Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Sprint SIP Trunking with Avaya Aura® Communication Manager R6.2, Avaya Aura® Session Manager R6.2, and Avaya Session Border Controller for Enterprise R4.0.5Q09 – Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between Sprint SIP Trunking and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager R6.2, Avaya Aura® Communication Manager R6.2, Avaya Session Border Controller for Enterprise R4.0.5-Q09 and various Avaya endpoints.

Sprint is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between Sprint SIP Trunking and an Avaya SIP-enabled enterprise solution. Sprint SIP Trunking is a business trunking product supported by the BroadWorks platform. The Avaya solution consists of Avaya Aura® Session Manager R6.2, Avaya Aura® Communication Manager Evolution Server R6.2, Avaya Session Border Controller for Enterprise (Avaya SBCE) R4.0.5-Q09 and various Avaya endpoints.

Avaya Aura® Session Manager is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager. The Avaya Session Border Controller for Enterprise is the point of connection between Avaya Aura® Session Manager and the Sprint SIP trunking service and is used to not only secure the SIP trunk, but also to make adjustments to SIP signaling for interoperability.

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

2. General Test Approach and Test Results

A simulated enterprise site using Communication Manager, Session Manager and the Avaya SBCE was connected to the Sprint test network via an IPSec VPN connection . The enterprise site was configured to connect to Sprint SIP trunking service through this VPN connection.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The 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:

- Response to SIP OPTIONS queries.
- Incoming PSTN calls to various phone types.
- Phone types included H.323, SIP, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types.
  Phone types included H.323, SIP, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.

- Inbound and outbound PSTN calls to/from Avaya one-X® Communicator (soft client).
- Avaya one-X® Communicator 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. Only the H.323 protocol version was tested.
- Various call types including: local, long distance, international, outbound toll-free, operator, local directory assistance (411), and 911 emergency.
- G.729A and G.711MU codecs.
- Voicemail navigation for inbound and outbound calls using DTMF transmission per RFC 2833.
- Caller ID presentation and Caller ID restriction.
- Response to incomplete call attempts and trunk errors.
- User features such as hold and resume, internal call forwarding, transfer, and conference
- Off-net call forwarding, transfer, conference and mobility (extension to cellular).
- Inbound and outbound T.38 faxing.

Items not supported or not tested included the following:

- Operator Assisted calling

2.2. Test Results

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

No limitations were observed in the compliance testing.

2.3. Support

For technical support on Sprint SIP Trunking, contact Sprint at http://www.Sprint.com

3. Reference Configuration

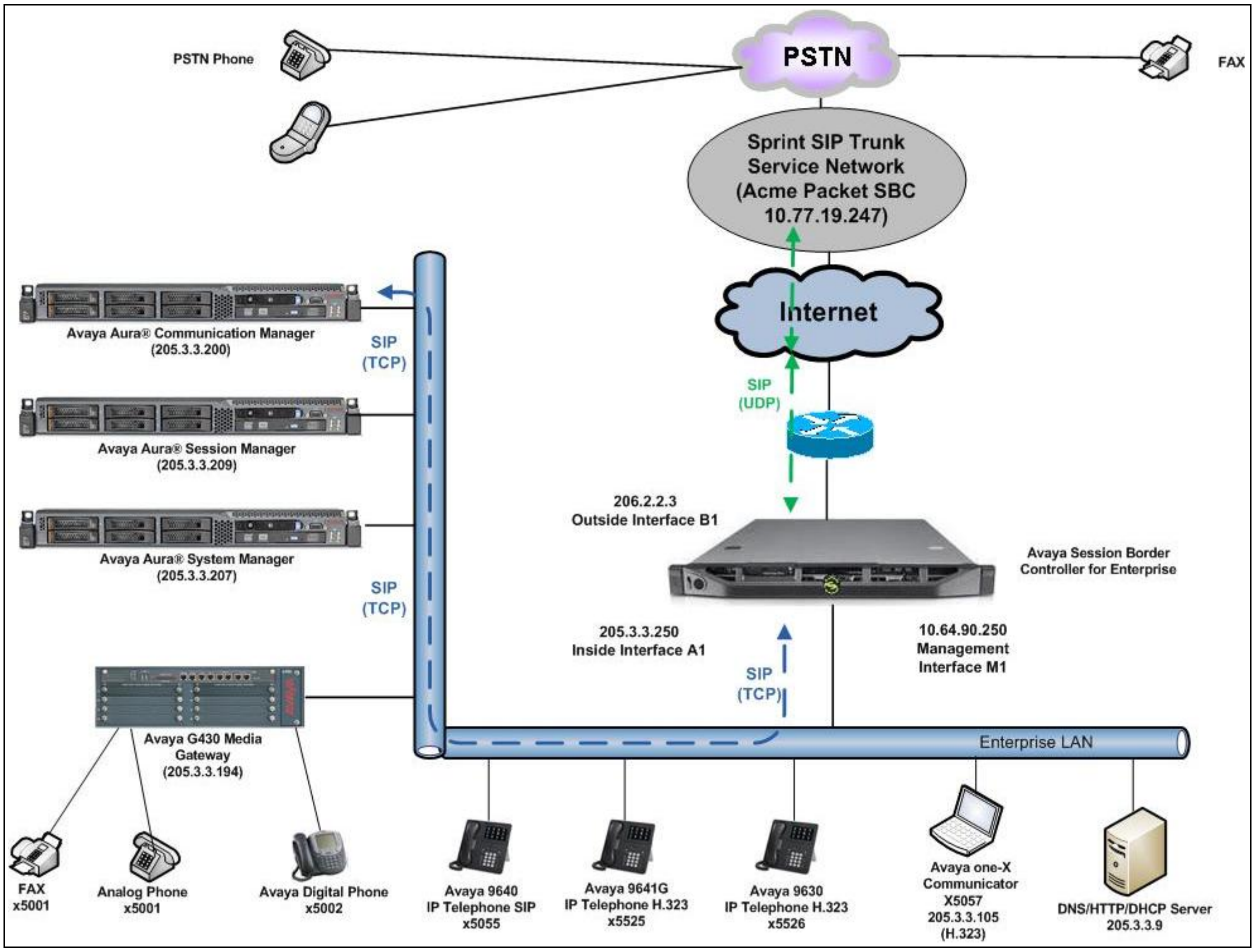
Figure 1 illustrates a sample Avaya SIP-enabled enterprise solution connected to Sprint SIP Trunking service (using a lab test circuit) through a public Internet IPSec VPN connection.

For security purposes, any actual public IP addresses and PSTN routable phone numbers used in the compliance test are masked in these Application Notes.

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

- Avaya S8800 Server running Communication Manager
- Avaya G430 Media Gateway
- Avaya S8800 Server running Session Manager
- Avaya S8800 Server running System Manager
- Avaya SBCE
- Avaya 9600-Series IP Telephones (H.323 and SIP)
- Avaya 96x1-Series IP Telephone (H.323)
- Avaya one-X® Communicator soft phones (H.323 and SIP)
- Avaya digital and analog telephones

Located at the edge of the enterprise is the Avaya SBC for Enterprise. It has a public interface that connects to the external network and a private interface that connects to the enterprise network. All SIP and RTP traffic entering or leaving the enterprise flows through this enterprise SBC. In this way, the SBC can protect the enterprise against any SIP-based attacks. The transport protocol between the enterprise SBC and Sprint across the public IP network is UDP; the transport protocol between the enterprise SBC and Session Manager across the enterprise IP network is TCP.



PSTN Phone
PSTN
FAX
Sprint SIP Trunk Service Network (Acme Packet SBC 10.77.19.247)
Internet
SIP (UDP)
Avaya Aura® Communication Manager (205.3.3.200)
SIP (TCP)
Avaya Aura® Session Manager (205.3.3.209)
Avaya Aura® System Manager (205.3.3.207)
SIP (TCP)
206.2.2.3
Outside Interface B1
Avaya Session Border Controller for Enterprise
205.3.3.250
Inside Interface A1
SIP (TCP)
10.64.90.250
Management Interface M1
Avaya G430 Media Gateway (205.3.3.194)
Enterprise LAN
FAX x5001
Analog Phone x5001
Avaya Digital Phone x5002
Avaya 9640 IP Telephone SIP x5055
Avaya 9641G IP Telephone H.323 x5525
Avaya 9630 IP Telephone H.323 x5526
Avaya one-X Communicator X5057 205.3.3.105 (H.323)
DNS/HTTP/DHCP Server 205.3.3.9

Figure 1: Avaya SIP Enterprise Solution with Sprint SIP Trunking

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

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

Outbound calls to the PSTN are first processed by Communication Manager and may be subject to outbound feature treatment such as automatic route selection, digit manipulation and class of service restrictions. Once Communication Manager selects the proper SIP trunk group, the call is routed to Session Manager. The Session Manager once again uses the configured dial patterns (or regular expressions) to determine the route to the Avaya SBCE. From the enterprise SBC, the call is sent to Sprint SIP Trunking service through the public Internet IPSec VPN connection.

The administration of Modular Messaging and Communication Manager extensions are standard for the enterprise. Since the configuration tasks for Modular Messaging and enterprise endpoints are not directly related to the interoperability with the Sprint SIP Trunking service, they are not included in these Application Notes.

4. Equipment and Software Validated

Avaya IP Telephony Solution Components	
Equipment/Software	**Release/Version**
Avaya Aura® Communication Manager running on Avaya S8800 Server	6.2 (R016x.02.0.823.0-19883)
Avaya G430 Media Gateway	31.22.0
– ANA MM711AP	HW33 FW091
– DCP MM712AP	HW07 FW007
Avaya Aura® Session Manager running on Avaya S8800 Server	6.2.2.0.622005
Avaya Aura® System Manager running on Avaya S8800 Server	6.2 Build 6.2.0.0.15669 Patch-6.2.12.202 Software Update Revision No: 6.2.14.1.1925
Avaya 96x0 Series IP Telephone (H.323)	Avaya one-X® Deskphone Edition 3.1.1
Avaya 96x0 Series IP Telephone (SIP)	Avaya one-X® Deskphone SIP Edition 2.6.6
Avaya 96x1 Series IP Telephone (H.323)	Avaya one-X® Deskphone Release S6.2119
Avaya one-X Communicator (H.323 & SIP)	6.1.5.07-SP5-37495
Avaya 8410D Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
Fax device	Ventafax Home Version 6.2.80.203
Avaya Session Border Controller for Enterprise	4.0.5.Q09
Avaya Modular Messaging	V5.2 (9.2.350.5019)
Sprint SIP Trunking Components	
Equipment/Software	**Release/Version**
Acme Packet 4250 SBC	SC6.2.0m6p2

The specific hardware and software listed in the table above were used for the compliance testing. Note that this solution will be compatible with other Avaya Server and Media Gateway platforms running similar versions of Communication Manager and Session Manager.

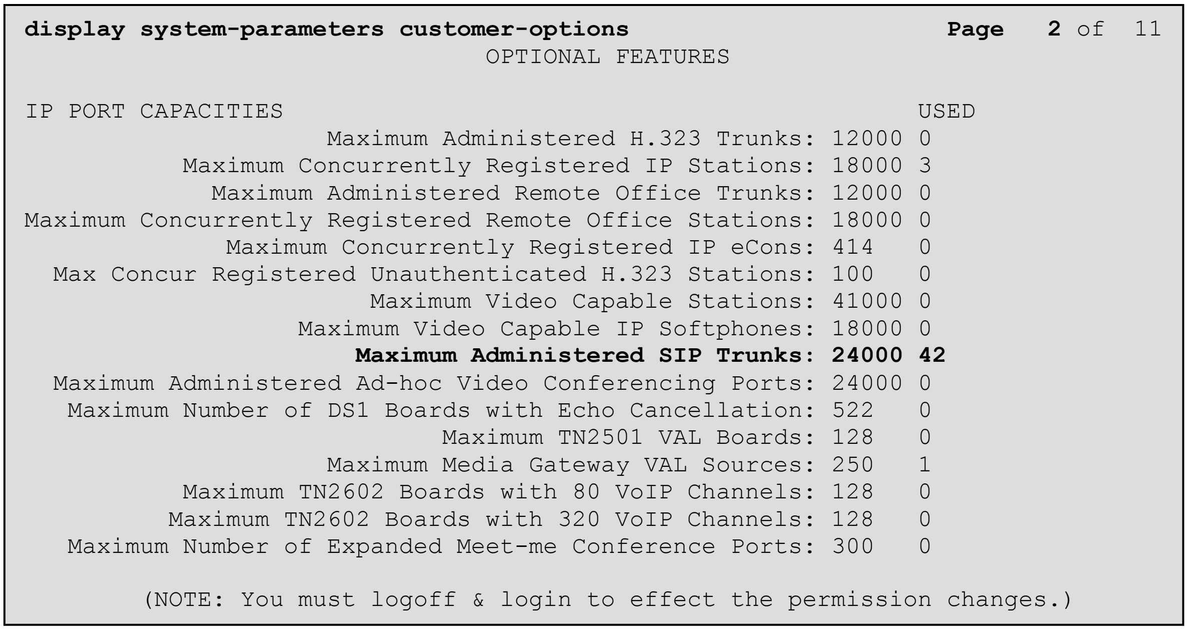
5. Configure Avaya Aura® Communication Manager

This section describes the procedures for configuring Communication Manager for Sprint SIP Trunking. A SIP trunk is established between Communication Manager and Session Manager for use by signaling traffic to and from Sprint. It is assumed the general installation of Communication Manager, Avaya G430 Media Gateway and Session Manager has been previously completed and is not discussed here.

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

5.1. Licensing and Capacity

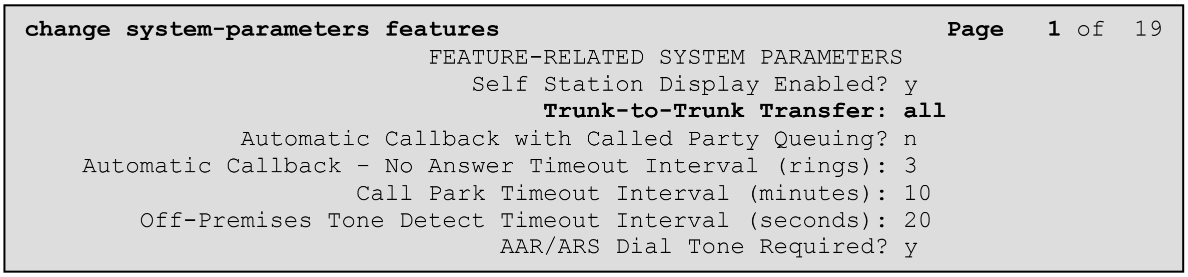
Use the **display system-parameters customer-options** command to verify that the **Maximum Administered SIP Trunks** value on **Page 2** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise including any trunks to the service provider. The example shows that 4000 licenses are available and 244 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: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 41000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 42
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   1
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

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

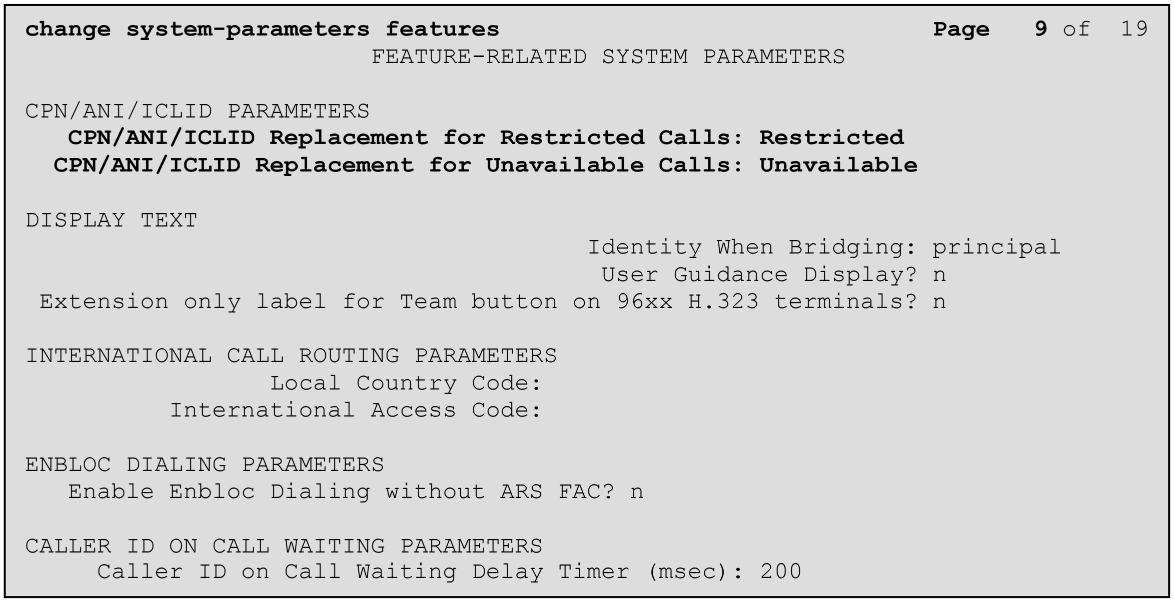
5.2. System Features

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



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

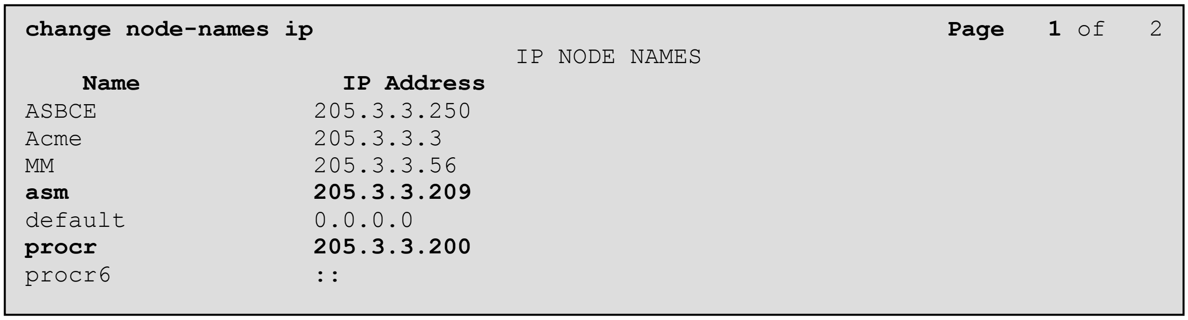
On **Page 9** verify that a text string has been defined to replace the Calling Party Number (CPN) for restricted or unavailable calls. This text string is entered in the two fields highlighted below. The compliance test used the values of ***Restricted*** for restricted calls and ***Unavailable*** for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



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

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

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

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

ENBLOC DIALING PARAMETERS
   Enable Enbloc Dialing without ARS FAC? n

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

5.3. IP Node Names

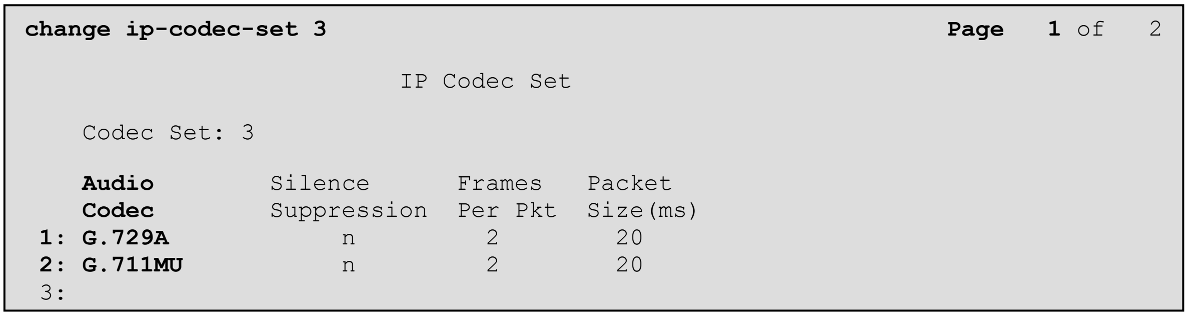
Use the **change node-names ip** command to verify that node names have been previously defined for the IP addresses of the Avaya S8800 Server running Communication Manager ***(procr***) and for Session Manager (***asm***). These node names will be needed for defining the service provider signaling group in **Section 5.6**.



```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address
ASBCE               205.3.3.250
Acme                205.3.3.3
MM                  205.3.3.56
asm                 205.3.3.209
default             0.0.0.0
procr               205.3.3.200
procr6              ::
```

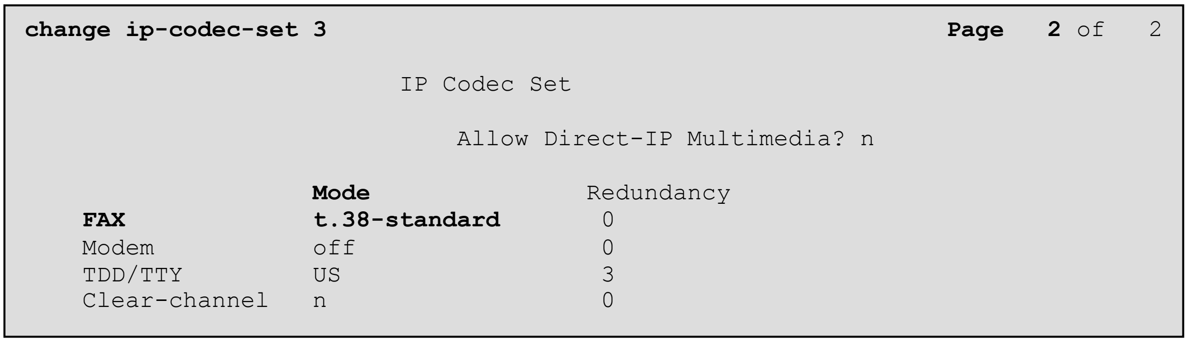
5.4. Codecs

Use the **change ip-codec-set** command to define a list of codecs to use for calls between the enterprise and the service provider. For the compliance test, **ip-codec-set *3*** was used for this purpose. Sprint officially supports G.729A and G.711MU. Thus, these codecs were included in the set. Enter ***G.729A*** first and then ***G.711MU*** in the **Audio Codec** column of the table. By listing the G.729A codec first, this tells the Sprint network that G.729A is preferred. Default values can be used for all other fields.



```
change ip-codec-set 3                                           Page   1 of   2

                          IP Codec Set

    Codec Set: 3

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

On **Page 2**, set the **Fax Mode** to ***t.38-standard***. This setting was used to test both inbound and outbound T.38 faxing.



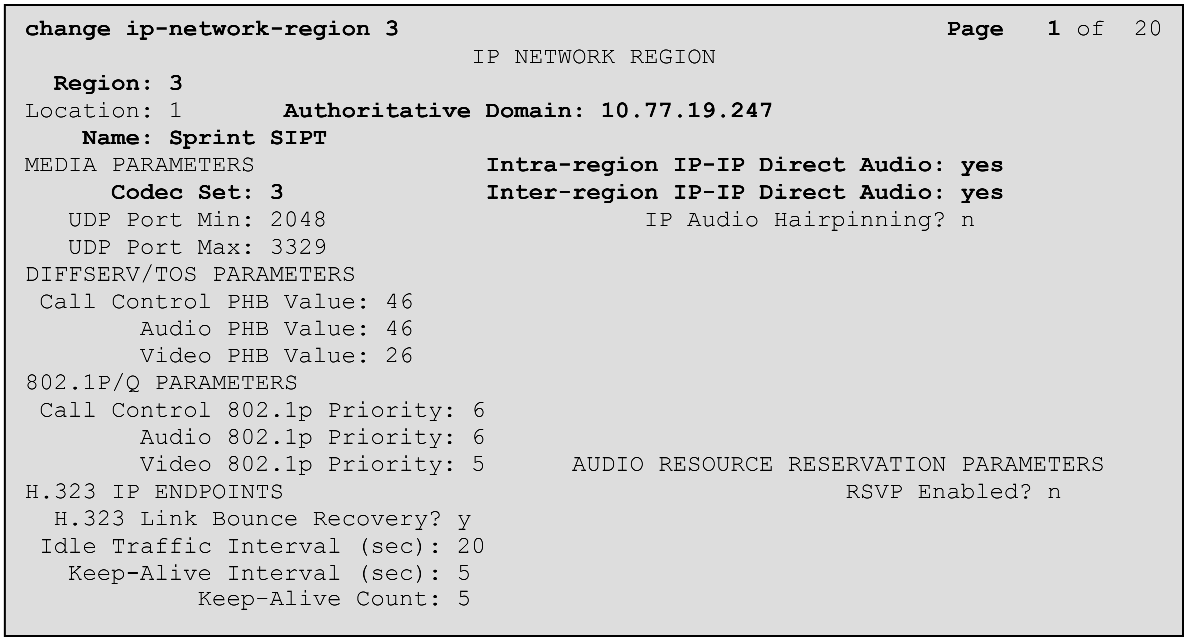
```
change ip-codec-set 3                                           Page   2 of   2

                          IP Codec Set

                              Allow Direct-IP Multimedia? n

                    Mode                 Redundancy
    FAX             t.38-standard         0
    Modem           off                   0
    TDD/TTY         US                    3
    Clear-channel   n                     0
```

For the enterprise, **ip-codec-set *1*** was used and was configured with the exact same settings as above except that the codec set contained ***G.711MU*** as the first codec and ***G.729A*** as the second.

5.5. IP Network Region

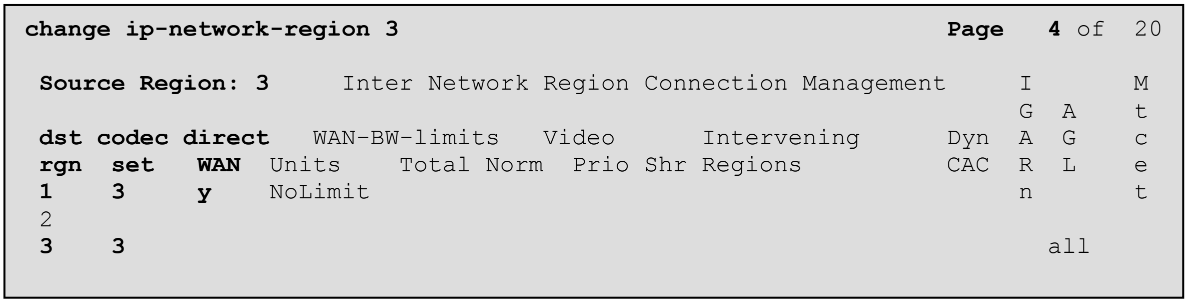
Create a separate IP network region for the service provider trunk. This allows for separate codec or quality of service settings to be used (if necessary) for calls between the enterprise and the service provider versus calls within the enterprise or elsewhere. For the compliance test, **IP-network-region 3** was chosen for the service provider trunk. Use the **change ip-network-region *3*** command to configure region 3 with the following parameters:

- Set the **Authoritative Domain** field to match the IP address of the Sprint SIP trunking SBC. In this configuration, the IP address was set to ***10.77.19.247***. This IP address appears in the "From" header of SIP messages originating from this IP region, namely the PSTN.
- Enter a descriptive name in the **Name** field.
- Enable **IP-IP Direct Audio** (shuffling) to allow audio to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. Set both **Intra-region** and **Inter-region IP-IP Direct Audio** to ***yes*.** This is the default setting. Shuffling can be further restricted at the trunk level on the Signaling Group form.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.4**.
- Default values can be used for all other fields.



```
change ip-network-region 3                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 3
Location: 1       Authoritative Domain: 10.77.19.247
    Name: Sprint SIPT
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 3              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

On **Page 4**, define the IP codec set to be used for traffic between region 3 and region 1. Enter the desired IP codec set in the **codec set** column of the row with destination region (**dst rgn**) 1. Default values may be used for all other fields. The example below shows the settings used for the compliance test. It indicates that codec set 3 will be used for calls between region 3 (the service provider region) and region 1 (the rest of the enterprise). Also, it is necessary to set the **direct WAN** field to ***y*** in order to connect Region 1 to Region 3.



```
change ip-network-region 3                                      Page   4 of  20

 Source Region: 3     Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
 1    3     y    NoLimit                                             n       t
 2
 3    3                                                                all
```

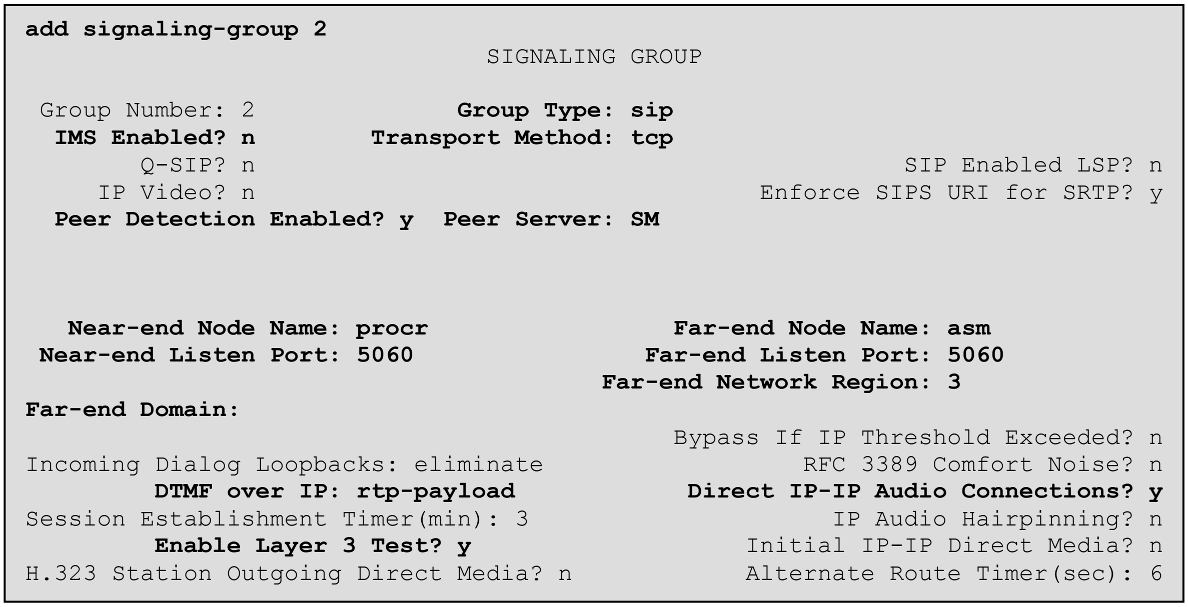
5.6. Signaling Group

Use the **add signaling-group** command to create a signaling group between Communication Manager and the 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***. This specifies the Communication Manager will serve as an Evolution Server for the Session Manager.
- Set the **Transport Method** to the value of ***tcp*** (Transport Layer Security). For ease of troubleshooting during testing, the compliance test was conducted with the **Transport Method** set to ***tcp.*** The transport method specified here is used between Communication Manager and Session Manager. For security purposes, the default Transport Method value of TLS is recommended for production environments.
- Set the **Near-end Listen Port** and **Far-end Listen Port** to a valid unused port. This is necessary for Session Manager to distinguish this trunk from the trunk used for other enterprise SIP traffic. The default well-known port value for SIP over TCP is 5060. The compliance test was conducted with the **Near-end Listen Port** and **Far-end Listen Port** set to ***5060***.
- Set the **Peer Detection Enabled** field to ***y***. The **Peer-Server** field will initially be set to ***Others*** and cannot be changed via administration. Later, the **Peer-Server** field will automatically change to ***SM*** once Communication Manager detects its peer as a Session Manager.
- Set the **Near-end Node Name** to ***procr***. This node name maps to the IP address of the Avaya S8800 Server running Communication Manager as defined in **Section 5.3**.
- Set the **Far-end Node Name** to ***asm***. This node name maps to the IP address of the S8800 Server running Session Manager as defined in **Section 5.3**.

- Set the **Far-end Network Region** to the IP network region defined for the service provider in **Section 5.5**.
- Leave the **Far-end Domain** field blank.
- Set **Direct IP-IP Audio Connections** to ***y***. This field will enable media shuffling on the SIP trunk allowing Communication Manager to redirect media traffic directly between the SIP trunk and the enterprise endpoint. If this value is set to ***n***, the Avaya Media Gateway will remain in the media path of all calls between the SIP trunk and the endpoint. Depending on the number of media resources available in the Avaya Media Gateway, these resources may be depleted during high call volume preventing additional calls from completing.
- 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.

Note that the **Initial IP-IP Direct Media** setting must be consistent with the setting in the signaling group used for general internal SIP traffic; otherwise unintended side effects could occur. The default setting is not to enable this feature.

