



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

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# **Application Notes for Configuring Avaya Aura® Communication Manager R10.1, Avaya Aura® Session Manager R10.1, Experience Portal R8.1 and Avaya Session Border Controller for Enterprise R10.1 to support Vodafone Germany SIP Trunk Service- Issue 1.0**

## **Abstract**

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between the Vodafone Germany SIP Trunk Service and an Avaya SIP enabled Enterprise Solution. The Avaya solution consists of Avaya Aura® Communication Manager R10.1, Avaya Aura® Session Manager R10.1, Avaya Experience Portal R8.1 and Avaya Session Border Controller for Enterprise R10.1.

The Vodafone Germany SIP Platform provides PSTN access via a SIP trunk connected to the Vodafone Germany Voice over Internet Protocol (VoIP) network as an alternative to legacy analogue or digital trunks.

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.

Vodafone Germany is a member of the DevConnect Service Provider program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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# 1. Introduction

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between the Vodafone Germany SIP Trunk and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of the following: Avaya Aura® Communication Manager R10.1 (Communication Manager); Avaya Aura® Session Manager R10.1 (Session Manager), Avaya Experience Portal (Experience Portal) and Avaya Session Border Controller for Enterprise R10.1 (Avaya SBCE).

Customers using this Avaya SIP-enabled enterprise solution with the Vodafone Germany SIP Trunk Service are able to place and receive PSTN calls via a dedicated Internet connection and the SIP protocol. 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, Experience Portal and Avaya SBCE. The enterprise site was configured to connect to the Vodafone Germany SIP platform.

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

## 2.1. Interoperability Compliance Testing

The interoperability test included the following:

- Incoming calls to the enterprise site from PSTN phones using the Vodafone Germany SIP Trunk Service, calls made to SIP and H.323 telephones at the enterprise.
- Outgoing calls from the enterprise site completed via the Vodafone Germany SIP Trunk Service to PSTN destinations, calls made from SIP and H.323 telephones.
- Incoming and Outgoing PSTN calls to/from Avaya one-X® Communicator and Avaya Workplace for Windows soft phones.
- Calls using the G.711A codec.
- Fax calls to/from a group 3 fax machine to a PSTN-connected fax machine using G.711 pass-through fax transmissions.
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, conference, call forwarding, etc.
- Caller ID Presentation and Caller ID Restriction.
- Call coverage and call forwarding for endpoints at the enterprise site.
- Inbound caller interaction with Experience Portal applications, including prompting, caller DTMF input, wait treatment (e.g., announcements and/or music on hold).
- Experience Portal use of SIP REFER to redirect inbound calls, via the Avaya SBCE, to the appropriate Communication Manager agents and extensions.
- Call and two-way talk path establishment between callers and Communication Manager agents and extensions following redirection from Experience Portal.
- Routing inbound vector call to call center agent queues.
- Transmission and response of SIP OPTIONS messages sent by Vodafone Germany requiring Avaya response and sent by Avaya requiring Vodafone Germany response.

## 2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the Vodafone Germany SIP Trunking Service with the following observations:

- It was observed when performing Blind Transfer to PSTN numbers on inbound calls (i.e. PSTN (A) -> Avaya (B) -> Blind Transfer -> PSTN (C)) from Avaya SIP handsets, that Vodafone Germany was responding with a “403 Forbidden”. The reason Vodafone Germany was responding with “403 Forbidden” is that the Avaya SIP handsets populate the P-Asserted-Identity Header with the originating caller (A) CLID and also insert a Header called “Referred-By” when completing the transfer. Vodafone Germany do not recognize this header “Referred-By” and Vodafone Germany require the P-Asserted-Identity Header to be populated with the CLID of a known Vodafone Germany number (B) on their SIP platform. In order for Blind Transfers to PSTN to complete successfully, a SigMa script was created on the Avaya SBCE to remove the “Referred-By” Header and also to populate the P-Asserted-Identity Header with a known Vodafone Germany CLID number on the Vodafone Germany SIP platform. The details of the Sigma Script are outlined in **Section 8.6**.
- It was observed during testing that Experience Portal uses REFER to complete Blind and Consultative transfers to internal Contact Center/ACD applications, such as agent routing, which led to signalling issues and transfer failures between Avaya and the Vodafone Germany SIP trunk. In order to complete Blind and Consultative transfers successfully within Experience Portal, REFER Handling needs to be enabled on the Vodafone Germany Server Interworking profile (**Section 8.5.2**) on the Avaya SBCE. When the REFER message comes from an Avaya enterprise element such as Experience Portal, the Avaya SBCE translates that REFER into a reINVITE which will then be routed towards the trunk server (i.e., Vodafone Germany) based on the trunk server interworking profile configuration.
- It was observed during testing that Blind and Consultative transfers from Experience Portal to external PSTN phones were failing due to lack of transmission of media. This is due to the handling of the signalling within the Vodafone Germany SIP platform when executing the Consultative and Blind transfers from Experience Portal to the external PSTN. Therefore, Blind and Consultative transfers from Experience Portal to the PSTN are not currently supported on the Vodafone Germany SIP platform.
- T.38 fax is not supported by Vodafone Germany 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.

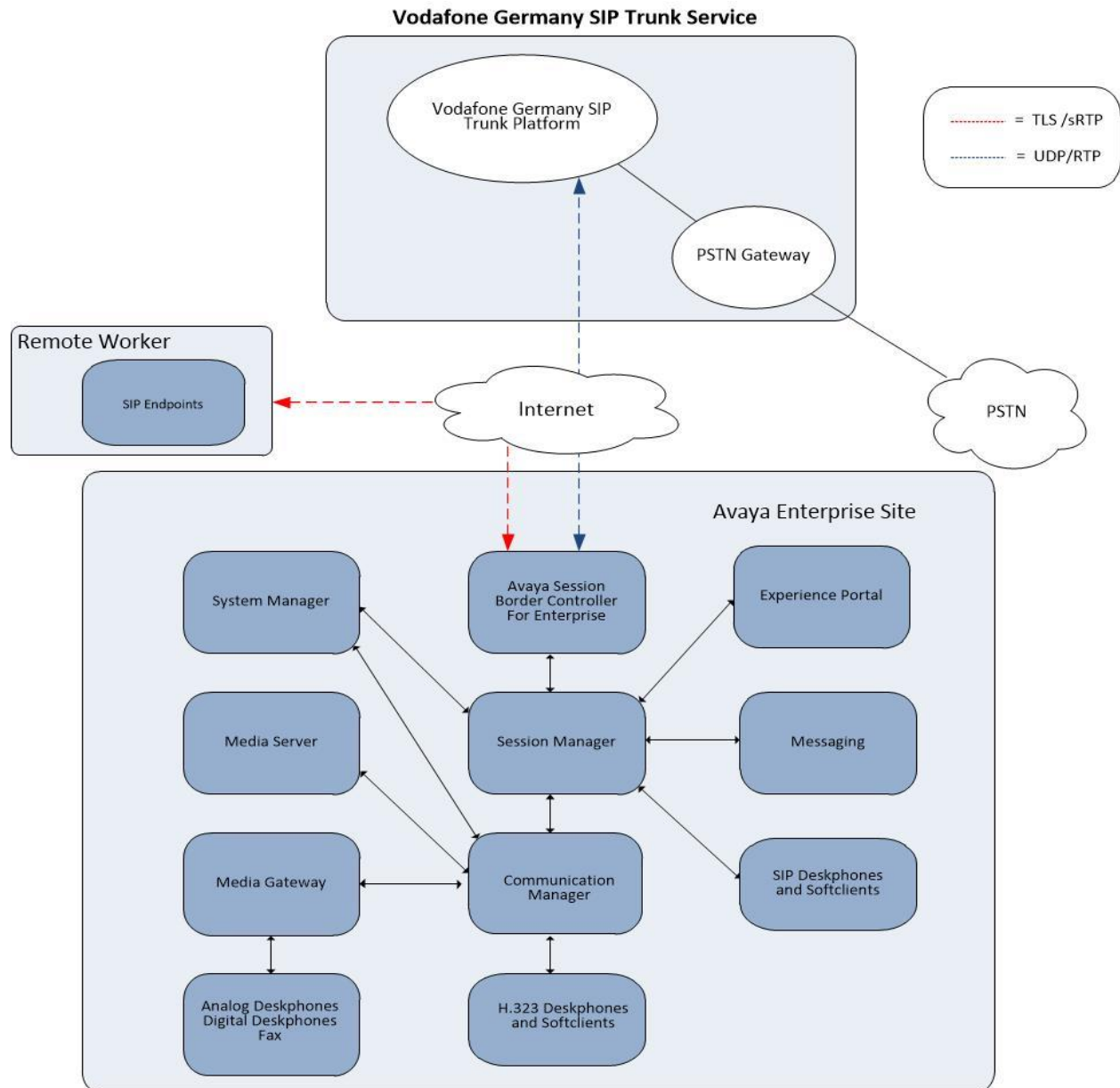
## 2.3. Support

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

For technical support on Vodafone Germany products please visit the website at [www.vodafone.de](http://www.vodafone.de) or contact an authorized Vodafone representative.

### 3. Reference Configuration

**Figure 1** illustrates the test configuration. The test configuration shows an Enterprise site connected to the Vodafone Germany SIP platform. Located at the Enterprise site is an Avaya SBCE, Session Manager, Experience Portal and Communication Manager. Endpoints are Avaya 96x1 series IP telephones (with SIP and H.323 firmware), Avaya J179 series IP telephone (with SIP 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 and Avaya Workplace for Windows running on laptop PCs



**Figure 1: Test Setup Vodafone Germany 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
<b>Avaya</b>	
Avaya Aura® System Manager	10.1.2.0 Build No. – 10.1.0.0.537353 Software Update Revision No: 10.1.2.0.0715474 – Feature Pack 2
Avaya Aura® Session Manager	10.1.2.0.1012016
Avaya Aura® Communication Manager	10.1 Feature Pack 2 - 27783
Avaya Session Border Controller for Enterprise	10.1.0.0-32-21432
Avaya Experience Portal	8.1.1
Avaya G430 Media Gateway	42.18.0
Avaya Aura® Media Server	10.1.0 SP2
Avaya Aura® Messaging	7.2 SP3
Avaya 96x1 IP Deskphone (H.323)	6.8.5
Avaya 9611 IP Deskphone (SIP)	7.1.15
Avaya 9608 IP Deskphone (SIP)	7.1.15
Avaya J179 IP Deskphone (SIP)	4.1.0
Avaya one-X® Communicator (H.323 & SIP)	6.2.14.17 -SP14-Patch 8
Avaya Workplace for Windows (SIP)	3.31.2
Avaya 1408 Digital Telephone	R48
Analogue Fax	N/A
<b>Vodafone Germany</b>	
A-SBC	N/A
Vodafone VoIP SoftSwitch	N/A



## 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 Vodafone Germany SIP Trunking Service. 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 Vodafone Germany 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 Vodafone Germany SIP Trunking Service and any other SIP trunks used.

<b>display system-parameters customer-options</b>		Page 2 of 12
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	4000	0
Maximum Concurrently Registered IP Stations:	2400	3
Maximum Administered Remote Office Trunks:	4000	0
Maximum Concurrently Registered Remote Office Stations:	2400	0
Maximum Concurrently Registered IP eCons:	68	0
Max Concur Registered Unauthenticated H.323 Stations:	100	0
Maximum Video Capable Stations:	2400	0
Maximum Video Capable IP Softphones:	2400	0
<b>Maximum Administered SIP Trunks:</b>	<b>4000</b>	<b>20</b>
Maximum Administered Ad-hoc Video Conferencing Ports:	4000	0
Maximum Number of DS1 Boards with Echo Cancellation:	80	0

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

display system-parameters customer-options		Page 5 of 12
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, **Session Manager** and **10.10.3.42** are the **Name** and **IP Address** for the Session Manager SIP interface. Also note the **procr** IP address 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	
AMS	10.10.3.45	
<b>Session_Manager</b>	<b>10.10.3.42</b>	
default	0.0.0.0	
<b>procr</b>	<b>10.10.3.44</b>	
procr6	::	

### 5.3. Administer IP Network Region

Use the **change ip-network-region n** command where **n** is the chosen value of the configuration for the SIP Trunk. 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 or the call is set up with initial IP-IP direct media, 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: 2
Location: Authoritative Domain: avaya.com
Name: Trunk 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** by typing **change ip-codec set n** where **n** is the chosen value of the configuration for the SIP Trunk. Enter the list of audio codec's eligible to be used in order of preference. For the interoperability test the codec supported by Vodafone Germany configured was **G.711A**.

In addition to the codec's, the **Media Encryption** is defined here. For the compliance test, a value of **srtp-aescm128-hmac80** was used.

change ip-codec-set 1 Page 1 of 2

IP MEDIA PARAMETERS

Codec Set: 1

Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)
1: G.711A	n	2	20

Media Encryption

1: srtp-aescm128-hmac80  
2: none

Encrypted SRTCP: enforce-unenc-srtcp

Vodafone Germany SIP Trunk supports G.711 pass-through for transmission of fax. Navigate to **Page 2** and define fax properties as follows:

- Set the **FAX - Mode** to **pass-through**.
- Leave **ECM** at default value of **y**.

change ip-codec-set 1 Page 2 of 2

IP MEDIA PARAMETERS

Allow Direct-IP Multimedia? n

	Mode	Redun- dancy	ECM: y	Packet Size (ms)
<b>FAX</b>	<b>pass-through</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 Signaling Groups

This signalling group (and trunk group) will be used for inbound and outbound PSTN calls to the Vodafone Germany0 SIP Trunking Service. Configure the **Signaling Group** using the **add signaling-group n** command as follows:

- Set **Group Type** to **sip**.
- Set **Transport Method** to **tls**.
- 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 interface (node name **Session\_Manager** as defined in the **IP Node Names** form shown in **Section 5.2**).
- Set **Near-end Listen Port** and **Far-end Listen Port** as required. The standard value for TLS is **5061**.
- 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 region **1**).
- Leave **Far-end Domain** blank to allow 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 **n**.
- 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: 2	Group Type: sip	
IMS Enabled? n	Transport Method: tls	
Q-SIP? n		
IP Video? n	Enforce SIPS URI for SRTP? n	
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		
Alert Incoming SIP Crisis Calls? n		
Near-end Node Name: procr	Far-end Node Name: Session_Manager	
Near-end Listen Port: 5061	Far-end Listen Port: 5061	
	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? n	Initial IP-IP Direct Media? n	
H.323 Station Outgoing Direct Media? n	Alternate Route Timer(sec): 6	

## 5.6. Administer SIP Trunk Groups

A trunk group is associated with the signalling group described in **Section 5.5**. Configure the trunk group using the **add trunk-group n** command, where **n** is an available trunk group for the SIP Trunk. 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	Night Service:	
Dial Access? n			
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 Vodafone Germany to prevent unnecessary SIP messages during call setup. During testing, a value of **900** was used that sets Min-SE and Session-Expires to 1800 in the SIP signalling.

