Avaya Solution & Interoperability Test Lab

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 1.1

Abstract

These Application Notes illustrate a sample configuration using Avaya Aura® Session Manager Release 6.1, Avaya Aura® Communication Manager Release 6.0.1, and a connection to the Verizon Interop Lab over an IP Trunk. The Verizon Business SIP trunk redundant architecture (2-CPE) is supported by dual Avaya Session Border Controllers for Enterprise.

The Verizon Business IP Trunk service offer 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 VPN connection to the Verizon Interop Lab.

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1. Introduction

These Application Notes illustrate a sample configuration using Avaya Aura® Session Manager Release 6.1 and Avaya Aura® Communication Manager Release 6.0.1 with the Verizon Business Internet Dedicated Access (IDA) Trunk service. The Verizon Business IP Trunk service provides local and/or long-distance calls (with PSTN endpoints) via standards-based SIP trunks. These Application Notes update previously published Application Notes with newer versions of Communication Manager and Session Manager. The Verizon Business SIP trunk redundant architecture (2-CPE) is supported by dual Avaya Session Border Controllers for Enterprise (ASBCE). The Verizon Business SIP Trunk redundant (2-CPE) architecture provides for redundant SIP trunk access between the Verizon Business IP Trunk service offer and the customer premises equipment (CPE).

Dual ASBCEs are used as edge devices between the Avaya CPE and the Verizon Business network, and provide for Verizon Business 2-CPE redundancy. In addition, the ASBCEs provide Network Address Translation (NAT) functionality to convert the addresses used within the enterprise to the Verizon routable addresses.

Note - The Verizon Business SIP Trunk Redundant (2-CPE) architecture is a service option and its use is not a requirement of the Verizon Business IP Trunk service offer.

Verizon Business and Avaya developed the SIP Trunk Redundant (2-CPE) architecture to ensure that SIP trunk calls can be automatically re-routed to bypass SIP trunk failures due to network or component outages. The 2-CPE architecture described in these Application Notes is based on a customer location having two ASBCEs. One ASBCE is designated as Primary and one as Secondary.

Avaya Aura® Session Manager is provisioned for fail-over of outbound calls from one ASBCE to the other, if there is a failure (e.g., timeout, or error response) associated with the first choice. Similarly, the Verizon Business Private IP Trunk service node will send inbound calls to the Primary Avaya ASBCE if there is a failure (e.g., timeout, or error response), and the call will be sent to the Secondary ASBCE.

2. General Test Approach and Test Results

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 Avaya Aura® Communication Manager and the PSTN can be made using G.711MU or G.729A codecs.
- Direct IP-to-IP Media (also known as "Shuffling") when applicable.
- DTMF using RFC 2833
    - Outbound call to PSTN application requiring DTMF navigation (e.g., an IVR or voice mail system)
    - Inbound call from PSTN to Avaya CPE requiring DTMF navigation(e.g., Avaya Modular Messaging, Avaya vector digit collection steps)
- Emergency calling (e.g. 911)
- 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
- Automatic fail-over testing associated with the 2-CPE redundancy (i.e., calls automatically re-routed around component outages).

2.2. Test Results

- **SIP OPTIONS:** While making multiple changes in the ASBCE, the SIP OPTIONS can possibly stop being proxied properly from the Inside to the Outside. This caused Session Manager to mark the links from Session Manager to the ASBCE as down and not allow any calls to/from the ASBCE. The work around is to reboot the ASBCE. Internal tracking issue AURORA-202 has been created for this issue.

- **SIP REFER/TRANSFER OFF-NET:** ASBCE: Server Configuration: If in a profile in the Global Profiles→Server Internetworking →Advanced settings, the "Topology Hiding: Change Call-ID" is set to y. When a call is referred to a PSTN extension using SIP REFER, the Refer-To Replaces value may be incorrect. This may cause the service provider to send a 603 DECLINE instead of a 202 ACCEPT on the REFER. This will allow the call to be transferred, but will not release media resources for the transfer, and the call will stay resident on the system. The recommended work-around is to set this feature to 'no'. A fix is expected in ASBCE Release 6.2. Internal tracking issue AURORA-411 has been created for this issue. See Section 7.2.1 and 7.2.2 for more details.

- **SIP REFER/TRANSFER OFF-NET:** When using SIP REFER and transferring a call to the PSTN, the Referred-By Header will incorrectly contain the service providers IP Address instead of the ASBCE outside address. This may cause the service provider to send a 603 DECLINE instead of a 202 ACCEPT on the REFER. This will allow the call to be transferred, but will not release media resources for the transfer, and the call will stay resident on the system. The recommended work-around is to use a Sigma Script detailed in Section 7.3.5. A fix is expected in ASBCE Release 6.2. Internal tracking issue AURORA-410 has been created for this issue.

- **SIP REFER/TRANSFER OFF-NET:** If on Communication Manager the public-unkonwn numbering table is being used to map local extensions to DIDs, and a transfer to the PSTN is attempted using a SIP REFER, the Referred-By Header will incorrectly contain the local extension instead of the DID. This may cause the service provider to send a 603 DECLINE instead of a 202 ACCEPT on the REFER. This will allow the call to be transferred, but will not release media resources for the transfer, and the call will stay resident on the system. The recommended work-around is to use a Sigma Script detailed in Section 7.3.5. Internal tracking issue defsw121205 has been created for this issue.

- **SIP REFER/TRANSFER OFF-NET:** If on Communication Manager the public-unkonwn numbering table is being used to map local extensions to DIDs and a transfer to the PSTN is attempted using a SIP REFER, the Contact Header will incorrectly contain the local extension instead of the DID. This may cause the service provider to send a 603 DECLINE instead of a 202 ACCEPT on the REFER. This will allow the call to be transferred but will not release media resources for the transfer and the call will stay resident on the system. The recommended work-around is to use a Sigma Script detailed in Section 7.3.5. . Internal tracking issue defsw121215 has been created for this issue.

- **DOMAIN NAME IN HEADERS**: If on the ASBCE, the Global Profiles→Topology Hiding option is set to "Overwrite" instead of "Auto" the initial INVITE will have the correct DNS name, but subsequent SIP messages will contain the Outside IP Address of the ASBCE. In testing, this did not create any immediate problems as DNS and IP Addresses were accepted by the Verizon network. See Section 7.3.3 and 7.3.4 for examples. Internal tracking issue AURORA-412 has been created. See Section 7.2.

- **2 – CPE TESTING:** Although the ASBCE will proxy OPTIONS messages from inside the network to outside, sourcing of OPTIONS must be turned on if a 2-CPE configuration is used or failover will not occur properly.

- **RE-INVITE:** When using an Avaya SIP phone with G.711 as the preferred codec and a call is established as G.711, when a re-invite is issued by Communication Manager for a shuffle, Verizon sends an ACK with just G.729 listed, so the SIP Phone will switch codecs to G.729 if G.729 is allowed in the codec list. The user experience will not be affected and the calls stays connected.

- **TRANSFER:** When a PSTN caller is transferred off-net (to another PSTN user) the second PSTN phone will see the Caller-ID of the CPE phone.

2.3. 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 in these Application Notes is based on a customer location having two Avaya Session Border Controllers for Enterprise. One ASBCE is designated as Primary and one as Secondary. The ASBCEs reside at the edge of the customer network.

Avaya Aura® Session Manager is provisioned to attempt outbound calls to the Primary ASBCE first. If that attempt fails, the Secondary ASBCE is used. Similarly, the Verizon Business Private IP Trunk service node will send inbound calls to the Primary ASBCE. If there is no response then the call will be sent to the Secondary ASBCE.

2.4. Support

2.4.1 Avaya

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

2.4.2 Verizon

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

2.5. Known Limitations

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

- Verizon Business IP Trunking service does not support G.729B codec.
- Although Verizon Business now supports T.38 for faxing, Verizon Business will never perform a SIP Re-Invite so on outbound faxing, it is the responsibility of the Avaya CPE to send a re-Invite to T38, therefore the codec setting must be set to T38. See codec settings in **Section 5.6** for an example.

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.

3. Reference Configuration

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

The ASBCEs receive traffic from the Verizon Business IP Trunk service on port 5060 and send traffic to the Verizon Business IP trunk service on port 5208 (domestic) and 5234(EMEA), using UDP protocol for network transport (required by the Verizon Business IP Trunk service). The Verizon Business IP Trunk service provided 10 digits Direct Inward Dial (DID) numbers. These DID numbers can be mapped by Session Manager or Communication Manager to Avaya telephone extensions.

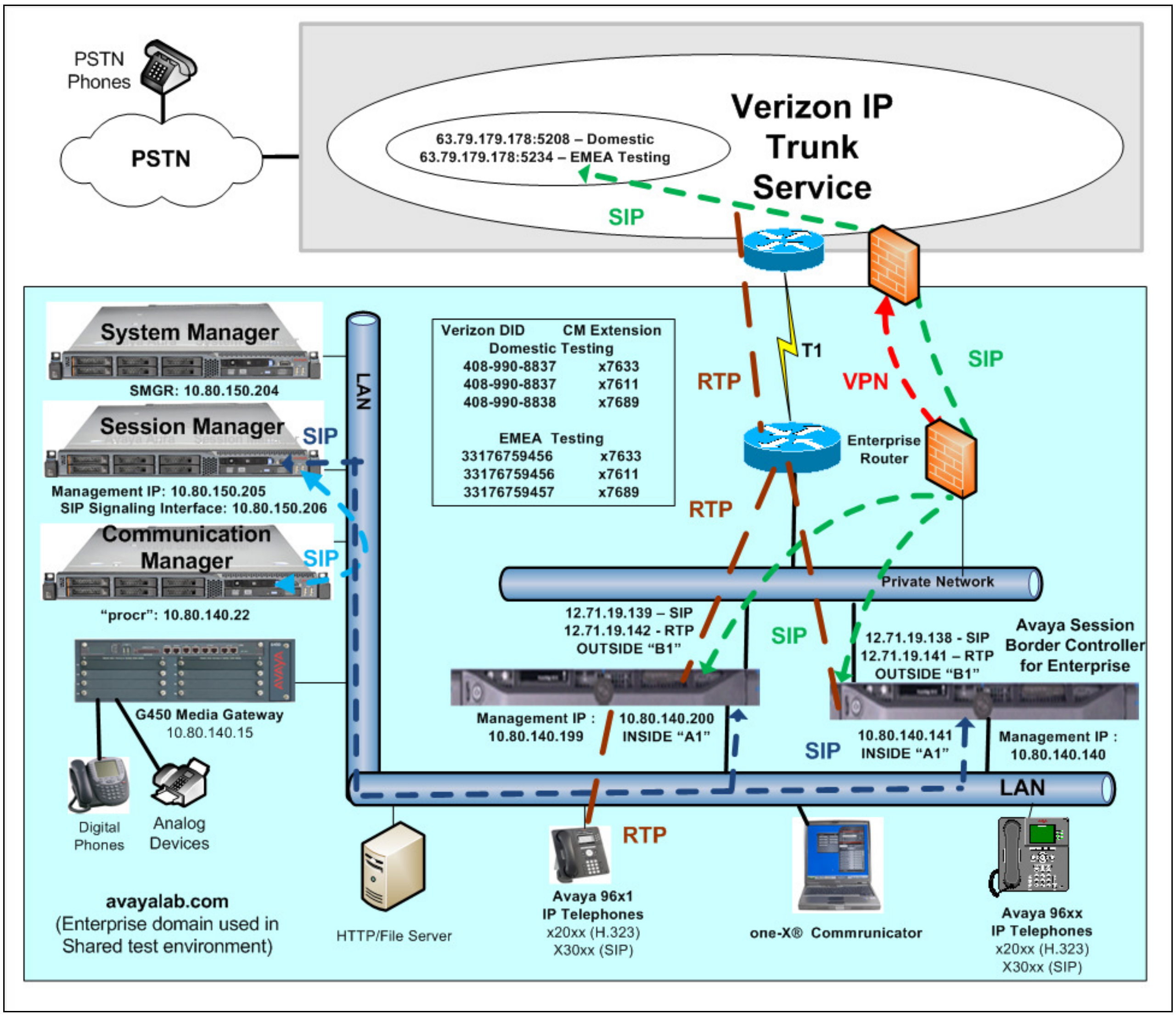


Figure 1: Avaya Interoperability Test Lab Configuration

The Verizon Business IP Trunk can use an IP Address or a domain name. The Avaya CPE environment was known to Verizon Business IP Trunk service as FQDN, *icrcn1n0002.customer08.tsengr.com* for domestic testing and *icrcn1n0002.customer34.tsengr.com* for EMEA testing. The Avaya CPE environment used the domain "avayalab.com" at the enterprise. As such, the ASBCEs are used to adapt the "avayalab.com" domain to the domain known to Verizon. These Application Notes indicate a configuration that would not be required in cases where the CPE domain in Communication Manager and Session Manager match the CPE domain known to the Verizon Business IP Trunk service.

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

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

- Verizon Business IP Trunk network IP Address
  - *63.79.179.178*
- Avaya CPE Fully Qualified Domain Name (FQDN) known to Verizon
  - *icrcn1n0002.customer08.tsengr.com – domestic*
  - *icrcn1n0002.customer34.tsengr.com - EMEA*

3.1. History Info and Diversion Headers

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

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

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 6.0.1
HP ProLiant DL360 G7	Avaya Aura® System Manager 6.1
HP ProLiant DL360 G7	Avaya Aura® Session Manager 6.1
G450 Gateway	3.1.20.1
DELL 210 RII	Avaya Session Border Controller for Enterprise Version 4.0.9Q02
Avaya 9600-Series Telephones (H.323)	96x1-IPT-H323-R6_0-09061
Avaya 9600-Series Telephones (SIP)	96xx-IPT-SIP-R2_6_3-101310
Avaya 96X1- Series Telephones (SIP)	96x1-IPT-SIP-R6_0_3-120511
Avaya 96X1- Series Telephones (H323)	96x1-IPT-H323-R6_0-090610
Avaya One-X Communicator (H.323)	6.1.2.06_SP2-35739
Avaya 2400-Series and 6400-Series Digital Telephones	N/A
Okidata Analog Fax	N/A

Table 1: Equipment and Software Used in the Sample Configuration

5. Configure Avaya Aura® Communication Manager

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 files control 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 2
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                       Maximum Video Capable Stations: 18000 0
                   Maximum Video Capable IP Softphones: 18000 1
                       Maximum Administered SIP Trunks: 24000 289
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0
```

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 four digit local extensions, such as 3xxx or 4xxx. 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: 2

    Dialed   Total  Call      Dialed   Total  Call      Dialed   Total  Call
    String   Length Type      String   Length Type      String   Length Type
   1           3   dac
   3           4   ext
   4           4   ext
   5           4   ext
   6           4   ext
   7           3   dac
   7           4   ext
   8           1   fac
   8           4   ext
   9           1   fac
   *           3   fac
   *10         4   dac
   #           3   fac
```

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 "**ASM**" with IP address **10.80.150.206**. The node name and IP address for the Processor Ethernet "**procr**" is **10.80.140.22**.

```
change node-names ip                                        Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
ASM                 10.80.150.206
Gateway1            10.80.140.1
default             0.0.0.0
procr               10.80.140.22
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 reference configuration.

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

```
change ip-interface pr                                            Page   1 of   2
                                  IP INTERFACES
                  Type: PROCR
                                                         Target socket load: 19660

      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: 10.80.140.22
```

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 4 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 (10.80.140.22), and that the gateway IP address is 10.80,140.15. 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
             Serial No: 11N510737929
          Encrypt Link? y                  Enable CF? n
        Network Region: 1                   Location: 1
                                           Site Data:
         Recovery Rule: 1

            Registered?  y
   FW Version/HW Vintage: 31 .18 .1  /1
      MGP IPV4 Address: 10.80.140.15
      MGP IPV6 Address:
   Controller IP Address: 10.80.140.22
           MAC Address: b4:b0:17:90:8c:30
```

The following screen shows **Page 2** for **Media Gateway 1**. The gateway has a **MM712** media module supporting Avaya digital phones in slot V6, a **MM711** supporting analog devices in slot V7 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:                                                 MP80      65   6
 V2:
 V3:
 V4:
 V5:                         WRG/OOS MM
 V6:   MM712                  DCP MM
 V7:   MM711                  ANA MM
 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 10.80.140.29 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: 10.80.140.0                              /24    1      n
   TO: 10.80.140.255
```

The following screen shows IP Network Region 4 configuration. In the shared test environment, network region 4 is used to allow unique behaviors for the Verizon test environment. In this example, codec set 4 will be used for calls within region 4. 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"). However, to illustrate the more typical case where Communication Manager domain matches the enterprise CPE domain known to Verizon, the **Authoritative Domain** in the following screen is "*icrcn1n0002.customer08.tsengr.com*", the domain known to Verizon, as shown in **Figure 1**. Even with this configuration, note that the domain in the PAI header sent by Communication Manager to Session Manager will contain "avayalab.com", the domain of the Far-end of the Avaya signaling group. The ASBCE will adapt "avayalab.com" to "*icrcn1n0002.customer08.tsengr.com*" in the PAI header, and the Diversion header.

```
change ip-network-region 4                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 4
Location:         Authoritative Domain: icrcn1n0002.customer08.tsengr.com
    Name: Verizon Domestic Tes
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 4              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 4. The first bold row shows that network region 4 is directly connected to network region 1, and that codec set 4 will also be used for any connections between region 4 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 4 for region 4 to region 1connectivity.

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

 Source Region: 4     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    4     y    NoLimit                                             n       t
 2
 3
 4    4                                                                 all
 5
```

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 4 will be used for connections between region 1 and region 4 as noted previously. In the shared test environment, network region 1 was in place prior to adding the Verizon test environment and already used **Authoritative Domain** “avayalab.com”. Where necessary, Session Manager or the ASBCE will adapt the domain from “avayalab.com” to “adevc.avaya.globalipcom.com”.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avayalab.com
    Name: Enterprise
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 4, and that codec set 4 will be used for any connections between region 4 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    1     y    NoLimit                                              n       t
 3
 4    4     y    NoLimit                                              n       t
```

5.6. IP Codec Sets

The following screen shows the configuration for codec set 4, the codec set configured to be used for calls within region 4 and for calls between region 1 and region 4. In general, an IP codec set is a list of allowable codecs in priority order. Using the example configuration shown below, all calls to and from the PSTN via the SIP trunks would use G.729A, since G.729A is preferred by both Verizon and the Avaya ip-codec-set. Any calls using this same codec set that are between devices capable of the G.722-64K codec (e.g., Avaya 9600-Series IP Telephone) can use G.722. Note that if G.711MU is omitted from the list of allowed codecs in ip-codec-set 4, calls from Verizon that are answered by Avaya Modular Messaging will use G450 VoIP resources to convert from G.729a (facing Verizon) to G.711MU (facing Modular Messaging). If G.711MU is included in ip-codec-set 4, then calls from Verizon that are answered by Modular Messaging will not use G450 VoIP resources, but rather be "ip-direct" using G.711MU from Modular Messaging to the inside of the ASBCE. Include G.711MU in the ip-codec-set if fax will be used.

```
change ip-codec-set 4                                           Page   1 of   2
                           IP Codec Set
    Codec Set: 4

    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:
```

On **Page 2** of the form:

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

```
change ip-codec-set 4                                           Page   2 of   2
                           IP Codec Set

                               Allow Direct-IP Multimedia? n

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

The following screen shows the configuration for codec set 1. This default configuration for codec set 1, using G.711MU, is used for Avaya Modular Messaging 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: 1.722-64K                   2        20
 2: G.711MU          n          2        20
 3: G.729A           n          2        20
 4:
```

5.7. SIP Signaling Groups

This section illustrates the configuration of the SIP Signaling Groups. Each signaling group has a **Group Type** of "sip", a **Near-end Node Name** of "procr", and a **Far-end Node Name** of "ASM". In the example screens, the **Transport Method** for all signaling groups is "tcp". In production, TLS transport between Communication Manager and Session Manager can be used. The **Enable Layer 3 Test** field is enabled on each of the signaling groups to allow Communication Manager to maintain the signaling group using the SIP OPTIONS method. Fields that are not referenced in the text below can be left at default values, including **DTMF over IP** set to "rtp-payload", which corresponds to RFC 2833.

The following screen shows signaling group 5. Signaling group 5 will be used for processing PSTN calls to / from Verizon via Session Manager. The **Far-end Network Region** is configured to region 4. Port 5060 has been configured as both the **Near-end Listen Port** and **Far-end Listen Port**. Session Manager will be configured to direct calls arriving from the PSTN with Verizon DID numbers to a route policy that uses a SIP entity link to Communication Manager specifying port 5060. The use of different ports is one means to allow Communication Manager to distinguish different types of calls arriving from the same Session Manager. In the sample configuration, the **Peer Detection Enabled** field was set to "n". Other parameters may be left at default values. Note that the **Alternate Route Timer** that defaults to 6 seconds has been changed to 12 seconds, this timer 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.

```
display signaling-group 5
                                SIGNALING GROUP

 Group Number: 5             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
       Q-SIP? n                                             SIP Enabled LSP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? n  Peer Server: Others



   Near-end Node Name: procr                 Far-end Node Name: ASM
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 4

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): 12
```

The following screen shows signaling group 6, 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 "ASM", the node name of 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. The **Far-end Domain** is set to "avayalab.com" matching the configuration in place prior to adding the Verizon IP SIP Trunking configuration.

```
change signaling-group 6                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 6             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
       Q-SIP? n                                             SIP Enabled LSP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                 Far-end Node Name: ASM
 Near-end Listen Port: 5070                Far-end Listen Port: 5070
                                        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): 10
```

5.8. SIP Trunk Groups

This section illustrates the configuration of the SIP Trunks Groups corresponding to the SIP signaling groups from the previous section.

The following shows **Page 1** for trunk group 5, which will be used for incoming and outgoing PSTN calls from Verizon. The **Number of Members** field defines how many simultaneous calls are permitted for the trunk group. The **Service Type** field should be set to "public-ntwrk" for the trunks that will handle calls with Verizon. The **Direction** has been configured to "two-way" to allow incoming and outgoing calls only in the sample configuration.

```
change trunk-group 5                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 5                    Group Type: sip           CDR Reports: y
  Group Name: OUTSIDE CALL                COR: 1        TN: 1        TAC: *105
   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: 5
                                                     Number of Members: 255
```

The following screen shows **Page 2** for trunk group 5. 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 Communication Manager default value, which can result in unnecessary SIP messages to re-establish a higher refresh interval for each call.

```
change trunk-group 5                                           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? n  Out? y

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

The following screen shows **Page 3** for trunk group 5. All parameters except those in bold are default values. The **Numbering Format** will use "public" numbering, meaning that the public numbering table would be consulted for any mappings of Communication Manager extensions to alternate numbers to be sent to Session Manager. Optionally, replacement text strings can be configured using the "system-parameters features" screen, such that incoming "private" (anonymous) or "restricted" calls can display an Avaya-configured text string on called party telephones.

```
change trunk-group 5                                            Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: public
                                                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 5. The **PROTOCOL VARIATIONS** page is one reason why it can be advantageous to configure incoming calls from Verizon to arrive on specific signaling groups and trunk groups. 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 Manager can send the History-Info header by setting **Support Request History** to "y", and Session Manager can adapt the History-Info header to the Diversion header using the "VerizonAdapter". In the testing associated with these Application Notes, call redirection testing with Communication Manager sending Diversion Header was completed successfully. Configuration for Communication Manager was then changed, and call redirection testing with Communication Manager sending History-Info and Session Manager adapting to Diversion Header was completed successfully.

```
change trunk-group 5                                              Page   4 of  21
                              PROTOCOL VARIATIONS

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

       Convert 180 to 183 for Early Media? y
 Always Use re-INVITE for Display Updates? n
       Identity for Calling Party Display: P-Asserted-Identity
                             Enable Q-SIP? n
```

The following screen shows **Page 1** for trunk group 6, 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 Modular Messaging, and does not reflect any unique Verizon configuration.

```
change trunk-group 6                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 6                   Group Type: sip           CDR Reports: y
  Group Name: to-ASM6.1                  COR: 1        TN: 1        TAC: *106
   Direction: two-way       Outgoing Display? y
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                            Member Assignment Method: auto
                                                     Signaling Group: 6
                                                   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 6                                              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 Modular Messaging.

```
change trunk-group 6                                          Page   4 of  21
                              PROTOCOL VARIATIONS

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

       Convert 180 to 183 for Early Media? n
 Always Use re-INVITE for Display Updates? 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. Digit manipulation can be performed on the called number, if needed, using **No. Del Dgts** and **Inserted Digits** parameters. Digit manipulation can also be performed by Session Manager.

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-VZ-IP-Trunk
                             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: 5   0                                                             n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                               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 Avaya Aura® Session Manager

Route pattern 6 contains trunk group 3, the "private" tie trunk group to Session Manager. The **Numbering Format**: ***lev0-pvt*** means all calls using this route pattern will use the private numbering table.

```
change route-pattern 6                                            Page   1 of   3
                    Pattern Number: 6   Pattern Name: SIP_Phones
                             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: 6    0                                                             n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                Dgts Format
                                                        Subaddress
 1: y y y y y n  n            rest                               lev0-pvt  none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
```

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 public-unknown-numbering, and incoming call handling treatment for the inbound trunk group).

In the bolded row shown in the example abridged output below, a specific Communication Manager extension (x7689) is mapped to a DID number that is known to Verizon for this SIP Trunk connection (4089908838), when the call uses trunk group 5. 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 public-unknown-numbering 0                                 Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext           Trk       CPN             CPN
Len Code          Grp(s)    Prefix          Len
                                                  Total Administered: 4
 4  7611          5         4089908837      10       Maximum Entries: 9999
 4  7633          5         33176759456     11
 4  7644          5         4089908839      10    Note: If an entry applies to
 4  7689          5         4089908838      10    a SIP connection to Avaya
                                                  Aura(tm) Session Manager,
                                                  the resulting number must
                                                  be a complete E.164 number.
```

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. Various example scenarios for a multi-location network with failover routing are provided in reference [PE]. 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 13035387022, 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
    13035387022           11   11      1        hnpa         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 13035387022
                           ARS ROUTE CHOSEN REPORT
     Location:  1                      Partitioned Group Number:  1

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

 13035387022          11    11       1       hnpa                 all
   Actual Outpulsed Digits by Preference (leading 35 of maximum 42 digit)

 1: 13035387022
```

5.13. Incoming Call Handling Treatment for Incoming Calls

In general, the "incoming call handling treatment" for a trunk group can be used to manipulate the digits received for an incoming call if necessary. Since Session Manager is present, Session Manager can also be used to perform digit conversion and digit manipulation; Communication Manager incoming call handling table may not be necessary. If the DID number sent by Verizon is unchanged by Session Manager, then the DID number can be mapped to an extension using the incoming call handling treatment of the receiving trunk group. As an example, the following screen illustrates a conversion of DID number 4089908838 to extension 7689. The EMEA testing using 33176759457 is also mapped to extension 7689. Both Session Manager digit conversion and Communication Manager incoming call handling treatment methods were tested successfully.

```
change inc-call-handling-trmt trunk-group 5                       Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    11 33176759456      11  7611
 public-ntwrk    11 33176759457      11  7689
 public-ntwrk    10 4089908837       10  7611
 public-ntwrk    10 4089908838       10  7689
```

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 7689. Use the command ***change off-pbx-telephone station mapping x*** where ***x*** is Communication Manager station (e.g. 7689).

