



# **Application Notes for Configuring Avaya Aura® Communication Manager 8.1, Avaya Aura® Session Manager 8.1 and Avaya Session Border Controller for Enterprise 8.0 to support Clearcom SIP Trunking Service – Issue 1.0**

## **Abstract**

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

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

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

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

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

## Table of Contents

1.	Introduction.....	4
2.	General Test Approach and Test Results.....	4
2.1.	Interoperability Compliance Testing.....	5
2.2.	Test Results .....	6
2.3.	Support .....	8
3.	Reference Configuration.....	9
4.	Equipment and Software Validated .....	12
5.	Configure Avaya Aura® Communication Manager .....	13
5.1.	Licensing and Capacity .....	13
5.2.	System Features.....	14
5.3.	IP Node Names.....	16
5.4.	Codecs .....	17
5.5.	IP Network Regions .....	19
5.6.	Signaling Group .....	20
5.7.	Trunk Group .....	22
5.8.	Calling Party Information.....	26
5.9.	Inbound Routing.....	27
5.10.	Outbound Routing .....	28
6.	Configure Avaya Aura® Session Manager .....	32
6.1.	System Manager Login and Navigation.....	33
6.2.	SIP Domain .....	35
6.3.	Locations .....	35
6.4.	Adaptations.....	39
6.5.	SIP Entities.....	41
6.6.	Entity Links .....	44
6.7.	Routing Policies .....	46
6.8.	Dial Patterns .....	47
7.	Configure Avaya Session Border Controller for Enterprise .....	50
7.1.	System Access.....	50
7.2.	Device Management.....	52
7.3.	TLS Management.....	54
7.4.	Network Management .....	54
7.5.	Media Interfaces.....	55
7.6.	Signaling Interfaces.....	57
7.7.	Server Interworking.....	59
7.7.1.	Server Interworking Profile – Enterprise .....	59
7.7.2.	Server Interworking Profile – Service Provider.....	61
7.8.	Signaling Manipulation.....	63
7.9.	Server Configuration .....	65
7.9.1.	Server Configuration Profile – Enterprise .....	65
7.9.2.	Server Configuration Profile – Service Provider .....	67
7.10.	Routing .....	70
7.10.1.	Routing Profile – Enterprise.....	70
7.10.2.	Routing Profile – Service Provider .....	72

7.11.	Topology Hiding.....	73
7.11.1.	Topology Hiding Profile – Enterprise .....	73
7.11.2.	Topology Hiding Profile – Service Provider.....	75
7.12.	Domain Policies.....	76
7.12.1.	Application Rules .....	76
7.12.2.	Media Rules.....	77
7.12.3.	Signaling Rules .....	79
7.13.	End Point Policy Groups .....	80
7.13.1.	End Point Policy Group – Enterprise .....	80
7.13.2.	End Point Policy Group – Service Provider.....	81
7.14.	End Point Flows.....	82
7.14.1.	End Point Flow – Enterprise .....	83
7.14.2.	End Point Flow – Service Provider .....	84
8.	Clearcom SIP Trunking Service Configuration.....	85
9.	Verification and Troubleshooting .....	85
9.1.	General Verification Steps .....	85
9.2.	Communication Manager Verification.....	85
9.3.	Session Manager Verification .....	86
9.4.	Avaya SBCE Verification .....	89
10.	Conclusion .....	94
11.	References.....	94

# 1. Introduction

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

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

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

## 2. General Test Approach and Test Results

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

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member’s solution.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products only (private network side). Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

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

For the testing associated with this Application Note, the interface between Avaya systems and the Clearcom SIP Trunking service did not include the use of any specific encryption features. Encryption (TLS/SRTP) was used internal to the enterprise between Avaya products wherever possible.

## 2.1. Interoperability Compliance Testing

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

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

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

The following items were not tested:

- Inbound toll-free calls, outbound Toll-Free calls, 911 calls (emergency), “0” calls (Operator), 0+10 digits calls (Operator Assisted) and local directory assistance calls were not tested.
- The SIP REFER method for call redirection was not tested for reasons noted in **Section 2.2**

## 2.2. Test Results

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

- **Music played to the user on incoming calls from the PSTN to H.323 endpoints:** On calls from the PSTN to enterprise H.323 endpoints music was played to the PSTN user when the call was answered at the enterprise H.323 endpoint. It was observed that the music originated from the PSTN and not from the enterprise, when **Direct IP-IP Audio Connections** was set to “y”, enabling media shuffling on the SIP trunk. Setting this field to “y” allows Communication Manager to redirect media traffic directly between the Avaya SBCE and the H.323 enterprise endpoint resulting in the release of Avaya Media Gateway or Media Server resources at the enterprise. Normally this field is set to “y” in order to free up Media resources at the enterprise, during the compliance test this field was set to “n” as a work around, refer to **Section 5.6**. This issue was reported to Clearcom.
- **SIP REFER method:** PSTN calls that were transferred back to the network using the SIP REFER method did not work properly. Attended call transfers dropped. On blind transfers, the REFER message was accepted by Clearcom with a “202” message, but the trunks were not released after the call transfer was completed. For these reasons testing was done with REFER disabled in Communication Manager (**Network Call Redirection** set to “n” under the **trunk-group**, refer to **Section 5.7**). With REFER disabled, blind and attended call transfers to the PSTN completed successfully, with the caveat that Communication Manager trunk channels were not released from the call path after the call was transferred, two trunks channels remained busy/connected for the entire duration of the call.
- **Fax support:** Fax calls using the T.38 protocol failed during the compliance test. G.711 pass-through fax was also tested, but it behaved unreliably. The issue related to G.711 pass-through fax failing during the compliance test may be related to the unpredictability of G.711 pass-through techniques, which only works well on networks with very few hops and with limited end-to-end delay. The issue related to T.38 fax calls failing is related to the PSTN carriers used by Clearcom in Mexico to route calls to the PSTN, not all PSTN carriers used by Clearcom in Mexico support T.38. This issue could be resolved by Clearcom selecting specific PSTN carriers that do support T.38 and routing T.38 fax traffic via these PSTN carriers.
- **Outbound Calling Party Number (CPN) Blocking:** To support user privacy on outbound calls (calling party number blocking), when enabled by the user, Communication Manager sends “anonymous” as the calling number in the SIP “From” header and includes “Privacy: id” in the INVITE message, while the actual number of the

caller is sent in the “P-Asserted-Identity” header. On the called PSTN phone, the calling party number was not blocked, the main DID number (pilot number) assigned to the trunk was displayed, instead of “anonymous”.

- **Caller ID display on Outbound Calls, Call Forwards and Call transfers to the local PSTN in Mexico:** For outbound calls, calls from the local PSTN in Mexico to Communication Manager that were Forwarded or calls that were transferred back out to the local PSTN in Mexico, the caller ID number displayed at the SIP softphone (local PSTN in Mexico) was always of the main DID number (pilot number) assigned to the trunk, regardless of the PSTN number being used to originate the call.
- **Caller ID display on EC500 extension to cellular:** For EC500 extension to cellular calls the Caller ID display at the Mobile/cellular station was always of the main DID number (pilot number) assigned to the trunk, regardless of the PSTN number being used to originate the call.
- **Outbound call from an enterprise extension to a busy PSTN number:** Clearcom did not send a “486 Busy Here” message on an outbound call to a PSTN number that was busy, as it was expected on this condition. There was no direct impact to the user, who heard busy tone.
- **From Header Manipulation:** Clearcom uses SIP trunk registration and digest authentication in order to accept calls from the enterprise into their network. Additionally, Clearcom requires the username associated with the SIP trunk credentials to be present in the “From” header of all outbound calls from the enterprise. Otherwise, the call is rejected with a “403 Username=From not allowed” message. A Signaling Script was created in the Avaya SBCE to include the SIP trunk credential’s username in the “From” header of all outbound calls. (refer to **Section 7.8**).
- **Request-URI Header Manipulation:** Clearcom sends the username associated with the SIP trunk credentials in the “Request URI” header of all inbound calls, while the actual DID number of the party dialed is sent in the “To” header. Since the routing decision in Session Manager is based on Dial Patterns, by inspecting the number present in the “Request URI” header of the incoming call, a Signaling Script was created in the Avaya SBCE to populate the “Request URI” header with the number present in the “To” header of inbound calls, refer to **Section 7.8**.
- **SIP OPTION Messages:** During the compliance test Clearcom did not send SIP OPTION messages to Avaya, Session Manager did send SIP OPTION messages to Clearcom, this was sufficient to keep the SIP trunk up in-service.
- **SIP header optimization:** There are multiple SIP headers and parameters used by Communication Manager and Session Manager, some of them Avaya proprietary, that had no significance in the service provider’s network. These headers were removed with the purpose of blocking enterprise information from being propagated outside of the enterprise boundaries, to reduce the size of the packets entering the service provider’s network and to improve the solution interoperability in general. The following headers were removed from outbound messages using an Adaptation in Session Manager: AV-Global-Session-ID, AV-Correlation-ID, Alert-Info, Endpoint-View, P-AV-Message-id, P-Charging-Vector and P-Location (**Section 6.4**). Additionally, the parameters “gsid” and “epv” were removed from outbound Contact headers using a Signaling Script in the Avaya SBCE, refer to **Section 7.8**.

## **2.3. Support**

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

<http://www.clearcom.mx/>

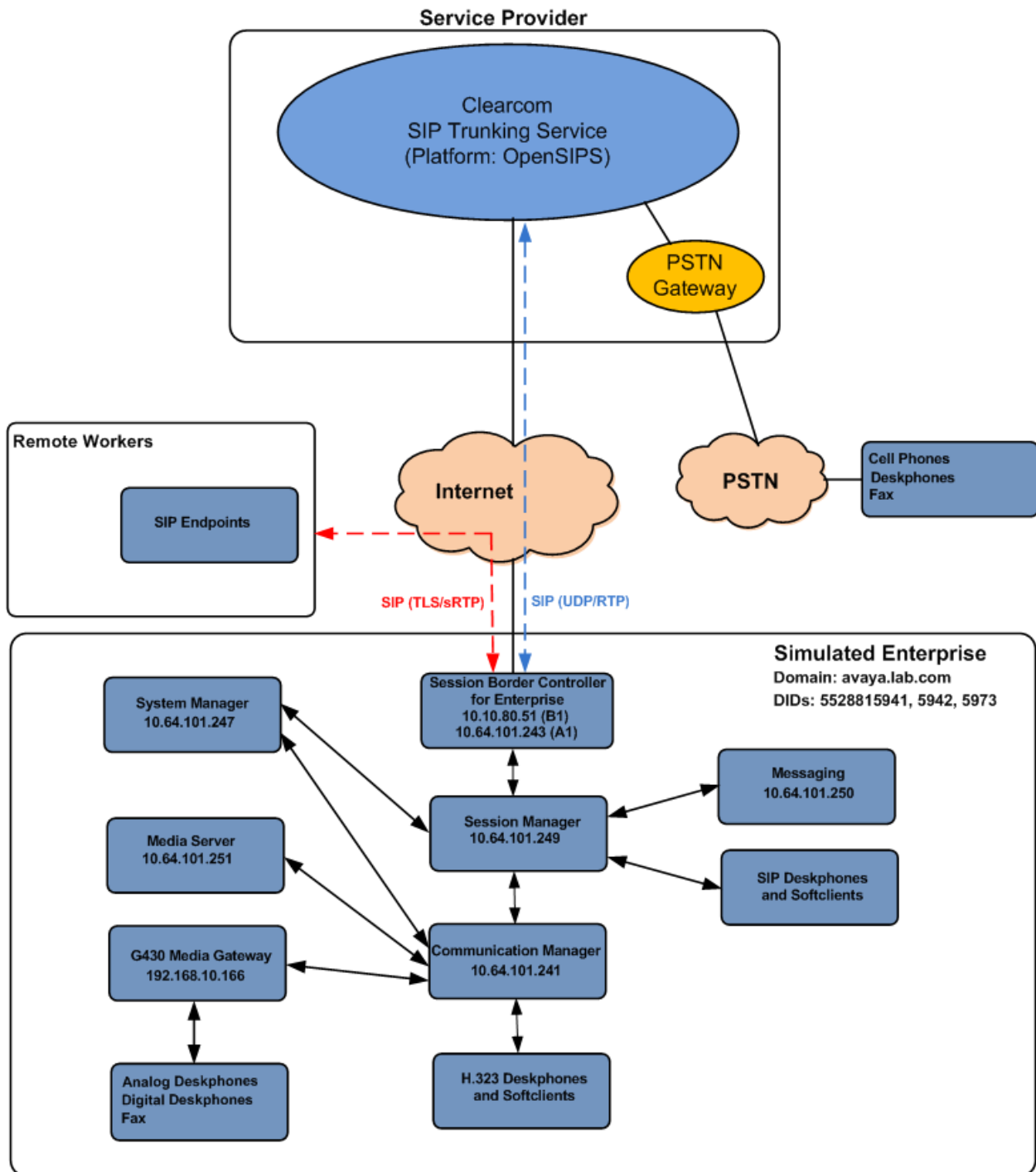
For technical support on the Avaya products described in these Application Notes visit

<http://support.avaya.com>



### 3. Reference Configuration

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



**Figure 1: Avaya SIP Enterprise Solution connected to Clearcom SIP Trunking Service**

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

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

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

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

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

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

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

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

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

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

## 4. Equipment and Software Validated

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

Equipment/Software	Release/Version
<b>Avaya</b>	
Avaya Aura® Communication Manager	8.1.1.0 (01.0.890.0-25763)
Avaya Aura® Session Manager	8.1.1.0 (8.1.1.0.811021)
Avaya Aura® System Manager	8.1.1.0 Build No. 8.1.0.0.733078 Software Update Rev. No. 8.1.1.0.0310504
Avaya Session Border Controller for Enterprise	ASBCE 8.0 8.0.1.0-10-17555
Avaya Aura® Messaging	7.1 Service Pack 2 (MSG-01.0.532.0-002_0204)
Avaya Aura® Media Server	8.0.2.61_2019.09.16
Avaya G430 Media Gateway	g430_sw_41_16_0
Avaya 96x1 Series IP Deskphones (SIP)	Version 7.1.7.0
Avaya 96x1 Series IP Deskphones (H.323)	Version 6.8202
Avaya J179 IP Deskphones (H.323)	Version 6.8202
Avaya one-X® Communicator (H.323, SIP)	6.2.14.1-SP14
Avaya Equinox for Windows (SIP)	3.6.4.31.2
Avaya 2420 Series Digital Deskphones	N/A
Avaya 6210 Analog Deskphones	N/A
<b>Clearcom</b>	
OpenSIPS Softswitch	1.9
OpenSIPS Session Border Controller	1.9

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

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

## 5. Configure Avaya Aura® Communication Manager

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

The Communication Manager configuration was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation. Some screens capture will show the use of the **change** command instead of the **add** command, since the configuration used for the testing was previously added.

