



Application Notes for Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1 and Avaya Session Border Controller for Enterprise 8.0 with Motto Communications SIP Trunking Service – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking Service on an enterprise solution consisting of Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1 and Avaya Session Border Controller for Enterprise 8.0 to interoperate with Motto Communications SIP Trunking service. These Application Notes update previously published Application Notes with newer versions of Communication Manager, Session Manager, and Avaya Session Border Controller for Enterprise.

The test was performed to verify SIP trunk features including basic calls, call forward (all calls, busy, no answer), call transfer (blind and consult), conference, and voice mail. The calls were placed to and from the PSTN with various Avaya endpoints.

The Motto Communications SIP Trunking service provides customers with PSTN access via a SIP trunk between the enterprise and the Motto Communications network, as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

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

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

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1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking Service between the Motto Communications network and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Communication Manager 8.1 (Communication Manager), Avaya Aura® Session Manager 8.1 (Session Manager), Avaya Session Border Controller for Enterprise 8.0 (Avaya SBCE) and various Avaya endpoints, listed in **Section 4**.

The Motto Communications SIP Trunking service referenced within these Application Notes is designed for business customers. Customers using this service with this Avaya enterprise solution are able to place and receive PSTN calls via a broadband WAN connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as analog and/or ISDN-PRI.

The terms “Service Provider”, “Motto Communications” or “Motto” will be used interchangeably throughout these Application Notes.

2. General Test Approach and Test Results

A simulated CPE site containing all the equipment for the Avaya SIP-enabled enterprise solution was installed at the Avaya Solution and Interoperability Lab. The enterprise site was configured to connect to the network via a broadband connection to the public Internet.

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 this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products only (private network side). 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.

For the testing associated with this Application Note, the interface between Avaya systems and the Motto Communications SIP Trunking service did not include the use of any specific encryption features.

Encryption (TLS/SRTP) was used internal to the enterprise between Avaya products wherever possible.

2.1. Interoperability Compliance Testing

To verify SIP trunk interoperability, the following features and functionality were covered during the interoperability compliance test:

- SIP Trunk Registration (Dynamic Authentication).
- Response to SIP OPTIONS queries.
- Incoming calls from the PSTN were routed to DID numbers assigned by Motto Communications. Incoming PSTN calls were terminated to the following endpoints: Avaya 96x1 Series IP Deskphones (H.323 and SIP), Avaya J179 IP Deskphones (H.323), Avaya 2420 Digital Deskphones, Avaya one-X® Communicator softphone (H.323 and SIP), Avaya Equinox softphone (SIP) and analog Deskphones.
- Inbound and outbound PSTN calls to/from Remote Workers using Avaya 96x1 Deskphones (SIP).
- Outgoing calls to the PSTN were routed via Motto Communications network to various PSTN destinations.
- Proper disconnect when the caller abandons the call before the call is answered.
- Proper disconnect via normal call termination by the caller or the called parties.
- Proper disconnect by the network for calls that are not answered (with voicemail off).
- Proper response to busy endpoints.
- Proper response/error treatment when dialing invalid PSTN numbers.
- Proper codec negotiation and two-way speech-path. Testing was performed with codecs: G.711A, G.711MU and G.729.
- No matching codecs.
- DTMF tone transmissions as out-of-band RTP events as per RFC2833:
 - Outbound call to PSTN application requiring DTMF (e.g., an IVR or voice mail system).
 - Inbound call from PSTN to Avaya CPE application requiring DTMF (e.g., Aura® Messaging, Avaya vector digit collection steps).
- Calling number blocking (Privacy).
- Call Hold/Resume (long and short duration).
- Call Forward (unconditional, busy, no answer).
- Blind Call Transfers.
- Consultative Call Transfers.
- Station Conference.
- EC500 (Extension to Cellular) calls.
- Routing inbound vector call to call center agent queues.
- Simultaneous active calls.
- Long duration calls (over one hour).
- Proper response/error treatment to all trunks busy.
- Proper response/error treatment when disabling SIP connection.

Note – Remote Worker was tested as part of this solution. The configuration necessary to support remote workers is beyond the scope of these Application Notes and is not included in these Application Notes. Consult reference [9] in the **References** section for additional information on this topic.

Items that are supported and that were not tested includes the following:

- Inbound toll-free calls were not tested.
- 0, 0+10 digits, 411 Directory Assistance, 911 Emergency and international calls were not tested.

2.2. Test Results

Interoperability testing of the Motto Communications SIP Trunking Service with the Avaya SIP-enabled enterprise solution was completed with successful results for all test cases with the observations/limitations noted below:

- **DNS SRV record queries** – The preferred method of connecting to the Motto Communications SIP Trunk network is via DNS SRV record queries. The DNS SRV record query functionality is not fully supported in the current release on the Avaya SBCE (8.0), simple DNS query was used instead. DNS SRV record query will be supported in the next release of the Avaya SBCE (release 8.1).
- **Caller ID on transferred calls** – On calls from the PSTN to the enterprise that were transferred back out to the PSTN, the caller ID number displayed at the PSTN endpoints was always of the transferring party instead of the originating PSTN number.
- **Caller ID on call-forward and EC500 calls** – On calls from the PSTN to the enterprise that were forwarded back out to the PSTN, the caller ID number displayed at the PSTN endpoint always showed “Restricted”. This included calls to “twinned” mobile phones (EC500).
- **OPTIONS** – Motto does not send OPTIONS messages to the Avaya enterprise network, but it does respond to OPTIONS messages it receives from the Avaya enterprise, this was sufficient to maintain the SIP trunk link up in service.
- **Fax support** – Fax call attempts using T.38 were rejected with a “488 Not Acceptable Here” response from Motto Communications. G.711 fax was also tested, but it behaved unreliably. The issue related to G.711 fax being unreliable during the compliance test may be related to the unpredictability of G.711 techniques, which only works well on networks with very few hops and with limited end-to-end delay.
- **TLS/SRTP used within the enterprise** – When TLS/SRTP is used within the enterprise; the SIP headers include the SIPS URI scheme for Secure SIP. The Avaya SBCE converts these header schemes from SIPS to SIP when it sends the SIP message toward Motto Communications. However, for call forward and EC500 calls, the Avaya SBCE was not changing the Diversion header scheme as expected. This anomaly is currently under investigation by the Avaya SBCE team. A workaround is to include a SigMa script for the Service Provider Server Configuration profile on the Avaya SBCE to convert “sips” to “sip” in the Diversion header (**Sections 7.8**).
- **SIP REFER method** – Calls from the PSTN to the enterprise that were transferred back out to the PSTN network using the SIP REFER method did not work properly. On blind transfers, the REFER message was accepted by Motto with a “202 Accepted message”,

but the SIP trunk resources were not released after the call transfer was completed. Motto did not send NOTIFY messages with SIP REFER during call redirection scenarios to the PSTN. Testing was done with REFER enabled in Communication Manager (**Network Call Redirection** set to “y” under the **trunk-group**, refer to **Section 5.7**). With REFER enabled, blind and attended call transfers to the PSTN completed successfully, with the caveat that Communication Manager trunk channels were not released from the call path after the call was transferred; two trunks channels remained busy/connected for the entire duration of the call. There was no impact to the user, it’s being mentioned here simply as an observation.

- **Removal of unwanted xml element information from the SDP in SIP messages sent to Motto Communications** – A Signaling Manipulation script (SigMa) was added to the Avaya SBCE to remove unwanted xml element information from the SDP in SIP messages sent to Motto Communications. (**Sections 7.8**).
- **SIP header optimization** – There are multiple SIP headers and parameters used by Communication Manager and Session Manager, some of them Avaya proprietary, that had no significance in the service provider’s network. These headers were removed with the purpose of blocking enterprise information from being propagated outside of the enterprise boundaries, to reduce the size of the packets entering the service provider’s network and to improve the solution interoperability in general. The following headers were removed from outbound messages using an Adaptation in Session Manager: AV-Correlation-ID, Alert-Info, Endpoint-View, P-AV-Message-id, P-Charging-Vector, AV-Global-Session-ID and P-Location (Refer to **Section 6.4**). To help reduce the packet size further, the Avaya SBCE can remove the “gsid” and “epv” parameters that may be included within the Contact header by applying a Sigma script to the Motto Communications server configuration. Refer to **Section 7.8**.

2.3. Support

For support of Motto Communications SIP Trunking Service visit the corporate Web page at:

<http://www.motto.nl>

For technical support on the Avaya products described in these Application Notes visit

<http://support.avaya.com>

3. Reference Configuration

Figure 1 illustrates the sample Avaya SIP-enabled enterprise solution, connected to the Motto Communications SIP Trunking Service through a public Internet WAN connection.

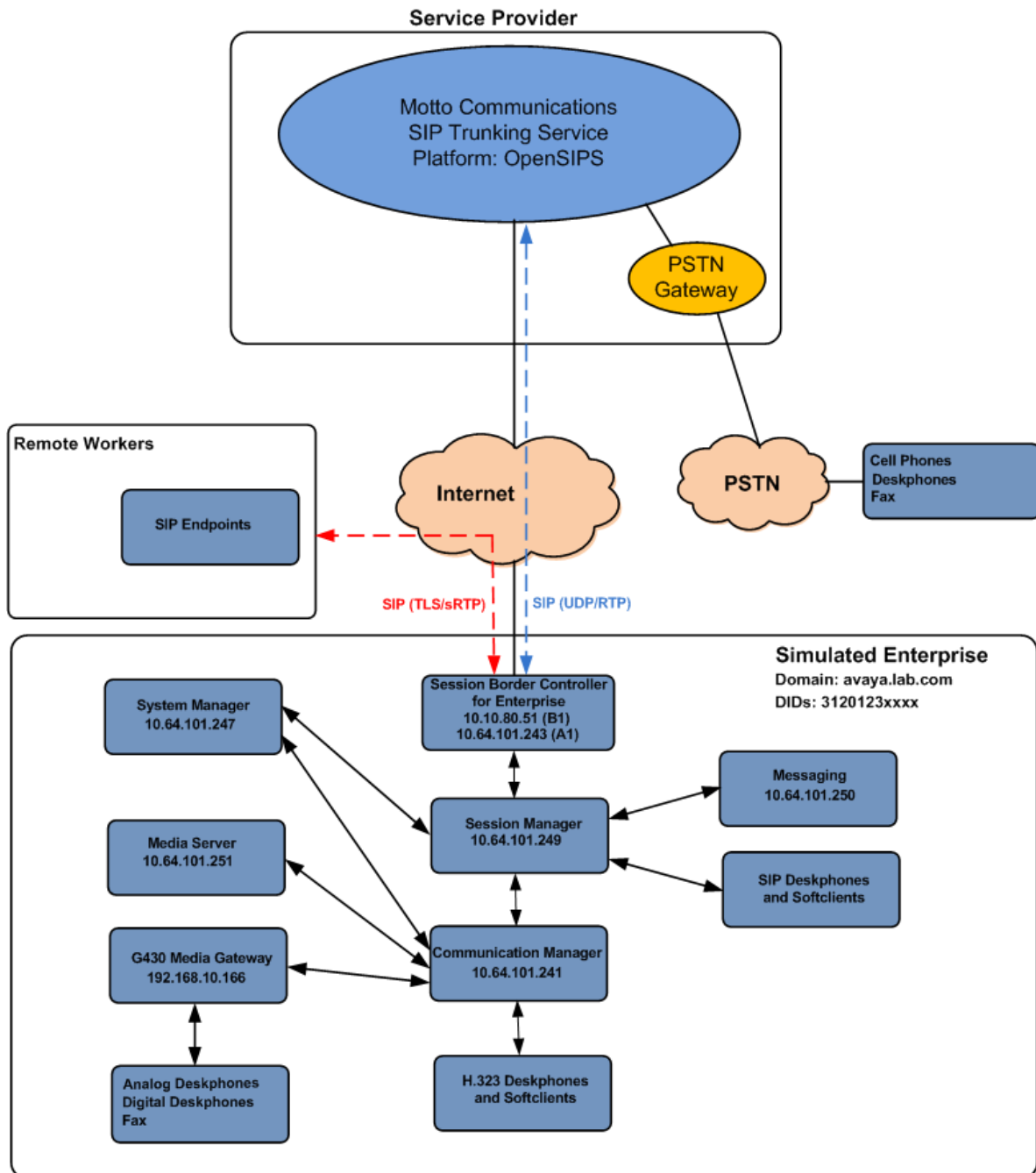


Figure 1: Avaya SIP Enterprise Solution connected to Motto Communications SIP Trunking Service

The Avaya components used to create the simulated enterprise customer site included:

- Avaya Aura® Communication Manager.
- Avaya Aura® Session Manager.
- Avaya Aura® System Manager.
- Avaya Session Border Controller for Enterprise.
- Avaya Aura® Messaging.
- Avaya Aura® Media Server.
- Avaya G430 Media Gateway.
- Avaya 96x1 Series IP Deskphones (H.323 and SIP).
- Avaya J179 IP Deskphones (H.323).
- Avaya one-X® Communicator softphones (H.323 and SIP).
- Avaya Equinox™ for Windows softphone (SIP).
- Avaya digital and analog telephones.
- Ventafax fax software.

Additionally, the reference configuration included remote worker functionality. A remote worker is a SIP endpoint that resides in the untrusted network, registered to Session Manager at the enterprise via the Avaya SBCE. Remote workers offer the same functionality as any other endpoint at the enterprise. This functionality was successfully tested during the compliance test using only the Avaya 96x1 SIP Deskphones. For signaling, Transport Layer Security (TLS) and for media, Secure Real-time Transport Protocol (SRTP) was used on Avaya 96x1 SIP Deskphones used to test remote worker functionality. Other Avaya SIP endpoints that are supported in a Remote Worker configuration deployment were not tested.

The configuration tasks required to support remote workers are beyond the scope of these Application Notes; hence they are not discussed in this document. Consult reference [9] in the **References** section for additional information on this topic.

The Avaya SBCE was located at the edge of the enterprise. Its public side was connected to the public Internet, while its private side was connected to the enterprise infrastructure. All signaling and media traffic entering or leaving the enterprise flowed through the Avaya SBCE, protecting in this way the enterprise against any SIP-based attacks. The Avaya SBCE also performed network address translation at both the IP and SIP layers.

For inbound calls, the calls flowed from the service provider to the Avaya SBCE then to Session Manager. Session Manager used the configured dial patterns (or regular expressions) and routing policies to determine the recipient (Communication Manager) and on which link to send the call.

Outbound calls to the PSTN were first processed by Communication Manager for outbound feature treatment such as automatic route selection and class of service restrictions. Once Communication Manager selected the proper SIP trunk, the call was routed to Session Manager. Session Manager once again used the configured dial patterns (or regular expressions) and

routing policies to determine the route to the Avaya SBCE for egress to the Motto Communications network.

A separate SIP trunk was created between Communication Manager and Session Manager to carry the service provider traffic. This was done so that any trunk or codec settings required by the service provider could be applied only to this trunk without affecting other enterprise SIP traffic. This trunk carried both inbound and outbound traffic.

As part of the Avaya Aura® version 8.0 release, Communication Manager incorporates the ability to use the Avaya Aura® Media Server (AAMS) as a media resource. The AAMS is a software-based, high density media server that provides DSP resources for IP-based sessions. Media resources from both the AAMS and a G430 Media Gateway were utilized during the compliance test. The configuration of the AAMS is not discussed in this document. For more information on the installation and administration of the AAMS in Communication Manager refer to the AAMS documentation listed in the **References** section.

The Avaya Aura® Messaging was used during the compliance test to verify voice mail redirection and navigation, as well as the delivery of Message Waiting Indicator (MWI) messages to the enterprise telephones. Since the configuration tasks for Messaging are not directly related to the interoperability tests with the Motto Communications network SIP Trunking service, they are not included in these Application Notes.

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Avaya Aura® Communication Manager	8.1.1.0 (01.0.890.0-25763)
Avaya Aura® Session Manager	8.1.1.0 (8.1.1.0.811021)
Avaya Aura® System Manager	.1.1.0 Build No. 8.1.0.0.733078 Software Update Rev. No. 8.1.1.0.0310504
Avaya Session Border Controller for Enterprise	ASBCE 8.0 8.0.1.0-10-17555
Avaya Aura® Messaging	7.1 Service Pack 2 (MSG-01.0.532.0-002_0204)
Avaya Aura® Media Server	8.0.2.61_2019.09.16
Avaya G430 Media Gateway	g430_sw_41_16_0
Avaya 96x1 Series IP Deskphones (SIP)	Version 7.1.7.0
Avaya 96x1 Series IP Deskphones (H.323)	Version 6.8202
Avaya J179 IP Deskphones (H.323)	Version 6.8202
Avaya one-X® Communicator (H.323, SIP)	6.2.14.1-SP14
Avaya Equinox for Windows (SIP)	3.6.4.31.2
Avaya 2420 Series Digital Deskphones	N/A
Avaya 6210 Analog Deskphones	N/A
Motto Communications	
OpenSIPS	2.2.1 (x86_64/linux)
Asterisk	11.14.0-motto3

The specific configuration above was used for the compliance testing. Note that this solution will be compatible with other Avaya Servers and Media Gateway platforms running similar versions of Communication Manager and Session Manager.

Note – The Avaya Aura® servers and the Avaya SBCE used in the reference configuration and shown on the previous table were deployed on a virtualized environment. These Avaya components ran as virtual machines over VMware® (ESXi 6.0.0) platforms. Consult the installation documentation on the **References** section for more information.

5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring Communication Manager to work with the Motto Communications SIP Trunking Service. A SIP trunk is established between Communication Manager and Session Manager for use by signaling traffic to and from the service provider. It is assumed that the general installation of Communication Manager, the Avaya G430 Media Gateway and the Avaya Media Server has been previously completed and is not discussed here.

The 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. Some screens capture will show the use of the **change** command instead of the **add** command, since the configuration used for the testing was previously added.

5.1. Licensing and Capacity

Use the **display system-parameters customer-options** command to verify that the **Maximum Administered SIP Trunks** value on **Page 2** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise including any trunks to and from the service provider. The example shows that **40000** licenses are available and **120** are in use. 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.

display system-parameters customer-options		Page 2 of 12
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:		12000 0
Maximum Concurrently Registered IP Stations:		18000 2
Maximum Administered Remote Office Trunks:		12000 0
Max Concurrently Registered Remote Office Stations:		18000 0
Maximum Concurrently Registered IP eCons:		414 0
Max Concur Reg Unauthenticated H.323 Stations:		100 0
Maximum Video Capable Stations:		41000 0
Maximum Video Capable IP Softphones:		18000 6
Maximum Administered SIP Trunks:		40000 120
Max Administered Ad-hoc Video Conferencing Ports:		24000 0
Max Number of DS1 Boards with Echo Cancellation:		999 0
(NOTE: You must logoff & login to effect the permission changes.)		

