



DevConnect Program

Application Notes for Configuring Avaya Aura® Communication Manager R10.1, Avaya Aura® Session Manager R10.1, Avaya Experience Portal R8.1 and Avaya Session Border Controller R10.1 to support VTX Connect PBX-IP SIP Trunk Service - Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between the VTX Connect PBX-IP SIP 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 R10.1.

The VTX Connect PBX-IP SIP Platform provides PSTN access via a SIP trunk connected to the VTX Voice over Internet Protocol (VoIP) network as an alternative to legacy analogue or digital trunks. VTX is a member of the DevConnect Service Provider program.

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

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program.

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1. Introduction

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between the VTX Connect PBX-IP SIP Service 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 R8.1 (Experience Portal) and Avaya Session Border Controller R10.1 (Avaya SBC).

Customers using this Avaya SIP-enabled enterprise solution with the VTX Connect PBX-IP SIP 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 SBC. The enterprise site was configured to connect to the VTX 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 VTX SIP platform, calls made to SIP, H.323, Digital and Analogue telephones at the enterprise.
- Outgoing calls from the enterprise site completed via the VTX SIP platform to PSTN destinations, calls made from SIP, H.323, Digital and Analogue telephones.
- Incoming and Outgoing PSTN calls to/from Avaya one-X® Communicator and Avaya Workplace Client for Windows soft phones.
- Calls using G.711A and G.729 codecs.
- Fax calls to/from Client a group 3 fax machine to a PSTN-connected fax machine using T.38 transmission.
- 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 SBC, 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 VTX requiring Avaya response and sent by Avaya requiring VTX response.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the VTX Connect PBX-IP SIP Trunking Service with the following observations:

- It was observed during testing that certain Call Forwarding All Calls and Call Forwarding No Answer scenarios from Communication Manager to a number of IP-PBX services hosted on the VTX SIP platform were failing. As a workaround, direct media setting “Initial IP-IP Direct Media” was set to “n” in the Communication Manager SIP Signalling Group as per **Section 5.5**. Once Direct Media was disabled on Communication Manager, all Call Forwarding All Calls and Call Forwarding No Answer calls terminated successfully on the multiple IP-PBX services hosted within the VTX SIP platform.
- 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 VTX SIP trunk. In order to complete Blind and Consultative transfers successfully within Experience Portal, REFER Handling needs to be enabled on the VTX Server Interworking profile (**Section 8.5.2**) on the Avaya SBC. When the REFER message comes from an Avaya enterprise element such as Experience Portal, the Avaya SBC translates that REFER into a reINVITE which will then be routed towards the trunk server (i.e., VTX) based on the trunk server interworking profile configuration.
- No Inbound Toll-Free access available for test.
- No Emergency Services test call booked with 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 VTX Services products please contact the VTX Services support team:
<https://www.vtx.ch/fr/support/contact>
<https://www.vtx.ch/de/support/kontakt>

3. Reference Configuration

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

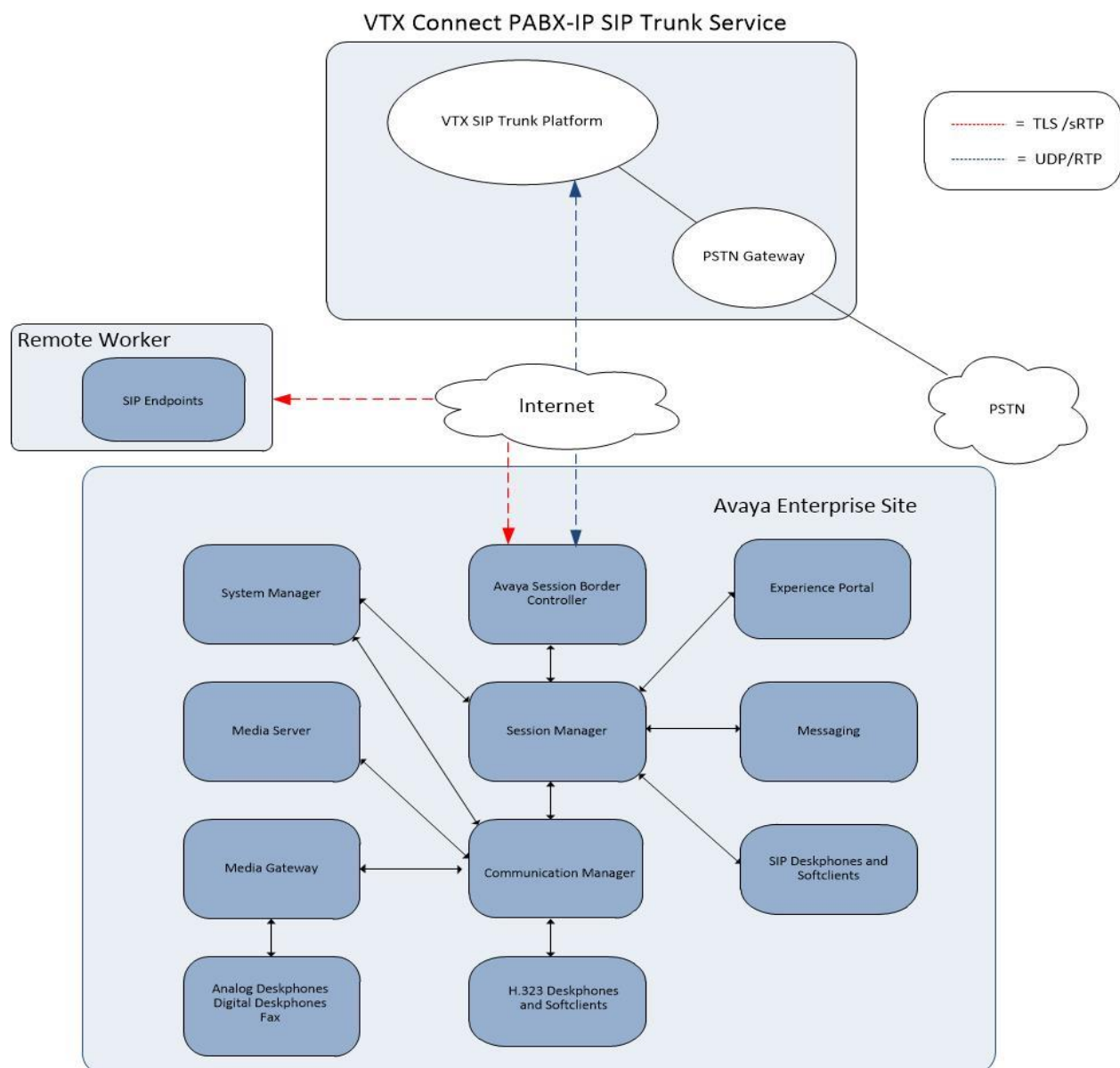


Figure 1: VTX Connect PBX-IP SIP Service to Avaya Enterprise

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Avaya Aura® System Manager	10.1.3.1 Build No. – 10.1.0.0.537353 Software Update Revision No: 10.1.3.1.0716149 – Service Pack 1
Avaya Aura® Session Manager	10.1.3.1.1013103
Avaya Aura® Communication Manager	10.1.3.1 Service Pack 1 - 27937
Avaya Session Border Controller	10.1.2.0-64-23285
Avaya Experience Portal	8.1.1
Avaya G430 Media Gateway	42.24.0
Avaya Aura® Media Server	v.10.1.0.125
Avaya Aura® Messaging	7.2 SP3
Avaya 1600 IP Deskphone (H.323)	1.3.12
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)	4.1.1.0
Avaya one-X® Communicator (H.323 & SIP)	6.2.14.18 -SP14-Patch 9
Avaya Workplace for Windows (SIP)	3.34.1.
Avaya 1408 Digital Telephone	R48
Analogue Handset	N/A.
VTX	
VTX CONNECT PBX IP	VTX CONNECT PBX-IP <X> (<X>: depending on number of channels: 4, 8, 15, 30, 60, etc.)
Platform: Communi5, VAS C5	7.4 udp7
SBC: Audiocodes	Mediant VE SBC, Version 7.40A.250.541

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 VTX Connect PBX-IP SIP Trunking Service. For incoming calls, Session Manager receives SIP messages from the Avaya SBC 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. The Session Manager directs the outbound SIP messages to the Avaya SBC at the enterprise site that then sends the SIP messages to the VTX 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 VTX SIP Trunk network, and any other SIP trunks used.

