



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

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# **Application Notes for Configuring the Cox Communications SIP Trunking Service with Avaya Aura® Communication Manager Evolution Server 6.3, Avaya Aura® Session Manager 6.3 and Avaya Session Border Controller for Enterprise 6.2 – Issue 1.0**

## **Abstract**

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between the Cox Communications SIP Trunking Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager 6.3, Avaya Aura® Communication Manager Evolution Server 6.3, Avaya Session Border Controller for Enterprise 6.2 and various Avaya endpoints. Cox Communications is a member of the Avaya DevConnect Service Provider program.

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

# 1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between the Cox Communications SIP Trunking Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager 6.3, Avaya Aura® Communication Manager Evolution Server 6.3, Avaya Session Border Controller for Enterprise 6.2 and various Avaya endpoints. In addition, Avaya Aura® System Manager 6.3 is used to configure Avaya Aura® Session Manager.

Customers using this Avaya SIP-enabled enterprise solution with the Cox Communications SIP Trunking Service are able to place and receive PSTN calls via a broadband WAN connection with SIP. This converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI.

## 2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the Cox Communications SIP Trunking Service via the public Internet and exercise the features and functionality listed in **Section 2.1**. It should be noted that for a real customer deployment, the connection between the enterprise and Cox Communications would not be done over the public Internet but instead would be via a Cox managed network IP connection. The simulated enterprise site was comprised of Communication Manager, Session Manager and Avaya Session Border Controller for Enterprise.

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

### 2.1. Interoperability Compliance Testing

To verify SIP trunking interoperability, the following features and functionality were covered during the interoperability compliance test.

- Sending and receiving SIP OPTIONS queries to the service provider
- Incoming PSTN calls to various phone types including Avaya H.323 and SIP telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types including H.323 and SIP telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from Avaya one-X® Communicator (soft client). Avaya one-X® Communicator can place calls from the local computer or control a remote phone. Both of these modes were tested. Avaya one-X® Communicator also

supports two Voice Over IP (VoIP) protocols: H.323 and SIP. Each protocol version of Avaya one-X® Communicator was also tested.

- Inbound and outbound calls to Avaya Flare® Experience for Windows.
- Various call types including: local, long distance, international, outbound toll-free, inbound toll-free, operator, operator-assisted and local directory assistance (411).
- Codec G.711MU
- DTMF transmission using RFC 2833
- Caller ID presentation and Caller ID restriction
- Response to incomplete call attempts and trunk errors
- Voicemail navigation for inbound and outbound calls
- Voicemail Message Waiting Indicator (MWI)
- User features such as hold and resume, internal call forwarding, transfer, and conference
- Off-net call transfer, conference, forwarding and mobility (extension to cellular – EC500)

Emergency 911 calls are supported but were not tested as part of the compliance test.

Items not supported included the following:

- G.729 codec
- T.38 fax
- REFER with replaces – Cox does not support REFER with replaces but does support REFER without replaces. However, it is not possible to enable use of REFER on Communication Manager (e.g., Network Call Redirection) for just those scenarios that use REFER without replaces. Thus, this solution will not use REFER (i.e., disable Network Call Redirection on Communication Manager).

In addition, the Cox Communications SIP Trunking Service requires the following behavior in the SIP messaging:

- Eleven (1+10) digits must be sent in the Request URI and To headers of an outbound SIP INVITE message. The 1+10 digits dialed by the user are passed unaltered through all enterprise components. See related routing **Sections 5.9, 6.8 and 7.12.2.**
- The **user=phone** parameter be set in the Request URI. See Communication Manager trunk group settings in **Section 5.7.**

## 2.2. Test Results

Interoperability testing of the Cox Communications SIP Trunking Service was completed with successful results for all test cases with the exception of the observations or limitations described below.

- **Calls on hold are torn down after 5 minutes:** Cox Communications will terminate a call on hold (without music on hold) after 5 minutes. This time is expected to increase in the future but is currently working as expected.

- **Calling Party Number (PSTN transfers):** The calling party number displayed on the PSTN phone is not updated to reflect the true connected party on calls that are transferred to the PSTN. After the call transfer is complete, the calling party number displays the number of the transferring party and not the actual connected party. Communication Manager provides the new connected party information by updating the Contact header in an UPDATE message but the far-end phone display is not updated. The PSTN phone display is ultimately controlled by the PSTN provider, thus this behavior is not necessarily indicative of a limitation of the combined Avaya/Cox solution. It is listed here simply as an observation.
- **Remote workers and DTMF digits:** An Avaya 9600 Series Deskphone configured as a remote worker was unable to pass DTMF digits to a PSTN Interactive Voice Response (IVR) system such as a PSTN voicemail system. This is being investigated as an Avaya issue. The workaround is to set the payload header value that the phone uses for DTMF digits to the value preferred by Cox Communications which is 101 instead of the default value of 120 used by the phone. To set the value, add the following line to the phone's 46xxsettings.txt configuration file: **SET DTMF\_PAYLOAD\_TYPE 101.**

## 2.3. Support

For technical support on the Cox Communications SIP Trunking Service, please contact Cox Communications via the following:

- Web: <http://www.cox.com> - Follow the support links for your particular service area.
- Phone: 1-800-620-6196

Avaya customers may obtain documentation and support for Avaya products by visiting <http://support.avaya.com>.

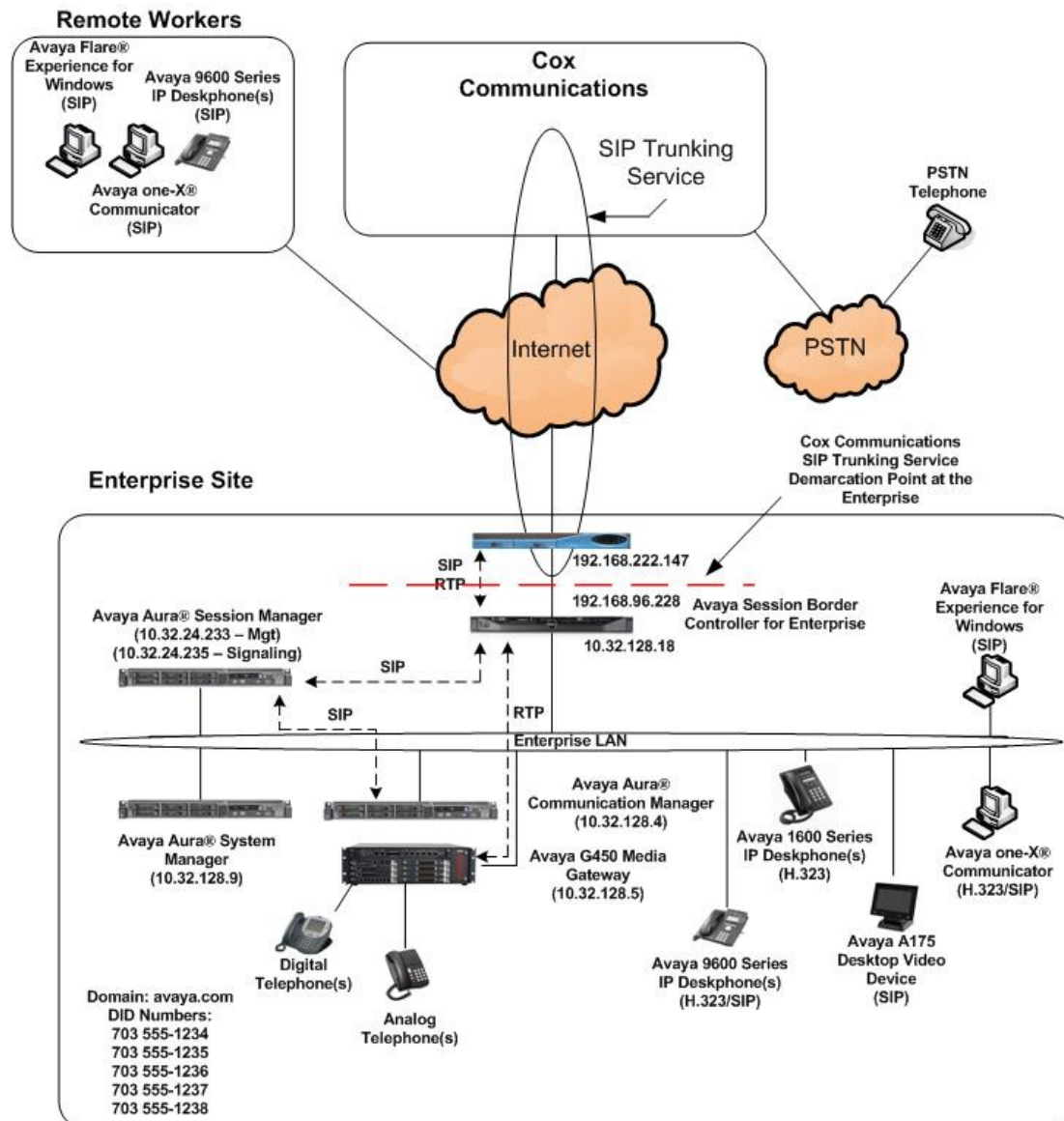
## 3. Reference Configuration

**Figure 1** illustrates a sample Avaya SIP-enabled enterprise solution connected to the Cox Communications SIP Trunking Service. This is the configuration used for compliance testing.

The components used to create the simulated customer site included:

- System Manager
- Session Manager
- Communication Manager
- Avaya G450 Media Gateway
- Avaya Session Border Controller for Enterprise (Avaya SBCE)
- Avaya 1600-Series IP telephones (H.323)
- Avaya 9600-Series IP telephones (H.323 and SIP)
- Avaya A175 Desktop Video Device
- Avaya one-X® Communicator (H.323 and SIP)
- Avaya Flare® Experience for Windows

The Cox Communications SIP Trunking Service deploys a session border controller at the enterprise site which is managed by Cox Communications and serves as the demarcation point for the service. The public side of the Avaya SBCE connects to the service and the private side connects to the enterprise network. All SIP and RTP traffic entering or leaving the enterprise flows through the Avaya SBCE. In 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. For security reasons, any actual public IP addresses used in the configuration have been replaced with private IP addresses. Similarly, any references to real routable PSTN numbers have also been changed to numbers that cannot be routed by the PSTN.



**Figure 1: Avaya IP Telephony Network using the Cox Communications SIP Trunking Service**

A separate trunk was created between Communication Manager and Session Manager to carry the service provider traffic. This was done so that any trunk or codec setting required by the service provider could be applied only to this trunk and not affect other enterprise SIP traffic. In addition, this trunk carried both inbound and outbound traffic.

For inbound calls, the calls flow from the service provider to the Avaya SBCE then to Session Manager. Session Manager uses the configured dial patterns (or regular expressions) and routing policies to determine the recipient (in this case Communication Manager) and on which link to send the call. Once the call arrives at Communication Manager, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed.

Outbound calls to the PSTN are first processed by Communication Manager and may be subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Communication Manager selects the proper SIP trunk, the call is routed to Session Manager. The Session Manager once again uses the configured dial patterns (or regular expressions) to determine the route to the Avaya SBCE. From the Avaya SBCE, the call is sent to the Cox Communications SIP Trunking Service.

For outbound calls, the Cox Communications SIP Trunking Service requires the enterprise send 11 (1+10) digits in the SIP destination headers (Request URI and To). In the SIP source headers (i.e., From, Contact, and P-Asserted-Identity), the enterprise sends 10 digits. For inbound calls, Cox Communications sends 10 digits in both the source headers and destination headers.

