



Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 5.2.1, Avaya Aura® Session Manager 6.2, and Avaya Session Border Controller for Enterprise 4.0.5 with Broadcore/Masergy SIP Trunk – Issue 1.1

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between Broadcore/Masergy SIP Trunk and Avaya Aura® Communication Manager 5.2.1, Avaya Aura® Session Manager 6.2, and Avaya Session Border Controller for Enterprise 4.0.5.

Broadcore/Masergy 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 in the Avaya Solutions and Interoperability Test Lab, utilizing Broadcore/Masergy SIP Trunk Services.

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1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between Broadcore/Masergy SIP Trunk Service and an Avaya SIP-enabled enterprise solution. The Avaya solutions consists of Avaya Aura® Communication Manager 5.2.1, Avaya Aura® Session Manager 6.2, Avaya Session Border Controller for Enterprise (SBCE) 4.0.5 and various Avaya endpoints.

Broadcore/Masergy offers SIP trunk services with either Single Number Registration offered service or through Static IP Authentication. These Application Notes illustrate Single Number Registration offered service, and includes Avaya SBCE configuration differences for Static IP Authentication in **Appendix A**.

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

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Communication Manager, Session Manager and Avaya SBCE to connect to the public Internet using a broadband connection. The enterprise site was configured to connect to Broadcore/Masergy SIP Trunk Service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 2.1**.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute 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:

- Incoming PSTN calls to various phone types. Phone types included H.323, 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. Phone types included H.323, 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 supports two modes (Road Warrior and Telecommuter). Each supported mode was tested. Avaya one-X Communicator also supports two Voice over IP (VoIP) protocols: H.323 and SIP. H.323 was the only protocol tested.
- Various call types including: local, long distance and outbound toll-free
- Codecs G.711MU and G.729A
- DTMF transmission using RFC 2833
- T.38 Fax
- Caller ID presentation and Caller ID restriction
- Voicemail navigation for inbound and outbound calls
- User features such as hold and resume, transfer, and conference
- Off-net call forwarding and mobility (extension to cellular)

Items not supported or not tested included the following:

- Inbound toll-free, international, operator, operator services (0 + 10 digits) and emergency calls (911) are supported but were not tested as part of the compliance test

2.2. Test Results

Interoperability testing of Broadcore/Masergy SIP Trunk Service was completed with successful results for all test cases with the observations/limitations described below.

- **Single Number Registration:** When using Broadcore/Masergy Single Number Registration offered service, the REQUEST-URI of an inbound call will include the main billing number of the SIP trunk, while the To header will include the actual called number. Communication Manager routes calls based on the REQUEST-URI, SIP manipulation was necessary to replace the User portion of the REQUEST-URI with information residing in the To header. Similarly outbound calls require the From header to include the main billing number and the P-Asserted-Identity (PAI) header to have the actual DID number. The Avaya SBCE is used to perform the required SIP manipulation. See **Section 7.5**.
- **Expires Header:** Broadcore/Masergy recommends including an Expires header with a value of 3600 to all Register requests. A Sigma script is used in the Avaya SBCE to insert an Expires header in all outbound SIP Register messages going towards Broadcore/Masergy SIP Trunk Service. See **Section 7.5**.
- **Fax:** When an outbound fax call is first setup with G.729 codec, Broadcore/Masergy will send a re-Invite to G.711 first before sending an Invite to T.38. If G.729 is the only codec listed by Communication Manager, the fax will fail with a 488 Not Acceptable Here response. To prevent this failure, it is necessary to always include G.711 as an available codec choice when fax will be used.
- **SendOnly SIP Parameter:** With the Network Call Redirection feature enabled, Communication Manager will use the SIP parameter “Sendonly” to signal any hold call conditions. Broadcore/Masergy will respond with an inactive media when it receives “Sendonly” instead of responding with “Recvonly”. As a result, the originating side hears music provided by Broadcore/Masergy instead of locally sourced music on hold.

The Avaya SBCE is used to remove the “Sendonly” parameter to allow local hold music to be received properly. See **Section 7.5**.

2.3. Support

For technical support on the Avaya products described in these Application Notes visit <http://support.avaya.com>.

For technical support on the Broadcore/Masergy SIP Trunk Service, contact Broadcore/Masergy using the Customer Care links at www.broadcore.com.

3. Reference Configuration

Figure 1 illustrates the sample configuration used for the DevConnect compliance testing. The configuration shown below simulates an enterprise location connected via T1 to the Internet and to the Broadcore/Masergy SIP Trunks service. At the edge of the enterprise is the Avaya SBCE providing NAT functionality and SIP header manipulation. Avaya SBCE receives traffic from Broadcore/Masergy SIP Trunk on port 5060 and sends traffic to the Broadcore/Masergy SIP Trunk using destination port 5060, using the UDP protocol. For security reasons, any actual public IP addresses used in the configuration have been either replaced with private IP addresses or have been blocked out. Similarly, any references to real routable PSTN numbers have also been changed to numbers that cannot be routed by the PSTN.

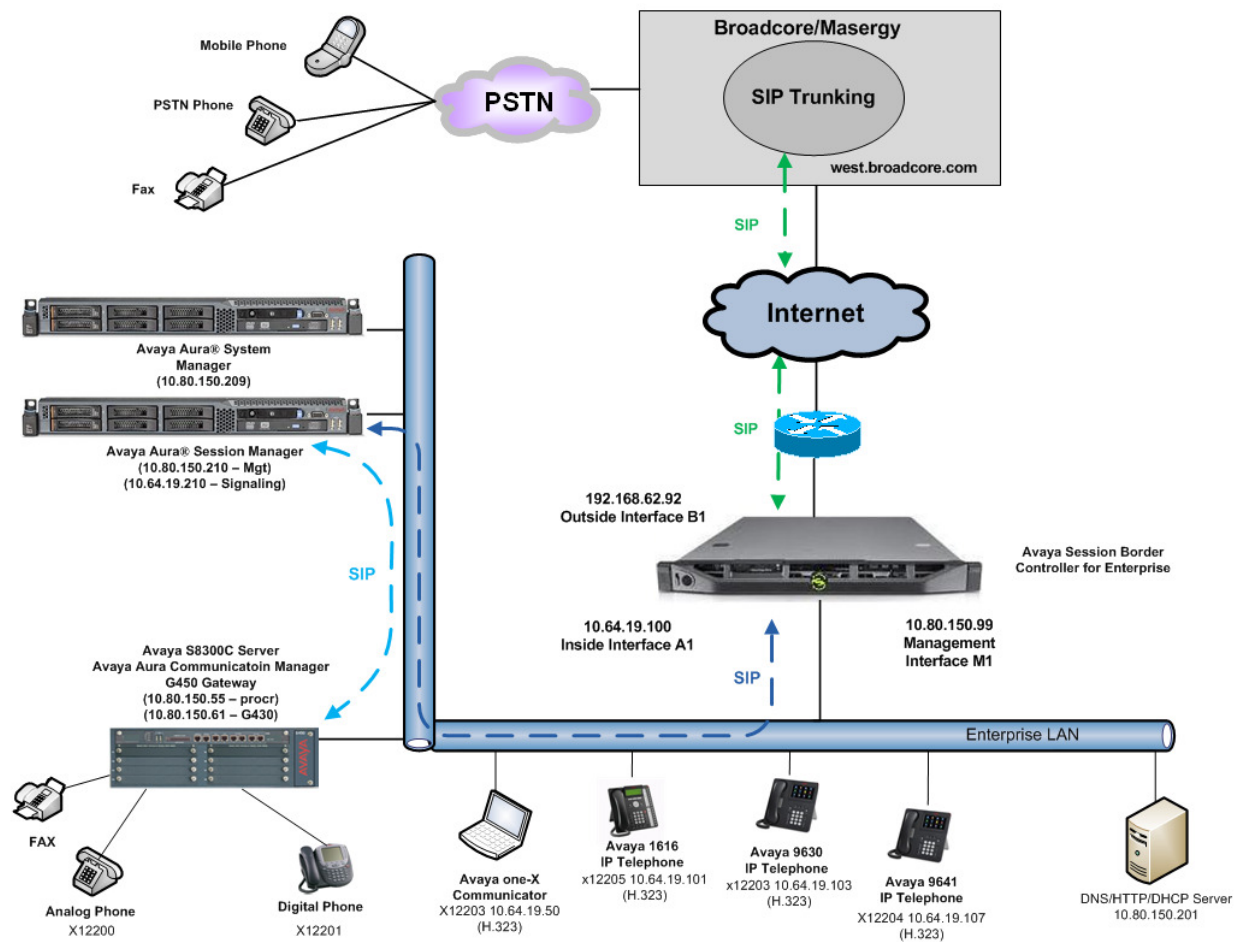


Figure 1: Avaya Interoperability Test Lab Configuration

4. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Component	Release
Avaya Aura® Communication Manger	R015x.02.1.016.4 -20445
Avaya Aura® System Manager	6.2.0 – SP3
Avaya Aura® Session Manager	6.2.3.0.623006
Avaya Session Border Controller for Enterprise	4.0.5Q19
Avaya G450	31.24.0
Avaya 9641 IP Telephone (H.323)	Avaya one-X® Deskphone Edition 6.2209
Avaya 9630 IP Telephone (H.323)	Avaya one-X® Deskphone Edition 3.104S
Avaya 1616 IP Telephone (H.323)	Avaya one-X® Deskphone Value Edition 1.302s
Avaya one-X® Communicator	6.1.5.07-SP5-37495
Avaya 2420 Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
Broadcore/Masergy SIP Trunking Solution Components	
Component	Release
Broadsoft	R17 SP4

Table 1: Equipment and Software Tested

The specific configuration above was used for the compatibility testing.

Note: 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 Broadcore/Masergy SIP Trunk Service. A SIP trunk is established between Communication Manager and Session Manager for use by signaling traffic to and from Broadcore/Masergy. It is assumed the general installation of Communication Manager, Avaya G450 Media Gateway and Session Manager has been previously completed and is not discussed here.

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: IP addresses and phone numbers shown throughout these Application Notes have been edited so that the actual 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 **450** licenses are available and **10** 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:		450	0
Maximum Concurrently Registered IP Stations:		450	4
Maximum Administered Remote Office Trunks:		0	0
Maximum Concurrently Registered Remote Office Stations:		0	0
Maximum Concurrently Registered IP eCons:		68	0
Max Concur Registered Unauthenticated H.323 Stations:		450	0
Maximum Video Capable Stations:		450	0
Maximum Video Capable IP Softphones:		450	0
Maximum Administered SIP Trunks:		450	10
Maximum Administered Ad-hoc Video Conferencing Ports:		450	0
Maximum Number of DS1 Boards with Echo Cancellation:		80	0
Maximum TN2501 VAL Boards:		0	0
Maximum Media Gateway VAL Sources:		50	1
Maximum TN2602 Boards with 80 VoIP Channels:		0	0
Maximum TN2602 Boards with 320 VoIP Channels:		0	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 18
      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 types of calls.

```
display 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: 1
      International Access Code: 011

      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 **display node-names ip** command to verify that node names have been previously defined for the IP addresses of Communication Manager (**procr**) and for Session Manager (**SM62**). These node names will be needed for defining the service provider signaling group in **Section 5.7**.

change node-names ip		Page 1 of 2
		IP NODE NAMES
Name	IP Address	
SM62	10.64.19.210	
default	0.0.0.0	
procr	10.64.19.55	

5.4. Codecs

Use the **change ip-codec-set** command to define a list of codecs to use for calls between the enterprise and the service provider. For the compliance test, ip-codec-set 2 was used for this purpose. The Broadcore/Masergy SIP Trunk Service supports G.729A, G.729AB and G.711MU. During compliance testing each of the supported codecs were tested independently by changing the order of preference to list the codec being tested as the first choice. The true order of preference is defined by the end customer. In the example below, **G.729A** and **G.711MU** were entered in the **Audio Codec** column of the table. Default values can be used for all other fields.

change ip-codec-set 2		Page 1 of 2
		IP Codec Set
Codec Set: 2		
Audio Codec	Silence Suppression	Frames Per Pkt
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 2		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 Interface for procr

The **add ip-interface procr** or **change ip-interface procr** command can be used to configure the Processor Ethernet (PE) parameters. The following screen shows the parameters used in the sample configuration. While the focus here is the use of the PE for SIP Trunk Signaling, observe that the Processor Ethernet will also be used for registrations from H.323 IP Telephones and H.248 gateways in the sample configuration.

change ip-interface procr		Page 1 of 2
IP INTERFACES		
Type: PROCR	Target socket load: 1700	
Enable Interface? y	Allow H.323 Endpoints? y	
Network Region: 1	Allow H.248 Gateways? y	
	Gatekeeper Priority: 5	
IPV4 PARAMETERS		
Node Name: procr		
Subnet Mask: /24		

5.6. IP Network Region

Create a separate IP network region for the service provider trunk. This allows for separate codec or quality of service settings to be used (if necessary) for calls between the enterprise and the service provider versus calls within the enterprise or elsewhere. For the compliance test, IP-network-region 2 was chosen for the service provider trunk. IP network region 1 is the default IP network region and encompasses the rest of the enterprise. Use the **change ip-network-region 2** command to configure region 2 with the following parameters:

- Set the **Location** field to match the enterprise location for this SIP trunk.
- Set the **Authoritative Domain** field to match the SIP domain of the enterprise. In this configuration, the domain name is **avayalab.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. To enable shuffling, 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**.
- Set the **UDP Port Min** and **UDP Port Max** fields to a range suitable for RTP traffic.
- Default values can be used for all other fields.

```

change ip-network-region 2                                     Page 1 of 20

                                IP NETWORK REGION

Region: 2
Location: 1      Authoritative Domain: avayalab.com
Name: SIP Trunks
MEDIA PARAMETERS
  Codec Set: 2      Intra-region IP-IP Direct Audio: yes
  UDP Port Min: 2048  Inter-region IP-IP Direct Audio: yes
  UDP Port Max: 3329      IP Audio Hairpinning? n
DIFFSERV/TOS PARAMETERS
  Call Control PHB Value: 46
  Audio PHB Value: 46
  Video PHB Value: 26
802.1P/Q PARAMETERS
  Call Control 802.1p Priority: 6
  Audio 802.1p Priority: 6
  Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS      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 2 and region 1 (the rest of the enterprise). Enter the desired IP codec set in the **codec set** column of the row with destination region (**dst rgn**) 1. Default values may be used for all other fields. The example below shows the settings used for the compliance test. It indicates that codec set 2 will be used for calls between region 2 (the service provider region) and region 1 (the rest of the enterprise).

```

change ip-network-region 2                                     Page 4 of 20

Source Region: 2      Inter Network Region Connection Management      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 2 y NoLimit                                                                n      t
2 2
3
4

```

5.7. Signaling Group

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

- Set the **Group Type** field to **sip**.
- Set the **IMS Enabled** field to **n**.

- In the sample configuration the **Transport Method** was set to **tcp** and the **Near-end Listen Port** and **Far-end Listen Port** was set to **5060**. In production, TLS transport between Communication Manager and Session Manager can be used.
- Set the **Near-end Node Name** to **procr**. This node name maps to the IP address of Communication Manager as defined in **Section 5.3**.
- Set the **Far-end Node Name** to **SM**. This node name maps to the IP address of Session Manager as defined in **Section 5.3**.
- Set the **Far-end Network Region** to the IP network region defined for the service provider in **Section 5.6**.
- 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.
- Set the **DTMF over IP** field to **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- Default values may be used for all other fields.

add signaling-group 1		Page 1 of 1
SIGNALING GROUP		
Group Number: 1	Group Type: sip	
	Transport Method: tcp	
IMS Enabled? n		
IP Video? n		
Near-end Node Name: procr	Far-end Node Name: SM62	
Near-end Listen Port: 5060	Far-end Listen Port: 5060	
	Far-end Network Region: 2	
Far-end Domain: avayalab.com		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
	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? n	Direct IP-IP Early Media? n	
H.323 Station Outgoing Direct Media? n	Alternate Route Timer(sec): 6	

5.8. Trunk Group

Use the **add trunk-group** command to create a trunk group for the signaling group created in **Section 5.7**. For the compliance test, trunk group 1 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 1                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 1                      Group Type: sip      CDR Reports: y
  Group Name: SIP Trunk to SP        COR: 1              TN: 1      TAC: *101
  Direction: two-way                Outgoing Display? n
  Dial Access? n                    Night Service:
  Queue Length: 0
  Service Type: public-ntwrk        Auth Code? n
                                     Signaling Group: 1
                                     Number of Members: 10
```

On **Page 2**, verify that the **Preferred Minimum Session Refresh Interval** is set to a value acceptable to the service provider. This value defines the interval that re-INVITEs must be sent to keep the active session alive. For the compliance test, the value of **900** seconds was used.

```
add trunk-group 1                                     Page 2 of 21
  Group Type: sip
TRUNK PARAMETERS
  Unicode Name: auto
                                     Redirect On OPTIM Failure: 5000
  SCCAN? n                            Digital Loss Group: 18
                                     Preferred Minimum Session Refresh Interval(sec): 900
Disconnect Supervision - In? y Out? y
```

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.

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.

add trunk-group 1		Page 3 of 21
TRUNK FEATURES		
ACA Assignment? n	Measured: none	Maintenance Tests? y
Numbering Format: private		
UI Treatment: service-provider		
Replace Restricted Numbers? y		
Replace Unavailable Numbers? y		
Modify Tandem Calling Number: no		
Show ANSWERED BY on Display? n		

On **Page 4**, set the **Network Call Redirection** field to **y**. This allows inbound calls transferred back to the PSTN to use the SIP REFER method, see **Reference [12]**. Setting the **Network Call Redirection** flag to “y” enables advanced services associated with the use of the REFER message, while also implicitly enabling Communication Manager to signal “send-only” media conditions for calls placed on hold at the enterprise site. If REFER signaling is not required, this field may be left at the default “n” value. In the testing associated with these Application Notes, transfer testing using REFER was successfully completed with the **Network Call Redirection** flag set to “y”, and transfer testing using INVITE was successfully completed with the **Network Call Redirection** flag set to “n”.

For redirected calls, Broadcore/Masergy supports the Diversion header, but not the History-Info header. Communication Manager can send the Diversion header by marking **Send Diversion Header** to “y”. Alternatively, Communication can send the History-Info header by setting **Support Request History** to “y”, and Session Manager can adapt the History-Info header to the Diversion header using the “DiversionTypeAdapter”. In the testing associated with these Application Notes, call redirection testing with Communication Manager sending History-Info and Session Manager adapting to Diversion Header was completed successfully. This allows for the same SIP trunk group to be used for Communication Manager Messaging, or any other SIP devices which requires the History-Info header.


```

add trunk-group 1
                                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? n
                                Support Request History? y
                                Telephone Event Payload Type: 101

```

5.9. Inbound Routing

In general, the incoming call handling treatment for a trunk group can be used to manipulate the digits received for an incoming call if necessary. Since Session Manager is present, Session Manager can be used to perform digit conversion using an Adaptation as shown in **Section 6.4**, and digit manipulation via Communication Manager incoming call handling table may not be necessary. If the DID number sent by Broadcore/Masergy is unchanged by Session Manager, then the DID number can be mapped to an extension using the incoming call handling treatment of the receiving trunk group. Both Session Manager digit conversion and Communication Manager incoming call handling treatment methods were created and tested successfully.

Use the **change inc-call-handling-trmt trunk-group** command to create an entry for any DID numbers unchanged by Session Manager. As an example, the following screen illustrates a conversion of DID number **213-555-2111** to extension **10000**.

