Avaya Solution & Interoperability Test Lab

Application Notes for SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service and Avaya IP Office Release 9.0 with Avaya Session Border Controller for Enterprise Release 6.2 – Issue 1.2

Abstract

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Trunk SIP Trunk Service Offer and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of Avaya Session Border Controller for Enterprise Release 6.2, an Avaya IP Office 500 V2 Release 9.0 Preferred Edition, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya SIP, H.323, digital, and analog endpoints.

These Application Notes complement previously published Application Notes by illustrating the configuration screens and Avaya testing of IP Office Release 9.0.

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

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

©2014 Avaya Inc. All Rights Reserved.

1. Introduction

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Trunk SIP Trunk Service Offer and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500 V2 Release 9.0 Preferred Edition; Avaya Voicemail Pro; Avaya IP Office Softphone; Avaya SIP, H.323, digital, and analog endpoints.

These Application Notes complement previously published Application Notes by illustrating the configuration screens and Avaya testing of IP Office Release 9.0.

Customers using Avaya IP Office with the Verizon Business IP Trunk SIP Trunk service are able to place and receive PSTN calls via the SIP protocol. The converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI.

In the sample configuration, An Avaya Session Border Controller for Enterprise (SBCE) is used as an edge device between the Avaya IP Office and Verizon business. The Avaya SBCE performs SIP header manipulation and provides topology hiding, as well as a variety of other functions providing security and the presentation of a standardized SIP interface.

Verizon Business IP Trunk service offer can be delivered to the customer premises via either a Private IP (PIP) or Internet Dedicated Access (IDA) IP network termination. Although the configuration documented in these Application Notes used Verizon's IP Trunk service terminated via a PIP network connection, the solution validated in this document also applies to IP Trunk services delivered via IDA service terminations.

For more information on the Verizon Business IP Trunk service, including access alternatives, visit http://www.verizonbusiness.com/us/products/voip/trunking/.

2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the Verizon Business IP Trunk service, as depicted in **Figure 1**. The Avaya SBCE and IP Office were configured to use the commercially available SIP Trunking solution provided by the Verizon Business IP Trunk SIP Trunk Service. This allowed Avaya IP Office users to make calls to the PSTN and receive calls from the PSTN via the Verizon Business IP Trunk SIP Trunk Service.

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

©2014 Avaya Inc. All Rights Reserved.

2.1. Interoperability Compliance Testing

To verify SIP trunking interoperability, the following features and functionality were covered during the interoperability compliance test:

- Incoming calls from the PSTN were routed to the DID numbers assigned by Verizon Business to the Avaya IP Office location. These incoming PSTN calls arrived via the SIP Line and were answered by Avaya SIP telephones, Avaya H.323 telephones, Avaya digital telephones, analog telephones, analog fax machines, Avaya IP Office Softphone, Avaya Flare® Experience and Avaya Voicemail Pro. The display of caller ID on display-equipped Avaya IP Office telephones was verified.
- Incoming calls answered by members of circular Hunt Groups were verified.
- Outgoing calls from the Avaya IP Office location to the PSTN were routed via the SIP Line to Verizon Business. These outgoing PSTN calls were originated from Avaya SIP telephones, Avaya H.323 telephones, Avaya digital telephones, analog endpoints, Avaya IP Office Softphone, Avaya Flare® Experience and Avaya Voicemail Pro. The display of caller ID on display-equipped PSTN telephones was verified.
- Inbound / Outbound fax using G.711 and T38 were verified.
- Proper disconnect when the caller abandoned a call before answer for both inbound and outbound calls.
- Proper disconnect when the IP Office party or the PSTN party terminated an active call.
- Proper busy tone heard when an IP Office user called a busy PSTN user, or a PSTN user called a busy IP Office user (i.e., if no redirection was configured for user busy conditions).
- Various outbound PSTN call types were tested including long distance, international, toll-free, operator assisted, and directory assistance calls.
- Requests for privacy (i.e., caller anonymity) for IP Office outbound calls to the PSTN were verified. That is, when privacy is requested by IP Office, outbound PSTN calls were successfully completed while withholding the caller ID from the displays of display-equipped PSTN telephones.
- Privacy requests for inbound calls from the PSTN to IP Office users were verified. That is, when privacy is requested by a PSTN caller, the inbound PSTN call was successfully completed to an IP Office user while presenting an “anonymous” display to the IP Office user.
- SIP OPTIONS monitoring of the health of the SIP trunk was verified. Both Verizon Business and IP Office were able to monitor SIP trunk health using SIP OPTIONS.
- IP Office outbound calls were placed with simple short codes as well as using ARS. Using ARS, the ability of IP Office to route-advance to an alternate route was exercised when the primary SIP line was not responding. The Line Group associated with the Verizon Business SIP Trunk was the primary line group chosen for a call, or an alternate line group was selected upon failure of a primary line.
- Incoming and outgoing calls using the G.729A and G.711MU codecs.
- DTMF transmission (RFC 2833) with successful voice mail navigation using G.729A and G.711MU for incoming and outgoing calls. Successful navigation of a simple auto-attendant application configured on Avaya Voicemail Pro.

©2014 Avaya Inc. All Rights Reserved.

- Inbound and outbound long holding time call stability.
- Telephony features such as call waiting, hold, transfer, and conference.
- Inbound calls from Verizon IP Trunk service that were call forwarded back to PSTN destinations, presenting true calling party information to the PSTN phone, via Verizon IP Trunk service.
- Mobile twinning to a mobile phone, presenting true calling party information to the mobile phone. Outbound mobile call control was also verified successfully (e.g., using DTMF on a twinned call to place new calls and create a conference via a mobile phone).
- DiffServ markings in accordance with network requirements for IP Office SIP signaling and RTP media.
- Mobility Features such as Mobile Callback and Mobile Call Control.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results. The following observations were noted.

1. **HOLD:** When a call is put on hold by an IP Office user, there is no indication sent via SIP messaging to Verizon. This is transparent to the users on the call.

2. **SIP PHONE TRANSFER:** When transferring an outbound call to the PSTN via Verizon using a SIP device registered to IP Office (e.g., Avaya 1140E or IP Softphone in the sample configuration), and the REFER transfer option is enabled on the SIP Line to Verizon, the IP Office user may briefly see the display "Transfer failed" after the final user operation, even if the transfer has actually succeeded. On the production circuit used for testing, Verizon did not send NOTIFY messages to IP Office to signal transfer completion.

3. **DTMF:** DTMF transmission from an Avaya 1140E SIP phone placed during an outbound PSTN call produces duplicate DTMF tones to the far end. For example, if the user presses 1234, the tones produced are 11223344. This occurs when VoIP Silence Suppression is checked on the VoIP tab of the SIP Extension form. With this setting unchecked, the DTMF tones play out correctly. See **Section 5.5.2**. This anomaly is under investigation by the IP Office product team as IPOFFICE-52193.

©2014 Avaya Inc. All Rights Reserved.

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 IP Trunk service offer, visit the online support site at http://www.verizonbusiness.com/us/customer/.

3. Reference Configuration

Figure 1 illustrates an example Avaya IP Office solution connected to the Verizon Business IP Trunk SIP Trunk Service. The Avaya equipment is located on a private IP subnet. An enterprise edge router provides access to the Verizon Business IP Trunk service network via a Verizon Business T1 circuit. This circuit is provisioned for the Verizon Business Private IP (PIP) service.

In the sample configuration, the Avaya SBCE receives traffic from the Verizon Business IP Trunk service on port 5060. The Avaya SBCE uses DNS SRV, using UDP for transport, to determine the IP Address and port to be used to send SIP signaling to Verizon. In the sample configuration, the DNS process will result in SIP signaling being sent to IP Address 172.30.209.21 and port 5071. As shown in **Table 1**, the Verizon Business IP Trunk service provided Direct Inward Dial (DID) numbers. These DID numbers were mapped to IP Office destinations via Incoming Call Routes in the IP Office configuration.

Verizon Business used FQDN *pcelban0001.avayalincroft.globalipcom.com*. The Avaya CPE environment was assigned FQDN *adevc.avaya.globalipcom.com* by Verizon Business.

©2014 Avaya Inc. All Rights Reserved.

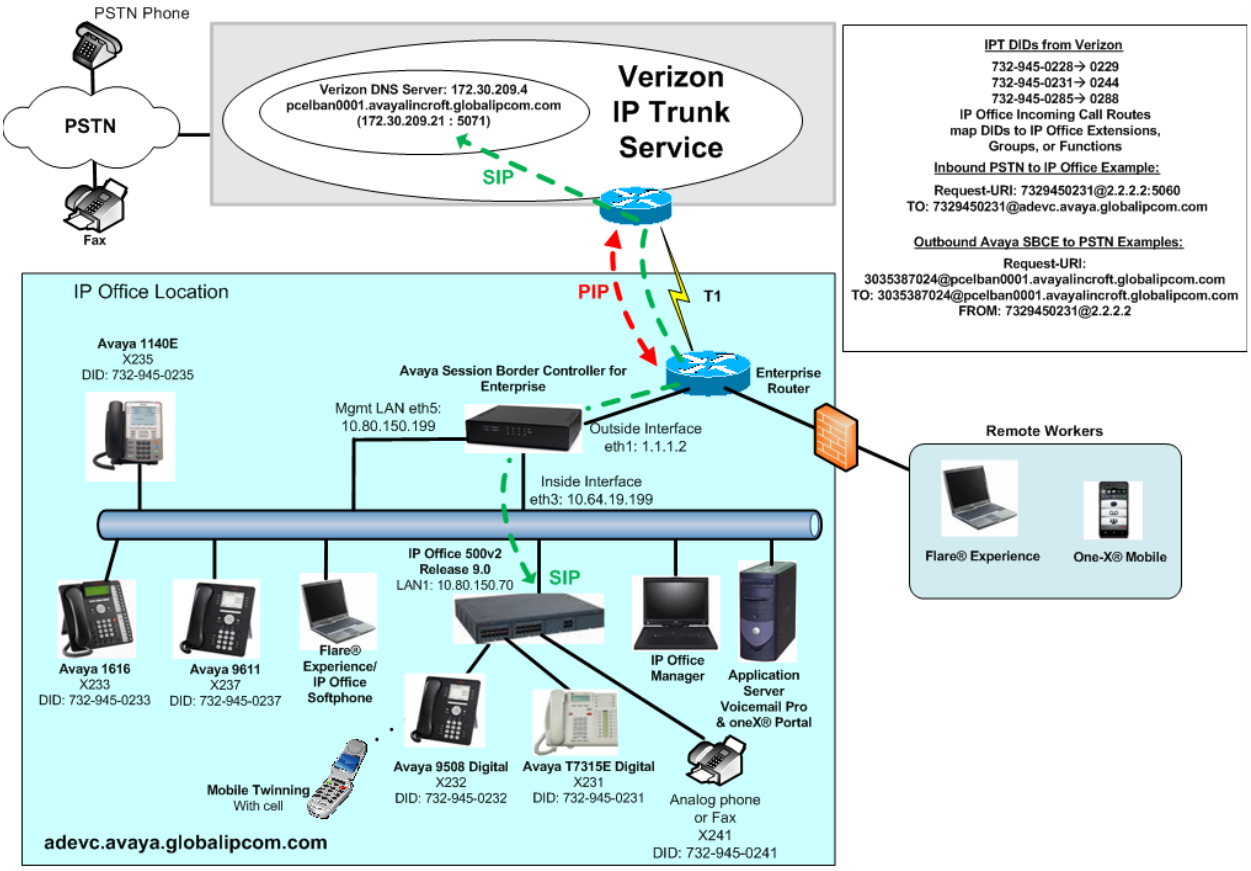


Figure 1: Avaya Interoperability Test Lab Configuration

Additionally, the reference configuration included remote worker functionality, introduced with Avaya IP Office 9.0 with Avaya SBCE. A remote worker is a SIP endpoint that resides in the untrusted network, registered to IP Office via the Avaya SBCE. Remote workers feature the same functionality as any other endpoint within the enterprise. This functionality was successfully tested during the compliance test, using the following endpoints and protocols:

- Avaya Flare® Experience for Windows (using TLS and SRTP)
- Avaya one-X® Mobile Preferred for IP Office on Android (using TCP and RTP)

The configuration tasks required to support remote workers are beyond the scope of these Application Notes; hence they are not discussed in this document. For more information on configuring the Avaya SBCE for IP Office remote workers, consult reference [**6**].

©2014 Avaya Inc. All Rights Reserved.

Table 1 shows the mapping of Verizon-provided DID numbers to IP Office users, groups, or functions. The associated IP Office configuration is shown in **Section 5**.

Verizon Provided DID	Avaya IP Office Destination	Notes
732-945-0231	X 231	T7316E Digital Telephone
732-945-0232	X 232	9508 Digital Telephone
732-945-0234	X 234	Avaya IP Office Softphone & Flare Experience
732-945-0235	X 235	Avaya SIP 1140E
732-945-0237	X 237	Avaya H.323 - 9611
732-945-0239	X 239	one-X® Mobile
732-945-0240	Short Code: FNE31	FNE Service 31 (Mobile Call Control)
732-945-0241	X 241	Analog telephone or Fax machine
732-945-0242	X 401 Hunt Group	Rotary Ring Mode to all Users
732-945-0243	Voicemail	Voicemail Pro Module

Table 1: Verizon DID to IP Office Mappings

4. Equipment and Software Validated

Table 2 shows the equipment and software used in the sample configuration

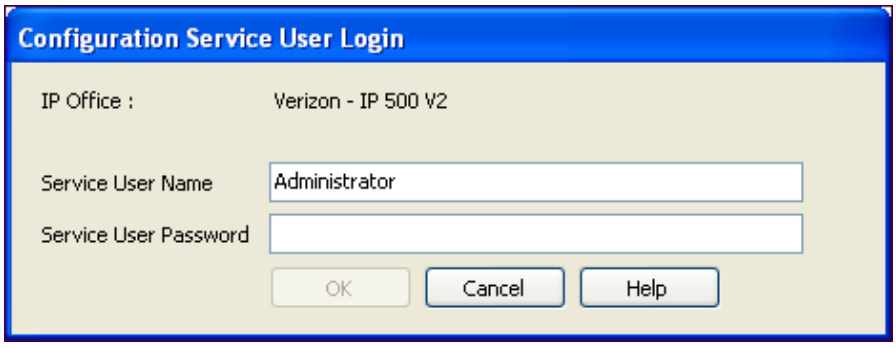
Avaya IP Telephony Solution Components	
Equipment	**Software**
Avaya Session Border Controller for Enterprise	Release 6.2.1 (Q07)
Avaya IP Office 500 V2	Release 9.0.0.829
Avaya Application Server	Release 9.0.0-829
Avaya IP Office Manager	Release 9.0.0.0 Build 829
Avaya 1616SW IP Telephone (H.323)	Release 1.33
Avaya 9621SW IP Telephone (H.323)	Release 6.2209
Avaya 1140E IP Telephone (SIP)	Release 04.03.18
Avaya 9508 Digital Telephone	Release 0.45
Avaya T7316E Digital Telephone	N/A
Avaya IP Office Softphone	Release 3.2.3.49
Avaya Flare® Experience for Windows	Release 1.1.4.23
Avaya one-X® Mobile Preferred for IP Office on Android	Release 1.8.0.9031

Table 2: Equipment and Software Tested

©2014 Avaya Inc. All Rights Reserved.

5. Avaya IP Office Configuration

IP Office is configured via the IP Office Manager program. For more information on IP Office Manager, consult reference [**2**]. From the IP Office Manager PC, select **Start → Programs → IP Office → Manager** to launch the Manager application. Provided that the IP Office system is accessible to IP Office Manager, the following will be displayed in the center of the opening screen:



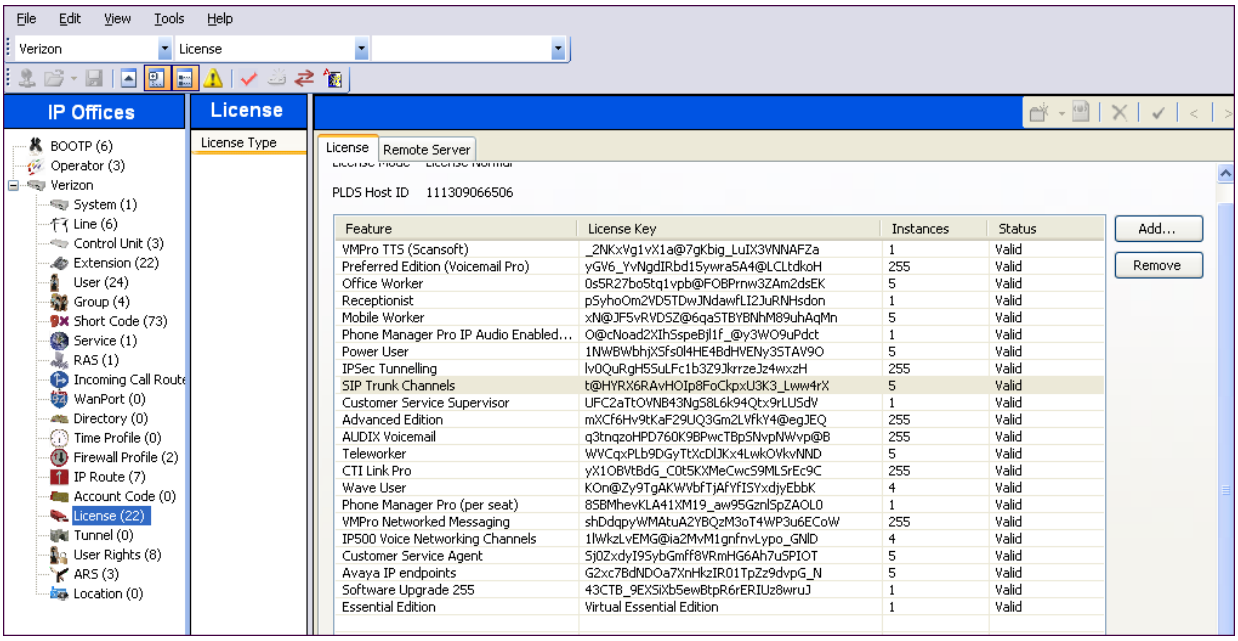
Log in with the appropriate configuration credentials. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details pane on the right side.

©2014 Avaya Inc. All Rights Reserved.

5.1. Licensing and Physical Hardware

The configuration and features described in these Application Notes require the IP Office system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

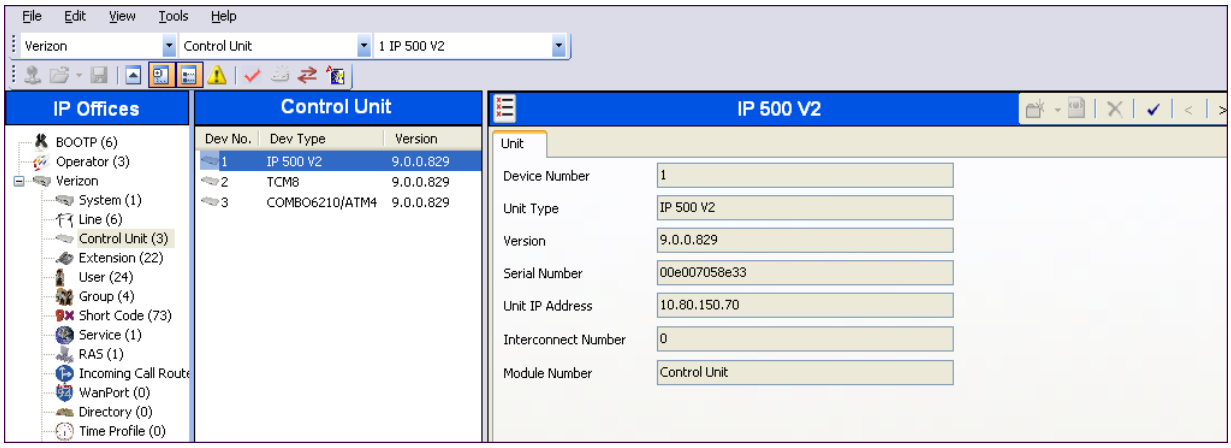
To verify that there is a SIP Trunk Channels License with sufficient capacity; click **License** in the Navigation pane. Confirm a valid **SIP Trunk Channels** license with sufficient **Instances** (trunk channels). If Avaya IP Telephones will be used as is the case in these Application Notes, verify the **Avaya IP endpoints** license.



©2014 Avaya Inc. All Rights Reserved.

In the sample configuration, looking at the IP Office 500 V2 from left to right, the first module is a TCM 8 Digital Station Module. This module supports BCM / Norstar T-Series and M-Series telephones. The second module is a COMBO6210/ATM4 module. This module is used to add a combination of ports to an IP500 V2 control unit and is not supported by IP500 control units. The module supports 10 voice compression channels. Codec support is G.711, G729A and G.723 with 64ms echo cancellation. G.722 is supported by IP Office Release 8.0 and higher. The "Combo" card will support 6 Digital Station ports for digital stations in slots 1-6 (except 3800, 4100, 4400, 7400, M and T-Series), 2 Analog Extension ports in slots 7-8, and 4 Analog Trunk ports in slots 9-12. Referring to **Figure 1**, the Avaya T7315E telephone with extension 231 is connected to port 1 of the TCM8 module, and the Avaya 9508 telephone with extension 232 is connected to port 1 of the "Combo" card. The analog extension or fax machine is connected to the "Combo" card on port 7.

The following screen shows the modules in the IP Office used in the sample configuration. To access such a screen, select **Control Unit** in the Navigation pane. The modules appear in the Group pane. In the screen below, **IP 500 V2** is selected in the Group pane, revealing additional information about the IP 500 V2 in the Details pane.



©2014 Avaya Inc. All Rights Reserved.

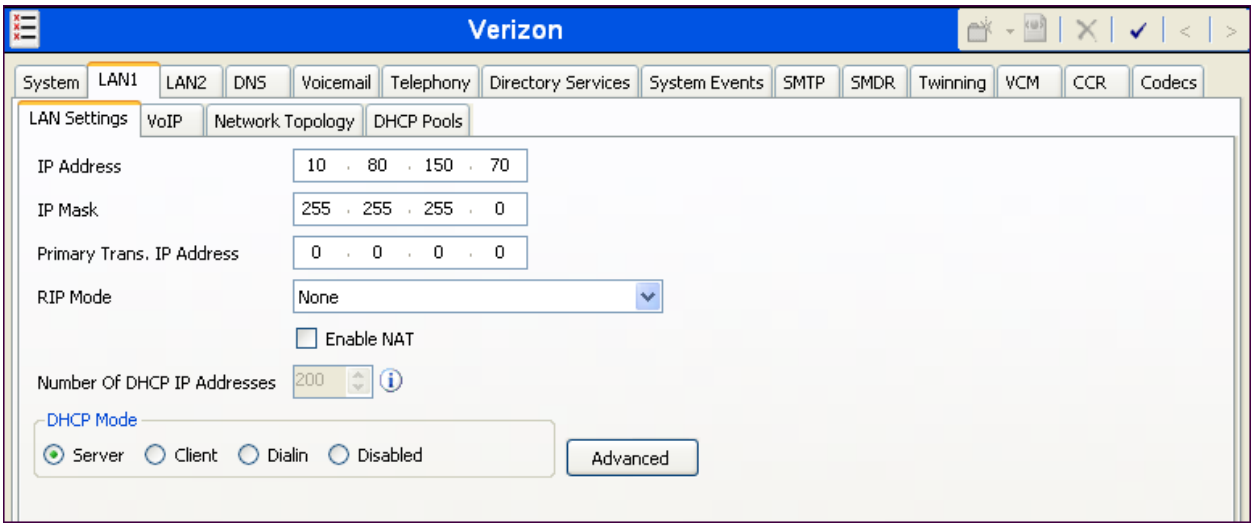
5.2. System Settings

This section illustrates the configuration of system settings. Select **System** in the Navigation pane to configure these settings. The subsection order corresponds to a left to right navigation of the tabs in the Details pane for System settings.

5.2.1. LAN Settings

The IP500/IP500 V2 control units have 2 RJ45 Ethernet ports, physically marked as LAN and WAN. Within the system configuration, the physical LAN port is LAN1, the physical WAN port is LAN2.

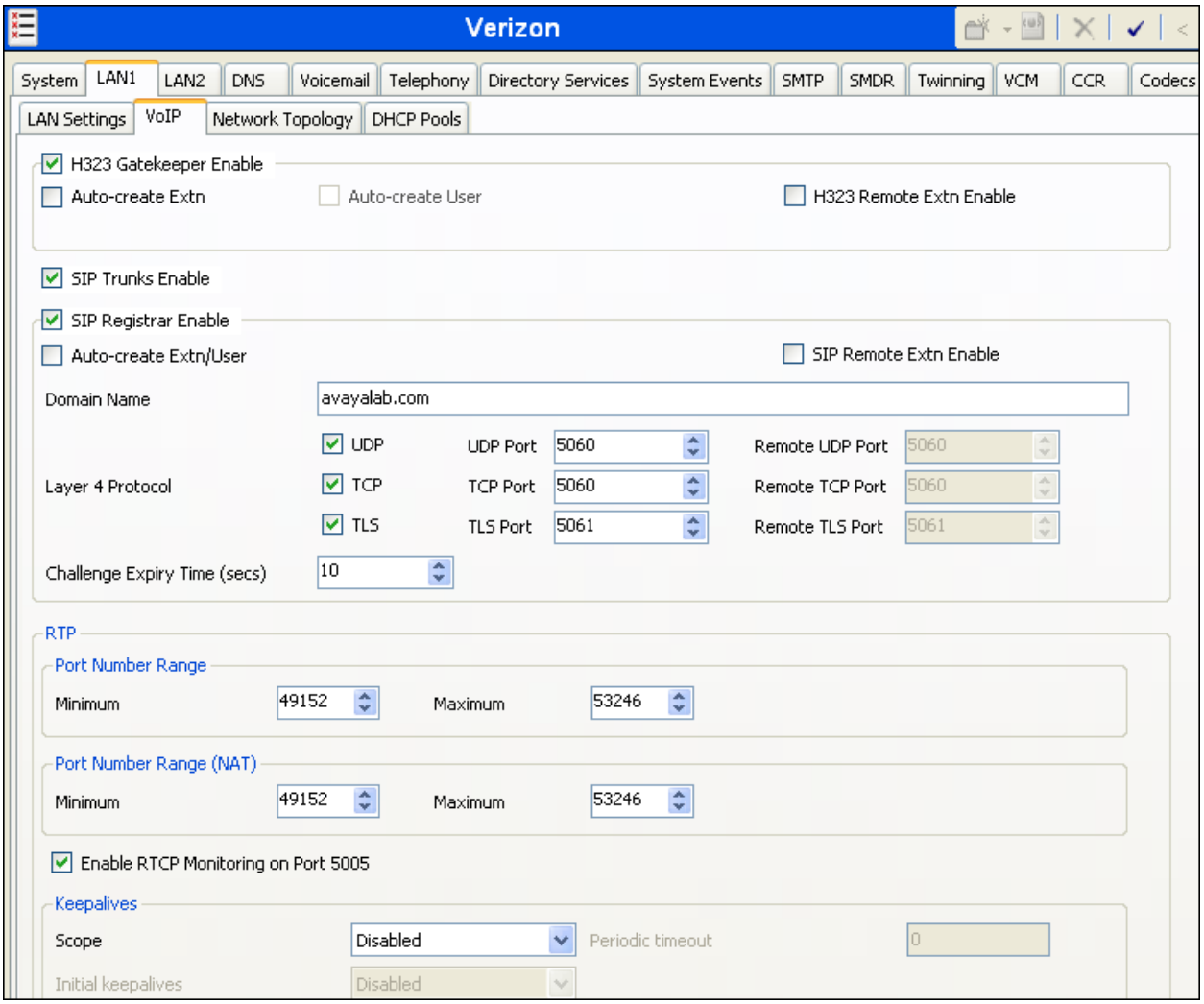
In the sample configuration, LAN1 is used to connect the IP Office to the enterprise network. To view or configure the **IP Address** of LAN1, select the **LAN1** tab followed by the **LAN Settings** tab. As shown in **Figure 1**, the IP Address of the IP Office is 10.80.150.70. **DHCP Mode** is set to "**Server**" so that IP phones will get an IP Address from the IP Office Server. Other parameters on this screen may be set according to customer requirements.



©2014 Avaya Inc. All Rights Reserved.

Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 1616 and 9611 used in the sample configuration. The **SIP Registrar Enable** box is checked to allow Avaya 1140E, Avaya IP Office Softphone, and Avaya Flare® Experience usage. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Verizon Business.

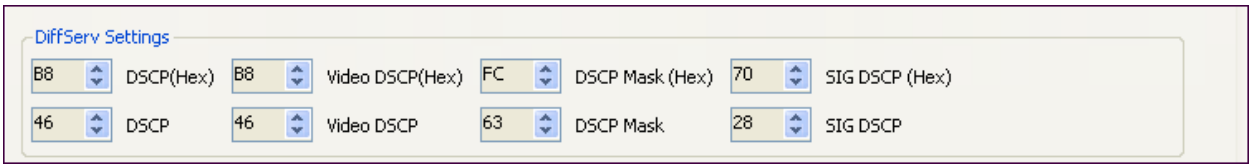
If desired, the **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media paths from Avaya SBCE to IP Office. The defaults are used here.