## 4. Equipment and Software Validated

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

<b>Avaya IP Telephony Solution Components</b>	
<b>Equipment/Software</b>	<b>Release/Version</b>
Avaya Aura® System Manager running on an Avaya S8800 Server	6.3 SP3 (Build 6.3.0.8.5682-6.3.8.1814) (Software Update Revision 6.3.3.5.1719) System Platform 6.3.0.0.18002
Avaya Aura® Session Manager running on an Avaya S8800 Server	6.3 SP3 (Build 6.3.3.0.633004)
Avaya Aura® Communication Manager running on an Avaya S8300 Server	6.3 SP1 (R016x.03.0.124.0-20850) System Platform 6.3.0.0.18002
Avaya G450 Media Gateway	33.13.0
Avaya Session Border Controller for Enterprise running on a Dell R210 V2 server	6.2.0.Q36
Avaya 1608 IP Deskphone (H.323) running Avaya one-X® Deskphone Value Edition	1.3 SP3 (1.3.3)
Avaya 9640G IP Deskphone (H.323) running Avaya one-X® Deskphone Edition	3.2 (S3.2)
Avaya 9641G IP Deskphone (H.323) running Avaya one-X® Deskphone Edition	6.2 SP4 (S6.2408)
Avaya 9611 IP Deskphone (SIP) running Avaya one-X® Deskphone SIP Edition	6.2 SP2 (6.2.2.17)
Avaya A175 Desktop Video Device with Avaya Flare® Experience	1.1.1
Avaya one-X® Communicator (H.323 or SIP)	6.1 SP8 (Build 6.1.8.06-SP8-40314)
Avaya Flare® Experience for Windows	1.1.2.11
<b>Cox Communications SIP Trunking Service Solution Components</b>	
<b>Equipment/Software</b>	<b>Release/Version</b>
Acme Packet Net-Net 3820 Session Border Controller (Enterprise)	nnSCX630m5
Acme Packet Net-Net 9200 Session Border Controller (Core)	SD710m4p2
Broadworks SIP Application Server	R17SP1

**Table 1: Equipment and Software Tested**

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

## 5. Configure Avaya Aura® Communication Manager

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

The Communication Manager configuration was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation. Note that the IP addresses and phone numbers shown throughout these Application Notes have been edited so that the actual public IP addresses of the network elements and public PSTN numbers are not revealed.

### 5.1. Licensing and Capacity

Use the **display system-parameters customer-options** command to verify that the **Maximum Administered SIP Trunks** value on **Page 2** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise including any trunks to the service provider. The example shows that **4000** SIP trunks are available and **60** are in use. The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity.

<b>display system-parameters customer-options</b>		Page	2 of 11
OPTIONAL FEATURES			
IP PORT CAPACITIES		USED	
Maximum Administered H.323 Trunks:		4000	36
Maximum Concurrently Registered IP Stations:		2400	2
Maximum Administered Remote Office Trunks:		4000	0
Maximum Concurrently Registered Remote Office Stations:		2400	0
Maximum Concurrently Registered IP eCons:		68	0
Max Concur Registered Unauthenticated H.323 Stations:		100	0
Maximum Video Capable Stations:		2400	1
Maximum Video Capable IP Softphones:		2400	4
<b>Maximum Administered SIP Trunks:</b>		<b>4000</b>	<b>60</b>
Maximum Administered Ad-hoc Video Conferencing Ports:		4000	0



## 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 be transferred back to the PSTN then leave the field set to **none**.

```
change system-parameters features                                     Page 1 of 20
      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
```

