



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager Rel. 7.0, Avaya Aura® Session Manager Rel. 7.0 and Avaya Session Border Controller for Enterprise Rel. 7.0 to support Clearcom SIP Trunking Services – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking service for an enterprise solution consisting of Avaya Aura® Communication Manager Rel. 7.0, Avaya Aura® Session Manager Rel. 7.0, and Avaya Session Border Controller for Enterprise Rel. 7.0 to support Clearcom SIP Trunking Services.

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.

Clearcom SIP Trunking Service provides PSTN access via SIP trunks between the enterprise and Clearcom's 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.....	4
2.2.	Test Results	5
2.3.	Support.....	7
3.	Reference Configuration.....	8
4.	Equipment and Software Validated	10
5.	Configure Avaya Aura® Communication Manager.....	11
5.1.	Licensing and Capacity	12
5.2.	System Features.....	13
5.3.	IP Node Names.....	14
5.4.	Codecs	15
5.5.	IP Network Region.....	16
5.6.	Signaling Group	17
5.7.	Trunk Group.....	19
5.8.	Calling Party Information.....	23
5.9.	Inbound Routing.....	24
5.10.	Outbound Routing	25
6.	Configure Avaya Aura® Session Manager	28
6.1.	System Manager Login and Navigation.....	29
6.2.	Specify SIP Domain	30
6.3.	Add Location.....	31
6.4.	Adaptations.....	34
6.5.	SIP Entities	36
6.6.	Entity Links	40
6.7.	Routing Policies	43
6.8.	Dial Patterns	44
6.9.	Add/View Avaya Aura® Session Manager	47
7.	Configure Avaya Session Border Controller for Enterprise.....	49
7.1.	Log in Avaya SBCE.....	49
7.2.	Global Profiles.....	52
7.2.1.	Server Interworking Avaya-SM.....	52
7.2.2.	Server Interworking SP-General.....	55
7.2.3.	Signaling Manipulation.....	56
7.2.4.	Server Configuration.....	58
7.2.5.	Routing Profiles	67
7.2.6.	Topology Hiding.....	71
7.3.	Domain Policies	75
7.3.1.	Application Rules.....	75
7.3.2.	Media Rules	77
7.3.3.	Signaling Rules	77

7.3.4.	End Point Policy Groups.....	78
7.4.	Device Specific Settings.....	80
7.4.1.	Network Management.....	80
7.4.2.	Media Interface	82
7.4.3.	Signaling Interface	84
7.4.4.	End Point Flows.....	87
8.	Clearcom SIP Trunking Service Configuration	91
9.	Verification and Troubleshooting.....	92
9.1.	Troubleshooting	92
9.1.1.	Communication Manager.....	92
9.1.2.	Session Manager	92
9.1.3.	Avaya SBCE	93
10.	Conclusion	98
11.	References.....	99
12.	Appendix A: SigMa Script.....	100

1. Introduction

These Application Notes describe the steps required to configure Session Initiation Protocol (SIP) trunk service between the service provider Clearcom in Mexico and an Avaya SIP-enabled enterprise solution.

In the sample configuration, the Avaya SIP-enabled enterprise solution consists of an Avaya Aura® Communication Manager Rel. 7.0 (hereafter referred to as Communication Manager), Avaya Aura® Session Manager Rel. 7.0 (hereafter referred to as Session Manager), Avaya Session Border Controller for Enterprise Rel. 7.0 (hereafter referred to as Avaya SBCE), and various Avaya endpoints. This solution does not extend to configurations without the Avaya Session Border Controller for Enterprise or Avaya Aura® Session Manager.

During the interoperability testing, feature test cases were executed to ensure interoperability between Clearcom and Communication Manager.

Customers using an Avaya SIP-enabled enterprise solution with Clearcom SIP Trunking Service are able to place and receive PSTN calls via the SIP protocol. The converged network solution is an alternative to traditional analog trunks and/or PSTN trunks such as ISDN-PRI. This approach generally results in lower cost for the enterprise.

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

2. General Test Approach and Test Results

The general test approach was to simulate an enterprise site in the Avaya Solution & Interoperability Test Lab by connecting Communication Manager, Session Manager and the Avaya SBCE to Clearcom SIP Trunking Service via the public internet, as depicted in **Figure 1**.

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 for 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.

2.1. Interoperability Compliance Testing

To verify SIP trunk interoperability, the following areas were tested for compliance:

- 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 96x0 Series IP Deskphones (H.323), Avaya 96x1 Series IP Deskphones (H.323 and SIP), Avaya 2420

Digital Deskphones, Avaya one-X® Communicator soft phone (H.323 and SIP), Avaya Communicator for Windows (SIP) soft phone, analog Deskphones.

- Inbound and outbound PSTN calls to/from Remote Workers using Avaya 96x1 deskphones (SIP), Avaya one-X® Communicator (SIP) and Avaya Communicator for Windows (SIP).
- Outgoing calls to the PSTN were routed via Clearcom's 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.729A, G.711MU and G.11A (Clearcom's preferred codec order).
- No matching codecs.
- Voicemail and DTMF tone support (leaving and retrieving voice mail messages, etc.).
- Outbound Toll-Free calls, interacting with IVR (Interactive Voice Response systems).
- 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.
- 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.

Items not supported or not tested included the following:

- Inbound toll-free calls, outbound Toll-Free calls, 911 calls (emergency), "0" calls (Operator) and 0+10 digits calls (Operator Assisted) were not tested.
- The SIP REFER method for call redirection was not tested for reasons noted in **Section 2.2**.
- T.38 fax was not tested for reasons noted in **Section 2.2**.

2.2. Test Results

Interoperability testing of Clearcom SIP Trunk service with an Avaya SIP-enabled enterprise solution was completed successfully with the following observations/limitations.

- **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.
- **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 first DID number assigned to the SIP trunk (5528810001) was displayed, instead of “anonymous”.
- **Caller ID on incoming calls from U.S. based PSTN numbers:** Calls originating from PSTN telephones based in the U.S. to Communication Manager displayed “Unavailable”. During the compliance test, Clearcom provided a local PSTN test number in Mexico, a SIP softphone was registered to this local PSTN number and was used to originate and terminate local PSTN calls to and from Communication Manager. The correct Caller ID was displayed at the Communication Manager extensions when calling from this local PSTN number. This behavior is not necessarily indicative of a limitation of the combined Avaya/Clearcom solution, this seems to be the expected behavior for international calls from the U.S., which is ultimately controlled by the PSTN providers, it is listed here simply as an observation.
- **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 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 first DID number assigned to the SIP Trunk (5528810001), 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 first DID number assigned to the SIP Trunk (5528810001), regardless of the PSTN number being used to originate the call.
- **Fax Support:** T.38 fax is the fax protocol officially supported by Communication Manager on SIP trunks. During the tests, Clearcom responded with “488 Not Acceptable Here” to the re-INVITE messages sent by Communication Manager to make the change from voice to T.38, causing the call to drop. Even though it was possible during the tests to complete G.711 fax pass-through calls using a local test number in Mexico, G.711 fax pass-through is available in Communication Manager on a “best effort” basis, and it’s not guaranteed that it will work in every instance, thus G.711 fax pass-through is not recommended in Communication Manager.

- **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. (**Section 7.2.3**).
- **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. (**Section 7.2.3**).
- **SIP Trunk Registration:** For the most current software release of the Avaya SBCE, used during the compliant test (**7.0.0-21-6602**), a patch was required for the Avaya SBCE to solve an issue with call processing when SIP Trunk registration is used, this patch is only required if SIP Trunk registration is required by the Service Provider, which was the case with Clearcom. The patch ID that was applied was: **SBCE0000039**, the Product Support Notice (PSN) that contains the patch installation instructions was: **PSN004619**. The patch and PSN can be obtained from the Avaya Support web site. This patch may not be required in future software releases of the Avaya SBCE.
- **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 (**Section 7.2.3**).

2.3. Support

For support on Clearcom systems visit the corporate Web page at: <http://www.clearcom.mx/>

Avaya customers may obtain documentation and support for Avaya products by visiting <http://support.avaya.com>. Alternatively, in the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus.

3. Reference Configuration

Figure 1 below illustrates the test configuration used. The test configuration simulates an enterprise site with an Avaya SIP-enabled enterprise solution connected to the Clearcom SIP Trunk service through the public Internet.

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

- Avaya Aura® Communication Manager running on VMware (ESXi 5.5) platform.
- Avaya Aura® Session Manager running on VMware (ESXi 5.5) platform.
- Avaya Aura® System Manager running on VMware (ESXi 5.5) platform.
- Avaya Session Border Controller for Enterprise running on a Dell R210 V2 Server.
- Avaya Aura® Messaging running on VMware (ESXi 5.5) platform.
- Avaya Aura® Media Server running on VMware (ESXi 5.5) platform.
- Avaya G450 Media Gateway.
- Avaya 96x0-Series IP Deskphones (H.323).
- Avaya 96x1-Series IP Deskphones (H.323 and SIP).
- Avaya one-X® Communicator soft phones (H.323 and SIP).
- Avaya Communicator for Windows soft phone (SIP)
- Avaya 2420 Digital Deskphones.
- Analog Deskphones.
- Desktop PC running administration interfaces.

Located at the edge of the enterprise is the Avaya SBCE. It has a public side that connects to the public network and a private side that connects to the enterprise network. All SIP and RTP traffic entering or leaving the enterprise flow through the Avaya SBCE. This way, the Avaya SBCE can protect the enterprise against any SIP-based attacks. The Avaya SBCE provides network address translation at both the IP and SIP layers. The transport protocol between the Avaya SBCE and Clearcom across the public Internet is SIP over UDP. The transport protocol between the Avaya SBCE and Session Manager across the enterprise network is SIP over TCP. The transport protocol between Session Manager and Communication Manager across the enterprise network is SIP over TLS. Note that for ease of troubleshooting during the testing, the compliance test was conducted with the transport protocol set to **tcp** between Session Manager and Communication Manager.

A separate SIP trunk group was created between Communication Manager and Session Manager to carry the traffic to and from the service provider (two-way trunk group). To separate the codec settings required by the service provider from the codec used by the telephones, two IP network regions were used each with dedicated signaling groups.

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 and routing policies to determine the recipient (in this case Communication Manager) and on which link to send the call. Once the

call arrived at Communication Manager, further incoming call treatment, such as incoming digit translations and class of service restrictions are performed.

Outbound calls to the PSTN were first processed by Communication Manager for outbound feature treatment such as Automatic Route Selection (ARS) 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 and routing policies to determine the route to the Avaya SBCE for egress to Clearcom's network.

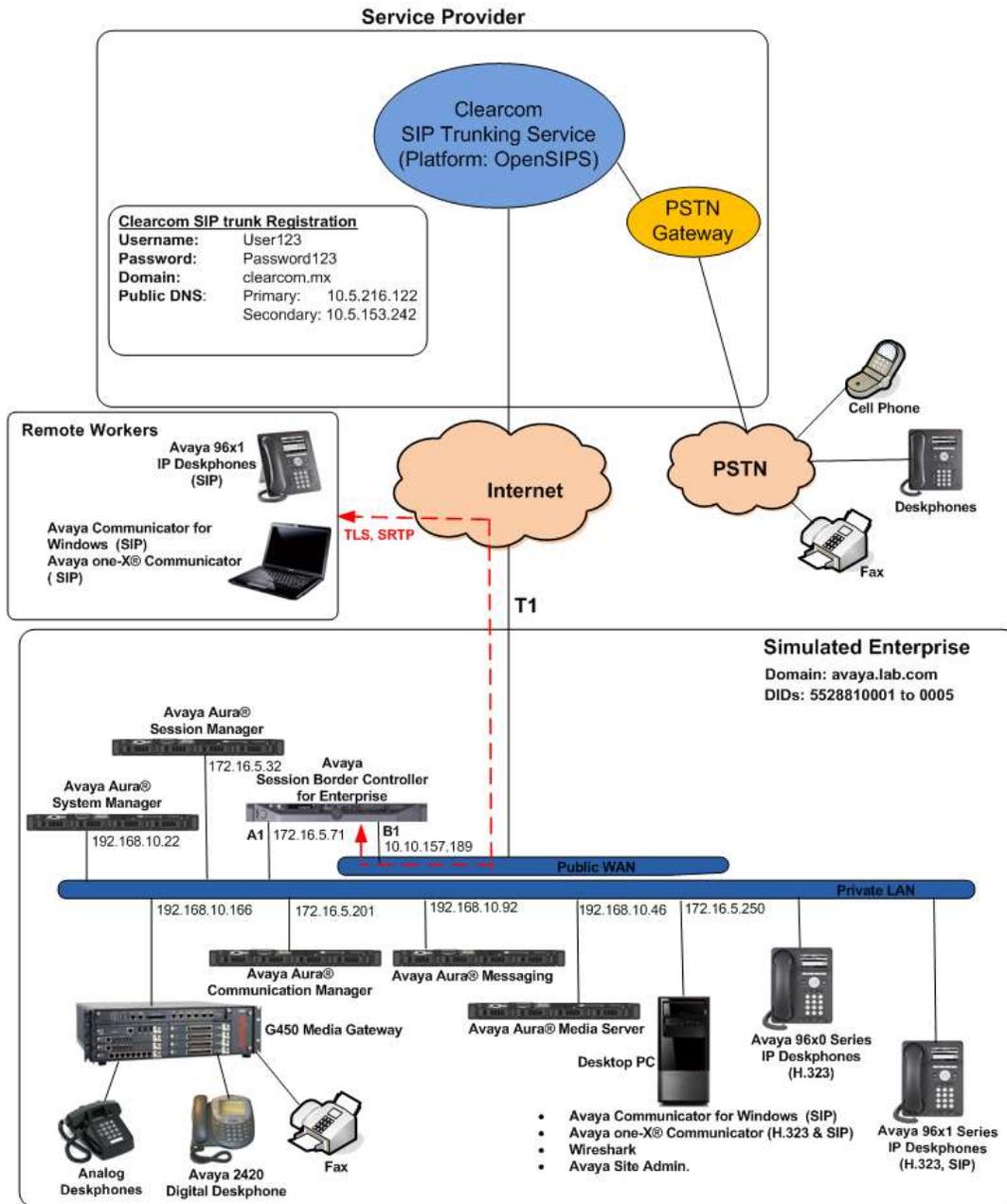


Figure 1: Avaya SIP-enabled Enterprise Solution and Clearcom SIP Trunking Service

4. Equipment and Software Validated

The following equipment and software were used for the compliance testing in the simulated enterprise:

Equipment/Software	Release/Version
Avaya	
Avaya Aura® Communication Manager running on VMware ESXi 5.5 platform	7.0.0.1.0 (00.0.441.0-22477)
Avaya Aura® Session Manager running on VMware ESXi 5.5 platform	7.0.0.0 (7.0.0.0.700007)
Avaya Aura® System Manager running on VMware ESXi 5.5 platform	7.0.0.0 Build No. 7.0.0.0.16266-7.0.9.912 Software Update Rev. No. 7.0.0.0.3929
G450 Gateway	37.19.0
Avaya Session Border Controller for Enterprise running on a DELL R210 V2 Server	7.0.0-21-6602
Avaya Aura® Media Server running on VMware ESXi 5.5 platform	7.7.0.226
Avaya Aura® Messaging running on VMware ESXi 5.5 platform	6.3.3 Service Pack 3 (MSG-03.0.141.0-348_0304)
Avaya Aura® Integrated Management Site Administrator	6.0.07
Avaya one-X® Communicator (SIP & H.323)	6.2.7.03-SP7
Avaya Communicator for Windows (SIP)	2.1.2.75
Avaya 96x0 Series IP Deskphones (H.323)	Avaya one-X® Desk phone Edition Version S3.250A
Avaya 96x1 Series IP Deskphones (H.323)	Avaya one-X® Deskphone H.323 Version 6.6029
Avaya 96x1 Series IP Deskphones (SIP)	Avaya one-X® Deskphone SIP Version 7.0.0.39
Avaya 2420 Series Digital Deskphone	--
Lucent Analog Deskphone	--
Clearcom	
OpenSIPS Softswitch	1.9
OpenSIPS Session Border Controller	1.9

Table 2 – Hardware and Software Components Tested

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

5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring Communication Manager. A SIP trunk is established between Communication Manager and Session Manager for use by signaling traffic to and from Clearcom. It is assumed that the general installation of Communication Manager, the Avaya G450 Media Gateway and the Avaya Aura® Media Server has been previously completed.

In configuring Communication Manager, various components such as ip-network-regions, signaling groups, trunk groups, etc. need to be selected or created for use with the SIP connection to the Service Provider. Unless specifically stated otherwise, any unused ip-network-region, signaling group, trunk group, etc. can be used for this purpose.

The Communication Manager configuration was performed using the Avaya Integrated Management Site Administrator. Some screens in this section have been abridged and highlighted for brevity and clarity in presentation. Note that the public IP addresses shown throughout these Application Notes have been edited so that the actual public IP addresses of the network elements are not revealed. Some screens captures 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 SIP trunks to the Service Provider. The example below shows one license with a capacity of **24000** trunks available and **122** 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 to add additional capacity.

