Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 7.0, Avaya Aura® Session Manager 7.0, and Avaya Session Border Controller for Enterprise 7.0 with Verizon Business IP Trunk SIP Trunk Service – Issue 1.0

Abstract

These Application Notes illustrate a sample configuration using Avaya Aura® Session Manager Release 7.0, Avaya Aura® Communication Manager Release 7.0, and Avaya Session Border Controller for Enterprise Release 7.0 with the Verizon Business IP Trunk SIP Trunk service. These Application Notes update previously published Application Notes with newer versions of Communication Manager, Session Manager, and Avaya Session Border Controller for Enterprise.

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

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

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as any observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab, utilizing a Verizon Business Private IP (PIP) circuit connection to the production Verizon Business IP Trunking service.

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1. Introduction

These Application Notes illustrate a sample configuration using Avaya Aura® Session Manager Release 7.0, Avaya Aura® Communication Manager Release 7.0, and Avaya Session Border Controller for Enterprise Release 7.0 with the Verizon Business IP Trunk SIP Trunk service (Verizon Business IP Trunk service). The Verizon Business IP Trunk service provides local and/or long-distance calls (with PSTN endpoints) via standards-based SIP trunks.

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

2. General Test Approach and Test Results

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

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

2.1. Interoperability Compliance Testing

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

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

- EC500
- Long hold time calls
- Remote Worker

2.2. Test Results

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

- Verizon provisioned T.38 fax on the production circuit used to verify these Application Notes. Verizon Business IP Trunk service requires all fax calls to start off with G.711 as the first codec choice, and relies on the CPE to send a re-Invite to T.38 when placing or receiving a fax call. Verizon Business IP Trunk service will never send a re-Invite to T.38. If the **FAX Mode** field on the Communication Manager ip-codec-set form page 2 is set to "**t.38-standard**" (see **Section 5.6**), Communication Manager will send the proper re-Invite to T.38 for both inbound and outbound fax calls, but will not failback to G.711 should the Verizon network reject the Communication Manager attempt to transition to T.38 by sending a 488 Not Acceptable message. Communication Manager Release 6.3 introduced the T.38 Fax with Fallback to G.711 Pass-Through feature. This provides the functionality for Communication Manager to interoperate with Verizon networks by re-Inviting to G.711 after receiving a 488 Not Acceptable message. If the **FAX Mode** is set to "**t.38-G711-fallback**" setting[1], Communication Manager will send a re-Invite to T.38 for inbound fax calls only and relies on the far end to send a re-Invite to T.38 for outbound calls. Communication Manager assumes T.38 fax is not supported for an outbound fax call unless an Invite for T.38 is received. The result is an outbound fax sent using G.711, even though the circuit is provisioned for T.38. Inbound fax calls negotiate properly to T.38. With the limitations of T.38 on Verizon's network and Verizon's requirement for fax calls to start off with G.711 as the first codec choice, it is recommended to use an AudioCodes MP-114 or MP-124 Gateway between Session Manager and the fax device when fax is used with Verizon Business IP Trunk service.

- When the **Initial IP-IP Direct Media** field on the Communication Manager signaling group form page 1 is set to "**y**", Communication Manager sends a "183 Session Progress" without SDP during an inbound PSTN call that is forwarded to another PSTN call just before a 183 is sent with SDP information to the far end. This is undesirable to Verizon and could result in no audio. The recommendation in **Section 5.7** is to leave the **Initial IP-IP Direct Media** field to "**n**".

- When an Avaya SBCE Interworking Profile is configured on the Verizon Server Configuration profile, Avaya SBCE inserts "Supported: replaces" header in the SIP message towards the Call Server. This can create an issue when the service provider includes a Supported header with no value within the SIP request messages, which is the case with Verizon. This will cause two Supported headers to be sent towards Session

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

Manager and Session Manager will convert these two SIP headers into one header, "Supported: **,** replaces". Communication Manager cannot parse a SIP message with a header starting with a comma ",". To prevent this issue, no Interworking Profile is set on the Trunk Server's Server Configuration on the Avaya SBCE. Therefore the extra Supported header was not inserted. See **Section 7.4.2**.

- When using the Avaya Media Server for VoIP resources, and G.722 as the first codec choice, inbound SIP trunk calls to SIP phones may result in anchored media through the Avaya Media Server for the duration of the call. As a workaround, omit the G.722 codec from the Communication Manager IP Codec Set between the SIP phones and the Avaya Media Server. Development is currently investigating this issue as CM-8086.

- Emergency 911/E911 Services Limitations and Restrictions - Although Verizon provides 911/E911 calling capabilities, 911 capabilities were not tested; therefore, it is the customer's responsibility to ensure proper operation with its equipment/software vendor.

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

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

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

2.3. History Info and Diversion Headers

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

If Communication Manager sends the History Info Header, Session Manager can convert the History Info header into the Diversion Header. This is performed by specifying the "*VerizonAdapter*" adaptation in Session Manager. See **Section 6.3**.

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

2.4. SIP Header Removal

To support advanced SIP telephony features in the Avaya Aura® enterprise environment, certain proprietary headers may be included in the SIP message sent toward Verizon. These extra headers can cause the SIP message to become larger than the specified Maximum Transmission Unit (MTU), and create fragmented UDP packets. These fragmented packets may not be re-assembled properly on the far-end by Verizon's equipment, for instance, when packets arrive out of order. To prevent fragmented packets, any unnecessary or proprietary headers should be removed from the

SIP message before being sent to Verizon. Session Manager can remove these headers by specifying the "*eRHdrs*" parameter within the "*VerizonAdapter*" adaptation. See **Section 6.3**.

In the Sample Configuration, the following headers were removed:

- AV-Global-Sesison-ID
- Alert-Info
- Endpoint-View
- P-AV-Message-Id
- P-Charging-vector
- P-Location

To help reduce the packet size further, the Avaya SBCE can remove the "*gsid*" and "*epv*" parameters that may be included within the Contact header by applying a Sigma script to the Verizon server configuration. See **Section 7.3** and **7.4.2**.

2.5. Support

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

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

3. Reference Configuration

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

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

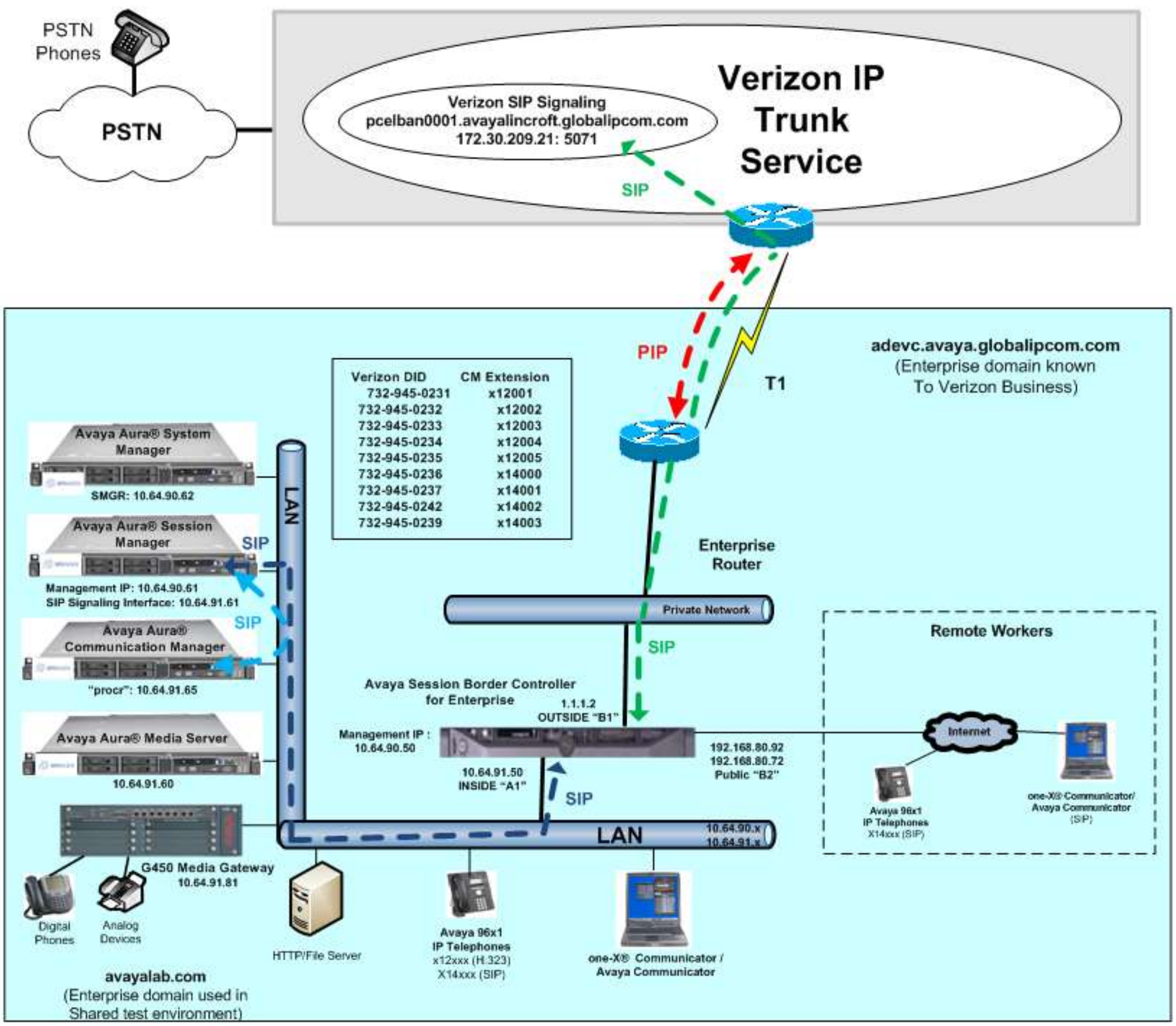


Figure 1: Avaya Interoperability Test Lab Configuration

The Verizon Business IP Trunk service used FQDN *pcelban0001.avayalincroft.globalipcom.com*. The Avaya CPE environment was known to Verizon Business IP Trunk service as FQDN *adevc.avaya.globalipcom.com.* Access to the Verizon Business IP Trunk service was added to a configuration that already used domain "avayalab.com" at the enterprise. As such, the Avaya SBCE is used to adapt the "avayalab.com" domain to the domain known to Verizon (see **Section 7.6**). These Application Notes indicate a configuration that would not be required in cases where the CPE domain in Communication Manager and Session Manager match the CPE domain known to the Verizon Business IP Trunk service.

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

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

- Verizon Business IP Trunk network Fully Qualified Domain Name (FQDN)
  - *pcelban0001.avayalincroft.globalipcom.com*
- Avaya CPE Fully Qualified Domain Name (FQDN) known to Verizon
  - *adevc.avaya.globalipcom.com*
- Avaya Session Border Controllers for Enterprise
- Avaya Aura® Communication Manager
- Avaya Aura® Session Manager Release
- Avaya G450 Media Gateway
- Avaya Media Server
- Avaya 96X1 Series IP telephones using the SIP and H.323 software bundle
- Avaya one-X® Communicator
- Avaya Communicator for Windows
- Avaya Digital Phones
- AudioCodes MP114

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	7.0-441.0-22477
Avaya Aura® System Manager	7.0.0.0.3929
Avaya Aura® Session Manager	7.0.0.0.700007
Avaya Session Border Controller for Enterprise	7.0.0-21-6602
Avaya Aura® Messaging	6.3.2 SP 2
Avaya Aura® Media Server	7.7.0.235
G450 Gateway	37.19.0
Avaya 96X1- Series Telephones (SIP)	R7.0.0.39
Avaya 96X1- Series Telephones (H323)	R6.2313
Avaya one-X® Communicator	6.2.7
Avaya Communicator for Windows	2.1.2.75
Avaya 2400-Series and 6400-Series Digital Telephones	N/A
AudioCodes MP-114	6.20A.035.001
Okidata Analog Fax	N/A

Table 1: Equipment and Software Used in the Sample Configuration

5. Configure Avaya Aura® Communication Manager Release 7.0

This section illustrates an example configuration allowing SIP signaling via the "Processor Ethernet" of Communication Manager to Session Manager.

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

5.1. Verify Licensed Features

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

On **Page 2** of the ***display system-parameters customer-options*** form, verify that the **Maximum Administered SIP Trunks** are sufficient for the combination of trunks to the Verizon Business IP Trunk service offer and any other SIP applications. Each call from a non-SIP endpoint to the Verizon Business IP Trunk service uses one SIP trunk for the duration of the call. 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  12
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                                USED
                     Maximum Administered H.323 Trunks: 4000  0
           Maximum Concurrently Registered IP Stations: 2400  1
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  3
                   Maximum Video Capable IP Softphones: 2400  4
                       Maximum Administered SIP Trunks: 4000  30
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0
```

On **Page 4** of the ***display system-parameters customer-options*** form, verify that **ARS** is enabled.

```
display system-parameters customer-options                        Page   4 of  12
                                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 5** 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 will be required verify that the **ISDN/SIP Network Call Redirection** feature is enabled.

```
display system-parameters customer-options                        Page   5 of  12
                                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
```

On **Page 6** 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   6 of  12
                                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
```

5.2. Dial Plan

In the reference configuration, the Avaya CPE environment uses five digit local extensions such as 12xxx, 14xxx or 20xxx. Trunk Access Codes (TAC) are 3 digits in length and begin with *. The Feature Access Code (FAC) to access ARS is the single digit 9. The Feature Access Code (FAC) to access AAR is the single digit 8. The dial plan illustrated here is not intended to be prescriptive; any valid dial plan may be used.

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

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

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

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 "**SM**" with IP address **10.64.91.61**. The node name and IP address for the Processor Ethernet "**procr**" is **10.64.91.65**.

```
change node-names ip                                               Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
AMS                 10.64.91.60
SM                  10.64.91.61
default             0.0.0.0
procr               10.64.91.65
procr6              ::
```

5.4. Processor Ethernet Configuration

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

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


                  Type: PROCR
                                                      Target socket load: 4800

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


                                 IPV4 PARAMETERS
             Node Name: procr                    IP Address: 10.64.91.65


           Subnet Mask: /24
```

5.5. IP Codec Sets

The following screen shows the configuration for codec set 2, the codec set configured to be used for calls within region 2 and for calls between region 1 and region 2. In general, an IP codec set is a list of allowable codecs in priority order. Using the example configuration shown below, calls to and from the PSTN via the SIP trunks would use G.729A, since G.729A is the preferred codec by both Verizon and the Avaya ip-codec-set. Include G.711MU to support calls to Messaging.

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

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

The following screen shows **Page 2** of the form. Configure the **Fax Mode** field to "**t.38-G711-fallback**", set the **Fax Redundancy** field to "**0**", and **ECM** to "**y**". See **Section 2.2** for more details regarding fax and the recommendation to use an AudioCodes MP-1xx for fax.

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

                          IP CODEC SET

                              Allow Direct-IP Multimedia? n

                                                                    Packet
                          Mode                   Redundancy         Size(ms)
    FAX                   t.38-G711-fallback       0           ECM: y
    Modem                 off                      0
    TDD/TTY               US                       3
    H.323 Clear-channel   n                        0
    SIP 64K Data          n                        0                20
```

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

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

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

5.6. Network Regions

Network regions provide a means to logically group resources. In the shared Communication Manager configuration used for the testing, the Avaya G450 Media Gateway and Avaya Media Server are in region 1. To provide testing flexibility, network region 2 was associated with other components used specifically for the Verizon testing.

Non-IP telephones (e.g., analog, digital) derive network region and location configuration from the Avaya gateway to which the device is connected. The following display command shows that **Media Gateway 1** is an Avaya G450 Media Gateway configured for network region 1. It can also be observed that the **Controller IP Address** is the Avaya Processor Ethernet (10.64.91.65), and that the gateway IP address is 10.64.91.81. These fields are not configured in this screen, but just display the current information for the Media Gateway.

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

                    Type: g450
                    Name: G450-1
               Serial No: 08IS38199678
    Link Encryption Type: any-ptls/tls       Enable CF? n
          Network Region: 1                   Location: 1
                                             Site Data:
           Recovery Rule: 1

              Registered?  y
   FW Version/HW Vintage: 36 .14 .0  /1
        MGP IPV4 Address: 10.64.91.81
        MGP IPV6 Address:
   Controller IP Address: 10.64.91.65
             MAC Address: 00:1b:4f:03:52:18
```

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

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

                                 Type: g450

Slot   Module Type             Name                    DSP Type  FW/HW version
 V1:                                                    MP80      134  3
 V2:   MM712                   DCP MM
 V3:   MM711                   ANA MM
 V4:
 V5:
 V6:
 V7:
 V8:                                                 Max Survivable IP Ext: 8
 V9:   gateway-announcements  ANN VMM
```

IP telephones can be assigned a network region based on an IP address mapping. The network region can also associate the IP telephone to a location for location-based routing decisions. The following screen illustrates a subset of the IP network map configuration used to verify these Application Notes. If the IP address of a registering IP Telephone does not appear in the ip-network-map, the phone is assigned the network region of the "gatekeeper" (e.g., CLAN or PE) to which it registers. When the IP address of a registering IP telephone is in the ip-network-map, the phone is assigned the network region assigned by the form shown below. For example, the IP address 10.64.91.39 would be mapped to network region 1, based on the configuration in bold below. In production environments, different sites will typically be on different networks, and ranges of IP addresses assigned by the DHCP scope serving the site can be entered as one entry in the network map, to assign all telephones in a range to a specific network region.

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

                                                Subnet Network     Emergency
 IP Address                                     Bits   Region VLAN Location Ext
 --------------------------------------------- ------ ------ ---- -------------
 FROM: 10.64.91.30                              /      1      n
   TO: 10.64.91.49
```

The following screen shows IP Network Region 2 configuration. In the test environment, network region 2 is used to allow unique behaviors for the Verizon test environment. In this example, codec set 2 will be used for calls within region 2. The Avaya Interoperability Lab test environment uses the domain "avayalab.com" (i.e., for network region 1 including the region of the Processor Ethernet "procr"). Session Manager also uses this domain to determined routes for calls based on the domain information of the calls and for SIP phone registration. Avaya SBCE will adapt "avayalab.com" to "adevc.avaya.globalipcom.com" for the From, PAI and Diversion headers.

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

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

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

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

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

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

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

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

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

5.7. SIP Signaling Group

This section illustrates the configuration of the SIP Signaling Groups. Each signaling group has a **Group Type** of "**sip**", a **Near-end Node Name** of "**procr**", and a **Far-end Node Name** of "**SM**". In the example screens, the **Transport Method** for all signaling groups is "**tls**". The **Peer Detection Enabled** field is set to "**y**" and a peer Session Manager has been previously detected. The **Far-end Domain** is set to "**avayalab.com**" matching the configuration in place prior to adding the Verizon IP SIP Trunking configuration. The **Enable Layer 3 Test** field is enabled on each of the signaling groups to allow Communication Manager to maintain the signaling group using the SIP OPTIONS method. Fields that are not referenced in the text below can be left at default values, including **DTMF over IP** set to "**rtp-payload**", which corresponds to RFC 2833.

The following screen shows signaling group 1. Signaling group 1 will be used for processing PSTN calls to / from Verizon via Session Manager. The **Far-end Network Region** is configured to region 2. Port 5081 has been configured as both the **Near-end Listen Port** and **Far-end Listen Port**. Session Manager will be configured to direct calls arriving from the PSTN with Verizon DID numbers to a route policy that uses a SIP entity link to Communication Manager specifying port 5081. The use of different ports is one means to allow Communication Manager to distinguish different types of calls arriving from the same Session Manager. The **Initial IP-IP Direct Media?** is set to "**n**". Other parameters may be left at default values.

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

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

 Group Number: 1             Group Type: sip
  IMS Enabled? n       Transport Method: tls
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                   Far-end Node Name: SM
 Near-end Listen Port: 5081                  Far-end Listen Port: 5081
                                          Far-end Network Region: 2

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

The following screen shows signaling group 3, the signaling group to Session Manager that was in place prior to adding the Verizon Business IP Trunk configuration to the shared Avaya Solutions and Interoperability Test Lab configuration. This signaling group reflects configuration not specifically related to Verizon Business 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 "**SM**", the node name of the Session Manager. Unlike the signaling group used for the Verizon Business IP Trunk signaling, the **Far-end Network Region** is "**1**". The **Peer Detection Enabled** field is set to "**y**" and a peer Session Manager has been previously detected.

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

 Group Number: 3             Group Type: sip
  IMS Enabled? n       Transport Method: tls
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                   Far-end Node Name: SM
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          Far-end Network Region: 1

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

5.8. SIP Trunk Group

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

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

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

Group Number: 1                    Group Type: sip          CDR Reports: y
  Group Name: OUTSIDE CALL                COR: 1        TN: 1        TAC: *01
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: public-ntwrk         Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 1
                                                    Number of Members: 10
```

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

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

TRUNK PARAMETERS

     Unicode Name: auto
                                             Redirect On OPTIM Failure: 5000

            SCCAN? n                                Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 900

 Disconnect Supervision - In? y  Out? y


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

The following screen shows **Page 3** for trunk group 1. All parameters except those in bold are default values. The **Numbering Format** will use "**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.

```
change trunk-group 1                                            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 1. The bold fields have non-default values. Setting the **Network Call Redirection** flag to "**y**" enables advanced services associated with the use of the REFER message, while also implicitly enabling Communication Manager to signal "send-only" media conditions for calls placed on hold at the enterprise site. If neither REFER signaling nor "send-only" media signaling is required, this field may be left at the default "n" value. In the testing associated with these Application Notes, transfer testing using REFER was successfully completed with the **Network Call Redirection** flag set to "**y**", and transfer testing using INVITE was successfully completed with the **Network Call Redirection** flag set to "**n**". For redirected calls, Verizon supports the Diversion header, but not the History-Info header. Communication Manager can send the Diversion header by marking **Send Diversion Header** to "**y**". Alternatively, Communication can send the History-Info header by setting **Support Request History** to "**y**", and Session Manager can adapt the History-Info header to the Diversion header using the "VerizonAdapter". In the testing associated with these Application Notes, call redirection testing with Communication Manager sending History-Info and Session Manager adapting to Diversion Header was completed successfully.

Although not strictly necessary, the **Telephone Event Payload Type** has been set to "**101**" to match Verizon configuration. Verizon recommends that inbound calls to the enterprise result in a 183 with SDP rather than a 180 with SDP, and setting **Convert 180 to 183 for Early Media** to "**y**" for the trunk group handling inbound calls from Verizon produces this result.

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

                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? y
          Build Refer-To URI of REFER From Contact For NCR? n
                                     Send Diversion Header? n
                                   Support Request History? y
                              Telephone Event Payload Type: 101


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

          Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                                Request URI Contents: may-have-extra-digits
```

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

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

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

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

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

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                 Modify Tandem Calling Number: no
```

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

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

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

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


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

5.9. Route Pattern Directing Outbound Calls to Verizon

Route pattern 1 will be used for calls destined for the PSTN via the Verizon Business IP Trunk service. Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of "**0**" is the least restrictive level. If desired, one or more alternate Communication Manager trunks can be listed in the route pattern so that the Look-Ahead Routing (**LAR**) "next" setting can route-advance to attempt to complete the call using alternate trunks should there be no response or an error response from the far-end.

```
change route-pattern 1                                           Page   1 of   3
                    Pattern Number: 1      Pattern Name: To PSTN SIP Trk
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 1    0       1                                                     n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                                          none
 2: y y y y y n  n            rest                                          none
 3: y y y y y n  n            rest                                          none
 4: y y y y y n  n            rest                                          none
 5: y y y y y n  n            rest                                          none
 6: y y y y y n  n            rest                                          none
```

5.10. Route Pattern for Internal Calls via Session Manager

Route pattern 3 contains trunk group 3, the "private" tie trunk group to Session Manager. The **Numbering Format** "**lev0-pvt**" insures proper numbering format for internal local calls to Session Manager.

```
change route-pattern 3                                           Page   1 of   3
                    Pattern Number: 3      Pattern Name: ToSM Enterprise
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 3    0                                                             n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y y  n            bothept                            lev0-pvt   none
 2: y y y y y n  n            rest                                          none
 3: y y y y y n  n            rest                                          none
 4: y y y y y n  n            rest                                          none
 5: y y y y y n  n            rest                                          none
 6: y y y y y n  n            rest                                          none
```

5.11. Public Numbering

The ***change public-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 form for outbound calls, and incoming call handling treatment form for the inbound trunk group).