5.2. System Features

Use the **change system-parameters features** command to set the **Trunk-to-Trunk Transfer** field to *all* to allow incoming calls from the PSTN to be transferred to another PSTN endpoint. If for security reasons incoming calls should not be allowed to transfer back to the PSTN, then leave the field set to *none*.

```
display system-parameters features                                     Page 1 of 19
FEATURE-RELATED SYSTEM PARAMETERS
  Self Station Display Enabled? n
  Trunk-to-Trunk Transfer: all
  Automatic Callback with Called Party Queuing? n
  Automatic Callback - No Answer Timeout Interval (rings): 3
  Call Park Timeout Interval (minutes): 10
  Off-Premises Tone Detect Timeout Interval (seconds): 20
  AAR/ARS Dial Tone Required? y

  Music (or Silence) on Transferred Trunk Calls? all
  DID/Tie/ISDN/SIP Intercept Treatment: attendant
  Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
  Automatic Circuit Assurance (ACA) Enabled? n

  Abbreviated Dial Programming by Assigned Lists? n
  Auto Abbreviated/Delayed Transition Interval (rings): 2
  Protocol for Caller ID Analog Terminals: Bellcore
  Display Calling Number for Room to Room Caller ID Calls? n
```

On **Page 9** verify that a text string has been defined to replace the Calling Party Number (CPN) for restricted or unavailable calls. This text string is entered in the two fields highlighted below. The compliance test used the value of *restricted* for restricted calls and *unavailable* for unavailable calls.

```
display system-parameters features                                     Page 9 of 19
                                FEATURE-RELATED SYSTEM PARAMETERS

CPN/ANI/ICLID PARAMETERS
  CPN/ANI/ICLID Replacement for Restricted Calls: restricted
  CPN/ANI/ICLID Replacement for Unavailable Calls: unavailable

DISPLAY TEXT
                                Identity When Bridging: principal
                                User Guidance Display? n
  Extension only label for Team button on 96xx H.323 terminals? n

INTERNATIONAL CALL ROUTING PARAMETERS
                                Local Country Code:
                                International Access Code:

SCCAN PARAMETERS
  Enable Enbloc Dialing without ARS FAC? n

CALLER ID ON CALL WAITING PARAMETERS
  Caller ID on Call Waiting Delay Timer (msec): 200
```

5.3. IP Node Names

Use the **change node-names ip** command to verify that node names have been previously defined for the IP addresses of Communication Manager (**procr**) and the Session Manager security module (**SM**). These node names will be needed for defining the service provider signaling group in **Section 5.6**.

change node-names ip		Page 1 of 2
IP NODE NAMES		
Name	IP Address	
ASBCE_A1	10.64.101.243	
SM	10.64.101.249	
default	0.0.0.0	
media server	10.64.101.251	
procr	10.64.101.241	
procr6	::	
(6 of 6 administered node-names were displayed)		
Use 'list node-names' command to see all the administered node-names		
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name		

5.4. Codecs

Use the **change ip-codec-set** command to define a list of codecs to use for calls between the enterprise and the service provider. For the compliance test, ip-codec-set 2 was used for this purpose. Enter the corresponding codec in the **Audio Codec** column of the table. Motto Communications supports audio codecs *G.711A*, *G.711MU* and *G.729*.

change ip-codec-set 2 Page 1 of 2

IP MEDIA PARAMETERS

Codec Set: 2

	Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)
1:	G.711A	n	2	20
2:	G.711MU	n	2	20
3:	G.729	n	2	20
4:		—	—	
5:		—	—	
6:		—	—	
7:		—	—	

Media Encryption

1: 1-srtp-aescm128-hmac80

2: none

3:

4:

5:

Encrypted SRTP: best-effort

On **Page 2**, set the **Fax Mode** to *off*.

Motto SIP Trunk supports G.711 for transmission of fax. As this is in-band and requires no interaction from Communication Manager, there is no specific configuration required (refer to **Section 2.2**).

change ip-codec-set 2 Page 2 of 2

IP MEDIA PARAMETERS

Allow Direct-IP Multimedia? n

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

Media Connection IP Address Type Preferences


1: IPv4

2:

5.5. IP Network Regions

Create a separate IP network region for the service provider trunk group. This allows for separate codec or quality of service settings to be used (if necessary) for calls between the enterprise and the service provider versus calls within the enterprise or elsewhere. For the compliance test, IP Network Region 2 was chosen for the service provider trunk. Use the **change ip-network-region 2** command to configure region 2 with the following parameters:

- Set the **Authoritative Domain** field to match the SIP domain of the enterprise. In this configuration, the domain name is **avaya.lab.com** as assigned to the shared test environment in the Avaya test lab. This domain name appears in the “From” header of SIP messages originating from this IP region.
- Enter a descriptive name in the **Name** field.
- Leave both **Intra-region** and **Inter-region IP-IP Direct Audio** set to **yes**, the default setting. This will enable **IP-IP Direct Audio** (shuffling), to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway and Media Server. Shuffling can be further restricted at the trunk level on the Signaling Group form if needed.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.4**.
- Default values may be used for all other fields.

change ip-network-region 2		Page 1 of 20
IP NETWORK REGION		
Region: 2	NR Group: 2	
Location: 1	Authoritative Domain: avaya.lab.com	
Name: SP Region	Stub Network Region: n	
MEDIA PARAMETERS		Intra-region IP-IP Direct Audio: yes
Codec Set: 2		Inter-region IP-IP Direct Audio: yes
UDP Port Min: 2048		IP Audio Hairpinning? n
UDP Port Max: 3349		
DIFFSERV/TOS PARAMETERS		
Call Control PHB Value: 46		
Audio PHB Value: 46		
Video PHB Value: 26		
802.1P/Q PARAMETERS		
Call Control 802.1p Priority: 6		
Audio 802.1p Priority: 6		
Video 802.1p Priority: 5		
H.323 IP ENDPOINTS		AUDIO RESOURCE RESERVATION PARAMETERS
H.323 Link Bounce Recovery? y		RSVP Enabled? 
Idle Traffic Interval (sec): 20		
Keep-Alive Interval (sec): 5		
Keep-Alive Count: 5		

On **Page 4**, define the IP codec set to be used for traffic between region 2 and region 1 (the rest of the enterprise). Enter the desired IP codec set in the **codec set** column of the row with destination region (**dst rgn**) 1. Default values may be used for all other fields. The following example shows the settings used for the compliance test. It indicates that codec set **2** will be used for calls between region 2 (the service provider region) and region 1 (the rest of the enterprise).

change ip-network-region 2										Page 4 of 20		
Source Region: 2		Inter Network Region Connection Management								I		M
dst	codec	direct	WAN-BW-limits		Video		Intervening		Dyn	G	A	t
rgn	set	WAN	Units	Total	Norm	Prio	Shr	Regions	CAC	R	L	c
1	2	y	NoLimit						n			t
2	2										all	
3												
4												
5												
6												
7												
8												
9												
10												
11												
12												
13												
14												
15												

5.6. Signaling Group

Use the **add signaling-group** command to create a signaling group between Communication Manager and Session Manager for use by the service provider trunk. This signaling group is used for inbound and outbound calls between the service provider and the enterprise. For the compliance test, signaling group 2 was used and was configured using the parameters highlighted below, shown on the screen on the next page:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*. This specifies the Communication Manager will serve as an Evolution Server for the Session Manager.
- Set the **Transport Method** to the transport protocol to be used between Communication Manager and Session Manager. For the compliance test, *tls* was used.
- Set the **Peer Detection Enabled** field to *y*. The **Peer-Server** field will initially be set to *Others* and cannot be changed via administration. Later, the **Peer-Server** field will automatically change to *SM* once Communication Manager detects its peer is a Session Manager.

Note: Once the **Peer-Server** field is updated to *SM*, the system changes the default values of the following fields, setting them to display-only:

- Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? is changed to *y*.
- Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? is changed to *n*.
- Set the **Near-end Node Name** to *procr*. This node name maps to the IP address of the Communication Manager as defined in **Section 5.3**.
- Set the **Far-end Node Name** to *SM*. This node name maps to the IP address of Session Manager, as defined in **Section 5.3**.
- Set the **Near-end Listen Port** and **Far-end Listen Port** to a valid unused port instead of the default well-known port value. (For TLS, the well-known port value is 5061). This is necessary so Session Manager can distinguish this trunk from the trunk used for other enterprise SIP traffic. The compliance test was conducted with the **Near-end Listen Port** and **Far-end Listen Port** set to *5071*.
- Set the **Far-end Network Region** to the IP network region defined for the Service Provider in **Section 5.5**.
- Set the **Far-end Domain** to the domain of the enterprise.
- Set the **DTMF over IP** field to *rtp-payload*. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- Set **Direct IP-IP Audio Connections** to *y*. This field will enable media shuffling on the SIP trunk allowing Communication Manager to redirect media traffic directly between the Avaya SBCE and the enterprise endpoint. If this value is set to *n*, then the Avaya Media Gateway or Media Server will remain in the media path of all calls between the SIP trunk and the endpoint. Depending on the number of media resources available in the Avaya Media Gateway and Media Server, these resources may be depleted during high call volume preventing additional calls from completing.
- Default values may be used for all other fields.

change signaling-group 2		Page 1 of 2
SIGNALING GROUP		
Group Number: 2	Group Type: sip	
IMS Enabled? <i>n</i>	Transport Method: <i>tls</i>	
Q-SIP? <i>n</i>		
IP Video? <i>n</i>	Enforce SIPS URI for SRTP? <i>y</i>	
Peer Detection Enabled? <i>y</i>	Peer Server: SM	Clustered? <i>n</i>
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? <i>y</i>		
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? <i>n</i>		
Alert Incoming SIP Crisis Calls? <i>n</i>		
Near-end Node Name: <i>procr</i>	Far-end Node Name: <i>SM</i>	
Near-end Listen Port: <i>5071</i>	Far-end Listen Port: <i>5071</i>	
	Far-end Network Region: <i>2</i>	
Far-end Domain: <i>avaya.lab.com</i>		
Bypass If IP Threshold Exceeded? <i>n</i>		
Incoming Dialog Loopbacks: <i>eliminate</i>	RFC 3389 Comfort Noise? <i>n</i>	
DTMF over IP: <i>rtp-payload</i>	Direct IP-IP Audio Connections? <i>y</i>	
Session Establishment Timer(min): <i>3</i>	IP Audio Hairpinning? <i>n</i>	
Enable Layer 3 Test? <i>n</i>	Initial IP-IP Direct Media? <i>n</i>	
H.323 Station Outgoing Direct Media? <i>n</i>	Alternate Route Timer(sec): <i>6</i>	

5.7. Trunk Group

Use the **add trunk-group** command to create a trunk group for the signaling group created in **Section 5.6**. For the compliance test, trunk group 2 was configured using the parameters highlighted below.

- Set the **Group Type** field to *sip*.
- Enter a descriptive name for the **Group Name**.
- Enter an available trunk access code (TAC) that is consistent with the existing dial plan in the **TAC** field.
- Set the **Service Type** field to *public-ntwrk*.
- Set the **Signaling Group** to the signaling group shown in **Section 5.6**.
- Set the **Number of Members** field to the number of trunk members in the SIP trunk group. This value determines how many simultaneous SIP calls can be supported by this trunk.
- Default values were used for all other fields.

```
change trunk-group 2                                     Page 1 of 4
                                     TRUNK GROUP
Group Number: 2      Group Type: sip      CDR Reports: y
Group Name: Service Provider      COR: 1      TN: 1      TAC: 602
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: 2
                                     Number of Members: 10
```

On **Page 2**, verify that the **Preferred Minimum Session Refresh Interval** is set to a value acceptable to the service provider. This value defines the interval that re-INVITEs must be sent to keep the active session alive. The default value of **600** seconds was used.

change trunk-group 2		Page 2 of 4
Group Type: sip		
TRUNK PARAMETERS		
Unicode Name: <u>auto</u>		
Redirect On OPTIM Failure: <u>5000</u>		
SCCAN? <u>n</u>	Digital Loss Group: <u>18</u>	
Preferred Minimum Session Refresh Interval(sec): <u>600</u>		
Disconnect Supervision - In? <u>y</u> Out? <u>y</u>		
XOIP Treatment: <u>auto</u> Delay Call Setup When Accessed Via IGAR? <u>n</u>		
Caller ID for Service Link Call to H.323 1xC: <u>Station-extension</u>		

On Page 3:

- Set the **Numbering Format** field to *public*. This field specifies the format of the calling party number (CPN) sent to the far-end. When *public* format is used, Communication Manager automatically inserts a “+” sign, preceding the numbers in the “From”, “Contact” and “P-Asserted Identity” (PAI) headers. The **Numbering Format** was set to *public* and the **Numbering Format** in the route pattern was set to *pub-unk* (see **Section 5.10**). Note that in the case of Motto Communications the + sign was removed from SIP messages with a SigMa script added to the Avaya SBCE before sending the SIP messages to Motto Communications (refer to **Section 7.8**).
- Set the **Replace Restricted Numbers** and **Replace Unavailable Numbers** fields to *y*. This will allow the CPN displayed on local endpoints to be replaced with the value set in **Section 5.2**, if the inbound call has enabled CPN block.

change trunk-group 2 Page 3 of 4

TRUNK FEATURES

ACA Assignment? n Measured: none Maintenance Tests? y

Suppress # Outpulsing? n **Numbering Format: public** UI Treatment: service-provider

Replace Restricted Numbers? y
Replace Unavailable Numbers? y

Hold/Unhold Notifications? y
Modify Tandem Calling Number: no

Show ANSWERED BY on Display? y

On Page 4:

- Set the **Network Call Redirection** field to **y**. With this setting, Communication Manager will use the SIP REFER method for the redirection of PSTN calls that are transferred back to the SIP trunk (refer to **Section 2.2**).
- Set the **Send Diversion Header** field to **y** and **Support Request History** to **n**.
- Set the **Telephone Event Payload Type** to **101**, the value preferred by Motto Communications.
- Verify that **Identity for Calling Party Display** is set to **P-Asserted-Identity**.
- Default values were used for all other fields.

change trunk-group 2		Page 4 of 4
PROTOCOL VARIATIONS		
	Mark Users as Phone?	<u>n</u>
Prepend '+' to Calling/Alerting/Diverting/Connected Number?		<u>n</u>
	Send Transferring Party Information?	<u>n</u>
	Network Call Redirection?	<u>y</u>
Build Refer-To URI of REFER From Contact For NCR?		<u>n</u>
	Send Diversion Header?	<u>y</u>
	Support Request History?	<u>n</u>
	Telephone Event Payload Type:	<u>101</u>
	Convert 180 to 183 for Early Media?	<u>n</u>
	Always Use re-INVITE for Display Updates?	<u>n</u>
	Identity for Calling Party Display:	<u>P-Asserted-Identity</u>
Block Sending Calling Party Location in INVITE?		<u>n</u>
	Accept Redirect to Blank User Destination?	<u>n</u>
	Enable Q-SIP?	<u>n</u>
Interworking of ISDN Clearing with In-Band Tones:		<u>keep-channel-active</u>
	Request URI Contents:	<u>may-have-extra-digits</u>

5.8. Calling Party Information

The calling party number is sent in the SIP “From”, “Contact” and “PAI” headers. Since public numbering was selected to define the format of this number (**Section 5.7**), use the **change public-unknown-numbering** command to create an entry for each extension which has a DID assigned. DID numbers are provided by the SIP service provider. Each DID number is assigned in this table to one enterprise internal extension or Vector Directory Numbers (VDNs). In the example below, four DID numbers assigned by the service provider are shown. These DID numbers were used as the outbound calling party information on the service provider trunk when calls were originated from the mapped extensions.

change public-unknown-numbering 1					Page 1 of 2
NUMBERING - PUBLIC/UNKNOWN FORMAT					
Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	Total CPN Len	
4	3			4	Total Administered: 6
4	5			4	Maximum Entries: 9999
4	3042	2	31201231111	11	Note: If an entry applies to a SIP connection to Avaya Aura(R) Session Manager, the resulting number must be a complete E.164 number.
4	3044	2	31201232222	11	
4	3045	2	31201233333	11	
4	3046	2	31201234444	11	
					Communication Manager automatically inserts a '+' digit in this case.

5.9. Inbound Routing

In general, the “incoming call handling treatment” form for a trunk group can be used to manipulate the digits received for an incoming call if necessary. Since Session Manager is present, Session Manager can be used to perform digit conversion using an Adaptation, and digit manipulation via the Communication Manager incoming call handling table may not be necessary. If the DID number sent by Motto Communications is left unchanged by Session Manager, then the DID number can be mapped to an extension using the incoming call handling treatment of the receiving trunk group. Use the **change inc-call-handling-trmt** command to create an entry for each DID.

change inc-call-handling-trmt trunk-group 2					Page 1 of 30
INCOMING CALL HANDLING TREATMENT					
Service/ Feature	Number Len	Number Digits	Del	Insert	
public-ntwrk	11	31201233333	11	3045	
public-ntwrk	11	31201231111	11	3042	
public-ntwrk	11	31201232222	11	3044	
public-ntwrk	11	31201234444	11	3046	
public-ntwrk					
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public					

5.10.Outbound Routing

In these Application Notes, the Automatic Route Selection (ARS) feature is used to route outbound calls via the SIP trunk to the service provider. In the sample configuration, the single digit 9 is used as the ARS access code. Enterprise callers will dial 9 to reach an “outside line”. This common configuration is illustrated below with little elaboration. Use the **change dialplan analysis** command to define a dialed string beginning with **9** of length **1**, as a feature access code (*fac*).

change dialplan analysis			DIAL PLAN ANALYSIS TABLE						Page 1 of 12
			Location: all			Percent Full: 2			
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	
0	13	udp							
1	4	dac							
2	4	ext							
3	4	ext							
4	4	udp							
5	4	ext							
6	3	dac							
7	4	ext							
8	1	fac							
9	1	fac							
*	3	dac							
#	2	dac							

Use the **change feature-access-codes** command to configure **9** 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: #7		
Answer Back Access Code: _____		
Attendant Access Code: _____		
Auto Alternate Routing (AAR) Access Code: 8		
Auto Route Selection (ARS) - Access Code 1: 9		Access Code 2: _____
Automatic Callback Activation: _____		Deactivation: _____
Call Forwarding Activation Busy/DA: _____	All: _____	Deactivation: _____
Call Forwarding Enhanced Status: _____	Act: _____	Deactivation: _____
Call Park Access Code: _____		
Call Pickup Access Code: _____		
CAS Remote Hold/Answer Hold-Unhold Access Code: _____		
CDR Account Code Access Code: _____		
Change COR Access Code: _____		
Change Coverage Access Code: _____		
Conditional Call Extend Activation: _____		Deactivation: _____
Contact Closure	Open Code: _____	Close Code: _____

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9. The example below shows a subset of the dialed strings tested as part of the compliance test. See **Section 2.1** for the complete list of call types tested. All dialed strings are mapped to route pattern 2, which contains the SIP trunk group to the service provider.

For international call to the U.S. (e.g., dialing: 90017863311234):

change ars analysis 001							Page 1 of 2
ARS DIGIT ANALYSIS TABLE							
Location: all							Percent Full: 1
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req'd	
001	13	18	2	intl		n	
01	12	12	2	natl		n	
011	10	18	2	intl		n	
040	3	3	2	svcl		n	
045	13	13	2	natl		n	
101xxxx0	8	8	deny	op		n	
101xxxx0	18	18	deny	op		n	
101xxxx01	16	24	deny	iop		n	
101xxxx011	17	25	deny	intl		n	
101xxxx1	18	18	deny	fnpa		n	
10xxx0	6	6	deny	op		n	
10xxx0	16	16	deny	op		n	
10xxx01	14	22	deny	iop		n	
10xxx011	15	23	deny	intl		n	
10xxx1	16	16	deny	fnpa		n	

The route pattern defines which trunk group will be used for the call and performs any necessary digit manipulation. Use the **change route-pattern** command to configure the parameters for the service provider trunk route pattern in the following manner. The example below shows the values used for route pattern 2 in the compliance test.