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

IP PORT CAPACITIES                               USED
          Maximum Administered H.323 Trunks: 12000 10
          Maximum Concurrently Registered IP Stations: 18000 2
          Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
          Maximum Concurrently Registered IP eCons: 414 0
Max Concur Registered Unauthenticated H.323 Stations: 100 0
          Maximum Video Capable Stations: 41000 1
          Maximum Video Capable IP Softphones: 18000 7
          Maximum Administered SIP Trunks: 24000 122
Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
Maximum Number of DS1 Boards with Echo Cancellation: 522 0

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

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

```
display system-parameters customer-options Page 4 of 12
OPTIONAL FEATURES

Abbreviated Dialing Enhanced List? y           Audible Message Waiting? y
Access Security Gateway (ASG)? n               Authorization Codes? y
Analog Trunk Incoming Call ID? y              CAS Branch? n
A/D Grp/Sys List Dialing Start at 01? y       CAS Main? n
Answer Supervision by Call Classifier? y       Change COR by FAC? n
ARS? y                                         Computer Telephony Adjunct Links? y
ARS/AAR Partitioning? y                       Cvg Of Calls Redirected Off-net? y
ARS/AAR Dialing without FAC? n                DCS (Basic)? y
ASAI Link Core Capabilities? n                DCS Call Coverage? y
ASAI Link Plus Capabilities? n                DCS with Rerouting? y
Async. Transfer Mode (ATM) PNC? n             Digital Loss Plan Modification? y
Async. Transfer Mode (ATM) Trunking? n        DS1 MSP? y
ATM WAN Spare Processor? n                    DS1 Echo Cancellation? y
ATMS? y
Attendant Vectoring? y

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

5.2. System Features

Use the **change system-parameters features** command to set the **Trunk-to-Trunk Transfer** field to *all* to allow 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 this field set to *none*.

```
change 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.

```

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

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

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

INTERNATIONAL CALL ROUTING PARAMETERS
Local Country Code: ___
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 the Avaya server running Communication Manager (**procr**), and for Session Manager (**Lab-HG-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 172.16.5.71
Lab-HG-SM 172.16.5.32
MA-CM 192.168.10.12
default 0.0.0.0
media_server 192.168.10.46
msgserver 172.16.5.12
procr 172.16.5.201
procr6 ::
_____
_____
_____
_____
_____
_____
_____
_____
_____
_____
( 8 of 8 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. Clearcom supports G.729A, G.711MU and G.711A. Thus, these codecs were included in this set. Enter **G.729A**, **G.711MU** and **G.711A** in the **Audio Codec** column of the table; this is Clearcom's preferred codec order. Default values can be used for all other fields.

```
change ip-codec-set 2 Page 1 of 2
```

IP CODEC SET

Codec Set: 2

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

On **Page 2**, set the **Fax Mode** to *off* (T.38 fax is currently not supported by Clearcom, refer **Section 2.2**).

```
change ip-codec-set 2 Page 2 of 2
```

IP CODEC SET

Allow Direct-IP Multimedia? n

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

5.5. IP Network Region

Create a separate IP network region for the Service Provider trunk. 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*. This name appears in the “From” header of SIP messages originating from this IP region.
- Enter a descriptive name in the **Name** field.
- Enable **IP-IP Direct Audio** (shuffling) to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. Set both **Intra-region** and **Inter-region IP-IP Direct Audio** to *yes*. This is the default setting. Shuffling can be further restricted at the trunk level on the Signaling Group form.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.4**.
- Default values can be used for all other fields.

```
change ip-network-region 2                                     Page 1 of 20
                                     IP NETWORK REGION
Region: 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      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS      RSUP Enabled? n
H.323 Link Bounce Recovery? y
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. 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 example below 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	
dst rgn	codec set	direct WAN	WAN Units	WAN-BW-limits	Video Total Norm	Video Prio	Intervening Shr	Intervening Regions	Dyn CAC	I G A R	M t c e
1	2	u	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 SIP 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 for this purpose and was configured using the parameters highlighted below.

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*. This specifies Communication Manager will serve as an Evolution Server for Session Manager.
- Set the **Transport Method** to the recommended default value of *tls* (Transport Layer Security). Note that for ease of troubleshooting during testing, the compliance test was conducted with the **Transport Method** set to *tcp*. The transport method specified here is used between Communication Manager and Session Manager. The transport method used between Session Manager and the Avaya SBCE is specified as TCP in **Sections 6.6** and **7.2.4**. Lastly, the transport method between the Avaya SBCE and Clearcom is UDP. This is defined in **Section 7.2.4**.
- 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 **5070**. (For TCP, the well-known port value for SIP is 5060).

- 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 as Session Manager.
- Set the **Near-end Node Name** to *procr*. This node name maps to the IP address of the Avaya Server running Communication Manager as defined in **Section 5.3**.
- Set the **Far-end Node Name** to *Lab-HG-SM*. This node name maps to the IP address of Session Manager as defined in **Section 5.3**.
- Set the **Far-end Network Region** to the IP network region defined for the Service Provider in **Section 5.5**.
- Set the **Far-end Domain** to the domain of the enterprise.
- Set **Direct IP-IP Audio Connections** to *y*. This field will enable media shuffling on the SIP trunk allowing Communication Manager to redirect media traffic directly between the inside IP of the Avaya SBCE and the enterprise endpoint. If this value is set to *n*, then the Avaya Media Gateway will remain in the media path of all calls between the SIP trunk and the endpoint.
- Set the **DTMF over IP** field to *rtp-payload*. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- 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? y   Transport Method: tcp
Q-SIP? n
IP Video? n
Peer Detection Enabled? y Peer Server: SM             Enforce SIPS URI for SRTP? y
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
Near-end Node Name: procr                             Far-end Node Name: Lab-HG-SM
Near-end Listen Port: 5070                           Far-end Listen Port: 5070
                                                        Far-end Network Region: 2
Far-end Domain: avaya.lab.com
Incoming Dialog Loopbacks: eliminate                 Bypass IF IP Threshold Exceeded? n
DTMF over IP: rtp-payload                           RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3                 Direct IP-IP Audio Connections? y
Enable Layer 3 Test? y                             IP Audio Hairpinning? n
H.323 Station Outgoing Direct Media? n             Initial IP-IP Direct Media? n
                                                        Alternate Route Timer(sec): 6

```

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 21
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 (sec)** 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. Note that the value assigned to the **Preferred Minimum Session Refresh Interval (sec)** field is doubled and assigned to the “Min-SE” Header Field in SIP INVITE messages for calls originating from Communication Manager. Using the default setting of **600** seconds as in the example, the “Min-SE” Header Field would be populated for 1200 seconds in SIP INVITE messages originating from Communication Manager.

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

On **Page 3**, set the **Numbering Format** field to *Private*. This field specifies the format of the calling party number (CPN) sent to the far-end. Beginning with Communication Manager 6.0, public numbers are automatically preceded with a + sign when passed in the SIP “From”, “Contact”, “P-Asserted Identity” and “Diversion” headers. The addition of the “+” sign impacted caller ID presentation on outbound calls sent to Clearcom. Thus, the **Numbering Format** was set to *private* and the **Numbering Format** in the route pattern was set to *unk-unk* (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 enabled CPN block.

Default values were used for all other fields.

```
change trunk-group 2                               Page 3 of 21
TRUNK FEATURES
  ACA Assignment? n                               Measured: none
                                                    Maintenance Tests? y

  Numbering Format: private
                                                    UII 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
```

Page 4 was configured using the parameters highlighted below.

- Set the **Network Call Redirection** field to *n*. This setting directs Communication Manager **not** to use the SIP REFER method for transferring calls off-net to the PSTN, refer to **Section 2.2**.
- Set the **Send Diversion Header** field to *n*.
- Set the **Support Request History** field to *n*.
- Set the **Telephone Event Payload Type** to *101*. The value preferred by Clearcom.
- Set the **Convert 180 to 183 for Early Media** to *y*.
- Set the **Identity for Calling Party Display** to *P-Asserted-Identity*.

```
change trunk-group 2                                     Page 4 of 21
PROTOCOL VARIATIONS
Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
Send Transferring Party Information? n
Network Call Redirection? n
Send Diversion Header? n
Support Request History? n
Telephone Event Payload Type: 101
Convert 180 to 183 for Early Media? y
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
```


5.9. 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, as shown below. 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
```

Service/ Feature	Number Len	Number Digits	Del	Insert
public-ntwrk	10	5528810001	10	3041
public-ntwrk	10	5528810002	10	3042
public-ntwrk	10	5528810003	10	3044
public-ntwrk	10	5528810004	10	3045
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: *44_____
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. Refer to **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 to the Service Provider (as defined next).

```

change ars analysis 0                                         Page 1 of 2
                                ARS DIGIT ANALYSIS TABLE
                                Location: all                    Percent Full: 0

```

Dialed String	Total		Route Pattern	Call Type	Node Num	ANI Reqd
	Min	Max				
0	1	11	2	op		n
0	13	13	1	hnpa		n
00	2	2	deny	op		n
001	13	13	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
101xxx0	8	8	deny	op		n
101xxx0	18	18	deny	op		n
101xxx01	16	24	deny	iop		n
101xxx011	17	25	deny	intl		n
101xxx1	18	18	deny	fnpa		n
10xxx0	6	6	deny	op		n
10xxx0	16	16	deny	op		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 during the compliance test.

- **Pattern Name:** Enter a descriptive name.
- **Grp No:** Enter the outbound trunk group for the Service Provider. For the compliance test, trunk group 2 was used.
- **FRL:** Set the Facility Restriction Level (FRL) field to a level that allows access to this trunk for all users that require it. The value of 0 is the least restrictive level.
- **Numbering Format:** Set to *unk-unk*. All calls using this route pattern will use the private 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 3																					
Pattern Number: 2											Pattern Name: Serv. Provider																					
SCCAN? n											Secure SIP? n											Used for SIP stations? n										
Grp No	FRL	NPA	Pfx Mrk	Hop Lmt	Toll List	No. Del	Inserted Dgts				DCS/ QSIG	IXC																				
											Intw																					
1:	2										n	user																				
2:											n	user																				
3:											n	user																				
4:											n	user																				
5:											n	user																				
6:											n	user																				
											BCC	VALUE	TSC	CA-TSC	ITC	BCIE	Service/Feature	PARM	Sub	Numbering	LAR											
											0	1	2	M	4	W	Request						Dgts	Format								
1:	y	y	y	y	y	n	n				rest									unk-unk	none											
2:	y	y	y	y	y	n	n				rest										none											
3:	y	y	y	y	y	n	n				rest										none											
4:	y	y	y	y	y	n	n				rest										none											
5:	y	y	y	y	y	n	n				rest										none											
6:	y	y	y	y	y	n	n				rest										none											

Note: To save all Communication Manager provisioning changes, enter the command **save translations**.

6. Configure Avaya Aura® Session Manager

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

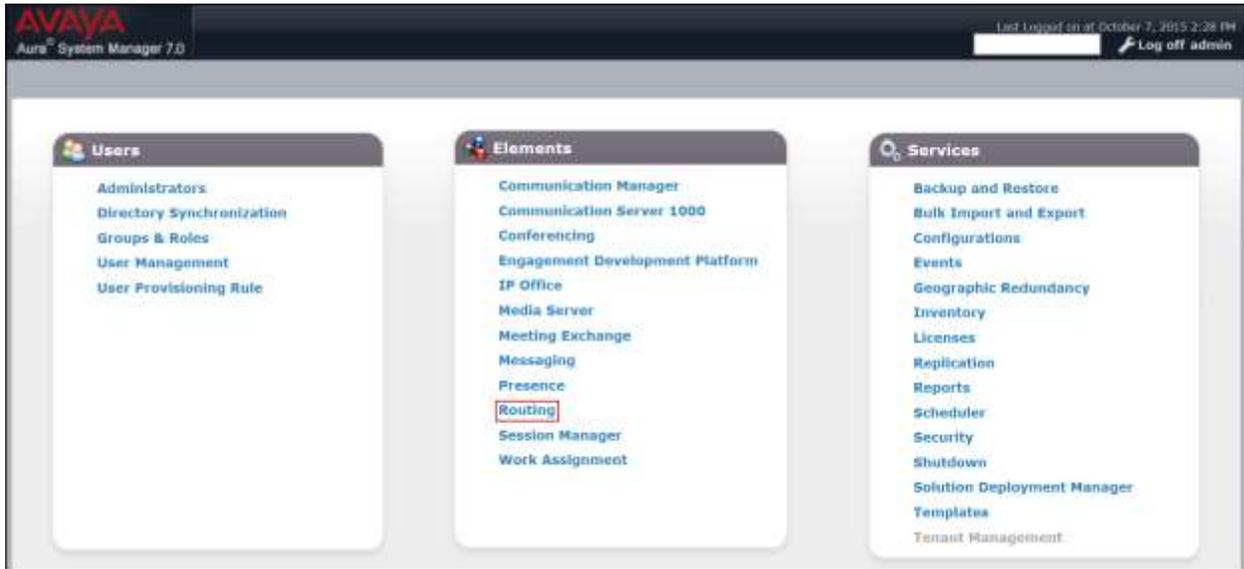
- SIP domain
- Logical/physical Location that can be occupied by SIP Entities
- Adaptation module to perform dial plan manipulation.
- SIP Entities corresponding to Communication Manager, the Avaya SBCE and Session Manager
- Entity Links, which define the SIP trunk parameters used by Session Manager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

It may not be necessary to create all the items above when configuring a connection to the Service Provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, Locations, Adaptations, SIP Entities, and Session Manager itself. However, each item should be reviewed to verify the configuration.

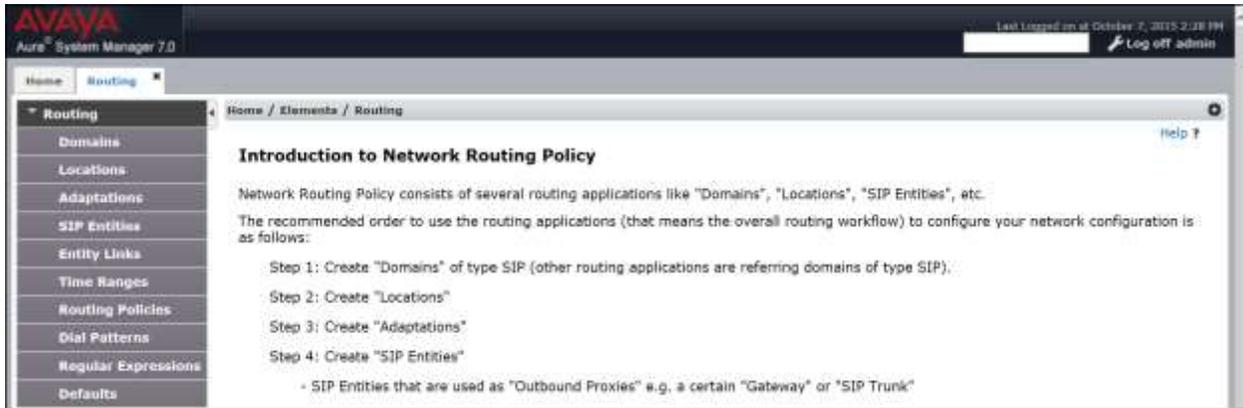
Note: Some of the default information in the screenshots that follow may have been cut out (not included) for brevity

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 (not shown). The screen shown below is then displayed. Click on **Routing**.



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



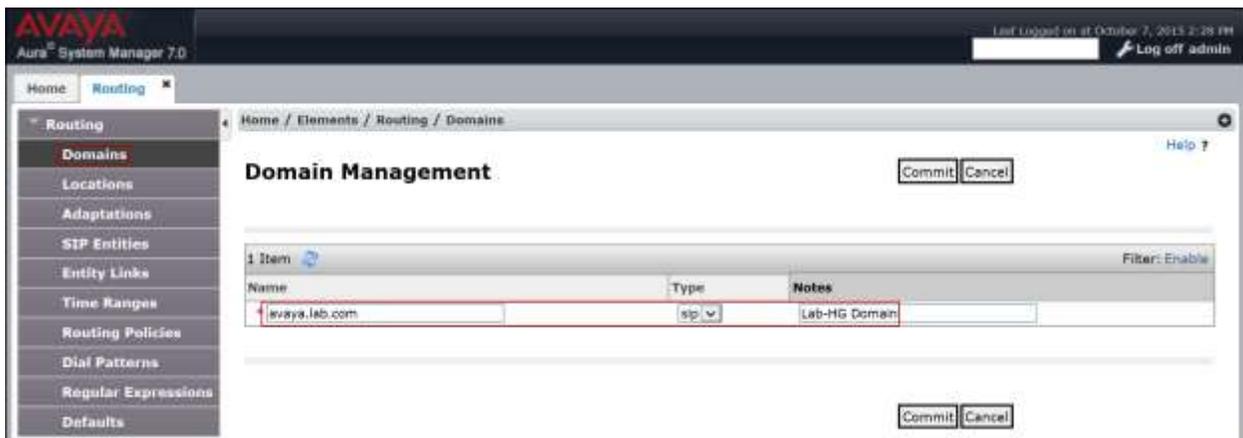
6.2. Specify SIP Domain

Create a SIP domain for each domain of which Session Manager will need to be aware in order to route calls. For the compliance test the enterprise domain *avaya.lab.com* was used.

To add a domain, 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 show).

The screen below shows the entry for the enterprise domain *avaya.lab.com*.



6.3. Add Location

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

In the **General** section, enter the following values. Use default values for all remaining fields:

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

The screen below shows the **HG Session Manager** location. This location will be assigned later to the SIP Entity corresponding to Session Manager.

