



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Level 3 SIP Trunking with Avaya Aura® Communication Manager Evolution Server 6.0.1, Avaya Aura® Session Manager 6.2 and Avaya Session Border Controller for Enterprise – Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between Level 3 SIP Trunking and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager 6.2, Avaya Aura® Communication Manager Evolution Server 6.0.1, Avaya Session Border Controller for Enterprise and various Avaya endpoints. Level 3 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 Level 3 SIP Trunking and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager 6.2, Avaya Aura® Communication Manager Evolution Server 6.0.1, Avaya Session Border Controller for Enterprise and various Avaya endpoints.

Customers using this Avaya SIP-enabled enterprise solution with Level 3 SIP Trunking 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 Level 3 SIP Trunking via the public Internet and exercise the features and functionality listed in **Section 2.1**. 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.

- Avaya SBCE provides Digest Authentication to Level 3 on behalf of Communication Manager. INVITE and REFER messages sent from Communication Manager are challenged by Level 3. The challenge is answered by the Avaya SBCE.
- Response to SIP OPTIONS queries
- Incoming PSTN calls to various phone types including Avaya H.323, SIP, digital, and analog 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, SIP, digital, and analog 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.

- Various call types including: local, long distance, international, outbound toll-free, operator assisted calls, and local directory assistance (411).
- Codec G.711MU and G.729A
- 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 forwarding and mobility (extension to cellular)
- T.38 Fax
- Network Call Redirection using the SIP REFER method

Items not supported or not tested included the following:

- Inbound toll-free and emergency calls are supported but were not tested.
- Call redirection requested by a 302 response is not supported by Level 3.

2.2. Test Results

Interoperability testing of Level 3 SIP Trunking was completed with successful results for all test cases with the exception of the observations/limitations described below.

- **Max-Forwards:** On incoming PSTN calls to the enterprise, the Max-Forwards value in the incoming SIP INVITE is set to a value of 9. This value was observed to be too small to allow the message to traverse all the SIP hops internal to the enterprise to reach the destination in all cases. Thus, the Avaya SBCE was used to increase this value when the INVITE arrived at the Avaya SBCE from the network. (See **Section 7.6.2**)
- **No Matching Codec Offered:** If the Communication Manager SIP trunk is improperly configured to have no matching codec with the service provider and an outbound call is placed, the service provider returns a “480 Temporarily Unavailable” response instead of a “488 Not Acceptable Here” response. The user hears fast busy.
- **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. Level 3 does not use the UPDATE message for this purpose.
- **Asymmetric DTMF payload header values are not supported:** Level 3 does not support the use of a different DTMF payload header value in each direction of a single call. This may occur if the media is re-directed from Communication Manager to an endpoint and the endpoint wishes to use a different DTMF payload header value than was negotiated when the call was initially established. During the compliance testing, this was observed when Avaya one-X® Communicator (SIP) transferred an inbound PSTN call to another PSTN number. The end result was that the call was dropped after a series

of re-INVITES to attempt to correct the asymmetric condition. There is currently no workaround for this problem other than to use the H.323 version of Avaya one-X® Communicator.

- **Avaya one-X® Communicator Conferencing:** When conferencing two PSTN calls using Avaya one-X® Communicator (H.323 and SIP), some call legs were observed to have no audio if the conference was initiated by clicking the Conference button during an PSTN active call, then entering the added PSTN party number. Conference calls were successful if the first PSTN active call was first put on hold, a new PSTN call was placed and answered and lastly the Conference button was selected to conference the two calls together.

2.3. Support

For technical support on Level 3 SIP Trunking, contact Level 3 using the Customer Center links at www.level3.com or by calling 1-877-2LEVEL3.

Avaya customers may obtain documentation and support for Avaya products by visiting <http://support.avaya.com>. Selecting the **Support Contact Options** link followed by **Maintenance Support** provides the worldwide support directory for Avaya Global Services. Specific numbers are provided for both customers and partners based on the specific type of support or consultation services needed. Some services may require specific Avaya service support agreements. Alternatively, in the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus.

3. Reference Configuration

Figure 1 illustrates a sample Avaya SIP-enabled enterprise solution connected to Level 3 SIP Trunking. This is the configuration used for compliance testing.

The components used to create the simulated customer site included:

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

Located at the edge of the enterprise is the Avaya SBCE. It has a public side that connects to the external network and a private side that 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.

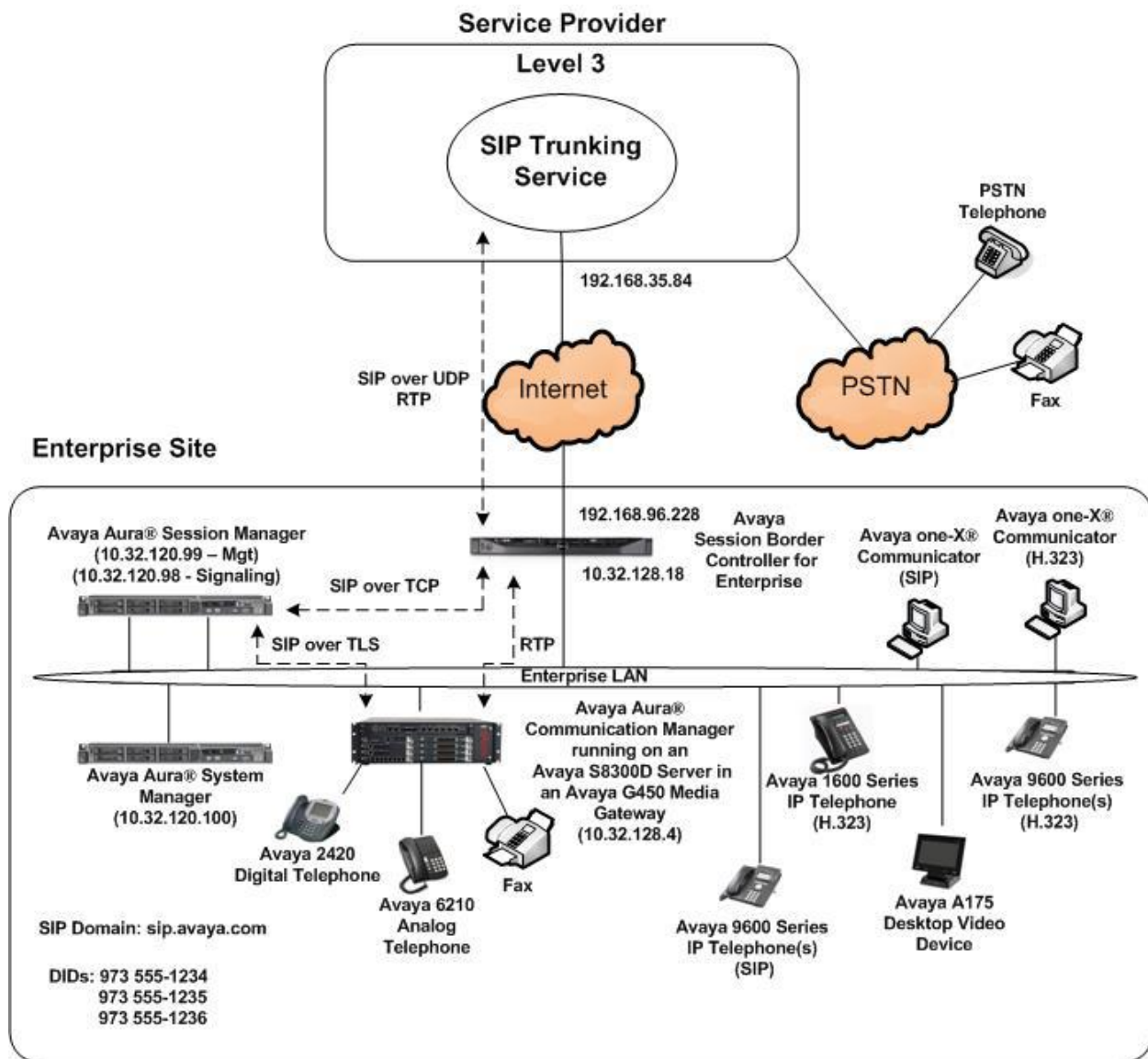


Figure 1: Avaya IP Telephony Network using Level 3 SIP Trunking

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 Level 3 SIP Trunking.

For the compliance test, 10 digit numbering was used for this purpose. Thus for outbound calls, the enterprise sent 10 digits in the SIP source headers (i.e., From, Contact, and P-Asserted-Identity). The enterprise was configured to send 11 digits in the SIP destination headers (Request URI and To). For inbound calls, Level 3 sent 10 digits in both the source headers and destination headers.

Level 3 listened for SIP signaling on port 5070 using UDP instead of the standard SIP port of 5060. The Avaya SBCE continued to listen on port 5060. Testing was conducted with and without the use of the SIP REFER method for redirected calls. Level 3 used digest authentication to challenge INVITE and REFER messages from the enterprise. The Avaya SBCE responded to the challenges on behalf of the enterprise to send the correct credentials to allow the calls to proceed.

4. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on Avaya S8300D Server	6.0.1 SP8.01 (R016x.00.1.510.0-19917) System Platform 6.0.3.9.3
Avaya G450 Media Gateway	31.20.0
Avaya Aura® System Manager running on an Avaya S8800 Server	6.2 SP2 (Build 6.2.0.0.15669-6.2.12.202) (Software Update Revision 6.2.14.1.1925) System Platform 6.2.1.0.9
Avaya Aura® Session Manager running on an Avaya S8800 Server	6.2 SP2 (Build 6.2.2.0.622005)
Avaya Session Border Controller for Enterprise running on a Dell R210 V2 server	4.0.5Q09
Avaya 1608 IP Telephone (H.323) running Avaya one-X® Deskphone Value Edition	1.3 SP1
Avaya 9640G IP Telephone (H.323) running Avaya one-X® Deskphone Edition	3.1 SP4 (3.1.04S)
Avaya 9641G IP Telephone (H.323) running Avaya one-X® Deskphone Edition	6.2 SP1 (S6.2119)
Avaya 9630 IP Telephone (SIP) running Avaya one-X® Deskphone SIP Edition	2.6 SP8 (2.6.8.4)
Avaya 9611 IP Telephone (SIP) running Avaya one-X® Deskphone SIP Edition	6.0 SP3 (6.0.3)
Avaya A175 Desktop Video Device with Avaya Flare® Experience	1.1
Avaya one-X® Communicator (H.323 or SIP)	6.1 SP5 (Build 6.1.5.07-SP5-37495)
Avaya 2420 Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
Level 3 SIP Trunking Solution Components	
Equipment/Software	Release/Version
Level 3 Enterprise Edge	Version 1

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 Level 3 SIP Trunking. A SIP trunk is established between Communication Manager and Session Manager for use by traffic to and from Level 3. 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.

display system-parameters customer-options		Page	2 of 11
OPTIONAL FEATURES			
IP PORT CAPACITIES		USED	
	Maximum Administered H.323 Trunks: 4000	36	
	Maximum Concurrently Registered IP Stations: 2400	3	
	Maximum Administered Remote Office Trunks: 4000	0	
	Maximum Concurrently Registered Remote Office Stations: 2400	0	
	Maximum Concurrently Registered IP eCons: 68	0	
	Max Concur Registered Unauthenticated H.323 Stations: 100	0	
	Maximum Video Capable Stations: 2400	0	
	Maximum Video Capable IP Softphones: 2400	1	
	Maximum Administered SIP Trunks: 4000	60	
	Maximum Administered Ad-hoc Video Conferencing Ports: 4000	0	
	Maximum Number of DS1 Boards with Echo Cancellation: 80	0	
	Maximum TN2501 VAL Boards: 10	0	
	Maximum Media Gateway VAL Sources: 50	0	
	Maximum TN2602 Boards with 80 VoIP Channels: 128	0	
	Maximum TN2602 Boards with 320 VoIP Channels: 128	0	
	Maximum Number of Expanded Meet-me Conference Ports: 300	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 transfer back to the PSTN then leave the field set to **none**.

```
change system-parameters features                                     Page 1 of 19
      FEATURE-RELATED SYSTEM PARAMETERS
      Self Station Display Enabled? 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 19
      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:

      ENBLOC DIALING PARAMETERS
      Enable Enbloc Dialing without ARS FAC? n

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

5.3. IP Node Names

Use the **change node-names ip** command to verify that node names have been previously defined for the IP addresses of the Avaya S8300D Server running Communication Manager (**procr**) and for Session Manager (**nwkSM**). These node names will be needed for defining the service provider signaling group in **Section 5.6**.

change node-names ip		Page 1 of 2
		IP NODE NAMES
Name	IP Address	
cmm	10.32.128.4	
default	0.0.0.0	
nwkSM	10.32.120.98	
procr	10.32.128.4	
procr6	::	

5.4. Codecs

Use the **change ip-codec-set** command to define a list of codecs to use for calls between the enterprise and the service provider. For the compliance test, G.729A and G.711MU were tested using IP codec set 3. To use these codecs, enter **G.729A** and **G.711MU** in the **Audio Codec** column of the table in the order of preference. Default values can be used for all other fields.

change ip-codec-set 3		Page 1 of 2
		IP Codec Set
Codec Set: 3		
Audio Codec	Silence Suppression	Frames Per Pkt
1: G.729A	n	2
2: G.711MU	n	2
3:		

On **Page 2**, set the **Fax Mode** to **t.38-standard**.

change ip-codec-set 3		Page 2 of 2
		IP Codec Set
Allow Direct-IP Multimedia? n		
FAX	Mode	Redundancy
	t.38-standard	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 **sip.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 19
                                                                IP NETWORK REGION
Region: 3
Location:                Authoritative Domain: sip.avaya.com
Name: SP Region
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
                                Inter-region IP-IP Direct Audio: yes
                                IP Audio Hairpinning? n
    Codec Set: 3
    UDP Port Min: 2048
    UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                RTCP Reporting Enabled? y
    Call Control PHB Value: 46          RTCP MONITOR SERVER PARAMETERS
    Audio PHB Value: 46                Use Default Server Parameters? y
    Video PHB Value: 26
802.1P/Q PARAMETERS
    Call Control 802.1p Priority: 6
    Audio 802.1p Priority: 6
    Video 802.1p Priority: 5          AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                RSVP Enabled? n
    H.323 Link Bounce Recovery? y
    Idle Traffic Interval (sec): 20
    Keep-Alive Interval (sec): 5
    Keep-Alive Count: 5
```

On **Page 4**, define the IP codec set to be used for traffic between region 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**.

change ip-network-region 3										Page	4	of	20
Source Region: 3 Inter Network Region Connection Management										I			M
										G	A		t
dst	codec	direct	WAN-BW-limits	Video	Intervening	Dyn	A	G					c
rgn	set	WAN	Units	Total Norm	Prio Shr Regions	CAC	R	L					e
1	3	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 12 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, part of 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.
- 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 Avaya S8300D Server running Communication Manager as defined in **Section 5.3**.
- Set the **Far-end Node Name** to **nwkSM**. 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 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 **5260**.

- 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 **6**. 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.

add signaling-group 12		Page 1 of 1
SIGNALING GROUP		
Group Number: 12	Group Type: sip	
IMS Enabled? n	Transport Method: tls	
Q-SIP? n	SIP Enabled LSP? n	
IP Video? n	Enforce SIPS URI for SRTP? y	
Peer Detection Enabled? y	Peer Server: SM	
Near-end Node Name: procr	Far-end Node Name: nwksM	
Near-end Listen Port: 5260	Far-end Listen Port: 5260	
	Far-end Network Region: 3	
	Far-end Secondary Node Name:	
Far-end Domain: sip.avaya.com	Bypass If IP Threshold Exceeded? n	
Incoming Dialog Loopbacks: eliminate	RFC 3389 Comfort Noise? n	
DTMF over IP: rtp-payload	Direct IP-IP Audio Connections? y	
Session Establishment Timer(min): 3	IP Audio Hairpinning? n	
Enable Layer 3 Test? y	Initial IP-IP Direct Media? n	
H.323 Station Outgoing Direct Media? n	Alternate Route Timer(sec): 6	

5.7. Trunk Group

Use the **add trunk-group** command to create a trunk group for the signaling group created in **Section 5.6**. For the compliance test, trunk group 12 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 the previous step.
- 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.

add trunk-group 12		Page 1 of 21	
TRUNK GROUP			
Group Number: 12	Group Type: sip	CDR Reports: y	
Group Name: SP Trunk	COR: 1	TN: 1	TAC: 1012
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n	Night Service:		
Queue Length: 0	Night Service:		
Service Type: public-ntwrk	Auth Code? n	Member Assignment Method: auto	
		Signaling Group: 12	
		Number of Members: 10	

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.

```

add trunk-group 12
    Group Type: sip

TRUNK PARAMETERS

    Unicode Name: auto

                                Redirect On OPTIM Failure: 6000

    SCCAN? n                                Digital Loss Group: 18
                                Preferred Minimum Session Refresh Interval(sec): 600

                                Delay Call Setup When Accessed Via IGAR? n

```

Page 2 of 21

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.

```

add trunk-group 12
TRUNK FEATURES
    ACA Assignment? n                Measured: none                Maintenance Tests? y

                                Numbering Format: private

                                UII Treatment: service-provider

                                Replace Restricted Numbers? y
                                Replace Unavailable Numbers? y

                                Modify Tandem Calling Number: no

    Show ANSWERED BY on Display? y

```

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On **Page 4**, the **Network Call Redirection** field may be set to **y** or **n**. Level 3 supports both settings. If set to **y**, Communication Manager will use the SIP REFER method to redirect calls back to the PSTN. Otherwise, a re-INVITE is used. 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 preferred by Level 3.

add trunk-group 12	Page 4 of 21
PROTOCOL VARIATIONS	
Mark Users as Phone? n	
Prepend '+' to Calling Number? n	
Send Transferring Party Information? n	
Network Call Redirection? y	
Send Diversion Header? y	
Support Request History? n	
Telephone Event Payload Type: 101	
Convert 180 to 183 for Early Media? n	
Always Use re-INVITE for Display Updates? n	
Identity for Calling Party Display: P-Asserted-Identity	
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, three DID numbers were assigned for testing. These three numbers were assigned to the three extensions 40003, 40021 and 40030. 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 three extensions.

