Avaya Solution & Interoperability Test Lab

# Application Notes for Avaya Aura™ Communication Manager 6.0, Avaya Aura™ Session Manager 6.0, and Avaya Aura™ Session Border Controller with Verizon Business IP Contact Center (IPCC) Services Suite – Issue 1.1

## Abstract


These Application Notes describe a sample configuration of Avaya Aura™ Communication Manager 6.0, Avaya Aura™ Session Manager 6.0, and Avaya Aura™ Session Border Controller (SBC) integration with Verizon Business IP Contact Center (IPCC) Services suite. The Verizon Business IPCC Services suite is comprised of the VoIP Inbound, IP Contact Center, and IP-IVR SIP trunk service offers. This service suite provides toll free inbound calling via standards-based SIP trunks as well as re-routing of inbound toll free calls to alternate destinations based upon SIP redirection messages generated by Avaya Aura™ Communication Manager. The Avaya Aura™ Communication Manager Network Call Redirection (NCR) and SIP User-to-User Information (UUI) features can be utilized together to transmit UUI within SIP signaling messages to alternate destinations via the Verizon network. These Application Notes update previously published Application Notes with newer versions of Communication Manager and Session Manager, **including a declaration of support for Communication Manager Release 6.0.1 and Session Manager Release 6.1, as noted in Section 3**.

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

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# 1. Introduction

These Application Notes describe a sample configuration of Avaya Aura™ Communication Manager 6.0, Avaya Aura™ Session Manager 6.0, and Avaya Aura™ Session Border Controller (SBC) integration with Verizon Business IP Contact Center (IPCC) Services suite. The Verizon Business IPCC Services suite is comprised of the VoIP Inbound, IP Contact Center, and IP-IVR SIP trunk service offers.

In the sample configuration, the SBC is used as an edge device between the Avaya CPE and Verizon Business. The SBC performs SIP header manipulation and provides Network Address Translation (NAT) functionality to convert the private Avaya CPE IP addressing to IP addressing appropriate for the Verizon access method.

Avaya Aura™ Session Manager is used as the Avaya SIP trunking "hub" connecting to Avaya Aura™ Communication Manager, the Avaya Aura™ Session Border Controller, and other applications such as Avaya Modular Messaging. Communication Manager SIP trunks and SBC "sip-gateways" are provisioned to terminate at Session Manager.

The Verizon Business IPCC Services suite described in these Application Notes is designed for business customers using Communication Manager and Session Manager. The service provides inbound toll-free service via standards-based SIP trunks. Using SIP Network Call Redirection (NCR), trunk-to-trunk connections of certain inbound calls at Communication Manager can be avoided by requesting that the Verizon network transfer the inbound caller to an alternate destination. In addition, the Communication Manager SIP User-to-User Information (UUI) feature can be utilized with the SIP NCR feature to transmit UUI within SIP signaling messages to alternate destinations. This capability allows the service to transmit a limited amount of call-related data between call centers to enhance customer service and increase call center efficiency. Examples of UUI data might include a customer account number obtained during a database query or the best service routing data exchanged between sites using Communication Manager.

Verizon Business IPCC Services suite is a portfolio of IP Contact Center (IPCC) interaction services that includes VoIP Inbound and IP Interactive Voice Response (IP IVR) service. Access to these features may use Internet Dedicated Access (IDA) or Private IP (PIP). PIP was used for the sample configuration described in these Application Notes. VoIP Inbound is the base service offering that offers core call routing and termination features. IP IVR is an enhanced service offering that includes features such as menu-routing, custom transfer, and additional media capabilities.

## 1.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows to Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager, and subsequent redirection of inbound calls to Verizon for re-routing to alternate destinations. See **Section 2.2** for an overview of key call flows and **Section 9** for detailed verifications of key call flows. Additional test objectives are listed in **Section 8**.

## 1.2. Support

### 1.2.1 Avaya

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus.

### 1.2.2 Verizon

For technical support, visit online support at http://www.verizonbusiness.com/us/customer/

## 1.3. Known Limitations

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

- Following a loss and restoration of Ethernet connectivity, the Avaya Aura™ Session Border Controller may not recover quickly without manual intervention. This problem has been reported to the product team (Ticket #28231, PD00014149) for resolution in a future software version. To trigger recovery of service following a loss and restoration of Ethernet connectivity, an arp request can be issued from the SBC for the default gateway IP address of the previously failed network interface. More specifically, the following actions will trigger recovery. Select the **Actions** tab from the menu shown in **Section 9.3**. From the left side menu, click the "arp" action. In the resultant right panel, select "request" from the **type** drop-down menu, and enter the IP Address of the default gateway for the previously failed interface. Click the **Invoke** button. Assuming the previously failed Ethernet connectivity has been restored, the arp request will succeed and stimulate full service recovery.
- Verizon Business IPCC Services suite does not support fax.
- Verizon Business IPCC Services suite does not support History Info or Diversion Headers.
- Verizon Business IPCC Services suite does not support G.729B codec.
- The following two potential problems have a similar root cause, and neither problem will be seen if the SIP header manipulation described in **Section 6.3.8** is implemented on the Avaya Aura™ Session Border Controller. Verizon has been notified of the Verizon SIP messaging triggering the problems, and Verizon is tracking the issue via **Thrupoint Case #00001146**. Independent of any future Verizon change, Avaya is also working towards a resolution via Communication Manager Modification Request **defsw102344**, targeted to a future Communication Manager 6.0 service pack. The SIP manipulation described in **Section 6.3.8** prevents Verizon from seeing a "sendonly" media attribute in SDP from the enterprise site. As a result, RTP will remain bi-directional when a call is put on hold at the enterprise site. This

has no user-visible consequence, since the enterprise site will not be listening to the media arriving from Verizon while the call is on hold at the enterprise site. The "sendonly" media attribute is only sent by Communication Manager when the Network Call Redirection (NCR) field on the SIP trunk group is enabled, meaning that bi-directional media flow for a call on hold is the normal case when NCR is disabled. To reiterate, the following problems will be seen only if the SIP header manipulation in **Section 6.3.8** is not configured:

  - If Avaya Aura™ Communication Manager Network Call Redirection (NCR) is enabled for the SIP trunk group used for the call, and a Verizon toll-free call is on hold listening to music on hold from the Avaya CPE, the music on hold will cease to be heard by the caller if a refresh INVITE is sent to Verizon while the call is on hold. After the exchange of SIP messages stimulated by a refresh INVITE while a call is on hold, if the Avaya CPE user tries to resume the held call, the audio path cannot be re-established.
  - If Avaya Aura™ Communication Manager Network Call Redirection (NCR) is enabled for the SIP trunk group used for the call, traditional telephone transfer and conference of an inbound toll-free call to another CPE telephone can result in no talk path conditions with the Verizon IPCC network after the transfer or conference operations are completed.

- If the Avaya Aura™ Communication Manager configuration described in **Section 4.11** is implemented for each Vector Directory Number (VDN) that may use SIP NCR and REFER, the additional messaging issue described by this bullet can be avoided for calls from Verizon routed directly to the VDN. After Verizon accepts the REFER from the enterprise equipment, Verizon sends an INVITE message to the enterprise, indicating a network hold state with connection address "0.0.0.0". If the Communication Manager configuration described in Section 4.11 is not implemented for the VDN associated with the vector issuing the REFER, Communication Manager will also send an INVITE message to the Verizon network. Verizon will respond to this INVITE message with a "491 Request Pending" response, which will trigger another INVITE message from Communication Manager to the Verizon network. A series of INVITE/491 message exchanges will continue for several seconds in this fashion. These messages do not impact the completion of the call to the refer-to destination, but the extra messaging can be avoided by implementing the configuration described in **Section 4.11** for each VDN associated with a vector that can issue a REFER. With the configuration shown in Section 4.11, for a Verizon IPCC call routed directly to a VDN, Communication Manager will not send the INVITE message to Verizon after the Verizon "hold" INVITE, thus preventing the trigger for the 491/INVITE series of messages.
- Although the Verizon IPCC Services suite defines call flows that would allow a call to remain in Communication Manager vector processing upon failure of a vector-triggered REFER attempt, such call scenarios could not be verified on the production Verizon circuit used for testing. See **Section 2.2.3** for additional information.
- Although Avaya Aura™ Session Manager 6.0 supports the use of SIP phones, and SIP phones were present in the sample configuration, the configuration of the SIP phones is not covered by these Application Notes.

## 2. Reference Configuration

**Figure 1** illustrates the sample configuration used for the DevConnect compliance testing. The configuration is comprised of the Avaya CPE location connected via a T1 Internet connection to the Verizon Business IPCC services node. The Avaya CPE location simulates a customer site. At the edge of the Avaya CPE location, an Avaya Aura™ Session Border Controller (SBC) provides NAT functionality and SIP header manipulation. The SBC receives traffic from Verizon Business IPCC Services on port 5060 and sends traffic to the Verizon Business IPCC Services using destination port 5072, using the UDP protocol. The PIP service defines a secure MPLS connection between the Avaya CPE T1 connection and the Verizon IPCC services node.

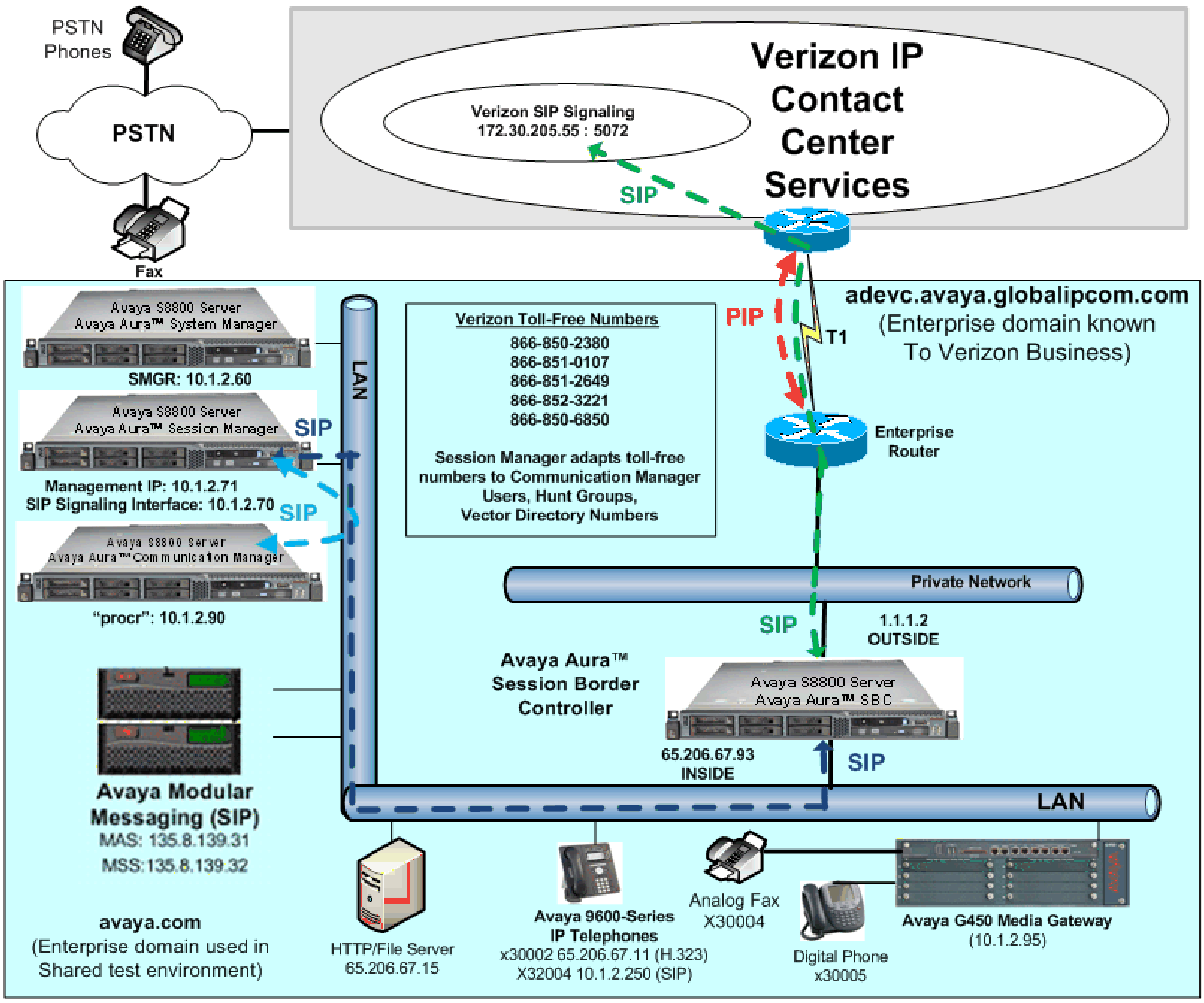



**Figure 1: Avaya Interoperability Test Lab Configuration**

The Verizon provided toll-free numbers were mapped by Avaya Aura™ Session Manager or Avaya Aura™ Communication Manager to various Communication Manager extensions. The extension mappings were varied during the testing to allow inbound toll-free calls to terminate directly on user extensions or indirectly through hunt groups, vector directory numbers (VDNs) and vectors to user extensions and contact center agents.

The Avaya CPE environment was known to Verizon Business IP Trunk Service as FQDN ***adevc.avaya.globalipcom.com***. For efficiency, the Avaya CPE environment utilizing Session Manager Release 6.0 and Communication Manager Release 6.0 was shared among many ongoing test efforts at the Avaya Solution Interoperability lab. Access to the Verizon Business IPCC services was added to a configuration that already used domain "avaya.com" at the enterprise. As such, Session Manager or the SBC can be used to adapt the "avaya.com" domain to the domain known to Verizon. These Application Notes indicate the configuration that would not be required in cases where the CPE domain in Communication Manager and Session Manager match the CPE domain known to Verizon.

The following summarizes various SIP header contents in SIP INVITE messages for inbound toll-free calls using the sample configuration:

- Verizon Business IPCC Services node sends the following to the SBC using destination port 5060 via UDP:
  - The CPE FQDN of ***adevc.avaya.globalipcom.com*** in the Request URI.
  - The Verizon IPCC Services gateway IP address in the From and PAI headers.
  - The SBC outside public IP address in the To header.
- The SBC sends the following to Session Manager using destination port 5060 via TCP:
  - The Request URI contains ***adevc.avaya.globalipcom.com***
  - The host portion of the From header and PAI header contain the Verizon IPCC Services gateway IP Address
  - The host portion of the To header contains IP address ***adevc.avaya.globalipcom.com***
- Avaya Aura™ Session Manager sends the following to Communication Manager using destination port 5062 via TCP to allow Communication Manager to distinguish Verizon traffic from other traffic arriving from the same instance of Session Manager
  - The Request URI contains ***avaya.com***, to match the shared Avaya SIL test environment.
  - The From, To and PAI headers match what was received from the SBC

**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 FQDNs and IP addressing as required.

## 2.1. History Info and Diversion Headers

The Verizon Business IPCC Services suite does not support SIP History Info Headers or Diversion Headers. Therefore, Avaya Aura™ Communication Manager was provisioned not to send History Info Headers or Diversion Headers.

## 2.2. Call Flows

To understand how inbound Verizon toll-free calls are handled by Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager, key call flows are summarized in this section.

### 2.2.1 Inbound Toll Free Call with no Network Call Redirection

The first call scenario illustrated in **Figure 22** is an inbound Verizon toll-free call that is routed to Avaya Aura™ Communication Manager, which in turn routes the call to a vector, agent, or phone. No redirection is performed in this simple scenario. A detailed verification of such a call with Communication Manager and Wireshark traces can be found in **Section 9.1.1**.

1. A PSTN phone originates a call to a Verizon IP Toll Free number.
2. The PSTN routes the call to the Verizon IP Toll Free service network.
3. The Verizon IP Toll Free service routes the call to the SBC.
4. The SBC performs SIP Network Address Translation (NAT) and any necessary SIP header modifications, and routes the call to Avaya Aura™ Session Manager.
5. Avaya Aura™ Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Routing Policies, determines where the call should be routed. In this case, Avaya Aura™ Session Manager routes the call to Avaya Aura™ Communication Manager.
6. Depending on the called number, Avaya Aura™ Communication Manager routes the call to a) a hunt group or vector, which in turn routes the call to an agent or phone, or b) directly to a phone.

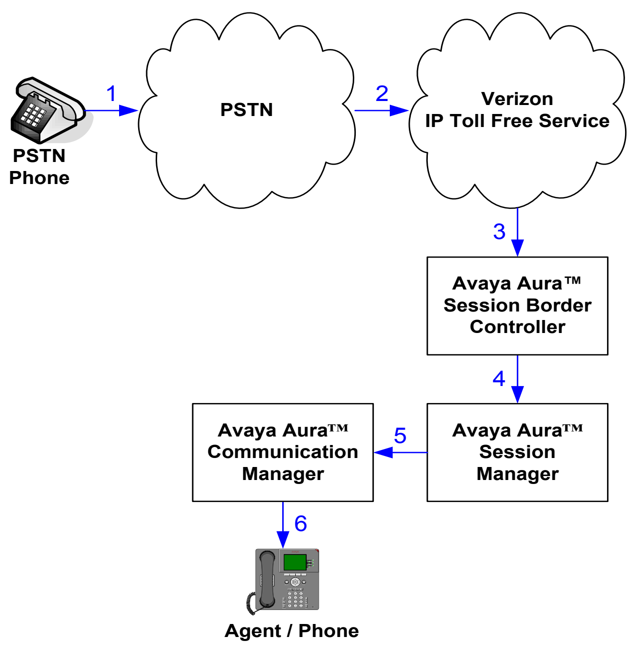



**Figure 2: Inbound Verizon IP Toll Free Call – No Redirection**

### 2.2.2 Inbound IP Toll Free Call with Post-Answer Network Call Redirection

The second call scenario illustrated in **Figure 3** is an inbound Verizon toll free call that is routed to an Avaya Aura™ Communication Manager Vector Directory Number (VDN) to invoke call handling logic in a vector. The vector answers the call and then redirects the call back to Verizon for routing to an alternate destination. Note that Verizon IP Toll Free service does not support redirecting a call before it is answered (using a SIP 302), and therefore the vector must include a step that results in answering the call, such as playing an announcement.

Detailed verifications of such calls with both Communication Manager and Wireshark traces can be found in **Section 9.1.2** for a PSTN destination and Section 9.1.3 for a Verizon SIP-connected alternate destination. In the latter case, the Verizon network can be used to pass User to User Information (UUI) from the redirecting site to the alternate destination.

1. Same as the first five steps in **Figure 2**.
2. Avaya Aura™ Communication Manager routes the call to a vector, which answers the call, plays an announcement, and attempts to redirect the call by sending a SIP REFER message out the SIP trunk upon which the inbound call arrived. The SIP REFER message specifies the alternate destination in the Refer-To header. The SIP REFER message passes back through Avaya Aura™ Session Manager and the SBC to the Verizon network.
3. The Verizon network places a call to the target party contained in the Refer-To header. Upon answer, the calling party is connected to the target party.
4. The call is cleared on the redirecting/referring party (Avaya Aura™ Communication Manager).

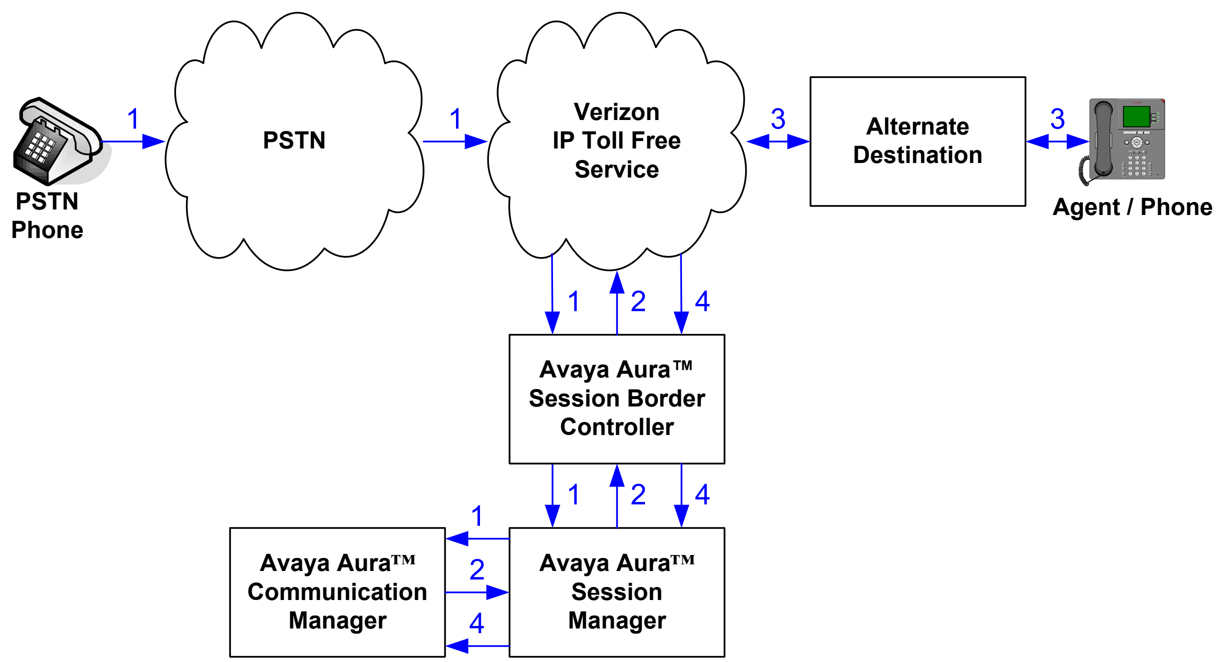



**Figure 3: Inbound Verizon Toll-Free Call – Post-Answer SIP REFER Redirection Successful**

## 2.2.3 Inbound IP Toll Free Call with Unsuccessful Network Call Redirection

The next call scenario illustrated in **Figure 4** is similar to the previous call scenario, except that the redirection is unsuccessful. As a result, Avaya Aura™ Communication Manager "takes the call back" and continues vector processing. For example, the call may route to an agent, phone, or announcement after unsuccessful NCR.

1. Same as **Figure 2**.

2. Same as **Figure 2**.
3. The Verizon IP Toll Free service places a call to the target party (alternate destination), but the target party is busy or otherwise unavailable.
4. The Verizon IP Toll Free service notifies the redirecting/referring party (Avaya Aura™ Communication Manager) of the error condition.
5. Avaya Aura™ Communication Manager routes the call to a local agent, phone, or announcement.

**Note:** As noted in **Section 1.3**, except for egregious configuration errors, this "error handling" scenario could not be verified on the production Verizon circuit used for testing. For example, on the production circuit, Verizon sends a SIP BYE which terminates Communication Manager vector processing for the call when the alternate destination is busy. In cases where misconfiguration is introduced such that the Refer-To header is malformed or the REFER times out, Communication Manager can continue vector processing.

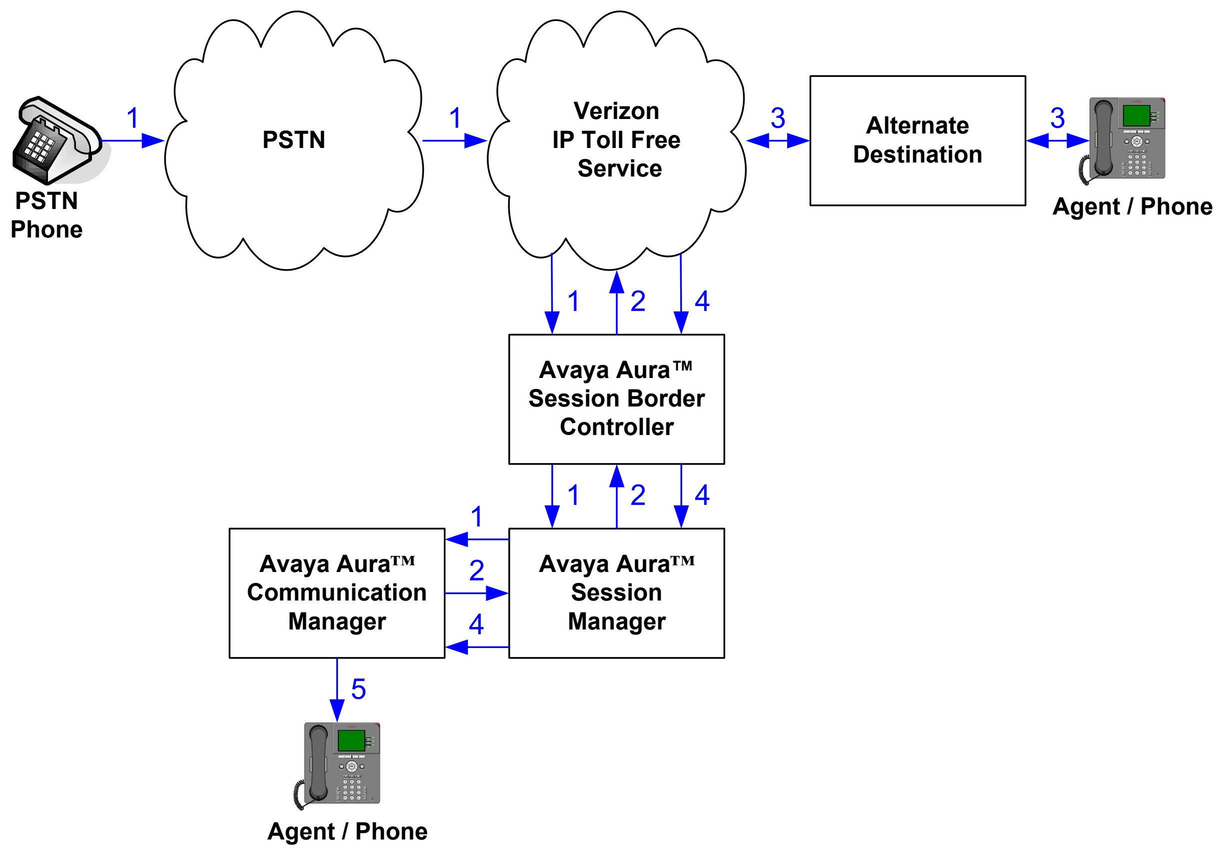



**Figure 4: Inbound Verizon Toll Free Call – Post-Answer SIP REFER Redirection Unsuccessful**

# 3. Equipment and Software Validated

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

| Equipment | Software |
|---|---|
| Avaya S8800 Server (Communication Manager) | Avaya Aura™ Communication Manager Release 6.0 load 345.0 Testing commenced with SP0, patch 18246, and concluded with SP1, patch 18444. |
| Avaya S8800 Server (System Manager) | Avaya Aura™ System Manager Release 6.0 |
| Avaya S8800 Server (Session Manager) | Avaya Aura™ Session Manager Release 6.0 (load 600020) |
| Avaya S8800 Server (Session Border Controller) | Avaya Aura™ Session Border Controller Release 6.0 SBC Template SBCT 6.0.0.1.4 |
| Avaya Modular Messaging (Application Server) | Avaya Modular Messaging (MAS) 5.2 Service Pack 3 Patch 1 |
| Avaya Modular Messaging (Storage Server) | Avaya Modular Messaging (MSS) 5.2, Build 5.2-11.0 |
| Avaya 9600-Series Telephones (H.323) | Release 3.1.1 – H.323 |
| Avaya 2400-Series and 6400-Series Digital Telephones | N/A |

**Table 1: Equipment and Software Used in the Sample Configuration**

**Note** - The solution integration validated in these Application Notes should be considered valid for deployment with Avaya Aura® Communication Manager release 6.0.1 and Avaya Aura® Session Manager release 6.1. Avaya agrees to provide service and support for the integration of Avaya Aura® Communication Manager release 6.0.1 and Avaya Aura® Session Manager release 6.1 with Verizon Business IP Contact Center service offer, in compliance with existing support agreements for Avaya Aura® Communication Manager release 6.0 and Avaya Aura® Session Manager 6.0, and in conformance with the integration guidelines as specified in this document.

# 4. Configure Avaya Aura™ Communication Manager Release 6

This section illustrates an example configuration allowing SIP signaling via the "Processor Ethernet" of the Avaya S8800 Server to Session Manager. In configurations that use an Avaya G650 Media Gateway, it is also possible to use an Avaya C-LAN in the Avaya G650 Media Gateway for SIP signaling to Session Manager.

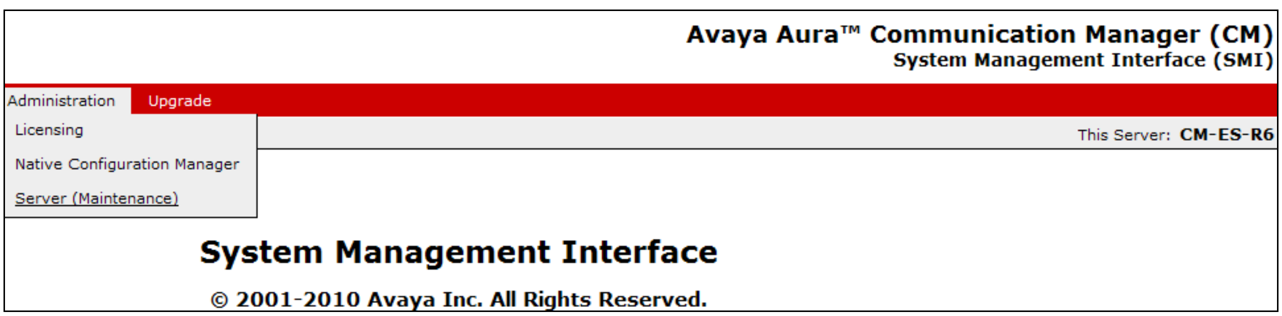
| |
|---|
| **Note** - The initial installation, configuration, and licensing of the Avaya servers and media gateways for Avaya Aura$^{TM}$ Communication Manager are assumed to have been previously completed and are not discussed in these Application Notes. |

Except for the web configuration shown in **Section 4.1**, all remaining configuration is performed via the Communication Manager SAT interface of the Avaya S8800 Server. Screens are abridged for brevity in presentation.

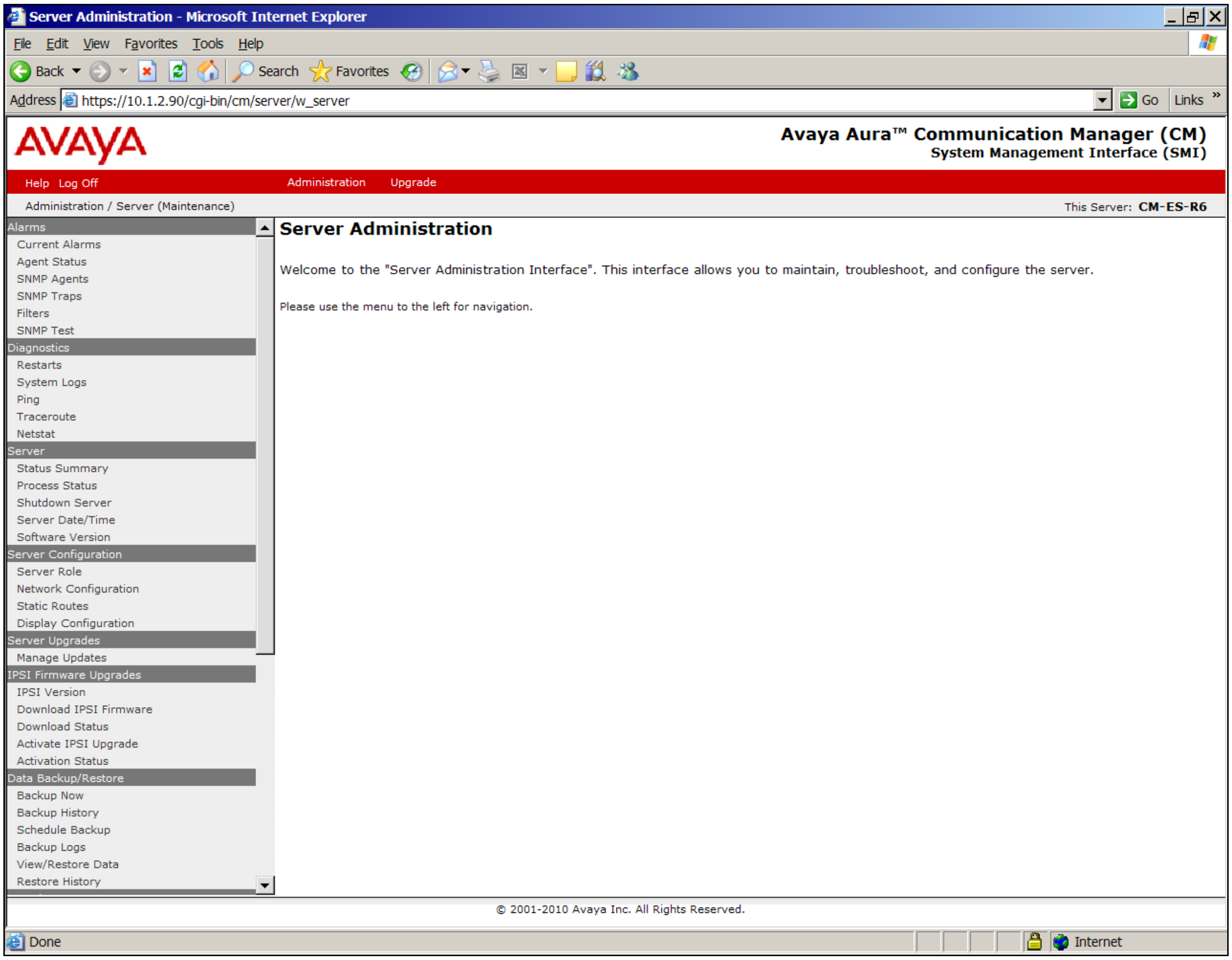
## 4.1. Processor Ethernet Configuration on S8800 Server

The screens in this section illustrate a previously completed configuration. Consult product documentation for further procedural guidance.

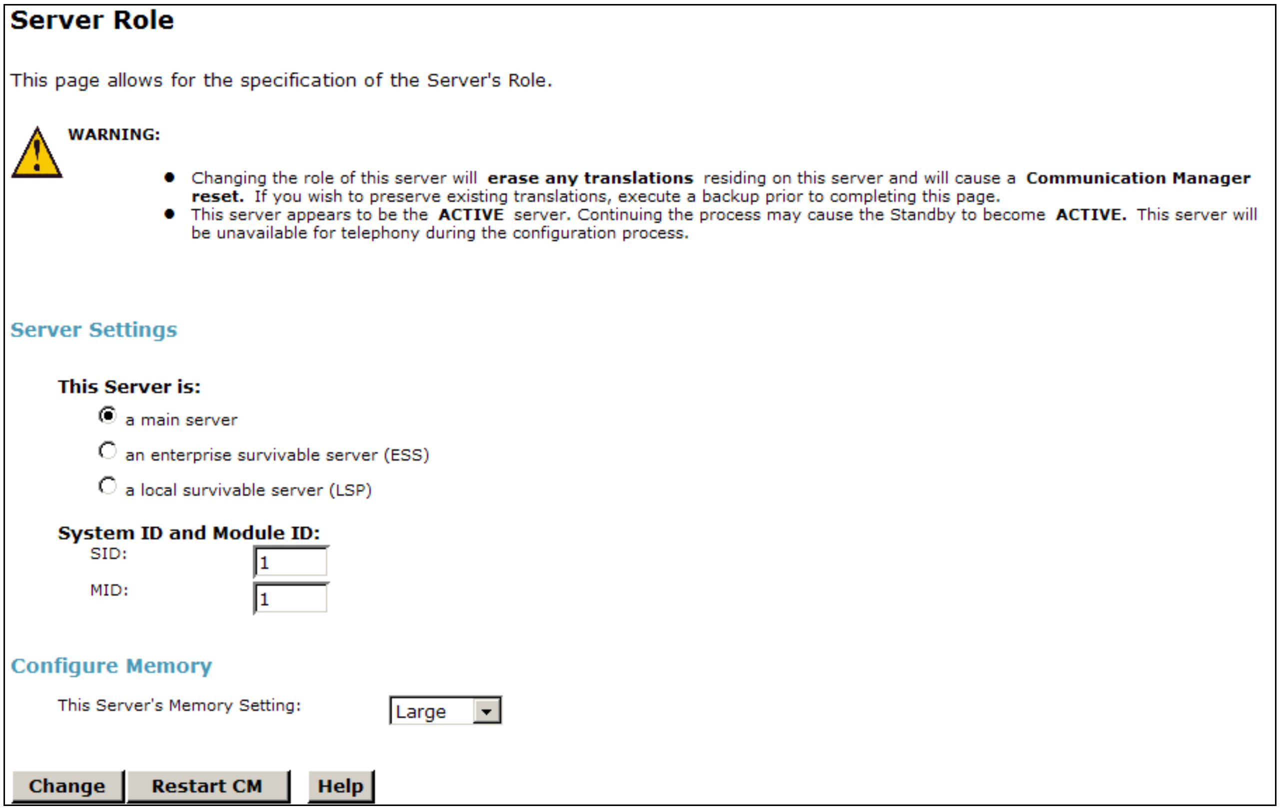
The S8800 Server can be accessed via a web interface in an internet browser. In the sample configuration, enter http://10.1.2.90 and log in with appropriate credentials (not shown). From the System Management Interface screen, select **Administration → Server (Maintenance)** as shown below.



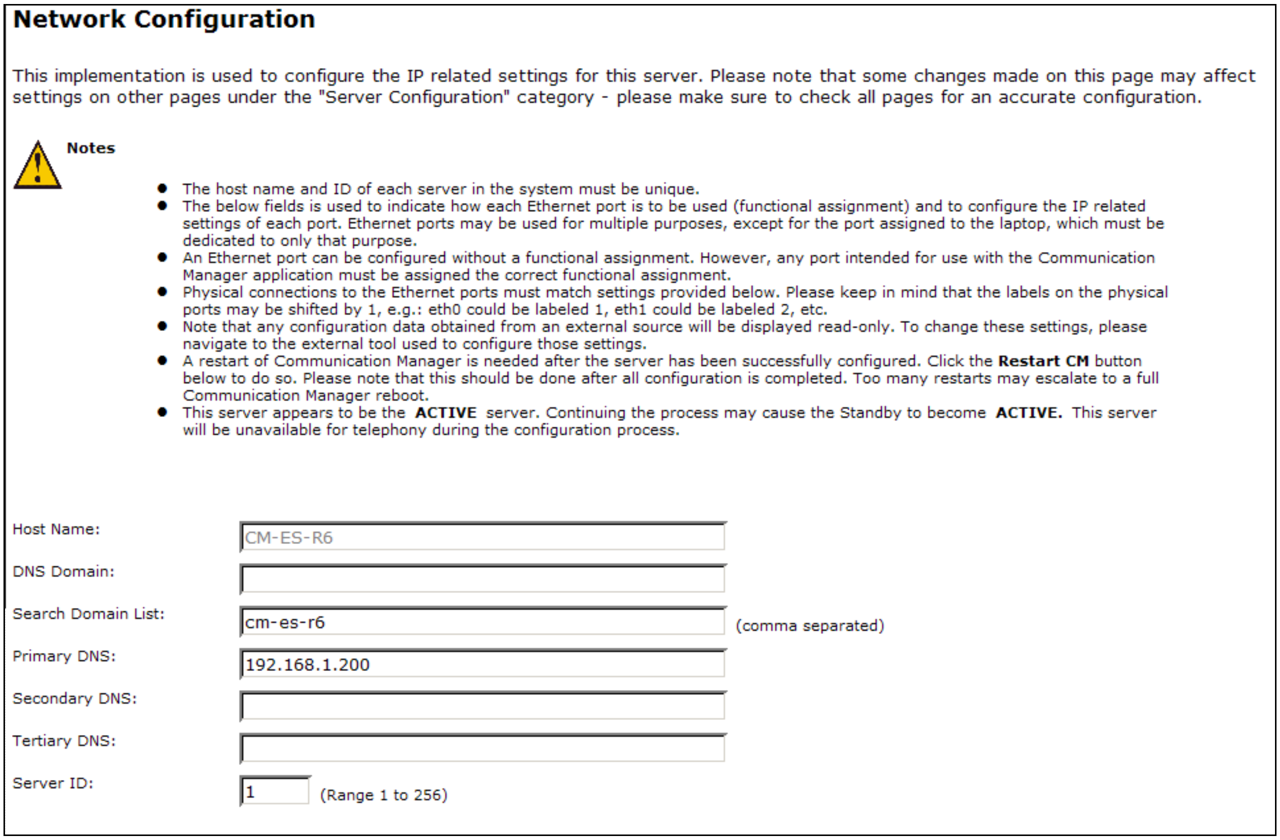
The resulting **Server Administration** screen is shown below.



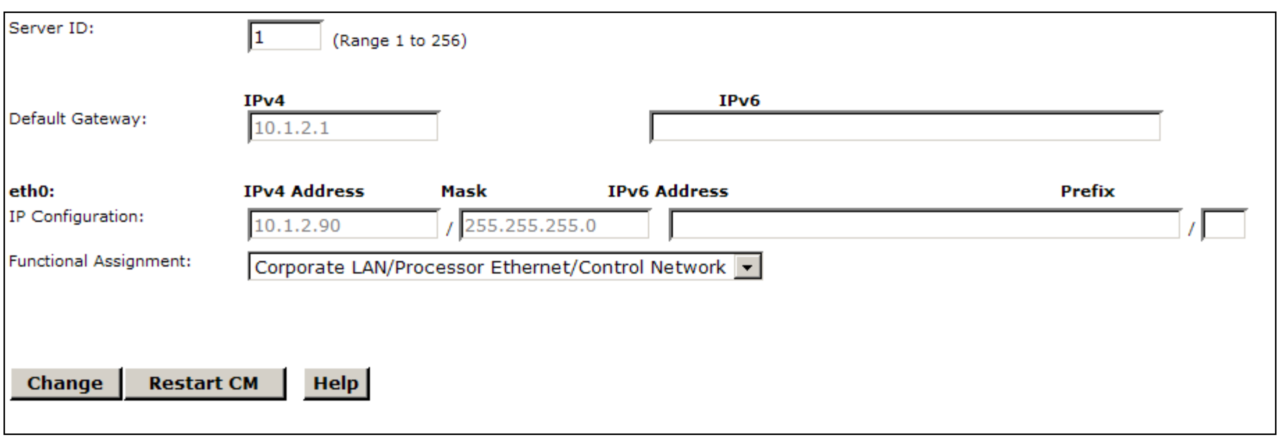
Under Server Configuration, select **Server Role** to view or configure the server role. In the sample configuration, the Avaya S8800 server is **a main server**, as shown below.