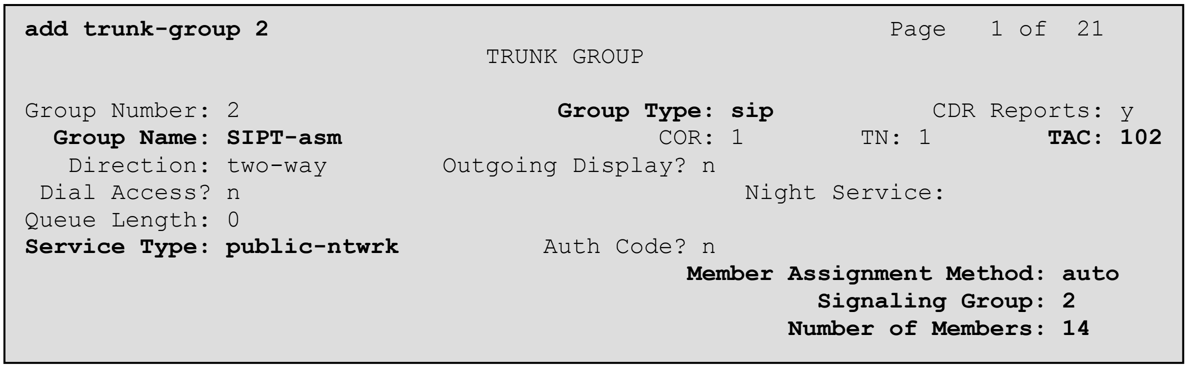


```
add signaling-group 2
                                SIGNALING GROUP

 Group Number: 2              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                             SIP Enabled LSP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                 Far-end Node Name: asm
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 3
Far-end Domain:
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                   Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

5.7. Trunk Group

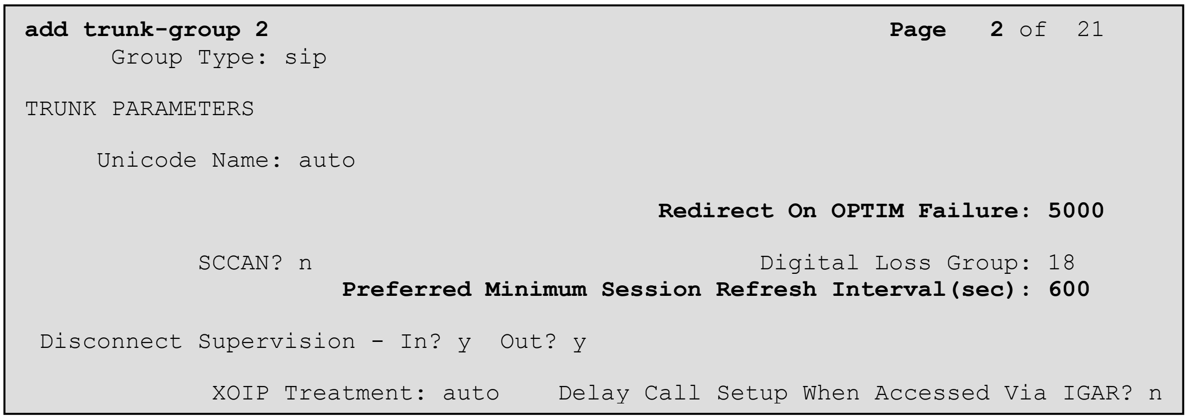
Use the **add trunk-group** command to create a trunk group for the signaling group created in **Section 5.6**. For the compliance test, trunk group 2 was configured using the parameters highlighted below.

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



```
add trunk-group 2                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 2                     Group Type: sip         CDR Reports: y
  Group Name: SIPT-asm                     COR: 1        TN: 1        TAC: 102
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 2
                                                     Number of Members: 14
```

On **Page 2**, leave the **Redirect On OPTIM Failure** timer set to the default value of ***5000***. Note that the **Redirect On OPTIM Failure** timer is defined in milliseconds. Verify that the **Preferred Minimum Session Refresh Interval** is set to a value acceptable to the service provider. This value defines the interval that re-INVITEs must be sent to keep the active session alive. For the compliance test, the value of ***600*** seconds was used.



```
add trunk-group 2                                          Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

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

 Disconnect Supervision - In? y  Out? y

             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n
```

On **Page 3**, set the **Numbering Format** field to ***public***. This field specifies the format of the calling party number (CPN) sent to the far-end. Beginning with Communication Manager 6.0, public numbers are automatically preceded with a + sign (E.164 numbering format) when passed in the SIP From, Contact and P-Asserted Identity headers. The compliance test used 10 digit numbering format.

Set the **Replace Restricted Numbers** and **Replace Unavailable Numbers** fields to ***y***. This will allow the CPN displayed on enterprise 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 an enterprise user requests CPN block on a particular call routed out this trunk. Default values were used for all other fields.

```
add trunk-group 2                                           Page   3 of  21
TRUNK FEATURES

          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y


                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? y

                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? y

 DSN Term? n
```

On **Page 4**, Set the **Mark Users as Phone field** to ***y*** as this is the preferred value for the Sprint SIP trunking service.

Set the **Network Call Redirection** field to ***y***. Setting the **Network Call Redirection** flag to ***y*** enables the use of the SIP REFER message for call redirection such as call transfers back out to the PSTN.

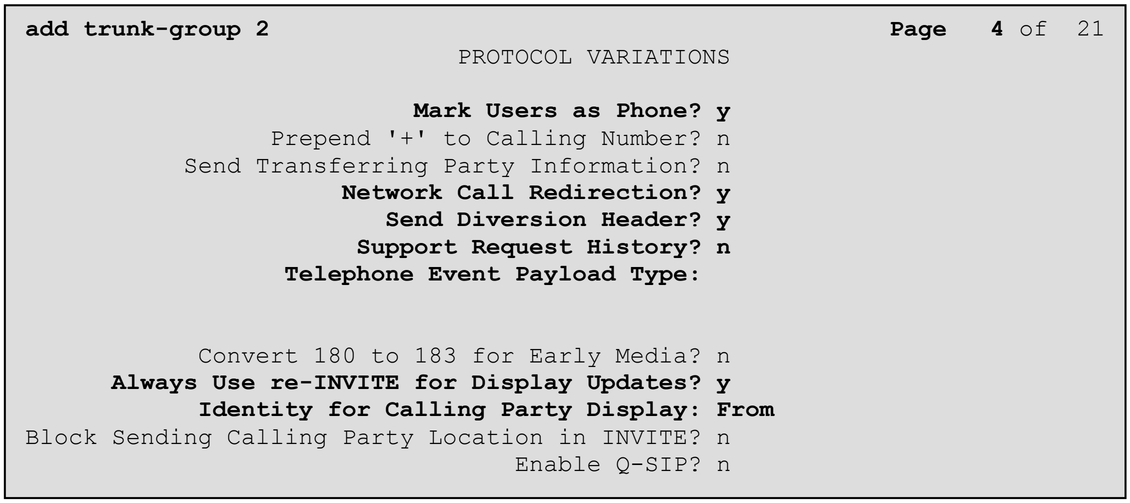
Set the **Send Diversion Header** field to ***y***. This field provides additional information to the network if the call has been re-directed. This is needed to support call forwarding of inbound calls back to the PSTN and some Extension to Cellular (EC500) call scenarios.

Set the **Support Request History** field to ***n***. This parameter determines whether the SIP History-Info header will be included in the call-redirection from the enterprise. Call-redirection of an inbound call from the PSTN back to the PSTN failed in the compliance test when the call re-direction contains the History-Info header.

Leave the **Telephone Event Payload Type** field blank.

Set the **Always Use re-INVITE for Display Updates** field to ***y***.

Set the **Identity for Calling Party Display** to ***From***. Since Sprint sends a cookie in the SIP Contact and PAI headers, it is necessary to set this field to ***From*** so that the proper Caller ID is displayed based on the From header rather than the PAI header.

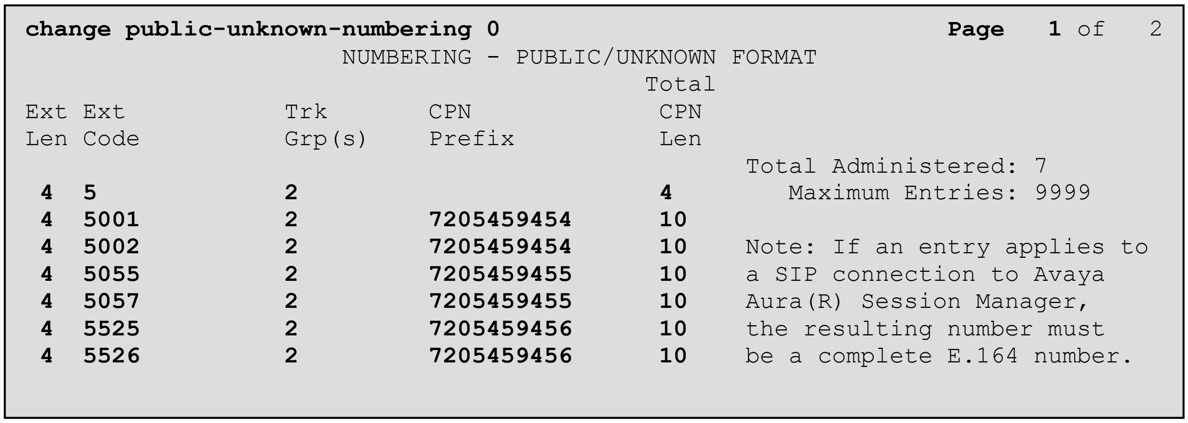


```
add trunk-group 2                                           Page   4 of  21
                              PROTOCOL VARIATIONS

                           Mark Users as Phone? y
                 Prepend '+' to Calling Number? n
           Send Transferring Party Information? n
                      Network Call Redirection? y
                         Send Diversion Header? y
                       Support Request History? n
                  Telephone Event Payload Type:


            Convert 180 to 183 for Early Media? n
      Always Use re-INVITE for Display Updates? y
            Identity for Calling Party Display: From
Block Sending Calling Party Location in INVITE? n
                                  Enable Q-SIP? n
```

5.8. Calling Party Information

The calling party number is sent in the SIP “From”, “Contact” and “PAI” headers. Since public numbering was selected to define the format of this number (**Section 5.7**), use the **change public-unknown-numbering** command to create an entry for each extension which has a DID assigned. The DID numbers are provided by the SIP service provider. Each DID number is assigned to an enterprise internal extension or Vector Directory Numbers (VDNs). It is also used to authenticate the caller.

The screen below shows the DID numbers assigned for testing by Sprint. It was necessary to assign theses 3 DID numbers to 6 enterprise extensions since only 3 DIDs were provisioned for the compliance testing. These DIDs were mapped to extensions 5001, 5002, 5055, 5057, 5525, and 5526. These same 10-digit numbers were used for the outbound calling party indentification on the service provider trunk when calls were originated from these 6 extensions.

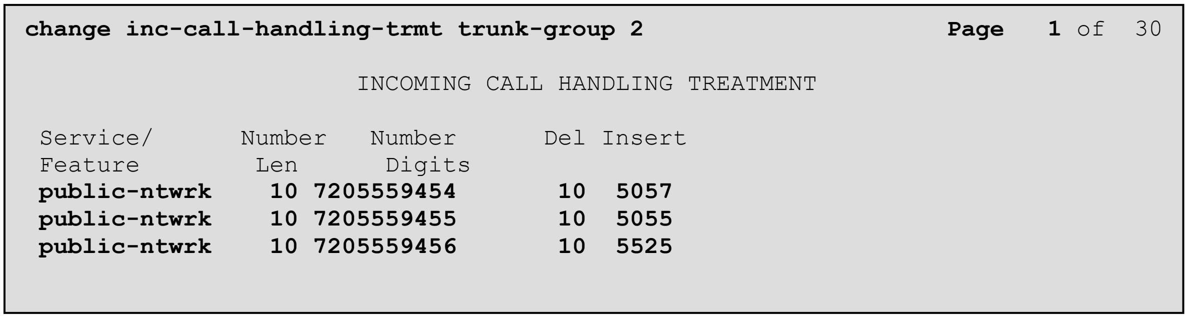


```
change public-unknown-numbering 0                               Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext           Trk       CPN             CPN
Len Code          Grp(s)    Prefix          Len
                                                  Total Administered: 7
 4  5             2                         4        Maximum Entries: 9999
 4  5001          2         7205459454      10
 4  5002          2         7205459454      10    Note: If an entry applies to
 4  5055          2         7205459455      10    a SIP connection to Avaya
 4  5057          2         7205459455      10    Aura(R) Session Manager,
 4  5525          2         7205459456      10    the resulting number must
 4  5526          2         7205459456      10    be a complete E.164 number.
```

In the example above, the top entry is necessary for Modular Messaging to receive the local extension number and properly identify the called party. With this entry, all stations with a 4-digit extension beginning with 5 will send the calling party number across trunk 2 to Modular Messaging, as the extension number.

5.9. Incoming Call Handling Treatment

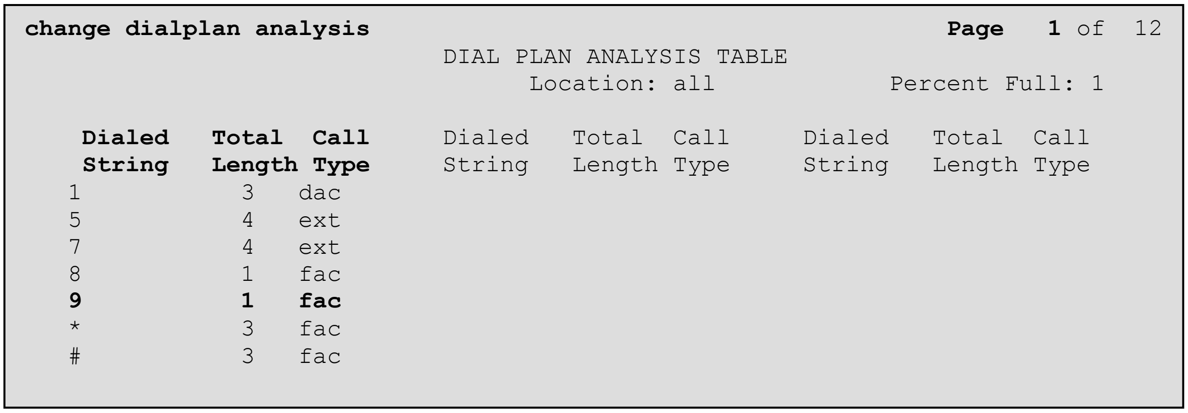
Use the **change incoming-call-handling-trmt trunk group *x*** command, where ***x*** is the SIP trunk configured for the service provider, to map inbound DIDs to extensions. During the compliance test, it was necessary to change these mappings between DIDs and extensions since there were only 3 DIDs assigned by the Service Provider.



```
change inc-call-handling-trmt trunk-group 2                     Page   1 of  30

                       INCOMING CALL HANDLING TREATMENT

 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    10 7205559454       10  5057
 public-ntwrk    10 7205559455       10  5055
 public-ntwrk    10 7205559456       10  5525
```

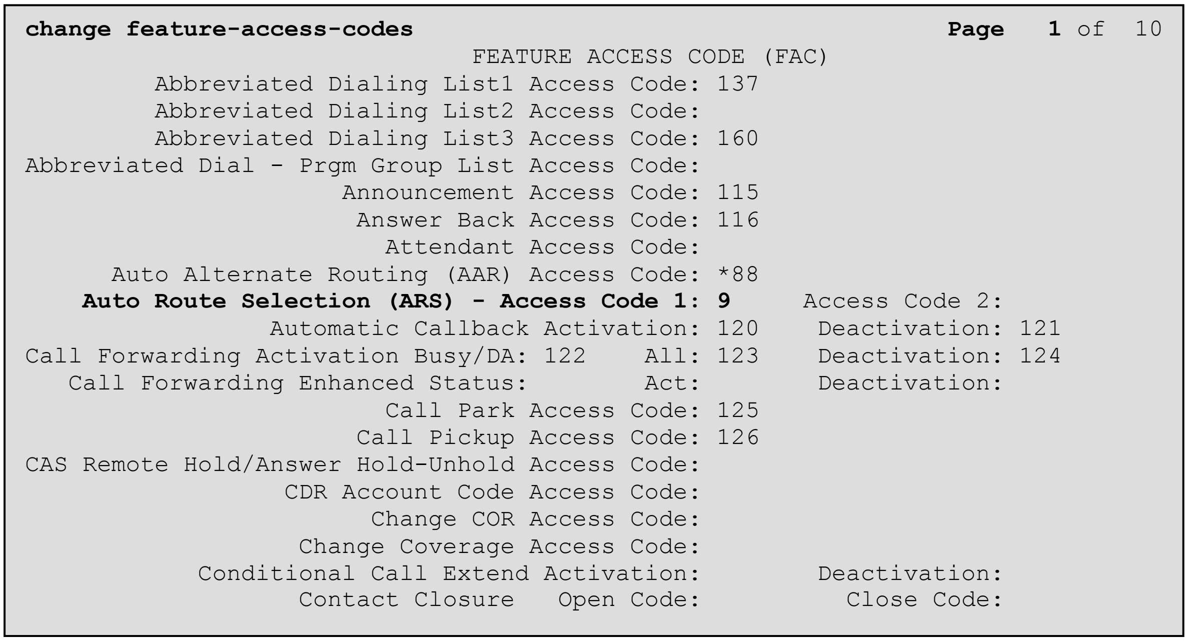
5.10. Outbound Routing

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



```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all          Percent Full: 1

    Dialed   Total  Call     Dialed   Total  Call     Dialed   Total  Call
    String   Length Type     String   Length Type     String   Length Type
   1           3    dac
   5           4    ext
   7           4    ext
   8           1    fac
   9           1    fac
   *           3    fac
   #           3    fac
```

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



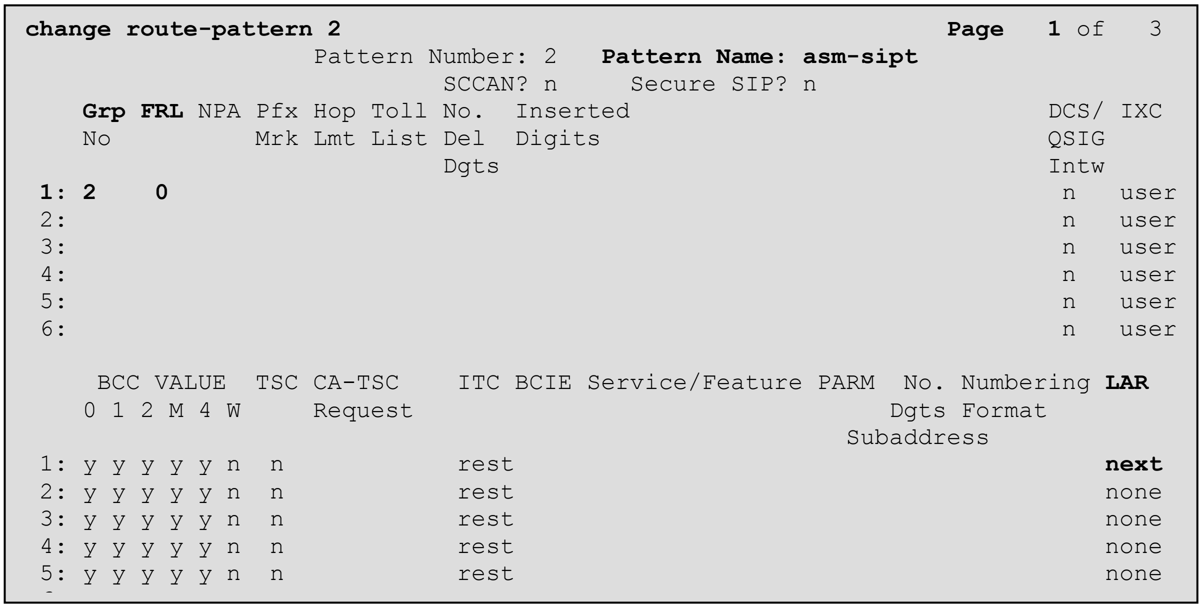
```
change feature-access-codes                                     Page   1 of  10
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: 137
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code: 160
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: 115
                       Answer Back Access Code: 116
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: *88
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation: 120    Deactivation: 121
Call Forwarding Activation Busy/DA: 122    All: 123    Deactivation: 124
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code: 125
                       Call Pickup Access Code: 126
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
            Conditional Call Extend Activation:        Deactivation:
                   Contact Closure   Open Code:          Close Code:
```

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

```
change ars analysis                                                     Page   1

                           ARS DIGIT ANALYSIS REPORT

                            Location:  all

               Dialed            Total        Route    Call      Node
               String          Min    Max    Pattern   Type     Number

         0                      1      1      2        op
         0                      8      8      2        op
         0                      11     11     2        op
         00                     2      2      2        op
         01                     9      17     deny     iop
         011                    10     18     2        intl
         130                    11     11     2        fnpa
         1300                   11     11     deny     fnpa
         131                    11     11     2        fnpa
         132                    11     11     2        fnpa
         1700                   11     11     deny     fnpa
         171                    11     11     2        fnpa
         172                    11     11     2        fnpa
         173                    11     11     2        fnpa
         174                    11     11     2        fnpa
         175                    11     11     2        fnpa
         180                    11     11     2        fnpa
         1800                   11     11     2        fnpa
         1800555                11     11     deny     fnpa
         181                    11     11     2        fnpa
         182                    11     11     2        fnpa
         183                    11     11     2        fnpa
         184                    11     11     2        fnpa
         185                    11     11     2        fnpa
         186                    11     11     2        fnpa
         187                    11     11     2        fnpa
         188                    11     11     2        fnpa
         189                    11     11     2        fnpa
         190                    11     11     2        fnpa
         1xxx555                11     11     deny     fnpa
         1xxx976                11     11     deny     fnpa
         2                      10     10     2        hnpa
         3                      10     10     2        hnpa
         4                      10     10     2        hnpa
         411                    3      3      2        svcl
         5                      10     10     2        hnpa
         555                    7      7      deny     hnpa
         6                      10     10     2        hnpa
         611                    3      3      2        svcl
         7                      10     10     2        hnpa
         8                      10     10     2        hnpa
         811                    3      3      2        svcl
         9                      10     10     2        hnpa
         911                    3      3      2        svcl
         976                    7      7      deny     hnpa
```

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

- **Pattern Name**: Enter a descriptive name.
- **Grp No**: Enter the outbound trunk group for the SIP service provider. For the compliance test, trunk group ***2*** was used.
- **FRL**: Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of ***0*** is the least restrictive level.
- **LAR**: ***next***