### 5.1. Licensing and Capacity

Use the **display system-parameters customer-options** command to verify that the **Maximum Administered SIP Trunks** value on **Page 2** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise including any trunks to and from the service provider. The example shows that **40000** licenses are available and **120** are in use. The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

```
display system-parameters customer-options                               Page 2 of 12
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                                    USED
      Maximum Administered H.323 Trunks: 12000                        0
      Maximum Concurrently Registered IP Stations: 18000              2
      Maximum Administered Remote Office Trunks: 12000               0
Max Concurrently Registered Remote Office Stations: 18000            0
      Maximum Concurrently Registered IP eCons: 414                  0
      Max Concur Reg Unauthenticated H.323 Stations: 100             0
      Maximum Video Capable Stations: 41000                         0
      Maximum Video Capable IP Softphones: 18000                    6
      Maximum Administered SIP Trunks: 40000                        120
Max Administered Ad-hoc Video Conferencing Ports: 24000              0
      Max Number of DS1 Boards with Echo Cancellation: 999           0

(NOTE: You must logoff & login to effect the permission changes.)
```

## 5.2. System Features

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

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

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

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

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

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

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

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

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

SCCAN PARAMETERS
  Enable Enbloc Dialing without ARS FAC? n

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

### 5.3. IP Node Names

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

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



## 5.4. Codecs

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

change ip-codec-set 2 Page 1 of 2

IP MEDIA PARAMETERS

Codec Set: 2

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

Media Encryption

1: 1-srtp-aescm128-hmac80

2: none

3:

4:

5:

Encrypted SRTP: best-effort

On **Page 2**, set the **Fax Mode** to *off* (refer to **Section 2.2**).

change ip-codec-set 2

Page 2 of 2

IP MEDIA PARAMETERS

Allow Direct-IP Multimedia? ☐

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

Media Connection IP Address Type Preferences


1: IPv4

2:

## 5.5. IP Network Regions

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

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

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

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

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

## 5.6. Signaling Group

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

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

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

- Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? is changed to *y*.
- Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? is changed to *n*.
- Set the **Near-end Node Name** to *procr*. This node name maps to the IP address of the Communication Manager as defined in **Section 5.3**.
- Set the **Far-end Node Name** to *SM*. This node name maps to the IP address of Session Manager, as defined in **Section 5.3**.
- Set the **Near-end Listen Port** and **Far-end Listen Port** to a valid unused port instead of the default well-known port value. (For TLS, the well-known port value is 5061). This is necessary so Session Manager can distinguish this trunk from the trunk used for other enterprise SIP traffic. The compliance test was conducted with the **Near-end Listen Port** and **Far-end Listen Port** set to *5071*.
- Set the **Far-end Network Region** to the IP network region defined for the Service Provider in **Section 5.5**.
- Set the **Far-end Domain** to the domain of the enterprise.
- Set the **DTMF over IP** field to *rtp-payload*. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- Set **Direct IP-IP Audio Connections** to *n* (refer to **Section 2.2**)
- Default values may be used for all other fields.

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

## 5.7. Trunk Group

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

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

```
change trunk-group 2                                     Page 1 of 4
                                     TRUNK GROUP
Group Number: 2      Group Type: sip      CDR Reports: y
Group Name: Service Provider      COR: 1      TN: 1      TAC: 602
Direction: two-way      Outgoing Display? n
Dial Access? n      Night Service: _____
Queue Length: 0
Service Type: public-ntwrk      Auth Code? n
                                     Member Assignment Method: auto
                                     Signaling Group: 2
                                     Number of Members: 10
```

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

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

On Page 3:

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

change trunk-group 2 Page 3 of 4

TRUNK FEATURES

ACA Assignment? ☐ Measured: none Maintenance Tests? y

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

**Replace Restricted Numbers? y**  
**Replace Unavailable Numbers? y**

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

Show ANSWERED BY on Display? y



On Page 4:

- Set the **Network Call Redirection** field to *n*. With this setting, Communication Manager will not use the SIP REFER method for the redirection of PSTN calls that are transferred back to the SIP trunk (refer to **Section 2.2**).
- Set the **Send Diversion Header** field to *n* and **Support Request History** to *n*.
- Set the **Telephone Event Payload Type** to **101**, the value preferred by Clearcom.
- Verify that **Identity for Calling Party Display** is set to *P-Asserted-Identity*.
- Default values were used for all other fields.

change trunk-group 2 Page 4 of 4 ^

PROTOCOL VARIATIONS

Mark Users as Phone? n

Prepend '+' to Calling/Alerting/Diverting/Connected Number? n

Send Transferring Party Information? n

Network Call Redirection? n

Send Diversion Header? n

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

Interworking of ISDN Clearing with In-Band Tones: keep-channel-active

Request URI Contents: may-have-extra-digits

## 5.8. Calling Party Information

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

change public-unknown-numbering 1					Page 1 of 2
NUMBERING - PUBLIC/UNKNOWN FORMAT					
Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	Total CPN Len	
4	3			4	Total Administered: 4
4	5			4	Maximum Entries: 9999
4	3042	2	525528815941	12	Note: If an entry applies to a SIP connection to Avaya Aura(R) Session Manager, the resulting number must be a complete E.164 number.
4	3044	2	525528815942	12	
—	—	—	—	—	Communication Manager automatically inserts a '+' digit in this case.
—	—	—	—	—	
—	—	—	—	—	
—	—	—	—	—	
—	—	—	—	—	
—	—	—	—	—	
—	—	—	—	—	
—	—	—	—	—	

## 5.9. Inbound Routing

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

change inc-call-handling-trmt trunk-group 2					Page 1 of 30
INCOMING CALL HANDLING TREATMENT					
Service/ Feature	Number Len	Number Digits	Del	Insert	
public-ntwrk	12	525528815941	12	3042	
public-ntwrk	12	525528815942	12	3044	
public-ntwrk					
public-ntwrk					
public-ntwrk					
public-ntwrk					
public-ntwrk					
public-ntwrk					
public-ntwrk					
public-ntwrk					
public-ntwrk					
public-ntwrk					
public-ntwrk					
public-ntwrk					
public-ntwrk					
public-ntwrk					
public-ntwrk					

## 5.10.Outbound Routing

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

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

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

change feature-access-codes		Page 1 of 10
FEATURE ACCESS CODE (FAC)		
Abbreviated Dialing List1 Access Code:	_____	
Abbreviated Dialing List2 Access Code:	_____	
Abbreviated Dialing List3 Access Code:	_____	
Abbreviated Dial - Prgm Group List Access Code:	_____	
Announcement Access Code:	#7	
Answer Back Access Code:	_____	
Attendant Access Code:	_____	
Auto Alternate Routing (AAR) Access Code:	8	
Auto Route Selection (ARS) - Access Code 1:	9	Access Code 2: _____
Automatic Callback Activation:	_____	Deactivation: _____
Call Forwarding Activation Busy/DA:	_____ All: _____	Deactivation: _____
Call Forwarding Enhanced Status:	_____ Act: _____	Deactivation: _____
Call Park Access Code:	_____	
Call Pickup Access Code:	_____	
CAS Remote Hold/Answer Hold-Unhold Access Code:	_____	
CDR Account Code Access Code:	_____	
Change COR Access Code:	_____	
Change Coverage Access Code:	_____	
Conditional Call Extend Activation:	_____	Deactivation: _____
Contact Closure Open Code:	_____	Close Code: _____

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

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

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

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

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

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

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

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

## 6. Configure Avaya Aura® Session Manager

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

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

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



## 6.1. System Manager Login and Navigation

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

The screenshot displays the Avaya Aura System Manager 8.1 web interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The 'Elements' menu is expanded, showing a list of system components. The 'Routing' option is highlighted in red, and its submenu is also expanded, with 'Domains' highlighted in red. The main dashboard area contains several widgets: 'System Resource Utilization' (a bar chart showing utilization for 'opt', 'var', and 'emdata'), 'Alarms' (an empty list), 'Notifications' (showing 'No data'), 'Application State' (a table with system status), 'Information' (a table of system components and their counts), and 'Shortcuts' (a drag-and-drop area). The 'Information' table is as follows:

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

The 'Current Usage' section shows two purple boxes: '6/250000 USERS' and '1/50 SIMULTANEOUS ADMINISTRATIVE LOGINS'.

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

The screenshot displays the Avaya Aura System Manager 8.1 web interface. The top navigation bar includes the Avaya logo, the version 'Aura® System Manager 8.1', and several menu items: 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. Below this, a secondary navigation bar shows 'Home' and 'Routing' (which is highlighted with a red box). On the left side, a dark navigation tree is visible, with 'Routing' expanded to show a list of sub-items: 'Domains', 'Locations', 'Conditions', 'Adaptations', 'SIP Entities', 'Entity Links', 'Time Ranges', 'Routing Policies', 'Dial Patterns', 'Regular Expressions', and 'Defaults'. The 'Domains' item is highlighted with a blue bar and a red box. The main content area is titled 'Domain Management' and features a toolbar with 'New', 'Edit', 'Delete', 'Duplicate', and 'More Actions' buttons. Below the toolbar, it indicates '1 Item' and displays a table with the following data:

	Name	Type	Notes
<input type="checkbox"/>	<a href="#">avaya.lab.com</a>	sip	HG V-Domain

At the bottom of the table, there is a selection control: 'Select : All, None'.

## 6.2. SIP Domain

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

- **Name:** Enter the domain name.
- **Type:** Select **sip** from the pull-down menu.
- **Notes:** Add a brief description (optional).
- Click **Commit** to save (not shown).

The screen below shows the entry for the enterprise domain.

The screenshot displays the Avaya Aura System Manager 8.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The left-hand navigation pane shows 'Routing' selected, with 'Domains' highlighted. The main content area is titled 'Domain Management' and features a table with one item. The table has columns for 'Name', 'Type', and 'Notes'. The item listed is 'avaya.lab.com' with type 'sip' and notes 'HG V-Domain'.

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

## 6.3. Locations

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

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

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

The screenshot displays the Avaya Aura System Manager 8.1 web interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 8.1', and menu items for Users, Elements, Services, Widgets, and Shortcuts. Below this, a secondary navigation bar shows 'Home' and 'Routing' (highlighted with a red box). A left-hand sidebar contains a tree view with 'Routing' expanded, showing sub-items: Domains, Locations (highlighted with a red box), Conditions, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Location Details' and includes 'Commit' and 'Cancel' buttons. It is divided into three sections: 'General' with fields for 'Name' (containing 'Session Manager' and highlighted with a red box) and 'Notes' (containing 'VMware Session Manager'); 'Dial Plan Transparency in Survivable Mode' with an 'Enabled' checkbox and fields for 'Listed Directory Number' and 'Associated CM SIP Entity'; and 'Overall Managed Bandwidth' with a 'Managed Bandwidth Units' dropdown (set to 'Kbit/sec'), and input fields for 'Total Bandwidth' and 'Multimedia Bandwidth'. At the bottom, the 'Audio Calls Can Take Multimedia Bandwidth' checkbox is checked.

**AVAYA**  
Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ | Widgets ▾ Shortcuts ▾

Home Routing

Routing ^

Domains

Locations

Conditions

Adaptations ▾

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns ▾

Regular Expressions

Defaults

**Location Details** Commit Cancel

**General**

\* Name: Session Manager

Notes: VMware Session Manager

**Dial Plan Transparency in Survivable Mode**

Enabled: ☐

Listed Directory Number:

Associated CM SIP Entity:

**Overall Managed Bandwidth**

Managed Bandwidth Units: Kbit/sec ▾

Total Bandwidth:

Multimedia Bandwidth:

Audio Calls Can Take Multimedia Bandwidth: ☒

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

The screenshot displays the Avaya Aura System Manager 8.1 web interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 8.1', and menu items for Users, Elements, Services, Widgets, and Shortcuts. A secondary navigation bar shows 'Home' and 'Routing' (highlighted with a red box). On the left, a sidebar menu lists various configuration options, with 'Locations' highlighted in blue and also marked with a red box. The main content area is titled 'Location Details' and includes 'Commit' and 'Cancel' buttons. It is divided into three sections: 'General', 'Dial Plan Transparency in Survivable Mode', and 'Overall Managed Bandwidth'. In the 'General' section, the 'Name' field is set to 'Communication Manager' (highlighted with a red box) and the 'Notes' field contains 'VMware Communication Manager'. The 'Dial Plan Transparency in Survivable Mode' section shows 'Enabled' as an unchecked checkbox, 'Listed Directory Number' as an empty field, and 'Associated CM SIP Entity' as an empty field. The 'Overall Managed Bandwidth' section shows 'Managed Bandwidth Units' as 'Kbit/sec', 'Total Bandwidth' as an empty field, 'Multimedia Bandwidth' as an empty field, and 'Audio Calls Can Take Multimedia Bandwidth' as a checked checkbox.