- **Pattern Name:** Enter a descriptive name.
- **Grp No:** Enter the outbound trunk group for the SIP service provider.
- **FRL:** Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of **0** is the least restrictive level.
- **Numbering Format:** Set to **pub-unk**. All calls using this route pattern will use the public numbering table. See setting of the **Numbering Format** in the trunk group form for full details in **Section 5.7**.

change route-pattern 2															Page 1 of 4	
Pattern Number: 2															Pattern Name: <u>Serv. Provider</u>	
SCCAN? <u>n</u>															Secure SIP? <u>n</u>	
															Used for SIP stations? <u>n</u>	
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted					DCS/	IXC			
No			Mrk	Lmt	List	Del	Digits					QSIG				
							Dgts					Intw				
1:	<u>2</u>	<u>0</u>										<u>n</u>	<u>user</u>			
2:												<u>n</u>	<u>user</u>			
3:												<u>n</u>	<u>user</u>			
4:												<u>n</u>	<u>user</u>			
5:												<u>n</u>	<u>user</u>			
6:												<u>n</u>	<u>user</u>			

BCC	VALUE	TSC	CA-TSC	ITC	BCIE	Service/Feature	PARM	Sub	Numbering	LAR	
0	1	2	M	4	W			Dgts	Format		
1:	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>n</u>	<u>n</u>		<u>rest</u>	<u>pub-unk</u>	<u>none</u>
2:	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>n</u>	<u>n</u>		<u>rest</u>		<u>none</u>
3:	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>n</u>	<u>n</u>		<u>rest</u>		<u>none</u>
4:	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>n</u>	<u>n</u>		<u>rest</u>		<u>none</u>
5:	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>n</u>	<u>n</u>		<u>rest</u>		<u>none</u>
6:	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>Y</u>	<u>n</u>	<u>n</u>		<u>rest</u>		<u>none</u>

Note - Enter the **save translation** command (not shown) to save all the changes made to the Communication Manager configuration in the previous sections.

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain.
- Logical/physical Locations that can be occupied by SIP Entities.
- Adaptation module to perform header manipulations.
- SIP Entities corresponding to Communication Manager, Session Manager and the Avaya SBCE.
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities.
- Routing Policies, which control call routing between the SIP Entities.
- Dial Patterns, which govern to which SIP Entity a call is routed.

The following sections assume that the initial configuration of Session Manager and System Manager has already been completed, and that network connectivity exists between System Manager and Session Manager.

6.1. System Manager Login and Navigation

Session Manager configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL “https://<ip-address>/SMGR”, where “<ip-address>” is the IP address of System Manager. Log in with the appropriate credentials and click on **Log On** (not shown). The screen shown below is then displayed; under **elements** select **Routing** → **Domains**.

The screenshot displays the Avaya Aura System Manager 8.0 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The 'Elements' menu is expanded, showing a list of system components. The 'Routing' option is highlighted, and its sub-menu is visible, with 'Domains' selected. The main dashboard area contains several widgets: 'System Resource Utilization' (a bar chart showing utilization for 'opt', 'var', and 'emdata'), 'Alarms' (an empty list), 'Notifications' (showing 'No data'), 'Application State' (a table with system status), 'Information' (a table of system components and their sync status), and 'Shortcuts' (a drag-and-drop area). The 'Current Usage' section shows '6/250000 USERS' and '1/50 SIMULTANEOUS ADMINISTRATIVE LOGINS'.

Category	Utilization
opt	~10
var	~5
emdata	~15

Property	Value
License Status	Active
Deployment Type	VMware
Multi-Tenancy	DISABLED
OoBM State	DISABLED
Hardening Mode	Standard

Elements	GNRL	Sync Status
CM	1	■
Messaging	1	■
Session Manager	1	■
System Manager	1	■
UCM Applications	16	■

Usage Type	Value
USERS	6/250000
SIMULTANEOUS ADMINISTRATIVE LOGINS	1/50

The navigation tree displayed in the left pane below will be referenced in subsequent sections to navigate to items requiring configuration. Most items discussed in this section will be located under the **Routing** link shown below.

The screenshot displays the Avaya Aura System Manager 8.0 web interface. The top navigation bar includes the Avaya logo, the text "Aura® System Manager 8.0", and several menu items: Users, Elements, Services, Widgets, and Shortcuts. A search bar and a user profile icon labeled "admin" are also present. Below the top bar, a breadcrumb trail shows "Home" and "Routing x". The left-hand navigation pane is expanded, showing a tree structure under the "Routing" category. The "Domains" item is highlighted in blue, and a red rectangle is drawn around the entire "Routing" section of the navigation tree. The main content area is titled "Domain Management" and features a "Help ?" link. Below the title, there are action buttons: "New", "Edit", "Delete", "Duplicate", and "More Actions". A table displays a single item with the following details:

Name	Type	Notes
avaya.lab.com	sip	HG V-Domain

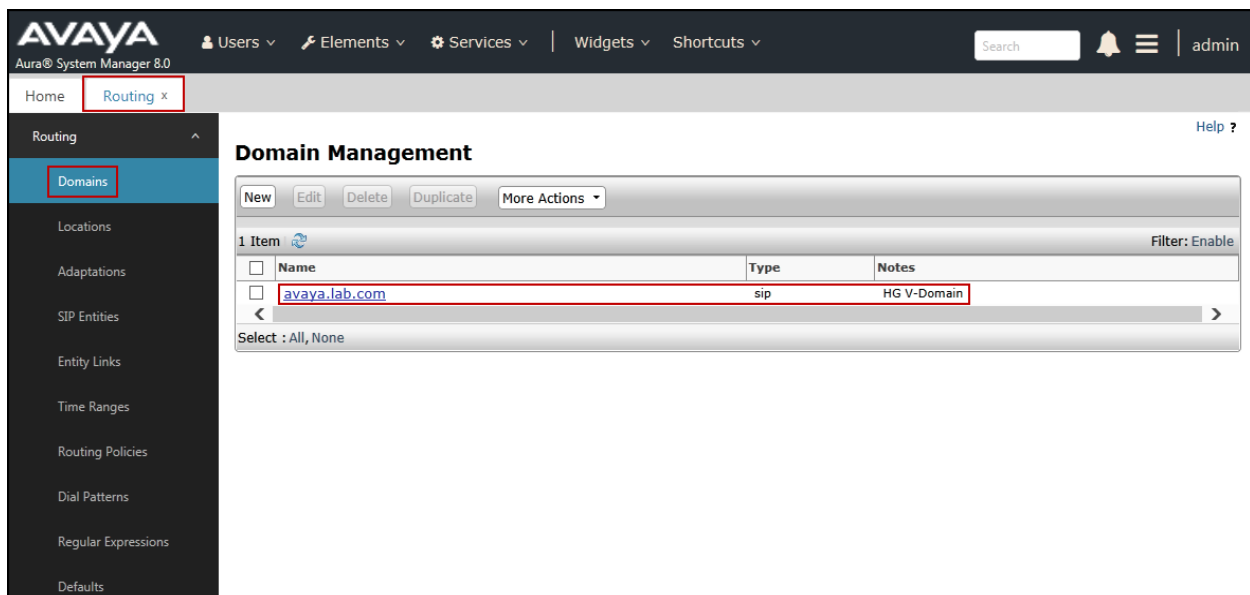
Below the table, there is a "Filter: Enable" option and a "Select : All, None" dropdown menu.

6.2. SIP Domain

Create an entry for each SIP domain for which Session Manager will need to be aware in order to route calls. For the compliance test, this was the enterprise domain, **avaya.lab.com**. Navigate to **Routing → Domains** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Name:** Enter the domain name.
- **Type:** Select **sip** from the pull-down menu.
- **Notes:** Add a brief description (optional).
- Click **Commit** to save.

The screen below shows the entry for the enterprise domain.



6.3. Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management, call admission control and location-based routing. 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:

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

The following screen shows the location details for the location named ***Session Manager***. Later, this location will be assigned to the SIP Entity corresponding to Session Manager. Other location parameters (not shown) retained the default values.

The screenshot displays the Avaya Aura System Manager 8.0 interface. The top navigation bar includes the Avaya logo, version information, and various menu items like Users, Elements, Services, Widgets, and Shortcuts. A search bar and a user profile icon are also present. The left sidebar shows a tree view with 'Routing' selected, and 'Locations' highlighted under it. The main content area is titled 'Location Details' and contains several sections: 'General' with fields for 'Name' (Session Manager) and 'Notes' (VMware Session Manager); 'Dial Plan Transparency in Survivable Mode' with an 'Enabled' checkbox and fields for 'Listed Directory Number' and 'Associated CM SIP Entity'; and 'Overall Managed Bandwidth' with a 'Managed Bandwidth Units' dropdown set to 'Kbit/sec' and fields for 'Total Bandwidth' and 'Multimedia Bandwidth'. 'Commit' and 'Cancel' buttons are located at the top right of the form area.

The following screen shows the location details for the location named ***Communication Manager***. Later, this location will be assigned to the SIP Entity corresponding to Communication Manager. Other location parameters (not shown) retained the default values.

This screenshot shows the same Avaya Aura System Manager 8.0 interface as the previous one, but with the 'Name' field set to 'Communication Manager' and the 'Notes' field set to 'VMware Communication Manager'. All other fields and the overall layout remain identical to the previous screenshot.

The following screen shows the location details for the location named **Avaya SBCE**. Later, this location will be assigned to the SIP Entity corresponding to the Avaya SBCE. Other location parameters (not shown) retained the default values.

The screenshot displays the Avaya Aura System Manager 8.0 web interface. The top navigation bar includes the Avaya logo, version information, and a menu with options like Users, Elements, Services, Widgets, and Shortcuts. A search bar and a user profile icon labeled 'admin' are also present. The left sidebar shows a navigation tree with 'Routing' selected, and 'Locations' highlighted under the 'Routing' section. The main content area is titled 'Location Details' and features a 'Commit' button and a 'Cancel' button. The 'General' section contains a redacted 'Name' field (containing 'Avaya SBCE') and a 'Notes' field (containing 'VMware Avaya SBCE'). The 'Dial Plan Transparency in Survivable Mode' section has an 'Enabled' checkbox and two empty input fields for 'Listed Directory Number' and 'Associated CM SIP Entity'. The 'Overall Managed Bandwidth' section includes a 'Managed Bandwidth Units' dropdown menu (set to 'Kbit/sec'), and two empty input fields for 'Total Bandwidth' and 'Multimedia Bandwidth'.

AVAYA
Aura® System Manager 8.0

Users ▾ Elements ▾ Services ▾ Widgets ▾ Shortcuts ▾

Search 🔍 🔔 ☰ | admin

Home Routing x

Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults

Location Details Commit Cancel Help ?

General

* Name: Avaya SBCE
Notes: VMware Avaya SBCE

Dial Plan Transparency in Survivable Mode

Enabled: ☐

Listed Directory Number:

Associated CM SIP Entity:

Overall Managed Bandwidth

Managed Bandwidth Units: Kbit/sec ▾

Total Bandwidth:

Multimedia Bandwidth:

6.4. Adaptations

In order to improve interoperability with third party elements, Session Manager 8.1 incorporates the ability to use Adaptation modules to remove specific headers that are either Avaya proprietary or deemed excessive/unnecessary for non-Avaya elements.

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

Navigate to **Routing → Adaptations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Adaptation Name:** Enter an appropriate name.
- **Module Name:** Select the ***DigitConversionAdapter*** option.
- **Module Parameter Type:** Select ***Name-Value Parameter***.

Click **Add** to add the name and value parameters, as follows:

- **Name:** Enter ***eRHdrs***. This parameter will remove the specified headers from messages in the egress direction.
- **Value:** Enter ***“Alert-Info, P-Charging-Vector, AV-Global-Session-ID, AV-Correlation-ID, P-AV-Message-Id, P-Location, Endpoint-View”***
- Click **Commit** to save.

The screen below shows the adaptation created for the compliance test. This adaptation will later be applied to the SIP Entity corresponding to the Avaya SBCE. All other fields were left at their default values.

AVAYA
Aura® System Manager 8.0

Users ▾ Elements ▾ Services ▾ Widgets ▾ Shortcuts ▾

Search 🔍 | admin

Home Routing × Routing ×

Routing ▾

Domains

Locations

Conditions

Adaptations ▾

Adaptations

Regular Expression...

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns ▾

Regular Expressions

Adaptation Details Commit Cancel Help ?

General

* Adaptation Name: CM_Outbound_Header_Removal

* Module Name: DigitConversionAdapter ▾

Module Parameter Type: Name-Value Parameter ▾

Name	Value
eRHdrs	Alert-Info, P-Charging-Vector, AV-Global-Session-ID, AV-Correlation-ID, P-AV-Message-id,

Select : All, None

Egress URI Parameters:

Notes:

Digit Conversion for Incoming Calls to SM

0 Items Filter: Enable

Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
------------------	-----	-----	---------------	---------------	---------------	-------------------	-----------------	-------

6.5. SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to it, which includes Communication Manager and the Avaya SBCE. Navigate to **Routing** → **SIP Entities** in the left navigation pane and click on 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.
- **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity that is used for SIP signaling (see **Figure 1**).
- **Type:** Select *Session Manager* for Session Manager, *CM* for Communication Manager and *SIP Trunk* (or *Other*) for the Avaya SBCE.
- **Adaptation:** This field is only present if **Type** is not set to **Session Manager**. If Adaptations were to be created, here is where they would be applied to the entity.
- **Location:** Select the location that applies to the SIP Entity being created, defined in **Section 6.3**.
- **Time Zone:** Select the time zone for the location above.
- Click **Commit** to save.

The following screen shows the addition of the *Session Manager* SIP Entity for Session Manager. The IP address of the Session Manager Security Module is entered in the **FQDN or IP Address** field.

The screenshot displays the Avaya Aura System Manager 8.0 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The left sidebar shows the 'Routing' section with 'SIP Entities' highlighted. The main content area is titled 'SIP Entity Details' and contains a 'General' tab. The form fields are as follows:

- Name:** Session Manager
- IP Address:** 10.64.101.249
- SIP FQDN:** (empty)
- Type:** Session Manager
- Notes:** VMware Session Manager
- Location:** Session Manager
- Outbound Proxy:** (empty)
- Time Zone:** America/New_York
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty)
- SIP Link Monitoring:** Use Session Manager Configuration
- CRLF Keep Alive Monitoring:** CRLF Monitoring Disabled

The 'Commit' and 'Cancel' buttons are located at the top right of the form area.

The following screen shows the addition of the *Communication Manager Trunk 2* SIP Entity for Communication Manager. In order for Session Manager to send SIP service provider traffic on a separate entity link to Communication Manager, the creation of a separate SIP entity for Communication Manager is required. This SIP Entity should be different than the one created during the Session Manager installation, used by all other enterprise SIP traffic. The **FQDN or IP Address** field is set to the IP address of the “**procr**” interface in Communication Manager, as seen in **Section 5.3**. Select the location that applies to the SIP Entity being created, defined in **Section 6.3**. Select the **Time Zone**.

The screenshot displays the Avaya Aura System Manager 8.0 web interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 8.0', and tabs for 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. A search bar and a user profile 'admin' are on the right. The left sidebar shows a menu with 'Routing' selected, and 'SIP Entities' highlighted. The main content area is titled 'SIP Entity Details' with a 'General' tab. The form contains the following fields: 'Name' (Communication Manager Trunk 2), 'FQDN or IP Address' (10.64.101.241), 'Type' (CM), 'Notes' (Used for SP Testing), 'Adaptation' (dropdown), 'Location' (Communication Manager), 'Time Zone' (America/New_York), 'SIP Timer B/F (in seconds)' (4), 'Minimum TLS Version' (Use Global Setting), 'Credential name' (text field), 'Securable' (checkbox), and 'Call Detail Recording' (none). 'Commit' and 'Cancel' buttons are at the top right of the form.

* Name:	Communication Manager Trunk 2
* FQDN or IP Address:	10.64.101.241
Type:	CM
Notes:	Used for SP Testing
Adaptation:	
Location:	Communication Manager
Time Zone:	America/New_York
* SIP Timer B/F (in seconds):	4
Minimum TLS Version:	Use Global Setting
Credential name:	
Securable:	<input type="checkbox"/>
Call Detail Recording:	none

The following screen shows the addition of the *Avaya SBCE* SIP Entity for the Avaya SBCE:

- The **FQDN or IP Address** field is set to the IP address of the SBC private network interface (see **Figure 1**).
- On the **Adaptation** field, the adaptation module *CM_Outbound_Header_Removal* previously defined in **Section 6.4** was selected.
- Select the location that applies to the SIP Entity being created, defined in **Section 6.3**.
- Select the **Time Zone**.

The screenshot displays the Avaya Aura System Manager 8.0 interface. The top navigation bar includes the Avaya logo, version information, and various menu items like Users, Elements, Services, Widgets, and Shortcuts. A search bar and a user profile icon are also present. The left sidebar shows a tree view with 'Routing' selected, and 'SIP Entities' highlighted under it. The main content area is titled 'SIP Entity Details' and features a 'General' tab. The form contains the following fields and values:

- Name:** Avaya SBCE
- FQDN or IP Address:** 10.64.101.243
- Type:** SIP Trunk
- Notes:** VMware Avaya SBCE
- Adaptation:** CM_Outbound_Header_Removal
- Location:** Avaya SBCE
- Time Zone:** America/New_York
- SIP Timer B/F (in seconds):** 4
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty field)
- Securable:** ☐
- Call Detail Recording:** none

The 'Commit' and 'Cancel' buttons are located at the top right of the form area.

6.6. Entity Links

A SIP trunk between Session Manager and a telephony system is described by an Entity Link. Two Entity Links were created; an entity link to Communication Manager for use only by service provider traffic and an entity link to the Avaya SBCE. To add an Entity Link, navigate to **Routing → Entity Links** in the left navigation pane and click on the **New** button in the right pane (not shown). Fill in the following fields in the new row that is displayed:

- **Name:** Enter a descriptive name.
- **SIP Entity 1:** Select the Session Manager from the drop-down menu (**Section 6.5**).
- **Protocol:** Select the transport protocol used for this link (**Section 5.6**).
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end (**Section 5.6**).
- **SIP Entity 2:** Select the name of the other system from the drop-down menu (**Section 6.5**).
- **Port:** Port number on which the other system receives SIP requests from Session Manager (**Section 5.6**).
- **Connection Policy:** Select **Trusted** to allow calls from the associated SIP Entity.
- Click **Commit** to save.

The screen below shows the Entity Link to Communication Manager. The protocol and ports defined here must match the values used on the Communication Manager signaling group form in **Section 5.6**. *TLS* transport and port *5071* were used.

	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	DNS Override	Connection Policy	Deny New Service	Notes
<input type="checkbox"/>	*Session_Manager_Ch	*Q Session Manager	TLS	*5071	*Q Communication Manager Trunk 2	*5071	<input type="checkbox"/>	trusted	<input type="checkbox"/>	

The Entity Link to the Avaya SBCE is shown below; *TLS* transport and port *5061* were used.

The screenshot shows the Avaya Aura System Manager 8.0 interface. The top navigation bar includes 'Home' and 'Routing' (highlighted with a red box). The left sidebar contains a menu with 'Entity Links' highlighted (red box). The main content area is titled 'Entity Links' and contains a table with one item. The table columns are: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, DNS Override, Connection Policy, Deny New Service, and Notes. The row data is: Name: Session_Manager_AS, SIP Entity 1: Session Manager, Protocol: TLS, Port: 5061, SIP Entity 2: Avaya SBCE, Port: 5061, DNS Override: (unchecked), Connection Policy: trusted, Deny New Service: (unchecked), Notes: (empty). The 'Routing' and 'Entity Links' menu items are highlighted with red boxes. The table row is also highlighted with a red box.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	DNS Override	Connection Policy	Deny New Service	Notes
Session_Manager_AS	Session Manager	TLS	5061	Avaya SBCE	5061	<input type="checkbox"/>	trusted	<input type="checkbox"/>	

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.5**. Two routing policies were added; an incoming policy with Communication Manager as the destination and an outbound policy with the Avaya SBCE as the destination. To add a routing policy, navigate to **Routing → Routing Policies** in the left navigation pane and click on the **New** button in the right pane (not shown). The following screen is displayed:

- The following screens show the Routing Policies for Communication Manager and the Avaya SBCE.