In the example abridged output below, a specific Communication Manager extension (x12001) is mapped to a DID number that is known to Verizon for this SIP Trunk connection (17329450231). As this applies to a SIP connection, the public numbering table will result in an E.164 formatted number (e.g., +17329450231). An adaptation in Session Manager will remove the “+1” from the number and present the 10 digit number format expected by Verizon (7329450231). See **Section 6.3**. In a real customer environment, normally the DID number may be comprised of the local extension plus a prefix. If this is true, then a single public numbering entry can be applied for a range of extensions. In the example below, all stations with a 5-digit extension beginning with 14 will send the calling party number as the **CPN Prefix** plus the extension number.

```
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: 16
 5  12002                   17329450232      11      Maximum Entries: 240
 5  12003                   17329450233      11
 5  14                      1732945          11   Note: If an entry applies to
                                                  a SIP connection to Avaya
                                                  Aura(R) Session Manager,
                                                  the resulting number must
                                                  be a complete E.164 number.

                                                  Communication Manager
                                                  automatically inserts
                                                  a '+' digit in this case.
```

5.12. ARS Routing For Outbound Calls

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

The following screen shows a specific ARS configuration as an example. If a user dials the ARS access code followed by 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       fnpa         n
```

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

```
list ars route-chosen 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        fnpa                 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 be used to perform digit conversion, and digit manipulation via the Communication Manager incoming call handling table may not be necessary. If the DID number sent by Verizon is unchanged by Session Manager, then the DID number can be mapped to an extension using the incoming call handling treatment of the receiving trunk group. As an example, the following screen illustrates a conversion of DID number 7329450285 to extension 14008. Both Session Manager digit conversion and Communication Manager incoming call handling treatment methods were tested successfully.

```
change inc-call-handling-trmt trunk-group 1                     Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    10 7329450285       all 14008
```

5.14. Avaya Aura® Communication Manager Stations

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

```
change station 12002                                            Page   1 of   5
                                     STATION

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

5.15. 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 12002. Use the command ***change off-pbx-telephone station mapping x*** where ***x*** is Communication Manager station (e.g., 12002).

- **Station Extension** – This field will automatically populate
- **Application** – Enter "**EC500**"
- **Dial Prefix** – Enter a prefix (e.g., 1) if required by the routing configuration
- **Phone Number** – Enter the phone that will also be called (e.g., **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 12002                  Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number     Trunk       Config  Dual
 Extension                 Prefix                      Selection   Set     Mode
  12002            EC500    1     -     3035387022       ars         1
```

5.16. Saving Communication Manager Configuration Changes

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

```
save translation all
```

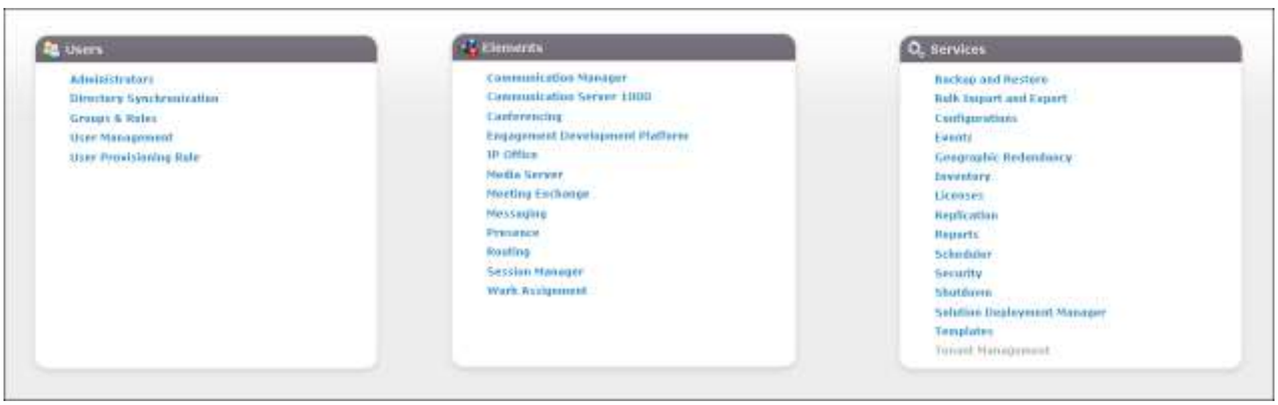
6. Configure Avaya Aura® Session Manager Release 7.0

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

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

Session Manager is managed via System Manager. Using a web browser, access "https://<ip-addr of System Manager>/SMGR". In the **Log On** screen, enter appropriate **User ID** and **Password** and press the **Log On** button (not shown).

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



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

Introduction to Network Routing Policy

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

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

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

Step 2: Create "Locations"

Step 3: Create "Adaptations"

Step 4: Create "SIP Entities"

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

Step 5: Create the "Entity Links"

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

Step 6: Create "Time Ranges"

- Align with the tariff information received from the Service Providers

Step 7: Create "Routing Policies"

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

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

Step 8: Create "Dial Patterns"

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

Step 9: Create "Regular Expressions"

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

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

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

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

"Dial Pattern driven approach to define Routing Policies"

That means (with regard to steps listed above):

Step 7: "Routing Polices" are defined

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

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

6.1. Domains

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

The following screen shows a list of configured SIP domains. The Session Manager used in the verification of these Application Notes was shared among other Avaya interoperability test efforts. The domain "**avayalab.com**" was used for communication with Avaya SIP Telephones and other

Avaya systems and applications. The domain "**avayalab.com**" is not known to the Verizon production service.

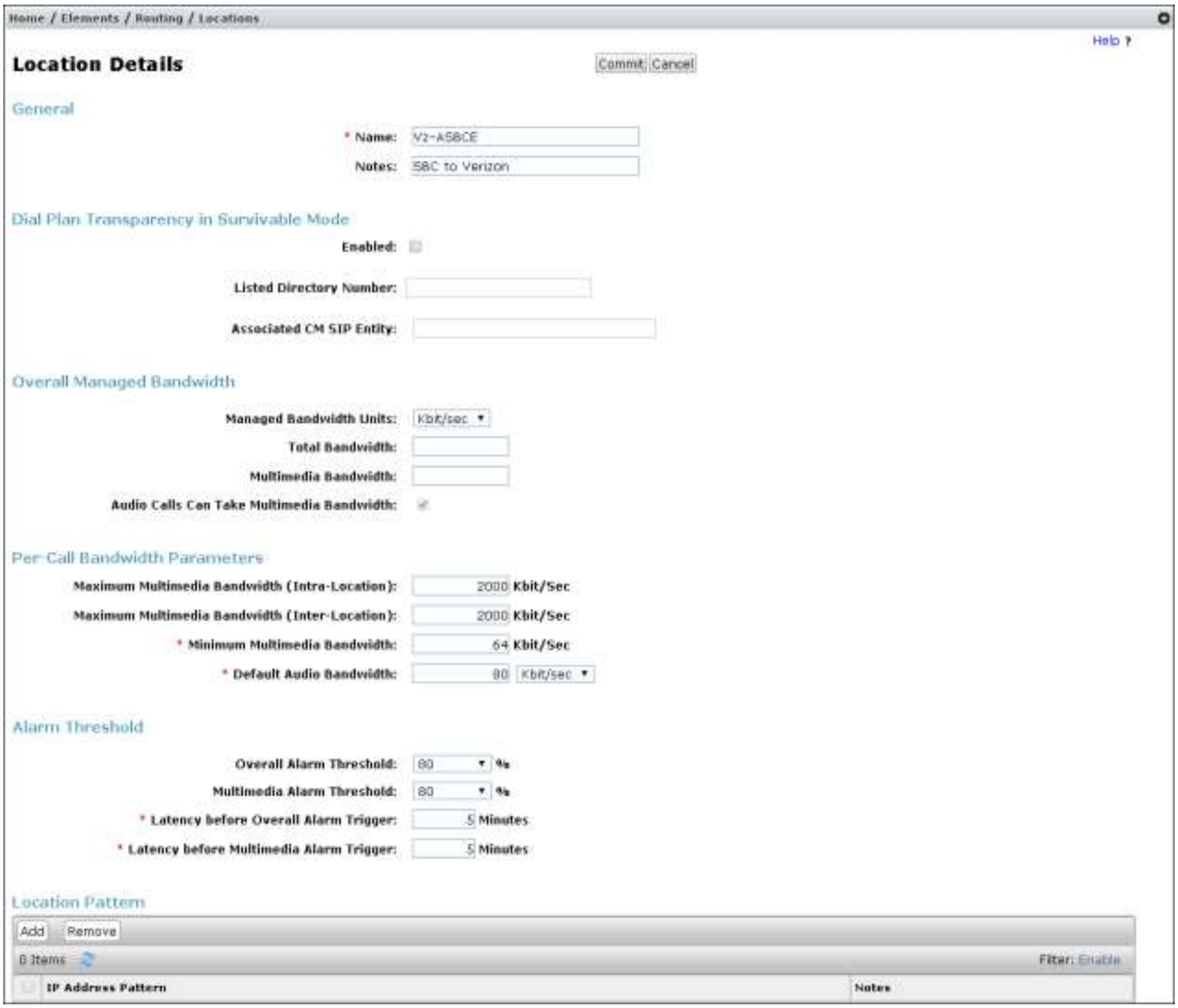
The domain "**adevc.avaya.globalipcom.com**" is the domain known to Verizon as the enterprise SIP domain. For example, for calls from the enterprise site to Verizon, this domain can appear in the From and P-Asserted-Identity headers in the INVITE message sent to Verizon.

6.2. Locations

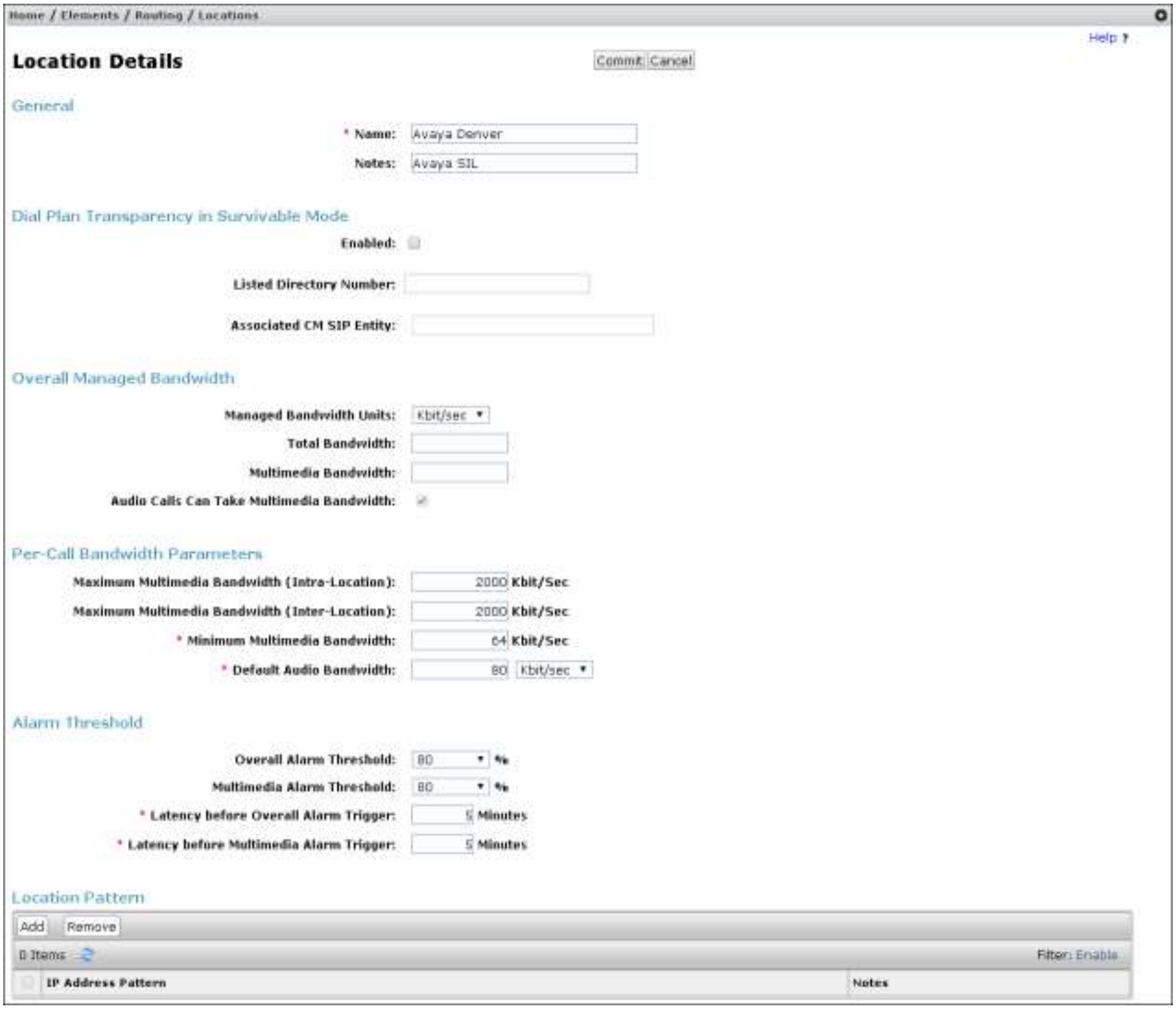
To view or change locations, select **Routing → Locations**. The following screen shows an abridged list of configured locations. Click on the checkbox corresponding to the name of a location and **Edit** to edit an existing location, or the **New** button to add a location. Click the **Commit** button (not shown) after changes are completed. Assigning unique locations can allow Session Manager to perform location-based routing, bandwidth management, and call admission control.

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

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



The following screen shows the location details for the location named "**Avaya Denver**", corresponding to SIP entities within the enterprise. Later, the location with name "**Avaya Denver**" will be assigned to the corresponding Communication Manager SIP Entity. In the sample configuration, other location parameters (not shown) retained the default values.

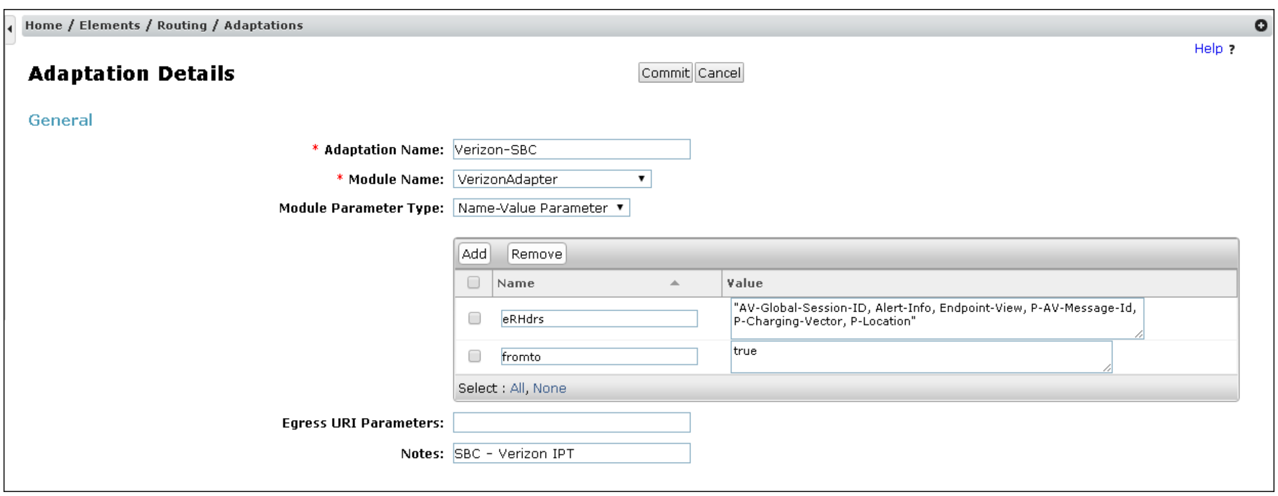


6.3. Adaptations

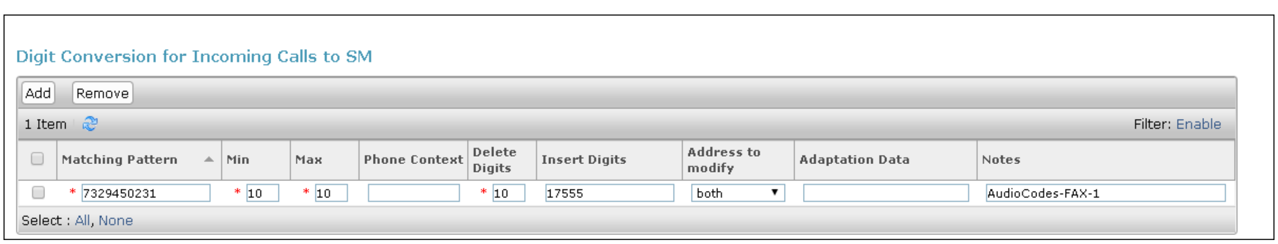
To view or change adaptations, select **Routing → Adaptations**. Click on the checkbox corresponding to the name of an adaptation and **Edit** to edit an existing adaptation, or the **New** button to add an adaptation. Click the **Commit** button after changes are completed (not shown).

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

This adaptation uses the "**VerizonAdapter**" module and specifies the "**eRHdrs**" and "**fromto**" parameters. The "**eRHdrs**" parameter will remove the specified headers during adaptation in the egress direction, i.e., towards Verizon. In the sample configuration, proprietary headers were removed to reduce the size of the SIP message to prevent packet fragmentation by adding the following **Value,** including the quotes, **"AV-Global-Sesison-ID, Alert-Info, Endpoint-View, P-AV-Message-Id, P-Charging-vector, P-Location"**. The "**fromto**" parameter, with a **Value** of "**true**", adapts the From and To headers for digit conversion along with the Request-Line and PAI headers.



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



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

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

Digit Conversion for Outgoing Calls from SM

Add Remove

3 Items Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
	* +	* 12	* 12		* 2		origination		E.164 to 10 digit Calling Party Number
	* +13035551234	* 12	* 12		* 2		origination	7329450821	Unscreened ANI - Diversion header
	* 17555	* 5	* 5		* 5	7329450231	origination		AudioCodes-FAX-1

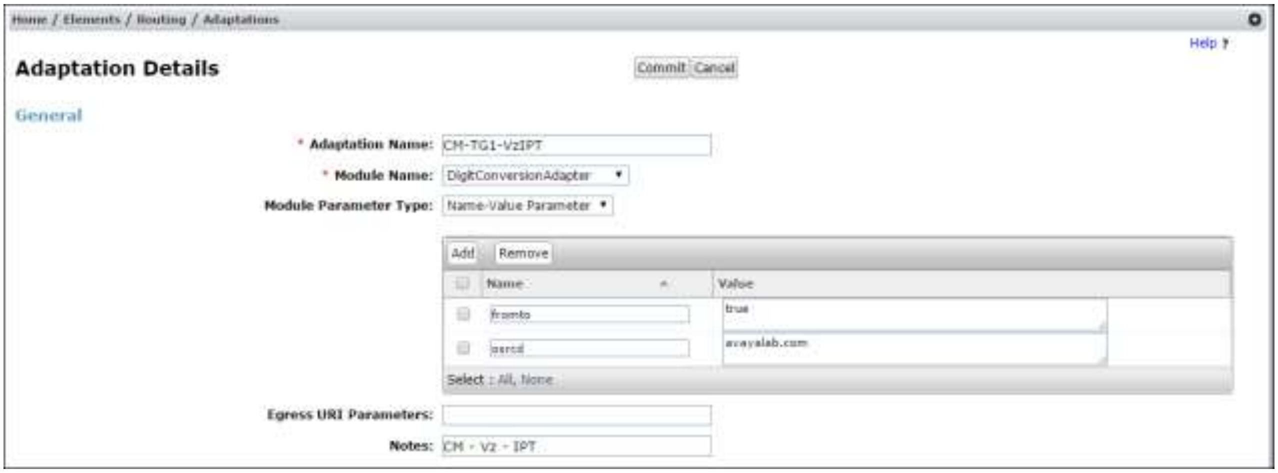
Select : All, None

The above screen also shows E.164 formatted numbers sent by Communication Manager's public-unknown numbering table (**Section 5.11**), **Matching Pattern** "+", will be converted to 10 digit numbers expected by Verizon by deleting the first two digits (i.e., +1). It also shows the addition of an extension number used by AudioCodes gateway (17555) being converted to a 10 digit DID number assigned by Verizon.

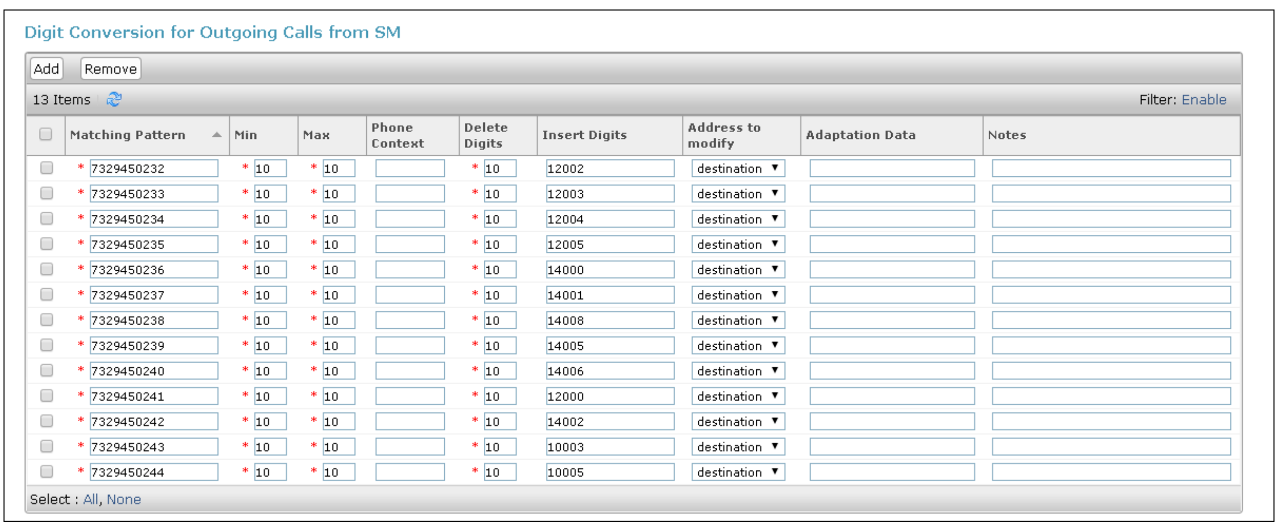
The adapter named "**CM-TG1-VzIPT**" shown in the following screen will later be assigned to the SIP Entity linking Session Manager to Communication Manager for calls involving Verizon Business IP Trunk service. This adaptation uses the "**DigitConversionAdapter**" and specifies the following parameters:

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

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



Scrolling down, the following screen shows a portion of the "**CM-TG1-VzIPT**" adapter that can be used to convert 10 digit DID numbers assigned by Verizon to the extension number used on Communication Manager. Since this adapter will be assigned to the SIP Entity sending calls to Communication Manager from the PSTN, the settings for **Digit Conversion for Outgoing Calls from SM** correspond to incoming calls from the PSTN to Communication Manager. In the example shown below, if a user on the PSTN dials 732-945-0232, Session Manager will convert the number to 12002 before sending the SIP INVITE to Communication Manager. In this case, digit conversion is done after the routing decision has been made based upon the user part of the SIP URI. As such, it would not be necessary to use the incoming call handling table of the receiving Communication Manager trunk group to convert the DID number to its corresponding extension.



Digit Conversion for Outgoing Calls from SM

Add Remove

13 Items Filter: Enable

Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
* 7329450232	* 10	* 10		* 10	12002	destination		
* 7329450233	* 10	* 10		* 10	12003	destination		
* 7329450234	* 10	* 10		* 10	12004	destination		
* 7329450235	* 10	* 10		* 10	12005	destination		
* 7329450236	* 10	* 10		* 10	14000	destination		
* 7329450237	* 10	* 10		* 10	14001	destination		
* 7329450238	* 10	* 10		* 10	14008	destination		
* 7329450239	* 10	* 10		* 10	14005	destination		
* 7329450240	* 10	* 10		* 10	14006	destination		
* 7329450241	* 10	* 10		* 10	12000	destination		
* 7329450242	* 10	* 10		* 10	14002	destination		
* 7329450243	* 10	* 10		* 10	10003	destination		
* 7329450244	* 10	* 10		* 10	10005	destination		

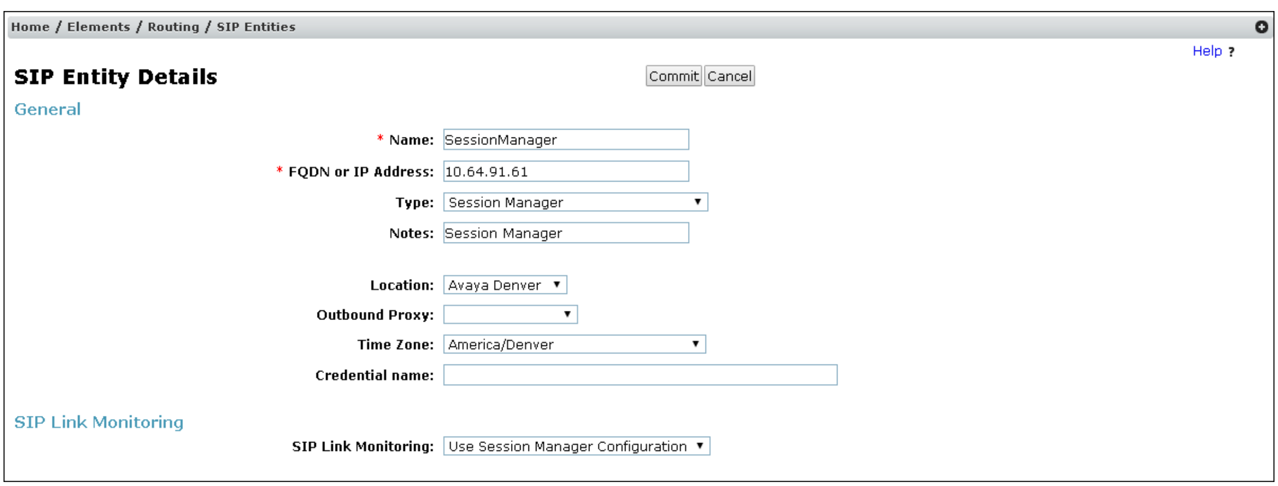
Select : All, None

6.4. SIP Entities

To view or change SIP entities, select **Routing → SIP Entities**. Click the checkbox corresponding to the name of an entity and **Edit** to edit an existing entity, or the **New** button to add an entity. Click the **Commit** button after changes are completed.

The following screen shows the list of configured SIP entities in the shared test environment.

The following screen shows the upper portion of the **SIP Entity Details** corresponding to "**SessionManager**". The **FQDN or IP Address** field for "**SessionManager**" is the Session Manager Security Module IP Address (10.64.91.61), which is used for SIP signaling with other networked SIP entities. The **Type** for this SIP entity is "**Session Manager**". Select an appropriate location for the Session Manager from the **Location** drop-down menu. In the test environment, the Session Manager used location "**Avaya Denver**". The default **SIP Link Monitoring** parameters may be used. Unless changed elsewhere, links from other SIP entities to this instance of Session Manager will use the default SIP Link Monitoring timers, configurable at the Session Manager level. If desired, these timers may be customized for each entity.