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

- General:** The 'Name' field is populated with 'HG Session Manager'. The 'Notes' field is empty.
- Dial Plan Transparency in Survivable Mode:** The 'Enabled' checkbox is unchecked. The 'Listed Directory Number' and 'Associated CM SIP Entity' fields are empty.
- Overall Managed Bandwidth:** The 'Managed Bandwidth Units' dropdown is set to 'Kbit/sec'. The 'Total Bandwidth' and 'Multimedia Bandwidth' fields are empty. The 'Audio Calls Can Take Multimedia Bandwidth' checkbox is checked.
- Per-Call Bandwidth Parameters:** The 'Maximum Multimedia Bandwidth (Intra-Location)' and 'Maximum Multimedia Bandwidth (Inter-Location)' fields are both set to '1000 Kbit/Sec'. The 'Minimum Multimedia Bandwidth' field is set to '64 Kbit/Sec'. The 'Default Audio Bandwidth' dropdown is set to '80 Kbit/sec'.
- Alarm Threshold:** The 'Overall Alarm Threshold' and 'Multimedia Alarm Threshold' dropdowns are both set to '80 %'. The 'Latency before Overall Alarm Trigger' and 'Latency before Multimedia Alarm Trigger' fields are both set to '5 Minutes'.
- Location Pattern:** A table with 'Add' and 'Remove' buttons is shown. It contains one entry: 'IP Address Pattern' with an empty 'Notes' field. The table shows '0 Items' and a 'Filter: Enable' option.

At the bottom right of the form, there are 'Commit' and 'Cancel' buttons.

The following screen shows the **HG Communication Manager** location. This location will be assigned later to the SIP Entity corresponding to Communication Manager.

The screenshot displays the Avaya Aura System Manager 7.0 interface. The top navigation bar includes the Avaya logo, the text 'Aura System Manager 7.0', and a user status indicator 'Last Logged on at: October 7, 2015 2:28 PM' with a 'Log off admin' link. The left sidebar contains a menu with options: Routing, Domains, Locations (highlighted), 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 a breadcrumb path 'Home / Elements / Routing / Locations'. The 'Name' field is set to 'HG Communication Manager'. Below this are sections for 'Dial Plan Transparency in Survivable Mode', 'Overall Managed Bandwidth', 'Per-Call Bandwidth Parameters', 'Alarm Threshold', and 'Location Pattern'. The 'Location Pattern' section shows a table with one entry: 'IP Address Pattern'. The interface includes 'Commit' and 'Cancel' buttons at the top right and bottom right.

Location Details

Commit Cancel

General

Name: HG Communication Manager

Notes:

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:

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location): 1000 Kbit/Sec

Maximum Multimedia Bandwidth (Inter-Location): 1000 Kbit/Sec

Minimum Multimedia Bandwidth: 64 Kbit/Sec

Default Audio Bandwidth: 80 Kbit/sec

Alarm Threshold

Overall Alarm Threshold: 80 %

Multimedia Alarm Threshold: 80 %

Latency before Overall Alarm Trigger: 5 Minutes

Latency before Multimedia Alarm Trigger: 5 Minutes

Location Pattern

Add Remove

0 Items Filter: Enable

IP Address Pattern Notes

Commit Cancel

The following screen shows the **HG ASBCE** location. This location will be assigned later to the SIP Entity corresponding to the Avaya SBCE.

The screenshot displays the Avaya Aura System Manager 7.0 interface. The top navigation bar includes the Avaya logo, the text 'Aura System Manager 7.0', and a user session indicator 'Last Logged on at: October 7, 2015 2:28 PM' with a 'Log off admin' link. The left sidebar contains a menu with 'Routing' selected, and sub-items: Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Location Details' and shows the configuration for a location named 'HG ASBCE'. The 'Name' field is highlighted with a red box. The configuration is organized into several sections: 'General' (Name, Notes), 'Dial Plan Transparency in Survivable Mode' (Enabled checkbox, Listed Directory Number, Associated CM SIP Entity), 'Overall Managed Bandwidth' (Managed Bandwidth Units dropdown, Total Bandwidth, Multimedia Bandwidth, Audio Calls Can Take Multimedia Bandwidth checkbox), 'Per-Call Bandwidth Parameters' (Maximum Multimedia Bandwidth for Intra-Location and Inter-Location, Minimum Multimedia Bandwidth, Default Audio Bandwidth), 'Alarm Threshold' (Overall Alarm Threshold, Multimedia Alarm Threshold, Latency before Overall Alarm Trigger, Latency before Multimedia Alarm Trigger), and 'Location Pattern' (Add/Remove buttons, 0 items, Filter: Enable, IP Address Pattern, Notes). 'Commit' and 'Cancel' buttons are present at the top right and bottom right of the configuration area.

6.4. Adaptations

In order to improve interoperability with third party elements, Session Manager 7.0 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: Alert-Info, P-Charging-Vector, AV-Global-Session-ID, AV-Correlation-ID, P-AV-Message-id, P-Location, and Endpoint-View. 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.

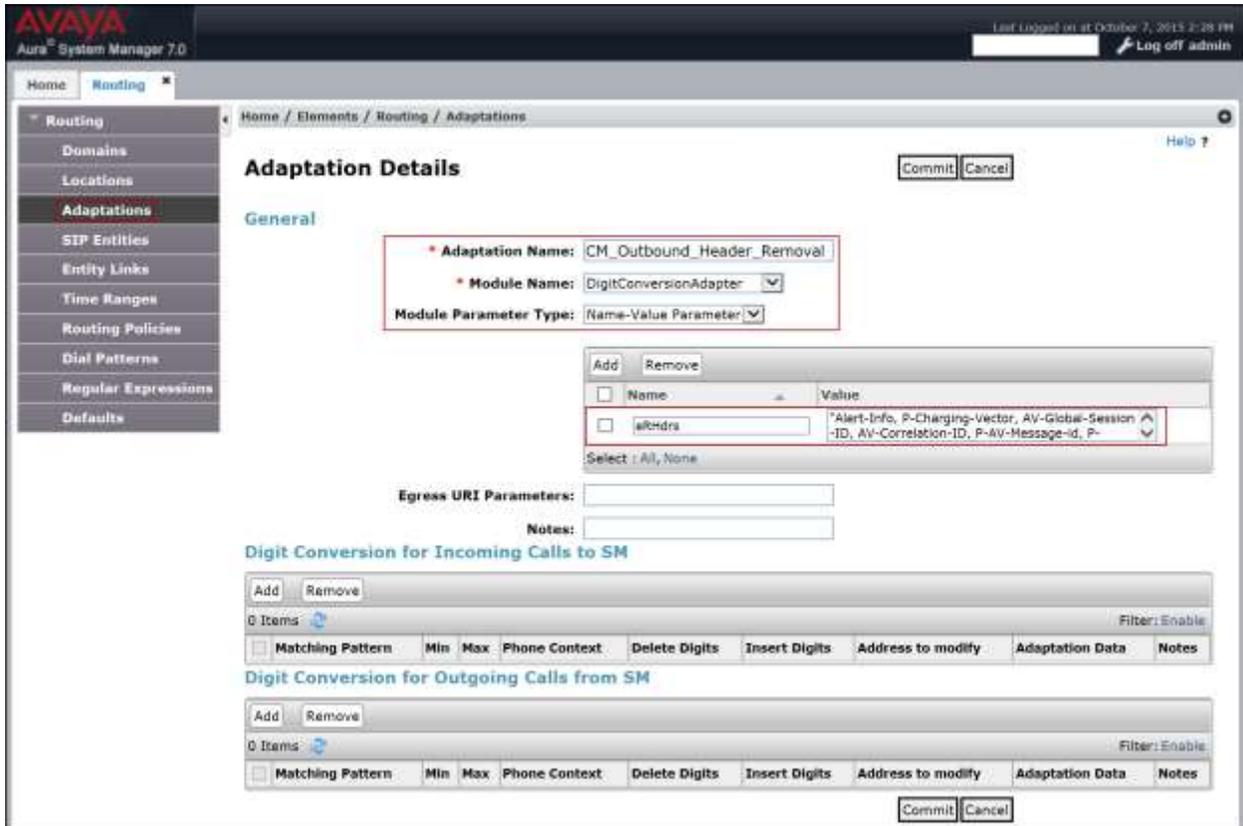
Navigate to **Routing** → **Adaptations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **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.

- **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*”

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 with their default values.



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-hand 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 interface that is used for SIP signaling.
- **Type:** Enter *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 applicable, select the **Adaptation Name**.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Select the time zone for the location above.

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

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

- **Port:** Port number on which the Session Manager will listen for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain:** The domain used for the enterprise.
- Click **Commit** to save.

The following screen shows the addition of the Session Manager SIP entity. The name **HG Session Manager**, the IP address of the Session Manager signaling interface, the Location **HG Session Manager** created in **Section 6.3** and the **Time Zone** were used.

For the compliance test, only two Ports were used:

- **5060** with **TCP** for connecting to the Avaya SBCE.
- **5070** with **TCP** for connecting to Communication Manager.

The screenshot displays the Avaya Aura System Manager 7.0 interface for configuring a SIP Entity. The 'SIP Entity Details' page is shown with the following configuration:

- Name:** HG Session Manager
- FQDN or IP Address:** 172.16.5.32
- Type:** Session Manager
- Notes:** Security Module
- Location:** HG Session Manager
- Outbound Proxy:** (empty)
- Time Zone:** America/New_York
- Credential name:** (empty)

The 'SIP Link Monitoring' section is set to 'Use Session Manager Configuration'. The 'Listen Ports' section shows a table of 10 items:

Listen Ports	Protocol	Default Domain	Notes
5060	TCP	avaya.lab.com	
5060	UDP	avaya.lab.com	
5061	TLS	avaya.lab.com	
5062	TCP	avaya.lab.com	
5065	TLS	avaya.lab.com	
5070	TCP	avaya.lab.com	
5080	TCP	avaya.lab.com	
5081	TCP	avaya.lab.com	
5085	UDP	avaya.lab.com	
5086	TCP	avaya.lab.com	

The 'SIP Responses to an OPTIONS Request' section is currently empty (0 items).

The following screen shows the addition of the Communication Manager SIP Entity.

A separate SIP entity for Communication Manager is required in order to route traffic from Communication Manager to the Service Provider.

The name **HG CM Trunk 2**, the IP of the Avaya Server running Communication Manager, the **Type** of **CM** for Communication Manager, the Location **HG Communication Manager** created in **Section 6.3** and the **Time Zone** were used.

The screenshot displays the Avaya Aura System Manager 7.0 interface for configuring a SIP Entity. The breadcrumb trail is Home / Elements / Routing / SIP Entities. The page title is 'SIP Entity Details' with 'Commit' and 'Cancel' buttons. The 'General' section contains the following fields:

- Name: HG CM Trunk 2
- FQDN or IP Address: 172.16.5.201
- Type: CM
- Notes: For Service Provider Calls
- Adaptation: (empty)
- Location: HG Communication Manager
- Time Zone: America/New_York
- SIP Timer B/F (in seconds): 4
- Credential name: (empty)
- Securable:
- Call Detail Recording: none
- Loop Detection Mode: Off
- SIP Link Monitoring: Use Session Manager Configuration
- Supports Call Admission Control:
- Shared Bandwidth Manager:
- Primary Session Manager Bandwidth Association: (empty)
- Backup Session Manager Bandwidth Association: (empty)

The 'SIP Responses to an OPTIONS Request' section features a table with the following structure:

Response Code & Reason Phrase	Mark Entity Up/Down	Notes
0 Items		

The page also includes 'Commit' and 'Cancel' buttons at the bottom.

The following screen shows the addition of the SIP entity for the Avaya SBCE.

The name **HG ASBCE**, the inside IP address of the Avaya SBCE, the **Type** of **Other**, the adaptation **CM_Outbound_Header_Removal** created in **Section 6.4**, the location **HG ASBCE** created in **Section 6.3** and the **Time Zone** were used.

Note: **Type: Other** was used during the testing; **SIP Trunk** could have been used instead.

AVAYA
Aura System Manager 7.0

Home Routing

Home / Elements / Routing / SIP Entities

SIP Entity Details

Commit Cancel Help ?

General

* Name: HG ASBCE

* FQDN or IP Address: 172.16.5.71

Type: Other

Notes: HG ASBCE

Adaptation: CM_Outbound_Header_Removal

Location: HG ASBCE

Time Zone: America/New_York

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

Credential name:

Securable:

Call Detail Recording: none

CommProfile Type Preference:

Loop Detection

Loop Detection Mode: Off

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

Supports Call Admission Control:

Shared Bandwidth Manager:

Primary Session Manager Bandwidth Association:

Backup Session Manager Bandwidth Association:

SIP Responses to an OPTIONS Request

Add Remove

0 Items Filter: Enable

Response Code & Reason Phrase	Mark Entity Up/Down	Notes
-------------------------------	---------------------	-------

Commit Cancel

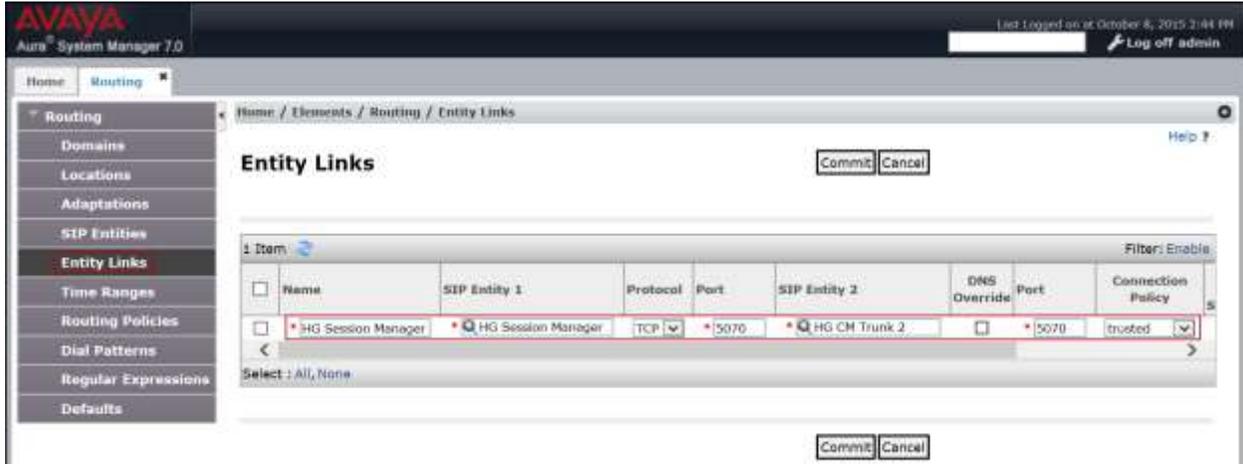
6.6. Entity Links

A SIP trunk between Session Manager and a telephony system is described by an Entity Link. Two entity links were created; one to Communication Manager and one to the Avaya SBCE, to be used only for Service Provider traffic. To add an entity link, navigate to **Routing → Entity Links** in the left-hand 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.
- **Protocol:** Select the transport protocol used for this link. For Communication Manager this was matched to the **Transport Method** defined on the Communication Manager signaling group in **Section 5.6**. For the Avaya SBCE, this was matched to the **Transport** defined on the **Server Configuration** for Session Manager (Call Server) in **Section 7.2.4**.
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end. For Communication Manager, this was matched to the **Far-end Listen Port** defined on the Communication Manager signaling group in **Section 5.6**. For the Avaya SBCE, this was matched to the **Port** defined on the **Server Configuration** for Session Manager (Call Server) in **Section 7.2.4**.
- **SIP Entity 2:** Select the name of the other system. For Communication Manager or the Avaya SBCE select the respective SIP Entity defined in **Section 6.5**.
- **Port:** Port number on which the other system will receive SIP requests from Session Manager. For Communication Manager, this was matched to the **Near-end Listen Port** defined on the Communication Manager signaling group in **Section 5.6**. For the Avaya SBCE, this was matched to the **TCP Port** defined for the private **Signaling Interface** on the Avaya SBCE in **Section 7.4.3**.
- **Connection Policy:** Select *Trusted*.
- Click **Commit** to save.

The following screens illustrate the entity links to Communication Manager and to the Avaya SBCE. It should be noted that in a customer environment the entity link to Communication Manager would normally use TLS. For the compliance test, TCP was used to aid in troubleshooting since the signaling traffic is not encrypted.

The following screen shows the entity link to Communication Manager:



The following screen shows the entity link to the Avaya SBCE:



The following screen shows the list of the newly added entity links. Note that only the highlighted entity links were created for the compliance test, and are the ones relevant to these Application Notes.

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy	Deny New Service	Notes
<input type="checkbox"/>	HG Session Manager AAC 5060 TCP	HG Session Manager	TCP	5060	AAC	<input type="checkbox"/>	5060	trusted	<input type="checkbox"/>	AAC Entity Link
<input type="checkbox"/>	HG Session Manager Acme Packet sIpI 5060 TCP	HG Session Manager	TCP	5060	Acme Packet sIpI	<input type="checkbox"/>	5060	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	HG Session Manager CS1K7.6 5085 UDP	HG Session Manager	UDP	5085	CS1K7.6	<input type="checkbox"/>	5085	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	HG Session Manager HG ASBCE 5060 TCP	HG Session Manager	TCP	5060	HG ASBCE	<input type="checkbox"/>	5060	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	HG Session Manager HG CM Trunk 1 5061 TCP	HG Session Manager	TLS	5061	HG CM Trunk 1	<input type="checkbox"/>	5061	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	HG Session Manager HG CM Trunk 2 5070 TCP	HG Session Manager	TCP	5070	HG CM Trunk 2	<input type="checkbox"/>	5070	trusted	<input type="checkbox"/>	

6.7. Routing Policies

Routing Policies describe the conditions under which calls are routed to the SIP entities specified in **Section 6.5**. Two routing policies must be added: one for Communication Manager and one for the Avaya SBCE. To add a routing policy, navigate to **Routing → Routing Policies** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). The following screen is displayed. Fill in the following:

In the **General** section, enter the following values. Use default values for all remaining fields:

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

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

- Click **Commit** to save.

