Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 5.2.1, Avaya Aura® Session Manager 6.2, and Avaya Session Border Controller for Enterprise with Verizon Business IP Trunk SIP Trunk Service – Issue 1.0

Abstract

These Application Notes illustrate a sample configuration using Avaya Aura® Session Manager Release 6.2 and Avaya Aura® Communication Manager Release 5.2.1 with the Verizon Business Private IP (PIP) IP Trunk service.

The Verizon Business IP Trunk service offer referenced within these Application Notes is designed for business customers with an Avaya SIP trunk solution. The service provides local and/or long distance PSTN calling via standards-based SIP trunks directly, without the need for additional TDM enterprise gateways or TDM cards and the associated maintenance costs.

The configuration and software versions described in these Application Notes have not yet been independently certified by Verizon labs. These Application Notes can be used to facilitate customer engagements via the Verizon field trial process, pending Verizon labs independent certification.

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., utilizing a Verizon Business Private IP (PIP) circuit connection to the production Verizon Business IP Trunking service.

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1. Introduction

These Application Notes illustrate a sample configuration using Avaya Aura® Session Manager Release 6.2 and Avaya Aura® Communication Manager Release 5.2.1 with the Verizon Business Private IP (PIP) IP Trunk service. The Verizon Business IP Trunk service provides local and/or long-distance calls (with PSTN endpoints) via standards-based SIP trunks.

The configuration and software versions described in these Application Notes have not yet been independently certified by Verizon labs. These Application Notes can be used to facilitate customer engagements via the Verizon field trial process, pending Verizon labs independent certification.

1.1. Interoperability Compliance Testing

Compliance testing scenarios for the configuration described in these Application Notes included the following:

- Inbound and outbound voice calls between telephones controlled by Avaya Aura® Communication Manager and the PSTN can be made using G.711MU or G.729A codecs.
- Direct IP-to-IP Media (also known as “Shuffling”) when applicable.
- DTMF using RFC 2833
  - Outbound call to PSTN application requiring post-answer DTMF (e.g., an IVR or voice mail system)
  - Inbound call from PSTN to Avaya CPE application requiring post-answer DTMF (e.g., Avaya Modular Messaging, Avaya vector digit collection steps)
- Additional PSTN numbering plans (e.g., International, operator assist, 411, 711)
- Hold / Retrieve with music on hold
- Call transfer using two approaches
  - REFER approach (Communication Manager Network Call Redirection flag on trunk group form set to “y”)
  - INVITE approach (Communication Manager Network Call Redirection flag on trunk group form set to “n”)
- Conference calls
- SIP Diversion Header for call redirection
  - Call Forwarding
  - EC500

1.2. Support

1.2.1 Avaya

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

1.2.2 Verizon

For technical support on Verizon Business IP Trunk service offer, visit online support at http://www.verizonbusiness.com/us/customer/

1.3. Known Limitations

The following limitations are noted for the sample configuration described in these Application Notes:

- Avaya Aura® Communication Manager 5.2.1 does not support the use of SIP phones and the H.323 IP phones simultaneously in the sample configuration; therefore, the configuration of SIP phones is not covered by these Application Notes.
- Emergency 911/E911 Services Limitations and Restrictions- Although Verizon provides 911/E911 calling capabilities, 911 capabilities were not tested, therefore it is the Customer's responsibility to ensure proper operation with its equipment/software vendor.
- If calls requiring in-band DTMF (rather than RFC 2833 signaling) will be required, the "DTMF over IP" parameter on the Avaya Aura® Communication Manager SIP signaling group carrying such calls can be set to "in-band" rather than "rtp-payload". If the Communication Manager SIP signaling group is set to "rtp-payload", and a call is established using RFC 2833, Communication Manager will not subsequently switch to using "in-band" procedures to signal DTMF. Avaya is considering an enhancement for a future release of Communication Manager that would allow a call initially established with RFC 2833 to switch to using in-band DTMF based on subsequent SIP SDP exchanges.
- Verizon Business IP Trunking service does not support G.729B codec.
- Verizon has recently begun to offer T.38 as a fax option for SIP trunks. This native T.38 implementation from Verizon has some restrictions for robust interoperability with Avaya products. There are both short-term and longer-term solution choices available.

  <u>Short-Term:</u>
  Use an approved SIP gateway to provide full T.38, and optionally, support Verizon's specialized G.711 offer for fax transport. One example is AudioCodes' MP-114 SIP gateway running version 6.20A.035.001 or higher with Communication Manager 5.2.1 SP-12, 6.0.1 SP-6 or higher. Other short-term options include using Verizon's TDM services or special TDM routing between locations.

  <u>Longer-Term:</u>
  By mid-2013, Avaya should have software options to fully interoperate with both Verizon fax offers of T.38 and their specialized G.711 service. More information is available in GRIP-4852. With GRIP-4852 functionality, there is no need to have a front-end SIP gateway.

Note – These Application Notes describe the provisioning used for the sample configuration shown in **Figure 1**. Other configurations may require modifications to the provisioning described in this document.

2. Reference Configuration

Figure 1 illustrates the sample configuration used for the testing. The Avaya CPE location simulates a customer site. The PIP service defines a secure MPLS connection between the Avaya CPE T1 connection and the Verizon service node.

The Avaya SBCE receive traffic from the Verizon Business IP Trunk service on port 5060 and send traffic to the Verizon Business IP trunk service on port 5071, using UDP protocol for network transport (required by the Verizon Business IP Trunk service). The Verizon Business IP Trunk service provided Direct Inward Dial (DID) 10 digit numbers. These DID numbers were mapped by Avaya Aura® Session Manager or Avaya Aura® Communication Manager to Avaya telephone extensions.

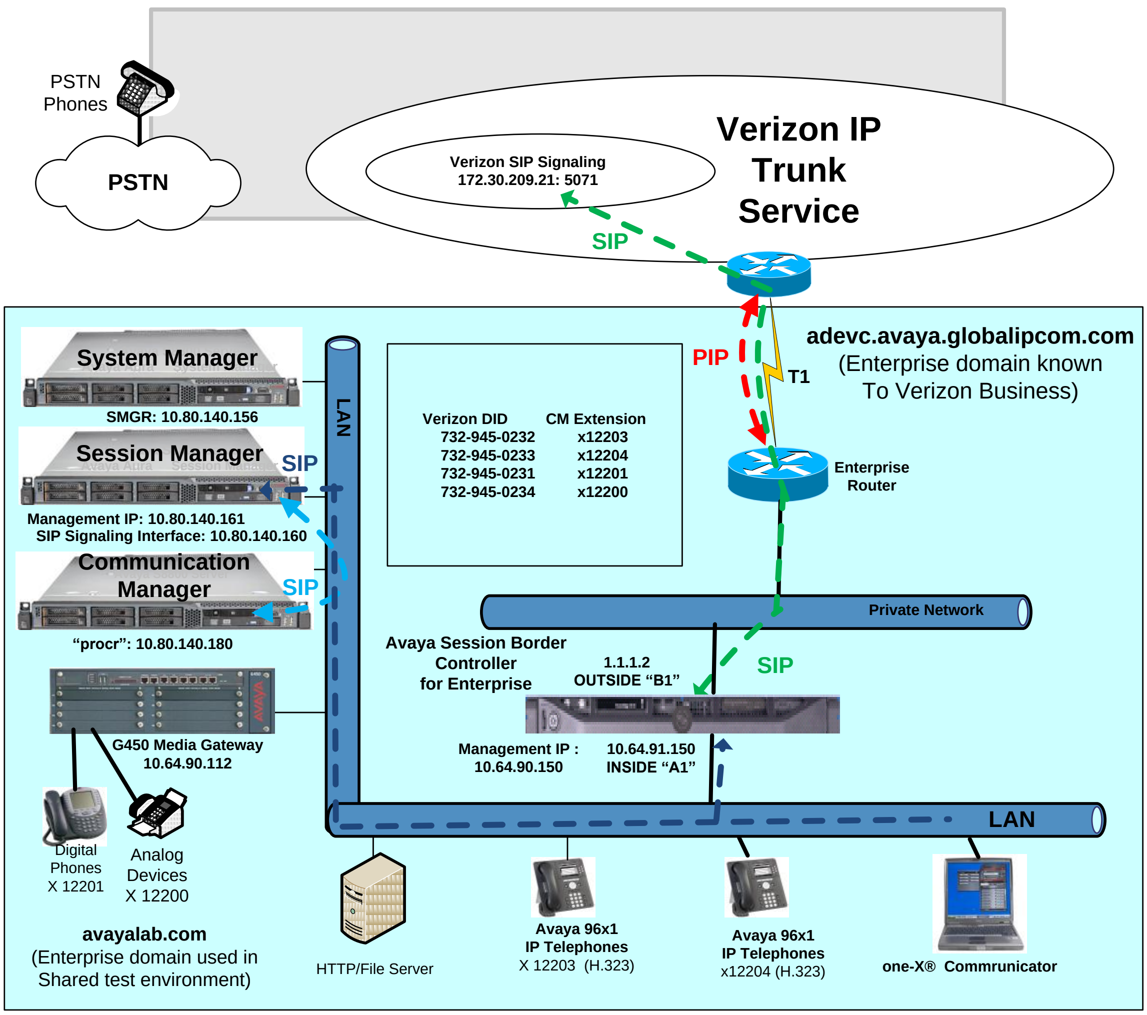


Figure 1: Avaya Interoperability Test Lab Configuration

The Verizon Business IP Trunk service used FQDN *pcelban0001.avayalincroft.globalipcom.com*. The Avaya CPE environment was known to Verizon Business IP Trunk service as FQDN *adevc.avaya.globalipcom.com.* Access to the Verizon Business IP Trunk service was added to a configuration that already used domain "avayalab.com" at the enterprise. As such, Session Manager or the SBCE are used to adapt the "avayalab.com" domain to the domain known to Verizon. These Application Notes indicate a configuration that would not be required in cases where the CPE domain in Communication Manager and Session Manager match the CPE domain known to the Verizon Business IP Trunk service.

Note – The Fully Qualified Domain Names and IP addressing specified in these Application Notes apply only to the reference configuration shown in **Figure 1**. Verizon Business customers will use their own FQDNs and IP addressing as required.

In summary, the following components were used in the reference configuration.

- Verizon Business IP Trunk network Fully Qualified Domain Name (FQDN)
  - *pcelban0001.avayalincroft.globalipcom.com*
- Avaya CPE Fully Qualified Domain Name (FQDN) known to Verizon
  - *adevc.avaya.globalipcom.com*
- Session Border Controllers for Enterprise
- Avaya Aura® Communication Manager Release 5.2.1, SP 13
- Avaya Aura® Session Manager Release 6.2
- Avaya 96X1 Series IP telephones using the H.323 software bundle
- Avaya 9600 Series IP telephones using the H.323 software bundle
- Avaya Digital Phones
- Avaya Analog Phones

2.1. History Info and Diversion Headers

The Verizon Business IP Trunk service does not support SIP History Info Headers. Instead, the Verizon Business IP Trunk service requires that SIP Diversion Header be sent for redirected calls. The Avaya Aura® Communication Manager SIP trunk group form provides options for specifying whether History Info Headers or Diversion Headers are sent.

If Avaya Aura® Communication Manager sends the History Info Header, Avaya Aura® Session Manager can convert the History Info header into the Diversion Header. This is performed by specifying the "*Verizon Adapter*" adaptation in Avaya Aura® Session Manager.

Communication Manager call forwarding or Extension to Cellular (EC500) features may be used for the call scenarios involving Diversion Header.

3. Equipment and Software Validated

The following equipment and software were used in the sample configuration.

Equipment:	Software:
Avaya Aura® Communication Manager	Release 5.2.1 load 016.4 SP 13
Avaya Aura® System Manager	6.2
Avaya Aura® Session Manager	6.2
G450 Gateway	3.1.20.1
Avaya Session Border Controller for Enterprise	4.0.5Q09
Avaya 9600-Series Telephones (H.323)	96xx-IPT-H323-R3_1_3-112211
Avaya 96X1- Series Telephones (H323)	96x1-IPT-H323-R6_0_5-091911
Avaya 2400-Series and 6400-Series Digital Telephones	N/A
Okidata Analog Fax	N/A
Avaya One-X Communicator (H.323)	6.1.3.08_SP3-Patch2-35791

Table 1: Equipment and Software Used in the Sample Configuration

4. Configure Avaya Aura® Communication Manager Release 5.2.1

This section describes the procedure for configuring Avaya Aura® Communication Manager for SIP Trunk service. A SIP trunk is established between Communication Manager and Avaya Aura® Session Manager for use by signaling traffic to and from Verizon.

Note - The initial installation, configuration, and licensing of the Avaya servers and media gateways for Communication Manager are assumed to have been previously completed and are not discussed in these Application Notes.

4.1. Verify Licensed Features

The Communication Manager License file controls customer capabilities. Contact an authorized Avaya representative for assistance if a required feature needs to be enabled.

On **Page 2** of the ***display system-parameters customer-options*** form, verify that the **Maximum Administered SIP Trunks** are sufficient for the combination of trunks to the Verizon Business IP Trunk service offer and any other SIP applications. Each call from a non-SIP endpoint to the Verizon Business IP Trunk service uses one SIP trunk for the duration of the call.

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 8000  0
           Maximum Concurrently Registered IP Stations: 18000 2
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 128   0
  Max Concur Registered Unauthenticated H.323 Stations: 18000 0
                        Maximum Video Capable Stations: 18000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 5000  283
  Maximum Administered Ad-hoc Video Conferencing Ports: 8000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 10    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.)
```

On **Page 3** of the ***display system-parameters customer-options*** form, verify that **ARS** is enabled.

```
display system-parameters customer-options                      Page   3 of  11
                                OPTIONAL FEATURES

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

On **Page 4** of the ***display system-parameters customer-options*** form, verify that the **Enhanced EC500**, **IP Trunks**, **IP Stations**, and **ISDN-PRI** features are enabled. If the use of SIP REFER messaging or send-only SDP attributes will be required verify that the **ISDN/SIP Network Call Redirection** feature is enabled.

```
display system-parameters customer-options                     Page   4 of  11
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                               IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                          ISDN Feature Plus? y
                  Enhanced EC500? y       ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                            ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                  ISDN-PRI? y
              ESS Administration? n              Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                     Malicious Call Trace? y
     External Device Alarm Admin? n                 Media Encryption Over IP? y
  Five Port Networks Max Per MCC? n    Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? n                 Multifrequency Signaling? y
      Global Call Classification? n          Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y       Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? n              Multimedia IP SIP Trunking? y
                       IP Trunks? y

           IP Attendant Consoles? y
```

On **Page 5** of the ***display system-parameters customer-options*** form, verify that the **Private Networking** and **Processor Ethernet** features are enabled.

```
display system-parameters customer-options                     Page   5 of  11
                                OPTIONAL FEATURES

               Multinational Locations? n          Station and Trunk MSP? y
 Multiple Level Precedence & Preemption? n     Station as Virtual Extension? y
                    Multiple Locations? y
                                           System Management Data Transfer? n
         Personal Station Access (PSA)? y              Tenant Partitioning? y
                       PNC Duplication? n      Terminal Trans. Init. (TTI)? y
                  Port Network Support? y              Time of Day Routing? y
                       Posted Messages? y      TN2501 VAL Maximum Capacity? y
                                                      Uniform Dialing Plan? y
                    Private Networking? y    Usage Allocation Enhancements? y
               Processor and System MSP? y
                    Processor Ethernet? y               Wideband Switching? n
                                                                  Wireless? n
                         Remote Office? n
         Restrict Call Forward Off Net? y
                 Secondary Data Module? y
```

4.2. Processor Ethernet Configuration on Common Server

The **add ip-interface procr** or **change ip-interface procr** command can be used to configure the Processor Ethernet (PE) parameters. The following screen shows the parameters used in the reference configuration.

- Verify that **Enable Interface?, Allow H.323 Endpoints?,** and **Allow H248 Gateways?** Fields are set to **Y**.
- Assign a network region (e.g. **1**).
- Use default values for the remaining parameters.

```
change ip-interface pro                                          Page   1 of 1
                                 IP INTERFACES


                 Type: PROCR
                                                   Target socket load: 1700

      Enable Interface? y                         Allow H.323 Endpoints? y
                                                   Allow H.248 Gateways? y
        Network Region: 1                           Gatekeeper Priority: 5
```

4.3. Dial Plan

In the reference configuration the Avaya CPE environment uses five digit local extensions, such as 12xxx . Trunk Access Codes (TAC) are 4 digits in length and begin with *. The Feature Access Code (FAC) to access ARS is the single digit 9. The Feature Access Code (FAC) to access AAR is the single digit 8. The dial plan illustrated here is not intended to be prescriptive; any valid dial plan may be used.

The dial plan is modified with the ***change dialplan analysis*** command as shown below.

```
change dialplan analysis                                      Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                  Location:  all          Percent Full:    0

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
     1            5    ext
     2            5    ext
     4            4    ext
     5            4    ext
     6            5    ext
     7            4    ext
     8            1    fac
     9            1    fac
     *            4    dac
     #            4    fac
```

4.4. Node Names

Node names are mappings of names to IP addresses that can be used in various screens. The following ***change node-names ip*** output shows relevant node-names in the sample configuration. As shown in bold, the node name for Session Manager is "SM" with IP address 10.80.140.160. The node name and IP address for the Processor Ethernet "procr" is 10.80.140.180.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
MM                  205.3.3.55
SM                  10.80.140.160
default             0.0.0.0
procr               10.80.140.180
```

4.5. Network Regions for Gateway, Telephones

Network regions provide a means to logically group resources. In the shared Communication Manager configuration used for the testing, the Avaya G450 Media Gateway is in region 1.

Non-IP telephones (e.g., analog, digital) derive network region and location configuration from the Avaya gateway to which the device is connected. The following display command shows that media gateway 3 is an Avaya G450 Media Gateway configured for network region 1. It can also be observed that the **Controller IP Address** is the Avaya Processor Ethernet (10.80.140.180), and that the gateway IP address is 10.64.90.112. These fields are not configured in this screen, but rather simply display the current information for the gateway.

```
change media-gateway 3                                          Page   1 of   1
                              MEDIA GATEWAY
         Number: 3                                 Registered?  y
           Type: g450               FW Version/HW Vintage: 31 .22 .0  /1
           Name: G450-1                    MGP IP Address: 10 .64 .90 .112
      Serial No: 11N510735839       Controller IP Address: 10 .80 .140.180
   Encrypt Link? y                            MAC Address: b4:b0:17:90:82:50
 Network Region: 1    Location: 1               Enable CF? n
                                                Site Data:
  Recovery Rule: 1

Slot   Module Type            Name                    DSP Type  FW/HW version
 V1:                                                  MP80      69   6
 V2:                                                  MP80      69   6
 V3:                                                  MP80      69   6
 V4:                                                  MP80      69   6
 V5:
 V6:
 V7:   MM712                  DCP MM
 V8:   MM711                  ANA MM                Max Survivable IP Ext: 8
 V9:   gateway-announcements  ANN VMM
```

The bottom of the screen shows the gateway has a MM712 media module supporting Avaya digital phones in slot v7, a MM711 supporting analog devices in slot v8, and the capability to provide announcements and music on hold via "gateway-announcements" in logical slot v9.

IP telephones can be assigned a network region based on an IP address mapping. The network region can also associate the IP telephone to a location for location-based routing decisions. The following screen illustrates a subset of the IP network map configuration used to verify these Application Notes. If the IP address of a registering IP Telephone does not appear in the ip-network-map, the phone is assigned the network region of the "gatekeeper" (e.g., CLAN or PE) to which it registers. When the IP address of a registering IP telephone is in the ip-network-map, the phone is assigned the network region assigned by the form shown below. For example, the IP address 10.80.150.101 would be mapped to network region 2, based on the bold configuration below. In production environments, different sites will typically be on different networks, and ranges of IP addresses assigned by the DHCP scope serving the site can be entered as one entry in the network map, to assign all telephones in a range to a specific network region.

```
change ip-network-map                                           Page   1 of  63
                               IP ADDRESS MAPPING

                                                Subnet Network     Emergency
 IP Address                                     Bits   Region VLAN Location Ext
 ---------------------------------------------- ------ ------ ---- -------------
 FROM: 10.80.150.100                            /      2      n
   TO: 10.80.150.199
 FROM:                                          /             n
   TO:
```

The following screen shows IP Network Region 2 configuration. In the shared test environment, network region 2 is used to allow unique behaviors for the Verizon test environment. In this example, codec set 1 will be used for calls within region 2. The shared test environment uses the domain "avayalab.com" (i.e., for network region 2 including the region of the Processor Ethernet "procr").

```
change ip-network-region 2                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 2
Location: 1       Authoritative Domain: avayalab.com
    Name: IP Phones
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
           Keep-Alive Count: 5
```

4.6. IP Codec Sets

The following screen shows the configuration for codec set 1, the codec set configured to be used for calls within region 1 and for calls within region 2. In general, an IP codec set is a list of allowable codecs in priority order. Using the example configuration shown below, all calls to and from the PSTN via the SIP trunks would use G.711MU, since G.711MU is preferred by both Verizon and the Avaya ip-codec-set. Any calls using this same codec set that are between devices capable of the G.722-64K codec (e.g., Avaya 9600-Series IP Telephone) can use G.722. Note that if G.711MU is omitted from the list of allowed codecs in ip-codec-set 1, calls from Verizon that are answered by Avaya Modular Messaging will use G450 VoIP resources to convert from G.729a (facing Verizon) to G.711MU (facing Modular Messaging). If G.711MU is included in ip-codec-set 1, then calls from Verizon that are answered by Modular Messaging will not use G450 VoIP resources, but rather be "ip-direct" using G.711MU from Modular Messaging to the inside of the SBCE. Include G.711MU in the ip-codec-set if fax will be used.

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

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.722-64K                   2        20
 2: G.711MU          n          2        20
 3: G.729            n          2        20
 4:
```

On **Page 2** of the form:

- Configure the Fax **Mode** field to "t.38-standard", T.38 is newly supported by Verizon and was tested successfully in this test configuration.
- Configure the Fa**x Redundancy** field to "**0".**

```
change ip-codec-set 1                                           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
```

4.7. SIP Signaling Groups

This section illustrates the configuration of the SIP Signaling Groups. Each signaling group has a **Group Type** of "sip", a **Near-end Node Name** of "procr", and a **Far-end Node Name** of "SM". In the example screens, the **Transport Method** for all signaling groups is "tcp". In production, TLS transport between Communication Manager and Session Manager can be used. The **Enable**

Layer 3 Test field is enabled on each of the signaling groups to allow Communication Manager to maintain the signaling group using the SIP OPTIONS method. Fields that are not referenced in the text below can be left at default values, including **DTMF over IP** set to "rtp-payload", which corresponds to RFC 2833.

The following screen shows signaling group 1. Signaling group 1 will be used for processing PSTN calls to / from Verizon via Session Manager. The **Far-end Network Region** is configured to region 10. Port 5060 has been configured as both the **Near-end Listen Port** and **Far-end Listen Port**. Session Manager will be configured to direct calls arriving from the PSTN with Verizon DID numbers to a route policy that uses a SIP entity link to Communication Manager specifying port 5060. The use of different ports is one means to allow Communication Manager to distinguish different types of calls arriving from the same Session Manager. Other parameters may be left at default values.

```
change signaling-group 1                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 1             Group Type: sip
                        Transport Method: tcp
  IMS Enabled? n
     IP Video? n




   Near-end Node Name: procr                Far-end Node Name: SM
 Near-end Listen Port: 5060               Far-end Listen Port: 5060
                                       Far-end Network Region: 10
Far-end Domain: avayalab.com

                                            Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                 RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                    IP Audio Hairpinning? n
         Enable Layer 3 Test? y                    Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n           Alternate Route Timer(sec): 6
```

4.8. SIP Trunk Groups

This section illustrates the configuration of the SIP Trunk Group corresponding to the SIP signaling group from the previous section.

The following shows **Page 1** for trunk group 1, which will be used for incoming and outgoing PSTN calls from Verizon. The **Number of Members** field defines how many simultaneous calls are permitted for the trunk group. The **Service Type** field should be set to "public-ntwrk" for the trunks that will handle calls with Verizon. The **Direction** has been configured to "two-way" to allow incoming and outgoing calls in the sample configuration.

```
change trunk-group 1                                                Page   1 of  21
                                TRUNK GROUP

Group Number: 1                  Group Type: sip           CDR Reports: y
  Group Name: SIP Trunk to SP           COR: 1        TN: 1        TAC: *101
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n

                                                       Signaling Group: 1
                                                     Number of Members: 10
```

The following shows **Page 2** for trunk group 1; all parameters shown are default values.

```
change trunk-group 1                                                Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

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

 Disconnect Supervision - In? y  Out? y
```

The following shows **Page 3** for trunk group 1. All parameters except those in bold are default values. The **Numbering Format** will use "private" numbering, meaning that the private numbering table would be consulted for any mappings of Communication Manager extensions to alternate numbers to be sent to Session Manager. Optionally, replacement text strings can be configured using the "system-parameters features" screen, such that incoming "private" (anonymous) or "restricted" calls can display an Avaya-configured text string on called party telephones.

```
change trunk-group 1                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                            Maintenance Tests? y



                     Numbering Format: private
                                                  UUI Treatment: service-provider

                                                   Replace Restricted Numbers? y
                                                  Replace Unavailable Numbers? y




 Show ANSWERED BY on Display? y
```

The following shows **Page 4** for trunk group 1. The **PROTOCOL VARIATIONS** page is one reason why it can be advantageous to configure incoming calls from Verizon to arrive on specific signaling groups and trunk groups. Although not strictly necessary, the **Telephone Event Payload Type** has been set to 101 to match Verizon configuration. Setting the **Network Call Redirection** flag to "y" enables advanced services associated with the use of the REFER message, while also implicitly enabling Communication Manager to signal "send-only" media conditions for calls placed on hold at the enterprise site. If neither REFER signaling nor "send-only" media signaling is required, this field may be left at the default "n" value. In the testing associated with these Application Notes, transfer testing using REFER was successfully completed with the **Network Call Redirection** flag set to "y", and transfer testing using INVITE was successfully completed with the **Network Call Redirection** flag set to "n".

For redirected calls, Verizon supports the Diversion header, but not the History-Info header. Communication Manager can send the Diversion header by marking **Send Diversion Header** to "y". Alternatively, Communication can send the History-Info header by setting **Support Request History** to "y", and Session Manager can adapt the History-Info header to the Diversion header using the "VerizonAdapter". In the testing associated with these Application Notes, call redirection testing with Communication Manager sending Diversion Header was completed successfully. Communication Manager configuration was then changed, and call redirection testing with Communication Manager sending History-Info and Session Manager adapting to Diversion Header was completed successfully.

```
change trunk-group 1                                              Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? y
                    Send Diversion Header? y
                  Support Request History? n
             Telephone Event Payload Type: 101
```

4.9. Route Pattern Directing Outbound Calls to Verizon

Route pattern 1 will be used for calls destined for the PSTN via the Verizon IP Trunk service. Digit manipulation can be performed on the called number, if needed, using the **No. Del Dgts** and **Inserted Digits** parameters. Digit manipulation can also be performed by Session Manager.

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1   Pattern Name: To SIP SP
                             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: 1    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                                         none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

4.10. Route Pattern for Internal Calls via Session Manager

Route pattern 3 contains trunk group 3, the "private" tie trunk group to Session Manager. The **Numbering Format**: ***lev0-pvt*** means all calls using this route pattern will use the private numbering table.

```
change route-pattern 3                                          Page   1 of   3
                    Pattern Number: 3   Pattern Name: Route to SM
                             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: 3    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                                lev0-pvt  none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
```

4.11. Private Numbering

The ***change private-unknown-numbering*** command may be used to define the format of numbers sent to Verizon in SIP headers such as the “From” and “PAI” headers. In general, the mappings of internal extensions to Verizon DID numbers may be done in Session Manager (via Digit Conversion in adaptations) or in Communication Manager (via public-unknown-numbering, and incoming call handling treatment for the inbound trunk group).

In the bolded row shown in the example abridged output below, a specific Communication Manager extension (12201) is mapped to a DID number that is known to Verizon for this SIP Trunk connection (7329450231), when the call uses trunk group 1. Alternatively, Communication Manager can send the five digit extension to Session Manager, and Session Manager can adapt the number to the Verizon DID. Both methods were tested successfully.

```
change private-numbering 0                              Page   1 of   2
                        NUMBERING - PRIVATE FORMAT

Ext Ext          Trk       Private        Total
Len Code          Grp(s)     Prefix          Len
 5  12                                 5     Total Administered: 3
 5  122          99                     5       Maximum Entries: 540
 5  12201         1        7329450231       10
 5  12203         1        7329450232       10
 5  12204         1        7329450233       10
```

4.12. ARS Routing for Outbound Calls

Although not illustrated in these Application Notes, location-based routing may be configured so that users at different locations that dial the same telephone number can have calls choose different route-patterns. In these Application Notes, the ARS "all locations" table directs ARS calls to specific SIP Trunks to Session Manager.

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9. The example below shows a subnet of the dialed strings tested as part of the testing. All dialed strings are mapped to route pattern 1 which contains SIP trunk to Session Manager.

```
change ars analysis 303                                        Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    0

          Dialed           Total      Route     Call    Node   ANI
          String          Min  Max   Pattern    Type    Num    Reqd
    303                   10   10     1         hnpa           n
    311                   3    3      1         svcl           n
    411                   3    3      1         svcl           n
    501                   10   10     1         hnpa           n
    511                   3    3      1         svcl           n
    611                   3    3      1         svcl           n
    711                   3    3      1         svcl           n
    720                   10   10     1         hnpa           n
    732                   10   10     1         hnpa           n
```

4.13. Incoming Call Handling Treatment for Incoming Calls

In general, the "incoming call handling treatment" for a trunk group can be used to manipulate the digits received for an incoming call if necessary. Since Avaya Aura® Session Manager is present, Session Manager can also be used to perform digit conversion, and digit manipulation and the Communication Manager incoming call handling table may not be necessary. If the DID number sent by Verizon is unchanged by Session Manager, then the DID number can be mapped to an extension using the incoming call handling treatment of the receiving trunk group. As an example, the following screen illustrates a conversion of DID number 7329450231-34 to extension 12200-12204 respectively.

```
change inc-call-handling-trmt trunk-group 1                    Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number       Del Insert
 Feature        Len      Digits
 public-ntwrk    11 17329450233       all 12204
 public-ntwrk    10 7329450231        all 12201
 public-ntwrk    10 7329450232        all 12203
 public-ntwrk    10 7329450233        all 12204
 public-ntwrk    10 7329450234        all 12200
 public-ntwrk
```

4.14. EC500 Configuration for Diversion Header Testing

When EC500 is enabled for a Communication Manager station, a call to that station will generate a new outbound call from Avaya Aura® Communication Manager to the configured EC500 destination, typically a mobile phone. The following screen shows an example EC500 configuration for the user with station extension 12203. Use the command ***change off-pbx-telephone station mapping x*** where ***x*** is the Communication Manager station (e.g. 12203).

- **Station Extension** – This field will automatically populate.
- **Application** – Enter **"**EC500".
- **Dial Prefix** – Enter a prefix (e.g., 1) if required by the routing configuration.
- **Phone Number** – Enter the phone that will also be called (e.g., 3035380023).
- **Trunk Selection** – Enter "ars". This means ARS will be used to determine how Communication Manager will route to the **Phone Number** destination.
- **Config Set** – Enter "1".
- Other parameters can retain default values.

```
change off-pbx-telephone station-mapping 12203                  Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                Prefix                      Selection   Set     Mode
 12203            EC500         -     3035380023       ars        1
                                -
                                -
```

4.15. Saving Communication Manager Configuration Changes

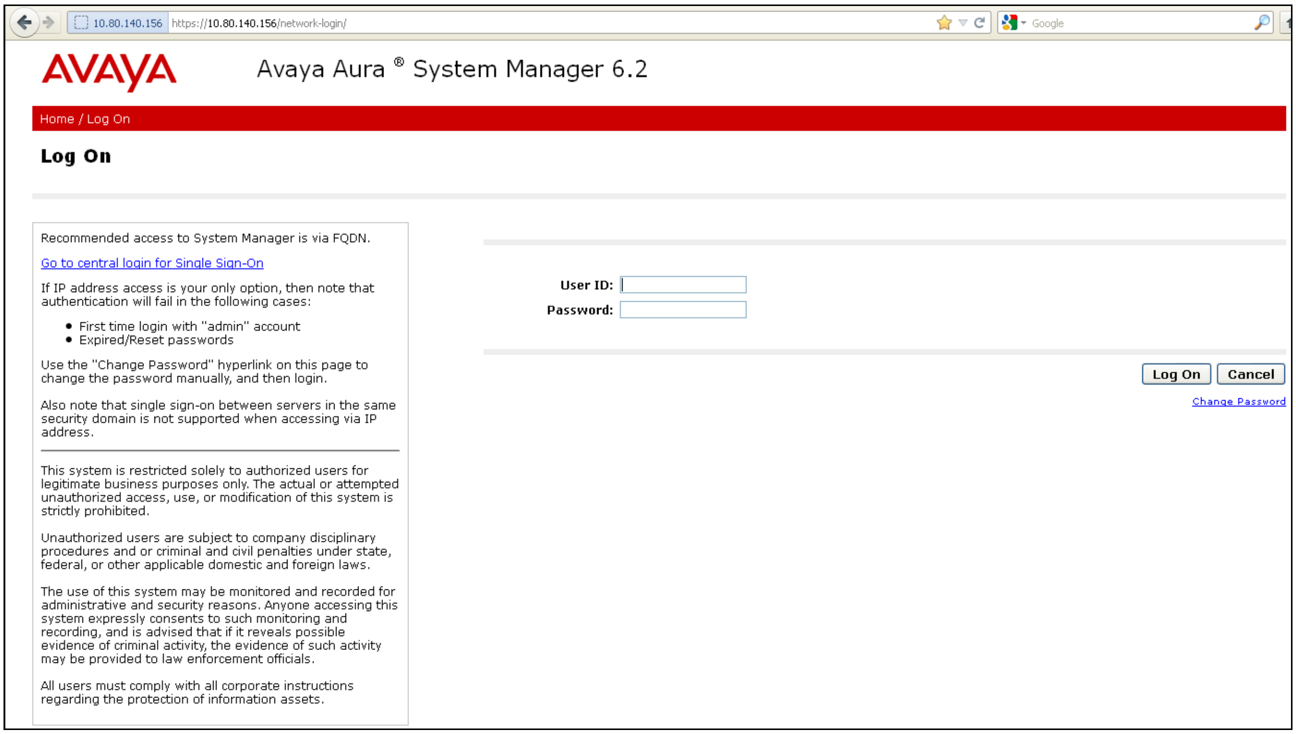
The command ***save translation all*** can be used to save the configuration.