display system-parameters customer-options		Page	2 of	11
OPTIONAL FEATURES				
IP PORT CAPACITIES		USED		
Maximum Administered H.323 Trunks:		12000	0	
Maximum Concurrently Registered IP Stations:		18000	3	
Maximum Administered Remote Office Trunks:		12000	0	
Maximum Concurrently Registered Remote Office Stations:		18000	0	
Maximum Concurrently Registered IP eCons:		414	0	
Max Concur Registered Unauthenticated H.323 Stations:		100	0	
Maximum Video Capable Stations:		41000	0	
Maximum Video Capable IP Softphones:		18000	0	
Maximum Administered SIP Trunks:		24000	10	
Maximum Administered Ad-hoc Video Conferencing Ports:		24000	0	
Maximum Number of DS1 Boards with Echo Cancellation:		522	0	
Maximum TN2501 VAL Boards:		128	0	
Maximum Media Gateway VAL Sources:		250	1	
Maximum TN2602 Boards with 80 VoIP Channels:		128	0	
Maximum TN2602 Boards with 320 VoIP Channels:		128	0	
Maximum Number of Expanded Meet-me Conference Ports:		300	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	
Session_Manager	10.10.3.42	
default	0.0.0.0	
procr	10.10.3.44	
procr6	::	

5.3. Administer IP Network Region

Use the **change ip-network-region 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 SBC.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this case, codec set **1** is used.
- The rest of the fields can be left at default values.

```
change ip-network-region 1                                     Page 1 of 20
                                                                IP NETWORK REGION
Region: 1
Location:               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 VTX were configured, namely **G.711A** and **G.729**.

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

change ip-codec-set 1

Page1 of 2

IP MEDIA PARAMETERS

Codec Set: 1

Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size(ms)
1: G.711A	n	2	20
2: G.729	n	2	20

Media Encryption

Encrypted SRTCP: enforce-unenc-srtcp

1: srtp-aescm128-hmac80

2: none

VTX SIP Trunk supports T.38 for transmission of fax. Navigate to **Page 2** and define fax properties as follows:

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

change ip-codec-set 2

Page2 of 2

IP MEDIA PARAMETERS

Allow Direct-IP Multimedia? n

	Mode	Redun- dancy	Packet Size (ms)
FAX	t.38-standard	0	
Modem	off	0	
TDD/TTY	US	3	
H.323 Clear-channel	n	0	
SIP 64K Data	n	0	20

Media Connection IP Address Type Preferences

1: IPv4

2:

5.5. Administer SIP Signaling Groups

This signalling group (and trunk group) will be used for inbound and outbound PSTN calls to the VTX SIP platform. 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** as per **Section 2.2**.
- Set **H.323 Station Outgoing Direct Media** to **y**.

The default values for the other fields may be used.

add signaling-group 1		Page 1 of 2
SIGNALING GROUP		
Group Number: 1	Group Type: sip	
IMS Enabled? n	Transport Method: 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? y	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		
Dial Access? n	Night Service:		
Queue Length: 0			
Service Type: public-ntwrk	Auth Code? n		
		Member Assignment Method: auto	
		Signaling Group: 1	
		Number of Members: 10	

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

add trunk-group 1		Page 2 of 21	
Group Type: sip			
TRUNK PARAMETERS			
Unicode Name: auto			
		Redirect On OPTIM Failure: 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**. This allows delivery of CLI in format of E.164 with leading “+”. Also, set the **Hold/Unhold Notifications** to **n**.

add trunk-group 1	Page 3 of 21
TRUNK FEATURES	
ACA Assignment? n	Measured: none
	Maintenance Tests? y
Suppress # Outpulsing? n	
Numbering Format: public	UUI Treatment: service-provider
	Replace Restricted Numbers? n
	Replace Unavailable Numbers? n
	Hold/Unhold Notifications? n
	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 Redirection** 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 VTX.
- 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	
	Mark Users as Phone? y
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n	
Send Transferring Party Information? n	
Network Call Redirection? n	
	Send Diversion Header? y
	Support Request History? n
	Telephone Event Payload Type: 101
	Convert 180 to 183 for Early Media? n
Always Use re-INVITE for Display Updates? y	
Identity for Calling Party Display: P-Asserted-Identity	
Block Sending Calling Party Location in INVITE? 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					
Ext	Trk	CPN	CPN	Total	
Len	Code	Grp(s)	Prefix	Len	
4	6100	1	4121xxxxx06	11	Total Administered: 4
4	6102	1	4121xxxxx07	11	Maximum Entries: 240
4	6010	1	4121xxxxx08	11	
4	6030	1	4121xxxxx09	11	Note: If an entry applies to
4	6104	1	4121xxxxx10	11	a SIP connection to Avaya
					Aura(R) Session Manager,
					the resulting number must
					be a complete E.164 number.
					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 VTX 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**.

change feature-access-codes		Page 1 of 10
FEATURE ACCESS CODE (FAC)		
Abbreviated Dialing List1 Access Code:		
Abbreviated Dialing List2 Access Code:		
Abbreviated Dialing List3 Access Code:		
Abbreviated Dial - Prgm Group List Access Code:		
Announcement Access Code: *69		
Answer Back Access Code:		
Attendant Access Code:		
Auto Alternate Routing (AAR) Access Code: 7		
Auto Route Selection (ARS) - Access Code 1: 9		Access Code 2:

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

change ars analysis 0							Page 1 of 2
ARS DIGIT ANALYSIS TABLE							
Location: all							Percent Full: 0
	Dialed String	Total Min Max	Route Pattern	Call Type	Node Num	ANI Req'd	
	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 **pub-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					pub-unk	none							
2:	y	y	y	y	y	n	n	rest						none							
3:	y	y	y	y	y	n	n	rest						none							
4:	y	y	y	y	y	n	n	rest						none							
5:	y	y	y	y	y	n	n	rest						none							
6:	y	y	y	y	y	n	n	rest						none							

5.9. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DID calls to the proper Communication Manager extension(s). The incoming digits sent in the INVITE message from VTX can be manipulated as necessary to route calls to the desired extension. In the examples used in the compliance testing, the incoming DID numbers provided by VTX correlate to the internal extensions assigned within Communication Manager. The entries displayed below translates incoming DID numbers **+4121xxxxxx** to a 4 digit extension by deleting all of the incoming digits and inserting an extension. Public DID numbers have been masked for security purposes.