The following screen shows the routing policy for Communication Manager:

The screenshot displays the Avaya System Manager 7.0 interface for configuring a Routing Policy. The left navigation pane shows 'Routing Policies' selected. The main content area is titled 'Routing Policy Details' and includes 'Commit' and 'Cancel' buttons. The 'General' section contains the following fields:

- Name:** To HG CM Trunk 2
- Disabled:**
- Retries:** 0
- Notes:** Inbound calls to HG CM Trunk 2

The 'SIP Entity as Destination' section features a table with the following data:

Select	Name	FQDN or IP Address	Type	Notes
<input type="checkbox"/>	HG CM Trunk 2	172.16.5.201	CM	For Service Provider Calls

The following screen shows the routing policy for the Avaya SBCE:



6.8. Dial Patterns

Dial Patterns are needed to route calls through Session Manager. For the compliance test, dial patterns were needed to route calls from Communication Manager to Clearcom 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-hand 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. Use default values for all remaining fields:

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

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

- Click **Commit** to save.

Examples of dial patterns used for the compliance testing are shown below.

The first example shows dial pattern **28**, with destination SIP Domain of **-ALL-**, Originating Location Name **HG Communication Manager** and Routing Policy name **To HG ASBCE**. This dial pattern was used for local outbound calls in Mexico.

Note: The SIP Domain was set to **-ALL-** since dial pattern 28 is shared among multiple SIP Domains in the Avaya lab, SIP Domain **Avaya.lab.com** could have been used instead.

Dial Pattern Details

General

Pattern: 28
 Min: 8
 Max: 8

Emergency Call:
 Emergency Priority: 1
 Emergency Type:
 SIP Domain: -ALL-
 Notes: Outbound to Clearcom Test Softphone

Originating Locations and Routing Policies

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/> HG Communication Manager		To HG ASBCE	0	<input type="checkbox"/>	HG ASBCE	For outbound calls to Service Provider
<input type="checkbox"/> MA Communication Manager	HP DL360	Outbound to MA ASBCE	0	<input type="checkbox"/>	MA_SBCE	Outbound to MA_SBCE

The following dial pattern used for the compliance testing was for inbound calls to the enterprise. It uses dial pattern **55** matching the first two digits of the DID numbers assigned to Communication Manager. This dial pattern was configured with the destination SIP Domain of **all**, Originating Location Name **HG ASBCE**, and Routing Policy name **To HG CM Trunk 2**.

Dial Pattern Details

General

Pattern: 55
 Min: 10
 Max: 10

Emergency Call:
 Emergency Priority:
 Emergency Type:
 SIP Domain: -ALL-
 Notes: Clearcom Incoming

Originating Locations and Routing Policies

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/> HG ASBCE		To HG CM Trunk 2	0	<input type="checkbox"/>	HG CM Trunk 2	Inbound calls to HG CM Trunk 2
<input type="checkbox"/> HA SBCE	Avaya SBCE 6.3	Incoming to HA CM trunk 2	0	<input type="checkbox"/>	HA_CM Trunk 2	

Note: The SIP Domain was set to -ALL- since dial pattern 55 is shared among multiple SIP Domains in the Avaya lab, SIP Domain **Avaya.lab.com** could have been used instead.

Note: The same procedure should be followed to add other required dial patterns.

6.9. Add/View Avaya Aura® Session Manager

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

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

- **SIP Entity Name:** Select the SIP Entity created for Session Manager.
- **Description:** Add a brief description (optional).
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

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

- **SIP Entity IP Address:** Should be filled in automatically based on the SIP Entity Name. Otherwise, enter IP address of the Session Manager signaling interface.
- **Network Mask:** Enter the network mask corresponding to the IP address of the Session Manager signaling interface.
- **Default Gateway:** Enter the IP address of the default gateway for Session Manager.

Use default values for the remaining fields.

- Click **Save** (not shown).

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

The screenshot displays the Avaya Aura System Manager 7.0 web interface. The top navigation bar includes the Avaya logo, the text 'Aura System Manager 7.0', and a user profile section with the name 'admin' and a 'Log off admin' link. The breadcrumb trail reads 'Home / Elements / Session Manager / Session Manager Administration'. The main content area is titled 'View Session Manager' and features a 'Return' button. Below the title, there are navigation links for 'General', 'Security Module', 'Monitoring', 'CDR', 'Personal Profile Manager (PPM)', 'Connection Settings', and 'Event Server', along with 'Expand All' and 'Collapse All' options. The 'General' section is expanded and contains the following configuration details:

SIP Entity Name	HG Session Manager
Description	Lab-HG SM
Management Access Point Host Name/IP	172.16.5.31
Direct Routing to Endpoints	Enable
Maintenance Mode	<input type="checkbox"/>

The 'Security Module' section is also expanded and contains the following configuration details:

SIP Entity IP Address	172.16.5.32
Network Mask	255.255.255.0
Default Gateway	172.16.5.254
Call Control PHB	46

At the bottom of the Security Module section, there is a '*SIP Firewall Configuration' dropdown menu set to 'Rule Set for HG Session Manager'.

7. Configure Avaya Session Border Controller for Enterprise

This section describes the required configuration of the Avaya SBCE to connect to Clearcom's SIP Trunking service.

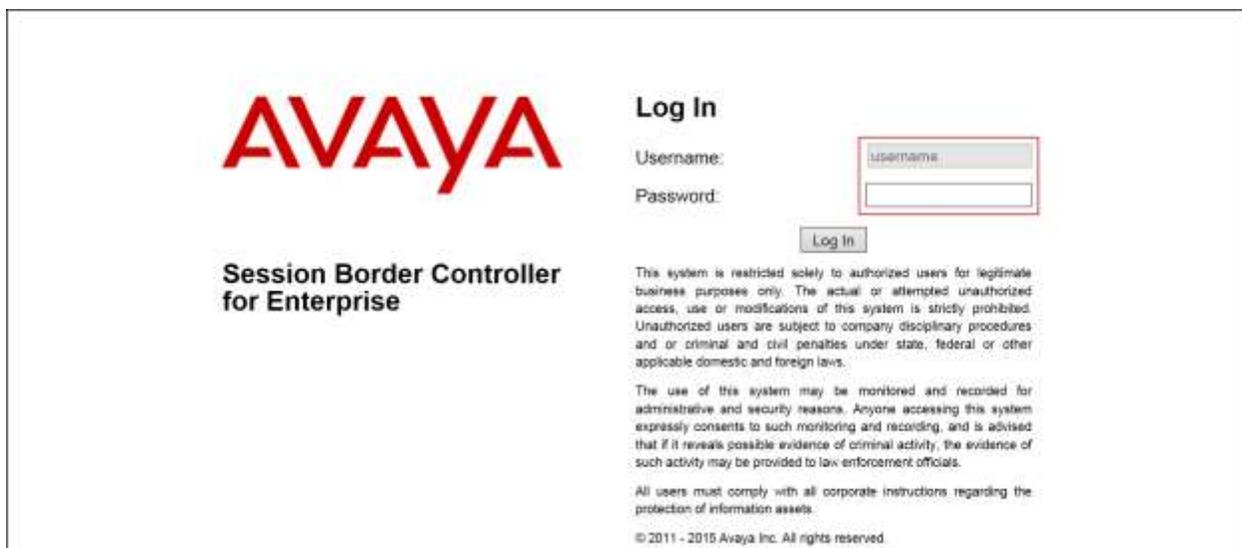
It is assumed that the Avaya SBCE was provisioned and is ready to be used; the configuration shown here is accomplished using the Avaya SBCE web interface.

Note: In the following pages, and for brevity in these Application Notes, not every provisioning step will have a screenshot associated with it. Some of the default information in the screenshots that follow may have been cut out (not included) for brevity.

7.1. Log in Avaya SBCE

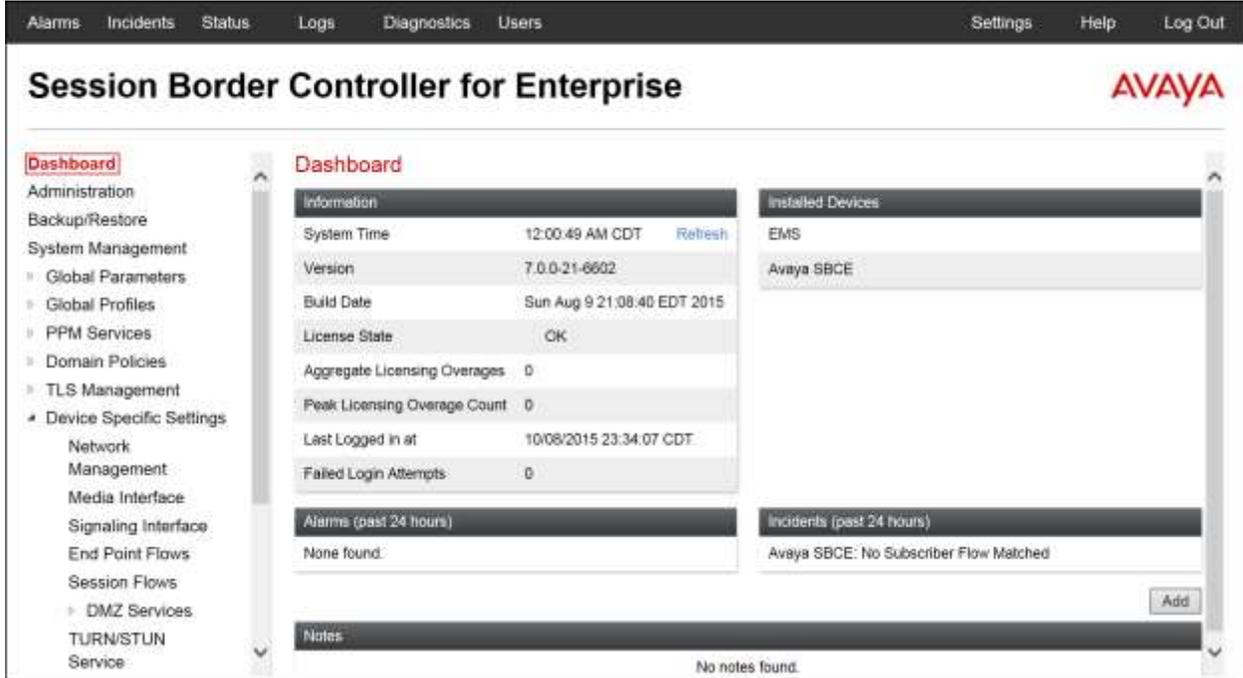
Use a web browser to access the Avaya SBCE web interface, enter `https://<ip-addr>/sbc` in the address field of the web browser, where `<ip-addr>` is the management IP address of the Avaya SBCE.

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

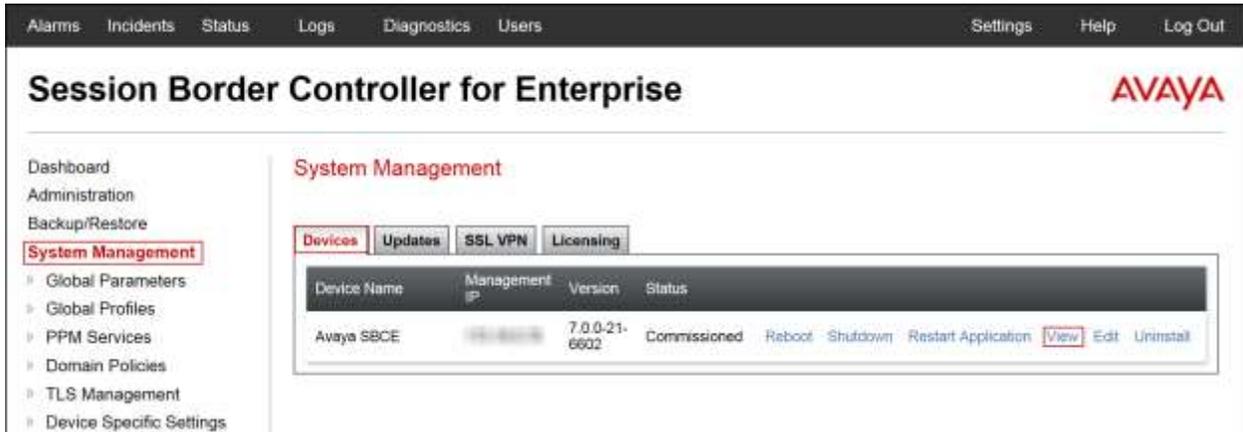


The screenshot shows the Avaya Session Border Controller for Enterprise login interface. On the left, the Avaya logo is displayed in red, with the text "Session Border Controller for Enterprise" below it. On the right, the "Log In" section contains a "Username:" label followed by a text input field containing the placeholder text "username", a "Password:" label followed by a password input field, and a "Log In" button. Below the login fields, there is a disclaimer: "This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use or modifications of this system is strictly prohibited. Unauthorized users are subject to company disciplinary procedures and/or criminal and civil penalties under state, federal or other applicable domestic and foreign laws." Below the disclaimer, another line of text states: "The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials." At the bottom, a copyright notice reads: "© 2011 - 2015 Avaya Inc. All rights reserved."

The **Dashboard** main page will appear as shown below.



To view the system information that has been configured during installation, navigate to **System Management**. A list of installed devices is shown in the right pane. In the compliance testing, a single Device Name **Avaya SBCE** was already added.



To view the network configuration assigned to the Avaya SBCE, click **View** as shown on the screen above. The **System Information** window is displayed as shown below.

The **System Information** screen shows **Network Configuration**, **DNS Configuration** and **Management IP** information provided during installation and corresponds to **Figure 1**. The **Box Type** was set to **SIP** and the **Deployment Mode** was set to **Proxy**. Default values were used for all other fields.

System Information: Avaya SBCE

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 <small>Requested: 2000</small>	2000
Advanced Sessions <small>Requested: 2000</small>	2000
Scopia Video Sessions <small>Requested: 500</small>	500
CES Sessions <small>Requested: 0</small>	0
Encryption	<input checked="" type="checkbox"/>

Network Configuration

IP	Public IP	Netmask	Gateway	Interface
172.16.5.71	172.16.5.71	255.255.255.0	172.16.5.254	A1
10.10.157.189	10.10.157.189	255.255.255.192	10.10.157.129	B1

DNS Configuration

Primary DNS	10.5.216.122
Secondary DNS	10.5.153.242
DNS Location	DMZ
DNS Client IP	10.10.157.189

Management IP(s)

IP	10.10.157.189
----	---------------

On the previous screen, note that the **A1** interface corresponds to the inside interface (Private Network side) and **B1** interface corresponds to the outside interface (Public Network side) of the Avaya SBCE. 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. Refer to **Figure 1** for the IP addresses for both the A1 and B1 interfaces on the Avaya SBCE.

DNS server configuration can be entered or modified as needed, by clicking **Edit** on the **System Management/Devices** tab shown on the previous page. Under **DNS Settings**, enter the IP addresses of the **Primary** and **Secondary** DNS servers. During the compliance test, public DNS servers were used, and the IP address corresponding to the public interface of the Avaya SBCE was selected from the **DNS Client IP** scroll down menu, as shown on the screen below. Click **Finish** (not shown) when done.

Edit Device: Avaya SBCE X

Address and interface changes must be made in Network Management.

Any changes to the management network on this device will reboot the device.

General Settings

Appliance Name X

Device Settings

High Availability (HA)

DNS Settings

Primary
Ex: 202.201.192.1

Secondary
Optional, Ex: 202.201.192.1

DNS Client IP ▼

Network Settings

IMPORTANT! – During the Avaya SBCE installation, the Management interface, (labeled “M1”), of the Avaya SBCE must be provisioned on a different subnet than either of the Avaya SBCE private and public network interfaces (e.g., A1 and B1). If this is not the case, contact your Avaya representative to have this resolved.

7.2. Global Profiles

The Global Profiles Menu, on the left navigation pane, allows the configuration of parameters across all Avaya SBCE appliances.