Under Server Configuration, select **Network Configuration** to view the network configuration. The following screen shows the upper portion of the **Network Configuration**.




Scrolling down, the following screen shows the lower portion of the Network Configuration. Note that the **IPv4 Address** of the server is 10.1.2.90, and that the **Functional Assignment** drop-down has assigned the **Corporate LAN/Processor Ethernet/Control Network** to the same "eth0" interface.



## 4.2. Verify Licensed Features

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

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

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

IP PORT CAPACITIES                                                USED
                     Maximum Administered H.323 Trunks: 12000 100
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 146
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   1
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0
```

On **Page 3** of the **System-Parameters Customer-Options** form, verify that **ARS** is enabled.

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

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

On **Page 4** of the **System-Parameters Customer-Options** form, verify that **IP Trunks**, **IP Stations**, and **ISDN-PRI** features are enabled. If the use of SIP REFER messaging will be required for the call flows as described in **Section 2.2**, verify that the **ISDN/SIP Network Call Redirection** feature is enabled.

```
display system-parameters customer-options                      Page   4 of  11
                                OPTIONAL FEATURES
   Emergency Access to Attendant? y                                IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                          ISDN Feature Plus? n
                  Enhanced EC500? y        ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                            ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                   ISDN-PRI? y
              ESS Administration? y                 Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                       Malicious Call Trace? y
     External Device Alarm Admin? y                   Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? y                   Multifrequency Signaling? y
      Global Call Classification? y         Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y      Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? y                Multimedia IP SIP Trunking? y
                       IP Trunks? y

           IP Attendant Consoles? y
```

On **Page 5** of the **System-Parameters Customer-Options** form, verify that the **Private Networking** and **Processor Ethernet** features are enabled if these features will be used, as is the case in the sample configuration.

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

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

On **Page 6** of the **System-Parameters Customer-Options** form, verify that any required call center features are enabled. In the sample configuration, vectoring is used to refer calls to alternate destinations using SIP NCR. Vector variables are used to include User-User Information (UUI) with the referred calls.

```
display system-parameters customer-options                      Page   6 of  11
                         CALL CENTER OPTIONAL FEATURES
                          Call Center Release: 5.0

                                    ACD? y                         Reason Codes? n
                           BCMS (Basic)? y             Service Level Maximizer? n
             BCMS/VuStats Service Level? n           Service Observing (Basic)? y
  BSR Local Treatment for IP & ISDN? n   Service Observing (Remote/By FAC)? n
                      Business Advocate? n           Service Observing (VDNs)? n
                        Call Work Codes? n                        Timed ACW? n
      DTMF Feedback Signals For VRU? n                  Vectoring (Basic)? y
                       Dynamic Advocate? n              Vectoring (Prompting)? y
       Expert Agent Selection (EAS)? y          Vectoring (G3V4 Enhanced)? y
                                EAS-PHD? y           Vectoring (3.0 Enhanced)? y
                       Forced ACD Calls? n   Vectoring (ANI/II-Digits Routing)? y
                   Least Occupied Agent? n   Vectoring (G3V4 Advanced Routing)? y
              Lookahead Interflow (LAI)? n                  Vectoring (CINFO)? n
Multiple Call Handling (On Request)? n    Vectoring (Best Service Routing)? y
    Multiple Call Handling (Forced)? n                  Vectoring (Holidays)? n
  PASTE (Display PBX Data on Phone)? n                  Vectoring (Variables)? y
```

## 4.3. Dial Plan

In the sample configuration, the Avaya CPE environment uses five digit local extensions, such as 3xxxx. Trunk Access Codes (TAC) are 3 digits in length and begin with 1. The Feature Access Code (FAC) to access ARS is the single digit 9. The Feature Access Code (FAC) to access AAR is the single digit 8. The dial plan illustrated here is not intended to be prescriptive; any valid dial plan may be used. The dial plan is modified with the **change dialplan analysis** command as shown below.

```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all          Percent Full: 2
    Dialed   Total  Call      Dialed   Total  Call      Dialed   Total  Call
    String   Length Type      String   Length Type      String   Length Type
   0           3   fac
   1           3   dac
   2           5   ext
   3           5   ext
   4           4   ext
   5           5   ext
   6           3   fac
   60          5   ext
   7           5   ext
   8           1   fac
   9           1   fac
   *           2   fac
   #           2   fac
```

## 4.4. Node Names

Node names are mappings of names to IP Addresses that can be used in various screens. The following abridged "change node-names ip" output shows relevant node-names in the sample configuration. As shown in bold, the node name for Avaya Aura™ Session Manager is "SM1" with IP Address 10.1.2.70. The node name and IP Address (10.1.2.90) for the Processor Ethernet "procr" appears automatically due to the web configuration in **Section 4.1**.

```
change node-names ip                                          Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
SM1                 10.1.2.70
procr               10.1.2.90
```

## 4.5. IP Interface for procr

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

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

                  Type: PROCR
                                                         Target socket load: 1700

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

                                 IPV4 PARAMETERS
             Node Name: procr                     IP Address: 10.1.2.90

           Subnet Mask: /24
```

## 4.6. Network Regions for Gateway, Telephones

Network regions provide a means to logically group resources. In the shared Communication Manager configuration used for the testing, the Avaya G450 Media Gateway is in region 1. To provide testing flexibility, network region 4 was associated with other components used specifically for the Verizon testing.

Non-IP telephones (e.g., analog, digital) derive network region and location configuration from the Avaya gateway to which the device is connected. The following display command shows that media gateway 1 is an Avaya G450 Media Gateway configured for network region 1. It can also be observed that the "Controller IP Address" is the Avaya S8800 processor Ethernet (10.1.2.90), and that the gateway IP Address is 10.1.2.95. These fields are not configured in this screen, but rather simply display the current information for the gateway.

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

                  Type: g450
                  Name: G450 Evolution Srvr
             Serial No: 08IS43202588
          Encrypt Link? y                  Enable CF? n
        Network Region: 1                   Location: 1
                                           Site Data:
         Recovery Rule: none

            Registered?  y
   FW Version/HW Vintage: 30 .13 .2  /1
      MGP IPV4 Address: 10.1.2.95
      MGP IPV6 Address:
   Controller IP Address: 10.1.2.90
           MAC Address: 00:1b:4f:03:57:b0
```

The following screen shows **Page 2** for media gateway 1. The gateway has an MM712 media module supporting Avaya digital phones in slot v3, an MM714 supporting analog devices in slot v5, 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      45   3
 V2:
 V3:   MM712                  DCP MM
 V4:
 V5:   MM714                  ANA MM
 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 following screen illustrates a subset of the IP network map configuration. 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 65.206.67.11 would be mapped to network region 4, based on the bold configuration below. In production environments, different sites will typically be on different networks, and ranges of IP Addresses assigned by the DHCP scope serving the site can be entered as one entry in the network map, to assign all telephones in a range to a specific network region.

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

                                                Subnet Network     Emergency
 IP Address                                     Bits   Region VLAN Location Ext
 ---------------------------------------------- ------ ------ ---- -------------
 FROM: 10.1.2.0                                 /24    1      n
   TO: 10.1.2.255
 FROM: 65.206.67.0                              /24    4      n
   TO: 65.206.67.255
```

The following screen shows IP Network Region 4 configuration. In the shared test environment, network region 4 is used to allow unique behaviors for the Verizon test environment. In this example, codec set 4 will be used for calls within region 4. The shared Avaya Interoperability Lab test environment uses the domain "avaya.com" (i.e., for network region 1 including the region of the processor ethernet "procr"). However, to illustrate the more typical case where the Communication Manager domain matches the enterprise CPE domain known to Verizon, the **Authoritative Domain** in the following screen is "adevc.avaya.globalipcom.com", the domain known to Verizon, as shown in **Figure 1**. Verizon supports domains that are longer than the maximum number of characters accepted by the **Authoritative Domain** field. If a domain is required that is longer than the maximum length of the **Authoritative Domain** field, a Session Manager adaptation can be used to manipulate the domain.

```
change ip-network-region 4                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 4
Location:          Authoritative Domain: adevc.avaya.globalipcom.com
    Name: Verizon testing
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 4              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                  RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

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

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

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

The following screen shows IP Network Region 1 configuration. In this example, codec set 1 will be used for calls within region 1 due to the **Codec Set** parameter on Page 1, but codec set 4 will be used for connections between region 1 and region 4 as noted previously. In the shared test environment, network region 1 was in place prior to adding the Verizon test environment and already used **Authoritative Domain** "avaya.com". Where necessary, Avaya Aura™ Session Manager or the SBC will adapt the domain.

```
change ip-network-region 1                                     Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: avaya.com
    Name: HQ CM and SIP Phones
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                   RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

The following screen shows the inter-network region connection configuration for region 1. The bold row shows that network region 1 is directly connected to network region 4, and that codec set 4 will be used for any connections between region 4 and region 1.

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

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

## 4.7. IP Codec Sets

The following screen shows the configuration for codec set 4, the codec set configured to be used for calls within region 4 and for calls between region 1 and region 4. In general, an IP codec set is a list of allowable codecs in priority order. Using the example configuration shown below, all calls with the PSTN via the SIP trunks would prefer to use G.729A, but also be capable of using G.711MU. Any calls using this same codec set that are between devices capable of the G.722-64K codec (e.g., Avaya 9600-Series IP Telephone) can use G.722. The specification of G.722 as the first choice is not required. That is, G.722 may be omitted from the codec set.

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

    Audio         Silence       Frames   Packet
    Codec         Suppression   Per Pkt  Size(ms)
 1: G.722-64K                     2        20
 2: G.729A             n          2        20
 3: G.711MU            n          2        20
 4:
 5:
 6:
 7:
```

On **Page 2** of the form:

- Configure the **Fax Mode** field to **off.** Verizon does not support T.38 fax.
- Configure the **Fax Redundancy** field to **0.**

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

                              Allow Direct-IP Multimedia? n

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

The following screen shows the configuration for codec set 1. The configuration for codec set 1 prefers G.711MU but also allows G.729A. Codec set 1 is used for Avaya Modular Messaging and other local Avaya CPE 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:
 4:
 5:
 6:
 7:
```

## 4.8. SIP Signaling Groups

This section illustrates the configuration of the SIP Signaling Groups. Each signaling group has a **Group Type** of “sip”, a **Near-end Node Name** of “procr”, and a **Far-end Node Name** of “SM1”. In the example screens, the **Transport Method** for all signaling groups is “tcp”. In production, TLS transport between Avaya Aura™ Communication Manager and Avaya Aura™ Session Manager can be used. The **Enable Layer 3 Test** field is enabled on each of the signaling groups to allow Communication Manager to maintain the signaling group using the SIP OPTIONS method. Fields that are not referenced in the text below can be left at default values, including **DTMF over IP** set to “rtp-payload”, which corresponds to RFC 2833.

The following screen shows signaling group 67. Signaling group 67 will be used for processing incoming calls from Verizon via Session Manager. The **Far-end Network Region** is configured to region 4. Port 5062 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 toll-free numbers to a route policy that uses a SIP entity link to Communication Manager specifying port 5062. The use of different ports is one means to allow Communication Manager to distinguish different types of calls arriving from the same Session Manager. In the sample configuration, the **Peer Detection Enabled** field was set to “n”. Other parameters may be left at default values.

```
change signaling-group 67                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 67             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                             SIP Enabled LSP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? n  Peer Server: Others

   Near-end Node Name: procr                  Far-end Node Name: SM1
 Near-end Listen Port: 5062                 Far-end Listen Port: 5062
                                         Far-end Network Region: 4

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

The following screen shows signaling group 60, the signaling group to Session Manager that was in place prior to adding the Verizon configuration to the shared Avaya Interoperability lab configuration. This signaling group reflects configuration not specifically related to Verizon trunking. For example, calls using Avaya SIP Telephones and calls routed to other Avaya applications, such as Avaya Modular Messaging, use this signaling group. Again, the **Near-end Node Name** is “procr” and the **Far-end Node Name** is “SM1”, the node name of the Session Manager. Unlike the signaling group used for the Verizon signaling, the **Far-end Network Region** is 1. The **Peer Detection Enabled** field is set to “y” and a peer Session Manager (SM) has been previously detected. The **Far-end Domain** is set to “avaya.com” matching the configuration in place prior to adding the Verizon SIP Trunking configuration.

```
change signaling-group 60                                       Page   1 of   1
                                SIGNALING GROUP

 Group Number: 60             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                           SIP Enabled LSP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM

   Near-end Node Name: procr                 Far-end Node Name: SM1
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 1

Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 10
```

## 4.9. SIP Trunk Groups

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

**NOTE:** For Verizon Business customers utilizing either Verizon **IP Contact Center** or **IP-IVR** service offers, at least one **Elite Agent license is required** to support the ability to utilize the Network Call Redirection capabilities of those services with Communication Manager. This license is required to enable the "**ISDN/SIP Network Call Redirection"** feature. This licensed feature must be turned **ON** (as shown in **Section 4.2**) to support Network Call Redirection.

The following shows **Page 1** for trunk group 67, which will be used for incoming toll-free calls from Verizon. The **Number of Members** field defines how many simultaneous calls are permitted for the trunk group. The **Service Type** field should be set to "public-ntwrk" for the trunks that will handle calls with Verizon. Although not strictly necessary, the **Direction** has been configured to "incoming" to emphasize that trunk group 67 is used for incoming calls only in the sample configuration.

```
change trunk-group 67                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 67                   Group Type: sip           CDR Reports: y
  Group Name: From-SM-Aura-VZ             COR: 1        TN: 1        TAC: 167
   Direction: incoming      Outgoing Display? n
 Dial Access? n                                 Night Service:

Service Type: public-ntwrk          Auth Code? n
                                                     Signaling Group: 67
                                                   Number of Members: 6
```

The following screen shows **Page 2** for trunk group 67. 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 Avaya Aura™ Communication Manager default value, which can result in unnecessary SIP messages to re-establish a higher refresh interval for each call.

```
change trunk-group 67                                              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

                                     Delay Call Setup When Accessed Via IGAR? n
```

The following screen shows **Page 3** for trunk group 67. All parameters except those in bold are default values. Optionally, replacement text strings can be configured using the "system-parameters features" screen, such that incoming "private" (anonymous) or "restricted" calls can display an Avaya-configured text string on called party telephones.

```
change trunk-group 67                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? y
 Show ANSWERED BY on Display? y
```

The following screen shows **Page 4** for trunk group 67. The **PROTOCOL VARIATIONS** page is one reason why it can be advantageous to configure incoming calls from Verizon to arrive on specific signaling groups and trunk groups. The bold fields have non-default values. The **Convert 180 to 183 for Early Media** field is new in Communication Manager Release 6. Verizon recommends that inbound calls to the enterprise result in a 183 with SDP rather than a 180 with SDP, and setting this field to "y" for the trunk group handling inbound calls from Verizon produces this result. Although not strictly necessary, the **Telephone Event Payload Type** has been set to 101 to match Verizon configuration. Setting the **Network Call Redirection** flag to "y" enables advanced services associated with the use of the REFER message, while also implicitly enabling Communication Manager to signal "sendonly" media conditions for calls placed on hold at the enterprise site. If neither REFER signaling nor "sendonly" media signaling is required, this field may be left at the default "n" value. In the testing associated with these Application Notes, the **Network Call Redirection** flag was set to "y" to allow REFER to be exercised.

The Verizon IPCC Services do not support the Diversion header or the History-Info header, and therefore both **Support Request History** and **Send Diversion Header** are set to "n".

```
change trunk-group 67                                           Page   4 of  21
                              PROTOCOL VARIATIONS

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

       Convert 180 to 183 for Early Media? y
 Always Use re-INVITE for Display Updates? n
                             Enable Q-SIP? n
```

The following screen shows **Page 1** for trunk group 60, 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 for communication with other Avaya applications, such as Avaya Modular Messaging, and does not reflect any unique Verizon configuration.

```
change trunk-group 60                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 60                   Group Type: sip           CDR Reports: y
  Group Name: SM1                         COR: 1        TN: 1        TAC: 160
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                                      Signaling Group: 60
                                                    Number of Members: 100
```

The following screen shows **Page 3** for trunk group 60. Note that unlike the trunks associated with Verizon calls that use "public" numbering, this tie trunk group uses a "private" **Numbering Format**.

```
change trunk-group 60                                           Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? y
```

The following screen shows **Page 4** for trunk group 60. 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 Avaya Modular Messaging.

```
change trunk-group 60                                           Page   4 of  21
                              PROTOCOL VARIATIONS

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

        Convert 180 to 183 for Early Media? n
  Always Use re-INVITE for Display Updates? n
                                Enable Q-SIP? n
```

## 4.10. Vector Directory Numbers (VDNs) and Vectors for SIP NCR

This section describes the basic commands used to configure Vector Directory Numbers (VDNs) and corresponding vectors. These vectors contain steps that invoke the Avaya Aura™ Communication Manager SIP Network Call Redirection (NCR) functionality. These Application Notes provide rudimentary vector definitions to demonstrate and test the SIP NCR and UUI functionalities. In general, call centers will use vector functionality that is more complex and tailored to individual needs. Call centers may also use customer hosts running applications used in conjunction with Avaya Aura™ Application Enablement Services (AES) to define call routing and provide associated UUI. The definition and documentation of those complex applications and associated vectors are beyond the scope of these Application Notes.

### 4.10.1 Post-Answer Redirection to a PSTN Destination

This section provides an example configuration of a vector that will use post-answer redirection to a PSTN destination. A corresponding detailed verification is provided in **Section 9.1.2**. In this example, the inbound toll-free call is routed to VDN 36998 shown in the following screen. The originally dialed Verizon IP Toll Free number may be mapped to VDN 36998 by Session Manager digit conversion, or via the incoming call handling treatment for the inbound trunk group.

```
display vdn 36998                                           Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                                Extension: 36998
                                    Name*: Refer-Vector
                              Destination: Vector Number        3
                      Attendant Vectoring? n
                     Meet-me Conferencing? n
                       Allow VDN Override? n
                                      COR: 1
```

VDN 36998 is associated with vector 3, which is shown below. Vector 3 plays an announcement (step 3) to answer the call. After the announcement, the "route-to number" (step 5) includes "~r+17326870755" where the number 732-687-0755 is a PSTN destination. This step causes a REFER message to be sent where the Refer-To header includes "+17326870755" as the user portion. Note that Verizon IP Contact Center services require the "+" in the Refer-To header for this type of call redirection.

```
display vector 3                                                Page   1 of   6
                                  CALL VECTOR

    Number: 3                Name: Refer-to-PSTN
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 #    Play announcement which answers call
03 announcement 36997
04 #    Refer the call to PSTN destination
05 route-to     number ~r+17326870755   with cov n if unconditionally
06 #    If Refer fails, play announcement and disconnect
07 disconnect   after announcement 36996
```

### 4.10.2 Post-Answer Redirection With UUI to a SIP Destination

This section provides an example of post-answer redirection with UUI passed to a SIP destination. A corresponding detailed verification is provided in **Section 9.1.3**. In this example, the inbound call is routed to VDN 36990 shown in the following screen. The originally dialed Verizon toll-free number may be mapped to VDN 36990 by Session Manager digit conversion, or via the incoming call handling treatment for the inbound trunk group.

```
display vdn 36990                                               Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 36990
                                 Name*: Refer-Vector-with-UUI
                           Destination: Vector Number        5
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
```

To facilitate testing of NCR with UUI, the following vector variables were defined.

```
change variables                                                Page   1 of  39
                              VARIABLES FOR VECTORS

Var Description                      Type    Scope Length Start Assignment       VAC
A   Test1                            asaiuui L     16     1
B   Test2                            asaiuui L     16     17
C   Test3                            asaiuui L     16     33
D   Test4                            asaiuui L     16     49
E   Test5                            asaiuui L     16     65
F   Test6                            asaiuui L     16     81
```

VDN 36990 is associated with vector 5, which is shown below. Vector 5 sets data in the vector variables A-F (step 1-6) and plays an announcement to answer the call (step 11). After the announcement, the "route-to" number step includes "~r+18668512649". This step causes a REFER message to be sent where the Refer-To header includes "+18668512649" as the user portion. The Refer-To header will also contain the UUI set in variables A-F. Verizon will include this UUI in the INVITE ultimately sent to the SIP-connected target of the Refer, which is toll-free number "18668512649". In the sample configuration, where only one location was used, 866-851-2649 is another toll-free number assigned to the same circuit as the original call. In practice, NCR with UUI allows Communication Manager to send call or customer-related data along with the call to another contact center.

```
display vector 5                                                Page   1 of   6
                                  CALL VECTOR

    Number: 5                Name: Refer-with-UUI
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 set          A      = none   CATR  1234567890123456
02 set          B      = none   CATR  7890123456789012
03 set          C      = none   CATR  3456789012345678
04 set          D      = none   CATR  9012345678901234
05 set          E      = none   CATR  5678901234567890
06 set          F      = none   CATR  1234567890123456
07
08
09
10 wait-time    2   secs hearing silence
11 announcement 36997
12 route-to     number ~r+18668512649   with cov n if unconditionally
13 disconnect   after announcement 36996
```

## 4.11. Public Numbering

The "change public-unknown-numbering" command may be used to define the format of numbers sent to Verizon in SIP headers.

In the first bolded row shown in the example abridged output below, a specific Communication Manager extension (x30002) is mapped to a Verizon toll-free number (866-851-2649), when the call uses trunk group 67. In the course of the testing, multiple Verizon toll-free numbers were associated with different Communication Manager extensions and functions in this fashion.

In the other bolded rows shown in the example abridged output below, entries are made for the specific Communication Manager Vector Directory Numbers (VDN) illustrated in the prior section. Making an entry such as this for each VDN will avoid unnecessary SIP messaging for toll-free calls to VDNs that use SIP NCR with REFER, as summarized in **Section 1.3**.

```
change public-unknown-numbering 5                                  Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext            Trk          CPN          CPN
Len Code           Grp(s)       Prefix       Len
                                                   Total Administered: 3
 5  3              60                        5        Maximum Entries: 9999
 5  30002          67           8668512649   10
 5  36990          67                        5
 5  36998          67                        5
```

## 4.12. Incoming Call Handling Treatment for Incoming Calls

In general, the "incoming call handling treatment" for a trunk group can be used to manipulate the digits received for an incoming call if necessary. Since Avaya Aura™ Session Manager is present, Session Manager can be used to perform digit conversion, and digit manipulation via the Communication Manager incoming call handling table may not be necessary. If the toll-free number sent by Verizon is unchanged by Session Manager, then the 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 toll-free number 8668512649 to extension 30002.

```
change inc-call-handling-trmt trunk-group 67                       Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature         Len      Digits
 public-ntwrk    10 8668512649      all 30002
```

## 4.13. Modular Messaging Hunt Group

Although not specifically related to Verizon, this section shows the hunt group used for access to Avaya Modular Messaging. In the sample configuration, users with voice mail have a coverage path containing hunt group 60. Users can dial extension 33000 to reach Modular Messaging (e.g., for message retrieval). The following screen shows Page 1 of hunt-group 60.

```
display hunt-group 60                                               Page   1 of  60
                                  HUNT GROUP

            Group Number: 60                              ACD? n
              Group Name: MM Coverage                   Queue? n
         Group Extension: 33000                        Vector? n
              Group Type: ucd-mia               Coverage Path:
                      TN: 1          Night Service Destination:
                     COR: 1                   MM Early Answer? n
           Security Code:               Local Agent Preference? n
 ISDN/SIP Caller Display: mbr-name
```

The following screen shows **Page 2** of hunt-group 60, which routes to the AAR access code 8 and **Voice Mail Number** 33000.

```
display hunt-group 60                                              Page   2 of  60
                                 HUNT GROUP
                     Message Center: sip-adjunct

     Voice Mail Number         Voice Mail Handle          Routing Digits
                                                   (e.g., AAR/ARS Access Code)
     33000                     33000                      8
```

## 4.14. AAR Routing to Modular Messaging via Session Manager

Although not specifically related to Verizon, this section shows the AAR routing for the number used in the hunt group in the previous section. The bold row shows that calls to the number range 33xxx, which includes the Modular Messaging hunt group 33000, will use **Route Pattern** 60. As can be observed from the other rows, various other dial strings also route to other internal destinations (i.e., not to Verizon) via route pattern 60.

```
change aar analysis 0                                             Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 0

          Dialed           Total     Route     Call   Node  ANI
          String         Min  Max   Pattern    Type   Num   Reqd
    32                   5    5      60        unku         n
    33                   5    5      60        unku         n
    3400                 5    5      60        unku         n
```

## 4.15. Uniform Dial Plan (UDP) Configuration

Although not specifically related to Verizon, this section shows the UDP configuration, with the bold row showing the calls of the form 33xxx will be routed via AAR.

```
change uniform-dialplan 3                                         Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                                Percent Full: 0
  Matching                    Insert              Node
  Pattern       Len Del       Digits      Net Conv Num

 31              5   0                    aar  n
 3100            5   0                    aar  n
 33              5   0                    aar  n
 3400            5   0                    aar  n
```

## 4.16. Route Pattern for Internal Calls via Session Manager

Although not specifically related to Verizon, this section shows the AAR routing for the number used in the hunt group for Modular Messaging. Route pattern 60 contains trunk group 60, the "private" tie trunk group to Session Manager.

```
change route-pattern 60                                         Page   1 of   3
                    Pattern Number: 60  Pattern Name: SM FS
                             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: 60   0                    0                                       n   user
 2:                                                                   n   user
 3:                                                                   n   user
 4:                                                                   n   user
 5:                                                                   n   user
 6:                                                                   n   user

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

## 4.17. Private Numbering

Although not specifically related to Verizon, this section shows the private numbering configuration associated with the calls using trunk group 60. The bold row configures any five digit number beginning with 3 (i.e., 3xxxx) that uses trunk group 60 to retain the original 5 digit number (i.e., no digit manipulation is specified, and the **Total Len** is 5).

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

Ext Ext              Trk        Private           Total
Len Code             Grp(s)     Prefix            Len
 5  2                                             5     Total Administered: 5
 5  3                60                           5        Maximum Entries: 540
 5  4                                             5
 5  5                                             5
```

## 4.18. Avaya Aura™ Communication Manager Stations

In the sample configuration, five digit station extensions were used with the format 3xxxx. The following abbreviated screen shows an example extension for an Avaya H.323 IP telephone. Coverage path 60 is assigned to give this user coverage to Avaya Modular Messaging.

```
change station 30002                                            Page   1 of   5
                                     STATION
Extension: 30002                         Lock Messages? n              BCC: 0
     Type: 9620                          Security Code: *               TN: 1
     Port: S00038                      Coverage Path 1: 60             COR: 1
     Name: Joey Votto                  Coverage Path 2:                COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 30002
```

On **Page 2**, the **MWI Served User Type** is set to "sip-adjunct" for the SIP integration to Avaya Modular Messaging.

```
change station 30002                                            Page   2 of   5
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe             Auto Select Any Idle Appearance? n
          LWC Activation? y                        Coverage Msg Retrieval? y
  LWC Log External Calls? n                                   Auto Answer:
none
             CDR Privacy? n                              Data Restriction? n
   Redirect Notification? y                    Idle Appearance Preference? n
 Per Button Ring Control? n                  Bridged Idle Line Preference? n
   Bridged Call Alerting? n                      Restrict Last Appearance? y
  Active Station Ringing: single
                                                        EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed                       EC500 State: enabled
         Multimedia Mode: enhanced                 Audible Message Waiting? n
    MWI Served User Type: sip-adjunct           Display Client Redirection? n
                                               Select Last Used Appearance? n
                                                 Coverage After Forwarding? s
                                                   Multimedia Early Answer? n
                                              Direct IP-IP Audio Connections? y
  Emergency Location Ext: 30002         Always Use? n IP Audio Hairpinning? n
```

## 4.19. Coverage Path

This section illustrates an example coverage path for a station with a mailbox on Avaya Modular Messaging.   Hunt group 60, the hunt group to Modular Messaging, is **Point1** in coverage path 60.