**AVAYA**  
Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ | Widgets ▾ Shortcuts ▾

Home Routing

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

**Location Details** Commit Cancel

**General**

\* Name: Communication Manager  
Notes: VMware Communication Manager

**Dial Plan Transparency in Survivable Mode**

Enabled: ☐

Listed Directory Number:

Associated CM SIP Entity:

**Overall Managed Bandwidth**

Managed Bandwidth Units: Kbit/sec ▾  
Total Bandwidth:   
Multimedia Bandwidth:   
Audio Calls Can Take Multimedia Bandwidth: ☒

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

The screenshot displays the Avaya Aura System Manager 8.1 web interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 8.1', and menu items for Users, Elements, Services, Widgets, and Shortcuts. Below this, a breadcrumb trail shows 'Home' and 'Routing'. The left sidebar contains a tree view with 'Routing' expanded, showing sub-items: Domains, Locations (highlighted in blue), Conditions, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Location Details' and includes 'Commit' and 'Cancel' buttons. It is divided into three sections: 'General' with fields for 'Name' (set to 'Avaya SBCE') and 'Notes' (set to 'VMware Avaya SBCE'); 'Dial Plan Transparency in Survivable Mode' with an 'Enabled' checkbox and fields for 'Listed Directory Number' and 'Associated CM SIP Entity'; and 'Overall Managed Bandwidth' with a 'Managed Bandwidth Units' dropdown (set to 'Kbit/sec'), fields for 'Total Bandwidth' and 'Multimedia Bandwidth', and a checked checkbox for 'Audio Calls Can Take Multimedia Bandwidth'.

**AVAYA**  
Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ | Widgets ▾ Shortcuts ▾

Home Routing

Routing ^

Domains

Locations

Conditions

Adaptations ▾

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns ▾

Regular Expressions

Defaults

### Location Details

Commit Cancel

#### General

\* Name: Avaya SBCE

Notes: VMware Avaya SBCE

#### Dial Plan Transparency in Survivable Mode

Enabled: ☐

Listed Directory Number:

Associated CM SIP Entity:

#### Overall Managed Bandwidth

Managed Bandwidth Units: Kbit/sec ▾

Total Bandwidth:

Multimedia Bandwidth:

Audio Calls Can Take Multimedia Bandwidth: ☒

## 6.4. Adaptations

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

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

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

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

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

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

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

The screenshot displays the Avaya Aura System Manager 8.1 interface. The top navigation bar includes the Avaya logo, version information, and user roles (Users, Elements, Services, Widgets, Shortcuts). A search bar and user profile (admin) are also present. The left sidebar shows a navigation menu with 'Routing' selected, and 'Adaptations' highlighted under the 'Routing' section. The main content area is titled 'Adaptation Details' and includes 'Commit' and 'Cancel' buttons. The 'General' tab is active, showing the following fields:

- Adaptation Name:** CM\_Outbound\_Header\_Removal
- Module Name:** DigitConversionAdapter
- Module Parameter Type:** Name-Value Parameter

Below these fields is a table for defining parameters:

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

At the bottom of the table, there is a 'Select' dropdown menu with options 'All' and 'None'.



## 6.5. SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to it, which includes Communication Manager and the Avaya SBCE. Navigate to **Routing** → **SIP Entities** in the left navigation pane and click on the **New** button in the right pane (not shown). In the **General** section, enter the following values. Use default values for all remaining fields:

- **Name:** Enter a descriptive name.
- **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity that is used for SIP signaling (see **Figure 1**).
- **Type:** Select *Session Manager* for Session Manager, *CM* for Communication Manager and *SIP Trunk* (or *Other*) for the Avaya SBCE.
- **Adaptation:** This field is only present if **Type** is not set to **Session Manager**. If Adaptations were to be created, here is where they would be applied to the entity.
- **Location:** Select the location that applies to the SIP Entity being created, defined in **Section 6.3**.
- **Time Zone:** Select the time zone for the location above.
- Click **Commit** to save.

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

The screenshot displays the Avaya Aura System Manager 8.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The left sidebar shows a tree view with 'Routing' selected, and 'SIP Entities' highlighted. The main content area is titled 'SIP Entity Details' with a 'General' sub-section. The form contains the following fields:

- Name:** Session Manager
- \* IP Address:** 10.64.101.249
- SIP FQDN:** (empty)
- Type:** Session Manager
- Notes:** VMware Session Manager
- Location:** Session Manager
- Outbound Proxy:** (empty)
- Time Zone:** America/New\_York
- Minimum TLS Version:** Use Global Setting

The 'Commit' and 'Cancel' buttons are located in the top right corner of the form.

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

**AVAYA**  
Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ | Widgets ▾ Shortcuts ▾

Home Routing

Routing  
Domains  
Locations  
Conditions  
Adaptations  
Adaptations  
Regular Expression ...  
**SIP Entities**  
Entity Links  
Time Ranges  
Routing Policies

**SIP Entity Details** Commit Cancel

**General**

\* Name: Communication Manager Trunk 2

\* FQDN or IP Address: 10.64.101.241

Type: CM ▾

Notes: Used for SP Testing

Adaptation: ▾

Location: Communication Manager ▾

Time Zone: America/New\_York ▾

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

Minimum TLS Version: Use Global Setting ▾

Credential name:

Securable: ☐

Call Detail Recording: none ▾

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

- The **FQDN or IP Address** field is set to the IP address of the SBC private network interface (see **Figure 1**).
- For **Type** select *SIP Trunk*.
- On the **Adaptation** field, the adaptation module *CM\_Outbound\_Header\_Removal* previously defined in **Section 6.4** was selected.
- Select the location that applies to the SIP Entity being created, defined in **Section 6.3**.
- Select the **Time Zone**.
- Click **Commit** to save.

**AVAYA**  
Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ | Widgets ▾ Shortcuts ▾

Home Routing

Routing  
Domains  
Locations  
Conditions  
Adaptations ▾  
**SIP Entities**  
Entity Links  
Time Ranges  
Routing Policies

### SIP Entity Details

Commit Cancel

**General**

\* Name: Avaya SBCE

\* FQDN or IP Address: 10.64.101.243

Type: SIP Trunk ▾

Notes: VMware Avaya SBCE

Adaptation: CM\_Outbound\_Header\_Removal ▾

Location: Avaya SBCE ▾

Time Zone: America/New\_York ▾

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

Minimum TLS Version: Use Global Setting ▾

Credential name:

## 6.6. Entity Links

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

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

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

The screenshot displays the Avaya Aura System Manager 8.1 interface. The left navigation pane shows the 'Entity Links' section selected. The main content area is titled 'Entity Links' and contains a table with one item. The table has the following columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, and DNS Override. The item in the table is 'Session\_Manager\_CM\_Trunk', which is linked to 'Session Manager' via 'TLS' on port '5071', and 'Communication Manager Trunk 2' on port '5071'. The table has a 'Filter: Enable' button and a 'Select: All, None' dropdown. Buttons for 'Commit' and 'Cancel' are visible at the top and bottom right of the table area.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	DNS Override
Session_Manager_CM_Trunk	Session Manager	TLS	5071	Communication Manager Trunk 2	5071	

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

The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes the Avaya logo, version 8.1, and tabs for Users, Elements, Services, Widgets, and Shortcuts. A search bar and a user profile (admin) are also present. The left sidebar contains a menu with items: Home, Routing, Domains, Locations, Conditions, Adaptations, SIP Entities, Entity Links (highlighted with a red box), Time Ranges, and Routing Policies. The main content area is titled 'Entity Links' and features a table with one item. The table columns are: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, and DNS Overr. The single row shows a link from 'Session\_Manager\_ASBC' to 'Avaya SBCE' using the 'TLS' protocol on port '5061'. The 'Name' and 'SIP Entity 2' fields are also highlighted with red boxes. At the bottom of the table, there is a 'Select : All, None' option and 'Commit' and 'Cancel' buttons.

	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	DNS Overr
<input type="checkbox"/>	Session_Manager_ASBC	Session Manager	TLS	5061	Avaya SBCE	5061	<input type="checkbox"/>

## 6.7. Routing Policies

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

- In the **General** section, enter a descriptive **Name** and add a brief description under **Notes** (optional).
- In the **SIP Entity as Destination** section, click **Select**. The **SIP Entity List** page opens (not shown). Choose the appropriate SIP entity to which this routing policy applies (**Section 6.5**) and click **Select**. The selected SIP Entity displays on the **Routing Policy Details** page as shown below.
- Use default values for remaining fields.
- Click **Commit** to save.

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

**AVAYA**  
Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ Widgets ▾ Shortcuts ▾

Search 🔍 admin

Home Routing

Routing Policy Details

Commit Cancel

**General**

\* Name: To CM Trunk 2

Disabled: ☐

\* Retries: 0

Notes: For inbound calls to CM via Trunk 2

**SIP Entity as Destination**

Select

Name	FQDN or IP Address	Type	Notes
Communication Manager Trunk 2	10.64.101.241	CM	Used for SP Testing

**Time of Day**

Add Remove View Gaps/Overlaps

1 Item

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

Select : All, None

**Routing Policy Details**

**General**

\* Name: Avaya SBCE

Disabled: ☐

\* Retries: 0

Notes: For outbound calls to SP via ASBCE

**SIP Entity as Destination**

Name	FQDN or IP Address	Type	Notes
Avaya SBCE	10.64.101.243	SIP Trunk	VMware Avaya SBCE

**Time of Day**

Add Remove View Gaps/Overlaps

1 Item Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None

## 6.8. Dial Patterns

Dial Patterns are needed to route specific calls through Session Manager. For the compliance test, dial patterns were needed to route calls from Communication Manager to the service provider and vice versa. Dial Patterns define which route 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 on the **New** button in the right pane (not shown). Fill in the following, as shown in the screens below:

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, or select “**ALL**” to route incoming calls to all SIP domains.
- **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 (**Section 6.3**).
- Lastly, select the routing policy from the list that will be used to route all calls that match the specified criteria (**Section 6.7**). Click **Select** (not shown).
- Click **Commit** to save.

AVAYA

Aura® System Manager 8.1

Users

Elements

Services

Widgets

Shortcuts

Search

admin

Home

Routing

Routing

Domains

Locations

Conditions

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Origination Dial Pat...

Dial Pattern Details

Commit Cancel

Help ?

General

\* Pattern: 525528

\* Min: 6

\* Max: 36

Emergency Call: ☐

SIP Domain: avaya.lab.com

Notes:

Originating Locations, Origination Dial Pattern Sets, and Routing Policies

Add Remove

1 Item

Filter: Enable

	Originating Location Name	Originating Location Notes	Origination Dial Pattern Set Name	Origination Dial Pattern Set Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Avaya SBCE	VMware Avaya SBCE			To CM Trunk 2	0	<input type="checkbox"/>	Communication Manager Trunk 2	For inbound calls to CM via Trunk 2

Select : All, None



The example in this screen shows the 13-digit dialed numbers for outbound calls, beginning with **001**, arriving from the **Communication Manager** location, will use route policy **Avaya SBCE**, which sends the call out to the PSTN via Avaya SBCE and the service provider SIP trunk. The SIP Domain was set to **avaya.lab.com**.

**AVAYA**  
Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ Widgets ▾ Shortcuts ▾

Search [ ] admin

Home Routing

Routing

Domains

Locations

Conditions

Adaptations ▾

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Origination Dial Pat...

**Dial Pattern Details** [Commit] [Cancel] [Help ?]

**General**

\* Pattern: 001

\* Min: 13

\* Max: 13

Emergency Call: ☐

SIP Domain: avaya.lab.com ▾

Notes: [ ]

**Originating Locations, Origination Dial Pattern Sets, and Routing Policies**

[Add] [Remove]

1 Item [Filter: Enable]

	Originating Location Name ▴	Originating Location Notes	Origination Dial Pattern Set Name	Origination Dial Pattern Set Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Communication Manager	VMware Communication Manager			Avaya SBCE	0	<input type="checkbox"/>	Avaya SBCE	For outbound calls to SP via ASBCE

Select : All, None

Repeat the above procedures as needed to define additional dial patterns.

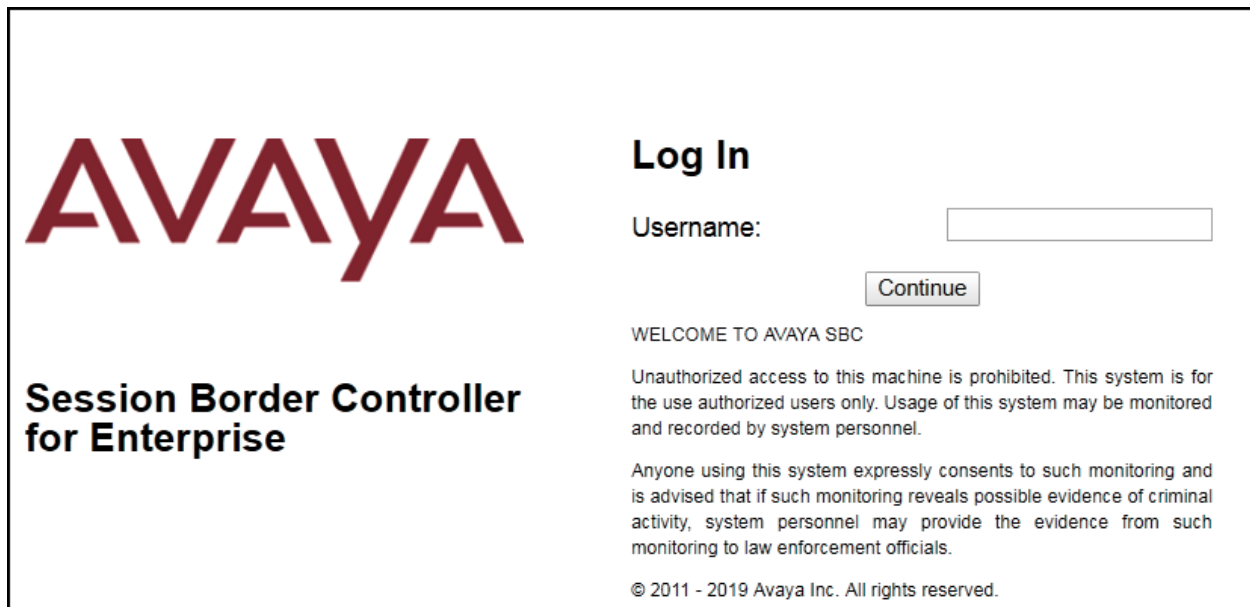
## 7. Configure Avaya Session Border Controller for Enterprise

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

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

### 7.1. System Access

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



The screenshot displays the login interface for the Avaya Session Border Controller for Enterprise. On the left, the Avaya logo is shown in a dark red color, with the text "Session Border Controller for Enterprise" below it. On the right, the "Log In" section includes a "Username:" label, a text input field, and a "Continue" button. Below the login fields, a "WELCOME TO AVAYA SBC" message is followed by a disclaimer: "Unauthorized access to this machine is prohibited. This system is for the use authorized users only. Usage of this system may be monitored and recorded by system personnel." A second paragraph states: "Anyone using this system expressly consents to such monitoring and is advised that if such monitoring reveals possible evidence of criminal activity, system personnel may provide the evidence from such monitoring to law enforcement officials." At the bottom, the copyright notice "© 2011 - 2019 Avaya Inc. All rights reserved." is displayed.