- **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., 3035387022)
- **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 7689                      Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number     Trunk        Config  Dual
 Extension                  Prefix                      Selection    Set     Mode
 7689              EC500         -      3035387022        ars         1
```

5.15. Saving Avaya Aura® Communication Manager Configuration Changes

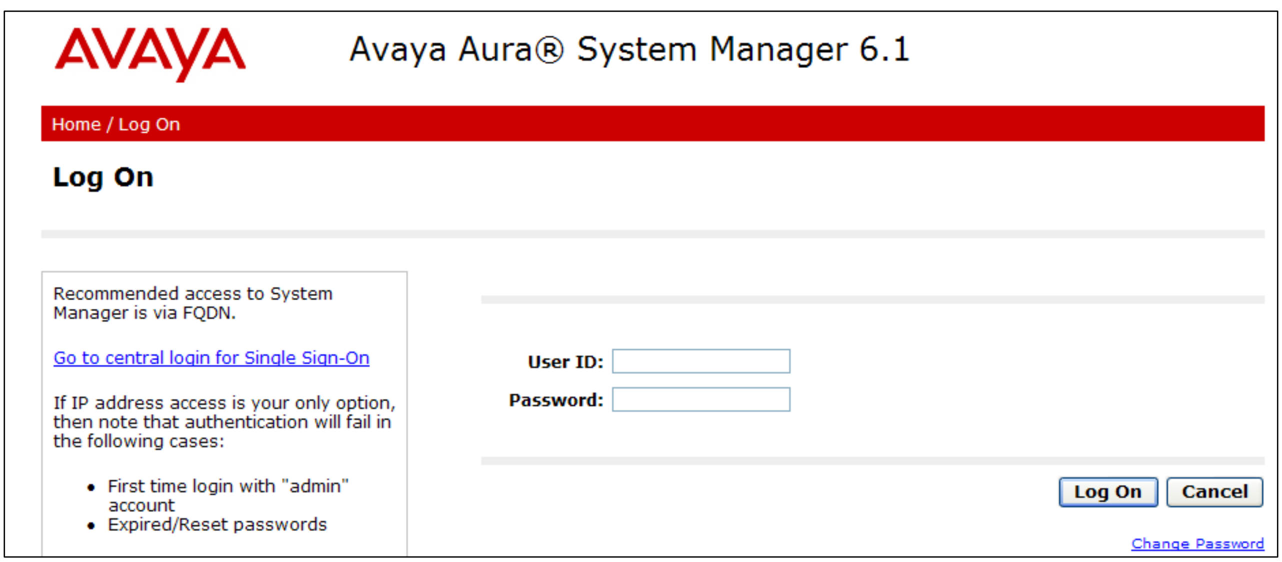
The command ***save translation all*** can be used to save the configuration.

6. Configure Avaya Aura® Session Manager

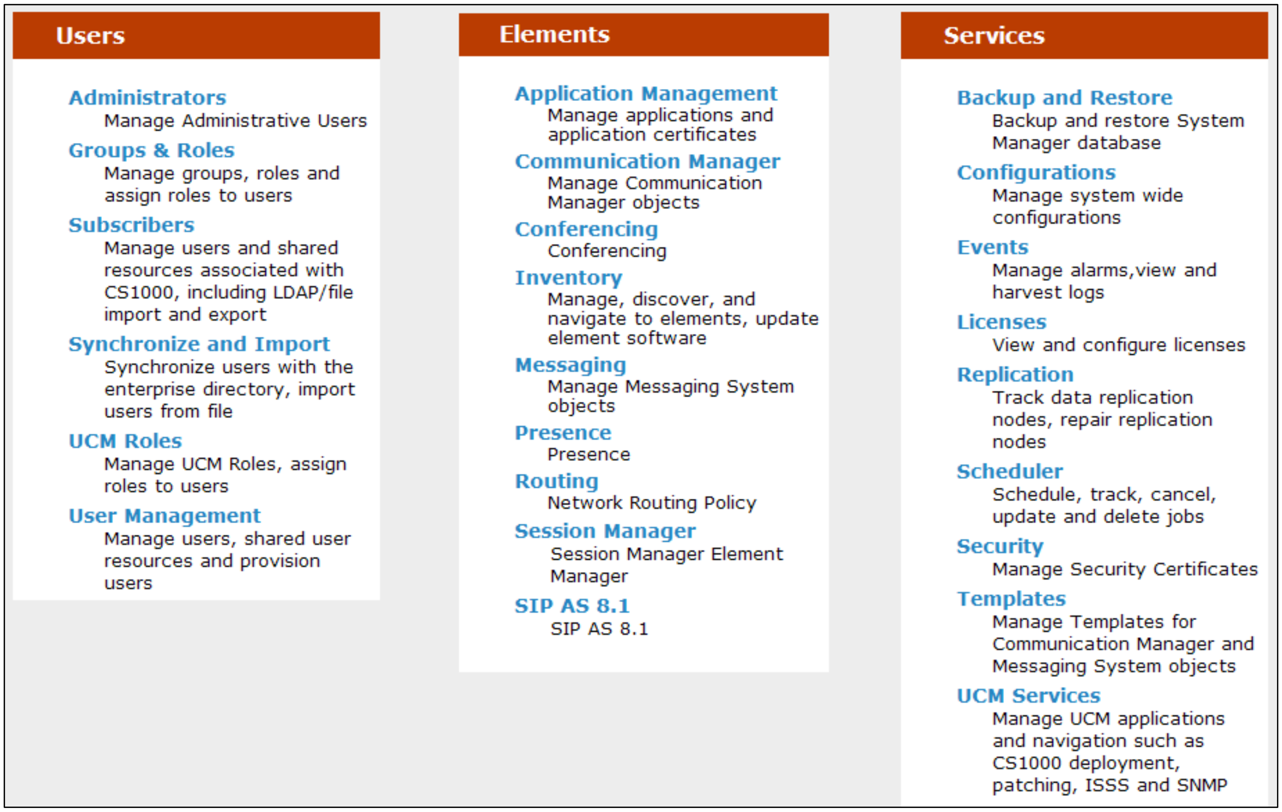
This section illustrates relevant aspects of the Session Manager configuration used in the verification of these Application Notes.

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

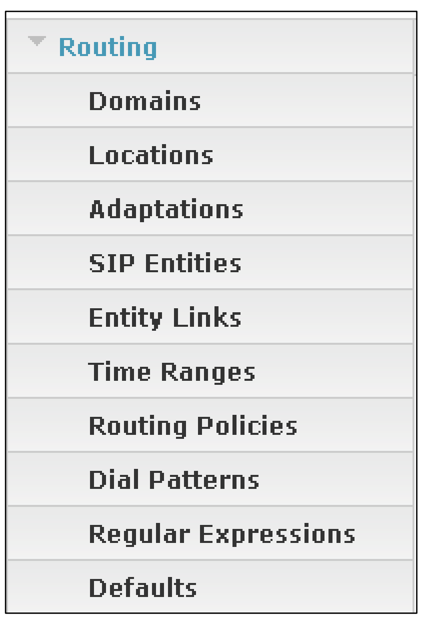
Session Manager is managed via System Manager. Using a web browser, access “https://<ip-addr of System Manager>/SMGR”. In the **Log On** screen, enter appropriate **User ID** and **Password** and press the **Log On** button as shown in the example System Manager 6.1 **Log On** screen below.



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



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



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

Introduction to Network Routing Policy

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

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

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

Step 2: Create "Locations"

Step 3: Create "Adaptations"

Step 4: Create "SIP Entities"

- SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"
- Create all "other SIP Entities" (Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)
- Assign the appropriate "Locations", "Adaptations" and "Outbound Proxies"

Step 5: Create the "Entity Links"

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

Step 6: Create "Time Ranges"

- Align with the tariff information received from the Service Providers

Step 7: Create "Routing Policies"

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

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

Step 8: Create "Dial Patterns"

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

Step 9: Create "Regular Expressions"

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

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

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

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

"Dial Pattern driven approach to define Routing Policies"

That means (with regard to steps listed above):

Step 7: "Routing Polices" are defined

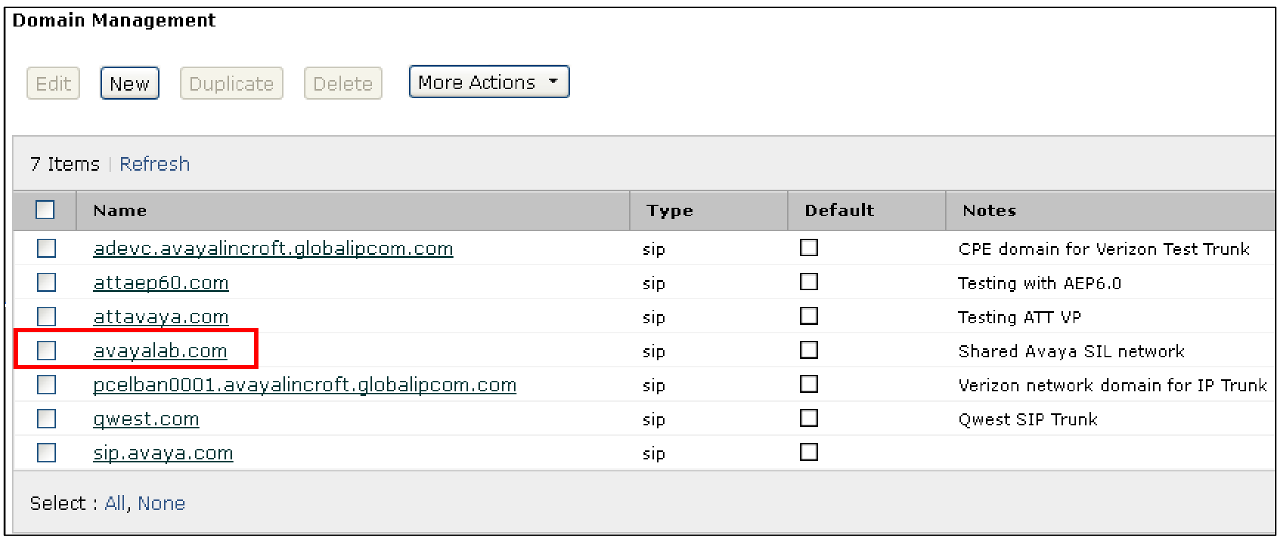
Step 8: "Dial Patterns" are defined and assigned to "Routing Policies" and "Locations" (one step)

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

6.1. Domains

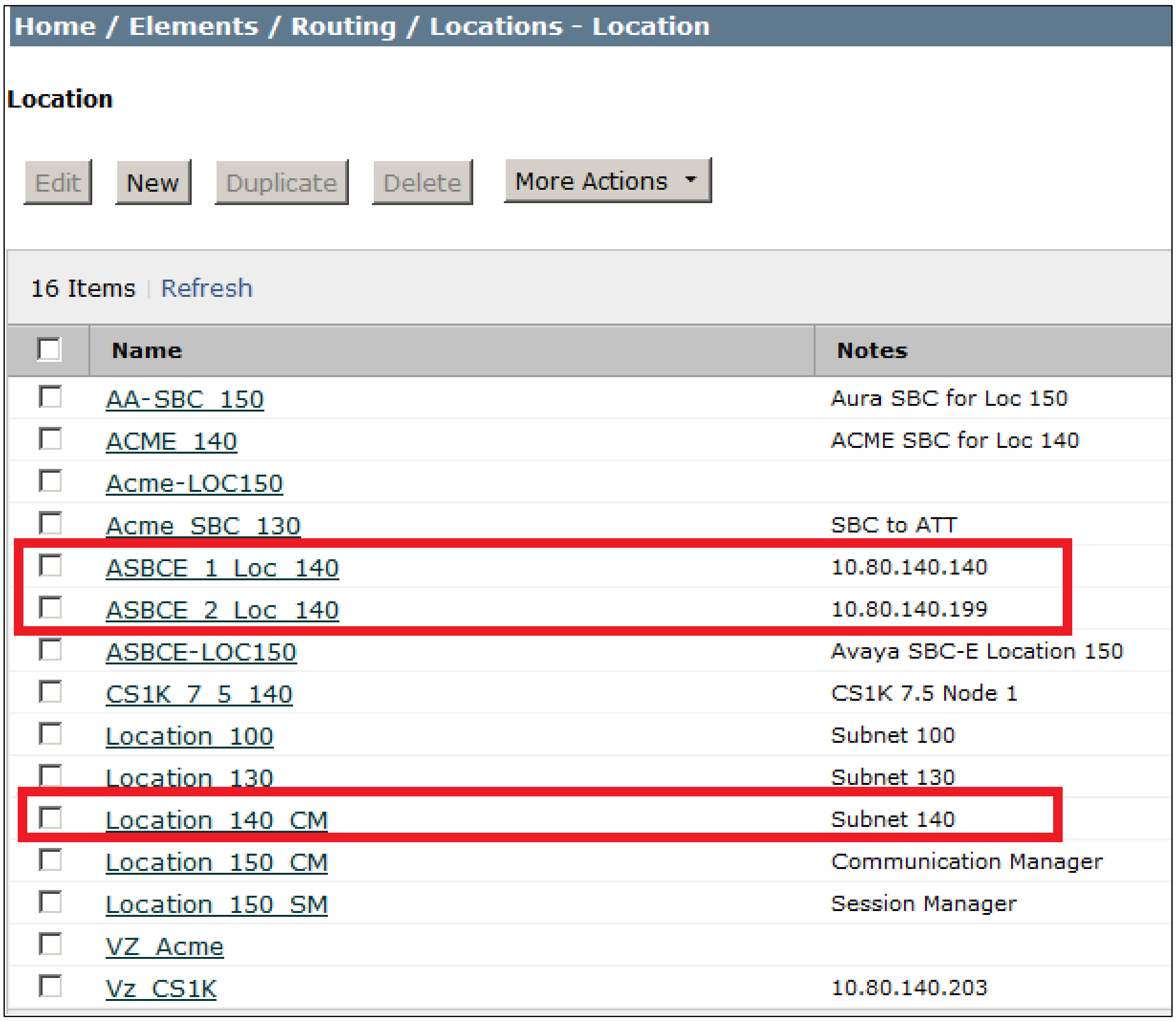
To view or change SIP domains, select **Routing → Domains**. Click on the checkbox next to the name of the SIP domain and **Edit** to edit an existing domain, or the **New** button to add a domain. Click the **Commit** button after changes are completed.

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 and will be manipulated at the SBC before sending traffic on to the Verizon network.

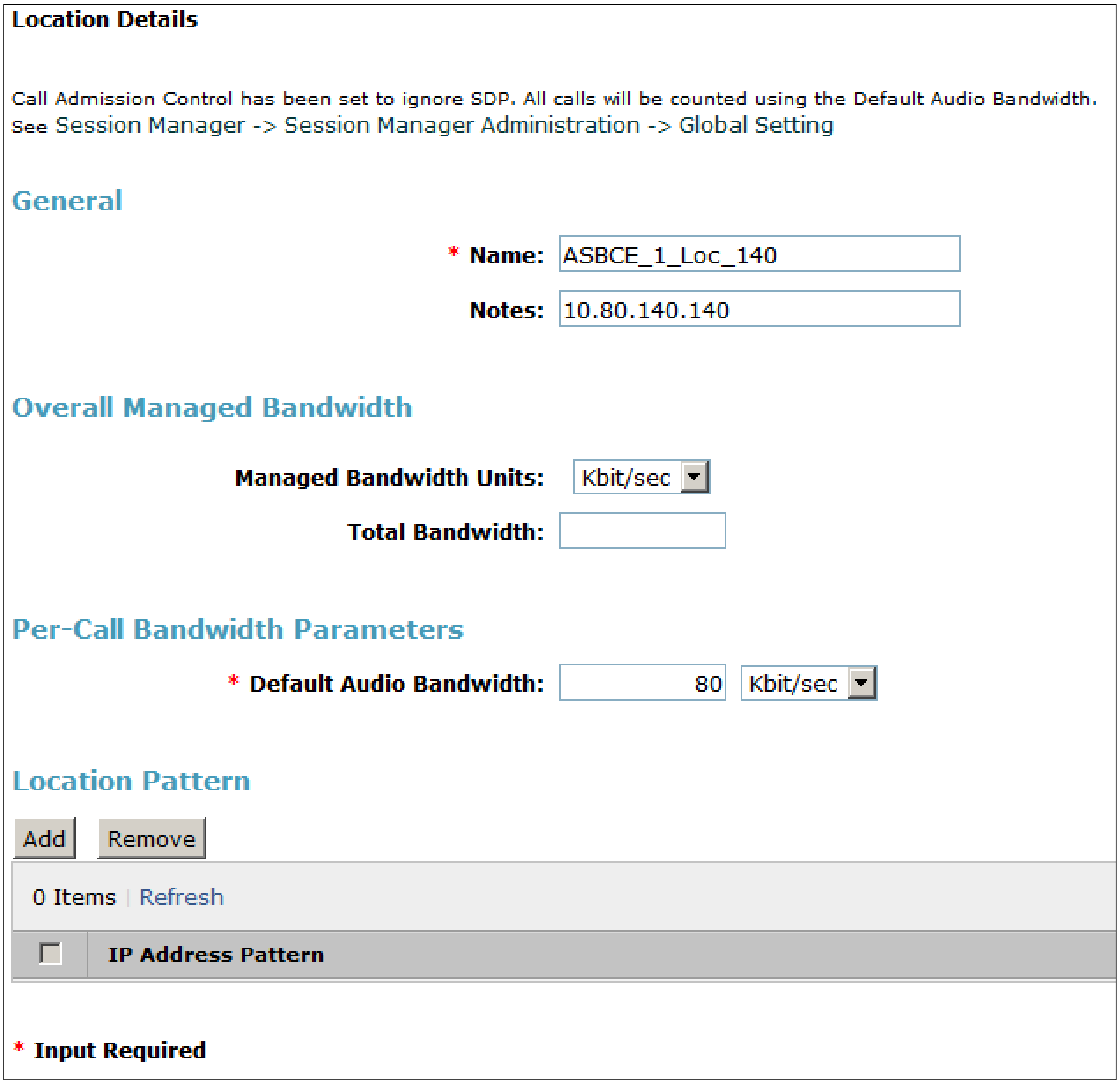


6.2. Locations

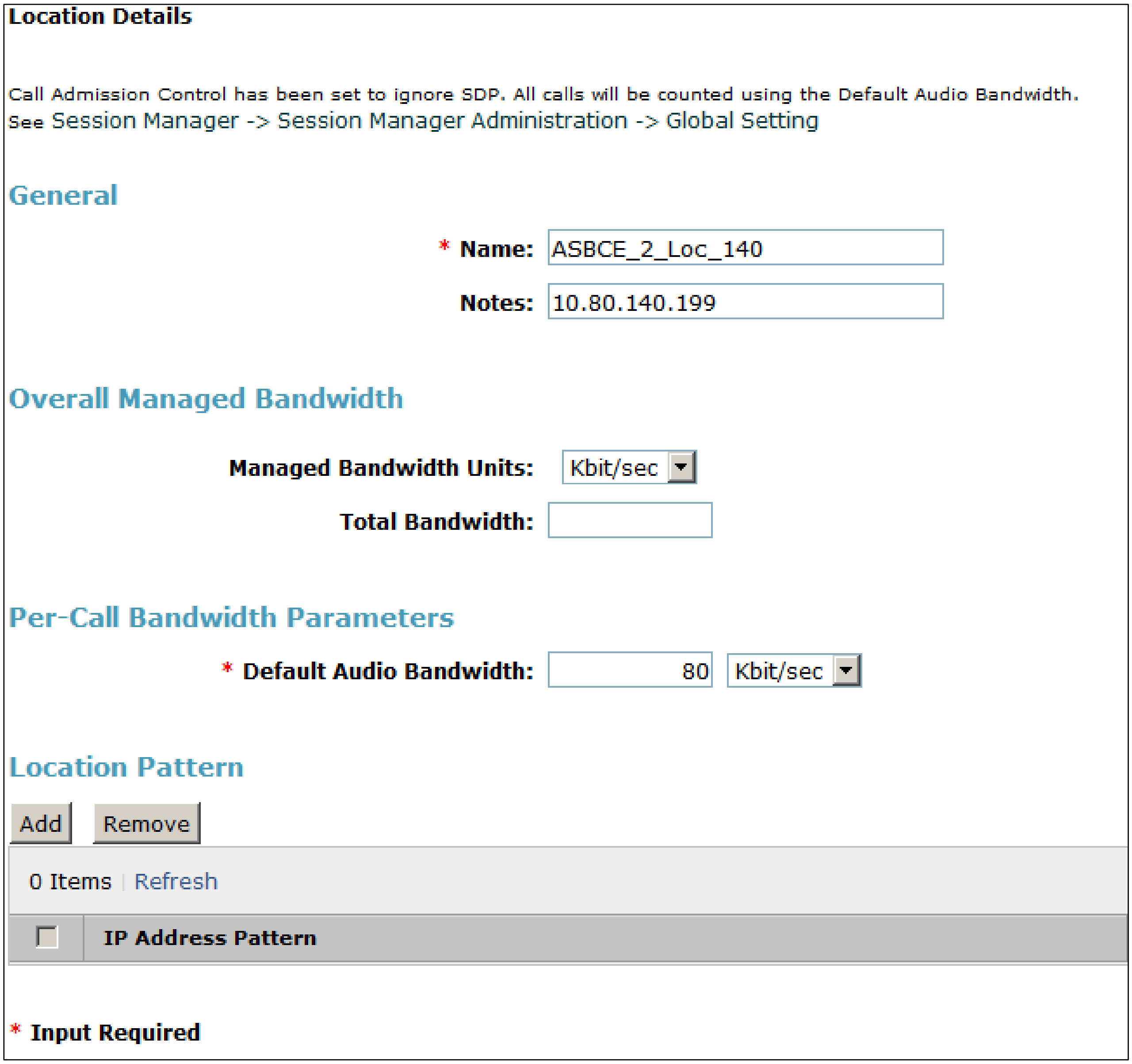
To view or change locations, select **Routing → Locations**. The following screen shows an abridged list of configured locations. Click on the checkbox corresponding to the name of a location and **Edit** to edit an existing location, or the **New** button to add a location. Click the **Commit** button after changes are completed. Assigning unique locations can allow Session Manager to perform location-based routing, bandwidth management, and call admission control. The locations used in this configuration are outlined in red. There were two ASBCEs used for the 2-CPE configuration, but configurations are identical except for IP Addresses so only one configuration will be shown.



The following image shows the top portion of the screen for the location details for the location named “ASBCE_1_Loc_140”, corresponding to the ASBCE relevant to these Application Notes. Later, the location with name “ASBCE_1_Loc_140” will be assigned to the corresponding SIP Entity.



The following image shows the location details for the location named "ASBCE_2_Loc_140". In the sample configuration, other location parameters retained default values.

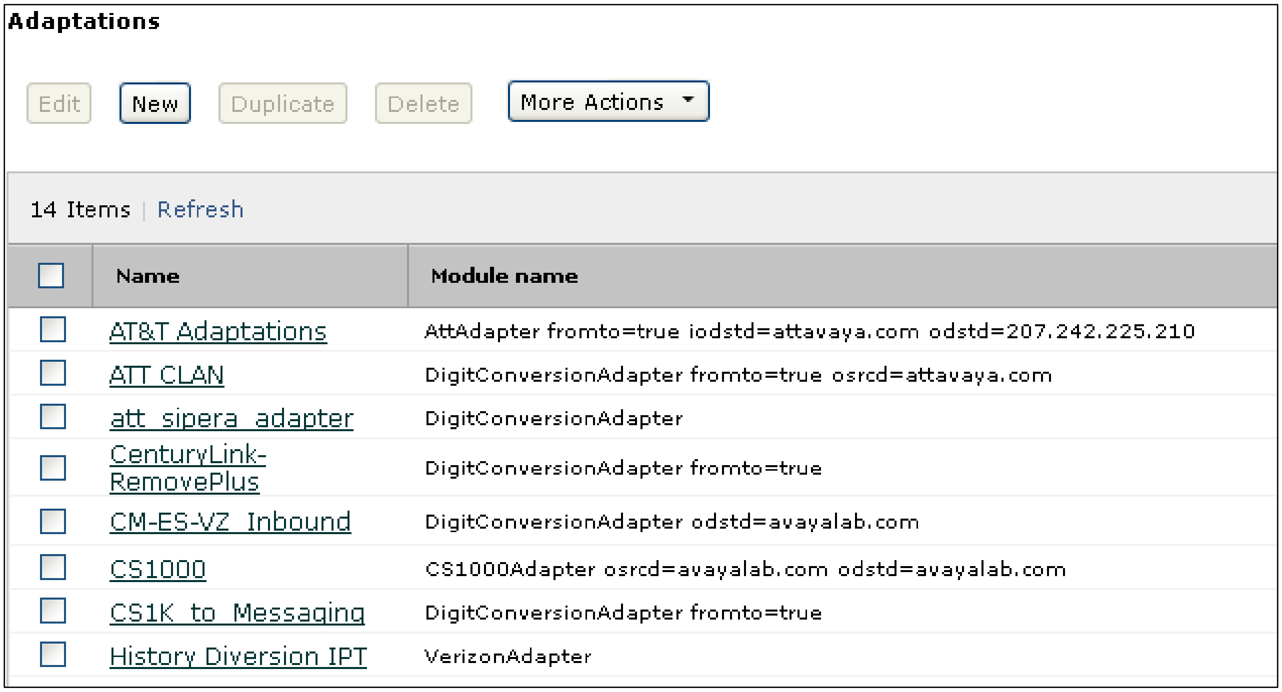


If desired, additional locations can be configured with IP Address Patterns corresponding to other elements in the configuration.

6.3. Adaptations

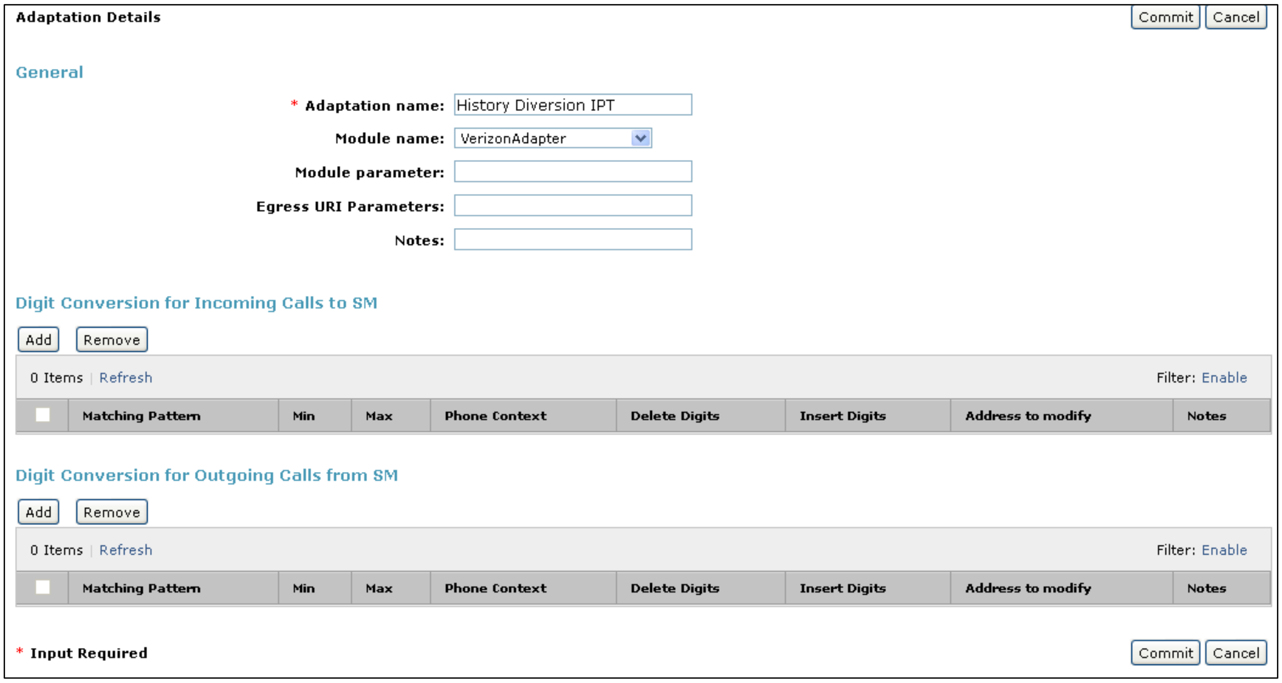
To view or change adaptations, select **Routing → Adaptations**. Click on the checkbox corresponding to the name of an adaptation and **Edit** to edit an existing adaptation, or the **New** button to add an adaptation. Click the **Commit** button after changes are completed (not shown).

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



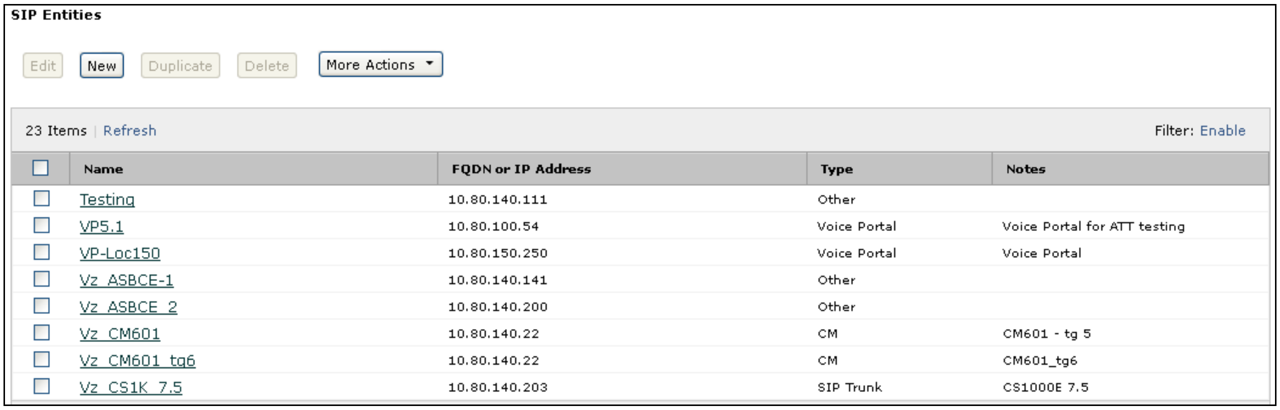
Name	Module name
AT&T Adaptations	AttAdapter fromto=true iodstd=attavaya.com odstd=207.242.225.210
ATT CLAN	DigitConversionAdapter fromto=true osrcd=attavaya.com
att_sipera_adapter	DigitConversionAdapter
CenturyLink-RemovePlus	DigitConversionAdapter fromto=true
CM-ES-VZ_Inbound	DigitConversionAdapter odstd=avayalab.com
CS1000	CS1000Adapter osrcd=avayalab.com odstd=avayalab.com
CS1K_to_Messaging	DigitConversionAdapter fromto=true
History Diversion IPT	VerizonAdapter

The following screen shows the adaptation details. The adapter named "History_Diversion_IPT" will later be assigned to the SIP Entity for the ASBCE, specifying that all communication from Session Manager to the ASBCE will use this adapter. As mentioned in Section 3.1, this adaptation uses the "VerizonAdapter" and specifies that if Communication Manager sends information in the History Info field to convert into a Diversion Header as expected by Verizon. The FQDN on all traffic is currently avayalab.com and could be adapted here to the FQDN or IP address known to Verizon, but will be changed on the ASBCE in this configuration.

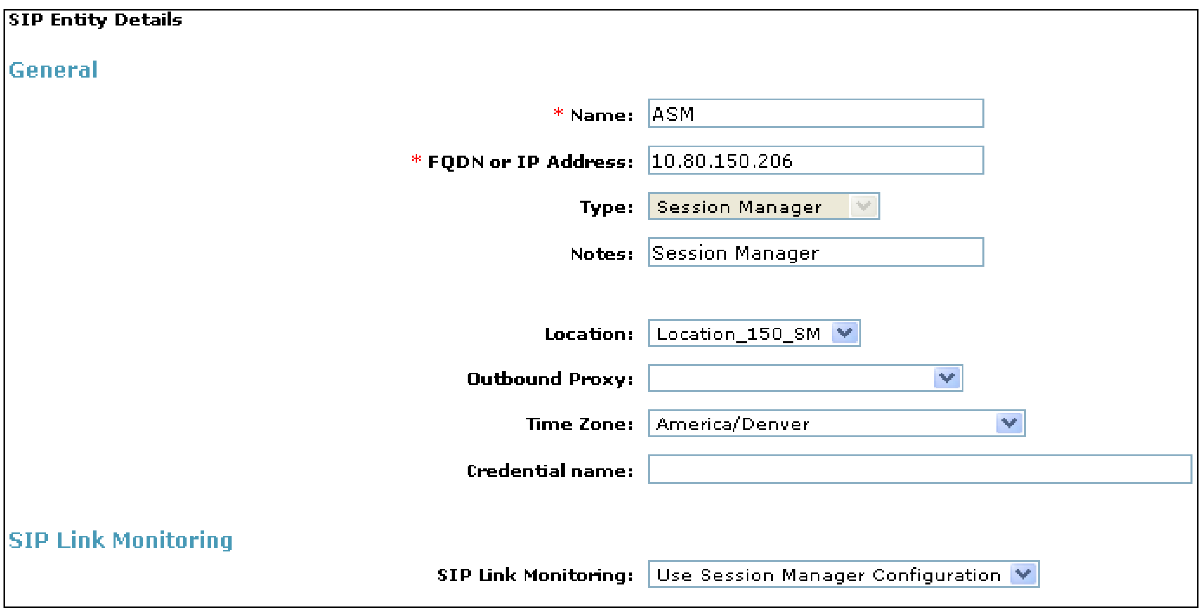


6.4. SIP Entities

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 a portion of the list of configured SIP entities. In this screen, the SIP Entities named "Avaya-SBCE-1", "Avaya-SBCE-2", "Vz_CM601", as well as "ASM" (not shown) are relevant to these Application Notes.



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 (10.80.150.206), 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 Session Manager from the **Location** drop-down menu. In the shared test environment, Session Manager used location "Location_150_SM". 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

20 Items | Refresh — Filter: Enable

	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
	ASM	TCP	* 5060	Vz_ASBCE_2	* 5060	Trusted
	ASM	TCP	* 5060	Vz_ASBCE-1	* 5060	Trusted
	ASM	TCP	* 5060	Vz_CM601	* 5060	Trusted
	ASM	TCP	* 5070	Vz_CM601_tg6	* 5070	Trusted
	ASM	TCP	* 5060	Vz_CS1K_7.5	* 5060	Trusted

Select : All, None — < Previous | Page 4 of 4 | Next >

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

Port

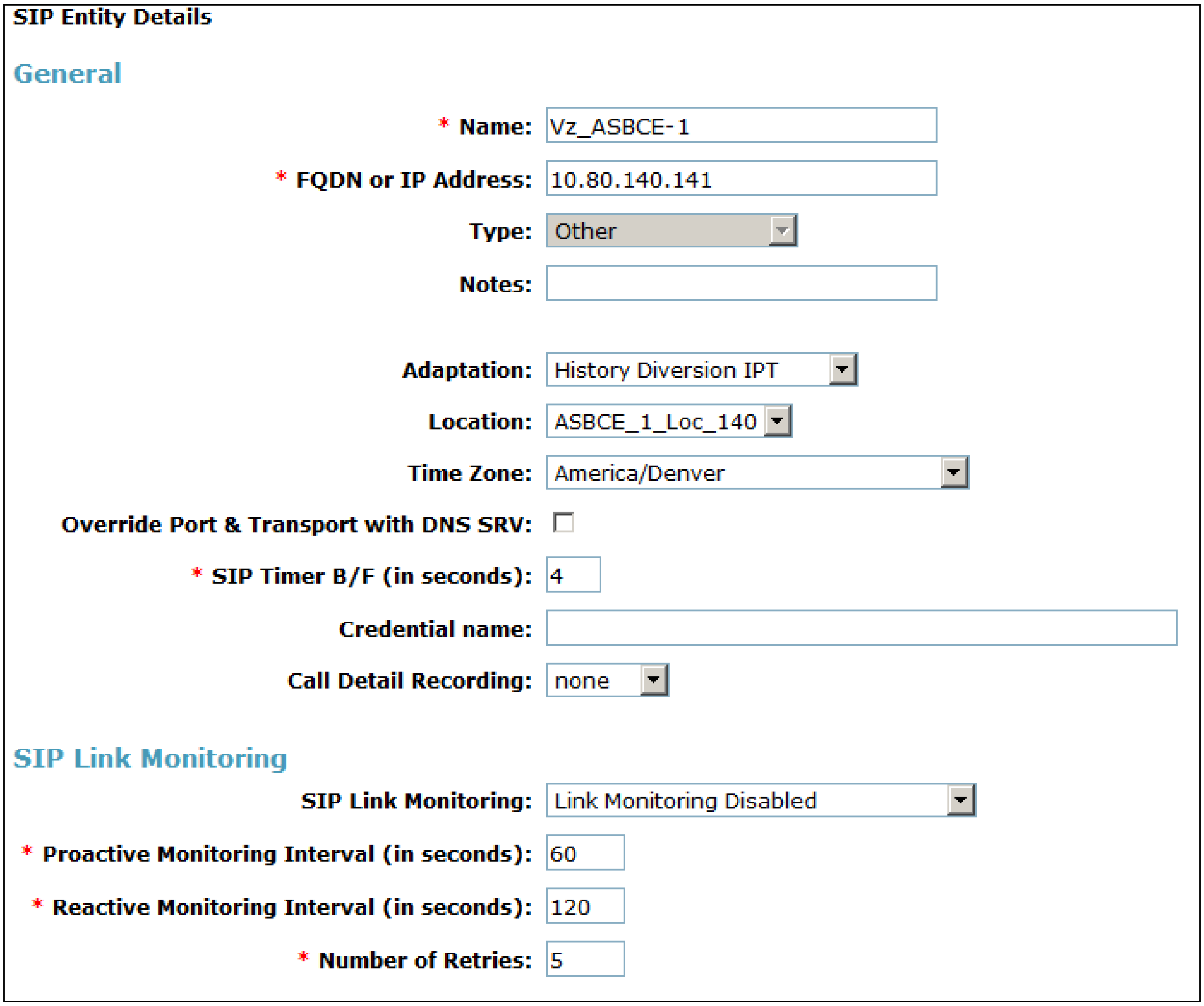
Add | Remove

7 Items | Refresh

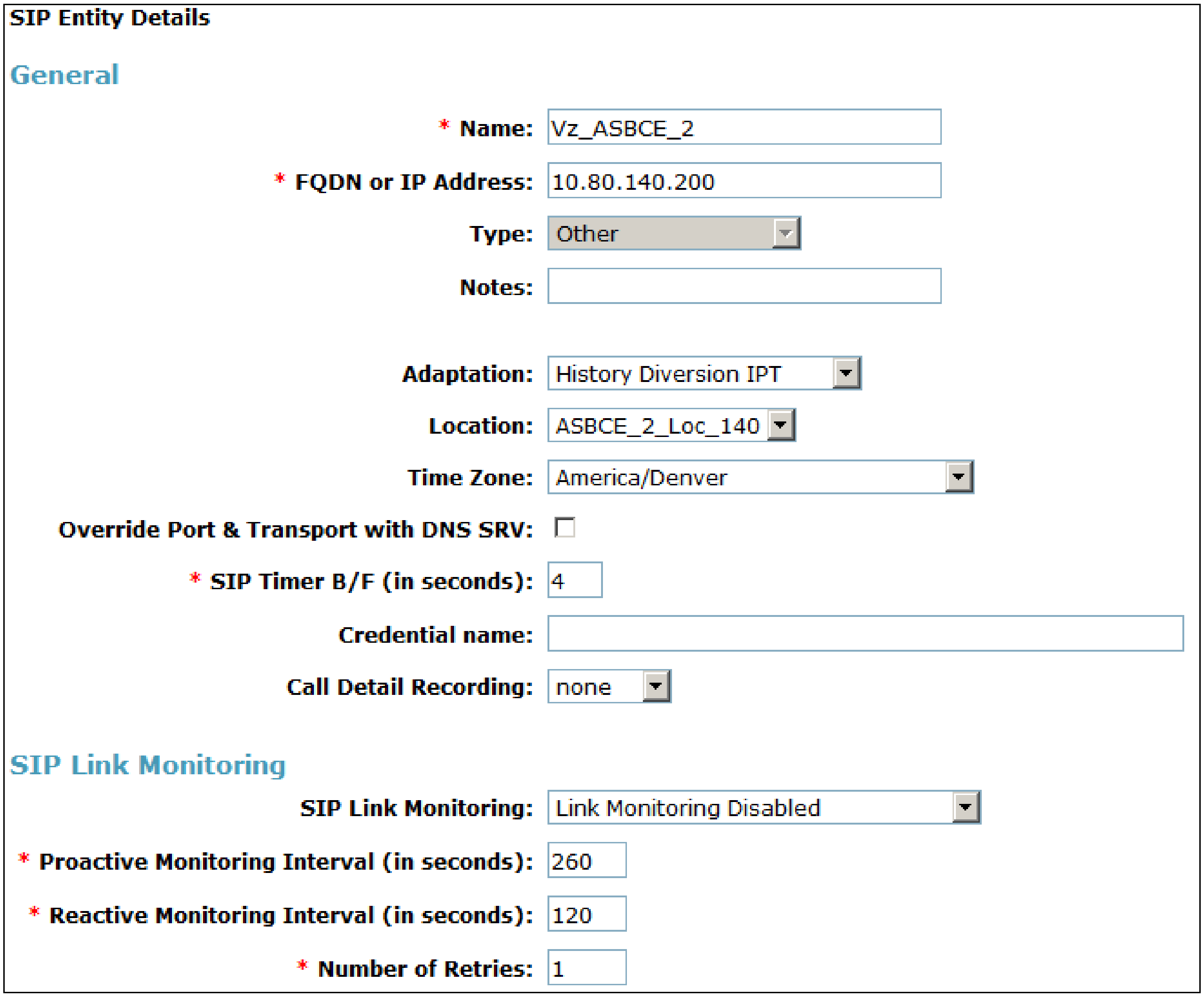
	Port	Protocol	Default Domain	Notes
	5060	UDP	avayalab.com	
	5060	TCP	avayalab.com	
	5061	TLS	avayalab.com	
	5070	TCP	avayalab.com	
	5080	TCP	avayalab.com	
	5081	TLS	avayalab.com	
	5090	TCP	attavaya.com	

Select : All, None

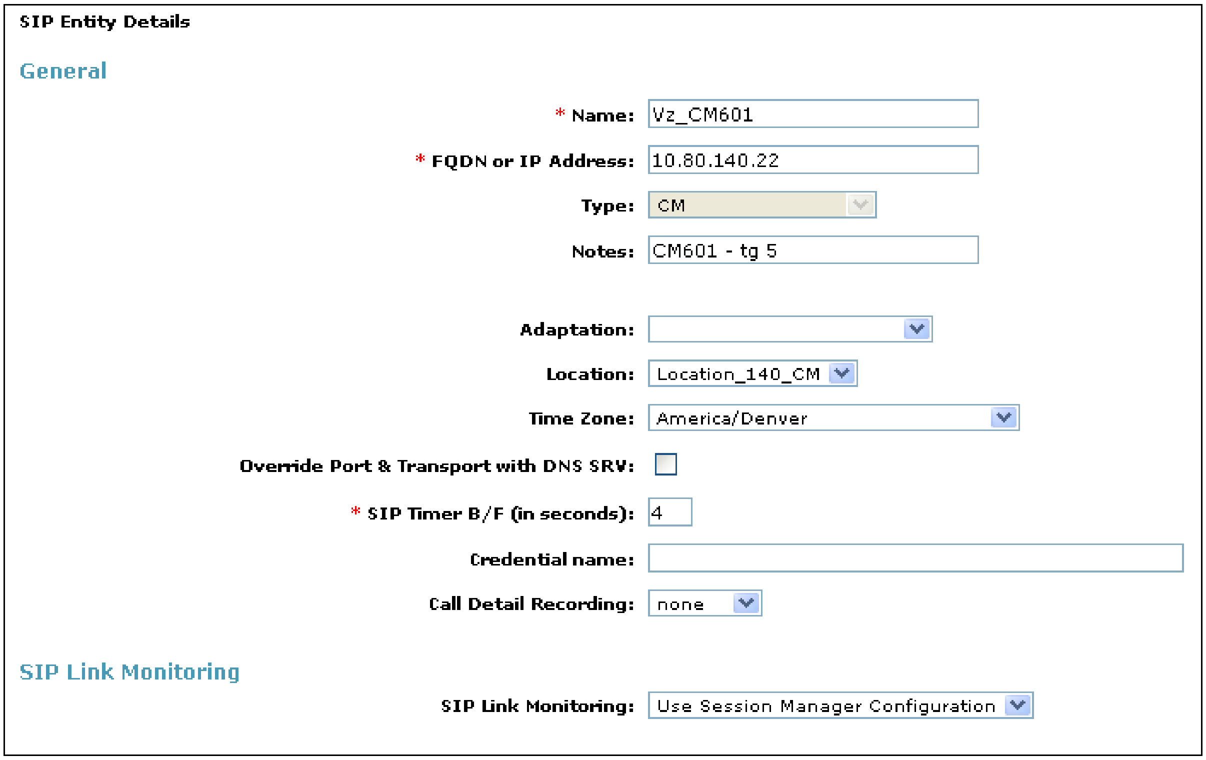