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

Entity Links

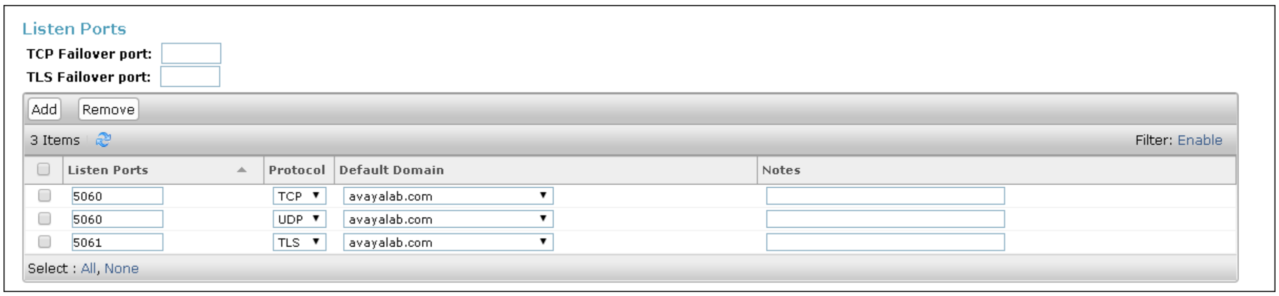
Add | Remove

6 Items | Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
* SM to AAM	SessionManager	TCP	* 5060	Aura Messaging	* 5060	trusted	
* SM to CM TG1	SessionManager	TLS	* 5081	CM-TG1	* 5081	trusted	
* SM to CM TG2	SessionManager	TLS	* 5071	CM-TG2	* 5071	trusted	
* SM to CM TG3	SessionManager	TLS	* 5061	CM-TG3	* 5061	trusted	
* SM to Vz SBC1	SessionManager	TCP	* 5060	Vz-SBC-1	* 5060	trusted	

Select : All, None | Page 1 of 2

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



Listen Ports

TCP Failover port:

TLS Failover port:

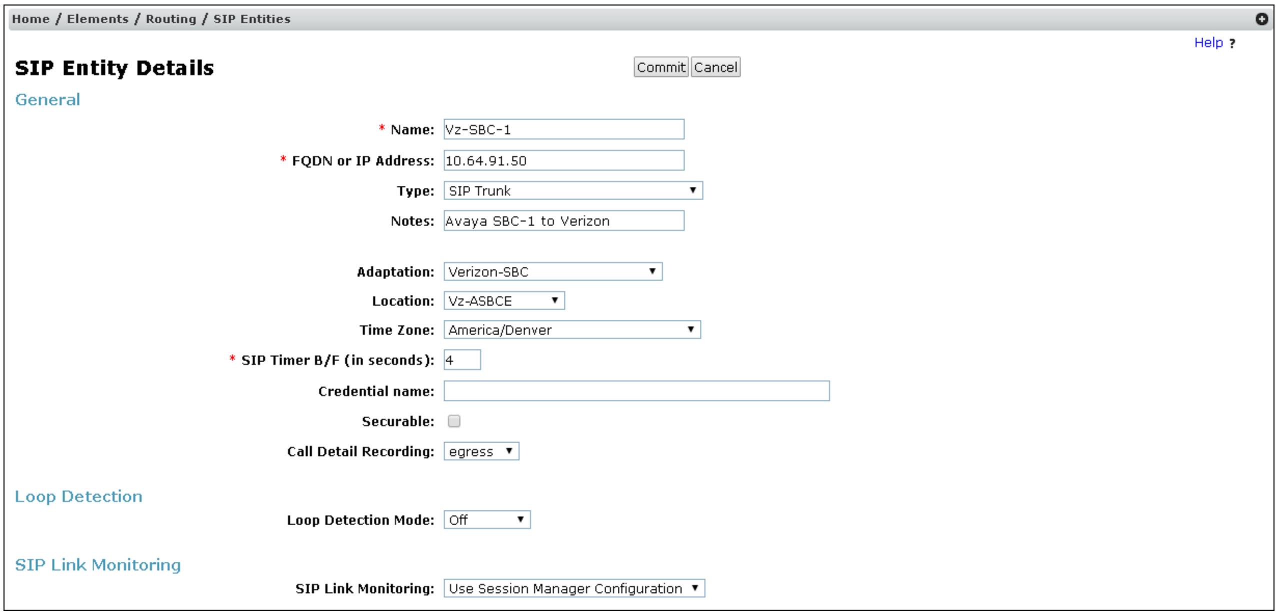
Add | Remove

3 Items | Filter: Enable

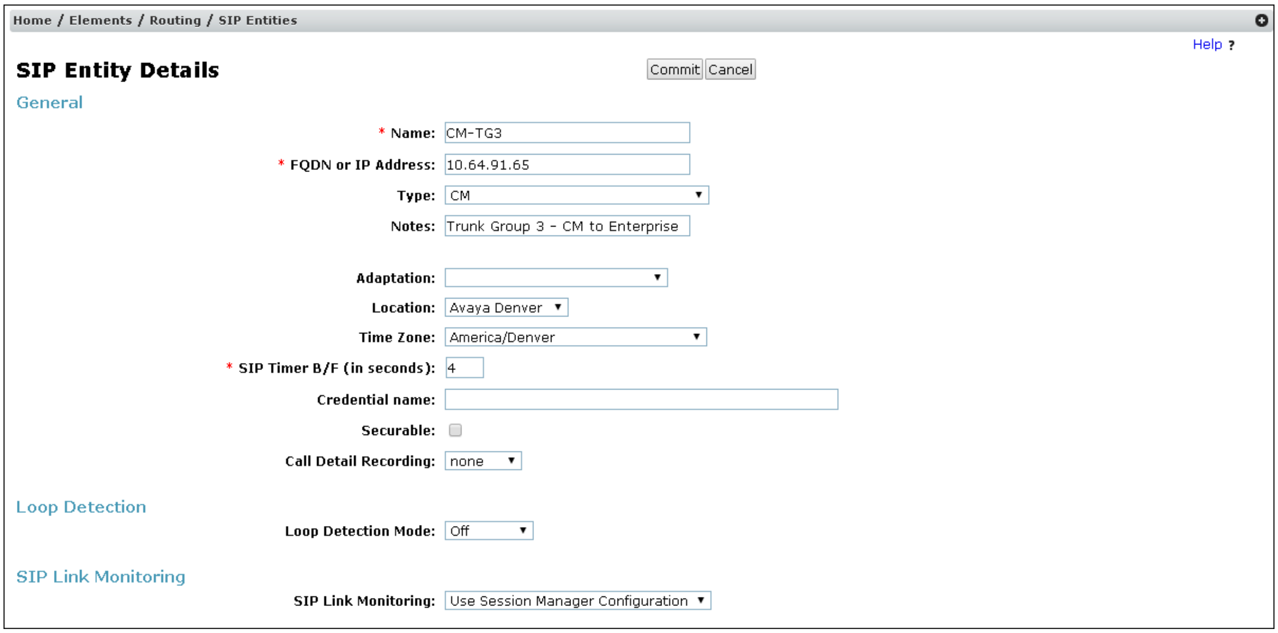
Listen Ports	Protocol	Default Domain	Notes
5060	TCP	avayalab.com	
5060	UDP	avayalab.com	
5061	TLS	avayalab.com	

Select : All, None

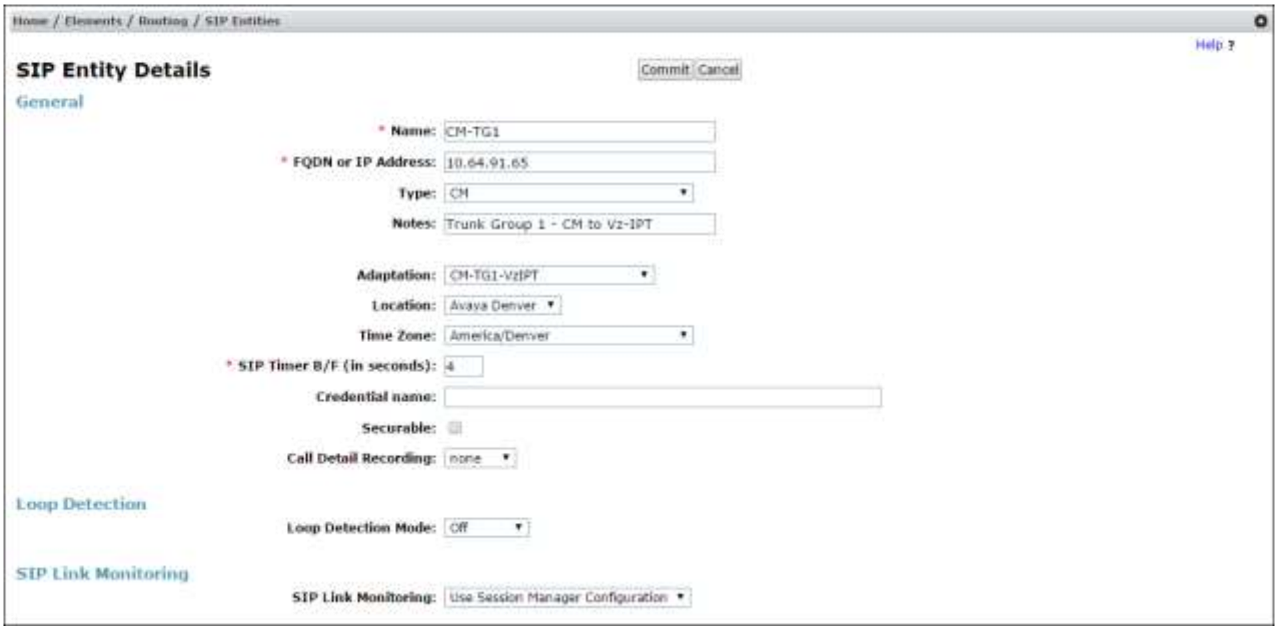
The following screen shows the upper portion of the **SIP Entity Details** corresponding to "**Vz-SBC-1**". The **FQDN or IP Address** field is configured with the Avaya SBCE inside IP Address (**10.64.91.50**). "**SIP Trunk**" is selected from the **Type** drop-down menu for Avaya SBCE SIP Entities. This Avaya SBCE has been assigned to **Location** "**Vz-ASBCE**", and the "**Verizon-SBC**" adapter is applied. Other parameters (not shown) retain default values.



The following screen shows a portion of the **SIP Entity Details** corresponding to a Communication Manager SIP Entity named "**CM-TG3**" This is the SIP Entity that was already in place in the Avaya Interoperability Test Lab environment, prior to adding the Verizon Business IP Trunk configuration. The **FQDN or IP Address** field contains the IP Address of the "processor Ethernet" (**10.64.91.65**). "**CM**" is selected from the **Type** drop-down menu and "**Avaya Denver**" is selected for the **Location**.



The following screen shows the **SIP Entity Details** for an entity named “**CM-TG1**”. This entity uses the same **FQDN or IP Address** (**10.64.91.65**) as the prior entity with name “**CM-TG3**”; both correspond to Communication Manager Processor Ethernet IP Address. Later, a unique port, 5081, will be used for the Entity Link to “**CM-TG1**”. Using a different port is one approach that will allow Communication Manager to distinguish traffic originally from Verizon Business IP Trunk from other SIP traffic arriving from the same IP Address of the Session Manager, such as SIP traffic associated with SIP Telephones or other SIP-integrated applications. “**CM**” is selected from the **Type** drop-down menu, and “**CM-TG1-VzIPT**” is selected for the **Adaptation**. “**Avaya Denver**” is selected for the **Location**.



6.5. Entity Links

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

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


Home / Elements / Routing / Entity Links

Help ?

Entity Links

New | More Actions

6 Items | Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy	Deny New Service	Notes
SM to AAM	SessionManager	TCP	5060	Aura Messaging	☐	5060	trusted	☐	
SM to CM TG1	SessionManager	TLS	5081	CM-TG1	☐	5081	trusted	☐	
SM to CM TG2	SessionManager	TLS	5071	CM-TG2	☐	5071	trusted	☐	
SM to CM TG3	SessionManager	TLS	5061	CM-TG3	☐	5061	trusted	☐	
SM to Vz SBC1	SessionManager	TCP	5060	Vz-SBC-1	☐	5060	trusted	☐	

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

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

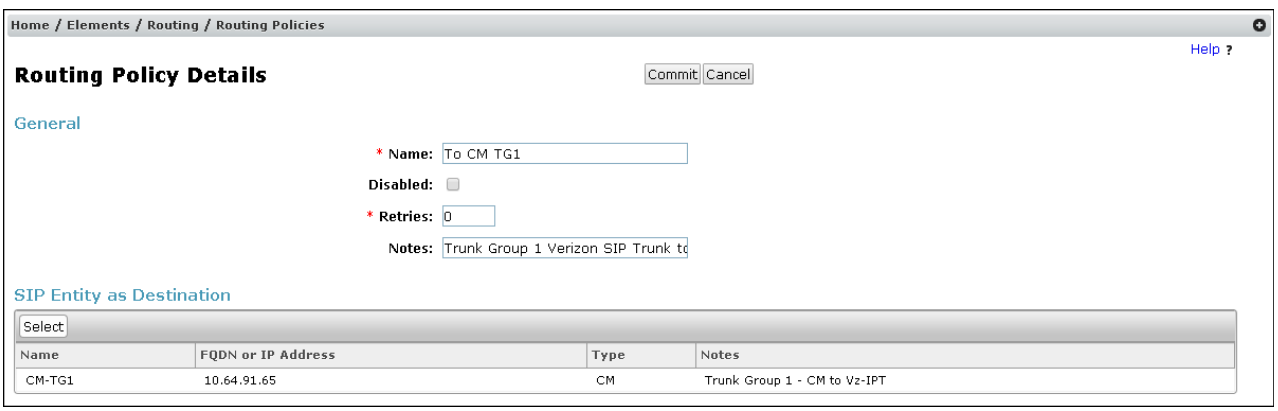
6.6. Time Ranges

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

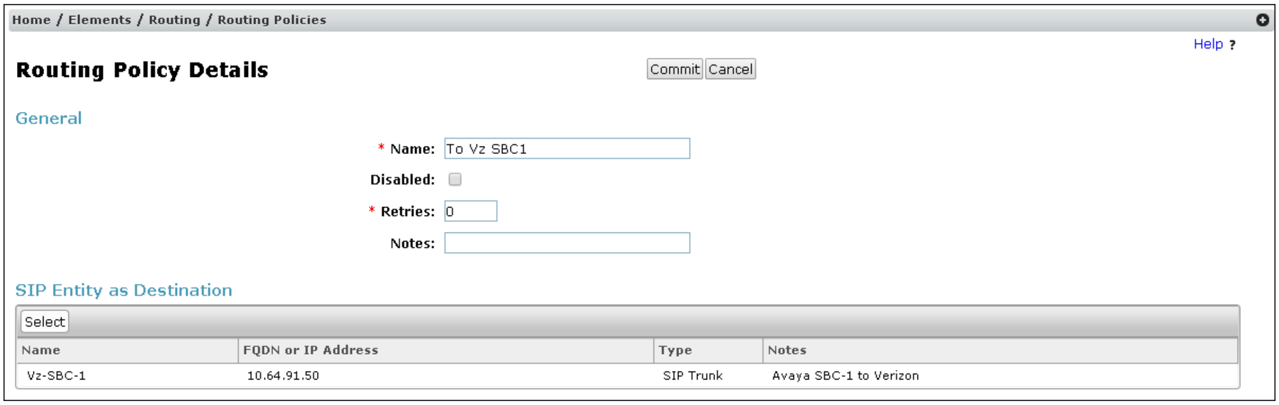
6.7. Routing Policies

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

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



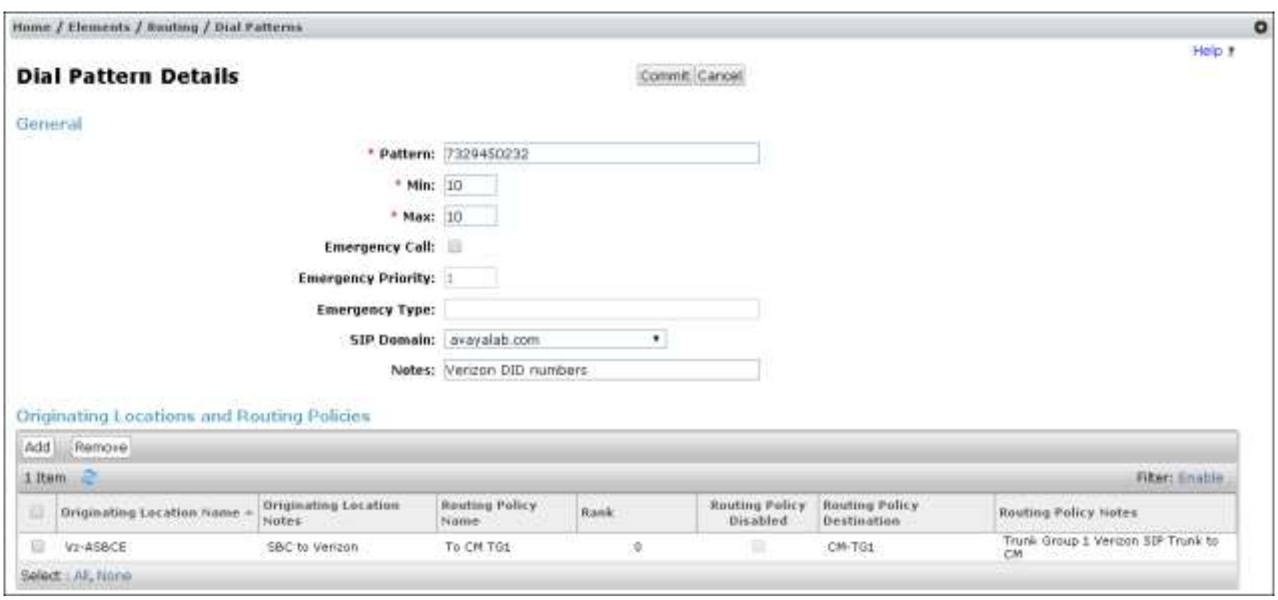
The following screen shows the **Routing Policy Details** for the policy named “**To Vz SBC1**” associated with outgoing calls from Communication Manager to the PSTN via Verizon through Avaya SBCE. Observe the **SIP Entity as Destination** as the entity named “**Vz-SBC-1**” that was created in **Section 6.4**.



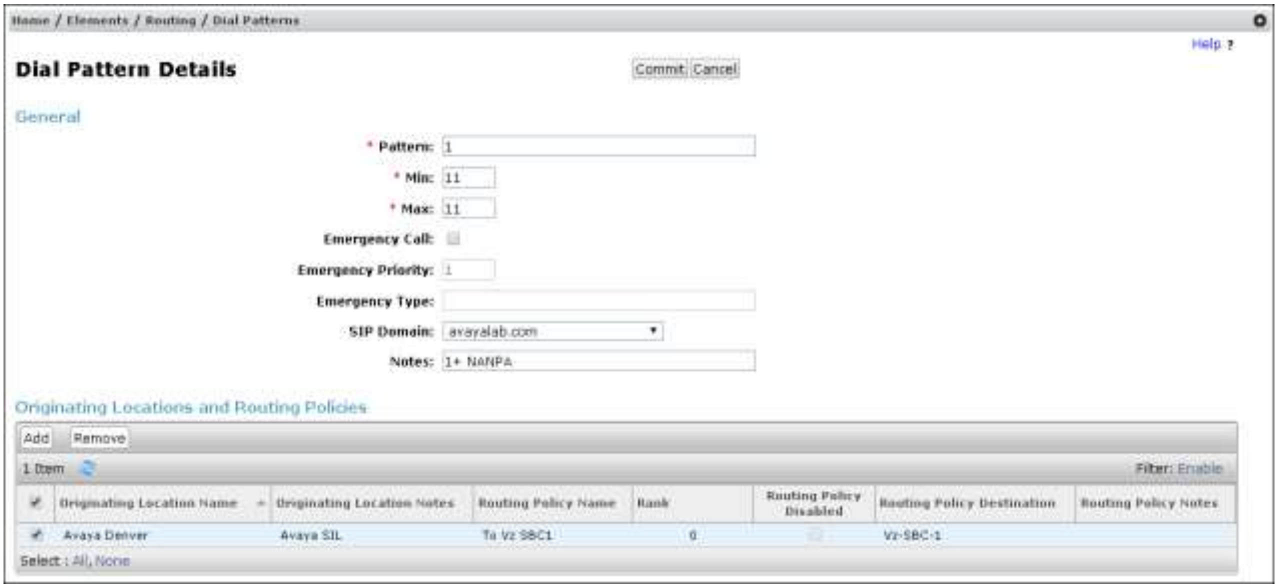
6.8. Dial Patterns

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

The following screen illustrates an example dial pattern used to verify inbound PSTN calls to the enterprise. When a user on the PSTN dials a number assigned to the Verizon Business IP Trunk service, such as 732-945-0232, Verizon delivers the number to the enterprise, and the Avaya SBCE sends the call to Session Manager. The pattern below matches on 732-945-0232 specifically. Dial patterns can alternatively match on ranges of number (e.g., a block of DID numbers). Under **Originating Locations and Routing Policies**, the routing policy named “**To CM TG1**” is chosen when the call originates from **Originating Location Name** “**Vz-ASBCE**”. This sends the call to Communication Manager using port 5081 as described previously.



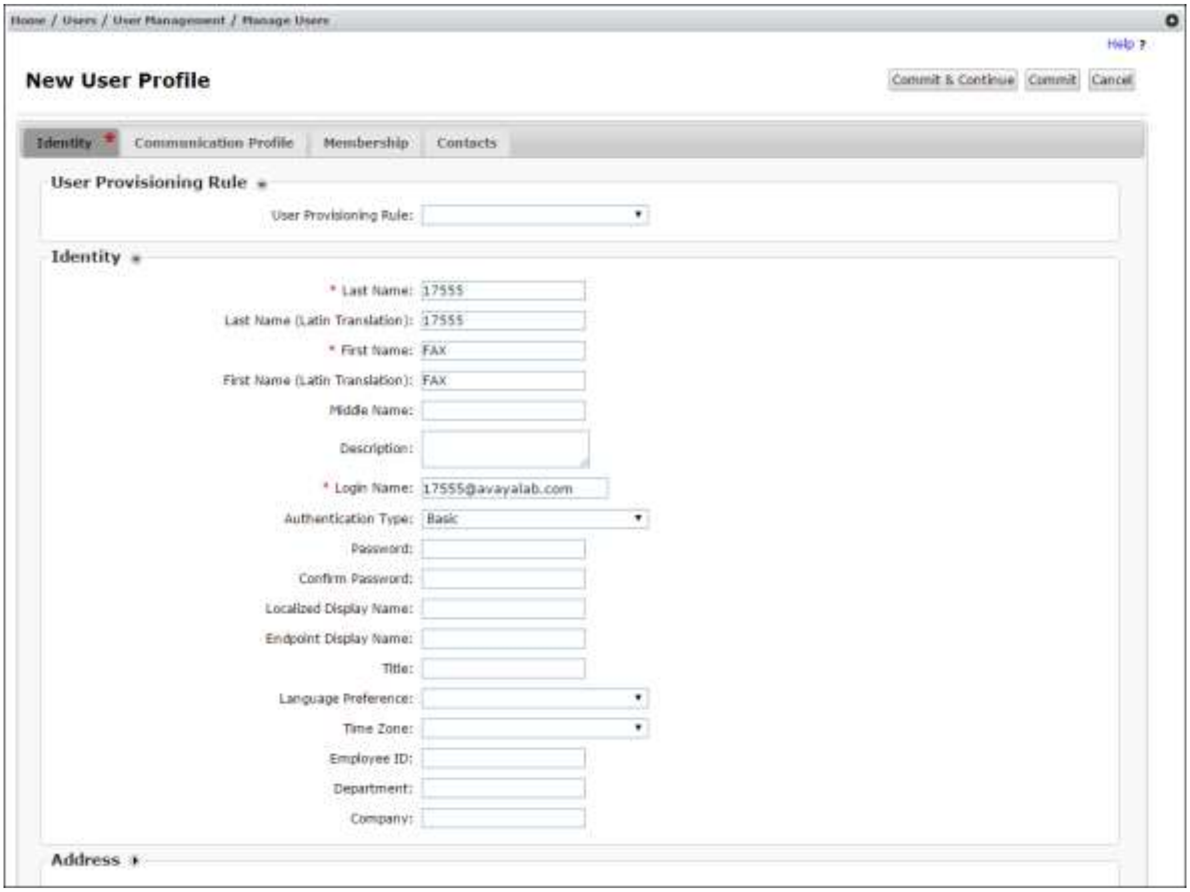
The following screen illustrates an example dial pattern used to verify outbound calls from the enterprise to the PSTN. When a Communication Manager user dials a PSTN number such as 9-1303-555-1234, Communication Manager sends the call to Session Manager as "13035551234". Session Manager will match the dial pattern shown below and send the call to the Avaya SBCE via the **Routing Policy Name** "**To Vz SBC1**".