```

change inc-call-handling-trmt trunk-group 1
                                Page 1 of 30
                                INCOMING CALL HANDLING TREATMENT
                                Service/      Number      Number      Del Insert
                                Feature      Len        Digits
                                public-ntwrk   10 2135552111    10 10000

```

5.10. 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 5.8**), the **change private-numbering** command may be used to define the format of numbers sent to Broadcore/Masergy in SIP headers such as the “From” and “PAI” headers. In general, the mappings of internal extensions to Broadcore/Masergy DID numbers may be done in Communication Manager (via private-numbering form for outbound calls, and incoming call handling treatment form for the inbound trunk group). Alternatively, Communication Manager can send the five digit extension to Session Manager, and Session Manager can adapt the number to the Broadcore/Masergy DID using an Adaptation as shown in **Section 6.4**. Both methods were tested successfully.

In the bolded row shown in the example below, a specific Communication Manager extension (x10000) is mapped to a DID number that is known to Broadcore/Masergy for this SIP Trunk connection (2135552111), when the call uses trunk group 1.

change private-numbering 0					Page 1 of 2
NUMBERING - PRIVATE FORMAT					
Ext	Ext	Trk	Private	Total	
Len	Code	Grp(s)	Prefix	Len	
5	122			5	Total Administered: 2
5	10000	1	2135552111	10	Maximum Entries: 540

5.11. 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. 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

Location: all

Percent Full: 0

	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type
1		5	ext						
2		5	ext						
4		4	ext						
5		4	ext						
6		5	ext						
7		4	ext						
8		1	fac						
9		1	fac						
*		4	dac						
#		4	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 9	
FEATURE ACCESS CODE (FAC)			
Abbreviated Dialing List1 Access Code: #110			
Abbreviated Dialing List2 Access Code: #111			
Abbreviated Dialing List3 Access Code: #112			
Abbreviated Dial - Prgm Group List Access Code: #113			
Announcement Access Code: #114			
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: #002 All:		Deactivation: #004	
Call Forwarding Enhanced Status:		Act: Deactivation:	

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9.

- **Dialed String:** enter the leading digits (e.g., **13**) necessary to uniquely select the desired route pattern.
- **Total Min:** enter the minimum number of digits (e.g., **11**) expected for this PSTN number.
- **Total Max:** enter the maximum number of digits (e.g., **11**) expected for this PSTN number.
- **Route Pattern:** enter the route pattern number (e.g., **1**) to be used. The route pattern (to be defined next) will specify the trunk group(s) to be used for calls matching the dialed number.
- **Call Type:** **fnpa** the call type for North American 1+10 digit calls. For local 7 or 10 digit calls enter **hnpa**. For 411 and 911 calls use **svcl** and **emer** respectively. The call type tells Communication Manager what kind of call is made to help decide how to handle the dialed string and whether or not to include a preceding 1. For more information and a complete list of Communication Manager call types, see **Reference [3]** and **[4]**.

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

change ars analysis 1							Page 1 of 2
ARS DIGIT ANALYSIS TABLE							
Location: all							Percent Full: 0
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req'd	
12	11	11	1	fnpa		n	
13	11	11	1	fnpa		n	
14	11	11	1	fnpa		n	
15	11	11	1	fnpa		n	
16	11	11	1	fnpa		n	
17	11	11	1	fnpa		n	
18	11	11	1	fnpa		n	
19	11	11	1	fnpa		n	
303	10	10	1	hnpa		n	
411	3	3	1	svcl		n	
4174089	7	7	1	hnpa		n	
501	10	10	1	hnpa		n	
720	10	10	1	hnpa		n	
911	3	3	1	emer		n	

The route pattern defines which trunk group will be used for the call and performs any necessary digit manipulation. Use the **change route-pattern** command to configure the parameters for the service provider trunk route pattern in the following manner. The example below shows the values used for route pattern 1 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 **1** 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:** The prefix mark (**Pfx Mrk**) of 1 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. All HNPA 10 digit numbers are left unchanged.
- The **Numbering Format unk-unk** means no special numbering format will be included.

change route-pattern 1												Page 1 of 3			
Pattern Number: 1												Pattern Name: Broadcore SIP TRK			
SCCAN? n												Secure SIP? n			
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted					DCS/	IXC		
No			Mrk	Lmt	List	Del	Digits					QSIG			
												Intw			
1:	1	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	none	
2:	y	y	y	y	y	n	n			rest				none	

5.12. Saving Communication Manager Configuration Changes

The command **save translation all** can be used to save the configuration.

save translation all	
SAVE TRANSLATION	
Command Completion Status	Error Code
Success	0

6. Configure Avaya Aura® Session Manager

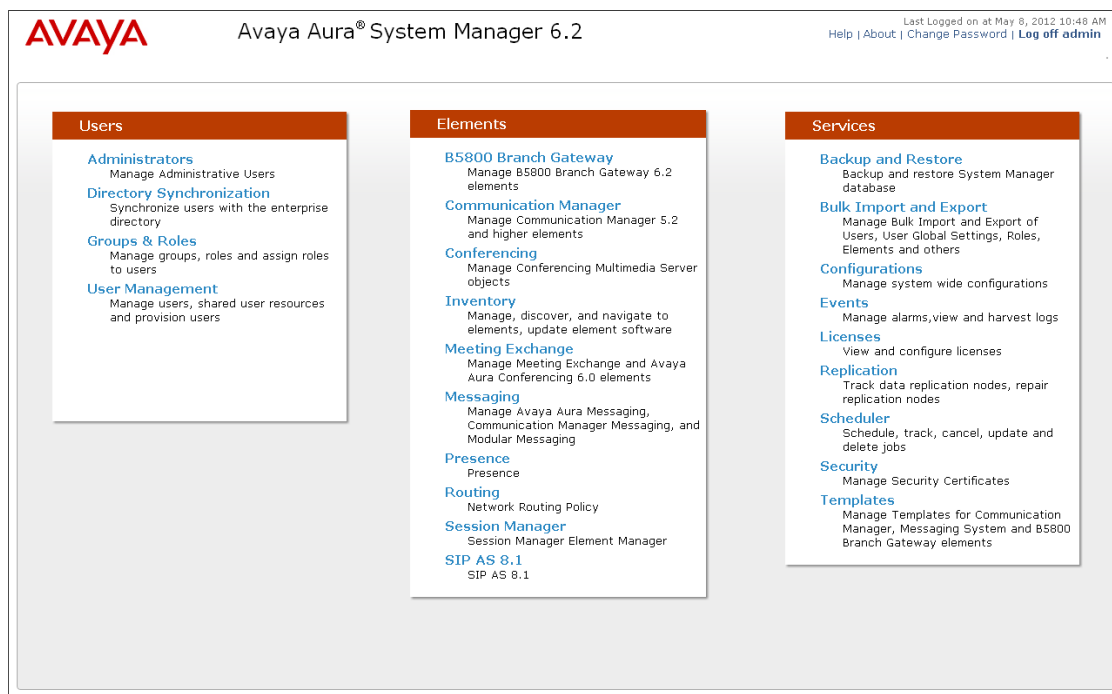
This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Location that can be occupied by SIP Entities
- SIP Entities corresponding to Communication Manager, Avaya SBCE and Session Manager
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Routing Policies, which control call routing between the SIP Entities
- Dial Patterns, which govern to which SIP Entity a call is routed
- Session Manager Instance, corresponding to the Session Manager server to be administered in 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 **Log On** (not shown). The screen shown below is then displayed.



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

AVAYA Avaya Aura® System Manager 6.2 Last Logged on at May 8, 2012 10:48 AM
Help | About | Change Password | Log off admin

Routing Home

Home / Elements / Routing

Introduction to Network Routing Policy Help ?

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

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

- Step 1: Create "Domains" of type SIP (other routing applications are referring domains of type SIP).
- Step 2: Create "Locations"
- Step 3: Create "Adaptations"
- Step 4: Create "SIP Entities"
 - SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"
 - Create all "other SIP Entities" (Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)
 - Assign the appropriate "Locations", "Adaptations" and "Outbound Proxies"
- Step 5: Create the "Entity Links"
 - Between Session Managers
 - Between Session Managers and "other 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 (**avayalab.com**). Navigate to **Routing → Domains** and click the **New** button in the right pane (not shown). In the new right pane that appears, 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 **avayalab.com** domain.

Home / Elements / Routing / Domains Help ?

Domain Management Commit Cancel

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.

1 Item [Refresh](#) Filter: Enable

Name	Type	Default	Notes
* avayalab.com	sip	<input type="checkbox"/>	

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. To add a location, navigate to **Routing → Locations** in the left-hand navigation pane and click the **New** button in the right pane (not shown).

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

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

The **Location Pattern** was not populated. The Location Pattern is used to identify call routing based on IP address. Session Manager matches the IP address against the patterns defined in this section. If a call is from a SIP Entity that does not match the IP address pattern then Session Manager uses the location administered for the SIP Entity. In this sample configuration Locations are added to SIP Entities (**Section 6.5**), so it was not necessary to add a pattern.

The following screen shows the addition of **SessionManager**, this location will be used for Session Manager. Click **Commit** to save.

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

Location Details [Commit](#) [Cancel](#)

General

* Name:

Notes:

Overall Managed Bandwidth

Managed Bandwidth Units:

Total Bandwidth:

Multimedia Bandwidth:

Audio Calls Can Take Multimedia Bandwidth: ☒

Per-Call Bandwidth Parameters

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

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

* Minimum Multimedia Bandwidth: Kbit/Sec

* Default Audio Bandwidth: Kbit/sec

Alarm Threshold

Overall Alarm Threshold: %

Multimedia Alarm Threshold: %

* Latency before Overall Alarm Trigger: Minutes

* Latency before Multimedia Alarm Trigger: Minutes

Location Pattern

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

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

IP Address Pattern	Notes
--------------------	-------

Note: Call bandwidth management parameters should be set per customer requirement.

Repeat the preceding procedure to create a separate Location for Communication Manager and Avaya SBCE. Displayed below is the screen for **Loc19-CMLab** used for Communication Manager.

[Home](#) / [Elements](#) / [Routing](#) / [Locations](#)

[Help ?](#)

Location Details

Commit

Cancel

General

*** Name:**

Loc19-CMLab

Notes:

Lab CM 10.64.19.205

Overall Managed Bandwidth

Managed Bandwidth Units:

Kbit/sec

Total Bandwidth:

Multimedia Bandwidth:

Audio Calls Can Take Multimedia Bandwidth:

☒

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location):

1000

Kbit/Sec

Maximum Multimedia Bandwidth (Inter-Location):

1000

Kbit/Sec

*** Minimum Multimedia Bandwidth:**

64

Kbit/Sec

*** Default Audio Bandwidth:**

80

Kbit/sec

Below is the screen for **Loc19-ASBCE** used for Avaya SBCE.

[Home](#) / [Elements](#) / [Routing](#) / [Locations](#)

[Help ?](#)

Location Details

Commit

Cancel

General

*** Name:**

Loc19-ASBCE

Notes:

Location 19 Avaya SBC

Overall Managed Bandwidth

Managed Bandwidth Units:

Kbit/sec

Total Bandwidth:

Multimedia Bandwidth:

Audio Calls Can Take Multimedia Bandwidth:

☒

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location):

1000

Kbit/Sec

Maximum Multimedia Bandwidth (Inter-Location):

1000

Kbit/Sec

*** Minimum Multimedia Bandwidth:**

64

Kbit/Sec

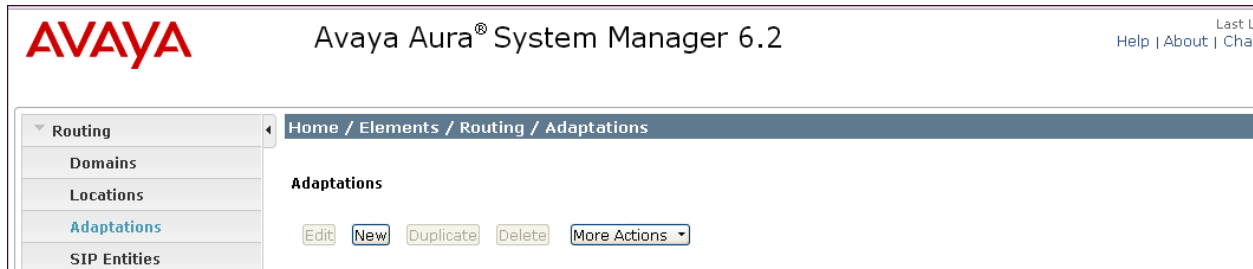
*** Default Audio Bandwidth:**

80

Kbit/sec

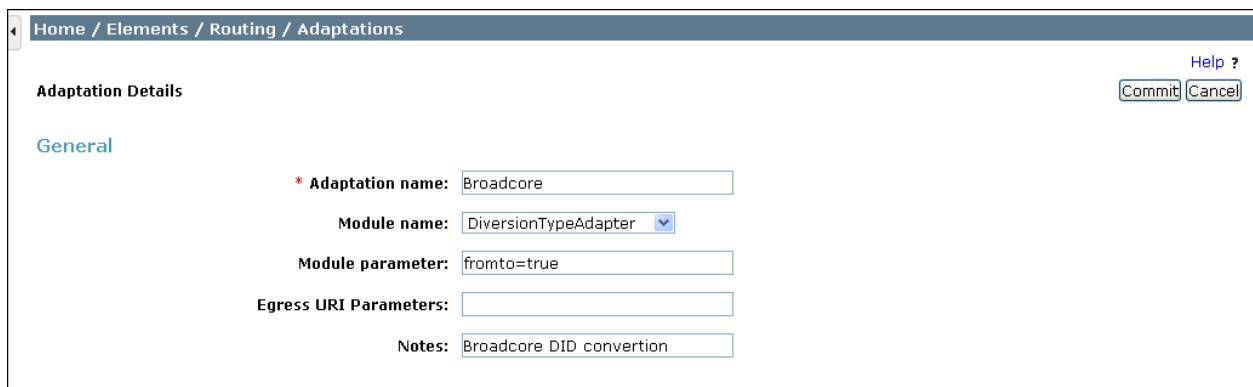
6.4. Adaptations

To view or change adaptations, select **Routing → Adaptations**. Click on the checkbox corresponding to the name of an adaptation and **Edit** to edit an existing adaptation, or the **New** button to add an adaptation. Click the **Commit** button after changes are completed.



The screenshot shows the Avaya Aura System Manager 6.2 web interface. The top header includes the Avaya logo, the product name "Avaya Aura® System Manager 6.2", and links for "Help | About | Chat". A left sidebar contains a navigation menu with "Routing" selected, and sub-items "Domains", "Locations", "Adaptations" (highlighted), and "SIP Entities". The main content area has a breadcrumb trail "Home / Elements / Routing / Adaptations" and a section titled "Adaptations". Below this title are buttons for "Edit", "New", "Duplicate", "Delete", and a "More Actions" dropdown menu.

The adapter named **Broadcore** will later be assigned to the SIP Entity linking Session Manager to Avaya SBCE for calls involving Broadcore/Masergy SIP Trunking. This adaptation uses the **DiversionTypeAdapter** to convert the History-Info header to a Diversion header and to convert digits between Communication Manager and Broadcore/Masergy. The **Module parameter fromto=true** will include the FROM and TO headers in the digit conversion.



The screenshot shows the "Adaptation Details" page in the Avaya Aura System Manager 6.2 interface. The breadcrumb trail is "Home / Elements / Routing / Adaptations". The page title is "Adaptation Details" with a "Help ?" link. There are "Commit" and "Cancel" buttons in the top right. The "General" tab is selected. The form contains the following fields:

- * Adaptation name:** Broadcore
- Module name:** DiversionTypeAdapter (selected from a dropdown)
- Module parameter:** fromto=true
- Egress URI Parameters:** (empty field)
- Notes:** Broadcore DID conversion

Scrolling down, the following screen shows a portion of the **Broadcore** adapter that can be used to convert digits between the Communication Manager extension numbers (user extensions, VDNs) and the 10 digit DID numbers assigned by Broadcore/Masergy. Since the adapter will be assigned to the SIP Entity receiving calls from Avaya SBCE for routing to Communication Manager, the settings for **Digit Conversion Incoming Calls to SM** correspond with incoming calls from Broadcore/Masergy to Communication Manager. Similarly, the settings for **Digit Conversion for Outgoing Calls from SM** correspond to outgoing calls from Communication Manager to the PSTN using the Broadcore/Masergy SIP Trunk service. In general, digit conversion such as this, that converts a Communication Manager extension (e.g., 12205) to a corresponding LDN or DID number known to the PSTN (e.g., 2135559111), can be performed in Session Manager as shown below.

Digit Conversion for Incoming Calls to SM

Add Remove

7 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"/>	* 2135559111	* 10	* 10		* 10	12205	destination ▼		
<input type="checkbox"/>	* 2135554089	* 10	* 10		* 10	12202	destination ▼		
<input type="checkbox"/>	* 3105557144	* 10	* 10		* 10	12203	destination ▼		
<input type="checkbox"/>	* 3235557673	* 10	* 10		* 10	12204	destination ▼		
<input type="checkbox"/>	* 4805551104	* 10	* 10		* 10	12201	destination ▼		
<input type="checkbox"/>	* 5035558019	* 10	* 10		* 10	12206	destination ▼		
<input type="checkbox"/>	* 8055552360	* 10	* 10		* 10	12299	destination ▼		

Select : All, None

Digit Conversion for Outgoing Calls from SM

Add Remove

8 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"/>	* 100xx	* 5	* 5		* 5	8055552360	origination ▼		
<input type="checkbox"/>	* 12200	* 5	* 5		* 5	8055552360	origination ▼		
<input type="checkbox"/>	* 12201	* 5	* 5		* 5	4805551104	origination ▼		
<input type="checkbox"/>	* 12202	* 5	* 5		* 5	2135554089	origination ▼		
<input type="checkbox"/>	* 12203	* 5	* 5		* 5	3105557144	origination ▼		
<input type="checkbox"/>	* 12204	* 5	* 5		* 5	3235557673	origination ▼		