The following screen shows the upper portion of the **SIP Entity Details** corresponding to "Vz_ASBCE-1" entity. The **FQDN or IP Address** field is configured with the ASBCE inside IP Address (10.80.140.141). "Other" is selected from the **Type** drop-down menu for ASBCE SIP Entities. This ASBCE has been assigned to **Location** "ASBCE_1_Loc_140", and the "History_Diversion_IPT" adapter is applied. Other parameters (not shown) retain default values.



The following screen shows the upper portion of the **SIP Entity Details** corresponding to "Vz_ASBCE-2". The **FQDN or IP Address** field is configured with the ASBCE inside IP Address (10.80.140.200). "Other" is selected from the **Type** drop-down menu for ASBCE SIP Entities. This ASBCE has been assigned to **Location** "ASBCE_2_Loc_140", and the "History_Diversion_IPT" adapter is applied. Other parameters (not shown) retain default values.


SIP Entity Details
General
* Name: Vz_ASBCE_2
* FQDN or IP Address: 10.80.140.200
Type: Other
Notes:
Adaptation: History Diversion IPT
Location: ASBCE_2_Loc_140
Time Zone: America/Denver
Override Port & Transport with DNS SRV:
* SIP Timer B/F (in seconds): 4
Credential name:
Call Detail Recording: none
SIP Link Monitoring
SIP Link Monitoring: Link Monitoring Disabled
* Proactive Monitoring Interval (in seconds): 260
* Reactive Monitoring Interval (in seconds): 120
* Number of Retries: 1

The following screen shows a portion of the **SIP Entity Details** corresponding to a Communication Manager SIP Entity named “Vz_CM6.0.1” The **FQDN or IP Address** field contains the IP Address of the “processor Ethernet” (10.80.140.22). 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.

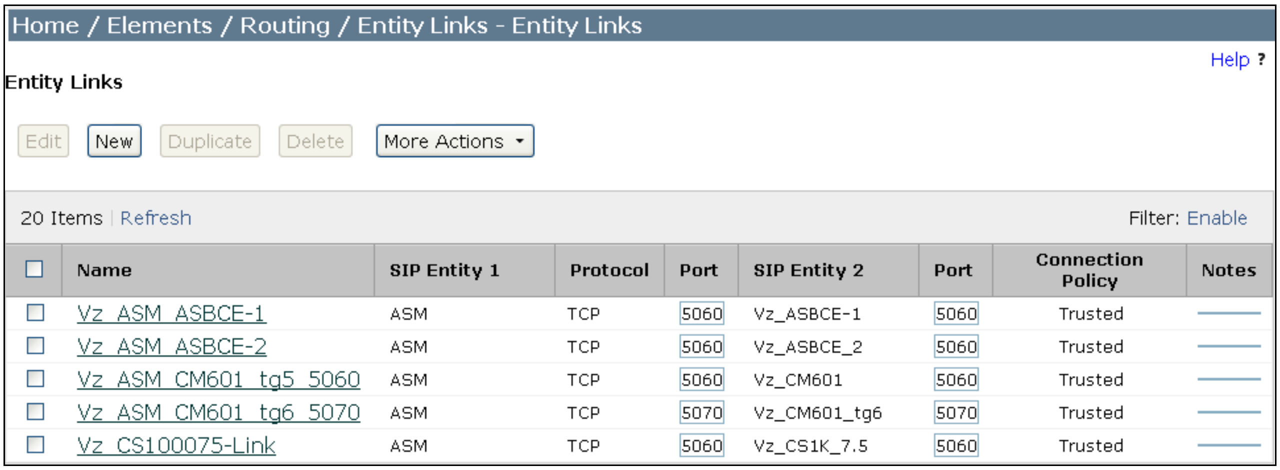


6.5. Entity Links

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.

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

The following screen shows a list of configured links. In the screen below, the links named "Vz_ASM_ASBCE-1", "Vz_ASM_ASBCE-2" and "Vz_ASM_CM601_tg5_5060" 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**. Note that there are multiple SIP Entity Links, using different TCP ports, linking the same "ASM" with the processor Ethernet of Communication Manager. For example, for one link, named "Vz_ASM_CM601_tg5_5060", both entities use TCP and port 5060. For the entity link used by Verizon IP Trunk named "Vz_ASM_CM601_tg6_5070", both entities use TCP and port 5070.



Home / Elements / Routing / Entity Links - Entity Links

Help ?

Entity Links

Edit | New | Duplicate | Delete | More Actions

20 Items | Refresh

Filter: Enable

	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
☐	Vz_ASM_ASBCE-1	ASM	TCP	5060	Vz_ASBCE-1	5060	Trusted	
☐	Vz_ASM_ASBCE-2	ASM	TCP	5060	Vz_ASBCE_2	5060	Trusted	
☐	Vz_ASM_CM601_tg5_5060	ASM	TCP	5060	Vz_CM601	5060	Trusted	
☐	Vz_ASM_CM601_tg6_5070	ASM	TCP	5070	Vz_CM601_tg6	5070	Trusted	
☐	Vz_CS100075-Link	ASM	TCP	5060	Vz_CS1K_7.5	5060	Trusted	

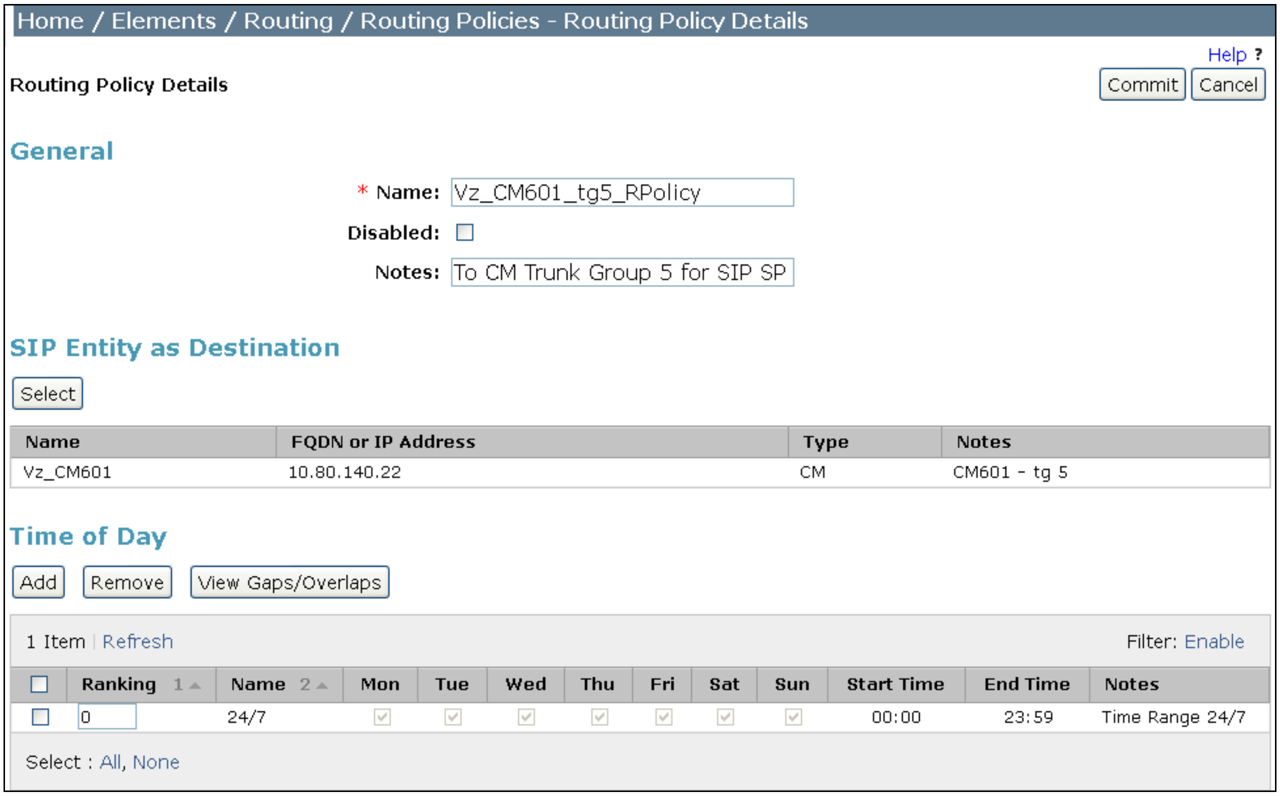
The link named "Vz_ASM_CM601_tg5_5060" links Session Manager "ASM" with Communication Manager processor Ethernet. However, this link uses port 5060 for both entities in the link. This link was created to allow Communication Manager to distinguish calls from Verizon IP Trunk from other calls that arrive from the same Session Manager. Other methods of distinguishing traffic could be used, if desired.

The link named "Vz_ASM_CM601_tg6_5070" also links Session Manager "ASM" with Communication Manager processor Ethernet. However, this link uses port 5070 for both entities in the link. This link existed in the configuration prior to adding the Verizon IP Trunk related configuration.

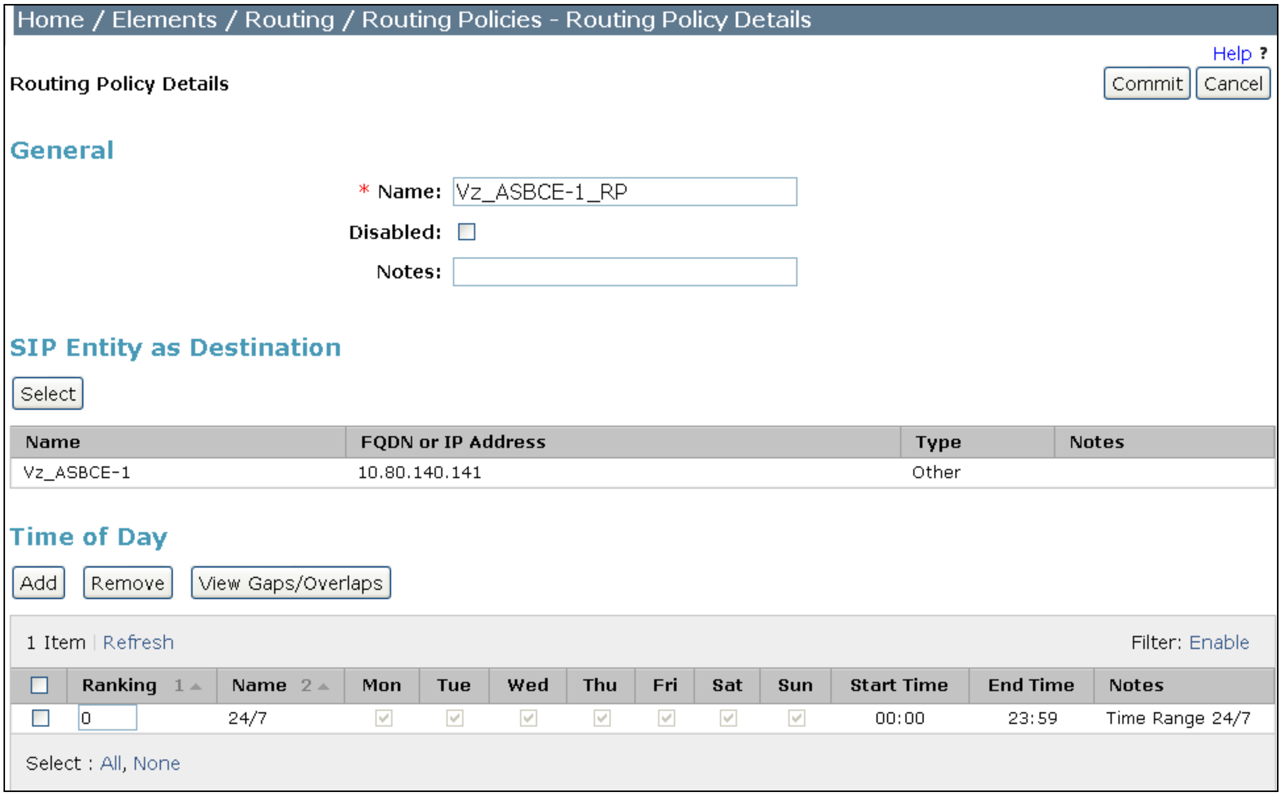
6.6. Routing Policies

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 "Vz_CM601_tg5_RPolicy" associated with incoming DID calls from Verizon IP Trunk to Communication Manager. Observe the **SIP Entity as Destination** is the entity named "Vz_CM601" which uses Communication Manager processor Ethernet IP Address (10.80.140.22).

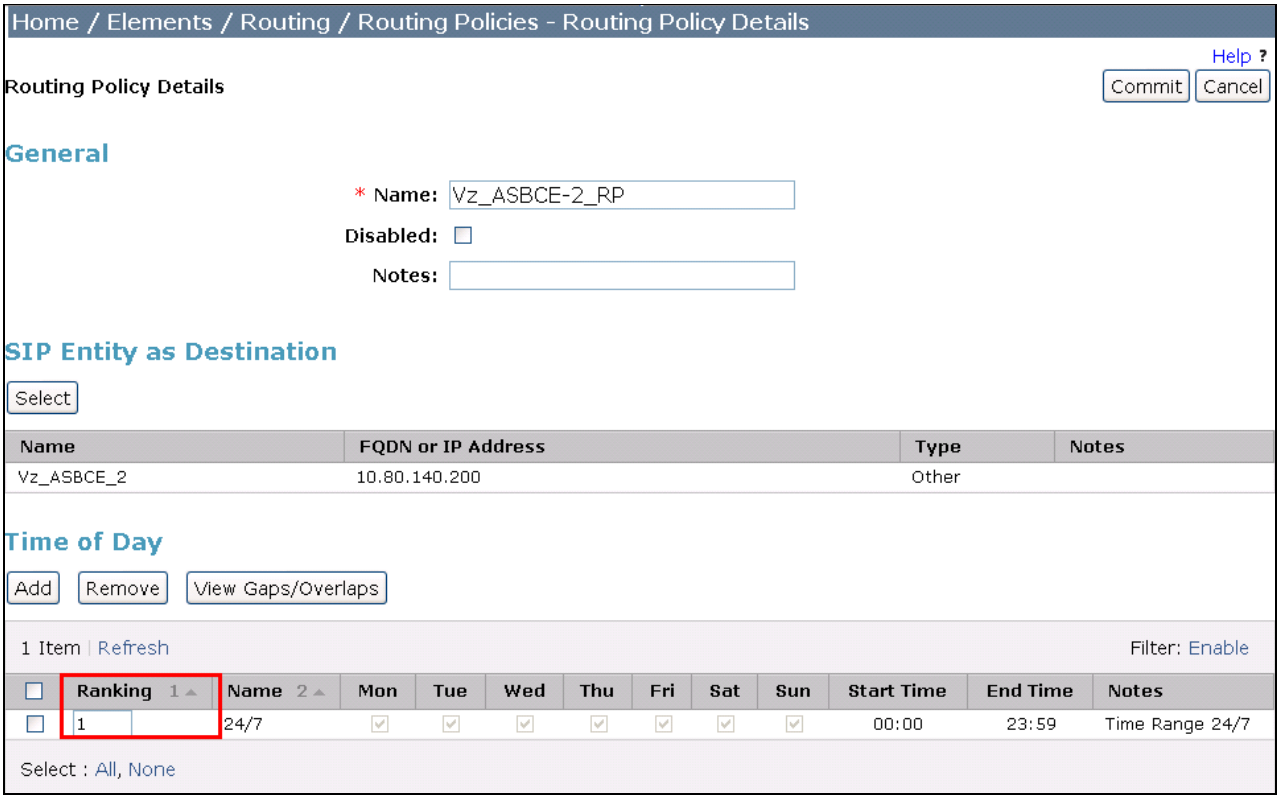

Home / Elements / Routing / Routing Policies - Routing Policy Details
Help ?
Routing Policy Details
Commit
Cancel
General
* Name: Vz_CM601_tg5_RPolicy
Disabled:
Notes: To CM Trunk Group 5 for SIP SP
SIP Entity as Destination
Select
Name
FQDN or IP Address
Type
Notes
Vz_CM601
10.80.140.22
CM
CM601 - tg 5
Time of Day
Add
Remove
View Gaps/Overlaps
1 Item | Refresh
Filter: Enable
Ranking 1
Name 2
Mon
Tue
Wed
Thu
Fri
Sat
Sun
Start Time
End Time
Notes
0
24/7
00:00
23:59
Time Range 24/7
Select : All, None

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



The **Routing Policy Details** for ASBCE-2 are the same except for the **SIP Entity as Destination** (shown below). However, observe the **SIP Entity as Destination** is the entity named "Avaya-SBCE-1" above. In the **Time of Day** area, note that a **Ranking** can be configured. To allow the "Avaya-SBCE-2" to receive calls from Session Manager even when the "Avaya-SBCE-1" is operational, the default rank of 0 (also assigned to "Avaya-SBCE-1") can be retained.

If it is intended that "Avaya-SBCE-1" should always be tried by Session Manager before "Avaya-SBCE-2", the rank of "Avaya-SBCE-2" can be changed to 1 as shown below. Both the "load sharing" approach where "Avaya-SBCE-1" and "Avaya-SBCE-2" use the same rank, and the strict rank order priority of "Avaya-SBCE-1" over "Avaya-SBCE-2" were successfully tested in the sample configuration.

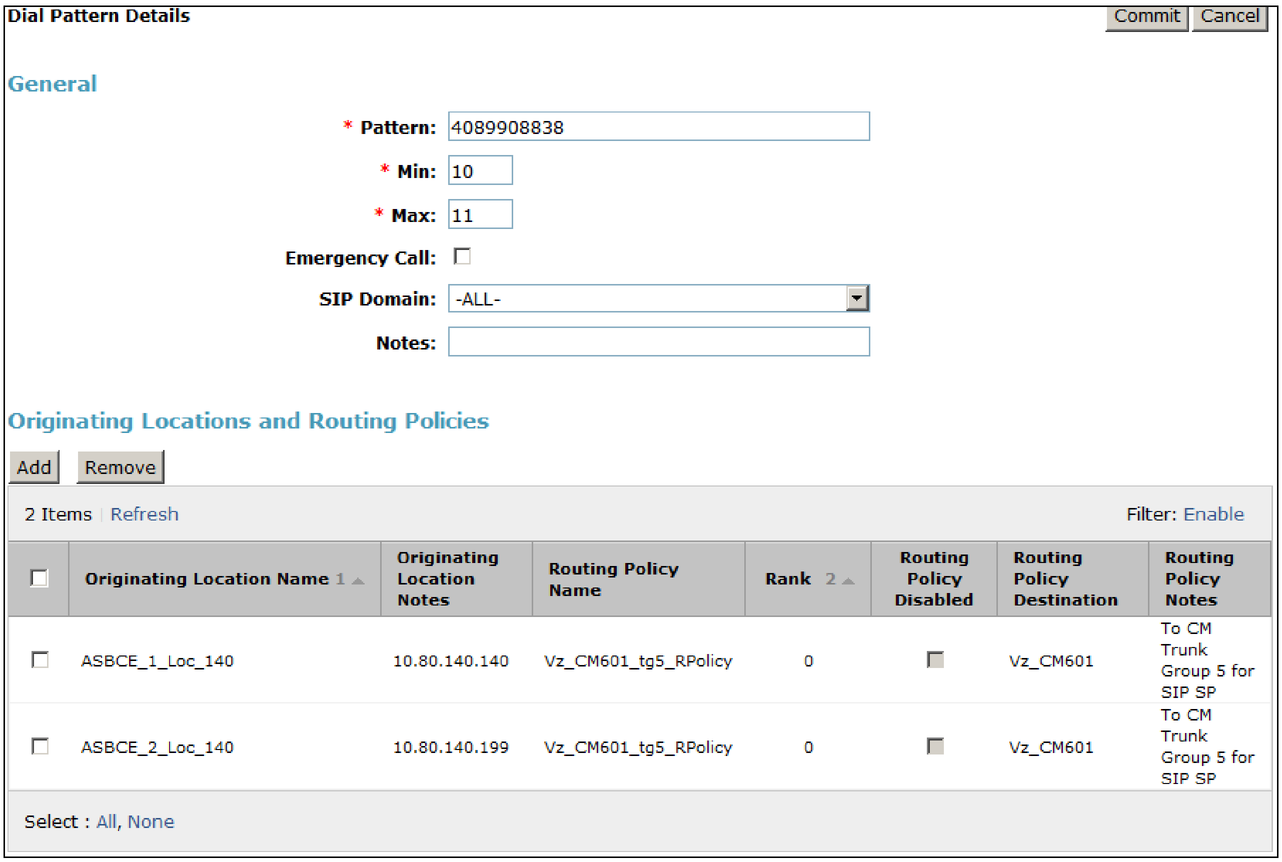


6.7. Dial Patterns

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.

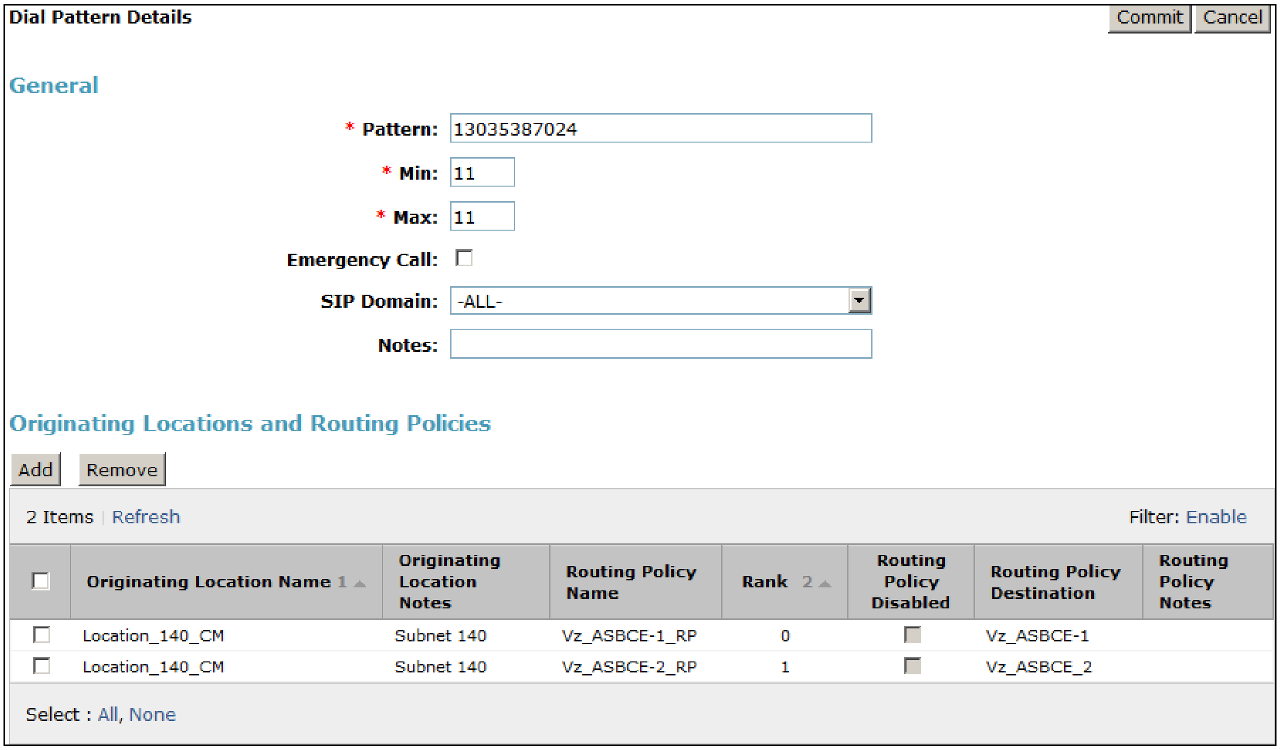
6.7.1 Inbound Call Dial Pattern

The following screen illustrates an example dial pattern used to verify inbound PSTN calls to the enterprise. When a user on the PSTN dials a number assigned to the Verizon IP Trunk service, such as 408-990-8838, Verizon delivers the number to the enterprise, and the ASBCE sends the call to Session Manager. The pattern below matches on 408-990-8838 specifically. Dial patterns can alternatively match on ranges of numbers (e.g., a DID block). Under **Originating Locations and Routing Policies**, the routing policy named "Vz_CM601_tg5_RPolicy" is selected, which sends the call to Communication Manager using port 5060 as described previously. In the Avaya Interoperability Lab configuration, calls to this number from any of the two originating locations, including the one with **Originating Location Name** "ASBCE_1_Loc_140", are routed to Communication Manager.



6.7.2 Outbound Call Dial Pattern

The following screen illustrates an example dial pattern used to verify outbound calls from the enterprise to the PSTN. When a Communication Manager user dials a PSTN number such as 9-1-303-538-7024, Communication Manager sends the call to Session Manager, via the HP Common Server Processor Ethernet. Session Manager will match the dial pattern shown below and send the call to the "Avaya-SBCE-1" or the "Avaya-SBCE-2" via the **Routing Policy Name** "Vz_ASBCE-1_RP"and "Vz_ASBCE-2_RP".



7. Avaya Session Border Controller for Enterprise

In the sample configuration, an ASBCE is used as the edge device between the CPE and Verizon Business.

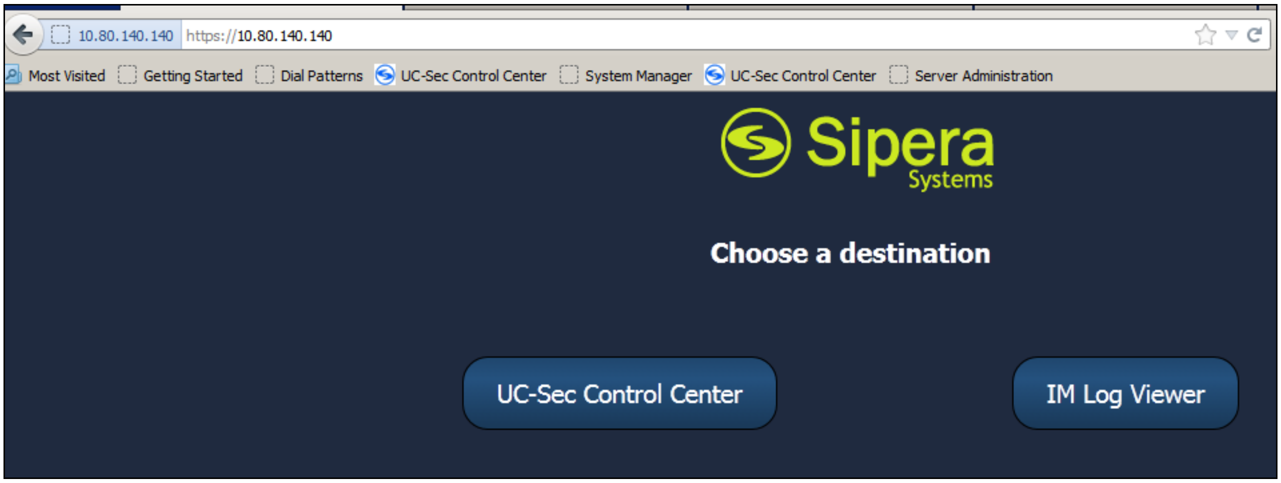
These Application Notes assume that the installation of the ASBCE, the assignment of a management IP Address, and commissioning of the system have already been completed.

As described in **Section 1**, Verizon Business IP Trunking supports a redundant (2-CPE) architecture that provides for redundant SIP trunk access between the Verizon Business IP Trunk service offer and the SIP trunk architecture customer premises equipment (CPE). In the reference configuration two (ASBCEs) were used to provide the 2-CPE redundant access.

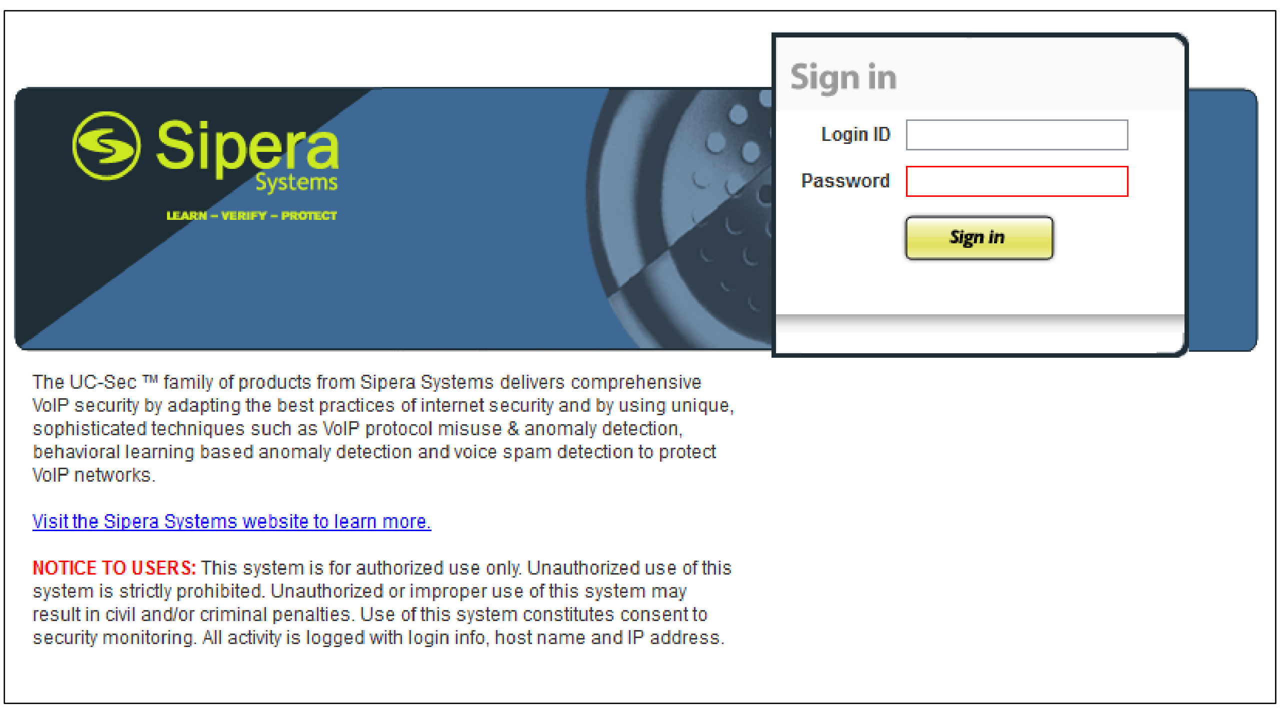
Note – The following Sections describe the provisioning of the Primary ASBCE. The configuration of the Secondary ASBCE is identical unless otherwise noted (e.g. IP addressing).

7.1. Access the Management Interface

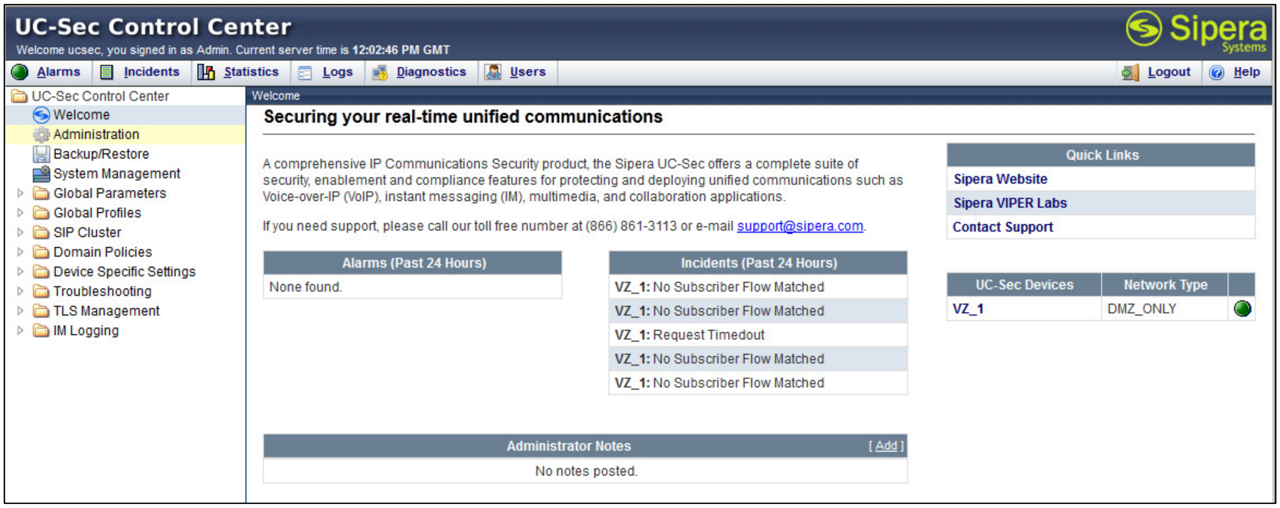