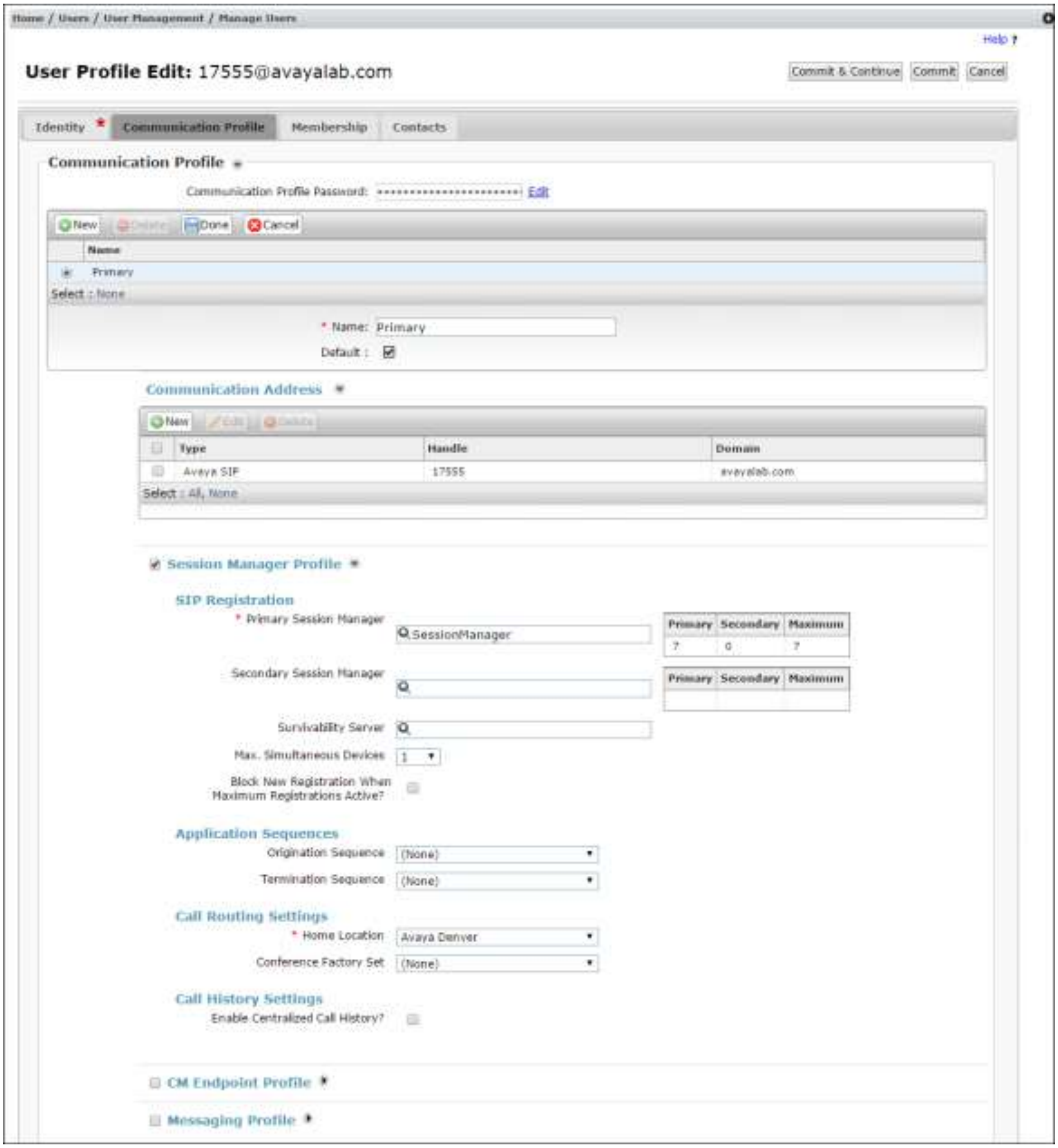
6.9. Fax Users

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

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



The following screen shows the **Communication Profile** tab of the sample user. The **Communication Profile Password** is the password used by the SIP device to register with Session Manager, and should match the password set on the AudioCodes MP-114 in **Section 8.2**. The **Application Sequences** section is set to "**(None)**", and the **CM Endpoint Profile** is unchecked. This allows for fax calls to be sent to the AudioCodes MP-114, without involving Communication Manager in the call setup. As stated in **Section 2.2**, Verizon requires fax calls to start off with G.711 as the first codec choice, and if all other voice calls prefer G.729 as the first codec, a separate Communication Manager trunk group dedicated for fax calls using an ip-codec-set with G.711 as the first codec choice would be required. Having the **Application Sequence** section set to "**(None)**" prevents the need for a separate fax dedicated trunk group on Communication Manager. As a result, fewer SIP re-Invites messages are sent during the beginning of a fax call, and voice calls to and from Communication Manager can use other preferred codecs. However, any functionality that would normally be controlled by Communication Manager, such as codec negotiation, calling restrictions, dial patterns, etc., will be controlled by the AudioCodes device, and therefore will need to be configured directly on the AudioCodes device. See **Section 8** and **Section 12.3** for information on AudioCodes MP-114 configuration.

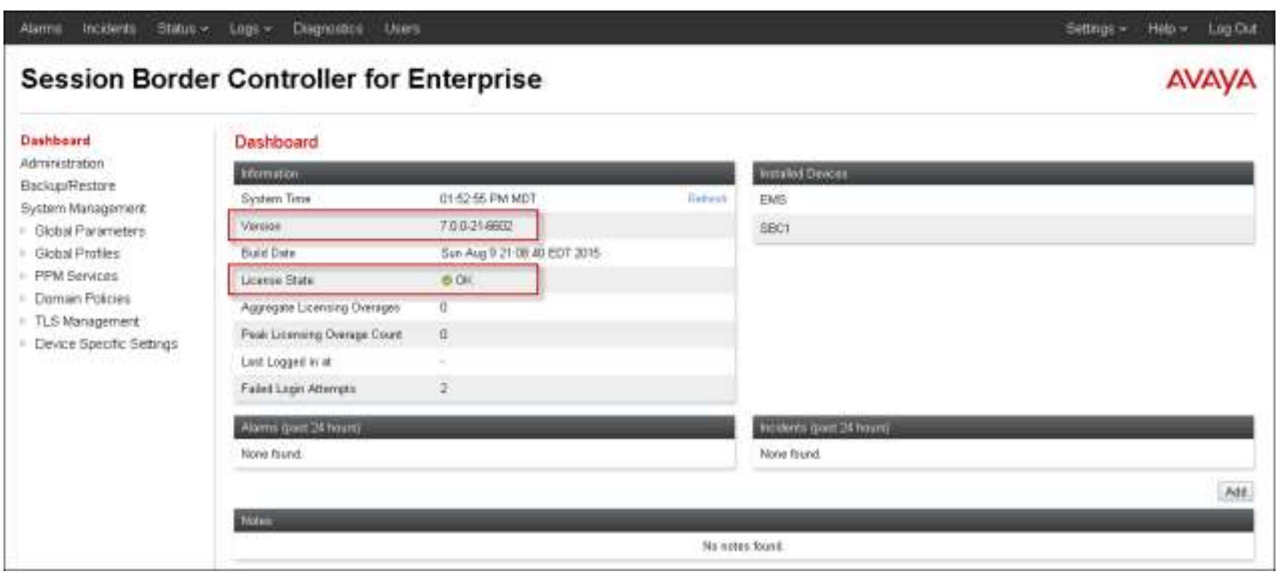

Home / Users / User Management / Manage Users
Help ?
User Profile Edit: 17555@avayalab.com
Commit & Continue
Commit
Cancel
Identity
Communication Profile
Membership
Contacts
Communication Profile
Communication Profile Password:
Edit
New
Done
Cancel
Name
Primary
Select : None
* Name:
Primary
Default :
Communication Address
New
Type
Handle
Domain
Avaya SIP
17555
avayalab.com
Select : All, None
Session Manager Profile
SIP Registration
* Primary Session Manager
SessionManager
Primary
Secondary
Maximum
7
0
7
Secondary Session Manager
Survivability Server
Max. Simultaneous Devices
1
Block New Registration When Maximum Registrations Active?
Application Sequences
Origination Sequence
(None)
Termination Sequence
Call Routing Settings
* Home Location
Avaya Denver
Conference Factory Set
Call History Settings
Enable Centralized Call History?
CM Endpoint Profile
Messaging Profile

7. Configure Avaya Session Border Controller for Enterprise Release 7.0

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

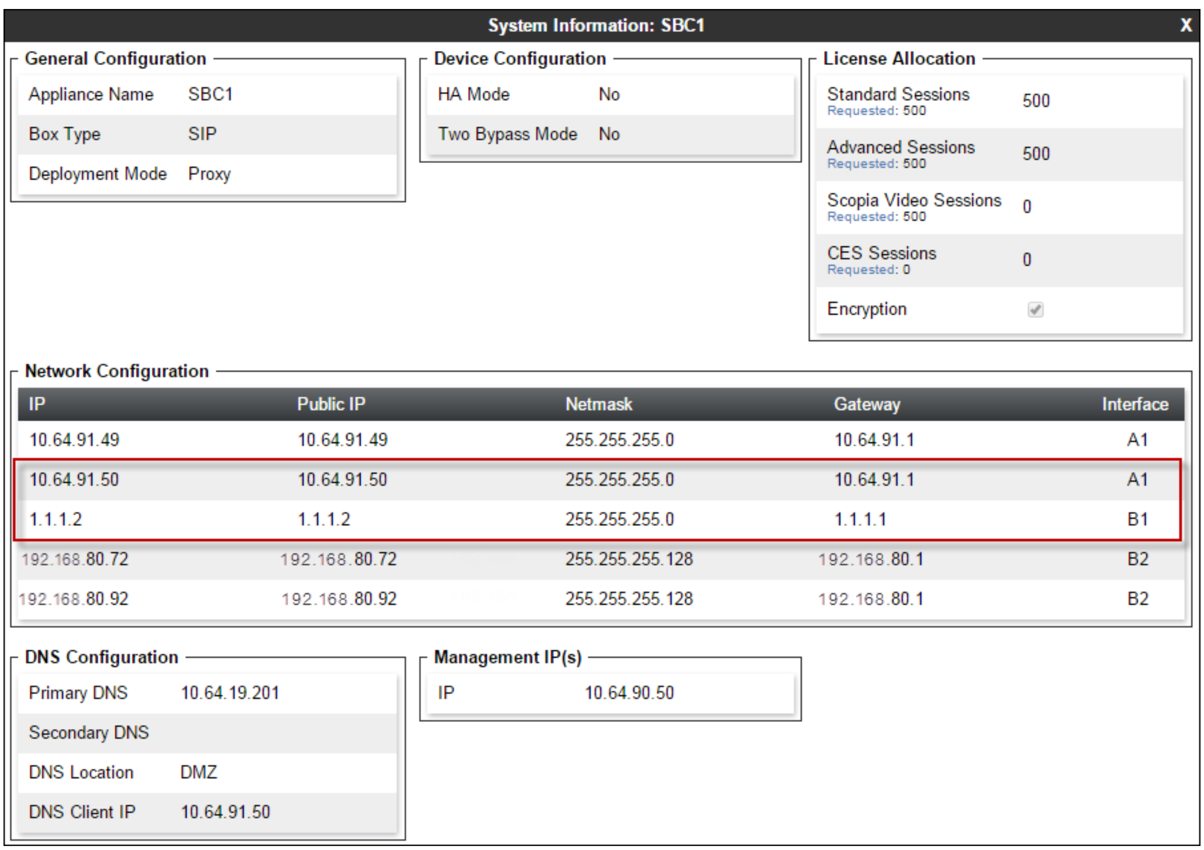
In the sample configuration, the management IP is 10.64.90.50. Access the web management interface by entering https://<ip-address> where <ip-address> is the management IP address assigned during installation. Log in with the appropriate credentials. Click **Log In**.

The main page of the Avaya SBCE will appear. Note that the installed software version is displayed. Verify that the **License State** is **"OK**". The SBCE will only operate for a short time without a valid license. Contact your Avaya representative to obtain a license



To view system information that was configured during installation, navigate to **System Management**. A list of installed devices is shown in the right pane. In the case of the sample configuration, a single device named "**SBC1**" is shown. To view the configuration of this device, click **View** as highlighted below.

The **System Information** screen shows the **Network Settings**, **DNS Configuration**, and **Management IP** information provided during installation and corresponds to **Figure 1**. The highlighted **A1** and **B1** IP addresses are the ones relevant to the configuration of the SIP trunk to Verizon. Other IP addresses assigned to these interfaces and interface **B2** on the screen below are used to support remote workers and are not the focus of these Application Notes. Note that the **Management IP** must be on a separate subnet from the IP interfaces designated for SIP traffic.



7.1. Network Management

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

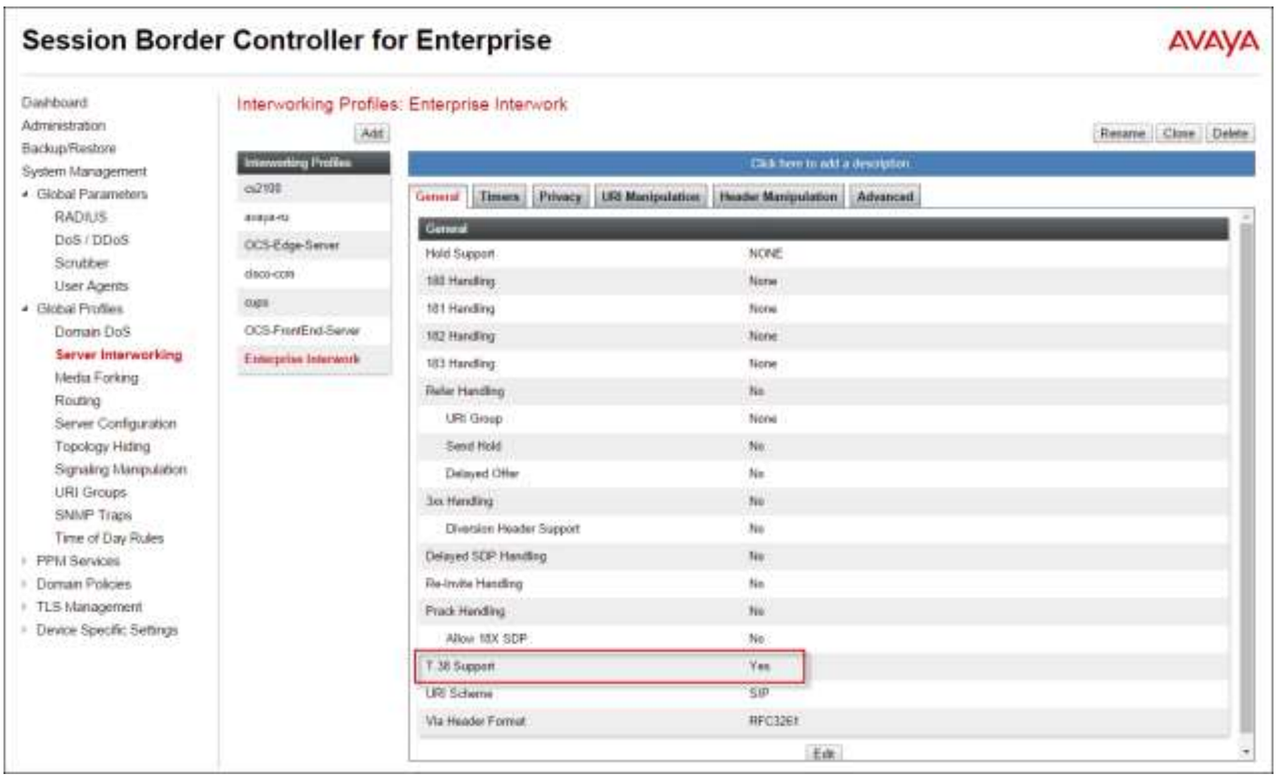
The following screen shows interface **A1** and **B1** are **Enabled**. To enable an interface click the corresponding **Toggle State** button.

7.2. Server Interworking Profile

Server Interworking allows users to configure and manage various SIP call server-specific capabilities such as call hold and T.38 faxing.

In the sample configuration, a single server interworking profile was created to define the connection to Session Manager. The Session Manager server interworking profile was cloned from the default **avaya-ru** profile. To clone a server interworking profile for, navigate to **Global Profiles → Server Interworking**, select the **avayu-ru** profile and click the **Clone** button. Enter a **Clone Name** and click **Finish** to continue.

The following screen shows the "**Enterprise Interwork**" profile used in the sample configuration, with **T.38 Support** set to "**Yes**". To modify the profile, scroll down to the bottom of the screen and click **Edit**. Select the **T.38 Support** parameter and then click **Next** and then **Finish** (not shown). Default values can be used for all other fields.



7.3. Signaling Manipulation

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

The sample script show below is used to remove the "**gsid**" and "**epv**" parameters Session Manager places in the Contact header. These parameters contain unnecessary information for Verizon, including the internal domain. Removing these parameters helps to mask the internal topology of the enterprise and reduces the size of the SIP packet sent to Verizon. The Endpoint-View header and other proprietary headers are removed using an adaptation as illustrated in **Section 6.3**.

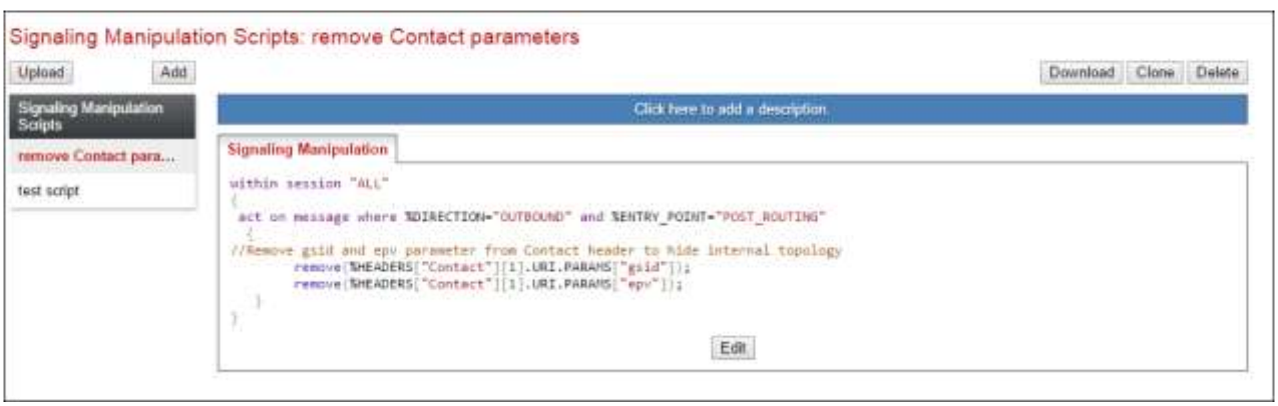
To create a new Signaling Manipulation, navigate to **Global Profiles → Signaling Manipulation** and click on **Add**. A new blank SigMa Editor window will pop up.

The following screen illustrates the "**remove Contact parameters**" script.

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

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

The following screen shows the finished Signaling Manipulation Script "**remove Contact parameters**" used during compliance testing. This script will later be applied to the Verizon Server Configuration in **Section 7.4.2**.



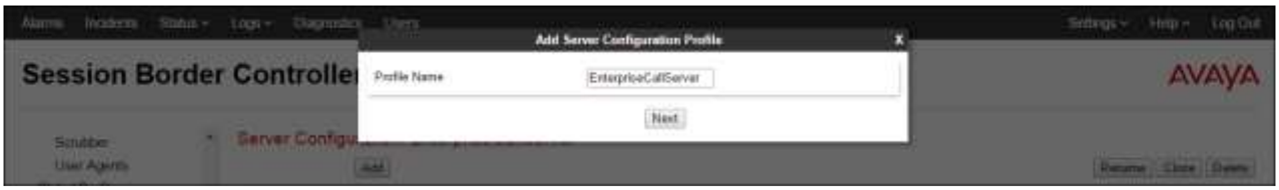
7.4. Server Configuration

The Server Configuration contains parameters to configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters, DoS security statistics, and trusted domains.

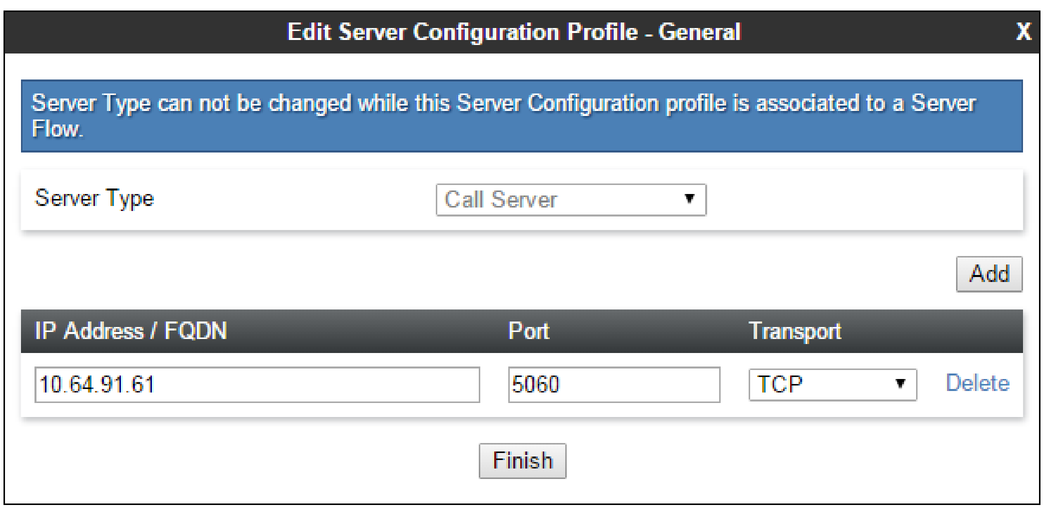
In the sample configuration, separate Server Configurations were created for Session Manager and Verizon Business IP Trunk service.

7.4.1 Server Configuration – Session Manager

To add a Server Configuration Profile for Session Manager, navigate to **Global Profiles → Server Configuration** and click **Add**. Enter a descriptive name for the new profile and click **Next**.

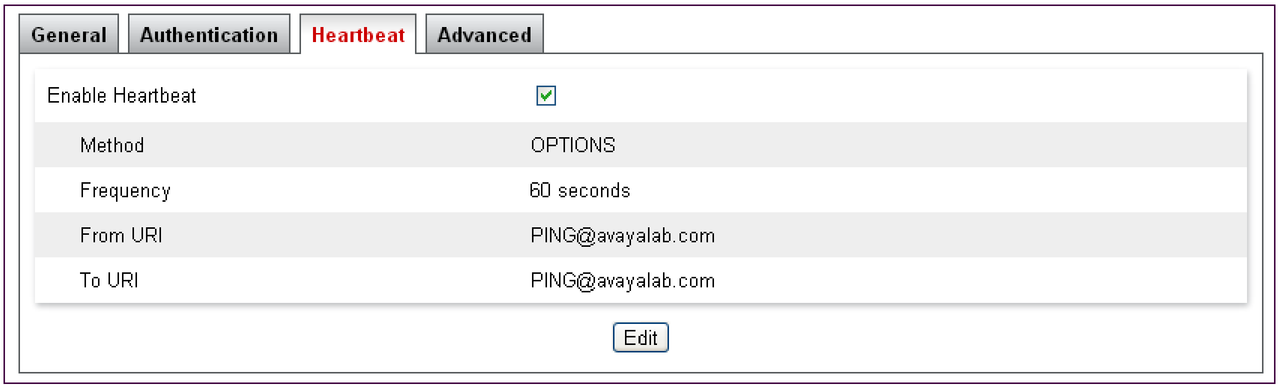


The following screens illustrate the Server Configuration for the Profile name "**EnterpriseCallServer**". In the **General** parameters, the **Server Type** is set to "**Call Server**". In the **IP Address / FQDN** field, the IP Address of the Session Manager SIP signaling interface is entered. In the sample configuration this IP Address is "**10.64.91.61**". Under **Port**, "**5060**" is entered, and the **Transport** parameter is set to "**TCP**". This configuration corresponds with the Session Manager entity link configuration for the entity link to the Avaya SBCE created in **Section 6.4**. If adding the profile, click **Next** (not shown) to proceed. If editing an existing profile, click **Finish**.

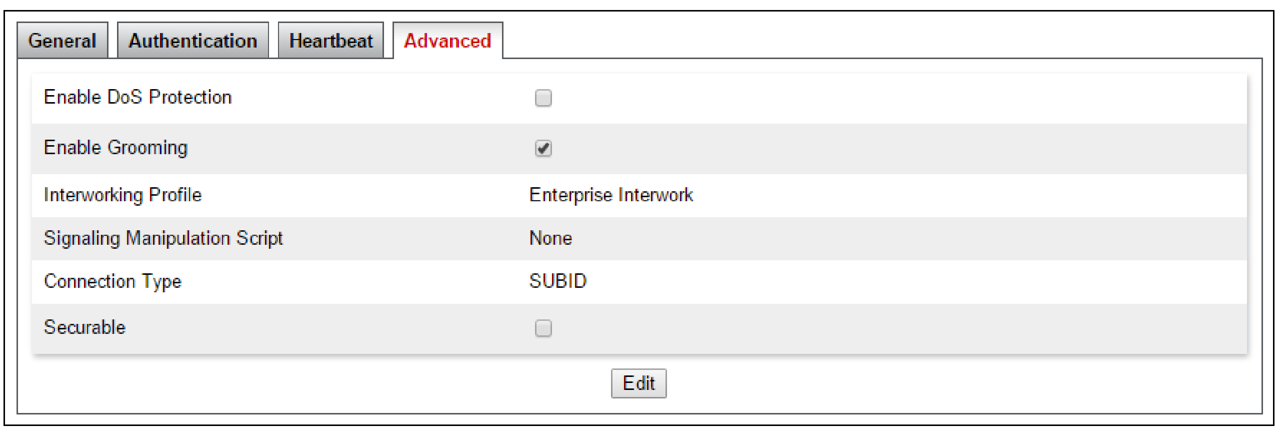


If adding the profile, click **Next** to accept default parameters for the **Authentication** tab (not shown), and advance to the Heartbeat area. If editing an existing profile, select the **Heartbeat** tab and click **Edit** (not shown).

Avaya SBCE can be configured to source “heartbeats” in the form of SIP OPTIONS. In the sample configuration, with one Session Manager, this configuration is optional. If Avaya SBCE-sourced OPTIONS messages are desired, check the **Enable Heartbeat** box. Select “**OPTIONS**” from the **Method** drop-down menu. Select the desired frequency that the Avaya SBCE will source OPTIONS to this server. The **From URI** and **To URI** may be filled in to configure easily identifiable URIs to appear in SIP OPTIONS sourced by the Avaya SBCE towards Session Manager. If adding a new profile, click **Next** (not shown). If editing an existing profile, click **Finish** (not shown).