5. Configure Avaya Aura® Session Manager Release 6.2

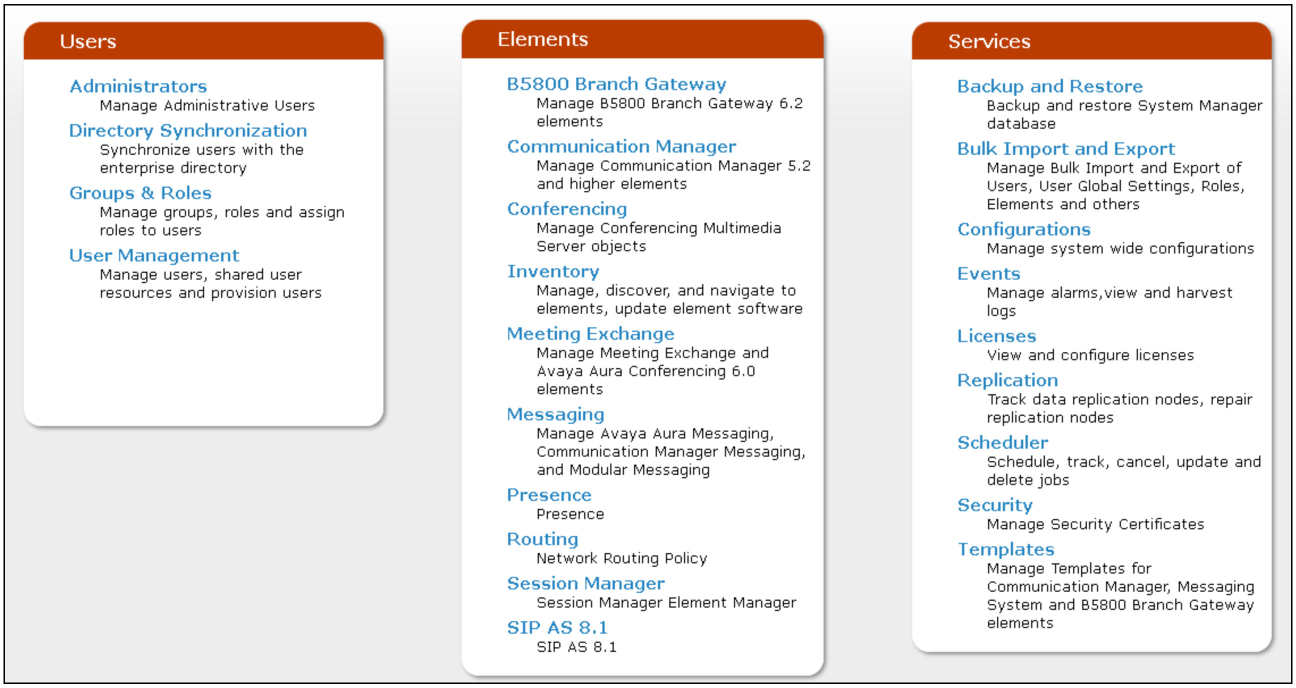
This section illustrates relevant aspects of the Session Manager configuration used in the verification of these Application Notes.

Note – The following sections assume that Session Manager and System Manager have been installed and that network connectivity exists between System Manager and Session Manager.

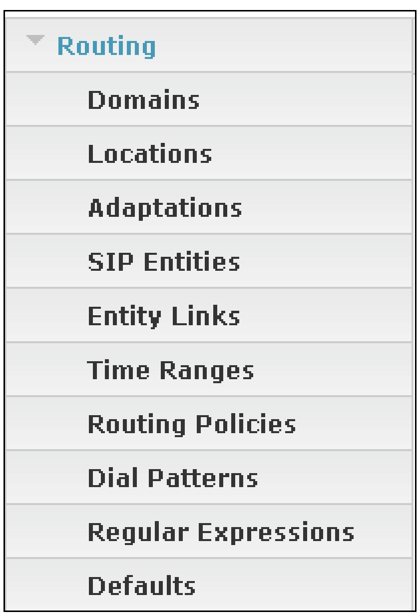
Session Manager is managed via System Manager. Using a web browser, access "https://<ip-addr of System Manager>/SMGR". In the **Log On** screen, enter appropriate **User ID** and **Password** and press the **Log On** button as shown in the example System Manager 6.2 **Log On** screen below.



Once logged in, a **Home Screen** is displayed. An abridged **Home Screen** is shown below.



Under the heading “Elements” in the center, select **Routing.** The screen shown below shows the various sub-headings available on the left hand side menu.



The right side of the screen, illustrated below, outlines a series of steps. The sub-sections that follow are in the same order as the steps outlined under **Introduction to Network Routing Policy** in the abridged screen shown below.

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"

Step 6: Create "Time Ranges"

- Align with the tariff information received from the Service Providers

Step 7: Create "Routing Policies"

- Assign the appropriate "Routing Destination" and "Time Of Day"

(Time Of Day = assign the appropriate "Time Range" and define the "Ranking")

Step 8: Create "Dial Patterns"

- Assign the appropriate "Locations" and "Routing Policies" to the "Dial Patterns"

Step 9: Create "Regular Expressions"

- Assign the appropriate "Routing Policies" to the "Regular Expressions"

Scroll down to review additional information as shown below. In these Application Notes, all steps are illustrated with the exception of Step 9, since “Regular Expressions” were not used.

Each "Routing Policy" defines the "Routing Destination" (which is a "SIP Entity") as well as the "Time of Day" and its associated "Ranking".

IMPORTANT: the appropriate dial patterns are defined and assigned afterwards with the help of the routing application "Dial patterns". That's why this overall routing workflow can be interpreted as

"Dial Pattern driven approach to define Routing Policies"

That means (with regard to steps listed above):

Step 7: "Routing Polices" are defined

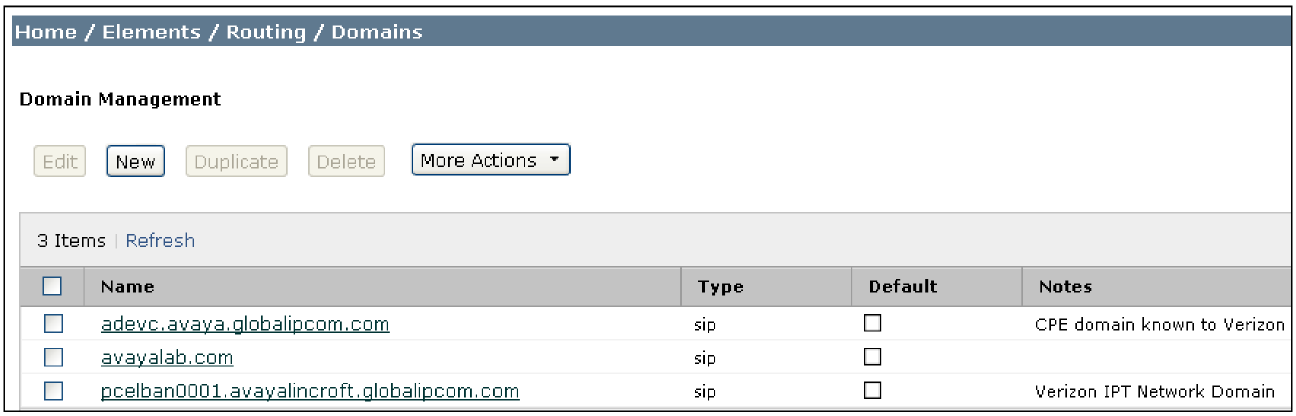
Step 8: "Dial Patterns" are defined and assigned to "Routing Policies" and "Locations" (one step)

Step 9: "Regular Expressions" are defined and assigned to "Routing Policies" (one step)

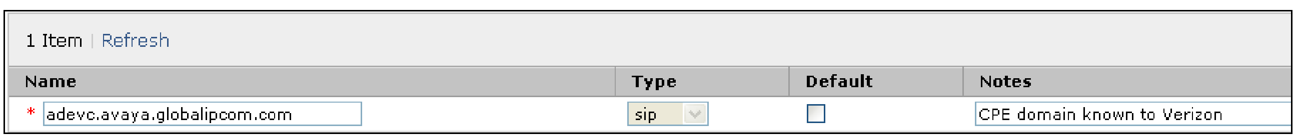
5.1. Domains

To view or change SIP domains, select **Routing → Domains**. Click on the checkbox next to the name of the SIP domain and **Edit** to edit an existing domain, or the **New** button to add a domain. Click the **Commit** button after changes are completed.

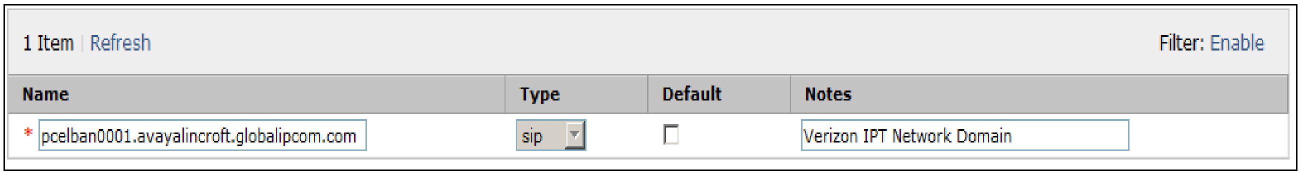
The following screen shows a list of configured SIP domains. The Session Manager used in the verification of these Application Notes was shared among other Avaya interoperability test efforts. The domain "avayalab.com" was used for communication with Avaya SIP Telephones and other Avaya systems and applications. The domain "avayalab.com" is not known to the Verizon production service.



The domain "adevc.avaya.globalipcom.com" is the domain known to Verizon as the enterprise SIP domain. In the sample configuration, Verizon included this domain as the host portion of the Request-URI for inbound DID calls.

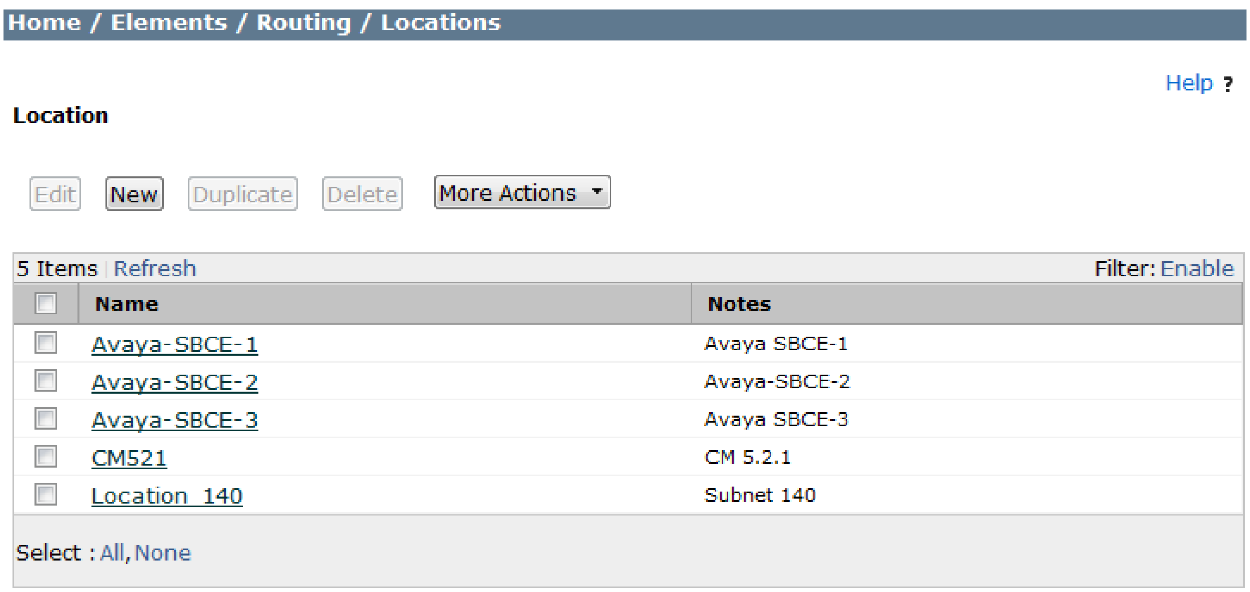


The domain "pcelban0001.avayalincroft.globalipcom.com" is associated with the Verizon network in the sample configuration. For example, for calls from the enterprise site to Verizon, this domain can appear in the Request-URI in the INVITE message sent to Verizon. The following screen shows the relevant configuration.

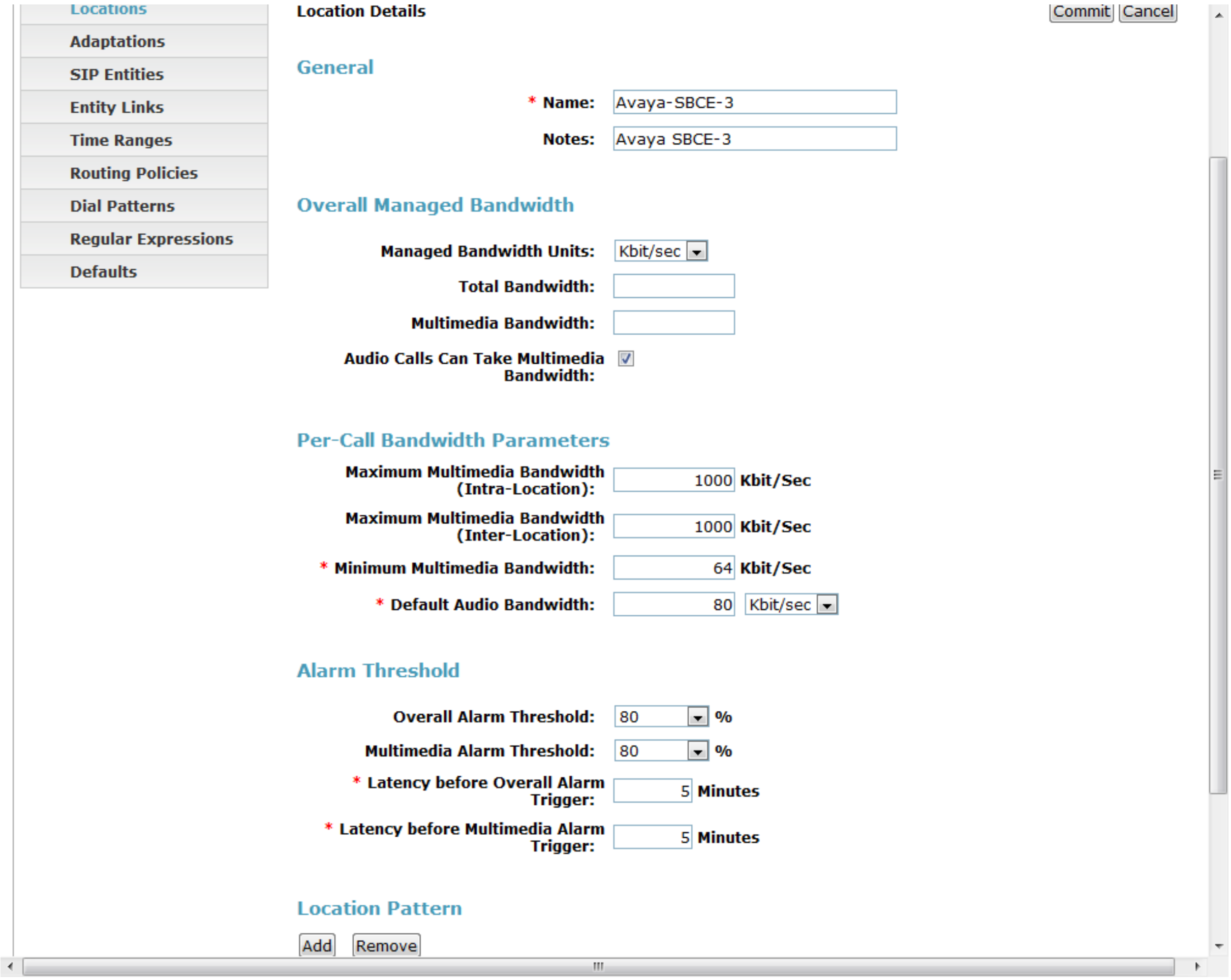


5.2. Locations

To view or change locations, select **Routing → Locations**. The following screen shows an abridged list of configured locations. Click on the checkbox corresponding to the name of a location and **Edit** to edit an existing location, or the **New** button to add a location. Click the **Commit** button after changes are completed. Assigning unique locations can allow Session Manager to perform location-based routing, bandwidth management, and call admission control.



The following image shows the top portion of the screen for the location details for the location named "Avaya-SBCE 3", corresponding to the Avaya SBC for Enterprise relevant to these Application Notes. Later, the location with name "Avaya-SBCE-3" will be assigned to the corresponding SIP Entity.



5.3. Adaptations

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

The following screen shows a portion of the list of adaptations that were available in the sample configuration, not all of which are applicable to these Application Notes.

Home / Elements / Routing / Adaptations

Help ?

Adaptations

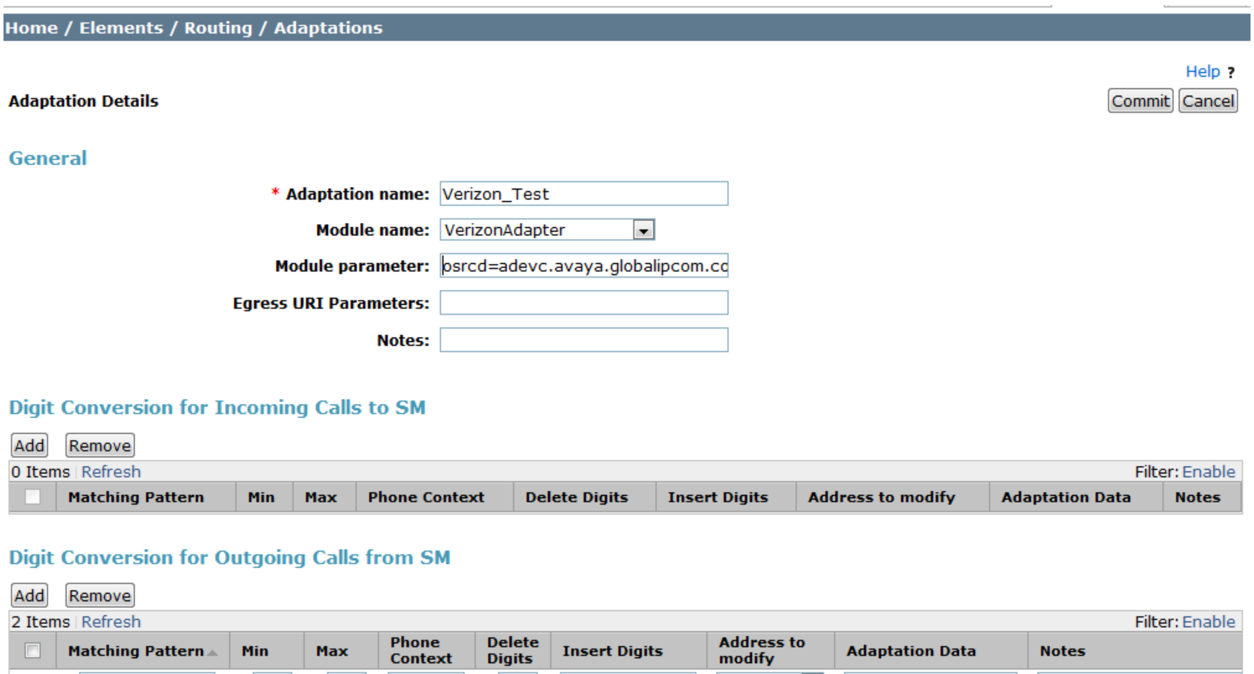
Edit New Duplicate Delete More Actions

9 Items | Refresh Filter: Enable

	Name	Module name	Egress URI Parameters	Notes
	CM-ES-VZ	DigitConversionAdapter		
	CM-ES-VZ-IPCC	DigitConversionAdapter odstd=avayalab.com fromto=true		Verizon IPCC to CM Numbers
	History_Diversion_IPT	VerizonAdapter osrcd=adevc.avaya.globalipcom.com odstd=pcelban0001.avayalincroft.globalipcom.com fromto=true		Verizon adaptation
	IPCC_Verizon_Interop_Lab	VerizonAdapter		
	MM_to_4digits	DigitConversionAdapter fromto=true		converting 5 to 4 digits VM
	SBC-VzB-IPCC	DigitConversionAdapter osrcd=adevc.avaya.globalipccom.com		
	To_VZ	VerizonAdapter odstd=172.30.209.21 fromto=true		
	Verizon_Test	VerizonAdapter		
	Verizon_Unscreened_ANI	VerizonAdapter osrcd=advec.avaya.globalipcom.com odstd=pcelban0001.avayalincroft.globalipcpm.com fromto=true		

Select : All, None

The following screen shows the adaptation details. The adapter named "Verizon_Test" will later be assigned to the SIP Entity for the Avaya SBCE-3, specifying that all communication from the Session Manager to the Avaya SBCE- 3 will use this adapter. This adaptation uses the "Verizon Adapter" and specifies three parameters that are used to adapt the FQDN to the domains expected by the Verizon network in the sample configuration. Again, this may not be required in all networks, but is used here to adapt the avayalab.com domain that is used in the shared test environment among other Avaya interoperability test efforts.



The "**Module parameter:**" line contains the following line:

osrcd=adevc.avaya.globalipcom.com odstd=pcelban0001.avayalincroft.globalipcom.com fromto=true

- `overrideDestinationDomain` : "**osrcd=adevc.avaya.globalipcom.com**". This configuration enables the source domain to be overwritten with "adevc.avaya.globalipcom.com". For example, for outbound PSTN calls from the Avaya CPE to Verizon, the PAI header will contain "adevc.avaya.globalipcom.com" as expected by Verizon.

- `overrideDestinationDomain` : "**odstd=pcelban0001.avayalincroft.globalipcom.com**" This configuration enables the destination domain to be overwritten with "pcelban0001.avayalincroft.globalipcom.com". For example, for outbound PSTN calls from the Avaya CPE to Verizon, the Request-URI header will contain "pcelban0001.avayalincroft.globalipcom.com" as expected by Verizon.

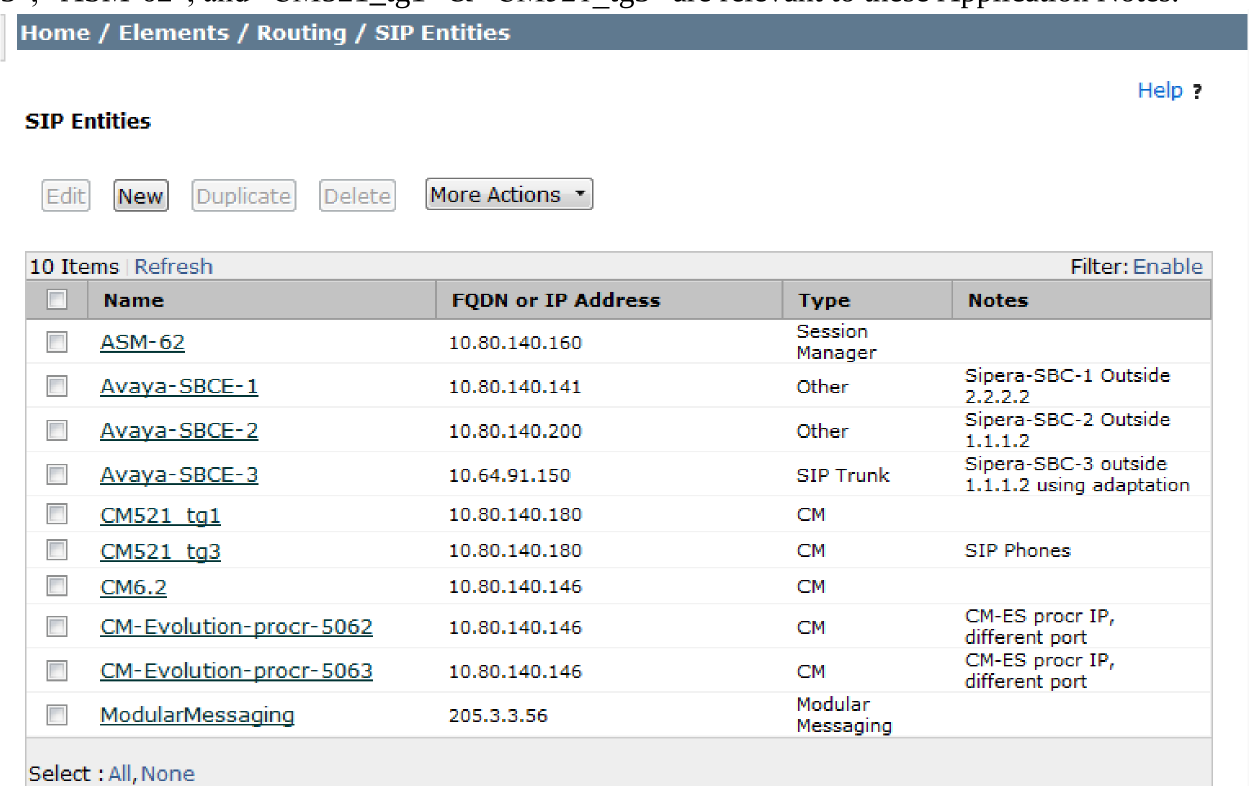
- `Fromto`: The parameter "**fromto=true**" enables Session to modify From and To headers of the message. If omitted or set to any other value, From and To headers will not be modified.

The "History_Diversion_IPT" Module Parameter statement above is overriding avayalab.com with the FQDNs know by Verizon towards the Avaya SBCE. It is also necessary to override the FQDNs known to Verizon back to avayalab.com towards the Communication Manager. This could be done on the next Adaptation "CM-ES-VZ" with the same parameters odstd and osrcd or here in the "History_Diversion_IPT" adapter with the statements:

- `ingressOverrideDestinationDomain:` "**iodstd=avayalab.com**"
- `ingressOverrideDestinationDomain:` "**iosrcd=avayalab.com**"

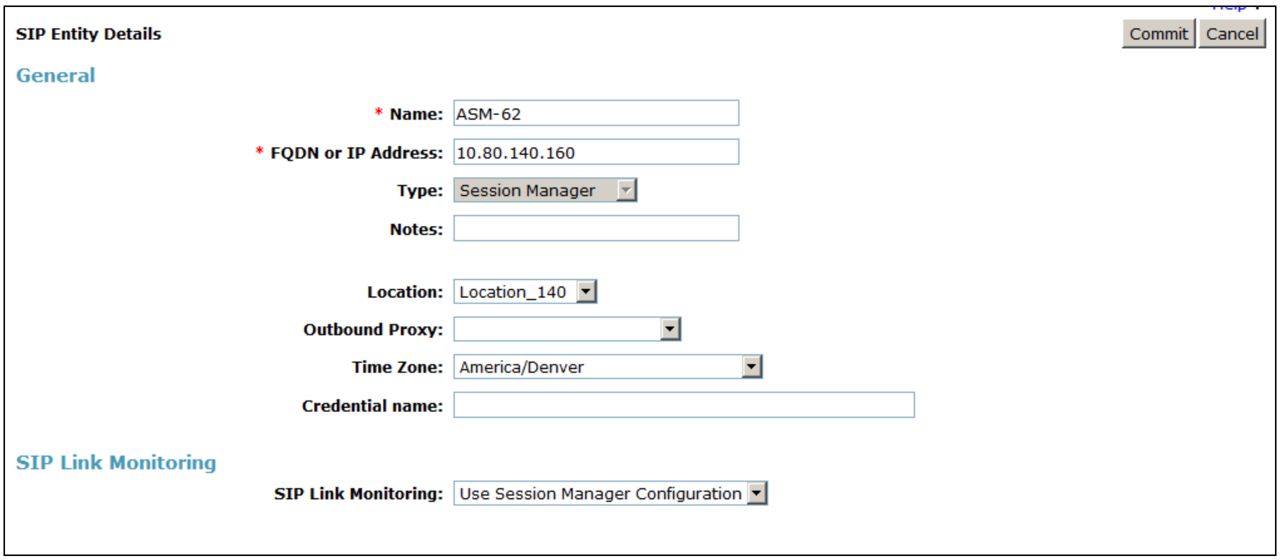
5.4. SIP Entities

To view or change SIP entities, select **Routing → SIP Entities**. Click the checkbox corresponding to the name of an entity and **Edit** to edit an existing entity, or the **New** button to add an entity. Click the **Commit** button after changes are completed. The following screen shows a portion of the list of configured SIP entities. In this screen, the SIP Entities named "Avaya-SBCE 3", "ASM-62", and "CM521_tg1" & "CM521_tg3" are relevant to these Application Notes.

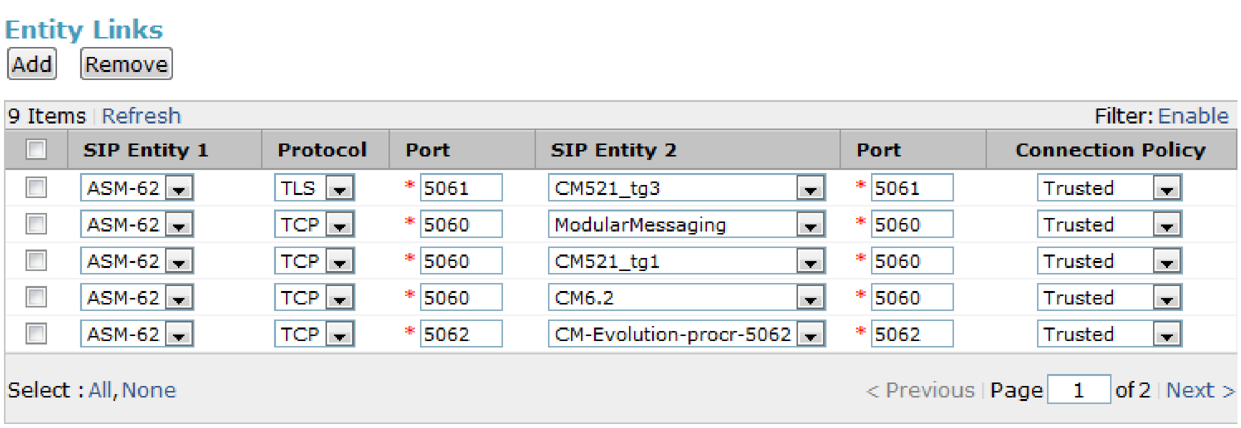


The following screen shows the upper portion of the **SIP Entity Details** corresponding to "ASM-62". The **FQDN or IP Address** field for "ASM-62" is the Session Manager Security Module IP Address (10.80.140.160), which is used for SIP signaling with other networked SIP entities. The **Type** for this SIP entity is "Session Manager". Select an appropriate location for the Session

Manager from the **Location** drop-down menu. In the shared test environment, the Session Manager used location "Location_140". The default **SIP Link Monitoring** parameters may be used. Unless changed elsewhere, links from other SIP entities to this instance of Session Manager will use the default SIP Link Monitoring timers, configurable at the Session Manager level. If desired, these timers may be customized for each entity.

