Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking Using Verizon Business IP Contact Center VoIP Inbound and Avaya IP Office Release 9.0 with Avaya Session Border Controller for Enterprise Release 6.2 – Issue 1.0

Abstract

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Contact Center VoIP Inbound SIP Trunk Service 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.

IP Office Release 9.0 with Avaya Session Border Controller for Enterprise Release 6.2 has not been independently certified by Verizon labs. These Application Notes can be used to facilitate customer engagements via the Verizon field trial process, pending Verizon labs independent certification.

The Verizon Business IP Contact Center VoIP Inbound offer referenced within these Application Notes enables a business to receive inbound toll free calls via standards-based SIP trunks, 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.

1. Introduction

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Contact Center VoIP Inbound Service and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya Session Border Controller for Enterprise Release 6.2, and Avaya IP Office 500 V2 Release 9 Preferred Edition, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, SIP, digital, and analog endpoints.

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

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

For more information on the Verizon Business IPCC service, visit http://www.verizonbusiness.com/Products/communications/contact-center/.

2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the Verizon Business IPCC 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 IPCC service. This allowed Avaya IP Office to receive inbound toll-free calls from the PSTN via the SIP protocol.

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

2.1. Interoperability Compliance Testing

The testing included executing the test cases detailed in Reference [VZ-Test-Plan], which contains the Verizon IPCC Interoperability Lab Test Plan. To summarize, the testing included the following successful SIP trunk interoperability compliance testing:

- SIP OPTIONS monitoring of the health of the SIP trunk was verified. Verizon Business, Avaya SBCE, and IP Office can all monitor health using SIP OPTIONS.
- Proper recovery from induced failure conditions such as IP Office reboots, and IP network outages between Verizon and IP Office, of short and long durations.
- Incoming calls from the PSTN were routed to the toll-free numbers assigned by Verizon Business to the Avaya IP Office location. These incoming calls arrived via the SIP Line configured in **Section 5.4** and were answered by Avaya H.323 telephones, Avaya SIP telephones, Avaya digital telephones, analog telephones, Avaya Flare® Experience, one-X® Mobile, Avaya IP Office Softphone, and Avaya IP Office Voicemail Pro.
- Proper disconnect when either party hangs up an active call.
- Proper disconnect when the PSTN caller abandons (i.e., hangs up) a toll-free call before the IP Office party has answered.
- Proper SIP 486 response and busy tone heard by the caller when a PSTN user calls a toll-free number directed to a busy IP Office user, an IP Office user with Do-not-disturb active, or an IP Office user that is logged out (i.e., assuming no redirection is configured for these conditions). Similarly, busy tone is heard when a PSTN user calls a toll-free number whose "SIP URI Max Calls per Channel" has been reached (see **Section 5.4.4**). Similarly, busy tone is heard when a PSTN user calls a toll-free number directed to a hunt group whose queue is "full" (i.e. if no redirection is configured for hunt group busy conditions, see **Section 5.5.3**).
- Proper termination of an inbound IP Toll Free call left in a ringing state for a relatively long duration.
- The display of caller ID on display-equipped Avaya IP Office telephones was verified. The IP Office capability to use the caller ID received from Verizon to look up and display a name from a configurable directory was also exercised successfully.
- Privacy requests for inbound toll-free calls from the PSTN were verified. That is, when privacy is requested by a PSTN caller (e.g., dialing *67 from a mobile phone), the inbound toll-free call can be successfully completed to an IP Office telephone user while presenting a "WITHHELD" or anonymous display to an IP Office user (i.e., rather than the caller's telephone number).
- Inbound toll-free long holding time call stability.