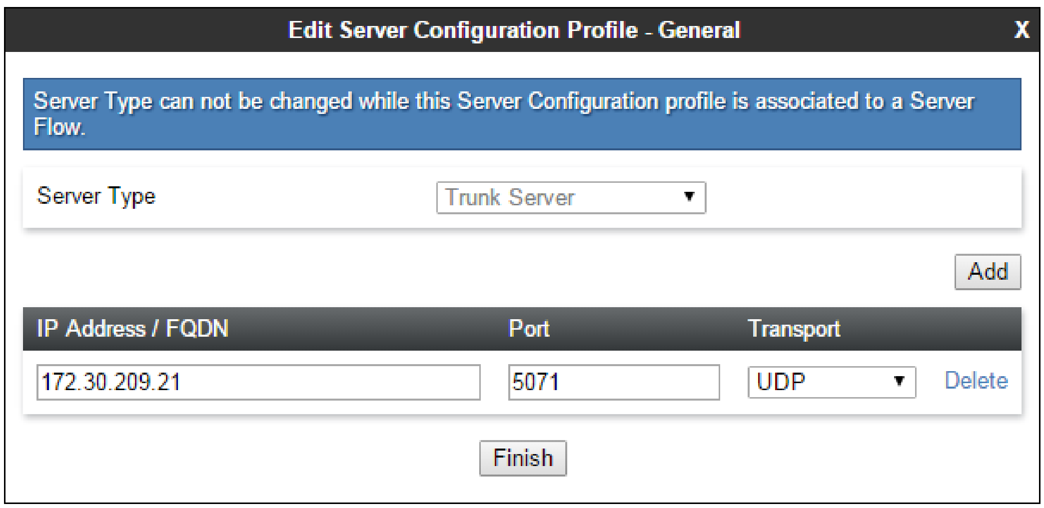
If adding a profile, click **Next** to continue to the **Advanced** settings (not shown). If editing an existing profile, select the **Advanced** tab and **Edit**. In the resultant screen, select **Enable Grooming** to allow the same TCP connection to be used for all SIP messages from this device. Select the **Interworking Profile** “**Enterprise Interwork**” created previously in **Section 7.2**. Click **Finish** (not shown).



7.4.2 Server Configuration - Verizon Business IP Trunk

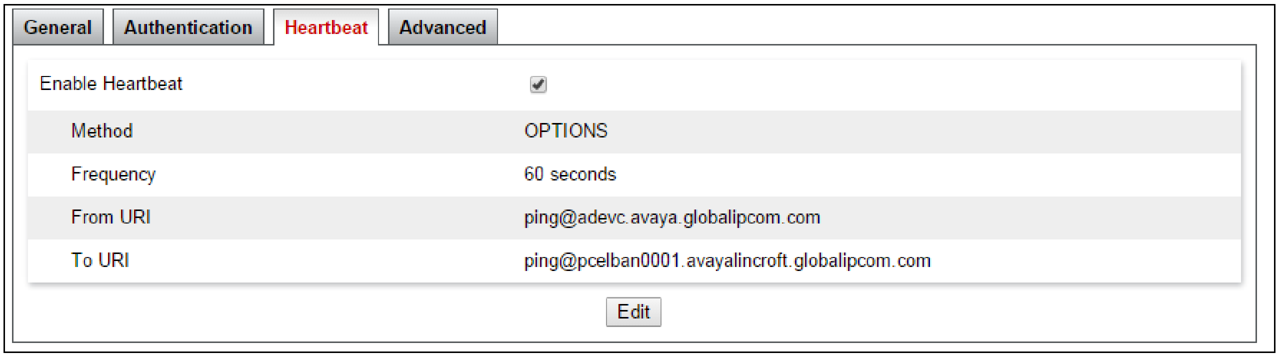
To add a Server Configuration Profile for Verizon, navigate to **Global Profiles → Server Configuration** and click **Add**. Enter a descriptive name for the new profile and click **Next**.

The following screens illustrate the Server Configuration for the Profile name "**Verizon Server**". In the **General** parameters, the **Server Type** is set to "**Trunk Server**". In the **IP Address / FQDN** field, the Verizon-provided IP address is entered. This is "**172.30.209.21**". Under **Port**, "**5071**" is entered, and the **Transport** parameter is set to "**UDP**". If adding the profile, click **Next** (not shown) to proceed. If editing an existing profile, click **Finish**.

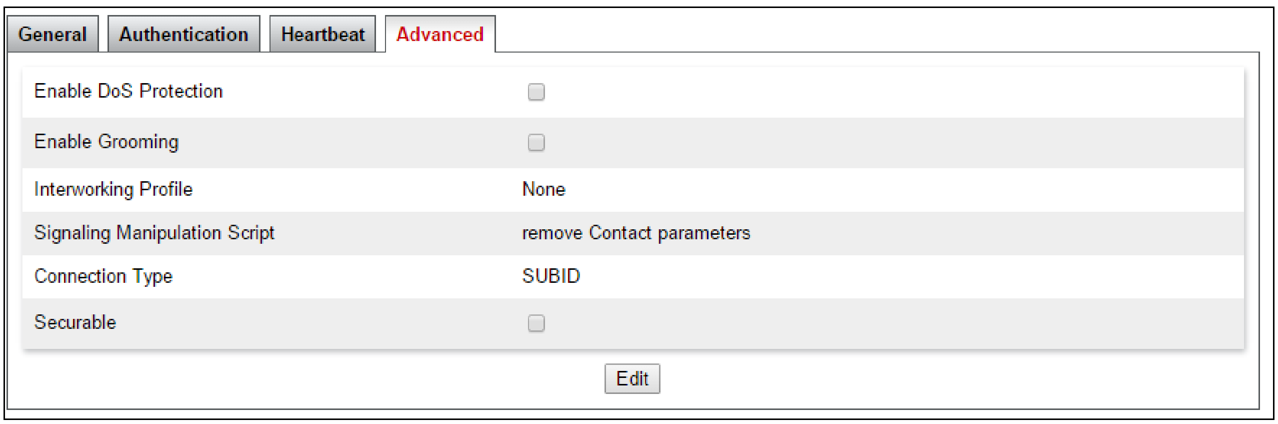


Default values can be used on the **Authentication** tab (not shown), click **Next** (not shown) to proceed to the **Heartbeats** tab. The ASBCE can be configured to source "heartbeats" in the form of SIP OPTIONS towards Verizon. This configuration is optional. Independent of whether the ASBCE is configured to source SIP OPTIONS towards Verizon, Verizon will receive OPTIONS from the enterprise site as a result of the SIP Entity Monitoring configured for Session Manager. When Session Manager sends SIP OPTIONS to the inside private IP Address of the Avaya SBCE, the Avaya SBCE will send SIP OPTIONS to Verizon. When Verizon responds, the Avaya SBCE will pass the response to Session Manager.

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



On the **Advanced** tab, **Enable Grooming** is not used for UDP connections and left unchecked. The Interworking Profile is left at its default setting of "**None**". This will prevent Avaya SBCE from inserting "Supported: replaces" in the SIP message toward Session Manager. See **Section 2.2** for additional information. Select the **Signaling Manipulation Script** created in **Section 7.3** titled "**remove Contact parameters**". Click **Finish** (not shown).

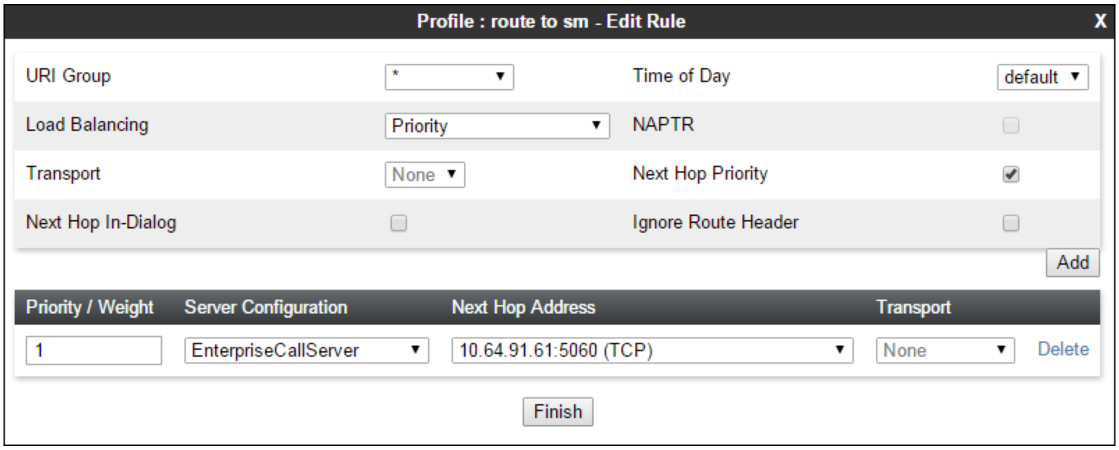


7.5. Routing Profile

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

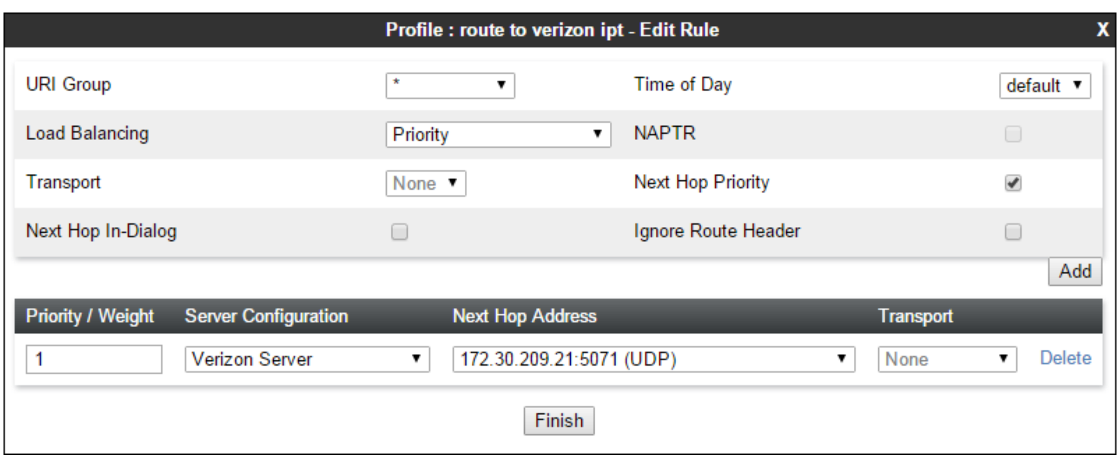
Create a Routing Profile for Session Manager and Verizon Business IP Trunk service. To add a routing profile, navigate to **Global Profiles → Routing** and select **Add**. Enter a **Profile Name** and click **Next** to continue.

The following screen shows the Routing Profile "**route to sm**" created in the sample configuration. The parameters in the top portion of the profile are left at their default settings. The **Priority / Weight** parameter is set to "**1**", and the Session Manager **Server Configuration**, created in **Section 7.4.1**, is selected from the drop-down menu. The **Next Hop Address** is automatically selected with the values from the Server Configuration, and **Transport** becomes greyed out. Click **Finish**.



Similarly add a Routing Profile to Verizon Business IP Trunk.

The following screen shows the Routing Profile "**route to verizon ipt**" created in the sample configuration. The parameters in the top portion of the profile are left at their default settings. The **Priority / Weight** parameter is set to "**1**", and the Verizon **Server Configuration**, created in **Section 7.4.2**, is selected from the drop-down menu. The **Next Hop Address** is automatically selected with the values from the Server Configuration, and **Transport** becomes greyed out. Click **Finish**.

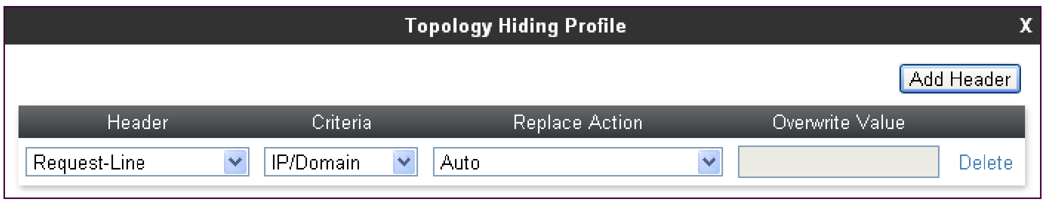


7.6. Topology Hiding Profile

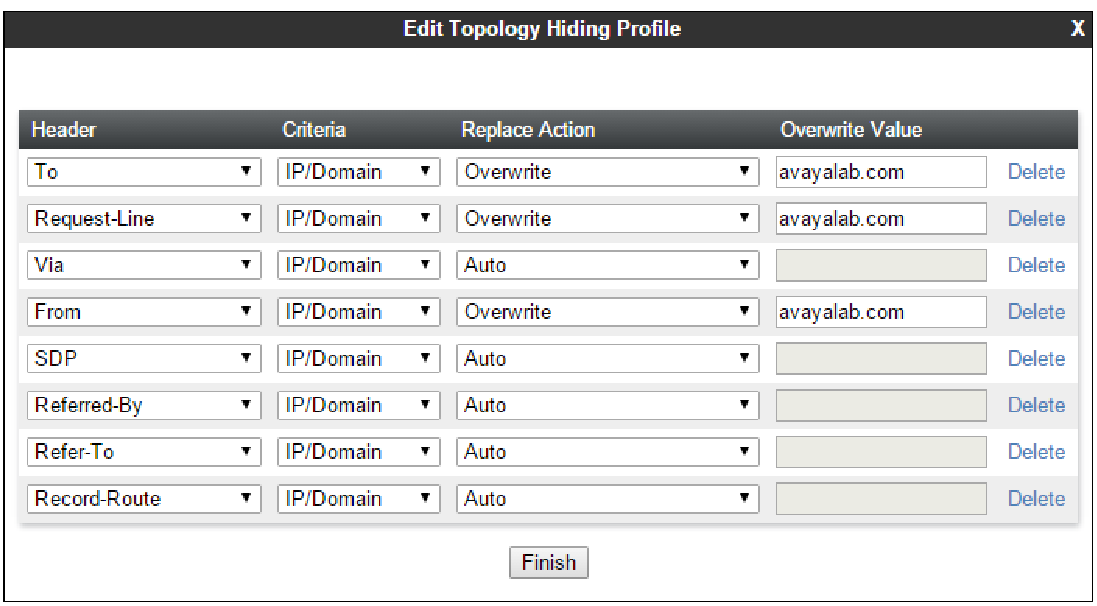
The Topology Hiding profile manages how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the integrity of the network. It hides the topology of the enterprise network from external networks.

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

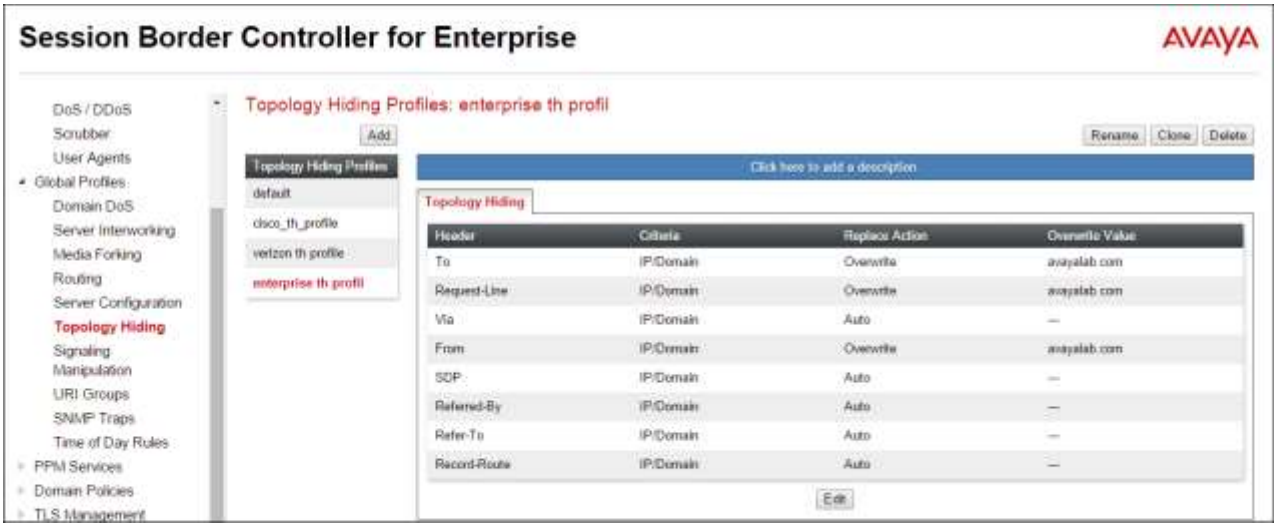
In the resultant screen, click the **Add Header** button in the upper right multiple times to reveal additional headers.



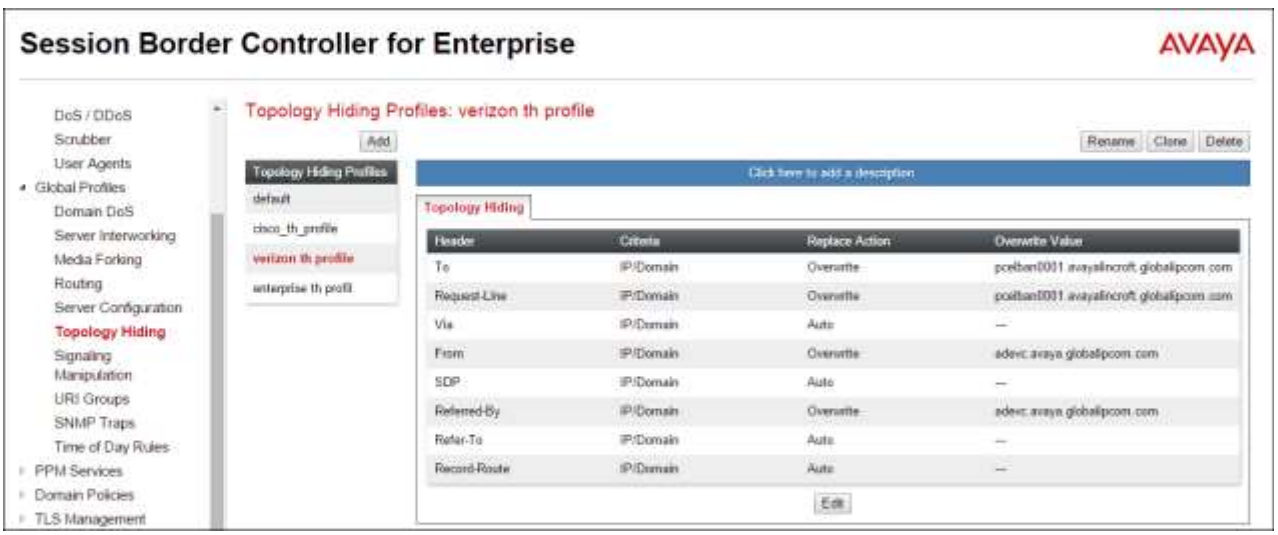
In the **Replace Action** column an action of "**Auto**" will replace the header field with the IP address of the Avaya SBCE interface and the "**Overwrite**" will use the value in the **Overwrite Value**. In the example shown, this profile will later be applied in the direction of the Session Manager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



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



Similarly, create a Topology Hiding profile for Verizon. Overwrite the headers as shown below with the FQDNs known by Verizon. The following screen shows Topology Hiding profile "**Verizon th profile**" created for Verizon. This profile will later be applied to the Server Flow for Verizon.



7.7. Application Rule

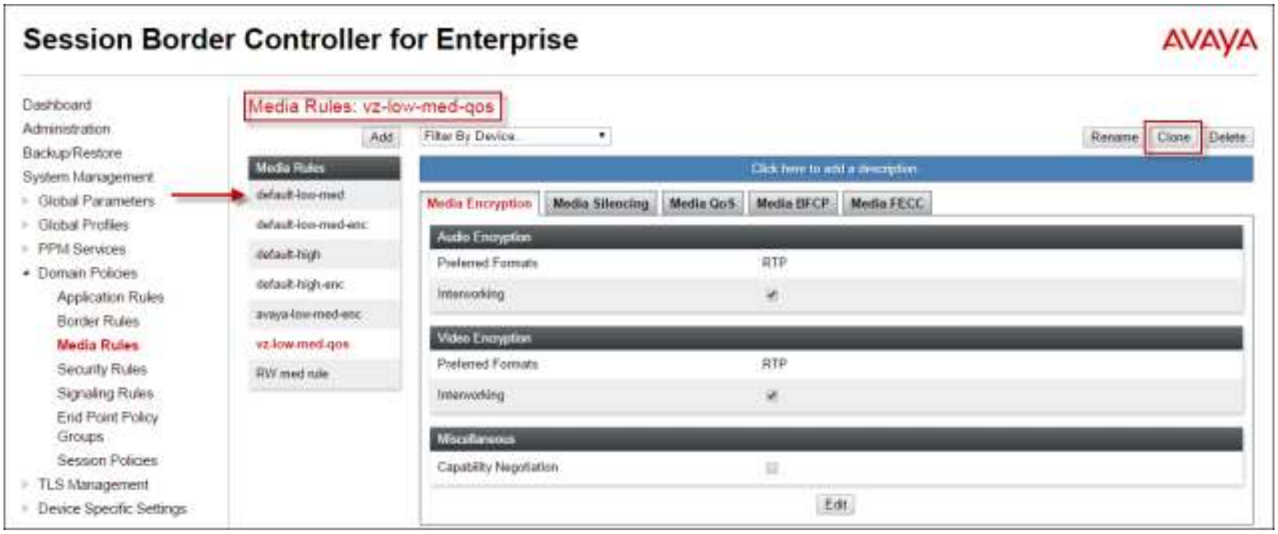
Application Rules define which types of SIP-based Unified Communications (UC) applications the Avaya SBCE security device will protect: voice, video, and/or Instant Messaging (IM). In addition, user can determine the maximum number of concurrent voice and video sessions the network will process in order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

Select **Domain Policies → Application Rules** from the left-side menu as shown below. Click the **Add** button to add a new profile, or select an existing topology hiding profile to edit. In the sample configuration, the "**sip-trunk**" profile was created from cloning the **default-trunk** application rule. In an actual customer installation, set the **Maximum Concurrent Sessions** for the **Audio** application to a value slightly larger than the licensed sessions. For example, if licensed for 1500 session set the values to "**2000**". The **Maximum Session Per Endpoint** should match the **Maximum Concurrent Sessions**.

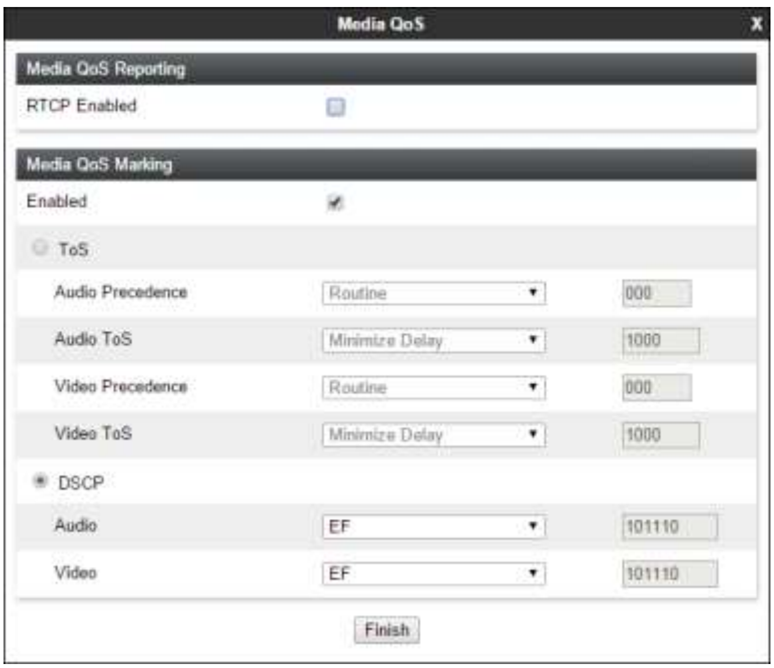
7.8. Media Rule

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

Select **Domain Policies → Media Rules** from the left-side menu as shown below. In the sample configuration, a single media rule is created by cloning the default rule called **default-low-med**. With the **default-low-med** rule chosen, click **Clone**. Enter a descriptive name for the new rule and click **Finish** (not shown).



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



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

7.9. Signaling Rule

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

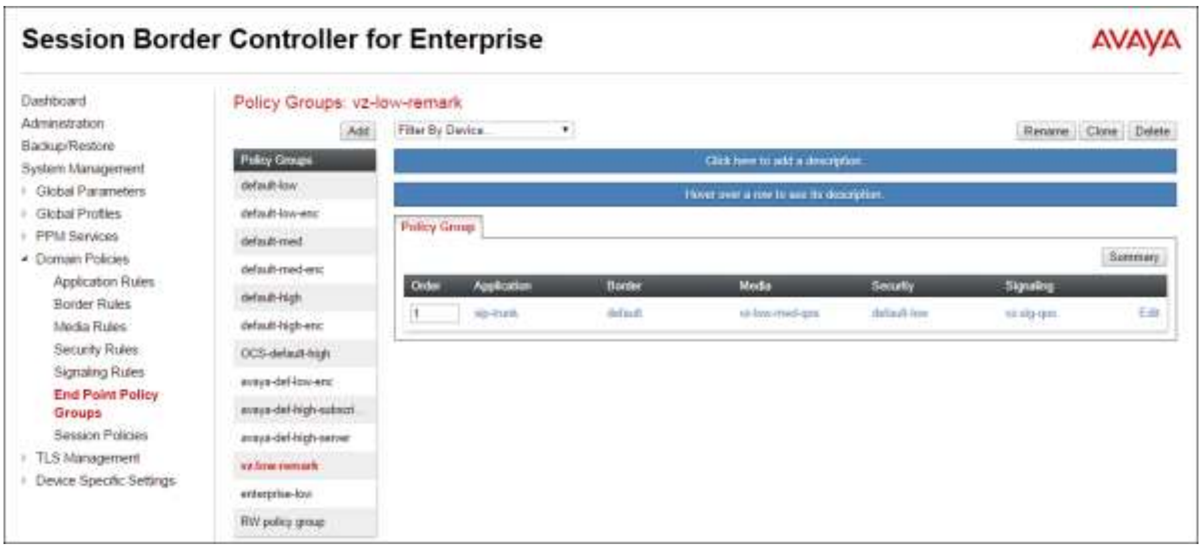
Clone and modify the **default** signaling rule to add the proper quality of service to the SIP signaling towards Verizon. To clone a signaling rule, navigate to **Domain Policies → Signaling Rules**. With the **default** rule chosen, click **Clone**. Enter a descriptive name for the new rule and click **Finish** (not shown).