7.2.1. Server Interworking Avaya-SM

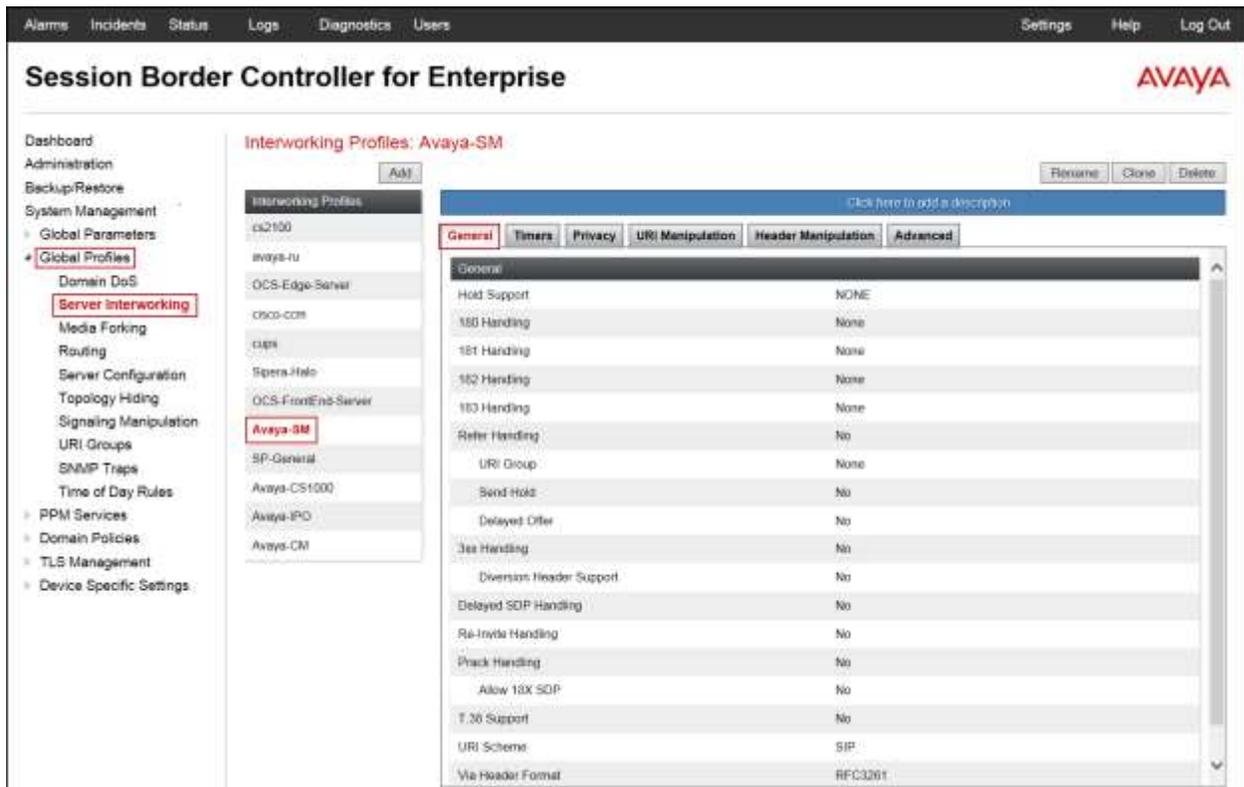
Interworking Profile features are configured to facilitate interoperability of implementations between enterprise SIP-enabled solutions and different SIP trunk Service Providers.

Several profiles have been already pre-defined and they populate the list under **Interworking Profiles** on the screen below. If a different profile is needed, a new Interworking Profile can be created, or an existing default profile can be modified or “cloned”. Since directly modifying a default profile is generally not recommended, for the test configuration the default **avaya-ru** profile was duplicated, or “cloned”, and then modified to meet specific requirements for the enterprise SIP-enabled solution.

On the left navigation pane, select **Global Profiles → Server Interworking**. From the **Interworking Profiles** list, select **avaya-ru**. Click **Clone** on top right of the screen.

Enter the new profile name in the **Clone Name** field, the name of **Avaya-SM** was chosen in this example. Click **Finish** (not shown).

The following screen capture shows the **General** tab of the newly created **Avaya-SM** Server Interworking Profile.



The following screen capture shows the **Advanced** tab of the newly created **Avaya-SM** Server Interworking Profile.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main header shows 'Session Border Controller for Enterprise' and the 'AVAYA' logo.

The left sidebar contains a navigation menu with categories like 'Dashboard', 'Administration', 'System Management', and 'Global Profiles'. Under 'Global Profiles', 'Server' and 'Interworking' are highlighted.

The main content area is titled 'Interworking Profiles: Avaya-SM'. It features a list of profiles on the left, including 'cs2100', 'avaya-nr', 'OCS-Edge-Server', 'cisco-com', 'cups', 'Spera-Halo', 'OCS-FrontEnd-Server', 'Avaya-SM', 'SP-General', 'Avaya-CS1000', 'Avaya-IPO', and 'Avaya-CM'. The 'Avaya-SM' profile is selected and highlighted.

The configuration details for the 'Avaya-SM' profile are shown in a table with tabs for 'General', 'Timers', 'Privacy', 'URI Manipulation', 'Header Manipulation', and 'Advanced'. The 'Advanced' tab is active, showing the following settings:

Setting	Value
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
DTMF	
DTMF Support	None

Buttons for 'Add', 'Rename', 'Clone', 'Delete', and 'Edit' are visible in the interface.

7.2.2. Server Interworking SP-General

A second Server Interworking profile named **SP-General** was created for the Service Provider.

On the left navigation pane, select **Global Profiles** → **Server Interworking**. From the **Interworking Profiles** list, select **Add**.

Enter the new profile name (not shown), the name of **SP-General** was chosen in this example. Click **Next**:

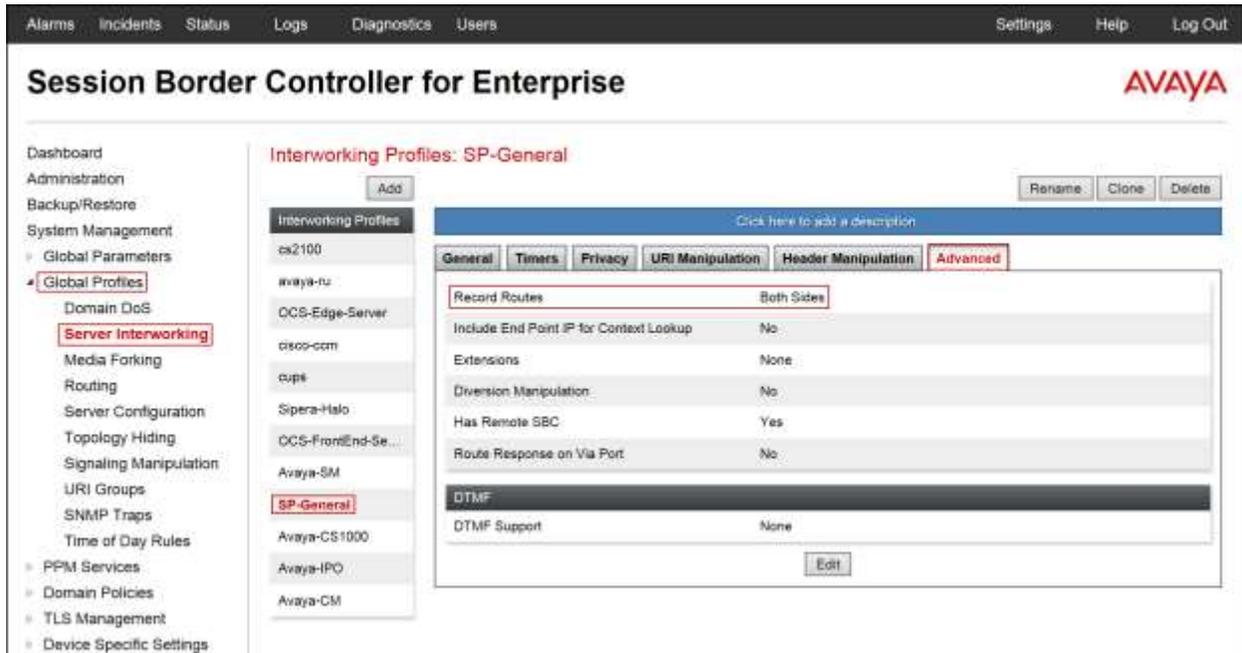
- Leaving other fields with their default values, click **Next** until the Advanced tab is reached, check **Both Sides** then click **Finish** on the Advanced tab.

The following screen capture shows the **General** tab of the newly created **SP-General** Server Interworking Profile.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The main content area is titled "Interworking Profiles: SP-General" and shows a list of profiles on the left and a configuration table on the right. The "General" tab is selected, showing various settings for the profile.

Setting	Value
Hold Support	NONE
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	No
URI Group	None
Send Host	No
Delayed Offer	No
Tex Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
Re-Invite Handling	No
Reack Handling	No
Allow 18X SDP	No
T.38 Support	No
URI Scheme	SIP
We Header Format	RFC3261

The following screen capture shows the **Advanced** tab of the newly created **SP-General** Server Interworking Profile.



7.2.3. 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 [6] in the **References** section for more information on this topic.

Sigma scripts were created during the compliance test to correct the following interoperability issues (refer to **Section 2.2**):

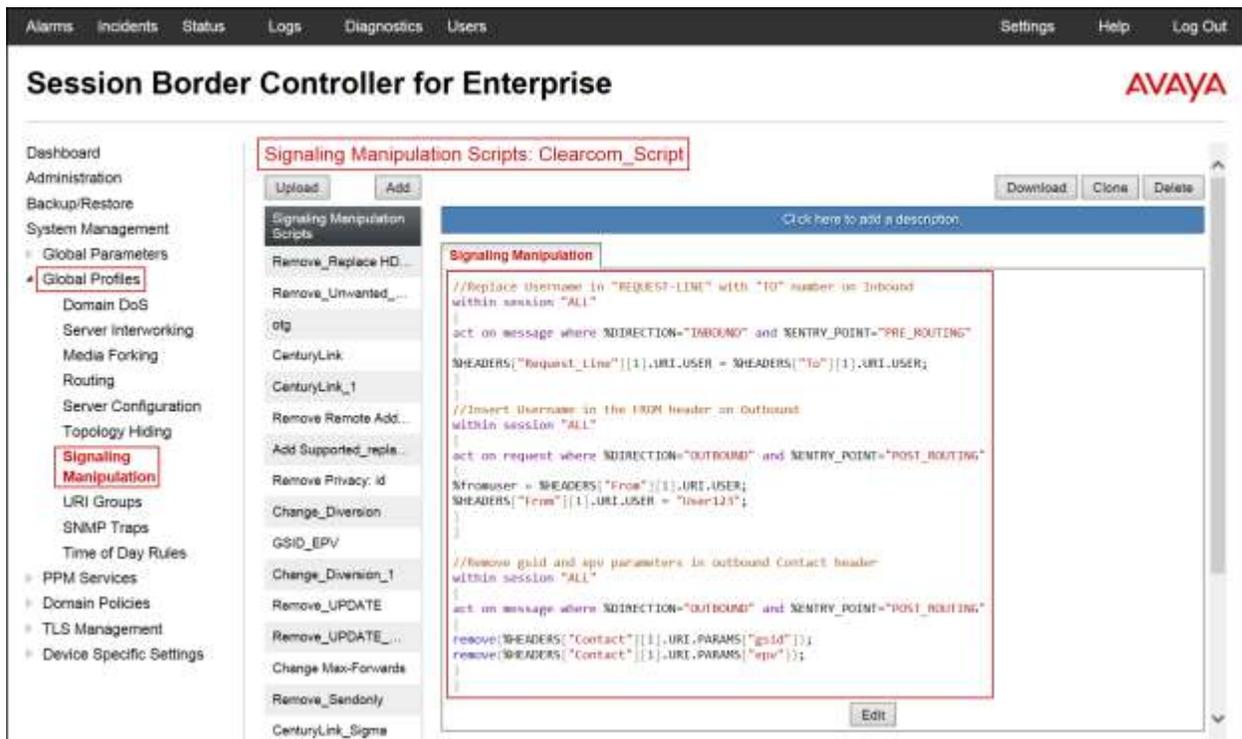
- Include the SIP trunk credential's username in the "From" header of all outbound calls.
- Copy the destination DID number present in the "To" header of incoming calls to the "Request-URI" header.
- Remove the "gsid" and "epv" parameters from outbound "Contact" headers.

The script will later be applied to the Server Configuration profile corresponding to the service provider in **Section 7.2.4**.

On the left navigation pane, select **Global 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 the complete script from **Appendix A**.
- Click **Save**.

The following screen capture shows the **Clearcom_Script** script after it was added.



7.2.4. Server Configuration

Server Profiles should be created for the Avaya SBCE's two peers, the Call Server (Session Manager) and the Trunk Server which is the SIP Proxy at the Service Provider's network.

To add the profile for the Call Server, from the **Global Profiles** menu on the left-hand navigation pane, select **Server Configuration**. Click **Add** in the **Server Profiles** section and enter the profile name: *Session Manager*.

On the **Edit Server Configuration Profile – General** window:

- **Server Type:** select *Call Server*.
- **IP Address / FQDN:** *172.16.5.32* (IP Address of the Session Manager SIP entity).
- **Port:** *5060* (This port must match the port number defined in **Section 6.6**).
- **Transports:** Select *TCP*.
- Click **Next**.

IP Address / FQDN	Port	Transport
172.16.5.32	5060	TCP

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

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

- Check **Enable Grooming**.
- Select **Avaya-SM** from the **Interworking Profile** drop down menu.
- Leave the **Signaling Manipulation Script** at the default **None**.
- Click **Finish**.

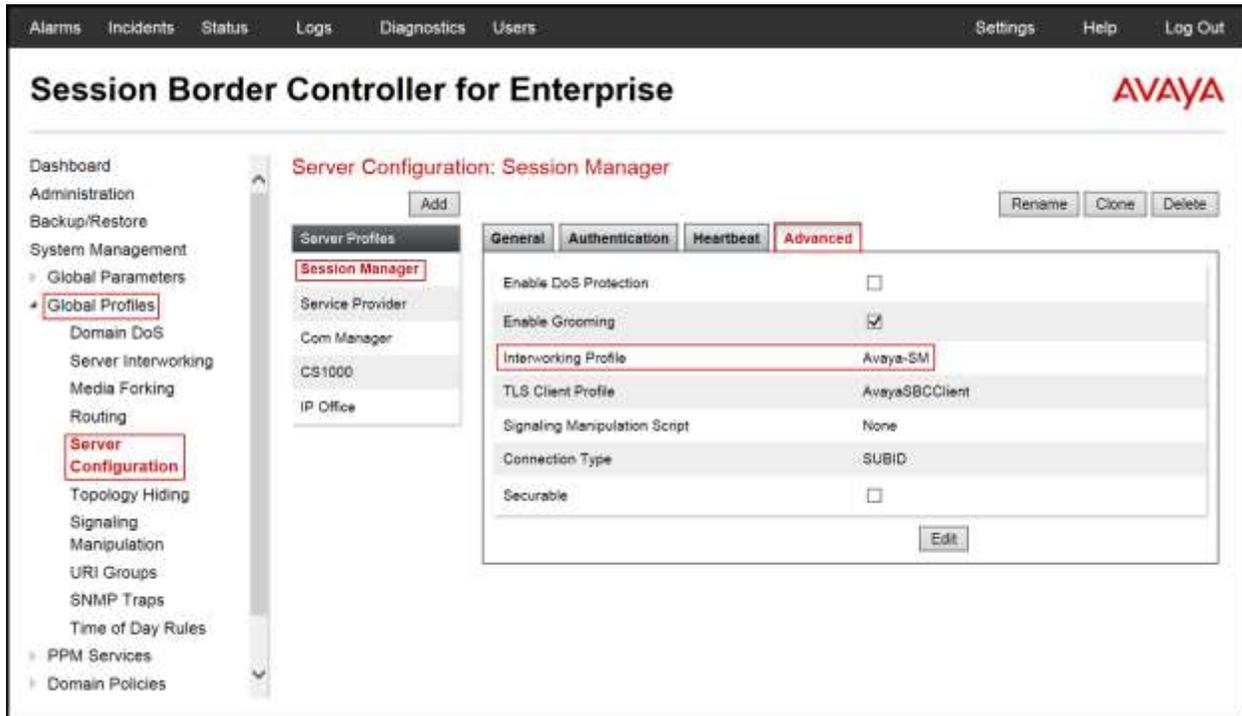
The screenshot shows the 'Add Server Configuration Profile - Advanced' window. The 'Enable Grooming' checkbox is checked. The 'Interworking Profile' dropdown is set to 'Avaya-SM'. The 'Signaling Manipulation Script' dropdown is set to 'None'. The 'Connection Type' dropdown is set to 'SUBID'. The 'Securable' checkbox is unchecked. The 'Back' and 'Finish' buttons are visible at the bottom.

The following screen capture shows the **General** tab of the newly created **Session Manager** Server Profile.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The 'Server Configuration: Session Manager' page is displayed. The 'General' tab is selected. The 'Server Type' is 'Call Server'. A table lists the IP Address / FQDN, Port, and Transport for the server profile.

IP Address / FQDN	Port	Transport
172.16.5.32	5060	TCP

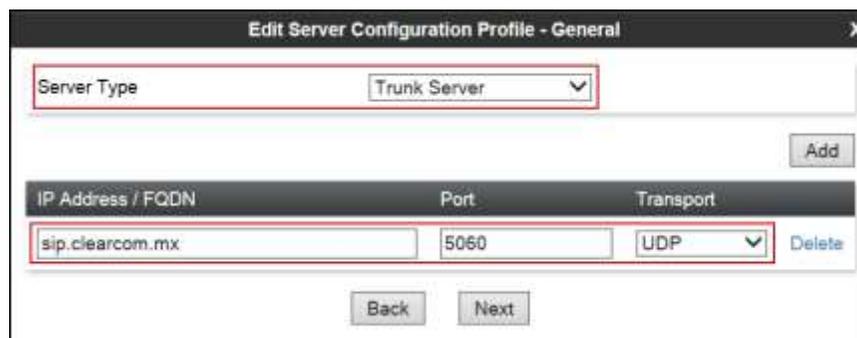
The following screen capture shows the **Advanced** tab of the newly created **Session Manager** Server Profile.