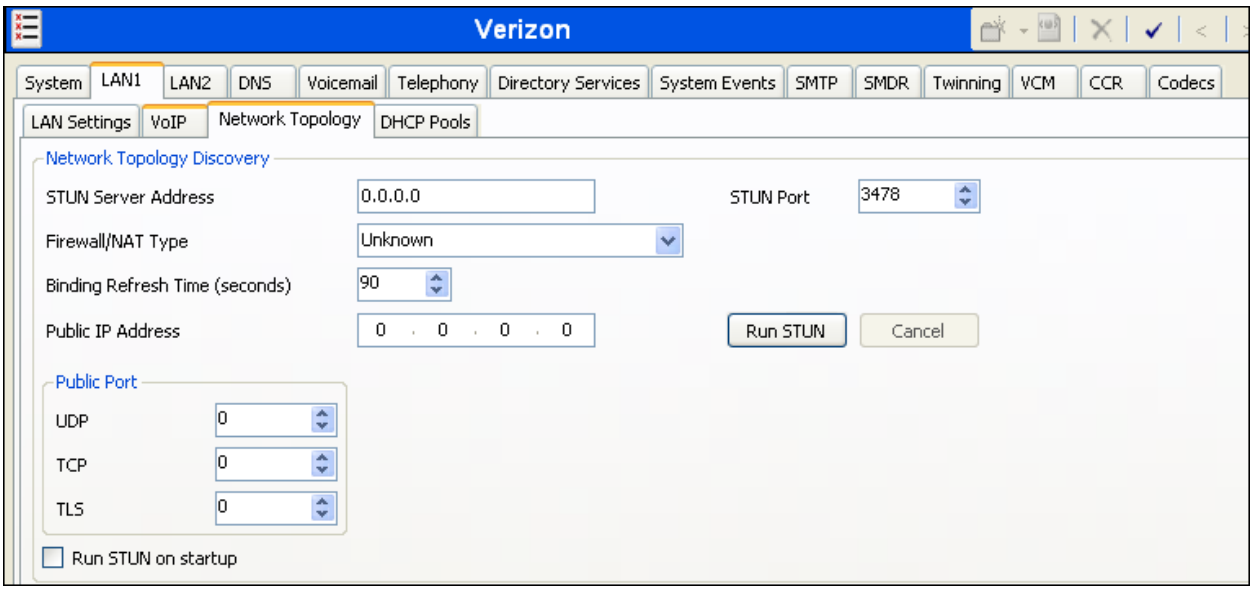
Scrolling down, IP Office can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Service policies. In the sample configuration shown below, IP Office will mark SIP signaling with a value associated with "Assured Forwarding" using DSCP decimal 28 (**SIG DSCP** parameter). IP Office will mark the RTP media with a value associated with "Expedited Forwarding" using DSCP decimal 46 (**DSCP** parameter). This screen enables flexibility in IP Office DiffServ markings (RFC 2474) to

©2014 Avaya Inc. All Rights Reserved.

allow alignment with network routing policies, which are outside the scope of these Application Notes. Other parameters on this screen may be set according to customer requirements.



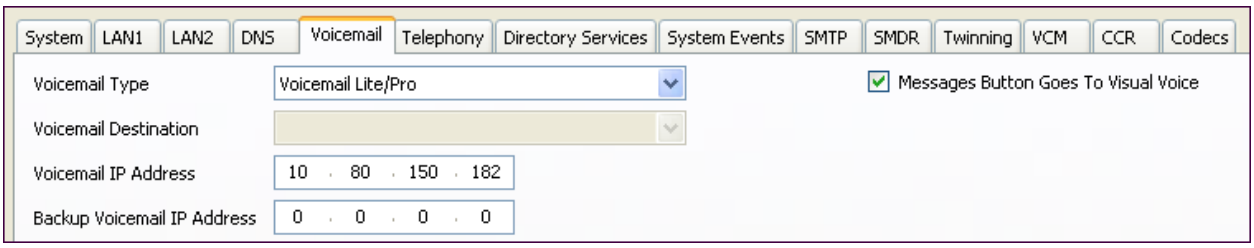
Select the **Network Topology** tab as shown in the following screen. In the sample configuration, the default settings are used and the **Use Network Topology Info** in the **SIP Line** is set to "**None**" in **Section 5.4.3**. The **Binding Refresh Time (seconds)** can be used to lower the SIP OPTIONS timing from the default of 300 seconds. During the testing, the Binding Refresh Time was varied (e.g., 90 seconds, 120 seconds) to test SIP OPTIONS timing.



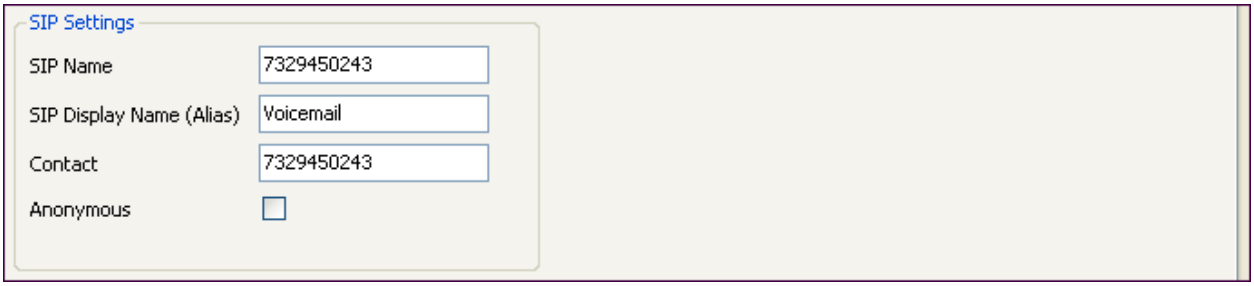
©2014 Avaya Inc. All Rights Reserved.

5.2.2. Voicemail Settings

To view or change voicemail settings, select the **Voicemail** tab as shown in the following screen. The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. The **Voicemail Type** in the sample configuration is "**Voicemail Lite/Pro**". Other Voicemail types may be used. The Voicemail IP Address in the sample configuration is 10.80.150.182, the IP Address of the IP Office Application Server running the Voicemail Pro software, as shown in **Figure 1**.



In the sample configuration, the "Callback" application of Avaya Voicemail Pro was used to allow Voicemail Pro to call out via the SIP Line to Verizon Business when a message is left in a voice mailbox. The **SIP Settings** shown in the screen below enable IP Office to populate the SIP headers for an outbound "callback" call from Voicemail Pro, similar to the way the fields with these same names apply to calls made from telephone users (e.g., see **Section 5.5**).

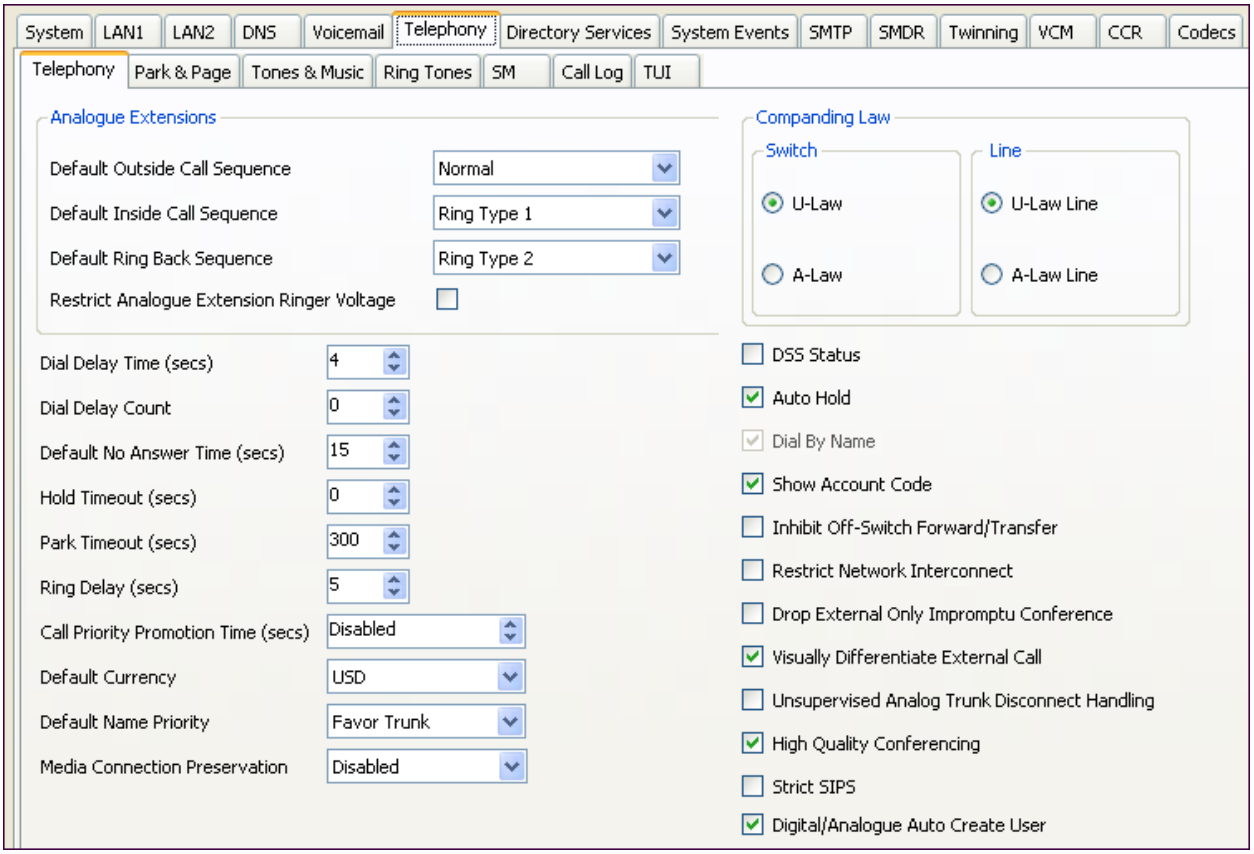


©2014 Avaya Inc. All Rights Reserved.

5.2.3. System Telephony Configuration

To view or change telephony settings, select the **Telephony** tab and **Telephony** sub-tab as shown in the following screen. The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. In the sample configuration, the **Inhibit Off-Switch Forward/Transfer** box is unchecked so that call forwarding and call transfer to PSTN destinations via the Verizon Business IP Trunk service can be tested. That is, a call can arrive to IP Office via the Verizon Business IP Trunk, and be forwarded or transferred back to the PSTN with the outbound leg of the call using the Verizon IP Trunk service. The **Companding Law** parameters are set to "**U-Law**" as is typical in North American locales. Other parameters on this screen may be set according to customer requirements.

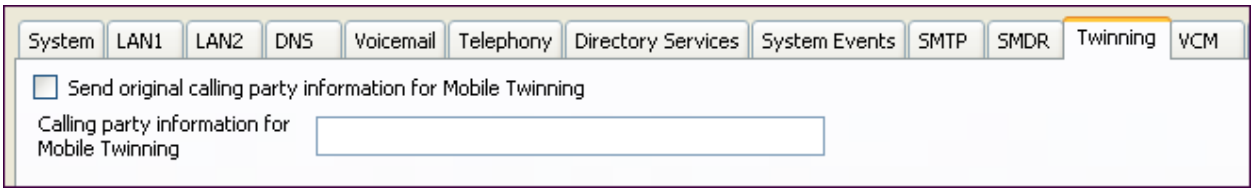
The **Default Name Priority** parameter was first introduced in IP Office Release 8 and can be relevant to SIP Trunking. The option to "Favor Trunk" or "Favor Directory" can be set system-wide using the screen below, or set uniquely for each line. With the option to "Favor Directory", IP Office will prefer to display names found in a personal or system directory over those arriving from the far-end, if there is a directory match to the caller ID. This capability will be illustrated further in the context of the SIP Line to Verizon.



©2014 Avaya Inc. All Rights Reserved.

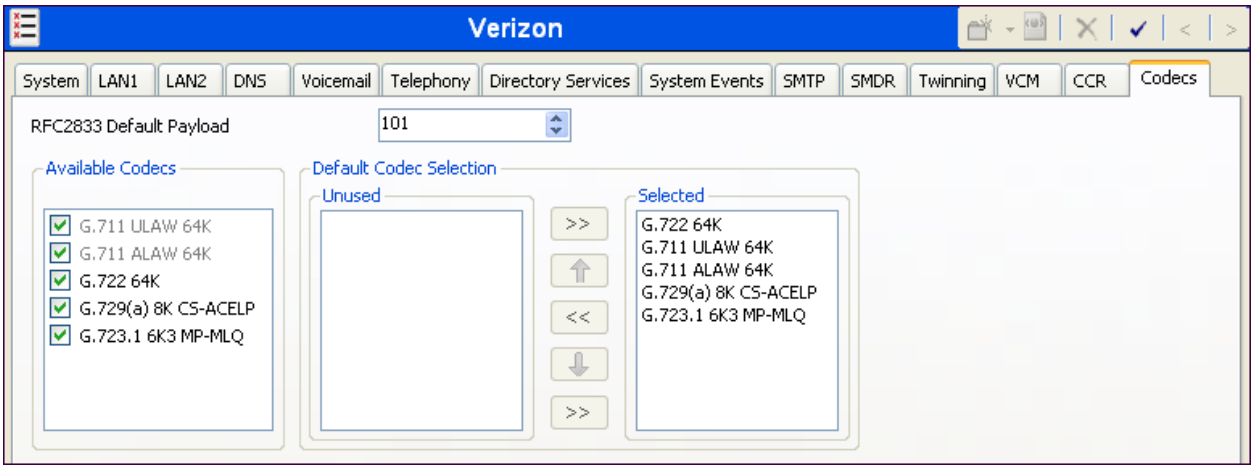
5.2.4. System Twinning Configuration

To view or change Twinning settings, select the **Twinning** tab as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning** is left blank. With this configuration, and related configuration of "Diversion header" on the SIP Line (**Section 5.4.2**), the true identity of a PSTN caller can be presented to the twinning destination (e.g., a user's mobile phone) when a call is twinned out via the Verizon Business IP Trunk service.



5.2.5. System Codecs Configuration

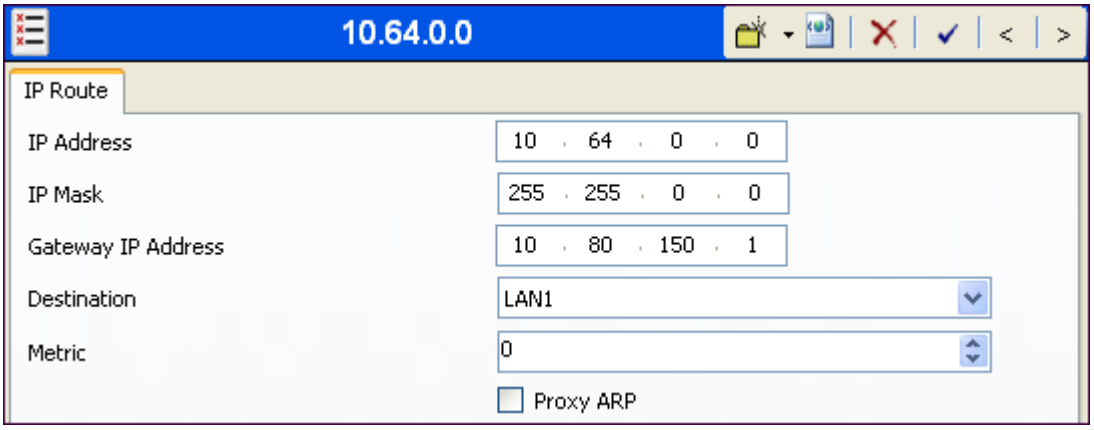
To view or change system codec settings, select the **Codecs** tab. On the left, observe the list of **Available Codecs**. In the example screen below, which is not intended to be prescriptive, the box next to each codec is checked, making all the codecs available in other screens where codec configuration may be performed (such as the SIP Line in **Section 5.4.5**). The **Default Codec Selection** area enables the codec preference order to be configured on a system-wide basis, using the up, down, left, and right arrows. By default, all IP (SIP and H.323) lines and extensions will assume the system default codec selection, unless configured otherwise for the specific line or extension. The **RFC2833 Default Payload** parameter is new in IP Office Release 9.0. Set the payload parameter to "**101**", the value preferred by Verizon Business.



©2014 Avaya Inc. All Rights Reserved.

5.3. IP Route

In the sample configuration, the IP Office LAN1 port is physically connected to the local area network switch at the IP Office customer site. The default gateway for this network is 10.80.150.1. The Avaya SBCE resides on a different subnet and requires an IP Route to allow SIP traffic between the two devices. To add an IP Route in IP Office, right-click **IP Route** from the Navigation pane, and select **New**. To view or edit an existing route, select **IP Route** from the Navigation pane, and select the appropriate route from the Group pane. The following screen shows the Details pane with the relevant route using **Destination** LAN1.



©2014 Avaya Inc. All Rights Reserved.

5.4. SIP Line

This section shows the configuration screens for the SIP Line in IP Office Release 9. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.4.1** to create the SIP Line from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:

- IP addresses
- SIP Credentials (if applicable)
- SIP URI entries
- Setting of the **Use Network Topology Info** field on the Transport tab.

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.4.2 – 5.4.5**.

Also, the following SIP Line settings are not supported on Basic Edition:

- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required

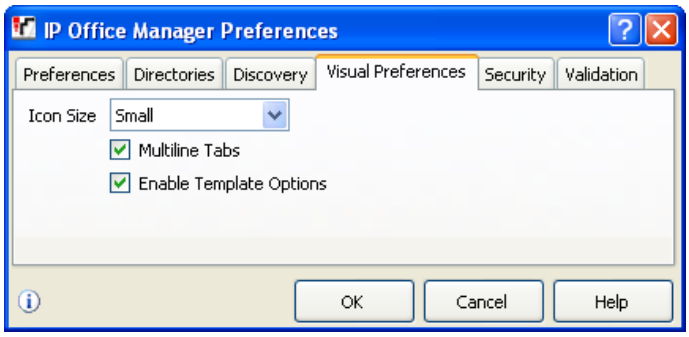
Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.4.2 – 5.4.5**.

©2014 Avaya Inc. All Rights Reserved.

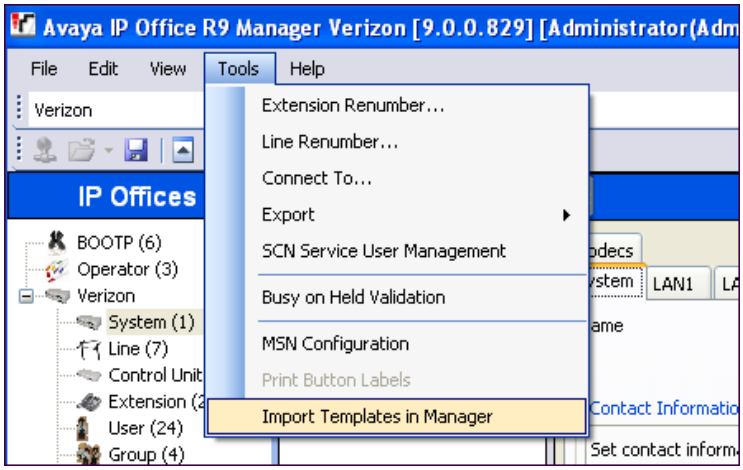
5.4.1. SIP Line From Template

1. Copy the template file to the computer where IP Office Manager is installed. Rename the template file to **US_VerizonIPT-ASBCE_SIPTrunk.xml**. The file name is important in locating the proper template file in **Step 5**.

2. Verify that template options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the IP Office Manager Preferences window that appears, select the Visual Preferences tab. Verify that the box is checked next to **Enable Template Options**. Click **OK**.



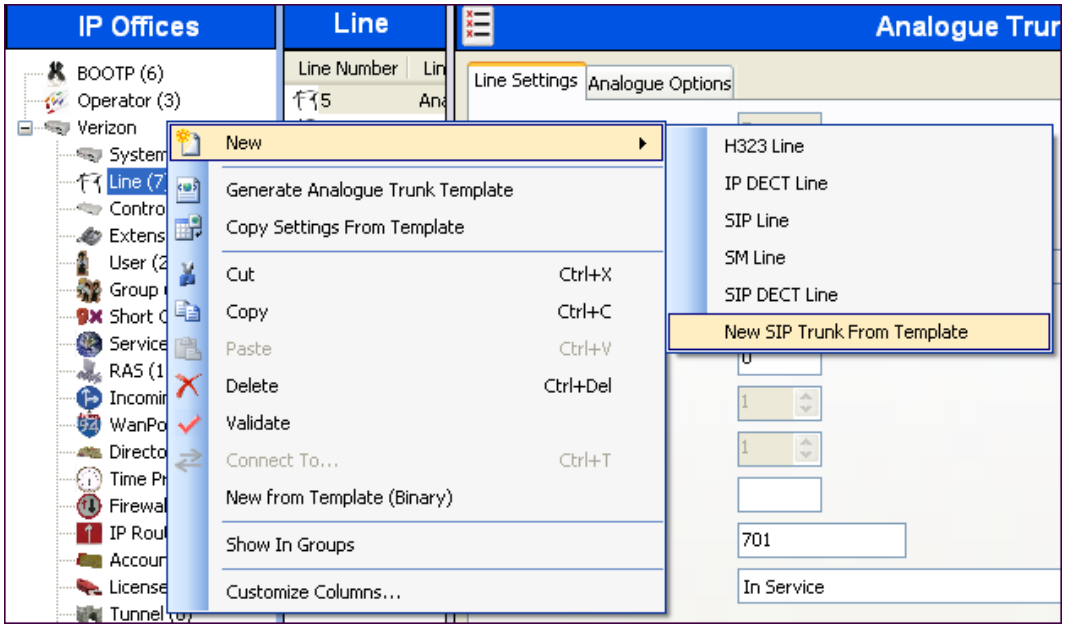
3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default template location is **C:\Program Files\Avaya\IP Office\Manager\Templates**.



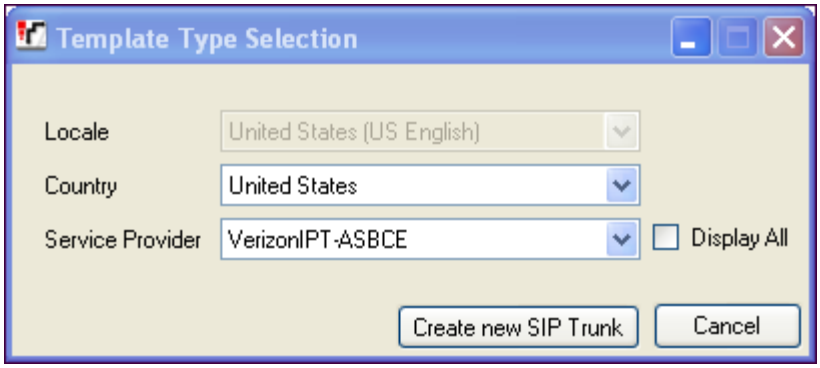
In the pop-up window (not shown) that appears, select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window (not shown) will appear stating success or failure. Click **OK** (not shown) to continue. If preferred, this step may be skipped if the template file is copied directly to the IP Office template directory.

©2014 Avaya Inc. All Rights Reserved.

4. To create the SIP Trunk from the template, right-click on **Line** in the Navigation Pane, then navigate to **New → New SIP Trunk From Template**.



5. In the subsequent Template Type Selection pop-up window, select "United States" from the **Country** pull-down menu and select "VerizonIPT-ASBCE" from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name (**US_VerizonIPT-ASBCE_SIPTrunk.xml)** created in **Step 1**. Click **Create new SIP Trunk** to finish creating the trunk.



Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.4.2 – 5.4.5**.

5.4.2. SIP Line – SIP Line Tab

The **SIP Line** tab in the Details pane is shown below for Line Number 18, used for Avaya SBCE to the Verizon Business IP Trunk service. The **ITSP Domain Name** is configured with the inside IP address of the Avaya SBCE as shown in **Figure 1**. By default, the **In Service** and **Check**

©2014 Avaya Inc. All Rights Reserved.

OOS boxes are checked. In the sample configuration, IP Office will use the SIP OPTIONS method to periodically check the SIP Line. The time between SIP OPTIONS sent by IP Office will use the **Binding Refresh Time** for LAN1, as shown in **Section 5.2.1**.

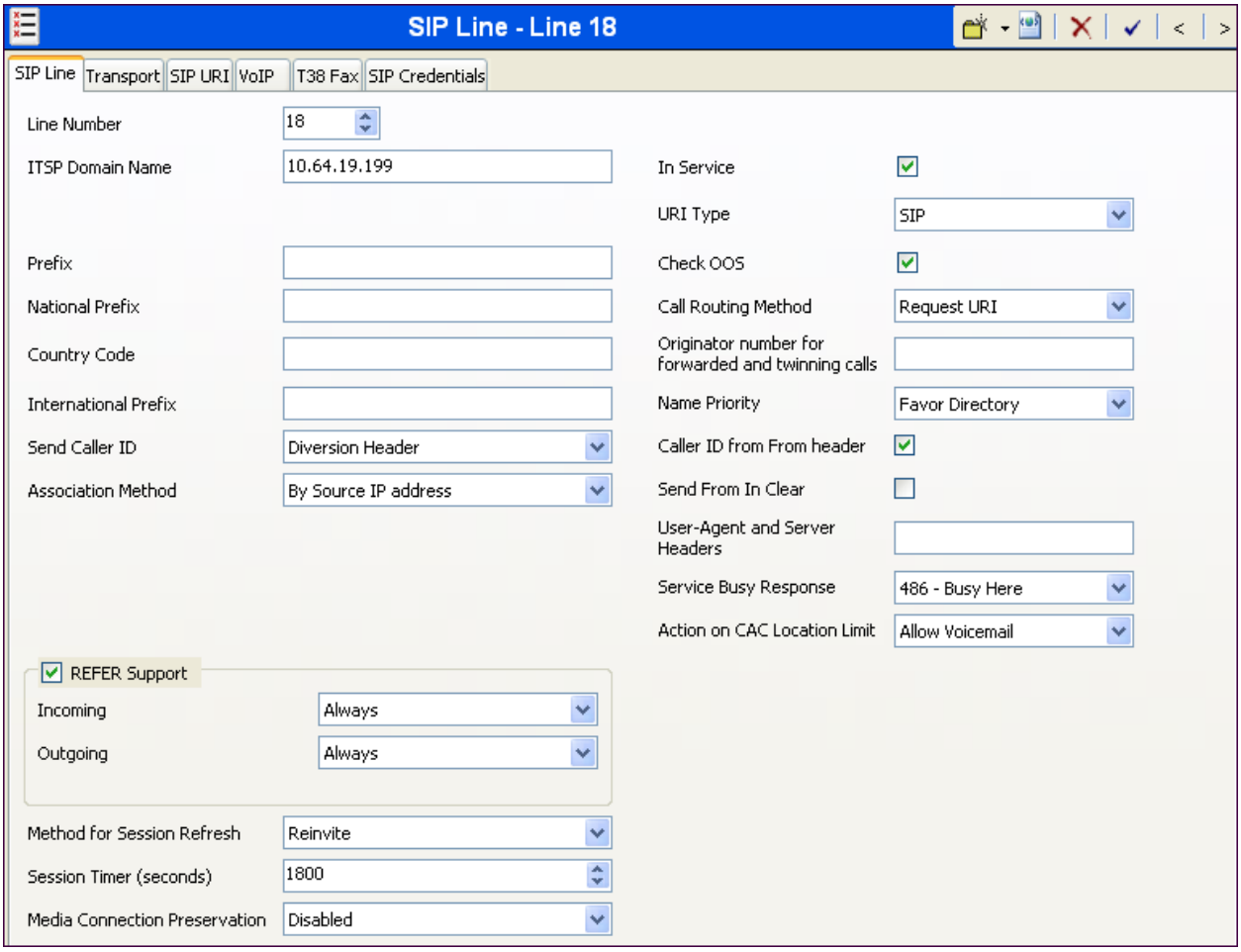
The **Send Caller ID** parameter is set to "Diversion Header". With this setting and the related configuration in **Section 5.2.4**, IP Office will include the Diversion Header for calls that are directed via Mobile Twinning out the SIP Line to Verizon. The Diversion Header will contain the number associated with the Twinning user, allowing Verizon to admit the call. The From Header will be populated with the true calling party identity, allowing the twinning destination (e.g., mobile phone) to see the true caller id. IP Office will also include the Diversion header for calls that are call forwarded out the SIP Line to Verizon.

The default automatic determination of **REFER Support** is "Auto". Alternatively, the default can be overridden with "Never" to explicitly disable use of REFER, or "Always" to explicitly enable use of REFER.

The **Method for Session Refresh** parameter and the related **Session Timer (seconds)** parameter are new with IP Office Release 9.0. The **Method for Session Refresh** is set to "Reinvite" and the **Session Timer (seconds)** is set to "1800". With this configuration, IP Office will send re-INVITEs every 15 minutes (half of the set value) to keep the active session alive. The **Media Connection Preservation** parameter retains the default setting "Disabled".