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): 900			
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**.

add trunk-group 1		Page 3 of 21
TRUNK FEATURES		
ACA Assignment? n	Measured: none	Maintenance Tests? y
Suppress # Outpulsing? n	<b>Numbering Format: public</b>	
		UI Treatment: service-provider
		Replace Restricted Numbers? n
		Replace Unavailable Numbers? n
		<b>Hold/Unhold Notifications? n</b>
Modify Tandem Calling Number: no		
Show ANSWERED BY on Display? y		

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** as requested by Vodafone Germany.
- Set **Always Use re-INVITE for Display Updates** to **y**.
- Set the **Identity for Calling Party Display** to **P-Asserted-Identity**.

add trunk-group 1		Page 4 of 21
PROTOCOL VARIATIONS		
<b>Mark Users as Phone? y</b>		
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n		
<b>Send Transferring Party Information? n</b>		
<b>Network Call Redirection? n</b>		
<b>Send Diversion Header? y</b>		
<b>Support Request History? n</b>		
<b>Telephone Event Payload Type: 101</b>		
Convert 180 to 183 for Early Media? n		
<b>Always Use re-INVITE for Display Updates? y</b>		
<b>Identity for Calling Party Display: P-Asserted-Identity</b>		
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		
Request URI Contents: may-have-extra-digits		

## 5.7. Administer Calling Party Number Information

Use the **change public-unknown-numbering** command to configure Communication Manager to send the calling party number in the format required. These calling party numbers are sent in the SIP From, Contact and PAI headers as well as the Diversion header for forwarded calls. The numbers are displayed on display-equipped PSTN telephones with any reformatting performed in the network. The public numbering table is used for numbers in E.164 format.

change public-unknown-numbering 0					Page 1 of 2
NUMBERING - PUBLIC/UNKNOWN FORMAT					
Total					
Ext	Trk	CPN			
Len Code	Grp(s)	Prefix	Len		
4 6102	1	4940xxxxxxx81	13	Total Administered: 4	
4 6010	1	4940xxxxxxx82	13	Maximum Entries: 240	
4 6020	1	4940xxxxxxx83	13	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	4940xxxxxxx84	13		
					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 the Vodafone Germany SIP Trunking Service. The single digit **9** was used as the ARS access code providing a facility for telephone users to dial 9 to invoke ARS directly. Use the **change feature-access-codes** command to configure a digit as the **Auto Route Selection (ARS) - Access Code 1**.

<b>change feature-access-codes</b>		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	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 does not 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:	v	v	v	v	v	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 Vodafone Germany 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 Vodafone Germany SIP platform correlate to the internal extensions assigned within Communication Manager. The entries displayed below translate incoming DDI numbers **+4940xxxxxxx81**, **+4940xxxxxxx82**, **+4940xxxxxxx83** and **+4940xxxxxxx84** to a 4-digit extension by deleting all of the incoming digits and inserting an extension.

change inc-call-handling-trmt trunk-group 1					Page	1 of	3
INCOMING CALL HANDLING TREATMENT							
Service/ Feature	Number Len	Del	Insert Digits				
public-ntwrk	14		+4940xxxxxxx81	all	6102		
public-ntwrk	14		+4940xxxxxxx82	all	6010		
public-ntwrk	14		+4940xxxxxxx83	all	6020		
public-ntwrk	14		+4940xxxxxxx84	all	6104		

## 5.10. EC500 Configuration

When EC500 is enabled on a 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 Communication Manager station.

- The **Station Extension** field will automatically populate with station extension.
- For **Application** enter **EC500**.
- Enter a **Dial Prefix** if required by the routing configuration, none was required during testing.
- For the **Phone Number** enter the phone that will also be called (e.g., **0035389434xxxx**).
- Set the **Trunk Selection** to **ars** so that the ARS table 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	ars	1	

**Note:** The phone number shown is for a mobile phone in the Avaya Lab. To use facilities for calls coming in from EC500 mobile phones, the calling party number received in Communication Manager must exactly match the number specified in the above table.

Save Communication Manager configuration by entering **save translation**.

## 6. Configuring Avaya Aura® Session Manager

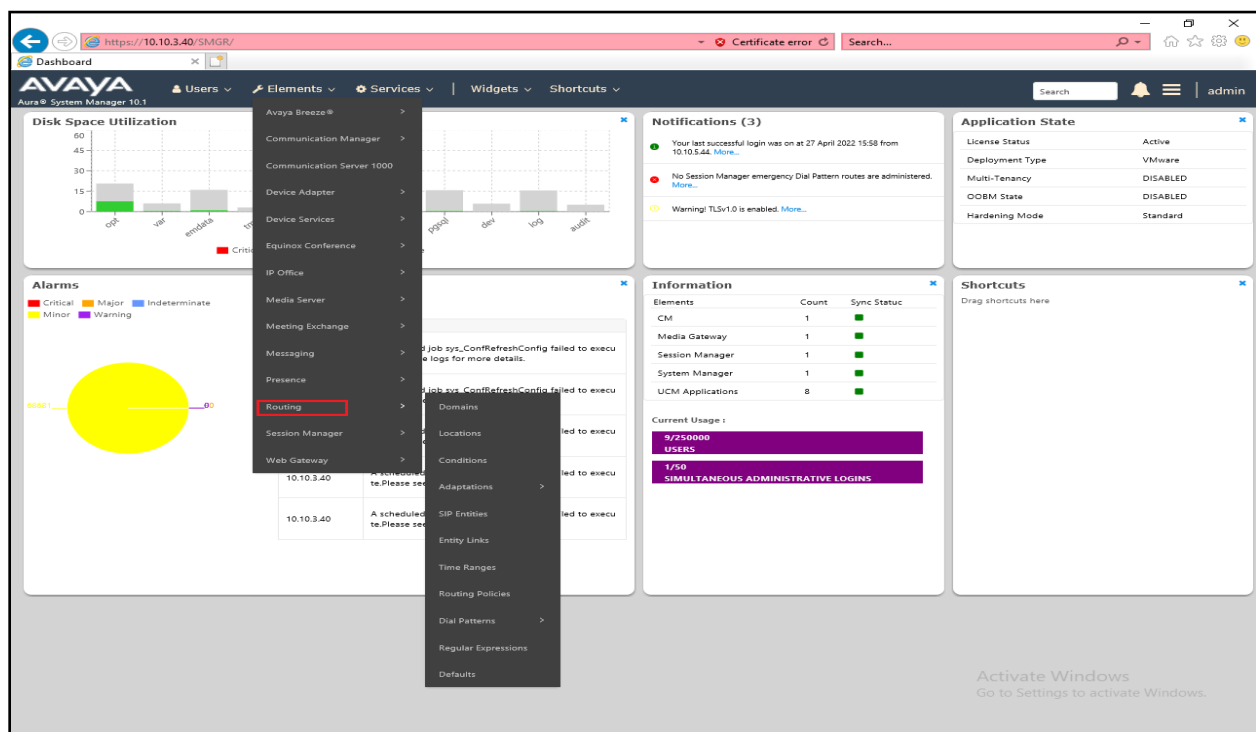
This section provides the procedures for configuring Session Manager. Session Manager is configured via System Manager. The procedures include the following areas:

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

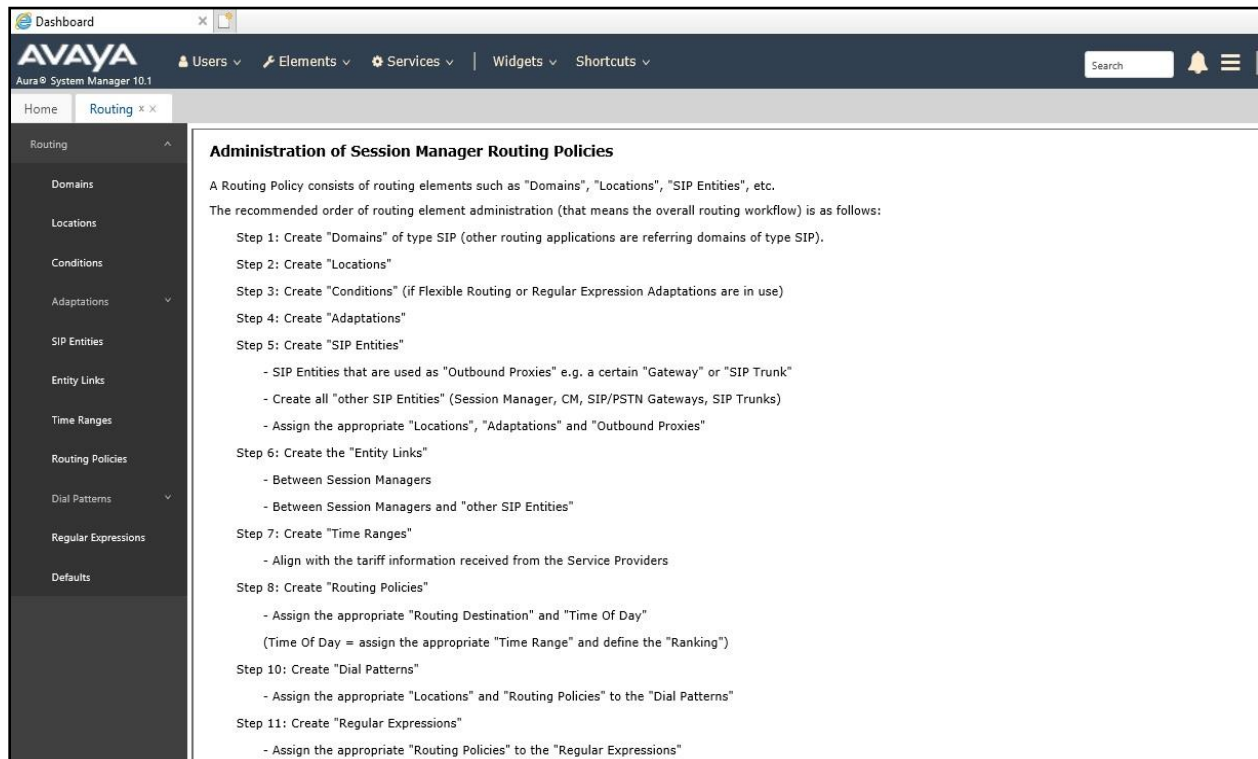
It may not be necessary to create all the items above when creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, locations, SIP entities, and Session 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 and entering **http://<FQDN>/SMGR**, where **<FQDN>** is the fully qualified domain name of System Manager. Log in using appropriate credentials (not shown) and the Dashboard 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 following screenshot shows the location details named **Session Manager**. This location is assigned to the SIP Entity called Session Manager in **Section 6.5.1**.

The screenshot displays the 'Location Details' configuration window. At the top right are 'Commit' and 'Cancel' buttons. The 'General' section contains a required 'Name' field with the value 'Session Manager' and an empty 'Notes' field. The 'Dial Plan Transparency in Survivable Mode' section has an 'Enabled' checkbox that is unchecked, and empty fields for 'Listed Directory Number' and 'Associated CM SIP Entity'. The 'Overall Managed Bandwidth' section shows 'Managed Bandwidth Units' set to 'Kbit/sec', with empty fields for 'Total Bandwidth' and 'Multimedia Bandwidth'. The 'Audio Calls Can Take Multimedia Bandwidth' checkbox is checked. The 'Per-Call Bandwidth Parameters' section includes four fields: 'Maximum Multimedia Bandwidth (Intra-Location)' and 'Maximum Multimedia Bandwidth (Inter-Location)' both set to '2000 Kbit/Sec', '\* Minimum Multimedia Bandwidth' set to '64 Kbit/Sec', and '\* Default Audio Bandwidth' set to '80 Kbit/sec' with a dropdown arrow.

Location Details		Commit	Cancel
<b>General</b>			
* Name:	Session Manager		
Notes:			
<b>Dial Plan Transparency in Survivable Mode</b>			
Enabled:	<input type="checkbox"/>		
Listed Directory Number:			
Associated CM SIP Entity:			
<b>Overall Managed Bandwidth</b>			
Managed Bandwidth Units:	Kbit/sec		
Total Bandwidth:			
Multimedia Bandwidth:			
Audio Calls Can Take Multimedia Bandwidth:	<input checked="" type="checkbox"/>		
<b>Per-Call Bandwidth Parameters</b>			
Maximum Multimedia Bandwidth (Intra-Location):	2000	Kbit/Sec	
Maximum Multimedia Bandwidth (Inter-Location):	2000	Kbit/Sec	
* Minimum Multimedia Bandwidth:	64	Kbit/Sec	
* Default Audio Bandwidth:	80	Kbit/sec	

The location pattern is a way of using subnets to further refine the location information, this may be useful for endpoints that could be logged in from different subnets. If required, scroll to the bottom of the page and under **Location Pattern**, click **Add**, then enter an **IP Address Pattern** in the resulting new row, \* is used to specify any number of allowed characters at the end of the string.

The screenshot shows a web interface titled "Location Pattern". At the top, there are "Add" and "Remove" buttons. Below them, it says "0 Items" with a refresh icon and a "Filter: Enable" link. A table with one row is visible, with a checkbox, the text "IP Address Pattern", and a "Notes" column. At the bottom right, there are "Commit" and "Cancel" buttons.

The following screenshot shows the location details named **Communication Manager**. This location is assigned to the SIP Entity called Communication Manager in **Section 6.5.2**.

The screenshot shows a web interface titled "Location Details" with "Commit" and "Cancel" buttons at the top right. It has three sections: "General" with fields for "Name" (set to "Communication Manager") and "Notes"; "Dial Plan Transparency in Survivable Mode" with an "Enabled" checkbox (unchecked), "Listed Directory Number" field, and "Associated CM SIP Entity" field; and "Overall Managed Bandwidth" with "Managed Bandwidth Units" set to "Kbit/sec", "Total Bandwidth" field, "Multimedia Bandwidth" field, and a checked checkbox for "Audio Calls Can Take Multimedia Bandwidth".

The following screenshot shows the location details named **Experience Portal**. This location is assigned to the SIP Entity called Experience Portal in **Section 6.5.3**.

The screenshot shows a web form titled "Location Details" with "Commit" and "Cancel" buttons in the top right. The form is divided into three sections: "General", "Dial Plan Transparency in Survivable Mode", and "Overall Managed Bandwidth".

- General**: Contains a required field "Name" with the value "Experience Portal" and an empty "Notes" field.
- Dial Plan Transparency in Survivable Mode**: Contains an "Enabled" checkbox (unchecked), a "Listed Directory Number" field, and an "Associated CM SIP Entity" field.
- Overall Managed Bandwidth**: Contains a "Managed Bandwidth Units" dropdown menu set to "Kbit/sec", "Total Bandwidth" and "Multimedia Bandwidth" input fields, and an "Audio Calls Can Take Multimedia Bandwidth" checkbox (checked).

The following screenshot shows the location details named **Avaya SBCE**. This location is assigned to the SIP Entity called Avaya SBCE in **Section 6.5.4**.

The screenshot shows a web form titled "Location Details" with "Commit" and "Cancel" buttons in the top right. The form is divided into three sections: "General", "Dial Plan Transparency in Survivable Mode", and "Overall Managed Bandwidth".