HG; Reviewed:
SPOC 1/21/2020

AVAYA
Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ | Widgets ▾ Shortcuts ▾

Search [] Help ? admin

Home Routing

Dial Pattern Details

Commit Cancel

General

* Pattern: 001
 * Min: 13
 * Max: 13

Emergency Call: ☐

SIP Domain: avaya.lab.com ▼

Notes:

Originating Locations, Origination Dial Pattern Sets, and Routing Policies

	Add	Remove									Filter: Enable
1 Item											
<input type="checkbox"/> Originating Location Name ▲	Originating Location Notes	Origination Dial Pattern Set Name	Origination Dial Pattern Set Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes			
<input type="checkbox"/> Communication Manager	VMware Communication Manager			Avaya SBCE	0	<input type="checkbox"/>	Avaya SBCE	For outbound calls to SP via ASBCE			

Select : All, None

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Motto_CM81SBC80

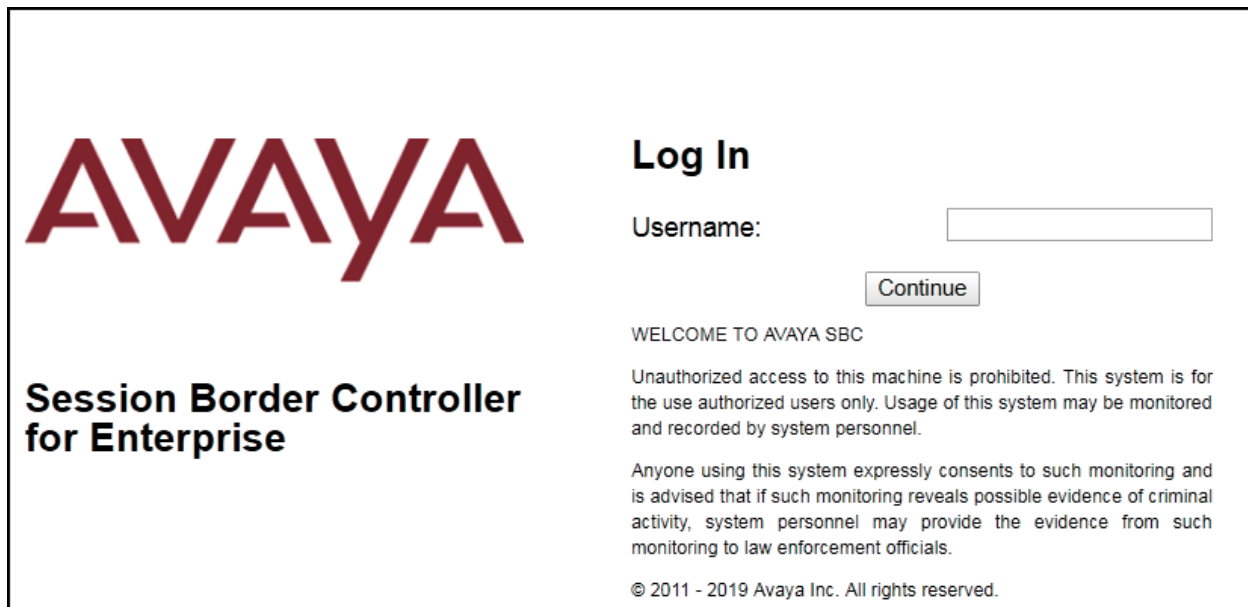
7. Configure Avaya Session Border Controller for Enterprise

This section describes the configuration of the Avaya SBCE. It is assumed that the initial installation of the Avaya SBCE, the assignment of the management interface IP Address and license installation have already been completed; hence these tasks are not covered in these Application Notes. For more information on the installation and initial provisioning of the Avaya SBCE consult the Avaya SBCE documentation in the **References** section.

Note - The configuration tasks required to support TLS transport for signaling and SRTP for media are beyond the scope of these Application Notes; hence it's not discussed in detail in this document. Consult reference [8] in the **References** section for additional information on this topic.

7.1. System Access

Access the Session Border Controller web management interface by using a web browser and entering the URL **https://<ip-address>**, where **<ip-address>** is the management IP address configured at installation. Log in using the appropriate credentials.



The screenshot shows the login interface of the Avaya Session Border Controller for Enterprise. On the left, the Avaya logo is displayed in a dark red color, with the text "Session Border Controller for Enterprise" below it. On the right, the "Log In" section contains a "Username:" label, a text input field, and a "Continue" button. Below the login fields, there is a "WELCOME TO AVAYA SBC" message, a disclaimer about unauthorized access, a consent statement, and a copyright notice: "© 2011 - 2019 Avaya Inc. All rights reserved."

Once logged in, on the top left of the screen, under **Device:** select the device being managed, *Avaya_SBCE* in the sample configuration.

Device: EMS | Alarms | Incidents | Status | Logs | Diagnostics | Users | Settings | Help | Log Out

Session Border Controller for Enterprise

AVAYA

EMS Dashboard

- Device Management
 - System Administration
 - Backup/Restore
 - Monitoring & Logging

Dashboard

Information	
System Time	03:45:41 PM EST Refresh
Version	8.0.1.0-10-17555
Build Date	Tue Jul 30 22:53:51 UTC 2019
License State	OK
Aggregate Licensing Overages	0
Peak Licensing Overage Count	0
Last Logged in at	12/09/2019 13:28:41 EST
Failed Login Attempts	0

Installed Devices
EMS
Avaya_SBCE

Active Alarms (past 24 hours)
None found.

Incidents (past 24 hours)
None found.

Notes
No notes found.

The left navigation pane contains the different available menu items used for the configuration of the Avaya SBCE. Verify that the status of the **License State** field is **OK**, indicating that a valid license is present. Contact an authorized Avaya sales representative if a license is needed.

Device: Avaya_SBCE | Alarms | Incidents | Status | Logs | Diagnostics | Users | Settings | Help | Log Out

Session Border Controller for Enterprise

AVAYA

EMS Dashboard

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

Dashboard

Information	
System Time	12:23:28 PM EST Refresh
Version	8.0.1.0-10-17555
Build Date	Tue Jul 30 22:53:51 UTC 2019
License State	OK
Aggregate Licensing Overages	0
Peak Licensing Overage Count	0
Last Logged in at	12/12/2019 13:38:24 EST
Failed Login Attempts	0

Installed Devices
EMS
Avaya_SBCE

Active Alarms (past 24 hours)
None found.

Incidents (past 24 hours)
Avaya_SBCE: No Subscriber Flow Matched

7.2. Device Management

To view current system information, select **Device Management** on the left navigation pane. In the reference configuration, the device named *Avaya_SBCE* is shown. The management IP address that was configured during installation is blurred out for security reasons, the current software version is shown. The management IP address needs to be on a subnet separate from the ones used in all other interfaces of the Avaya SBCE, segmented from all VoIP traffic. Verify that the **Status** is *Commissioned*, indicating that the initial installation process of the device has been previously completed, as shown on the screen below.

The screenshot displays the Avaya Session Border Controller for Enterprise (SBCE) management interface. At the top, a dark navigation bar includes 'Device: Avaya_SBCE' and links for Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header reads 'Session Border Controller for Enterprise' with the AVAYA logo. A left-hand navigation pane lists various system management options, with 'Device Management' highlighted in red. The central 'Device Management' section features tabs for Devices, Updates, SSL VPN, Licensing, and Key Bundles. The 'Devices' tab is active, showing a table with columns for Device Name, Management IP, Version, and Status. A single device, 'Avaya_SBCE', is listed with a blurred management IP, version '8.0.1.0-10-17555', and a status of 'Commissioned'. To the right of the status are action buttons: Reboot, Shutdown, Restart Application, View (highlighted in red), Edit, and Uninstall.

Device Name	Management IP	Version	Status
Avaya_SBCE	[Blurred]	8.0.1.0-10-17555	Commissioned

To view the network configuration assigned to the Avaya SBCE, click **View** on the screen above. The **System Information** window is displayed, containing the current device configuration and network settings. Note that **DNS configuration** is required for this solution.

System Information: Avaya_SBCE

X

General Configuration

Appliance NameAvaya_SBCE

Box TypeSIP

Deployment ModeProxy

Device Configuration

HA ModeNo

Two Bypass ModeNo

License Allocation

Standard Sessions
Requested: 20001000

Advanced Sessions
Requested: 20001000

Scopia Video Sessions
Requested: 500500

CES Sessions
Requested: 00

Transcoding Sessions
Requested: 00

CLID---

Encryption
Available: Yes☒

Network Configuration

IP	Public IP	Network Prefix or Subnet Mask	Gateway	Interface
10.64.101.243	10.64.101.243	255.255.255.0	10.64.101.1	A1
				A1
				A1
				B1
				B1
10.10.80.51	10.10.80.51	255.255.255.128	10.10.80.1	B1

DNS Configuration

Primary DNS8.8.8.8

Secondary DNS7.7.7.7

DNS LocationDMZ

DNS Client IP10.10.80.51

Management IP(s)

IP #1 (IPv4)

The highlighted IP addresses in the **System Information** screen shown above are the ones used for the SIP trunk to Motto Communications and are the ones relevant to these Application Notes. Other IP addresses assigned to the Avaya SBCE **A1** and **B1** interfaces are used to support remote workers and other SIP trunks, and they are not discussed in this document. Also note that for security purposes, any public IP addresses used during the compliance test have been masked in this document.

In the reference configuration, the private interface of the Avaya SBCE (10.64.101.243) was used to connect to the enterprise network, while its public interface (10.10.80.51) was used to connect to the public network. See **Figure 1**.

On the **License Allocation** area of the **System Information**, verify that the number of **Standard Sessions** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise. The number of sessions and encryption features are primarily controlled by the license file installed.

7.3. TLS Management

Transport Layer Security (TLS) is a standard protocol that is used extensively to provide a secure channel by encrypting communications over IP networks. It enables clients to authenticate servers or, optionally, servers to authenticate clients. UC-Sec security products utilize TLS primarily to facilitate secure communications with remote servers.

It is assumed that generation and installation of certificates and the creation of TLS Profiles on the Avaya SBCE have been previously completed, as it's not discussed in this document. Refer to item [8] in **Section 11**.

7.4. Network Management

The network configuration parameters should have been previously specified during installation of the Avaya SBCE. In the event that changes need to be made to the network configuration, they can be entered here.

Select **Network Management** from the **Network & Flows** on the left-side menu. On the **Networks** tab, verify or enter the network information as needed.

Note that in the configuration used during the compliance test, the IP addresses assigned to the private (**10.64.101.243**) and public (**10.10.80.51**) sides of the Avaya SBCE are the ones relevant to these Application Notes.

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

Session Border Controller for Enterprise

AVAYA

EMS Dashboard

Device Management

Backup/Restore

▸ System Parameters

▸ Configuration Profiles

▸ Services

▸ Domain Policies

▸ TLS Management

▸ **Network & Flows**

Network Management

 Media Interface

 Signaling Interface

Network Management

Interfaces **Networks**

Add

Name	Gateway	Subnet Mask / Prefix Length	Interface	IP Address	
Network_A1	10.64.101.1	255.255.255.0	A1	10.64.101.243	Edit Delete
Network_B1	10.10.80.1	255.255.255.128	B1	10.10.80.51	Edit Delete

On the **Interfaces** tab, verify the **Administrative Status** is **Enabled** for the **A1** and **B1** interfaces. Click the buttons under the **Status** column if necessary to enable the interfaces.

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

Session Border Controller for Enterprise

AVAYA

EMS Dashboard

Device Management

Backup/Restore

▸ System Parameters

▸ Configuration Profiles

▸ Services

▸ Domain Policies

▸ TLS Management

▸ **Network & Flows**

Network Management

 Media Interface

 Signaling Interface

Network Management

Interfaces Networks

Add VLAN

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

7.5. Media Interfaces

Media Interfaces were created to specify the IP address and port range in which the Avaya SBCE will accept media streams on each interface. Packets leaving the interfaces of the Avaya SBCE will advertise this IP address, and one of the ports in this range as the listening IP address and port in which it will accept media from the Call Server or the trunk server.

To add the Media Interface in the enterprise direction, select **Media Interface** from the **Network & Flows** menu on the left-hand side, click the **Add** button (not shown).

- On the **Add Media Interface** screen, enter an appropriate **Name** for the Media Interface.

- Under **IP Address**, select from the drop-down menus the network and IP address to be associated with this interface.
- The **Port Range** was left at the default values of **35000-40000**.
- Click **Finish**.

The screenshot shows the 'Add Media Interface' dialog box with the following fields:

- Name:** Private_med
- IP Address:** Network_A1 (A1, VLAN 0) (selected from a dropdown menu)
- IP Address:** 10.64.101.243 (selected from a dropdown menu)
- Port Range:** 35000 - 40000
- Finish:** A button at the bottom right.

A red rectangular box highlights the Name, IP Address dropdowns, and Port Range fields.

A Media Interface facing the public side was similarly created with the name **Public_med**, as shown below.

- Under **IP Address**, the network and IP address to be associated with this interface was selected.
- The **Port Range** was left at the default values.
- Click **Finish**.

The screenshot shows the 'Add Media Interface' dialog box with the following fields:

- Name:** Public_med
- IP Address:** Network_B1 (B1, VLAN 0) (selected from a dropdown menu)
- IP Address:** 10.10.80.51 (selected from a dropdown menu)
- Port Range:** 35000 - 40000
- Finish:** A button at the bottom right.

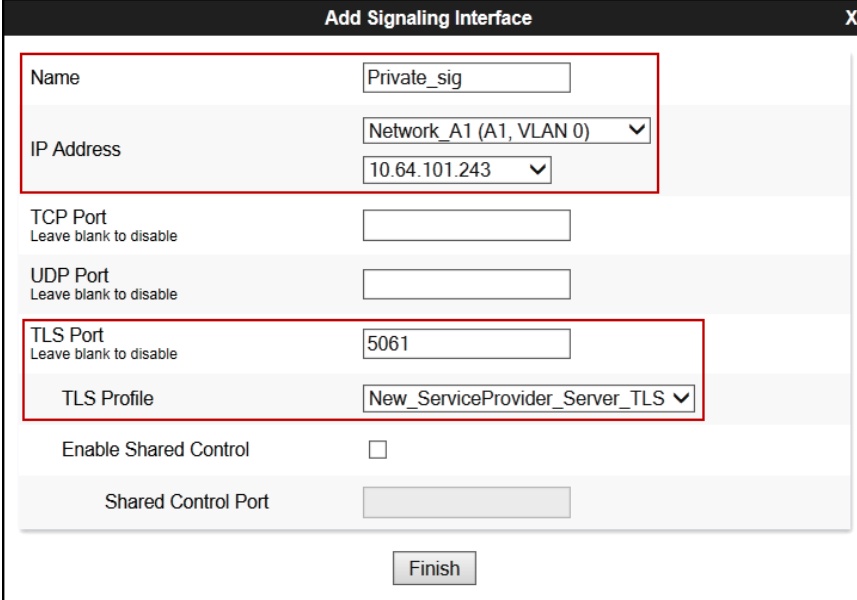
A red rectangular box highlights the Name, IP Address dropdowns, and Port Range fields.

7.6. Signaling Interfaces

Signaling Interfaces are created to specify the IP addresses and ports in which the Avaya SBCE will listen for signaling traffic in the connected networks.

To add the Signaling Interface in the enterprise direction, select **Signaling Interface** from the **Network & Flows** menu on the left-hand side, click the **Add** button (not shown).

- On the **Add Signaling Interface** screen, enter an appropriate **Name** for the interface.
- Under **IP Address**, select from the drop-down menus the network and IP address to be associated with this interface.
- Enter **5061** for **TLS Port**, since TLS port 5061 is used to listen for signaling traffic from Session Manager in the sample configuration, as defined in **Section 6.6**.
- Select a **TLS Profile**.
- Click **Finish**.



The screenshot shows the 'Add Signaling Interface' configuration window. The window has a title bar with 'Add Signaling Interface' and a close button 'X'. The form contains several fields and a 'Finish' button at the bottom. Two red boxes highlight specific sections: the first box encloses the 'Name' field (containing 'Private_sig'), the 'IP Address' dropdown (showing 'Network_A1 (A1, VLAN 0)' and '10.64.101.243'), and the 'TCP Port' field (which is empty). The second box encloses the 'TLS Port' field (containing '5061') and the 'TLS Profile' dropdown (showing 'New_ServiceProvider_Server_TLS'). Other fields include 'UDP Port' (empty), 'Enable Shared Control' (checkbox), and 'Shared Control Port' (empty).

Name	Private_sig
IP Address	Network_A1 (A1, VLAN 0) 10.64.101.243
TCP Port	
UDP Port	
TLS Port	5061
TLS Profile	New_ServiceProvider_Server_TLS
Enable Shared Control	<input type="checkbox"/>
Shared Control Port	
Finish	

A second Signaling Interface with the name **Public_sig** was similarly created in the service provider's direction.

- Under **IP Address**, select from the drop-down menus the network and IP address to be associated with this interface.
- Enter **5060** for **UDP Port**, since UDP port 5060 is used to listen for signaling traffic from Motto Communications in the sample configuration.
- Click **Finish**.

Add Signaling Interface X

Name: Public_sig

IP Address: Network_B1 (B1. VLAN 0) 10.10.80.51

TCP Port: Leave blank to disable

UDP Port: 5060 Leave blank to disable

TLS Port: Leave blank to disable

TLS Profile: None

Enable Shared Control: ☐

Shared Control Port:

Finish

7.7. Server Interworking

Interworking Profile features are configured to facilitate the interoperability between the enterprise SIP-enabled solution (Call Server) and the SIP trunk service provider (Trunk Server).

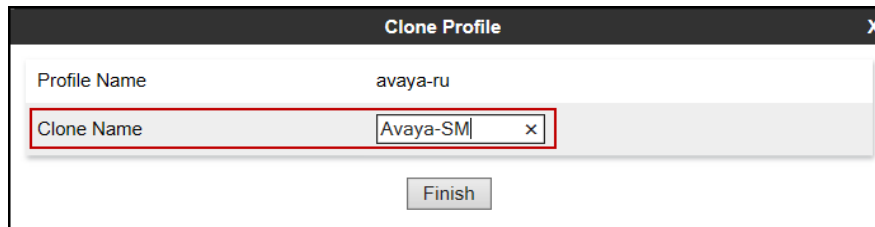
7.7.1. Server Interworking Profile – Enterprise

Interworking profiles can be created by cloning one of the pre-defined default profiles, or by adding a new profile. To configure the interworking profile in the enterprise direction, select **Configuration Profiles → Server Interworking** on the left navigation pane. Under **Interworking Profiles**, select *avaya-ru* from the list of pre-defined profiles. Click **Clone** (not shown).

The screenshot displays the configuration interface for a Session Border Controller for Enterprise. The top navigation bar includes 'Device: Avaya_SBCE', 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', and 'Settings'. The main title is 'Session Border Controller for Enterprise'. The left navigation pane shows a tree structure with 'Configuration Profiles' expanded, and 'Server Interworking' selected. The main content area is titled 'Interworking Profiles: avaya-ru' and features an 'Add' button. A warning message states: 'It is not recommended to edit the defaults. Try cloning or adding a new profile instead.' Below this, there are tabs for 'General', 'Timers', 'Privacy', 'URI Manipulation', 'Header Manipulation', and 'Advanced'. The 'General' tab is active, showing a table of configuration parameters. An 'Edit' button is located at the bottom right of the configuration area.