```
change coverage path 60                                         Page   1 of   1
                                 COVERAGE PATH
                   Coverage Path Number: 60
     Cvg Enabled for VDN Route-To Party? y        Hunt after Coverage? n
                       Next Path Number:          Linkage
COVERAGE CRITERIA
    Station/Group Status    Inside Call     Outside Call
             Active?             n               n
               Busy?             y               y
       Don't Answer?             y               y        Number of Rings: 2
                All?             n               n
 DND/SAC/Goto Cover?             y               y
   Holiday Coverage?             n               n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h60            Rng:    Point2:
  Point3:                        Point4:
  Point5:                        Point6:
```

## 4.20. Saving Communication Manager Configuration Changes

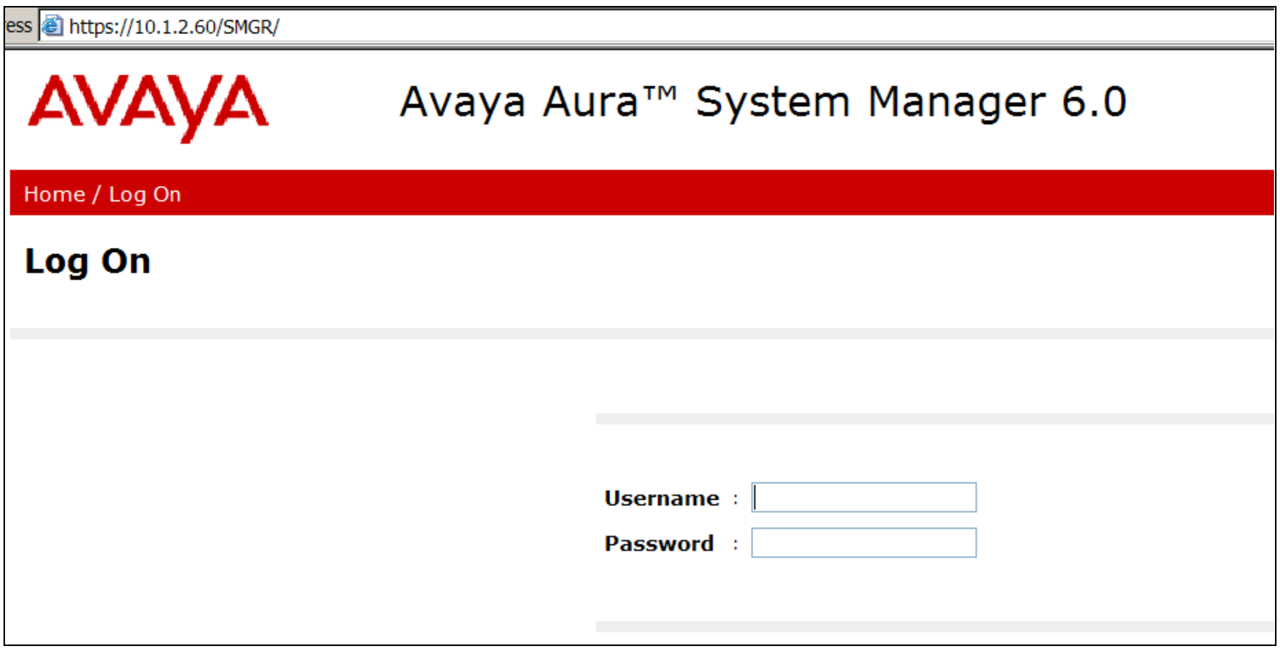
The command "save translation all" can be used to save the configuration.

# 5. Avaya Aura™ Session Manager Provisioning

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

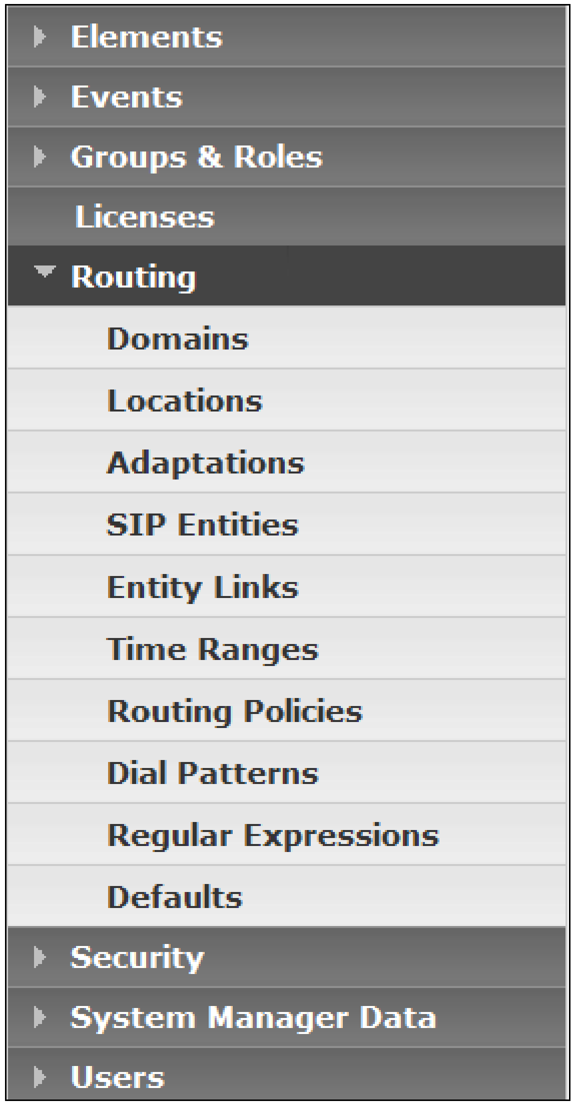
| |
|---|
| **Note** – The following sections assume that Avaya Aura™ Session Manager and Avaya Aura™ System Manager have been installed and that network connectivity exists between the two. For more information on Avaya Aura™ Session Manager see **[3, 4]**. |

Session Manager is managed via Avaya Aura™ System Manager. Using a web browser, access "https://<ip-addr of System Manager>/SMGR". In the **Log On** screen, enter appropriate **Username** 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.

For readers familiar with prior releases of Session Manager, the configurable items under **Routing** in Release 6 were located under a heading called **Network Routing Policy** in prior releases. Select **Routing.** The screen shown below shows the various sub-headings.



When Routing is selected, the right side 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 (NRP)** 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"

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

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"

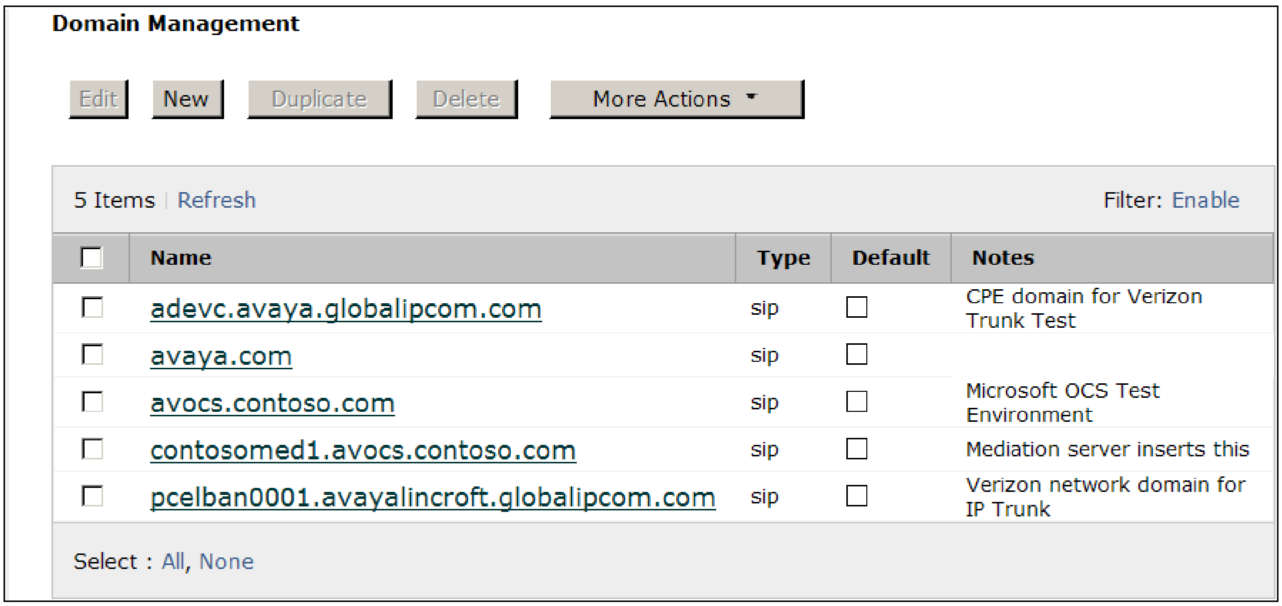
- Assign the appropriate "Routing Policies" to the "Regular Expressions"

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

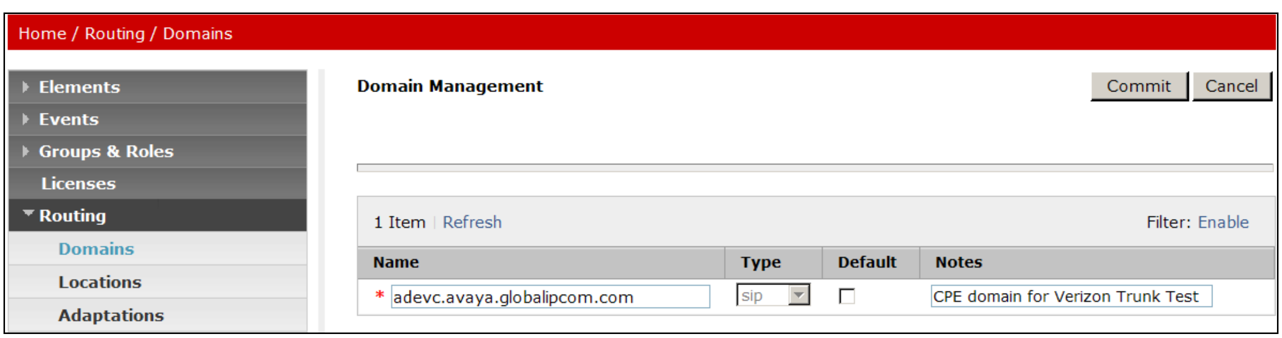
## 5.1. Domains

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

The following screen shows a list of configured SIP domains. The Session Manager used in the verification of these Application Notes was shared among many Avaya interoperability test efforts. The domain "avaya.com" was already being used for communication among a number of Avaya systems and applications, including an Avaya Modular Messaging system with SIP integration to Session Manager. The domain "avaya.com" is not known to the Verizon production service.

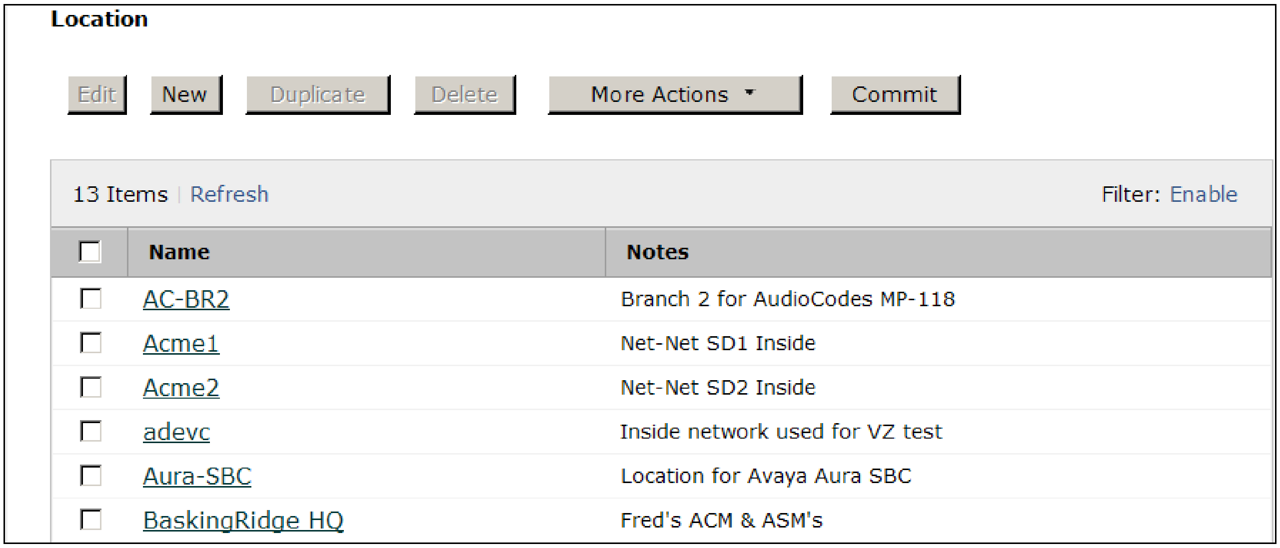



The domain "adevc.avaya.globalipcom.com" is the domain known to Verizon as the enterprise SIP domain. In the sample configuration, Verizon included this domain as the host portion of the Request-URI for inbound toll-free calls.



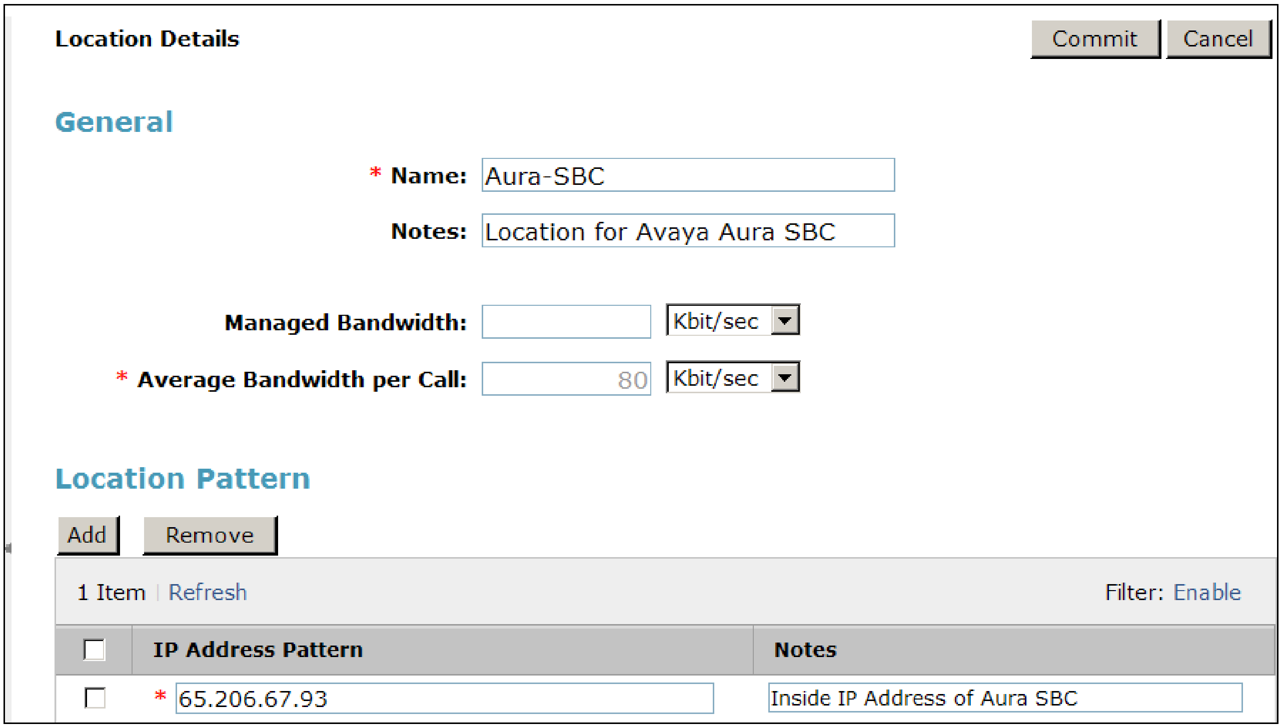
## 5.2. Locations

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

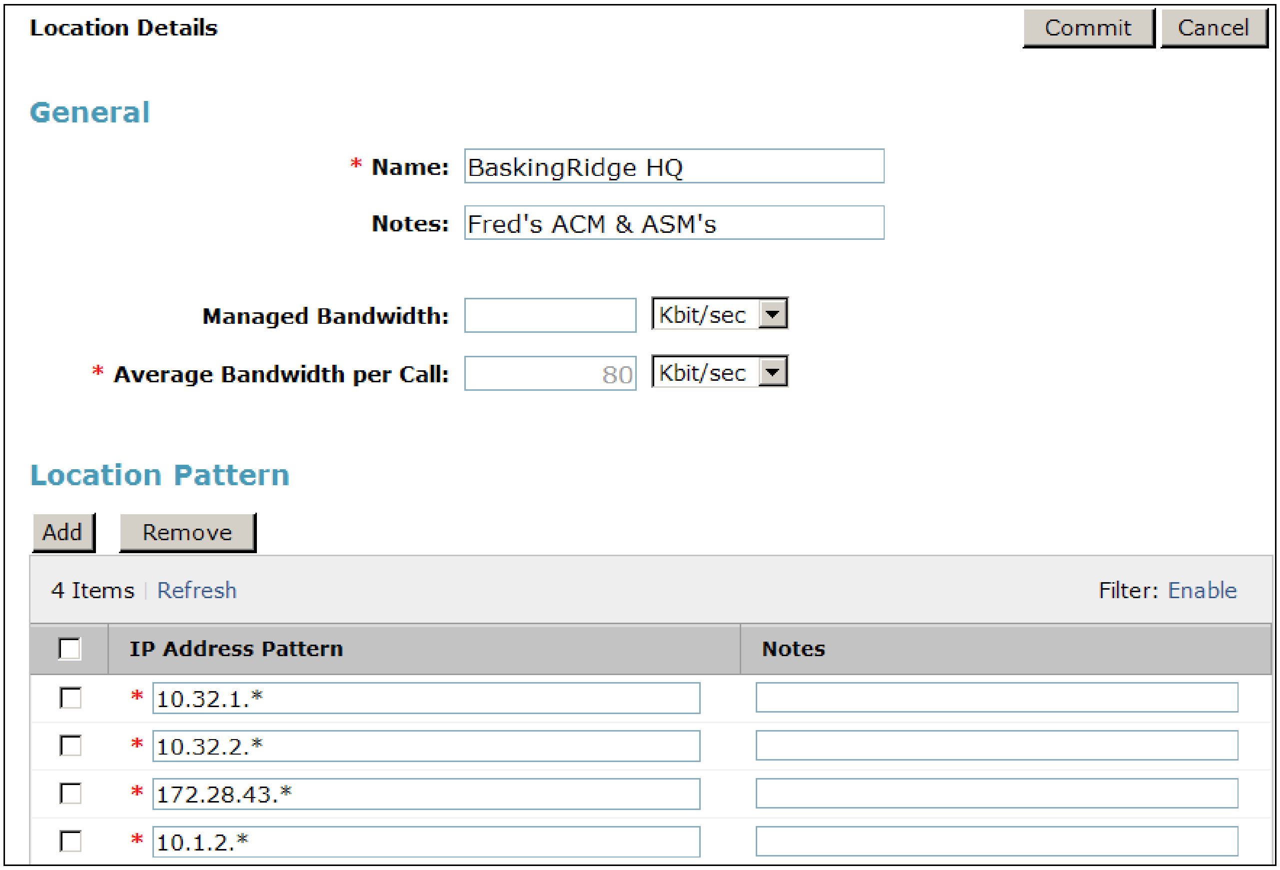



| Name | Notes |
|---|---|
| AC-BR2 | Branch 2 for AudioCodes MP-118 |
| Acme1 | Net-Net SD1 Inside |
| Acme2 | Net-Net SD2 Inside |
| adevc | Inside network used for VZ test |
| Aura-SBC | Location for Avaya Aura SBC |
| BaskingRidge HQ | Fred's ACM & ASM's |

The following screen shows the location details for the location named “Aura-SBC”, corresponding to the Avaya Aura™ Session Border Controller. Later, the location with name “Aura-SBC” will be assigned to the corresponding SIP Entity. The IP Address 65.206.67.93 of the inside (private) interface of the SBC is entered in the **IP Address Pattern** field.



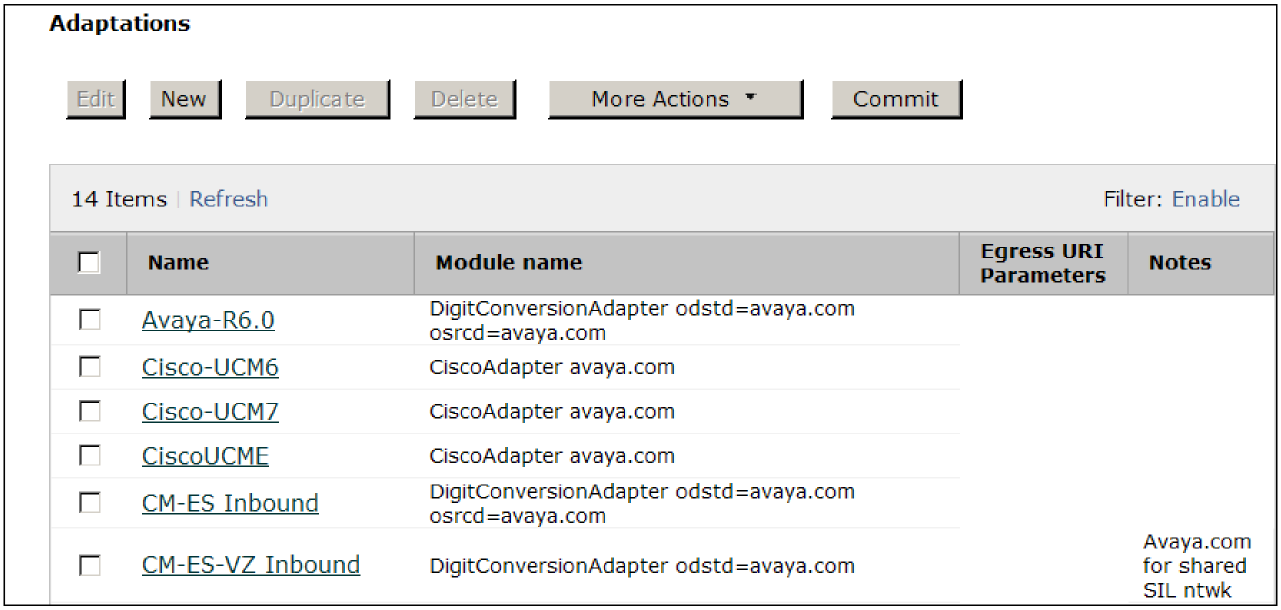
The following screen shows the location details for the location named "BaskingRidgeHQ". The SIP Entities and associated IP Addresses for this location correspond to the shared components of the Avaya Interoperability Lab test environment, such as Avaya Aura™ Communication Manager Release 6, Avaya Aura™ Session Manager Release 6, and Avaya Modular Messaging servers.



## 5.3. Adaptations

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

The following screen shows a portion of the list of adaptations in the sample configuration.

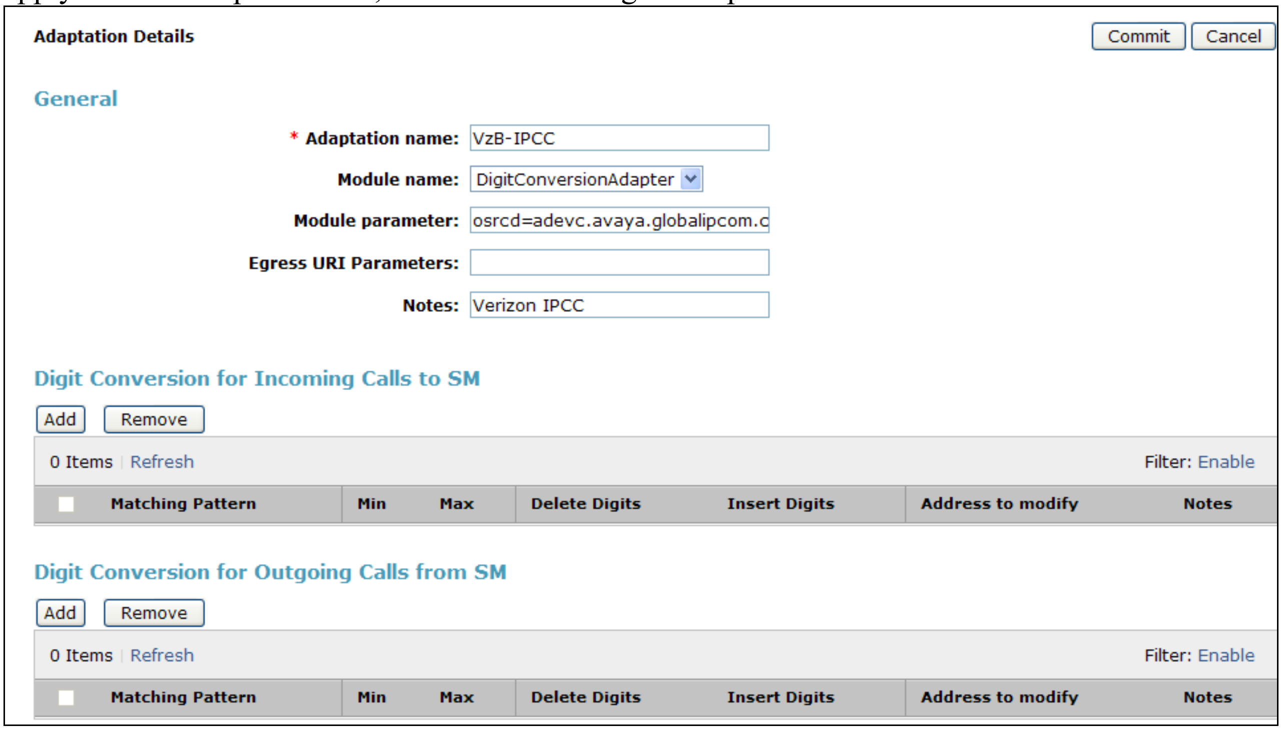



After scrolling down, the following screen shows another portion of the list of adaptations in the sample configuration.

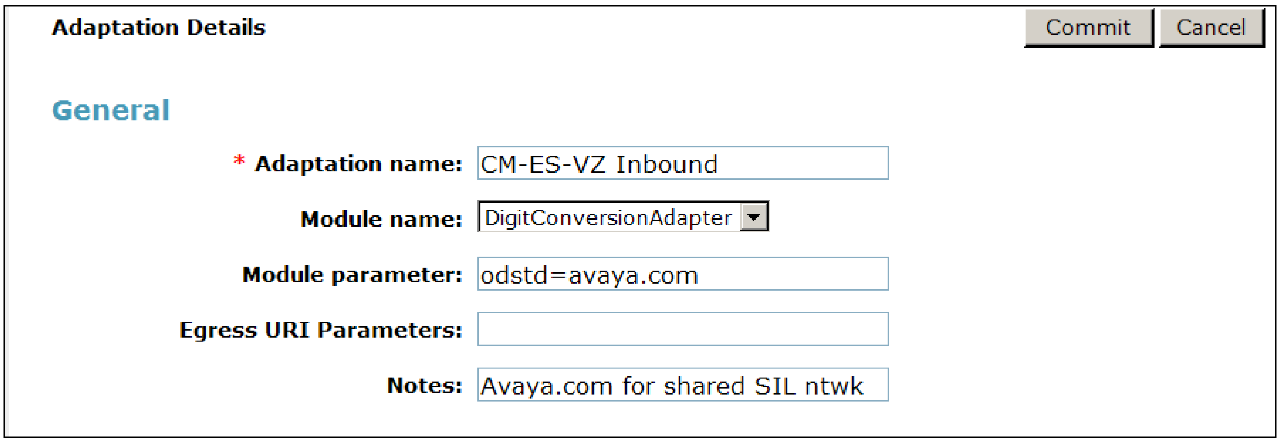

The adapter named "VzB-IPCC" shown above will later be assigned to the SBC SIP Entity. The adapter is configured to apply two parameters:

- "osrcd=adevc.avaya.globalipcom.com". This configuration enables the source domain to be overwritten with "adevc.avaya.globalipcom.com". For example, for inbound toll-free calls from Verizon, the PAI header sent to Verizon in the 200 OK will contain "adevc.avaya.globalipcom.com". Depending on the Communication Manager configuration, it may not be necessary for Session Manager to adapt the domain in this fashion. In the sample configuration, where "avaya.com" was already in use in a shared Avaya environment, it was appropriate for Session Manager to adapt the domain from "avaya.com" to "adevc.avaya.globalipcom.com" where the latter is the CPE domain known to Verizon.
- "odstd=172.30.205.55" This configuration enables the destination domain to be overwritten with "172.30.205.55", the Verizon IPCC service node IP Address. The similar configuration including rationale is provided in **Section 4.3.2.2** of reference [JF-VZIPCC].

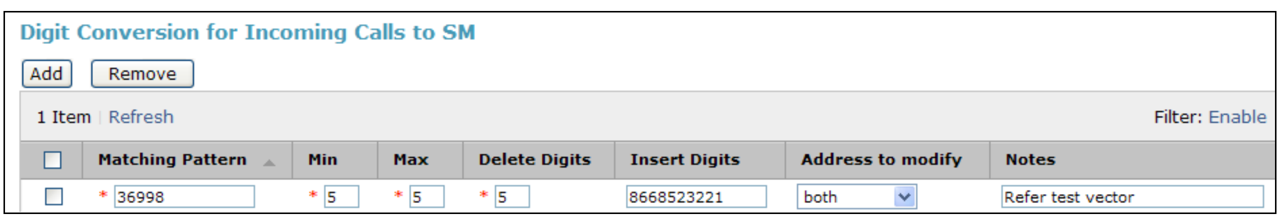
The following screen shows the complete adaptation details. Although the "DigitConversionAdapter" is used, no conversion of digits is required. The adapter is used to apply the module parameters, and not for true digit manipulation.



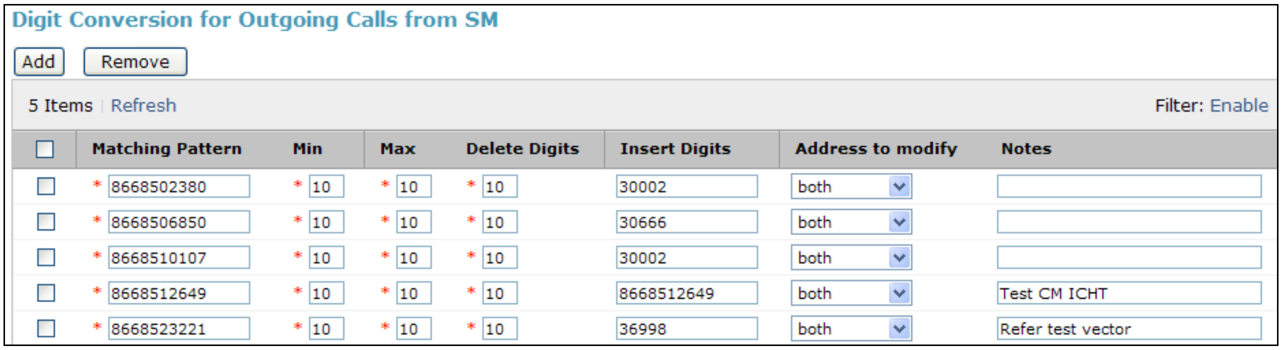
The adapter named "CM-ES-VZ Inbound" shown below will later be assigned to the SIP Entity linking Session Manager to Communication Manager for calls involving Verizon. This adaptation uses the "DigitConversionAdapter" and specifies the "odstd=avaya.com" parameter to adapt the domain to the domain expected by Communication Manager in the sample configuration. More specifically, this configuration enables the destination domain to be overwritten with "avaya.com" for calls that egress to a SIP entity using this adapter. For example, for inbound toll-free calls from Verizon to the Avaya CPE, the Request-URI header sent to Communication Manager will contain "avaya.com" as expected by Communication Manager in the shared Avaya Interoperability Lab configuration. 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-ES-VZ Inbound" adapter that can be used to convert digits between the extension numbers used on Communication Manager and the toll-free numbers assigned by Verizon. An example portion of the settings for "Digit Conversion for Incoming Calls to SM" is shown below.




An example portion of the settings for "Digit Conversion for Outgoing Calls from SM" (i.e., inbound to Communication Manager) is shown below. During the testing, the digit conversion was varied to allow the same toll-free number to be used to test different Communication Manager call destinations.



**Digit Conversion for Outgoing Calls from SM**

Add | Remove

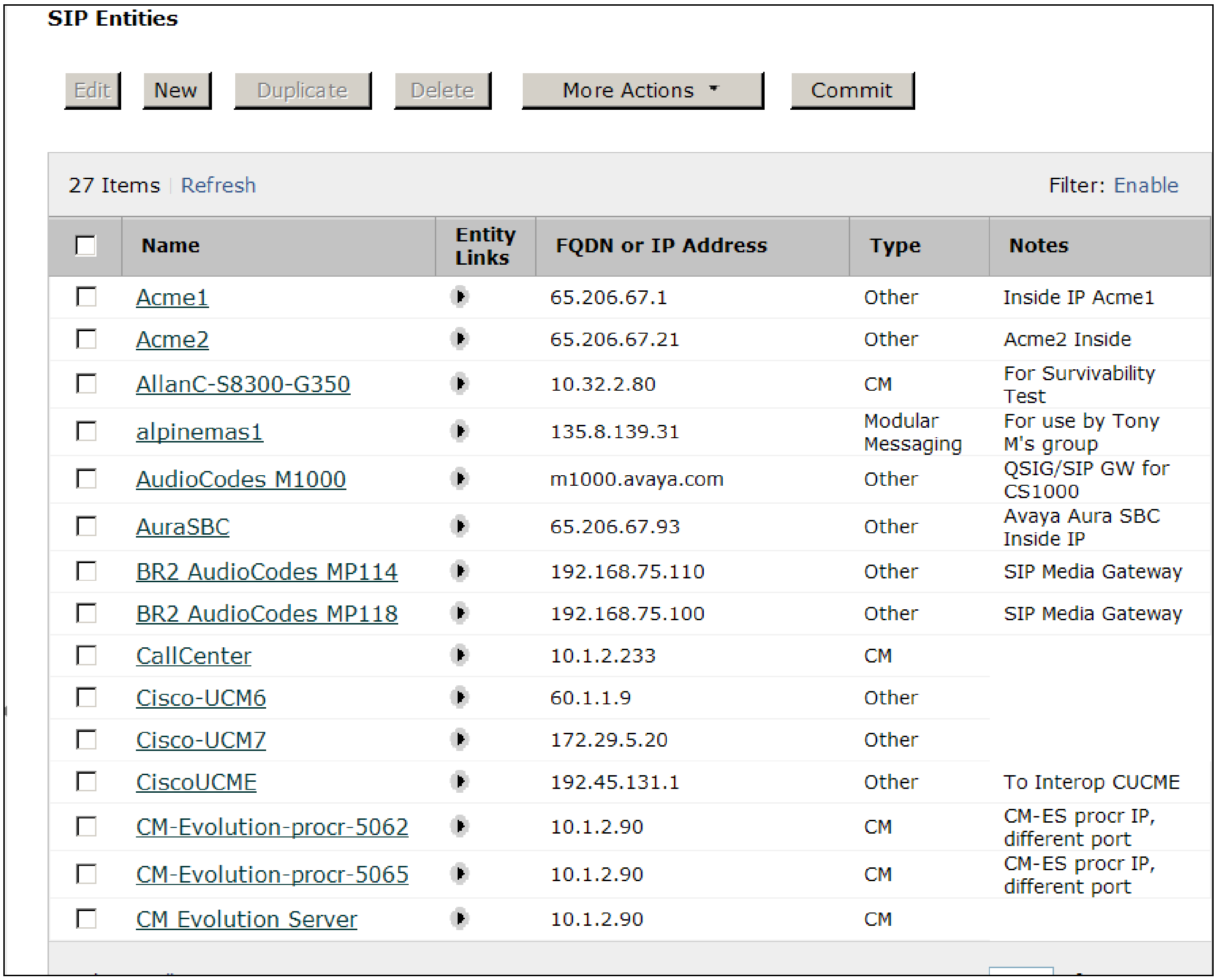
5 Items | Refresh — Filter: Enable

| | Matching Pattern | Min | Max | Delete Digits | Insert Digits | Address to modify | Notes |
|---|---|---|---|---|---|---|---|
| ☐ | * 8668502380 | * 10 | * 10 | * 10 | 30002 | both | |
| ☐ | * 8668506850 | * 10 | * 10 | * 10 | 30666 | both | |
| ☐ | * 8668510107 | * 10 | * 10 | * 10 | 30002 | both | |
| ☐ | * 8668512649 | * 10 | * 10 | * 10 | 8668512649 | both | Test CM ICHT |
| ☐ | * 8668523221 | * 10 | * 10 | * 10 | 36998 | both | Refer test vector |

In general, digit conversion such as this, that associates a Communication Manager extension (e.g., 36998, in this case, a VDN) with a corresponding toll-free number (e.g., 866-852-3221), can be performed in Communication Manager or in Session Manager. In the example shown above, if a user on the PSTN dials 866-852-3221, Session Manager will convert the number to 36998 before sending the SIP INVITE to Communication Manager. As such, it would not be necessary to use the incoming call handling table of the receiving Communication Manager trunk group to convert the toll-free number to its corresponding extension.

## 5.4. SIP Entities

To view or change SIP entities, select **Routing → SIP Entities**. Click the checkbox corresponding to the name of an entity and **Edit** to edit an existing entity, or the **New** button to add an entity. Click the **Commit** button after changes are completed. The following screen shows a portion of the list of configured SIP entities. In this screen, the SIP Entities named "AuraSBC", "alpinemas1", "CM-Evolution-procr-5062", and "CM Evolution Server" are relevant to these Application Notes.


**SIP Entities**

Edit | New | Duplicate | Delete | More Actions | Commit

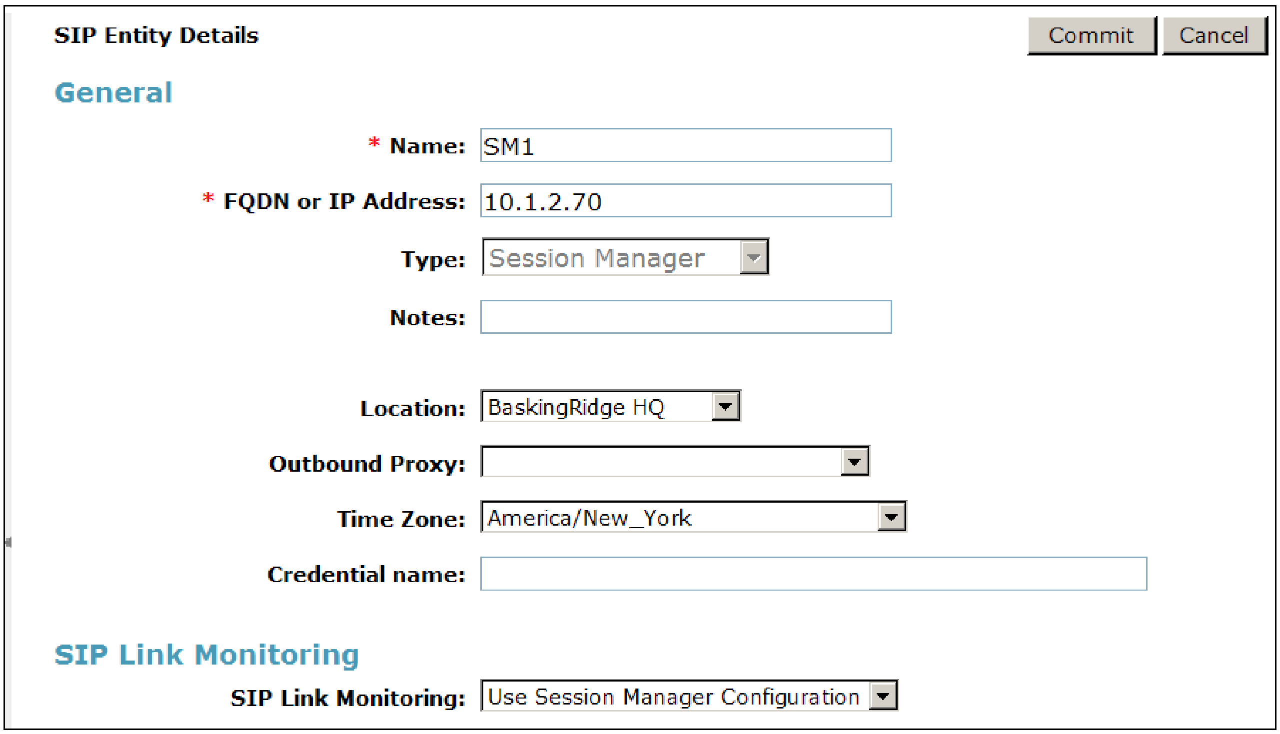
27 Items | Refresh — Filter: Enable

| | Name | Entity Links | FQDN or IP Address | Type | Notes |
|---|---|---|---|---|---|
| | Acme1 | | 65.206.67.1 | Other | Inside IP Acme1 |
| | Acme2 | | 65.206.67.21 | Other | Acme2 Inside |
| | AllanC-S8300-G350 | | 10.32.2.80 | CM | For Survivability Test |
| | alpinemas1 | | 135.8.139.31 | Modular Messaging | For use by Tony M's group |
| | AudioCodes M1000 | | m1000.avaya.com | Other | QSIG/SIP GW for CS1000 |
| | AuraSBC | | 65.206.67.93 | Other | Avaya Aura SBC Inside IP |
| | BR2 AudioCodes MP114 | | 192.168.75.110 | Other | SIP Media Gateway |
| | BR2 AudioCodes MP118 | | 192.168.75.100 | Other | SIP Media Gateway |
| | CallCenter | | 10.1.2.233 | CM | |
| | Cisco-UCM6 | | 60.1.1.9 | Other | |
| | Cisco-UCM7 | | 172.29.5.20 | Other | |
| | CiscoUCME | | 192.45.131.1 | Other | To Interop CUCME |
| | CM-Evolution-procr-5062 | | 10.1.2.90 | CM | CM-ES procr IP, different port |
| | CM-Evolution-procr-5065 | | 10.1.2.90 | CM | CM-ES procr IP, different port |
| | CM Evolution Server | | 10.1.2.90 | CM | |

The following screen shows Page 2 of the list of SIP Entities. In this screen, only the SIP Entity named "SM1" (corresponding to Avaya Aura™ Session Manager) is relevant to these Application Notes.

| ☐ | Name | Entity Links | FQDN or IP Address | Type | Notes |
|---|---|---|---|---|---|
| ☐ | Denver Nortel CS1000e | ▸ | CS1KGateway.avaya.com | Other | |
| ☐ | Juniper-SRX240 | ▸ | 1.0.0.2 | Other | |
| ☐ | Microsoft-OCS-Mediation-Server | ▸ | 135.8.19.139 | SIP Trunk | MS OCS Mediation Server in WM |
| ☐ | MikeH-S8300-G450 | ▸ | 10.32.2.20 | CM | For Survivability Test |
| ☐ | OITT Test Tool | ▸ | 135.8.19.109 | Other | OITT Test Tool |
| ☐ | RobertIP500 | ▸ | 10.1.2.190 | SIP Trunk | Robert's IP500 |
| ☐ | S8300-G250-JRWB | ▸ | 172.28.40.5 | CM | S8300-in-G250 at JRR workbench |
| ☐ | S8300-G450-BR1 | ▸ | 135.8.139.118 | CM | S8300 is an LSP |
| ☐ | S87x0-Procr-CM521-VZ | ▸ | 65.206.67.3 | CM | CM 5.2.1 Verizon Testbed |
| ☐ | SM1 | ▸ | 10.1.2.70 | Session Manager | |

The following screen shows the upper portion of the **SIP Entity Details** corresponding to "SM1". The **FQDN or IP Address** field for "SM1" is the Avaya Aura™ Session Manager Security Module IP Address (10.1.2.70), which is used for SIP signaling with other networked SIP entities. The **Type** for this SIP entity is "Session Manager". Select an appropriate location for the Session Manager from the **Location** drop-down menu. In the shared test environment, the Session Manager used location "BaskingRidge HQ". 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 "SM1". The links relevant to these Application Notes are described in the following section.

**Entity Links**

Add Remove

27 Items | Refresh Filter: Enable

| ☐ | SIP Entity 1 | Protocol | Port | SIP Entity 2 | Port | Trusted |
|---|---|---|---|---|---|---|
| ☐ | SM1 | TCP | * 5060 | Acme1 | * 5060 | ☑ |
| ☐ | SM1 | TCP | * 5060 | Acme2 | * 5060 | ☑ |
| ☐ | SM1 | TCP | * 5060 | AuraSBC | * 5060 | ☑ |
| ☐ | SM1 | TCP | * 5060 | CallCenter | * 5060 | ☑ |
| ☐ | SM1 | TCP | * 5060 | Cisco-UCM6 | * 5060 | ☑ |
| ☐ | SM1 | TCP | * 5060 | Cisco-UCM7 | * 5060 | ☑ |
| ☐ | SM1 | TCP | * 5060 | CiscoUCME | * 5060 | ☑ |
| ☐ | SM1 | TCP | * 5060 | CM Evolution Server | * 5060 | ☑ |
| ☐ | SM1 | TCP | * 5062 | CM-Evolution-procr-5062 | * 5062 | ☑ |
| ☐ | SM1 | TCP | * 5060 | Denver Nortel CS1000e | * 5060 | ☑ |
| ☐ | SM1 | TCP | * 5060 | alpinemas1 | * 5060 | ☑ |

Scrolling down, the following screen shows the lower portion of the **SIP Entity Details**, a listing of the configured ports for "SM1". In the sample configuration, TCP port 5060 was already in place for the shared test environment, using **Default Domain** "avaya.com". To enable calls with Verizon to be distinguished from other types of SIP calls using the same Session Manager, TCP port 5062 was added, with **Default Domain** "adevc.avaya.globalipcom.com". Click the **Add** button to configure a new port. TCP is used in the sample configuration for improved visibility during testing.

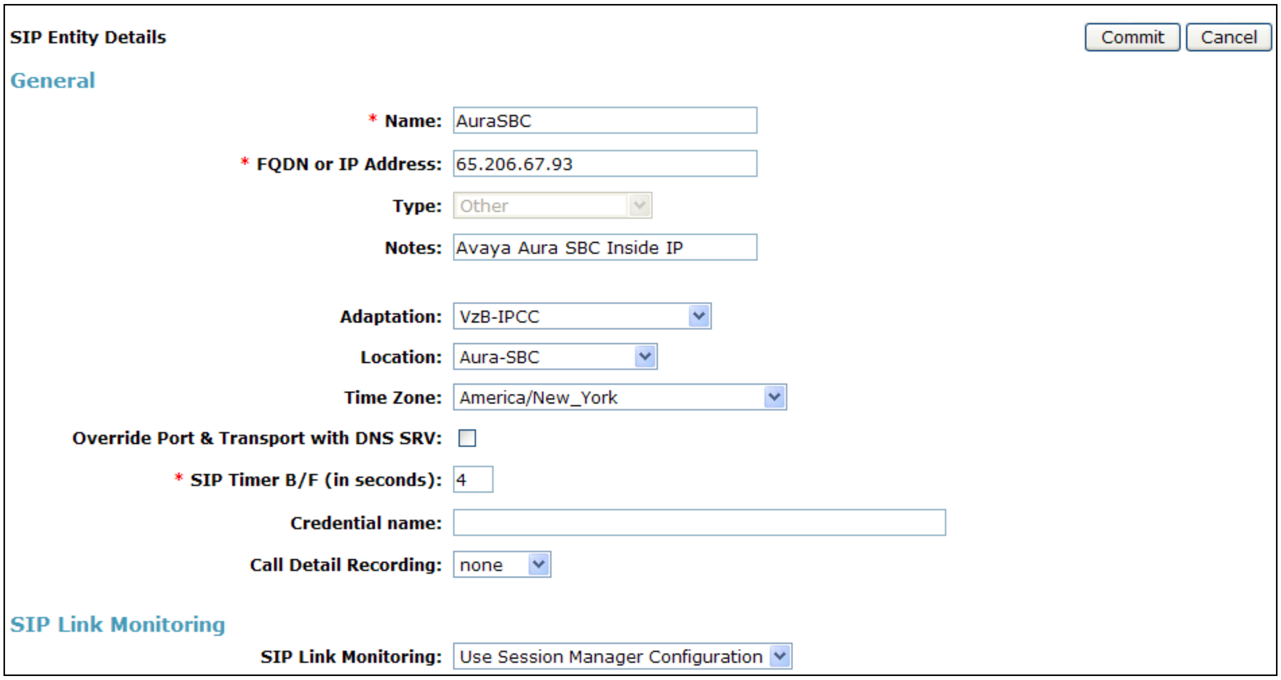
**Port**

Add Remove

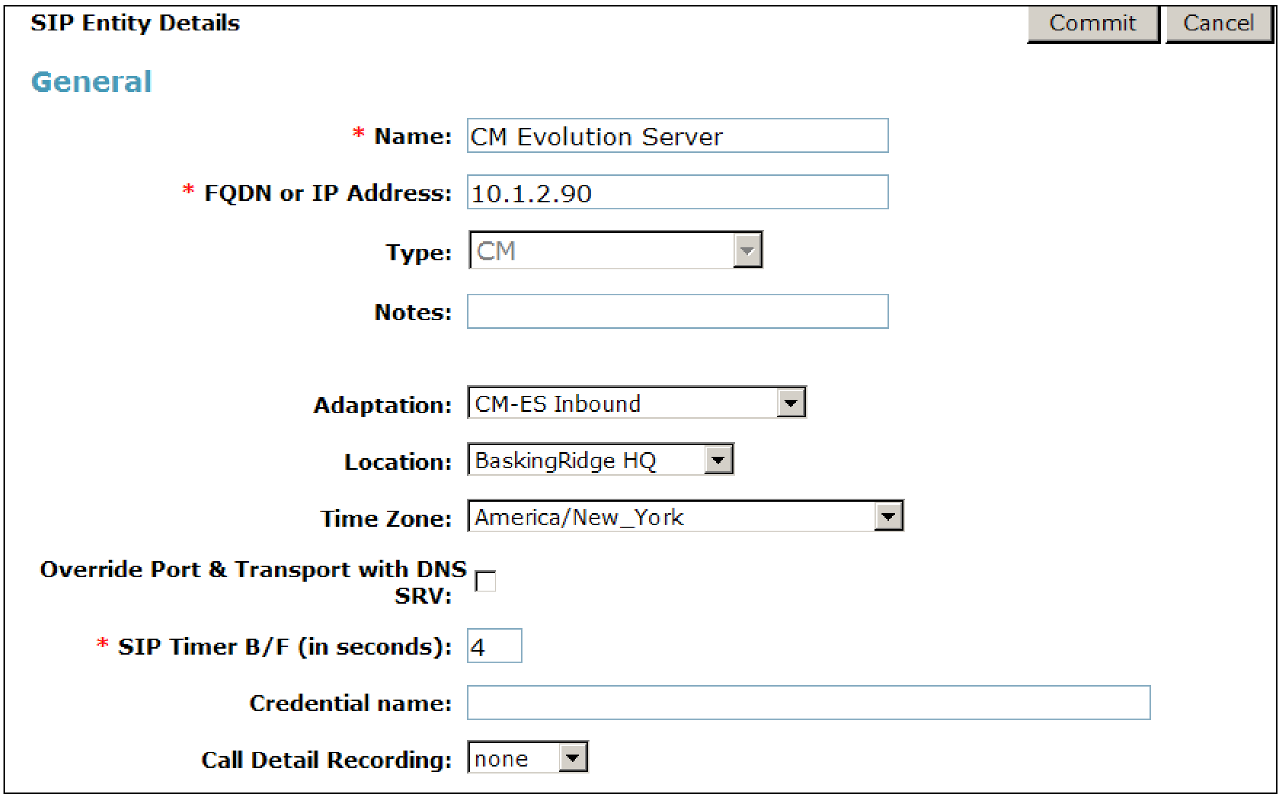
5 Items | Refresh Filter: Enable

| ☐ | Port | Protocol | Default Domain | Notes |
|---|---|---|---|---|
| ☐ | 5060 | TCP | avaya.com | |
| ☐ | 5060 | UDP | avaya.com | |
| ☐ | 5061 | TLS | avaya.com | |
| ☐ | 5062 | TCP | adevc.avaya.globalipcom.com | Verizon testing CPE-domain |
| ☐ | 5070 | TCP | avocs.contoso.com | |

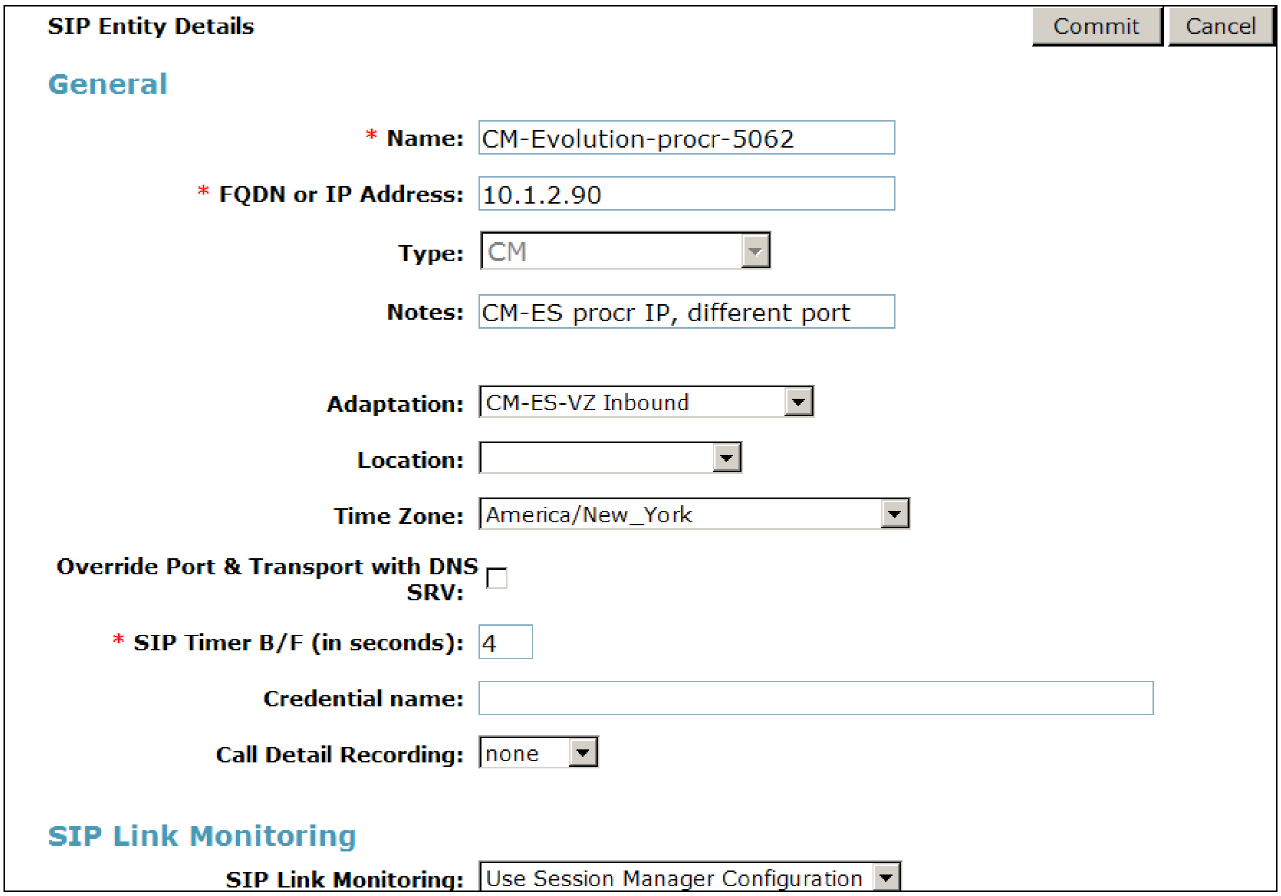
The following screen shows the **SIP Entity Details** corresponding to “AuraSBC”. The **FQDN or IP Address** field is configured with the SBC inside private IP Address (65.206.67.93). “Other” is selected from the **Type** drop-down menu for SBC SIP Entities. This SBC has been assigned to **Location** “Aura-SBC”, and the “VzB-IPCC” adapter is applied.



The following screen shows a portion of the **SIP Entity Details** corresponding to an Avaya Aura™ Communication Manager SIP Entity named "CM Evolution Server" This is the SIP Entity that was already in place in the shared Avaya Interoperability Lab test environment, prior to adding the Verizon IP Trunk configuration. The **FQDN or IP Address** field contains the IP Address of the "processor ethernet" (10.1.2.90). In systems with Avaya G650 Media Gateways containing C-LAN cards, C-LAN cards may also be used as SIP entities, instead of, or in addition to, the "processor ethernet". "CM" is selected from the **Type** drop-down menu. In the shared test environment, the **Adaptation** "CM-ES Inbound" and **Location** "BaskingRidge HQ" had already been assigned to the Communication Manager SIP entity.



The following screen shows the **SIP Entity Details** for an entity named "CM-Evolution-procr-5062". This entity uses the same **FQDN or IP Address** (10.1.2.90) as the prior entity with name "CM Evolution Server"; both correspond to the S8800 Processor Ethernet. Later, a unique port, 5062, will be used for the Entity Link to "CM-Evolution-procr-5062". Using a different port is one approach that will allow Avaya Aura™ Communication Manager to distinguish traffic originally from Verizon from other SIP traffic arriving from the same IP Address of the Avaya Aura™ Session Manager. The adapter "CM-ES-VZ Inbound" is applied to this SIP entity. Recall that this adapter is used to adapt the domain as well as map the Verizon toll-free numbers to the corresponding Communication Manager extensions. If desired, a location can be assigned if location-based routing criteria will be used.




## 5.5. Entity Links

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

The following screen shows a partial list of configured links. In the screen below, the links named "AuraSBC", "CM-ES-VZ-5062", and "CM Evolution Server" are relevant to these Application Notes. Each of the links uses the entity named "SM1" as **SIP Entity 1**, and the appropriate entity, such as "AuraSBC" for **SIP Entity 2**. Note that there are two SIP Entity Links, using different TCP ports, linking the same SM1 with the processor Ethernet of Avaya Aura™ Communication

Manager. For one link, named “CM Evolution Server”, both entities use port 5060. For the other, named “CM-ES-VZ-5062”, both entities use port 5062.

**Entity Links**

Edit | New | Duplicate | Delete | More Actions ▾ | Commit

27 Items | Refresh    Filter: Enable

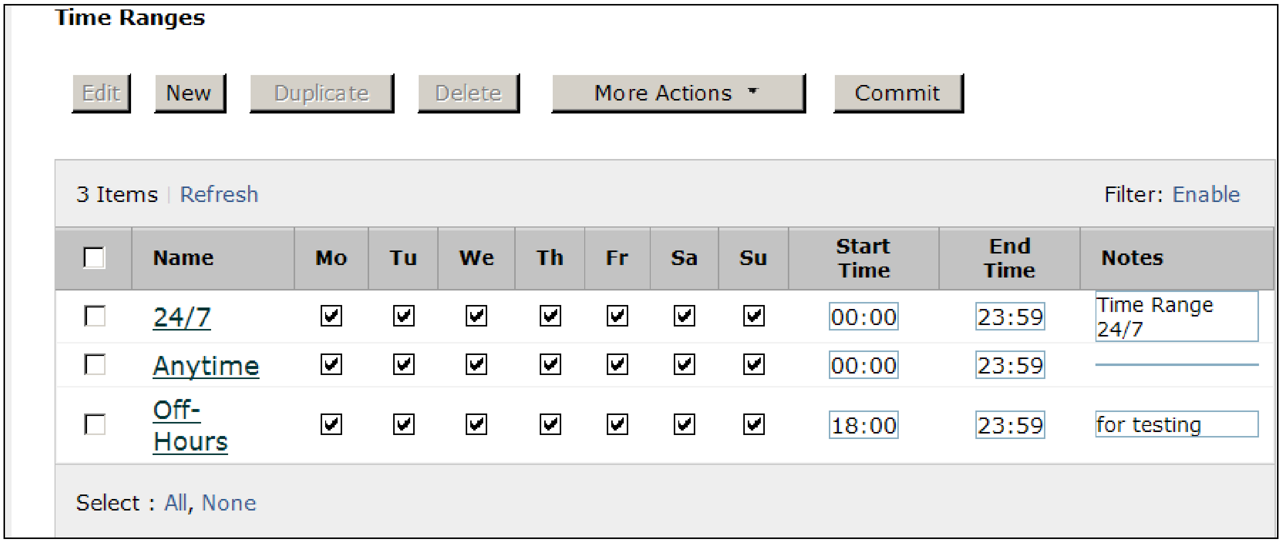
| ☐ | Name | SIP Entity 1 | Protocol | Port | SIP Entity 2 | Port | Trusted | Notes |
|---|---|---|---|---|---|---|---|---|
| ☐ | Acme1 | SM1 | TCP | 5060 | Acme1 | 5060 | ☑ | |
| ☐ | Acme2 | SM1 | TCP | 5060 | Acme2 | 5060 | ☑ | |
| ☐ | AuraSBC | SM1 | TCP | 5060 | AuraSBC | 5060 | ☑ | |
| ☐ | Call Center | SM1 | TCP | 5060 | CallCenter | 5060 | ☑ | |
| ☐ | Cisco-UCM6 | SM1 | TCP | 5060 | Cisco-UCM6 | 5060 | ☑ | |
| ☐ | Cisco-UCM7 | SM1 | TCP | 5060 | Cisco-UCM7 | 5060 | ☑ | |
| ☐ | CiscoUCME-Link | SM1 | TCP | 5060 | CiscoUCME | 5060 | ☑ | |
| ☐ | CM-ES-VZ-5062 | SM1 | TCP | 5062 | CM-Evolution-procr-5062 | 5062 | ☑ | Same IP, diff port |
| ☐ | CM Evolution Server | SM1 | TCP | 5060 | CM Evolution Server | 5060 | ☑ | |
| ☐ | Denver CS1000e | SM1 | TCP | 5060 | Denver Nortel CS1000e | 5060 | ☑ | |

The link named “CM Evolution Server” links Session Manager “SM1” with the Communication Manager processor Ethernet. This link existed in the shared configuration prior to adding the Verizon-related configuration. This link, using port 5060, 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, or traffic related to Avaya Modular Messaging, which has SIP integration to Session Manager.

The link named “CM-ES-VZ-5062” also links Session Manager “SM1” with the Communication Manager processor Ethernet. However, this link uses port 5062 for both entities in the link. This link was created to allow Communication Manager to distinguish Verizon inbound calls from other calls that arrive from the same Session Manager. Other methods of distinguishing traffic could be used, if desired. For example, in a configuration using G650 Media Gateways, the use of one or more C-LAN interface cards can provide additional Communication Manager SIP Signaling alternatives.

## 5.6. Time Ranges

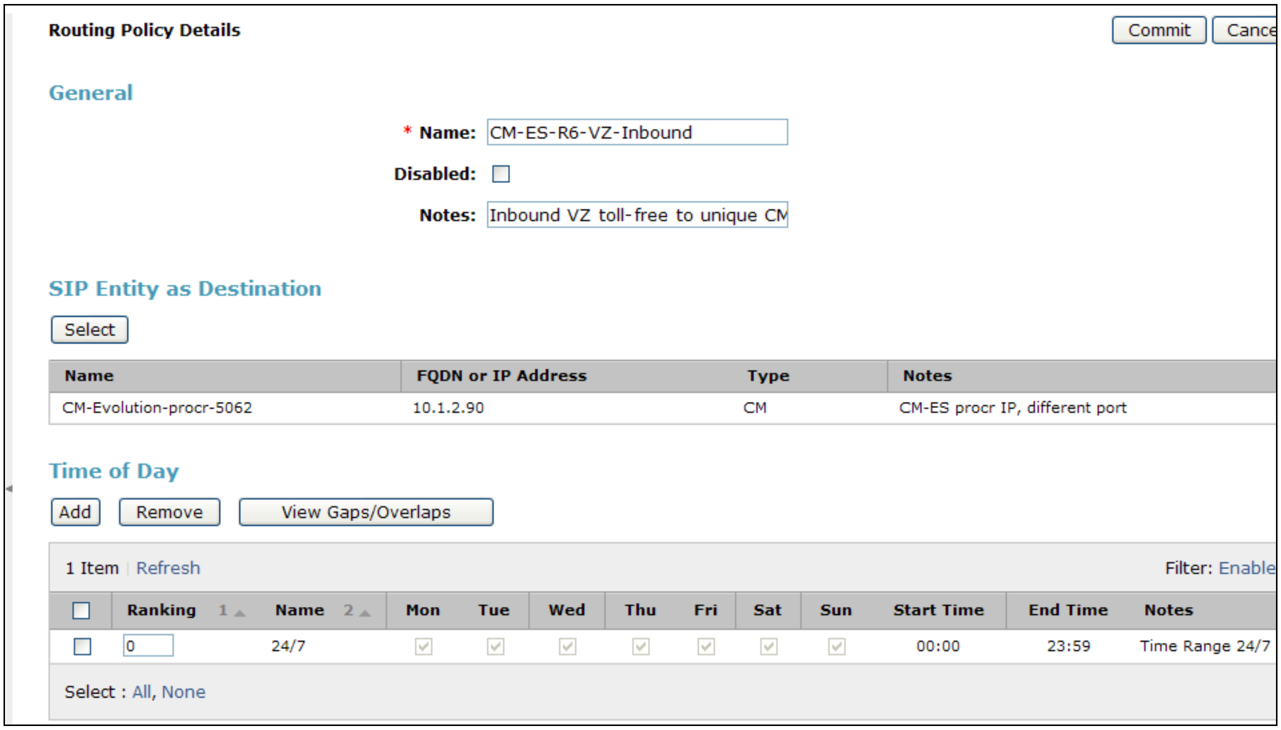
To view or change Time Ranges, select **Routing → Time Ranges**. The Routing Policies shown subsequently will use the "24/7" range since time-based routing was not the focus of these Application Notes. Click the **Commit** button after changes are completed.



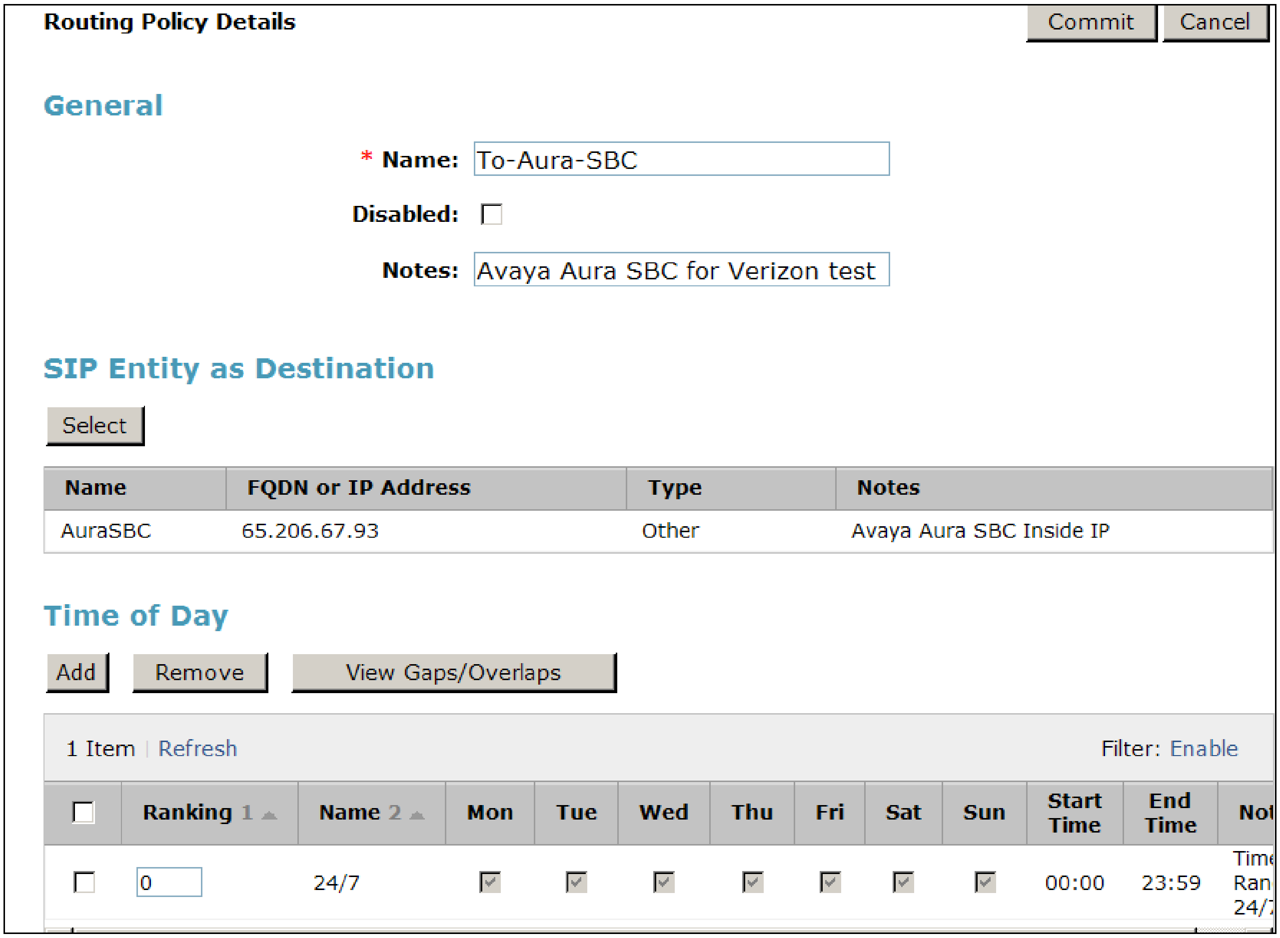
## 5.7. Routing Policies

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

The following screen shows the **Routing Policy Details** for the policy named "CM-ES-R6-VZ-Inbound" associated with incoming toll-free calls from Verizon to Communication Manager, using the Avaya S8800 PE. Observe the **SIP Entity as Destination** is the entity named "CM-Evolution-procr-5062".



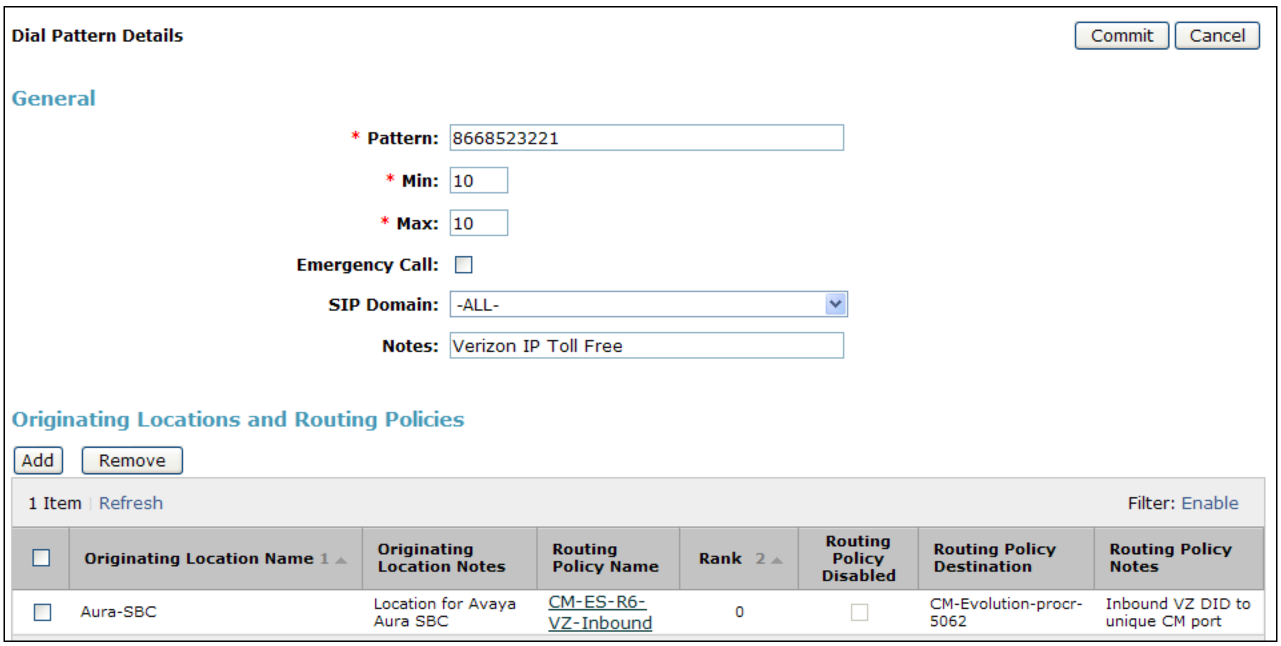
The following screen shows the **Routing Policy Details** for the policy named “To-Aura-SBC”. Observe the **SIP Entity as Destination** is the entity named “AuraSBC”.



## 5.8. Dial Patterns

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

The following screen illustrates an example dial pattern used to verify an inbound toll-free call to the enterprise via the Avaya S8800 Processor Ethernet. When a user on the PSTN dials a toll-free number such as 866-852-3221, Verizon delivers the number to the enterprise, and the SBC sends the call to Session Manager. The dial pattern below matches on 866-852-3221 specifically. Dial patterns can alternatively match on ranges of numbers. Under **Originating Location and Routing Policies**, the routing policy named "CM-ES-R6-VZ-Inbound" is selected, which sends the call to Communication Manager using the routing policy "CM-Evolution-procr-5062" as described previously. The **Originating Location Name** is "Aura-SBC".




Once Dial Patterns are configured that associate dialed numbers with routing policies, a return to the routing policy screen will list the Dial Patterns associated with the policy.

# 6. Avaya Aura™ Session Border Controller (SBC)

Reference [AuraSBC-IP-Trunk] is a companion Application Notes that illustrates the initial installation, licensing, and wizard configuration of the SBC that formed the starting point for the SBC configuration shown in these Application Notes. In **Section 5** of reference [AuraSBC-IP-Trunk], the installation, licensing, and initial wizard configuration of the SBC are shown. These steps will not be repeated here.

The Avaya Aura™ Session Border Controller includes a configuration wizard that can be used as part of the installation of the SBC template on System Platform. The wizard pre-configures the underlying SBC for much of the required provisioning. The configuration shown in this section assumes that the configuration of the connection to the Verizon IP Contact Center Service is being added to the SBC configuration previously documented in reference [AuraSBC-IP-Trunk]. As an alternative, the procedures using the installation wizard from reference [AuraSBC-IP-Trunk] can be used to connect to the Verizon IPCC Service. The wizard can be used for one SIP Service Provider trunk connection only.

In the example configuration in these Application Notes, the wizard configuration shown in [AuraSBC-IP-Trunk] was already run to configure the SBC. Although the SBC configuration for connection to the Verizon IPCC Service is added to a configuration where the SBC installation wizard was previously run for connection to the Verizon IP Trunk service, these Application Notes are intended to cover only the Verizon IPCC Service. That is, these Application Notes do not intend to cover both services being used at the same time.

After the SBC has been installed, any subsequent changes to the network configuration (e.g.,. IP address, network mask, hostname) for the SBC eth0 or eth2 interfaces must be done via the System Platform webconsole Network Configuration page. Any backup and restore actions should also use System Platform. Configuration of SBC behaviors (e.g., header manipulations) can be performed through the element manager GUI as shown in **Section 6.3**.