In the sample configuration, signaling rule “**vz-sig-qos**” is shown with the **DSCP** value “**AF32**” for assured forwarding, changed from the default settings under the **Signaling QoS** tab.

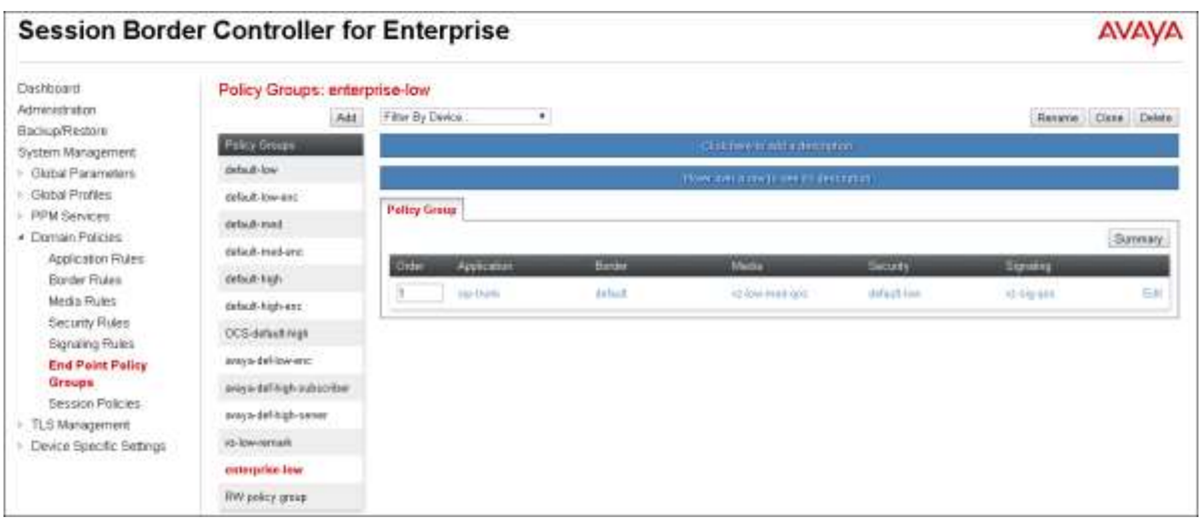
7.10. Endpoint Policy Group

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

To create a new policy group, navigate to **Domain Policies → Endpoint Policy Groups** and click on **Add** as shown below. The following screen shows the "**vz-low-remark**" created for Verizon Business IP Trunk service. The details of the non-default rules chosen are shown in previous sections.



The following screen shows the "**enterprise-low**" created for the enterprise. The details of the non-default rules chosen are shown in previous sections.



7.11. Media Interface

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

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

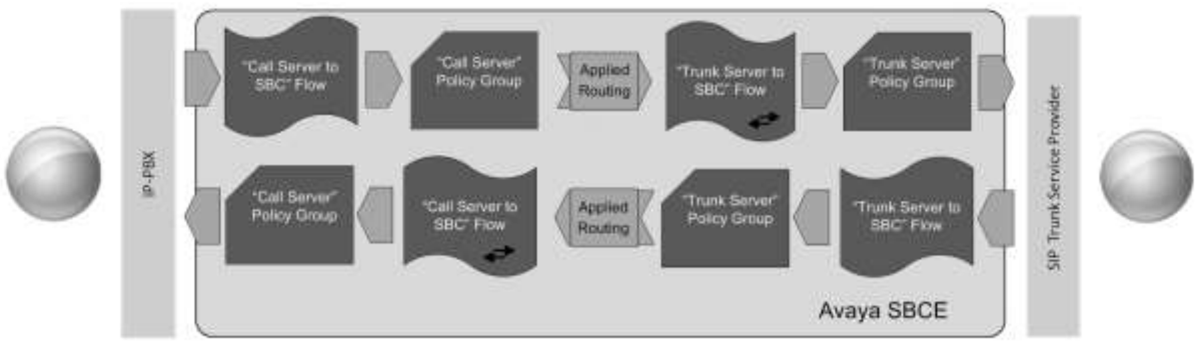
7.12. Signaling Interface

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

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

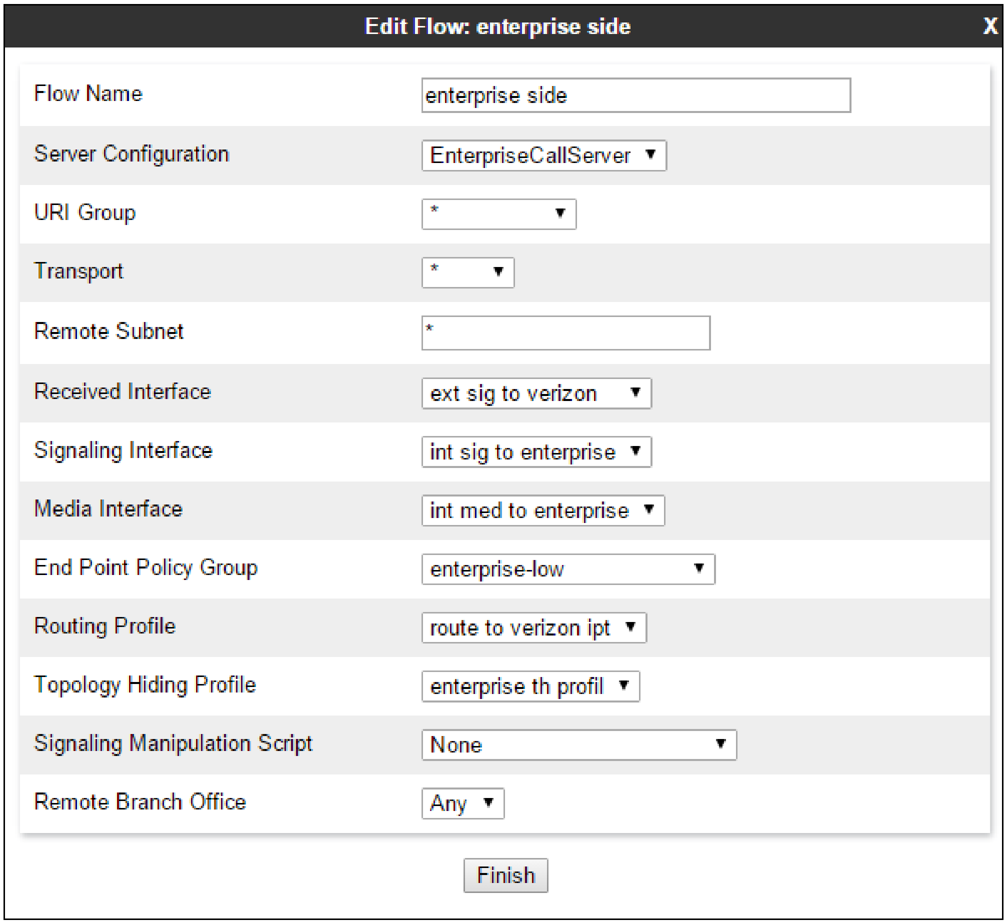
7.13. End Point Flows - Server Flow

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

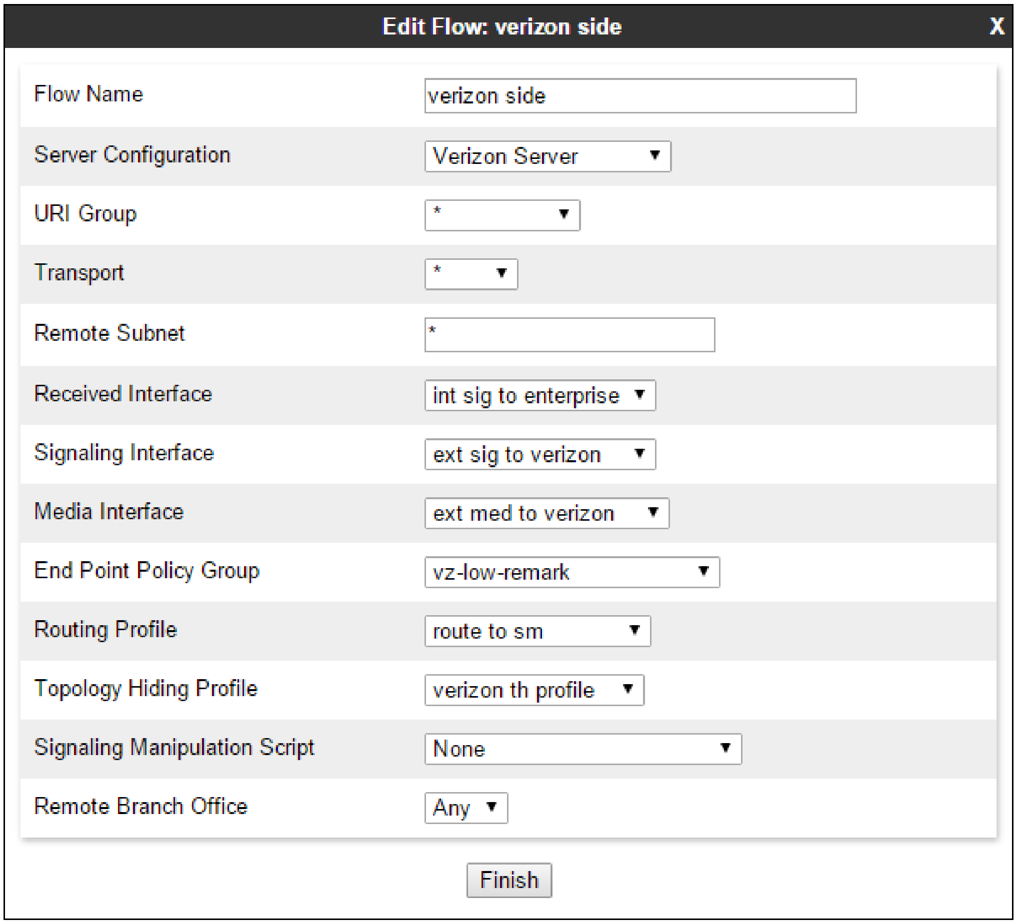


Create a Server Flow for Session Manager and the Verizon Business IP Trunk. To create a Server Flow, navigate to **Device Specific Settings → End Point Flows**. Select the **Server Flows** tab and click **Add** as shown in below.

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



The following screen shows the flow named “**verizon side**” used in the sample configuration. This flow uses the interfaces, policies, and profiles defined in previous sections. Click **Finish**.



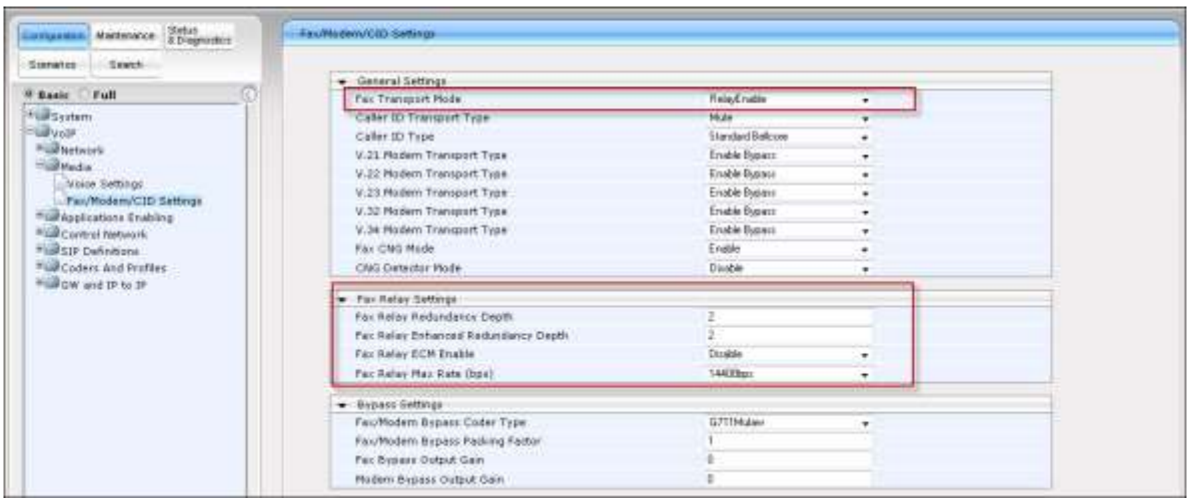
8. AudioCodes MP-114

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

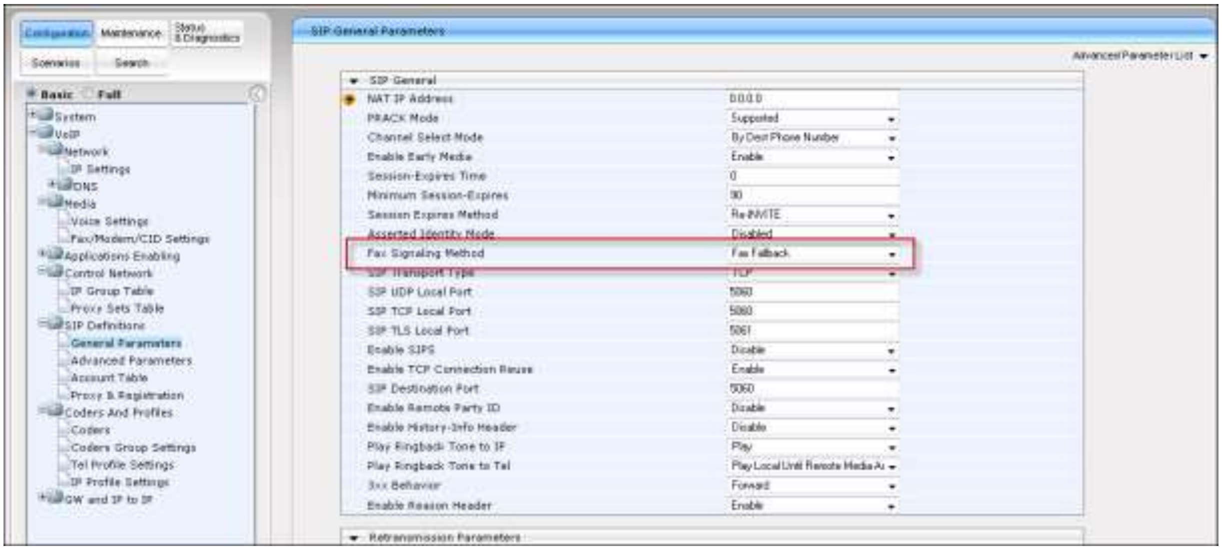
Note - Although the MP-114 is described in these Application Notes, other AudioCodes Telephone Adapters such as the MP-202 or MP-124 may be used.

8.1. Fax Configuration Settings

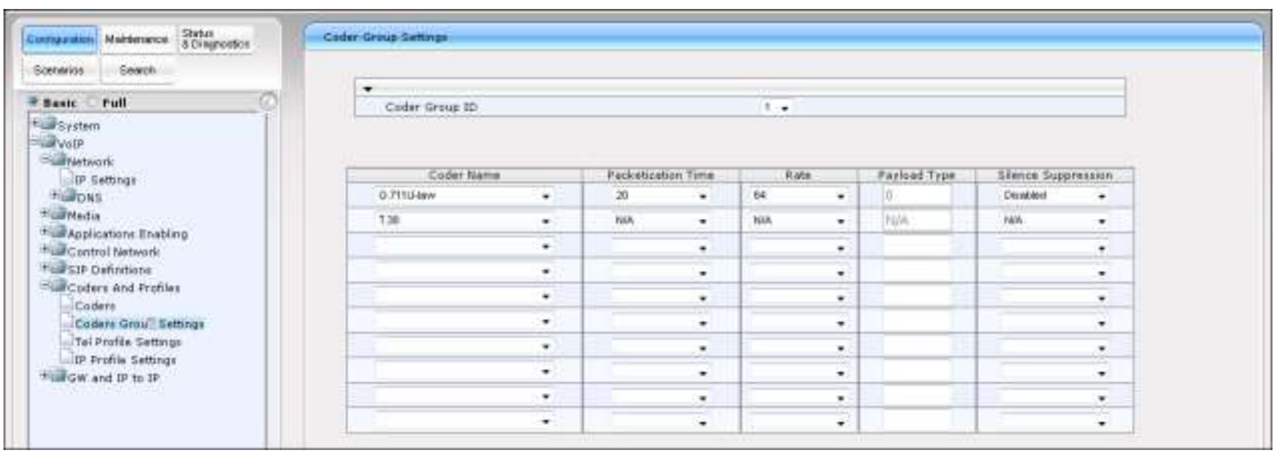
Select **Configuration** menu on the top left of the screen, and navigate to **VoIP→Media→Fax/Modem/CID Settings**. Set the **Fax Transport Mode** to "**RelayEnable**" and set the **Fax Relay Settings** as highlighted below.



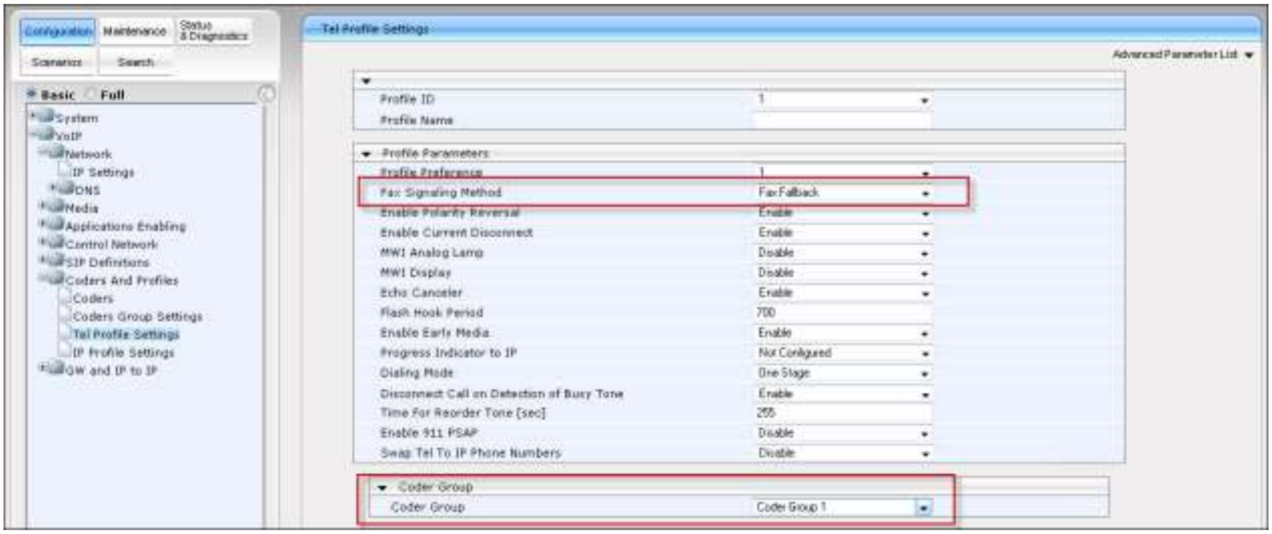
Navigate to **VoIP→SIP Definitions→General Parameters**. Set the **Fax Signaling Method** to "**Fax Fallback**".



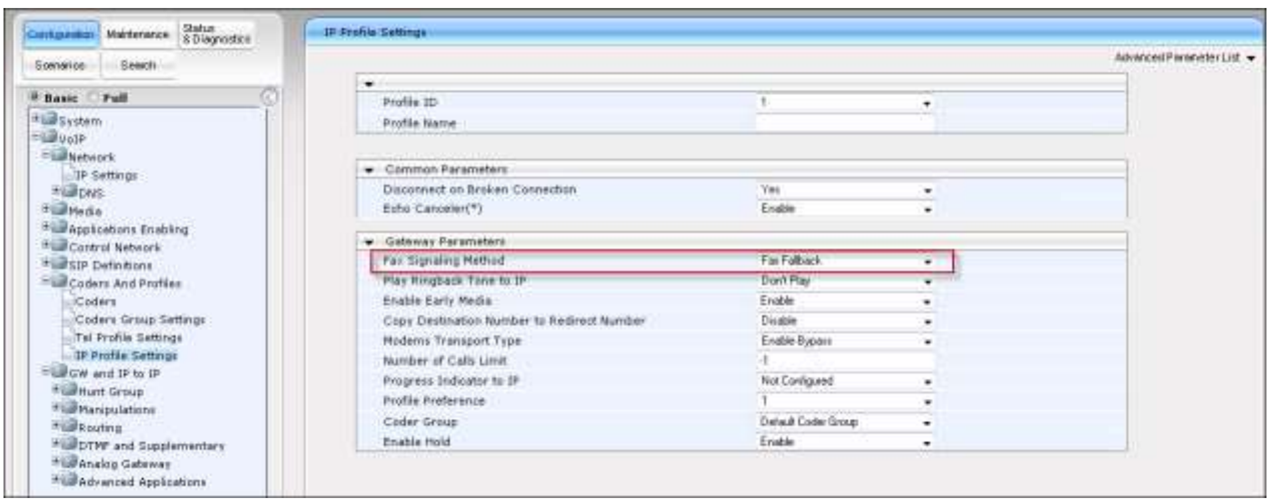
Navigate to **VoIP→Coders and Profiles→Coders Group Settings**. Select **Coder Group ID** "**1**" and under the **Coder Name** column, select "**G.711U-Law**" as the first choice and "**T.38**" as the second choice as shown below. This will allow calls to and from the fax to begin with G.711 as the first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



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

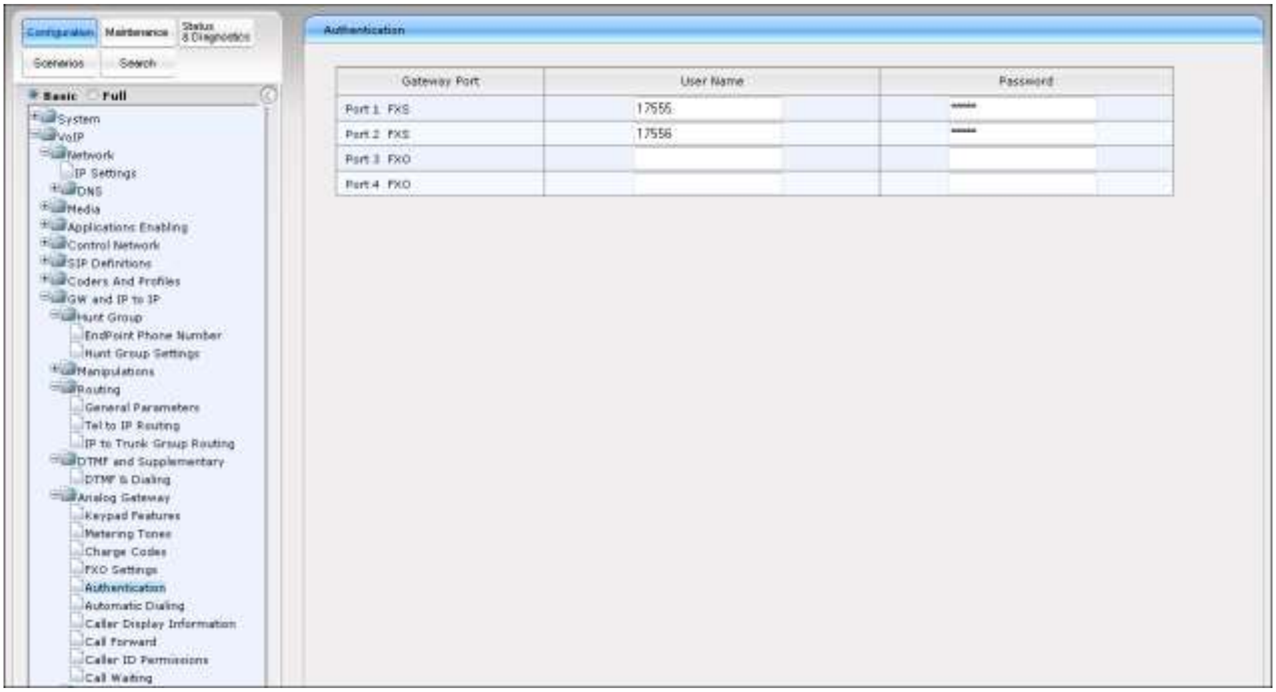


Navigate to **VoIP→Coders and Profiles→IP Profile Settings**. Select **Profile ID** “**1**” and set **Fax Signaling** Method to “**Fax Fallback**”.