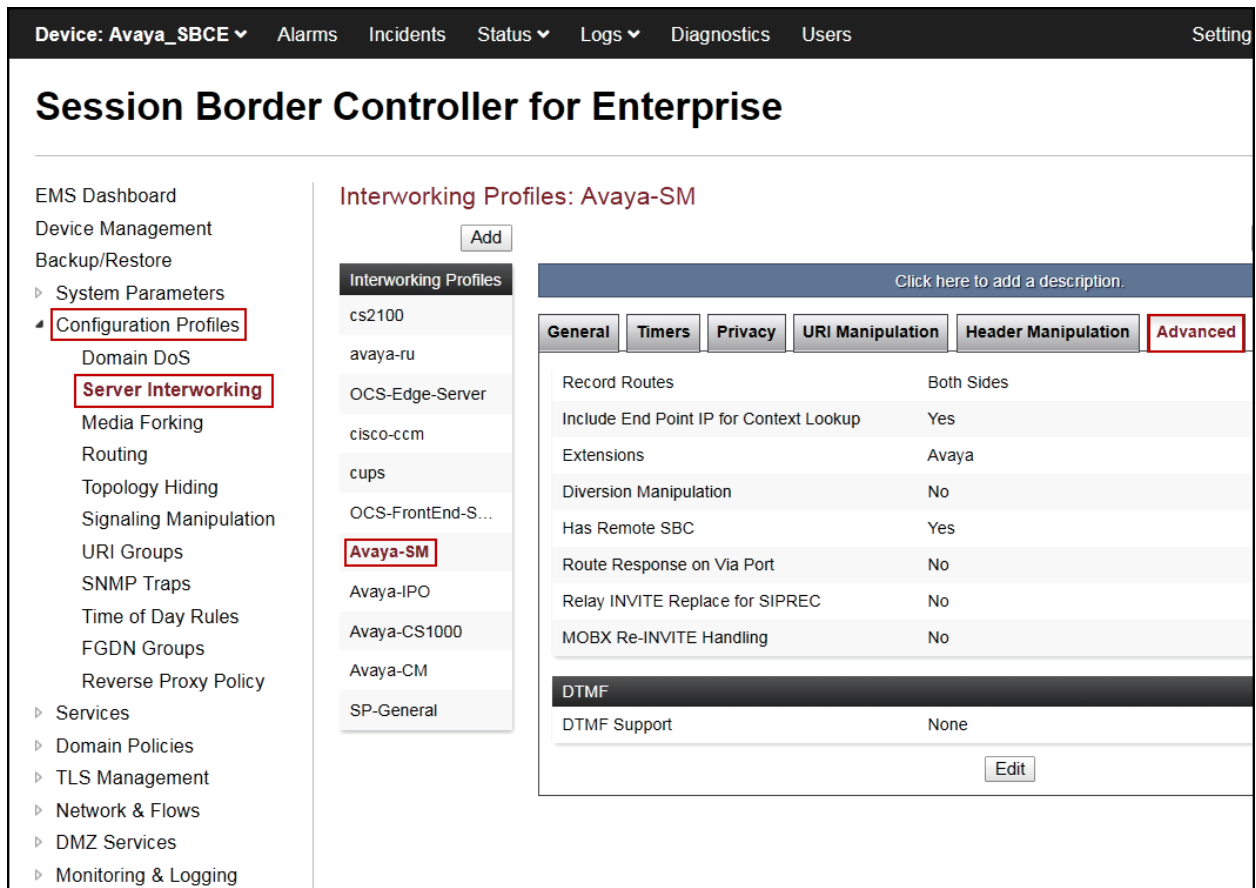
General	
Hold Support	NONE
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	No
URI Group	None
Send Hold	No
Delayed Offer	Yes
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
Re-Invite Handling	No
Prack Handling	No
Allow 18X SDP	No
T.38 Support	No
URI Scheme	SIP
Via Header Format	RFC3261

- Enter a descriptive name for the cloned profile.
- Click **Finish**.



The image shows a 'Clone Profile' dialog box. It has a title bar with 'Clone Profile' and a close button 'X'. Inside, there are two input fields: 'Profile Name' with the value 'avaya-ru' and 'Clone Name' with the value 'Avaya-SM'. The 'Clone Name' field is highlighted with a red border. Below the fields is a 'Finish' button.

The **Advanced** tab settings are shown on the screen below:

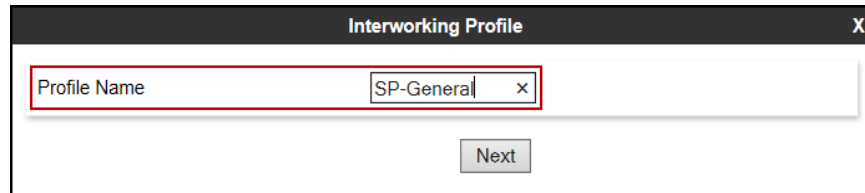


The image shows the 'Session Border Controller for Enterprise' web interface. The top navigation bar includes 'Device: Avaya_SBCE', 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', and 'Setting'. The main heading is 'Session Border Controller for Enterprise'. On the left is a sidebar menu with categories like 'EMS Dashboard', 'Device Management', 'Backup/Restore', 'System Parameters', 'Configuration Profiles', 'Domain DoS', 'Media Forking', 'Routing', 'Topology Hiding', 'Signaling Manipulation', 'URI Groups', 'SNMP Traps', 'Time of Day Rules', 'FGDN Groups', 'Reverse Proxy Policy', 'Services', 'Domain Policies', 'TLS Management', 'Network & Flows', 'DMZ Services', and 'Monitoring & Logging'. The 'Configuration Profiles' section is expanded, showing 'Server Interworking' as a sub-category. Under 'Server Interworking', there is a list of profiles: 'cs2100', 'avaya-ru', 'OCS-Edge-Server', 'cisco-ccm', 'cups', 'OCS-FrontEnd-S...', 'Avaya-SM' (highlighted with a red border), 'Avaya-IPO', 'Avaya-CS1000', 'Avaya-CM', and 'SP-General'. An 'Add' button is above this list. The main content area is titled 'Interworking Profiles: Avaya-SM'. It has a tabbed interface with 'General', 'Timers', 'Privacy', 'URI Manipulation', 'Header Manipulation', and 'Advanced' (highlighted with a red border). The 'Advanced' tab shows settings for 'Record Routes' (Both Sides), 'Include End Point IP for Context Lookup' (Yes), 'Extensions' (Avaya), 'Diversion Manipulation' (No), 'Has Remote SBC' (Yes), 'Route Response on Via Port' (No), 'Relay INVITE Replace for SIPREC' (No), and 'MOBX Re-INVITE Handling' (No). Below this is a 'DTMF' section with 'DTMF Support' set to 'None'. An 'Edit' button is at the bottom right.

7.7.2. Server Interworking Profile – Service Provider

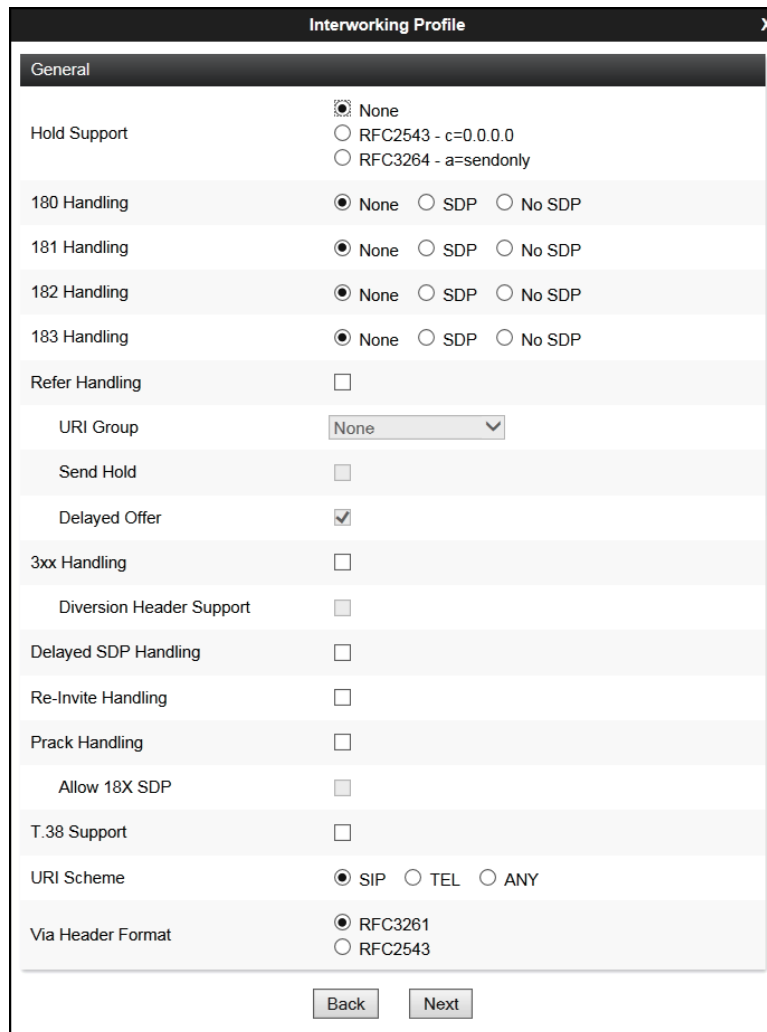
A second interworking profile in the direction of the SIP trunk was created, by adding a new profile in this case. Select **Global Profiles → Server Interworking** on the left navigation pane and click **Add** (not shown).

- Enter a descriptive name for the new profile.
- Click **Next**.



The screenshot shows a dialog box titled "Interworking Profile" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Profile Name" which contains the text "SP-General". To the right of the input field is a small "x" icon. Below the input field is a "Next" button.

- Click **Next** until the last tab is reached then click **Finish** on the last tab leaving remaining fields with default values (not shown).



The screenshot shows the "Interworking Profile" configuration page, General tab. The page lists various settings with radio buttons and checkboxes. The "Delayed Offer" checkbox is checked. The "URI Scheme" is set to SIP. The "Via Header Format" is set to RFC3261. The "Back" and "Next" buttons are at the bottom.

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

The **Advanced** tab settings are shown on the screen below:

Device: Avaya_SBCE ▾

Alarms

Incidents

Status ▾

Logs ▾

Diagnostics

Users

Settings ▾

Session Border Controller for Enterprise

EMS Dashboard

Device Management

Backup/Restore

▸ System Parameters

▾ Configuration Profiles

Domain DoS

Server Interworking

Media Forking

Routing

Topology Hiding

Signaling Manipulation

URI Groups

SNMP Traps

Time of Day Rules

FGDN Groups

Reverse Proxy Policy

▸ Services

▸ Domain Policies

▸ TLS Management

▸ Network & Flows

Interworking Profiles: SP-General

Add

Interworking Profiles

cs2100

avaya-ru

OCS-Edge-Server

cisco-ccm

cups

OCS-FrontEnd-...

Avaya-SM

Avaya-IPO

Avaya-CS1000

Avaya-CM

SP-General

Click here to add a description.

General

Timers

Privacy

URI Manipulation

Header Manipulation

Advanced

Record Routes

Both Sides

Include End Point IP for Context Lookup

No

Extensions

None

Diversion Manipulation

No

Has Remote SBC

Yes

Route Response on Via Port

No

Relay INVITE Replace for SIPREC

No

MOBX Re-INVITE Handling

No

DTMF

DTMF Support

None

Edit

7.8. Signaling Manipulation

The Signaling Manipulation feature of the Avaya SBCE allows an administrator to perform granular header manipulations on the headers of the SIP messages, which sometimes is not possible by direct configuration on the web interface. This ability to configure header manipulation in such a highly flexible manner is achieved by the use of a proprietary scripting language called SigMa.

The script can be created externally as a regular text file and imported in the Signaling Manipulation screen, or they can be written directly in the page using the embedded Sigma Editor. In the reference configuration, the Editor was used. A detailed description of the structure of the SigMa scripting language and details on its use is beyond the scope of these Application Notes. Consult reference [8] in the **References** section for more information on this topic.

A single Sigma script was created during the compliance test to correct the following interoperability issues (refer to **Section 2.2**):

- Remove + sign from SIP messages before sending to Motto Communications.
- Remove unwanted “gsid” and “epv” parameter from being sent to the Service Provider in the Contact header.
- Remove the P-Location parameter from being sent to the Service Provider.
- Change the Diversion header scheme from SIPS to SIP.
- Remove unwanted xml element information from the SDP in SIP messages sent to the Service Provider.

The scripts will later be applied to the Server Configuration profile corresponding to the Service Provider (toward Motto Communications) in **Section 7.9.2**.

To create the SigMa script on the left navigation pane, select **Configuration Profiles** → **Signaling Manipulation**. From the **Signaling Manipulation Scripts** list, select **Add**.

- For **Title** enter a name, the name *Motto_Sigma* was chosen in this example.
- Copy and paste the entire script shown below.
- Click **Save**.

within session "ALL"

```
{
act on request where %DIRECTION="OUTBOUND" and
%ENTRY_POINT="POST_ROUTING"
{

//Removes + signs from headers
%HEADERS["To"][1].URI.USER.regex_replace("\+", "");
%HEADERS["From"][1].URI.USER.regex_replace("\+", "");
%HEADERS["Contact"][1].URI.USER.regex_replace("\+", "");
%HEADERS["Diversion"][1].URI.USER.regex_replace("\+", "");
%HEADERS["P-Asserted-Identity"][1].URI.USER.regex_replace("\+", "");
```

```
//Remove gsid and epv parameters from Contact header.  
remove(%HEADERS["Contact"][1].URI.PARAMS["gsid"]);  
remove(%HEADERS["Contact"][1].URI.PARAMS["epv"]);  
  
//Remove P-Location parameter.  
remove(%HEADERS["P-Location"][1]);  
  
//Changes the Diversion header scheme from SIPS to SIP.  
%HEADERS["Diversion"][1].regex_replace("sips","sip");  
  
//Remove unwanted xml element information from the SDP in SIP messages sent to the Service  
Provider.  
remove(%BODY[1]);  
  
}  
}
```

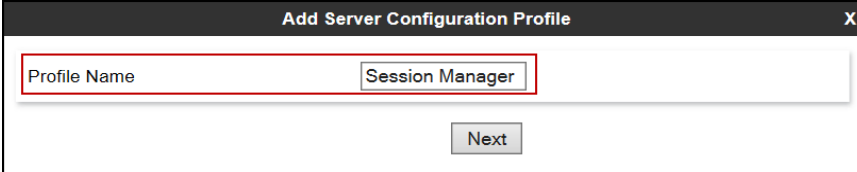
7.9. Server Configuration

Server Profiles are created to define the parameters for the Avaya SBCE peers; Session Manager (Call Server) at the enterprise and Motto Communications SIP Proxy (Trunk Server).

7.9.1. Server Configuration Profile – Enterprise

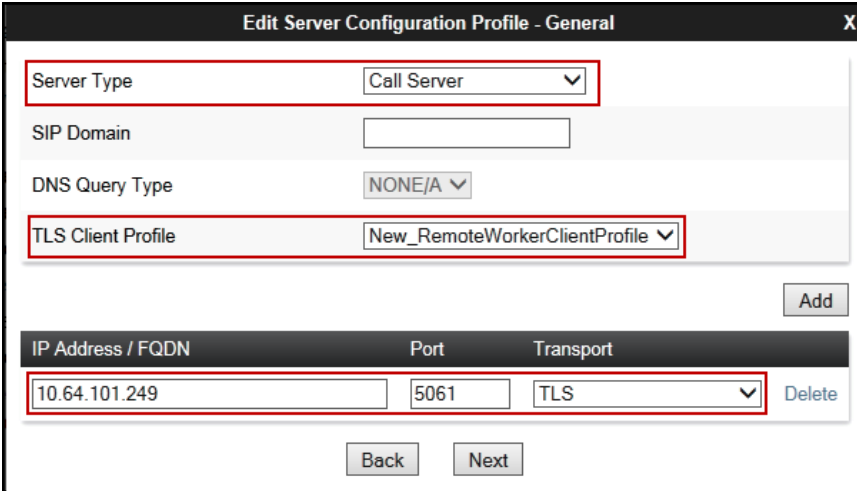
From the **Services** menu on the left-hand navigation pane, select **SIP Servers** and click the **Add** button (not shown) to add a new profile for the Call Server.

- Enter an appropriate **Profile Name** similar to the screen below.
- Click **Next**.



The screenshot shows a dialog box titled "Add Server Configuration Profile". It has a close button (X) in the top right corner. The main area contains a text input field labeled "Profile Name" which has "Session Manager" entered. Below the field is a "Next" button.

- On the **Edit SIP Server Profile – General** tab select **Call Server** from the drop-down menu under the **Server Type**.
- On the **IP Addresses / FQDN** field, enter the IP address of the Session Manager Security Module (**Section 6.5**).
- Enter **5061** under **Port** and select **TLS** for **Transport**. The transport protocol and port selected here must match the values defined for the Entity Link to the Session Manager previously created in **Section 6.6**.
- Select a **TLS Profile**.
- Click **Next**.



The screenshot shows a dialog box titled "Edit Server Configuration Profile - General". It has a close button (X) in the top right corner. The form contains several fields: "Server Type" (dropdown menu set to "Call Server"), "SIP Domain" (text field), "DNS Query Type" (dropdown menu set to "NONE/A"), and "TLS Client Profile" (dropdown menu set to "New_RemoteWorkerClientProfile"). Below these is an "Add" button. A table-like section contains three columns: "IP Address / FQDN", "Port", and "Transport". The "IP Address / FQDN" field contains "10.64.101.249", the "Port" field contains "5061", and the "Transport" dropdown menu is set to "TLS". There is a "Delete" button next to the "Transport" field. At the bottom are "Back" and "Next" buttons.

- Click **Next** until the **Add Server Configuration Profile – Advanced** tab is reached (not shown).
- On the **Add Server Configuration Profile – Advanced** tab:
 - Check **Enable Grooming**.
 - Select **Avaya-SM** from the **Interworking Profile** drop-down menu (**Section 7.7.1**).
- Click **Finish**.

Add SIP Server Profile - Advanced

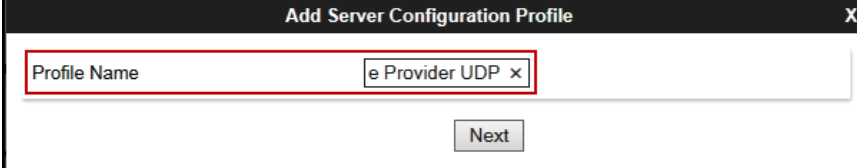
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input checked="" type="checkbox"/>
Interworking Profile	Avaya-SM ▼
Signaling Manipulation Script	None ▼
Securable	<input type="checkbox"/>
Enable FGDN	<input type="checkbox"/>
TCP Failover Port	5060
TLS Failover Port	5061
Tolerant	<input type="checkbox"/>
URI Group	None ▼

Back Finish

7.9.2. Server Configuration Profile – Service Provider

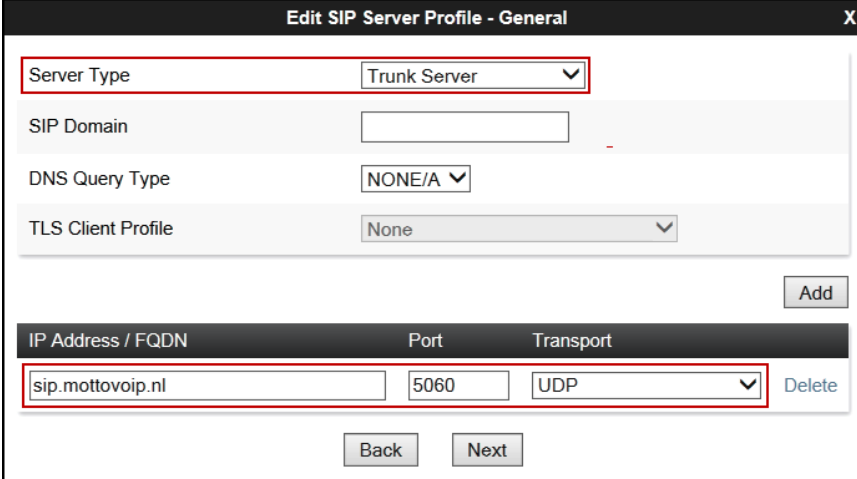
Similarly, to add the profile for the Trunk Server, click the **Add** button on the **Server Configuration** screen (not shown).

- Enter an appropriate **Profile Name** similar to the screen below (*Service Provider UDP* was used).
- Click **Next**.