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

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

EMS  
Avaya\_SBCE

### Session Border Controller for Enterprise

AVAYA

**EMS Dashboard**  
Device Management  
‣ System Administration  
Backup/Restore  
‣ Monitoring & Logging

**Dashboard**

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

Installed Devices
EMS
Avaya_SBCE

Active Alarms (past 24 hours)  
None found.

Incidents (past 24 hours)  
None found.

Notes  
No notes found.

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

Device: Avaya\_SBCE Alarms Incidents Status Logs Diagnostics Users Settings Help Log Out

### Session Border Controller for Enterprise

AVAYA

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

**Dashboard**

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

Installed Devices
EMS
Avaya_SBCE

Active Alarms (past 24 hours)  
None found.

Incidents (past 24 hours)  
Avaya\_SBCE: No Subscriber Flow Matched

## 7.2. Device Management

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

The screenshot displays the Avaya Session Border Controller for Enterprise (SBCE) web interface. The top navigation bar includes links for Device: Avaya\_SBCE, Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header reads "Session Border Controller for Enterprise" with the AVAYA logo. The left sidebar lists various management options, with "Device Management" highlighted. The main content area is titled "Device Management" and contains a sub-menu with "Devices", "Updates", "SSL VPN", "Licensing", and "Key Bundles". The "Devices" tab is active, showing a table of managed devices. The table has columns for Device Name, Management IP, Version, and Status. A single device, "Avaya\_SBCE", is listed with a blurred Management IP, version "8.0.1.0-10-17555", and status "Commissioned". Below the table, there are action buttons: Reboot, Shutdown, Restart Application, View (highlighted with a red box), Edit, and Uninstall.

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

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

System Information: Avaya\_SBCE

X

General Configuration

Appliance Name

Avaya\_SBCE

Box Type

SIP

Deployment Mode

Proxy

Device Configuration

HA Mode

No

Two Bypass Mode

No

License Allocation

Standard Sessions

Requested: 2000

1000

Advanced Sessions

Requested: 2000

1000

Scopia Video Sessions

Requested: 500

500

CES Sessions

Requested: 0

0

Transcoding Sessions

Requested: 0

0

CLID

---

Encryption

Available: Yes

☒

Network Configuration

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

DNS Configuration

Primary DNS

8.8.8.8

Secondary DNS

7.7.7.7

DNS Location

DMZ

DNS Client IP

10.10.80.51

Management IP(s)

IP #1 (IPv4)

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

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

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

### 7.3. TLS Management

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

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

### 7.4. Network Management

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

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

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

Device: Avaya\_SBCE
Alarms 1
Incidents
Status
Logs
Diagnostics
Users
Settings
Help
Log Out

## Session Border Controller for Enterprise

EMS Dashboard
Device Management
Backup/Restore
System Parameters
Configuration Profiles
Services
Domain Policies
TLS Management
Network & Flows
Network Management
Media Interface
Signaling Interface

### Network Management

Interfaces
Networks

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

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

Device: Avaya\_SBCE
Alarms 1
Incidents
Status
Logs
Diagnostics
Users
Settings
Help
Log Out

## Session Border Controller for Enterprise

EMS Dashboard
Device Management
Backup/Restore
System Parameters
Configuration Profiles
Services
Domain Policies
TLS Management
Network & Flows
Network Management
Media Interface
Signaling Interface

### Network Management

Interfaces
Networks

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

## 7.5. Media Interfaces

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

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

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

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

The screenshot shows the 'Add Media Interface' dialog box. The 'Name' field is 'Private\_med'. The 'IP Address' section has a dropdown menu set to 'Network\_A1 (A1, VLAN 0)' and a text field set to '10.64.101.243'. The 'Port Range' section has two text fields set to '35000' and '40000'. A 'Finish' button is at the bottom.

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

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

The screenshot shows the 'Add Media Interface' dialog box. The 'Name' field is 'Public\_med'. The 'IP Address' section has a dropdown menu set to 'Network\_B1 (B1, VLAN 0)' and a text field set to '10.10.80.51'. The 'Port Range' section has two text fields set to '35000' and '40000'. A 'Finish' button is at the bottom.



## 7.6. Signaling Interfaces

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

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

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

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

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

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

The screenshot shows the 'Add Signaling Interface' configuration window. The 'Name' field is set to 'Public\_sig'. The 'IP Address' section shows a dropdown menu for the network, currently displaying 'Network\_B1 (B1. VLAN 0)', and a sub-dropdown for the IP address, displaying '10.10.80.51'. The 'UDP Port' field is set to '5060'. The 'Finish' button is located at the bottom right of the window.

## 7.7. Server Interworking

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

### 7.7.1. Server Interworking Profile – Enterprise

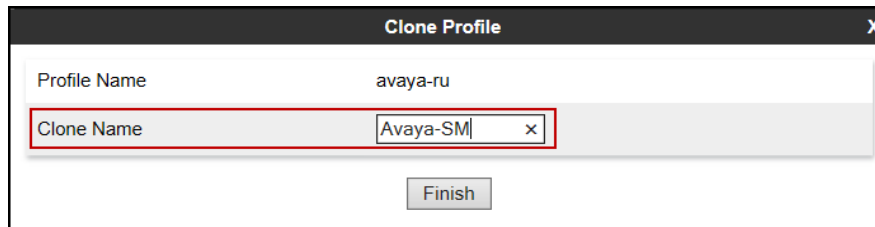
Interworking profiles can be created by cloning one of the pre-defined default profiles, or by adding a new profile. To configure the interworking profile in the enterprise direction, select **Configuration Profiles → Server Interworking** on the left navigation pane. Under **Interworking Profiles**,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

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

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

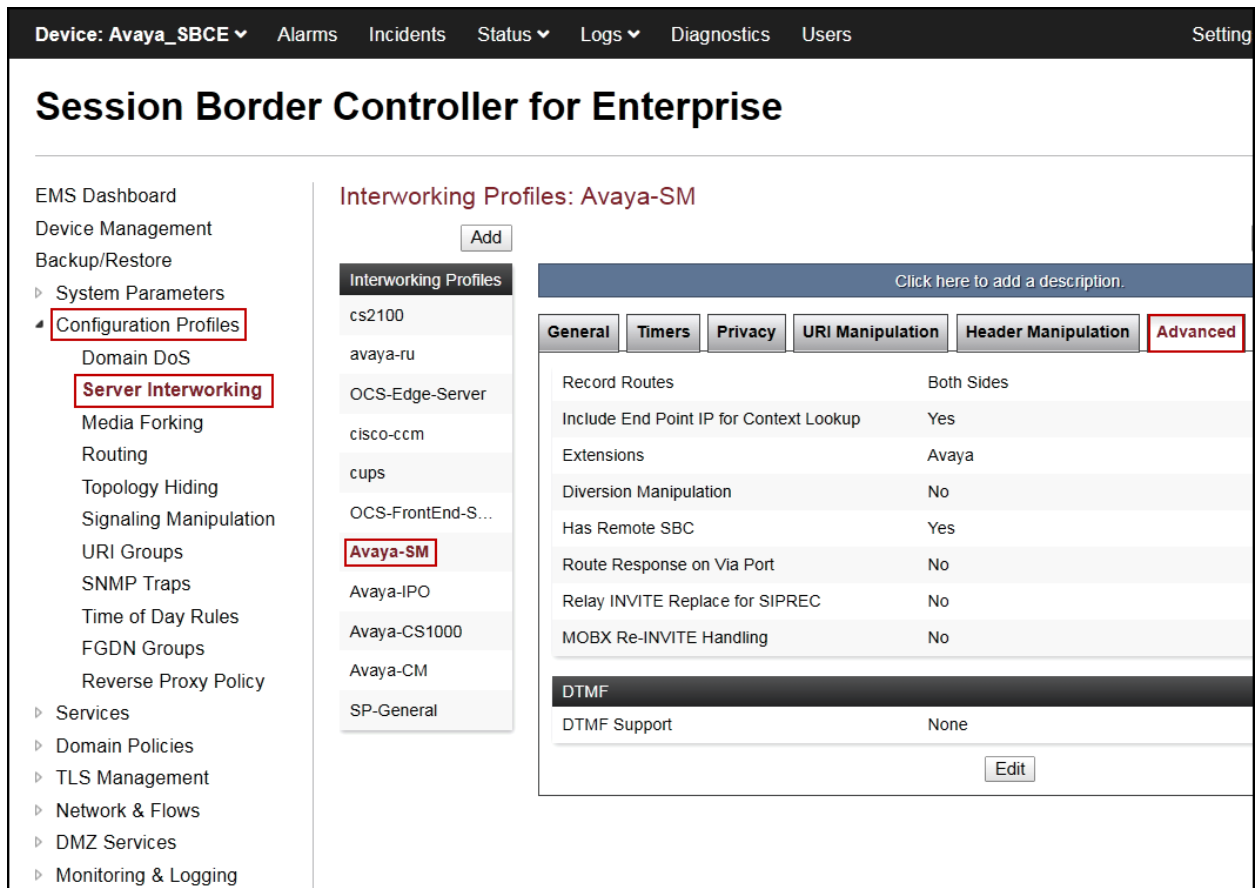
An 'Edit' button is located at the bottom right of the configuration table.

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



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

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

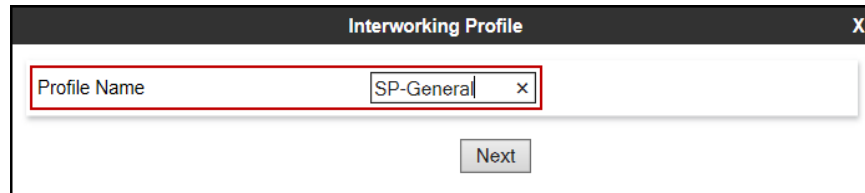


The image shows the 'Session Border Controller for Enterprise' web interface. The top navigation bar includes 'Device: Avaya\_SBCE', 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', and 'Setting'. The main heading is 'Session Border Controller for Enterprise'. On the left is a sidebar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

### 7.7.2. Server Interworking Profile – Service Provider

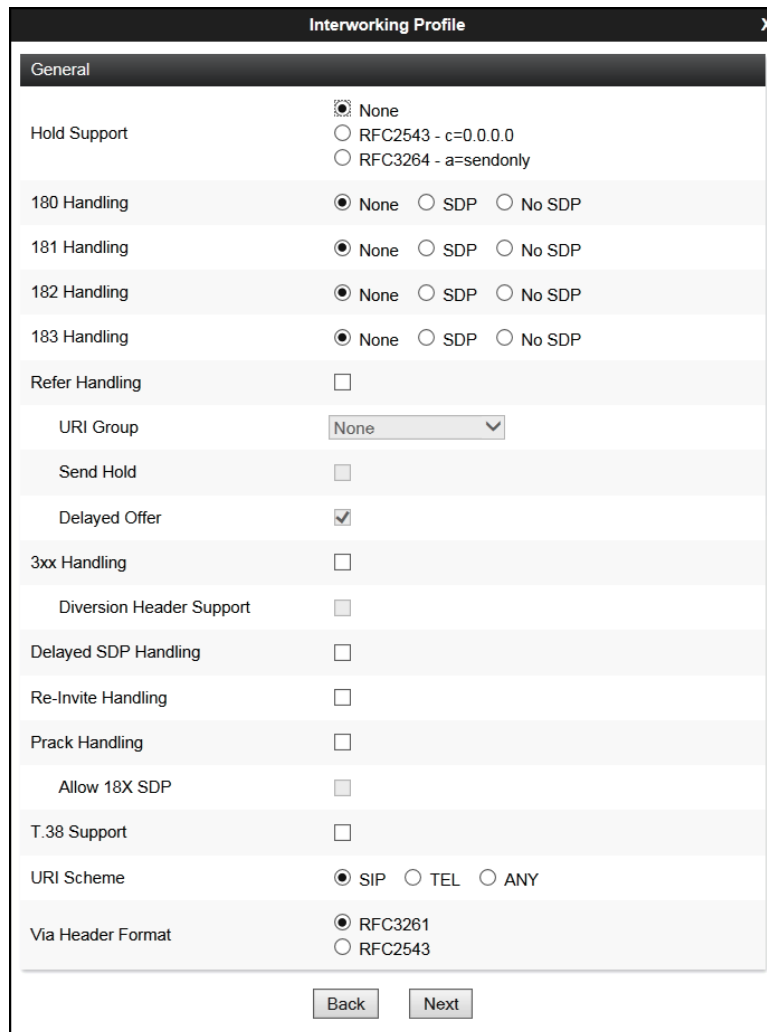
A second interworking profile in the direction of the SIP trunk was created, by adding a new profile in this case. Select **Global Profiles → Server Interworking** on the left navigation pane and click **Add** (not shown).