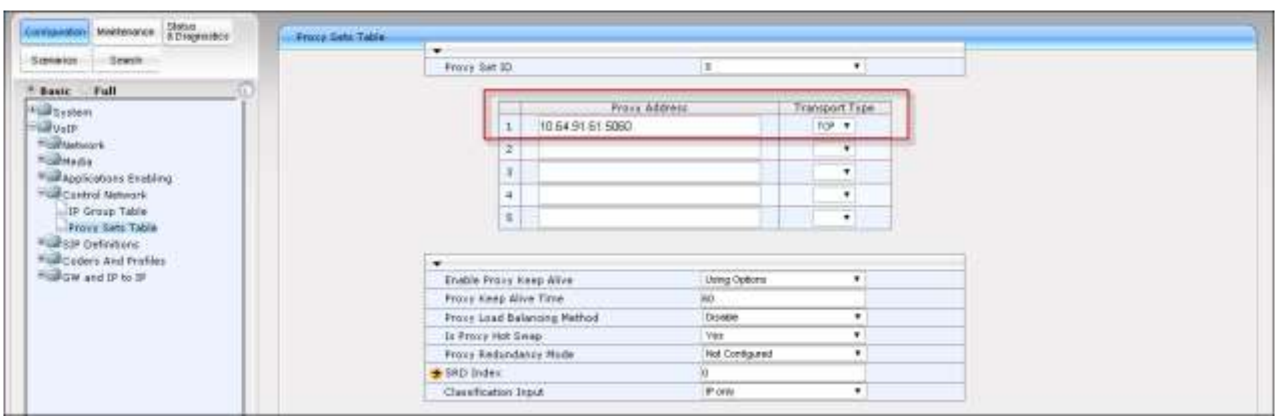


8.2. SIP Endpoint Registration and Proxy Settings

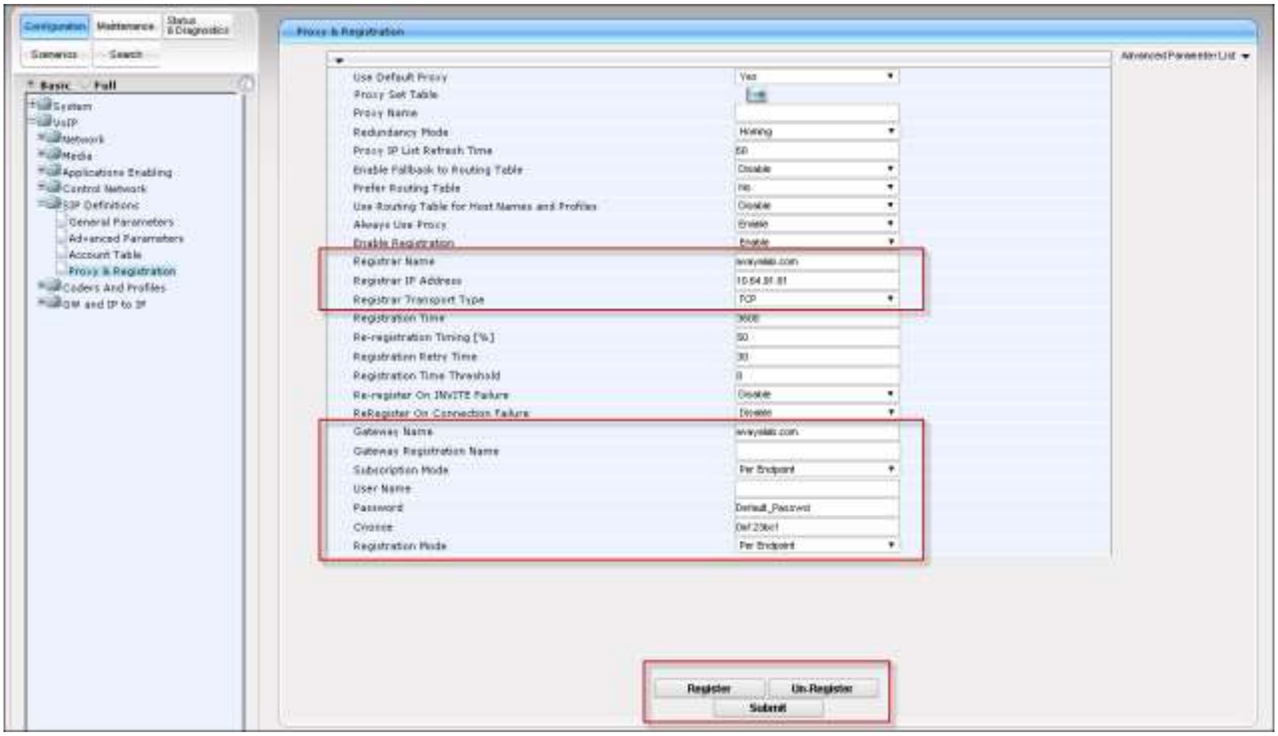
Navigate to **VoIP→Analog Gateway→Authentication**. Set the **User Name** and **Password** for each FXS port used for fax. The **User Name** corresponds to the **Avaya SIP Handle** of the SIP User created in System Manager and the **Password** corresponds to the **Communication Profile Password** as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



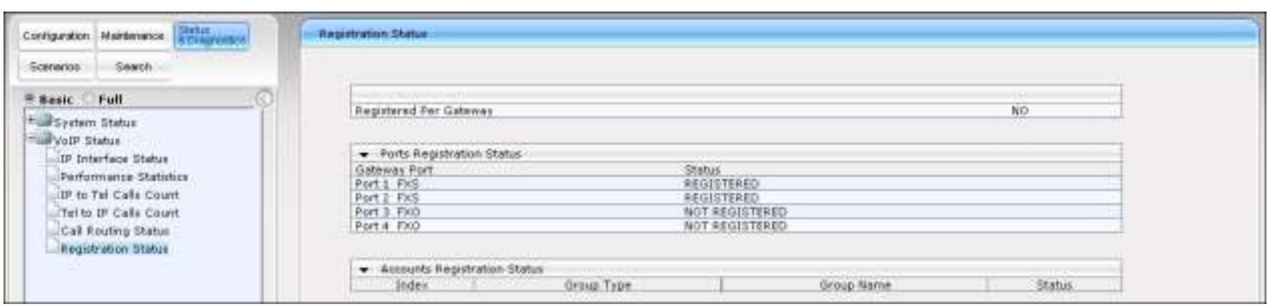
Navigate to **VoIP→Control Network→Proxy Sets Table**. Set the **Proxy Address** to the IP address and port used by Session Manager to listen for SIP REGSTER requests. In the sample configuration, this is "**10.64.91.61:5060**". Set the **Transport Type** to "**TCP**".



Navigate to **VoIP→SIP Definitions→Proxy & Registration**. Set the **Registrar Name** and **Gateway Name** to the domain name used by Session Manager as set in **Section 6.1**. Set the **Registrar IP Address** to Session Manager Security Module IP Address (**10.64.91.61**). Set the **Subscription Mode** and **Registration Mode** to "**Per Endpoint**" and verify the **Cnonce** setting. Click **Submit** and then **Register** on the bottom of the screen.

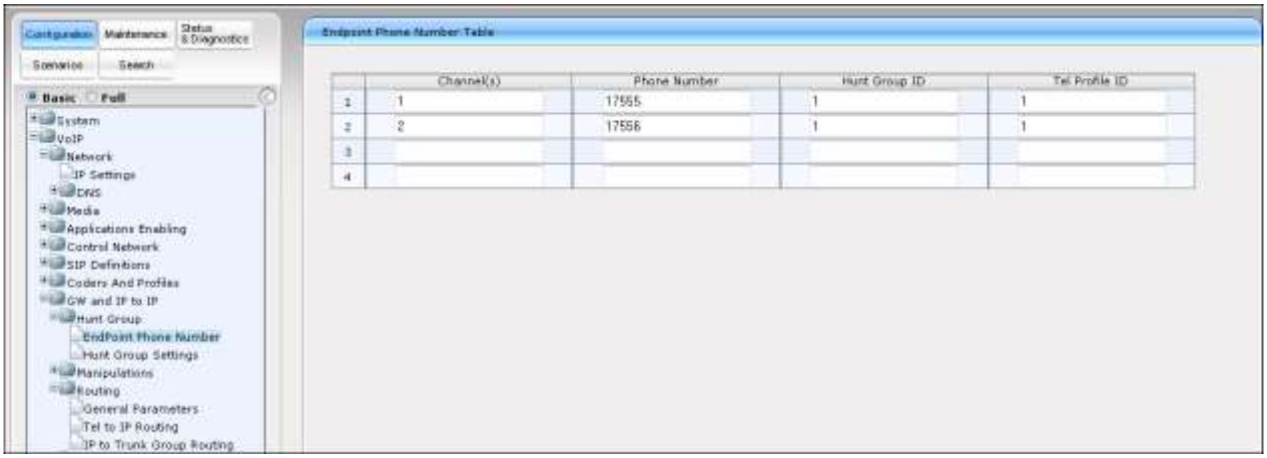


Select the **Status & Diagnostics** menu, and navigate to **VoIP Status→Registration Status**. At this point, the **Gateway Port**(s) used for fax should show a **Status** of "**REGISTERED**".

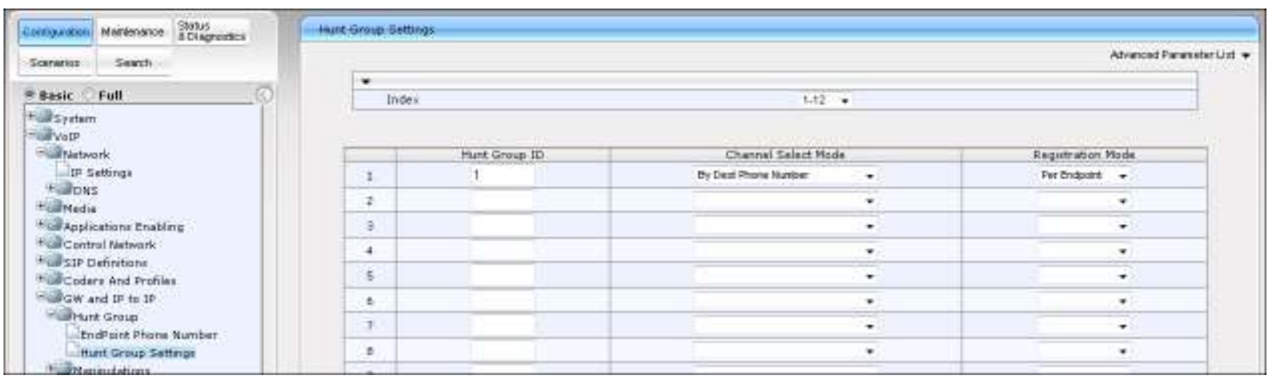


8.3. Routing

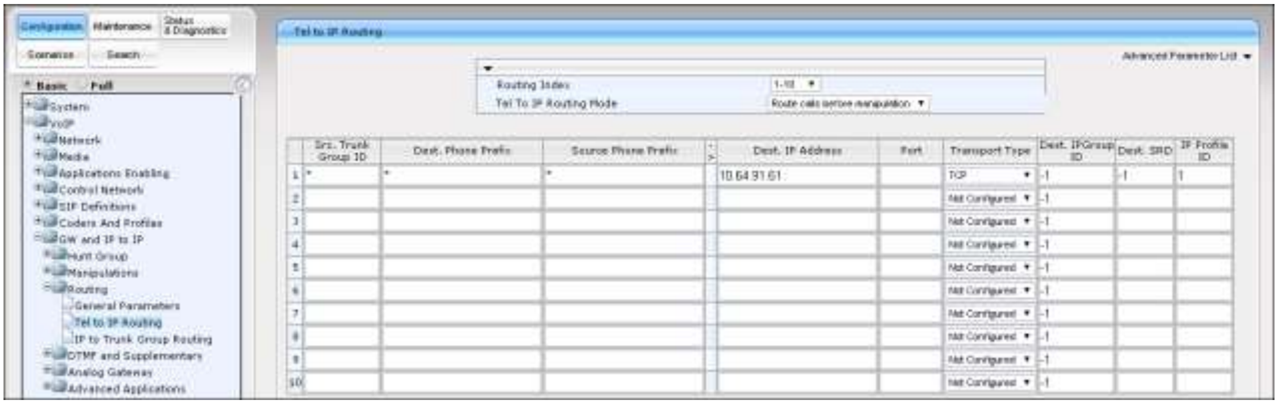
Select **Configuration** menu again on the top left of the screen, and navigate to **VoIP→GW and IP to IP→Hunt Group→EndPoint Phone Number**. Configure a Channel for each FXS port used for fax as shown below. Set the **Hunt Group ID** to "**1**". Set the **Tel Profile ID** to the ID modified in **Section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



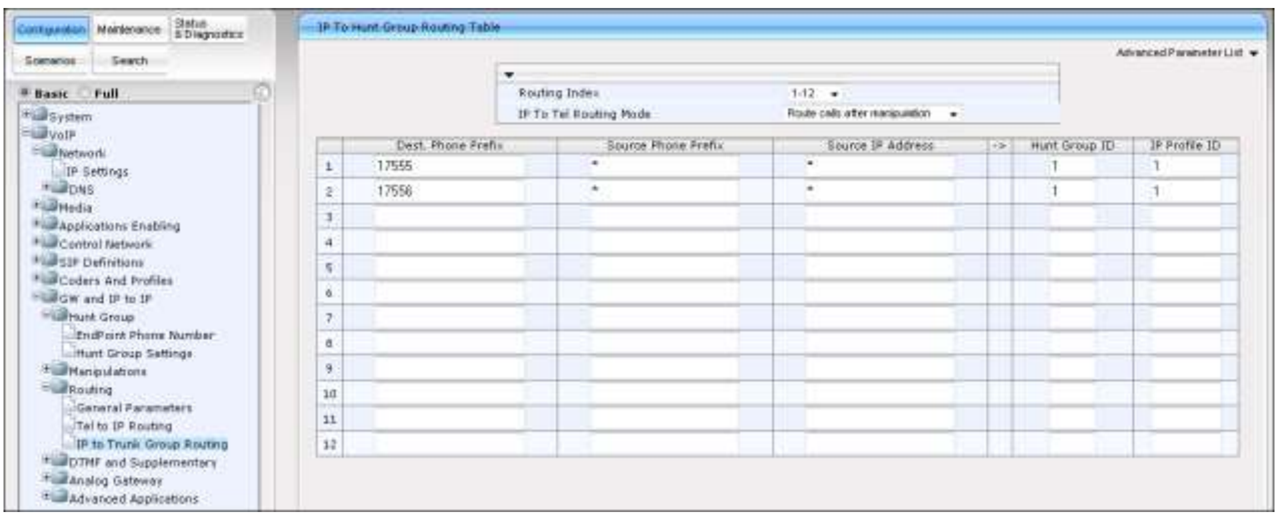
Navigate to **VoIP→GW and IP to IP→Hunt Group→Hunt Group Settings**. Configure **Hunt Group ID** "**1**" with **Channel Select Mode** set to "**By Dest Phone Number**" and **Registration Mode** set to "**Per Endpoint**".



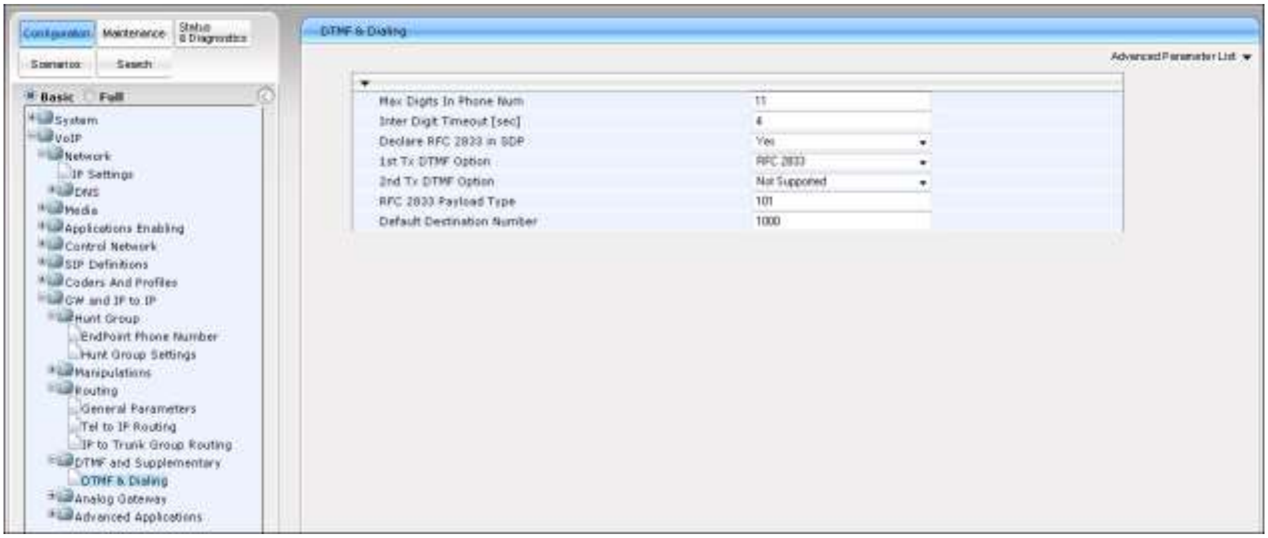
Navigate to **VoIP→GW and IP to IP→Routing→Tel to IP Routing**. Set the **Src. Trunk Group ID**, **Dest. Phone Prefix**, and **Source Phone Prefix** to "*". Set the **Dest. IP Address** to the Session Manager Security Module IP Address (**10.64.91.61**).



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



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



Solution & Interoperability Test Lab Application Notes

9. Verizon Business IP Trunk Services Suite Configuration

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

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

9.1. Service Access Information

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

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

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

10. Verification Steps

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

10.1. Avaya Aura® Communication Manager Verifications

This section illustrates verifications from Communication Manager.

10.1.1 Example Incoming Call from PSTN via Verizon SIP Trunk

Incoming PSTN calls arrive from Verizon at Avaya SBCE, which sends the call to Session Manager. Session Manager sends the call to Communication Manager. On Communication Manager, the incoming call arrives via signaling group 1 and trunk group 1.

The following edited Communication Manager ***list trace tac*** trace output shows a call incoming on trunk group 1. The PSTN telephone dialed 732-945-0233. Session Manager mapped the number received from Verizon to the extension of a Communication Manager telephone (x12003). Extension 12003 is an IP Telephone with IP address 10.64.91.32 in Region 1. The RTP media path is “ip-direct” from the IP Telephone (**10.64.91.32**) to the “inside” of the Avaya SBCE (**10.64.91.50**) in Region 2.

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

15:02:31 TRACE STARTED 08/07/2015 CM Release String cold-00.0.440.0-1004
15:02:35 SIP<INVITE sip:12003@avayalab.com SIP/2.0
15:02:35     Call-ID: 51e93360fb64f603badb8472d415fa04
15:02:35     active trunk-group 1 member 1    cid 0x376
15:02:35 SIP>SIP/2.0 180 Ringing
15:02:35     Call-ID: 51e93360fb64f603badb8472d415fa04
15:02:35     dial 12003
15:02:35     ring station      12003 cid 0x376
15:02:38 SIP>SIP/2.0 200 OK
15:02:38     Call-ID: 51e93360fb64f603badb8472d415fa04
15:02:38     active station      12003 cid 0x376
15:02:38     G729A ss:off ps:20
             rgn:1 [10.64.91.32]:2896
             rgn:2 [10.64.91.50]:35484
15:02:38     G729A ss:off ps:20
             rgn:2 [10.64.91.50]:35484
             rgn:1 [10.64.91.32]:2896
15:02:38 SIP<ACK sip:+17329450233@10.64.91.65:5081;transport=tls SIP
15:02:38 SIP</2.0
15:02:38     Call-ID: 51e93360fb64f603badb8472d415fa04
15:02:48 SIP<BYE sip:+17329450233@10.64.91.65:5081;transport=tls SIP
15:02:48 SIP</2.0
15:02:48     Call-ID: 51e93360fb64f603badb8472d415fa04
15:02:48 SIP>SIP/2.0 200 OK
15:02:48     Call-ID: 51e93360fb64f603badb8472d415fa04
15:02:48     idle trunk-group 1 member 1    cid 0x376
```

The following screen shows **Page 2** of the output of the ***status trunk*** command pertaining to this same call. Note the signaling using port 5081 between Communication Manager and Session Manager. Note the media is "ip-direct" from the IP Telephone (**10.64.91.32**) to the inside IP address of Avaya SBCE (**10.64.91.50**) using codec G.729a.

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

Near-end Signaling Loc: PROCR
  Signaling   IP Address                               Port
   Near-end:  10.64.91.65                            : 5081
    Far-end:  10.64.91.61                            : 5081
 H.245 Near:
  H.245 Far:
   H.245 Signaling Loc:           H.245 Tunneled in Q.931? no

 Audio Connection Type: ip-direct     Authentication Type: None
    Near-end Audio Loc:                        Codec Type: G.729A
   Audio      IP Address                               Port
   Near-end:  10.64.91.32                            : 2896
    Far-end:  10.64.91.50                            : 35486

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

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

```
status trunk 1/249                                              Page   3 of   3
                      SRC PORT TO DEST PORT TALKPATH
src port: T00001
T00001:TX:10.64.91.50:35486/g729a/20ms
S00003:RX:10.64.91.32:2896/g729a/20ms
```

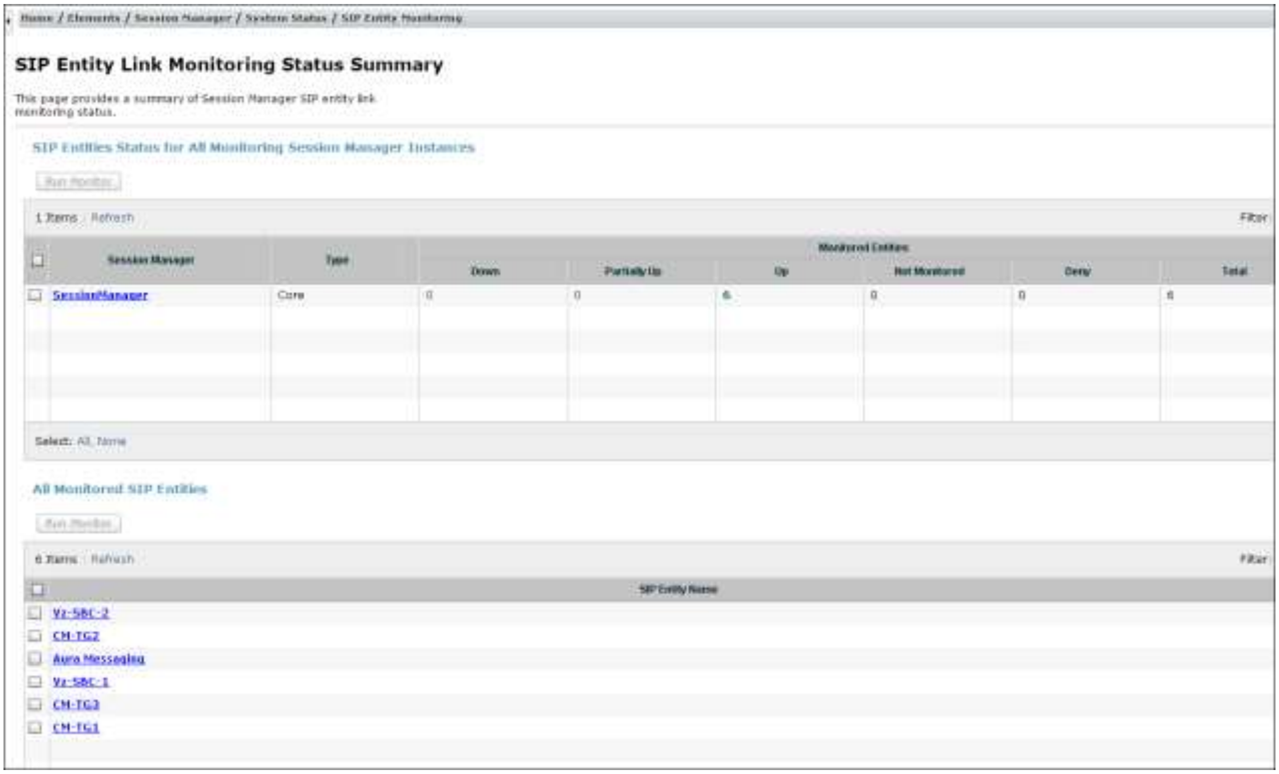
10.1.2 Example Outgoing Calls to PSTN via Verizon SIP Trunk

The following edited trace shows an outbound ARS call from IP Telephone x12003 to the PSTN number 9-1-303-538-2177. The call is routed to route pattern 1 and trunk group 1. The call initially uses the Avaya Media Server (**10.64.91.60**), but after the call is answered, the call is "shuffled" to become an "ip-direct" connection between the IP Telephone (**10.64.91.32**) and the "inside" of the Avaya SBCE (**10.64.91.50**).

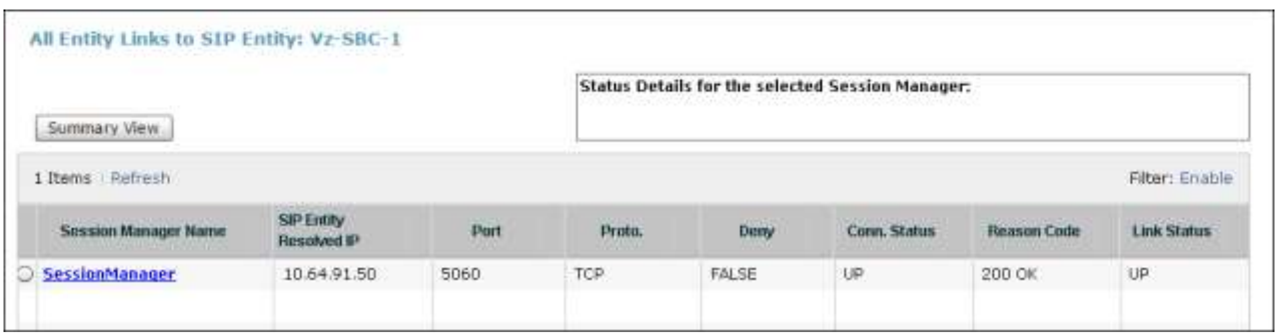
```
list trace tac *01                                                    Page   1
                                LIST TRACE
time            data