```
change route-pattern 2                                                Page   1 of   3
                    Pattern Number: 2   Pattern Name: asm-sipt
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                                DCS/ IXC
    No          Mrk Lmt List Del  Digits                                  QSIG
                             Dgts                                         Intw
 1: 2    0                                                                 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                                         next
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
```

6. Configure Avaya Aura® Session Manager

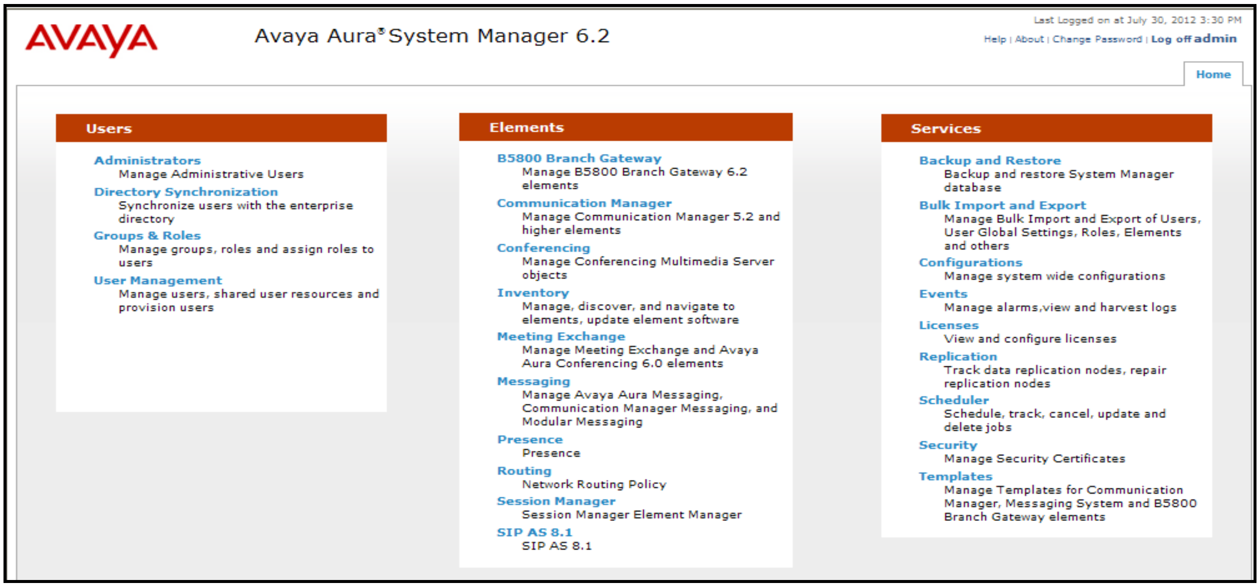
This section provides the procedures for configuring Session Manager. The procedures include the following items:

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

It may not be necessary to create all the items above when configuring for 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. 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. At the **System Manager Log On** screen, provide the appropriate credentials and click on **Login** (not shown). The initial screen shown below is then displayed.



AVAYA Avaya Aura® System Manager 6.2

Last Logged on at July 30, 2012 3:30 PM
Help | About | Change Password | Log off admin

Home

Users

Administrators
Manage Administrative Users
Directory Synchronization
Synchronize users with the enterprise directory
Groups & Roles
Manage groups, roles and assign roles to users
User Management
Manage users, shared user resources and provision users

Elements

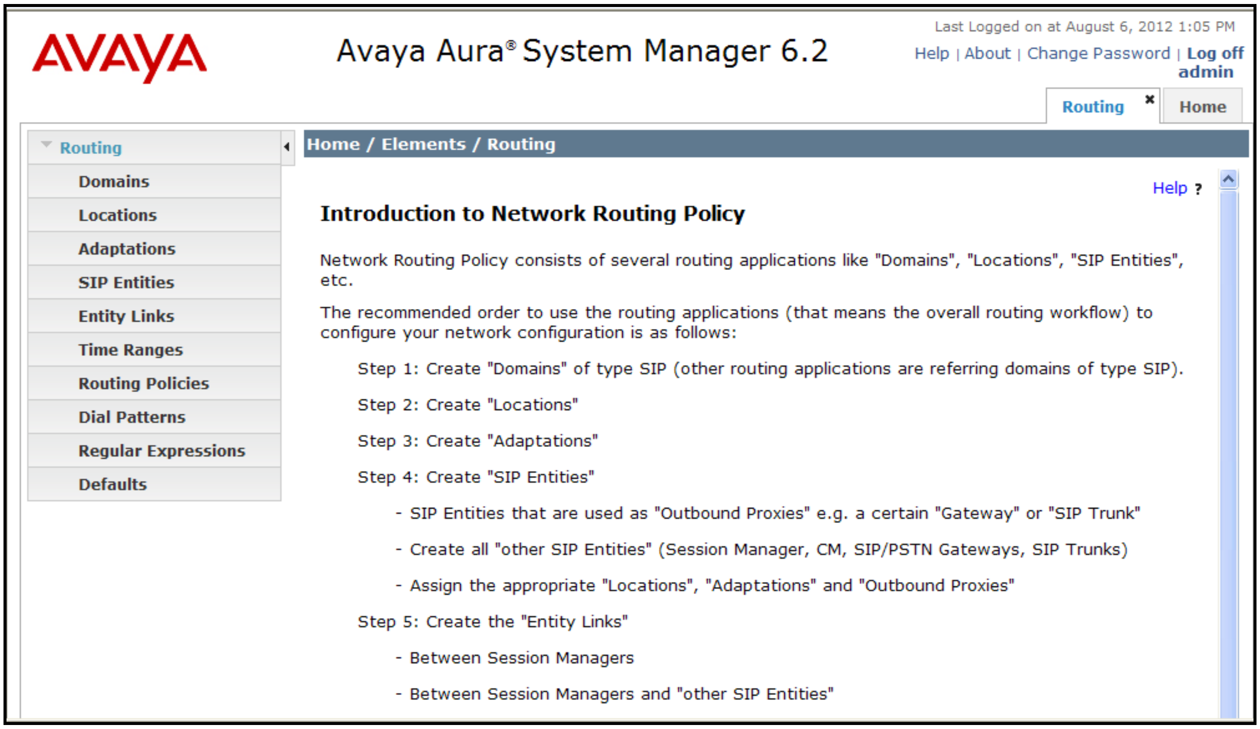
B5800 Branch Gateway
Manage B5800 Branch Gateway 6.2 elements
Communication Manager
Manage Communication Manager 5.2 and higher elements
Conferencing
Manage Conferencing Multimedia Server objects
Inventory
Manage, discover, and navigate to elements, update element software
Meeting Exchange
Manage Meeting Exchange and Avaya Aura Conferencing 6.0 elements
Messaging
Manage Avaya Aura Messaging, Communication Manager Messaging, and Modular Messaging
Presence
Presence
Routing
Network Routing Policy
Session Manager
Session Manager Element Manager
SIP AS 8.1
SIP AS 8.1

Services

Backup and Restore
Backup and restore System Manager database
Bulk Import and Export
Manage Bulk Import and Export of Users, User Global Settings, Roles, Elements and others
Configurations
Manage system wide configurations
Events
Manage alarms,view and harvest logs
Licenses
View and configure licenses
Replication
Track data replication nodes, repair replication nodes
Scheduler
Schedule, track, cancel, update and delete jobs
Security
Manage Security Certificates
Templates
Manage Templates for Communication Manager, Messaging System and B5800 Branch Gateway elements

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

The navigation tree displayed in the left pane will be referenced in subsequent sections to navigate to items requiring configuration.



Introduction to Network Routing Policy

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

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

Step 1: Create "Domains" of type SIP (other routing applications are referring domains of type SIP).

Step 2: Create "Locations"

Step 3: Create "Adaptations"

Step 4: Create "SIP Entities"

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

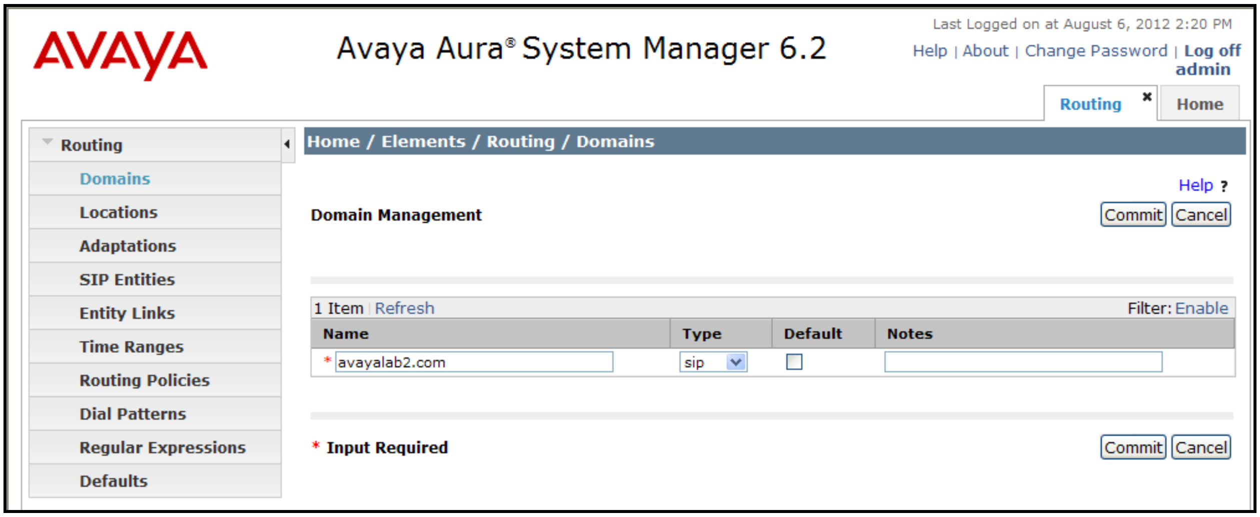
- Between Session Managers
- Between Session Managers and "other SIP Entities"

6.2. Specify SIP Domain

Create a SIP domain for each domain of which Session Manager will need to be aware in order to route calls. For the compliance test, this includes the enterprise domain (***avayalab2.com***). Navigate to **Routing → Domains** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Name:** Enter the domain name.
- **Type:** Select **sip** from the pull-down menu.
- **Notes:** Add a brief description (optional).

Click **Commit**. The screen below shows the entry for the enterprise domain.



Domain Management

Name	Type	Default	Notes
* avayalab2.com	sip	□	

* Input Required

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).

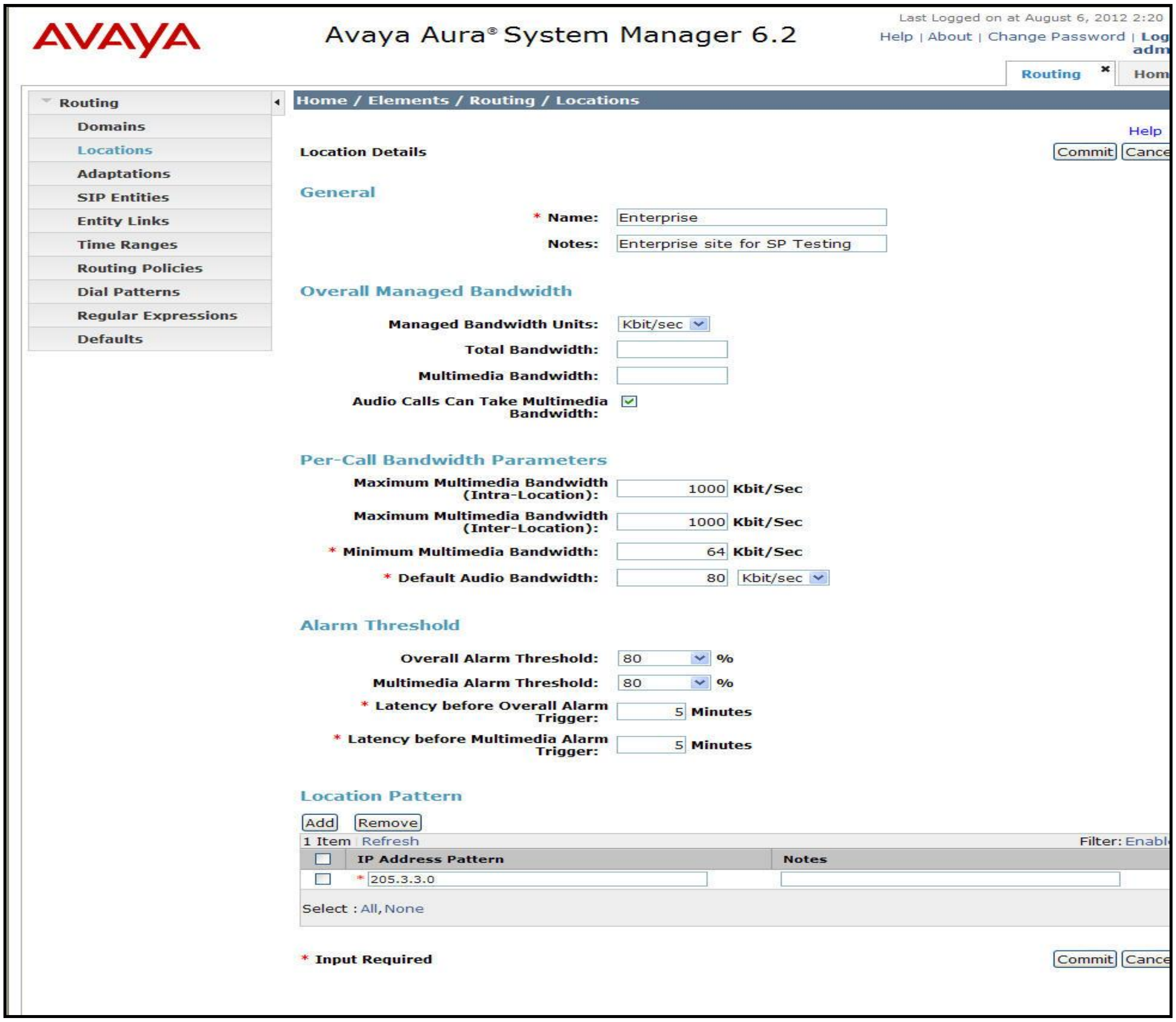
In the **General** section, enter the following values:

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

In the **Location Pattern** section (see screen below), click **Add** and enter the following values:

- **IP Address Pattern:** An IP address pattern used to identify the Location.
- **Notes:** Add a brief description (optional).

Displayed below is the screen for the addition of the **Enterprise** Location, which includes all equipment on the enterprise network including Communication Manager and Session Manager itself. Click **Commit** to save.



Location Details

General

* Name:	Enterprise
Notes:	Enterprise site for SP Testing

Overall Managed Bandwidth

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

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location):	1000 Kbit/Sec
Maximum Multimedia Bandwidth (Inter-Location):	1000 Kbit/Sec
* Minimum Multimedia Bandwidth:	64 Kbit/Sec
* Default Audio Bandwidth:	80 Kbit/sec

Alarm Threshold

Overall Alarm Threshold:	80 %
Multimedia Alarm Threshold:	80 %
* Latency before Overall Alarm Trigger:	5 Minutes
* Latency before Multimedia Alarm Trigger:	5 Minutes

Location Pattern

☐	IP Address Pattern	Notes
☐	* 205.3.3.0	

* Input Required

Note that call bandwidth management parameters should be set per customer requirement.

6.4. Add Adaptation Module

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

For interoperability with Sprint SIP Trunking no Adaptations were necessary.

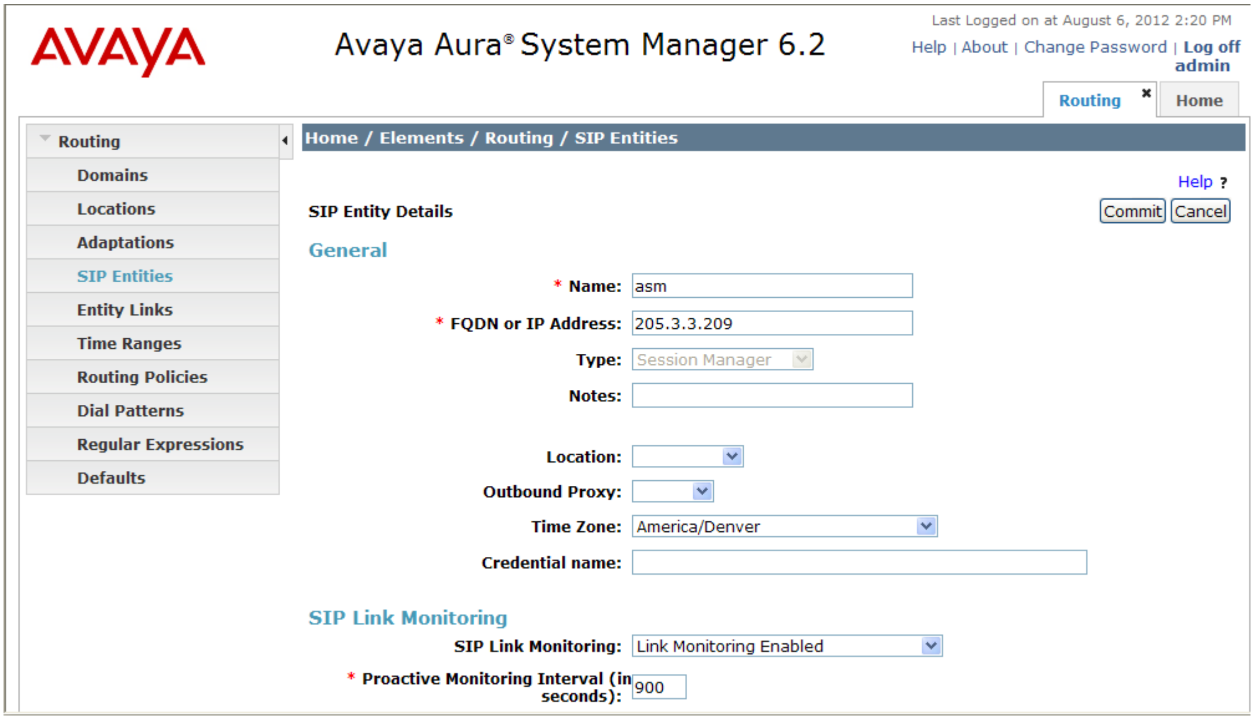
6.5. Add SIP Entities

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

- **Name:** Enter a descriptive name.

- **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity that is used for SIP signaling.
- **Type:** Select **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 created in **Section 6.4** 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 the Session Manager SIP Entity. The IP address of the Session Manager signaling interface is entered for **FQDN or IP Address**.



SIP Entity Details

General

* Name:	asm
* FQDN or IP Address:	205.3.3.209
Type:	Session Manager
Notes:	
Location:	
Outbound Proxy:	
Time Zone:	America/Denver
Credential name:	

SIP Link Monitoring

SIP Link Monitoring:	Link Monitoring Enabled
* Proactive Monitoring Interval (in seconds):	900

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 the **Session Manager** SIP Entity.

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

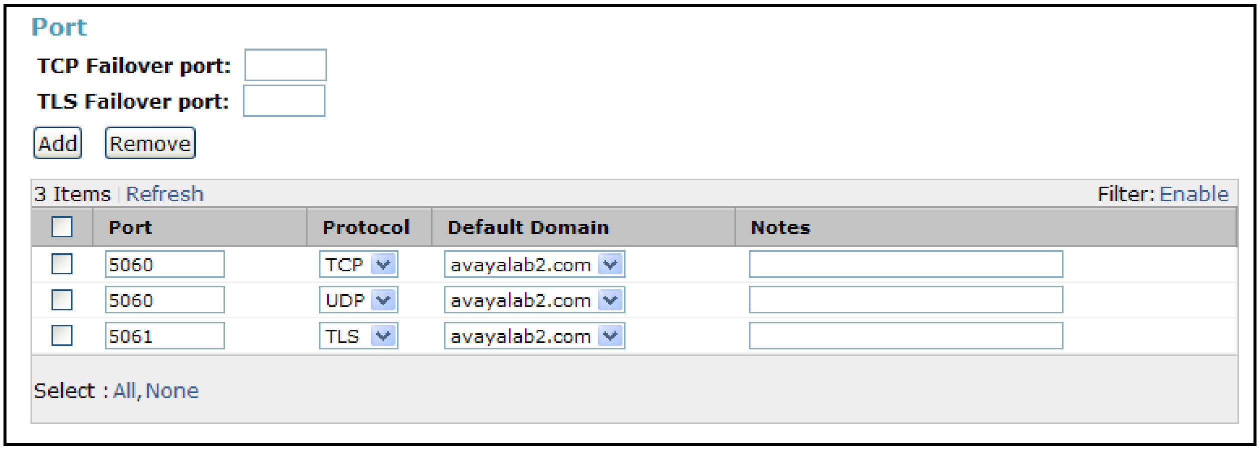
- **Port:** Port number on which the Session Manager can listen for SIP requests.
- **Protocol:** Transport protocol to be used to send and receive SIP requests.
- **Default Domain:** The domain used for the enterprise.

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

Although the default port values for SIP over UDP and TLS were configured, they were not used in this compliance test. One **Port** entry was used for three SIP Entities:

- **5060** with **TCP** for connecting to Avaya SBCE
- **5060** with **TCP** for connecting to Communication Manager
- **5060** with **TCP** for connecting to Modular Messaging

In addition, port 5060 with TCP was also used between Session Manager and Communication Manager for Avaya SIP telephones and SIP soft clients. This SIP Link was part of the standard configuration on Session Manager and was not directly relevant to the inter-operability with Sprint SIP Trunking.



TCP Failover port:

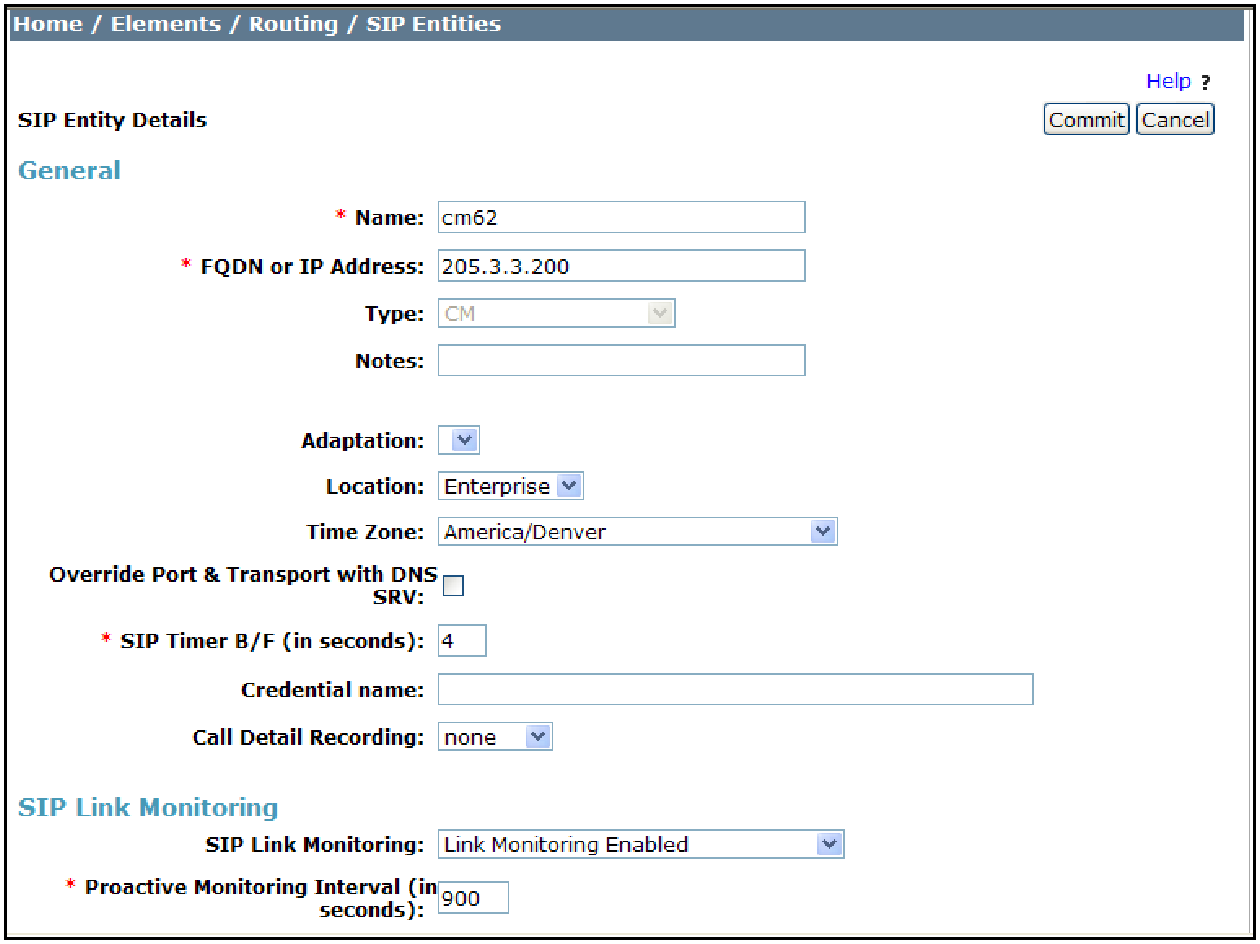
TLS Failover port:

□	Port	Protocol	Default Domain	Notes
□	5060	TCP	avayalab2.com	
□	5060	UDP	avayalab2.com	
□	5061	TLS	avayalab2.com	

The following screen shows the addition of the Communication Manager SIP Entity. In order for Session Manager to send SIP service provider traffic on a separate entity link to Communication Manager, it is necessary to create a separate SIP Entity for Communication.

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

- **Name:** Enter a descriptive name.
- **FQDN or IP Address:** Enter the FQDN or IP address of the Communication Manager SIP signaling interface.
- **Type:** Select **CM** for Communciation Manager.
- **Adaptation:** For the **Adaptation** field, you would select an Adaptation created in **Section 6.4**; however, no Adaptations were necessary for this compliance test.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Select the time zone for the location above.



SIP Entity Details

General

* Name:	cm62
* FQDN or IP Address:	205.3.3.200
Type:	CM
Notes:	
Adaptation:	
Location:	Enterprise
Time Zone:	America/Denver
Override Port & Transport with DNS SRV:	□
* SIP Timer B/F (in seconds):	4
Credential name:	
Call Detail Recording:	none

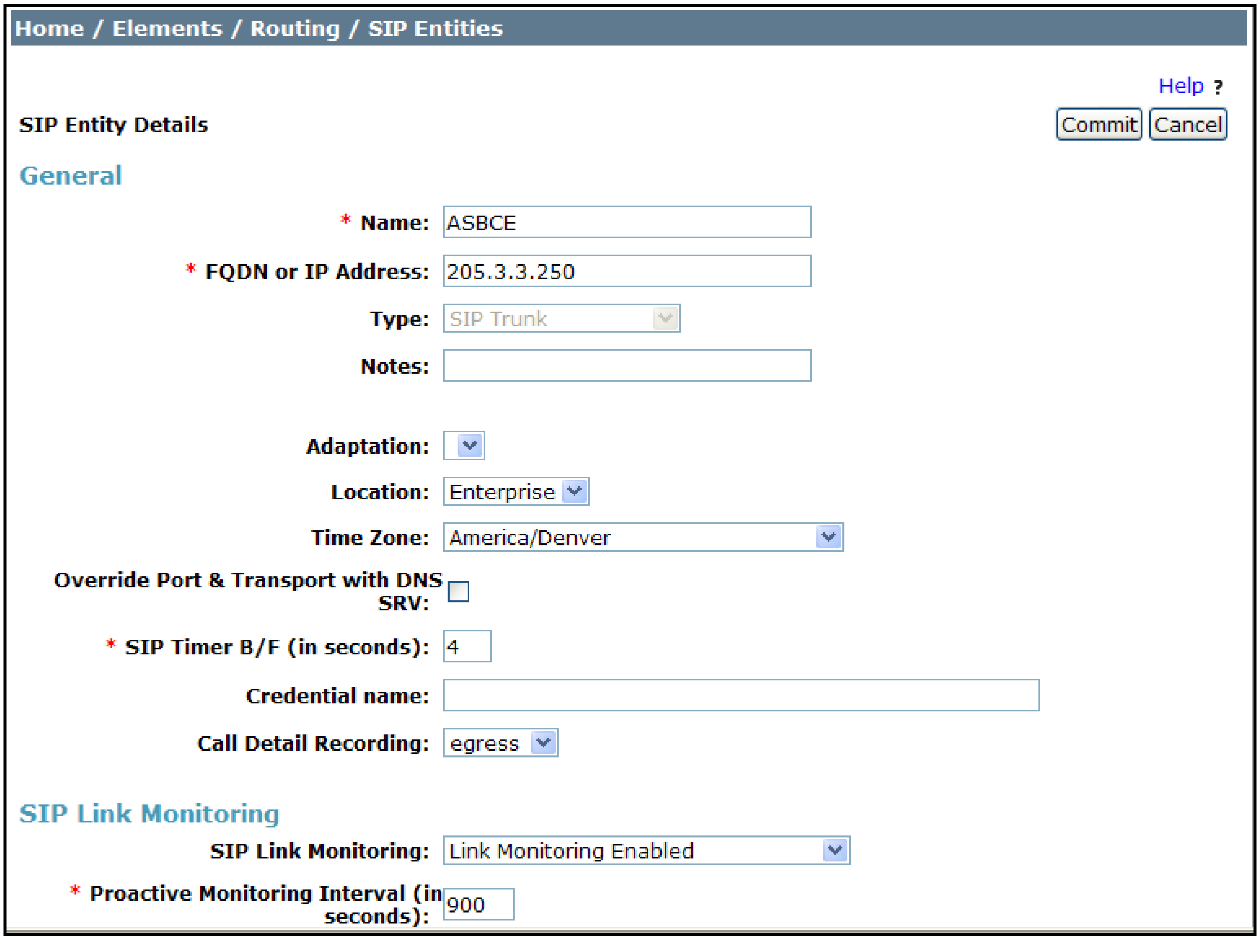
SIP Link Monitoring

SIP Link Monitoring:	Link Monitoring Enabled
* Proactive Monitoring Interval (in seconds):	900

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

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

- **Name:** Enter a descriptive name.
- **FQDN or IP Address:** This field is set to the IP address of the SBC's inside network interface (see **Figure 1**).
- **Type:** Select **Other** for the Avaya SBCE.
- **Adaptation:** For the **Adaptation** field, you would select an Adaptation created in **Section 6.4**; however, no Adaptations were necessary for this compliance test.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Select the time zone for the location above.



SIP Entity Details

General

* Name: ASBCE

* FQDN or IP Address: 205.3.3.250

Type: SIP Trunk

Notes:

Adaptation:

Location: Enterprise

Time Zone: America/Denver

Override Port & Transport with DNS SRV: □

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

Credential name:

Call Detail Recording: egress

SIP Link Monitoring

SIP Link Monitoring: Link Monitoring Enabled

* Proactive Monitoring Interval (in seconds): 900

6.6. Add Entity Links

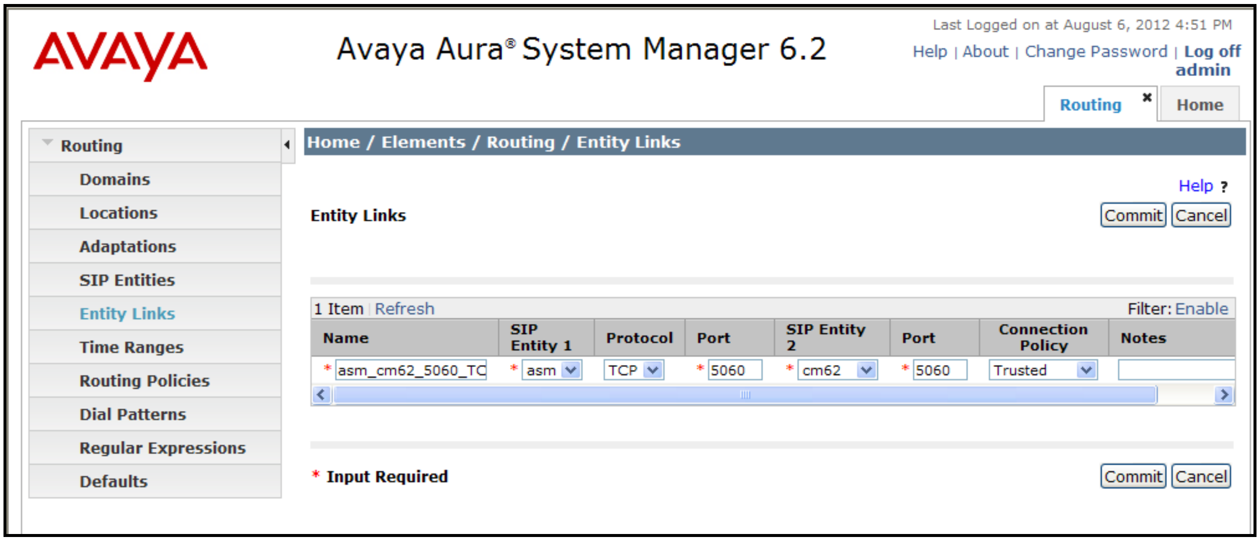
A SIP trunk between Session Manager and a telephony system is described by an Entity Link. For the compliance test, three Entity Links were created; one to Communication Manager, one to the Avaya SBCE and one to Avaya Modular Messaging. To add an Entity Link, navigate to **Routing → Entity Links** in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

- **Name:** Enter a descriptive name.
- **SIP Entity 1:** Select the Session Manager SIP Entity.
- **Protocol:** Select the transport protocol used for this link.
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end. For the Communication Manager, this must match the **Far-end Listen Port** defined on the Communication Manager signaling group in **Section 5.6**.
- **SIP Entity 2:** Select the name of the other system. For 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 5.6**.
- **Trusted:** Check this box. *Note: If this box is not checked, calls from the associated SIP Entity specified in **Section 6.5** will be challenged / denied.*