change inc-call-handling-trmt trunk-group 1				Page	1 of	3
INCOMING CALL HANDLING TREATMENT						
Service/ Feature	Number Len	Number Digits	Del Insert			
public-ntwrk	12	+4121xxxxx06	all 6010			
public-ntwrk	12	+4121xxxxx07	all 6102			
public-ntwrk	12	+4121xxxxx08	all 6020			
public-ntwrk	12	+4121xxxxx09	all 6030			
public-ntwrk	12	+4121xxxxx10	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** (e.g., 9) if required by the routing configuration.
- For the **Phone Number** enter the phone that will also be called (e.g. **0035389434xxxx**).
- Set the **Trunk Selection** to **ARS**.
- 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

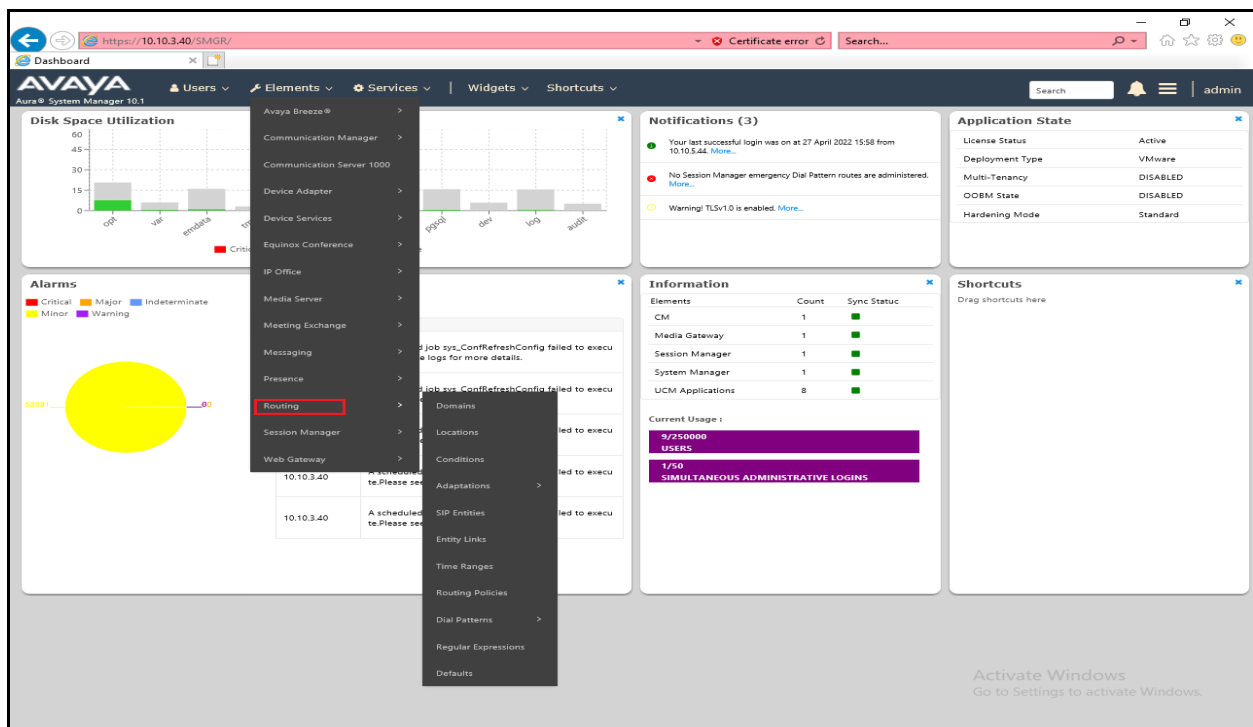
This section provides the procedures for configuring Session Manager. Session Manager is configured via System Manager. The procedures include the following areas:

- 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.

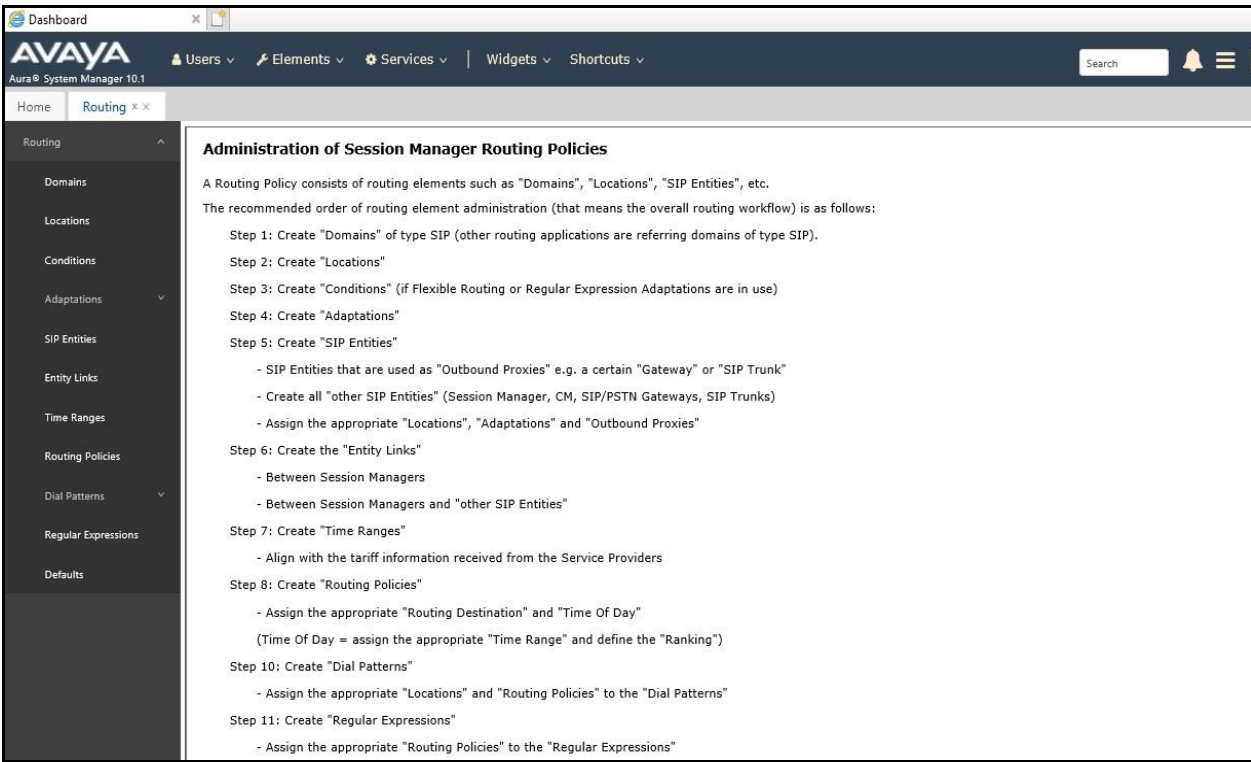
It may not be necessary to create all the items above when creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, locations, SIP entities, and Session 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 **Administration of Session Manager Routing Policies** screen.

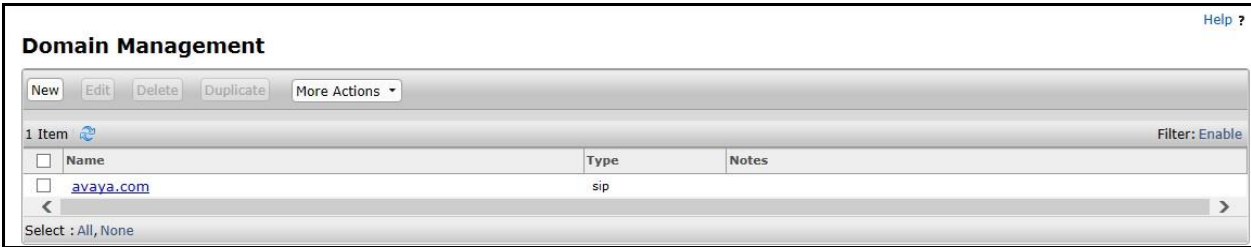


6.2. Administer SIP Domain

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 page for a location named 'Session Manager'. The page is divided into several sections:

- General:** Contains fields for 'Name' (set to 'Session Manager') and 'Notes'.
- Dial Plan Transparency in Survivable Mode:** Includes an 'Enabled' checkbox (unchecked), a 'Listed Directory Number' field, and an 'Associated CM SIP Entity' field.
- Overall Managed Bandwidth:** Features a 'Managed Bandwidth Units' dropdown (set to 'Kbit/sec'), 'Total Bandwidth' and 'Multimedia Bandwidth' input fields, and an 'Audio Calls Can Take Multimedia Bandwidth' checkbox (checked).
- Per-Call Bandwidth Parameters:** Includes '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).

Buttons for 'Commit' and 'Cancel' are located in the top right corner.

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. On the right, there is 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 main sections: "General", "Dial Plan Transparency in Survivable Mode", and "Overall Managed Bandwidth".

- General**: Includes a required field for "Name" (set to "Communication Manager") and a "Notes" field.
- Dial Plan Transparency in Survivable Mode**: Includes an "Enabled" checkbox (unchecked), a "Listed Directory Number" field, and an "Associated CM SIP Entity" field.
- Overall Managed Bandwidth**: Includes a "Managed Bandwidth Units" dropdown (set to "Kbit/sec"), "Total Bandwidth" and "Multimedia Bandwidth" fields, 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 SBC**. This location is assigned to the SIP Entity called Avaya SBC 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. 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 **DigitConversionAdapter** 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 either Avaya proprietary unnecessary for non-Avaya elements

For the compliance test, an Adaptation named “**VTX**” was created to block the following headers from outbound messages, before they were forwarded to the Avaya SBC: 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 **VTX**.
- **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**.

Adaptation Details Commit Cancel Help ?

General

* **Adaptation Name:** VTX

Notes:

* **Module Name:** DigitConversionAdapter

Type: digit

State: enabled

Module Parameter Type: Name-Value Parameter

Add		Remove	
<input type="checkbox"/>	Name		Value
<input type="checkbox"/>	eRHdrs		"P-AV-Message-Id, P-Charging-Vector, P-Location, Endpoint-View, P-Conference, Alert-Info, Correlation-ID, Accept-Language"
<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

<input type="checkbox"/>	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

Commit Cancel

This will ensure any outgoing national numbers matching 0 will have a prefix 0 inserted and any outgoing international numbers matching 00 will have prefix's 00 deleted and have + inserted being converted to E.164 format before being forwarded to the Avaya SBC.

6.5. Administer SIP Entities

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

Under **General**:

- In the **Name** field enter an informative name.
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signalling interface on the connecting system.
- In the **Type** field use **Session Manager** for a Session Manager SIP Entity, **CM** for a Communication Manager SIP Entity and **SIP Trunk** for the Avaya SBC 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 SBC 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

Commit

Cancel

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:

Add

Remove

3 Items

Filter: Enable

	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. 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

Commit

Cancel

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 SIP Entity

The following screen shows the SIP Entity for the Avaya SBC used for PSTN destinations. The **FQDN or IP Address** field is set to the IP address of the Avaya SBC 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 SBC in **Section 6.3** and the **Time Zone** to the appropriate time zone.

SIP Entity Details

CommitCancel

General

* Name: Avaya SBC

* FQDN or IP Address: 10.10.3.35

Type: SIP Trunk

Notes:

Adaptation: VTX

Location: Avaya SBC

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

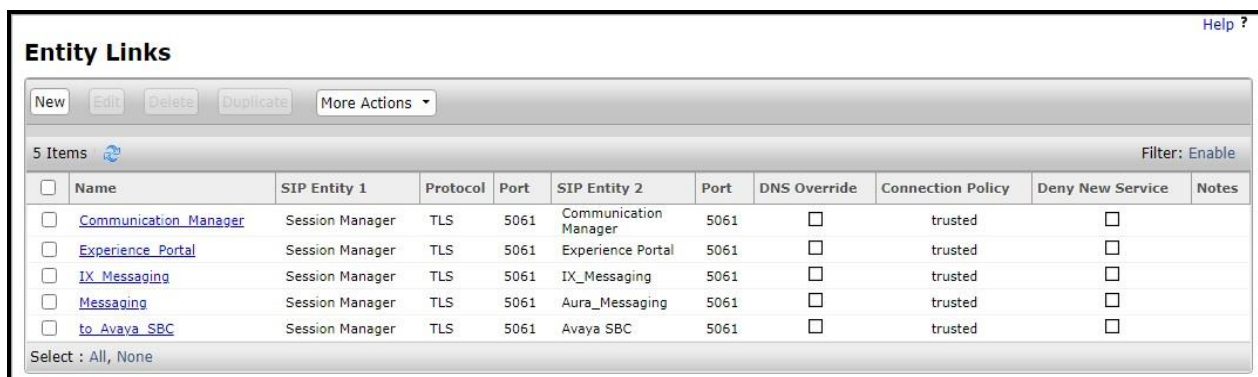
Loop Detection Interval (in msec): 200