In the sample configuration, the management IP is 10.80.140.140. Access the web management interface by entering https://<ip-address> where <ip-address> is the management IP address assigned during installation. Select **UC-Sec Control Center**.



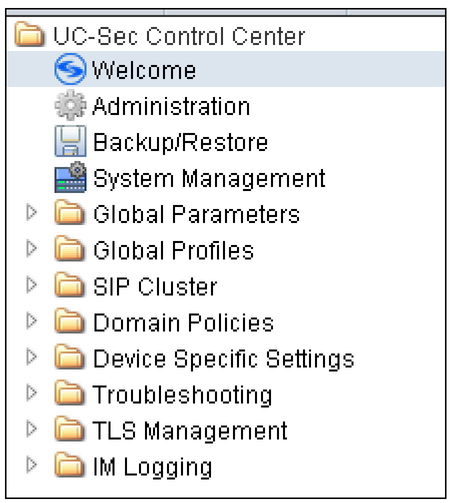
A log in screen is presented. Enter an appropriate **Login ID** and **Password.**



Once logged in, a UC-Sec Control Center screen will be presented.

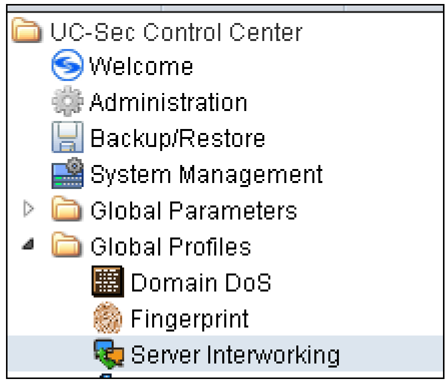


The following image illustrates the menu items available on the left-side of the UC-Sec Control Center screen.



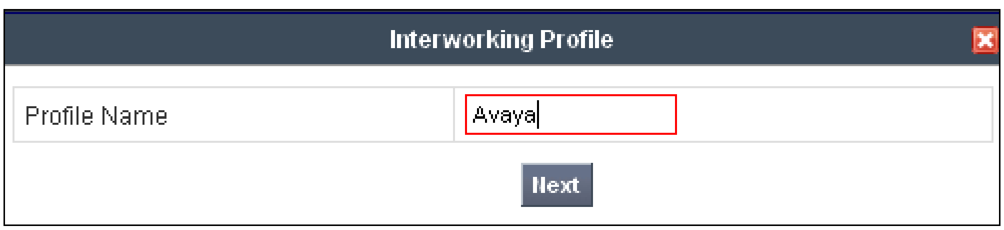
7.2. Global Profiles – Server Interworking

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

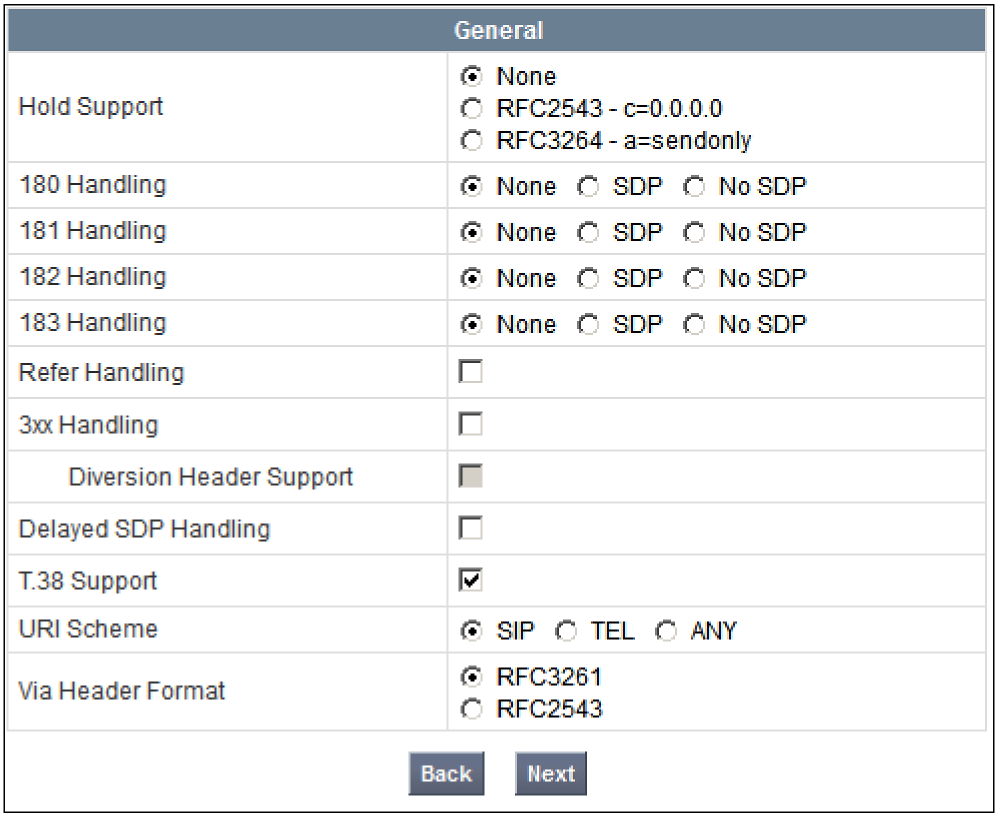


7.2.1 Server Interworking - Avaya

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



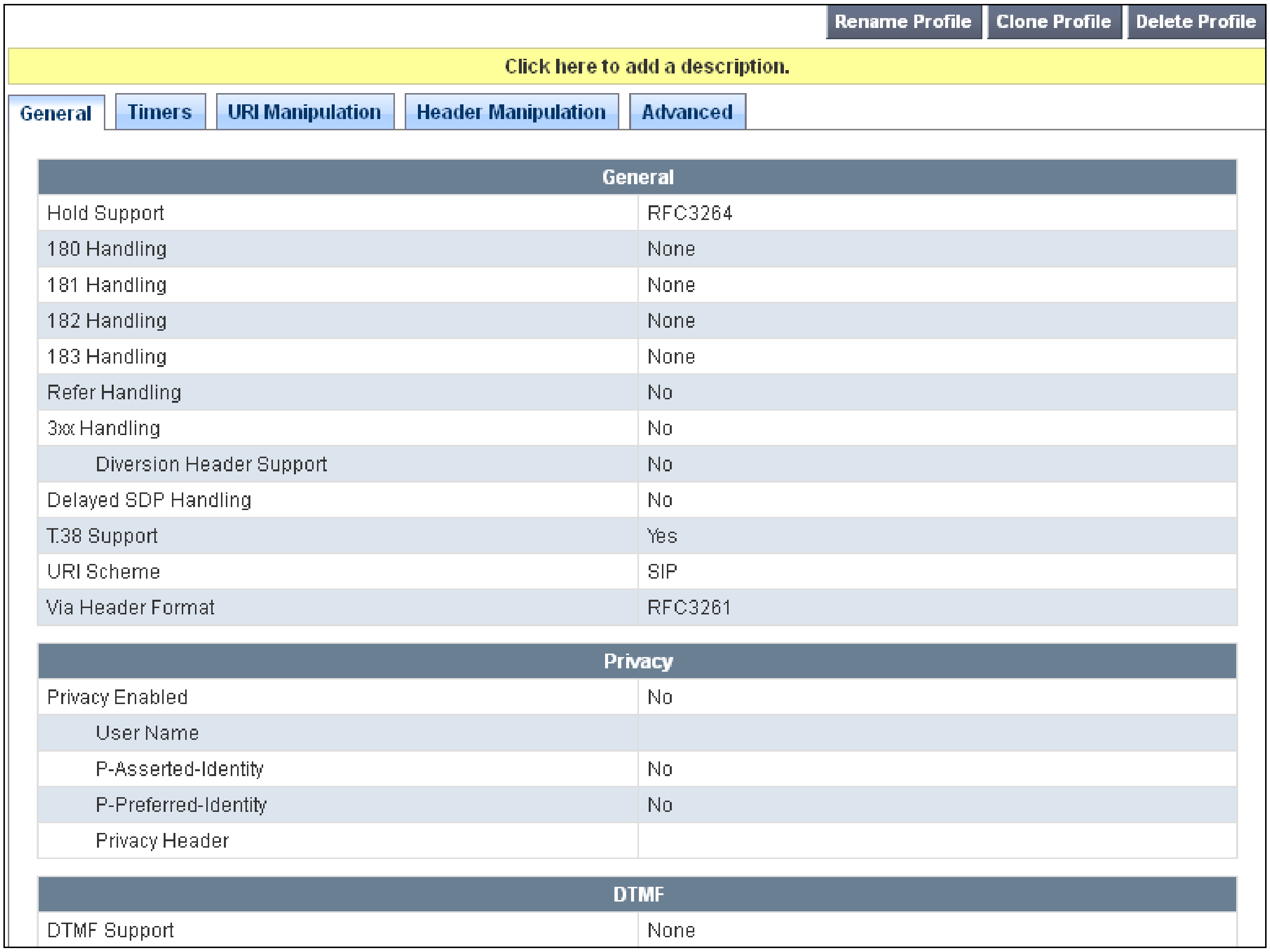
The following screens illustrate the "General" parameters used in the sample configuration for the Interworking Profile named "Avaya". Most parameters retain default values. In the sample configuration, **T.38 support** was checked (optional), and **Hold Support** was set for RFC3264.



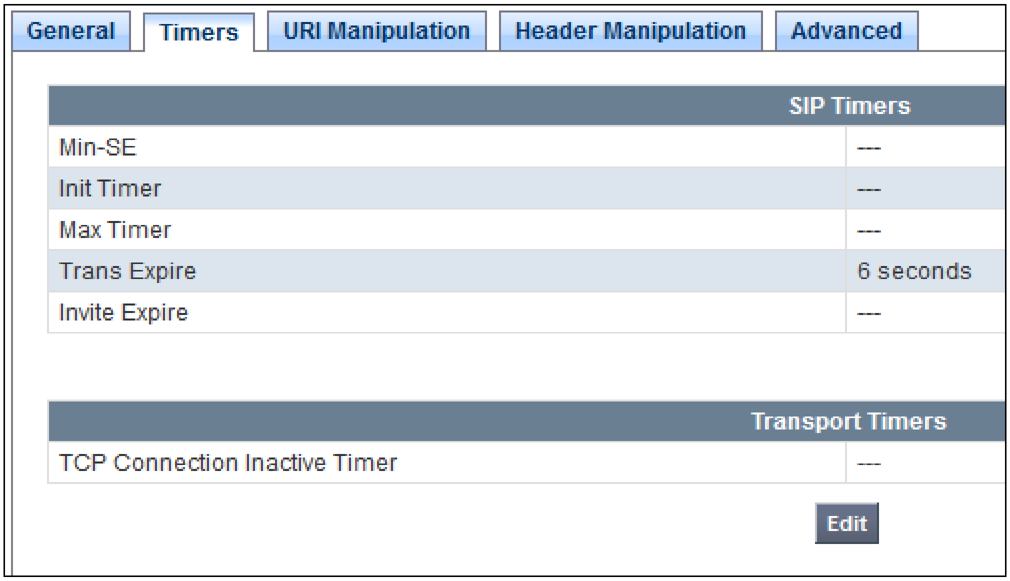
General	
Hold Support	(•) None ( ) RFC2543 - c=0.0.0.0 ( ) RFC3264 - a=sendonly
180 Handling	(•) None ( ) SDP ( ) No SDP
181 Handling	(•) None ( ) SDP ( ) No SDP
182 Handling	(•) None ( ) SDP ( ) No SDP
183 Handling	(•) None ( ) SDP ( ) No SDP
Refer Handling	☐
3xx Handling	☐
Diversion Header Support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

Back Next

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



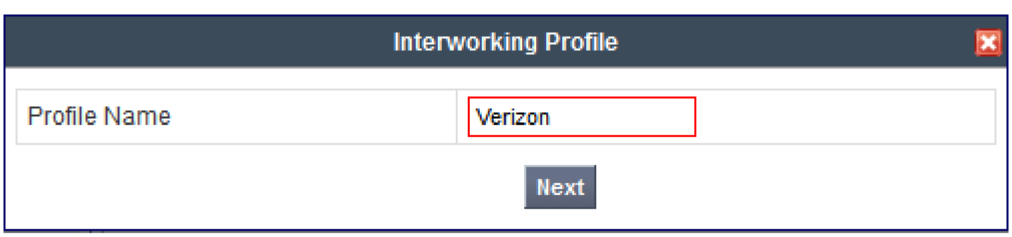
The 2-CPE configuration requires that certain timers be set to assist the failover process to happen smoothly. One of the timers is the **Trans Expire** timer. This timer is set to 6 seconds as shown below on the Avaya side only.



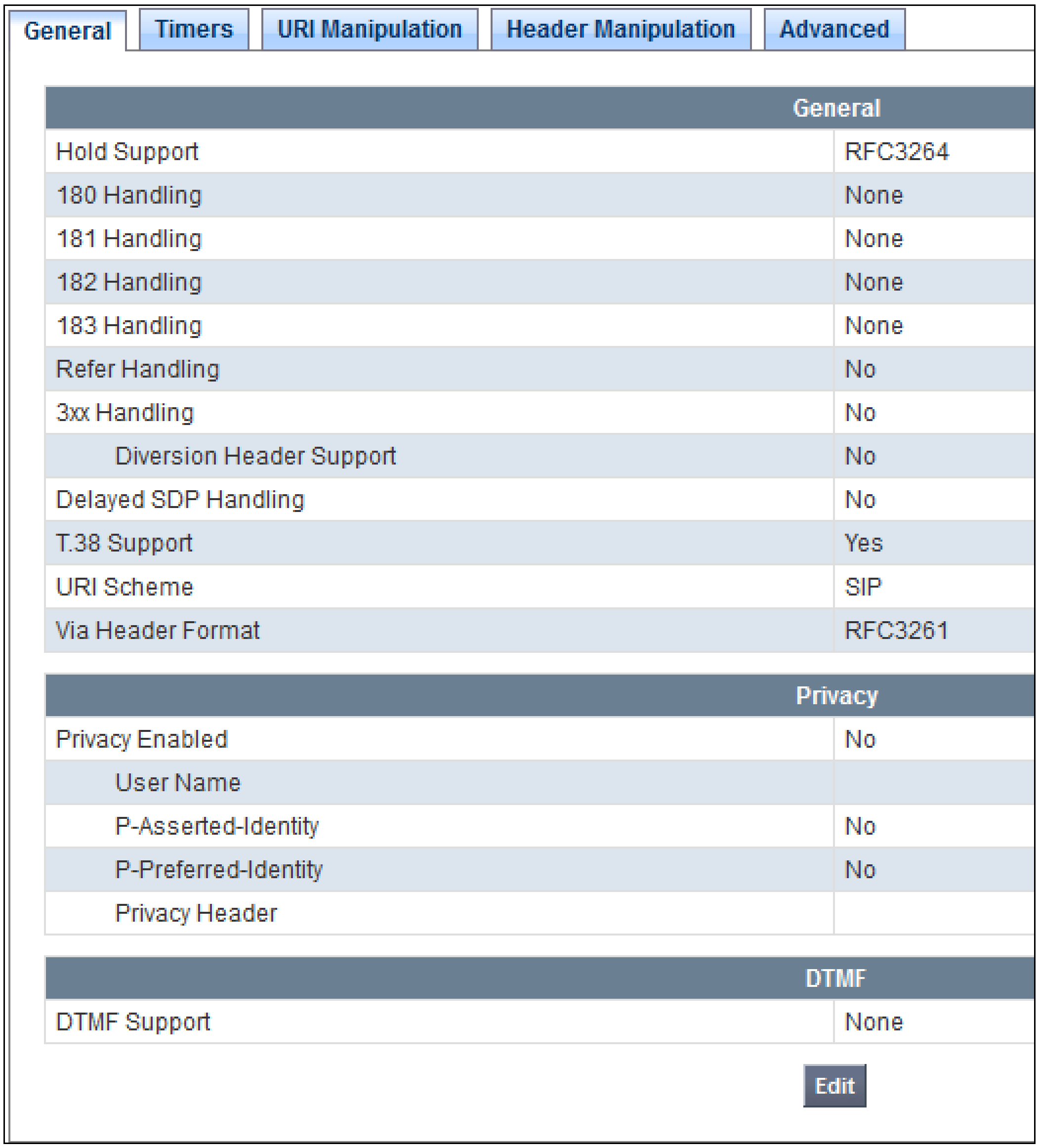
The following screen illustrates the **Advanced Settings** configuration. The “Topology Hiding: Change Call-ID” defaults to Yes, but was changed in the test configuration to allow for easier correlation of data. This value is set in the field at the discretion of the user. Both settings were tested. If using “Topology Hiding: Change Call-ID” = Yes, there could be a problem when using REFER on transferred calls. Please see **Section 2.2** for more details. All other parameters shown are default values. Note that the default configuration will result in Record-Route headers in SIP messages.

7.2.2 Server Interworking – Verizon IP Trunk

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



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

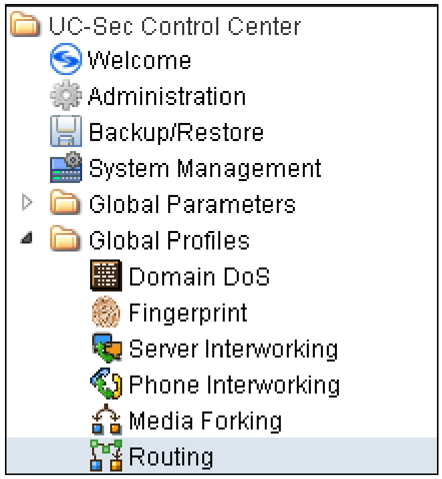


The following screen illustrates the **Advanced Settings** configuration. All parameters shown are default values except for “Topology Hiding: Change Call-ID” which was changed to No.

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

7.3. Global Profiles – Routing

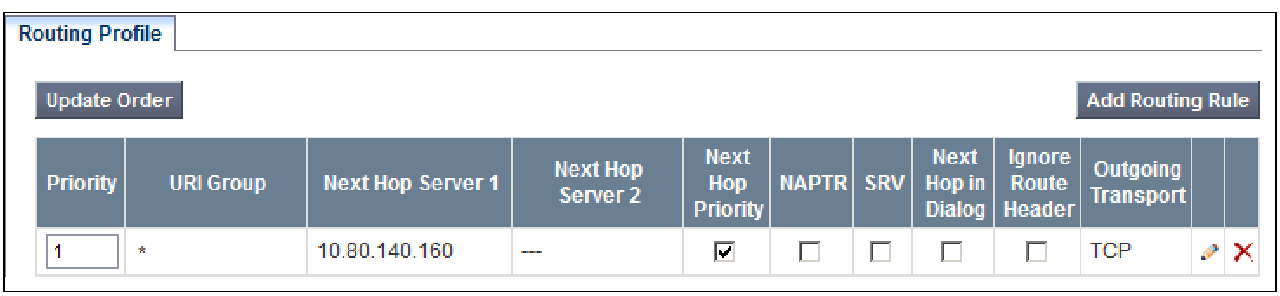
Select **Global Profiles → Routing** from the left-side menu as shown below.



7.3.1 Routing Configuration for Session Manager

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

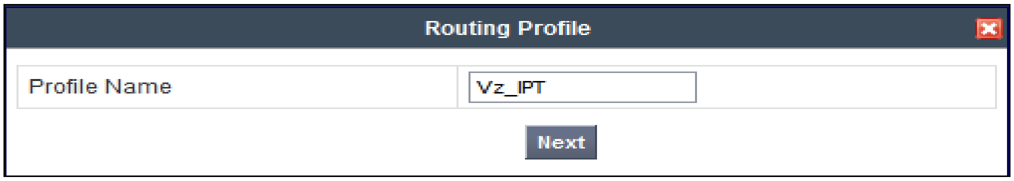
For the **Next Hop Routing**, enter the IP Address of the Session Manager SIP signaling interface as **Next Hop Server 1**, as shown below. Check **Next Hop Priority**. Choose **TCP** for **Outgoing Transport**. Then click **Finish**.



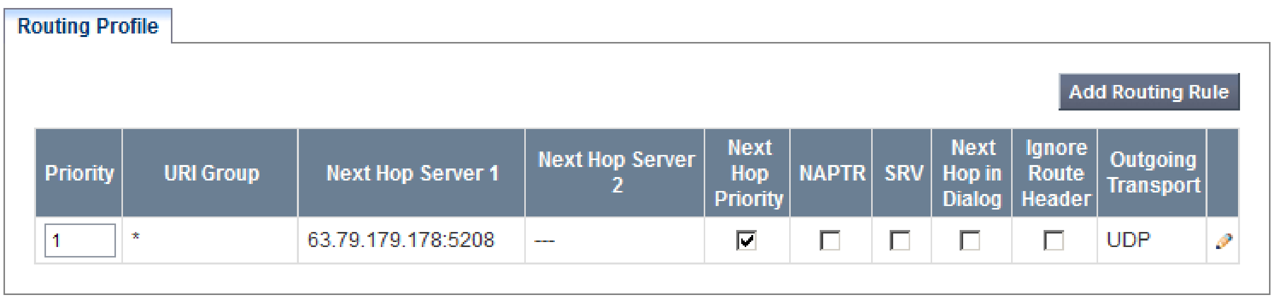
Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport		
1	*	10.80.140.160	---	☑	☐	☐	☐	☐	TCP		

7.3.2 Routing Configuration for Verizon IP Trunk

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



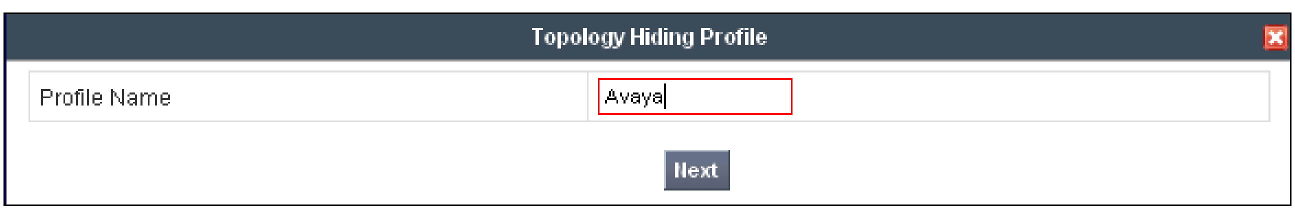
For the **Next Hop Routing**, enter the IP Address of the Verizon SIP signaling interface as **Next Hop Server 1**, as shown below. Check **Next Hop Priority**. Choose **UDP** for **Outgoing Transport**, then click **Finish (not shown)**.



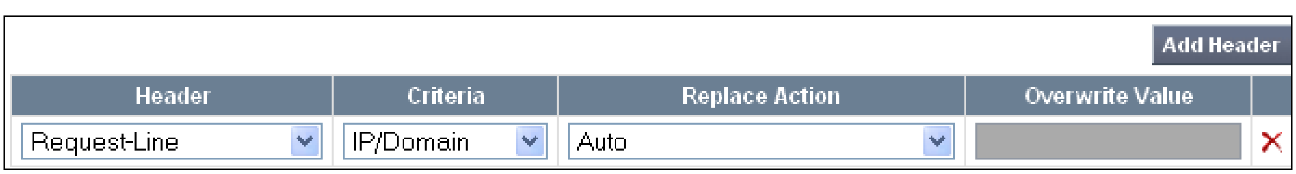
7.3.3 Topology Hiding for Session Manager

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

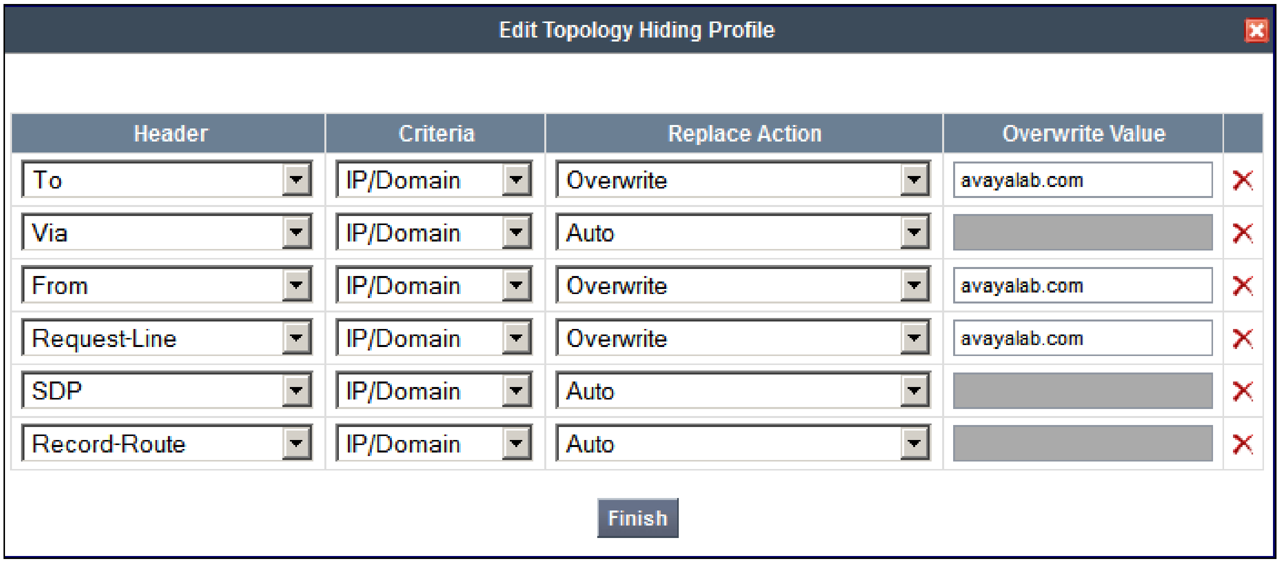
Click the **Add Profile** button (not shown) to add a new profile, or select an existing topology hiding profile to edit. If adding a profile, a screen such as the following is displayed. Enter a **Profile Name** such as "Avaya" shown below. Click **Next**.



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



In the Replace Action column an action of “Auto” will replace the header field with the IP address of the ASBCE interface and Overwrite will use the value in the “Overwrite Value”. In the example shown, this profile will later be applied in the direction of 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. This action can also be done in the Adaptations section of Session Manager. Click **Finish**.

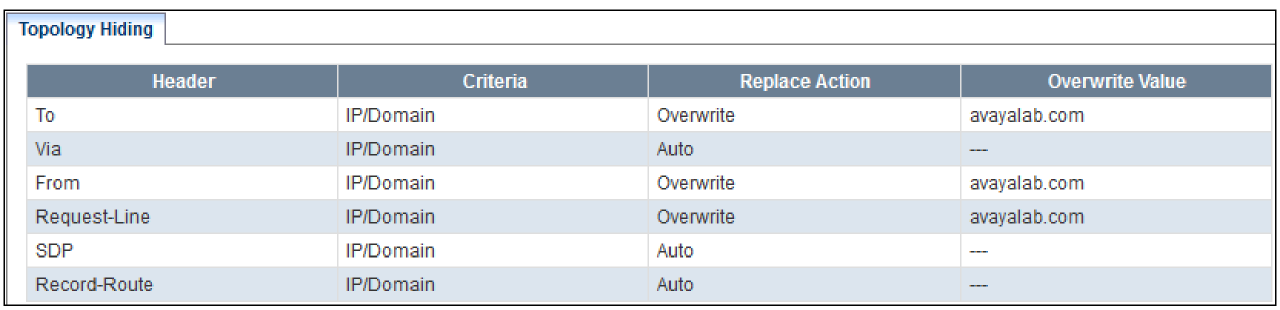


Edit Topology Hiding Profile

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

Finish

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

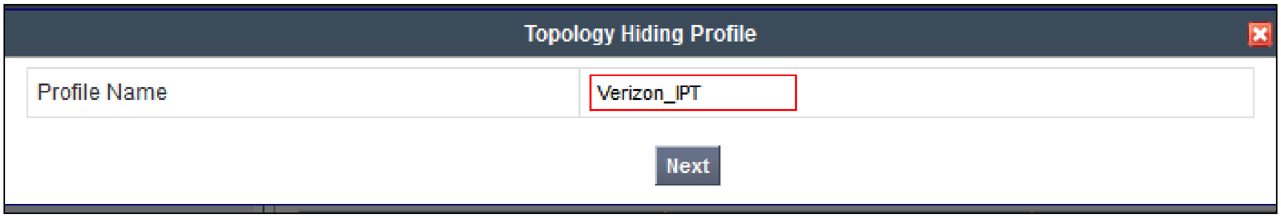


Topology Hiding

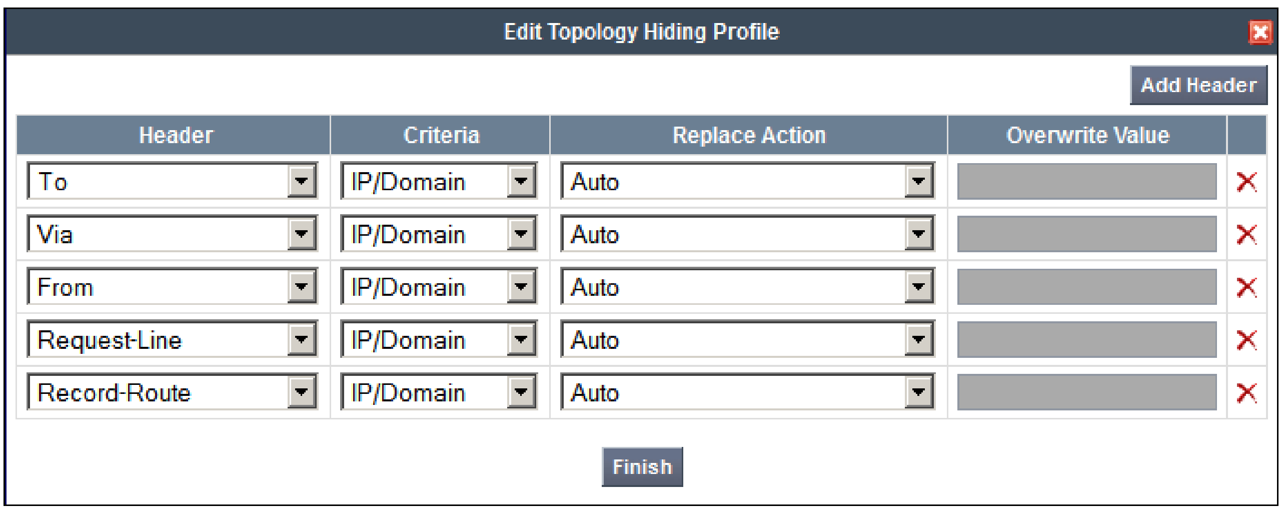
Header	Criteria	Replace Action	Overwrite Value
To	IP/Domain	Overwrite	avayalab.com
Via	IP/Domain	Auto	---
From	IP/Domain	Overwrite	avayalab.com
Request-Line	IP/Domain	Overwrite	avayalab.com
SDP	IP/Domain	Auto	---
Record-Route	IP/Domain	Auto	---

7.3.4 Topology Hiding for Verizon IP Trunk

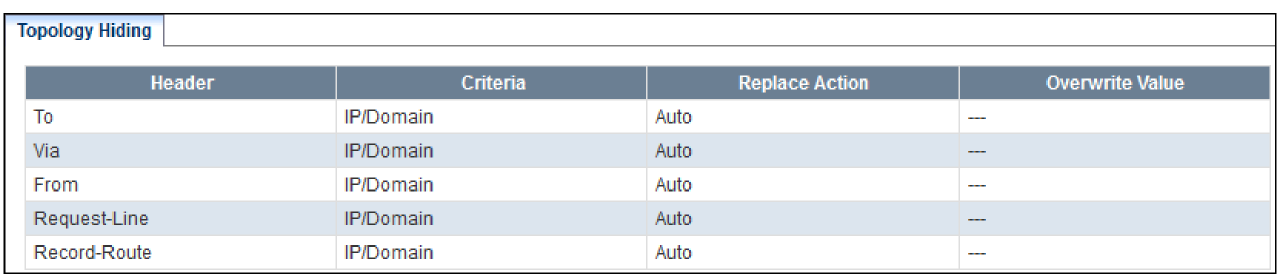
Click the **Add Profile** button (not shown) to add a new profile, or select an existing topology hiding profile to edit. If adding a profile, a screen such as the following is displayed. Enter a **Profile Name** such as "Verizon_IPT" shown below. Click **Next**.



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



After configuration is completed, the **Topology Hiding** for profile "Verizon_IPT" will appear as follows.

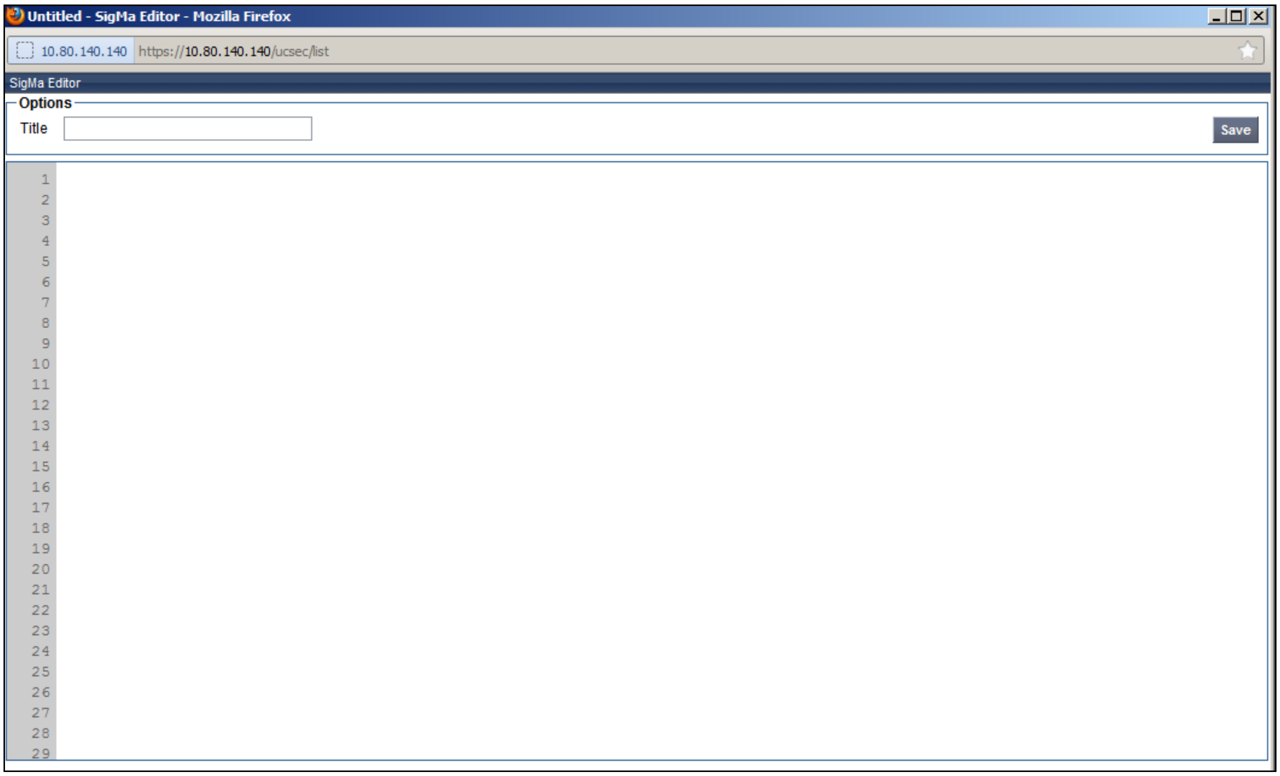

Topology Hiding

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

7.3.5 Signaling Manipulation

This feature adds the ability to add, change and delete any of the headers and other information in a SIP message on each flow in a highly flexible manner using a proprietary scripting language.

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



1) **REMOVE UNWANTED HEADERS:** In Communication Manager and Session Manager 6.1, there are proprietary headers (e.g., P-Location, Endpoint-View) and three standard headers (Alert-Info, User-Agent, Server) that contain internal information and that are not applicable to a service provider that need to be stripped. These headers were stripped with a Sigma script and applied in the server configuration section. The script “Example2” is shown here. This script will be applied in the next section, ‘Server Configuration’. The script was applied on requests and responses as shown below.

2) **SIP REFER HEADERS**: The SIP Header manipulations highlighted below were used to overcome the defects listed in **Section 2.2** for SIP REFER issues. In the testing environment, Verizon required the SIP Header “Referred-By” to contain a valid phone number and IP Address and the issues highlighted in Section 2.2 needed to be overcome with this script. The extensions 7688 and 7633 were the local extensions and were being translated to valid DIDs 4089908838 and 4089908837 respectively for both the Referred-By and Contact headers. The IP address in the domain of the Referred-By header is also being translated from the incorrect IP address and port of the service provider to just the IP address of the ASBCE outside IP address.

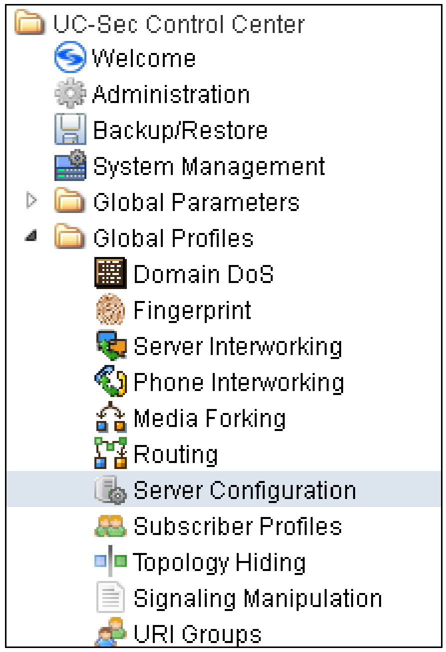
This Sigma Script was used during testing and was named Example2. It will be applied to the Verizon Server Configuration created in the next section.

Signaling Manipulation