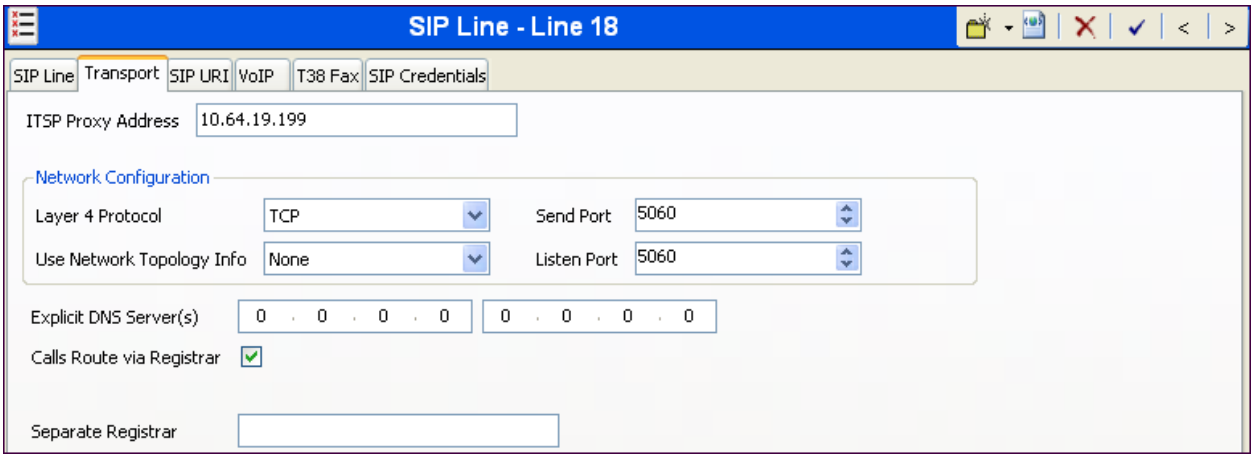
The **Name Priority** parameter can retain the default "System Default" setting, or can be configured to "Favor Trunk" or "Favor Directory" as shown in the sample screen below. "System Default" will use the setting displayed on the **System → Telephony→ Telephony** tab. The "Favor Directory" setting enables IP Office to match the caller's telephone number against available system or personal directories, and display the name obtained from a match in the directory, if any, rather than name information received in the SIP signaling from Verizon. The **Caller ID from From header** is checked to insure the incoming caller ID information in the From field is used rather than the P-Asserted-Identity or Contact fields. Click **OK** (not shown).

©2014 Avaya Inc. All Rights Reserved.



5.4.3. SIP Line - Transport Tab

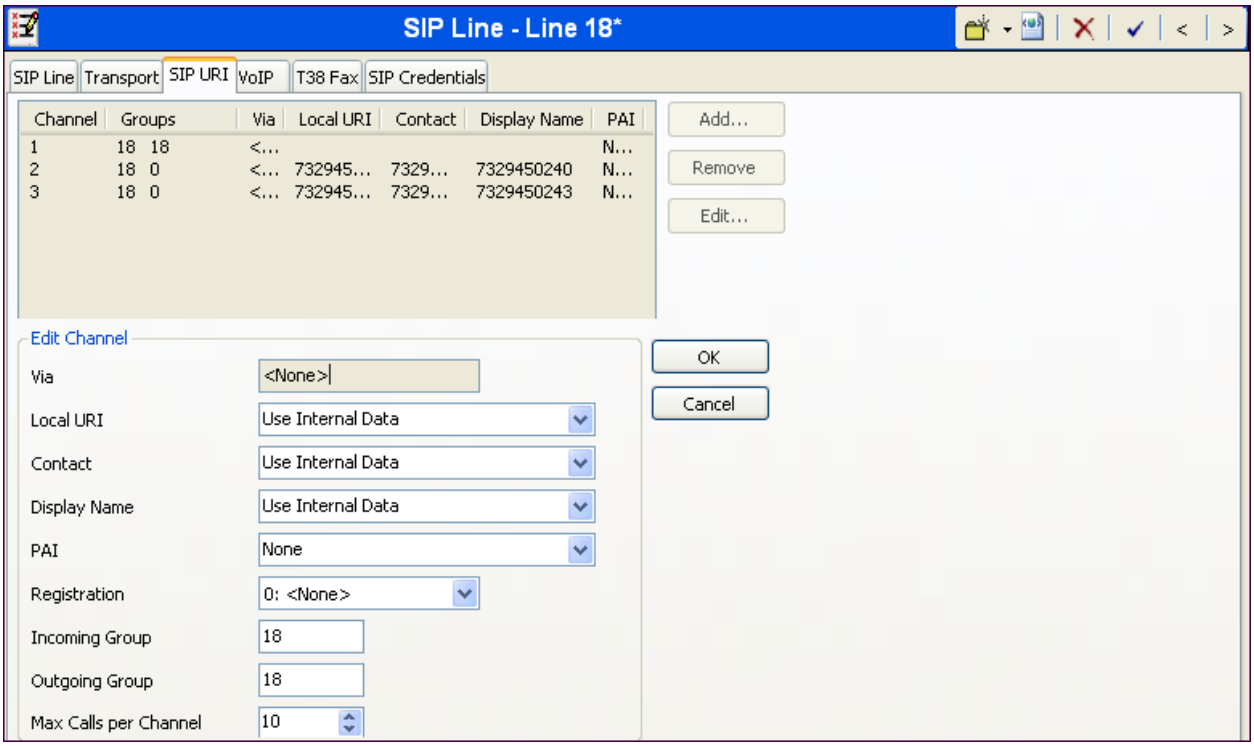
Select the **Transport** tab. The **ITSP Proxy Address** is set to the inside IP address of the Avaya SBCE as shown in **Figure 1**. In the **Network Configuration** area, "**TCP**" is selected as the **Layer 4 Protocol**. The **Send Port** and **Listen Port** can retain the default value 5060. The **Use Network Topology Info** parameter is set to "**None**".



©2014 Avaya Inc. All Rights Reserved.

5.4.4. SIP Line - SIP URI Tab

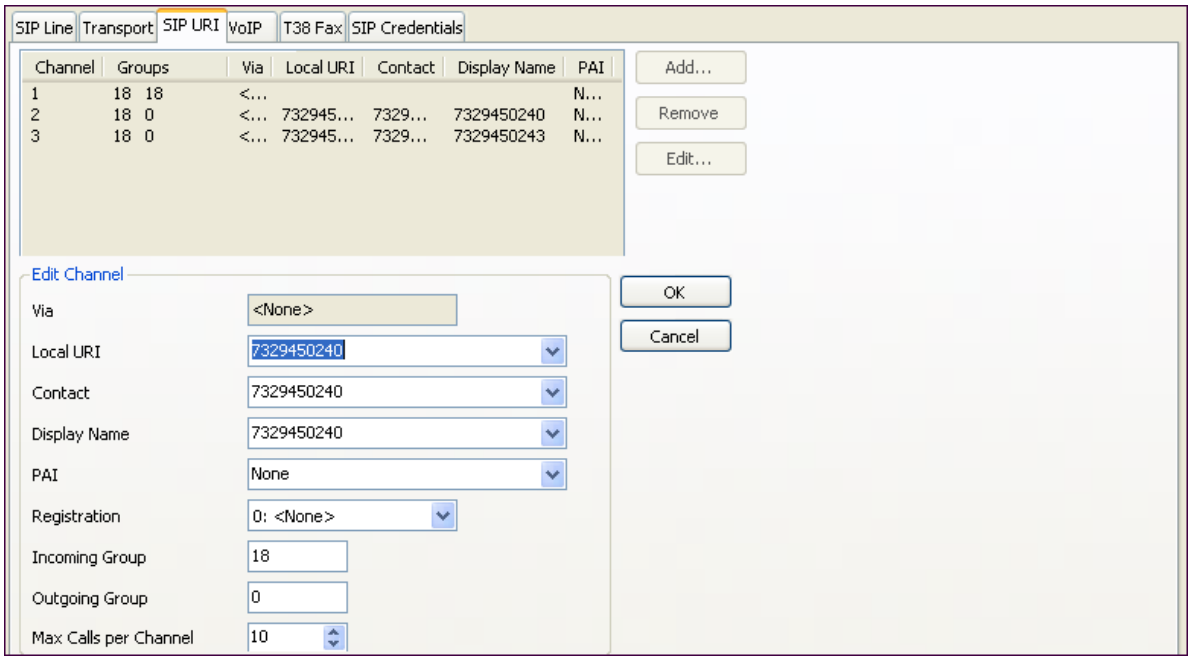
Select the **SIP URI** tab. To add a new SIP URI, click the **Add…** button. In the bottom of the screen, a New Channel area will be opened. To edit an existing entry, click an entry in the list at the top, and click the **Edit…** button. In the bottom of the screen, the Edit Channel area will be opened. In the example screen below, a previously configured entry is edited. "**Use Internal Data**" is selected for the **Local URI**, **Contact**, and **Display Name**. Information configured on the SIP Tab for individual users will be used to populate the SIP headers. The **PAI** parameter is set to "**None**". The **Registration** parameter is set to the default "**0: <None>**" since Verizon Business IP Trunk service does not require registration. The **Incoming Group** parameter, set here to 18, will be referenced when configuring Incoming Call Routes to map inbound SIP trunk calls to IP Office destinations in **Section 5.7**. The **Outgoing Group** parameter, set here to 18, will be used for routing outbound calls to Verizon via the Short Codes (**Section 5.6**). The **Max Calls per Channel** parameter, configured here to 10, sets the maximum number of simultaneous calls that can use the URI before IP Office returns busy to any further calls. Click **OK**.

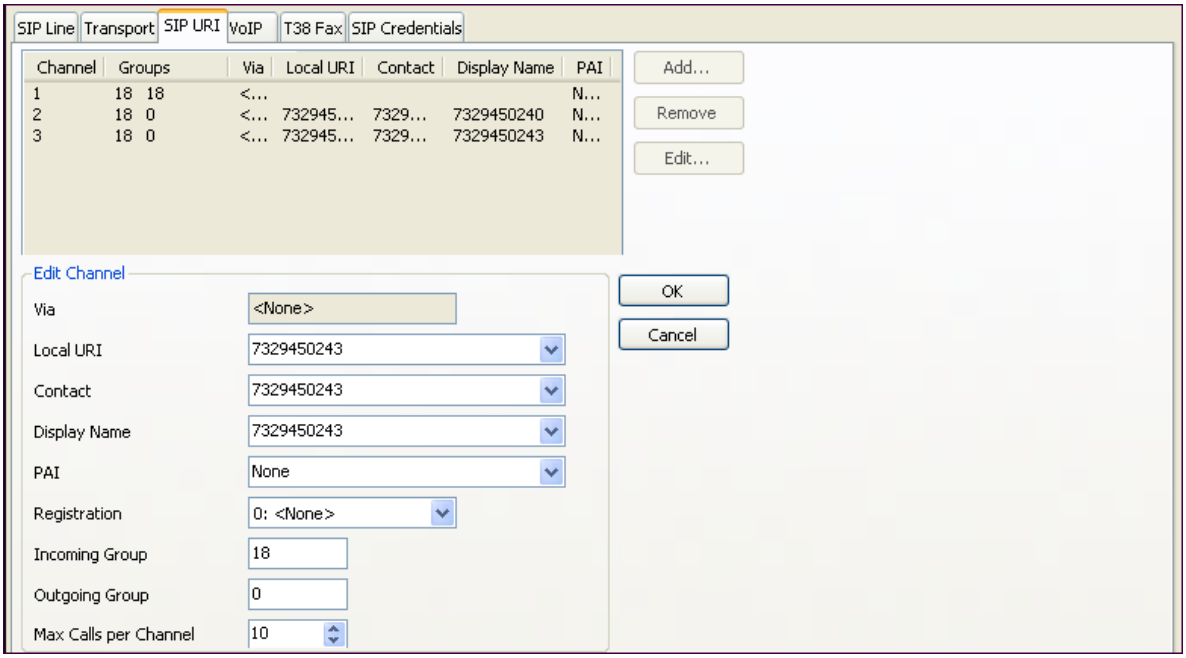


In the sample configuration, the single SIP URI shown above was sufficient to allow incoming calls for Verizon DID numbers destined for specific IP Office users or IP Office hunt groups. The calls are accepted by IP Office since the incoming number will match the SIP Name configured for the user or hunt group that is the destination for the call.

©2014 Avaya Inc. All Rights Reserved.

Channels 2 and 3 display service numbers, such as a DID number routed directly to voicemail or DID used for Mobile Call Control. DID numbers that IP Office should admit can be entered into the **Local URI** and **Contact** fields instead of "Use Internal Data". The numbers 732-945-0240 and 732-945-0243 will be assigned as service numbers in the Incoming Call Routes in **Section 5.7**.

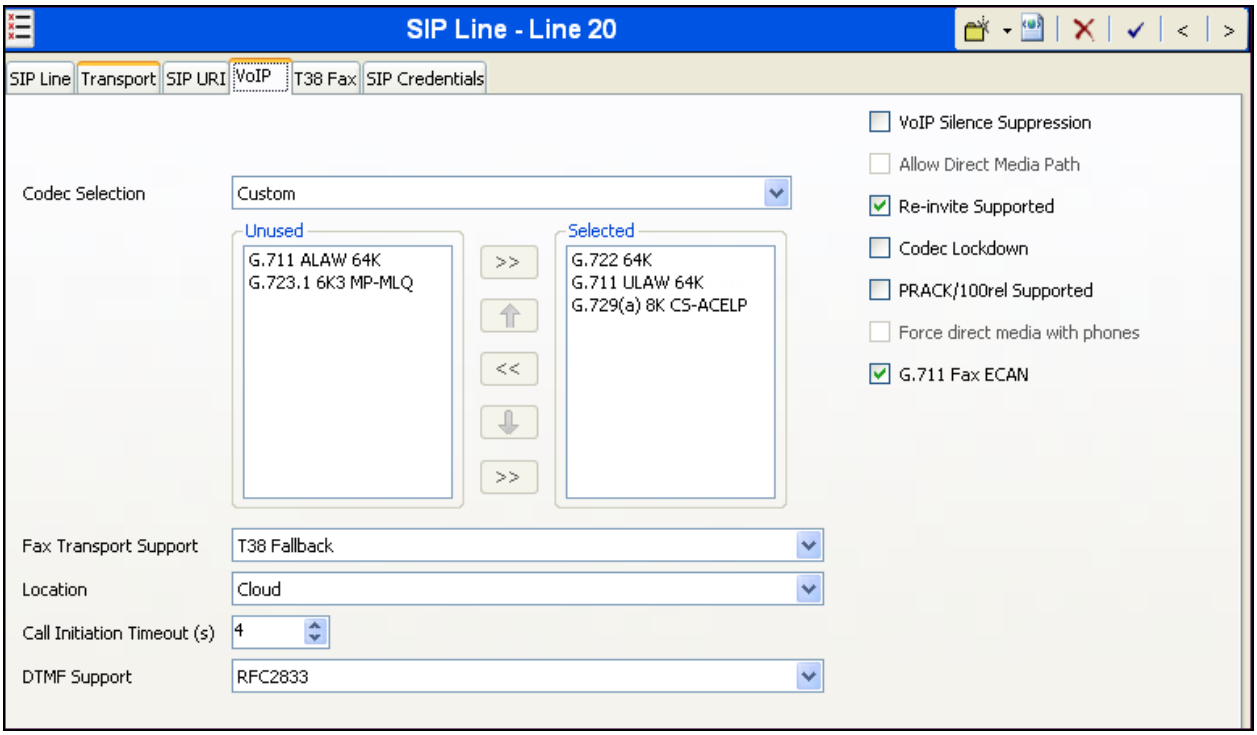




©2014 Avaya Inc. All Rights Reserved.

5.4.5. SIP Line - VoIP Tab

Select the **VoIP** tab. The **Codec Selection** drop-down box **System Default** (default) will match the codecs set in the system wide Default Selection list (**System → Codecs**). In the sample configuration, **Custom** is selected and codecs preferred by Verizon are included as well as the newly supported G.722 codec (i.e., G.722 64K, G729(a) 8K CS-ACELP and G.711 ULAW 64K). This will cause IP Office to include G.722, G.711MU, and G.729a in the Session Description Protocol (SDP) offer, in that order. Set the **Fax Transport Support** drop-down to "**T38 Fallback**". This enables T.38 to be used if supported and will fall-back to G.711 if not. The **DTMF Support** parameter can remain set to the default value "**RFC2833**". The **Re-invite Supported** parameter can be checked to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk. The **Re-invite Supported** parameter and the **G.711 Fax ECAN** parameter should be checked and **G.711 ULAW 64K** as the first codec preference if the SIP Line will be used for fax. For PSTN originations, Verizon preferred the G.729a codec in the SDP, while also allowing the G.711MU codec. However, if an originator is at a SIP connected location and offers G.722, Verizon will preserve this offer and allow G.722 to be negotiated and used end to end. During testing, the IP Office configuration was varied such that G.729a was the preferred or only codec listed, and G.729a calls were also successfully verified. Click **OK** (not shown).



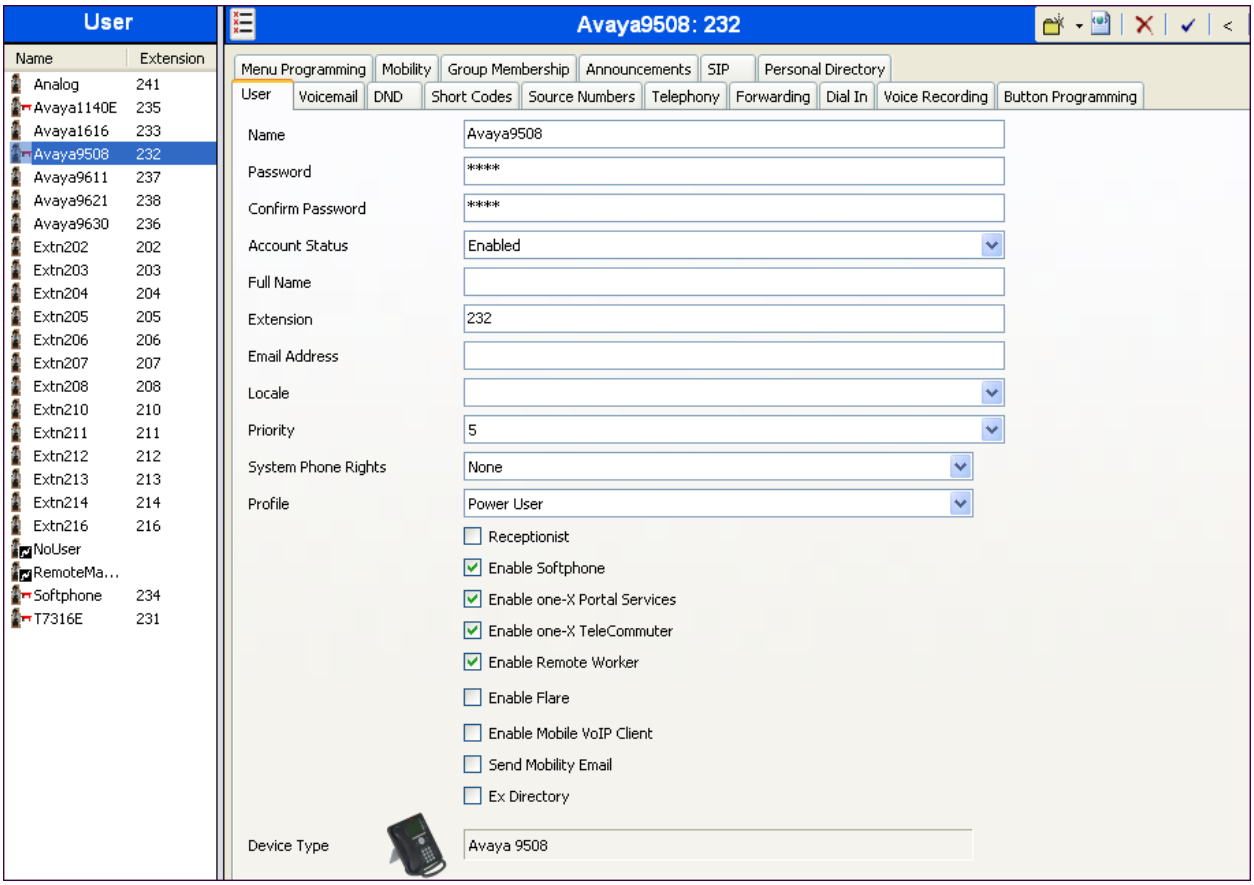
©2014 Avaya Inc. All Rights Reserved.

5.5. Users, Extensions, and Hunt Groups

In this section, examples of an IP Office User, Extension, and Hunt Group will be illustrated. In the interests of brevity, not all users and extensions shown in **Figure 1** will be presented, since the configuration can be easily extrapolated to other users. To add a User, right click on **User** in the Navigation pane, and select **New**. To edit an existing User, select **User** in the Navigation pane, and select the appropriate user to be configured in the Group pane.

5.5.1. Digital User 232

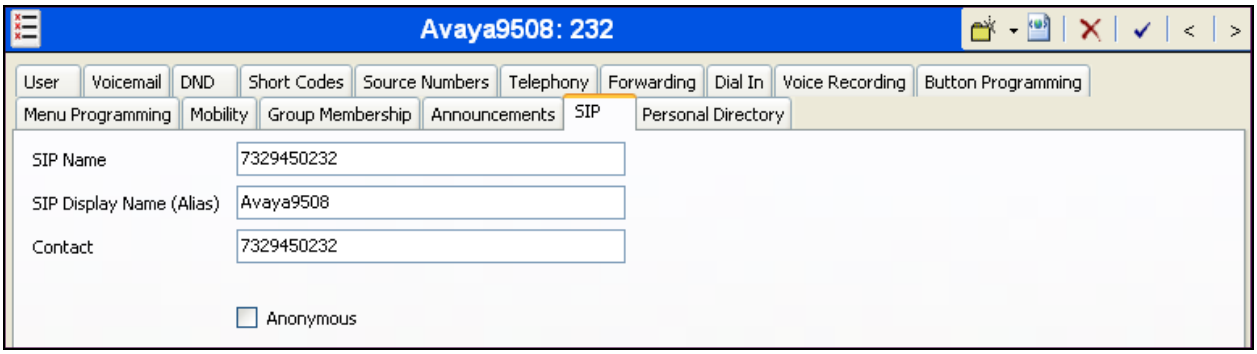
The following screen shows the **User** tab for User 232. As shown in **Figure 1**, this user corresponds to the Avaya Digital 9508.



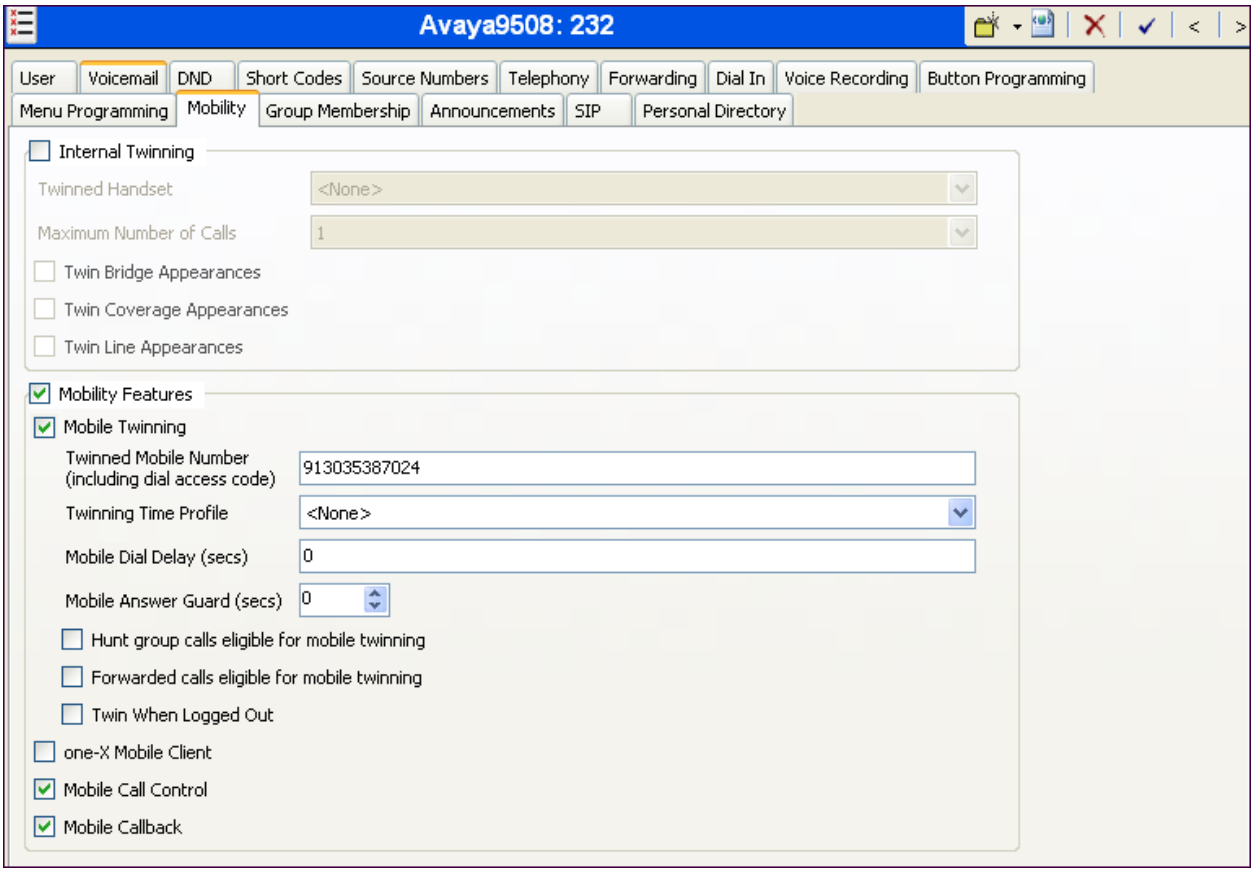
The following screen shows the **SIP** tab for User 232. The **SIP Name** and **Contact** parameters are configured with the DID number of the user, 732-945-0232. These parameters configure the user part of the SIP URI in the From header for outgoing SIP trunk calls, and allow matching of the SIP URI for incoming calls, without having to enter this number as an explicit SIP URI for the SIP Line. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network.

©2014 Avaya Inc. All Rights Reserved.

See **Section 5.6** for a method of using a short code (rather than static user provisioning) to place an anonymous call.



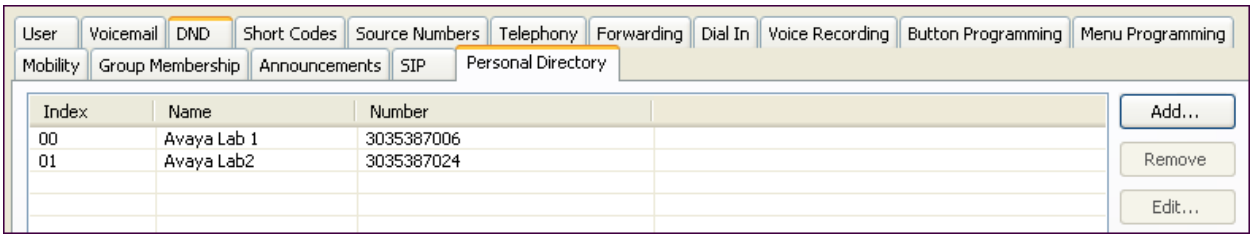
From **Figure 1**, note that user 232 will use the Mobile Twinning feature. The following screen shows the **Mobility** tab for User 232. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case 913035387024. Other options can be set according to customer requirements.



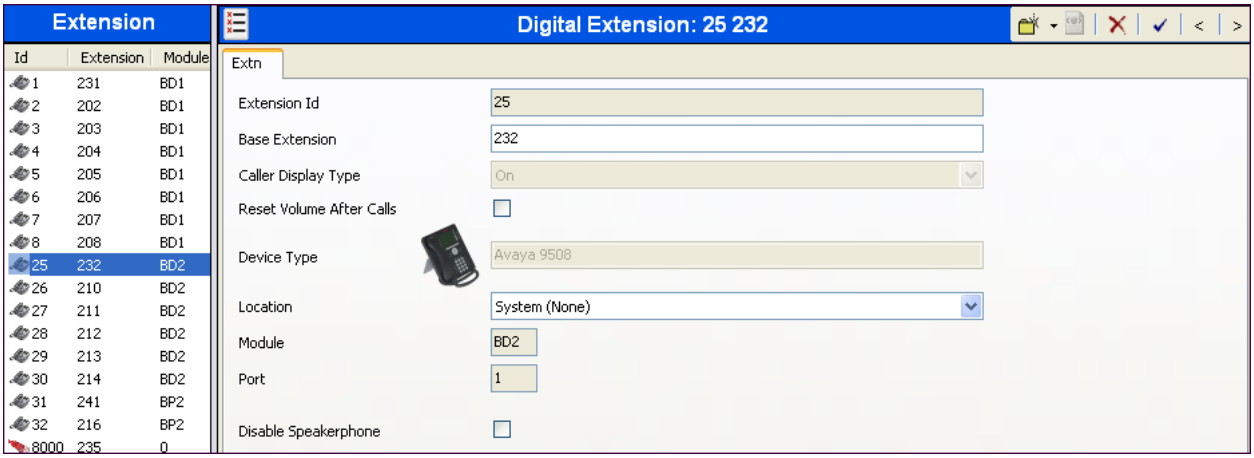
As described in **Section 5.2.3**, names can be entered in directories to allow IP Office to match the caller ID for incoming calls and display the names from the directory. The following screen

©2014 Avaya Inc. All Rights Reserved.

shows the **Personal Directory** tab for user 232. With the configuration shown below and on the SIP Line in **Section 5.4.2** (where "Favor Directory" is selected), if user 232 receives an inbound Verizon IP Trunk call from the telephone number 3035387006, the phone will display the name "Avaya Lab 1" (along with the number).



