Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 6.3, Avaya Aura® Session Manager 6.3, and Avaya Session Border Controller for Enterprise 6.2 with Verizon Business IP Trunk SIP Trunk Service – Issue 1.2

Abstract

These Application Notes illustrate a sample configuration using Avaya Aura® Session Manager Release 6.3, Avaya Aura® Communication Manager Release 6.3, and Avaya Session Border Controller for Enterprise Release 6.2 with the Verizon Business Private IP (PIP) IP Trunk service. These Application Notes update previously published Application Notes with newer versions of Communication Manager and Session Manager.

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.

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 6.3, Avaya Aura® Communication Manager Release 6.3, and Avaya Session Border Controller for Enterprise Release 6.2 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.

2. General Test Approach and Test Results

The test approach was manual testing of inbound and outbound calls using the Verizon IP Trunk SIP Trunk Service on a production Verizon PIP access circuit, as shown in **Figure 1**.

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

2.1. Interoperability Compliance Testing

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

- Inbound and outbound voice calls between telephones controlled by 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., Communication Manager Messaging, Avaya vector digit collection steps)
- Additional PSTN numbering plans (e.g., International, operator assist, 411)
- 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
- Long hold time calls

2.2. Test Results

Interoperability testing of Verizon Business IP Trunk SIP Trunk Service was completed with successful results for all test cases. The following limitations are noted for the sample configuration described in these Application Notes.

- Verizon provisioned T.38 Fax on the production circuit used to verify these Application Notes. Verizon Business IP Trunk service requires all fax calls to start off with G.711 as the first codec choice, and relies on the CPE to send a re-Invite to T.38 when placing or receiving a fax call. If the **FAX Mode** field on the Communication Manager ip-codec-set form page 2 is set to "t.38-standard" (see **Section 5.6**), Communication Manager will send the proper re-Invite to T.38, but will not failback to G.711 should the Verizon network reject the Communication Manager attempt to transition to T.38 by sending a 488 Not Acceptable message. Communication Manager Release 6.3 introduces the T.38 Fax with Fallback to G.711 Pass-Through feature. This provides the functionality for Communication Manager to interoperate with Verizon networks by re-Inviting to G.711 after receiving a 488 Not Acceptable message. If the **FAX Mode** is set to the new "t.38-G711-fallback" setting[1], Communication Manager will send a re-Invite to T.38 for inbound fax calls only and relies on the far end to send a re-Invite to T.38 for outbound calls. Communication Manager assumes T.38 fax is not supported for an outbound fax call unless an Invite for T.38 is received. The result is an outbound fax sent using G.711, even though the circuit is provisioned for T.38. Inbound fax calls negotiate properly to T.38. With the limitations of T.38 on Verizon's network and Verizon's requirement for fax calls to start off with G.711 as the first codec choice, it is recommended to use an AudioCodes MP-114 or MP-124 Gateway between Session Manager and the fax device when fax is used with Verizon IP Trunk service.

- When the **Initial IP-IP Direct Media** field on the Communication Manager signaling group form page 1 is set to 'y", Communication Manager sends a "183 Session Progress" without SDP during an inbound PSTN call that is forwarded to another PSTN call just before a 183 is sent with SDP information to the far end. This is undesirable to Verizon and results in no audio. The recommendation in **Section 5.7** is to leave the **Initial IP-IP Direct Media** field to "n". As a safeguard, an Avaya SBCE Server Interworking Profile in **Section 7.4.2** includes a parameter to insure Verizon always receives "183 Session Progress" with SDP.

- 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.

- Verizon Business IP Trunking service does not support G.711a codec for domestic service (EMEA only).

- Verizon Business IP Trunking service does not support G.729B codec.

[1] The "T.38 Fax with Fallback to G.711 Pass-Through" feature requires G430 or G450 Media Gateways with release 33.13 or higher.

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.

2.3. 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 the SIP Diversion Header be sent for redirected calls. The Communication Manager SIP trunk group form provides the options for specifying whether History Info Headers or Diversion Headers are sent.

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

The Communication Manager Call Forwarding or Extension to Cellular (EC500) features may be used for the call scenarios testing the Diversion Header.

2.4. The SIP Trunk Redundant (2-CPE) Architecture Option

Verizon Business and Avaya developed the SIP Trunk Redundant (2-CPE) architecture to ensure that SIP trunk calls can be automatically rerouted to bypass SIP trunk failures due to network or component outages. The 2-CPE architecture described is based on a customer location having two Avaya Session Border Controllers for Enterprise. One Avaya SBCE is designated as Primary and one as Secondary. The Avaya SBCEs reside at the edge of the customer network.

Avaya Aura® Session Manager is provisioned to attempt outbound calls to the Primary Avaya SBCE first. If that attempt fails, the Secondary Avaya SBCE is used. Similarly, the Verizon Business Private IP Trunk service node will send inbound calls to the Primary Avaya SBCE. If there is no response then the call will be sent to the Secondary Avaya SBCE. For more information on how to configure 2-CPE see [MO-VZIPT-SM62].

2.5. Support

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

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

3. 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 receives traffic from the Verizon Business IP Trunk service on port 5060 and sends 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 can be mapped by Avaya Aura® Session Manager or Avaya Aura® Communication Manager to Avaya telephone extensions.

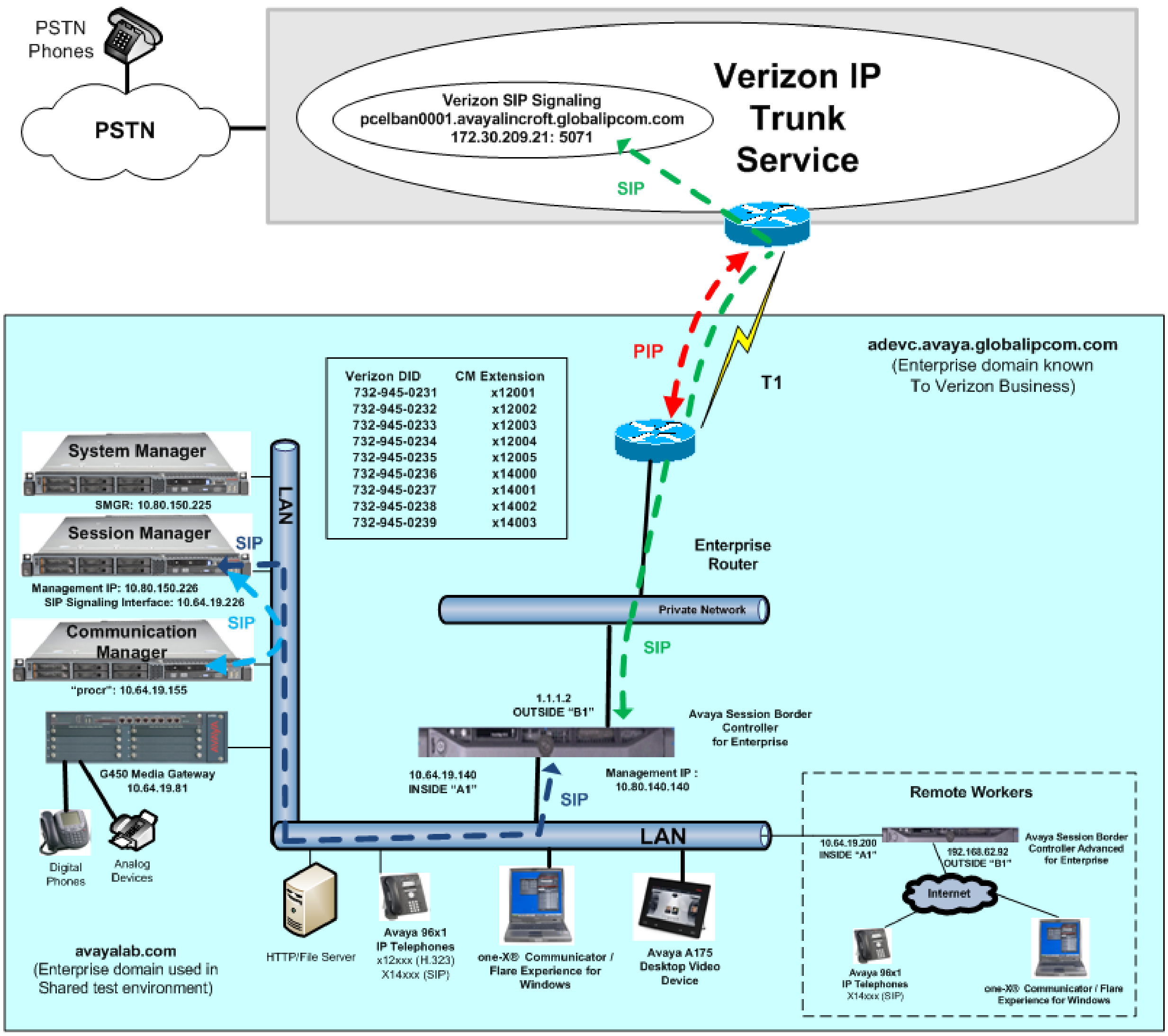


Figure 1: Avaya Interoperability Test Lab Configuration

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, the Avaya SBCE is 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*
- Avaya Session Border Controllers for Enterprise Release 6.2
- Avaya Aura® Communication Manager Release 6.3
- Avaya Aura® Session Manager Release 6.3
- Avaya 96X1 Series IP telephones using the SIP and H.323 software bundle
- Avaya 96X0 Series IP telephones using the H.323 software bundle
- Avaya Digital Phones
- Remote Workers
  - Avaya Session Border Controller Advanced for Enterprise Release 6.2
  - Avaya 96X1 Series IP telephones using the SIP software bundle
  - Avaya Flare® Experience for Windows

3.1. Remote Workers

In the sample configuration, remote Avaya SIP endpoints connected through Avaya SBCE with Advanced Services licensing were used along with local Avaya endpoints in the verification of these Application Notes. The figure below illustrates a detailed view of the Remote Workers section previously shown in **Figure 1**. Although not the primary focus of these Application Notes, relevant configuration parameters of the Avaya SBCE for use with Remote Worker are illustrated in **Appendix A**.

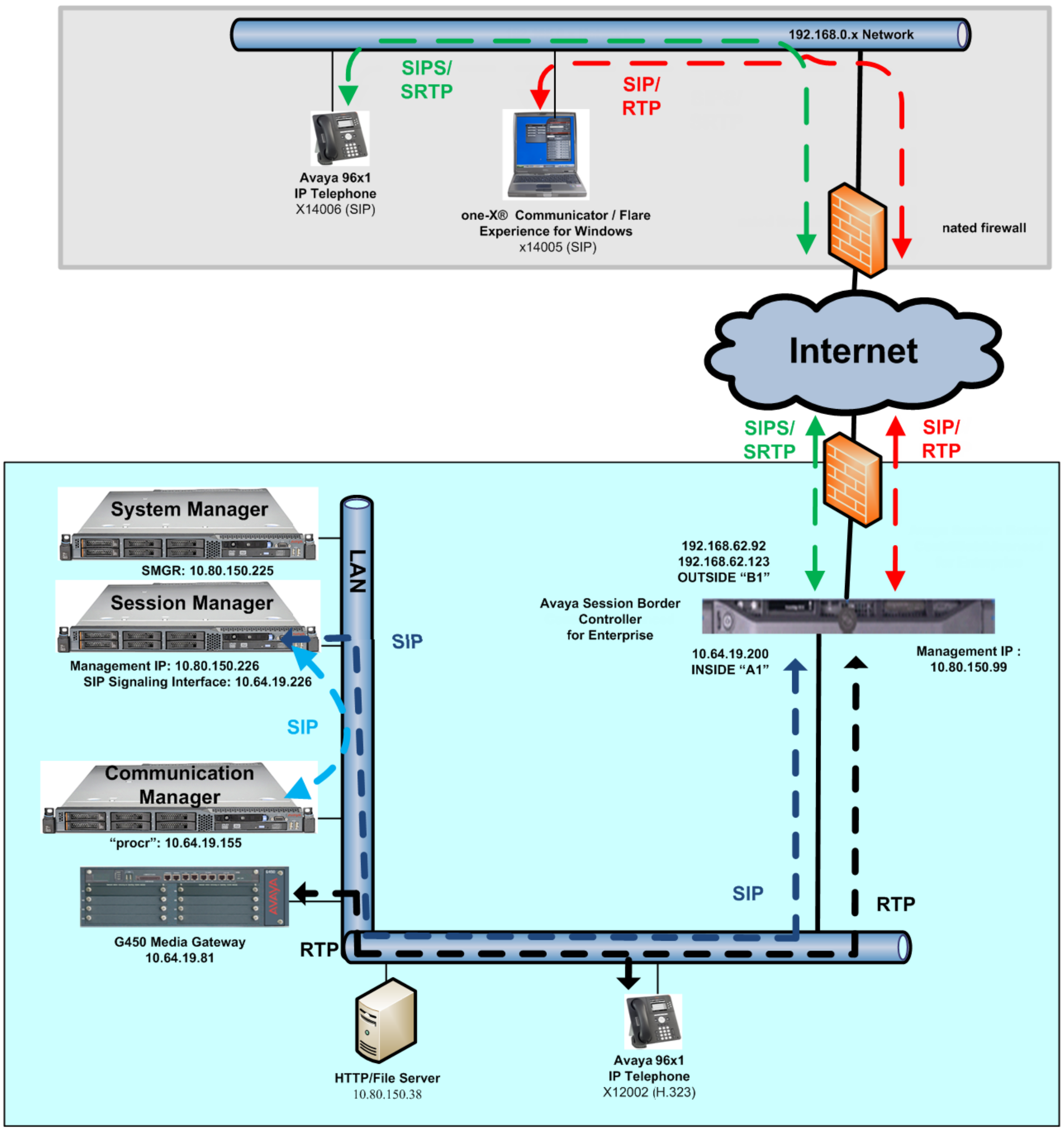


Figure 2: Remote Worker Lab Configuration

4. Equipment and Software Validated

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

Equipment:	Software:
HP ProLiant DL360 G7	Avaya Aura® Communication Manager Release 6.3 SP0
HP ProLiant DL360 G7	Avaya Aura® System Manager 6.3 SP2
HP ProLiant DL360 G7	Avaya Aura® Session Manager 6.3 SP2
G450 Gateway	33.13.0
DELL 210 RII	Avaya Session Border Controller for Enterprise Version 6.2 Q36
Avaya 96X0-Series Telephones (H.323)	R 3.2
Avaya 96X1- Series Telephones (SIP)	R6.2.2.17
Avaya 96X1- Series Telephones (H323)	R6.2313
Avaya One-X Communicator (H.323)	6.1.8.06-SP8-40314
Avaya Flare® Experience for Windows	1.1.2.11
Avaya Desktop Video Device	Flare 1.1.3
Avaya 2400-Series and 6400-Series Digital Telephones	N/A
AudioCodes MP-114	6.20A.035.001
Okidata Analog Fax	N/A

Table 1: Equipment and Software Used in the Sample Configuration

5. Configure Avaya Aura® Communication Manager Release 6.3

This section illustrates an example configuration allowing SIP signaling via the "Processor Ethernet" of the Avaya HP Server to Session Manager. In configurations that use an Avaya G650 Media Gateway, it is also possible to use an Avaya C-LAN in the Avaya G650 Media Gateway for SIP signaling to Session Manager.

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.

5.1. Verify Licensed Features

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** is 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. Each call from a SIP endpoint to the Verizon Business IP Trunk service uses two SIP trunks 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: 12000 0
             Maximum Concurrently Registered IP Stations: 18000 3
               Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
                Maximum Concurrently Registered IP eCons: 128   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                       Maximum Video Capable Stations: 36000 3
                   Maximum Video Capable IP Softphones: 18000 1
                       Maximum Administered SIP Trunks: 12000 40
  Maximum Administered Ad-hoc Video Conferencing Ports: 12000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 10    0
                     Maximum Media Gateway VAL Sources: 250   2
           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
```

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? n                      DCS (Basic)? y
          ASAI Link Core Capabilities? n                DCS Call Coverage? y
          ASAI Link Plus Capabilities? n               DCS with Rerouting? y
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n    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? n
                  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? y             Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                   Malicious Call Trace? y
     External Device Alarm Admin? y               Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n    Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? y               Multifrequency Signaling? y
      Global Call Classification? y        Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y     Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? y              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? n
                                              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? y
                                                                    Wireless? n
                          Remote Office? y
          Restrict Call Forward Off Net? y
                  Secondary Data Module? y
```

5.2. Dial Plan

In the reference configuration, the Avaya CPE environment uses five digit local extensions such as 12xxx, 14xxx or 20xxx. Trunk Access Codes (TAC) are 3 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: 1

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

5.3. 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 “SM63” with IP address “10.64.19.226”. The node name and IP address for the Processor Ethernet “procr” is “10.64.19.155”.

```
change node-names ip                                               Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
SM63                10.64.19.226
default             0.0.0.0
procr               10.64.19.155
procr6              ::
```

5.4. Processor Ethernet Configuration on HP 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 sample configuration. While the focus here is the use of the PE for SIP Trunk Signaling, observe that the Processor Ethernet will also be used for registrations from H.323 IP Telephones and H.248 gateways in the sample configuration.

```
change ip-interface procr                                       Page   1 of   2
                                IP INTERFACES


                   Type: PROCR
                                                    Target socket load: 1700

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

                               IPV4 PARAMETERS
             Node Name: procr                    IP Address: 10.64.19.155

           Subnet Mask: /24
```

5.5. 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. To provide testing flexibility, network region 2 was associated with other components used specifically for the Verizon testing.

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 1** 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 (10.64.19.155), and that the gateway IP address is 10.64.19.81. These fields are not configured in this screen, but just display the current information for the Media Gateway.

```
change media-gateway 1                                          Page   1 of   2
                             MEDIA GATEWAY 1

                    Type: g450
                    Name: G450-1
               Serial No: 08IS38199678
            Encrypt Link? y                   Enable CF? n
          Network Region: 1                    Location: 1
                                              Site Data:
           Recovery Rule: 1

              Registered?  y
   FW Version/HW Vintage: 33 .13 .0  /1
       MGP IPV4 Address: 10.64.19.81
       MGP IPV6 Address:
   Controller IP Address: 10.64.19.155
             MAC Address: 00:1b:4f:03:52:18
```

The following screen shows **Page 2** for **Media Gateway 1**. The gateway has an **S8300** in slot V1 (unused), an **MM712** media module supporting Avaya digital phones in slot V2, an **MM711** supporting analog devices in slot V3, and the capability to provide announcements and music on hold via "gateway-announcements" in logical slot V9.

```
change media-gateway 1                                          Page   2 of   2
                             MEDIA GATEWAY 1

                                 Type: g450

Slot   Module Type             Name                    DSP Type  FW/HW version
 V1:   S8300                   ICC MM                  MP80      110  3
 V2:   MM712                   DCP MM
 V3:   MM711                   ANA MM
 V4:
 V5:
 V6:
 V7:
 V8:                                                 Max Survivable IP Ext: 8
 V9:   gateway-announcements   ANN VMM
```

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 10.64.19.109 would be mapped to network region 1, based on the configuration in bold 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: 10.64.19.100                            /      1      n
   TO: 10.64.19.120
```

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 2 will be used for calls within region 2. The shared Avaya Interoperability Lab test environment uses the domain "avayalab.com" (i.e., for network region 1 including the region of the Processor Ethernet "procr"). Session Manager also uses this domain to determined routes for calls based on the domain information of the calls and for SIP phone registration. Avaya SBCE will adapt "avayalab.com" to "adevc.avaya.globalipcom.com" for the From, PAI and Diversion headers.

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

The following screen shows the inter-network region connection configuration for region 2. The first bold row shows that network region 2 is directly connected to network region 1, and that codec set 2 will also be used for any connections between region 2 and region 1. For configurations where multiple remote gateways are used, each gateway will typically be configured for a different region, and this screen can be used to specify unique codec or call admission control parameters for the pairs of regions. If a different codec should be used for inter-region connectivity than for intra-region connectivity, a different codec set can be entered in the **codec set** column for the appropriate row in the screen shown below. Once submitted, the configuration becomes symmetric, meaning that network region 1, **Page 4** will also show codec set 2 for region 2 to region 1 connectivity.

```
change ip-network-region 2                                      Page   4 of  20

 Source Region: 2     Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
 1    2     y    NoLimit                                             n       t
 2    2                                                                all
 3
 4
```

The following screen shows IP Network Region 1 configuration. In this example, codec set 1 will be used for calls within region 1 due to the **Codec Set** parameter on **Page 1**, but codec set 2 will be used for connections between region 1 and region 2 as noted previously.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avayalab.com
    Name: Enterprise             Stub Network Region: n
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
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                      RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

The following screen shows the inter-network region connection configuration for region 1. The bold row shows that network region 1 is directly connected to network region 2, and that codec set 2 will be used for any connections between region 2 and region 1.

```
change ip-network-region 1                                      Page   4 of  20

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

5.6. IP Codec Sets

The following screen shows the configuration for codec set 2, the codec set configured to be used for calls within region 2 and for calls between region 1 and region 2. In general, an IP codec set is a list of allowable codecs in priority order. Using the example configuration shown below, calls to and from the PSTN via the SIP trunks between devices capable of the G.722-64K codec (e.g., Avaya 9600-Series IP Telephone) can use G.722. While all other calls would use G.729A, since G.729A is the next preferred codec by both Verizon and the Avaya ip-codec-set. Include G.711MU in the ip-codec-set if fax will be used.

```
change ip-codec-set 2                                           Page   1 of   2
                          IP Codec Set
   Codec Set: 2

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

The following screen shows **Page 2** of the form. Configure the **Fax Mode** field to “off” and set the Fax **Redundancy** field to “0”. See **Section 2.2** for more details regarding fax and the recommendation to use an AudioCodes MP-1xx for fax.

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n

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

The following screen shows the configuration for codec set 1. This configuration for codec set 1 is used for analog, digital, H.323, SIP phones and other connections within region 1.

```
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:
 4:
```

5.7. SIP Signaling Group

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 “SM63”. In the example screens, the **Transport Method** for all signaling groups is “tls”. The **Peer Detection Enabled** field is set to “y” and a peer Session Manager has been previously detected. The **Far-end Domain** is set to “avayalab.com” matching the configuration in place prior to adding the Verizon IP SIP Trunking configuration. 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 2. Port 5081 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 5081. The use of different ports is one means to allow Communication Manager to distinguish different types of calls arriving from the same Session Manager. The **Initial IP-IP Direct Media?** is set to “n”. Other parameters may be left at default values.

The **Alternate Route Timer** that defaults to 6 seconds impacts fail-over timing for outbound calls. If Communication Manager does not get an expected response, Look-Ahead Routing (LAR) can be triggered, after the expiration of the Alternate Route Timer.

```
change signaling-group 1                                        Page   1 of   2
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                  Far-end Node Name: SM63
 Near-end Listen Port: 5081                 Far-end Listen Port: 5081
                                         Far-end Network Region: 2

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                    Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 6
```

The following screen shows signaling group 3, the signaling group to Session Manager that was in place prior to adding the Verizon IP Trunk configuration to the shared Avaya Solutions and Interoperability Test Lab configuration. This signaling group reflects configuration not specifically related to Verizon IP Trunk but will be used to enable SIP phones to register to Session Manager and to use features from Communication Manager. Again, the **Near-end Node Name** is "procr" and the **Far-end Node Name** is "SM63", the node name of the Session Manager. Unlike the signaling group used for the Verizon IP Trunk signaling, the **Far-end Network Region** is "1". The **Peer Detection Enabled** field is set to "y" and a peer Session Manager has been previously detected.

```
change signaling-group 3                                        Page   1 of   2
                                SIGNALING GROUP

 Group Number: 3              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                  Far-end Node Name: SM63
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1

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                    Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 6
```

5.8. SIP Trunk Group

This section illustrates the configuration of the SIP Trunk Groups 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 is 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: OUTSIDE CALL              COR: 1        TN: 1        TAC: *01
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 1
                                                     Number of Members: 10
```

The following screen shows **Page 2** for trunk group 1. All parameters shown are default values, except for the **Preferred Minimum Session Refresh Interval**, which has been changed from the default 600 to 900. Although not strictly necessary, some SIP products prefer a higher session refresh interval than the Communication Manager default value, which can result in unnecessary SIP messages to re-establish a higher refresh interval for each call.

```
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): 900

 Disconnect Supervision - In? y  Out? y


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

The following screen 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.

```
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? n
                                                Replace Unavailable Numbers? n

                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? y
```

The following screen shows **Page 4** for trunk group 1. The bold fields have non-default values. The **Convert 180 to 183 for Early Media** field was new in Communication Manager Release 6. Verizon recommends that inbound calls to the enterprise result in a 183 with SDP rather than a 180 with SDP, and setting this field to “y” for the trunk group handling inbound calls from Verizon produces this result. 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 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/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? y
          Build Refer-To URI of REFER From Contact For NCR? n
                                     Send Diversion Header? n
                                   Support Request History? y
                              Telephone Event Payload Type: 101


                        Convert 180 to 183 for Early Media? y
                  Always Use re-INVITE for Display Updates? n
                        Identity for Calling Party Display: P-Asserted-Identity
            Block Sending Calling Party Location in INVITE? n
                 Accept Redirect to Blank User Destination? n
                                              Enable Q-SIP? n
```

The following screen shows **Page 1** for trunk group 3, the bi-directional “tie” trunk group to Session Manager that existed before adding the Verizon SIP Trunk configuration to the shared Avaya Interoperability Lab network. Recall that this trunk is used to enable SIP phones to use features from Communication Manager and to communicate with other Avaya applications, such as Avaya Aura® Messaging, and does not reflect any unique Verizon configuration.

```
change trunk-group 3                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 3                    Group Type: sip          CDR Reports: y
  Group Name: To SM Enterprise           COR: 1        TN: 1        TAC: *03
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 3
                                                     Number of Members: 20
```

The following shows **Page 3** for trunk group 3. Note that this tie trunk group uses a “private” **Numbering Format**.

```
change trunk-group 3                                              Page   3 of  21
                                TRUNK FEATURES
          ACA Assignment? n           Measured: none
                                                          Maintenance Tests? y
                     Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                Modify Tandem Calling Number: no
```

The following screen shows **Page 4** for trunk group 3. Note that unlike the trunks associated with Verizon calls that have non-default “protocol variations”, this trunk group maintains all default values. **Support Request History** must remain set to the default “y” to support proper subscriber mailbox identification by Communication Manager Messaging.

```
change trunk-group 3                                               Page   4 of  21
                              PROTOCOL VARIATIONS

                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? n

                                     Send Diversion Header? n
                                   Support Request History? y
                              Telephone Event Payload Type: 120


                        Convert 180 to 183 for Early Media? y
                  Always Use re-INVITE for Display Updates? n
                        Identity for Calling Party Display: P-Asserted-Identity
            Block Sending Calling Party Location in INVITE? n
                 Accept Redirect to Blank User Destination? n
                                              Enable Q-SIP? n
```

5.9. 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. Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of “0” is the least restrictive level. The **Numbering Format** “unk-unk” means no special numbering format will be included.

If desired, one or more alternate Communication Manager trunks can be listed in the route pattern so that the Look-Ahead Routing (**LAR**) “next” setting can route-advance to attempt to complete the call using alternate trunks should there be no response or an error response from the far-end.

```
change route-pattern 1                                         Page   1 of   3
                    Pattern Number: 1      Pattern Name: To PSTN SIP Trk
                             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       1                                                     n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                unk-unk   next
 2: y y y y y n  n            rest                                          none
 3: y y y y y n  n            rest                                          none
 4: y y y y y n  n            rest                                          none
 5: y y y y y n  n            rest                                          none
 6: y y y y y n  n            rest                                          none
```

5.10. 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" insures proper numbering format for internal local calls to Session Manager.

```
change route-pattern 3                                          Page   1 of   3
                    Pattern Number: 3      Pattern Name: ToSM Enterprise
                             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 y  n            bothept                               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
 5: y y y y y n  n            rest                                            none
 6: y y y y y n  n            rest                                            none
```

5.11. 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 Communication Manager (via private-numbering form for outbound calls, and incoming call handling treatment form for the inbound trunk group).

In the example abridged output below, a specific Communication Manager extension (x12001) 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  10                                          5     Total Administered: 5
 5  12                                          5        Maximum Entries: 540
 5  14                                          5
 5  20                                          5
 5  12001           1          7329450231       10
```

5.12. 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.

The following screen shows a specific ARS configuration as an example. If a user dials the ARS access code followed by 13035387024, the call will select route pattern 1. Of course, matching of the dialed string need not be this specific. The ARS configuration shown here is not intended to be prescriptive.

```
change ars analysis 13035387022                               Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 1

          Dialed            Total     Route    Call   Node  ANI
          String           Min  Max  Pattern   Type   Num   Reqd
    13035387024            11   11    1       fnpa          n
```

The ***list ars route-chosen*** command can be used on a target dialed number to check whether routing will behave as intended. An example is shown below.

```
list ars route-chosen 13035387024
                           ARS ROUTE CHOSEN REPORT
     Location:  1                      Partitioned Group Number:  1

      Dialed             Total         Route    Call      Node
      String           Min    Max    Pattern   Type     Number    Location

 13035387024           11    11       1        fnpa                all
   Actual Outpulsed Digits by Preference (leading 35 of maximum 42 digit)

 1: 13035387024
```

5.13. Avaya Aura® Communication Manager Stations

In the sample configuration, five digit station extensions were used with the format 12xxx, and 14xxx. The following abbreviated screen shows an example extension for an Avaya H.323 IP telephone.

```
change station 12002                                          Page   1 of   5
                                     STATION

Extension: 12002                        Lock Messages? n              BCC: 0
     Type: 9621                         Security Code: *               TN: 1
     Port: S00025                     Coverage Path 1:                COR: 1
     Name: test IP                    Coverage Path 2:                COS: 1
                                      Hunt-to Station:             Tests? y
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 19        Personalized Ringing Pattern: 1
                                                Message Lamp Ext: 12002
            Speakerphone: 2-way             Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
```

5.14. 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 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 12005. Use the command ***change off-pbx-telephone station mapping x*** where ***x*** is Communication Manager station (e.g. 12002).

- **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., 3035387024)
- **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 12002                     Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 12002            EC500        - 1   3035387024       ars         1
```

5.15. Saving Communication Manager Configuration Changes

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

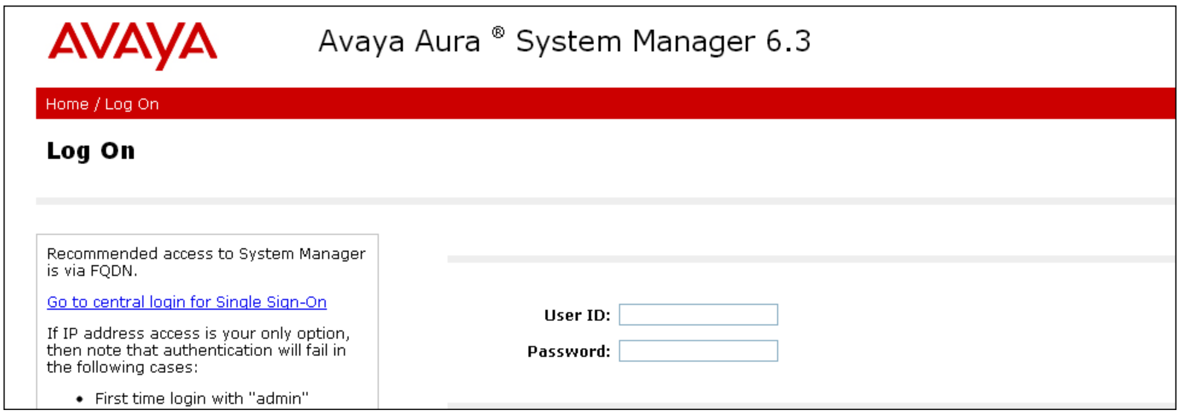
```
save translation all

                             SAVE TRANSLATION

          Command Completion Status                                    Error Code

          Success                                                      0
```

6. Configure Avaya Aura® Session Manager Release 6.3

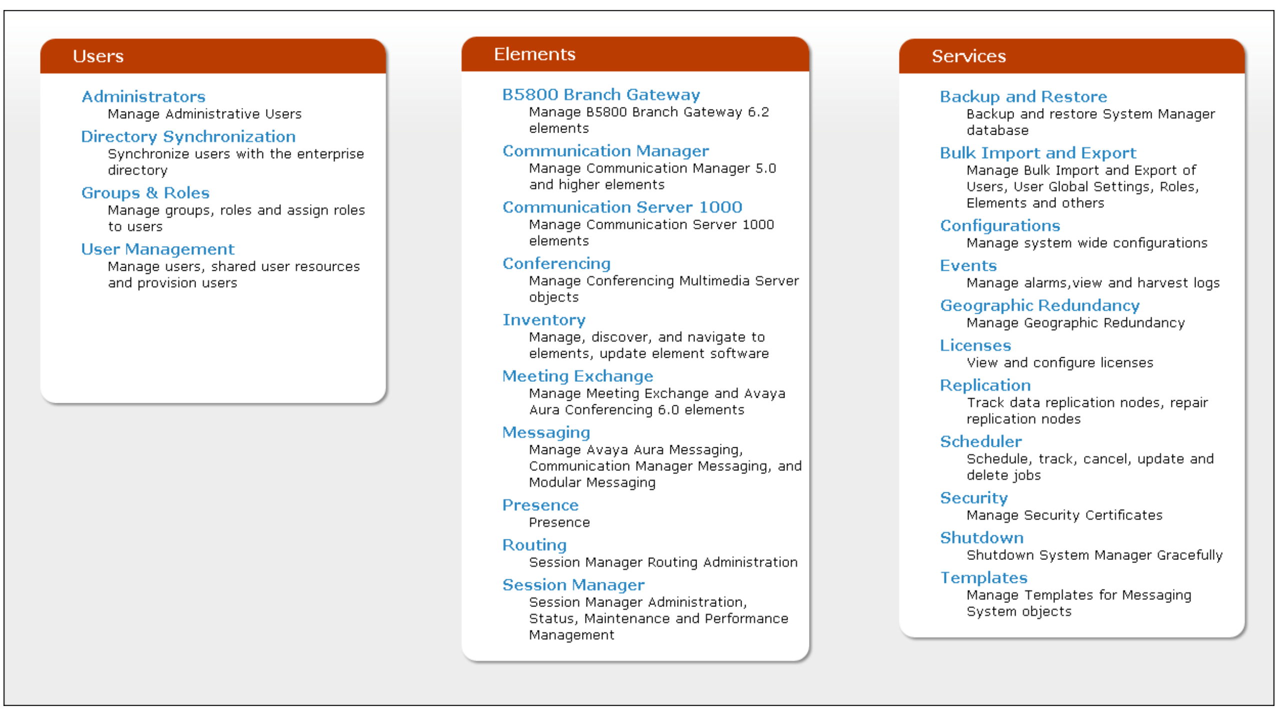
This section illustrates relevant aspects of the Session Manager configuration used in the verification of these Application Notes.

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 (not shown).



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 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)

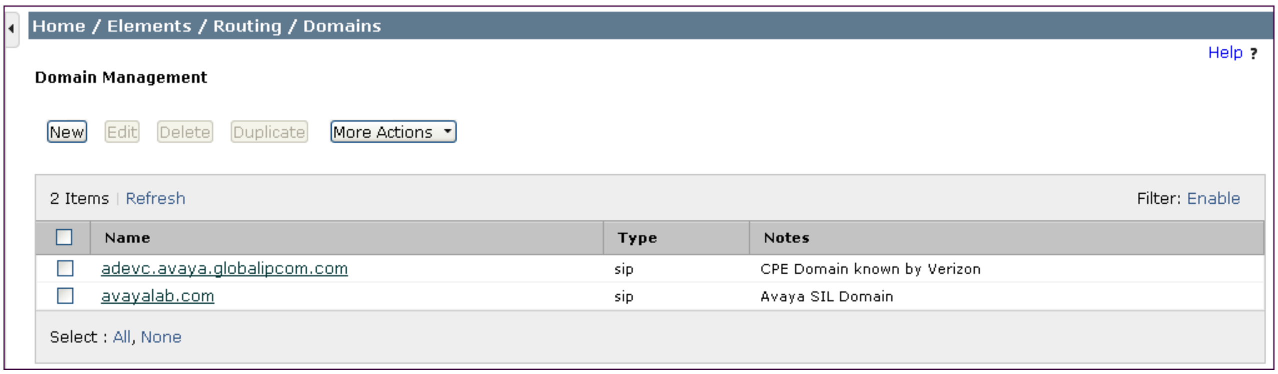
Step 9: "Regular Expressions" are defined and assigned to "Routing Policies" (one step)

6.1. 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.

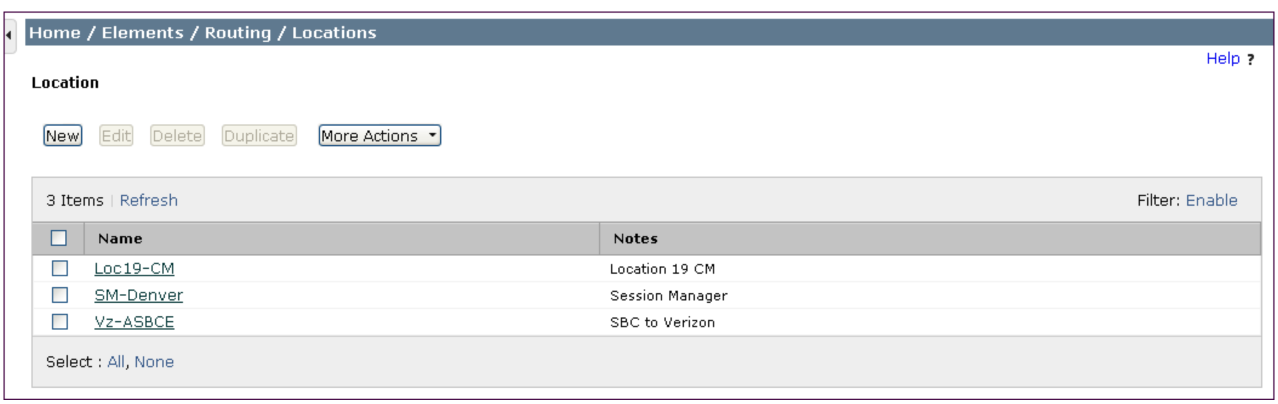
The following screen shows a list of configured SIP domains. The Session Manager used in the verification of these Application Notes was shared among other Avaya interoperability test efforts. The domain "avayalab.com" was used for communication with Avaya SIP Telephones and other Avaya systems and applications. 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. For example, for calls from the enterprise site to Verizon, this domain can appear in the From and P-Asserted-Identity headers in the INVITE message sent to Verizon.



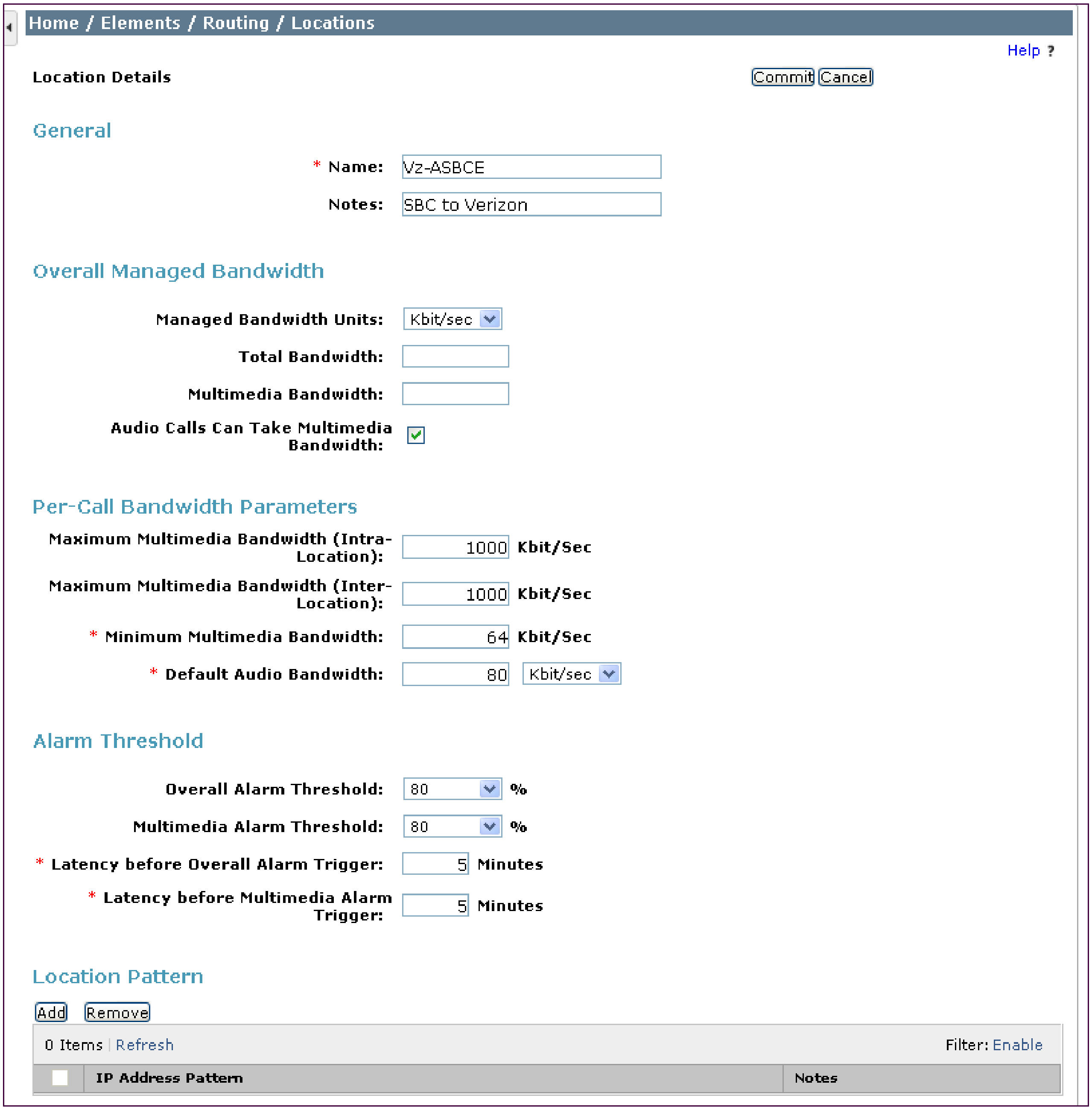
6.2. Locations

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 (not shown) after changes are completed. Assigning unique locations can allow Session Manager to perform location-based routing, bandwidth management, and call admission control.

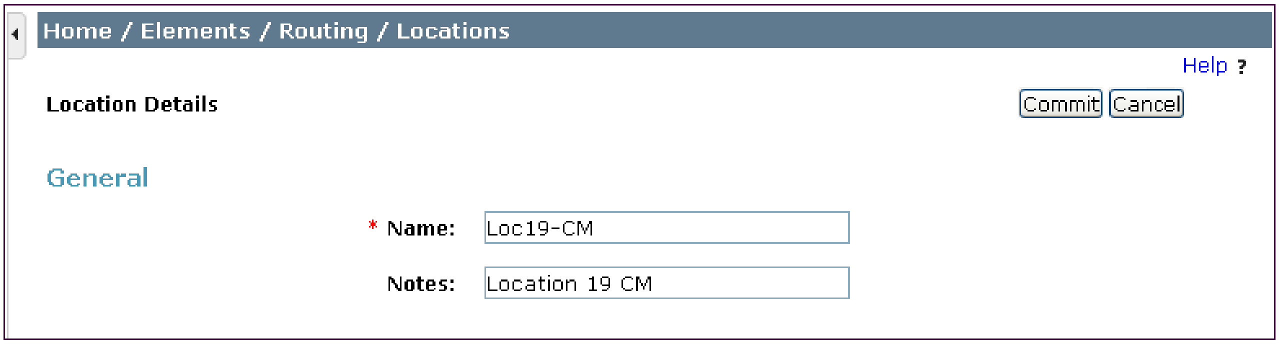


The following screen shows the location details for the location named "Vz-ASBCE", corresponding to the Avaya SBCE relevant to these Application Notes. Later, the location with name "Vz-ASBCE" will be assigned to the corresponding Avaya SBCE SIP Entity.