In the sample configuration, the Avaya S8800 Server has four physical network interfaces, labeled 1 through 4. The port labeled “1” (virtual “eth0”) is used for the management and private (inside) network interface of the SBC. The port labeled “4” (virtual “eth2”) is used for the public (outside) network interface of the SBC.

## 6.1. Avaya Aura™ Session Border Controller (SBC) Installation

For the installation procedures used in the sample configuration, please refer to **Section 5.1** of reference [AuraSBC-IP-Trunk].

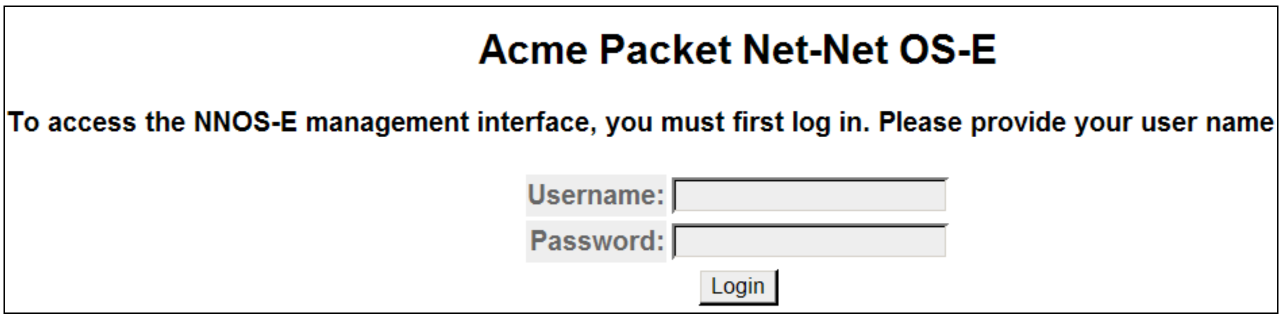
## 6.2. Avaya Aura™ Session Border Controller (SBC) Licensing

For the licensing procedures used in the sample configuration, please refer to Section 5.2 of reference [AuraSBC-IP-Trunk].

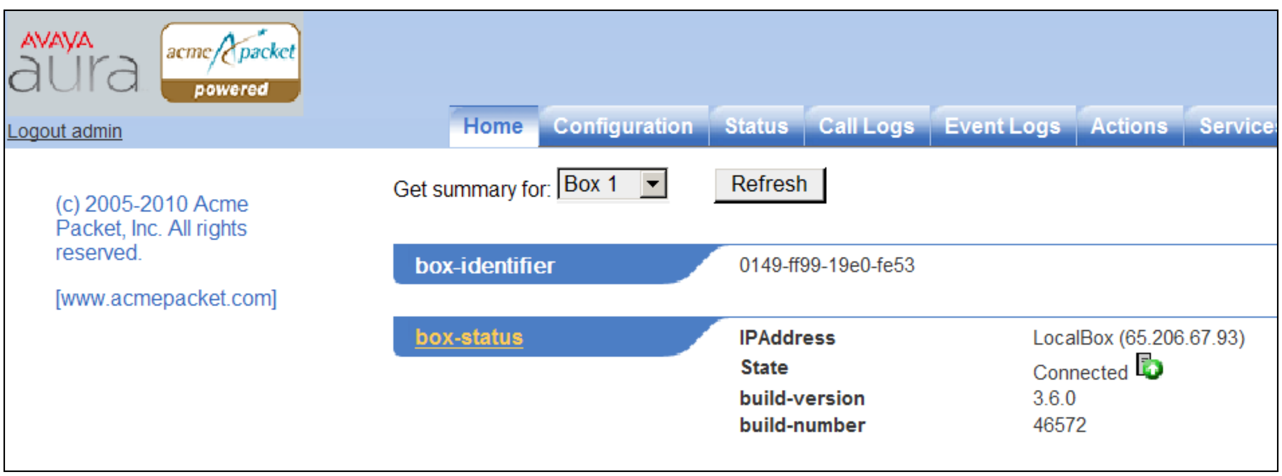
## 6.3. Avaya Aura™ Session Border Controller (SBC) Element Manager Configuration

This section presents the incremental configuration using the element manager of the SBC. It is assumed that the installation, licensing, and configuration shown in **Section 5.1 – Section 5.3** of reference [AuraSBC-IP-Trunk] has been completed. The configuration screens will be familiar to the reader experienced with the Acme Packet Net-Net OS-E.

To log in, either select the wrench sbc icon from System Platform, or enter https://<ip-addr> where <ip-addr> is the management IP Address of the SBC. In the example configuration, the IP Address 65.206.67.93 can be used Address https://65.206.67.93/ to access a log in screen. Enter appropriate **Username** and **Password** and click **Login**.

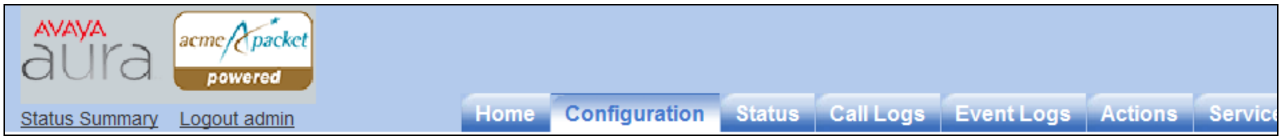



The following shows an abridged **Home** screen after logging in. Note the tabs at the top.

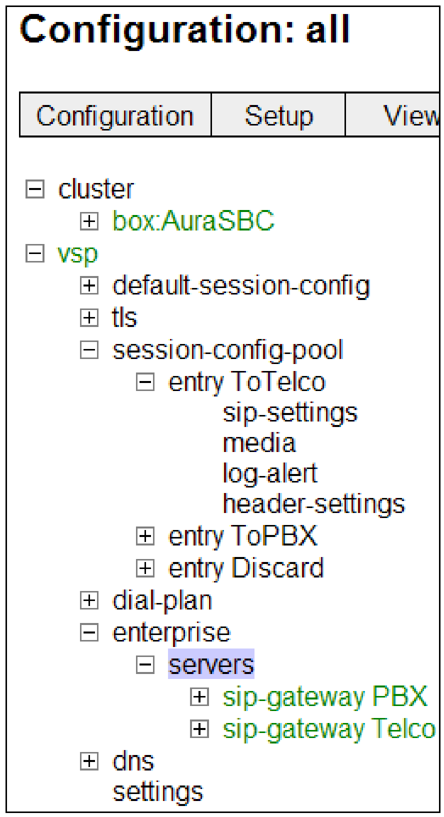


### 6.3.1 Adding SIP Gateway to Verizon IP Contact Center Service

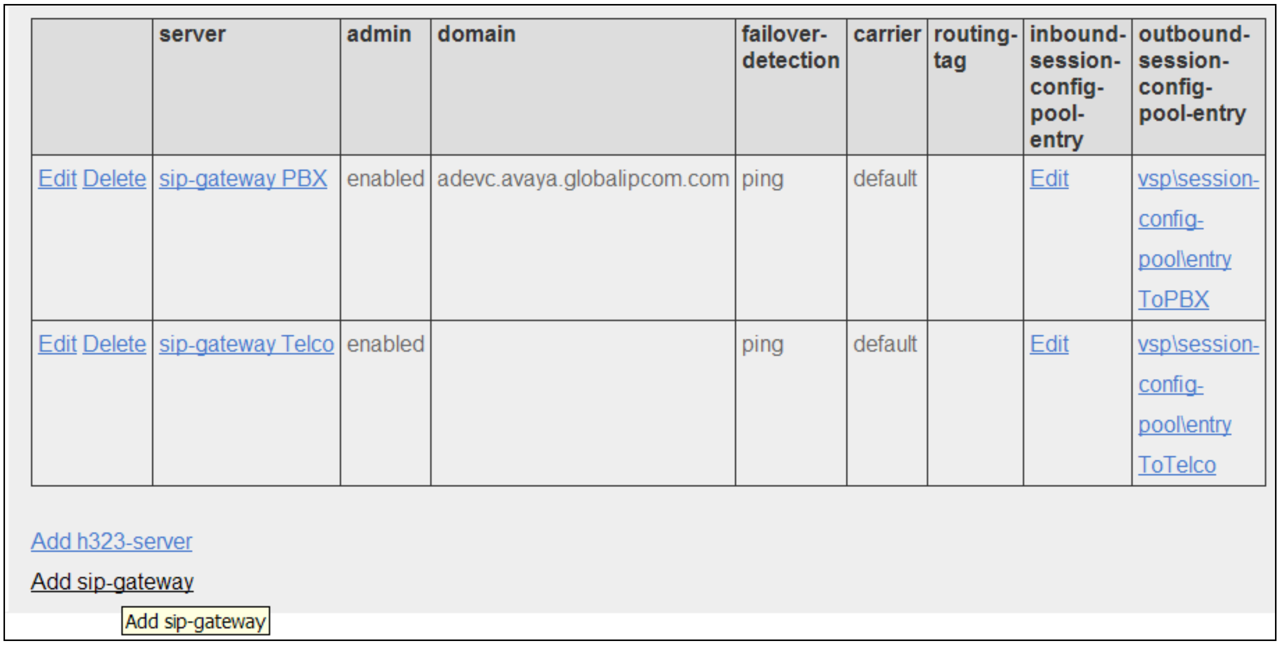
After logging in, select the **Configuration** tab.




Using the menu on the left hand side, expand **vsp → enterprise → servers** as shown below.



On the right hand side, the following screen shows the foundational configuration of "sip-gateways" already in place from reference [AuraSBC-IP-Trunk]. Note that there is already a "sip-gateway PBX" that will be used for connectivity towards Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager on the inside or private side of the SBC. There is also a "sip-gateway Telco" previously configured for connectivity to the Verizon IP Trunk Service on the outside or public side of the SBC. Although not the focus of these Application Notes, the connectivity to the Verizon IP Trunk Service will remain in place, and connectivity to the Verizon IP Contact Center service will be added. Click **Add sip-gateway** as shown below.



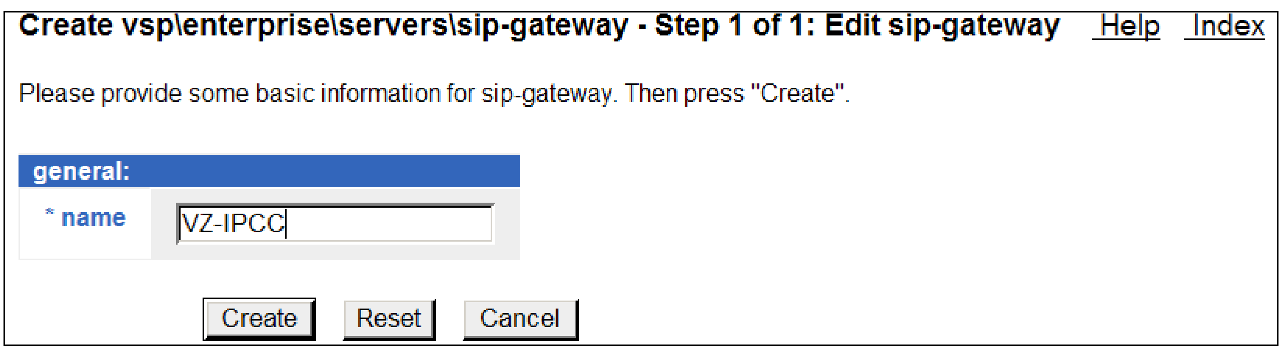
| | server | admin | domain | failover-detection | carrier | routing-tag | inbound-session-config-pool-entry | outbound-session-config-pool-entry |
|---|---|---|---|---|---|---|---|---|
| Edit Delete | sip-gateway PBX | enabled | adevc.avaya.globalipcom.com | ping | default | | Edit | vsp\session-config-pool\entry ToPBX |
| Edit Delete | sip-gateway Telco | enabled | | ping | default | | Edit | vsp\session-config-pool\entry ToTelco |

Add h323-server

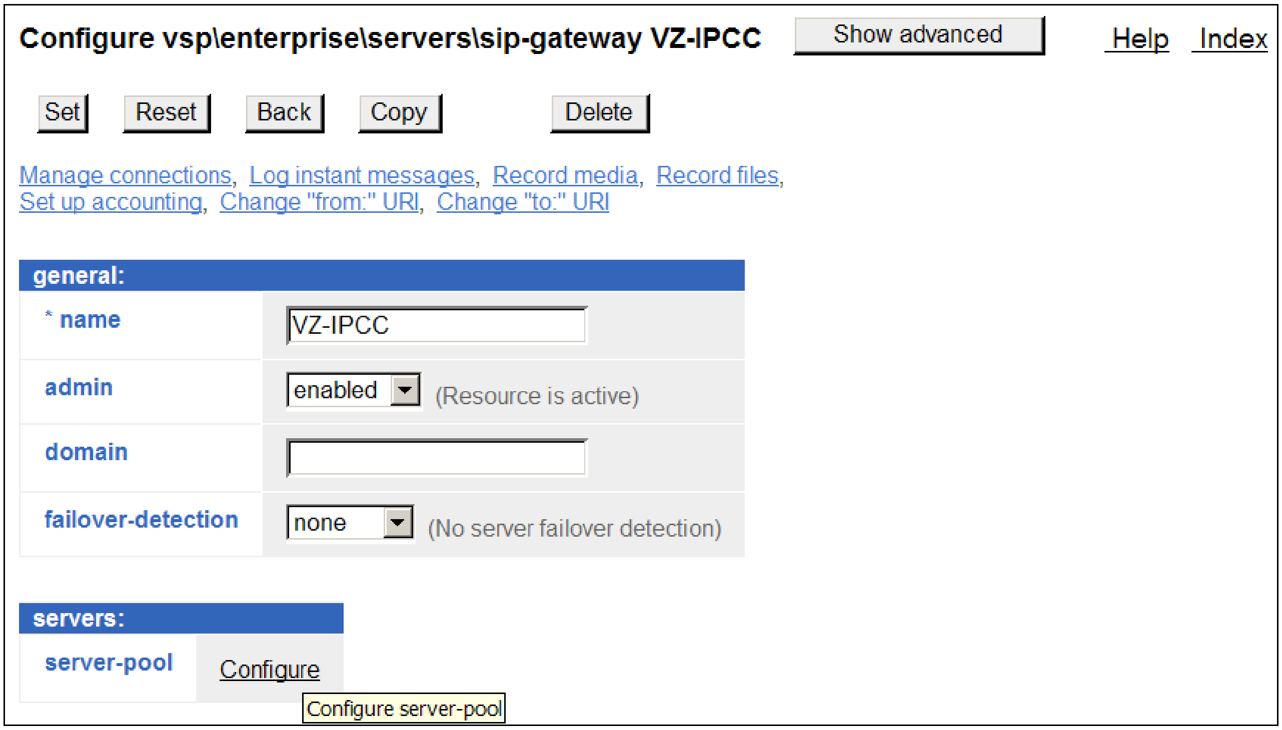
Add sip-gateway

Add sip-gateway

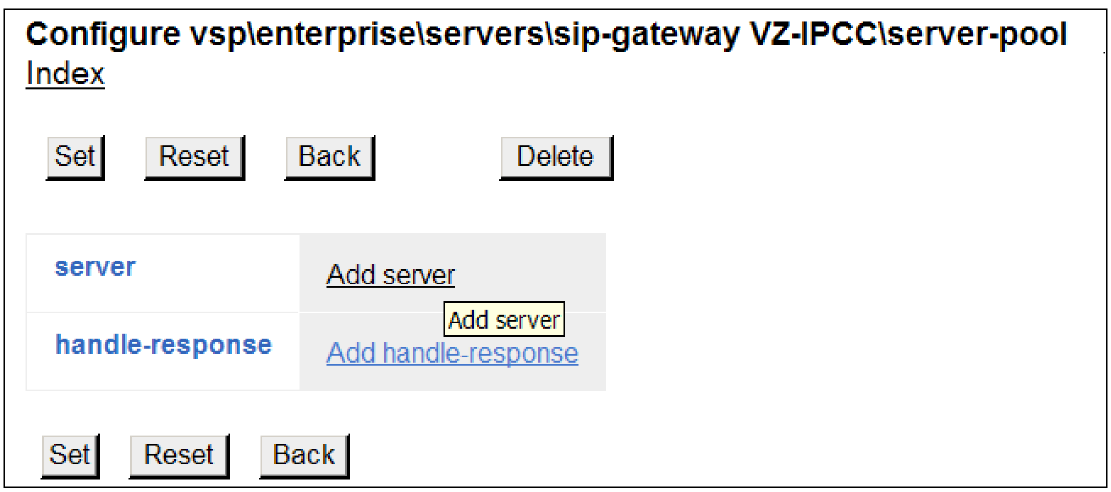
In the resultant screen shown below, enter an appropriate **name** for the new sip-gateway to the Verizon IP Contact Center service and click **Create**.



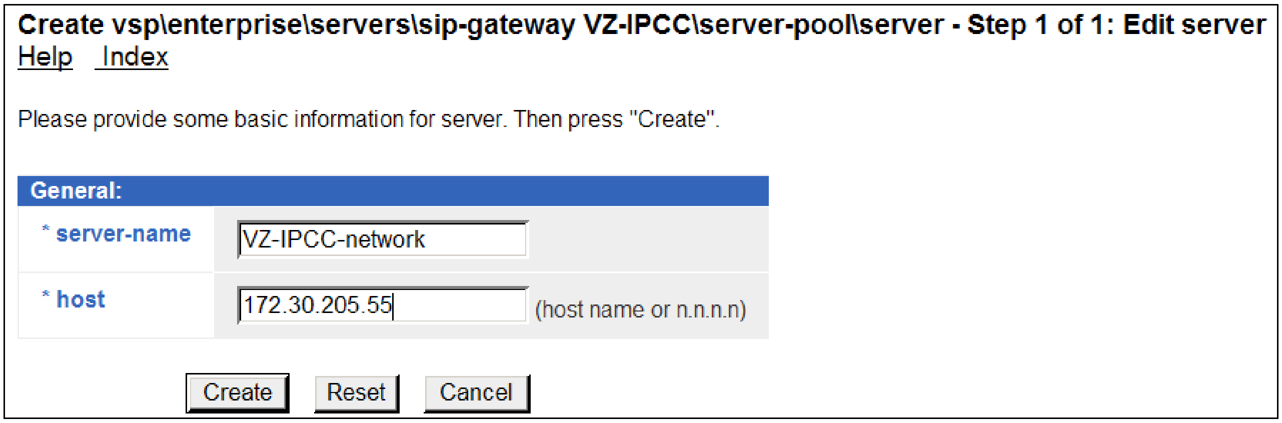
In the resultant screen, click **Configure** under the "servers: server-pool" heading, as shown below.



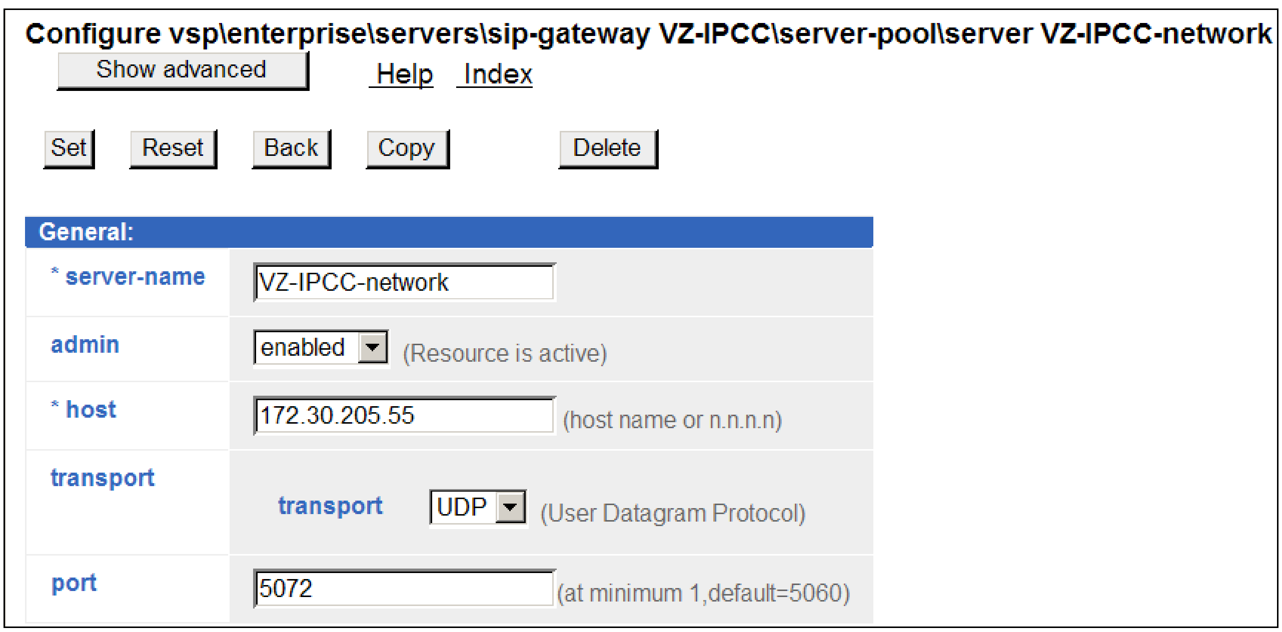

In the resultant screen, click **Add server** as shown below.



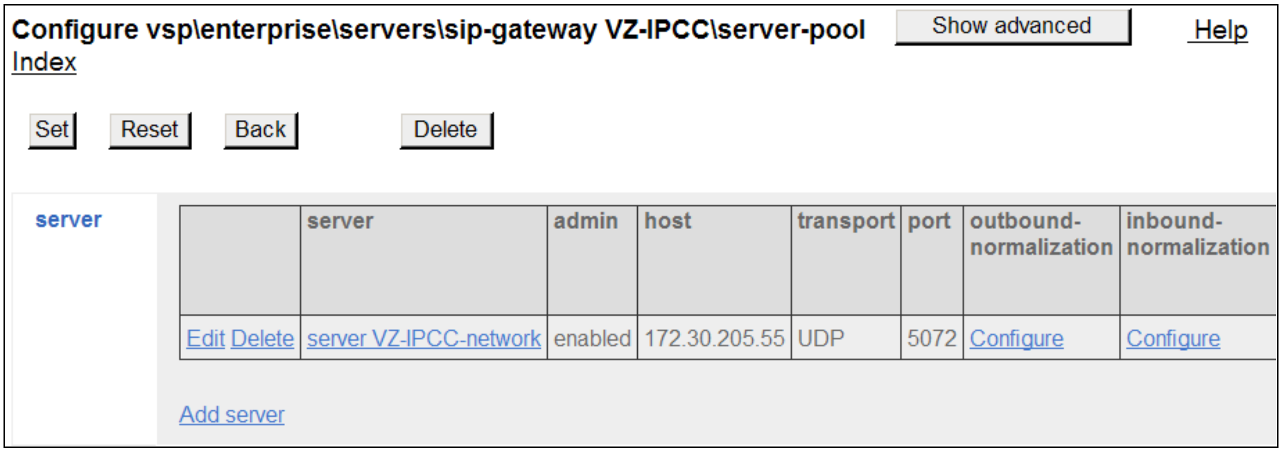
In the resultant screen, enter an appropriate **server-name** and **hostname** for the Verizon IP Contact Center service. In the screen shown below, the IP Address 172.30.205.55 was provided by Verizon as the SIP signaling IP Address of the IP Contact Center service. Click **Create**.




In the resultant screen, select UDP as the **transport** and enter an appropriate **port**. In the sample configuration, Verizon IP Contact Center service expected the enterprise to send SIP signaling to IP Address 172.30.205.55 and port 5072, as shown below. Click **Set**.



After clicking **Set**, a screen such as the following is displayed.


Configure vsp\enterprise\servers\sip-gateway VZ-IPCC\server-pool Show advanced Help
Index

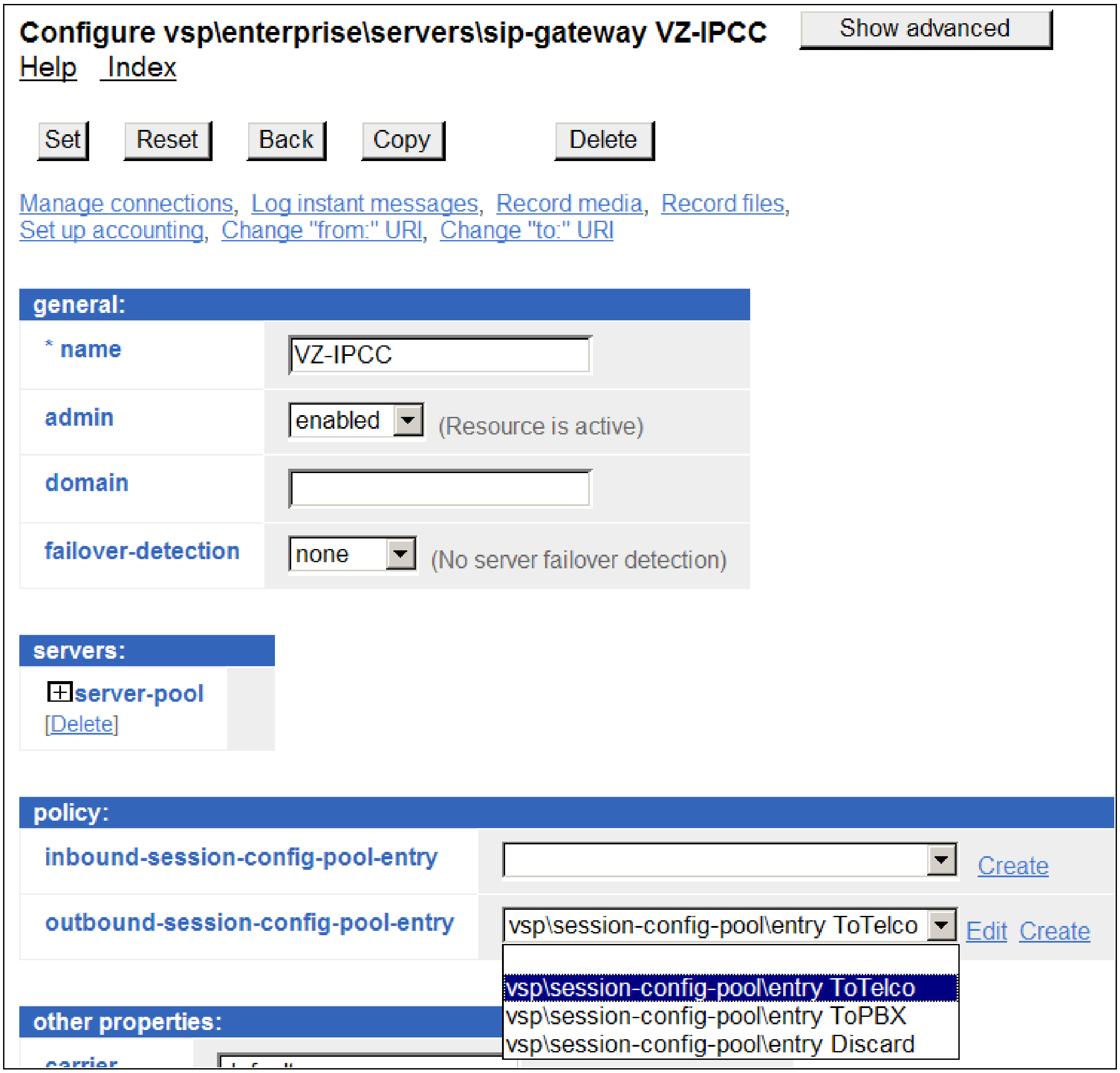
Set Reset Back Delete

server

| | server | admin | host | transport | port | outbound-normalization | inbound-normalization |
|---|---|---|---|---|---|---|---|
| Edit Delete | server VZ-IPCC-network | enabled | 172.30.205.55 | UDP | 5072 | Configure | Configure |

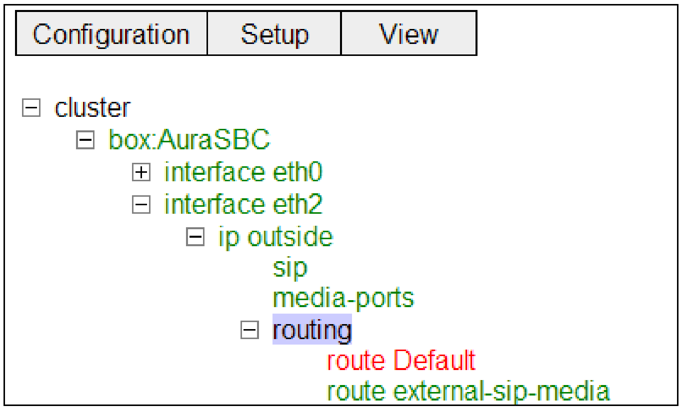
Add server

Using the left-side menu, navigate to **vsp → enterprise → servers → sip-gateway** and select the newly created "VZ-IPCC" entry. Scroll down to the policy heading. Using the **outbound-session-config-pool-entry** drop-down menu, select the entry "vsp\session-config-pool\entry To-Telco" as shown in the screen below. This session-config-pool entry was created by the wizard configuration shown in reference [AuraSBC-IP-Trunk].

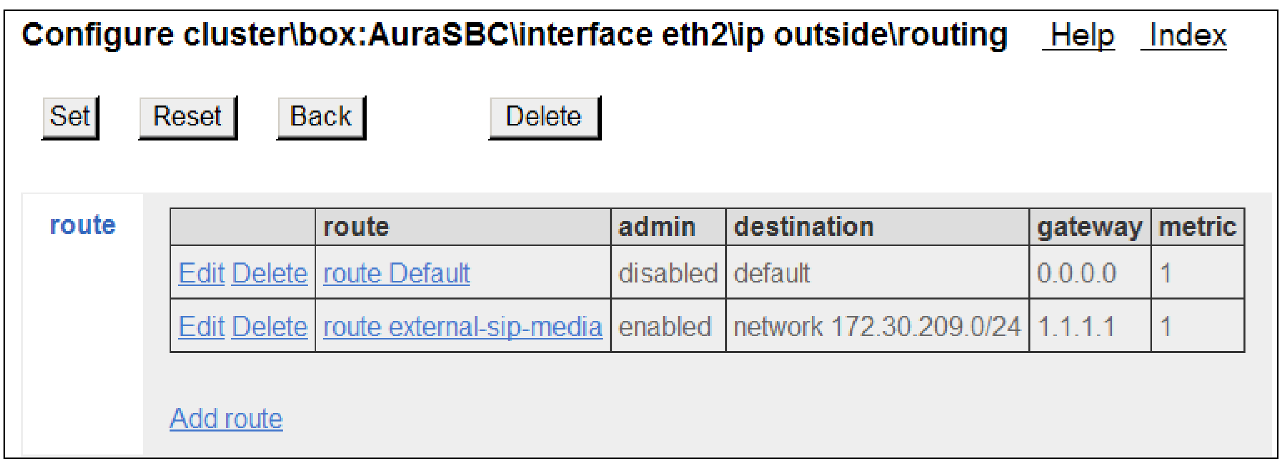


### 6.3.2 Adding IP Routing for Verizon IP Contact Center Network

From the left-side menu, select **routing** for the interface to the outside network, which is interface virtual "eth2" in the sample configuration.




In the right-side, a screen such as the following is displayed. The screen below shows the IP route established from reference [AuraSBC-IP-Trunk]. The Verizon IP Trunk Service on network 172.30.209.0/24 used gateway 1.1.1.1. A new route will be added for the Verizon IP Contact Center service using the same gateway. In the sample configuration, Verizon IP Trunk service and Verizon IP Contact Center service shared the same PIP access circuit. Click **Add route**.



**Configure cluster\box:AuraSBC\interface eth2\ip outside\routing** Help Index

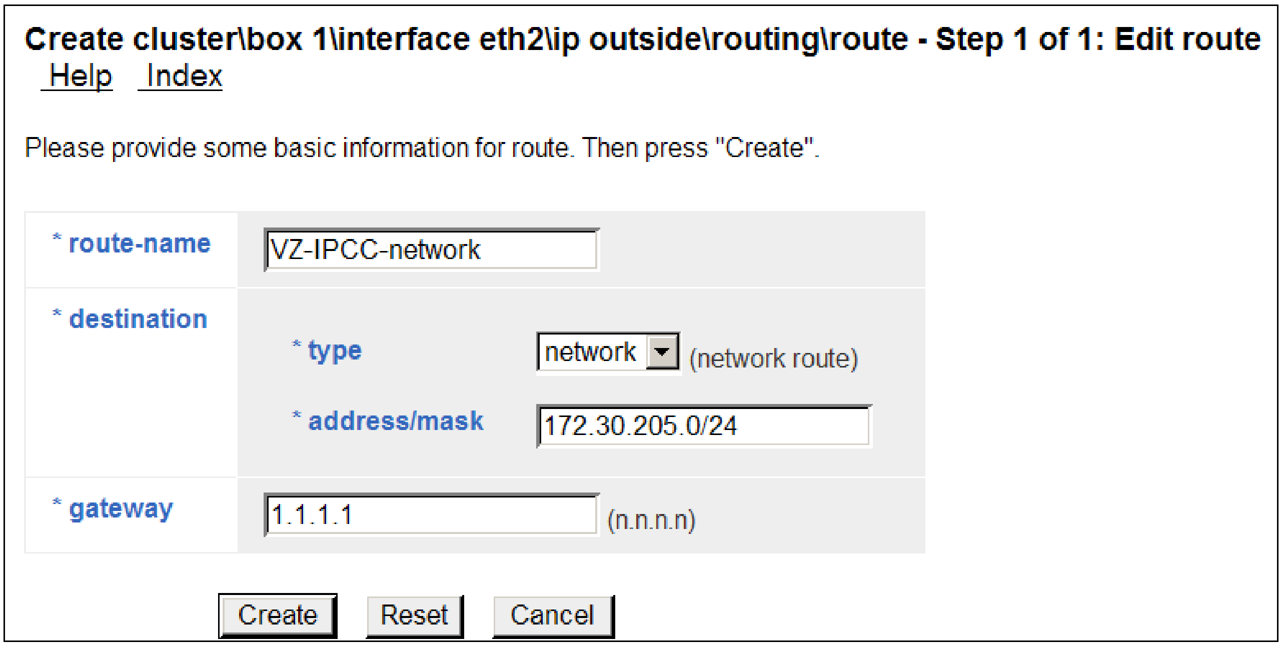
Set Reset Back Delete

**route**

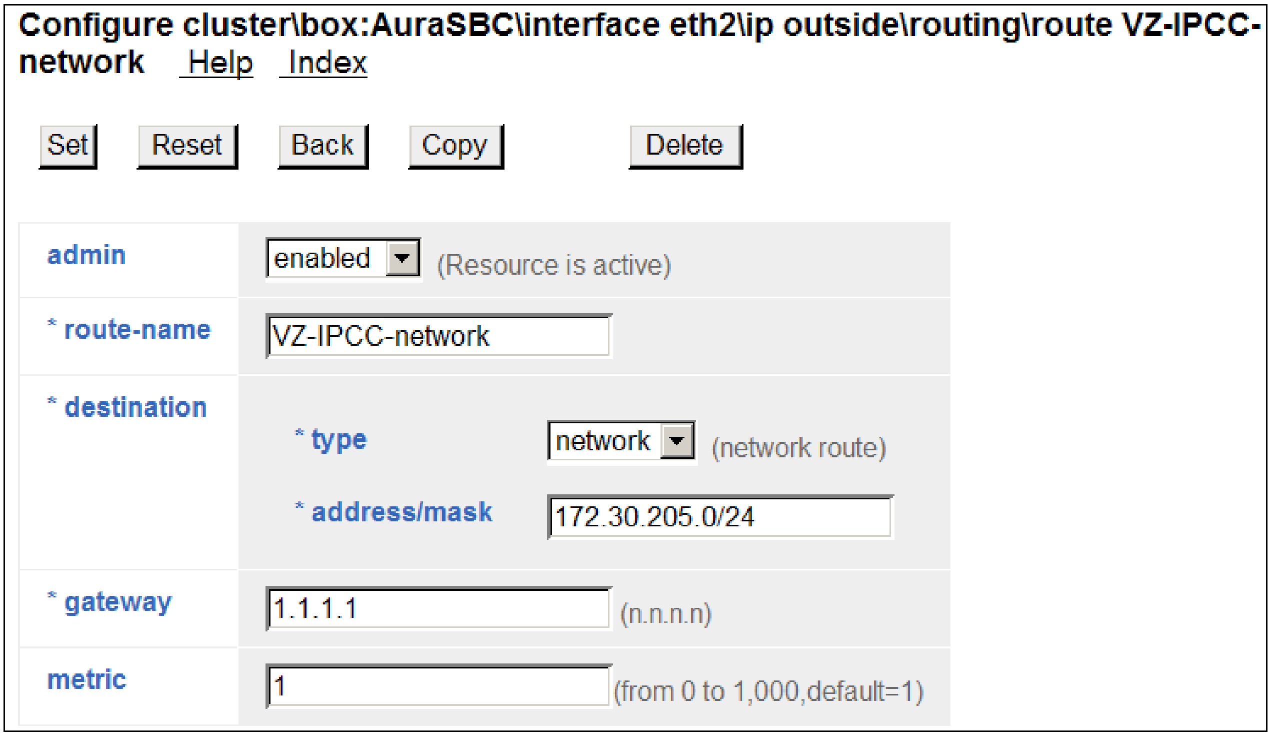
| | route | admin | destination | gateway | metric |
|---|---|---|---|---|---|
| Edit Delete | route Default | disabled | default | 0.0.0.0 | 1 |
| Edit Delete | route external-sip-media | enabled | network 172.30.209.0/24 | 1.1.1.1 | 1 |

Add route

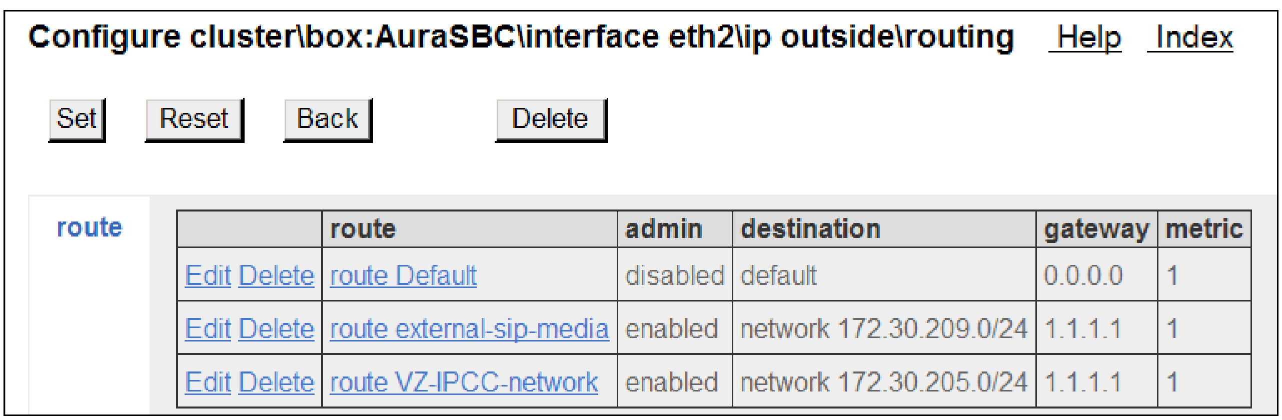
In the resultant screen shown below, enter an appropriate **route-name**. Using the **type** drop-down, select "network". In the **address/mask** field, enter the IP address and network mask associated with the Verizon IP Contact Center service. In the sample configuration, the Verizon IP Contact Center service uses 172.30.205.0/24 as shown below. In the **gateway** field, enter the IP address that is the gateway for the public side of the SBC to Verizon. In the sample configuration, the gateway is 1.1.1.1, the same gateway used with the Verizon IP Trunk service, since both share the same PIP access circuit. Click **Create**.



In the resultant screen shown below, click the **Set** button.

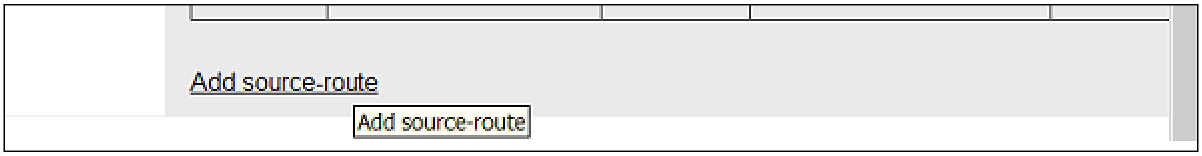



The following screen summarizes the updated routing configuration.

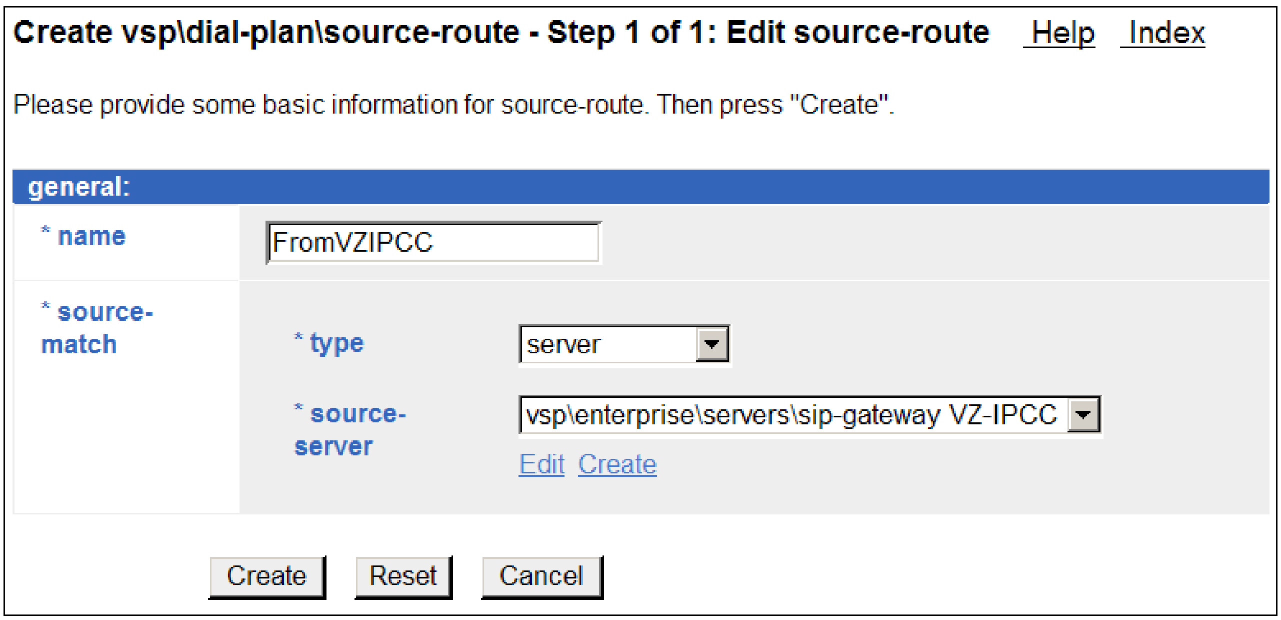


### 6.3.3 Configure Dial-Plan

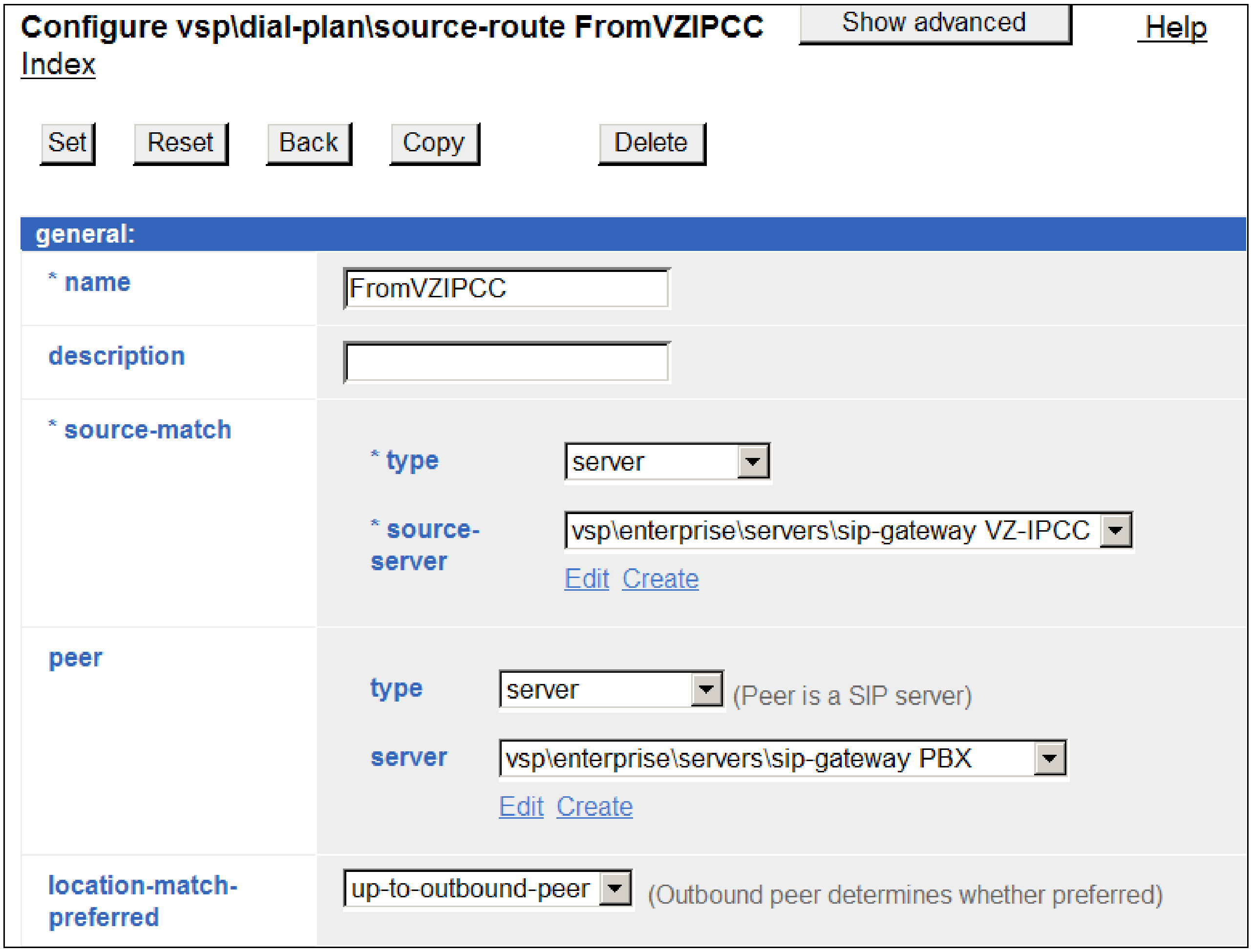
From the left-side menu, select **vsp → dial-plan.** In the right-hand side, scroll down and click **Add source-route** as shown below.



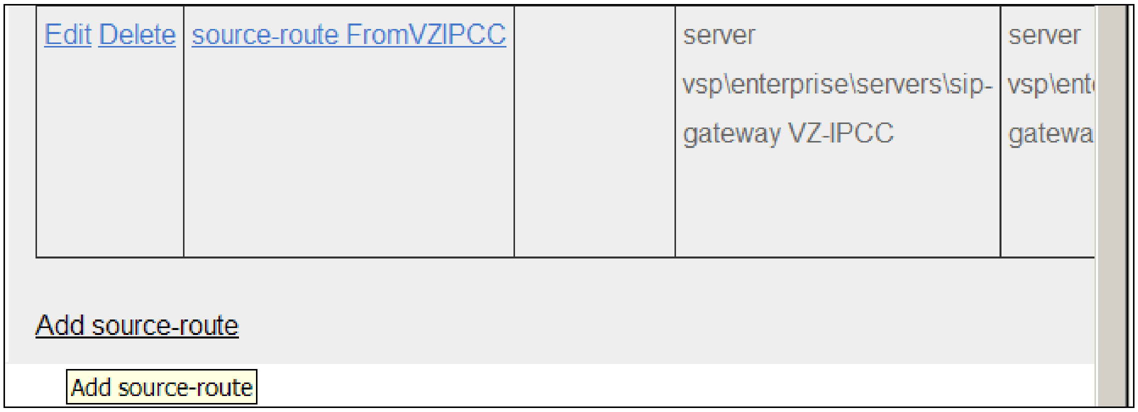

In the resultant screen, enter an appropriate name in the **name** field. In the **type** field drop-down menu, select “server”, and in the **source-server** drop-down menu, select the sip-gateway entry previously created in **Section 6.3.1**, as shown below. Click **Create**.



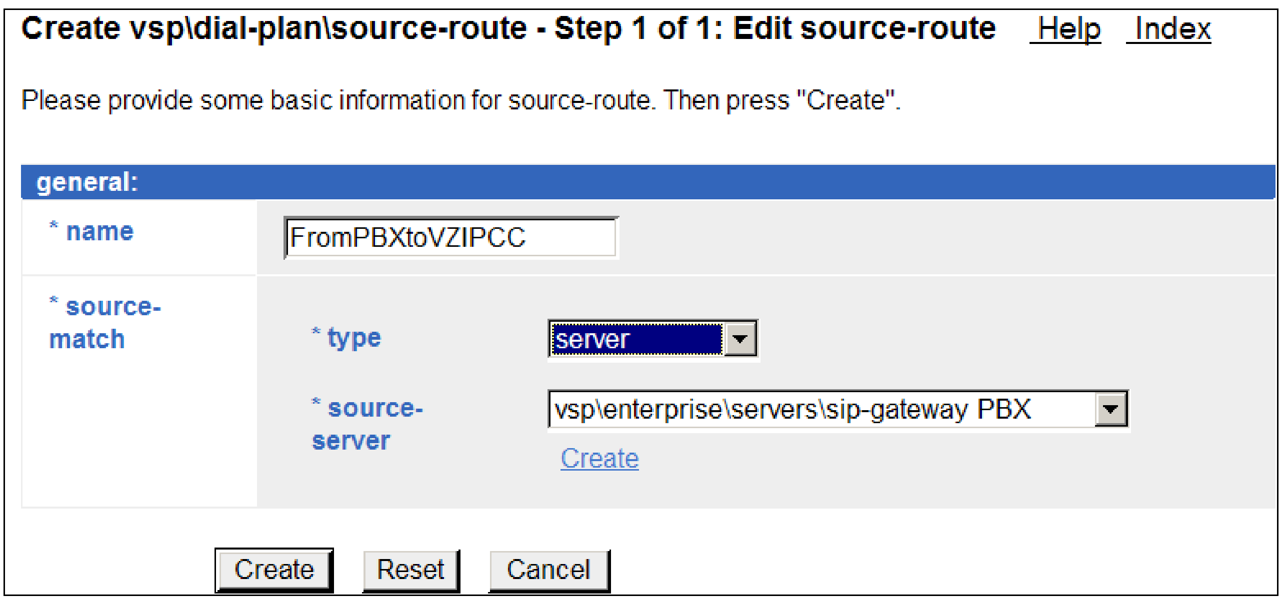
In the resultant screen, in the **peer** area, select “server” from the **type** drop-down. In the **server** drop-down, select the sip-gateway representing the enterprise SIP equipment. In the sample configuration, “vsp\enterprise\servers\sip-gateway PBX” already existed from the wizard configuration in reference [AuraSBC-IP-Trunk]. Incoming toll-free calls from the Verizon IP Contact Center service will route to Avaya Aura™ Session Manager as in reference [AuraSBC-IP-Trunk]. Click **Set**.



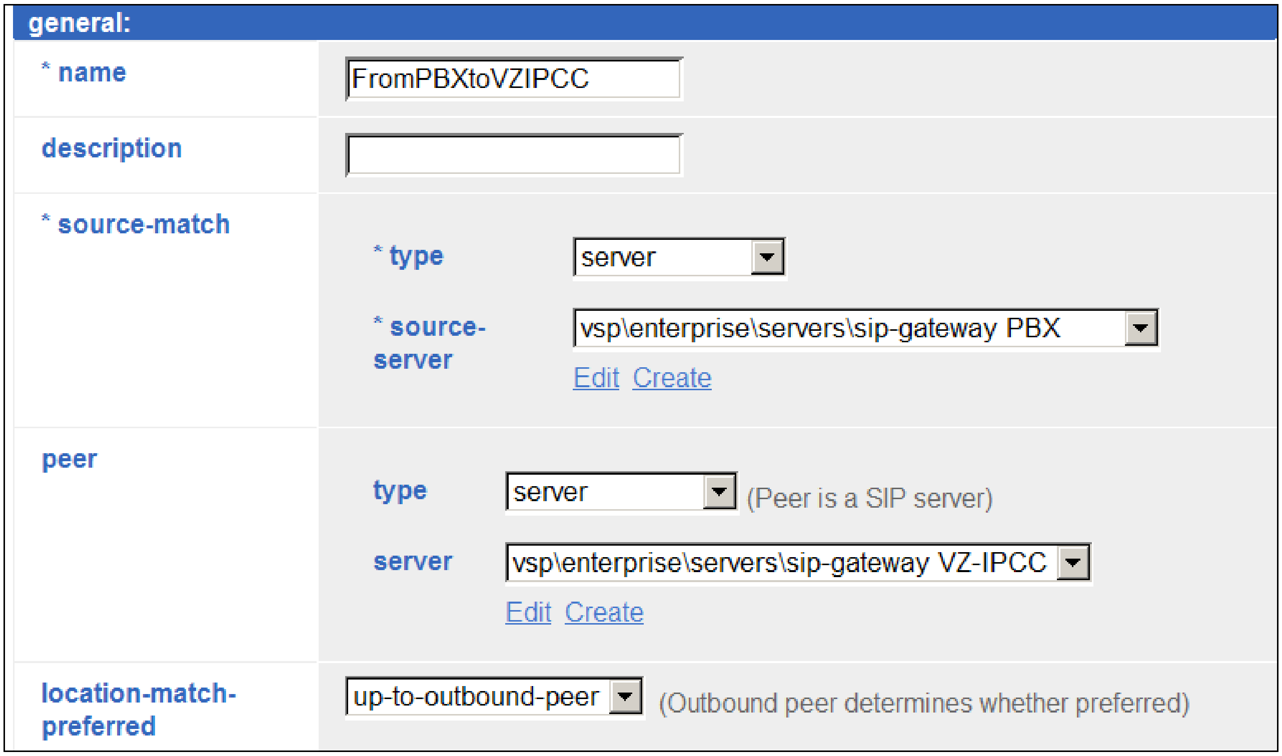
These same procedures can be repeated to create another source-route. Scroll down in the source-route area and click **Add source route** as shown below.




In the resultant screen, enter an appropriate name in the **name** field. Using the **type** drop-down menu, select “server”. Using the **source-server** drop-down, select the sip-gateway corresponding to the Avaya enterprise equipment. In the sample configuration, “vsp\enterprise\servers\sip-gateway PBX” is selected, which represents the connection to Avaya Aura™ Session Manager. Click **Create**.



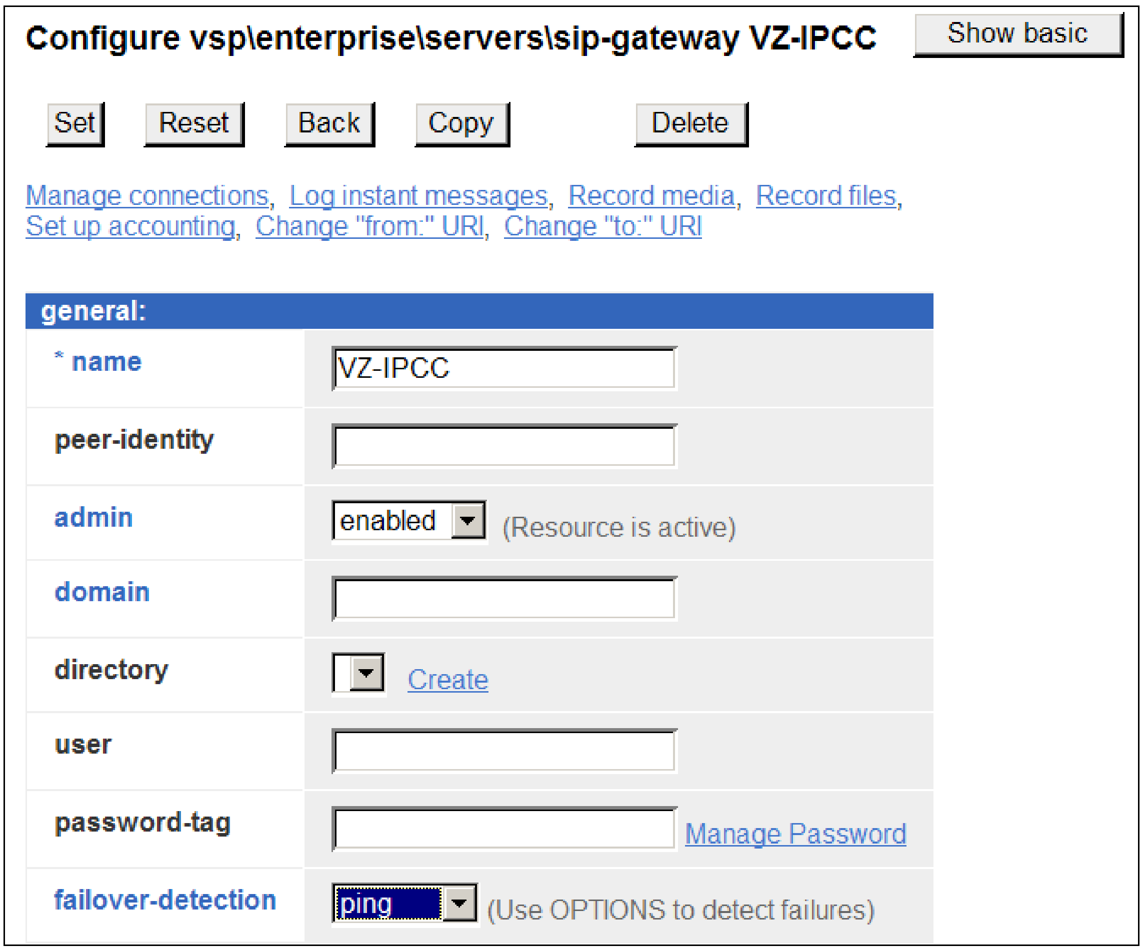
In the **peer** area, select "server" from the **type** drop-down. From the **server** drop-down, select the sip-gateway corresponding to the Verizon IP Contact Center service created in **Section 6.3.1**. Click **Set** (not shown).



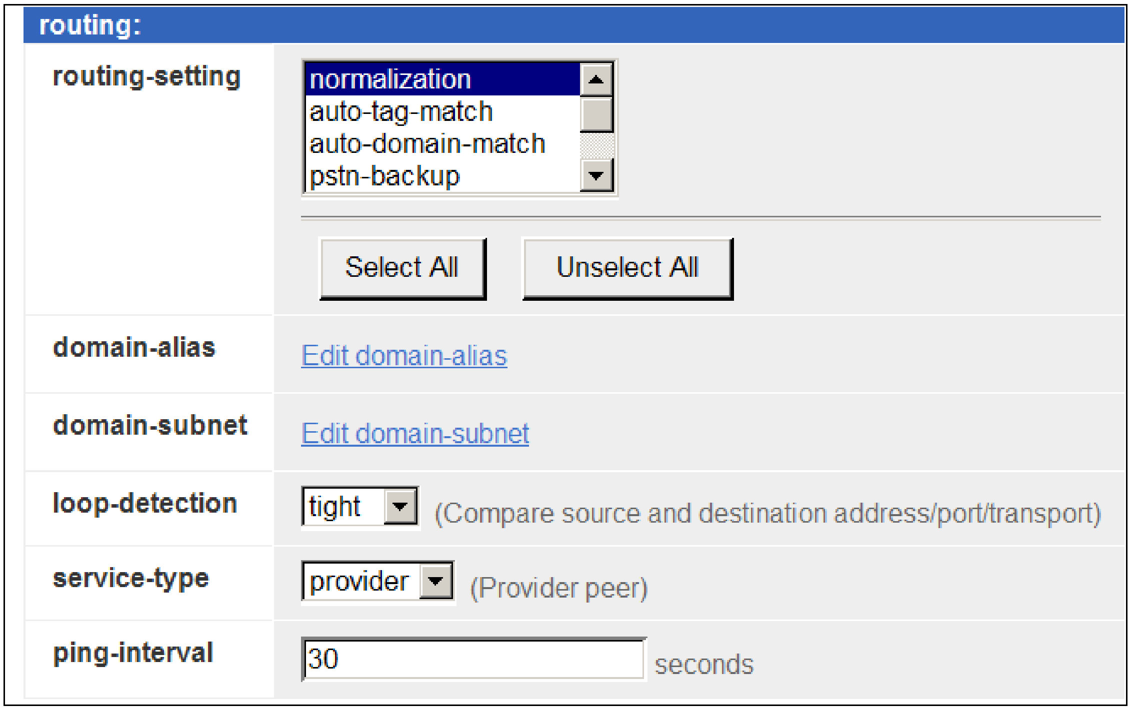
### 6.3.4 Configure OPTIONS ping to Verizon IP Contact Center

From the left-side menu, select **vsp → enterprise → servers → sip-gateway.** Select the sip-gateway to the Verizon IP Contact Center service added in **Section 6.3.1**. Click the **Show Advanced** button (not shown). In general, clicking this button reveals additional configuration parameters, and a **Show basic** button is presented, as shown below.

In the **failover-detection** drop-down, select "ping" as shown below.



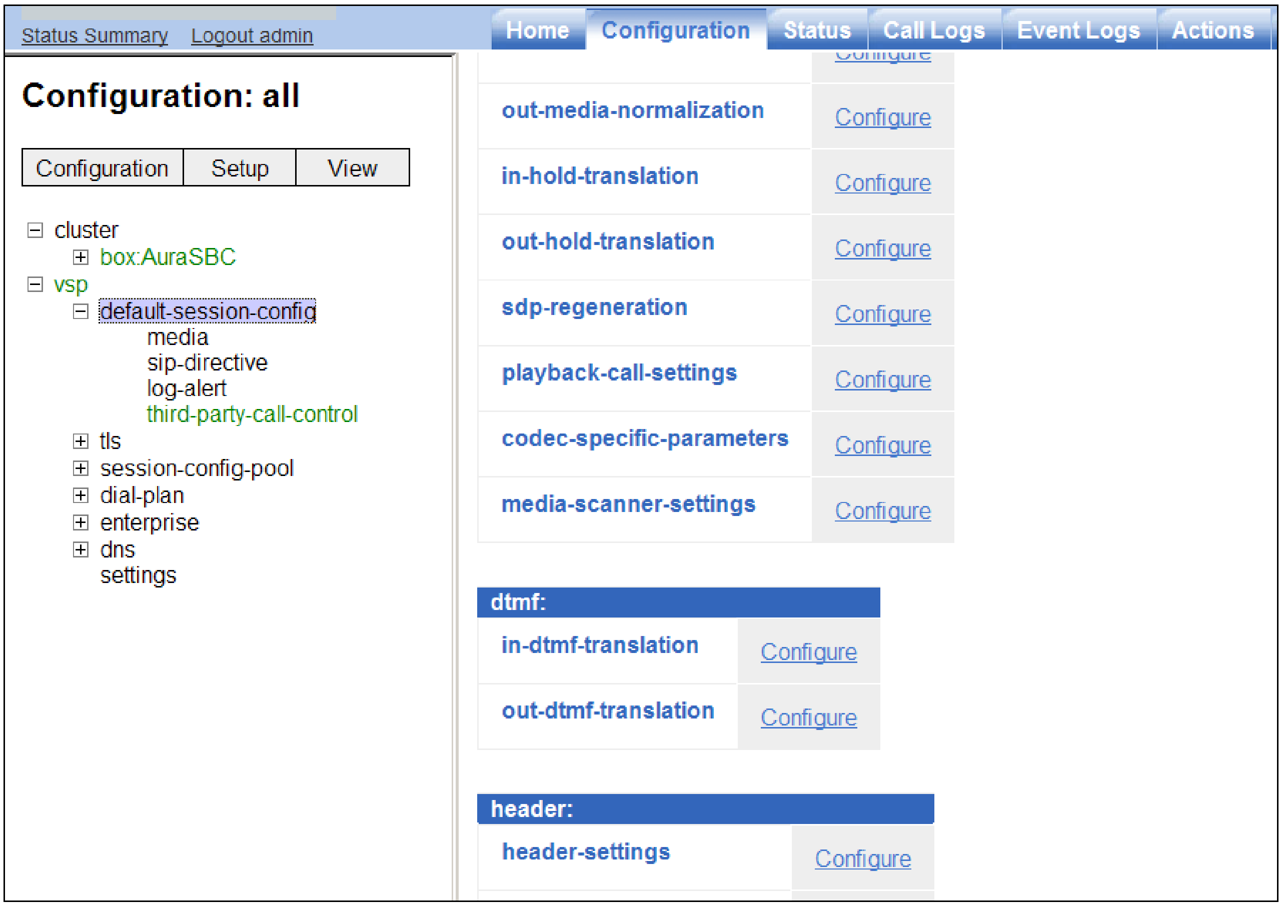
Scroll down and locate the **ping-interval** parameter, which is considered an "advanced" parameter (i.e., only available after the **Show Advanced** button has been clicked). Enter the desired period, in seconds, that the SBC will use to source SIP OPTIONS messages towards the Verizon IP Contact Center service. In the sample configuration shown below, the SBC will send OPTIONS every 30 seconds. This is not intended to be prescriptive; other intervals may be used.



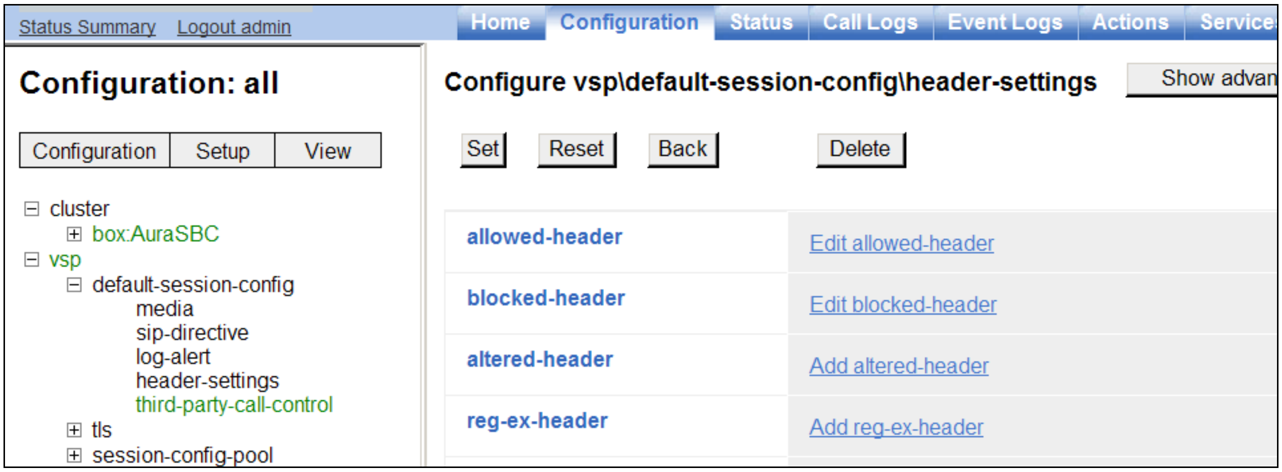
### 6.3.5 Stripping SIP Headers using P-Site as an Example

The SBC can be used to strip SIP headers. For headers that have relevance only within the enterprise, it may be desirable to prevent the header from being sent to the public SIP Service Provider. For example, Avaya Aura™ Session Manager Release 6 inserts the P-Site header. The following procedures may be used to strip the P-Site header.

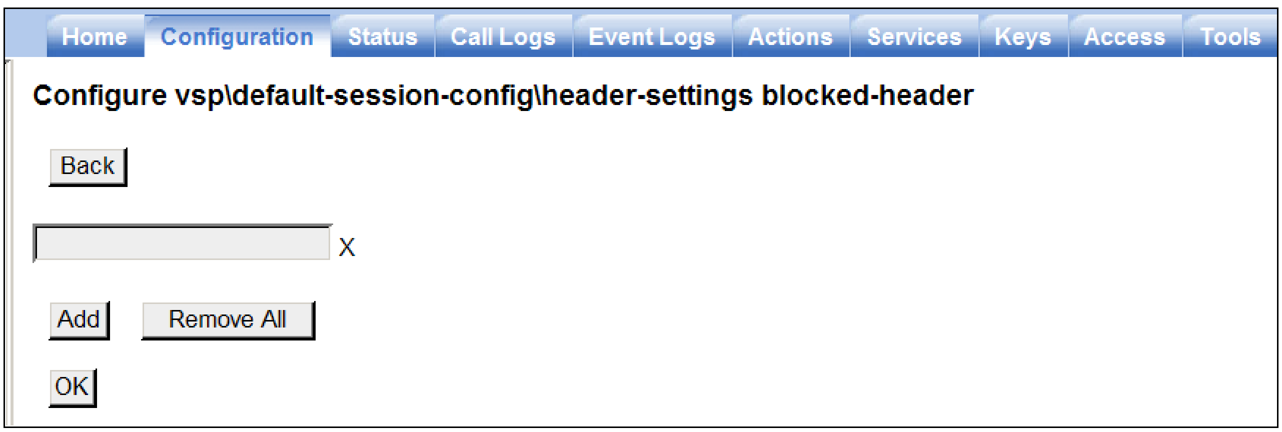
Select the **Configuration** tab. Using the menu on the left hand side, select **vsp → default-session-config.** Scroll down on the right and select **header-settings** as shown in the screen below.



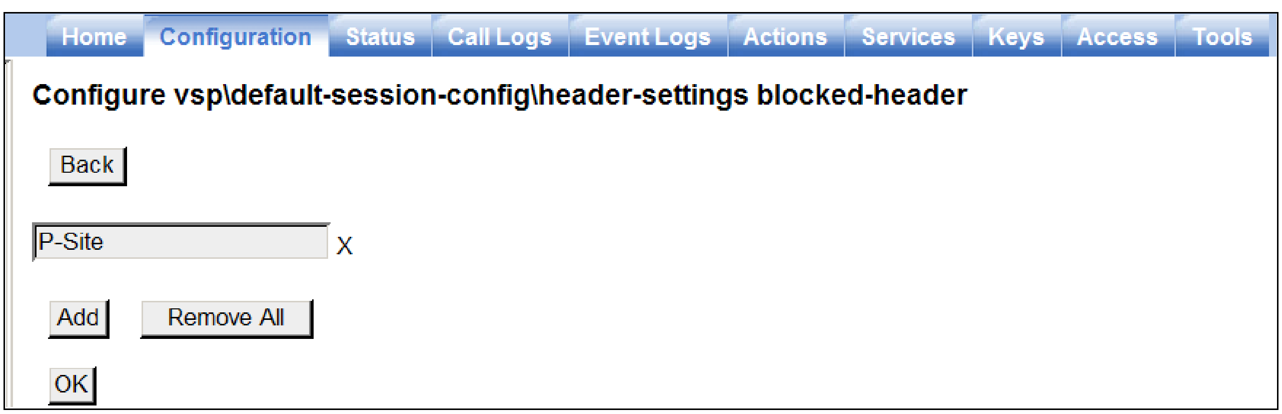
Select the **blocked-header** link on the right.



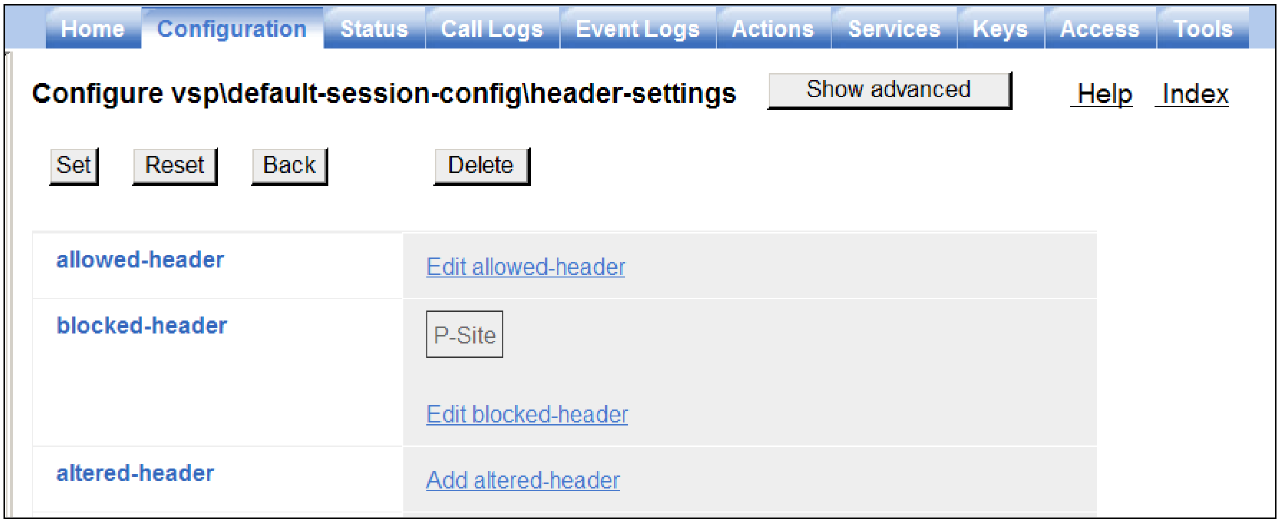

The following screen appears allowing configuration of the header to block.




To block the P-Site header, enter "P-Site" and click **OK** as shown in the screen below.



The following screen shows the resulting configuration.  The P-Site header is a blocked-header.

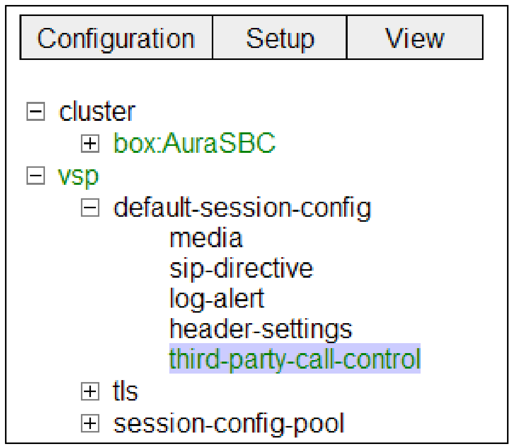



Similar procedures can be used to strip headers in a more specific session-config-pool.

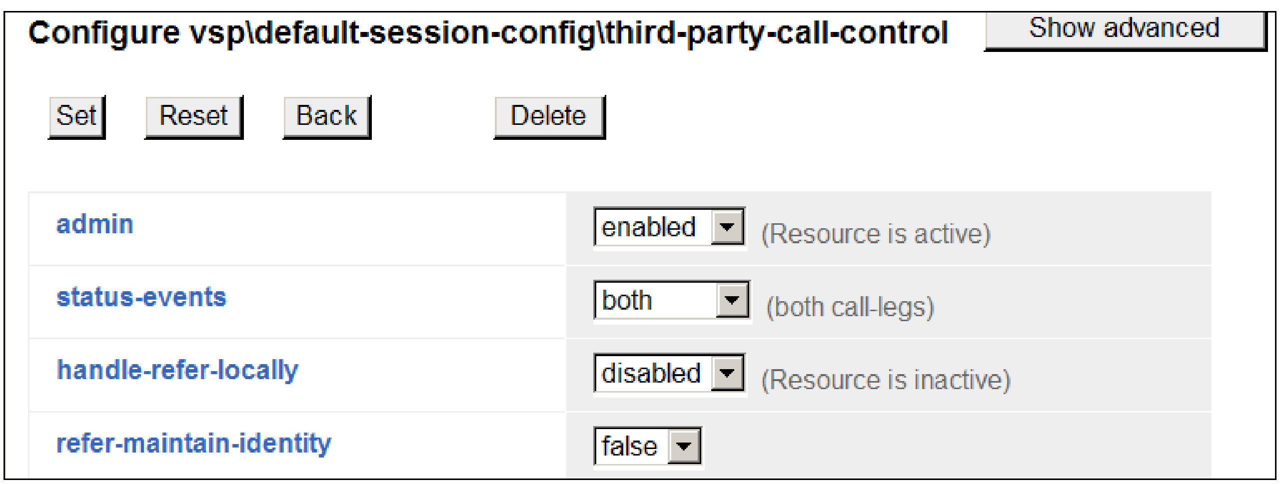
### 6.3.6 Use of REFER With Verizon

After running the installation wizard with the Verizon service provider profile as shown in **Section 5.1** of Reference [AuraSBC-IP-Trunk], the default configuration of the SBC will not use REFER messages towards Verizon. That is, REFER messages received from the private side of the SBC will result in INVITE messages on the public side to Verizon. This section shows how the configuration can be changed to enable the use of REFER messages towards Verizon.

To cause a REFER sent by Communication Manager to result in a REFER sent to Verizon, the following change can be made to the SBC. Navigate to **vsp → default-session-config → third-party-call-control** as shown below.



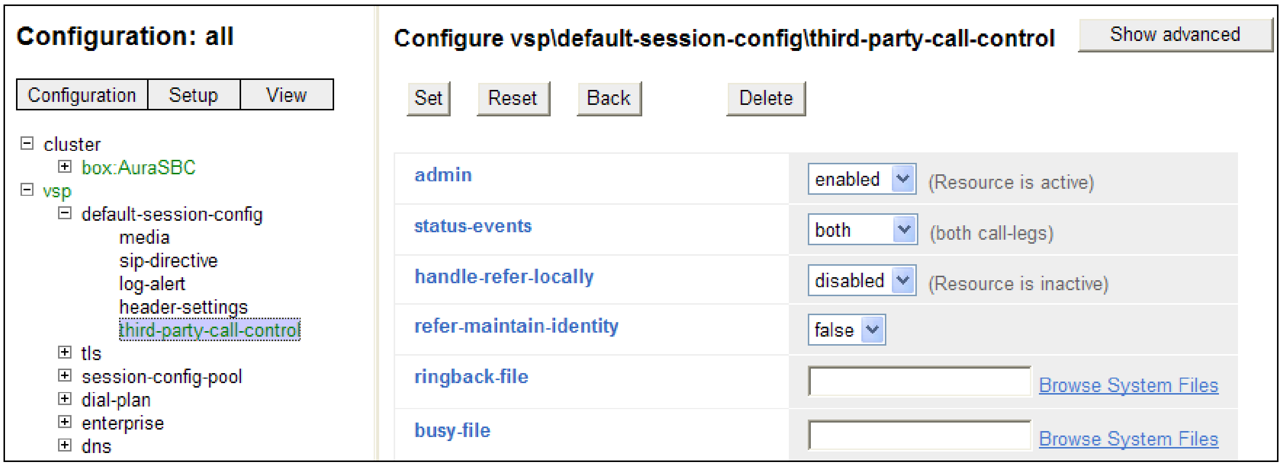

On the right, select "disabled" from the **handle-refer-locally** drop-down menu. Click the **Set** button. Proceed to save and activate the configuration as described in **Section 6.4**.



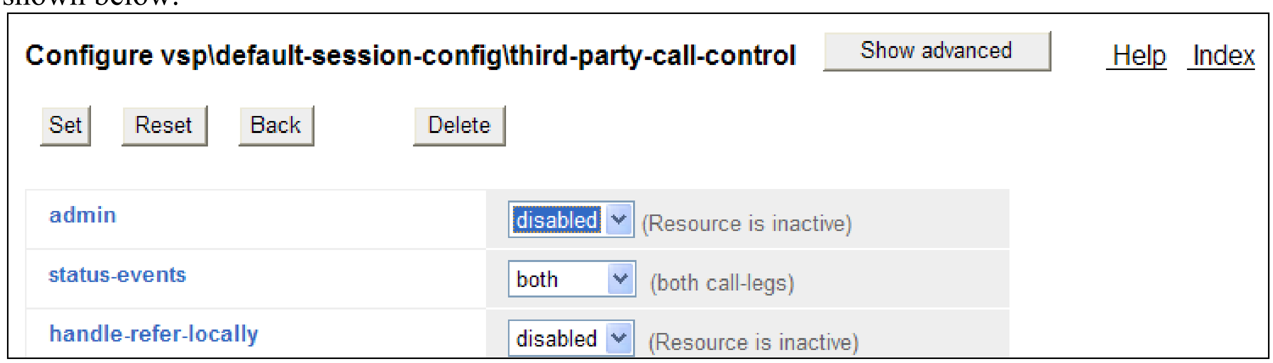
### 6.3.7 Disabling Third Party Call Control

The installation wizard for Verizon in the release documented in these Application Notes will enable the **admin** field for third party call control.

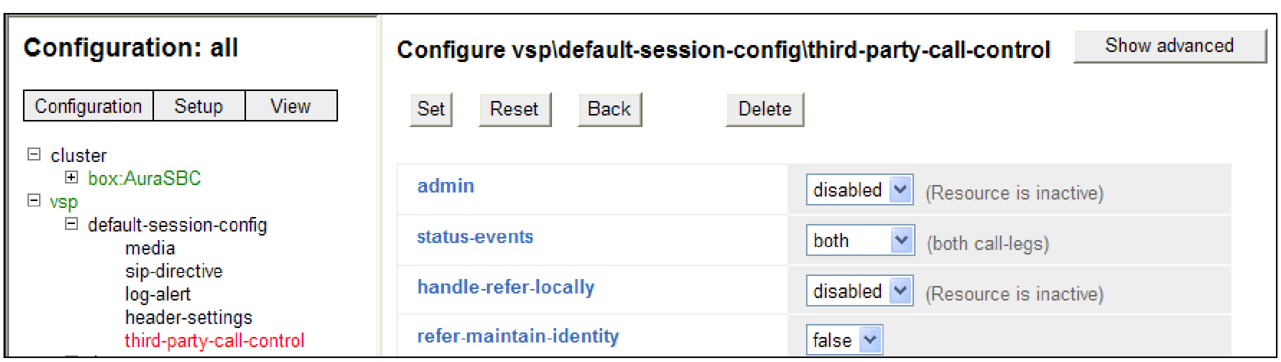
Navigate to **vsp → default-session-config → third-party-call-control**. As shown below, the installation wizard in the release covered by these Application Notes sets the **admin** field to enabled.




To disable third-party-call-control, select disabled from the **admin** drop-down and click **Set** as shown below.




After disabling, the third-party-call-control link becomes red as shown below.

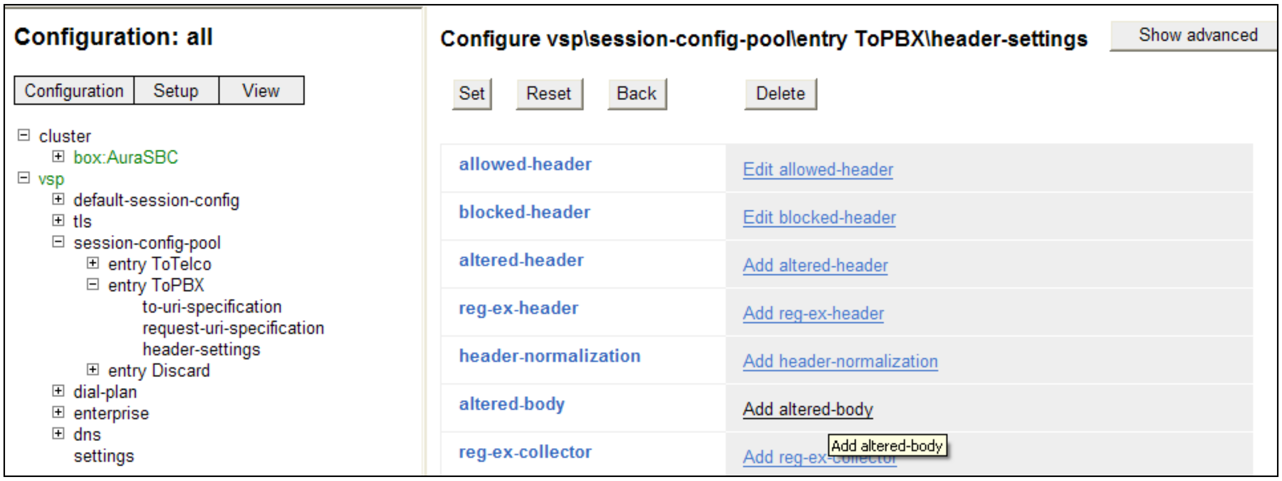


Proceed to save and activate the configuration as described in **Section 6.4**.

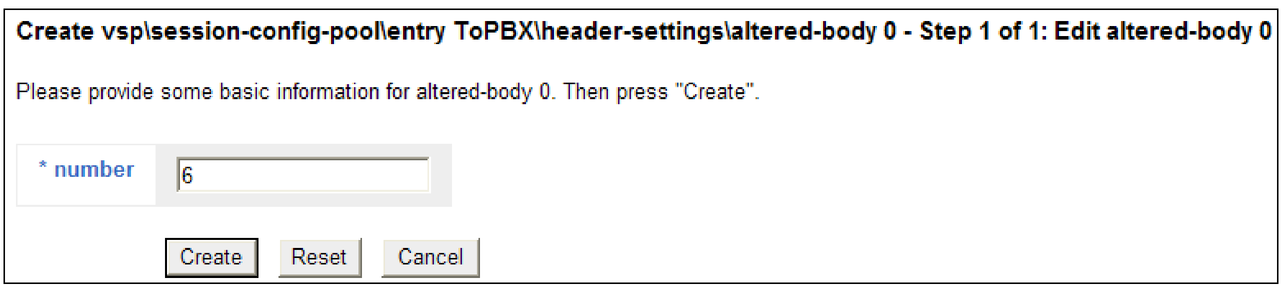
### 6.3.8 SDP Modification From Sendonly to Sendrecv

In **Section 1.3**, potential problems are described that can be avoided by implementing the SIP header manipulation described in this section. This manipulation will replace "sendonly" with "sendrecv" in the SDP. With this manipulation configured and activated, Verizon will not receive "sendonly" in the SDP from the enterprise site, avoiding a specific Verizon response that can lead to a subsequent loss of media paths.

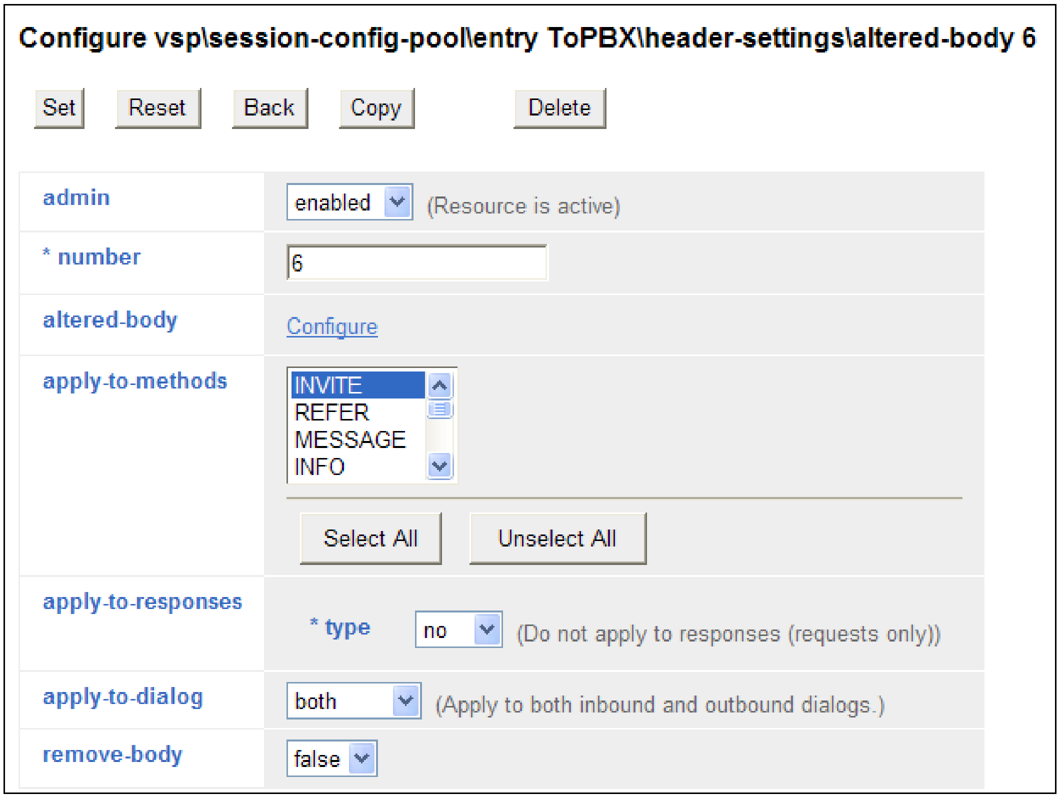
In the left side menu, navigate to **vsp → session-config-pool → entry ToPBX → header-settings**. On the right panel, select **Add altered-body** as shown below.



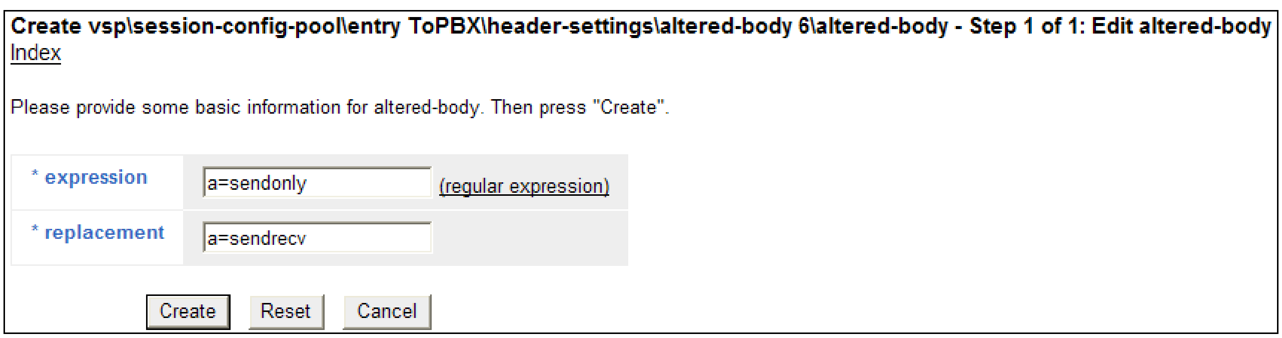

In the resultant screen, enter a **number** as shown below, then click **Create**.



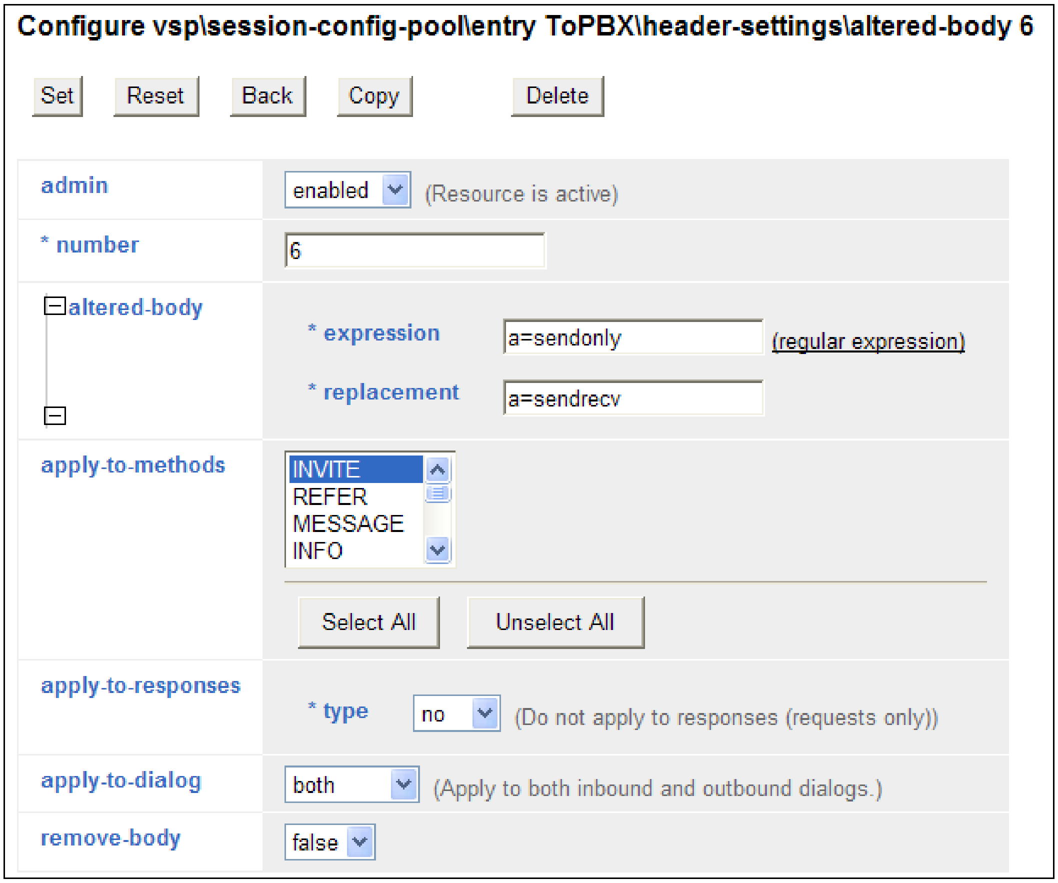
The resultant screen is presented below. Retain the default parameters, and click **Configure** next to altered-body.




In the resultant screen, enter "a=sendonly" for the **expression**, and "a=sendrecv" for the **replacement**, as shown below. Click **Create**.



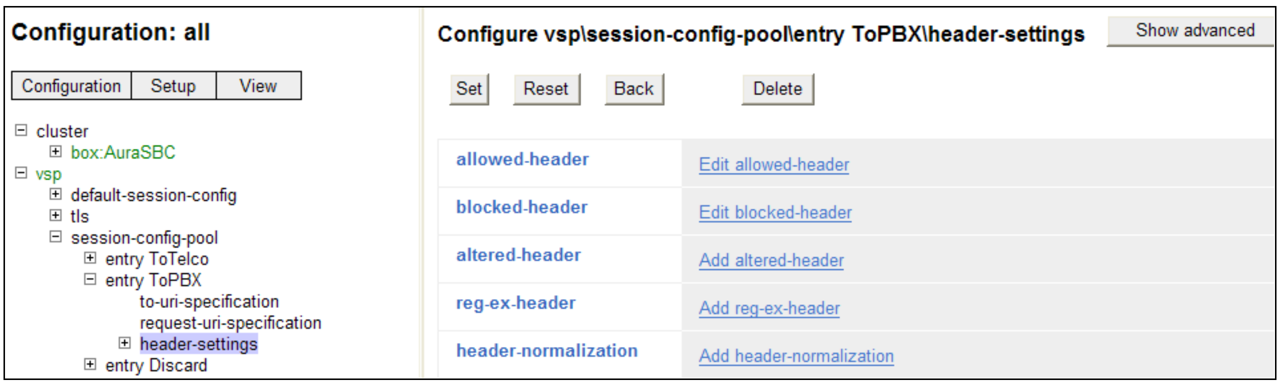
The new altered-body is summarized below. Click the **Set** button. Proceed to save and activate the configuration as described in **Section 6.4**.



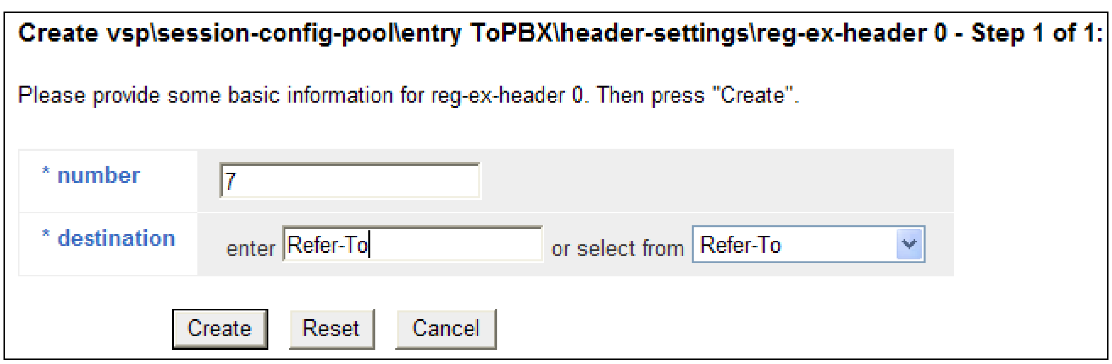
### 6.3.9 Refer-To Header in REFER Message

This section presents a sample configuration that will cause the SBC to modify the host portion of the Refer-To header in a REFER message, while preserving the user portion (containing the Refer-To destination telephone number) and any User-User Information. In this example, the host portion was changed such that Verizon would receive the Verizon IPCC service IP Address and port as the host portion. On the production circuit used to verify these Application Notes, this header manipulation was not strictly required. That is, Verizon would route the call to the Refer-To destination even if the host portion contained contents other than the Verizon IPCC service IP Address and port.

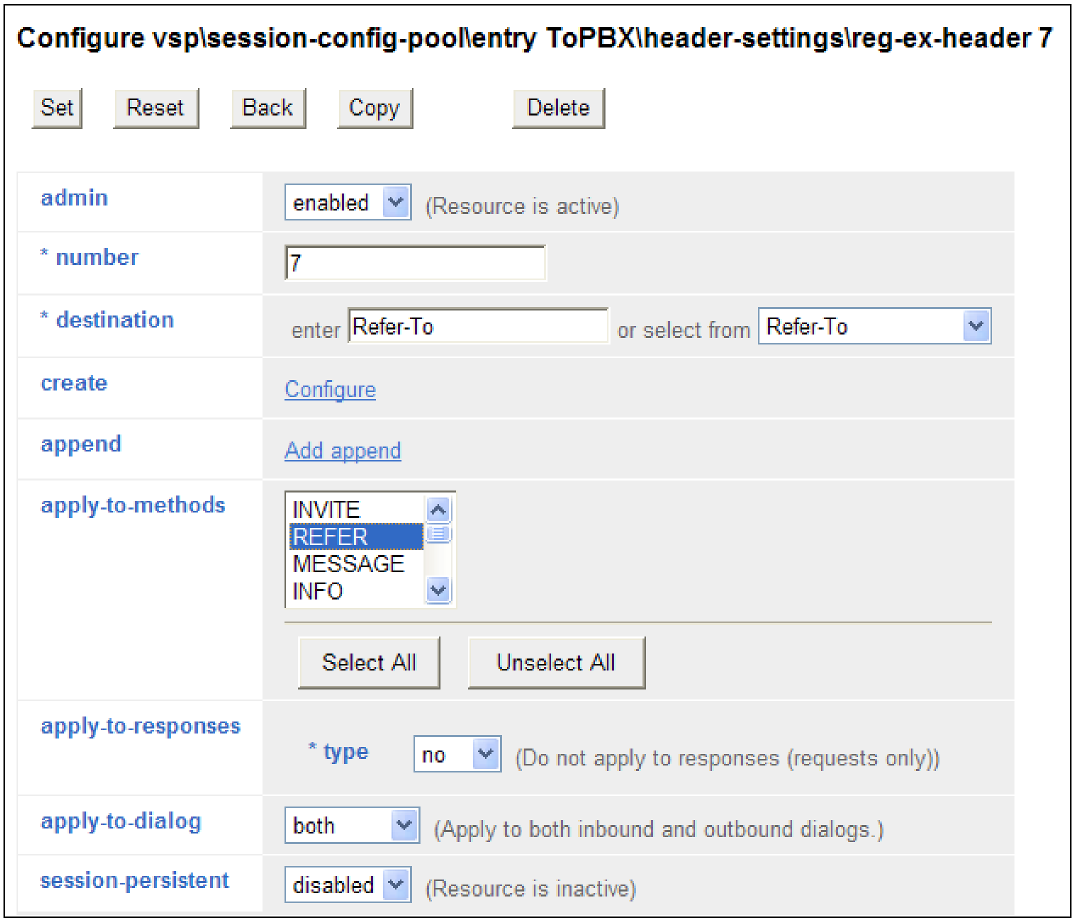
In the left side menu, navigate to **vsp → session-config-pool → entry ToPBX → header-settings**. On the right panel, select **Add reg-ex-header** as shown below.




In the resultant screen, enter a number in the **number** field and enter "Refer-To" as the **destination** as shown in the example screen below. Click **Create**.



In the resultant screen, select "REFER" for **apply-to-methods** as shown in the screen below. Select the **Configure** link to the right of **create**.

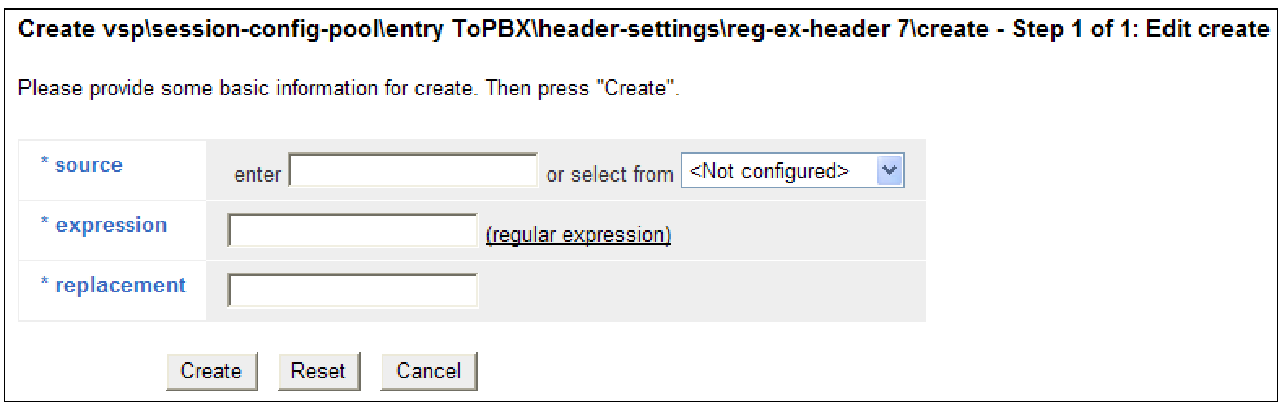


The following screen is presented. In the **source** area, select “Refer-To” from the drop-down list or type “Refer-To” in the **enter** field.

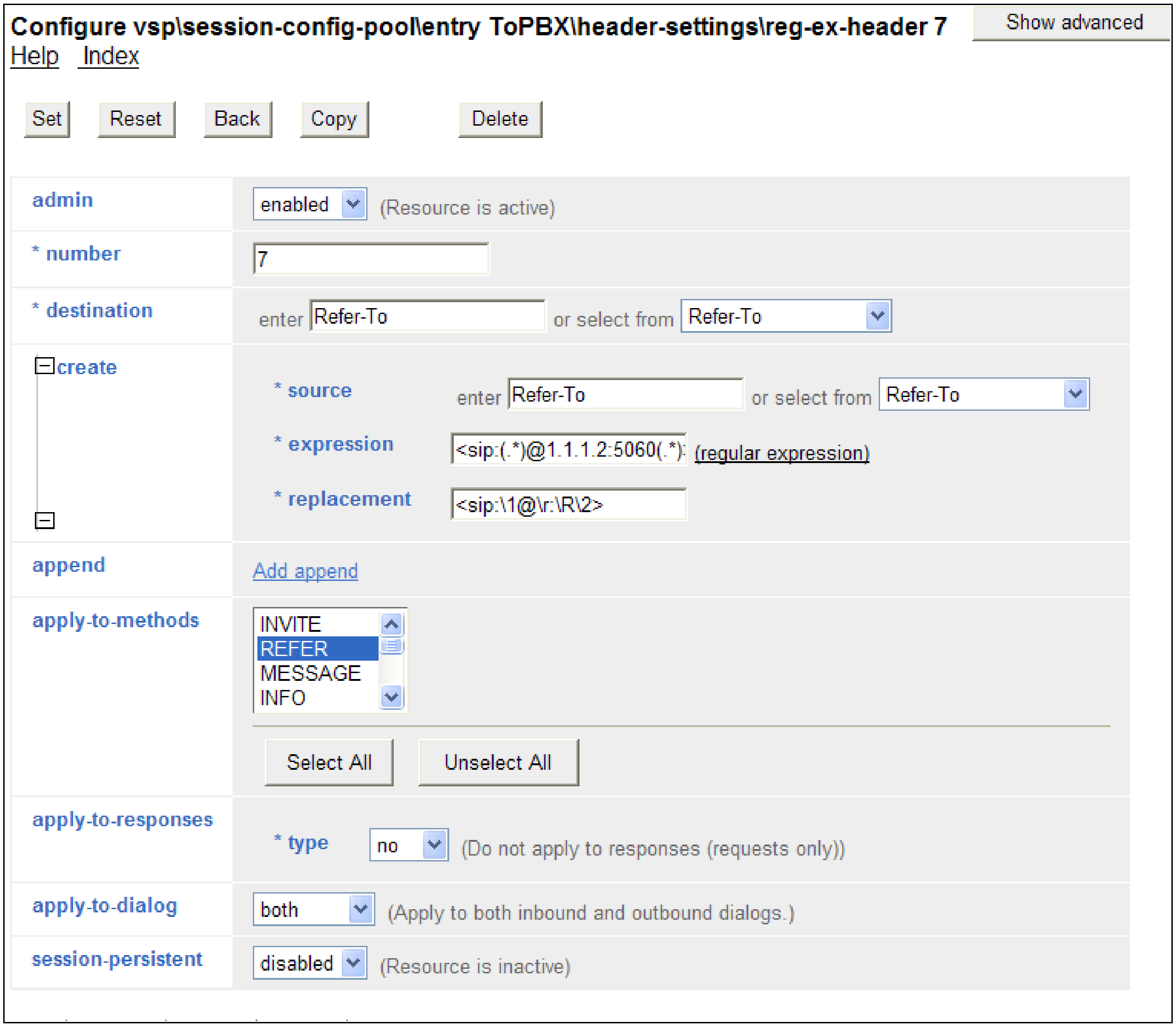
In the **expression** field, enter a regular expression to match. In the sample configuration, “<sip:(.*)@1.1.1.2:5060(.*)>” was entered. In this expression, the first (.*) will match and store any user part of the Refer-To header, meaning that any Refer-To destination number will match and be stored. The second instance of (.*) matches and stores any UUI if present. The address “1.1.1.2:5060” is what the SBC would otherwise put in the Refer-To header host part, and is an example of the statement earlier that the Refer-To header manipulation in this section was not strictly necessary on the Verizon production circuit used for testing.

In the **replacement** field, “<sip:\1@\r:\R\2>” was entered in the sample configuration. The variable “\1” is the stored user part from the original Refer-To header containing the Refer-To number, corresponding to the first instance of “(.*)” from the **expression**. The variable “\2” is any stored UUI from the original Refer-To header, corresponding to the second instance of “(.*)” from the **expression**. The “\r” inserts the “remote IP Address” corresponding to the Verizon IPCC Service IP Address, which is 173.30.205.55. This is followed by a colon and “\R” corresponding to the Verizon IPCC SIP signaling port, which is 5072 in this case.

After completing the **source**, **expression** and **replacement** fields as appropriate, click **Create**.



The following screen shows the completed rule. Click the **Set** button. Proceed to save and activate the configuration as described in **Section 6.4**.




With this rule activated, an example Refer-To header sent to Verizon for a Refer-To with UUI is as follows. See **Section 9.1.3** for additional detailed verification information. The following is a portion of a bitmap image from a Wireshark trace for such a call:

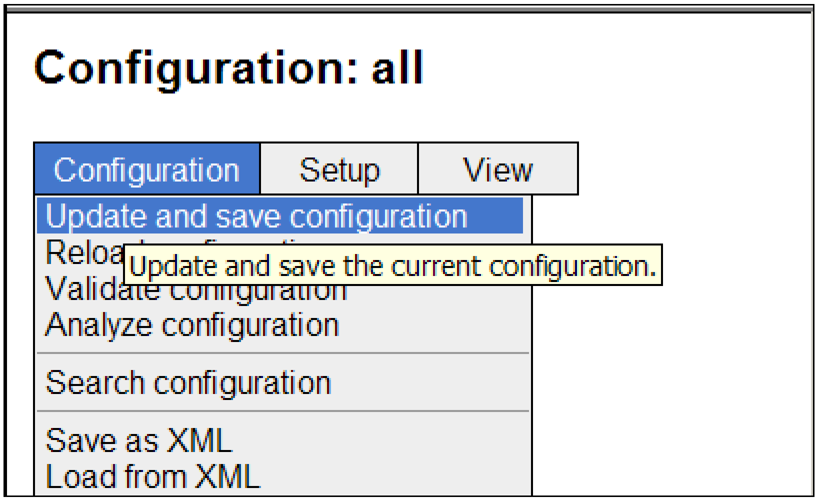
```
Refer-To: <sip:+18668512649@172.30.205.55:5072?User-to-User=0431323334353637383930313233343536373839303132
```

With this rule activated, an example Refer-To header sent to Verizon for a Refer-To without UUI (e.g., to a PSTN destination) is as follows. See **Section 9.1.2** for additional detailed verification information. The following is a portion of a bitmap image from a Wireshark trace for such a call:

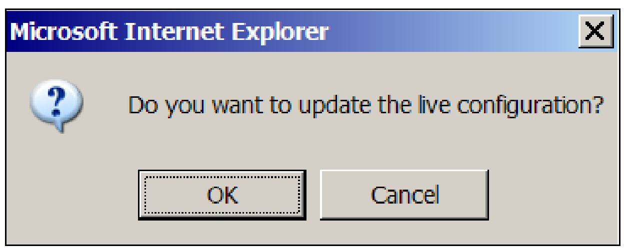
```
Refer-To: <sip:+17326870755@172.30.205.55:5072>
```

## 6.4. Saving and Activating Configuration Changes

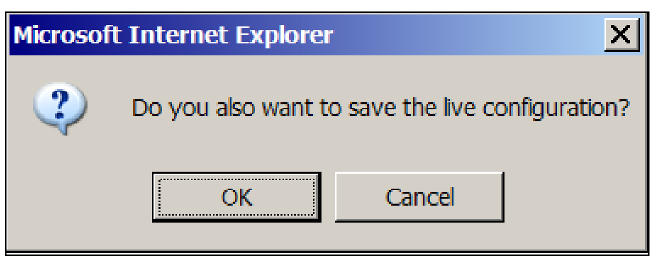
To save and activate configuration changes, select **Configuration → Update and save configuration** from the upper left hand side of the user interface, as shown below.



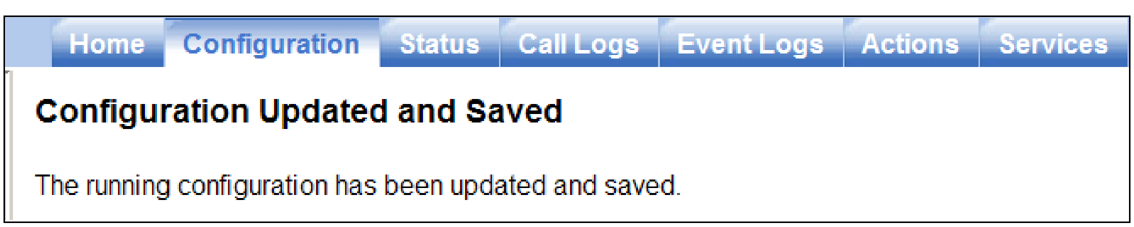

Click **OK** to update the live configuration.




Click **OK** to save the live configuration.




A screen that includes the following should appear.



# 7. Verizon Business IPCC Services Suite Configuration

Information regarding Verizon Business IPCC Services suite offer can be found at http://www.verizonbusiness.com/products/contactcenter/ip/ or by contacting a Verizon Business sales representative.

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

## 7.1. Service access information

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

| CPE (Avaya) | Verizon Network |
|---|---|
| ***adevc.avaya.globalipcom.com***<br>***UDP port 5060*** | ***172.30.205.55***<br>***UDP Port 5072*** |

| Toll Free Numbers |
|---|
| 866-850-2380 |
| 866-851-0107 |
| 866-851-2649 |
| 866-852-3221 |
| 866-850-6850 |

# 8. General Test Approach and Test Results

The test approach was manual testing of inbound and referred calls using the Verizon IPCC Services on a production Verizon PIP access circuit, as shown in **Figure 1**.

The main test objectives were to verify the following features and functionality:

- Inbound Verizon toll-free calls to Communication Manager telephones and VDNs/Vectors
- Inbound private toll-free calls (e.g., PSTN caller uses *67 followed by the toll-free number)
- Inbound Verizon toll-free calls redirected using Communication Manager SIP NCR (via SIP REFER/Refer-To) to PSTN alternate destinations
- Inbound Verizon toll-free calls redirected using Communication Manager SIP NCR with UUI (via SIP REFER/Refer-To with UUI) to a SIP-connected destination
- Inbound toll-free voice calls can use G.711MU or G.729A codecs.
- Inbound toll-free voice calls can use DTMF using RFC 2833
- Inbound toll-free voice calls to Communication Manager stations can be covered to Avaya Modular Messaging.

Testing was successful, except as noted in the limitations described in **Section 1.3.**

Examples of representative verified call scenarios are detailed in **Section 9**.

# 9. Verification Steps

This section provides example verifications of the sample configuration illustrated in these Application Notes.

## 9.1. Avaya Aura™ Communication Manager and Wireshark Verifications

This section illustrates verifications using Avaya Aura™ Communication Manager and Wireshark to illustrate key SIP messaging.

### 9.1.1 Example Incoming Call from PSTN via Verizon SIP Trunk

Incoming toll-free calls arrive from Verizon at the SBC, which sends the call to Avaya Aura™ Session Manager.  Session Manager sends the call to Avaya Aura™ Communication Manager via the entity link corresponding to the Avaya S8800 PE using port 5062.  On Communication Manager, the incoming call arrives via signaling group 67 and trunk group 67.

The following abridged Communication Manager "list trace" trace output shows a call incoming on trunk group 67.  The PSTN telephone dialed 866-851-2649.  Session Manager can map the number received from Verizon to the extension of a Communication Manager telephone (x30002), or the incoming call handling table for trunk group 67 can do the same.  In the trace below, Session Manager had already mapped the Verizon number to the Communication Manager extension. Extension 30002 is an IP Telephone with IP Address 65.206.67.11 in Region 4.  Initially, the G450 Media Gateway (10.1.2.95) is used, but as can be seen in the final trace output, once the call is answered, the final RTP media path is "ip-direct" from the IP Telephone (65.206.67.11) to the "inside" of the SBC (65.206.67.93).

In Communication Manager Release 6, the tracing prints the Communication Manager release version at the start of the trace, and intersperses the SIP messaging with the Communication Manager processing.