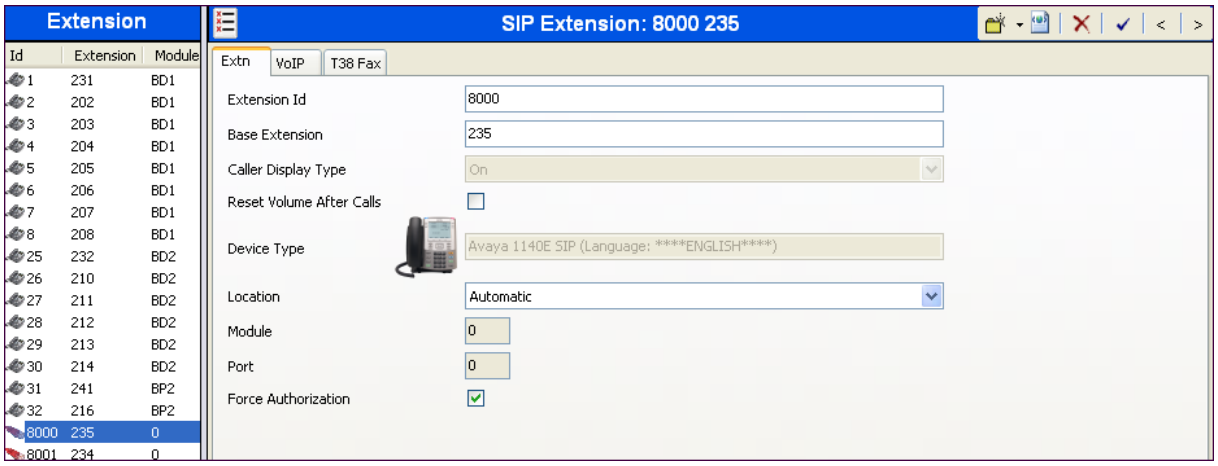
The following screen shows the Extension information for this user. To view, select **Extension** from the Navigation pane, and the appropriate extension from the Group pane.



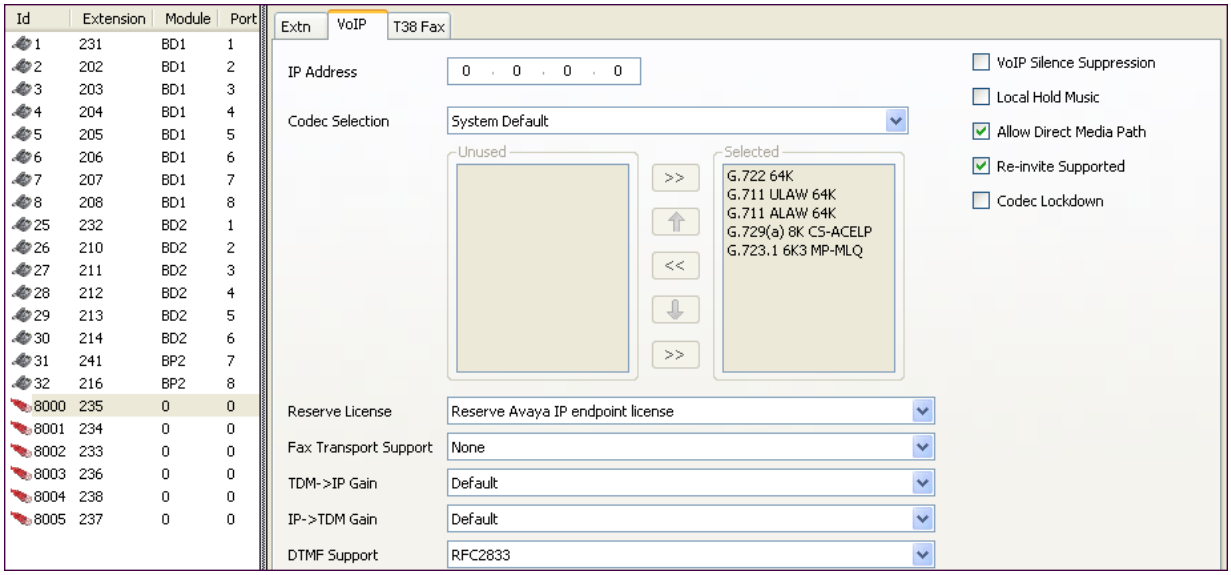
5.5.2. SIP Telephone User (Avaya 1140E)

A new SIP extension may be added by right-clicking on **Extension** in the Navigation pane and selecting **New SIP Extension**. Alternatively, an existing SIP extension may be selected in the group pane. The following screen shows the **Extn** tab for the extension corresponding to an Avaya 1140E. The **Base Extension** field is populated with 235, the extension assigned to the Avaya 1140E. Ensure the **Force Authorization** box is checked.

©2014 Avaya Inc. All Rights Reserved.

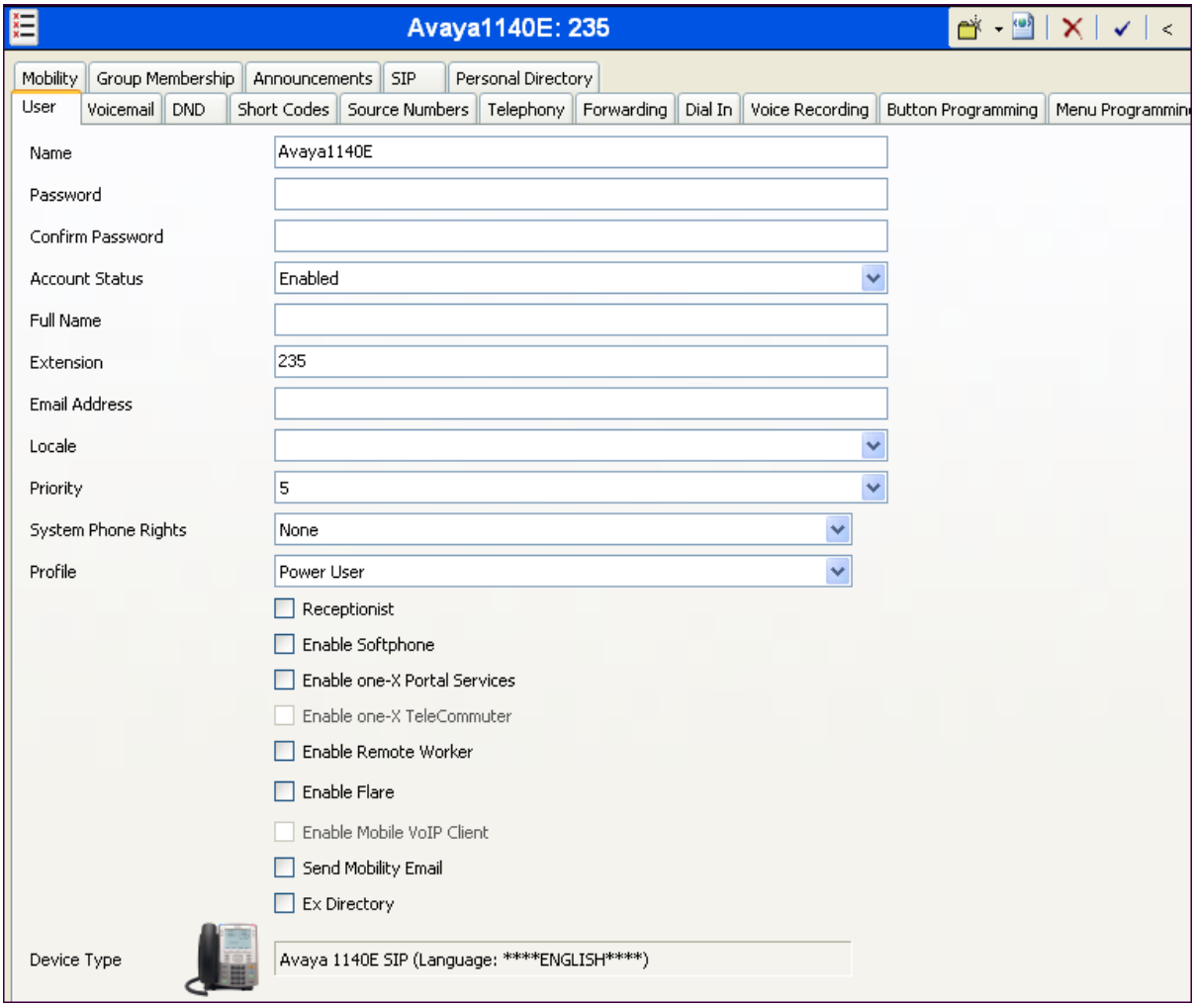


The following screen shows the **VoIP** tab for the extension. The **IP Address** field may be left blank. For the **Reserve License** parameter, select "Reserve Avaya IP endpoint license" from the drop-down box. The **Codec Selection** parameter may retain the default setting "System Default" to follow the system configuration shown in **Section 5.2.5**. Alternatively, "Custom" may be selected to allow the codecs to be configured for this extension, using the arrow keys to select and order the codecs. Ensure the **VoIP Silence Suppression** box is unchecked. See **Section 2.2** for limitations with VoIP Silence Suppression. Other fields may retain default values.



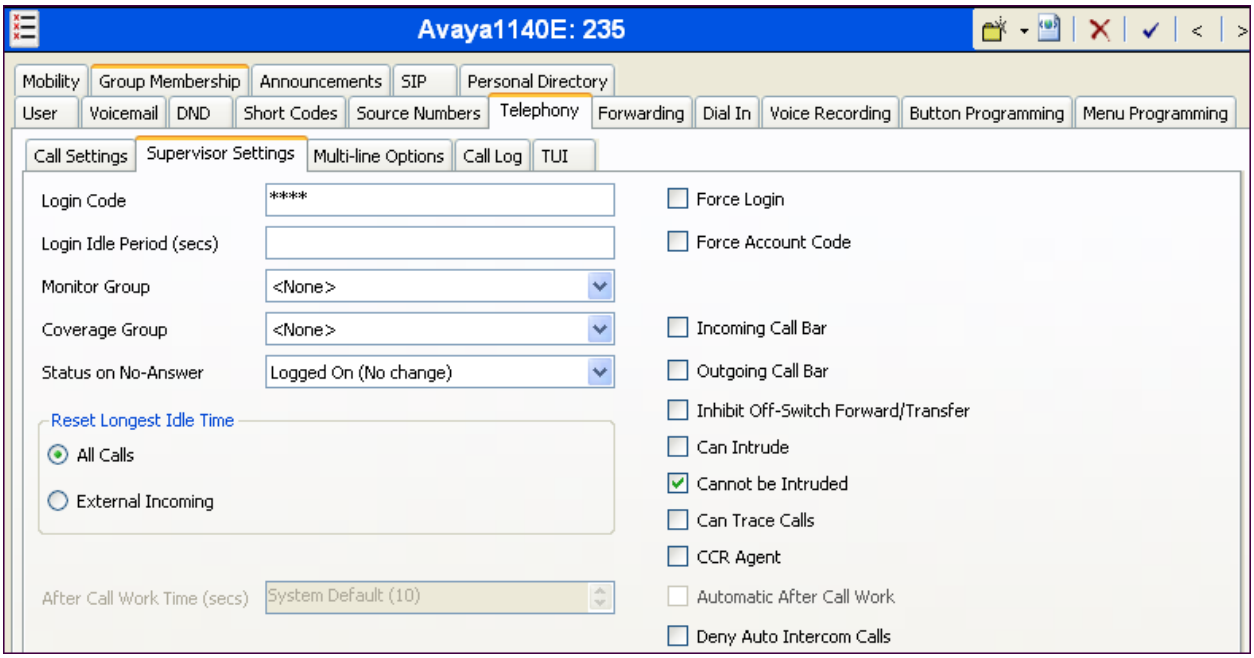
The following screen shows the **User** tab for User 235 corresponding to an Avaya 1140E. The **Extension** parameter is populated with extension 235.

©2014 Avaya Inc. All Rights Reserved.

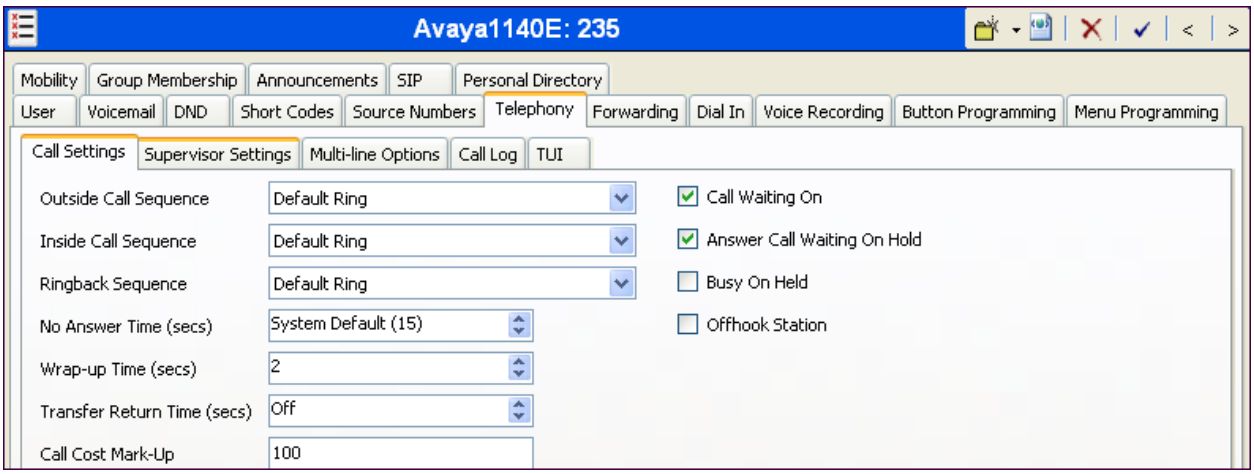


©2014 Avaya Inc. All Rights Reserved.

Select the **Telephony** tab. Then select the **Supervisor Settings** tab as shown below. The **Login Code** will be used by the Avaya 1140E telephone user as the login password.

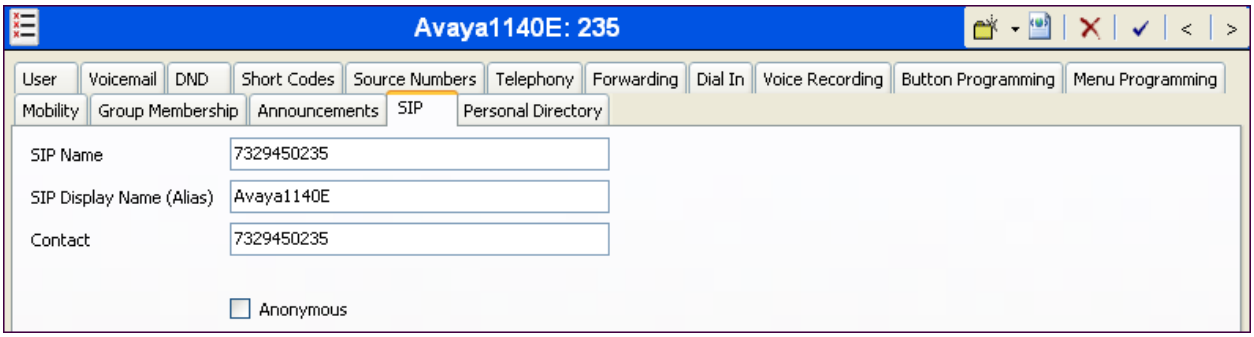


Remaining in the **Telephony** tab for the user, select the **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow multiple call appearances and transfer operations.



©2014 Avaya Inc. All Rights Reserved.

Like other users previously illustrated, the **SIP** tab for the user with extension 235 is configured with a **SIP Name** and **Contact** specifying the user's Verizon IP Trunk service DID number.

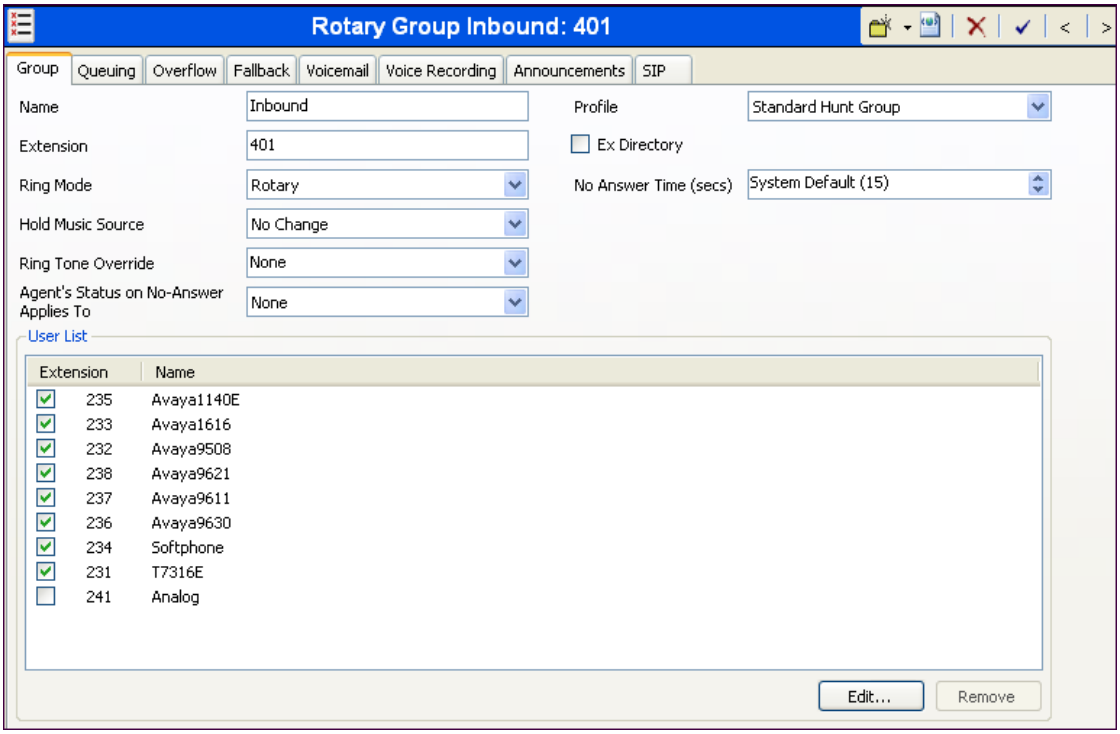


©2014 Avaya Inc. All Rights Reserved.

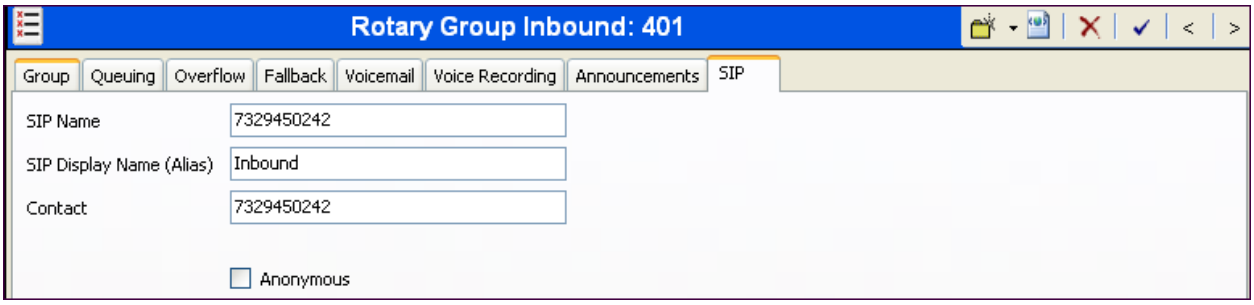
5.5.3. Hunt Groups

During the verification of these Application Notes, users could also receive incoming calls as members of a hunt group. To configure a new hunt group, right-click **Group** from the Navigation pane, and select **New**. To view or edit an existing hunt group, select **Group** from the Navigation pane, and the appropriate hunt group from the Group pane.

The following screen shows the **Group** tab for hunt group 401. The telephone extensions in the **User List** are rung in order, one after the other. However, the last extension used is remembered. The next call received rings the next extension in the list, due to the **Ring Mode** setting "**Rotary**". Click the **Edit** button to change the **User List**.



The following screen shows the **SIP** tab for hunt group 401. The **SIP Name** and **Contact** are configured with Verizon DID 7329450242. Later, in **Section 5.7**, an Incoming Call Route will map 7329450242 to this hunt group based on the information entered on this tab.



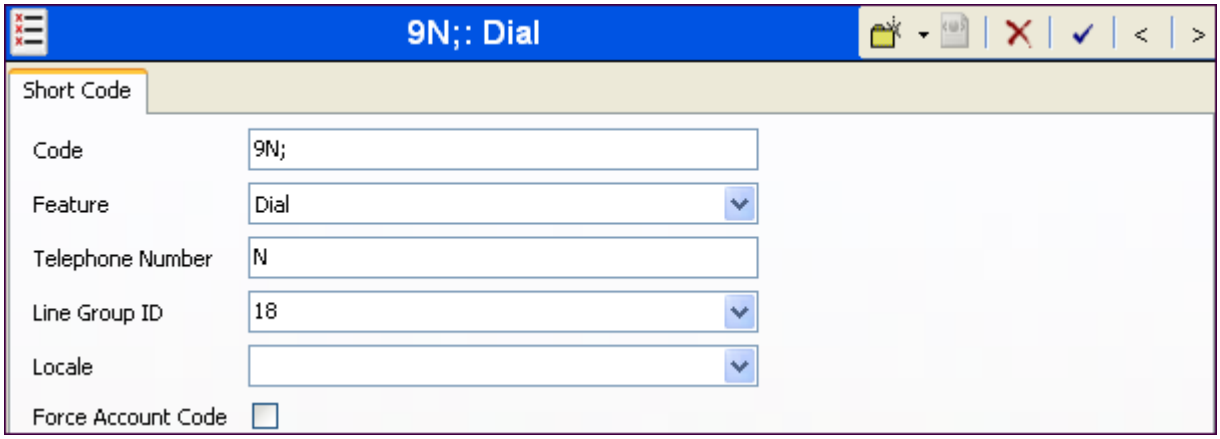
©2014 Avaya Inc. All Rights Reserved.

5.6. Short Codes

In this section, various examples of IP Office short codes will be illustrated. To add a short code, right click on **Short Code** in the Navigation pane, and select **New**. To edit an existing short code, click **Short Code** in the Navigation pane, and the short code to be configured in the Group pane.

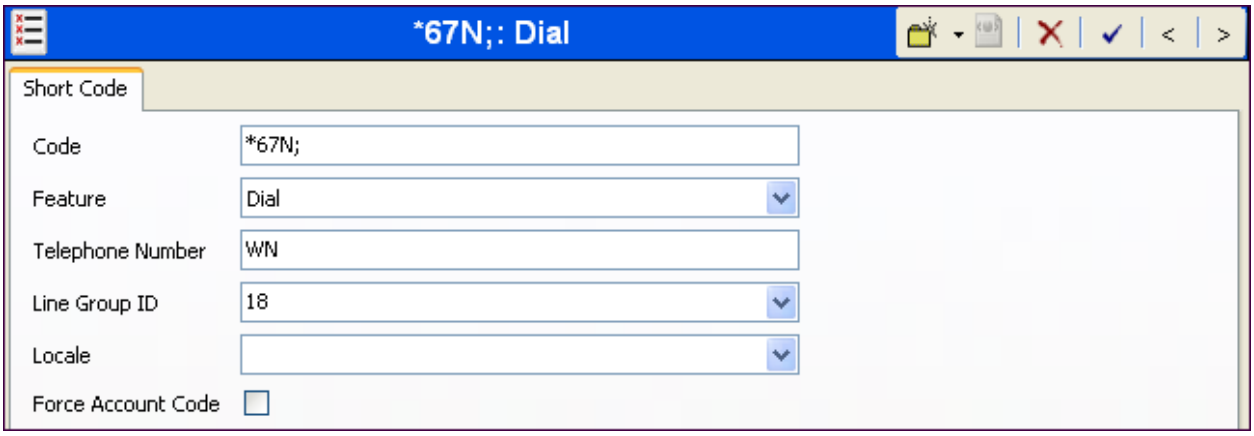
In the screen shown below, the short code "**9N;**" is illustrated. The **Code** parameter is set to "**9N;**". The **Feature** parameter is set to "**Dial**". The **Telephone Number** parameter is set to "**N**". The **Telephone Number** field is used to construct the Request URI and To Header in the outgoing SIP INVITE message. The **Line Group ID** parameter is set to "**18**", matching the number of the **Outgoing Group** configured on the **SIP URI** tab of SIP Line 18 to Avaya SBCE (**Section 5.4.4**).

This simple short code will allow an IP Office user to dial the digit 9 followed by any telephone number, symbolized by the letter N, to reach the SIP Line to Verizon business. "N" can be any number such as a 10-digit number, a 1+10 digit number, a toll free number, directory assistance (e.g., 411), etc..

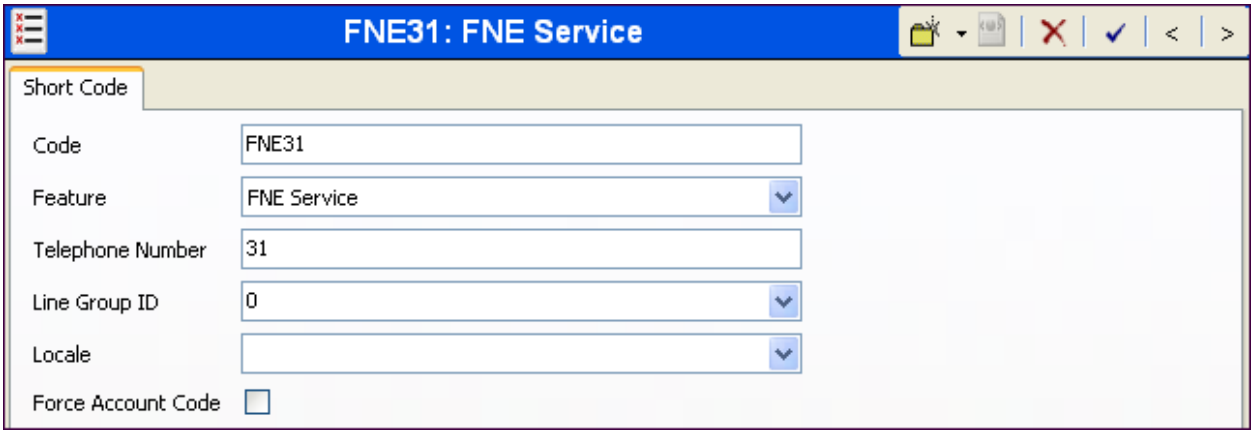


©2014 Avaya Inc. All Rights Reserved.

Optionally, add or edit a short code that can be used to access the SIP Line anonymously. In the screen shown below, the short code "***67N;**" is illustrated. This short code is similar to the "9N;" short code except that the **Telephone Number** field begins with the letter "W", which means "withhold the outgoing calling line identification". In the case of the SIP Line connecting to Verizon documented in these Application Notes, when a user dials *67 plus any number "N", IP Office will include the user's telephone number in the P-Asserted-Identity (PAI) header along with "Privacy: Id". Verizon will allow the call due to the presence of a valid DID in the PAI header, but will prevent presentation of the caller id to the called PSTN destination.



The following screen illustrates a short code that acts like a feature access code rather than a means to access a SIP Line. In this case, the **Code** "**FNE31**" is defined for **Feature** "**FNE Service**" to **Telephone Number** "**31**" (Mobile Call Control). This short code will be used as means to allow a Verizon DID to be programmed to route directly to this feature, via inclusion of this short code as the destination of an Incoming Call Route. See **Section 5.7**. This feature is used to provide dial tone to twinned mobile devices (e.g., cell phone) directly from IP Office; once dial tone is received the user can perform dialing actions including making calls and activating Short Codes.

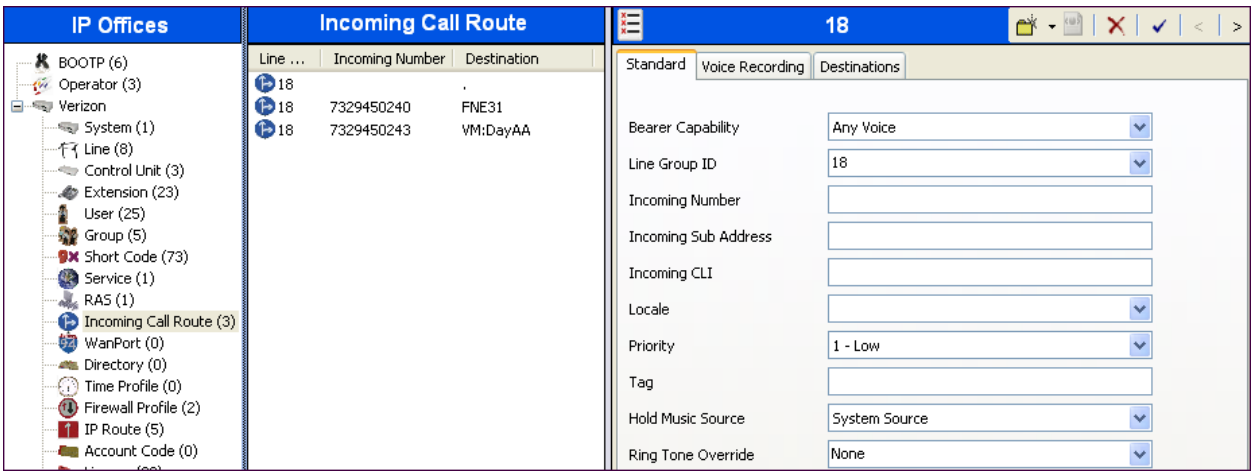


©2014 Avaya Inc. All Rights Reserved.