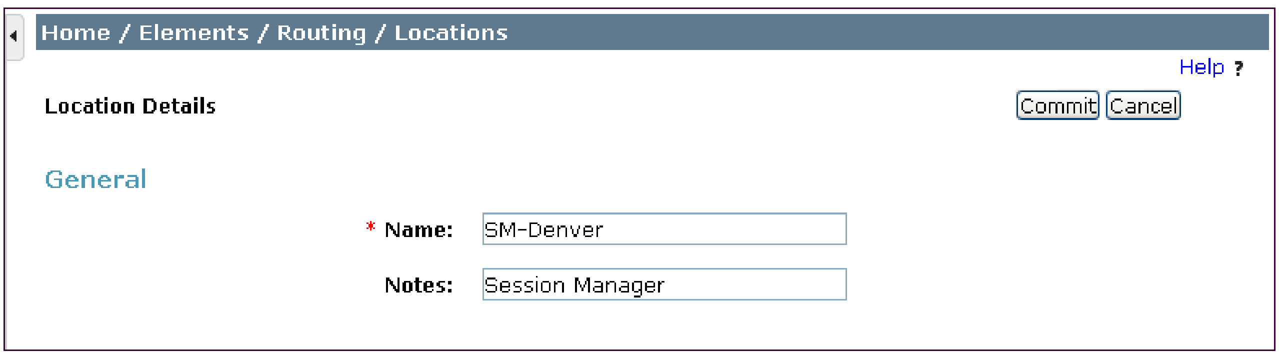
The **Location Pattern** is used to identify call routing based on IP address. Session Manager matches the IP address of SIP Entities against the patterns defined in this section. If a call is from a SIP Entity that does not match the IP address pattern then Session Manager uses the Location administered in the SIP Entity form. In this sample configuration Locations are added to SIP Entities in **Section 6.4**, so it is not necessary to add a pattern.



The following screen shows the location details for the location named “Loc19-CM”, corresponding to Communication Manager. Later, the location with name “Loc19-CM” will be assigned to the corresponding Communication Manager SIP Entity. In the sample configuration, other location parameters (not shown) retained the default values.

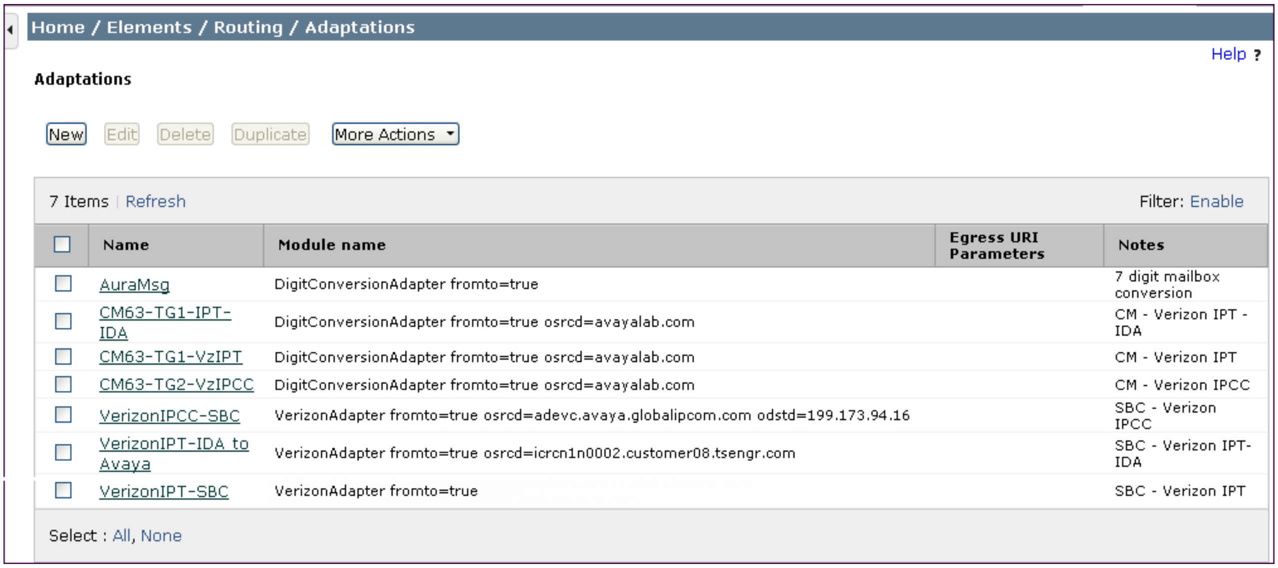


The following screen shows the location details for the location named “SM-Denver”, corresponding to Session Manager. This location was created during the installation of Session Manager and was assigned to the Session Manager SIP Entity. In the sample configuration, other location parameters (not shown) retained the default values.



6.3. Adaptations

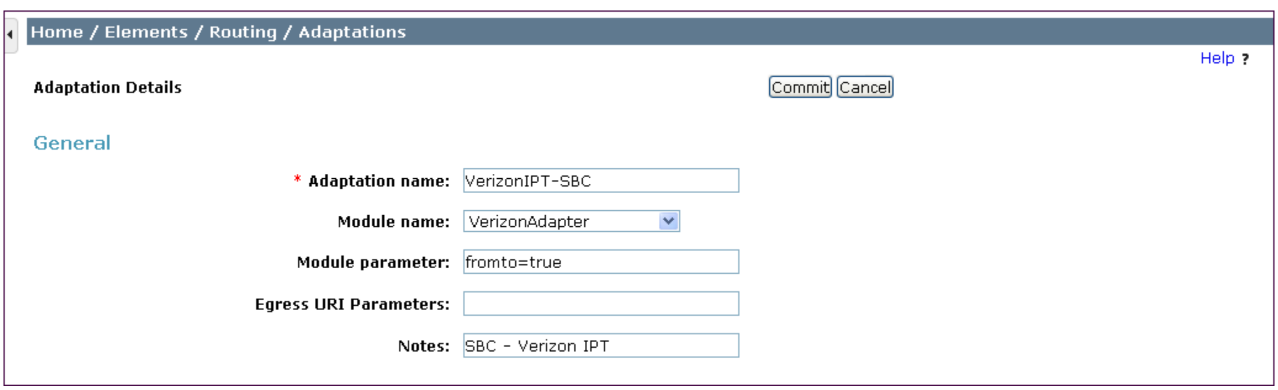
To view or change adaptations, select **Routing → Adaptations**. Click on the checkbox corresponding to the name of an adaptation and **Edit** to edit an existing adaptation, or the **New** button to add an adaptation. Click the **Commit** button after changes are completed (not shown).



Name	Module name	Egress URI Parameters	Notes
AuraMsg	DigitConversionAdapter fromto=true		7 digit mailbox conversion
CM63-TG1-IPT-IDA	DigitConversionAdapter fromto=true osrcd=avayalab.com		CM - Verizon IPT - IDA
CM63-TG1-VzIPT	DigitConversionAdapter fromto=true osrcd=avayalab.com		CM - Verizon IPT
CM63-TG2-VzIPCC	DigitConversionAdapter fromto=true osrcd=avayalab.com		CM - Verizon IPCC
VerizonIPCC-SBC	VerizonAdapter fromto=true osrcd=adevc.avaya.globalipcom.com odstd=199.173.94.16		SBC - Verizon IPCC
VerizonIPT-IDA to Avaya	VerizonAdapter fromto=true osrcd=icrcn1n0002.customer08.tsengr.com		SBC - Verizon IPT-IDA
VerizonIPT-SBC	VerizonAdapter fromto=true		SBC - Verizon IPT

The adapter named "VerizonIPT-SBC" shown below will later be assigned to the SIP Entity for the Avaya SBCE, specifying that all communication from Session Manager to the Avaya SBCEs will use this adapter.

This adaptation uses the "VerizonAdapter" module and specifies the "fromto=true" parameter. This parameter adapts the From and To headers along with the Request-Line and PAI headers.



Scrolling down to the **Digit Conversion for Incoming Calls to SM** section, the following screen shows the addition of the 10 digit DID numbers assigned by Verizon intended for fax calls converted to the extension numbers used by the AudioCodes gateway.

Digit Conversion for Incoming Calls to SM

Add Remove

1 Item | Refresh Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
☐	* 7329450231	* 10	* 10		* 10	17555	both		

Select : All, None

Scrolling down to the **Digit Conversion for Outgoing Calls from SM** section, the following screen shows an example configuration for Verizon's Unscreened ANI feature. This optional configuration allows customers to send an "unscreened" ANI to Verizon's network which is then displayed to the called party as Caller ID. An "unscreened" ANI can be any telephone number that the customer passes through Verizon's network for Caller ID display purposes only. If this feature is enabled on the Verizon IP Trunk services, Verizon will designate one of the assigned telephone numbers as a "Screened Telephone Number" for each unique location. Verizon will use this Screened Telephone Number to determine call origination for billing, call routing, and E911.

The Screened Telephone Number (STN) provided by Verizon for this test is 732-945-0821. Typically, customers would have one or more STN; one for every location. A central Session Manager could be used to pass multiple STNs to Verizon based on a **Matching Pattern** (i.e., a user's Calling Line Identification). The STN would then be entered in the **Adaptation Data** field as shown below.

Digit Conversion for Outgoing Calls from SM

Add Remove

1 Item | Refresh Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
☐	* 408990883x	* 10	* 10		* 0		origination	7329450821	Unscreened ANI - Diversion

Select : All, None

The following screen shows the addition of extension numbers used on Communication Manager that are being converted to the 10 digit DID numbers assigned by Verizon. Since this adapter will be assigned to the SIP Entity sending calls to Avaya SBCE for routing to the PSTN, the settings for **Digit Conversion for Outgoing Calls from SM** correspond with outgoing calls from Communication Manager to the PSTN using the Verizon IP Trunk service. In general, digit conversion such as this, that converts a Communication Manager extension (e.g., 12xxx) to a corresponding LDN or DID number known to the PSTN (e.g., 73294502xx), can be performed in Session Manager as shown below. For example, if extension 12001 dials the PSTN, and if Communication Manager sends the extension 12001 to Session manager as the calling number, Session Manager would convert the calling number to 7329450231.

Digit Conversion for Outgoing Calls from SM

Add Remove

6 Items | Refresh Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
☐	* 10000	* 5	* 5		* 5	7329450243	origination		
☐	* 12000	* 5	* 5		* 5	7329450241	origination		
☐	* 12001	* 5	* 5		* 5	7329450231	origination		
☐	* 12002	* 5	* 5		* 5	7329450232	origination		
☐	* 12003	* 5	* 5		* 5	7329450233	origination		
☐	* 408990883x	* 10	* 10		* 10		origination	7329450821	Unscreened ANI - Diversic

Select : All, None

The adapter named "CM63-TG1-VzIPT" shown in the following screen will later be assigned to the SIP Entity linking Session Manager to Communication Manager for calls involving Verizon Business IP Trunk service. This adaptation uses the "DigitConversionAdapter" and specifies the following parameters:

- "fromto=true". This adapts the From and To headers along with the Request-Line and PAI headers.
- "osrcd=avayalab.com". This enables the source domain to be overwritten with "avayalab.com". For example, for inbound PSTN calls from Verizon to Communication Manager, the PAI header will contain "avayalab.com".

Depending on the Communication Manager configuration, it may not be necessary for Session Manager to adapt the domain in this fashion.

Home / Elements / Routing / Adaptations

Help ?

Adaptation Details Commit Cancel

General

* Adaptation name: CM63-TG1-VzIPT

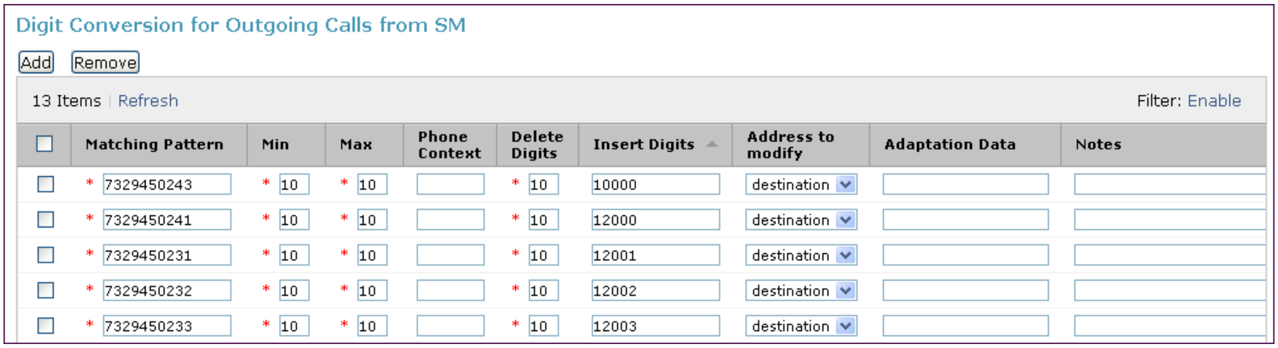
Module name: DigitConversionAdapter

Module parameter: fromto=true osrcd=avayalab.com

Egress URI Parameters:

Notes: CM - Verizon IPT

Scrolling down, the following screen shows a portion of the "CM63-TG1-VzIPT" adapter that can be used to convert 10 digit DID numbers assigned by Verizon to the extension number used on Communication Manager. Since this adapter will be assigned to the SIP Entity sending calls to Communication Manager from the PSTN, the settings for **Digit Conversion for Outgoing Calls from SM** correspond to incoming calls from the PSTN to Communication Manager. In the example shown below, if a user on the PSTN dials 732-945-0231, Session Manager will convert the number to 12001 before sending the SIP INVITE to Communication Manager. In this case, digit conversion is done after the routing decision has been made based upon the user part of the SIP URI. As such, it would not be necessary to use the incoming call handling table of the receiving Communication Manager trunk group to convert the DID number to its corresponding extension.


Digit Conversion for Outgoing Calls from SM

Add Remove

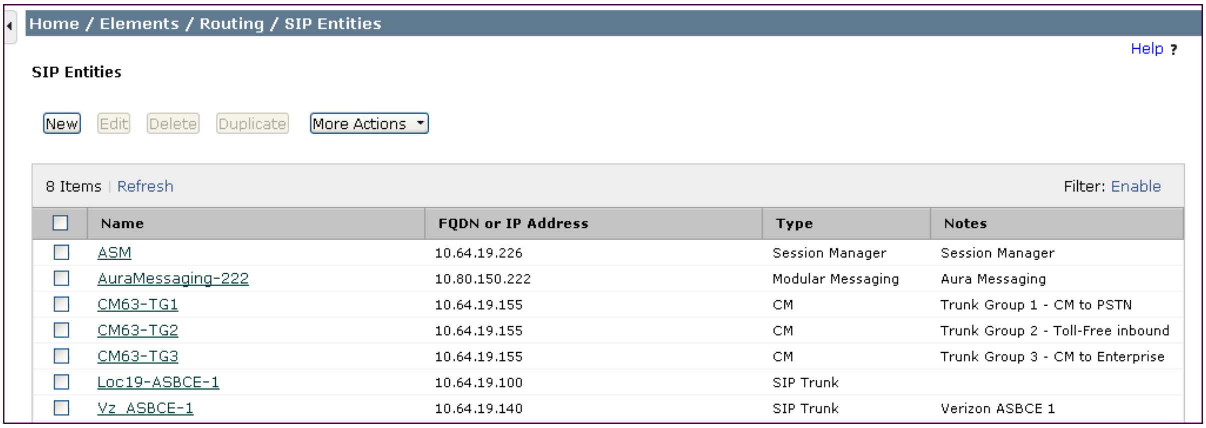
13 Items | Refresh — Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
	* 7329450243	* 10	* 10		* 10	10000	destination		
	* 7329450241	* 10	* 10		* 10	12000	destination		
	* 7329450231	* 10	* 10		* 10	12001	destination		
	* 7329450232	* 10	* 10		* 10	12002	destination		
	* 7329450233	* 10	* 10		* 10	12003	destination		

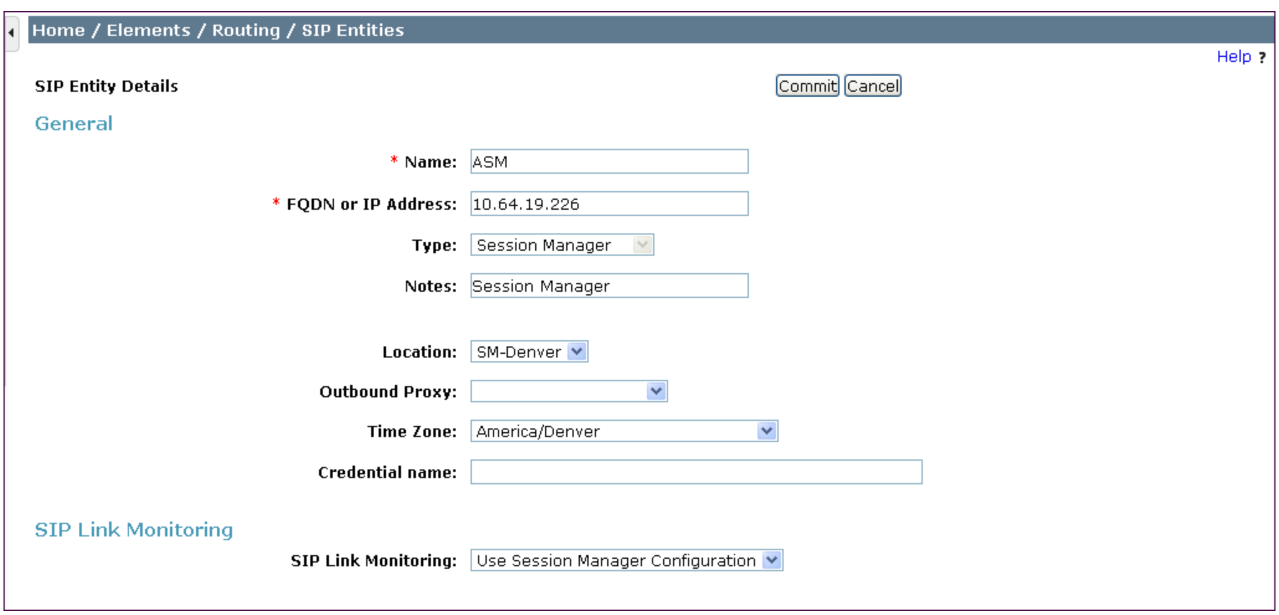
6.4. SIP Entities

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 the list of configured SIP entities in the shared test environment.



The following screen shows the upper portion of the **SIP Entity Details** corresponding to “ASM”. The **FQDN or IP Address** field for “ASM” is the Session Manager Security Module IP Address (10.64.19.226), 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 “SM-Denver”. 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". The links relevant to these Application Notes are described in the subsequent section.

Entity Links

Add Remove

7 Items | Refresh Filter: Enable

	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
☐	ASM	TLS	* 5061	AuraMessaging-222	* 5061	trusted	☐
☐	ASM	TLS	* 5081	CM63-TG1	* 5081	trusted	☐
☐	ASM	TLS	* 5071	CM63-TG2	* 5071	trusted	☐
☐	ASM	TLS	* 5061	CM63-TG3	* 5061	trusted	☐
☐	ASM	TCP	* 5060	Vz_ASBCE-1	* 5060	trusted	☐

Select : All, None < Previous | Page 1 of 2 | Next >

Scrolling down, the following screen shows the lower portion of the **SIP Entity Details**, illustrating the configured ports for "ASM". This section is only present for Session Manager SIP entities. This section defines a default set of ports that Session Manager will use to listen for SIP requests, typically from registered SIP endpoints. Session Manager can also listen on additional ports defined elsewhere such as the ports specified in the SIP Entity Link definition in **Section 6.5**.

Port

TCP Failover port:

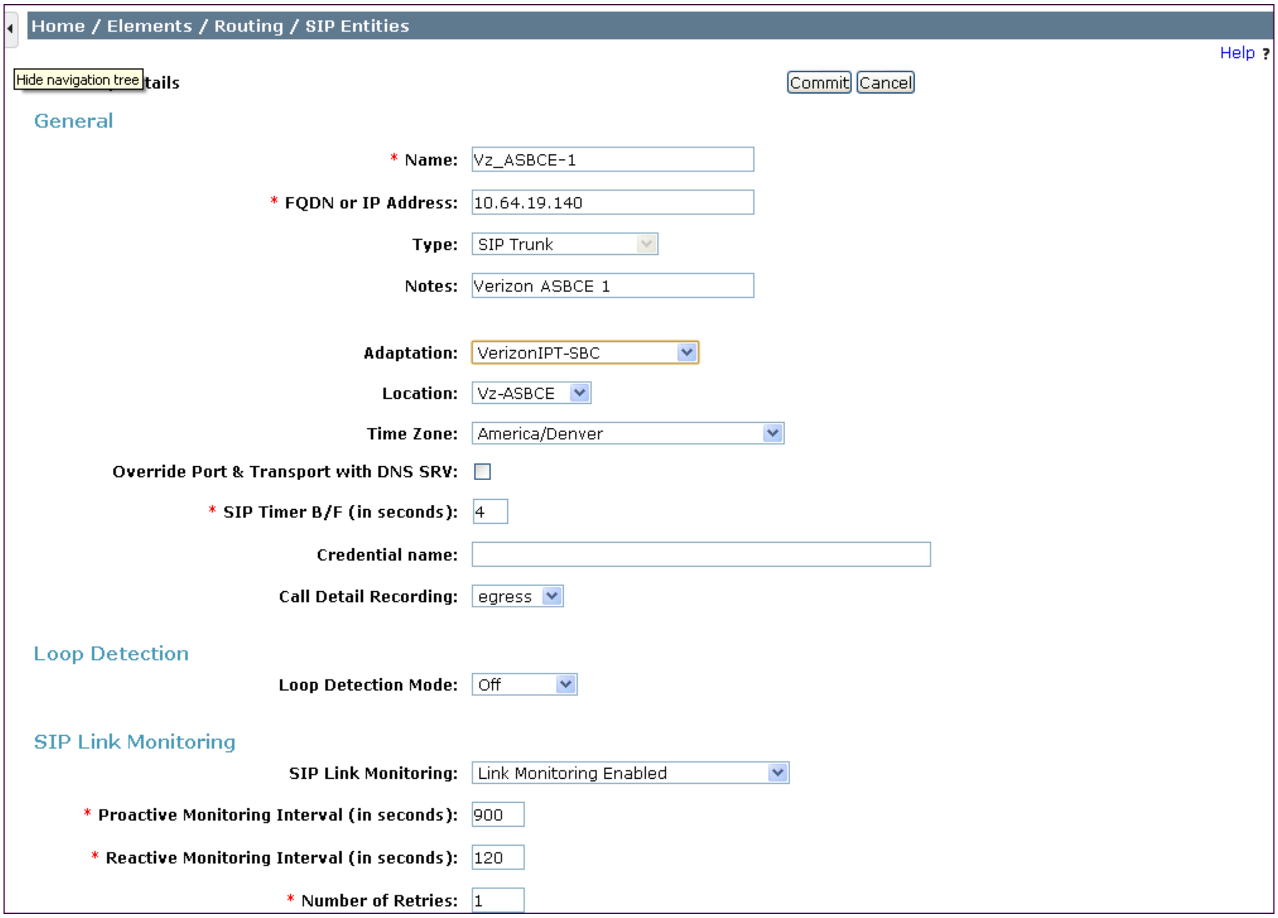
TLS Failover port:

Add Remove

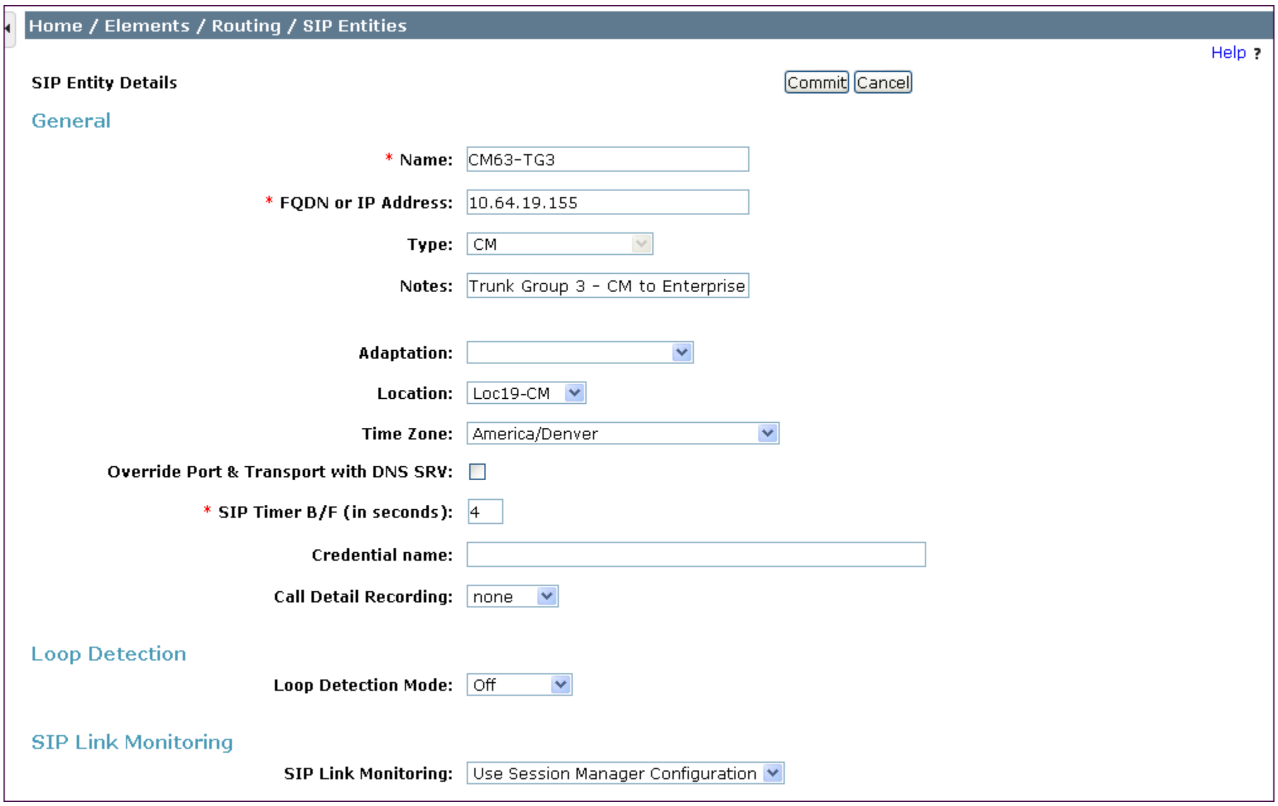
5 Items | Refresh Filter: Enable

	Port	Protocol	Default Domain	Notes
☐	5060	TCP	avayalab.com	
☐	5060	UDP	avayalab.com	
☐	5061	TLS	avayalab.com	
☐	5071	TLS	avayalab.com	
☐	5081	TLS	avayalab.com	

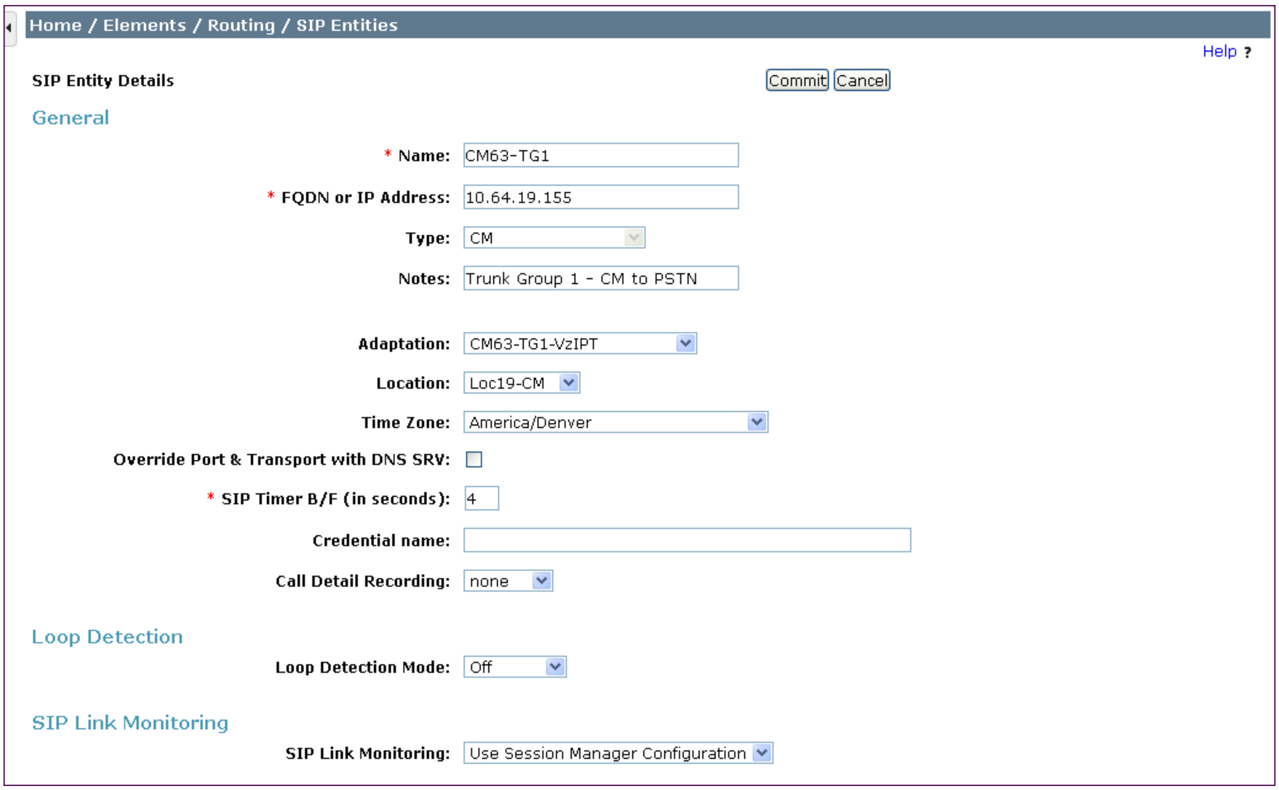
The following screen shows the upper portion of the **SIP Entity Details** corresponding to "Vz_ASBCE-1". The **FQDN or IP Address** field is configured with the Avaya SBCE inside IP Address (10.64.19.140). "SIP Trunk" is selected from the **Type** drop-down menu for Avaya SBCE SIP Entities. This Avaya SBCE has been assigned to **Location** "Vz-ASBCE", and the "VerizonIPT-SBC" 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 “CM63-TG3” This is the SIP Entity that was already in place in the shared Avaya Interoperability Test Lab environment, prior to adding the Verizon IP Trunk configuration. The **FQDN or IP Address** field contains the IP Address of the “processor Ethernet” (10.64.19.155). In systems with Avaya G650 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 and “Loc19-CM” is selected for the **Location**.



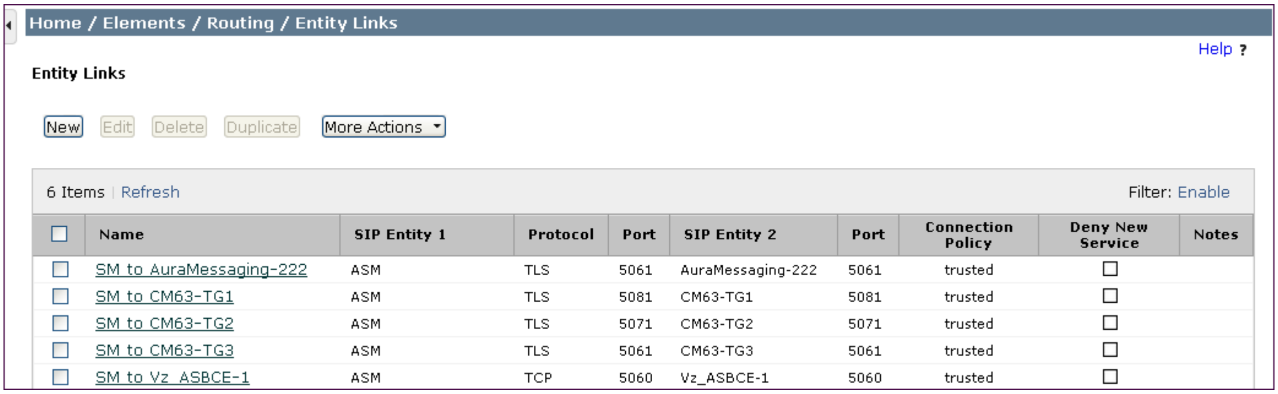
The following screen shows the **SIP Entity Details** for an entity named “CM63-TG1”. This entity uses the same **FQDN or IP Address** (10.64.19.155) as the prior entity with name “CM63-TG3”; both correspond to Communication Manager Processor Ethernet IP Address. Later, a unique port, 5081, will be used for the Entity Link to “CM63-TG1”. Using a different port is one approach that will allow Communication Manager to distinguish traffic originally from Verizon IP Trunk from other SIP traffic arriving from the same IP Address of the Session Manager, such as SIP traffic associated with SIP Telephones or other SIP-integrated applications. “CM” is selected from the **Type** drop-down menu, and “Loc19-CM” is selected for the **Location**.



6.5. 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.

The following screen shows a list of configured links. In the screen below, the links named "SM to Vz_ASBCE-1"and "SM to CM63-TG1" are most relevant to these Application Notes. Each link uses the entity named "ASM" as **SIP Entity 1**, and the appropriate entity, such as "Vz_ASBCE-1", for **SIP Entity 2**.


Home / Elements / Routing / Entity Links

Help ?

Entity Links

New Edit Delete Duplicate More Actions

6 Items | Refresh   Filter: Enable

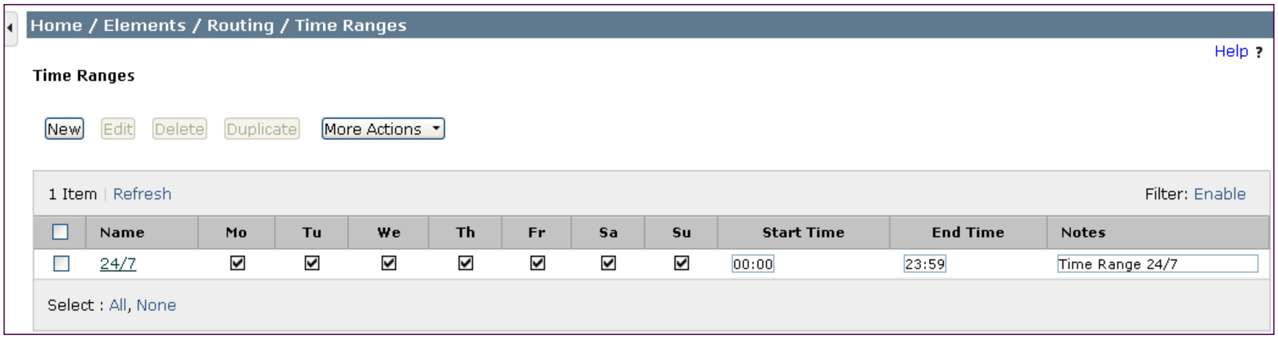
	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service	Notes
☐	SM to AuraMessaging-222	ASM	TLS	5061	AuraMessaging-222	5061	trusted	☐	
☐	SM to CM63-TG1	ASM	TLS	5081	CM63-TG1	5081	trusted	☐	
☐	SM to CM63-TG2	ASM	TLS	5071	CM63-TG2	5071	trusted	☐	
☐	SM to CM63-TG3	ASM	TLS	5061	CM63-TG3	5061	trusted	☐	
☐	SM to Vz_ASBCE-1	ASM	TCP	5060	Vz_ASBCE-1	5060	trusted	☐	

The link named "SM to CM63-TG3" links Session Manager "ASM" with Communication Manager processor Ethernet. This link existed in the configuration prior to adding the Verizon IP Trunk related configuration. This link, using port 5061, can carry traffic between Session Manager and Communication Manager that is not necessarily related to calls with Verizon, such as traffic related to SIP Telephones registered to Session Manager.

The link named "SM to CM63-TG1" also links Session Manager "ASM" with Communication Manager processor Ethernet. However, this link uses port 5081 for both entities in the link. This link was created to allow Communication Manager to distinguish calls from Verizon IP Trunk from other calls that arrive from the same Session Manager. Other methods of distinguishing traffic could be used, if desired.

6.6. Time Ranges

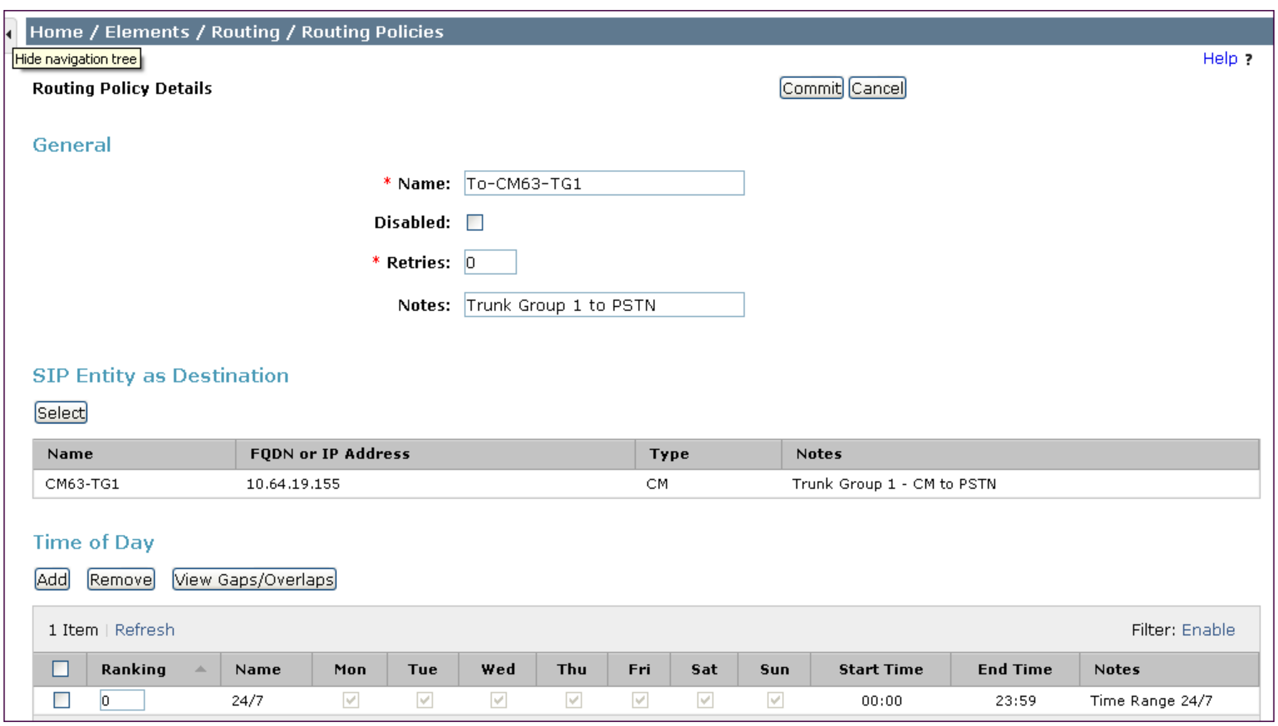
To view or change Time Ranges, select **Routing → Time Ranges**. The Routing Policies shown subsequently will use the "24/7" range since time-based routing was not the focus of these Application Notes. Click the **Commit** button (not shown) after changes are completed.



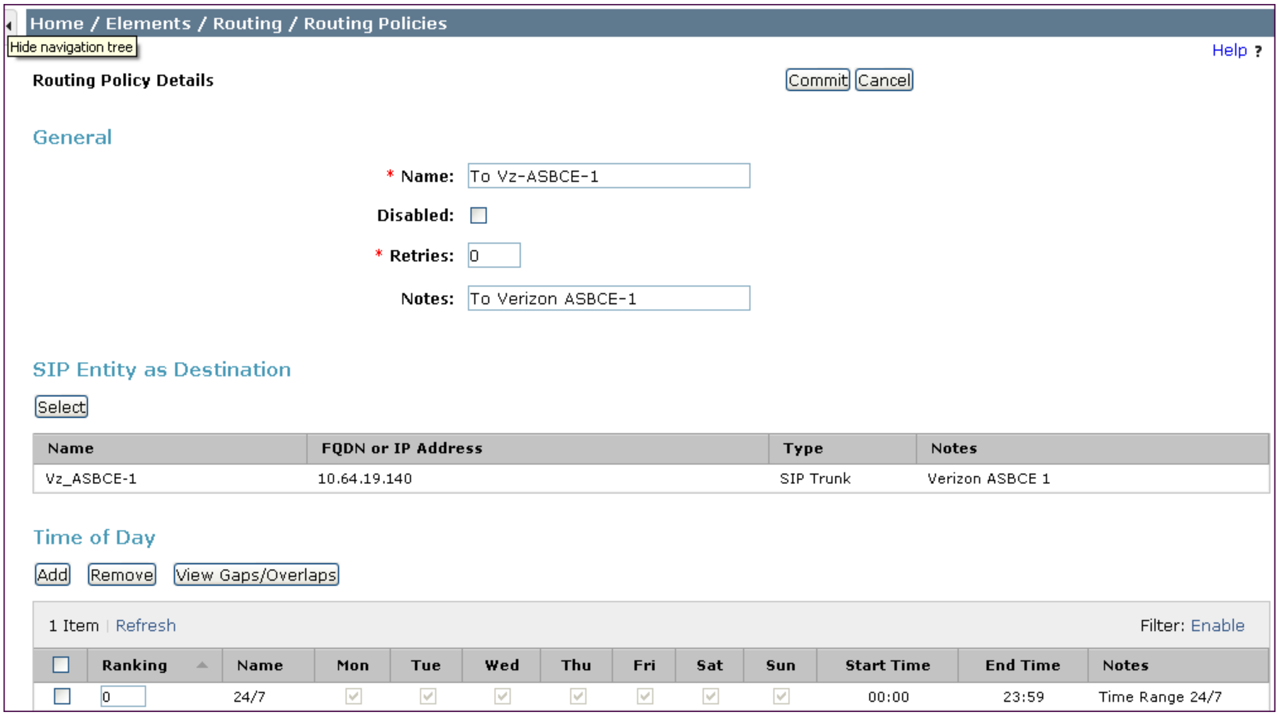
6.7. Routing Policies

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 (not shown).

The following screen shows the **Routing Policy Details** for the policy named "To-CM63-TG1" associated with incoming PSTN calls from Verizon to Communication Manager. Observe the **SIP Entity as Destination** is the entity named "CM63-TG1".