```
within session "ALL"
{
 act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
  {
// Topology Hiding of P-Location header for subsequent re-INVITEs

   remove(%HEADERS["P-Location"][1]);
   remove(%HEADERS["Endpoint-View"][1]);
   remove(%HEADERS["Alert-Info"][1]);
   remove(%HEADERS["User-Agent"][1]);
   remove(%HEADERS["Server"][1]);
   %HEADERS["Referred-By"][1].regex_replace("7689@63.79.179.178:5208","4089908838@12.71.19.138");
   %HEADERS["Contact"][1].regex_replace("7689","4089908838");
   %HEADERS["Referred-By"][1].regex_replace("7633@63.79.179.178:5208","4089908837@12.71.19.138");
   %HEADERS["Contact"][1].regex_replace("7633","4089908837");

   }
 }
```

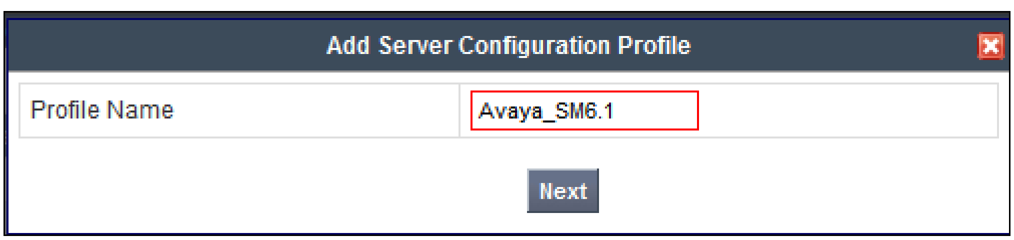
7.4. Global Profiles – Server Configuration

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

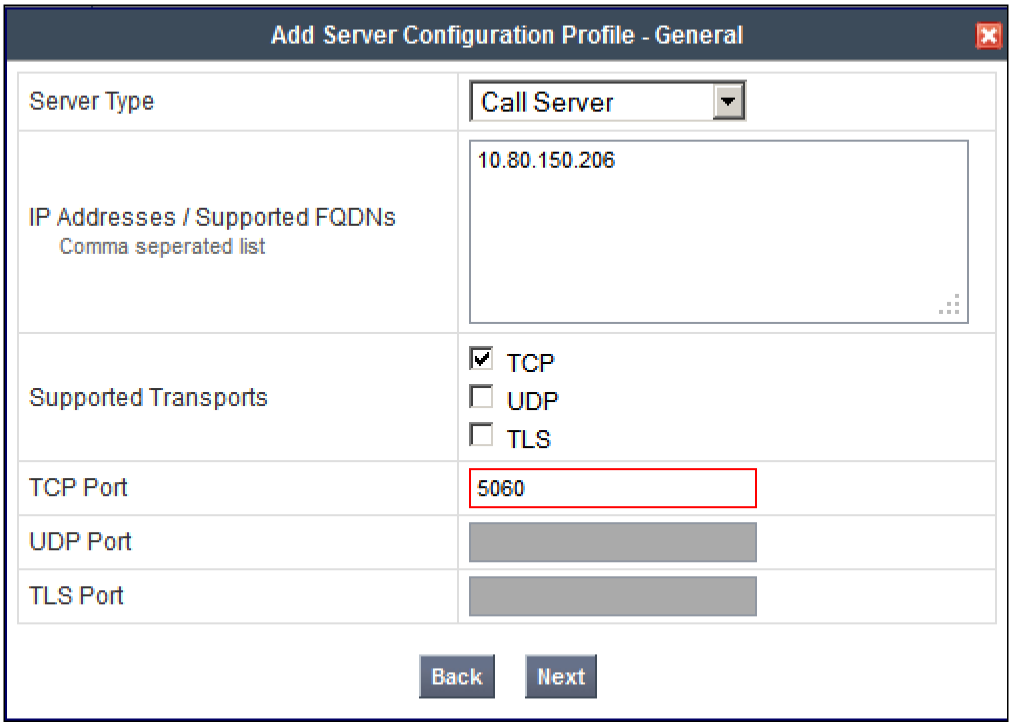


7.4.1 Server Configuration for Session Manager

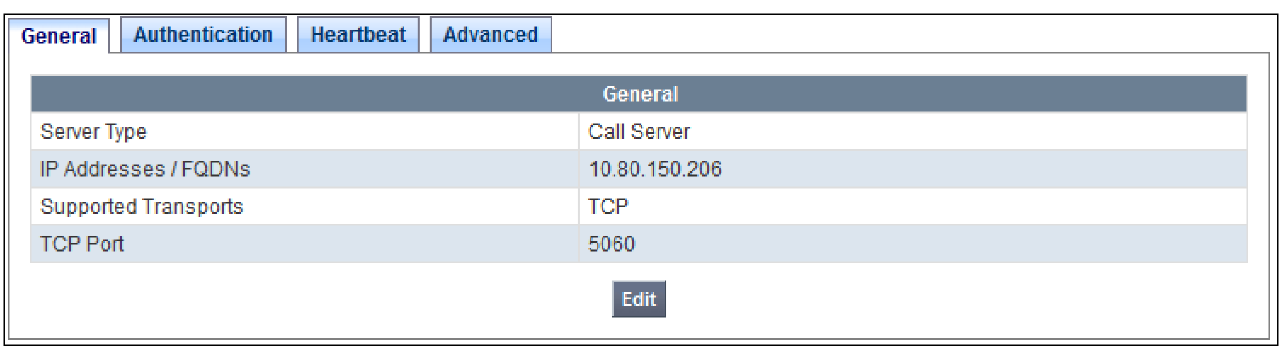
Click the **Add Profile** button (not shown) to add a new profile, or select an existing profile to edit. If adding a profile, a screen such as the following is displayed. Enter an appropriate Profile Name such as "Avaya_SM6.1" shown below. Click **Next**.



The following screens illustrate the Server Configuration for the Profile name "Avaya_SM6.1". 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 10.80.150.206. 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 ASBCE created in **Section 6.5**. If adding a new profile, click **Next** (not shown). If editing an existing profile, click **Finish** (not shown).



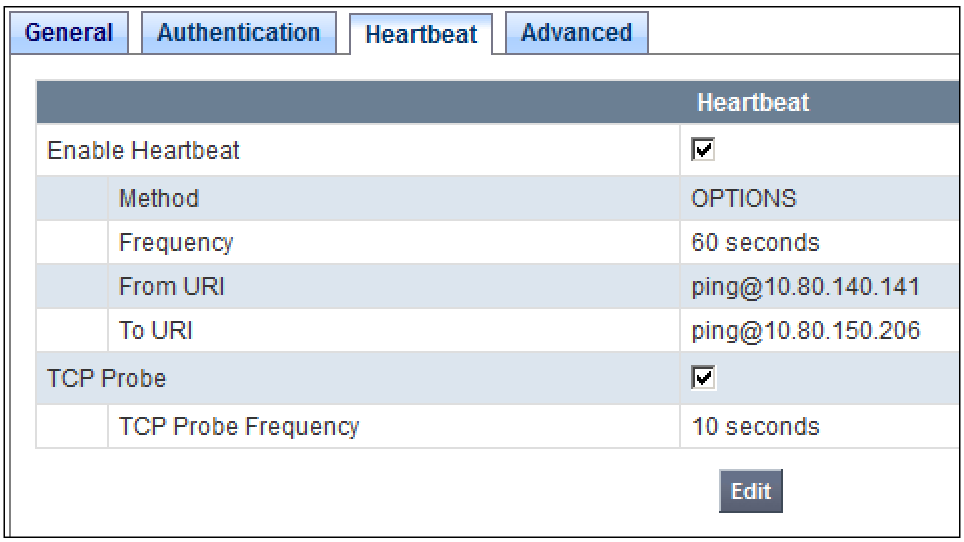
Once configuration is completed, the **General** tab for "Avaya_SM6.1" will appear as shown below.



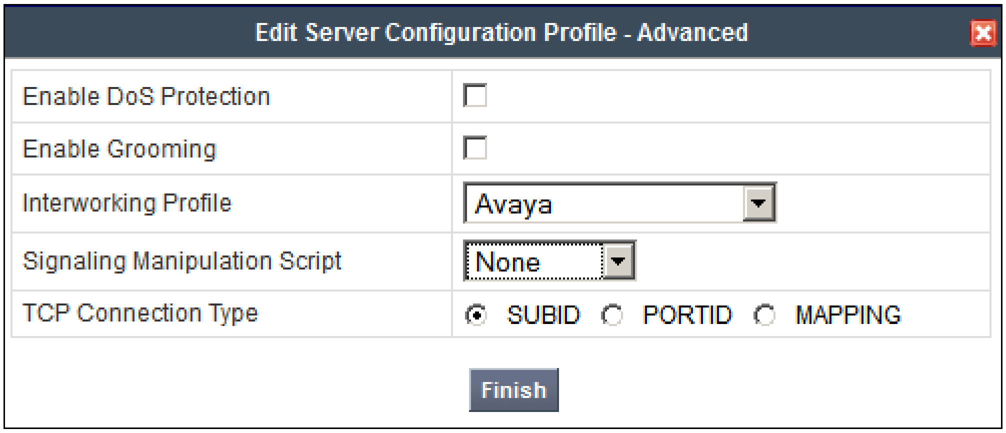
If adding the profile, click **Next** to accept default parameters for the Authentication tab (not shown), and advance to the Heartbeat area. If editing an existing profile, select the **Heartbeat** tab and click **Edit** (not shown).

The ASBCE can be configured to source "heartbeats" in the form of SIP OPTIONS. In the sample configuration with one Session Manager, this configuration is unnecessary unless 2- CPE is used. If 2-CPE is used, the OPTIONS must be configured along with the **TCP Probe Frequency** at 10 seconds.

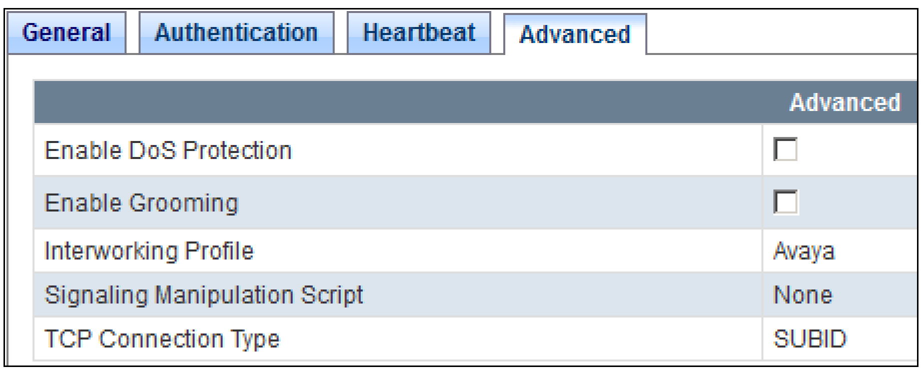
If ASBCE-sourced OPTIONS messages are desired, check the **Enable Heartbeat** box. Select "OPTIONS" from the **Method** drop-down menu. Select the desired frequency that the ASBCE will source OPTIONS to this server. The **From URI** and **To URI** may be filled in to configure easily identifiable URIs to appear in SIP OPTIONS sourced by the ASBCE 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 the **Interworking Profile** "Avaya" created previously. Click **Finish**.

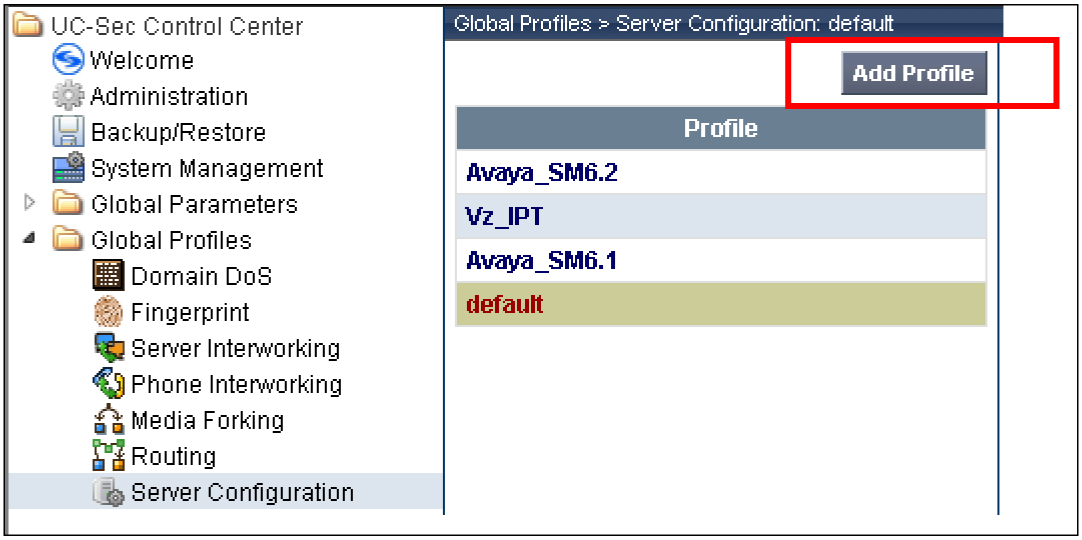


Once configuration is completed, the **Advanced** tab for the profile "Avaya_SM6.1" will appear as shown below.

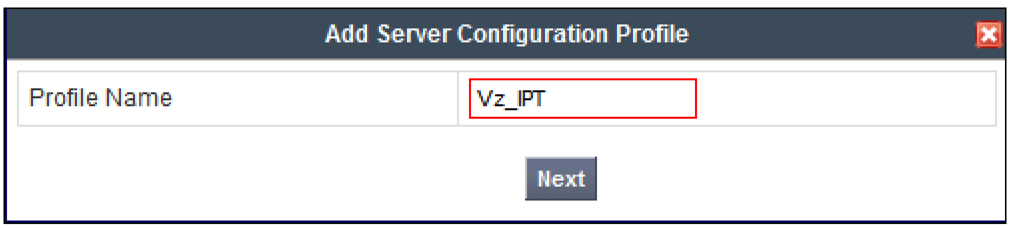


7.4.2 Server Configuration for Verizon IP Trunk

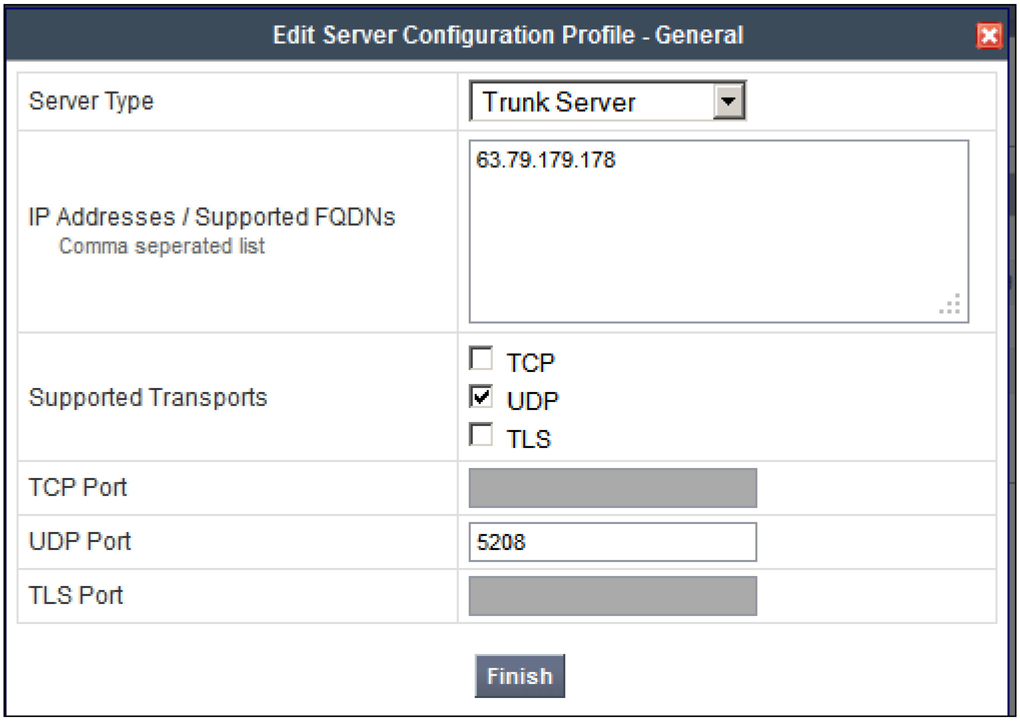
Click the **Add Profile** button to add a new profile, or select an existing profile to edit.



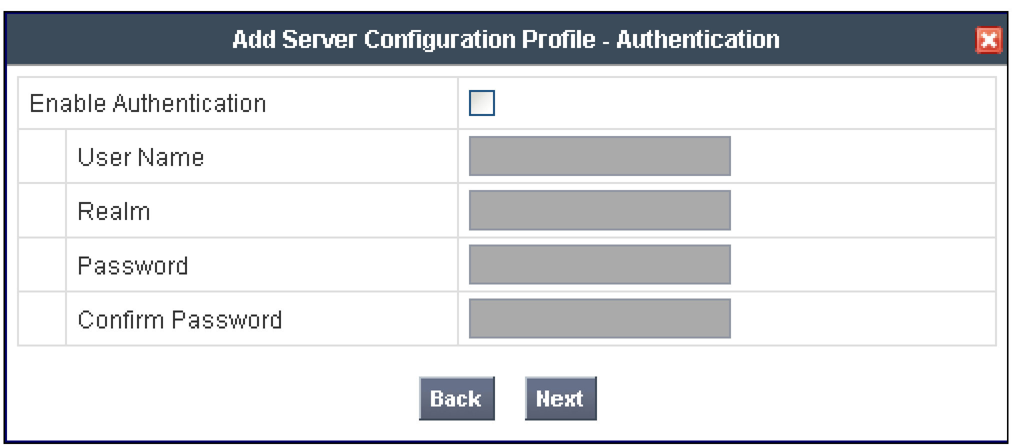
If adding a profile, a screen such as the following is displayed. Enter an appropriate Profile Name such as "Vz_IPT" shown below. Click **Next**.



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 5208(domestic). Click **Next** to proceed to the **Authentication** Tab.



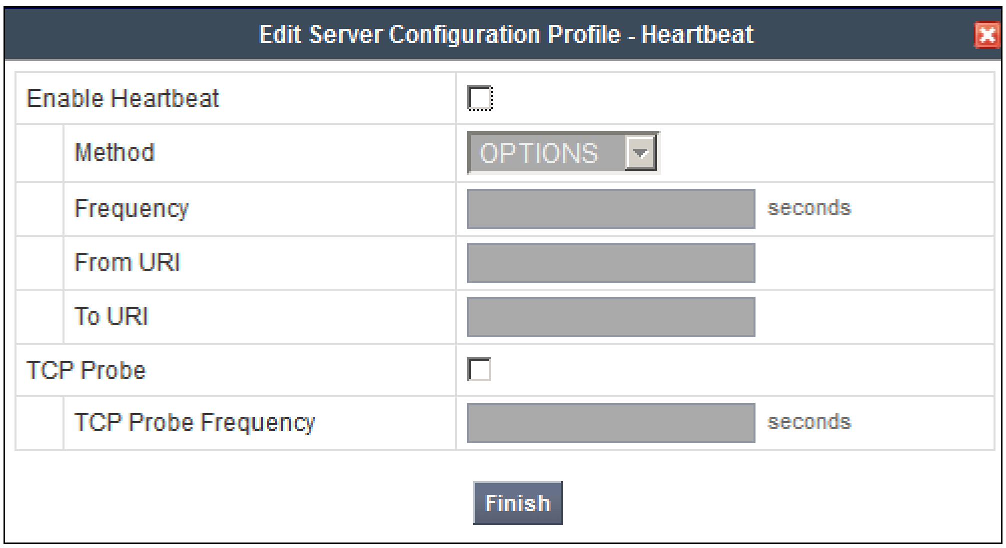
If adding the profile, click **Next** to accept default parameters for the **Authentication t**ab (below), and advance to the Heartbeat area. No authentication was used in the test configuration.



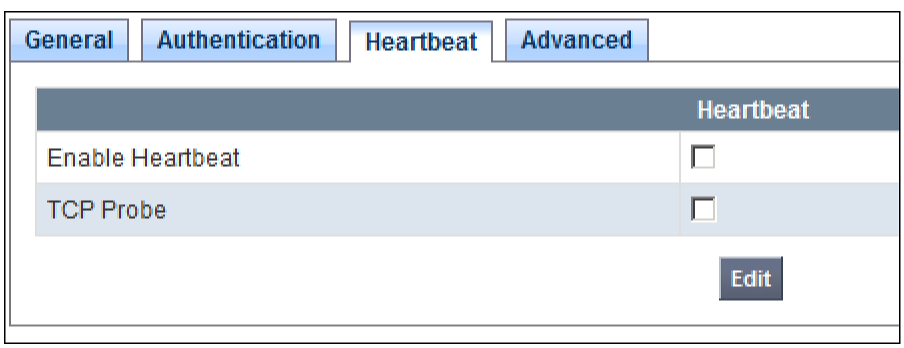
The ASBCE can be configured to source “heartbeats” in the form of SIP OPTIONS towards Verizon. This configuration is optional and was not used in this configuration. Independent of whether the ASBCE is configured to source SIP OPTIONS towards Verizon, Verizon will receive OPTIONS from the enterprise site as a result of the SIP Entity Monitoring configured for Session

Manager. When Session Manager sends SIP OPTIONS to the private IP Address of the ASBCE, the ASBCE will send SIP OPTIONS to Verizon. When Verizon responds, the ASBCE will pass the response to Session Manager.

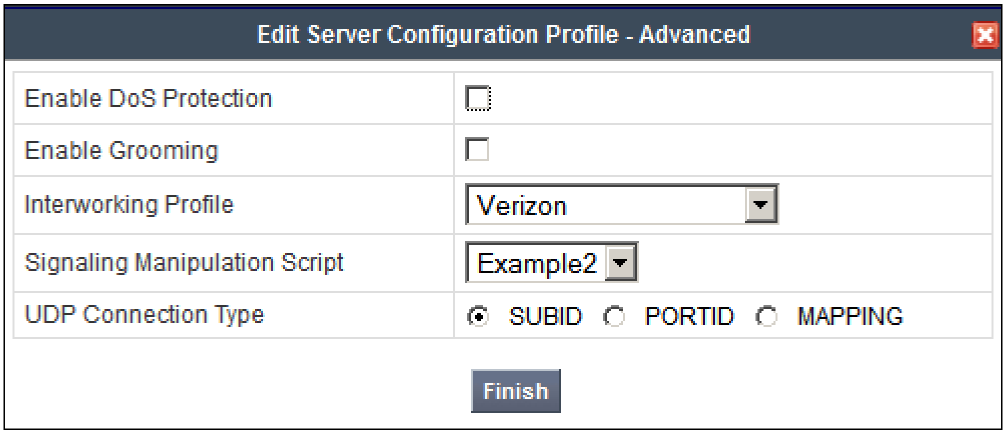
If ASBCE-sourced OPTIONS are desired, select "OPTIONS" from the **Method** drop-down menu. Select the desired frequency that the SBCE will source OPTIONS. The **From URI** and **To URI** may be filled in to configure easily identifiable URIs to appear in SIP OPTIONS sourced by the ASBCE. If adding a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



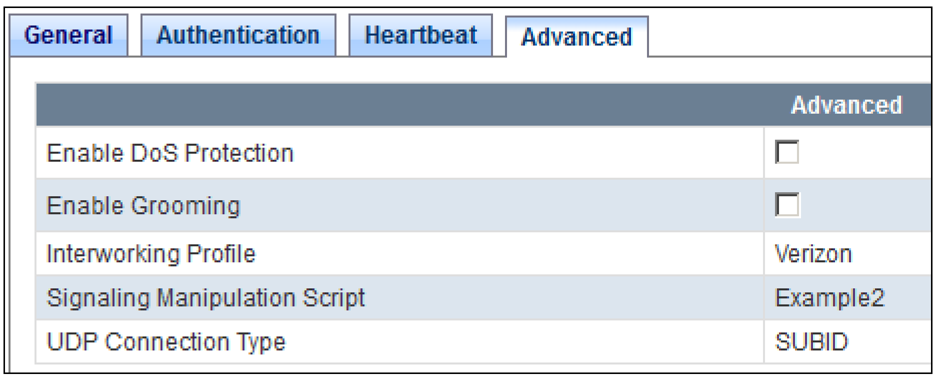
If the optional ASBCE sourced OPTIONS configuration is completed, the **Heartbeat** tab for "Vz_IPT" will appear as shown below.



If editing an existing profile, highlight the desired profile, select the **Advanced** tab, and click the **Edit button** (not shown). In the resultant screen, select the **Interworking Profile** "Verizon" created previously, and Signaling Manipulation Script will be the script shown in the previous section titled "Example2". Other ASBCE features, such as DoS Protection and Grooming, can be configured according to customer preference. Click **Finish**.

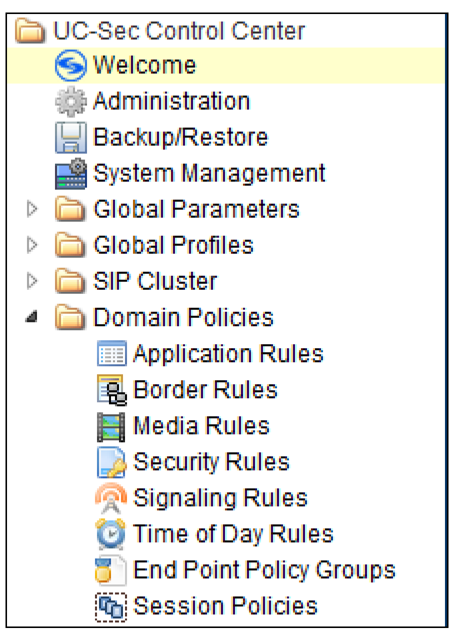


Once configuration is completed, the **Advanced** tab for “Vz_-IPT” will appear as shown below.

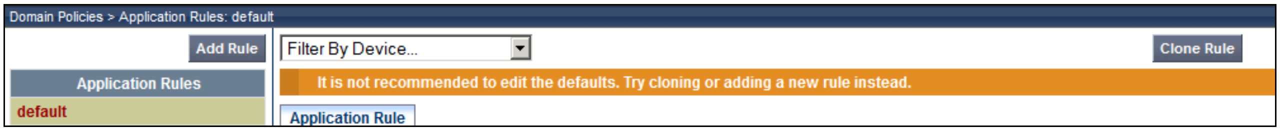


7.5. Domain Policies – Application Rule

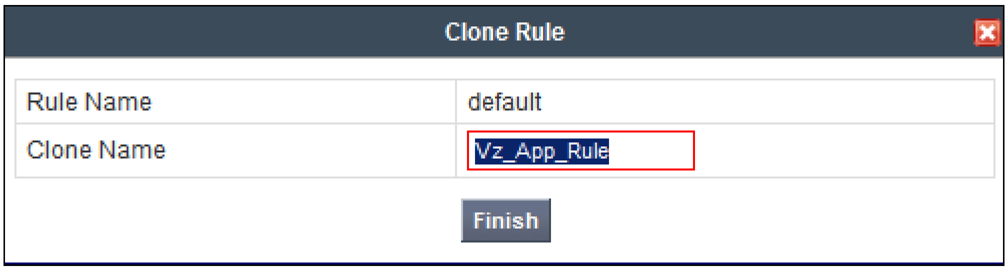
Select **Domain Policies → Application Rules** from the left-side menu as shown below.



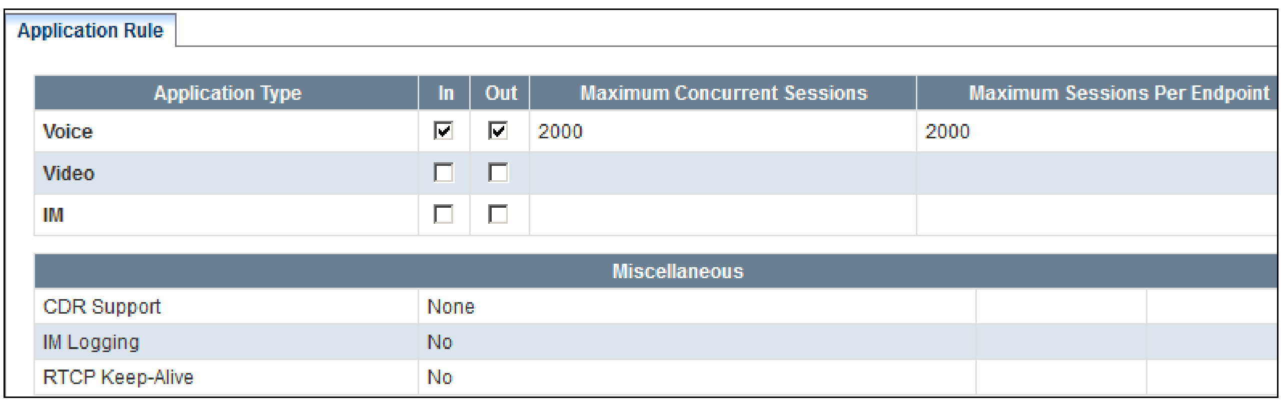
In the sample configuration, a single application rule was created by cloning the default rule called "default". Select the default rule and click the **Clone Rule** button.



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



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



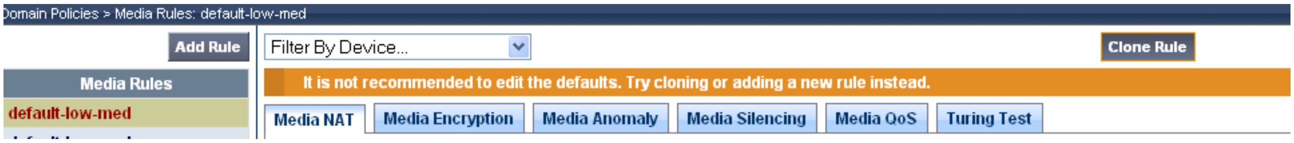
Application Rule

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

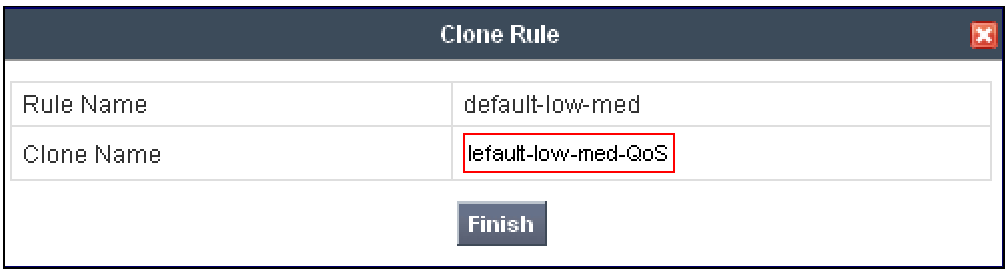
Miscellaneous	
CDR Support	None
IM Logging	No
RTCP Keep-Alive	No

7.6. Domain Policy – Media Rules

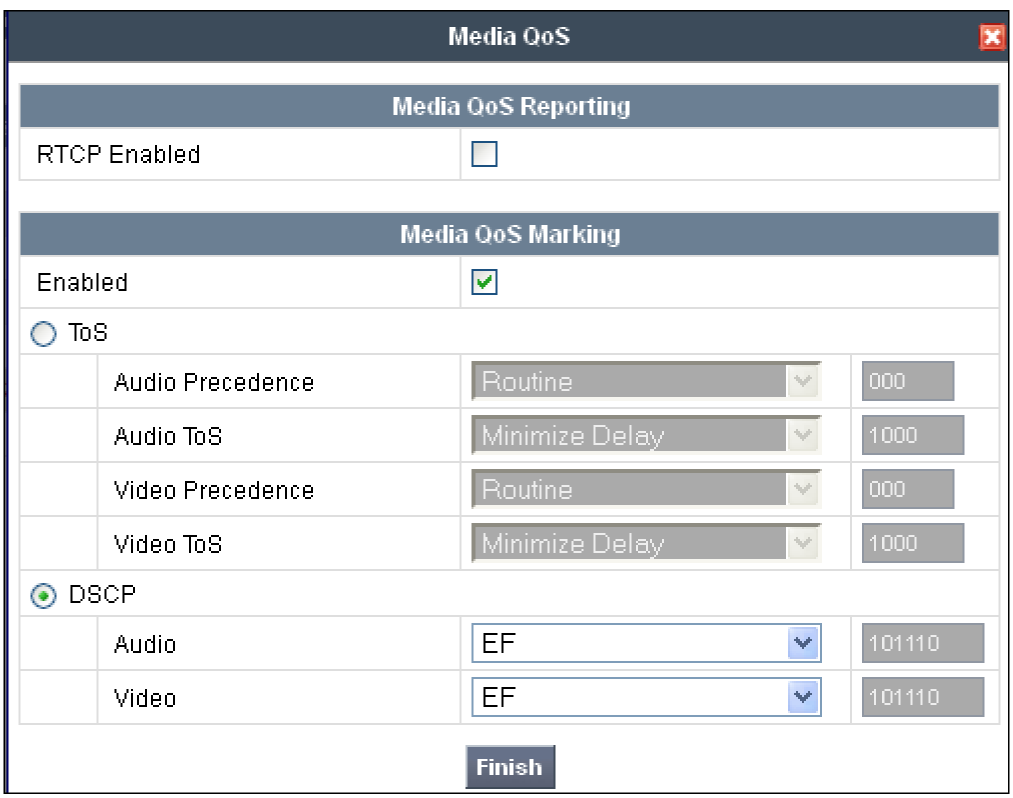
In the sample configuration, a single media rule was created by cloning the default rule called "default-low-med". Select the default-low-med rule and click the **Clone Rule** button.



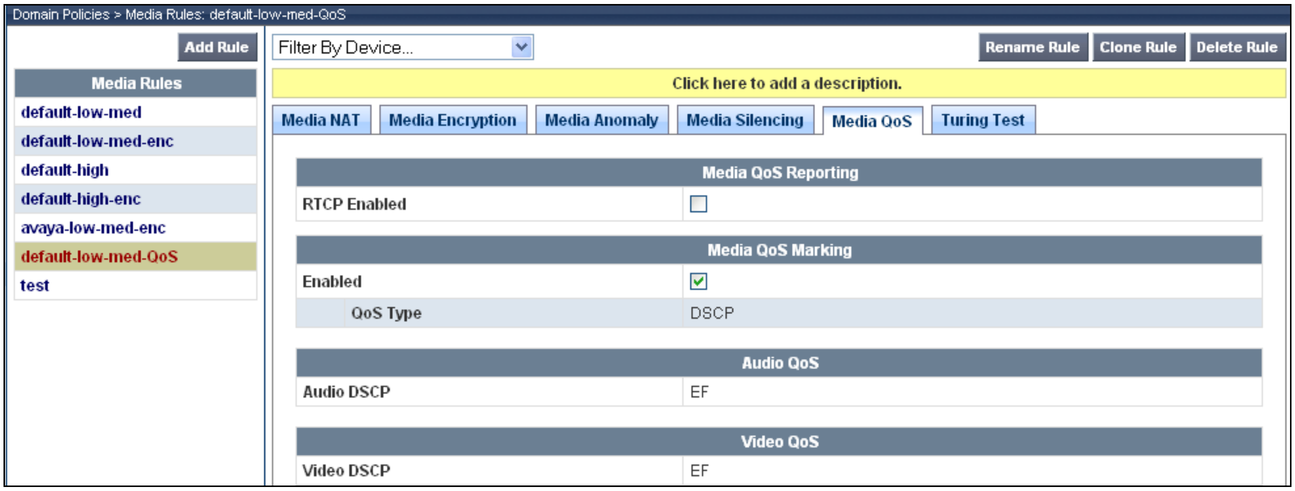
Enter a name in the **Clone Name** field, such as "default-low-med-QoS" as shown below. Click **Finish**.



Select the newly created rule, select the **Media QoS** tab (shown in previous screen), 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**.

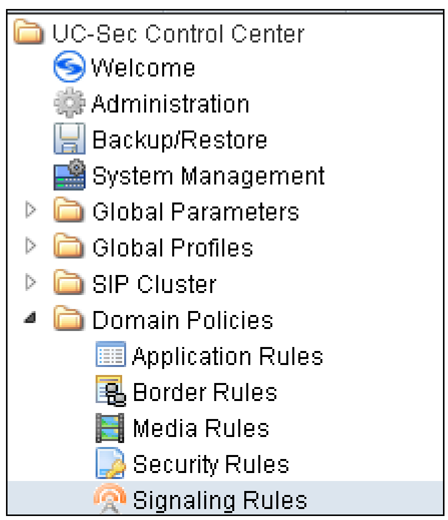


When configuration is complete, the “default-low-med-QoS” media rule **Media QoS** tab appears as follows.

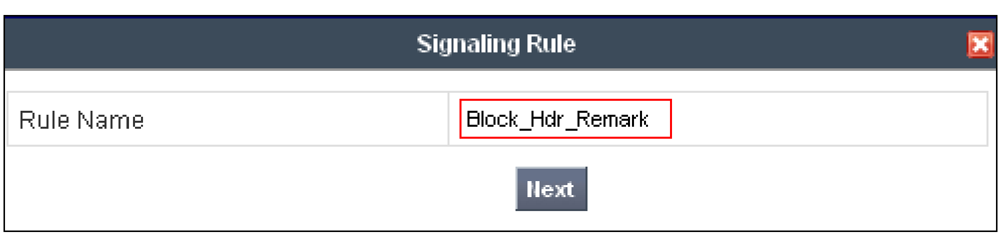


7.7. Domain Policies – Signaling Rules

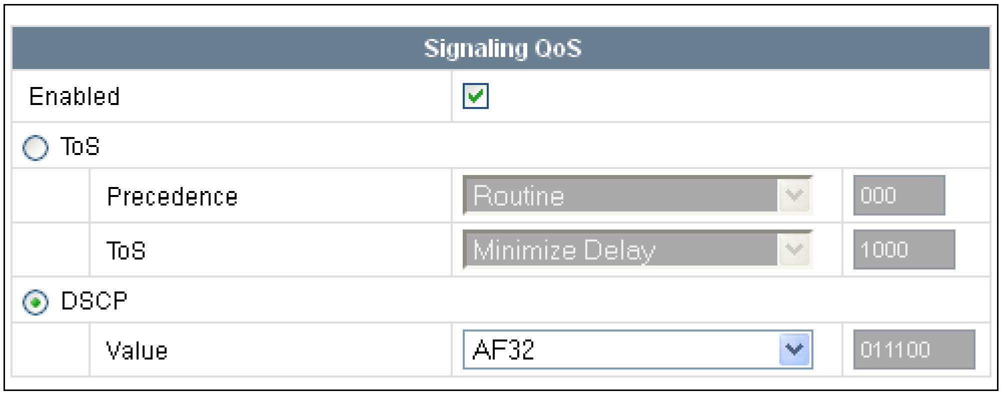
Select **Domain Policies → Signaling Rules** from the left-side menu as shown below.



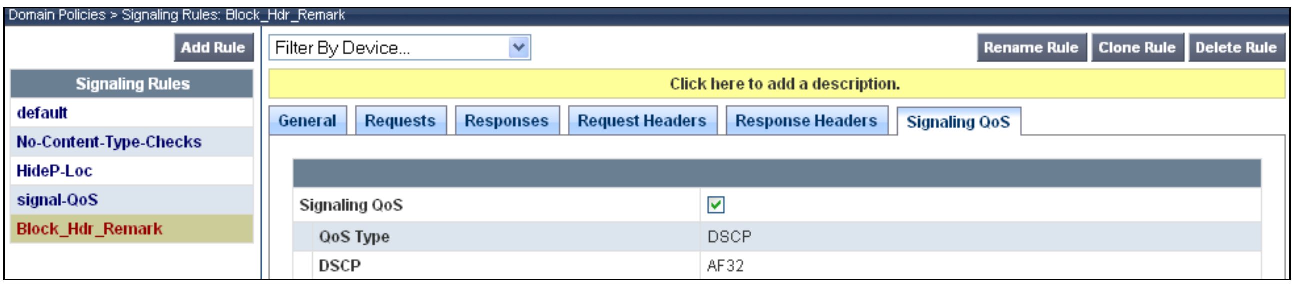
Click the **Add Rule** button (not shown) to add a new signaling rule. In the Rule Name field, enter an appropriate name, such as “Block_Hdr_Remark” and click **Next**.



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

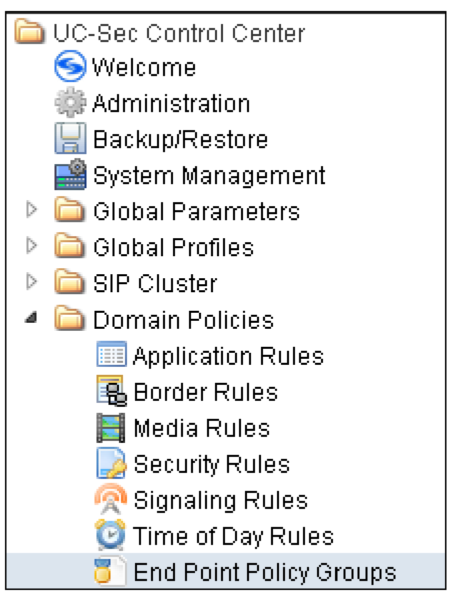


After this configuration, the new "Block_Hdr_Remark" **Signaling QoS** tab will appear as follows.

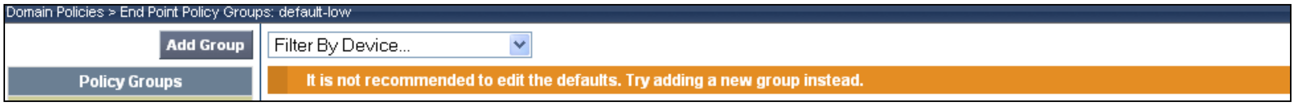


7.8. Domain Policies – End Point Policy Groups

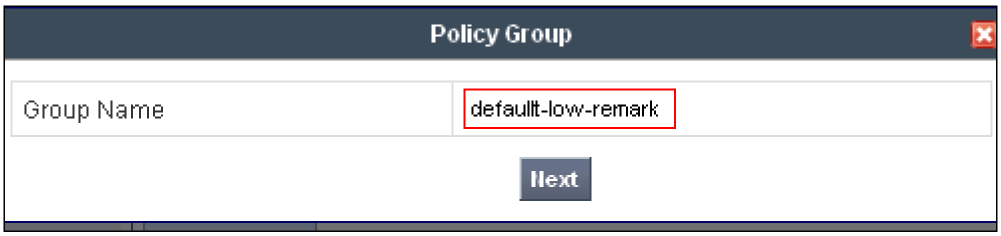
Select **Domain Policies → End Point Policy Groups** from the left-side menu as shown below.



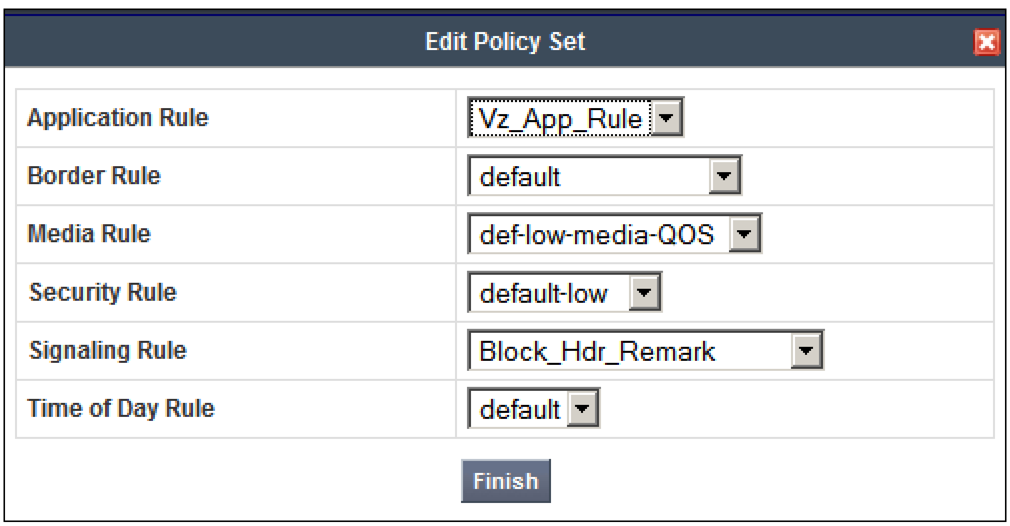
Select the **Add Group** button.



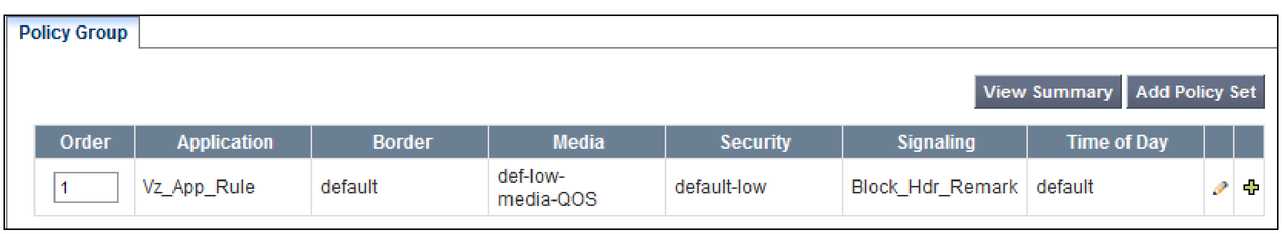
Enter a name in the **Group Name** field, such as "default-low-remark" as shown below. Click **Next**.



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

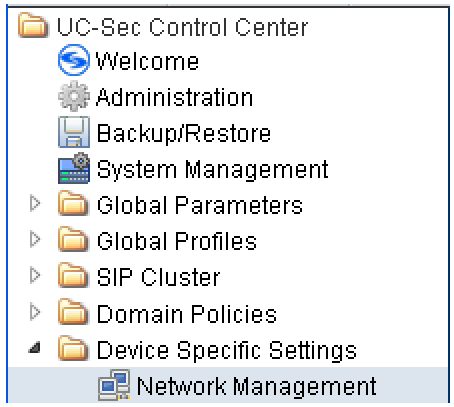


Once configuration is completed, the "default-low-remark" policy group will appear as follows.

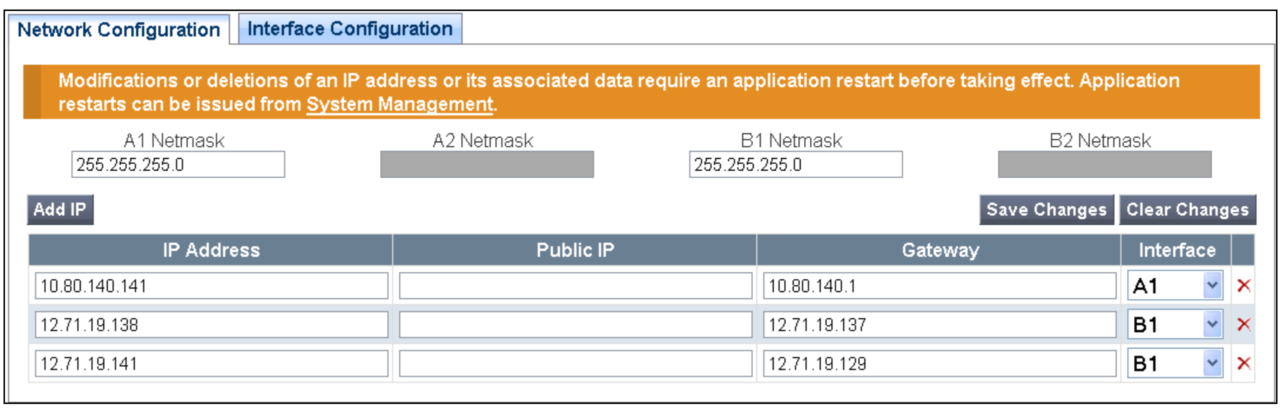


7.9. Device Specific Settings - Network Management

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



Under **UC-Sec Devices**, select the device being managed, which was named "VZ_1" in the sample configuration (not shown). The **Network Configuration** tab is shown below. Observe the **IP Address**, **Netmask (A1 and B1)**, **Gateway**, and **Interface** information previously assigned. In this test configuration, there were two IP Addresses assigned to the outside interface (B1), one for SIP signaling to the service provider that was routed to a VPN tunnel (12.71.19.138), and one for RTP (12.71.19.141) that was routed to the network's default gateway. This may not be necessary in all configurations, but was required for the specific test environment.