To add the profile for the Trunk Server, from the **Server Configuration** screen, click **Add** in the **Server Profiles** section and enter the profile name: *Service Provider*.

On the **Edit Server Configuration Profile – General** window

- **Server Type:** select *Trunk Server*.
- **IP Address/FQDN:** *sip.clearcom.mx* (the Fully Qualified Domain Name of the service provider SIP proxy server. This information was provided by Clearcom.).
- **Port:** *5060*.
- **Transports:** Select *UDP*.
- Click **Next**.



On the **Add Server Configuration Profile - Authentication** window:

- 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 (Must be entered, currently cannot be detected automatically from the challenge).
- Enter **Password** credential provided by the service provider for SIP trunk registration.
- Click **Next**.

The screenshot shows a window titled "Add Server Configuration Profile - Authentication". The window contains a form with the following fields and values:

- Enable Authentication:**
- User Name:** User123
- Realm:** clearcom.mx (Note: Leave blank to detect from server challenge)
- Password:** [Masked with dots]
- Confirm Password:** [Masked with dots]

At the bottom of the form, there are two buttons: "Back" and "Next".

On the **Add Server Configuration Profile - Heartbeat** window:

- Check the **Enable Heartbeat** box.
- Under **Method**, select **REGISTER** from the drop down menu.
- **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 configuration window titled "Add Server Configuration Profile - Heartbeat". The window contains a form with the following fields and values:

Enable Heartbeat	<input checked="" type="checkbox"/>
Method	REGISTER
Frequency	120 seconds
From URI	Jser123@clearcom.mx
To URI	Jser123@clearcom.mx

At the bottom of the form, there are two buttons: "Back" and "Next".

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

- Select *SP-General* from the **Interworking Profile** drop down menu.
- Select the *Clearcom_Script* from the **Signaling Manipulation Script** drop down menu.
- Click **Finish**

The screenshot shows a configuration window titled "Add Server Configuration Profile - Advanced". It contains several settings:

- Enable DoS Protection:
- Enable Grooming:
- Interworking Profile: SP-General (dropdown menu)
- Signaling Manipulation Script: Clearcom_Script (dropdown menu)
- Connection Type: SUBID (dropdown menu)
- Securable:

At the bottom, there are two buttons: "Back" and "Finish".

The following screen capture shows the **General** tab of the newly created **Service Provider** Server Configuration Profile.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The main content area is titled "Server Configuration: Service Provider". On the left, a navigation menu includes "Global Profiles" and "Server Configuration". The "General" tab is selected, showing the following configuration:

- Server Type: Trunk Server
- IP Address / FQDN: sip.clearcom.mx
- Port: 5060
- Transport: UDP

An "Edit" button is located below the table.

The following screen capture shows the **Authentication** tab of the newly created **Service Provider** Server Configuration Profile.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes links for Alarms, Incidents, Status, Logs, Diagnostics, Users, iftings, Help, and Log Out. The main header shows the product name and the Avaya logo. A left-hand navigation menu lists various configuration categories, with 'Server Configuration' highlighted. The main content area is titled 'Server Configuration: Service Provider' and features an 'Add' button and 'Rename', 'Clone', and 'Delete' buttons. Below this, there are tabs for 'General', 'Authentication', 'Heartbeat', and 'Advanced'. The 'Authentication' tab is active, showing a table with the following data:

Enable Authentication	<input checked="" type="checkbox"/>
User Name	User123
Realm	clearcom.mx

An 'Edit' button is located at the bottom right of the table. The left navigation menu includes: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles (selected), Domain DoS, Server Interworking, Media Forking, Routing, Server Configuration (highlighted), Topology Hiding, Signaling Manipulation, URI Groups, SNMP Traps, Time of Day Rules, PPM Services, Domain Policies, TLS Management, and Device Specific Settings.

The following screen capture shows the **Heartbeat** tab of the newly created **Service Provider** Server Configuration Profile.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes links for Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header shows the product name and the Avaya logo. A left-hand navigation menu lists various system management options, with 'Server Configuration' highlighted. The main content area is titled 'Server Configuration: Service Provider' and features an 'Add' button and 'Rename', 'Clone', and 'Delete' buttons. Below this, there are tabs for 'General', 'Authentication', 'Heartbeat', and 'Advanced'. The 'Heartbeat' tab is active, showing a configuration table with the following details:

Enable Heartbeat	<input checked="" type="checkbox"/>
Method	REGISTER
Frequency	120 seconds
From URI	User123@clearcom.mx
To URI	User123@clearcom.mx

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

The following screen capture shows the **Advanced** tab of the newly created **Service Provider** Server Configuration Profile.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes links for Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header shows the product name and the Avaya logo. A left-hand navigation menu lists various system management options, with 'Server Configuration' highlighted. The main content area is titled 'Server Configuration: Service Provider' and features an 'Add' button and 'Rename', 'Clone', and 'Delete' buttons. A sub-menu on the left lists 'Server Profiles' including Session Manager, Service Provider (highlighted), Com Manager, CS1000, and IP Office. The 'Advanced' tab is selected, showing a configuration table with the following settings:

Setting	Value
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	SP-General
Signaling Manipulation Script	Clearcom_Script
Connection Type	SUBID
Securable	<input type="checkbox"/>

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

7.2.5. Routing Profiles

Routing profiles define a specific set of routing criteria that are used, in conjunction with other types of domain policies, to determine the route that SIP packets should follow to arrive at their intended destination.

Two Routing profiles were created; one for inbound calls, with Session Manager as the destination, and the second one for outbound calls, which are sent to the service provider.

To create the inbound route, from the **Global Profiles** menu on the left-hand side:

- Select **Routing**.
- Click **Add** in the **Routing Profiles** section.
- Enter Profile Name: **Route_to_SM**.
- Click **Next**.

On the **Routing Profile** screen complete the following:

- Click on the **Add** button to add a **Next-Hop Address**.
- **Priority / Weight: 1**
- **Server Configuration:** Select **Session Manager**.
- The **Next Hop Address** is populated automatically with **172.16.5.32:5060 (TCP)** (Session Manager IP address, Port and Transport).
- Click **Finish**.

Priority / Weight	Server Configuration	Next Hop Address	Transport
1	Session Manager	172.16.5.32:5060 (TCP)	None

The following screen capture shows the newly created **Route_to_SM** Routing Profile.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes links for Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header shows the product name and the AVAYA logo. A left-hand navigation menu lists various system management options, with 'Global Profiles' and 'Routing' highlighted. The main content area is titled 'Routing Profiles: Route_to_SM' and features an 'Add' button. Below this, a list of routing profiles is shown, with 'Route_to_SM' selected. The configuration details for 'Route_to_SM' are displayed in a table with the following columns: Priority, URI Group, Time of Day, Load Balancing, Next Hop Address, and Transport. The table contains one entry with a priority of 1, a URI Group of *, a Time of Day of default, a Load Balancing of Priority, a Next Hop Address of 172.16.5.32, and a Transport of TCP. There are 'Edit' and 'Delete' buttons for this entry. A 'Routing Profile' section above the table includes an 'Update Priority' button and an 'Add' button. A description field above the table contains the text 'Click here to add a description.' and buttons for 'Rename', 'Clone', and 'Delete'.

Priority	URI Group	Time of Day	Load Balancing	Next Hop Address	Transport	
1	*	default	Priority	172.16.5.32	TCP	Edit Delete

Similarly, for the outbound route:

- Select **Routing**.
- Click **Add** in the **Routing Profiles** section.
- Enter Profile Name: **Route_to_SP**.
- Click **Next**.

On the **Routing Profile** screen complete the following:

- **Load Balancing:** Select **DNS/SRV**
- Click on the **Add** button to add a **Next-Hop Address**.
- **Server Configuration:** Select **Service Provider**.
- The **Next Hop Address** is populated automatically with **sip.clearcom.mx:5060 (UDP)** (Service Provider FQDN, Port and Transport).
- Click **Finish**.

The screenshot shows the 'Routing Profile' configuration window. It includes fields for 'URI Group' (set to '*') and 'Time of Day' (set to 'default'). The 'Load Balancing' dropdown is set to 'DNS/SRV'. The 'Transport' dropdown is set to 'None'. The 'Next Hop In-Dialog' checkbox is unchecked. The 'NAPTR' checkbox is unchecked. The 'Next Hop Priority' field is empty. The 'Ignore Route Header' checkbox is unchecked. An 'Add' button is located at the bottom right of the configuration area. Below the configuration area is a table with the following columns: 'Priority / Weight', 'Server Configuration', 'Next Hop Address', and 'Transport'. The table contains one row with the following values: 'Priority / Weight' is empty, 'Server Configuration' is 'Service Provider', 'Next Hop Address' is 'sip.clearcom.mx:5060 (UDP)', and 'Transport' is 'None'. A 'Delete' button is located to the right of the 'Transport' dropdown. At the bottom of the window are 'Back' and 'Finish' buttons.

The following screen capture shows the newly created **Route_to_SP** Routing Profile.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main header shows 'Session Border Controller for Enterprise' and the 'AVAYA' logo.

The left sidebar contains a navigation menu with categories like 'Dashboard', 'Administration', 'Backup/Restore', 'System Management', 'Global Profiles', 'Domain DoS', 'Server Interworking', 'Media Forking', 'Routing', 'Server', 'Configuration', 'Topology Hiding', 'Signaling Manipulation', 'URI Groups', 'SNMP Traps', 'Time of Day Rules', 'PPM Services', 'Domain Policies', and 'TLS Management'. The 'Routing' option is highlighted.

The main content area is titled 'Routing Profiles: Route_to_SP'. It features an 'Add' button and 'Rename', 'Clone', and 'Delete' buttons. Below this is a list of routing profiles: 'default', 'Route_to_SM', 'Route_to_SP' (highlighted), 'Route_to_CM', 'Route_to_CS1000', and 'Route_to_IP0'. There are also links for 'To SM from Rem.W' and 'To IPO from Rem.W'.

The 'Route_to_SP' profile is expanded, showing a table with the following configuration:

Priority	URI Group	Time of Day	Load Balancing	Next Hop Address	Transport	
1	*	default	DNS/SRV	sip.clearcom.mx	UDP	Edit Delete

7.2.6. Topology Hiding

Topology Hiding is a security feature which allows changing several parameters of the SIP packets, 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 SIP headers like To, From, Request-URI, Via, Record-Route and SDP to the IP addresses or domains expected by Session Manager and the SIP trunk Service Provider, allowing the call to be accepted in each case.

For the compliance test, 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.

To add the Topology Hiding profile in the enterprise direction, select **Topology Hiding** from the **Global Profiles** menu on the left-hand side:

- Select the **default** profile in the **Topology Hiding Profiles** list, then click **Clone** on top right of the screen.
- Enter the **Profile Name: *Session_Manager***.
- Click **Finish**.
- Click **Edit** on the newly added **Session_Manager** Topology Hiding profile.
- For **To** under **Header**, choose **Overwrite** from the pull-down menu under **Replace Action**, enter the domain name for the Enterprise (***avaya.lab.com***) under **Overwrite Value**.
- For **From** under **Header**, choose **Overwrite** from the pull-down menu under **Replace Action**, enter the domain name for the enterprise (***avaya.lab.com***) under **Overwrite Value**.
- For **Request-Line** under **Header**, choose **Overwrite** from the pull-down menu under **Replace Action**; enter the domain name for the Enterprise (***avaya.lab.com***) under **Overwrite Value**.

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

Finish

The following screen capture shows the newly created **Session_Manager** Topology Hiding Profile.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The left-hand navigation menu is expanded to show 'Global Profiles', with 'Topology Hiding' selected. The main content area is titled 'Topology Hiding Profiles: Session_Manager'. A list of profiles is shown on the left, with 'Session_Manager' highlighted. The main configuration area shows a table of header configurations for the 'Topology Hiding' profile.

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

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 the **default** profile in the **Topology Hiding Profiles** list, then click **Clone** on top right of the screen.
- Enter the **Profile Name: *Service_Provider***.
- Click **Finish**.
- Click **Edit** on the newly added **Service_Provider** Topology Hiding profile.
- For **To** under **Header**, choose **Overwrite** from the pull-down menu under **Replace Action**, enter the domain name for the service provider (***Clearcom.mx***) under **Overwrite Value**.
- For **From** under **Header**, choose **Overwrite** from the pull-down menu under **Replace Action**, enter the domain name for the service provider (***Clearcom.mx***) under **Overwrite Value**.
- For **Request-Line** under **Header**, choose **Overwrite** from the pull-down menu under **Replace Action**, enter the domain name for the service provider (***Clearcom.mx***) under **Overwrite Value**.

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

Finish

The following screen capture shows the newly created **Service_Provider** Topology Hiding Profile.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The left-hand navigation menu includes sections like Dashboard, Administration, System Management, Global Profiles, and Signaling Manipulation. The 'Global Profiles' section is expanded, and 'Topology Hiding' is selected. The main content area shows the configuration for a 'Topology Hiding Profile: Service_Provider'. A table lists various headers and their corresponding actions and values.

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

7.3. 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.

Note: The **default-trunk** Application Rule could have been used instead of creating a new one, but a new Application Rule was created to allow changes in the future.

7.3.1. 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 and video 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.
- **Rule Name:** enter the name of the profile, e.g., *2000 Sessions*.
- Under **Audio** check *In* and *Out* and set the **Maximum Concurrent Sessions** and **Maximum Sessions Per Endpoint** to recommended values, the value of *2000* was used in the sample configuration.
- Click **Finish**.

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

Miscellaneous

CDR Support: None, CDR w/ RTP, CDR w/o RTP

RTCP Keep-Alive:

Back Finish

The following screen capture shows the newly created **2000 Sessions** Application Rule.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main header shows 'Session Border Controller for Enterprise' and the 'AVAYA' logo.

The left sidebar contains a navigation menu with categories: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, PPM Services, Domain Policies (highlighted), Border Rules, Media Rules, Security Rules, Signaling Rules, End Point Policy Groups, Session Policies, TLS Management, and Device Specific Settings. Under 'Domain Policies', 'Application Rules' is highlighted.

The main content area is titled 'Application Rules: 2000 Sessions'. It features an 'Add' button, a 'Filter By Device...' dropdown, and 'Rename', 'Clone', and 'Delete' buttons. A blue bar prompts the user to 'Click here to add a description'. Below this is a table of application rules:

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

Below the table is a 'Miscellaneous' section with the following settings:

- CDR Support: None
- RTCP Keep-Alive: No

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

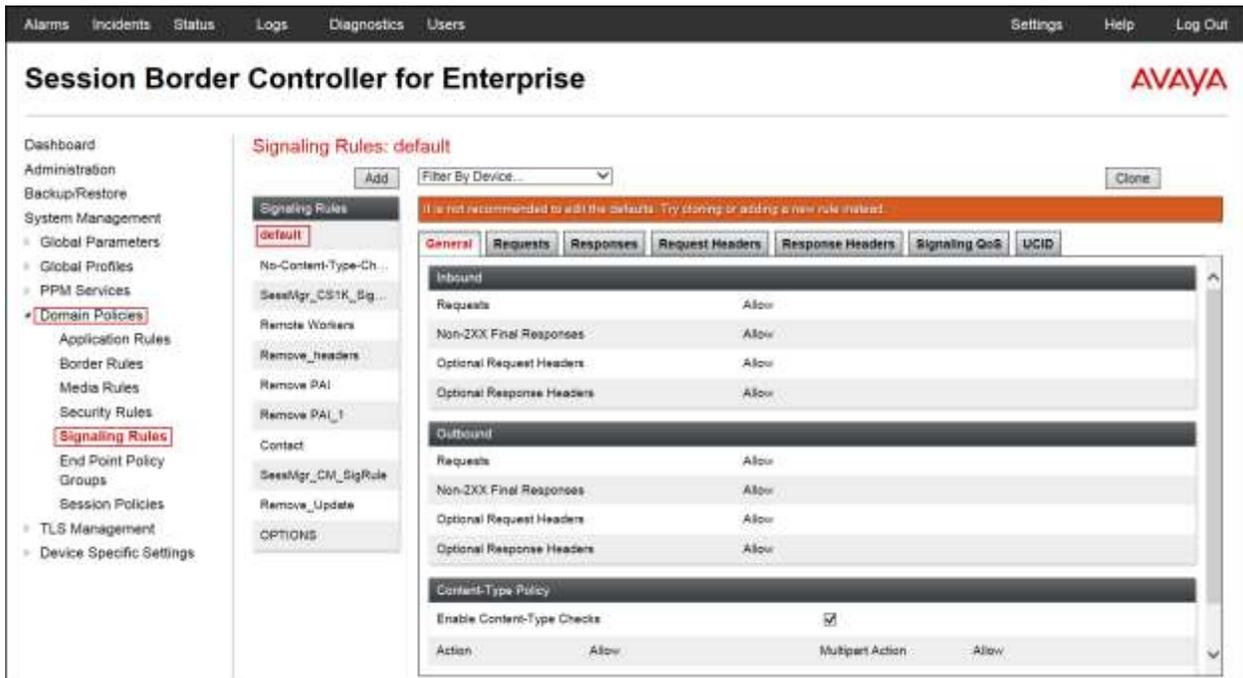
7.3.2. Media Rules

For the compliance test, the **default-low-med** Media Rule was used.