Click **Commit** to save.

The following screens illustrate the Entity Links to Communication Manager and the Avaya SBCE. It should be noted that in a customer environment the Entity Link to Communication Manager would normally use TLS. For the compliance test, TCP was used to aid in troubleshooting since the signaling traffic would not be encrypted. The protocol and ports defined here must match the values used on the Communication Manager signaling group form in **Section 5.6**.

Entity Link to Communication Manager:

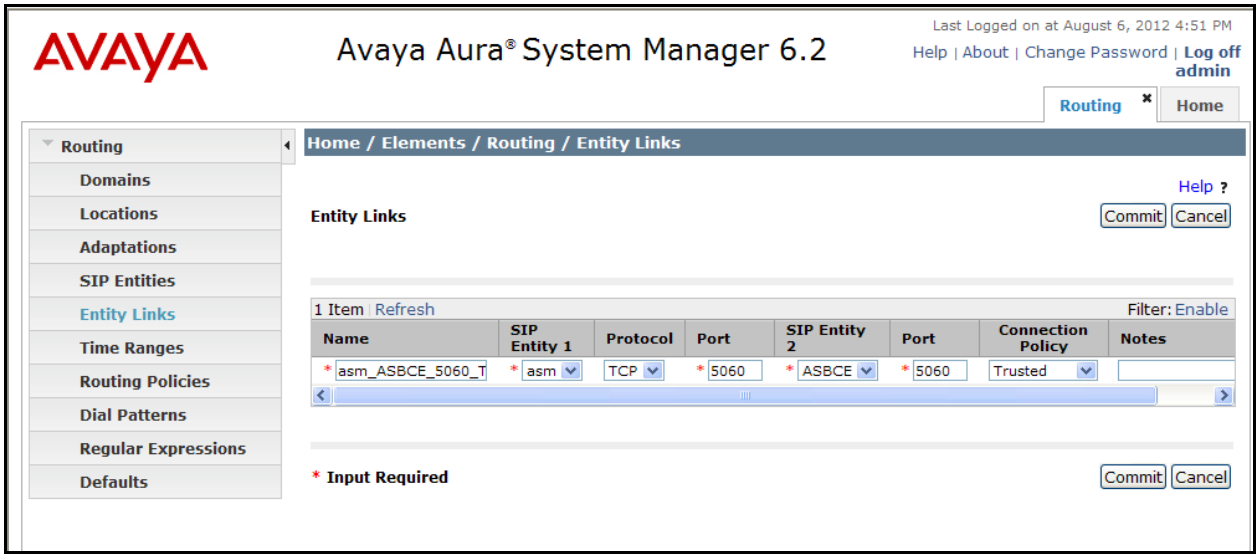


Entity Links

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* asm_cm62_5060_TC	* asm	TCP	* 5060	* cm62	* 5060	Trusted	

* Input Required

Entity Link to Avaya SBCE:



Entity Links

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* asm_ASBCE_5060_T	* asm	TCP	* 5060	* ASBCE	* 5060	Trusted	

* Input Required

6.7. Add Routing Policies

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

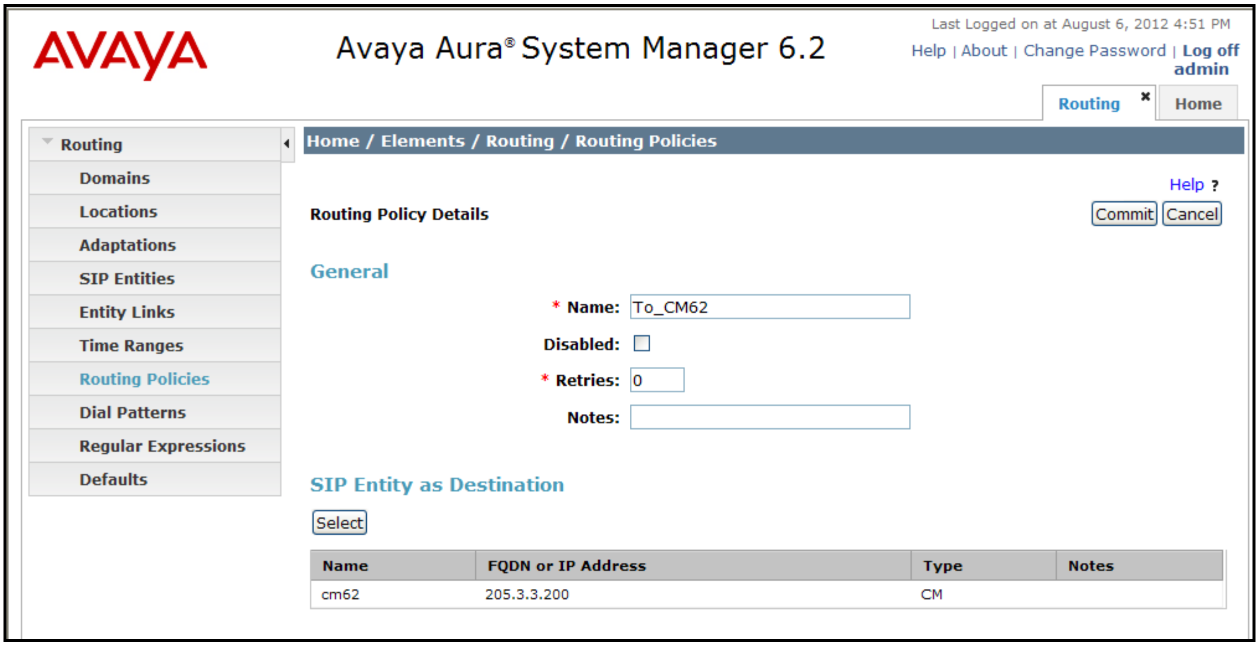
In the **General** section, enter the following values. Use default values for all remaining fields:

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

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

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

Routing Policy for Communication Manager:



Routing Policy Details

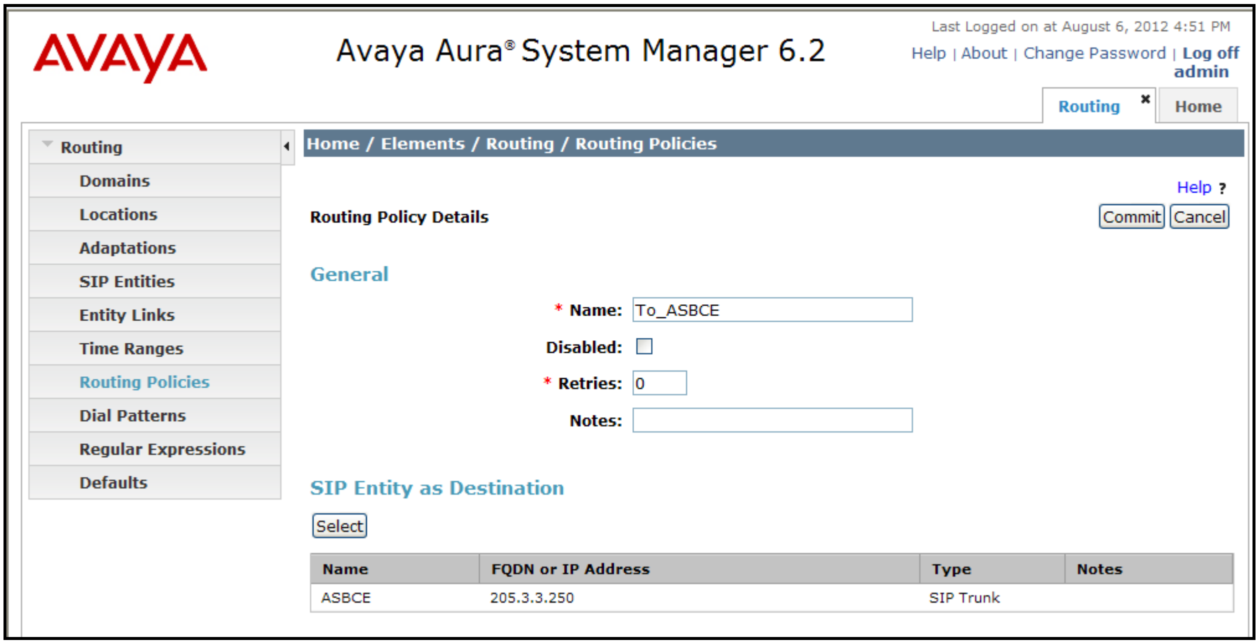
General

* Name:	To_CM62
Disabled:	☐
* Retries:	0
Notes:	

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
cm62	205.3.3.200	CM	

Routing Policy for Avaya SBCE:



Routing Policy Details

General

* Name:	To_ASBCE
Disabled:	□
* Retries:	0
Notes:	

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
ASBCE	205.3.3.250	SIP Trunk	

6.8. Add Dial Patterns

Dial Patterns are needed to route calls through Session Manager. For the compliance test, Dial Patterns were needed to route calls from Communication Manager to Sprint and vice versa. Dial Patterns define which Routing 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:

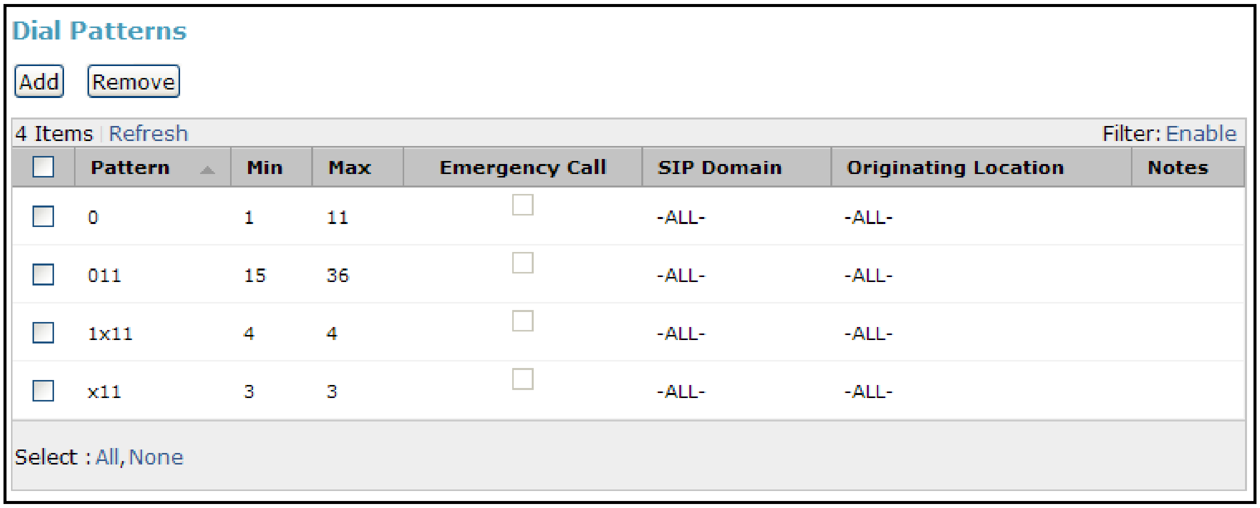
- **Pattern:** Enter a dial string that will be matched against the Request-URI of the call.
- **Min:** Enter a minimum length used in the match criteria.
- **Max:** Enter a maximum length used in the match criteria.
- **SIP Domain:** Enter the destination SIP Domain used in the match criteria.
- **Notes:**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

Examples of the Dial Patterns used for the compliance test are shown below, one for outbound calls from the enterprise to the PSTN and one for inbound calls from the PSTN to the enterprise. Other dial patterns were similarly defined.

The first example shows the Dial Patterns for outbound calls that belong to the Routing Policy **To_ASBCE** as defined in **Section 6.7**. These Dial Patterns cover Operator and Operator Assisted calls, International calls, and any 1x11 and x11 services, respectively. There is the option to define a Dial Pattern of ***x*** here to cover any and all possible Dial Patterns for outbound calls; however, for the compliance test a Regular Expression was used for this purpose which will be covered in the Regular Expression section that follows. The **SIP domain** was set to ***-ALL-*** since this Session Manager was not being shared in this environment, but could have been set specifically to ***avayalab2.com*** if necessary.



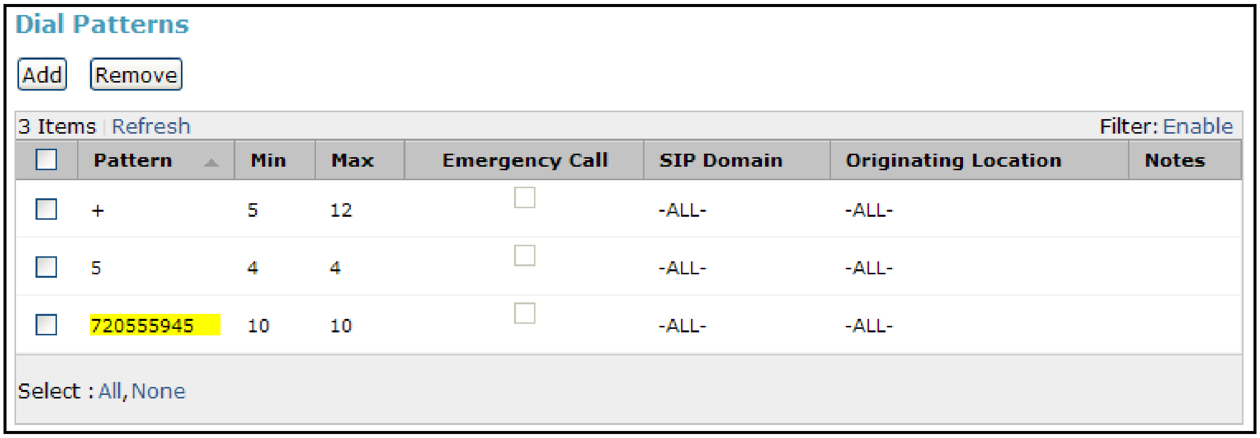
Dial Patterns

□	Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
□	0	1	11	□	-ALL-	-ALL-	
□	011	15	36	□	-ALL-	-ALL-	
□	1x11	4	4	□	-ALL-	-ALL-	
□	x11	3	3	□	-ALL-	-ALL-	

Note that the above Dial Pattern configuration did not restrict outbound calls to specific US area codes. In real deployments, appropriate restriction can be exercised (e.g., use Pattern 1908, 1303, etc. with 11 digits) per customer business policies.

Also note that *–**ALL**-* was selected for **Originating Location**. This selection was to accommodate certain off-net call forward scenarios where the inbound call was re-directed outbound back to the PSTN. For straight outbound calls, like 411 local directory, the enterprise Location ***Enterprise*** could have been selected.

The screen below shows the Dial Patterns for inbound calls that belong to the Routing Policy **To_CM62** as defined in **Section 6.7**. These Dial Patterns cover any e.164 numbering that carries the preceding + sign, the 4 digit extension range of 5xxx, and the DIDs assigned to the enterprise by Sprint which all begin with ***720555945***. The **SIP domain** was set to ***-ALL-*** since this Session Manager was not being shared in this environment, but could have been set specifically to ***avayalab2.com*** if necessary. **Originating location** was also set to ***-ALL-*** but could have been set to a more specific location, had one been defined for the Sprint network.



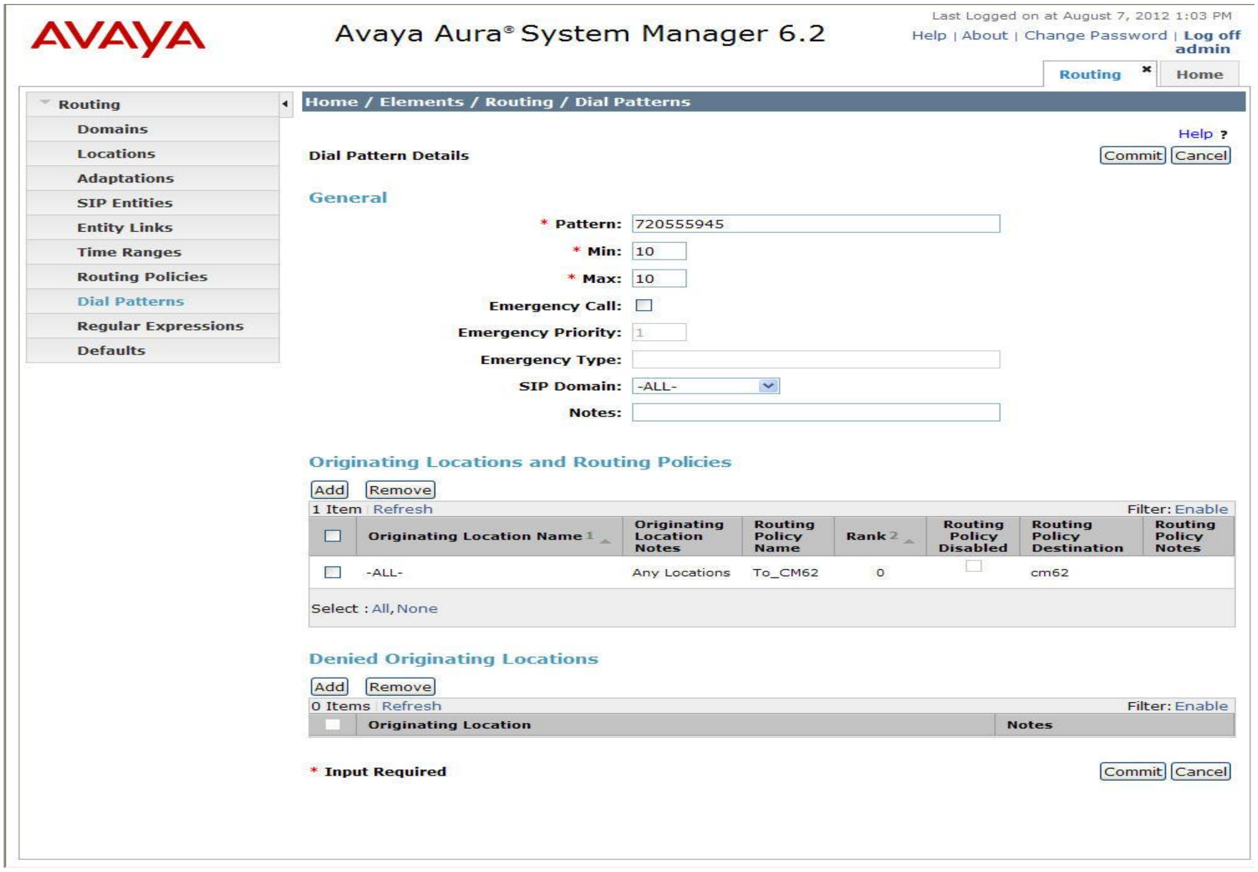
Dial Patterns

3 Items | Refresh — Filter: Enable

□	Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
□	+	5	12	□	-ALL-	-ALL-	
□	5	4	4	□	-ALL-	-ALL-	
□	720555945	10	10	□	-ALL-	-ALL-	

Select : All, None

Below shows an example of the **Dial Pattern Details** for the DIDs that were assigned by Sprint. Inbound 10-digit numbers that start with ***720555945*** will use Routing Policy ***To_CM62*** as defined in **Section 6.7**. This Dial Pattern matches the DID numbers assigned to the enterprise by Sprint.



Dial Pattern Details

General

* Pattern:	720555945
* Min:	10
* Max:	10
Emergency Call:	□
Emergency Priority:	1
Emergency Type:	
SIP Domain:	-ALL-
Notes:	

Originating Locations and Routing Policies

1 Item | Refresh — Filter: Enable

□	Originating Location Name 1	Originating Location Notes	Routing Policy Name	Rank 2	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
□	-ALL-	Any Locations	To_CM62	0	□	cm62	

Select : All, None

Denied Originating Locations

0 Items | Refresh — Filter: Enable

□	Originating Location	Notes

* Input Required

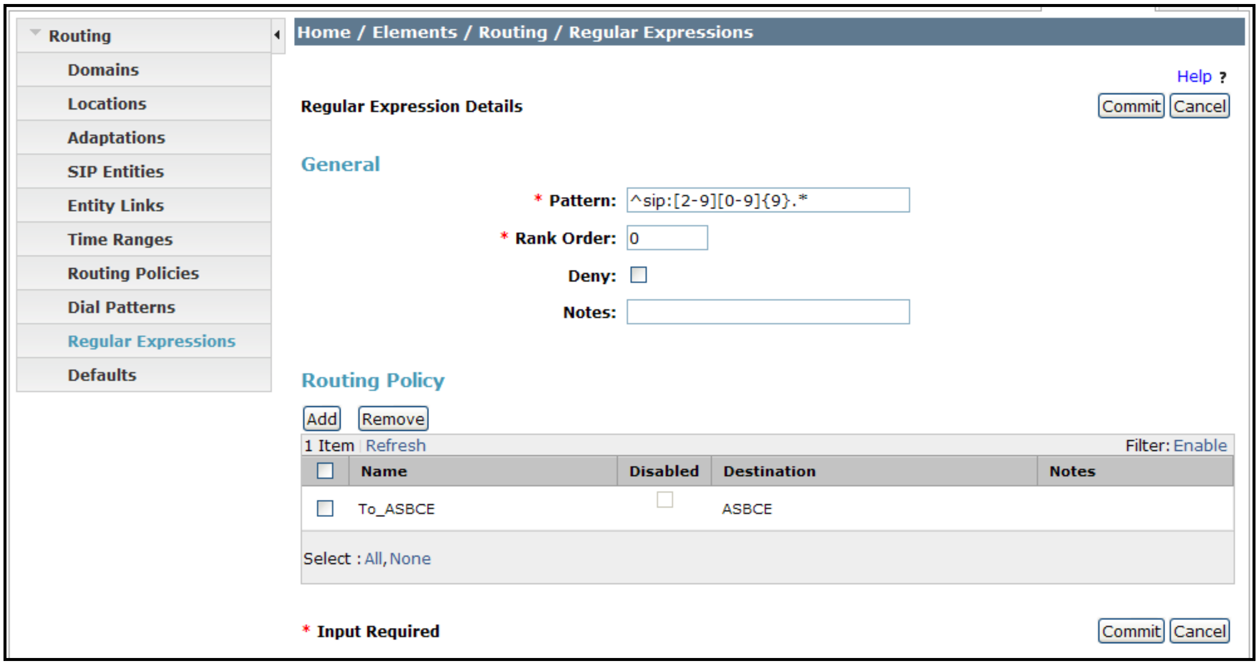
6.9. Regular Expressions

A **Regular Expression** was created to route any possible combination of outbound dialed digits to the Sprint SIP trunk service via the Avaya SBCE. This regular expression covers all possible 10 digit dialed strings starting with any digit except ***0*** or ***1***.

The screen below shows the **Regular Expression Details** screen and the values that were used to create the Regular Expression:

- **Pattern:** Enter a regular expression that covers the requirements.
- **Rank Order:** Priority of the pattern. Lower numbers mean higher priority. For the compliance test, ***0*** was used as it is the highest priority possible.
- **Routing Policy:** Enter the routing policy that should be associated with the regular expression. For the compliance test, the ***To_ASBCE*** **Routing Policy** was chosen since this regular expression covers outbound calls.

Click on **Commit** when finished.



Regular Expression Details

General

* Pattern: ^sip:[2-9][0-9]{9}.*

* Rank Order: 0

Deny: □

Notes:

Routing Policy

□	Name	Disabled	Destination	Notes
□	To_ASBCE	□	ASBCE	

* Input Required

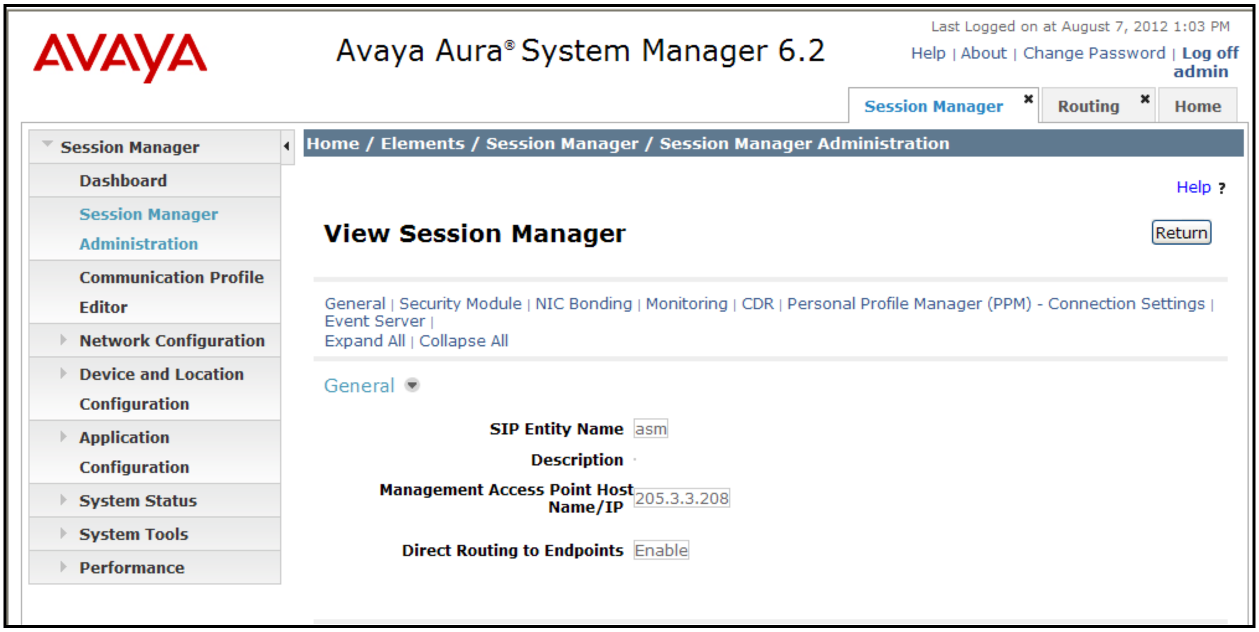
6.10. Add/View Session Manager

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

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

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

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


AVAYA
Avaya Aura® System Manager 6.2
Last Logged on at August 7, 2012 1:03 PM
Help | About | Change Password | Log off admin
Session Manager
Routing
Home
Session Manager
Dashboard
Session Manager Administration
Communication Profile Editor
Network Configuration
Device and Location Configuration
Application Configuration
System Status
System Tools
Performance
Home / Elements / Session Manager / Session Manager Administration
Help ?
Return
General | Security Module | NIC Bonding | Monitoring | CDR | Personal Profile Manager (PPM) - Connection Settings | Event Server |
Expand All | Collapse All

View Session Manager

General

SIP Entity Name asm

Description

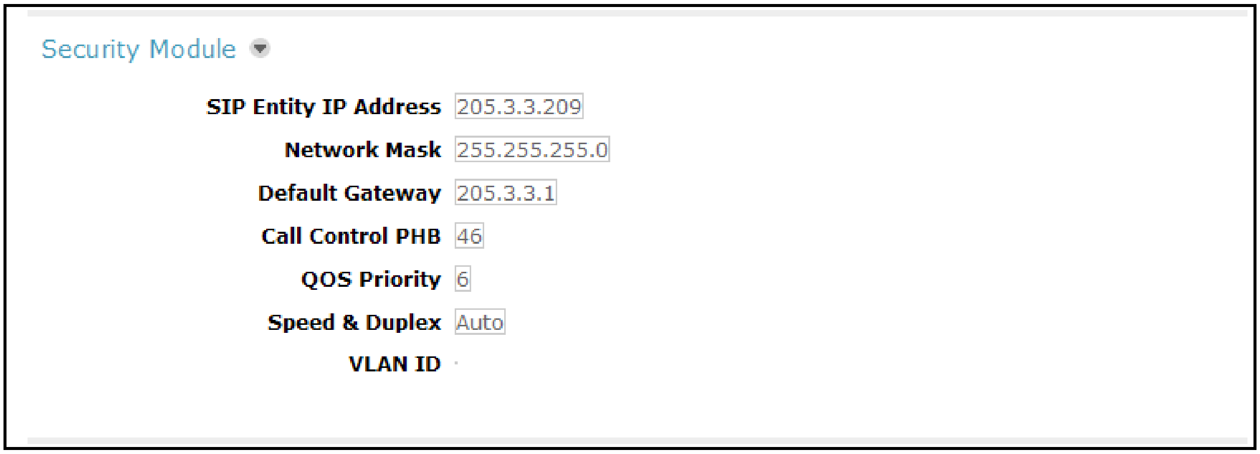
Management Access Point Host Name/IP 205.3.3.208

Direct Routing to Endpoints Enable

In the **Security Module** section, the following values should be filled in automatically since these parameters are generally setup at the time of installation. If not, use the values below:

- **SIP Entity IP Address:** Enter the IP address of the 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, or values appropriate for the specific customer, for the remaining fields. Click **Save** (not shown) to add this Session Manager. The screen below shows the Session Manager values used for the compliance test.



Security Module

SIP Entity IP Address	205.3.3.209
Network Mask	255.255.255.0
Default Gateway	205.3.3.1
Call Control PHB	46
QOS Priority	6
Speed & Duplex	Auto
VLAN ID	

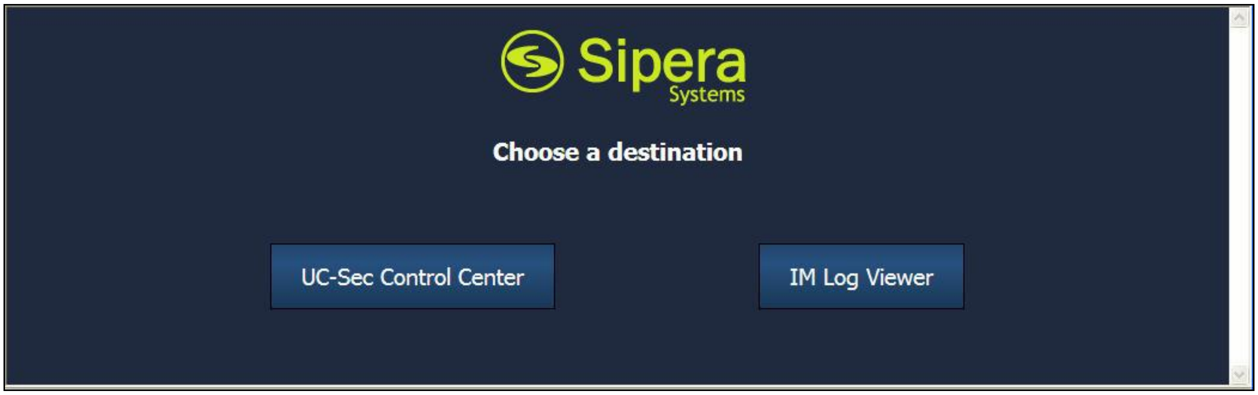
7. Configure Avaya Session Border Controller for Enterprise

In the sample configuration, an Avaya SBCE is used as the edge device between the Avaya CPE and the Sprint SIP Trunking service.

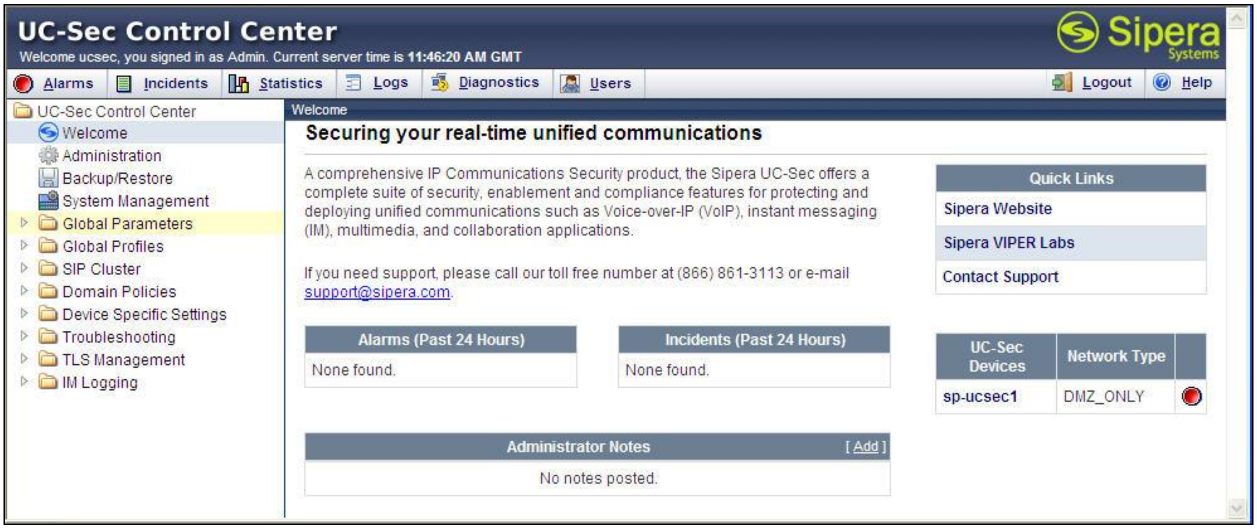
These Application Notes assume that the installation of the SBC and the assignment of a management IP Address have already been completed.

7.1. Access Management Interface

Use a WEB browser to access the web management interface by entering URL https://<ip-addr>, where <ip-addr> is the management LAN IP address assigned during installation. Select **UC-Sec Control Center** on the displayed web page, and log in using proper login credentials (not shown).



Once logged in, a Welcome screen will be presented. The following image illustrates the menu items available on the left-side of the UC-Sec Control Center screen.



Welcome ucsec, you signed in as Admin. Current server time is **11:46:20 AM GMT**

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.

Quick Links
Sipera Website
Sipera VIPER Labs
Contact Support

Alarms (Past 24 Hours)
None found.

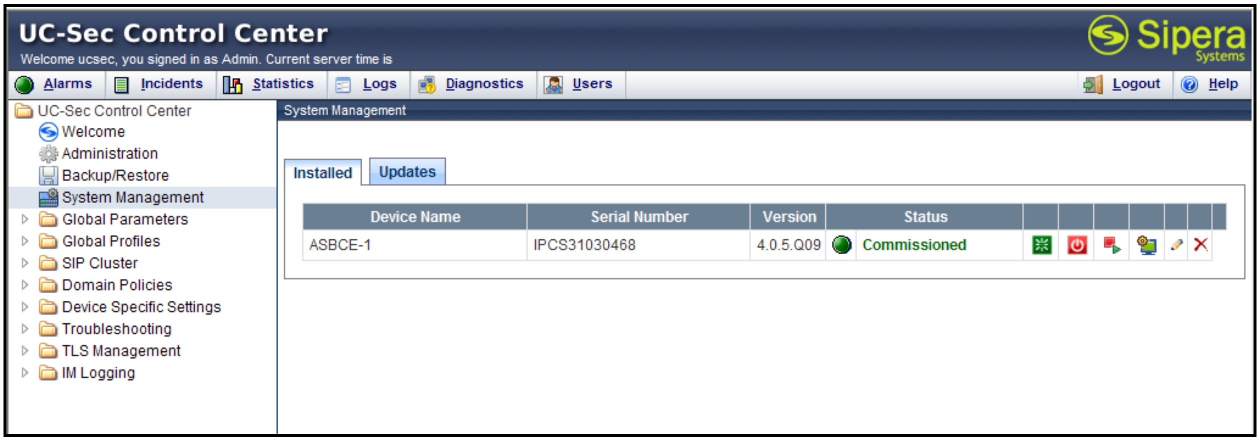
Incidents (Past 24 Hours)
None found.

UC-Sec Devices	Network Type	
sp-ucsec1	DMZ_ONLY	●

Administrator Notes [Add]
No notes posted.

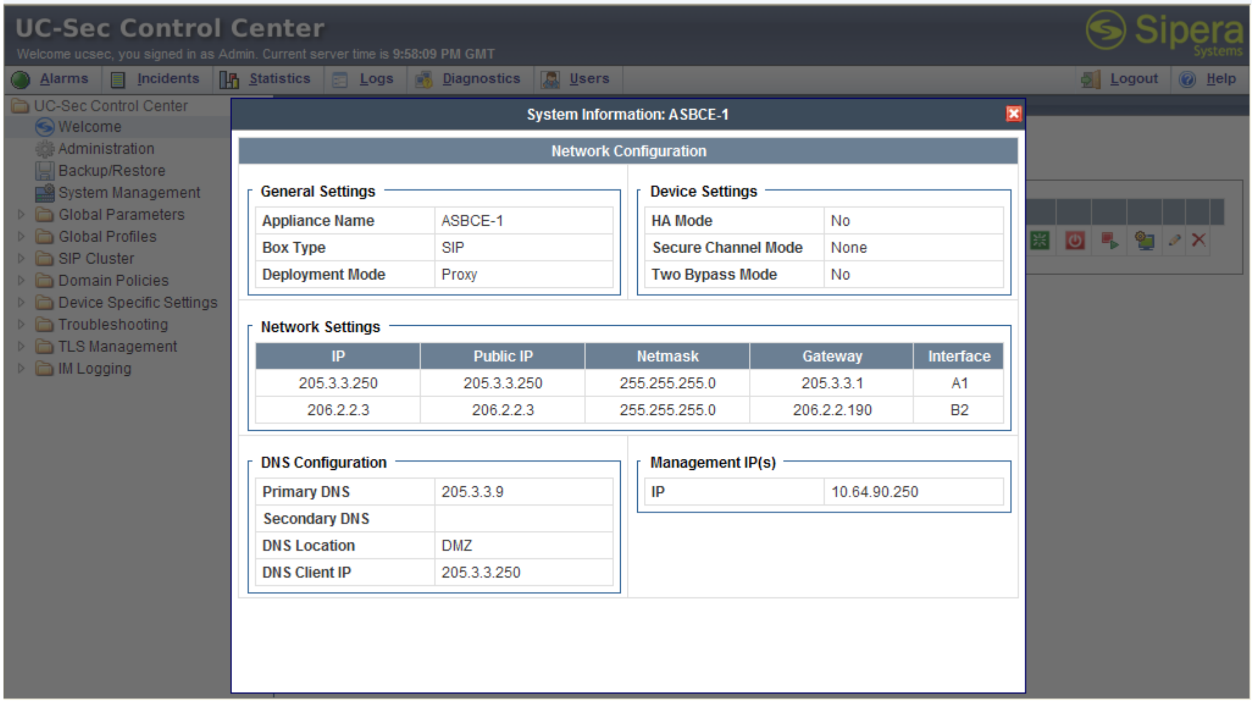
7.2. System Status

Navigate to **UC-Sec Control Center → System Management**. A list of installed devices is shown in the right pane. For the sample configuration, a single device named ***ASBCE-1*** is shown. Device **Status** "Commissioned" should be displayed as shown below.



Device Name	Serial Number	Version	Status
ASBCE-1	IPCS31030468	4.0.5.Q09	● Commissioned

To view the network information of this device, which was assigned during installation, click the **View Config** icon button (the third icon from the right). A **Network Configuration** window is displayed as shown below. Note that the A1 and B2 interface IP addresses correspond to the inside and outside interfaces, respectively, for the Avaya SBCE as shown in **Figure 1**.



System Information: ASBCE-1

Network Configuration

General Settings

Appliance Name	ASBCE-1
Box Type	SIP
Deployment Mode	Proxy