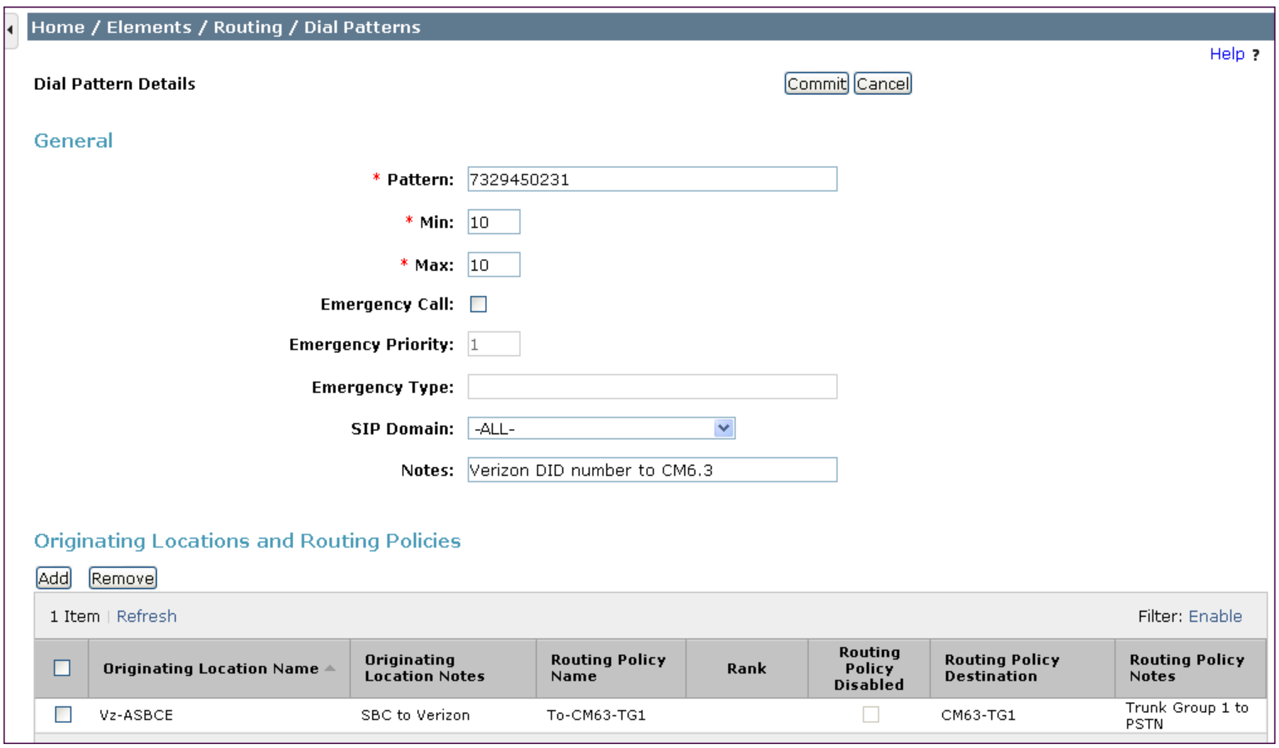
The following screen shows the **Routing Policy Details** for the policy named "To Vz-ASBCE-1" associated with outgoing calls from Communication Manager to the PSTN via Verizon through Avaya SBCE. Observe the **SIP Entity as Destination** as the entity named "Vz_ASBCE-1" that was created in **Section 6.4**.



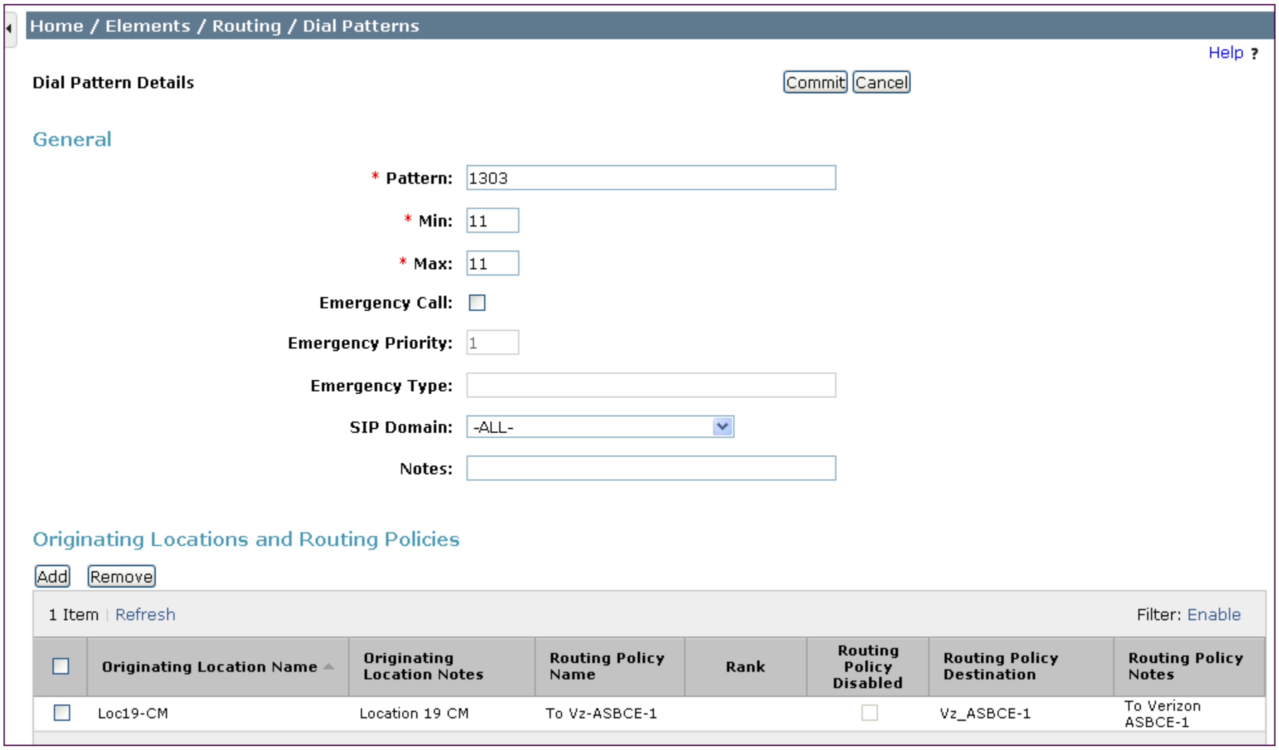
6.8. Dial Patterns

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.

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 732-945-0231, Verizon delivers the number to the enterprise, and the Avaya SBCE sends the call to Session Manager. The pattern below matches on 732-945-0231 specifically. Dial patterns can alternatively match on ranges of number (e.g., a DID block). Under **Originating Locations and Routing Policies**, the routing policy named "To-CM63-TG1" is chosen when the call originates from **Originating Location Name** "Vz-ASBCE". This sends the call to Communication Manager using port 5081 as described previously.



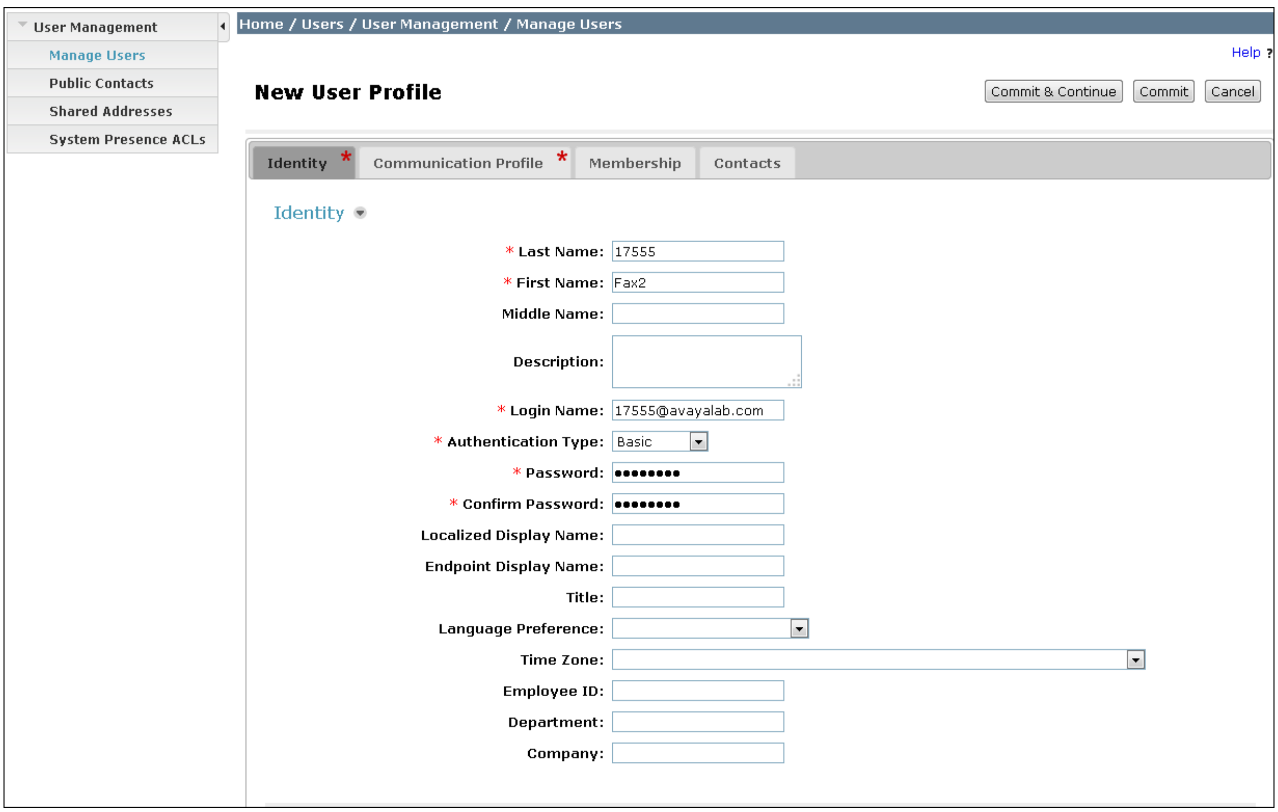
The following screen illustrates an example dial pattern used to verify outbound calls from the enterprise to the PSTN. When a Communication Manager user dials a PSTN number such as 9-1303-XXX-XXX, Communication Manager sends the call to Session Manager. Session Manager will match the dial pattern shown below and send the call to the Avaya SBCE via the **Routing Policy Name** "To Vz-ASBCE-1".



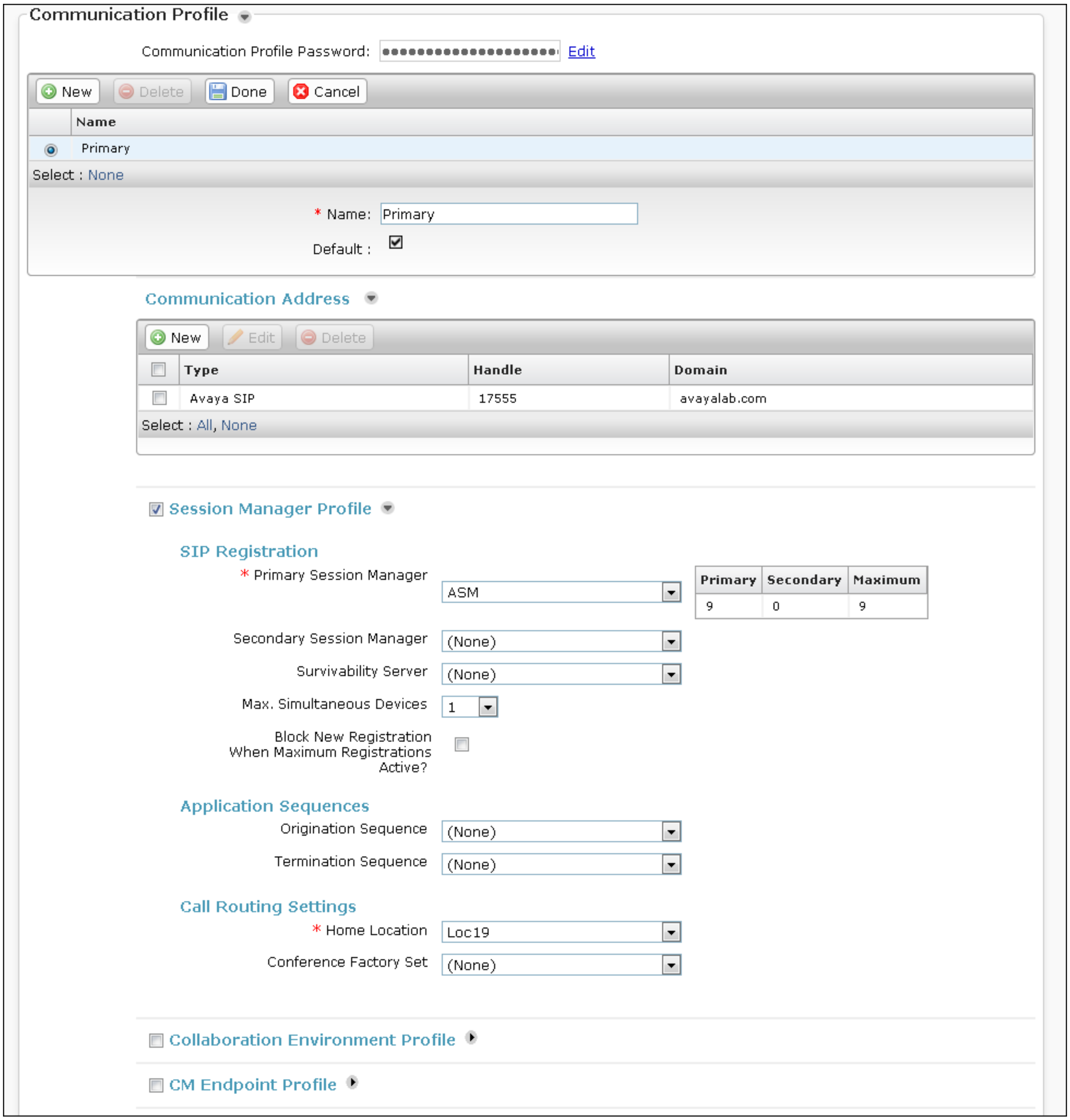
6.9. Fax Users

The following is an example SIP user created on System Manager to register an AudioCodes MP-114 port with Session Manager. On the Home screen, under the heading "Users", select **User Management**. On the left side, select **Manager Users** and click **New** as shown below.

The following screen shows the **Identity** tab of a sample SIP user created for fax calls.



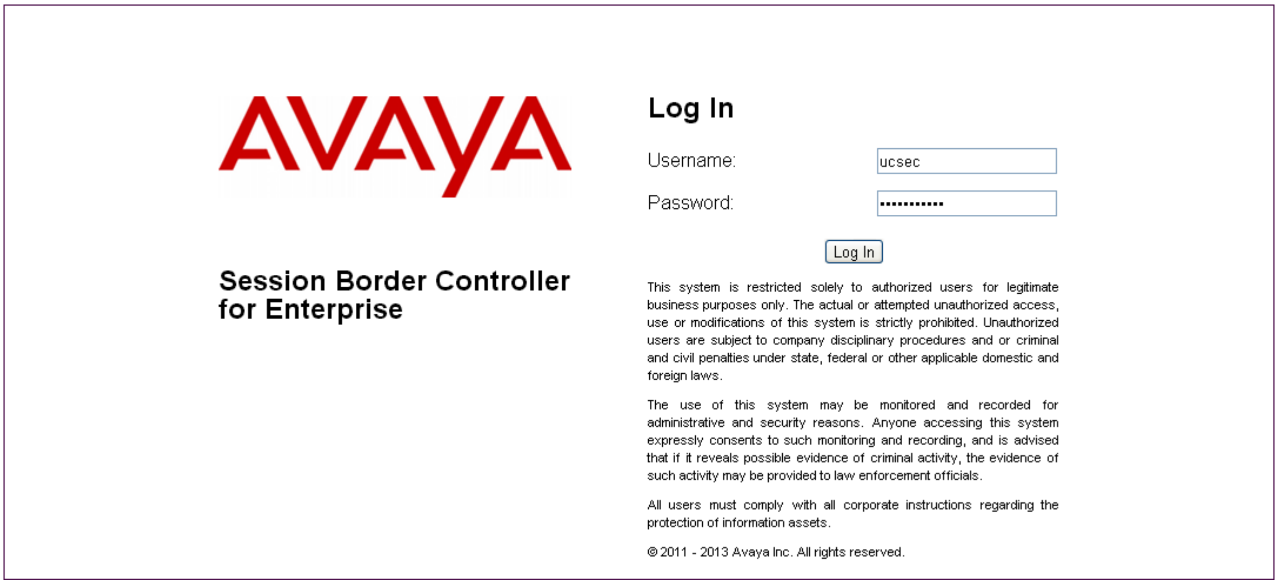
The following screen shows the **Communication Profile** tab of the sample user. The **Communication Profile Password** is the password used by the SIP device to register with Session Manager, and should match the password set on the AudioCodes MP-114 in **Section 8.2**. The **Application Sequences** section is set to "(None)", and the **CM Endpoint Profile** is unchecked. This allows for fax calls to be sent to the AudioCodes MP-114, without involving Communication Manager in the call setup. As stated in **Section 2.**2, Verizon requires fax calls to start off with G.711 as the first codec choice, and if all other voice calls prefer G.729 as the first codec, a separate Communication Manager trunk group dedicated for fax calls using an ip-codec-set with G.711 as the first codec choice would be required. Having the **Application Sequence** section set to "(None)" prevents the need for a separate fax dedicated trunk group on Communication Manager. As a result, fewer SIP re-Invites messages are sent during the beginning of a fax call, and voice calls to and from Communication Manager can use other preferred codecs. However, any functionality that would normally be controlled by Communication Manager, such as codec negotiation, calling restrictions, dial patterns, etc., will be controlled by the AudioCodes device, and therefore will need to be configured directly on the AudioCodes device. See **Section 8** and **Section 12.3** for information on AudioCodes MP-114 configuration.



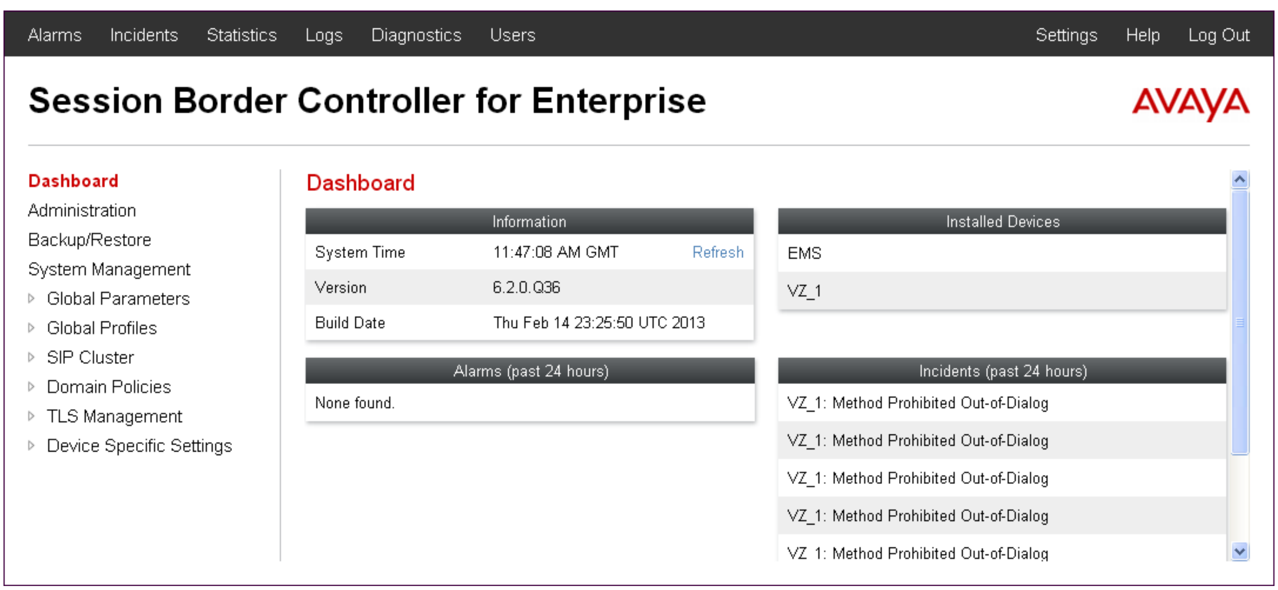
7. Configure Avaya Session Border Controller for Enterprise Release 6.2

These Application Notes assume that the installation of the Avaya SBCE and the assignment of all IP addresses have already been completed, including the management IP address.

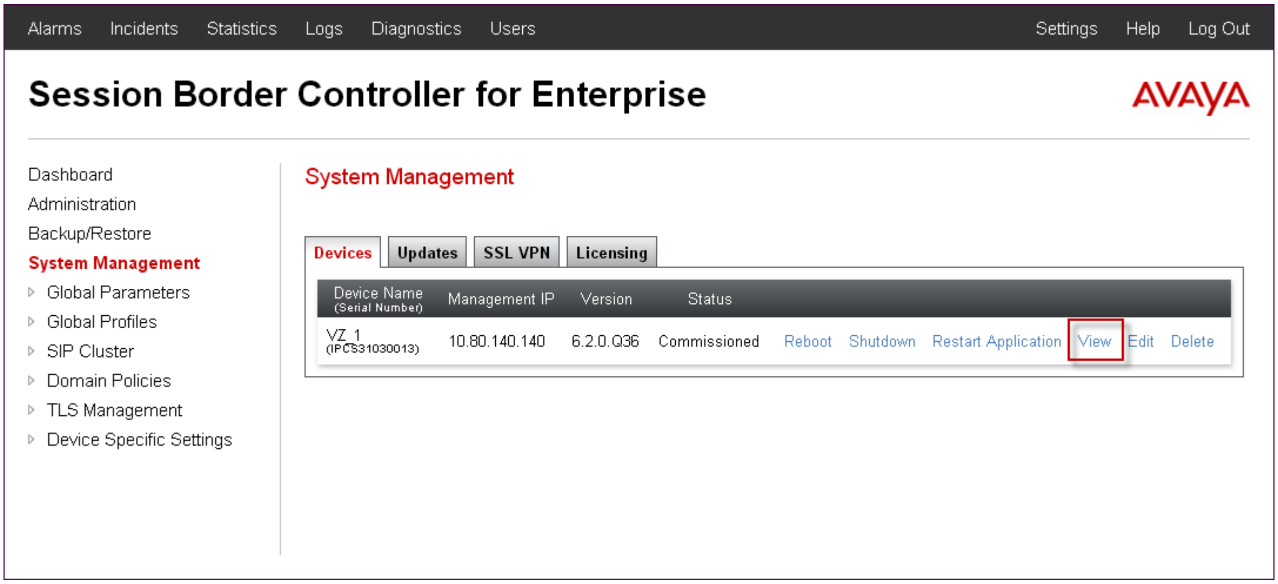
In the sample configuration, the management IP is 10.80.140.140. Access the web management interface by entering https://<ip-address> where <ip-address> is the management IP address assigned during installation. Log in with the appropriate credentials. Click **Log In**.



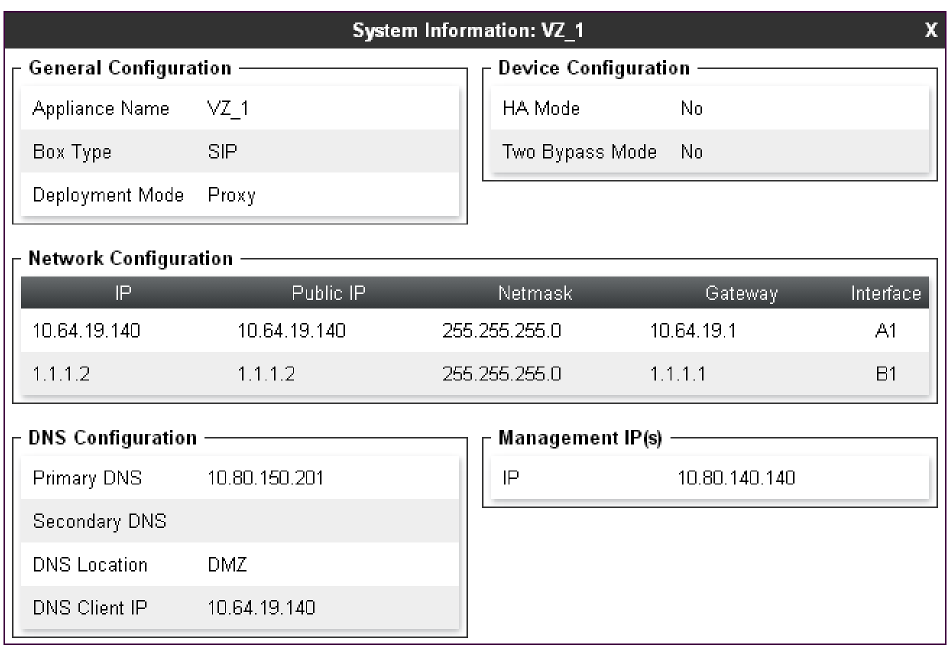
The main page of the Avaya SBCE will appear.



To view system information that was configured during installation, navigate to **System Management**. A list of installed devices is shown in the right pane. In the case of the sample configuration, a single device named “VZ_1” is shown. To view the configuration of this device, click **View** as highlighted below.

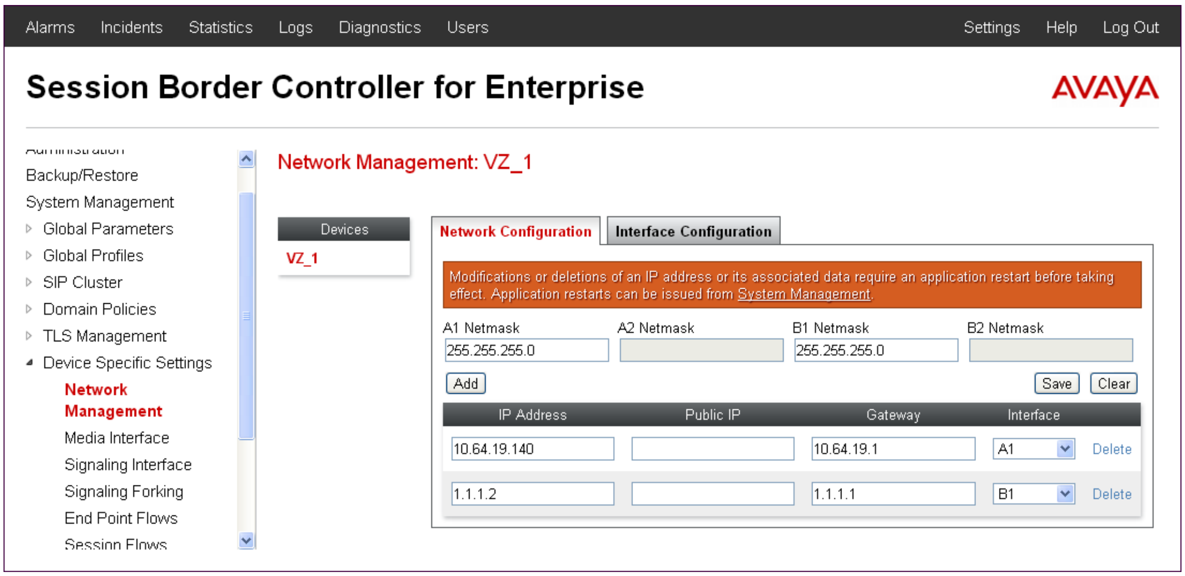


The **System Information** screen shows the **Network Settings**, **DNS Configuration**, and **Management IP** information provided during installation and corresponds to **Figure 1**. The **Box Type** was set to “SIP” and the **Deployment Mode** was set to “Proxy”. Default values were used for all other fields. Note that the **Management IP** must be on a separate subnet from the IP interfaces designated for SIP traffic.

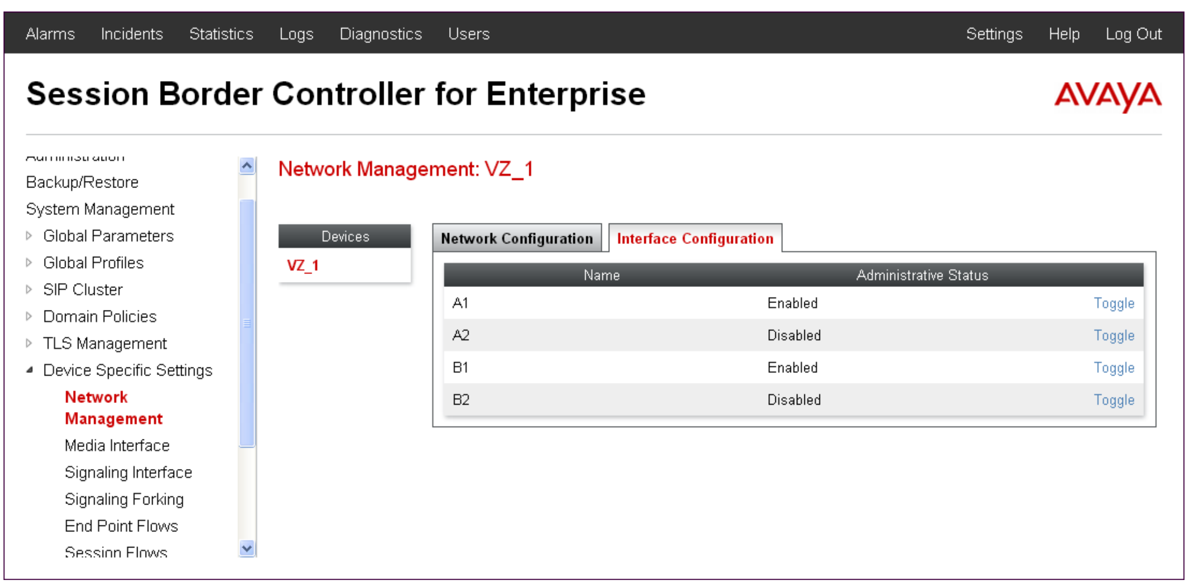


7.1. Network Management

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



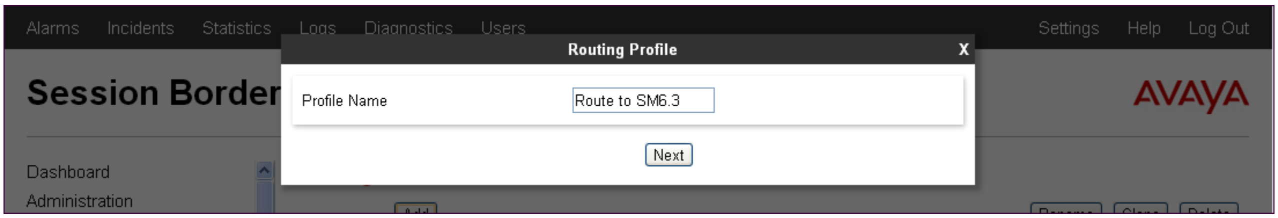
The following screen shows interface **A1** and **B1** are **Enabled**. To enable an interface click the corresponding **Toggle State** button.



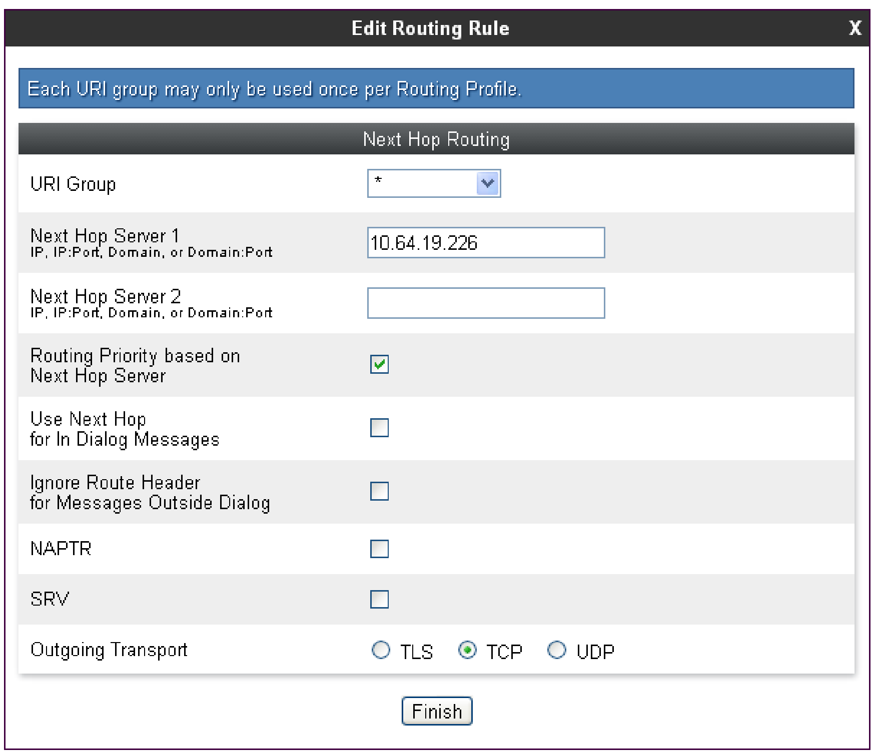
7.2. Routing Profile

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

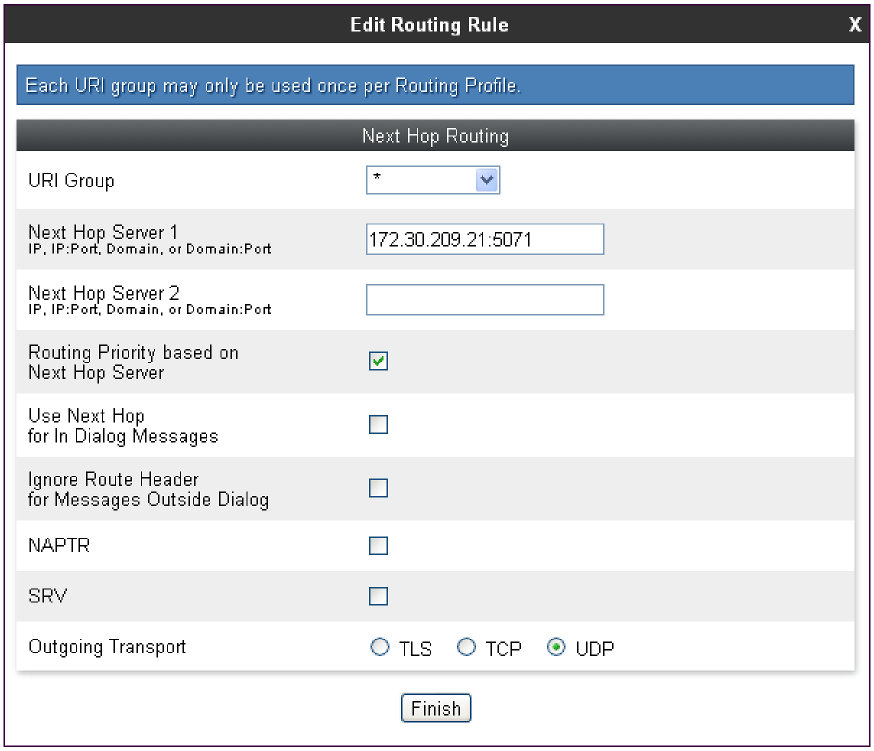
Create a Routing Profile for Session Manager and Verizon IP Trunk service. To add a routing profile, navigate to **Global Profiles → Routing** and select **Add**. Enter a **Profile Name** and click **Next** to continue.



In the shared test environment the following screen shows Routing Profile "Route to SM6.3" created for Session Manager. The **Next Hop Server 1** IP address must match the IP address of Session Manager Entity created in **Section 6.4**. The **Outgoing Transport** is set to **TCP** and matched the **Protocol** set in the Session Manager Entity Link for Avaya SBCE in **Section 6.5**.



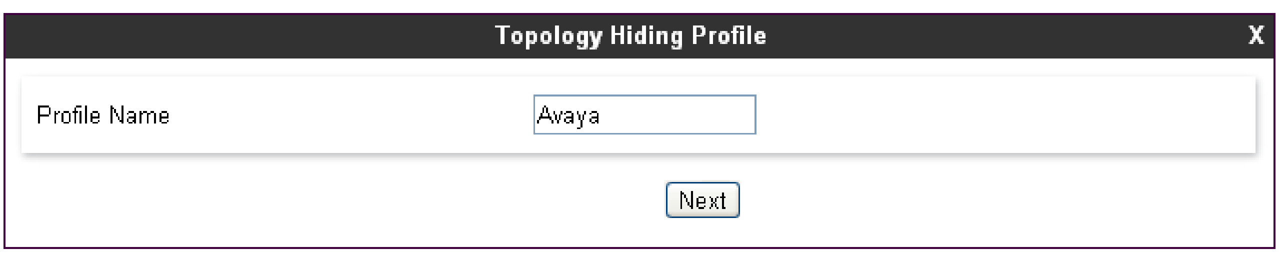
The following screen shows Routing Profile "Route To Vz_IPT" created for Verizon. Enter the IP address and port of the Verizon SIP signaling interface as **Next Hop Server 1**, as shown below. It is only necessary to include the port after the IP address when it is not the default SIP port. Choose **UDP** for **Outgoing Transport**, and click **Finish**.



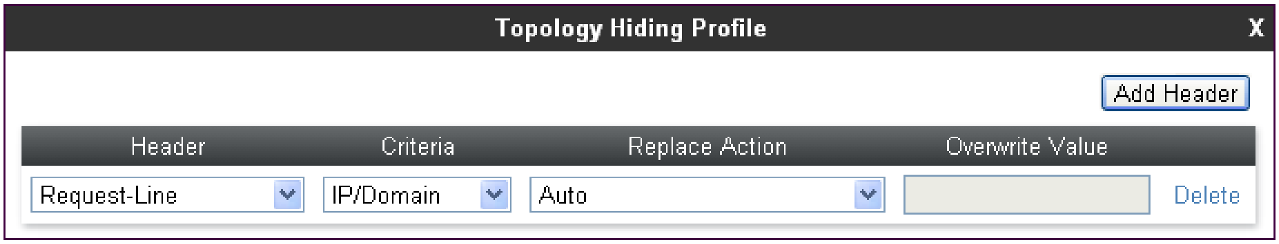
7.3. Topology Hiding Profile

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

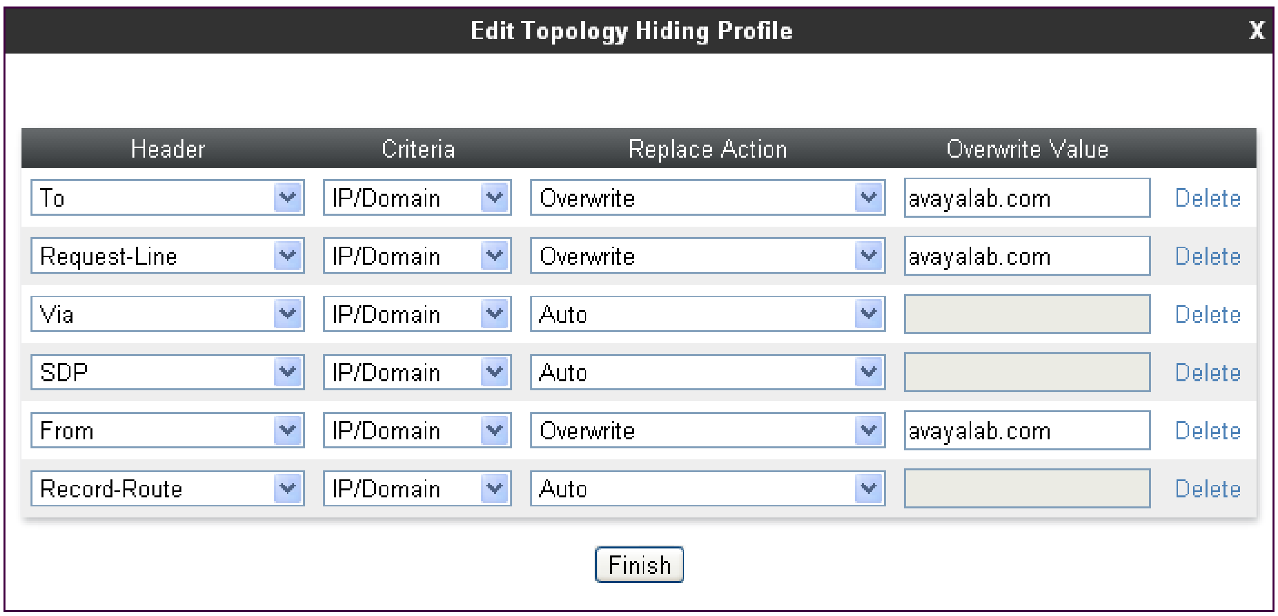
Click the **Add** button (not shown) to add a new profile, or select an existing topology hiding profile to edit. If adding a profile, a screen such as the following is displayed. Enter a **Profile Name** such as "Avaya" shown below. Click **Next**.



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



In the **Replace Action** column an action of "Auto" will replace the header field with the IP address of the Avaya SBCE interface and the "Overwrite" will use the value in the **Overwrite Value**. In the example shown, this profile will later be applied in the direction of the Session Manager and "Overwrite" has been selected for the To/From and Request-Line headers and the shared interop lab domain of "avayalab.com" has been inserted. Click **Finish**.



After configuration is completed, the Topology Hiding for profile "Avaya" will appear as follows. This profile will later be applied to the Server Flow for Avaya.

Topology Hiding

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

Edit

Similarly, create a Topology Hiding profile for Verizon. The following screen shows Topology Hiding profile "VzIPT-TopoHiding" created for Verizon. This profile will later be applied to the Server Flow for Verizon.

Topology Hiding Profiles: VzIPT-TopoHiding

Add | Rename | Clone | Delete

Topology Hiding Profiles
- default
- cisco_th_profile
- Avaya
- IPCC_Topology_Hiding
- **VzIPT-TopoHiding**

Click here to add a description.

Topology Hiding

Header	Criteria	Replace Action	Overwrite Value
SDP	IP/Domain	Auto	---
To	IP/Domain	Overwrite	pcelban0001.avayalincroft.globalipcom.com
Record-Route	IP/Domain	Auto	---
From	IP/Domain	Overwrite	adevc.avaya.globalipcom.com
Via	IP/Domain	Auto	---
Request-Line	IP/Domain	Overwrite	pcelban0001.avayalincroft.globalipcom.com

Edit

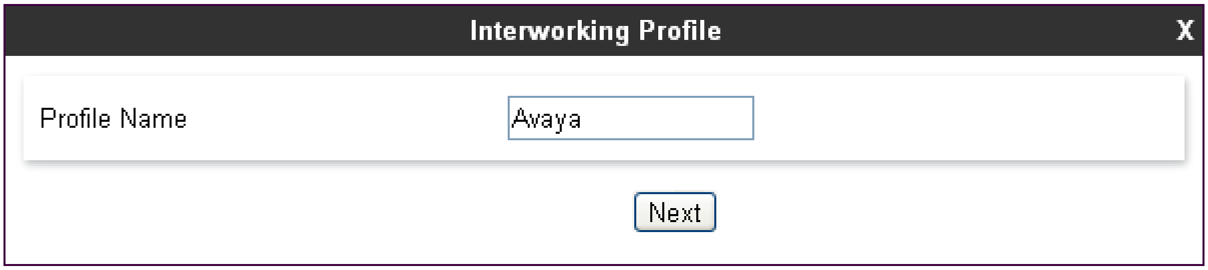
7.4. Server Interworking Profile

The Server Internetworking profile includes parameters to make the Avaya SBCE function in an enterprise VoIP network using different implementations of the SIP protocol. There are default profiles available that may be used as is, or modified, or new profiles can be configured as described below.

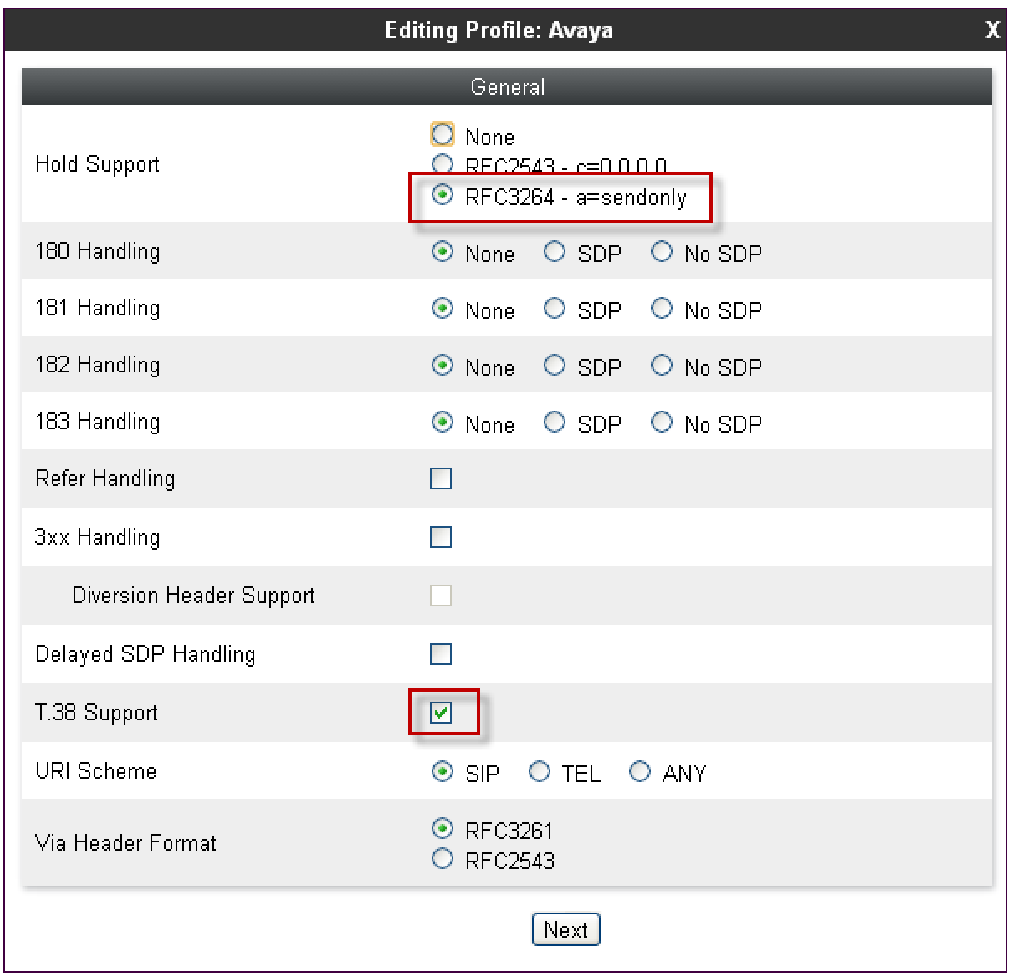
In the sample configuration, separate Server Interworking Profiles were created for Avaya and Verizon IP Trunk.

