



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

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# **Application Notes for Pure IP SIP Trunking Service with Avaya Aura® Communication Manager Release 7.0, Avaya Aura® Session Manager Release 7.0 and Avaya Session Border Controller for Enterprise Release 7.1 – Issue 1.0**

## **Abstract**

These Application Notes describe the steps to configure a Session Initiation Protocol (SIP) trunk between Pure IP SIP Trunking Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Communication Manager 7.0, Avaya Aura® Session Manager 7.0, Avaya Session Border Controller for Enterprise 7.1, Avaya Aura® Media Server 7.7, Avaya Aura® Messaging 6.3 and various Avaya endpoints. This documented solution does not extend to configurations without Avaya Session Border Controller for 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.

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

## Table of Contents

1.	Introduction.....	4
2.	General Test Approach and Test Results.....	4
2.1.	Interoperability Compliance Testing .....	4
2.2.	Test Results .....	5
2.3.	Support.....	6
3.	Reference Configuration .....	7
4.	Equipment and Software Validated .....	9
5.	Configure Avaya Aura® Communication Manager .....	10
5.1.	Licensing and Capacity .....	10
5.2.	System Features .....	12
5.3.	IP Node Names .....	13
5.4.	Codecs.....	14
5.5.	IP Network Region .....	15
5.6.	Signaling Group.....	17
5.7.	Trunk Group.....	19
5.8.	Calling Party Information .....	21
5.9.	Incoming Call Handling Treatment .....	21
5.10.	Outbound Routing.....	22
5.11.	Saving Communication Manager Configuration Changes .....	23
5.12.	TLS Management on Communication Manager.....	24
6.	Configure Avaya Aura® Session Manager .....	27
6.1.	System Manager Login and Navigation .....	27
6.2.	Specify SIP Domain.....	28
6.3.	Add Location .....	29
6.4.	Add SIP Entities.....	30
6.5.	Add Entity Links.....	33
6.6.	Add Routing Policies .....	34
6.7.	Add Dial Patterns.....	36
6.8.	Add/View Session Manager .....	39
6.9.	TLS Certificate Management on System Manager.....	40
7.	Configure Avaya Session Border Controller for Enterprise .....	41
7.1.	Avaya Session Border Controller for Enterprise Login.....	41
7.2.	TLS Management.....	42
7.2.1.	Certificates .....	43
7.2.2.	Client Profiles .....	44
7.2.3.	Server Profiles .....	45
7.3.	Global Profiles .....	46
7.3.1.	Uniform Resource Identifier (URI) Groups .....	46
7.3.2.	Server Interworking Profile .....	46
7.3.3.	Server Configuration .....	50
7.3.4.	Routing Profiles .....	53
7.3.5.	Topology Hiding.....	55
7.4.	Domain Policies .....	57
7.4.1.	Media Rules .....	57

7.4.2.	Signaling Rules .....	58
7.4.3.	Endpoint Policy Groups.....	59
7.5.	Device Specific Settings .....	61
7.5.1.	Network Management .....	61
7.5.2.	Media Interface.....	63
7.5.3.	Signaling Interface.....	63
7.5.4.	End Point Flows - Server Flow.....	65
8.	Pure IP Service Configuration .....	68
9.	Verification and Troubleshooting .....	68
9.1.	Verification Steps.....	68
9.2.	Protocol Traces .....	68
9.3.	Troubleshooting: .....	69
9.3.1.	The Avaya SBCE.....	69
9.3.2.	Communication Manager .....	69
10.	Conclusion .....	70
11.	References .....	71

# 1. Introduction

These Application Notes describe the steps to configure a SIP trunk between Pure IP SIP Trunking Service and an Avaya SIP-enabled enterprise solution. Avaya Aura® release 7.0 is being deployed in virtualized environment that includes Avaya Aura® Communication Manager 7.0 (Communication Manager), Avaya Aura® Session Manager 7.0 (Session Manager), Avaya Aura® Media Server 7.7, Avaya Aura® Messaging and Avaya Session Border Controller for Enterprise 7.1 (Avaya SBCE). Various Avaya endpoints are also used in the test configuration.

For privacy and security, TLS for Signaling and SRTP for media encryption were used inside of the enterprise (private network side). Outside of the enterprise (public network side) to Pure IP was using UDP and RTP.

Customers using this Avaya SIP-enabled enterprise solution with Pure IP are able to place and receive PSTN calls via a broadband Internet connection. This converged network solution is an alternative to a traditional PSTN trunk such as analog and/or ISDN-PRI.

## 2. General Test Approach and Test Results

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.

Pure IP is a member of the Avaya DevConnect Service Provider Program. The general test approach is to connect a simulated enterprise to Pure IP via the Internet and exercise the features and functionalities listed in **Section 2.1**.

### 2.1. Interoperability Compliance Testing

To verify Pure IP interoperability, the following features and functionalities are covered in the compliance testing:

- Inbound PSTN calls to various phone types including H.323, SIP, digital and analog telephone at the enterprise. All inbound calls from PSTN are routed to the enterprise across the SIP trunk from the service provider.
- Outbound PSTN calls from various phone types including H.323, SIP, digital and analog telephone at the enterprise. All outbound calls to PSTN are routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from Avaya one-X® Communicator (1XC) soft phone.
- Dialing plans including local, long distance, international, outbound toll-free calls etc.
- Calling Party Name presentation and Calling Party Name restriction.
- Codecs G.711A, G.711U and G.729.
- Fax G.711 pass-through.

- Media and Early Media transmissions.
- DTMF tone transmissions as out-of-band RTP events as per RFC2833.
- Voicemail navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, forward and conference.
- Off-net call forward.
- EC500 mobility (extension to cellular).
- Routing inbound vector call to call center agent queues.
- Response to OPTIONS heartbeat, Authentication and Registration.
- Response to incomplete call attempts and trunk errors.
- Avaya Communicator for Windows.
- Session Timers implementation.
- Remote Worker, which allows Avaya SIP endpoints to connect directly to the public Internet as enterprise phones.

Items, that are not supported, include the following:

- Fax T.38 is not supported by Pure IP.

Items, that are not tested, include the following:

- Directory assistance, Operator Assistance, Emergency.

## 2.2. Test Results

Interoperability testing of Pure IP with the Avaya SIP-enabled enterprise solution was completed with successful results for all test cases with the exception of the observations and limitations described below:

- **No Ring-back Tone Using G.729** – Outbound call to service provider, there is no ring-back tone heard on the originator when using G.729 codec only. Pure IP system trans-coding set up was miss-matching with Avaya system. The issue was resolved by Pure IP.
- **Calling Party Number (CPN) Was Not Blocked with Caller ID Restricted** – Even outbound call to service provider with caller ID restricted (CPN blocked) on Avaya desk-phone but the caller ID was still observed on the PSTN telephone. The issue was resolved by Pure IP.
- **Call Redirection (Blind Call Transfer) Using REFER Method** – Outbound call to PSTN telephone from Avaya IP desk-phone (H.323 or SIP) was being blind transferred to another PSTN. After the call being transferred, it was dropped. The Pure IP system sent a NOTIFY messages with Subscription-State: terminated; reason=noresource to Avaya's system. Issue has been resolved by Pure IP team. It has been configuration changed.
- **Call Conference with PSTN users Using REFER Method** – Inbound PSTN call to Avaya IP desk-phone (H.323 or SIP), Avaya IP desk-phone performed 3-way conference with another PSTN user. As soon as Avaya IP desk-phone hanged up, the conference was dropped. The Pure IP system sent a NOTIFY messages with Subscription-State: terminated; reason=noresource to Avaya system. Issue has been resolved by Pure IP team. It has been configuration changed.
- **Fax T.38 Not Working** – Even though fax t.38 negotiated properly and signaling was correctly observed. But when re-INVITE being sent by Avaya system to negotiate T.38 fax, the SDP contained the internal IP address of Avaya voice gateway instead of Avaya

SBCE external interface IP address. This caused the confusion in Pure IP system and resulting in no fax data being sent or received. Issue has been raised with Avaya design team and will be addressed in the next service update of Avaya SBCE. The issue tracking number is Aurora-10203. The recommendation is to use Pass-Through mode for fax as it has been tested and worked consistently. Refer to **Section 5.4** for set up.

## **2.3. Support**

For technical support on Pure IP SIP Trunking, contact Pure IP at <http://www.pure-ip.com/>

### 3. Reference Configuration

**Figure 1** illustrates the sample Avaya SIP-enabled enterprise solution connected to the Pure IP (Vendor Validation circuit) through a public Internet connection.

For security purposes, the real public IP addresses and PSTN routable phone numbers used in the compliance test are not shown in these Application Notes.

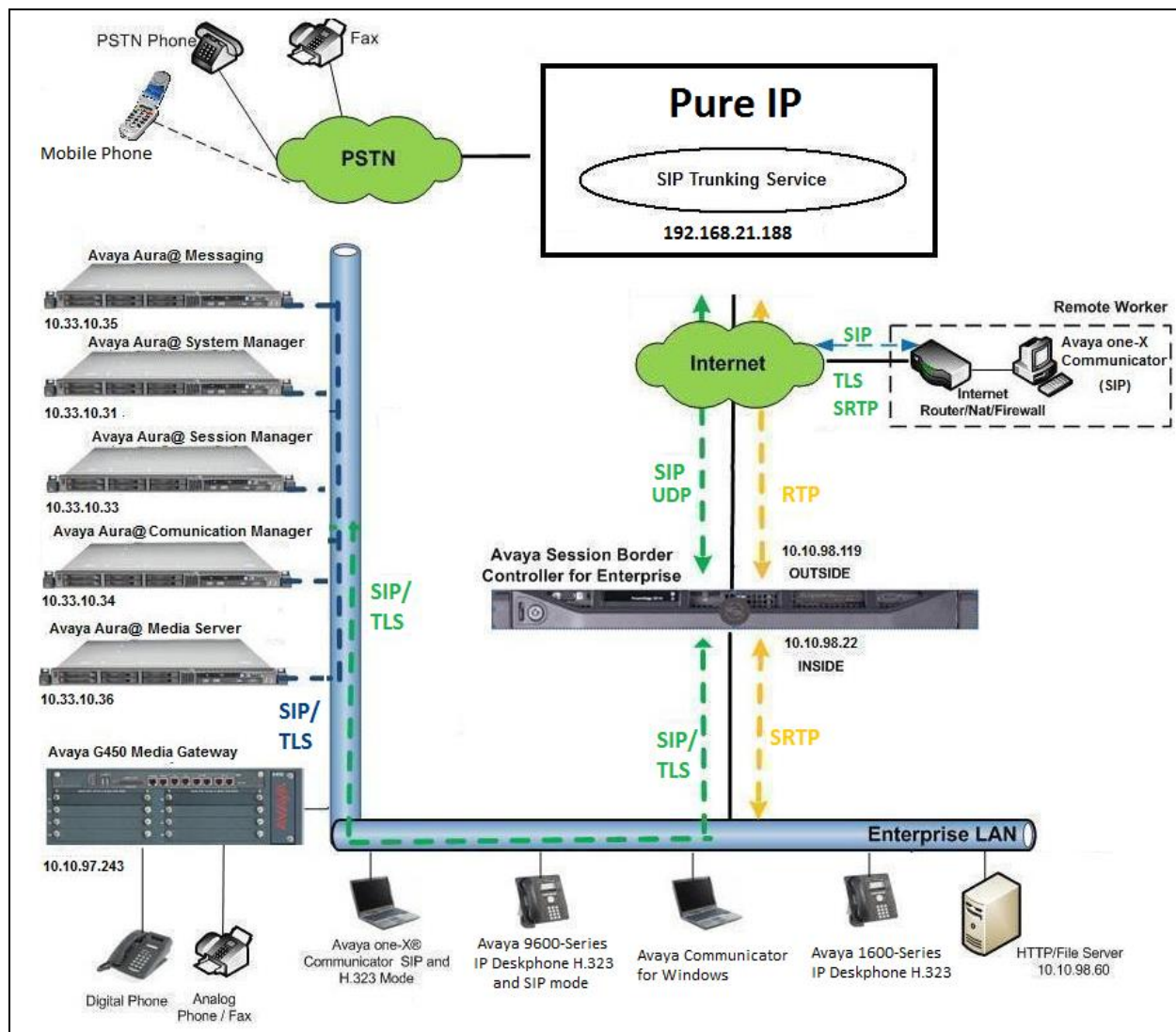
The Avaya components used to create the simulated customer site included:

- Avaya Aura® Communication Manager running in Virtualized environment.
- Avaya Aura® System Manager running in Virtualized environment.
- Avaya Aura® Session Manager running in Virtualized environment.
- Avaya Aura® Messaging running in Virtualized environment.
- Avaya Aura® Media Server running in Virtualized environment.
- Avaya G450 Media Gateway.
- Avaya Session Border Controller for Enterprise.
- Avaya 9600Series IP Deskphones (H.323, SIP).
- Avaya one-X® Communicator soft phones (H.323, SIP).
- Avaya digital and analog telephones.
- Avaya Communicator for Windows.

Located at the edge of the enterprise network is the Avaya SBCE. It has a public side that connects to Pure IP via Internet and a private side that connects to the enterprise network. All SIP and RTP traffic entering or leaving the enterprise network flows through the Avaya SBCE which can protect the enterprise against any outside SIP-based attacks. The Avaya SBCE provides network address translation at both the IP and SIP layers. The transport protocol between the Avaya SBCE and Pure IP across the public network is UDP. The transport protocol between the Avaya SBCE, Session Manager and Communication Manager is TLS.

In the compliance testing, the Avaya Customer-Premises Equipment (CPE) environment was configured with SIP domain “avayalab.com” for the enterprise. The Avaya SBCE is used to adapt the enterprise SIP domain to the IP address based URI-Host known to Pure IP. **Figure 1** below illustrates the network diagram for the enterprise. All voice application elements are connected to internal trusted LAN.

Additionally, a remote worker is included in the reference configuration **Figure 1**. A remote worker is a SIP endpoint that resides in the un-trusted network, registered to Session Manager via the Avaya SBCE. Remote workers feature the same functionality as any other endpoint within the enterprise. This functionality was successfully tested during the compliance test, using the Avaya Communicator for Windows using TLS/SRTP. The configuration tasks required to support remote workers are referenced in **Section 11**.



**Figure 1: Avaya IP Telephony Network connecting to Pure IP Networks**

This testing uses Pure IP SIP trunking system (United Kingdom). For outbound call from Avaya System to PSTN, user dials 9 follow 11 digits including a 1 (Canada/US country code) since the testing is performed in North America (Canada). For inbound call to Avaya system from PSTN, user needs to dial international call beginning with 011 plus 12 digits number (011 441189099574).



## 4. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Component	Release
Avaya Aura® Communication Manager running on Virtualized Environment	7.0.1.1 (FP1SP1, 00.0.441.0-23169)
Avaya G450 Media Gateway	37.39.0
Avaya Aura® System Manager running on Virtualized Environment	7.0.1.1 (Build No. - 7.0.0.0.16266)
Avaya Aura® Session Manager running on Virtualized Environment	7.0.1.1 (7.0.1.1.701114)
Avaya Aura® Messaging running on Virtualized Environment	6.3.124.335-1.253373
Avaya Aura® Media Server running on Virtualized Environment	7.7.0.334
Avaya Session Border Controller for Enterprise	7.1.0.1-07-12090
Avaya 9621G IP Deskphone (H.323)	6.6.302
Avaya 9641G IP Deskphone (SIP)	7.0.1.29
Avaya one-X® Communicator (H.323/SIP)	6.2.12.04-SP12
Avaya Communicator for Windows	2.1.3.8
Avaya 1608 IP Deskphone (H.323)	1.380B
Avaya 1408 Digital Telephone	1408D02A-003
Avaya Analog Telephone	n/a
Pure IP SIP Trunking Service Components	
Component	Release
Sonus SBC5200	V04.02.03R000
PortaOne	MR40

**Table 1: Equipment and Software Tested**

**Note:** This solution will be compatible with other Avaya Server and Media Gateway platforms running similar version of Communication Manager.

## 5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring Communication Manager for the Pure IP SIP Trunking service. It is assumed the general installation of Communication Manager, Avaya G450 Media Gateway and Media Server has been previously completed and is not discussed here.

The configuration of Communication Manager was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation.