Device Settings

HA Mode	No
Secure Channel Mode	None
Two Bypass Mode	No

Network Settings

IP	Public IP	Netmask	Gateway	Interface
205.3.3.250	205.3.3.250	255.255.255.0	205.3.3.1	A1
206.2.2.3	206.2.2.3	255.255.255.0	206.2.2.190	B2

DNS Configuration

Primary DNS	205.3.3.9
Secondary DNS	
DNS Location	DMZ
DNS Client IP	205.3.3.250

Management IP(s)

IP	10.64.90.250

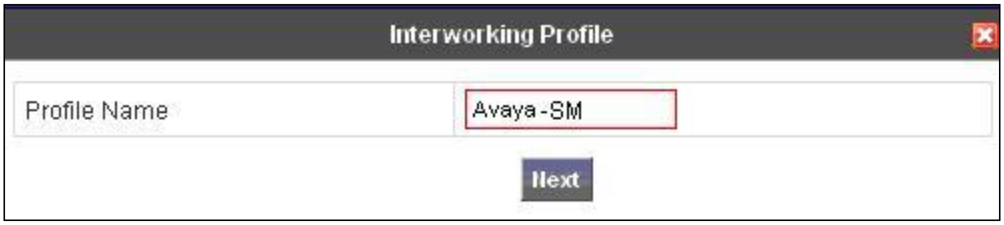
7.3. Global Profiles – Server Interworking

Server interworking is defined for each server connected to the Avaya SBCE. For the compliance test, the Sprint SIP trunk network-edge SBC serves as the Trunk Server and Session Manager serves as the Call Server.

Navigate to **Global Profiles → Server Interworking** from the left-side menu (not shown) to configure Server Interworking profiles.

7.3.1. Server Interworking: Avaya-SM

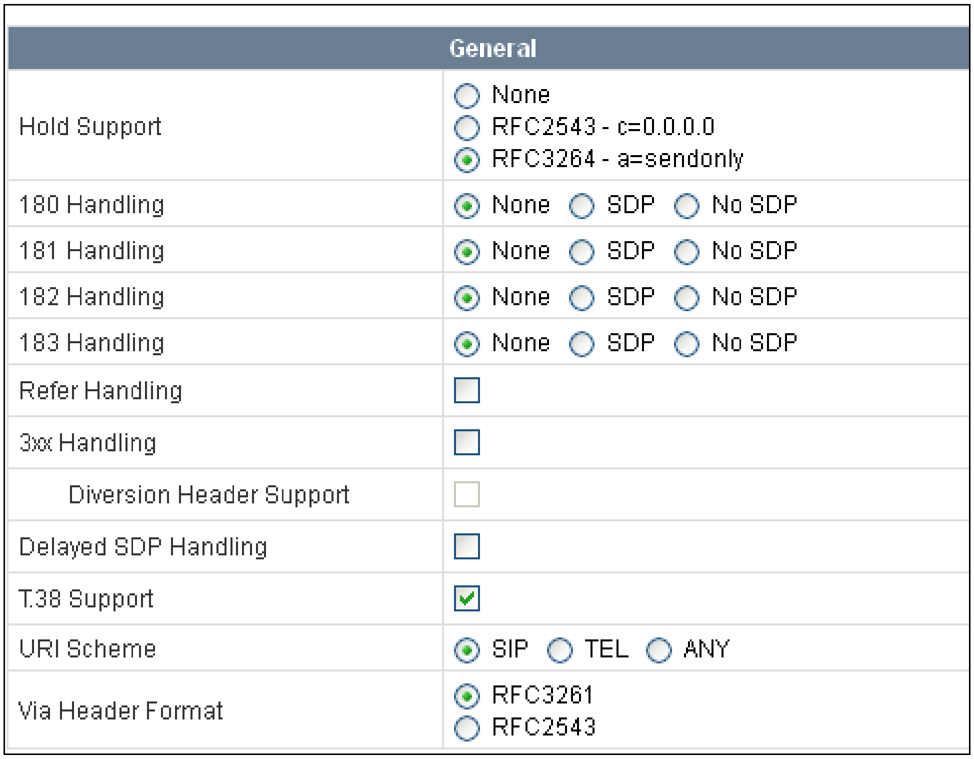
Click the **Add Profile** button (not shown) to add a new profile or select an existing Server Interworking profile to edit. If adding a profile, a screen such as the following is displayed. Enter an appropriate **Profile Name** such as ***Avaya-SM*** shown below. Click **Next**.



Interworking Profile

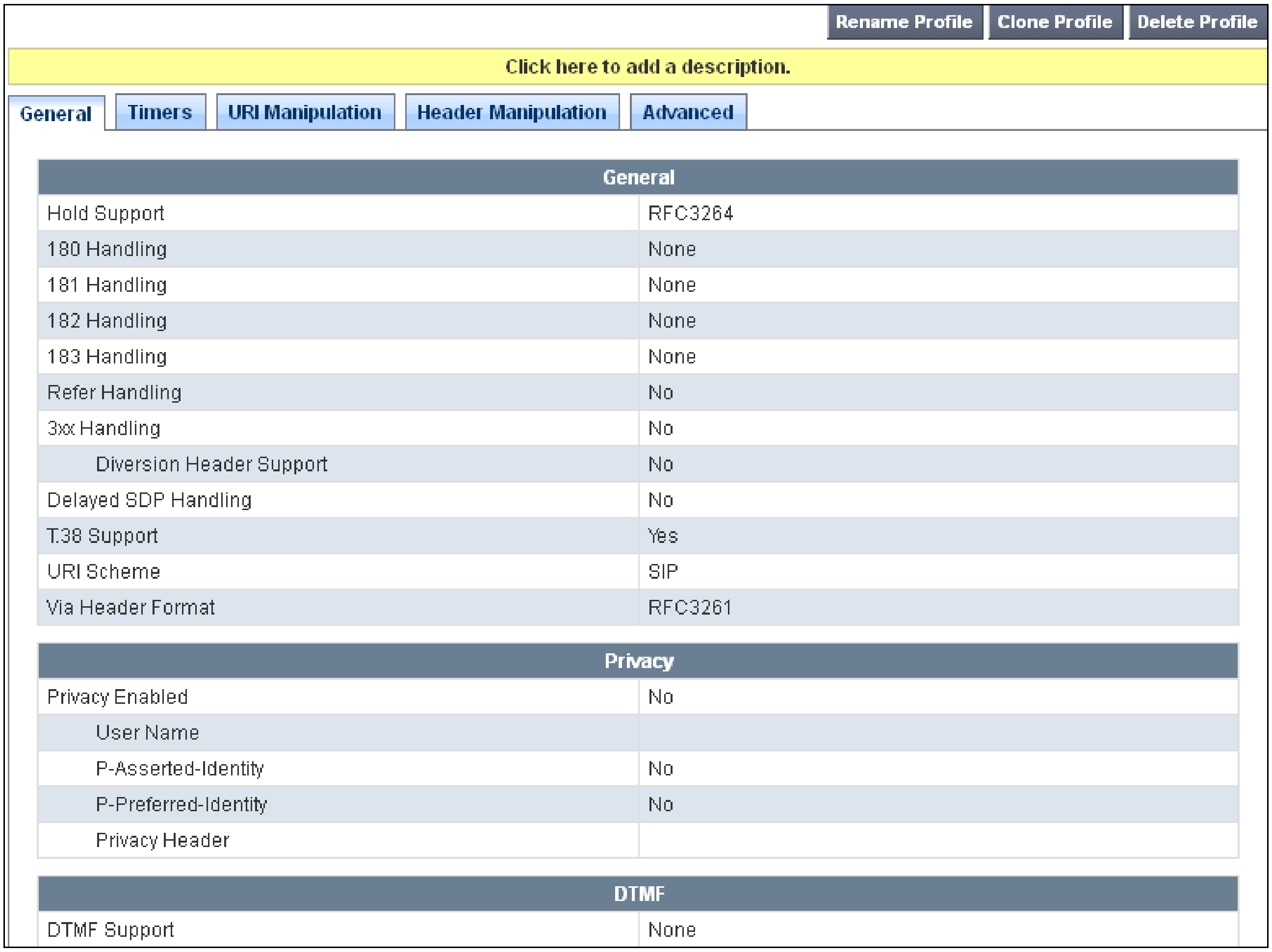
Profile Name	Avaya-SM

The following screens illustrate the **General** parameters used in the sample configuration for the Interworking Profile named ***Avaya-SM***. Most parameters retain default values. In the sample configuration, **T.38 Support** was checked to enable T.38 faxing, and **Hold Support** was set for ***RFC3264***.



General	
Hold Support	○ None ○ RFC2543 - c=0.0.0.0 ● RFC3264 - a=sendonly
180 Handling	● None ○ SDP ○ No SDP
181 Handling	● None ○ SDP ○ No SDP
182 Handling	● None ○ SDP ○ No SDP
183 Handling	● None ○ SDP ○ No SDP
Refer Handling	□
3xx Handling	□
Diversion Header Support	□
Delayed SDP Handling	□
T.38 Support	☑
URI Scheme	● SIP ○ TEL ○ ANY
Via Header Format	● RFC3261 ○ RFC2543

Click **Next** (not shown) to advance to configure **Privacy** and **DTMF** general parameters, which can retain default values. The following screen shows the complete **General** tab used in the sample configuration for the interworking profile named ***Avaya-SM***.



General	
Hold Support	RFC3264
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	No
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
T.38 Support	Yes
URI Scheme	SIP
Via Header Format	RFC3261

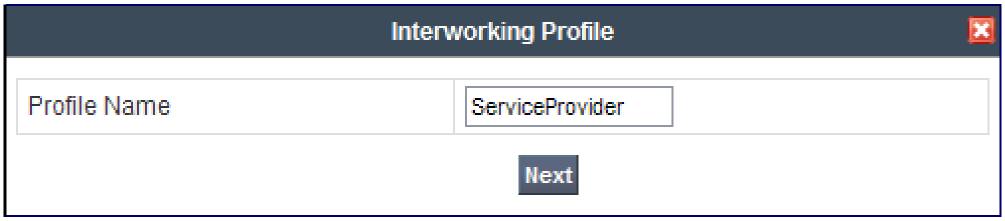
Privacy	
Privacy Enabled	No
User Name	
P-Asserted-Identity	No
P-Preferred-Identity	No
Privacy Header	

DTMF	
DTMF Support	None

The parameters in all other tabs may retain default settings.

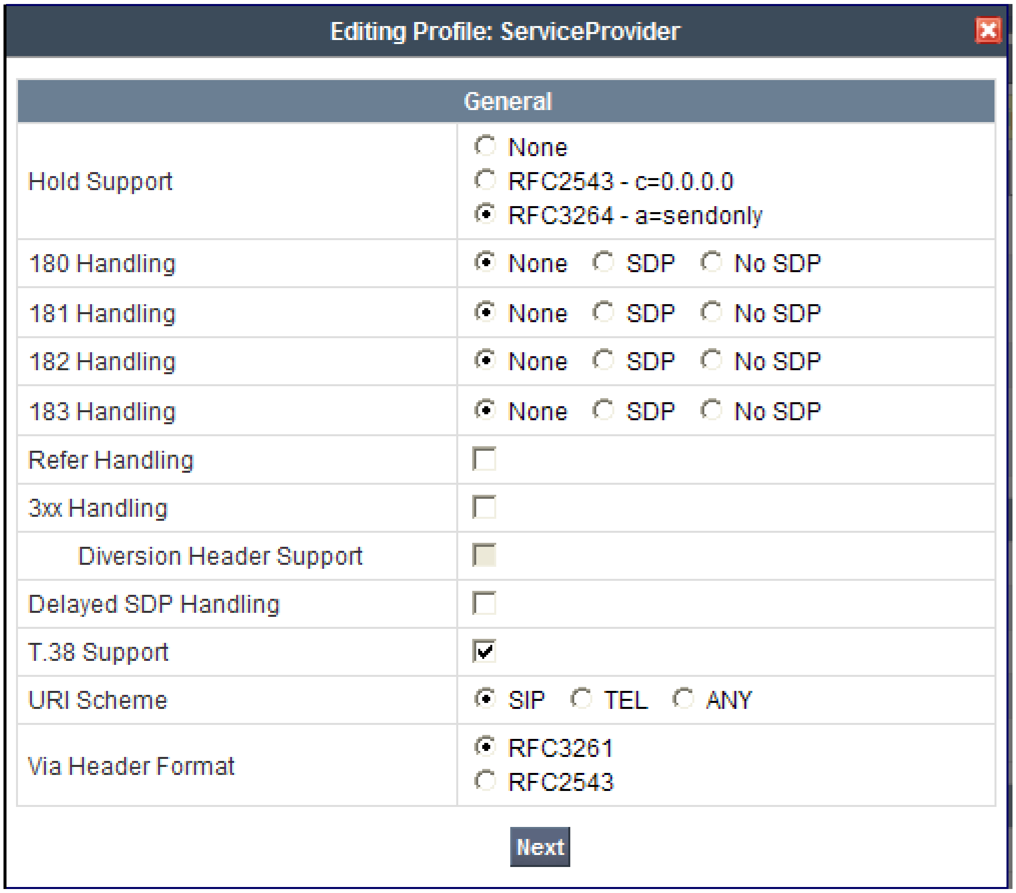
7.3.2. Server Interworking: ServiceProvider

A second Server Interworking profile named ***ServiceProvider*** was similarly created. Click the **Add Profile** button (not shown) to add a new profile or select an existing Server Interworking profile to edit. If adding a profile, a screen such as the following is displayed. Enter an appropriate **Profile Name** such as ***ServiceProvider*** as shown below. Click **Next**.



Profile Name	ServiceProvider

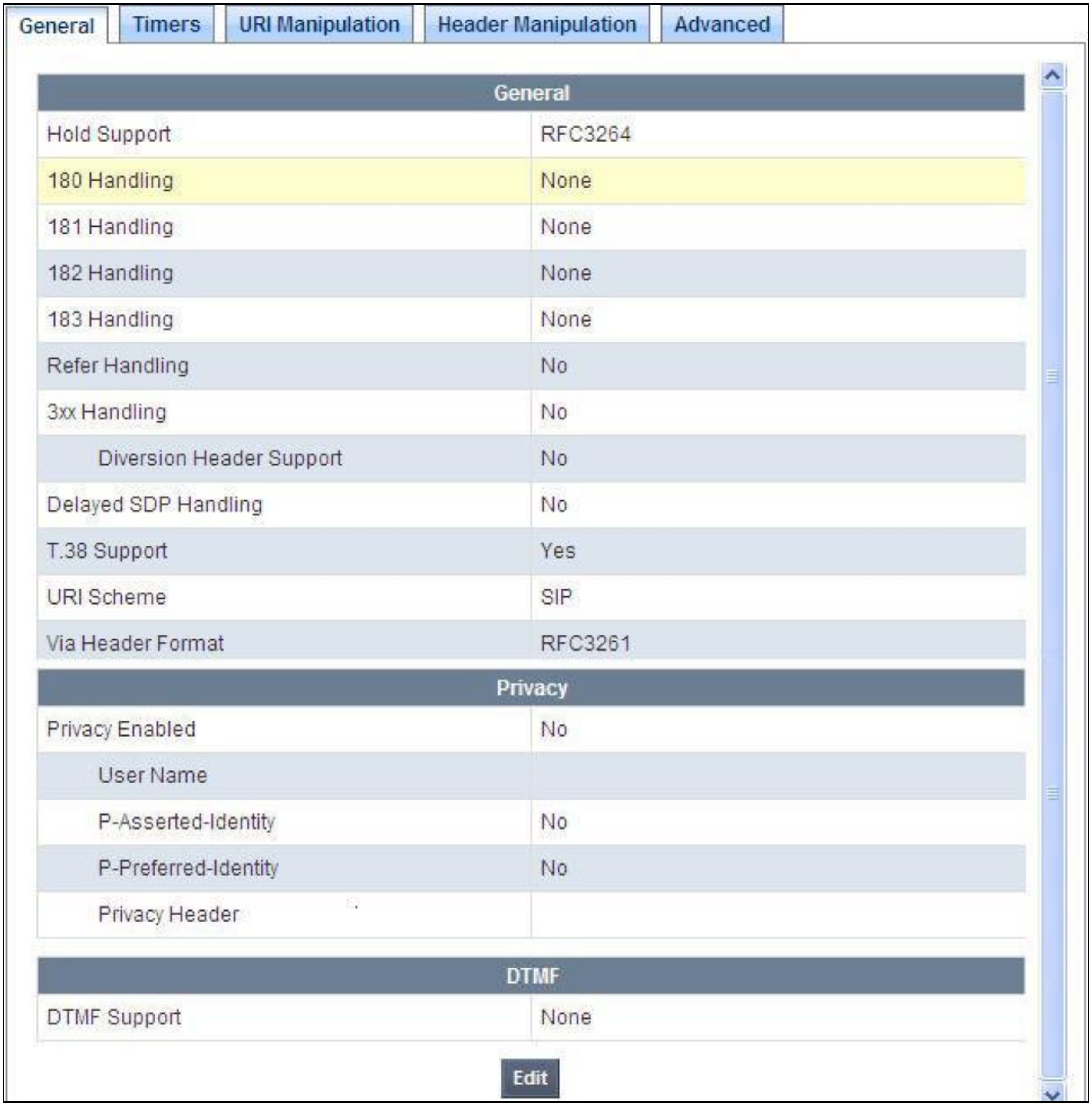
The following screen illustrates the **General** parameters used in the sample configuration for the ***ServiceProvider*** Server Interworking profile. In the sample configuration, **T.38 Support** was checked and **Hold Support** was set for ***RFC3264***. Other parameters can retain their default values.



General	
Hold Support	○ None ○ RFC2543 - c=0.0.0.0 ● RFC3264 - a=sendonly
180 Handling	● None ○ SDP ○ No SDP
181 Handling	● None ○ SDP ○ No SDP
182 Handling	● None ○ SDP ○ No SDP
183 Handling	● None ○ SDP ○ No SDP
Refer Handling	□
3xx Handling	□
Diversion Header Support	□
Delayed SDP Handling	□
T.38 Support	☑
URI Scheme	● SIP ○ TEL ○ ANY
Via Header Format	● RFC3261 ○ RFC2543

Click **Next** to advance to configure **Privacy** and **DTMF** general parameters, which can retain default values.

The following screen shows the complete **General** tab used in the sample configuration for the interworking profile named ***ServiceProvider***.

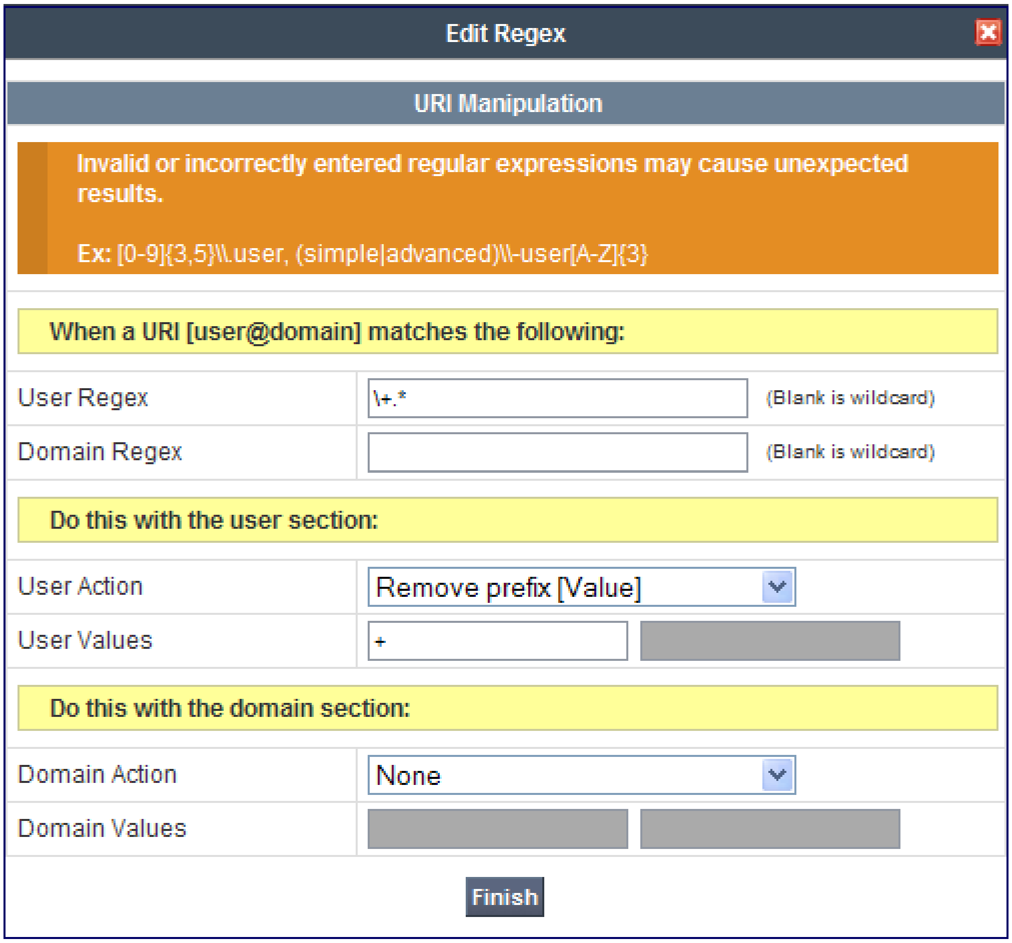


General	
Hold Support	RFC3264
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	No
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
T.38 Support	Yes
URI Scheme	SIP
Via Header Format	RFC3261
Privacy	
Privacy Enabled	No
User Name	
P-Asserted-Identity	No
P-Preferred-Identity	No
Privacy Header	
DTMF	
DTMF Support	None

Next go to the **URI Manipulation** tab and click on ***Add Regex***. The screen below shows the regex values used for the compliance test. Enter the following:

- **User Regex:** Enter **\+.***
- **User Action** Select ***Remove prefix [value]*** from the drop-down menu.
- **User Values** Enter **+**

Default values may be retained for all other fields. Click **Finish** when completed.



Edit Regex

URI Manipulation

Invalid or incorrectly entered regular expressions may cause unexpected results.

Ex: [0-9]{3,5}\\.user, (simple|advanced)\\-user[A-Z]{3}

When a URI [user@domain] matches the following:

User Regex	\+.*	(Blank is wildcard)
Domain Regex		(Blank is wildcard)

Do this with the user section:

User Action	Remove prefix [Value]
User Values	+

Do this with the domain section:

Domain Action	None
Domain Values	

The parameters in all other tabs may retain their default settings.

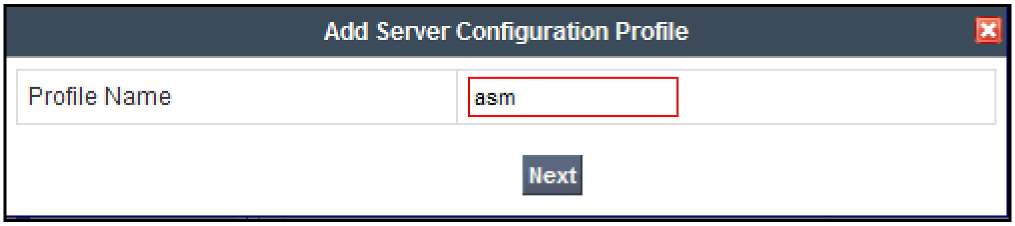
7.4. Global Profiles – Server Configuration

In the compliance test, the Sprint SIP trunk network-edge SBC is connected as the Trunk Server and the enterprise Session Manager is connected as the Call Server.

Navigate to **Global Profiles → Server Configuration** from the left-side menu to configure the two servers.

7.4.1. Server Configuration for Session Manager

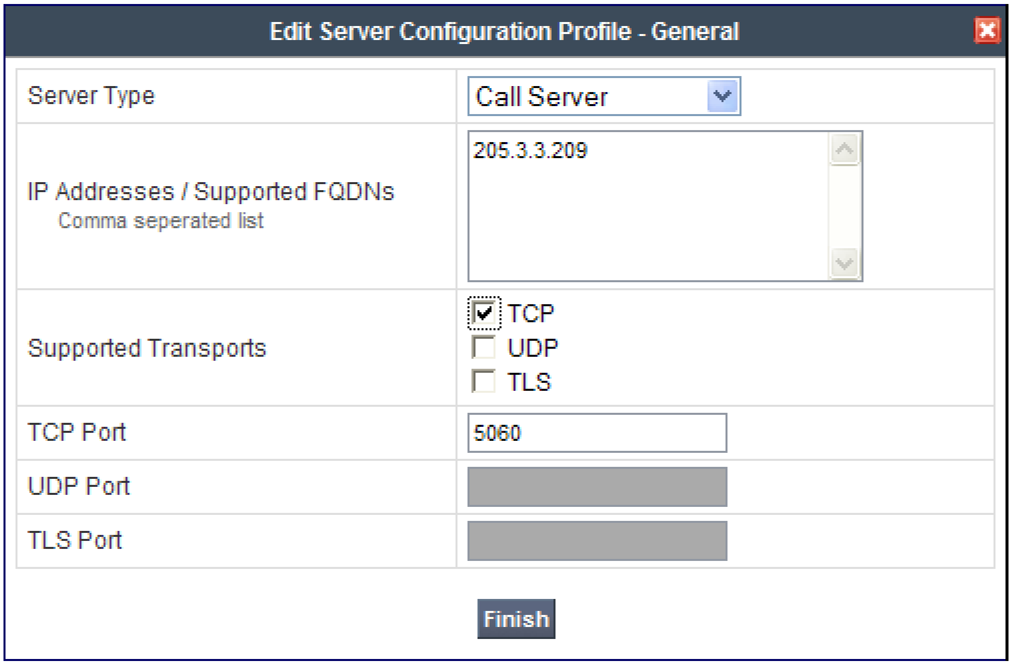
Click the **Add Profile** button (not shown) to add a new profile, or select an existing profile to edit. If adding a profile, a screen such as the following is displayed. Enter an appropriate Profile Name such as ***asm*** shown below. Click **Next**.



Add Server Configuration Profile

Profile Name	asm

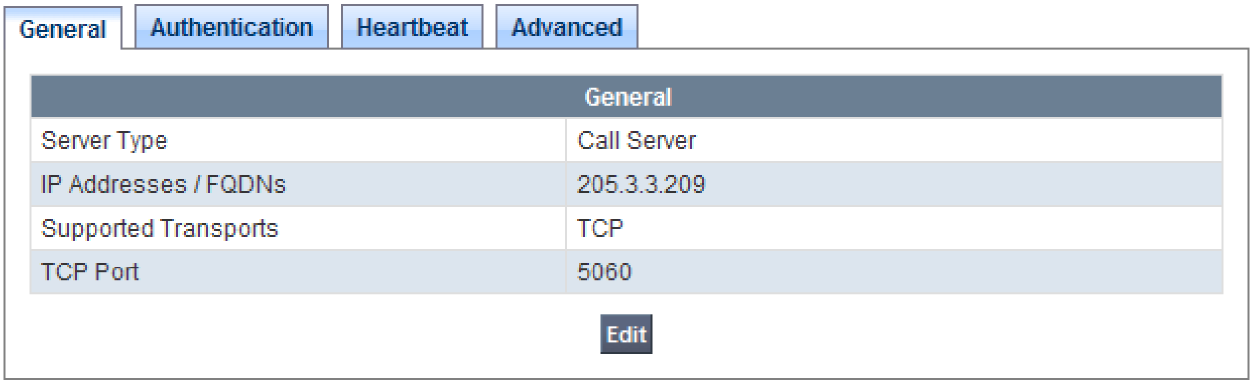
The following screens illustrate the Server Configuration with Profile name ***asm***. Select ***Call Server*** from the **Server Type** drop-down menu. In the **IP Addresses / Supported FQDNs** area, the IP Address of the Session Manager SIP signaling interface should be entered. In the **Supported 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 connecting to the SBC. If adding a new profile, click **Next** (not shown). If editing an existing profile, click **Finish**.



Edit Server Configuration Profile - General

Server Type	Call Server
IP Addresses / Supported FQDNs Comma seperated list	205.3.3.209
Supported Transports	☑ TCP □ UDP □ TLS
TCP Port	5060
UDP Port	
TLS Port	

Once configuration is completed, the **General** tab for the configured ***asm*** call server will appear as shown below:

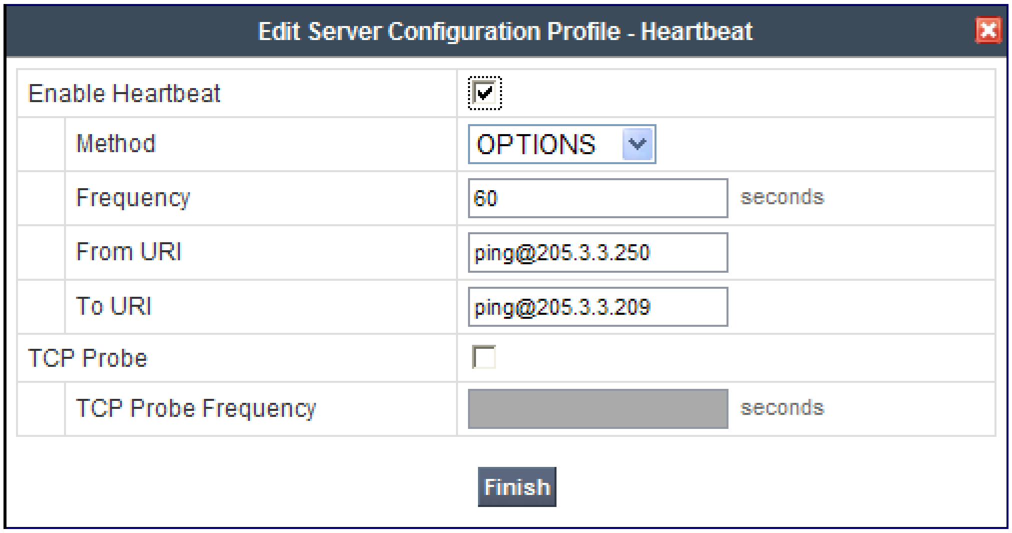


General	
Server Type	Call Server
IP Addresses / FQDNs	205.3.3.209
Supported Transports	TCP
TCP Port	5060

If adding the profile, click **Next** to accept default parameters for the **Authentication** tab, and advance to the **Heartbeat** area (not shown). If editing an existing profile, select the **Heartbeat** tab and click **Edit**.

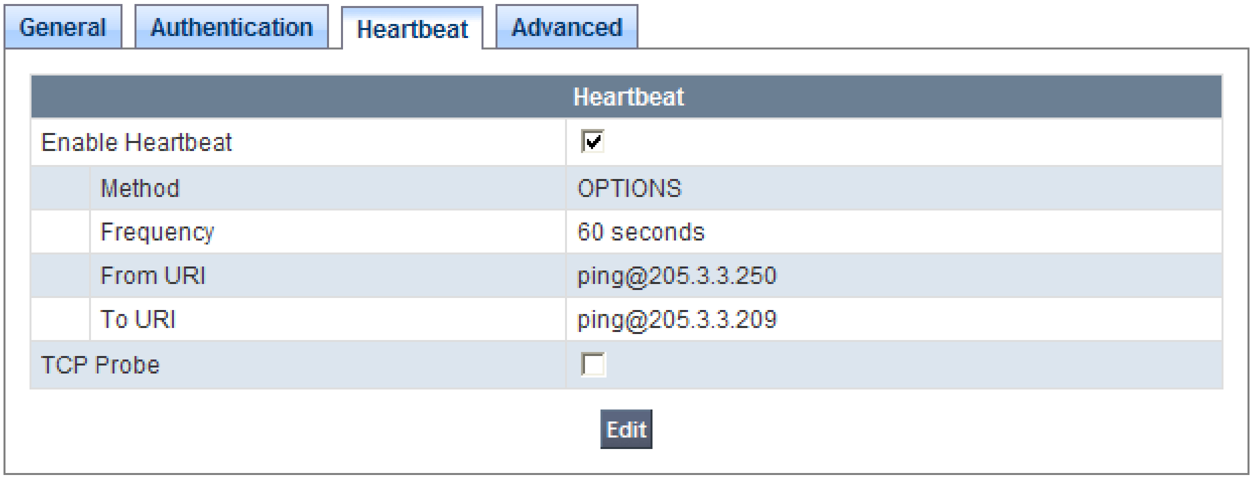
The SBC can be configured to source "heartbeats" in the form of SIP OPTIONS. In the sample configuration, with one connected Session Manager, this configuration is optional.

If SBC-sourced OPTIONS messages are desired, check the **Enable Heartbeat** box. Select ***OPTIONS*** from the **Method** drop-down menu. Select the desired frequency (in seconds) that the SBC will source OPTIONS to this server. The **From URI** and **To URI** may be filled in to configure easily identifiable URIs to appear in SIP OPTIONS sourced by the SBC toward Session Manager. If adding a new profile, click **Next** (not shown). If editing an existing profile, click **Finish**.



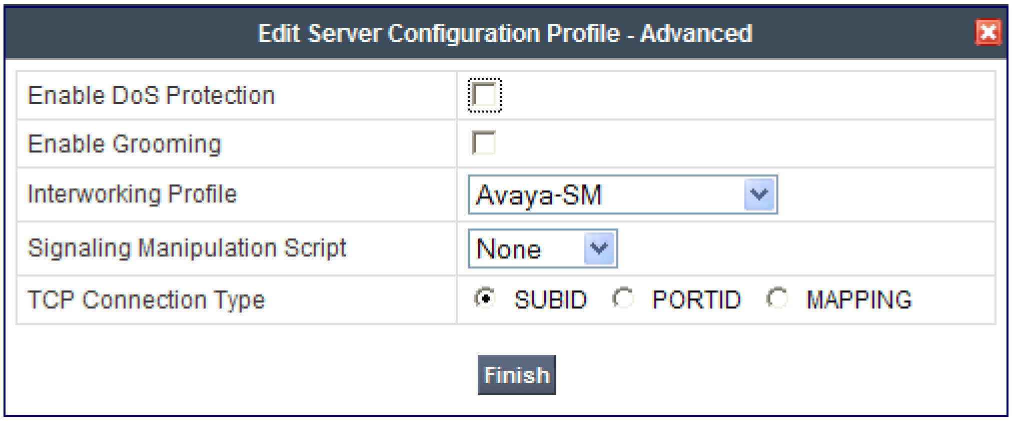
Enable Heartbeat	☑
Method	OPTIONS
Frequency	60 seconds
From URI	ping@205.3.3.250
To URI	ping@205.3.3.209
TCP Probe	☐
TCP Probe Frequency	seconds

If SBC sourced OPTIONS are configured, the **Heartbeat** tab for the ***asm*** server profile will appear as shown below:



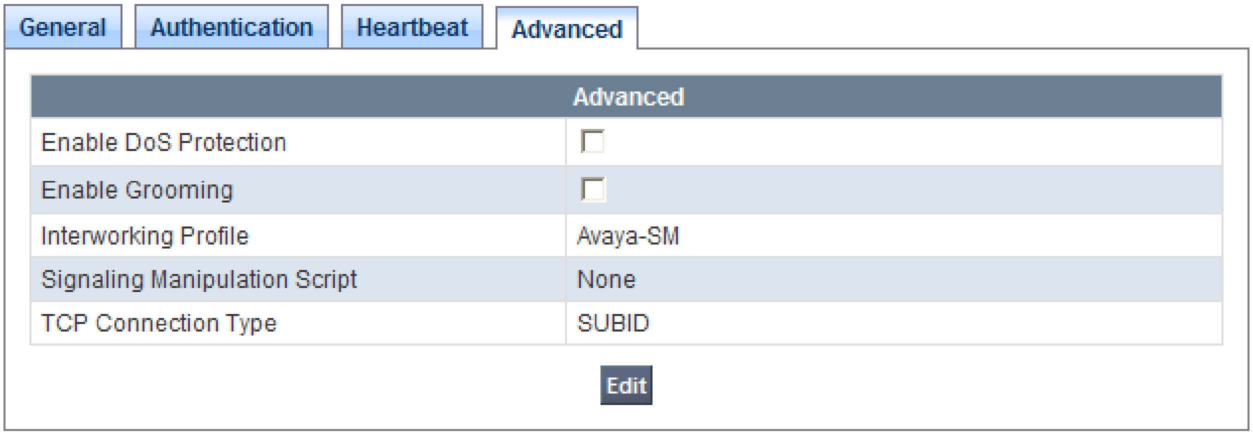
Heartbeat	
Enable Heartbeat	☑
Method	OPTIONS
Frequency	60 seconds
From URI	ping@205.3.3.250
To URI	ping@205.3.3.209
TCP Probe	□

If adding a profile, click **Next** to continue to the ***Advanced*** settings. If editing an existing profile, select the **Advanced** tab and click **Edit**. In the resultant screen, select the **Interworking Profile *Avaya-SM*** created in **Section 7.3.1**. Click **Finish**.



Enable DoS Protection	□
Enable Grooming	□
Interworking Profile	Avaya-SM
Signaling Manipulation Script	None
TCP Connection Type	● SUBID ○ PORTID ○ MAPPING

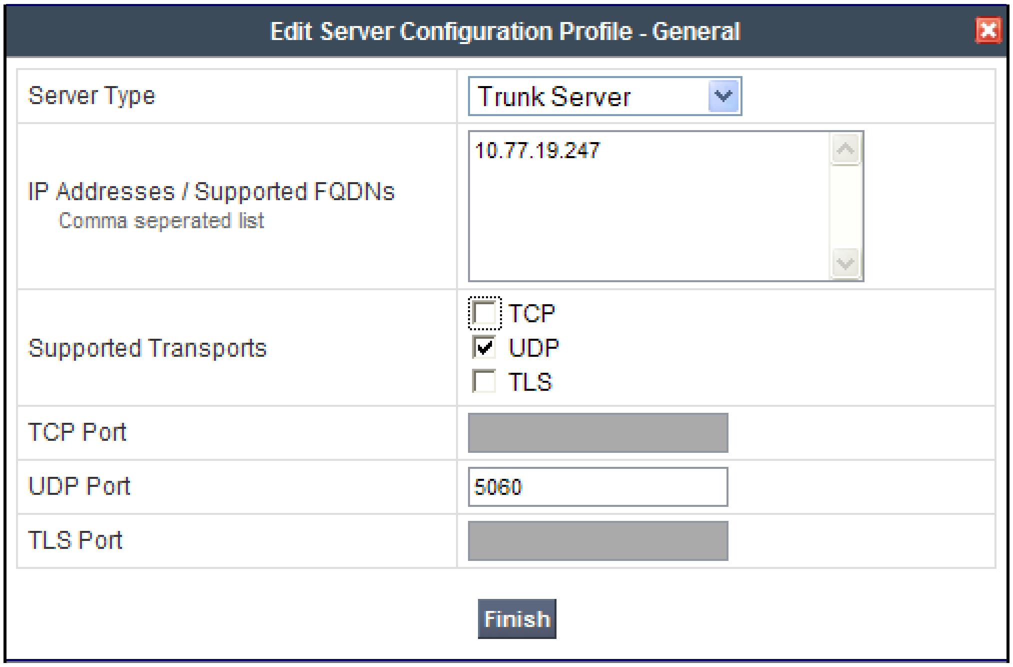
Once configuration is completed, the **Advanced** tab for the call server ***asm*** will appear as shown below.



Advanced	
Enable DoS Protection	□
Enable Grooming	□
Interworking Profile	Avaya-SM
Signaling Manipulation Script	None
TCP Connection Type	SUBID

7.4.2. Server Configuration for Sprint SIP Trunking

A second Server Configuration profile named ***Sprint-SIPT*** was similarly created. The following screens illustrate the ***Sprint-SIPT*** Server Configuration profile. In the **General** parameters, select ***Trunk Server*** from the **Server Type** drop-down menu. In the **IP Addresses / Supported FQDNs** area, the Sprint-provided SIP Trunking SBC IP Address is entered. In the **Supported Transports** area, ***UDP*** is selected, and the **UDP Port** is set to ***5060*** as specified by Sprint.



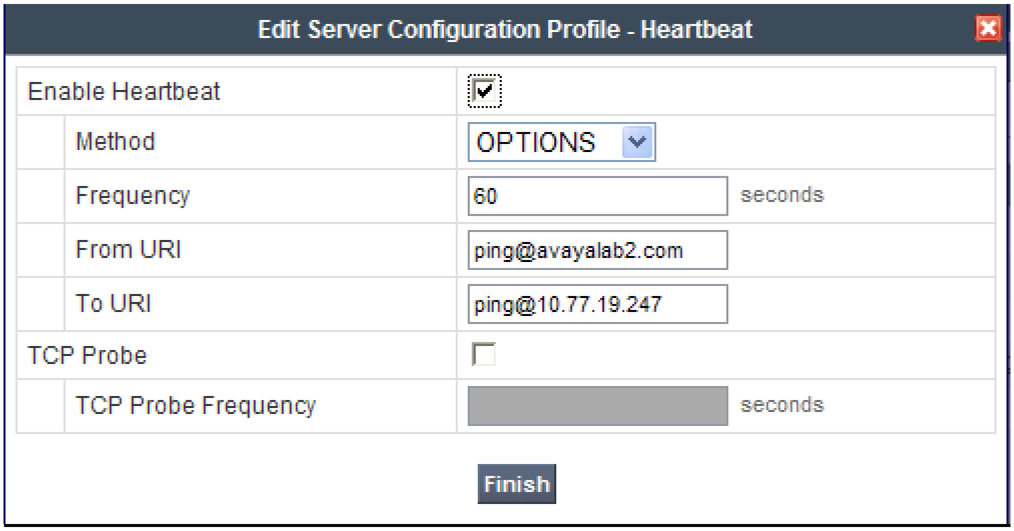
Edit Server Configuration Profile - General

Server Type	Trunk Server
IP Addresses / Supported FQDNs Comma seperated list	10.77.19.247
Supported Transports	□ TCP ☑ UDP □ TLS
TCP Port	
UDP Port	5060
TLS Port	

If adding the profile, click **Next** to accept default parameters for the **Authentication** tab, and advance to the **Heartbeat** area (not shown). If editing an existing profile, select the **Heartbeat** tab and click edit.

The SBC can be configured to source "heartbeats" in the form of SIP OPTIONS towards Sprint. This configuration is optional. Independent of whether the SBC is configured to source SIP OPTIONS towards Sprint, Sprint will receive OPTIONS from the enterprise site as a result of the SIP Entity Monitoring configured for Session Manager. When Session Manager sends SIP OPTIONS to the inside private IP Address of the SBC, the SBC will forward those SIP OPTIONS to Sprint. When Sprint responds, the SBC will pass the response back to Session Manager.

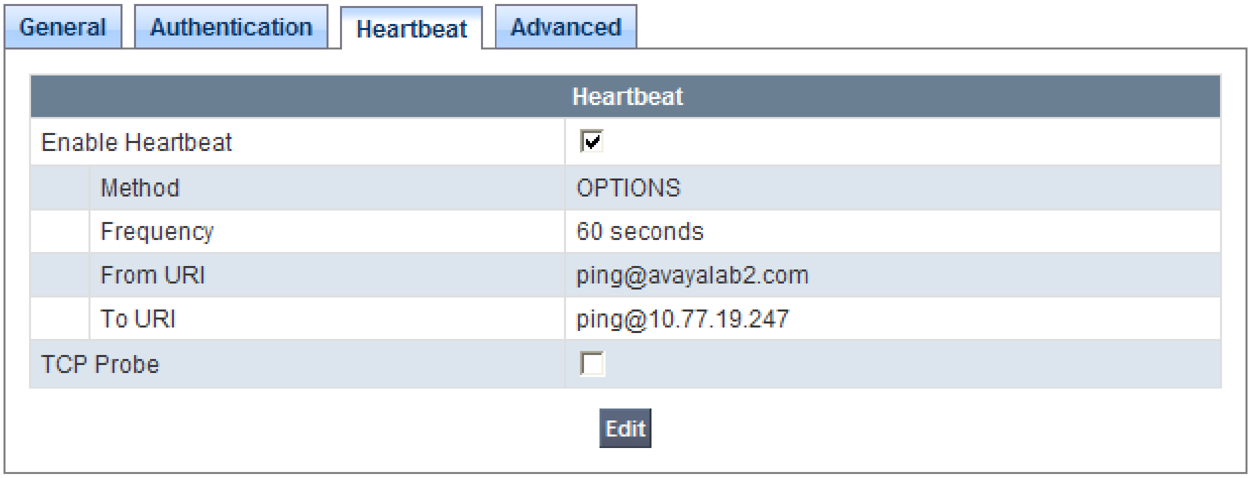
If SBC-sourced OPTIONS are desired, select ***OPTIONS*** from the **Method** drop-down menu. Select the desired frequency that the SBC will source OPTIONS. The **From URI** and **To URI** may be filled in to configure easily identifiable URIs to appear in SIP OPTIONS sourced by the SBC. If adding a new profile, click **Next** (not shown). If editing an existing profile, click **Finish**.