The screenshot shows a dialog box titled "Add Server Configuration Profile". It has a close button (X) in the top right corner. The main area contains a text input field labeled "Profile Name" with the text "e Provider UDP" and a small 'x' icon to its right. Below the input field is a "Next" button.

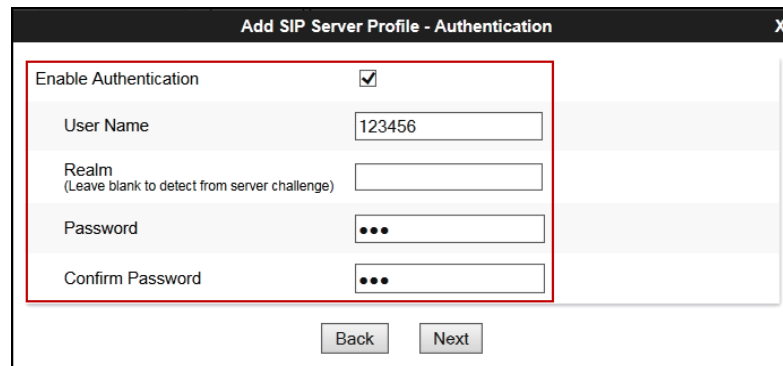
- On the **Edit Server Configuration Profile - General** Tab select *Trunk Server* from the drop-down menu for the **Server Type**.
- On the **IP Addresses / FQDN** field, enter *sip.mottovoip.nl* (Motto Communications SIP proxy server FQDN). This information was provided by Motto Communications.
- Enter *5060* under **Port** and select **UDP** for **Transport**.
- Click **Next**.



The screenshot shows a dialog box titled "Edit SIP Server Profile - General". It has a close button (X) in the top right corner. The main area contains several fields: "Server Type" (dropdown menu set to "Trunk Server"), "SIP Domain" (text input field), "DNS Query Type" (dropdown menu set to "NONE/A"), and "TLS Client Profile" (dropdown menu set to "None"). Below these fields is an "Add" button. At the bottom, there is a table with three columns: "IP Address / FQDN", "Port", and "Transport". The table contains one row with the values "sip.mottovoip.nl", "5060", and "UDP". A red box highlights the "IP Address / FQDN", "Port", and "Transport" fields. To the right of the table is a "Delete" button. Below the table are "Back" and "Next" buttons.

On the **Add SIP Server Profile - Authentication** tab:

- Check the **Enable Authentication** box.
- Enter the **User Name** credential provided by the service provider for SIP trunk registration.
- Leave the **Realm** blank.
- Enter **Password** credential provided by the service provider for SIP trunk registration.
- Click **Next**.



The screenshot shows a window titled "Add SIP Server Profile - Authentication". Inside the window, there is a red rectangular box highlighting the following fields:

- Enable Authentication**: A checkbox that is checked.
- User Name**: A text input field containing "123456".
- Realm**: A text input field containing the text "(Leave blank to detect from server challenge)".
- Password**: A text input field containing three dots.
- Confirm Password**: A text input field containing three dots.

Below the highlighted fields, there are two buttons: "Back" and "Next".

- Click **Next** on the **Add Server Configuration Profile - Heartbeat** window (not shown).

On the **Add SIP Server Profile - Registration** tab:

- Check the **Register with All Servers** box.
- **Frequency**: Enter the amount of time (in seconds) between REGISTER messages that will be sent from the enterprise to the Service Provider Proxy Server to refresh the registration binding of the SIP trunk. This value should be chosen in consultation with the service provider. **60** seconds was the value used during the compliance test.
- The **From URI** and **To URI** entries for the REGISTER messages are built using the following:
 - **From URI**: Use the **User Name** entered above in the **Authentication** screen (**123456**) and the Service Provider's SIP proxy server FQDN (**sip.mottovoip.nl**), as shown on the screen below.
 - **To URI**: Use the **User Name** entered above in the **Authentication** screen (**123456**) and the Service Provider's SIP proxy server FQDN (**sip.mottovoip.nl**), as shown on the screen below.
 - Click **Next**.

The screenshot shows the 'Add SIP Server Profile - Registration' window. It contains the following fields and controls:

- Register with All Servers**: A checkbox that is checked.
- Register with Priority Server**: A checkbox that is unchecked.
- Refresh Interval**: A text box containing '60' followed by the label 'seconds'.
- From URI**: A text box containing '123456@sip.mottovoip.'.
- To URI**: A text box containing '.56@sip.mottovoip.nl' with a clear button 'x' to its right.
- Back** and **Next** buttons at the bottom.

A red rectangle highlights the 'Register with All Servers', 'Refresh Interval', 'From URI', and 'To URI' fields.

Click **Next** on the **Add SIP Server Profile - Ping** window (not shown).

On the **Add SIP Server Profile - Advanced** window:

- Uncheck **Enable Grooming**.
- Select **SP-General** from the **Interworking Profile** drop-down menu (**Section 7.7.2**).
- Select the **Motto_Sigma** from the **Signaling Manipulation Script** drop down menu (**Sections 7.8**).
- Click **Finish**.

Add SIP Server Profile - Advanced

Enable DoS Protection ☐

Enable Grooming ☐

Interworking Profile SP-General

Signaling Manipulation Script Motto_Sigma

Securable ☐

Enable FGDN ☐

TCP Failover Port 5060

TLS Failover Port 5061

Tolerant ☐

URI Group None

Back Finish

7.10.Routing

Routing profiles define a specific set of routing criteria that is used, in addition to other types of domain policies, to determine the path that the SIP traffic will follow as it flows through the Avaya SBCE interfaces. Two Routing Profiles were created in the test configuration, one for inbound calls, with Session Manager as the destination, and the second one for outbound calls, which are routed to the service provider SIP trunk.

7.10.1. Routing Profile – Enterprise

To create the inbound route, select the **Routing** tab from the **Configuration Profiles** menu on the left-hand side and select **Add** (not shown).

- Enter an appropriate **Profile Name** similar to the example below.
- Click **Next**.

Routing Profile

Profile Name Route_to_SM

Next

- On the **Routing Profile** tab, click the **Add** button to enter the next-hop address.
- Under **Priority/Weight** enter **1**.
- Under **SIP Server Profile**, select **Session Manager**. The **Next Hop Address** field will be populated with the IP address, port and protocol defined for the Session Manager Server Configuration Profile in **Section 7.9.1**.
- Defaults were used for all other parameters.
- Click **Finish**.

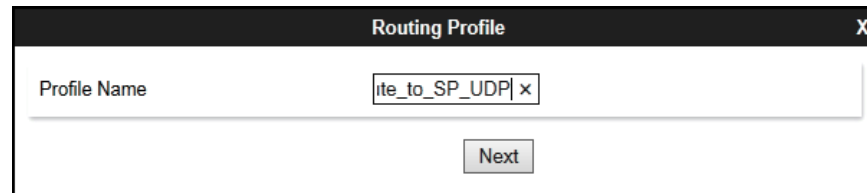
The screenshot shows the 'Routing Profile' configuration window. The main configuration area includes fields for URI Group, Load Balancing (set to Priority), Transport (set to None), LDAP Server Profile (set to None), Matched Attribute Priority (checked), Next Hop Priority (checked), Ignore Route Header (unchecked), ENUM (unchecked), and ENUM Suffix. Below this is a table for adding routing entries. The table has columns for Priority / Weight, LDAP Search Attribute, LDAP Search Regex Pattern, LDAP Search Regex Result, SIP Server Profile, Next Hop Address, and Transport. A new entry is being added with Priority / Weight '1', SIP Server Profile 'Session Manager', and Next Hop Address '10.64.101.249:5061 (TLS)'. The 'Add' button is highlighted, and the 'Finish' button is visible at the bottom.

Priority / Weight	LDAP Search Attribute	LDAP Search Regex Pattern	LDAP Search Regex Result	SIP Server Profile	Next Hop Address	Transport	
1				Session Manager	10.64.101.249:5061 (TLS)	None	Delete

7.10.2. Routing Profile – Service Provider

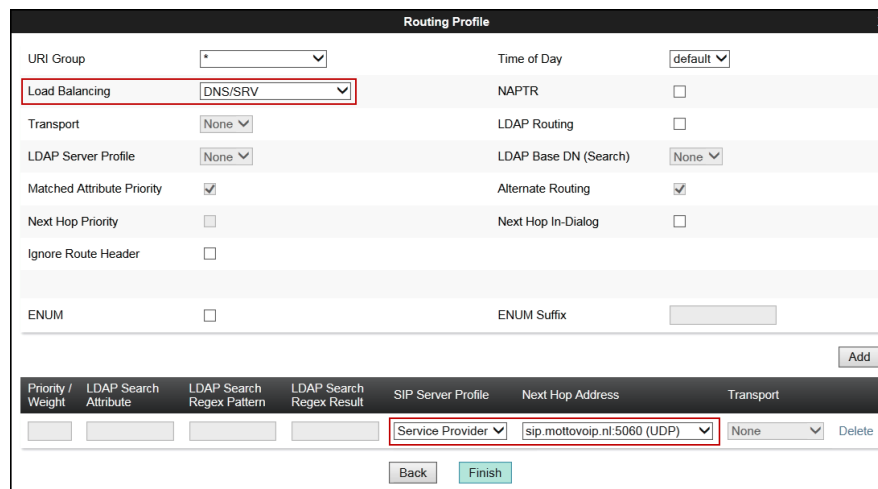
Back at the **Routing** tab, select **Add** (not shown) to repeat the process in order to create the outbound route.

- Enter an appropriate **Profile Name** similar to the example below (*Route_to_SP_UDP* was used).
- Click **Next**.



The image shows a 'Routing Profile' dialog box. It has a title bar with 'Routing Profile' and a close button 'X'. Inside, there is a text input field for 'Profile Name' containing 'ite_to_SP_UDP' with a small 'x' icon to its right. Below the input field is a 'Next' button.

- Under **Load Balancing** select *DNS/SRV*.
- Click the **Add** button to enter the next-hop address.
- Under **SIP Server Profile**, select *Service Provider UDP*.
- The **Next Hop Address** is populated automatically with *sip.mottovoip.nl:5060 (UDP)*. Motto Communications SIP Proxy FQDN, Port and Transport, Server Configuration Profile defined in **Section 7.9.2**.
- Click **Finish**



The image shows a 'Routing Profile' dialog box with various configuration options. The 'Load Balancing' dropdown is set to 'DNS/SRV'. The 'SIP Server Profile' dropdown is set to 'Service Provider'. The 'Next Hop Address' dropdown is set to 'sip.mottovoip.nl:5060 (UDP)'. The 'Transport' dropdown is set to 'None'. The 'Add' button is visible. Below the configuration options is a table with columns: Priority / Weight, LDAP Search Attribute, LDAP Search Regex Pattern, LDAP Search Regex Result, SIP Server Profile, Next Hop Address, Transport, and Delete. The table contains one row with the following values: (empty), (empty), (empty), (empty), Service Provider, sip.mottovoip.nl:5060 (UDP), None, and Delete. The 'Back' and 'Finish' buttons are at the bottom.

Priority / Weight	LDAP Search Attribute	LDAP Search Regex Pattern	LDAP Search Regex Result	SIP Server Profile	Next Hop Address	Transport	Delete
				Service Provider	sip.mottovoip.nl:5060 (UDP)	None	Delete

7.11.Topology Hiding

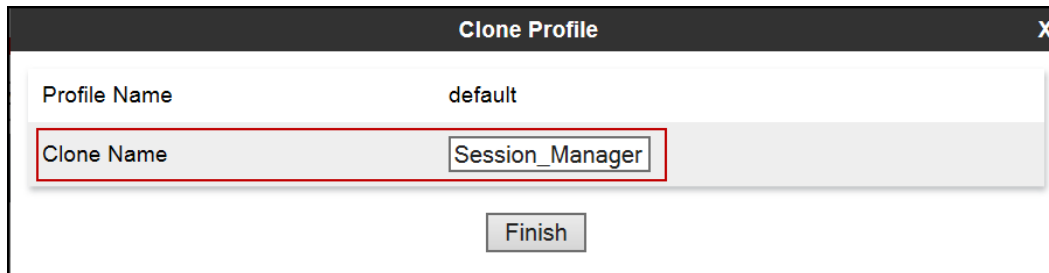
Topology Hiding is a security feature that allows the modification of several SIP headers, preventing private enterprise network information from being propagated to the untrusted public network.

Topology Hiding can also be used as an interoperability tool to adapt the host portion in the SIP headers to the IP addresses or domains expected on the service provider and the enterprise networks. For the compliance test, the default Topology Hiding Profile was cloned and modified accordingly. Only the minimum configuration required to achieve interoperability on the SIP trunk was performed. Additional steps can be taken in this section to further mask the information that is sent from the enterprise to the public network.

7.11.1. Topology Hiding Profile – Enterprise

To add the Topology Hiding Profile in the enterprise direction, select **Topology Hiding** from the **Configuration Profiles** menu on the left-hand side, select *default* from the list of pre-defined profiles and click the **Clone** button (not shown).

- Enter a **Clone Name** such as the one shown below.
- Click **Finish**.



The screenshot shows a 'Clone Profile' dialog box. It has a title bar with 'Clone Profile' and a close button 'X'. Inside, there are two input fields: 'Profile Name' with the value 'default' and 'Clone Name' with the value 'Session_Manager'. The 'Clone Name' field is highlighted with a red rectangular border. Below these fields is a 'Finish' button.

On the newly cloned *Session_Manager* profile screen, click the **Edit** button (not shown).

- For the, **From**, **To** and **Request-Line** headers, select **Overwrite** in the **Replace Action** column and enter the enterprise SIP domain *avaya.lab.com*, in the **Overwrite Value** column of these headers, as shown below. This is the domain known by Session Manager, defined in **Section 6.2**.
- Default values were used for all other fields.
- Click **Finish**.

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

Finish

7.11.2. Topology Hiding Profile – Service Provider

To add the Topology Hiding Profile in the service provider direction, select **Topology Hiding** from the **Global Profiles** menu on the left-hand side, select *default* from the list of pre-defined profiles and click the **Clone** button (not shown).

- Enter a **Clone Name** such as the one shown below.
- Click **Finish**.



The screenshot shows a 'Clone Profile' dialog box with a dark header bar containing the title 'Clone Profile' and a close button 'X'. The main area is white and contains two input fields. The first field, labeled 'Profile Name', has the value 'default'. The second field, labeled 'Clone Name', is highlighted with a red border and contains the text 'Service_Provider'. Below these fields is a 'Finish' button.

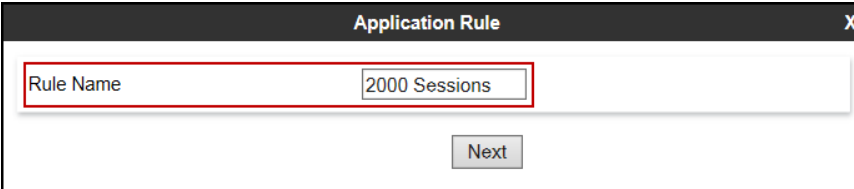
7.12.Domain Policies

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.

7.12.1.Application Rules

Application Rules define which types of SIP-based Unified Communications (UC) applications the UC-Sec security device will protect: voice, video, and/or Instant Messaging (IM). In addition, Application Rules define the maximum number of concurrent voice sessions the network will process in order to prevent resource exhaustion. From the menu on the left-hand side, select **Domain Policies** → **Application Rules**, click on the **Add** button to add a new rule.

- Under **Rule Name** enter the name of the profile, e.g., *2000 Sessions*.
- Click **Next**.

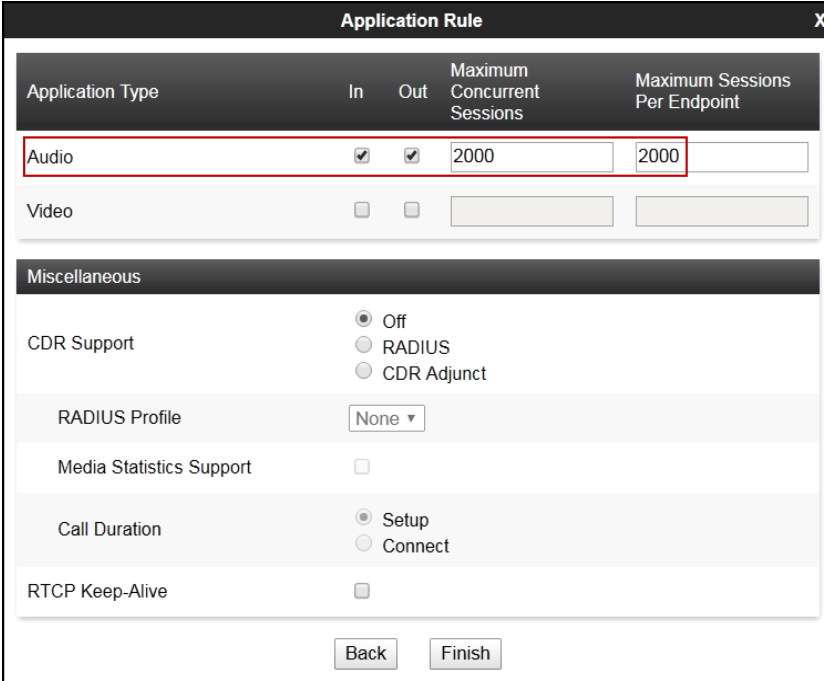


Application Rule

Rule Name: 2000 Sessions

Next

- Under **Audio** check *In* and *Out* and set the **Maximum Concurrent Sessions** and **Maximum Sessions Per Endpoint** to recommended values, the value of *2000* for Audio. Repeat for video if needed.
- Click **Finish**.



Application Rule

Application Type	In	Out	Maximum Concurrent Sessions	Maximum Sessions Per Endpoint
Audio	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	2000	2000
Video	<input type="checkbox"/>	<input type="checkbox"/>		

Miscellaneous

CDR Support: ☒ Off, ☐ RADIUS, ☐ CDR Adjunct

RADIUS Profile: None

Media Statistics Support: ☐

Call Duration: ☒ Setup, ☐ Connect

RTCP Keep-Alive: ☐

Back Finish

7.12.2. Media Rules

Media Rules allow one to define RTP media packet parameters such as prioritizing encryption techniques and packet encryption techniques. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies to determine how media packets matching these criteria will be handled by the Avaya SBCE security product. For the compliance test, one media rule (shown below) was created toward Session Manager and a default media rule was used toward the Service Provider.

To add a media rule in the Session Manager direction, from the menu on the left-hand side, select **Domain Policies → Media Rules**.

- Click on the **Add** button to add a new media rule (not shown).
- Under **Rule Name** enter **SM_SRTP**.
- Click **Next** (not shown).
- Under Audio Encryption, **Preferred Format #1**, select **SRTP_AES_CM_128_HMAC_SHA1_80**.
- Under Audio Encryption, **Preferred Format #2**, select **RTP**.
- Under Audio Encryption, uncheck **Encrypted RTCP**.
- Under Audio Encryption, check **Interworking**.
- Repeat the above steps under Video Encryption, if needed.
- Under Miscellaneous verify that **Capability Negotiation** is checked.
- Click **Next**.

The screenshot shows the 'Media Rule' configuration window. It is divided into three main sections: Audio Encryption, Video Encryption, and Miscellaneous. In the Audio Encryption section, Preferred Format #1 is set to 'SRTP_AES_CM_128_HMAC_SHA1_80', Preferred Format #2 is set to 'RTP', Preferred Format #3 is set to 'NONE', Encrypted RTCP is unchecked, MKI is unchecked, Lifetime is set to '2^4', and Interworking is checked. The Video Encryption section has identical settings. In the Miscellaneous section, Capability Negotiation is checked. At the bottom, there are 'Back' and 'Next' buttons.