7.3.3. Signaling Rules

For the compliance test, the **default** Signaling Rule was used.



7.3.4. End Point Policy Groups

End Point Policy Groups are associations of different sets of rules (Media, Signaling, Security, etc.) to be applied to specific SIP messages traversing through the Avaya SBCE.

To create an End Point Policy Group for the Enterprise, from the **Domain Policies** menu, select **End Point Policy Groups**. Select **Add Group**, under **Group Name** enter *Enterprise*.

- **Application Rule:** *2000 Sessions*.
- **Border Rule:** *default*.
- **Media Rule:** *default-low-med*.
- **Security Rule:** *default-low*.
- **Signaling Rule:** *default*
- Click **Finish**.

Application Rule	2000 Sessions
Border Rule	default
Media Rule	default-low-med
Security Rule	default-low
Signaling Rule	default

The following screen capture shows the newly created **Enterprise** End Point Policy Group.

Order	Application	Border	Media	Security	Signaling	Edit
1	2000 Sessions	default	default-low-med	default-low	default	Edit

Similarly, to create an End Point Policy Group for the Service Provider SIP Trunk, select **Add Group**, under **Group Name** enter *Service Provider*.

- **Application Rule:** *2000 Sessions*.
- **Border Rule:** *default*.
- **Media Rule:** *default-low-med*.
- **Security Rule:** *default-low*.
- **Signaling Rule:** *default*.
- Click **Finish**.

The following screen capture shows the newly created **Service Provider** End Point Policy Group.

Order	Application	Border	Media	Security	Signaling	
1	2000 Sessions	default	default-low-med	default-low	default	Edit

7.4. Device Specific Settings

The **Device Specific Settings** allow the management of various device-specific parameters, which determine how a particular device will function when deployed in the network. Specific server parameters, like network and interface settings, as well as call flows, etc. are defined here.

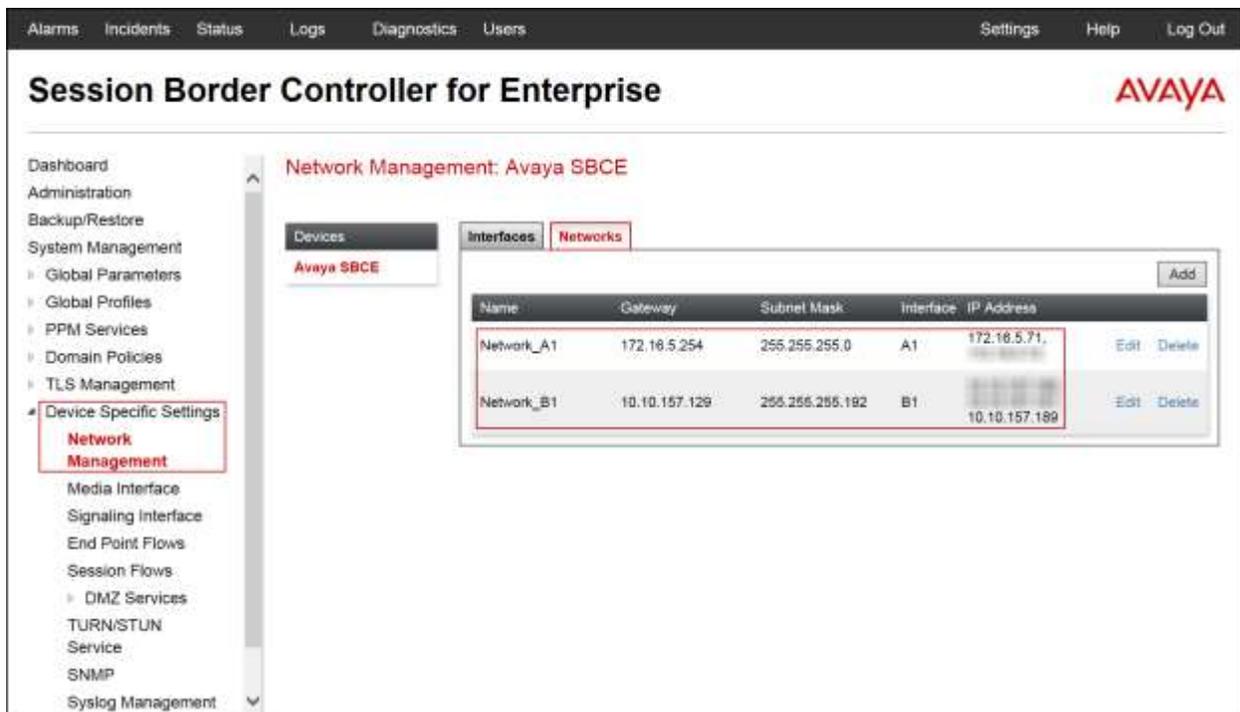
7.4.1. Network Management

The network information should have been previously completed. To verify the network configuration, from the **Device Specific Settings** on the left hand side, select **Network Management**. Select the **Network Configuration** tab.

In the event that changes need to be made to the network configuration information, they can be entered here.

Use **Figure 1** as reference for IP address assignments.

Note: Only the highlighted entity items were created for the compliance test, and are the ones relevant to these Application Notes. Blurred out items are part of the Remote Worker configuration, which is not discussed in this Application Notes.



The screenshot shows the Avaya SBCE management console. The top navigation bar includes '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 categories, with 'Device Specific Settings' expanded to show 'Network Management'. The main content area is titled 'Network Management: Avaya SBCE' and has tabs for 'Devices', 'Interfaces', and 'Networks'. The 'Networks' tab is active, displaying a table with the following data:

Name	Gateway	Subnet Mask	Interface	IP Address		
Network_A1	172.16.5.254	255.255.255.0	A1	172.16.5.71	Edit	Delete
Network_B1	10.10.157.129	255.255.255.192	B1	10.10.157.189	Edit	Delete

On the Interface Configuration tab, click the **Toggle** control for interfaces **A1** and **B1** to change the status to **Enabled**. It should be noted that the default state for all interfaces is **disabled**, so it is important to perform this step or the Avaya SBCE will not be able to communicate on any of its interfaces.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header displays "Session Border Controller for Enterprise" and the AVAYA logo. The left sidebar contains a navigation menu with categories like Dashboard, Administration, Backup/Restore, System Management, and Device Specific Settings. The "Device Specific Settings" category is expanded, and "Network Management" is selected. The main content area is titled "Network Management: Avaya SBCE" and shows a "Devices" tab with "Avaya SBCE" selected. Below this, there are tabs for "Interfaces" and "Networks". The "Interfaces" tab is active, showing a table with the following data:

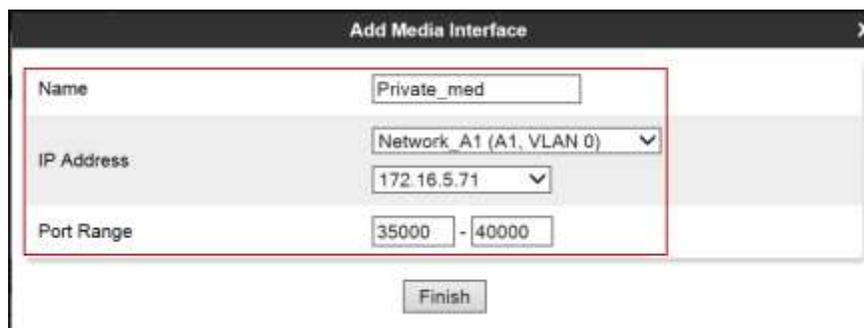
Interface Name	VLAN Tag	Status
A1		Enabled
A2		Disabled
B1		Enabled
B2		Disabled

7.4.2. Media Interface

Media Interfaces were created to adjust the port range assigned to media streams leaving the interfaces of the Avaya SBCE. On the Private and Public interfaces of the Avaya SBCE, port range 35000 to 40000 was used.

From the **Device Specific Settings** menu on the left-hand side, select **Media Interface**. Below is the configuration of the inside, private Media Interface of the Avaya SBCE.

- Select **Add** in the **Media Interface** area.
- **Name:** *Private_med*.
- Under **IP Address** select: **Network_A1 (A1, VLAN 0)**
Select **IP Address:** *172.16.5.71* (Private or A1 IP Address of the Avaya SBCE, toward Session Manager).
- Enter **Port Range:** *35000-40000*.
- Click **Finish**.



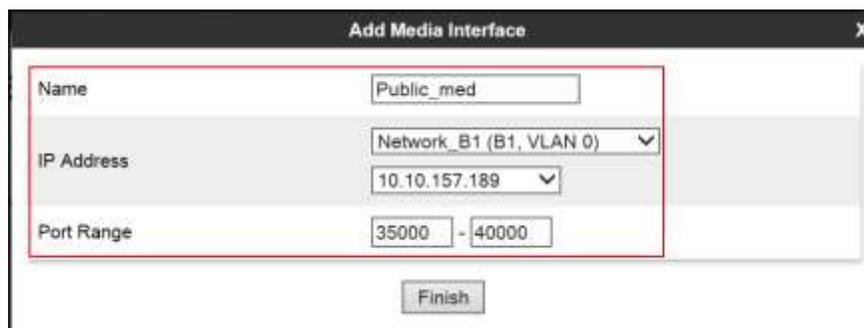
The screenshot shows a window titled "Add Media Interface" with a close button (X) in the top right corner. The window contains the following configuration fields:

- Name:** Private_med
- IP Address:** Network_A1 (A1, VLAN 0) (dropdown menu)
- IP Address:** 172.16.5.71 (dropdown menu)
- Port Range:** 35000 - 40000

A "Finish" button is located at the bottom center of the window.

Below is the configuration of the outside, public Media Interface of the Avaya SBCE.

- Select **Add** in the **Media Interface** area.
- **Name:** *Public_med*.
- Under **IP Address** select: **Network_B1 (B1, VLAN 0)**
Select **IP Address:** *10.10.157.189* (Public or B1 IP Address of the Avaya SBCE toward the Service Provider).
- **Port Range:** *35000-40000*.
- Click **Finish**.



The screenshot shows a window titled "Add Media Interface" with a close button (X) in the top right corner. The window contains the following configuration fields:

- Name:** Public_med
- IP Address:** Network_B1 (B1, VLAN 0) (dropdown menu)
- IP Address:** 10.10.157.189 (dropdown menu)
- Port Range:** 35000 - 40000

A "Finish" button is located at the bottom center of the window.

The following screen capture shows the newly created **Media Interfaces**.

The screenshot displays the Avaya SBCE web interface. The top navigation bar includes links for Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header reads "Session Border Controller for Enterprise" and the AVAYA logo is in the top right corner.

The left navigation menu is expanded to "Device Specific Settings", with "Media Interface" highlighted. The main content area is titled "Media Interface: Avaya SBCE". Below this title, there is a "Media Interface" tab and a table of configured interfaces. A warning message states: "Modifying or deleting an existing media interface will require an application restart before taking effect. Application restarts can be viewed from System Management." An "Add" button is located to the right of the warning.

Name	Media IP Network	Port Range	Edit	Delete
Private_med	172.16.5.71 Network_A1 (A1 - VLAN 0)	35000 - 40000	Edit	Delete
Public_med	10.10.157.189 Network_B1 (B1 - VLAN 0)	35000 - 40000	Edit	Delete
...	Edit	Delete
...	Edit	Delete

7.4.3. Signaling Interface

To create the Signaling Interface toward Session Manager, from the **Device Specific** menu on the left hand side, select **Signaling Interface**.

Below is the configuration of the inside private Signaling Interface of the Avaya SBCE.

- Select **Add** in the **Signaling Interface** area.
- **Name:** *Private_sig*.
- Under **IP Address** select: **Network_A1 (A1, VLAN 0)**
Select **IP Address:** *172.16.5.71* (Inside or A1 IP Address of the Avaya SBCE, toward Session Manager).
- **TCP Port:** *5060*
- Click **Finish**.

The screenshot shows a configuration window titled "Add Signaling Interface". The fields are as follows:

Name	Private_sig
IP Address	Network_A1 (A1, VLAN 0) 172.16.5.71
TCP Port	5060 <small>Leave blank to disable</small>
UDP Port	 <small>Leave blank to disable</small>
TLS Port	 <small>Leave blank to disable</small>
TLS Profile	None
Enable Shared Control	<input type="checkbox"/>
Shared Control Port	

Finish

Below is the configuration of the outside, public Signaling Interface of the Avaya SBCE.

- Select **Add** in the **Signaling Interface** area.
- **Name:** *Public_sig*.
- Under **IP Address** select: **Network_B1 (B1, VLAN 0)**
- Select **IP Address:** *10.10.157.189* (Public or B1 IP Address of the Avaya SBCE toward the Service Provider).
- **UDP Port:** *5060*.
- Click **Finish**.

The screenshot shows a configuration window titled "Add Signaling Interface". The fields are as follows:

Name	Public_sig
IP Address	Network_B1 (B1, VLAN 0) 10.10.157.189
TCP Port	Leave blank to disable
UDP Port	5060
TLS Port	Leave blank to disable
TLS Profile	None
Enable Shared Control	<input type="checkbox"/>
Shared Control Port	

Finish

The following screen capture shows the newly created **Signaling Interfaces**.

Alarms Incidents Status Logs Diagnostics Users Settings Help Log Out

Session Border Controller for Enterprise

AVAYA

Dashboard
Administration
Backup/Restore
System Management
Global Parameters
Global Profiles
PPM Services
Domain Policies
TLS Management
Device Specific Settings

Signaling Interface: Avaya SBCE

Devices
Avaya SBCE

Signaling interface

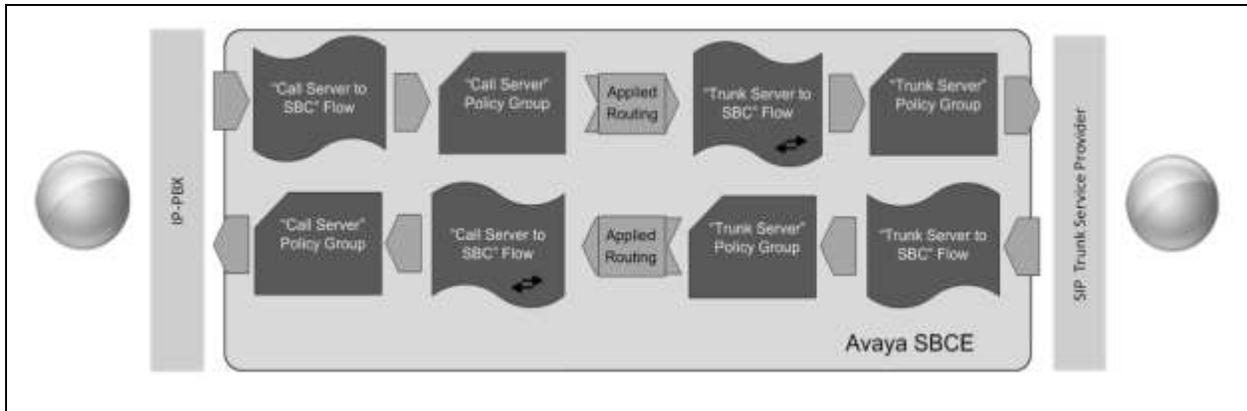
Modifying or deleting an existing signaling interface will require an application restart before taking effect. Application restarts can be issued from [System Management](#).

Add

Name	Signaling IP Network	TCP Port	UDP Port	TLS Port	TLS Profile	
Private_sig	172.16.5.71 Network_A1 (A1, VLAN 0)	5060	---	---	None	Edit Delete
Public_sig	10.10.157.186 Network_B1 (B1, VLAN 0)	---	5060	---	None	Edit Delete
...
...

7.4.4. 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.

To create the call flow toward the Service Provider SIP trunk, from the **Device Specific Settings** menu, select **End Point Flows**, and then the **Server Flows** tab. Click **Add** (not shown).

- **Flow Name:** *SIP_Trunk_Flow*.
- **Server Configuration:** *Service Provider*.
- **URI Group:** *
- **Transport:** *
- **Remote Subnet:** *
- **Received Interface:** *Private_sig*.
- **Signaling Interface:** *Public_sig*.
- **Media Interface:** *Public_med*.
- **End Point Policy Group:** *Service Provider*.
- **Routing Profile:** *Route_to_SM* (Note that this is the reverse route of the flow).
- **Topology Hiding Profile:** *Service_Provider*.
- **Signaling Manipulation Script:** *None*.
- **Remote Branch Office:** *Any*.
- Click **Finish**.