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 **anonymous** for both.

```
change system-parameters features                                     Page 9 of 20
      FEATURE-RELATED SYSTEM PARAMETERS

      CPN/ANI/ICLID PARAMETERS
        CPN/ANI/ICLID Replacement for Restricted Calls: anonymous
        CPN/ANI/ICLID Replacement for Unavailable Calls: anonymous

      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 server running Communication Manager (**procr**) and for Session Manager (**sessionMgr**). These node names will be needed for defining the service provider signaling group in **Section 5.6**.

<b>change node-names ip</b>		Page 1 of 2
		IP NODE NAMES
Name	IP Address	
cmm	10.32.128.4	
default	0.0.0.0	
<b>procr</b>	<b>10.32.128.4</b>	
procr6	::	
<b>sessionMgr</b>	<b>10.32.24.235</b>	

### 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. The list should include the codecs and preferred order defined by the service provider. For the compliance test, codec G.711MU was tested using ip-codec-set 3. To configure the codecs, enter the codecs in the **Audio Codec** column of the table in the order of preference. Default values can be used for all other fields.

<b>change ip-codec-set 3</b>		Page 1 of 2
		IP Codec Set
Codec Set: 3		
<b>Audio Codec</b>	Silence Suppression	Frames Per Pkt
1: <b>G.711MU</b>	n	2
2:		
3:		
		Packet Size (ms)
		20

On **Page 2**, set the **Fax Mode** to **off** since T.38 fax calls are not supported with this solution.

<b>change ip-codec-set 3</b>		Page 2 of 2
		IP Codec Set
Allow Direct-IP Multimedia? n		
<b>FAX</b>	<b>Mode</b>	Redundancy
	<b>off</b>	0
Modem	off	0
TDD/TTY	US	3

## 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 3 was chosen for the service provider trunk. Use the **change ip-network-region 3** command to configure region 3 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.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 3                                     Page 1 of 20

                                IP NETWORK REGION

Region: 3
Location:                Authoritative Domain: avaya.com
Name: SP Region          Stub Network Region: n
MEDIA PARAMETERS        Intra-region IP-IP Direct Audio: yes
Codec Set: 3            Inter-region IP-IP Direct Audio: yes
                        IP Audio Hairpinning? n
UDP Port Min: 2048
UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 46
Audio PHB Value: 46
Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
Audio 802.1p Priority: 6
Video 802.1p Priority: 5
H.323 IP ENDPOINTS      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 Link Bounce Recovery? y      RSVP Enabled? n
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 3 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 3 will be used for calls between region 3 (the service provider region) and region 1 (the rest of the enterprise). Creating this table entry for IP network region 3 will automatically create a complementary table entry on the IP network region 1 form for destination region 3. This complementary table entry can be viewed using the **display ip-network-region 1** command and navigating to **Page 4** (not shown).

change ip-network-region 3										Page	4	of	20
Source Region: 3      Inter Network Region Connection Management										I			M
										G	A		t
<b>dst</b>	<b>codec</b>	direct	WAN-BW-limits	Video	Intervening	Dyn	A	G					c
<b>rgn</b>	<b>set</b>	WAN	Units	Total Norm	Prio Shr Regions	CAC	R	L					e
<b>1</b>	<b>3</b>	y	NoLimit				n						t
2													
3	3											all	

## 5.6. Signaling Group

Use the **add signaling-group** command to create a signaling group between Communication Manager and Session Manager for use by the service provider trunk. This signaling group is used for inbound and outbound calls between the service provider and the enterprise. For the compliance test, signaling group 3 was used for this purpose and was configured using the parameters highlighted below.

- Set the **Group Type** field to **sip**.
- Set the **Transport Method** to the recommended default value of **tls** (Transport Layer Security). 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. If TLS is used here, it must also be used on the Session Manager entity link defined in **Section 6.6**.
- Set the **IMS Enabled** field to **n**. This specifies Communication Manager will serve as an Evolution Server for Session Manager.
- 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 a Session Manager.
- Set the **Near-end Node Name** to **procr**. This node name maps to the IP address of the Communication Manager as defined in **Section 5.3**.
- Set the **Far-end Node Name** to **sessionMgr**. This node name maps to the IP address of Session Manager as defined in **Section 5.3**.
- Set the **Near-end Listen Port** and **Far-end Listen Port** to a valid unused port instead of the default well-known port value. (For TLS, the well-known port value is 5061 and for TCP the well-known port value is 5060). At the time of Session Manager installation, a SIP connection between Communication Manager and Session Manager would have been established for use by all Communication Manager SIP traffic using the well-known port

value for TLS or TCP. By creating a new signaling group with a separate port value, a separate SIP connection is created between Communication Manager and Session Manager for SIP traffic to and from the service provider. As a result, any signaling group or trunk group settings (**Section 5.7**) will only affect the service provider traffic and not other SIP traffic at the enterprise. The compliance test was conducted with the **Near-end Listen Port** and **Far-end Listen Port** set to **5062**.

- 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 SIP trunk and the enterprise endpoint.
- Set the **DTMF over IP** field to **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- Set the **Alternate Route Timer** to **15**. This defines the number of seconds that Communication Manager will wait for a response (other than “100 Trying”) to an outbound INVITE before selecting another route. If an alternate route is not defined, then the call is cancelled after this interval.
- Default values may be used for all other fields.

<b>add signaling-group 3</b>		Page 1 of 2
SIGNALING GROUP		
Group Number: 3	Group Type: sip	
IMS Enabled? n	Transport Method: tcp	
Q-SIP? n		
IP Video? n	Enforce SIPS URI for SRTP? y	
Peer Detection Enabled? y	Peer Server: SM	
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y		
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n		
Near-end Node Name: procr	Far-end Node Name: sessionMgr	
Near-end Listen Port: 5062	Far-end Listen Port: 5062	
	Far-end Network Region: 3	
	Far-end Secondary Node Name:	
Far-end Domain: avaya.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? n	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Initial IP-IP Direct Media? n	
	Alternate Route Timer(sec): 15	

## 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 3 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 **Member Assignment Method** to **auto**.
- 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.

<b>add trunk-group 3</b>		Page 1 of 21	
TRUNK GROUP			
Group Number: 3	<b>Group Type: sip</b>	CDR Reports: y	
<b>Group Name: SP Trunk</b>	COR: 1	TN: 1	<b>TAC: 1003</b>
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n			
Queue Length: 0			
<b>Service Type: public-ntwrk</b>	Auth Code? n		
		<b>Member Assignment Method: auto</b>	
		<b>Signaling Group: 3</b>	
		<b>Number of Members: 10</b>	

On **Page 2**, the **Redirect On OPTIM Failure** value is the amount of time (in milliseconds) that Communication Manager will wait for a response (other than “100 Trying”) to a pending INVITE sent to an EC500 remote endpoint before selecting another route. If another route is not defined, then the call is cancelled after this interval. This time interval should be set to a value equal to the **Alternate Route Timer** on the signaling group form described in **Section 5.6**.

Verify that the **Preferred Minimum Session Refresh Interval** is set to a value acceptable to the service provider. This value defines the interval that re-INVITEs must be sent to keep the active session alive. For the compliance test, the value of **600** seconds was used.

<b>add trunk-group 3</b>		Page 2 of 21
Group Type: sip		
TRUNK PARAMETERS		
Unicode Name: auto		
		Redirect On OPTIM Failure: 15000
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

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 (E.164 numbering format) when passed in the SIP From, Contact and P-Asserted Identity headers. To remove the + sign, the **Numbering Format** was set to **private** and the **Numbering Format** in the route pattern was set to **unk-unk** (see **Section 5.9**).

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. For outbound calls, these same settings request that CPN block be activated on the far-end destination if a local user requests CPN block on a particular call routed out this trunk. Default values were used for all other fields.

<b>add trunk-group 3</b>		Page 3 of 21
TRUNK FEATURES		
ACA Assignment? n	Measured: none	Maintenance Tests? y
<b>Numbering Format: private</b>		UI Treatment: service-provider
		<b>Replace Restricted Numbers? y</b>
		<b>Replace Unavailable Numbers? y</b>
Modify Tandem Calling Number: no		
Show ANSWERED BY on Display? y		
DSN Term? n	SIP ANAT Supported? N	

On **Page 4**, set the **Mark Users as Phone** field to **y** which is required by Cox Communications. Since Cox Communications does not support all REFER scenarios (e.g., REFER with replaces) then the **Network Call Redirection** field must be set to **n**. Set the **Send Diversion Header** field to **y** and the **Support Request History** field to **n**. The **Send Diversion Header** field provides additional information to the network if the call has been redirected. These settings are needed to support call forwarding of inbound calls back to the PSTN and some Extension to Cellular (EC500) call scenarios.

Set the **Telephone Event Payload Type** to **101**, the value used by Cox Communications.

<b>add trunk-group 3</b>	<b>Page 4 of 21</b>
PROTOCOL VARIATIONS	
<b>Mark Users as Phone? y</b>	
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n	
Send Transferring Party Information? n	
<b>Network Call Redirection? n</b>	
<b>Send Diversion Header? y</b>	
<b>Support Request History? n</b>	
<b>Telephone Event Payload Type: 101</b>	
Shuffling with SDP? n	
Convert 180 to 183 for Early Media? n	
Always Use re-INVITE for Display Updates? n	
Identity for Calling Party Display: P-Asserted-Identity	
Block Sending Calling Party Location in INVITE? n	
Accept Redirect to Blank User Destination? n	
Enable Q-SIP? n	

To ensure interoperability with Avaya SIP endpoints, the **Mark Users as Phone** field must also be set to **y** on the SIP trunk used by the SIP endpoints to register and communicate with the Session Manager. In most cases, this will be the trunk created during the initial installation of the Session Manager. In the case of the compliance test, this was trunk-group 1.

<b>change trunk-group 1</b>	<b>Page 4 of 21</b>
PROTOCOL VARIATIONS	
<b>Mark Users as Phone? y</b>	
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n	
Send Transferring Party Information? n	
Network Call Redirection? n	
Send Diversion Header? n	
Support Request History? y	
<b>Telephone Event Payload Type:</b>	
Shuffling with SDP? n	
Convert 180 to 183 for Early Media? n	
Always Use re-INVITE for Display Updates? n	
Identity for Calling Party Display: P-Asserted-Identity	
Block Sending Calling Party Location in INVITE? n	
Accept Redirect to Blank User Destination? n	
Enable Q-SIP? n	



## 5.8. Calling Party Information

The calling party number is sent in the SIP “From”, “Contact” and “PAI” headers. Since private numbering was selected to define the format of this number (**Section 5.7**), use the **change private-numbering** command to create an entry for each extension which has a DID assigned. The DID number will be assigned by the SIP service provider. It is used to authenticate the caller.

In the sample configuration, five DID numbers were assigned for testing. These five numbers were assigned to the five extensions 40003, 40006, 40022, 40023 and 40024. Thus, these same 10-digit numbers were used in the outbound calling party information on the service provider trunk when calls were originated from these extensions.

change private-numbering 5					Page 1 of 2
NUMBERING - PRIVATE FORMAT					
Ext Len	Ext Code	Trk Grp (s)	Private Prefix	Total Len	
5	4			5	Total Administered: 6
5	40003	3	7035551234	10	Maximum Entries: 540
5	40006	3	7035551235	10	
5	40022	3	7035551236	10	
5	40023	3	7035551237	10	
5	40024	3	7025551238	10	

In a real customer environment, normally the DID number is comprised of the local extension plus a prefix. If this is true, then a single private numbering entry can be applied for all extensions. In the example below, all stations with a 5-digit extension beginning with 4 will send the calling party number as the **Private Prefix** plus the extension number.

change private-numbering 5					Page 1 of 2
NUMBERING - PRIVATE FORMAT					
Ext Len	Ext Code	Trk Grp (s)	Private Prefix	Total Len	
5	4			5	Total Administered: 2
5	4	3	70355	10	Maximum Entries: 540

## 5.9. 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: 3			
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	
1	4	dac							
3	5	ext							
4	5	ext							
8	1	fac							
<b>9</b>	<b>1</b>	<b>fac</b>							
*	3	fac							
#	3	fac							

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

change feature-access-codes			FEATURE ACCESS CODE (FAC)						Page 1 of 11
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:									
Answer Back Access Code:									
Attendant Access Code:									
Auto Alternate Routing (AAR) Access Code: 8									
<b>Auto Route Selection (ARS) – Access Code 1: 9</b>			Access Code 2:						
Automatic Callback Activation:			Deactivation:						
Call Forwarding Activation Busy/DA: *01 All: *02			Deactivation: *03						
Call Forwarding Enhanced Status: Act:			Deactivation:						