15:08:01 TRACE STARTED 08/07/2015 CM Release String cold-00.0.440.0-1004
15:08:18     dial 913035382177 route:PREFIX|FNPA|ARS
15:08:18     route-pattern  1 preference 1 location 1/ALL  cid 0x378
15:08:18     seize trunk-group 1 member 2    cid 0x378
15:08:18     Calling Number & Name 12003 IP Phone 9630
15:08:18 SIP>INVITE sip:+13035382177@avayalab.com SIP/2.0
15:08:18     Call-ID: 6d4e38c23d4841e5b54d0c29f8f3f3
15:08:18     Setup digits +13035382177
15:08:18     Calling Number & Name +17329450233 IP Phone 9630
15:08:18     Proceed trunk-group 1 member 2    cid 0x378
15:08:21     G729 ss:off ps:20
             rgn:2 [10.64.91.50]:35488
             rgn:1 [10.64.91.60]:6024
15:08:24 SIP>ACK sip:13035382177@10.64.91.50:5060;transport=tcp;gsid
15:08:24 SIP>=6d4e34e4-3d48-41e5-b54a-000c29f8f3f3 SIP/2.0
15:08:24     Call-ID: 6d4e38c23d4841e5b54d0c29f8f3f3
15:08:24     active trunk-group 1 member 2    cid 0x378
15:08:24 SIP>INVITE sip:13035382177@10.64.91.50:5060;transport=tcp;g
15:08:24 SIP>sid=6d4e34e4-3d48-41e5-b54a-000c29f8f3f3 SIP/2.0
15:08:24     Call-ID: 6d4e38c23d4841e5b54d0c29f8f3f3
15:08:24     G729 ss:off ps:20
             rgn:1 [10.64.91.32]:2896
             rgn:2 [10.64.91.50]:35488
15:08:24 SIP>ACK sip:13035382177@10.64.91.50:5060;transport=tcp;gsid
15:08:24 SIP>=6d4e34e4-3d48-41e5-b54a-000c29f8f3f3 SIP/2.0
15:08:24     Call-ID: 6d4e38c23d4841e5b54d0c29f8f3f3
15:08:24     G729A ss:off ps:20
             rgn:2 [10.64.91.50]:35488
             rgn:1 [10.64.91.32]:2896
15:09:16 SIP<BYE sip:+17329450233@10.64.91.65:5081;transport=tls SIP
15:09:16 SIP</2.0
```

10.2. Avaya Aura® System Manager and Avaya Aura® Session Manager Verification

This section contains verification steps that may be performed using System Manager for Session Manager. Log in to System Manager. Expand **Elements** → **Session Manager** → **System Status** → **SIP Entity Monitoring**. A screen such as the following is displayed.



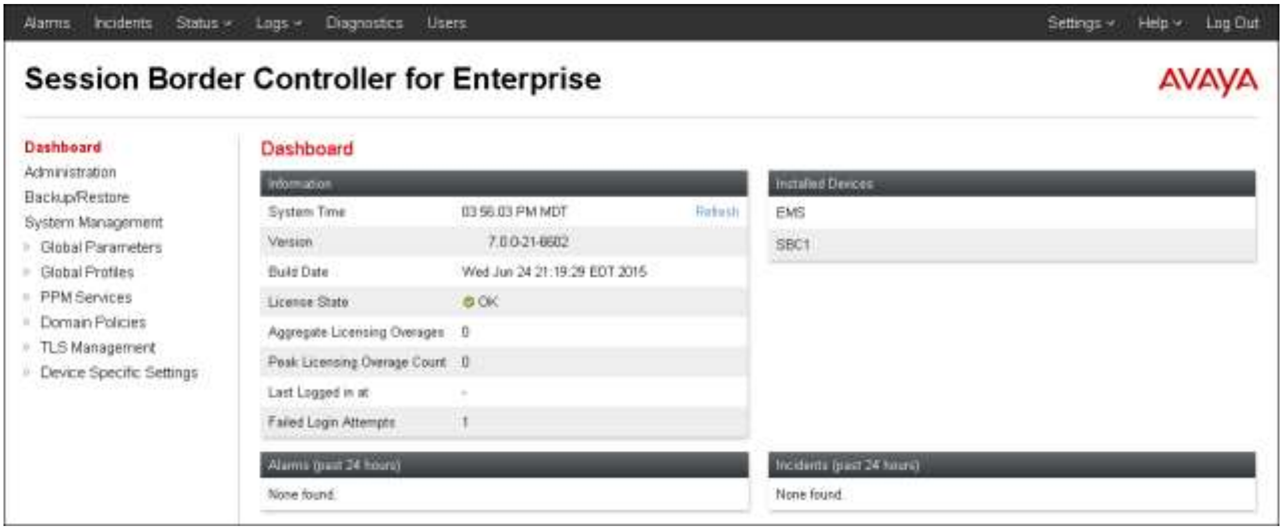
From the list of monitored entities, select an entity of interest, such as "**Vz_SBC-1**". Under normal operating conditions, the **Link Status** should be "**UP**" as shown in the example screen below.



10.3. Avaya Session Border Controller for Enterprise Verification

10.3.1 Welcome Screen

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



10.3.2 Alarms

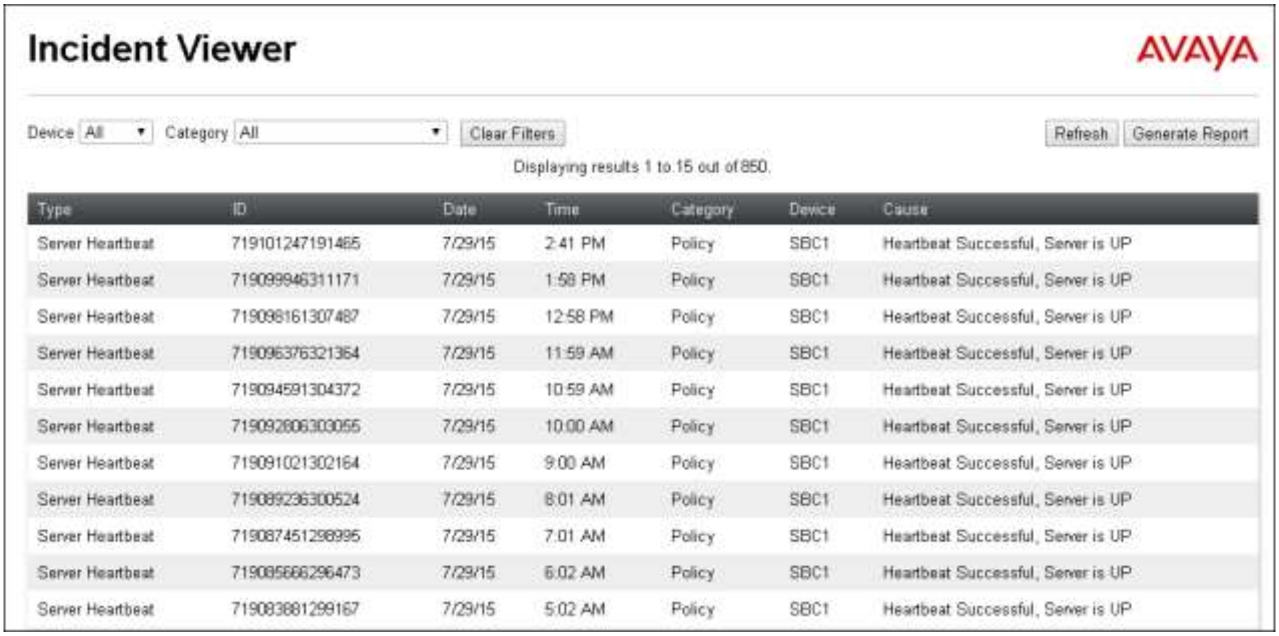
A list of the most recent alarms can be found under the **Alarms** tab on the top left bar.

Alarm Viewer:

10.3.3 Incidents

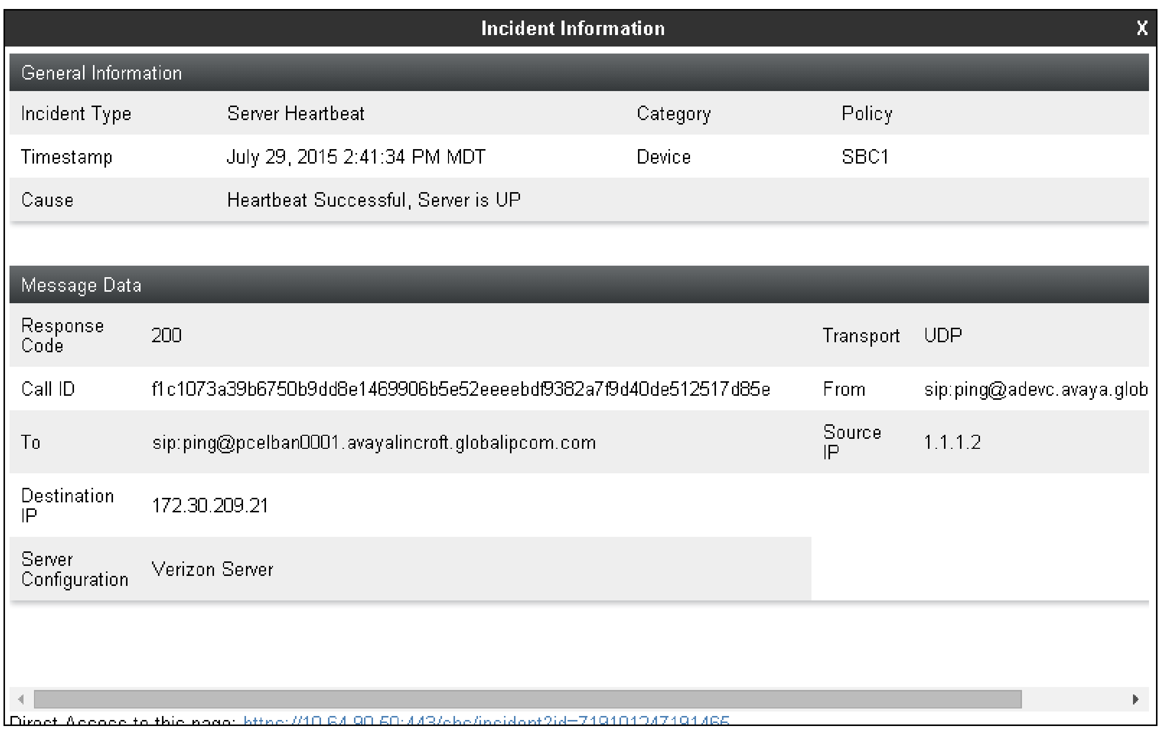
A list of all recent incidents can be found under the **Incidents** tab at the top left next to the Alarms.

Incident Viewer:



Type	ID	Date	Time	Category	Device	Cause
Server Heartbeat	719101247191465	7/29/15	2:41 PM	Policy	SBC1	Heartbeat Successful, Server is UP
Server Heartbeat	719099946311171	7/29/15	1:58 PM	Policy	SBC1	Heartbeat Successful, Server is UP
Server Heartbeat	719098161307487	7/29/15	12:58 PM	Policy	SBC1	Heartbeat Successful, Server is UP
Server Heartbeat	719096376321364	7/29/15	11:59 AM	Policy	SBC1	Heartbeat Successful, Server is UP
Server Heartbeat	719094591304372	7/29/15	10:59 AM	Policy	SBC1	Heartbeat Successful, Server is UP
Server Heartbeat	719092806303055	7/29/15	10:00 AM	Policy	SBC1	Heartbeat Successful, Server is UP
Server Heartbeat	719091021302164	7/29/15	9:00 AM	Policy	SBC1	Heartbeat Successful, Server is UP
Server Heartbeat	719089236300524	7/29/15	8:01 AM	Policy	SBC1	Heartbeat Successful, Server is UP
Server Heartbeat	719087451298995	7/29/15	7:01 AM	Policy	SBC1	Heartbeat Successful, Server is UP
Server Heartbeat	719085666296473	7/29/15	6:02 AM	Policy	SBC1	Heartbeat Successful, Server is UP
Server Heartbeat	719083881299167	7/29/15	5:02 AM	Policy	SBC1	Heartbeat Successful, Server is UP

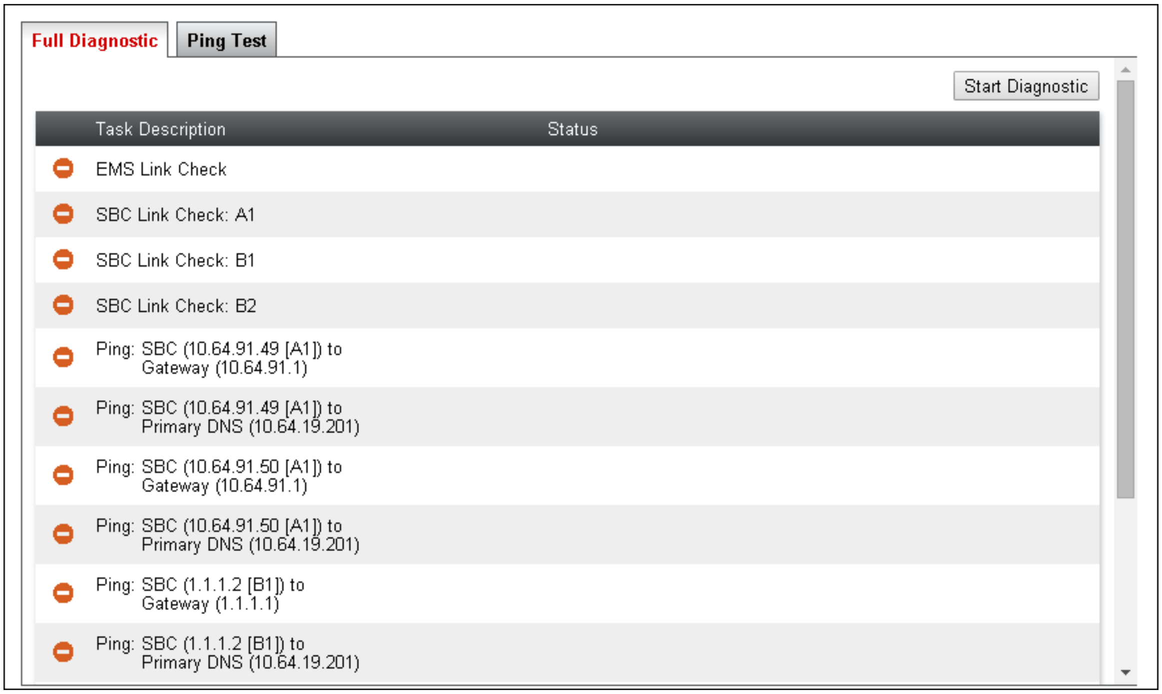
Further Information can be obtained by clicking on an incident in the incident viewer.



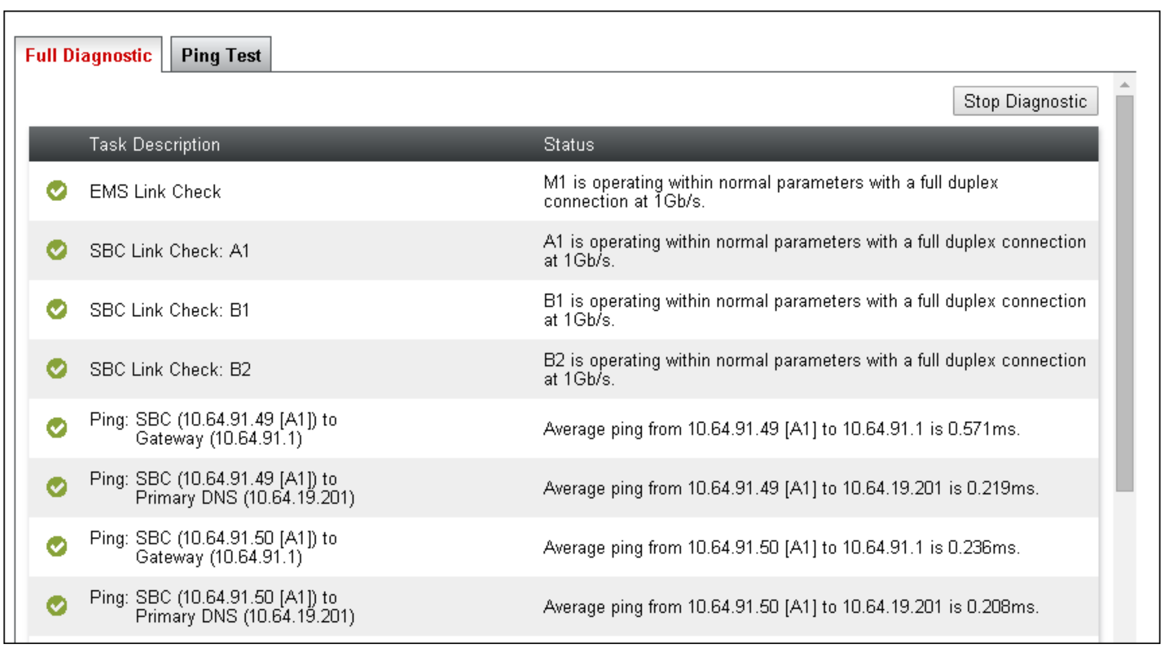
10.3.4 Diagnostics

The full diagnostics check will verify the link of each interface, and ping the configured next-hop gateways and DNS servers.

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

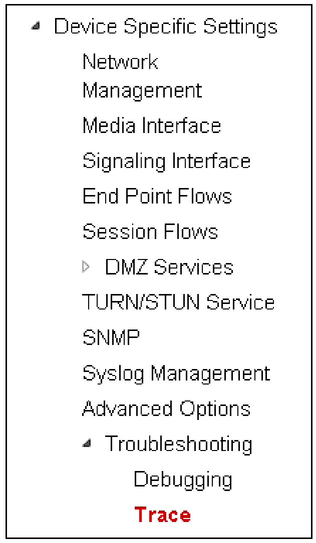


A green check mark or a red x will indicate success or failure.

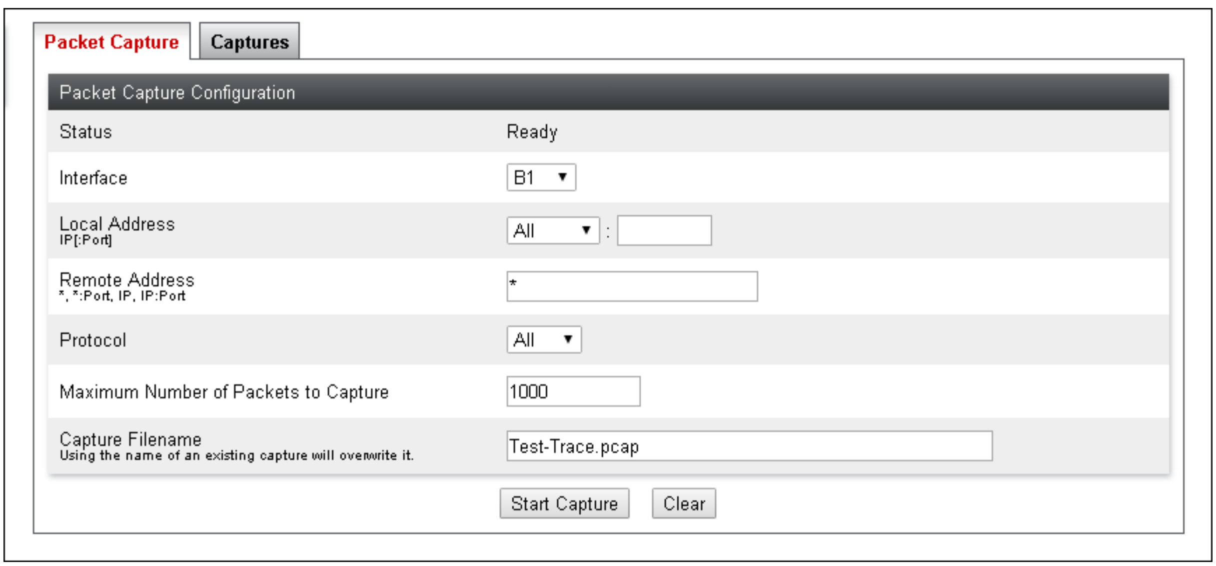


10.3.5 Tracing

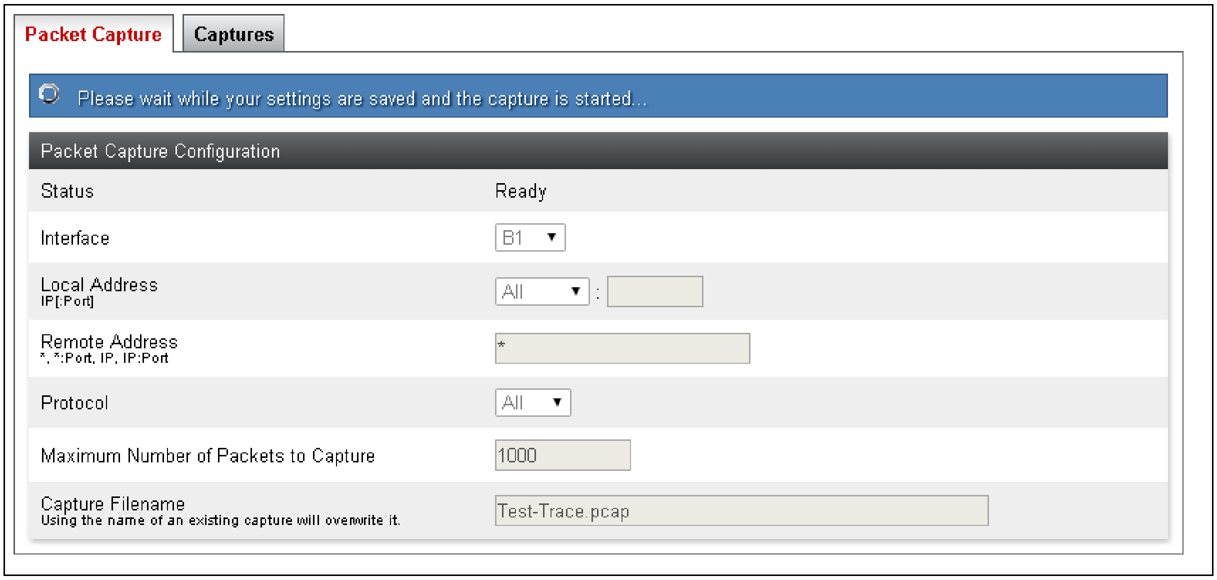
To take a call trace, Select **Device Specific Settings → Troubleshooting → Tracing** from the left-side menu as shown below.



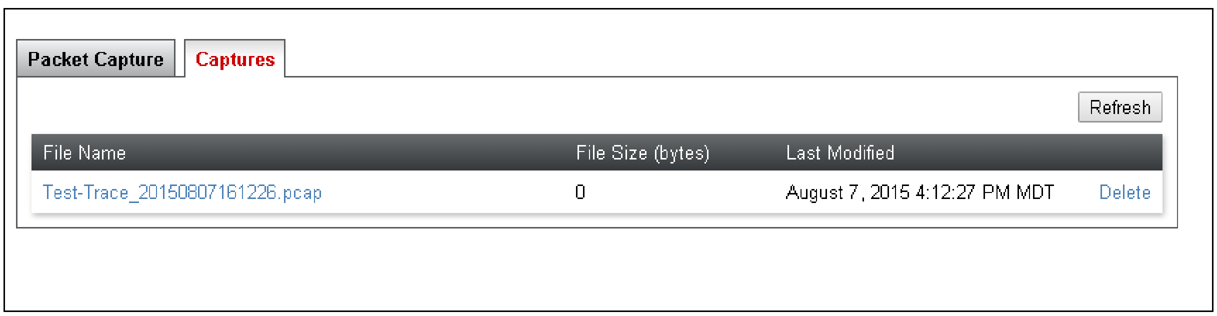
Select the **Packet Capture** tab and set the desired configuration for a call trace and click **Start Capture**.



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



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



11. Conclusion

As illustrated in these Application Notes, Avaya Aura® Communication Manager 7.0, Avaya Aura® Session Manager 7.0, and Avaya Session Border Controller for Enterprise 7.0 can be configured to interoperate successfully with Verizon Business IP Trunk service. This solution allows Avaya Aura® Communication Manager and Avaya Aura® Session Manager users access to the PSTN using a Verizon Business IP Trunk public SIP trunk service connection.

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

12. Additional References

12.1. Avaya

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

[1] *Deploying Avaya Aura® Communication Manager in Virtualized Environment*, Release 7.0
[2] *Administering Avaya Aura® Communication Manager*, Doc ID 03-300509, Release 7.0
[3] *Deploying Avaya Aura® Session Manager on VMware®*, Release 7.0
[4] *Installing Service Packs for Avaya Aura® Session Manager*, Release 7.0
[5] *Upgrading Avaya Aura® Session Manager*, Release 7.0
[6] *Maintaining and Troubleshooting Avaya Aura® Session Manager*, Release 7.0
[7] *Deploying Avaya Aura® System Manager*, Release 7.0
[8] *Deploying Avaya Session Border Controller for Enterprise*, Release 7.0
[9] *Administering Avaya Session Border Controller for Enterprise*, Release 7.0

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

The following Application Notes cover Session Manager 6.3 with Verizon IP SIP Trunk Service using the Avaya Session Border Controller for Enterprise.
[DT-VZIPT-SM63] Application Notes for Avaya Aura® Communication Manager 6.3, Avaya Aura® Session Manager 6.3, and Avaya Session Border Controller for Enterprise with Verizon Business IP Trunk SIP Trunk Service – Issue 1.2

The following Application Notes cover Session Manager 6.2 with Verizon IP SIP Trunk Service using the Avaya Session Border Controller for Enterprise.
[MO-VZIPT-SM62] Application Notes for Avaya Aura® Communication Manager 6.2, Avaya Aura® Session Manager 6.2, and Avaya Session Border Controller for Enterprise with Verizon Business IP Trunk SIP Trunk Service – Issue 1.0

12.2. Verizon Business

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

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

12.3. AudioCodes

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

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

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.