change private-numbering 0					Page 1 of 2
NUMBERING - PRIVATE FORMAT					
Ext Len	Ext Code	Trk Grp (s)	Private Prefix	Total Len	
5	4			5	Total Administered: 4
5	40003	12	9735551234	10	Maximum Entries: 540
5	40021	12	9735551235	10	
5	40030	12	9735551236	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 0					Page 1 of 2
NUMBERING - PRIVATE FORMAT					
Ext Len	Ext Code	Trk Grp (s)	Private Prefix	Total Len	
5	4	12	97355	10	Total Administered: 1
					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: 2			
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	
1	4	dac							
4	5	ext							
8	1	fac							
9	1	fac							
*	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		Page 1 of 10
FEATURE ACCESS CODE (FAC)		
Abbreviated Dialing List1 Access Code:		
Abbreviated Dialing List2 Access Code:		
Abbreviated Dialing List3 Access Code:		
Abbreviated Dial - Prgm Group List Access Code:		
Announcement Access Code:		
Answer Back Access Code:		
Attendant Access Code:		
Auto Alternate Routing (AAR) Access Code: 8		
Auto Route Selection (ARS) – Access Code 1: 9		Access Code 2:
Automatic Callback Activation:		Deactivation:
Call Forwarding Activation Busy/DA: *01 All: *02		Deactivation: *03

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 12 which contains the SIP trunk to the service provider (as defined next).

change ars analysis 0						Page 1 of 2	
ARS DIGIT ANALYSIS TABLE							
Location: all						Percent Full: 2	
	Dialed	Total		Route	Call	Node	ANI
	String	Min	Max	Pattern	Type	Num	Reqd
0		1	1	12	op		n
0		11	11	12	op		n
011		10	18	12	intl		n
1800		11	11	12	fpna		n
1877		11	11	12	fpna		n
1908		11	11	12	fpna		n
411		3	3	12	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 12 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 12 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 service provider 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 12														Page 1 of 3													
Pattern Number: 12														Pattern Name: SP route													
SCCAN? n														Secure SIP? n													
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted							DCS/	IXC												
No			Mrk	Lmt	List	Del	Digits							QSIG													
														Dgts		Intw											
1:	12	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										

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.

AVAYA

Avaya Aura® System Manager 6.2

Last Logged on at June 25, 2012 11:58 AM
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Users	Elements	Services
Administrators Manage Administrative Users	B5800 Branch Gateway Manage B5800 Branch Gateway 6.2 elements	Backup and Restore Backup and restore System Manager database
Directory Synchronization Synchronize users with the enterprise directory	Communication Manager Manage Communication Manager 5.2 and higher elements	Bulk Import and Export Manage Bulk Import and Export of Users, User Global Settings, Roles, Elements and others
Groups & Roles Manage groups, roles and assign roles to users	Conferencing Manage Conferencing Multimedia Server objects	Configurations Manage system wide configurations
User Management Manage users, shared user resources and provision users	Inventory Manage, discover, and navigate to elements, update element software	Events Manage alarms, view and harvest logs
	Meeting Exchange Manage Meeting Exchange and Avaya Aura Conferencing 6.0 elements	Licenses View and configure licenses
	Messaging Manage Avaya Aura Messaging, Communication Manager Messaging, and Modular Messaging	Replication Track data replication nodes, repair replication nodes
	Presence Presence	Scheduler Schedule, track, cancel, update and delete jobs
	Routing Network Routing Policy	Security Manage Security Certificates
	Session Manager Session Manager Element Manager	Templates Manage Templates for Communication Manager, Messaging System and B5800 Branch Gateway elements
	SIP AS 8.1 SIP AS 8.1	

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 6.2 web interface. At the top left is the Avaya logo. The main header shows 'Avaya Aura® System Manager 6.2'. On the top right, it indicates 'Last Logged on at June 25, 2012 11:58 AM' and provides links for 'Help', 'About', 'Change Password', and 'Log off admin'. Below the header, there are tabs for 'Routing' (active) and 'Home'. The left-hand navigation pane lists various configuration items under the 'Routing' category: 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 (other routing applications are referring domains of type SIP), Step 2: Create 'Locations', Step 3: Create 'Adaptations', and Step 4: Create 'SIP Entities'.

6.2. Specify SIP Domain

Create a SIP domain for each domain for which Session Manager will need to be aware in order to route calls. For the compliance test, this includes the enterprise domain (**sip.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". To the right of the breadcrumb is a "Help ?" link. Below the breadcrumb, the title "Domain Management" is displayed. To the right of the title are two buttons: "Commit" and "Cancel". A warning message is shown: "Warning: SIP Domain name change will cause login failure for Communication Address handles with this domain. Consult release notes or Support for steps to reset login credentials." Below the warning, there is a table with one item. The table has four columns: "Name", "Type", "Default", and "Notes". The "Name" column contains "sip.avaya.com" with a red asterisk to its left. The "Type" column contains a pull-down menu with "sip" selected. The "Default" column contains an unchecked checkbox. The "Notes" column contains "Auto CS domain". Above the table, there is a "1 Item | Refresh" link and a "Filter: Enable" link.

Name	Type	Default	Notes
* sip.avaya.com	sip	<input type="checkbox"/>	Auto CS 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 **Belleville**, 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

Help ?

Commit Cancel

General

* Name: Belleville

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 test environment included two subnets as shown below.
- **Notes:** Add a brief description (optional).

Click **Commit** to save.

Location Pattern

Add Remove

2 Items Refresh Filter: Enable

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.32.120.*	CPE CM, SM and other devices
<input type="checkbox"/>	* 10.32.128.*	SBCs

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, an adaptation was applied to the Communication Manager SIP entity. This adaptation mapped inbound DID numbers from Level 3 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 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**.

Home / Elements / Routing / Adaptations

Help ?

Adaptation Details

General

* **Adaptation name:** PRT CM Adaptation

Module name: DigitConversionAdapter

Module parameter:

Egress URI Parameters:

Notes: Princeton CM Adaptation

Commit Cancel

To map inbound DID numbers from Level 3 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

3 Items | [Refresh](#)
Filter: [Enable](#)

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
<input type="checkbox"/>	* 9735551234	* 10	* 10		* 10	40003	destination ▼		
<input type="checkbox"/>	* 9735551235	* 10	* 10		* 10	40021	destination ▼		
<input type="checkbox"/>	* 9735551236	* 10	* 10		* 10	40030	destination ▼		

Select : All, None

* Input Required

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 **Belleville**.
- **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'. 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: 'nwk-sm'), 'FQDN or IP Address' (required, value: '10.32.120.98'), 'Type' (dropdown menu, value: 'Session Manager'), 'Notes' (text area), 'Location' (dropdown menu, value: 'Belleville'), '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 5260 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"/>	<input type="text" value="sip.avaya.com"/> <input type="button" value="v"/>	<input type="text" value="for enterprise SBC's"/>
<input type="checkbox"/>	<input type="text" value="5060"/>	UDP <input type="button" value="v"/>	<input type="text" value="sip.avaya.com"/> <input type="button" value="v"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5061"/>	TLS <input type="button" value="v"/>	<input type="text" value="sip.avaya.com"/> <input type="button" value="v"/>	<input type="text" value="for nwk-cm & nwk-aes1"/>
<input type="checkbox"/>	<input type="text" value="5260"/>	TLS <input type="button" value="v"/>	<input type="text" value="sip.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 **Belleville** which is the location defined for the subnet where Communication Manager resides.

The screenshot shows a web interface for configuring SIP entities. At the top, a breadcrumb trail reads 'Home / Elements / Routing / SIP Entities'. Below this is a 'SIP Entity Details' section with a 'Help ?' link and 'Commit' and 'Cancel' buttons. The 'General' tab is active, displaying various configuration fields. Required fields are marked with a red asterisk. The 'Name' field contains 'prt-cm-trk12'. The 'FQDN or IP Address' field contains '10.32.128.4'. The 'Type' dropdown is set to 'CM'. The 'Notes' field contains 'Princeton CM Trk 12 - TM'. The 'Adaptation' dropdown is set to 'PRT CM Adaptation'. The 'Location' dropdown is set to 'Belleville'. The 'Time Zone' dropdown is set to 'America/New_York'. There is an unchecked checkbox for 'Override Port & Transport with DNS SRV:'. The 'SIP Timer B/F (in seconds)' field contains '4'. The 'Credential name' field is empty. The 'Call Detail Recording' dropdown is set to 'none'. Below the 'General' section is the 'SIP Link Monitoring' section, which has a dropdown set to 'Use Session Manager Configuration'.

Home / Elements / Routing / SIP Entities

Help ?

Commit Cancel

SIP Entity Details

General

* Name: prt-cm-trk12

* FQDN or IP Address: 10.32.128.4

Type: CM

Notes: Princeton CM Trk 12 - TM

Adaptation: PRT CM Adaptation

Location: Belleville

Time Zone: America/New_York

Override Port & Transport with DNS SRV: ☐

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

Credential name:

Call Detail Recording: none

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 **Belleville** which is the location defined for the subnet where the Avaya SBCE resides.

The screenshot shows a web-based configuration interface for 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 "General" section contains the following fields: "Name" (ASBCE), "FQDN or IP Address" (10.32.128.18), "Type" (SIP Trunk), "Notes" (Avaya SBC for Enterprise), "Adaptation" (empty), "Location" (Belleville), and "Time Zone" (America/New_York). There is an unchecked checkbox for "Override Port & Transport with DNS SRV". The "SIP Timer B/F (in seconds)" is set to 4. The "Credential name" field is empty. The "Call Detail Recording" is set to egress. The "SIP Link Monitoring" section at the bottom has a dropdown menu set to "Use Session Manager Configuration".