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

change ars analysis 0				Page 1 of 2			
ARS DIGIT ANALYSIS TABLE				Percent Full: 1			
Location: all							
	Dialed String	Total Min Max	Route Pattern	Call Type	Node Num	ANI Reqd	
	0	1 1	2	op		n	
	0	11 11	2	op		n	
	011	10 18	2	intl		n	
	1703	11 11	2	fnpa		n	
	1732	11 11	2	fnpa		n	
	1800	11 11	2	fnpa		n	
	1877	11 11	2	fnpa		n	
	1908	11 11	2	fnpa		n	
	411	3 3	2	svcl		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 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 SIP service provider. For the compliance test, trunk group **3** 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.
- **Pfx Mrk: 1** The prefix mark (**Pfx Mrk**) of one will prefix any FNPA 10-digit number with a 1 and leave numbers of any other length unchanged. This will ensure 1 + 10 digits are sent to the Session Manager for long distance North American Numbering Plan (NANP) numbers.
- **Numbering Format: 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**.
- **LAR: next**

change route-pattern 2													Page 1 of 3								
Pattern Number: 4													Pattern Name: SP Route								
SCCAN? n													Secure SIP? n								
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted						DCS/	IXC							
No			Mrk	Lmt	List	Del	Digits						QSIG								
													Intw								
1:	3	0	1										n	user							
2:													n	user							
3:													n	user							
4:													n	user							
5:													n	user							
6:													n	user							
BCC VALUE													TSC	CA-TSC	ITC	BCIE	Service/Feature	PARM	No.	Numbering	LAR
0	1	2	M	4	W								Request					Dgts	Format		
																				Subaddress	
1:	y	y	y	y	y	n	n						rest					unk-unk	next		
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		

## 6. Configure Avaya Aura® Session Manager

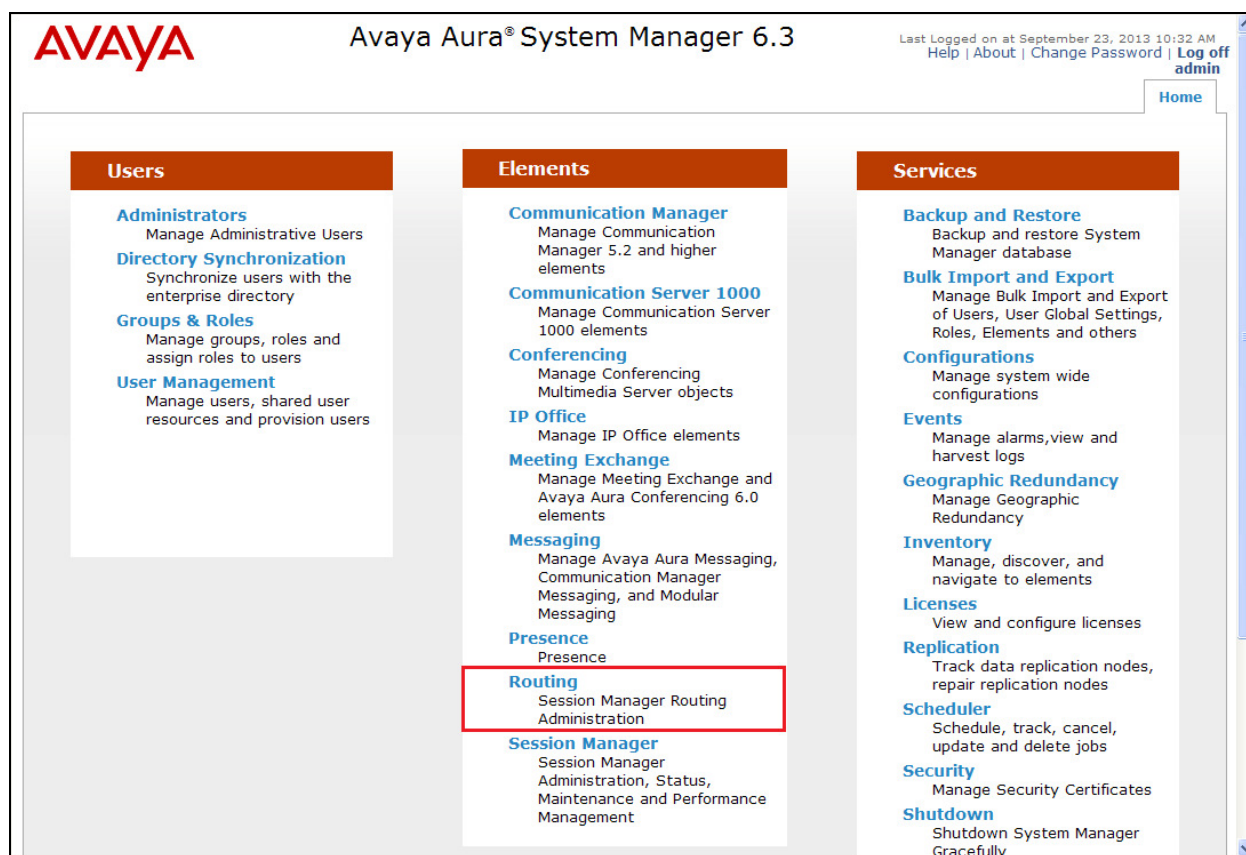
This section provides the procedures for configuring Session Manager. The procedures include configuring the following items:

- 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 governs which Routing Policy is used to service a call.
- 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 creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, locations, SIP entities, and Session Manager itself. However, each item should be reviewed to verify the configuration.

## 6.1. Avaya Aura® System Manager Login and Navigation

Session Manager configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL <https://<ip-address>/SMGR>, where **<ip-address>** is the IP address of System Manager. Log in with the appropriate credentials and click on **Login** (not shown). The **Home** page is displayed. The links displayed below will be referenced in subsequent sections to navigate to items requiring configuration. Most items will be located under the **Elements** → **Routing** link highlighted below.



Clicking the **Elements** → **Routing** link, displays the **Introduction to Network Routing Policy** page. In the left-hand pane is a navigation tree containing many of the items to be configured in the following sections.

The screenshot displays the Avaya Aura System Manager 6.3 web interface. At the top left is the Avaya logo. The title bar reads 'Avaya Aura® System Manager 6.3'. On the top right, it shows the last login time: 'Last Logged on at September 23, 2013 10:32 AM', along with links for 'Help', 'About', 'Change Password', and a 'Log off admin' button. Below the title bar, there are tabs for 'Routing' (active) and 'Home'. The left-hand navigation pane is expanded to show the 'Routing' section, which includes a list of items: Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Introduction to Network Routing Policy' and includes a 'Help ?' link. The text explains that Network Routing Policy consists of several routing applications like 'Domains', 'Locations', 'SIP Entities', etc., and provides a recommended order for configuration: Step 1: Create 'Domains' of type SIP; Step 2: Create 'Locations'; Step 3: Create 'Adaptations'; Step 4: Create 'SIP Entities'. A note specifies that SIP Entities used as 'Outbound Proxies' include a certain 'Gateway' or 'SIP Trunk'.

**AVAYA** Avaya Aura® System Manager 6.3 Last Logged on at September 23, 2013 10:32 AM  
Help | About | Change Password | Log off admin

Routing x Home

Home / Elements / Routing Help ?

### Introduction to Network Routing Policy

Network Routing Policy consists of several routing applications like "Domains", "Locations", "SIP Entities", etc.

The recommended order to use the routing applications (that means the overall routing workflow) to configure your network configuration is as follows:

- Step 1: Create "Domains" of type SIP (other routing applications are referring domains of type SIP).
- Step 2: Create "Locations"
- Step 3: Create "Adaptations"
- Step 4: Create "SIP Entities"

- SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"

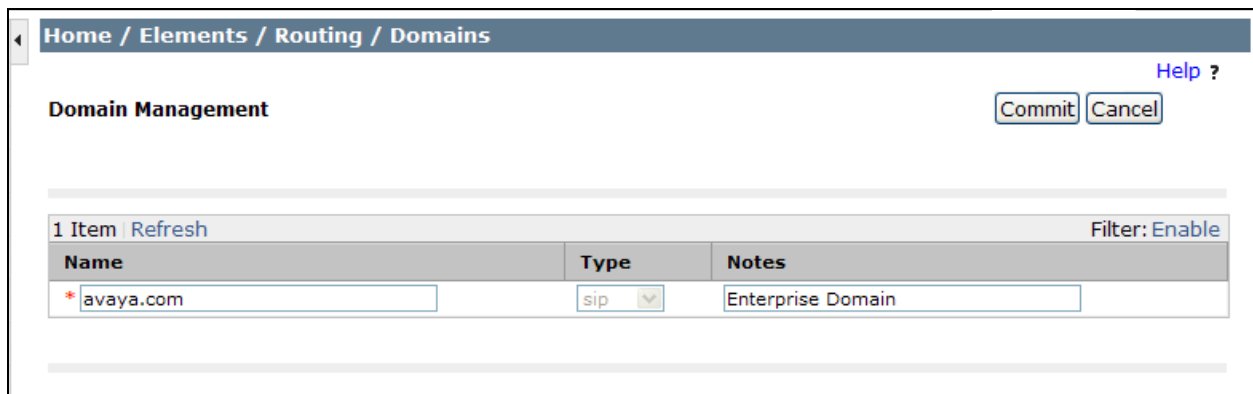
## 6.2. Specify SIP Domain

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, this includes the enterprise domain (**avaya.com**).

Navigate to **Routing → Domains** in the left-hand navigation pane (**Section 6.1**) 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**. The screen below shows the entry for the enterprise domain.



The screenshot shows a web interface for "Domain Management". At the top, there is a breadcrumb trail: "Home / Elements / Routing / Domains". Below this, the title "Domain Management" is displayed. To the right of the title are two buttons: "Commit" and "Cancel", and a "Help ?" link. Below the title bar, there is a table with the following structure:

1 Item <a href="#">Refresh</a> <span style="float: right;">Filter: <a href="#">Enable</a></span>		
Name	Type	Notes
* <input type="text" value="avaya.com"/>	<input type="text" value="sip"/> ▼	<input type="text" value="Enterprise Domain"/>



### 6.3. Add Location

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. A single location was defined for the enterprise even though multiple subnets were used. The screens below show the addition of the location named **Location 1**, which includes all equipment on the enterprise including Communication Manager, Session Manager and the Avaya SBCE.

To add a location, navigate to **Routing → Locations** in the left-hand navigation pane (**Section 6.1**) and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

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).

Home / Elements / Routing / Locations

Location Details

Commit Cancel

Help ?

General

\* Name: Location 1

Notes: Enterprise Site for SP Testing

Scroll down to the **Location Pattern** section. Click **Add** and enter the following values. Use default values for all remaining fields.

- **IP Address Pattern:** Add all IP address patterns used to identify the location. The compliance test used the two subnets highlighted below.
- **Notes:** Add a brief description (optional).

Click **Commit** to save.

Location Pattern

Add Remove

5 Items Refresh Filter: Enable

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.1.2.*	Trenton CM 5.2.1 Environment
<input type="checkbox"/>	* 10.32.120.*	AAM and other CPE devices
<input type="checkbox"/>	* 10.32.128.*	Princeton CM and other CPE devices
<input type="checkbox"/>	* 10.32.24.235	SM (devcon-asm)
<input type="checkbox"/>	* 192.168.49.*	CPE endpoints

Select : All, None

## 6.4. Add Adaptation Module

Session Manager can be configured with adaptation modules that can modify SIP messages before or after routing decisions have been made. A generic adaptation module **DigitConversionAdapter** supports digit conversion of telephone numbers in specific headers of SIP messages. Other adaptation modules are built on this generic, and can modify other headers to permit interoperability with third party SIP products.

For the compliance test, one adaptation was created for Communication Manager. The adaptation mapped inbound DID numbers from Cox Communications to local Communication Manager extensions.

To create the adaptation that will be applied to the Communication Manager SIP entity, navigate to **Routing → Adaptations** in the left-hand navigation pane (**Section 6.1**) and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

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

- **Adaptation name:** Enter a descriptive name for the adaptation.
- **Module name:** Enter **DigitConversionAdapter**.
- **Notes:** Enter a description (optional).

Home / Elements / Routing / Adaptations [Help ?](#)

**Adaptation Details**

**General**

\* **Adaptation name:**

**Module name:**  ▼

**Module parameter:**

**Egress URI Parameters:**

**Notes:**

To map inbound DID numbers from Cox Communications to Communication Manager extensions, scroll down to the **Digit Conversion for Outgoing Calls from SM** section. Create an entry for each DID to be mapped. Click **Add** and enter the following values for each mapping. Use default values for all remaining fields.

- **Matching Pattern:** Enter a digit string used to match the inbound DID number.
- **Min:** Enter a minimum dialed number length used in the match criteria.
- **Max:** Enter a maximum dialed number length used in the match criteria.
- **Delete Digits** Enter the number of digits to delete from the beginning of the received number.
- **Insert Digits:** Enter the number of digits to insert at the beginning of the received number.
- **Address to modify:** Select **destination** since this digit conversion only applies to the destination number.

Click **Commit** to save.

**Digit Conversion for Outgoing Calls from SM**

[Add](#) [Remove](#)

5 Items [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	Matching Pattern ▲	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Da
<input type="checkbox"/>	* 7035551234	* 10	* 10		* 10	40003	destination ▼	
<input type="checkbox"/>	* 7035551235	* 10	* 10		* 10	40006	destination ▼	
<input type="checkbox"/>	* 7035551236	* 10	* 10		* 10	40022	destination ▼	
<input type="checkbox"/>	* 7035551237	* 10	* 10		* 10	40023	destination ▼	
<input type="checkbox"/>	* 7035551238	* 10	* 10		* 10	40024	destination ▼	

In a real customer environment, often the DID number is comprised of the local extension plus a prefix. If this is true, then a single digit conversion entry can be created for all extensions. In the example below, a 5 digit prefix is deleted from each incoming DID number leaving a 5 digit extension to be routed by Session Manager.

**Digit Conversion for Outgoing Calls from SM**

[Add](#) [Remove](#)

1 Item [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	Matching Pattern ▲	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Da
<input type="checkbox"/>	* 70355	* 10	* 10		* 5		destination ▼	

## 6.5. Add SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to Session Manager which includes Communication Manager and the Avaya SBCE. Navigate to **Routing → SIP Entities** in the left-hand navigation pane (**Section 6.1**) and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

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

- **Name:** Enter a descriptive name.
- **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity that is used for SIP signaling.
- **Type:** Enter **Session Manager** for Session Manager, **CM** for Communication Manager and **SIP Trunk** for the Avaya SBCE.
- **Adaptation:** This field is only present if **Type** is not set to **Session Manager**. If applicable, select the appropriate **Adaptation name** created in **Section 6.4** that will be applied to this entity.
- **Location:** Select the location that applies to the SIP entity being created. For the compliance test, all components were located in location **Location 1**.
- **Time Zone:** Select the time zone for the location above.

The following screen shows the addition of Session Manager. The IP address of the virtual SM-100 Security Module is entered for **FQDN or IP Address**.

The screenshot displays the 'SIP Entities' configuration page. The breadcrumb trail at the top reads 'Home / Elements / Routing / SIP Entities'. The page title is 'SIP Entity Details' with 'General' selected. There are 'Commit' and 'Cancel' buttons in the top right corner, along with a 'Help ?' link. The 'General' section contains the following fields: 'Name' (required, value: devcon-asm), 'FQDN or IP Address' (required, value: 10.32.24.235), 'Type' (dropdown menu, value: Session Manager), 'Notes' (text area, value: Session Manager for SP testing), 'Location' (dropdown menu, value: Location 1), 'Outbound Proxy' (dropdown menu), 'Time Zone' (dropdown menu, value: America/New\_York), and 'Credential name' (text area). The 'SIP Link Monitoring' section at the bottom has a 'SIP Link Monitoring' dropdown menu with the value 'Use Session Manager Configuration'.

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 can listen for SIP requests.
- **Protocol:** Transport protocol to be used with this port.
- **Default Domain:** The default domain associated with this port. For the compliance test, this was the enterprise SIP domain.

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

For the compliance test, four port entries were used. The first three are the standard ports used for SIP traffic: port 5060 for UDP/TCP and port 5061 for TLS. In addition, port 5062 defined in **Section 5.6** for use with service provider SIP traffic between Communication Manager and Session Manager was added to the list.

**Port**

TCP Failover port:

TLS Failover port:

4 Items [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	<input type="text" value="5060"/>	TCP <input type="button" value="v"/>	avaya.com <input type="button" value="v"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5060"/>	UDP <input type="button" value="v"/>	avaya.com <input type="button" value="v"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5061"/>	TLS <input type="button" value="v"/>	avaya.com <input type="button" value="v"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5062"/>	TCP <input type="button" value="v"/>	avaya.com <input type="button" value="v"/>	<input type="text"/>

Select : All, None