7.4.1 Server Interworking– Avaya

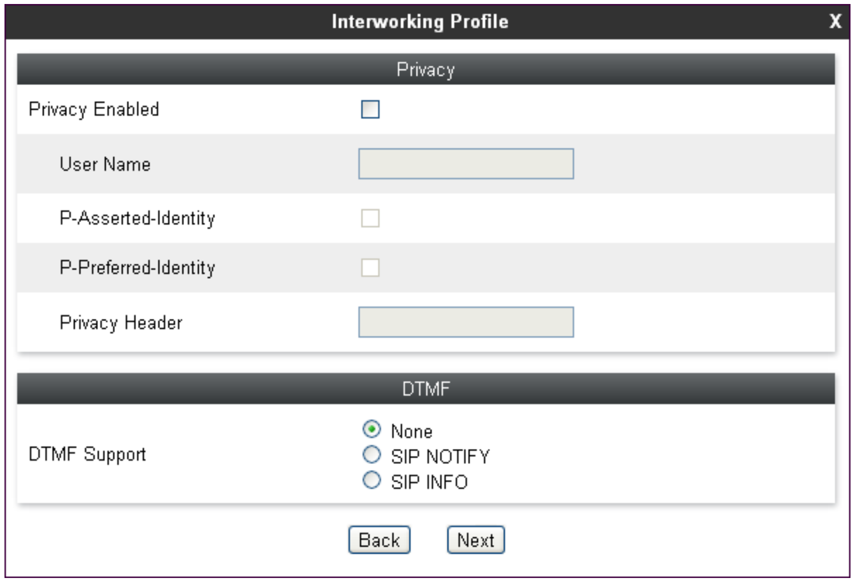
Navigate to **Global Profiles → Server Interworking** and click the **Add Profile** button (not shown) to add a new profile or select an existing interworking profile. If adding a profile, a screen such as the following is displayed. Enter an appropriate **Profile Name** such as “Avaya” shown below. Click **Next**.

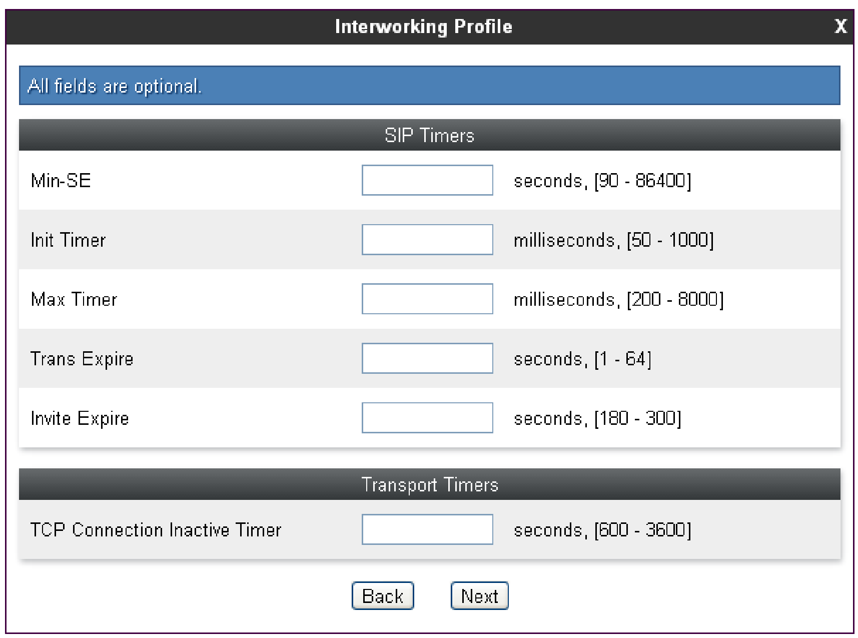


The following screens illustrate the “General” parameters used in the sample configuration for the Interworking Profile named “Avaya”. Most parameters retain default values. In the sample configuration, **RFC3264 – a=sendonly** is selected and **T.38 support** is checked.

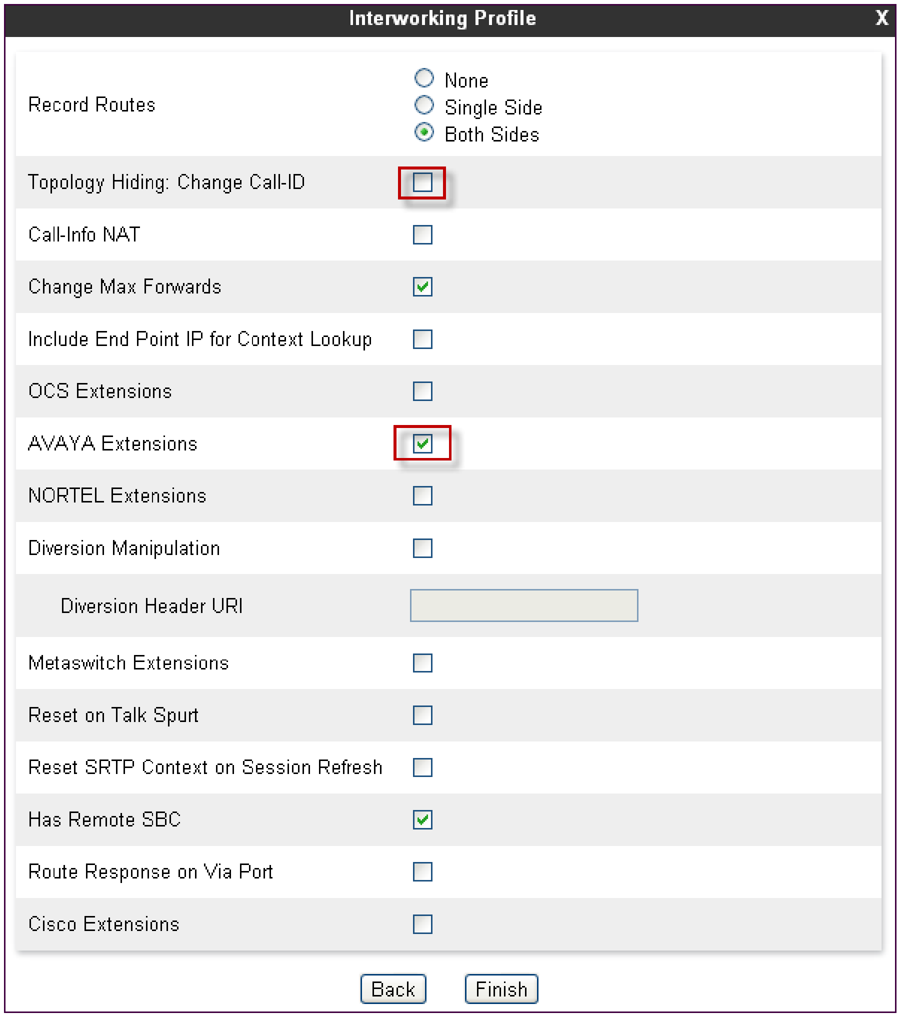


Click **Next** to advance to through both the Privacy / DTMF parameters screen, and the SIP / Transport Timers parameters screen, which may retain default values.



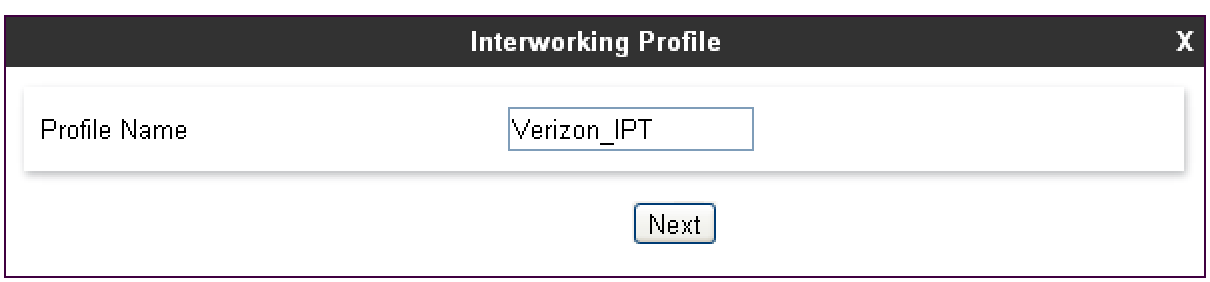


The following screen illustrates the **Advanced Settings** configuration. The **Topology Hiding: Change Call-ID** is unchecked and the **AVAYA Extensions** is checked. All other parameters shown are default values. Note that the default configuration will result in Record-Route headers in SIP messages.

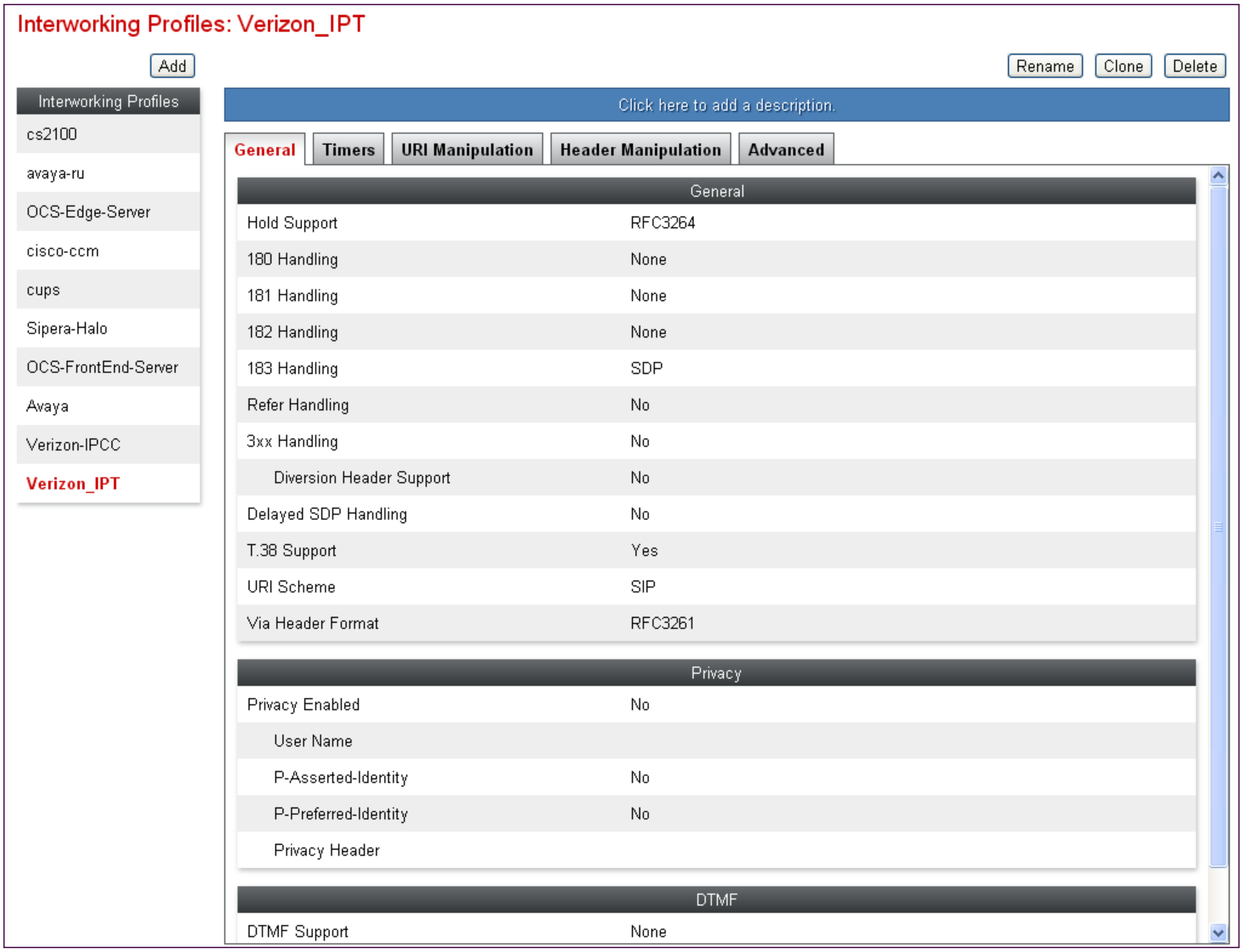


7.4.2 Server Interworking – Verizon IP Trunk

Click the **Add** 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_IPT" shown below. Click **Next**.



The following screens illustrate the **General** parameters used in the sample configuration for the Interworking Profile named “Verizon_IPT”. Most parameters retain default values. In the sample configuration, **183 Handling** is set to “SDP” to make sure all “183 Session Progress” messages include SDP. Verizon requires SDP to be included for all “183 Session Progress” messages. **T.38 support** is set to “Yes”, **Hold Support** is set for RFC3264, and all other fields retained default values.



On the Timers tab, select 6 seconds for the **Trans Expire** timer as shown below.

General | Timers | URI Manipulation | Header Manipulation | Advanced

SIP Timers	
Min-SE	---
Init Timer	---
Max Timer	---
Trans Expire	6 seconds
Invite Expire	---

Transport Timers	
TCP Connection Inactive Timer	---

Edit

The following screen illustrates the **Advanced Settings** configuration. The **Topology Hiding: Change Call-ID** and **Change Max Forwards** defaults were changed to “No”. All other parameters shown are default values. Note that the default configuration will result in Record-Route headers in SIP messages.

General | Timers | URI Manipulation | Header Manipulation | Advanced

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

Edit

7.5. Signaling Manipulation

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

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

These Application Notes will not discuss the full feature of Signaling Manipulation but will show an example of a script created during compliance testing. The sample script is used to remove the “epv” parameter Session Manager places in the Contact header. This parameter contains Endpoint-View information, including the internal domain. Removing this parameter helps mask the internal topology of the enterprise. The Endpoint-View header and other proprietary headers are removed using a Signaling Rule as illustrated in **Section 7.8**. This configuration is optional, in that the “epv” parameter does not cause any user-perceivable problems if presented to Verizon.

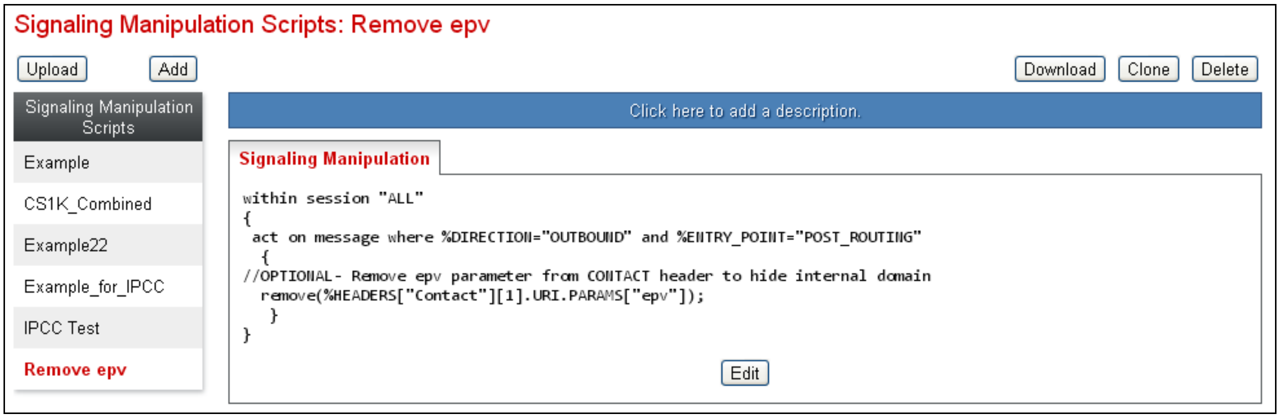
To create a new Signaling Manipulation, navigate to **Global Profiles → Signaling Manipulation** and click on **Add**. A new blank SigMa Editor window will pop up.

The following screen illustrates the “Remove epv” script.

```
within session "ALL"
{
 act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
  {
//OPTIONAL- Remove epv parameter from CONTACT header to hide internal domain
  remove(%HEADERS["Contact"][1].URI.PARAMS["epv"]);
   }
}
```

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

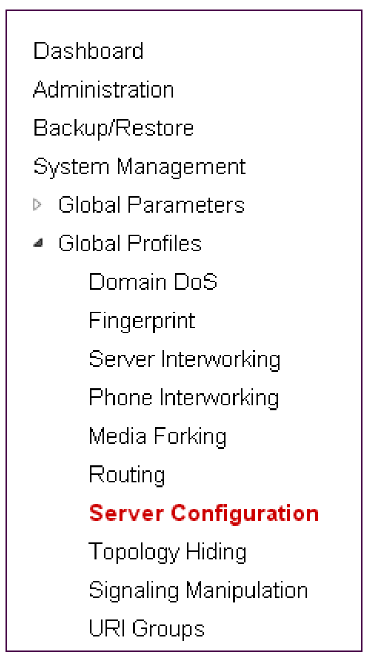
The following screen shows the finished Signaling Manipulation Script “Remove epv” used during compliance testing. This script will later be applied to the Verizon Server Configuration in **Section 7.6.2**.



7.6. Server Configuration

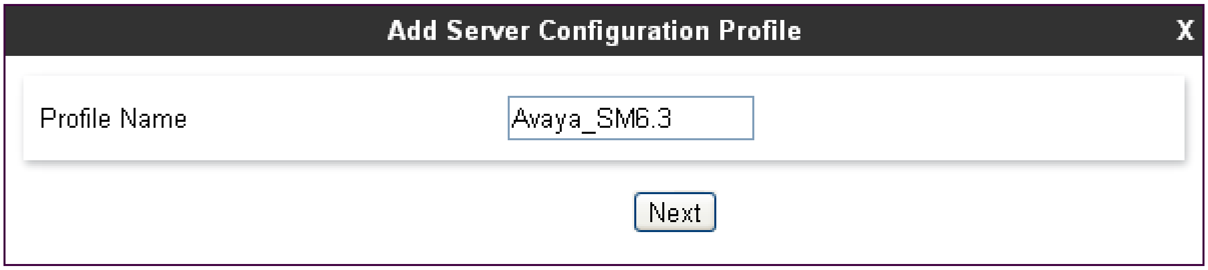
The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters, DoS security statistics,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

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

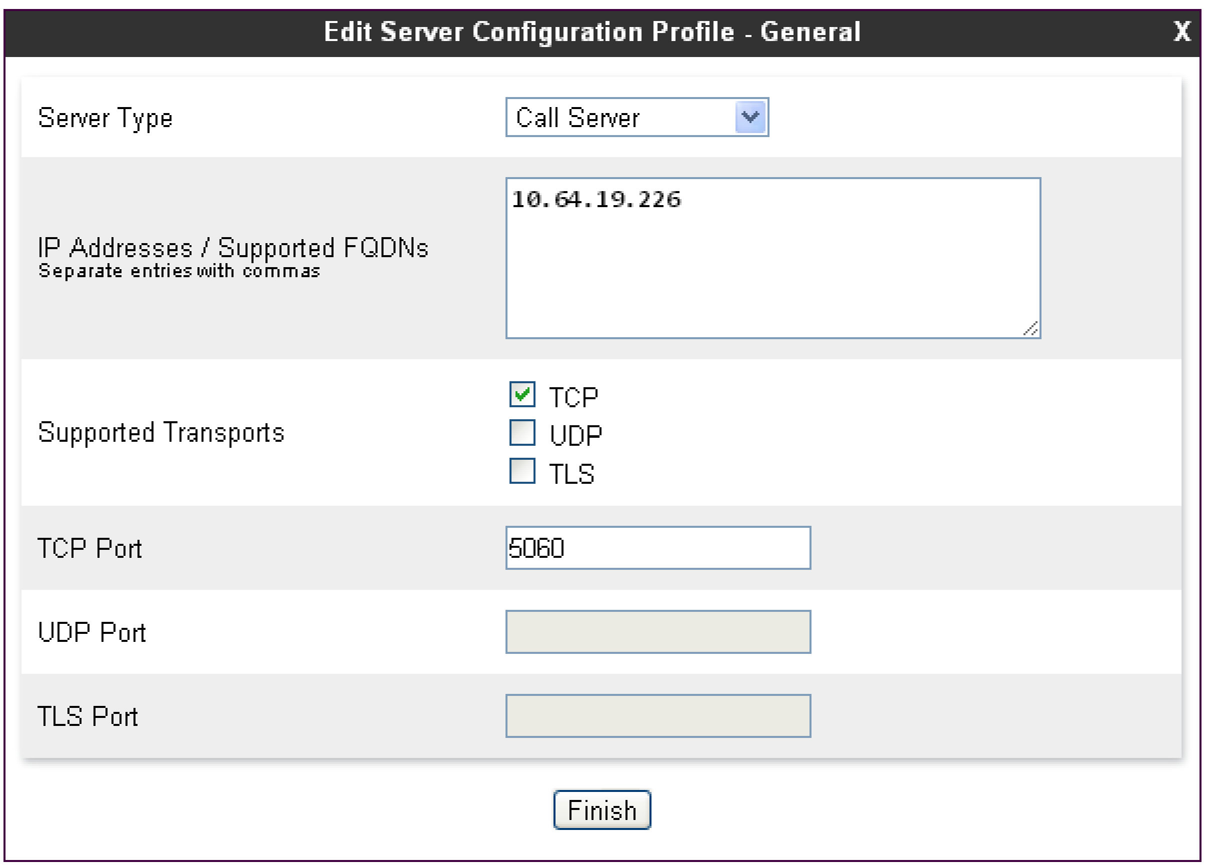


7.6.1 Server Configuration for Session Manager

Click the **Add** 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_SM6.3” shown below. Click **Next**.

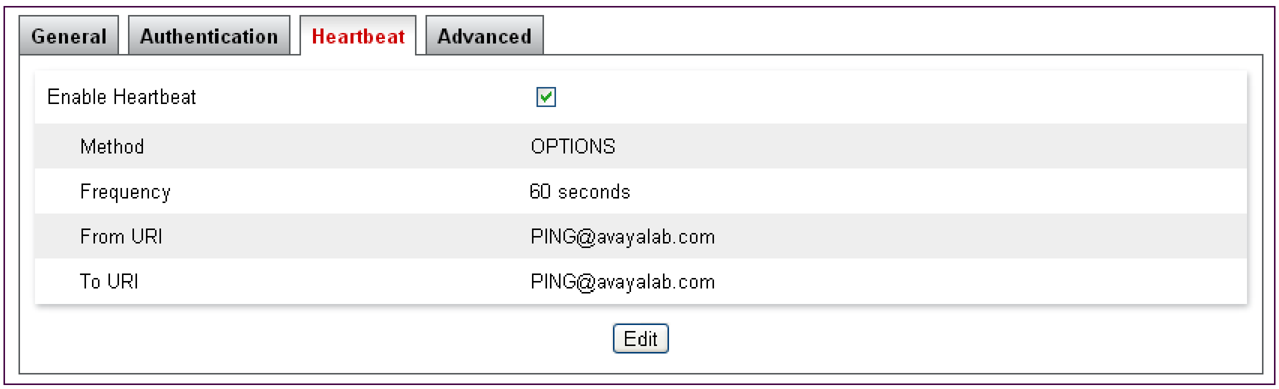


The following screens illustrate the Server Configuration for the Profile name “Avaya_SM6.3”. On the **General** tab, select “Call Server” from the **Server Type** drop-down menu. In the **IP Addresses / Supported FQDNs** area, the IP Address of the Session Manager SIP signaling interface in the sample configuration is entered. This IP Address is 10.64.19.226. In the **Supported Transports** area, **TCP** is selected, and the **TCP Port** is set to 5060. This configuration corresponds with the Session Manager entity link configuration for the entity link to the Avaya SBCE created in **Section 6.4**. If adding a new profile, click **Next** (not shown). If editing an existing profile, click **Finish**.

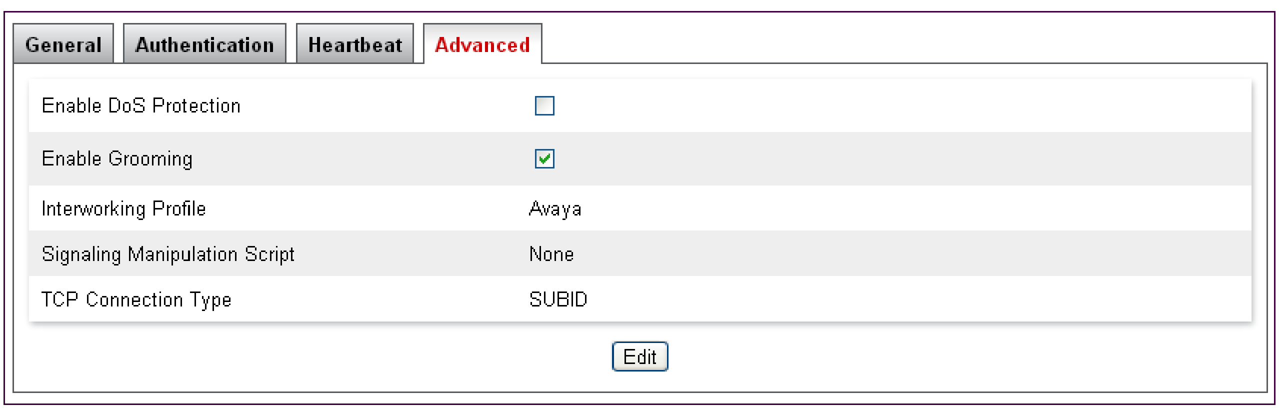


If adding the profile, click **Next** to accept default parameters for the **Authentication** tab (not shown), and advance to the Heartbeat area. If editing an existing profile, select the **Heartbeat** tab and click **Edit** (not shown).

Avaya SBCE can be configured to source "heartbeats" in the form of SIP OPTIONS. In the sample configuration, with one Session Manager, this configuration is optional. If Avaya SBCE-sourced OPTIONS messages are desired, check the **Enable Heartbeat** box. Select "OPTIONS" from the **Method** drop-down menu. Select the desired frequency that the Avaya SBCE 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 Avaya SBCE towards Session Manager. If adding a new profile, click **Next** (not shown). If editing an existing profile, click **Finish** (not shown).

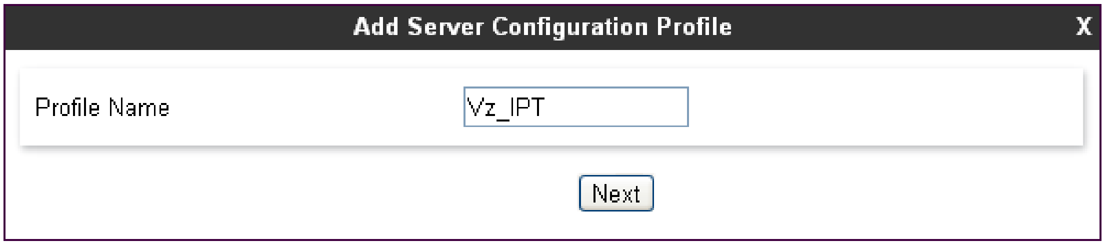


If adding a profile, click **Next** to continue to the "Advanced" settings (not shown). If editing an existing profile, select the **Advanced** tab and **Edit** (not shown). In the resultant screen, select **Enable Grooming** to allow the same TCP connection to be used for all SIP messages from this device. Select the **Interworking Profile** "Avaya" created previously. Click **Finish**.

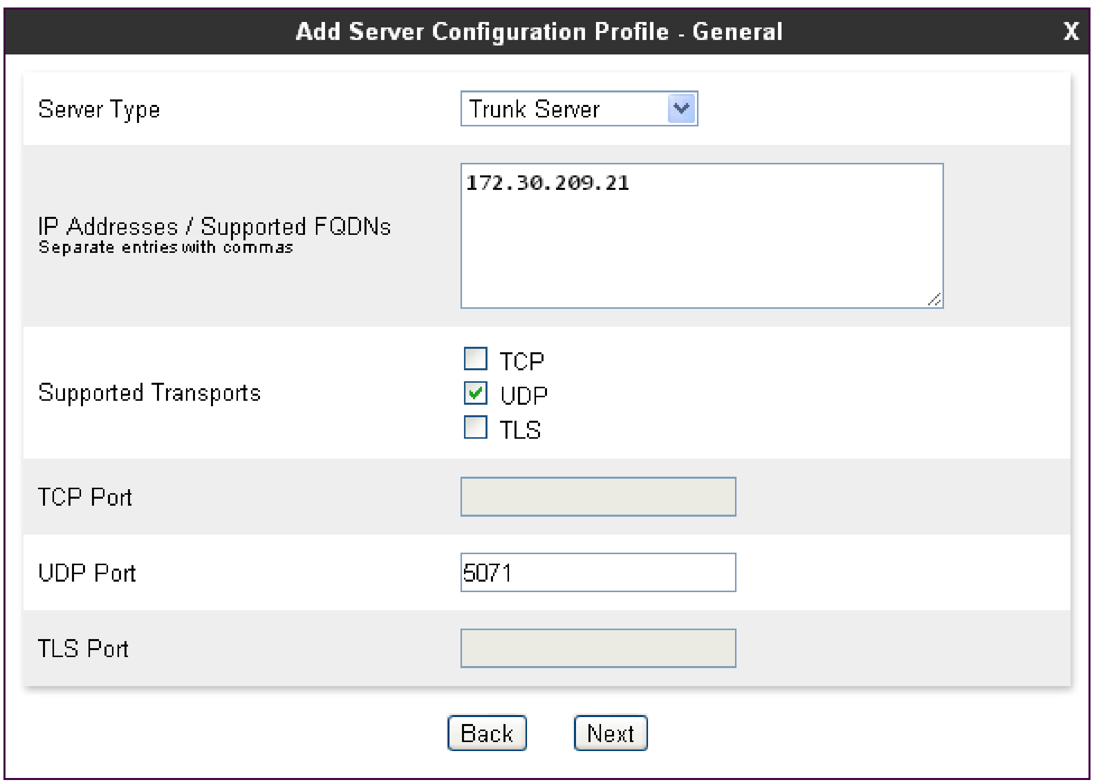


7.6.2 Server Configuration for Verizon IP Trunk

Click the **Add** 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 "Vz_IPT" shown below. Click **Next**.



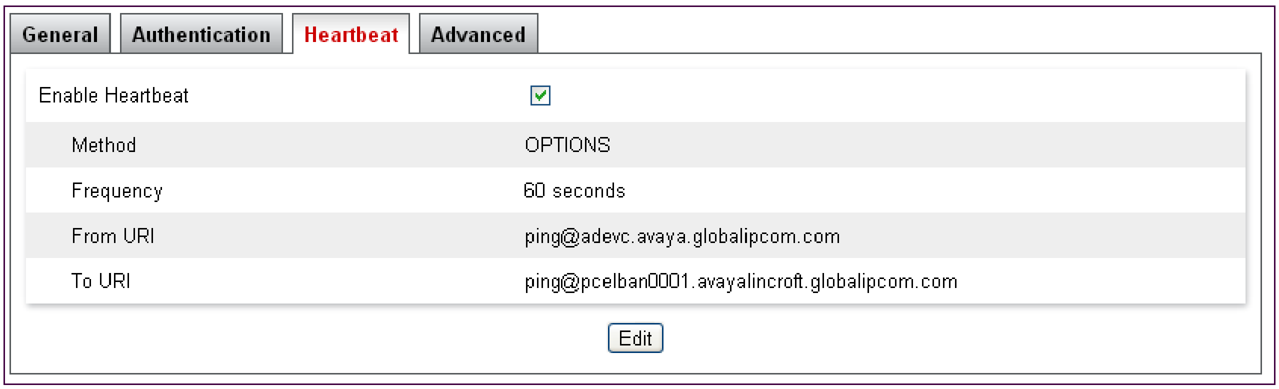
The following screens illustrate the Server Configuration with Profile name "Vz_IPT". In the "General" parameters, select "Trunk Server" from the **Server Type** drop-down menu. In the **IP Addresses / Supported FQDNs** area, the Verizon-provided IP Trunk IP Address is entered. This IP Address is 172.30.209.21.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



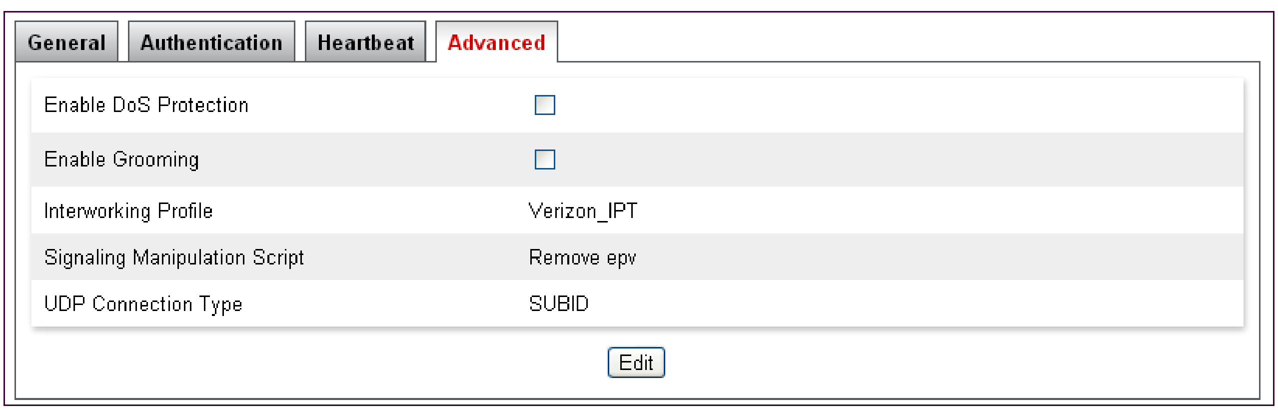
If adding the profile, click **Next** to accept default parameters for the **Authentication** tab (not shown), and advance to the Heartbeat area. If editing an existing profile, select the **Heartbeat** tab and click **Edit** (not shown).

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

Select “OPTIONS” from the **Method** drop-down menu. Select the desired frequency that the SBCE 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 Avaya SBCE. If adding a new profile, click **Next** to continuing to the “Advanced” settings. If editing an existing profile, click Finish (not shown).



If editing an existing profile, highlight the desired profile and select the **Advanced** tab and then click the **Edit button**. In the resultant screen, **Enable Grooming** is not used for UDP connections and left unchecked. Select the **Interworking Profile** “Verizon_IPT” created previously, and **Signaling Manipulation Script** will be the script shown in the previous section titled “Remove epv”. Click **Finish**.

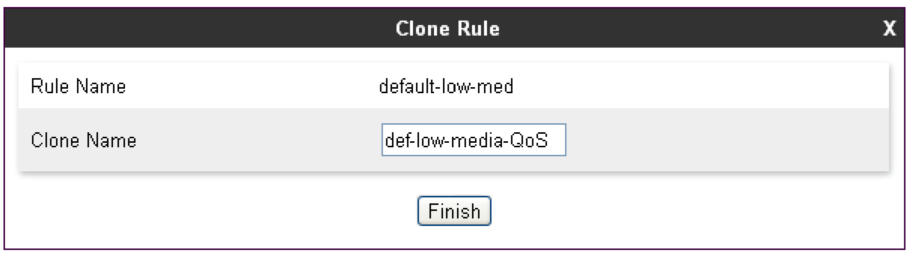


7.7. Media Rule

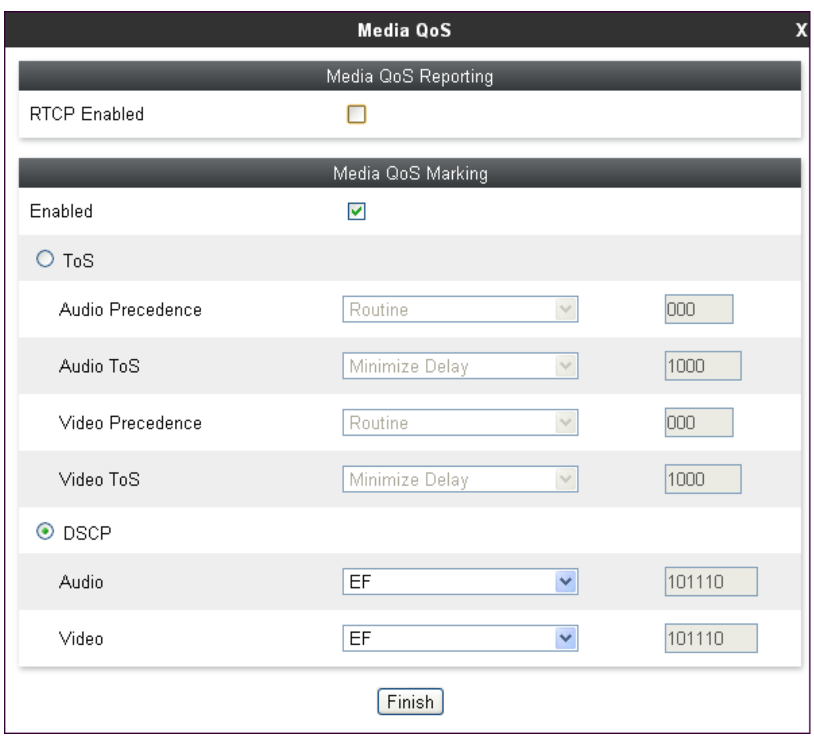
Media Rules define RTP media packet parameters such as prioritizing encryption techniques and packet encryption techniques. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies to determine how media packets matching these criteria will be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

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

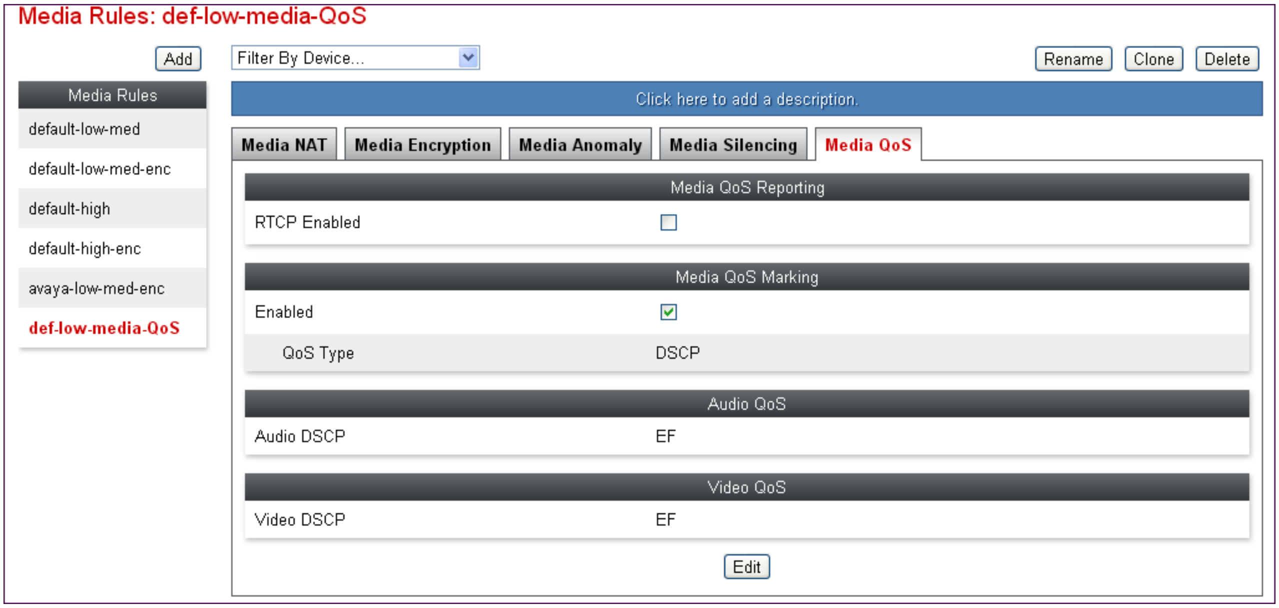
Enter a name in the **Clone Name** field, such as “def-low-media-QoS” as shown below. Click **Finish**.



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



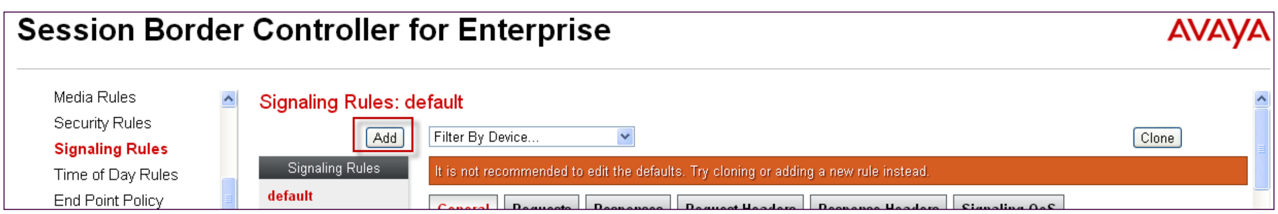
When configuration is complete, the “default-low-media-QoS” media rule **Media QoS** tab appears as follows.



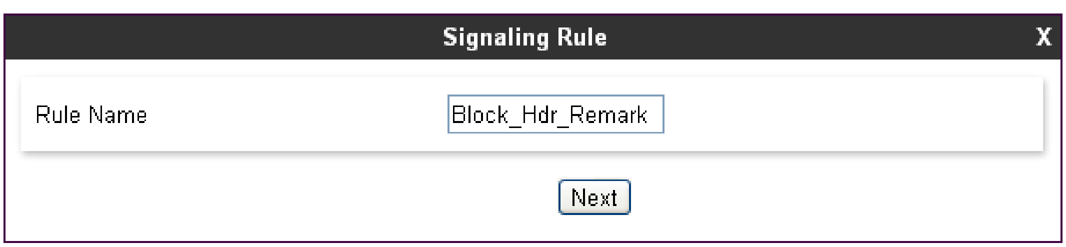
7.8. Signaling Rule

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

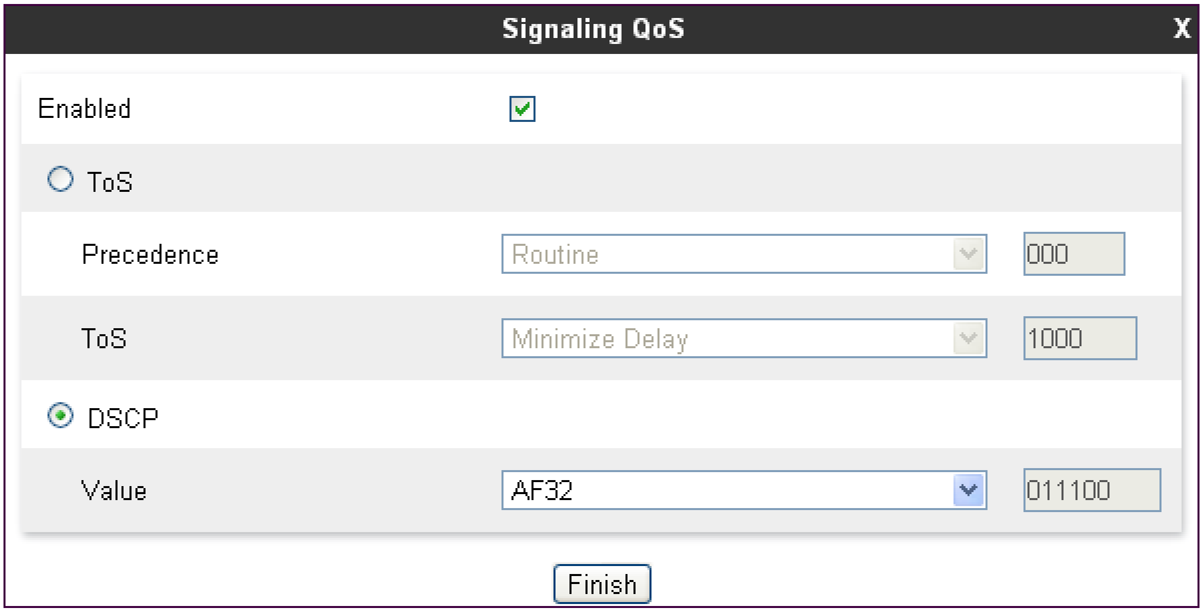
To add a signaling rule, navigate to **Domain Policies → Signaling Rules**. Click the **Add** button to add a new signaling rule.