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"/>	Communication Manager	Session Manager	TLS	5061	Communication Manager	5061	<input type="checkbox"/>	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	Experience Portal	Session Manager	TLS	5061	Experience Portal	5061	<input type="checkbox"/>	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	IX_Messaging	Session Manager	TLS	5061	IX_Messaging	5061	<input type="checkbox"/>	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	Messaging	Session Manager	TLS	5061	Aura_Messaging	5061	<input type="checkbox"/>	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	to_Avaya_SBC	Session Manager	TLS	5061	Avaya SBC	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 VTX SIP Trunk to Communication Manager.

Routing Policy Details

CommitCancel

Help ?

General

* Name: to_Communication_Manager

Disabled: ☐

* Retries: 0

Notes: Inbound calls to CM.

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Communication Manager	10.10.3.44	CM	

Time of Day

AddRemoveView 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 outbound calls from Communication Manager via Avaya SBC to the VTX SIP trunk.

Routing Policy Details

CommitCancel

Help ?

General

* Name: to_Avaya_SBC

Disabled: ☐

* Retries: 0

Notes: Outbound calls to SP via ASBC.

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Avaya SBC	10.10.3.35	SIP Trunk	

Time of Day

AddRemoveView 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

CommitCancel

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

AddRemoveView 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 VTX SIP Trunk.

Dial Pattern Details

CommitCancelHelp ?

General

* Pattern: 00353

* Min: 5

* Max: 16

Emergency Call: ☐

SIP Domain: avaya.com

Notes:

Originating Locations and Routing Policies

AddRemove

1 Item

Filter: Enable

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Communication Manager		to_Avaya_SBC	0	<input type="checkbox"/>	Avaya SBC	Outbound calls to SP via Avaya SBC

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

CommitCancel

Help ?

General

* Pattern: +41

* Min: 3

* Max: 15

Emergency Call: ☐

SIP Domain: avaya.com

Notes:

Originating Locations and Routing Policies

AddRemove

1 Item

Filter: Enable

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

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 Experience Portal.

Dial Pattern Details

CommitCancel

Help ?

General

* Pattern: +4989xxxxxxx15

* Min: 14

* Max: 14

Emergency Call: ☐

SIP Domain: avaya.com

Notes:

Originating Locations and Routing Policies

AddRemove

1 Item

Filter: Enable

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Avaya SBC		to_Experience_Portal	0	<input type="checkbox"/>	Experience Portal	Inbound calls to Experience Portal

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 [16] 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 VTX 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.

Avaya Experience Portal 8.1.1 (ExperiencePortal)

You are here: Home

Avaya Experience Portal Manager

Avaya Experience Portal Manager (EPM) is the consolidated web-based application for administering Experience Portal. Through the EPM interface you can configure Experience Portal, check of an Experience Portal component, and generate reports related to system operation.

Installed Components

Media Processing Platform
Media Processing Platform (MPP) is an Avaya media processing server. When an MPP receives a call from a PBX, it invokes a VoiceXML (or CCXML) application on an application server. It then communicates with ASR and TTS servers as necessary to process the call.

Email Service
Email Service is an Experience Portal feature which provides e-mail capabilities.

HTML Service
HTML Service is an Experience Portal feature which supports web applications with HTML5 capabilities. It includes support for browser based services for mobile devices.

SMS Service
SMS Service is an Experience Portal feature which provides SMS capabilities.

Legal Notice

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REVISED: June 1st, 2020

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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

Product	Count
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:

SIP Domain:

P-Asserted-Identity:

Maximum Redirection Attempts:

Consultative Transfer: ☒ INVITE with REPLACES ☐ REFER

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

SIP Timers

T1: milliseconds

T2: milliseconds

B and F: milliseconds

Call Capacity

Maximum Simultaneous Calls:

- ☒ 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

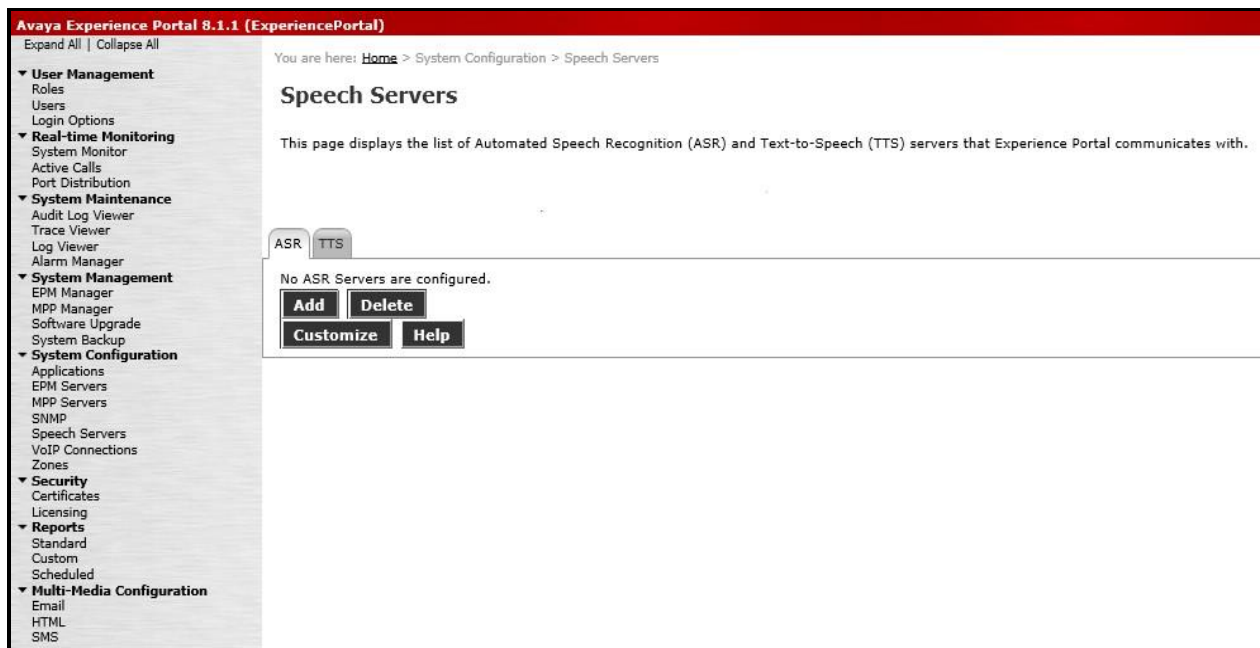
Help

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:

Mutual Certificate Authentication: ☐ Yes ☒ No

Basic Authentication: ☐ Yes ☒ No

ASR Speech Servers ▾

Engine Types	Selected Engine Types
ASR: <input type="text" value="<None>"/>	<input type="text" value="<None>"/>

TTS Speech Servers ▾

TTS:

Application Launch ▾

☒ Inbound ☐ Inbound Default ☐ Outbound

☒ Number ☐ Number Range ☐ URI

Called Number:

SIP Header Source:

Speech Parameters ▸

Reporting Parameters ▸

Advanced Parameters ▸

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

▼ User Management

Roles

Users

Login Options

▼ Real-time Monitoring

System Monitor

Active Calls

Port Distribution

▼ System Maintenance

Audit Log Viewer

Trace Viewer

Log Viewer

Alarm Manager

▼ System Management

EPM Manager

MPP Manager

Software Upgrade

System Backup

▼ System Configuration

Applications

EPM Servers

MPP Servers

SNMP

Speech Servers

VoIP Connections

Zones

▼ Security

Certificates

Licensing

▼ Reports

Standard

Custom

Scheduled

▼ Multi-Media Configuration

Email

HTML

SMS

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.

<input type="checkbox"/>	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: 952c116c181b7815
Signature Algorithm: SHA256withRSA
Valid from: 28 February 2019 13:17:17 GMT until 28 February 2029 13:17:17 GMT
Certificate Fingerprints
MD5: 8b:17:0c:92:49:ef:64:3d:f6:b2:60:6a:bb:f5:09:69
SHA: 9a:90:a4:2c:48:21:46:ac:e4:18:e0:35:b0:e6:c1:42:3c:2b:d1:be
SHA-256: 09:cb:da:73:0d:e6:ae:02:98:80:eb:92:56:0c:15:17:b2:f6:9e:f6:f9:2e:90:6d: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: 20 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

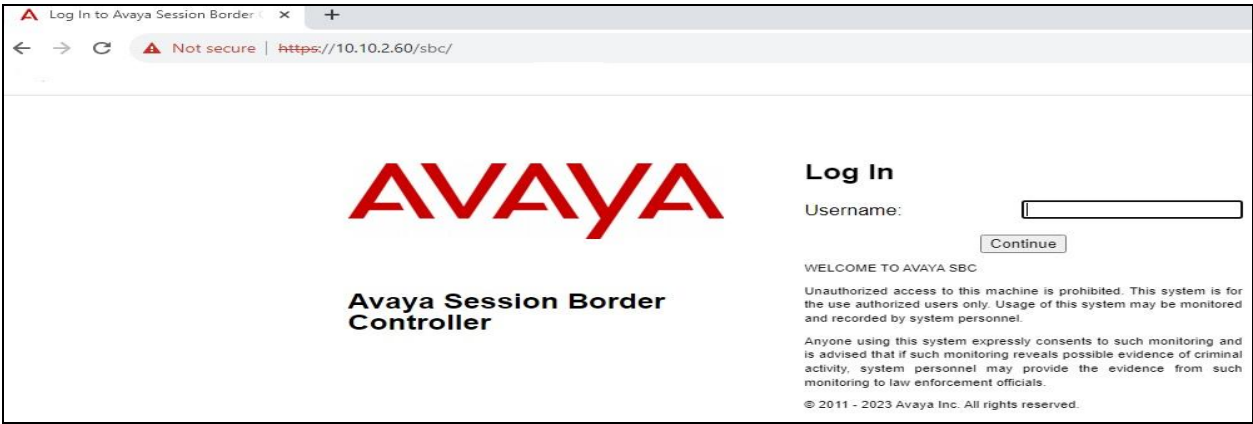
Help

8. Configure Avaya Session Border Controller

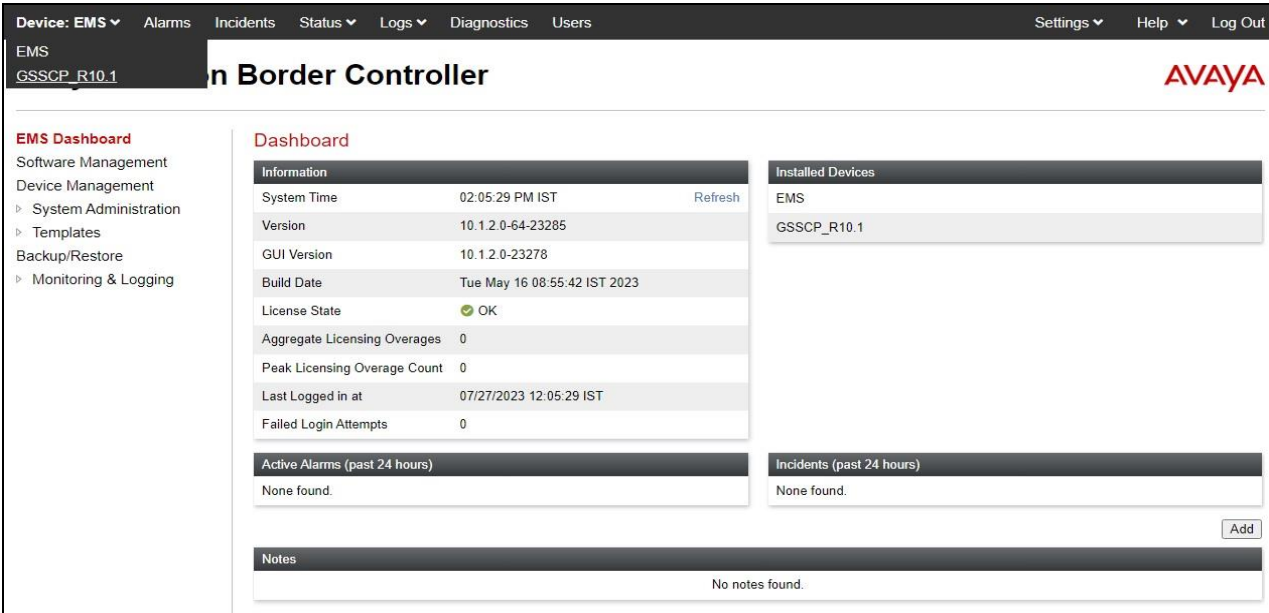