The following screen shows the addition of Communication Manager. In order for Session Manager to send SIP service provider traffic on a separate entity link to Communication Manager, this requires the creation of a separate SIP entity for Communication Manager other than the one created at Session Manager installation for use with all other SIP traffic. The **FQDN or IP Address** field is set to the IP address of Communication Manager. For the **Adaptation** field, select the adaptation module previously defined for dial plan digit manipulation in **Section 6.4**. The **Location** field is set to **Location 1** which is the location defined for the subnet where Communication Manager resides.

The screenshot shows a web interface for configuring SIP entities. The breadcrumb navigation at the top reads "Home / Elements / Routing / SIP Entities". The page title is "SIP Entity Details", and there are "Commit" and "Cancel" buttons in the top right corner. A "Help ?" link is also present. The configuration is organized into sections: "General", "Loop Detection", and "SIP Link Monitoring".

**SIP Entity Details**

**General**

- Name:** sp3-cm-2
- \* FQDN or IP Address:** 10.32.128.4
- Type:** CM
- Notes:** Princeton CM Trk 3
- Adaptation:** PRT-CM-TRK3
- Location:** Location 1
- Time Zone:** America/New\_York
- Override Port & Transport with DNS SRV:** ☐
- \* SIP Timer B/F (in seconds):** 4
- Credential name:**
- Call Detail Recording:** none

**Loop Detection**

- Loop Detection Mode:** Off

**SIP Link Monitoring**

- SIP Link Monitoring:** Use Session Manager Configuration

The following screen shows the addition of the Avaya SBCE. The **FQDN or IP Address** field is set to the IP address of its private network interface (see **Figure 1**). The **Location** field is set to **Location 1** which is the location defined for the subnet where the Avaya SBCE resides.

Home / Elements / Routing / SIP Entities

Help ?

SIP Entity Details

CommitCancel

General

\* Name: ASBCE

\* FQDN or IP Address: 10.32.128.18

Type: SIP Trunk

Notes: Avaya SBCE

Adaptation:

Location: Location 1

Time Zone: America/New\_York

Override Port & Transport with DNS SRV:

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

Credential name:

Call Detail Recording: egress

Loop Detection

Loop Detection Mode: Off

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

## 6.6. Add 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 for use only by service provider traffic and one to the Avaya SBCE. To add an Entity Link, navigate to **Routing → Entity Links** in the left-hand navigation pane (**Section 6.1**) and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Name:** Enter a descriptive name.
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the transport protocol used for this link. This must match the protocol used in the Communication Manager signaling group in **Section 5.6**.
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end. For the Communication Manager Entity Link, this must match the **Far-end Listen Port** defined on the Communication Manager signaling group in **Section 5.6**.
- **SIP Entity 2:** Select the name of the other system. For the Communication Manager Entity Link, select the Communication Manager SIP Entity defined in **Section 6.5**.
- **Port:** Port number on which the other system receives SIP requests from the Session Manager. For the Communication Manager Entity Link, this must match the **Near-end Listen Port** defined on the Communication Manager signaling group in **Section 5.6**.
- **Connection Policy:** Select **Trusted** from pull-down menu.

Click **Commit** to save.

The following screen illustrates the Entity Link to Communication Manager. The protocol and ports defined here must match the values used on the Communication Manager signaling group form in **Section 5.6**. For the compliance test, the TCP protocol was used but the recommended configuration is to use TLS.

Home / Elements / Routing / Entity Links

Entity Links

Commit Cancel

1 Item Refresh Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
<input type="checkbox"/>	* devcon-asm_sp3-cm	* devcon-asm	TCP	* 5062	* sp3-cm-2	* 5062	trusted	<input type="checkbox"/>

Select : All, None



The following screen illustrates the Entity Link to the Avaya SBCE.

Home / Elements / Routing / Entity Links

Entity Links

CommitCancel

Help ?

1 Item RefreshFilter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
<input type="checkbox"/>	*devcon-asm_ASBCE	*devcon-asm	TCP	*5060	*ASBCE	*5060	trusted	<input type="checkbox"/>

Select : All, None

## 6.7. Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.5**. Two routing policies 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 (**Section 6.1**) and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), 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 screens show the Routing Policies for Communication Manager and the Avaya SBCE.

Home / Elements / Routing / Routing Policies [Help ?](#)

**Routing Policy Details** [Commit](#) [Cancel](#)

**General**

\* **Name:**

**Disabled:** ☐

\* **Retries:**

**Notes:**

**SIP Entity as Destination**

[Select](#)

Name	FQDN or IP Address	Type	Notes
sp3-cm-2	10.32.128.4	CM	Princeton CM Trk 3

Home / Elements / Routing / Routing Policies

Routing Policy Details

CommitCancel

Help ?

General

\* Name:

ASBCE-route

Disabled:

☐

\* Retries:

0

Notes:

Outbound to ASBCE for SP testing

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
ASBCE	10.32.128.18	SIP Trunk	Avaya SBCE

## 6.8. Add 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 Cox Communications 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 (**Section 6.1**) and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

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**.

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

Two examples of the dial patterns used for the compliance test are shown below. The first example shows that numbers that begin with 1 and have a destination domain of **avaya.com** from **ALL** locations use route policy **ASBCE-route**.

Home / Elements / Routing / Dial Patterns
[Help ?](#)

Dial Pattern Details
Commit Cancel

General

\* Pattern: 1

\* Min: 11

\* Max: 11

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: avaya.com

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item Refresh Filter: Enable

<input type="checkbox"/>	Originating Location Name ▲	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-		ASBCE-route		<input type="checkbox"/>	ASBCE	Outbound to ASBCE for SP testing

Select : All, None

The second example shows that 10 digit numbers that start with **703555** to domain **avaya.com** and originating from **ALL** locations use route policy **sp3-cm Route 2**. These are the DID numbers assigned to the enterprise from Cox Communications. All other dial patterns used as part of the compliance test were configured in a similar manner.

Home / Elements / Routing / Dial Patterns
[Help ?](#)

Dial Pattern Details
Commit Cancel

General

\* Pattern: 703555

\* Min: 10

\* Max: 10

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: avaya.com

Notes: Cox DID numbers

Originating Locations and Routing Policies

Add Remove

1 Item Refresh Filter: Enable

<input type="checkbox"/>	Originating Location Name ▲	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-		sp3-cm Route 2		<input type="checkbox"/>	sp3-cm-2	

Select : All, None

## 6.9. Add/View 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 a Session Manager, from the **Home** page, navigate to **Elements → Session Manager → Session Manager Administration** in the left-hand navigation pane (**Section 6.1**) and click on the **New** button in the right pane (not shown). If the Session Manager already exists, select the appropriate Session Manager and 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 host name or IP address of the Session Manager.

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

Home / Elements / Session Manager [Help ?](#)

### View Session Manager [Return](#)

[General](#) | [Security Module](#) | [NIC Bonding](#) | [Monitoring](#) | [CDR](#) | [Personal Profile Manager \(PPM\) - Connection Settings](#) | [Event Server](#)  
[Expand All](#) | [Collapse All](#)

**General** ▼

**SIP Entity Name**

**Description**

**Management Access Point Host Name/IP**

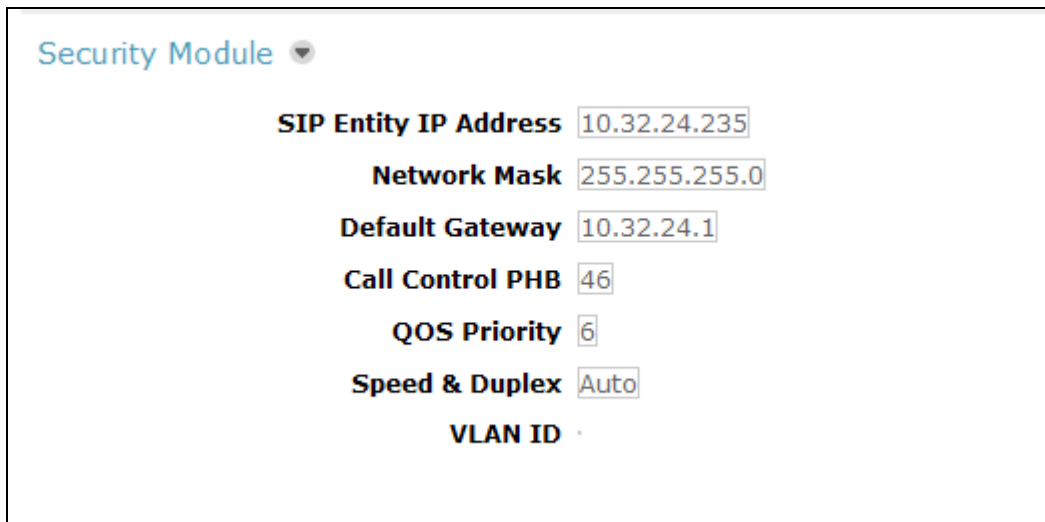
**Direct Routing to Endpoints**

**VMware Virtual Machine** ☐

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 Session Manager signaling interface.
- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager.
- **Default Gateway:** Enter the IP address of the default gateway for Session Manager.

Use default values for the remaining fields. Click **Save** (not shown) to add this Session Manager. The screen below shows the remaining Session Manager values used for the compliance test.



The screenshot displays the 'Security Module' configuration interface. It features a title bar with the text 'Security Module' and a dropdown arrow. Below the title, there are seven configuration fields, each with a label and a text input box. The fields are: 'SIP Entity IP Address' with the value '10.32.24.235', 'Network Mask' with '255.255.255.0', 'Default Gateway' with '10.32.24.1', 'Call Control PHB' with '46', 'QOS Priority' with '6', 'Speed & Duplex' with 'Auto', and 'VLAN ID' which is currently empty.

Field	Value
SIP Entity IP Address	10.32.24.235
Network Mask	255.255.255.0
Default Gateway	10.32.24.1
Call Control PHB	46
QOS Priority	6
Speed & Duplex	Auto
VLAN ID	



## 7. Configure Avaya Session Border Controller for Enterprise

This section describes the configuration of the Avaya SBCE. It is assumed that the initial installation of the Avaya SBCE has been completed including the assignment of a management IP address. The management interface **must** be provisioned on a different subnet than either the Avaya SBCE private or public network interfaces (e.g., A1 and B1). If the management interface has not been configured on a separate subnet, then contact your Avaya representative for guidance in correcting the configuration.

On all screens described in this section, it is to be assumed that parameters are left at their default values unless specified otherwise.

### 7.1. Access the Management Interface

Use a web browser to access the web interface by entering the URL **https://<ip-addr>**, where **<ip-addr>** is the management IP address assigned during installation. The Avaya SBCE login page will appear as shown below. Log in with appropriate credentials.



The image shows the login page for the Avaya Session Border Controller for Enterprise. On the left, there is a large red 'AVAYA' logo and the text 'Session Border Controller for Enterprise' in bold. On the right, under the heading 'Log In', there are input fields for 'Username:' and 'Password:', followed by 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.' This is followed by a statement: '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, it says 'All users must comply with all corporate instructions regarding the protection of information assets.' and '© 2011 - 2013 Avaya Inc. All rights reserved.'

After logging in, the Dashboard screen will appear as shown below. All configuration screens of the Avaya SBCE are accessed by navigating the menu tree in the left pane.

Alarms **1**IncidentsStatisticsLogsDiagnosticsUsersSettingsHelpLog Out

Session Border Controller for EnterpriseAVAYA

DashboardAdministrationBackup/RestoreSystem ManagementGlobal ParametersGlobal ProfilesSIP ClusterDomain PoliciesTLS ManagementDevice Specific Settings

Dashboard

Information

System Time01:42:35 PM GMTRefreshVersion6.2.0.Q36Build DateWed May 22 22:52:47 UTC 2013

Alarms (past 24 hours)None found.

Installed Devices

EMS1sp-ucsec1

Incidents (past 24 hours)None found.

Add

Notes

No notes found.

CTM; Reviewed:  
SPOC 10/14/2013

Solution & Interoperability Test Lab Application Notes  
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## 7.2. Verify Network Configuration and Enable Interfaces

To view the network information provided during installation, navigate to **System Management**. In the right pane, click **View** highlighted below.

**Session Border Controller for Enterprise** AVAYA

Dashboard  
Administration  
Backup/Restore  
**System Management**  
‣ Global Parameters  
‣ Global Profiles  
‣ SIP Cluster  
‣ Domain Policies  
‣ TLS Management  
‣ Device Specific Settings

**System Management**

Devices Updates SSL VPN Licensing

Device Name (Serial Number)	Management IP	Version	Status					
sp-ucsec1 (IFCS31030012)	10.32.101.10	6.2.0.Q36	Commissioned	Reboot	Shutdown	Restart Application	<b>View</b>	Edit Delete

A System Information page will appear showing the information provided during installation. In the **Appliance Name** field is the name of the device (**sp-ucsec1**). This name will be referenced in other configuration screens. The two **Network Configuration** entries highlighted below are the only two IP addresses that are directly related to the SIP trunking solution described in these Application Notes. Interfaces **A1** and **B1** represent the private and public interfaces of the Avaya SBCE. Each of these interfaces must be enabled after installation.

**System Information: sp-ucsec1**

**General Configuration**

Appliance Name sp-ucsec1  
Box Type SIP  
Deployment Mode Proxy

**Device Configuration**

HA Mode No  
Two Bypass Mode No

**Network Configuration**

IP	Public IP	Netmask	Gateway	Interface
10.32.128.18	10.32.128.18	255.255.255.0	10.32.128.254	A1
192.168.96.228	192.168.96.228	255.255.255.224	192.168.96.254	B1
192.168.96.230	192.168.96.230	255.255.255.224	192.168.96.254	B1
192.168.96.229	192.168.96.229	255.255.255.224	192.168.96.254	B1
10.32.128.19	10.32.128.19	255.255.255.0	10.32.128.254	A1

**DNS Configuration**


Primary DNS 10.32.128.200  
Secondary DNS  
DNS Location DMZ  
DNS Client IP 10.32.128.18

**Management IP(s)**

IP 10.32.101.10