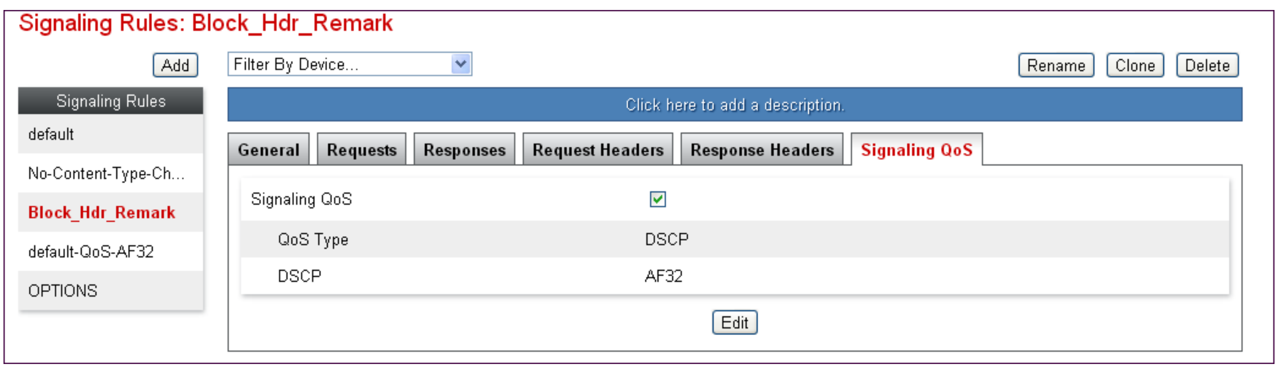
In the **Rule Name** field, enter an appropriate name, such as “Block_Hdr_Remark” and click **Next**.



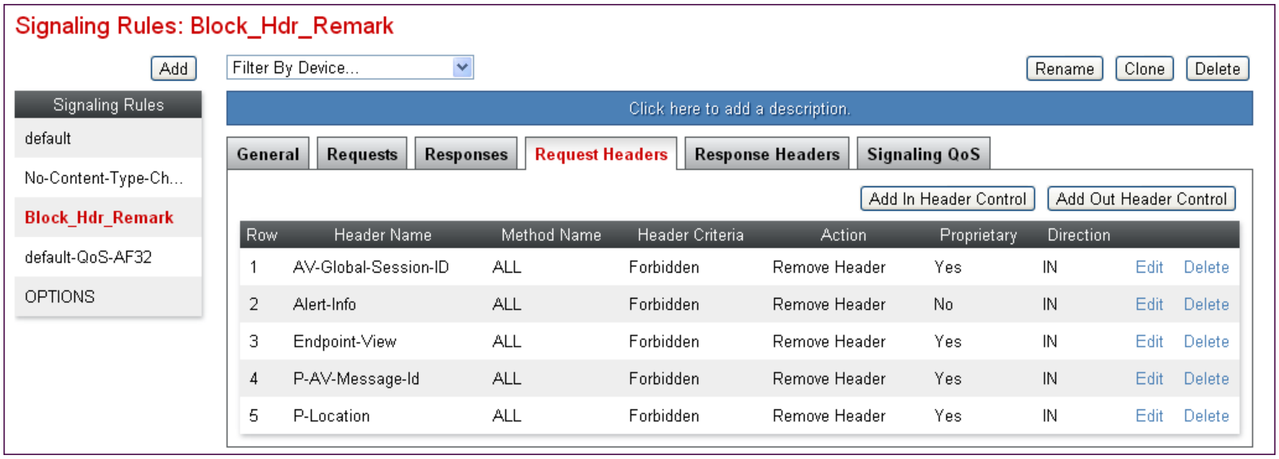
In the subsequent screen (not shown), click **Next** to accept defaults. In the **Signaling QoS** screen below, select **DSCP** and select the desired **Value** for Signaling QoS from the drop-down box. In the sample configuration, “AF32” is selected for Assured Forwarding 32. Click **Finish**.



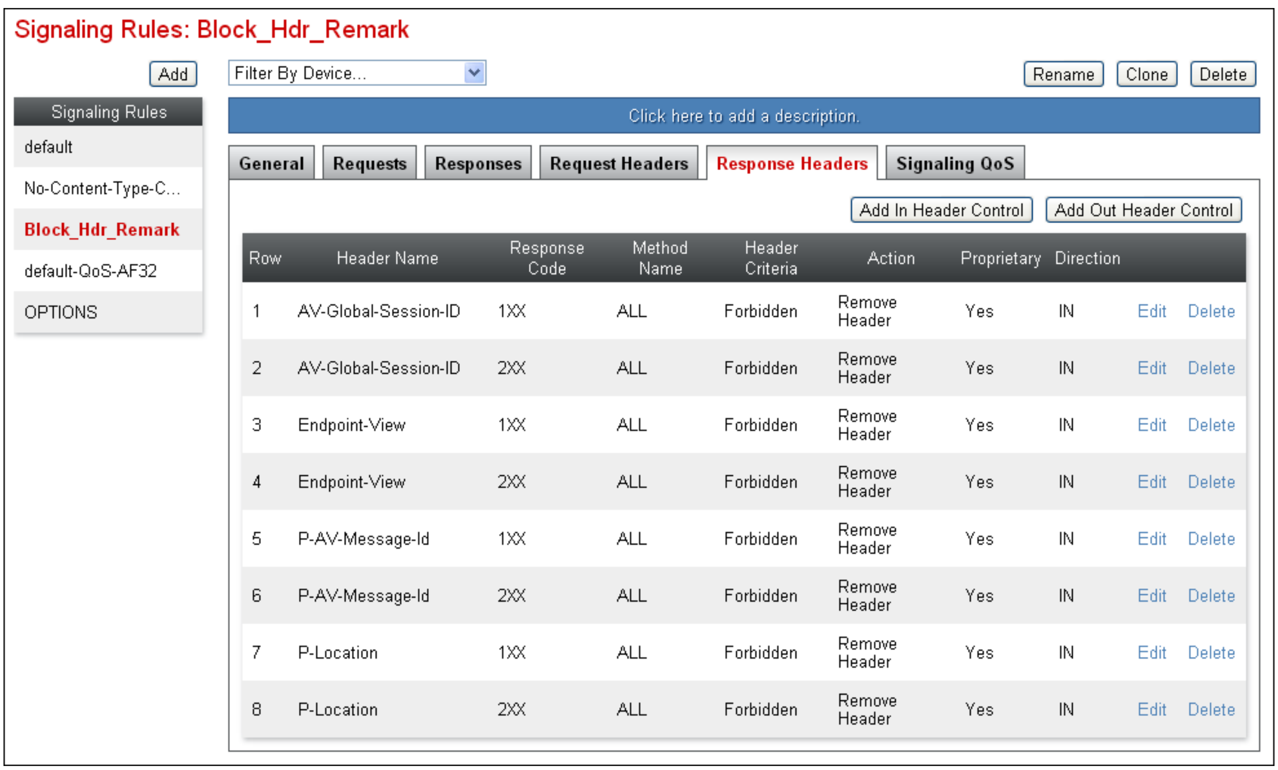
After this configuration, the new “Block_Hdr_Remark” will appear as follows.



Select this rule in the center pane, then select the **Request Headers** tab to view the manipulations performed on the request messages such as the initial INVITE or UPDATE message. The following screen shows the “Alert-Info”, “Endpoint-View”, “P-Location" and other proprietary headers removed during the compliance test. This configuration is optional in that these headers do not cause any user-perceivable problems if presented to Verizon.



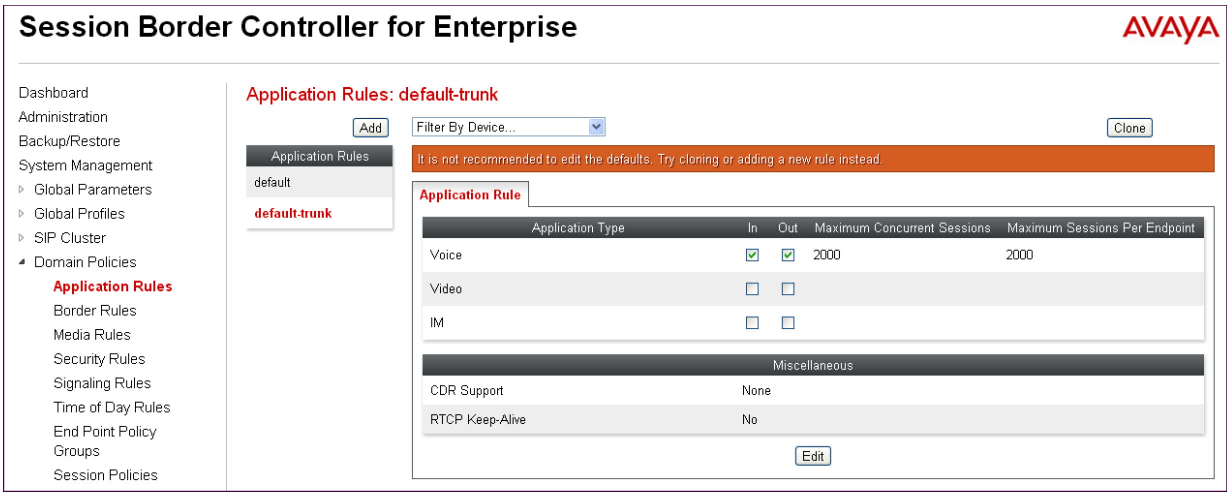
Similarly, manipulations can be performed on the SIP response messages. These can be viewed by selecting the **Response Headers** tab as shown below.



7.9. Application Rule

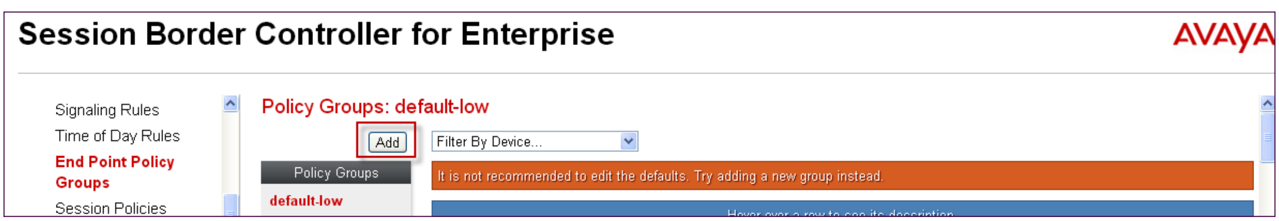
Application Rules define which types of SIP-based Unified Communications (UC) applications the Avaya SBCE security device will protect: voice, video, and/or Instant Messaging (IM). In addition, user can determine the maximum number of concurrent voice and video sessions the network will process in order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

Select **Domain Policies → Application Rules** from the left-side menu as shown below. In the sample configuration, a single default application rule "default-trunk" is used and will be applied to the Endpoint Policy Group in the next section.

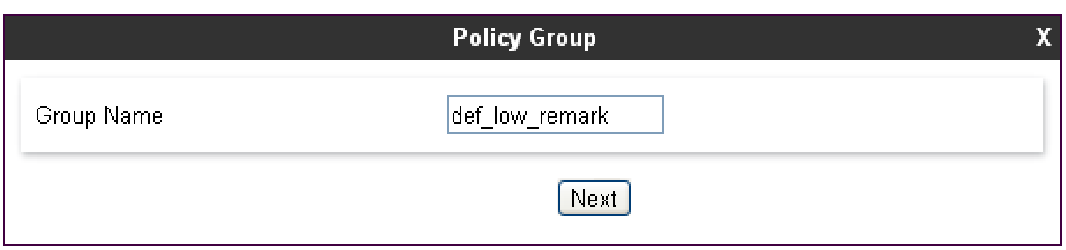


7.10. Endpoint Policy Group

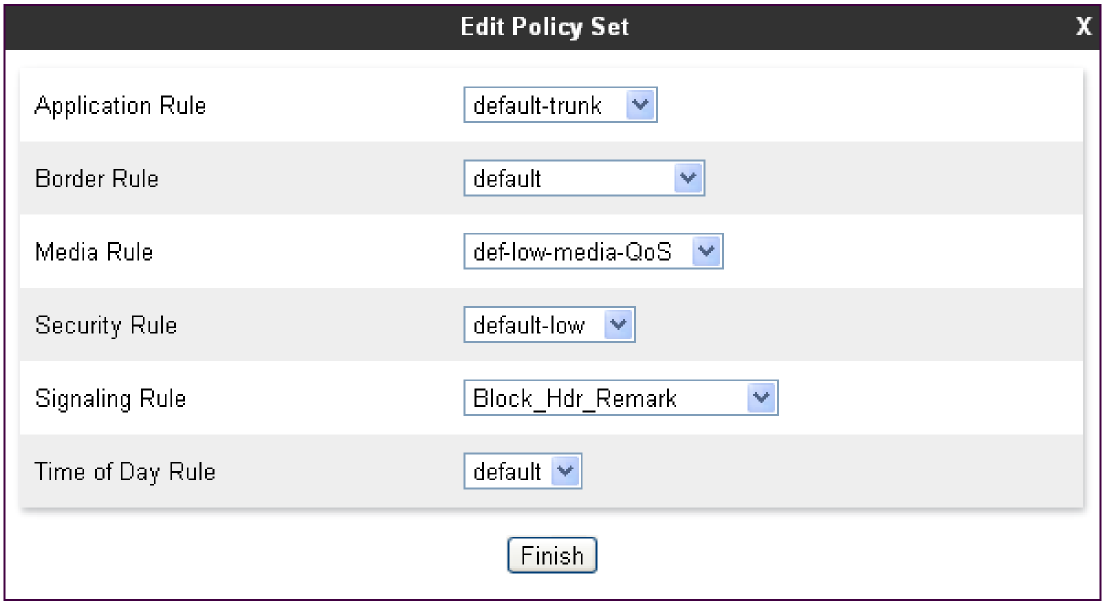
The rules created within the Domain Policy section are assigned to an Endpoint Policy Group. The Endpoint Policy Group is then applied to a Server Flow in **Section 7.13**. Create a separate Endpoint Policy Group for the enterprise and the Verizon IP Trunk. To create a new policy group, navigate to **Domain Policies → Endpoint Policy Groups**. Select the **Add** button.



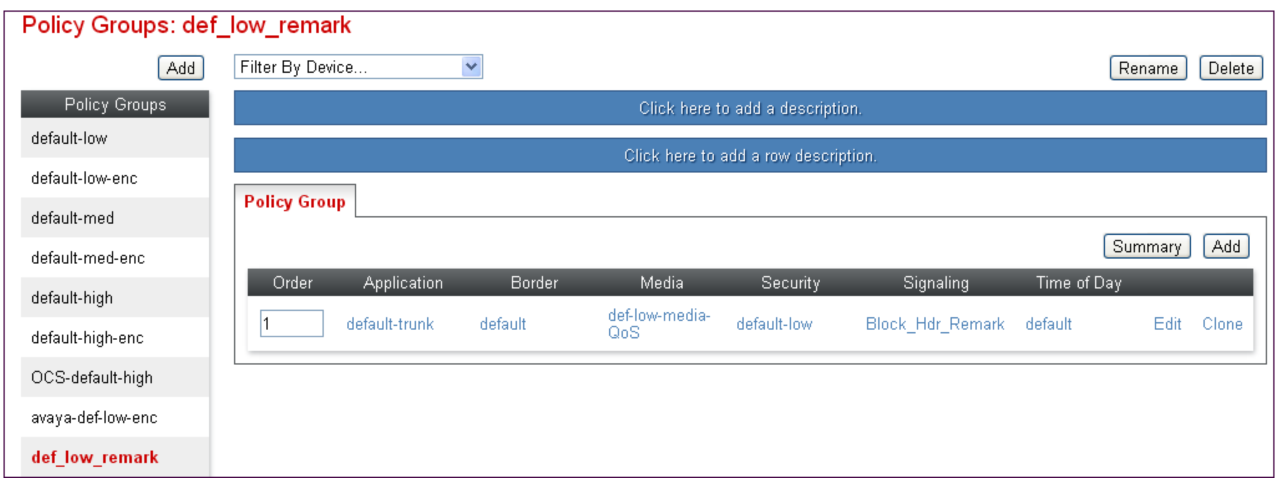
Enter a name in the **Group Name** field, such as "def_low_remark" as shown below. Click **Next**.



In the sample configuration, defaults were selected for all fields, with the exception of the **Application Rule** which is set to "default-trunk", **Media Rule** which is set to "default-low-media-QoS", and the **Signaling Rule**, which is 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**.



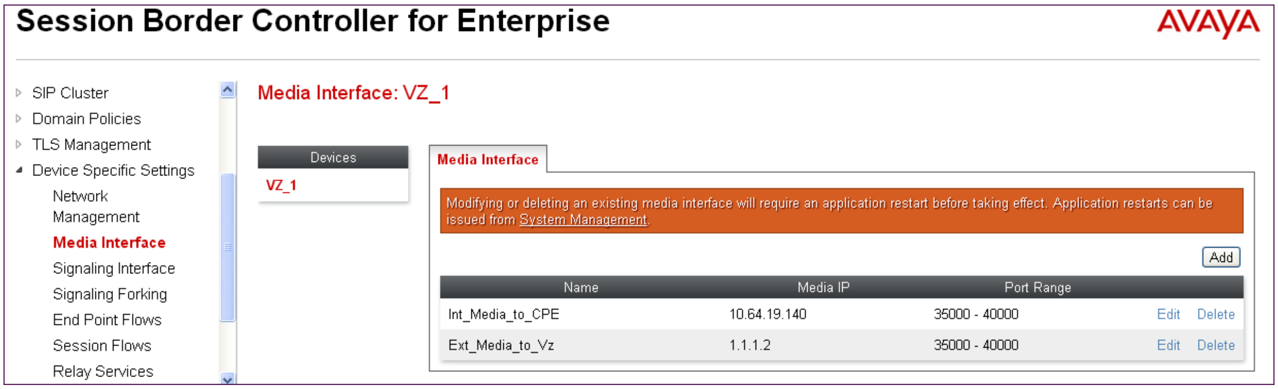
Once configuration is completed, the "default-low-remark" policy group will appear as follows.



7.11. Media Interface

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

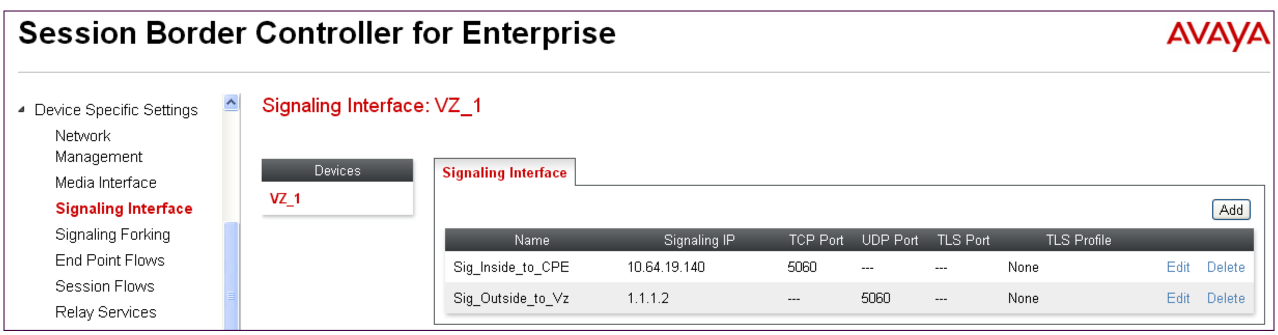
To create a new Signaling Interface, navigate to **Device Specific Settings → Media Interface** and click **Add Media Interface**. The following screen shows the media interfaces created in the sample configuration for the inside and outside IP interfaces.



7.12. Signaling Interface

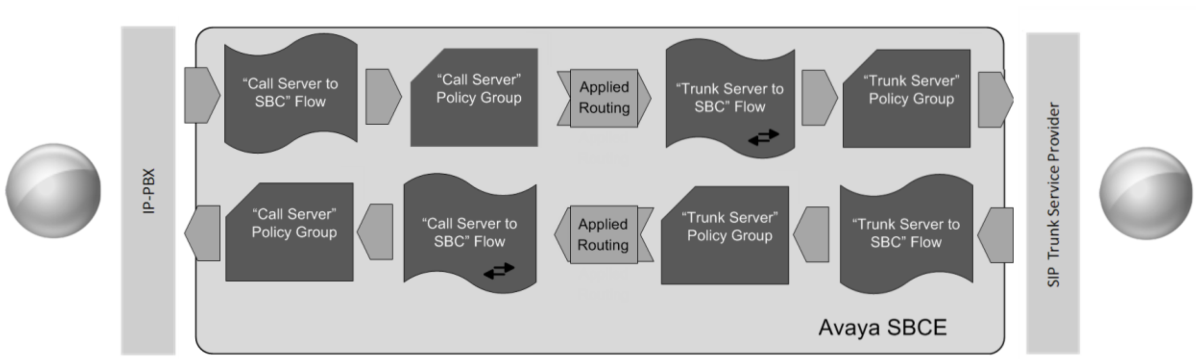
The Signaling Interface screen is where the SIP signaling ports are defined. Avaya SBCE will listen for SIP requests on the defined ports. Create a Signaling Interface for both the inside and outside IP interfaces.

To create a new Signaling Interface, navigate to **Device Specific Settings → Signaling Interface** and click **Add Signaling Interface**. The following screen shows the signaling interfaces created in the sample configuration for the inside and outside IP interfaces.

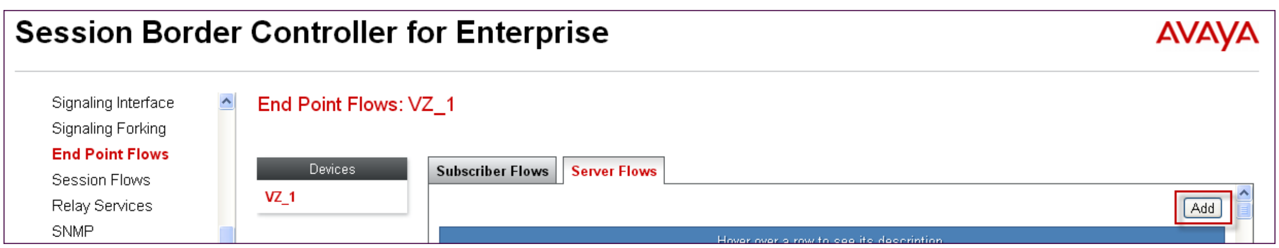


7.13. End Point Flows - Server Flow

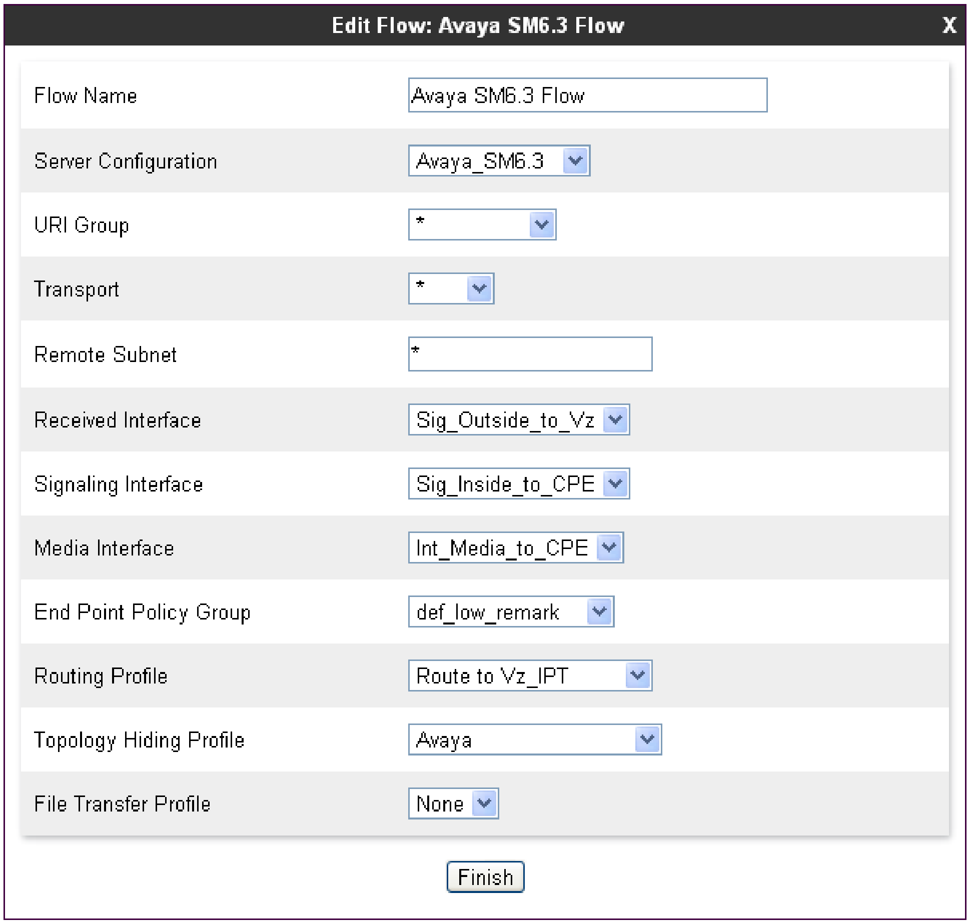
When a packet is received by Avaya SBCE, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to a policy which contains several rules concerning processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for this destination endpoint are applied. The context is maintained, so as to be applied to future packets in the same flow. The following screen illustrates the flow through the Avaya SBCE to secure a SIP Trunk call.



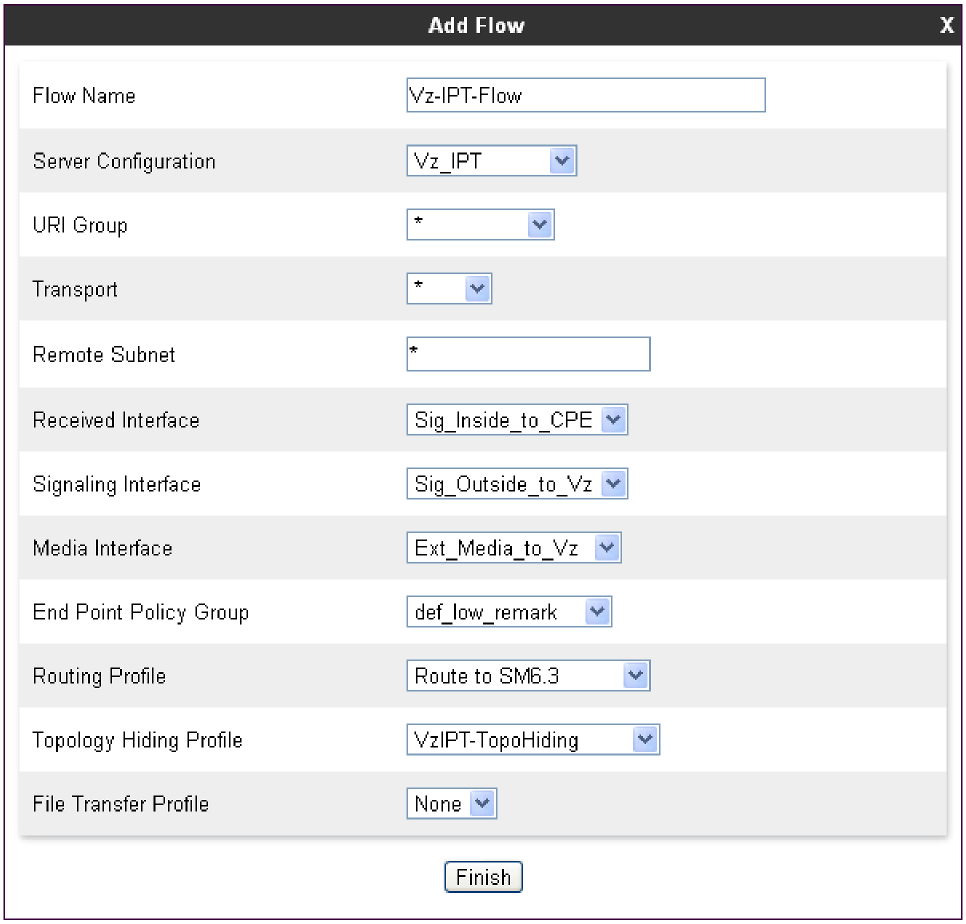
Create a Server Flow for Session Manager and the Verizon IP Trunk. To create a Server Flow, navigate to **Device Specific Settings → End Point Flows**. Select the **Server Flows** tab and click **Add Flow** as shown in below.



The following screen shows the flow named "Avaya SM6.3 Flow" used in the sample configuration. This flow uses the interfaces, policies, and profiles defined in previous sections. Click **Finish**.



The following screen shows the flow named "Vz-IPT-Flow" used in the sample configuration. This flow uses the interfaces, policies, and profiles defined in previous sections. Click **Finish**.



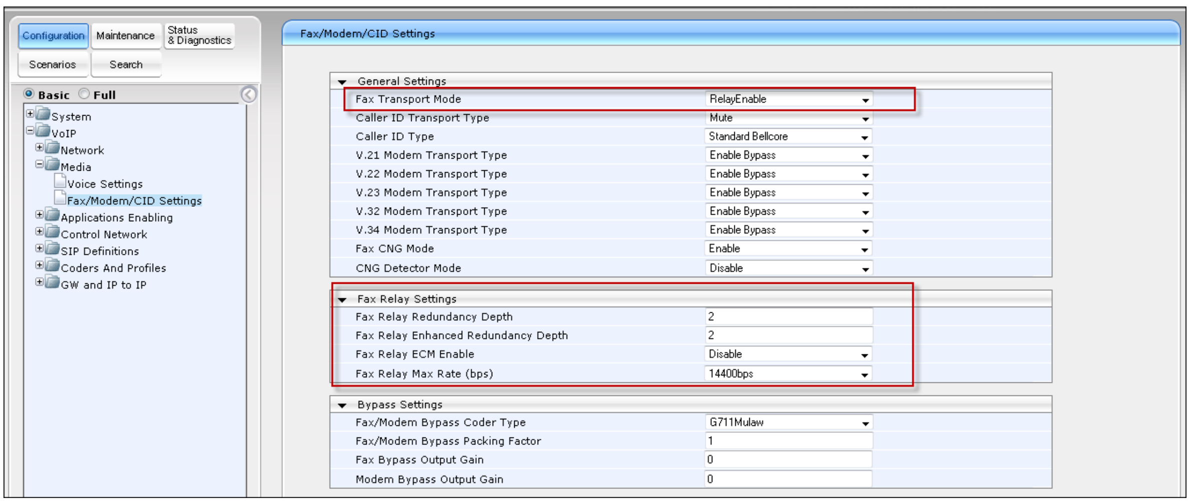
8. AudioCodes MP-114

During the verification these Application Notes, an AudioCodes MP-114 was used for fax calls to and from the PSTN. This section will show the necessary settings to incorporate fax calls with Verizon IP Trunk service and to register the MP-114 with Session Manager. These Application Notes assume that the installation of the AudioCodes MP-114 and the assignment of an IP address have already been completed. See **Section 12.3** for information regarding the installation of the AudioCodes MP-114.

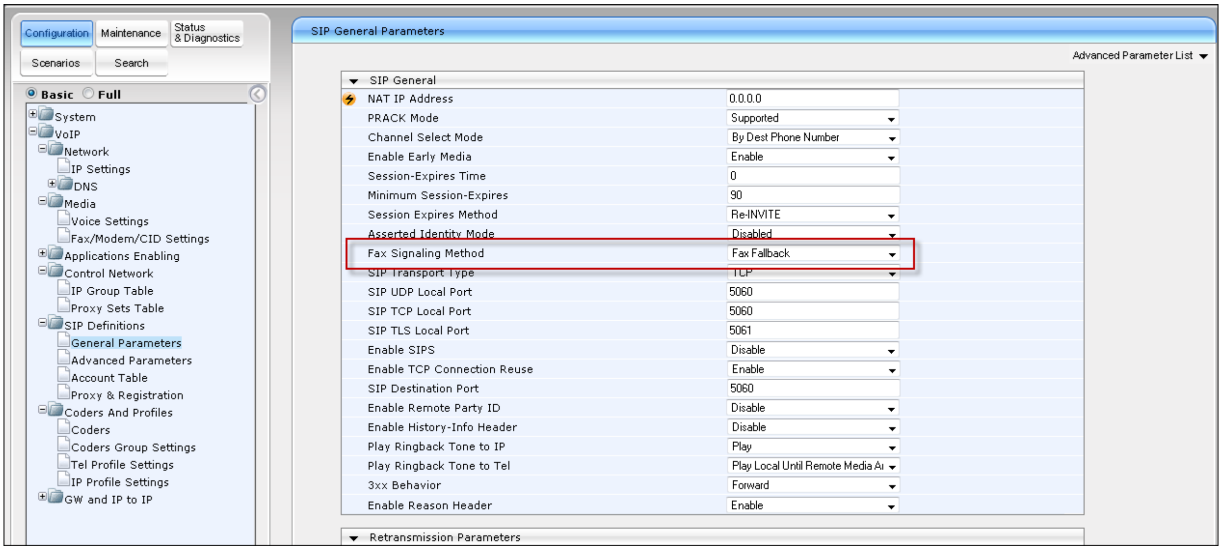
Note - Although the MP-114 is described in these Application Notes, other AudioCodes Telephone Adapters such as the MP-202 or MP-124 may be used.

8.1. Fax Configuration Settings

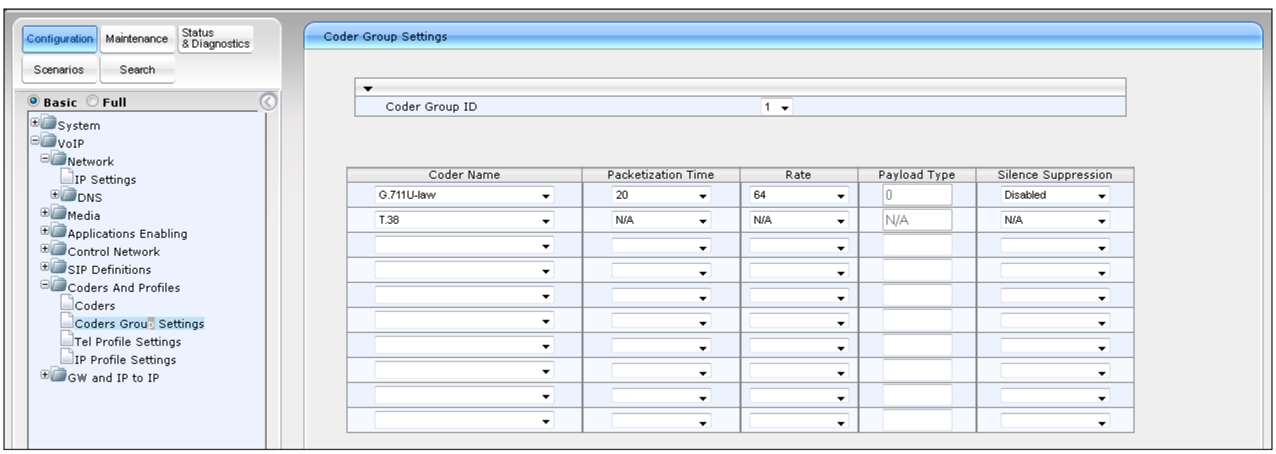
Select **Configuration** menu on the top left of the screen, and navigate to **VoIP→Media→Fax/Modem/CID Settings**. Set the **Fax Transport Mode** to "RelayEnable" and set the **Fax Relay Settings** as highlighted below.



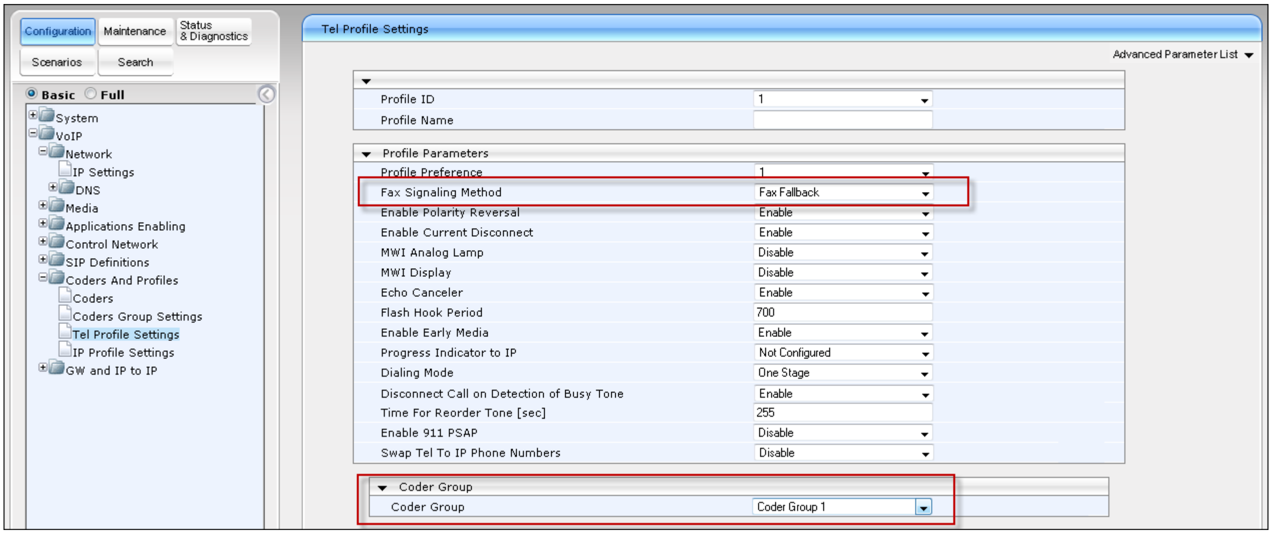
Navigate to **VoIP→SIP Definitions→General Parameters**. Set the **Fax Signaling Method** to "Fax Fallback".



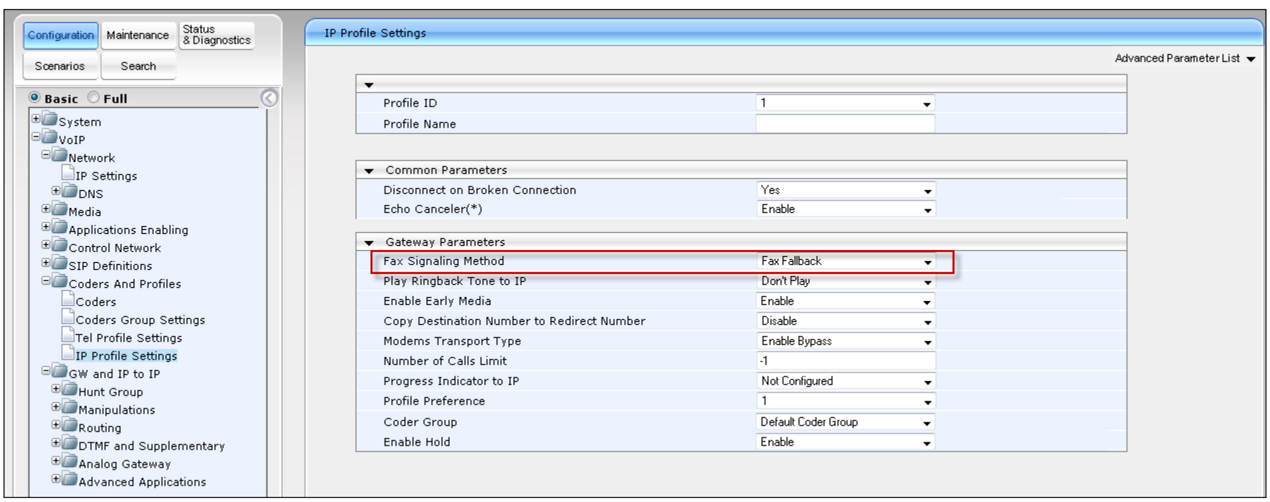
Navigate to **VoIP→Coders and Profiles→Coders Group Settings**. Select **Coder Group ID** "1" and under the **Coder Name** column, select "G.711U-Law" as the first choice and "T.38" as the second choice as shown below. This will allow calls to and from the fax to begin with G.711 as the first codec choice and re-Invite to T.38 when fax tones are detected.



Navigate to **VoIP→Coders and Profiles→Tel Profile Settings**. Select **Profile ID** “1” and set **Fax Signaling** Method to “Fax Fallback”. Select “Coder Group 1” for the **Coder Group**.

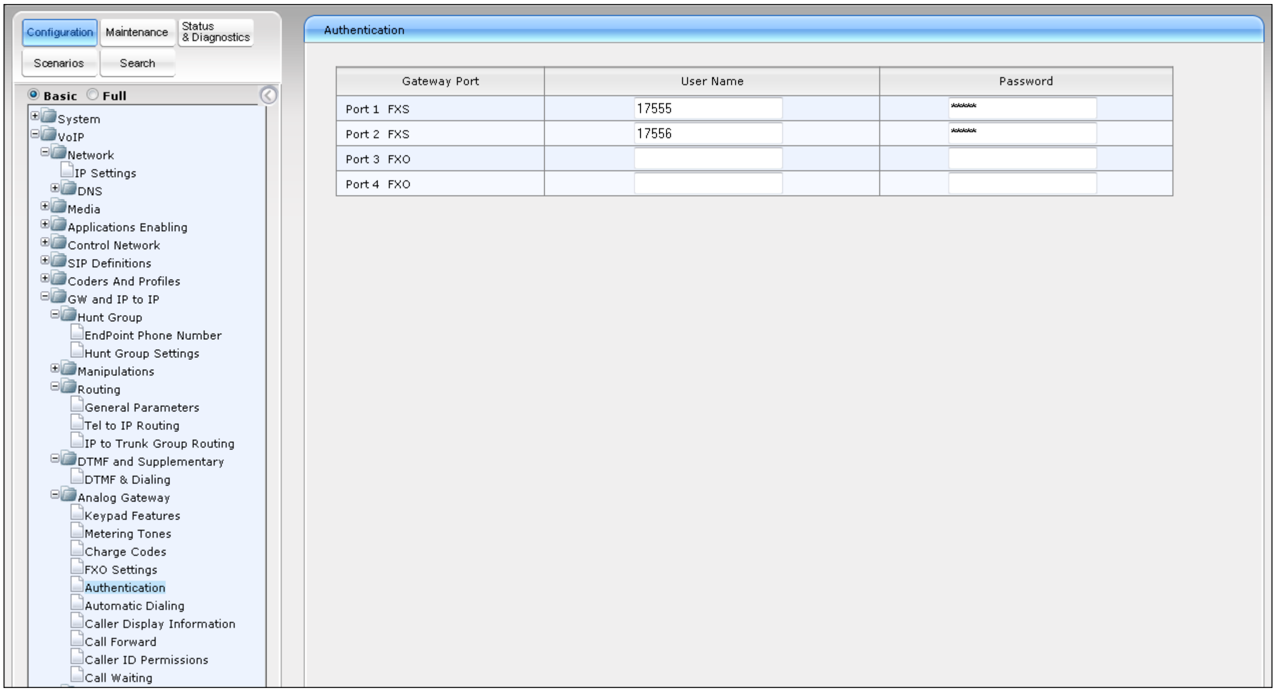


Navigate to **VoIP→Coders and Profiles→IP Profile Settings**. Select **Profile ID** “1” and set **Fax Signaling** Method to “Fax Fallback”.

