Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Communication Server 1000E Release 7.5, Avaya Aura® Session Manager 6.1, and Avaya Aura® Session Border Controller 6.0 with Verizon Business IP Contact Center (IPCC) Services Suite – Issue 1.1

Abstract

These Application Notes illustrate a sample configuration using Avaya Communication Server 1000E Release 7.5, Avaya Aura® Session Manager Release 6.1, and the Avaya Aura® Session Border Controller Release 6.0. The enterprise equipment is integrated with the Verizon Business IP Contact Center (IPCC) Services suite. The Verizon Business IPCC Services suite is comprised of the IP Toll Free VoIP Inbound and IP-IVR SIP trunk service offers. This service suite provides toll free inbound calling via standards-based SIP trunks. Using the sample configuration, PSTN callers may dial toll-free numbers associated with the IP Toll Free and IP-IVR services to reach Avaya Communication Server 1000E telephone users.

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

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1. Introduction

These Application Notes illustrate a sample configuration using Avaya Communication Server 1000E Release 7.5, Avaya Aura® Session Manager Release 6.1, and the Avaya Aura® Session Border Controller Release 6.0. The enterprise equipment is integrated with the Verizon Business IP Contact Center (IPCC) Services suite. The Verizon Business IPCC Services suite is comprised of the IP Toll Free VoIP Inbound and IP-IVR SIP trunk service offers. This service suite provides toll free inbound calling via standards-based SIP trunks. Using the sample configuration, PSTN callers may dial toll-free numbers associated with the IP Toll Free and IP-IVR services to reach Avaya Communication Server 1000E telephone users.

Access to the IPCC Services suite may use Internet Dedicated Access (IDA) or Private IP (PIP). The configuration documented in these Application Notes used the Verizon IPCC service terminated via a PIP network connection, but the solution validated in this document can also be applied to IPCC services delivered via IDA service terminations. IP Toll Free 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.

In the sample configuration, an Avaya Aura® Session Border Controller (SBC) is used as an edge device between the Avaya CPE and Verizon Business. The SBC performs SIP header manipulation and topology hiding to convert the private Avaya CPE IP addressing to IP addressing appropriate for the Verizon access method.

Customers using Avaya Communication Server 1000E with the Verizon Business IP Contact Center services are able to receive inbound toll-free calls from the PSTN via the SIP protocol. The converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI.

For more information on the Verizon Business IP Contact Center service, including access alternatives, visit http://www.verizonbusiness.com/products/contactcenter/ip/

2. General Test Approach and Test Results

Avaya CS1000E location was connected to the Verizon Business IPCC Service, as depicted in **Figure 1.** Avaya equipment was configured to use the commercially available IP Toll Free VoIP Inbound and IP-IVR services that comprise the Verizon Business IPCC services suite.

2.1. Interoperability Compliance Testing

The testing included executing the test cases detailed in Reference [VZ-Test-Plan]. To summarize, the testing included the following successful compliance testing:

- Incoming calls from the PSTN were routed to the toll-free numbers assigned by Verizon Business to the Avaya CS1000E location. These incoming were answered by Avaya IP-UNIStim telephones, Avaya SIP telephones, Avaya digital telephones, and analog telephones. The display of caller ID on display-equipped Avaya telephones was verified.
- Proper disconnect when the PSTN caller abandons a call before answer.
- Proper disconnect when either party hangs up an active call.
- Proper busy tone heard when a PSTN user calls a toll-free number directed to a busy CS1000E user (i.e., if no redirection is configured for user busy conditions).
- Privacy requests for inbound toll-free calls from the PSTN were verified. That is, when privacy is requested by a PSTN caller (e.g., dialing *67 from a mobile phone), the inbound toll-free call can be successfully completed to a CS1000E user while presenting an anonymous display to the CS1000E user.
- SIP OPTIONS monitoring of the health of the SIP trunk was verified. Both Verizon Business and the enterprise SBC can monitor health using SIP OPTIONS. The Avaya Aura® SBC configurable control of SIP OPTIONS timing was exercised successfully.
- Calls using the G.729A (IP Toll Free) and G.711 ULAW (IP-IVR) codecs, and proper protocol procedures related to media
- DTMF transmission using RFC 2833.
- Inbound toll-free call long holding time call stability
- Telephony features such as call waiting, hold, transfer, and conference. Note that CS1000E will not send REFER to the Verizon network.
- Proper DiffServ markings for SIP signaling and RTP media sent to Verizon

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results as described in Section 2.1. The following observations may be noteworthy:

1. The Verizon IPCC Service does not support fax.

2. Although the Verizon Business IP Contact Center service supports transfer using the SIP REFER method, Avaya CS1000E does not support sending REFER to Verizon.

3. The SIP protocol allows sessions to be refreshed for calls that remain active for some time. In the tested configuration, neither Verizon nor CS1000E send re-INVITE or UPDATE

messages to refresh a session. In the tested configuration, this is transparent to the users that are party to the call in that the media paths remain established.

4. When Avaya Aura® Session Border Controller generates a SIP response (e.g., 180 Ringing or 200 OK for an incoming toll-free call from Verizon), an empty "Request:" header is included. Although this does not have a negative effect on calls (i.e., no user-perceivable problem was observed), product defect **PD00016834** is expected to correct this in a forthcoming SBC service pack.

5. Since the Avaya CPE will respond to an incoming IP-IVR call with 180 Ringing (without SDP), the Verizon IP-IVR service programming must provide pre-answer call treatments (e.g., ring back tone or other network-provided call treatments).

2.3. Support

2.3.1 Avaya

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

2.3.2 Verizon

For technical support on Verizon Business IPCC service, visit the online support site at http://www.verizonbusiness.com/us/customer/.

3. Reference Configuration

Figure 1 illustrates an example Avaya CS1000E solution connected to the Verizon Business IPCC service. Avaya equipment is located on a private IP network. An enterprise edge router provides access to the Verizon IPCC service network via a T1 circuit provisioned for the Verizon Business Private IP (PIP) service. At the edge of the Avaya CPE location, an Avaya Aura® Session Border Controller (SBC) provides topology hiding 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.

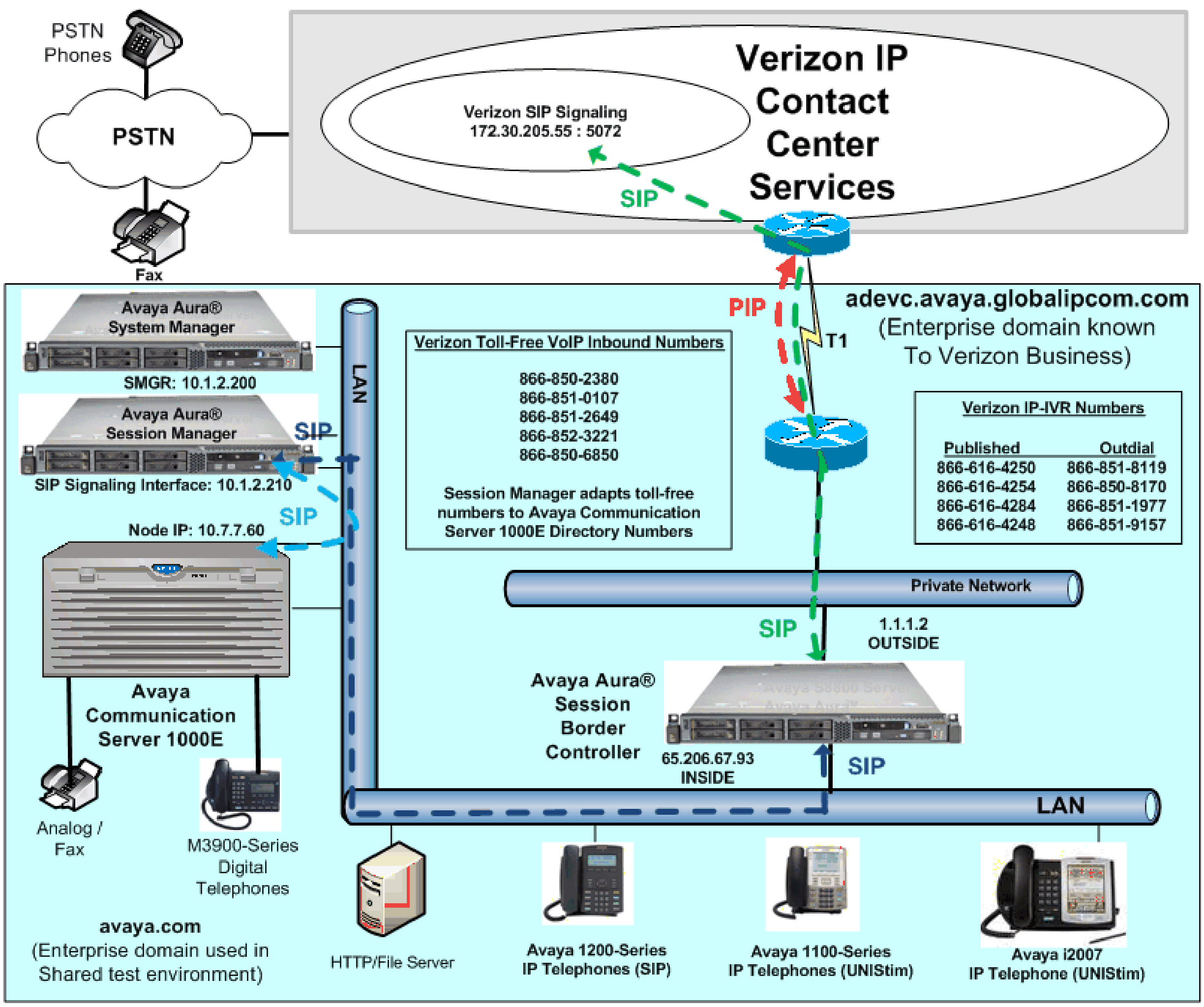


Figure 1: Verizon IP Contact Center Avaya Interoperability Test Lab Configuration

The Avaya CPE was known to Verizon Business as FQDN *adevc.avaya.globalipcom.com*. For efficiency, the Avaya environment utilizing Session Manager Release 6.1 and Communication Server 1000E Release 7.5 was shared among many ongoing test efforts at the Avaya Solution and Interoperability Test lab. Access to the Verizon Business IPCC service was added to a configuration that already used domain "avaya.com" at the enterprise. Session Manager is used to adapt the "avaya.com" domain to the domains known to Verizon. These Application Notes indicate the configuration that would not be required in cases where the CPE domain in Communication Server 1000E and Session Manager match the CPE domain known to Verizon.

Table 1 lists a sampling of Verizon Business IP Toll-Free numbers that terminated at the Avaya CS1000E location. These toll-free numbers were mapped to Avaya CS1000E users via an Avaya Aura® Session Manager adaptation.

Verizon IP Toll-Free Number	Avaya CS1000E Destination	Notes
866-850-2380	x57005	Avaya M3903 Digital Telephone
866-850-6850	x57003	Avaya IP Phone 2007 (UNIStim)
866-852-3221	x57007	Avaya 1200-Series IP Deskphone (SIP)

Table 1: Sample Verizon IP Toll Free Number to CS1000E Telephone Mappings

Table 2 lists a sampling a sampling of Verizon Business IP-IVR numbers that terminated at the Avaya CS1000E location. The IP-IVR Outdial numbers were mapped to Avaya CS1000E users via an Avaya Aura® Session Manager adaptation.

Verizon IP IVR Published Number	Verizon IP IVR Outdial Number	Avaya CS1000E Destination	Notes
866-616-4250	866-851-8119	x57005	Avaya M3903 Digital Telephone
866-616-4254	866-850-8170	x57007	Avaya 1200-Series IP Deskphone (SIP)
866-616-4284	866-851-1977	x57003	Avaya IP Phone 2007 (UNIStim)

Table 2: Sample Verizon IP-IVR Number to CS1000E Telephone Mappings

In the sample configuration, the Verizon Business PIP circuit previously shown in **Figure 1** enabled access to the Verizon IP Trunk Service as well as the Verizon IPCC Service. The companion Application Notes available in Reference [AuraSBC-IP-Trunk] detail the overall configuration for access to the Verizon IP Trunk service. Although Verizon IP Trunk service is not the focus of these Application Notes, the figure below is included because the Verizon IPCC Service configuration builds upon the configuration detailed in Reference [AuraSBC-IP-Trunk], particularly Avaya Aura® Session Border Controller configuration in Section 7.

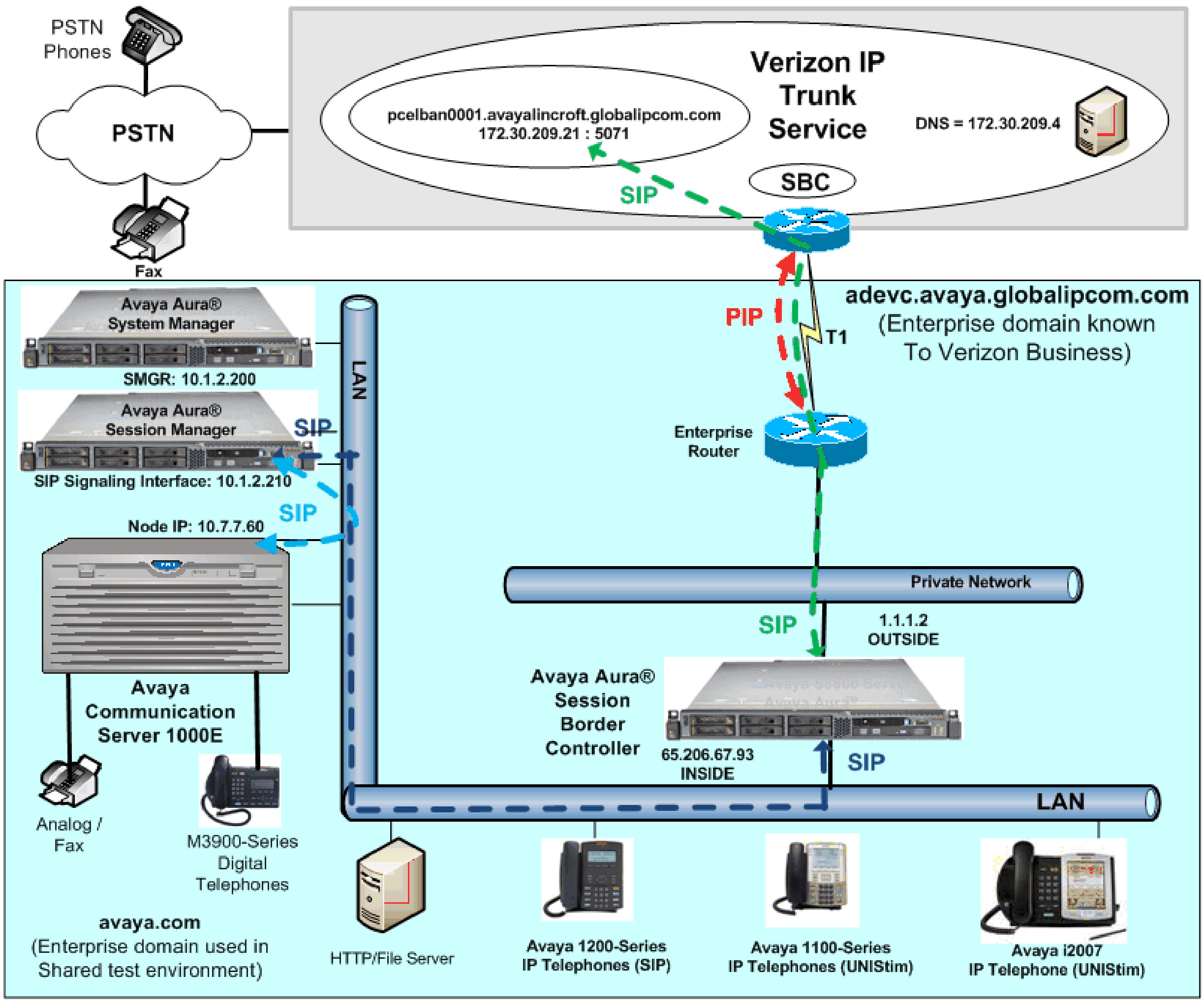


Figure 2: Verizon IP Trunk Avaya Interoperability Test Lab Configuration

The following components were used in the sample configuration:

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

- Avaya CPE Fully Qualified Domain Name (FQDN) known to Verizon
  - *adevc.avaya.globalipcom.com*
- Avaya Aura® Session Border Controller (SBC)
- Avaya Communication Server 1000E Release 7.5
- Avaya Aura® System Manager Release 6.1
- Avaya Aura® Session Manager Release 6.1
- Avaya IP-2007 UNIStim telephones
- Avaya 1100-Series IP Deskphones using UNIStim software
- Avaya 1200-Series IP Deskphones using SIP software, registered to CS1000E
- Avaya M3900-Series Digital phones
- Analog telephones

4. Equipment and Software Validated

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

Equipment	Software
Avaya Communication Server 1000E running on CP+PM server as co-resident configuration	Release 7.5, Version 7.50.17 (with latest Patches and Deplist) Plug-in 201 Enabled Plug-in 501 Enabled
Avaya S8800 Server (System Manager)	Avaya Aura® System Manager Release 6.1.5.0 (Build Number 6.1.0.0.7345 Patch 6.1.5.7)
Avaya S8800 Server (Session Manager)	Avaya Aura® Session Manager Release 6.1 (Load 6.1.1.0.611023)
Avaya S8800 Server (Session Border Controller)	Avaya Aura® Session Border Controller Release 6.0 SBC Template SBCT 6.0.0.1.5
Avaya 1100-Series IP Deskphones (UNIStim)	FW 0624C8A
Avaya 1200-Series IP Deskphones (SIP)	SIP 04.00.04.00
Avaya IP Phone 2007 (UNIStim)	FW 0621C8A
Avaya M3900-Series Digital Telephone	N/A

Table 3: Equipment and Software Used in the Sample Configuration

5. Configure Avaya Communication Server 1000E

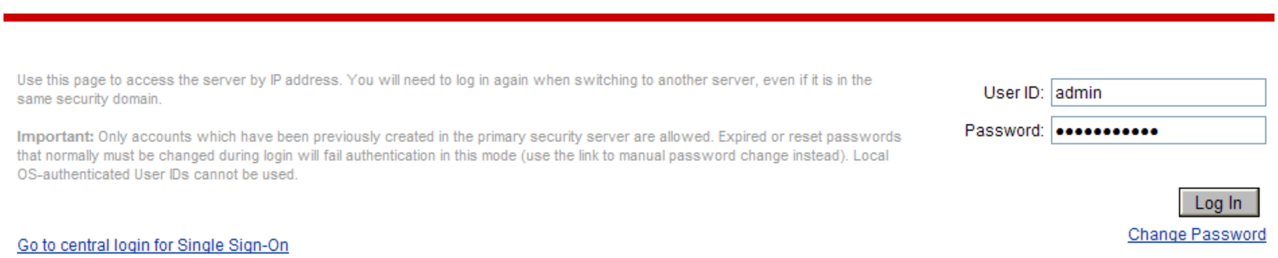
This section describes the Avaya Communication Server 1000E configuration, focusing on the SIP Trunk to Session Manager. As described in Section 3, the same Avaya Communication Server 1000E SIP Trunking configuration was used to test both Verizon IP Trunk Service and Verizon IPCC Service. The configuration for outbound calling using Verizon IP Trunk Service is more fully documented in the companion Application Notes in Reference [AuraSBC-IP-Trunk].

In the sample configuration, Avaya Communication Server 1000E Release 7.5 was deployed as a co-resident system with the SIP Signaling Server and Call Server applications all running on the same CP-PM server platform.

Avaya Aura® Session Manager Release 6.1 provides all the SIP Proxy Service (SPS) and Network Connect Services (NCS) functions previously provided by the Network Routing Service (NRS). As a result, the NRS application is not required to configure a SIP trunk between Avaya Communication Server 1000E and Session Manager Release 6.1.

This section focuses on the SIP Trunking configuration. Although sample screens are illustrated to document the overall configuration, it is assumed that the basic configuration of the Call Server and SIP Signaling Server applications has been completed, and that the Avaya Communication Server 1000E is configured to support analog, digital, UNIStim, and SIP telephones. For references on how to administer these functions of Avaya Communication Server 1000E, see Section 11**.**

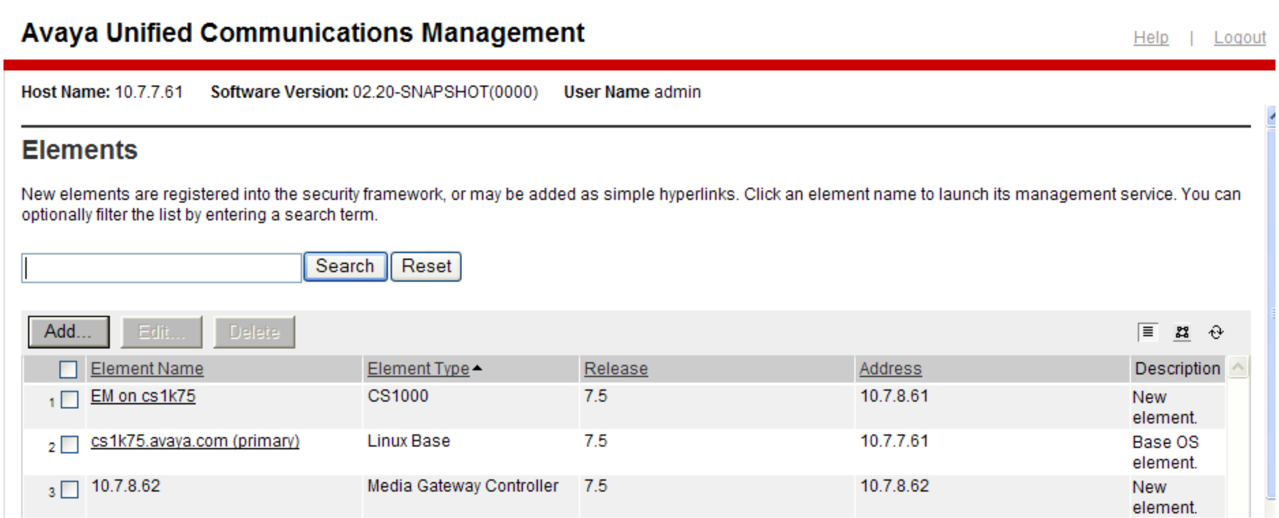
Configuration will be shown using the web based Avaya Unified Communications Management GUI. The Avaya Unified Communications Management GUI may be launched directly via **https://<ip-address>** where the relevant <ipaddress> in the sample configuration is 10.7.7.61. The following screen shows an abridged log in screen. Log in with appropriate credentials.



Alternatively, if Avaya Aura® System Manager has been configured as the Primary Security Server for the Avaya Unified Communications Management application and Avaya Communication Server 1000E is registered as a member of the System Manager Security framework, the GUI may be accessed via System Manager. In this case, access the web based GUI of Avaya Aura® System Manager by using the URL **"http://<ip-address>/SMGR"**, where **<ip-address>** is the IP address of Avaya Aura® System Manager. Log in with appropriate credentials.

The Avaya Aura® System Manager Home Page will be displayed. Under the **Services** category on the right side of the page, click the **UCM Services** link.

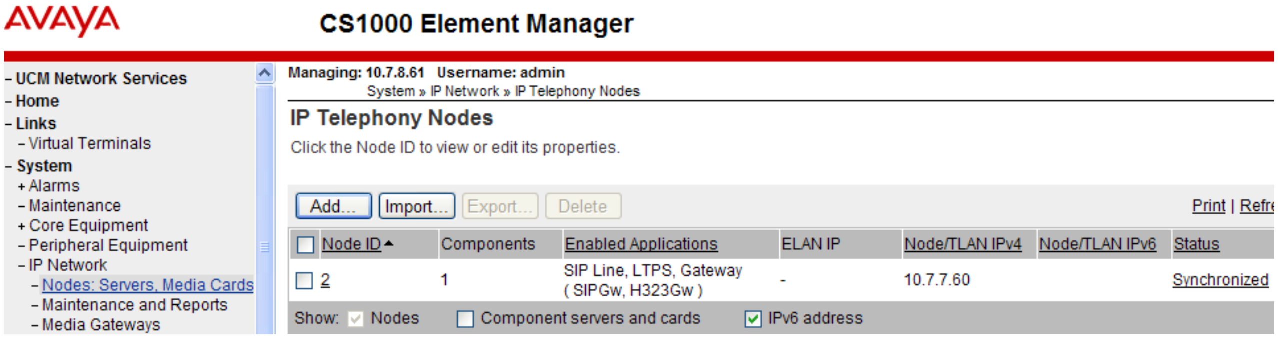
Whether the CS1000E is accessed directly or via System Manager, the Avaya Unified Communications Management **Elements** page will be used for configuration. Click on the **Element Name** corresponding to "CS1000" in the **Element Type** column. In the abridged screen below, the user would click on the **Element Name** "EM on cs1k75".



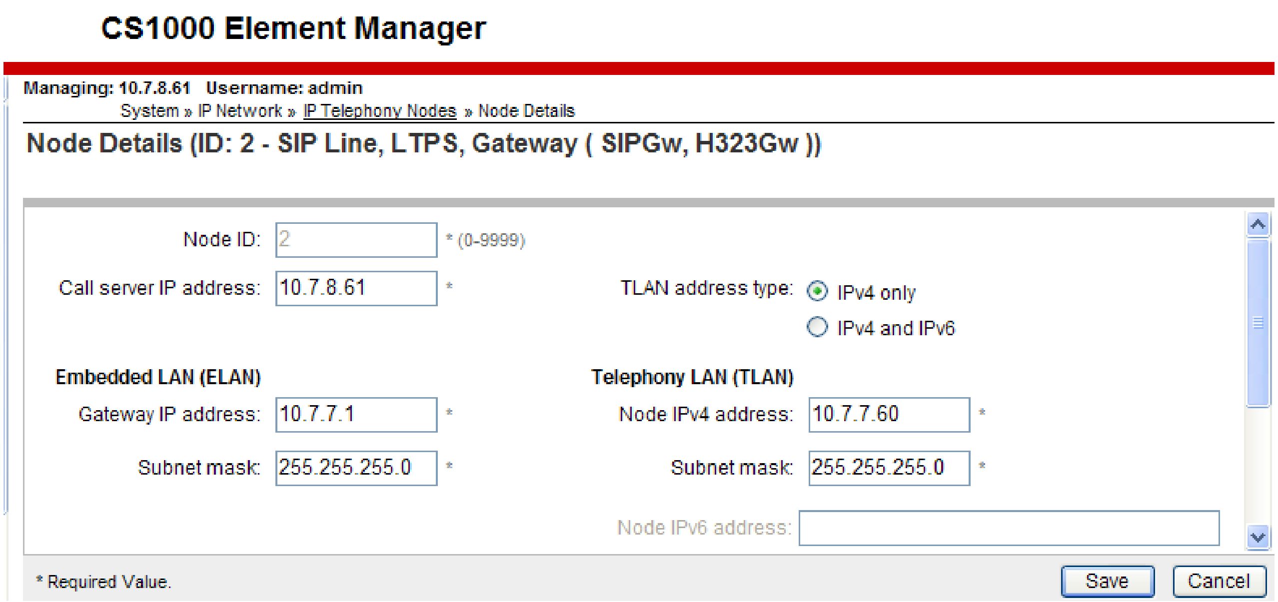
5.1. Node and Key IP Addresses

Expand **System → IP Network** on the left panel and select **Nodes: Servers, Media Cards**.

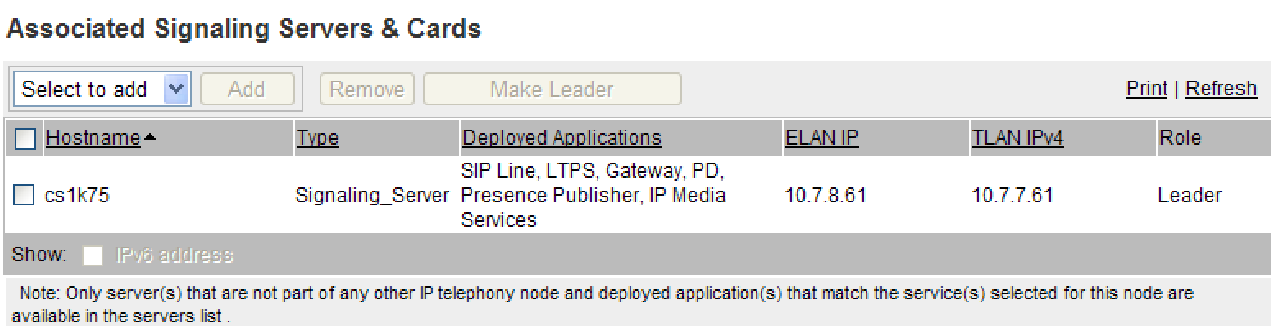
The **IP Telephony Nodes** page is displayed as shown below. Click "**<Node id>**" in the **Node ID** column to view details of the node. In the sample configuration, **Node ID "2"** was used.



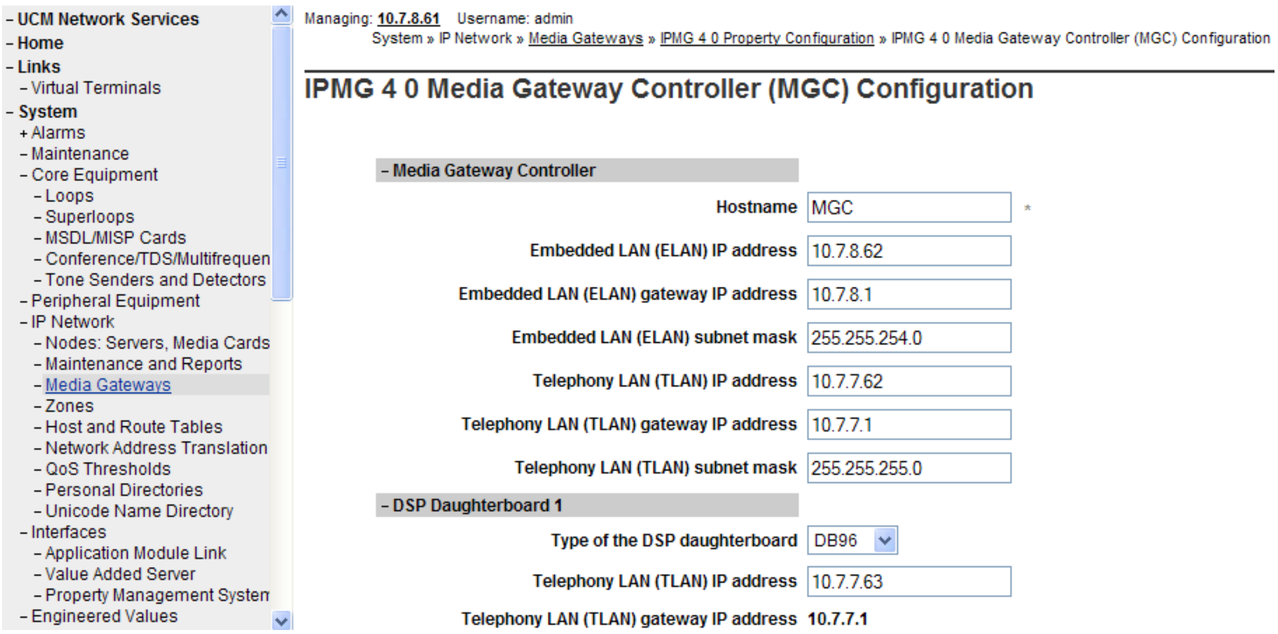
The **Node Details** screen is displayed with additional details as shown below. Under the **Node Details** heading at the top of the screen, make a note of the **TLAN Node IPV4 address.** In the sample screen below, the **Node IPV4 address** is **"**10.7.7.60**".** This IP address will be needed when configuring Session Manager with a SIP Entity for the CS1000E.



The following screen shows the **Associated Signaling Servers & Cards** heading at the bottom of the screen, simply to document the configuration.



Expand **System → IP Network** on the left panel and select **Media Gateways**. The **Telephony LAN (TLAN) IP Address** under the **DSP Daughterboard 1** heading will be the IP Address in the SDP portion of SIP messages, for calls requiring a gateway resource. For example, for an inbound Verizon toll-free call to a digital telephone, the IP Address in the SDP in the 200 OK message will be 10.7.7.63 in the sample configuration.

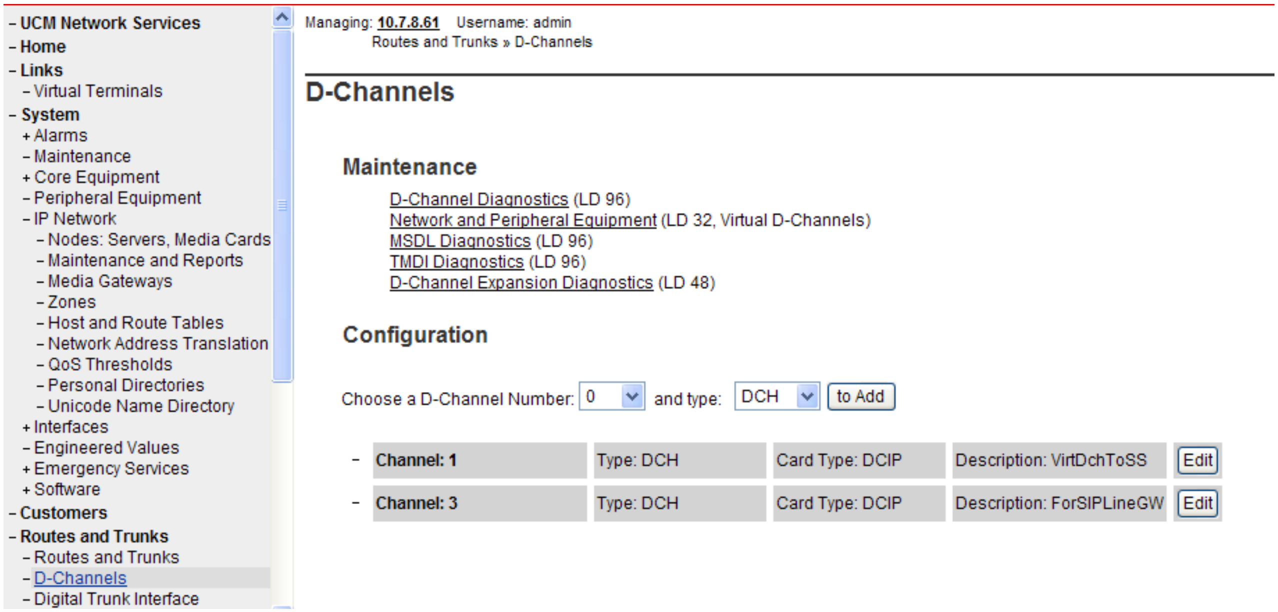


5.2. Virtual D-Channel, Routes and Trunks

Avaya Communication Server 1000E Call Server utilizes a virtual D-channel and associated Route and Trunks to communicate with the Signaling Server.

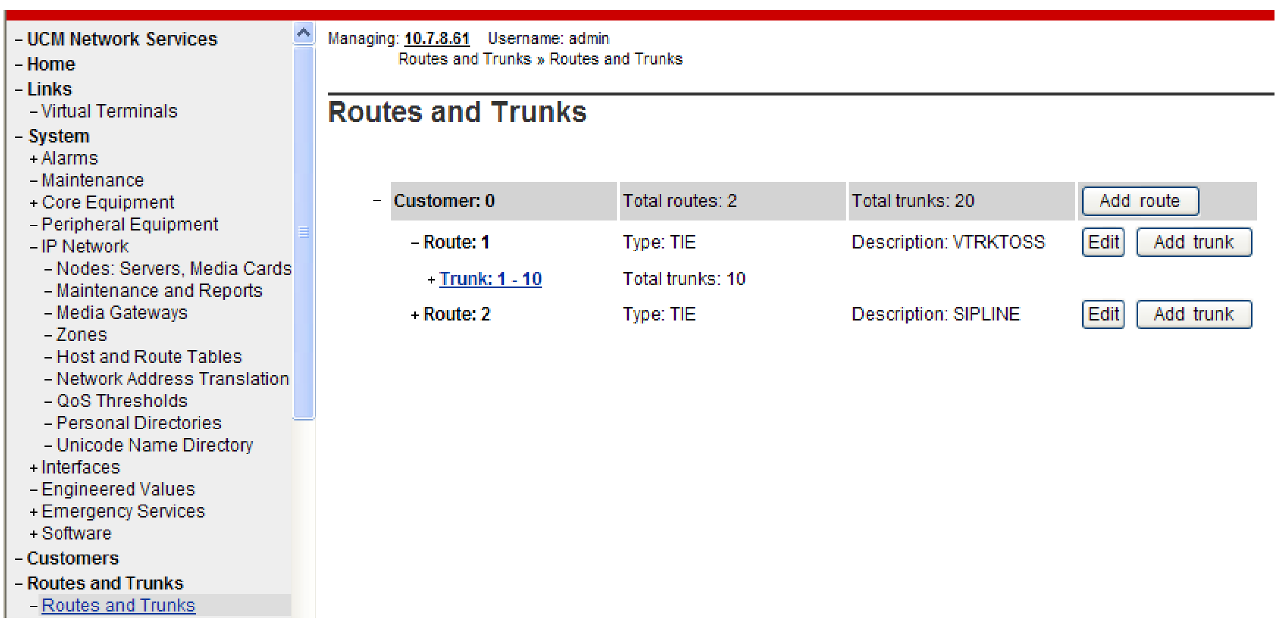
5.2.1 Virtual D-Channel Configuration

Expand **Routes and Trunks** on the left navigation panel and select **D-Channels**. In the sample configuration, there is a virtual D-Channel 1 associated with the Signaling Server.

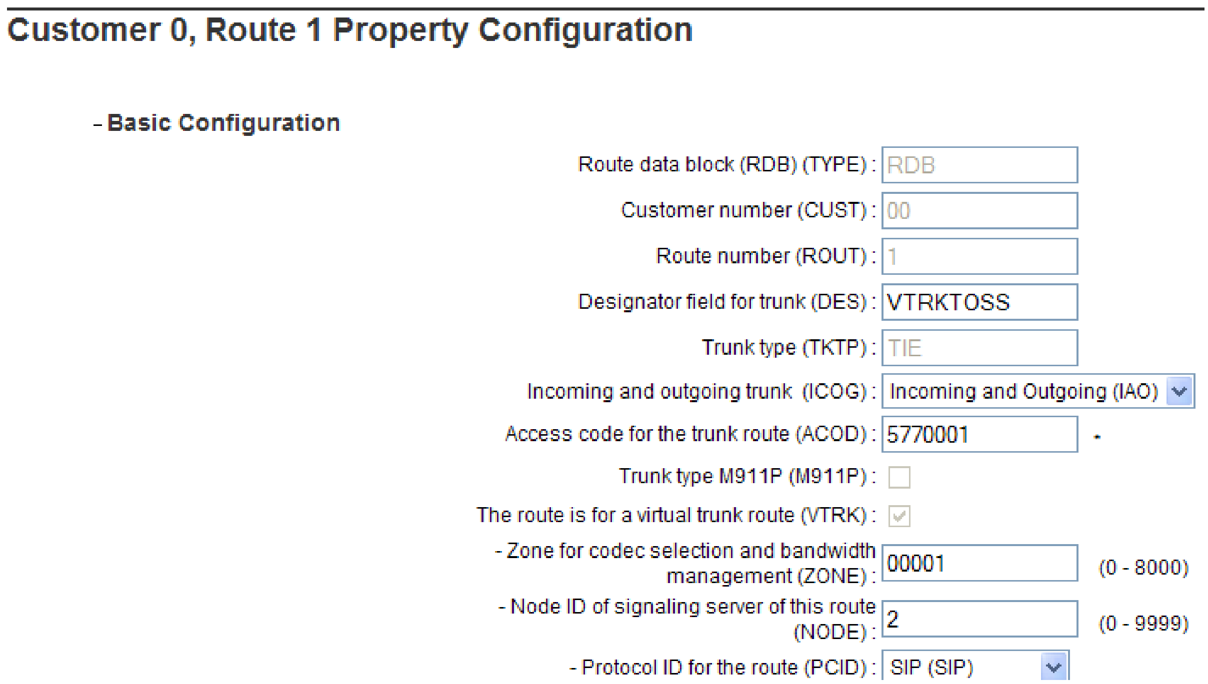


5.2.2 Routes and Trunks Configuration

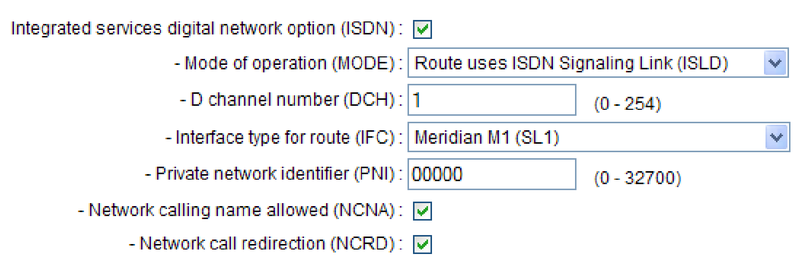
In addition to configuring a virtual D-channel, a **Route** and associated **Trunks** must be configured. Expand **Routes and Trunks** on the left navigation panel and expand the customer number. In the screen that follows, it can be observed that Route 1 has 10 trunks in the sample configuration.



Select **Edit** to verify the configuration, as shown below. Verify "**SIP (SIP)**" has been selected for **Protocol ID for the route (PCID)** field and the **Node ID of signaling server of this route (NODE)** matches the node shown in Section 5.1. As can be observed in the **Incoming and outgoing trunk (ICOG)** parameter, incoming and outgoing calls are allowed. Recall that the same configuration is used for Verizon IP Trunk Service, which supports outbound dialing to the PSTN. The **Access code for the trunk route (ACOD)** will in general not be dialed, but the number that appears in this field may be observed on Avaya CS1000E display phones if an incoming call on the trunk is anonymous or marked for privacy. The **Zone for codec selection and bandwidth management (ZONE)** parameter can be used to associate the route with a zone for configuration of the audio codec preferences sent via the Session Description Protocol (SDP) in SIP messaging.



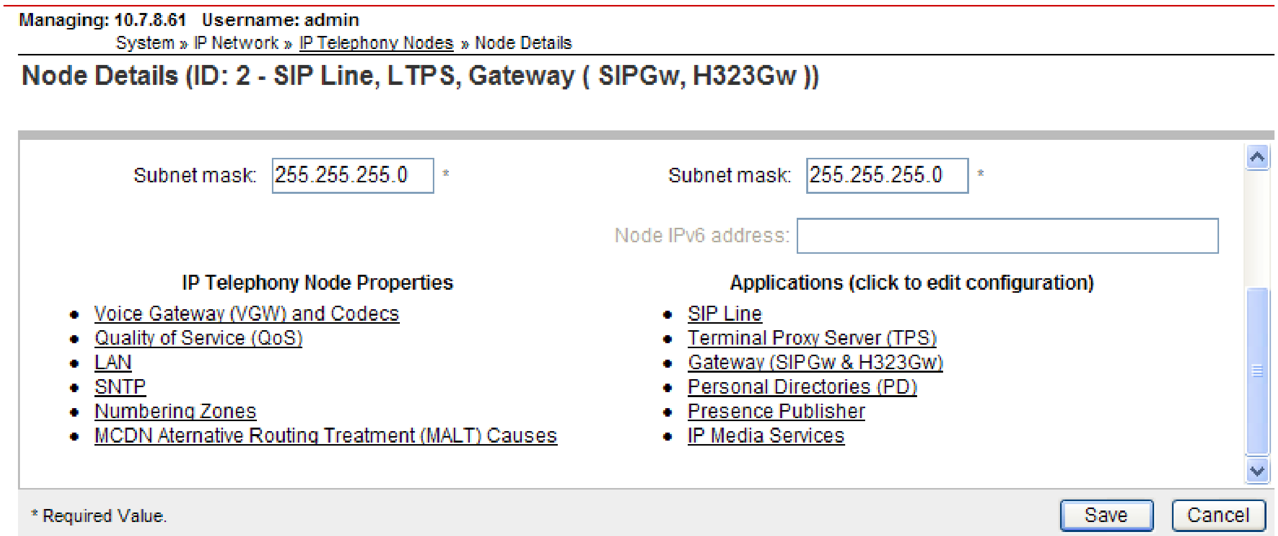
Scrolling down, other parameters may be observed. The **D channel number (DCH)** field must match the D-Channel number shown in Section 5.2.1.