### 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 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 sale representative to add the additional capacity or feature.

display system-parameters customer-options		Page	2 of 12
OPTIONAL FEATURES			
IP PORT CAPACITIES		USED	
Maximum Administered H.323 Trunks:	4000	0	
Maximum Concurrently Registered IP Stations:	2400	1	
Maximum Administered Remote Office Trunks:	4000	0	
Maximum Concurrently Registered Remote Office Stations:	2400	0	
Maximum Concurrently Registered IP eCons:	68	0	
Max Concur Registered Unauthenticated H.323 Stations:	100	0	
Maximum Video Capable Stations:	2400	0	
Maximum Video Capable IP Softphones:	2400	3	
<b>Maximum Administered SIP Trunks:</b>	<b>4000</b>	<b>74</b>	
Maximum Administered Ad-hoc Video Conferencing Ports:	4000	0	
Maximum Number of DS1 Boards with Echo Cancellation:	80	0	
(NOTE: You must logoff & login to effect the permission changes.)			

On **Page 4**, verify that **ARS** is set to **y**.

display system-parameters customer-options		Page	4 of 12
OPTIONAL FEATURES			
Abbreviated Dialing Enhanced List? y	Audible Message Waiting? y		
Access Security Gateway (ASG)? n	Authorization Codes? y		
Analog Trunk Incoming Call ID? y	CAS Branch? n		
A/D Grp/Sys List Dialing Start at 01? y	CAS Main? n		
Answer Supervision by Call Classifier? y	Change COR by FAC? n		
<b>ARS? y</b>	Computer Telephony Adjunct Links? y		
ARS/AAR Partitioning? y	Cvg Of Calls Redirected Off-net? y		
ARS/AAR Dialing without FAC? n	DCS (Basic)? y		
ASAI Link Core Capabilities? n	DCS Call Coverage? y		
ASAI Link Plus Capabilities? n	DCS with Rerouting? y		
Async. Transfer Mode (ATM) PNC? n			
Async. Transfer Mode (ATM) Trunking? n	Digital Loss Plan Modification? y		
ATM WAN Spare Processor? n	DS1 MSP? y		
ATMS? y	DS1 Echo Cancellation? y		
Attendant Vectoring? y			
(NOTE: You must logoff & login to effect the permission changes.)			

On **Page 5**, verify that **IP Trunks** field is set to **y** and **Media Encryption Over IP** field is set to **y**.

(Note: The Media Encryption option is only available if Media Encryption Over IP is enabled on the installed license)

display system-parameters customer-options		Page	5 of 12
OPTIONAL FEATURES			
Emergency Access to Attendant? y	IP Stations? y		
Enable 'dadmin' Login? y			
Enhanced Conferencing? y	ISDN Feature Plus? n		
Enhanced EC500? y	ISDN/SIP Network Call Redirection? y		
Enterprise Survivable Server? n	ISDN-BRI Trunks? y		
Enterprise Wide Licensing? n	ISDN-PRI? y		
ESS Administration? y	Local Survivable Processor? n		
Extended Cvg/Fwd Admin? y	Malicious Call Trace? y		
External Device Alarm Admin? y	<b>Media Encryption Over IP? y</b>		
Five Port Networks Max Per MCC? n	Mode Code for Centralized Voice Mail? n		
Flexible Billing? n			
Forced Entry of Account Codes? y	Multifrequency Signaling? y		
Global Call Classification? y	Multimedia Call Handling (Basic)? y		
Hospitality (Basic)? y	Multimedia Call Handling (Enhanced)? y		
Hospitality (G3V3 Enhancements)? y	Multimedia IP SIP Trunking? y		
<b>IP Trunks? y</b>			
IP Attendant Consoles? y			
(NOTE: You must logoff & login to effect the permission changes.)			

On **Page 6**, verify that **Private Networking** and **Processor Ethernet** are set to **y**.

display system-parameters customer-options		Page	6 of	12
OPTIONAL FEATURES				
Multinational Locations?	n	Station and Trunk MSP?	y	
Multiple Level Precedence & Preemption?	n	Station as Virtual Extension?	y	
Multiple Locations?	n			
Personal Station Access (PSA)?	y	System Management Data Transfer?	n	
PNC Duplication?	n	Tenant Partitioning?	y	
Port Network Support?	n	Terminal Trans. Init. (TTI)?	y	
Posted Messages?	y	Time of Day Routing?	y	
		TN2501 VAL Maximum Capacity?	y	
		Uniform Dialing Plan?	y	
<b>Private Networking?</b>	<b>y</b>	Usage Allocation Enhancements?	y	
Processor and System MSP?	y			
<b>Processor Ethernet?</b>	<b>y</b>	Wideband Switching?	y	
		Wireless?	n	
Remote Office?	y			
Restrict Call Forward Off Net?	y			
Secondary Data Module?	y			
(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 an incoming call 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**.

change system-parameters features		Page	1 of 19
FEATURE-RELATED SYSTEM PARAMETERS			
Self Station Display Enabled? y			
<b>Trunk-to-Trunk Transfer: all</b>			
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			

On **Page 9** verify that a text string has been defined to replace the Calling Party Number (CPN) for restricted or unavailable calls. The compliance test used the value of ***Restricted*** for restricted calls and ***Unavailable*** for unavailable calls.

```

change 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: 1
    International Access Code: 001

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**), Session Manager (**SM**) and Media Server (**AMS**). These node names will be needed for defining the signaling groups in **Section 5.6**.

```

change node-names ip                                               Page 1 of 2
                                IP NODE NAMES

    Name          IP Address
SM              10.33.10.33
AMS             10.33.10.36
default          0.0.0.0
procr           10.33.10.34
procr6           ::

```

## 5.4. Codecs

Use the **change ip-codec-set** command to define a list of codecs to be used for calls between the enterprise and the service provider. This compliance test used ip-codec-set 1. Pure IP supports G.711MU and G729. To use these codecs, enter **G.711A**, **G.711MU** and **G.729** in the **Audio Codec**. For media encryption used within Avaya system, the **1-srtp-aescm128-hmac80**, **2-srtp-aescm128-hmac32** and **none** are used in **Media Encryption** and **best-effort** in **Encrypted SRTCP** columns of the table in the order of preference.

The following screen shows the configuration for ip-codec-set 1. During testing, the codec set specifications are varied to test for individual codec support as well as codec negotiation between the enterprise and the network at call setup time.

change ip-codec-set 1

Page 1 of 2

IP CODEC SET

Codec Set: 1

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

Media Encryption

Encrypted SRTCP: best-effort

1:	1-srtp-aescm128-hmac80
2:	2-srtp-aescm128-hmac32
3:	none

On **Page 2**, set the **Fax Mode** to **pass-through** faxing which is supported by Pure IP (refer to **Section 2.2**).

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

## 5.5. IP Network Region

For the compliance testing, ip-network-region 1 was created by the **change ip-network-region 1** command with the following parameters:

- Set the **Authoritative Domain** field to match the SIP domain of the enterprise. In the compliance testing, the domain name is *avayalab.com*. This domain name appears in the “From” header of SIP message originating from this IP region.
- Enter a descriptive name in the **Name** field.
- Enable **IP-IP Direct Audio** (shuffling) to allow audio traffic to be sent directly between IP endpoints without using media resources in the Media Gateway. By default, both **Intra-region** and **Inter-region IP-IP Direct Audio** are set to *yes*. Shuffling can be further restricted at the trunk level under the Signaling Group form.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.4**.
- Default values can be used for all other fields.

```
change ip-network-region 1                                     Page 1 of 20

                                IP NETWORK REGION

Region: 1
Location: 1      Authoritative Domain: avayalab.com
Name: ToSM
MEDIA PARAMETERS                                Intra-region IP-IP Direct Audio: yes
Codec Set: 1      Inter-region IP-IP Direct Audio: yes
UDP Port Min: 2048                                IP Audio Hairpinning? n
UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 46
Audio PHB Value: 46
Video PHB Value: 26
...
```

On **Page 4**, define the IP codec set to be used for traffic between region 1 and other regions. In the compliance testing, Communication Manager, the Avaya G450 Media Gateway, IP/SIP phones and Session Manager were assigned to the same region 1.

```
change ip-network-region 1                                     Page 4 of 20

Source Region: 1      Inter Network Region Connection Management      I      M
                                                                G      A      t
dst codec direct      WAN-BW-limits      Video      Intervening      Dyn      A      G      c
rgn set      WAN Units      Total Norm      Prio Shr Regions      CAC      R      L      e
1      1                                                                all
2      1      y      NoLimit      n      t
3                                                                n      t
```

Non-IP telephones (e.g., analog, digital) derive network region from the IP interface of the Avaya G450 Media Gateway to which the device is connected. IP telephones can be assigned a network region based on an IP address mapping.

To define network region 1 for IP interface **procr**, use **change ip-interface procr** command as shown in the following screen.

<b>change ip-interface procr</b>	Page 1 of 2
IP INTERFACES	
Type: PROCR	Target socket load: 4800
Enable Interface? y	Allow H.323 Endpoints? y
Network Region: 1	Allow H.248 Gateways? y
...	Gatekeeper Priority: 5

To define network region 1 for the Avaya G450 Media Gateway, use **change media-gateway** command as shown in the following screen.

<b>change media-gateway 1</b>	Page 1 of 2
MEDIA GATEWAY 1	
Type: g450	
Name: g450	
Serial No: 11N526797797	
Link Encryption Type: any-ptls/tls	Enable CF? n
Network Region: 1	Location: 1
	Site Data:
Recovery Rule: none	
...	

If Avaya Media Server is used in parallel of Avaya Media Gateway G450, then it is needed to define network region 1 for the Avaya Media Server. Use **change media-server** command as shown in the following screen.

<b>change media-server 1</b>	Page 1 of 1
MEDIA SERVER	
Media Server ID: 1	
Signaling Group: 3	
Voip Channel License Limit: 30	
Dedicated Voip Channel Licenses: 30	
Node Name: AMS	
Network Region: 1	
Location: 1	
Announcement Storage Area:	
...	



## 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 Avaya SBCE 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.

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- Set the **Transport Method** to *tls* (*Transport Layer Security*). The transport method specified here is used between Communication Manager and Session Manager.
- Set the **Near-end Listen Port** and **Far-end Listen Port** to *5061*.
- Set the **Near-end Node Name** to *procr*. This node name maps to the IP interface of *procr* 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 **Far-end Network Region** to the IP network region *1* defined for the service provider in **Section 5.5**.
- Set the **Far-end Domain** to *avayalab.com*.
- Set the **DTMF over IP** to *rtp-payload*. This setting enables Communication Manager to send or receive the DTMF transmissions using RFC2833.
- Set **Enable Layer 3 Test?** to *y*. This setting allows Communication Manager to send OPTIONS heartbeat to Session Manager on the SIP trunk.
- 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 SIP trunk and the enterprise endpoint. If this value is set to *n*, then the Avaya G450 Media Gateway will remain in the media path between the SIP trunk and the endpoint for the duration of the call. Depending on the number of media resources available in the Avaya G450 Media Gateway, these resources may be depleted during high call volume preventing additional calls from completing.
- Set the **Alternate Route Timer** to *30*. This defines the number of seconds Communication Manager will wait for a response (other than 100 Trying) to an outbound INVITE before canceling the call.
- Default values may be used for all other fields.

## Signaling Group 2:

add signaling-group 2		Page 1 of 2
SIGNALING GROUP		
Group Number: 2	Group Type: sip	
IMS Enabled? n	Transport Method: tls	
Q-SIP? n		
IP Video? n		Enforce SIPS URI for SRTP? y
Peer Detection Enabled? y	Peer Server: SM	
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y		
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n		
Alert Incoming SIP Crisis Calls? n		
Near-end Node Name: procr	Far-end Node Name: SM	
Near-end Listen Port: 5061	Far-end Listen Port: 5061	
	Far-end Network Region: 1	
Far-end Domain: avayalab.com		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? y	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Initial IP-IP Direct Media? n	
	Alternate Route Timer(sec): 30	

Another signaling group is created between Communication Manager and the Media Server to provide media resources for IP telephony in parallel of the media gateway G450. For the compliance test, signaling group 3 was used for this purpose and was configured as shown in the capture below.

## Signaling Group 3:

add signaling-group 3		Page 1 of 2
SIGNALING GROUP		
Group Number: 3	Group Type: sip	
	Transport Method: tls	
Peer Detection Enabled? n Peer Server: AMS		
Near-end Node Name: procr	Far-end Node Name: AMS	
Near-end Listen Port: 5061	Far-end Listen Port: 5061	
	Far-end Network Region: 1	
Far-end Domain: 10.33.10.36		

## 5.7. Trunk Group

Use the **add trunk-group** command to create a trunk group for the signaling group created in **Section 0**. For the compliance testing, 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 **Outgoing Display** to *y* to enable name display on the trunk.
- Set the **Service Type** field to *public-ntwrk*.
- Set the **Signaling Group** to the signaling group **2** shown in **Section 0**.
- Set the **Number of Members** field to customer requirement. It is the number of trunk members in the SIP trunk group. This value determines how many simultaneous SIP calls can be supported by this trunk group.
- Default values are used for all other fields.

```
add trunk-group 2                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 2                      Group Type: sip      CDR Reports: y
  Group Name: SIP-Carrier             COR: 1             TN: 1       TAC: #02
  Direction: two-way                 Outgoing Display? y
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: 32
```

On **Page 2**, verify that the **Preferred Minimum Session Refresh Interval (sec)** is set to a value acceptable to the service provider. This value defines the interval a re-INVITEs must be sent to refresh the Session Timer. For the compliance testing, a default value of **600** seconds was used.