```
list trace tac 167                                                  Page   1
                                LIST TRACE
time            data
12:57:13 TRACE STARTED 09/08/2010 CM Release String cold-00.0.345.0-18444
12:58:34 SIP<INVITE sip:30002@avaya.com:5060 SIP/2.0
12:58:34     active trunk-group 67 member 1  cid 0x1fd
12:58:34 SIP>SIP/2.0 183 Session Progress
12:58:34     dial 30002
12:58:34     ring station    30002 cid 0x1fd
12:58:34     G729A ss:off ps:20
             rgn:4 [65.206.67.11]:2504
             rgn:1 [10.1.2.95]:2050
12:58:34     G729 ss:off ps:20
             rgn:4 [65.206.67.93]:21624
             rgn:1 [10.1.2.95]:2104
12:58:34     xoip options: fax:off modem:off tty:US  uid:0x500f1
             xoip ip: [10.1.2.95]:2104
12:58:41 SIP>SIP/2.0 200 OK
12:58:41     active station    30002 cid 0x1fd
12:58:41 SIP<ACK sip:8668512649@10.1.2.90:5062;transport=tcp SIP
12:58:41 SIP</2.0
12:58:41 SIP>INVITE sip:+19088485704@199.173.95.16:5060;transpor
12:58:41 SIP>t=tcp;maddr=65.206.67.93 SIP/2.0
12:58:41 SIP<SIP/2.0 100 Trying
12:58:41 SIP<SIP/2.0 200 OK
12:58:41 SIP>ACK sip:+19088485704@199.173.95.16:5060;transport=t
12:58:41 SIP>cp;maddr=65.206.67.93 SIP/2.0
12:58:41     G729A ss:off ps:20
             rgn:4 [65.206.67.93]:21624
             rgn:4 [65.206.67.11]:2504
12:58:41     G729 ss:off ps:20
             rgn:4 [65.206.67.11]:2504
             rgn:4 [65.206.67.93]:21624
```

The following screen shows **Page 2** of the output of the "status trunk" command pertaining to the same call. Note the signaling using port 5062 between Communication Manager and Session Manager. Note the media is "ip-direct" from the IP Telephone (65.206.67.11) to the inside IP Address of the SBC (65.206.67.93) using G.729.

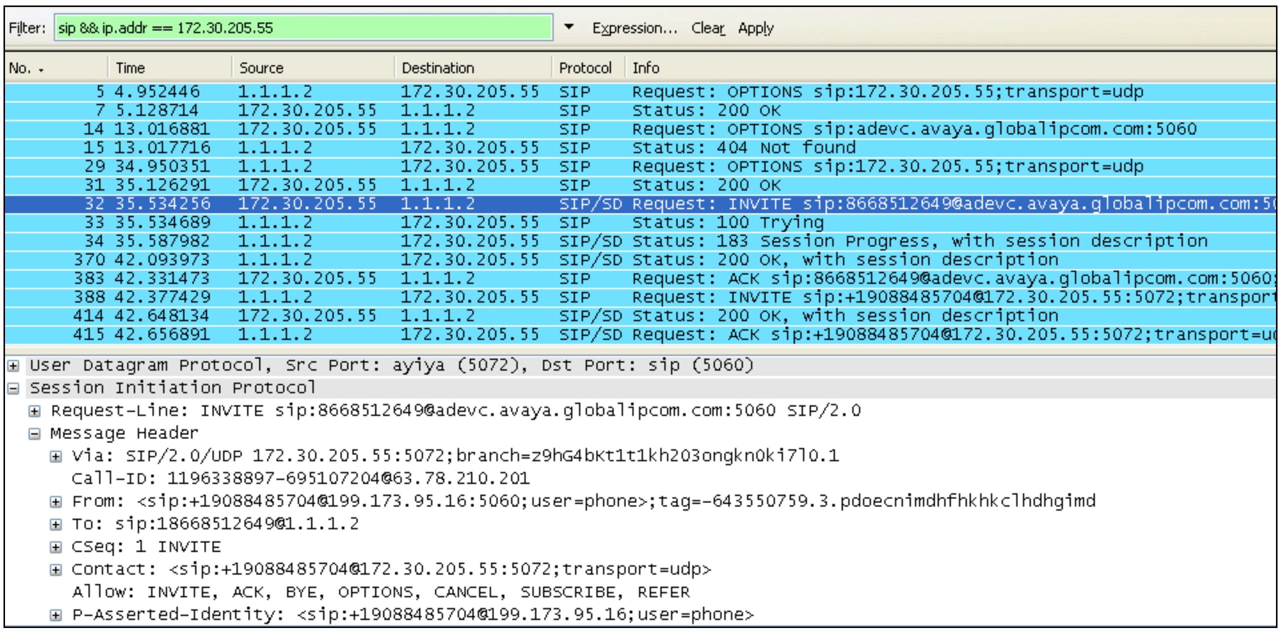
```
status trunk 67/1                                               Page   2 of   3
                                CALL CONTROL SIGNALING