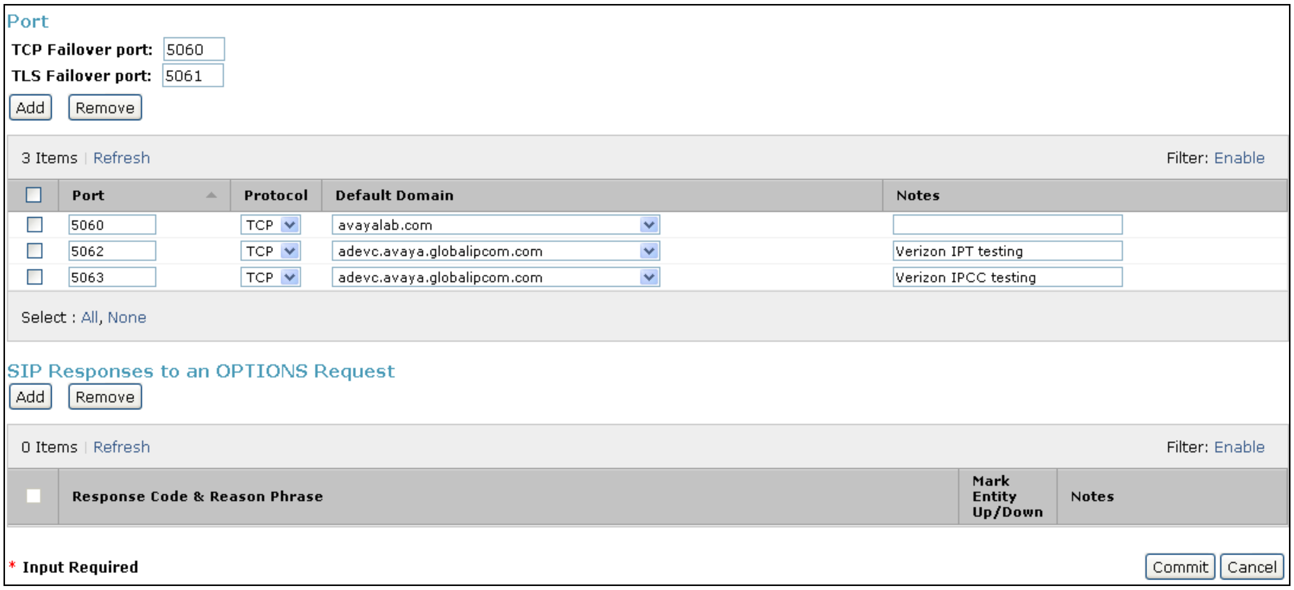


Scrolling down, the following screen shows the middle portion of the **SIP Entity Details**, a listing of the **Entity Links** previously configured for "ASM-62". The links relevant to these Application Notes are described in the subsequent section.

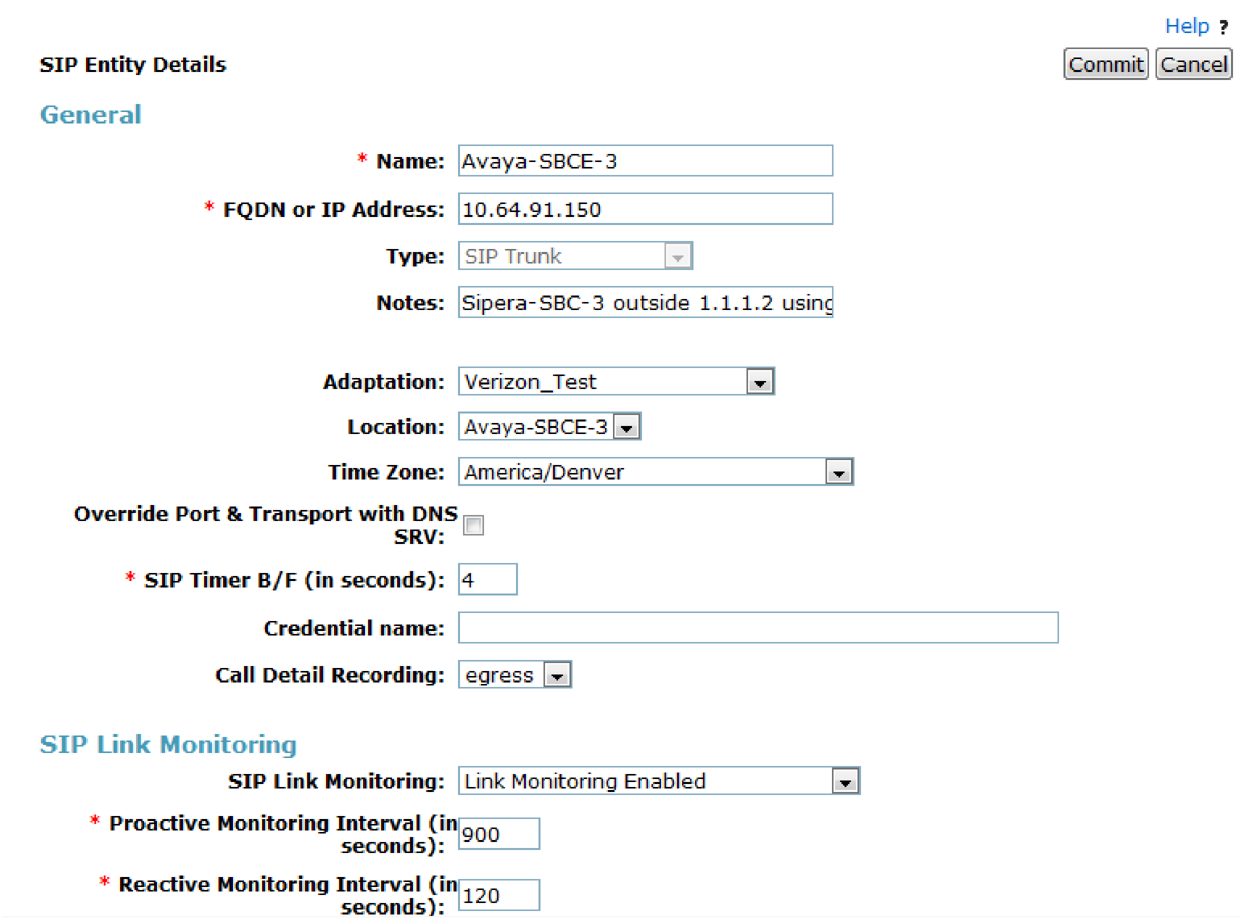


	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
	ASM-62	TLS	* 5061	CM521_tg3	* 5061	Trusted
	ASM-62	TCP	* 5060	ModularMessaging	* 5060	Trusted
	ASM-62	TCP	* 5060	CM521_tg1	* 5060	Trusted
	ASM-62	TCP	* 5060	CM6.2	* 5060	Trusted
	ASM-62	TCP	* 5062	CM-Evolution-procr-5062	* 5062	Trusted

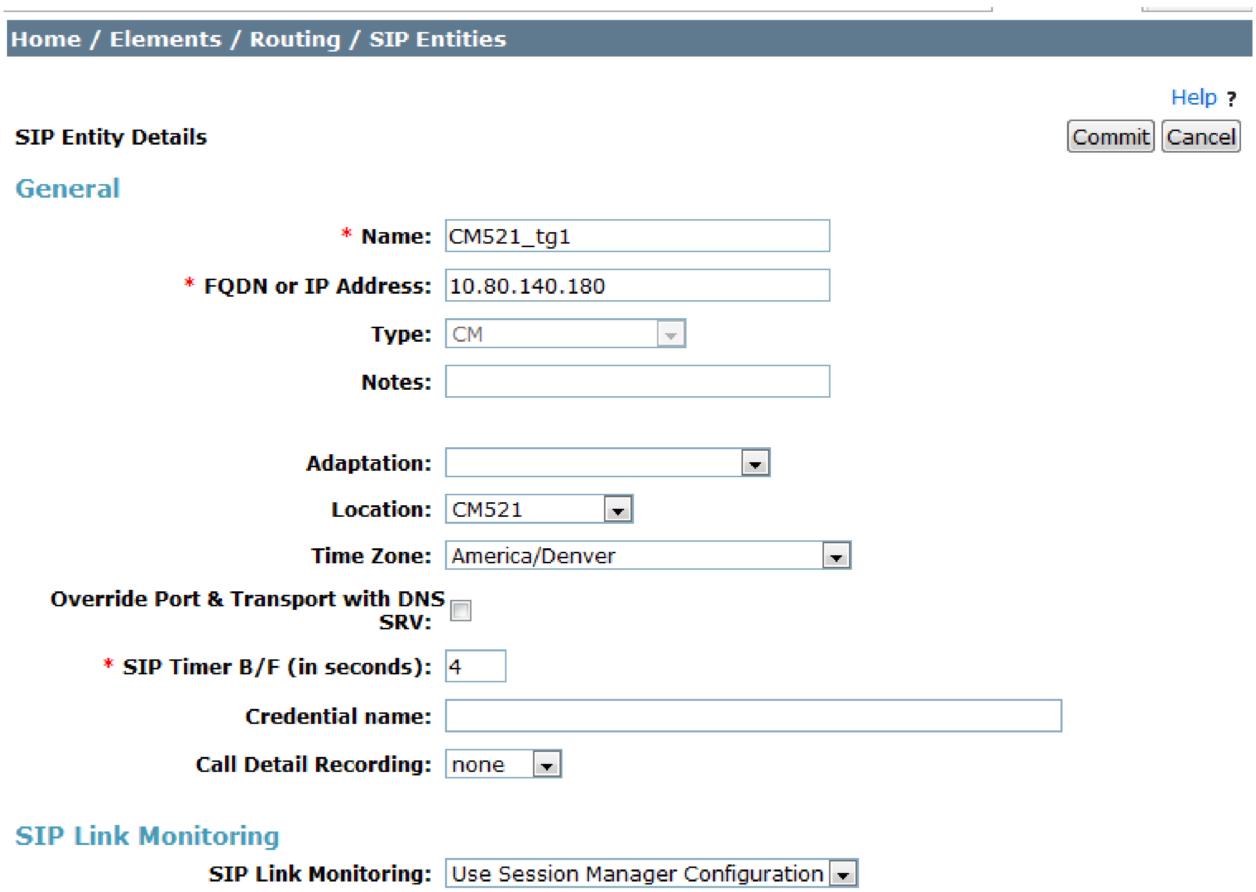
Scrolling down, the following screen shows the lower portion of the **SIP Entity Details**, illustrating the configured ports for "ASM-62". In the sample configuration, TCP port 5060 was already in place for the shared test environment, using **Default Domain** "avayalab.com". Click the **Add** button to configure a new port.



The following screen shows the upper portion of the **SIP Entity Details** corresponding to "Avaya-SBCE 3". The **FQDN or IP Address** field is configured with the Avaya SBC inside IP Address (10.64.91.150). "SIP Trunk" is selected from the **Type** drop-down menu for SBC SIP Entities. This SBCE has been assigned to **Location** "Avaya-SBCE3", and the "Verizon_Test" adapter is applied. Other parameters (not shown) retain default values.



The following screen shows a portion of the **SIP Entity Details** corresponding to a Communication Manager SIP Entity named "CM521_tg1" This is the SIP Entity that was added in the shared environment, after adding the Verizon IP Trunk configurations. The **FQDN or IP Address** field contains the IP Address of the "processor Ethernet" (10.80.140.180). In systems with Avaya G450 Media Gateways containing C-LAN cards, C-LAN cards may also be used as SIP entities, instead of, or in addition to, the "processor Ethernet". "CM" is selected from the **Type** drop-down menu.



5.5. Entity Links

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

Note – In the Entity Link configurations below (and in the Communication Manager SIP trunk configuration), TCP was selected as the transport protocol for the Avaya CPE in the sample configuration. TCP was used to facilitate trace analysis during network verification. TLS may be used between Communication Manager and Session Manager in customer deployments.

The following screen shows a list of configured links. In the screen below, the links named "Sipera-SBCE-3" and "ASM-CM521_tg1" are most relevant to these Application Notes. Each link uses the entity named “ASM-62” as **SIP Entity 1**, and the appropriate entity, such as “CM521_tg1” for **SIP Entity 2**. Note that there are multiple SIP Entity Links, using different TCP ports, linking the same “ASM-62” with the Processor Ethernet of Communication Manager. For example, for one link, named “ASM_to_CM”, both entities use TCP and port 5060. For the entity link used by Verizon IP Trunk named “CM-ES-VZ-5062”, both entities use TCP and port 5062.

Help ?

Entity Links

Edit | New | Duplicate | Delete | More Actions ▾

9 Items | Refresh   Filter: Enable

	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
☐	ASM-62_CM521_tg3_5061_TLS	ASM-62	TLS	5061	CM521_tg3	5061	Trusted	
☐	ASM-62_ModularMessaging_5060_TCP	ASM-62	TCP	5060	ModularMessaging	5060	Trusted	
☐	ASM-CM521_tg1	ASM-62	TCP	5060	CM521_tg1	5060	Trusted	
☐	ASM_to_CM	ASM-62	TCP	5060	CM6.2	5060	Trusted	
☐	CM-ES-VZ-5062	ASM-62	TCP	5062	CM-Evolution-procr-5062	5062	Trusted	
☐	CM-ES-VZ-5063	ASM-62	TCP	5063	CM-Evolution-procr-5063	5063	Trusted	VZ IPCC
☐	Sipera-SBC-1	ASM-62	TCP	5060	Avaya-SBCE-1	5060	Trusted	SBC-Outside-2222
☐	Sipera-SBC-2	ASM-62	TCP	5060	Avaya-SBCE-2	5060	Trusted	SBC-Outisde-1112
☐	Sipera-SBC-3	ASM-62	TCP	5060	Avaya-SBCE-3	5060	Trusted	SBC outside 1112

Select : All, None

5.6. Time Ranges

To view or change Time Ranges, select **Routing → Time Ranges**. The Routing Policies shown subsequently will use the “24/7” range since time-based routing was not the focus of these Application Notes. Click the **Commit** button after changes are completed.

Home / Elements / Routing / Time Ranges

Time Ranges

Edit | New | Duplicate | Delete | More Actions ▾

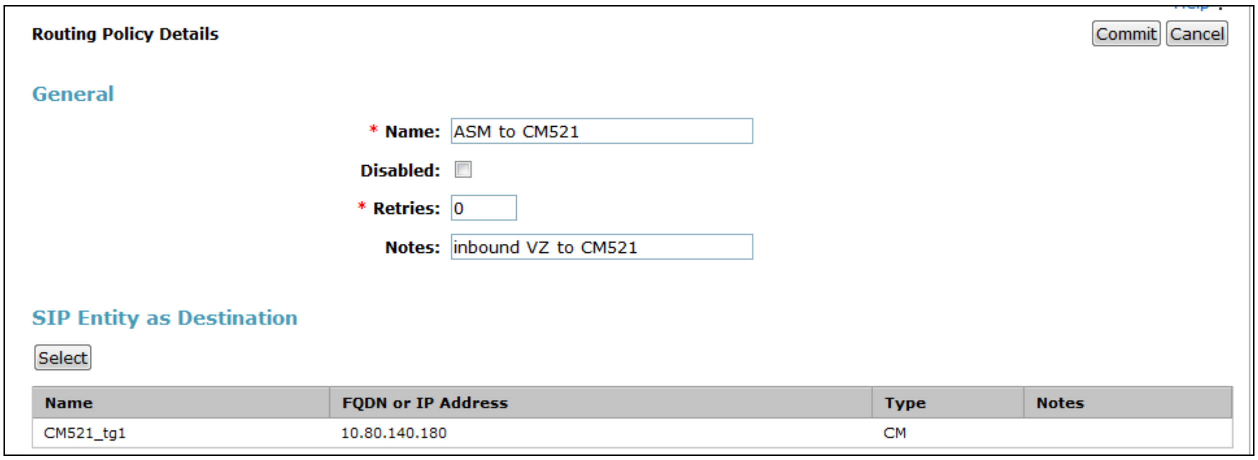
2 Items | Refresh   Filter

	Name	Mo	Tu	We	Th	Fr	Sa	Su	Start Time	End Time	Notes
☐	24/7	☑	☑	☑	☑	☑	☑	☑	00:00	23:59	Time Range 24/7
☐	Anytime	☑	☑	☑	☑	☑	☑	☑	00:00	23:59	24/7

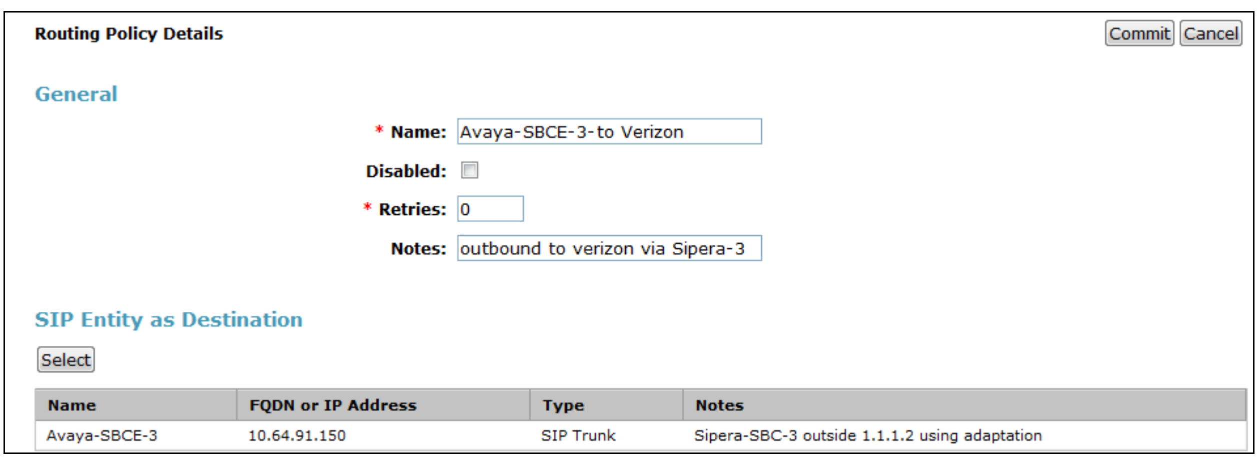
5.7. Routing Policies

To view or change routing policies, select **Routing → Policies**. Click on the checkbox corresponding to the name of a policy and **Edit** to edit an existing policy, or **New** to add a policy. Click the **Commit** button after changes are completed.

The following screen shows the **Routing Policy Details** for the policy named "ASM to CM521" associated with incoming calls from Verizon IP Trunk to Communication Manager. Observe the **SIP Entity as Destination** is the entity named "CM521_tg1" which uses the Communication Manager processor Ethernet IP Address (10.80.140.180).



The following screen shows the **Routing Policy Details** for the policy named "Avaya-SBCE-3-to-Verizon" associated with outgoing calls from Communication Manager to the PSTN via Verizon through the Avaya SBCE. Observe the **SIP Entity as Destination** is the entity named "Avaya-SBCE-3".

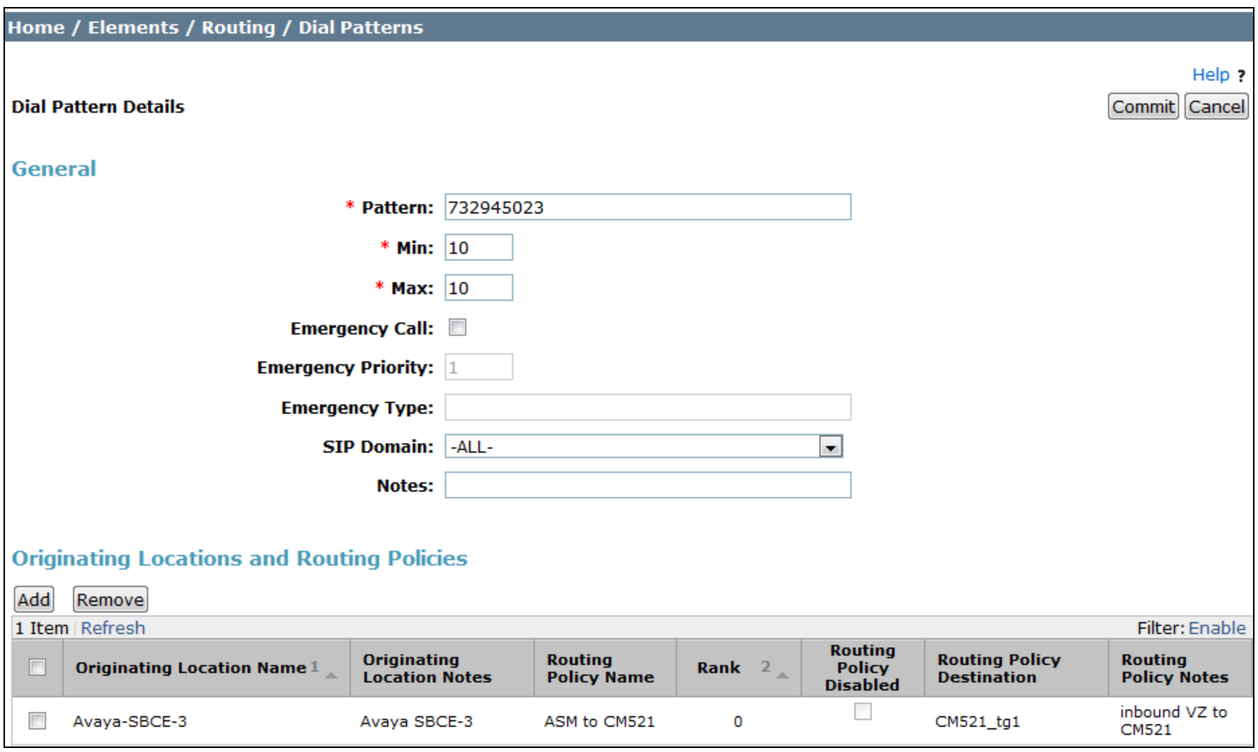


5.8. Dial Patterns

To view or change dial patterns, select **Routing → Dial Patterns**. Click on the checkbox corresponding to the name of a pattern and **Edit** to edit an existing pattern, or **New** to add a pattern. Click the **Commit** button after changes are completed.

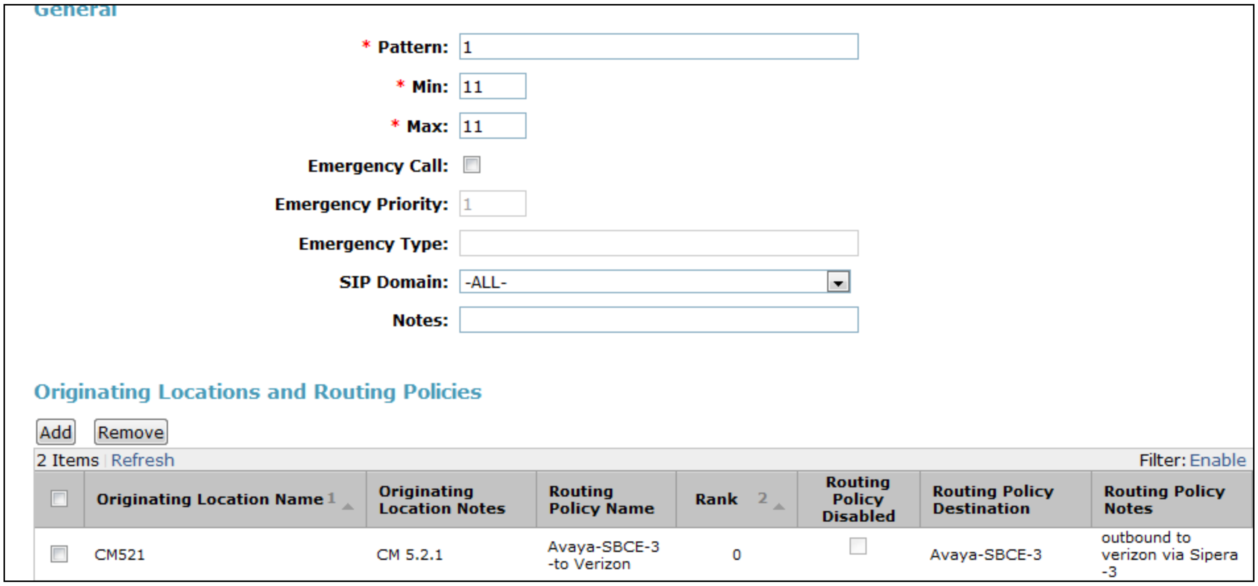
5.8.1 Inbound Call Dial Pattern

The following screen illustrates an example dial pattern used to verify inbound PSTN calls to the enterprise. When a user on the PSTN dials a number assigned to the Verizon IP Trunk service, such as 732-945-0233, Verizon delivers the number to the enterprise, and the Avaya SBCE sends the call to Session Manager. The pattern below matches on 732-945-023. Dial patterns can alternatively match on ranges of numbers (e.g., a DID block). Under **Originating Locations and Routing Policies**, the routing policy named "ASM to CM521 is selected, which sends the call to Communication Manager using port 5062 as described previously. In the configuration, calls to this number from **Originating Location Name** "Avaya-SBCE-3", are routed to Communication Manager.



5.8.2 Outbound Call Dial Pattern

The following screen illustrates an example dial pattern used to verify outbound calls from the enterprise to the PSTN. When a Communication Manager user dials a PSTN number such as 9-1-XXX-XXX-XXXX, Communication Manager sends the call to Session Manager. Session Manager will match the dial pattern shown below and send the call to the Avaya SBCE-3 via the **Routing Policy Name** “Avaya-SBCE-3-to-Verizon”.



6. Avaya Session Border Controller for Enterprise

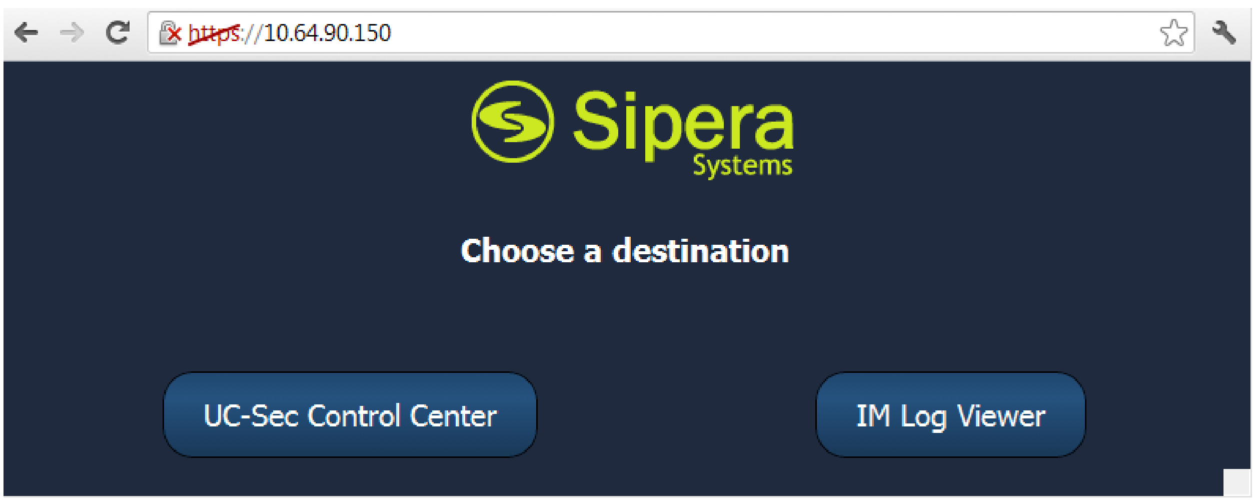
In the sample configuration, an Avaya Session Border Controller for Enterprise is used as the edge device between the Avaya CPE and Verizon Business.

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

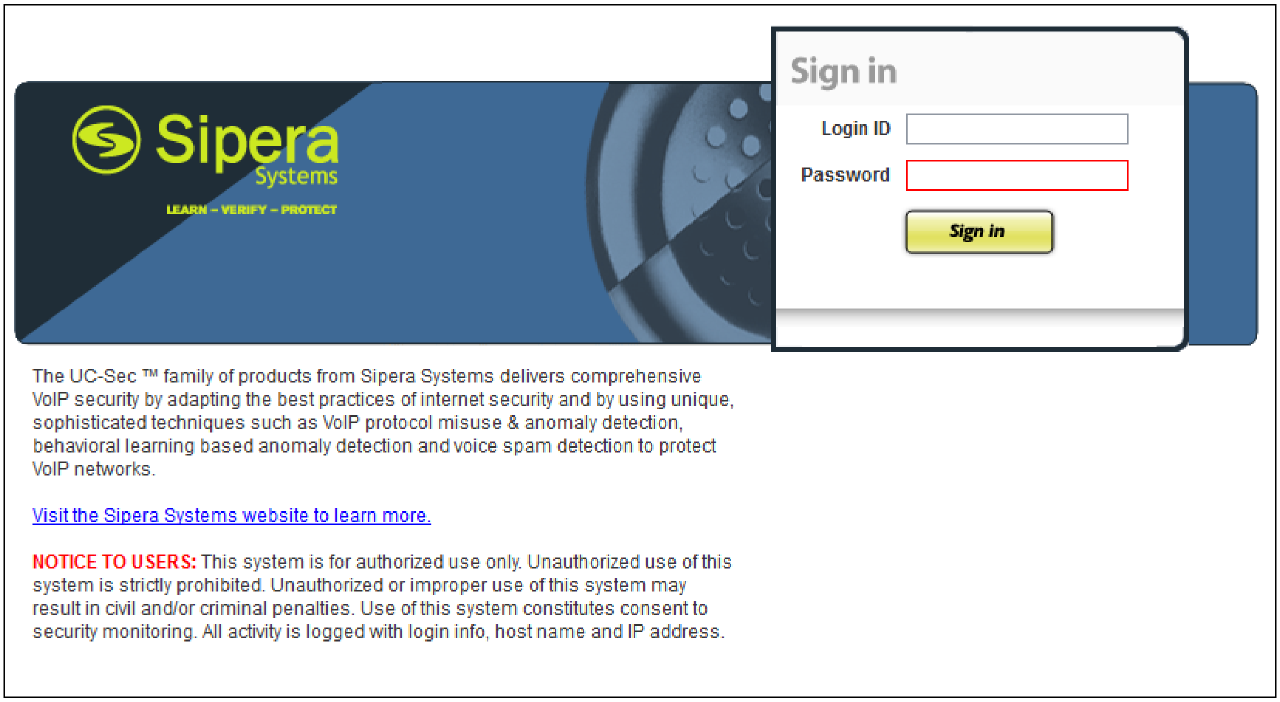
Note – The following Sections describe the provisioning of the Primary SBCE. The configuration of the Secondary SBCE is identical unless otherwise noted (e.g. IP addressing).

6.1. Access the Management Interface

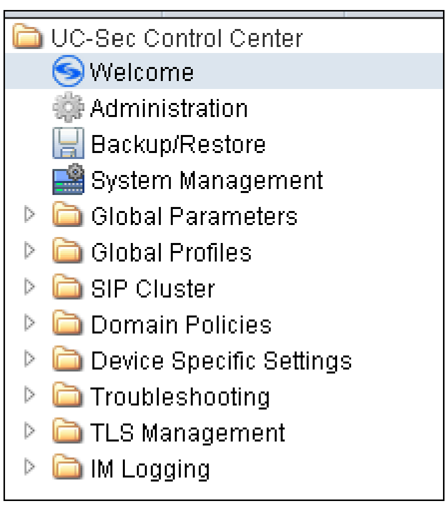
In the sample configuration, the management IP is 10.64.90.150. Access the web management interface by entering https://<ip-address> where <ip-address> is the management IP address assigned during installation. Select **UC-Sec Control Center**.



A log in screen is presented. Enter an appropriate **Login ID** and **Password**.

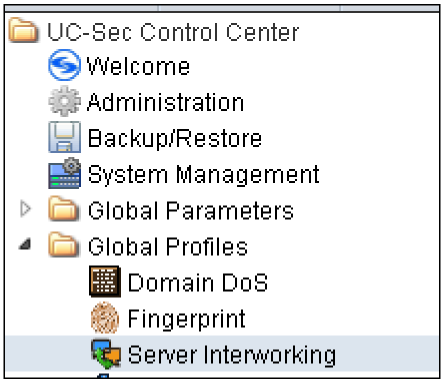


Once logged in, a UC-Sec Control Center screen will be presented. The following image illustrates the menu items available on the left-side of the UC-Sec Control Center screen.



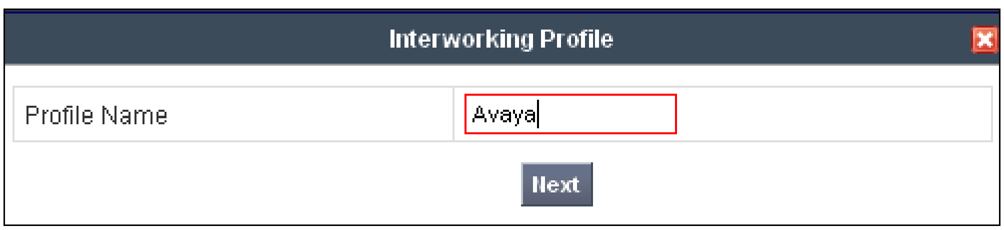
6.2. Global Profiles – Server Interworking

Select **Global Profiles → Server Interworking** from the left-side menu as shown below.