5.3. SIP Trunk to Session Manager

Expand **System → IP Network → Nodes: Servers, Media Cards**. Click "**2**" in the **Node ID** column (not shown) to edit configuration settings for the configured node.

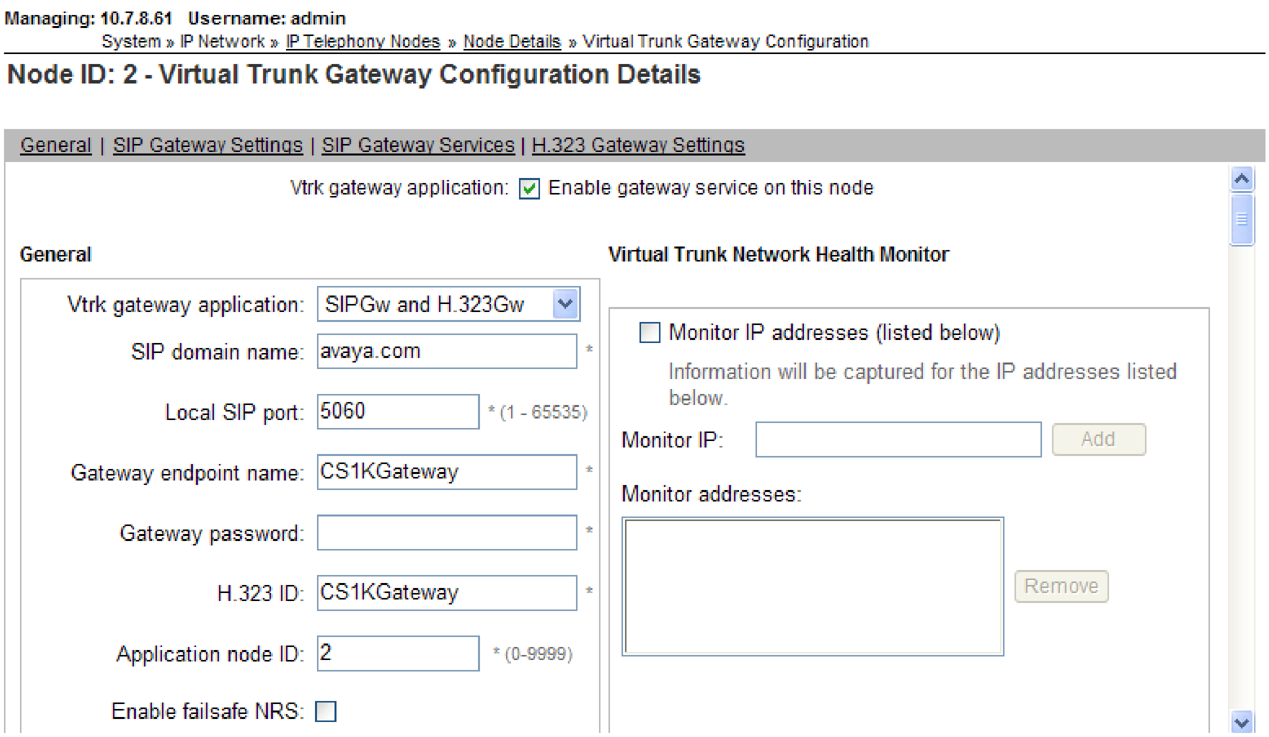
Using the scroll bar on the right side of the screen, navigate to the **Applications** section on the screen and select the **Gateway (SIPGw & H323Gw)** link to view or edit the SIP Gateway configuration.



On the **Node ID: 2 - Virtual Trunk Gateway Configuration Details** page, enter the following values and use default values for remaining fields.

- **SIP domain name:** Enter the appropriate SIP domain for the customer network. In the sample configuration, "**avaya.com**" was used in the shared Avaya Solution and Interoperability Test lab environment. The SIP domain name for the enterprise known to Verizon is "adevc.avaya.globalipcom.com", and the SIP domain will be adapted by Session Manager for calls to and from the Avaya CS1000E. If no such adaptation is required, enter the domain known to Verizon.
- **Local SIP port:** Enter "**5060**"
- **Gateway endpoint name:** Enter descriptive name
- **Application node ID:** Enter "**<Node id>**". In the sample configuration, "**2**" was used, matching the node shown in Section 5.1.

The values defined for the sample configuration are shown below.

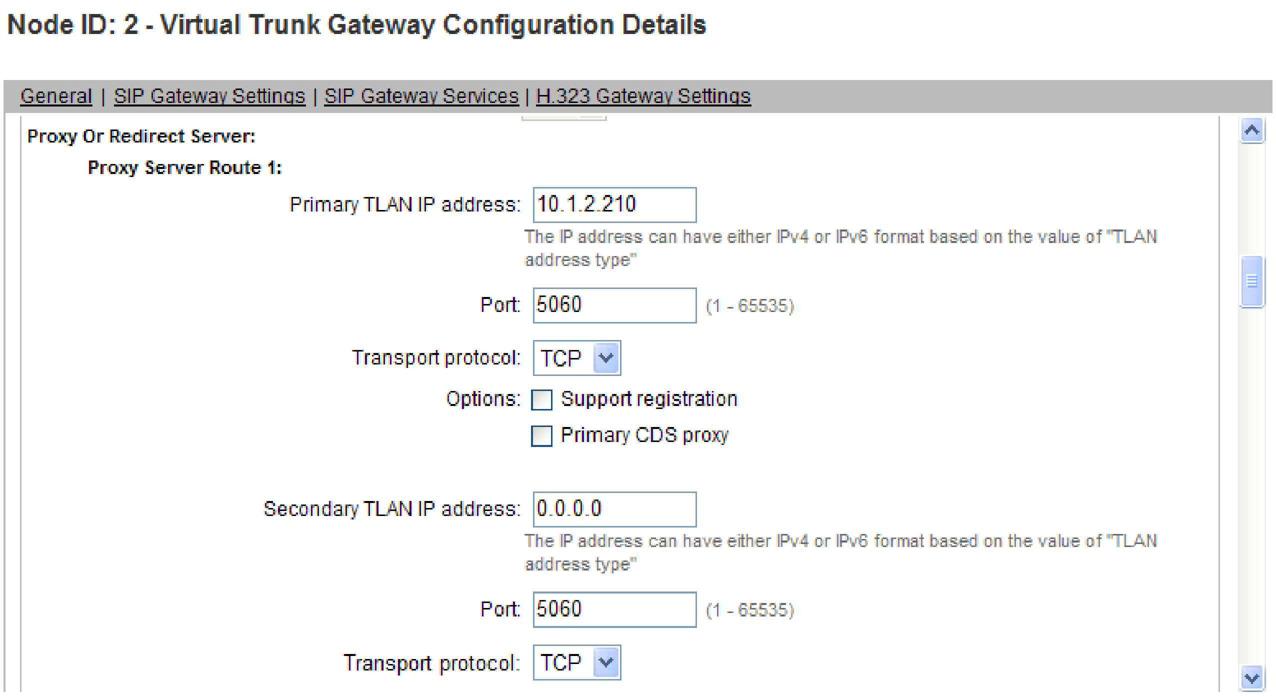


Scroll down to the **SIP Gateway Settings** → **Proxy or Redirect Server**: section.

Under **Proxy Server Route 1**, enter the following and use default values for remaining fields.

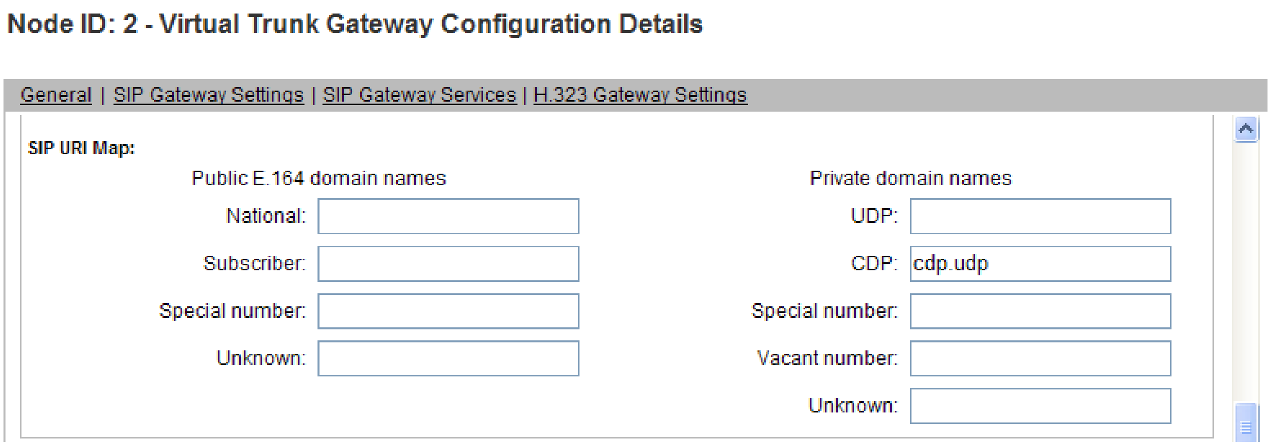
- **Primary TLAN IP address:** Enter the IP address of the Session Manager SIP signaling interface. In the sample configuration, "**10.1.2.210**" was used.
- **Port:** Enter "**5060**"
- **Transport protocol:** Select "**TCP**"

The values defined for the sample configuration are shown below.



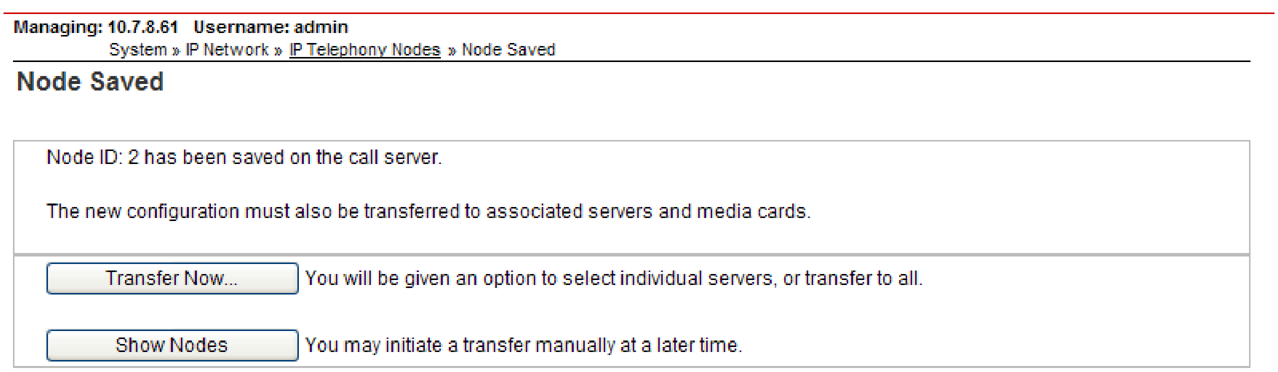
Scroll down and repeat these steps for the **Proxy Server Route 2**.

Scroll down to the **SIP URI Map** section. The values defined for the sample configuration are shown below. In general, the **SIP URI Map** values have been set to blank for call types that may ultimately interact with Verizon. The Avaya CS1000E will put the "string" entered in the **SIP URI Map** in the "phone-context=<string>" parameter in SIP headers such as the P-Asserted-Identity. If the value is configured to blank, the CS1000E will omit the "phone-context=" in the SIP header altogether.

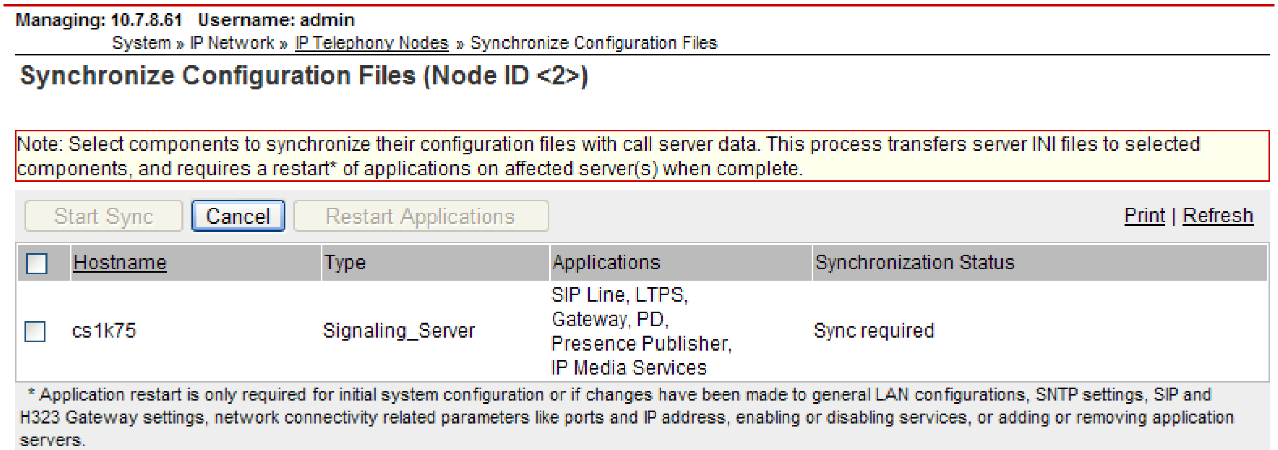


Scroll to the bottom of the page and click **Save** (not shown) to save SIP Gateway configuration settings. This will return the interface to the **Node Details** screen. Click **Save** on the **Node Details** screen (not shown).

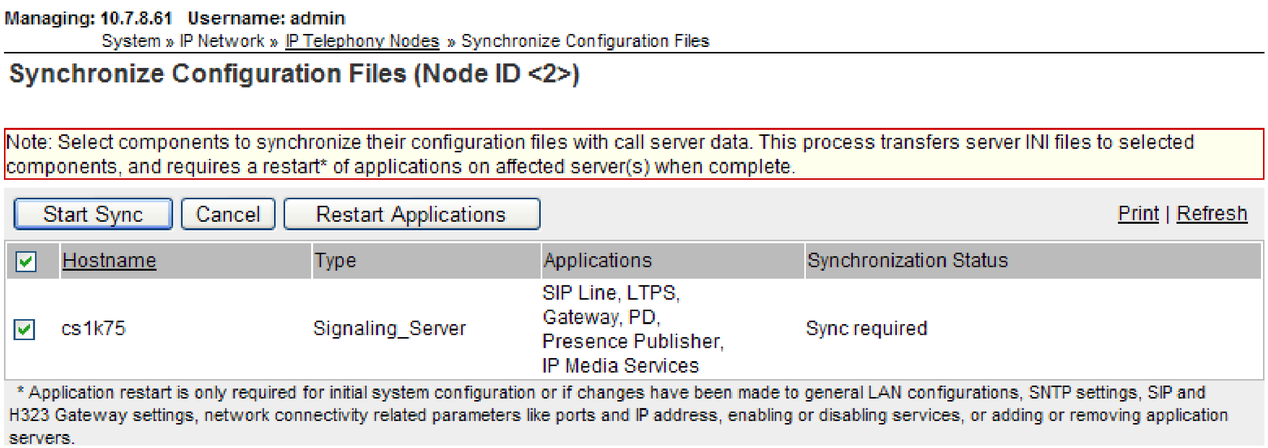
Select **Transfer Now** on the **Node Saved** page as shown below.



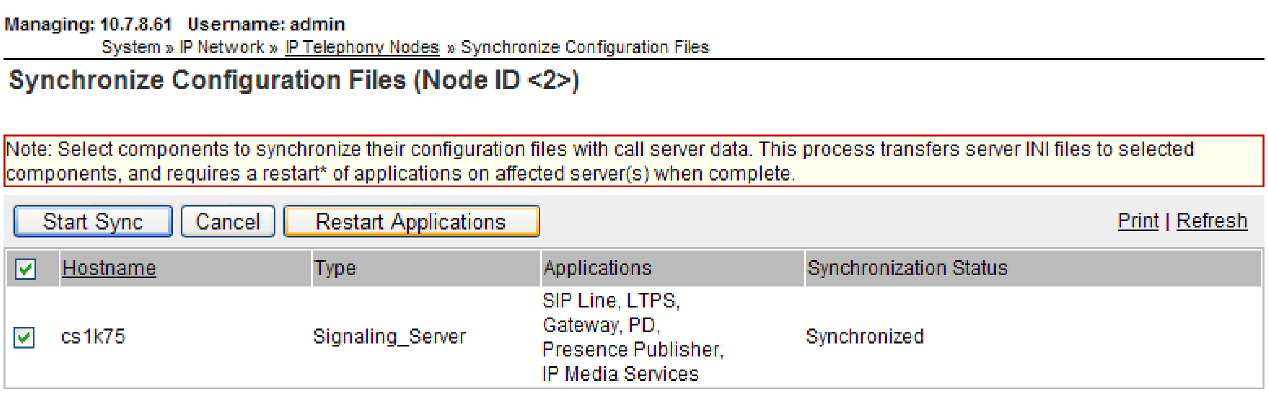
Once the transfer is complete, the **Synchronize Configuration Files (Node ID <id>)** page is displayed.



Enter associated with the appropriate Hostname and click **Start Sync.** The screen will automatically refresh until the synchronization is finished.



The **Synchronization Status** field will update from **Sync required** (as shown above) to **Synchronized** (as shown below). After synchronization completes, enter associated with the appropriate Hostname and click **Restart Applications**.

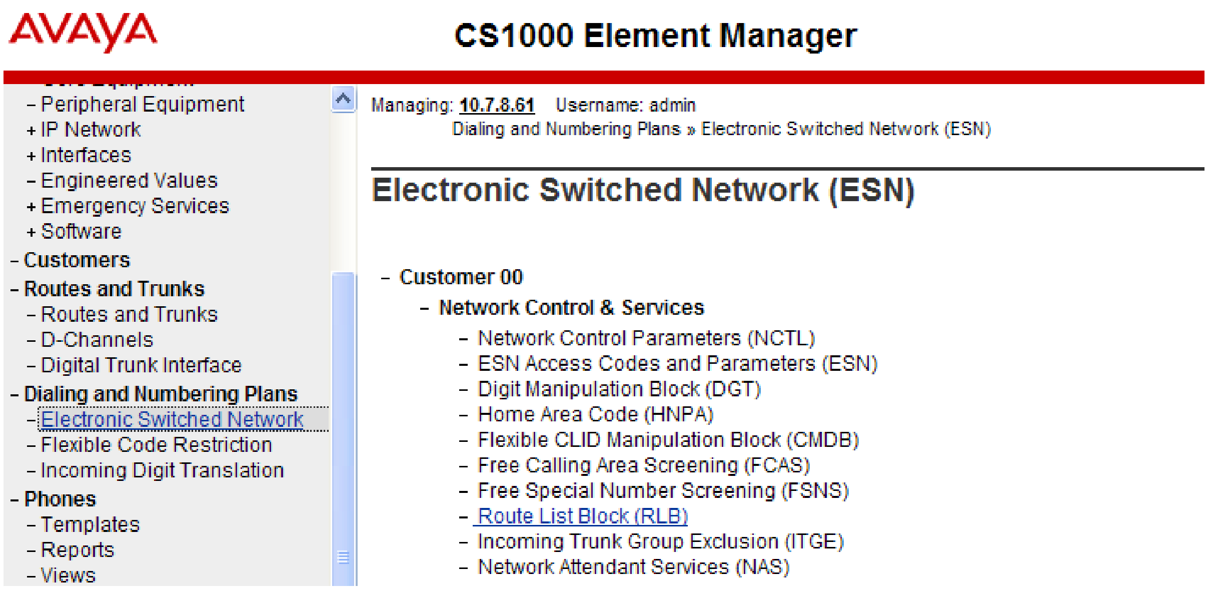


5.4. Routing of Dialed Numbers to Session Manager

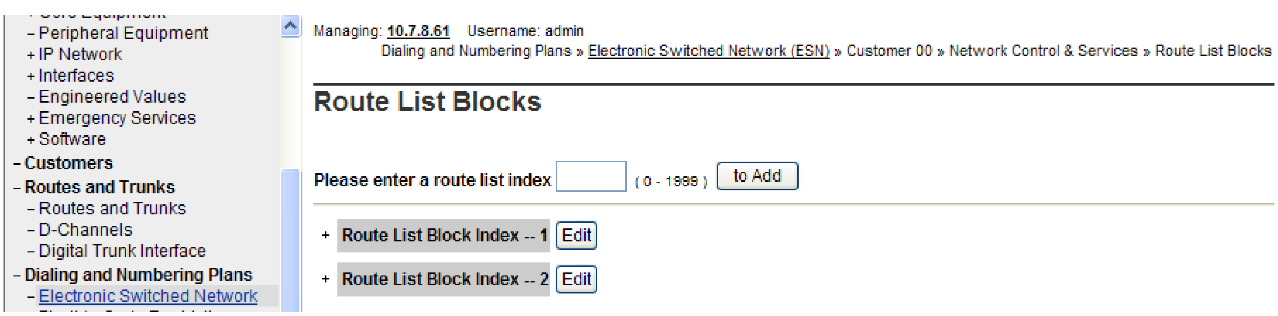
This section illustrates routing used in the sample configuration for routing calls over the SIP Trunk between Avaya Communication Server 1000E and Session Manager for calls destined for the Verizon IP Trunk service, which was available from the same PIP circuit as the Verizon IPCC Service. The routing defined in this section is simply informational, and not intended to be prescriptive.

5.4.1 Route List Block

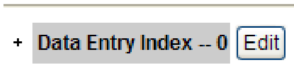
Expand **Dialing and Numbering Plans** on the left navigational panel and select **Electronic Switched Network.** Select **Route List Block (RLB)** on the **Electronic Switched Network (ESN)** page as shown below.



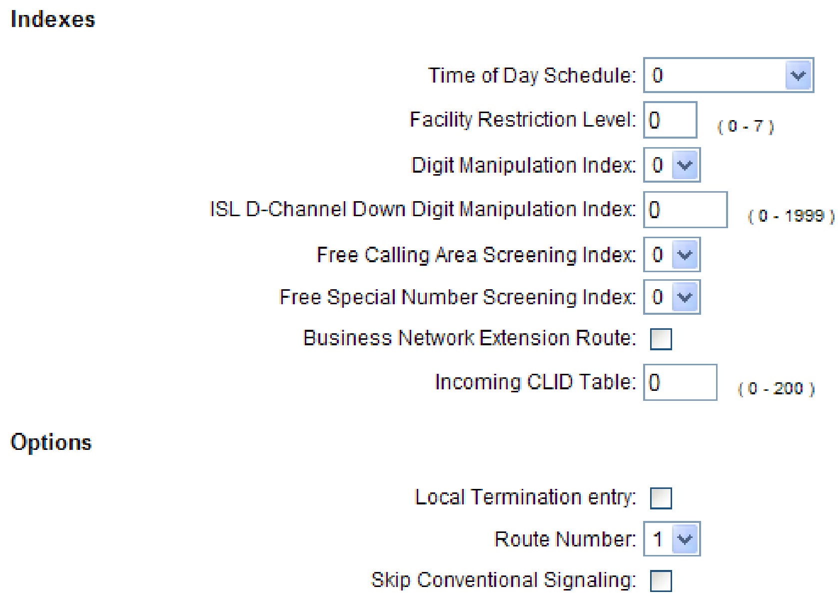
The **Route List Blocks** screen is displayed. Enter an available route list index number in the **Please enter a route list index** field and click **to Add**, or edit an existing entry by clicking the corresponding Edit button. In the sample configuration, route list block index 1 is used.



If adding the route list index anew, scroll down to the **Options** area of the screen. If editing an existing route list block index, select the **Edit** button next to the appropriate **Data Entry Index** as shown below, and scroll down to the **Options** area of the screen.



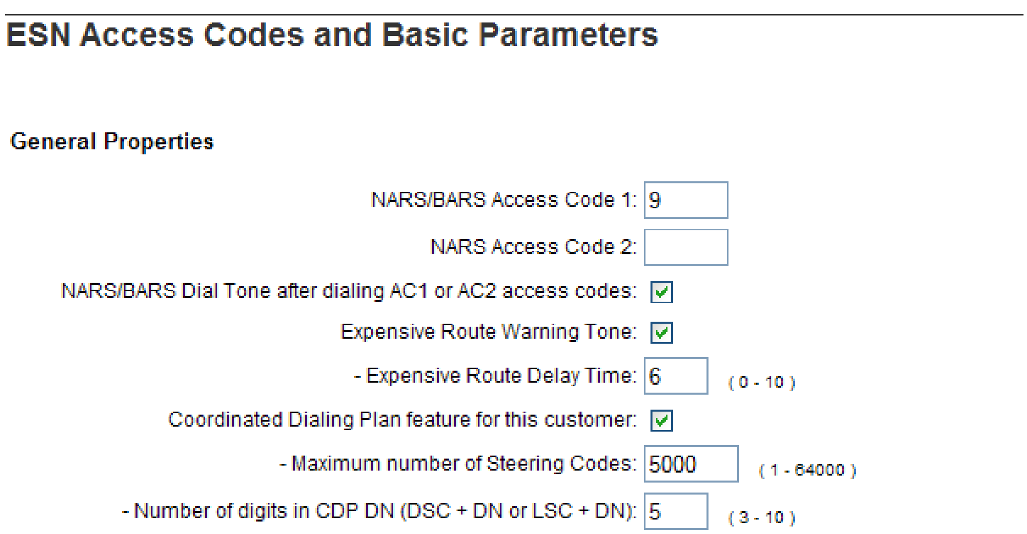
Under the **Options** section, select **“<Route id>”** in the **Route Number** field. In the sample configuration route number 1 was used. Default values may be retained for remaining fields as shown below.



Click **Save** (not shown) to save the Route List Block definition.

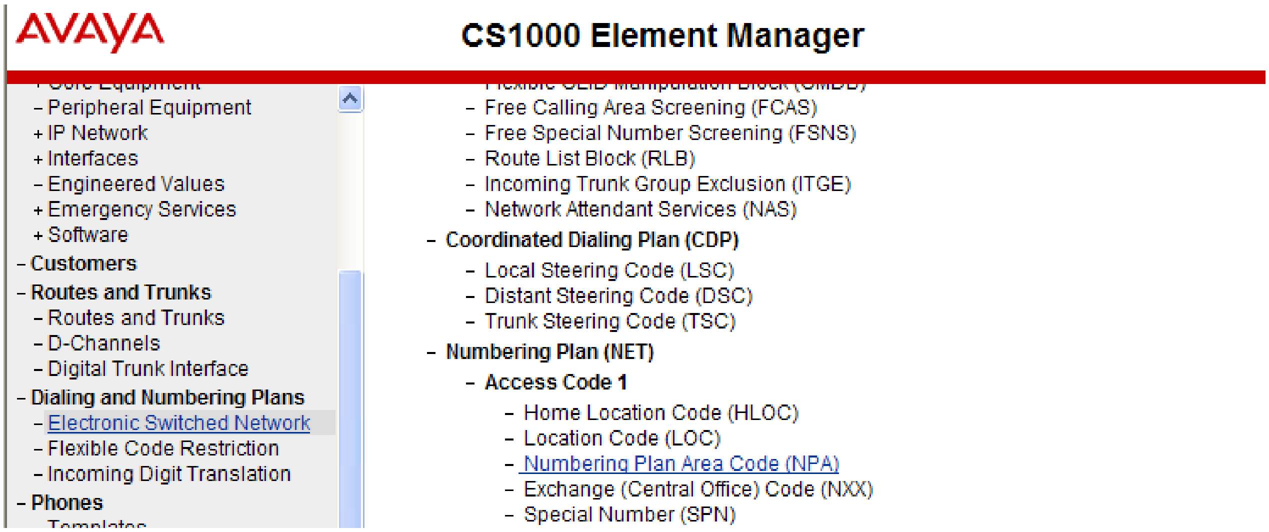
5.4.2 NARS Access Code

Expand **Dialing and Numbering Plans** on the left navigational panel and select **Electronic Switched Network.** Select **ESN Access Codes and Parameters (ESN).** Although not repeated below, this link can be observed in the first screen in Section 5.4.1. In the **NARS/BARS Access Code 1** field, enter the number the user will dial before the target PSTN number. In the sample configuration, the single digit “9” was used.

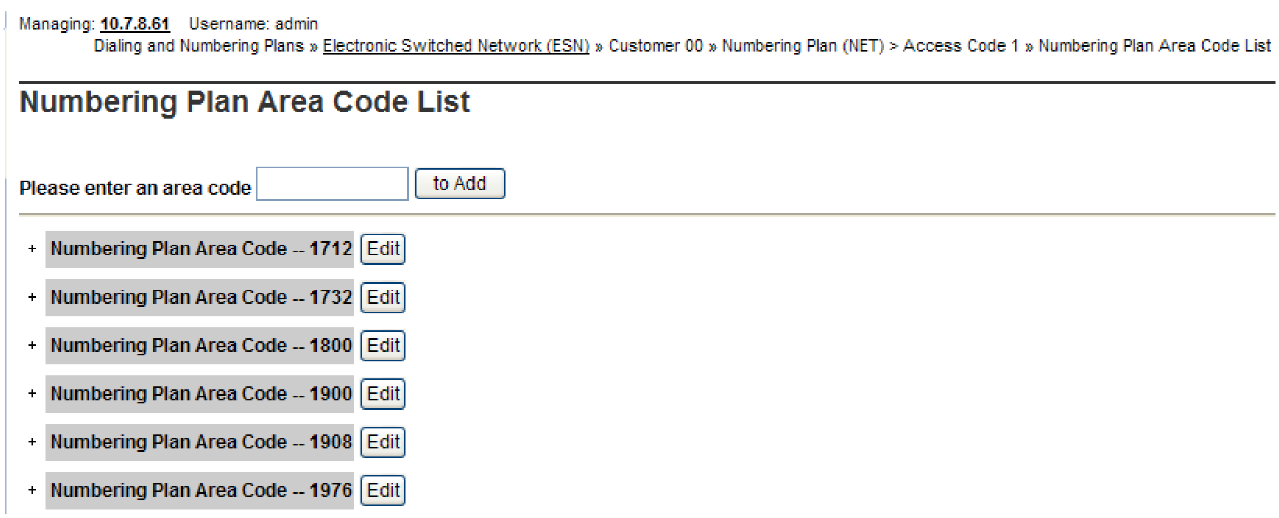


5.4.3 Numbering Plan Area Codes

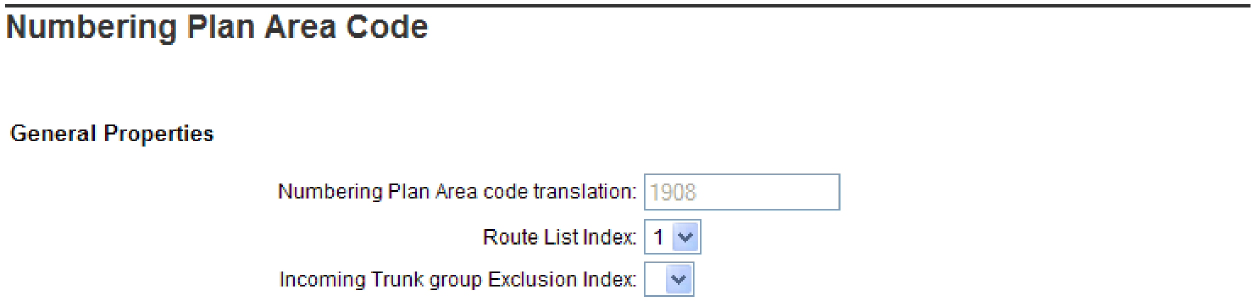
Expand **Dialing and Numbering Plans** on the left navigational panel and select **Electronic Switched Network.** Scroll down and select **Numbering Plan Area Code (NPA)** under the appropriate access code heading. In the sample configuration, this is **Access Code 1**, as shown in below.



Add a new NPA by entering it in the **Please enter an area code** box and click **to Add** or click **Edit** to view or change an NPA that has been previously configured. In the screen below, it can be observed that various dial strings such as 1800 and 1908 are configured.

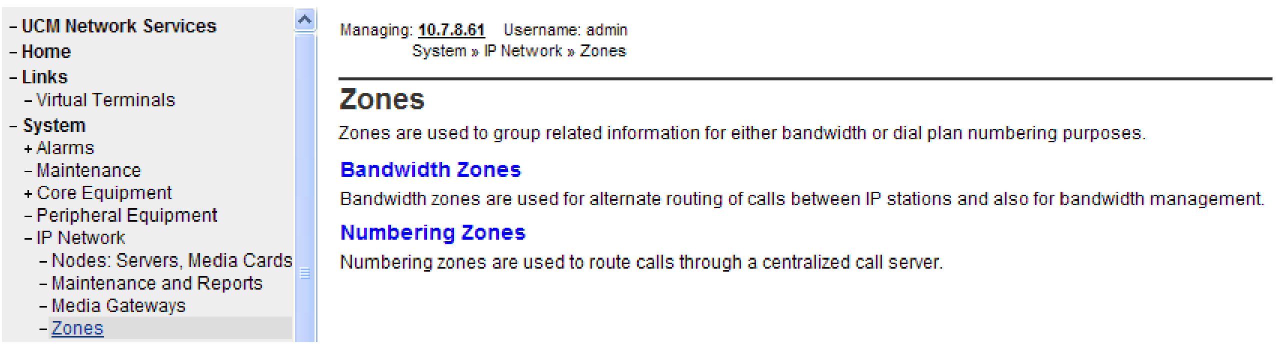


In the screen below, the entry for “1908” is displayed. In the Route List Index, “1” is selected to use the route list associated with the SIP Trunk to Session Manager. Default parameters may be retained for other parameters. Repeat this procedure for the dial strings associated with other numbering plan area codes that should route to the SIP Trunk to Session Manager.

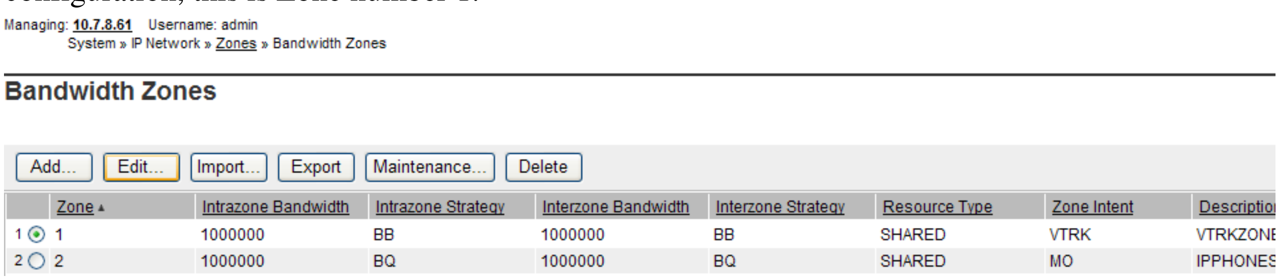


5.5. Zones

Zone configuration can be used to control codec selection and for bandwidth management. To configure, expand **System → IP Network** and select **Zones** as shown below.

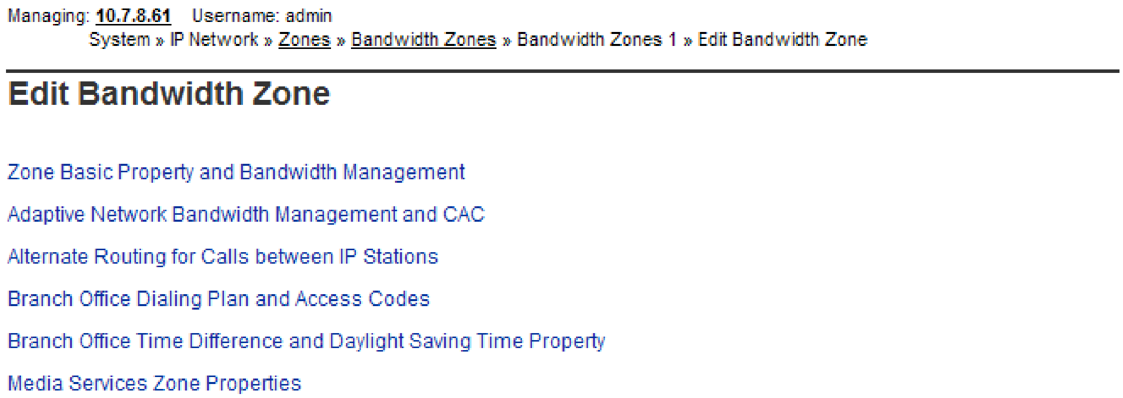


Select **Bandwidth Zones**. In the sample configuration, two zones are configured as shown below. In production environments, it is likely that more zones will be required. Select the zone associated with the virtual trunk to Session Manager and click **Edit** as shown below. In the sample configuration, this is Zone number 1.

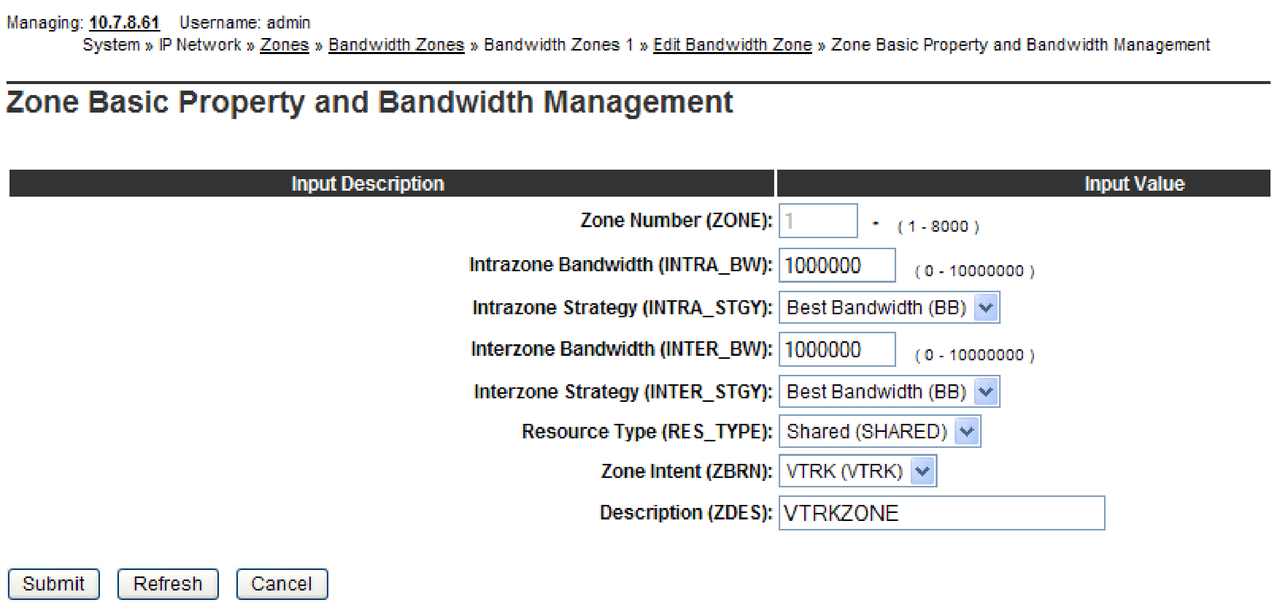


	Zone	Intrazone Bandwidth	Intrazone Strategy	Interzone Bandwidth	Interzone Strategy	Resource Type	Zone Intent	Descriptio
1	1	1000000	BB	1000000	BB	SHARED	VTRK	VTRKZONE
2	2	1000000	BQ	1000000	BQ	SHARED	MO	IPPHONES

In the resultant screen shown below, select **Zone Basic Property and Bandwidth Management**.



The following screen shows the Zone 1 configuration. Note that "Best Bandwidth (BB)" is selected for the zone strategy parameters so that codec G.729A is preferred over codec G.711MU for calls with Verizon. Using the production circuit, inbound Verizon IP Toll Free calls preferred and used G.729A while Verizon offered only G.711MU for inbound IP-IVR calls.

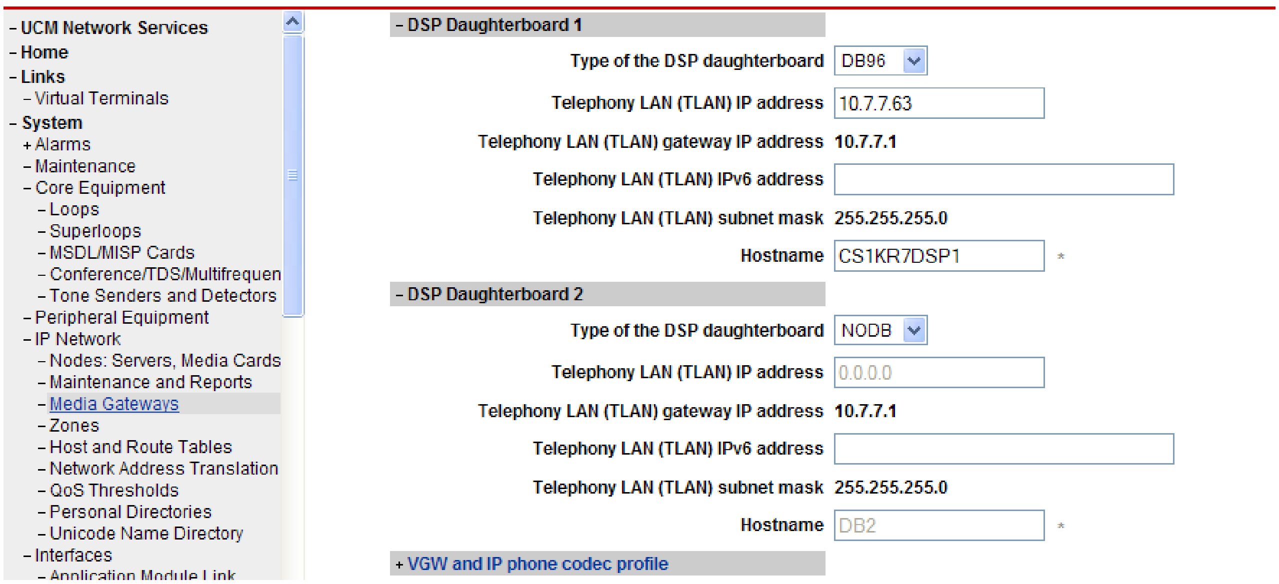


5.6. Codec Parameters, Including Ensuring Annexb=no for G.729

Verizon IPCC Service does not support G.729 Annex B, and Verizon requires that SDP offers and SDP answers in SIP messages include the "annexb=no" attribute when G.729 is used. This section includes the configuration that ensures that the "annexb=no" attribute is included.

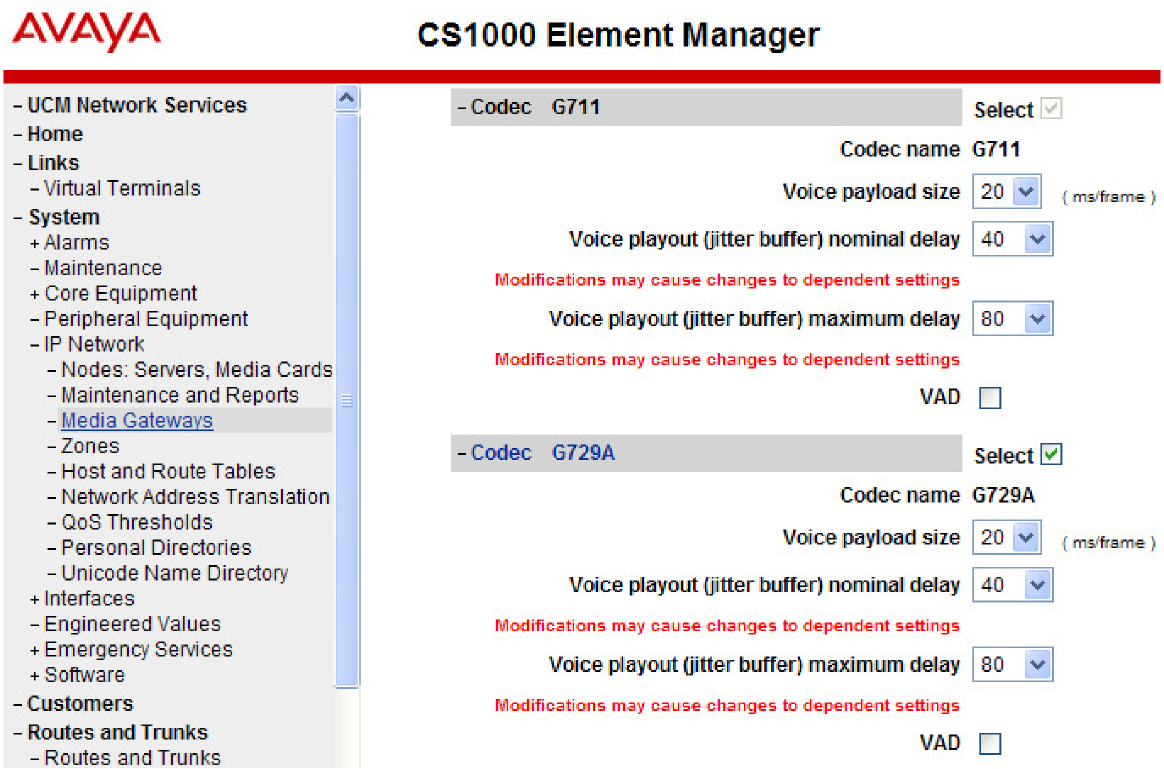
5.6.1 Media Gateway Configuration

To ensure that the "annexb=no" attribute is included, expand **System → IP Network** on the left panel and select **Media Gateways**. Select the appropriate media gateway (not shown), and scroll down to the area of the screen containing **VGW and IP phone codec profile** as shown below.