```
add trunk-group 2                                     Page 2 of 21
  Group Type: sip
TRUNK PARAMETERS
  Unicode Name: auto
                                     Redirect On OPTIM Failure: 5000
SCCAN? n                               Digital Loss Group: 18
                                     Preferred Minimum Session Refresh Interval(sec): 600
Disconnect Supervision - In? y Out? y
XOIP Treatment: auto      Delay Call Setup When Accessed Via IGAR? N
Caller ID for Service Link Call to H.323 1xC: station-extension
```

On **Page 3**, set the **Numbering Format** field to *public*. This field specifies the format of the CPN sent to the far-end. The public numbers are automatically preceded with a + sign when passed in the “From”, “Contact” and “P-Asserted Identity” headers.

Set the **Replace Restricted Numbers** and **Replace Unavailable Numbers** fields to *y*. This will allow the CPN displayed on the local endpoint to be replaced with the value set in **Section 5.2**, if the inbound call enabled CPN block. Default values are used for all other fields.

add trunk-group 2		Page 3 of 21
TRUNK FEATURES		
ACA Assignment? n	Measured: none	Maintenance Tests? y
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**, the settings are as follow:

- Set of **Network Call Redirection** flag to *y* to enable the use of SIP REFER message to transfer calls back to the PSTN as service provider does support it. It can also be set to *n* if the use of re-INVITE for call re-direction is preferred.
- Set the **Send Diversion Header** field to *y* as service provider does support it.
- Set the **Support Request History** field to *n*.
- Set the **Telephone Event Payload Type** to *101*.

add trunk-group 2		Page 4 of 21
PROTOCOL VARIATIONS		
Mark Users as Phone? n		
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n		
Send Transferring Party Information? n		
Network Call Redirection? y		
Build Refer-To URI of REFER From Contact For NCR? n		
Send Diversion Header? y		
Support Request History? n		
Telephone Event Payload Type: 101		
Convert 180 to 183 for Early Media? n		
Always Use re-INVITE for Display Updates? n		
Identity for Calling Party Display: P-Asserted-Identity		
Block Sending Calling Party Location in INVITE? n		
Accept Redirect to Blank User Destination? n		
Enable Q-SIP? n		
...		

## 5.8. Calling Party Information

The calling party number is sent in the SIP “From”, “Contact” and “PAI” headers. Since public numbering is selected to define the format of this number (**Section 0**), 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 service provider. They are used to authenticate the caller.

The screen below shows a subset of the 10 digits DID numbers assigned for testing. These 4 numbers were mapped to the 4 enterprise extensions 60396, 60397, 60379 and 60398. These same 12-digit numbers were used in the outbound calling party information on the service provider trunk when calls were originated from these extensions.

change public-unknown-numbering 0					Page 1 of 2
NUMBERING - PUBLIC/UNKNOWN FORMAT					
Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	Total Len	
5	60396	2	441189099571	12	Total Administered: 6
5	60397	2	441189099572	12	Maximum Entries: 240
5	60379	2	441189099573	12	
5	60398	2	441189099574	12	

## 5.9. Incoming Call Handling Treatment

In general, the incoming call handling treatment for a trunk group can be used to manipulate the digits received for an incoming call if necessary. DID number sent by Pure IP can be mapped to an extension using the incoming call handling treatment of the receiving trunk-group. Use the **change inc-call-handling-trmt trunk-group** 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	12	441189099571	12	60396	
public-ntwrk	12	441189099572	12	60397	
public-ntwrk	12	441189099573	12	60379	
public-ntwrk	12	441189099574	12	60398	

## 5.10. Outbound Routing

In these Application Notes, the **Automatic Route Selection (ARS)** feature is used to route an outbound call via the SIP trunk to the service provider via the Avaya SBCE. In the compliance testing, a single digit 9 was used as the ARS access code. An enterprise caller will dial 9 to reach an outside line. To define feature access code (**fac**) **9**, use the **change dialplan analysis** command as shown below.

<b>change dialplan analysis</b>			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	
1	4	ext							
30	4	ext							
39	5	udp							
60	5	ext							
<b>9</b>	<b>1</b>	<b>fac</b>							
*	3	dac							
#	3	dac							

Use the **change feature-access-codes** command to define **9** as the **Auto Route Selection (ARS)** – **Access Code 1**.

<b>change feature-access-codes</b>			FEATURE ACCESS CODE (FAC)						Page 1 of 10
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:			*05						
Answer Back Access Code:									
Attendant Access Code:									
Auto Alternate Routing (AAR) Access Code:									
<b>Auto Route Selection (ARS) - Access Code 1: 9</b>			Access Code 2:						

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9. The example pattern below shows a sample of the dialed strings calling on service provider. All dialed strings are mapped to route pattern **2** for an outbound call which contains the SIP trunk to the service provider (as defined next).

change ars analysis 0							Page 1 of 2
ARS DIGIT ANALYSIS TABLE							
Location: all							Percent Full: 0
	Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
	011	3	36	2	intl		n
	1	11	11	2	svcl		n
	44	5	10	2	svcl		n

As mentioned above, the route pattern defines which trunk group will be used for the outbound calls and performs necessary digit manipulation. Use the **change route-pattern** command to configure the parameters for route pattern **2** in the following manner.

- **Pattern Name:** Enter a descriptive name.
- **Grp No:** Enter the outbound trunk group for the SIP service provider. For the compliance testing, trunk group **2** was used.
- **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:** *pub-unk*. All calls using this route pattern will use the public numbering table as shown in **Section 5.8**.

change route-pattern 2														Page 1 of 3
Pattern Number: 2							Pattern Name: SP Route							
SCCAN? n							Secure SIP? n							
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted					DCS/	IXC	
No			Mrk	Lmt	List	Del	Digits					QSIG		
							Dgts					Intw		
1:	2	0										n	user	
2:												n	user	
....														
	BCC	VALUE	TSC	CA-TSC		ITC	BCIE	Service/Feature	PARM	No.	Numbering	LAR		
	0	1	2	M	4	W	Request			Dgts	Format			
										Subaddress				
1:	y	y	y	y	y	n	n	rest			pub-unk	none		
...														

## 5.11. Saving Communication Manager Configuration Changes

The command “**save translation all**” can be used to save the configuration changes made on Communication Manager.

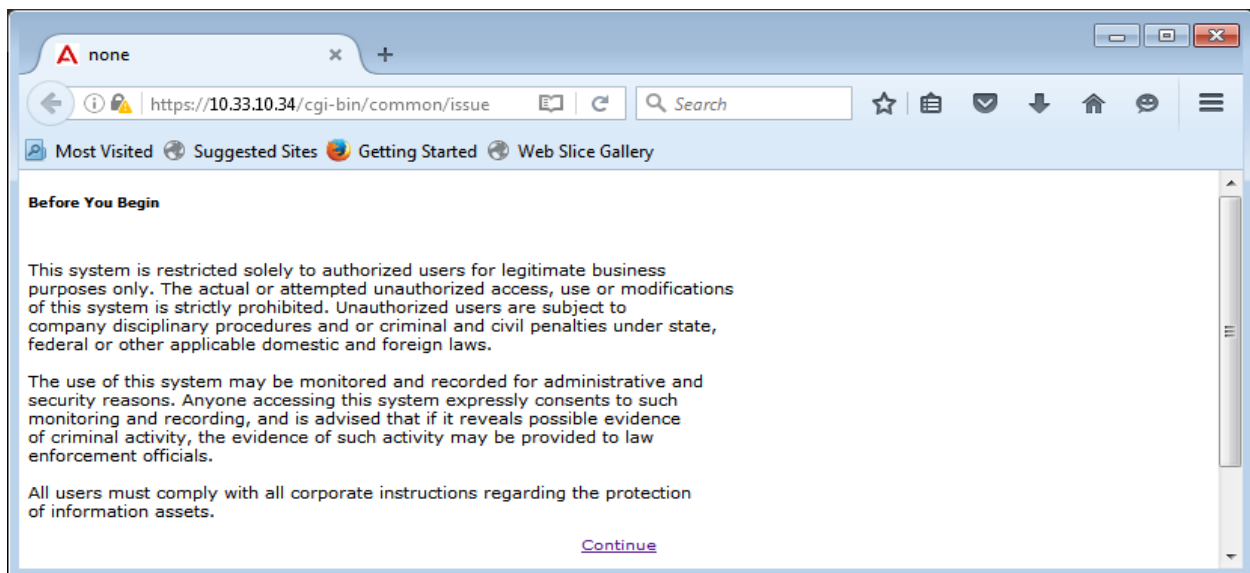
## 5.12. TLS Management on Communication Manager

It is (or maybe) necessary to install System Manager CA certificate on Communication Manager for the TLS signalling to work between Avaya Session Manager and Avaya Communication Manager if it is not previously installed.

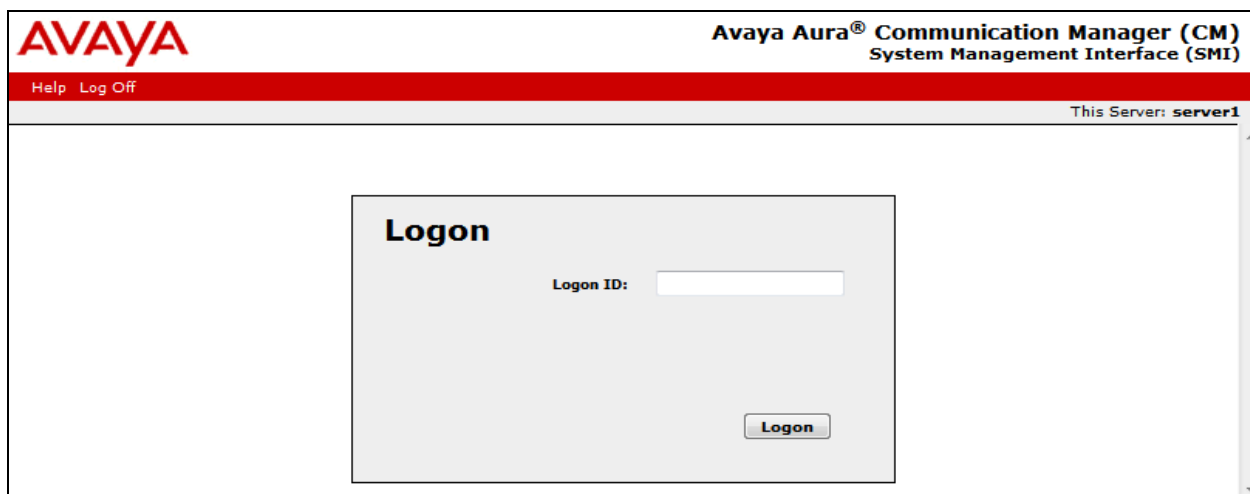
This section is to show how to install System Manager CA certificate on Communication Manager using Web console.

System Manager CA certificate is obtained using procedure provided in **Section 6.9**.

From a web browser, type in “https://<ip-address>”, where “<ip-address>” is the IP address or FQDN of Communication Manager. Click on **Continue** and it will be redirect to login page.



At login page, type in the login ID and its password credential.





Click on **Continue** again (not shown), navigate to **Administration** → **Server** → **Trusted Certificates** to verify if the System Manager CA certificate is present or not. If it is not, then continue to the next step.

The screenshot shows the Avaya Aura Communication Manager (CM) System Management Interface (SMI) for server1. The left sidebar contains a navigation menu with categories: Administration / Server (Maintenance), Security, and Miscellaneous. The main content area is titled "Trusted Certificates" and includes a description: "This page provides management of the trusted security certificates present on this server." Below this, there are "Trusted Repositories" defined: A = Authentication, Authorization and Accounting Services (e.g. LDAP), C = Communication Manager, W = Web Server, and R = Remote Logging. A table lists three certificates:

Select File	Issued To	Issued By	Expiration Date	Trusted By
<input type="radio"/> apr-ca.crt	Avaya Product Root CA	Avaya Product Root CA	Sun Aug 14 2033	C W R
<input type="radio"/> motorola_sscca_root.crt	SCCAN Server Root CA	SCCAN Server Root CA	Sun Dec 04 2033	C
<input type="radio"/> sip_product_root.crt	SIP Product Certificate Authority	SIP Product Certificate Authority	Tue Aug 17 2027	C W R

At the bottom of the table are buttons: Display, Add, Remove, Copy, and Help.

Navigate to **Miscellaneous** → **Download Files**, click on **File** to download from the machine I'm using to connect to the server and click on **Browse** to where the System Manager CA is being located. Then click on **Download** button to load the System Manager CA on Communication Manager Server.

The screenshot shows the Avaya Aura Communication Manager (CM) System Management Interface (SMI) for server1, specifically the "Download Files" page. The left sidebar is the same as the previous screenshot, but the "Download Files" option under the Miscellaneous category is selected. The main content area is titled "Download Files" and includes a description: "The Download Files SMI page lets you download files to the server." There are two radio buttons for selection:

- ☐ File(s) to download from the machine I'm using to connect to the server. Below this are four "Browse..." buttons, each followed by the text "No file selected."
- ☐ File(s) to download from the LAN using URL. Below this are four empty text input fields.

At the bottom, there is a "Proxy Server" label followed by a text input field and the text "(e.g proxy.domain:3152)". Below these are "Download" and "Help" buttons.

Navigate to **Security** → **Trusted Certificates**, click on **Add** button and enter the certificate name which has been downloaded from above step. Then click **Open**.

**AVAYA** Avaya Aura® Communication Manager (CM) System Management Interface (SMI)

Help Log Off Administration This Server: server1

Administration / Server (Maintenance)

**Trusted Certificates - Add**

This page allows for the addition of a trusted certificate to this server.

SystemManagerCA.pem PEM file containing certificate

Open Cancel Help

Enter the name of the System Manager CA certificate to store the certificate in Communication Manager. Check the Communication Manager check-box. Then click **Add**.

**AVAYA** Avaya Aura® Communication Manager (CM) System Management Interface (SMI)

Help Log Off Administration This Server: server1

Administration / Server (Maintenance)

**Trusted Certificates**

This page provides management of the trusted security certificates present on this server.

**Add this certificate**

Issued To	Issued By	Expiration Date
System Manager CA	System Manager CA	Sat Aug 23 2025

SystemManagerCA Store the certificate in this file in each repository selected below

**Add to these trusted repositories**

- ☐ Authentication, Authorization and Accounting Services (e.g. LDAP)
- ☒ Communication Manager
- ☐ Web Server
- ☐ Remote Logging

Add Cancel Help

Navigate to **Security** → **Trusted Certificates** again. It now shows the System Manager CA in the **Trusted Repositories**.

**AVAYA** Avaya Aura® Communication Manager (CM) System Management Interface (SMI)

Help Log Off Administration This Server: server1

Administration / Server (Maintenance)

**Trusted Certificates**

This page provides management of the trusted security certificates present on this server.

**Trusted Repositories**

A = Authentication, Authorization and Accounting Services (e.g. LDAP)  
C = Communication Manager  
W = Web Server  
R = Remote Logging

Select File	Issued To	Issued By	Expiration Date	Trusted By
<input type="radio"/> SystemManagerCA.crt	System Manager CA	System Manager CA	Sat Aug 23 2025	C
<input type="radio"/> apr-ca.crt	Avaya Product Root CA	Avaya Product Root CA	Sun Aug 14 2033	C W R
<input type="radio"/> motorola_sseca_root.crt	SCCAN Server Root CA	SCCAN Server Root CA	Sun Dec 04 2033	C
<input type="radio"/> sip_product_root.crt	SIP Product Certificate Authority	SIP Product Certificate Authority	Tue Aug 17 2027	C W R

Display Add Remove Copy Help

## 6. Configure Avaya Aura® Session Manager

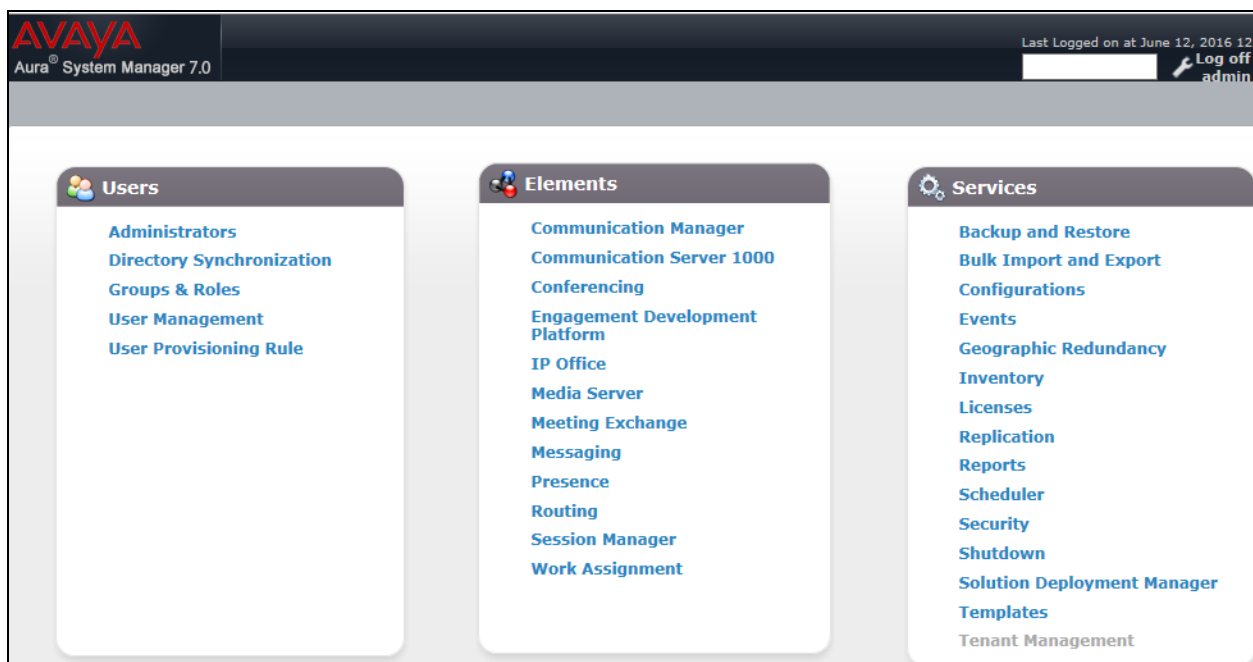
This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain.
- Logical/physical Location that can be used by SIP Entities.
- 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.
- Session Manager, corresponding to the Session Manager server to be managed by System Manager.

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

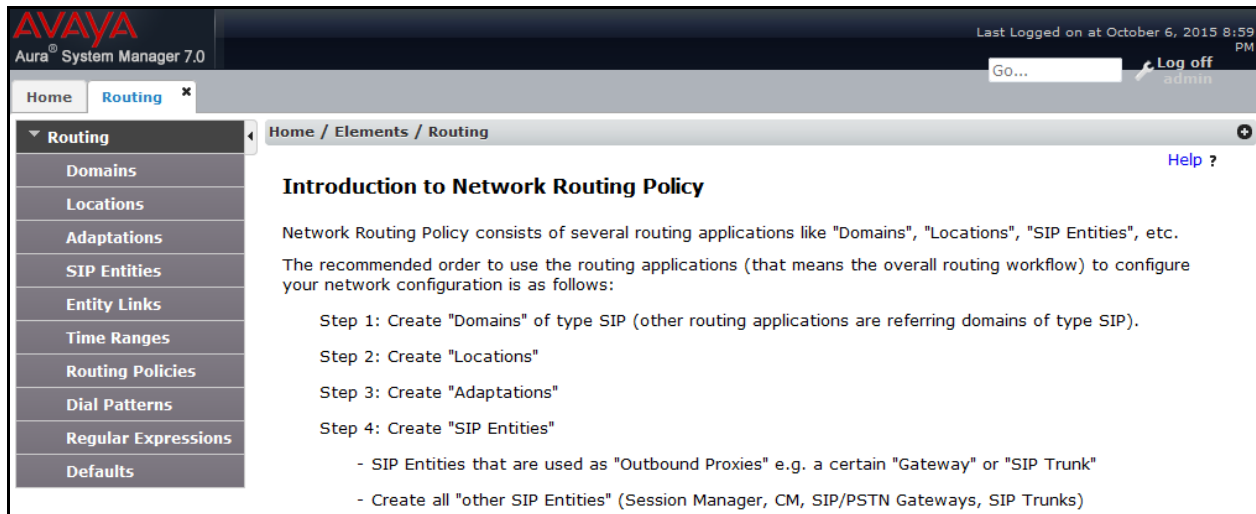
### 6.1. System Manager Login and Navigation

Session Manager configuration is accomplished by accessing the Web GUI of System Manager, using the URL “https://<ip-address>/SMGR”, where “<ip-address>” is the IP address or FQDN of System Manager. At the **System Manager Log On** screen, provide the appropriate credentials and click on **Login** (not shown). The initial screen shown below is then displayed.



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

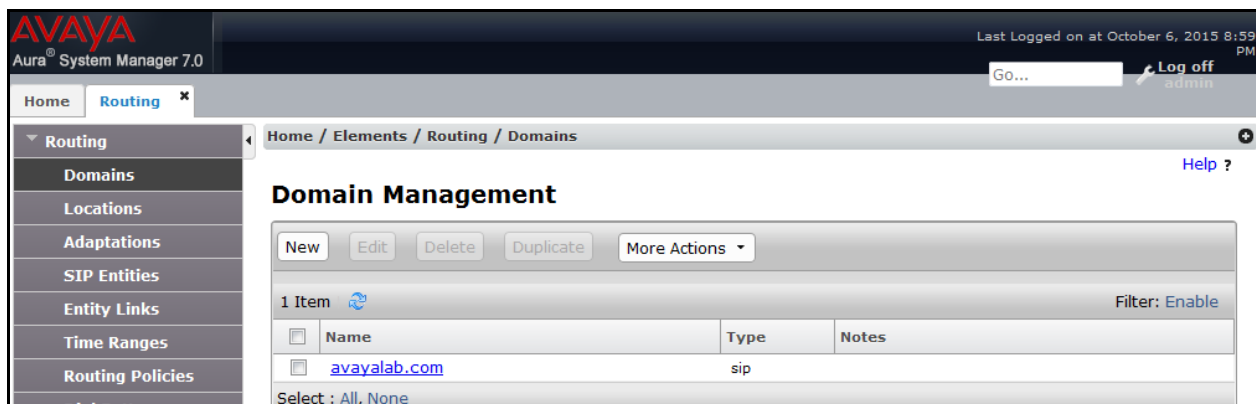
The navigation tree displayed in the left pane will be referenced in subsequent sections to navigate to items requiring configuration.



## 6.2. Specify SIP Domain

To view or to change SIP domains, select **Routing** → **Domains**. Click on the checkbox next to the name of the SIP domain and **Edit** to edit an existing domain, or the **New** button to add a domain. Click the **Commit** button (not shown) after changes are completed.

The following screen shows the list of configured SIP domains. The domain, **avayalab.com** was already created for communication between Session Manager and Communication Manager. The domain **avayalab.com** is not known to Pure IP. It will be adapted by the Avaya SBCE to IP address based URI-Host to meet the SIP specification of Pure IP system.



## 6.3. Add Location

Locations can be used to identify logical and/or physical locations where SIP Entities reside for bandwidth management and call admission control purposes. To add a location, navigate to **Routing → Locations** in the left-hand navigation pane and click **New** button in the right pane (not shown).

In **General** section, enter the following values:

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

In the **Location Pattern** section (see the screen below), click **Add** and configure following fields:

- **IP Address Pattern:** An IP address pattern used to identify the location.
- **Notes:** Add a brief description (optional).

Displayed below are the screenshots for location **Belleville**, which includes all equipment on the **10.33.\***, **10.10.98.\*** and **10.10.97.\*** subnet including Communication Manager, Session Manager and Avaya SBCE. Click **Commit** to save.

The screenshot displays the Avaya Aura System Manager 7.0 web interface. The left-hand navigation pane shows the 'Routing' menu expanded, with 'Locations' selected. The main content area is titled 'Location Details' and contains the following sections:

- General:** Includes fields for 'Name' (set to 'Belleville') and 'Notes' (set to 'GSSCP Belleville').
- Dial Plan Transparency in Survivable Mode:** Includes an 'Enabled' checkbox (unchecked), a 'Listed Directory Number' field, and an 'Associated CM SIP Entity' field.
- Overall Managed Bandwidth:** Includes 'Managed Bandwidth Units' (set to 'Kbit/sec'), 'Total Bandwidth' (set to '10000000'), 'Multimedia Bandwidth' (set to '10000000'), and an 'Audio Calls Can Take Multimedia Bandwidth' checkbox (checked).
- Location Pattern:** Includes an 'Add' button, a 'Remove' button, and a table listing 3 items. The table has columns for 'IP Address Pattern' and 'Notes'. The listed items are:
 

IP Address Pattern	Notes
* 10.33.*	
* 10.10.97.*	
* 10.10.98.*	

The interface also shows a 'Commit' button and a 'Cancel' button at the top right of the form area.

## 6.4. Add 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 Avaya SBCE.

To add a new SIP Entity, navigate to **Routing** → **SIP Entities** in the left navigation pane and click **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.
- **Type:** Select *Session Manager* for Session Manager, *CM* for Communication Manager and *SIP Trunk* for the Avaya SBCE.
- **Location:** Select the location defined in **Section Error! Reference source not found.**
- **Time Zone:** Select the time zone for the location above.

The following screen shows the addition of Session Manager SIP Entity. The IP address of the Session Manager signaling interface is entered for **FQDN or IP Address**.

The screenshot displays the Avaya Aura System Manager 7.0 web interface. The top navigation bar includes the Avaya logo, the text 'Aura System Manager 7.0', and a 'Last Logged on at October 6, 2015 8:59 PM' timestamp. A search bar with 'Go...' and a 'Log off' button are also present. The left sidebar shows a navigation menu with 'Routing' selected, and a sub-menu with 'SIP Entities' highlighted. The main content area is titled 'SIP Entity Details' and contains a 'General' section. The form fields are as follows: 'Name' (SM7), 'FQDN or IP Address' (10.33.10.33), 'Type' (Session Manager), 'Notes' (empty), 'Location' (Belleville), 'Outbound Proxy' (empty), 'Time Zone' (America/Toronto), and 'Credential name' (empty). At the bottom, there is a 'SIP Link Monitoring' section with a dropdown menu set to 'Use Session Manager Configuration'. 'Commit' and 'Cancel' buttons are located at the top right of the form area.

To define the ports used by Session Manager, scroll down to the **Port** section of the **SIP Entity Details** screen. This section is only present for the **Session Manager** SIP Entity.

In the **Port** section, click **Add** and enter following values. Use default values for all remaining fields:

- **Listen Ports:** Port number on which the Session Manager can listen for SIP requests.
- **Protocol:** Transport protocol to be used to receive SIP requests.

- **Default Domain:** The domain used for the enterprise.

Defaults can be used for the remaining fields. Click **Commit** to save (not shown).

The compliance test used **Listen Ports** entry **5061** with **TLS** for connecting to Communication Manager and for connecting to the Avaya SBCE.

**Listen Ports**

TCP Failover port:

TLS Failover port:

Add Remove

6 Items Filter: Enable

<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	TCP	avayalab.com	
<input type="checkbox"/>	5060	UDP	avayalab.com	
<input type="checkbox"/>	5061	TLS	avayalab.com	

Select : All, None

The following screen shows the addition of the Communication Manager SIP Entity. In order for Session Manager to send SIP traffic on an entity link to Communication Manager, it is necessary to create a SIP Entity for Communication Manager. The **FQDN or IP Address** field is set to IP address of Communication Manager and **Type** to **CM**. The **Location** and **Time Zone** parameters are set as shown in screen below.

**AVAYA**  
Aura® System Manager 7.0

Last Logged on at October 6, 2015 8:59 PM

Go... Log off admin

Home Routing

Home / Elements / Routing / SIP Entities

**SIP Entity Details**

Commit Cancel

**General**

\* Name: CM7

\* FQDN or IP Address: 10.33.10.34

Type: CM

Notes:

Adaptation:

Location: Belleville

Time Zone: America/Toronto

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

The following screen shows the addition of the SIP Entity for the Avaya SBCE. The **FQDN or IP Address** field is set to the IP address of its private network interface (see **Figure 1**). Select **Type** as *SIP Trunk*. Select **SIP Link Monitoring** as **Link Monitoring Enabled** with the interval of **120** seconds. This setting allows Session Manager to send outbound OPTIONS heartbeat every **120** seconds to service provider (which is forwarded by the Avaya SBCE) to query the status of the SIP trunk connecting to service provider.

**AVAYA**  
Aura® System Manager 7.0

Last Logged on at January 15, 2016 12:03 PM  
Go... Log off admin

Home Routing

Home / Elements / Routing / SIP Entities

**SIP Entity Details** Commit Cancel Help ?

**General**

\* Name: SBCE22

\* FQDN or IP Address: 10.10.98.22

Type: SIP Trunk

Notes: Avaya Aura SBC-E using IP 98.22

Adaptation:

Location: Belleville

Time Zone: America/Toronto

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

Credential name:

Securable:

Call Detail Recording: none

**Loop Detection**

Loop Detection Mode: Off

**SIP Link Monitoring**

SIP Link Monitoring: Link Monitoring Enabled

\* Proactive Monitoring Interval (in seconds): 120

\* Reactive Monitoring Interval (in seconds): 120

\* Number of Retries: 5

Supports Call Admission Control:



Similarly, a SIP Entity is added for Avaya Messaging server as shown in the capture below.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The top header includes the Avaya logo, 'Aura System Manager 7.0', and a 'Last Logged on' timestamp. The left navigation pane has 'Routing' expanded, with 'SIP Entities' selected. The main content area is titled 'SIP Entity Details' and contains a 'General' tab. The form fields are as follows:

- Name:** AAM
- \* FQDN or IP Address:** 10.33.10.35
- Type:** Modular Messaging
- Notes:** (empty text area)
- Adaptation:** (empty dropdown)
- Location:** Belleville
- Time Zone:** America/Toronto
- \* SIP Timer B/F (in seconds):** 4
- Credential name:** (empty text area)
- Securable:** (checkbox, unchecked)
- Call Detail Recording:** none
- Loop Detection Mode:** Off
- SIP Link Monitoring:** Use Session Manager Configuration

## 6.5. Add Entity Links

A SIP trunk between Session Manager and a telephony entity is described by an Entity Link. During compliance testing, three Entity Links were created, one for Communication Manager, Avaya Messaging and other for Avaya SBCE. To add an Entity Link, navigate to **Routing** → **Entity Links** in the left navigation pane and click **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 entity defined in **Section 6.4**.
- **Protocol:** Select the transport protocol used for this link, **TLS** for the Entity Link to Communication Manager and Avaya Messaging and **TLS** for the Entity Link to the Avaya SBCE.
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end. For Communication Manager, this must match the **Far-end Listen Port** defined on the Communication Manager in **Section 5.6**.
- **SIP Entity 2:** Select the name of the other systems. For Communication Manager, select the Communication Manager SIP Entity defined in **Section Error! Reference source not found..** For Avaya SBCE, select Avaya SBCE SIP Entity defined in **Section Error! Reference source not found..**

- **Port:** Port number on which the other system receives SIP requests from Session Manager. For Communication Manager, this must match the **Near-end Listen Port** defined on the Communication Manager in **Section 5.6**.
- **Connection Policy:** Select **Trusted**. **Note:** If this is not selected, calls from the associated SIP Entity specified in **Section Error! Reference source not found**. will be denied.
- Click **Commit** to save.

The following screens illustrate the Entity Links to Communication Manager and to the Avaya SBCE.

### Entity Link to Communication Manager

The screenshot shows the Avaya Aura System Manager 7.0 interface. The left sidebar contains a navigation menu with options: Home, Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links (selected), Time Ranges, and Routing Policies. The main content area is titled 'Entity Links' and includes 'Commit' and 'Cancel' buttons. Below the title, there is a table with 1 item. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, DNS Override, Port, and Connection Policy. The row shows: Name: \*SM7\_CM7\_5061\_TLS, SIP Entity 1: \*Q SM7, Protocol: TLS, Port: \*5061, SIP Entity 2: \*Q CM7, DNS Override: (empty), Port: \*5061, and Connection Policy: trusted.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy
*SM7_CM7_5061_TLS	*Q SM7	TLS	*5061	*Q CM7		*5061	trusted

### Entity Link to Avaya SBCE

The screenshot shows the Avaya Aura System Manager 7.0 interface. The left sidebar contains a navigation menu with options: Home, Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links (selected), Time Ranges, and Routing Policies. The main content area is titled 'Entity Links' and includes 'Commit' and 'Cancel' buttons. Below the title, there is a table with 1 item. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, DNS Override, Port, and Connection Policy. The row shows: Name: \*SM7\_SBCE22\_5061\_, SIP Entity 1: \*Q SM7, Protocol: TLS, Port: \*5061, SIP Entity 2: \*Q SBCE22, DNS Override: (empty), Port: \*5061, and Connection Policy: trusted.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy
*SM7_SBCE22_5061_	*Q SM7	TLS	*5061	*Q SBCE22		*5061	trusted

### Entity Link to Avaya Messaging

The screenshot shows the Avaya Aura System Manager 7.0 interface. The left sidebar contains a navigation menu with options: Home, Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links (selected), Time Ranges, and Routing Policies. The main content area is titled 'Entity Links' and includes 'Commit' and 'Cancel' buttons. Below the title, there is a table with 1 item. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, DNS Override, Port, and Connection Policy. The row shows: Name: \*SM-SP\_SP-AAM\_5061, SIP Entity 1: \*Q SM7, Protocol: TLS, Port: \*5061, SIP Entity 2: \*Q AAM, DNS Override: (empty), Port: \*5061, and Connection Policy: trusted.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy
*SM-SP_SP-AAM_5061	*Q SM7	TLS	*5061	*Q AAM		*5061	trusted

## 6.6. Add Routing Policies

Routing Policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section** Error! Reference source not found.. Three routing policies were added, one for Communication Manager, Avaya Messaging and other for Avaya SBCE. To add a routing policy, navigate to **Routing → Routing Policies** in the left navigation pane and click **New** button in the right pane (not shown). The following screen is displayed.

In the **General** section, configure the following fields:

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

In **SIP Entity as Destination** section, click **Select**. The **SIP Entity List** page opens (not shown). Select the appropriate SIP entity to which this routing policy applies and click **Select**. The selected SIP Entity is displayed in the **Routing Policy Details** page as shown below. Use default values for remaining fields. Click **Commit** to save.

The following screens show the Routing Policy for Communication Manager.

AVAYA  
Aura® System Manager 7.0

Last Logged on at October 6, 2015 8:59 PM

Go... Log off admin

Home Routing

Home / Elements / Routing / Routing Policies

Routing Policy Details

Commit Cancel Help ?

General

\* Name: To-CM7

Disabled: ☐

\* Retries: 0

Notes: Route to CM

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
CM7	10.33.10.34	CM	

The following screens show the Routing Policy for the Avaya SBCE.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The left navigation pane is expanded to 'Routing', and the 'Routing Policies' sub-menu is selected. The main content area displays the 'Routing Policy Details' for a policy named 'To-SBCE22'. The 'General' tab is active, showing fields for Name, Disabled, Retries, and Notes. Below this, the 'SIP Entity as Destination' section shows a table with one entry: 'SBCE22' with FQDN or IP Address '10.10.98.22', Type 'Other', and Notes 'Avaya Aura SBC-E using IP 98.22'.

**Routing Policy Details**

**General**

\* Name: To-SBCE22

Disabled: ☐

\* Retries: 0

Notes:

**SIP Entity as Destination**

Name	FQDN or IP Address	Type	Notes
SBCE22	10.10.98.22	Other	Avaya Aura SBC-E using IP 98.22

The following screens show the Routing Policy for the Avaya Messaging.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The left navigation pane is expanded to 'Routing', and the 'Routing Policies' sub-menu is selected. The main content area displays the 'Routing Policy Details' for a policy named 'To-AAM'. The 'General' tab is active, showing fields for Name, Disabled, Retries, and Notes. Below this, the 'SIP Entity as Destination' section shows a table with one entry: 'AAM' with FQDN or IP Address '10.33.10.35', Type 'Modular Messaging', and Notes 'Routing from SM to AAM'.

**Routing Policy Details**

**General**

\* Name: To-AAM

Disabled: ☐

\* Retries: 0

Notes: Routing from SM to AAM

**SIP Entity as Destination**

Name	FQDN or IP Address	Type	Notes
AAM	10.33.10.35	Modular Messaging	

## 6.7. Add Dial Patterns

Dial Patterns are needed to route specific calls through Session Manager. For the compliance testing, dial patterns were needed to route calls from Communication Manager to Avaya Messaging and from Communication Manager to Pure IP and vice versa. Dial Patterns define which routing policy will be selected for a particular call based on the dialed digits, destination domain and originating location. To add a dial pattern, navigate to **Routing → Dial Patterns** in the left navigation pane and click **New** button in the right pane (not shown).

In the **General** section, enter the following values:

- **Pattern:** Enter a dial string that will be matched against the “Request-URI” of the call.