- General**: Contains a required field "Name" with the value "Avaya SBCE" and an empty "Notes" field.
- Dial Plan Transparency in Survivable Mode**: Contains an "Enabled" checkbox (unchecked), a "Listed Directory Number" field, and an "Associated CM SIP Entity" field.
- Overall Managed Bandwidth**: Contains a "Managed Bandwidth Units" dropdown menu set to "Kbit/sec", "Total Bandwidth" and "Multimedia Bandwidth" input fields, and an "Audio Calls Can Take Multimedia Bandwidth" checkbox (checked).



## 6.4. Administer Adaptations

Session Manager Adaptations can be used to alter parameters in the SIP message headers. An Adaptation was used during testing to remove Avaya proprietary headers from messages sent and remove headers from messages received from Vodafone Germany. 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 R10.1 incorporates the ability to use Adaptation modules to remove specific SIP headers that are Avaya proprietary unnecessary for non-Avaya elements.

For the compliance test, an Adaptation named “**VFDE**” 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 and 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 **VFDE**.
- **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**.

Help ?

## Adaptation Details

### General

\* Adaptation Name:

Notes:

\* Module Name:

Type:

State:

Module Parameter Type:

Add
Remove

	Name	Value
<input type="checkbox"/>	eRHdrs	"P-AV-Message-Id, P-Charging-Vector, P-Location, Endpoint-View, P-Conference, Alert-Info, Accept-Language, Av-Global-
<input type="checkbox"/>	fromto	true
<input type="checkbox"/>	MIME	no

Select : All, None

Egress URI Parameters:

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

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
<input type="checkbox"/>	*00	*2	*15		*2	+	both		

Select : All, None

This will ensure any outgoing numbers matching 00 will be deleted and have + inserted being converted to E.164 format before being forwarded to the Avaya SBCE.

## 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, **Voice Portal** for an Experience Portal SIP Entity and **SIP Trunk** for the Avaya SBCE SIP Entities.
- 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 four SIP Entities.

- Session Manager SIP Entity.
- Communication Manager SIP Entity.
- Experience Portal 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 for Session Manager in **Section 6.3** and the **Time Zone** to the appropriate time zone.

#### SIP Entity Details

CommitCancel

##### General

\* Name:Session Manager

\* IP Address:10.10.3.42

SIP FQDN:

Type:Session Manager

Notes:

Location:Session Manager

Outbound Proxy:

Time Zone:Europe/Dublin

Minimum TLS Version:Use Global Setting

Credential name:

##### Monitoring

SIP Link Monitoring:Use Session Manager Configuration

CRLF Keep Alive Monitoring:Use Session Manager Configuration

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.

#### Port

TCP Failover port:

TLS Failover port:

AddRemove

3 Items

Filter: Enable

<input type="checkbox"/>	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	

Select : All, None

### 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. This SIP Entity is used for the SIP Trunk. The **FQDN or IP Address** field is set to the IP address of the interface on Communication Manager that will be providing SIP signalling. Set the **Location** to that defined for Communication Manager in **Section 6.3** and the **Time Zone** to the appropriate time zone.

#### SIP Entity Details

CommitCancel

##### General

\* Name: Communication Manager

\* FQDN or IP Address: 10.10.3.44

Type: CM

Notes:

Adaptation:

Location: Communication Manager

Time Zone: Europe/Dublin

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

Minimum TLS Version: Use Global Setting

Credential name:

Securable:

Call Detail Recording: none

##### Loop Detection

Loop Detection Mode: On

Loop Count Threshold: 5

Loop Detection Interval (in msec): 200

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 Experience Portal SIP Entity

The following screen shows the SIP entity for Experience Portal. The **FQDN or IP Address** field is set to the IP address of the Experience Portal. Set the **Location** to that defined for Experience Portal in **Section 6.3** and the **Time Zone** to the appropriate time zone.

#### SIP Entity Details

CommitCancel

##### General

\* Name: Experience\_Portal

\* FQDN or IP Address: 10.10.3.50

Type: Voice Portal

Notes:

Adaptation:

Location: Experience Portal

Time Zone: Europe/Dublin

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

Minimum TLS Version: Use Global Setting

Credential name:

Securable: ☐

Call Detail Recording: none

##### Loop Detection

Loop Detection Mode: On

Loop Count Threshold: 5

Loop Detection Interval (in msec): 200

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

The following screen shows the SIP Entity for the Avaya SBCE used for PSTN destinations. The **FQDN or IP Address** field is set to the IP address of the Avaya SBCE private network interface (See **Section 8.4.1**). Set the **Adaptation** to that defined in **Section 6.4**, the **Location** to that defined for Avaya SBCE in **Section 6.3** and the **Time Zone** to the appropriate time zone.

### SIP Entity Details

CommitCancel

#### General

\* Name: Avaya SBCE

\* FQDN or IP Address: 10.10.3.35

Type: SIP Trunk

Notes:

Adaptation: VFDE

Location: Session Manager

Time Zone: Europe/Dublin

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

Minimum TLS Version: Use Global Setting

Credential name:

Securable: ☐

Call Detail Recording: egress

#### Loop Detection

Loop Detection Mode: On

Loop Count Threshold: 5

Loop Detection Interval (in msec): 200

## 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.

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	DNS Override	Connection Policy	Deny New Service	Notes
<input type="checkbox"/>	<a href="#">Aura_Messaging</a>	Session Manager	TLS	5061	Aura_Messaging	5061	<input type="checkbox"/>	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	<a href="#">Avaya_SBCE</a>	Session Manager	TLS	5061	Avaya_SBCE	5061	<input type="checkbox"/>	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	<a href="#">Communication_Manager</a>	Session Manager	TLS	5061	Communication_Manager	5061	<input type="checkbox"/>	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	<a href="#">Experience_Portal</a>	Session Manager	TLS	5061	Experience_Portal	5061	<input type="checkbox"/>	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 inbound calls from the Vodafone Germany SIP Trunk to Communication Manager.

**Routing Policy Details** Commit Cancel Help ?

**General**

\* Name:

Disabled: ☐

\* Retries:

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 outbound calls from Communication Manager via Avaya SBCE to the Vodafone Germany SIP trunk.

Routing Policy Details
Commit
Cancel

Help ?

General

\* Name: to\_Avaya\_SBCE

Disabled: ☐

\* Retries: 0

Notes: Outbound calls to SP via ASBCE.

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Avaya_SBCE	10.10.3.35	SIP Trunk	

Time of Day

Add Remove View Gaps/Overlaps

1 Item Filter: Enable

<input type="checkbox"/> 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 calls inbound from the SIP Trunk to Experience Portal.

Routing Policy Details
Commit
Cancel

Help ?

General

\* Name: to\_Experience\_Portal

Disabled: ☐

\* Retries: 0

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Experience_Portal	10.10.3.50	Voice Portal	

Time of Day

Add Remove View Gaps/Overlaps

1 Item Filter: Enable

<input type="checkbox"/> 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 outbound calls to the Vodafone Germany SIP Trunk.

Dial Pattern Details

CommitCancel

General

\* Pattern: 00353

\* Min: 5

\* Max: 16

Emergency Call: ☐

SIP Domain: avaya.com

Notes:

Originating Locations and Routing Policies

AddRemove

1 Item

<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"/>	Communication Manager		to_Avaya_SBCE	0	<input type="checkbox"/>	Avaya_SBCE	Outbound calls to SP via ASBCE.

Select : All, None

Denied Originating Locations

AddRemove

0 Items

<input type="checkbox"/>	Originating Location	Notes
--------------------------	----------------------	-------

CommitCancel

The following screen shows the dial pattern configured for inbound calls to Communication Manager.

**Dial Pattern Details**
Commit Cancel
Help ?

**General**

\* Pattern:

+49

\* Min:

6

\* Max:

16

Emergency Call:

☐

SIP Domain:

avaya.com

Notes:

**Originating Locations and Routing Policies**

Add Remove

1 Item

	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Session Manager		to_Communication_Manager	0	<input type="checkbox"/>	Communication Manager	Inbound calls to CM.

Select : All, None

**Denied Originating Locations**

Add Remove

0 Items

Originating Location	Notes

Commit Cancel

The following screen shows the dial pattern configured for inbound calls to Experience Portal.

Help ?

## Dial Pattern Details

Commit Cancel

### General

\* Pattern:

\* Min:

\* Max:

Emergency Call: ☐

SIP Domain:

Notes:

### Originating Locations and Routing Policies

Add Remove
Filter: Enable

1 Item

<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"/>	Session Manager		to_Experience_Portal	0	<input type="checkbox"/>	Experience Portal	Inbound calls to Experience Portal

Select : All, None

### Denied Originating Locations

Add Remove

0 Items

<input type="checkbox"/>	Originating Location	Notes
--------------------------	----------------------	-------

Commit Cancel

The following screen shows the dial pattern configured for outbound calls from Experience Portal to the Vodafone Germanys SIP Trunk.

Dial Pattern Details

CommitCancel

Help ?

General

\* Pattern:0035391

\* Min:7

\* Max:16

Emergency Call:☐

SIP Domain:avaya.com

Notes:

Originating Locations and Routing Policies

AddRemove

1 Item

<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"/>	Experience Portal		to_Avaya_SBCE	0	<input type="checkbox"/>	Avaya SBCE	Outbound calls to SP via ASBCE.

Select : All, None

Denied Originating Locations

AddRemove

0 Items

<input type="checkbox"/>	Originating Location	Notes
--------------------------	----------------------	-------

CommitCancel

## 7. Configure Avaya Experience Portal

These Application Notes assume that the necessary Experience Portal licenses have been installed and basic Experience Portal administration has already been performed. Consult [13] in the **References** section for further details if necessary.

### 7.1. Background

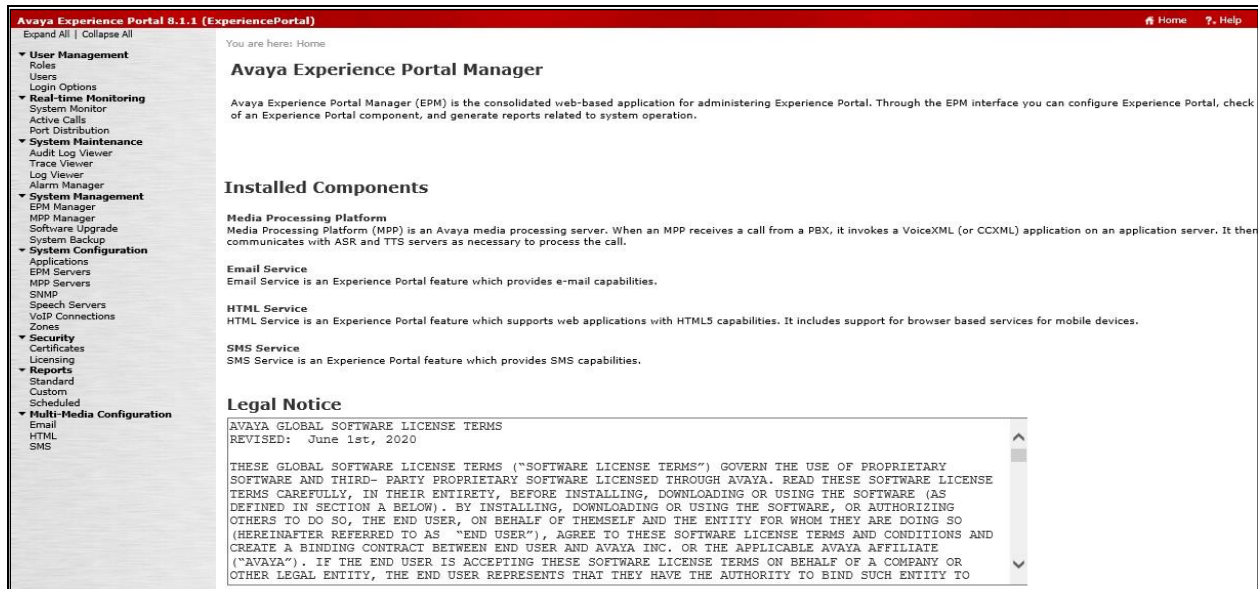
Experience Portal consists of one or more Media Processing Platform (MPP) servers and an Experience Portal Manager (EPM) server. A single “server configuration” was used in the reference configuration. This consisted of a single MPP and EPM, running on a VMware environment, including an Apache Tomcat Application Server (hosting the Voice XML (VXML) and/or Call Control XML (CCXML) application scripts), that provide the directives to Experience Portal for handling the inbound calls.

References to the Voice XML and/or Call Control XML applications are administered on Experience Portal, along with one or more called numbers for each application reference. When an inbound call arrives at Experience Portal, the called party DDI number is matched against those administered called numbers. If a match is found, then the corresponding application is accessed to handle the call. If no match is found, Experience Portal informs the caller that the call cannot be handled, and disconnects the call sample configuration described in these Application Notes. A simple VXML test application was used to exercise various SIP call flow scenarios with the Vodafone Germany SIP Trunk service. In production, enterprises can develop their own VXML and/or CCXML applications to meet specific customer self-service needs, or consult Avaya Professional Services and/or authorized Avaya Business Partners. The development and deployment of VXML and CCXML applications is beyond the scope of these Application Notes.

## 7.2. Logging In and Licensing

This section describes the steps on Experience Portal for administering a SIP connection to the Session Manager.

**Step 1** - Launch a web browser, enter `http://<IP address of the Avaya EPM server>/` in the URL, log in with the appropriate credentials and the following screen is displayed.



**Step 2** - In the left pane, navigate to **Security**→**Licensing**. On the **Licensing** page, verify that Experience Portal is properly licensed. If required licenses are not enabled, contact an authorized Avaya representative to obtain the licenses.



You are here: [Home](#) > Security > Licensing

## Licensing

This page displays the Experience Portal license information that is currently in effect. Experience Portal uses Avaya License Manager (WebLM).

### License Server Information ▾

License Server URL:	https://10.10.9.19:52233/WebLM/LicenseServer
Last Updated:	01-Mar-2019 12:22:58 GMT
Last Successful Poll:	27-Sep-2022 11:36:44 IST

### Licensed Products ▾

#### Experience Portal

Announcement Ports:	100
ASR Connections:	100
Email Units:	10
Enable Media Encryption:	100
Enhanced Call Classification:	100
Google ASR Connections:	10
HTML Units:	100
SIP Signaling Connections:	1,000
SMS Units:	10
Telephony Ports:	1,000
TTS Connections:	100
Video Server Connections:	100
Zones:	10

Last Successful Poll:	27-Sep-2022 11:36:44 IST
Last Changed:	10-Mar-2020 10:54:53 GMT

[Allocations](#)

[Help](#)

## 7.3. VoIP Connection

This section defines a SIP trunk between Experience Portal and Session Manager.

**Step 1** - In the left pane, navigate to **System Configuration**→**VoIP Connections**. On the **VoIP Connections** page, select the **SIP** tab and click **Add** to add a SIP trunk. **Note** – Only one SIP trunk can be active at any given time on Experience Portal.

You are here: [Home](#) > [System Configuration](#) > [VoIP Connections](#)

### VoIP Connections

This page displays a list of Voice over Internet Protocol (VoIP) servers that Experience Portal communicates with. You can configure multiple SIP connections, but only one SIP connection can be enabled at any one given time.