<input type="checkbox"/>	* 12205	* 5	* 5		* 5	2135559111	origination ▼		
<input type="checkbox"/>	* 12206	* 5	* 5		* 5	5035558019	origination ▼		

Select : All, None

6.5. Add SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to it which includes Communication Manager and Avaya SBCE. Navigate to **Routing → SIP Entities** in the left-hand navigation pane and click on the **New** button in the right pane (not shown).

- **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 **Other** for Avaya SBCE.
- **Adaptation** This field is only present if **Type** is not set to **Session Manager**. If applicable, select the **Adaptation Name** that will be applied to this entity
- **Location** Select one of the locations defined previously
- **Time Zone** Select the time zone for the location above

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

Home / Elements / Routing / SIP Entities

SIP Entity Details [Help ?](#)

General

* Name:

* FQDN or IP Address:

Type:

Notes:

Location:

Outbound Proxy:

Time Zone:

Credential name:

SIP Link Monitoring

SIP Link Monitoring:

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. This section defines a default set of ports that Session Manager will use to listen for SIP requests, typically from registered SIP endpoints. Session Manager can also listen on additional ports defined elsewhere such as the ports specified in the SIP Entity Link definition in **Section 6.6**.

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

- **Port** Port number on which Session Manager can list for SIP Requests
- **Protocol** Transport protocol to be used to send SIP Requests
- **Default Domain** The domain used for the enterprise

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

For the compliance test, four **Port** entries were added.

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="5081"/>	TLS <input type="button" value="v"/>	avayalab.com <input type="button" value="v"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5071"/>	TLS <input type="button" value="v"/>	avayalab.com <input type="button" value="v"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5060"/>	TCP <input type="button" value="v"/>	avayalab.com <input type="button" value="v"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5061"/>	TLS <input type="button" value="v"/>	avayalab.com <input type="button" value="v"/>	<input type="text"/>

Select : All, None

The following screen shows the addition of Communication Manager. The **FQDN or IP Address** field is set to the IP address defined in **Section 5.3** of the procr interface on Communication Manager. The Location is set to the one defined for Communication Manager in **Section 6.3**.

[Home](#) / [Elements](#) / [Routing](#) / [SIP Entities](#)

[Help ?](#)

SIP Entity Details

Commit

Cancel

General

*** Name:**

Loc19-CM 5.2.1

*** FQDN or IP Address:**

10.64.19.55

Type:

CM

Notes:

Adaptation:

Location:

Loc19-CMLab

Time Zone:

America/Denver

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 Avaya SBCE SIP Entity. The **FQDN or IP Address** field is set to the IP address of its private network interface (see **Figure 1**). The **Adaptation** field is set to the Adaptation created in **Section 6.4** and the **Location** is set to the one defined for Avaya SBCE in **Section 6.3**. **Link Monitoring Enabled** was selected for **SIP Link Monitoring** using the specific time settings for **Proactive Monitoring Interval (in seconds)** and **Reactive Monitoring Interval (in seconds)** for the compliance test. These time settings should be adjusted or left at their default values per customer needs and requirements.

The screenshot displays the 'SIP Entity Details' configuration page for 'Loc19-ASBCE'. The page is divided into two main sections: 'General' and 'SIP Link Monitoring'. In the 'General' section, fields include Name (Loc19-ASBCE), FQDN or IP Address (10.64.19.100), Type (Other), Notes (Avaya SBC), Adaptation (Broadcore), Location (Loc19-ASBCE), Time Zone (America/Denver), and an unchecked 'Override Port & Transport with DNS SRV' checkbox. Below these are 'SIP Timer B/F (in seconds)' set to 4, a 'Credential name' field, 'Call Detail Recording' set to none, and 'CommProfile Type Preference' set to a default value. The 'SIP Link Monitoring' section shows 'SIP Link Monitoring' set to 'Link Monitoring Enabled', 'Proactive Monitoring Interval (in seconds)' set to 900, 'Reactive Monitoring Interval (in seconds)' set to 120, and 'Number of Retries' set to 1. Navigation links at the top include 'Home / Elements / Routing / SIP Entities', and action buttons 'Commit' and 'Cancel' are in the top right.

Home / Elements / Routing / SIP Entities

SIP Entity Details

Help ?

Commit Cancel

General

* Name: Loc19-ASBCE

* FQDN or IP Address: 10.64.19.100

Type: Other

Notes: Avaya SBC

Adaptation: Broadcore

Location: Loc19-ASBCE

Time Zone: America/Denver

Override Port & Transport with DNS SRV: ☐

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

Credential name:

Call Detail Recording: none

CommProfile Type Preference:

SIP Link Monitoring

SIP Link Monitoring: Link Monitoring Enabled

* Proactive Monitoring Interval (in seconds): 900

* Reactive Monitoring Interval (in seconds): 120

* Number of Retries: 1

6.6. Add Entity Links

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

- **Name** Enter a descriptive name
- **SIP Entity 1** Select the SIP Entity for 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 Communication Manager, this must match the **Far-end Listen Port** defined on the Communication Manager signaling group in **Section 5.7**
- **SIP Entity 2** Select the name of the other system. For Communication Manager, 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 Communication Manager, this must match the **Near-end Listen Port** defined on the Communication Manager signaling group in **Section 5.7**
- **Trusted** Check this box. **Note:** If this box is not checked, calls from the associated SIP Entity specified in **Section 6.5** will be denied.

Click **Commit** to save. The following screens illustrate the Entity Links to Communication Manager and Avaya SBCE.

Entity Link to Communication Manager:

Entity Links								Commit	Cancel
1 Item Refresh								Filter: Enable	
Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes		
* SM to Loc19-CM5.2.1	* DenverSM	TCP	* 5060	* Loc19-CM 5.2.1	* 5060	Trusted	Loc 19 CM 5.2.1		

Entity Link to Avaya SBCE:

Entity Links								Commit	Cancel
1 Item Refresh								Filter: Enable	
Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes		
* SM to Loc19-ASBCE	* DenverSM	TCP	* 5060	* Loc19-ASBCE	* 5060	Trusted	To Avaya SBC		

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 Avaya SBCE. To add a routing policy, navigate to **Routing → Routing Policies** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). The screen below is displayed. Fill in the following:

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

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

In the **SIP Entity as Destination** section, click **Select**. The **SIP Entity List** page opens (not shown). Select the appropriate SIP entity to which this routing policy applies and click **Select** (not shown). 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 Avaya SBCE.

Routing Policy for Communication Manger:

Home / Elements / Routing / Routing Policies

Routing Policy Details

Help ?

Commit Cancel

General

* Name: To Loc 19 CM5.2.1

Disabled: ☐

* Retries: 0

Notes: Location 19 CM 5.2.1

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Loc19-CM 5.2.1	10.64.19.55	CM	

Routing Policy for Avaya SBCE:

The screenshot shows the 'Routing Policy Details' form in a web application. The breadcrumb trail is 'Home / Elements / Routing / Routing Policies'. The form title is 'Routing Policy Details'. There are 'Commit' and 'Cancel' buttons in the top right, and a 'Help ?' link. The 'General' section contains the following fields: 'Name' (To-ASBCE), 'Disabled' (checkbox), 'Retries' (0), and 'Notes' (To Avaya SBCE). Below this is the 'SIP Entity as Destination' section with a 'Select' button. At the bottom is a table with the following data:

Name	FQDN or IP Address	Type	Notes
Loc19-ASBCE	10.64.19.100	Other	Avaya SBC

6.8. Add Dial Patterns

Dial Patterns are needed to route calls through Session Manager. For the compliance test, dial patterns were created to route calls from Communication Manager to Broadcore/Masergy and vice versa. Dial Patterns define which route policy will be selected for a particular call based on the dialed digits, destination domain and originating location. To add a dial pattern, navigate to **Routing → Dial Patterns** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). Fill in the following, as shown in the screens below:

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

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

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

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

The following screen illustrates an example dial pattern used to verify outbound calls from the enterprise to the PSTN. When a Communication Manager user dials a PSTN number such as 9-1-303-XXX-XXX, Communication Manager sends the call to Session Manager, via the processor Ethernet. Session Manager will match the dial pattern shown below and send the call to the Avaya SBCE using route policy **To-ASBCE**.

Home / Elements / Routing / Dial Patterns

[Help ?](#)

Dial Pattern Details

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 <small>1 ▲</small>	Originating Location Notes	Routing Policy Name	Rank <small>2 ▲</small>	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Loc19-CMLab	Lab CM 10.64.19.205	To-ASBCE	0	<input type="checkbox"/>	Loc19-ASBCE	

Select : [All](#), [None](#)

The following screen illustrates an example dial pattern used to verify inbound PSTN calls to the enterprise. When a user on the PSTN dials a number assigned to the Broadcore/Masergy SIP Trunk service, such as 213-555-9111, Broadcore/Masergy delivers the number to the enterprise, and the Avaya SBCE sends the call to Session Manager. Session Manager will then convert the digits to the corresponding five digit extension number using an Adaptation created in **Section 6.4**, in this case 12205. The pattern below matches on a range of numbers 122XX. Under **Originating Locations and Routing Policies**, the routing policy named **To Loc19 CM5.2.1** is chosen. This sends the call to Communication Manager as described previously.

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 <small>1 ▲</small>	Originating Location Notes	Routing Policy Name	Rank <small>2 ▲</small>	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	To Loc 19 CM5.2.1	0	<input type="checkbox"/>	Loc19-CM 5.2.1	Location 19 CM 5.2.1

Select : All, None

6.9. Add/Verify Avaya Aura® Session Manager Instance

The creation of a Session Manager Instance 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, navigate to **Elements → Session Manager → Session Manager Administration** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). If the Session Manager instance already exists, click **View** (not shown) to view the configuration. Enter/verify the data as described below and shown in the screen below:

In the **General** section, enter or verify 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 displays the 'Edit Session Manager' configuration page. At the top, there is a breadcrumb trail: 'Home / Elements / Session Manager'. A 'Help ?' link is located in the top right corner. Below the breadcrumb, the title 'Edit Session Manager' is centered, with 'Commit' and 'Cancel' buttons to its right. A horizontal menu bar contains several options: 'General', 'Security Module', 'NIC Bonding', 'Monitoring', 'CDR', 'Personal Profile Manager (PPM)', 'Connection Settings', 'Event Server', 'Expand All', and 'Collapse All'. The 'General' tab is currently selected. Under the 'General' tab, the following fields are visible: 'SIP Entity Name' with the value 'DenverSM', 'Description' with the value 'Session Manager', '*Management Access Point Host Name/IP' with the value '10.80.150.210', and '*Direct Routing to Endpoints' with a dropdown menu set to 'Enable'.

In the **Security Module** section, enter or verify 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.

Security Module ▾

SIP Entity IP Address

10.64.19.210

*Network Mask

255.255.255.0

*Default Gateway

10.64.19.1

*Call Control PHB

46

*QOS Priority

6

*Speed & Duplex

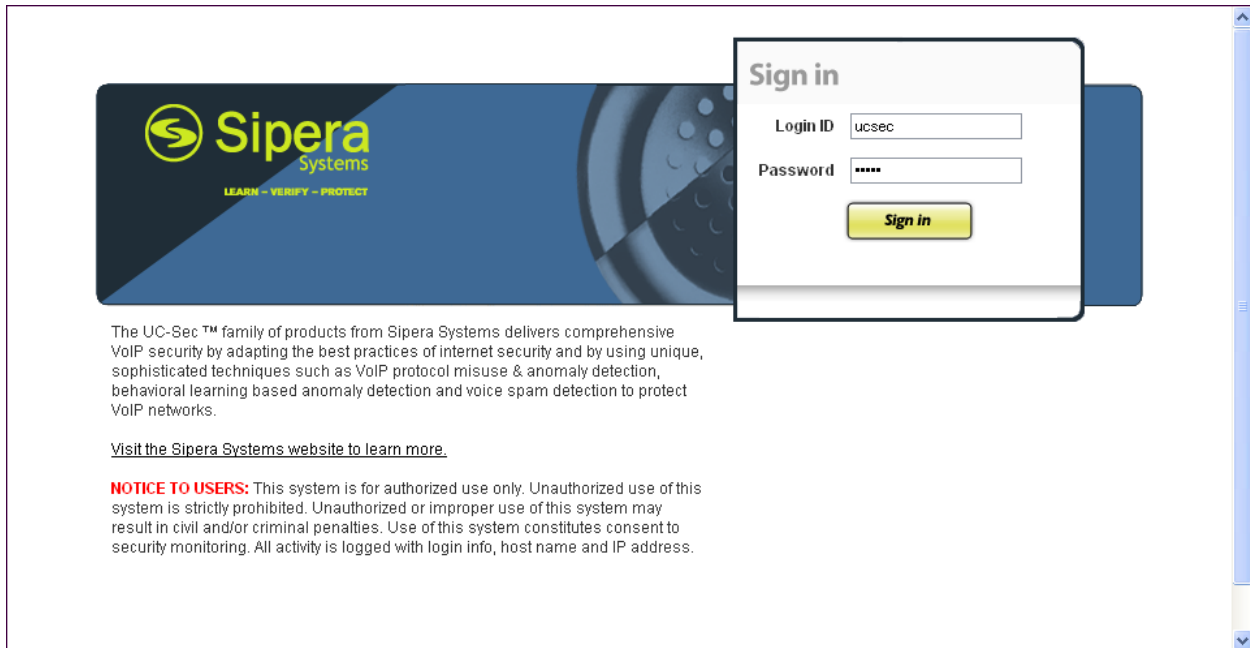
Auto ▾

VLAN ID

7. Configure Avaya Session Border Controller for Enterprise

This section covers the configuration of the Avaya SBCE. It is assumed that the Avaya SBCE software has already been installed.

Log in with the appropriate credentials. Click **Sign In**.

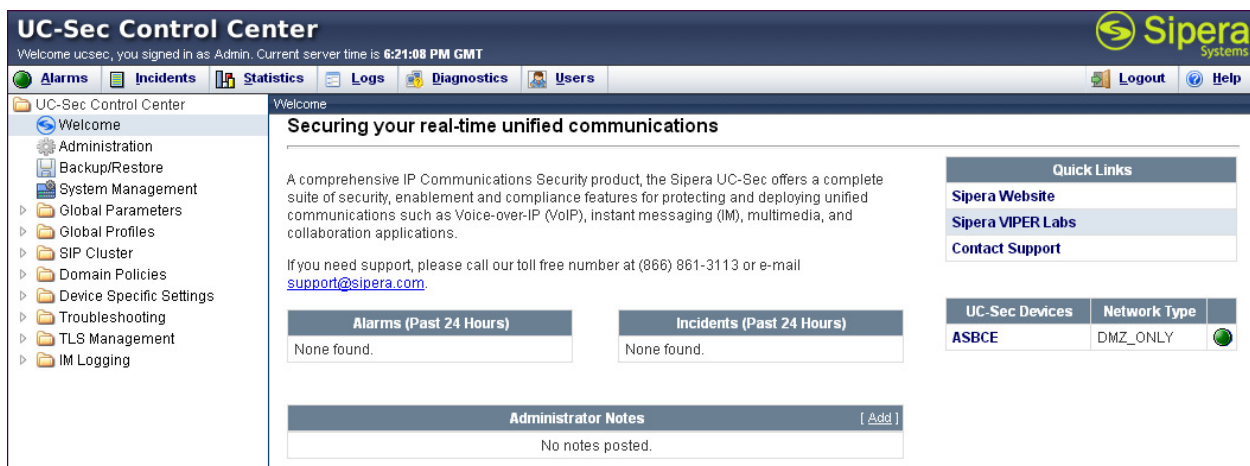


The UC-Sec™ family of products from Sipera Systems delivers comprehensive VoIP security by adapting the best practices of internet security and by using unique, sophisticated techniques such as VoIP protocol misuse & anomaly detection, behavioral learning based anomaly detection and voice spam detection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

The main page of the UC-Sec Control Center will appear.



UC-Sec Control Center
Welcome ucsec, you signed in as Admin. Current server time is 6:21:08 PM GMT

Alarms **Incidents** **Statistics** **Logs** **Diagnostics** **Users** **Logout** **Help**

UC-Sec Control Center
Welcome

Securing your real-time unified communications

A comprehensive IP Communications Security product, the Sipera UC-Sec offers a complete suite of security, enablement and compliance features for protecting and deploying unified communications such as Voice-over-IP (VoIP), instant messaging (IM), multimedia, and collaboration applications.

If you need support, please call our toll free number at (866) 861-3113 or e-mail support@sipera.com.

Alarms (Past 24 Hours)
None found.

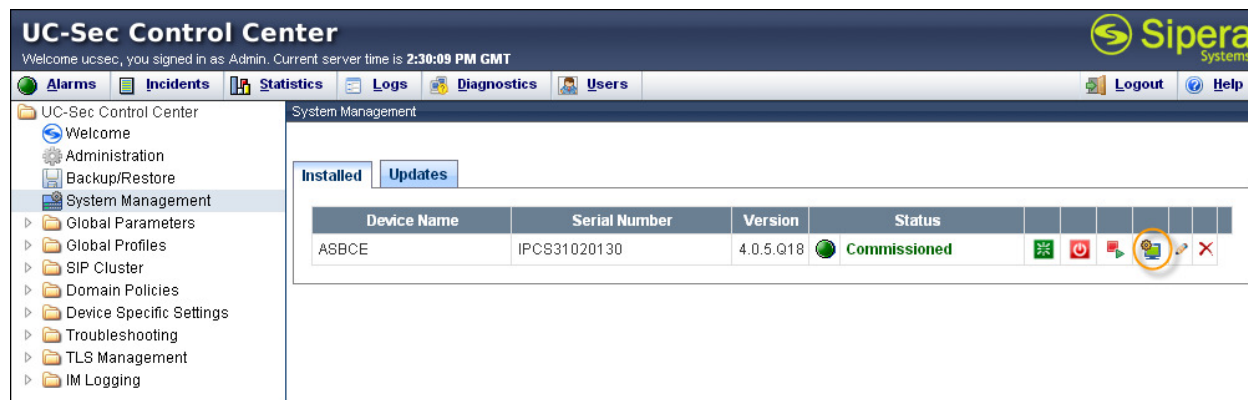
Incidents (Past 24 Hours)
None found.

Administrator Notes [Add]
No notes posted.

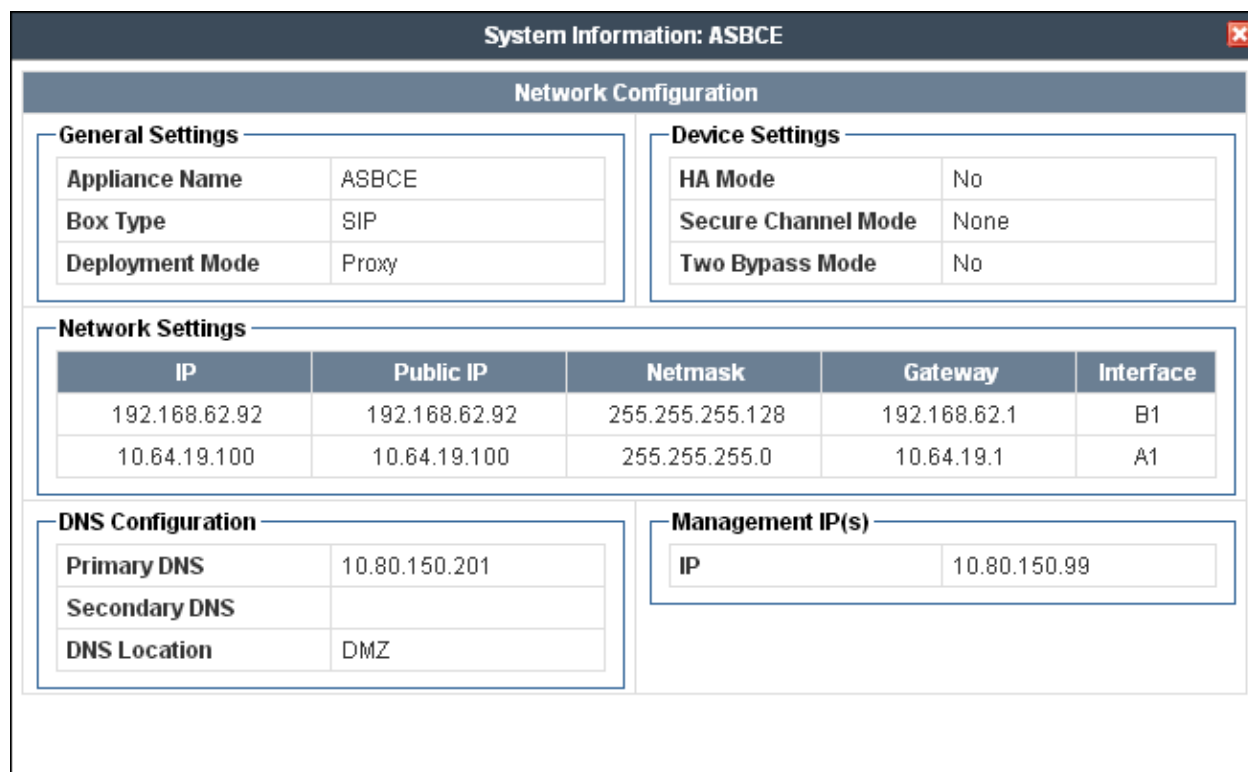
Quick Links
[Sipera Website](#)
[Sipera VIPER Labs](#)
[Contact Support](#)

UC-Sec Devices	Network Type
ASBCE	DMZ_ONLY

To view system information that was configured during installation, navigate to **UC-Sec Control Center → System Management**. A list of installed devices is shown in the right pane. In the case of the sample configuration, a single device named **ASBCE** is shown. To view the configuration of this device, click the monitor icon as highlighted below.