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



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

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



The screenshot shows the "Interworking Profile" dialog box with the "General" tab selected. The dialog contains the following configuration options:

- Hold Support: ☒ None, ☐ RFC2543 - c=0.0.0.0, ☐ RFC3264 - a=sendonly
- 180 Handling: ☒ None, ☐ SDP, ☐ No SDP
- 181 Handling: ☒ None, ☐ SDP, ☐ No SDP
- 182 Handling: ☒ None, ☐ SDP, ☐ No SDP
- 183 Handling: ☒ None, ☐ SDP, ☐ No SDP
- Refer Handling: ☐
- URI Group:  (dropdown)
- Send Hold: ☐
- Delayed Offer: ☒
- 3xx Handling: ☐
- Diversion Header Support: ☐
- Delayed SDP Handling: ☐
- Re-Invite Handling: ☐
- Prack Handling: ☐
- Allow 18X SDP: ☐
- T.38 Support: ☐
- URI Scheme: ☒ SIP, ☐ TEL, ☐ ANY
- Via Header Format: ☒ RFC3261, ☐ RFC2543

At the bottom of the dialog, there are "Back" and "Next" buttons.

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

Device: Avaya\_SBCE ▾

Alarms

Incidents

Status ▾

Logs ▾

Diagnostics

Users

Settings ▾

## Session Border Controller for Enterprise

EMS Dashboard

Device Management

Backup/Restore

▸ System Parameters

▾ Configuration Profiles

Domain DoS

**Server Interworking**

Media Forking

Routing

Topology Hiding

Signaling Manipulation

URI Groups

SNMP Traps

Time of Day Rules

FGDN Groups

Reverse Proxy Policy

▸ Services

▸ Domain Policies

▸ TLS Management

▸ Network & Flows

Interworking Profiles: SP-General

Add

Rename

Interworking Profiles

cs2100

avaya-ru

OCS-Edge-Server

cisco-ccm

cups

OCS-FrontEnd-...

Avaya-SM

Avaya-IPO

Avaya-CS1000

Avaya-CM

**SP-General**

Click here to add a description.

GeneralTimersPrivacyURI ManipulationHeader Manipulation**Advanced**

Record RoutesBoth Sides

Include End Point IP for Context LookupNo

ExtensionsNone

Diversion ManipulationNo

Has Remote SBCYes

Route Response on Via PortNo

Relay INVITE Replace for SIPRECNo

MOBX Re-INVITE HandlingNo

DTMF

DTMF SupportNone

Edit

## 7.8. Signaling Manipulation

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

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

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

- Copy the destination DID number present in the “To” header of incoming calls to the “Request-URI” header.
- Include the SIP trunk credential’s username in the “From” header of all outbound calls.
- Remove the “gsid” and “epv” parameters from outbound “Contact” headers.
- Remove the P-Location header.

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

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

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

```
//Replace Username in "REQUEST-LINE" with "TO" number on Inbound
within session "ALL"
{
act on message where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING"
{
%HEADERS["Request_Line"][1].URI.USER = %HEADERS["To"][1].URI.USER;
}
}
```

```
//Insert Username in the FROM header on Outbound
within session "ALL"
{
act on request where %DIRECTION="OUTBOUND" and
%ENTRY_POINT="POST_ROUTING"
{
```

```

%fromuser = %HEADERS["From"][1].URI.USER;
%HEADERS["From"][1].URI.USER = "user123";
}
}

//Remove gsid and epv parameters in outbound Contact header
//Remove P-Location parameter
within session "ALL"
{
act on message where %DIRECTION="OUTBOUND" and
%ENTRY_POINT="POST_ROUTING"
{
remove(%HEADERS["Contact"][1].URI.PARAMS["gsid"]);
remove(%HEADERS["Contact"][1].URI.PARAMS["epv"]);
remove(%HEADERS["P-Location"][1]);

}
}

```



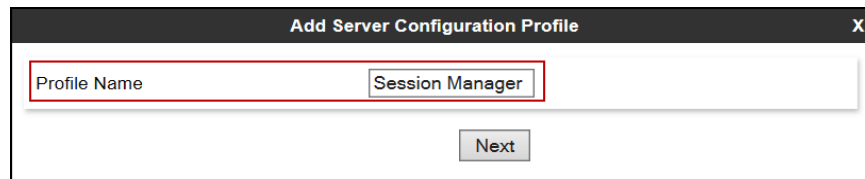
## 7.9. Server Configuration

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

### 7.9.1. Server Configuration Profile – Enterprise

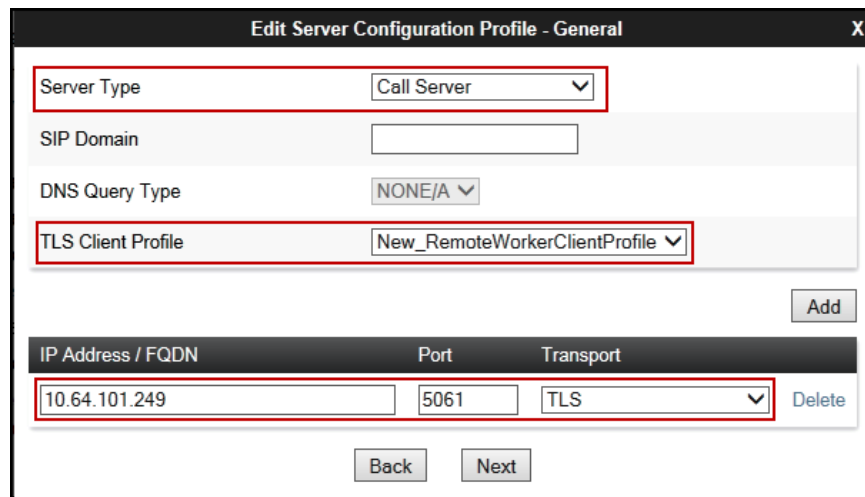
From the **Services** menu on the left-hand navigation pane, select **SIP Servers** and click the **Add** button (not shown) to add a new profile for the Call Server.

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



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

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



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

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

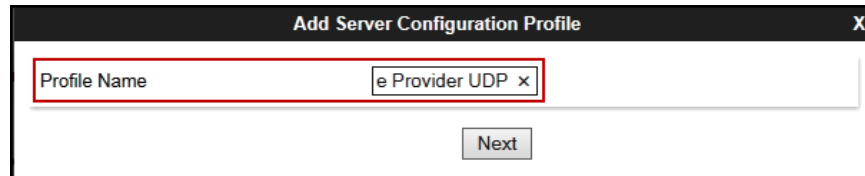
**Add SIP Server Profile - Advanced**

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

### 7.9.2. Server Configuration Profile – Service Provider

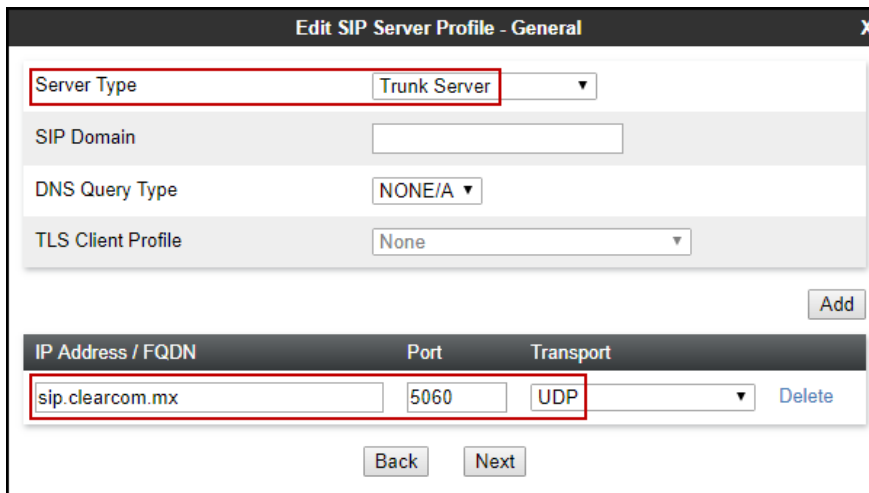
Similarly, to add the profile for the Trunk Server, click the **Add** button on the **Server Configuration** screen (not shown).

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



The screenshot shows a dialog box titled "Add Server Configuration Profile". It has a close button (X) in the top right corner. The main area contains a text input field labeled "Profile Name" which has the text "e Provider UDP" entered. A red rectangular box highlights this text. Below the input field is a "Next" button.

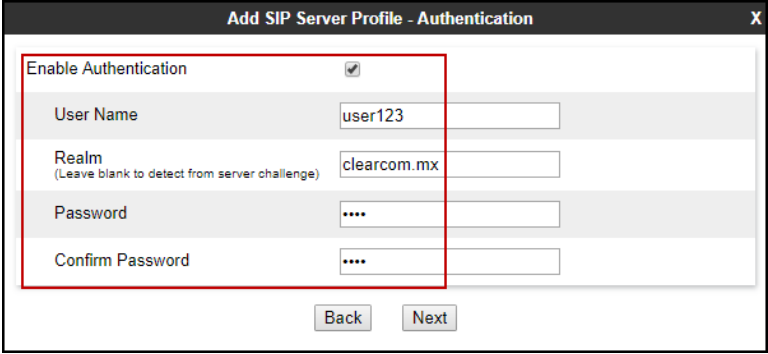
- On the **Edit Server Configuration Profile - General** Tab select *Trunk Server* from the drop-down menu for the **Server Type**.
- On the **IP Addresses / FQDN** field, enter *sip.clearcom.mx* (Clearcom SIP proxy server FQDN). This information was provided by Clearcom.
- Enter *5060* under **Port**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



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

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

- Check the **Enable Authentication** box.
- Enter the **User Name** credential provided by the service provider for SIP trunk registration.
- Enter the **Realm** credential provided by the service provider for SIP trunk registration. Note that the Service Provider's Domain Name was used.
- Enter **Password** credential provided by the service provider
- for SIP trunk registration.
- Click **Next**.



The screenshot shows a web-based configuration window titled "Add SIP Server Profile - Authentication". It contains a red-bordered box with the following fields:

- Enable Authentication**: A checkbox that is checked.
- User Name**: A text input field containing "user123".
- Realm**: A text input field containing "clearcom.mx". Below this field is the text "(Leave blank to detect from server challenge)".
- Password**: A text input field with masked characters "\*\*\*\*".
- Confirm Password**: A text input field with masked characters "\*\*\*\*".

Below the red-bordered box are two buttons: "Back" and "Next".

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

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

- Check the **Register with All Servers** box.
- **Frequency**: Enter the amount of time (in seconds) between REGISTER messages that will be sent from the enterprise to the Service Provider Proxy Server to refresh the registration binding of the SIP trunk. This value should be chosen in consultation with the service provider. **120** seconds was the value used during the compliance test.
- The **From URI** and **To URI** entries for the REGISTER messages are built using the following:
  - **From URI**: Use the **User Name** entered above in the **Authentication** screen (*user123*) and the Service Provider's domain name (*clearcom.mx*), as shown on the screen below.
  - **To URI**: Use the **User Name** entered above in the **Authentication** screen (*user123*) and the Service Provider's domain name (*clearcom.mx*), as shown on the screen below.
  - Click **Next**.

The screenshot shows a window titled "Add SIP Server Profile - Registration". It contains the following elements:

- Register with All Servers**: A checkbox that is checked.
- Register with Priority Server**: A checkbox that is unchecked.
- Refresh Interval**: A text box containing "120" followed by the label "seconds".
- From URI**: A text box containing "user123@clearcom.mx".
- To URI**: A text box containing "user123@clearcom.mx".
- Back** and **Next** buttons at the bottom.

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

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

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

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

**Add SIP Server Profile - Advanced**

Enable DoS Protection ☐

Enable Grooming ☐

Interworking Profile SP-General ▼

Signaling Manipulation Script Clearcom\_Script ▼

Securable ☐

Enable FGDN ☐

TCP Failover Port

TLS Failover Port

Tolerant ☐

URI Group None ▼

Back Finish

## 7.10.Routing

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

### 7.10.1. Routing Profile – Enterprise

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

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

**Routing Profile**

Profile Name

Next

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

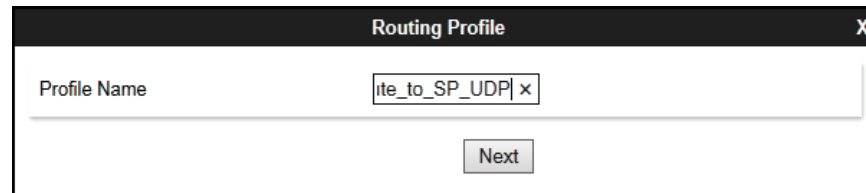
The screenshot shows the 'Routing Profile' configuration window. The 'SIP Server Profile' is set to 'Session Manager', and the 'Next Hop Address' is '10.64.101.249:5061 (TLS)'. The 'Priority / Weight' is set to '1'. The 'Add' button is visible at the bottom right of the configuration area.