Audio Encryption	
Preferred Format #1	SRTP_AES_CM_128_HMAC_SHA1_80
Preferred Format #2	RTP
Preferred Format #3	NONE
Encrypted RTCP	<input type="checkbox"/>
MKI	<input type="checkbox"/>
Lifetime <small>Leave blank to match any value.</small>	2 ⁴
Interworking	<input checked="" type="checkbox"/>

Video Encryption	
Preferred Format #1	SRTP_AES_CM_128_HMAC_SHA1_80
Preferred Format #2	RTP
Preferred Format #3	NONE
Encrypted RTCP	<input type="checkbox"/>
MKI	<input type="checkbox"/>
Lifetime <small>Leave blank to match any value.</small>	2 ⁴
Interworking	<input checked="" type="checkbox"/>

Miscellaneous	
Capability Negotiation	<input checked="" type="checkbox"/>

Back Next

- Accept default values in the remaining sections by clicking **Next** (not shown), and then click **Finish** (not shown).

- For the compliance test, the **default-low-med** Media Rule was used in the Service Provider direction.

Media Encryption

Audio Encryption

Preferred Format #1

RTP

Preferred Format #2

NONE

Preferred Format #3

NONE

Encrypted RTCP

☐

MKI

☐

Lifetime

Leave blank to match any value.

2^

Interworking

☒

Video Encryption

Preferred Format #1

RTP

Preferred Format #2

NONE

Preferred Format #3

NONE

Encrypted RTCP

☐

MKI

☐

Lifetime

Leave blank to match any value.

2^

Interworking

☒

Miscellaneous

Capability Negotiation

☐

Finish

7.12.3. Signaling Rules

For the compliance test, the **default** signaling rule was used.

The screenshot displays the Avaya Session Border Controller for Enterprise (SBCE) web interface. The top navigation bar includes links for Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header shows the device name 'Avaya_SBCE' and the title 'Session Border Controller for Enterprise' with the Avaya logo.

The left sidebar contains a navigation menu with categories like EMS Dashboard, Device Management, Backup/Restore, System Parameters, Configuration Profiles, Services, Domain Policies (expanded), Application Rules, Border Rules, Media Rules, Security Rules, Signaling Rules (highlighted), Charging Rules, End Point Policy Groups, Session Policies, TLS Management, Network & Flows, DMZ Services, and Monitoring & Logging.

The main content area is titled 'Signaling Rules: default' and features an 'Add' button. A warning message states: 'It is not recommended to edit the defaults. Try cloning or adding a new rule instead.' Below this, there are tabs for General, Requests, Responses, Request Headers, Response Headers, Signaling QoS, and UCID. The 'General' tab is active, showing configuration options for Inbound and Outbound traffic, including Requests, Non-2XX Final Responses, Optional Request Headers, and Optional Response Headers, all set to 'Allow'. A 'Content-Type Policy' section includes a checkbox for 'Enable Content-Type Checks' (checked) and a table for 'Action' (Allow) and 'Multipart Action' (Allow). An 'Exception List' section is also present. An 'Edit' button is located at the bottom right of the configuration area.

7.13.End Point Policy Groups

End Point Policy Groups associate the different sets of rules under Domain Policies (Media, Signaling, Security, etc.) to be applied to specific SIP messages traversing through the Avaya SBCE. Please note that changes should not be made to any of the default rules used in these End Point Policy Groups.

7.13.1. End Point Policy Group – Enterprise

To create an End Point Policy Group for the enterprise, select **End Point Policy Groups** under the **Domain Policies** menu and select **Add** (not shown).

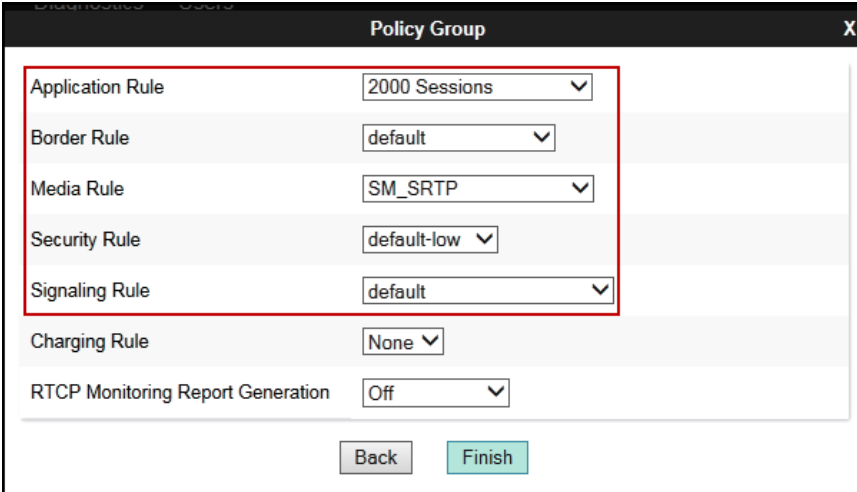
- Enter an appropriate name in the **Group Name** field.
- Click **Next**.



The screenshot shows a dialog box titled "Policy Group" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Group Name" which contains the text "Enterprise". A red rectangular box highlights this field. Below the input field, there is a button labeled "Next".

Under the **Policy Group** tab enter the following:

- **Application Rule:** *2000 Sessions* (Section 7.12.1).
- **Border Rule:** *default*.
- **Media Rule:** *SM_SRTP* (Section 7.12.2).
- **Security Rule:** *default-low*.
- **Signaling Rule:** *default* (Section 7.12.3).
- Click **Finish**.

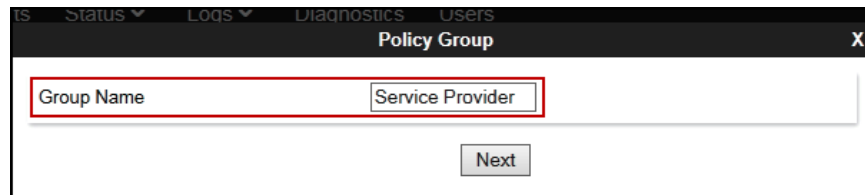


The screenshot shows a dialog box titled "Policy Group" with a close button (X) in the top right corner. Inside the dialog, there are several dropdown menus for configuring rules. A red rectangular box highlights the first five rules: Application Rule (2000 Sessions), Border Rule (default), Media Rule (SM_SRTP), Security Rule (default-low), and Signaling Rule (default). Below these, there is a Charging Rule (None) and an RTCP Monitoring Report Generation (Off) dropdown. At the bottom of the dialog, there are two buttons: "Back" and "Finish".

7.13.2. End Point Policy Group – Service Provider

To create an End Point Policy Group for the Service Provider, select **End Point Policy Groups** under the **Domain Policies** menu and select **Add** (not shown).

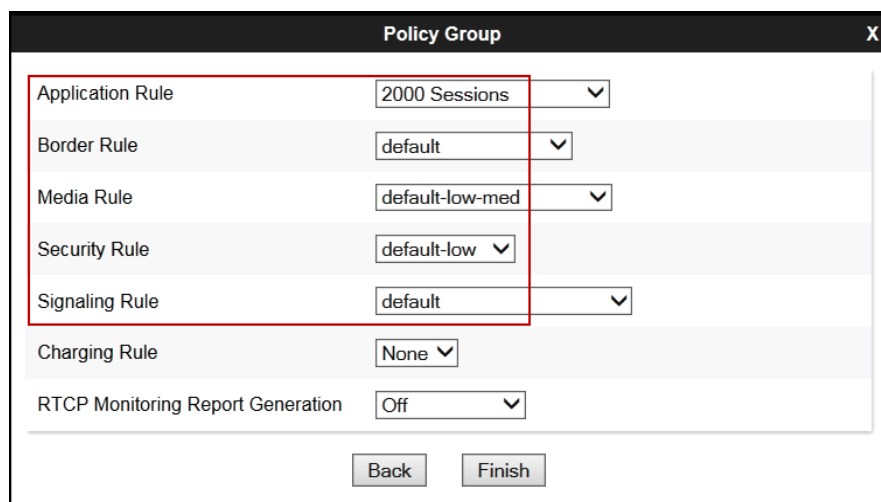
- Enter an appropriate name in the **Group Name** field (*Service Provider* was used).
- Click **Next**.



The screenshot shows a dialog box titled "Policy Group" with a close button (X) in the top right corner. Below the title bar, there is a text input field labeled "Group Name" containing the text "Service Provider". A red rectangular box highlights the "Group Name" field and the "Service Provider" text. Below the input field, there is a "Next" button.

Under the **Policy Group** tab enter the following:

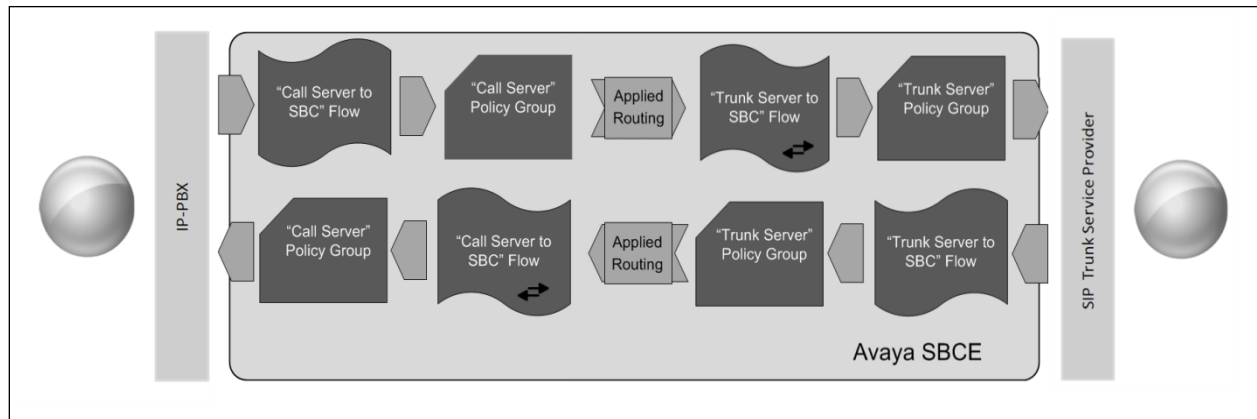
- **Application Rule:** *2000 Sessions* (Section 7.12.1).
- **Border Rule:** *default*.
- **Media Rule:** *default-low-med* (Section 7.12.2).
- **Security Rule:** *default-low*.
- **Signaling Rule:** *default* (Section 7.12.3).
- Click **Finish**.



The screenshot shows a dialog box titled "Policy Group" with a close button (X) in the top right corner. Below the title bar, there is a list of rules with their corresponding values in dropdown menus. A red rectangular box highlights the first five rules: Application Rule (2000 Sessions), Border Rule (default), Media Rule (default-low-med), Security Rule (default-low), and Signaling Rule (default). Below these, there are two more rules: Charging Rule (None) and RTCP Monitoring Report Generation (Off). At the bottom of the dialog, there are "Back" and "Finish" buttons.

7.14.End Point Flows

When a packet is received by Avaya SBCE, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to a policy group which contains several rules concerning processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for this destination endpoint are applied. The context is maintained, so as to be applied to future packets in the same flow. The following screen illustrates the flow through the Avaya SBCE to secure a SIP trunk call.



The **End-Point Flows** defines certain parameters that pertain to the signaling and media portions of a call, whether it originates from within the enterprise or outside of the enterprise.

7.14.1. End Point Flow – Enterprise

To create the call flow toward the enterprise, from the **Device Specific** menu, select **End Point Flows**, then select the **Server Flows** tab. Click **Add** (not shown). The screen below shows the flow named *Session_Manager_Flow* created in the sample configuration. The flow uses the interfaces, policies, and profiles defined in previous sections. Note that the **Routing Profile** selection is the profile created for the Service Provider in **Section 7.10.2**, which is the reverse route of the flow. Click **Finish**.

The screenshot shows the 'Edit Flow: Session_Manager_Flow' configuration window. The window has a title bar with tabs: Alarms, Incidents, Status, Logs, Diagnostics, Users, and Settings. The main content area contains the following fields:

Flow Name	Session_Manager_Flow
SIP Server Profile	Session Manager
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Public_sig
Signaling Interface	Private_sig
Media Interface	Private_med
Secondary Media Interface	None
End Point Policy Group	Enterprise
Routing Profile	Route_to_SP_UDP
Topology Hiding Profile	Session_Manager
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	<input type="checkbox"/>

At the bottom of the window is a 'Finish' button.

7.14.2. End Point Flow – Service Provider

A second Server Flow with the name *SIP_Trunk_Flow_UDP* was similarly created in the Service Provider direction. The flow uses the interfaces, policies, and profiles defined in previous sections. Note that the **Routing Profile** selection is the profile created for Session Manager in **Section 7.10.1**, which is the reverse route of the flow. Also note that there is no selection under the **Signaling Manipulation Script** field. Click **Finish**.

Edit Flow: SIP_Trunk_Flow_UDP	
Flow Name	SIP_Trunk_Flow_UDP
SIP Server Profile	Service Provider UDP
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Private_sig
Signaling Interface	Public_sig
Media Interface	Public_med
Secondary Media Interface	None
End Point Policy Group	Service Provider
Routing Profile	Route_to_SM
Topology Hiding Profile	Service_Provider
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	<input type="checkbox"/>
Finish	

8. Motto Communications SIP Trunking Service Configuration

To use Motto Communications SIP Trunking Service, a customer must request the service from Motto Communications using the established sales processes. The process can be started by contacting Motto Communications via the corporate web site at: <http://www.motto.nl>

During the signup process, Motto Communications and the customer will discuss details about the preferred method to be used to connect the customer's enterprise network to Motto Communications network.

Motto Communications will provide the following information:

- FQDN to be used for public DNS record queries.
- SIP Trunk registration credentials (User Name, Password, etc.).
- DID numbers.
- Etc.

9. Verification and Troubleshooting

This section provides verification steps that may be performed in the field to verify that the solution is configured properly. This section also provides a list of commands that can be used to troubleshoot the solution.

9.1. General Verification Steps

- Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.
- Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active for more than 35 seconds.
- Verify that the user on the PSTN can end an active call by hanging up.
- Verify that an endpoint at the enterprise site can end an active call by hanging up.

9.2. Communication Manager Verification

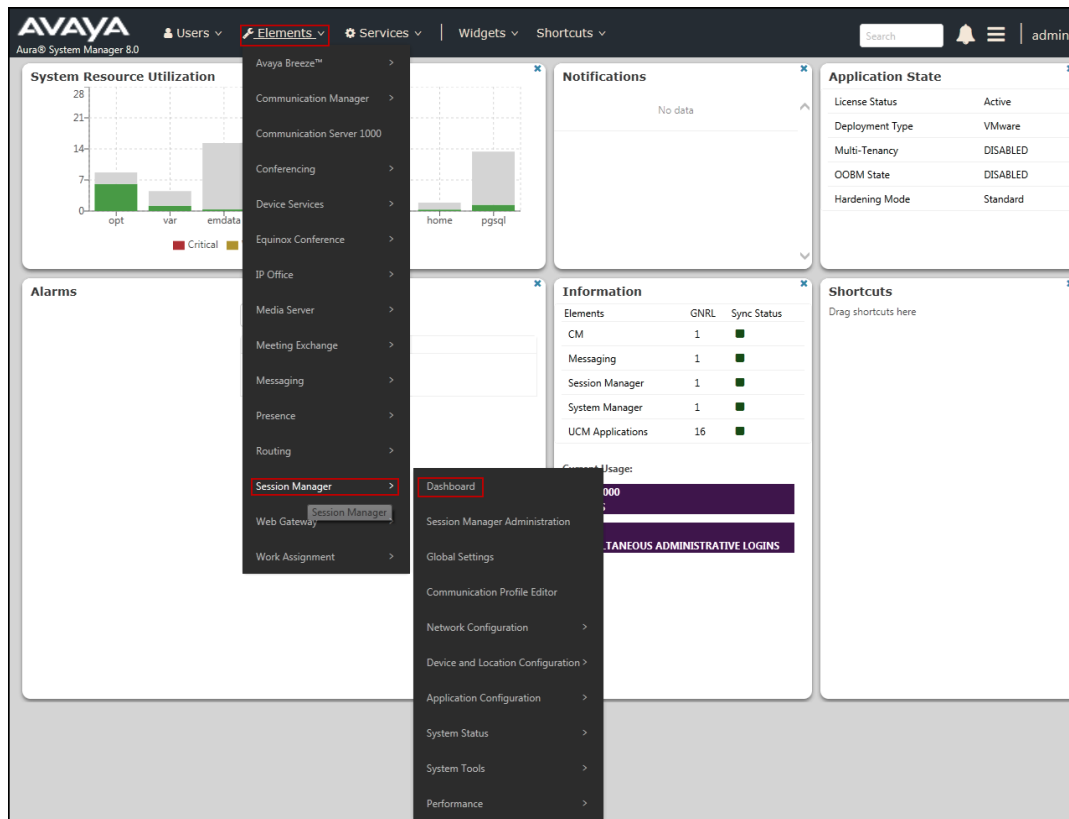
The following commands can be entered in the Communication Manager SAT terminal to verify the SIP trunk functionality:

- **list trace station** <extension number>
Traces calls to and from a specific station.
- **list trace tac** <trunk access code number>
Trace calls over a specific trunk group.
- **status signaling-group** <signaling group number>
Displays signaling group service state.
- **status trunk** <trunk group number>
Displays trunk group service state.
- **status station** <extension number>
Displays signaling and media information for an active call on a specific station.

9.3. Session Manager Verification

The Session Manager configuration may be verified via System Manager.

Step 1 - Using the procedures described in **Section 6**, access the System Manager GUI. From the **Home** screen, under the **Elements** heading, select **Session Manager**, then select **Dashboard**.



Step 2 - The Session Manager Dashboard is displayed. Note that the **Test Passed**, **Alarms**, **Service State**, and **Data Replication** columns all show good status.

In the **Entity Monitoring** column, Session Manager shows that there are **2** alarms out of the **7** Entities defined.

Session Manager Dashboard

This page provides the overall status and health summary of each administered Session Manager.