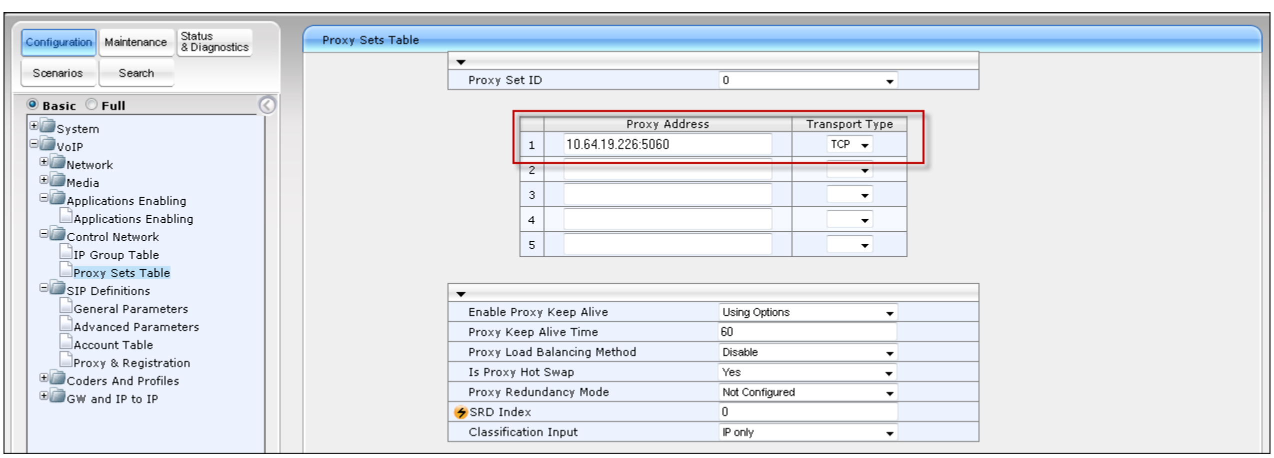


8.2. SIP Endpoint Registration and Proxy Settings

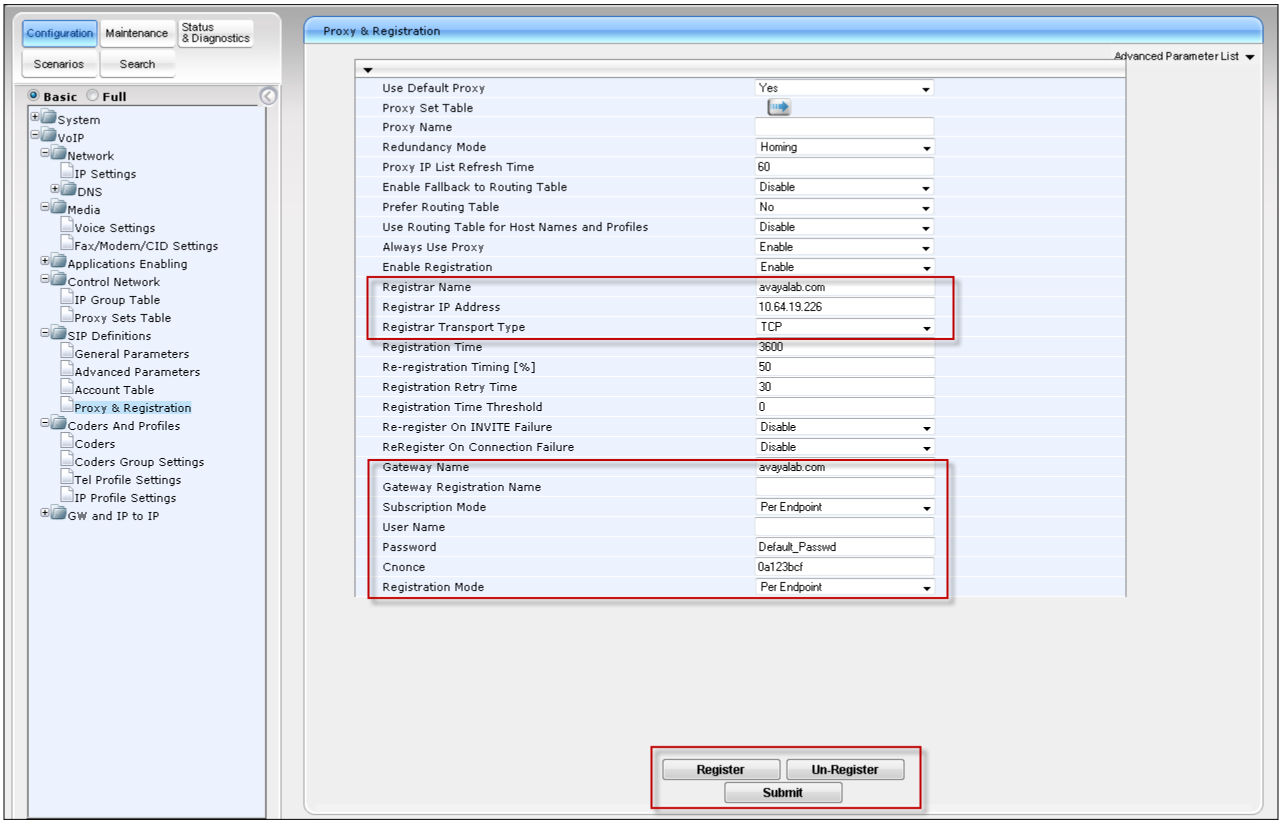
Navigate to **VoIP→Analog Gateway→Authentication**. Set the **User Name** and **Password** for each FXS port used for fax. The **User Name** corresponds to the **Avaya SIP Handle** of the SIP User created in System Manager and the **Password** corresponds to the **Communication Profile Password** as shown in **Section 6.9**.



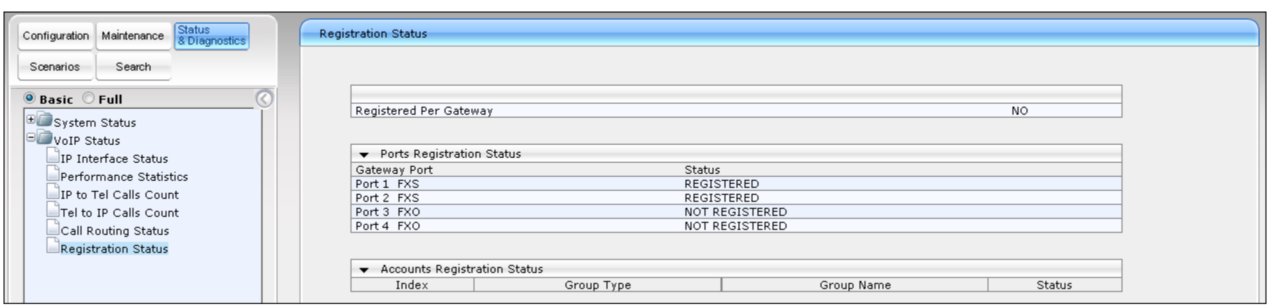
Navigate to **VoIP→Control Network→Proxy Sets Table**. Set the **Proxy Address** to the IP address and port used by Session Manager to listen for SIP REGSTER requests. In the sample configuration, this is "10.64.19.226:5060". Set the **Transport Type** to "TCP".



Navigate to **VoIP→SIP Definitions→Proxy & Registration**. Set the **Registrar Name** and **Gateway Name** to the domain name used by Session Manager as set in **Section 6.1**. Set the **Registrar IP Address** to Session Manager Security Module IP Address (10.64.19.226). Set the **Subscription Mode** and **Registration Mode** to "Per Endpoint" and verify the **Cnonce** setting. Click **Submit** and then **Register** on the bottom of the screen.

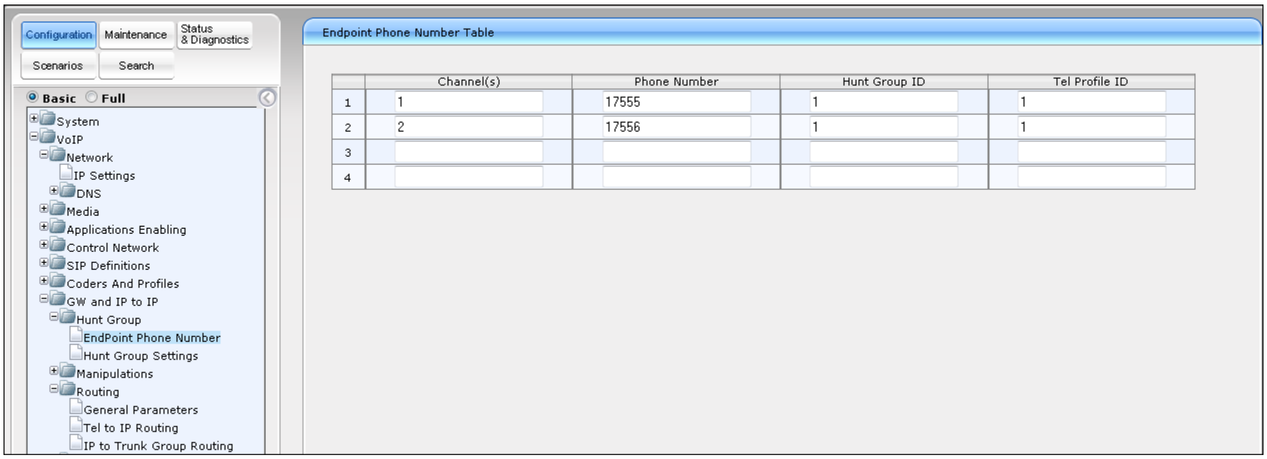


Select the **Status & Diagnostics** menu, and navigate to **VoIP Status→Registration Status**. At this point, the the **Gateway Port**(s) used for fax should show a **Status** of "REGISTERED".

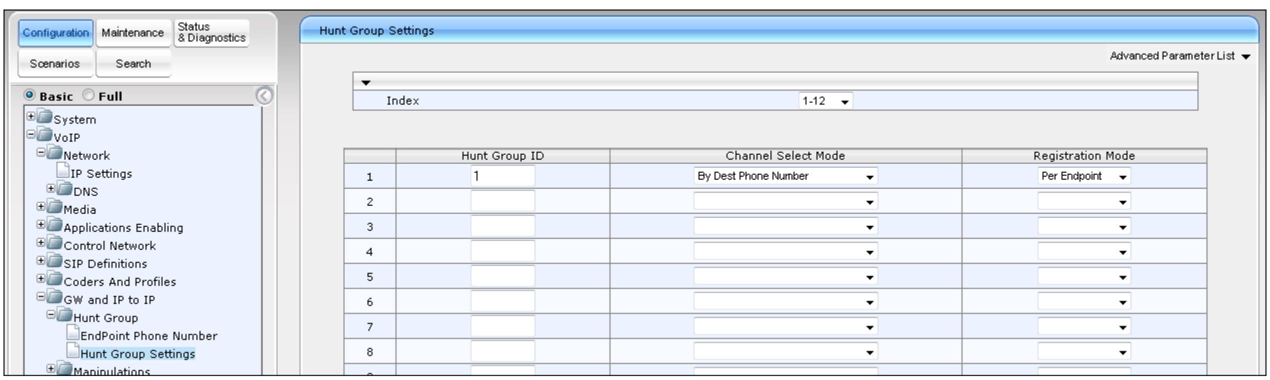


8.3. Routing

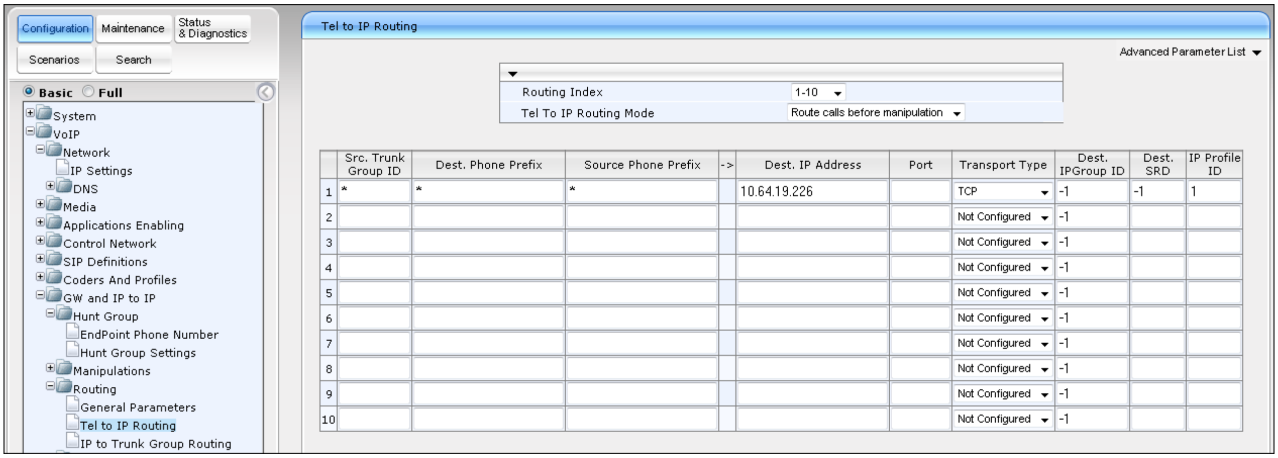
Select **Configuration** menu again on the top left of the screen, and navigate to **VoIP→GW and IP to IP→Hunt Group→EndPoint Phone Number**. Configure a Channel for each FXS port used for fax as shown below. Set the **Hunt Group ID** to “1”. Set the **Tel Profile ID** to the ID modified in **Section 8.1**.



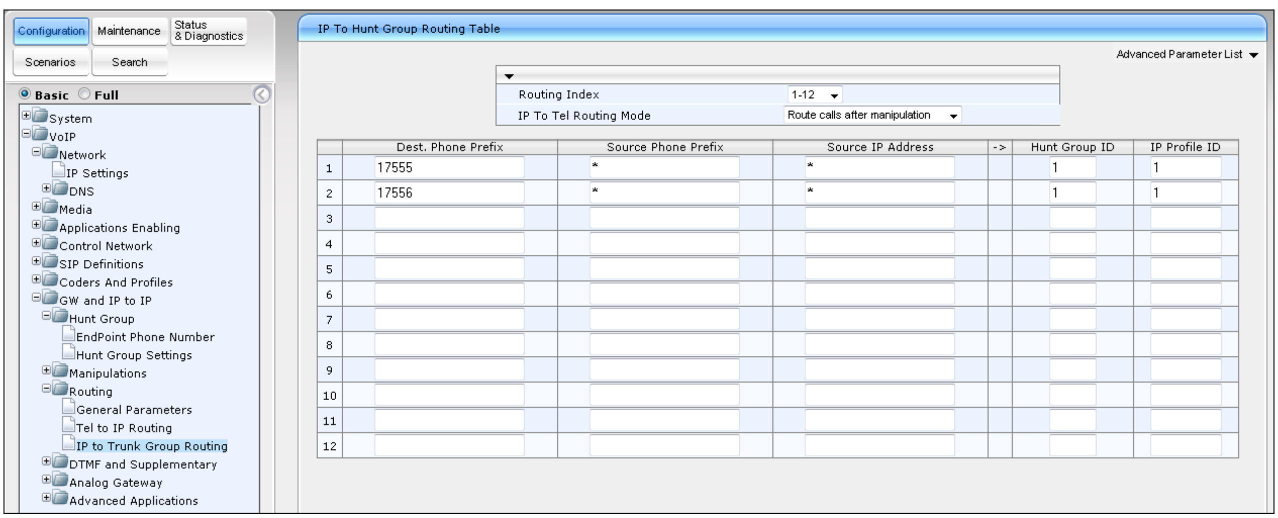
Navigate to **VoIP→GW and IP to IP→Hunt Group→Hunt Group Settings**. Configure **Hunt Group ID** “1” with **Channel Select Mode** set to “By Dest Phone Number” and **Registration Mode** set to “Per Endpoint”.



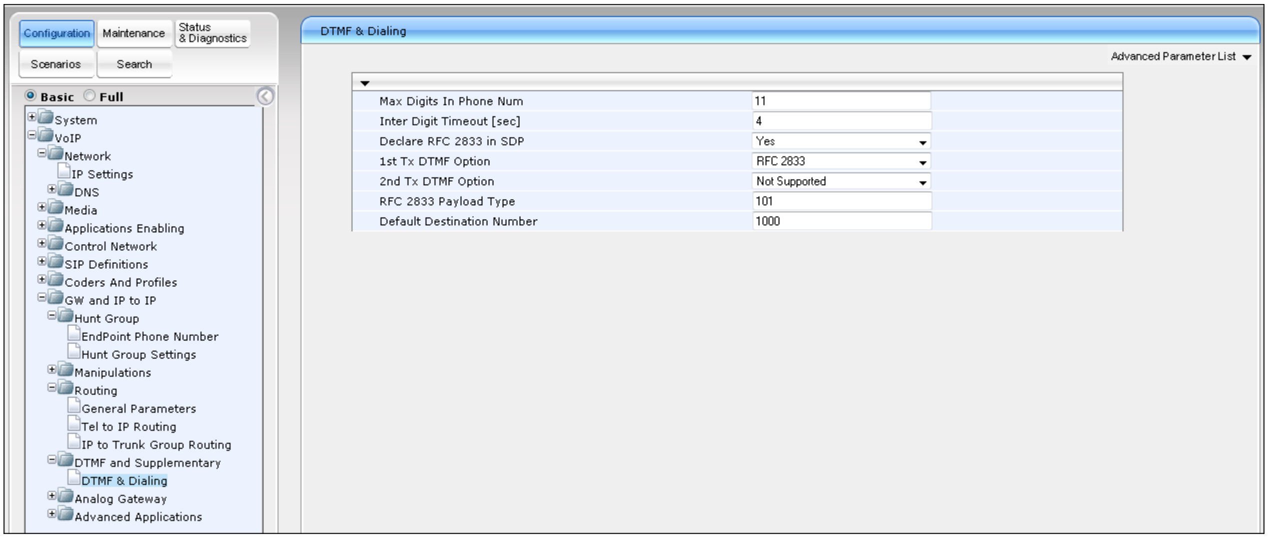
Navigate to **VoIP→GW and IP to IP→Routing→Tel to IP Routing**. Set the **Src. Trunk Group ID**, **Dest. Phone Prefix**, and **Source Phone Prefix** to "*". Set the **Dest. IP Address** to the Session Manager Security Module IP Address (10.64.19.226).



Navigate to **VoIP→GW and IP to IP→Routing→IP to Trunk Group Routing**. Set the **Dest. Phone Prefix** for each FXS port used for fax with the appropriate extension number as shown below. Set the **Source Phone Prefix** and **Source IP Address** to "*". Set the **Hunt Group ID** to "1" and **IP Profile ID** to "1" for each extension number.



Navigate to **VoIP→GW and IP to IP→DTMF and Supplementary→DTMF & Dialing**. Set the **Max Digits In Phone Num** to the maximum amount of digits the fax machine will use to dial a PSTN fax machine.



9. Verizon Business IP Trunk Services Suite Configuration

Information regarding the 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 is located in the Avaya Solutions 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.

9.1. Service Access Information

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

CPE (Avaya)	Verizon Network
adevc.avaya.globalipcom.com UDP port 5060	***pcelban0001.avayalincroft.globalipcom.com UDP Port 5071***

IP DID Numbers
732-945-0231
732-945-0232
732-945-0233
732-945-0234
732-945-0235
732-945-0236
732-945-0237
732-945-0238
732-945-0239

10. Verification Steps

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

10.1. Avaya Aura® Communication Manager Verifications

This section illustrates verifications from Communication Manager.

10.1.1 Example Incoming Call from PSTN via Verizon SIP Trunk

Incoming PSTN calls arrive from Verizon at Avaya SBCE, which sends the call to Session Manager. Session Manager sends the call to Communication Manager. 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 mapped the number received from Verizon to the extension of a Communication Manager telephone (x12002). Extension 12002 is an IP Telephone with IP address 10.64.19.109 in Region 1. The RTP media path is “ip-direct” from the IP Telephone (10.64.19.109) to the “inside” of the Avaya SBCE (10.64.19.140) in Region 2.

```
list trace tac *01                                                  Page   1
                                LIST TRACE
time            data

14:30:19 TRACE STARTED 03/26/2013 CM Release String cold-02.0.823.0-20396
14:30:26 SIP<INVITE sip:12002@avayalab.com SIP/2.0
14:30:26     Call-ID: BW203026076260313-1913181969@65.211.120.226
14:30:26     active trunk-group 1 member 249    cid 0x32d
14:30:26 SIP>SIP/2.0 180 Ringing
14:30:26     Call-ID: BW203026076260313-1913181969@65.211.120.226
14:30:26     dial 12002
14:30:26     ring station      12002 cid 0x32d
14:30:28 SIP>SIP/2.0 200 OK
14:30:28     Call-ID: BW203026076260313-1913181969@65.211.120.226
14:30:28     active station      12002 cid 0x32d
14:30:28     G729A ss:off ps:20
             rgn:1 [10.64.19.109]:3132
             rgn:2 [10.64.19.140]:35022
14:30:28     G729A ss:off ps:20
             rgn:2 [10.64.19.140]:35022
             rgn:1 [10.64.19.109]:3132
14:30:28 SIP<ACK sip:12002@10.64.19.155:5061;transport=tls SIP/2.0
14:30:28     Call-ID: BW203026076260313-1913181969@65.211.120.226
14:30:35 SIP>BYE sip:3035387006@10.64.19.140:5060;transport=tcp;gsid
14:30:35 SIP>=fded8570-9653-11e2-b83f-9c8e992b0a68 SIP/2.0
14:30:35     Call-ID: BW203026076260313-1913181969@65.211.120.226
14:30:35     idle station      12002 cid 0x32d
```

The following screen shows **Page 2** of the output of the ***status trunk*** command pertaining to this same call. Note the signaling using port 5061 between Communication Manager and Session Manager. Note the media is “ip-direct” from the IP Telephone (10.64.19.109) to the inside IP address of Avaya SBCE (10.64.19.140) using codec G.729a.

```
status trunk 1/249                                            Page   2 of   3
                                CALL CONTROL SIGNALING

Near-end Signaling Loc: PROCR
  Signaling   IP Address                                   Port
   Near-end:  10.64.19.155                               : 5061
    Far-end:  10.64.19.226                               : 5061
 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.729A
   Audio      IP Address                                   Port
   Near-end:  10.64.19.109                               : 3132
    Far-end:  10.64.19.140                               : 35024

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

The following screen shows **Page 3** of the output of the ***status trunk*** command pertaining to this same call. Here it can be observed that G.729a codec is used.

```
status trunk 1/249                                            Page   3 of   3
                      SRC PORT TO DEST PORT TALKPATH
src port: T00249
T00249:TX:10.64.19.140:35024/g729a/20ms
S00025:RX:10.64.19.109:3132/g729a/20ms
```

10.1.2 Example Outgoing Calls to PSTN via Verizon IP Trunk

The following edited trace shows an outbound ARS call from IP Telephone x12002 to the PSTN number 9-1-303-538-7024. The call is routed to route pattern 1 and trunk group 1. The call initially uses the G450 gateway (10.64.19.81), but after the call is answered, the call is "shuffled" to become an "ip-direct" connection between the IP Telephone (10.64.19.109) and the "inside" of the Avaya SBCE (10.64.19.140).

```
list trace tac *01                                                   Page   1
                                LIST TRACE
time            data