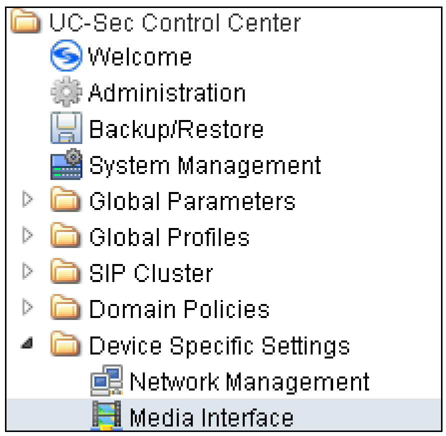
Select the **Interface Configuration** tab. The Administrative Status can be toggled between "Enabled" and "Disabled" in this screen. The following screen was captured after the interfaces had already been enabled. To enable the interface if it is disabled, click the **Toggle State** button.

Network Configuration | Interface Configuration

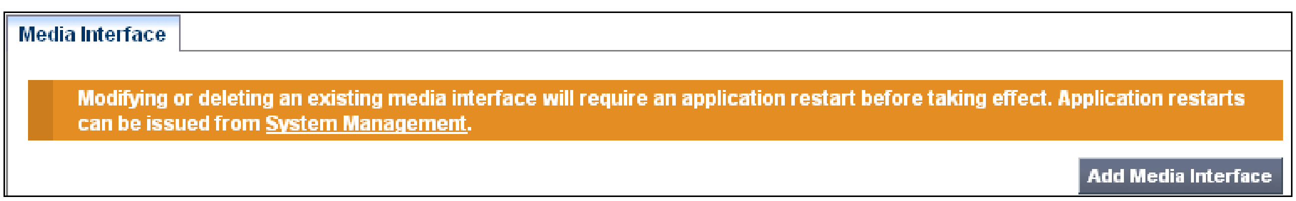
Name	Administrative Status	
A1	Enabled	Toggle State
A2	Disabled	Toggle State
B1	Enabled	Toggle State
B2	Disabled	Toggle State

7.10. Device Specific Settings – Media Interface

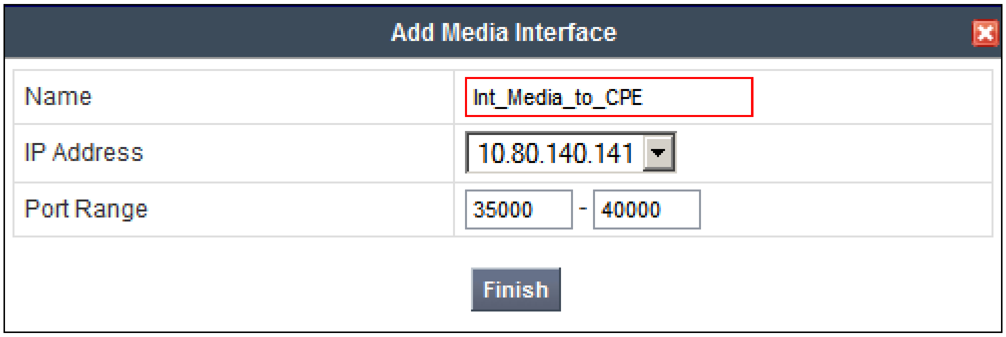
Select **Device Specific Setting → Media Interface** from the left-side menu as shown below.



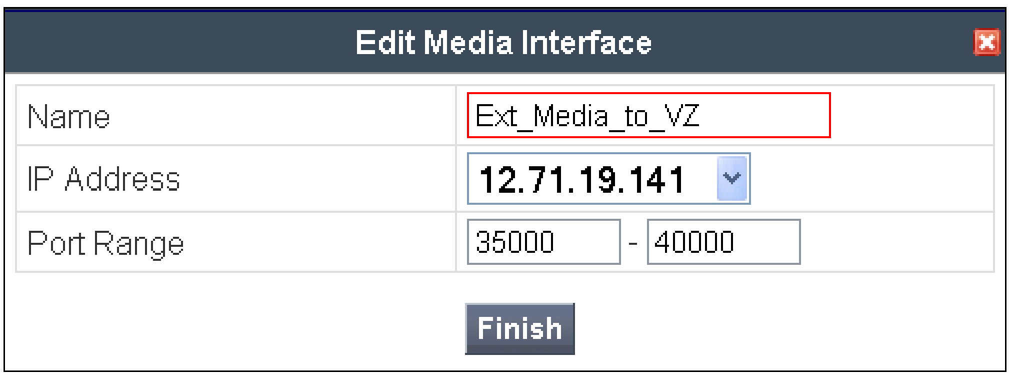
Under **UC-Sec Devices**, select the device being managed, which was named "VZ_1" in the sample configuration (not shown). Click **Add Media Interface**.



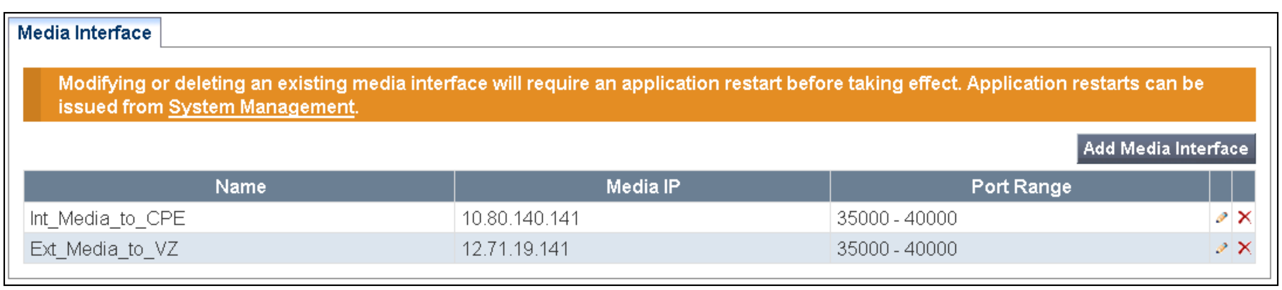
Enter an appropriate **Name** for the media interface for the Avaya CPE and select the inside private IP Address from the **IP Address** drop-down menu. In the sample configuration, "Int_Media_to_CPE" is chosen as the Name, and the "inside" IP Address of the ASBCE is "10.80.140.141". For the **Port Range**, default values are shown. Click **Finish**.



Once again, select **Add Media Interface**. Enter an appropriate **Name** for the media interface for the public "outside" of the ASBCE, and select the outside public IP Address from the **IP Address** drop-down menu. In the sample configuration, "Ext_Media_to_VZ" is chosen as the name, and the "outside" public IP Address of the ASBCE is "2.2.2.2". For the **Port Range**, default values are shown. Verizon IP Trunk does not require that the RTP ports be chosen within a specific range. Click **Finish**.

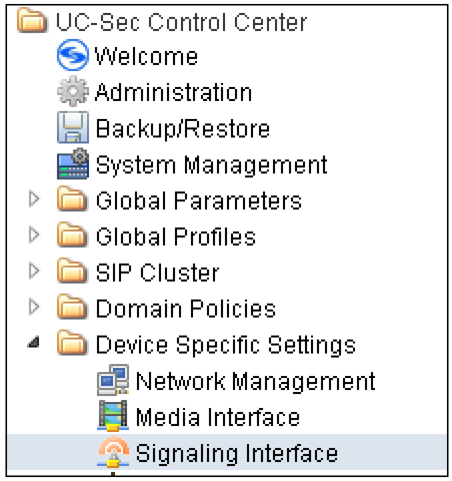


The resultant Media Interface configuration used in the sample configuration is shown below.

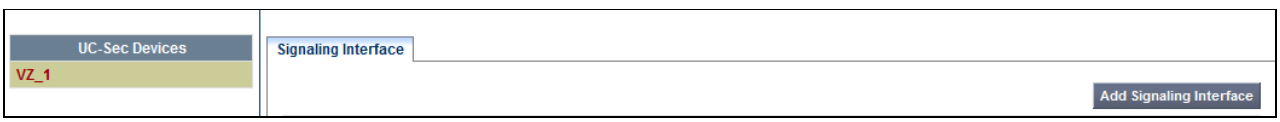


7.11. Device Specific Settings – Signaling Interface

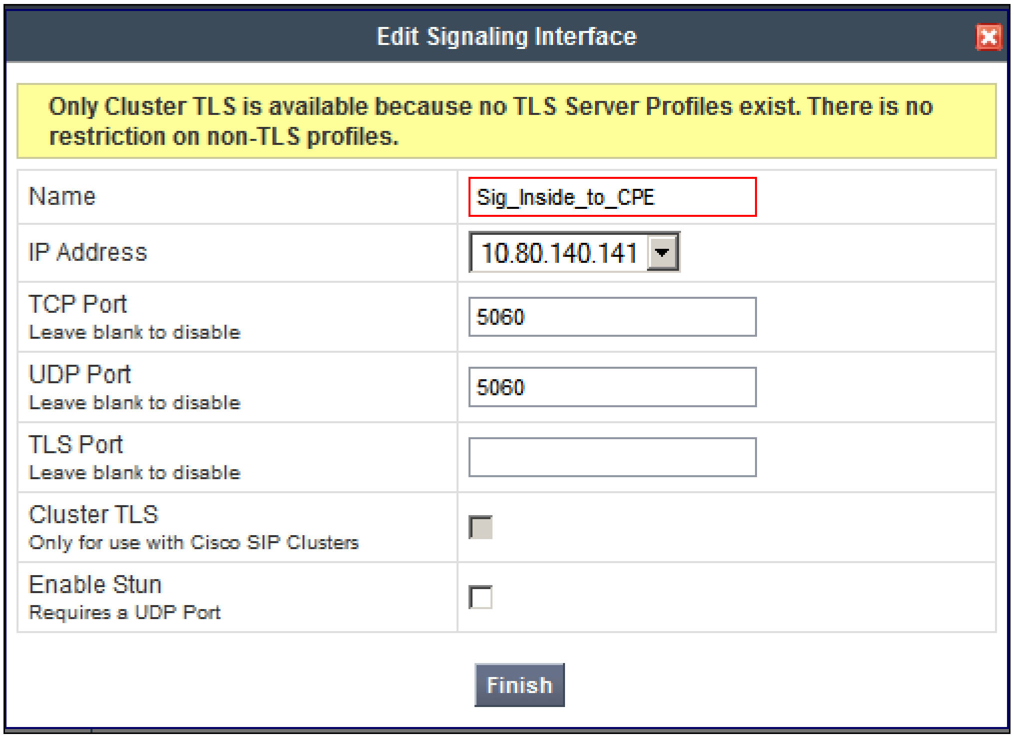
Select **Device Specific Setting → Signaling Interface** from the left-side menu as shown below.



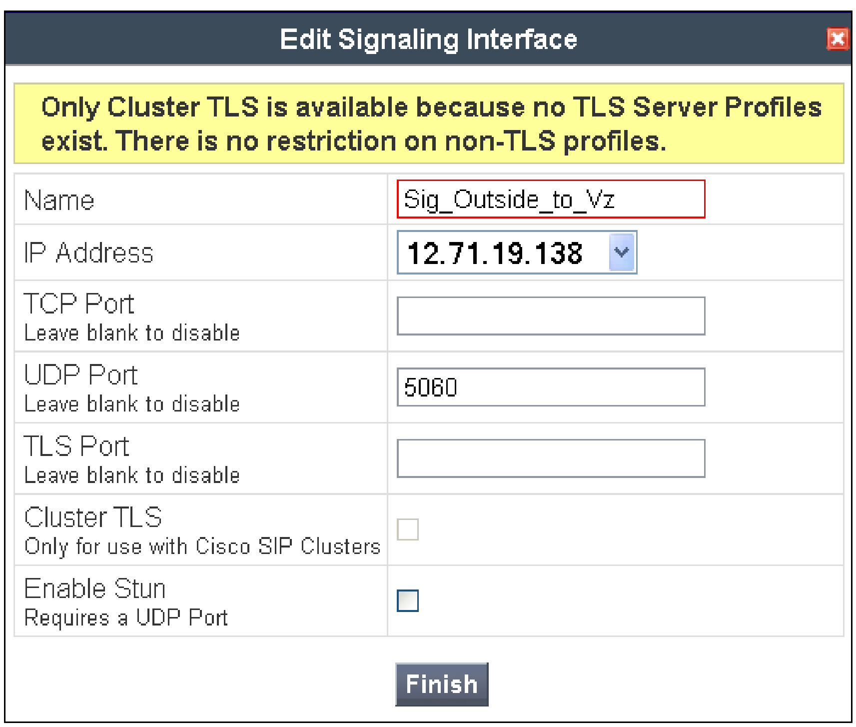
Under **UC-Sec Devices**, select the device being managed, which was named "VZ_1" in the sample configuration. Select **Add Signaling Interface**.



In the **Edit Signaling Interface** screen, enter an appropriate **Name** (e.g., "Sig_Inside_to_CPE") for the "inside" private interface, and choose the private inside IP Address (e.g., 10.80.140.141) from the **IP Address** drop-down menu. Choose **TCP Port** "5060" since TCP and port 5060 is used between Session Manager and the ASBCE in the sample configuration. Click **Finish**.



Once again, select **Add Signaling Interface**. In the Add Signaling Interface screen, enter an appropriate **Name** (e.g., "Sig_Outside_to_VZ") for the "outside" public interface, and choose the public IP Address for signaling (e.g., "12.71.19.138") from the **IP Address** drop-down box. Choose **UDP Port** "5060". In the sample configuration, Verizon will send SIP signaling using UDP to the CPE IP Address 12.71.19.138 and to UDP Port 5060. Click **Finish**.



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

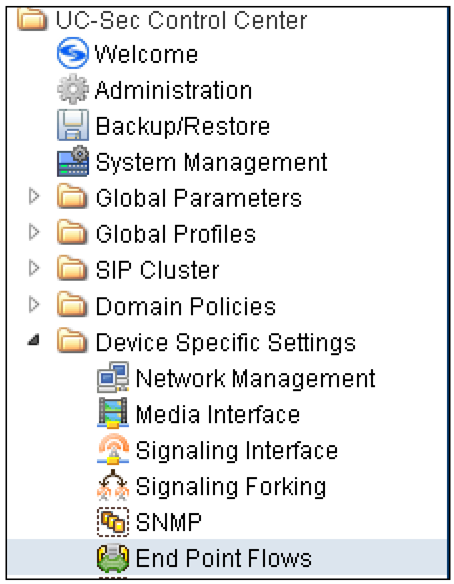
Signaling Interface

Add Signaling Interface

Name	Signaling IP	TCP Port	UDP Port	TLS Port	TLS Profile
Sig_Inside_to_CPE	10.80.140.141	5060	5060	---	None
Sig_Outside_to_Vz	12.71.19.138	---	5060	---	None

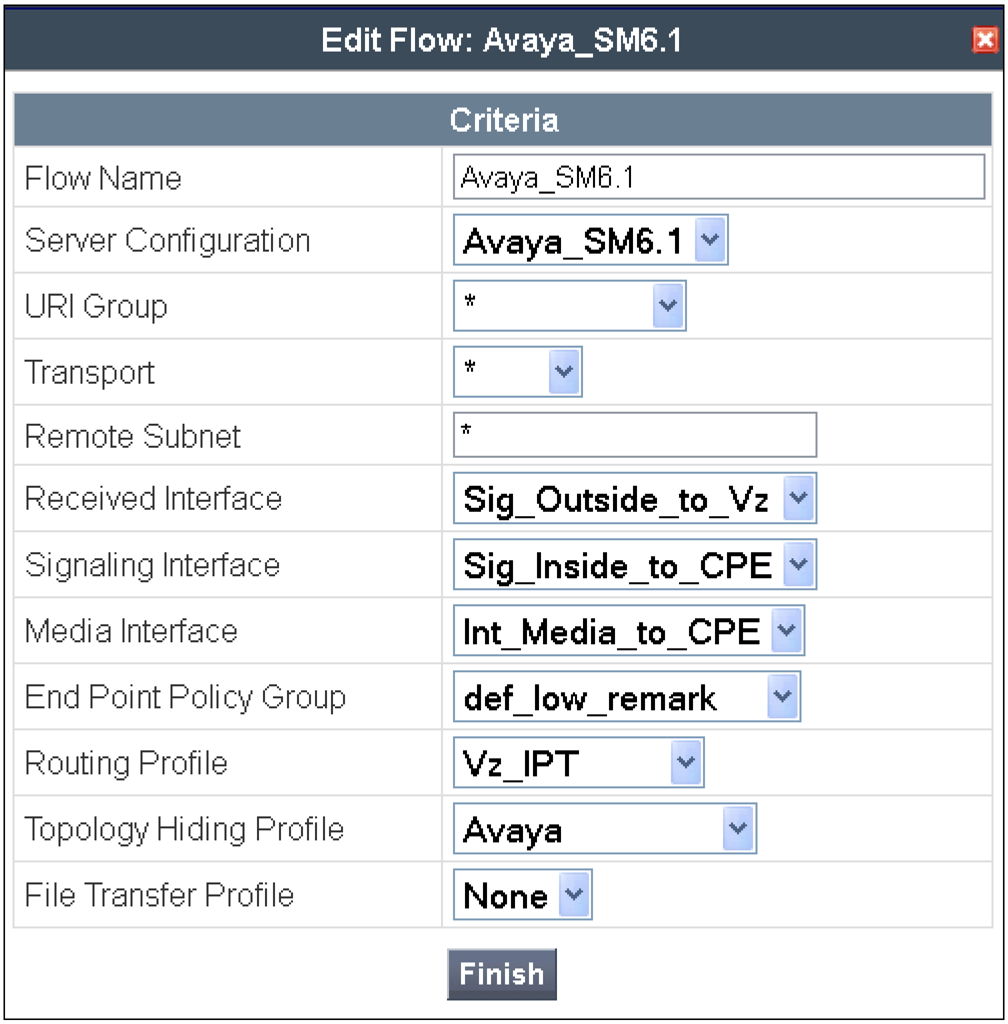
7.12. Device Specific Settings – End Point Flows

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

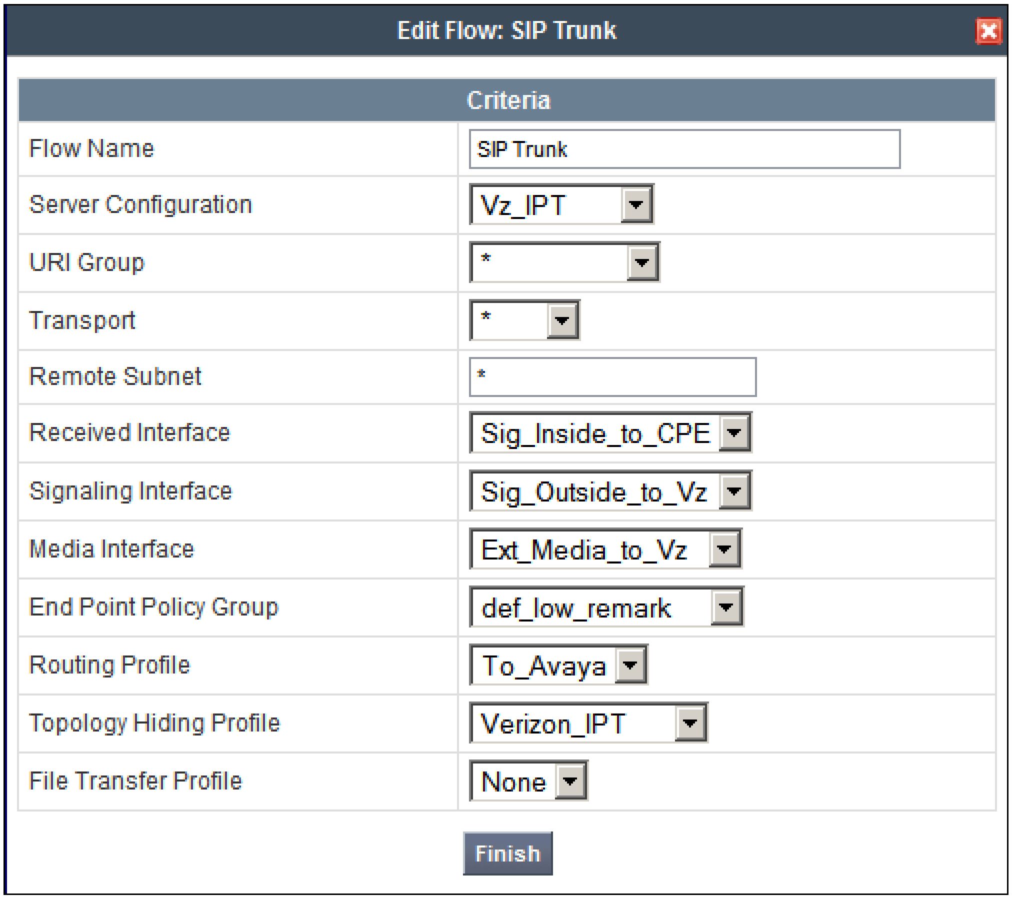


Under **UC-Sec Devices**, select the device being managed, which was named "VZ_1" in the sample configuration (not shown). Select the **Server Flows** tab. Select **Add Flow**.

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



Once again, select the **Server Flows** tab. Select **Add Flow**. The following screen shows the flow named "Vz_IPT" being added to the sample configuration. This flow uses the interfaces, policies, and profiles defined in previous sections. Click **Finish**.



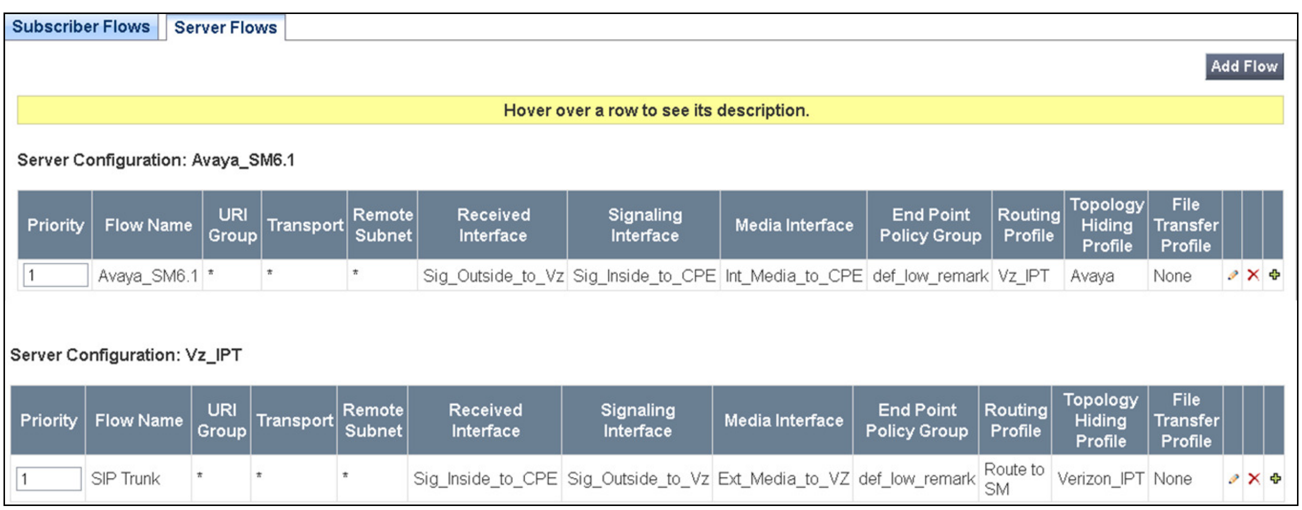
Edit Flow: SIP Trunk

Criteria

Flow Name	SIP Trunk
Server Configuration	Vz_IPT
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Sig_Inside_to_CPE
Signaling Interface	Sig_Outside_to_Vz
Media Interface	Ext_Media_to_Vz
End Point Policy Group	def_low_remark
Routing Profile	To_Avaya
Topology Hiding Profile	Verizon_IPT
File Transfer Profile	None

Finish

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



Subscriber Flows | Server Flows

Add Flow

Hover over a row to see its description.

Server Configuration: Avaya_SM6.1

Priority	Flow Name	URI Group	Transport	Remote Subnet	Received Interface	Signaling Interface	Media Interface	End Point Policy Group	Routing Profile	Topology Hiding Profile	File Transfer Profile
1	Avaya_SM6.1	*	*	*	Sig_Outside_to_Vz	Sig_Inside_to_CPE	Int_Media_to_CPE	def_low_remark	Vz_IPT	Avaya	None

Server Configuration: Vz_IPT

Priority	Flow Name	URI Group	Transport	Remote Subnet	Received Interface	Signaling Interface	Media Interface	End Point Policy Group	Routing Profile	Topology Hiding Profile	File Transfer Profile
1	SIP Trunk	*	*	*	Sig_Inside_to_CPE	Sig_Outside_to_Vz	Ext_Media_to_VZ	def_low_remark	Route to SM	Verizon_IPT	None

8. Verizon Business IP Trunk Services Suite Configuration

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

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

8.1. Service Access Information

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

CPE (Avaya)	Verizon Network
12.71.19.138 UDP port 5060	***icrcn1n0002.customer08.tsengr.com UDP Port 5208***
12.71.19.138 UDP port 5060	***icrcn1n0002.customer34.tsengr.com UDP Port 5234***

IP DID Numbers
408-990-8838
408-990-8837
33176759456
33176759457

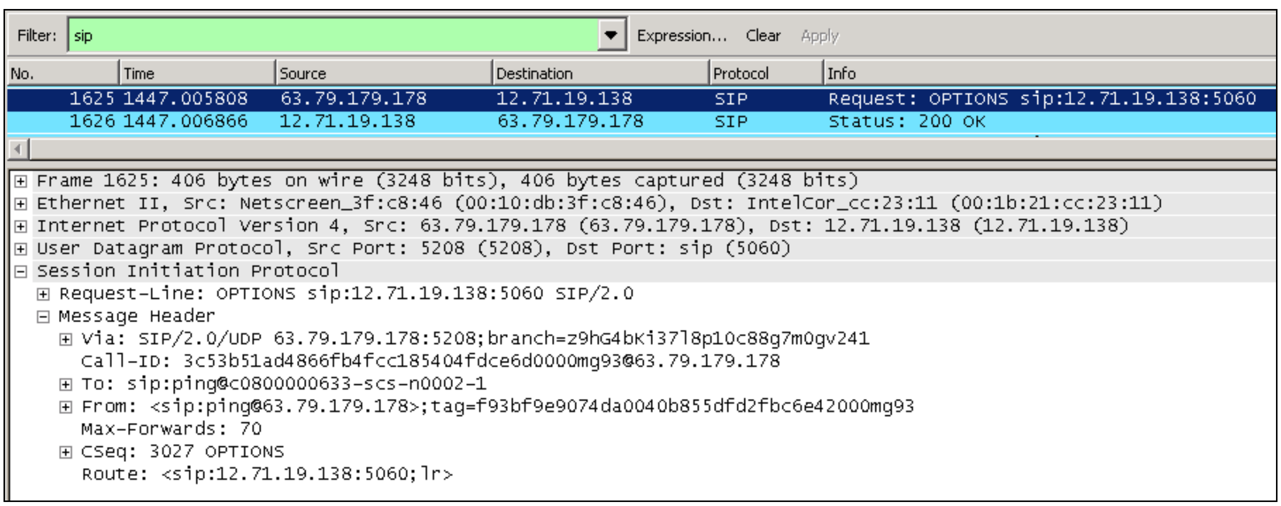
9. Verification Steps

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

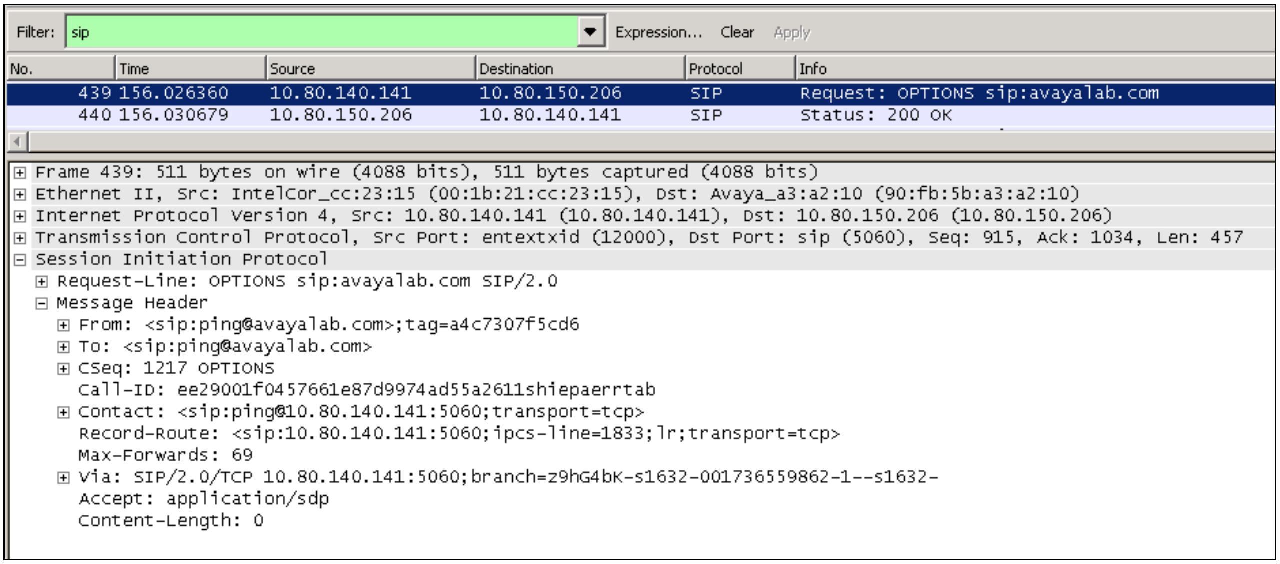
9.1. Illustration of OPTIONS Handling

This section illustrates SIP OPTIONS monitoring of the SIP trunk from Verizon to the CPE and from the CPE to Verizon through the ASBCE.

The following screens from a filtered Wireshark trace illustrate OPTIONS sent by Verizon to the CPE. Verizon IP Trunk service uses OPTIONS to determine whether the CPE is available to receive inbound calls. Therefore, proper OPTIONS response is necessary. In the trace shown below, taken from the outside public side of the ASBCE, frame 1625 is highlighted and expanded to show OPTIONS sent from the Verizon IPC Trunk (63.79.179.178) to the ASBCE (12.71.19.138). Observe the use of UDP for transport, from source port 5060 (Avaya) to destination port 5208 (Verizon). Note that Max-Forwards is 70.



Before the ASBCE replies to Verizon, the ASBCE sends OPTIONS to Session Manager on the inside private interface. In the trace shown below, taken from the private side of the ASBCE, frame 439 is highlighted and expanded to show OPTIONS sent from the inside interface of the ASBCE (10.80.140.141) to Session Manager (10.80.150.206). Observe the use of TCP for transport, using port 5060. Observe that the ASBCE has changed the Request-URI, From and To headers per the previous configuration such that “avayalab.com” now appears. Note that Max-Forwards has been decremented by 1 and is now 69.



9.2. Avaya Aura® Communication Manager Verifications

This section illustrates verifications from Communication Manager.

9.2.1 Example Incoming Call from PSTN via Verizon SIP Trunk

Incoming PSTN calls arrive from Verizon at ASBCE, which sends the call to Session Manager. In the sample configuration, when the ASBCE is in-service, Verizon sends all inbound calls to ASBCE-1 (i.e., not load balanced). Session Manager sends the call to Communication Manager via the entity link corresponding to the Avaya HP Common Server using port 5062. On Communication Manager, the incoming call arrives via signaling group 5 and trunk group 5.

The following edited Communication Manager ***list trace tac*** trace output shows a call incoming on trunk group 68. The PSTN telephone dialed 408-990-8838. Session Manager can map the number received from Verizon to the extension of a Communication Manager telephone (x7689), or the incoming call handling table for trunk group 5 can do the same. In the trace below, Communication Manager had already mapped the Verizon DID to Communication Manager extension. Extension 7689 is an IP Telephone with IP address 10.80.140.133 in Region 1. Initial IP-IP media is set to y, so the call RTP media path is “ip-direct” from the IP Telephone (10.80.140.133) to the “inside” of the ASBCE (10.80.140.141).