This section describes the configuration of the Session Border Controller (Avaya SBC). The Avaya SBC provides security and manipulation of signalling to deliver an interface to the Service Provider’s SIP Trunk that is standard where possible and adapted to the Service Provider’s SIP implementation where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

8.1. Access Avaya Session Border Controller

Access the Avaya SBC 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 SBC.



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).

Device: GSSCP_R10.1 ▾ Alarms Incidents Status ▾ Logs ▾ Diagnostics Users Settings ▾ Help ▾ Log Out

Avaya Session Border Controller

AVAYA

EMS Dashboard
Software Management
Device Management
Backup/Restore
System Parameters
Configuration Profiles
Services
Domain Policies
TLS Management
Network & Flows
DMZ Services
Monitoring & Logging

Device Management

Devices Updates Licensing Key Bundles License Compliance

Device Name	Management IP	Version	Status							
GSSCP_R10.1	10.10.2.60	10.1.2.0-64-23285	Commissioned	Reboot	Shutdown	Restart Application	View	Edit	Uninstall	

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

Alarms Incidents Status ▾ Logs ▾ Diagnostics Users Settings ▾ Help ▾ Log Out

System Information: GSSCP_R10.1

General Configuration

Appliance Name: GSSCP_R10.1
Box Type: SIP
Deployment Mode: Proxy
HA Mode: No

Management IP(s)

IP #1 (IPv4): 10.10.2.60

DNS Configuration

Primary DNS: 8.8.8.8
Secondary DNS: 8.8.4.4
DNS Location: DMZ
DNS Client IP: 192.168.122.57

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
192.168.122.57	192.168.122.57	255.255.255.0	192.168.122.9	B1

8.2. Define Network Management

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

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 SBC 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' configuration window. At the top, a warning message states: '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 this, there are four input fields: 'Name' (A1_Internal), 'Default Gateway' (10.10.3.1), 'Network Prefix or Subnet Mask' (255.255.255.0), and 'Interface' (A1). An 'Add' button is located to the right of the 'Interface' field. Below these fields is a table with four columns: 'IP Address', 'Public IP', 'Gateway Override', and 'Passthrough'. The first row contains the values '10.10.3.35', 'Use IP Address', 'Use Default', and an unchecked checkbox. A 'Delete' button is to the right of the checkbox. At the bottom of the window is a 'Finish' button.

IP Address	Public IP	Gateway Override	Passthrough
10.10.3.35	Use IP Address	Use Default	<input type="checkbox"/>

Click on **Add** to define the internal interfaces or Edit if it was defined during installation of the Avaya SBC. 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 SBC 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.

Edit Network

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

B1_External

Default Gateway

192.168.122.9

Network Prefix or Subnet Mask

255.255.255.0

Interface

B1

Add

IP Address

192.168.122.57

Public IP

Use IP Address

Gateway Override

Use Default

Passthrough

☐

Delete

Finish

The following screenshot shows the completed Network Management configuration:

Network Management

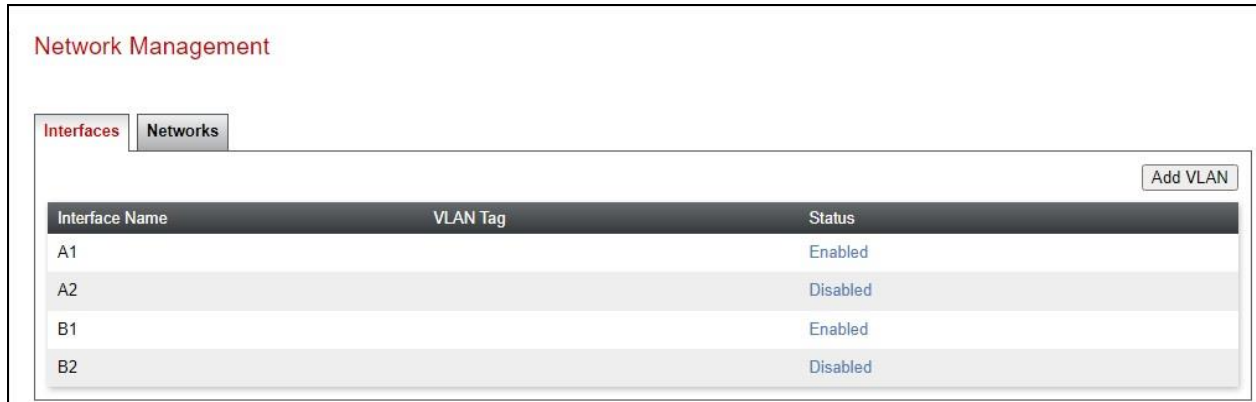
Interfaces

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	192.168.122.57	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.



The screenshot shows a web interface titled "Network Management". It has two tabs: "Interfaces" (selected) and "Networks". In the top right corner of the interface is a button labeled "Add VLAN". Below the tabs is a table with three columns: "Interface Name", "VLAN Tag", and "Status". The table contains four rows of data:

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

Note: to ensure that the Avaya SBC 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 SBC. 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 SBC, navigate to **TLS Management** → **Certificates**:

- Verify that an Avaya SBC identity certificate (**asbce60.crt**) is present under **Installed Certificates**.
- Verify that certificate authority root certificate (**SystemManagerCA.crt**) is present under **Installed CA certificates**.
- Verify that private key associated with the identity certificate (**asbce60.key**) is present under **Installed Keys**.

The screenshot displays the 'Certificates' management page. At the top right, there are 'Install' and 'Generate CSR' buttons. The main content area is divided into several sections:

- Installed Certificates:** A table with one entry, 'asbce60.crt', which has 'View' and 'Delete' links.
- Installed CA Certificates:** A table listing several certificates: 'avayaaitrootca2.pem', 'entrust_g2_ca.cer', 'AvayaDeviceEnrollmentCAchain.crt', 'DigiCertGlobalRootG2.crt', 'Mnet.crt', and 'SystemManagerCA.crt'. Each entry has 'View' and 'Delete' links.
- Installed Certificate Revocation Lists:** A message stating 'No certificate revocation lists have been installed.'
- Installed Certificate Signing Requests:** A table with one entry, 'asbce60.avaya.com.req', which has a 'Delete' link.
- Installed Keys:** A table with one entry, 'asbce60.key', which has a 'Delete' link.

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 **asbce60.crt** 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 shows the 'Client Profiles: GSSCP_Client' configuration window. On the left, a sidebar lists 'Client Profiles' with 'GSSCP_Client' selected. The main area contains a 'Client Profile' tab with a description field. Below this are several sections: 'TLS Profile' with fields for Profile Name (GSSCP_Client), Certificate (asbce60.crt), and SNI (Enabled checkbox); 'Certificate Verification' with fields for Peer Verification (Required), Peer Certificate Authorities (SystemManagerCA.crt), Peer Certificate Revocation Lists (---), Verification Depth (1), and Extended Hostname Verification (checkbox); 'Renegotiation Parameters' with fields for Renegotiation Time (0) and Renegotiation Byte Count (0); and 'Handshake Options' with fields for Version (TLS 1.3 and TLS 1.2 checked), Ciphers (Default selected), and a Value field set to DEFAULT:SHA. An 'Edit' button is at the bottom right.

Client Profile	
Click here to add a description.	
TLS Profile	
Profile Name	GSSCP_Client