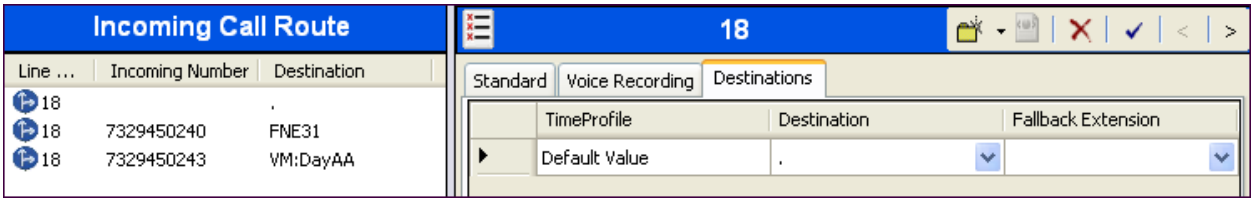
5.7. Incoming Call Routes

In this section, IP Office Incoming Call Routes are illustrated. To add an incoming call route, right click on **Incoming Call Route** in the Navigation pane, and select **New**. To edit an existing incoming call route, select **Incoming Call Route** in the Navigation pane, and the appropriate incoming call route to be configured in the Group pane.

In the screen shown below, a simple incoming call route is illustrated. The **Line Group Id** is "**18**", matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to Verizon Business in **Section 5.4.4**. The **Incoming Number** field is left blank to match all details of the To header.

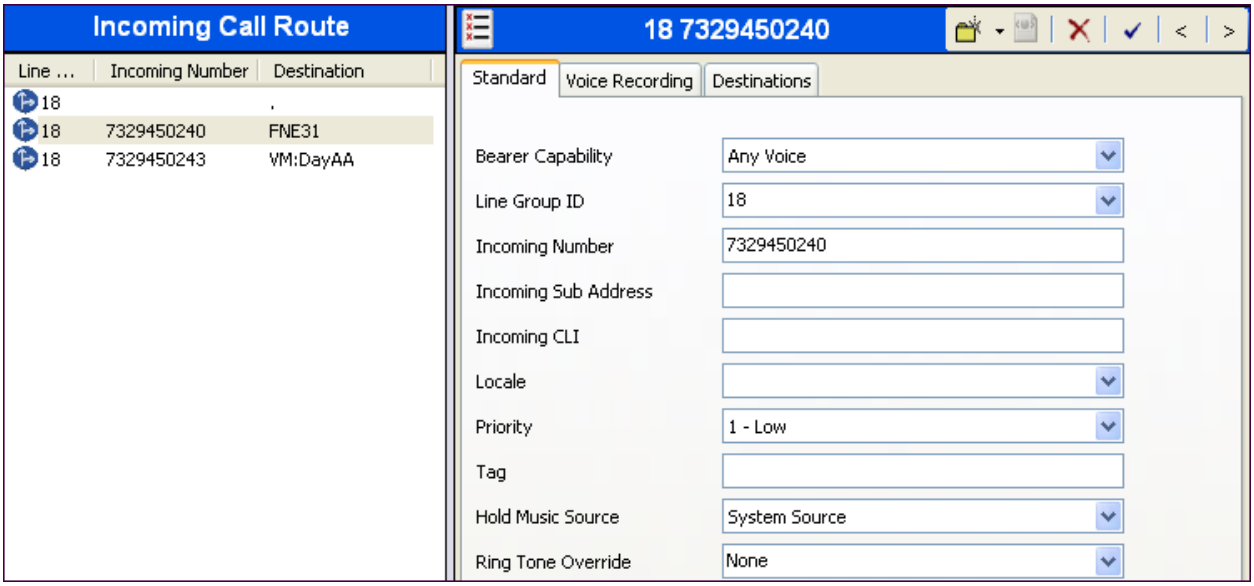


The following **Destinations** tab for the incoming call route contains the **Destination** "**.**" entered manually. This will match the **Incoming Number** field as the Destination and route the call based on the information in the SIP tab for the User or hunt group as illustrated in **Section 5.5**.

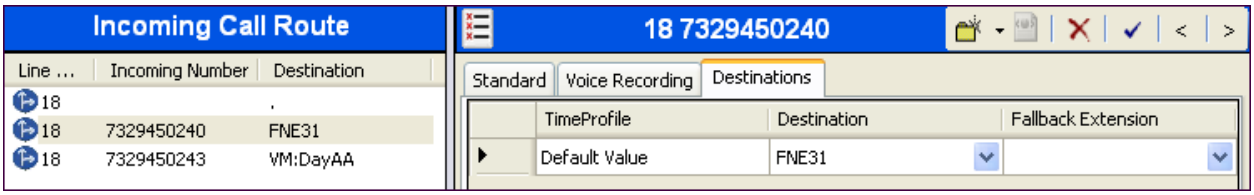


©2014 Avaya Inc. All Rights Reserved.

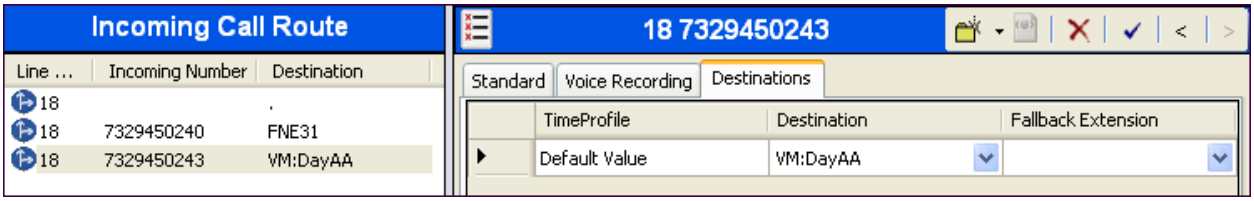
In the screen shown below, the incoming call route for **Incoming Number** "**7329450240**" is illustrated. The **Line Group Id** is "**18**", matching the Incoming Group field configured in the **SIP URI** tab for the SIP Line to Verizon Business in **Section 5.4.4**.



The following **Destinations** tab for the incoming call route contains the **Destination** "**FNE31**" entered manually. The name "FNE31" is the short code for accessing the "Mobile Call Control" application and 732-945-0240 was configured in **Section 5.4.4** on the SIP URI tab as an incoming number. An incoming call to 732-945-0240 will be delivered directly to internal dial tone from the IP Office, allowing the caller to perform dialing actions including making calls and activating Short Codes. The incoming caller ID must match the Twinned Mobile Number entered in the User Mobility tab in **Section 5.5.1**; otherwise the IP Office responds with a "486 Busy Here" and the caller will hear a busy tone



Similarly, the following **Destinations** tab for an incoming call route contains the **Destination** "**VM:DayAA**" entered manually. An incoming call to 732-945-0243 will be delivered directed to the Voicemail Pro Module "DayAA".

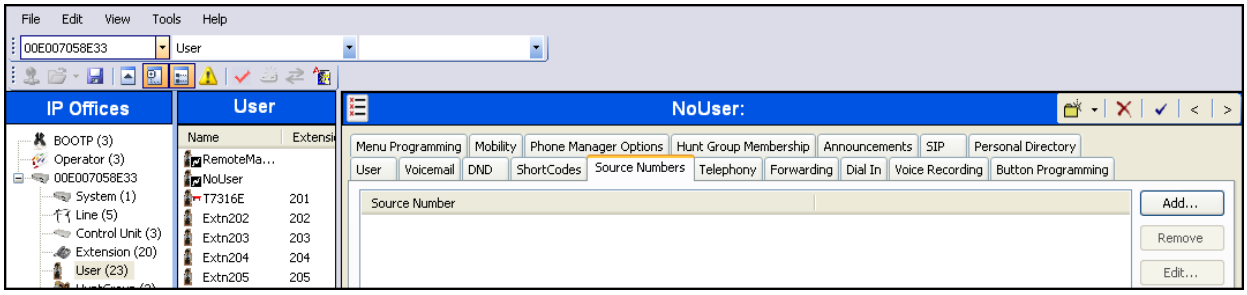


©2014 Avaya Inc. All Rights Reserved.

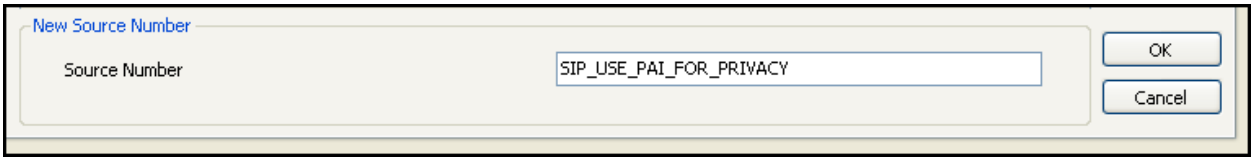
5.8. Privacy/Anonymous Calls

For outbound calls with privacy (anonymous) enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with "restricted" and "anonymous" respectively. Avaya IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing. By default, Avaya IP Office will use PPI for privacy. For the compliance test, PAI was used for the purposes of privacy.

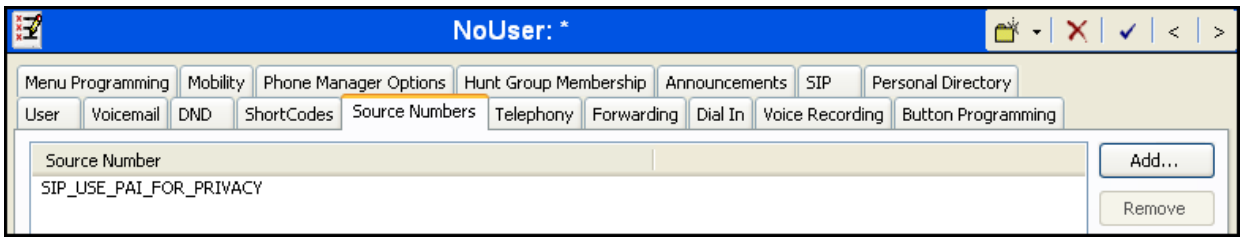
To configure Avaya IP Office to use PAI for privacy calls, navigate to **User** in the Navigation Pane, then **NoUser** in the Group Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



At the bottom of the Details Pane, the **Source Number** field will appear. Enter ***SIP_USE_PAI_FOR_PRIVACY***. Click **OK**.



The **SIP_USE_PAI_FOR_PRIVACY** parameter will appear in the list of Source Numbers as shown below. Click **OK** at the bottom of the screen (not shown).

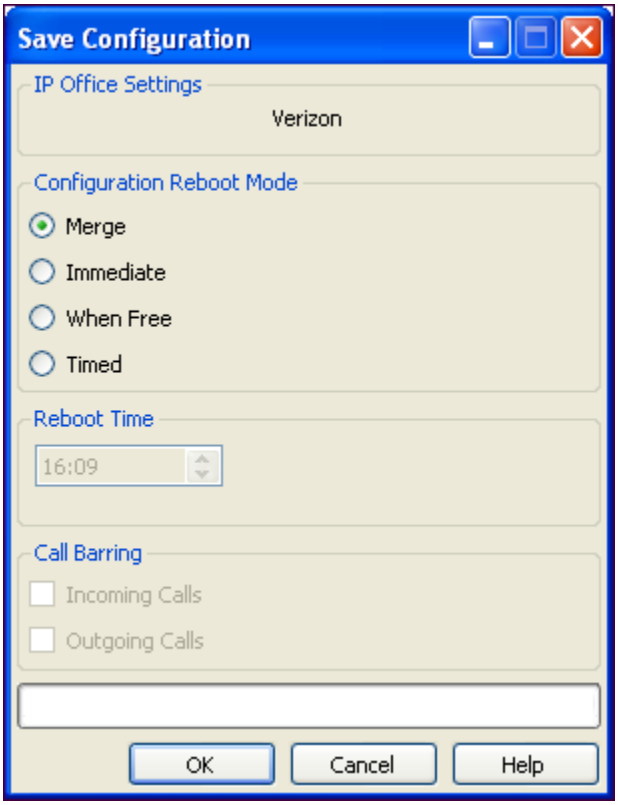


©2014 Avaya Inc. All Rights Reserved.

5.9. Save Configuration

Navigate to **File → Save Configuration** in the menu bar at the top of the screen to save the configuration performed in the preceding sections.

The following will appear, with either **Merge** or **Immediate** selected, based on the nature of the configuration changes made since the last save. Note that clicking **OK** may cause a service disruption. Click **OK** if desired.



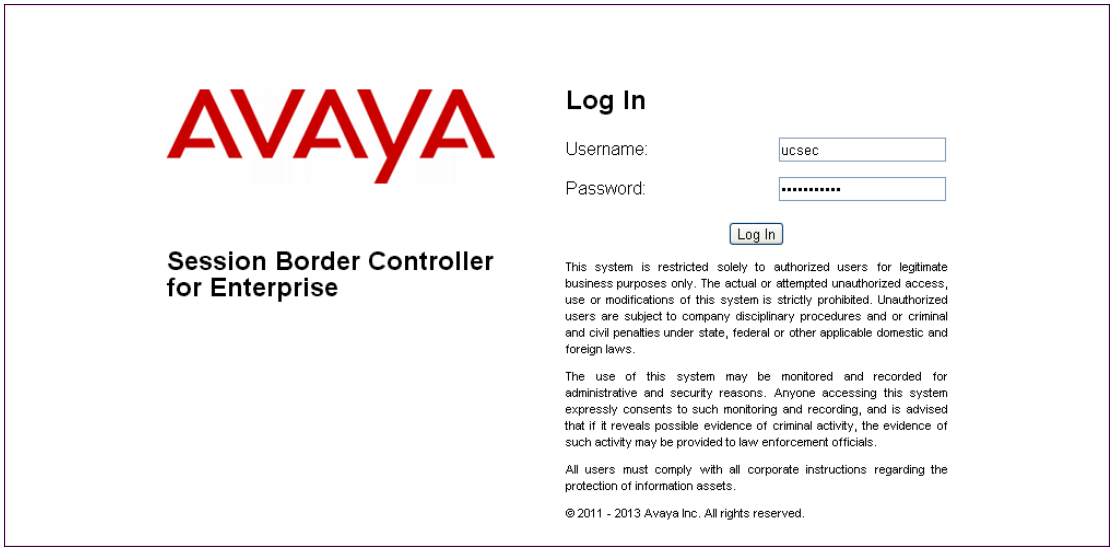
©2014 Avaya Inc. All Rights Reserved.

6. Configure Avaya Session Border Controller for Enterprise

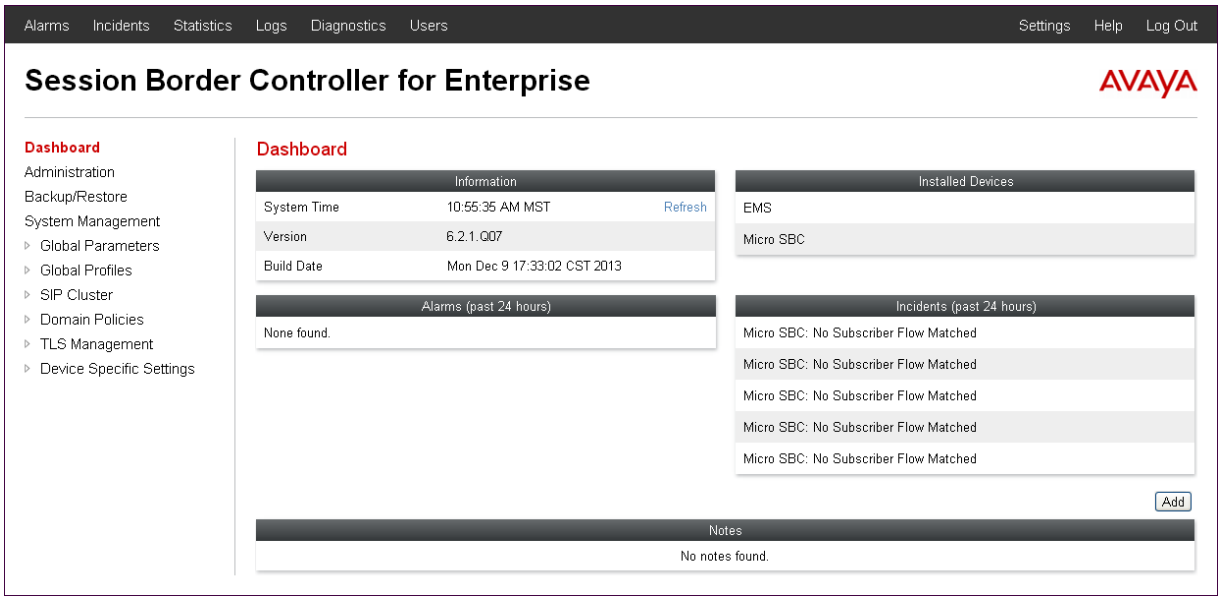
This section covers the configuration of the Avaya SBCE. It is assumed that the Avaya SBCE software has already been installed.

Use a WEB browser to access the Element Management Server (EMS) web interface, and enter https://*ipaddress*/sbc in the address field of the web browser, where *ipaddress* is the management LAN IP address of the Avaya SBCE.

Log in with the appropriate credentials. Click **Log In**.

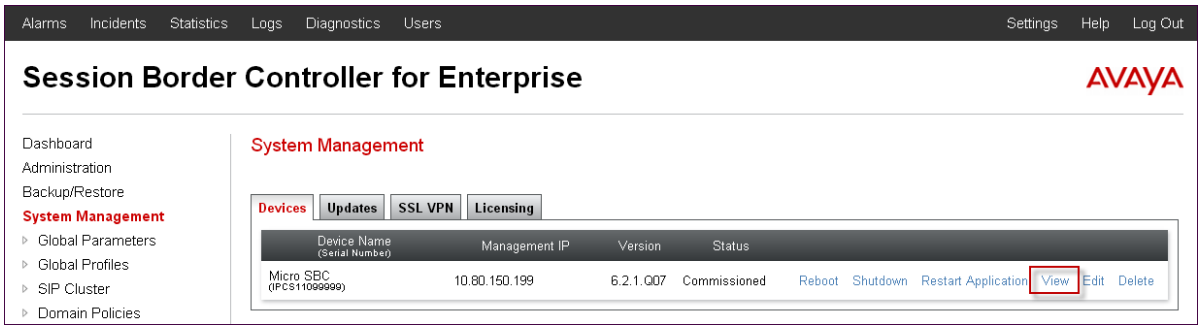


The Dashboard for the Avaya SBCE will appear.

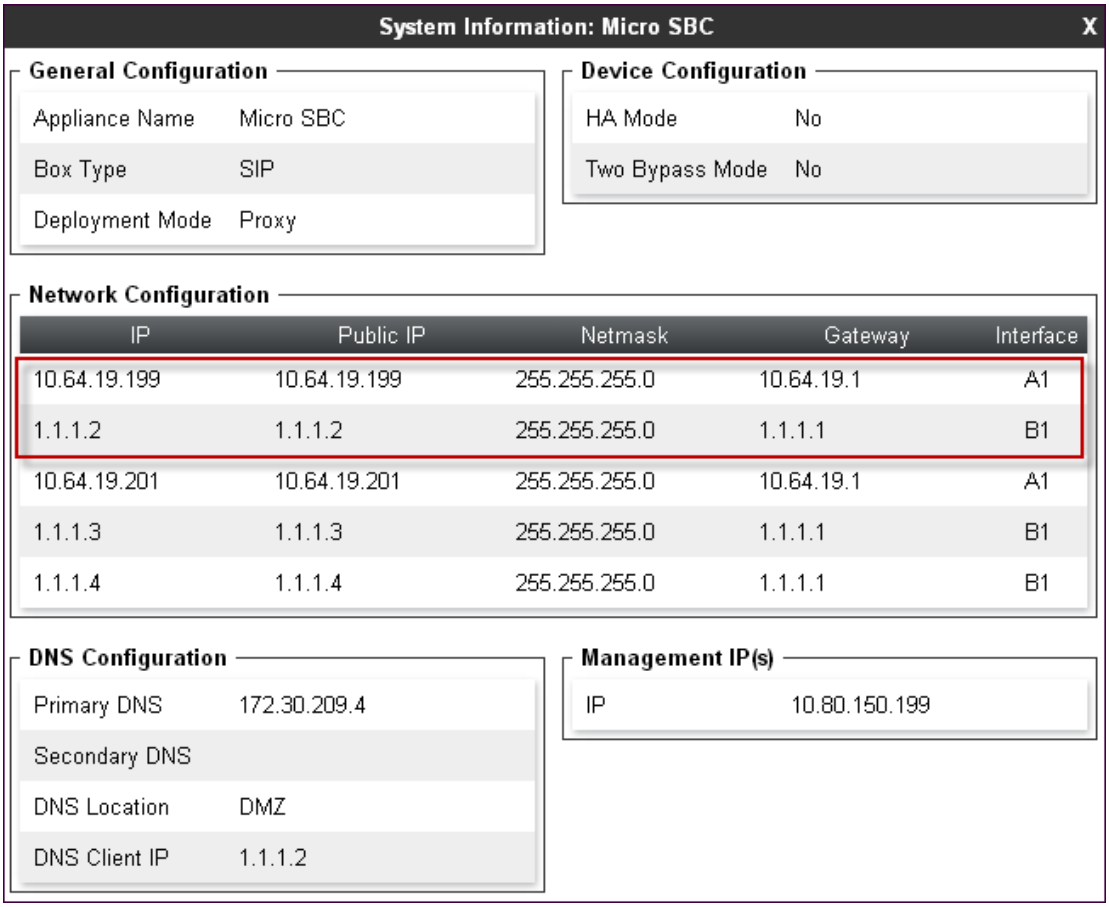


©2014 Avaya Inc. All Rights Reserved.

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



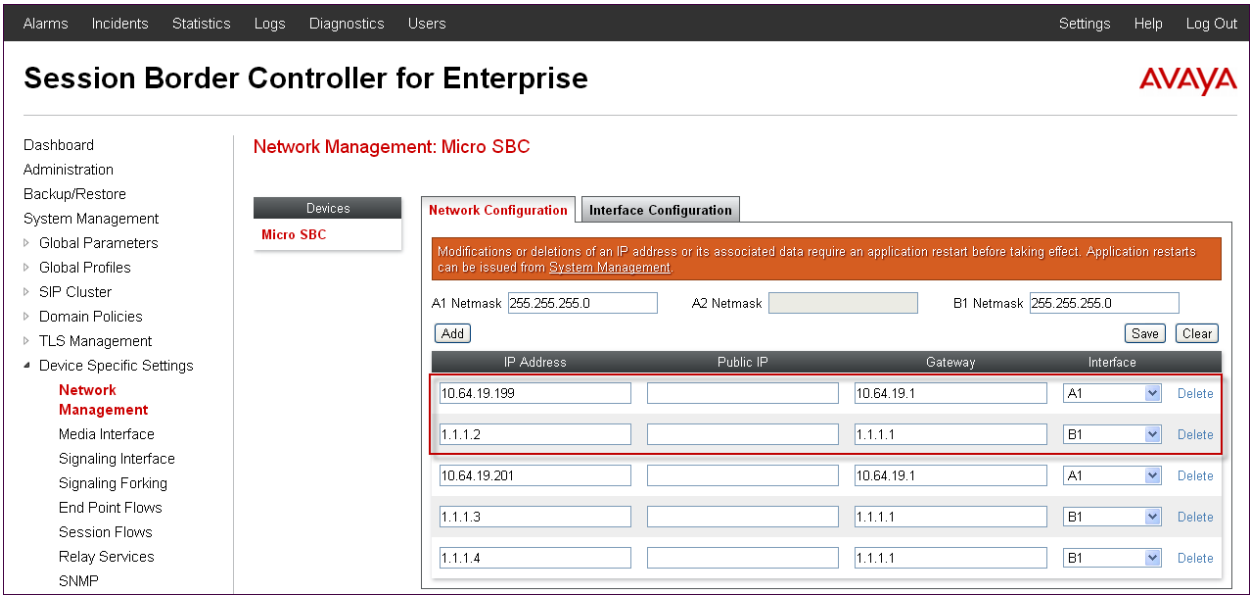
The **System Information** screen shows the **Network Configuration**, **DNS Configuration** and **Management IP(s)** information provided during installation and corresponds to **Figure 1**. The highlighted **A1** and **B1** IP addresses are the ones relevant to the configuration of the SIP trunk to Verizon. Other IP addresses assigned to these interfaces on the screen below are used to support remote workers and they are not discussed in these Application Notes.



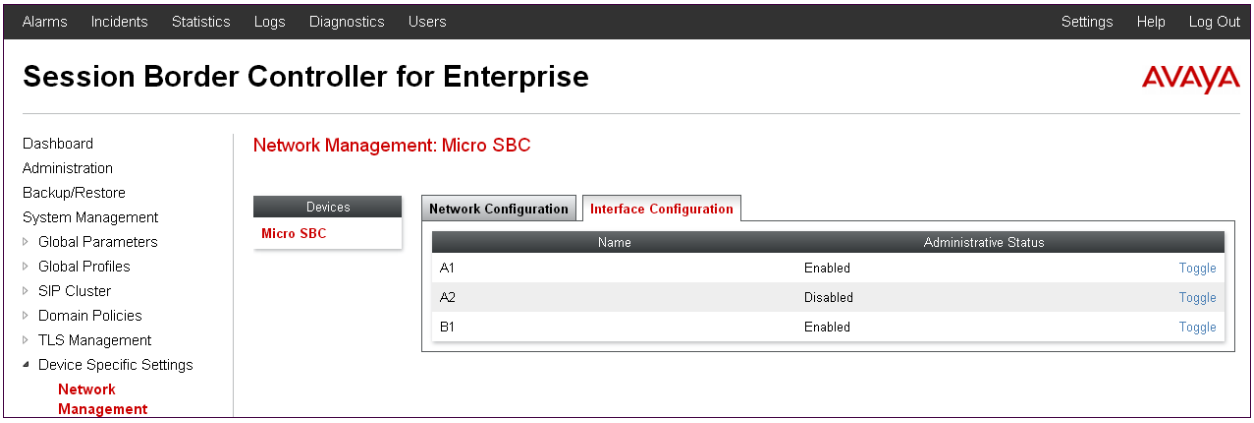
©2014 Avaya Inc. All Rights Reserved.

6.1. Network Management

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



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

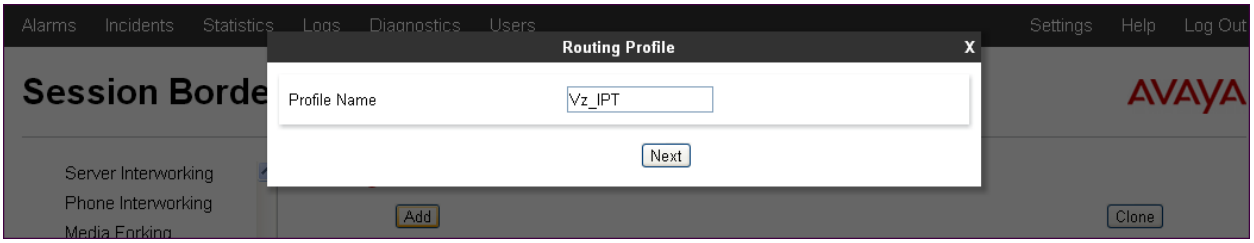


©2014 Avaya Inc. All Rights Reserved.