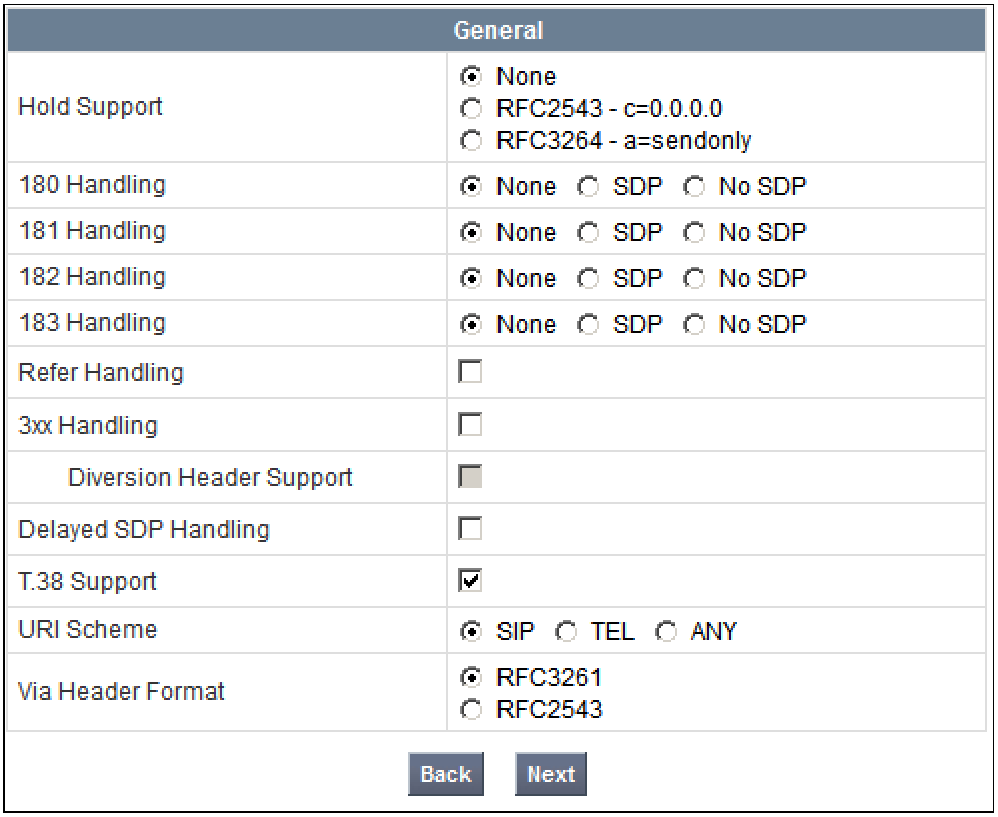


6.2.1 Server Interworking - Avaya

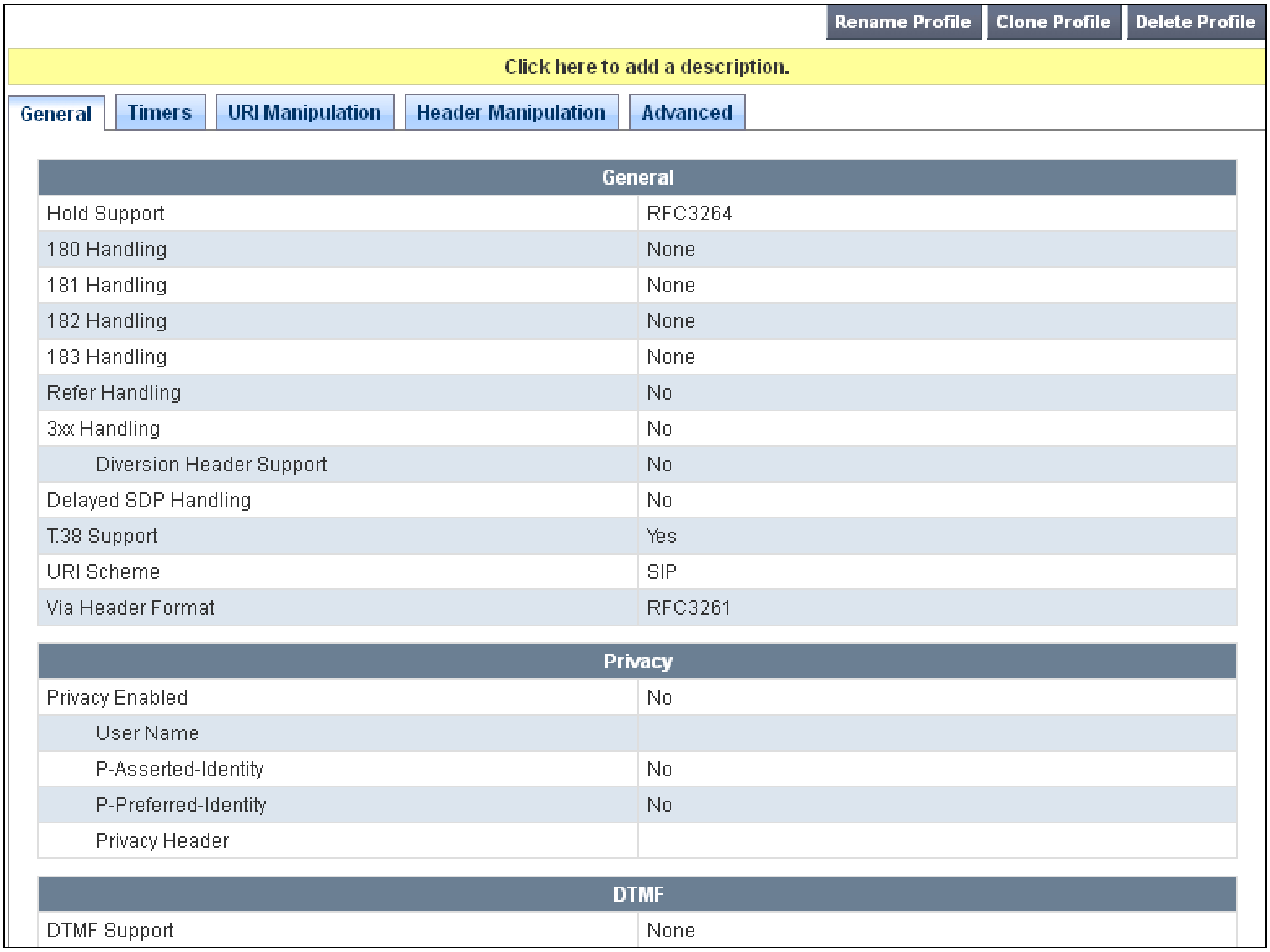
Click the **Add Profile** button (not shown) to add a new profile or select an existing interworking profile. If adding a profile, a screen such as the following is displayed. Enter an appropriate **Profile Name** such as "Avaya" shown below. Click **Next**.



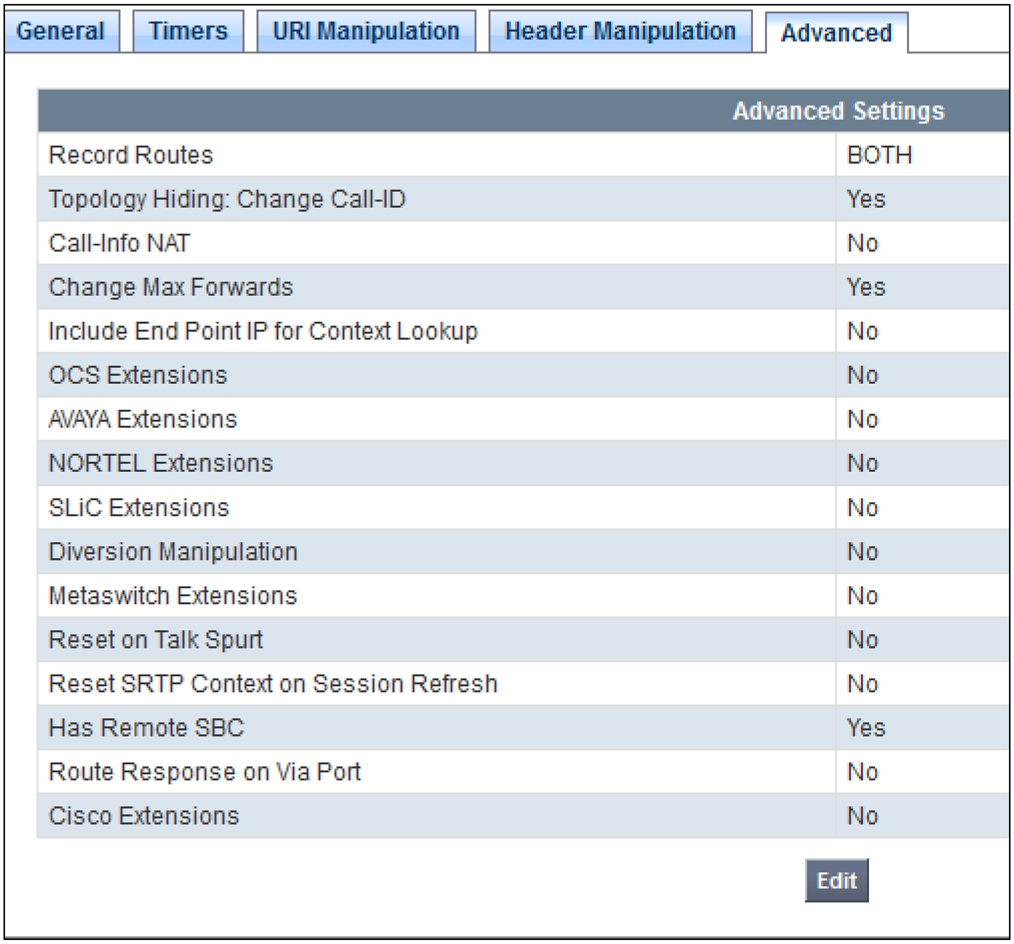
The following screens illustrate the “General” parameters used in the sample configuration for the Interworking Profile named “Avaya”.   Most parameters retain default values. In the sample configuration, **T.38 support** was checked (optional), and **Hold Support** was set for None.



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 interworking profile named “Avaya.”



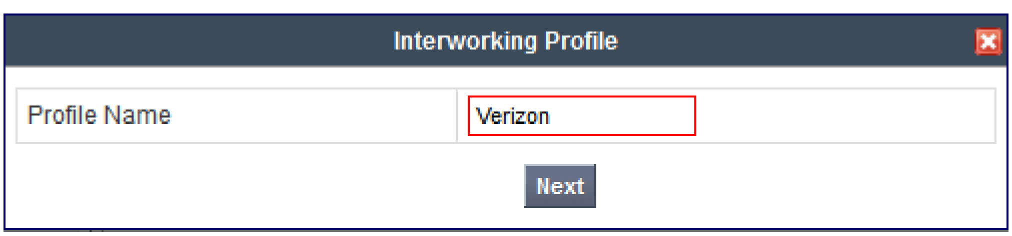
The following screen illustrates the **Advanced Settings** configuration. The “Topology Hiding: Change Call-ID” defaults to Yes, but was changed in the test configuration to allow for easier correlation of data. This setting in this field is at the discretion of the user. Both settings were tested. All other parameters shown are default values. Note that the default configuration will result in Record-Route headers in SIP messages.



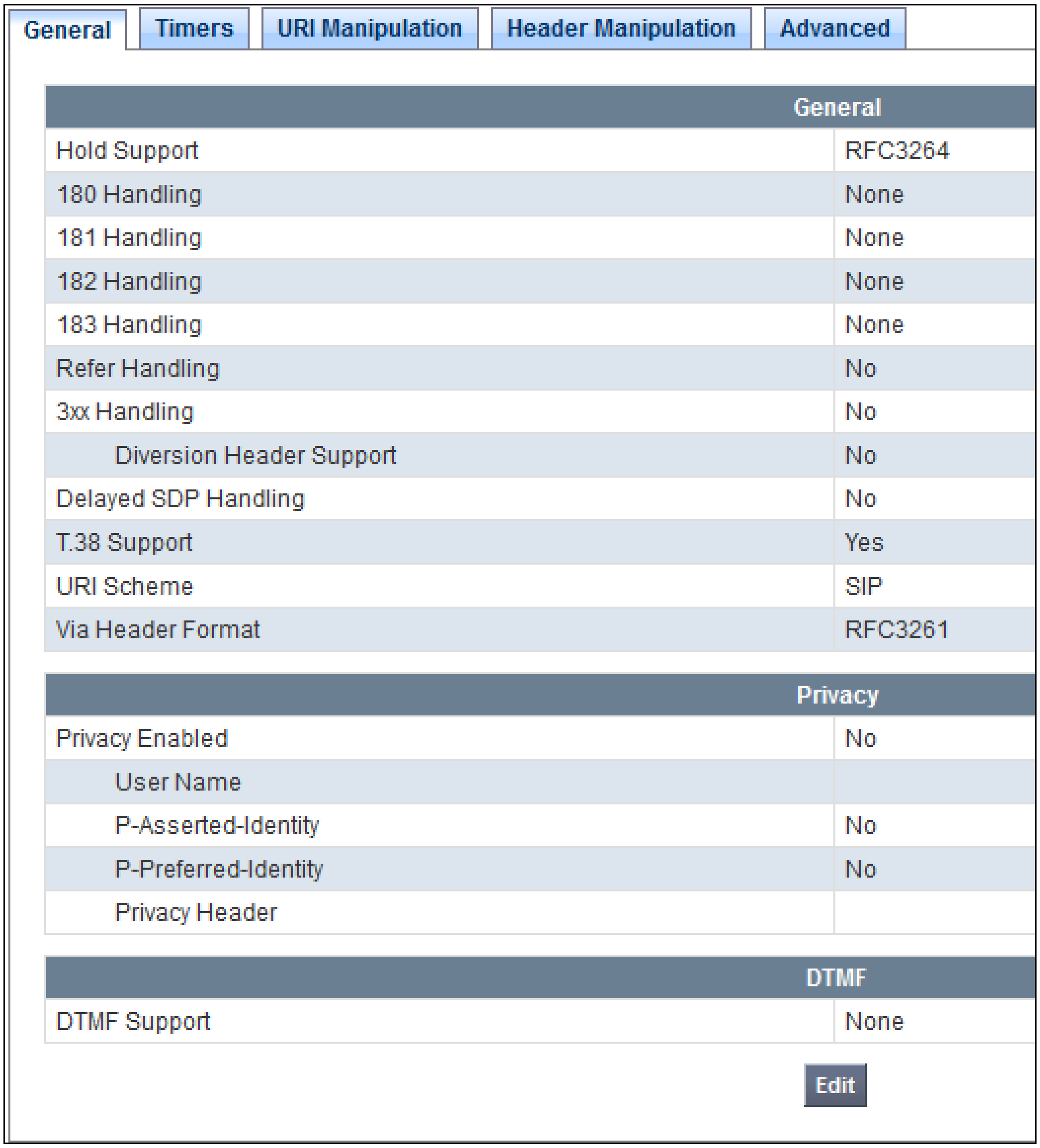
Advanced Settings	
Record Routes	BOTH
Topology Hiding: Change Call-ID	Yes
Call-Info NAT	No
Change Max Forwards	Yes
Include End Point IP for Context Lookup	No
OCS Extensions	No
AVAYA Extensions	No
NORTEL Extensions	No
SLiC Extensions	No
Diversion Manipulation	No
Metaswitch Extensions	No
Reset on Talk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

6.2.2 Server Interworking – Verizon IP Trunk

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



The following screens illustrate the “General” parameters used in the sample configuration for the Interworking Profile named “Verizon”. Most parameters retain default values. In the sample configuration, **T.38 support** was set to “Yes”, **Hold Support** was set for RFC3264, all other fields retained default values.



The following screen illustrates the **Advanced Settings** configuration. All parameters shown are default values. Note that the default configuration will result in Record-Route headers in SIP messages.

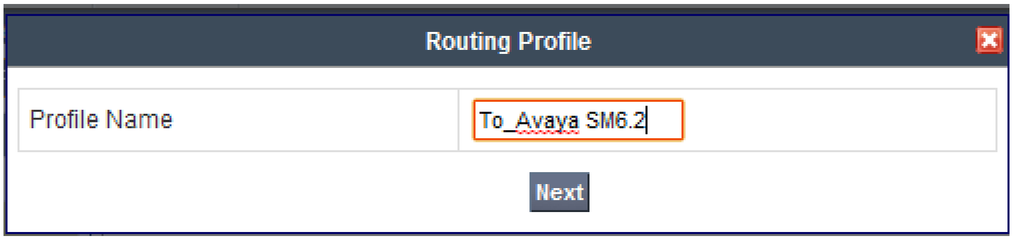
Advanced Settings	
Record Routes	BOTH
Topology Hiding: Change Call-ID	Yes
Call-Info NAT	Yes
Change Max Forwards	Yes
Include End Point IP for Context Lookup	No
OCS Extensions	No
AVAYA Extensions	No
NORTEL Extensions	No
SLiC Extensions	No
Diversion Manipulation	No
Metaswitch Extensions	No
Reset on Talk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

6.3. Global Profiles – Routing

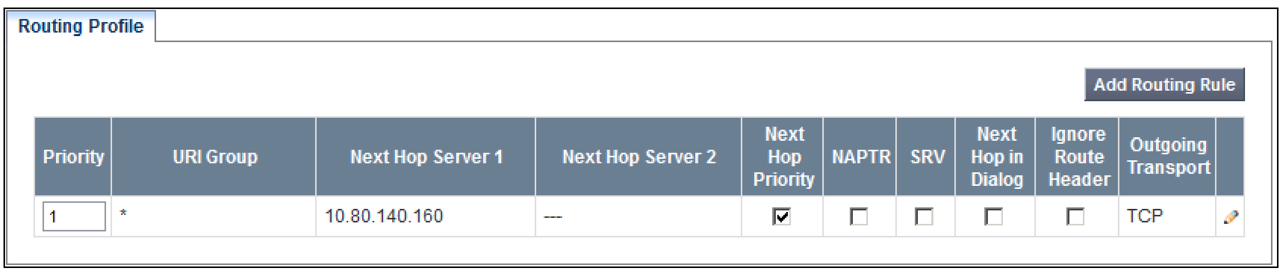
Select **Global Profiles → Server Configuration** from the left-side menu as shown below.

6.3.1 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_SM6.2" shown below. Click **Next**.

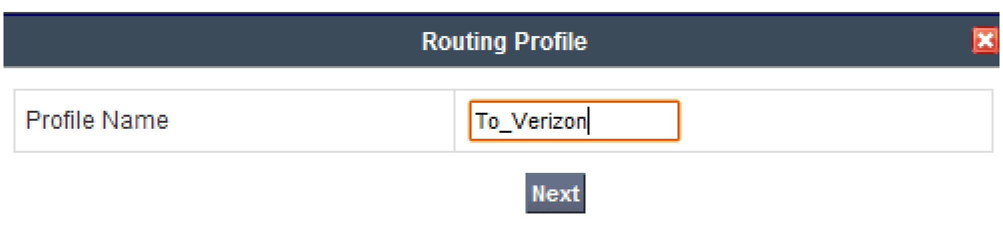


For the **Next Hop Routing**, enter the IP Address of the Session Manager SIP signaling interface as **Next Hop Server 1**, as shown below. Check **Next Hop Priority**. Choose **TCP** for **Outgoing Transport**.

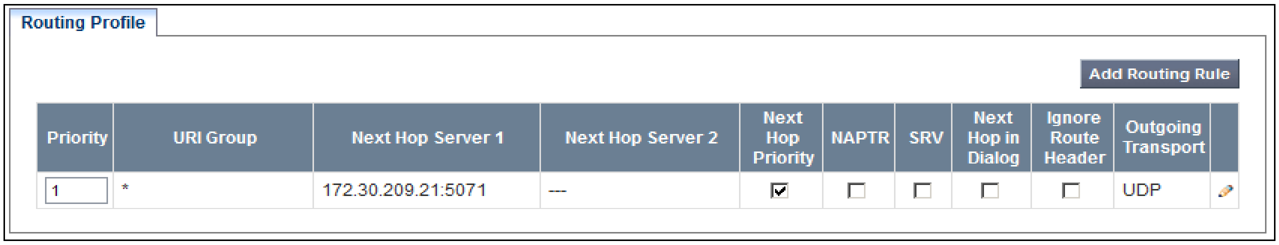


6.3.2 Routing Configuration for Verizon IP Trunk

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_Verizon" shown below. Click **Next**.



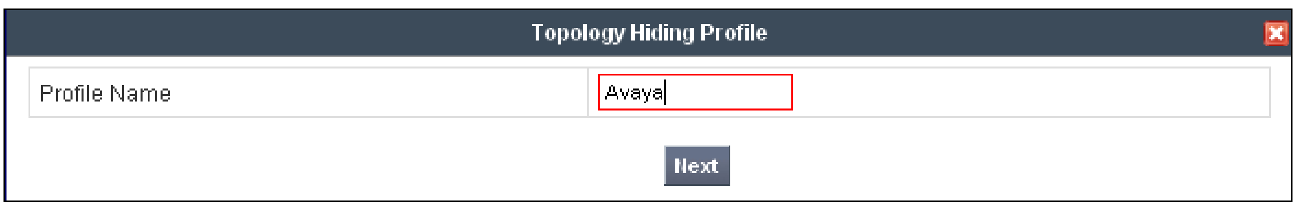
For the **Next Hop Routing**, enter the IP Address of the Verizon SIP signaling interface as **Next Hop Server 1**, as shown below. Check **Next Hop Priority**. Choose **UDP** for **Outgoing Transport**.



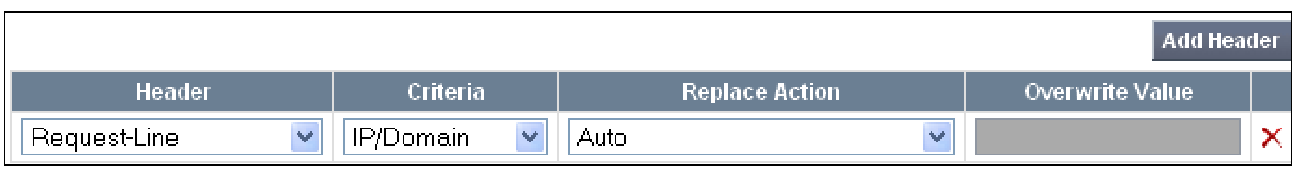
6.3.3 Topology Hiding for Session Manager

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

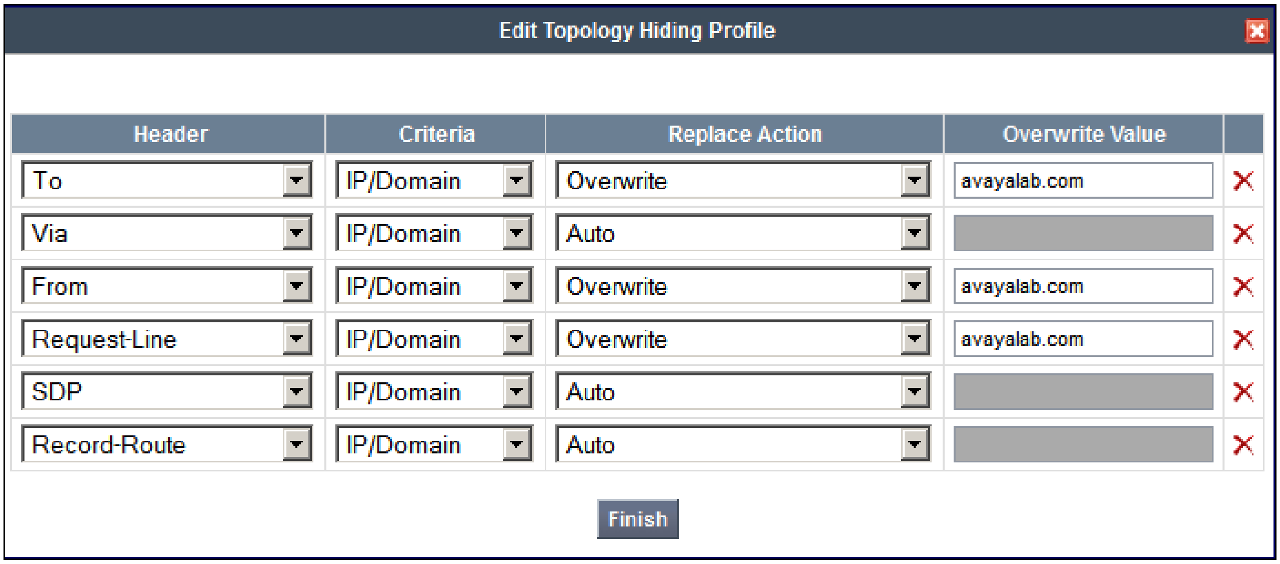
Click the **Add Profile** button (not shown) to add a new profile, or select an existing topology hiding profile to edit. If adding a profile, a screen such as the following is displayed. Enter a **Profile Name** such as "Avaya" shown below. Click **Next**.



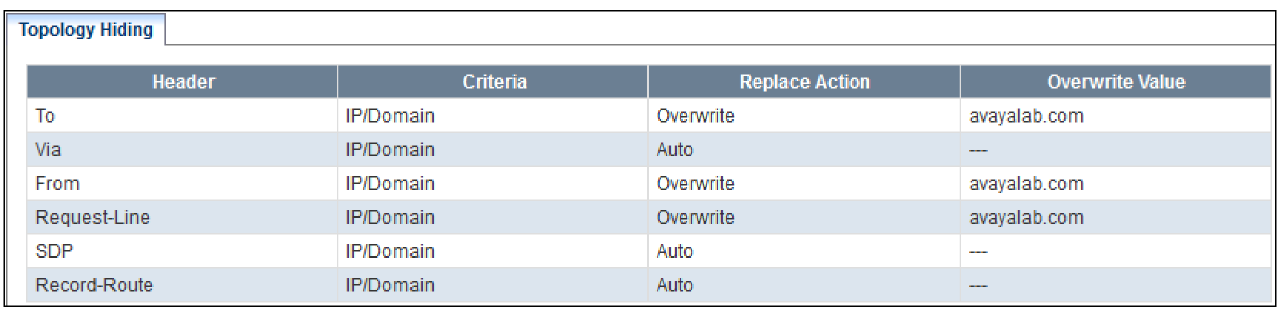
In the resultant screen, click the **Add Header** button in the upper right multiple times to reveal additional headers.



If it is desired to ensure that the domain received by Session Manager from the SBCE is the expected enterprise domain, select “Overwrite” as the **Replace Action** for the To, From, 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 is changed by the SBCE to “avayalab.com”. Click **Finish**.



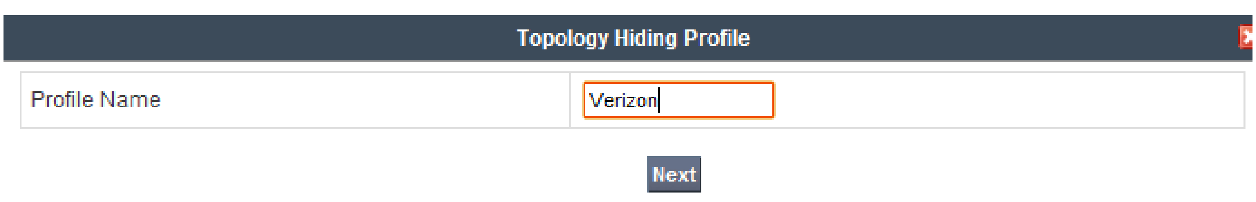
After configuration is completed, the Topology Hiding for profile “Avaya” will appear as follows.


Topology Hiding

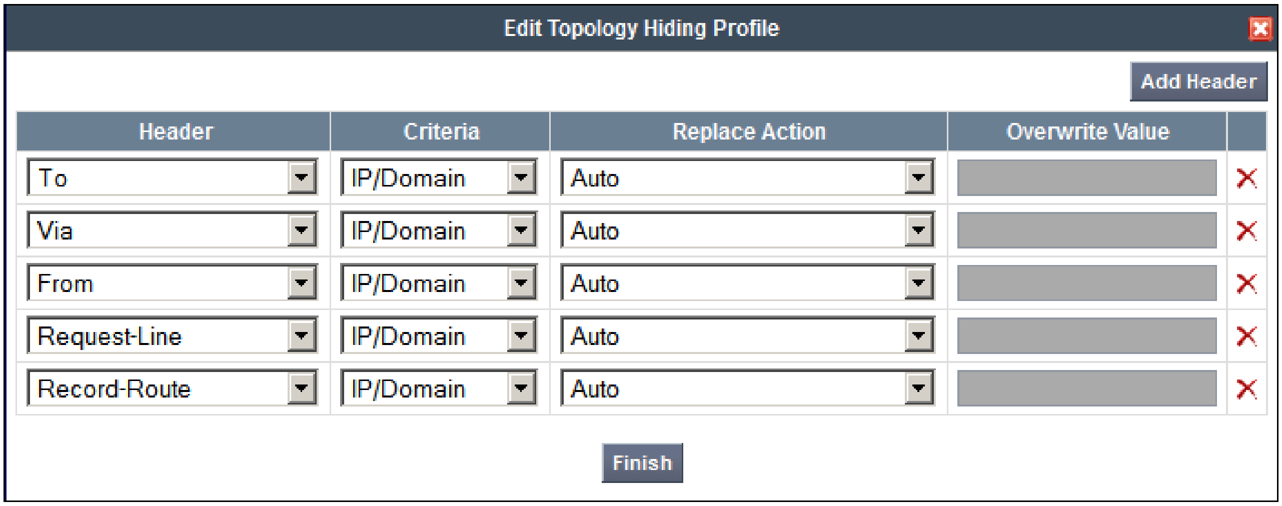
Header	Criteria	Replace Action	Overwrite Value
To	IP/Domain	Overwrite	avayalab.com
Via	IP/Domain	Auto	---
From	IP/Domain	Overwrite	avayalab.com
Request-Line	IP/Domain	Overwrite	avayalab.com
SDP	IP/Domain	Auto	---
Record-Route	IP/Domain	Auto	---

6.3.4 Topology Hiding for Verizon IP Trunk

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 “Verizon” shown below. Click **Next**.



Again, in the resultant screen, click the **Add Header** button in the upper right multiple times to reveal additional headers. The default “Auto” behaviors are sufficient. Click **Finish.**

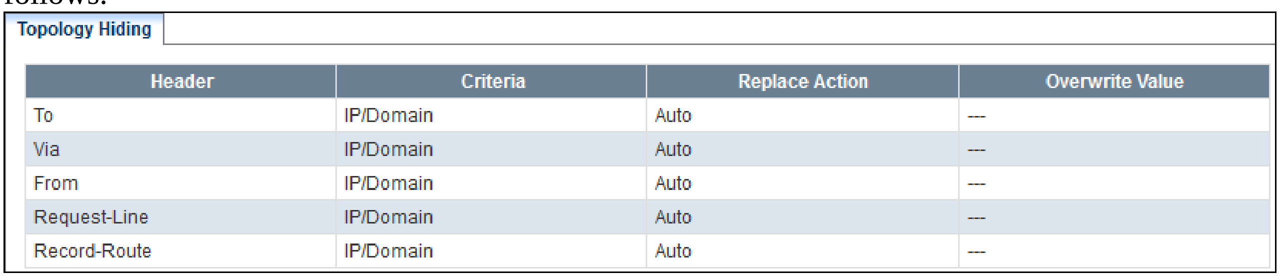

Edit Topology Hiding Profile

Add Header

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

Finish

After configuration is completed, the **Topology Hiding** for profile “Verizon” will appear as follows.

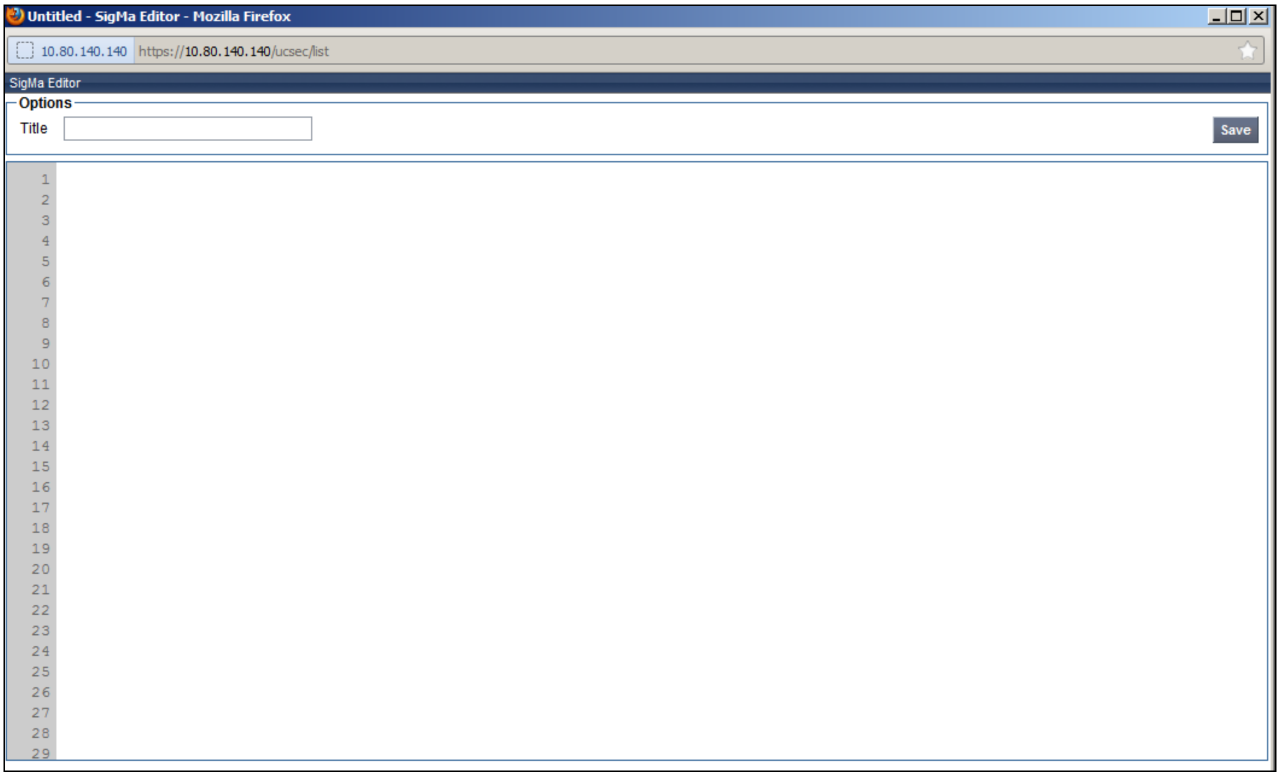

Topology Hiding

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

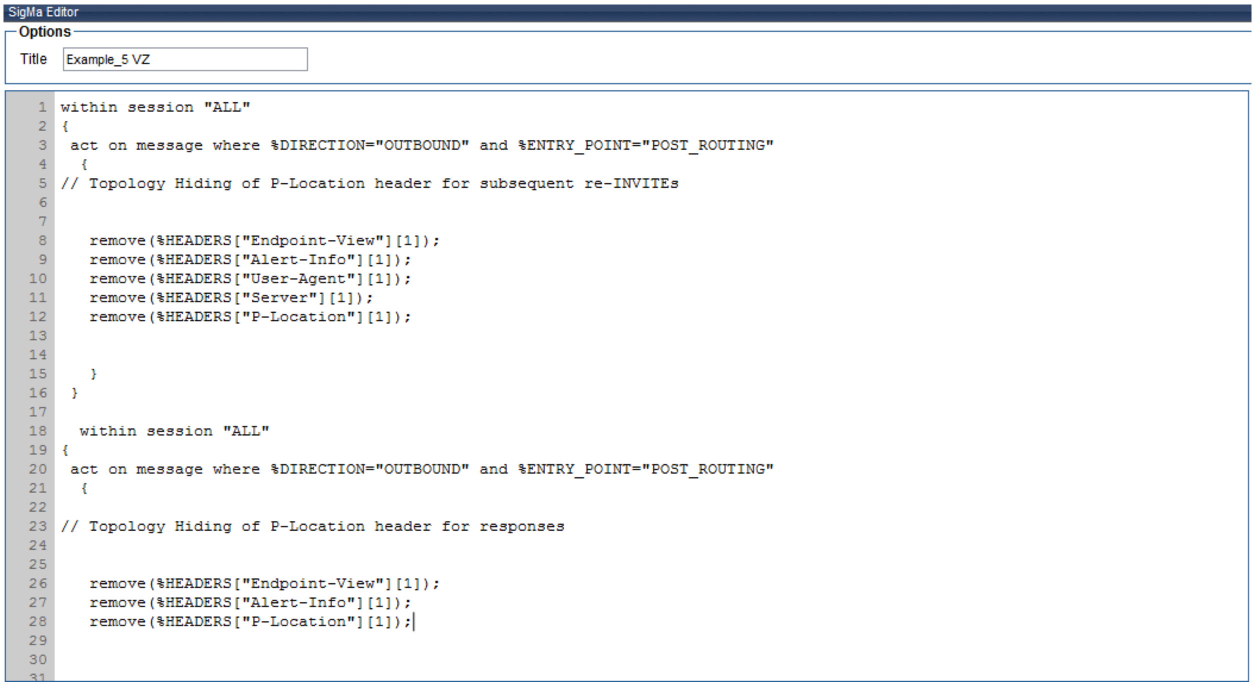
6.3.5 Signaling Manipulation

This feature adds the ability to add, change and delete any of the headers and other information in a SIP message. The feature will add the ability to configure such manipulation at each flow level in a highly flexible manner using a proprietary scripting language.