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



7.1. Network Management

The Network Management screen is where the network interface settings are configured and enabled. During the installation process of Avaya SBCE, certain network-specific information is defined such as device IP address(es), public IP address(es), netmask, gateway, etc., to interface the device to the network. It is this information that populates the various Network Management tab displays, which can be edited as needed to optimize device performance and network efficiency. Navigate to **UC-Sec Control Center → Device Specific Settings → Network Management** and verify the IP addresses assigned to the interfaces and that the interfaces are enabled. The following screen shows the private interface is assigned to **A1** and the external interface is assigned to **B1**.

The screenshot shows the 'UC-Sec Control Center' interface with the 'Network Configuration' tab selected. The left sidebar shows the navigation tree with 'Device Specific Settings' expanded. The main content area displays the 'Network Configuration' tab for device 'ASBCE'. 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 input fields for 'A1 Netmask' (255.255.255.0), 'A2 Netmask', 'B1 Netmask' (255.255.255.128), and 'B2 Netmask'. An 'Add IP' button is present. Below the netmask fields is a table with columns: IP Address, Public IP, Gateway, Interface, and a status icon. The table contains two entries: one for IP 192.168.62.92 assigned to interface B1, and another for IP 10.64.19.100 assigned to interface A1. 'Save Changes' and 'Clear Changes' buttons are at the bottom right of the configuration area.

IP Address	Public IP	Gateway	Interface	
192.168.62.92		192.168.62.1	B1	✗
10.64.19.100		10.64.19.1	A1	✗

The following screen shows interface **A1** and **B1** are **Enabled**. To enable an interface click its **Toggle State** button.

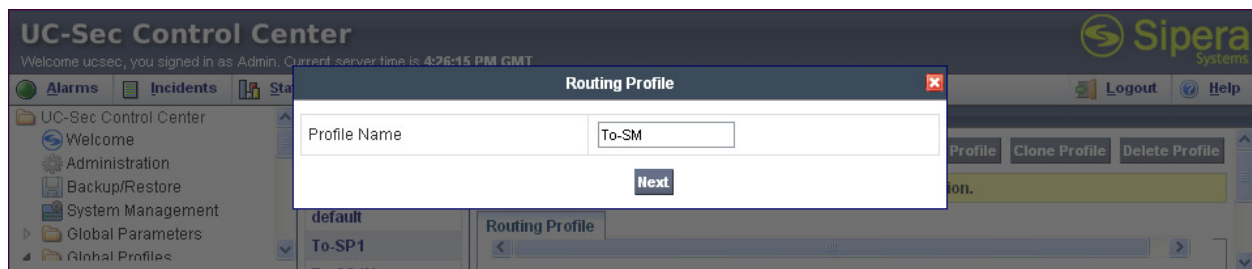
The screenshot shows the 'UC-Sec Control Center' interface with the 'Interface Configuration' tab selected. The left sidebar is the same as the previous screenshot. The main content area displays the 'Interface Configuration' tab for device 'ASBCE'. It shows a table with columns: Name, Administrative Status, and a 'Toggle State' button. The table lists four interfaces: A1 (Enabled), A2 (Disabled), B1 (Enabled), and B2 (Disabled). Each interface has a 'Toggle State' button next to it.

Name	Administrative Status	
A1	Enabled	Toggle State
A2	Disabled	Toggle State
B1	Enabled	Toggle State
B2	Disabled	Toggle State

7.2. Routing Profile

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

Create a Routing Profile for Session Manager and Broadcore/Masergy SIP Trunk Service. To add a routing profile, navigate to **UC-Sec Control Center → Global Profiles → Routing** and select **Add Profile**. Enter a **Profile Name** and click **Next** to continue.

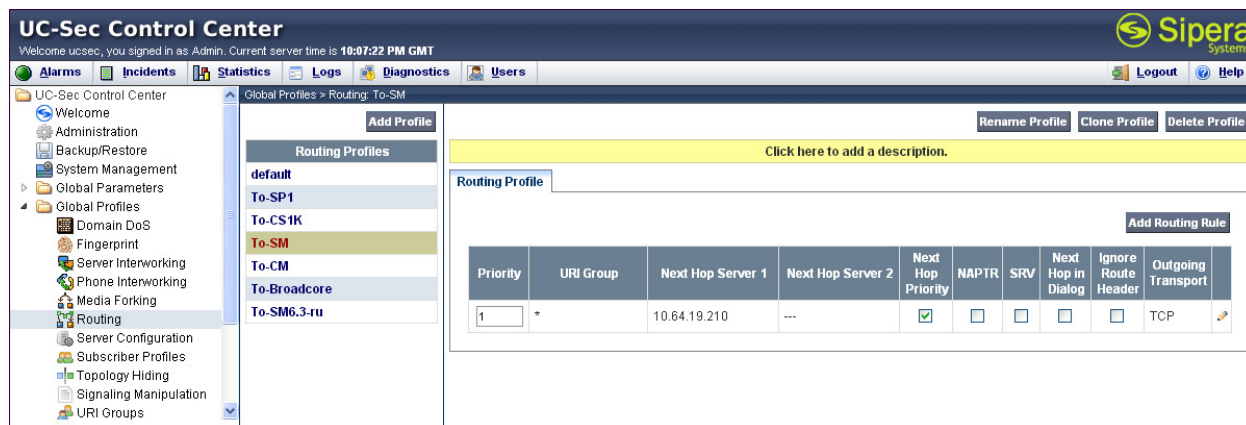


In the new window that appears (not shown), enter the following values. Use default values for all remaining fields:

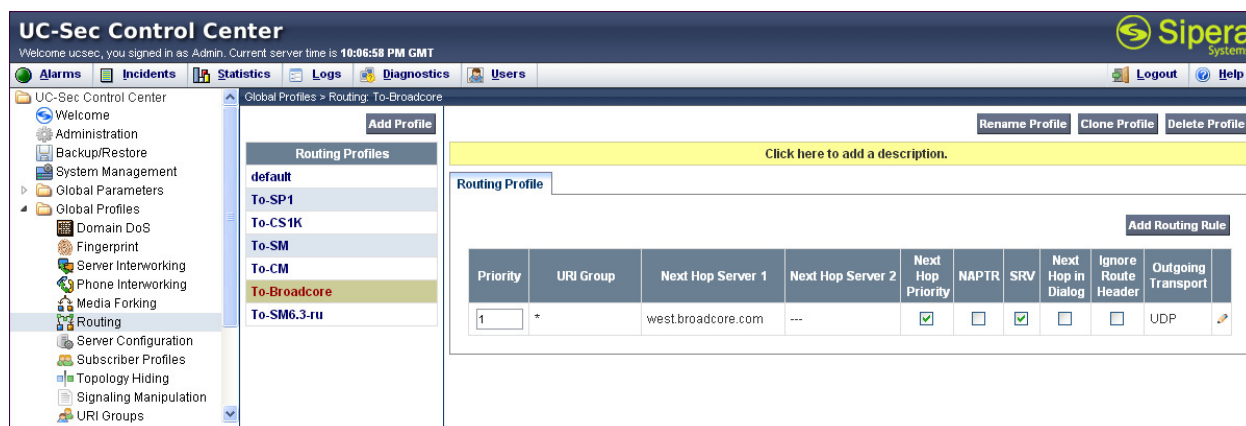
- **URI Group:** Select "*" from the drop down box.
- **Next Hop Server 1:** Enter the Domain Name or IP address of the Primary Next Hop server.
- **Next Hop Server 2:** (Optional) Enter the Domain Name or IP address of the secondary Next Hop server.
- **Routing Priority Based on Next Hop Server:** Checked.
- **Outgoing Transport:** Choose the protocol used for transporting outgoing signaling packets.

Click **Finish**.

In the shared test environment the following screen shows Routing Profile **To-SM** created for Session Manager. The **Next Hop Server 1** IP address must match the IP address of Session Manager Entity created in **Section 6.5**. The **Outgoing Transport** is set to **TCP** and matched the **Protocol** set in the Session Manager Entity Link for Avaya SBCE in **Section 6.6**.



The following screen shows Routing Profile **To-Broadcore** created for Broadcore/Masergy. In the **Next Hop Server 1** field enter the Fully Qualified Domain Name that Broadcore/Masergy uses to listen for SIP traffic. In the sample configuration **west.broadcore.com** was used. Select **SRV** and enter **UDP** for the **Outgoing Transport** field.



7.3. Topology Hiding Profile

The Topology Hiding profile manages how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the integrity of the network. It hides the topology of the enterprise network from external networks.

Create a Topology Hiding Profile for the enterprise and Broadcore/Masergy SIP Trunk Service. In the sample configuration, the **Enterprise** and **Broadcore Topology** profiles were cloned from the default profile. To clone a default profile, navigate to **UC-Sec Control Center → Global Profiles → Topology Hiding**. Select the **default** profile and click on **Clone Profile** as shown below.

UC-Sec Control Center
 Welcome ucsec, you signed in as Admin. Current server time is 10:41:18 PM GMT

Alarms Incidents Statistics Logs Diagnostics Users Logout Help

UC-Sec Control Center
 Administration
 Backup/Restore
 System Management
 Global Parameters
 Global Profiles
 Domain DoS
 Fingerprint
 Server Interworking
 Phone Interworking
 Media Forking
 Routing
 Server Configuration
 Subscriber Profiles
 Topology Hiding
 Signaling Manipulation
 URI Groups

Global Profiles > Topology Hiding: default

Add Profile

Topology Hiding Profiles
 default

Clone Profile

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

Topology Hiding

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

Edit

Enter a descriptive name for the new profile and click **Finish**.

Clone Profile

Profile Name	default
Clone Name	Enterprise

Finish

Edit the **Enterprise** profile to overwrite the headers shown below to the enterprise domain. The **Overwrite Value** should match the Domain set in Session Manager (**Section 6.2**). Click **Finish** to save the changes.

Edit Topology Hiding Profile ✕

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

Finish

Use the same procedure to clone the default profile for Broadcore/Masergy. Edit the profile to change the **FROM** header's **Criteria** to **Domain** and **Replace Action** to **Next Hop** as shown below.

UC-Sec Control Center
Welcome ucsec, you signed in as Admin. Current server time is 4:50:42 PM GMT

Alarms Incidents Statistics Logs Diagnostics Users Logout Help

- UC-Sec Control Center
- Welcome
- Administration
- Backup/Restore
- System Management
- Global Parameters
- Global Profiles
 - Domain DoS
 - Fingerprint
 - Server Interworking
 - Phone Interworking
 - Media Forking
 - Routing
 - Server Configuration
 - Subscriber Profiles
 - Topology Hiding
 - Signaling Manipulation
 - URI Groups

Global Profiles > Topology Hiding: Broadcore Topology

Add Profile

Rename Profile
Clone Profile
Delete Profile

Click here to add a description.

Topology Hiding

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

Edit

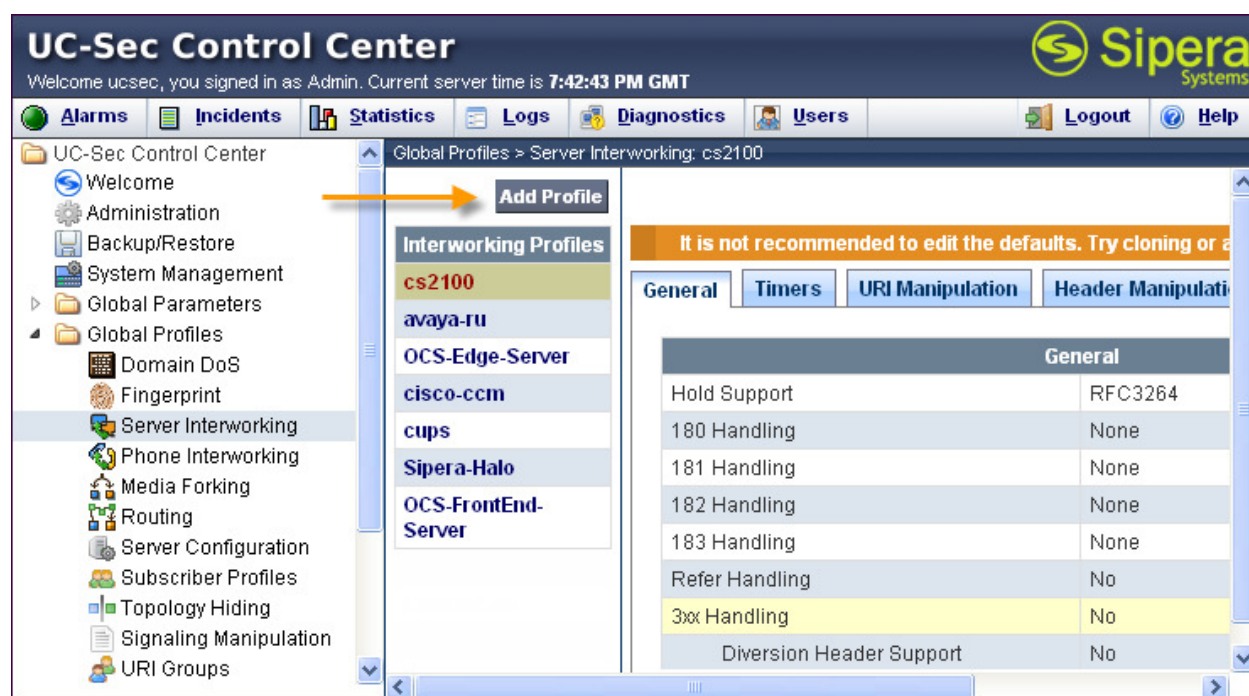
7.4. Server Interworking Profile

The Server Interworking profile configures and manages various SIP call server-specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters (for HA deployments), DoS security statistics, and trusted domains. Interworking Profile features are configured based on different Trunk Servers. There are default profiles available that may be used as is, or modified, or new profiles can be configured as described below.

In the sample configuration, separate Server Interworking Profiles were created for Enterprise and Broadcore/Masergy.

7.4.1. Server Interworking Profile – Enterprise

To create a new Server Interworking Profile for the enterprise, navigate to **UC-Sec Control Center** → **Global Profiles** → **Server Interworking** and click on **Add Profile** as shown below.



Enter a descriptive name for the new profile and click **Next** to continue.

Interworking Profile

Profile Name

Lab Interworking

Next

In the new window that appears, select **RFC3264 - a=sendonly** and check the **T.38 Support** field. Use default values for all remaining fields. Click **Next** to continue.

Editing Profile: Lab Interworking

General

Hold Support	<input type="radio"/> None <input type="radio"/> RFC2543 - c=0.0.0.0 <input checked="" type="radio"/> RFC3264 - a=sendonly
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input type="checkbox"/>
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input type="checkbox"/>
T.38 Support	<input checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543

Next

Default values can be used for the next two windows that appear. Click **Next** to continue.

Interworking Profile

Privacy

Privacy Enabled	<input type="checkbox"/>
User Name	
P-Asserted-Identity	<input type="checkbox"/>
P-Preferred-Identity	<input type="checkbox"/>
Privacy Header	

DTMF

DTMF Support	<input checked="" type="radio"/> None <input type="radio"/> SIP NOTIFY <input type="radio"/> SIP INFO
--------------	---

Back Next

Interworking Profile

Configuration is not required. All fields are optional.

SIP Timers

Min-SE		seconds, [90 - 86400]
Init Timer		milliseconds, [50 - 1000]
Max Timer		milliseconds, [200 - 8000]
Trans Expire		seconds, [1 - 64]
Invite Expire		seconds, [180 - 300]

Transport Timers

TCP Connection Inactive Timer		seconds, [600 - 3600]
-------------------------------	--	-----------------------

Back Next

On the **Advanced Settings** window uncheck the following default settings:

- **Topology Hiding: Change Call-ID**
- **Change Max Forwards**

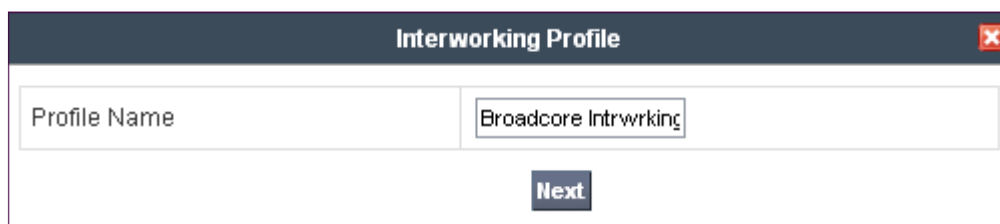
Click **Finish** to save changes.

Advanced Settings	
Record Routes	<input type="radio"/> None <input type="radio"/> Single Side <input checked="" type="radio"/> Both Sides
Topology Hiding: Change Call-ID	<input checked="" type="checkbox"/>
Call-Info NAT	<input type="checkbox"/>
Change Max Forwards	<input checked="" type="checkbox"/>
Include End Point IP for Context Lookup	<input type="checkbox"/>
OCS Extensions	<input type="checkbox"/>
AVAYA Extensions	<input type="checkbox"/>
NORTEL Extensions	<input type="checkbox"/>
SLIC Extensions	<input type="checkbox"/>
Diversion Manipulation	<input type="checkbox"/>
Diversion Header URI	<input type="text"/>
Metaswitch Extensions	<input type="checkbox"/>
Reset on Talk Spurt	<input type="checkbox"/>
Reset SRTP Context on Session Refresh	<input type="checkbox"/>
Has Remote SBC	<input checked="" type="checkbox"/>
Route Response on Via Port	<input type="checkbox"/>
Cisco Extensions	<input type="checkbox"/>

Back **Finish**

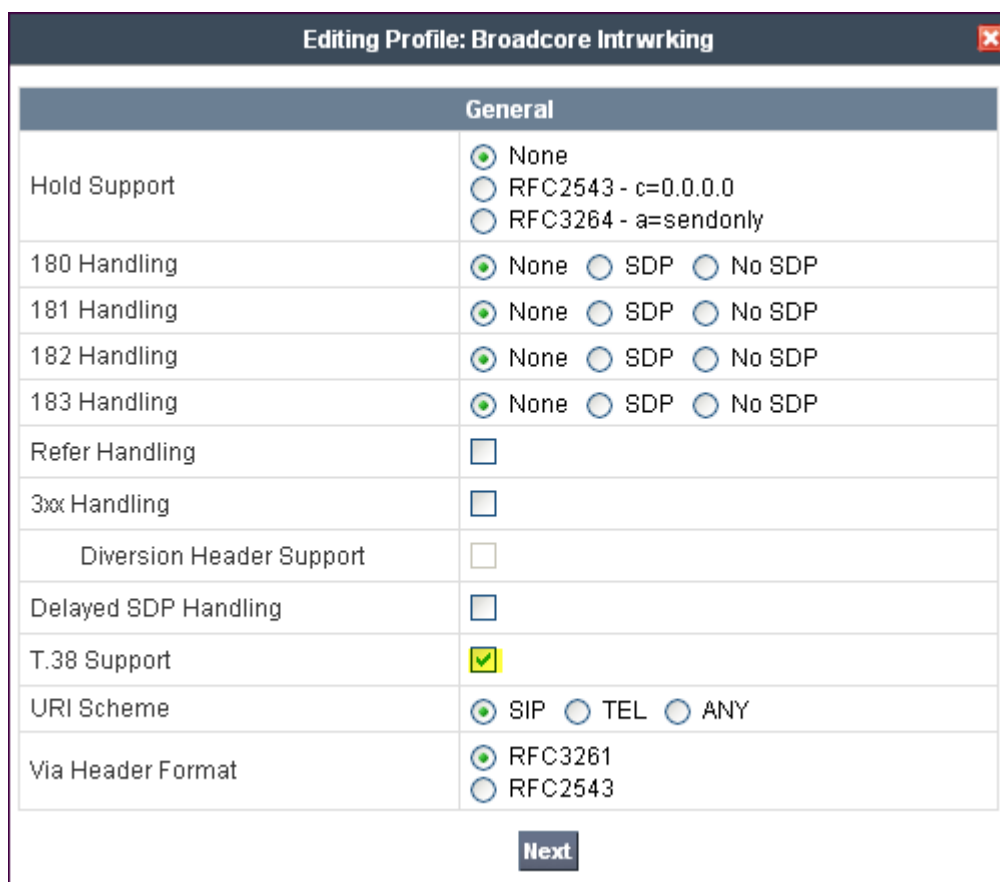
7.4.2. Server Interworking Profile – Broadcore/Masergy

Click the **Add Profile** button (not shown) to add a new profile or select an existing interworking profile. If adding a profile, a screen such as the following is displayed. Enter an appropriate **Profile Name** such as **Broadcore Intrwrking** shown below. Click **Next**.



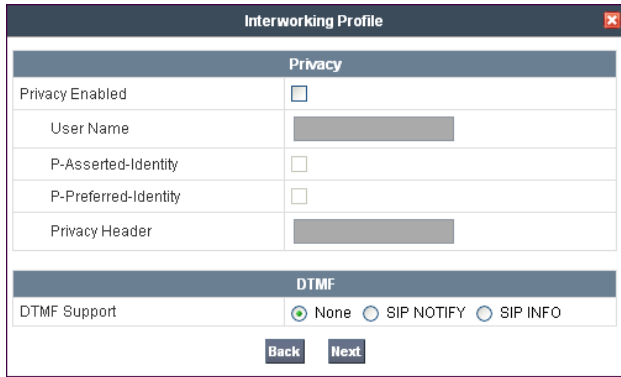
Interworking Profile	
Profile Name	Broadcore Intrwrking
Next	

In the new window that appears, check the **T.38 Support** field. Use default values for all remaining fields. Click **Next** to continue.



Editing Profile: Broadcore Intrwrking	
General	
Hold Support	<input checked="" type="radio"/> None <input type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input type="checkbox"/>
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input type="checkbox"/>
T.38 Support	<input checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543
Next	

Default values can be used for the next two windows that appear. Click **Next** to continue.



Interworking Profile

Privacy

Privacy Enabled ☐

User Name

P-Asserted-Identity ☐

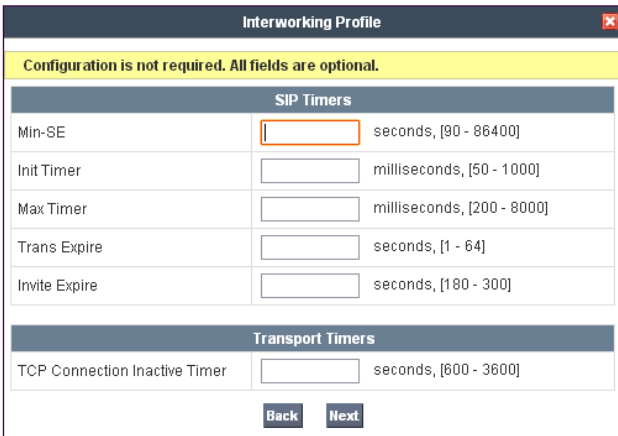
P-Preferred-Identity ☐

Privacy Header

DTMF

DTMF Support ☒ None ☐ SIP NOTIFY ☐ SIP INFO

Back **Next**



Interworking Profile

Configuration is not required. All fields are optional.

SIP Timers

Min-SE seconds, [90 - 86400]

Init Timer milliseconds, [50 - 1000]

Max Timer milliseconds, [200 - 8000]

Trans Expire seconds, [1 - 64]

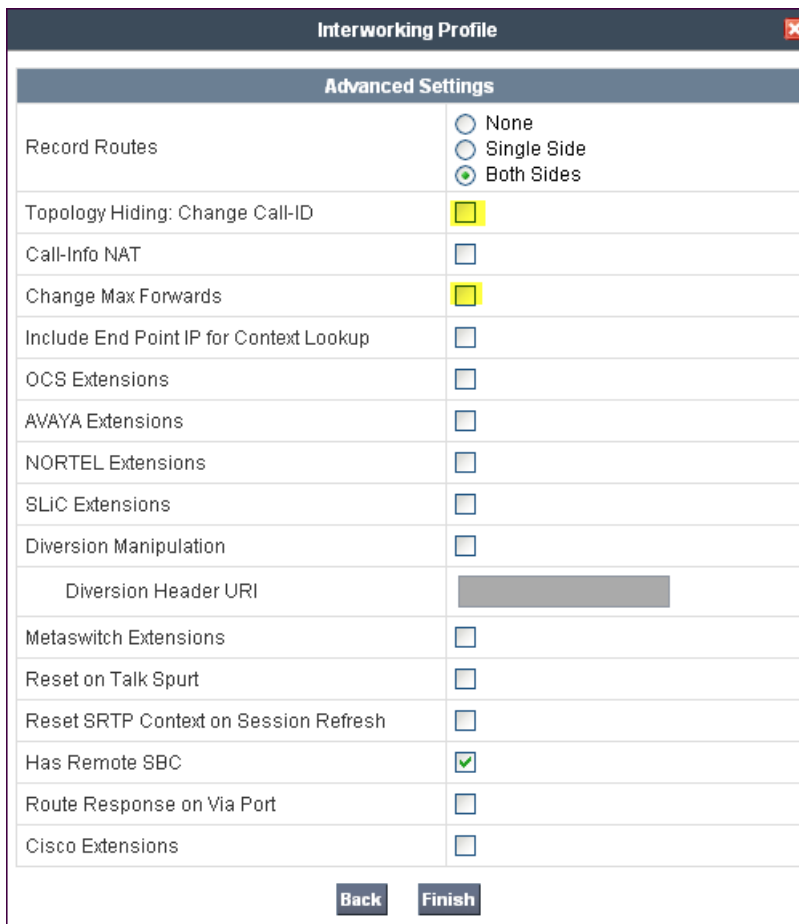
Invite Expire seconds, [180 - 300]

Transport Timers

TCP Connection Inactive Timer seconds, [600 - 3600]

Back **Next**

On the **Advanced Settings** window uncheck **Topology Hiding: Change Call-ID** and **Change Max Forwards**. Click **Finish** to save changes.



Interworking Profile

Advanced Settings

Record Routes ☐ None ☐ Single Side ☒ Both Sides

Topology Hiding: Change Call-ID ☐

Call-Info NAT ☐

Change Max Forwards ☐

Include End Point IP for Context Lookup ☐

OCS Extensions ☐

AVAYA Extensions ☐

NORTEL Extensions ☐

SLIC Extensions ☐

Diversion Manipulation ☐

Diversion Header URI

Metaswitch Extensions ☐

Reset on Talk Spurt ☐

Reset SRTP Context on Session Refresh ☐

Has Remote SBC ☒

Route Response on Via Port ☐

Cisco Extensions ☐

Back **Finish**

7.5. Signaling Manipulation

The Signaling Manipulation feature allows the ability to add, change and delete any of the headers in a SIP message. This feature will add the ability to configure such manipulation in a highly flexible manner using a proprietary scripting language called SigMa.

The SigMa scripting language is designed to express any of the SIP header manipulation operations to be done by the Avaya SBCE. Using this language, a script can be written and tied to a given flow through the Avaya SBCE GUI. The Avaya SBCE appliance then interprets this script at the given entry point or “hook point”.

These application notes will not discuss the full feature of the Signaling Manipulation but will show an example of a script created during compliance testing. The sample script was used to change the From user to the pilot number for outbound calls in order to be authenticated on the Broadcore/Masergy network. It also adds an Expires header with a value recommended by Broadcore/Masergy. For inbound calls the script was used to change the SIP trunk pilot number presented in the Request URI to the number in the To header so calls can be routed properly through Communication Manger.

To create a new Signaling Manipulation, navigate to **UC-Sec Control Center → Global Profiles → Signaling Manipulation** and click on **Add Script**. A new blank SigMa Editor window will pop up.

In this compliance testing, the script named **Broadcore Script** was created as shown below:

```

within session "ALL"
{
  act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
  {
    //Insert Pilot number in the FROM header
    %fromuser = %HEADERS["From"][1].URI.USER;
    %HEADERS["From"][1].URI.USER = "2135559111";

    //Remove "sendonly" attribute for Music on Hold
    if(%SDP[1]["s"]["m"][1].ATTRIBUTES["sendonly"][1]="") then
    {
      remove(%SDP[1]["s"]["m"][1].ATTRIBUTES["sendonly"][1]);
    }
  }
}

//Add "Expires" header to REGISTER message
within session "ALL"
{
  act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING" and
  %METHOD="REGISTER"
  {
    %HEADERS["Expires"][1] = "3600";
  }
}

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

// Return FROM header to original form
within session "ALL"
{
  act on response where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING"
  {
    %HEADERS["From"][1].URI.USER = %fromuser;
  }
}

```

In the Signaling Manipulation script named **Broadcore Script** above, the statement **act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"** specifies the portion of the script that will take effect on request SIP messages for an outbound call and the manipulation will be done after routing. The manipulation will be according to the rules contained in this statement.

-SigMa rules to populate the Pilot DID in the From header. All calls must have the Pilot DID in the From header in order to be authenticated on the Broadcore/Masergy network. The original FROM user is saved as variable “%fromuser” so it can be converted back later on in the script. Then it is changed to the pilot number.

```
//Insert Pilot number in the FROM header
%fromuser = %HEADERS["From"][1].URI.USER;
%HEADERS["From"][1].URI.USER = "2135559111";
```

-SigMa rules to delete the Sendonly attribute. This will remove the media attribute sent by Communication Manager when a call is placed on hold. The Broadcore/Masergy SIP Trunk Service will play its own music source when the “sendonly” media attribute is received. The “sendrecv” media attribute is assumed as the default for the session when no other attribute is sent. So rather than replacing “sendonly” with “sendrecv”, the “sendonly” media attribute was simply removed. This allows internal music/message on hold to be played while the call is on hold.

```
//Remove "sendonly" attribute for Music on Hold
if(%SDP[1]["s"]["m"][1].ATTRIBUTES["sendonly"][1]="") then
{
    remove(%SDP[1]["s"]["m"][1].ATTRIBUTES["sendonly"][1]);
}
}
```

In the Signaling Manipulation script named **Broadcore Script** further above, the statement **act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING" and %METHOD="REGISTER"** is similar to the previous statement with the exception that with the addition of **and %METHOD="REGISTER"** it will only act on SIP Register messages. The manipulation will be according to the rules contained in this statement.

-SigMa rules to add an Expires header to Register Methods. This will add an Expires header with a value recommended by Broadcore/Masergy to all SIP Register messages. This is required to set the proper register expiration for the Broadcore/Masergy application server.

```
//Add "Expires" header to REGISTER message
within session "ALL"
{
    act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
    and %METHOD="REGISTER"
    {
        %HEADERS["Expires"][1] = "3600";
    }
}
```

In the Signaling Manipulation script named **Broadcore Script** further above, the statement **act on request where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING"** specifies the portion of the script that will take effect on request SIP messages (i.e., initial INVITE) for an inbound call and the manipulation will be done before routing. The manipulation will be according to the rules contained in this statement.

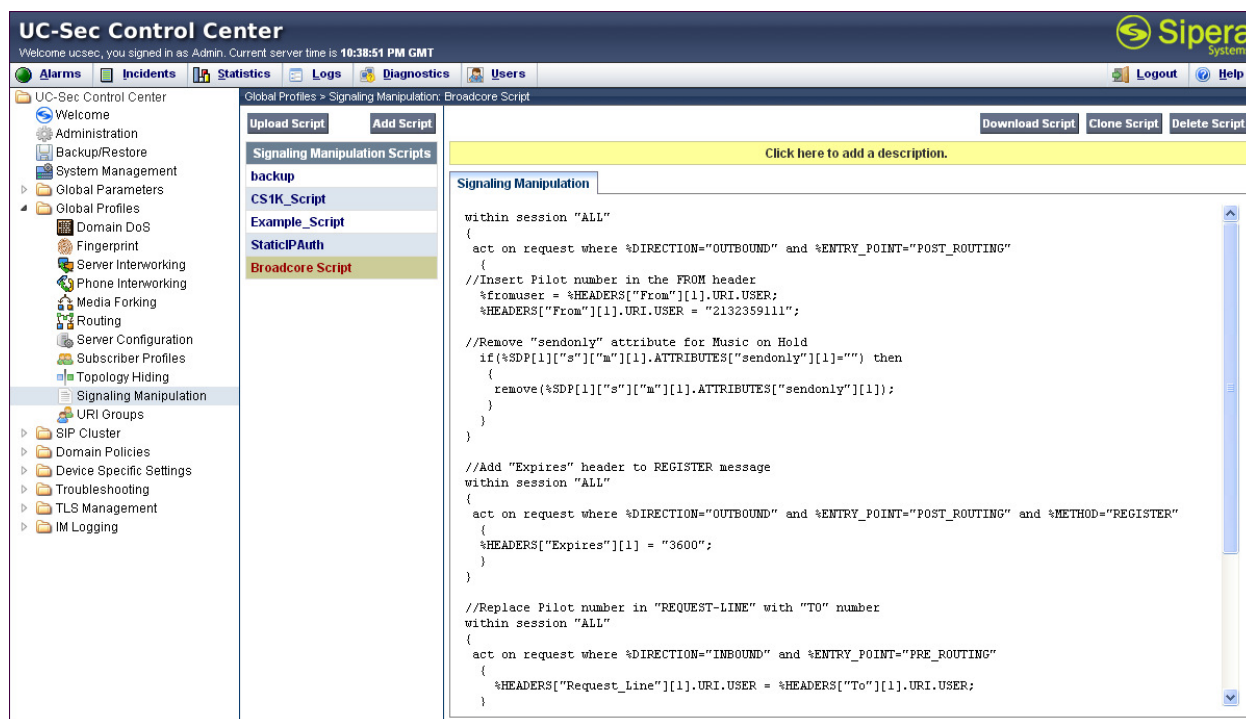
-SigMa rules to manipulate the calling number in Request URI header. For incoming calls the Request URI will always be the Pilot DID as defined by Broadcore/Masergy. The Pilot DID needs to be removed and the actual called number should be populated in its place. The called number is populated in the To header. The statement below will copy the To URI User into the Request URI header so the call can be properly processed by the Avaya network.

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

-SigMa rules to return From header to original form. The From header changed in outbound request messages need to be changed back for inbound responses. This is done by saving the original From User to variable “%fromuser” created previously in the script and applying the variable to the From header for inbound responses (i.e., 180 Ringing and 200 OK).