To enable the interfaces, first navigate to **Device Specific Settings** → **Network Management** in the left pane and select the device being managed in the center pane. In the right pane, click on the **Interface Configuration** tab. Verify the **Administrative Status** is **Enabled** for both the **A1** and **B1** interfaces. If not, click **Toggle** to enable the interface.

## Session Border Controller for Enterprise



Dashboard

Administration

Backup/Restore

System Management

▸ Global Parameters

▸ Global Profiles

▸ SIP Cluster

▸ Domain Policies

▸ TLS Management

▸ Device Specific Settings

**Network Management**

Media Interface

Network Management: sp-ucsec1

Devices

sp-ucsec1

Network Configuration

Interface Configuration

Name	Administrative Status	
A1	Enabled	<a href="#">Toggle</a>
A2	Disabled	<a href="#">Toggle</a>
B1	Enabled	<a href="#">Toggle</a>
B2	Disabled	<a href="#">Toggle</a>

## 7.3. Signaling Interface

A signaling interface defines an IP address, protocols and listen ports that the Avaya SBCE can use for signaling. Create a signaling interface for both the internal and external sides of the Avaya SBCE.

To create a new interface, navigate to **Device Specific Settings → Signaling Interface** in the left pane. In the center pane, select the Avaya SBCE device (**sp-ucsec1**) to be managed. In the right pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new interface, followed by a series of pop-up windows in which the interface parameters can be configured. Once complete, the settings are shown in the far right pane.

For the compliance test, signaling interface **Int\_Sig\_Intf** was created for the Avaya SBCE internal interface and signaling interface **Ext\_Sig\_Intf** was created for the Avaya SBCE external interface. Each is highlighted below. When configuring the interfaces, configure the parameters as follows:

- Set **Name** to a descriptive name.
- For the internal interface, set the **Signaling IP** to the IP address associated with the private interface (A1) defined in **Section 7.2**. For the external interface, set the **Signaling IP** to the IP address associated with the public interface (B1) defined in **Section 7.2**.
- In the **UDP Port**, **TCP Port** and **TLS Port** fields, enter the port the Avaya SBCE will listen on for each transport protocol. For the internal interface, the Avaya SBCE was configured to listen for TCP on port 5060. For the external interface, the Avaya SBCE was configured to listen for UDP or TCP on port 5060. Since Cox Communications uses UDP on port 5060, it would have been sufficient to simply configure the Avaya SBCE for UDP.

**Session Border Controller for Enterprise** AVAYA

Dashboard  
Administration  
Backup/Restore  
System Management  
‣ Global Parameters  
‣ Global Profiles  
‣ SIP Cluster  
‣ Domain Policies  
‣ TLS Management  
‣ Device Specific Settings  
  Network Management  
  Media Interface  
  **Signaling Interface**

**Signaling Interface: sp-ucsec1**

Devices  
sp-ucsec1

**Signaling Interface** Add

Name	Signaling IP	TCP Port	UDP Port	TLS Port	TLS Profile		
Int_Sig_Intf	10.32.128.18	5060	---	---	None	Edit	Delete
Ext_Sig_Intf	192.168.96.228	5060	5060	---	None	Edit	Delete
RW_Sig_Outside_229	192.168.96.229	5060	---	5061	AvayaSBCServer	Edit	Delete
RW_Sig_Inside_19	10.32.128.19	5060	---	5061	AvayaSBCServer	Edit	Delete

## 7.4. Media Interface

A media interface defines an IP address and port range for transmitting media. Create a media interface for both the internal and external sides of the Avaya SBCE.

To create a new interface, navigate to **Device Specific Settings → Media Interface** in the left pane. In the center pane, select the Avaya SBCE device (**sp-ucsec1**) to be managed. In the right pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new interface, followed by a series of pop-up windows in which the interface parameters can be configured. Once complete, the settings are shown in the far right pane.

For the compliance test, media interface **Int\_Media\_Intf** was created for the Avaya SBCE internal interface and media interface **Ext\_Media\_Intf** was created for the Avaya SBCE external interface. Each is highlighted below. When configuring the interfaces, configure the parameters as follows:

- Set **Name** to a descriptive name.
- For the internal interface, set the **Media IP** to the IP address associated with the private interface (A1) defined in **Section 7.2**. For the external interface, set the **Media IP** to the IP address associated with the public interface (B1) defined in **Section 7.2**.
- Set **Port Range** to a range of ports acceptable to both the Avaya SBCE and the far-end. For the compliance test, the default port range was used for both interfaces.

**Session Border Controller for Enterprise** AVAYA

Dashboard  
Administration  
Backup/Restore  
System Management  
‣ Global Parameters  
‣ Global Profiles  
‣ SIP Cluster  
‣ Domain Policies  
‣ TLS Management  
‣ **Device Specific Settings**  
  Network Management  
  **Media Interface**  
  Signaling Interface  
  Signaling Forking

**Media Interface: sp-ucsec1**

Devices  
sp-ucsec1

**Media Interface**

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

[Add](#)

Name	Media IP	Port Range	Edit	Delete
Int_Media_Intf	10.32.128.18	35000 - 40000	<a href="#">Edit</a>	<a href="#">Delete</a>
Ext_Media_Intf	192.168.96.228	35000 - 40000	<a href="#">Edit</a>	<a href="#">Delete</a>
RW_Med_Outside_229	192.168.96.229	35000 - 40000	<a href="#">Edit</a>	<a href="#">Delete</a>
RW_Med_Inside_19	10.32.128.19	35000 - 40000	<a href="#">Edit</a>	<a href="#">Delete</a>

## 7.5. Server Interworking

A server interworking profile defines a set of parameters that aid in interworking between the Avaya SBCE and a connected server. Create a server interworking profile for Session Manager and the service provider SIP server. These profiles will be applied to the appropriate server in **Section 7.7.1** and **7.7.2**.

To create a new profile, navigate to **Global Profiles → Server Interworking** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by a series of pop-up windows in which the profile parameters can be configured. Once complete, the settings are shown in the far right pane. Alternatively, a new profile may be created by selecting an existing profile in the center pane and clicking the **Clone** button in the right pane. This will create a copy of the selected profile which can then be edited as needed. To view the settings of an existing profile, select the profile from the center pane. The settings will appear in the right pane.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The left navigation pane shows the hierarchy: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, and Server Interworking (highlighted). The main content area is titled "Interworking Profiles: cs2100" and includes an "Add" button and a "Clone" button. A warning message states: "It is not recommended to edit the defaults. Try cloning or adding a new profile instead." Below this, there are tabs for "General", "Timers", "URI Manipulation", "Header Manipulation", and "Advanced". The "General" tab is active, showing a table of settings:

General	
Hold Support	RFC3264
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	No

### 7.5.1. Server Interworking – Session Manager

For the compliance test, server interworking profile **Avaya-SM** was created for Session Manager by cloning the existing profile **avaya-ru**. The values highlighted below are values that differ from the default. In addition, T.38 is set to **No** since Cox Communications does not support T.38 fax. The **General** tab parameters are shown below.

General	Timers	URI Manipulation	Header Manipulation	Advanced
<b>General</b>				
Hold Support	RFC2543			
180 Handling	None			
181 Handling	None			
182 Handling	None			
183 Handling	None			
Refer Handling	No			
3xx Handling	No			
Diversion Header Support	No			
Delayed SDP Handling	No			
T.38 Support	No			
URI Scheme	SIP			
Via Header Format	RFC3261			
<b>Privacy</b>				
Privacy Enabled	No			
User Name				
P-Asserted-Identity	No			
P-Preferred-Identity	No			
Privacy Header				
<b>DTMF</b>				
DTMF Support	None			
<a href="#">Edit</a>				

The **Timers**, **URI Manipulation**, **Header Manipulation** tabs have no entries.



The **Advanced** tab parameters are shown below.

General	Timers	URI Manipulation	Header Manipulation	Advanced
Record Routes		Both		
Topology Hiding: Change Call-ID		No		
Call-Info NAT		No		
Change Max Forwards		Yes		
Include End Point IP for Context Lookup		No		
OCS Extensions		No		
AVAYA Extensions		Yes		
NORTEL Extensions		No		
Diversion Manipulation		No		
Metaswitch Extensions		No		
Reset on Talk Spurt		No		
Reset SRTP Context on Session Refresh		No		
Has Remote SBC		Yes		
Route Response on Via Port		No		
Cisco Extensions		No		

[Edit](#)

### 7.5.2. Server Interworking – Cox Communications

For the compliance test, server interworking profile **SP-General** was created for the Cox Communications SIP server. When creating the profile, the default values were used for all parameters including the setting of **T.38 Support** to **No**. The **General** tab parameters are shown below.

General	Timers	URI Manipulation	Header Manipulation	Advanced
<b>General</b>				
Hold Support	NONE			
180 Handling	None			
181 Handling	None			
182 Handling	None			
183 Handling	None			
Refer Handling	No			
3xx Handling	No			
Diversion Header Support	No			
Delayed SDP Handling	No			
T.38 Support	No			
URI Scheme	SIP			
Via Header Format	RFC3261			
<b>Privacy</b>				
Privacy Enabled	No			
User Name				
P-Asserted-Identity	No			
P-Preferred-Identity	No			
Privacy Header				
<b>DTMF</b>				
DTMF Support	None			
<a href="#">Edit</a>				

The **Timers**, **URI Manipulation**, **Header Manipulation** tabs have no entries.

The **Advanced** tab parameters are shown below.

<b>General</b>	<b>Timers</b>	<b>URI Manipulation</b>	<b>Header Manipulation</b>	<b>Advanced</b>
Record Routes				Both
Topology Hiding: Change Call-ID				Yes
Call-Info NAT				No
Change Max Forwards				Yes
Include End Point IP for Context Lookup				No
OCS Extensions				No
AVAYA Extensions				No
NORTEL Extensions				No
Diversion Manipulation				No
Metaswitch Extensions				No
Reset on Talk Spurt				No
Reset SRTP Context on Session Refresh				No
Has Remote SBC				Yes
Route Response on Via Port				No
Cisco Extensions				No
<input type="button" value="Edit"/>				

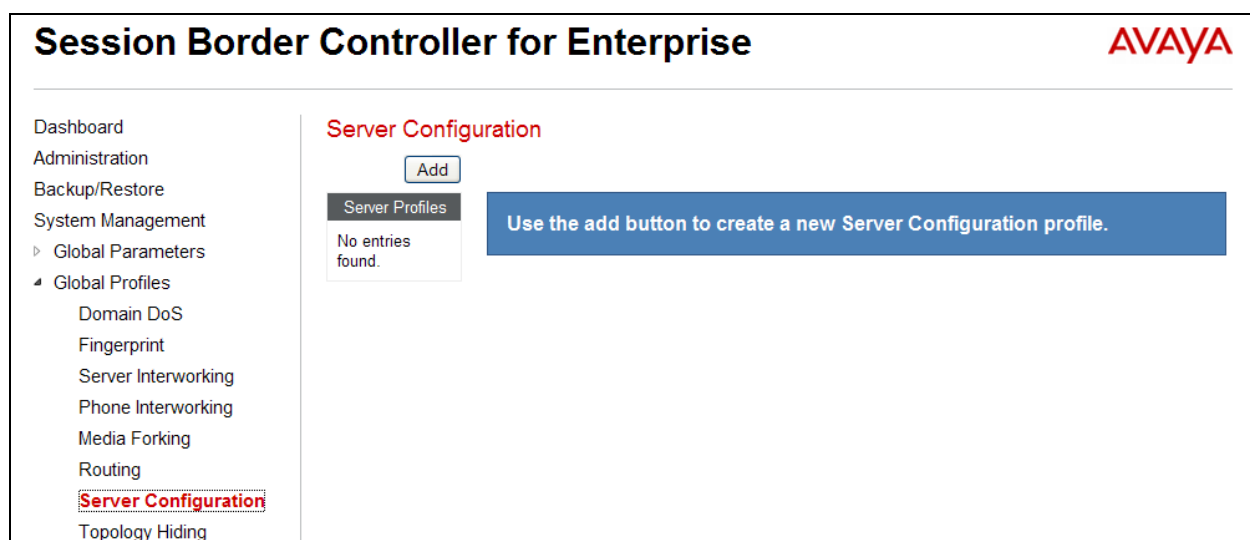
## 7.6. Signaling Manipulation

Signaling manipulation scripts provides for the manipulation of SIP messages which cannot be done by other configuration within the Avaya SBCE. It was not necessary to create any signaling manipulation scripts for interoperability with Cox Communications.

## 7.7. Server Configuration

A server configuration profile defines the attributes of the physical server. Create a server configuration profile for Session Manager and the service provider SIP server.

To create a new profile, navigate to **Global Profiles → Server Configuration** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by a series of pop-up windows in which the profile parameters can be configured. Once complete, the settings are shown in the far right pane. To view the settings of an existing profile, select the profile from the center pane. The settings will appear in the right pane.



### 7.7.1. Server Configuration – Session Manager

For the compliance test, server configuration profile **Avaya-SM** was created for Session Manager. When creating the profile, configure the **General** tab parameters as follows:

- Set **Server Type** to **Call Server**.
- Set **IP Addresses / FQDNs** to the IP address of the Session Manager signaling interface.
- Set **Supported Transports** to the transport protocol used for SIP signaling between Session Manager and the Avaya SBCE.
- Set the **TCP Port** to the port Session Manager will listen on for SIP requests from the Avaya SBCE.

The screenshot shows the 'General' tab of a configuration interface. At the top right are buttons for 'Rename', 'Clone', and 'Delete'. Below these are tabs for 'General', 'Authentication', 'Heartbeat', and 'Advanced'. The 'General' tab is active, displaying a table with the following configuration:

Server Type	Call Server
IP Addresses / FQDNs	10.32.24.235
Supported Transports	TCP
TCP Port	5060

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

On the **Advanced** tab, check **Enable Grooming** and set the **Interworking Profile** field to the interworking profile for Session Manager defined in **Section 7.5.1**.

The screenshot shows the 'Advanced' tab of the same configuration interface. The 'Advanced' tab is active, displaying a table with the following configuration:

Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input checked="" type="checkbox"/>
Interworking Profile	Avaya-SM
Signaling Manipulation Script	None
TCP Connection Type	SUBID

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

### 7.7.2. Server Configuration – Cox Communications

For the compliance test, server configuration profile **SP-Cox** was created for Cox Communications. When creating the profile, configure the **General** tab parameters as follows:

- Set **Server Type** to **Trunk Server**.
- Set **IP Addresses / FQDNs** to the IP address of the Cox Communications SIP server (e.g., **192.168.222.147** as shown in **Figure 1** in **Section 3**).
- Set **Supported Transports** to the transport protocol used for SIP signaling between Cox Communications and the Avaya SBCE. In the compliance test, UDP was tested.
- Set the **UDP Port** to the standard SIP port of 5060. This is the port Cox Communications will listen on for SIP requests from the Avaya SBCE.

The screenshot shows the 'General' tab of a server configuration interface. At the top right are buttons for 'Rename', 'Clone', and 'Delete'. Below the tabs are four configuration rows: 'Server Type' set to 'Trunk Server', 'IP Addresses / FQDNs' set to '192.168.222.147', 'Supported Transports' set to 'UDP', and 'UDP Port' set to '5060'. An 'Edit' button is located at the bottom center of the configuration area.

General	Authentication	Heartbeat	Advanced
Server Type			
Trunk Server			
IP Addresses / FQDNs			
192.168.222.147			
Supported Transports			