```
list trace tac *105                                                  Page   1
                                LIST TRACE
time            data
13:50:35TRACE STARTED 05/24/2012 CM Release String cold-00.1.510.1-19528
13:50:42 SIP<INVITE sip:7689@avayalab.com SIP/2.0
13:50:42     Call-ID: BW154843154260212214185365l
13:50:42     active trunk-group 5 member 1    cid 0xcc2
13:50:42 SIP>SIP/2.0 180 Ringing
13:50:42     Call-ID: BW154843154260212214185365l
13:50:42     dial 2011
13:50:42     ring station      2011 cid 0xcc2
13:50:50 SIP>SIP/2.0 200 OK
13:50:50     Call-ID: BW154843154260212214185365l
13:50:50     active station      2011 cid 0xcc2
13:50:50     G729A ss:off ps:20
             rgn:1 [10.80.140.133]:2890
             rgn:4 [10.80.140.141]:35072
13:50:50     G729A ss:off ps:20
             rgn:4 [10.80.140.141]:35072
             rgn:1 [10.80.140.133]:2890
13:50:50 SIP<ACK sip: 4089908838@10.80.140.146:5062;transport=tcp SIP
13:50:50 SIP</2.0
13:50:50     Call-ID: BW154843154260212214185365l
13:50:54 SIP<BYE sip:4089908838@10.80.140.146:5062;transport=tcp SIP
13:50:54 SIP</2.0
13:50:54     Call-ID: BW154843154260212214185365l
13:50:54 SIP>SIP/2.0 200 OK
13:50:54     Call-ID: BW154843154260212214185365l
13:50:54     idle trunk-group 68 member 1    cid 0xcc2
```

The following screen shows **Page 2** of the output of the ***status trunk*** command pertaining to this same call. Note the signaling using port 5060 between Communication Manager and Session Manager. Note the media is "ip-direct" from the IP Telephone (10.80.140.133) to the inside IP address of ASBCE (10.80.140.141) using G.729.