Near-end Signaling Loc: PROCR
  Signaling   IP Address                               Port
   Near-end:  10.1.2.90                              : 5062
    Far-end:  10.1.2.70                              : 5062
 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.729
   Audio      IP Address                               Port
   Near-end:  65.206.67.11                           : 2504
    Far-end:  65.206.67.93                           : 21624
```

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 is used.

```
status trunk 67/1                                               Page   3 of   3
                      SRC PORT TO DEST PORT TALKPATH
src port: T00241
T00241:TX:65.206.67.93:21624/g729/20ms
S00038:RX:65.206.67.11:2504/g729a/20ms
dst port: S00038
```

The following portion of a filtered Wireshark trace (tracing only SIP messages on the public interface on the “outside” of the SBC) shows the same incoming PSTN call. In frame 32, Verizon sends the INVITE to the SBC (1.1.1.2). Frame 32 is selected and expanded so that the middle portion of the screen can illustrate the contents of the R-URI, From, To, Contact, and PAI headers sent by Verizon. The trace shows that the SIP message uses UDP with source port 5072 and destination port 5060. The subsequent call processing of this call will be illustrated in the context of the “inside” trace analysis (private side of SBC) that follows.

Note that this trace also shows exchanges of SIP OPTIONS messages. In frame 14, Verizon sends OPTIONS, and the SBC responds with 404 Not Found in frame 15. This SIP response is sufficient for Verizon to keep the link in-service. In frame 29, the SBC sends OPTIONS, and Verizon responds with 200 OK in frame 31.



The following portion of a filtered Wireshark trace (tracing SIP messages on the private inside interface of the SBC only) shows the same incoming toll-free call. In frame 1878, the inside interface of the SBC (65.206.67.93) sends an INVITE to Session Manager (10.1.2.70). In highlighted frame 1882, Session Manager sends the INVITE to the S8800 PE (10.1.2.90). Observe that Session Manager has already adapted the Verizon toll-free number to its corresponding Communication Manager extension (30002). In the center portion, observe the use of TCP and destination port 5062 on the S8800 PE. Communication Manager can apply Verizon-appropriate behaviors, such as the use of 183 with SDP, since it can distinguish that the call is inbound from Verizon by the use of port 5062 (i.e., arriving from the same Session Manager as other non-Verizon traffic). In frame 1887, Communication Manager sends a 183 Session Progress with SDP. Note that in prior releases of Communication Manager, a 180 with SDP would have been sent, but enhancements in Communication Manager Release 6 allow a 183 with SDP to be configured to be sent, as desired by Verizon.

Filter: sip  Expression... Clear Apply

| No. | Time | Source | Destination | Protocol | Info |
|---|---|---|---|---|---|
| 1878 | 68.863637 | 65.206.67.93 | 10.1.2.70 | SIP/SD | Request: INVITE sip:8668512649@adevc.avaya.globalipcom.com:50 |
| 1879 | 68.865428 | 10.1.2.70 | 65.206.67.93 | SIP | Status: 100 Trying |
| 1882 | 68.908334 | 10.1.2.70 | 10.1.2.90 | SIP/SD | Request: INVITE sip:30002@avaya.com:5060, with session descri |
| 1884 | 68.909177 | 10.1.2.90 | 10.1.2.70 | SIP | Status: 100 Trying |
| 1887 | 68.911343 | 10.1.2.90 | 10.1.2.70 | SIP/SD | Status: 183 Session Progress, with session description |
| 1890 | 68.913311 | 10.1.2.70 | 65.206.67.93 | SIP/SD | Status: 183 Session Progress, with session description |