Home / Elements / Routing / SIP Entities

SIP Entity Details [Help ?](#)

General

* Name: ASBCE

* FQDN or IP Address: 10.32.128.18

Type: SIP Trunk

Notes: Avaya SBC for Enterprise

Adaptation:

Location: Belleville

Time Zone: America/New_York

Override Port & Transport with DNS SRV: ☐

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

Credential name:

Call Detail Recording: egress

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.
- **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**.

Home / Elements / Routing / Entity Links

Entity Links [Help ?](#)

1 Item | Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* SM to PRT CM TRK 1	* nwk-sm	TLS	* 5260	* prt-cm-trk12	* 5260	Trusted	

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

Home / Elements / Routing / Entity Links

Entity Links

[Help ?](#)

Commit

Cancel

1 Item | [Refresh](#)

Filter: [Enable](#)

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* SM to ASBCE	* nwk-sm	TCP	* 5060	* ASBCE	* 5060	Trusted	

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:** PRT CM TRK 12 Policy

Disabled: ☐

* **Retries:** 0

Notes: TM SP Testing

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
prt-cm-trk12	10.32.128.4	CM	Princeton CM Trk 12 - TM

Home / Elements / Routing / Routing Policies

Routing Policy Details

Commit

Cancel

Help ?

General

* Name:

ASBCE Policy

Disabled:

☐

* Retries:

0

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
ASBCE	10.32.128.18	SIP Trunk	Avaya SBC for Enterprise

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 Level 3 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 **sip.avaya.com** from **ALL** locations use route policy **ASBCE Policy**.

Home / Elements / Routing / Dial Patterns

Dial Pattern Details
[Help ?](#)

General

* Pattern:

* Min:

* Max:

Emergency Call: ☐

Emergency Priority:

Emergency Type:

SIP Domain:

Notes:

Originating Locations and Routing Policies

1 Item Refresh Filter: [Enable](#)

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	ASBCE Policy	0	<input type="checkbox"/>	ASBCE	

Select : All, None

The second example shows that 10 digit numbers that start with **973555** to domain **sip.avaya.com** and originating from **ALL** locations use route policy **PRT CM TRK 12 Policy**. These are the DID numbers assigned to the enterprise from Level 3.

Home / Elements / Routing / Dial Patterns

Dial Pattern Details
[Help ?](#)

General

* Pattern:

* Min:

* Max:

Emergency Call: ☐

Emergency Priority:

Emergency Type:

SIP Domain:

Notes:

Originating Locations and Routing Policies

1 Item Refresh
Filter: [Enable](#)

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	PRT CM TRK 12 Policy	0	<input type="checkbox"/>	prt-cm-trk12	TM SP Testing

Select : All, None

The complete list of dial patterns defined for the compliance test is shown below.

Home / Elements / Routing / Dial Patterns

Help ?

Dial Patterns

Edit

New

Duplicate

Delete

More Actions ▾

11 Items | Refresh

Filter: Enable

<input type="checkbox"/>	Pattern	Min	Max	Emergency Call	Emergency Type	Emergency Priority	SIP Domain	Notes
<input type="checkbox"/>	0	1	1	<input type="checkbox"/>			sip.avaya.com	Outbound call to operator
<input type="checkbox"/>	011	10	18	<input type="checkbox"/>			sip.avaya.com	Outbound international call
<input type="checkbox"/>	1	11	11	<input type="checkbox"/>			sip.avaya.com	Outbound
<input type="checkbox"/>	4	5	5	<input type="checkbox"/>			sip.avaya.com	
<input type="checkbox"/>	411	3	3	<input type="checkbox"/>			sip.avaya.com	Outbound call for local directory assistance
<input type="checkbox"/>	5	5	5	<input type="checkbox"/>			sip.avaya.com	
<input type="checkbox"/>	973555	10	10	<input type="checkbox"/>			sip.avaya.com	DID Numbers

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 IP address of the Session Manager management interface.

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

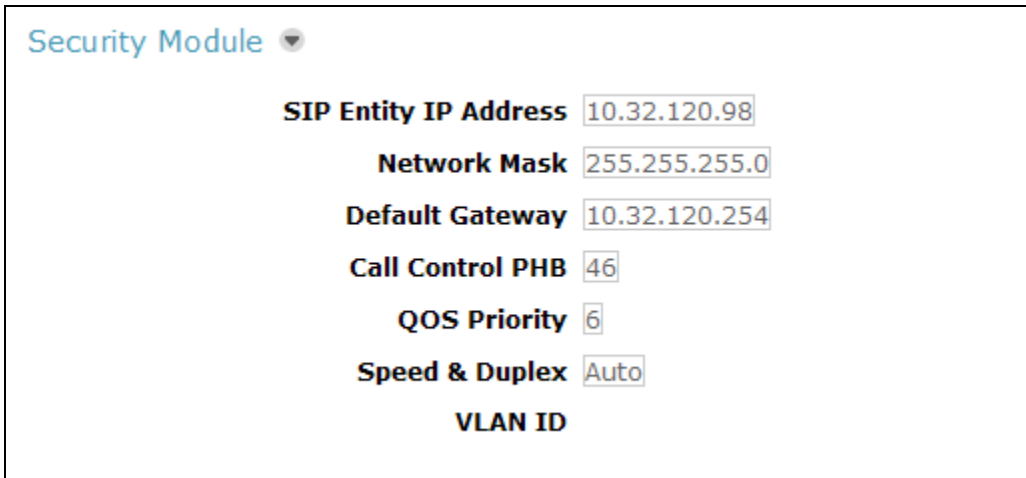
The screenshot shows a web interface for 'Session Manager Administration'. The breadcrumb trail at the top is 'Home / Elements / Session Manager / Session Manager Administration'. The main heading is 'View Session Manager' with a 'Return' button. Below the heading is a navigation menu with links: 'General | Security Module | NIC Bonding | Monitoring | CDR | Personal Profile Manager (PPM) - Connection Settings | Event Server |'. There are also links for 'Expand All' and 'Collapse All'. The 'General' section is expanded, showing the following fields:

SIP Entity Name	nwk-sm
Description	
Management Access Point Host Name/IP	nwk-sm.avaya.com
Direct Routing to Endpoints	Disable

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 '10.32.120.98', 'Network Mask' with '255.255.255.0', 'Default Gateway' with '10.32.120.254', '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	10.32.120.98
Network Mask	255.255.255.0
Default Gateway	10.32.120.254
Call Control PHB	46
QOS Priority	6
Speed & Duplex	Auto
VLAN ID	

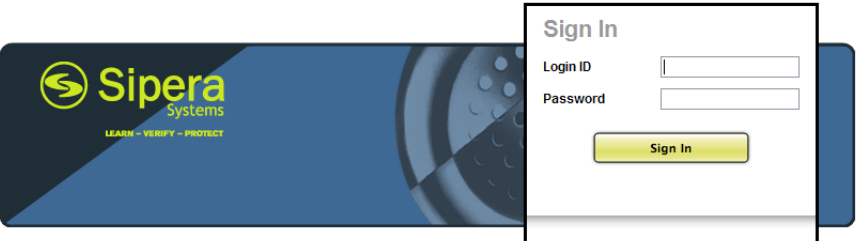
7. Configure Avaya Session Border Controller for Enterprise

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. For the compliance test, the Avaya SBCE management interface was on the same subnet as the private interface A1. However at a customer site, 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. A screen will appear (not shown) requesting the user to **Choose a destination**. Select **UC-Sec Control Center** and the Avaya SBCE login page will appear as shown below. Log in with appropriate credentials.

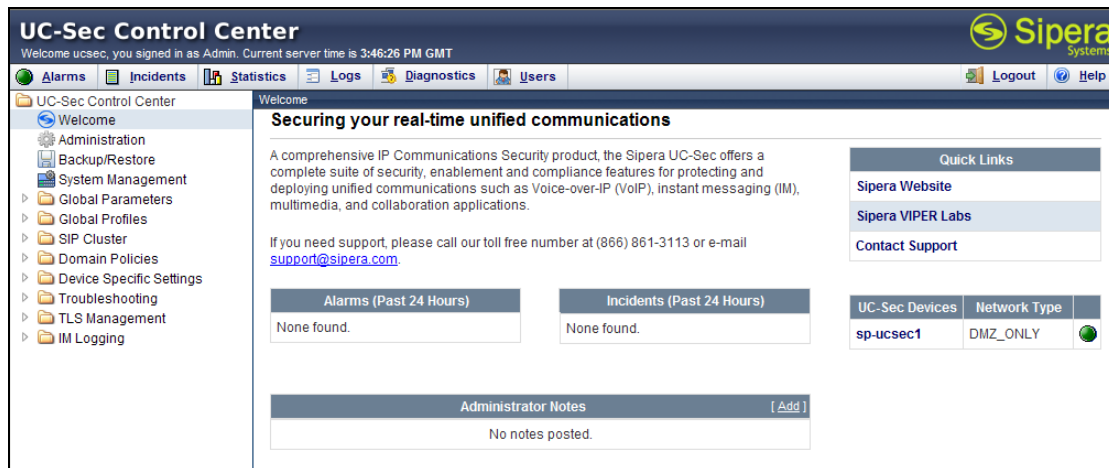


The UC-Sec™ family of products from Sipera Systems delivers comprehensive VoIP security by adapting the best practices of internet security and by using unique, sophisticated techniques such as VoIP protocol misuse & anomaly detection, behavioral learning based anomaly detection and voice spam detection to protect VoIP networks.