UDP			
UDP Port			
5060			

Edit

On the **Advanced** tab, set the **Interworking Profile** field to the interworking profile for Cox Communications defined in **Section 7.5.2**.

The screenshot shows the 'Advanced' tab of the same server configuration interface. At the top right are buttons for 'Rename', 'Clone', and 'Delete'. Below the tabs are five configuration rows: 'Enable DoS Protection' with an unchecked checkbox, 'Enable Grooming' with an unchecked checkbox, 'Interworking Profile' set to 'SP-General', 'Signaling Manipulation Script' set to 'None', and 'UDP Connection Type' set to 'SUBID'. An 'Edit' button is located at the bottom center of the configuration area.

General	Authentication	Heartbeat	Advanced
Enable DoS Protection			
<input type="checkbox"/>			
Enable Grooming			
<input type="checkbox"/>			
Interworking Profile			
SP-General			
Signaling Manipulation Script			
None			
UDP Connection Type			
SUBID			

Edit

## 7.8. Application Rules

An application rule defines the allowable SIP applications and associated parameters. An application rule is one component of the larger endpoint policy group defined in **Section 7.11**. For the compliance test, the predefined **default-trunk** application rule (shown below) was used for both Session Manager and the Cox Communications SIP server.

To view an existing rule, navigate to **Domain Policies** → **Application Rules** in the left pane. In the center pane, select the rule (e.g., **default-trunk**) to be viewed.

### Session Border Controller for Enterprise

Dashboard

Administration

Backup/Restore

System Management

▸ Global Parameters

▸ Global Profiles

▸ SIP Cluster

▸ Domain Policies

Application Rules

Border Rules

Media Rules

Security Rules

Signaling Rules

Time of Day Rules

End Point Policy Groups

Session Policies

▸ TLS Management

Application Rules: default-trunk

Add

Filter By Device...

Clone

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

Application Rule

Application Type	In	Out	Maximum Concurrent Sessions	Maximum Sessions Per Endpoint
Voice	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	2000	2000
Video	<input type="checkbox"/>	<input type="checkbox"/>		
IM	<input type="checkbox"/>	<input type="checkbox"/>		
Miscellaneous				
CDR Support	None			
RTCP Keep-Alive	No			

Edit

## 7.9. Media Rules

A media rule defines the processing to be applied to the selected media. A media rule is one component of the larger endpoint policy group defined in **Section 7.11**. For the compliance test, the predefined **default-low-med** media rule (shown below) was used for both Session Manager and the Cox Communications SIP server.

To view an existing rule, navigate to **Domain Policies** → **Media Rules** in the left pane. In the center pane, select the rule (e.g., **default-low-med**) to be viewed.

Each of the tabs of the **default-low-med** media rule containing data is shown below.

The **Media NAT** tab has no entries.

The screenshot shows the 'Session Border Controller for Enterprise' web interface. On the left is a navigation menu with 'Media Rules' highlighted. The main area is titled 'Media Rules: default-low-med' and includes an 'Add' button, a 'Filter By Device...' dropdown, and a 'Clone' button. A warning message states: 'It is not recommended to edit the defaults. Try cloning or adding a new rule instead.' Below this are tabs for 'Media NAT', 'Media Encryption', 'Media Anomaly', 'Media Silencing', and 'Media QoS'. The 'Media NAT' tab is active, showing a 'Media NAT' section with the text 'Learn Media IP dynamically' and an 'Edit' button.

The **Media Encryption** tab indicates that no encryption was used.

This screenshot shows the 'Media Encryption' configuration tab. It features a header with tabs for 'Media NAT', 'Media Encryption' (which is selected), 'Media Anomaly', 'Media Silencing', and 'Media QoS'. The configuration is organized into three sections: 'Audio Encryption', 'Video Encryption', and 'Miscellaneous'. Under 'Audio Encryption', 'Preferred Formats' is set to 'RTP' and 'Interworking' is checked with a green checkmark. Under 'Video Encryption', 'Preferred Formats' is also 'RTP' and 'Interworking' is checked. In the 'Miscellaneous' section, 'Capability Negotiation' is unchecked. An 'Edit' button is located at the bottom of the configuration area.



The **Media Anomaly** tab shows **Media Anomaly Detection** was enabled.

Media NAT	Media Encryption	<b>Media Anomaly</b>	Media Silencing	Media QoS
Media Anomaly Detection <input checked="" type="checkbox"/>				
Detect RTP Injection Attack <input checked="" type="checkbox"/>				
Asymmetric RTP <input type="checkbox"/>				
Action		Alert		
<a href="#">Edit</a>				

The **Media Silencing** tab has no entries.

The **Media QoS** settings are shown below.

Media NAT	Media Encryption	Media Anomaly	Media Silencing	<b>Media QoS</b>
Media QoS Reporting				
RTCP Enabled		<input type="checkbox"/>		
Media QoS Marking				
Enabled		<input type="checkbox"/>		
<a href="#">Edit</a>				

## 7.10. Signaling Rules

A signaling rule defines the processing to be applied to the selected signaling traffic. A signaling rule is one component of the larger endpoint policy group defined in **Section 7.11**. A specific signaling rule was created for Session Manager. The Cox Communications SIP server used the **default** rule.

To create a new rule, navigate to **Domain Policies → Signaling Rules** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new rule, followed by a series of pop-up windows in which the rule parameters can be configured. Once complete, the settings are shown in the far right pane. To view the settings of an existing rule, select the rule from the center pane. The settings will appear in the right pane.

The screenshot displays the 'Session Border Controller for Enterprise' web interface. The left navigation pane shows the hierarchy: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, SIP Cluster, Domain Policies (expanded), and its sub-items: Application Rules, Border Rules, Media Rules, Security Rules, **Signaling Rules** (highlighted), Time of Day Rules, End Point Policy Groups, Session Policies, TLS Management, and Device Specific Settings. The main content area is titled 'Signaling Rules: default' and includes an 'Add' button, a 'Filter By Device...' dropdown, and a 'Clone' button. A warning message states: 'It is not recommended to edit the defaults. Try cloning or adding a new rule instead.' Below this, there are tabs for 'General', 'Requests', 'Responses', 'Request Headers', 'Response Headers', and 'Signaling QoS'. The 'General' tab is active, showing settings for 'Inbound' and 'Outbound' traffic. For both directions, 'Requests', 'Non-2XX Final Responses', 'Optional Request Headers', and 'Optional Response Headers' are all set to 'Allow'. At the bottom, the 'Content-Type Policy' section has 'Enable Content-Type Checks' checked, and the 'Action' is set to 'Allow' for both 'Multipart Action' and 'Allow'.

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

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

Content-Type Policy			
Enable Content-Type Checks <input checked="" type="checkbox"/>			
Action	Allow	Multipart Action	Allow

### 7.10.1. Signaling Rules – Session Manager

For the compliance test, signaling rule **SessMgr\_SigRules** was created for Session Manager to prevent proprietary headers in the SIP messages sent from the Session Manager from being propagated to Cox Communications. These headers may contain internal addresses or other information about the internal network. Select this rule in the center pane, then select the **Request Headers** tab to view the manipulations performed on request messages such as the initial INVITE or UPDATE message.

Click **Add** to create the rule. Keep the default values on all tabs except the **Request Headers** and **Response Headers** tabs. Select the **Request Headers** tab to create the manipulations performed on request messages such as the initial INVITE or UPDATE message. An entry is created by clicking the **Add In Header Control** or **Add Out Header Control** button depending on the direction (relative to the Avaya SBCE) of the message to be modified. Entries were created to perform the following actions:

1. Removes the **AV-Correlation-ID** header from **INVITE** messages in the **IN** direction (Session Manager to Avaya SBCE).
2. Removes the **Endpoint-View** header from **ALL** messages in the **IN** direction.

General Requests Responses Request Headers Response Headers Signaling QoS							
				Add In Header Control		Add Out Header Control	
Row	Header Name	Method Name	Header Criteria	Action	Proprietary	Direction	
1	AV-Correlation-ID	INVITE	Forbidden	Remove Header	Yes	IN	Edit Delete
2	Endpoint-View	ALL	Forbidden	Remove Header	Yes	IN	Edit Delete

Similarly, manipulations can be performed on SIP response messages. These can be created by selecting the **Response Header** tab as shown below. Entries were created in the same manner as was done on the **Request Headers** tab. The entries shown perform the following actions:

1. Removes the **Endpoint-View** header from any **2XX** response to **ALL** messages in the **IN** direction (Session Manager to Avaya SBCE).
2. Removes the **Endpoint-View** header from the **1XX** response to an **INVITE** message in the **IN** direction.

General Requests Responses Request Headers Response Headers Signaling QoS									
Add In Header Control Add Out Header Control									
Row	Header Name	Response Code	Method Name	Header Criteria	Action	Proprietary	Direction		
1	Endpoint-View	2XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
2	Endpoint-View	1XX	INVITE	Forbidden	Remove Header	Yes	IN	Edit	Delete

## 7.10.2. Signaling Rules – Cox Communications

The predefined **default** signaling rule (shown below) was used for the Cox Communications SIP server. The **General** tab settings are shown below.

Session Border Controller for Enterprise

AVAYA

Dashboard
Administration
Backup/Restore
System Management
Global Parameters
Global Profiles
SIP Cluster
Domain Policies
Application Rules
Border Rules
Media Rules
Security Rules
**Signaling Rules**
Time of Day Rules
End Point Policy Groups
Session Policies
TLS Management
Device Specific Settings

Signaling Rules: default

Add Filter By Device... Clone

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

General Requests Responses Request Headers Response Headers Signaling QoS

Inbound

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

Outbound

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

Content-Type Policy

Enable Content-Type Checks ☒

Action Allow Multipart Action Allow

The **Requests**, **Responses**, **Request Headers**, and **Response Headers** tabs have no entries. The **Signaling QoS** tab is shown below.

General	Requests	Responses	Request Headers	Response Headers	Signaling QoS						
<div>Signaling QoS <input checked="" type="checkbox"/></div> <table><tr><td>QoS Type</td><td>TOS</td></tr><tr><td>Precedence</td><td>Routine</td></tr><tr><td>ToS</td><td>Minimize Delay</td></tr></table> <div>Edit</div>						QoS Type	TOS	Precedence	Routine	ToS	Minimize Delay
QoS Type	TOS										
Precedence	Routine										
ToS	Minimize Delay										

## 7.11. Endpoint Policy Groups

An endpoint policy group is a set of policies that will be applied to traffic between the Avaya SBCE and a signaling endpoint (connected server). Thus, an endpoint policy group must be created for Session Manager and the service provider SIP server. The endpoint policy group is applied to the traffic as part of the endpoint flow defined in **Section 7.14**.

To create a new group, navigate to **Domain Policies → End Point Policy Groups** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new group, followed by a series of pop-up windows in which the group parameters can be configured. Once complete, the settings are shown in the far right pane. To view the settings of an existing group, select the group from the center pane. The settings will appear in the right pane.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The left navigation pane includes: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, SIP Cluster, Domain Policies (selected), Application Rules, Border Rules, Media Rules, Security Rules, Signaling Rules, Time of Day Rules, End Point Policy Groups, and Groups. The main content area is titled 'Policy Groups: default-low' and features an 'Add' button and a 'Filter By Device...' dropdown. A message states: 'It is not recommended to edit the defaults. Try adding a new group instead.' Below this is a table of policy groups. The 'Policy Group' tab is active, showing a table with columns: Order, Application, Border, Media, Security, Signaling, Time of Day, and actions (Edit, Clone). The table contains one row with the following values: Order 1, Application default, Border default, Media default-low-med, Security default-low, Signaling default, Time of Day default. Buttons for 'Summary' and 'Add' are visible in the top right of the table area.

Order	Application	Border	Media	Security	Signaling	Time of Day	
1	default	default	default-low-med	default-low	default	default	Edit Clone

### 7.11.1. Endpoint Policy Group – Session Manager

For the compliance test, endpoint policy group **TM-Test** was created for Session Manager. Default values were used for each of the rules which comprise the group with the exception of **Application** and **Signaling**. For **Application**, enter the application rule described in **Section 7.8**. For **Signaling**, enter the signaling rule created in **Section 7.10.1**. The details of the default settings for **Media** are showed in **Section 7.9**.

The screenshot shows the 'Policy Group' configuration window. It includes a 'Summary' button and an 'Add' button. The table below shows the configuration for the 'TM-Test' group.

Order	Application	Border	Media	Security	Signaling	Time of Day	
1	default-trunk	default	default-low-med	default-low	SessMgr_SigRules	default	Edit Clone