Click the **Add Script** button (not shown) to add a new script, or select an existing script to edit. If adding a script, a screen such as the following is displayed. Enter a title in the upper left and then enter the text to manipulate headers and click **Save**.

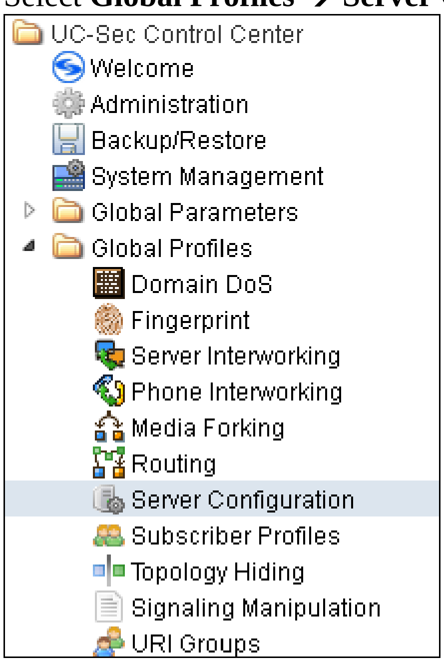


In 6.2, there are two proprietary headers (P-Location and Endpoint View) and one standard header (Alert-Info) that contain internal information and that are not applicable to a service provider that need to be stripped. These headers were stripped with a Sigma script and applied in the server configuration section. The script “Example_5 VZ” is shown here. This script will be applied in the next section, ‘Server Configuration’.



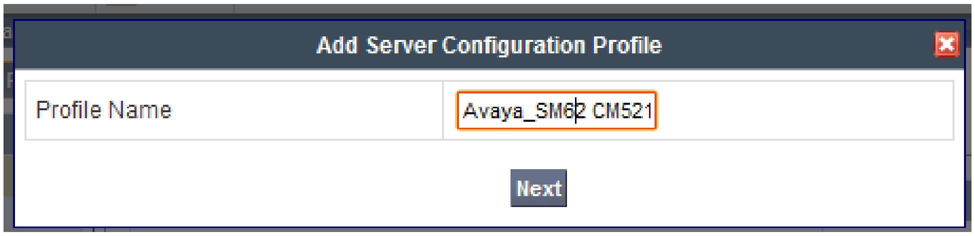
6.4. Global Profiles – Server Configuration

Select **Global Profiles → Server Configuration** from the left-side menu as shown below.

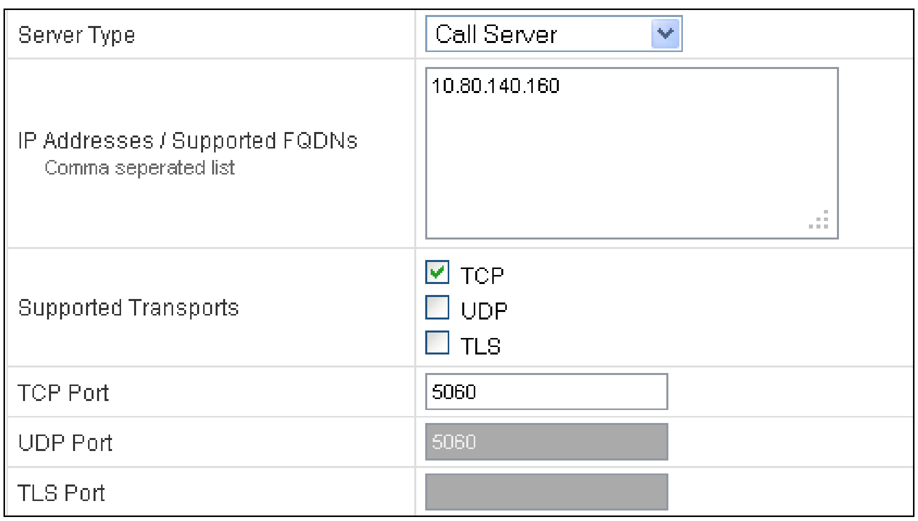


6.4.1 Server Configuration for Session Manager

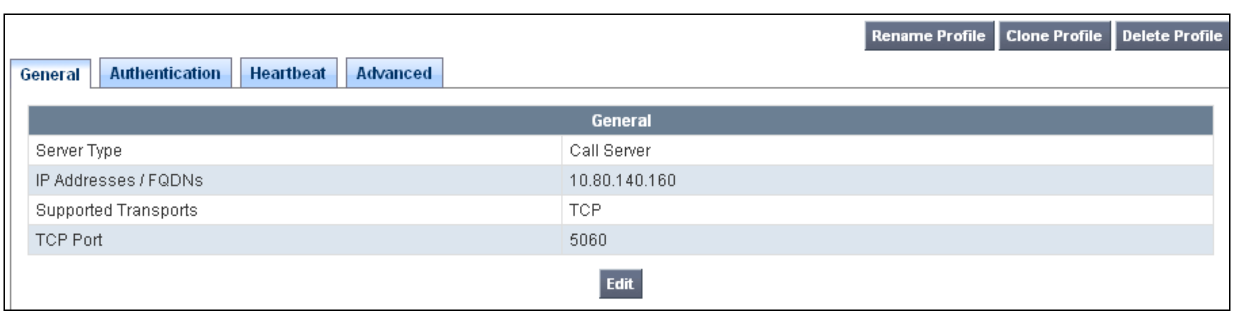
Click the **Add Profile** button (not shown) to add a new profile, or select an existing profile to edit. If adding a profile, a screen such as the following is displayed. Enter an appropriate Profile Name such as "Avaya_SM62 CM521" shown below. Click **Next**.



The following screens illustrate the Server Configuration with Profile name "Avaya_SM". In the "General" parameters, select "Call Server" from the **Server Type** drop-down menu. In the **IP Addresses / Supported FQDNs** area, the IP Address of the Session Manager SIP signaling interface in the sample configuration is entered. This IP Address is 10.80.140.160. In the **Supported Transports** area, TCP is selected, and the **TCP Port** is set to 5060. This configuration corresponds with the Session Manager entity link configuration for the entity link to the SBC. If adding a new profile, click **Next**. If editing an existing profile, click **Finish**.

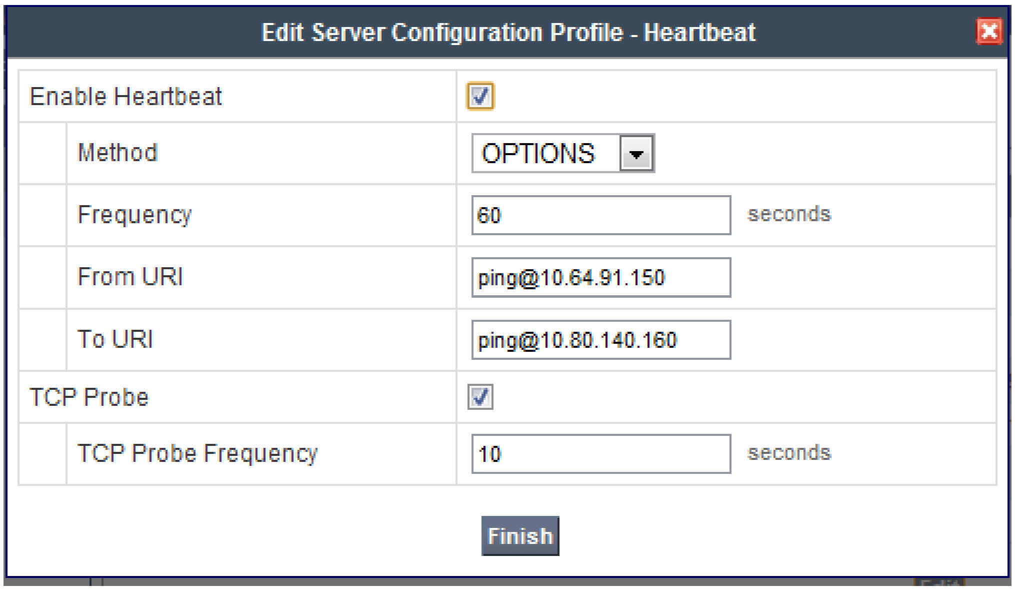


Once configuration is completed, the **General** tab for "Avaya_SM62 CM521" will appear as shown below.

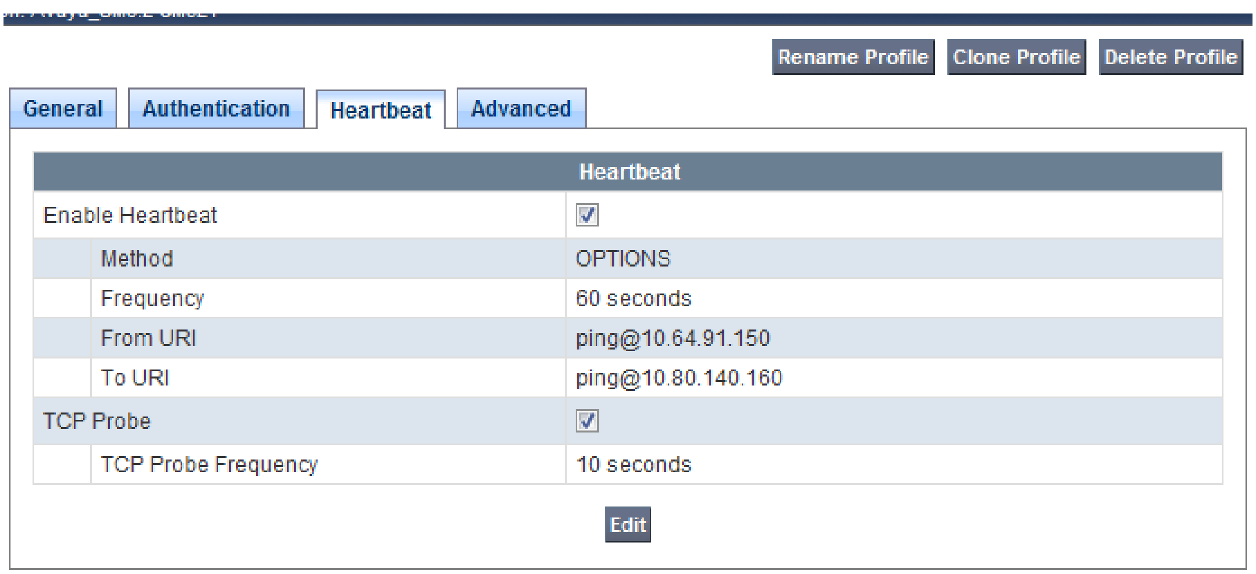


If adding the profile, click **Next** to accept default parameters for the Authentication tab, and advance to the Heartbeat area. If editing an existing profile, select the **Heartbeat** tab and click edit.

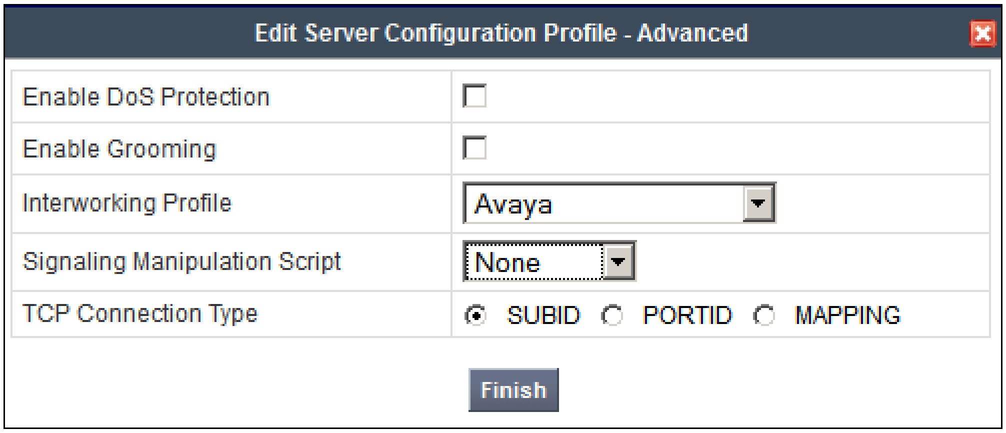
The SBC can be configured to source “heartbeats” in the form of SIP OPTIONS. If SBC-sourced OPTIONS messages are desired, check the **Enable Heartbeat** box. Select “OPTIONS” from the **Method** drop-down menu. Select the desired frequency that the SBC will source OPTIONS to this server. The **From URI** 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**. If editing an existing profile, click **Finish**.



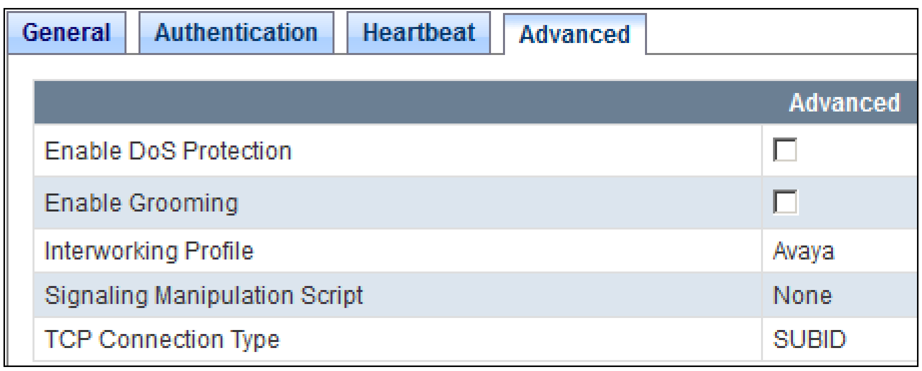
If SBC sourced OPTIONS are configured, the **Heartbeat** tab for “Avaya_SM62 CM521” will appear as shown below.



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

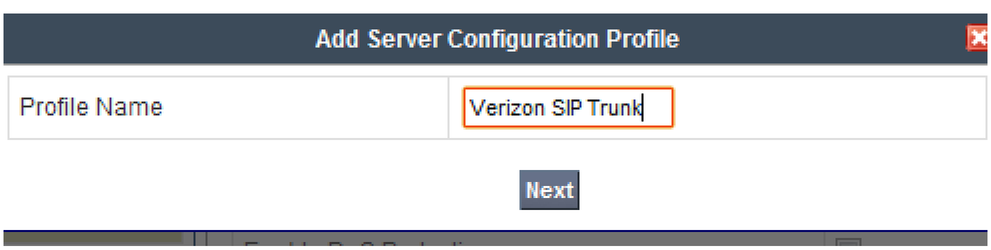


Once configuration is completed, the **Advanced** tab for "Avaya_SM62 CM521" will appear as shown below.

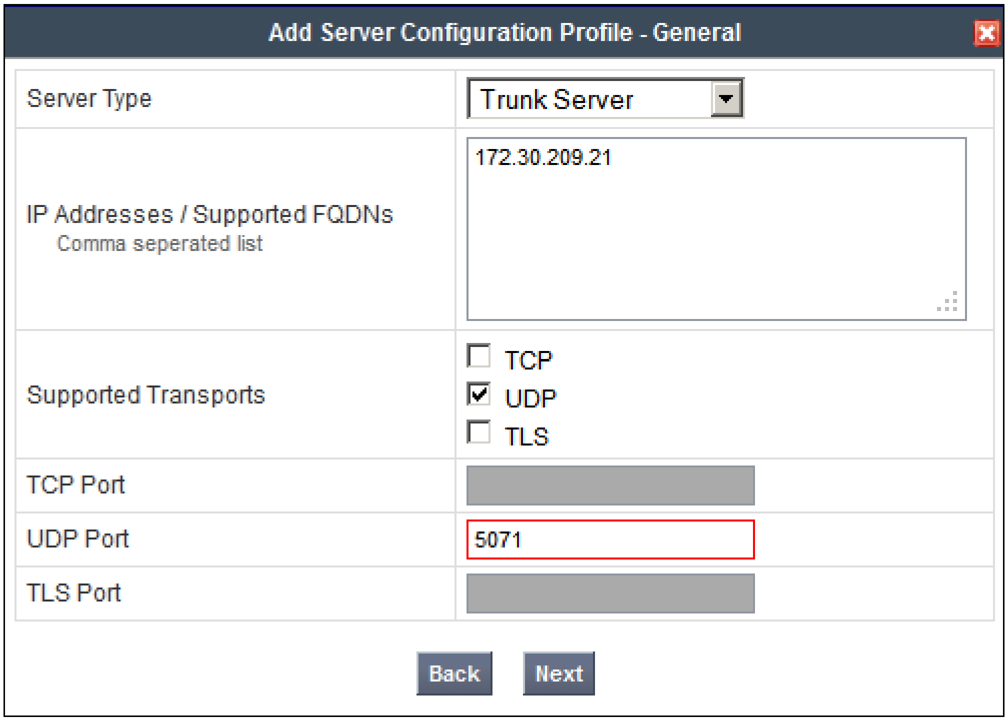


6.4.2 Server Configuration for Verizon IP Trunk

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 "Verizon SIP Trunk" shown below. Click **Next**.



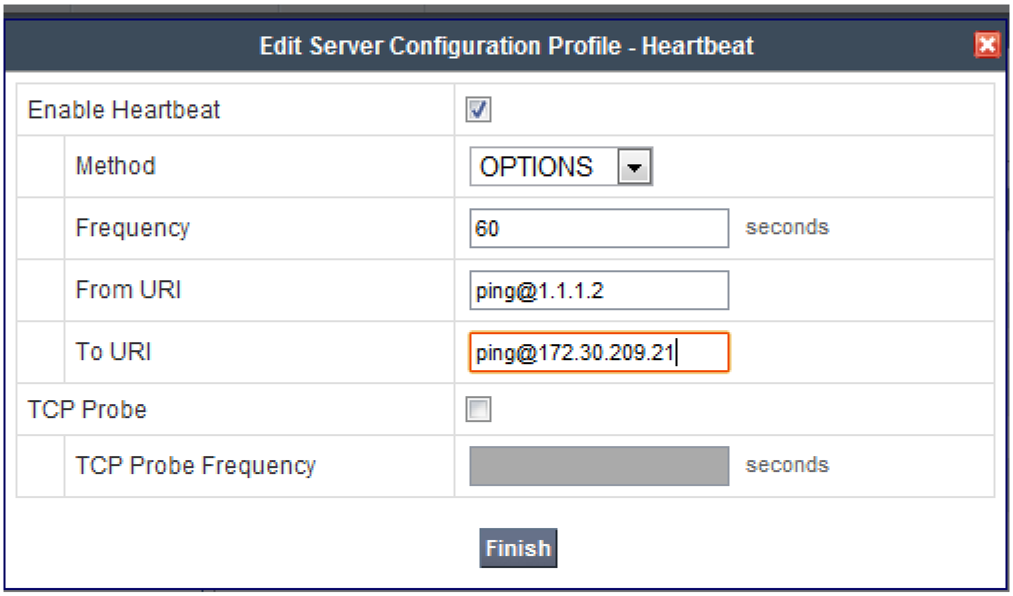
The following screens illustrate the Server Configuration with Profile name "Verizon SIP Trunk". In the "General" parameters, select "Trunk Server" from the **Server Type** drop-down menu. In the **IP Addresses / Supported FQDNs** area, the Verizon-provided Verizon IP Trunk IP Address is entered. This IP Address is 172.30.209.21. In the **Supported Transports** area, UDP is selected, and the **UDP Port** is set to 5071.



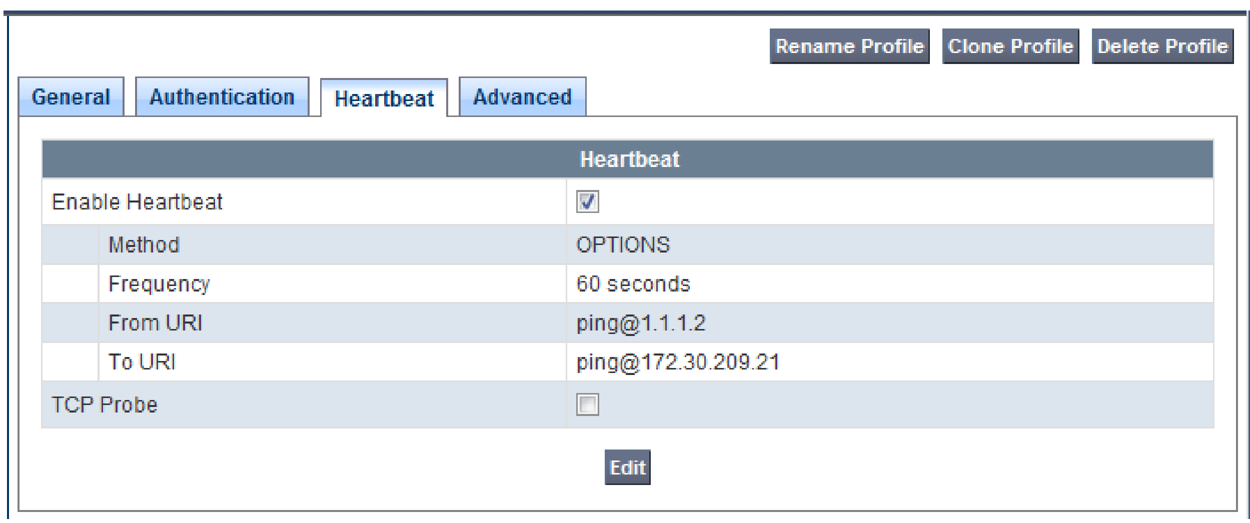
If adding the profile, click **Next** to accept default parameters for the Authentication tab, and advance to the Heartbeat area. If editing an existing profile, select the **Heartbeat** tab and click edit.

The SBC can be configured to source "heartbeats" in the form of SIP OPTIONS towards Verizon. This configuration is optional. Independent of whether the SBC is configured to source SIP OPTIONS towards Verizon, Verizon 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 send SIP OPTIONS to Verizon. When Verizon responds, the SBC will pass the response to Session Manager.

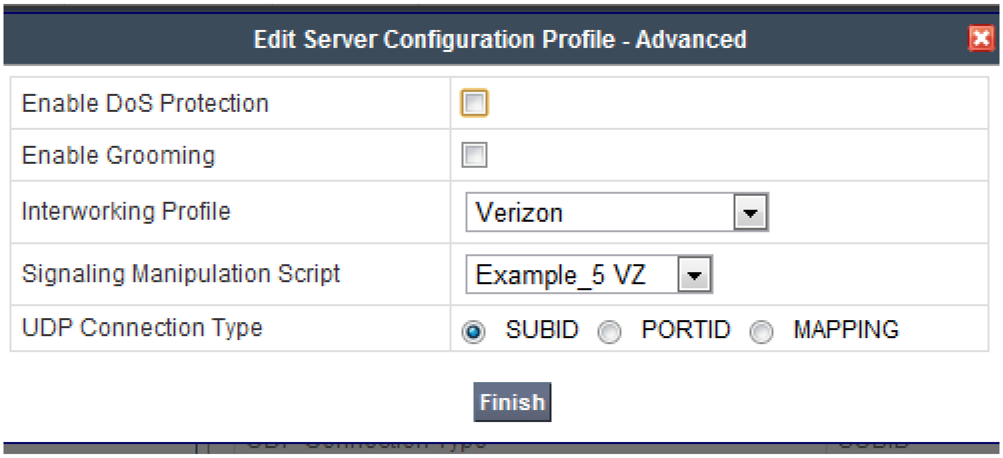
If SBC-sourced OPTIONS are desired, select "OPTIONS" from the **Method** drop-down menu. Select the desired frequency that the SBC will source OPTIONS. The **From URI** and **To URI** may be filled in to configure easily identifiable URIs to appear in SIP OPTIONS sourced by the SBC. If adding a new profile, click **Next**. If editing an existing profile, click **Finish**.



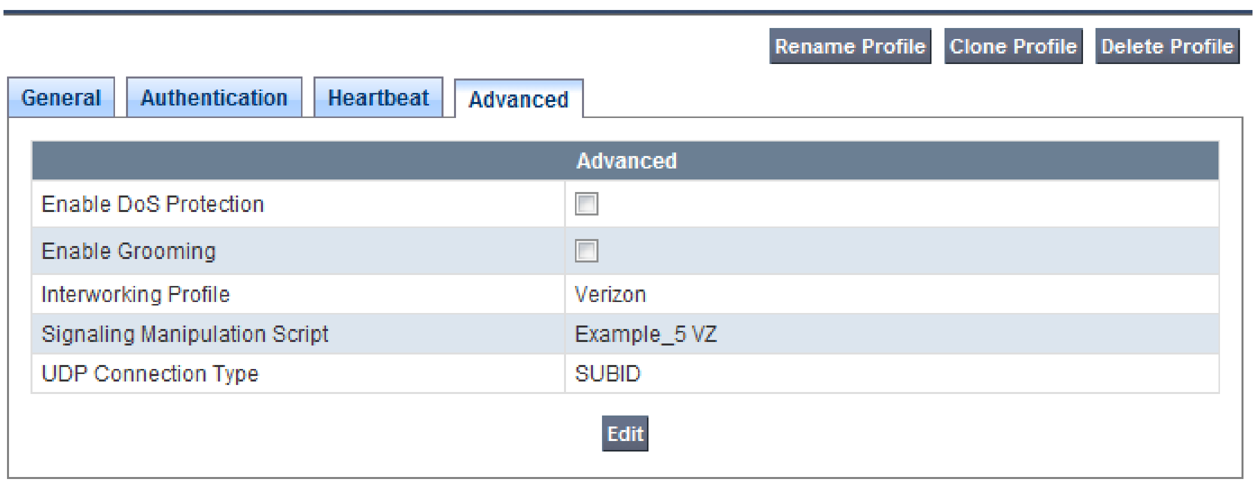
If the optional SBC sourced OPTIONS configuration is completed, the **Heartbeat** tab for "Verizon SIP Trunk" will appear as shown below.



If adding a profile, click **Next** to continuing to the “Advanced” settings. If editing an existing profile, select the **Advanced** tab and **Edit**. In the resultant screen, select the **Interworking Profile** “Verizon” created previously, and Signaling Manipulation Script will be the script shown in the previous section titled “Example_5VZ”. Other SBC features, such as DoS Protection and Grooming, can be configured according to customer preference. Click **Finish**.

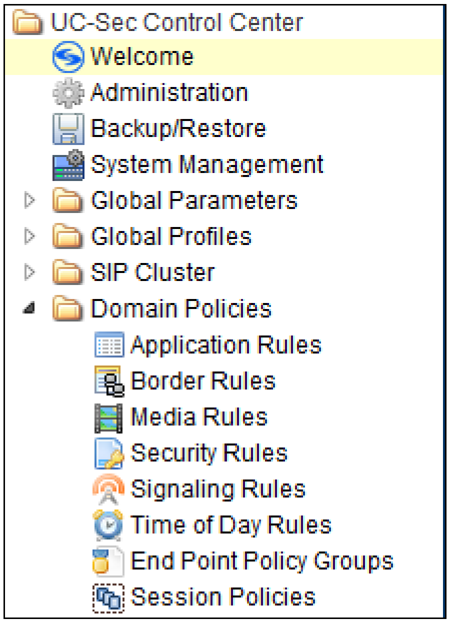


Once configuration is completed, the **Advanced** tab for “Verizon SIP Trunk” will appear as shown below.

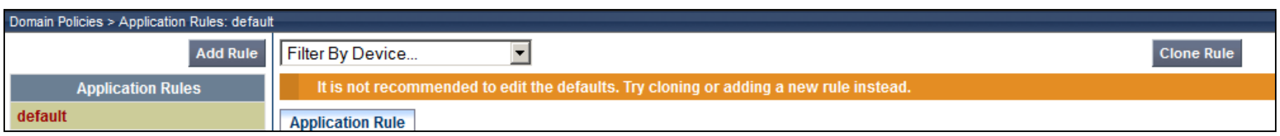


6.5. Domain Policies – Application Rule

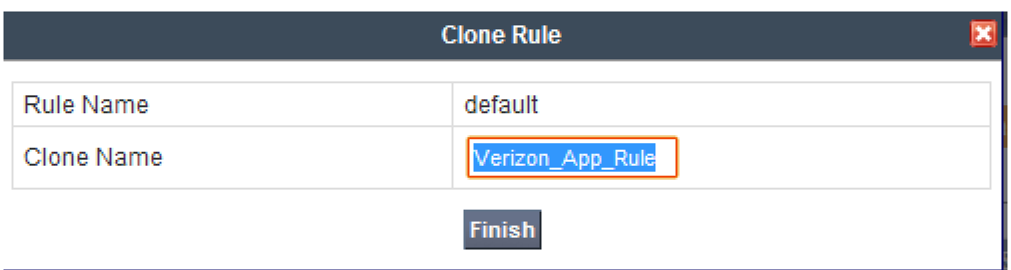
Select **Domain Policies → Application Rules** from the left-side menu as shown below.



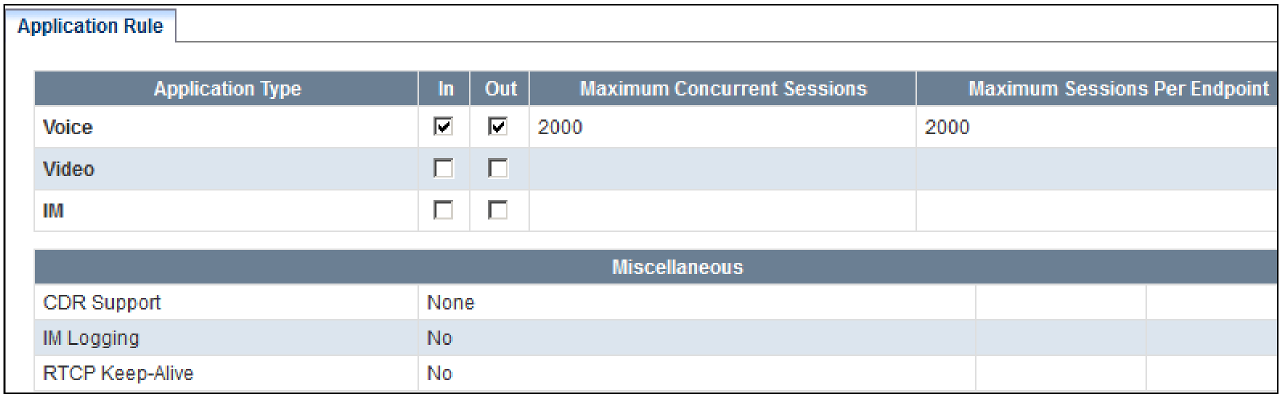
In the sample configuration, a single application rule was created by cloning the default rule called "default". Select the default rule and click the **Clone Rule** button.



Enter a name in the **Clone Name** field, such as "Verizon_App_Rule" as shown below. Click **Finish**.



Select the newly created rule and click the **Edit** button (not shown). In the resulting screen, change the default **Maximum Concurrent Sessions** to 2000, the **Maximum Session per Endpoint** to 2000. Click **Finish**.