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

## 7.10.2. Routing Profile – Service Provider

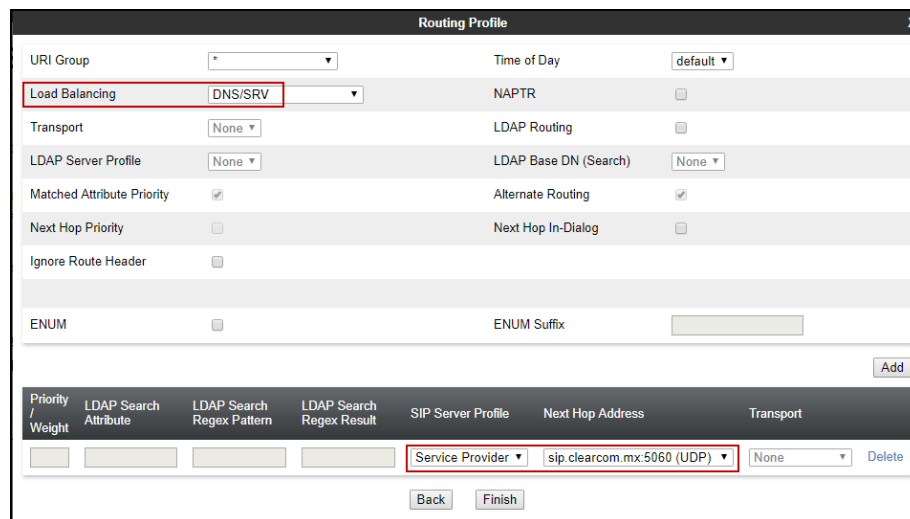
Back at the **Routing** tab, select **Add** (not shown) to repeat the process in order to create the outbound route.

- Enter an appropriate **Profile Name** similar to the example below (*Route\_to\_SP\_UDP* was used).
- Click **Next**.



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

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



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

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



## 7.11.Topology Hiding

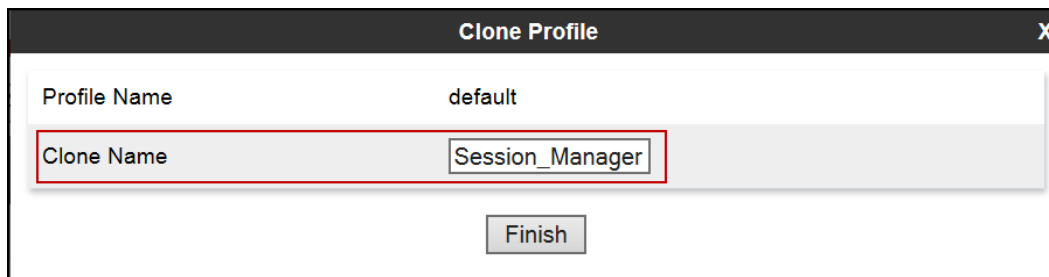
Topology Hiding is a security feature that allows the modification of several SIP headers, preventing private enterprise network information from being propagated to the untrusted public network.

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

### 7.11.1. Topology Hiding Profile – Enterprise

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

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



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

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

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

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

Finish

### 7.11.2. Topology Hiding Profile – Service Provider

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

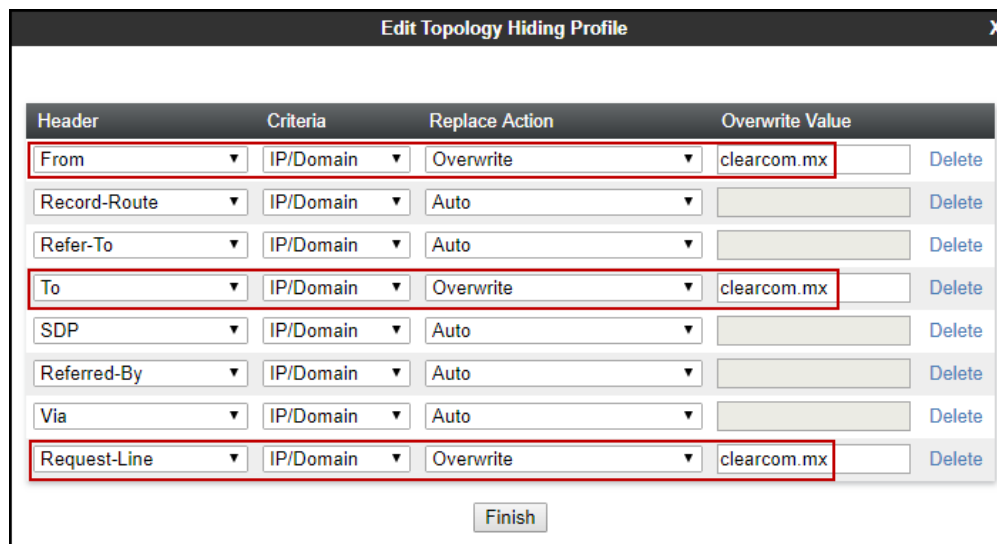
- Enter a **Clone Name** such as the one shown below.
- Click **Finish**.



The 'Clone Profile' dialog box has a title bar with 'Clone Profile' and a close button 'X'. It contains two input fields: 'Profile Name' with the value 'default' and 'Clone Name' with the value 'Service\_Provider'. The 'Clone Name' field is highlighted with a red border. Below the fields is a 'Finish' button.

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

- For the, **From**, **To** and **Request-Line** headers, select *Overwrite* in the **Replace Action** column and enter the enterprise SIP domain *clearcom.mx*, in the **Overwrite Value** column of these headers, as shown below. This is the service provider's domain name.
- Default values were used for all other fields.
- Click **Finish**.



The 'Edit Topology Hiding Profile' dialog box has a title bar with 'Edit Topology Hiding Profile' and a close button 'X'. It contains a table with the following columns: Header, Criteria, Replace Action, Overwrite Value, and a Delete button. The 'From', 'To', and 'Request-Line' rows are highlighted with red borders. The 'Replace Action' for these rows is 'Overwrite' and the 'Overwrite Value' is 'clearcom.mx'.

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

Below the table is a 'Finish' button.

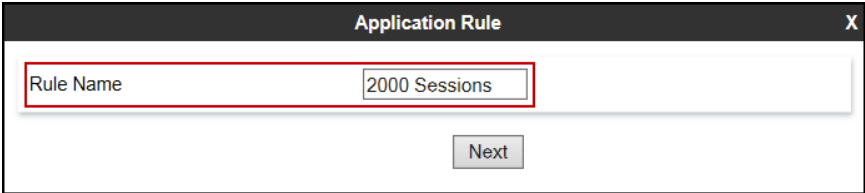
## 7.12.Domain Policies

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

### 7.12.1.Application Rules

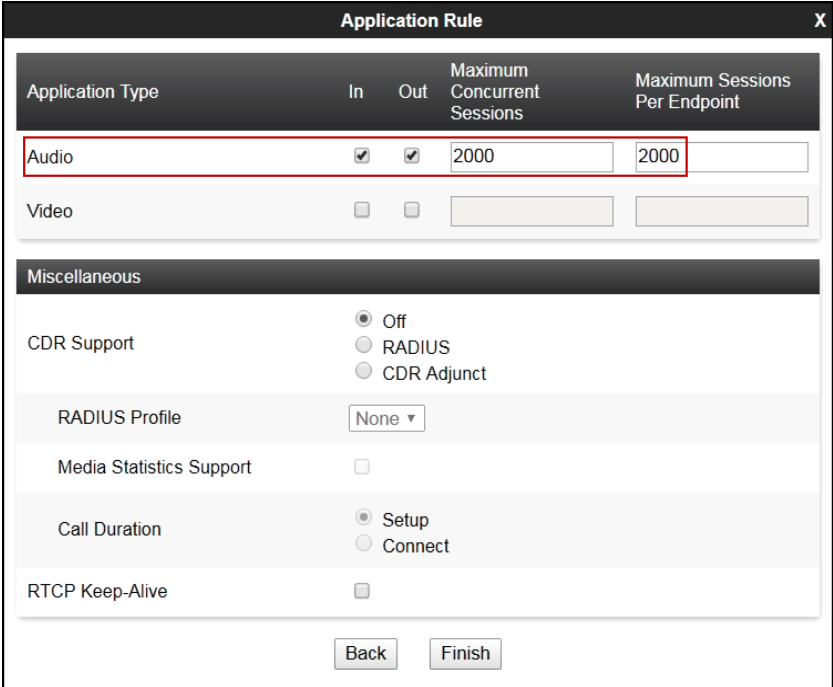
Application Rules define which types of SIP-based Unified Communications (UC) applications the UC-Sec security device will protect voice, video, and/or Instant Messaging (IM). In addition, Application Rules define the maximum number of concurrent voice sessions the network will process in order to prevent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

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



The screenshot shows a window titled "Application Rule" with a close button (X) in the top right corner. Inside the window, there is a text input field labeled "Rule Name" which contains the text "2000 Sessions". A red rectangular box highlights this input field. Below the input field, there is a "Next" button.

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



The screenshot shows a window titled "Application Rule" with a close button (X) in the top right corner. The window contains a table with the following columns: "Application Type", "In", "Out", "Maximum Concurrent Sessions", and "Maximum Sessions Per Endpoint". The "Audio" row is highlighted with a red box, showing "In" and "Out" checkboxes checked, and "Maximum Concurrent Sessions" and "Maximum Sessions Per Endpoint" both set to "2000". Below the table, there is a "Miscellaneous" section with various options: "CDR Support" (radio buttons for Off, RADIUS, CDR Adjunct), "RADIUS Profile" (dropdown menu set to None), "Media Statistics Support" (checkbox), "Call Duration" (radio buttons for Setup, Connect), and "RTCP Keep-Alive" (checkbox). At the bottom of the window, there are "Back" and "Finish" buttons.

## 7.12.2. Media Rules

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

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

- Click on the **Add** button to add a new media rule (not shown).
- Under **Rule Name** enter **SM\_SRTP**.
- Click **Next** (not shown).
- Under Audio Encryption, **Preferred Format #1**, select **SRTP\_AES\_CM\_128\_HMAC\_SHA1\_80**.
- Under Audio Encryption, **Preferred Format #2**, select **RTP**.
- Under Audio Encryption, uncheck **Encrypted RTCP**.
- Under Audio Encryption, check **Interworking**.
- Repeat the above steps under Video Encryption, if needed.
- Under Miscellaneous verify that **Capability Negotiation** is checked.
- Click **Next**.

The screenshot shows the 'Media Rule' configuration window. It is divided into three main sections: Audio Encryption, Video Encryption, and Miscellaneous. In the Audio Encryption section, Preferred Format #1 is set to 'SRTP\_AES\_CM\_128\_HMAC\_SHA1\_80', Preferred Format #2 is set to 'RTP', Preferred Format #3 is set to 'NONE', Encrypted RTCP is unchecked, MKI is unchecked, Lifetime is set to '2^4', Interworking is checked, and Encrypted RTCP is unchecked. In the Video Encryption section, Preferred Format #1 is set to 'SRTP\_AES\_CM\_128\_HMAC\_SHA1\_80', Preferred Format #2 is set to 'RTP', Preferred Format #3 is set to 'NONE', Encrypted RTCP is unchecked, MKI is unchecked, Lifetime is set to '2^4', Interworking is checked, and Encrypted RTCP is unchecked. In the Miscellaneous section, Capability Negotiation is checked. At the bottom, there are 'Back' and 'Next' buttons.

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

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

Media Encryption

Audio Encryption

Preferred Format #1

RTP

Preferred Format #2

NONE

Preferred Format #3

NONE

Encrypted RTCP

☐

MKI

☐

Lifetime

Leave blank to match any value.

2^

Interworking

☒

Video Encryption

Preferred Format #1

RTP

Preferred Format #2

NONE

Preferred Format #3

NONE

Encrypted RTCP

☐

MKI

☐

Lifetime

Leave blank to match any value.

2^

Interworking

☒

Miscellaneous

Capability Negotiation

☐

Finish

### 7.12.3. Signaling Rules

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

Device: Avaya\_SBCE Alarms Incidents Status Logs Diagnostics Users Settings Help Log Out

## Session Border Controller for Enterprise

EMS Dashboard  
Device Management  
Backup/Restore  
System Parameters  
Configuration Profiles  
Services  
Domain Policies  
Application Rules  
Border Rules  
Media Rules  
Security Rules  
Signaling Rules  
Charging Rules  
End Point Policy Groups  
Session Policies  
TLS Management  
Network & Flows  
DMZ Services  
Monitoring & Logging

### Signaling Rules: default

Add Clone

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

General Requests Responses Request Headers Response Headers Signaling QoS UCID

**Inbound**

Requests	Allow
Non-2XX Final Responses	Allow
Optional Request Headers	Allow
Optional Response Headers	Allow

**Outbound**

Requests	Allow
Non-2XX Final Responses	Allow
Optional Request Headers	Allow
Optional Response Headers	Allow

**Content-Type Policy**

Enable Content-Type Checks ☒

Action	Allow	Multipart Action	Allow
Exception List		Exception List	

Edit

## 7.13.End Point Policy Groups

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

### 7.13.1. End Point Policy Group – Enterprise

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

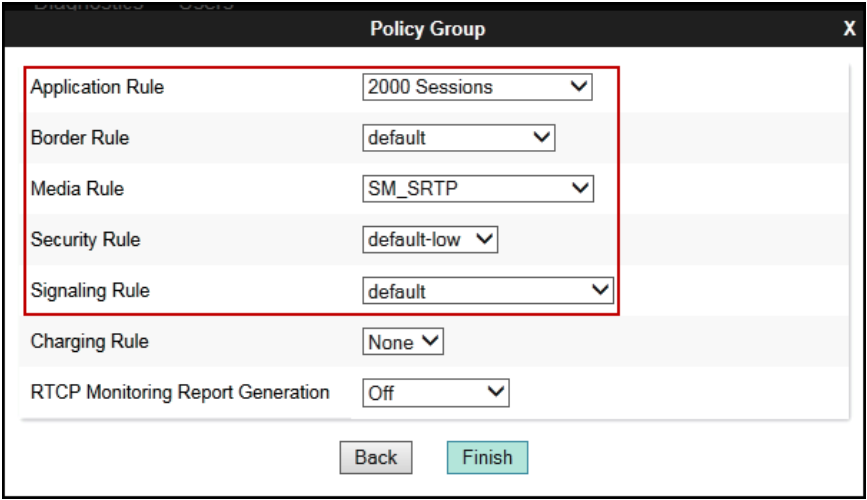
- Enter an appropriate name in the **Group Name** field.
- Click **Next**.