```
// Return FROM header to original form
within session "ALL"
{
  act on response where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING"
  {
    %HEADERS["From"][1].URI.USER = %fromuser;
  }
}
```

The following screen shows the finished Signaling Manipulation Script **Broadcore Script** used during compliance testing. This script will later be applied to the Broadcore/Masergy Server Configuration in **Section 7.6.2**.



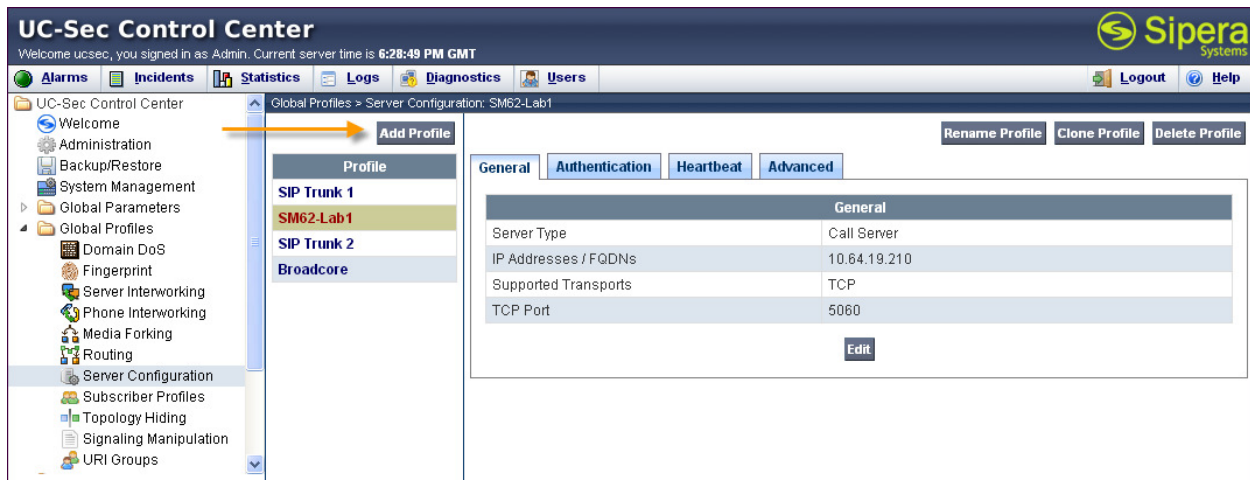
7.6. Server Configuration

The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters, DoS security statistics, and trusted domains.

In the sample configuration, separate Server Configurations were created for Session Manager and Broadcore/Masergy.

7.6.1. Server Configuration – Session Manager

To add a Server Configuration Profile for Session Manager, navigate to **UC-Sec Control Center** → **Global Profiles** → **Server Configuration** and click on **Add Profile**.



Enter a descriptive name for the new profile and click **Next**.

Add Server Configuration Profile

Profile Name

SM62-Lab1

Next

The following screens illustrate the Server Configuration for the Profile name **SM62-Lab1**. On the **General** tab, select **Call Server** from the **Server Type** drop-down menu. In the **IP Addresses / Supported FQDNs** area, the IP Address of the Session Manager SIP signaling interface in the sample configuration is entered. This IP Address is 10.64.19.210. In the **Supported Transports** area, **TCP** is selected, and the **TCP Port** is set to **5060**. This configuration corresponds with the Session Manager entity link configuration for the entity link to the Avaya SBCE created in **Section 6.6**. If adding a new profile, click **Next**. If editing an existing profile, click **Finish** (not shown).

Add Server Configuration Profile - General

Server Type

IP Addresses / Supported FQDNs
Comma seperated list

Supported Transports

TCP Port

UDP Port

TLS Port

Call Server

10.64.19.210

☒ TCP

☐ UDP

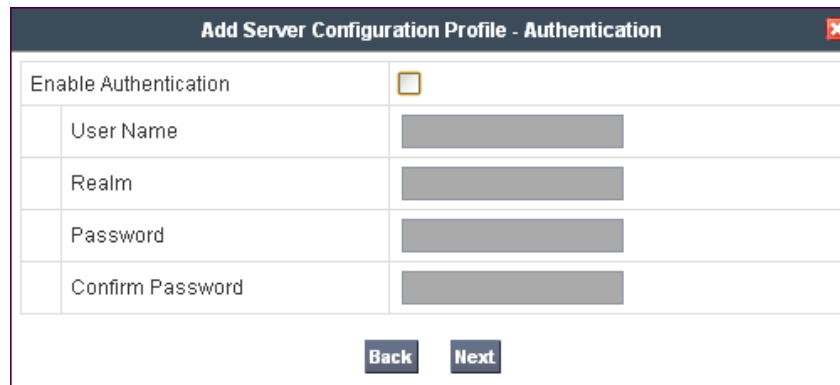
☐ TLS

5060

Back

Next

Verify **Enable Authentication** is unchecked as Session Manager does not require authentication. Click **Next** to continue.

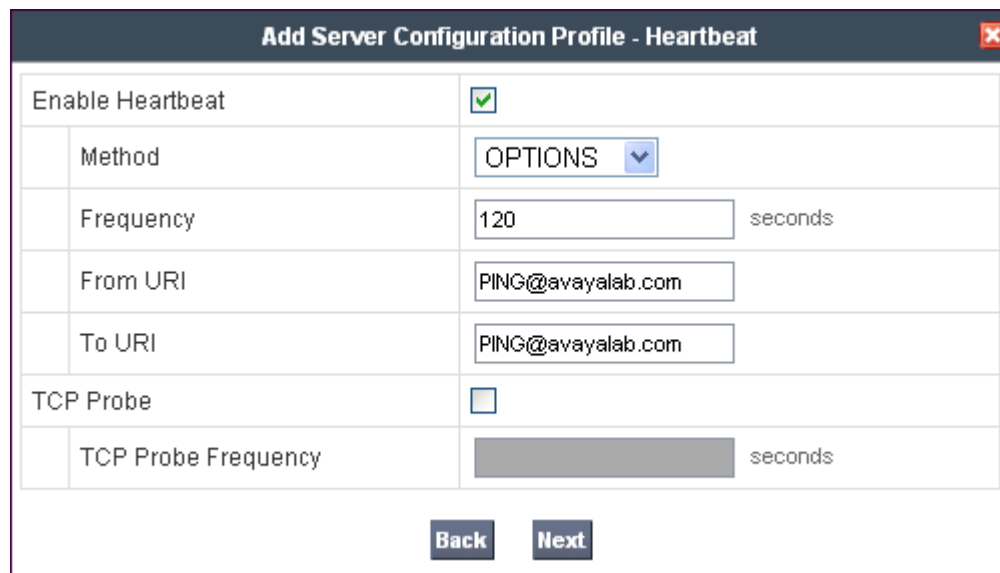


The screenshot shows a dialog box titled "Add Server Configuration Profile - Authentication". It contains a form with the following fields:

Enable Authentication	<input type="checkbox"/>
User Name	<input type="text"/>
Realm	<input type="text"/>
Password	<input type="password"/>
Confirm Password	<input type="password"/>

At the bottom of the dialog are two buttons: "Back" and "Next".

Avaya SBCE can be configured to source “heartbeats” in the form of SIP OPTIONS. In the sample configuration, with one Session Manager, this configuration is optional. If Avaya SBCE-sourced OPTIONS messages are desired, check the **Enable Heartbeat** box. Select **OPTIONS** from the **Method** drop-down menu. Select the desired frequency that the SBC will source OPTIONS. The **From URI** and **To URI** may be filled in to configure easily identifiable URIs to appear in SIP OPTIONS sourced by the SBC. Click **Next** to continue.

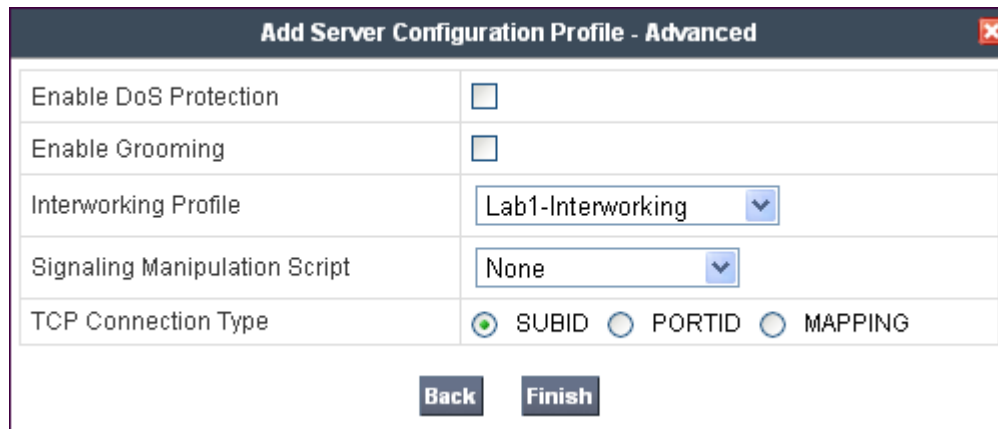


The screenshot shows a dialog box titled "Add Server Configuration Profile - Heartbeat". It contains a form with the following fields:

Enable Heartbeat	<input checked="" type="checkbox"/>
Method	OPTIONS <input type="button" value="v"/>
Frequency	120 seconds
From URI	PING@avayalab.com
To URI	PING@avayalab.com
TCP Probe	<input type="checkbox"/>
TCP Probe Frequency	<input type="text"/> seconds

At the bottom of the dialog are two buttons: "Back" and "Next".

In the new window that appears, select the **Interworking Profile** created for the enterprise in **Section 7.4.1**. Use default values for all remaining fields. Click **Finish** to save the configuration.



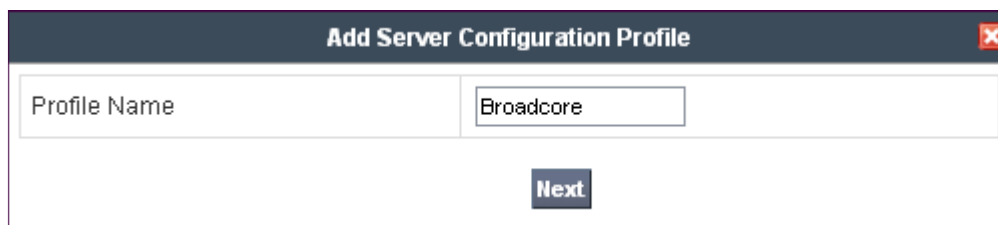
The screenshot shows a dialog box titled "Add Server Configuration Profile - Advanced". It contains five rows of configuration options:

Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	Lab1-Interworking
Signaling Manipulation Script	None
TCP Connection Type	<input checked="" type="radio"/> SUBID <input type="radio"/> PORTID <input type="radio"/> MAPPING

At the bottom of the dialog are two buttons: "Back" and "Finish".

7.6.2. Server Configuration - Broadcore/Masergy

To add a Server Configuration Profile for Broadcore/Masergy, navigate to **UC-Sec Control Center → Global Profiles → Server Configuration** and click on **Add Profile** (not shown). Enter a descriptive name for the new profile and click **Next**.



The screenshot shows a dialog box titled "Add Server Configuration Profile". It contains a single row with a text input field for the "Profile Name" containing the text "Broadcore". Below the input field is a "Next" button.

The following screens illustrate the Server Configuration for the Profile name **Broadcore**. In the **General** parameters, select **Trunk Server** from the **Server Type** drop-down menu. In the **IP Addresses / Supported FQDNs** area, the Broadcore/Masergy provided Fully Qualified Domain Name is entered. This is **west.broadcore.com**. In the **Supported Transports** area, UDP is selected, and the **UDP Port** is set to **5060**. If adding a new profile, click Next. If editing an existing profile, click Finish (not shown).

The screenshot shows a dialog box titled "Add Server Configuration Profile - General". It contains the following fields and options:

Server Type	Trunk Server (dropdown)
IP Addresses / Supported FQDNs <small>Comma seperated list</small>	west.broadcore.com
Supported Transports	<input type="checkbox"/> TCP <input checked="" type="checkbox"/> UDP <input type="checkbox"/> TLS
TCP Port	[disabled text box]
UDP Port	5060
TLS Port	[disabled text box]

At the bottom are "Back" and "Next" buttons.

Select **Enable Authentication**. Enter the user name provided by Broadcore/Masergy in the **User Name** field. Leave the **Realm** blank to have it detected from the server challenge. Enter the password provided by Broadcore/Masergy in the **Password** field. Click **Next** to continue.

The screenshot shows a dialog box titled "Add Server Configuration Profile - Authentication". It contains the following fields and options:

Enable Authentication	<input checked="" type="checkbox"/>
User Name	user1234
Realm <small>(Leave blank to detect from server challenge)</small>	[empty text box]
Password	****
Confirm Password	****

At the bottom are "Back" and "Next" buttons.

In the new window that appears, check the **Enable Heartbeat** box. Select **REGISTER** from the **Method** drop-down menu. Select the desired frequency that the SBC will source REGISTERS. The **From URI** and **To URI** are filled in with <number>@west.broadcore.com, where <number> is the pilot number provided by Broadcore/Masergy. Click **Next** to continue.

Add Server Configuration Profile - Heartbeat	
Enable Heartbeat	<input checked="" type="checkbox"/>
Method	REGISTER ▼
Frequency	120 seconds
From URI	4245556553@west.broac
To URI	4245556553@west.broac
TCP Probe	<input type="checkbox"/>
TCP Probe Frequency	seconds
<div>Back Next</div>	

In the new window that appears, select the **Interworking Profile** created for Broadcore/Masergy in **Section 7.4.2**. Select the **Signaling Manipulation Script** created in **Section 7.5**. Use default values for all remaining fields. Click **Finish** to save the configuration.

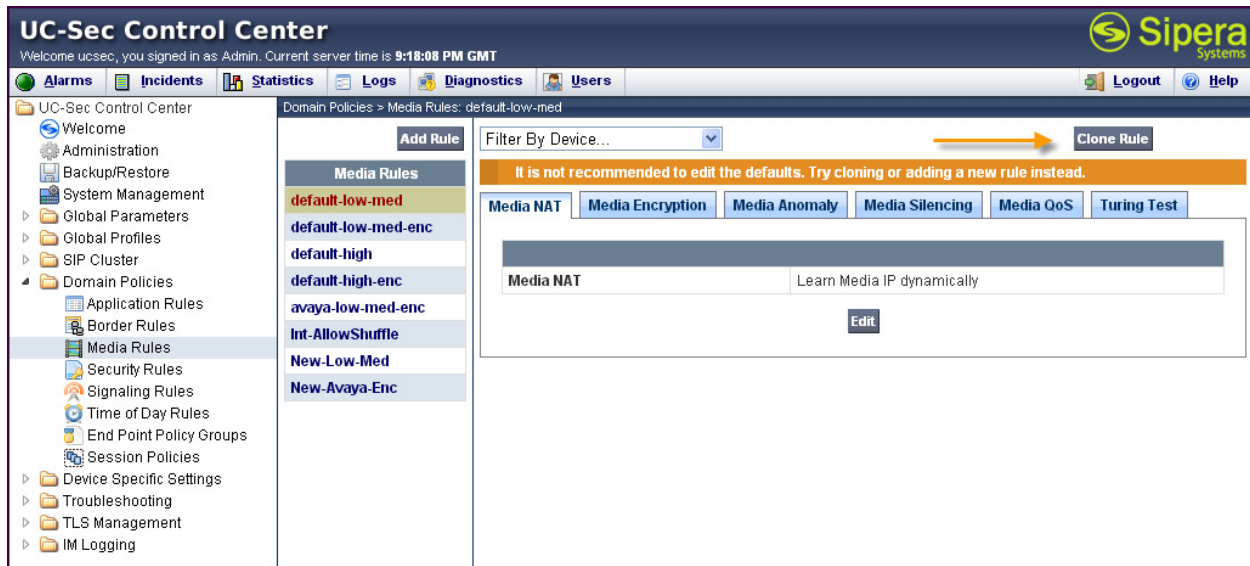
Add Server Configuration Profile - Advanced	
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	Broadcore Intrwrking ▼
Signaling Manipulation Script	Broadcore Script ▼
UDP Connection Type	<input checked="" type="radio"/> SUBID <input type="radio"/> PORTID <input type="radio"/> MAPPING
<div>Back Finish</div>	

7.7. Media Rule

Media Rules define RTP media packet parameters such as prioritizing encryption techniques and packet encryption techniques. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies to determine how media packets matching these criteria will be handled by the Avaya SBCE security product.

Create a custom Media Rule to set the Quality of Service. The sample configuration shows a custom Media Rule **New-Low-Med** created for Broadcore/Masergy SIP Trunk Service and the enterprise.

To create a custom Media Rule, navigate to **UC-Sec Control Center → Domain Policies → Media Rules**. With **default-low-med** selected, click **Clone Rule** as shown below.



Enter a descriptive name for the new rule and click **Finish**.

Clone Rule	
Rule Name	default-low-med
Clone Name	<input type="text" value="New-Low-Med"/>
<input type="button" value="Finish"/>	

On the **Media QoS** tab select the proper Quality of Service (QoS). Avaya SBCE can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Service policies for the media. The following screen shows the QoS values used for compliance testing.

UC-Sec Control Center
Welcome ucsec, you signed in as Admin. Current server time is 3:45:44 PM GMT

Alarms Incidents Statistics Logs Diagnostics Users Logout Help

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
Session Policies
Device Specific Settings
Troubleshooting
TLS Management
IM Logging

Domain Policies > Media Rules: New-Low-Med

Add Rule Filter By Device... Rename Rule Clone Rule Delete Rule

Click here to add a description.

Media NAT Media Encryption Media Anomaly Media Silencing Media QoS Turing Test

Media QoS Reporting
RTCP Enabled ☐

Media QoS Marking
Enabled ☒
QoS Type DSCP

Audio QoS
Audio DSCP EF

Video QoS
Video DSCP EF

Edit

7.8. Signaling Rule

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

Clone and modify the default signaling rule to remove unnecessary SIP headers and add the proper quality of service to the SIP message. To clone a signaling rule, navigate to **UC-Sec Control Center → Domain Policies → Signaling Rules**. With the **default** rule chosen, click on **Clone Rule** (not shown). Enter a descriptive name for the new rule and click **Finish**.

Clone Rule

Rule Name default

Clone Name Avaya

Finish

In the sample configuration, signaling rule **Avaya** was created for Session Manager to prevent certain headers in the SIP messages sent from Session Manager from being propagated to Broadcore/Masergy. Select this rule in the center pane, then select the **Request Headers** tab to view the manipulations performed on the request messages such as the initial INVITE or UPDATE message. The following screen shows the **Alert-Info**, **Endpoint-View**, **History-Info** and **P-Location** headers removed during the compliance test.

UC-Sec Control Center
Welcome ucsec, you signed in as Admin. Current server time is 10:43:48 PM GMT

Domain Policies > Signaling Rules: Avaya

Filter By Device... [v] [Rename Rule] [Clone Rule] [Delete Rule]

Click here to add a description.

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	Alert-Info	ALL	Forbidden	Remove Header	No	IN		
2	Endpoint-View	ALL	Forbidden	Remove Header	Yes	IN		
3	History-Info	ALL	Forbidden	Remove Header	No	IN		
4	P-Location	ALL	Forbidden	Remove Header	Yes	IN		

Similarly, manipulations can be performed on the SIP response messages. These can be viewed by selecting the **Response Headers** tab as shown below.

UC-Sec Control Center
Welcome ucsec, you signed in as Admin. Current server time is 10:50:44 PM GMT

Domain Policies > Signaling Rules: Avaya

Filter By Device... [v] [Rename Rule] [Clone Rule] [Delete Rule]

Click here to add a description.

General [Requests] [Responses] [Request Headers] [Response Headers] [Signaling QoS]

Add In Header Control Add Out Header Control

Row	Header Name	Response Code	Method Name	Header Criteria	Action	Proprietary	Direction		
1	Endpoint-View	1XX	ALL	Forbidden	Remove Header	Yes	IN		
2	Endpoint-View	2XX	ALL	Forbidden	Remove Header	Yes	IN		
3	P-Location	1XX	ALL	Forbidden	Remove Header	Yes	IN		
4	P-Location	2XX	ALL	Forbidden	Remove Header	Yes	IN		

On the **Signaling QoS** tab select the proper Quality of Service (QoS). The Avaya SBCE can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Service policies for signaling. The following screen shows the QoS values used for compliance testing.

The screenshot shows the UC-Sec Control Center interface. The left sidebar contains a tree view with categories like Alarms, Incidents, Statistics, Logs, Diagnostics, and Users. Under 'Domain Policies', 'Signaling Rules' is expanded, showing a list of rules including 'default', 'No-Content-Type-Checks', 'Avaya', and 'SIPTrunk Sig Rule'. The main panel displays the configuration for the 'Avaya' rule. The 'Signaling QoS' tab is active, showing a table with the following data:

QoS Type	DSCP
DSCP	EF

An 'Edit' button is located below the table.