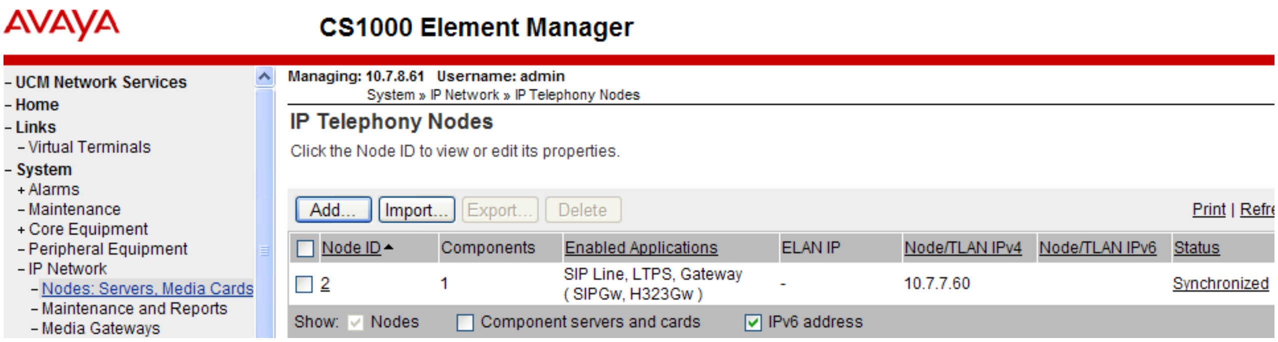
Expand **VGW and IP phone codec profile**. To use G.729A, ensure that the **Select** box is checked for **Codec G729A**, and the **VAD** (Voice Activity Detection) box is un-checked.

Note that **Codec G.711** is enabled by default. **Voice payload size** "20" can be used with Verizon for both G.729A and G.711. The following screen shows the parameters used.

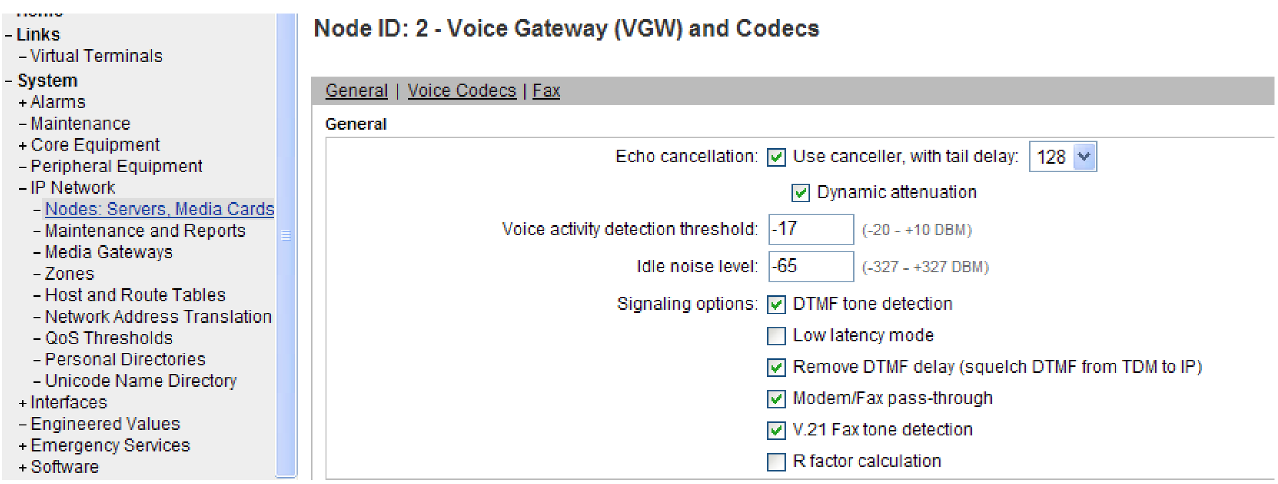


5.6.2 Node Voice Gateway and Codec Configuration

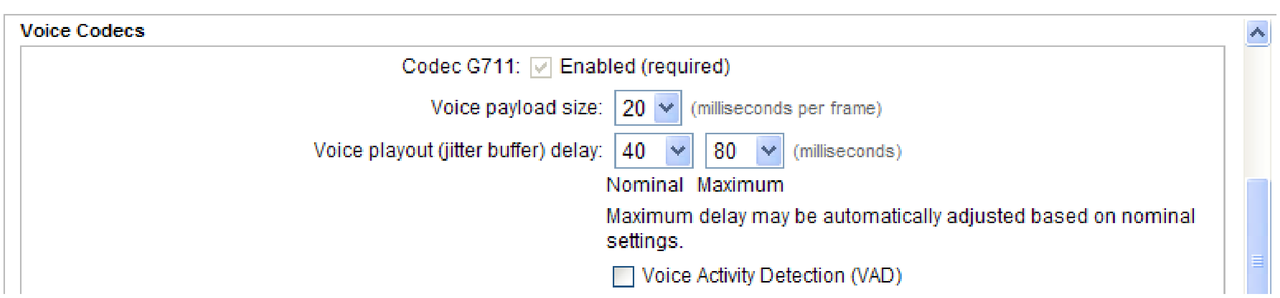
Expand **System → IP Network** and select **Node, Server, Media Cards**. Select the appropriate **Node Id** "2" as shown below.



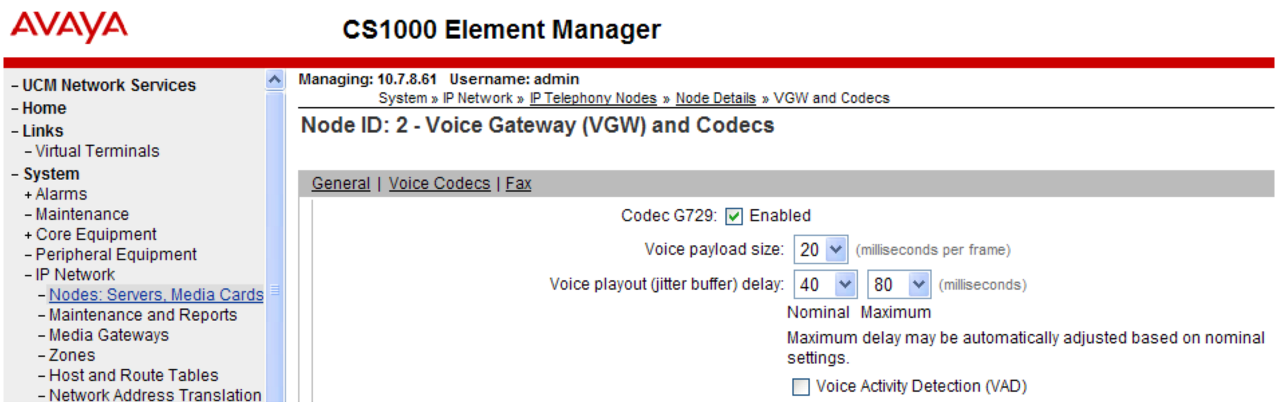
In the resultant screen (not shown) use the scroll bar on the right to select **Voice Gateway (VGW) and Codecs**. The following screen shows the **General** parameters used in the sample configuration.



Use the scroll bar on the right to find the area with heading **Voice Codecs**. Note that **Codec G.711** is enabled by default. The following screen shows the G.711 parameters used in the sample configuration.



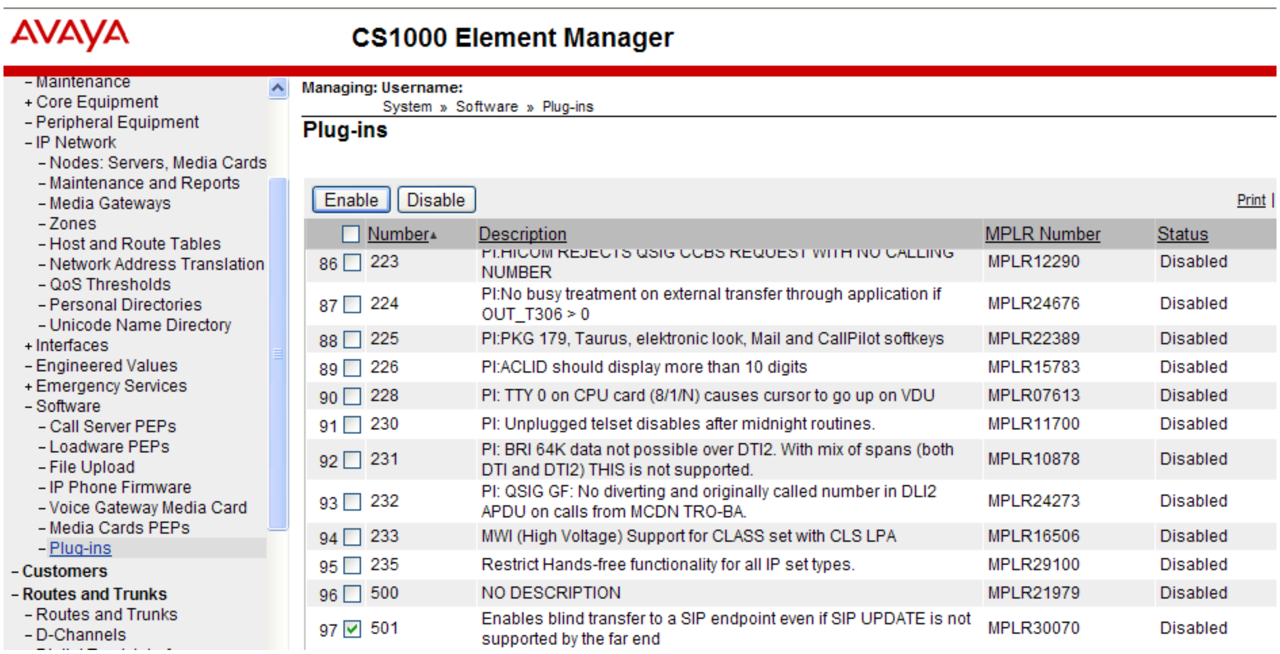
To allow the use of G.729, ensure that the **Enabled** box is checked for the **Codec G.729**, and the **Voice Activity Detection (VAD)** box is un-checked, as shown below.



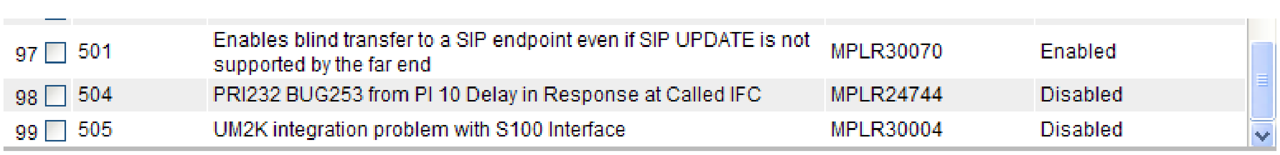
5.7. Enabling Plug-Ins for Call Transfer Scenarios

The procedures in this section are optional for Verizon IPCC deployments, but are included for completeness, since the same Avaya CS1000E configuration was used in testing both Verizon IP Trunk Service and Verizon IPCC Service. Plug-ins allow specific CS1000E software feature behaviors to be changed. In the testing associated with these Application Notes, two plug-ins were enabled as shown in this section.

To view or enable a plug-in, from the left navigation menu, expand **System → Software**, and select **Plug-ins**. In the right side screen, a list of available plug-ins will be displayed along with the associated MPLR Number and Status. Use the scroll bar on the right to scroll down so that Plug-in 501 is displayed as shown in the screen below. If the **Status** is "Disabled", select the check-box next to Number 501 and click the **Enable** button at the top, if it is desirable to allow CS1000E users to complete call transfer to PSTN destinations via the Verizon IP Trunk service before the call has been answered by the PSTN user. Note that enabling plug-in 501 will allow the user to complete the transfer while the call is in a ringing state, but no audible ring back tone will be heard after the transfer is completed.

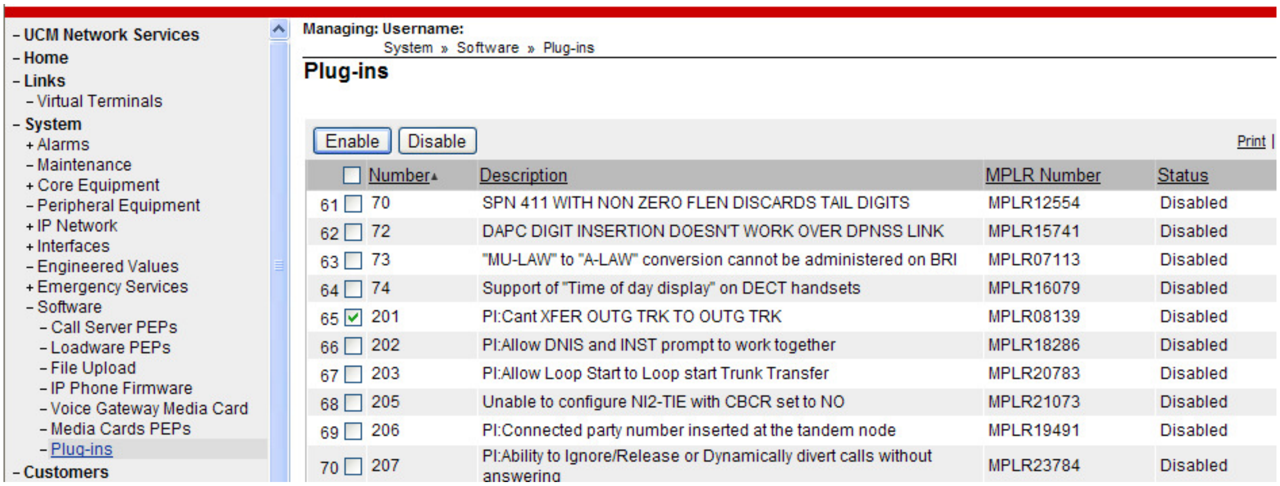


The following screen shows the relevant portion of this same screen after plug-in 501 has been enabled.



The same procedure may be used to enable plug-in 201 if desired. Plug-in 201 will allow a CS1000E user to make a call to the PSTN using the Verizon IP Trunk service, and then subsequently perform an attended transfer of the call to another PSTN destination via the Verizon IP Trunk service.

Expand **System → Software**, and select **Plug-ins**. Use the scroll bar to scroll down so that Plug-in 201 is displayed as shown in the screen below. If the **Status** is "Disabled", and it is desirable to allow attended transfer of an outbound trunk call to another outbound trunk, select the check-box next to Number 201 and click the **Enable** button at the top.

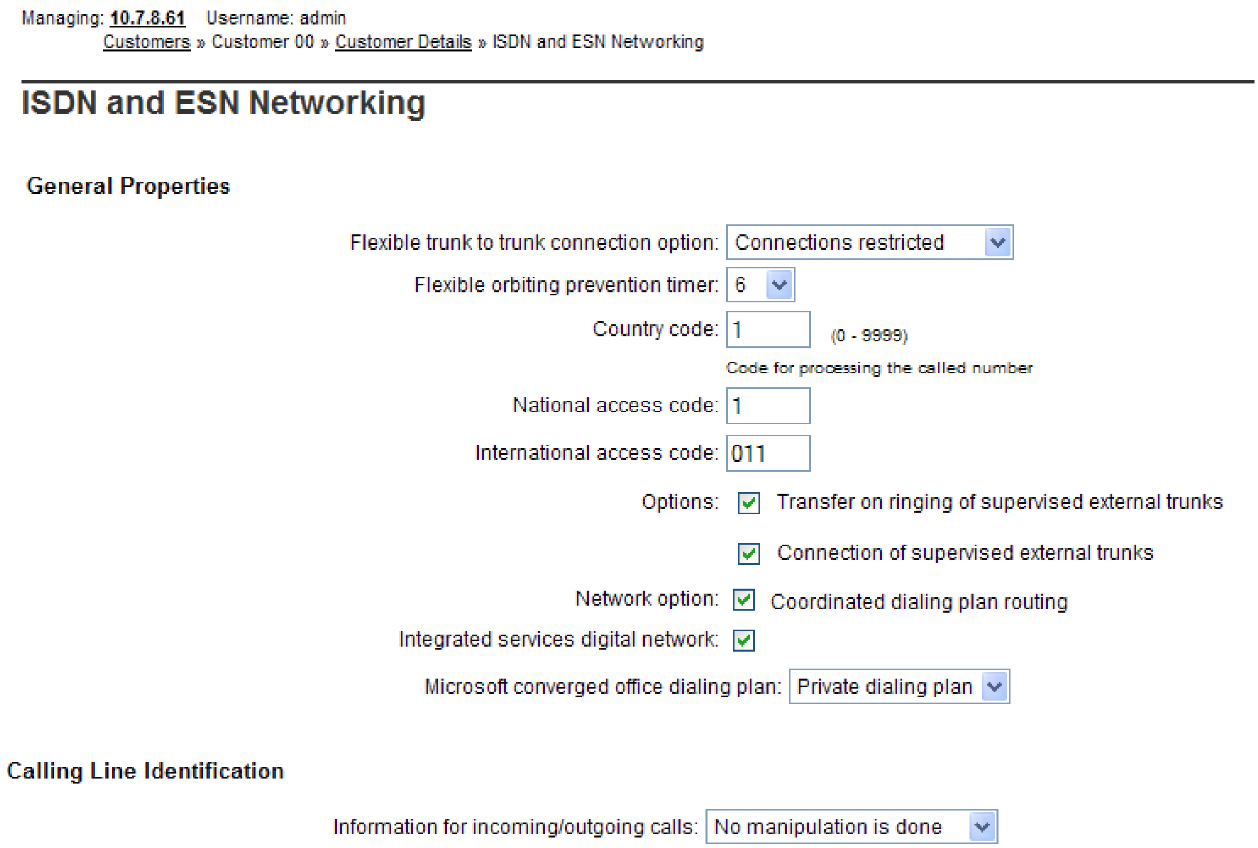


The following screen shows the relevant portion of this same screen after plug-in 201 has been enabled.

63	73	"MU-LAW" to "A-LAW" conversion cannot be administered on BRI	MPLR07113	Disabled
64	74	Support of "Time of day display" on DECT handsets	MPLR16079	Disabled
65	201	PI:Cant XFER OUTG TRK TO OUTG TRK	MPLR08139	Enabled

5.8. Customer Information

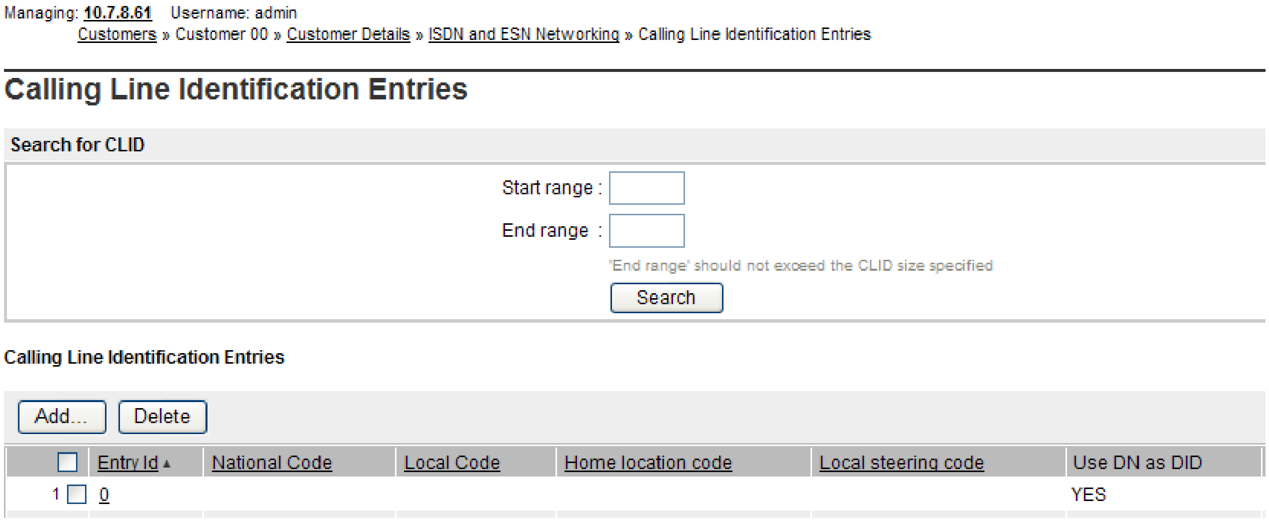
This section documents basic configuration relevant to the sample configuration. This section is not intended to be prescriptive. Select **Customers** from the left navigation menu, click on the appropriate **Customer Number** and select **ISDN and ESN Networking** (not shown). The following screen shows the **General Properties** used in the sample configuration.



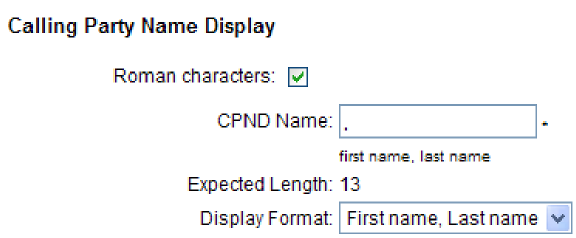
5.8.1 Caller ID Related Configuration

Although not intended to be prescriptive, in the sample configuration, the CS1000E would send the user's five-digit directory number in SIP headers such as the From and PAI headers. Avaya Aura® Session Manager would adapt the user's directory number to an appropriate number before passing the message to the Avaya Aura® SBC towards Verizon.

Scroll down from the screen shown in Section 5.8, click the **Calling Line Identification Entries** link (now shown), and search for the **Calling Line Identification Entries** by **Entry ID.** As shown below, the **Use DN as DID parameter** was set to "YES" for the **Entry ID** "0" used in the sample configuration.



Click on **Entry Id** 0 to view or change further details. The following shows the **Calling Party Name Display** configuration used in the sample configuration.

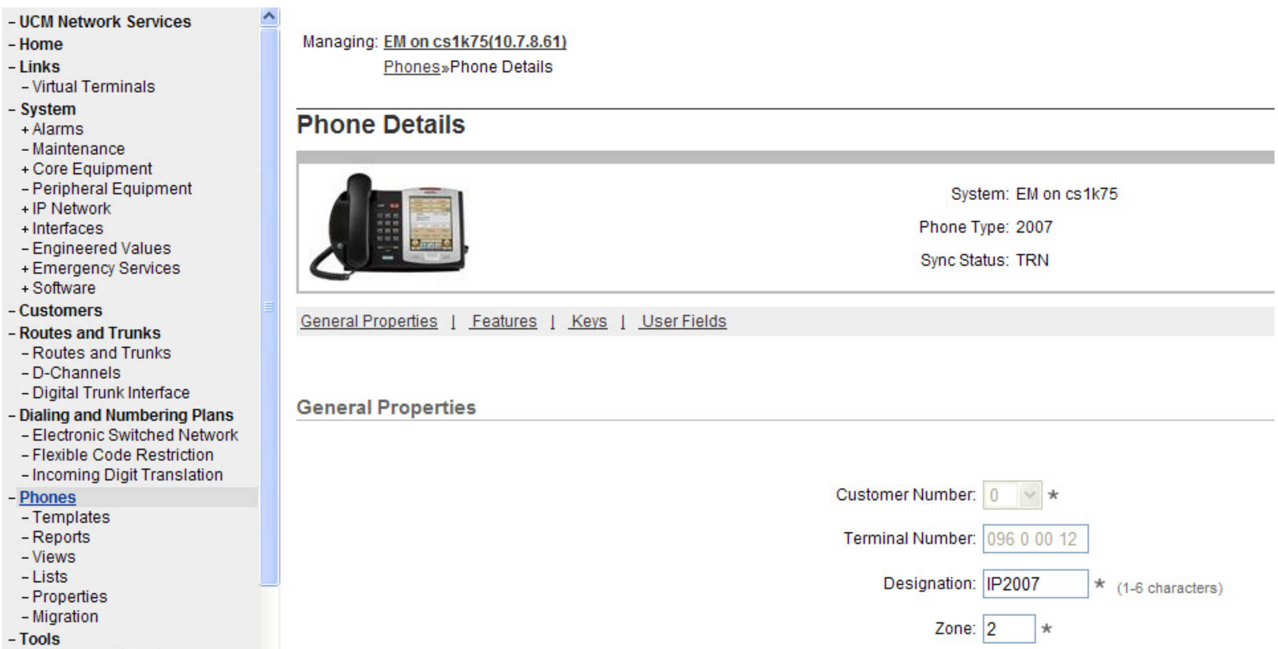


5.9. Example CS1000 Telephone Users

This section is not intended to be prescriptive, but simply illustrates a sampling of the telephone users in the sample configuration. These telephone directory numbers can be observed in the Session Manager configuration, since Session Manager is used to adapt the Verizon IP toll free numbers to Avaya CS1000E user telephone numbers.

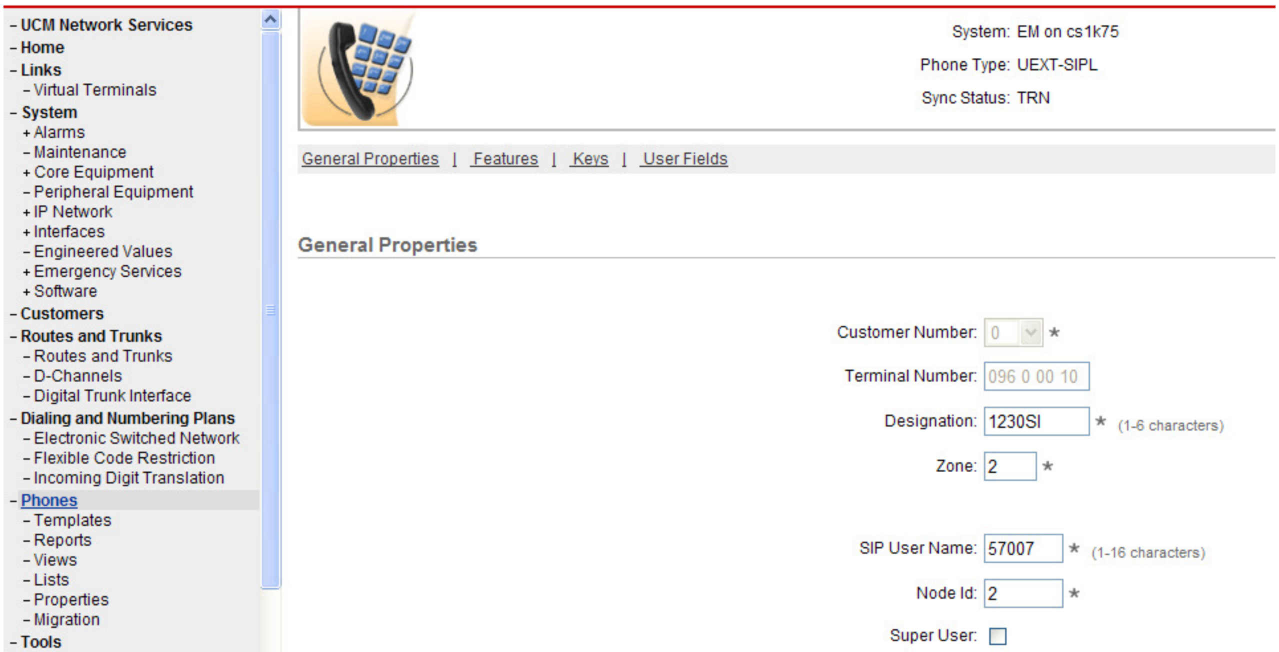
5.9.1 Example IP UNIStim Phone DN 57003, Codec Considerations

The following screen shows basic information for an IP UNIStim phone in the configuration. The telephone is configured as Directory Number 57003. Note that the telephone is in Zone 2. A call between this telephone and another telephone in Zone 2 will use a "best quality" strategy (see Section 5.5) and therefore can use G.711MU. If this same telephone calls out to the PSTN via the Verizon IP Trunk service or receives an inbound Verizon IP Toll Free call, the call would use a "best bandwidth" strategy, and the call would use G.729A.



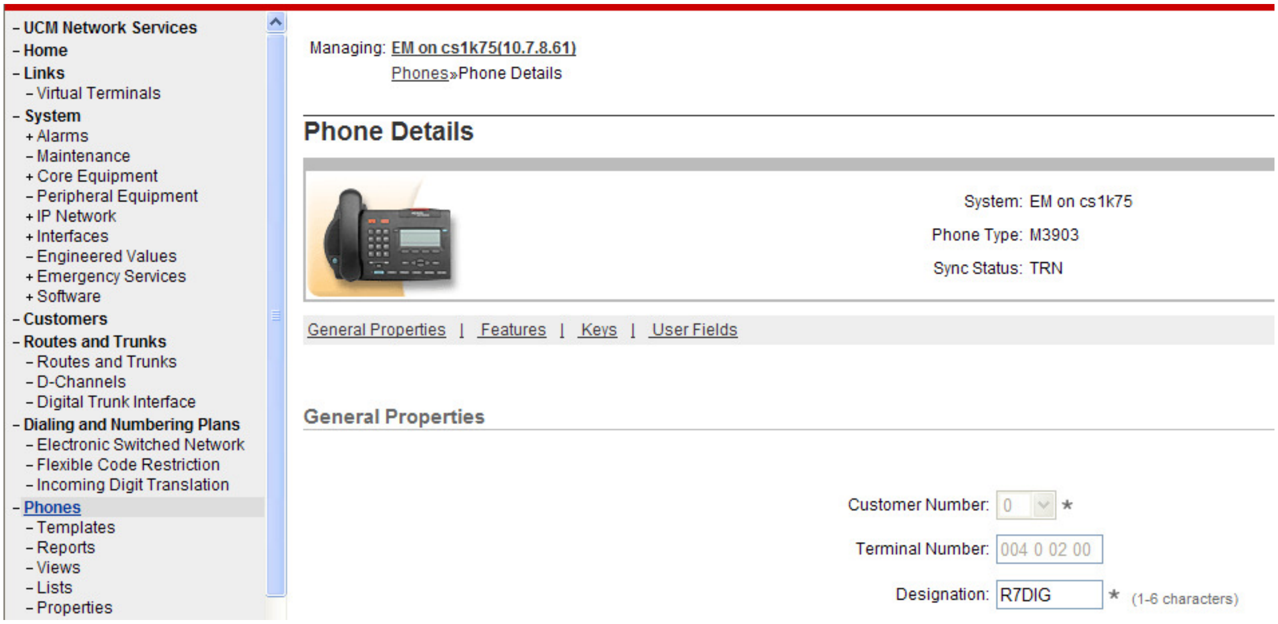
5.9.2 Example SIP Phone DN 57007, Codec Considerations

The following screen shows basic information for a SIP phone in the configuration. The telephone is configured as Directory Number 57007. Note that the telephone is in Zone 2 and is associated with Node 2 (see Section 5.1). A call between this telephone and another telephone in Zone 2 will use a "best quality" strategy (see Section 5.5) and therefore can use G.711MU. If this same telephone calls out to the PSTN via the Verizon IP Trunk service, the call would use a "best bandwidth" strategy, and the call would use G.729A. Similarly, if the user receives a call from Verizon IP Toll Free service, the call will use G.729A.

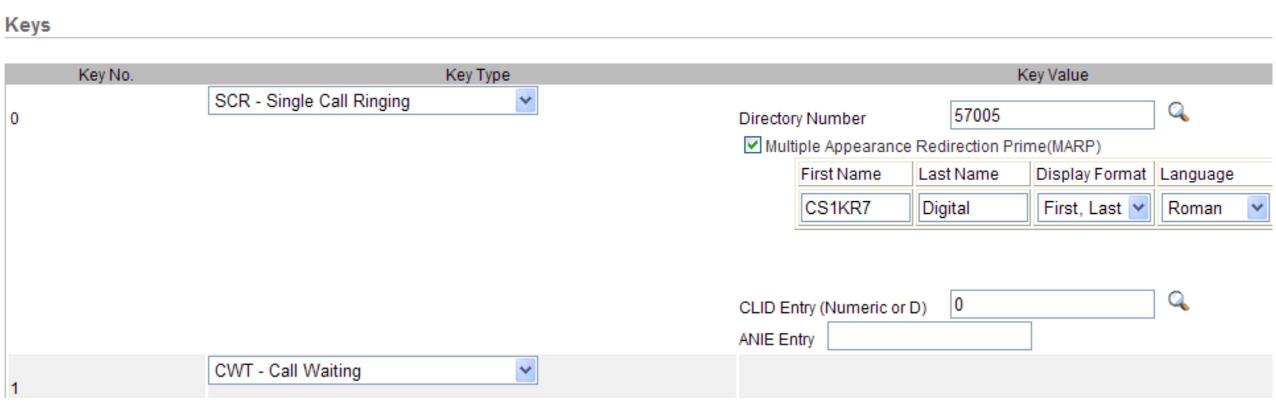


5.9.3 Example Digital Phone DN 57005 with Call Waiting

The following screen shows basic information for a digital phone in the configuration. The telephone is configured as Directory Number 57005.

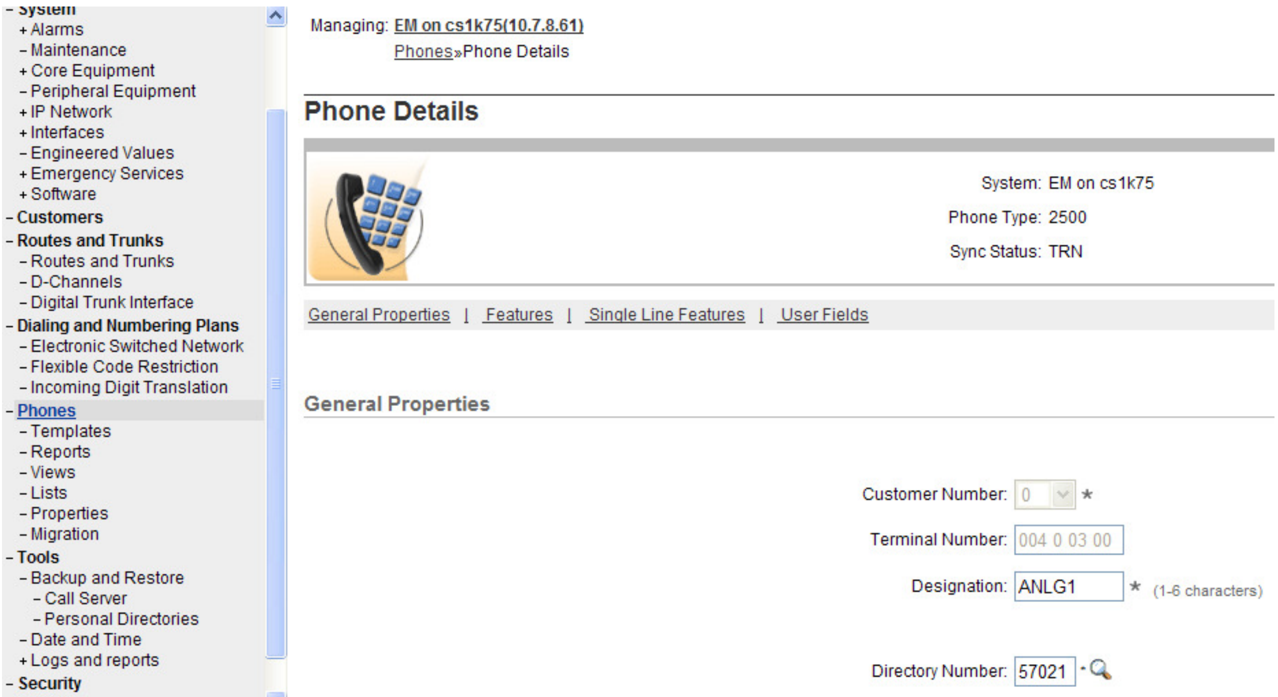


The following screen shows basic key information for the telephone. It can be observed that the telephone can support call waiting with tone, and uses CLID Entry 0 (see Section 5.8). Although not shown in detail below, to use call waiting with tone, assign a key "CWT – Call Waiting", set the feature "SWA – Call waiting from a Station" to "Allowed", and set the feature "WTA – Warning Tone" to "Allowed".



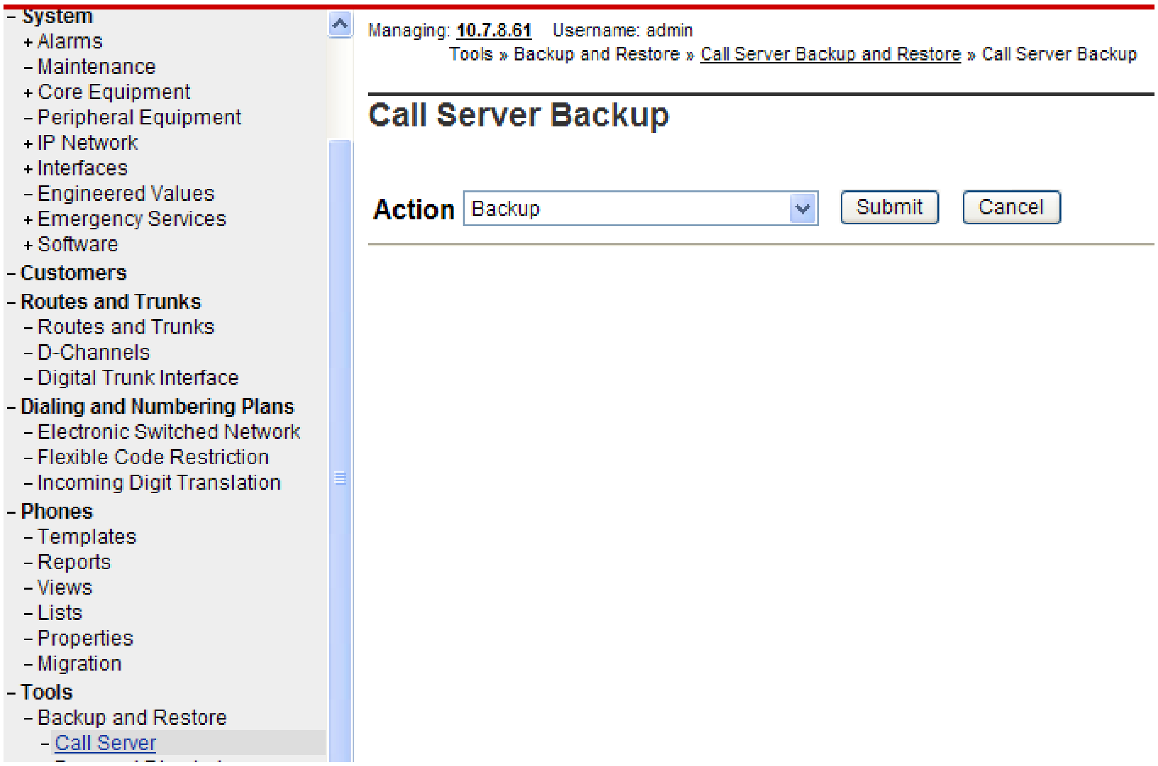
5.9.4 Example Analog Port with DN 57021

The following screen shows basic information for an analog port in the configuration that may be used with a basic analog telephone. The port is configured as Directory Number 57021.

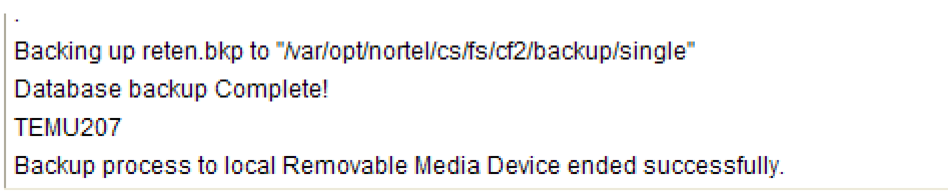


5.10. Save Configuration

Expand **Tools → Backup and Restore** on the left navigation panel and select **Call Server.** Select **Backup** (not shown) and click **Submit** to save configuration changes as shown below.



The backup process may take several minutes to complete. Scroll to the bottom of the page to verify the backup process completed successfully as shown below.



The configuration of Avaya Communication Server 1000E is complete.

6. Configure Avaya Aura® Session Manager Release 6.1

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

Note – The following sections assume that Session Manager and System Manager have been installed and that network connectivity exists between the two. For more information, consult the references in Section 11.

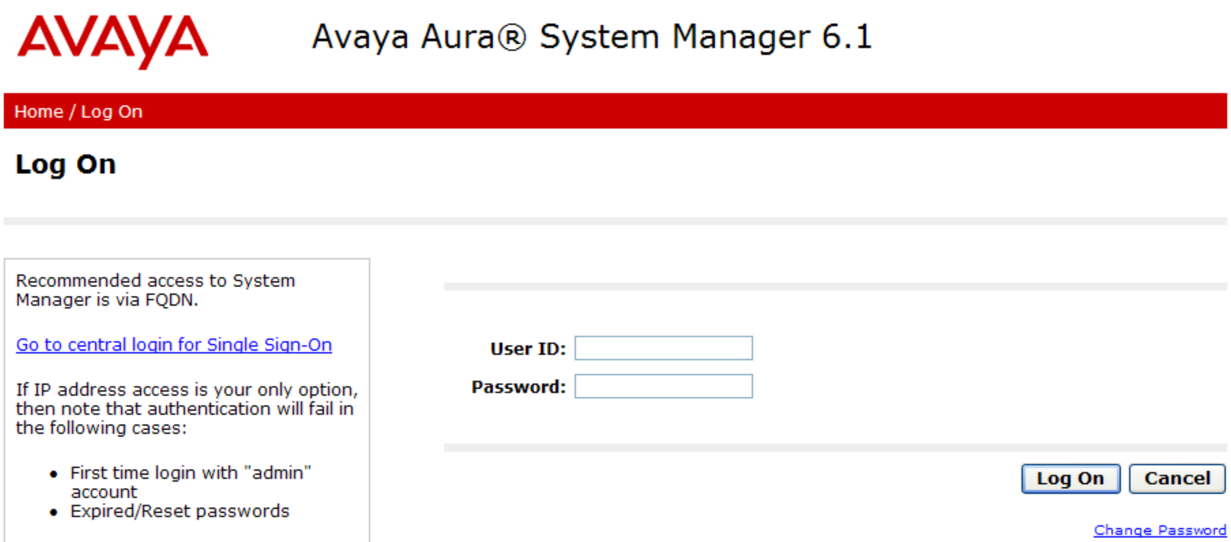
As described in Section 3, the same Session Manager configuration was used for testing both Verizon IP Trunk Service and Verizon IPCC Service. Although not the focus of these Application Notes, Verizon IP Trunk Service supports outbound dialing to the PSTN, and this section includes the procedures to allow outbound routing, for completeness. Consult Reference [AuraSBC-IP-Trunk] for more information on connecting to Verizon IP Trunk Service.