H.323

SIP

<input type="checkbox"/>	Name	Enable	Proxy Transport	Proxy/DNS Server Address	Proxy Server Port	Listener Port	SIP Domain	Maximum Simultaneous Calls
<input type="checkbox"/>	SM	Yes	TLS	10.10.3.42	5061	5061	avaya.com	10

Add

Delete

Help

**Step 2** - Configure a SIP connection as follows:

- **Name** – Set to a descriptive name (e.g., **SM**).
- **Enable** – Set to **Yes**.
- **Proxy Server Transport** – Set to **TLS**.
- Select **Proxy Servers**, and enter:
  - **Proxy Server Address** = **10.10.3.42** (the IP address of the Session Manager signaling interface defined in **Section 6.5.1**).
  - **Port** = **5061**
  - **Priority** = **0** (default)
  - **Weight** = **0** (default)
- **Listener Port** – Set to **5061**.
- **SIP Domain** – Set to **avaya.com** (see **Section 6.2**).
- **Consultative Transfer** – Select **INVITE with REPLACES**.
- **SIP Reject Response Code** – Select **ASM (503)**.
- **Maximum Simultaneous Calls** – Set to a number in accordance with licensed capacity. In the reference configuration a value of **10** was used.
- Select **All Calls can be either inbound or outbound**.
- **SRTP Enable** = **Yes**
- **Encryption Algorithm** = **AES\_CM\_128**
- **Authentication Algorithm** = **HMAC\_SHA1\_80**
- **RTCP Encryption Enabled** = **No**
- **RTP Authentication Enabled** = **Yes**
- Click **Add**.
- Use default values for all other fields and click **Save**.

You are here: [Home](#) > [System Configuration](#) > [VoIP Connections](#) > [Change SIP Connection](#)

## Change SIP Connection

Use this page to change the configuration of a SIP connection.

Name: SM

Enable: ☒ Yes ☐ No

Proxy Transport: TLS ▼

☒ Proxy Servers ☐ DNS SRV Domain

Address	Port	Priority	Weight	
10.10.3.42	5061	0	0	Remove

### Additional Proxy Server

Listener Port: 5061

SIP Domain: avaya.com

P-Asserted-Identity:

Maximum Redirection Attempts: 2

Consultative Transfer: ☒ INVITE with REPLACES ☐ REFER

SIP Reject Response Code: ☒ ASM (503) ☐ SES (480) ☐ Custom 503

### SIP Timers

T1: 250 milliseconds

T2: 2000 milliseconds

B and F: 4000 milliseconds

### Call Capacity

Maximum Simultaneous Calls: 10

☒ All Calls can be either inbound or outbound  
☐ Configure number of inbound and outbound calls allowed

### SRTP

Enable: ☒ Yes ☐ No

Encryption Algorithm: ☒ AES\_CM\_128 ☐ NONE

Authentication Algorithm: ☒ HMAC\_SHA1\_80 ☐ HMAC\_SHA1\_32

RTCP Encryption Enabled: ☐ Yes ☒ No

RTP Authentication Enabled: ☒ Yes ☐ No

Add

### Configured SRTP List

SRTP-Yes,AES\_CM\_128,HMAC\_SHA1\_80,RTCP Encryption-No,RTP Authentication-Yes

Remove

Save

Apply

Cancel

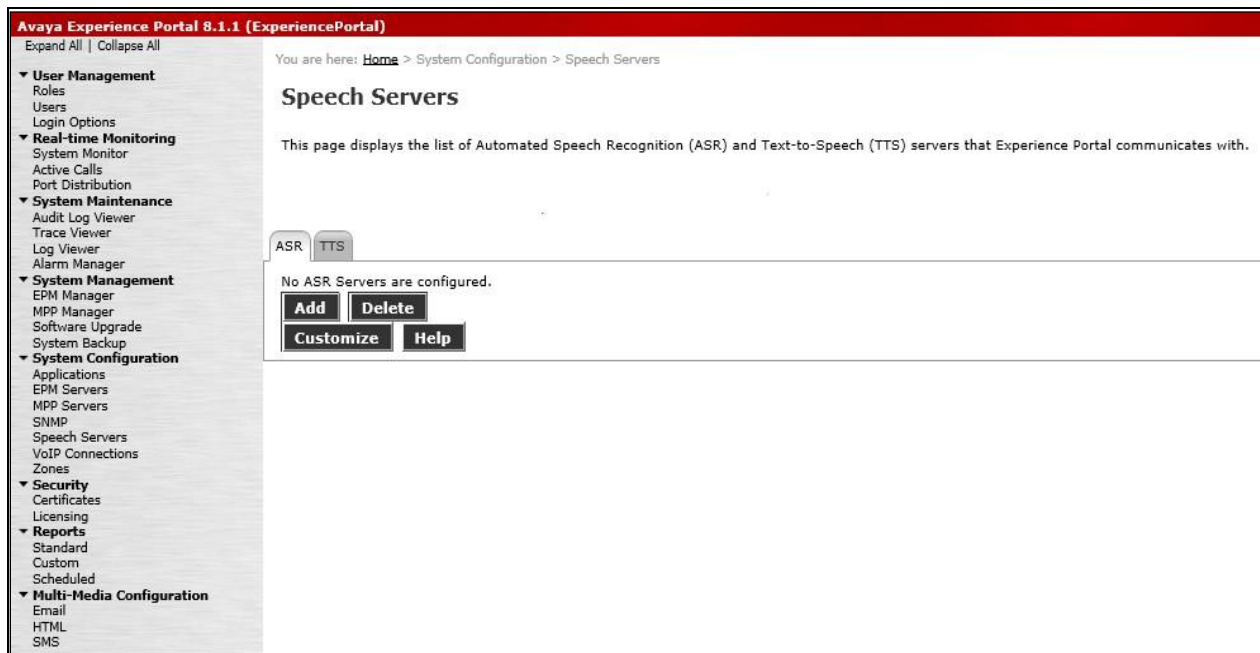
Help

## 7.4. Speech Servers

Avaya Experience Portal system integrates with two types of third-party speech servers:

- Automatic Speech Recognition (ASR): This technology enables an interactive voice response (IVR) system to collect verbal responses from callers.
- Text-to-Speech (TTS): This technology enables an IVR system to render text content into synthesized speech output according to algorithms within the TTS software.

No speech servers were required as part of the test configuration. The installation and administration of the ASR and TTS Speech Servers are beyond the scope of this document



## 7.5. Application References

This section describes the steps for administering a reference to the VXML and/or CCXML applications residing on the application server. In the sample configuration, the applications were co-resident on one Experience Portal server, with IP Address 10.10.3.50.

**Step 1** - In the left pane, navigate to **System Configuration→Applications**. On the **Applications** page (not shown), click **Add** to add an application and configure as follows:

- **Name** – Set to a descriptive name (e.g., **Test\_App**).
- **Enable** – Set to **Yes**. This field determines which application(s) will be executed based on their defined criteria.
- **Type** – Select **VoiceXML**, **CCXML**, or **CCXML/VoiceXML** according to the application type. CCXML was used in the test configuration.
- **VoiceXML** and/or **CCXML URL** – Enter the necessary URL(s) to access the VXML and/or CCXML application(s) on the application server. In the sample screen below, the Experience Portal test application on a single server is referenced. CCXML was used in the test configuration.
- **Speech Servers ASR** and **TTS** – Select the appropriate ASR and/or TTS servers as necessary.
- **Application Launch** – Set to **Inbound**.
- **Called Number** – Enter the number to match against an inbound SIP INVITE message, and click **Add**.

You are here: [Home](#) > [System Configuration](#) > [Applications](#) > [Change Application](#)

## Change Application

Use this page to change the configuration of an application.

Name: Test\_App  
Enable: ☒ Yes ☐ No  
Type: CCXML  
Reserved SIP Calls: ☒ None ☐ Minimum ☐ Maximum  
Requested:

### URI

☒ Single ☐ Fail Over ☐ Load Balance

CCXML URL:

**Verify**

Mutual Certificate Authentication: ☐ Yes ☒ No

Basic Authentication: ☐ Yes ☒ No

### ASR Speech Servers

Engine Types	Selected Engine Types
ASR: <input type="text" value="&lt;None&gt;"/>	<input type="text" value="&lt;None&gt;"/>

### TTS Speech Servers

TTS:

### Application Launch

☒ Inbound ☐ Inbound Default ☐ Outbound

☒ Number ☐ Number Range ☐ URI

Called Number:  **Add**

8000	<b>Remove</b>
49000000000087	

### Speech Parameters

### Reporting Parameters

### Advanced Parameters

**Save** **Apply** **Cancel** **Help**

## 7.6. MPP Servers and VoIP Settings

This section illustrates the procedure for viewing or changing the MPP Settings. In the sample configuration, the MPP Server is co-resident on a single server with the Experience Portal Management server (EPM).

**Step 1** - In the left pane, navigate to **System Configuration**→**MPP Servers** and the following screen is displayed. Click **Add**.

Avaya Experience Portal 8.1.1 (ExperiencePortal)

Expand All | Collapse All

You are here: [Home](#) > System Configuration > MPP Servers

### MPP Servers

This page displays the list of Media Processing Platform (MPP) servers in the Experience Portal system. When an MPP receives a call from a PBX, server and communicates with ASR and TTS servers as necessary to process the call.

Name	Host Address	Network Address (VoIP)	Network Address (MRCP)	Network Address (AppSvr)	Maximum Simultaneous Calls	Trace Level
<input type="checkbox"/> mpp1	10.10.3.50	<Default>	<Default>	<Default>	1	Use MPP Settings

**Add** **Delete**

**MPP Settings** **Browser Settings** **Video Settings** **VoIP Settings** **Help**

**Step 2** - Enter any descriptive name in the **Name** field (e.g., **mpp1**) and the IP address of the MPP server in the **Host Address** field and click **Continue** (not shown).

**Step 3** - The certificate page will open. Check the **Trust this certificate** box (not shown). Once complete, click **Save**.

You are here: [Home](#) > System Configuration > [MPP Servers](#) > Change MPP Server

## Change MPP Server

Use this page to change the configuration of an MPP. Take care when changing the MPP Trace Logging Thresholds. Do not set Trace Levels to Finest if your Experience Portal system has heavy call traffic. The system might experience performance issues if Trace Levels are set to Finest. Set Trace Levels to Finest only when you are troubleshooting the system.

Name: mpp1  
 Host Address: 10.10.3.50  
 Network Address (VoIP): <Default>  
 Network Address (MRCP): <Default>  
 Network Address (AppSvr): <Default>  
 Maximum Simultaneous Calls: 10  
 Restart Automatically: ☐ Yes ☒ No

### MPP Certificate

```

Owner: CN=ep7cmn.avaya.com,O=Avaya,OU=EPM
Issuer: CN=ep7cmn.avaya.com,O=Avaya,OU=EPM
Serial Number: 952c116c191b7519
Signature Algorithm: SHA256withRSA
Valid from: 28 February 2019 12:17:17 GMT until: 28 February 2029 12:17:17 GMT
Certificate Fingerprints
MD5: 8b:17:0c:92:43:ef:64:3d:86:b2:60:6a:bb:f5:09:69
SHA: 9a:90:a4:2c:48:21:46:ac:e4:18:c0:35:b0:e6:c1:43:9c:9b:di:be
SHA-256: 09:cb:da:73:0d:e6:ae:02:95:80:eb:92:56:0c:15:17:b2:f6:9e:f6:f9:2e:90:63:8e:06:be:98:96:cc:6a:26
Subject Alternative Names
DNS Name: ep7cmn
DNS Name: ep7cmn.avaya.com
IP Address: 10.10.3.50
  
```

Categories and Trace Levels ▶

**Step 4** - Click **VoIP Settings** tab on the screen displayed in **Step 1**, and the following screen is displayed.

- In the Port Ranges section, default ports were used.

You are here: [Home](#) > System Configuration > [MPP Servers](#) > VoIP Settings

## VoIP Settings

Voice over Internet Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols such as H.323 and Real-time Transfer Protocol (RTP). Use this page to configure parameters that affect how voice data is transferred through the network. Note that if you make any changes to this page, you must restart all MPPs.

### Port Ranges

	Low	High
UDP:	11000	30999
TCP:	31000	33499
MRCP:	34000	36499
H.323 Station:	37000	39499

### RTCP Monitor Settings

Host Address:   
 Port:

### VoIP Audio Formats

MPP Native Format:

In the **Codecs** section set:

- Set **Packet Time** to **20**.
- Verify the **G711alaw**, **G729** and **G711ulaw** codecs are enabled.
- Set **G729 Discontinuous Transmission** to **No (G.729A)**.
- Set the **Offer Order** to the preferred codec.
- Use default values for all other fields.

**Step 5** - Click on **Save**.



Codecs ▾

Offer

Enable	Codec	Order
<input checked="" type="checkbox"/>	G711aLaw	1
<input checked="" type="checkbox"/>	G729	2
<input checked="" type="checkbox"/>	G711uLaw	3

Packet Time:  milliseconds

G729 Discontinuous Transmission: ☐ Yes ☒ No

Answer

Enable	Codec	Order
<input checked="" type="checkbox"/>	G711uLaw	1
<input checked="" type="checkbox"/>	G711aLaw	1
<input checked="" type="checkbox"/>	G729	1

G729 Discontinuous Transmission: ☐ Yes ☐ No ☒ Either

G729 Reduced Complexity Encoder: ☒ Yes ☐ No

QoS Parameters ▸  
Out of Service Threshold (% of VoIP Resources) ▸  
Call Progress ▸  
Miscellaneous ▸

Save Apply Cancel Help

After saving the configuration changes, restart the MPP server for the change to take effect. As shown below, the MPP may be restarted using the **Restart** button available via the Experience Portal GUI at **System Management → MPP Manager**. Note that the **State** column shows when the MPP is running after the restart.

You are here: [Home](#) > System Management > MPP Manager

## MPP Manager (27-Sep-2022 12:11:58 IST)

This page displays the current state of each MPP in the Experience Portal system. To enable the state and mode commands, select one or more MPPs. must also be stopped.