A separate signaling rule **SIPTrunk Sig Rule** was created for Broadcore/Masergy SIP Trunk Service by cloning the **default** signaling rule and changing the **Signaling QoS** parameters as shown below.

The screenshot shows the UC-Sec Control Center interface. The left sidebar is the same as the previous screenshot. Under 'Domain Policies', 'Signaling Rules' is expanded, and 'SIPTrunk Sig Rule' is selected. The main panel displays the configuration for the 'SIPTrunk Sig Rule'. The 'Signaling QoS' tab is active, showing a table with the following data:

QoS Type	DSCP
DSCP	EF

An 'Edit' button is located below the table.

7.9. Application Rule

Application Rules define which types of SIP-based Unified Communications (UC) applications the Avaya SBCE security device will protect: voice, video, and/or Instant Messaging (IM). In addition, you can determine the maximum number of concurrent voice and video sessions the network will process in order to prevent resource exhaustion.

Create an Application Rule to increase the number of concurrent voice traffic. The sample configuration cloned and modified the default application rule to increase the number of **Maximum Concurrent Session** and **Maximum Sessions Per Endpoint**. To clone an application rule, navigate to **UC-Sec Control Center → Domain Policies → Application Rules**. With the **default** rule chosen, click on **Clone Rule** (not shown). Enter a descriptive name for the new rule and click **Finish**.

Clone Rule ✕

Rule Name	default
Clone Name	<input style="width: 90%;" type="text" value="MaxVoiceSession"/>

Finish

Modify the rule by clicking the **Edit** button. The following screen shows the modified Application Rule **MaxVoiceSession** created in the sample configuration. Set the **Maximum Concurrent Sessions** and **Maximum Session Per Endpoint** to **2000**.

UC-Sec Control Center
Welcome ucsec, you signed in as Admin. Current server time is 7:55:24 PM GMT

Alarms Incidents Statistics Logs Diagnostics Users Logout Help

- 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
 - Session Policies
- Device Specific Settings
- Troubleshooting
- TLS Management
- IM Logging

Domain Policies > Application Rules: MaxVoiceSession

Add Rule

Filter By Device...

Rename Rule

Clone Rule

Delete Rule

Click here to add a description.

Application Rule

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

Miscellaneous

CDR Support	None		
IM Logging	No		
RTCP Keep-Alive	No		

Edit

DDT; Reviewed:
SPOC 8/9/2013

Solution & Interoperability Test Lab Application Notes
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7.10. Endpoint Policy Group

The rules created within the Domain Policy section are assigned to an Endpoint Policy Group. The Endpoint Policy Group is then applied to a Server Flow in **Section 7.13**. Create a separate Endpoint Policy Group for the enterprise and the Broadcore/Masergy SIP Trunk Service.

To create a new policy group, navigate to **UC-Sec Control Center → Domain Policies → Endpoint Policy Groups** and click on **Add Group** as shown below.

The screenshot shows the UC-Sec Control Center interface. The left sidebar contains a tree view with 'Domain Policies' expanded, and 'End Point Policy Groups' selected. The main area displays the 'Add Group' dialog for 'default-low'. The dialog includes a 'Filter By Device...' dropdown, a warning message, and a table for defining the policy group.

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

The following screen shows **Lab1_DomPolicy** created for the enterprise. Set the **Application**, **Media**, and **Signaling** rules to the ones previously created for the enterprise. Set the **Border**, **Security** and **Time of Day** rules to either the **default** or **default-low** policies.

The screenshot shows the UC-Sec Control Center interface with 'Lab1_DomPolicy' selected in the 'End Point Policy Groups' list. The main area displays the configuration for this policy group, including a table for defining the policy group.

Order	Application	Border	Media	Security	Signaling	Time of Day	
1	MaxVoiceSession	default	New-Low-Med	default-low	Avaya	default	

The following screen shows **SIP Trunk_DomPolicy** created for Broadcore/Masergy. Set the **Application**, **Media**, and **Signaling** rules to the one previously created for Broadcore/Masergy. Set the **Border**, **Security**, and **Time of Day** rules to either the **default** or **default-high** policies.

The screenshot shows the UC-Sec Control Center interface. The left sidebar contains a tree view with categories like Administration, Global Parameters, SIP Cluster, Domain Policies, and End Point Policy Groups. The main area displays the configuration for 'SIP Trunk_DomPolicy'. It includes a 'Policy Groups' list on the left with options like 'default-low', 'default-low-enc', 'default-med', 'default-med-enc', 'default-high', 'default-high-enc', 'OCS-default-high', 'avaya-def-low-enc', 'SIP Trunk_DomPolicy' (highlighted), 'Enterprise_enc', and 'Lab1_DomPolicy'. The main configuration area has a table with columns: Order, Application, Border, Media, Security, Signaling, Time of Day, and an action column. The table contains one row with the following values: Order 1, Application MaxVoiceSession, Border default, Media New-Low-Med, Security default-high, Signaling SIPTrunk Sig Rule, Time of Day default, and an edit/delete icon.

Order	Application	Border	Media	Security	Signaling	Time of Day	
1	MaxVoiceSession	default	New-Low-Med	default-high	SIPTrunk Sig Rule	default	

7.11. Media Interface

The Media Interface screen is where the SIP media ports are defined. Avaya SBCE will send SIP media on the defined ports. Create a SIP Media Interface for both the inside and outside IP interfaces.

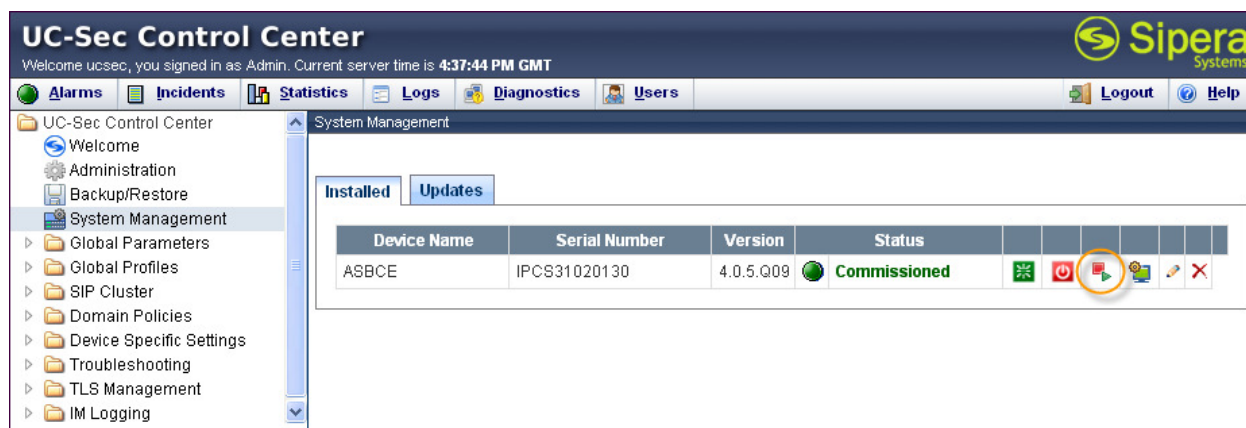
To create a new Media Interface, navigate to **UC-Sec Control Center → Device Specific Settings → Media Interface** and click **Add Media Interface**.

The following screen shows the media interfaces created in the sample configuration for the inside and outside IP interfaces.

The screenshot shows the UC-Sec Control Center interface for 'Device Specific Settings > Media Interface: ASBCE'. The left sidebar shows the navigation tree with 'Media Interface' selected under 'Device Specific Settings'. The main area displays the 'Media Interface' configuration. It includes a warning message: 'Modifying or deleting an existing media interface will require an application restart before taking effect. Application restarts can be issued from System Management.' Below this is a table with columns: Name, Media IP, Port Range, and an action column. The table contains two rows: 'Media_Inside' with Media IP 10.64.19.100 and Port Range 2048 - 5059, and 'Media_Outside_92' with Media IP 192.168.62.92 and Port Range 8000 - 8999. Both rows have edit and delete icons in the action column.

Name	Media IP	Port Range	
Media_Inside	10.64.19.100	2048 - 5059	
Media_Outside_92	192.168.62.92	8000 - 8999	

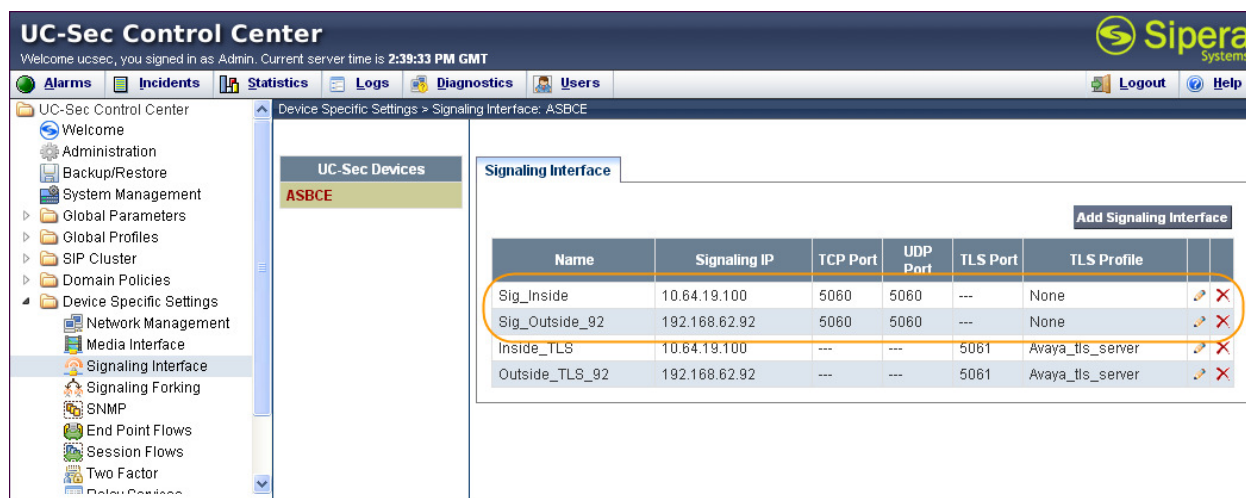
After the media interfaces are created, an application restart is necessary before the changes will take effect. Navigate to **UC-Sec Control Center** → **System Management** and click the forth icon from the right to restart the applications as highlighted below.



7.12. Signaling Interface

The Signaling Interface screen is where the SIP signaling ports are defined. Avaya SBCE will listen for SIP requests on the defined ports. Create a Signaling Interface for both the inside and outside IP interfaces. To create a new Signaling Interface, navigate to **UC-Sec Control Center** → **Device Specific Settings** → **Signaling Interface** and click **Add Signaling Interface**.

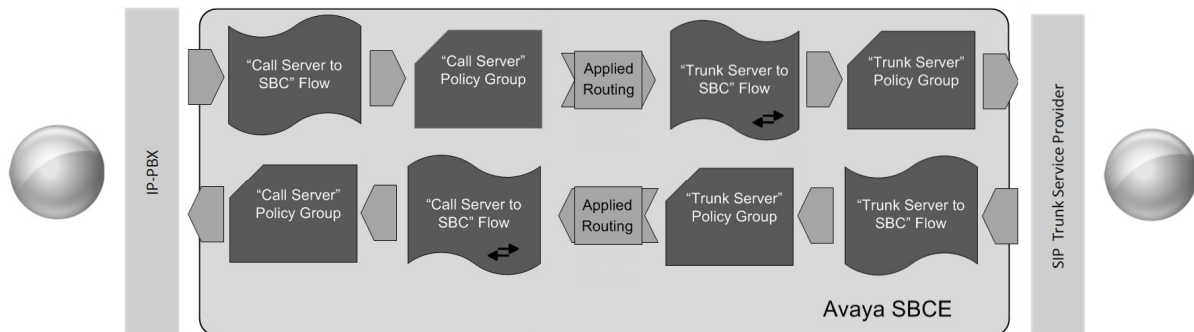
In the shared test environment the following screen shows the signaling interfaces created in the sample configuration for the inside and outside IP interfaces.



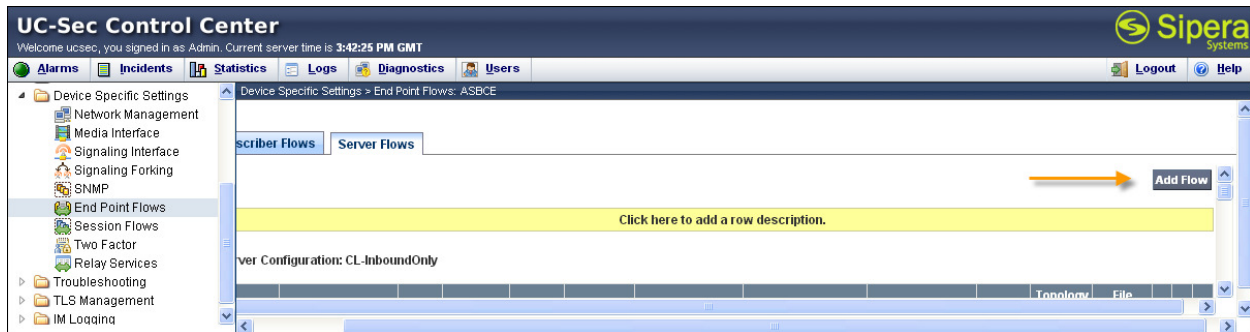
7.13. End Point Flows - Server Flow

When a packet is received by Avaya SBCE, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to a policy which contains several rules concerning processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for this

destination endpoint are applied. The context is maintained, so as to be applied to future packets in the same flow. The following screen illustrates the flow through the Avaya SBCE to secure a SIP Trunk call.



Create a Server Flow for Session Manager and Broadcore/Masergy SIP Trunk Service. To create a Server Flow, navigate to **UC-Sec Control Center** → **Device Specific Settings** → **End Point Flows**. Select the **Server Flows** tab and click **Add Flow** as shown in below.



The following screen show the flow named **Broadcore Flow** created in the sample configuration. This flow uses the interfaces, policies, and profiles defined in previous sections. Click **Finish**.

Edit Flow: Broadcore Flow

Criteria	
Flow Name	Broadcore Flow
Server Configuration	Broadcore
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Sig_Inside
Signaling Interface	Sig_Outside_92
Media Interface	Media_Outside_92
End Point Policy Group	SIP Trunk_DomPolicy
Routing Profile	To-SM
Topology Hiding Profile	Broadcore Topology
File Transfer Profile	None

Finish

Once again, select the **Server Flows** tab and click **Add Flow**. The following screen shows the flow named **SM62-Lab1-Flow** created in the sample configuration. This flow uses the interfaces, policies, and profiles defined in previous sections. Click **Finish**.

Edit Flow: SM62-Lab1-Flow
✖

Criteria	
Flow Name	<input type="text" value="SM62-Lab1-Flow"/>
Server Configuration	<input type="text" value="SM62-Lab1"/> ▼
URI Group	<input type="text" value="*"/> ▼
Transport	<input type="text" value="*"/> ▼
Remote Subnet	<input type="text" value="*"/>
Received Interface	<input type="text" value="Sig_Outside_92"/> ▼
Signaling Interface	<input type="text" value="Sig_Inside"/> ▼
Media Interface	<input type="text" value="Media_Inside"/> ▼
End Point Policy Group	<input type="text" value="Lab1_DomPolicy"/> ▼
Routing Profile	<input type="text" value="To-Broadcore"/> ▼
Topology Hiding Profile	<input type="text" value="Enterprise"/> ▼
File Transfer Profile	<input type="text" value="None"/> ▼

Finish

8. Broadcore/Masergy SIP Trunk Configuration

To use Broadcore/Masergy SIP Trunk Service, a customer must request the service from Broadcore/Masergy using their sales processes. This process can be initiated by contacting Broadcore/Masergy via the corporate web site at www.broadcore.com and requesting information via the online sales links or telephone numbers.

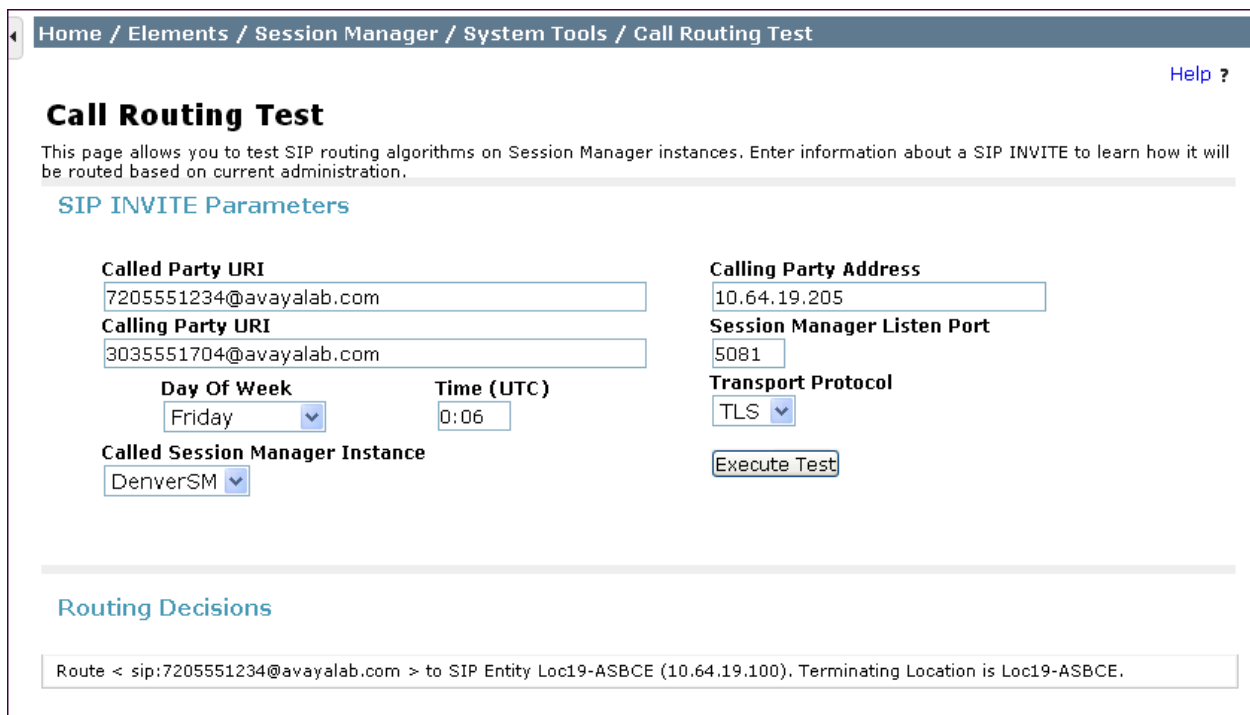
9. Verification and Troubleshooting

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

9.1. Verification

The following steps may be used to verify the configuration:

1. Verify the call routing administration on Session Manager by logging in to System Manager and executing the Call Routing Test. Expand **Elements** → **Session Manager** → **System Tools** → **Call Routing Test**. Populate the field for the call parameters of interest. For example, the following screen shows an example call routing test for an outbound call to PSTN via Broadcore/Masergy. Under **Routing Decisions**, observe the call will route via the Avaya SBCE SIP Entity to Broadcore/Masergy. Scroll down to inspect the details of the **Routing Decision Process** if desired (not shown).



Home / Elements / Session Manager / System Tools / Call Routing Test [Help ?](#)

Call Routing Test

This page allows you to test SIP routing algorithms on Session Manager instances. Enter information about a SIP INVITE to learn how it will be routed based on current administration.

SIP INVITE Parameters

Called Party URI	7205551234@avayalab.com	Calling Party Address	10.64.19.205
Calling Party URI	3035551704@avayalab.com	Session Manager Listen Port	5081
Day Of Week	Friday	Time (UTC)	0:06
Called Session Manager Instance	DenverSM	Transport Protocol	TLS
		Execute Test	

Routing Decisions

Route < sip:7205551234@avayalab.com > to SIP Entity Loc19-ASBCE (10.64.19.100). Terminating Location is Loc19-ASBCE.

2. 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.
3. 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.
4. Verify that the user on the PSTN can end an active call by hanging up.
5. Verify that an endpoint at the enterprise site can end an active call by hanging up.

Use the SAT interface on Communication Manager to verify status of SIP trunks. Specifically use the **status trunk n** command to verify the active call has ended, where **n** is the trunk group number used for Broadcore/Masergy SIP Trunk Service defined in **Section 5.8**.

Below is an example of an active call.