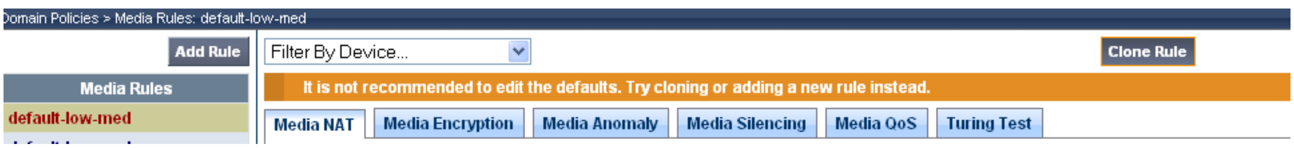

Application Rule

Application Type	In	Out	Maximum Concurrent Sessions	Maximum Sessions Per Endpoint
Voice	☑	☑	2000	2000
Video	☐	☐		
IM	☐	☐		

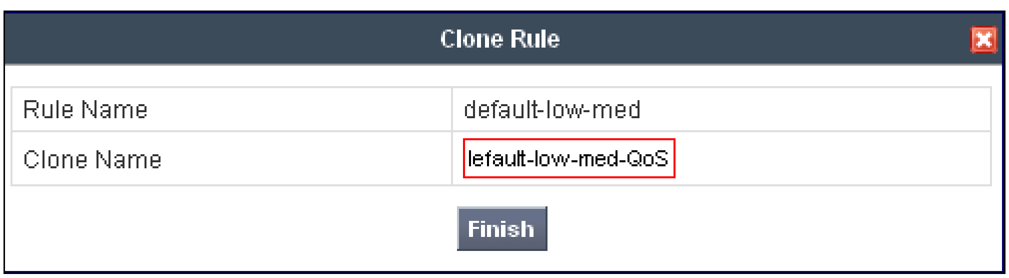
Miscellaneous	
CDR Support	None
IM Logging	No
RTCP Keep-Alive	No

6.6. Domain Policy – Media Rules

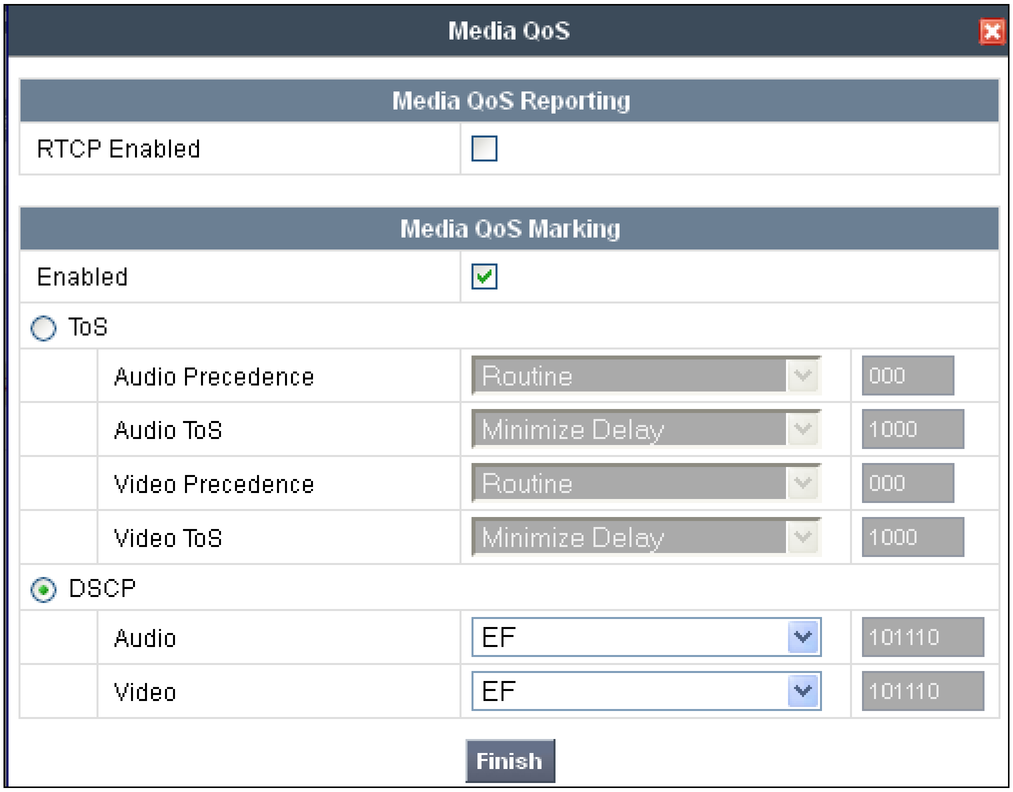
In the sample configuration, a single media rule was created by cloning the default rule called "default-low-med". Select the default-low-med rule and click the **Clone Rule** button.



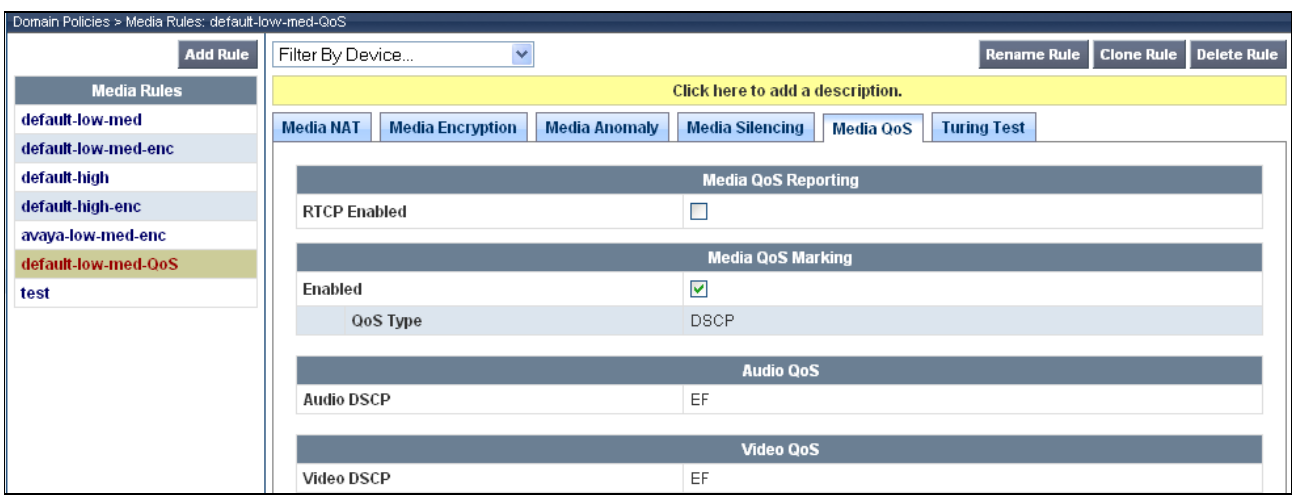
Enter a name in the **Clone Name** field, such as "default-low-med-QoS" as shown below. Click **Finish**.



Select the newly created rule, select the **Media QoS** tab, and click the **Edit** button (not shown). In the resulting screen, check the **Media QoS Marking Enabled** checkbox. Select **DSCP** and select "EF" for expedited forwarding as shown below. Click **Finish**.

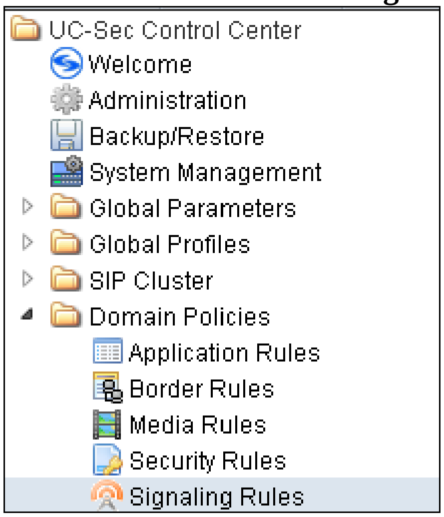


When configuration is complete, the "default-low-med-QoS" media rule **Media QoS** tab appears as follows.

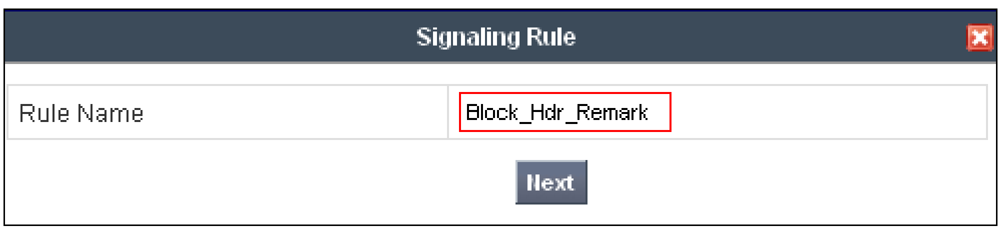


6.7. Domain Policies – Signaling Rules

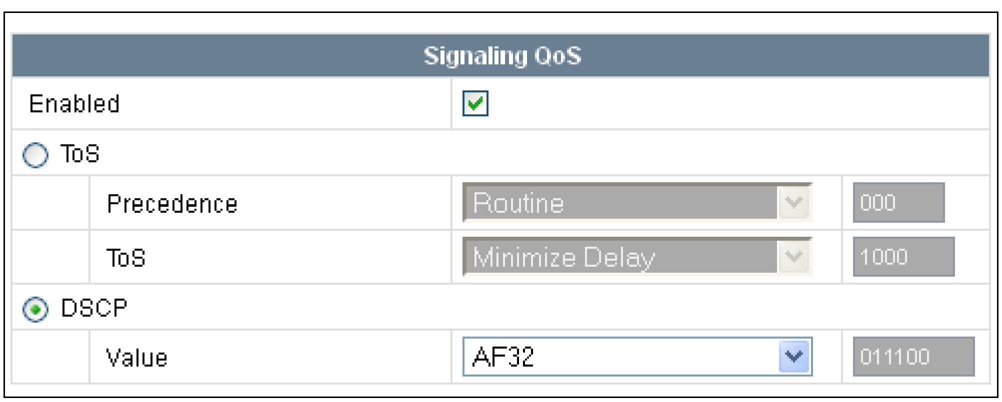
Select **Domain Policies → Signaling Rules** from the left-side menu as shown below.



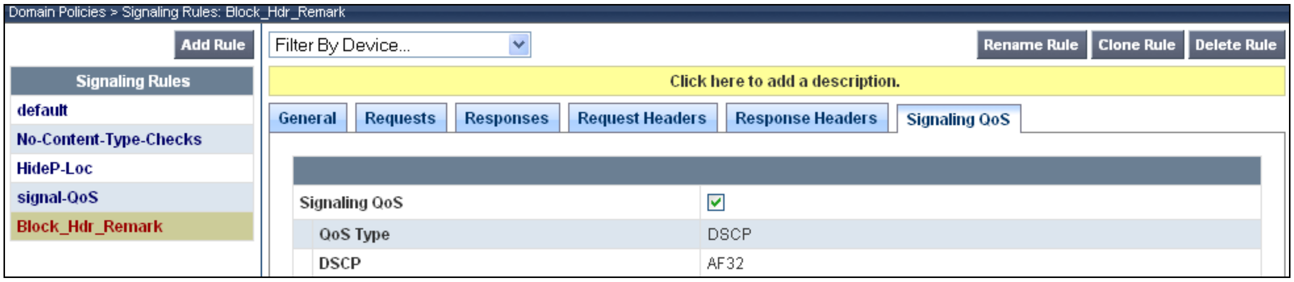
Click the Add Rule button to add a new signaling rule. In the Rule Name field, enter an appropriate name, such as "Block_Hdr_Remark".



In the subsequent screen (not shown), click **Next** to accept defaults. In the Signaling QoS screen, select **DSCP** and select the desired **Value** for Signaling QoS from the drop-down menu. In the sample configuration, "AF32" was selected for "Assured Forwarding 32." Click **Finish** (not shown).

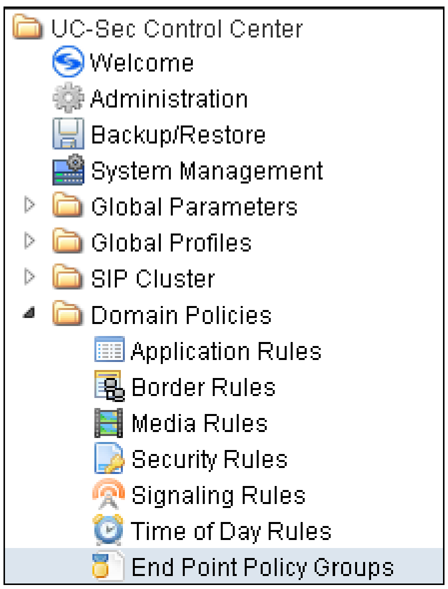


After this configuration, the new “Block_Hdr_Remark” will appear as follows.

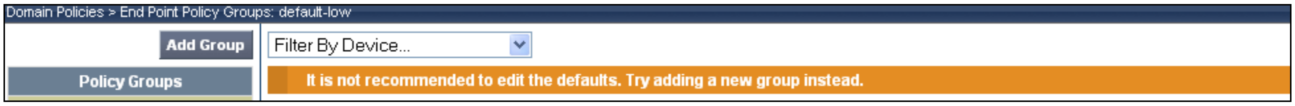


6.8. Domain Policies – End Point Policy Groups

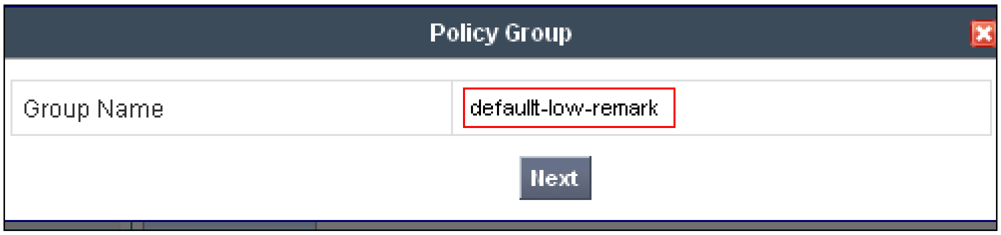
Select **Domain Policies → End Point Policy Groups** from the left-side menu as shown below.



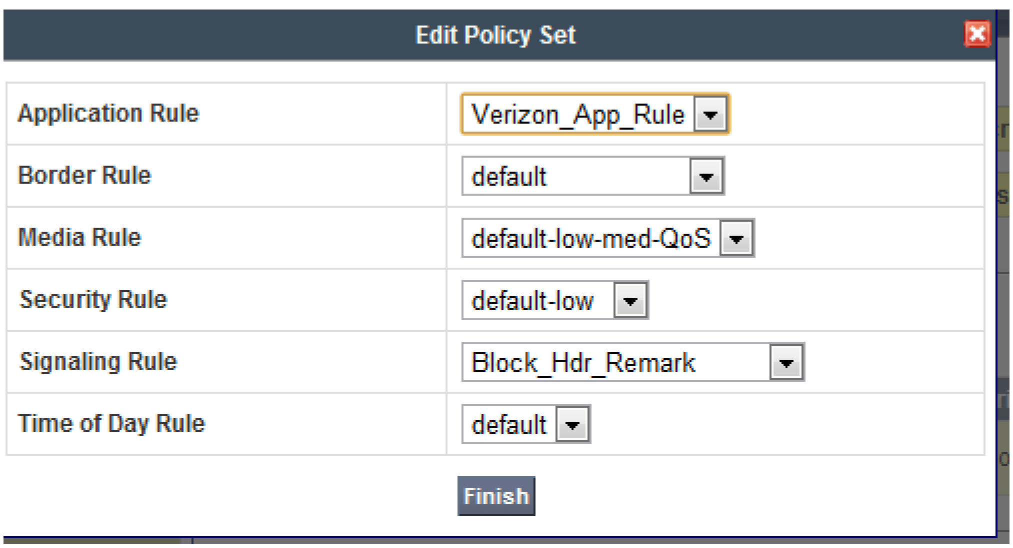
Select the **Add Group** button.



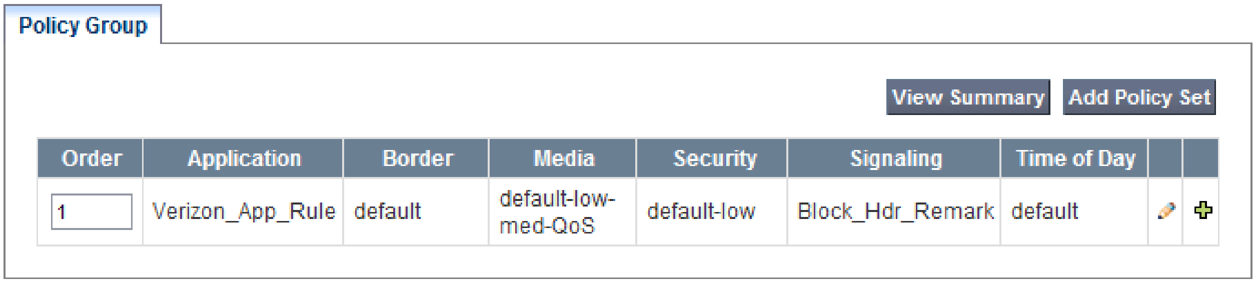
Enter a name in the **Group Name** field, such as "default-low-remark" as shown below. Click **Next**.



In the sample configuration, defaults were selected for all fields, with the exception of the **Application Rule** which was set to "Verizon_App_Rule", **Media Rule** which was set to "default-low-med-QoS", and the **Signaling Rule**, which was set to "Block_Hdr_Remark" as shown below. The selected non-default media rule and signaling rule chosen were created in previous sections. Click **Finish**.

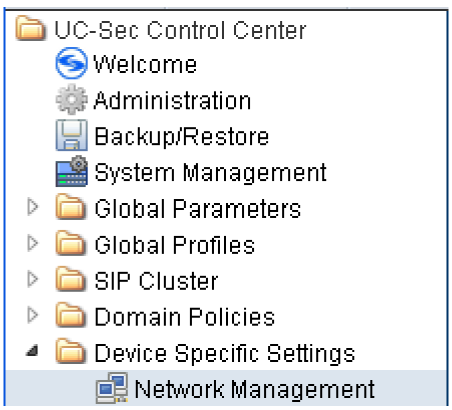


Once configuration is completed, the “default-low-remark” policy group will appear as follows.

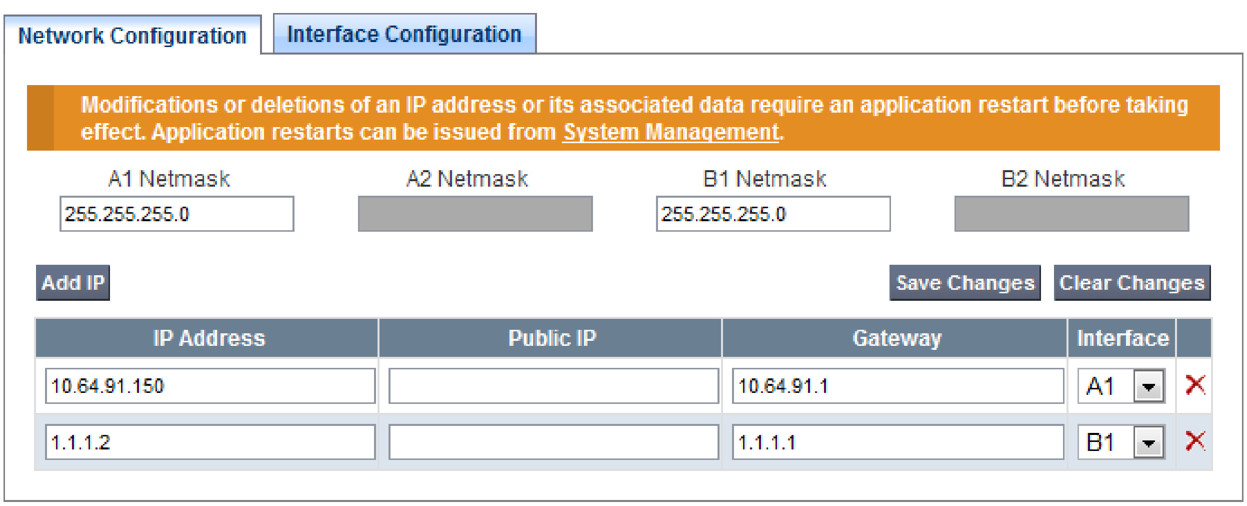


6.9. Device Specific Settings - Network Management

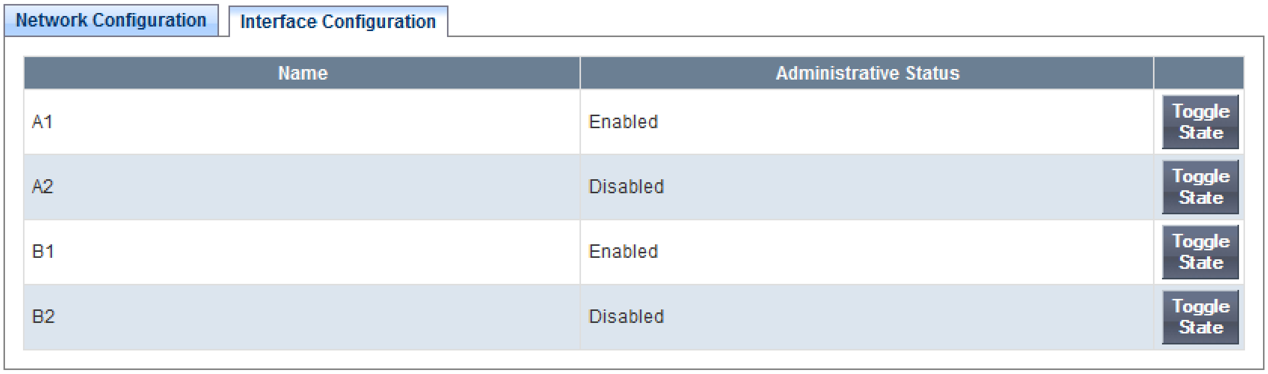
Select **Device Specific Setting → Network Management** from the left-side menu as shown below.



Under **UC-Sec Devices**, select the device being managed, which was named “ASBCE-3” in the sample configuration (not shown). The **Network Configuration** tab is shown below. Observe the **IP Address**, **Netmask**, **Gateway**, and **Interface** information previously assigned.

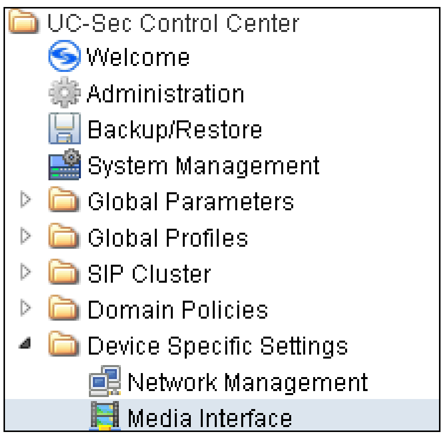


Select the **Interface Configuration** tab. The Administrative Status can be toggled between "Enabled" and "Disabled" in this screen. The following screen was captured after the interfaces had already been enabled. To enable the interface if it is disabled, click the **Toggle State** button.

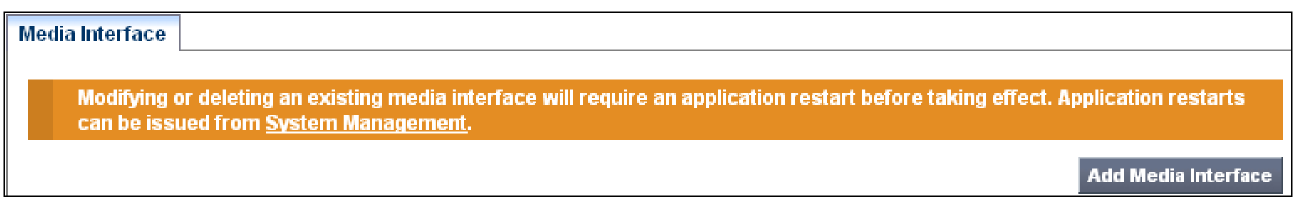


6.10. Device Specific Settings – Media Interface

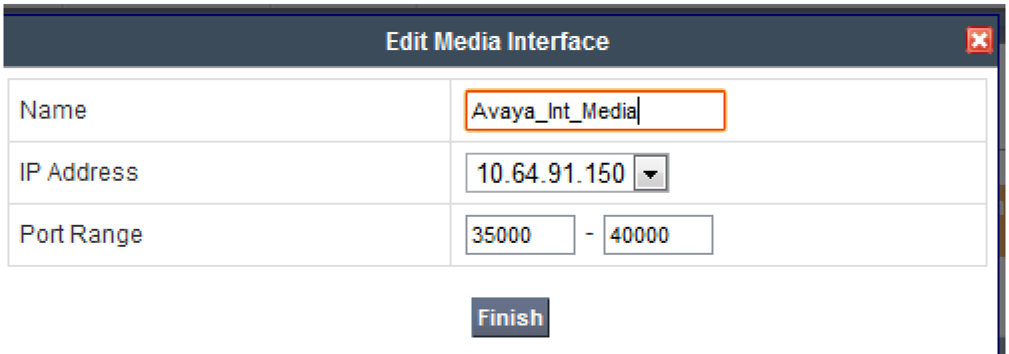
Select **Device Specific Setting → Media Interface** from the left-side menu as shown below.



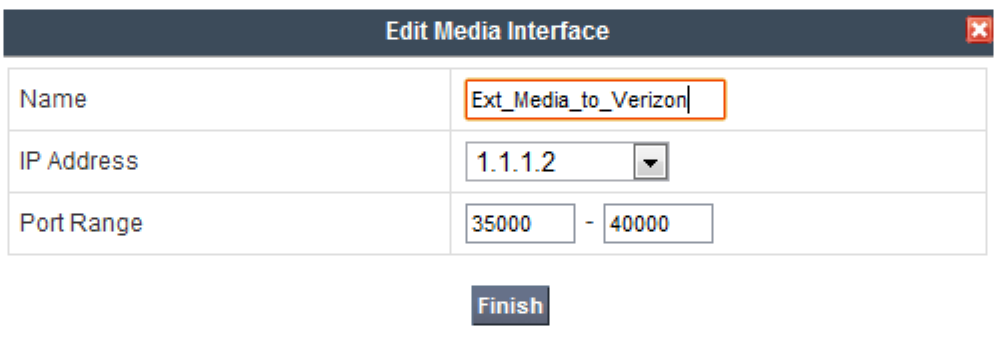
Under **UC-Sec Devices**, select the device being managed, which was named "ASBCE-3" in the sample configuration (not shown). Select **Add Media Interface**.



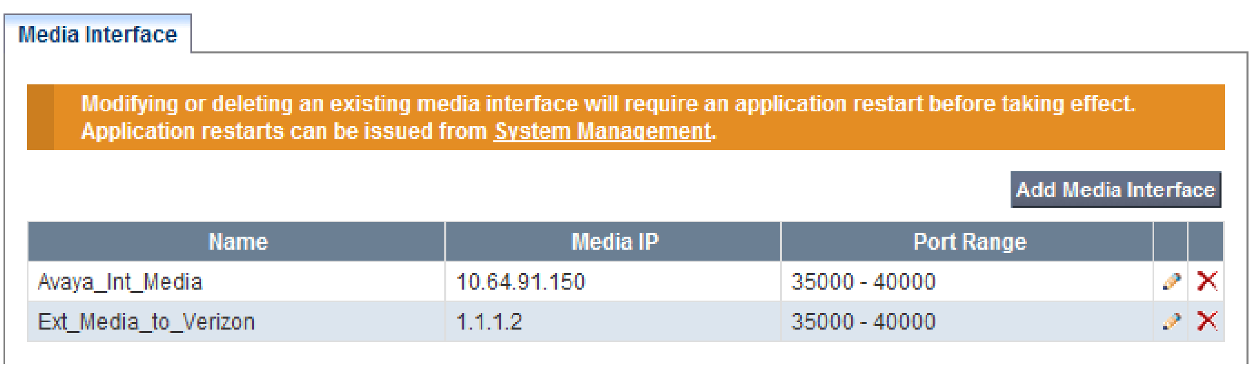
Enter an appropriate **Name** for the media interface for the Avaya CPE and select the inside private IP Address from the **IP Address** drop-down menu. In the sample configuration, "Avaya_Int_Media" is chosen as the name, and the "inside" IP Address of the SBCE is "10.64.91.150". For the **Port Range**, default values are shown. Click **Finish**.



Once again, select **Add Media Interface**. Enter an appropriate **Name** for the media interface for the public "outside" of the SBC, and select the outside public IP Address from the **IP Address** drop-down menu. In the sample configuration, "Ext_Media_to_Verizon" is chosen as the name, and the "outside" public IP Address of the SBC is "1.1.1.2". For the **Port Range**, default values are shown. Verizon IP Trunk does not require that the RTP ports be chosen within a specific range. Click **Finish**.

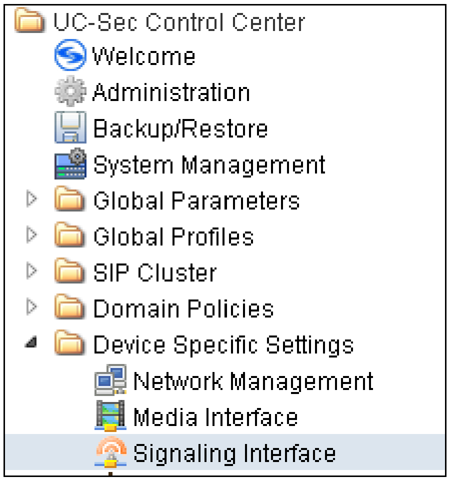


The resultant Media Interface configuration used in the sample configuration is shown below.



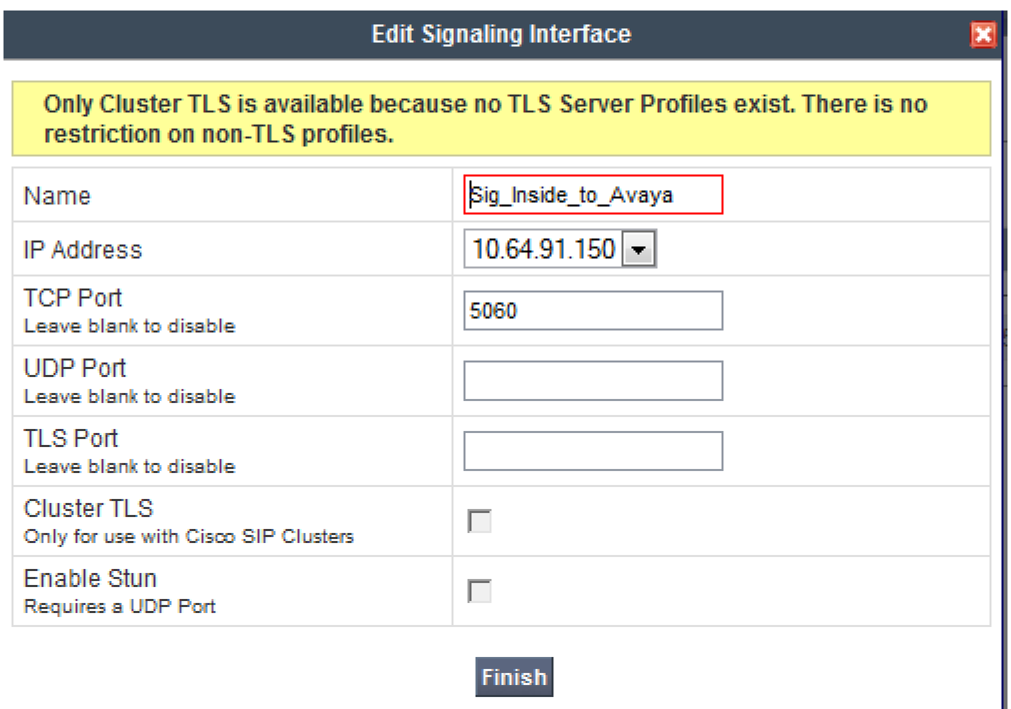
6.11. Device Specific Settings – Signaling Interface

Select **Device Specific Setting → Signaling Interface** from the left-side menu as shown below.

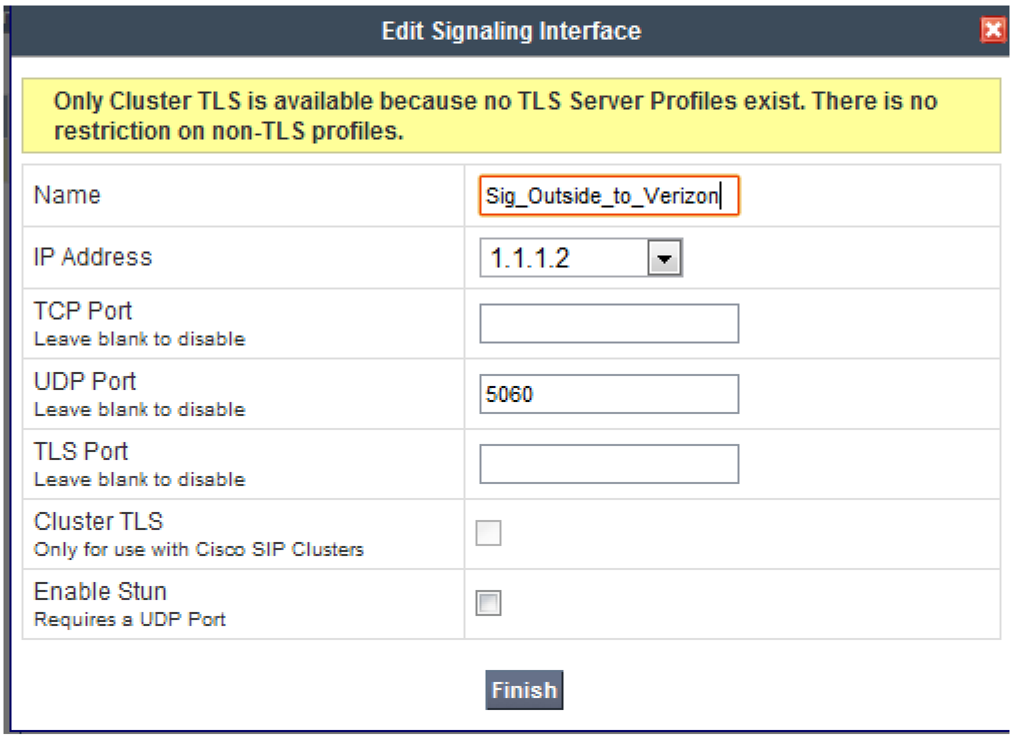


Under **UC-Sec Devices**, select the device being managed, which was named "ASBCE-3" in the sample configuration (not shown). Select **Add Signaling Interface**.