14:40:29 TRACE STARTED 03/26/2013 CM Release String cold-02.0.823.0-20396
14:40:34     dial 913035387024 route:PREFIX|FNPA|ARS
14:40:34     route-pattern  1 preference 1 location 1/ALL  cid 0x330
14:40:34     seize trunk-group 1 member 20    cid 0x330
14:40:34     Calling Number & Name 12002 test IP
14:40:34 SIP>INVITE sip:3035387024@avayalab.com SIP/2.0
14:40:34     Call-ID: 070bf25995e2188225156b4a00
14:40:34     Setup digits 13035387024
14:40:34     Calling Number & Name 12002 test IP
14:40:34 SIP<SIP/2.0 100 Trying
14:40:34     Call-ID: 070bf25995e2188225156b4a00
14:40:34     Proceed trunk-group 1 member 20    cid 0x330
14:40:37 SIP<SIP/2.0 183 Session Progress
14:40:37     Call-ID: 070bf25995e2188225156b4a00
14:40:37     G729 ss:off ps:20
             rgn:2 [10.64.19.140]:35026
             rgn:1 [10.64.19.81]:2052
14:40:37     xoip options: fax:T38 modem:off tty:US  uid:0x5000c
             xoip ip: [10.64.19.81]:2052
14:40:39 SIP<SIP/2.0 200 OK
14:40:39     Call-ID: 070bf25995e2188225156b4a00
14:40:39 SIP>ACK sip:3035387024@10.64.19.140:5060;transport=tcp;gsi
14:40:39 SIP>d=68c5b470-9655-11e2-b83f-9c8e992b0a68 SIP/2.0
14:40:39     Call-ID: 070bf25995e2188225156b4a00
14:40:39     active trunk-group 1 member 20    cid 0x330
14:40:39 SIP>INVITE sip:13035387024@10.64.19.140:5060;transport=tcp;
14:40:39 SIP>gsid=68c5b470-9655-11e2-b83f-9c8e992b0a68 SIP/2.0
14:40:39     Call-ID: 070bf25995e2188225156b4a00
14:40:39 SIP<SIP/2.0 100 Trying
14:40:39     Call-ID: 070bf25995e2188225156b4a00
14:40:39 SIP<SIP/2.0 200 OK
14:40:39     Call-ID: 070bf25995e2188225156b4a00
14:40:39     G729 ss:off ps:20
             rgn:1 [10.64.19.109]:3132
             rgn:2 [10.64.19.140]:35026
14:40:39 SIP>ACK sip:3035387024@10.64.19.140:5060;transport=tcp;gsi
14:40:39 SIP>d=68c5b470-9655-11e2-b83f-9c8e992b0a68 SIP/2.0
14:40:39     Call-ID: 070bf25995e2188225156b4a00
14:40:39     G729A ss:off ps:20
             rgn:2 [10.64.19.140]:35026
             rgn:1 [10.64.19.109]:3132
14:41:16 SIP<BYE sip:12002@10.64.19.155:5061;transport=tls SIP/2.0
14:41:16     Call-ID: 070bf25995e2188225156b4a00
14:41:16 SIP>SIP/2.0 200 OK
14:41:16     Call-ID: 070bf25995e2188225156b4a00
14:41:16     idle trunk-group 1 member 20    cid 0x330
```

10.2. 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.

10.2.1 Verify SIP Entity Link Status

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

Home / Elements / Session Manager / System Status / SIP Entity Monitoring

Help ?

SIP Entity Link Monitoring Status Summary

This page provides a summary of Session Manager SIP entity link monitoring status.

SIP Entities Status for All Monitoring Session Manager Instances

Run Monitor

1 Items | Refresh Filter: Enable

	Session Manager	Type	Monitored Entities					
			Down	Partially Up	Up	Not Monitored	Deny	Total
☐	ASM	Core	0	0	5	0	0	5

All Monitored SIP Entities

Run Monitor

5 Items (1 Selected) | Refresh Filter: Enable

	SIP Entity Name
☐	Loc19-CM-TG1
☐	Loc19-CM Messaging
☐	CS1K
☑	Vz_ASBCE-1
☐	Vz_ASBCE-2

From the list of monitored entities, select an entity of interest, such as "Vz_ASBCE-1". Under normal operating conditions, the **Link Status** should be "UP" as shown in the example screen below.

All Entity Links to SIP Entity: Vz_ASBCE-1

Summary View

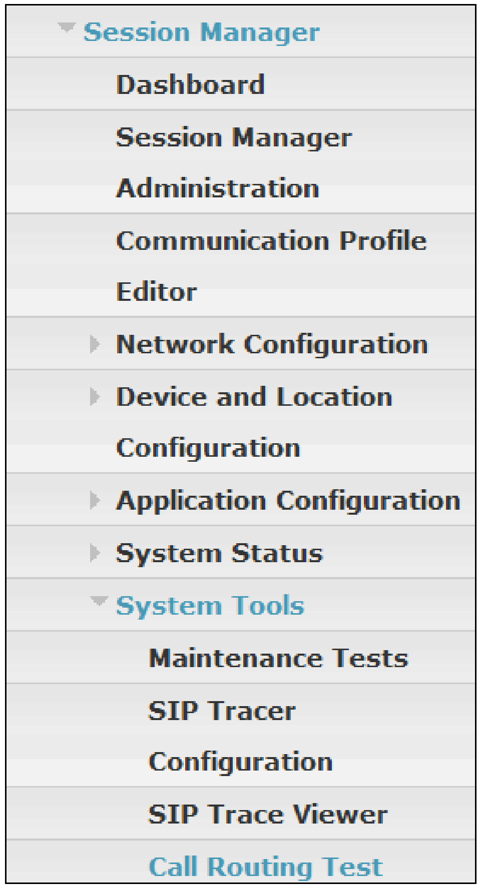
Status Details for the selected Session Manager:

1 Items | Refresh Filter: Enable

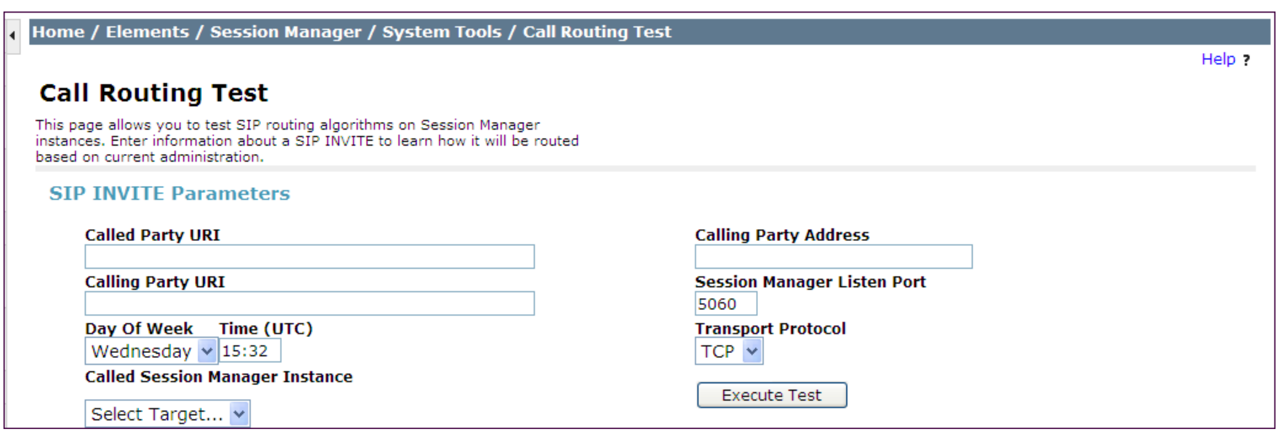
	Session Manager Na	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
○	ASM	10.64.19.140	5060	TCP	FALSE	UP	200 OK	UP

10.2.2 Call Routing Test

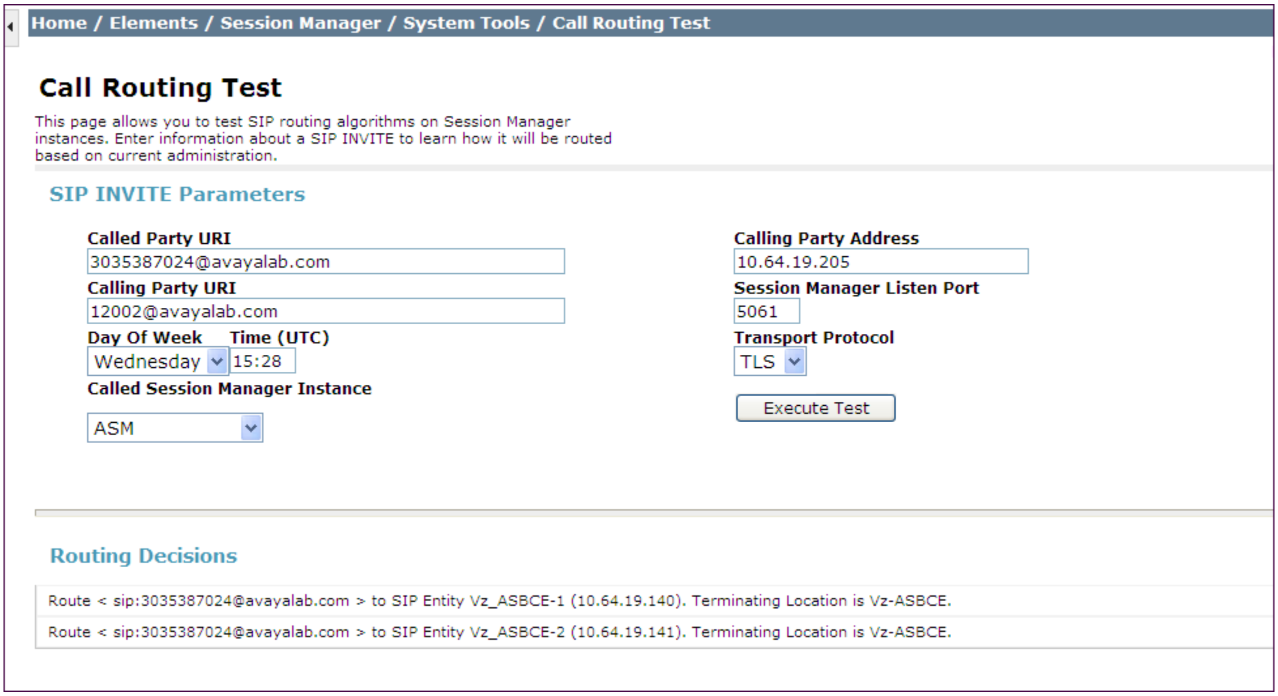
The **Call Routing Test** verifies the routing for a particular source and destination. To run the routing test, expand **Elements** → **Session Manager** → **System Tools** → **Call Routing Test**, as shown below.



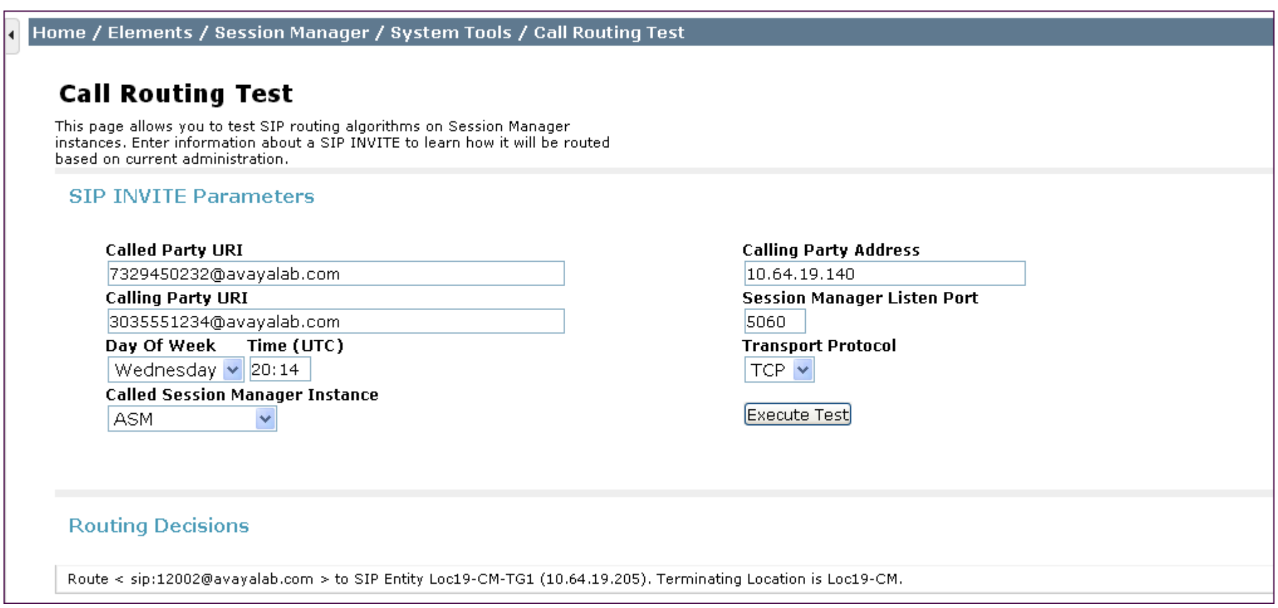
A screen such as the following is displayed.



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).



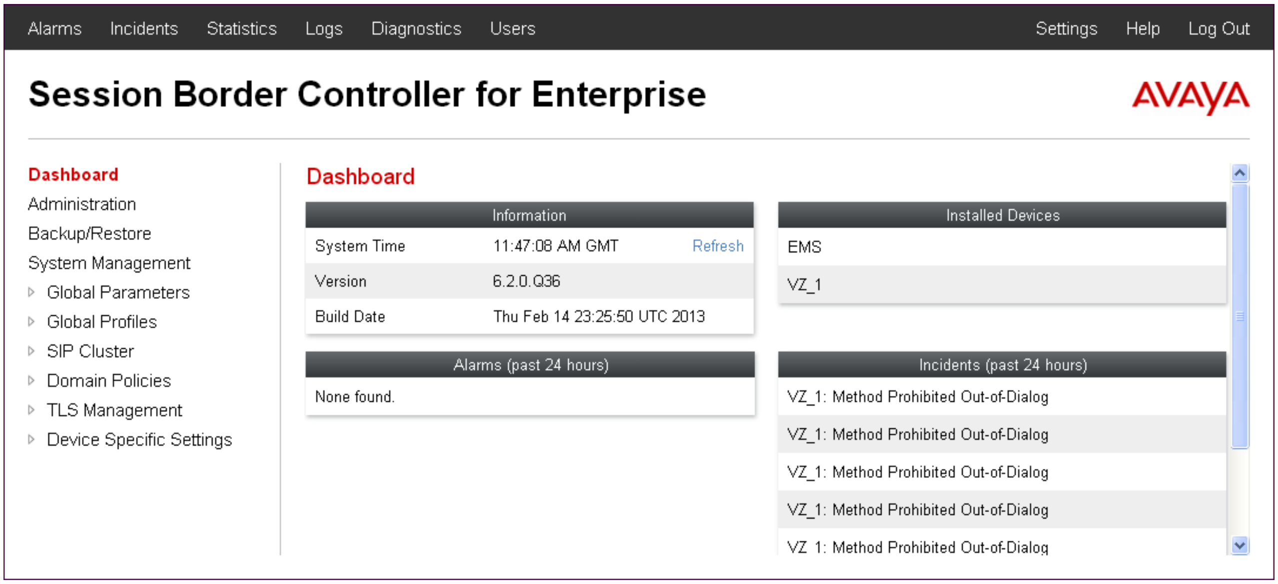
Another example shows an inbound call to one of Verizon assigned DID numbers. Observe that the DID number 732-945-0232 has been converted to Communication Manager extension 12002 under **Routing Decisions** and will be routed to Communication Manager.



10.3. Avaya Session Border Controller for Enterprise Verification

10.3.1 Welcome Screen

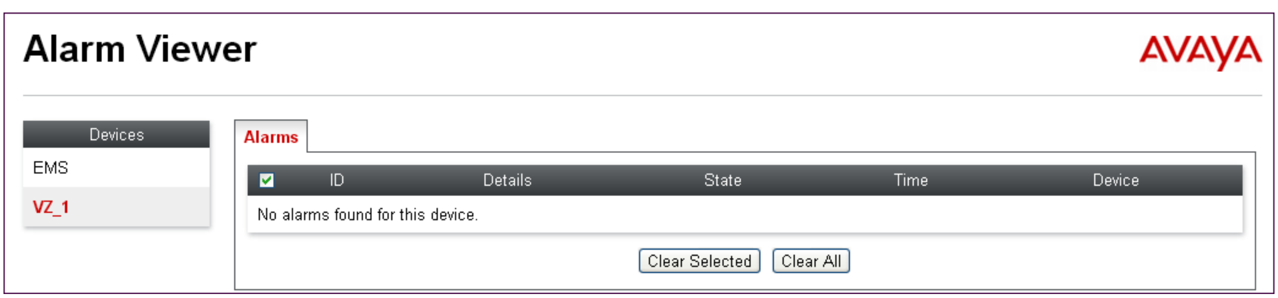
The welcome screen shows alarms, incidents, and the status of all managed Avaya SBCEs at a glance.



10.3.2 Alarms

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

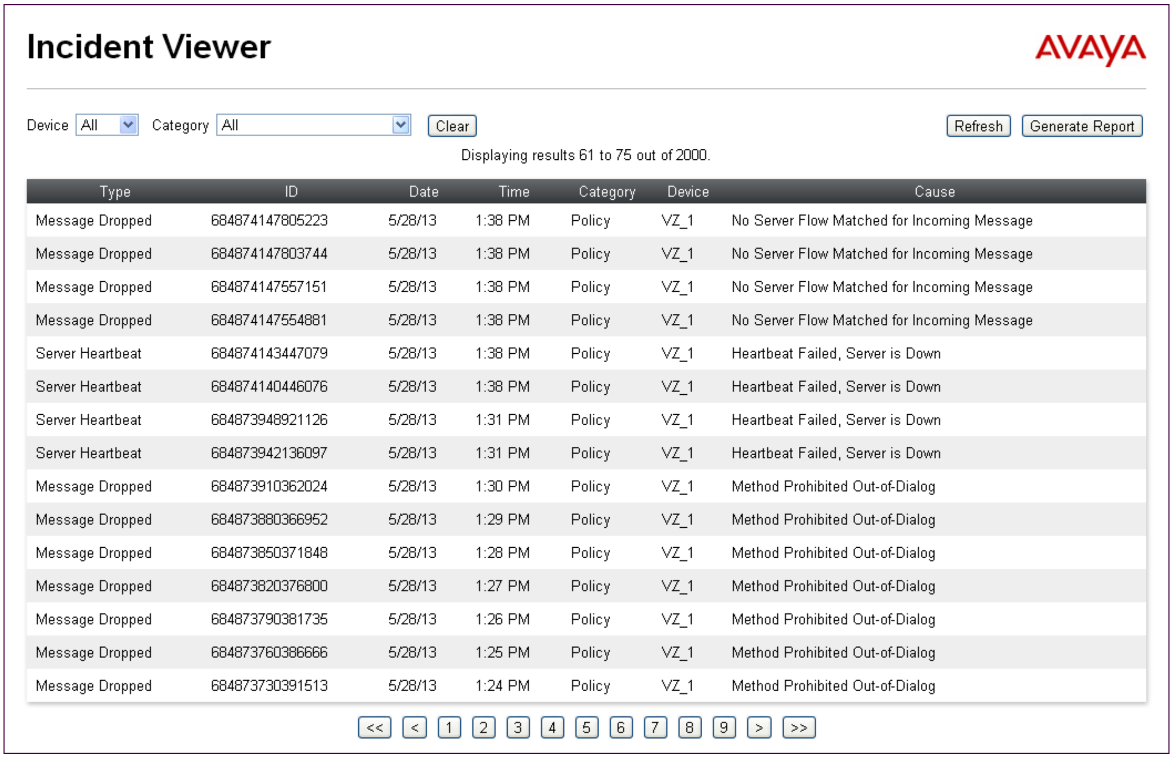
Alarm Viewer.



10.3.3 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 Viewer

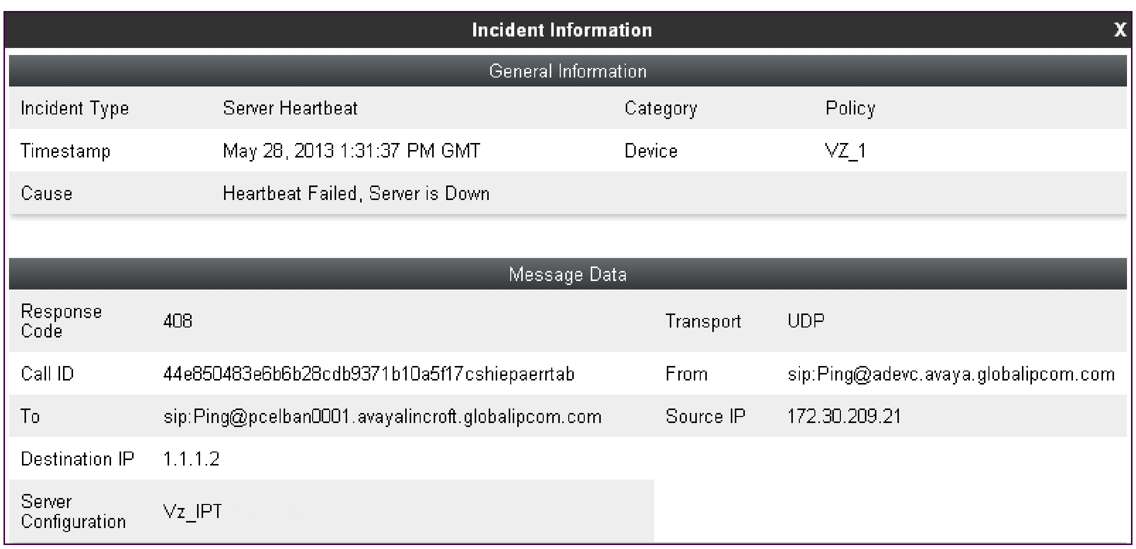
Device All Category All Clear Refresh Generate Report

Displaying results 61 to 75 out of 2000.

Type	ID	Date	Time	Category	Device	Cause
Message Dropped	684874147805223	5/28/13	1:38 PM	Policy	VZ_1	No Server Flow Matched for Incoming Message
Message Dropped	684874147803744	5/28/13	1:38 PM	Policy	VZ_1	No Server Flow Matched for Incoming Message
Message Dropped	684874147557151	5/28/13	1:38 PM	Policy	VZ_1	No Server Flow Matched for Incoming Message
Message Dropped	684874147554881	5/28/13	1:38 PM	Policy	VZ_1	No Server Flow Matched for Incoming Message
Server Heartbeat	684874143447079	5/28/13	1:38 PM	Policy	VZ_1	Heartbeat Failed, Server is Down
Server Heartbeat	684874140446076	5/28/13	1:38 PM	Policy	VZ_1	Heartbeat Failed, Server is Down
Server Heartbeat	684873948921126	5/28/13	1:31 PM	Policy	VZ_1	Heartbeat Failed, Server is Down
Server Heartbeat	684873942136097	5/28/13	1:31 PM	Policy	VZ_1	Heartbeat Failed, Server is Down
Message Dropped	684873910362024	5/28/13	1:30 PM	Policy	VZ_1	Method Prohibited Out-of-Dialog
Message Dropped	684873880366952	5/28/13	1:29 PM	Policy	VZ_1	Method Prohibited Out-of-Dialog
Message Dropped	684873850371848	5/28/13	1:28 PM	Policy	VZ_1	Method Prohibited Out-of-Dialog
Message Dropped	684873820376800	5/28/13	1:27 PM	Policy	VZ_1	Method Prohibited Out-of-Dialog
Message Dropped	684873790381735	5/28/13	1:26 PM	Policy	VZ_1	Method Prohibited Out-of-Dialog
Message Dropped	684873760386666	5/28/13	1:25 PM	Policy	VZ_1	Method Prohibited Out-of-Dialog
Message Dropped	684873730391513	5/28/13	1:24 PM	Policy	VZ_1	Method Prohibited Out-of-Dialog

<< < 1 2 3 4 5 6 7 8 9 > >>

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



Incident Information

General Information

Incident Type	Server Heartbeat	Category	Policy
Timestamp	May 28, 2013 1:31:37 PM GMT	Device	VZ_1
Cause	Heartbeat Failed, Server is Down		

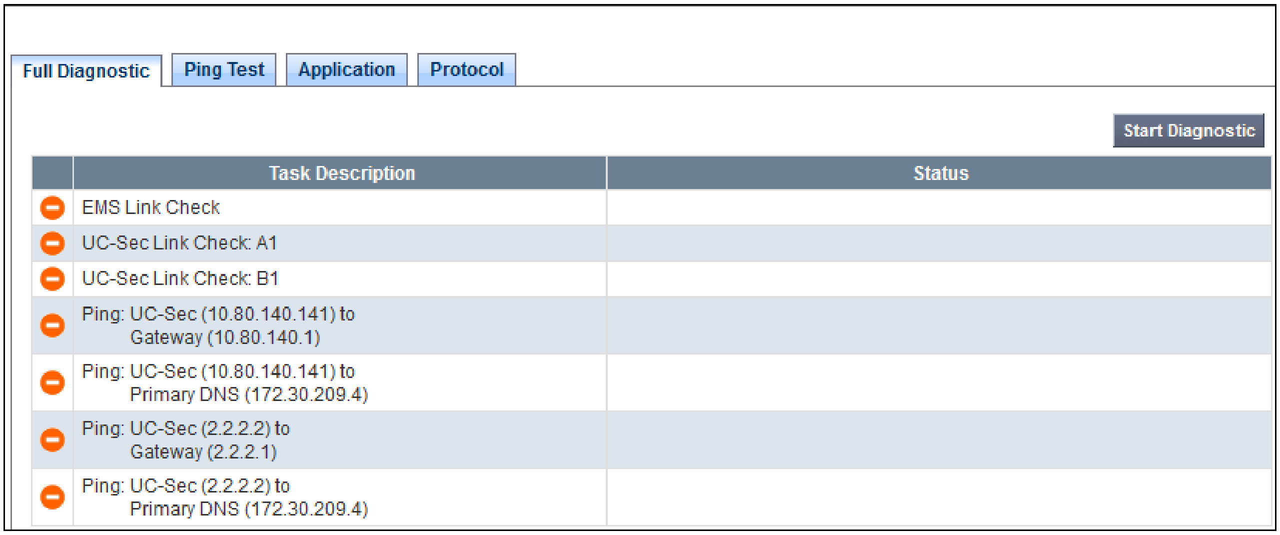
Message Data

Response Code	408	Transport	UDP
Call ID	44e850483e6b6b28cdb9371b10a5f17cshiepaerrtab	From	sip:Ping@adevc.avaya.globalipcom.com
To	sip:Ping@pcelban0001.avayalincroft.globalipcom.com	Source IP	172.30.209.21
Destination IP	1.1.1.2		
Server Configuration	Vz_IPT		

10.3.4 Diagnostics

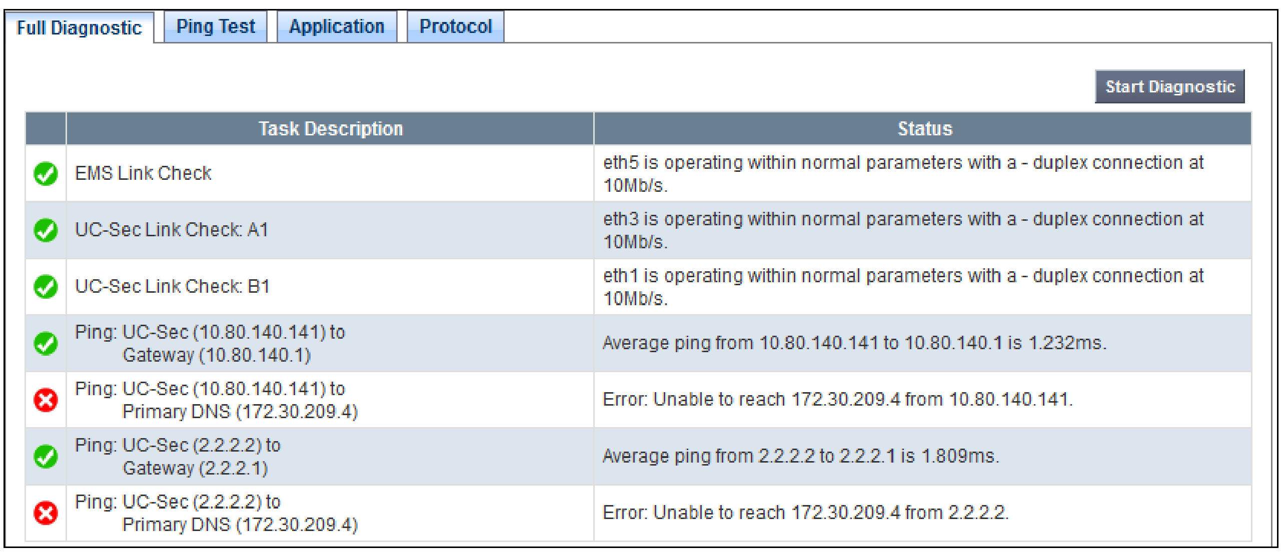
The full diagnostics check that can be run can run line checks in both directions.

Click on **Diagnostics** on the top bar, select the Avaya SBCE from the list of devices and then click "Start Diagnostics".



	Task Description	Status
	EMS Link Check	
	UC-Sec Link Check: A1	
	UC-Sec Link Check: B1	
	Ping: UC-Sec (10.80.140.141) to Gateway (10.80.140.1)	
	Ping: UC-Sec (10.80.140.141) to Primary DNS (172.30.209.4)	
	Ping: UC-Sec (2.2.2.2) to Gateway (2.2.2.1)	
	Ping: UC-Sec (2.2.2.2) to Primary DNS (172.30.209.4)	

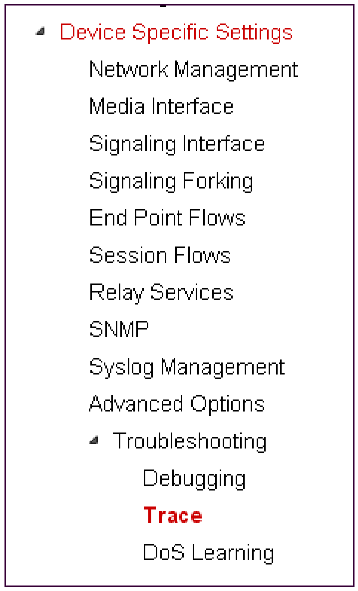
A green check mark or a red x will indicate success or failure.



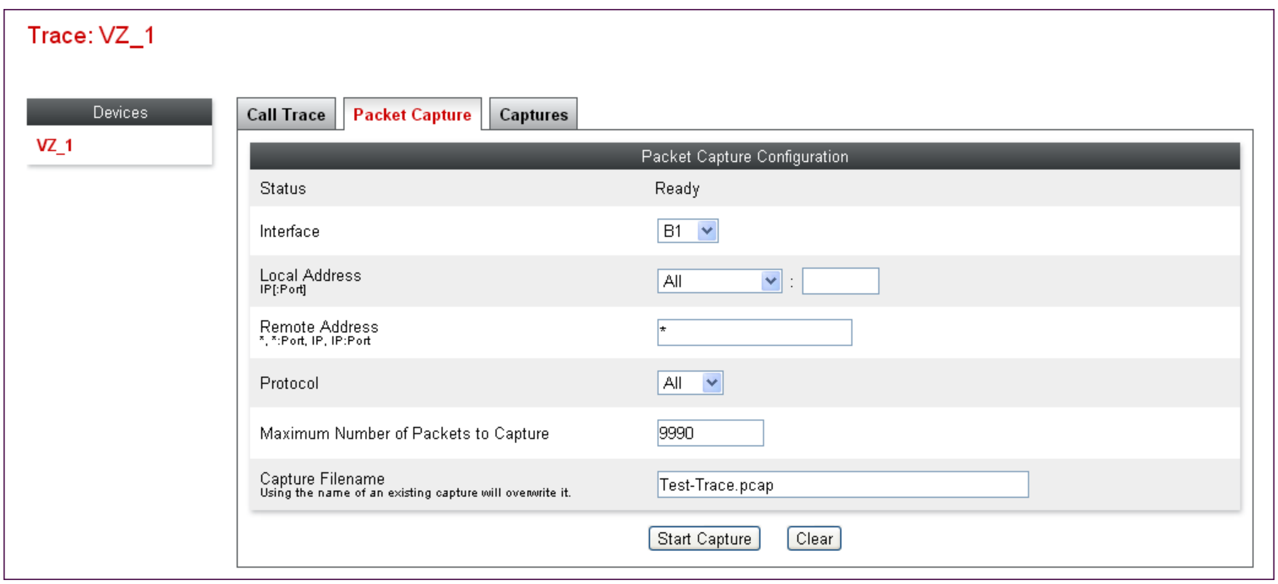
	Task Description	Status
	EMS Link Check	eth5 is operating within normal parameters with a - duplex connection at 10Mb/s.
	UC-Sec Link Check: A1	eth3 is operating within normal parameters with a - duplex connection at 10Mb/s.
	UC-Sec Link Check: B1	eth1 is operating within normal parameters with a - duplex connection at 10Mb/s.
	Ping: UC-Sec (10.80.140.141) to Gateway (10.80.140.1)	Average ping from 10.80.140.141 to 10.80.140.1 is 1.232ms.
	Ping: UC-Sec (10.80.140.141) to Primary DNS (172.30.209.4)	Error: Unable to reach 172.30.209.4 from 10.80.140.141.
	Ping: UC-Sec (2.2.2.2) to Gateway (2.2.2.1)	Average ping from 2.2.2.2 to 2.2.2.1 is 1.809ms.
	Ping: UC-Sec (2.2.2.2) to Primary DNS (172.30.209.4)	Error: Unable to reach 172.30.209.4 from 2.2.2.2.

10.3.5 Tracing

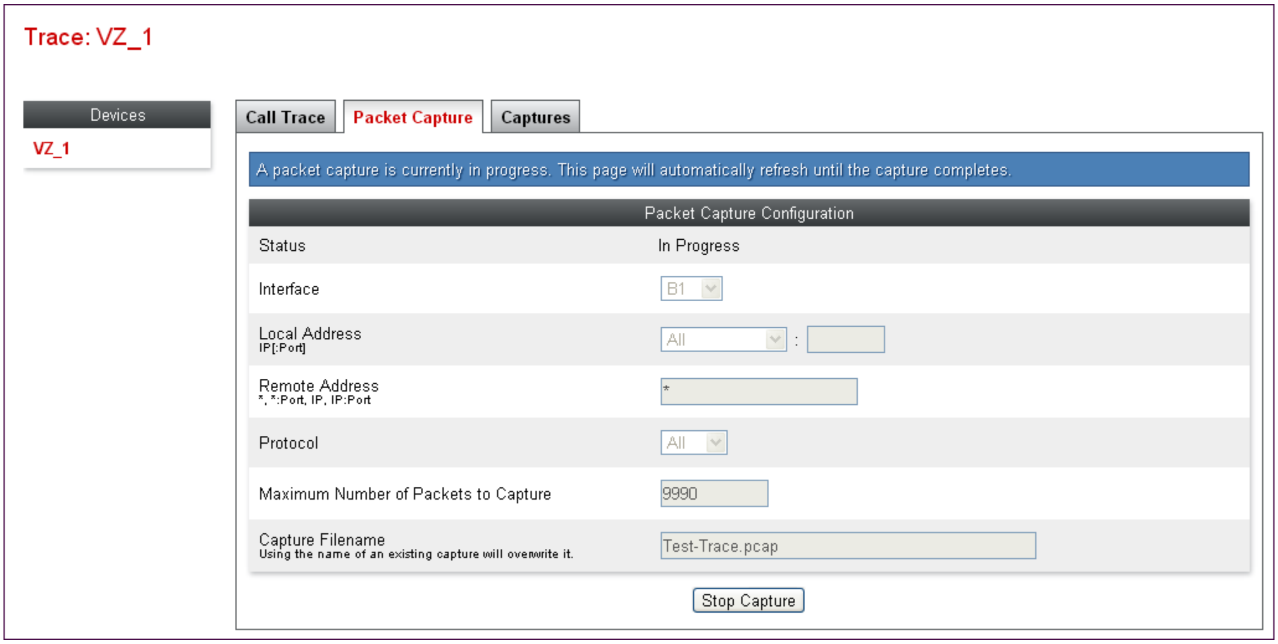
To take a call trace, Select **Device Specific Settings → Troubleshooting → Tracing** from the left-side menu as shown below.



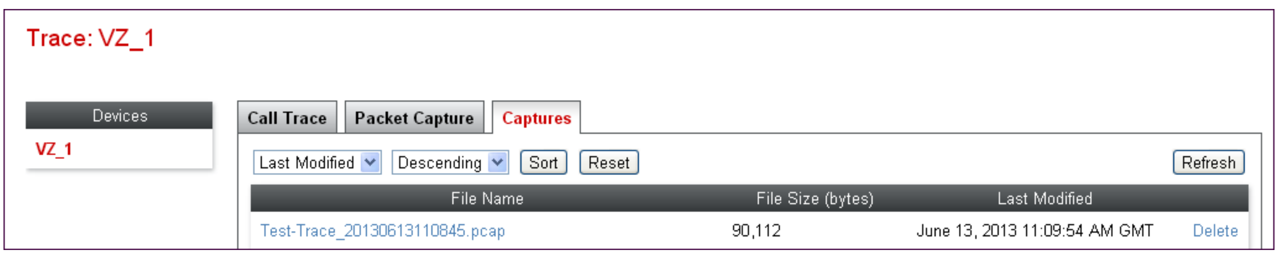
Select the **Packet Capture** tab and set the desired configuration for a call trace and click **Start Capture**.



When tracing has reached the desired number of packets the trace will stop automatically, or alternatively, click the **Stop Capture** button at the bottom.



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



11. Conclusion

As illustrated in these Application Notes, Avaya Aura® Communication Manager 6.3, Avaya Aura® Session Manager 6.3, and 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 users access to the PSTN using a Verizon Business IP Trunk public SIP trunk service connection.

12. Additional References

12.1. Avaya

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

[1] *Implementing Avaya Aura® Communication Manager*, Doc ID 03-603558, Release 6.3
[2] *Administering Avaya Aura® Communication Manager*, Doc ID 03-300509, Release 6.3
[3] *Implementing Avaya Aura® Session Manager*, Release 6.3
[4] *Installing Service Packs for Avaya Aura® Session Manager*, Release 6.3
[5] *Upgrading Avaya Aura® Session Manager*, Release 6.3
[6] *Maintaining and Troubleshooting Avaya Aura® Session Manager*, Release 6.3
[7] *Implementing Avaya Aura® System Manager*, Release 6.3
[8] *Installing Avaya Session Border Controller for Enterprise*, Release 6.2
[9] *Administering Avaya Session Border Controller for Enterprise*, Release 6.2

Avaya Application Notes, including the following, are also available at http://support.avaya.com

The following Application Notes cover Session Manager 6.2 with Verizon IP SIP Trunk Service using the Avaya Session Border Controller for Enterprise.

[MO-VZIPT-SM62] Application Notes for Avaya Aura® Communication Manager 6.2, Avaya Aura® Session Manager 6.2, and Avaya Session Border Controller for Enterprise with Verizon Business IP Trunk SIP Trunk Service – Issue 1.0
http://downloads.avaya.com/css/P8/documents/100162132

The following Application Notes cover Session Manager 6.1 with Verizon IP SIP Trunk Service using the Avaya Session Border Controller for Enterprise.

[MO-VZIPT-SM61] Application Notes for Avaya Aura® Communication Manager 6.0.1, Avaya Aura® Session Manager 6.1, and Avaya Session Border Controller for Enterprise with Verizon Business IP Trunk SIP Trunk Service – Issue 1.0
http://downloads.avaya.com/css/P8/documents/100164354

12.2. Verizon Business

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

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

12.3. AudioCodes

The following document is available at http://audiocodes.com

- *Verizon T.38 FAX Configuration Guide for AudioCodes MP-11x*

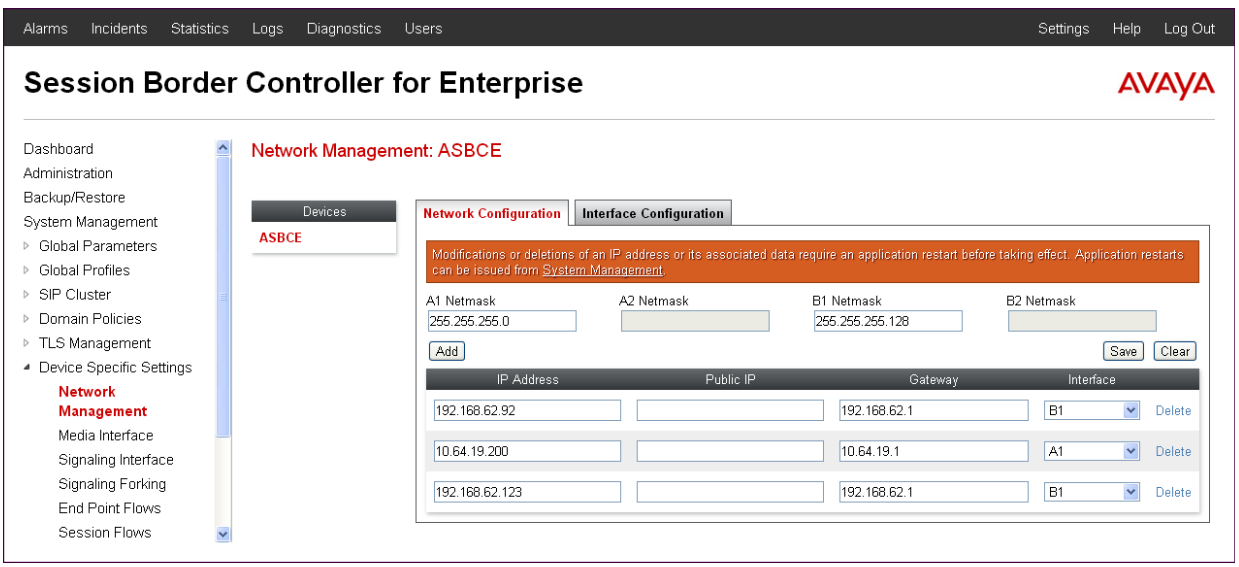
Appendix A

This section covers the configuration settings of Avaya SBCE, Session Manager, and a sample endpoint as used for Remote Workers during compliance testing. In the test environment, a dedicated Avaya SBCE with private IP addresses was used to access the Verizon Business Private IP (PIP) IP Trunk service. To allow remote SIP endpoints access to the test environment through a public network, a separate Avaya SBCE with public IP addresses was utilized. The settings presented here simply illustrate the sample configuration used during compliance testing with Verizon IP Trunk service, and are not intended to be prescriptive. Other routing criteria and policies may be appropriate based on different customer needs.

Standard and Advanced Session Licenses are required for Remote Worker. Contact an authorized Avaya representative for assistance if additional licensing is required.

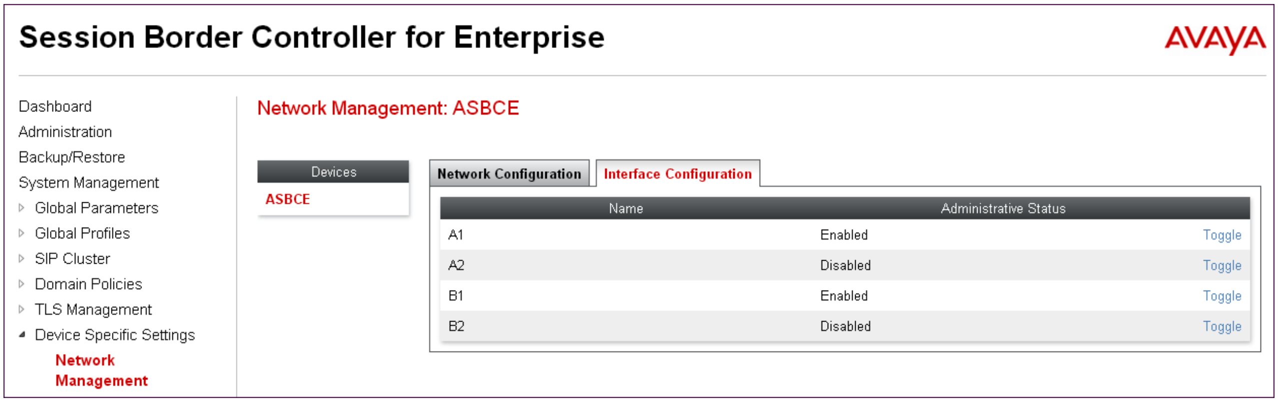
The following screen shows the **Network Management** screen of the Avaya SBCE. The internal interface is assigned to **A1** and the external interfaces are assigned to **B1**. Avaya SIP endpoints registered to IP address “192.168.62.92” and retrieved firmware and configuration data from IP address “192.168.63.123”. For security reasons, any actual public IP addresses used in the configuration have been replaced with private IP addresses.

Note – A SIP Entity in Session Manager was *not* configured for the Avaya SBCE’s internal IP address used for Remote Worker. This keeps the interface untrusted in Session Manager, thereby allowing Session Manager to properly challenge user registration requests.

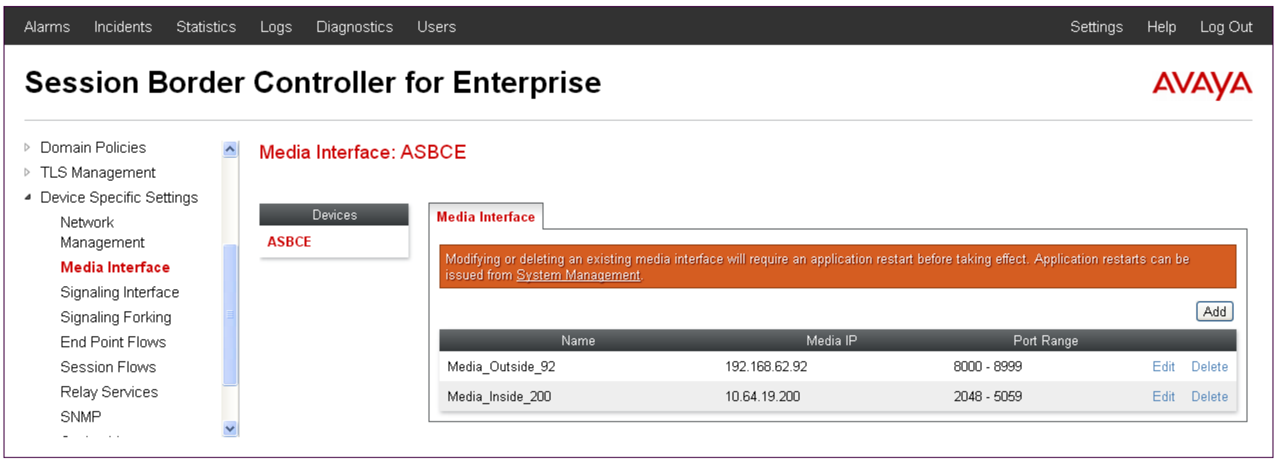


It is possible to deploy Remote Worker using the same Avaya SBCE as the one used for SIP trunking. However, separate IP addresses are needed for each function. This allows the Avaya SBCE to enforce proper security policies as if it were two different Avaya SBCEs. Only two network interfaces on the Avaya SBCE may be active at one time, so this requires all external IP addresses to be on the same subnet so they may be applied to the same network interface. Similarly, additional internal IP addresses must be on the same internal subnet.

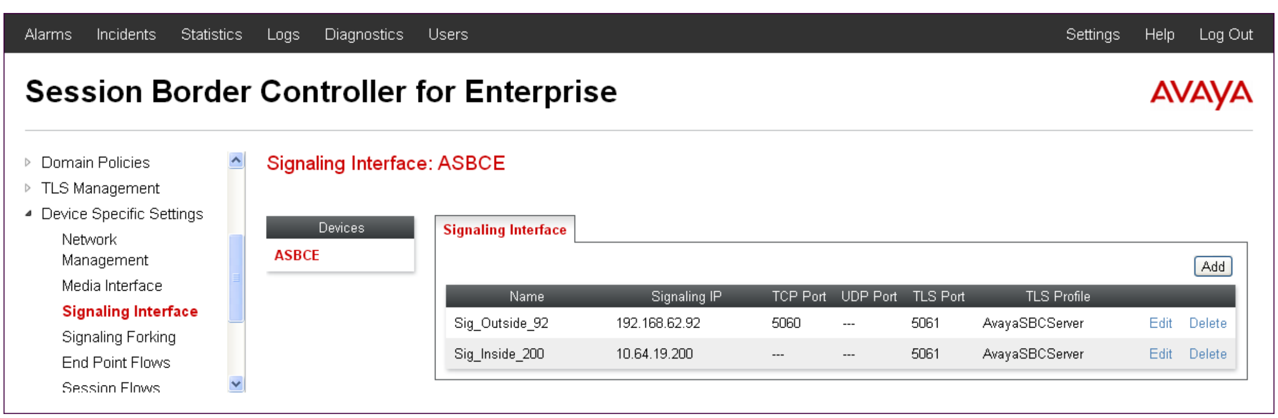
Interfaces **A1** and **B1** were both set to **Enabled**.



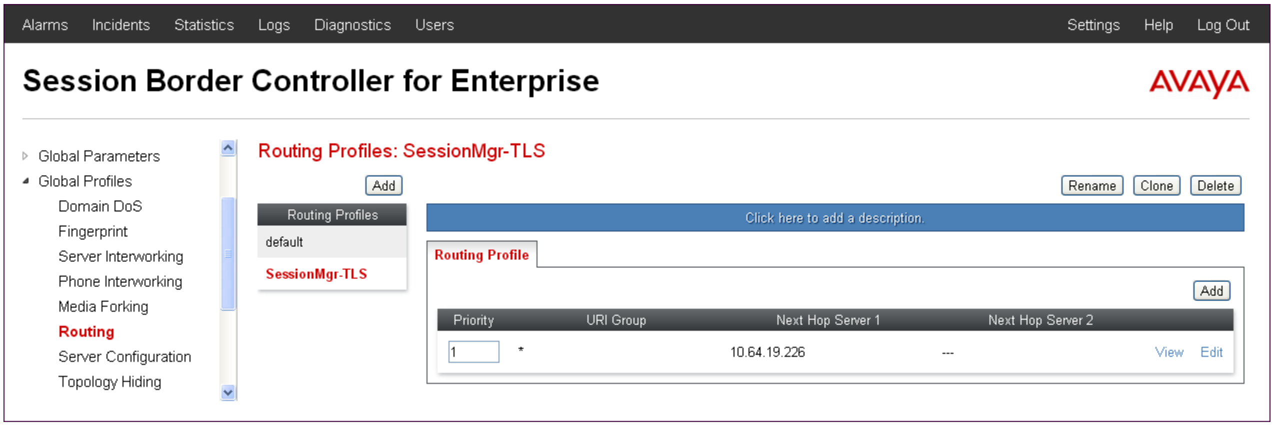
The following screen shows the **Media Interface** settings. Media interfaces were created for the inside and outside IP interfaces used for Remote Worker SIP traffic.

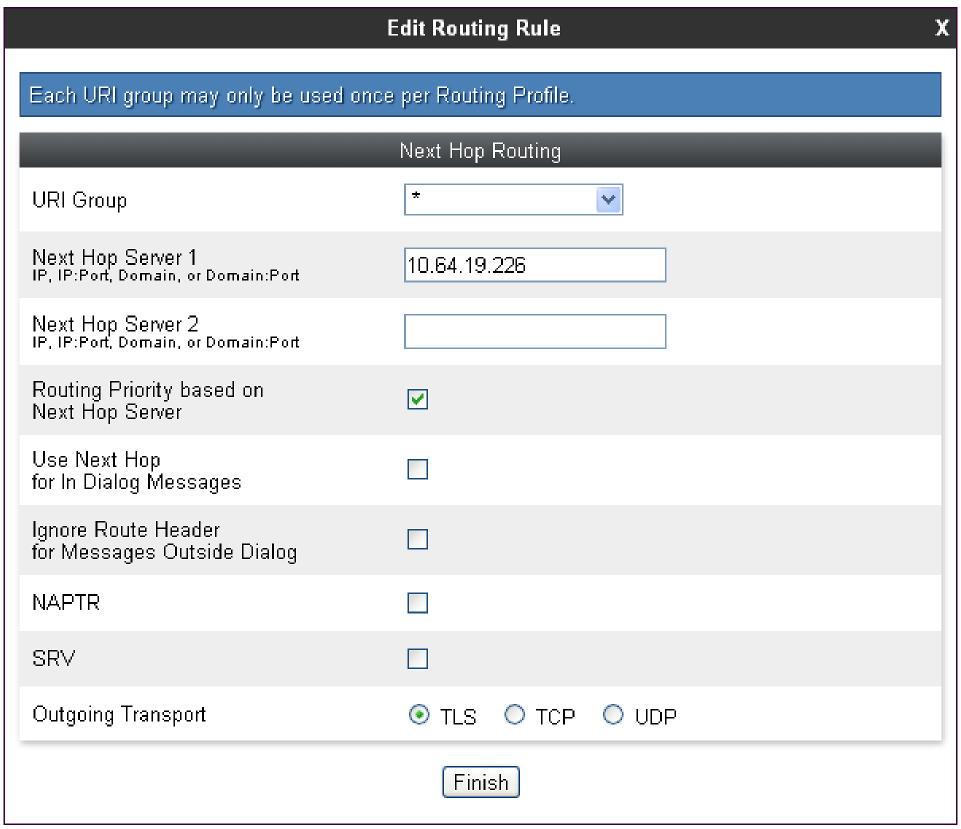


The following screen shows the **Signaling Interface** settings. Signaling interfaces were also created for the inside and outside IP interfaces used for Remote Worker SIP traffic. The interface named "Sig_Outside_92" supports TCP and TLS, while the interface named "Sig_Inside_200" supports TLS only.

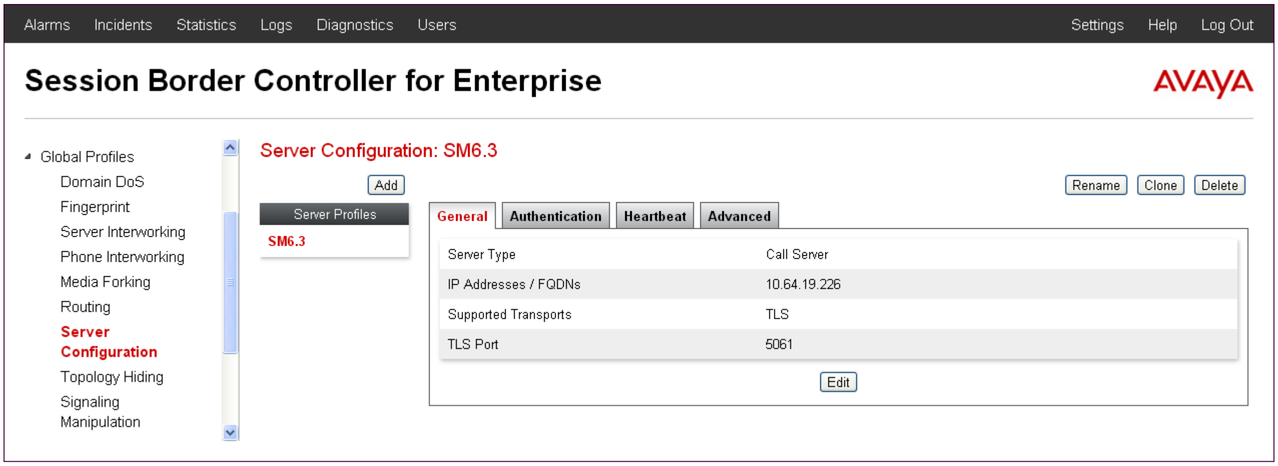


Routing Profile "SessionMgr-TLS" was created for Session Manager as shown in the following screens.

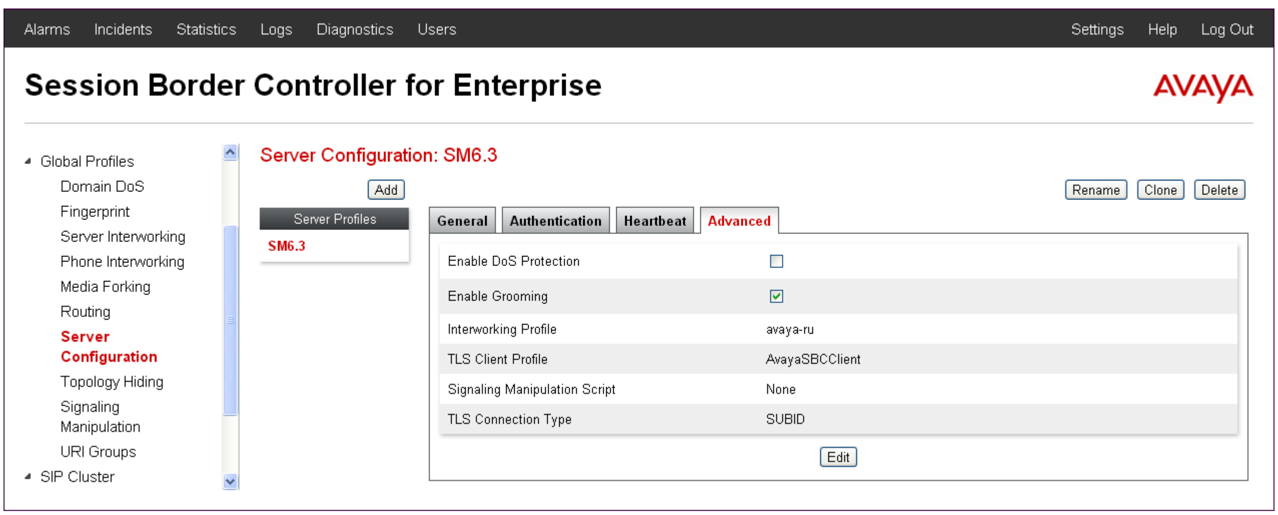




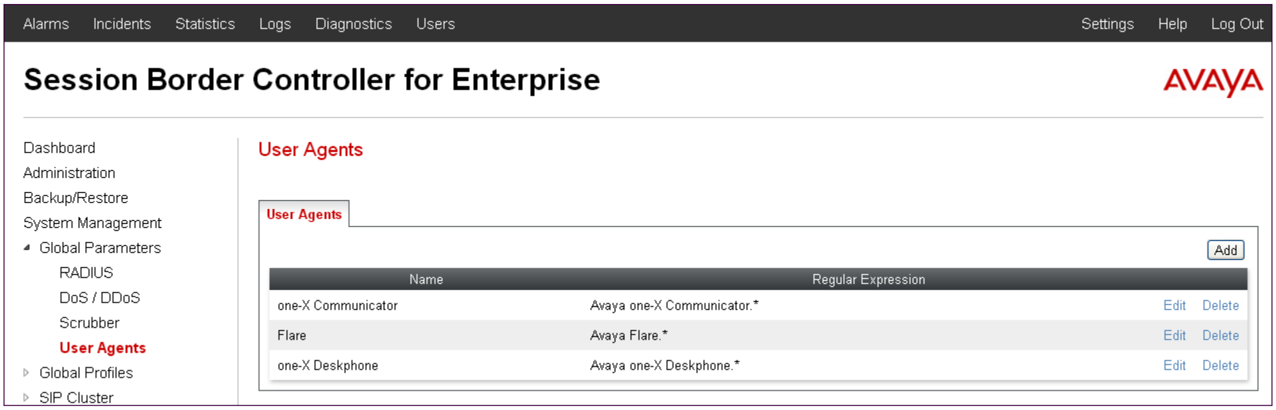
The following screens illustrate the **Server Configuration** for the Profile named “SM6.3” created for Session Manager. The **Authentication** and **Heartbeat** tabs were kept at the default disabled setting.



On the **Advanced** tab, default profiles were specified that applies to traffic between the Avaya SBCE and Session Manager as shown below.



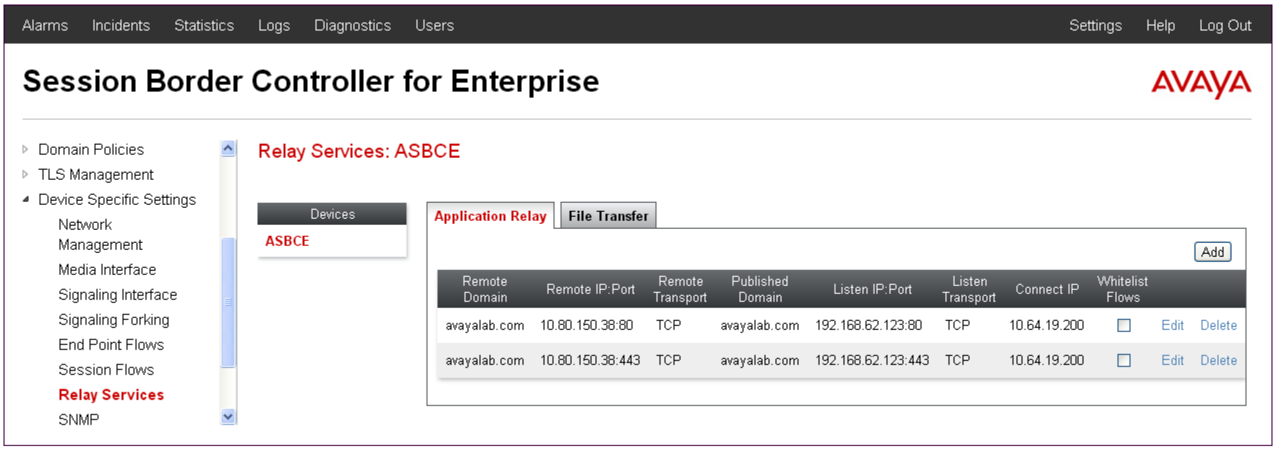
User Agents were created for each type of endpoint tested. This allows for different policies to be applied based on the type of device. For example, Avaya one-X Deskphones will use TLS and SRTP, while one-X® Communicator and Avaya Flare® will use TCP and RTP.



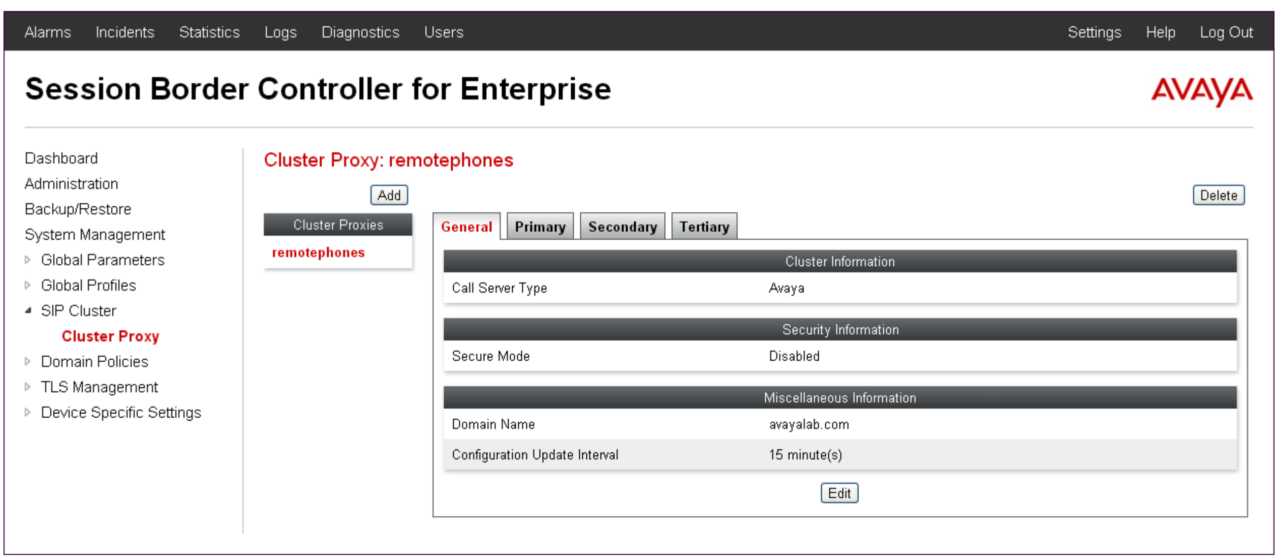
The following abridged output of Session Manager's traceSM command shows the details of an INVITE message from an Avaya one-X Deskphone. The **User-Agent** shown in this trace will match "one-X Deskphone" shown above with a **Regular Expression** of "Avaya one-X Deskphone.*". In this expression, ".*" will match any software version listed after the user agent name.