Certificate	asbce60.crt
SNI	<input checked="" type="checkbox"/> Enabled
Certificate Verification	
Peer Verification	Required
Peer Certificate Authorities	SystemManagerCA.crt
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 1.3 <input checked="" type="checkbox"/> TLS 1.2
Ciphers	<input checked="" type="radio"/> Default <input type="radio"/> FIPS <input type="radio"/> Custom
Value	DEFAULT:SHA

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 **asbce60.crt** 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 'Server Profiles' management interface. On the left, a sidebar shows 'Server Profiles' with a sub-item 'GSSCP_Server'. The main area is titled 'Server Profiles: GSSCP_Server' and includes 'Add' and 'Delete' buttons. A blue bar at the top of the main area says 'Click here to add a description.' Below this, the 'Server Profile' tab is active, showing a configuration form for the 'GSSCP_Server' profile. The form is organized into sections: 'TLS Profile', 'Certificate Verification', 'Renegotiation Parameters', and 'Handshake Options'. The 'TLS Profile' section includes 'Profile Name' (GSSCP_Server), 'Certificate' (asbce60.crt), and 'SNI Options' (None). The 'Certificate Verification' section includes 'Peer Verification' (Optional), 'Peer Certificate Authorities' (SystemManagerCA.crt), 'Peer Certificate Revocation Lists' (---), 'Verification Depth' (1), and 'Extended Hostname Verification' (unchecked). The 'Renegotiation Parameters' section includes 'Renegotiation Time' (0) and 'Renegotiation Byte Count' (0). The 'Handshake Options' section includes 'Version' (TLS 1.3 and TLS 1.2 checked), 'Ciphers' (Default selected, FIPS and Custom unselected), and 'Value' (DEFAULT:SHA). An 'Edit' button is located at the bottom right of the form.

TLS Profile	
Profile Name	GSSCP_Server
Certificate	asbce60.crt
SNI Options	None

Certificate Verification	
Peer Verification	Optional
Peer Certificate Authorities	SystemManagerCA.crt
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 1.3 <input checked="" type="checkbox"/> TLS 1.2
Ciphers	<input checked="" type="radio"/> Default <input type="radio"/> FIPS <input type="radio"/> Custom
Value	DEFAULT:SHA

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 SBC, 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 VTX SIP Trunk.
- Click **Finish**.

Signaling Interface

Signaling Interface

Add

Name	Signaling IP Network	TCP Port	UDP Port	TLS Port	TLS Profile	
Signaling_External	192.168.122.57 B1_External (B1, VLAN 0)	---	5060	---	None	Edit Delete
Signaling_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 SBC, 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	192.168.122.57 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 SBC. In this case, VTX 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**.
- Check **T.38 Support**.
- Uncheck **SIPS Required**.
- All other options on the **General** Tab can be left at default.

The screenshot displays the 'General' configuration tab for a Server Interworking profile. The interface includes a title bar 'General' and a list of configuration options with their respective settings:

Configuration Option	Value / Setting
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 type="checkbox"/>
URI Group	None (dropdown menu)
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 checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543
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**.

The screenshot displays the 'Advanced' configuration tab with the following settings:

- 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:** Avaya (dropdown)
- Diversion Manipulation:** ☐
- Diversion Condition:** None (dropdown)
- Diversion Header URI:** (empty text field)
- Has Remote SBC:** ☒
- Route Response on Via Port:** ☐
- Relay INVITE Replace for SIPREC:** ☐
- MOBX Re-INVITE Handling:** ☐
- NATing for 301/302 Redirection:** ☒

DTMF (Section Header)

- DTMF Support:** ☒ None>, ☐ SIP Notify>, ☐ RFC 2833 Relay & SIP Notify>, ☐ SIP Info>, ☐ RFC 2833 Relay & SIP Info>, ☐ Inband>

Finish (button)

8.5.2. Server Interworking – VTX

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 **VTX** and click **Next** (Not Shown).
- Check **Hold Support** = **None**.
- Check **Refer Handling** as per **Section 2.2**.
- Check **T.38 Support**.
- 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 checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543
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. Define Servers

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

8.6.1. 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.

SIP Server Profile - General

Server Type can not be changed while this SIP Server Profile is associated to a Server Flow.

Server Type: Call Server

SIP Domain:

DNS Query Type: NONE/A

TLS Client Profile: GSSCP_Client

Add

IP Address / FQDN	Port	Transport	Whitelist
10.10.3.42	5061	TLS	<input type="checkbox"/>

Delete

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:

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	<input type="text"/>
TLS Failover Port	<input type="text"/>
Tolerant	<input type="checkbox"/>
URI Group	None ▼
NG911 Support	<input type="checkbox"/>

At the bottom center of the window is a button labeled "Finish".

8.6.2. Server Configuration – VTX

To define the VTX 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 **s1.xxxxxx.trk.ipvoip.ch** (VTX SIP Platform).
- For **Port**, enter **5060**.
- For **Transport**, select **UDP**.
- Click on **Next** (not shown).

SIP Server Profile - General

Server Type can not be changed while this SIP Server Profile is associated to a Server Flow.

Server Type: Trunk Server

SIP Domain:

DNS Query Type: NONE/A

TLS Client Profile: None

Add

IP Address / FQDN / CIDR Range	Port	Transport	Whitelist
s1.xxxxxx.trk.ipvoip.ch	5060	UDP	<input type="checkbox"/>

Delete

In the new Authentication window that appears, enter the following values as VTX require authentication to connect to their network:

- **Enabled Authentication:** Checked
- **User Name:** Enter username provided by the Service Provider.
- **Realm:** Enter realm details provided by the Service Provider or leave blank to be detected by the server challenge.
- **Password** Enter password provided by the Service Provider.