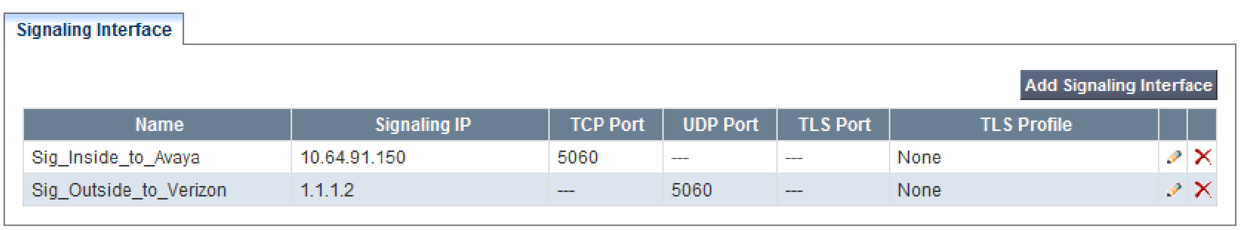
In the Add Signaling Interface screen, enter an appropriate **Name** (e.g., "Sig_Inside_to_Avaya) for the "inside" private interface, and choose the private inside IP Address (e.g., 10.64.91.150) from the **IP Address** drop-down menu. Choose **TCP Port** "5060" since TCP and port 5060 is used between Session Manager and the ASBC in the sample configuration. Click **Finish**.



Once again, select **Add Signaling Interface**. In the Add Signaling Interface screen, enter an appropriate **Name** (e.g., “Sig_Outside_to_Verizon” for the “outside” public interface, and choose the public IP Address (e.g., 1.1.1.2) from the **IP Address** drop-down menu. Choose **UDP Port** “5060”. In the sample configuration, Verizon will send SIP signaling using UDP to the CPE IP Address 1.1.1.2 and to UDP Port 5060. Click **Finish**.



The following screen shows the signaling interfaces defined for the sample configuration.



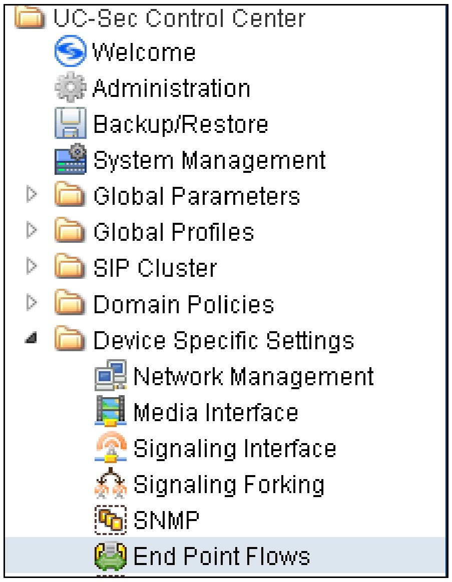
Signaling Interface

Add Signaling Interface

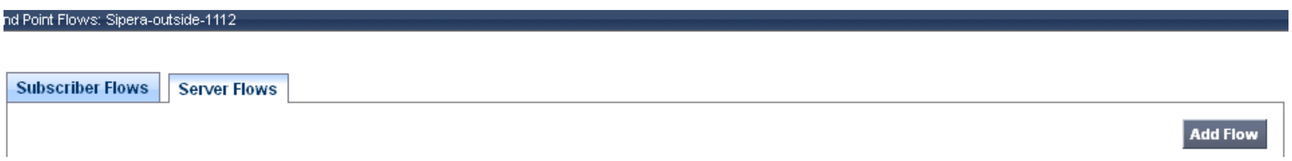
Name	Signaling IP	TCP Port	UDP Port	TLS Port	TLS Profile
Sig_Inside_to_Avaya	10.64.91.150	5060	---	---	None
Sig_Outside_to_Verizon	1.1.1.2	---	5060	---	None

6.12. Device Specific Settings – End Point Flows

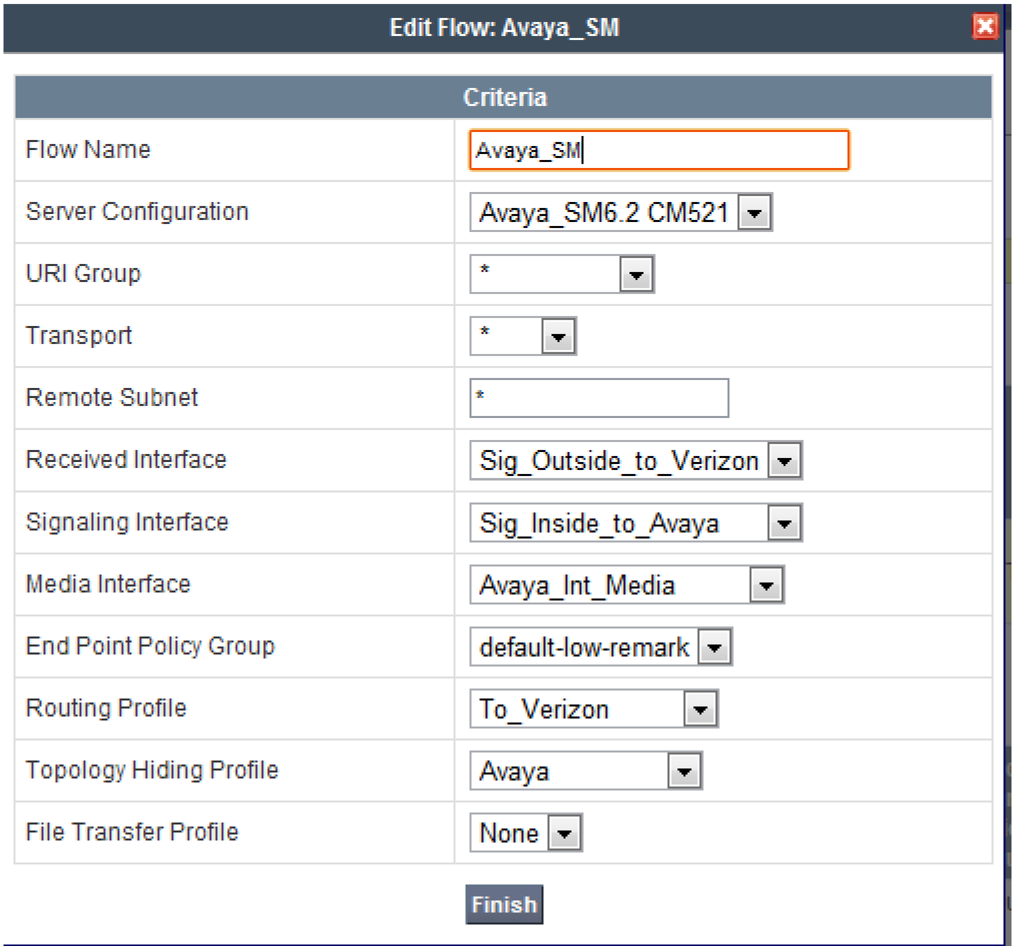
Select **Device Specific Setting → End Point Flows** from the left-side menu as shown below.



Under **UC-Sec Devices**, select the device being managed, which was named "ASBCE-3" in the sample configuration (not shown). Select the **Server Flows** tab. Select **Add Flow**.

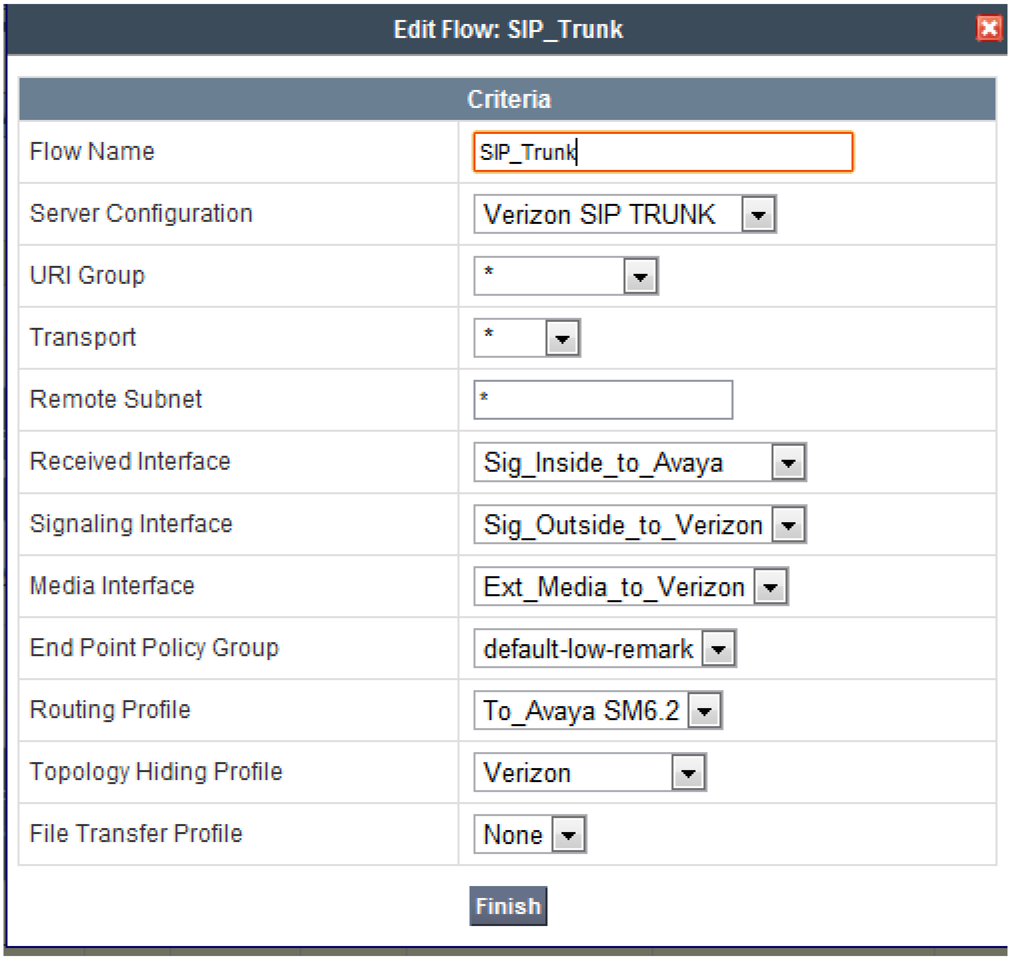


The following screen shows the flow named “Avaya_SM” being added to the sample configuration. This flow uses the interfaces, policies, and profiles defined in previous sections. Click **Finish**.



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. Click **Finish**.



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

Subscriber Flows | Server Flows

Add Flow

Click here to add a row description.

Server Configuration: Avaya_SM6.2 CM521

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_to_Verizon	Sig_Inside_to_Avaya	Avaya_Int_Media	default-low-remark	To_Verizon	Avaya	None			

Server Configuration: Verizon SIP TRUNK

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_to_Avaya	Sig_Outside_to_Verizon	Ext_Media_to_Verizon	default-low-remark	To_Avaya SM6.2	Verizon	None			

7. Verizon Business IP Trunk Services Suite Configuration

Information regarding Verizon Business IP Trunk Services suite offer can be found at http://www.verizonbusiness.com/Products/communications/ip-telephony/ or by contacting a Verizon Business sales representative.

The reference configuration described in these Application Notes was located in the Avaya Solution and Interoperability Test Lab. Access to the Verizon Business IP Trunk Services suite was via a Verizon Private IP (PIP) T1 connection. Verizon Business provided all of the necessary service provisioning.

7.1. Service Access Information

The following service access information (FQDN, IP addressing, ports, IP toll free numbers) was provided by Verizon for the sample configuration.

CPE (Avaya)	Verizon Network
adevc.avaya.globalipcom.com ***UDP port 5060***	***172.30.209.21*** ***UDP Port 5071***

IP DID Numbers
732-945-0231
732-945-0232
732-945-0233
732-945-0234

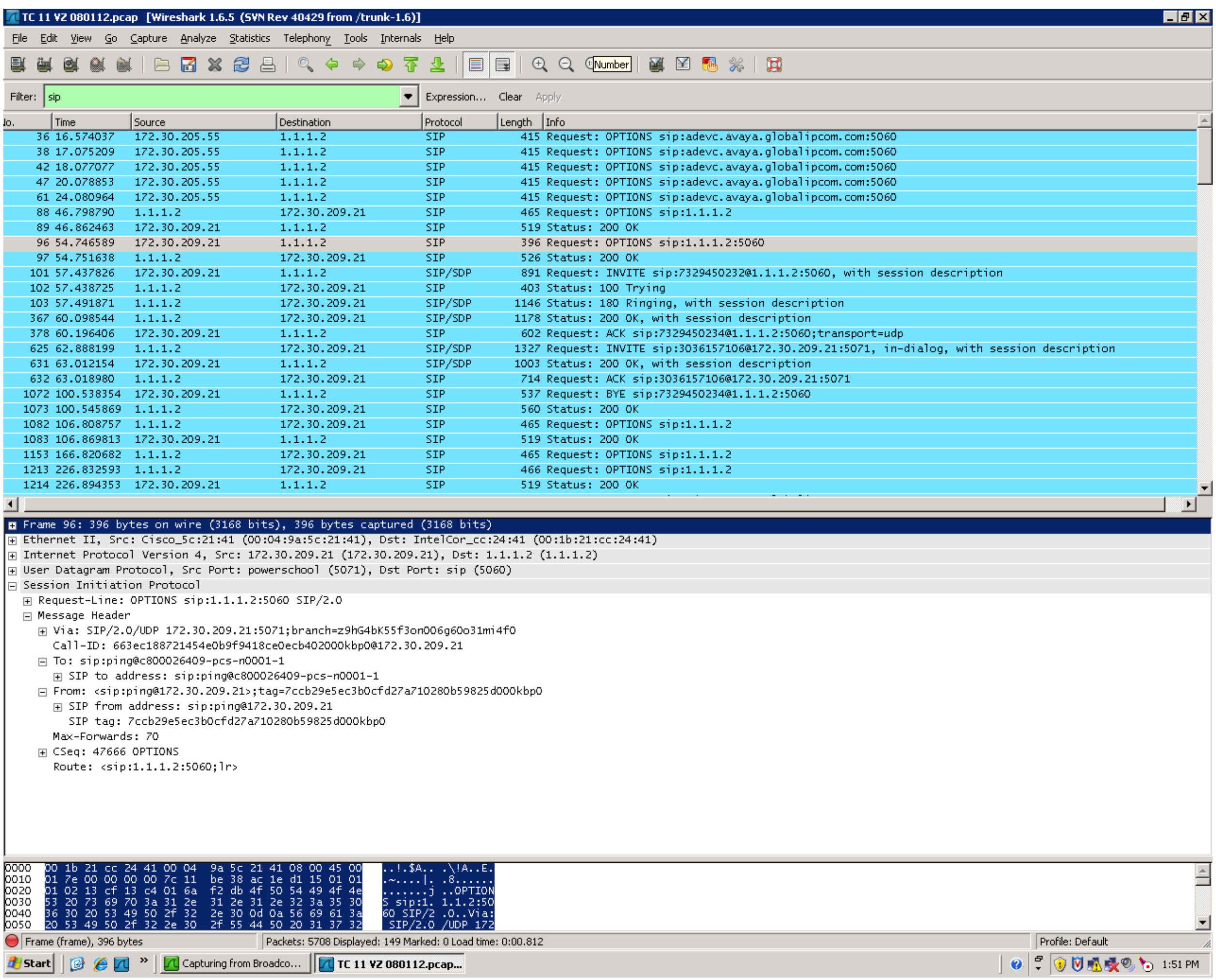
8. Verification Steps

This section provides example verifications of the Avaya configuration with Verizon Business Private IP (PIP) IP Trunk service.

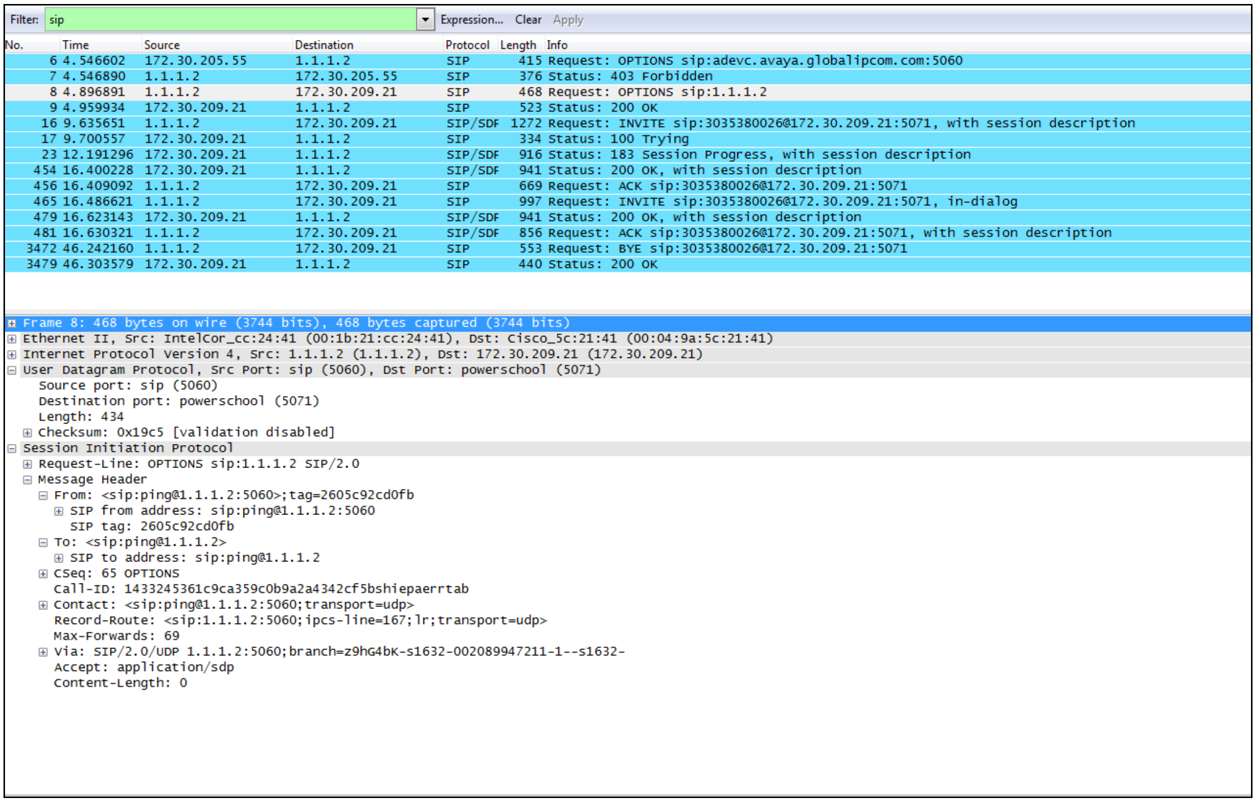
8.1. Illustration of OPTIONS Handling

This section illustrates SIP OPTIONS monitoring of the SIP trunk from Avaya CPE to Verizon the CPE and from the CPE to Verizon through the Avaya Session Border Controller for Enterprise.

The following screens from a filtered Wireshark trace illustrate OPTIONS sent by Verizon to the Avaya CPE.   Verizon IP Trunk service uses OPTIONS to determine whether the CPE is available to receive inbound calls. Therefore, proper OPTIONS response is necessary. In the trace shown below, taken from the outside public side of the SBCE, frame 96 is highlighted and expanded to show OPTIONS sent from Verizon IPC Trunk (172.30.209.21) to the SBCE (1.1.1.2). Observe the use of UDP for transport, from source port 5071 (Verizon) to destination port 5060 (Avaya). Verizon sends the Avaya domain “1.1.1.2” in the Request-Line. Note that Max-Forward is 70.



Before the SBCE replies to Verizon, the SBCE sends OPTIONS to Session Manager on the inside private interface. In the trace shown below, taken from the inside private side of the SBCE, frame 8 is highlighted and expanded to show OPTIONS sent from the inside interface of the SBCE (1.1.1.2) to Session Manager (10.80.140.160). Observe the use of UDP for transport, using port 5060. Observe that the SBCE has changed the Request-URI, From, and To headers per the previous configuration such that “avayalab.com” now appears. Note that Max-Forwards has been decremented by 1 and is now 69.



In this same trace, frame 9 below shows Verizon responding to the OPTIONS with 200 OK. The receipt of a valid OPTIONS response from the CPE is necessary for Verizon to route inbound calls to the CPE. Since the SBCE proxies the OPTIONS received from Verizon to Session Manager, the end to end path from Verizon through to Session Manager must be in-service for OPTIONS (and ultimately calls) to be successful.

8.2. Avaya Aura® Communication Manager Verifications

This section illustrates verifications from Communication Manager.

8.2.1 Example Incoming Call from PSTN via Verizon SIP Trunk

Incoming PSTN calls arrive from Verizon at Avaya ASBCE, which sends the call to Session Manager. In the sample configuration, when the ASBCE is in-service, Verizon sends all inbound calls to ASBCE-3. Session Manager sends the call to Communication Manager via the entity link corresponding to the Avaya Common Server using port 5062. On Communication Manager, the incoming call arrives via signaling group 1 and trunk group 1.

The following edited Communication Manager ***list trace tac*** trace output shows a call incoming on trunk group 1. The PSTN telephone dialed 732-945-0232. Session Manager can map the number received from Verizon to the extension of a Communication Manager telephone (x12203), or the incoming call handling table for trunk group 1 can do the same. In the trace below, Communication Manager had already mapped the Verizon DID to the Communication Manager extension. Extension 12203 is an IP Telephone with IP address 10.64.90.75 in Region 1. Initially, the G450 Media Gateway (10.64.90.112) is used, but as can be seen in the final trace output, once the call is answered, the final RTP media path is "ip-direct" from the IP Telephone (10.64.90.75) to the "inside" of the Avaya SBCE (10.64.91.150).

```
list trace previous                                                 Page   1

                                LIST TRACE

time            data

15:30:36 SIP<INVITE sip:7329450232@avayalab.com  SIP/2.0
15:30:36     Call-ID: d2dbca47c8812b411b41193fc677c650
15:30:36     active trunk-group 1 member 1    cid 0x193
15:30:36 SIP>SIP/2.0 180 Ringing
15:30:36     Call-ID: d2dbca47c8812b411b41193fc677c650
15:30:36     dial 12203
15:30:36     ring station      12203 cid 0x193
15:30:36     G711MU ss:off ps:20
             rgn:1 [10.64.90.75]:2782
             rgn:1 [10.64.90.112]:2056
15:30:36     G711MU ss:off ps:20
             rgn:10 [10.64.91.150]:35112
             rgn:1 [10.64.90.112]:2052
15:30:36     xoip options: fax:T38 modem:off tty:US  uid:0x50112
             xoip ip: [10.64.90.112]:2052
```

The following screen shows **Page 2** of the output of the ***status trunk*** command pertaining to this same call. Note the signaling using port 5060 between Communication Manager and Session Manager. Note the media is "ip-direct" from the IP Telephone (10.64.90.75) to the inside IP address of SBCE (10.64.91.150) using G.711MU.

```
status trunk 1/1                                                Page   2 of   3
                                CALL CONTROL SIGNALING

Near-end Signaling Loc: 01A0017
  Signaling   IP Address                                Port
   Near-end:  10.80.140.180                           : 5060
    Far-end:  10.80.140.160                           : 5060
 H.245 Near:
  H.245 Far:
   H.245 Signaling Loc:           H.245 Tunneled in Q.931? no

 Audio Connection Type: ip-direct     Authentication Type: None
    Near-end Audio Loc:                        Codec Type: G.711MU
   Audio      IP Address                                Port
   Near-end:  10.64.90.75                             : 2782
    Far-end:  10.64.91.150                            : 35112

 Video Near:
  Video Far:
 Video Port:
  Video Near-end Codec:               Video Far-end Codec:
```

8.2.2 Example Outgoing Calls to PSTN via Verizon IP Trunk

The following edited trace shows an outbound ARS call from IP Telephone x12204 to the PSTN number 9-303-538-0026. The call is routed to route pattern 1 and trunk group 1. The call initially uses the gateway (10.64.90.112), but after the call is answered, the call is "shuffled" to become an "ip-direct" connection between the IP Telephone (10.64.90.74) and the "inside" of the Avaya SBCE-3 (10.64.91.150).

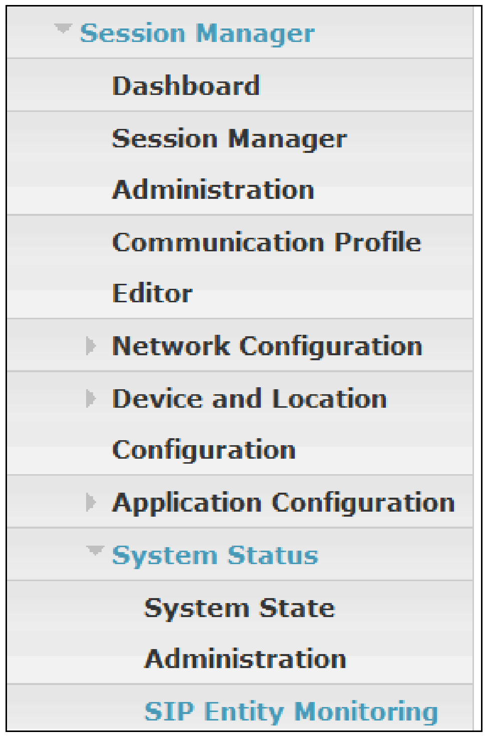
```
list trace tac *101                                             Page   1

                                LIST TRACE

time            data

16:02:50     dial 93035380026 route:HNPA|ARS
16:02:50     route-pattern  1 preference 1 location 1/ALL  cid 0x195
16:02:50     seize trunk-group 1 member 6    cid 0x195
16:02:50     Calling Number & Name 12204 IP 9641
16:02:50 SIP>INVITE sip:3035380026@avayalab.com SIP/2.0
16:02:50     Call-ID: 0c461a560f0e1175b503f896600
16:02:50     Setup digits 3035380026
16:02:50     Calling Number & Name 7329450233 IP 9641
16:02:50 SIP<SIP/2.0 100 Trying
16:02:50     Call-ID: 0c461a560f0e1175b503f896600
16:02:50     Proceed trunk-group 1 member 6    cid 0x195
16:02:53 SIP<SIP/2.0 183 Session Progress
16:02:53     Call-ID: 0c461a560f0e1175b503f896600
16:02:53     G711MU ss:off ps:20
             rgn:10 [10.64.91.150]:35114
             rgn:1 [10.64.90.112]:2050
16:02:53     xoip options: fax:T38 modem:off tty:US  uid:0x50117
             xoip ip: [10.64.90.112]:2050
16:02:53 SIP<SIP/2.0 200 OK
16:02:53     Call-ID: 0c461a560f0e1175b503f896600
16:02:53 SIP>ACK sip:3035380026@10.64.91.150:5060;transport=tcp SIP/
16:02:53 SIP>2.0
16:02:53     Call-ID: 0c461a560f0e1175b503f896600
16:02:53     active trunk-group 1 member 6    cid 0x195
16:02:53 SIP>INVITE sip:3035380026@10.64.91.150:5060;transport=tcp S
16:02:53 SIP>IP/2.0
16:02:53     Call-ID: 0c461a560f0e1175b503f896600
16:02:53 SIP<SIP/2.0 100 Trying
16:02:53     Call-ID: 0c461a560f0e1175b503f896600
16:02:53 SIP<SIP/2.0 200 OK
16:02:53     Call-ID: 0c461a560f0e1175b503f896600
time            data
16:02:53     G711MU ss:off ps:20
             rgn:1 [10.64.90.74]:2904
             rgn:10 [10.64.91.150]:35114
16:02:53 SIP>ACK sip:3035380026@10.64.91.150:5060;transport=tcp SIP/
16:02:53 SIP>2.0
16:02:53     Call-ID: 0c461a560f0e1175b503f896600
16:02:53     G711MU ss:off ps:20
             rgn:10 [10.64.91.150]:35114
             rgn:1 [10.64.90.74]:2904
16:03:06 TRACE COMPLETE trunk-group  1 cid 0x195
```

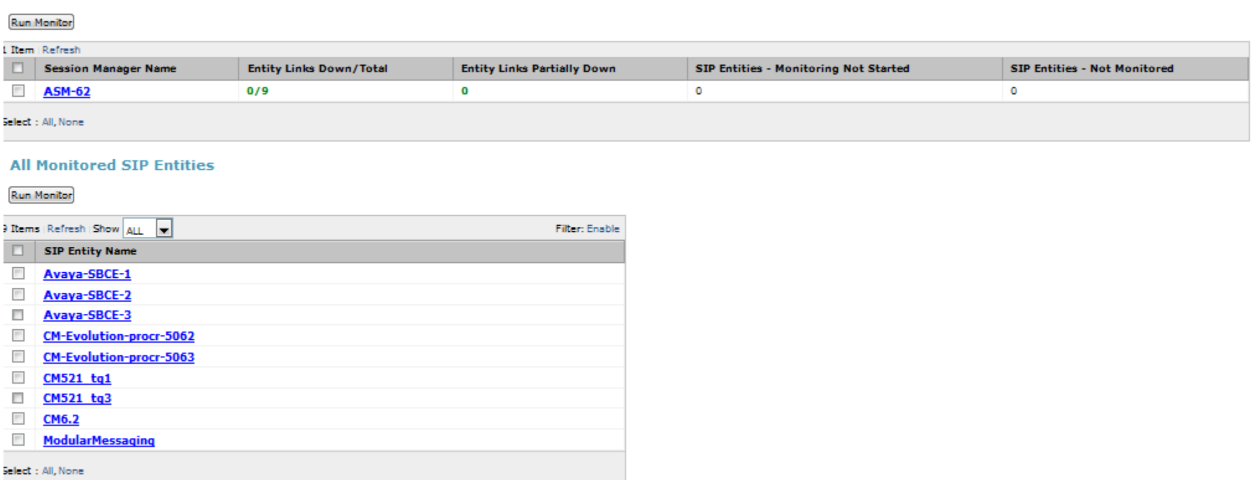
8.3. Avaya Aura® System Manager and Avaya Aura® Session Manager Verifications

This section contains verification steps that may be performed using System Manager for Session Manager.

8.3.1 Verify SIP Entity Link Status

Log in to System Manager. Expand **Elements** → **Session Manager** → **System Status** → **SIP Entity Monitoring**, as shown below.





Run Monitor

1 Item | Refresh

Session Manager Name	Entity Links Down/Total	Entity Links Partially Down	SIP Entities - Monitoring Not Started	SIP Entities - Not Monitored
ASM-62	0/9	0	0	0

Select : All, None

All Monitored SIP Entities

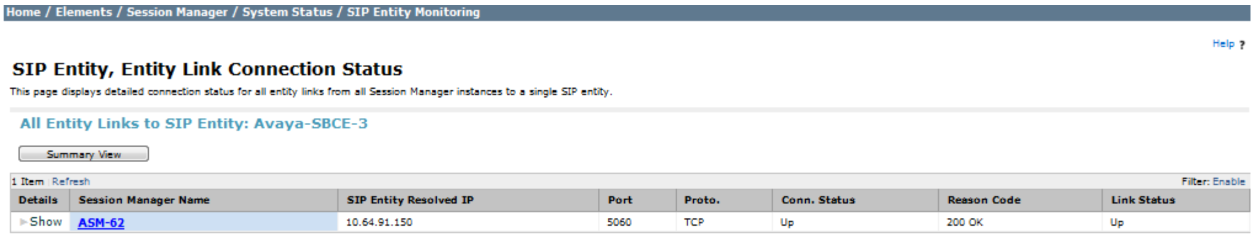
Run Monitor

9 Items | Refresh | Show ALL | Filter: Enable

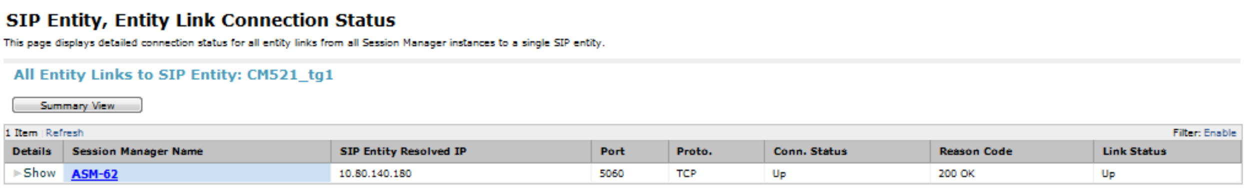
SIP Entity Name
Avaya-SBCE-1
Avaya-SBCE-2
Avaya-SBCE-3
CM-Evolution-procr-5062
CM-Evolution-procr-5063
CM521_tg1
CM521_tg3
CM6.2
ModularMessaging

Select : All, None

From the list of monitored entities, select an entity of interest, such as “Avaya-SBCE-3”. Under normal operating conditions, the **Link Status** should be “Up” as shown in the example screen below.