Session Manager Instances

Service State: Shutdown System: EASG: As of 1:40 PM

1 Item Show All Filter: Enable

	Session Manager	Type	Tests Pass	Alarms	Security Module	Service State	Entity Monitoring	Active Call Count	Registrations	Data Replication	User Data Storage Status	License Mode	EASG	Version
<input type="checkbox"/>	Session Manager	Core	✓	0/0/0	Up	Accept New Service	2/7	0	1/1	✓	✓	Normal	Enabled	8.0.1.1.801103

Select: All, None

Verify that the state of the Session Manager links under the **Conn. Status** and **Link Status** columns are **UP**, like shown on the screen below

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:

All Entity Links for Session Manager: Session Manager

Summary View

7 Items Filter: Enable

SIP Entity Name	IP Address Family	SIP Entity Resolved IP	Port	Proto	Deny	Conn. Status	Reason Code	Link Status
Avaya SBCE	IPv4	10.64.101.243	5061	TLS	FALSE	UP	200 OK	UP
Avaya Experience Portal	IPv4	10.64.101.252	5061	TLS	FALSE	UP	200 OK	UP
Communication Manager Trunk 1	IPv4	10.64.101.241	5061	TLS	FALSE	UP	200 OK	UP
AA-Messaging	IPv4	10.64.101.250	5060	TCP	FALSE	UP	200 OK	UP
Communication Manager Trunk 2	IPv4	10.64.101.241	5071	TLS	FALSE	UP	200 OK	UP
Communication Manager Trunk 98	IPv4	10.64.101.241	5065	TLS	FALSE	UP	200 OK	UP
CS1K7.6	IPv4	172.16.5.60	5085	UDP	FALSE	DOWN	408 Request Timeout	DOWN

Select: None

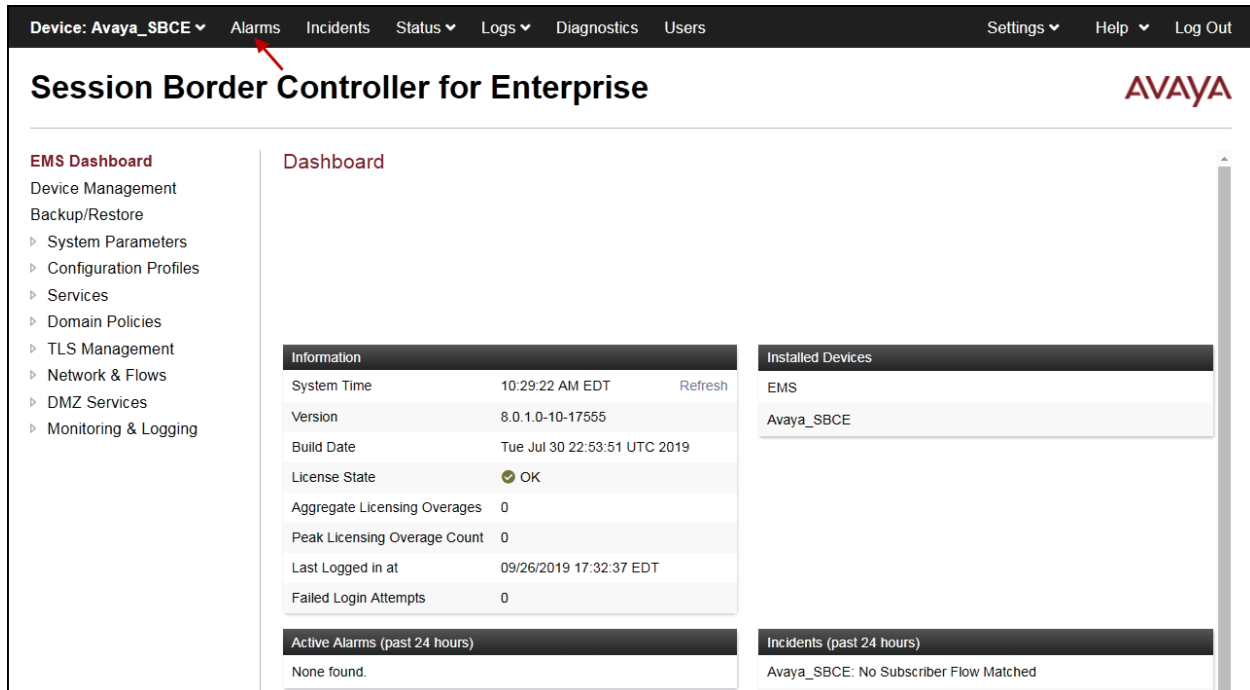
Other Session Manager useful verification and troubleshooting tools include:

- **traceSM** – Session Manager command line tool for traffic analysis. Login to the Session Manager command line management interface to run this command.
- **Call Routing Test** – The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, from the System Manager Home screen navigate to **Elements → Session Manager → System Tools → Call Routing Test**. Enter the requested data to run the test.

9.4. Avaya SBCE Verification

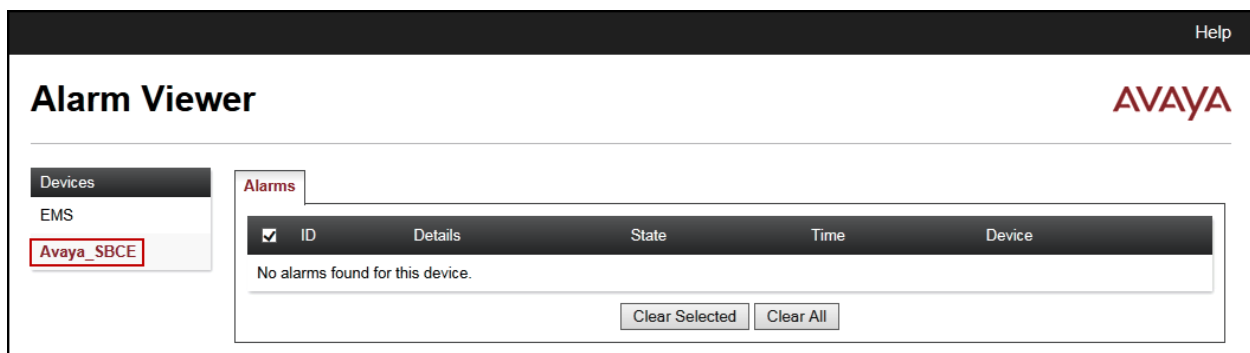
There are several links and menus located on the taskbar at the top of the screen of the web interface that can provide useful diagnostic or troubleshooting information.

Alarms: This screen provides information about the health of the SBC.



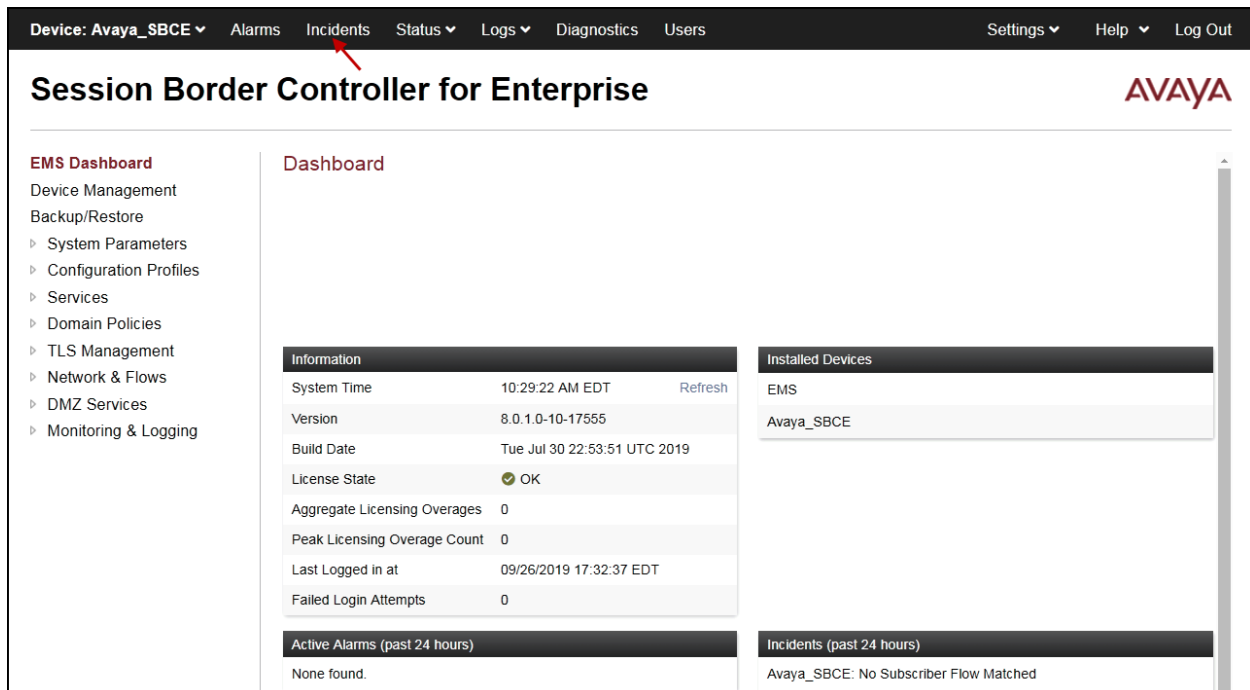
The screenshot shows the Avaya SBCE Dashboard. The top navigation bar includes links for Device: Avaya_SBCE, Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header reads "Session Border Controller for Enterprise" with the Avaya logo. On the left is the "EMS Dashboard" menu with options like Device Management, Backup/Restore, System Parameters, Configuration Profiles, Services, Domain Policies, TLS Management, Network & Flows, DMZ Services, and Monitoring & Logging. The central "Dashboard" area contains several widgets: "Information" (System Time: 10:29:22 AM EDT, Version: 8.0.1.0-10-17555, Build Date: Tue Jul 30 22:53:51 UTC 2019, License State: OK, Aggregate Licensing Overages: 0, Peak Licensing Overage Count: 0, Last Logged in at: 09/26/2019 17:32:37 EDT, Failed Login Attempts: 0), "Installed Devices" (listing EMS and Avaya_SBCE), "Active Alarms (past 24 hours)" (None found), and "Incidents (past 24 hours)" (Avaya_SBCE: No Subscriber Flow Matched).

The following screen shows the **Alarm Viewer** page.

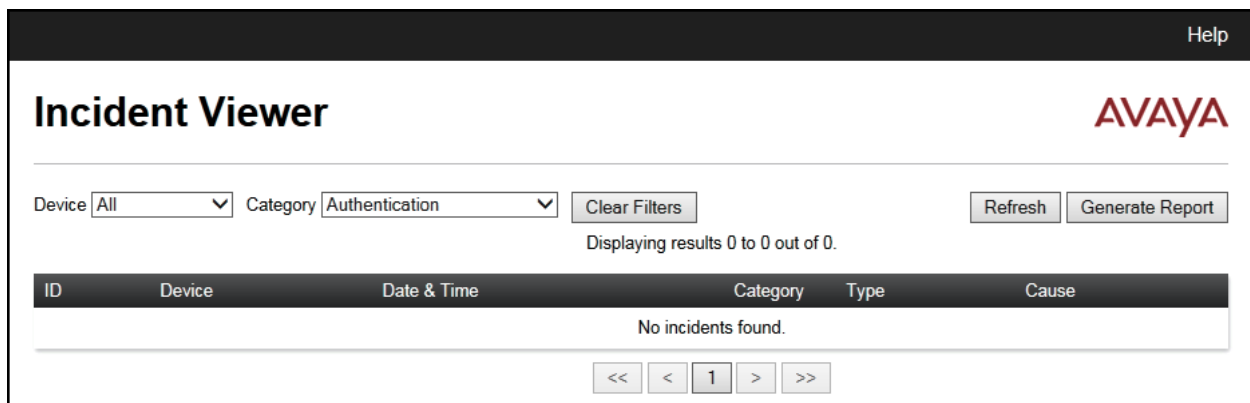


The screenshot shows the Avaya Alarm Viewer page. The top navigation bar includes a Help link. The main header reads "Alarm Viewer" with the Avaya logo. On the left is the "Devices" menu with options for EMS and Avaya_SBCE. The central "Alarms" section has a table with columns: ID, Details, State, Time, and Device. The table is currently empty, displaying "No alarms found for this device." Below the table are buttons for "Clear Selected" and "Clear All".

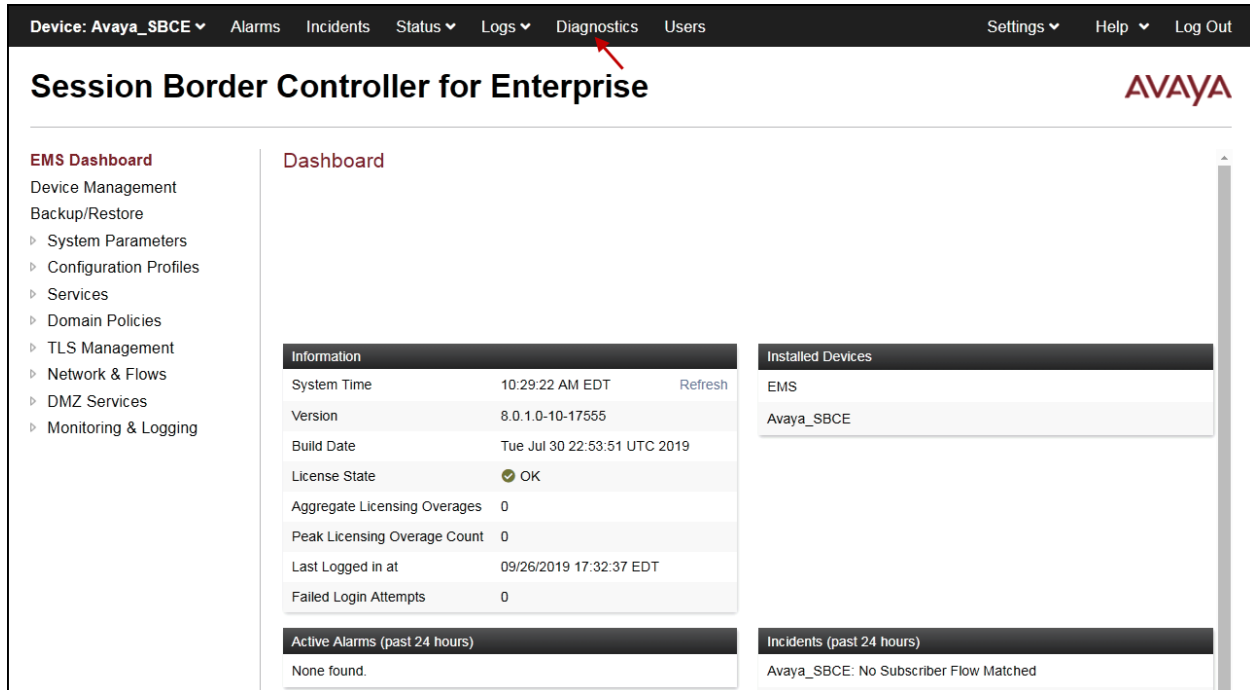
Incidents : Provides detailed reports of anomalies, errors, policies violations, etc.



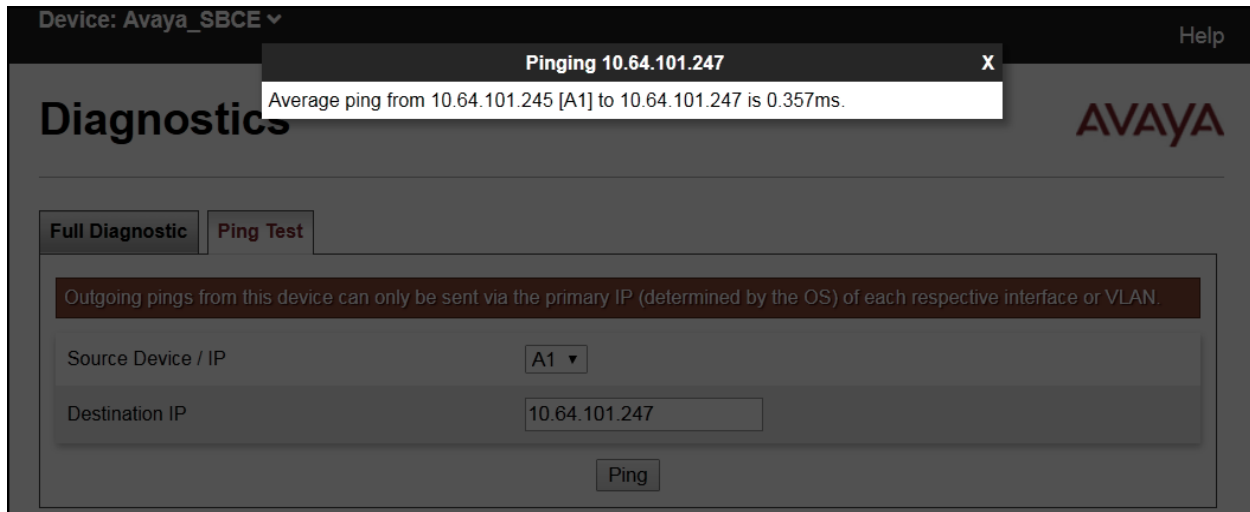
The following screen shows the Incident Viewer page.



Diagnostics: This screen provides a variety of tools to test and troubleshoot the Avaya SBCE network connectivity.



The following screen shows the Diagnostics page with the results of a ping test.



Additionally, the Avaya SBCE contains an internal packet capture tool that allows the capture of packets on any of its interfaces, saving them as *pcap* files. Navigate to **Monitor & Logging** → **Trace**. Select the **Packet Capture** tab, set the desired configuration for the trace and click **Start Capture**.

Device: Avaya_SBCE ▾

Alarms

Incidents

Status ▾

Logs ▾

Diagnostics

Users

Settings ▾

Help ▾

Log Out

Session Border Controller for Enterprise

AVAYA

EMS Dashboard

Device Management

Backup/Restore

▸ System Parameters

▸ Configuration Profiles

▸ Services

▸ Domain Policies

▸ TLS Management

▸ Network & Flows

▸ DMZ Services

▸ **Monitoring & Logging**

SNMP

Syslog Management

Debugging

Trace

Log Collection

DoS Learning

CDR Adjunct

Trace: Avaya_SBCE

Packet Capture

Captures

Packet Capture Configuration

Status

Ready

Interface

Any ▾

Local Address
IP[:Port]

All ▾ :

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

Protocol

All ▾

Maximum Number of Packets to Capture

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

Start Capture

Clear

Once the capture is stopped, click the **Captures** tab and select the proper *pcap* file. Note that the date and time is appended to the filename specified previously. The file can now be saved to the local PC, where it can be opened with an application such as Wireshark.

The screenshot displays the Avaya Session Border Controller for Enterprise (SBCE) web interface. The top navigation bar includes links for Device: Avaya_SBCE, Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header reads 'Session Border Controller for Enterprise' with the AVAYA logo. On the left, a sidebar menu lists various management options, with 'Monitoring & Logging' and its sub-item 'Trace' highlighted. The main content area, titled 'Trace: Avaya_SBCE', features two tabs: 'Packet Capture' and 'Captures' (the active tab). Below the tabs is a table listing captured files. A single entry is shown: 'Motto_DNS_SRV_20191118143144.pcap' with a size of 1,388,544 bytes and a timestamp of November 18, 2019 2:33:31 PM EST. A 'Delete' button is visible next to the entry. A 'Refresh' button is located in the top right corner of the table area.

File Name	File Size (bytes)	Last Modified
Motto_DNS_SRV_20191118143144.pcap	1,388,544	November 18, 2019 2:33:31 PM EST

Also, the **traceSBC** tool can be used to monitor the SIP signaling messages between the Service provider and the Avaya SBCE.

10. Conclusion

These Application Notes describe the procedures required to configure Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1 and Avaya Session Border Controller for Enterprise 8.0, to connect to the Motto Communications SIP Trunking service, as shown in **Figure 1**.

Interoperability testing of the sample configuration was completed with successful results for all test cases with the observations/limitations described in **Sections 2.1** and **2.2**.

11. 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 Aura® Communication Manager in a Virtualized Environment*, Release 8.1.x, Issue 2, August 2019.
- [2] *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 3, August 2019.
- [3] *Administering Avaya Aura® System Manager* for Release 8.1.x, Issue 3, July 2019.
- [4] *Deploying Avaya Aura® System Manager in a Virtualized Environment*, Release 8.1.x, Issue 2, July 2019.
- [5] *Deploying Avaya Aura® Session Manager and Avaya Aura® Branch Session Manager in a Virtualized Environment*, Release 8.1., Issue 1, June 2019.
- [6] *Administering Avaya Aura® Session Manager*, Release 8.1, Issue 1, June 2019.
- [7] *Deploying Avaya Session Border Controller for Enterprise*, Release 8.0, Issue 3, July 2019.
- [8] *Administering Avaya Session Border Controller for Enterprise*, Release 8.0, Issue 1, February 2019.
- [9] *Configuring Remote Workers with Avaya Session Border Controller for Enterprise Rel. 7.0, Avaya Aura® Communication Manager Rel. 7.0 and Avaya Aura® Session Managers Rel. 7.0 - Issue 1.0*.
- [10] *Deploying and Updating Avaya Aura® Media Server Appliance*, Release 8.0.x, Issue 7, June 2019.
- [11] *Implementing and Administering Avaya Aura® Media Server*. Release 8.0.x, Issue 5, June 2019.
- [12] *Planning for and Administering Avaya Equinox for Android, iOS, Mac, and Windows*. Release 3.6, Issue 1, July 2019.
- [13] *Administering Avaya one-X® Communicator*. Release 6.2, Feature Pack 10, November 2015.
- [14] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>
- [15] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, <http://www.ietf.org/>

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