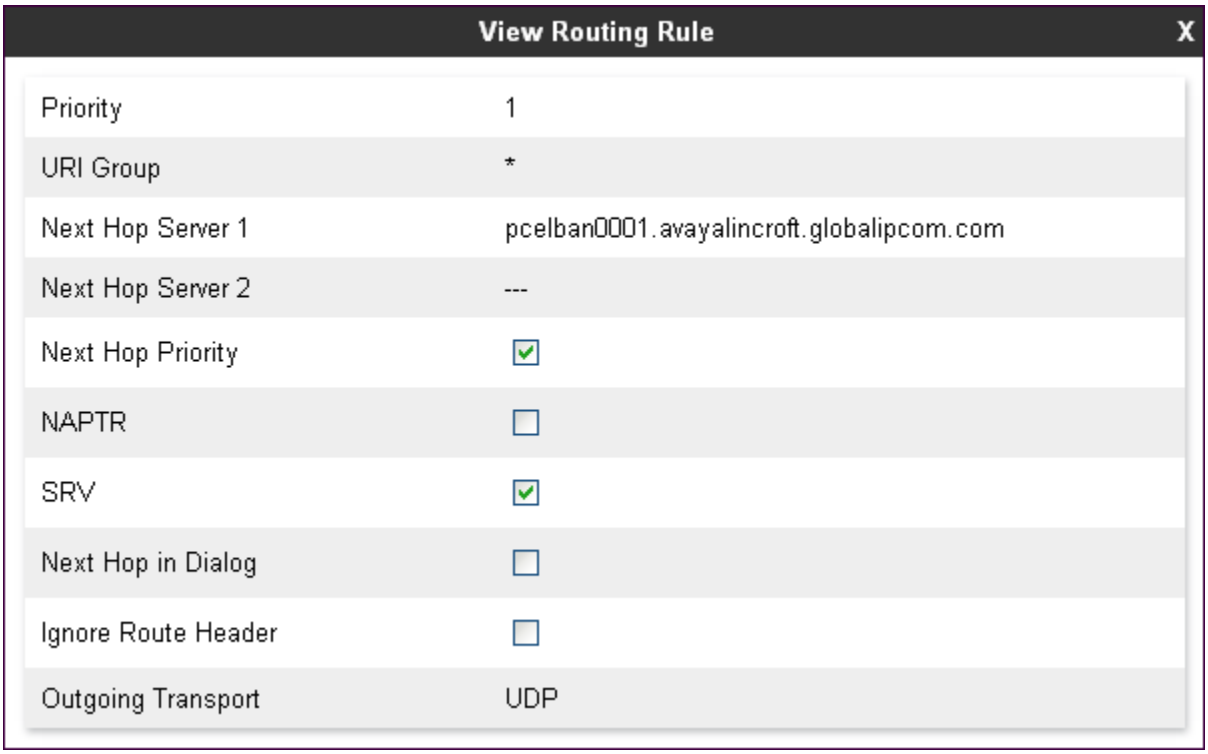
6.2. Routing Profile

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

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

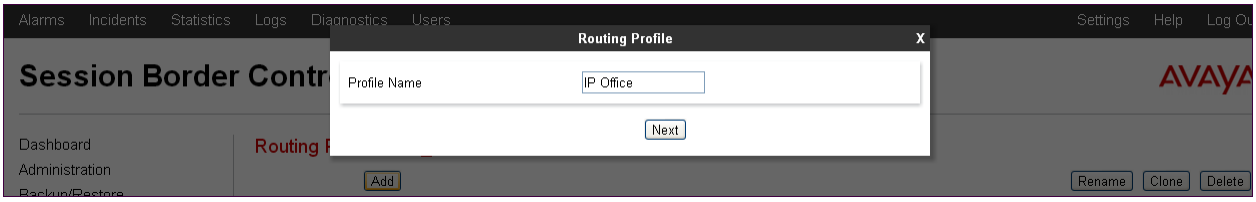


The following screen shows the Routing Profile to Verizon. In the **Next Hop Server 1** field enter the Fully Qualified Domain Name that Verizon uses to listen for SIP traffic. In the sample configuration "**pcelban0001.avayalincroft.globalipcom.com**" is used. Select **SRV** and enter "**UDP**" for the **Outgoing Transport** field.

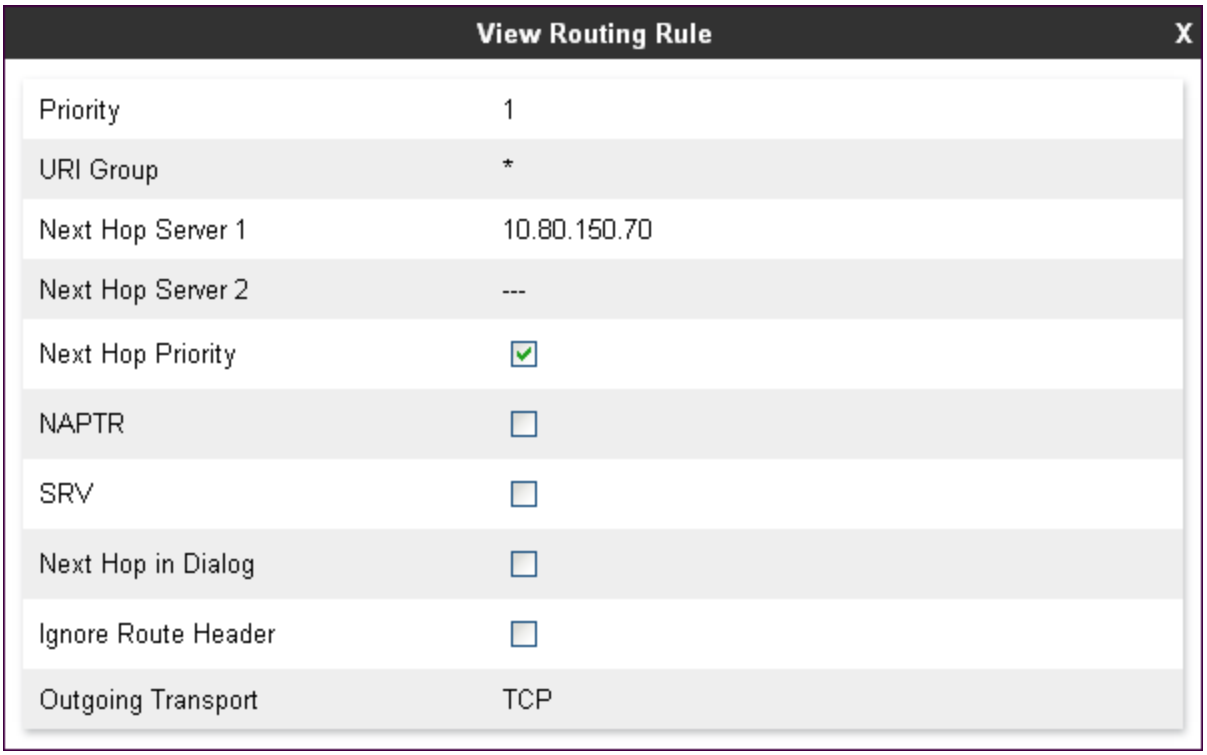


©2014 Avaya Inc. All Rights Reserved.

Similarly add a Routing Profile to IP Office.



The following screen shows the Routing Profile to IP Office. The **Next Hop Server 1** IP address must match the IP address of the IP Office LAN settings entered in **Section 5.2.1**. The **Outgoing Transport** is set to "**TCP**" and matches the **Layer 4 Protocol** set in IP Office **SIP Line →Transport** in **Section 5.4.3**.



View Routing Rule X

Priority	1
URI Group	*
Next Hop Server 1	10.80.150.70
Next Hop Server 2	---
Next Hop Priority	☑
NAPTR	☐
SRV	☐
Next Hop in Dialog	☐
Ignore Route Header	☐
Outgoing Transport	TCP

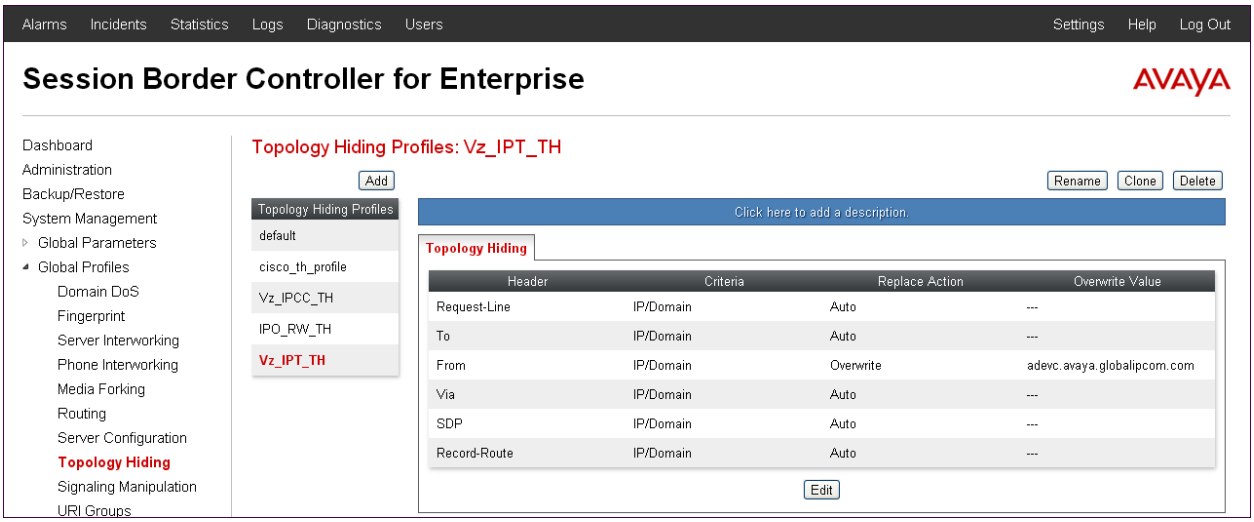
©2014 Avaya Inc. All Rights Reserved.

6.3. Topology Hiding Profile

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

Click the **Add** button to add a new profile, or select an existing topology hiding profile to edit. In the sample configuration, the "**default**" profile was cloned and modified for Verizon.

In the **Replace Action** column an action of **Auto** will replace the header field with the IP address of the Avaya SBCE interface and the **Overwrite** will use the value in the **Overwrite Value**. In the example shown, "**Vz_IPT_TH**" was cloned from the default profile and will later be applied in the direction of Verizon. **Overwrite** is selected for the From header and domain of "**adevc.avaya.globalipcom.com**" is inserted. This is the enterprise domain known to Verizon Business IP Trunk service.



6.4. Server Interworking Profile

The Server Interworking profile configures and manages various SIP call server-specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters (for HA deployments), DoS security statistics, and trusted domains. Interworking Profile features are configured based on different Trunk Servers. There are default profiles available that may be used as is, or new profiles can be configured as described below.

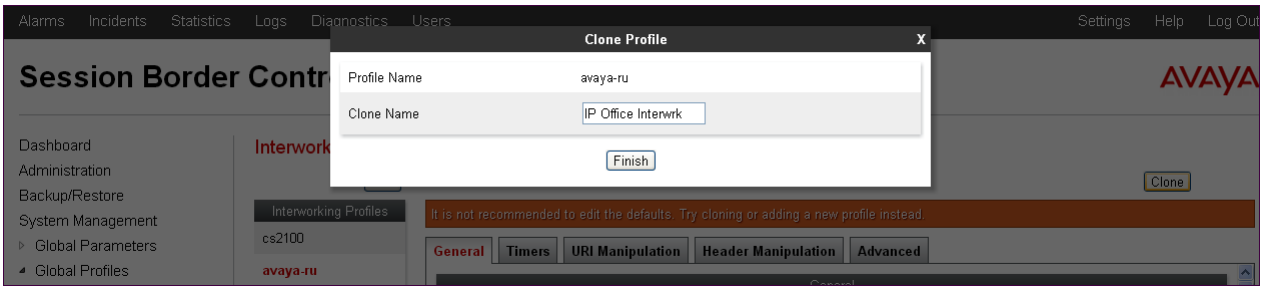
In the sample configuration, separate Server Interworking Profiles were created for IP Office and Verizon Business IP Trunk service.

6.4.1. Server Interworking Profile – IP Office

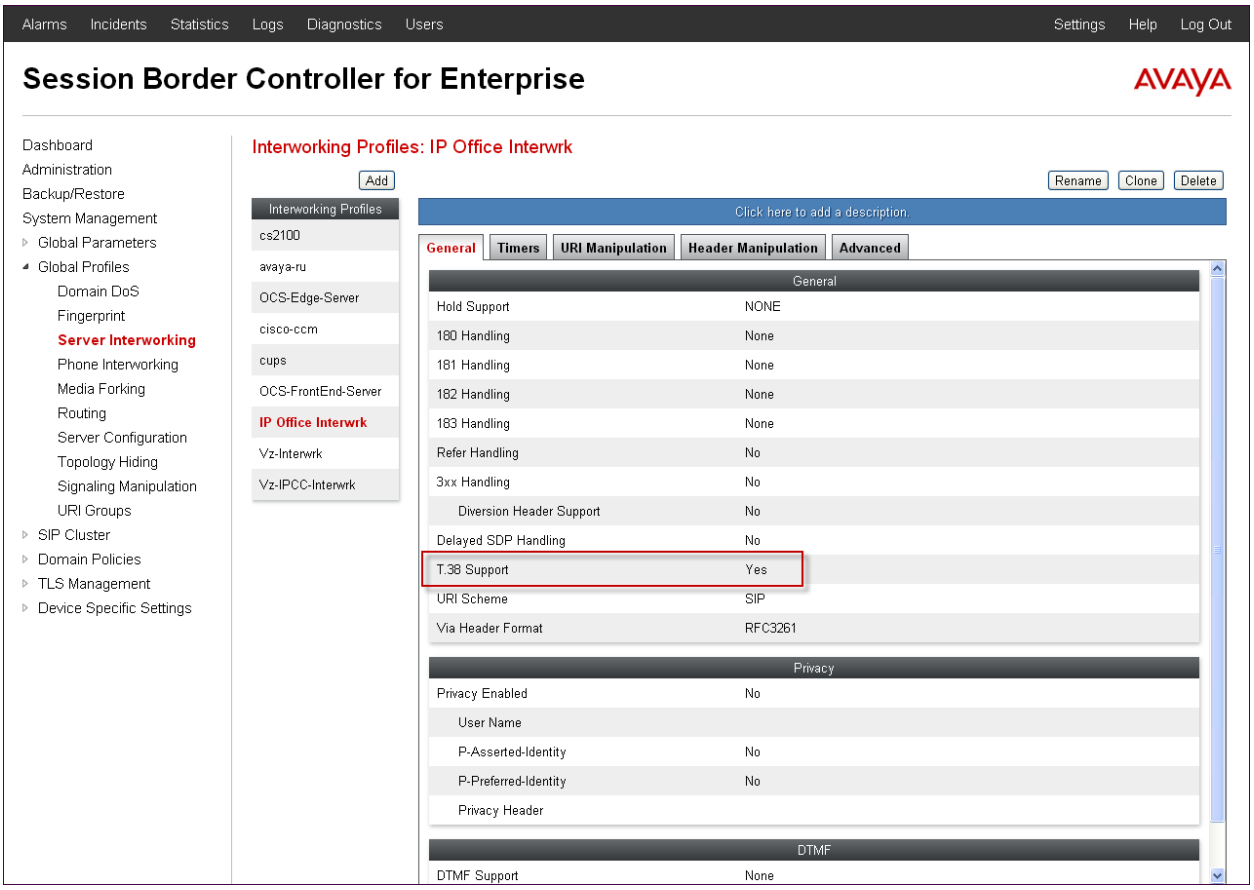
In the sample configuration, the IP Office Server Interworking profile was cloned from the default **avaya-ru** profile. To clone a Server Interworking Profile for IP Office, navigate to

©2014 Avaya Inc. All Rights Reserved.

Global Profiles → Server Interworking, select the **avayu-ru** profile and click the **Clone** button. Enter a **Clone Name** and click **Next** to continue.



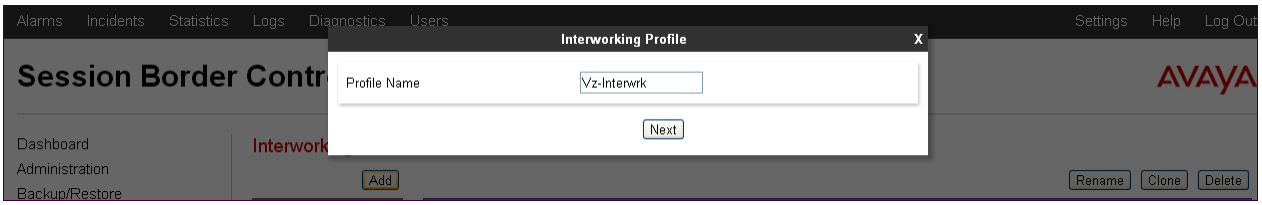
The following screen shows the "**IP Office Interwrk**" profile used in the sample configuration, with **T.38 Support** set to "**Yes**". Default values can be used for all other fields.



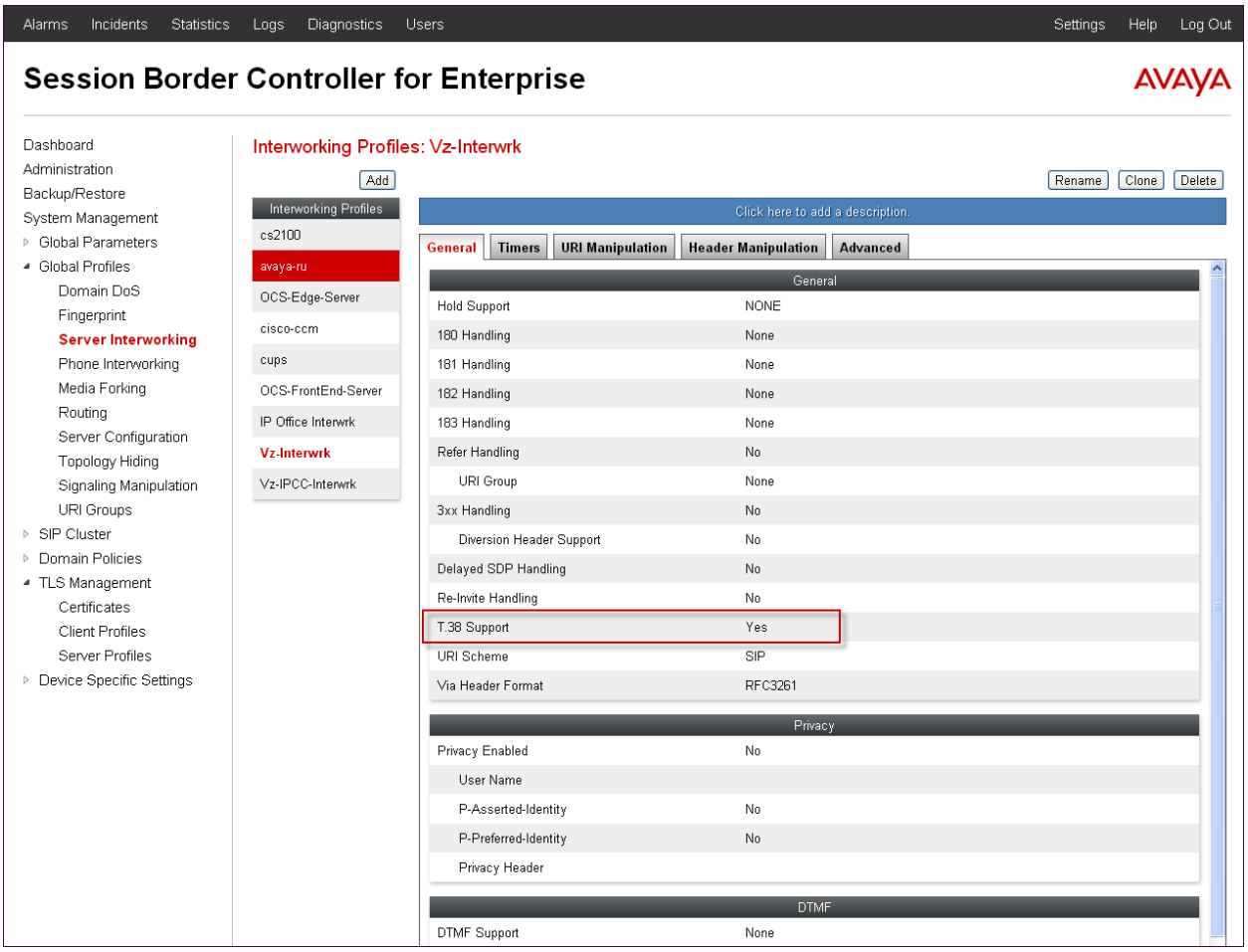
6.4.2. Server Interworking Profile – Verizon

To create a new Server Interworking Profile for Verizon, navigate to **Global Profiles → Server Interworking** and click **Add** as shown below. Enter a **Profile Name** and click **Next**.

©2014 Avaya Inc. All Rights Reserved.

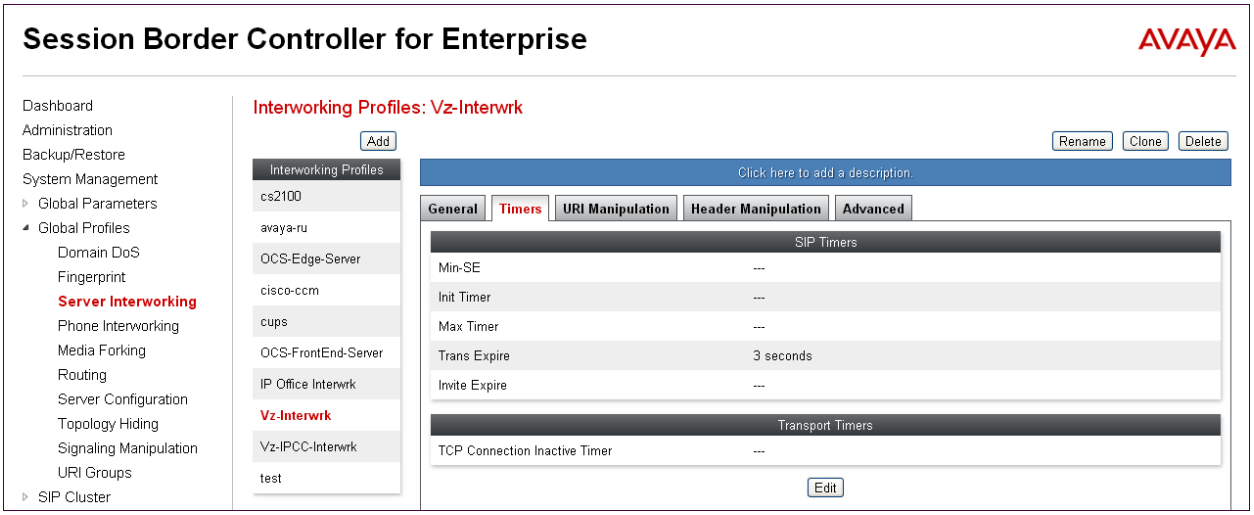


The following screens show the "**Vz-Interwrk**" profile used in the sample configuration. On the **General** tab, default values are used with the exception of **T.38 Support** set to "**Yes**".

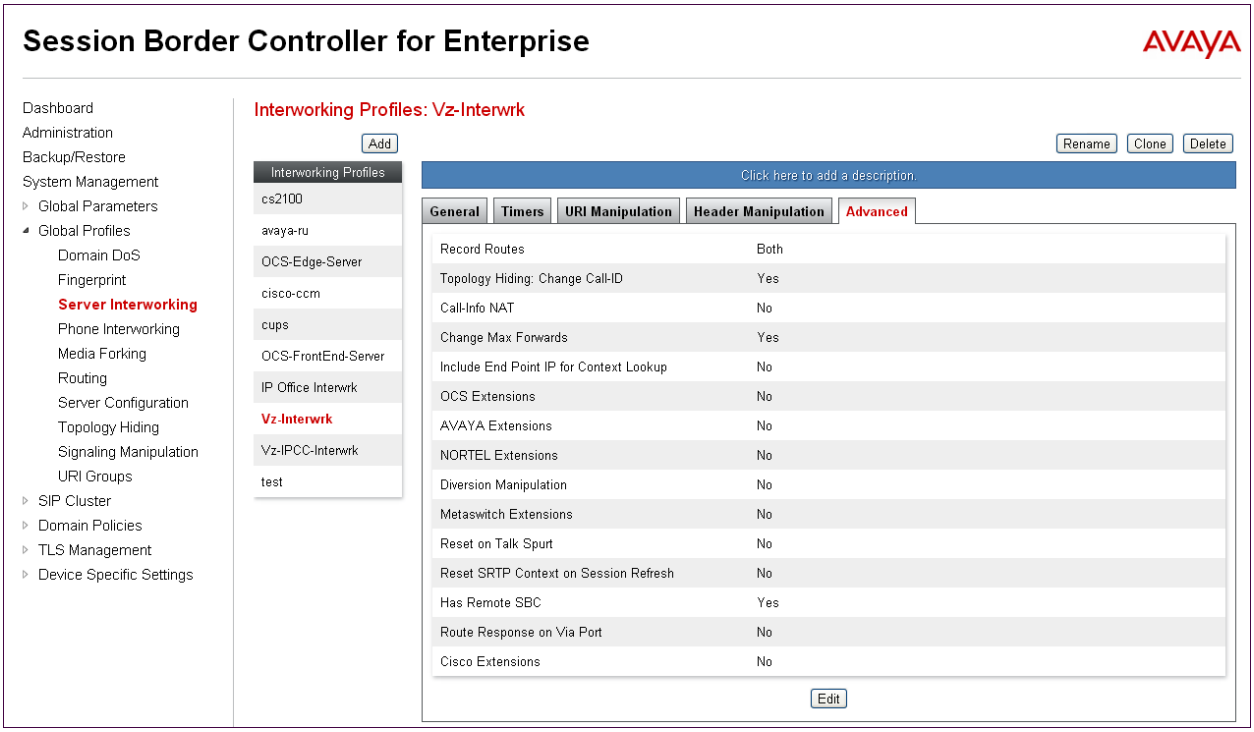


©2014 Avaya Inc. All Rights Reserved.

The **Timers** tab shows the values used for compliance testing for the **Trans Expire** field. The **Trans Expire** timer sets the allotted time the Avaya SBCE will try the first primary server before trying the secondary server.



Advanced tab:



©2014 Avaya Inc. All Rights Reserved.

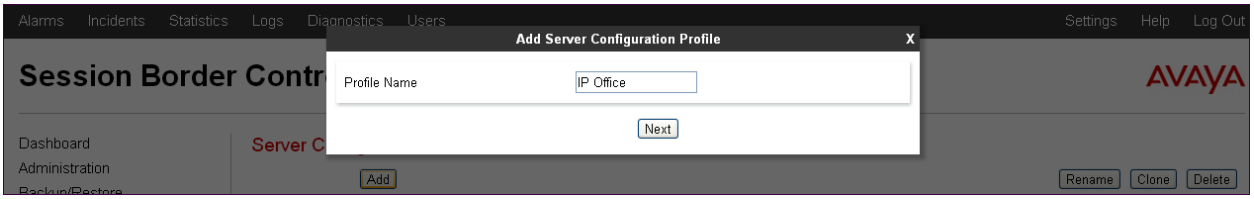
6.5. Server Configuration

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

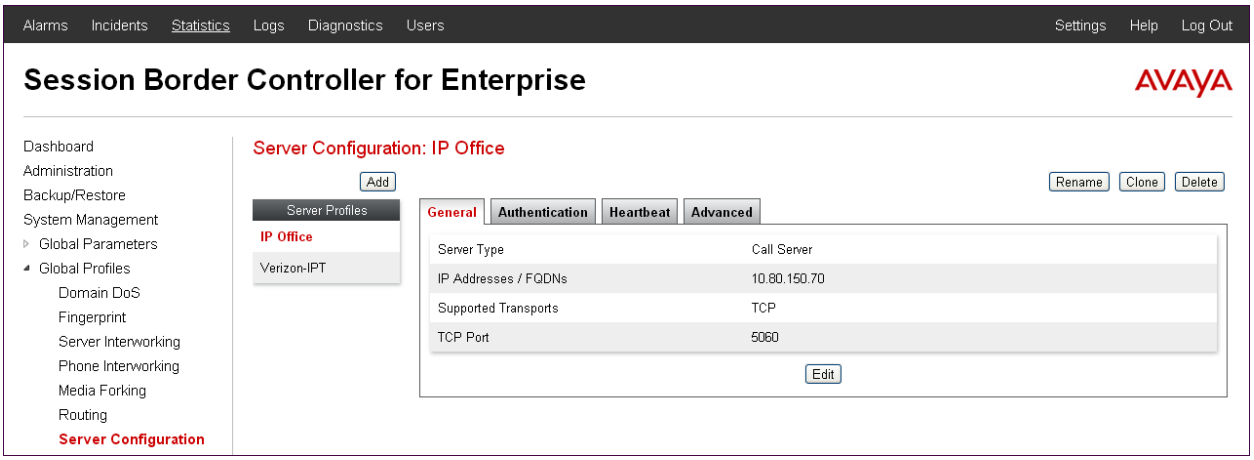
In the sample configuration, separate Server Configurations were created for IP Office and Verizon Business IP Trunk service.