The following administration activities will be described:

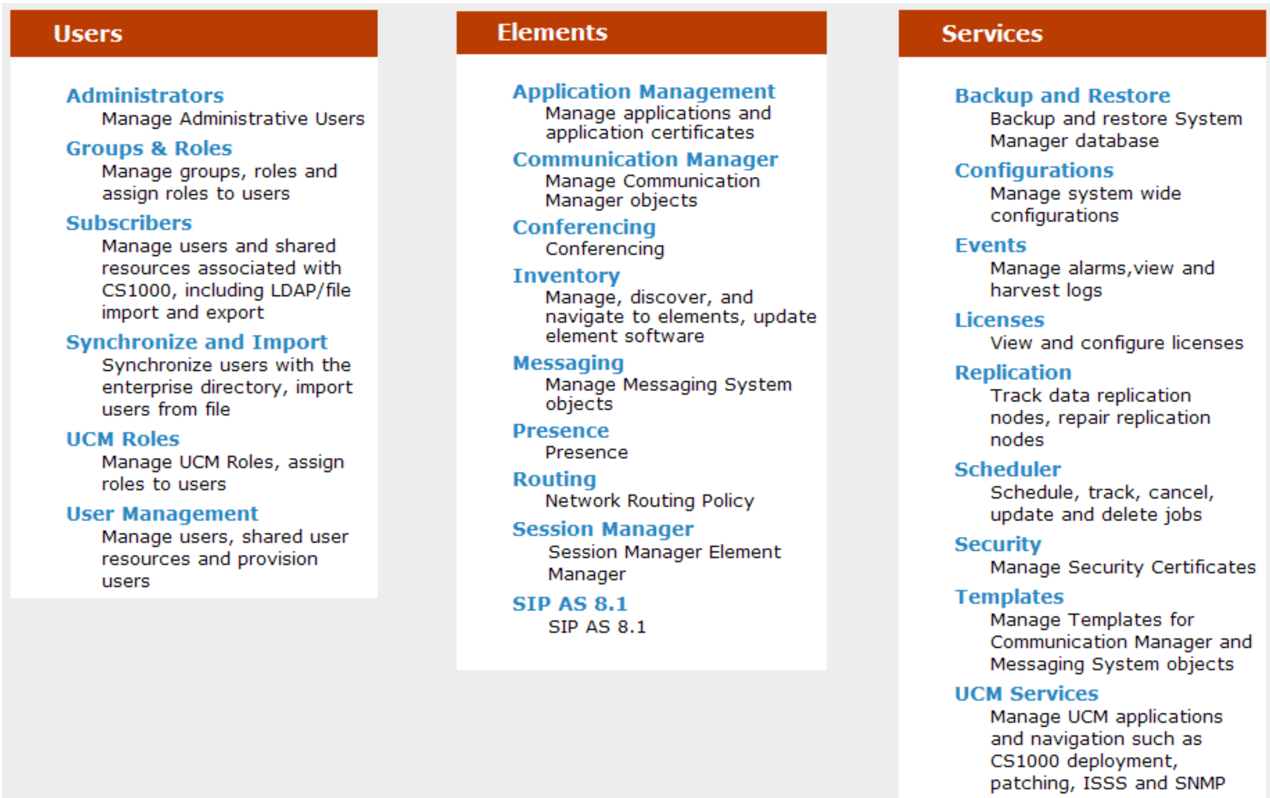
- Define SIP Domain
- Define Locations for Avaya Communication Server 1000E and for the SBC
- Configure the Adaptation Modules that will be associated with the SIP Entities for Avaya Communication Server 1000E and the SBC
- Define SIP Entities corresponding to Avaya Communication Server 1000E and the SBC
- Define Entity Links describing the SIP trunk between Avaya Communication Server 1000E and Session Manager, and the SIP Trunk between Session Manager and the SBC.
- Define Routing Policies associated with the Avaya Communication Server 1000E and the SBC.
- Define Dial Patterns, which govern which routing policy will be selected for call routing.

Configuration is accomplished by accessing the browser-based GUI of Avaya Aura® System Manager, using the URL "**http://<ip-address>/SMGR**", where **<ip-address>** is the IP address of Avaya Aura® System Manager. Log in with the appropriate credentials.

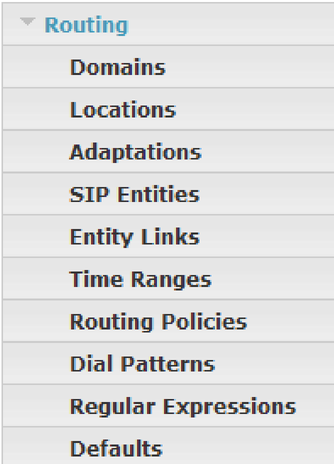
In the **Log On** screen, enter appropriate **User ID** and **Password** and press the **Log On** button (not shown).



Once logged in, a Release 6.1 **Home** screen like the following is displayed. From the **Home** screen below, under the **Elements** heading in the center, select **Routing**.



The screen shown below shows the various sub-headings of the left navigation menu that will be referenced in this section.

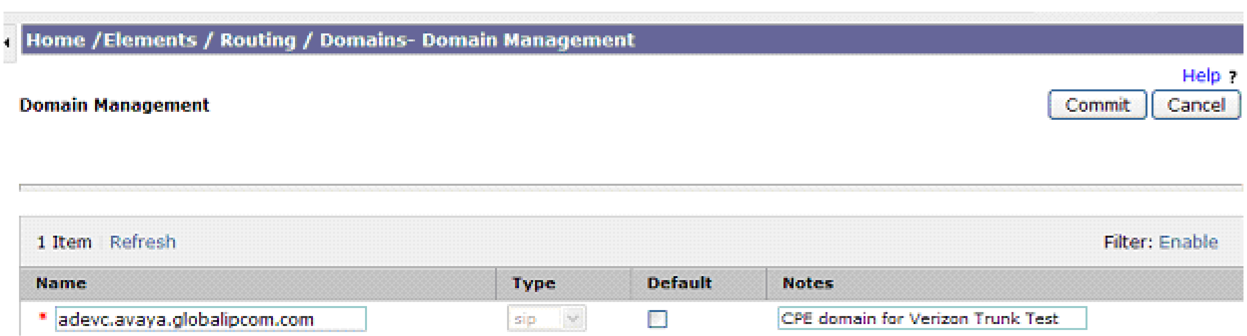


6.1. SIP Domain

Select **Domains** from the left navigation menu. Two domains can be added, one for the enterprise SIP domain, and one for the Verizon network SIP domain, if needed. In the shared environment of the Avaya Solution and Interoperability Test lab, a domain "avaya.com" is also defined and used by the shared Avaya equipment.

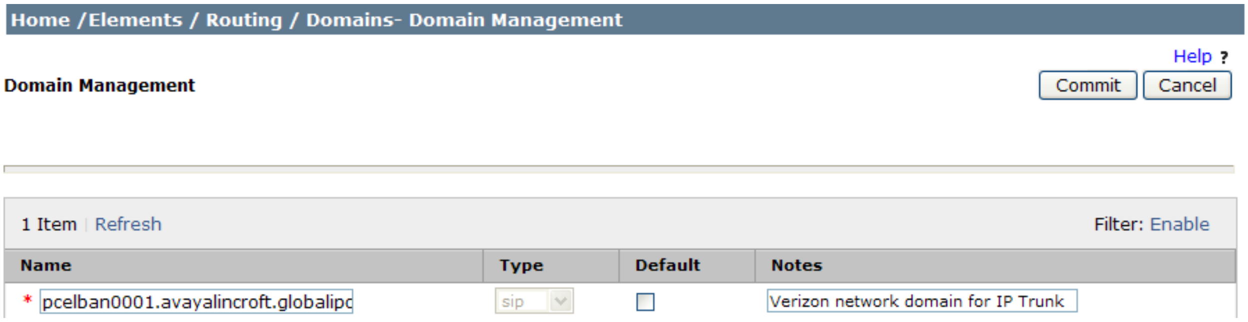
Click **New** (not shown)**.** Enter the following values and use default values for remaining fields**.**

- **Name** Enter the enterprise SIP Domain Name. In the sample screen below, "**adevc.avaya.globalipcom.com**" is shown, the CPE domain known to Verizon.
- **Type** Verify "**SIP**" is selected.
- **Notes** Add a brief description. [Optional]



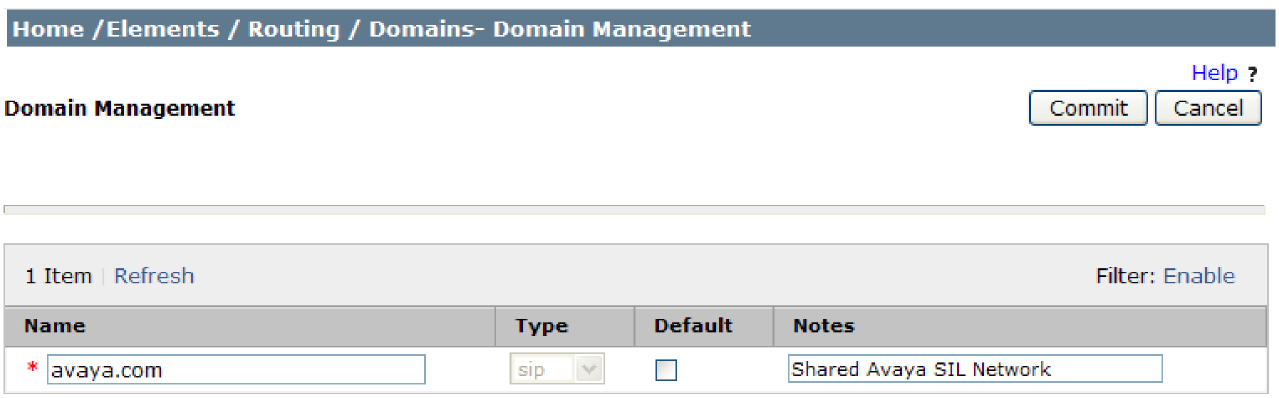
Click **New** (not shown)**.** Enter the following values and use default values for remaining fields. The domain shown below is associated with the Verizon IP Trunk Service available from the same PIP access circuit. This domain can be omitted if configuring only the Verizon IPCC Service.

- **Name** Enter the Domain Name used for the Verizon network. In the sample screen below, "**pcelban0001.avayalincroft.globalipcom.com**" is shown.
- **Type** Verify "**SIP**" is selected.
- **Notes** Add a brief description. [Optional]

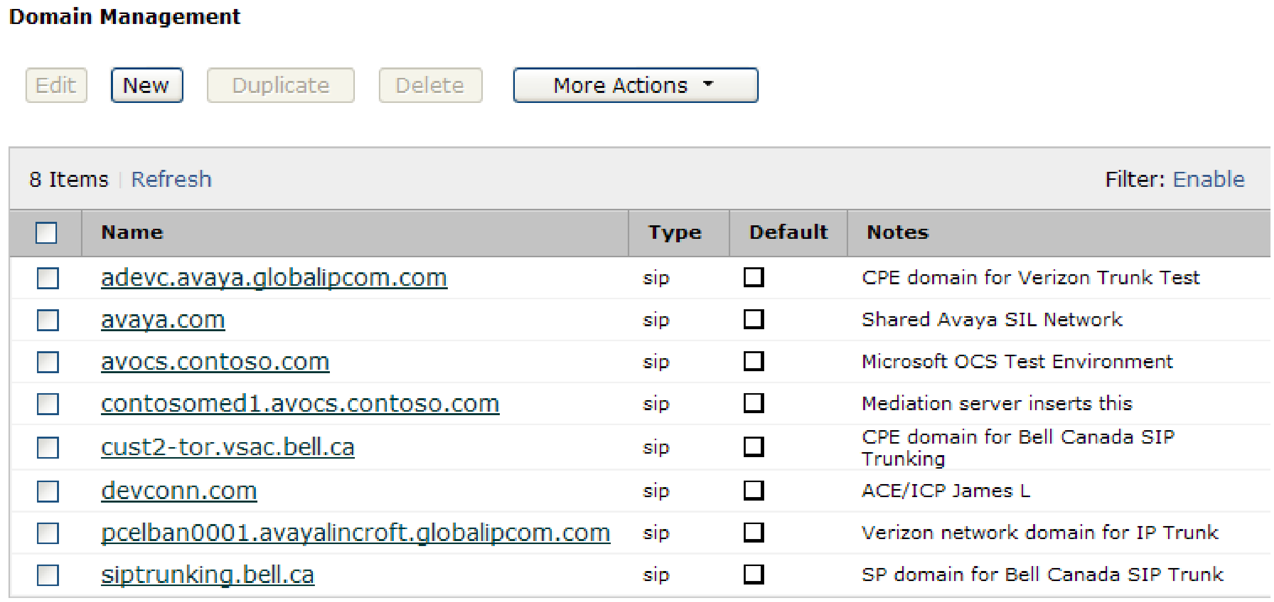


Click **Commit** to save.

The following screen shows the "avaya.com" SIP domain that was already configured in the shared laboratory network.



The screen below shows an example SIP Domain list after SIP Domains are configured. Many SIP Domains can be configured, distinguished, and adapted by the same Session Manager as needed.



Domain Management

Edit | New | Duplicate | Delete | More Actions

8 Items | Refresh Filter: Enable

	Name	Type	Default	Notes
☐	adevc.avaya.globalipcom.com	sip	☐	CPE domain for Verizon Trunk Test
☐	avaya.com	sip	☐	Shared Avaya SIL Network
☐	avocs.contoso.com	sip	☐	Microsoft OCS Test Environment
☐	contosomed1.avocs.contoso.com	sip	☐	Mediation server inserts this
☐	cust2-tor.vsac.bell.ca	sip	☐	CPE domain for Bell Canada SIP Trunking
☐	devconn.com	sip	☐	ACE/ICP James L
☐	pcelban0001.avayalincroft.globalipcom.com	sip	☐	Verizon network domain for IP Trunk
☐	siptrunking.bell.ca	sip	☐	SP domain for Bell Canada SIP Trunk

6.2. Locations

Locations are used to identify logical and/or physical locations where SIP Entities reside. Location identifiers can be used for bandwidth management or location-based routing.

6.2.1 Location for Avaya Communication Server 1000E

Select **Locations** from the left navigational menu. Click **New** (not shown)**.** In the **General** section**,** enter the following values and use default values for remaining fields**.**

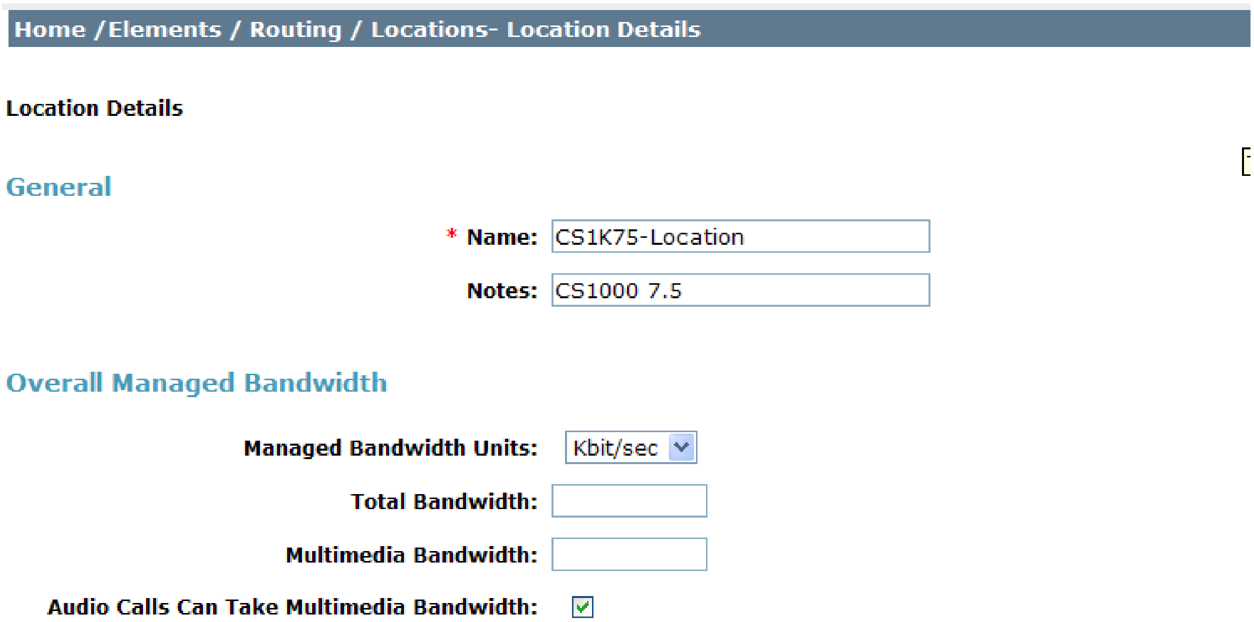
- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description. [Optional]

In the **Location Pattern** section, click **Add** and enter the following values.

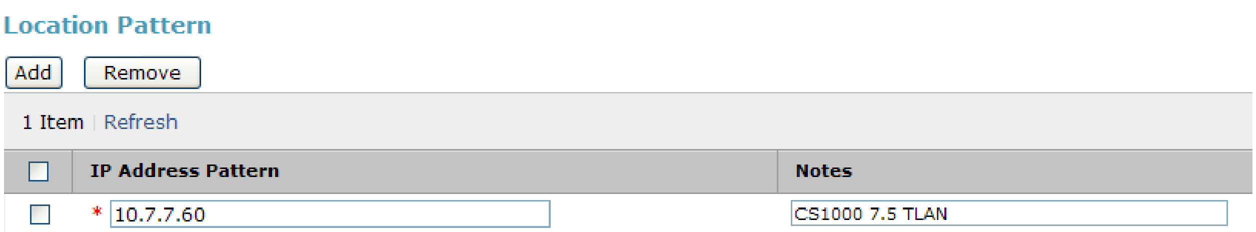
- **IP Address Pattern** Enter the IP Address or IP Address pattern used to identify the location.
- **Notes** Add a brief description. [Optional]

Click **Commit** to save.

The screen below shows the top portion of the screen for the Location defined for Avaya Communication Server 1000E.



Scrolling down, the following screen shows the lower portion of the Location for the CS1000E.



6.2.2 Location for Session Border Controller

Select **Locations** from the left navigational menu. Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

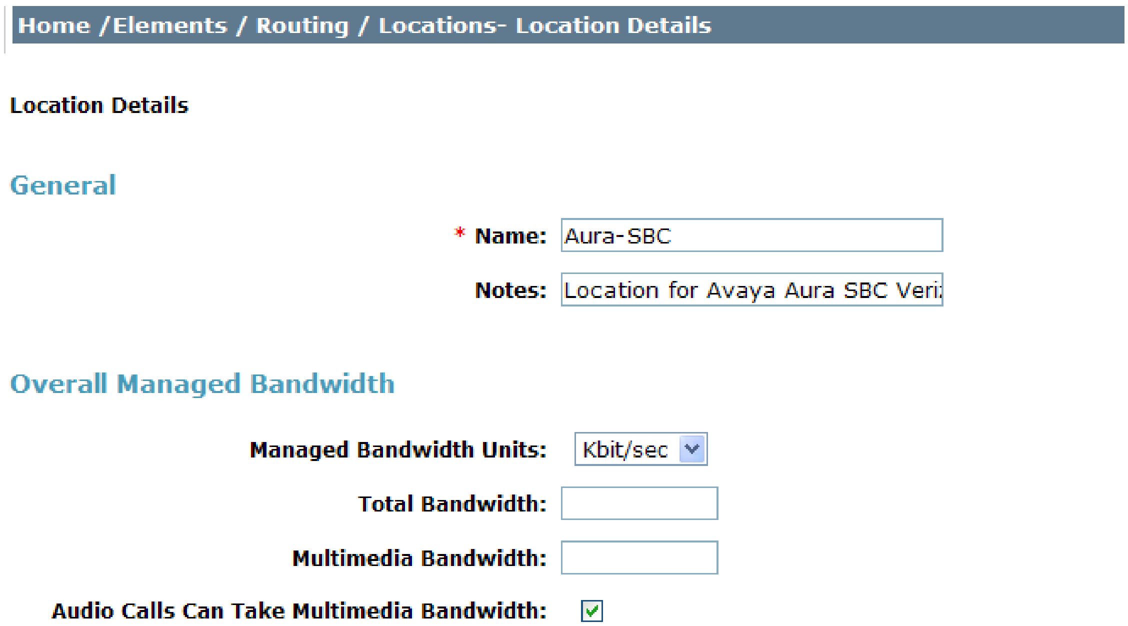
- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description. [Optional]

In the **Location Pattern** section, click **Add** and enter the following values.

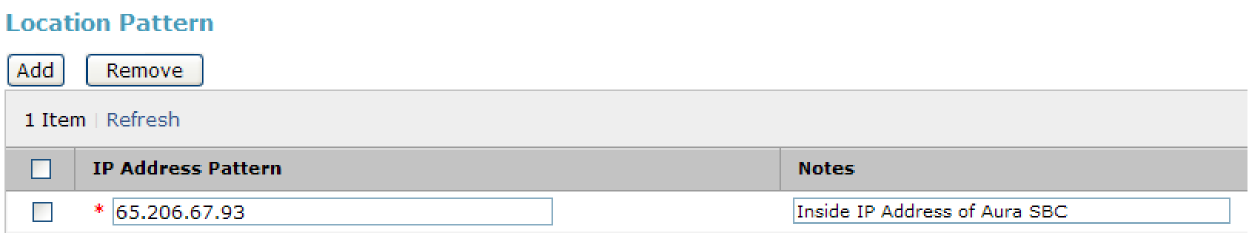
- **IP Address Pattern** Enter the IP Address or IP Address pattern used to identify the location.
- **Notes** Add a brief description. [Optional]

Click **Commit** to save.

The screen below shows the top portion of the screen for the Location defined for the SBC.



Scrolling down, the following screen shows the lower portion of the Location for the SBC.



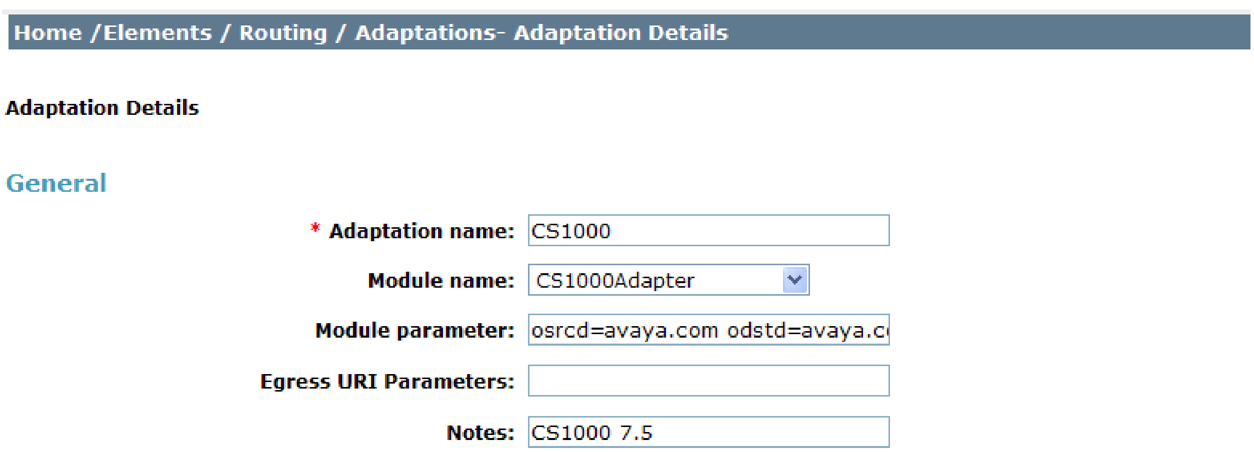
6.3. Configure Adaptations

Session Manager can be configured to use an Adaptation Module designed for Avaya Communication Server 1000E to convert SIP headers in messages sent by Avaya Communication Server to the format used by other Avaya products and endpoints.

6.3.1 Adaptation for Avaya Communication Server 1000E Entity

Select **Adaptations** from the left navigational menu. Click **New** (not shown**).** In the **General** section, enter the following values and use default values for remaining fields**.**

- **Adaptation Name:** Enter an identifier for the Adaptation Module (e.g., "CS1000")
- **Module Name:** Select "**CS1000Adapter**" from drop-down menu (or add an adapter with name "CS1000Adapter" if not previously defined)
- **Module Parameter:** Enter "osrcd=<cs1000-domain>.com" and "odstd=<cs1000-domain>.com" where <cs1000-domain> is the SIP domain configured in the CS1000E system. Enter "fromto=true" to allow the From and To headers to be modified by Session Manager (i.e., in addition to other headers such as the P-Asserted-Identity and Request-URI headers).



Scroll down to the **Digit Conversion for Incoming Calls to SM** section. Click **Add** to configure entries for calls from CS1000E users to Verizon, a necessary step for outbound calls to the Verizon IP Trunk service, but an optional step if only inbound calls from the Verizon IPCC services will be configured. If only Verizon IPCC service is used, this area of the screen need not be configured, but may optionally be used to map the CS1000 DN to a Verizon IP Toll Free number in the PAI header sent to Verizon in 180 Ringing and 200 OK for an inbound toll-free call.

- **Matching Pattern** Enter Avaya CS1000E extensions (or extension ranges via wildcard pattern matching).
- **Min** Enter minimum number of digits (e.g., 5)
- **Max** Enter maximum number of digits (e.g., 5)
- **Phone Context** Enter value of **Private CDP domain name** defined in the CS1000E for any patterns matching SIP endpoints registered to Session Manager (if any).
- **Delete Digits** Enter "**0**", unless digits should be removed before routing by Session Manager. For CS1000E extension conversion to the corresponding Verizon number, enter the number of digits in the extension to remove all digits.
- **Insert Digits** Enter the Verizon number corresponding to the matched extension. The numbers shown below are Verizon IP Trunk DID numbers. For inbound IPCC calls, it is not imperative that the PAI in responses contain a Verizon IPCC number. If only IPCC service is used, an IP Toll Free number or IP-IVR number may be used.
- **Address to modify** Select **"both"**

Notes: CS1000 7.5

Digit Conversion for Incoming Calls to SM

Add Remove

5 Items | Refresh Filter: Ena

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes
	* 3	* 5	* 5	cdp.udp	* 0		both	3xxxx on CM/SM
	* 57003	* 5	* 5		* 5	7329450235	both	CS1K IP-Unistim to Verizon I
	* 57005	* 5	* 5		* 5	7329450231	both	CS1K Digital to Verizon DID
	* 57007	* 5	* 5		* 5	7329450236	both	CS1K SIP phone to Verizon I
	* 57021	* 5	* 5		* 5	7329450288	both	CS1K Analog port (fax)

Scroll down to the **Digit Conversion for Outgoing Calls from SM** section, corresponding to inbound Verizon toll-free calls to CS1000E. In the **Matching Pattern**, enter a Verizon toll-free number, with **Min** and **Max** set to 10, the length of the number to match. In the **Delete Digits** field, enter the number of digits to delete. In the sample configuration, the entire 10 digit toll-free number is deleted and replaced by the desired CS1000E Directory Number.

Digit Conversion for Outgoing Calls from SM

Add | Remove

11 Items | Refresh — Filter: Ena

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes
☐	* 7329450231	* 10	* 10		* 10	57005	both	Verizon DID to CS1K Digital
☐	* 7329450232	* 10	* 10		* 10	57001	both	Verizon DID to CS1K IP-Unis
☐	* 7329450235	* 10	* 10		* 10	57003	both	Verizon DID to CS1K IP-Unis
☐	* 7329450236	* 10	* 10		* 10	57007	both	Verizon DID to CS1K SIP ph
☐	* 7329450288	* 10	* 10		* 10	57021	both	Verizon DID to CS1K analog
☐	* 8668502380	* 10	* 10		* 10	57005	both	Verizon IPTF to CS1K Digital
☐	* 8668506850	* 10	* 10		* 10	57003	both	Verizon IPTF to CS1K IP-Unis
☐	* 8668508170	* 10	* 10		* 10	57007	both	Verizon IP-IVR to CS1K SIP
☐	* 8668511977	* 10	* 10		* 10	57003	both	Verizon IP-IVR to CS1K IP-U
☐	* 8668518119	* 10	* 10		* 10	57005	both	Verizon IP-IVR to CS1K Digit
☐	* 8668523221	* 10	* 10		* 10	57007	both	Verizon IPTF to CS1K SIP

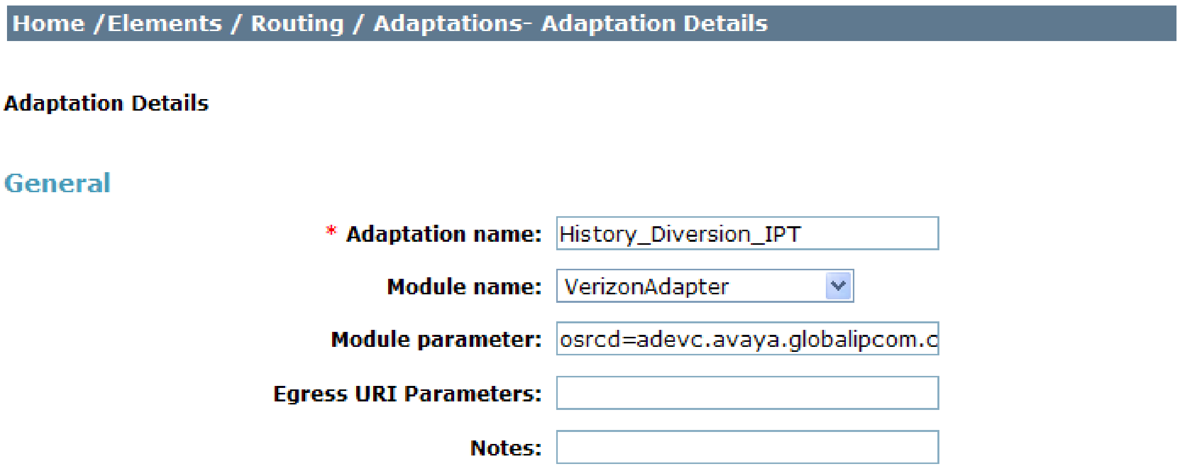
As an example, using these screens, if a PSTN user dials Verizon IP Toll Free number 866-850-6850, and the call is routed to the CS1000E, then this adapter will change the number sent to the CS1000E to directory number 57003 Other mappings of IP Toll Free and IP-IVR numbers from **Table 1** and **Table 2** in Section 3 can also be observed.

Click **Commit** (not shown)**.**

6.3.2 Adaptation for SBC Entity

Select **Adaptations** from the left navigational menu. Click **New** (not shown)**.** In the **General** section, enter the following values and use default values for remaining fields**.**

- **Adaptation Name:** Enter an identifier for the Adaptation Module
- **Module Name:** Select "**VerizonAdapter**" from drop-down menu (or add an adapter with name "VerizonAdapter" if not previously defined)
- **Module Parameter:** Enter "osrcd=<CPE-domain-known-to-Verizon>.com" and (optionally) "odstd=<Verizon-domain>.com". The <CPE-domain-known-to-Verizon > is the SIP domain for the CPE configured in the Verizon network (i.e., the SIP domain Verizon would include in the Request-URI for an inbound toll-free call). <Verizon-domain> is the Verizon network SIP domain (i.e., the SIP domain Verizon would expect in the Request-URI for an INVITE sent from the CPE to the PSTN for the Verizon IP Trunk Service, if used). Enter "fromto=true" to allow the From and To headers to be modified by Session Manager (i.e., in addition to other headers such as the P-Asserted-Identity and Request-URI headers).



Click **Commit** (not shown).

6.3.3 List of Adaptations

Select **Adaptations** from the left navigational menu. A partial list of the Adaptation Modules defined for the sample configuration is shown below. In list form, the module parameters assigned to the adapters named "CS1000" and "History_Diversion_IPT" are more evident than the screens presented in the prior sections.

☐	Name	Module name	Egress URI Parameters	Notes
☐	AcmeAdapt	DigitConversionAdapter odstd=138.210.71.242		Change RURI To Dest IP
☐	Avaya-R6.0	DigitConversionAdapter odstd=avaya.com osrcd=avaya.com		
☐	BC AA-SBC	DigitConversionAdapter osrcd=cust2-tor.vsac.bell.ca odstd=siptrunking.bell.ca		convert to BC's domains
☐	BC CM-ES	DigitConversionAdapter odstd=avaya.com		avaya.com for shared SIL ntwk
☐	BCM_Adapter	DigitConversionAdapter avaya.com		Delete prefix
☐	Cisco-UCM513	CiscoAdapter 192.45.130.105		
☐	Cisco-UCM6	CiscoAdapter avaya.com		
☐	Cisco-UCM7	CiscoAdapter avaya.com		
☐	CiscoUCME	CiscoAdapter iosrcd=avaya.com odstd=192.45.131.1		
☐	CM5-2-1 Adapt	DigitConversionAdapter osrcd=avaya.com		Tim For CLink Testing
☐	CM-ES Inbound	DigitConversionAdapter odstd=avaya.com osrcd=avaya.com		
☐	CM-ES-VZ Inbound	DigitConversionAdapter odstd=avaya.com		Avaya.com for shared SIL ntwk
☐	CS1000	CS1000Adapter osrcd=avaya.com odstd=avaya.com fromto=true		CS1000 7.5
☐	Digit_Conversion_VZ	DigitConversionAdapter osrcd=adevc.avaya.globalipcom.com odstd=pcelban0001.avayalincroft.globalipcom.com		Verizon DID to CM Extn map, param above should be on VZ-adapter
☐	History_Diversion_IPT	VerizonAdapter osrcd=adevc.avaya.globalipcom.com odstd=pcelban0001.avayalincroft.globalipcom.com fromto=true		

6.4. SIP Entities

SIP Entities must be added for the Avaya Communication Server 1000E and for the SBC.

6.4.1 SIP Entity for Avaya Communication Server 1000E

Select **SIP Entities** from the left navigation menu.

Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

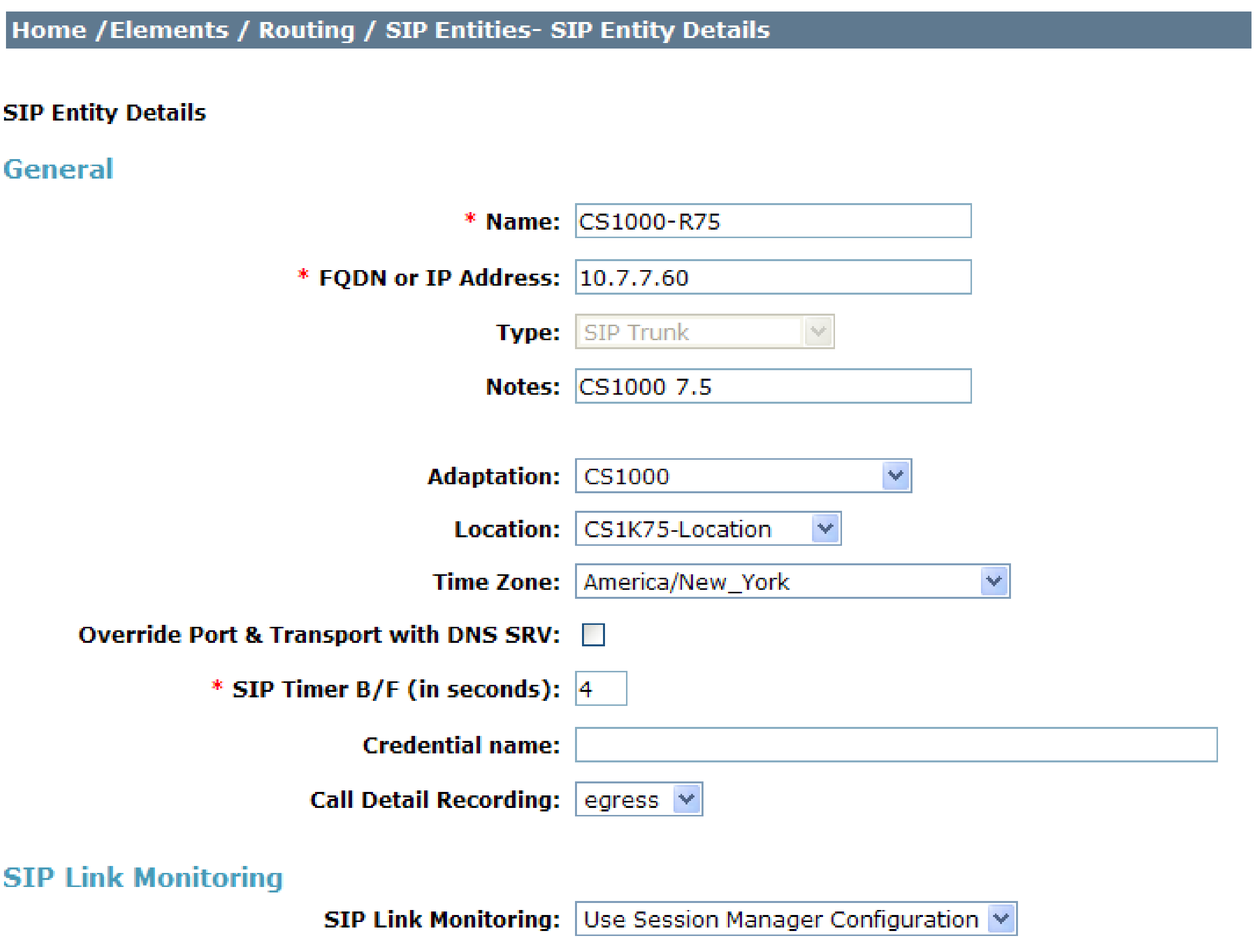
- **Name:** Enter an identifier for the SIP Entity
- **FQDN or IP Address:** Enter the TLAN IP address of the CS1000E Node.
- **Type:** Select "**SIP Trunk**"
- **Notes:** Enter a brief description. [Optional]
- **Adaptation:** Select the Adaptation Module for the CS1000E
- **Location:** Select the Location for the CS1000E

In the **SIP Link Monitoring** section:

- **SIP Link Monitoring:** Select "**Use Session Manager Configuration**" (or choose an alternate Link Monitoring approach for this entity, if desired).

Click **Commit** (not shown) to save the definition of the new SIP Entity.

The following screen shows the SIP Entity defined for Avaya Communication Server 1000E in the sample configuration.



6.4.2 SIP Entity for SBC

Select **SIP Entities** from the left navigation menu.

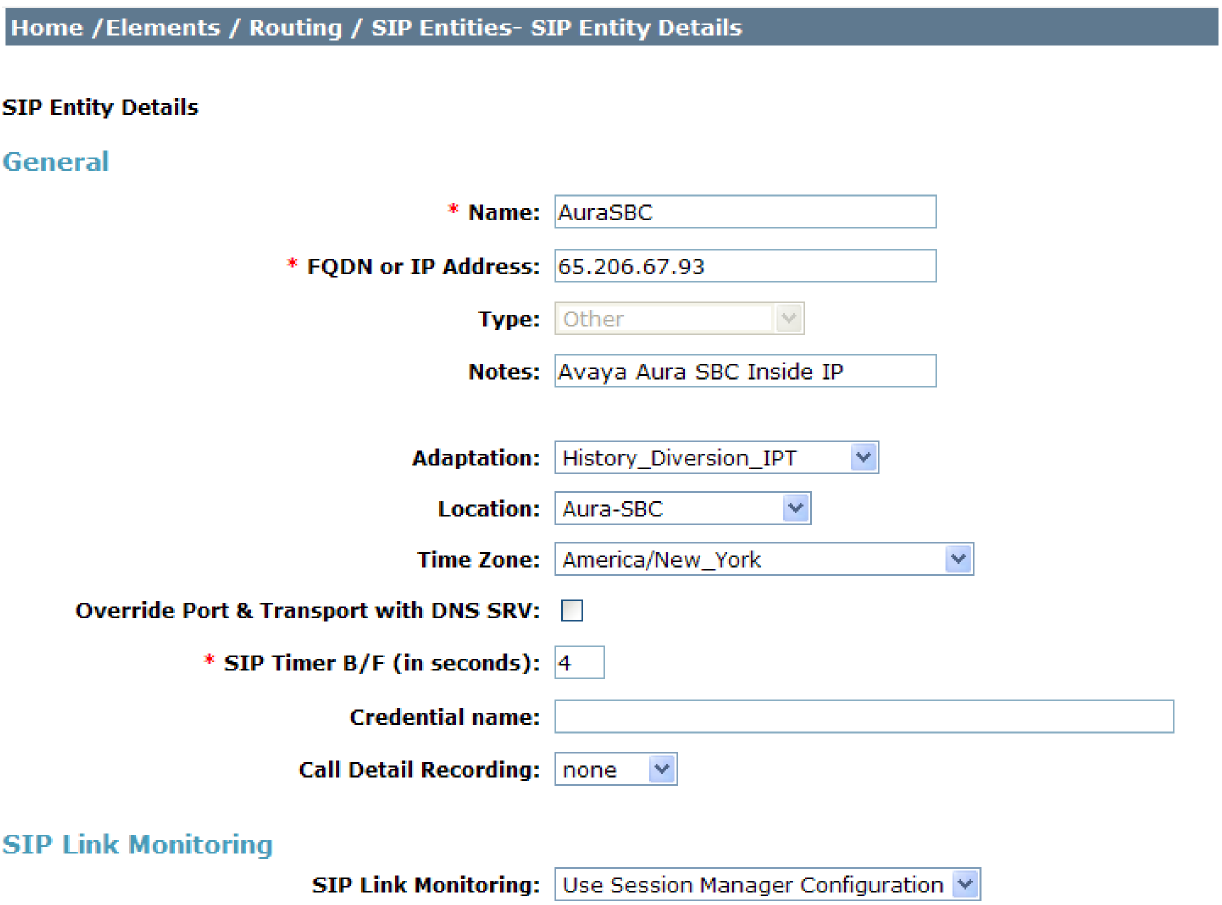
Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **Name:** Enter an identifier for the SIP Entity
- **FQDN or IP Address:** Enter the private side IP Address of the SBC.
- **Type:** Select "**Other**"
- **Notes:** Enter a brief description. [Optional]
- **Adaptation:** Select the Adaptation Module for the SBC
- **Location:** Select the Location for the SBC

In the **SIP Link Monitoring** section:

- **SIP Link Monitoring:** Select "**Use Session Manager Configuration**" (or choose an alternate Link Monitoring approach for this entity, if desired).

The following screen shows the SIP Entity defined for the SBC in the sample configuration.



6.5. Entity Links

The SIP trunk between Session Manager and Avaya Communication Server 1000E is described by an Entity Link, as is the SIP trunk between Session Manager and the SBC.

6.5.1 Entity Link to Avaya Communication Server 1000E Entity

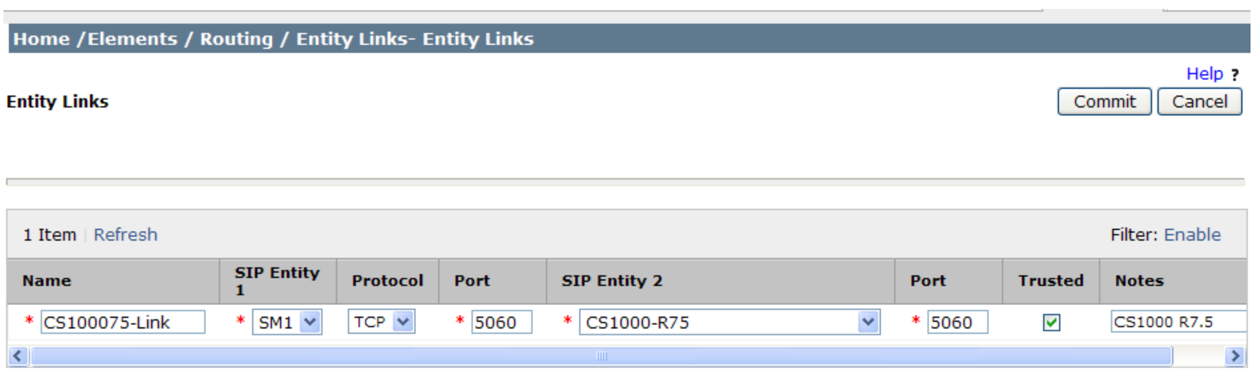
Select **Entity Links** from the left navigation menu.