```
⊞ Transmission Control Protocol, Src Port: 22576 (22576), Dst Port: 5062 (5062), Seq: 1462, Ack: 1, Len: 234
⊞ [Reassembled TCP Segments (1694 bytes): #1881(1460), #1882(234)]
⊟ Session Initiation Protocol
  ⊞ Request-Line: INVITE sip:30002@avaya.com:5060 SIP/2.0
```

Scrolling down in the same trace in the screen below, in frame 2068, Communication Manager sends the 200 OK when the user answers the call. In frame 2079, Communication Manager sends the INVITE to begin the process of shuffling the media paths to "ip-direct", which concludes with the ACK in frame 2101.

| No. | Time | Source | Destination | Protocol | Info |
|---|---|---|---|---|---|
| 2068 | 75.416606 | 10.1.2.90 | 10.1.2.70 | SIP/SD | Status: 200 OK, with session description |
| 2071 | 75.419250 | 10.1.2.70 | 65.206.67.93 | SIP/SD | Status: 200 OK, with session description |
| 2076 | 75.659827 | 65.206.67.93 | 10.1.2.70 | SIP | Request: ACK sip:8668512649@10.1.2.90:5062;transport=tcp |
| 2077 | 75.662214 | 10.1.2.70 | 10.1.2.90 | SIP | Request: ACK sip:8668512649@10.1.2.90:5062;transport=tcp |
| 2079 | 75.663184 | 10.1.2.90 | 10.1.2.70 | SIP | Request: INVITE sip:+19088485704@199.173.95.16:5060;transpo |
| 2084 | 75.701827 | 10.1.2.70 | 10.1.2.90 | SIP | Status: 100 Trying |
| 2086 | 75.702853 | 10.1.2.70 | 65.206.67.93 | SIP | Request: INVITE sip:+19088485704@199.173.95.16:5060;transpo |
| 2088 | 75.704172 | 65.206.67.93 | 10.1.2.70 | SIP | Status: 100 Trying |
| 2097 | 75.977206 | 65.206.67.93 | 10.1.2.70 | SIP/SD | Status: 200 OK, with session description |
| 2098 | 75.978947 | 10.1.2.70 | 10.1.2.90 | SIP/SD | Status: 200 OK, with session description |
| 2100 | 75.980335 | 10.1.2.90 | 10.1.2.70 | SIP/SD | Request: ACK sip:+19088485704@199.173.95.16:5060;transport= |
| 2101 | 75.982562 | 10.1.2.70 | 65.206.67.93 | SIP/SD | Request: ACK sip:+19088485704@199.173.95.16:5060;transport= |

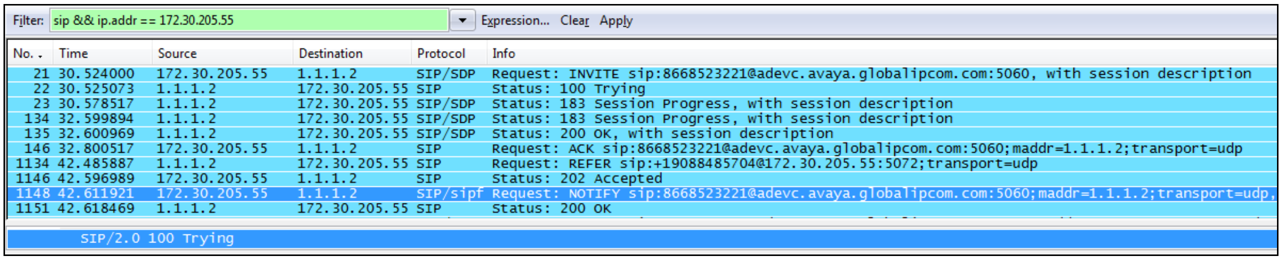
## 9.1.2 Example Inbound Call Referred via Call Vector to PSTN Destination

The following edited and annotated Communication Manager "list trace" trace output shows a call incoming on trunk group 67. The call was routed to a Communication Manager vector directory number (VDN 36998, **Section 4.10.1**) associated with a call vector (call vector 3, Section 4.10.1). The vector answers the call, plays an announcement to the caller, and then uses a "route-to" step to cause a REFER message to be sent with a Refer-To header containing the number configured in the vector "route-to" step (this number in the Refer-To can not be deduced via the trace command below). The originating PSTN telephone dialed 866-852-3221. Session Manager can map the number received from Verizon to the VDN extension (x36998), or the incoming call handling table for trunk group 67 can do the same. In the trace below, Session Manager had already mapped the

Verizon number to the Communication Manager VDN extension. The annotations in the edited trace highlight the behaviors. At the conclusion, the PSTN caller that dialed the Verizon toll-free number is talking to the Referred-to PSTN destination, and no trunks (i.e., from trunk 67 handling the call) are in use.

```
list trace tac 167                                                    Page   1
                                LIST TRACE
time            data
13:10:20 TRACE STARTED 09/08/2010 CM Release String cold-00.0.345.0-18444
13:11:49 SIP<INVITE sip:36998@avaya.com:5060 SIP/2.0
13:11:49     active trunk-group 67 member 1  cid 0x210
13:11:49 SIP>SIP/2.0 183 Session Progress
13:11:49     dial 36998
13:11:49     ring vector 3    cid 0x210
13:11:49     G729 ss:off ps:20
             rgn:4 [65.206.67.93]:21568
             rgn:1 [10.1.2.95]:2054
13:11:49     xoip options: fax:off modem:off tty:US  uid:0x500f1
             xoip ip: [10.1.2.95]:2054
13:11:51 SIP>SIP/2.0 183 Session Progress
/** Call is answered by Communication Manager to play announcement **/
13:11:51 SIP>SIP/2.0 200 OK
13:11:51     active announcement    36997 cid 0x210
13:11:51     hear annc board 001V9 ext 36997 cid 0x210
13:11:51 SIP<ACK sip:18668523221@10.1.2.90:5062;transport=tcp SI
13:11:51 SIP<P/2.0
13:12:01     idle announcement     cid 0x210
/** Announcement completes and route-to step in vector follows **/
13:12:01 SIP>REFER sip:+19088485704@199.173.95.16:5060;transport
13:12:01 SIP>=tcp;maddr=65.206.67.93 SIP/2.0
/** Verizon sends 202 Accepted and NOTIFY with 100-Trying **/
13:12:01 SIP<SIP/2.0 202 Accepted
13:12:01 SIP<NOTIFY sip:18668523221@10.1.2.90:5062;transport=tcp
13:12:01 SIP< SIP/2.0
13:12:01 SIP>SIP/2.0 200 OK
/** Referred-To PSTN Number answers, Verizon sends NOTIFY with 200 OK **/
13:12:13 SIP<NOTIFY sip:18668523221@10.1.2.90:5062;transport=tcp
13:12:13 SIP< SIP/2.0
13:12:13 SIP>SIP/2.0 200 OK
/** Call using enterprise trunks is cleared **/
13:12:13 SIP>BYE sip:+19088485704@199.173.95.16:5060;transport=t
13:12:13 SIP>cp;maddr=65.206.67.93 SIP/2.0
```

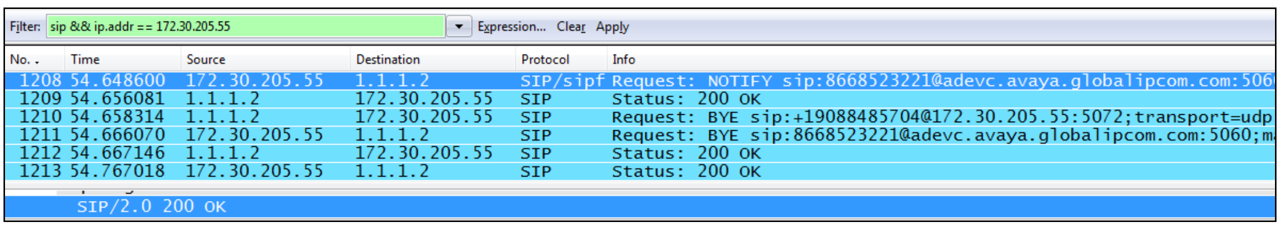
The following portion of a filtered Wireshark trace (tracing SIP messages on the public outside interface of the SBC only) shows the same incoming PSTN call. The call vector answers the call (frame 135), plays an announcement to the caller (note elapsed time between frames 135 and 1134 when RTP carrying the announcement is flowing), and then uses a “route-to” step to cause a REFER message to be sent (frame 1134) with a Refer-To header containing the number configured in the “route-to” step. In frame 1146, Verizon sends a 202 Accepted message for the REFER. In highlighted frame 1148, Verizon sends a NOTIFY message, where the abridged center area illustrates the NOTIFY is for a “100 Trying”.


Filter: sip && ip.addr == 172.30.205.55 Expression... Clear Apply

| No. | Time | Source | Destination | Protocol | Info |
|---|---|---|---|---|---|
| 21 | 30.524000 | 172.30.205.55 | 1.1.1.2 | SIP/SDP | Request: INVITE sip:8668523221@adevc.avaya.globalipcom.com:5060, with session description |
| 22 | 30.525073 | 1.1.1.2 | 172.30.205.55 | SIP | Status: 100 Trying |
| 23 | 30.578517 | 1.1.1.2 | 172.30.205.55 | SIP/SDP | Status: 183 Session Progress, with session description |
| 134 | 32.599894 | 1.1.1.2 | 172.30.205.55 | SIP/SDP | Status: 183 Session Progress, with session description |
| 135 | 32.600969 | 1.1.1.2 | 172.30.205.55 | SIP/SDP | Status: 200 OK, with session description |
| 146 | 32.800517 | 172.30.205.55 | 1.1.1.2 | SIP | Request: ACK sip:8668523221@adevc.avaya.globalipcom.com:5060;maddr=1.1.1.2;transport=udp |
| 1134 | 42.485887 | 1.1.1.2 | 172.30.205.55 | SIP | Request: REFER sip:+19088485704@172.30.205.55:5072;transport=udp |
| 1146 | 42.596989 | 172.30.205.55 | 1.1.1.2 | SIP | Status: 202 Accepted |
| 1148 | 42.611921 | 172.30.205.55 | 1.1.1.2 | SIP/sipf | Request: NOTIFY sip:8668523221@adevc.avaya.globalipcom.com:5060;maddr=1.1.1.2;transport=udp, |
| 1151 | 42.618469 | 1.1.1.2 | 172.30.205.55 | SIP | Status: 200 OK |

SIP/2.0 100 Trying

Verizon routes the call to the number specified in the Route-To header (i.e., the number in the route-to step in the vector). Scrolling down in this same trace, when the PSTN destination answers, Verizon sends the NOTIFY message in highlighted frame 1208, where the abridged center area illustrates the NOTIFY is for a “200 OK”. Observe the BYE messages clear the call to the enterprise site. Although the PSTN caller who dialed the toll-free number is talking to the Referred-to destination, no trunks are in use to the enterprise site that received the call.


Filter: sip && ip.addr == 172.30.205.55 Expression... Clear Apply

| No. | Time | Source | Destination | Protocol | Info |
|---|---|---|---|---|---|
| 1208 | 54.648600 | 172.30.205.55 | 1.1.1.2 | SIP/sipf | Request: NOTIFY sip:8668523221@adevc.avaya.globalipcom.com:506 |
| 1209 | 54.656081 | 1.1.1.2 | 172.30.205.55 | SIP | Status: 200 OK |
| 1210 | 54.658314 | 1.1.1.2 | 172.30.205.55 | SIP | Request: BYE sip:+19088485704@172.30.205.55:5072;transport=udp |
| 1211 | 54.666070 | 172.30.205.55 | 1.1.1.2 | SIP | Request: BYE sip:8668523221@adevc.avaya.globalipcom.com:5060;m |
| 1212 | 54.667146 | 1.1.1.2 | 172.30.205.55 | SIP | Status: 200 OK |
| 1213 | 54.767018 | 172.30.205.55 | 1.1.1.2 | SIP | Status: 200 OK |

SIP/2.0 200 OK

### 9.1.3 Example Inbound Call Referred with UUI to Alternate SIP Destination

The following Communication Manager “list trace vector” trace output shows a different example incoming Verizon toll-free call. The call was routed to a Communication Manager vector directory number (VDN 36990) associated with a call vector (call vector 5). As in previous illustrations, this vector will answer the call, play an announcement to the caller, and then use a “route-to” step to cause a REFER message to be sent to Verizon. In this case, the Refer-To number will cause Verizon to route the call to another SIP-connected destination. In the sample configuration, where only one site is available, this was tested by including a different toll-free number (866-851-2649) assigned to the same site in the Route-To header. The vector also sets 96 bytes of UUI data that will be included in the Refer-To header. When Verizon routes the call to the “alternate” destination, the INVITE message will contain a User-To-User header containing the UUI data sent in the Refer-To header. In practice, this would allow a Communication Manager at one site to pass call or customer-related data to another site via the Verizon network.

```
list trace vector 5                                                  Page   1
                                LIST TRACE
time      vec st data
13:26:06 TRACE STARTED 09/08/2010 CM Release String cold-00.0.345.0-18444
13:26:16    0  0 ENTERING TRACE cid 536
13:26:16    5  1 vdn e36990 bsr appl   0 strategy 1st-found override n
13:26:16    5  1 set A = none CATR 1234567890123456
13:26:16    5  1      operand    = []
13:26:16    5  1      operand    = [1234567890123456]
13:26:16    5  1      ========= CATR =========
13:26:16    5  1      variable A = [1234567890123456] asaiuui local
13:26:16    5  1      asaiuui chg from [] to [1234567890123456]
13:26:16    5  2 set B = none CATR 7890123456789012
13:26:16    5  2      operand    = []
13:26:16    5  2      operand    = [7890123456789012]
13:26:16    5  2      ========= CATR =========
13:26:16    5  2      variable B = [7890123456789012] asaiuui local
13:26:16    5  2      asaiuui chg from [] to [7890123456789012]
13:26:16    5  3 set C = none CATR 3456789012345678
13:26:16    5  3      operand    = []
13:26:16    5  3      operand    = [3456789012345678]
13:26:16    5  3      ========= CATR =========
13:26:16    5  3      variable C = [3456789012345678] asaiuui local
13:26:16    5  3      asaiuui chg from [] to [3456789012345678]
13:26:16    5  4 set D = none CATR 9012345678901234
13:26:16    5  4      operand    = []
13:26:16    5  4      operand    = [9012345678901234]
13:26:16    5  4      ========= CATR =========
13:26:16    5  4      variable D = [9012345678901234] asaiuui local
13:26:16    5  4      asaiuui chg from [] to [9012345678901234]
13:26:16    5  5 set E = none CATR 5678901234567890
13:26:16    5  5      operand    = []
13:26:16    5  5      operand    = [5678901234567890]
13:26:16    5  5      ========= CATR =========
13:26:16    5  5      variable E = [5678901234567890] asaiuui local
13:26:16    5  5      asaiuui chg from [] to [5678901234567890]
13:26:16    5  6 set F = none CATR 1234567890123456
13:26:16    5  6      operand    = []
13:26:16    5  6      operand    = [1234567890123456]
13:26:16    5  6      ========= CATR =========
13:26:16    5  6      variable F = [1234567890123456] asaiuui local
13:26:16    5  6      asaiuui chg from [] to [1234567890123456]
13:26:16    5  9 <blank>
13:26:16    5 10 wait 2 secs hearing silent
13:26:18    5 11 announcement 36997
13:26:18    5 11      announcement: board 001V9 ann ext: 36997
13:26:28    5 12 route-to number ~r+18668512649 cov n if unconditionally
13:26:35    5 12 LEAVING VECTOR PROCESSING cid 536
13:26:35    5 12 TRACE COMPLETE cid 536
```

The following screen, which is the beginning of a filtered Wireshark trace, (tracing SIP messages on the public outside interface of the SBC only) shows another call to this same Verizon toll-free number. At the start, the trace looks very similar to the one shown in the previous section. The user dials the same number (866-852-3221), but in this case, Session Manager has adapted the number to Communication Manager vector directory number 36990 associated with call vector 5. The call vector answers the call (frame 133), plays an announcement to the caller (note elapsed time between frames 134 and 1128), and then uses a “route-to” step to cause a REFER message to be sent (frame 1128). The REFER includes a Refer-To header containing the number configured in the “route-to” step, which in this case contains another toll-free number (866-851-2649). The

REFER also contains the 96 bytes of UUI data set in vector 5. In frame 1147, Verizon sends a 202 Accepted message for the REFER, and in frame 1149, Verizon sends a NOTIFY with “100 Trying” as illustrated previously.

Filter: sip && ip.addr == 172.30.205.55 Expression... Clear Apply

| No. | Time | Source | Destination | Protocol | Info |
|---|---|---|---|---|---|
| 22 | 27.344000 | 172.30.205.55 | 1.1.1.2 | SIP/SDP | Request: INVITE sip:8668523221@adevc.avaya.globalipcom.com:5060, with session |
| 23 | 27.344546 | 1.1.1.2 | 172.30.205.55 | SIP | Status: 100 Trying |
| 24 | 27.402025 | 1.1.1.2 | 172.30.205.55 | SIP/SDP | Status: 183 Session Progress, with session description |
| 132 | 29.406676 | 1.1.1.2 | 172.30.205.55 | SIP/SDP | Status: 183 Session Progress, with session description |
| 133 | 29.407612 | 1.1.1.2 | 172.30.205.55 | SIP/SDP | Status: 200 OK, with session description |
| 144 | 29.606868 | 172.30.205.55 | 1.1.1.2 | SIP | Request: ACK sip:8668523221@adevc.avaya.globalipcom.com:5060;maddr=1.1.1.2;tra |
| 1128 | 39.292858 | 1.1.1.2 | 172.30.205.55 | SIP | Request: REFER sip:+19088485704@172.30.205.55:5072;transport=udp |
| 1147 | 39.461923 | 172.30.205.55 | 1.1.1.2 | SIP | Status: 202 Accepted |
| 1149 | 39.475683 | 172.30.205.55 | 1.1.1.2 | SIP/sip | Request: NOTIFY sip:8668523221@adevc.avaya.globalipcom.com:5060;maddr=1.1.1.2; |
| 1151 | 39.482494 | 1.1.1.2 | 172.30.205.55 | SIP | Status: 200 OK |

Although not expanded in the Wireshark trace above, the format of the Refer-To header in the REFER message will be like the following, where the host portion labeled “Verizon-IPCC” can be manipulated by the SBC as needed:
Refer-To: <sip:+18668512649@Verizon-IPCC?User-to-User=0431323334353637 38<UUI abridged for brevity> %3Bencoding%3Dhex>.

If the SBC is used to manipulate the Refer-To header as shown in **Section 6.3.9**, an example Refer-To header copied from a Wireshark trace of a successful call will look like the following (UUI abridged for brevity):

```
Refer-To: <sip:+18668512649@172.30.205.55:5072?User-to-User=04313233343536373839303132333435363738393031323
```

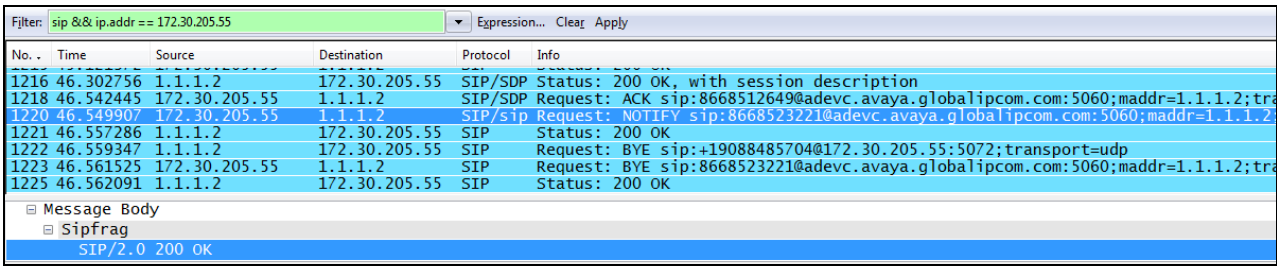
Verizon routes the call to the number specified in the Route-To header which in this case is another Verizon toll-free number assigned to this same site (i.e., in production, this would typically be used to route to an alternate site). Scrolling down in this same trace, frame 1187 is selected below to show the INVITE from Verizon that was stimulated by the REFER/Refer-To. From the highlighted message summary, it can be observed that the R-URI contains 866-851-2649, the toll-free number used in the Refer-To step in the vector. In the center, where details of the contents of the INVITE are shown, note that the PAI contains the original caller ID of the true PSTN caller (908-848-5704), and the User-to-User header contains the contents of the UUI previously sent by the Avaya CPE to Verizon in the Refer-To header in the REFER message. The reader may also observe that this INVITE from Verizon does not contain SDP.

Filter: sip && ip.addr == 172.30.205.55 Expression... Clear Apply

| No. | Time | Source | Destination | Protocol | Info |
|---|---|---|---|---|---|
| 1187 | 40.214234 | 172.30.205.55 | 1.1.1.2 | SIP | Request: INVITE sip:8668512649@adevc.avaya.globalipcom.com:5060 |
| 1188 | 40.215309 | 1.1.1.2 | 172.30.205.55 | SIP | Status: 100 Trying |
| 1189 | 40.228758 | 1.1.1.2 | 172.30.205.55 | SIP/SDP | Status: 183 Session Progress, with session description |