The screenshot shows a 'Policy Group' dialog box. It has a title bar with 'Policy Group' and a close button 'X'. Inside, there is a 'Group Name' label and a text input field containing 'Enterprise'. A red rectangle highlights the 'Group Name' label and the input field. Below the input field is a 'Next' button.

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

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



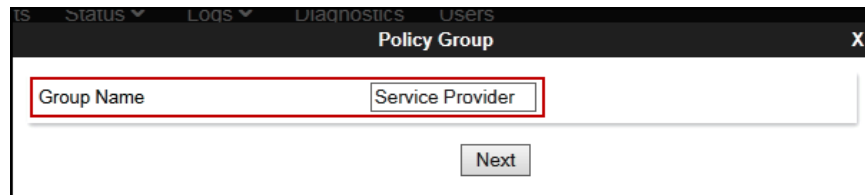
The screenshot shows the 'Policy Group' dialog box with several dropdown menus. A red rectangle highlights the first five rules: Application Rule (2000 Sessions), Border Rule (default), Media Rule (SM\_SRTP), Security Rule (default-low), and Signaling Rule (default). Below these are Charging Rule (None) and RTCP Monitoring Report Generation (Off). At the bottom are 'Back' and 'Finish' buttons. The 'Finish' button is highlighted in light blue.



### 7.13.2. End Point Policy Group – Service Provider

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

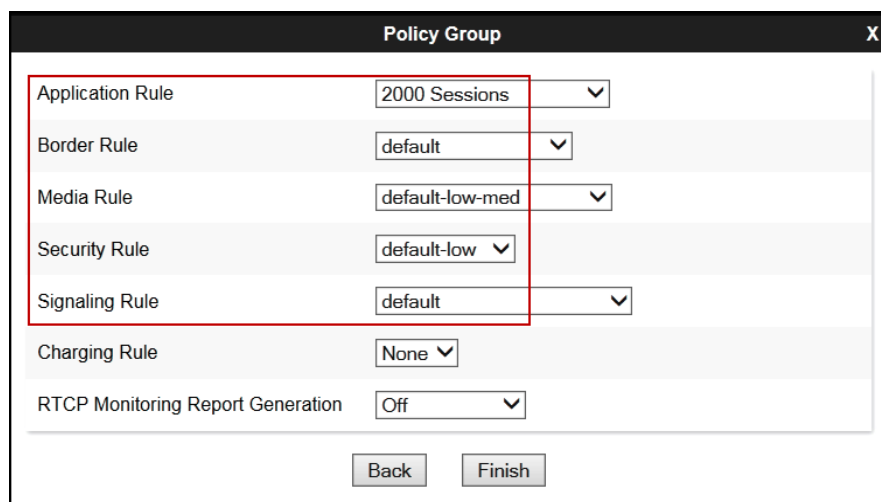
- Enter an appropriate name in the **Group Name** field (*Service Provider* was used).
- Click **Next**.



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

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

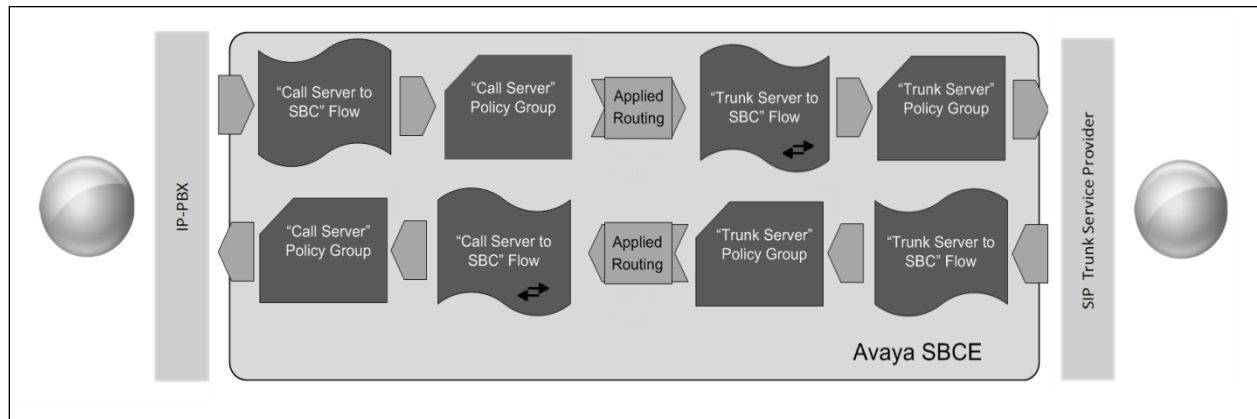
- **Application Rule:** *2000 Sessions* (Section 7.12.1).
- **Border Rule:** *default*.
- **Media Rule:** *default-low-med* (Section 7.12.2).
- **Security Rule:** *default-low*.
- **Signaling Rule:** *default* (Section 7.12.3).
- Click **Finish**.



The screenshot shows the "Policy Group" dialog box with several configuration options. A red rectangular box highlights the first five options: "Application Rule" (2000 Sessions), "Border Rule" (default), "Media Rule" (default-low-med), "Security Rule" (default-low), and "Signaling Rule" (default). Below these are "Charging Rule" (None) and "RTCP Monitoring Report Generation" (Off). At the bottom are "Back" and "Finish" buttons.

## 7.14.End Point Flows

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



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

### 7.14.1. End Point Flow – Enterprise

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

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

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

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

### 7.14.2. End Point Flow – Service Provider

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

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

## 8. Clearcom SIP Trunking Service Configuration

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

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

Clearcom will provide the following information:

- SIP Trunk registration credentials (user name, password, SIP domain).
- Fully Qualified Domain Name of the Clearcom SIP proxy server.
- DID numbers.
- Public DNS IP addresses.
- Supported codecs and order of preference.
- Any IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices (firewall).

## 9. Verification and Troubleshooting

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

### 9.1. General Verification Steps

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

### 9.2. Communication Manager Verification

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

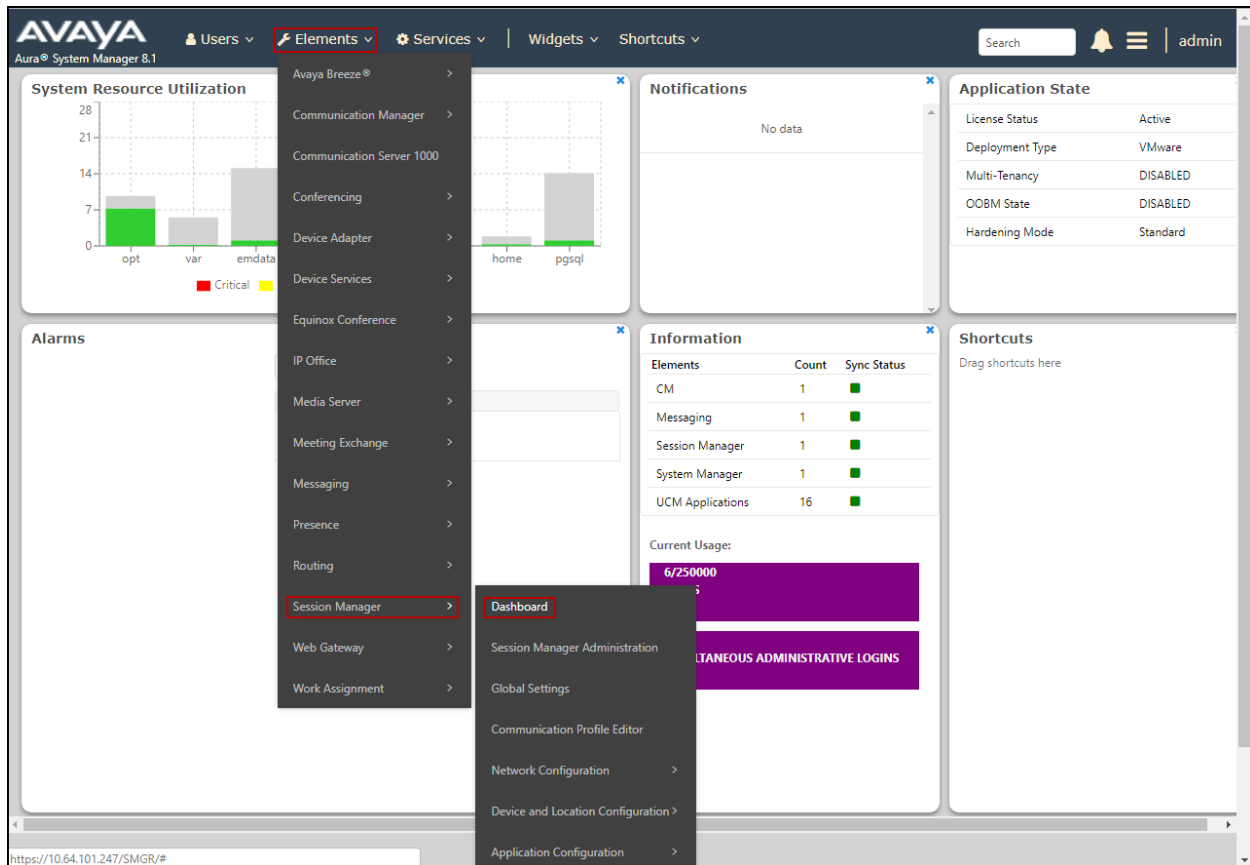
- **list trace station** <extension number>  
Traces calls to and from a specific station.
- **list trace tac** <trunk access code number>  
Trace calls over a specific trunk group.
- **status signaling-group** <signaling group number>  
Displays signaling group service state.
- **status trunk** <trunk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

Displays signaling and media information for an active call on a specific station.

### 9.3. Session Manager Verification

The Session Manager configuration may be verified via System Manager.

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



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

In the **Entity Monitoring** column, Session Manager shows that there are **1** alarms out of the **7** Entities defined.

**Session Manager Dashboard**

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

**Session Manager Instances**

Service State: Up Shutdown System: Disabled EASG: Disabled As of 11:32 AM

1 Item Show All Filter: Enable

	Session Manager	Type	Tests Pass	Alarms	Security Module	Service State	Entity Monitoring	Active Call Count	Registrations	Data Replication	User Data Storage Status	License Mode	EASG	Version
<input type="checkbox"/>	<a href="#">Session Manager</a>	Core	✓	0/0/0	Up	Accept New Service	1/7	0	0/0	✓	✓	Normal	Disabled	8.1.1.0.811021

Select : All, None

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

**Session Manager Entity Link Connection Status**

This page displays detailed connection status for all entity links from a Session Manager.

Status Details for the selected Session Manager:

**All Entity Links for Session Manager: Session Manager**

Summary View

7 Items Filter: Enable

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

Select : None

Other Session Manager useful verification and troubleshooting tools include:

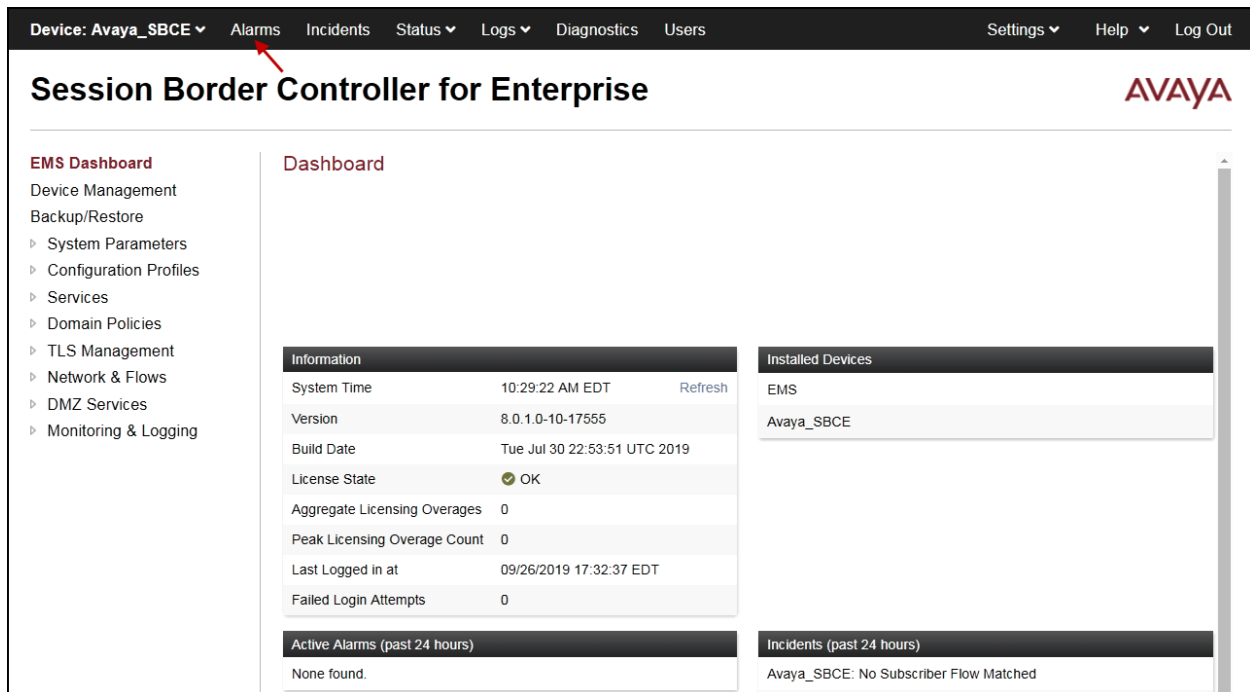
- **traceSM** – Session Manager command line tool for traffic analysis. Login to the Session Manager command line management interface to run this command.
- **Call Routing Test** – The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, from the System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



## 9.4. Avaya SBCE Verification

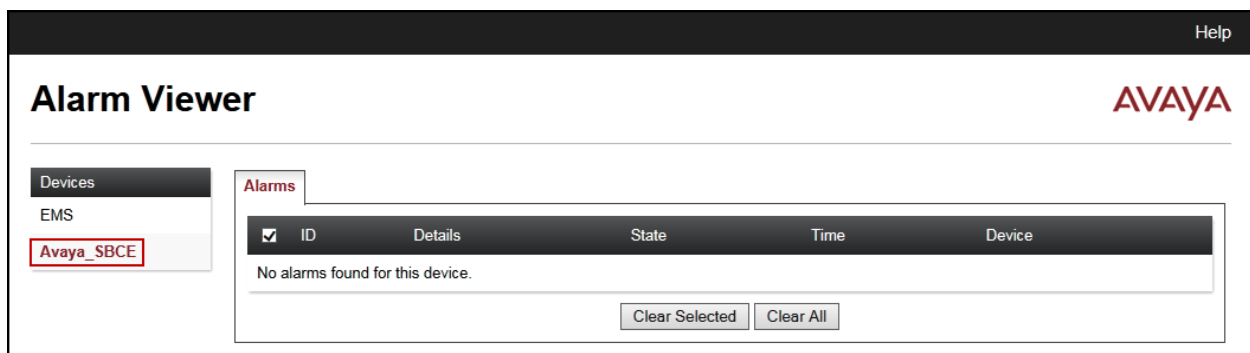
There are several links and menus located on the taskbar at the top of the screen of the web interface that can provide useful diagnostic or troubleshooting information.