```
INVITE sip:12002@avayalab.com SIP/2.0
From: <sip:14006@avayalab.com>;tag=-76557dff51bb3900-5c89896d_F1400610.80.150.111
To: <sip:12002@avayalab.com>
CSeq: 357 INVITE
Call-ID: 161_51bb3900-2ff0c2ff-5c898dda_I@10.80.150.111
Contact: <sip:14006@10.64.19.200:5061;transport=tls;subid_ipcs=448140782>;+avaya-cm-line=1
Record-Route: <sip:10.64.19.200:5061;ipcs-line=930;lr;transport=tls>
Record-Route: <sips:205.168.62.92:5061;ipcs-line=930;lr;transport=tls>
Allow:
INVITE,ACK,BYE,CANCEL,SUBSCRIBE,NOTIFY,MESSAGE,REFER,INFO,PRACK,PUBLISH,UPDATE
Supported: 100rel,eventlist,feature-ref,replaces
User-Agent: Avaya one-X Deskphone 6.2.2.17 (40235)
Max-Forwards: 69
Via: SIP/2.0/TLS 10.64.19.200:5061;branch=z9hG4bK-s1632-001744755540-1--s1632-
Via: SIP/2.0/TLS 10.80.150.111:5061;branch=z9hG4bK165_51bb39027017c28d-5c899bb5_I14006
Accept-Language: en
Content-Type: application/sdp
Content-Length: 416
```

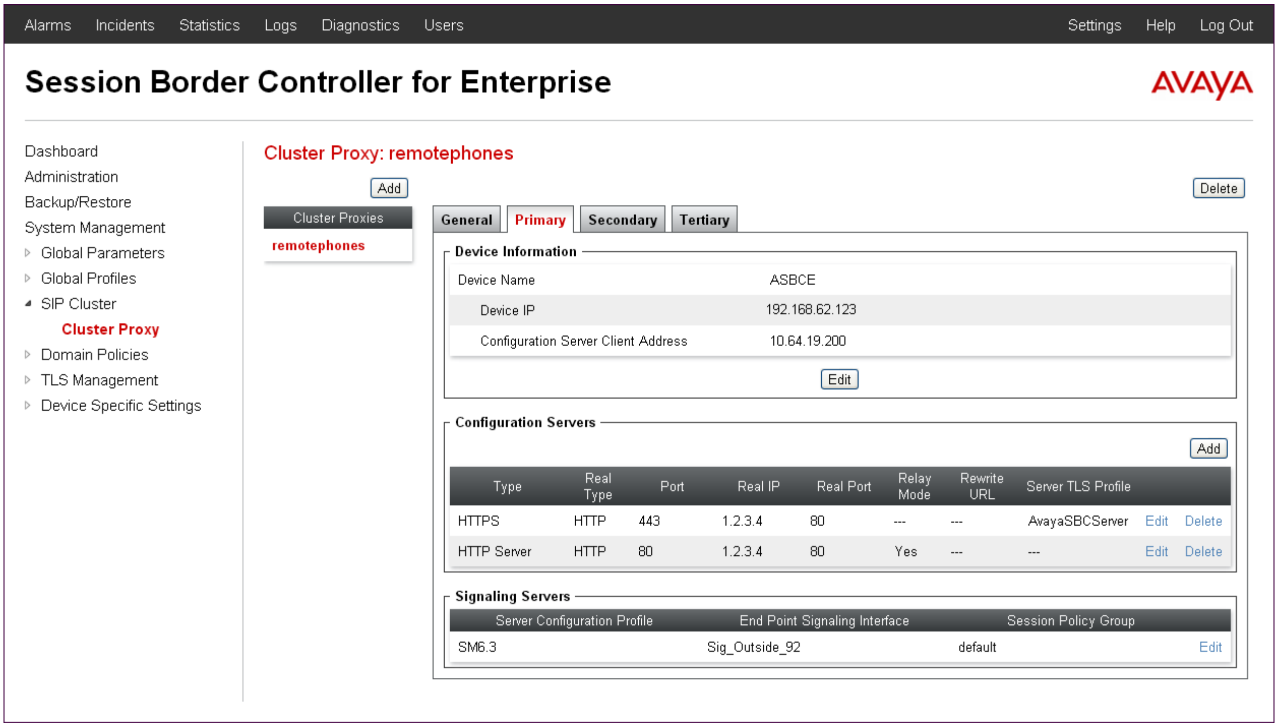
Relay Services are used to define how firmware updates and configuration data are routed for remote endpoints. The following screen shows the **Application Relay** tab with the two application relays created for the sample configuration. This allows for both HTTP and HTTPS traffic to be routed to the appropriate internal file server. The **Remote IP:Port** was set to the IP address and port of the internal file server used to provide the firmware updates and configuration data for the remote endpoints. The **Listen IP:Port** was set to the IP address and port of the Avaya SBCE's external IP address designated for file transfers. The **Connected IP** was set to the internal IP address of the Avaya SBCE.



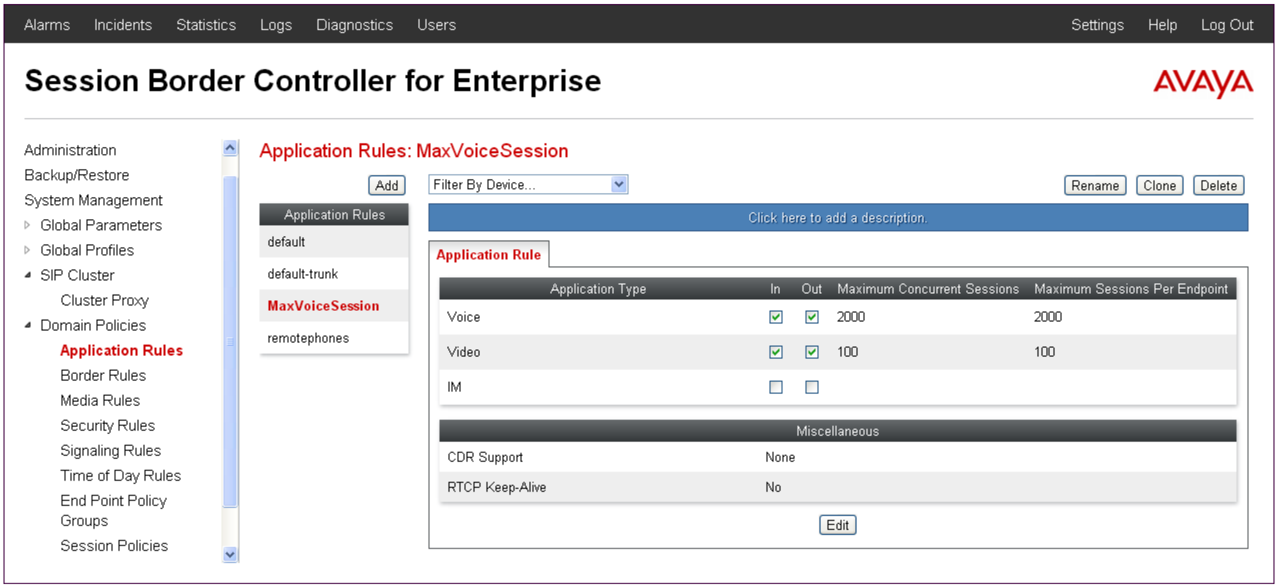
A **Cluster Proxy** is used for Personal Profile Manager (PPM) data and Presence services. The following screen shows the cluster proxy "remotephones" created in the sample configuration. A Presence Services server was not part of the sample configuration. Therefore, configuration of the cluster proxy for use with Presence is not shown and only configuration related to PPM data is present.

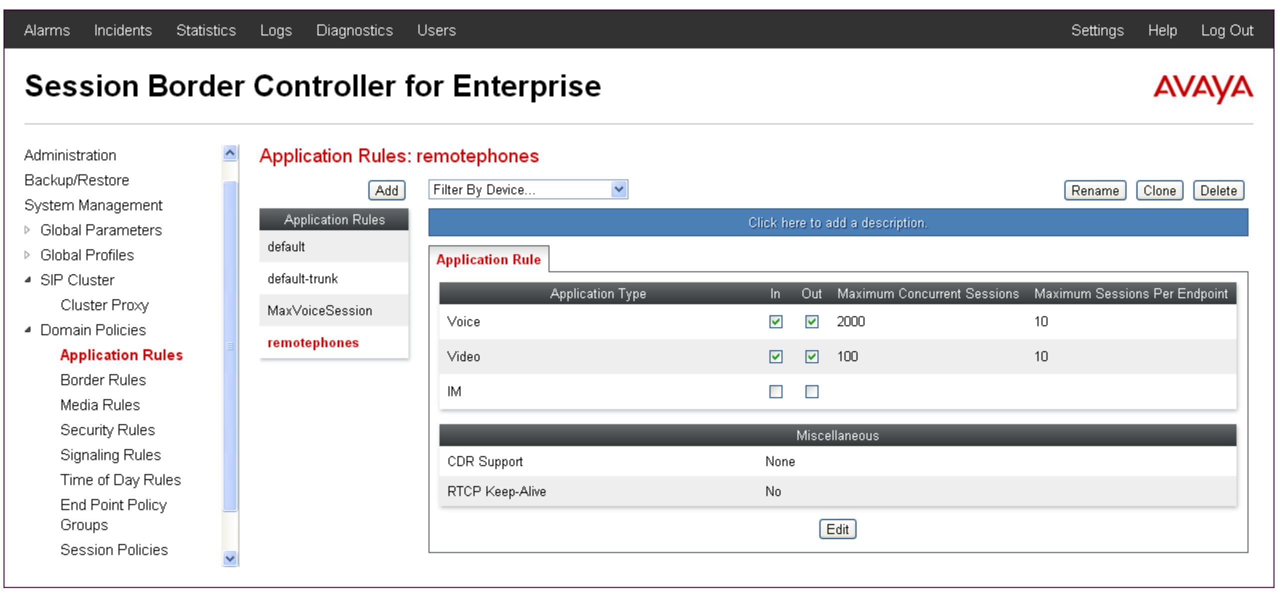


On the **Primary** tab, PPM traffic received on **Device IP** “192.168.62.92” will be routed to the **Configuration Server Client Address** “10.64.19.200”. The **Real IP** field is not used for PPM, so any IP address can be entered, e.g., “1.2.3.4”. This enables the remote Avaya SIP endpoints to send and receive PPM information to and from Session Manager via the Avaya SBCE.

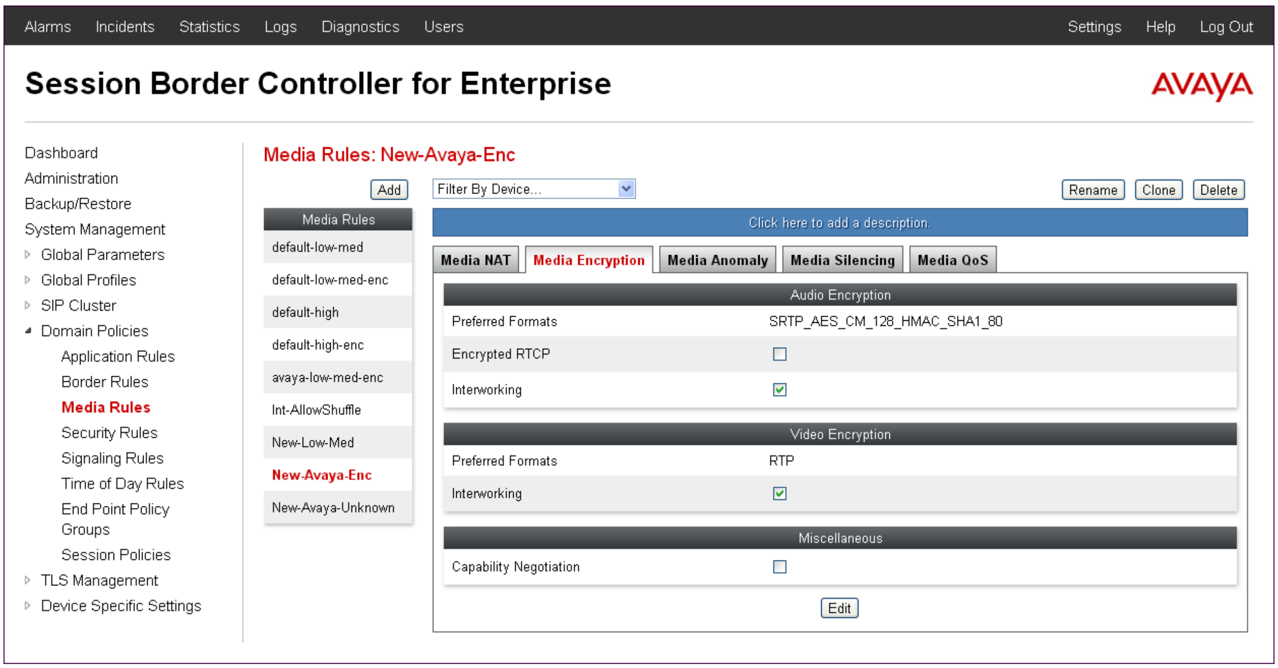


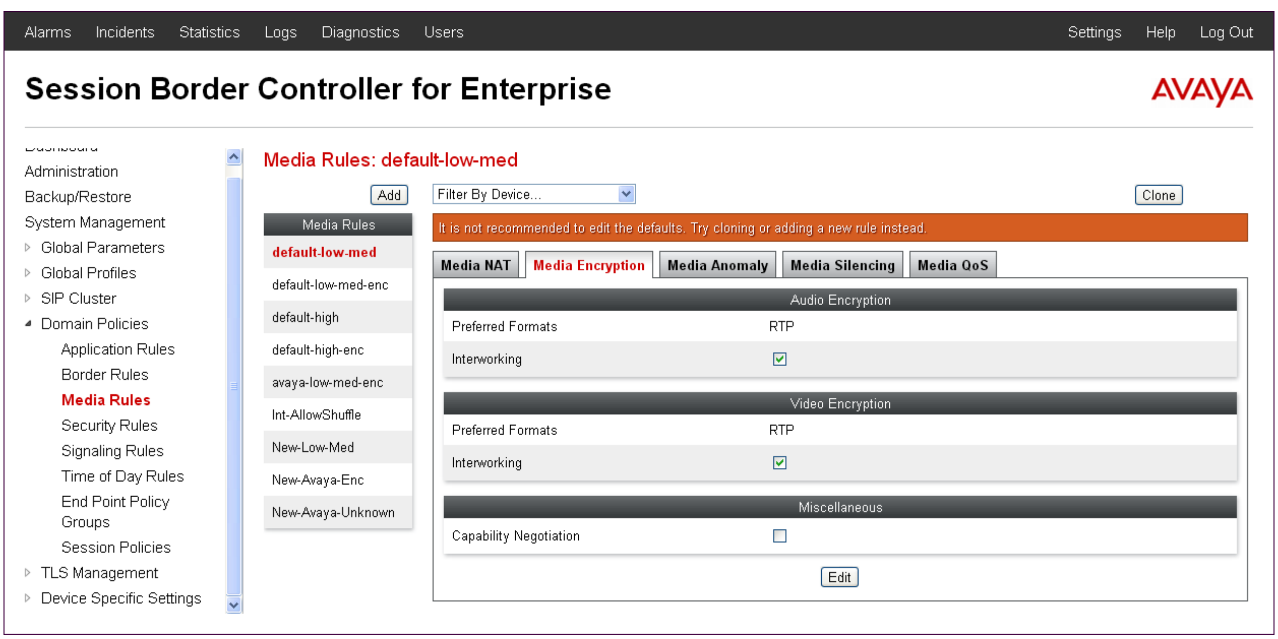
The following screens show the **Application Rules** "MaxVoiceSession" and "remotephones" used in the sample configuration. In an actual customer installation, set the **Maximum Concurrent Sessions** for the **Voice** application to a value slightly larger than the licensed sessions. For example, if licensed for 300 session set the values to "500". For the "MaxVoiceSession" rule, the **Maximum Session Per Endpoint** matches the **Maximum Concurrent Sessions**. For the application rule applied to the Remote Workers subscriber flows, a value of "10" is recommended for the **Maximum Session Per Endpoint**.



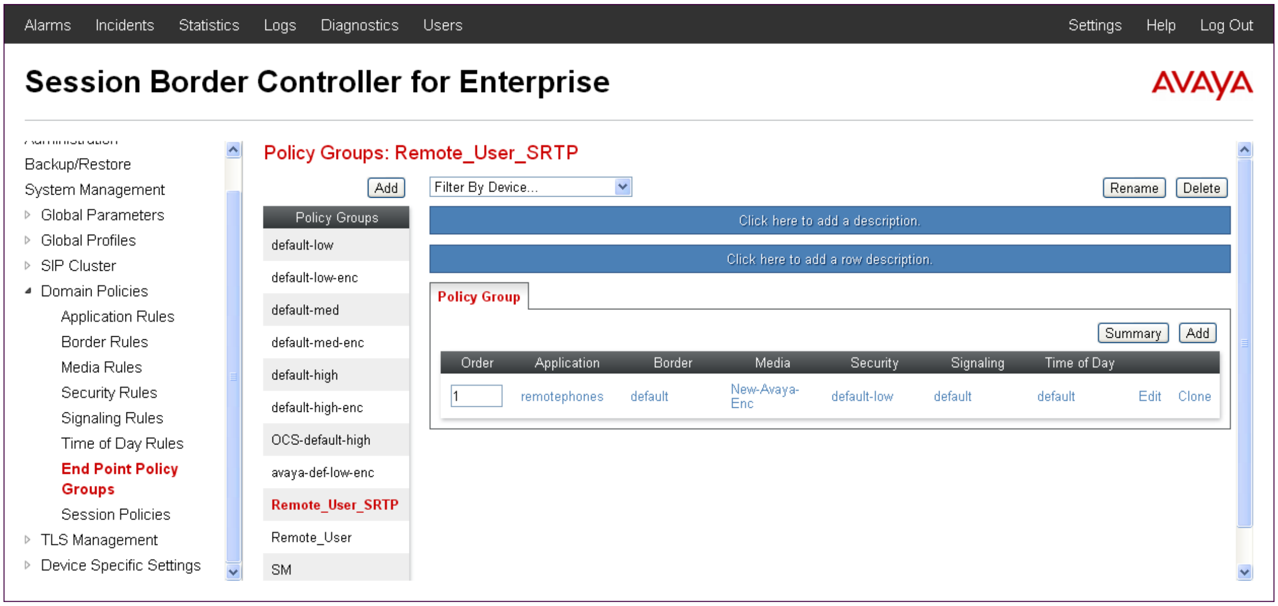


The following screens show **Media Rules** “New-Avaya-Enc” and “default-low-med” that will later be assigned to the End Point Policy Groups. Note that both rules have **Interworking** checked. Based on how calls are routed through Avaya SBCE, this will convert SRTP media to RTP and vice versa. In the sample configuration, Avaya SBCE will convert the SRTP media stream from remote Avaya 96x1 SIP Telephones to RTP towards the enterprise and also towards remote endpoints using TCP. Avaya SBCE will also convert RTP media from calls originating from Session Manager to SRTP towards Avaya 96x1 SIP Telephones using TLS through the external IP interface. The “New-Avaya-Enc” policy was cloned from the existing “default-low-med” policy. The parameters on the other tabs not shown retained their default values.

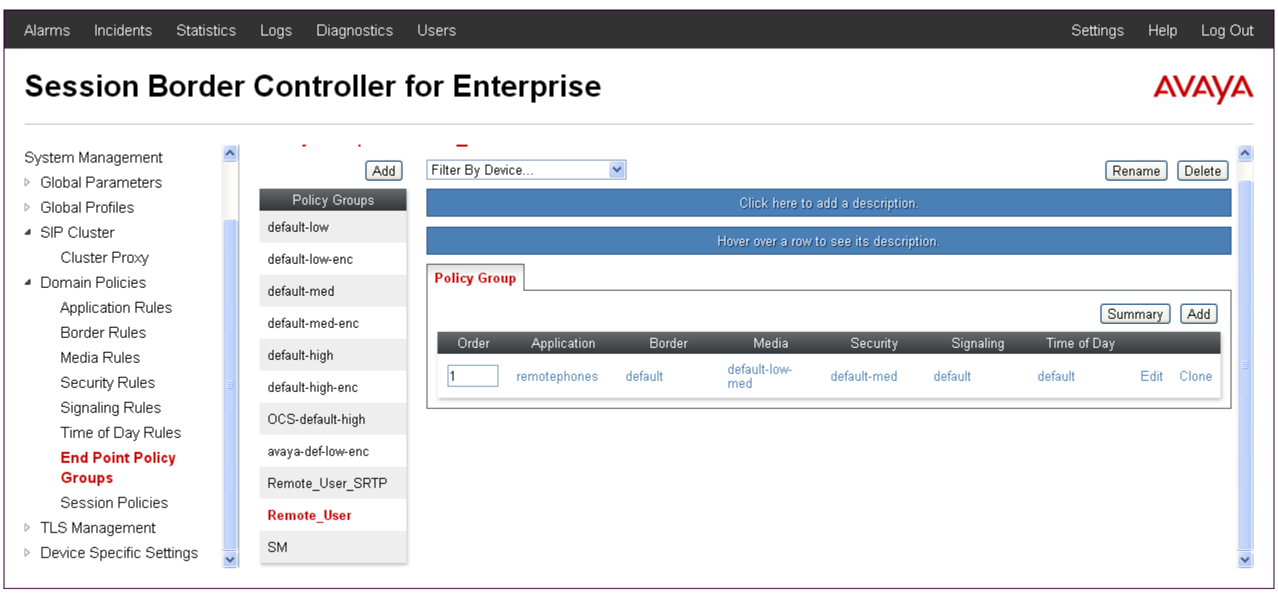




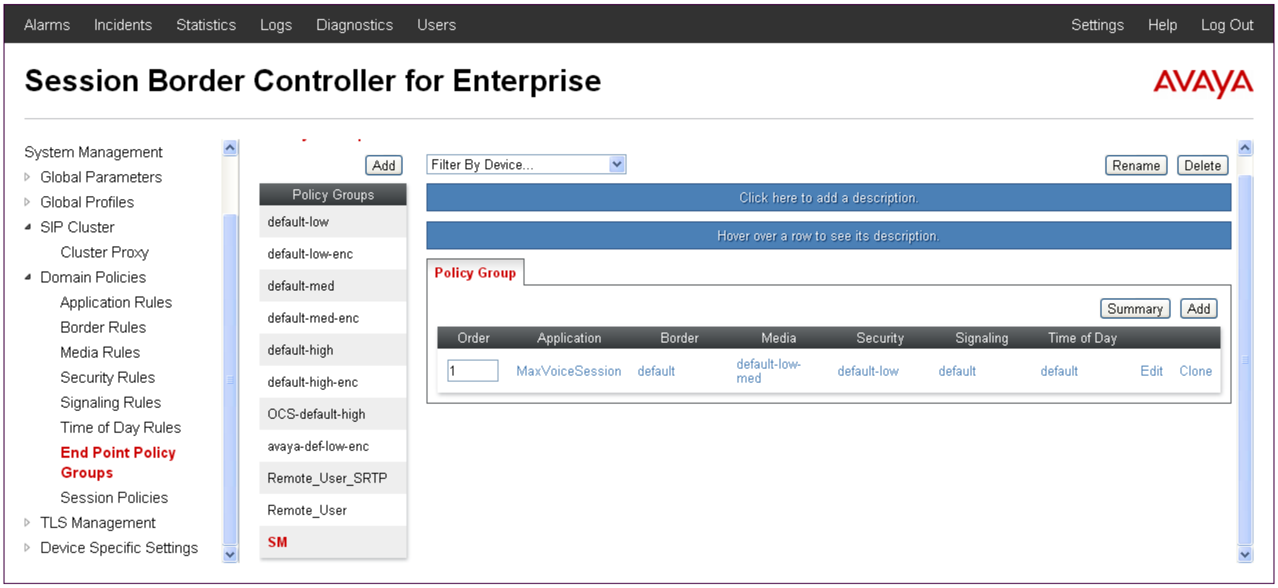
The following **End Point Policy Groups** will later be assigned to the subscriber and server flows. The “Remote_User_SRTP” policy uses the “remotephones” **Application** rule and the “New-Avaya-Enc” **Media** rule.



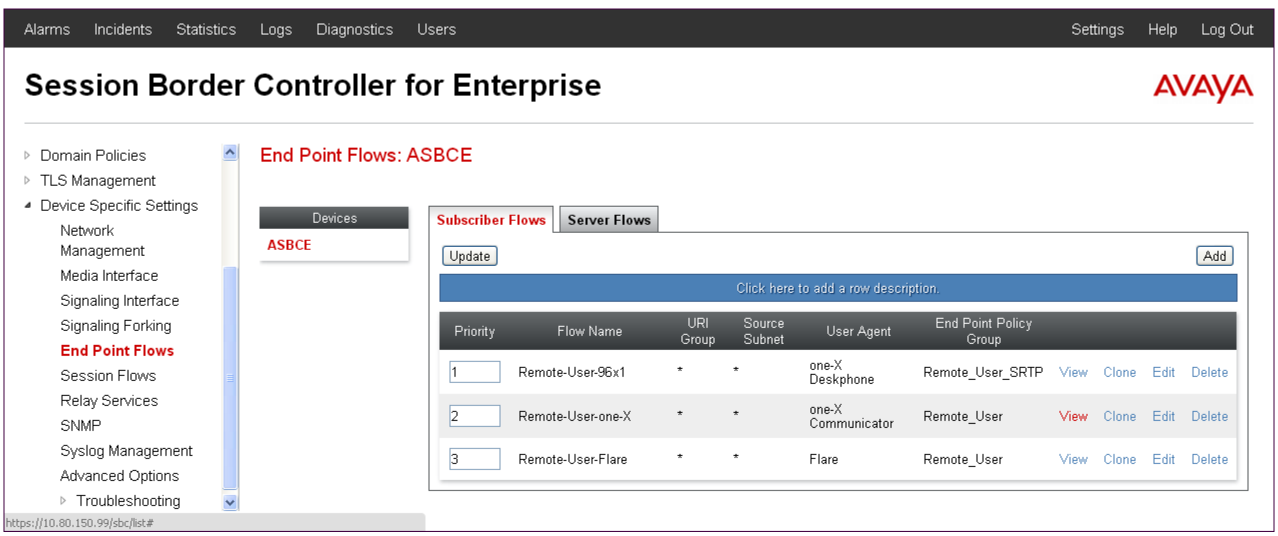
The “Remote_User” policy uses the “remotephones” **Application** rule and the “default-low-med” **Media** rule.



The “SM” policy uses the “MaxVoiceSession” **Application** rule and the “default-low-med” **Media** rule.



Three **Subscriber Flows** were created for Remote Workers. One for each **User Agent** previously created.



The following screen shows the details of the flow named “Remote-User-96x1” used in the sample configuration. This flow will match traffic from remote Avaya 96x1 Series IP Telephones set to use TLS. Note that the **User Agent** was set to “one-X Deskphone” and that the **End Point Policy Group** was set to “Remote_User_SRTP”.

View Flow: Remote-User-96x1

Criteria

Flow Name	Remote-User-96x1
URI Group	*
User Agent	one-X Deskphone
Source Subnet	*
Via Host	*
Contact Host	*
Signaling Interface	Sig_Outside_92

Optional Settings

Topology Hiding Profile	None
Phone Interworking Profile	Avaya-Ru
TLS Client Profile	AvayaSBCClient
RADIUS Profile	None
File Transfer Profile	None
Signaling Manipulation Script	None

Profile

Source	Subscriber
Methods Allowed Before REGISTER	
User Agent	one-X Deskphone
Media Interface	Media_Outside_92
End Point Policy Group	Remote_User_SRTP
Routing Profile	SessionMgr-TLS

The “Remote-User-one-X” flow will match traffic from remote one-X® Communicator devices set to use TCP. Note that the **User Agent** was set to “one-X Communicator” and that the **End Point Policy Group** was set to “Remote_User”.

View Flow: Remote-User-one-X

Criteria

Flow Name	Remote-User-one-X
URI Group	*
User Agent	one-X Communicator
Source Subnet	*
Via Host	*
Contact Host	*
Signaling Interface	Sig_Outside_92

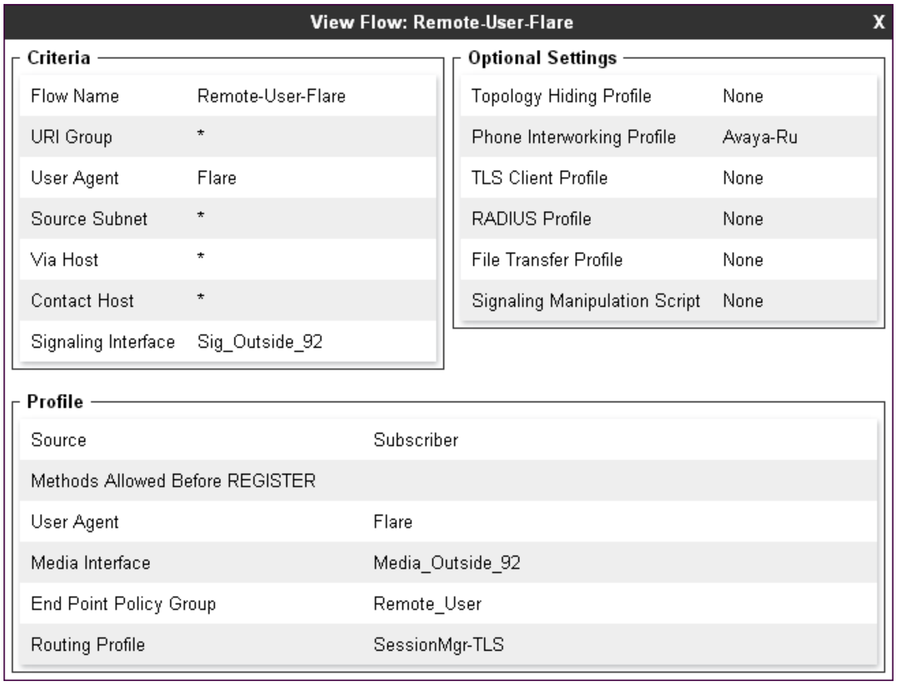
Optional Settings

Topology Hiding Profile	None
Phone Interworking Profile	Avaya-Ru
TLS Client Profile	None
RADIUS Profile	None
File Transfer Profile	None
Signaling Manipulation Script	None

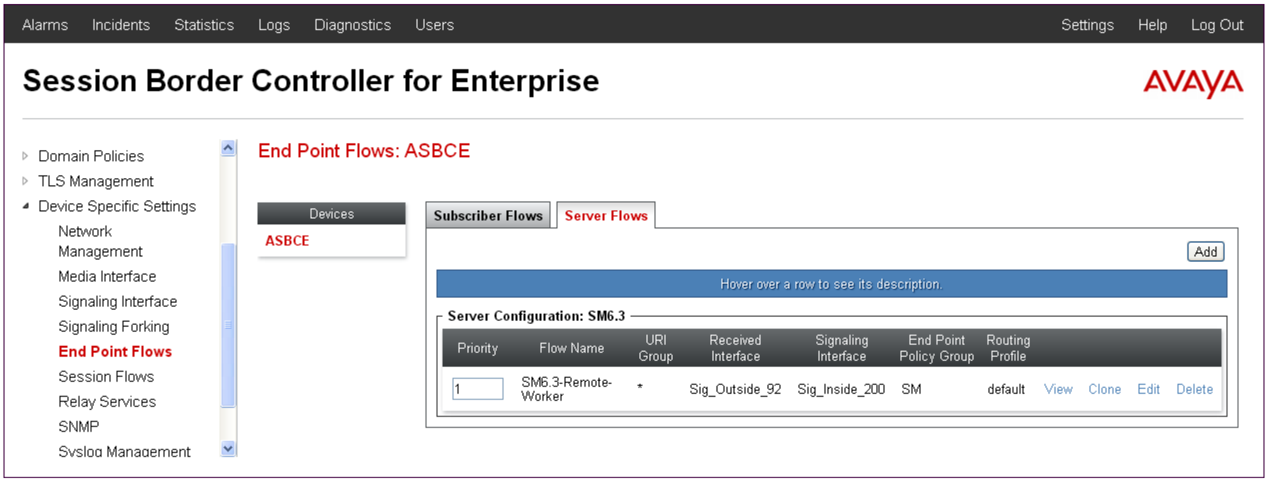
Profile

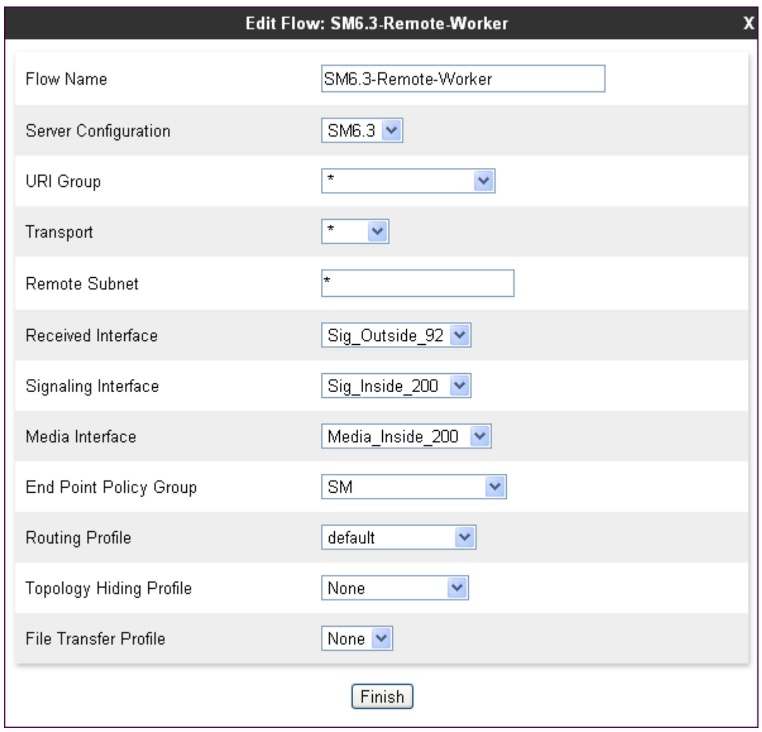
Source	Subscriber
Methods Allowed Before REGISTER	
User Agent	one-X Communicator
Media Interface	Media_Outside_92
End Point Policy Group	Remote_User
Routing Profile	SessionMgr-TLS

The “Remote-User-Flare” flow will match traffic from remote Avaya Flare® devices set to use TCP. Note that the **User Agent** was set to “Flare” and that the **End Point Policy Group** was set to “Remote_User”.



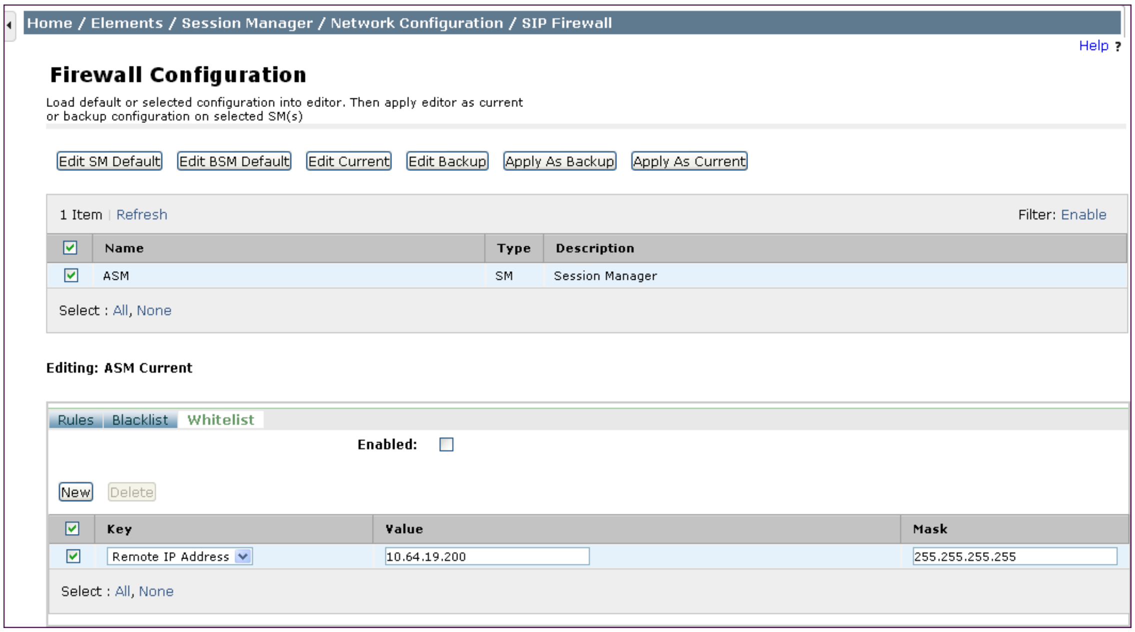
The following screens show the **Server Flows** settings for Session Manager.



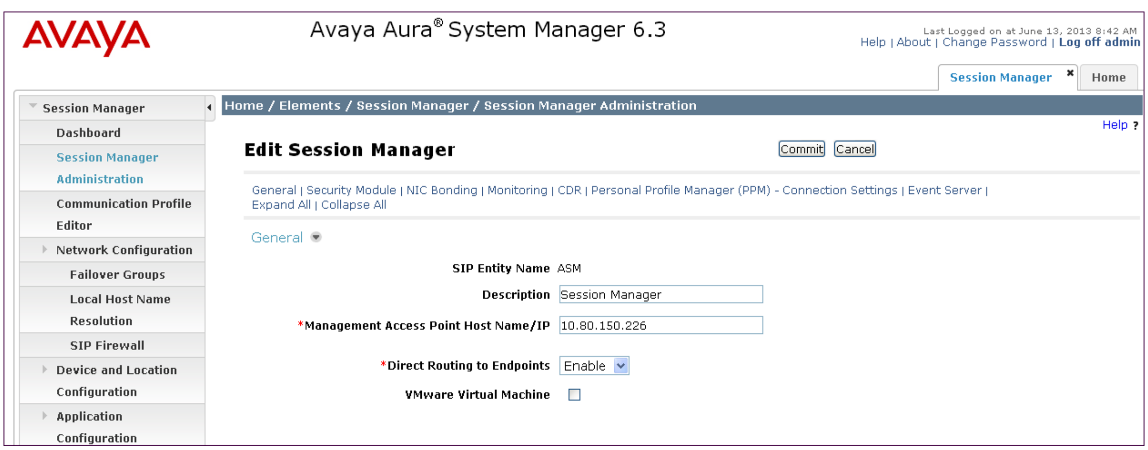


In the sample configuration, the internal IP address of the Avaya SBCE used for Remote Worker was added to Session Manager's SIP Firewall Whitelist and PPM limiting was disabled.

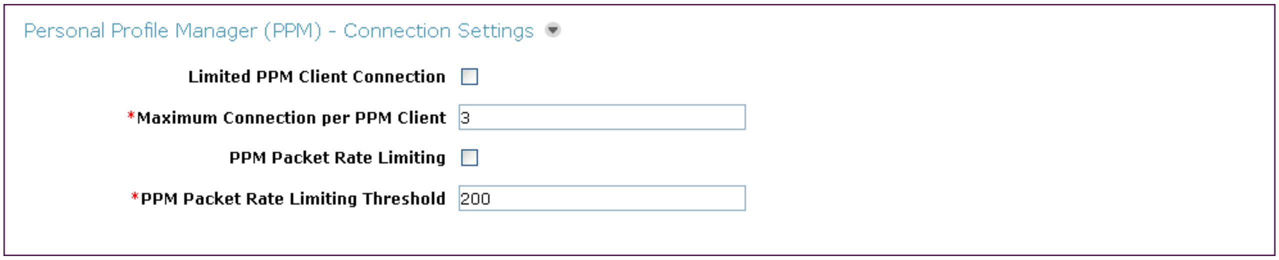
To add an IP address to the Whitelist, log into System Manager and navigate to **Session Manager → Network Configuration → SIP Firewall**. Select the Session Manager listed in the top section and click **Edit SM Default**. Select the **Whitelist** tab towards the bottom of the screen and click **New**. Enter the internal IP address of Avaya SBCE used for Remote Workers in the **Value** field and "255.255.255.255" in the **Mask** field. Click **Apply As Current** to save the configuration.



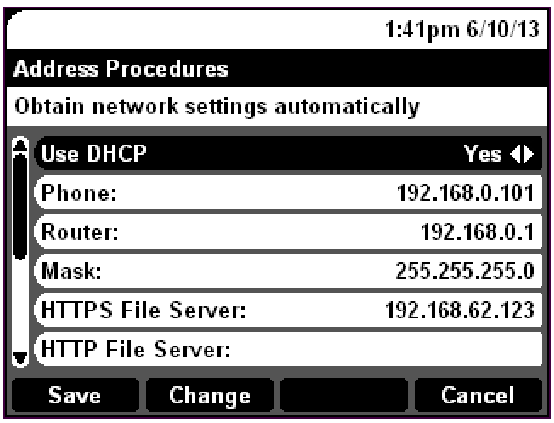
To disable PPM limiting, navigate to **Session Manager → Session Manager Administration** in the left-hand navigation pane and click **View** (not shown). A screen such as the following is displayed.

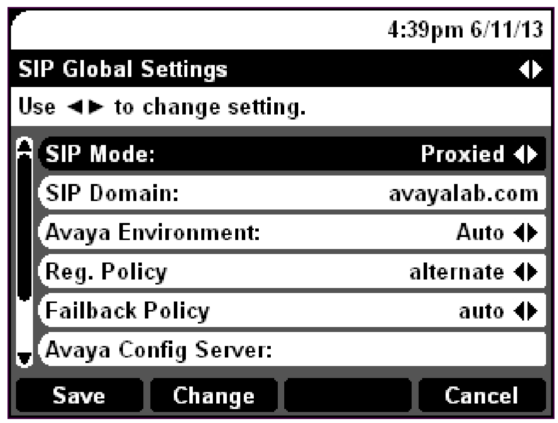


Scroll down to the **Personal Profile manager (PPM) – Connection Settings**. Uncheck **Limited PPM Client Connections** and **PPM Packet Rate Limiting**.

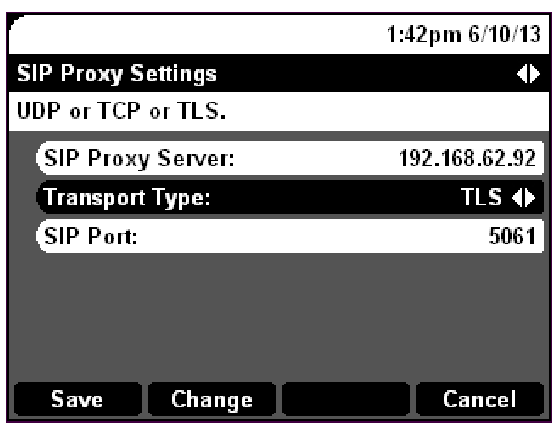


The following screens show an Avaya one-X® Deskphone SIP Emulator illustrating the administration settings of a SIP endpoint used for Remote Worker. Note that the **HTTPS File Server** is set to the external IP address of the Avaya SBCE designated for firmware and configuration file transfers. Under **SIP Global Settings**, the **SIP Domain** is set to "avayalab.com". The domain expected by Session Manager.





Under **SIP Proxy Settings**, the **SIP Proxy Server** is set to the external IP address of Avaya SBCE designated for Remote Worker SIP traffic. The Transport Type and SIP Port should be set according to device type. For example, "TLS" and "5061" for one-X® Deskphones, and "TCP" and "5060" for one-X® Communicator and Flare® Experience.



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.