- IP Office sends SIP 180 RINGING (no SDP in 180) for inbound calls and ring back tone is heard by the caller.
- Telephony features such as hold and resume, transfer of toll-free calls to other IP Office users, and conference of toll-free calls.
- Incoming voice calls using the G.729(a) and G.711 ULAW codecs, and proper protocol procedures related to media.
- DTMF transmission using RFC 2833. Successful IP Office Voicemail Pro menu navigation for incoming toll-free calls. Successful use of IP Office Mobile Call Control, where DTMF sequences can be performed remotely using the SIP Line.
- Incoming toll-free calls directed to the Hunt Groups configured in **Section 5.5.3** were verified. Incoming calls could be queued, queued with priority, and be answered by members of the hunt group as members become available.
- Proper DiffServ markings for Avaya SBCE SIP signaling and RTP media.

2.2. Test Results

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

1. The Verizon IPCC Service does not support fax.

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

3. Although the Verizon Business IPCC service supports transfer using the SIP REFER method and IP Office supports sending REFER, IP Office will not send REFER to Verizon in the verified configuration. When configured, IP Office sends REFER with Replaces, and Verizon IPCC requires REFER without Replaces.

4. Although Verizon IPCC, the focus of these Application Notes, is used for inbound toll-free numbers, inbound toll-free calls can be twinned, forwarded, or transferred back to the PSTN via the Verizon IP Trunk SIP Line, See Reference [VZBIPT-IPO9SBC]. If an inbound Verizon IPCC call to an IP Office user is forwarded to a PSTN number using a separate SIP service through the same Avaya SBC, IP Office may be configured to send a REFER to Verizon IPCC service when the call is answered. Verizon IPCC service will respond with a 202 Accepted message followed by 603 Server Internal Error in the NOTIFY message. This causes the call to drop. To prevent IP Office from sending a REFER to Verizon IPCC service in this scenario a signaling rule was created in the Avaya SBCE, as shown in **Section 6.7**.

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 Contact Center SIP Trunk service. The Avaya equipment is located on a private IP subnet. An enterprise edge router provides access to the Verizon Business 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 Contact Center service on port 5060 and sends traffic to port 5072, using UDP for network transport, as required by the Verizon Business IP Contact Center service. Verizon provided five toll-free numbers associated with the IP Contact Center service. These toll-free numbers were mapped to IP Office destinations via Incoming Call Routes as summarized in **Table 1**. The Avaya IP Office environment domain known to Verizon was *adevc.avaya.globalipcom.com*.