- **Min:** Enter a minimum length used in the match criteria.
- **Max:** Enter a maximum length used in the match criteria.
- **SIP Domain:** Enter the destination domain used in the match criteria.
- **Notes:** Add a brief description (optional).

In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate originating location for use in the match criteria. Lastly, select the routing policy from the list that will be used to route all calls that match the specified criteria. Click **Select**.

Default values can be used for the remaining fields. Click **Commit** to save.

Two examples of the dial patterns used for the compliance testing are shown below, one for outbound calls from the enterprise to the PSTN and one for inbound calls from the PSTN to the enterprise.

The first example shows that 11-digit dialed numbers that have a destination domain of “avayalab.com” uses route policy to Avaya SBCE as defined in **Section** Error! Reference source not found..

AVAYA  
Aura® System Manager 7.0

Last Logged on at November 24, 2016 9:53 AM

Go... Log off

Home Routing

Home / Elements / Routing / Dial Patterns

**Dial Pattern Details**

Commit Cancel

**General**

\* Pattern: 1

\* Min: 1

\* Max: 36

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: avayalab.com

Notes: Outgoing to SBCE to SP

**Originating Locations and Routing Policies**

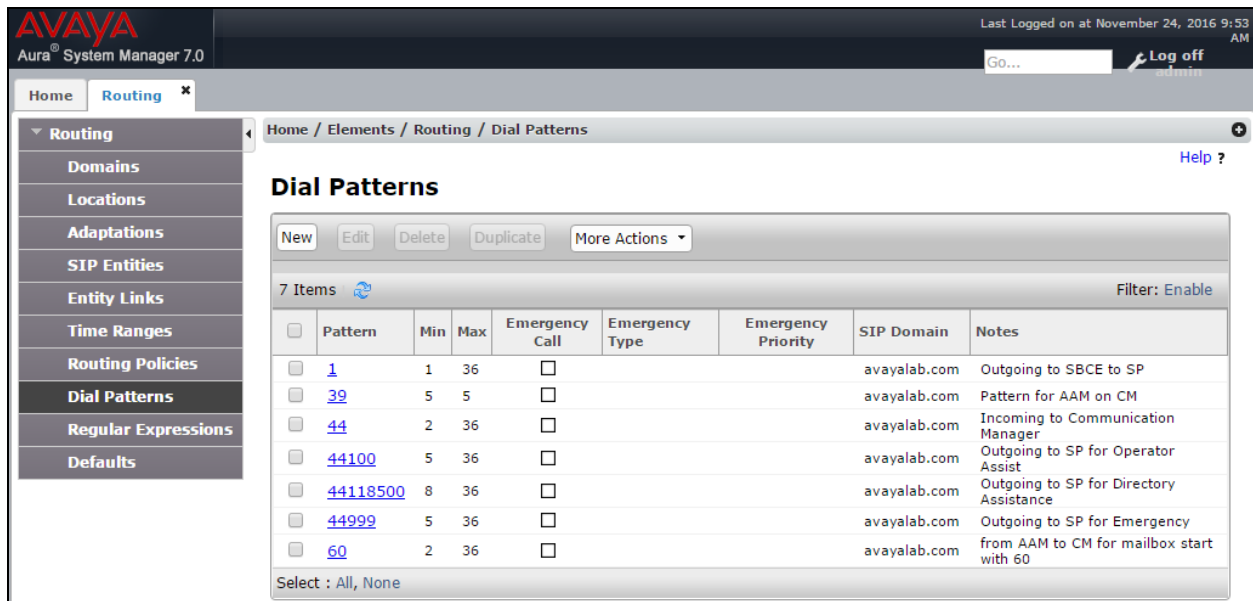
Add Remove

1 Item Filter: Enable

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Belleville	GSSCP Belleville	To-SBCE22	0	<input type="checkbox"/>	SBCE22	

Select : All, None

Similarly, the some other dial patterns can be defined for numbering plan mentioning in **Section 5.10**.



The screenshot shows the Avaya Aura System Manager 7.0 interface. The left sidebar contains a navigation menu with options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, **Dial Patterns**, Regular Expressions, and Defaults. The main content area is titled 'Dial Patterns' and shows a table with 7 items. The table columns are: Pattern, Min, Max, Emergency Call, Emergency Type, Emergency Priority, SIP Domain, and Notes. The SIP Domain for all listed patterns is 'avayalab.com'.

Pattern	Min	Max	Emergency Call	Emergency Type	Emergency Priority	SIP Domain	Notes
1	1	36	<input type="checkbox"/>			avayalab.com	Outgoing to SBCE to SP
39	5	5	<input type="checkbox"/>			avayalab.com	Pattern for AAM on CM
44	2	36	<input type="checkbox"/>			avayalab.com	Incoming to Communication Manager
44100	5	36	<input type="checkbox"/>			avayalab.com	Outgoing to SP for Operator Assist
44118500	8	36	<input type="checkbox"/>			avayalab.com	Outgoing to SP for Directory Assistance
44999	5	36	<input type="checkbox"/>			avayalab.com	Outgoing to SP for Emergency
60	2	36	<input type="checkbox"/>			avayalab.com	from AAM to CM for mailbox start with 60

The second example shows that inbound 12-digit numbers assigned by Pure IP with domain “avayalab.com” to use route policy to Communication Manager as defined in **Section Error!** Reference source not found..

**AVAYA**  
Aura® System Manager 7.0

Last Logged on at November 24, 2016 9:53 AM

Go... Log off admin

Home Routing x

Home / Elements / Routing / Dial Patterns

**Dial Pattern Details** Commit Cancel [Help ?](#)

**General**

\* Pattern: 44

\* Min: 2

\* Max: 36

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: avayalab.com

Notes: Incoming to Communication Manager

**Originating Locations and Routing Policies**

Add Remove

1 Item [Filter: Enable](#)

<input type="checkbox"/>	Originating Location Name ▲	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Belleville	GSSCP Belleville	To-CM7	0	<input type="checkbox"/>	CM7	

Select : All, None

## 6.8. Add/View Session Manager

The creation of a Session Manager element provides the linkage between System Manager and Session Manager. This is most likely done as part of the initial Session Manager installation. To add a Session Manager, navigate to **Home → Elements → Session Manager → Session Manager Administration** in the left navigation pane and click **New** button in the right pane (not shown). If the Session Manager Instance already exists, click **View** (not shown) to view the configuration. Enter/verify the data as described below and shown in the following screen:

In the **General** section, configure the following fields:

- **SIP Entity Name:** Select the SIP Entity created for Session Manager.
- **Description:** Add a brief description (optional).
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.
- **Directs Routing to Endpoints:** Enabled, to enable call routing on the Session Manager.

In the **Security Module** section, enter the following values:

- **SIP Entity IP Address:** Should be filled in automatically based on the SIP Entity Name.
- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager.
- **Default Gateway:** Enter the IP address of the default gateway for Session Manager.
- Use default values for the remaining fields. Click **Commit** to save (not shown).

The screen below shows the Session Manager values used for the compliance testing.

The screenshot displays the Avaya Aura System Manager 7.0 interface. The top header shows the Avaya logo and 'Aura System Manager 7.0'. The user is logged in as 'admin' and the session expires at 8:59 PM on October 6, 2015. The breadcrumb trail is 'Home / Elements / Session Manager / Session Manager Administration'. The left sidebar contains a navigation menu with options like Session Manager, Dashboard, Session Manager Administration, Communication Profile Editor, Network Configuration, Device and Location Configuration, Application Configuration, System Status, System Tools, and Performance. The main content area is titled 'View Session Manager' and includes a 'Return' button. Below the title is a navigation bar with links: General | Security Module | Monitoring | CDR | Personal Profile Manager (PPM) - Connection Settings | Event Server | Expand All | Collapse All. The 'General' tab is active, showing fields for SIP Entity Name (SM7), Description, Management Access Point Host Name/IP (10.33.10.32), Direct Routing to Endpoints (Enable), and Maintenance Mode (unchecked). The 'Security Module' tab is also visible, showing fields for SIP Entity IP Address (10.33.10.33), Network Mask (255.255.255.0), Default Gateway (10.33.10.1), Call Control PHB (46), and SIP Firewall Configuration (SM 6.3.8.0).

AVAYA  
Aura System Manager 7.0

Last Logged on at October 6, 2015 8:59 PM

Go... Log off admin

Home / Elements / Session Manager / Session Manager Administration

Help ?

### View Session Manager

Return

General | Security Module | Monitoring | CDR | Personal Profile Manager (PPM) - Connection Settings | Event Server | Expand All | Collapse All

#### General

SIP Entity Name SM7

Description

Management Access Point Host Name/IP 10.33.10.32

Direct Routing to Endpoints Enable

Maintenance Mode ☐

#### Security Module

SIP Entity IP Address 10.33.10.33

Network Mask 255.255.255.0

Default Gateway 10.33.10.1

Call Control PHB 46


\*SIP Firewall Configuration SM 6.3.8.0

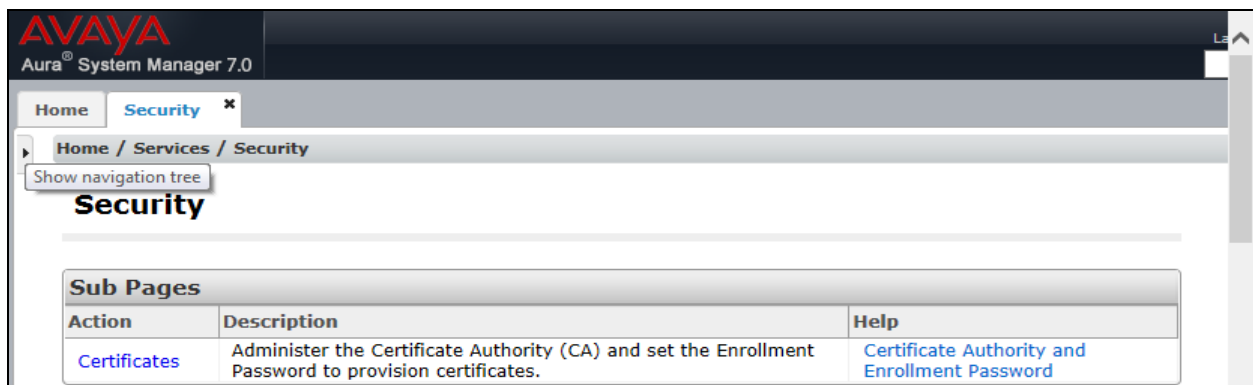


## 6.9. TLS Certificate Management on System Manager

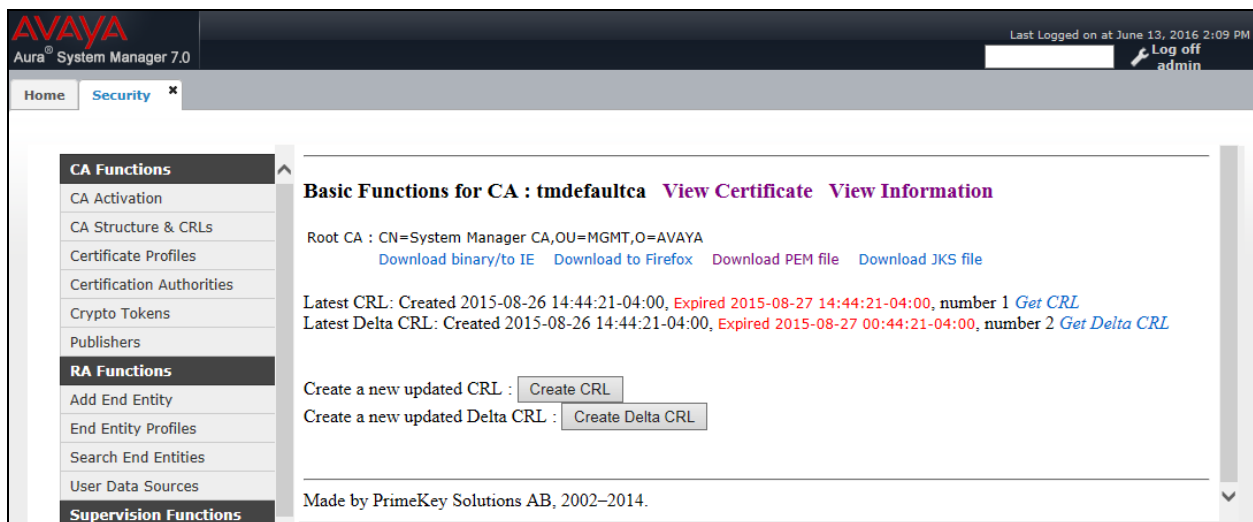
This section is to provide a procedure how to download System Manager CA certificate which is being installed on Avaya Communication Manager and Avaya SBCE for the communication between Avaya system components using TLS connectivity.

### How to download System Manager CA certificate from Avaya System Manager

From System Manager Menu in **Section 6.1**, navigate to **Services → Security**. Click on arrow tab  to show navigation tree as shown.



Navigate to **Certificates → Authority → CA Functions → CA Structure & CRLs**. Then click on **Download PEM file** to download the System Manager CA and save it as *SystemManagerCA.pem* to a directory on local management PC.



## 7. Configure Avaya Session Border Controller for Enterprise

In the sample configuration, an Avaya SBCE is used as the edge device between the Avaya CPE and Pure IP SIP Trunking Service.

These Application Notes assume that the installation of the Avaya SBCE and the assignment of a management IP Address have already been completed.

In this session, the naming convention used for Pure IP is Service Provider (SP), which is connected to the external interface of the Avaya SBCE. And for the Avaya side is Enterprise (EN), which is connected to the internal interface of the Avaya SBCE.

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

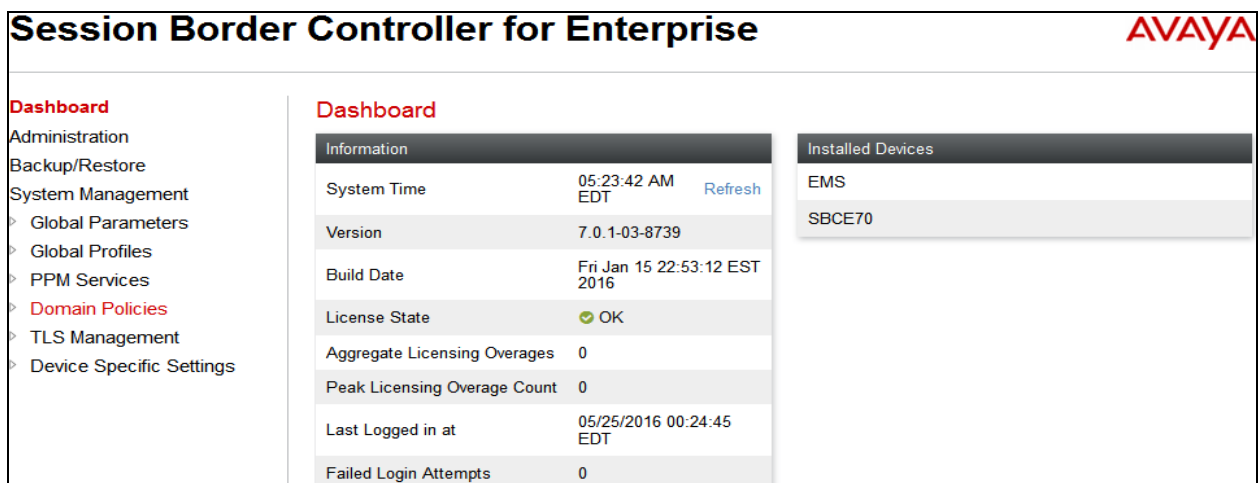
Use a Web browser to access the Avaya SBCE web interface, enter “https://<ip-addr>/ucsec” in the address field of the web browser (not shown), where “<ip-addr>” is the management LAN IP address of Avaya SBCE.

Enter appropriate credentials and click **Log In**.



The login page features the Avaya logo in red at the top left. Below it, the text "Session Border Controller for Enterprise" is displayed. To the right, under the heading "Log In", there is a "Username:" label followed by a text input field and a "Continue" button. Below the input field, there is a disclaimer: "This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use or modifications of this system is strictly prohibited. Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal or other applicable domestic and foreign laws." followed by a monitoring notice: "The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials." and a compliance statement: "All users must comply with all corporate instructions regarding the protection of information assets." At the bottom right, it says "© 2011 - 2015 Avaya Inc. All rights reserved."

The main page of the Avaya SBCE will appear as shown below.



The dashboard page has the title "Session Border Controller for Enterprise" and the Avaya logo in the top right. On the left is a navigation menu with items: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, PPM Services, Domain Policies, TLS Management, and Device Specific Settings. The main content area is titled "Dashboard" and contains two panels. The "Information" panel is a table with system details, and the "Installed Devices" panel lists connected devices.

Information	
System Time	05:23:42 AM EDT <a href="#">Refresh</a>
Version	7.0.1-03-8739
Build Date	Fri Jan 15 22:53:12 EST 2016
License State	OK
Aggregate Licensing Overages	0
Peak Licensing Overage Count	0
Last Logged in at	05/25/2016 00:24:45 EDT
Failed Login Attempts	0

Installed Devices
EMS
SBCE70

## 7.2. 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. The Avaya SBCE utilizes TLS primarily to facilitate secure communications with remote users.

Avaya SBCE is preinstalled with several certificates and profiles that can be used to quickly set up secure communication using TLS, which are listed in the Pre-installed Avaya Profiles and Certificates section. Session Manager, Avaya SBCE and the 96x1 IP Deskphones are shipped with default identity certificate to enable out-of-box support for TLS sessions. Do not use this default certificate in a production/customer environment since this certificate is common across all instances of Session Manager, Avaya SBCE and 96x1 IP Deskphones. Avaya SBCE supports the configuration of third-party certificates and TLS settings. For optimum security, Avaya recommends using third-party CA certificates for enhanced security

Testing was done with default identity certificates, the procedure to obtain and install 3<sup>rd</sup> party CA certificates is outside the scope of these application notes.

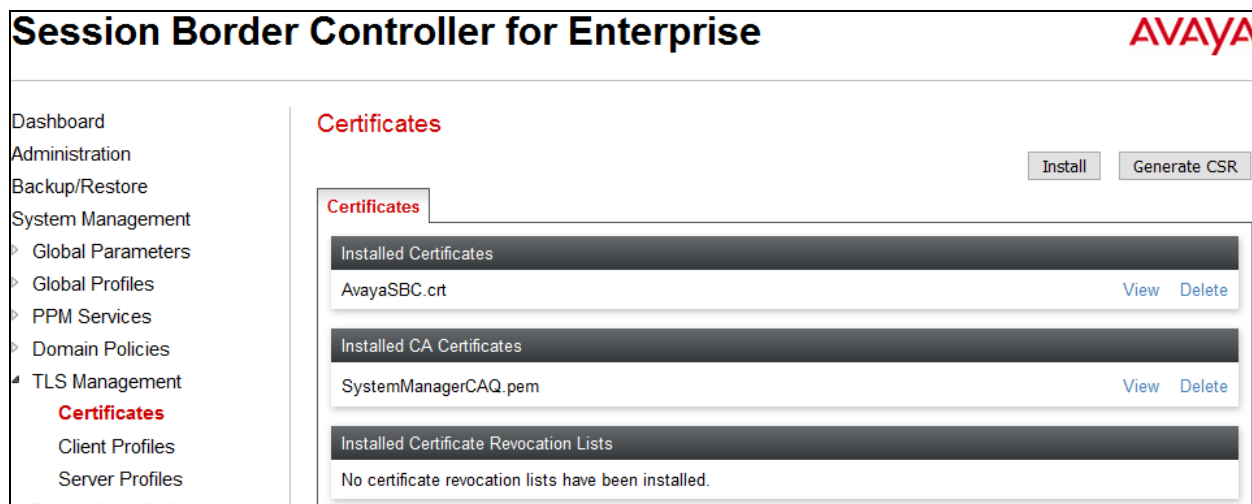
In this compliance testing, TLS transport is used for the communication between Avaya Session Manager and Avaya SBCE. The following procedures show how to create the client and server profiles.

### 7.2.1. Certificates

You can use the certificate management functionality that is built into the Avaya SBCE to control all certificates used in TLS handshakes. You can access the Certificates screen from **TLS Management → Certificates**.

Ensure the preinstalled certificates are presented in the system as shown below.

- *AvayaSBCcrt* is Avaya SBCE Certificate Authority root certificate.
- *SystemManagerCAQ.pem* is System Manager Certificate Authority root certificate.



If System Manager Certificate Authority certificate (SystemManagerCAQ.pem) is not present, the following procedure shows how to install it on the Avaya SBCE.

System Manager CA certificate is obtained using procedure provided in **Section 6.9**. Then on the Avaya SBCE, navigate to **TLS Management → Certificates**. Click on **Install** button.

- Select **CA Certificate**.
- Provide a descriptive **Name**.
- **Browse** to the directory where the System Manager CA previously saved and select it.
- Click **Upload**.

## 7.2.2. Client Profiles

This section describes the procedure to create client profile for Avaya SBCE to communicate with Avaya Session Manager via TLS signalling.

To create Client profile, navigate to **TLS Management** → **Client Profiles**, click on **Add**.

- Enter descriptive name in **Profile Name**.
- Select *AvayaSBC.crt* from pull down menu of **Certificate**.
- Select *SystemManagerCAQ.pem* from pull down of **Peer Certificate Authorities**.
- Enter **5** as **Verification Depth**.
- Click **Finish**.

The screenshot displays the 'Session Border Controller for Enterprise' web interface. On the left is a navigation menu with options like Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, PPM Services, Domain Policies, TLS Management (selected), Certificates, Client Profiles (highlighted), Server Profiles, and Device Specific Settings. The main content area is titled 'Client Profiles: AvayaSBCCclient-Q' and includes an 'Add' button. Below this is a list of client profiles: COLTClient, AvayaSBCCclient, AvayaSBCCclient-H, and AvayaSBCCclient-Q (selected). The 'Edit Profile' window for 'AvayaSBCCclient-Q' is open, showing a warning about OpenSSL cipher checking. The configuration fields are as follows:

TLS Profile	
Profile Name	AvayaSBCCclient-Q
Certificate	AvayaSBC.crt

Certificate Verification	
Peer Verification	Required
Peer Certificate Authorities	AvayaSBCCA.crt coltroot.crt Cisco_phone_CA.crt SystemManagerCAQ.pem (selected)
Peer Certificate Revocation Lists	
Verification Depth	5
Extended Hostname Verification	<input type="checkbox"/>
Custom Hostname Override	

At the bottom right of the 'Edit Profile' window is a 'Next' button.

### 7.2.3. Server Profiles

This section describes the procedure to create server profile for Avaya SBCE to communicate with Avaya Session Manager via TLS signalling.

To create Server profile, navigate to **TLS Management** → **Server Profiles**, click on **Add**.

- Enter descriptive name in **Profile Name**.
- Select **AvayaSBC.crt** from pull down menu of **Certificate**.