- **Confirm Password** Re-enter password provided by the Service Provider.

Click **Next** to continue (not shown).



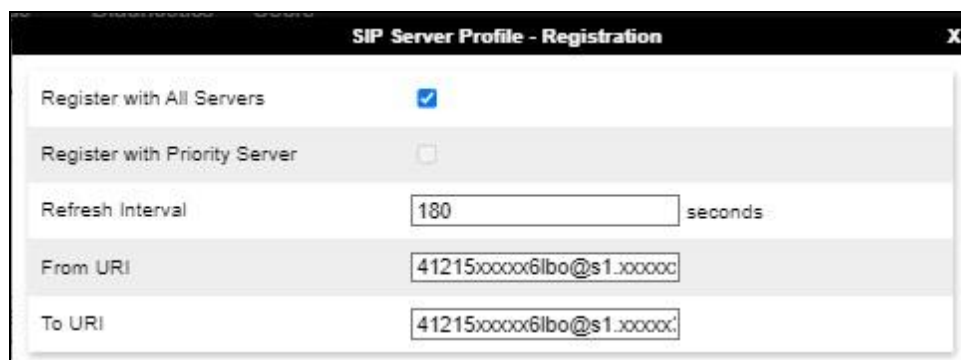
The screenshot shows a window titled "SIP Server Profile - Authentication". It contains the following fields and controls:

- Enable Authentication:** A checkbox that is checked.
- User Name:** A text input field containing "412156xxxxlbo".
- Realm:** A text input field with the placeholder text "(Leave blank to detect from server challenge)".
- Password:** A text input field with the placeholder text "(Leave blank to keep existing password)".
- Confirm Password:** A text input field.

In the new Registration window that appears, enter the following values.

- **Register with Priority Server:** Check.
- **Refresh Interval** Choose the desired frequency in seconds the Avaya SBC will send SIP REGISTERS.
- **From URI:** Enter an URI to be sent in the FROM header for SIP REGISTERS.
- **TO URI:** Enter an URI to be sent in the TO header for SIP REGISTERS.

Click **Next** to continue (not shown).



The screenshot shows a window titled "SIP Server Profile - Registration". It contains the following fields and controls:

- Register with All Servers:** A checkbox that is checked.
- Register with Priority Server:** An unchecked checkbox.
- Refresh Interval:** A text input field containing "180" followed by the label "seconds".
- From URI:** A text input field containing "41215xxxxx6lbo@s1.xxxxxx".
- To URI:** A text input field containing "41215xxxxx6lbo@s1.xxxxxx".

On the Advanced tab:

- Select **VTX** 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, each with a label and a control element (checkbox or dropdown menu). The settings are as follows:

Setting	Value
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	VTX
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 center of the window is a button labeled "Finish".

8.7. 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 VTX 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.

8.7.1. Routing – Avaya

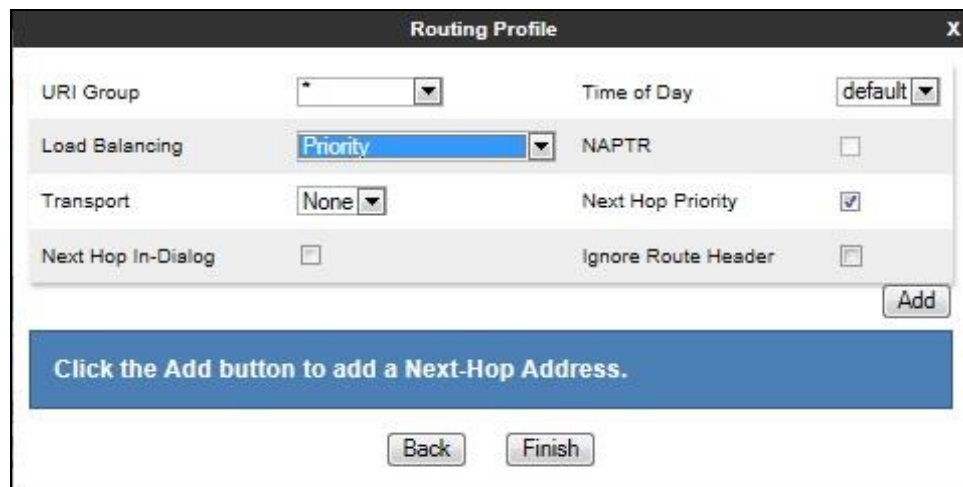
Create a Routing Profile for Session Manager.

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



The image 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 image 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 8.6.1)** from drop down menu.
- **Next Hop Address = Select 10.10.3.42:5061 (TLS)** from drop down menu.
- Click **Finish**.

Profile : Avaya

URI Group: *
Time of Day: default
Load Balancing: Priority
Transport: None
LDAP Server Profile: None
LDAP Base DN (Search): None
Matched Attribute Priority: ☐
Next Hop Priority: ☒
Ignore Route Header: ☐
ENUM: ☐
ENUM Suffix:
Add
Priority / Weight: 1
LDAP Search Attribute:
LDAP Search Regex Pattern:
LDAP Search Regex Result:
SIP Server Profile: Avaya
Next Hop Address: 10.10.3.42:5061
Transport: None
Delete
Finish

8.7.2. Routing – VTX

Create a Routing Profile for VTX SIP network.

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

Routing Profile

Profile Name: VTX
Next

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

Routing Profile

URI Group

*

Time of Day

default

Load Balancing

Priority

NAPTR

Transport

None

Next Hop Priority

Next Hop In-Dialog

Ignore Route Header

Add

Click the Add button to add a Next-Hop Address.

Back

Finish

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

- **Priority/Weight = 1.**
- **SIP Server Profile = VTX** (Section 8.6.2) from drop down menu.
- **Next Hop Address = Select s1.xxxxxx.trk.ipvoip.ch (UDP)** from drop down menu.
- Click **Finish**.

Profile : VTX

URI Group

*

Time of Day

default

Load Balancing

Priority

NAPTR

Transport

None

LDAP Routing

LDAP Server Profile

None

LDAP Base DN (Search)

None

Matched Attribute Priority

Alternate Routing

Next Hop Priority

Next Hop In-Dialog

Ignore Route Header

ENUM

ENUM Suffix

Add

Priority / Weight

LDAP Search Attribute

LDAP Search Regex Pattern

LDAP Search Regex Result

SIP Server Profile

Next Hop Address

Transport

1

VTX

s1.xxxxxx.trk.ipv

None

Delete

Finish

8.8. 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 SBC 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

VTX

Click here to add a description.

Topology Hiding

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

Edit

To define Topology Hiding for VTX, 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 VTX and click **Next**.
- If the required Header is not shown, click on **Add Header**.
- Under the **Header** field for, **From**, select **IP/Domain** under **Criteria** and **Overwrite** under **Replace Action**. For Overwrite value, insert **s1.xxxxxx.trk.ipvoip.ch**.
- Click **Finish** (not shown).

Topology Hiding Profiles: VTX

Add Rename Clone Delete

Topology Hiding Profiles

default

cisco_th_profile

Avaya

VTX

Click here to add a description.

Topology Hiding

Header	Criteria	Replace Action	Overwrite Value
Request-Line	IP/Domain	Auto	---
Via	IP/Domain	Auto	---
From	IP/Domain	Overwrite	s1.xxxxxx.trk.ipvoip.ch
To	IP/Domain	Auto	---
Referred-By	IP/Domain	Auto	---
Refer-To	IP/Domain	Auto	---
Record-Route	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---

Edit

8.9. 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 and Signalling 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 of the defaults and then make the necessary changes to the new rule.

8.9.1. Media Rules

A media rule defines the processing to be applied to the selected media. For the compliance test, media rules were created for Session Manager to SRTP and VTX to use RTP.

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** (not shown).

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 list of media rules: 'default-low-med', 'default-low-med-enc', 'default-high', 'default-high-enc', 'avaya-low-med-enc', and 'Avaya_SRTP' (highlighted in red). Above the list is an 'Add' button. The main area has a title bar with 'Rename', 'Clone', and 'Delete' buttons, and a description field with the placeholder 'Click here to add a description.' Below the title bar are 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' (SRTP_AES_CM_128_HMAC_SHA1_80, RTP), 'SRTP Context Reset on SSRC Change' (checkbox), 'Encrypted RTCP' (checkbox), 'MKI' (checkbox), 'Lifetime' (Any), and 'Interworking' (checkbox). The 'Video Encryption' section has fields for 'Preferred Formats' (RTP) and 'Interworking' (checkbox).