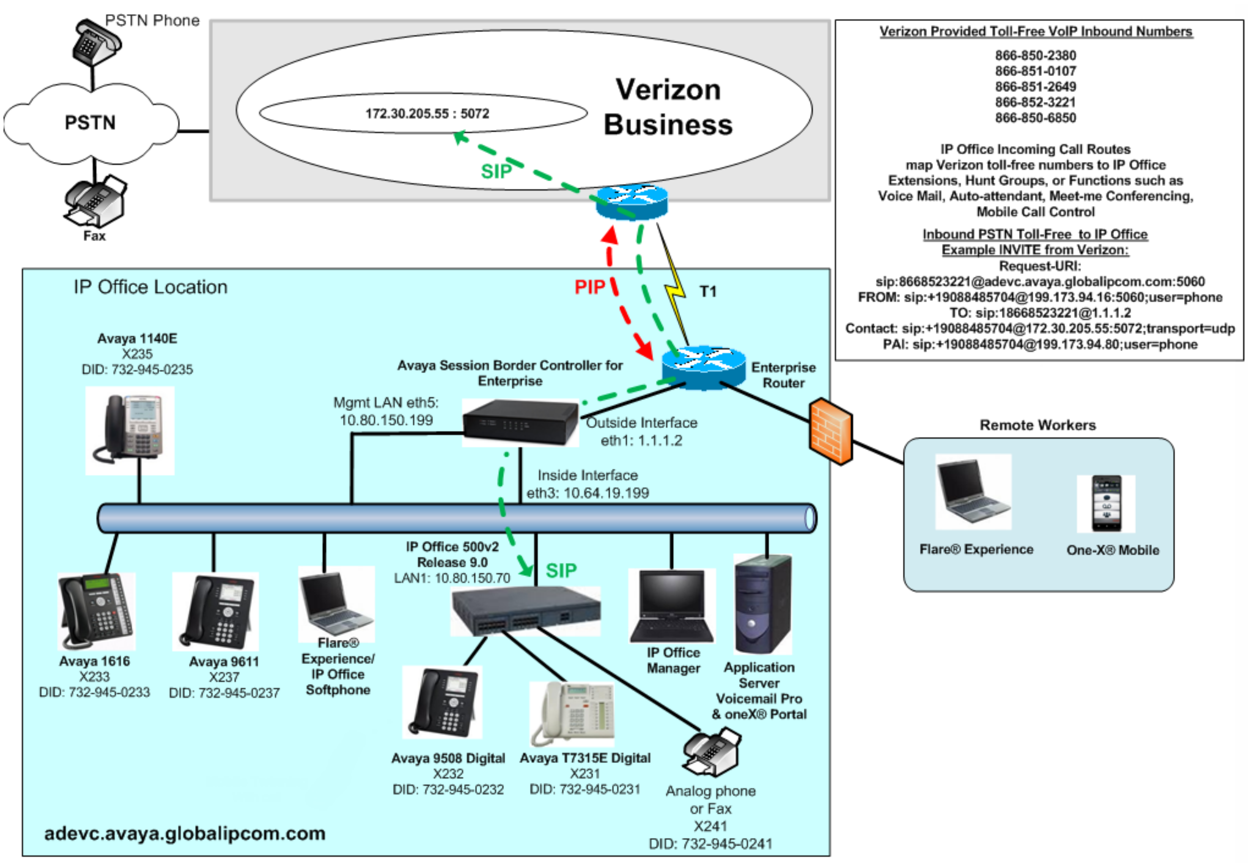


Figure 1: Avaya Interoperability Test Lab Configuration

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**].

Table 1 shows an example mapping of toll-free numbers to IP Office users, groups, or functions. The associated IP Office configuration is shown in **Section 5**. Since the quantity of toll-free numbers was limited in the test configuration relative to the desired test coverage, the same toll-free number was routed to different IP Office destinations (i.e., IP Office configuration changes were made to the Incoming Call Route destination as needed between successive tests).

Verizon Provided Toll-Free Number	Configured Avaya IP Office Destination(s)	Notes
866-851-0107	x235, x239	Avaya 1140E, Avaya one-X® Mobile
866-850-2380	x232, x241, x234	Digital Telephone with Mobile Twinning and Mobile Call Control permission. Also used to test analog telephone and Avaya Flare Experience capabilities.
866-851-2649	x233, x237	Avaya 1616 Telephone, Avaya 9611 Telephone
866-850-6850	Voicemail Collect on Voicemail Pro	Allow external callers to access voice mail toll-free
866-850-6850	Inbound Mobile Call Control	Allow toll-free calls from pre-configured twinning numbers to access mobile call control
866-850-6850	Conference Bridge on Voicemail Pro	Allow external callers to access conference bridge toll-free
866-852-3221 (any caller)	"401 Sales" Hunt Group (with default priority)	Hunt Group with queuing
866-852-3221 (specific callers)	"400 Overdue Account" Hunt Group	Show IP Office destination selection based on caller ID
866-852-3221 (specific priority callers)	"401 Sales" Hunt Group (with High Priority)	Show IP Office priority queuing based on caller ID