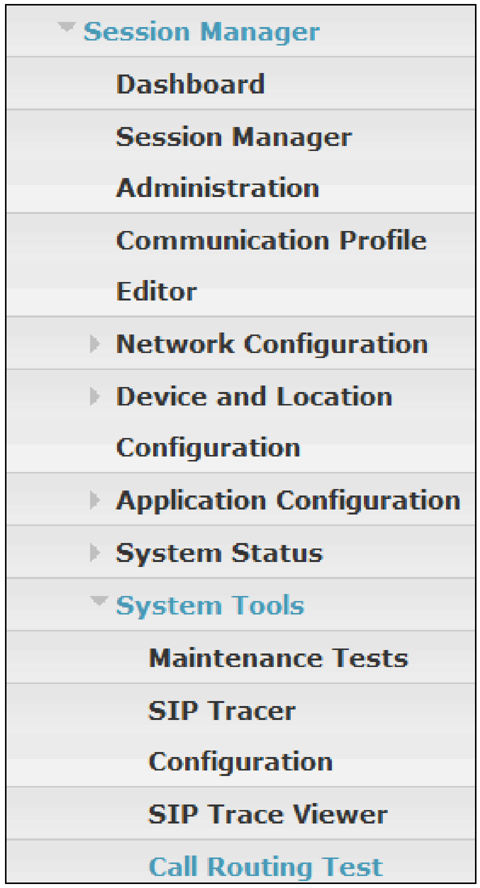


Return to the list of monitored entities, and select another entity of interest, such as “CM521-tg1”. Under normal operating conditions, the **Link Status** should be “Up” as shown in the example screen below.

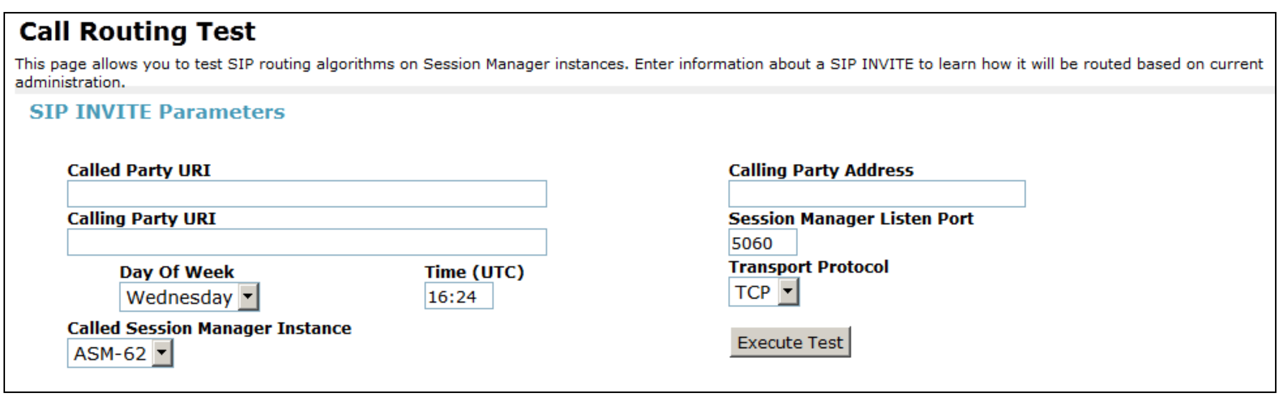


8.3.2 Call Routing Test

The **Call Routing Test** verifies the routing for a particular source and destination. To run the routing test, expand **Elements** → **Session Manager** → **System Tools** → **Call Routing Test**, as shown below.

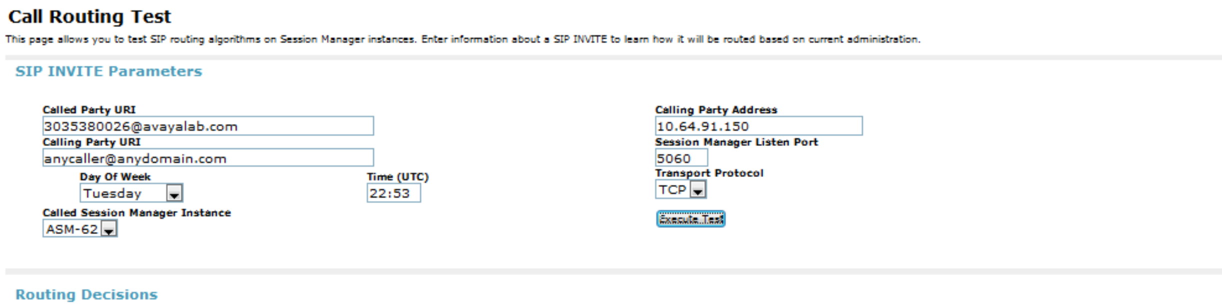


A screen such as the following is displayed.



Populate the fields for the call parameters of interest. For example, the following screen shows an example call routing test for an outbound call to the PSTN via Verizon. Under **Routing Decisions**, observe that the call will route via an Avaya SBCE on the path to Verizon

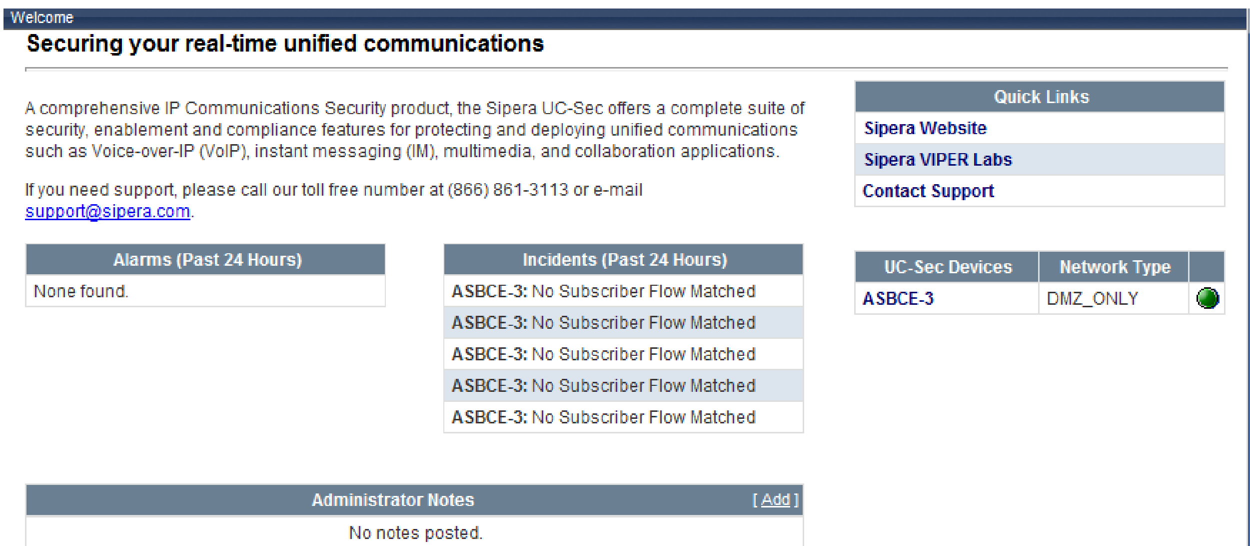
Scroll down to inspect the details of the **Routing Decision Process** if desired (not shown).



8.4. Avaya Session Border Controller for Enterprise Verification

8.4.1 Welcome Screen

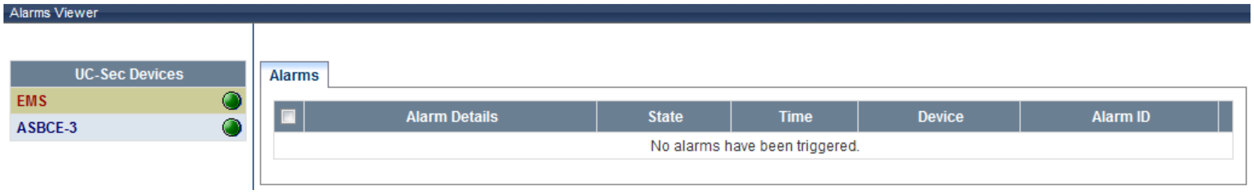
The welcome screen shows alarms, incidents, and the status of all managed ASBCEs at a glance.



8.4.2 Alarms

A list of the most recent alarms can be found under the Alarm tab on the top left bar.

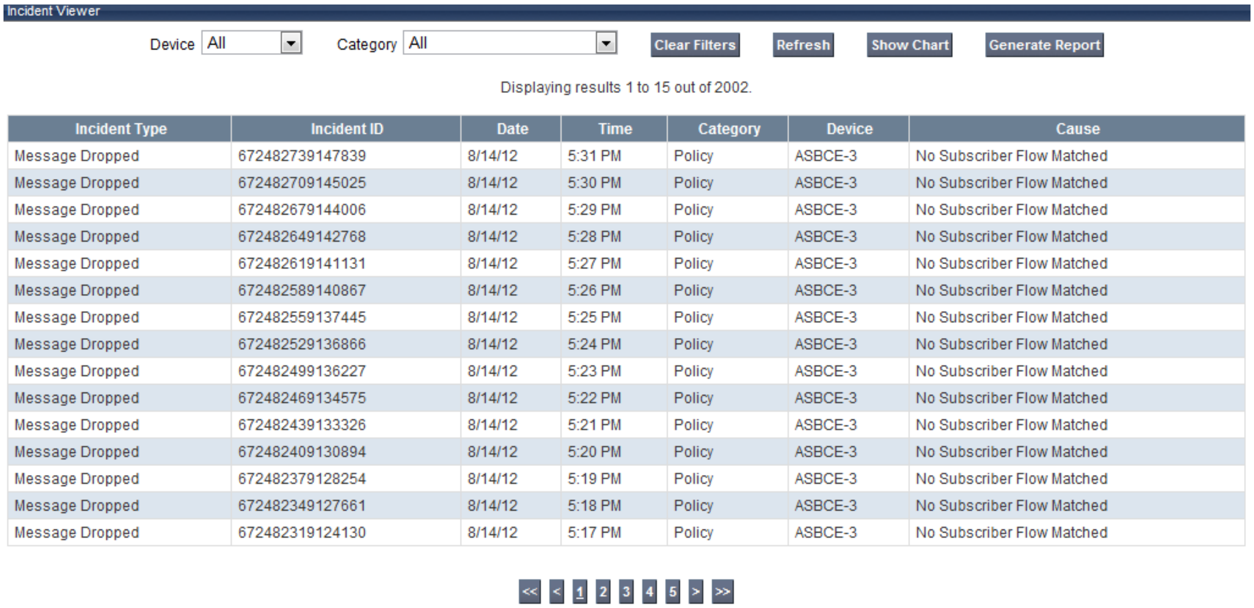
Alarms Viewer.



8.4.3 Incidents

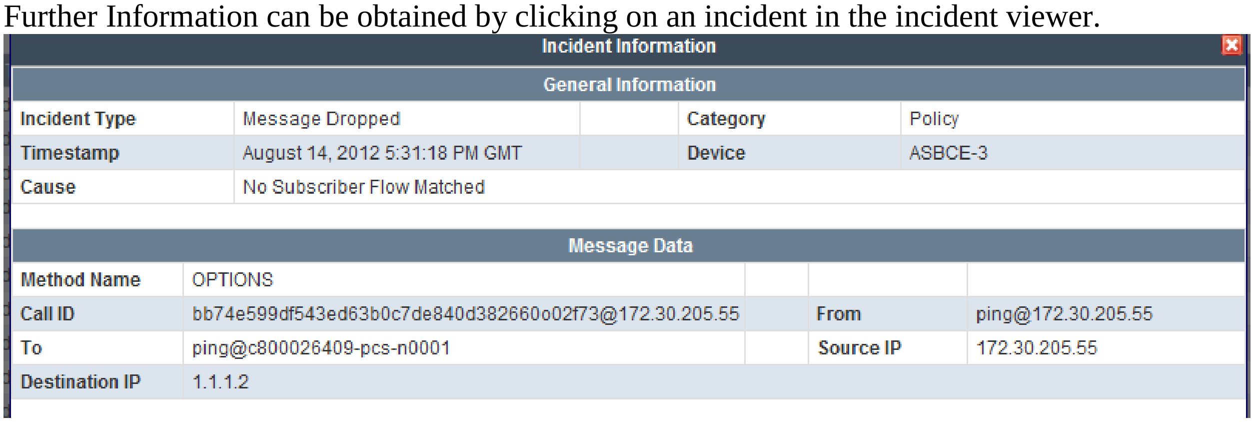
A list of all recent incidents can be found under the incidents tab at the top left next to the Alarms.

Incident Viewer



Incident Type	Incident ID	Date	Time	Category	Device	Cause
Message Dropped	672482739147839	8/14/12	5:31 PM	Policy	ASBCE-3	No Subscriber Flow Matched
Message Dropped	672482709145025	8/14/12	5:30 PM	Policy	ASBCE-3	No Subscriber Flow Matched
Message Dropped	672482679144006	8/14/12	5:29 PM	Policy	ASBCE-3	No Subscriber Flow Matched
Message Dropped	672482649142768	8/14/12	5:28 PM	Policy	ASBCE-3	No Subscriber Flow Matched
Message Dropped	672482619141131	8/14/12	5:27 PM	Policy	ASBCE-3	No Subscriber Flow Matched
Message Dropped	672482589140867	8/14/12	5:26 PM	Policy	ASBCE-3	No Subscriber Flow Matched
Message Dropped	672482559137445	8/14/12	5:25 PM	Policy	ASBCE-3	No Subscriber Flow Matched
Message Dropped	672482529136866	8/14/12	5:24 PM	Policy	ASBCE-3	No Subscriber Flow Matched
Message Dropped	672482499136227	8/14/12	5:23 PM	Policy	ASBCE-3	No Subscriber Flow Matched
Message Dropped	672482469134575	8/14/12	5:22 PM	Policy	ASBCE-3	No Subscriber Flow Matched
Message Dropped	672482439133326	8/14/12	5:21 PM	Policy	ASBCE-3	No Subscriber Flow Matched
Message Dropped	672482409130894	8/14/12	5:20 PM	Policy	ASBCE-3	No Subscriber Flow Matched
Message Dropped	672482379128254	8/14/12	5:19 PM	Policy	ASBCE-3	No Subscriber Flow Matched
Message Dropped	672482349127661	8/14/12	5:18 PM	Policy	ASBCE-3	No Subscriber Flow Matched
Message Dropped	672482319124130	8/14/12	5:17 PM	Policy	ASBCE-3	No Subscriber Flow Matched

Further Information can be obtained by clicking on an incident in the incident viewer.

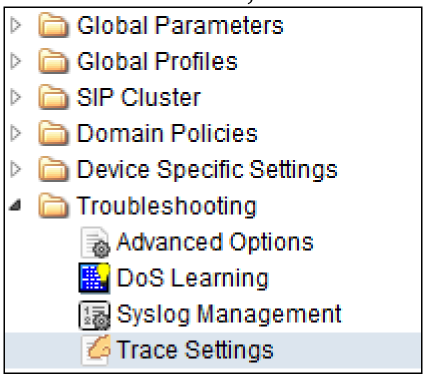


Incident Type	Message Dropped	Category	Policy
Timestamp	August 14, 2012 5:31:18 PM GMT	Device	ASBCE-3
Cause	No Subscriber Flow Matched		

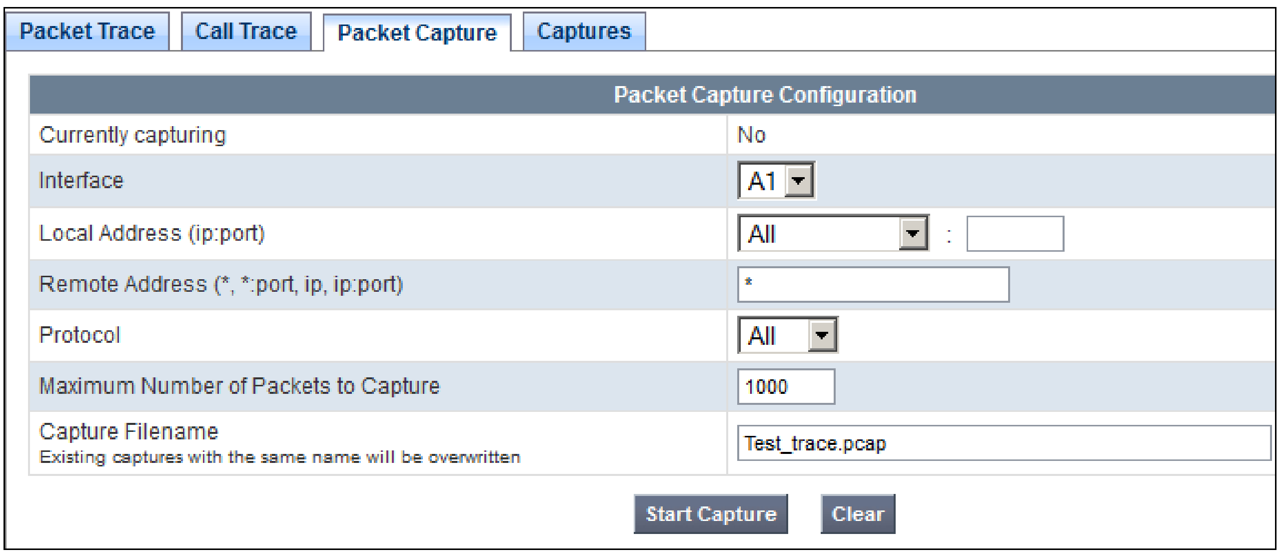
Method Name	OPTIONS		
Call ID	bb74e599df543ed63b0c7de840d382660o02f73@172.30.205.55	From	ping@172.30.205.55
To	ping@c800026409-pcs-n0001	Source IP	172.30.205.55
Destination IP	1.1.1.2		

8.4.4 Tracing

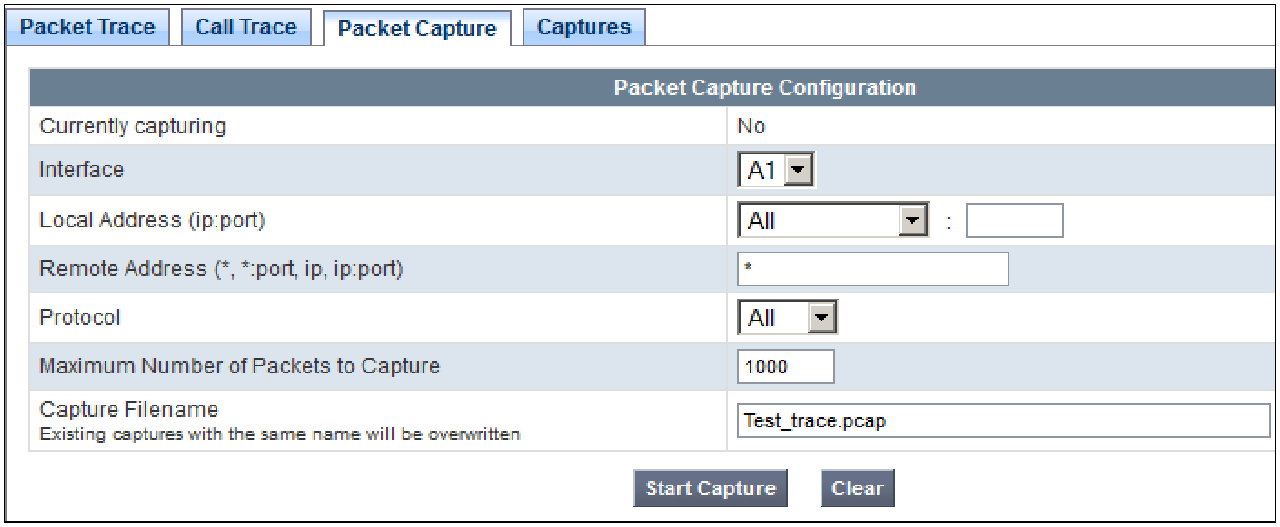
To take a call trace, Select **Troubleshooting → Tracing** from the left-side menu as shown below.



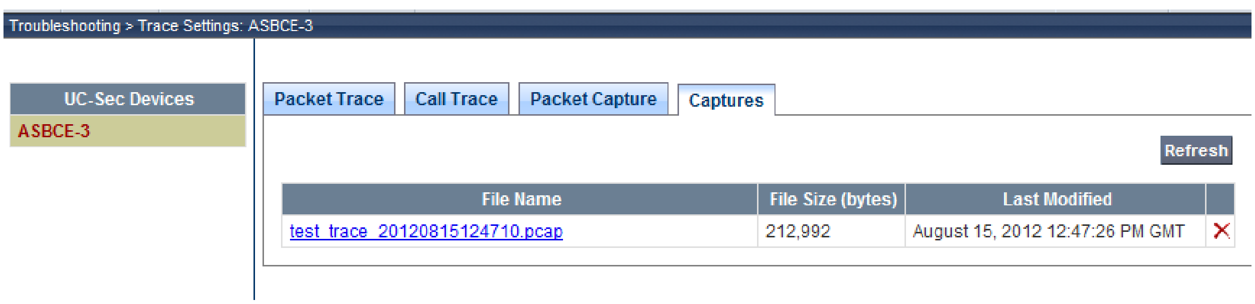
Select the Packet Capture tab and set the desired configuration for a call trace, hit **Start Capture**. Only one interface can be selected at once, so only an inside or only an outside trace is possible.



When tracing has reached the desired number of packets the trace will stop automatically, or alternatively, hit the Stop Capture button at the bottom not shown.



Select the Captures tab at the top and your capture will be listed, you can select the File Name and choose to open it with an application like Wireshark.



9. Conclusion

As illustrated in these Application Notes, Avaya Aura® Communication Manager 5.2.1, Avaya Aura® Session Manager 6.2, and the Avaya Session Border Controller for Enterprise can be configured to interoperate successfully with Verizon Business IP Trunk service. This solution allows Avaya Aura® Communication Manager and Avaya Aura® Session Manager user's access to the PSTN using a Verizon Business IP Trunk public SIP trunk service connection.

10. Additional References

10.1. Avaya

Avaya product documentation, including the following, is available at http://support.avaya.com

[1] *Installing and Configuring Avaya Aura® Communication Manager*, Doc ID 03-603558, Release 5.2.1
[2] *Administering Avaya Aura® Communication Manager*, Doc ID 03-300509
[3] *Administering Avaya Aura® Session Manager*, Doc ID 03-603324
[4] *Installing and Configuring Avaya Aura® Session Manager*, Doc ID 03-603473
[5] *Maintaining and Troubleshooting Avaya Aura® Session Manager*, Doc ID 03-603325
[6] *Administering Avaya Aura® System Manager*, Document Number 03-603324

Avaya Application Notes are also available at http://support.avaya.com

Application Notes Reference [LAR] contains additional information on Communication Manager Look-Ahead Routing.
[LAR] Sample Configuration for SIP Private Networking and SIP Look-Ahead Routing Using Avaya Communication Manager, Issue 1.0
http://www.avaya.com/master-usa/en-us/resource/assets/applicationnotes/sip-pvt-lar.pdf

10.2. Verizon Business

The following documents may be obtained by contacting a Verizon Business Account Representative.

[7] *Retail VoIP Interoperability Test Plan*
[8] *Network Interface Specification Retail VoIP Trunk Interface (for non-registering devices)*

Appendix A: Unscreened ANI Testing and Configuration

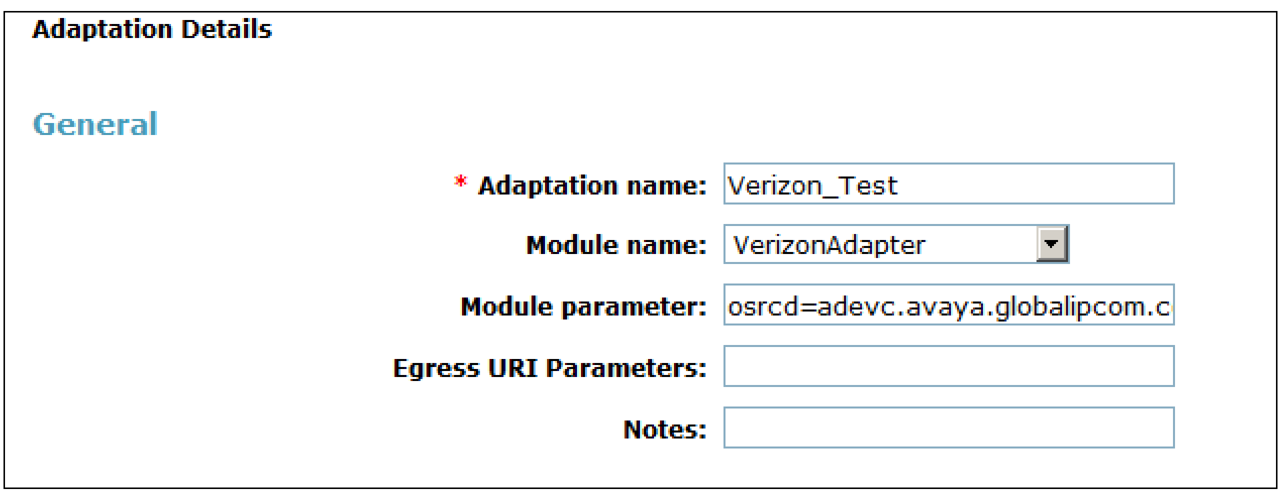
Unscreened ANI is a Verizon offered service (available with VoIP IP Integrated Access and VoIP IP Trunking) and is a new feature being offered with Session Manager 6.2. This service was tested successfully in this test configuration and can be implemented by following the steps here.

This feature allows Customer to send an "unscreened" ANI to the Company's network which is then displayed to the called party as Caller ID. An "unscreened" ANI can be any telephone number that Customer passes through the Company's network for Caller ID display purposes only. There is no charge for this feature. If Customer selects this feature, Verizon will designate one of Customer's assigned telephone numbers as a "Screened Telephone Number" for each Customer unique location. Verizon will use the Screened Telephone Number to determine call origination for billing, call routing and E911 support. The customer is responsible for configuring its IP-PBX, PBX or other devices to accommodate and properly process the Screened Telephone Number.

The screened telephone number provided by Verizon for this test is 732-945-0821. Typically customers would have one or more screened telephone number, one for every location and a central Session Manager could be used to pass multiple screened telephone numbers to Verizon based on a Matching Pattern (i.e. a user's CLID).

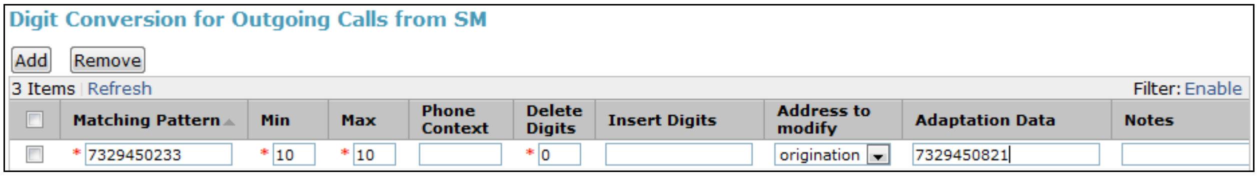
Login to Session Manager as shown above, navigate to Routing→Adaptations, and select "New".

Create a unique name for the Adaptation, here "Verizon_Test". Select the "VerizonAdapter" for the **Module Name**. In module parameter enter any domain adaptions that may be needed. Here the domains known to Verizon needed to overwrite the internal lab environment name of "avayalab.com" so a **Module Parameter** of "osrcd=adevc.avaya.globalipcom.com odstd=pcelban0001.avayalincroft.globalipcom.com fromto=true" was used.

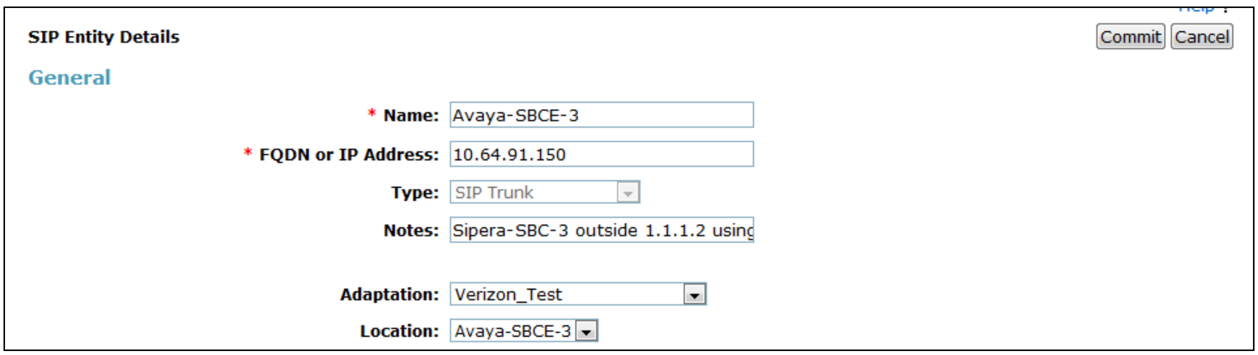


Scroll down to the **Digit Conversion for Outgoing Calls from SM** section, enter a **Matching Pattern** (e.g. 732-945-0233), with the **Min** and **Max** number of digits to match on, in **Address to**

modify, select origination, and in the **Adaptation Data** enter the screened telephone number (e.g. 732-945-0821) provided by Verizon. Hit **Commit**.



Once the Adaptation has been committed it needs to be applied to a SIP Entity. Back at the Routing screen, select SIP Entities as shown in the Session manager section above, and select the Avaya SBCE-1. Under Adaptation, change to the newly created “Verizon_Test” adaptation.

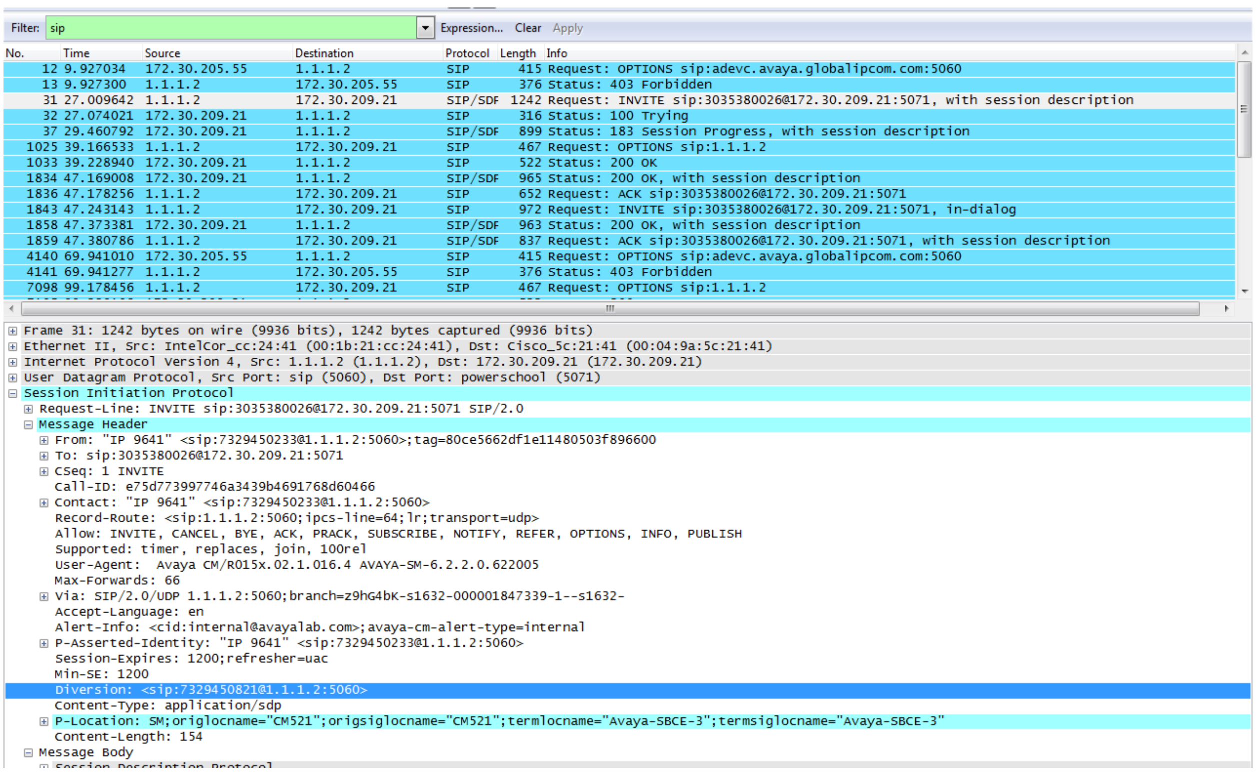


Verification

In the following filter Wireshark trace, you can see that the From line contains the DID number, 732-945-0233 and in the p-asserted identity section, a Diversion headeer has been added with the screened ANI (732-945-0821).

From: "IP 9641 - SIP" <sip:**7329450233**@1.1.1.2:5060>;tag=80ce5662df1e11480503f896600

Diversion: sip:**7329450821**@1.1.1.2:5060>



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.