Last Poll: 27-Sep-2022 12:11:54 IST

<input checked="" type="checkbox"/>	Server Name	Mode	State	Config	Auto Restart	Restart Schedule		Active Calls	
						Today	Recurring	In	Out
<input checked="" type="checkbox"/>	mpp1	Online	Running	OK	Yes	No	None	0	0

State Commands

Start Stop Restart Reboot Halt Cancel

Mode Commands

Offline Test Online

Restart/Reboot Options

☒ One server at a time  
☐ All servers

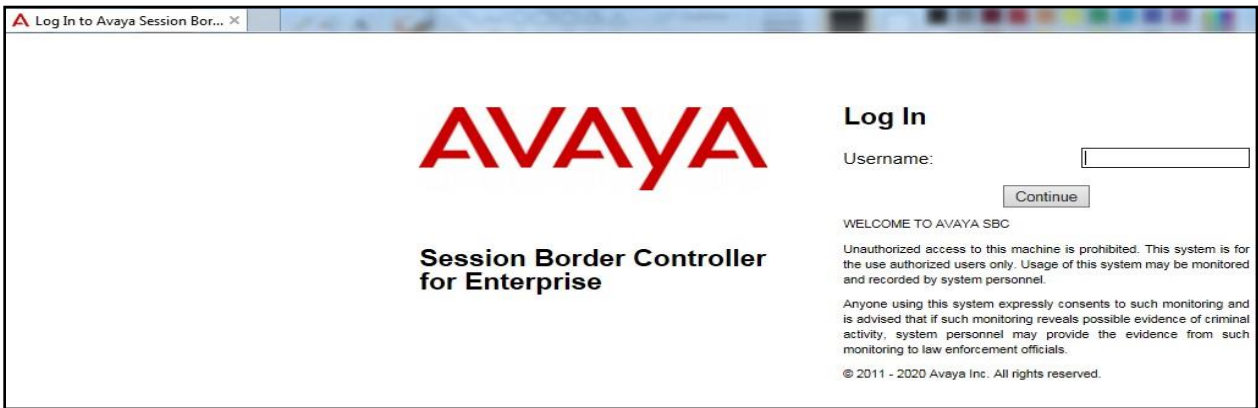
Help

## 8. Configure Avaya Session Border Controller for Enterprise

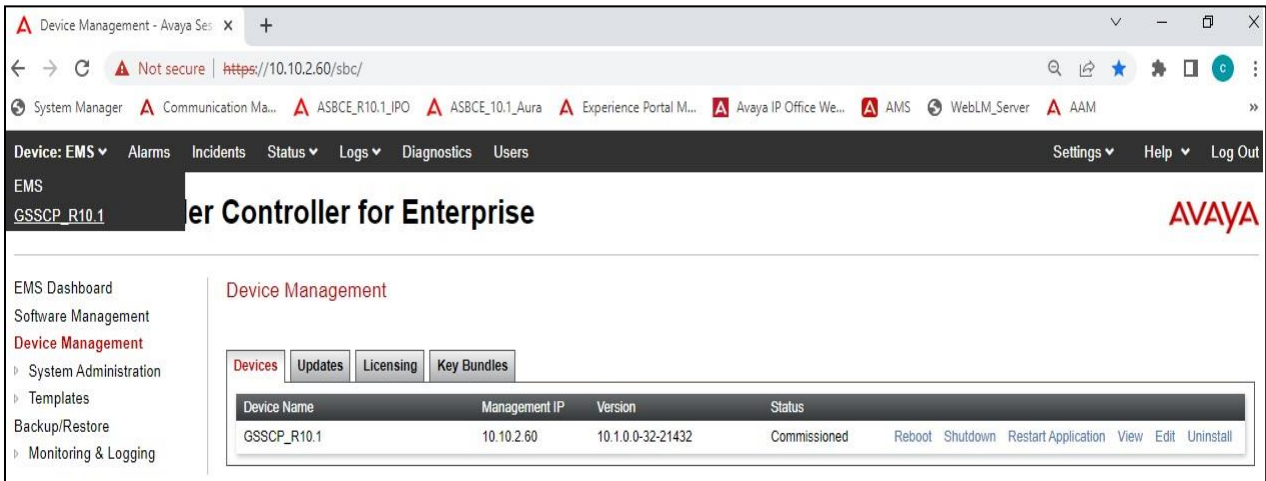
This section describes the configuration of the Session Border Controller for Enterprise (Avaya SBCE). The Avaya SBCE provides security and manipulation of signalling to deliver an interface to the Service Provider’s SIP Trunk that is standard where possible and adapted to the Service Provider’s SIP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

### 8.1. Access 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, on the top-left of the screen, under **Device:** select the required device from the drop-down menu. with a menu on the left-hand side. In this case, **GSSCP\_R10.1** is used as a starting point for all configuration of the Avaya SBCE.



To view system information that was configured during installation, navigate to **Device Management**. A list of installed devices is shown in the right pane. In the case of the sample configuration, a single device named **GSSCP\_R10.1** is shown. To view the configuration of this device, click **View** (the third option from the right).

The screenshot shows the Avaya Session Border Controller (SBC) web interface. The top navigation bar includes links for Alarms, Incidents, Status, Logs, Diagnostics, and Users. The main heading is "Session Border Controller for Enterprise". The left sidebar lists various management options, with "Device Management" highlighted. The main content area displays a table of installed devices:

Device Name	Management IP	Version	Status
GSSCP_R10.1	10.10.2.60	10.1.0.0-32-21432	Commissioned

Below the table, there are action buttons: Reboot, Shutdown, Restart Application, View, Edit, and Uninstall.

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

The screenshot shows the "System Information: GSSCP\_R10.1" screen. It contains the following sections:

- General Configuration:**
  - Appliance Name: GSSCP\_R10.1
  - Box Type: SIP
  - Deployment Mode: Proxy
- Device Configuration:**
  - HA Mode: No
  - Two Bypass Mode: No
- License Allocation:**
  - Standard Sessions Requested: 0
  - Advanced Sessions Requested: 0
  - Scopia Video Sessions Requested: 0
  - CES Sessions Requested: 0
  - Transcoding Sessions Requested: 0
  - AMR: ☒
  - Premium Sessions Requested: 0
  - CLID: ---
  - Encryption Available: Yes ☒
- Network Configuration:**

IP	Public IP	Network Prefix or Subnet Mask	Gateway	Interface
10.10.3.35	10.10.3.35	255.255.255.0	10.10.3.1	A1
192.168.122.55	192.168.122.55	255.255.255.128	192.168.122.9	B1
- DNS Configuration:**
  - Primary DNS: 8.8.8.8
  - Secondary DNS: 8.8.4.4
  - DNS Location: DMZ
  - DNS Client IP: 192.168.122.55
- Management IP(s):**
  - IP #1 (IPv4): 10.10.2.60

## 8.2. Define Network Management

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 **Network & Flows → Network Management** in the main menu on the left-hand side and click on **Add**. Enter details for the external interfaces in the dialogue box:

- Enter a descriptive name in the **Name** field.
- Enter the default gateway IP address for the external interfaces in the **Default Gateway** field.
- Enter the subnet mask in the **Network Prefix or Subnet Mask** field.
- Select the external physical interface to be used from the **Interface** drop down menu. In the test environment, this was **B1**.
- Click on **Add** and an additional row will appear allowing an IP address to be entered.
- Enter the external IP address of the Avaya SBCE on the SIP trunk in the **IP Address** field and leave the **Public IP** and **Gateway Override** fields blank.
- Click on **Finish** to complete the interface definition.

The screenshot shows a 'Network' dialog box with a warning message at the top: 'Modifications to the interfaces and IP addresses are service impacting and take effect immediately. If changes are made, sessions using this network will be dropped.' Below the warning, there are four input fields: 'Name' (B1\_External), 'Default Gateway' (192.168.122.9), 'Network Prefix or Subnet Mask' (255.255.255.0), and 'Interface' (B1). An 'Add' button is located to the right of the 'Interface' field. Below these fields is a table with three columns: 'IP Address', 'Public IP', and 'Gateway Override'. The first row contains the values '192.168.122.55', 'Use IP Address', and 'Use Default'. A 'Delete' button is located to the right of the first row. At the bottom of the dialog is a 'Finish' button.

IP Address	Public IP	Gateway Override
192.168.122.55	Use IP Address	Use Default

Click on **Add** to define the internal interfaces or Edit if it was defined during installation of the Avaya SBCE. Enter details in the dialogue box:

- Enter a descriptive name in the **Name** field.
- Enter the default gateway IP address for the internal interfaces in the **Default Gateway** field.
- Enter the subnet mask in the **Network Prefix or Subnet Mask** field.
- Select the internal physical interface to be used from the **Interface** drop down menu. In the test environment, this was **A1**.
- Click on **Add** and an additional row will appear allowing an IP address to be entered.
- Enter the internal IP address of the Avaya SBCE on the SIP trunk in the **IP Address** field and leave the **Public IP** and **Gateway Override** fields blank.
- Click on **Finish** to complete the interface definition.

**Network** [X]

Modifications to the interfaces and IP addresses are service impacting and take effect immediately. If changes are made, sessions using this network will be dropped.

Name: A1\_Internal

Default Gateway: 10.10.3.1

Network Prefix or Subnet Mask: 255.255.255.0

Interface: A1

Add

IP Address	Public IP	Gateway Override
10.10.3.35	Use IP Address	Use Default

Delete

Finish

The following screenshot shows the completed Network Management configuration:

**Network Management**

Interfaces Networks

Add

Name	Gateway	Subnet Mask / Prefix Length	Interface	IP Address	
A1_Internal	10.10.3.1	255.255.255.0	A1	10.10.3.35	Edit Delete
B1_External	192.168.122.9	255.255.255.0	B1	192.168.122.55	Edit Delete

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



Network Management		
Interfaces Networks		
Interface Name	VLAN Tag	Status
A1		Enabled
A2		Disabled
B1		Enabled
B2		Disabled

**Note:** to ensure that the Avaya SBCE uses the interfaces defined, the Application must be restarted.

- Click on **Device Management** in the main menu (not shown).
- Select **Restart Application** indicated by an icon in the status bar (not shown).

A status box will appear that will indicate when the restart is complete.

## 8.3. Define TLS Profiles

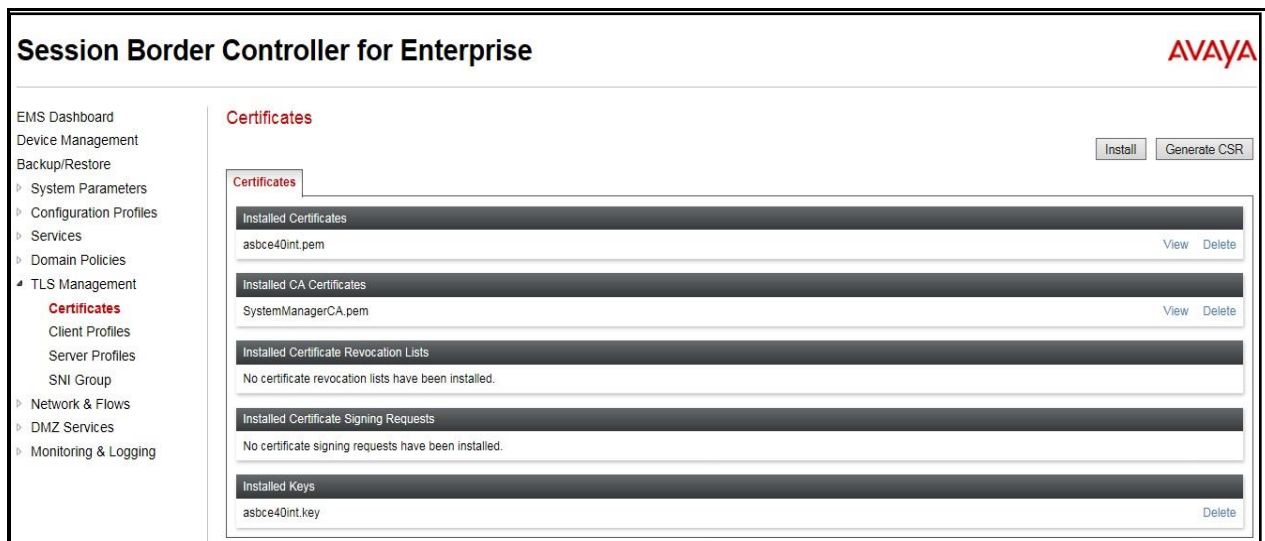
For the compliance test, TLS transport is used for signalling on the SIP trunk between Session Manager and the Avaya SBCE. Compliance testing was done using identity certificates signed by a local certificate authority. The generation and installation of these certificates are beyond the scope of these Application Notes.

The following procedures show how to view the certificates and configure the Client and Server profiles to support the TLS connection.

### 8.3.1. Certificates

To view the certificates currently installed on the Avaya SBCE, navigate to **TLS Management** → **Certificates**:

- Verify that an Avaya SBCE identity certificate (**asbce40int.pem**) is present under **Installed Certificates**.
- Verify that certificate authority root certificate (**SystemManagerCA.pem**) is present under **Installed CA certificates**.
- Verify that private key associated with the identity certificate (**asbce40int.key**) is present under **Installed Keys**.





### 8.3.2. Client Profile

To create a new client profile, navigate to **TLS Management** → **Client Profile** in the left pane and click **Add** (not shown).

- Set **Profile Name** to a descriptive name. **GSSCP\_Client** was used in the compliance testing.
- Set **Certificate** to the identity certificate **asbce40int.pem** used in the compliance testing.
- **Peer Verification** is automatically set to **Required**.
- Set **Peer Certificate Authorities** to the **SystemManagerCA.pem** identity certificate.
- Set **Verification Depth** to **1**.

Click **Next** to accept default values for the next screen and click **Finish** (not shown).

The screenshot displays the 'Client Profiles: GSSCP\_Client' configuration window. On the left, a sidebar shows 'Client Profiles' with 'GSSCP\_Client' selected. The main area is divided into two sections. The top section, titled 'Client Profile', contains a 'TLS Profile' table with fields: Profile Name (GSSCP\_Client), Certificate (asbce40int.pem), and SNI (Enabled). Below this is a 'Certificate Verification' table with fields: Peer Verification (Required), Peer Certificate Authorities (SystemManagerCA.pem), Peer Certificate Revocation Lists (---), Verification Depth (1), and Extended Hostname Verification (disabled). The bottom section contains 'Renegotiation Parameters' (Renegotiation Time: 0, Renegotiation Byte Count: 0) and 'Handshake Options' (Version: TLS 1.2, TLS 1.1, TLS 1.0; Ciphers: Default, FIPS, Custom; Value: HIGH:!DH:!ADH:!MD5:!aNULL:!eNULL:@STRENGTH). An 'Edit' button is at the bottom right.

Client Profiles: GSSCP_Client	
Click here to add a description.	
<b>Client Profile</b>	
<b>TLS Profile</b>	
Profile Name	GSSCP_Client
Certificate	asbce40int.pem
SNI	<input type="checkbox"/> Enabled
<b>Certificate Verification</b>	
Peer Verification	Required
Peer Certificate Authorities	SystemManagerCA.pem
Peer Certificate Revocation Lists	---
Verification Depth	1
Extended Hostname Verification	<input type="checkbox"/>
<b>Renegotiation Parameters</b>	
Renegotiation Time	0
Renegotiation Byte Count	0
<b>Handshake Options</b>	
Version	<input checked="" type="checkbox"/> TLS 1.2 <input checked="" type="checkbox"/> TLS 1.1 <input checked="" type="checkbox"/> TLS 1.0
Ciphers	<input checked="" type="radio"/> Default <input type="radio"/> FIPS <input type="radio"/> Custom
Value	HIGH:!DH:!ADH:!MD5:!aNULL:!eNULL:@STRENGTH



### 8.3.3. Server Profile

To create a new server profile, navigate to **TLS Management** → **Server Profile** in the left pane and click **Add** (not shown).

- Set **Profile Name** to a descriptive name. **GSSCP\_Server** was used in the compliance testing
- Set **Certificate** to the identity certificate **asbce40int.pem** used in the compliance testing.
- Set **Peer Verification** to **Optional**.