Table 1: Example Verizon Toll Free Number to IP Office Destination Mappings

4. Equipment and Software Validated

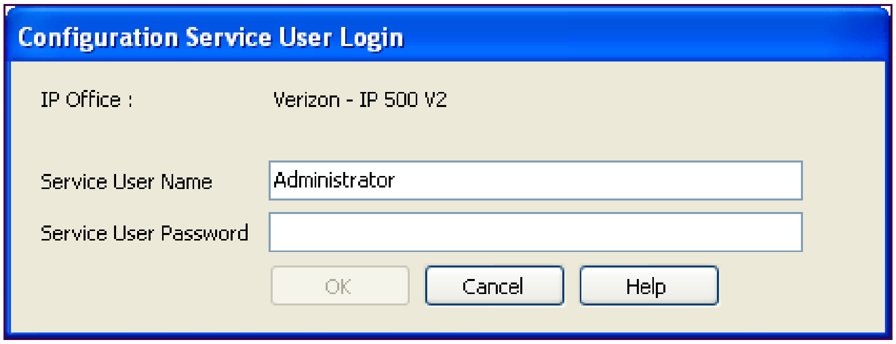
Table 2 shows the equipment and software used in the sample configuration

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 running Voicemail Pro and one-X® Portal for IP Office	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 9611SW 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

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:

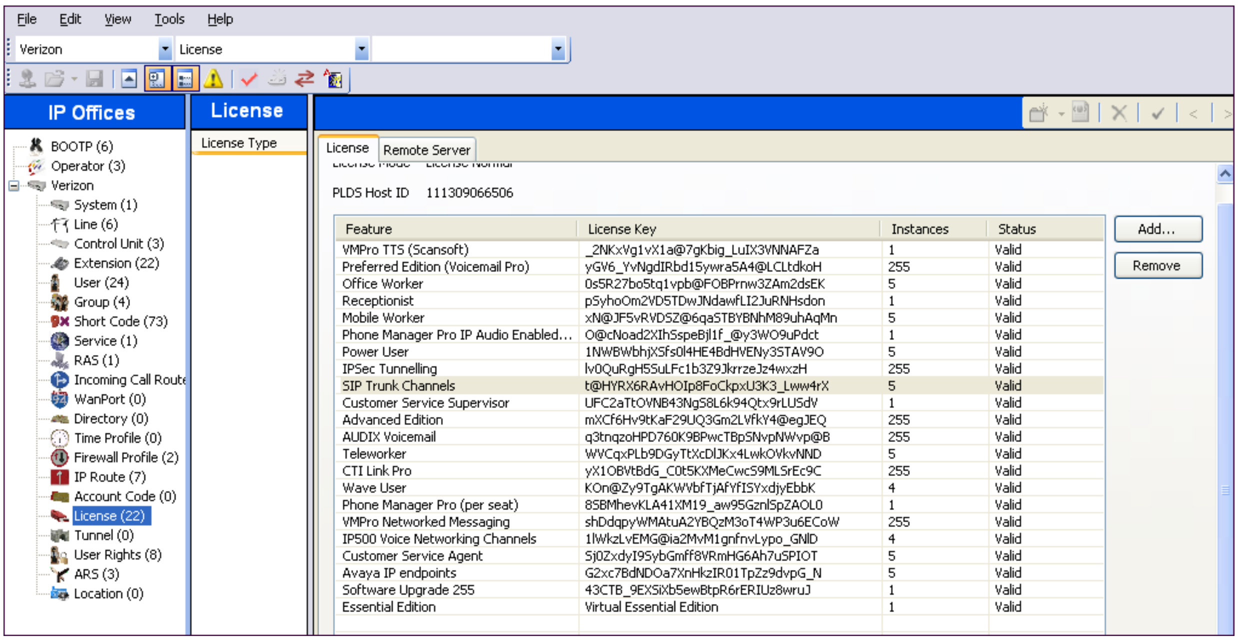


Log in with the appropriate configuration credentials. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details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