[Visit the Sipera Systems website to learn more.](#)

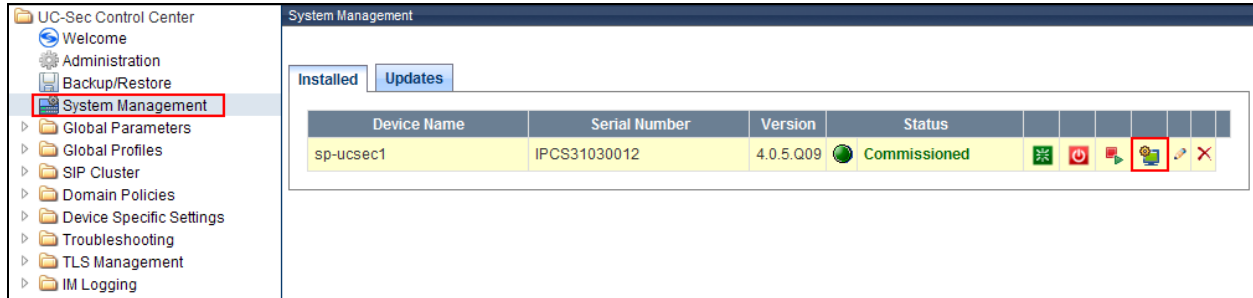
NOTICE TO USERS: This system is for authorized use only. Unauthorized use of this system is strictly prohibited. Unauthorized or improper use of this system may result in civil and/or criminal penalties. Use of this system constitutes consent to security monitoring. All activity is logged with login info, host name and IP address.

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



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 the **View Config** icon highlighted below.



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. 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

Network Configuration

General Settings

Appliance Name	sp-ucsec1
Box Type	SIP
Deployment Mode	Proxy

Device Settings

HA Mode	No
Secure Channel Mode	None
Two Bypass Mode	No

Network Settings

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

DNS Configuration

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

Management IP(s)

IP	10.32.128.17
----	--------------

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. The right pane will show the same **A1** and **B1** interfaces displayed in the previous screen. Click on the **Interface Configuration** tab.

The screenshot shows the UC-Sec Control Center interface. On the left, the 'Device Specific Settings' menu is expanded, and 'Network Management' is selected. In the center pane, 'sp-ucsec1' is selected under 'UC-Sec Devices'. The right pane shows the 'Interface Configuration' tab for 'sp-ucsec1'. A warning message states: 'Modifications or deletions of an IP address or its associated data require an application restart before taking effect. Application restarts can be issued from System Management.' Below this, there are fields for A1 Netmask (255.255.255.0), A2 Netmask, B1 Netmask (255.255.255.224), and B2 Netmask. There are 'Add IP', 'Save Changes', and 'Clear Changes' buttons. A table lists IP addresses and their associated interfaces:

IP Address	Public IP	Gateway	Interface	
10.32.128.18		10.32.128.254	A1	✗
192.168.96.228		192.168.96.254	B1	✗

On the **Interface Configuration** tab, verify the **Administrative Status** is **Enabled** for both the **A1** and **B1** interfaces. If not, click the **Toggle State** button to enable the interface.

Network Configuration		Interface Configuration
Name	Administrative Status	
A1	Enabled	Toggle State
A2	Disabled	Toggle State
B1	Enabled	Toggle State
B2	Disabled	Toggle State

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 Signaling Interface**. A pop-up window (not shown) will appear requesting the name of the new interface, followed by 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. When configuring the interface, configure the parameters as follows:

- Set **Name** to a descriptive name.
- Set the **Signaling IP** to the IP address associated with the private interface (A1) defined in **Section 7.2**.
- Set **TCP port** to the port the Avaya SBCE will listen on for SIP requests from Session Manager.

Signaling interface **Ent_Sig_Intf** was created for the Avaya SBCE external interface. When configuring the interface, configure the parameters as follows:

- Set **Name** to a descriptive name.
- Set the **Signaling IP** to the IP address associated with the public interface (B1) defined in **Section 7.2**.
- Set **UDP port** to the port the Avaya SBCE will listen on for SIP requests from the service provider.

The screenshot displays the UC-Sec Control Center interface. On the left, the navigation pane shows the hierarchy: UC-Sec Control Center > Device Specific Settings > Signaling Interface. The 'Signaling Interface' option is highlighted. The main pane is titled 'Device Specific Settings > Signaling Interface: sp-ucsec1'. It features a table of configured signaling interfaces and an 'Add Signaling Interface' button.

Name	Signaling IP	TCP Port	UDP Port	TLS Port	TLS Profile		
Int_Sig_Intf	10.32.128.18	5060	---	---	None		
Ext_Sig_Intf	192.168.96.228	---	5060	---	None		

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 Media Interface**. A pop-up window (not shown) will appear requesting the name of the new interface, followed by 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_Media_Intf** was created for the Avaya SBCE internal interface. When configuring the interface, configure the parameters as follows:

- Set **Name** to a descriptive name.
- Set the **Media IP** to the IP address associated with the private interface (A1) defined in **Section 7.2**.
- Set **Port Range** to a range of ports acceptable to both the Avaya SBCE and Session Manager. For the compliance test, the port range used was selected arbitrarily.

Signaling interface **Ent_Media_Intf** was created for the Avaya SBCE external interface. When configuring the interface, configure the parameters as follows:

- Set **Name** to a descriptive name.
- 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 service provider. For the compliance test, the port range used was selected arbitrarily.

UC-Sec Control Center

Device Specific Settings > Media Interface: sp-ucsec1

UC-Sec 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 Media Interface

Name	Media IP	Port Range		
Int_Media_Intf	10.32.128.18	35000 - 40000		
Ext_Media_Intf	192.168.96.228	35000 - 40000		

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 the 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 Profile**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by 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.

The screenshot displays the UC-Sec Control Center interface. On the left, the navigation pane shows the hierarchy: UC-Sec Control Center > Global Profiles > Server Interworking. The 'Server Interworking' item is highlighted. The center pane, titled 'Global Profiles > Server Interworking: Avaya-SM', contains a list of interworking profiles. The 'Avaya-SM' profile is selected and highlighted. Above the list are buttons for 'Add Profile', 'Rename Profile', 'Clone Profile', and 'Delete Profile'. The right pane shows the configuration for the selected 'Avaya-SM' profile, with tabs for 'General', 'Timers', 'URI Manipulation', 'Header Manipulation', and 'Advanced'. The 'General' tab is active, displaying a table of parameters.

General	
Hold Support	RFC3264
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	No
3xx Handling	No
Diversion Header Support	No

7.5.1. Server Interworking – Session Manager

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

- Set **Hold Support** to **RFC3264**.
- Enable **T.38 Support**.

General	Timers	URI Manipulation	Header Manipulation	Advanced
General				
Hold Support	RFC3264			
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	Yes			
URI Scheme	SIP			
Via Header Format	RFC3261			
Privacy				
Privacy Enabled	No			
User Name				
P-Asserted-Identity	No			
P-Preferred-Identity	No			
Privacy Header				
DTMF				
DTMF Support	None			
Edit				

On the Advanced tab, disable **Topology Hiding: Change Call-ID** and enable the **Avaya Extensions**. It is necessary to disable **Topology Hiding: Change Call-ID** when Network Call Redirection (NCR) is enabled on Communication Manager (Section 5.7). See Section 7.5.2 for more details.

General	Timers	URI Manipulation	Header Manipulation	Advanced
---------	--------	------------------	---------------------	----------

Advanced Settings	
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
SLiC 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 – Level 3

For the compliance test, server interworking profile **SP-General** was created for the Level 3 SIP server. When creating the profile, configure the General tab parameters as follows:

- Set **Hold Support** to **RFC3264**.
- Enable **T.38 Support**.

General	Timers	URI Manipulation	Header Manipulation	Advanced
General				
Hold Support	RFC3264			
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	Yes			
URI Scheme	SIP			
Via Header Format	RFC3261			
Privacy				
Privacy Enabled	No			
User Name				
P-Asserted-Identity	No			
P-Preferred-Identity	No			
Privacy Header				
DTMF				
DTMF Support	None			
Edit				

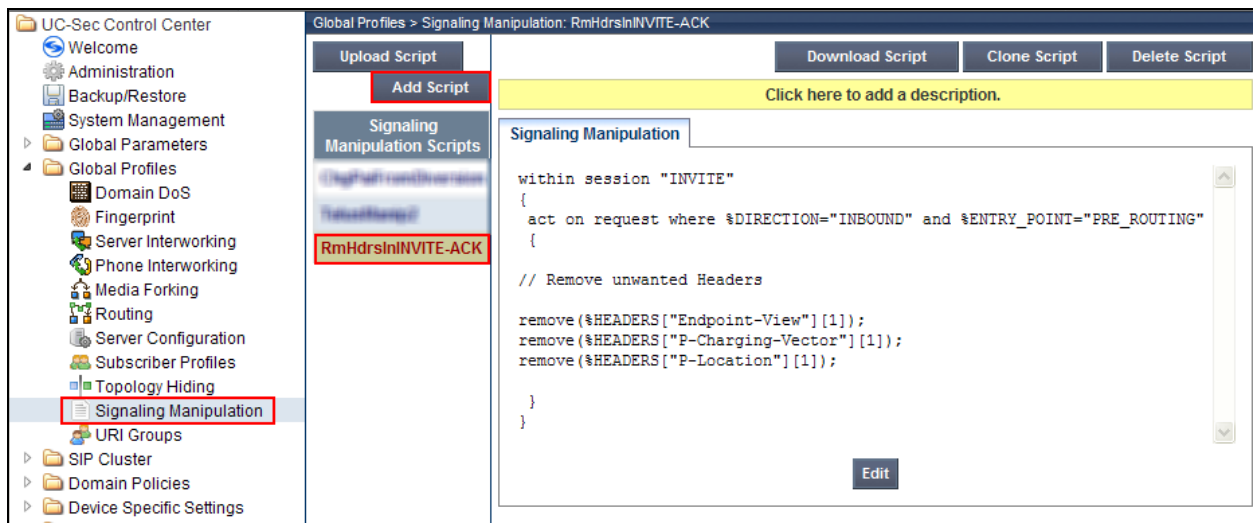
On the Advanced tab, disable **Topology Hiding: Change Call-ID** and disable the **Avaya Extensions**. It is necessary to disable **Topology Hiding: Change Call-ID** when Network Call Redirection (NCR) is enabled on Communication Manager. Otherwise, in some call transfer scenarios, the wrong Call-ID is passed in the Replaces parameter of the Refer-To header in the REFER message sent to Level 3. If this occurs, the transferred call fails with a 603 Decline response from Level 3 and the original call has one-way audio.