Edit Server Configuration Profile - Heartbeat

Enable Heartbeat	☑
Method	OPTIONS
Frequency	60 seconds
From URI	ping@avayalab2.com
To URI	ping@10.77.19.247
TCP Probe	□
TCP Probe Frequency	seconds

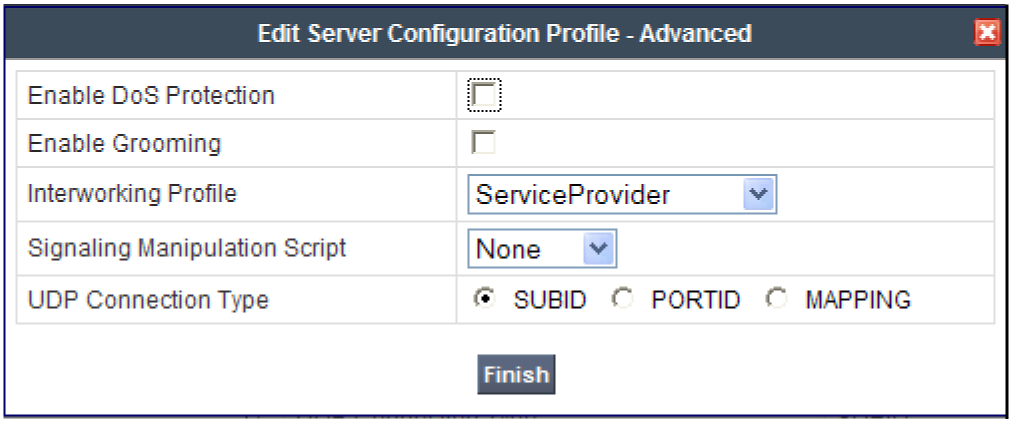
If the optional SBC sourced OPTIONS configuration is completed, the **Heartbeat** tab for the ***Sprint-SIPT*** server profile will appear as shown below.



Heartbeat

Enable Heartbeat	☑
Method	OPTIONS
Frequency	60 seconds
From URI	ping@avayalab2.com
To URI	ping@10.77.19.247
TCP Probe	□

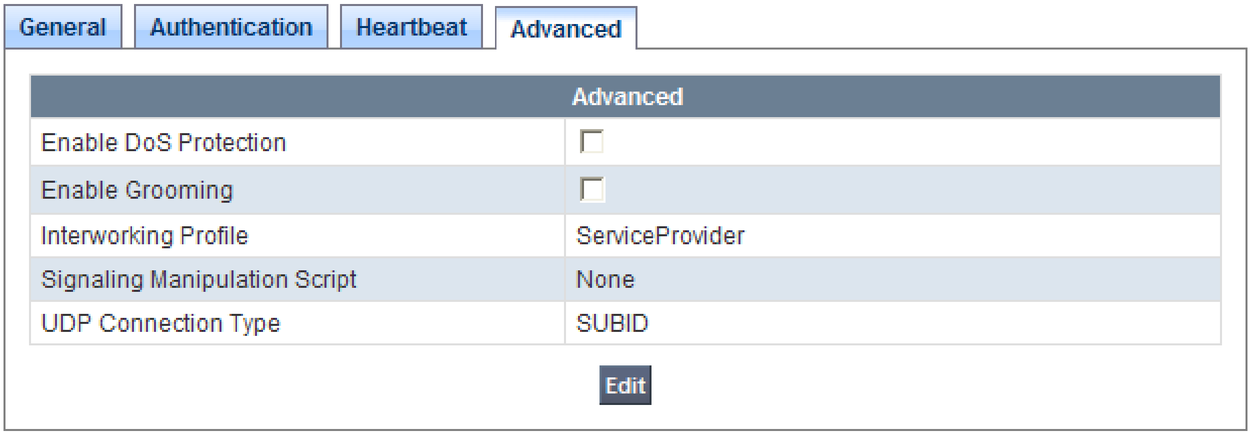
If adding a profile, click **Next** to continuing to the **Advanced** settings (not shown). If editing an existing profile, select the **Advanced** tab and click **Edit**. In the resultant screen, select the **Interworking Profile *ServiceProvider*** created in **Section 7.3.2**. Click **Finish**.



Edit Server Configuration Profile - Advanced

Enable DoS Protection	□
Enable Grooming	□
Interworking Profile	ServiceProvider
Signaling Manipulation Script	None
UDP Connection Type	● SUBID ○ PORTID ○ MAPPING

Once configuration is completed, the **Advanced** tab for ***ServiceProvider*** will appear as shown below.



Advanced	
Enable DoS Protection	□
Enable Grooming	□
Interworking Profile	ServiceProvider
Signaling Manipulation Script	None
UDP Connection Type	SUBID

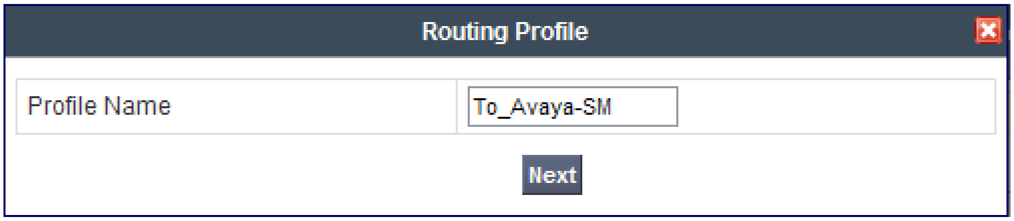
7.5. Global Profiles – Routing

Routing information is required for traffic to be routed to Session Manager on the internal side, and to the Sprint network on the external side. The IP addresses and ports defined here will be used as the destination addresses for signaling. If no port is specified, the default SIP port of 5060 is used.

Navigate to **Global Profiles → Routing** from the left-side menu to configure Routing profiles.

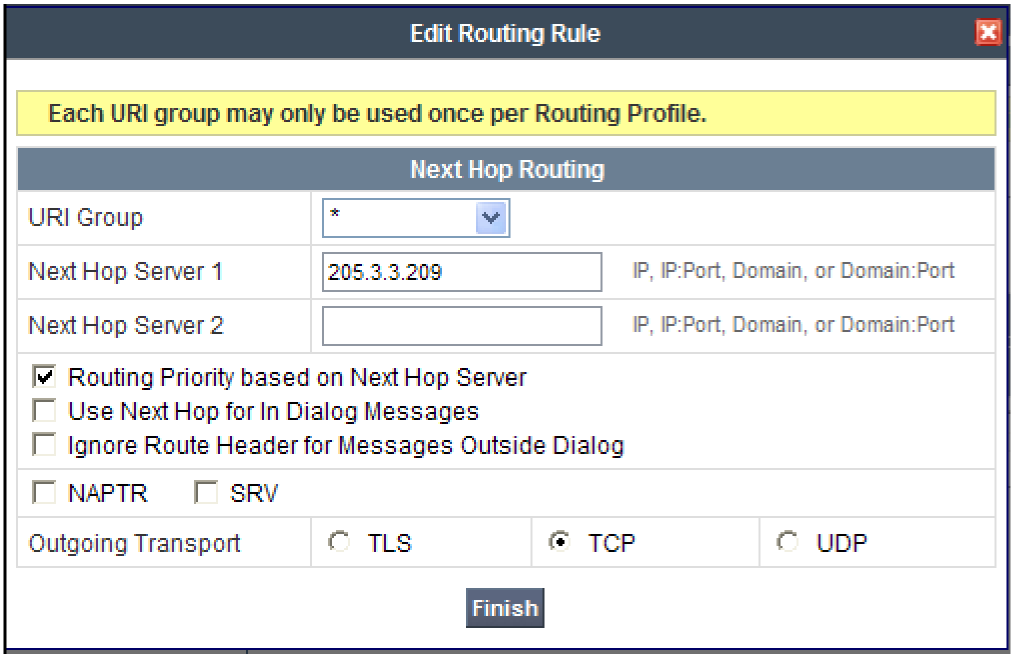
7.5.1. Routing Configuration for Session Manager

Click the **Add Profile** button (not shown) to add a new profile, or select an existing routing profile to edit. If adding a profile, a screen such as the following is displayed. Enter a **Profile Name** such as ***To_Avaya-SM*** shown below. Click **Next**.



Profile Name	To_Avaya-SM

In the **Next Hop Routing** configuration, enter the IP Address of the Session Manager SIP signaling interface with port number (optional if port number is 5060) as **Next Hop Server 1**, as shown below. Check **Routing Priority based on Next Hop Server**. Choose ***TCP*** for **Outgoing Transport**.



Each URI group may only be used once per Routing Profile.

Next Hop Routing

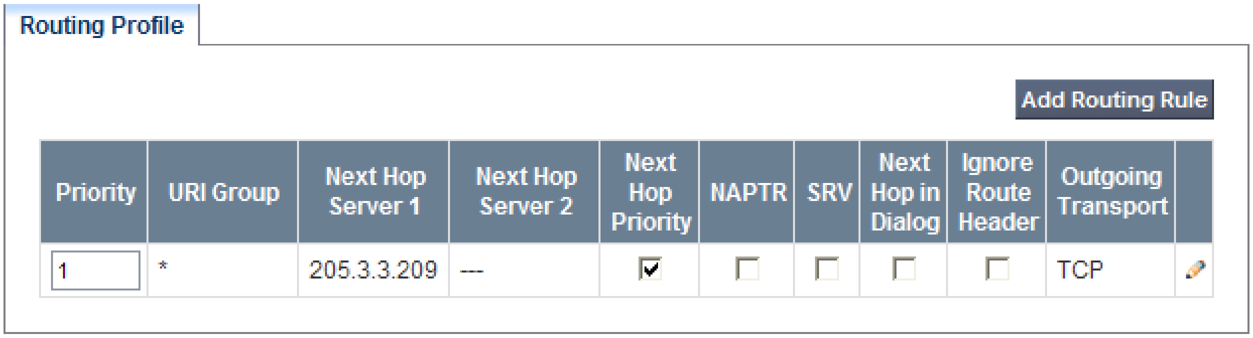
URI Group	*	
Next Hop Server 1	205.3.3.209	IP, IP:Port, Domain, or Domain:Port
Next Hop Server 2		IP, IP:Port, Domain, or Domain:Port

☑ Routing Priority based on Next Hop Server
□ Use Next Hop for In Dialog Messages
□ Ignore Route Header for Messages Outside Dialog

□ NAPTR □ SRV

Outgoing Transport ○ TLS ● TCP ○ UDP

Once configuration is completed, the **Routing Profile** for ***To_Avaya-SM*** will appear as follows:

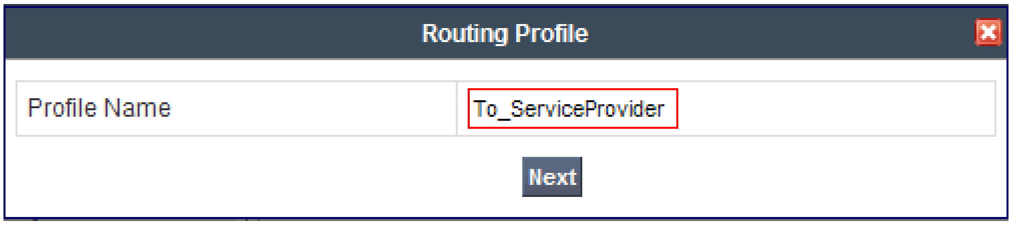


Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport	
1	*	205.3.3.209	---	☑	□	□	□	□	TCP	✎

7.5.2. Routing Configuration for Sprint SIP Trunking

A Routing Profile named ***To_ServiceProvider*** for the trunk server was similarly configured as shown below.

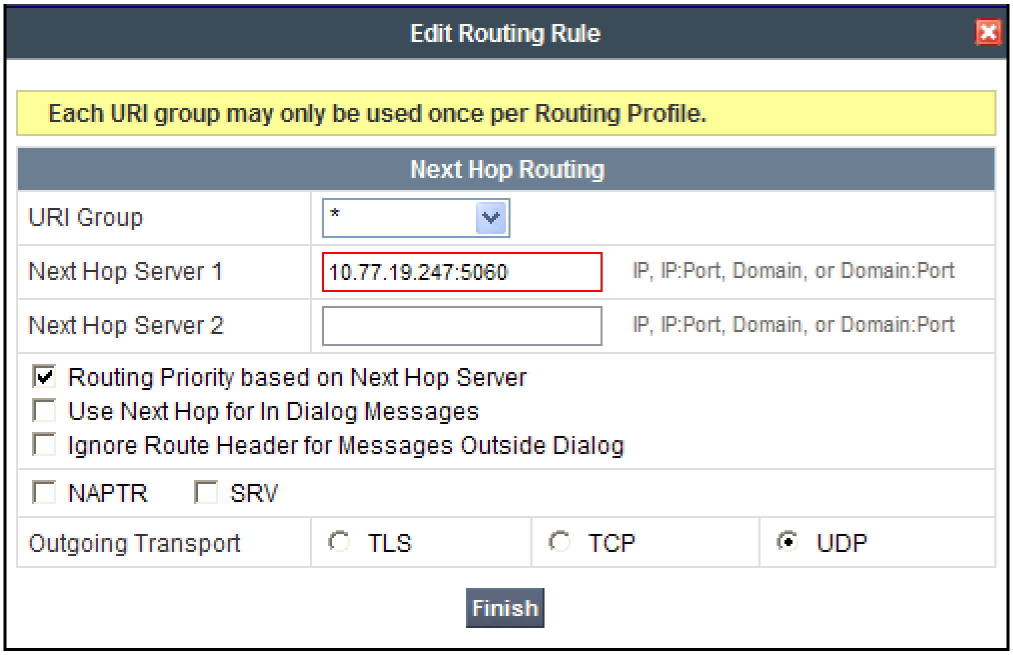
Click the **Add Profile** button (not shown) to add a new profile, or select an existing routing profile to edit. If adding a profile, a screen such as the following is displayed. Enter a **Profile Name** such as ***To_ServiceProvider*** shown below. Click **Next**.



Routing Profile

Profile Name	To_ServiceProvider

In the **Next Hop Routing** configuration, enter the IP Address of the Sprint SIP trunking SBC signaling interface with port number (optional if port number is 5060) as **Next Hop Server 1**, as shown below. Check **Routing Priority based on Next Hop Server**. Choose ***UDP*** for **Outgoing Transport** as Sprint will only accept SIP traffic over UDP.



Edit Routing Rule

Each URI group may only be used once per Routing Profile.

Next Hop Routing

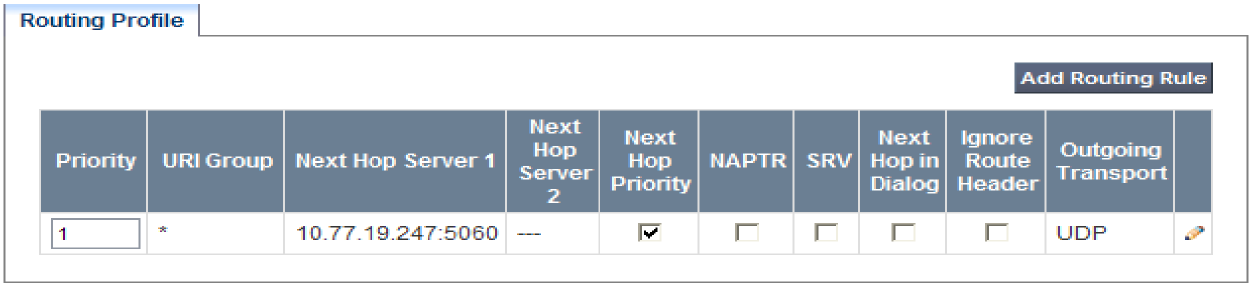
URI Group	*	
Next Hop Server 1	10.77.19.247:5060	IP, IP:Port, Domain, or Domain:Port
Next Hop Server 2		IP, IP:Port, Domain, or Domain:Port

☑ Routing Priority based on Next Hop Server
☐ Use Next Hop for In Dialog Messages
☐ Ignore Route Header for Messages Outside Dialog

☐ NAPTR ☐ SRV

Outgoing Transport ○ TLS ○ TCP ● UDP

Once configuration is completed, the **Routing Profile** for ***To_ServiceProvider*** will appear as follows:



Routing Profile

Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport	
1	*	10.77.19.247:5060	---	☑	☐	☐	☐	☐	UDP	

7.6. Global Profiles – Topology Hiding

Topology Hiding is a security feature which allows the changing of several parameters within SIP packets, preventing the private enterprise network information from being propagated to the un-trusted public network.

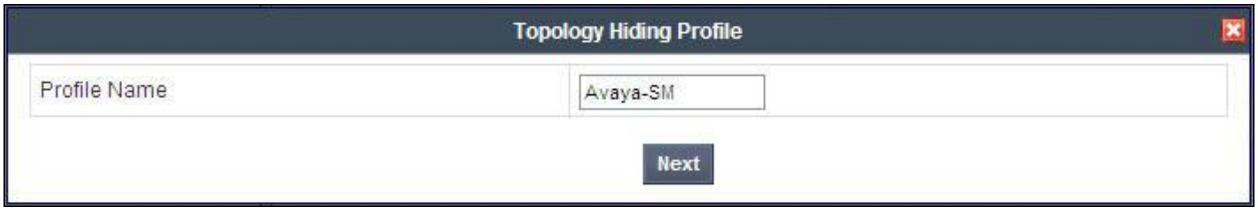
Topology Hiding can also be used as an interoperability tool to adapt certain parameters in selected SIP headers to meet expectations by Session Manager and the SIP trunk service provider, allowing the call to be accepted in each case.

For the compliance test, only the minimum configuration required to achieve interoperability was performed.

Navigate to **Global Profiles → Topology Hiding** from the left-side menu for configuring Topology Hiding profiles.

7.6.1. Topology Hiding for Session Manager

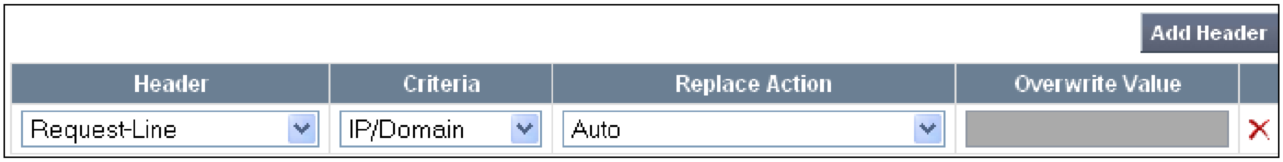
Click the **Add Profile** button (not shown) to add a new profile, or select an existing topology hiding profile to edit. If adding a profile, a screen such as the following is displayed. Enter a **Profile Name** such as ***Avaya-SM*** shown below. Click **Next**.



Topology Hiding Profile

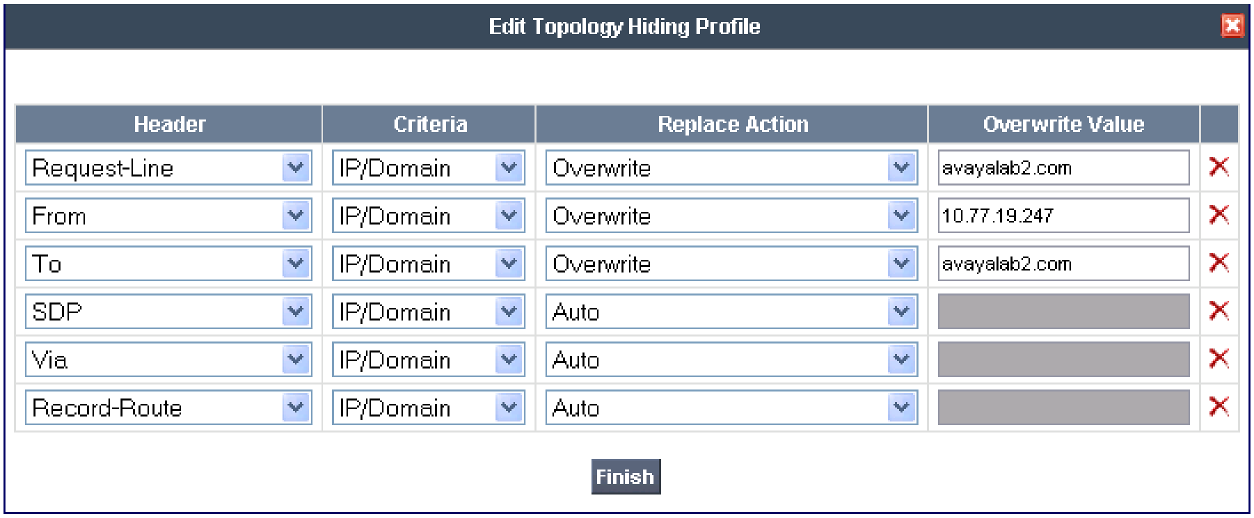
Profile Name: Avaya-SM

In the resultant screen, click the **Add Header** button to reveal additional headers.



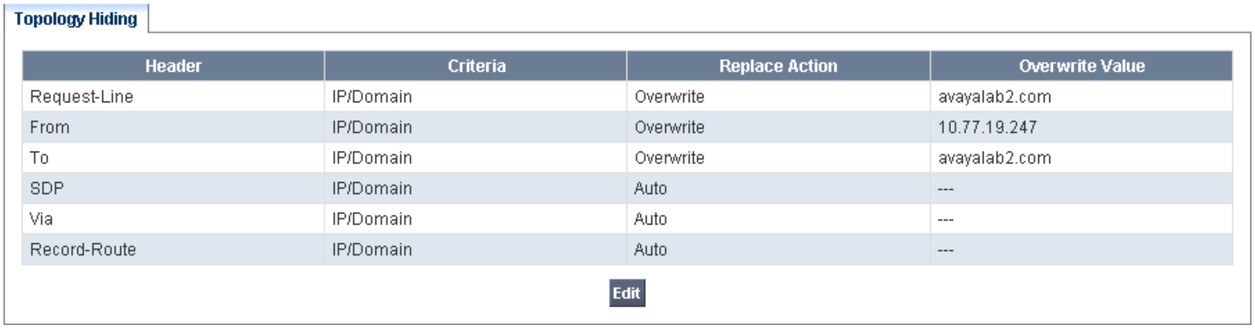
Header	Criteria	Replace Action	Overwrite Value	
Request-Line	IP/Domain	Auto		✕

To ensure that the domain received by Session Manager from the SBC is the expected enterprise domain, select ***Overwrite*** from the drop-down menu as the **Replace Action** for the To and Request-Line headers. Enter the enterprise domain in the **Overwrite Value** column as shown below. In the example below, the domain received by Session Manager has been changed by the Avaya SBCE to ***avayalab2.com***. Next, select ***Overwrite*** from the drop-down menu as the **Replace Action** for the From header. Enter the IP address of the Sprint SIP trunking SBC as the ***Overwrite value***. Click **Finish**.



Header	Criteria	Replace Action	Overwrite Value	
Request-Line	IP/Domain	Overwrite	avayalab2.com	✕
From	IP/Domain	Overwrite	10.77.19.247	✕
To	IP/Domain	Overwrite	avayalab2.com	✕
SDP	IP/Domain	Auto		✕
Via	IP/Domain	Auto		✕
Record-Route	IP/Domain	Auto		✕

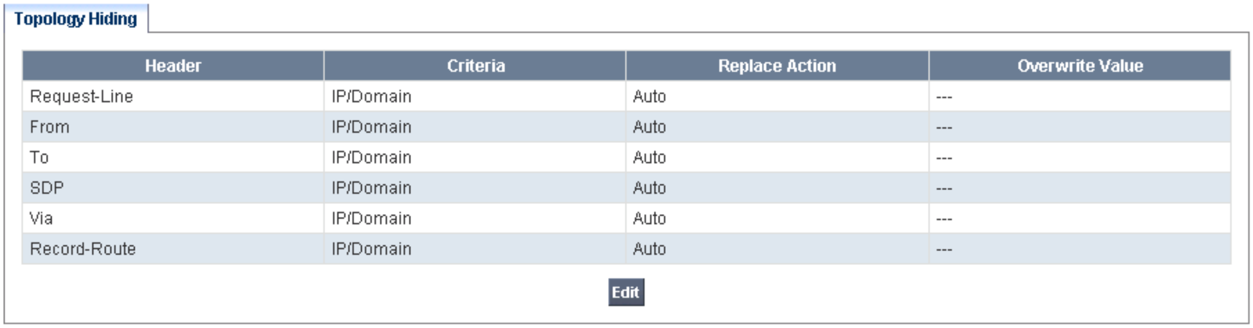
After configuration is completed, the Topology Hiding for profile ***Avaya-SM*** will appear as follows.



Header	Criteria	Replace Action	Overwrite Value
Request-Line	IP/Domain	Overwrite	avayalab2.com
From	IP/Domain	Overwrite	10.77.19.247
To	IP/Domain	Overwrite	avayalab2.com
SDP	IP/Domain	Auto	---
Via	IP/Domain	Auto	---
Record-Route	IP/Domain	Auto	---

7.6.2. Topology Hiding for Sprint SIP Trunking

A Topology Hiding profile named ***ServiceProvider*** for Sprint was similarly configured as shown below. Note that it was not necessary to configure any **Topology Hiding** for SIP signaling that was forwarded to the Service Provider. Default values were used for all fields.



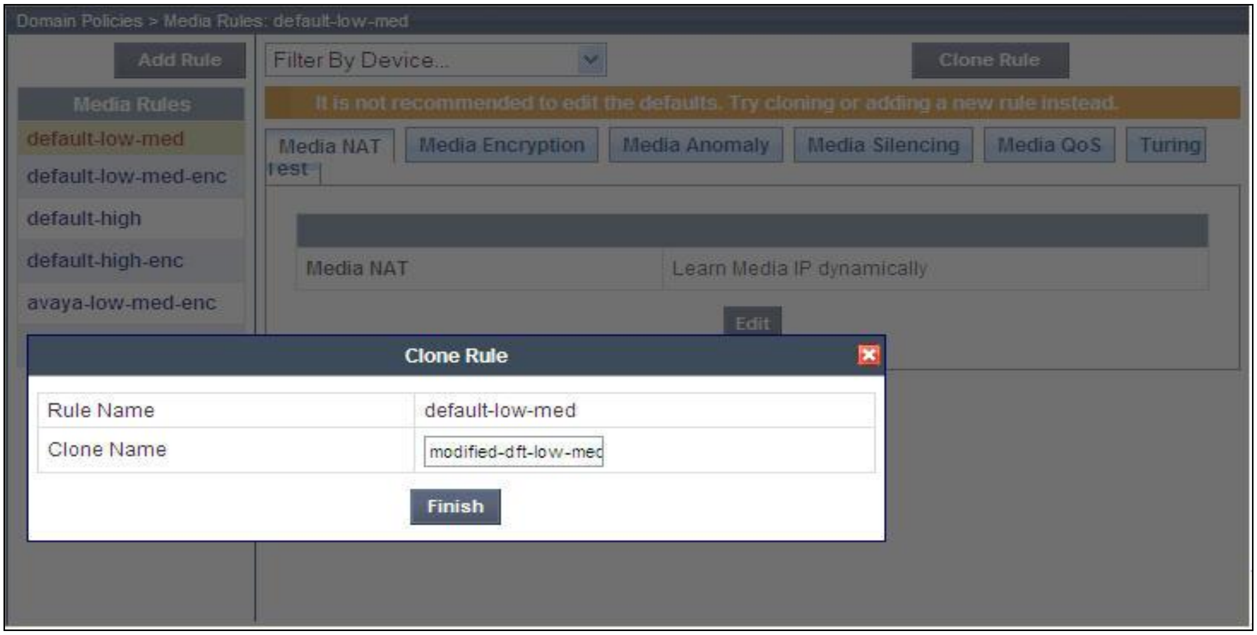
Header	Criteria	Replace Action	Overwrite Value
Request-Line	IP/Domain	Auto	---
From	IP/Domain	Auto	---
To	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---
Via	IP/Domain	Auto	---
Record-Route	IP/Domain	Auto	---

7.7. Domain Policies – Media Rules

Domain Policies allow configuring, managing and applying various sets of rules designed to control and normalize the behavior of call flows, based upon various criteria of communication sessions originating from or terminating to the enterprise.

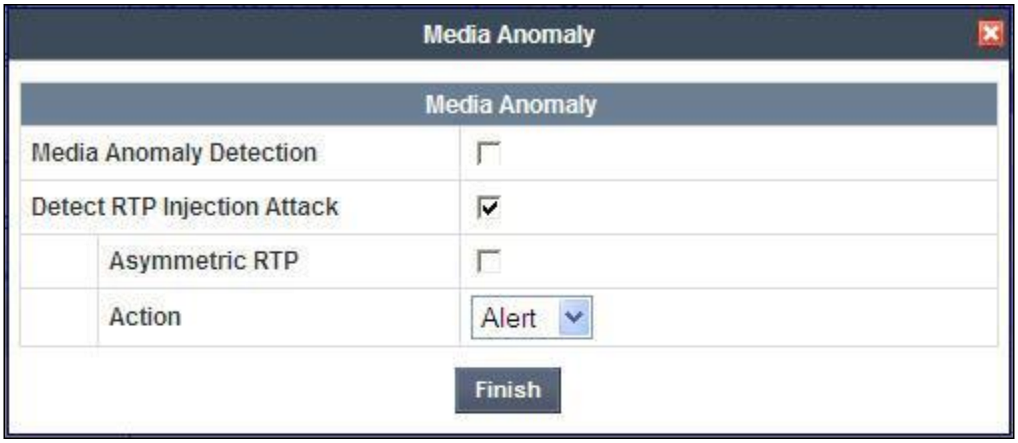
Navigate to **Domain Policies → Media Rules** from the left-side menu to configure Media Rules.

In the sample configuration, a single media rule was used. This media rule was cloned from the default rule ***default-low-med*** by selecting the default rule ***default-low-med*** then clicking the **Clone Rule** button in the upper right corner as shown below:



Rule Name	default-low-med
Clone Name	modified-dft-low-med

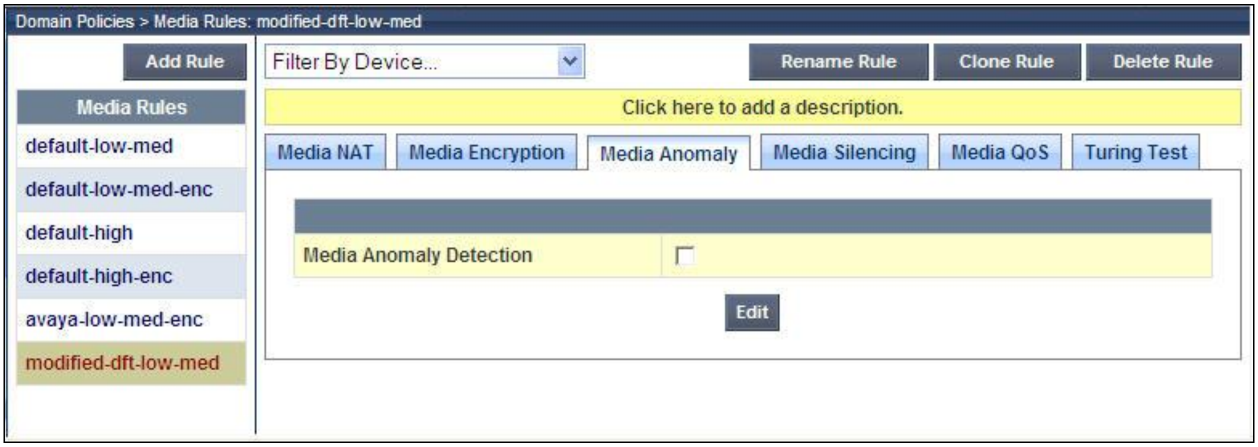
Enter a descriptive **Clone Name**, and then click **Finish**. The cloned media rule will be displayed in the **Media Rules** list on the left. Select this cloned rule from the list, then select **Media Anomaly** tab and click **Edit**. In the displayed Media Anomaly edit window, uncheck **Media Anomaly Detection** as shown below.



Media Anomaly

Media Anomaly	
Media Anomaly Detection	□
Detect RTP Injection Attack	☑
Asymmetric RTP	□
Action	Alert

Click **Finish**. The rule named ***modified-dft-low-med*** is shown below with the Media Anomaly tab selected. This rule is sufficient for the compliance test.



Domain Policies > Media Rules: modified-dft-low-med

Media Anomaly Detection	□

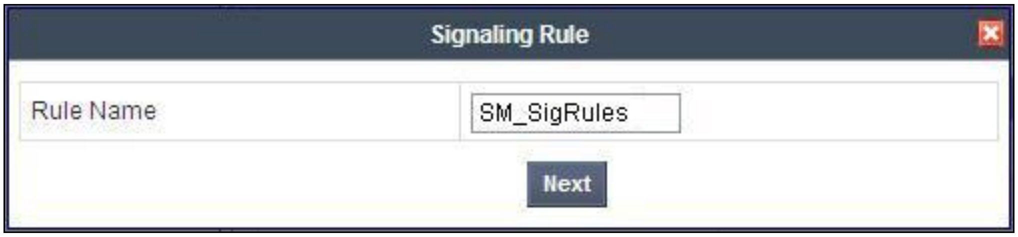
7.8. Domain Policies – Signaling Rules

Signaling Rules define the actions to be taken (*Allow*, *Block*, *Block with Response*, etc.) on signaling request and response messages. They also allow the setting of Quality of Service markings for the signaling packets.

The P-Location and P-Charging-Vector headers are sent in SIP messages from Session Manager to the service provider network via the Avaya SBCE. These headers should not be exposed outside the enterprise. For simplicity, these headers were simply removed (blocked) from both request and response messages that originated from Session Manager.

Navigate to **Domain Policies → Signaling Rules** from the left-side menu to configure Signaling Rules.

Click the Add Rule button (not shown) to add a new signaling rule. In the **Rule Name** field, enter an appropriate name, such as ***SM_SigRules***. Click **Next**.

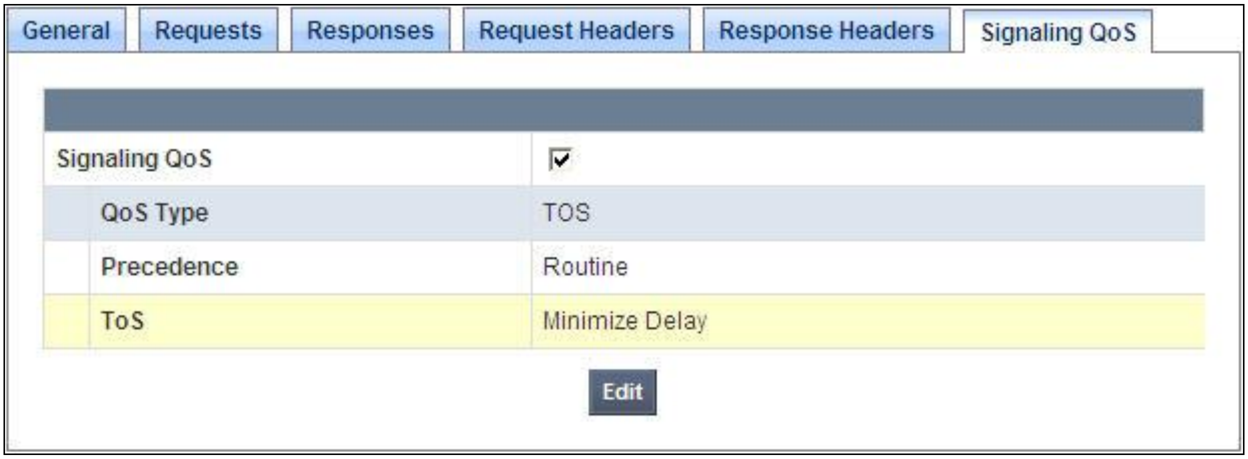


Signaling Rule

Rule Name	SM_SigRules

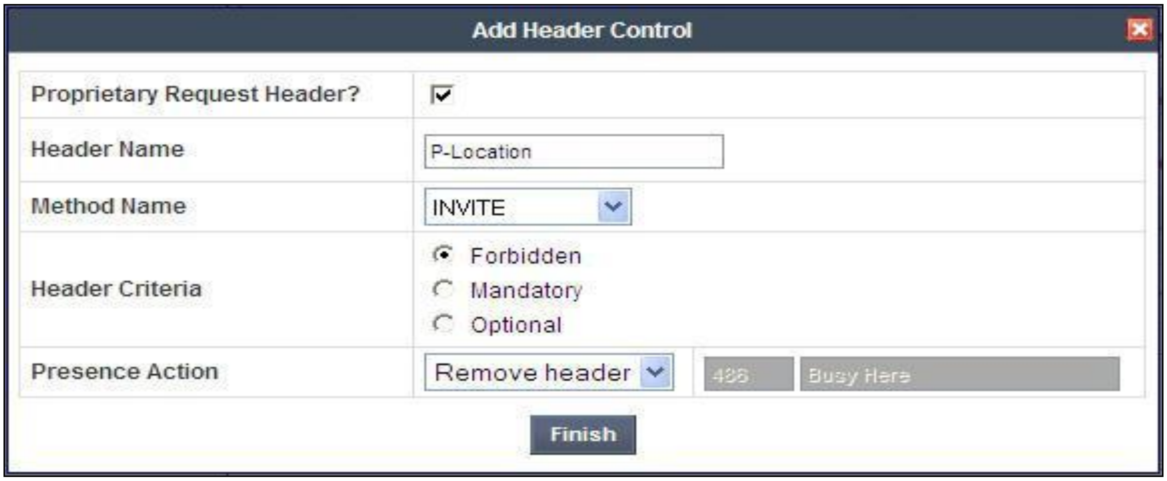
In the subsequent screen (not shown), click **Next** to accept defaults. In the Signaling QoS screen, click **Finish** (not shown).

After this configuration, the new ***SM_SigRules*** rule will appear as follows.



Signaling QoS	☑
QoS Type	TOS
Precedence	Routine
ToS	Minimize Delay