Click **New** (not shown). Enter the following values**.**

- **Name** Enter an identifier for the link.
- **SIP Entity 1** Select SIP Entity defined for Session Manager
- **SIP Entity 2** Select the SIP Entity defined for the CS1000E
- **Protocol** After selecting both SIP Entities, select "**TCP"**.
- **Port** Verify **Port** for both SIP entities is the default listen port. For the sample configuration, default listen port is "**5060**".
- **Trusted** Enter ☑
- **Notes** Enter a brief description. [Optional]

Click **Commit** to save the **Entity Link** definition.

The following screen shows the entity link defined for the SIP trunk between Session Manager and Avaya Communication Server 1000E.



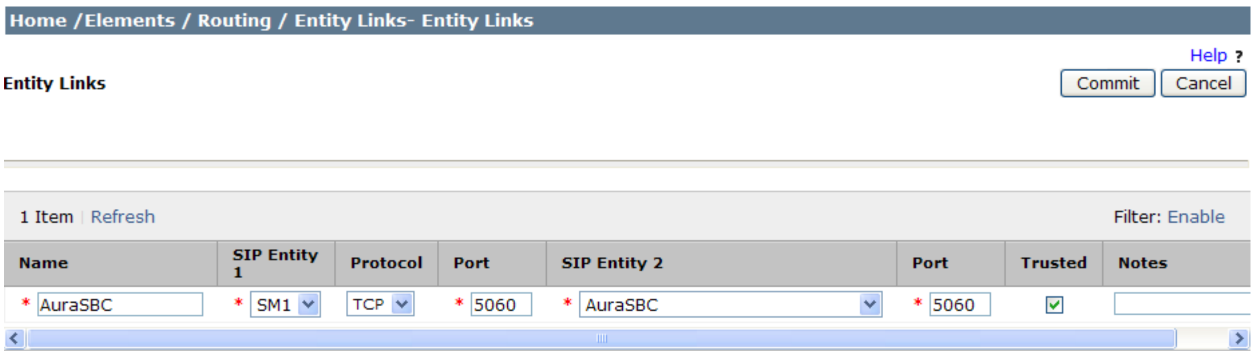
6.5.2 Entity Link to SBC

Select **Entity Links** from the left navigation menu. Click **New** (not shown). Enter the following values**.**

- **Name** Enter an identifier for the link.
- **SIP Entity 1** Select SIP Entity defined for Session Manager
- **SIP Entity 2** Select the SIP Entity defined for the SBC.
- **Protocol** After selecting both SIP Entities, select "**TCP"**.
- **Port** Verify **Port** for both SIP entities is the default listen port. For the sample configuration, default listen port is "**5060**".
- **Trusted** Enter ☑
- **Notes** Enter a brief description. [Optional]

Click **Commit** to save the **Entity Link** definition.

The following screen shows the entity link defined for the SIP trunk between Session Manager and the SBC.



6.6. Routing Policies

Routing policies describe the conditions under which calls will be routed to the Avaya Communication Server 1000E or SBC.

6.6.1 Routing Policy to Avaya Communication Server 1000E

To add a new routing policy, select **Routing Policies.** Click **New** (not shown). In the **General** section, enter the following values.

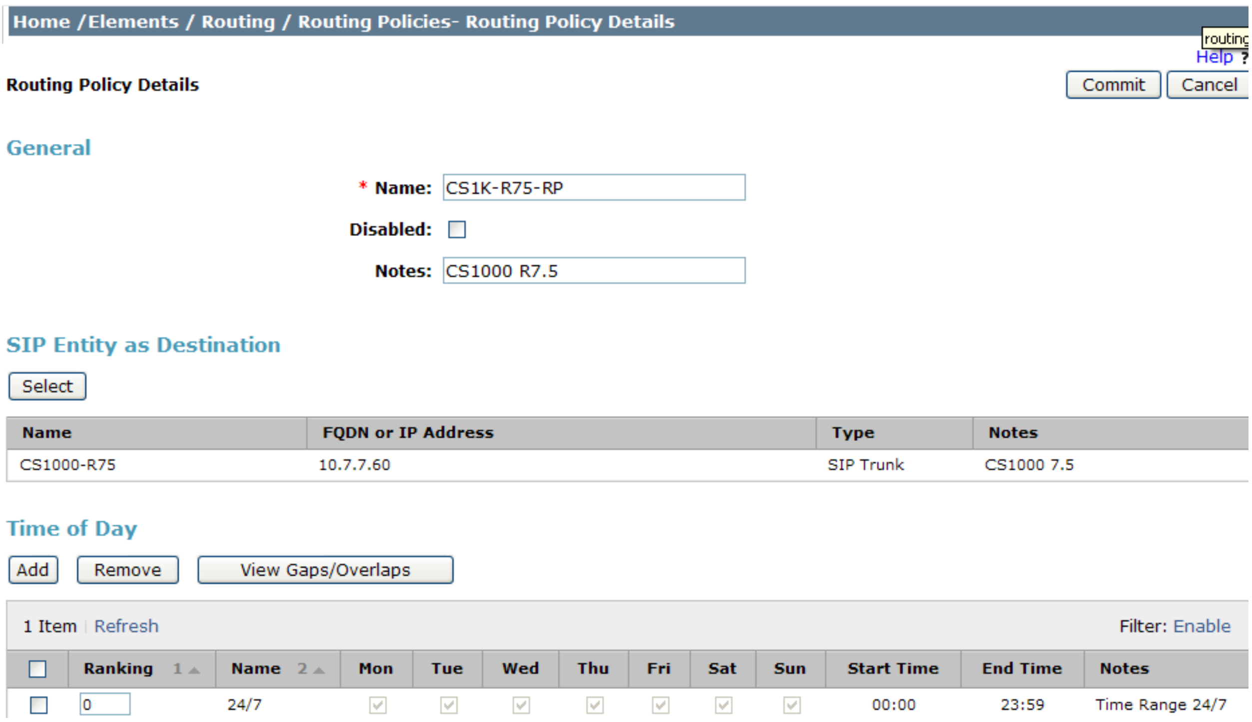
- **Name:** Enter an identifier to define the routing policy
- **Disabled:** Leave unchecked.
- **Notes:** Enter a brief description. [Optional]

In the **SIP Entity as Destination** section, click **Select.** The **SIP Entity List** page opens (not shown).

- Select the SIP Entity associated with the CS1000E and click **Select.**
- The selected SIP Entity displays on the **Routing Policy Details** page.

In the **Time of Day** section, add an appropriate time of day. In the sample configuration, time of day was not a relevant routing criteria, so the "24/7" range was chosen. Use default values for remaining fields. Click **Commit** to save the Routing Policy definition.

The following screen shows the Routing Policy for Avaya Communication Server 1000E.



6.6.2 Routing Policy to SBC

The configuration in this section is not required if only inbound toll-free calls will be configured for the Verizon IPCC Services. In the sample configuration, Verizon IP Trunk service, which supports outbound dialing to the PSTN, was also available.

To add a new routing policy, select **Routing Policies.** Click **New** (not shown). In the **General** section, enter the following values.

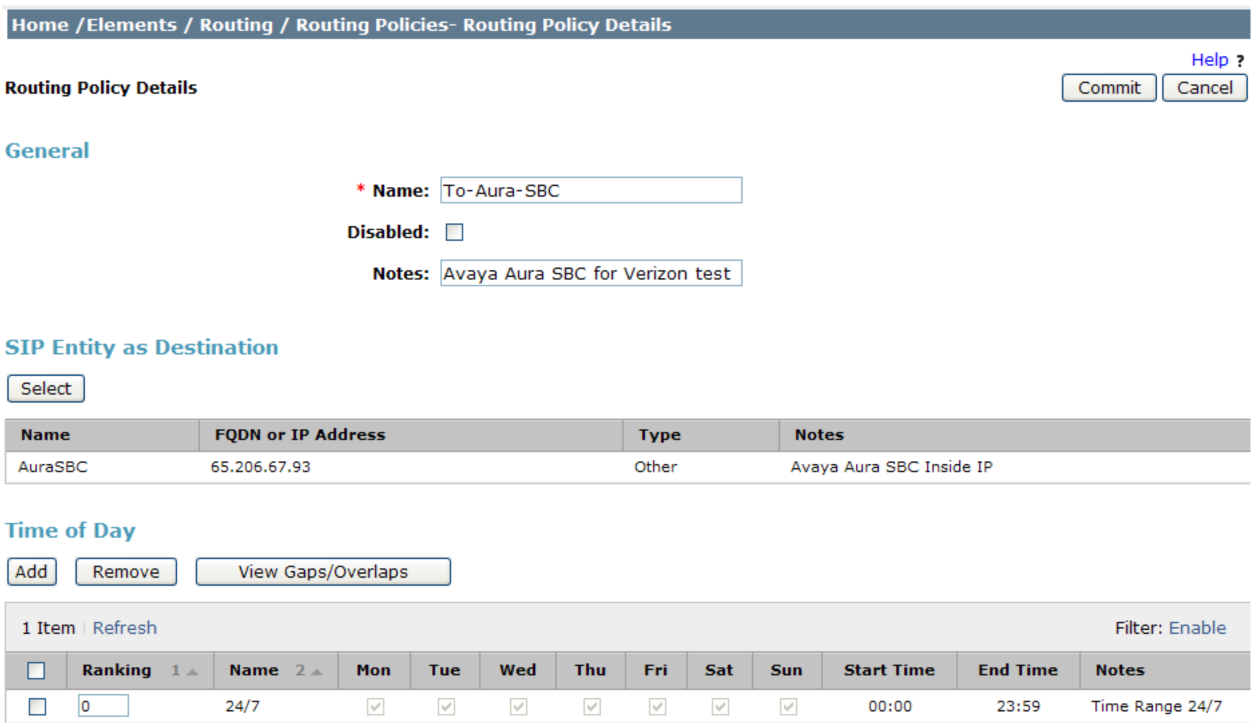
- **Name:** Enter an identifier to define the routing policy
- **Disabled:** Leave unchecked.
- **Notes:** Enter a brief description. [Optional]

In the **SIP Entity as Destination** section, click **Select.** The **SIP Entity List** page opens (not shown).

- Select the SIP Entity associated with the SBC and click **Select.**
- The selected SIP Entity displays on the **Routing Policy Details** page.

In the **Time of Day** section, add an appropriate time of day. In the sample configuration, time of day was not a relevant routing criteria, so the “24/7” range was chosen. Use default values for remaining fields. Click **Commit** to save the Routing Policy definition.

The following screen shows the Routing Policy for the SBC.



6.7. Dial Patterns

Dial patterns are used to route calls to the appropriate routing policies, and ultimately to the appropriate SIP Entities.

6.7.1 Inbound Verizon Calls to CS1000E Users

To define a dial pattern, select **Dial Patterns** from the navigation menu. Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

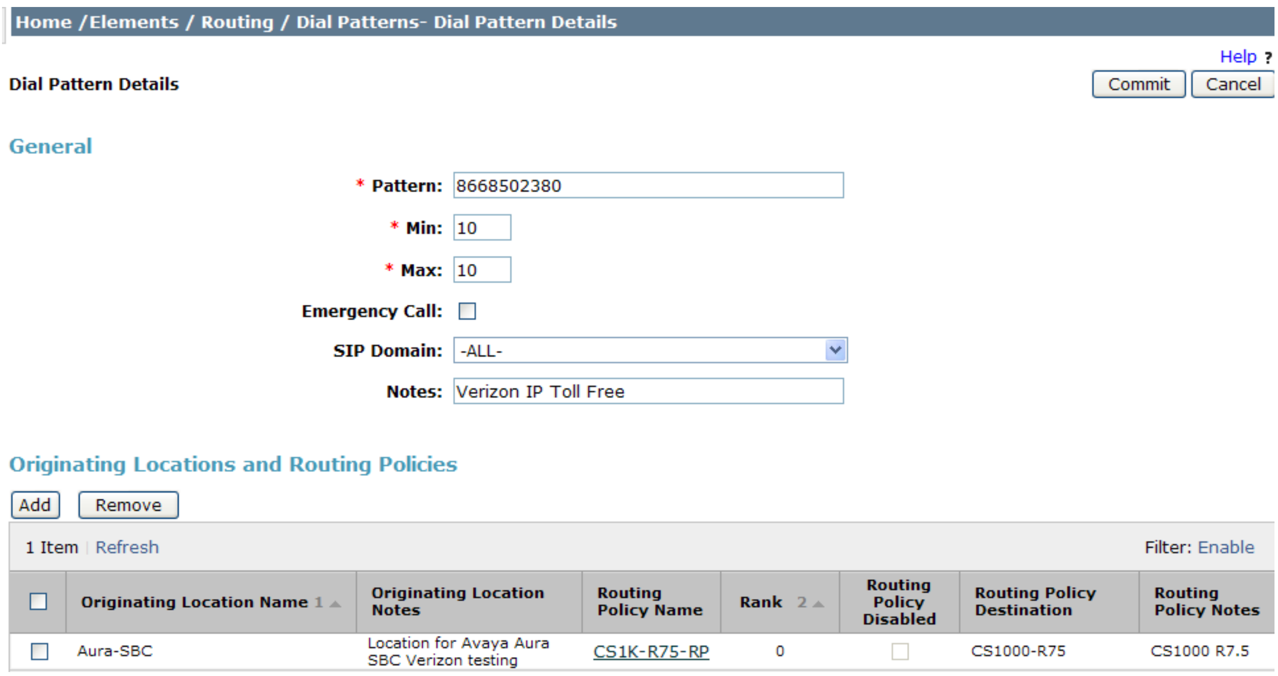
- **Pattern:** Enter dial pattern for calls to Avaya Communication Server 1000E (e.g., a Verizon toll-free number)
- **Min:** Enter the minimum number of digits.
- **Max:** Enter the maximum number of digits.
- **SIP Domain:** Select a SIP Domain from drop-down menu or select **"All"** if Session Manager should route incoming calls from all SIP domains.
- **Notes:** Enter a brief description. [Optional]

In the **Originating Locations and Routing Policies** section, click **Add.**

The **Originating Locations and Routing Policy List** page opens (not shown).

- In the **Originating Location** list, select "**Apply the Selected Routing Policies to All Originating Locations**" or alternatively, select a specific location. In the example below, the SBC location was selected as the originating location.
- In the **Routing Policies** table, select the Routing Policy defined for Avaya Communication Server 1000E**.**
- Click **Select** to save these changes and return to **Dial Pattern Details** page.

Click **Commit** to save. The following screen shows an example Dial Pattern. In the screen, Verizon IP Toll Free number 866-850-2380 is routed to the Avaya CS1000E. The adapter assigned to the Avaya CS1000E (in Section 6.3) will map the toll-free number to the desired CS1000E Directory Number. Repeat this procedure as needed to allow additional Verizon toll-free numbers to be routed to the CS1000E. Wildcards may be used in the **Pattern** field so that blocks of matching numbers are routed based on a single dial pattern.



6.7.2 Outbound Calls to Verizon (Optional)

The configuration in this section is not required if only inbound toll-free calls will be configured for the Verizon IPCC Services. In the sample configuration, Verizon IP Trunk service, which supports outbound dialing to the PSTN, was also available.

To define a dial pattern, select **Dial Patterns** from the navigation menu. Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **Pattern:** Enter dial pattern for calls destined for the Verizon network
- **Min:** Enter the minimum number of digits.
- **Max:** Enter the maximum number of digits.
- **SIP Domain:** Select a SIP Domain from drop-down menu or select **"All"** if Session Manager should route outgoing calls from all SIP domains.
- **Notes:** Enter a brief description. [Optional]

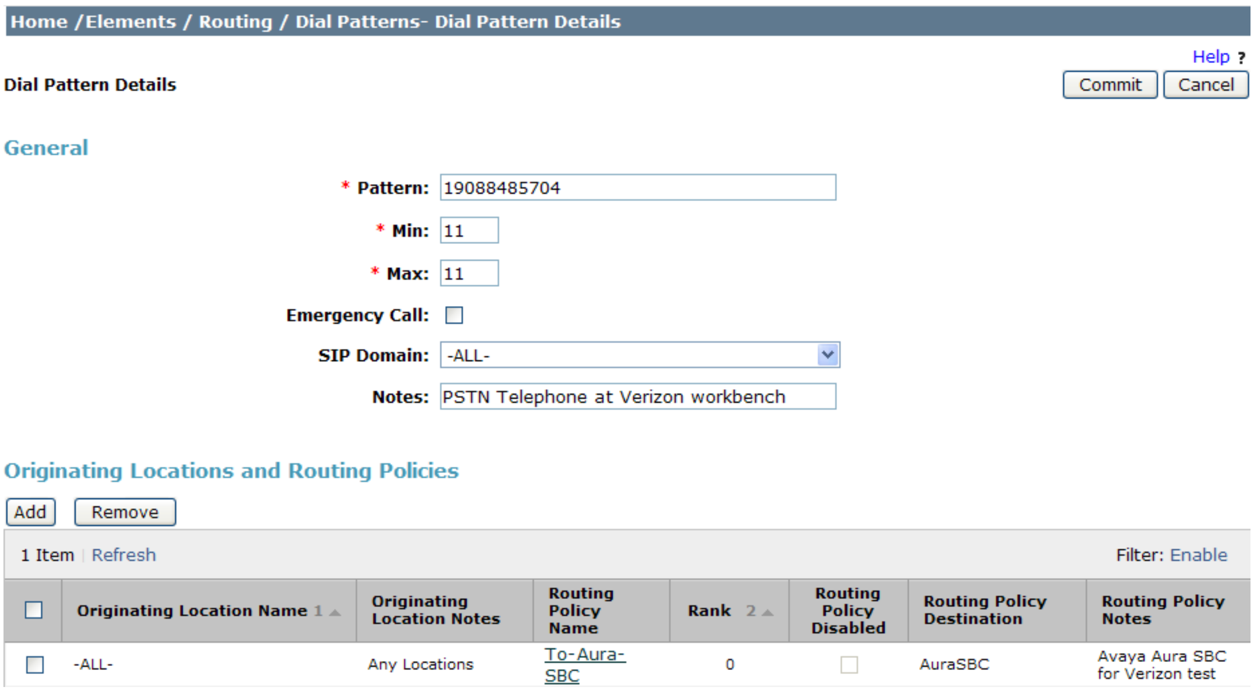
In the **Originating Locations and Routing Policies** section, click **Add.**

The **Originating Locations and Routing Policy List** page opens (not shown).

- In the **Originating Location** list, select "**Apply the Selected Routing Policies to All Originating Locations**" or alternatively, select a specific originating location. In the **Routing Policies** table, select the Routing Policy defined for the SBC.

- Click **Select** to save these changes and return to **Dial Pattern Details** page.

Click **Commit** to save. The following screen shows an example Dial Pattern defined for the sample configuration. Repeat this procedure as needed to allow additional PSTN numbers to be routed to the Verizon network via the SBC. Wildcards may be used in the **Pattern** field so that blocks of matching numbers are routed based on a single dial pattern.



7. Configure Avaya Aura® Session Border Controller (SBC)

This section illustrates an example configuration of the Avaya Aura® SBC. In the sample configuration, the Avaya Aura® SBC runs on its own S8800 Server as an application template using System Platform. The installation of the System Platform is assumed to have been previously completed. 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.

Reference [AuraSBC-IP-Trunk] is a companion Application Notes 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. In Section 7 of reference [AuraSBC-IP-Trunk], the installation, licensing, and initial wizard configuration of the SBC are detailed. These procedures will not be repeated here.

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, if only the Verizon IPCC Service is necessary, the procedures that use the installation wizard from reference [AuraSBC-IP-Trunk] can be used to connect to the Verizon IPCC Service only, using the appropriate Verizon-provided IPCC Service IP Address and port information. The wizard can be used for one SIP Service Provider service connection only.

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 specific SBC behaviors (e.g., header manipulations) can be performed through the element manager GUI as shown in Section 7.3.

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.

7.1. Avaya Aura® Session Border Controller (SBC) Installation

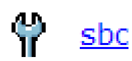
For the installation procedures used in the sample configuration, please refer to Section 7.1 of reference [AuraSBC-IP-Trunk].

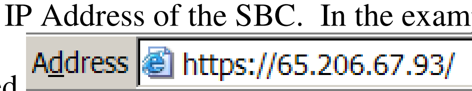
7.2. Avaya Aura® Session Border Controller (SBC) Licensing

For the licensing procedures used in the sample configuration, please refer to Section 7.2 of reference [AuraSBC-IP-Trunk].

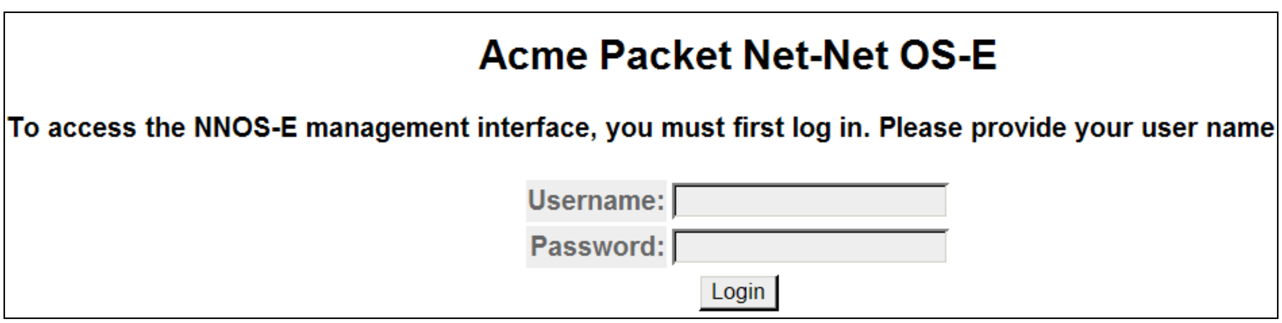
7.3. SBC Element Manager Configuration

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 7.1 – Section 7.3 of reference [AuraSBC-IP-Trunk] has already been completed. In the screens below, it can also be observed that Section 12 (an optional procedure covering use of DNS-SRV with Verizon) from reference [AuraSBC-IP-Trunk] has also already been completed.

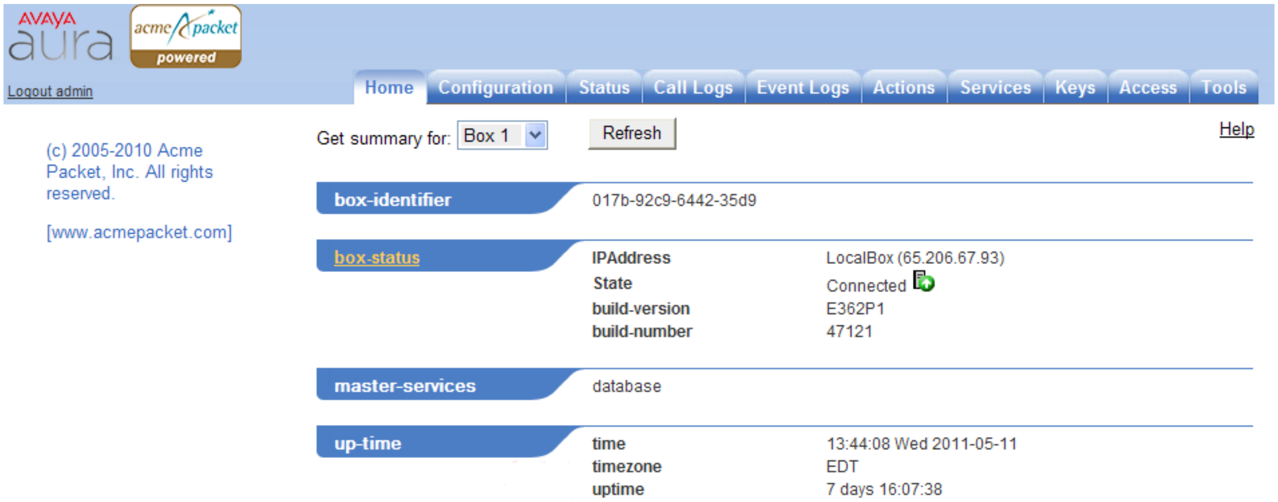
The configuration screens will be familiar to the reader experienced with the Acme Packet Net-Net OS-E. To log in, either select the wrench 

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 

to access a log in screen. Enter appropriate **Username** and **Password** and click **Login**.

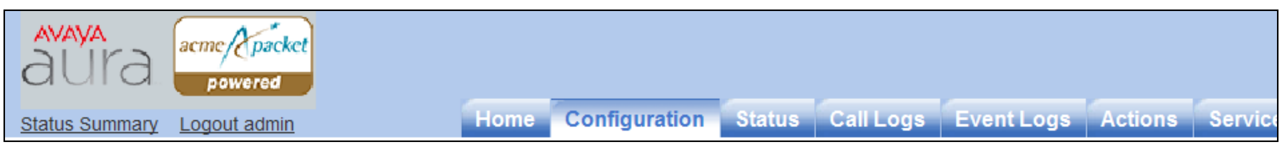


The following shows an abridged **Home** screen after logging in. Note the tabs at the top.



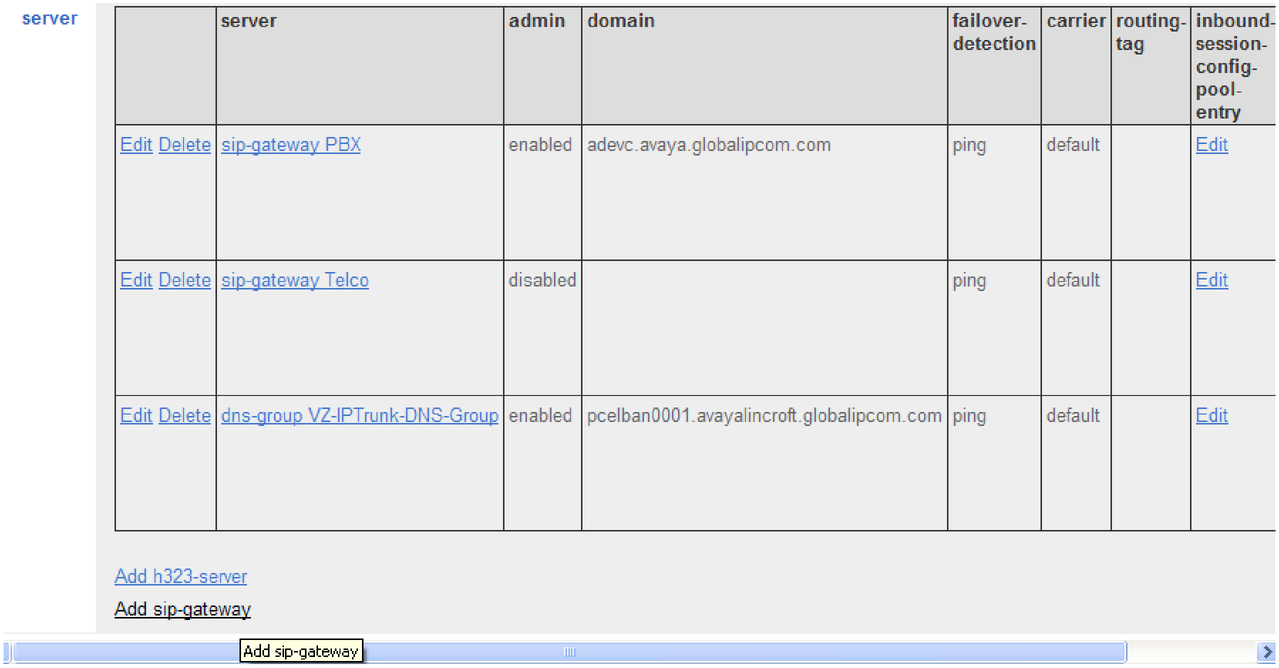
7.3.1 Adding SIP Gateway to Verizon IP Contact Center Service

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



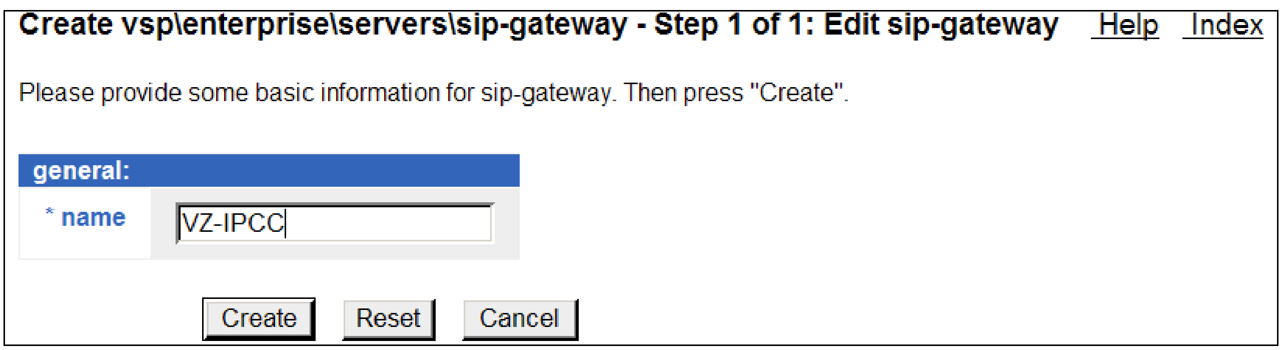
Using the menu on the left hand side, expand **vsp → enterprise → servers** as shown below.

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 Session Manager. There is also a "sip-gateway Telco" previously configured for connectivity to the Verizon IP Trunk Service, and an optional "DNS Group" illustrated in Section 12 of reference [AuraSBC-IP-Trunk]. 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.

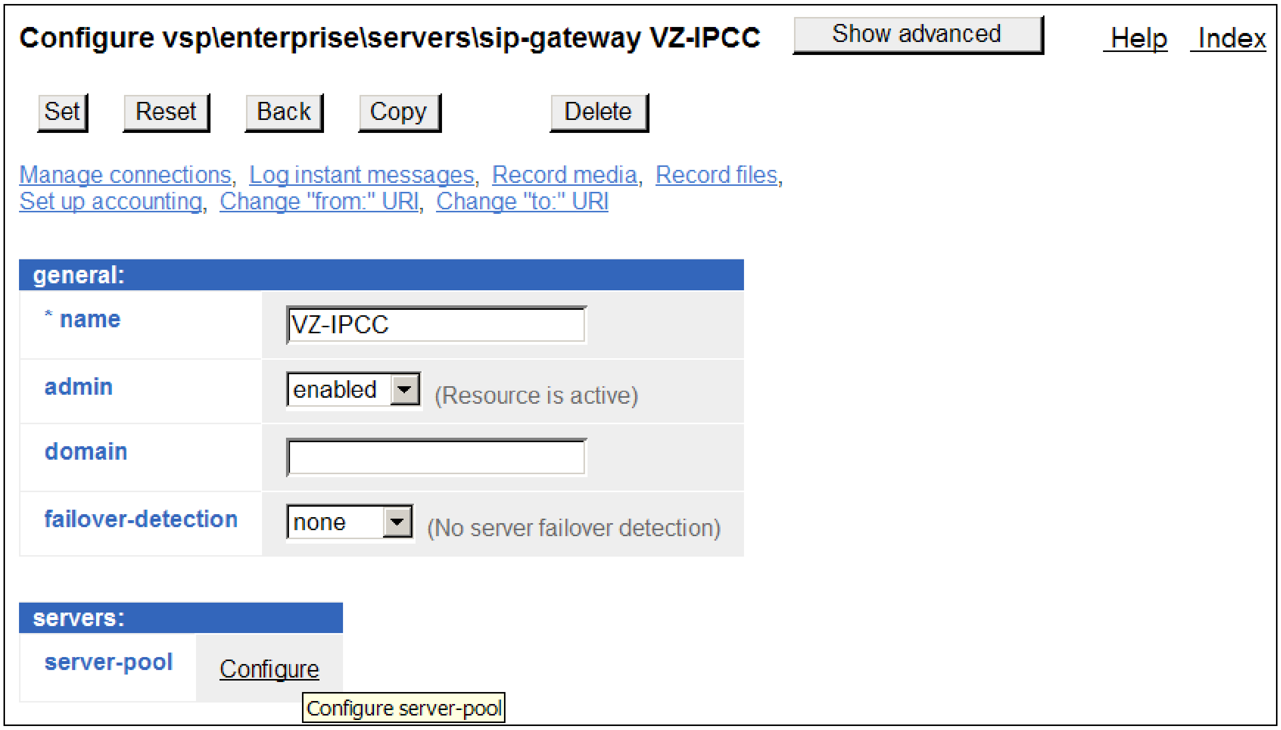


	server	admin	domain	failover-detection	carrier	routing-tag	inbound-session-config-pool-entry
Edit Delete	sip-gateway PBX	enabled	adevc.avaya.globalipcom.com	ping	default		Edit
Edit Delete	sip-gateway Telco	disabled		ping	default		Edit
Edit Delete	dns-group VZ-IPTrunk-DNS-Group	enabled	pcelban0001.avayalincroft.globalipcom.com	ping	default		Edit

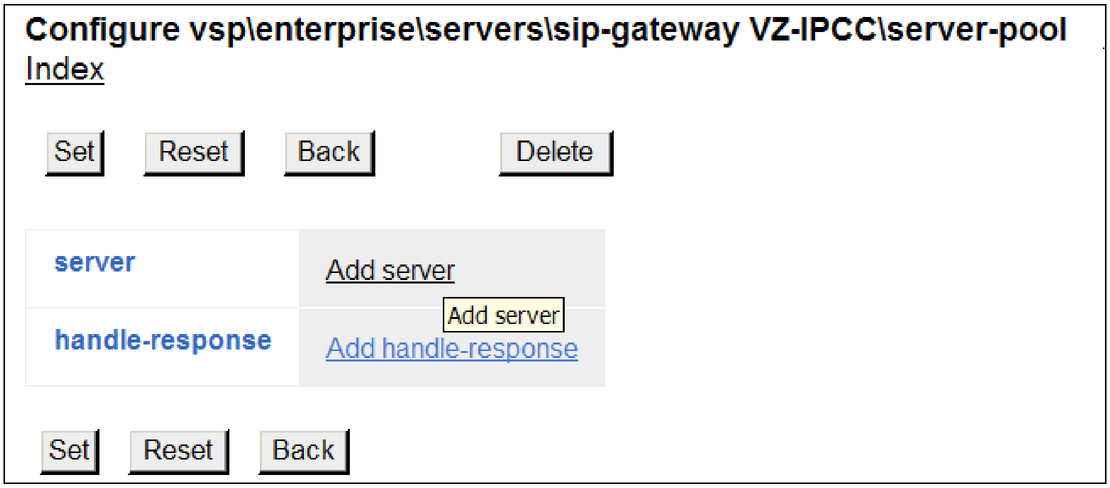
In the resultant screen shown below, enter an appropriate **name** for the new sip-gateway to the Verizon IP Contact Center service and click **Create**.



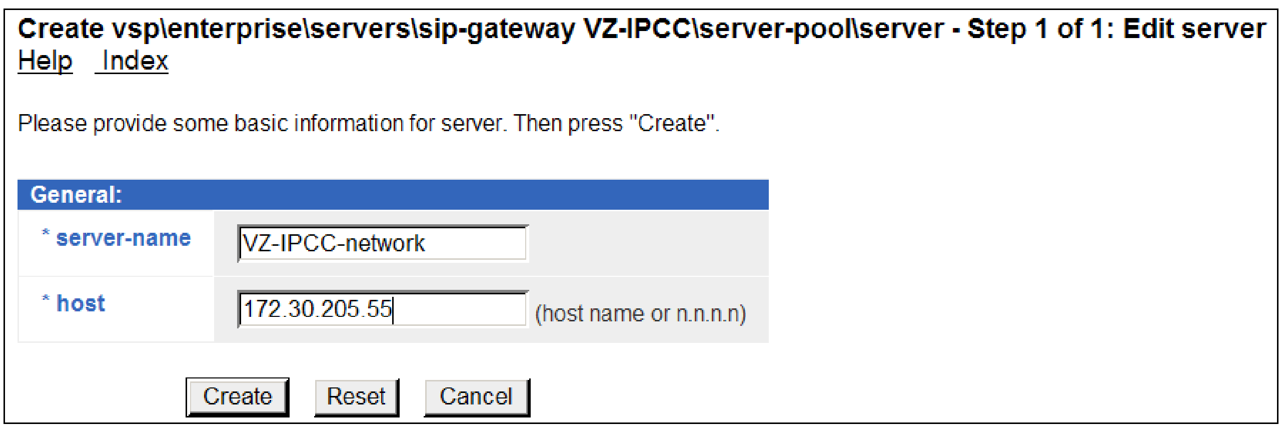
In the resultant screen, click **Configure** under the “servers: server-pool” heading, as shown below.



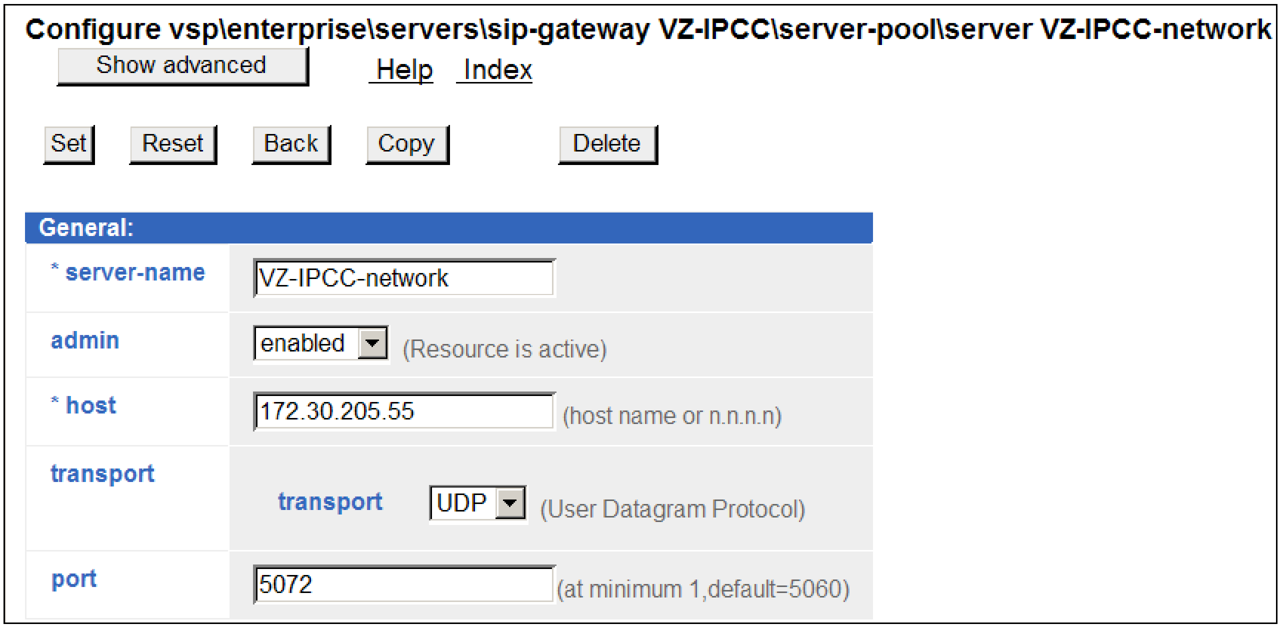
In the resultant screen, click **Add server** as shown below.



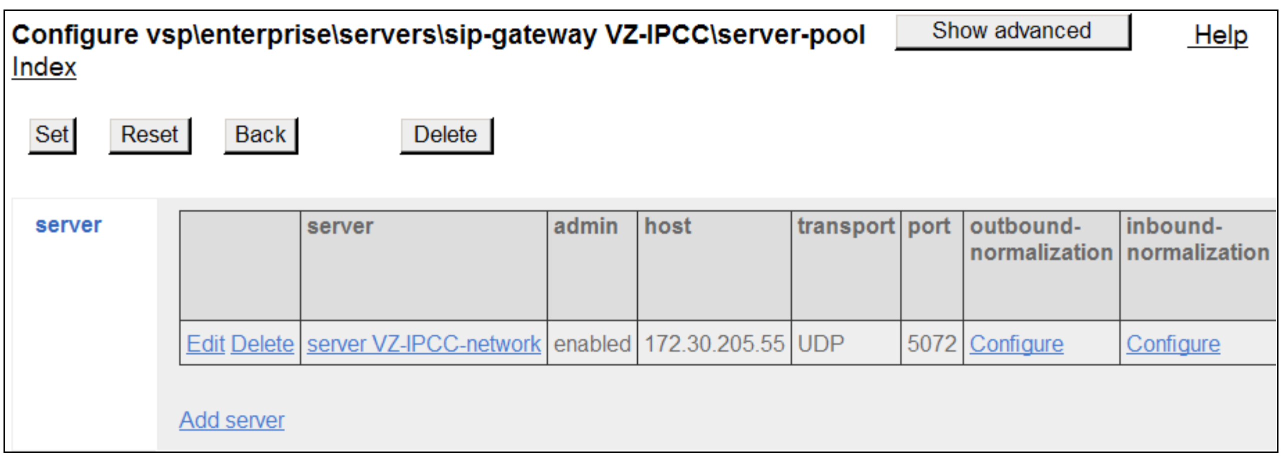
In the resultant screen, enter an appropriate **server-name** and **host** for the Verizon IP Contact Center service. In the screen shown below, the IP Address 172.30.205.55 was provided by Verizon as the SIP signaling IP Address of the IP Contact Center service. Click **Create**.



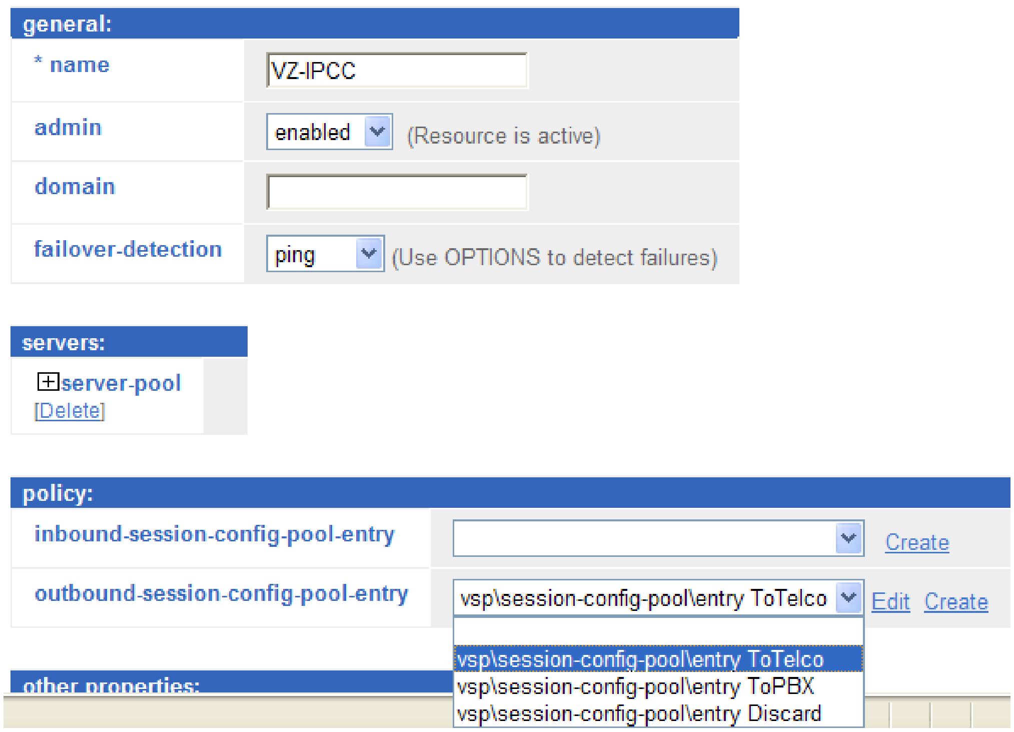
In the resultant screen, select UDP as the **transport** and enter an appropriate **port**. In the sample configuration, Verizon IP Contact Center service expected the enterprise to send SIP signaling to IP Address 172.30.205.55 and port 5072, as shown below. Click **Set**.