General	Timers	URI Manipulation	Header Manipulation	Advanced
Advanced Settings				
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		No		
NORTEL Extensions		No		
SLiC 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				

7.6. Signaling Manipulation

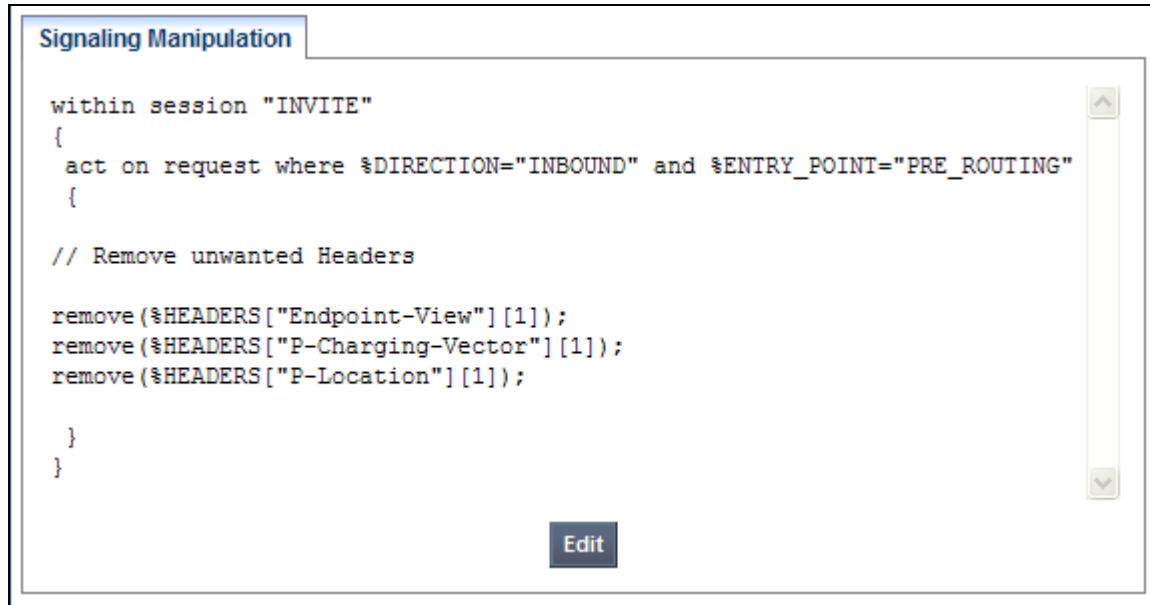
Signaling manipulation scripts provides for the manipulation of SIP messages which cannot be done by other configuration within the Avaya SBCE. Session Manager required the signaling manipulation script defined in **Section 7.6.1**. It is applied to the Session Manager server in **Section 7.7.1**. Level 3 required the signaling manipulation script defined in **Section 7.6.2**. It is applied to the Level 3 server in **Section 7.7.2**.

To create a script, navigate to **Global Profiles → Signaling Manipulation** in the left pane. In the center pane, select **Add Script**. A script editor window (not shown) will appear in which the script can be entered line by line. The **Title** box at the top of the editor window (not shown) is where the name of the script is entered. Once complete, the script is shown in the far right pane. To view an existing script, select the script from the center pane. The settings will appear in the right pane.



7.6.1. Signaling Manipulation – Session Manager

For the compliance test, signaling manipulation script **RmHdrsInINVITE-ACK** was created for Session Manager. The script removes unwanted headers in the ACK sent from Session Manager to the Avaya SBCE in response to a 200 OK associated with an INVITE request. This is in addition to header manipulations performed in the signaling rules defined in **Section 7.8.1**.



7.6.2. Signaling Manipulation – Level 3

For the compliance test, signaling manipulation script **IncMaxFwd** was created for the Level 3 SIP server. The script increases the value of the Max-Forwards header in an inbound INVITE message from Level 3. The initial Max-Forwards value is too small to allow the message to traverse all the SIP hops internal to the enterprise to reach the destination in all cases. Thus, the Avaya SBCE was used to increase this value when the INVITE arrived at the Avaya SBCE from the network.

Signaling Manipulation

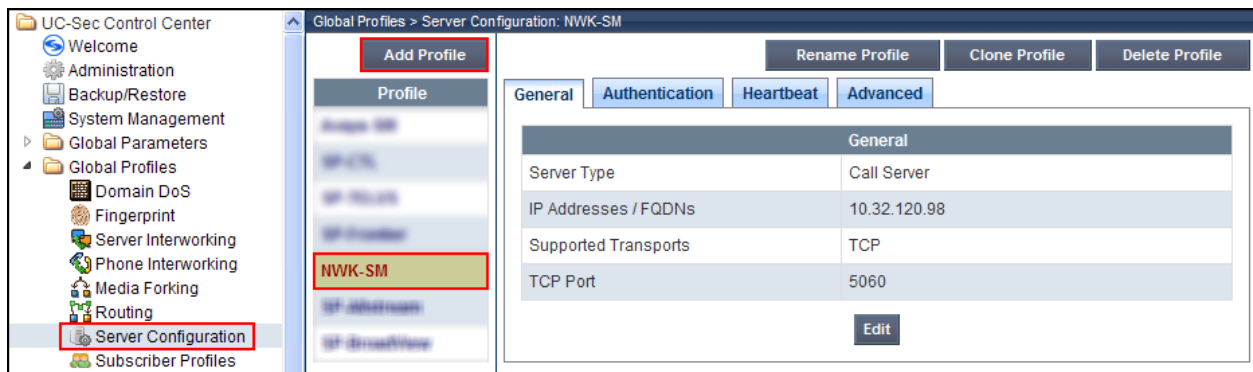
```
within session "INVITE"
{
  act on request where $DIRECTION="INBOUND" and $ENTRY_POINT="PRE_ROUTING"
  {
    if (exists($HEADERS["Max-Forwards"][1])) then
    {
      $HEADERS["Max-Forwards"][1] = "70";
    }
  }
}
```

Edit

7.7. Server Configuration

A server configuration profile defines the attributes of the physical server. Create a server configuration profile for the 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 Profile**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by 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 **NWK-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 Session Manager signaling interface.
- Set **Supported Transports** to the transport protocol used for SIP signaling between the Session Manager and the Avaya SBCE.
- Set the **TCP Port** to the port the Session Manager will listen on for SIP requests from the Avaya SBCE.

The screenshot shows a configuration interface with three buttons at the top: "Rename Profile", "Clone Profile", and "Delete Profile". Below these are four tabs: "General", "Authentication", "Heartbeat", and "Advanced". The "General" tab is selected. It contains a table with the following data:

General	
Server Type	Call Server
IP Addresses / FQDNs	10.32.120.98
Supported Transports	TCP
TCP Port	5060

Below the table is an "Edit" button.

On the Advanced tab, set the **Interworking Profile** field to the interworking profile for the Session Manager defined in **Section 7.5.1**. Set the **Signaling Manipulation Script** field to the signaling manipulation script for Session Manager defined in **Section 7.6.1**.

The screenshot shows the same configuration interface as above, but with the "Advanced" tab selected. It contains a table with the following data:

Advanced	
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	Avaya-SM
Signaling Manipulation Script	RmHdrsInINVITE-ACK
TCP Connection Type	SUBID

Below the table is an "Edit" button.

7.7.2. Server Configuration – Level 3

For the compliance test, server configuration profile **SP-Level3** was created for Level 3. 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 Level 3 SIP server.
- Set **Supported Transports** to the transport protocol used for SIP signaling between Level 3 and the Avaya SBCE.
- Set the **UDP Port** to the port Level 3 will listen on for SIP requests from the Avaya SBCE.

The screenshot shows the 'General' tab of a configuration interface. At the top, there are three buttons: 'Rename Profile', 'Clone Profile', and 'Delete Profile'. Below these are four tabs: 'General' (selected), 'Authentication', 'Heartbeat', and 'Advanced'. The 'General' tab contains a table with the following data:

General	
Server Type	Trunk Server
IP Addresses / FQDNs	192.168.35.84
Supported Transports	UDP
UDP Port	5070

Below the table is an 'Edit' button.