```
status trunk 1
```

TRUNK GROUP STATUS				
Member	Port	Service State	Mtce Connected Ports Busy	
0001/001	T00001	in-service/active	no	S00000
0001/002	T00002	in-service/idle	no	
0001/003	T00003	in-service/idle	no	
0001/004	T00004	in-service/idle	no	

Verify the port returns to **in-service/idle** after the call has ended.

```
status trunk 1
```

TRUNK GROUP STATUS				
Member	Port	Service State	Mtce Connected Ports Busy	
0001/001	T00001	in-service/idle	no	
0001/002	T00002	in-service/idle	no	
0001/003	T00003	in-service/idle	no	
0001/004	T00004	in-service/idle	no	

9.2. 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.
2. Session Manager: **traceSM -x -uni** – Session Manager command line tool for traffic analysis. Login to the Session Manager management interface to run this command.

3. Avaya SBCE:

- **Incidences** – Displays alerts captured by the UC-Sec appliance.

Incident Viewer

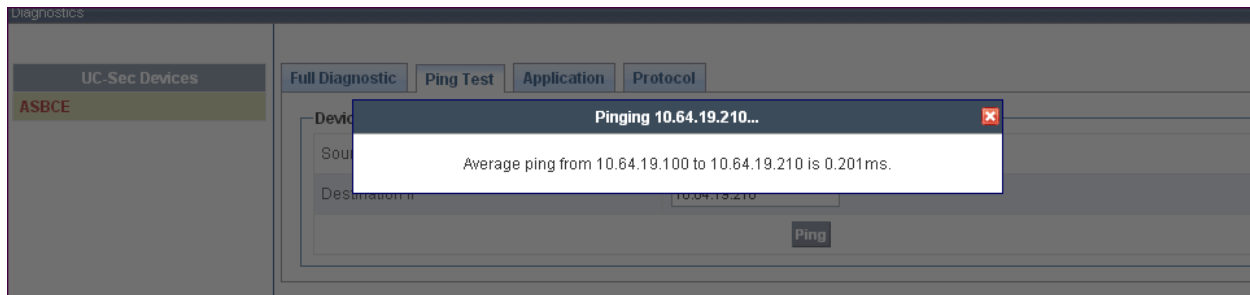
Device Category

Displaying results 1 to 15 out of 829.

Incident Type	Incident ID	Date	Time	Category	Device	Cause
Server Heartbeat	677626218369560	12/11/12	7:00 PM	Policy	ASBCE	Heartbeat Failed, Server is Down
Server Heartbeat	677625332521293	12/11/12	6:31 PM	Policy	ASBCE	Heartbeat Failed, Server is Down
Server Heartbeat	677625276170295	12/11/12	6:29 PM	Policy	ASBCE	Heartbeat Failed, Server is Down
Server Heartbeat	677623965751299	12/11/12	5:45 PM	Policy	ASBCE	Heartbeat Failed, Server is Down
Server Heartbeat	677623890418242	12/11/12	5:43 PM	Policy	ASBCE	Heartbeat Failed, Server is Down
Server Heartbeat	677585247087809	12/10/12	8:14 PM	Policy	ASBCE	Heartbeat Failed, Server is Down
Server Heartbeat	677583086633859	12/10/12	7:02 PM	Policy	ASBCE	Heartbeat Failed, Server is Down
Routing Failure	677574431620086	12/10/12	2:14 PM	Policy	ASBCE	Server Config Found. But no server flow matched, Sending 403 Forbidden
Message Dropped	677574431620044	12/10/12	2:14 PM	Policy	ASBCE	No Server Flow Matched for Outgoing Message
Routing Failure	677574416592850	12/10/12	2:13 PM	Policy	ASBCE	Server Config Found. But no server flow matched, Sending 403 Forbidden
Message Dropped	677574416592807	12/10/12	2:13 PM	Policy	ASBCE	No Server Flow Matched for Outgoing Message
Routing Failure	677574401570113	12/10/12	2:13 PM	Policy	ASBCE	Server Config Found. But no server flow matched, Sending 403 Forbidden
Message Dropped	677574401570070	12/10/12	2:13 PM	Policy	ASBCE	No Server Flow Matched for Outgoing Message
Routing Failure	677574386540355	12/10/12	2:12 PM	Policy	ASBCE	Server Config Found. But no server flow matched, Sending 403 Forbidden
Message Dropped	677574386540292	12/10/12	2:12 PM	Policy	ASBCE	No Server Flow Matched for Outgoing Message

<< < 1 2 3 4 5 > >>

- **Diagnostics** – Allows for PING tests and displays application and protocol use.



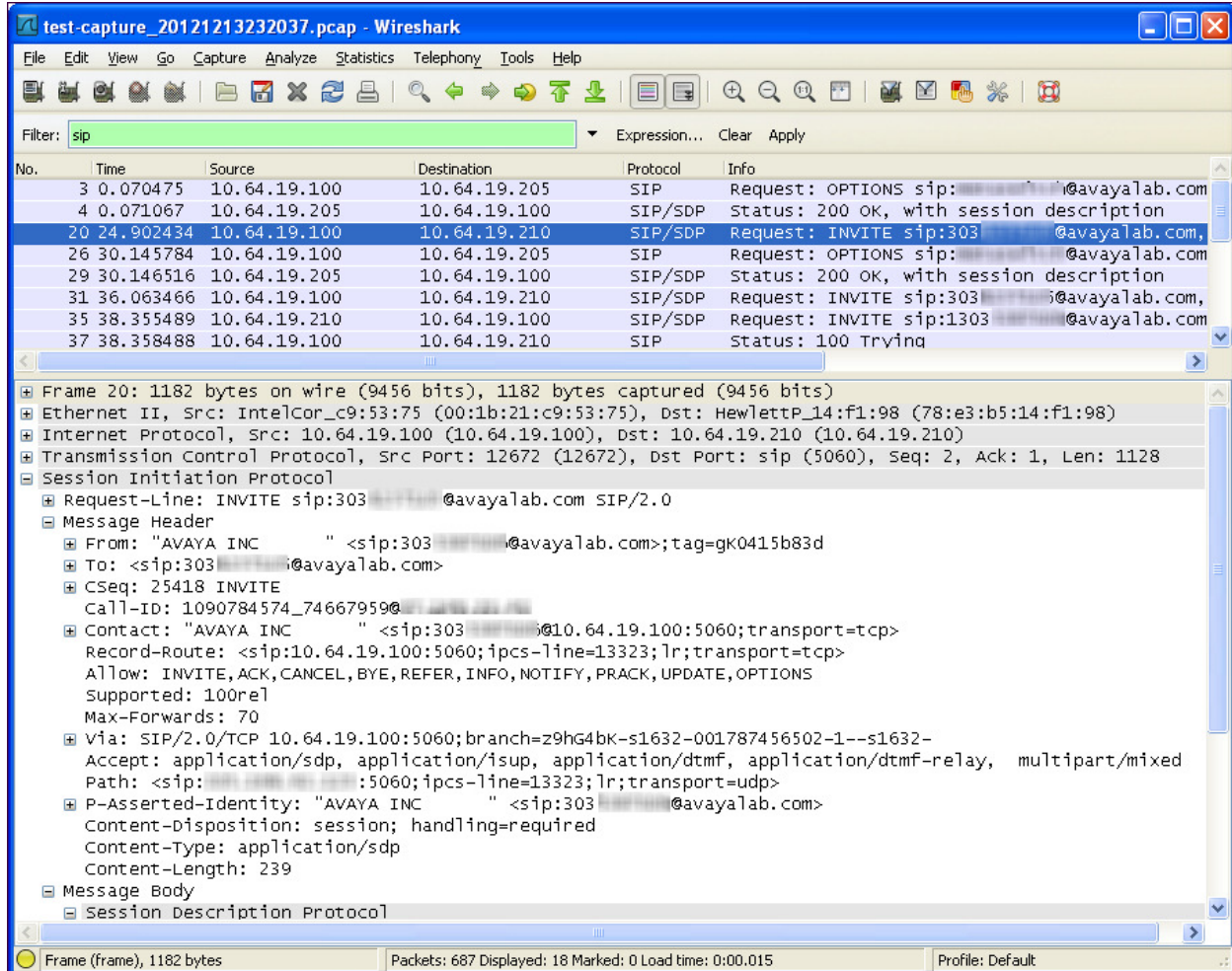
- **Troubleshooting → Trace Settings** – Configure and display call traces and packet captures for the UC-Sec appliance.

The screenshot shows the UC-Sec Control Center interface. The left sidebar contains a navigation menu with options like Welcome, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, SIP Cluster, Domain Policies, Device Specific Settings, Troubleshooting, Advanced Options, DoS Learning, Syslog Management, Trace Settings, TLS Management, and IM Logging. The main content area is titled 'Troubleshooting > Trace Settings: ASBCE'. It features a 'UC-Sec Devices' list with 'ASBCE' selected. Below this, there are tabs for 'Packet Trace', 'Call Trace', 'Packet Capture', and 'Captures'. The 'Packet Capture' tab is active, displaying the 'Packet Capture Configuration' form. The form includes fields for 'Currently capturing' (No), 'Interface' (A1), 'Local Address (ip:port)' (10.64.19.100), 'Remote Address (*, *:port, ip, ip:port)' (*), 'Protocol' (All), 'Maximum Number of Packets to Capture' (1200), and 'Capture Filename' (test-capture.pcap). There are 'Start Capture' and 'Clear' buttons at the bottom.

The screenshot shows the UC-Sec Control Center interface, specifically the 'Captures' tab. The left sidebar is the same as in the previous screenshot. The main content area is titled 'Troubleshooting > Trace Settings: ASBCE'. It features a 'UC-Sec Devices' list with 'ASBCE' selected. Below this, there are tabs for 'Packet Trace', 'Call Trace', 'Packet Capture', and 'Captures'. The 'Captures' tab is active, displaying a table of captured files. The table has columns for 'File Name', 'File Size (bytes)', and 'Last Modified'. A 'Refresh' button is located at the top right of the table. The table contains one entry: 'test-capture_20121213232037.pcap' with a file size of 167,936 bytes and a last modified time of December 13, 2012 11:21:26 PM GMT. There is a red 'X' icon in the last column of the table.

File Name	File Size (bytes)	Last Modified
test-capture_20121213232037.pcap	167,936	December 13, 2012 11:21:26 PM GMT

The packet capture file can be downloaded and viewed using a Network Protocol Analyzer like Wireshark:



10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Session Manager, Avaya Aura® Communication Manager, and Avaya Session Border Controller for Enterprise to the Broadcore/Masergy SIP Trunk Service. The Broadcore/Masergy SIP Trunk Service is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. The Broadcore/Masergy SIP Trunk Service provides businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks.

Broadcore/Masergy SIP Trunk Service passed compliance testing.

11. Additional 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 6.2.0*, March 2012.
- [2] *Administering Avaya Aura® System Platform, Release 6.2.0*, February 2012.
- [3] *Administering Avaya Aura® Communication Manager*, June 2010, Document Number 03-300509.
- [4] *Avaya Aura® Communication Manager Feature Description and Implementation*, June 2010, Document Number 555-245-205.
- [5] *Implementing Avaya Aura® System Manager*, Release 6.2, March 2012
- [6] *Installing Service Packs for Avaya Aura® Session Manager*, February 2012, Document Number 03-603863
- [7] *Implementing Avaya Aura® Session Manager*, February 2012, Document Number 03-603473.
- [8] *Avaya one-X Deskphone H.323 Administrator Guide*, May 2011, Document Number 16-300698.
- [9] *Administering Avaya one-X Communicator*, July 2011
- [10] *Administering Avaya Session Border Controller*, Document Number 08-604063, Sept. 2012
- [11] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>
- [12] *RFC 3515, The Session Initiation Protocol (SIP) Refer Method*, <http://www.ietf.org/>
- [13] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, <http://www.ietf.org/>
- [14] *RFC 4244, An Extension to the Session Initiation Protocol (SIP) for Request History Information*, <http://www.ietf.org/>

Appendix A: Static IP Authentication

Static IP Authentication is a Broadcore/Masergy offered service that is an alternative to Single Number Registration. This feature allows Customers to register a SIP trunk by using the IP address of the Avaya SBCE outside interface rather than sending REGISTER messages using a username and password. The Avaya SBCE will also route calls based on a static IP address rather than using DNS to discover the IP address.

The procedures outlined in these Application Notes are used to support static IP authentication with the exception of the changes outlined in this section for the Avaya SBCE.

Login to Avaya SBCE as shown in **Section 7** above, navigate to **UC-Sec Control Center** → **Global Profiles** → **Routing** and select **Add Profile**. Enter a **Profile Name** and click **Next** to continue (not shown).

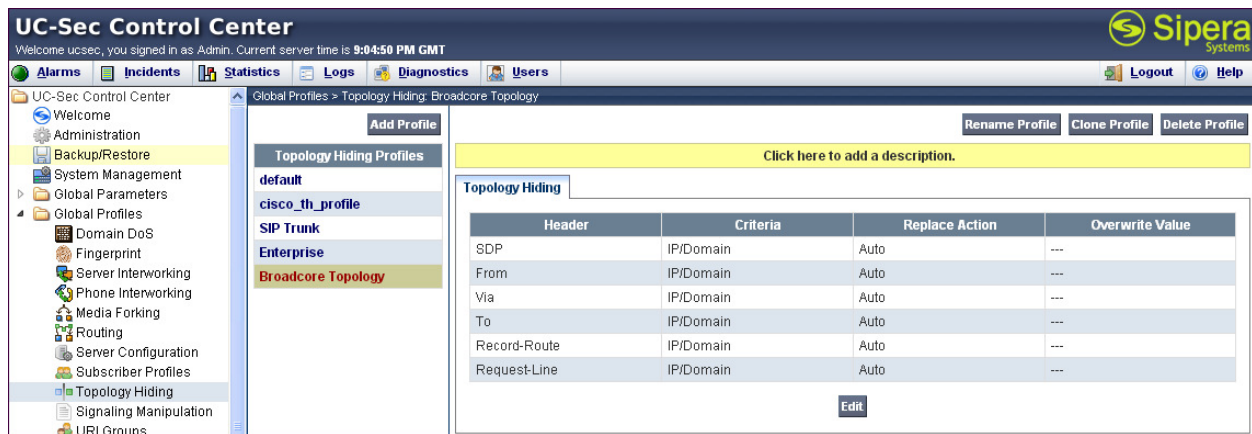
The following screen shows the Routing Profile **To-Broadcore** created for Static IP Authentication. In the **Next Hop Server 1** field enter the IP Address that Broadcore/Masergy uses to listen for SIP traffic. In the sample configuration **192.168.11.69** was used. Select **Next Hop Priority** and enter **UDP** for the **Outgoing Transport** field.

The screenshot shows the UC-Sec Control Center interface. The left sidebar contains a tree view with categories like Welcome, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, Domain DoS, Fingerprint, Server Interworking, Phone Interworking, Media Forking, Routing, Server Configuration, Subscriber Profiles, and Topology Hiding. The 'Routing' category is selected. The main area displays the 'Global Profiles > Routing: To-Broadcore' configuration. At the top, there are buttons for 'Add Profile', 'Rename Profile', 'Clone Profile', and 'Delete Profile'. Below these is a yellow bar with the text 'Click here to add a description.' The 'Routing Profile' section contains a table with columns: Priority, URI Group, Next Hop Server 1, Next Hop Server 2, Next Hop Priority, NAPTR, SRV, Next Hop in Dialog, Ignore Route Header, and Outgoing Transport. A single row is visible with the following values: Priority 1, URI Group *, Next Hop Server 1 192.168.11.69, Next Hop Server 2 ---, Next Hop Priority checked, NAPTR unchecked, SRV unchecked, Next Hop in Dialog unchecked, Ignore Route Header unchecked, and Outgoing Transport UDP.

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	*	192.168.11.69	---	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	UDP

Navigate to **UC-Sec Control Center** → **Global Profiles** → **Topology Hiding**. Select the **default** profile and click on **Clone Profile** (not shown).

The following screen shows the Topology Hiding Profile **Broadcore Topology** created for Static IP Authentication.

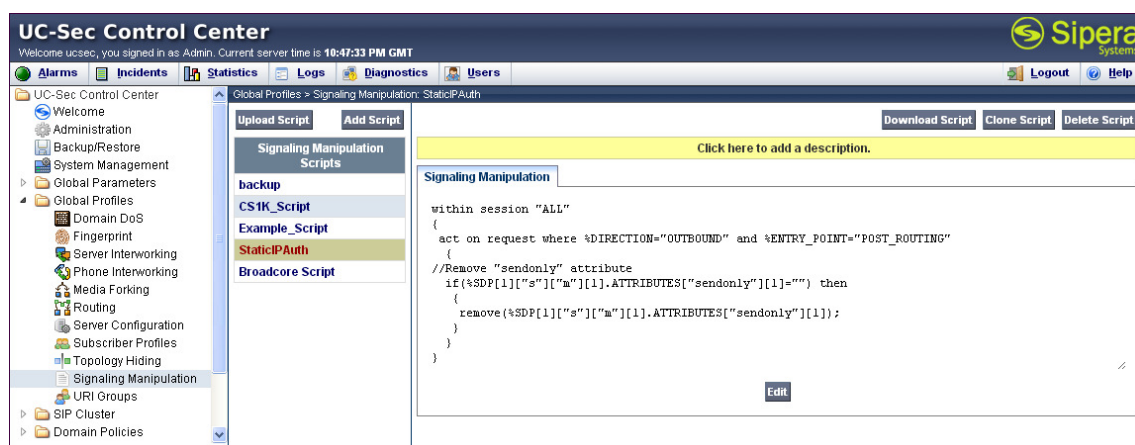


Navigate to **UC-Sec Control Center** → **Global Profiles** → **Signaling Manipulation** and click on **Add Script**. A new blank SigMa Editor window will pop up.

In this compliance testing, the script named **StaticIPAuth** was created as shown below. See **Section 7.5** for more information regarding signaling manipulation.

```
within session "ALL"
{
  act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
  {
    //Remove "sendonly" attribute
    if(%SDP[1]["s"]["m"][1].ATTRIBUTES["sendonly"][1]="") then
    {
      remove(%SDP[1]["s"]["m"][1].ATTRIBUTES["sendonly"][1]);
    }
  }
}
```

The following screen shows the finished Signaling Manipulation **StaticIPAuth** used in the sample configuration.



Navigate to **UC-Sec Control Center** → **Global Profiles** → **Server Configuration** and click on **Add Profile** (not shown).

The following screens illustrate the Server Configuration for the Profile name **Broadcore** used for Static IP Authentication. In the **General** parameters, select **Trunk Server** from the **Server Type** drop-down menu. In the **IP Addresses / Supported FQDNs** area, the Broadcore/Masergy provided IP address is entered. In the sample configuration **192.168.11.69** was used. In the **Supported Transports** area, UDP is selected, and the **UDP Port** is set to **5060**.

The screenshot shows a dialog box titled "Edit Server Configuration Profile - General". It contains the following fields and options:

Server Type	Trunk Server (dropdown)
IP Addresses / Supported FQDNs Comma separated list	192.168.11.69
Supported Transports	<input type="checkbox"/> TCP <input checked="" type="checkbox"/> UDP <input type="checkbox"/> TLS
TCP Port	
UDP Port	5060
TLS Port	

At the bottom of the dialog is a "Finish" button.

On the Authentication tab, verify **Enable Authentication** is unchecked as Broadcore/Masergy does not require authentication for this type of configuration.

The screenshot shows the UC-Sec Control Center interface. The left sidebar contains a tree view with the following items:

- UC-Sec Control Center
 - Welcome
 - Administration
 - Backup/Restore
 - System Management
 - Global Parameters
 - Global Profiles
 - Domain DoS
 - Fingerprint
 - Server Interworking
 - Phone Interworking

The main area shows the "Global Profiles > Server Configuration: Broadcore" view. It includes a list of profiles on the left:

- SIP Trunk 1
- SM62-Lab1
- Broadcore** (selected)
- Communication Mgr62
- CS1K

At the top of the main area are buttons: "Add Profile", "Rename Profile", "Clone Profile", and "Delete Profile". Below these are tabs: "General", "Authentication", "Heartbeat", and "Advanced". The "Authentication" tab is selected, showing a section titled "Authentication" with the following content:

Enable Authentication ☐

At the bottom of the "Authentication" section is an "Edit" button.

On the Advanced tab, check the **Enable Heartbeat** box. Select **OPTIONS** from the **Method** drop-down menu. Select the desired frequency that the SBC will source OPTIONS. The **From URI** and **To URI** may be filled in to configure easily identifiable URIs to appear in SIP OPTIONS sourced by the SBC.

The screenshot shows the UC-Sec Control Center interface. The left sidebar lists various configuration categories, with 'Server Configuration' selected. The main panel displays the 'Global Profiles > Server Configuration: Broadcore' configuration page. The 'Advanced' tab is active, showing the 'Heartbeat' section. The 'Enable Heartbeat' checkbox is checked. The 'Method' is set to 'OPTIONS', the 'Frequency' is '120 seconds', the 'From URI' is 'PING@broadcore.com', and the 'To URI' is 'PING@broadcore.com'. The 'TCP Probe' checkbox is unchecked. An 'Edit' button is visible at the bottom right of the configuration area.

Heartbeat	
Enable Heartbeat	<input checked="" type="checkbox"/>
Method	OPTIONS
Frequency	120 seconds
From URI	PING@broadcore.com
To URI	PING@broadcore.com
TCP Probe	<input type="checkbox"/>

On the Advanced tab, select the **Interworking Profile** created for Broadcore/Masergy in **Section 7.4.2**. Select the **Signaling Manipulation Script** created in this section. Use default values for all remaining fields.

The screenshot shows the UC-Sec Control Center interface. The left sidebar lists various configuration categories, with 'Server Configuration' selected. The main panel displays the 'Global Profiles > Server Configuration: Broadcore' configuration page. The 'Advanced' tab is active, showing the 'Advanced' section. The 'Enable DoS Protection' and 'Enable Grooming' checkboxes are unchecked. The 'Interworking Profile' is set to 'Masergy Intrwrking', the 'Signaling Manipulation Script' is 'StaticIPAuth', and the 'UDP Connection Type' is 'SUBID'. An 'Edit' button is visible at the bottom right of the configuration area.

Advanced	
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	Masergy Intrwrking
Signaling Manipulation Script	StaticIPAuth
UDP Connection Type	SUBID

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