- Select **None** from pull down menu of **Peer Verification**.
- Others are left at default.
- Click Next and **Finish** (not shown).

The screenshot displays the 'Session Border Controller for Enterprise' web interface. On the left is a navigation menu with options: Dashboard, Administration, Backup/Restore, System Management (with sub-items: Global Parameters, Global Profiles, PPM Services, Domain Policies, TLS Management, and Device Specific Settings), Certificates, Client Profiles, and **Server Profiles**. The main content area is titled 'Server Profiles: AvayaSBCServer-Q' and includes an 'Add' button. Below this is a list of server profiles: COLTServer, AvayaSBCServer, AvayaSBCServer-H, and **AvayaSBCServer-Q**. The 'Edit Profile' window for 'AvayaSBCServer-Q' is open, showing a warning message: 'The selected certificate is known to have been compromised and should not be used in a production environment.' Below the warning is a 'WARNING' box stating: 'Due to the way OpenSSL handles cipher checking, Cipher Suite validation will pass even if one or more of the ciphers are invalid as long as at least one cipher is valid. Make sure to carefully check your entry as invalid or incorrectly entered Cipher Suite custom values may cause catastrophic problems.' The configuration fields are: TLS Profile (Profile Name: AvayaSBCServer-Q, Certificate: AvayaSBC.crt), Certificate Verification (Peer Verification: None, Peer Certificate Authorities: SystemManagerCA-H.pem, AvayaSBCCA.crt, coltroot.crt, Cisco\_phone\_CA.crt, Peer Certificate Revocation Lists: empty), and Verification Depth: 0. A 'Next' button is at the bottom right.

## 7.3. Global Profiles

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

### 7.3.1. Uniform Resource Identifier (URI) Groups

URI Group feature allows a user to create any number of logical URI Groups that are comprised of individual SIP subscribers located in that particular domain or group. These groups are used by the various domain policies to determine which actions (Allow, Block, or Apply Policy) should be used for a given call flow.

For this configuration testing, “\*” is used for all incoming and outgoing traffic.

### 7.3.2. Server Interworking Profile

Interworking Profile features are configured differently for Call Server and Trunk Server.

To create a Server Interworking profile, select **Global Profiles → Server Interworking**. Click on the **Add** button.

In the compliance testing, two Server Interworking profiles were created for SP and EN respectively.

#### Server Interworking profile for SP


Profile **SP-SI** was defined to match the specification of SP. The **General** and **Advanced** tabs are configured with the following parameters while the other tabs for **Timers**, **Privacy**, **URI Manipulation** and **Header Manipulation** are kept as default.

**General** tab:

- **Hold Support** = *NONE*. The Avaya SBCE will not modify the hold/ resume signaling from EN to SP.
- **18X Handling** = *None*. The Avaya SBCE will not handle 18X, it will keep the 18X messages unchanged from EN to SP.
- **Refer Handling** = *No*. The Avaya SBCE will not handle REFER. It will keep the REFER message unchanged from EN to SP.
- **T.38 Support** = *No*. SP does not support T.38 fax in the compliance testing.
- Others are left as default values.

The screenshots below illustrate the Server Interworking profile **SP-SI, General**.

# Session Border Controller for Enterprise



Dashboard

Administration

Backup/Restore

System Management

▸ Global Parameters

▾ Global Profiles

Domain DoS

**Server Interworking**

Media Forking

Routing

Server Configuration

Topology Hiding

Signaling Manipulation

URI Groups

SNMP Traps

Time of Day Rules

▸ PPM Services

▸ Domain Policies

▸ TLS Management

▸ Device Specific Settings

## Interworking Profiles: SP-SI

Add

Interworking Profiles

cs2100

EN-SI

**SP-SI**

Rename

Clone

Delete

Click here to add a description.

**General**

Timers

Privacy

URI Manipulation

Header Manipulation

Advanced

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	No
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

Edit



### Advanced tab:

- **Record Routes:** *Both Sides*.
- **Include End Point IP for Context Lookup:** *No*.
- **Extensions:** *None*.
- **Has Remote SBC:** *Yes*. SP has a SBC which interfaces its Central Office (CO) to the enterprise SIP trunk. This setting allows the Avaya SBCE to always use the SDP received from SP for the media.
- **DTMF Support:** *None*. The Avaya SBCE will send original DTMF method from EN to SP.
- Others are left as default values.

The screenshots below illustrate the Server Interworking profile **SP-SI**, **Advanced**.

The screenshot displays the Avaya Session Border Controller for Enterprise (SBCE) configuration interface. The left sidebar contains a navigation menu with categories like Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, and PPM Services. Under Global Profiles, 'Server Interworking' is highlighted. The main content area is titled 'Interworking Profiles: SP-SI' and includes an 'Add' button and action buttons (Rename, Clone, Delete). A list of profiles (cs2100, EN-SI, SP-SI) is shown, with 'SP-SI' selected. The 'Advanced' tab is active, displaying configuration settings for the selected profile. The settings are organized into sections: General (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) and DTMF (DTMF Support: None). An 'Edit' button is located at the bottom right of the configuration area.

Section	Setting	Value
General	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
DTMF	DTMF Support	None

## Server Interworking profile for EN

Profile **EN-SI** was defined to match the specification of EN. The **General** and **Advanced** tabs are configured with the following parameters while the other settings for **Timers**, **Privacy**, **URI Manipulation** and **Header Manipulation** are kept as default.

### General tab:

- **Hold Support:** *None*.
- **18X Handling:** *None*. The Avaya SBCE will not handle 18X, it will keep the 18X messages unchanged from SP to EN.
- **Refer Handling:** *No*. The Avaya SBCE will not handle REFER, it will keep the REFER messages unchanged from SP to EN.
- **T.38 Support:** *No*.
- Others are left as default values.

The screenshots below illustrate the Server Interworking profile **EN-SI**, **General**.

The screenshot displays the Avaya Session Border Controller for Enterprise (SBCE) web interface. The left sidebar contains a navigation menu with options like Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, Domain DoS, Server Interworking (highlighted), Media Forking, Routing, Server Configuration, Topology Hiding, Signaling Manipulation, URI Groups, SNMP Traps, Time of Day Rules, PPM Services, Domain Policies, TLS Management, and Device Specific Settings. The main content area is titled 'Interworking Profiles: EN-SI' and features a list of profiles (cs2100, EN-SI, SP-SI) with an 'Add' button. The 'EN-SI' profile is selected, and its configuration is shown in a tabbed interface. The 'General' tab is active, displaying a table of parameters and their values. The 'Advanced' tab is also visible. The top right corner of the interface shows the Avaya logo and buttons for 'Rename', 'Clone', and 'Delete'.

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	No
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

### Advanced tab:

- **Record Routes: Both Sides.** The Avaya SBCE will send Record-Route header to both call and trunk servers.
- **Include End Point IP for Context Lookup = Yes.**
- **Extensions: Avaya.**
- **Has Remote SBC: Yes.** This setting allows the Avaya SBCE to always use the SDP received from EN for the media.
- **DTMF Support: None.** The Avaya SBCE will send original DTMF method from SP to EN.
- Others are left as default values.

The screenshots below illustrate the Server Interworking profile **EN-SI**, **Advanced**.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The left sidebar contains a navigation menu with options like Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, Domain DoS, **Server Interworking**, Media Forking, Routing, Server Configuration, Topology Hiding, Signaling Manipulation, URI Groups, SNMP Traps, Time of Day Rules, and PPM Services. The main content area is titled 'Interworking Profiles: EN-SI' and features an 'Add' button and 'Rename', 'Clone', and 'Delete' buttons. Below this is a list of profiles: 'EN-SI' (selected) and 'SP-SI'. The 'Advanced' tab is active, showing 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), and 'DTMF Support' (None). An 'Edit' button is at the bottom right of the settings table.

General	Timers	Privacy	URI Manipulation	Header Manipulation	Advanced
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					
<b>DTMF</b>					
DTMF Support None					

### 7.3.3. Server Configuration

The Server Configuration screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. These tabs are used to configure and manage various SIP Call Server specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters, DoS security statistics and trusted domains. No configuration of **Heartbeat** is required.

To create a Server Configuration entry, select **Global Profiles → Server Configuration**. Click on the **Add** button.

In the compliance testing, two separate Server Configurations were created, server entry **SP-SC** for SP and server entry **EN-SC** for EN.

## Server Configuration for SP

Server Configuration named **SP-SC** was created for SP. All tabs are provisioned for SP on the SIP trunk for every outbound call from enterprise to PSTN.

### General tab:

Click on the **Add** button and enter the following information.

- Enter **Profile Name** *SP-SC* and click **Next**.
- Set **Server Type** for SP as *Trunk Server*.
- Enter **IP Addresss/FQDN** provided by SP.
- In the compliance testing, SP supported **UDP** and listened on port **5060**.
- Click **Next**, then **Next** and **Finish**.

The completed server profile is shown below.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The left sidebar contains a navigation menu with options like System Management, Global Parameters, Global Profiles, and Server Configuration. The main area is titled "Server Configuration: SP-SC" and features an "Add" button and "Rename", "Clone", and "Delete" buttons. Below these are tabs for "General", "Authentication", "Heartbeat", and "Advanced". The "General" tab is active, displaying a table with the following data:

IP Address / FQDN	Port	Transport
192.168.21.188	5060	UDP

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

### Advanced tab:

Click on the **Edit** button and enter following information.

- **Interworking Profile** drop down list, select *SP-SI* as defined in **Section 7.3.2**.
- The other settings are kept as default.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface, specifically the "Advanced" tab of the "Server Configuration: SP-SC" configuration. The left sidebar is the same as in the previous screenshot. The main area has tabs for "General", "Authentication", "Heartbeat", and "Advanced". The "Advanced" tab is active, displaying a list of settings:

Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	SP-SI
Signaling Manipulation Script	None
Securable	<input type="checkbox"/>
Enable FGDN	<input type="checkbox"/>

An "Edit" button is located at the bottom right of the settings list.

## Server Configuration for EN

Server Configuration named **EN-SC** created for EN is discussed in detail below. **General** and **Advanced** tabs are provisioned but no configuration is done for **Authentication** tab. The **Heartbeat** tab is kept as *disabled* as default to allow the Avaya SBCE to forward the OPTIONS heartbeat from SP to EN to query the status of the SIP trunk.

### General tab:

Click on the **Add** button and enter the following information.

- Enter **Profile Name** as *EN-SC* and click **Next**.
- **Server Type** for EN as *Call Server*.
- Select *AvayaSBCClien-Q* for **TLS Client Profile**.
- **IP Address/FQND** is Session Manager IP address.
- **Transport**, the link between the Avaya SBCE and EN was *TLS*.
- Listened on **Port 5061**.
- Click **Next**, **Next** and then **Finish**.

The completed server profile is shown below.

The screenshot displays the Avaya Session Border Controller for Enterprise (SBCE) web interface. The title bar at the top reads "Session Border Controller for Enterprise" with the AVAYA logo on the right. A left-hand navigation menu includes links for Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles (with sub-links for Domain DoS, Server Interworking, Media Forking, Routing, and Server Configuration), and Topology Hiding. The "Server Configuration: EN-SC" page is active, featuring an "Add" button and tabs for General, Authentication, Heartbeat, and Advanced. The "General" tab is selected, showing the following configuration:

Server Type	Call Server	
TLS Client Profile	AvayaSBCClien-Q	
IP Address / FQDN	Port	Transport
10.33.10.33	5060	TCP
10.33.10.33	5061	TLS

Buttons for "Rename", "Clone", "Delete", and "Edit" are also visible.

### Advanced tab:

Click on the **Edit** button to enter the following information.

- **Interworking Profile** drop down list select **EN-SI** as defined in **Section Error!** Reference source not found..
- The other settings are kept as default.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The title bar at the top reads "Session Border Controller for Enterprise" with the Avaya logo on the right. On the left is a navigation menu with categories: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, Domain DoS, Server Interworking, Media Forking, Routing, **Server Configuration**, Topology Hiding, Signaling Manipulation, and URL Groups. The main content area is titled "Server Configuration: EN-SC" and includes an "Add" button. Below this is a "Server Profiles" dropdown menu with "SP-SC" and "EN-SC" (highlighted with a red border). To the right are tabs for "General", "Authentication", "Heartbeat", and "Advanced" (selected). The "Advanced" tab contains a table of settings:

Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	EN-SI
Signaling Manipulation Script	None
Securable	<input type="checkbox"/>
Enable FGDN	<input type="checkbox"/>

At the bottom right of the settings table is an "Edit" button. Above the tabs are "Rename", "Clone", and "Delete" buttons.

### 7.3.4. Routing Profiles

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

To create a Routing Profile, select **Global Profiles → Routing**. Click on the **Add** button.

In the compliance testing, a Routing Profile **EN-to-SP** was created to use in conjunction with the server flow defined for EN. This entry is to route the outbound call from the enterprise to the service provider.