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



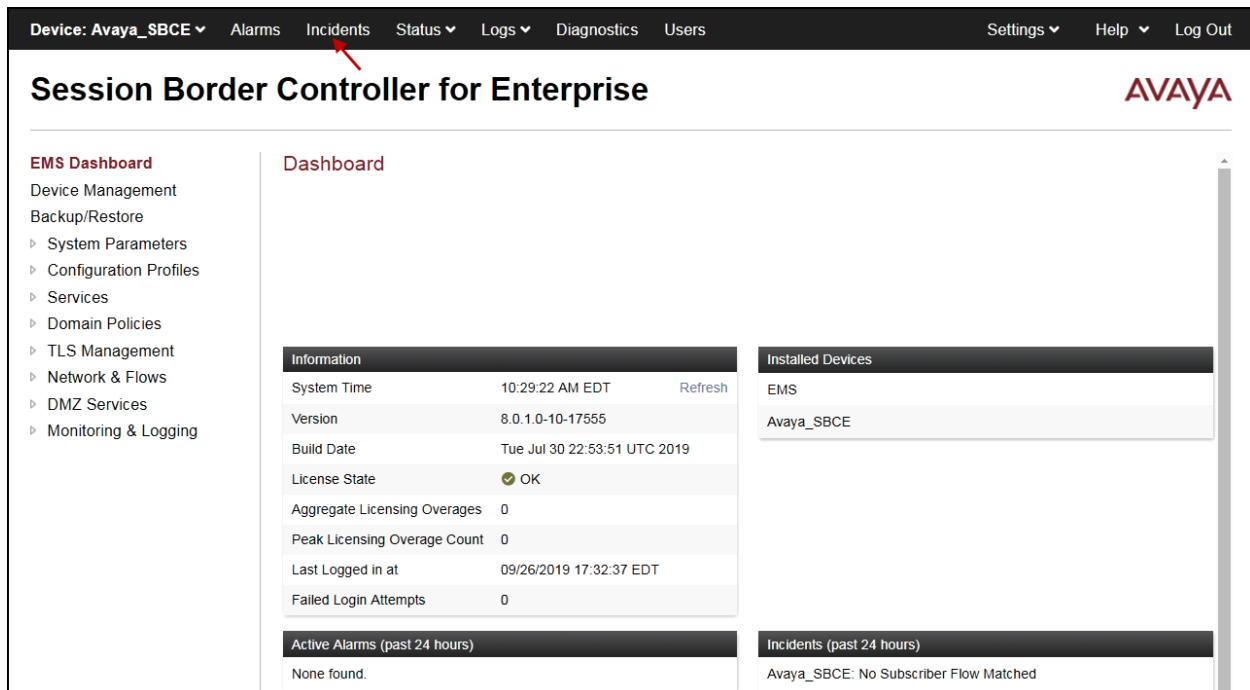
The screenshot shows the Avaya SBCE Dashboard. At the top, a navigation bar includes 'Device: Avaya\_SBCE', 'Alarms' (highlighted with a red arrow), 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main header reads 'Session Border Controller for Enterprise'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

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

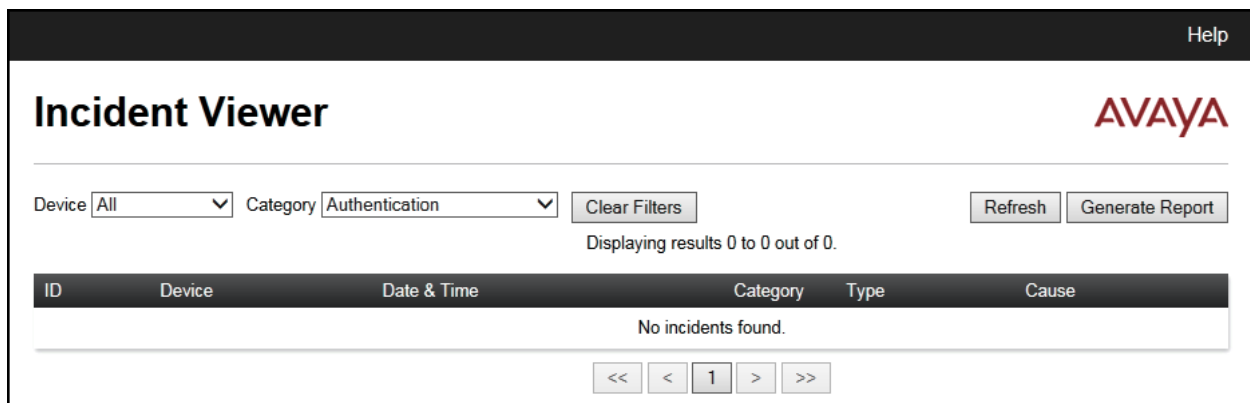


The screenshot shows the Avaya Alarm Viewer page. The top navigation bar includes 'Help'. The main header reads 'Alarm Viewer' with the Avaya logo. On the left, the 'Devices' menu lists 'EMS' and 'Avaya\_SBCE' (highlighted with a red box). The central 'Alarms' section features a table with columns: ID, Details, State, Time, and Device. Below the table, it states 'No alarms found for this device.' and includes 'Clear Selected' and 'Clear All' buttons.

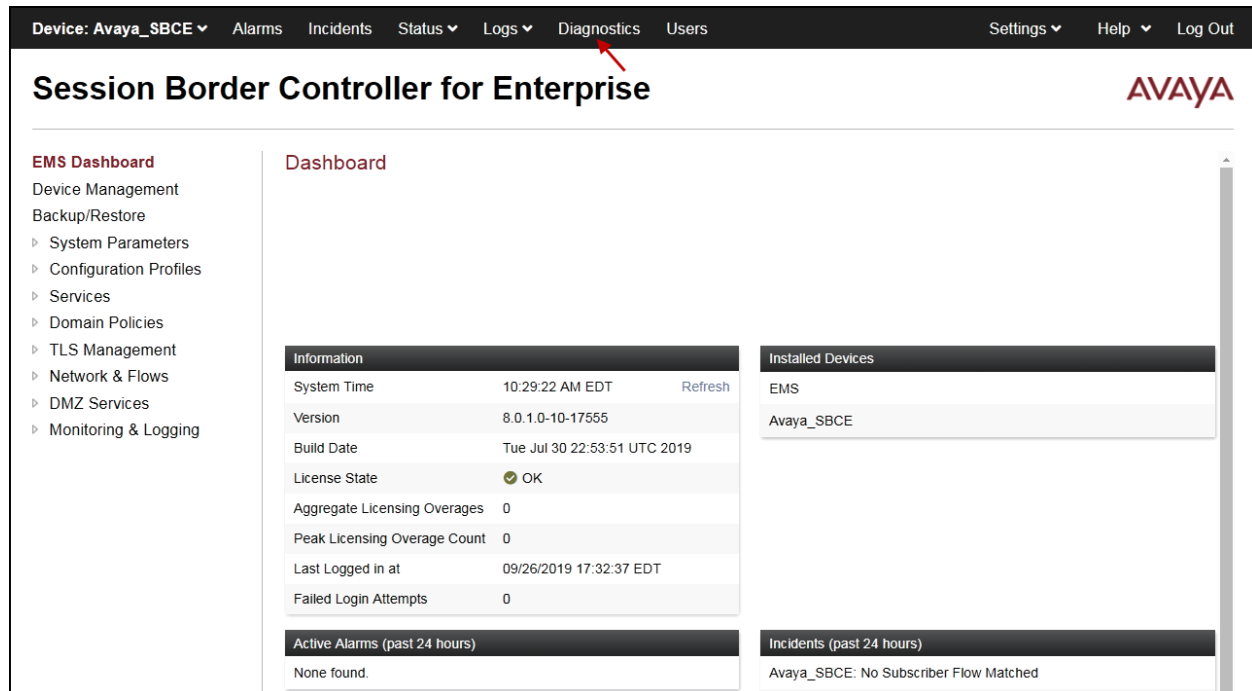
**Incidents** : Provides detailed reports of anomalies, errors, policies violations, etc.



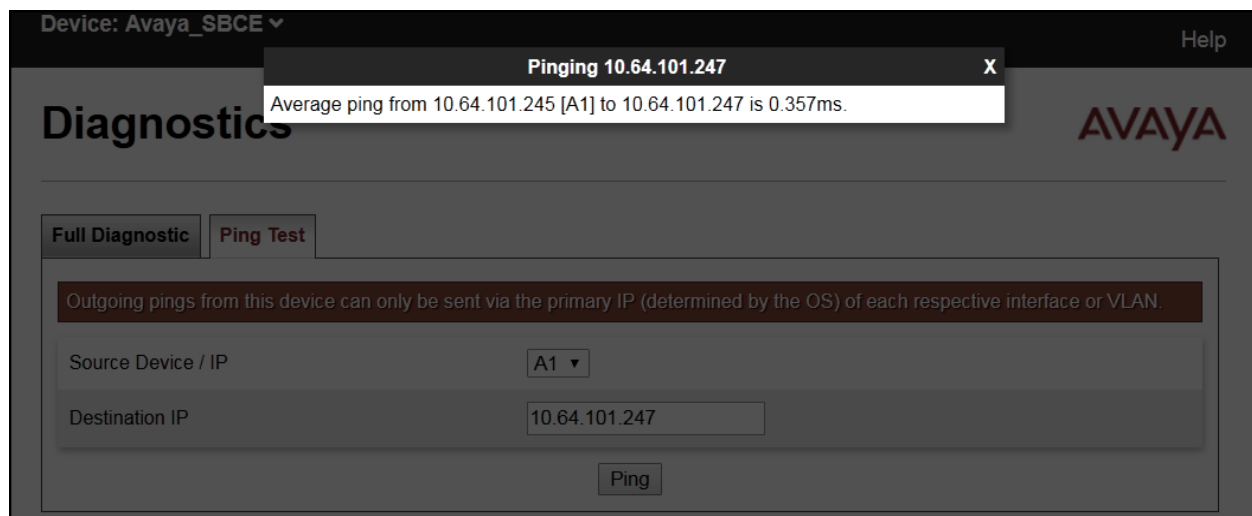
The following screen shows the Incident Viewer page.



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



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



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

**Device: Avaya\_SBCE** ▾ Alarms Incidents Status ▾ Logs ▾ Diagnostics Users

## Session Border Controller for Enterprise

EMS Dashboard

Device Management

Backup/Restore

▸ System Parameters

▸ Configuration Profiles

▸ Services

▸ Domain Policies

▸ TLS Management

▸ Network & Flows

▸ DMZ Services

▸ **Monitoring & Logging**

SNMP

Syslog Management

Debugging

**Trace**

Log Collection

DoS Learning

CDR Adjunct

Trace: Avaya\_SBCE

**Packet Capture**

Captures

**Packet Capture Configuration**

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

Start Capture

Clear

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

The screenshot displays the Avaya Session Border Controller for Enterprise (SBCE) web interface. The top navigation bar includes links for Device: Avaya\_SBCE, Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header reads "Session Border Controller for Enterprise" with the Avaya logo on the right. A left-hand navigation menu lists various management options, with "Monitoring & Logging" expanded and "Trace" highlighted. The main content area, titled "Trace: Avaya\_SBCE", features two tabs: "Packet Capture" and "Captures" (the active tab). Below the tabs is a table of captured files. The table has columns for "File Name", "File Size (bytes)", "Last Modified", and a "Delete" button. One file is listed: "Test\_Capture\_20200225113535.pcap" with a size of 139,264 bytes and a timestamp of February 25, 2020 11:35:46 AM EST. A "Refresh" button is located in the top right corner of the table area.

File Name	File Size (bytes)	Last Modified	
Test_Capture_20200225113535.pcap	139,264	February 25, 2020 11:35:46 AM EST	Delete

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

## 10. Conclusion

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

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

## 11. References

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

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

---

**©2020 Avaya Inc. All Rights Reserved.**

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at [devconnect@avaya.com](mailto:devconnect@avaya.com).