Select the **Request Headers** tab, and select the **Add In Header Control** button. Check the **Proprietary Request Header?** checkbox. In the **Header Name** field, type ***P-Location***. Select ***INVITE*** as the **Method Name** from the drop-down menu. In the Header Criteria, select ***Forbidden***. Retain **Presence Action *Remove header***. The intent is to remove the P-Location header which is inserted by Session Manager but not needed by the Sprint SIP trunking service. This configuration is optional in that the P-Location and P-Charging-Vector headers do not cause any user-perceivable problem if presented to Sprint.



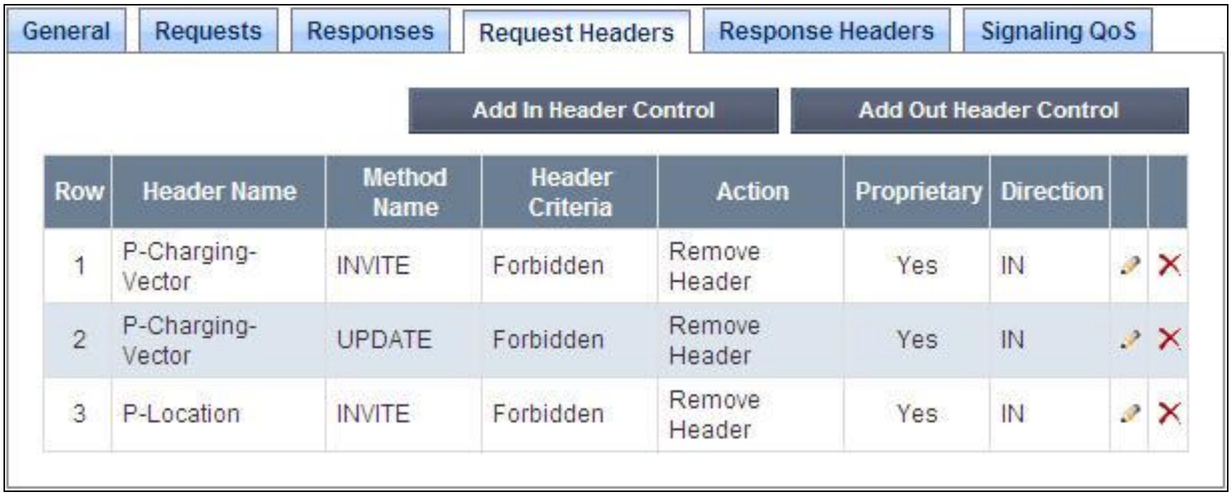
Add Header Control

Proprietary Request Header?	☑
Header Name	P-Location
Method Name	INVITE
Header Criteria	● Forbidden ○ Mandatory ○ Optional
Presence Action	Remove header 486 Busy Here

Similarly, configure additional header control rules to:

- Remove the P-Charging-Vector header in the outbound INVITE
- Remove the P-Charging-Vector header in the outbound UPDATE

Once complete, the **Request Headers** tab appears as follows.

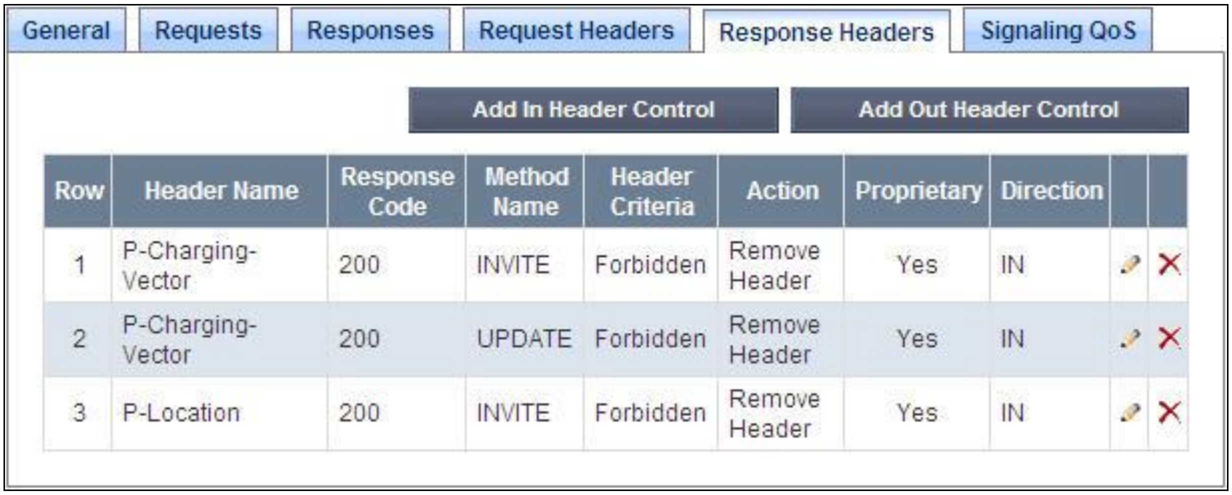


Row	Header Name	Method Name	Header Criteria	Action	Proprietary	Direction		
1	P-Charging-Vector	INVITE	Forbidden	Remove Header	Yes	IN	✎	✕
2	P-Charging-Vector	UPDATE	Forbidden	Remove Header	Yes	IN	✎	✕
3	P-Location	INVITE	Forbidden	Remove Header	Yes	IN	✎	✕

Select the **Response Headers** tab and repeat the above configuration steps to:

- Remove the P-Charging-Vector header in the 200 OK response to INVITEs
- Remove the P-Charging-Vector header in the 200 OK response to UPDATEs
- Remove the P-Location header in the 200 OK response to INVITEs

Once configuration is complete, the **Response Headers** tab for the ***SM_SigRules*** signaling rule will appear as follows.



Row	Header Name	Response Code	Method Name	Header Criteria	Action	Proprietary	Direction		
1	P-Charging-Vector	200	INVITE	Forbidden	Remove Header	Yes	IN	✎	✕
2	P-Charging-Vector	200	UPDATE	Forbidden	Remove Header	Yes	IN	✎	✕
3	P-Location	200	INVITE	Forbidden	Remove Header	Yes	IN	✎	✕

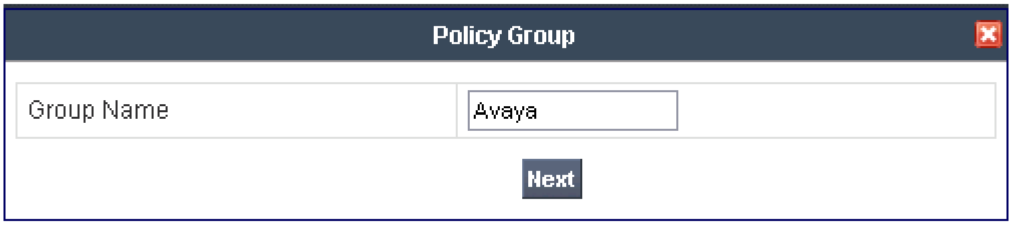
Again, this configuration is optional in that the P-Location and P-Charging-Vector headers do not cause any user-perceivable problem if presented to Sprint.

7.9. Domain Policies – End Point Policy Groups

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

Navigate to **Domain Policies → End Point Policy Groups** from the left-side menu to configure End Point Policy Groups.

Select the **Add Group** button (not shown). Enter a name in the **Group Name** field, such as ***Avaya*** as shown below. Click **Next**.

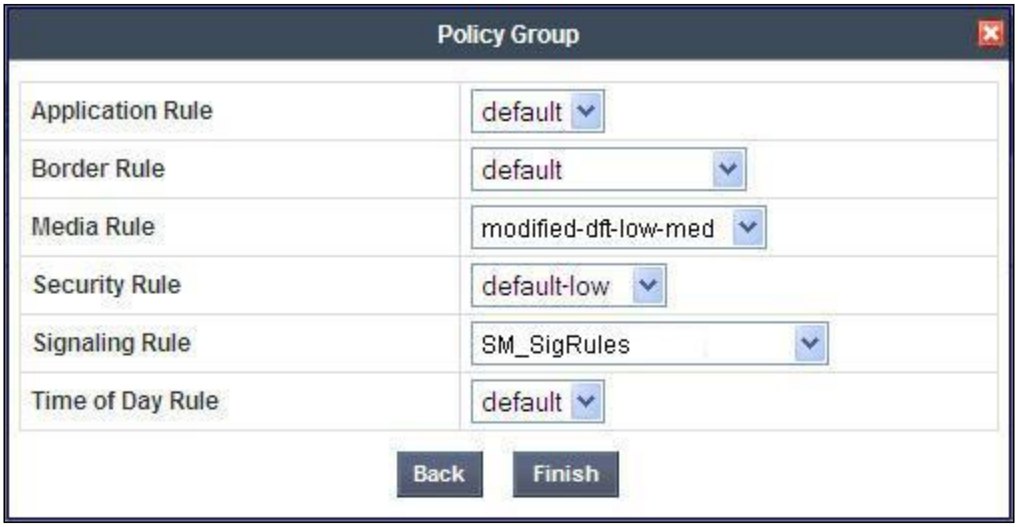


Group Name	Avaya

In the sample configuration, defaults were selected for all fields, with the exception of:

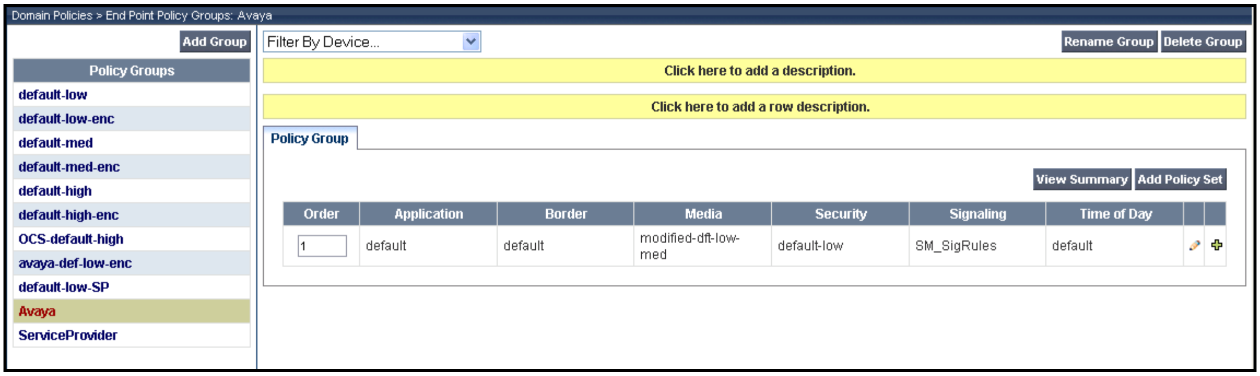
- **Media Rule**, which was set to the ***modified-dft-low-med*** media rule as defined in **Section 7.7**
- **Signaling Rule**, which was set to the ***SM_SigRules*** signaling rule as defined in **Section 7.8**

Click **Finish**.



Application Rule	default
Border Rule	default
Media Rule	modified-dft-low-med
Security Rule	default-low
Signaling Rule	SM_SigRules
Time of Day Rule	default

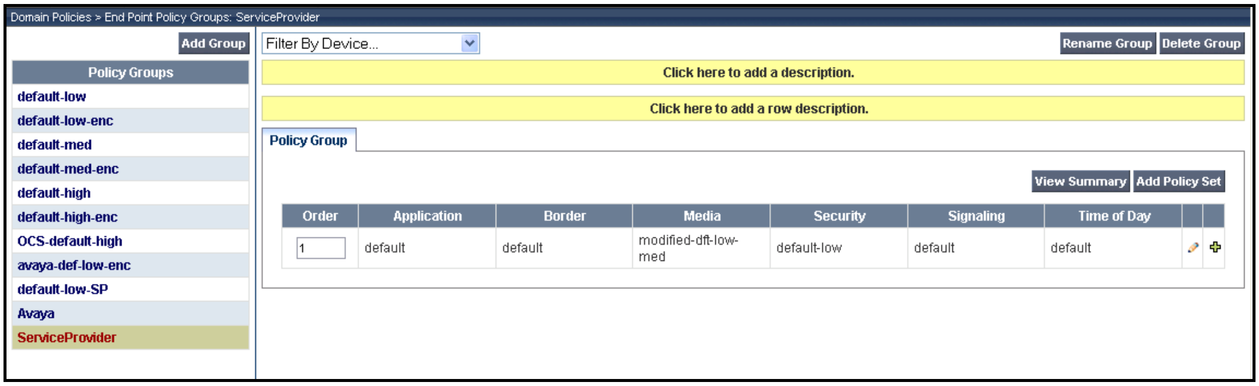
Once configuration is completed, the ***Avaya*** End Point Policy Group will appear as follows.



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

Repeat the configuration steps above to create a 2nd **End Point Policy Group** named ***ServiceProvider*** for the network side as shown below.

Note that this End Point Policy Group uses the same Media Rule (***modified-dft-low-med***) for disabling Media Anomaly Detection and the default Signaling Rule since no header manipulations are required for messages to and from the outside interface of the SBC.



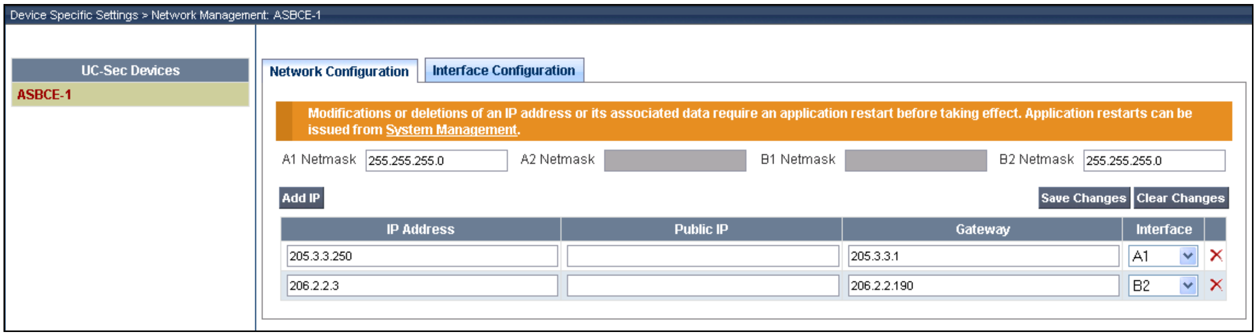
Order	Application	Border	Media	Security	Signaling	Time of Day
1	default	default	modified-dft-low-med	default-low	default	default

7.10. Device Specific Settings – Network Management

The network information should have been previously specified during installation of Avaya SBCE.

Navigate to **Device Specific Setting → Network Management** from the left-side menu.

Under **UC-Sec Devices**, select the device being managed, which was named ***ASBCE-1*** in the sample configuration. The **Network Configuration** tab is shown below. Observe the **IP Address**, **Netmask**, **Gateway**, and **Interface** information previously assigned. Note that only the **A1** and **B2** interfaces are used, typically the **A** interfaces are used for the internal side and **B** interfaces are used for the external side of the Avaya SBCE.


Device Specific Settings > Network Management: ASBCE-1

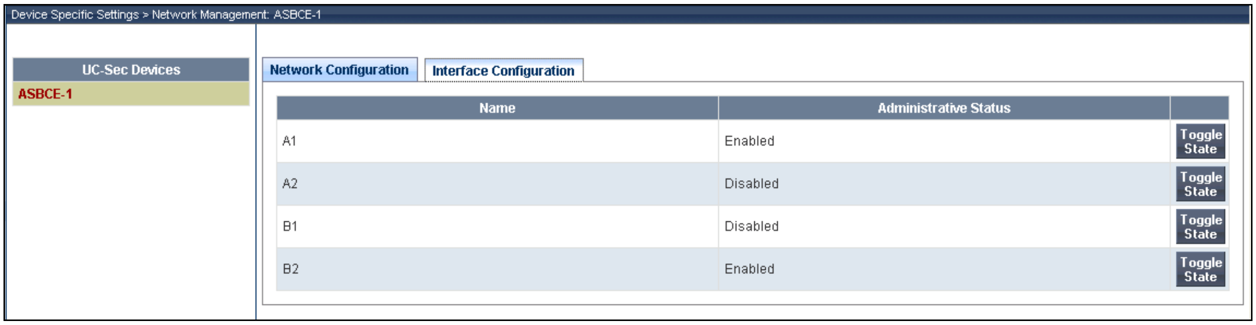
UC-Sec Devices
ASBCE-1

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.

A1 Netmask 255.255.255.0 A2 Netmask B1 Netmask B2 Netmask 255.255.255.0

IP Address	Public IP	Gateway	Interface	
205.3.3.250		205.3.3.1	A1	✕
206.2.2.3		206.2.2.190	B2	✕

Select the **Interface Configuration** tab. The **Administrative Status** can be toggled between **Enabled** and **Disabled** in this screen. The following screen was captured after the interfaces had already been enabled. To enable the interface if it is disabled, click the **Toggle State** button.


Device Specific Settings > Network Management: ASBCE-1

UC-Sec Devices
ASBCE-1

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

When IP addresses and network masks are assigned to interfaces, these are then configured as signaling and media interfaces.

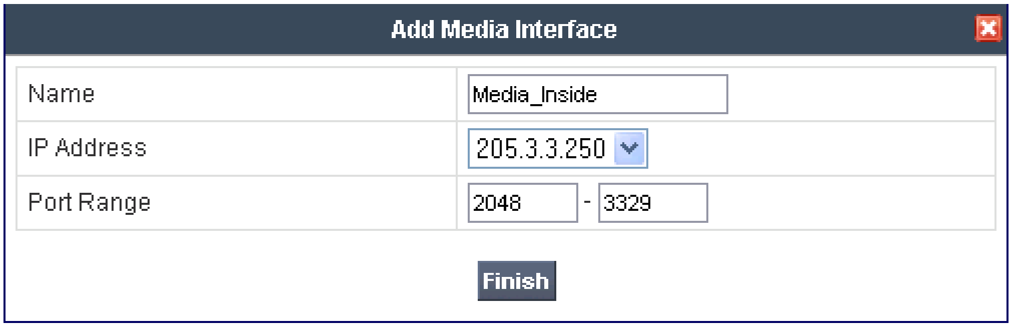
7.11. Device Specific Settings – Media Interface

Media Interfaces are created to adjust the port range assigned to media streams leaving the interfaces of the SBC. The compliance test used the port range 2048 to 3329 for the inside, private interface to match the default media port range for Communication Manger. The public interface was set to use the Avaya SBCE default media port range of 35000 to 400.

Navigate to **Device Specific Setting → Media Interface** from the left-side menu to configure Media Interfaces; one for internal and one for external.

Under **UC-Sec Devices**, select the device being managed, which was named ***ASBCE-1*** in the sample configuration (not shown). Select **Add Media Interface**.

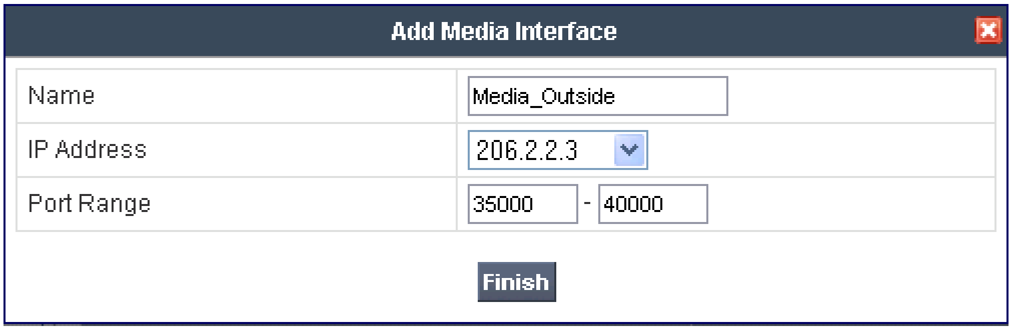
Enter an appropriate **Name** for the Media Interface facing the enterprise and select the inside, private IP Address for the Avaya SBCE from the **IP Address** drop-down menu. In the sample configuration, **Media_Inside** was chosen as the name, and the inside IP Address of the SBC is ***205.3.3.250***. For the **Port Range** the default Communication Manager media port range of ***2048*** to ***3329*** are shown. Click **Finish**.



Add Media Interface

Name	Media_Inside
IP Address	205.3.3.250
Port Range	2048 - 3329

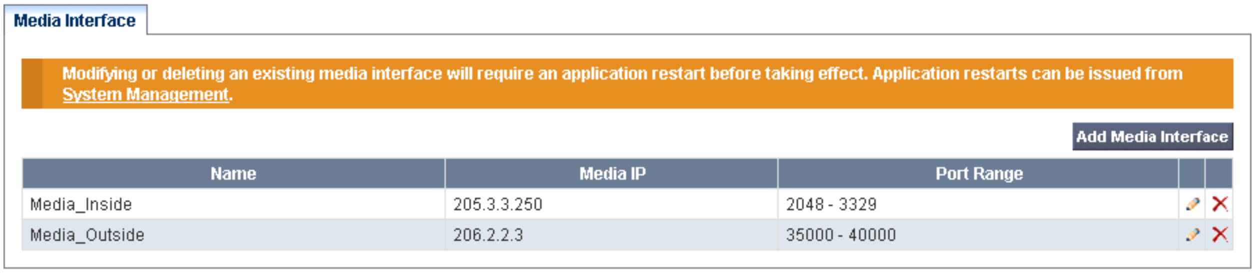
An external Media Interface facing the network was similarly created with the name ***Media_Outside***. The outside IP Address of the SBC (***206.2.2.3***) was selected from the drop-down menu and the **Port Range** setting was left at the Avaya SBCE default value of ***35000*** to ***40000***.



Add Media Interface

Name	Media_Outside
IP Address	206.2.2.3
Port Range	35000 - 40000

The resultant Media Interface configuration used in the sample configuration is shown below.



Media Interface

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

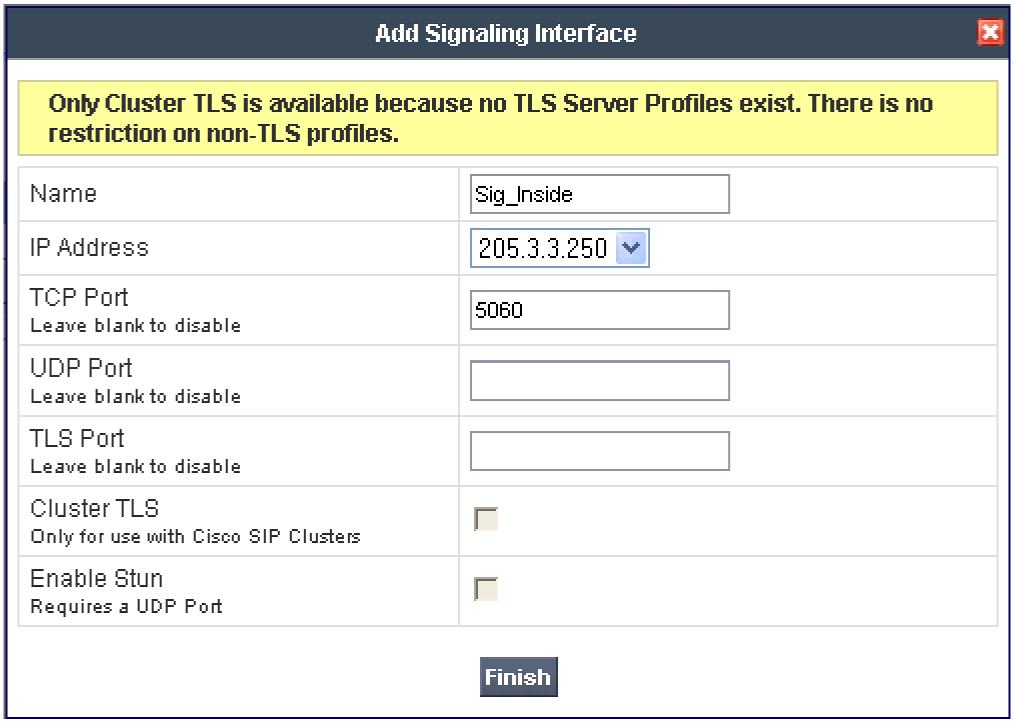
Name	Media IP	Port Range
Media_Inside	205.3.3.250	2048 - 3329
Media_Outside	206.2.2.3	35000 - 40000

7.12. Device Specific Settings – Signaling Interface

Navigate to **Device Specific Setting → Signaling Interface** from the left-side menu to configure Signaling Interfaces; one for internal and one for external.

Under **UC-Sec Devices**, select the device being managed, which was named ***ASBCE-1*** in the sample configuration (not shown). Select **Add Signaling Interface**.

In the **Add Signaling Interface** screen, enter an appropriate **Name** (e.g., ***Sig_Inside***) for the inside interface, and choose the private, inside IP Address of the Avaya SBCE from the **IP Address** drop-down menu. Enter ***5060*** for **TCP Port** since TCP port 5060 is used between Session Manager and the Avaya SBCE in the sample configuration. Click **Finish**.

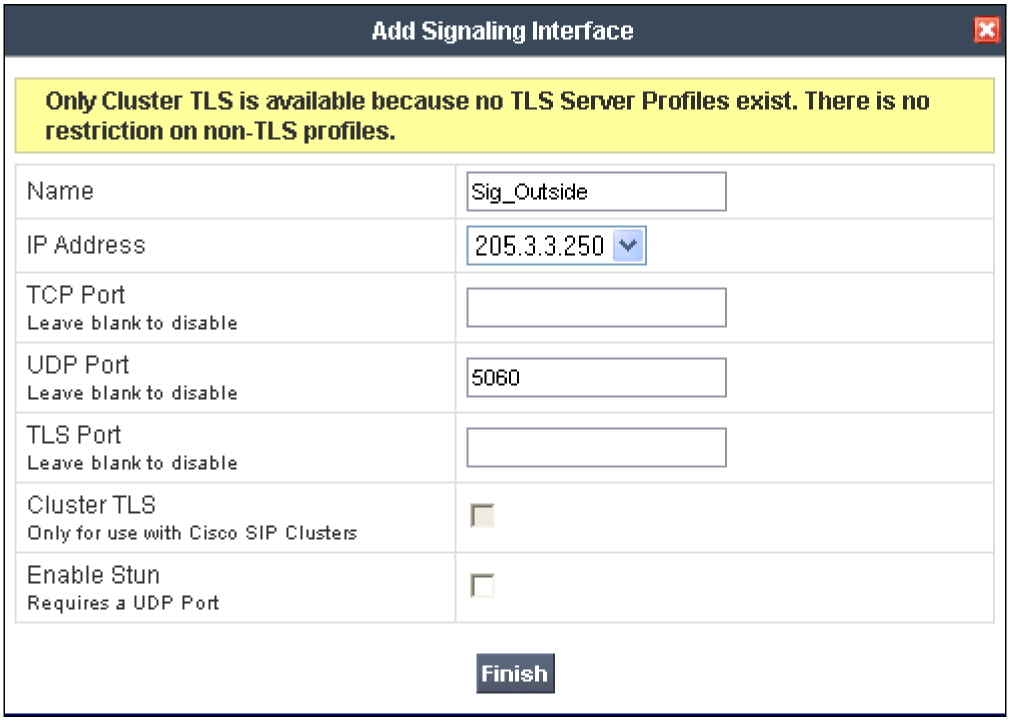


Add Signaling Interface

Only Cluster TLS is available because no TLS Server Profiles exist. There is no restriction on non-TLS profiles.

Name	Sig_Inside
IP Address	205.3.3.250
TCP Port Leave blank to disable	5060
UDP Port Leave blank to disable	
TLS Port Leave blank to disable	
Cluster TLS Only for use with Cisco SIP Clusters	□
Enable Stun Requires a UDP Port	□

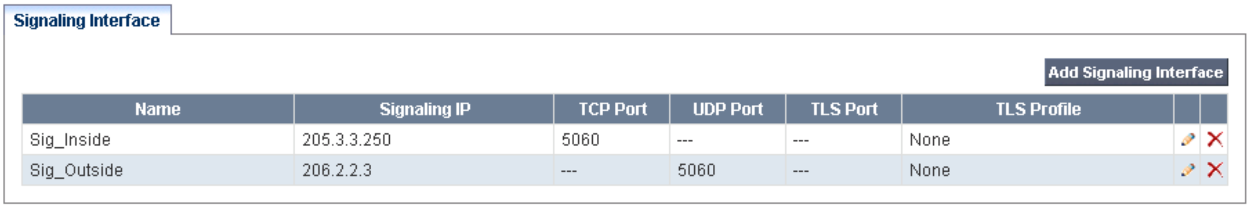
An external Signaling Interface facing the network was similarly created with the name ***Sig_Outside***. Select the outside, public IP Address of the Avaya SBCE (***206.2.2.3***) from the drop-down menu. Note that ***5060*** was specified as the **UDP Port** since UDP was used between the Avaya SBCE and the Sprint network.



Only Cluster TLS is available because no TLS Server Profiles exist. There is no restriction on non-TLS profiles.

Name	Sig_Outside
IP Address	205.3.3.250
TCP Port Leave blank to disable	
UDP Port Leave blank to disable	5060
TLS Port Leave blank to disable	
Cluster TLS Only for use with Cisco SIP Clusters	□
Enable Stun Requires a UDP Port	□

The following screen shows the Signaling Interfaces defined for the sample configuration.



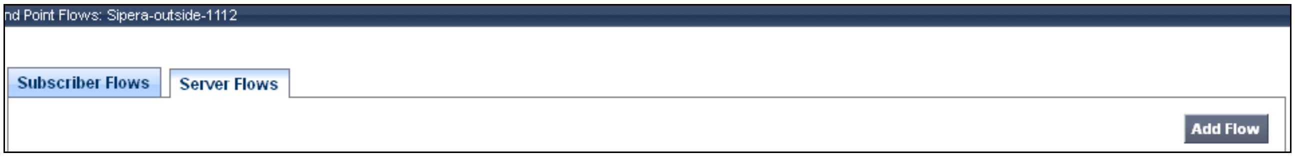
Name	Signaling IP	TCP Port	UDP Port	TLS Port	TLS Profile	
Sig_Inside	205.3.3.250	5060	---	---	None	✎ ✕
Sig_Outside	206.2.2.3	---	5060	---	None	✎ ✕

7.13. Device Specific Settings – End Point Server Flows

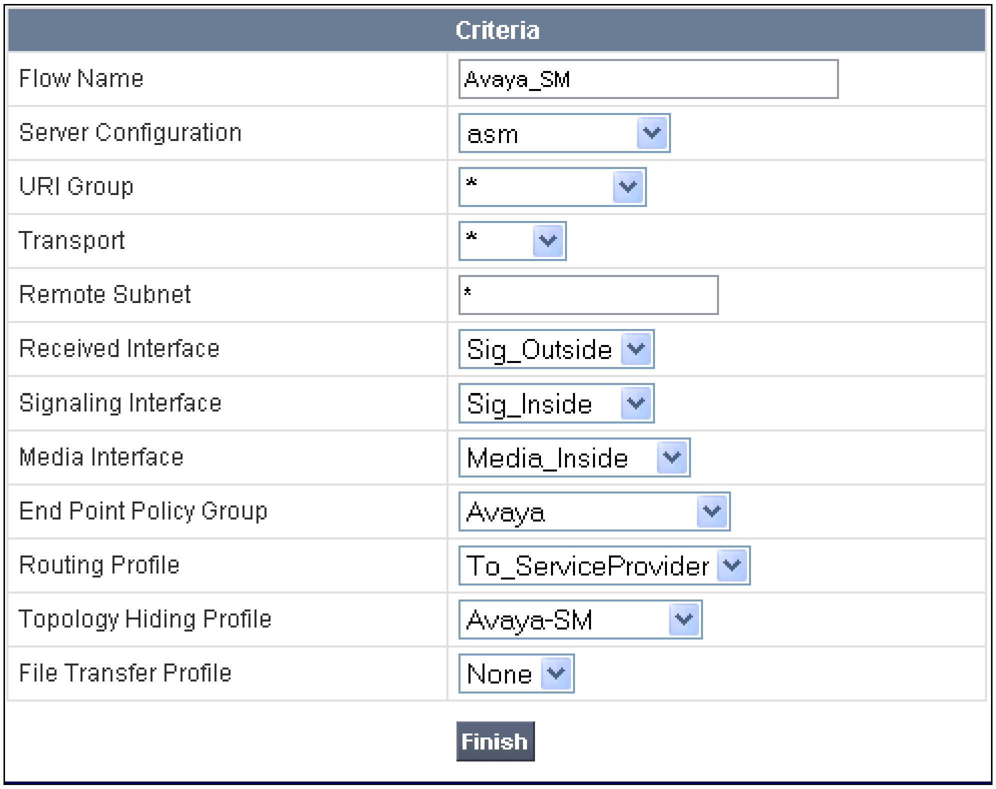
End Point Server Flows combine the previously defined profiles into an outgoing flow from the Call Server (Session Manager) to the Trunk Server (service provider network) and an incoming flow from the Trunk Server to the Call Server. This configuration ties all the previously entered information together so that calls can be routed from Session Manager to the service provider network and vice versa.

Select **Device Specific Setting → End Point Flows** from the left-side menu to configure End Point Flows.

Under **UC-Sec Devices**, select the device being managed, which was named ***ASBCE-1*** in the sample configuration (not shown). Select the **Server Flows** tab. Select **Add Flow**.



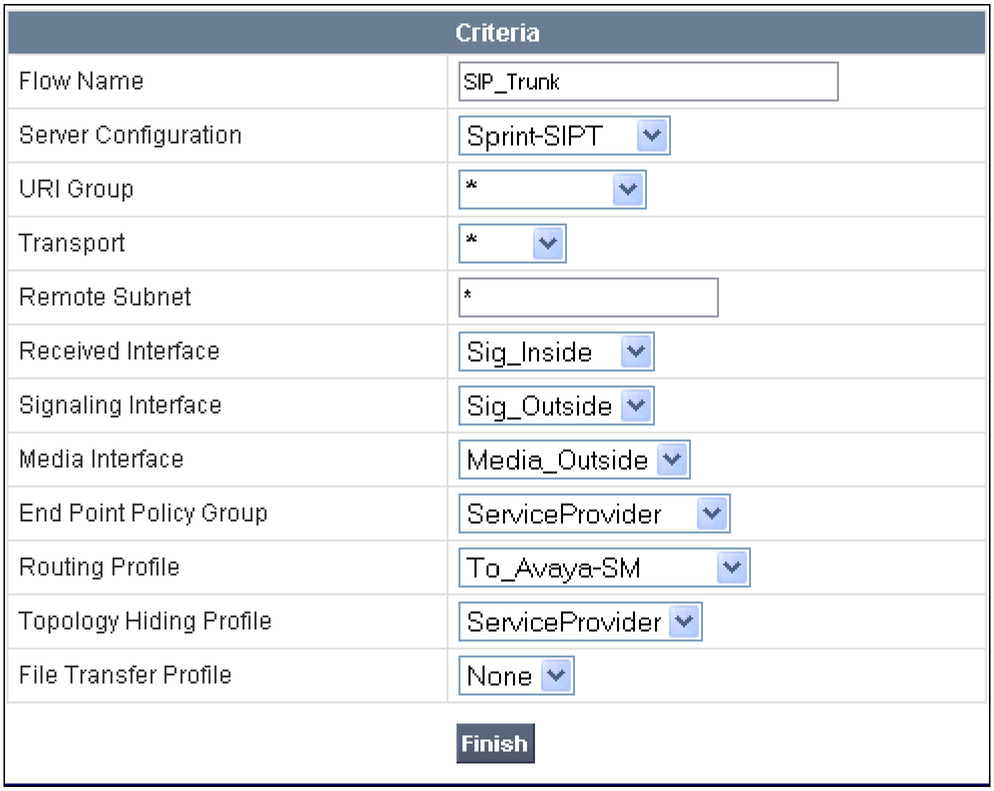
The following screen shows the flow named ***Avaya-SM*** being added to the sample configuration. This flow uses the interfaces, policies, and profiles defined in previous sections. Note the **Routing Profile** selection which is the reverse route of the flow. Click **Finish**.



Criteria	
Flow Name	Avaya_SM
Server Configuration	asm
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Sig_Outside
Signaling Interface	Sig_Inside
Media Interface	Media_Inside
End Point Policy Group	Avaya
Routing Profile	To_ServiceProvider
Topology Hiding Profile	Avaya-SM