```
status trunk 68/1                                             Page   2 of   3
                                CALL CONTROL SIGNALING
Near-end Signaling Loc: PROCR
  Signaling   IP Address                               Port
   Near-end:  10.80.140.146                          : 5060
    Far-end:  10.80.150.206                          : 5060
 H.245 Near:
  H.245 Far:
   H.245 Signaling Loc:           H.245 Tunneled in Q.931? no

 Audio Connection Type: ip-direct     Authentication Type: None
    Near-end Audio Loc:                        Codec Type: G.729A
   Audio      IP Address                               Port
   Near-end:  10.80.140.133                          : 2890
    Far-end:  10.80.140.141                          : 35070

 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.

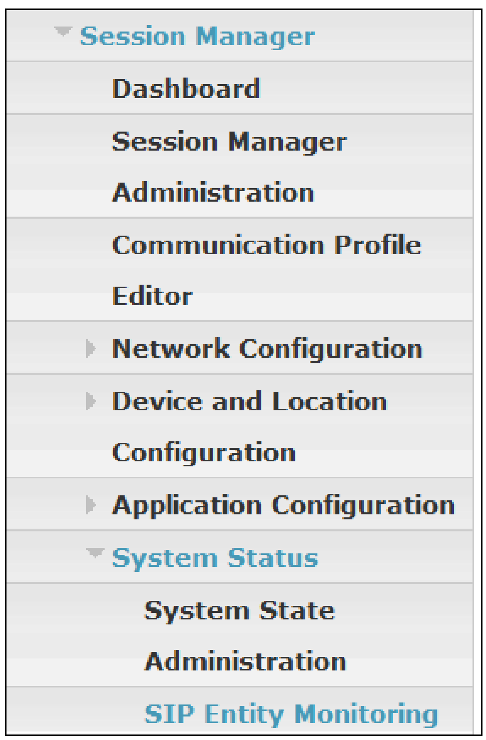
```
status trunk 68/1                                              Page    3 of    3
                      SRC PORT TO DEST PORT TALKPATH
src port: T00031
T00031:TX:10.80.140.141:35070/g729a/20ms
S00001:RX:10.80.140.133:2890/g729a/20ms

dst port: S00001
```

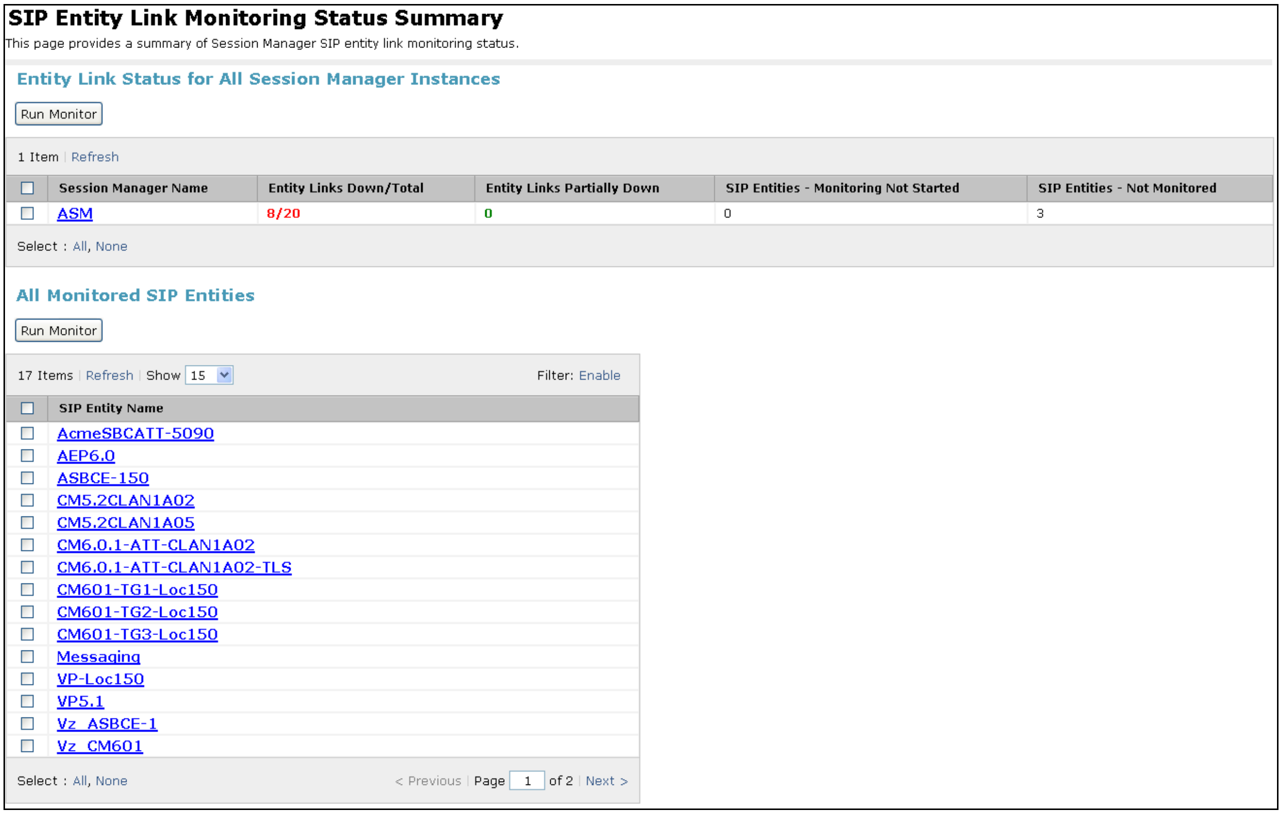
9.3. Avaya Aura® System Manager and Avaya Aura® Session Manager Verifications

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

9.3.1 Verify SIP Entity Link Status

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





SIP Entity Link Monitoring Status Summary

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

Entity Link Status for All Session Manager Instances

Run Monitor

1 Item | Refresh

	Session Manager Name	Entity Links Down/Total	Entity Links Partially Down	SIP Entities - Monitoring Not Started	SIP Entities - Not Monitored
	ASM	8/20	0	0	3

Select : All, None

All Monitored SIP Entities

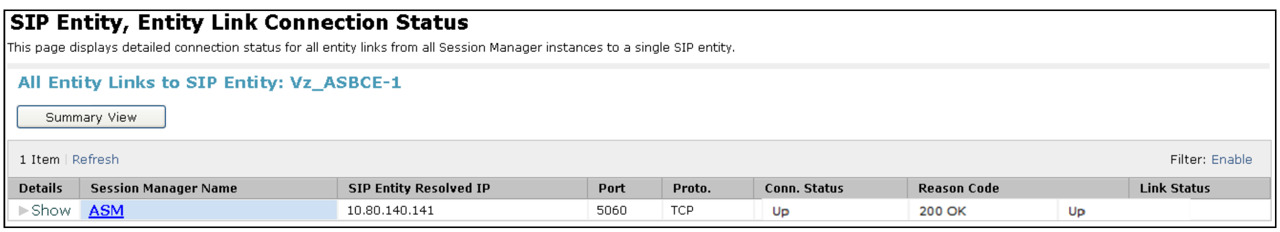
Run Monitor

17 Items | Refresh | Show 15 Filter: Enable

	SIP Entity Name
	AcmeSBCATT-5090
	AEP6.0
	ASBCE-150
	CM5.2CLAN1A02
	CM5.2CLAN1A05
	CM6.0.1-ATT-CLAN1A02
	CM6.0.1-ATT-CLAN1A02-TLS
	CM601-TG1-Loc150
	CM601-TG2-Loc150
	CM601-TG3-Loc150
	Messaging
	VP-Loc150
	VP5.1
	Vz_ASBCE-1
	Vz_CM601

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

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.



SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

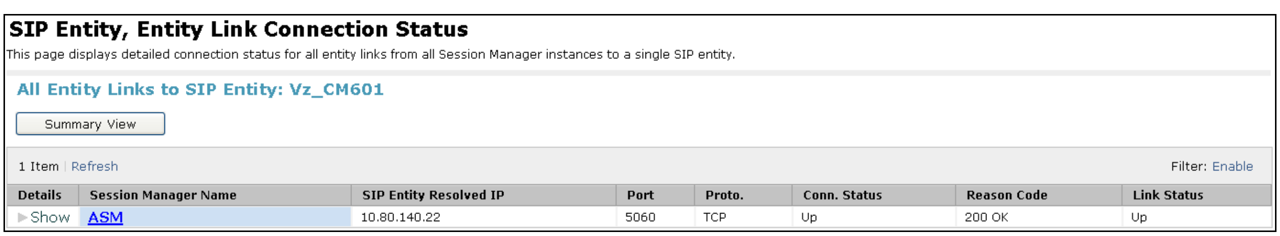
All Entity Links to SIP Entity: Vz_ASBCE-1

Summary View

1 Item | Refresh Filter: Enable

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
▶Show	ASM	10.80.140.141	5060	TCP	Up	200 OK	Up

Return to the list of monitored entities, and select another entity of interest, such as “Vz_CM601”. Under normal operating conditions, the **Link Status** should be “Up” as shown in the example screen below. Note the use of port 5060.



SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

All Entity Links to SIP Entity: Vz_CM601

Summary View

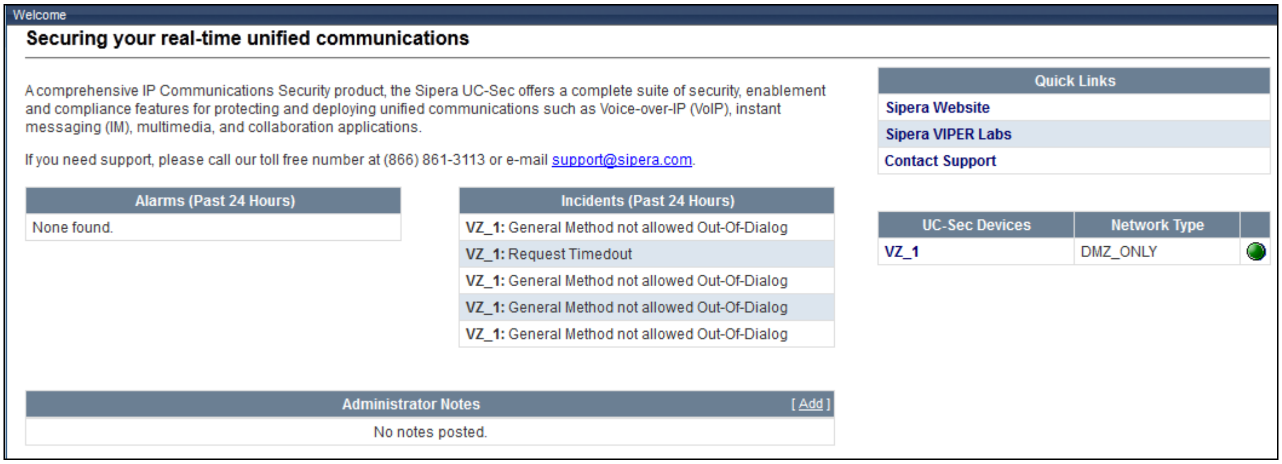
1 Item | Refresh Filter: Enable

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
▶Show	ASM	10.80.140.22	5060	TCP	Up	200 OK	Up

9.4. Avaya Session Border Controller for Enterprise Verification

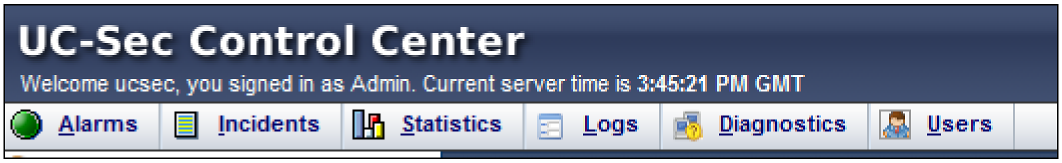
9.4.1 Welcome Screen

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

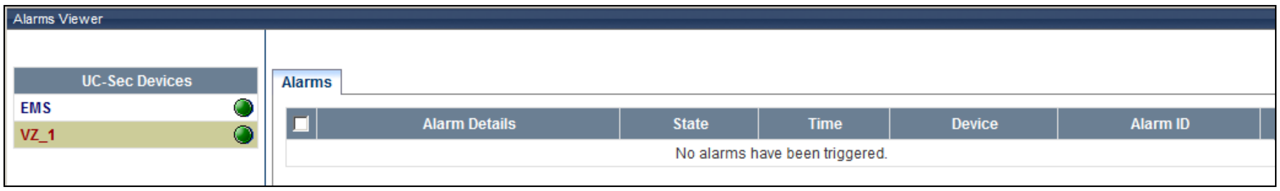


9.4.2 Alarms

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



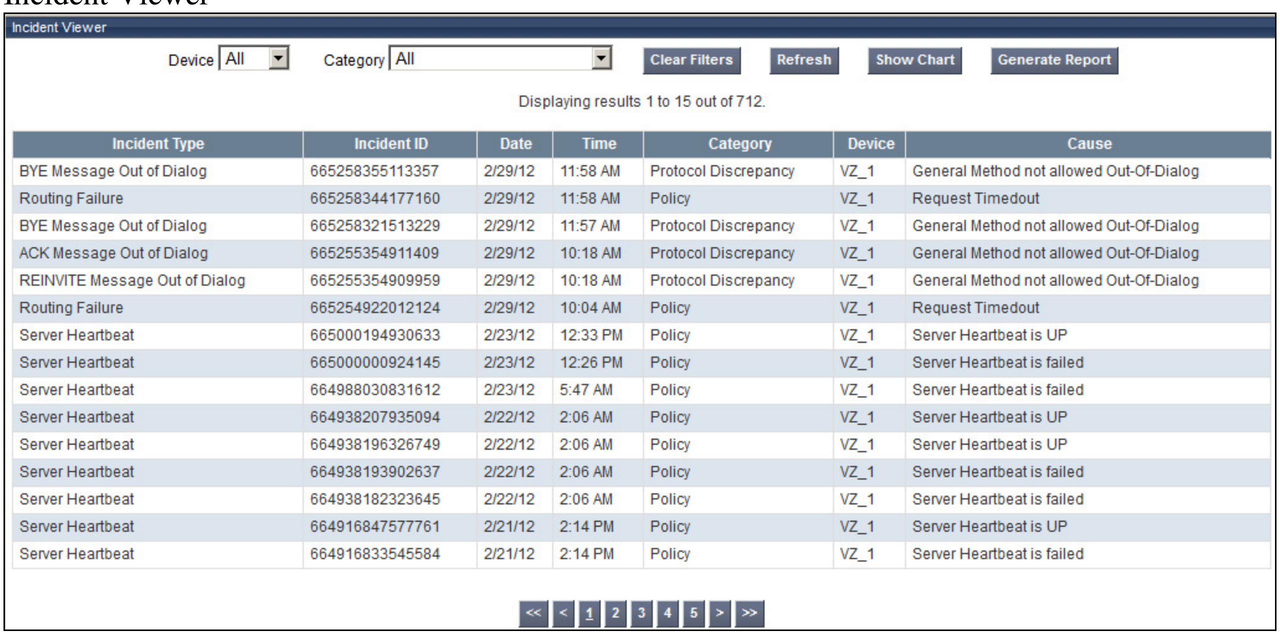
Alarms Viewer.



9.4.3 Incidents

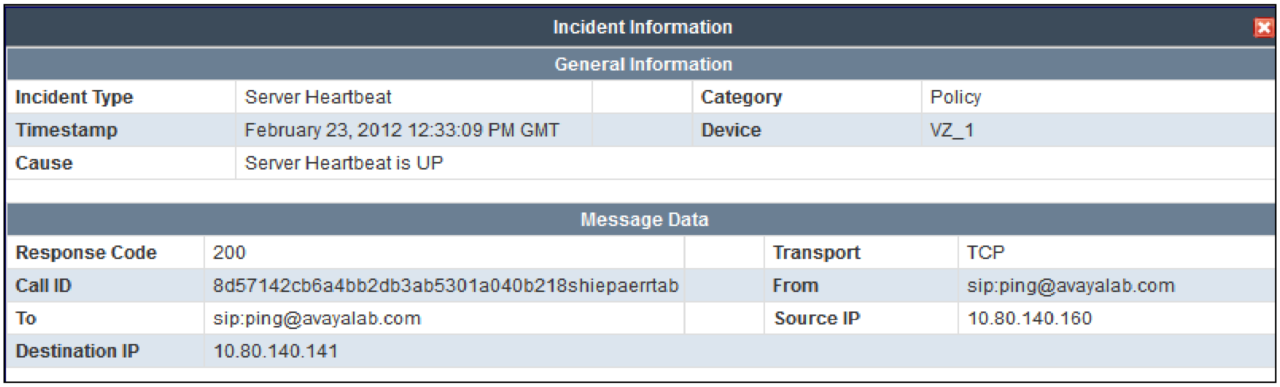
A list of all recent incidents can be found under the incidents tab at the top left next to the Alarms.

Incident Viewer



Incident Type	Incident ID	Date	Time	Category	Device	Cause
BYE Message Out of Dialog	665258355113357	2/29/12	11:58 AM	Protocol Discrepancy	VZ_1	General Method not allowed Out-Of-Dialog
Routing Failure	665258344177160	2/29/12	11:58 AM	Policy	VZ_1	Request Timedout
BYE Message Out of Dialog	665258321513229	2/29/12	11:57 AM	Protocol Discrepancy	VZ_1	General Method not allowed Out-Of-Dialog
ACK Message Out of Dialog	665255354911409	2/29/12	10:18 AM	Protocol Discrepancy	VZ_1	General Method not allowed Out-Of-Dialog
REINVITE Message Out of Dialog	665255354909959	2/29/12	10:18 AM	Protocol Discrepancy	VZ_1	General Method not allowed Out-Of-Dialog
Routing Failure	665254922012124	2/29/12	10:04 AM	Policy	VZ_1	Request Timedout
Server Heartbeat	665000194930633	2/23/12	12:33 PM	Policy	VZ_1	Server Heartbeat is UP
Server Heartbeat	665000000924145	2/23/12	12:26 PM	Policy	VZ_1	Server Heartbeat is failed
Server Heartbeat	664988030831612	2/23/12	5:47 AM	Policy	VZ_1	Server Heartbeat is failed
Server Heartbeat	664938207935094	2/22/12	2:06 AM	Policy	VZ_1	Server Heartbeat is UP
Server Heartbeat	664938196326749	2/22/12	2:06 AM	Policy	VZ_1	Server Heartbeat is UP
Server Heartbeat	664938193902637	2/22/12	2:06 AM	Policy	VZ_1	Server Heartbeat is failed
Server Heartbeat	664938182323645	2/22/12	2:06 AM	Policy	VZ_1	Server Heartbeat is failed
Server Heartbeat	664916847777761	2/21/12	2:14 PM	Policy	VZ_1	Server Heartbeat is UP
Server Heartbeat	664916833545584	2/21/12	2:14 PM	Policy	VZ_1	Server Heartbeat is failed

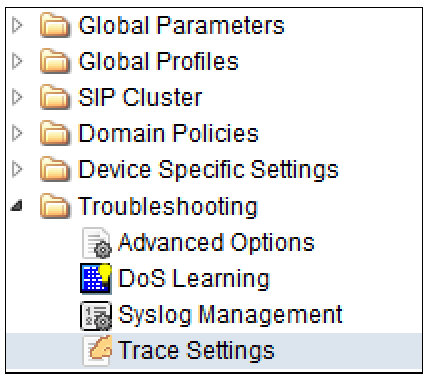
Further Information can be obtained by clicking on an incident in the incident viewer.



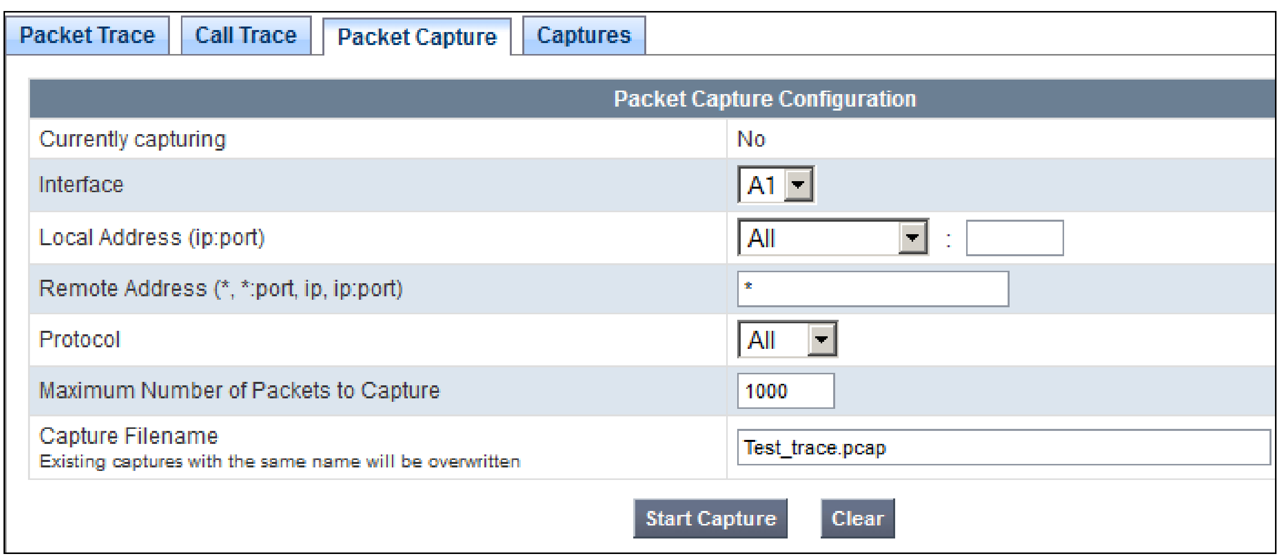
Incident Information			
General Information			
Incident Type	Server Heartbeat	Category	Policy
Timestamp	February 23, 2012 12:33:09 PM GMT	Device	VZ_1
Cause	Server Heartbeat is UP		
Message Data			
Response Code	200	Transport	TCP
Call ID	8d57142cb6a4bb2db3ab5301a040b218shiepaerrtab	From	sip:ping@avayalab.com
To	sip:ping@avayalab.com	Source IP	10.80.140.160
Destination IP	10.80.140.141		

9.4.4 Tracing

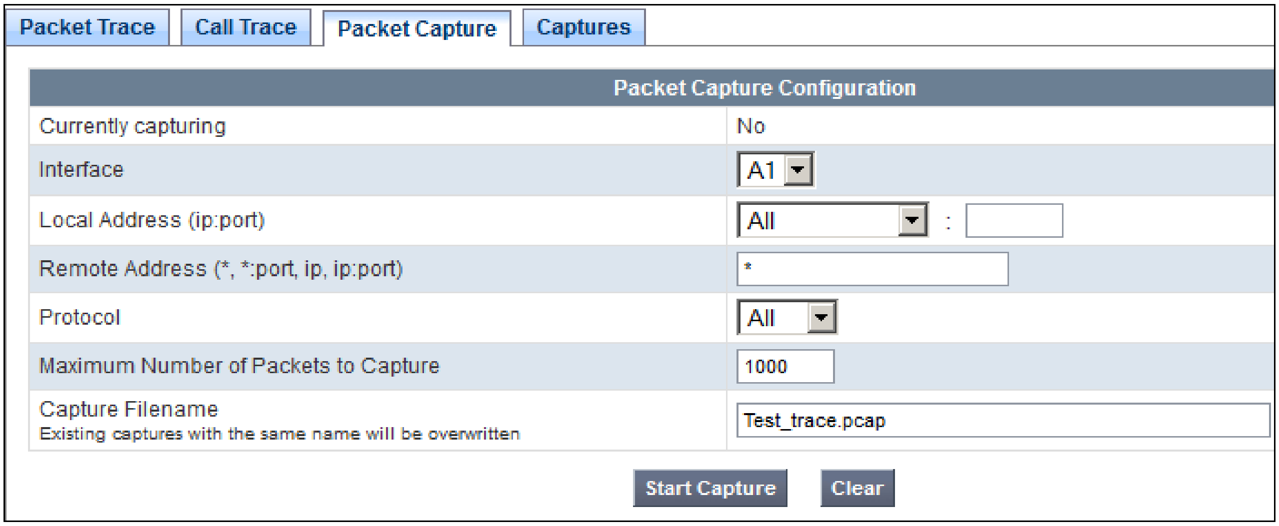
To take a call trace, Select **Troubleshooting → Tracing** from the left-side menu as shown below.



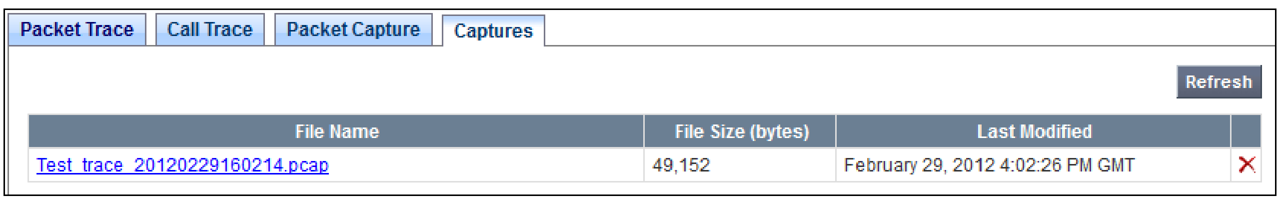
Select the Packet Capture tab and set the desired configuration for a call trace, hit **Start Capture**. Only one interface can be selected at once, so only an inside or only an outside trace is possible.



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



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



10. Conclusion

As illustrated in these Application Notes, Avaya Aura® Communication Manager 6.0.1, Avaya Aura® Session Manager 6.1, and Avaya Session Border Controller for Enterprise can be configured to interoperate successfully with Verizon Business IP Trunk service, inclusive of the "2-CPE" SIP trunk redundancy architecture. This solution allows Avaya Aura® Communication Manager and Avaya Aura® Session Manager users to access the PSTN using a Verizon Business IP Trunk public SIP trunk service connection.

11. Additional References

11.1. Avaya

[1] *Administering Avaya Aura® Communication Manager*, (Aug 2010), Document Number 03-300509.

[2] *Avaya Aura® Communication Manager Feature Description and Implementation,* Release 6.0, 555-245-205, Issue 8.0, June 2010

[3] *Installing and Configuring Avaya Aura® Session Manager,* Doc ID 03-603473 Release 6.

[4] *Administering Avaya Aura® Session Manager*, Doc ID 03-603324, Release 6.0, June 2010
[5] *Avaya 1600 Series IP Deskphones Administrator Guide Release 1.2.x,* February 2010, Document Number 16-601443.
[6] *4600 Series IP Telephone LAN Administrator Guide,* October 2007, Document Number 555-233-507.
[7] *Avaya one-X® Deskphone Edition for 9600 Series IP Telephones Administrator Guide,* November 2009, Document Number 16-300698.
[8] *Avaya one-X® Communicator Getting Started, November 2009.*
[9] RFC 3261 *SIP: Session Initiation Protocol,* http://www.ietf.org/
[10] RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals, http://www.ietf.org/
[11] RFC 4244, An Extension to the Session Initiation Protocol (SIP) for Request History Information, http://www.ietf.org/

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

11.2. Verizon Business

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

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

Appendix A: Avaya Session Border Control for Enterprise – Sigma Script "EXAMPLE 2"

```
within session "ALL"
{
 act on message where %DIRECTION="OUTBOUND" and
%ENTRY_POINT="POST_ROUTING"
  {
// Topology Hiding of P-Location header for subsequent re-INVITEs

   remove(%HEADERS["P-Location"][1]);
   remove(%HEADERS["Endpoint-View"][1]);
   remove(%HEADERS["Alert-Info"][1]);
   remove(%HEADERS["User-Agent"][1]);
   remove(%HEADERS["Server"][1]);
   %HEADERS["Referred-
By"][1].regex_replace("7689@63.79.179.178:5208","4089908838@12.71.19.138");
   %HEADERS["Contact"][1].regex_replace("7689","4089908838");
   %HEADERS["Referred-
By"][1].regex_replace("7633@63.79.179.178:5208","4089908837@12.71.19.138");
   %HEADERS["Contact"][1].regex_replace("7633","4089908837");

   }
 }
```

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.