In the opposite direction, a Routing Profile named **SP-to-EN** was created to be used in conjunction with the server flow defined for SP. This entry is to route the inbound call from the service provider to the enterprise.

## Routing Profile for SP

The screenshot below illustrate the routing profile from Avaya SBCE to the SP network, **Global Profiles → Routing: EN-to-SP**. As shown in **Figure 1**, the SP SIP trunk is connected with transport protocol **UDP** (not shown). If there is a match in the “To” or “Request URI” headers with the URI Group “\*” as described in **Section 7.3.1**, the call will be routed to the **Next Hop Address** which is the IP address of SP SIP trunk.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The left sidebar contains a navigation menu with options: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles (selected), Domain DoS, Server Interworking, Media Forking, and Routing (highlighted in red). The main content area is titled "Routing Profiles: EN-to-SP" and includes an "Add" button, "Rename", "Clone", and "Delete" buttons. Below this is a description field with the text "Click here to add a description." and a "Routing Profile" tab. A table lists the routing profile configuration:

Priority	URI Group	Time of Day	Load Balancing	Next Hop Address	Transport	
1	*	default	Priority	192.168.21.188	UDP	Edit Delete

## Routing Profile for EN

The Routing Profile for SP to EN, **SP-to-EN**, was defined to route call where the “To” header matches the URI Group **SP** defined in **Section 7.3.1** to **Next Hop Address** which is the IP address of Session Manager as a destination. As shown in **Figure 1**, the SIP trunk between EN and the Avaya SBCE is connected with transport protocol **TLS**.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The left sidebar contains a navigation menu with options: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles (selected), Domain DoS, Server Interworking, Media Forking, Routing, and Server Configuration. The main content area is titled "Routing Profiles: SP-to-EN" and includes an "Add" button, "Rename", "Clone", and "Delete" buttons. Below this is a description field with the text "Click here to add a description." and a "Routing Profile" tab. A table lists the routing profile configuration:

Priority	URI Group	Time of Day	Load Balancing	Next Hop Address	Transport	
1	*	default	Priority	10.33.10.33	TLS	Edit Delete

### 7.3.5. Topology Hiding

Topology Hiding is an Avaya SBCE security feature which allows changing certain key SIP message parameters to ‘hide’ or ‘mask’ how the enterprise network may appear to an unauthorized or malicious user.

To create a Topology Hiding profile, select **Global Profiles → Topology Hiding**. Click on the **Add** button.

In the compliance testing, two Topology Hiding profiles **EN-to-SP** and **SP-to-EN** were created.

#### Topology Hiding Profile for SP

Profile **EN-to-SP** was defined to mask the enterprise SIP domain avayalab.com in the “Request-Line”, “From” and “To” headers to SP provided full qualified domain name. This is done to secure the enterprise network topology and to meet the SIP requirement of the service provider.

**Notes:**

- The **Criteria** should be selected as **IP/Domain** to give the Avaya SBCE the capability to mask both domain name and IP address present in URI-Host.
- The masking applied on “From” header.
- The masking applied on “To” header.

The screenshots below illustrate the Topology Hiding profile **EN-to-SP**.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The left sidebar contains a navigation menu with options like Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, Domain DoS, Server Interworking, Media Forking, Routing, Server Configuration, Topology Hiding (highlighted), Signaling Manipulation, URI Groups, SNMP Traps, Time of Day Rules, FGDN Groups, and Reverse Proxy Policy. The main content area is titled 'Topology Hiding Profiles: EN-to-SP' and includes an 'Add' button, a list of profiles (default, cisco\_th\_profile, EN-to-SP, SP-to-EN), and buttons for Rename, Clone, and Delete. A table titled 'Topology Hiding' shows the configuration for the EN-to-SP profile, with columns for Header, Criteria, Replace Action, and Overwrite Value. The table lists headers: Via, To, SDP, Record-Route, Refer-To, Request-Line, Referred-By, and From, all with criteria of IP/Domain. The Replace Action for Via, SDP, Record-Route, Refer-To, and Referred-By is Auto, while for To, Request-Line, and From, it is Overwrite. The Overwrite Value for To, Request-Line, and From is 192.168.21.188. An 'Edit' button is located at the bottom right of the table.

Header	Criteria	Replace Action	Overwrite Value
Via	IP/Domain	Auto	---
To	IP/Domain	Overwrite	192.168.21.188
SDP	IP/Domain	Auto	---
Record-Route	IP/Domain	Auto	---
Refer-To	IP/Domain	Auto	---
Request-Line	IP/Domain	Overwrite	192.168.21.188
Referred-By	IP/Domain	Auto	---
From	IP/Domain	Overwrite	192.168.21.188



## Topology Hiding Profile for EN

Profile **SP-to-EN** was also created to mask SP URI-Host in “Request-Line”, “From” and “To”, headers to the enterprise domain **avayalab.com**, replace Record-Route, Via headers and SDP added by SP to internal IP address known to EN.

### Notes:

- The **Criteria** should be **IP/Domain** to give the Avaya SBCE the capability to mask both domain name and IP address present in URI-Host.
- The masking applied on “From” header.
- The masking applied on “To” header.

The screenshots below illustrate the Topology Hiding profile **SP-to-EN**.

The screenshot displays the Avaya Session Border Controller for Enterprise (SBCE) web interface. The left sidebar contains a navigation menu with options: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles (expanded), Domain DoS, Server Interworking, Media Forking, Routing, Server Configuration, **Topology Hiding** (highlighted), Signaling Manipulation, URI Groups, SNMP Traps, Time of Day Rules, FGDN Groups, and Reverse Proxy Policy. The main content area is titled 'Topology Hiding Profiles: SP-to-EN' and includes an 'Add' button, 'Rename', 'Clone', and 'Delete' buttons. Below this is a list of profiles: 'default', 'cisco\_th\_profile', 'EN-to-SP', and 'SP-to-EN' (highlighted). The 'SP-to-EN' profile is selected, showing a table of configuration rules. The table has columns: Header, Criteria, Replace Action, and Overwrite Value. The rules are as follows:

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

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

## 7.4. Domain Policies

Domain Policies configures various rule sets (policies) to control unified communications based upon criteria of communication sessions originating from or terminating at the enterprise. These criteria can be used to trigger policies which, in turn, activate various security features of the Avaya SBCE security device to aggregate, monitor, control and normalize call flow. There are default policies available for use, or a custom domain policy can be created.

### 7.4.1. Media Rules

Media rules can be used 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. You can also define how Avaya SBCE must handle media packets that adhere to the set parameters.

To clone a Media Rule, navigate to **Domain Policies** → **Media Rules**. With *default-low-med* rule chosen, click on the **Clone** button.

### Media Rules for EN

In this compliance testing, Secure Real-Time Transport Protocol (SRTP, media encryption) is used within enterprise network only. Therefore, it is necessary to create a media rule to apply to the internal interface of Avaya SBCE and EN. Created **sRTP-MR** rule is shown below.

## Session Border Controller for Enterprise

AVAYA

Dashboard

Administration

Backup/Restore

System Management

Global Parameters

Global Profiles

PPM Services

Domain Policies

- Application Rules
- Border Rules
- Media Rules
- Security Rules
- Signaling Rules
- End Point Policy Groups
- Session Policies

TLS Management

Device Specific Settings

Media Rules: sRTP-MR

Add

Filter By Device...

Rename

Clone

Delete

Media Rules

sRTP-MR

SMVM\_RW\_...

Click here to add a description.

Encryption

Codec Prioritization

Advanced

QoS

Audio Encryption

Preferred FormatsSRTP\_AES\_CM\_128\_HMAC\_SHA1\_80  
SRTP\_AES\_CM\_128\_HMAC\_SHA1\_32  
RTP

Encrypted RTCP☒

MKI☐

LifetimeAny

Interworking☒

Video Encryption

Preferred FormatsRTP

Interworking☒

Miscellaneous

Capability Negotiation☐

Edit

QT; Reviewed:  
SPOC 1/26/2017

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58 of 73  
PICMSM701SBCE71

## 7.4.2. Signaling Rules

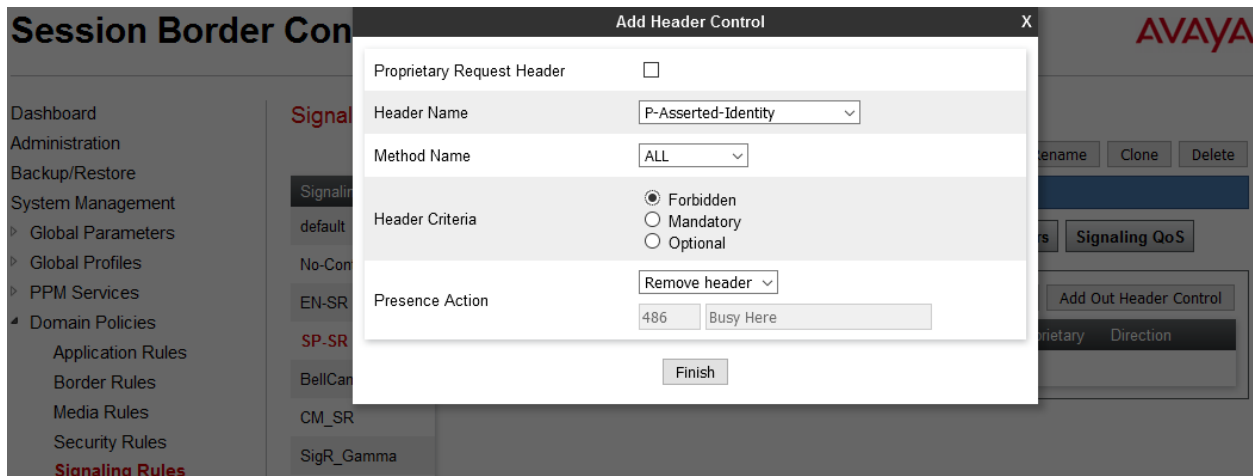
Signaling Rules define the action to be taken (Allow, Block, Block with Response, etc.) for each type of SIP-specific signaling request and response message. When SIP signaling packets are received by the Avaya SBCE, they are parsed and “pattern-matched” against the particular signaling criteria defined by these rules. Packets matching the criteria defined by the Signaling Rules are tagged for further policy matching.

To clone a Signaling Rule, navigate to **Domain Policies → Signaling Rules**. With the **default** rule chosen, click on the **Clone** button.

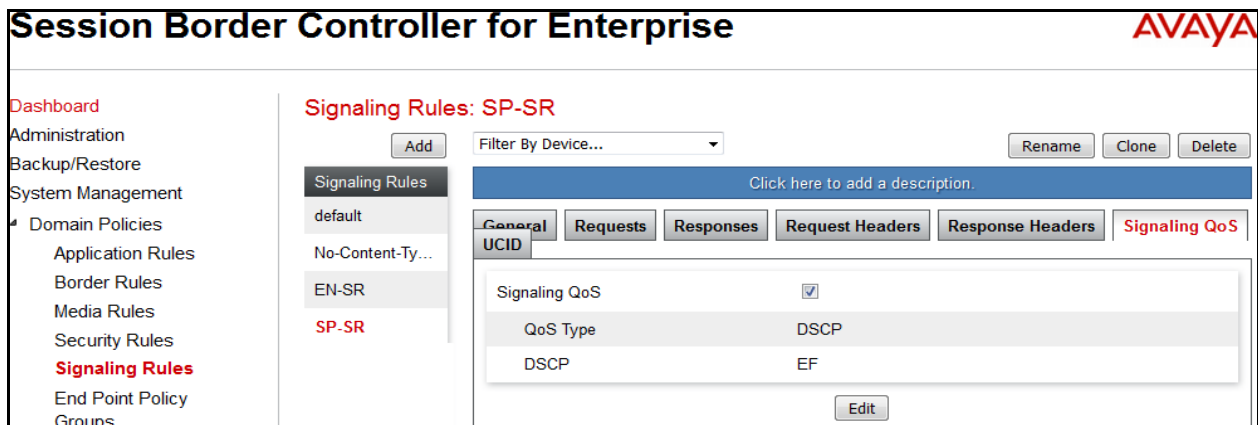
### Signaling Rules for SP

In the compliance testing, created signaling rule **SP-SR** is discussed below. All the tabs are kept as default values except the **Signaling QoS** tab.

In **Request Headers** tab, click **Add Out Header Control** button then enter information as shown in capture below.



In the **Signaling QoS** tab, click on **Edit** button then check on checkbox. Then select **EF** value for **DSCP** option.



## Signaling Rules for EN

In the compliance testing, created signaling rule **EN-SR** is discussed below. All the tabs are kept as default values except **Signaling QoS** tab.

In **Signaling QoS** tab, click on **Edit** button then check on checkbox. Then select **EF** value for **DSCP** option.

The screenshot shows the 'Session Border Controller for Enterprise' web interface. The left sidebar contains a navigation menu with 'Domain Policies' expanded, showing 'Signaling Rules' as the selected option. The main content area is titled 'Signaling Rules: EN-SR'. It features an 'Add' button, a 'Filter By Device...' dropdown, and 'Rename', 'Clone', and 'Delete' buttons. Below these is a 'Click here to add a description.' link. The 'General' tab is active, showing a table with columns 'UCID' and 'Signaling QoS'. The 'Signaling QoS' checkbox is checked. The 'QoS Type' is set to 'DSCP' and the 'DSCP' value is 'EF'. An 'Edit' button is at the bottom right of the table.

UCID	Signaling QoS
	<input checked="" type="checkbox"/>
	QoS Type: DSCP
	DSCP: EF

### 7.4.3. Endpoint Policy Groups

The rules created within the **Domain Policies** section are assigned to an **Endpoint Policy Group**. The **Endpoint Policy Group** is then applied to a **Server Flow** defined in the next section. Endpoint Policy Groups were created for SP and EN. To create a new policy group, navigate to **Domain Policies** → **Endpoint Policy Groups** and click on **Add**.

### Endpoint Policy Group for SP

The following screen shows **SP-PG** created for SP:

- Set Application Rule to *default-trunk*.
- Set Border Rule to *default*.
- Set Media Rule to *default-low-med* as created in **Section 7.4.1**.
- Set Security Rule to *default-high*
- Set Signaling Rule to *SP-SR* as created in **Section 7.4.2**.

The screenshot shows the 'Session Border Controller for Enterprise' web interface. The left sidebar contains a navigation menu with 'Domain Policies' expanded, showing 'Endpoint Policy Groups' as the selected option. The main content area is titled 'Policy Groups: SP-PG'. It features an 'Add' button, a 'Filter By Device...' dropdown, and 'Rename', 'Clone', and 'Delete' buttons. Below these is a 'Click here to add a description.' link. The 'Policy Group' tab is active, showing a table with columns 'Order', 'Application', 'Border', 'Media', 'Security', 'Signaling', and 'Summary'. The table contains one row with the following values: Order: 1, Application: default-trunk, Border: default, Media: default-low-med, Security: default-high, Signaling: SP-SR. An 'Edit' button is at the bottom right of the table.

Order	Application	Border	Media	Security	Signaling	Summary
1	default-trunk	default	default-low-med	default-high	SP-SR	Edit

## Endpoint Policy Group for EN

The following screen shows **EN-PG** created for EN:

- Set Application Rule to *default-trunk*.
- Set Border Rule to *default*.
- Set Media Rule to *sRTP-MR* as created in **Section 7.4.1**.
- Set Security Rule to *default-high*.
- Set Signaling Rule to *EN-SR* as created in **Section 7.4.2**.

### Session Border Controller for Enterprise

Dashboard

Administration

Backup/Restore

Domain Policies

- Application Rules
- Border Rules
- Media Rules
- Security Rules
- Signaling Rules
- End Point Policy Groups**

#### Policy Groups: EN-PG

Add

Filter By Device...

Rename

Clone

Delete

Policy Groups

EN-PG

SP-PG

Click here to add a description.

Hover over a row to see its description.

Policy Group

Summary

Order	Application	Border	Media	Security	Signaling	
1	default-trunk	default	sRTP-MR	default-high	EN-SR	Edit

## 7.5. Device Specific Settings

Device Specific Settings allows aggregate system information to be viewed and various device-specific parameters to be managed to determine how a particular device will function when deployed in the network. Specifically, it gives the ability to define and administer various device-specific protection features such as Message Sequence Analysis (MSA) functionality and protocol scrubber rules, end-point and session call flows, as well as the ability to manage system logs and control security features.

### 7.5.1. Network Management

The Network Management screen is where the network interface settings are configured and enabled. During the installation process of the Avaya SBCE, certain network-specific information was defined such as; device IP address(es), public IP address(es), netmask, gateway, etc. to interface the device to the network. This information populates the **Network Management** tab, which can be edited as needed to optimize device performance and network efficiency.

Enable the interfaces used to connect to the inside and outside networks on the **Interface** tab. The following screen shows Interface Names, **A1** and **B1** are **Enabled**. To enable an interface, click on its **Status** corresponding to the interface names.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The title bar at the top reads "Session Border Controller for Enterprise" with the AVAYA logo on the right. A left-hand navigation menu lists various system management options, with "Network Management" highlighted in red. The main content area is titled "Network Management: SBCE70" and contains two tabs: "Interfaces" (active) and "Networks". Under the "Interfaces" tab, there is a table with three columns: "Interface Name", "VLAN Tag", and "Status". The table lists four interfaces: A1 (Enabled), A2 (Disabled), B1 (Enabled), and B2 (Disabled). An "Add VLAN" button is located in the top right corner of the table area.