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

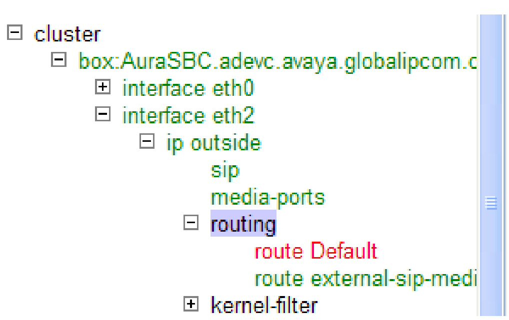


Using the left-side menu, navigate to **vsp → enterprise → servers → sip-gateway** and select the newly created "VZ-IPCC" entry. Scroll down to the policy heading. Using the **outbound-session-config-pool-entry** drop-down menu, select the entry "vsp\session-config-pool\entry To-Telco" as shown in the screen below. This session-config-pool entry was created by the wizard configuration shown in reference [AuraSBC-IP-Trunk]. Using the **failover-detection** drop-down, select "ping" to cause the SBC to periodically send SIP OPTIONS messages to the Verizon IPCC Service to verify the health of the connection.



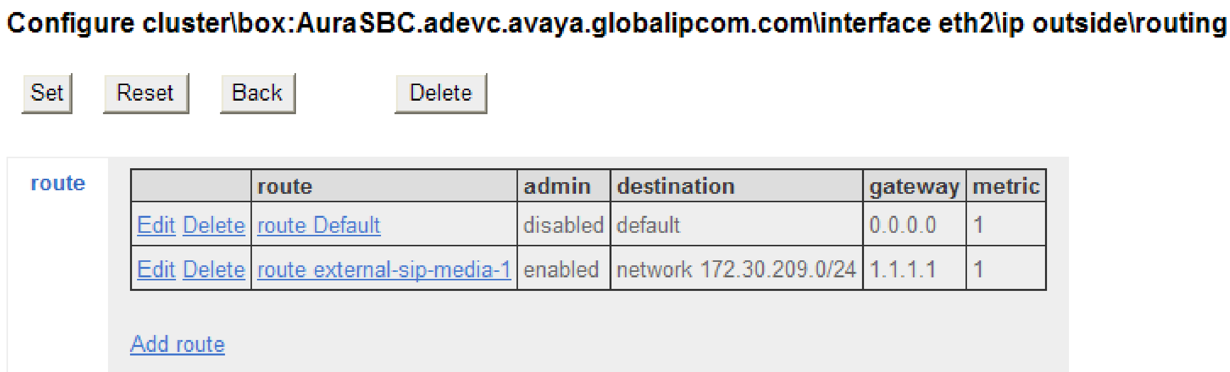
7.3.2 Adding IP Routing for Verizon IP Contact Center Network

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

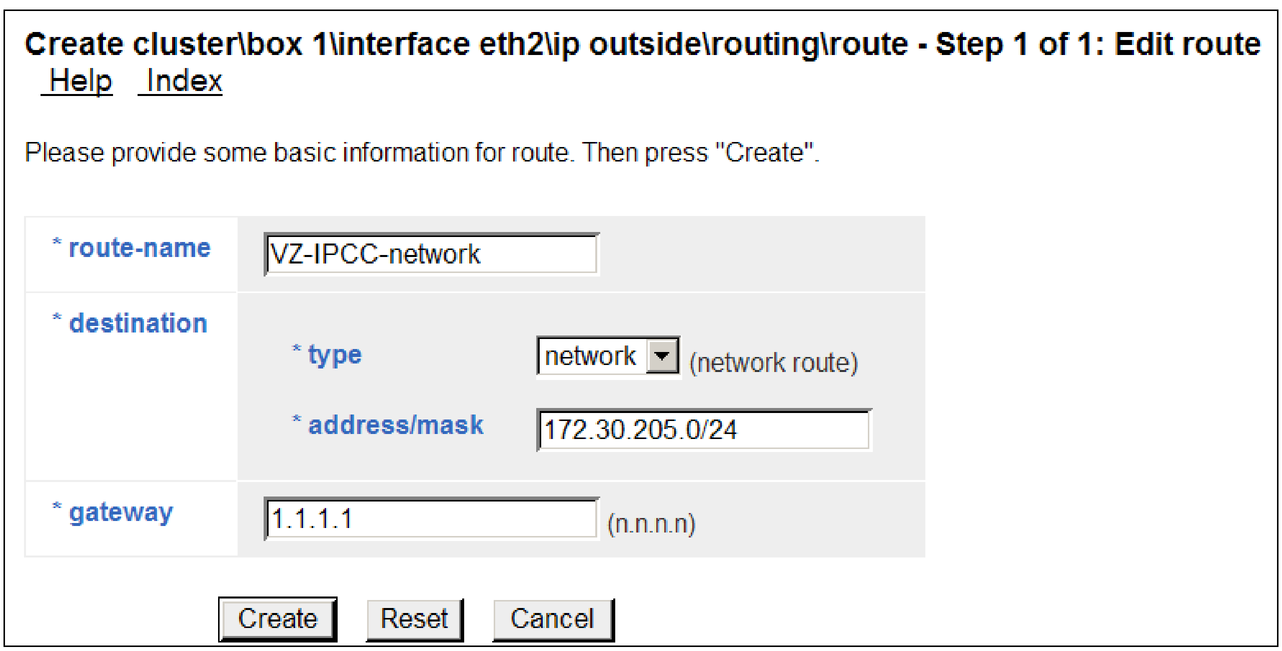


In the right-side, a screen such as the following is displayed. The screen below shows the IP route established from reference [AuraSBC-IP-Trunk]. The Verizon IP Trunk Service on network 172.30.209.0/24 used gateway 1.1.1.1. A new route will be added for the Verizon IP Contact

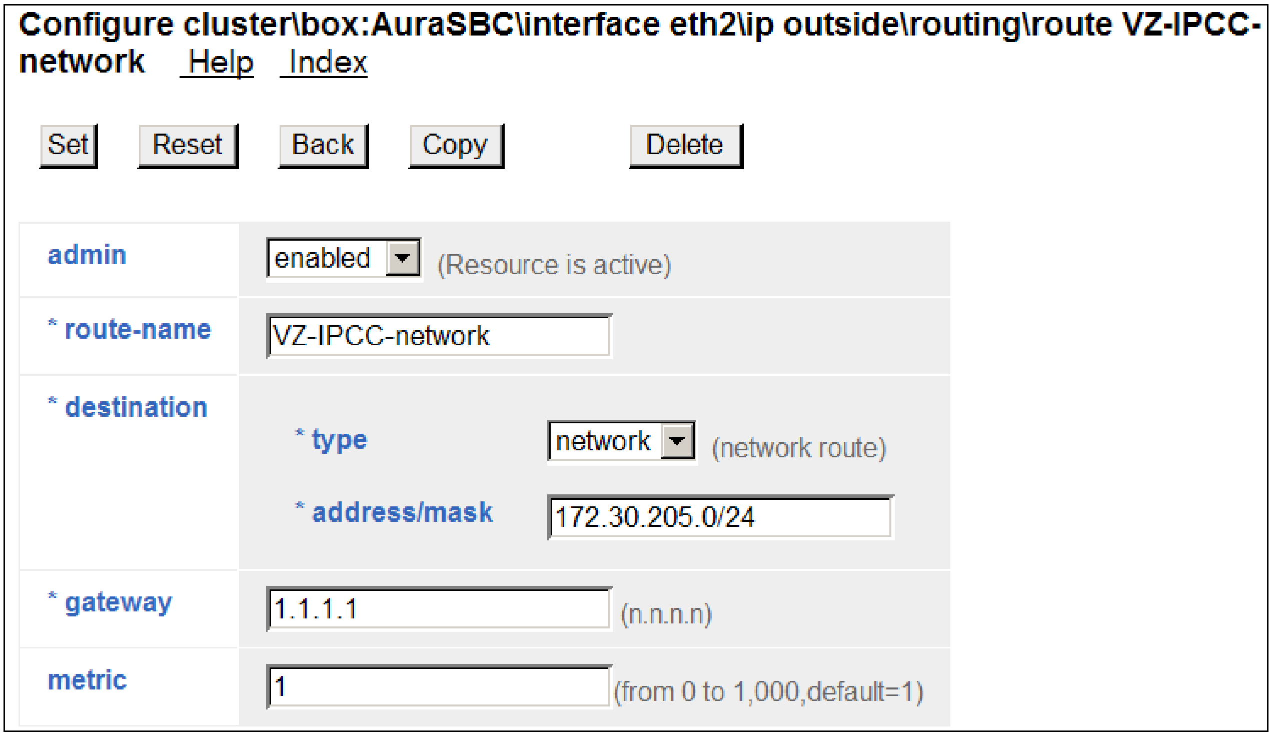
Center service using the same gateway. In the sample configuration, Verizon IP Trunk service and Verizon IP Contact Center service shared the same PIP access circuit. Click **Add route**.



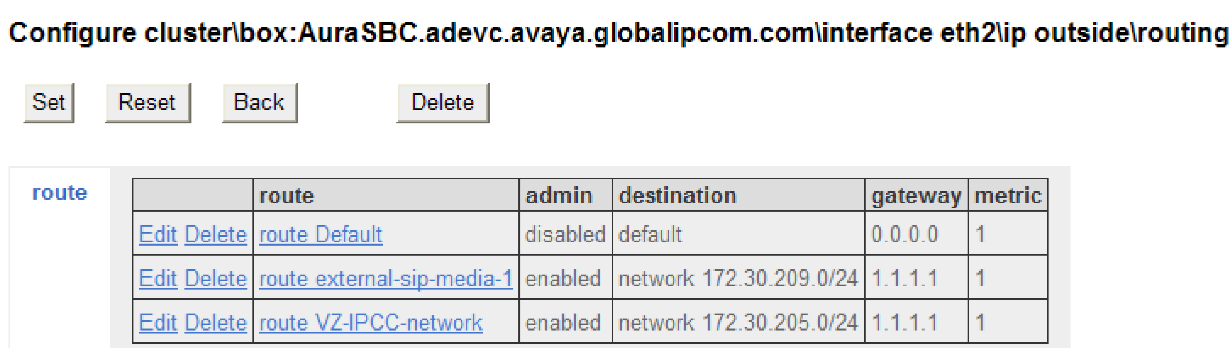
In the resultant screen shown below, enter an appropriate **route-name**. Using the **type** drop-down, select "network". In the **address/mask** field, enter the IP address and network mask associated with the Verizon IP Contact Center service. In the sample configuration, the Verizon IP Contact Center 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.

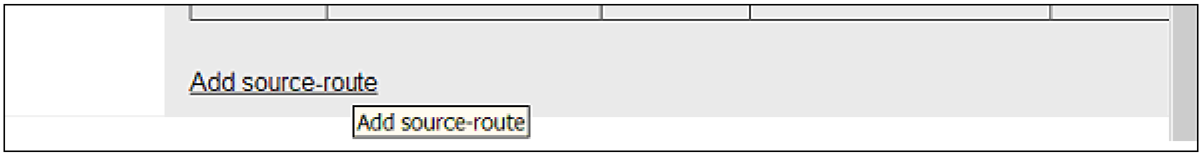


The following screen summarizes the updated routing configuration.

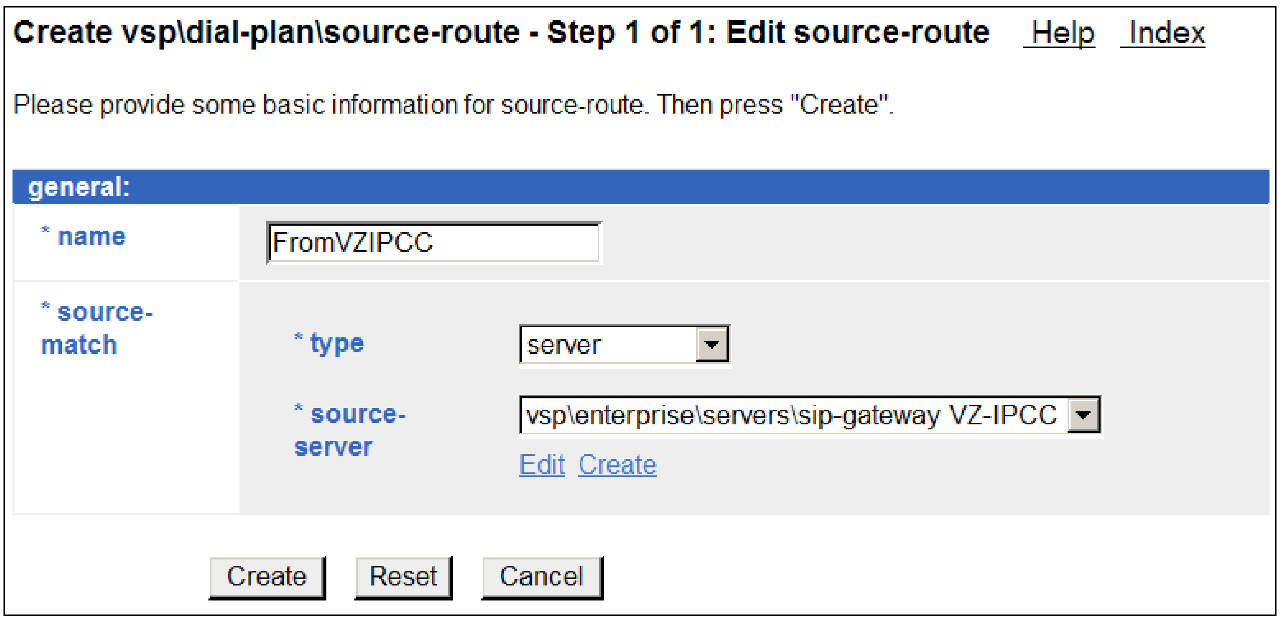


7.3.3 Configure Dial-Plan

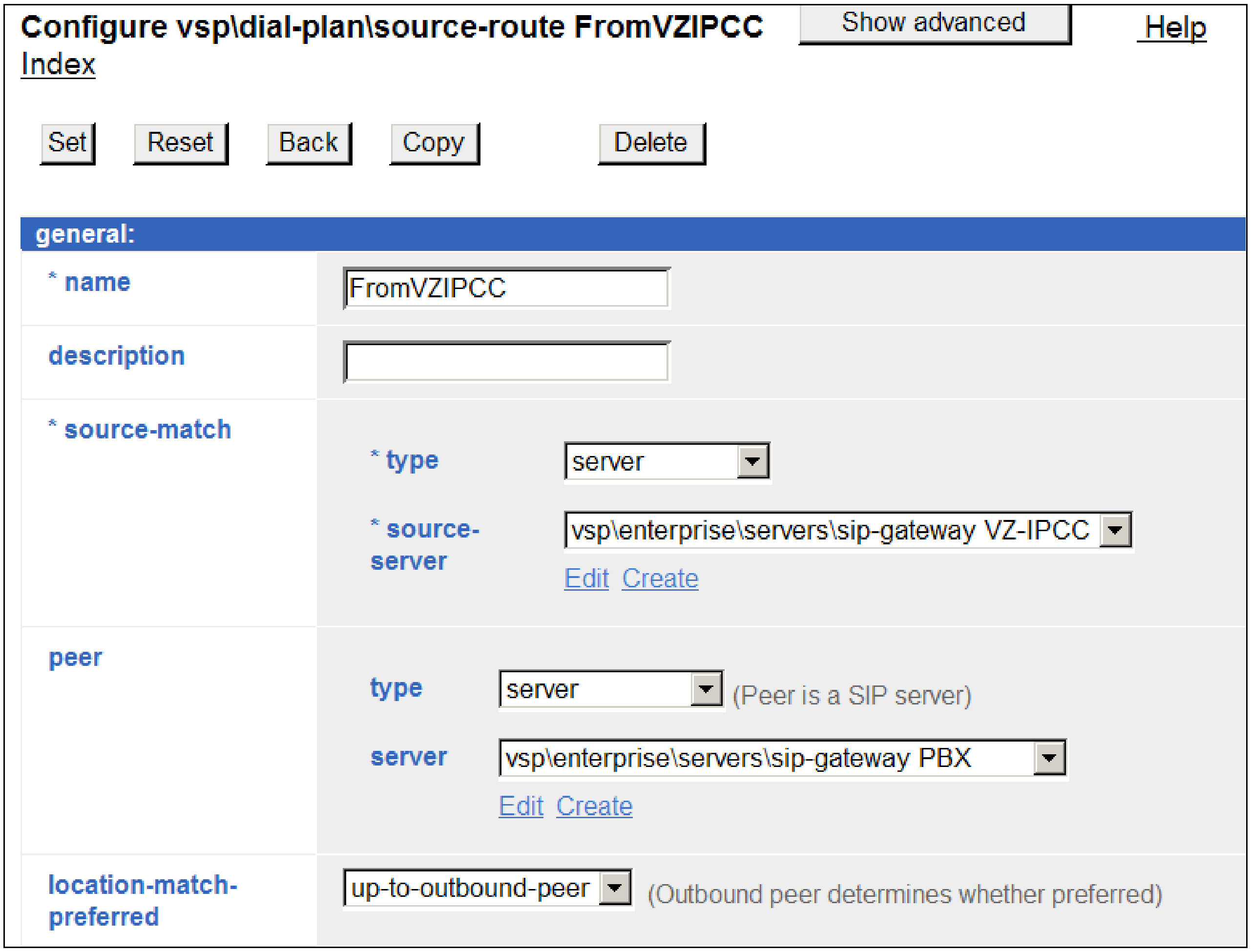
From the left-side menu, select **vsp → dial-plan.** In the right-hand side, scroll down and click **Add source-route** as shown below.



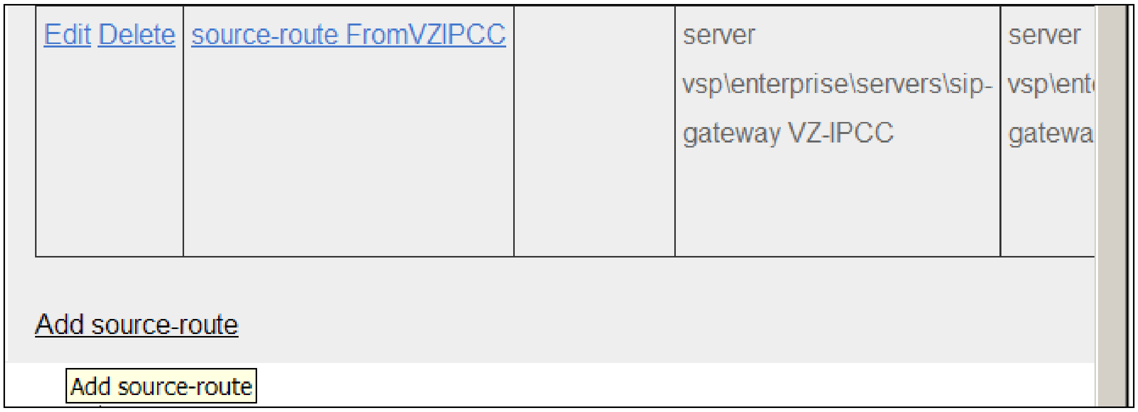
In the resultant screen, enter an appropriate name in the **name** field. In the **type** field drop-down menu, select "server", and in the **source-server** drop-down menu, select the sip-gateway entry previously created in **Section 7.3.1**, as shown below. Click **Create**.



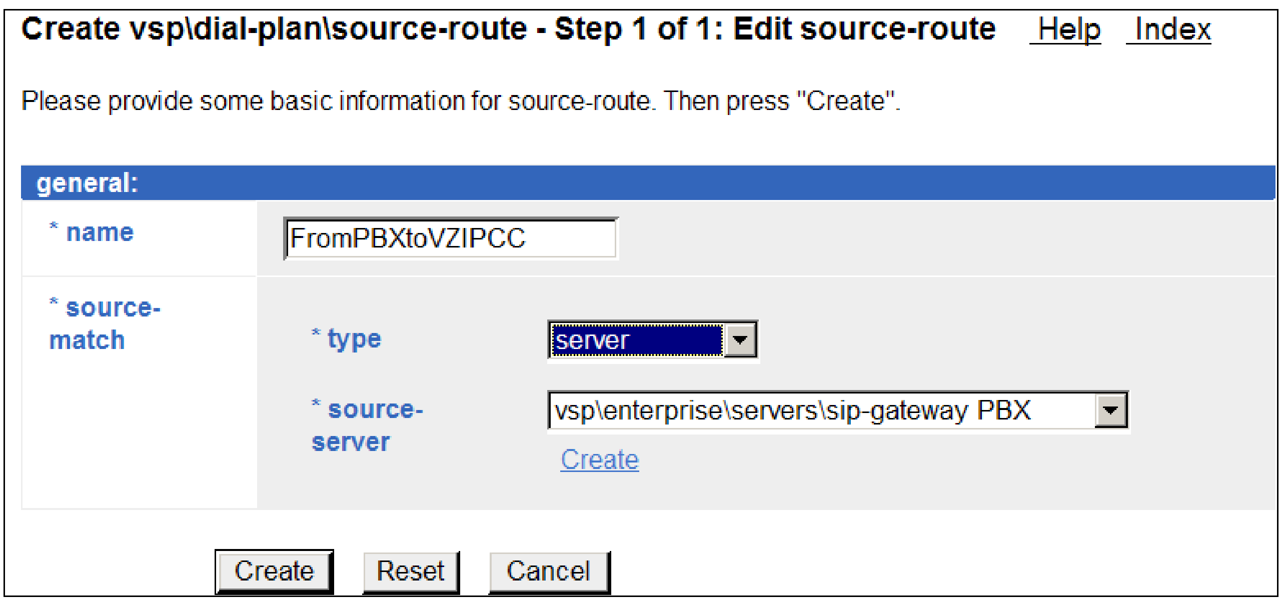
In the resultant screen, in the **peer** area, select "server" from the **type** drop-down. In the **server** drop-down, select the sip-gateway representing the enterprise SIP equipment. In the sample configuration, "vsp\enterprise\servers\sip-gateway PBX" already existed from the wizard configuration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



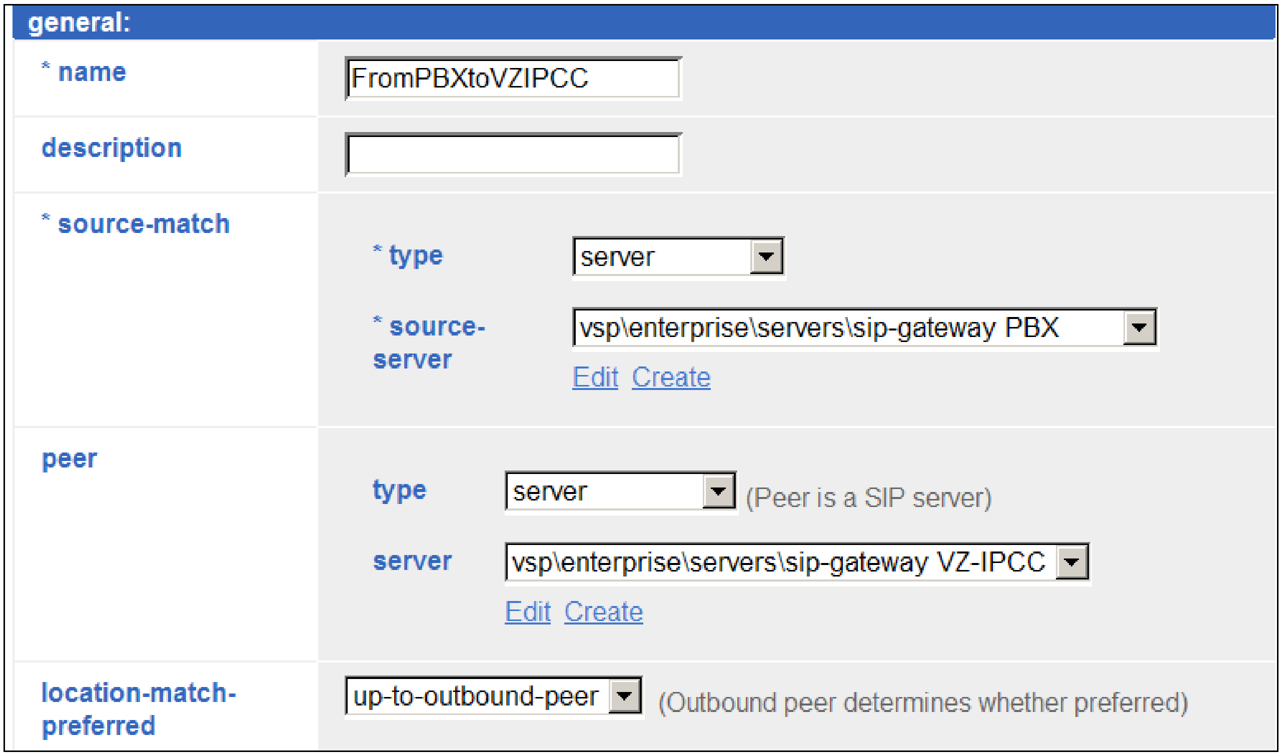
These same procedures can be repeated to create another source-route. Scroll down in the source-route area and click **Add source route** as shown below.



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 Session Manager. Click **Create**.



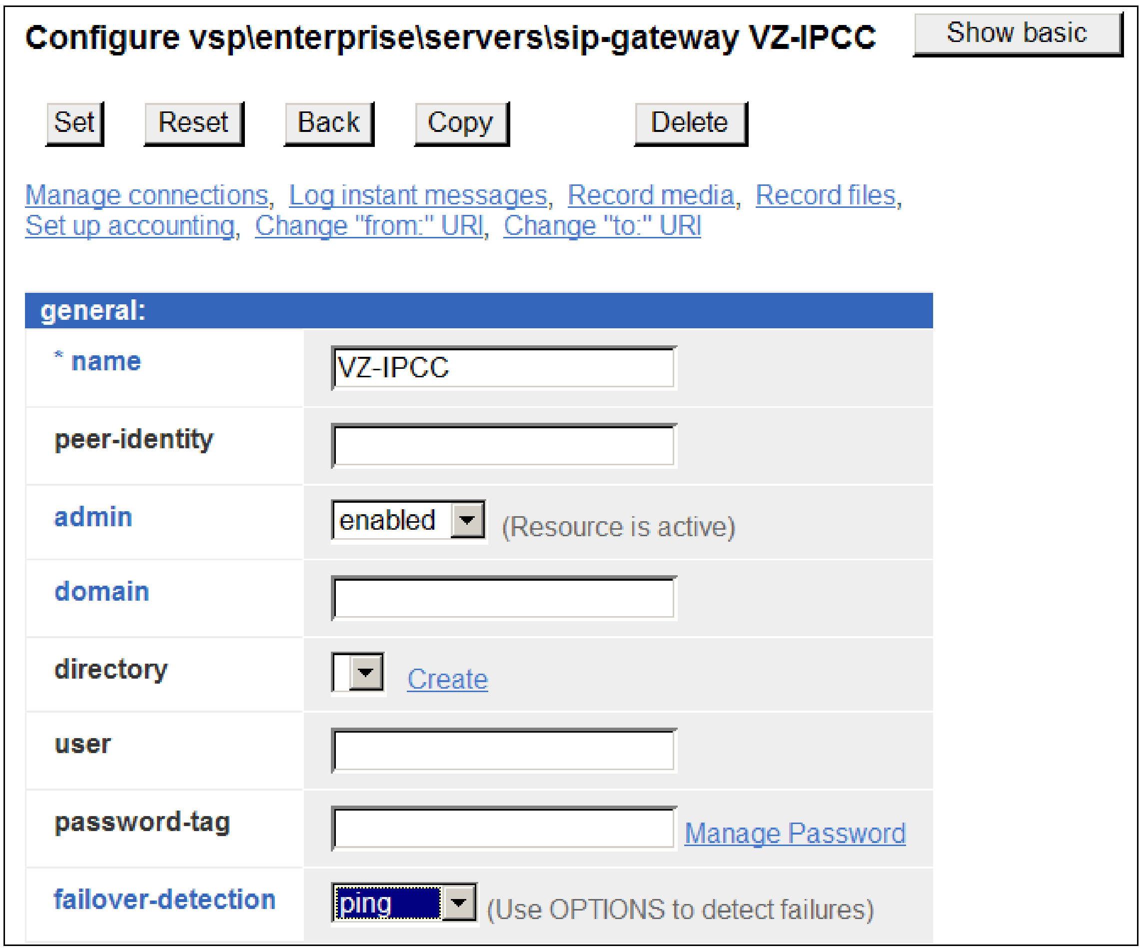
In the **peer** area, select "server" from the **type** drop-down. From the **server** drop-down, select the sip-gateway corresponding to the Verizon IP Contact Center service created in **Section 7.3.1**. Click **Set** (not shown).



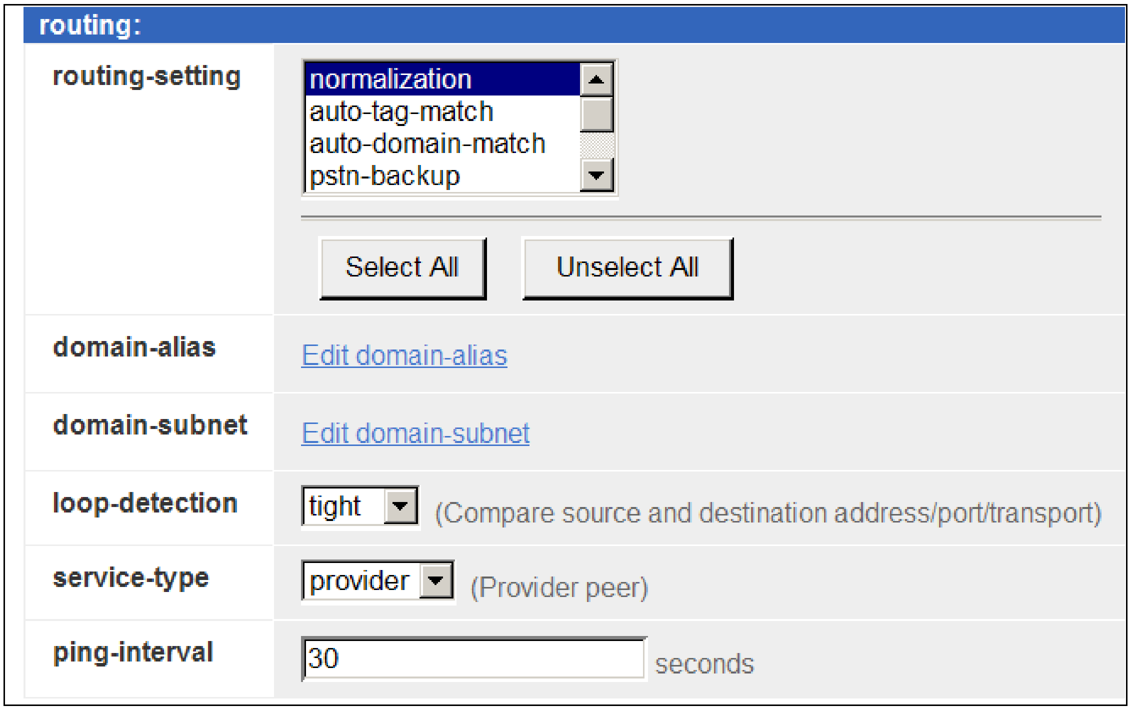
7.3.4 Configure OPTIONS ping to Verizon IP Contact Center

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 7.3.1. 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, verify "ping" as selected as shown below.

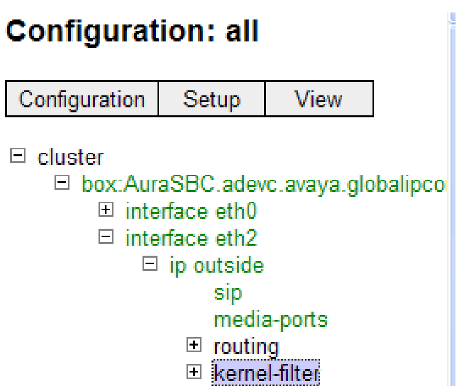


Scroll down and locate the **ping-interval** parameter, which is considered an "advanced" parameter (i.e., only available after the **Show Advanced** button has been clicked). Enter the desired period, in seconds, that the SBC will use to source SIP OPTIONS messages towards the Verizon IP Contact Center 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.



7.3.5 Configure Kernel-Filter for Verizon IPCC

Using the left-hand side navigation menu, select **kernel-filter** as shown in the menu tree below.



On the right, the kernel-filters that were established by the wizard configuration at installation time can be observed. In the sample configuration, the wizard configuration in reference [AuraSBC-IP-Trunk] has established an "allow-rule" that permits traffic from the Verizon IP Trunk service network 172.30.209.0/24, as shown below. Had the wizard been run for Verizon IPCC service, an allow-rule would already exist for the Verizon IPCC network, and this step would not be required.

Configure cluster\box:AuraSBC.adevc.avaya.globalipcom.com\interface eth2\ip outside\kernel-filter Help Index

Set Reset Back Delete

allow-rule

	allow-rule	admin	destination-port	source-address/mask	source-port	protocol
Edit Delete	allow-rule allow-sip-udp-from-peer-1	enabled	5060	172.30.209.0/24	0	udp

Add allow-rule

deny-rule

	deny-rule	admin	destination-port	source-address/mask	source-port	protocol
Edit Delete	deny-rule deny-all-sip	enabled	5060	0.0.0.0/0	0	all

Click **Add allow-rule**. Enter an appropriate **name**. In the **source-address/mask** field, enter the network information corresponding to the Verizon IPCC Service network. In the sample configuration, "172.30.205.0/24" was entered. Click **Create**.

Create cluster\box 1\interface eth2\ip outside\kernel-filter\allow-rule - Step 1 of 1: Edit allow-rule

Please provide some basic information for allow-rule. Then press "Create".

* name	allow-sip-udp-from-peer-2	
* source-address/mask	172.30.205.0/24	(n.n.n.n/n)

Create Reset Cancel

In the resultant screen, enter "5060" in the **destination-port** field. Verizon IPCC will signal to port 5060. In the **protocol** field, select "udp". In the source-port field, the default "0" may be retained to mimic the configuration that would be performed by the wizard. Alternatively, a more specific port may be entered, such as source-port "5072" used in the sample configuration. Both approaches were tested successfully. Click **Set**.

Configure cluster\box:AuraSBC.adevc.avaya.globalipcom.com\interface eth2\ip outside\kernel-filter\allow-rule allow-sip-udp-from-peer-2 Help Index

Set Reset Back Copy Delete

* name	allow-sip-udp-from-peer-2	
admin	enabled	(Resource is active)
destination-port	5060	(from 0 to 65,535)
* source-address/mask	172.30.205.0/24	(n.n.n.n/n)
source-port	5072	(from 0 to 65,535)
protocol	udp	(User Datagram Protocol)

The following screen shows the resulting kernel-filter, which allows SIP traffic from both the Verizon IP Trunk service (from the wizard configuration shown in Reference [AuraSBC-IP-Trunk]) and the Verizon IPCC service (from the manual configuration in this section).

Configure cluster\box:AuraSBC.adevc.avaya.globalipcom.com\interface eth2\ip outside\kernel-filter Help Index

Set Reset Back Delete

allow-rule

	allow-rule	admin	destination-port	source-address/mask	source-port	protocol
Edit Delete	allow-rule allow-sip-udp-from-peer-1	enabled	5060	172.30.209.0/24	0	udp
Edit Delete	allow-rule allow-sip-udp-from-peer-2	enabled	5060	172.30.205.0/24	5072	udp

Add allow-rule

deny-rule

	deny-rule	admin	destination-port	source-address/mask	source-port	protocol
Edit Delete	deny-rule deny-all-sip	enabled	5060	0.0.0.0/0	0	all

Add deny-rule

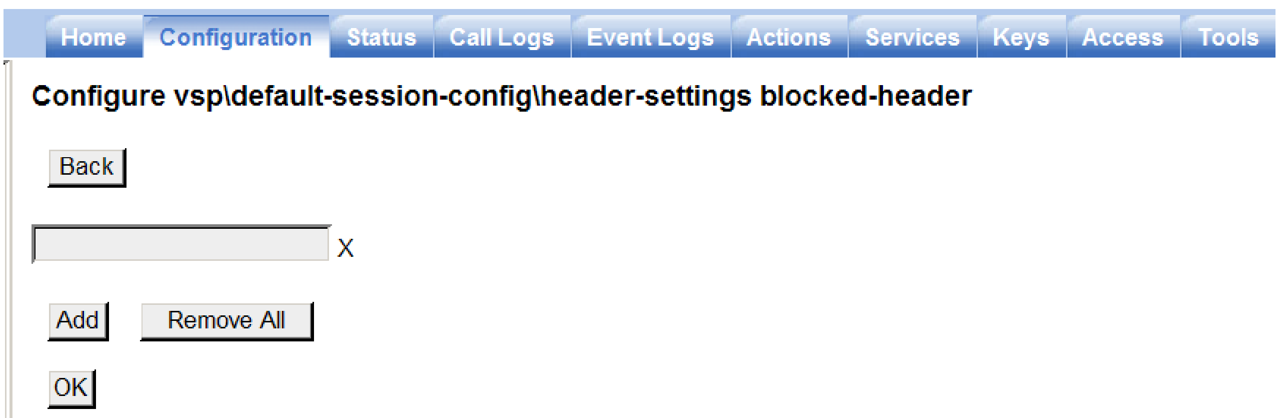
Proceed to save and activate the configuration as described in Section 7.4.

7.3.6 Stripping Unnecessary SIP Headers

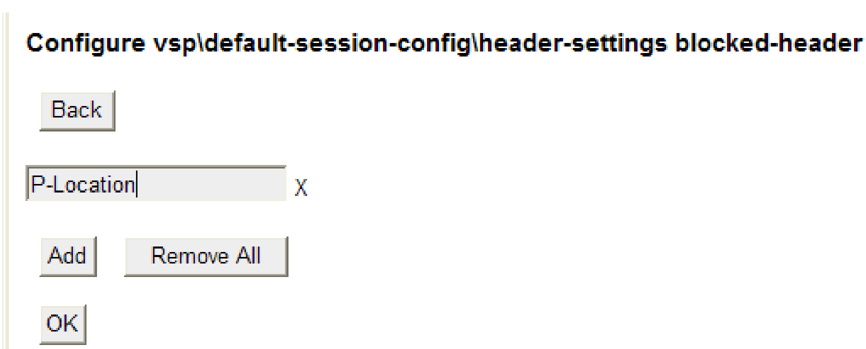
The SBC can be used to strip SIP headers that are not required or expected by Verizon. For headers that have relevance only within the enterprise, it may be desirable to prevent the header from being sent to the public SIP Service Provider. For example, Session Manager Release 6.1 may insert the P-Location header. The Avaya CS1000E may send the "x-nt-e164-clid", "x-nt-corr-id", and "Alert-Info" headers. While allowing these headers to be sent to Verizon does not cause any user-perceivable problem, the following procedures may be used to strip such headers that Verizon does not require.

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** or expand **default-session-config** and click on **header-settings**. Select the **blocked-header** link on the right.

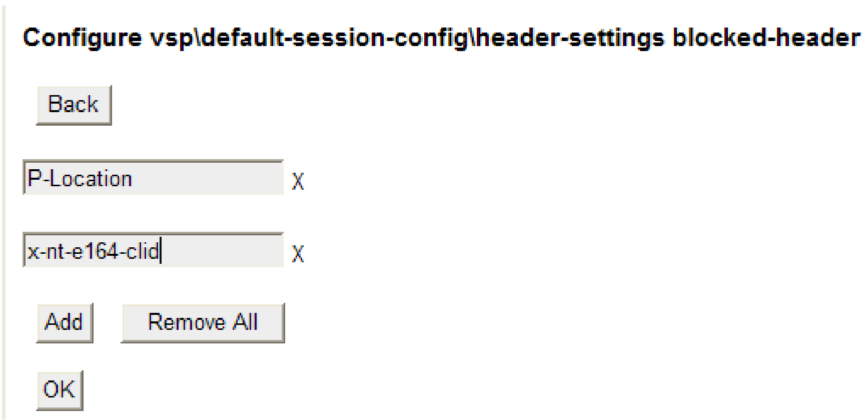
The following screen appears allowing configuration of the header to block.



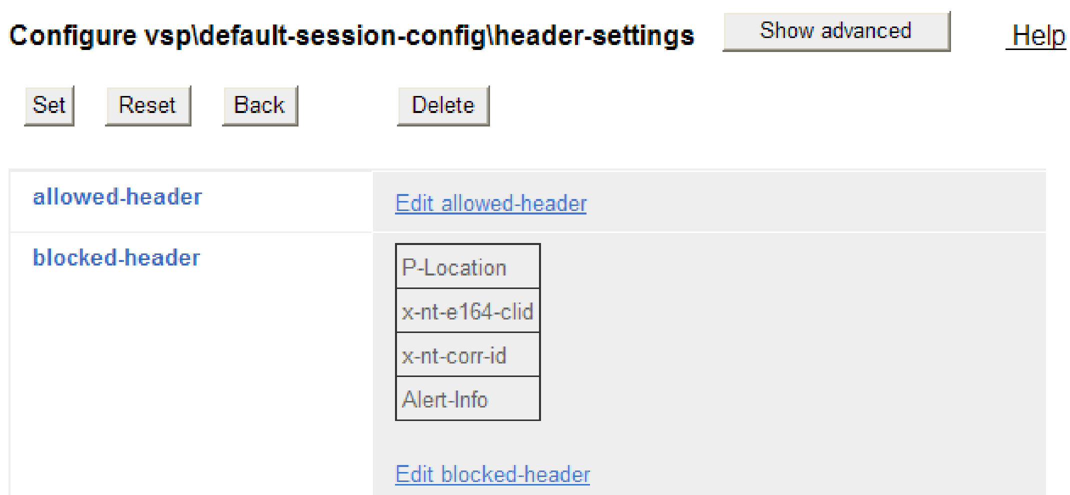
To block the P-Location header, enter "P-Location" and click **OK**, or click **Add** to configure more blocked headers.



The following screen shows the screen after the **Add** button was clicked and the addition blocked-header "x-nt-e164-clid" was entered.



Continue to add the desired blocked-headers in this fashion. When finished, click **OK** and **Set**. The following screen shows the blocked-headers used in the sample configuration.



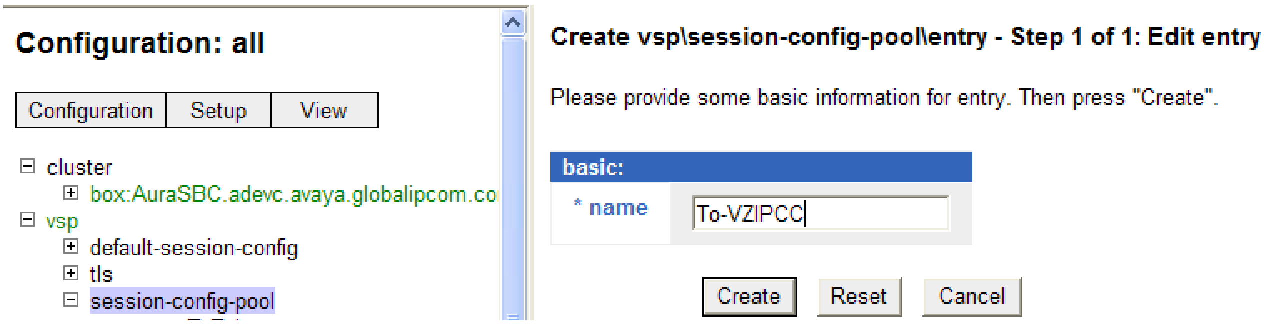
If the default-session-config does not apply, similar procedures can be used to strip headers in a more specific session-config-pool. For example, to strip the P-Location header in the session-config-pool “To-Telco”, navigate to **vsp → session-config-pool → entry ToTelco → header-settings**. In the resultant screen, click **Edit blocked-header** and proceed to add the P-Location and other blocked headers as described in this section.

Proceed to save and activate the configuration as described in Section 7.4.