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

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.

Encryption

Codec Prioritization

Advanced

QoS

Audio Encryption

Preferred Formats

RTP

Interworking

☒

Symmetric Context Reset

☒

Key Change in New Offer

☐

Video Encryption

Preferred Formats

RTP

Interworking

☒

Symmetric Context Reset

☒

Key Change in New Offer

☐

Miscellaneous

Capability Negotiation

☐

Edit

CMN; Reviewed:
SPOC 1/17/2024

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VTX_AuraEPSBC10

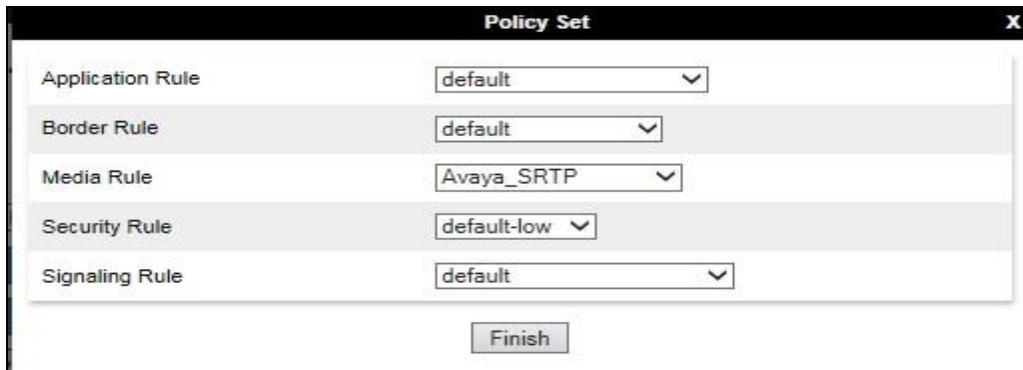
8.10. End Point Policy Groups

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

8.10.1. 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**.



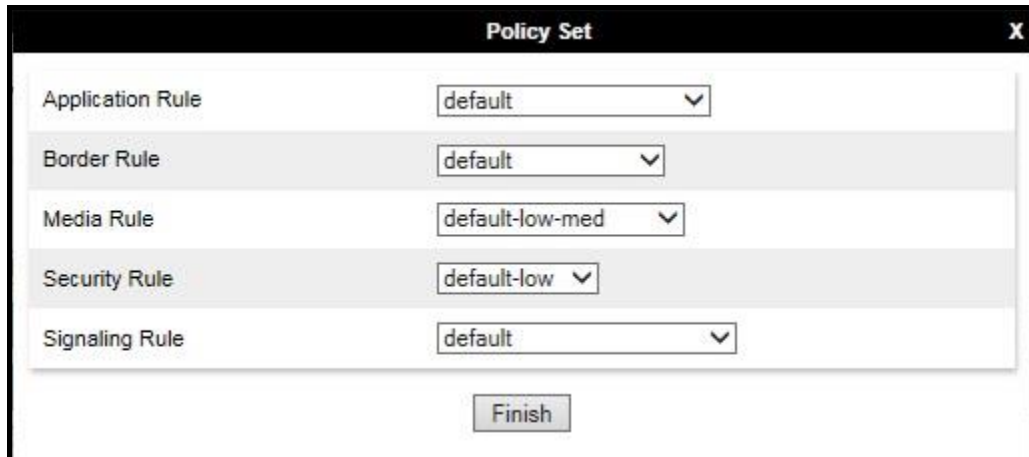
The screenshot shows a 'Policy Set' dialog box with a close button (X) in the top right corner. It contains five rows of configuration options, each with a label and a dropdown menu:

Field	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.

8.10.2. End Point Policy Group – VTX

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



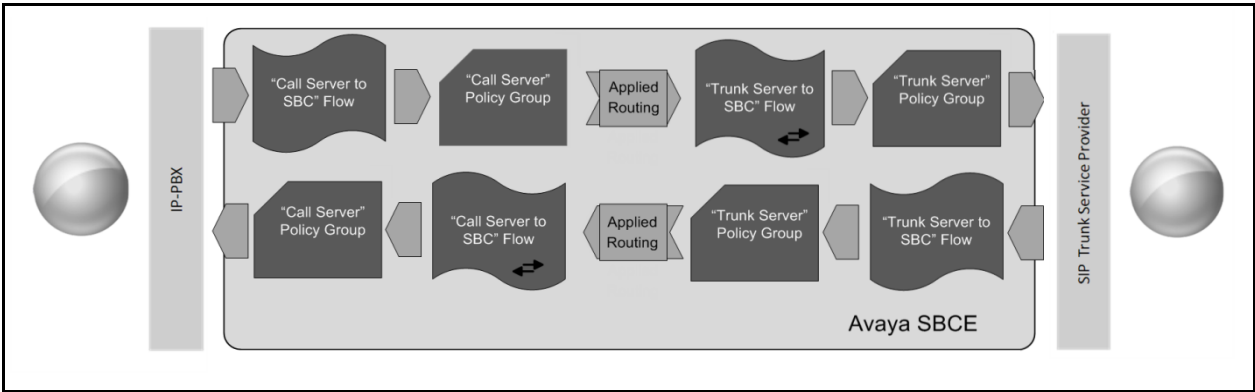
The screenshot shows a 'Policy Set' configuration window with a black title bar and a close button (X) in the top right corner. The window contains a list of five rules, each with a corresponding dropdown menu. The rules and their selected values are: Application Rule (default), Border Rule (default),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

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

Finish

8.11. Server Flows

Server Flows combine the previously defined profiles into outgoing flows from Session Manager to VTX’s SIP Trunk and incoming flows from VTX’s SIP Trunk to Session Manager. The following screen illustrates the flow through the Avaya SBC 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 VTX 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	VTX	View Clone Edit Delete

SIP Server: VTX

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 a Server Flow for the VTX 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 VTX SIP Trunk, in the test environment **Trunk_Server** was used.
- In the **Server Configuration** drop-down menu, select the VTX 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 8.7.1**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the VTX SIP Trunk defined in **Section 8.8** and click **Finish** (not shown).

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

Criteria Section:

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

Profile Section:

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

To define an incoming server flow for Session Manager from the VTX 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 VTX SIP Trunk defined in **Section 8.7.2**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of Session Manager defined in **Section 8.8** 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" and "Profile".

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	VTX
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. VTX Connect PBX-IP SIP Trunk Configuration

The configuration of the VTX equipment used to support VTX’s SIP Trunk is outside of the scope of these Application Notes and will not be covered. To obtain further information on VTX equipment and system configuration please contact an authorized VTX representative as listed in **Section 2.3**.

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 Filter: Enable									
	SIP Entity Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
<input type="radio"/>	Avaya SBCE	IPv4	10.10.3.30	5061	TLS	FALSE	UP	200 OK	UP
<input type="radio"/>	Communication Manager	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 SBC trace facility to check that the OPTIONS requests sent from Session Manager via the Avaya SBC to the network SBCs are receiving a response.

To define the trace, navigate to **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 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**, **1000** 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

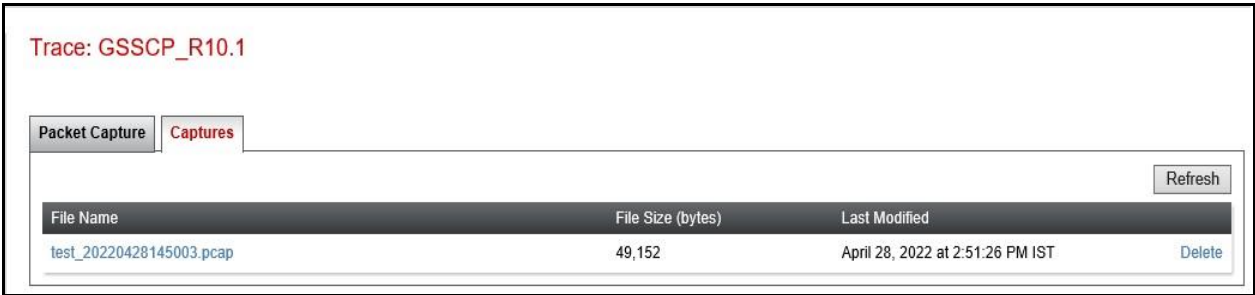
Packet Capture
Captures

Packet Capture Configuration

Status	Ready
Interface	B1
Local Address <small>IP[:Port]</small>	All :
Remote Address <small>*, *:Port, IP, IP:Port</small>	*
Protocol	All
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 VTX network.

11. Conclusion

These Application Notes describe the configuration necessary to successfully connect Avaya Aura ® Communication Manager R10.1, Avaya Aura ® Session Manager 10.1 and Avaya Session Border Controller R10.1 to the VTX Connect PBX-IP SIP platform.

The reference configuration shown in these Application Notes is representative of a basic enterprise customer configuration and demonstrates Avaya Aura ® Communication Manager R10.1, Avaya Aura ® Session Manager 10.1 with Avaya Session Border Controller can be configured to interoperate successfully with VTX Connect PBX-IP SIP Trunk Service. The VTX Connect PBX-IP SIP Service is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. The service was successfully tested with a number of observations listed in **Section 2.2**.

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, Sep 2023
- [2] *Upgrading Avaya Aura® applications*, Release 10.1, Sep 2023
- [3] *Deploying Avaya Aura® applications from System Manager*, Release 10.1, Feb 2023
- [4] *Deploying Avaya Aura® Communication Manager*, Release 10.1, May 2023
- [5] *Administering Avaya Aura® Communication Manager*, Release 10.1, May 2023
- [6] *Upgrading Avaya Aura® Communication Manager*, Release 10.1, May 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*, Release 10.1, Jun 2023
- [14] *Upgrading Avaya Session Border Controller*, Release 10.1 Jun 2023
- [15] *Administering Avaya Session Border Controller*, Release 10.1, Jun 2023
- [16] *Deploying Avaya Experience Portal*, Release 8.1.2, Oct 2022
- [17] *Administering Avaya Experience Portal*, Release 8.1.2, Oct 2022
- [18] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>

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