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

Navigate to **Device Specific Settings** → **Network** and under the **Network Configuration** tab verify the IP addresses assigned to the interfaces. The following screens show the private interface is assigned to **A1** and the public interface is assigned to **B1** respectively.

The screenshot shows the 'Session Border Controller for Enterprise' web interface. The left sidebar contains a navigation menu with 'Device Specific Settings' expanded and 'Network Management' selected. The main content area is titled 'Edit Network' and displays configuration for 'Network\_A1'. A warning message at the top states: 'This Network contains one or more IP Address entries which are in use. If the Interface, an IP Address, or Public IP which is in use is modified, the application must be restarted or the device may stop functioning.' The configuration fields are: Name (Network\_A1), Default Gateway (10.10.98.1), Subnet Mask (255.255.255.192), and Interface (A1). Below these fields is an 'Add' button. A table at the bottom shows the IP configuration:

IP Address	Public IP	Gateway Override
10.10.98.22	Use IP Address	Use Default

Buttons for 'Delete' and 'Finish' are also present.

The screenshot shows the 'Session Border Controller for Enterprise' web interface. The left sidebar contains a navigation menu with 'Device Specific Settings' expanded and 'Network Management' selected. The main content area is titled 'Edit Network' and displays configuration for 'Network\_B1'. A warning message at the top states: 'This Network contains one or more IP Address entries which are in use. If the Interface, an IP Address, or Public IP which is in use is modified, the application must be restarted or the device may stop functioning.' The configuration fields are: Name (Network\_B1), Default Gateway (10.10.98.97), Subnet Mask (255.255.255.224), and Interface (B1). Below these fields is an 'Add' button. A table at the bottom shows the IP configuration:

IP Address	Public IP	Gateway Override
10.10.98.119	Use IP Address	Use Default

Buttons for 'Delete' and 'Finish' are also present.

## 7.5.2. Media Interface

The Media Interface screen is where the media ports are defined. The Avaya SBCE will open a connection for RTP on the defined ports.

To create a new Media Interface, navigate to **Device Specific Settings → Media Interface** and click **Add**.

Separate Media Interfaces were created for both inside and outside interfaces. The following screen shows the Media Interfaces created in the compliance testing.

**Note:** After the media interfaces are created, an application restart is necessary before the changes will take effect.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The left sidebar contains a navigation menu with options: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, PPM Services, Domain Policies, TLS Management, Device Specific Settings (selected), and Network Management. Under Device Specific Settings, the 'Media Interface' option is highlighted in red. The main content area is titled 'Media Interface: SBCE70'. Below this, there is a 'Media Interface' tab and a warning message: 'Modifying or deleting an existing media interface will require an application restart before taking effect. Application restarts can be issued from System Management.' Below the warning is a table with two columns: 'Name' and 'Media IP Network'. The table lists two interfaces: 'InsideMedia' and 'OutsideMedia'. The 'Port Range' column shows '35000 - 40000' for both. Each interface has 'Edit' and 'Delete' links. An 'Add' button is located at the top right of the table.

Name	Media IP Network	Port Range	Edit	Delete
InsideMedia	10.10.98.22 Network_A1 (A1, VLAN 0)	35000 - 40000	Edit	Delete
OutsideMedia	10.10.98.119 Network_B1 (B1, VLAN 0)	35000 - 40000	Edit	Delete

## 7.5.3. Signaling Interface

The Signaling Interface screen is where the SIP signaling port is defined. The Avaya SBCE will listen for SIP requests on the defined port.

To create a new Signaling Interface, navigate to **Device Specific → Settings → Signaling Interface** and click **Add**.

Separate Signaling Interfaces were created for both inside and outside interfaces.



## Signaling Interface for SP

The outside interface to service provider is created with UDP/5083 as shown below.

The screenshot shows the 'Edit Signaling Interface' dialog box with the following configuration:

Field	Value
Name	OutsideSignalingUDP
IP Address	Network_B1 (B1, VLAN 0) 10.10.98.119
TCP Port	Leave blank to disable
UDP Port	5083
TLS Port	Leave blank to disable
TLS Profile	None
Enable Shared Control	<input type="checkbox"/>
Shared Control Port	

Buttons: Finish

## Signaling Interface for EN

The inside to service provider interface is created with TLS/5061 as shown below.

- Enter descriptive name for **Name** field.
- Select **IP Address** from pull down menu defined as internal network interface **Section 7.5.1**.
- Specified **5061** for **TLS Port**. Then select **TLS profile** from pull down menu as defined in **Section 7.2.3**.
- Click **Finish**.

The screenshot shows the 'Edit Signaling Interface' dialog box with the following configuration:

Field	Value
Name	InsideSignalingTLS
IP Address	Network_A1 (A1, VLAN 0) 10.10.98.22
TCP Port	Leave blank to disable
UDP Port	Leave blank to disable
TLS Port	5061
TLS Profile	AvayaSBCServer-Q
Enable Shared Control	<input type="checkbox"/>
Shared Control Port	

Buttons: Finish

#### 7.5.4. End Point Flows - Server Flow

When a packet is received by the 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 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 screens illustrate the flow through the Avaya SBCE to secure a SIP Trunk call.

In the compliance testing, separate Server Flows were created for SP and EN. To create a Server Flow, navigate to **Device Specific Settings → End Point Flows**. Select the **Server Flows** tab and click **Add** (not shown). In the new window that appears, enter the following values. The other fields are kept default.

- **Flow Name:** Enter a descriptive name.
- **Server Configuration:** Select a Server Configuration created in **Section 7.3.4** to assign to the Flow.
- **URI Group:** Select the URI Group created in **Section 7.3.1** to assign to the Flow.  
**Note:** URI Group can be set to “\*” to match all calls.
- **Received Interface:** Select the Signaling Interface created in **Section 7.5.3** that the Server Configuration is allowed to receive SIP messages from.
- **Signaling Interface:** Select the Signaling Interface created in **Section 7.5.3** used to communicate with the Server Configuration.
- **Media Interface:** Select the Media Interface created in **Section 7.5.2** used to communicate with the Server Configuration.
- **End Point Policy Group:** Select the End Point Policy Group created in **Section 7.4.3** to assign to the Server Configuration.
- **Routing Profile:** Select the Routing Profile created in **Section 7.3.2** that the Server Configuration will use to route SIP messages to.
- **Topology Hiding Profile:** Select the Topology-Hiding profile created in **Section 7.3.5** to apply to the Server Configuration.
- Click **Finish**.

The following screen shows the Server Flow **SP-SF** configured for SP.

## Session Border Controller for Enterprise

Dashboard

Administration

Backup/Restore

System Management

▸ Global Parameters

▸ Global Profiles

▸ PPM Services

▸ Domain Policies

▸ TLS Management

▸ Device Specific Settings

Network Management

Media Interface

Signaling Interface

**End Point Flows**

Session Flows

▸ DMZ Services

TURN/STUN Service

SNMP

Syslog Management

Advanced Options

▸ Troubleshooting

End Point Flows: SBCE71

Devices

Subscriber Flows

Server Flows

SBCE71

Edit Flow: SP-SF

X

Flow Name

SP-SF

Server Configuration

SP-SC

URI Group

\*

Transport

\*

Remote Subnet

\*

Received Interface

InsideSignalingTLS

Signaling Interface

OutsideSignalingUDP

Media Interface

OutsideMedia

Secondary Media Interface

None

End Point Policy Group

SP-PG

Routing Profile

SP-to-EN

Topology Hiding Profile

EN-to-SP

Signaling Manipulation Script

None

Remote Branch Office

Any

Finish

Similarly, the following screen shows the Server Flow **EN-SF** configured for EN.

## Session Border Controller for Enterprise

Dashboard

Administration

Backup/Restore

System Management

▸ Global Parameters

▸ Global Profiles

▸ PPM Services

▸ Domain Policies

▸ TLS Management

▸ Device Specific Settings

Network Management

Media Interface

Signaling Interface

**End Point Flows**

Session Flows

▸ DMZ Services

TURN/STUN Service

SNMP

Syslog Management

Advanced Options

▸ Troubleshooting

End Point Flows: SBCE71

Devices

Subscriber Flows

Server Flows

SBCE71

Edit Flow: EN-SF X

Flow Name

EN-SF

Server Configuration

EN-SC ▾

URI Group

\* ▾

Transport

\* ▾

Remote Subnet

\* ▾

Received Interface

OutsideSignalingUDP ▾

Signaling Interface

InsideSignalingTLS ▾

Media Interface

InsideMedia ▾

Secondary Media Interface

None ▾

End Point Policy Group

EN-PG ▾

Routing Profile

EN-to-SP ▾

Topology Hiding Profile

SP-to-EN ▾

Signaling Manipulation Script

None ▾

Remote Branch Office

Any ▾

Finish

## 8. Pure IP Service Configuration

Pure IP is responsible for the configuration of its SIP Trunking Service. The customer will need to provide the IP address used to reach the Avaya SBCE at the enterprise. Pure IP will provide the customer with the necessary information to configure the SIP connection from the enterprise to Pure IP. The information provided by Pure IP includes:

- IP address and port number used for signaling through security devices (if any).
- IP address and port number used for media through security devices (if any).
- Pure IP SIP domain. In the compliance testing, Pure IP preferred to use IP address as an URI-Host.
- CPE SIP domain. In the compliance testing, Pure IP preferred to use IP address of the Avaya SBCE as an URI-Host.
- Supported codecs.
- DID numbers.

The sample configuration between Pure IP and the enterprise for the compliance testing is a static configuration. There is no registration on the SIP trunk implemented on either Pure IP or enterprise side.

## 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 useful troubleshooting commands.

### 9.1. Verification Steps

- Verify that endpoints at the enterprise site can place call to PSTN and that the call remains active for more than 35 seconds. This time period is included to satisfy SIP protocol timers.
- Verify that endpoints at the enterprise site can receive call from PSTN and that the call can remain active for more than 35 seconds. This time period is included satisfy SIP protocol timers.
- Verify that the user on 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. Protocol Traces

The following SIP headers are inspected using Wireshark trace analysis:

- Request-URI: verify the called party number and SIP domain.
- From: verify the calling party name and number.
- To: verify the called party name and number.
- P-Asserted-Identity: verify the calling party name and number.
- Privacy: verify the value “user” and/or “id” presents the private call scenario.

The following attributes in SIP message body are inspected using Wireshark trace analysis:

- Connection Information (c line): verify IP address of near end and far end endpoints.
- Time Description (t line): verify session timeout value of near end and far end endpoints.
- Media Description (m line): verify audio port, codec, DTMF event description.
- Media Attribute (a line): verify specific audio port, codec, ptime, send/ receive ability, DTMF event and fax attributes.

## 9.3. Troubleshooting:

### 9.3.1. The Avaya SBCE

Use Avaya SBCE trace tool, traceSBC to monitor the SIP signaling messages between Pure IP and the Avaya SBCE.

### 9.3.2. Communication Manager

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

## 10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager 7.0, Avaya Aura® Session Manager 7.0 and Avaya Session Border Controller for Enterprise 7.1 to Pure IP SIP Trunking Service. Pure IP SIP Trunking Service is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. Pure IP provides a flexible, cost-saving alternative to traditional analog and ISDN-PRI trunks.

All of the test cases were executed. Despite the observation seen during testing as noted in **Section 2.2**, the test results met the objectives outlined in **Section 2.1**. The Pure IP SIP Trunking Service is considered **compliant** with Avaya Aura® Communication Manager 7.0, Avaya Aura® Session Manager 7.0 and Avaya Session Border Controller for Enterprise 7.1.

## 11.References

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

- [1] *What's New in Avaya Aura Release 7.0*, Release 7.0, 03-601818, Issue 1, August 2015.
- [2] *Deploying Avaya Aura® System Manager*, Release 7.0, Issue 1, October 2015.
- [3] *Administering Avaya Aura® System Manager for Release 7.0*, Issue 1, August 2015.
- [4] *Administering Avaya Aura® Session Manager*, Release 7.0, Issue 1, August 2015.
- [5] *Deploying Avaya Aura Communication Manager in Virtualized Environment*, Release 7.0, Issue 1, August 2015.
- [6] *Avaya Session Border Controller for Enterprise Overview and Specification*, Release 7.0, Issue 1, August 2015.
- [7] *Deploying Avaya Session Border Controller for Enterprise*, Release 7.0, Issue 1, August 2015.
- [8] *Deploying Avaya Session Border Controller in Virtualized Environment*, Release 7.0, Issue 1, August 2015.
- [9] *Administering Avaya Session Border Controller for Enterprise*, Release 7.0, Issue 1, August 2015.
- [10] *Deploying and Updating Avaya Aura Media Server Appliance*, Release 7.7, Issue 1, August 2015.
- [11] *9600 Series IP Deskphones Overview and Specification*, Release 7.0, Issue 1, August 2015.
- [12] *Installing and Maintaining Avaya 9601/9608/9611G/9621G/9641G/9641GS IP Deskphones SIP*, Release 7.0, Issue 1, August 2015.
- [13] *Administering Avaya 9601/9608/9611G/9621G/9641G/9641GS IP Deskphones SIP*, Release 7.0, Issue 2, August 2015.
- [14] *Administering Avaya one-X® Communicator*, Release 6.2, April 2015.
- [15] *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*
- [16] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>
- [17] *RFC 3515, The Session Initiation Protocol (SIP) Refer Method*, <http://www.ietf.org/>
- [18] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, <http://www.ietf.org/>

Product documentation for Pure IP Networks' SIP Trunking Solution is available from Pure IP.



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