On the Authentication tab, configure the parameters using the credentials provided by Level 3 as follows:

- Click the **Enable Authentication** check box.
- Set **User Name** to the user name provided by Level 3.
- Set **Password** to the password provided by Level 3. This field is not shown in the summary screenshot below.
- Set **Realm** to the realm provided by Level 3.

The screenshot shows the 'Authentication' tab of the same configuration interface. At the top, there are three buttons: 'Rename Profile', 'Clone Profile', and 'Delete Profile'. Below these are four tabs: 'General', 'Authentication' (selected), 'Heartbeat', and 'Advanced'. The 'Authentication' tab contains a table with the following data:

Authentication	
Enable Authentication	<input checked="" type="checkbox"/>
User Name	1-23Q-3413
Realm	BroadWorks

Below the table is an 'Edit' button.

Session Manager will generate OPTIONS messages that will be forwarded by the Avaya SBCE to the service provider. Alternatively, the Avaya SBCE can be configured to send OPTIONS messages to the service provider. Both approaches were tested by the compliance test. To configure the Avaya SBCE to generate OPTIONS messages configure the parameters on the Heartbeat tab as follows:

- Click the **Enable Heartbeat** check box.
- Set **Method** to **OPTIONS**.
- Set **Frequency** to the desired interval between OPTIONS messages.
- Set **From URI** to a valid URI in the form of **number@domain** where the **number** is a valid DID assigned to the enterprise and the **domain** is the enterprise SIP domain.
- Set **To URI** to a valid URI in the form of **number@domain** where the **number** is the same number used in the **From URI** and the **domain** is the service provider domain.

Rename Profile Clone Profile Delete Profile

General Authentication Heartbeat Advanced

Heartbeat	
Enable Heartbeat	<input checked="" type="checkbox"/>
Method	OPTIONS
Frequency	60 seconds
From URI	9732418445@sip.avaya.com
To URI	9732418445@192.168.35.84
TCP Probe	<input type="checkbox"/>

Edit

On the Advanced tab, set the **Interworking Profile** field to the interworking profile for Level 3 defined in **Section 7.5.2**. Set the **Signaling Manipulation Script** field to the signaling manipulation script for the Level 3 SIP server defined in **Section 7.6.2**.

Advanced	
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	SP-Level3
Signaling Manipulation Script	IncMaxFwd
UDP Connection Type	SUBID

[Edit](#)

7.8. 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 7.10**. A specific signaling rule was created for Session Manager. The Level 3 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 Rule**. A pop-up window (not shown) will appear requesting the name of the new rule, followed by 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.

UC-Sec Control Center

- Welcome
- 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

Domain Policies > Signaling Rules: SessMgr_SigRules

[Add Rule](#) [Filter By Device...](#) [Rename Rule](#) [Clone Rule](#) [Delete Rule](#)

Click here to add a description.

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	
----------	--

7.8.1. 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 Level 3. 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.

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. The entries perform the following actions:

1. Removes the **Endpoint-View** header from the **BYE** message in the **IN** direction (Session Manager to Avaya SBCE).
2. Removes the **Endpoint-View** header from the **PRACK** message in the **IN** direction.
3. Removes the **P-Charging Vector** header from the **UPDATE** 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	Method Name	Header Criteria	Action	Proprietary	Direction		
1	Endpoint-View	BYE	Forbidden	Remove Header	Yes	IN		
2	Endpoint-View	PRACK	Forbidden	Remove Header	Yes	IN		
3	P-Charging-Vector	UPDATE	Forbidden	Remove Header	Yes	IN		















Similarly, manipulations can be performed on SIP response messages. These can be viewed 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 the **200** response to an **INVITE** message in the **IN** direction (Session Manager to Avaya SBCE).
2. Removes the **P-Charging Vector** header from the **200** response to an **INVITE** message in the **IN** direction.
3. Removes the **P-Charging Vector** header from the **200** response to an **UPDATE** message in the **IN** direction.
4. Removes the **P-Location** header from the **181** response to an **INVITE** message in the **IN** direction.
5. Removes the **P-Location** header from the **182** response to an **INVITE** message in the **IN** direction.

6. Removes the **P-Location** header from the **183** response to an **INVITE** message in the **IN** direction.
7. Removes the **P-Location** header from the **200** response to an **INVITE** message in the **IN** direction.

GeneralRequestsResponsesRequest HeadersResponse HeadersSignaling QoS

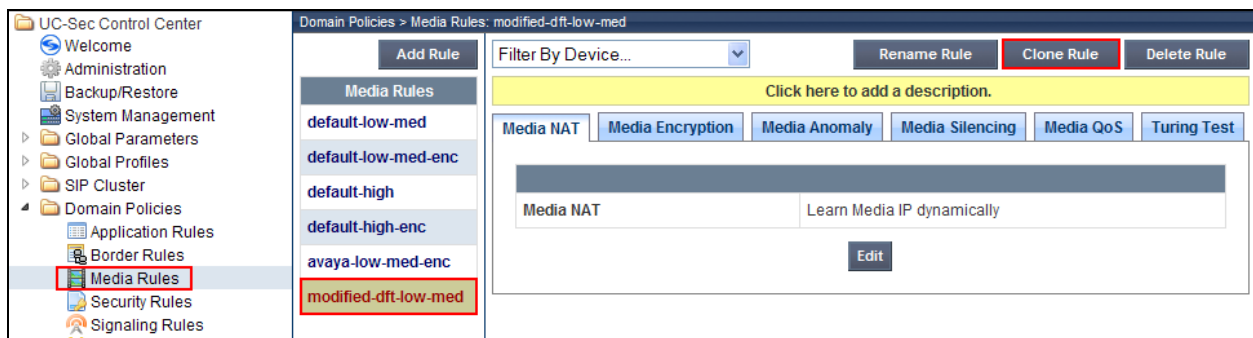
Add In Header ControlAdd Out Header Control

Row	Header Name	Response Code	Method Name	Header Criteria	Action	Proprietary	Direction		
1	Endpoint-View	200	INVITE	Forbidden	Remove Header	Yes	IN		
2	P-Charging-Vector	200	INVITE	Forbidden	Remove Header	Yes	IN		
3	P-Charging-Vector	200	UPDATE	Forbidden	Remove Header	Yes	IN		
4	P-Location	181	INVITE	Forbidden	Remove Header	Yes	IN		
5	P-Location	182	INVITE	Forbidden	Remove Header	Yes	IN		
6	P-Location	183	INVITE	Forbidden	Remove Header	Yes	IN		
7	P-Location	200	INVITE	Forbidden	Remove Header	Yes	IN		

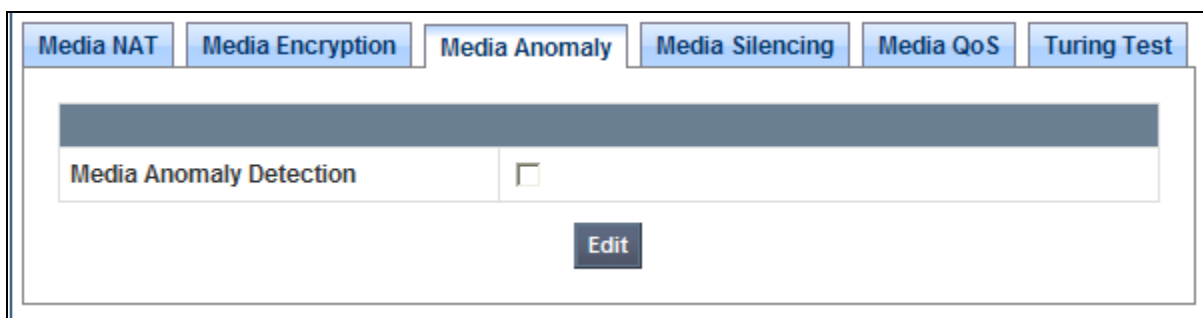
7.9. Media Rules

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 7.10**.

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



For the compliance test, a single media rule **modified-dft-low-med** was created that was used for both the Session Manager and the Level 3 SIP server. It was created by cloning the existing rule **default-low-med** which uses unencrypted media and then disabling **Media Anomaly Detection** on the Media Anomaly tab. This was done to prevent some false media errors from impacting the RTP media stream.



7.10. 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 7.13**.

To create a new group, navigate to **Domain Policies → End Point Policy Groups** in the left pane. In the center pane, select **Add Group**. A pop-up window (not shown) will appear requesting the name of the new group, followed by 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 UC-Sec Control Center interface. On the left is a navigation tree with 'End Point Policy Groups' selected. The main area is titled 'Domain Policies > End Point Policy Groups: SM'. It features an 'Add Group' button, a 'Filter By Device...' dropdown, and buttons for 'Rename Group' and 'Delete Group'. Below these are two yellow boxes with 'Click here to add a description.' and 'Click here to add a row description.' A 'Policy Group' tab is active, showing a table with columns: Order, Application, Border, Media, Security, Signaling, Time of Day, and two empty columns. The table contains one row with values: 1, default, default, modified-dft-low-med, default-low, SessMgr_SigRules, default, and two icons (a pencil and a plus sign). Buttons for 'View Summary' and 'Add Policy Set' are located above the table.