```
⊞ Request-Line: INVITE sip:8668512649@adevc.avaya.globalipcom.com:5060 SIP/2.0
⊟ Message Header
  ⊞ Via: SIP/2.0/UDP 172.30.205.55:5072;branch=z9hG4bKfhei9m102o5giokap2r0.1
    Call-ID: -20161471376346634940@65.210.180.195
  ⊞ From: <sip:+19088485704@199.173.94.208:5060;user=phone>;tag=942921061.2.becndlcmnefaejmgphgmkcao
  ⊞ To: sip:18668523221@1.1.1.2
  ⊞ CSeq: 1 INVITE
  ⊞ Contact: <sip:+19088485704@172.30.205.55:5072;transport=udp>
    Content-Length: 0
    Allow: INVITE, ACK, BYE, OPTIONS, CANCEL, SUBSCRIBE, REFER
  ⊞ P-Asserted-Identity: <sip:+19088485704@199.173.94.208;user=phone>
  ⊞ User-to-User: 04313233343536373839303132333435363738393031323334353637383930313233343536373839303132333435363738393031323334353
```

Scrolling down further in this same trace, in frame 1216, the enterprise site sends the 200 OK with SDP when the new inbound call to 866-851-2649 is answered. Verizon responds with an ACK with SDP in frame 1218. Once the referred-to destination has answered, Verizon sends the NOTIFY containing the “200 OK” result in frame 1220, which is highlighted to show the 200 OK in the center. The original call (i.e., the original call to 866-852-3221 that stimulated the REFER) is then cleared. The PSTN caller and the answering party of the referred-to call are now talking. If the answering party of a referred-to call is a Communication Manager user who has a “uui-info” button, and the answering user’s Class of Restriction (COR) allows “Station Button Display of UUI IE data”, the answering user can see UUI data on the display phone by pressing the “uui-info” button. In a multi-site contact center setting, a contact center agent answering a call at site B can see UUI sent in the REFER from site A.

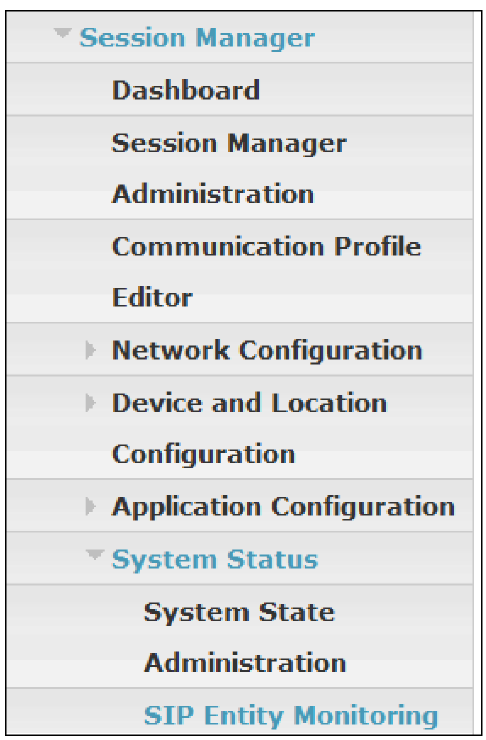



## 9.2. Avaya Aura™ System Manager and Session Manager Verifications

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

### 9.2.1 Verify SIP Entity Link Status

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



From the list of monitored entities, select an entity of interest, such as “AuraSBC”. Under normal operating conditions, the **Link Status** should be “Up” as shown in the example screen below. As can be observed below, the SBC has responded with a “404 Not Found” to the SIP OPTIONS from Session Manager. This response is sufficient for Session Manager to consider the SIP entity up.

**All Entity Links to SIP Entity: AuraSBC**

Refresh | Summary View

1 Item — Filter: Enable

| Details | Session Manager Name | SIP Entity Resolved IP | Port | Proto. | Conn. Status | Reason Code | Link Status |
|---|---|---|---|---|---|---|---|
| Hide | SM1 | 65.206.67.93 | 5060 | TCP | Up | 404 Not found | Up |

| Time Last Down | Time Last Up | Last Message Sent | Last Response Latency (ms) |
|---|---|---|---|
| Aug 11, 2010 1:06:38 PM EDT | Aug 11, 2010 1:38:49 PM EDT | Sep 8, 2010 2:13:20 PM EDT | 8 |

Return to the list of monitored entities, and select another entity of interest, such as “CM-Evolution-procr-5062”. Under normal operating conditions, the **Link Status** should be “Up” as shown in the example screen below. In this case, “Show” under Details was selected to view additional information. Note the use of port 5062.

**All Entity Links to SIP Entity: CM-Evolution-procr-5062**

Refresh | Summary View

1 Item — Filter: Enable

| Details | Session Manager Name | SIP Entity Resolved IP | Port | Proto. | Conn. Status | Reason Code | Link Status |
|---|---|---|---|---|---|---|---|
| Hide | SM1 | 10.1.2.90 | 5062 | TCP | Up | 200 OK | Up |

| Time Last Down | Time Last Up | Last Message Sent | Last Response Latency (ms) |
|---|---|---|---|
| Jul 20, 2010 9:03:20 AM EDT | Jul 20, 2010 9:05:18 AM EDT | Sep 8, 2010 2:09:33 PM EDT | 8 |

Return to the list of monitored entities, and select another entity of interest, such as “CM Evolution Server”. Under normal operating conditions, the **Link Status** should be “Up” as shown in the example screen below. In this case, “Show” under Details was selected to view additional information. Note the use of port 5060 using the same IP Address as “CM-Evolution-procr-5062” shown in the prior screen.

**All Entity Links to SIP Entity: CM Evolution Server**

Refresh | Summary View

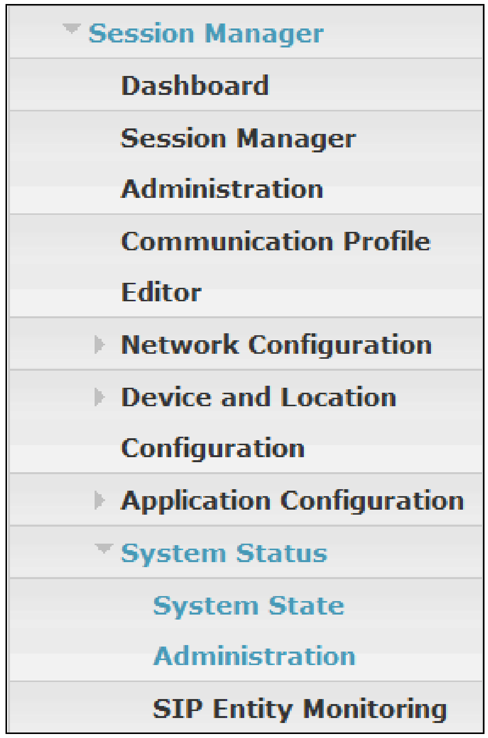
1 Item — Filter: Enable

| Details | Session Manager Name | SIP Entity Resolved IP | Port | Proto. | Conn. Status | Reason Code | Link Status |
|---|---|---|---|---|---|---|---|
| Hide | SM1 | 10.1.2.90 | 5060 | TCP | Up | 200 OK | Up |

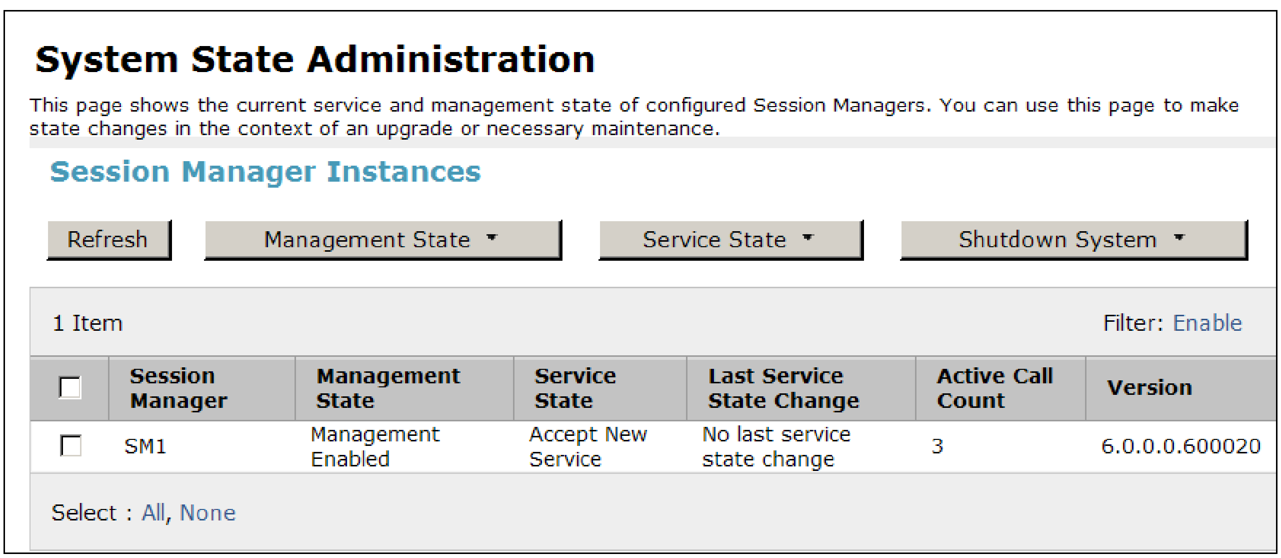
| Time Last Down | Time Last Up | Last Message Sent | Last Response Latency (ms) |
|---|---|---|---|
| Sep 7, 2010 2:19:13 PM EDT | Sep 7, 2010 2:21:10 PM EDT | Sep 8, 2010 2:16:23 PM EDT | 7 |

### 9.2.2 Verify System State

Expand **Elements → Session Manager → System Status → System State Administration**, as shown below.

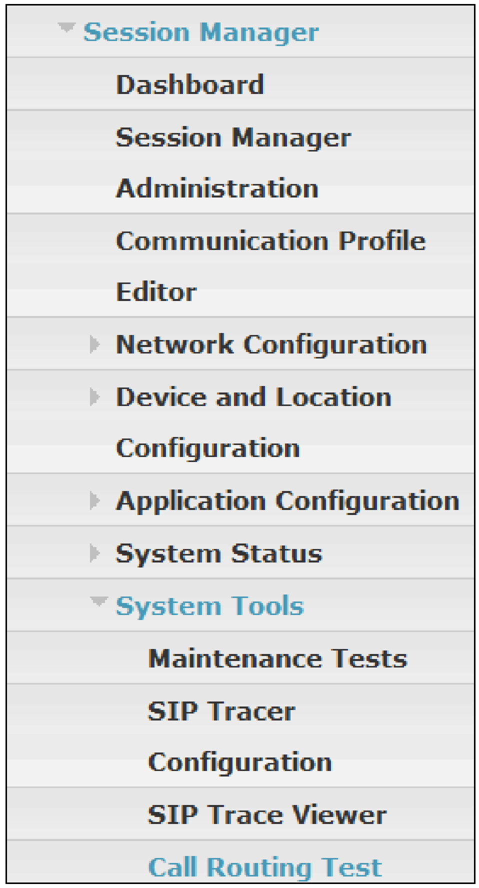



Verify that the **Management State** is "Management Enabled" and the **Service State** is "Accept New Service." The **Version** can also be observed.

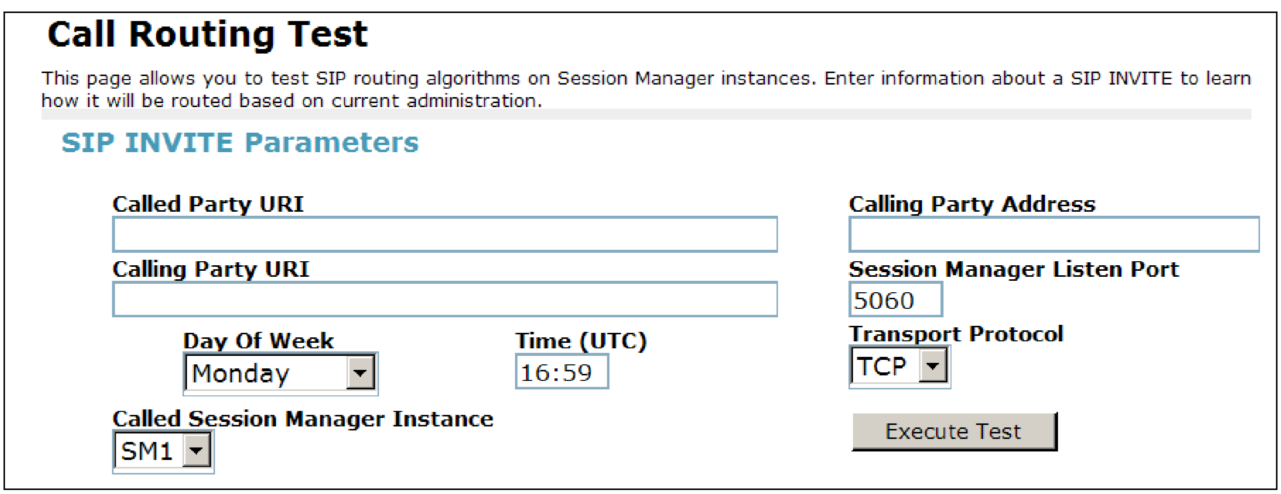


### 9.2.3 Call Routing Test

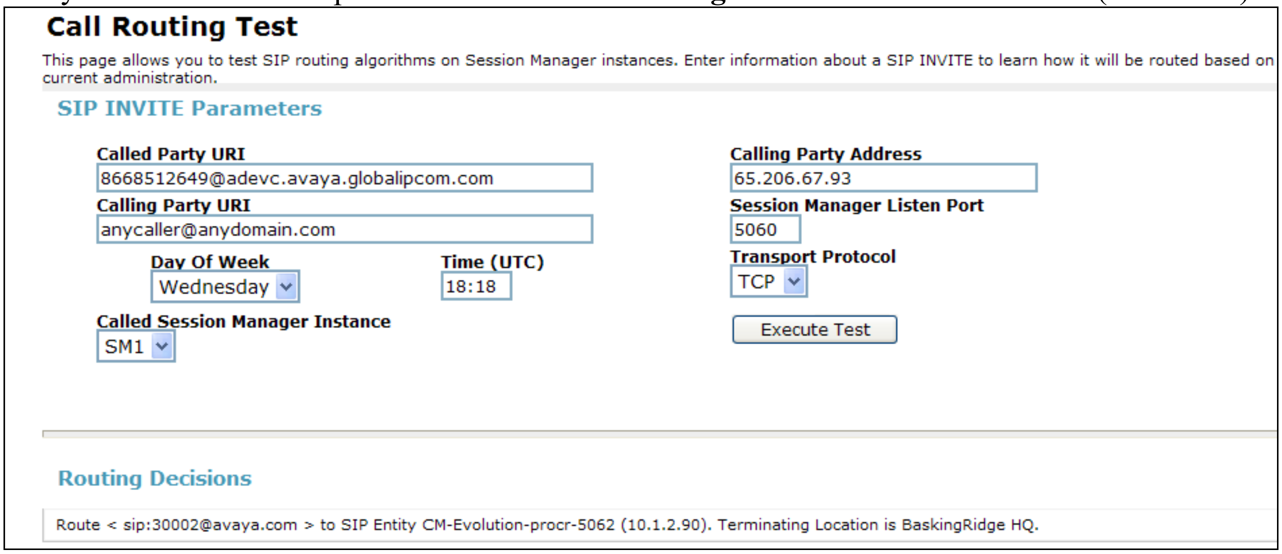
The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, expand **Elements → Session Manager → System Tools → Call Routing Test**, as shown below.



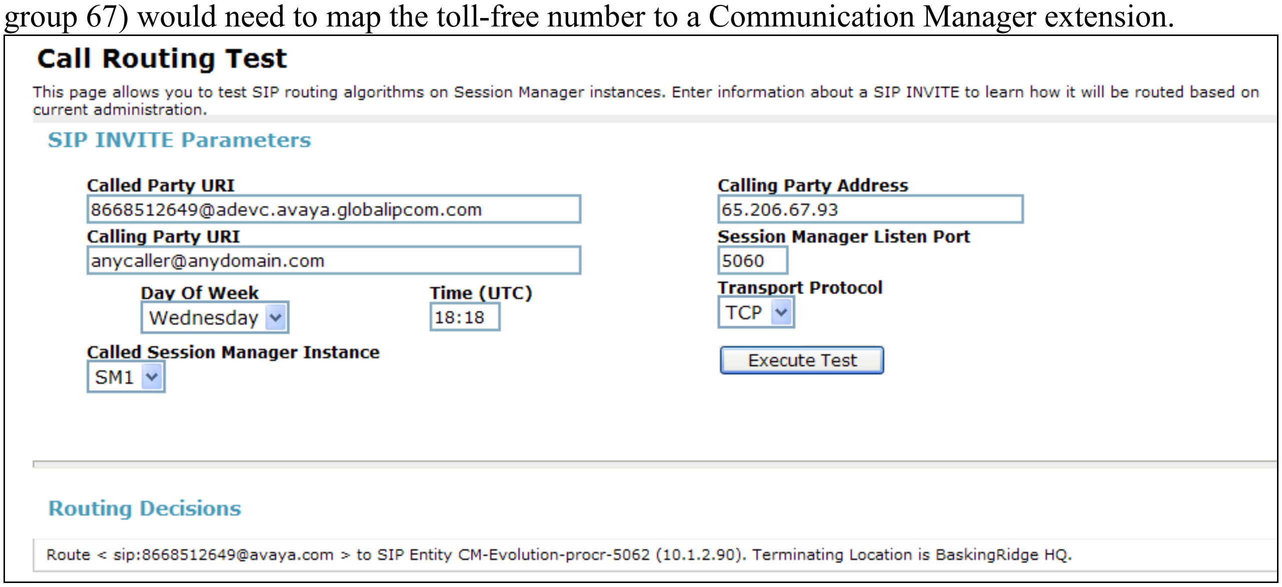

A screen such as the following is displayed.



Populate the fields for the call parameters of interest and click **Execute Test**. For example, the following shows a call routing test for an inbound toll-free call from the PSTN to the enterprise via the SBC (65.206.67.93). Under **Routing Decisions**, observe that the call will route to the S8800 processor ethernet (10.1.2.90) using the SIP entity named "CM-Evolution-procr-5062". The domain in the Request-URI is converted to "avaya.com", and the digits are manipulated such that the Verizon toll-free number (i.e., 866-851-2649) is converted to a Communication Manager extension (i.e., 30002) by the Session Manager adapter assigned to the Communication Manager entity. Scroll down to inspect the details of the **Routing Decision Process** if desired (not shown).



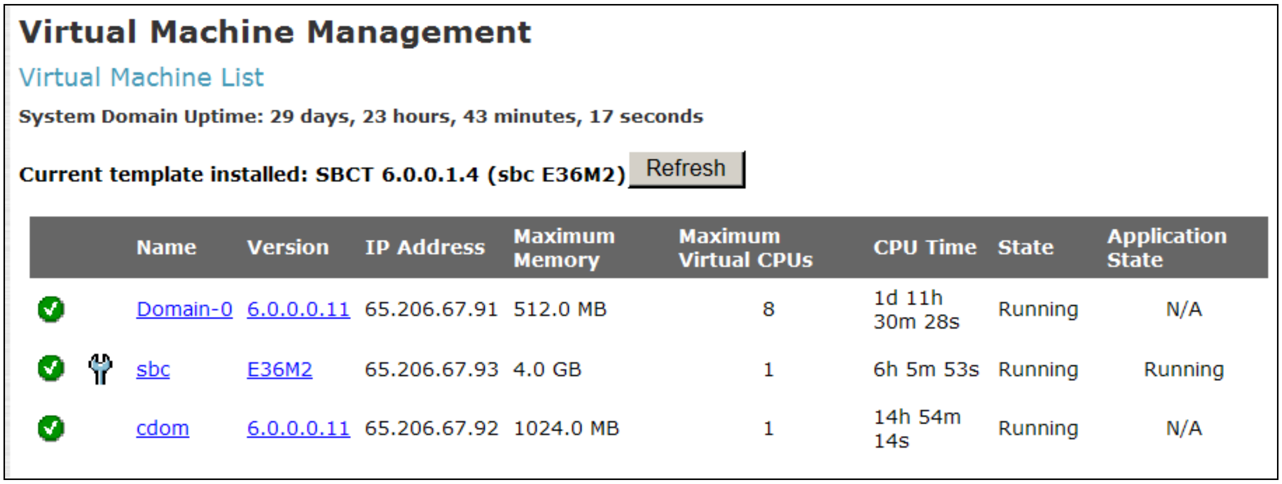

After a configuration change that removed the Verizon toll-free number to Communication Manager extension digit manipulation from the Session Manager adapter, the following example shows the same call routing test. Under **Routing Decisions**, observe the call will still route to the S8800 processor ethernet using the SIP entity named "CM-Evolution-procr-5062", but the Request-URI now contains the full 10 digit number. With this configuration, the incoming call handling table of the Communication Manager trunk group receiving the incoming call (i.e., trunk group 67) would need to map the toll-free number to a Communication Manager extension.



## 9.3. Avaya Aura™ Session Border Controller Verification

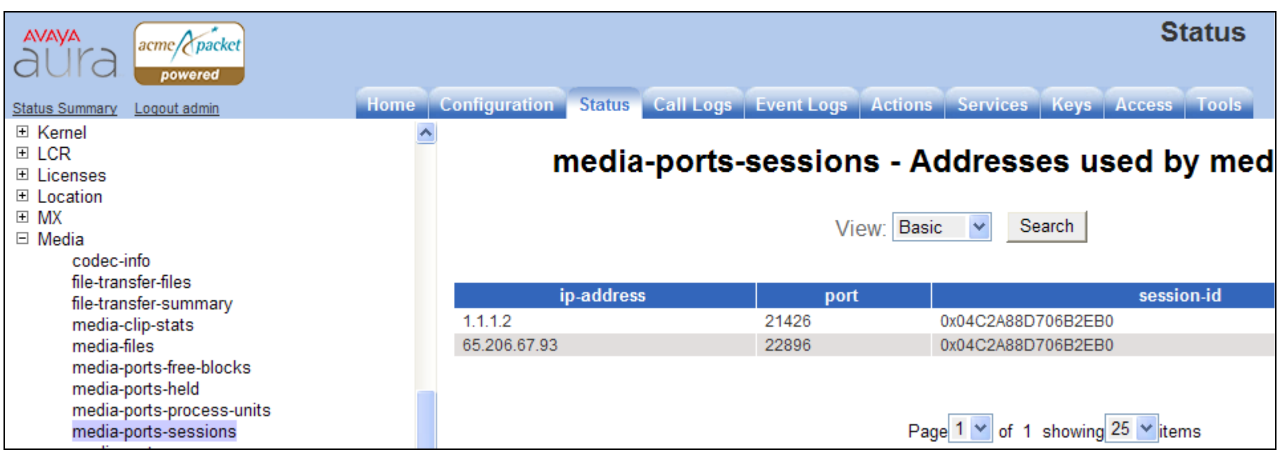
This section contains verification steps that may be performed using the Avaya Aura™ Session Border Controller.

The status of the virtual machines can be checked via the System Platform Console Domain of the S8800 Server running the SBC. The following screen, available via the **Virtual Machine Management** link in the console domain, shows the "Running" State of the SBC.

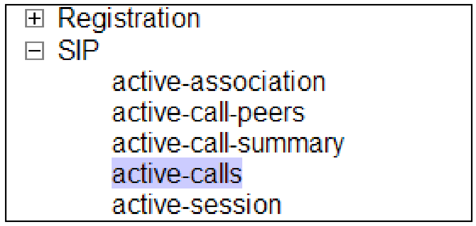



Click on the wrench icon to the left of the name "sbc" to access the element manager user interface of the SBC.

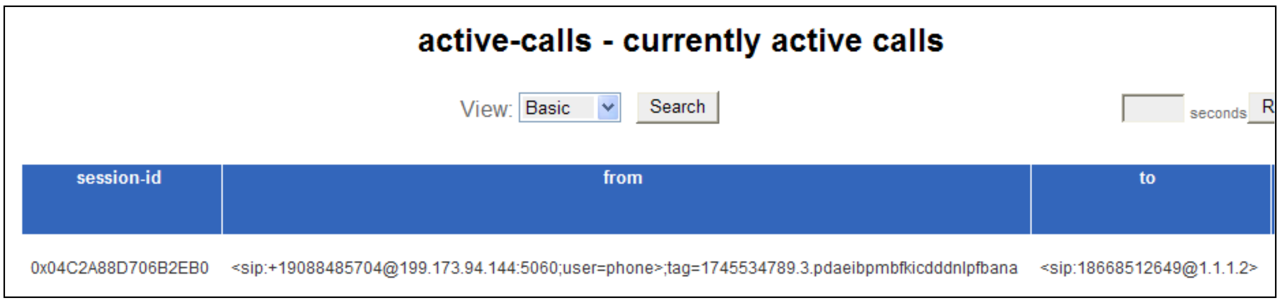
A wealth of status information is available via the **Status** tab. For example, in the following screen, the left side menu expands **Media** and **media-port-sessions** is selected, revealing the information on the right about an active call.



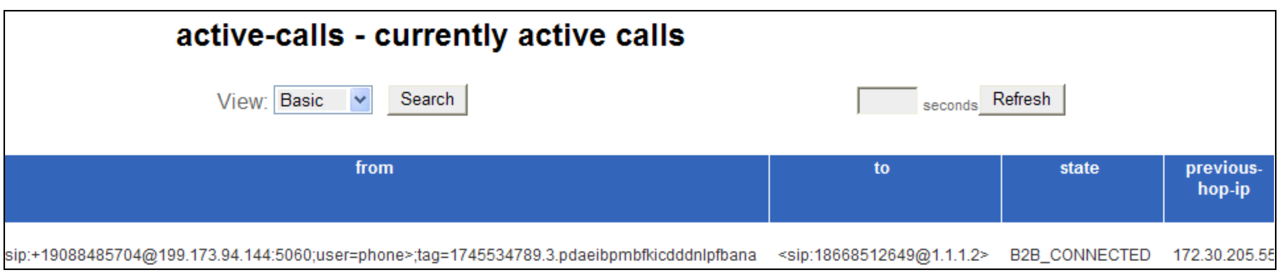
In the same **Status** tab, there is a SIP heading on the left that can be expanded as shown below.




In the example screen below, **active-calls** was selected, revealing details about an active incoming call on the right.



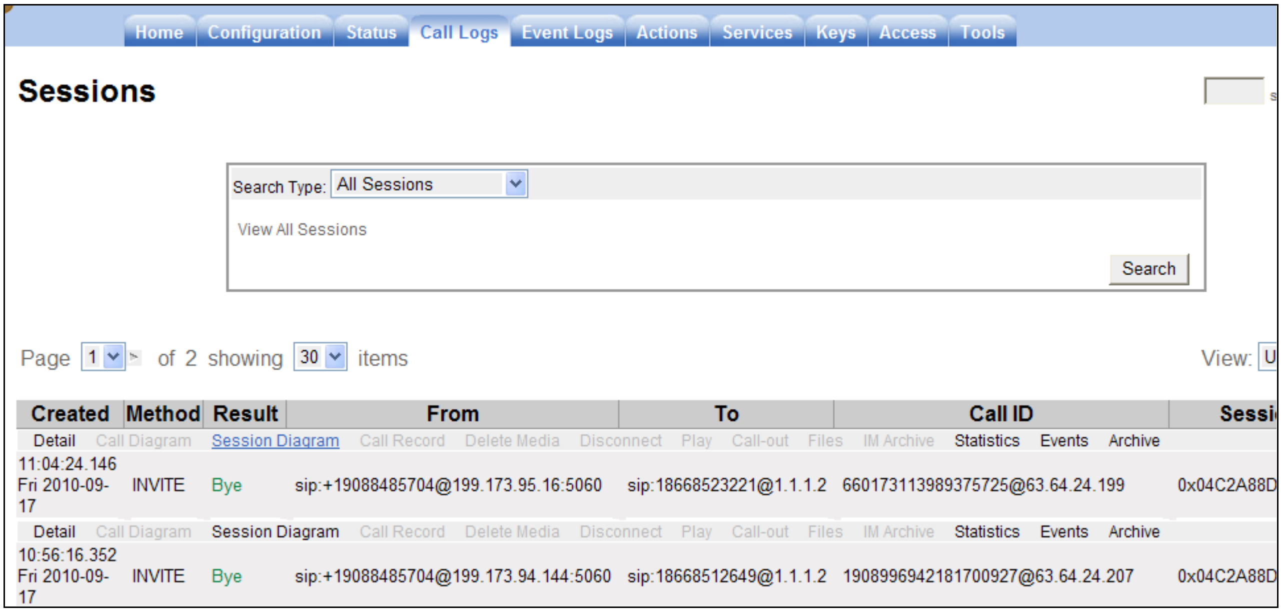

A scroll bar allows viewing of additional information about the active call. An example screen after scrolling to the right is shown below. Additional information is available by continuing to scroll to the right (not shown).



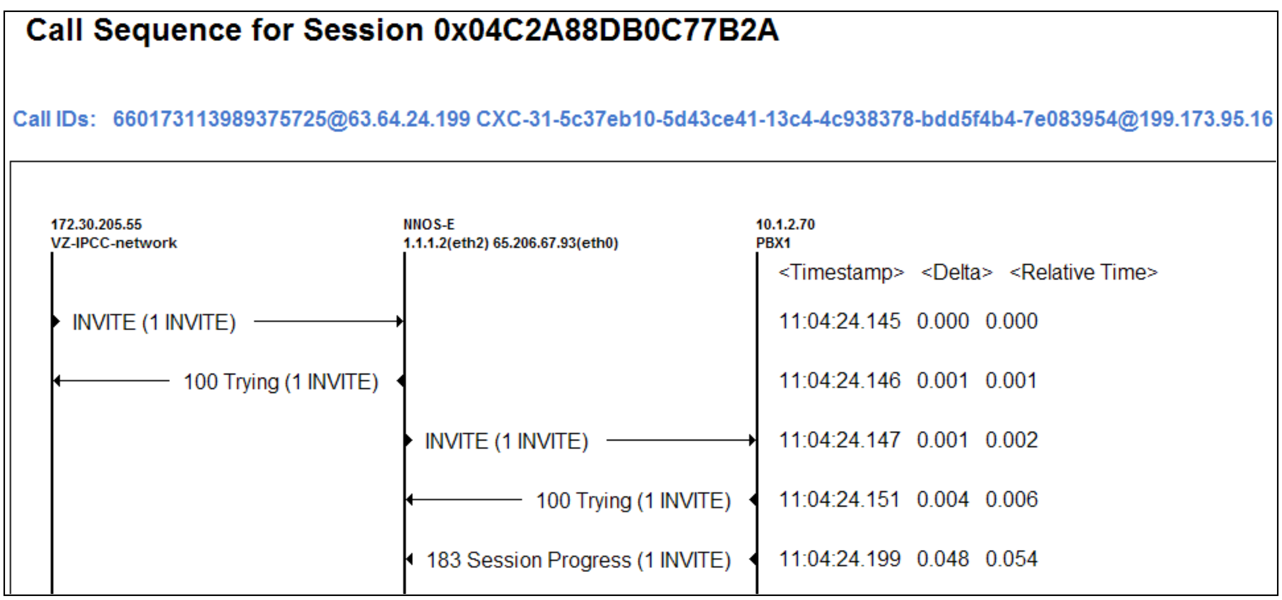
### 9.3.1 Avaya Aura™ Session Border Controller Call Logs

The **Call Logs** tab can provide useful diagnostic or troubleshooting information. In the following screen, the **SIP Messages** search capability can be observed.

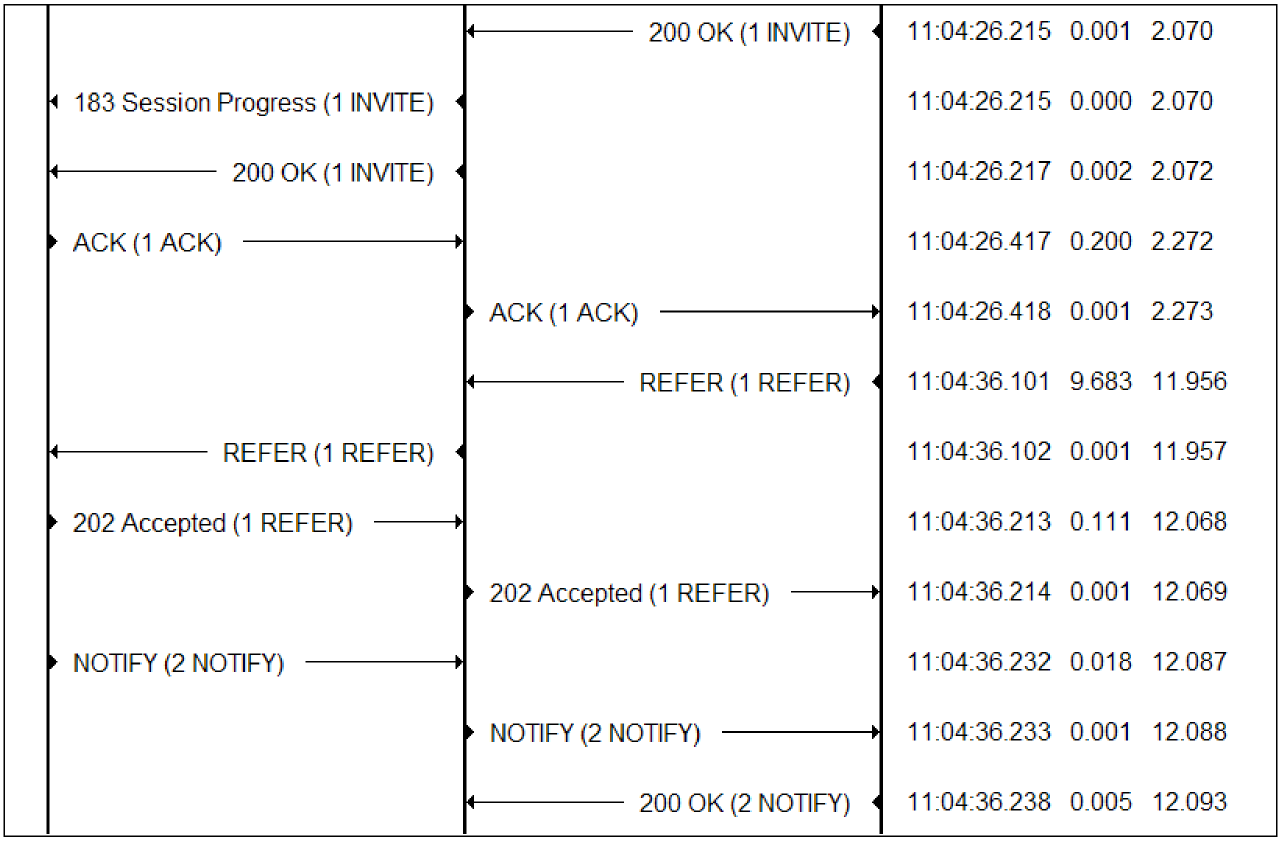
The following screen shows the **Call Logs** tab selected after making an inbound toll-free call. As shown below, select the **Session Diagram** link to view a ladder diagram for the session.



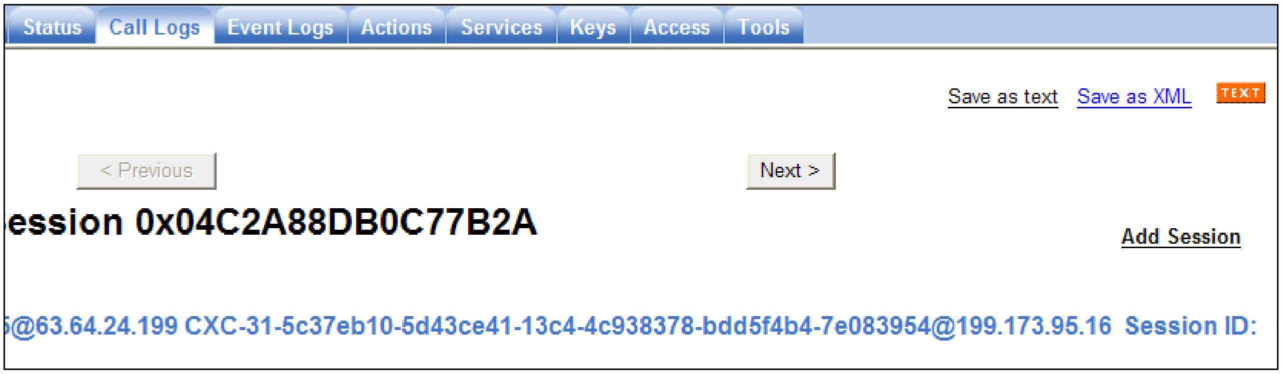

The following screen shows a portion of the ladder diagram for an inbound toll-free call. Note that the activity for both the inside private and outside public side of the SBC can be seen.



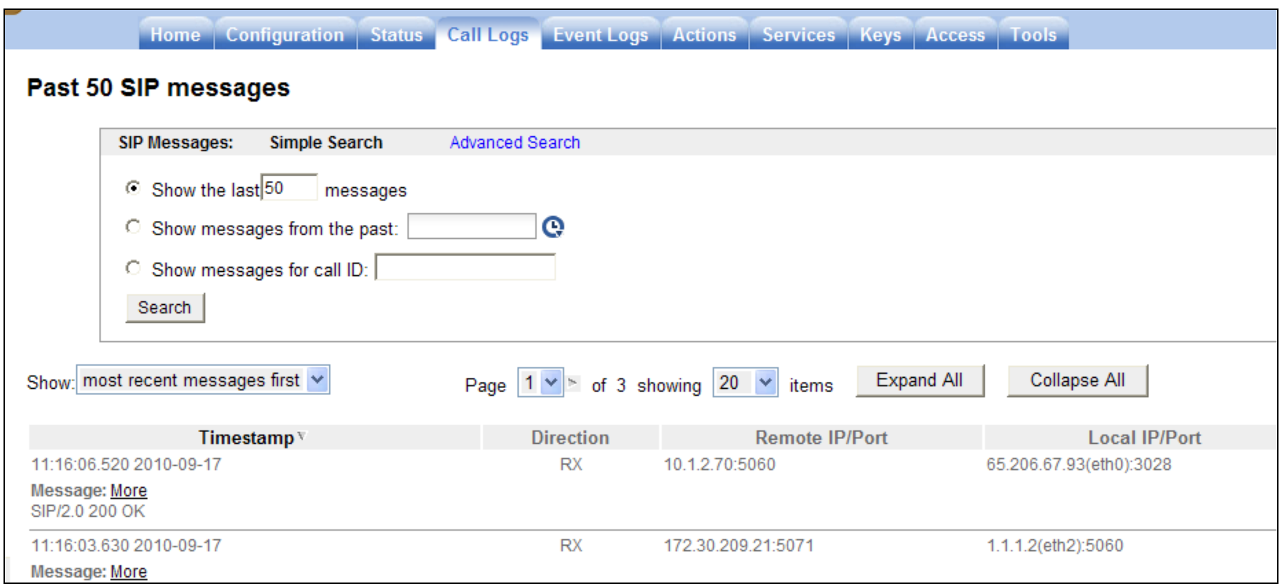
Scroll down to continue the ladder diagram. The following screen shows the portion of the ladder diagram for a call that is answered by a Communication Manager vector and subsequently referred back to Verizon.



At the top right of the screen, the session may be saved as a text or XML file. If the session is saved as an XML file, using the **Save as XML** link, the xml file can be provided to support personnel that can open the session on another Avaya Aura™ Session Border Controller for analysis.




The **Call Logs** tab also provides search capabilities. The following screen shows the result of selecting the **SIP Messages** link (not shown) within the left-side menu of the **Call Logs** tab. The number "50" was entered to view the last 50 SIP messages.



Scrolling down, the following screen shows a sampling of SIP messages for an inbound toll-free call to a call vector that triggers a REFER. **More** may be clicked to reveal more information about a particular message, as is the case below for the REFER message that was transmitted (Direction TX) to Verizon. In this case, the Refer-To was to a PSTN destination (as described in **Section 9.1.2**).

| Timestamp | Direction | Remote IP/Port | Local IP/Port | Transp |
|---|---|---|---|---|
| 11:18:28.319 2010-09-17<br>**Message:** More<br>SIP/2.0 200 OK | RX | 10.1.2.70:5060 | 65.206.67.93(eth0):3028 | TCP |
| 11:18:28.313 2010-09-17<br>**Message:** More<br>NOTIFY sip:18668523221@10.1.2.90:5062;transport=tcp SIP/2.0 | TX | 10.1.2.70:5060 | 65.206.67.93(eth0):3028 | TCP |
| 11:18:28.312 2010-09-17<br>**Message:** More<br>NOTIFY sip:8668523221@adevc.avaya.globalipcom.com:5060;maddr=1.1.1.2;transport=udp SIP/2.0 | RX | 172.30.205.55:5072 | 1.1.1.2(eth2):5060 | UDP |
| 11:18:28.295 2010-09-17<br>**Message:** More<br>SIP/2.0 202 Accepted | TX | 10.1.2.70:54190 | 65.206.67.93(eth0):5060 | TCP |
| 11:18:28.295 2010-09-17<br>**Message:** More<br>SIP/2.0 202 Accepted | RX | 172.30.205.55:5072 | 1.1.1.2(eth2):5060 | UDP |
| 11:18:28.121 2010-09-17<br>**Message:** Less<br>Call-ID: -954515655177363214l@63.78.210.198<br>CSeq: 1 REFER<br>Via: SIP/2.0/UDP 1.1.1.2:5060;branch=z9hG4bK-b45ea-4c9386c4-bde2d57a-5c8abf5f<br>User-Agent: Avaya CM/R016x.00.0.345.0 AVAYA-SM-6.0.0.0.600020<br>Max-Forwards: 66<br>Refer-To: <sip:+17326870755@172.30.205.55:5072> | TX | 172.30.205.55:5072 | 1.1.1.2(eth2):5060 | UDP |
| 11:18:28.120 2010-09-17<br>**Message:** More<br>REFER sip:+19088485704@199.173.94.208:5060;transport=tcp;maddr=65.206.67.93 SIP/2.0 | RX | 10.1.2.70:54190 | 65.206.67.93(eth0):5060 | TCP |
| 11:18:26.522 2010-09-17<br>**Message:** More<br>SIP/2.0 200 OK | RX | 10.1.2.70:5060 | 65.206.67.93(eth0):3028 | TCP |

# 10. Conclusion

As illustrated in these Application Notes, Avaya Aura™ Communication Manager 6.0, Avaya Aura™ Session Manager 6.0, and the Avaya Aura™ Session Border Controller can be configured to interoperate successfully with Verizon Business IP Contact Center Services suite. This solution provides users of Avaya Aura™ Communication Manager the ability to support inbound toll free calls over a Verizon Business VoIP Inbound SIP trunk service connection. In addition, these Application Notes further demonstrate that the Avaya Aura™ Communication Manager implementation of SIP Network Call Redirection (SIP-NCR) can work in conjunction with Verizon's Business IP Contact Center services implementation of SIP-NCR to support call redirection over SIP trunks inclusive of passing User-User Information (UUI).

Please note that the sample configuration shown in these Application Notes is intended to provide configuration guidance as a supplement to other Avaya product documentation.

# 11. Additional References

## 11.1. Avaya

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


[1] *Installing and Configuring Avaya Aura™ Communication Manager*, Doc ID 03-603558, Release 6.0 June, 2010 available at http://support.avaya.com/css/P8/documents/100089133
[2] *Administering Avaya Aura™ Communication Manager*, Doc ID 03-300509, Issue 6.0 June 2010 available at http://support.avaya.com/css/P8/documents/100089333
[3] *Administering Avaya Aura™ Session Manager*, Doc ID 03-603324, Release 6.0, June 2010 available at http://support.avaya.com/css/P8/documents/100082630
[4] *Installing and Configuring Avaya Aura™ Session Manager,* Doc ID 03-603473 Release 6.0, June 2010 available at http://support.avaya.com/css/P8/documents/100089152
[5] *Maintaining and Troubleshooting Avaya Aura™ Session Manager,* Doc ID 03-603325, Release 6.0, June 2010 available at http://support.avaya.com/css/P8/documents/100089154
[6] *Administering Avaya Aura™ System Manager*, Document Number 03-603324, Release 5.2, November 2009 available at http://support.avaya.com/css/P8/documents/100089681
[7] *Avaya Aura™ SBC System Administration Guide* September 2010 available at http://support.avaya.com/css/P8/documents/100111137


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

Application Notes reference [AuraSBC-IP-Trunk] is a companion document that illustrates the initial installation, licensing, and wizard configuration of the SBC that formed the starting point for the SBC configuration shown in these Application Notes. If Verizon IPCC Services will be the first service configured on the SBC, the wizard configuration approach shown in reference [AuraSBC-IP-Trunk] may be run for the Verizon IPCC Services as an alternative to the procedures shown in this document that adds the Verizon IPCC Services via the OS-E GUI.
[AuraSBC-IP-Trunk] Application Notes for Avaya Aura™ Communication Manager 6.0, Avaya Aura™ Session Manager 6.0, and Avaya Aura™ Session Border Controller with Verizon Business IP Trunk Service – Issue 1.0

Application Notes Reference [JF-VZIPCC] documents Verizon IPCC Services with previous versions of Avaya Aura™ Communication Manager and Avaya Aura™ Session Manager.
[JF-VZIPCC] Application Notes for Avaya Aura™ Communication Manager 5.2, Avaya Aura™ Session Manager 1.1, and Acme Packet 3800 Net-Net Session Director with Verizon Business IP Contact Centers Services Suite – Issue 1.2
https://devconnect.avaya.com/public/download/dyn/AvayaSM_VzBIPCC.pdf

## 11.2. Verizon Business

Information in the following documents was also used for these Application Notes:


[8] *Verizon Business IPCC Interoperability Test Plan, Revision 1.7, Aug 27, 2009*
[9] *Verizon Business IP Contact Center Trunk Interface Network Interface Specification, Document Version 2.2.1.9, Aug 25, 2009*
[10] *Additional information regarding Verizon Business IPCC Services suite offer can be found at http://www.verizonbusiness.com/products/contactcenter/ip/*