Click **Next** to accept default values for the next screen and click **Finish** (not shown).

The screenshot displays the configuration interface for a server profile named 'GSSCP\_Server'. The interface is divided into two main sections. The top section, titled 'Server Profile', contains fields for 'Profile Name' (GSSCP\_Server), 'Certificate' (asbce40int.pem), and 'SNI Options' (None). Below these is a 'Certificate Verification' section with 'Peer Verification' set to 'Optional', and fields for 'Peer Certificate Authorities', 'Peer Certificate Revocation Lists', 'Verification Depth' (1), and 'Extended Hostname Verification' (unchecked). The bottom section contains 'Renegotiation Parameters' with 'Renegotiation Time' and 'Renegotiation Byte Count' both set to 0. The 'Handshake Options' section shows 'Version' with checkboxes for TLS 1.2, TLS 1.1, and TLS 1.0 (all checked), 'Ciphers' set to 'Default' (with radio buttons for Default, FIPS, and Custom), and a 'Value' field containing the string 'HIGH:!DH:!ADH:!MD5:!aNULL:!eNULL:@STRENGTH'. An 'Edit' button is located at the bottom right of the configuration area.

Server Profiles: GSSCP_Server	
Click here to add a description.	
Server Profile	
TLS Profile	
Profile Name	GSSCP_Server
Certificate	asbce40int.pem
SNI Options	None
Certificate Verification	
Peer Verification	Optional
Peer Certificate Authorities	---
Peer Certificate Revocation Lists	---
Verification Depth	1
Extended Hostname Verification	<input type="checkbox"/>
Renegotiation Parameters	
Renegotiation Time	0
Renegotiation Byte Count	0
Handshake Options	
Version	<input checked="" type="checkbox"/> TLS 1.2 <input checked="" type="checkbox"/> TLS 1.1 <input checked="" type="checkbox"/> TLS 1.0
Ciphers	<input checked="" type="radio"/> Default <input type="radio"/> FIPS <input type="radio"/> Custom
Value	HIGH:!DH:!ADH:!MD5:!aNULL:!eNULL:@STRENGTH

## 8.4. Define Interfaces

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

### 8.4.1. Signalling Interfaces

To define the signalling interfaces on the Avaya SBCE, navigate to **Network & Flows** → **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 **A1\_Internal** signalling interface IP addresses defined in **Section 8.2**.
- Select **TLS** port number, **5061** is used for Session Manager.
- Select a **TLS Profile** defined in **Section 8.3.3** from the drop-down menu.
- Click **Finish**.

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 **B1\_external** signalling interface IP address defined in **Section 8.2**.
- Select **UDP** port number, **5060** is used for the Vodafone Germany SIP Trunk.
- Click **Finish**.

Signaling Interface						
Signaling Interface						
Name	Signaling IP Network	TCP Port	UDP Port	TLS Port	TLS Profile	
Signalling_External	192.168.122.55 B1_External (B1, VLAN 0)	---	5060	---	None	Edit Delete
Signalling_Internal	10.10.3.35 A1_Internal (A1, VLAN 0)	---	---	5061	GSSCP_Server	Edit Delete

## 8.4.2. Media Interfaces

To define the media interfaces on the Avaya SBCE, navigate to **Network & Flows → 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 for the internal interface to be used in the server flow:

- Select **Add Media Interface** 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 **A1\_Internal** media interface IP address defined in **Section 8.2**.
- For **Port Range**, enter **35000-40000**.
- Click **Finish**.

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.
- In the **Name** field enter a descriptive name for the external media interface.
- For **Media IP**, select the **B1\_External** media interface IP address defined in **Section 8.2**.
- Select **Port Range**, enter **35000-40000**.
- Click **Finish**.

Media Interface			
Media Interface			Add
Name	Media IP Network	Port Range	
Media_Internal	10.10.3.35 A1_Internal (A1, VLAN 0)	35000 - 40000	Edit Delete
Media_External	192.168.122.55 B1_External (B1, VLAN 0)	35000 - 40000	Edit Delete

## 8.5. Define Server Interworking

Server interworking is defined for each server connected to the Avaya SBCE. In this case, Vodafone Germany is connected as the Trunk Server and Session Manager is connected as the Call Server.

### 8.5.1. 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 **Configuration Profiles** → **Server Interworking** and click on **Add**.

- Enter profile name such as Avaya and click **Next** (Not Shown).
- Check **Hold Support** = **None**.
- Uncheck **SIPS Required**.
- 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=sendsonly <input type="radio"/> Microsoft Teams
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 checked="" 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
SIPS Required	<input type="checkbox"/>
Mediasec Handling	<input type="checkbox"/>

On the **Advanced** Tab:

- Check **Record Routes = Both Sides**.
- Ensure **Extensions = Avaya**.
- Check **Has Remote SBC**.
- All other options on the **Advanced** Tab can be left at default.

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 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"/>
MOBX Re-INVITE Handling	<input type="checkbox"/>
NATing for 301/302 Redirection	<input checked="" type="checkbox"/>
<b>DTMF</b>	
DTMF Support	<input checked="" type="radio"/> None> <input type="radio"/> SIP Notify> <input type="radio"/> RFC 2833 Relay & SIP Notify> <input type="radio"/> SIP Info> <input type="radio"/> RFC 2833 Relay & SIP Info> <input type="radio"/> Inband>
<input type="button" value="Finish"/>	

### 8.5.2. Server Interworking – Vodafone Germany

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 **Configuration Profiles** → **Server Interworking** and click on **Add**.

- Enter profile name such as **VFDE** and click **Next** (Not Shown).
- Check **Hold Support** = **None**.
- Check **Refer Handling** as per **Section 2.2**
- Uncheck **SIPS Required**
- 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 <input type="radio"/> Microsoft Teams
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 checked="" 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 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
SIPS Required	<input type="checkbox"/>
Mediasec Handling	<input type="checkbox"/>

On the **Advanced** Tab:

- Check **Record Routes = Both Sides**.
- Ensure **Extensions = None**.
- Check **Has Remote SBC**.
- All other options on the **Advanced** Tab can be left at default.

Click **Finish**.

Record Routes

☐ None

☐ Single Side

☒ Both Sides

☐ Dialog-Initiate Only (Single Side)

☐ Dialog-Initiate Only (Both Sides)

Include End Point IP for Context Lookup ☐

Extensions None ▾

Diversion Manipulation ☐

Diversion Condition None ▾

Diversion Header URI

Has Remote SBC ☒

Route Response on Via Port ☐

Relay INVITE Replace for SIPREC ☐

MOBX Re-INVITE Handling ☐

NATing for 301/302 Redirection ☒

**DTMF**

DTMF Support

☒ None>

☐ SIP Notify>

☐ RFC 2833 Relay & SIP Notify>

☐ SIP Info>

☐ RFC 2833 Relay & SIP Info>

☐ Inband>

Finish

## 8.6. Signalling 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

During compliance testing, it was observed when performing Blind Transfer to PSTN numbers on inbound calls (i.e. PSTN (A) -> Avaya (B) -> Blind Transfer -> PSTN (C)) from Avaya SIP handsets, that Vodafone Germany was responding with a “403 Forbidden”. The reason Vodafone Germany was responding with “403 Forbidden” is that the Avaya SIP handsets populate the P-Asserted-Identity Header with the originating caller (A) CLID and also insert a Header called “Referred-By” when completing the transfer. Vodafone Germany do not recognize this header “Referred-By” and Vodafone Germany require the P-Asserted-Identity Header to be populated with the CLID of a known Vodafone Germany number (B) on their SIP platform.

In order for Blind Transfers to PSTN to complete successfully, a SigMa script was created on the Avaya SBCE. This script checks to see if a Referred-By Header is present and if present, the Referred-By Header will be deleted and it will populate the P-Asserted-Identity Header with the From Header CLID and the Blind Transfer is executed successfully.

To define the signalling manipulation, navigate to **Configuration Profiles → Signaling Manipulations** and click on **Add** and enter a title. A new blank Signaling Manipulation Editor window will pop up. The script text is as follows:

```
/*Script to copy From Header to PAI Header and delete Referred-By Header for Blind Xfer */

within session "INVITE"
{
  act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
  {
    if (exists(%HEADERS["Referred-By"][1])) then
    {
      %DivUser = %HEADERS["From"][1].URI.USER;
      %HEADERS["P-Asserted-Identity"][1].URI.USER = %DivUser;
      remove(%HEADERS["Referred-By"][1]);
    }
  }
}
```



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



## 8.7. Define Servers

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

### 8.7.1. Server Configuration – Avaya

From the left-hand menu select **Services → SIP Servers** and click on **Add** and enter a descriptive name. On the **Add Server Configuration Profiles** tab, set the following:

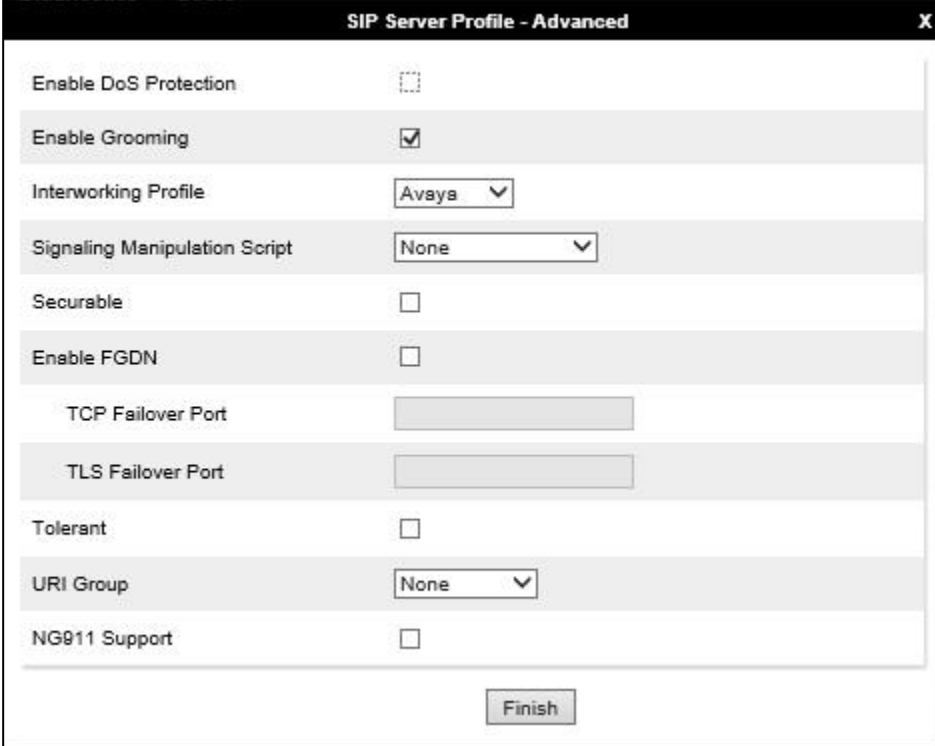
- Select **Server Type** to be **Call Server**.
- Select **TLS Client Profile** to be **GSSCP\_Client** as defined in **Section 8.3.2**.
- Enter **IP Address / FQDN** to **10.10.3.42** (Session Manager IP Address).
- For **Port**, enter **5061**.
- For **Transport**, select **TLS**.
- Click on **Next** (not shown) to use default entries on the **Authentication** and **Heartbeat** tabs.

The screenshot shows the 'SIP Server Profile - General' configuration window. At the top, a blue banner states: 'Server Type can not be changed while this SIP Server Profile is associated to a Server Flow.' Below this, the 'Server Type' is set to 'Call Server'. The 'SIP Domain' field is empty. The 'DNS Query Type' is set to 'NONE/A'. The 'TLS Client Profile' is set to 'GSSCP\_Client'. An 'Add' button is located to the right of these fields. Below the 'Add' button is a table with three columns: 'IP Address / FQDN', 'Port', and 'Transport'. The first row contains the values '10.10.3.42', '5061', and 'TLS'. A 'Delete' button is located to the right of the first row.

IP Address / FQDN	Port	Transport
10.10.3.42	5061	TLS

On the **Advanced** tab:

- Check **Enable Grooming**.
- Select **Avaya** for **Interworking Profile**.
- Click **Finish**.



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

Setting	Value
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input checked="" type="checkbox"/>
Interworking Profile	Avaya
Signaling Manipulation Script	None
Securable	<input type="checkbox"/>
Enable FGDN	<input type="checkbox"/>
TCP Failover Port	
TLS Failover Port	
Tolerant	<input type="checkbox"/>
URI Group	None
NG911 Support	<input type="checkbox"/>

At the bottom right of the window is a "Finish" button.

### 8.7.2. Server Configuration – Vodafone Germany

To define the Vodafone Germany Trunk Server, navigate to **Services → SIP Servers** and click on **Add** and enter a descriptive name. On the **Add Server Configuration Profile** tab, set the following:

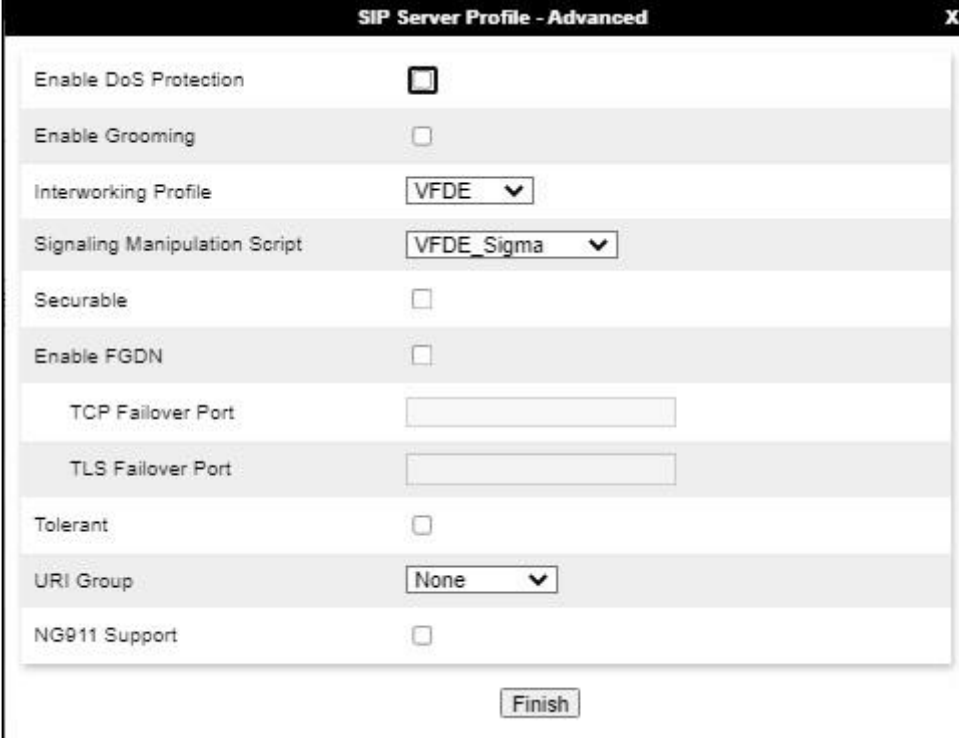
- Select **Server Type** to be **Trunk Server**.
- Enter **IP Address / FQDN** to **192.168.165.116** (Vodafone Germany SIP Platform).
- For **Port**, enter **5060**.
- For **Transport**, select **UDP**.
- Click on **Next** (not shown) to use default entries on the **Authentication** and **Heartbeat** tabs.