Order	Application	Border	Media	Security	Signaling	Time of Day		
1	default	default	modified-dft-low-med	default-low	SessMgr_SigRules	default		

7.10.1. Endpoint Policy Group – Session Manager

For the compliance test, endpoint policy group **SM** was created for Session Manager. Default values were used for each of the rules which comprise the group with the exception of **Media** and **Signaling**. For **Media**, select the media rule created in **Section 7.9**. For **Signaling**, select the signaling rule created for the Session Manager in **Section 7.8.1**.

This is a close-up of the 'Policy Group' table from the previous screenshot. It shows the table structure and the single data row.

Order	Application	Border	Media	Security	Signaling	Time of Day		
1	default	default	modified-dft-low-med	default-low	SessMgr_SigRules	default		

7.10.2. Endpoint Policy Group – Level 3

For the compliance test, endpoint policy group **General-SP** was created for the Level 3 SIP server. Default values were used for each of the rules which comprise the group with the exception of **Media**. For **Media**, select the media rule created in **Section 7.9**.

Policy Group								
				View Summary		Add Policy Set		
Order	Application	Border	Media	Security	Signaling	Time of Day		
1	default	default	modified-dft-low-med	default-low	default	default		

7.11. Routing

A routing profile defines where traffic will be directed based on the contents of the URI. A routing profile is applied only after the traffic has matched an endpoint server flow defined in **Section 7.13**. Create a routing profile for the Session Manager and the service provider SIP server.

To create a new profile, navigate to **Global Profiles → Routing** in the left pane. In the center pane, select **Add Profile**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by 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.

The screenshot displays the UC-Sec Control Center interface. On the left, the navigation pane shows the 'Routing' option under 'Global Profiles' highlighted. The main area is titled 'Global Profiles > Routing: To_SM'. It features buttons for 'Add Profile', 'Rename Profile', 'Clone Profile', and 'Delete Profile'. Below these, a yellow bar prompts the user to 'Click here to add a description.' The 'Routing Profile' section includes an 'Update Order' button and an 'Add Routing Rule' button. A table lists the routing rules:

Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport	
1	*	10.32.120.98	---	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	TCP	

7.11.1. Routing – Session Manager



For the compliance test, routing profile **To_SM** 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 the Session Manager signaling interface.
- Enable **Next Hop Priority**.
- Set the **Outgoing Transport** field to **TCP**.

Routing Profile

Update Order

Add Routing Rule

Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport		
1	*	10.32.120.98	---	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	TCP		

7.11.2. Routing – Level 3



For the compliance test, routing profile **To_Trunks** was created for Level 3. 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 Level 3 SIP server followed by the port number.
- Enable **Next Hop Priority**.
- Set the **Outgoing Transport** field to **UDP**.

Routing Profile

Update Order

Add Routing Rule

Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport		
1	*	192.168.35.84:5070	---	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	UDP		

7.12. 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 7.13**.

To create a new profile, navigate to **Global Profiles → Topology Hiding** in the left pane. In the center pane, select **Add Profile**. 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, select the profile from the center pane. The settings will appear in the right pane.

The screenshot shows the UC-Sec Control Center interface. On the left is a navigation tree with 'Topology Hiding' selected. The center pane shows 'Global Profiles > Topology Hiding: NWK-SM' with an 'Add Profile' button highlighted. The right pane displays the configuration for the 'NWK-SM' profile, including a table for 'Topology Hiding' settings.

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

7.12.1. Topology Hiding – Session Manager

For the compliance test, topology hiding profile **NWK_SM** 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 (**sip.avaya.com**).

Topology Hiding

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

Edit

7.12.2. Topology Hiding – Level 3

For the compliance test, topology hiding profile **SP-General** was created for Level 3. This profile will be applied to traffic from the Avaya SBCE to Level 3. 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**. Set the **Replace Action** for the **Request-Line** and **To** headers to **Next Hop** which is the IP address of the Level 3 SIP server. Set the **Replace Action** for the **From** header to **Signaling Interface** which is the IP address of the public interface of the Avaya SBCE.

Topology Hiding

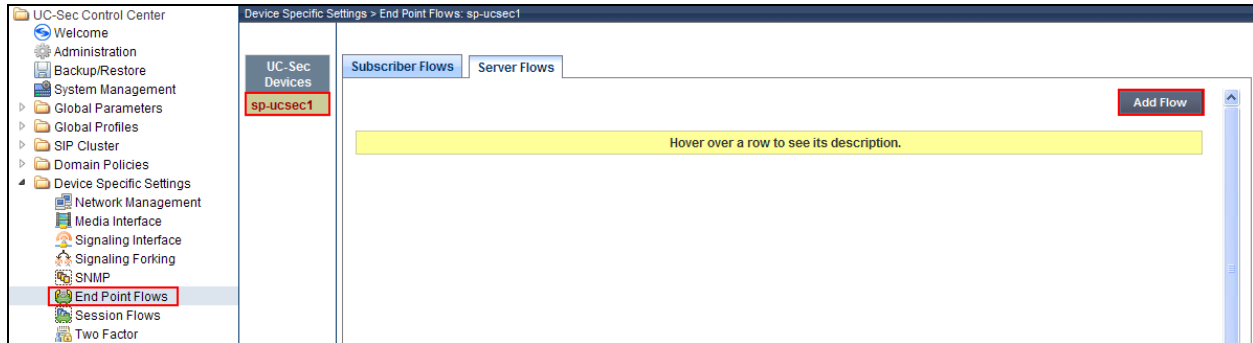
Header	Criteria	Replace Action	Overwrite Value
From	IP/Domain	Signaling Interface	---
To	IP/Domain	Next Hop	---
Via	IP/Domain	Auto	---
Request-Line	IP/Domain	Next Hop	---
Record-Route	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---

Edit

7.13. 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 SIP trunking, the signaling endpoints are the 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 Flow** 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.






7.13.1. End Point Flow – Session Manager

For the compliance test, endpoint flow **SM** was created for the Session Manager. All traffic from the 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 7.10.1**.
- Set the **Routing Profile** to the routing profile defined in **Section 7.11.2** used to direct traffic to the Level 3 SIP server.




- Set the **Topology Hiding Profile** to the topology hiding profile defined for Session Manager in **Section 7.12.1**.

Server Configuration: NWK-SM												
Priority	Flow Name	URI Group	Transport	Remote Subnet	Received Interface	Signaling Interface	Media Interface	End Point Policy Group	Routing Profile	Topology Hiding Profile	File Transfer Profile	
1	NWK-SM	*	*	*	Ext_Sig_Intf	Int_Sig_Intf	Int_Media_Intf	SM	To_Trunks	NWK-SM	None	  

7.13.2. End Point Flow – Level 3

For the compliance test, endpoint flow **Level3** was created for the Level 3 SIP server. All traffic from Level 3 will match this flow as the source flow and use the specified **Routing Profile To_SM** 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 Level 3 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 Level 3 in **Section 7.10.2**.
- Set the **Routing Profile** to the routing profile defined in **Section 7.11.1** used to direct traffic to the Session Manager.
- Set the **Topology Hiding Profile** to the topology hiding profile defined for Level 3 in **Section 7.12.2**.

Server Configuration: SP-Level3												
Priority	Flow Name	URI Group	Transport	Remote Subnet	Received Interface	Signaling Interface	Media Interface	End Point Policy Group	Routing Profile	Topology Hiding Profile	File Transfer Profile	
1	Level3	*	*	*	Int_Sig_Intf	Ext_Sig_Intf	Ext_Media_Intf	General-SP	To_SM	SP-General	None	  

8. Configure 9600 Series IP Telephones

For the compliance test, the DTMF payload header value for 9600 Series IP Telephones was set to 101 by adding the command **SET DTMF_PAYLOAD_TYPE=101** in the phone 46xxsettings.txt configuration file. Only the 9600 and 1600 SIP Telephones use this setting. The value of 101 is the value used by Level 3. The purpose of this configuration is to avoid a situation where a call between Level 3 and the SIP phone could be established with a DTMF payload header value that is different in each direction of the call. This scenario can cause interoperability issues.

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 access code number> - Displays trunk group information.
 - **status trunk** <trunk access code 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 Level 3 SIP Trunking. Level 3 SIP Trunking is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. Level 3 SIP Trunking 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 6.2.1, July 2012.
- [2] *Administering Avaya Aura® System Platform*, Release 6.2.1, July 2012.
- [3] *Administering Avaya Aura® Communication Manager*, August 2010, Document Number 03-300509.
- [4] *Avaya Aura® Communication Manager Feature Description and Implementation*, August 2010, Document Number 555-245-205.
- [5] *Upgrading Avaya Aura® System Manager*, Release 6.2, July 2012.
- [6] *Administering Avaya Aura® System Manager*, Release 6.2, July 2012.
- [7] *Installing and Configuring Avaya Aura® Session Manager*, Release 6.1, April 2011, Document Number 03-603473.
- [8] *Administering Avaya Aura® Session Manager*, Release 6.2, July 2012, Document Number 03-603324.
- [9] *Avaya 1600 Series IP Deskphones Administrator Guide Release 1.3.x*, May 2010, Document Number 16-601443.
- [10] *4600 Series IP Telephone LAN Administrator Guide*, October 2007, Document Number 555-233-507.
- [11] *Avaya one-X® Deskphone Edition for 9600 Series IP Telephones Administrator Guide*, November 2009, Document Number 16-300698.
- [12] *Administering Avaya one-X® Communicator*, July 2011.
- [13] RFC 3261 *SIP: Session Initiation Protocol*, <http://www.ietf.org/>
- [14] RFC 2833 *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, <http://www.ietf.org/>

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