6.5.1. Server Configuration – IP Office

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

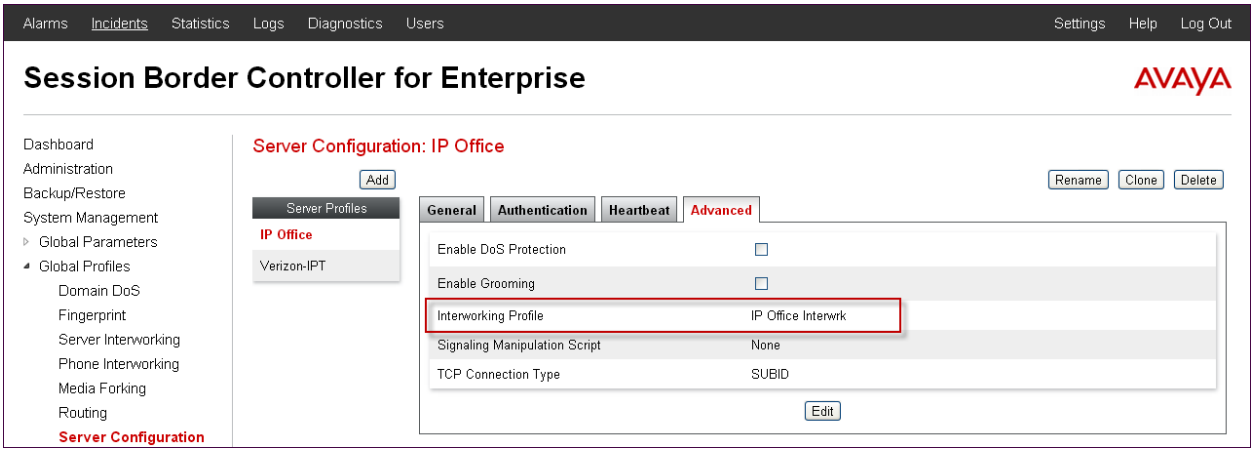


The following screens illustrate the Server Configuration for the Profile name "**IP Office**". In the **General** parameters, the **Server Type** is set to "**Call Server**". In the **IP Addresses / Supported FQDNs** area, the IP Address of the IP Office LAN 1 interface in the sample configuration is entered. This IP address is 10.80.150.70. In the **Supported Transports** area, **TCP** is selected, and the **TCP Port** is set to "**5060**".



©2014 Avaya Inc. All Rights Reserved.

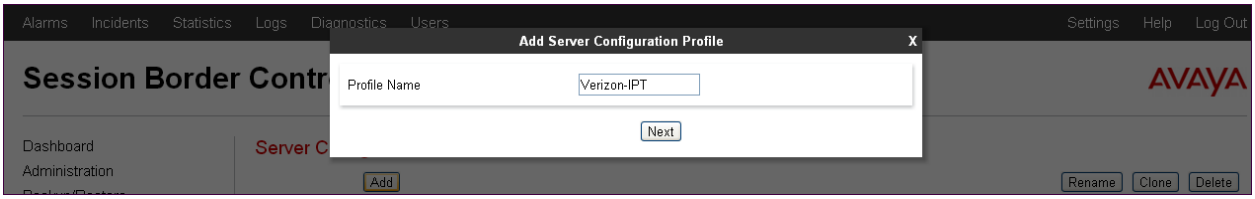
Default values can be used on the **Authentication** and **Heartbeat** tabs (not shown). On the **Advanced** tab, the **Interworking Profile** is set to "**IP Office Interwrk**" created in **Section 6.4.1** for IP Office. If remote workers will not be used, **Enable Grooming** can be selected to allow the same TCP connection to be used for all SIP messages from this device. In the sample configuration, this server configuration is also used for remote workers and IP Office uses different TCP connections to each endpoint, therefore "Grooming" is disabled.



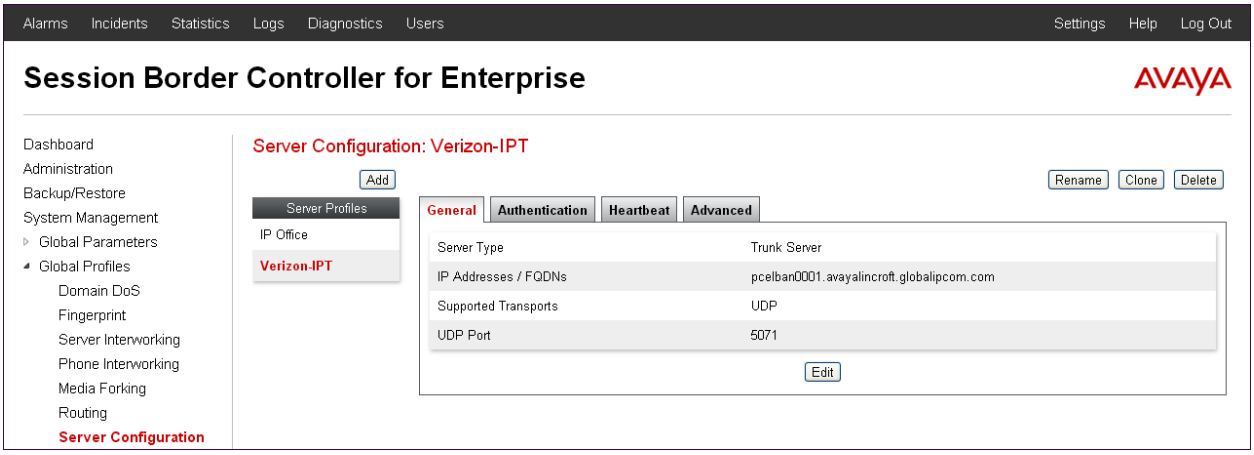
©2014 Avaya Inc. All Rights Reserved.

6.5.2. Server Configuration - Verizon

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

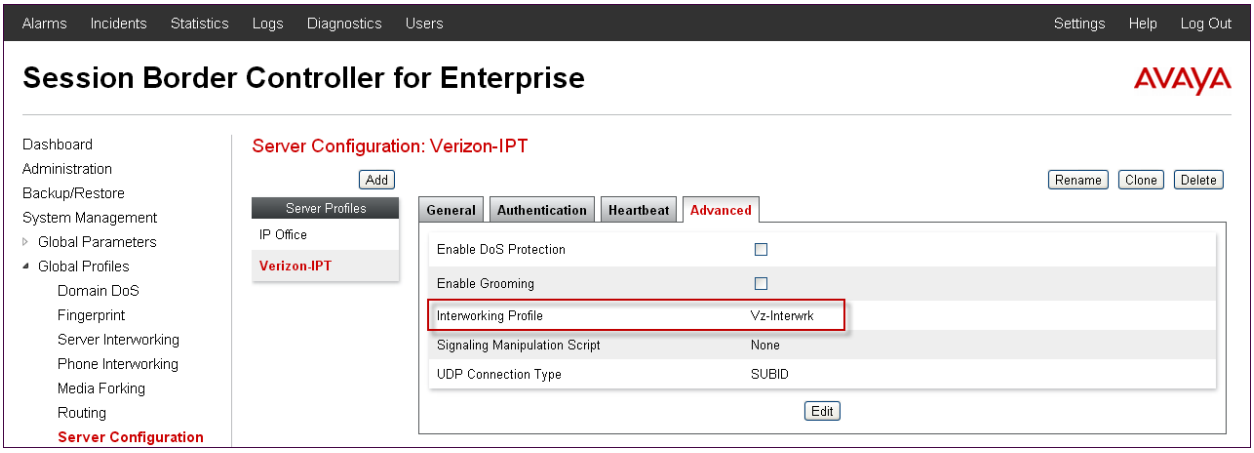


The following screens illustrate the Server Configuration for the Profile name "**Verizon-IPT**". In the **General** parameters, the **Server Type** is set to "**Trunk Server**". In the **IP Addresses / Supported FQDNs** area, the Verizon-provided IP trunk Fully Qualified Domain Name is entered. This is "**pcelban0001.avayalincroft.globalipcom.com**". In the **Supported Transports** area, **UDP** is selected, and the **UDP Port** is set to "**5071**".



©2014 Avaya Inc. All Rights Reserved.

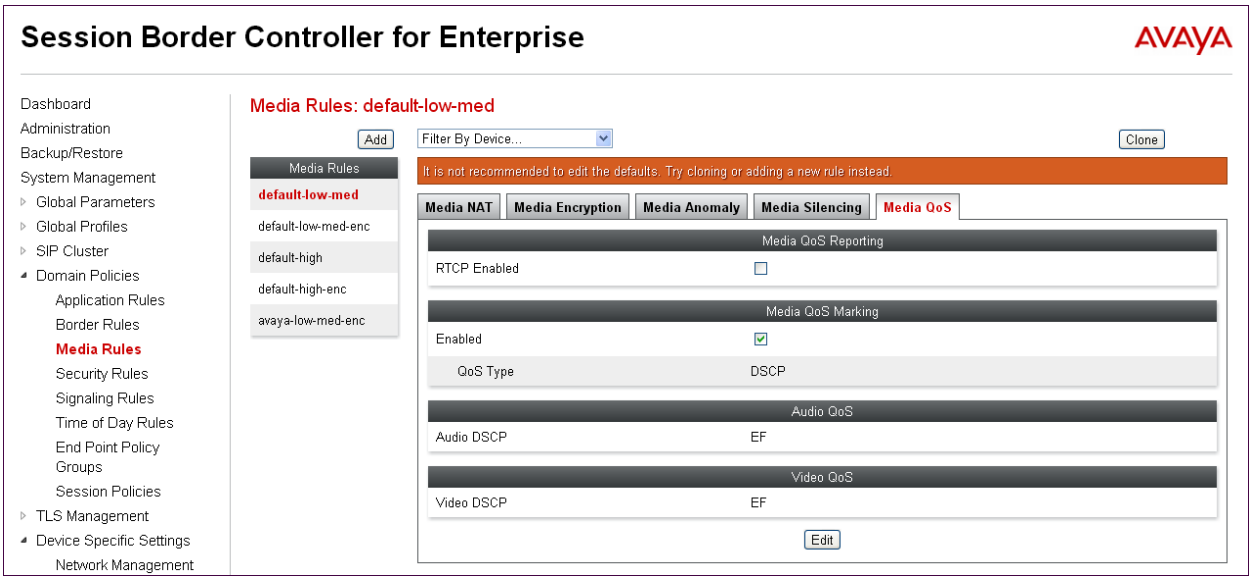
Default values can be used on the **Authentication** and **Heartbeat** tabs (not shown). On the **Advanced** tab, **Enable Grooming** is not used for UDP connections and left unchecked. The **Interworking Profile** is set to "**Vz-Interwrk**" created in **Section 6.4.2** for Verizon.



6.6. Media Rule

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

Select **Domain Policies → Media Rules** from the left-side menu as shown below. In the sample configuration, a single default media rule "**default-low-med**" was used with the DSCP values "**EF**" for expedited forwarding (default value) for **Media QoS** as shown below.

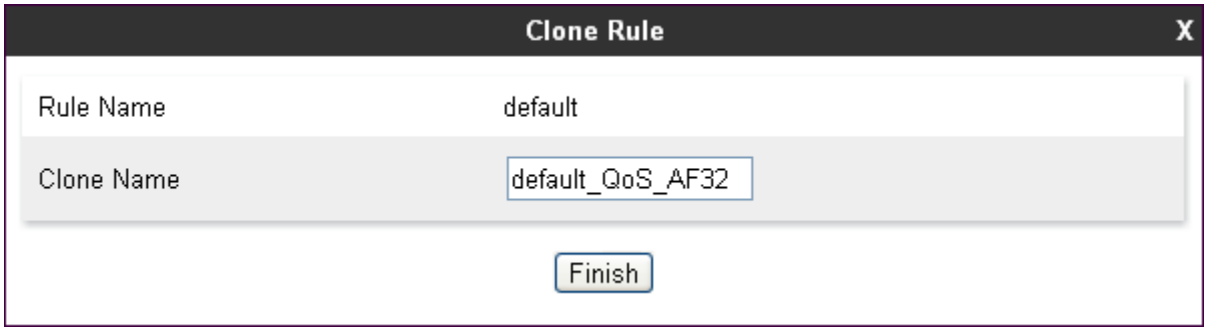


©2014 Avaya Inc. All Rights Reserved.

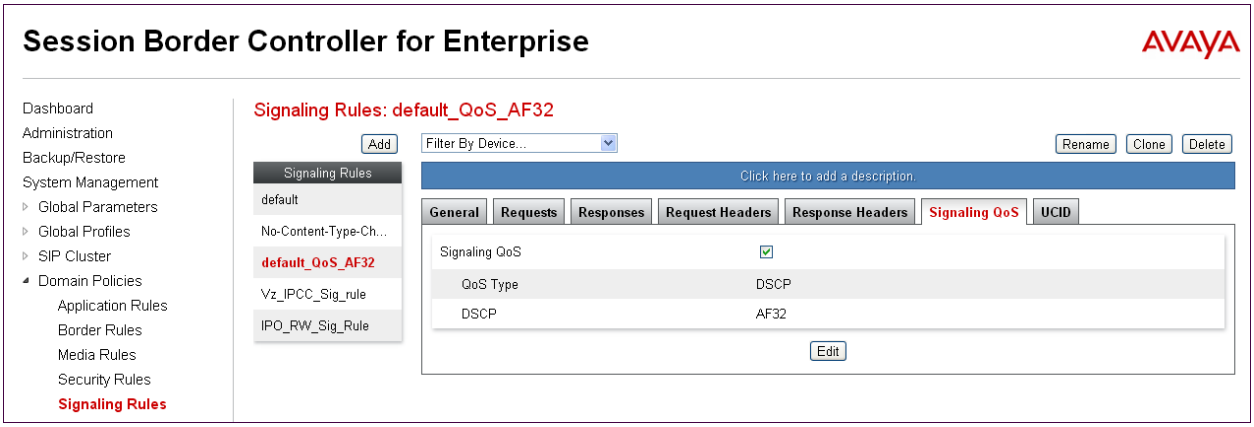
6.7. Signaling Rule

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

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



In the sample configuration, signaling rule "**default_QoS_AF32**" was used with the DSCP values "**AF32**" for assured forwarding set for **Signaling QoS** as shown below.

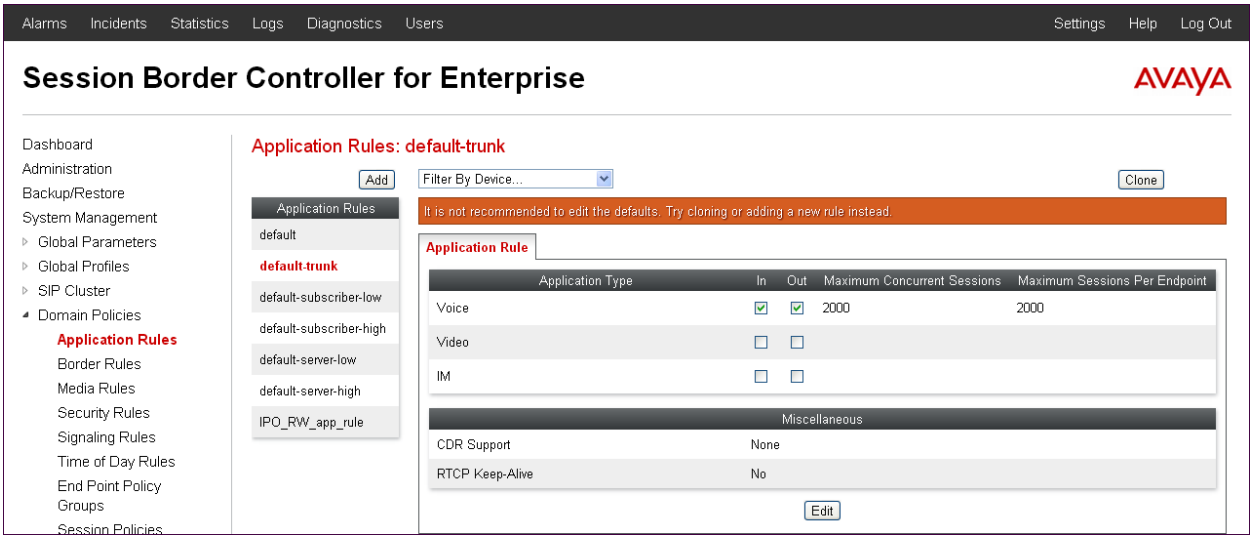


6.8. Application Rule

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

©2014 Avaya Inc. All Rights Reserved.

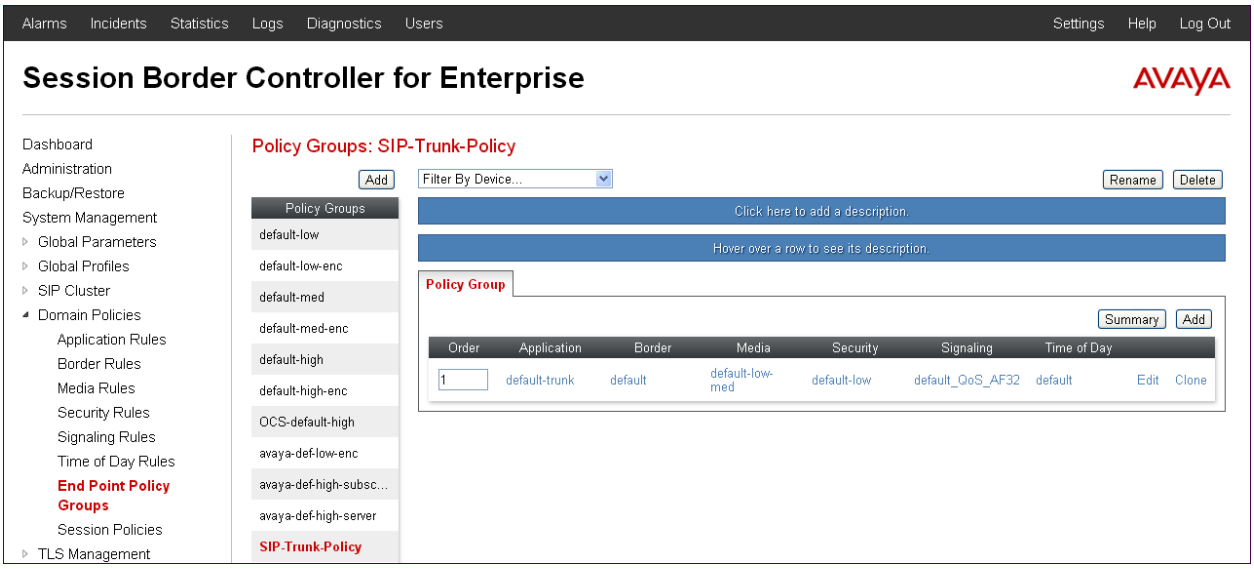
Select **Domain Policies → Application Rules** from the left-side menu as shown below. In the sample configuration, a single default application rule "**default-trunk**" is used and will be applied to the Endpoint Policy Group in the next section.



6.9. Endpoint Policy Groups

The rules created within the Domain Policy section are assigned to an Endpoint Policy Group. The Endpoint Policy Group is then applied to a Server Flow in **Section 6.12**.

To create a new policy group, navigate to **Domain Policies → Endpoint Policy Groups** and click on **Add** as shown below. The following screen shows the "**SIP-Trunk-Policy**" with defaults selected for all fields, with the exception of **Application** set to "**default-trunk**", and **Signaling**, which is set to "**default_QoS_AF32**" as shown below. The details of the non-default rules chosen are shown in previous sections.

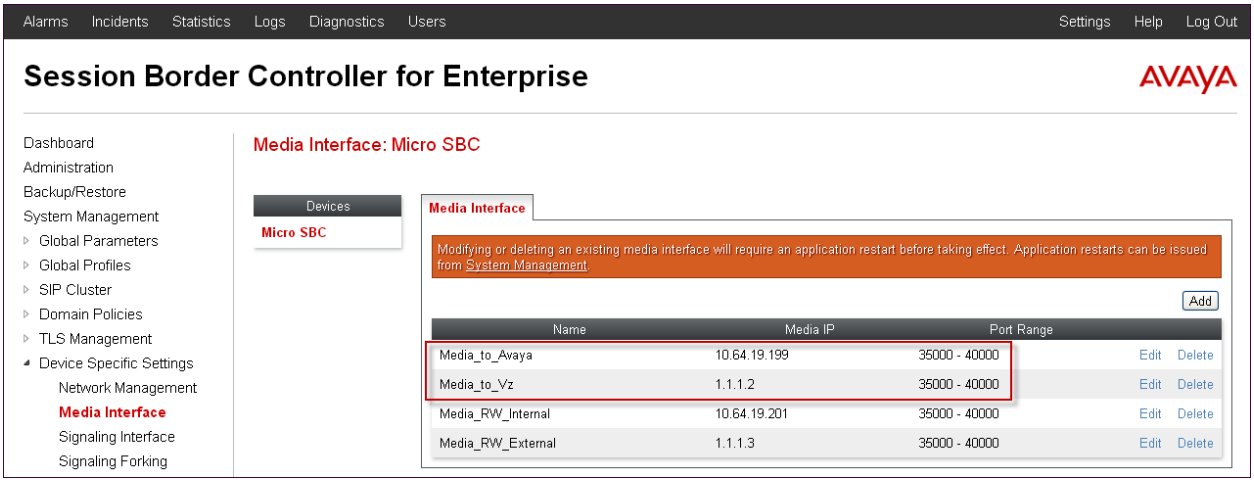


©2014 Avaya Inc. All Rights Reserved.

6.10. Media Interface

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

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



6.11. Signaling Interface

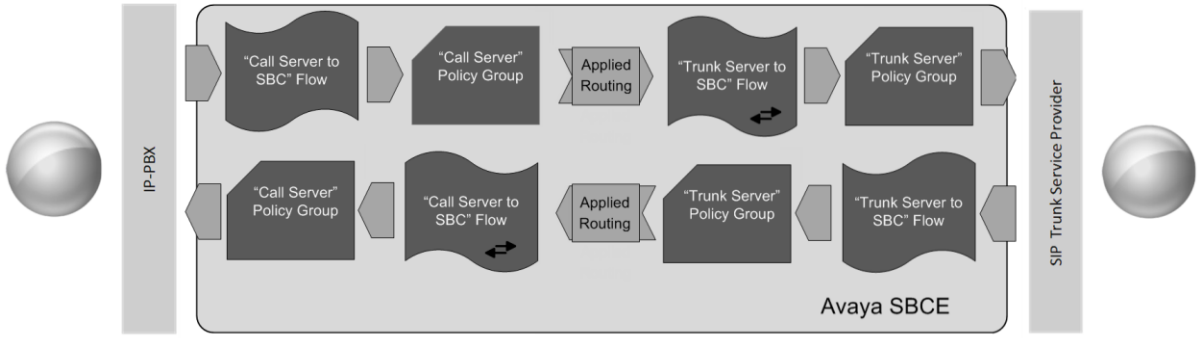
The Signaling Interface screen is where the SIP signaling ports are defined. Avaya SBCE will listen for SIP requests on the defined ports. Create a signaling interface for the inside and outside IP interfaces.

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

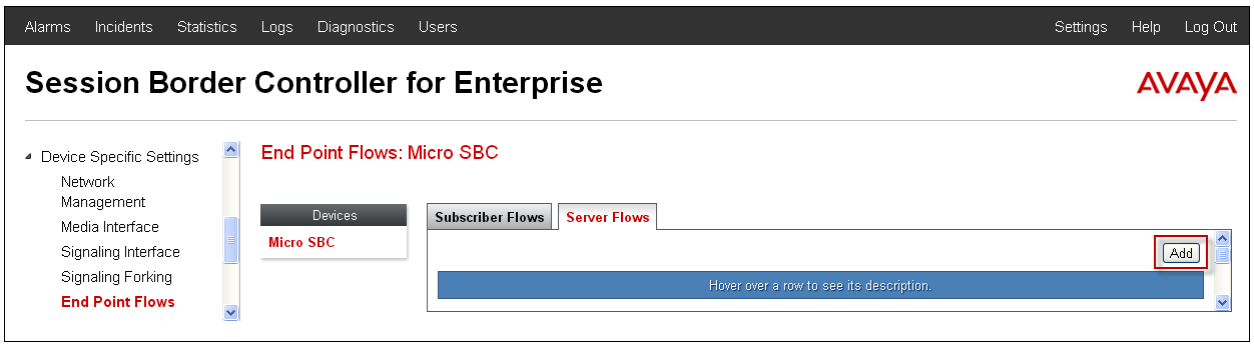
©2014 Avaya Inc. All Rights Reserved.

6.12. End Point Flows - Server Flow

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



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



©2014 Avaya Inc. All Rights Reserved.

The following screen shows the flow named "**Verizon IPT Flow**" viewed from the sample configuration. This flow uses the interfaces, polices, and profiles defined in previous sections.

View Flow: Verizon IPT Flow X

Criteria		Profile	
Flow Name	Verizon IPT Flow	Signaling Interface	Sig_to_Vz
Server Configuration	Verizon-IPT	Media Interface	Media_to_Vz
URI Group	*	End Point Policy Group	SIP-Trunk-Policy
Transport	*	Routing Profile	IP Office
Remote Subnet	*	Topology Hiding Profile	Vz_IPT_TH
Received Interface	Sig_to_Avaya	File Transfer Profile	None

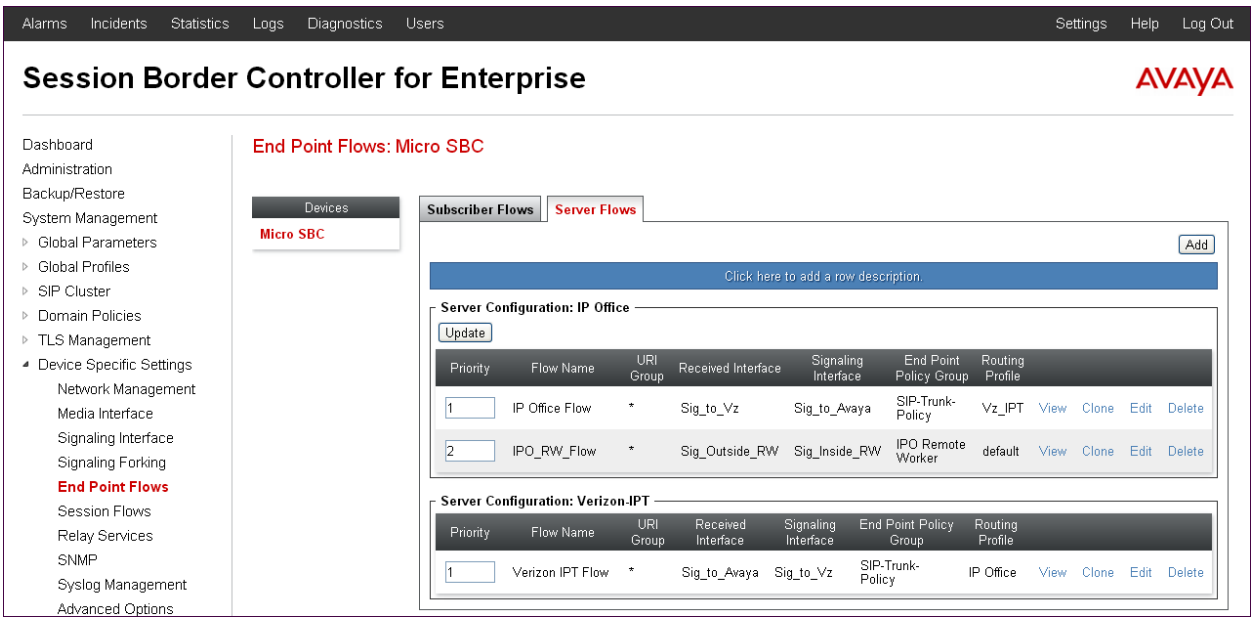
Once again, select the **Server Flows** tab and click **Add**. The following screen shows the flow named "**IP Office Flow**" viewed from the sample configuration. This flow uses the interfaces, polices, and profiles defined in previous sections.

View Flow: IP Office Flow X

Criteria		Profile	
Flow Name	IP Office Flow	Signaling Interface	Sig_to_Avaya
Server Configuration	IP Office	Media Interface	Media_to_Avaya
URI Group	*	End Point Policy Group	SIP-Trunk-Policy
Transport	*	Routing Profile	Vz_IPT
Remote Subnet	*	Topology Hiding Profile	default
Received Interface	Sig_to_Vz	File Transfer Profile	None

©2014 Avaya Inc. All Rights Reserved.

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



©2014 Avaya Inc. All Rights Reserved.

7. Verizon Business Configuration

Information regarding Verizon Business IP Trunk service offer can be found by contacting a Verizon Business sales representative, or by visiting http://www.verizonbusiness.com/us/products/voip/trunking/.

The reference configuration described in these Application Notes was located in the Avaya Solutions and Interoperability Lab. The Verizon Business IP trunk service was accessed via a Verizon Private IP (PIP) T1 connection. Verizon Business provided the necessary service provisioning.

The following Fully Qualified Domain Names (FQDNs) were provided by Verizon for the reference configuration.

CPE (Avaya)	Verizon Network
adevc.avaya.globalipcom.com	*pcelban0001.avayalincroft.globalipcom.com*

For service provisioning, Verizon will require the customer IP address used to reach the Avaya SBCE. Verizon provided the following information for the compliance testing: the IP address and port used by the Verizon SIP SBC, DNS server information, and the Direct Inward Dialed (DID) numbers shown in **Figure 1** and **Table 1**. This information was used to complete the Avaya IP Office and Avaya SBCE configuration.

©2014 Avaya Inc. All Rights Reserved.

8. Verifications

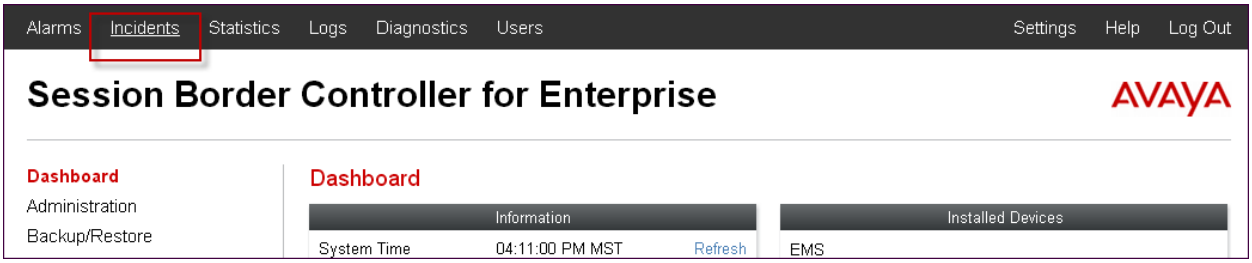
This section provides example verifications of the Avaya configuration with Verizon Business Private IP (PIP) Trunk service.