The screenshot shows the 'SIP Server Profile - General' configuration window. At the top, a blue banner states: 'Server Type can not be changed while this SIP Server Profile is associated to a Server Flow.' Below this, there are four configuration fields: 'Server Type' (set to 'Trunk Server'), 'SIP Domain' (empty), 'DNS Query Type' (set to 'NONE/A'), and 'TLS Client Profile' (set to 'None'). An 'Add' button is located to the right of these fields. Below the fields is a table with three columns: 'IP Address / FQDN / CIDR Range', 'Port', and 'Transport'. The table contains one entry with the IP '192.168.165.116', port '5060', and transport 'UDP'. A 'Delete' button is next to the entry.

IP Address / FQDN / CIDR Range	Port	Transport
192.168.165.116	5060	UDP

On the Advanced tab:

- Select **VFDE** for **Interworking Profile**.
- Select **VFDE\_Sigma** for **Signaling Manipulation Script**.
- Click **Finish**.



The screenshot shows the 'SIP Server Profile - Advanced' configuration window. It contains several settings with checkboxes and dropdown menus. The 'Interworking Profile' is set to 'VFDE' and the 'Signaling Manipulation Script' is set to 'VFDE\_Sigma'. The 'Finish' button is located at the bottom right of the window.

Setting	Value
Enable DoS Protection	<input checked="" type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	VFDE
Signaling Manipulation Script	VFDE_Sigma
Securable	<input type="checkbox"/>
Enable FGDN	<input type="checkbox"/>
TCP Failover Port	
TLS Failover Port	
Tolerant	<input type="checkbox"/>
URI Group	None
NG911 Support	<input type="checkbox"/>

Finish

## 8.8. 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 Vodafone Germany address 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.

### 8.8.1. Routing – Avaya

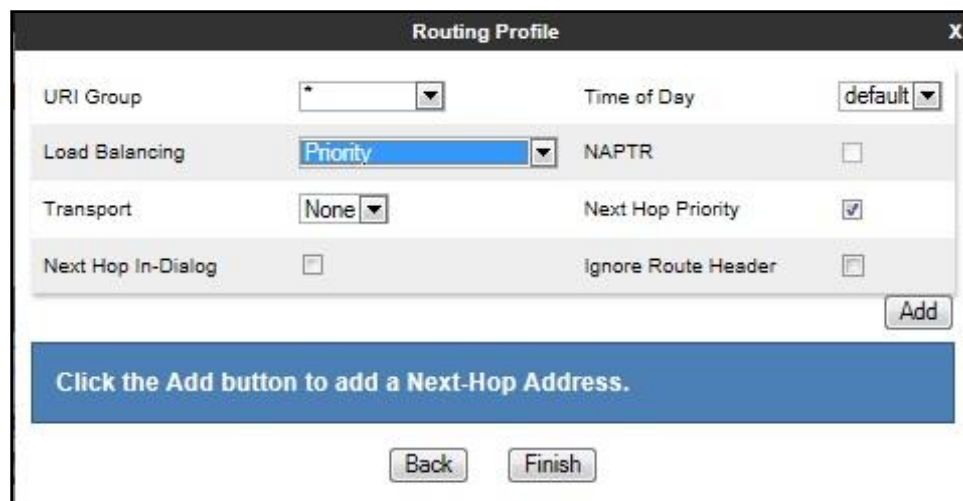
Create a Routing Profile for Session Manager.

- Navigate to **Configuration Profiles → Routing** and select **Add Profile**.
- 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: \* (dropdown)
- Time of Day: default (dropdown)
- Load Balancing: Priority (dropdown)
- NAPTR: ☐
- Transport: None (dropdown)
- Next Hop Priority: ☒
- Next Hop In-Dialog: ☐
- Ignore Route Header: ☐

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

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

- **Priority/Weight = 1.**
- **SIP Server Profile = Avaya (Section 8.7.1)** from drop down menu.
- **Next Hop Address = Select 10.10.3.42:5061 (TLS)** from drop down menu.
- Click **Finish**.

Priority / Weight	LDAP Search Attribute	LDAP Search Regex Pattern	LDAP Search Regex Result	SIP Server Profile	Next Hop Address	Transport
1				Avaya	10.10.3.42:5061 (TLS)	None

### 8.8.2. Routing – Vodafone Germany

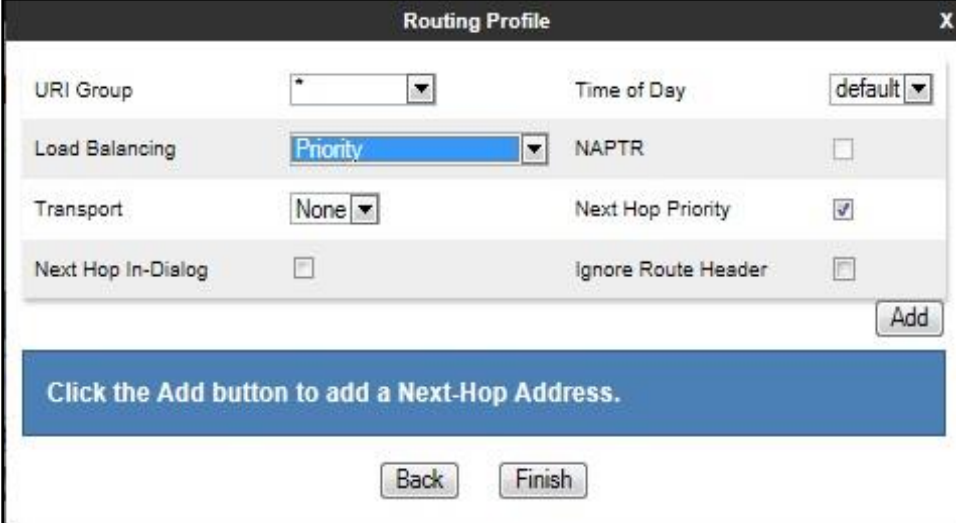
Create a Routing Profile for Vodafone Germany SIP network.

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

Profile Name: VFDE

Next

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



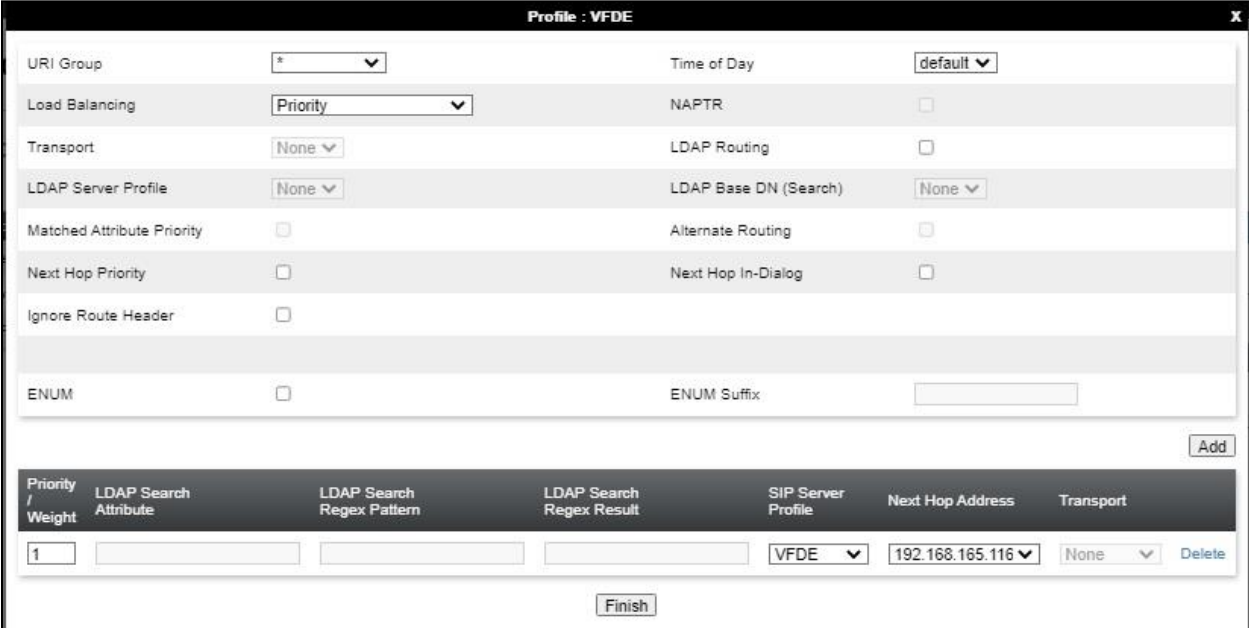
The Routing Profile window is a configuration dialog with a title bar and a close button. It contains several settings:

- 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 unchecked.
- Transport: A dropdown menu with "None" as the selected value.
- Next Hop Priority: A checkbox that is checked.
- Next Hop In-Dialog: A checkbox that is unchecked.
- Ignore Route Header: A checkbox that is unchecked.

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

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

- **Load Balancing = Priority.**
- **SIP Server Profile = VFDE (Section 8.7.2)** from drop down menu.
- **Next Hop Address = Select 192.168.165.116 (UDP)** from drop down menu.
- Click **Finish**.



The Profile : VFDE window is a configuration dialog with a title bar and a close button. It contains several settings:

- 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 unchecked.
- Transport: A dropdown menu with "None" as the selected value.
- LDAP Server Profile: A dropdown menu with "None" as the selected value.
- LDAP Routing: A checkbox that is unchecked.
- LDAP Base DN (Search): A dropdown menu with "None" as the selected value.
- Matched Attribute Priority: A checkbox that is unchecked.
- Alternate Routing: A checkbox that is unchecked.
- Next Hop Priority: A checkbox that is unchecked.
- Next Hop In-Dialog: A checkbox that is unchecked.
- Ignore Route Header: A checkbox that is unchecked.
- ENUM: A checkbox that is unchecked.
- ENUM Suffix: A text input field.

At the bottom right is an "Add" button. Below the settings is a table with the following columns: Priority / Weight, LDAP Search Attribute, LDAP Search Regex Pattern, LDAP Search Regex Result, SIP Server Profile, Next Hop Address, Transport, and a Delete button.

Priority / Weight	LDAP Search Attribute	LDAP Search Regex Pattern	LDAP Search Regex Result	SIP Server Profile	Next Hop Address	Transport	Delete
1				VFDE	192.168.165.116	None	

At the bottom center is a "Finish" button.



## 8.9. 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 cannot 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 **Configuration 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

RenameCloneDelete

Topology Hiding Profiles

default

cisco\_th\_profile

Avaya

VFDE

Click here to add a description.

Topology Hiding

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

Edit

To define Topology Hiding for Vodafone Germany, navigate to **Configuration 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 Vodafone Germany 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: VFDE

Add
Rename
Clone
Delete

Topology Hiding Profiles

default

cisco\_th\_profile

Avaya

VFDE

Click here to add a description.

Topology Hiding

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

Edit

## 8.10.Domain Policies

Domain Policies allow the configuration of sets of rules designed to control and normalize the behavior of call flows, based upon various criteria of communication sessions originating from or terminating in the enterprise. Domain Policies include rules for Application, Media, Signaling, Security, etc.

In the reference configuration, only new Media Rules were defined. All other rules under Domain Policies, linked together on End Point Policy Groups later in this section, used one of the default sets already pre-defined in the configuration. Please note that changes should not be made to any of the defaults. If changes are needed, it is recommended to create a new rule by cloning one the defaults and then make the necessary changes to the new rule.

CMN; Reviewed:  
SPOC 4/25/2023

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### 8.10.1. Media Rules

A media rule defines the processing to be applied to the selected media. For the compliance test, a media rule was created for Session Manager to use SRTP, while the predefined **default-low-med** media rule was used for the Vodafone Germany SIP trunk.

To define the Media Rule for Session Manager, navigate to **Domain Policies → Media Rules** in the main menu on the left-hand side. Click on **Add** and enter details in the Media Rule pop-up box (not shown)

- In the **Rule Name** field enter a descriptive name such as **Avaya\_SRTP**.
- Set **Preferred Format #1** to **SRTP\_AES\_CM\_128\_HMAC\_SHA1\_80**.
- Set **Preferred Format #2** to **RTP**.
- Uncheck **Encrypted RTCP**.
- Check **Capability Negotiation** under **Miscellaneous**.

Default values were used for all other fields. Click **Finish** (not shown).

The screenshot shows the 'Media Rules: Avaya\_SRTP' configuration window. On the left is a sidebar with a 'Media Rules' section containing a list of rules: 'default-low-med', 'default-low-med-enc', 'default-high', 'default-high-enc', 'avaya-low-med-enc', and 'Avaya\_SRTP' (which is highlighted in red). Above this list is an 'Add' button. To the right of the sidebar is the main configuration area. At the top right of this area are 'Rename', 'Clone', and 'Delete' buttons. Below them is a blue bar with the text 'Click here to add a description.' The configuration area is divided into four tabs: 'Encryption' (selected), 'Codec Prioritization', 'Advanced', and 'QoS'. The 'Encryption' tab contains two sections: 'Audio Encryption' and 'Video Encryption'. The 'Audio Encryption' section has fields for 'Preferred Formats' (set to 'SRTP\_AES\_CM\_128\_HMAC\_SHA1\_80' and 'RTP'), 'Encrypted RTCP' (unchecked), 'MKI' (unchecked), 'Lifetime' (set to 'Any'), 'Interworking' (unchecked), 'Symmetric Context Reset' (checked), and 'Key Change in New Offer' (unchecked). The 'Video Encryption' section has fields for 'Preferred Formats' (set to 'RTP'), 'Interworking' (unchecked), 'Symmetric Context Reset' (checked), and 'Key Change in New Offer' (unchecked). Below these sections is a 'Miscellaneous' section with 'Capability Negotiation' checked. An 'Edit' button is located at the bottom right of the configuration area.

Media Rules	Configuration
default-low-med	
default-low-med-enc	
default-high	
default-high-enc	
avaya-low-med-enc	
<b>Avaya_SRTP</b>	<b>Encryption</b> Preferred Formats: SRTP_AES_CM_128_HMAC_SHA1_80, RTP Encrypted RTCP: <input type="checkbox"/> MKI: <input type="checkbox"/> Lifetime: Any Interworking: <input type="checkbox"/> Symmetric Context Reset: <input checked="" type="checkbox"/> Key Change in New Offer: <input type="checkbox"/> Video Encryption Preferred Formats: RTP Interworking: <input type="checkbox"/> Symmetric Context Reset: <input checked="" type="checkbox"/> Key Change in New Offer: <input type="checkbox"/> Miscellaneous Capability Negotiation: <input checked="" type="checkbox"/>

For the compliance test, the default media rule **default-low-med** was used for Vodafone Germany.