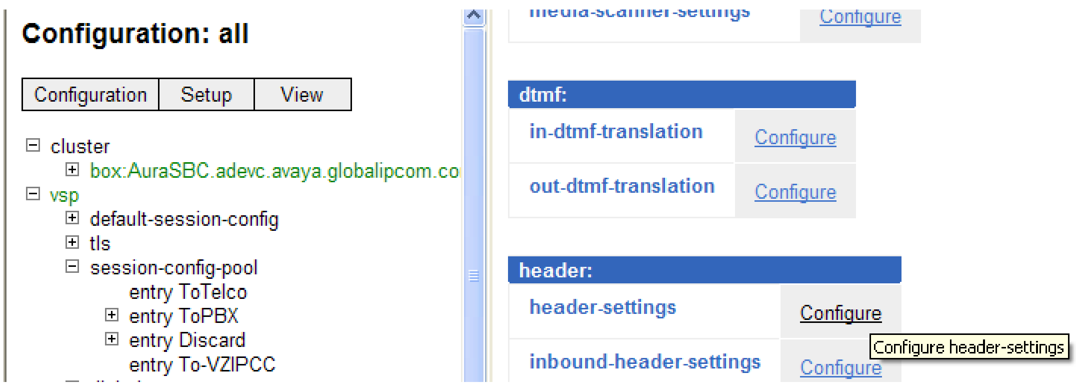
7.3.6.1 Stripping Diversion for IP Contact Center Only

The Verizon IP Contact Center service does not support receiving Diversion header. However, the Verizon IP Trunk service does support Diversion header and expects to receive Diversion header in specific scenarios, such as call forwarding to Verizon IP Trunk service. To ensure that a Diversion header is never sent to Verizon IP Contact Center, the following procedure may be followed. On the production circuit used for testing, this procedure was not required.

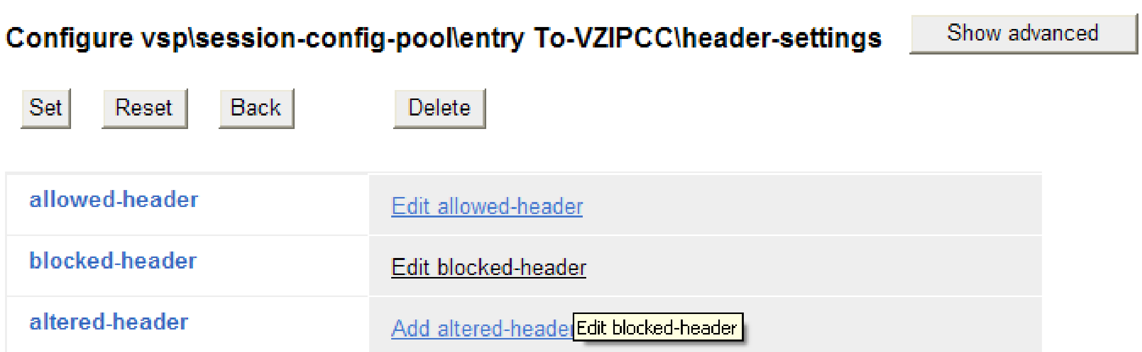
Navigate to **vsp → session-config-pool** as shown below on the left. In the right-hand side, enter a **name** for a new session-config-pool entry that will later be assigned for use by Verizon IPCC. Click **Create**.



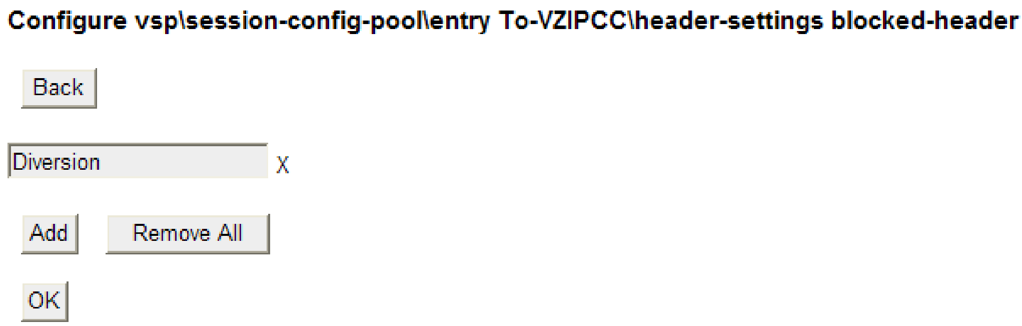
Navigate to the newly created "To-VZIPCC" session-config-pool entry, and scroll down on the right to the "header" area. Select **Configure** next to **header-settings** as shown below.



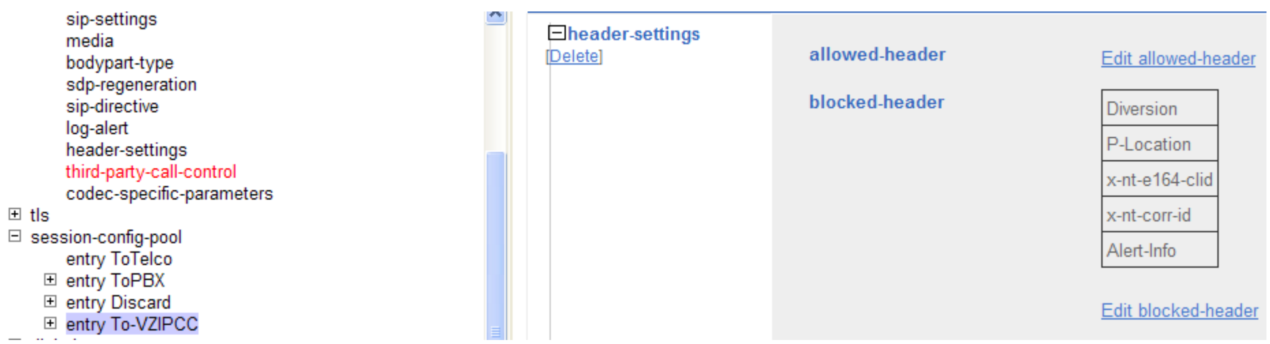
Select **Edit blocked-header** as shown below.



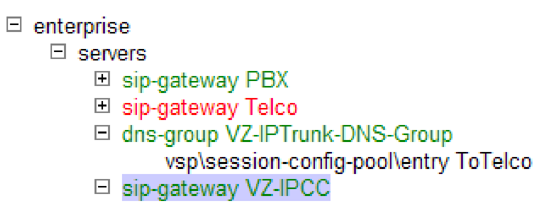
In the resultant screen, type in "Diversion" and click **Add**. Since this session-config-pool entry will be used instead of the default session-config-pool entry, any blocked-headers previously assigned to the default session-config-pool entry should also be added, as shown earlier in this section. After competing data entry for all blocked headers, click **OK** and **Set**.



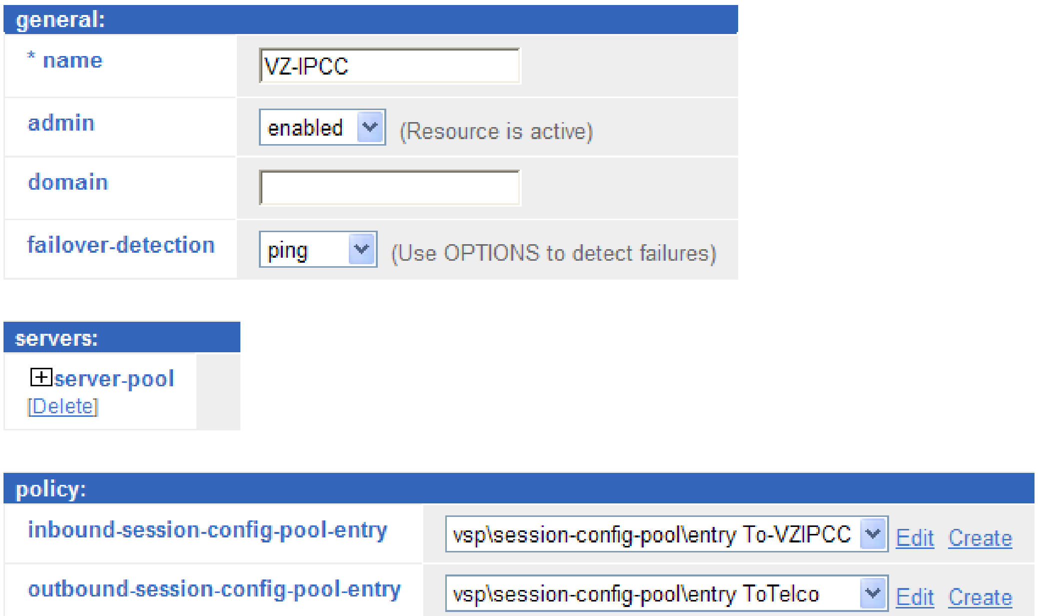
The following screen shows the list of blocked-headers assigned to the session-config-pool entry To-VZIPCC at the end of the procedure.



On the left menu, navigate to **vsp → enterprise → servers** and select the sip-gateway "VZ-IPCC" for the Verizon IP Contact Center service, as shown in the abridged screen below.



On the right, scroll down to the **policy** area. In the inbound-session-config-pool-entry, select the newly created session-config-pool that blocks Diversion header, as shown below.



Proceed to save and activate the configuration as described in Section 7.4.

7.3.7 Stripping Unnecessary SIP Message Body Information (Optional)

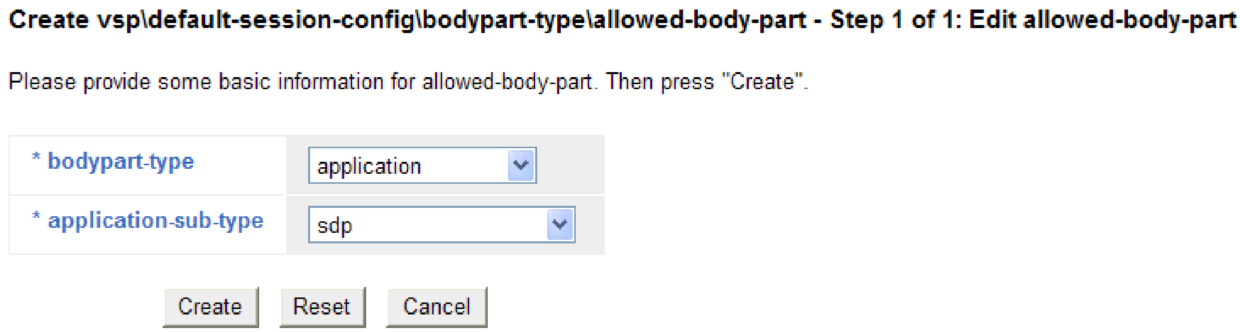
The procedures in this section are optional.

The SBC can be used to strip information from the message body that is not required or expected by Verizon. For example, if Verizon IP Trunk service will be used for outbound calls, the message body of an INVITE message sent from the Avaya CS1000E will contain a MIME Multipart message body containing the SDP information expected by Verizon, but also containing "x-nt-mcdn-frag-hex" and "x-nt-epid-frag-hex" application parts that are not processed by Verizon.   On the production circuit used for testing, Verizon was able to properly parse the Multipart MIME message body, and outgoing calls from the CS1000E to Verizon IP Trunk Service could be completed successfully without the configuration in this section. Nevertheless, since Verizon has no use for this information, the following procedures may be used to strip out unnecessary information and send only SDP in the message body to Verizon.

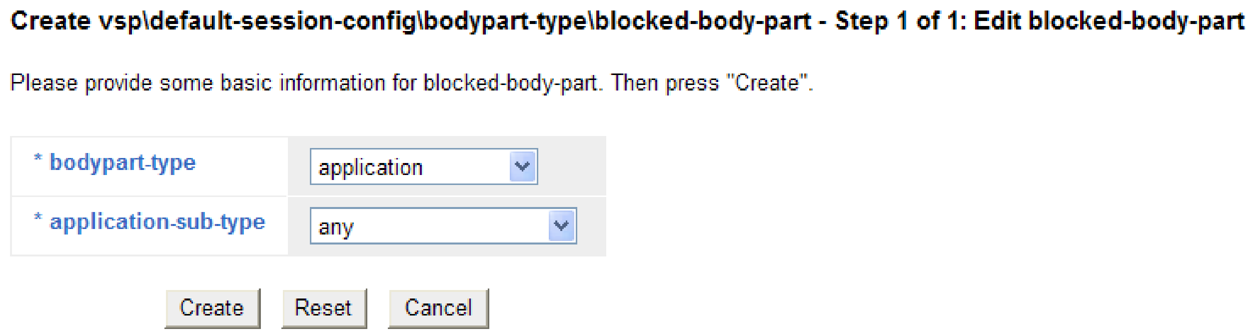
Two alternative approaches were tested successfully. In one approach, the SBC is used to specifically block the "x-nt-mcdn-frag-hex" and "x-nt-epid-frag-hex" parts.   In another approach, the SBC is used to block any body part that is not SDP.

7.3.7.1 Block Any body part but SDP Approach

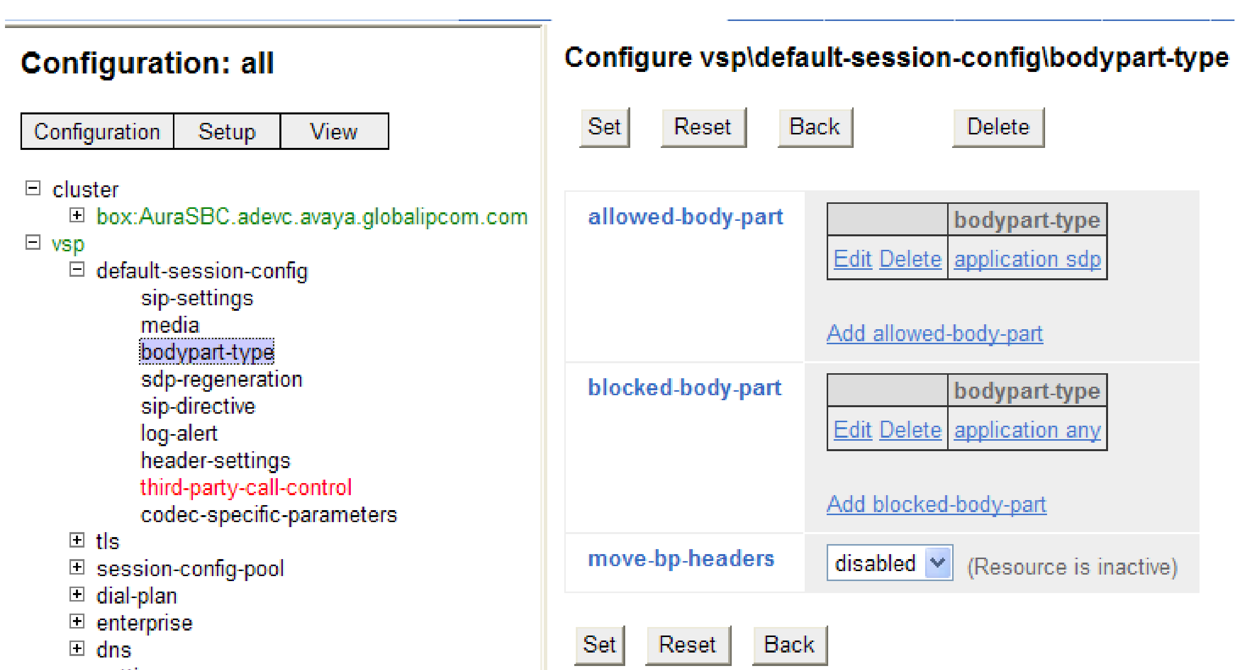
To block any body part but SDP, navigate to **vsp → default-session-config → bodypart-type**. Click **Add allowed-body-part**. In the **bodypart-type** drop-down menu, select "application". In the application-sub-type menu, select "sdp" as shown in the screen below. Click **Create**.



Then navigate to **vsp → default-session-config → bodypart-type**. Click **Add blocked-body-part**. In the **bodypart-type** drop-down menu, select “application”. In the application-sub-type menu, select “any” as shown in the screen below. Click **Create**.



The following screen shows the resulting configuration, where any application except SDP is blocked. Click **Set**.



Proceed to save and activate the configuration as described in Section 7.4.

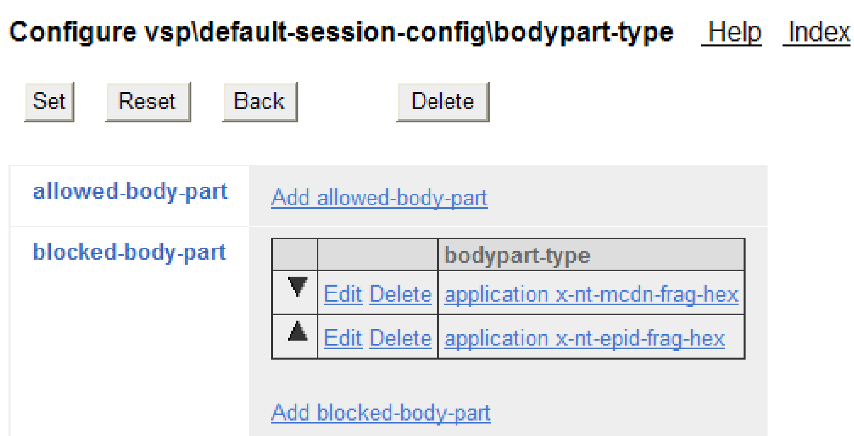
7.3.7.2 Block Specific Body Part Approach

This is an alternative to the approach documented in the previous sub-section. That is, it is shown as if the procedures in the prior section were not followed. In this section, the specific body parts that the CS1000E inserts in the message body are blocked rather than blocking anything but SDP.

Navigate to **vsp → default-session-config → bodypart-type**. Click **Add blocked-body-part**. In the **bodypart-type** drop-down menu, select “application”. In the application-sub-type menu, type in or select “x-nt-mcdn-frag-hex”. Click **Create**.

Again click **Add blocked-body-part.** In the **bodypart-type** drop-down menu, select “application”. In the application-sub-type menu, type in or select “x-nt-epid-frag-hex”. Click **Create**.

The following screen shows the resulting configuration. Click **Set**.

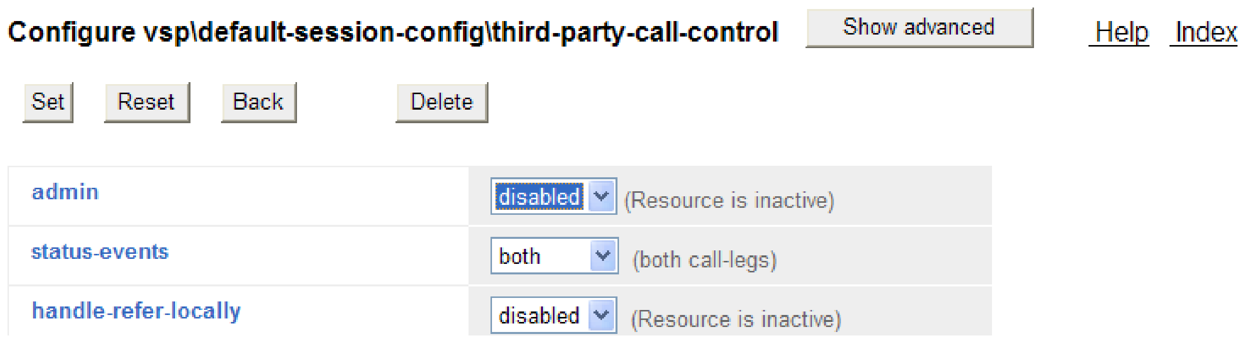


Proceed to save and activate the configuration as described in Section 7.4.

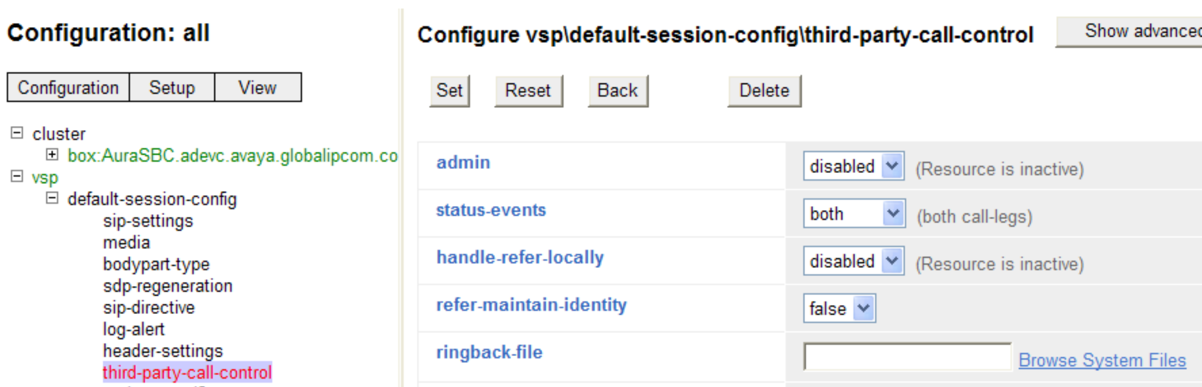
7.3.8 Disable Third Party Call Control

The installation wizard for the Avaya Aura® SBC in the release documented in these Application Notes will enable the **admin** field for third party call control. However, with third party call control enabled, the SBC is not able to properly reformat the message body with only the SDP information as described in the previous section. See Section 2.2 of Reference [AuraSBC-IP-Trunk].

To disable third party call control, navigate to **vsp → default-session-config → third-party-call-control**. 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.

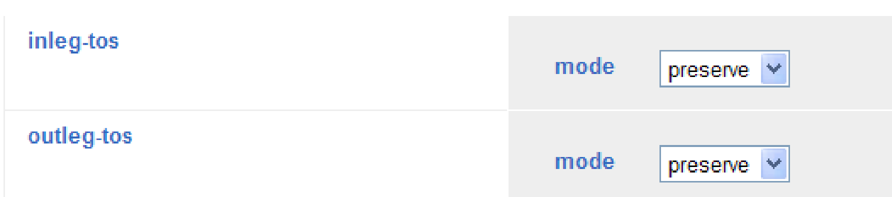


Proceed to save and activate the configuration as described in Section 7.4.

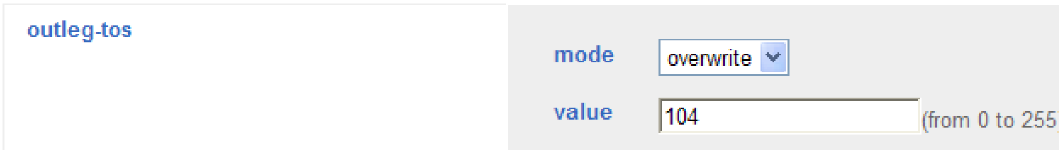
7.3.9 Quality Of Service (QoS) Markings for SIP Signaling

The procedure in this section is optional. The procedure can be used to achieve SIP signaling re-marking using the Avaya Aura® SBC.

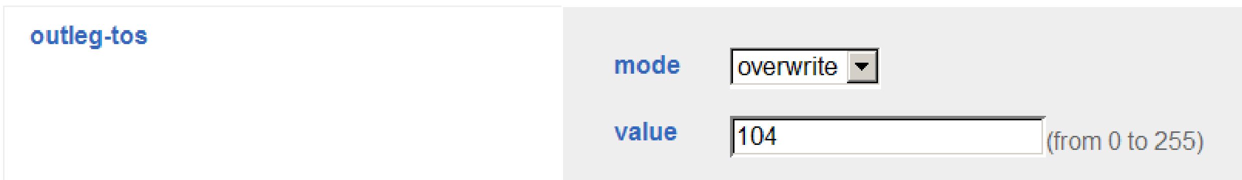
The default QoS behavior after using the installation wizard will be to preserve the TOS values. That is, the TOS value received from the private side of the SBC will be transmitted to Verizon on the public side of the SBC. For example, if Session Manager sends a SIP message to the SBC with a Differentiated Services Code Point (DSCP) value of 46, then the SBC will send the SIP message to Verizon with a DSCP of 46. To change this behavior, navigate to **vsp → session-config-pool → entry default-session-config → sip-settings** and scroll down to the **message-options** heading. The following portion of the screen shows the settings configured by the installation wizard, where the **inleg-tos** and **outleg-tos** are set to "preserve".



If it is desired to have the SBC re-mark SIP signaling to a different DSCP, the **inleg-tos** and **outleg-tos** parameters can be changed to desired DSCP values. For example, select "overwrite" from the **outleg-tos mode** drop-down menu.



In the **value** field that appears after selecting "overwrite", enter the decimal value corresponding to the byte containing the ToS field. For example, if the value is set to 104 (0x68) as shown above, the DSCP value 26 (0x1A) will be sent to Verizon (decoded by Wireshark as "Assured Forwarding 31").



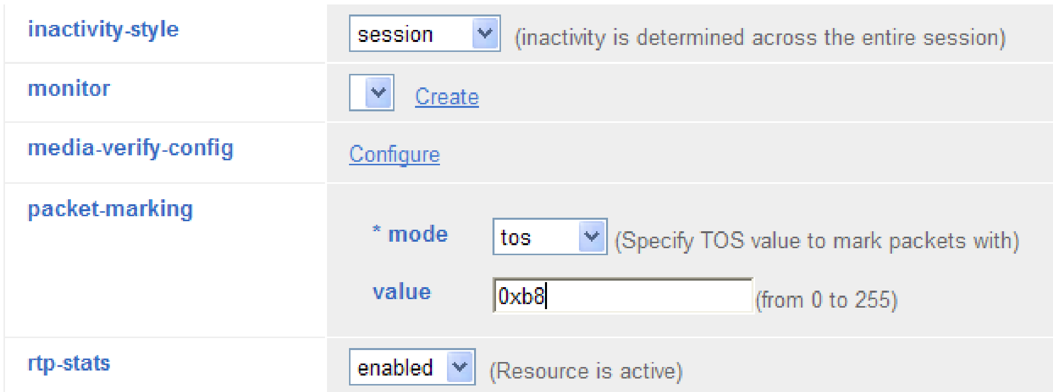
If desired, make the same change for the **inleg-tos**. Click the **Set** button. If DSCP value 28 (0x1C) is desired (decoded by Wireshark as "Assured Forwarding 32"), then the **value** field can be set to 112 instead.

Proceed to save and activate the configuration as described in Section 7.4.

7.3.10 Quality Of Service (QoS) Markings for Media

The procedure in this section is optional. If it is desired to have the SBC re-mark the DSCP in RTP media packets, navigate to **vsp → default-session-config → media**. Scroll down on the right until the **packet-marking** section is visible. The following screen shows the relevant area.

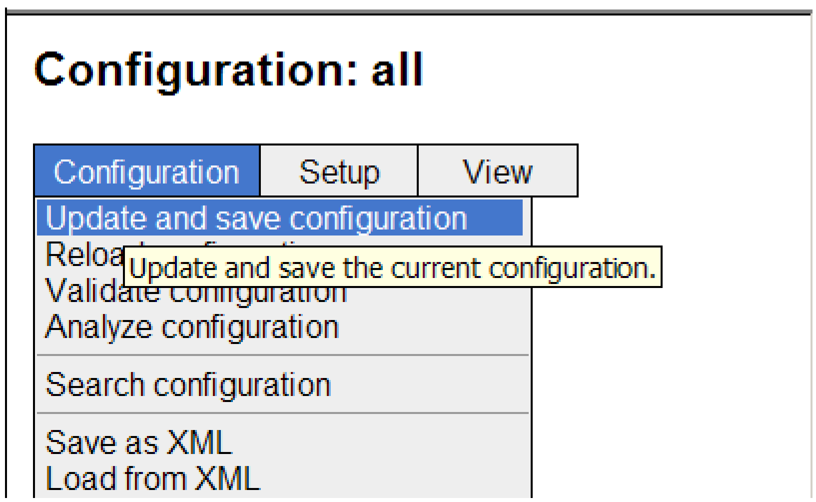
Set the **packet-marking mode** to **"**tos**"**. In the **value** field, enter the desired value of the ToS byte. The following screen uses the **value** "0xb8". With this configuration, RTP packets flowing to Verizon will contain DSCP 0x2e (decoded by Wireshark as "Expedited Forwarding").



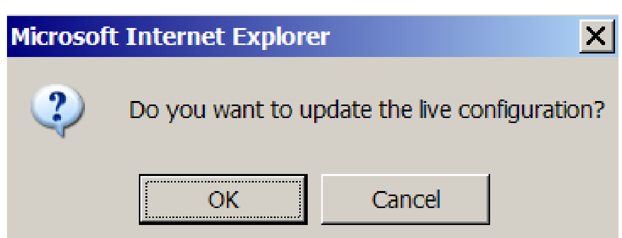
Proceed to save and activate the configuration as described in Section 7.4.

7.4. Saving and Activating Configuration Changes

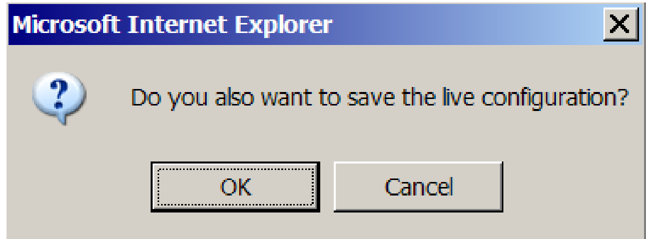
To save and activate configuration changes, select **Configuration → Update and save configuration** from the upper left hand side of the user interface, as shown below.



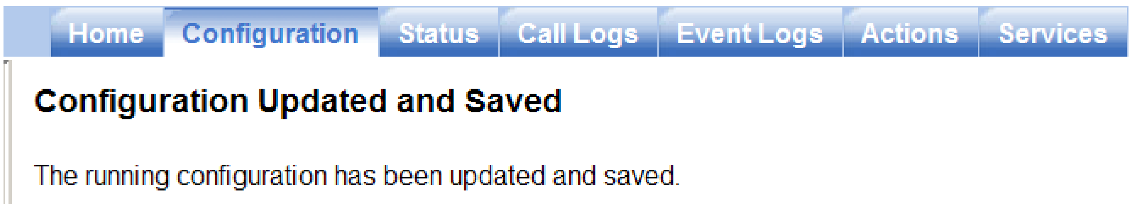
Click **OK** to update the live configuration.



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



A screen that includes the following should appear.



7.5. Example Configuration File

The configuration changes made by the installation wizard and the element manager OS-E GUI result in a text configuration file. A copy of the configuration file can be retrieved from the SBC by selecting the **Tools** tab and selecting **Download saved configuration file** from the left-side menu. An example configuration file resulting from the configuration in Section 7 is included below. This file includes configuration for Verizon IP Trunk service (from Reference [AuraSBC-IP-Trunk]) as well as configuration for Verizon IPCC service.

```
#
#  Copyright (c) 2004-2010  Acme Packet Inc.
#  All Rights Reserved.
#
#  File: /cxc/cxc.cfg
#  Date: 10:11:00 Wed 2011-05-18
#
config cluster
 config box 1
  set hostname AuraSBC.adevc.avaya.globalipcom.com
  set timezone America/New_York
  set name AuraSBC.adevc.avaya.globalipcom.com
  set identifier 00:ca:fe:69:46:13
  config interface eth0
   config ip inside
    set ip-address static 65.206.67.93/24
    config ssh
    return
    config snmp
     set trap-target 65.206.67.92 162
     set trap-filter generic
     set trap-filter dos
     set trap-filter sip
     set trap-filter system
    return
    config web
```

```
    return
    config web-service
     set protocol https 8443
     set authentication certificate "vsp\tls\certificate ws-cert"
    return
    config sip
     set udp-port 5060 "" "" any 0
     set tcp-port 5060 "" "" any 0
     set tls-port 5061 "" "" TLS 0 "vsp\tls\certificate aasbc.p12"
    return
    config icmp
    return
    config media-ports
    return
    config routing
     config route Default
      set gateway 65.206.67.254
     return
     config route Static0
      set destination network 192.11.13.4/30
      set gateway 65.206.67.91
     return
     config route Static1
      set admin disabled
     return
     config route Static2
      set admin disabled
     return
     config route Static3
      set admin disabled
     return
     config route Static4
      set admin disabled
     return
     config route Static5
      set admin disabled
     return
     config route Static6
      set admin disabled
     return
     config route Static7
      set admin disabled
     return
    return
    config dns-server
    return
   return
  return
  config interface eth2
   config ip outside
    set ip-address static 1.1.1.2/24
    config sip
     set udp-port 5060 "" "" any 0
    return
    config media-ports
    return
    config routing
```

```
      config route Default
       set admin disabled
      return
      config route external-sip-media-1
       set destination network 172.30.209.0/24
       set gateway 1.1.1.1
      return
      config route VZ-IPCC-network
       set destination network 172.30.205.0/24
       set gateway 1.1.1.1
      return
     return
     config kernel-filter
      config allow-rule allow-sip-udp-from-peer-2
       set destination-port 5060
       set source-address/mask 172.30.205.0/24
       set source-port 5072
       set protocol udp
      return
      config allow-rule allow-sip-udp-from-peer-1
       set destination-port 5060
       set source-address/mask 172.30.209.0/24
       set protocol udp
      return
      config deny-rule deny-all-sip
       set destination-port 5060
      return
     return
    return
   return
   config cli
    set prompt AuraSBC.adevc.avaya.globalipcom.com
   return
  return
return

config services
 config event-log
  config file access
   set filter access info
   set count 3
  return
  config file system
   set filter system info
   set count 3
  return
  config file errorlog
   set filter all error
   set count 3
  return
  config file db
   set filter db debug
   set filter dosDatabase info
   set count 3
  return
  config file management
   set filter management info
```

```
   set count 3
  return
  config file peer
   set filter sipSvr info
   set count 3
  return
  config file dos
   set filter dos alert
   set filter dosSip alert
   set filter dosTransport alert
   set filter dosUrl alert
   set count 3
  return
  config file krnlsys
   set filter krnlsys debug
   set count 3
  return
 return
return

config master-services
 config database
  set media enabled
 return
return

config vsp
 set admin enabled
 config default-session-config
  config sip-settings
   set inleg-tos overwrite 104
   set outleg-tos overwrite 104
  return
  config media
   set anchor enabled
   set packet-marking tos 0xb8
   set rtp-stats enabled
  return
  config bodypart-type
   set allowed-body-part application sdp
   set blocked-body-part application any
  return
  config sdp-regeneration
   set regenerate disabled
   set name Aura-SBC
  return
  config sip-directive
   set directive allow
  return
  config log-alert
   set apply-to-methods-for-filtered-logs
  return
  config header-settings
   set blocked-header P-Location
   set blocked-header x-nt-e164-clid
   set blocked-header x-nt-corr-id
   set blocked-header Alert-Info
```

```
  return
  config third-party-call-control
   set handle-refer-locally disabled
  return
  config codec-specific-parameters
  return
 return
 config tls
  config default-ca
   set ca-file /cxc/certs/sipca.pem
  return
  config certificate ws-cert
   set certificate-file /cxc/certs/ws.cert
  return
  config certificate aasbc.p12
   set certificate-file /cxc/certs/aasbc.p12
   set passphrase-tag aasbc-cert-tag
  return
 return
 config session-config-pool
  config entry ToTelco
  return
  config entry ToPBX
   config to-uri-specification
    set host next-hop-domain
   return
   config request-uri-specification
    set host next-hop-domain
   return
  return
  config entry Discard
   config sip-directive
   return
  return
  config entry To-VZIPCC
   config header-settings
    set blocked-header Diversion
    set blocked-header P-Location
    set blocked-header x-nt-e164-clid
    set blocked-header x-nt-corr-id
    set blocked-header Alert-Info
   return
  return
 return
 config dial-plan
  config route Default
   set priority 500
   set location-match-preferred exclusive
   set session-config vsp\session-config-pool\entry Discard
  return
  config source-route FromTelco
   set peer server "vsp\enterprise\servers\sip-gateway PBX"
   set source-match server "vsp\enterprise\servers\dns-group VZ-IPTrunk-DNS-
Group"
  return
  config source-route FromPBX
   set peer server "vsp\enterprise\servers\dns-group VZ-IPTrunk-DNS-Group"
```

```
   set source-match server "vsp\enterprise\servers\sip-gateway PBX"
  return
  config source-route FromVZIPCC
   set peer server "vsp\enterprise\servers\sip-gateway PBX"
   set source-match server "vsp\enterprise\servers\sip-gateway VZ-IPCC"
  return
  config source-route FromPBXtoVZIPCC
   set peer server "vsp\enterprise\servers\sip-gateway VZ-IPCC"
   set source-match server "vsp\enterprise\servers\sip-gateway PBX"
  return
 return
 config enterprise
  config servers
   config sip-gateway PBX
    set domain adevc.avaya.globalipcom.com
    set failover-detection ping
    set outbound-session-config-pool-entry vsp\session-config-pool\entry ToPBX
    config server-pool
     config server PBX1
      set host 10.1.2.210
      set transport TCP
     return
    return
   return
   config sip-gateway Telco
    set admin disabled
    set failover-detection ping
    set outbound-session-config-pool-entry vsp\session-config-pool\entry
ToTelco
    config server-pool
     config server Telco1
      set host 172.30.209.21
      set port 5071
     return
    return
   return
   config dns-group VZ-IPTrunk-DNS-Group
    set domain pcelban0001.avayalincroft.globalipcom.com
    set failover-detection ping
    set outbound-session-config-pool-entry vsp\session-config-pool\entry
ToTelco
   return
   config sip-gateway VZ-IPCC
    set failover-detection ping
    set ping-interval 30
    set inbound-session-config-pool-entry vsp\session-config-pool\entry To-
VZIPCC
    set outbound-session-config-pool-entry vsp\session-config-pool\entry
ToTelco
    config server-pool
     config server VZ-IPCC-network
      set host 172.30.205.55
      set port 5072
     return
    return
   return
  return
```

```
 return
 config dns
  config resolver
   config server 172.30.209.4
    set name VZ-IPTrunk-DNS
   return
  return
 return
 config settings
  set read-header-max 8191
 return
return

config external-services
return

config preferences
 config gui-preferences
 return
return

config access
 config permissions superuser
  set cli advanced
 return
 config permissions read-only
  set config view
  set actions disabled
 return
 config users
  config user admin
   set password 0x00e9a8385c963a64b97c9efd745dca47a89d67a2ff039b08cefbbe8c6b
   set permissions access\permissions superuser
  return
  config user cust
   set password 0x0077cc723ccd18d052a3ce58a8f47712d1c49d99963a7b8086a554d15e
   set permissions access\permissions read-only
  return
  config user init
   set password 0x00527bc64d625298d3d82aecb06b5b82d74e6c74a212e7d7783276bd46
   set permissions access\permissions superuser
  return
  config user craft
   set password 0x00623332bf3f6d7069f443dcdc98b8d4aa67bb3d4e3bebed35fd2f09f8
   set permissions access\permissions superuser
  return
  config user dadmin
   set password 0x00f6240b8d3a025fdf273432b58036947f461243c56717ec8379432867
   set permissions access\permissions read-only
  return
 return
return

config features
return
```

8. Verizon Business IP Contact Center 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.

8.1. 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 IPCC Network
adevc.avaya.globalipcom.com ***UDP port 5060***	***172.30.205.55*** ***UDP Port 5072***

8.2. Numbers Assigned by Verizon

Verizon will provide IP Toll Free VoIP Inbound numbers and/or Verizon IP-IVR numbers as part of service provisioning. **Table 2** and **Table 3** in Section 3 show the Verizon-provided IP Toll Free and IP-IVR numbers used in the sample configuration.

9. Verification Steps

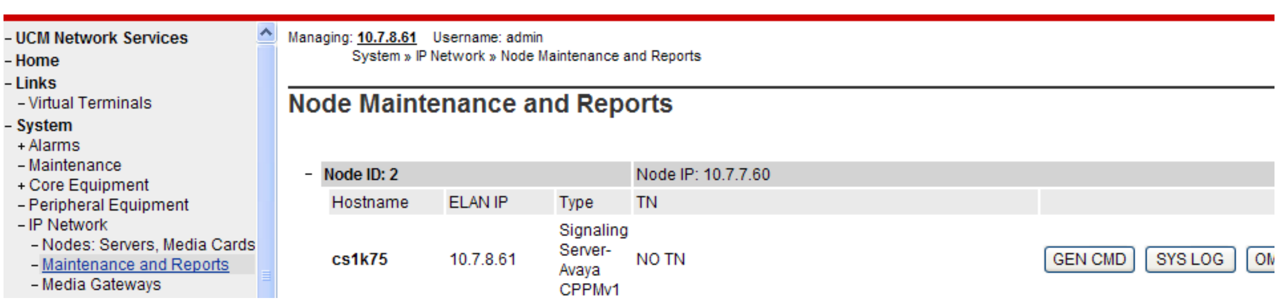
This section provides example verifications of the Avaya configuration with Verizon Business IP Contact Center service.

9.1. Avaya Communication Server 1000E Verifications

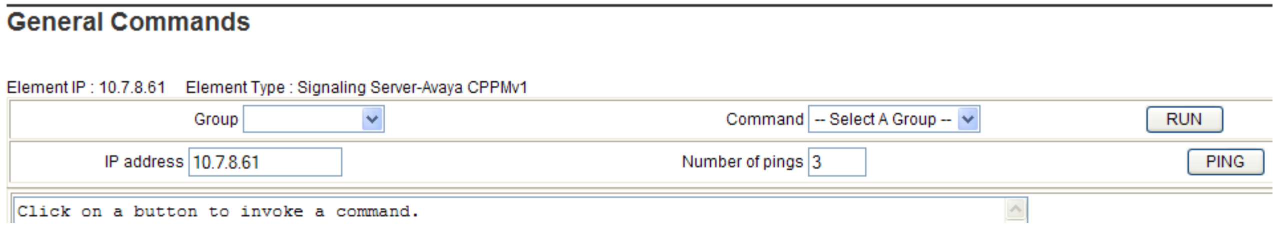
This section illustrates sample verifications that may be performed using the Avaya CS1000E Element Manager GUI. Additional commands are illustrated in Section 9.1 of [AuraSBC-IP-Trunk].

9.1.1 IP Network Maintenance and Reports Commands

From Element Manager, navigate to **System → IP Network → Maintenance and Reports** as shown below. In the resultant screen on the right, click the **Gen CMD** button.

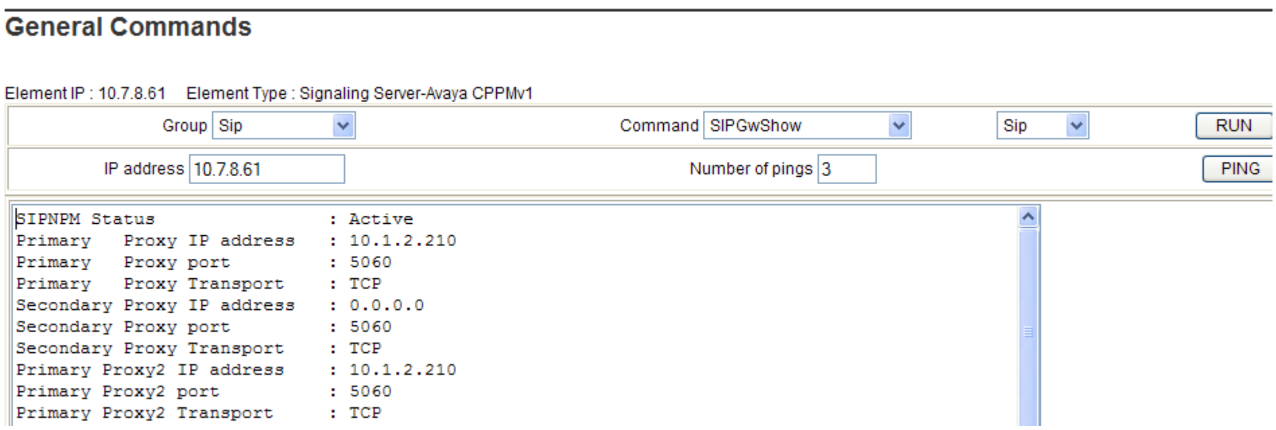


The **General Commands** page is displayed as shown below.

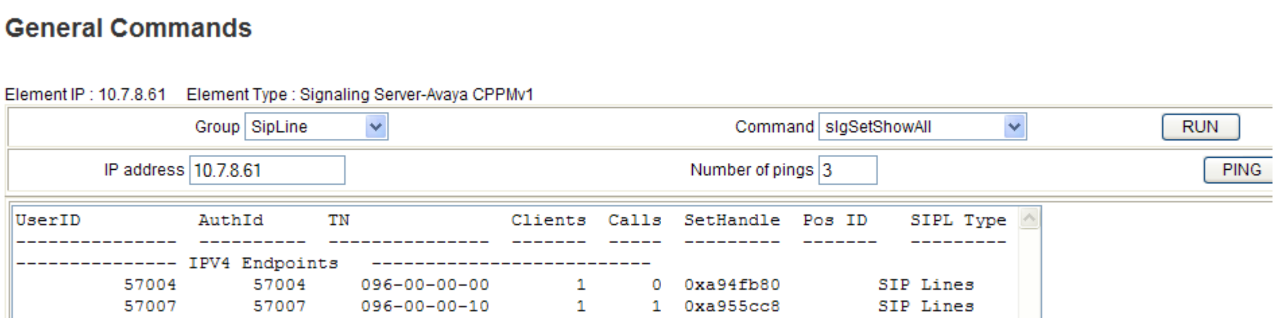


A variety of commands are available by selecting an appropriate **Group** and **Command** from the drop-down menus, and selecting **Run**.