File Transfer Profile	None

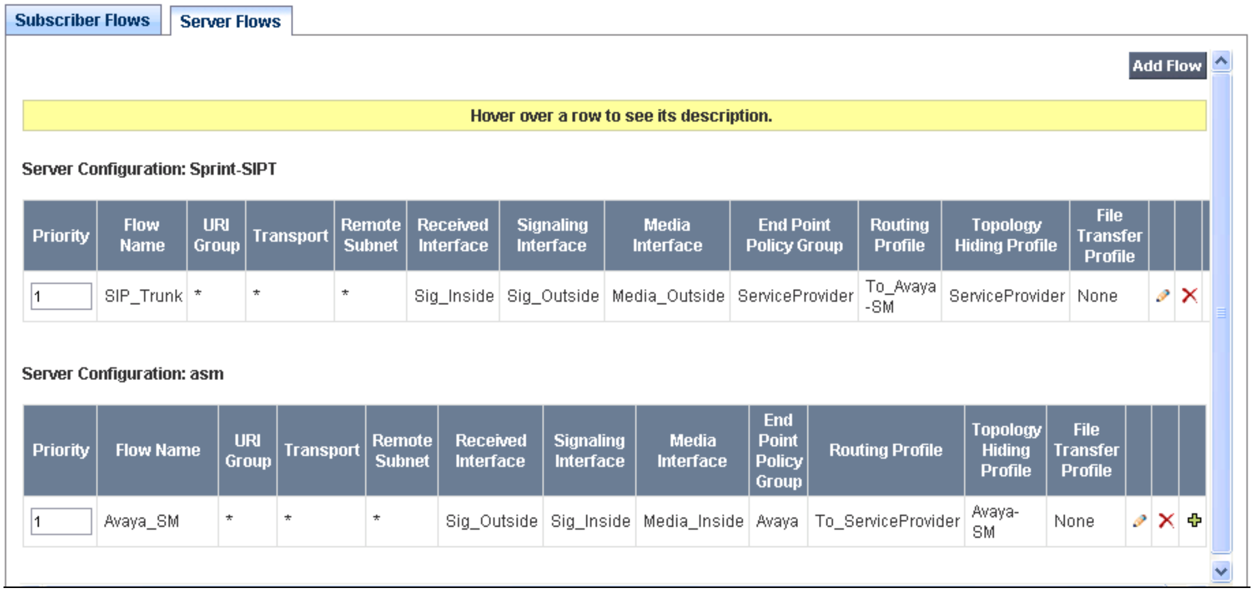
Once again, select the **Server Flows** tab. Select **Add Flow**.

The following screen shows the flow named ***SIP_Trunk*** being added to the sample configuration. This flow uses the interfaces, policies, and profiles defined in previous sections. Note the **Routing Profile** selection which is the reverse route of the flow. Click **Finish**.



Criteria	
Flow Name	SIP_Trunk
Server Configuration	Sprint-SIPT
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Sig_Inside
Signaling Interface	Sig_Outside
Media Interface	Media_Outside
End Point Policy Group	ServiceProvider
Routing Profile	To_Avaya-SM
Topology Hiding Profile	ServiceProvider
File Transfer Profile	None

The following screen summarizes the Server Flows configured in the sample configuration.



Hover over a row to see its description.

Server Configuration: Sprint-SIPT

Priority	Flow Name	URI Group	Transport	Remote Subnet	Received Interface	Signaling Interface	Media Interface	End Point Policy Group	Routing Profile	Topology Hiding Profile	File Transfer Profile		
1	SIP_Trunk	*	*	*	Sig_Inside	Sig_Outside	Media_Outside	ServiceProvider	To_Avaya -SM	ServiceProvider	None	✎	✕

Server Configuration: asm

Priority	Flow Name	URI Group	Transport	Remote Subnet	Received Interface	Signaling Interface	Media Interface	End Point Policy Group	Routing Profile	Topology Hiding Profile	File Transfer Profile			
1	Avaya_SM	*	*	*	Sig_Outside	Sig_Inside	Media_Inside	Avaya	To_ServiceProvider	Avaya-SM	None	✎	✕	✚

7.14. Signaling Manipulations

The Signaling Manipulation feature allows the ability to add, change and delete any of the headers in a SIP message. This feature allows configuration of such manipulations 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 EMS GUI. The Avaya SBCE appliance then interprets this script at the given entry point or "hook point" of the call flow.

For the compliance test it was necessary to create a SigMa script so that the mobile phone configured for EC500 (Extension to Cellular) would be recognized as an extension on the system for any inbound calls placed by the mobile phone. Sprint sends a cookie in the SIP Contact header of the initial INVITE of an inbound call with no SIP PAI header. When there is no PAI header present, Communication Manager looks at the Contact header to match the number of the caller for EC500 origination, and therefore was not able to recognize the mobile phone number due to the cookie in the Contact header.

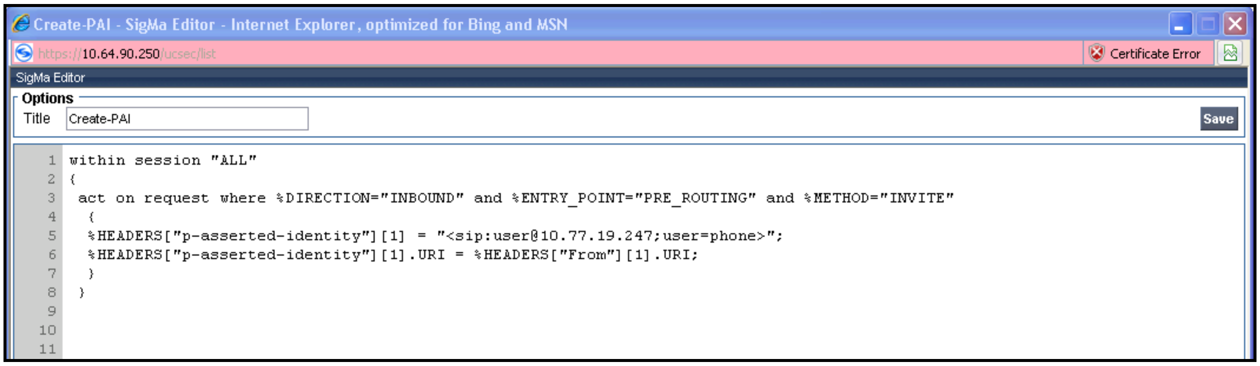
Rather than change the Contact header, which could have unintended results, it was a better solution to simply create a PAI header and modify it with information present in the SIP From header.

To create a new Signaling Manipulation, navigate to **UC-Sec Control Center →Global Profiles → Signaling Manipulation** and click on **Add Script** (not shown). A new blank SigMa Editor window will pop up. For more information on Signaling Manipulation see **Reference** [**8**].

First the SigMa script will create the PAI header and then it will modify the PAI header to contain information present in the From header. The script is broken down as follows:

- **within session "All"** Manipulations are applied to all SIP sessions.
- **act on request** Manipulations will be applied to requests.
- **%DIRECTION="INBOUND"** Applied to messages entering the Avaya SBCE.
- **%ENTRY_POINT="PRE_ROUTING"** The "hook point" to apply the script before the SIP message has routed through Avaya SBCE.
- **%METHOD="INVITE"** Apply manipulations only to the SIP INVITE method.

In the body of the Signaling Manipulation script, the P-Asserted Identity (PAI) header will first be created (first **%HEADERS** line) and then modified (second **%HEADERS** line). The PAI header will be created with a format of ***<sip:user@10.77.19.247;user=phone>*** and will then be modified by replacing the information in the PAI header with information contained in the SIP From header.

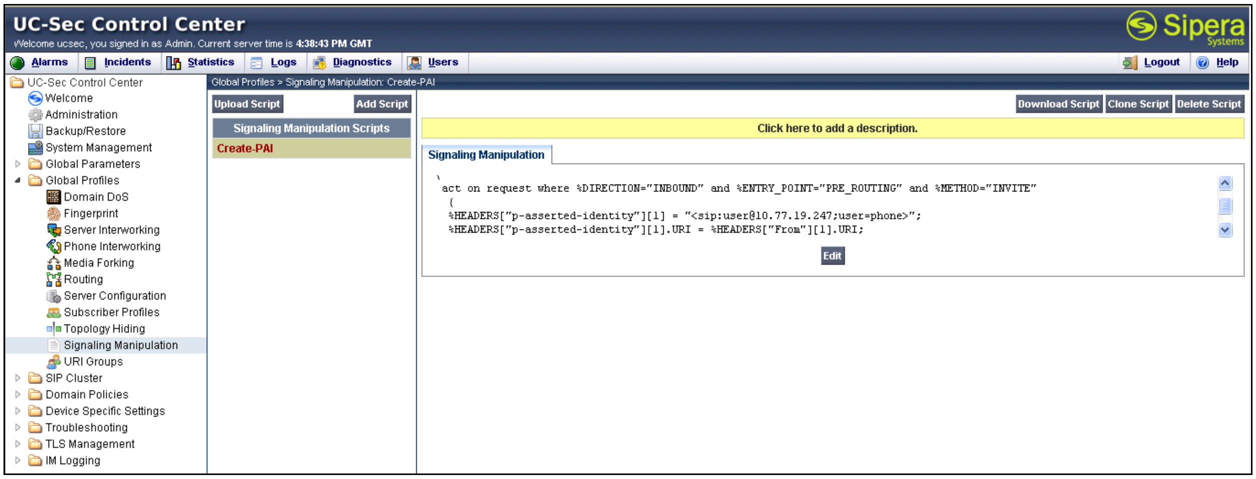


Create-PAI - SigMa Editor - Internet Explorer, optimized for Bing and MSN
https://10.64.90.250/ucsec/list
Certificate Error
SigMa Editor
Options
Save

Title: Create-PAI

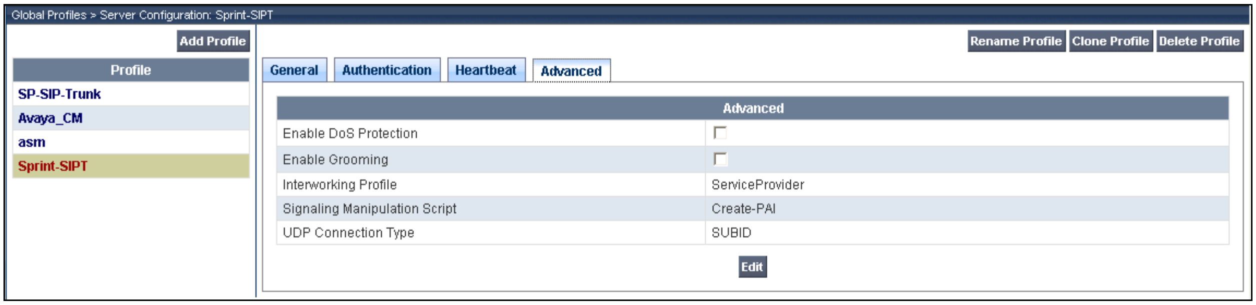
```
within session "ALL"
{
 act on request where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING" and %METHOD="INVITE"
  {
  %HEADERS["p-asserted-identity"][1] = "<sip:user@10.77.19.247;user=phone>";
  %HEADERS["p-asserted-identity"][1].URI = %HEADERS["From"][1].URI;
  }
 }
```

The following screen shows the finished Signaling Manipulation Script **Create-PAI**.



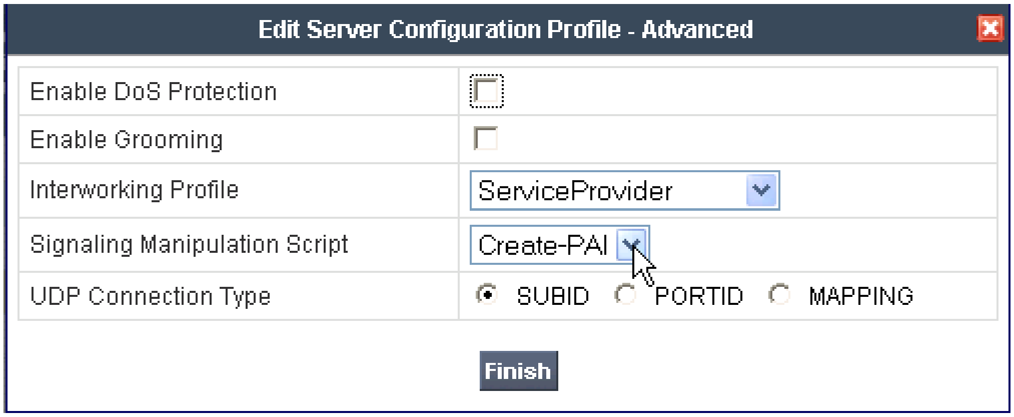
```
 \
 act on request where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING" and %METHOD="INVITE"
  {
  %HEADERS["p-asserted-identity"][1] = "<sip:user@10.77.19.247;user=phone>";
  %HEADERS["p-asserted-identity"][1].URI = %HEADERS["From"][1].URI;
```

After the SigMa script has been created it needs to be applied to the **Server Configuration**. Navigate to **Global Profiles→Server Configuration** (not shown). Click on the **Advanced** tab for the service provider, in this case ***Sprint-SIPT***, and then click **Edit** as shown below:



Advanced	
Enable DoS Protection	□
Enable Grooming	□
Interworking Profile	ServiceProvider
Signaling Manipulation Script	Create-PAI
UDP Connection Type	SUBID

Next, select the appropriate SigMa script from the drop-down menu. For the compliance test, the SigMa script ***Create-PAI*** was used as shown below:



Enable DoS Protection	□
Enable Grooming	□
Interworking Profile	ServiceProvider
Signaling Manipulation Script	Create-PAI
UDP Connection Type	● SUBID ○ PORTID ○ MAPPING

Note, the script is applied to the service provider **Server Configuration** so that manipulations can occur as the SIP messages arrive at the Avaya SBCE, and before any routing decisions have been made.

8. Sprint SIP Trunking Configuration

To use Sprint SIP Trunking, a customer must request the service from Sprint using the established sales and provisioning processes.

During the signup process, Sprint will require that the customer provide the public IP address used to reach the Avaya SBCE at the edge of the enterprise and information related to SIP configuration supported by the enterprise. Sprint will provide the IP address of the Sprint SIP trunk proxy/SBC, transport protocol and listening port for the SIP connection to the enterprise, and Direct Inward Dialed (DID) numbers assigned to the enterprise. This information is used to complete the configurations of Communication Manager, Session Manager, and the Avaya SBCE discussed in the previous sections.

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

9. Verification and Troubleshooting

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

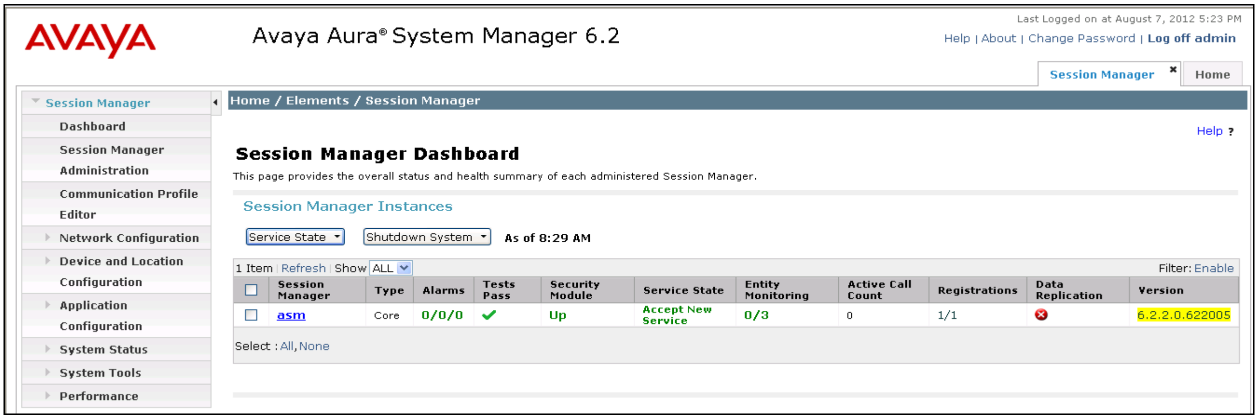
Verification Steps:

1. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active with 2-way audio path.
2. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call remains active with 2-way audio path.
3. Verify that the user on the PSTN can end an active call by hanging up.
4. Verify that an endpoint at the enterprise site can end an active call by hanging up.

Troubleshooting:

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

2. Session Manager:
    - **System State** – Navigate to **Home → Elements → Session Manager**, as shown below. Verify that for the Session Manager of interest, all Entity Links are in service, a green check mark is placed under **Tests Pass** and the **Service State** is **Accept New Service**.



Last Logged on at August 7, 2012 5:23 PM

Home / Elements / Session Manager

Session Manager Dashboard

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

Session Manager Instances

As of 8:29 AM

1 Item | Refresh | Show ALL

Filter: Enable

☐	Session Manager	Type	Alarms	Tests Pass	Security Module	Service State	Entity Monitoring	Active Call Count	Registrations	Data Replication	Version
☐	asm	Core	0/0/0	✓	Up	Accept New Service	0/3	0	1/1	☒	6.2.2.0.622005

Select : All, None

AVAYA
Avaya Aura® System Manager 6.2
Help | About | Change Password | Log off admin
Session Manager
Home
Session Manager
Dashboard
Session Manager Administration
Communication Profile Editor
Network Configuration
Device and Location Configuration
Application Configuration
System Status
System Tools
Performance
Help ?
Service State
Shutdown System

- **traceSM -x** – Session Manager command line tool for traffic analysis. Log in via SSH to the Session Manager management interface and become the root user. Run the command ***traceSM –x*** to start the Session Manager traceSM tool.
- **Call Routing Test** - The Call Routing Test verifies routing for a particular source and destination. To run the routing test, navigate to **Home → Elements →**

Session Manager → System Tools → Call Routing Test. Enter the requested data to run tests.

3. Avaya SBC for Enterprise
    - **OPTIONS** – Disable the SBC-sourced OPTIONS to the trunk server (see **Section 7.4.2**) and use a network sniffer like Wireshark to verify that the service provider network will receive OPTIONS forwarded by the SBC from the enterprise site as a result of the SIP Entity Monitoring configured for Session Manager. Reversely, when the service provider network responds to the OPTIONS from Session Manager, the SBC will pass the response to Session Manager.
    - **Incidents** – From the admin web interface of the Avaya SBCE, open the Incidents report by clicking the **Incidents** button on the menu bar. Look for any errors.

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager R6.2, Avaya Aura® Session Manager R6.2 and Avaya Session Border Controller for Enterprise R4.0.5-Q09 to Sprint SIP Trunking. Sprint SIP Trunking is a SIP-based Voice over IP service for customers ranging from small businesses to large enterprises providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks.

11. References

The Avaya product documentation is available at http://support.avaya.com unless otherwise noted.

Avaya Aura® Session Manager/System Manager

[1] *Administering Avaya Aura® Session Manager*, Document ID 03-603324, Release 6.2, July 2012
[2] *Implementing Avaya Aura® Session Manager*, Doc ID 03-603473 Release 6.2, July 2012
[3] *Maintaining and Troubleshooting Avaya Aura® Session Manager*, Doc ID 03-603325, Issue 4.1, March 2012
[4] *Administering Avaya Aura® System Manager*, Document Number 03-603324, Release 6.2, July 2012

Avaya Aura® Communication Manager

[5] *Administering Avaya Aura® Communication Manager*, Document ID 03-300509, Issue 7.0, Release 6.2, July 2012
[6] *Programming Call Vectors in Avaya Aura® Call Center*, 6.0, June 2010

Avaya Session Border Controller for Enterprise

Product documentation for UC-Sec can be obtained from Sipera using the link at http://www.sipera.com.

[7] *E-SBC 1U Installation Guide, Release 4.0.5,* Part Number: 101-5225-405v1.00, Release Date: November 2011
[8] *E-SBC Administration Guide, Release 4.0.5,* Part Number: 010-5424-405v1.00, Release Date: November 2011

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