8.1. Avaya SBCE

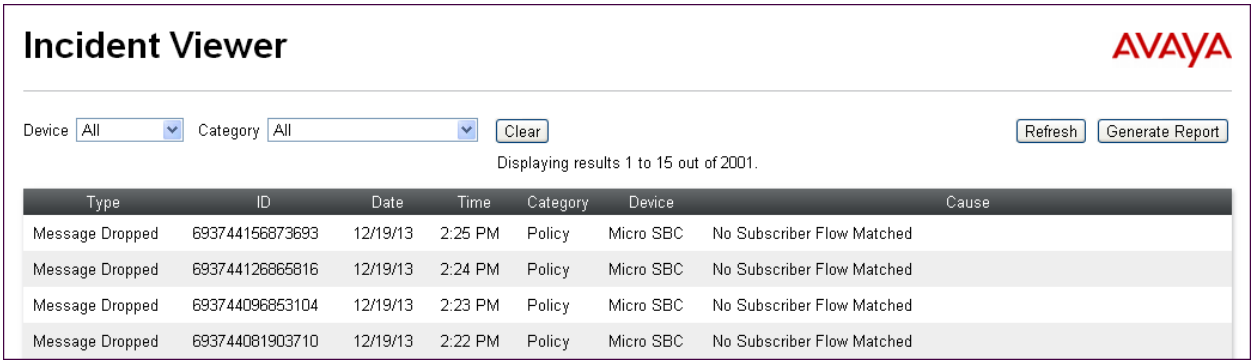
This section provides verification steps that may be performed with the Avaya SBCE.

8.1.1. Incidents

The Incident Viewer can be accessed from the Avaya SBCE Dashboard as highlighted in the screen shot below.



Use the Incident Viewer to verify Server Heartbeat and to troubleshoot routing failures.

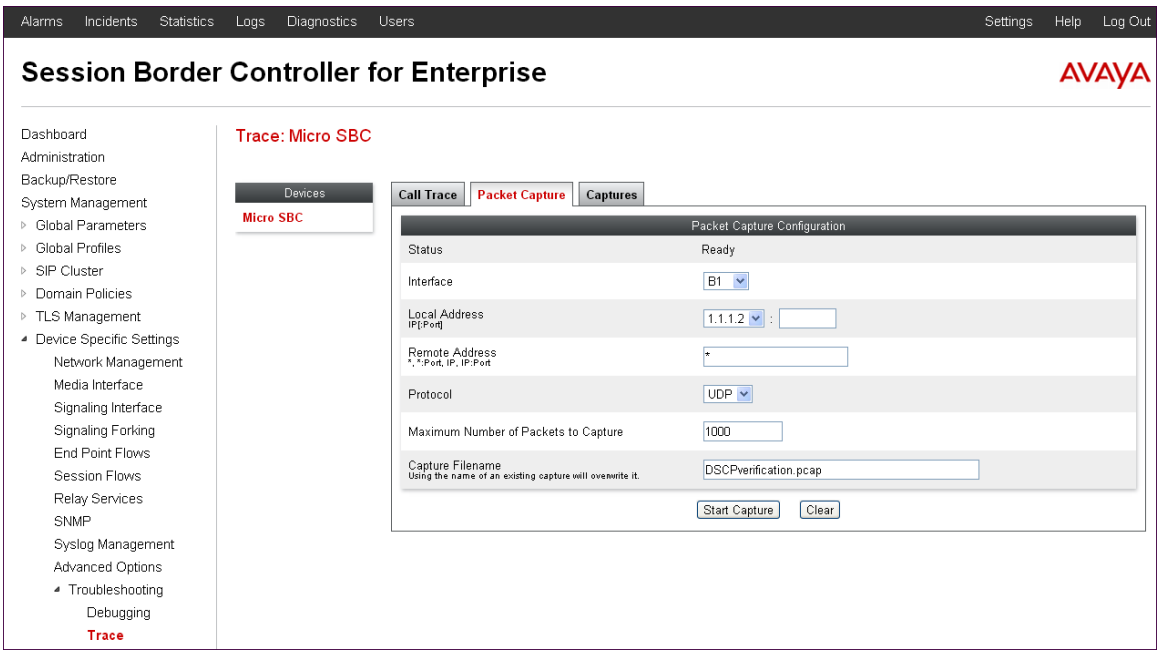


Type	ID	Date	Time	Category	Device	Cause
Message Dropped	693744156873693	12/19/13	2:25 PM	Policy	Micro SBC	No Subscriber Flow Matched
Message Dropped	693744126865816	12/19/13	2:24 PM	Policy	Micro SBC	No Subscriber Flow Matched
Message Dropped	693744096853104	12/19/13	2:23 PM	Policy	Micro SBC	No Subscriber Flow Matched
Message Dropped	693744081903710	12/19/13	2:22 PM	Policy	Micro SBC	No Subscriber Flow Matched

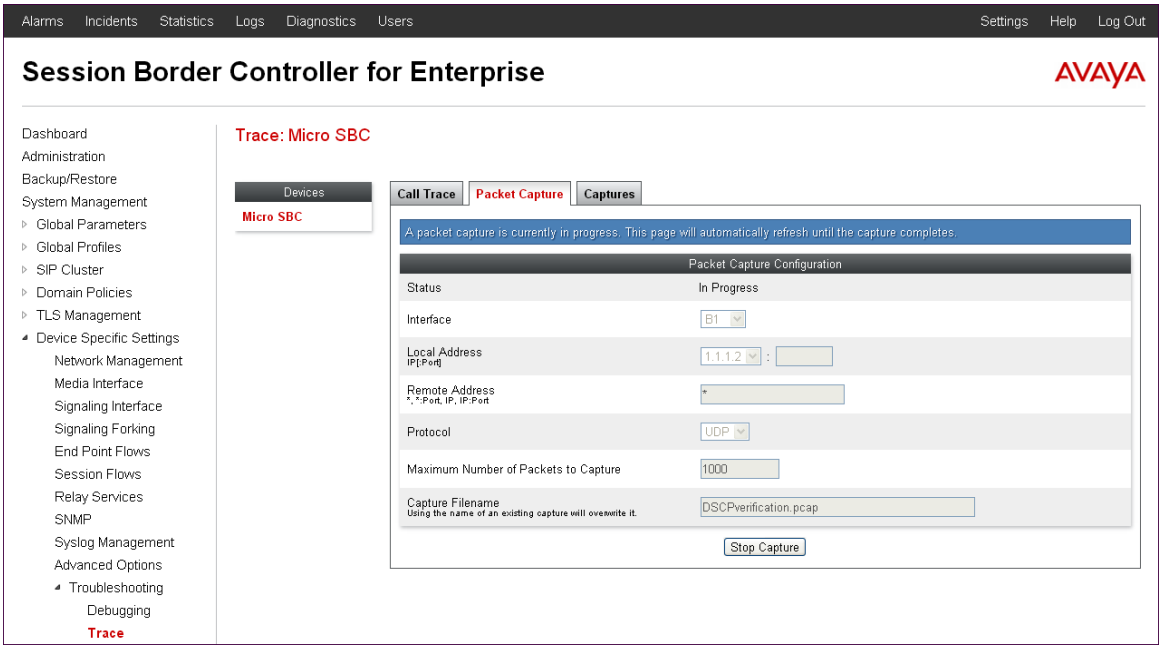
©2014 Avaya Inc. All Rights Reserved.

8.1.2. Tracing

To take a call trace, navigate to **Device Specific Settings → Trace** and select the **Packet Capture** tab. Populate the fields for the capture parameters and click **Start Capture** as shown below.

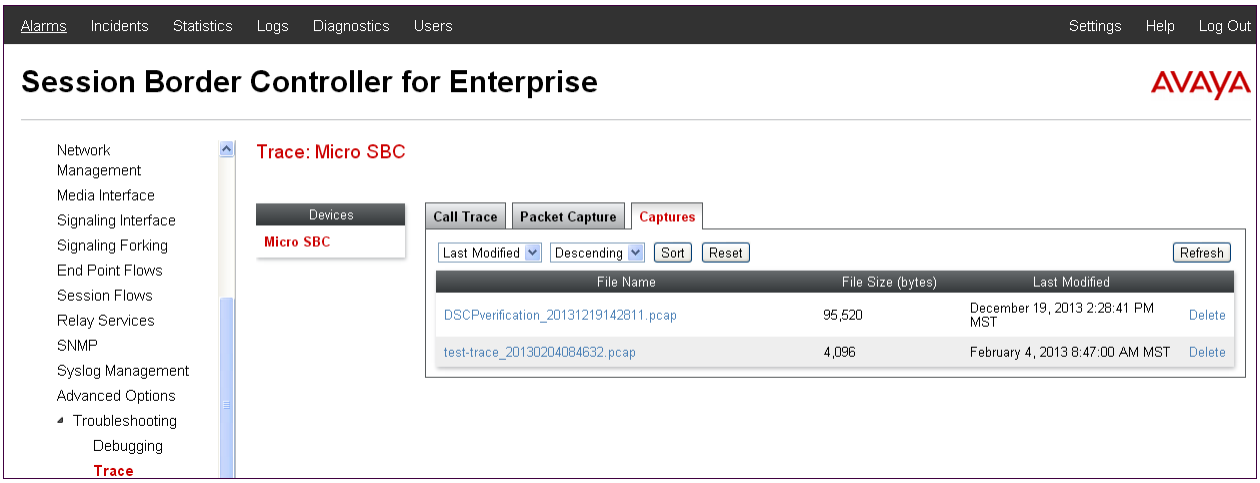


When tracing has reached the desired number of packets the trace will stop automatically, or alternatively, hit the **Stop Capture** button at the bottom.

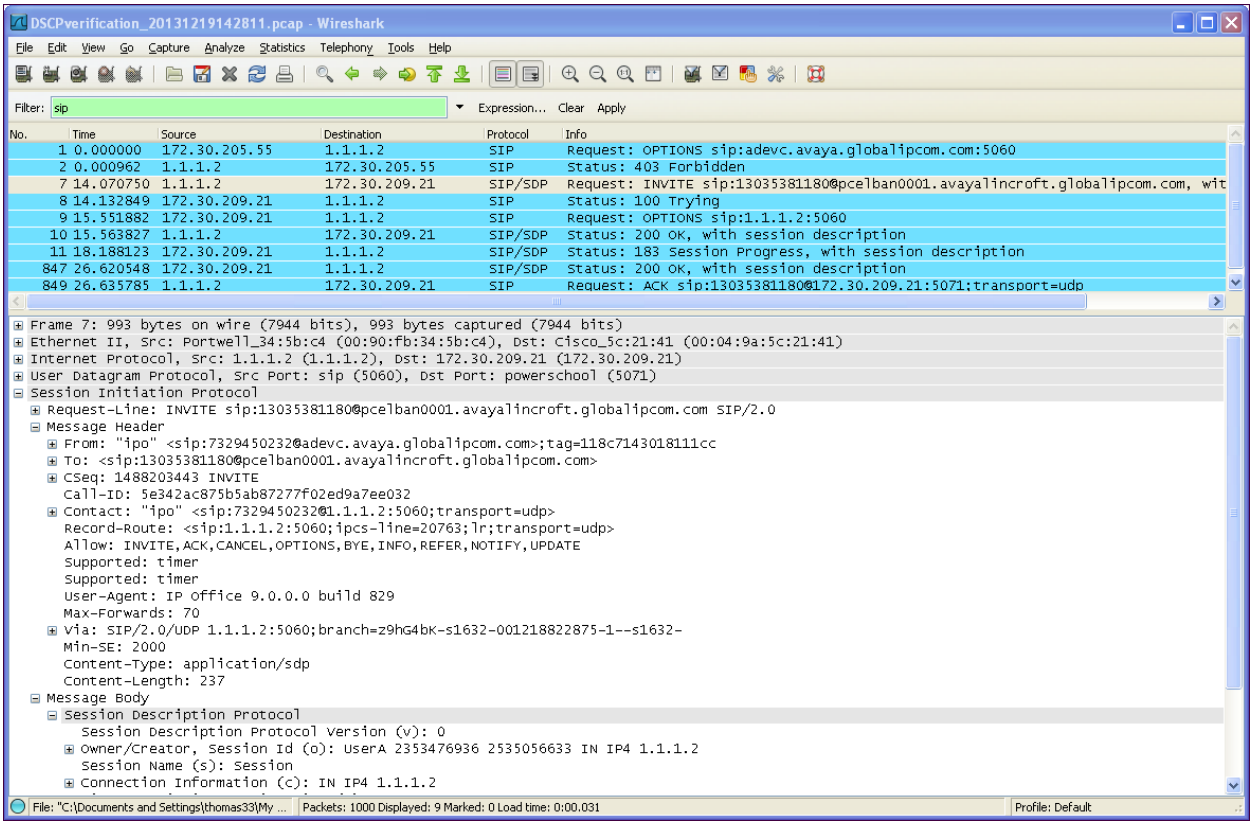


Select the **Captures** tab to view the files created during the packet capture.

©2014 Avaya Inc. All Rights Reserved.



The packet capture file can be downloaded and then viewed using a Network Protocol Analyzer like WireShark.



©2014 Avaya Inc. All Rights Reserved.

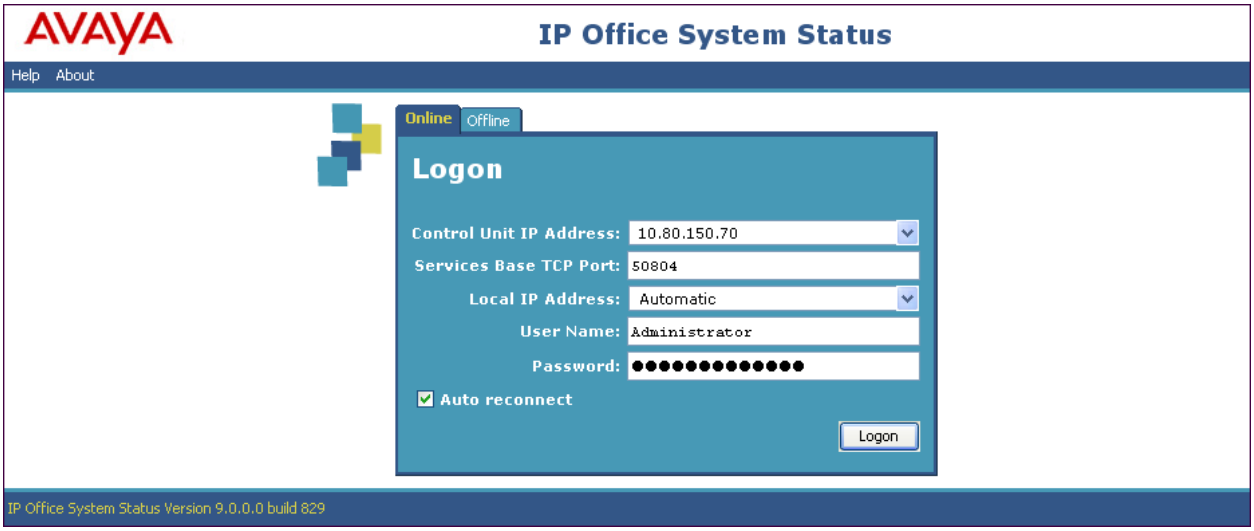
8.2. IP Office

This section provides verification steps that may be performed with the IP Office.

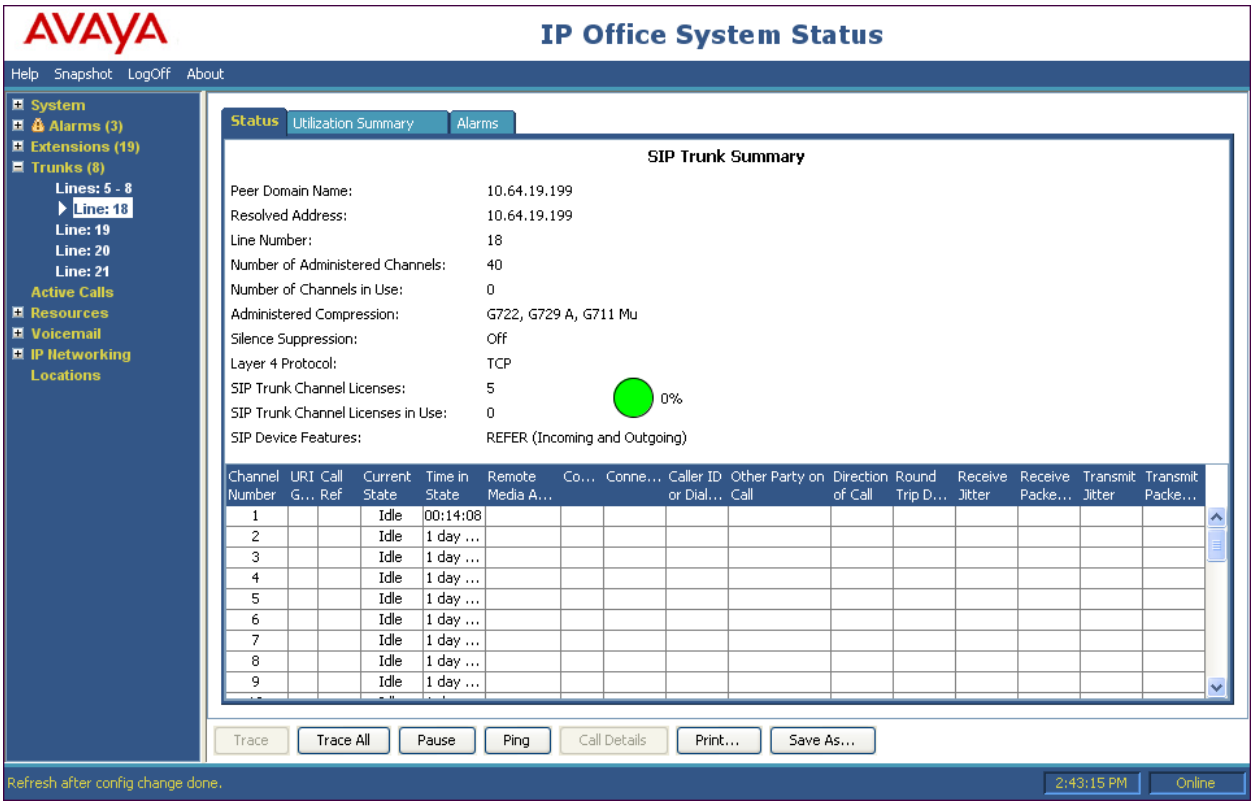
8.2.1. System Status

The System Status application is used to monitor and troubleshoot IP Office. Use the System Status application to verify the state of the SIP trunk. System Status can be accessed from **Start → Programs → IP Office → System Status**. Or by opening an Internet browser and type the URL: http://*ipaddress* where *ipaddress* is the IP address of the Avaya IP Office LAN1 interface. Click on **System Status** to launch the application.

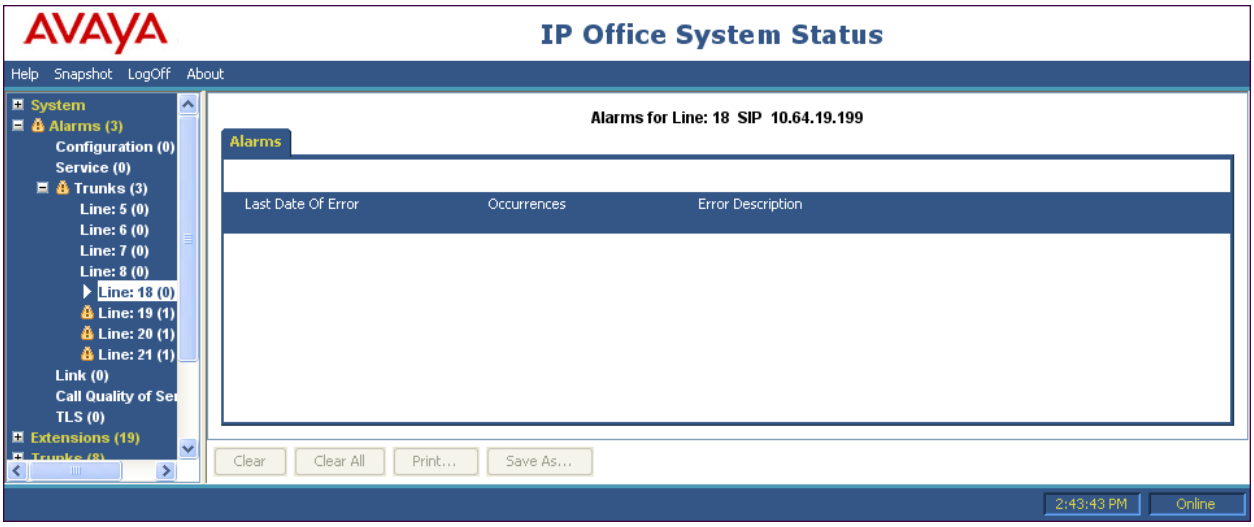
The following screen shows an example **Logon** screen. Enter the IP Office IP address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.



Select the SIP line under **Trunks** from the left pane. On the **Status** tab in the right pane, verify the **Current State** is ***Idle*** for each channel.



Select the **Alarms** tab and verify that no alarms are active on the SIP line.



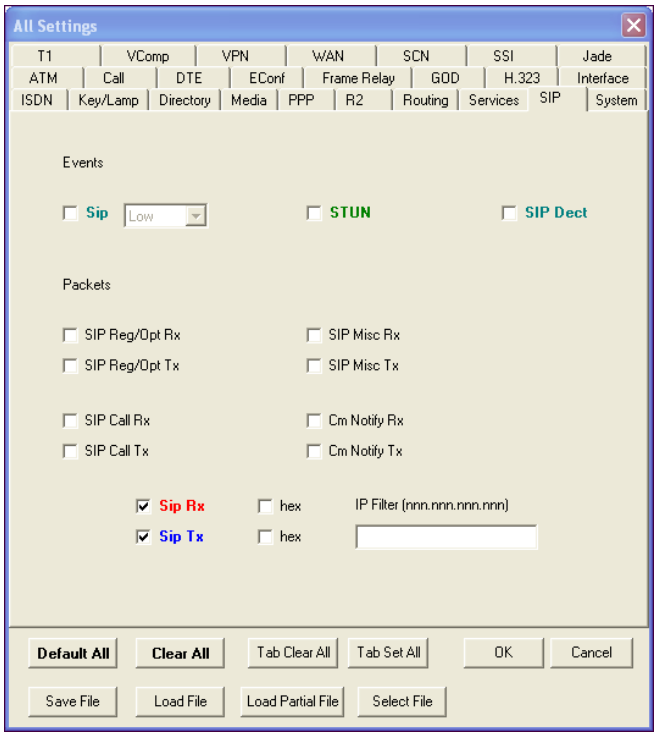
8.2.2. Monitor

The Monitor application can also be used to monitor and troubleshoot IP Office. Monitor can be accessed from **Start → Programs → IP Office → Monitor**. The application allows the

©2014 Avaya Inc. All Rights Reserved.

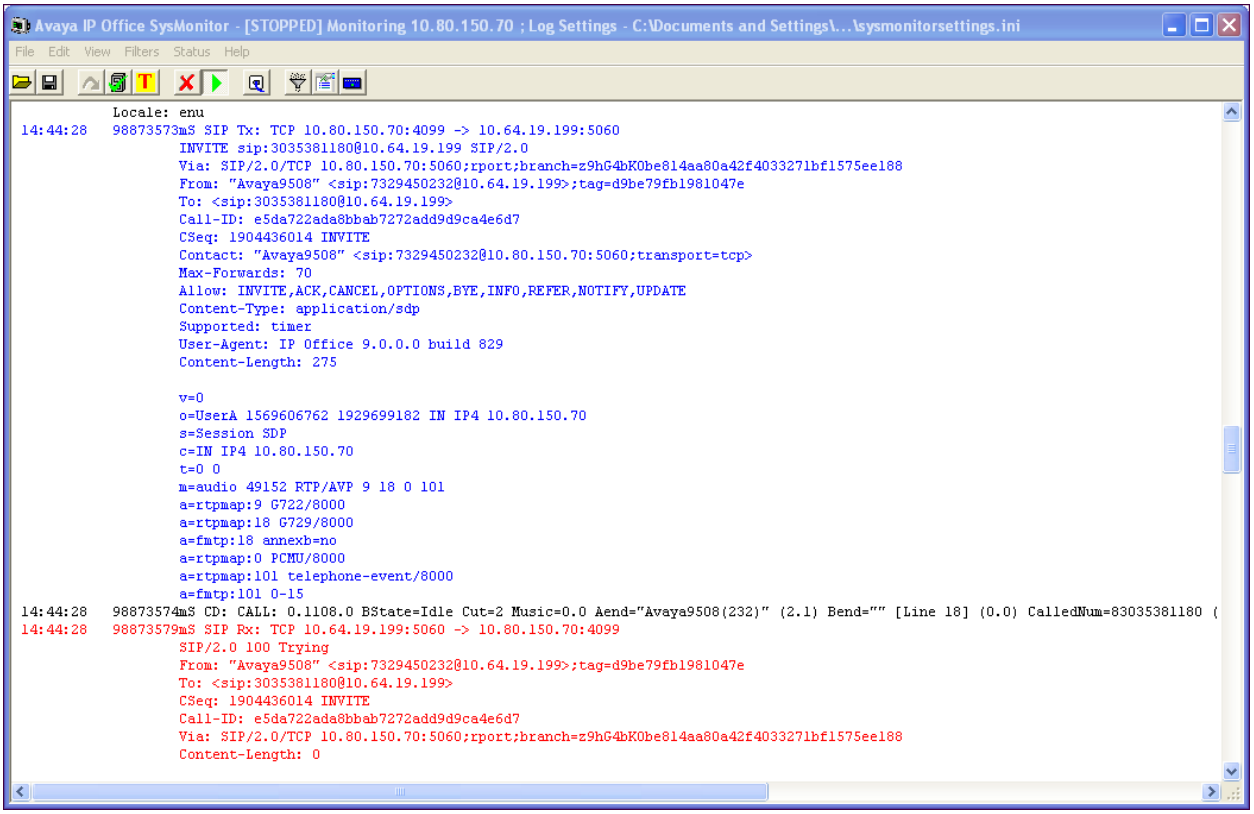
monitored information to be customized. To customize, select the button that is third from the right in the screen below, or select **Filters →Trace Options**.

The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, the **SIP Rx** and **SIP Tx** boxes are checked. All SIP messages will appear in the trace with the color blue. To customize the color, right-click on **SIP Rx** or **SIP Tx** and select the desired color.



©2014 Avaya Inc. All Rights Reserved.

As an example, the following shows a portion of the monitoring window for an outbound call from extension 233, whose DID is 732-945-0232, calling out to the PSTN via the Verizon Business IP Trunk service. The telephone user dialed 9-1-303-538-1180.



©2014 Avaya Inc. All Rights Reserved.

9. Conclusion

IP Office is a highly modular IP telephone system designed to meet the needs of home offices, standalone businesses, and networked branch and head offices for small and medium enterprises. These Application Notes demonstrated how IP Office Release 9.0 with Avaya Session Border Controller for Enterprise Release 6.2 can be successfully combined with a Verizon Business IP Trunk SIP Trunk Service connection to create an end-to-end SIP Telephony business solution. By following the example configurations provided in this document, customers using Avaya IP Office and Avaya SBCE can connect to the PSTN via a Verizon Business IP Trunk SIP Trunk service connection, thus eliminating the costs of analog or digital trunk connections previously required to access the PSTN. Utilizing this solution, IP Office customers can leverage the operational efficiencies and cost savings associated with SIP trunking while gaining the advanced technical features provided through the marriage of best of breed technologies from Avaya and Verizon.

10. Additional References

This section references documentation relevant to these Application Notes. In general, Avaya product documentation is available at http://support.avaya.com

[1] *IP Office 9.0 Installing IP500/IP 500 V2*, Document Number 15-601042, November 2013

[2] *IP Office Manager*, Document Number 15-601011, November 2013

[3] *IP Office Application server 9.0 Installation and Maintenance*, August 2013

[4] *IP Office 9.0 Using System Status*, Document Number 15-601758, August 2013

[5] *Administering Avaya Flare® Experience for iPad Devices and Windows*, September 2013

[6] *Configuring the Avaya Session Border Controller for IP office Remote Workers*, Sept 2013

[7] *Installing Avaya Session Border Controller for Enterprise*, June 2013

[8] *Administering Avaya Session Border Controller*, December 2013

[9] RFC 3261 *SIP: Session Initiation Protocol* http://www.ietf.org/rfc/rfc3261.txt

Additional IP Office documentation can be found at:
http://marketingtools.avaya.com/knowledgebase/

©2014 Avaya Inc. All Rights Reserved.

©2014 Avaya Inc. All Rights Reserved.
Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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.