Media Rules: default-low-med

Add

Clone

Media Rules

default-low-med

default-low-med-enc

default-high

default-high-enc

avaya-low-med-enc

Avaya\_SRTP

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

EncryptionCodec PrioritizationAdvancedQoS

Audio Encryption

Preferred FormatsRTP

Interworking☒

Symmetric Context Reset☒

Key Change in New Offer☐

Video Encryption

Preferred FormatsRTP

Interworking☒

Symmetric Context Reset☒

Key Change in New Offer☐

Miscellaneous

Capability Negotiation☐

Edit

## 8.11. End Point Policy Groups

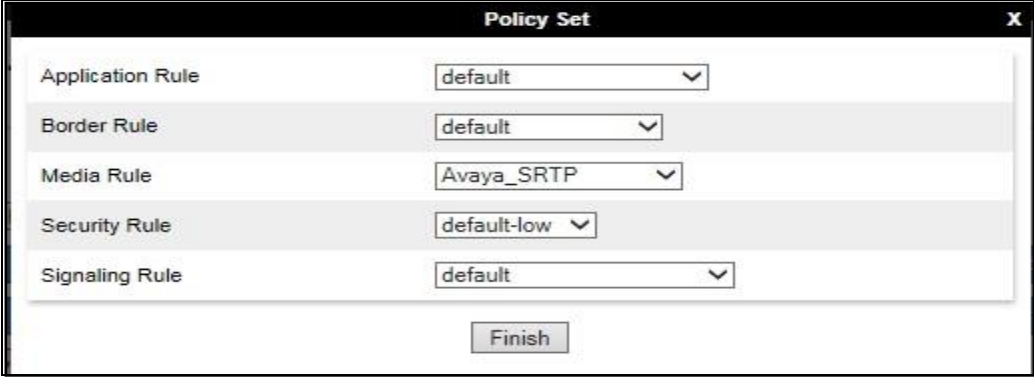
An end point policy group is a set of policies that will be applied to traffic between the Avaya SBCE and a signaling endpoint (connected server). Thus, one end point policy group must be created for Session Manager and another for the Vodafone Germany SIP trunk. The end point policy group is applied to the traffic as part of the end point flow defined in **Section 8.11.**

### 8.11.1. End Point Policy Group – Session Manager

To define an End Point policy for Session Manager, navigate to **Domain Policies → End Point Policy Groups** in the main menu on the left-hand side. Click on **Add** and enter details in the Policy Group pop-up box (not shown).

- In the **Group Name** field enter a descriptive name, in this case **Avaya**, and click **Next** (not shown).
- Leave the **Application Rule**, **Border Rule**, **Security Rule** and **Signalling Rule** fields at their default values.
- In the **Media Rule** drop down menu, select the recently added Media Rule called **Avaya\_SRTP**.

Click **Finish**.



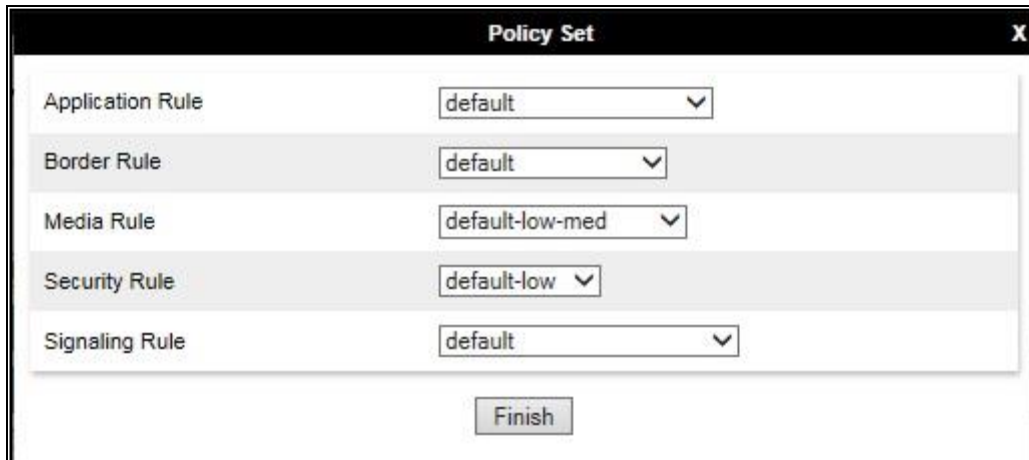
The screenshot shows a 'Policy Set' dialog box with a close button (X) in the top right corner. It contains five rows of configuration options, each with a label and a dropdown menu:

Label	Value
Application Rule	default
Border Rule	default
Media Rule	Avaya_SRTP
Security Rule	default-low
Signaling Rule	default

At the bottom center of the dialog is a 'Finish' button.

### 8.11.2. End Point Policy Group – Vodafone Germany

For the compliance test, the predefined End Point Policy **default-low** was used for the Vodafone Germany End Point Policy Group.



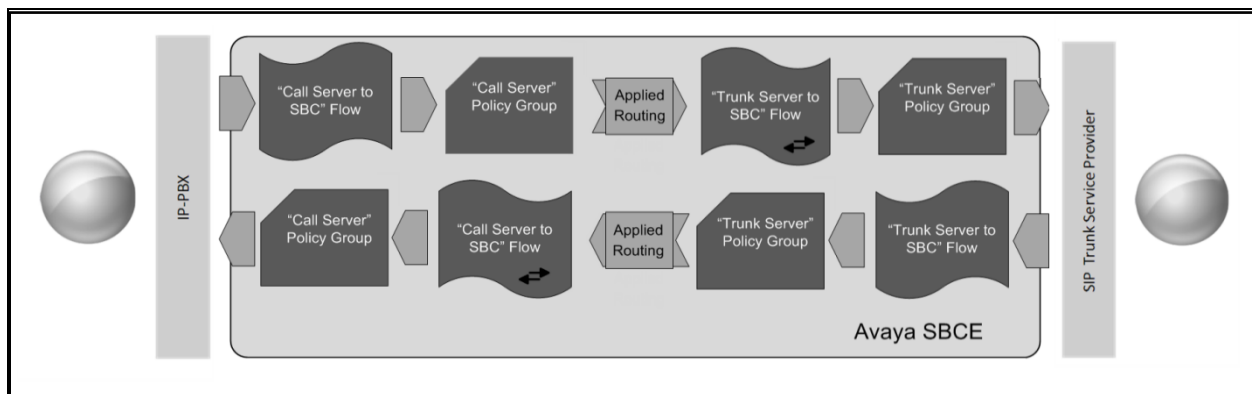
The screenshot shows a 'Policy Set' configuration window with a black title bar and a close button (X) in the top right corner. The window contains a list of five rules, each with a corresponding dropdown menu. The rules and their selected values are: Application Rule (default), Border Rule (default), Media Rule (default-low-med), Security Rule (default-low), and Signaling Rule (default). The rows for Border Rule, Security Rule, and Signaling Rule are highlighted in light gray. A 'Finish' button is located at the bottom center of the window.

Rule Type	Selected Policy
Application Rule	default
Border Rule	default
Media Rule	default-low-med
Security Rule	default-low
Signaling Rule	default

Finish

## 8.12. Server Flows

Server Flows combine the previously defined profiles into outgoing flows from Session Manager to Vodafone Germany's SIP Trunk and incoming flows from Vodafone Germany's SIP Trunk to Session Manager. 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 Vodafone Germany SIP Trunk and vice versa. The following screenshot shows all configured flows.

**End Point Flows**

Subscriber Flows | **Server Flows** | Add

Modifications made to a Server Flow will only take effect on new sessions.

Hover over a row to see its description.

SIP Server: Avaya

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile	
1	Call_Server	*	Signaling_External	Signaling_Internal	Avaya	VFDE	View Clone Edit Delete

SIP Server: VFDE

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

To define the inbound Server Flow for the Vodafone Germany SIP Trunk, navigate to **Network & Flows → 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 Vodafone Germany SIP Trunk, in the test environment **Trunk\_Server** was used.
- In the **Server Configuration** drop-down menu, select the Vodafone Germany server configuration defined in **Section 8.7.2**.
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 8.4.1**.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface defined in **Section 8.4.1**.
- In the **Media Interface** drop-down menu, select the external media interface defined in **Section 8.4.2**.
- Set the **End Point Policy Group** to the endpoint policy group **default-low**.
- In the **Routing Profile** drop-down menu, select the routing profile of the Session Manager defined in **Section 8.8.1**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the Vodafone Germany SIP Trunk defined in **Section 8.9** and click **Finish** (not shown).

The screenshot shows a configuration window titled "Flow: Trunk\_Server". It is divided into two main sections: "Criteria" and "Profile".

Criteria	
Flow Name	Trunk_Server
Server Configuration	VFDE
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Signaling_Internal

Profile	
Signaling Interface	Signaling_External
Media Interface	Media_External
Secondary Media Interface	None
End Point Policy Group	default-low
Routing Profile	Avaya
Topology Hiding Profile	VFDE
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	<input type="checkbox"/>
FQDN Support	<input type="checkbox"/>



To define the outbound server flow for Session Manager to the Vodafone Germany network, navigate to **Network & Flows → 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 server configuration for Session Manager defined in **Section 8.7.1**.
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 8.4.1**.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface defined in **Section 8.4.1**.
- In the **Media Interface** drop-down menu, select the external media interface defined in **Section 8.4.2**.
- Set the **End Point Policy Group** to the endpoint policy group **Avaya**.
- In the **Routing Profile** drop-down menu, select the routing profile of the Vodafone Germany SIP Trunk defined in **Section 8.8.2**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of Session Manager defined in **Section 8.9** and click **Finish** (not shown).

The screenshot shows a configuration window titled "Flow: Call\_Server" with a close button (X) in the top right corner. The window is divided into two main sections: "Criteria" on the left and "Profile" on the right.

**Criteria Section:**

Flow Name	Call_Server
Server Configuration	Avaya
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Signaling_External

**Profile Section:**

Signaling Interface	Signaling_Internal
Media Interface	Media_Internal
Secondary Media Interface	None
End Point Policy Group	Avaya
Routing Profile	VFDE
Topology Hiding Profile	Avaya
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	<input type="checkbox"/>
FQDN Support	<input type="checkbox"/>

## 9. Vodafone Germany SIP Trunk Configuration

The configuration of the Vodafone Germany equipment used to support Vodafone Germany's SIP Trunk is outside of the scope of these Application Notes and will not be covered. To obtain further information on Vodafone Germany equipment and system configuration please contact an authorized Vodafone Germany representative.

## 10. 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**.

Session Manager Entity Link Connection Status									
This page displays detailed connection status for all entity links from a Session Manager.									
Status Details for the selected Session Manager: Time Last Down: 12/09/19 11:10:34 Last Message Sent: 12/10/19 10:44:38 Time Last Up: 12/09/19 11:25:56 Last Response Latency (ms): 21									
All Entity Links for Session Manager: Session Manager									
Summary View									
4 Items <span>Filter: Enable</span>									
	SIP Entity Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
<input type="radio"/>	<a href="#">Avaya SBCE</a>	IPv4	10.10.3.30	5061	TLS	FALSE	UP	200 OK	UP
<input type="radio"/>	<a href="#">Communication Manager</a>	IPv4	10.10.3.44	5061	TLS	FALSE	UP	200 OK	UP

2. From 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
0002/001	T00011	in-service/idle	no
0002/002	T00012	in-service/idle	no
0002/003	T00013	in-service/idle	no
0002/004	T00014	in-service/idle	no
0002/005	T00015	in-service/idle	no
0002/006	T00016	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 Management → Monitoring & Logging → 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 the signalling interface IP address or “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\_R10.1

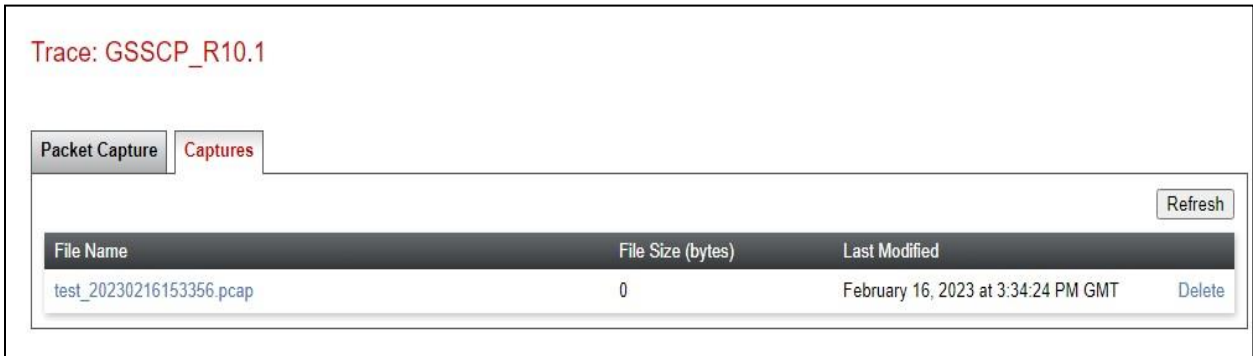
Packet Capture
Captures

**Packet Capture Configuration**

Status	Ready
Interface	B1 ▼
Local Address IP[:Port]	All ▼ :
Remote Address *, *:Port, IP, IP:Port	*
Protocol	UDP ▼
Maximum Number of Packets to Capture	10000
Capture Filename <small>Using the name of an existing capture will overwrite it.</small>	test.pcap

Start Capture
Clear

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



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

## 11. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager R10.1, Avaya Aura® Session Manager R10.1, Avaya Experience Portal R8.1 and Avaya Session Border Controller for Enterprise R10.1 to the Vodafone Germany SIP Trunk Service.

The Vodafone Germany 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**.

## 12. Additional References

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

- [1] *Deploying Avaya Appliance Virtualization Platform*, Release 10.1, Jan 2023
- [2] *Upgrading Avaya Aura® applications*, Release 10.1, Jan 2023
- [3] *Deploying Avaya Aura® applications from System Manager*, Release 10.1, Jan 2023
- [4] *Deploying Avaya Aura® Communication Manager*, Release 10.1, Jan 2023
- [5] *Administering Avaya Aura® Communication Manager*, Release 10.1, Feb 2023
- [6] *Upgrading Avaya Aura® Communication Manager*, Release 10.1, Jan 2023
- [7] *Deploying Avaya Aura® System Manager*, Release 10.1, Feb 2023
- [8] *Upgrading Avaya Aura® System Manager*, Release 10.1, Feb 2023
- [9] *Administering Avaya Aura® System Manager*, Release 10.1, Feb 2023
- [10] *Deploying Avaya Aura® Session Manager*, Release 10.1 Feb 2023
- [11] *Upgrading Avaya Aura® Session Manager*, Release 10.1, Feb 2023
- [12] *Administering Avaya Aura® Session Manager*, Release 10.1, Feb 2023
- [13] *Deploying Avaya Session Border Controller for Enterprise*, Release 10.1, Jan 2023
- [14] *Upgrading Avaya Session Border Controller for Enterprise*, Release 10.1 Jan 2023
- [15] *Administering Avaya Session Border Controller for Enterprise*, Release 10.1, Jan 2023
- [16] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>

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