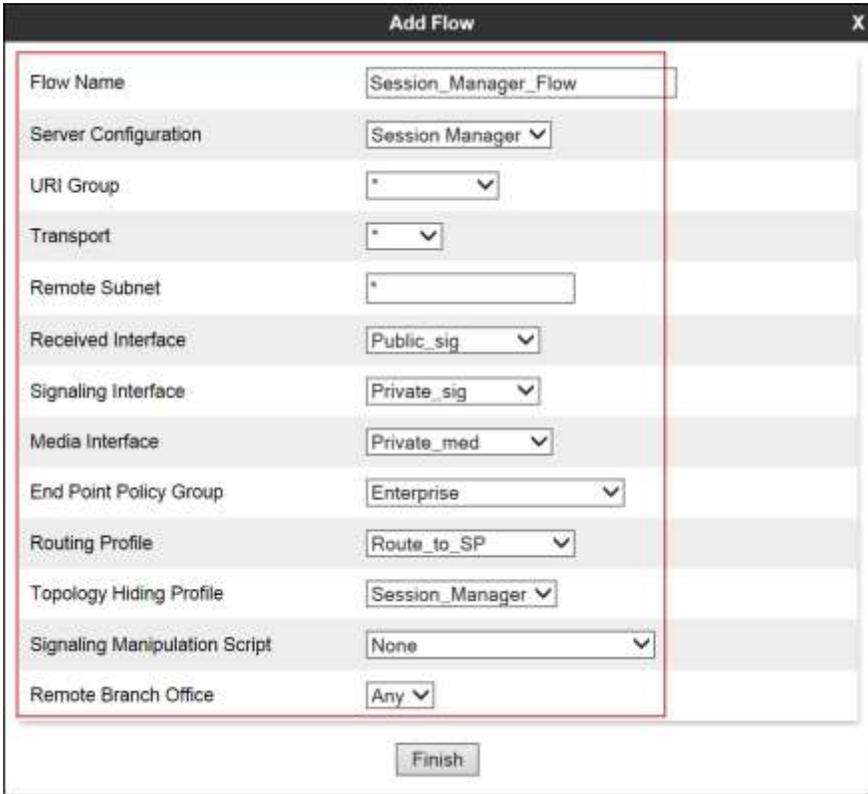
The screenshot shows a window titled "Add Flow" with a close button (X) in the top right corner. The window contains a list of configuration fields, each with a label and a corresponding input field or dropdown menu. A red rectangular box highlights the entire configuration area. The fields are as follows:

Field Label	Value
Flow Name	SIP_Trunk_Flow
Server Configuration	Service Provider
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Private_sig
Signaling Interface	Public_sig
Media Interface	Public_med
End Point Policy Group	Service Provider
Routing Profile	Route_to_SM
Topology Hiding Profile	Service_Provider
Signaling Manipulation Script	None
Remote Branch Office	Any

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

To create the call flow toward the Session Manager, click **Add**.

- **Flow Name:** *Session_Manager_Flow*.
- **Server Configuration:** *Session Manager*.
- **URI Group:** *
- **Transport:** *
- **Remote Subnet:** *
- **Received Interface:** *Public_sig*.
- **Signaling Interface:** *Private_sig*.
- **Media Interface:** *Private_med*.
- **End Point Policy Group:** *Enterprise*.
- **Routing Profile:** *Route_to_SP* (Note that this is the reverse route of the flow).
- **Topology Hiding Profile:** *Session_Manager*.
- **Signaling Manipulation Script:** *None*.
- **Remote Branch Office:** *Any*.
- Click **Finish**.



The screenshot shows a window titled "Add Flow" with a close button (X) in the top right corner. The window contains a list of configuration fields, each with a label and a corresponding input field or dropdown menu. A red rectangular box highlights the entire configuration area. At the bottom of the window, there is a "Finish" button.

Field	Value
Flow Name	Session_Manager_Flow
Server Configuration	Session Manager
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Public_sig
Signaling Interface	Private_sig
Media Interface	Private_med
End Point Policy Group	Enterprise
Routing Profile	Route_to_SP
Topology Hiding Profile	Session_Manager
Signaling Manipulation Script	None
Remote Branch Office	Any

The following screen capture shows the newly created **End Point Flows**.

The screenshot displays the Avaya Session Border Controller for Enterprise (SBCE) configuration interface. The main title is "Session Border Controller for Enterprise" with the AVAYA logo in the top right. The navigation menu on the left includes "Device Specific Settings" and "End Point Flows". The main content area is titled "End Point Flows: Avaya SBCE" and shows two tabs: "Subscriber Flows" and "Server Flows".

Under "Server Flows", there are two sections:

- Server Configuration: Service Provider**: Contains a table with one row:

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile	
1	SIP_Trunk_Flow	*	Private_sig	Public_sig	Service Provider	Route_to_SM	View Clone Edit Delete
- Server Configuration: Session Manager**: Contains a table with one row:

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile	
2	Session_Manager_Flow	*	Public_sig	Private_sig	Enterprise	Route_to_SP	View Clone Edit

8. Clearcom SIP Trunking Service Configuration

To use Clearcom's 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/> and requesting information.

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's 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.
- 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.

9. Verification and Troubleshooting

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

Verification Steps:

1. 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.
2. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active with two-way audio for more than 35 seconds.
3. Verify that the user on the PSTN can end an active call by hanging up.
4. Verify that an endpoint at the enterprise site can end an active call by hanging up.

9.1. Troubleshooting

9.1.1. Communication Manager

- **list trace station** <extension number>
Traces calls to and from a specific station.
- **list trace tac** <trunk access code number>
Traces 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.

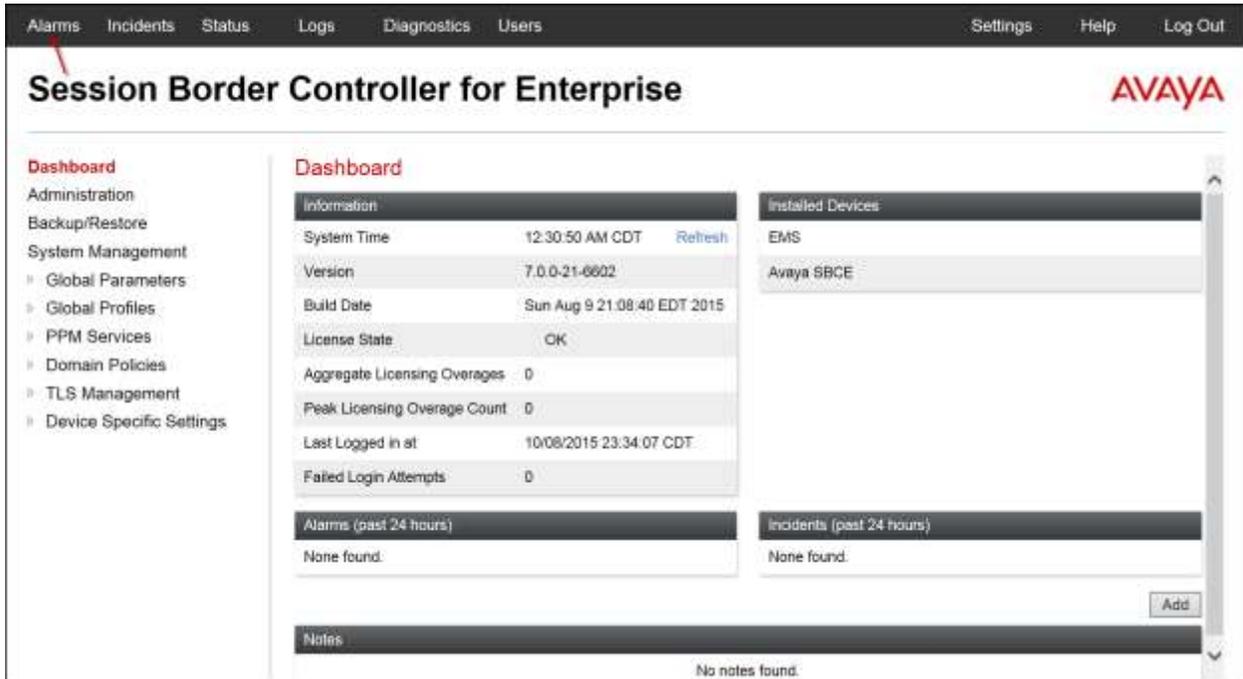
9.1.2. Session Manager

- **traceSM -x** – Session Manager command line tool for traffic analysis. Login to the Session Manager management CLI 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, navigate to **Home → Elements → Session Manager → System Tools → Call Routing Test**. Enter the requested data to run the test.

9.1.3. Avaya SBCE

There are several links and menus located on the taskbar at the top of the screen of the web interface that can be used for diagnostic and troubleshooting.

Alarms: Provides information about the health of the Avaya SBCE.



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



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

The screenshot shows the Avaya Session Border Controller for Enterprise dashboard. The navigation menu at the top includes Alarms, Incidents (highlighted with a red arrow), Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main content area is divided into several sections:

- Information:** System Time (12:30:50 AM CDT), Version (7.0.0-21-6602), Build Date (Sun Aug 9 21:08:40 EDT 2015), License State (OK), Aggregate Licensing Overages (0), Peak Licensing Overage Count (0), Last Logged in at (10/08/2015 23:34:07 CDT), and Failed Login Attempts (0).
- Installed Devices:** Lists EMS and Avaya SBCE.
- Alarms (past 24 hours):** None found.
- Incidents (past 24 hours):** None found.
- Notes:** No notes found.

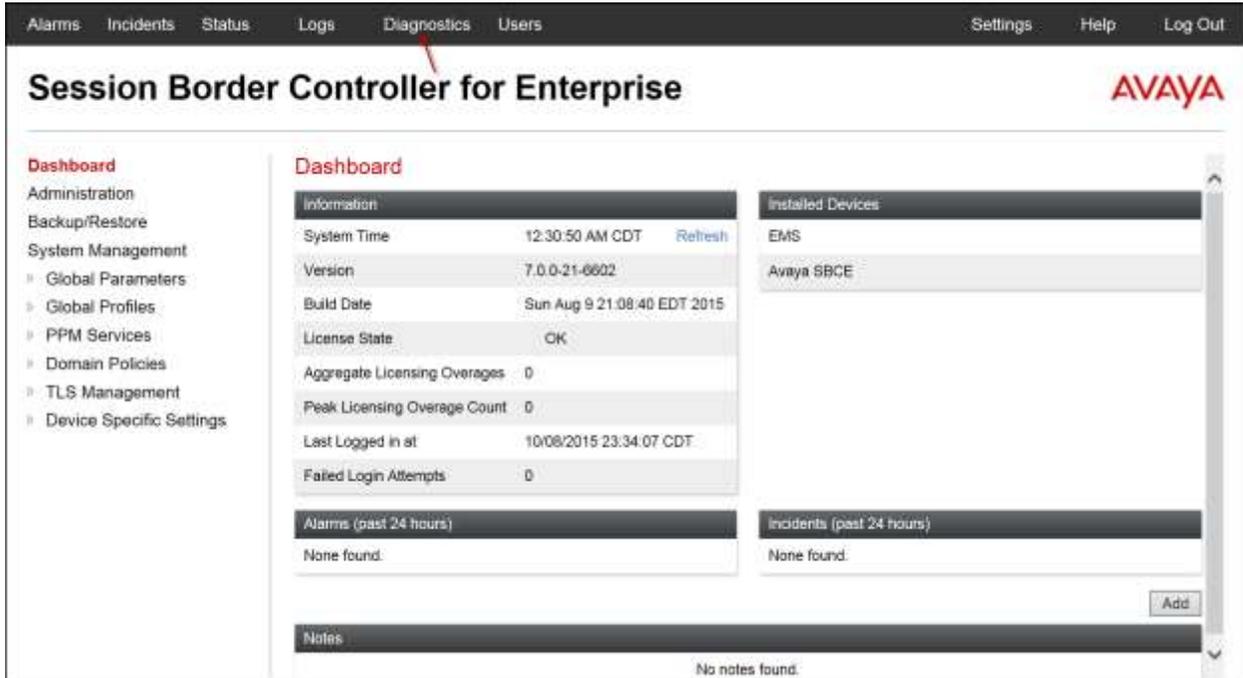
The following screen shows the Incident Viewer page.

The Incident Viewer page shows a table of incidents with the following data:

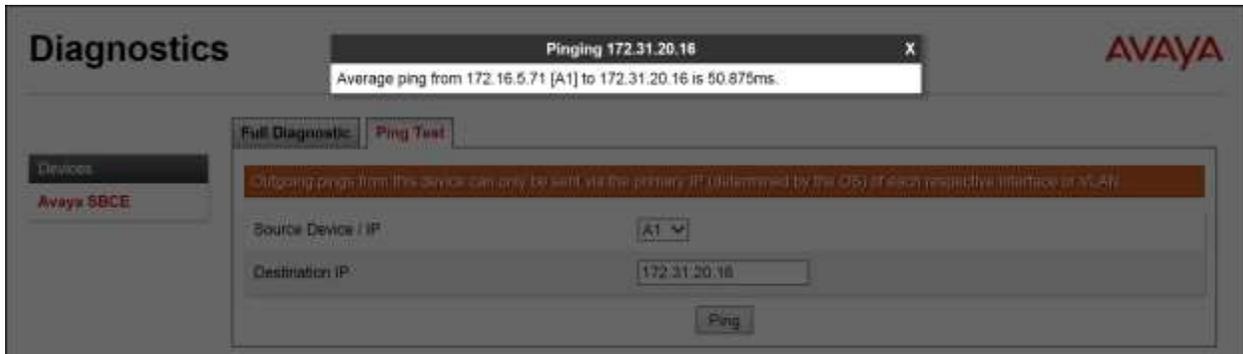
Type	ID	Date	Time	Category	Device	Cause
Message Dropped	722182809923738	10/8/15	11:40 PM	Policy	Avaya SBCE	No Subscriber Flow Matched
Server Heartbeat	721578665668258	9/24/15	10:55 PM	Policy	Avaya SBCE	Heartbeat Failed, Server is Down
Server Heartbeat	720627871533350	9/2/15	11:49 PM	Policy	Avaya SBCE	Heartbeat Failed, Server is Down
Server Heartbeat	720627092366599	9/2/15	11:23 PM	Policy	Avaya SBCE	Heartbeat Failed, Server is Down
Server Heartbeat	720581909185100	9/1/15	10:16 PM	Policy	Avaya SBCE	Heartbeat Failed, Server is Down

The page also includes filter controls for Device (Avaya SBCE) and Category (Policy), a 'Clear Filters' button, and 'Refresh' and 'Generate Report' buttons. The status 'Displaying results 1 to 5 out of 5' is shown above the table.

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 **Device Specific Settings** → **Troubleshooting** → **Trace**. Select the **Packet Capture** tab, set the desired configuration for the trace and click **Start Capture**.



Once the capture is stopped, click on 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 web interface. The top navigation bar includes 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main header shows 'Session Border Controller for Enterprise' and the 'AVAYA' logo. On the left, a navigation menu is expanded to 'Device Specific Settings', with sub-items like 'Network Management', 'Media Interface', 'Signaling Interface', 'End Point Flows', 'Session Flows', 'DMZ Services', 'TURN/STUN Service', 'SNMP', 'Syslog Management', 'Advanced Options', 'Troubleshooting', 'Debugging', 'Trace', 'DoS', and 'Learning'. The 'Trace' option is highlighted. The main content area is titled 'Trace: Avaya SBCE' and contains two tabs: 'Packet Capture' and 'Captures'. The 'Captures' tab is active, showing a table with the following data:

File Name	File Size (bytes)	Last Modified	
Test_20151012004900.pcap	12,288	October 12, 2015 12:49:10 AM CDT	Delete

A 'Refresh' button is located in the top right corner of the table area.

10. Conclusion

These Application Notes describe the procedures necessary for configuring Session Initiation Protocol (SIP) Trunk service for an enterprise solution consisting of Avaya Aura® Communication Manager Release 7.0, Avaya Aura® Session Manager Release 7.0, and Avaya Session Border Controller for Enterprise Release 7.0 to support Clearcom SIP Trunking Service, as shown in **Figure 1**.

Interoperability testing was completed successfully with the observations/limitations outlined in the scope of testing in **Section 2.1** as well as under test results in **Section 2.2**.

11.References

This section references the documentation relevant to these Application Notes.

Product documentation for Avaya Aura® Communication Manager, including the following, is available at: <http://support.avaya.com/>

- [1] *Administering Avaya Aura® Communication Manager*, Release 7.0, August 2015, Document Number 03-300509.
- [2] *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 7.0, August 2015, Document Number 555-245-205.

Product documentation for Avaya Aura® System Manager, including the following, is available at: <http://support.avaya.com/>

- [3] *Administering Avaya Aura® System Manager for Release 7.0*, Release 7.0, Issue 1, August 2015.

Product documentation for Avaya Aura® Session Manager, including the following, is available at: <http://support.avaya.com/>

- [4] *Administering Avaya Aura® Session Manager*, Release 7.0, August 2015.

Product documentation for the Avaya Session Border Controller for Enterprise, including the following, is available at: <http://support.avaya.com/>

- [5] *Deploying Avaya Session Border Controller for Enterprise*, Release 7.0, August 2015.
- [6] *Administering Avaya Session Border Controller for Enterprise*, Release 7.0, August 2015.

Product documentation for Avaya Aura® Media Server, is available at: <http://support.avaya.com/>

- [7] *Implementing and Administering Avaya Aura® Media Server*. Release 7.7. August 2015.
- [8] *Quick Start Guide to Using the Avaya Aura® Media Server with Avaya Aura® Communication Manager. White Paper*. August 2015.

Other resources:

- [9] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>.
- [10] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, <http://www.ietf.org/>

12. Appendix A: SigMa Script

Following is the Signaling Manipulation script that was used in the configuration of the Avaya SBCE, **Section 7.2.3**. When adding this script as instructed in **Section 7.2.4** enter a name for the script in the Title (e.g., **Clearcom_Script**) and copy/paste the entire script. Note that the user name shown below as "User123" will need to be changed with the correct user name provided by Clearcom for registration purpose.

Title: Clearcom_Script

```
//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
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"]);
}
}
```

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