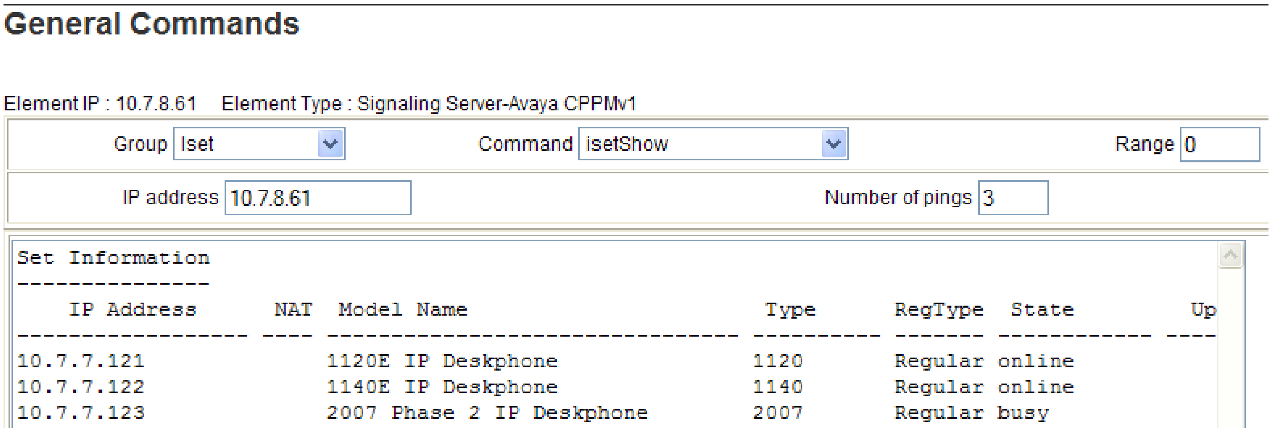
To check the status of the SIP Gateway to Session Manager in the sample configuration, select "Sip" from the **Group** menu and "SIPGwShow" from the **Command** menu. Click **Run**. The example output below shows that the Session Manager (10.1.2.210, port 5060, TCP) has "SIPNPM Status" Active.



The following screen shows a means to view registered SIP telephones. The screen shows the output of the **Command** "sigSetShowAll" in **Group** "SipLine". At the time this screen was captured, the SIP telephone with DN 57007 was involved in an active call.



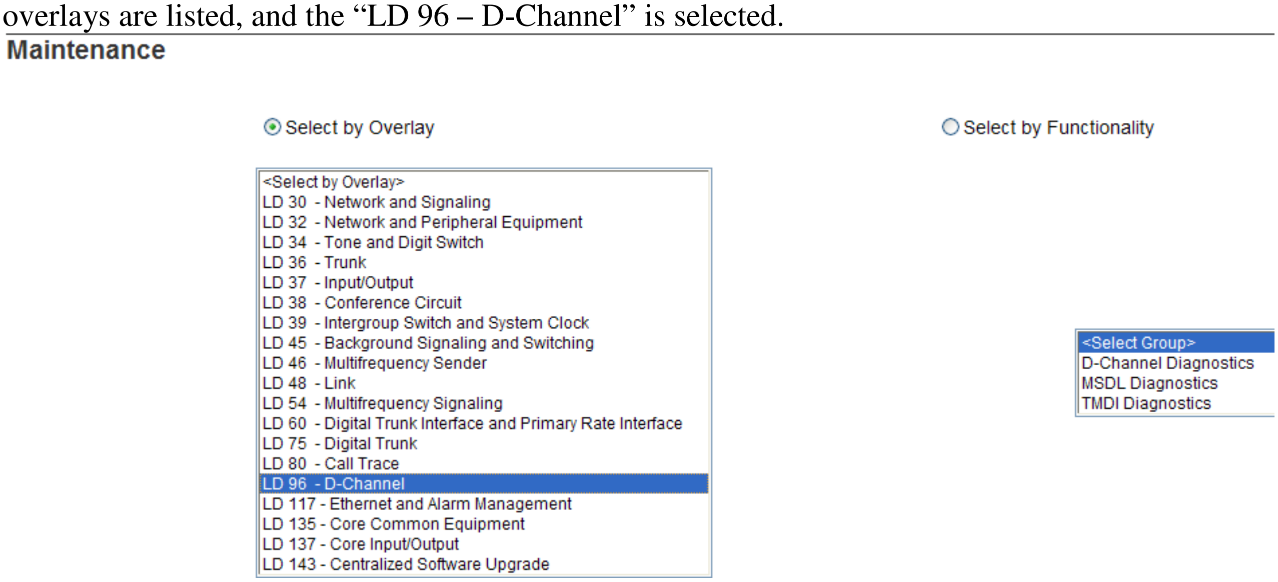
The following screen shows a means to view IP UNIStim telephones. The screen shows the output of the **Command** "isetShow" in **Group** "Iset". At the time this screen was captured, the "2007 Phase 2 IP Deskphone" UNIStim telephone was involved in an active call.



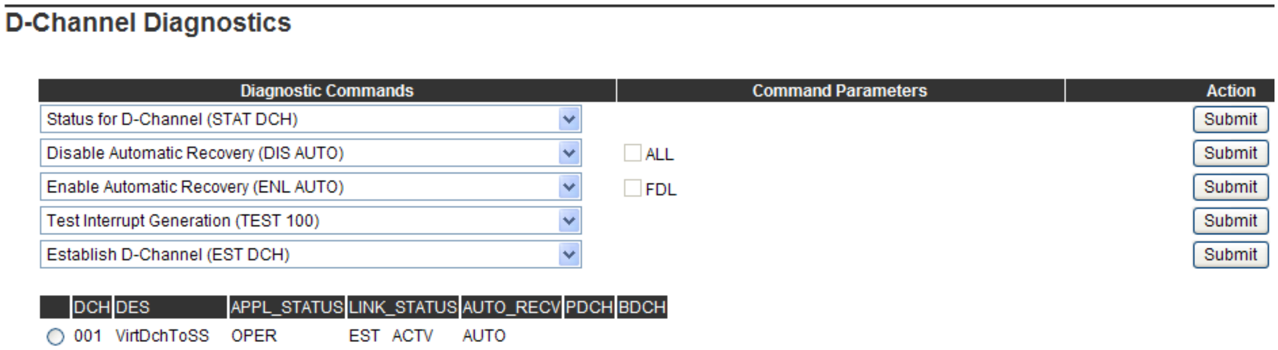
9.1.2 System Maintenance Commands

A variety of system maintenance commands are available by navigating to **System → Maintenance** using Element Manager. The user can navigate the maintenance commands using either the "Select by Overlay" approach or the "Select by Functionality" approach.

The following screen shows an example where "Select by Overlay" has been chosen. The various overlays are listed, and the "LD 96 – D-Channel" is selected.



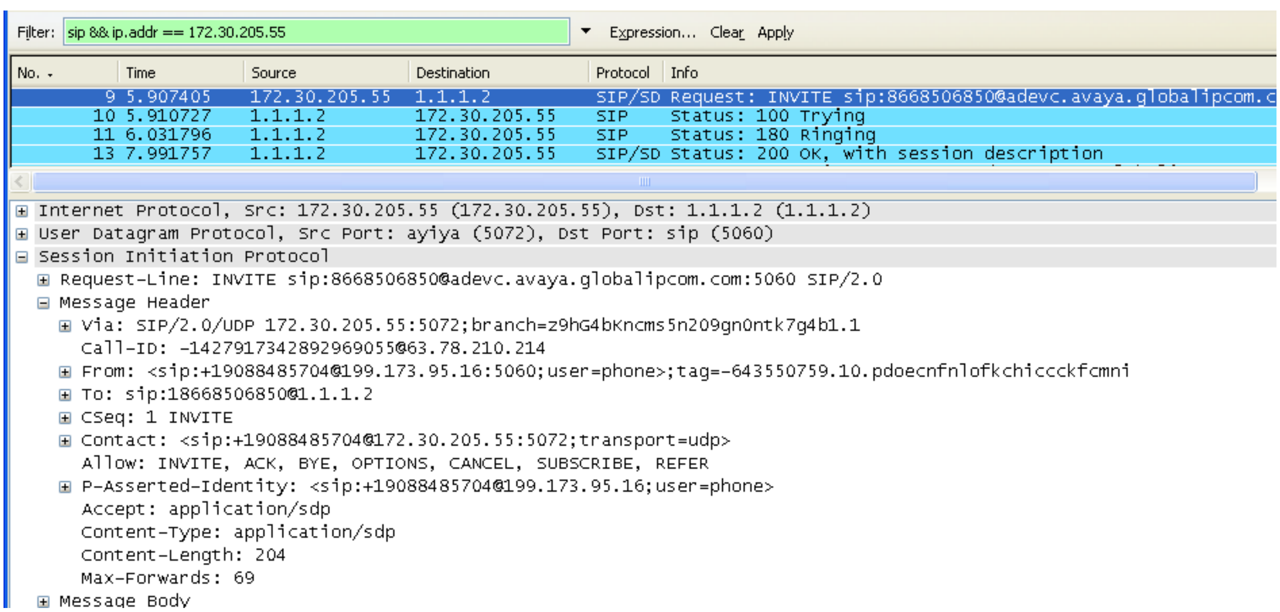
On the preceding screen, if **D-Channel Diagnostics** is selected on the right, a screen such as the following is displayed. D-Channel number 1, which is used in the sample configuration, is established "EST" and active "ACTV".



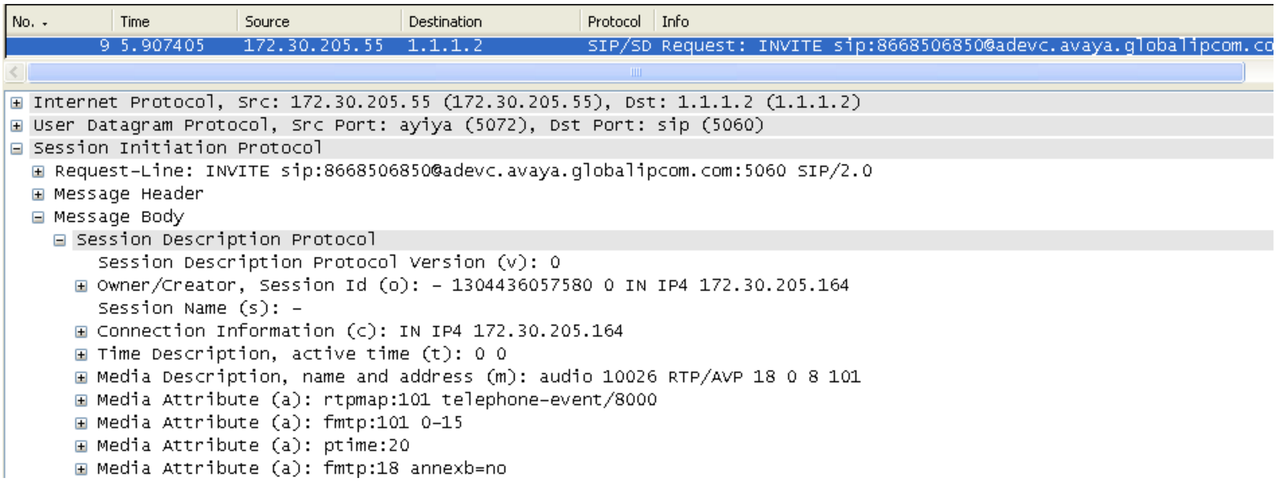
9.2. Wireshark Verification

This section illustrates Wireshark traces for an inbound IP Toll-Free VoIP Inbound call using the sample configuration. The PSTN telephone 908-848-5704 dials the IP Toll Free number 866-850-6850.

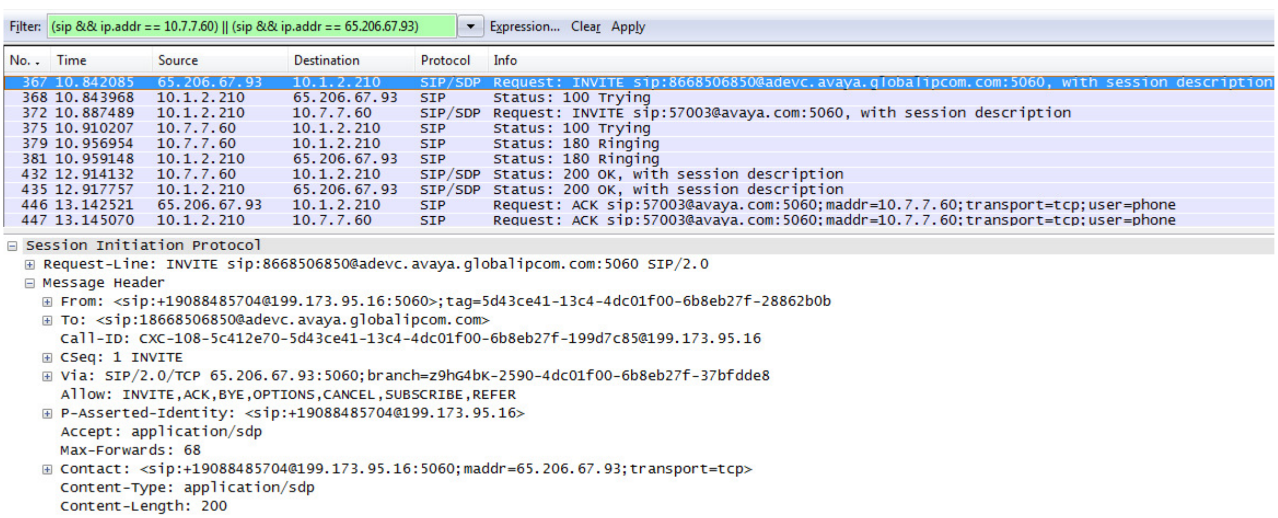
The following screen shows a Wireshark trace taken from the outside of the SBC. The use of UDP and port 5072 can be observed. The INVITE from Verizon in frame 9 is selected and expanded to illustrate the contents of the message header. The Request-URI contains the dialed IP Toll Free number 8668506850. Note that Verizon prefixes the calling party number 9088485704 in the From and PAI headers with +1. The overall flow for an incoming call can be observed. In frame 11, a 180 Ringing (without SDP) response is sent to Verizon. In frame 13, the 200 OK with SDP answering the inbound call is sent to Verizon.



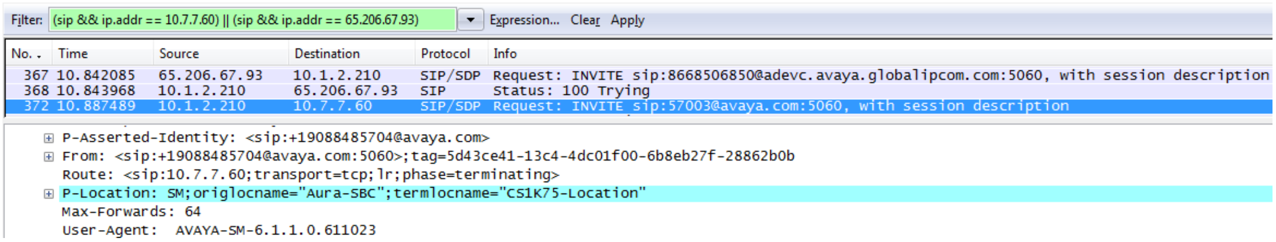
The following screen shows the same Wireshark trace and message, focusing on the message body of the INVITE in frame 9. Note that the Verizon SDP offer lists G.729 first, followed by G.711. The value "101" is specified for "DTMF" using RFC 2833.



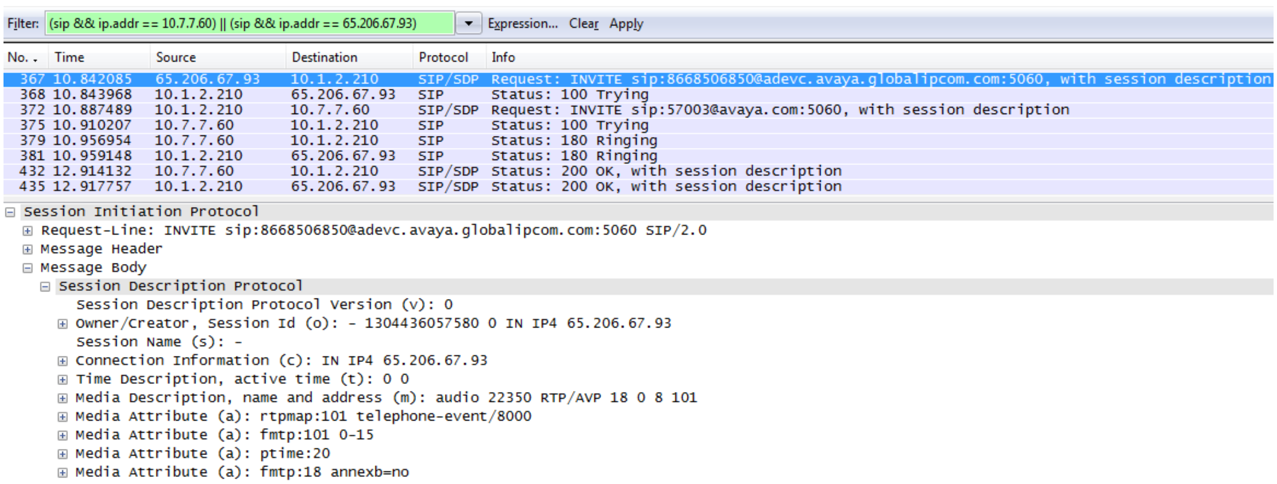
The following screen shows a filtered Wireshark trace taken from the inside of the SBC. The INVITE sent from the SBC to Session Manager in frame 367 is selected and expanded to illustrate the message headers. The toll-free number 8668506850 can be observed in the Request-URI along with the domain "adevc.avaya.globalipcom.com". Session Manager will use the Request-URI for routing. The From and PAI headers contain the calling PSTN telephone number 908-848-5704, prefixed with "+1". This screen also shows the SIP messaging through answer. In frame 379, the CS1000E sends 180 Ringing (without SDP) when alerting the telephone, and in frame 432, the CS1000E sends the 200 OK with SDP when the call is answered.



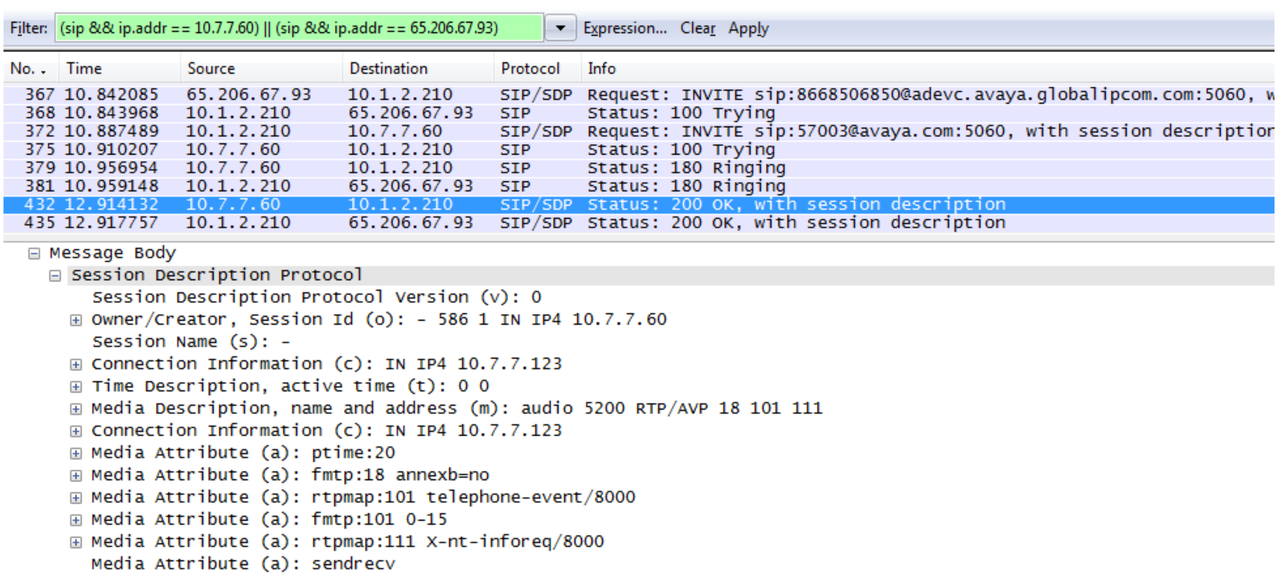
The following screen shows the same filtered Wireshark trace. The INVITE sent from Session Manager to the CS1000E in Frame 372 is selected and expanded to illustrate select message headers. From the selected frame (in blue), it can be observed that the toll-free number 8668506850 in the original Request-URI has been adapted by Session Manager to the CS1000E Directory Number 57003. The original domain "adevc.avaya.globalipcom.com" has been adapted to "avaya.com" in the Request-URI, From, and To headers by this same Session Manager adapter.



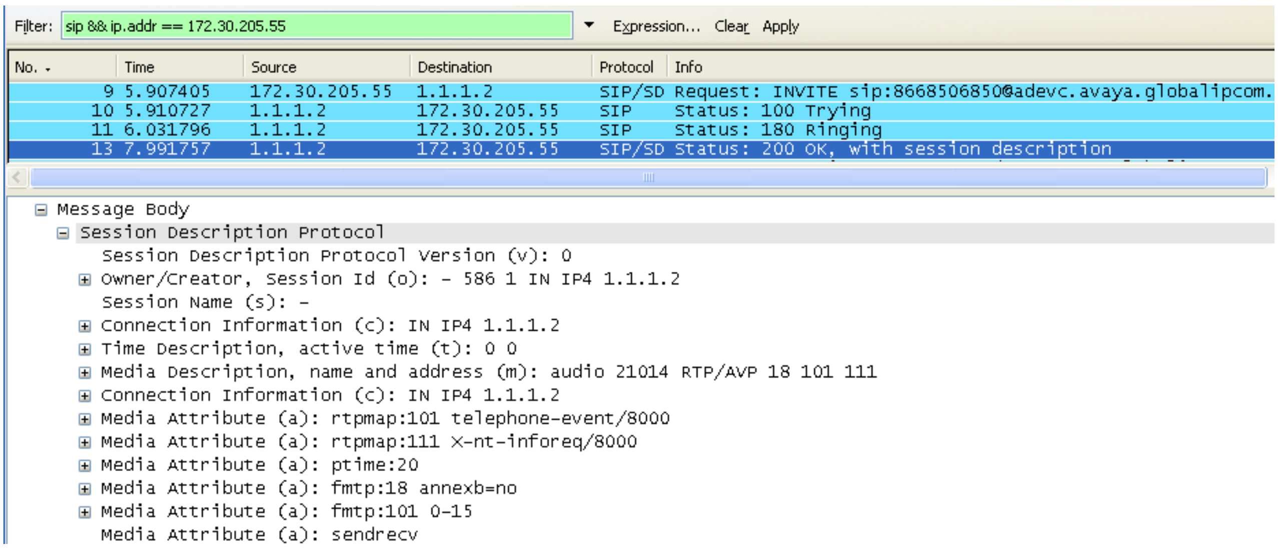
The following screens show the same Wireshark trace, focused on the message bodies. The SDP offer in the INVITE in frame 367 is expanded in the screen below. Note that inbound IP Toll-Free VoIP Inbound calls prefer G.729A (18, annexb=no) and allow G.711 (0). For "DTMF" events, the value "101" is used. The IP Address (65.206.67.93) is the inside private IP address of the SBC.



In the next screen, the 200 OK sent by the CS1000E in frame 432 is expanded to illustrate the SDP answer. The CS1000E answers with G.729A (18, annexb=no), and the IP Address (10.7.7.123) is the IP Address of the answering IP UNIStim telephone. The use of RFC 2833 value 101 for "DTMF" can be observed.



The following screen shows a portion of the 200 OK sent to Verizon from the outside of the SBC, focusing on the SDP in the message body. The mapping of the IP Address in the connection information to the outside IP address of the SBC (1.1.1.2) can be observed. The SDP answer contains G.729A (i.e., 18, with annexb=no) and uses "101" for "DTMF" telephone events using RFC2833.



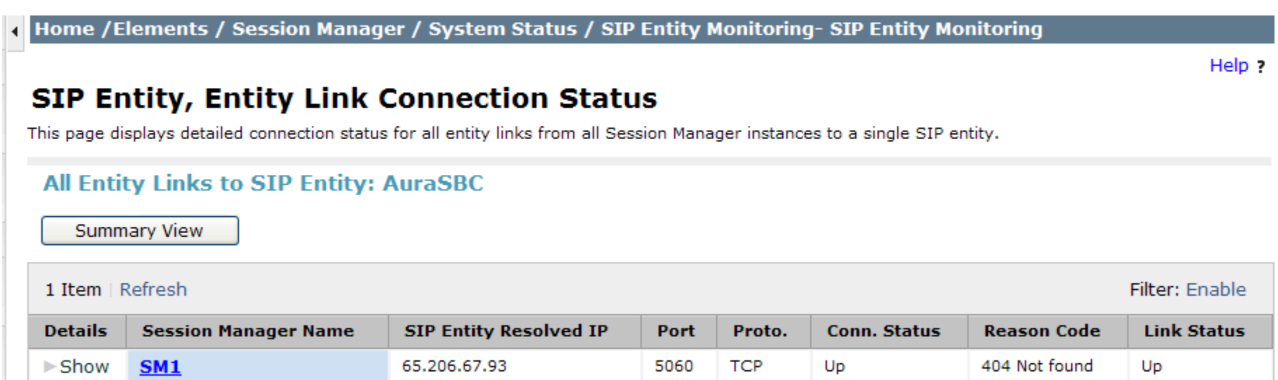
9.3. System Manager and Session Manager Verification

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

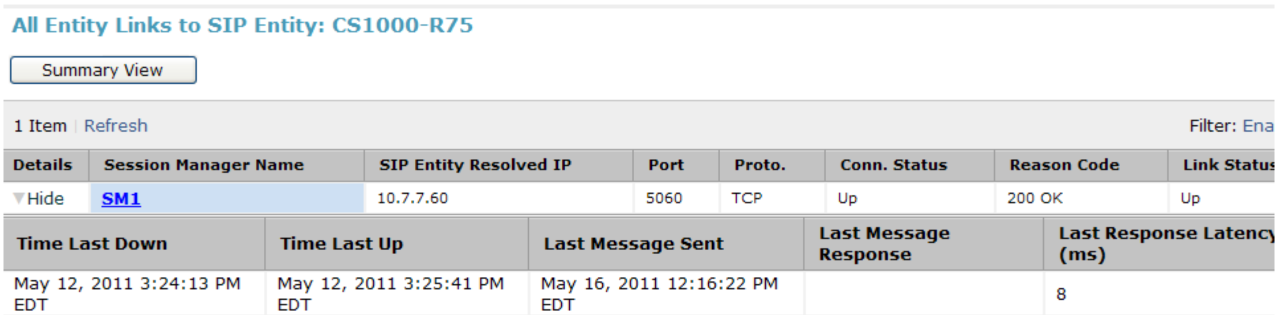
9.3.1 Verify SIP Entity Link Status

Log in to System Manager. Expand **Elements → Session Manager → System Status → SIP Entity Monitoring**.

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. The **Reason Code** column indicates that the SBC has responded to SIP OPTIONS from Session Manager with a SIP 404 message, which is sufficient for SIP Link Monitoring to consider the link up.



Return to the list of monitored entities, and select another entity of interest, such as "CS1000-R75". 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.


All Entity Links to SIP Entity: CS1000-R75

Summary View

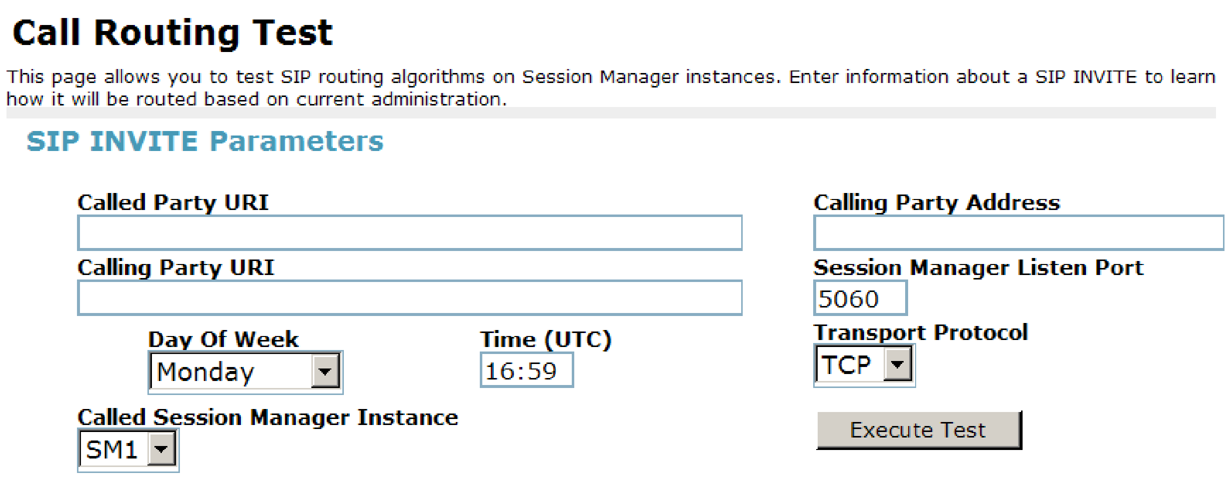
1 Item | Refresh Filter: Ena

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Hide	SM1	10.7.7.60	5060	TCP	Up	200 OK	Up

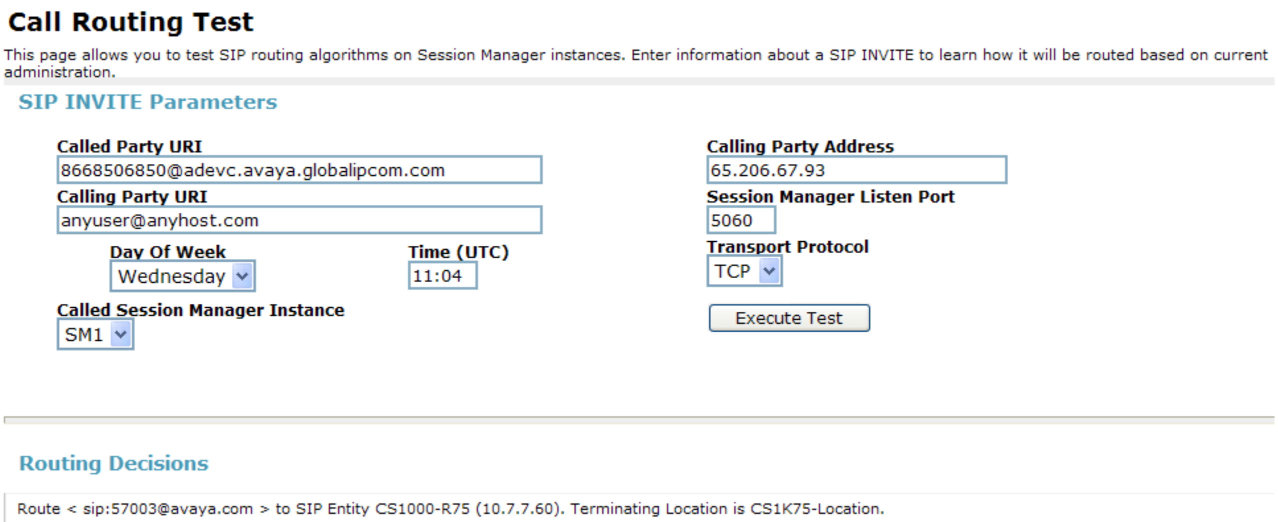
Time Last Down	Time Last Up	Last Message Sent	Last Message Response	Last Response Latency (ms)
May 12, 2011 3:24:13 PM EDT	May 12, 2011 3:25:41 PM EDT	May 16, 2011 12:16:22 PM EDT		8

9.3.2 Call Routing Test

The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, expand **Elements → Session Manager → System Tools → Call Routing Test**. A screen such as the following is displayed.



As an example, the following screen shows a call routing test for an inbound IP Toll Free call from the PSTN to the enterprise, arriving via the Avaya Aura® SBC. Under **Routing Decisions**, observe that the call will route to the CS1000E (10.7.7.60) using the SIP entity named "CS1000-R75". The user part of the Request-URI is adapted from the Verizon IP Toll Free number "8668506850" to the CS1000E Directory Number 57003. The host part of the Request-URI is adapted from the enterprise domain known to Verizon "adevc.avaya.globalipcom.com" to the domain "avaya.com" configured for the shared Avaya Solution and Interoperability Lab test network. Scroll down to inspect the details of the **Routing Decision Process** if desired (not shown).



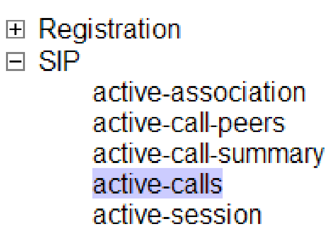
9.4. Avaya Aura® Session Border Controller Verification

This section contains verification steps that may be performed using the Avaya Aura® Session Border Controller. Section 9.4 of reference [AuraSBC-IP-Trunk] contains additional methods of verification.

9.4.1 Status Tab

A wealth of status information is available via the **Status** tab. This section provides several examples of status information that may be obtained. Select the **Status** tab as shown below.

There is a **SIP** heading on the left that can be expanded as shown in the abridged screen below.



In the example screen below, **active-calls** was selected from the left, revealing details about an active incoming call on the right. A scroll bar allows viewing of information about the active call. The following screen was captured when an incoming Verizon IP Toll Free call was active.

active-calls - currently active calls

View: Basic Search

session-id	from	to	state	previous-hop-ip
0x04C2D405F816E193	<sip:+19088485704@199.173.94.16:5060;user=phone>;tag=-643550759.10.pdoecnfnpbgnicpjbmadmobk	<sip:18668506850@1.1.1.2>	B2B_CONNECTED	172.30.205.55

Additional information about the call is available by continuing to scroll right, as shown below.

seconds Refresh

previous-hop-ip	next-hop-domain	duration (seconds)	inbound-connection	outbound-connection	header-value	subject-to-CAC	contact
172.30.205.55	avaya.com	128		65.206.67.93:4868-10.1.2.210:5060 TCP		true	<sip:+19088485704@172.30.205.55:5072;transport=udp>

Taken May 25, 2011 8:46:39 AM XML

The following screen shows an example screen output when **sdp-session-stats** was selected from the navigation menu on the left (not shown), and "Verbose" was selected for the **View**. This screen was captured when the same inbound Verizon IP Toll Free call was active. Observe that media is anchored at the SBC, and the codec in use is G.729.

sdp-session-stats - Active SDP session information

View: Verbose Search seconds Re

session-id	stream	stream-type	anchor-setting	anchor-state	num-answers	associated-session	sdp-state	on-hold	codecs
0x04C2D405F816E193	1	audio	enabled	anchored	1	0x00	answered	false	g729, telephone-event, inforeq

The following screen shows an example screen output when **media → media-ports-sessions** was selected from the navigation menu on the left (not shown) and "Verbose" was selected for the **View**. This screen was captured when the same inbound Verizon IP Toll Free call was active.

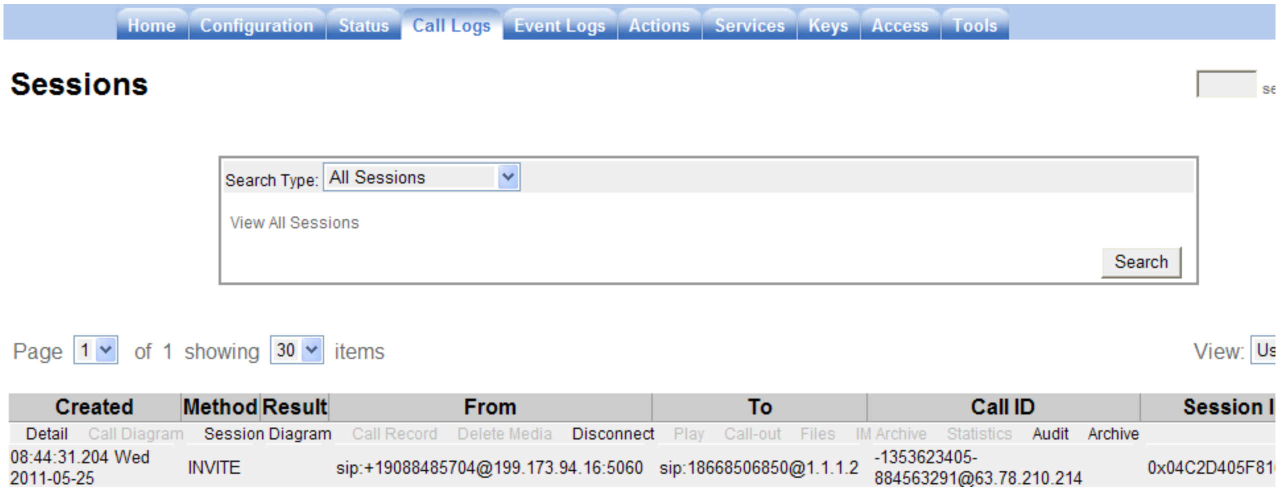
media-ports-sessions - Addresses used by media stream sessions

View: Verbose Search seconds Refresh

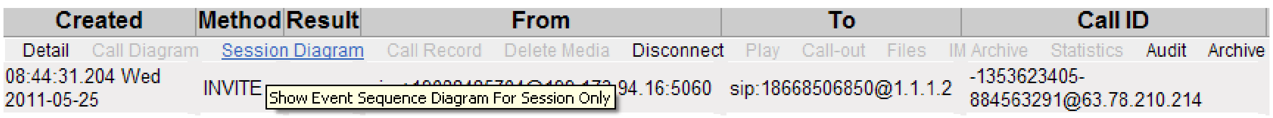
	ip-address	port	session-id	call-leg	anchor-state
	1.1.1.2	22124	0x04C2D405F816E193	1	anchored
	65.206.67.93	21190	0x04C2D405F816E193	2	anchored

9.4.2 Call Logs

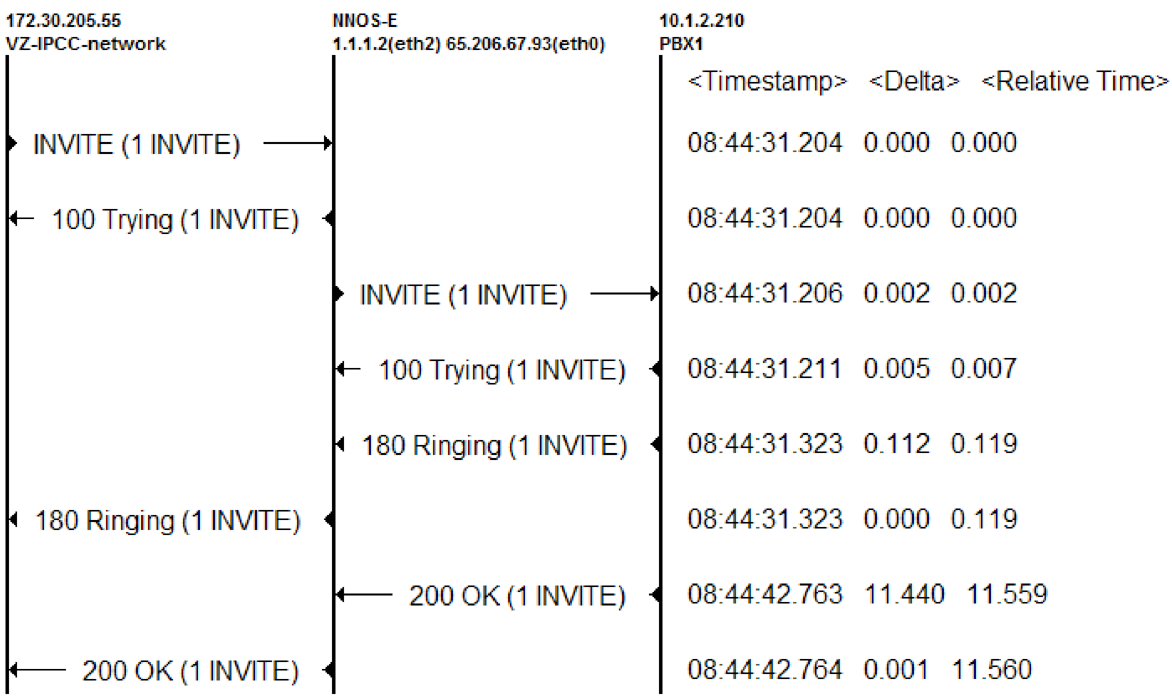
The **Call Logs** tab can provide useful diagnostic or troubleshooting information. The following screen shows a portion of the **Call Logs** tab selected after making an inbound Verizon IP Toll Free call.



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 Verizon IP Toll Free call. Note that the activity for both the inside private and outside public side of the SBC can be seen. Scroll down (not shown) to see additional information for the session.



Select the **Back** button (not shown). 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® SBC for analysis.

10. Conclusion

As illustrated in these Application Notes, Avaya Communication Server Release 7.5, Avaya Aura® Session Manager 6.1, and the Avaya Aura® Session Border Controller Release 6 can be configured to interoperate successfully with Verizon Business IP Contact Center service. This solution enables callers on the PSTN to dial Verizon toll-free numbers to reach the Avaya Communication Server 1000E via the SIP protocol.

11. Additional References

This section references documentation relevant to these Applications.

11.1. Avaya

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

[1] *Administering Avaya Aura™ Session Manager*, Doc ID 03-603324, Issue 4, Feb 2011 available at http://support.avaya.com/css/P8/documents/100082630

[2] *Installing and Configuring Avaya Aura™ Session Manager,* Doc ID 03-603473 Issue 2, November 2010 available at http://support.avaya.com/css/P8/documents/100089152

[3] *Maintaining and Troubleshooting Avaya Aura™ Session Manager,* Doc ID 03-603325, Issue 3.1, March 2011 available at http://support.avaya.com/css/P8/documents/100089154

[4] *Administering Avaya Aura™ System Manager*, Document Number 03-603324, June 2010 available at http://support.avaya.com/css/P8/documents/100089681

Avaya Communication Server 1000E

1) IP Peer Networking Installation and Commissioning, Release 7.5, Document Number NN43001-313
2) Unified Communications Management Common Services Fundamentals, Avaya Communication Server 1000E Release 7.5, Document Number NN43001-116
3) Network Routing Service Fundamentals, Release 7.5, Document Number NN43001-130, Issue 03.02
4) Co-resident Call Server and Signaling Server Fundamentals, Avaya Communication Server 1000E Release 7.5, Document Number NN43001-509
5) Signaling Server and IP Line Fundamentals, Avaya Communication Server 1000E Release 7.5, Document Number NN43001-125

A variety of Avaya Application Notes on Verizon solutions tested via Avaya DevConnect are available at the following link:

http://devconnect.avaya.com/dc/Public/WebListings/v2/CompanyWebListing.aspx?CompanyId=1236

Reference [AuraSBC-IP-Trunk] below is a companion to these Application Notes. The document is among those available at the above link.

[AuraSBC-IP-Trunk] Application Notes for Avaya Communication Server 1000E Release 7.5, Avaya Aura® Session Manager 6.1, and Avaya Aura® Session Border Controller 6.0 with Verizon Business IP Trunk SIP Trunk Service
https://devconnect.avaya.com/public/download/dyn/CS1K75-VZIPT.pdf

11.2. Verizon Business

Information in the following Verizon documents was also used for these Application Notes. Contact a Verizon Business Account Representative for additional information.

- [VZ-Test-Plan] Verizon Business IPCC Interoperability Lab Test Plan, Revision 1.7
- [VZ-Spec] Verizon Business IPCC Trunk Interface Network Interface Specification, Document Version 2.2.1.9

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.