For the compliance test, endpoint policy group **Cox-Policy-Grp** was created for the Cox Communications SIP server. Default values were used for each of the rules which comprise the group with the exception of **Application**. For **Application**, enter the application rule described in **Section 7.8**. The details of the default settings for **Media** and **Signaling** are showed in **Section 7.9** and **Section 7.10.2** respectively.

## 7.12. Routing

To create a new profile, navigate to **Global Profiles → Routing** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by a series of pop-up windows in which the profile parameters can be configured. Once complete, the settings are shown in the far right pane. To view the settings of an existing profile, select the profile from the center pane. The settings will appear in the right pane.

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CoxC63S63SBCE62

### 7.12.1. Routing – Session Manager

For the compliance test, routing profile **To-PrtSM** was created for Session Manager. When creating the profile, configure the parameters as follows:

- Set the **URI Group** to the wild card \* to match on any URI.
- Set the **Next Hop Server 1** field to the IP address of Session Manager signaling interface.
- Enable **Next Hop Priority**.
- Set the **Outgoing Transport** field to **TCP**.

View Routing Rule		X
Priority	1	
URI Group	*	
Next Hop Server 1	10.32.24.235	
Next Hop Server 2	---	
Next Hop Priority	<input checked="" type="checkbox"/>	
NAPTR	<input type="checkbox"/>	
SRV	<input type="checkbox"/>	
Next Hop in Dialog	<input type="checkbox"/>	
Ignore Route Header	<input type="checkbox"/>	
Outgoing Transport	TCP	



### 7.12.2. Routing – Cox Communications

For the compliance test, routing profile **To-Trunks** was created for Cox Communications. When creating the profile, configure the parameters as follows:

- Set the **URI Group** to the wild card \* to match on any URI.
- Set the **Next Hop Server 1** field to the IP address of the Cox Communications SIP server.
- Enable **Next Hop Priority**.
- Set the **Outgoing Transport** field to **UDP**.

View Routing Rule		X
Priority	1	
URI Group	*	
Next Hop Server 1	192.168.222.147	
Next Hop Server 2	---	
Next Hop Priority	<input checked="" type="checkbox"/>	
NAPTR	<input type="checkbox"/>	
SRV	<input type="checkbox"/>	
Next Hop in Dialog	<input type="checkbox"/>	
Ignore Route Header	<input type="checkbox"/>	
Outgoing Transport	UDP	

## 7.13. Topology Hiding

Topology hiding allows the host part of some SIP message headers to be modified in order to prevent private network information from being propagated to the untrusted public network. It can also be used as an interoperability tool to adapt the host portion of these same headers to meet the requirements of the connected servers. The topology hiding profile is applied as part of the endpoint flow in **Section 7.14**.

To create a new profile, navigate to **Global Profiles → Topology Hiding** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by a pop-up window in which a header can be selected and configured. Additional headers can be added in this window. Once complete, the settings are shown in the far right pane. To view the settings of an existing profile (e.g., **default**), select the profile from the center pane. The settings will appear in the right pane.

### Session Border Controller for Enterprise

AVAYA

Dashboard

Administration

Backup/Restore

System Management

▸ Global Parameters

▾ Global Profiles

Domain DoS

Fingerprint

Server Interworking

Phone Interworking

Media Forking

Routing

Server Configuration

**Topology Hiding**

Signaling Manipulation

URI Groups

Topology Hiding Profiles: default

AddClone

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

Topology Hiding

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

Edit

Topology Hiding Profiles

default

cisco\_th\_profile

### 7.13.1. Topology Hiding – Session Manager

For the compliance test, topology hiding profile **PRT-Domain** was created for Session Manager. This profile will be applied to traffic from the Avaya SBCE to Session Manager. When creating the profile, configure the parameters as follows:

- Set **Header** to the header whose host part of the URI is to be modified.
- Set **Criteria** to **IP/Domain** to indicate that the host part should be modified if it is an IP address or a domain.
- Set **Replace Action** to **Auto** for all headers except **Request-Line**, **From** and **To** which should be set to **Overwrite**.
- For those headers to be overwritten, the **Overwrite Value** is set to the enterprise domain (**avaya.com**).

Topology Hiding			
Header	Criteria	Replace Action	Overwrite Value
Request-Line	IP/Domain	Overwrite	avaya.com
Via	IP/Domain	Auto	---
To	IP/Domain	Overwrite	avaya.com
SDP	IP/Domain	Auto	---
From	IP/Domain	Overwrite	avaya.com
Record-Route	IP/Domain	Auto	---
<div>Edit</div>			

### 7.13.2. Topology Hiding – Cox Communications

For the compliance test, topology hiding profile **SP-General** was created for Cox Communications. This profile will be applied to traffic from the Avaya SBCE to Cox Communications. When creating the profile, configure the parameters as follows:

- Set **Header** to the header whose host part of the URI is to be modified.
- Set **Criteria** to **IP/Domain** to indicate that the host part should be modified if it is an IP address or a domain.
- Set **Replace Action** to **Auto** for all headers.

**Topology Hiding**

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

Edit

## 7.14. End Point Flows

Endpoint flows are used to determine the signaling endpoints involved in a call in order to apply the appropriate policies. When a packet arrives at the Avaya SBCE, the content of the packet (IP addresses, URIs, etc) is used to determine which flow it matches. Once the flow is determined, the flow points to policies and profiles which control processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for the destination endpoint are applied. Thus, two flows are involved in every call: the source endpoint flow and the destination endpoint flow. In the case of the compliance test, the signaling endpoints are Session Manager and the service provider SIP server.

To create a new flow for a server endpoint, navigate to **Device Specific Settings → End Point Flows** in the left pane. In the center pane, select the Avaya SBCE device (**sp-ucsec1**) to be managed. In the right pane, select the **Server Flows** tab and click the **Add** button. A pop-up window (not shown) will appear requesting the name of the new flow and the flow parameters. Once complete, the settings are shown in the far right pane.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The left sidebar contains a navigation menu with options like Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, SIP Cluster, Domain Policies, TLS Management, and Device Specific Settings. Under Device Specific Settings, End Point Flows is selected. The main content area is titled 'End Point Flows: sp-ucsec1' and features two tabs: 'Subscriber Flows' and 'Server Flows'. The 'Server Flows' tab is active, showing a table of server configurations. Above the table is a button to 'Add' a new flow and a section for 'Server Configuration: Avaya-SM' with an 'Update' button. The table lists two flows with their respective priorities, names, URI groups, received and signaling interfaces, end point policy groups, and routing profiles.

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile	View	Close
1	Avaya-SM	*	Ext_Sig_Intf	Int_Sig_Intf	TM_Test	To_Trunks	View	Close
2	RW-Avaya-SM	*	RW_Sig_Outside_229	RW_Sig_Inside_19	Remote_User_SM	default	View	Close

### 7.14.1. End Point Flow – Session Manager

For the compliance test, endpoint flow **Avaya-SM** was created for Session Manager. All traffic from Session Manager will match this flow as the source flow and use the specified **Routing Profile To-Trunks** to determine the destination server and corresponding destination flow. The **End Point Policy** and **Topology Hiding Profile** will be applied as appropriate. When creating the flow, configure the parameters as follows:

- For the **Flow Name**, enter a descriptive name.
- For **Server Configuration**, select the Session Manager server created in **Section 7.7.1**.
- To match all traffic, set the **URI Group**, **Transport**, and **Remote Subnet** to \*.
- Set the **Received Interface** to the external signaling interface.
- Set the **Signaling Interface** to the internal signaling interface.
- Set the **Media Interface** to the internal media interface.
- Set the **End Point Policy Group** to the endpoint policy group defined for Session Manager in **Section 7.11.1**.
- Set the **Routing Profile** to the routing profile defined in **Section 7.12.2** used to direct traffic to the Cox Communications SIP server.
- Set the **Topology Hiding Profile** to the topology hiding profile defined for Session Manager in **Section 7.13.1**.

View Flow: Avaya-SM

X

Criteria	
Flow Name	Avaya-SM
Server Configuration	Avaya-SM
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Ext_Sig_Intf

Profile	
Signaling Interface	Int_Sig_Intf
Media Interface	Int_Media_Intf
End Point Policy Group	TM_Test
Routing Profile	To_Trunks
Topology Hiding Profile	PRT-Domain
File Transfer Profile	None

### 7.14.2. End Point Flow – Cox Communications

For the compliance test, endpoint flow **Cox** was created for the Cox Communications SIP server. All traffic from Cox Communications will match this flow as the source flow and use the specified **Routing Profile To-PrtSM** to determine the destination server and corresponding destination flow. The **End Point Policy** and **Topology Hiding Profile** will be applied as appropriate. When creating the flow, configure the parameters as follows:

- For the **Flow Name**, enter a descriptive name.
- For **Server Configuration**, select the Cox Communications SIP server created in **Section 7.7.2**.
- To match all traffic, set the **URI Group**, **Transport**, and **Remote Subnet** to \*.
- Set the **Received Interface** to the internal signaling interface.
- Set the **Signaling Interface** to the external signaling interface.
- Set the **Media Interface** to the external media interface.
- Set the **End Point Policy Group** to the endpoint policy group defined for Cox Communications in **Section 7.11.2**.
- Set the **Routing Profile** to the routing profile defined in **Section 7.12.1** used to direct traffic to Session Manager.
- Set the **Topology Hiding Profile** to the topology hiding profile defined for Cox Communications in **Section 7.13.2**.

View Flow: Cox			
Criteria		Profile	
Flow Name	Cox	Signaling Interface	Ext_Sig_Intf
Server Configuration	SP-Cox	Media Interface	Ext_Media_Intf
URI Group	*	End Point Policy Group	Cox-Policy-Grp
Transport	*	Routing Profile	To_PrtSM
Remote Subnet	*	Topology Hiding Profile	SP-General
Received Interface	Int_Sig_Intf	File Transfer Profile	None

## 8. Cox Communications SIP Trunking Service Configuration

Cox Communications is responsible for the network configuration and deployment of the Cox Communications SIP Trunking Service.

Cox Communications will require that the customer provide the IP address and port number used to reach the Avaya SBCE at the edge of the enterprise. Cox Communications will provide the IP address and port number of the Cox Communications SIP proxy/SBC, IP addresses/ports of media sources and Direct Inward Dialed (DID) numbers assigned to the enterprise. This information is used to complete the Communication Manager, Session Manager and the Avaya SBCE configuration discussed in the previous sections.

The configuration between Cox Communications and the enterprise is a static configuration. There is no registration of the SIP trunk or enterprise users to the Cox Communications network.

## 9. Verification Steps

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 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.

Troubleshooting:

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



- **Call Routing Test** - The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, navigate to **Elements → Session Manager → System Tools → Call Routing Test**. Enter the requested data to run the test.

## 10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager, Avaya Aura® Session Manager and Avaya Session Border Controller for Enterprise to the Cox Communications SIP Trunking Service. The Cox Communications SIP Trunking Service provides businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. Please refer to **Section 2.2** for any exceptions or workarounds.

## 11. References

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

- [1] *Installing and Configuring Avaya Aura® System Platform*, Release 6.3, May 2013.
- [2] *Administering Avaya Aura® System Platform*, Release 6.3, May 2013.
- [3] *Administering Avaya Aura® Communication Manager*, Release 6.3, May 2013, Document Number 03-300509.
- [4] *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 6.3, May 2013, Document Number 555-245-205.
- [5] *Upgrading Avaya Aura® System Manager*, Release 6.3, May 2013.
- [6] *Administering Avaya Aura® System Manager*, Release 6.3, May 2013.
- [7] *Installing and Configuring Avaya Aura® Session Manager*, Release 6.1, April 2011, Document Number 03-603473.
- [8] *Administering Avaya Aura® Session Manager*, Release 6.3, June 2013.
- [9] *Avaya 1600 Series IP Deskphones Administrator Guide Release 1.3.x*, May 2010, Document Number 16-601443.
- [10] *Avaya one-X® Deskphone Edition H.323 for 9600 Series IP Deskphones Administrator Guide*, Release 3.2, January 2013, Document Number 16-300698.
- [11] *Avaya one-X® Deskphone Edition H.323 9608,9611G,9621G and 9641G Administrator Guide*, Release 6.2 SP3, January 2013, Document Number 16-300698.
- [12] *Avaya one-X® Deskphone Edition SIP 9608/9611G/9621G/9641G Administrator Guide*, Release 6.2, April 2013, Document Number 16-601944.
- [13] *Administering Avaya one-X® Communicator*, July 2013.
- [14] RFC 3261 *SIP: Session Initiation Protocol*, <http://www.ietf.org/>
- [15] RFC 2833 *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, <http://www.ietf.org/>

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