5.1. Licensing and Physical Hardware

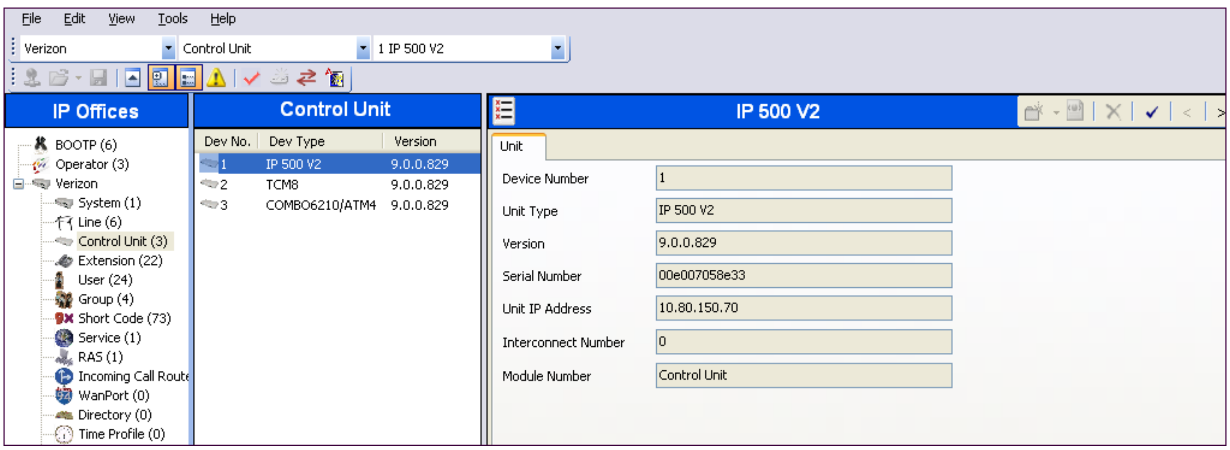
The configuration and features described in these Application Notes require the IP Office system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

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.



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.



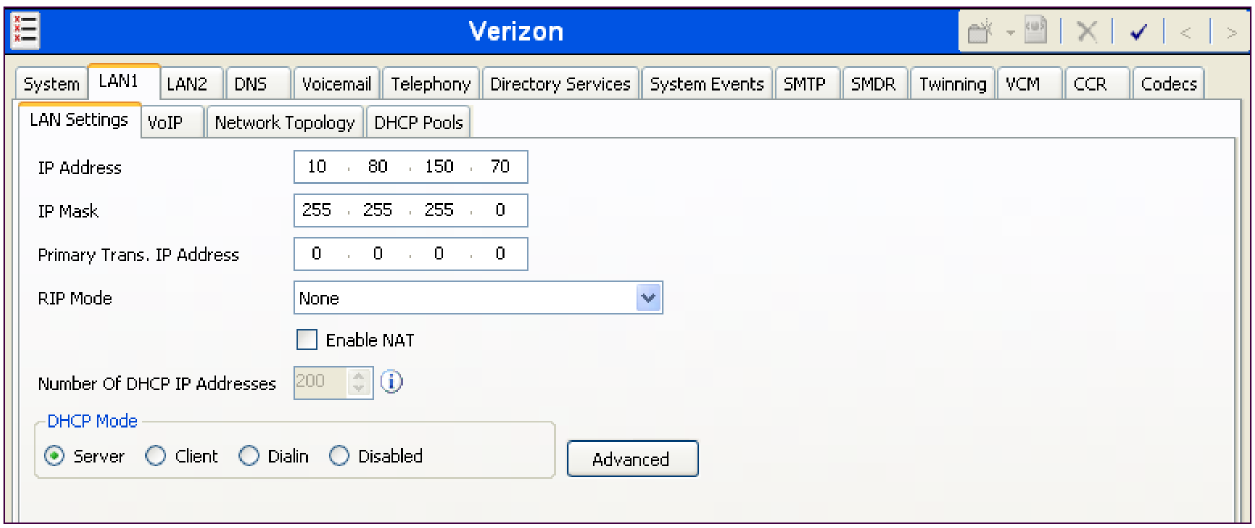
5.2. System Settings

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

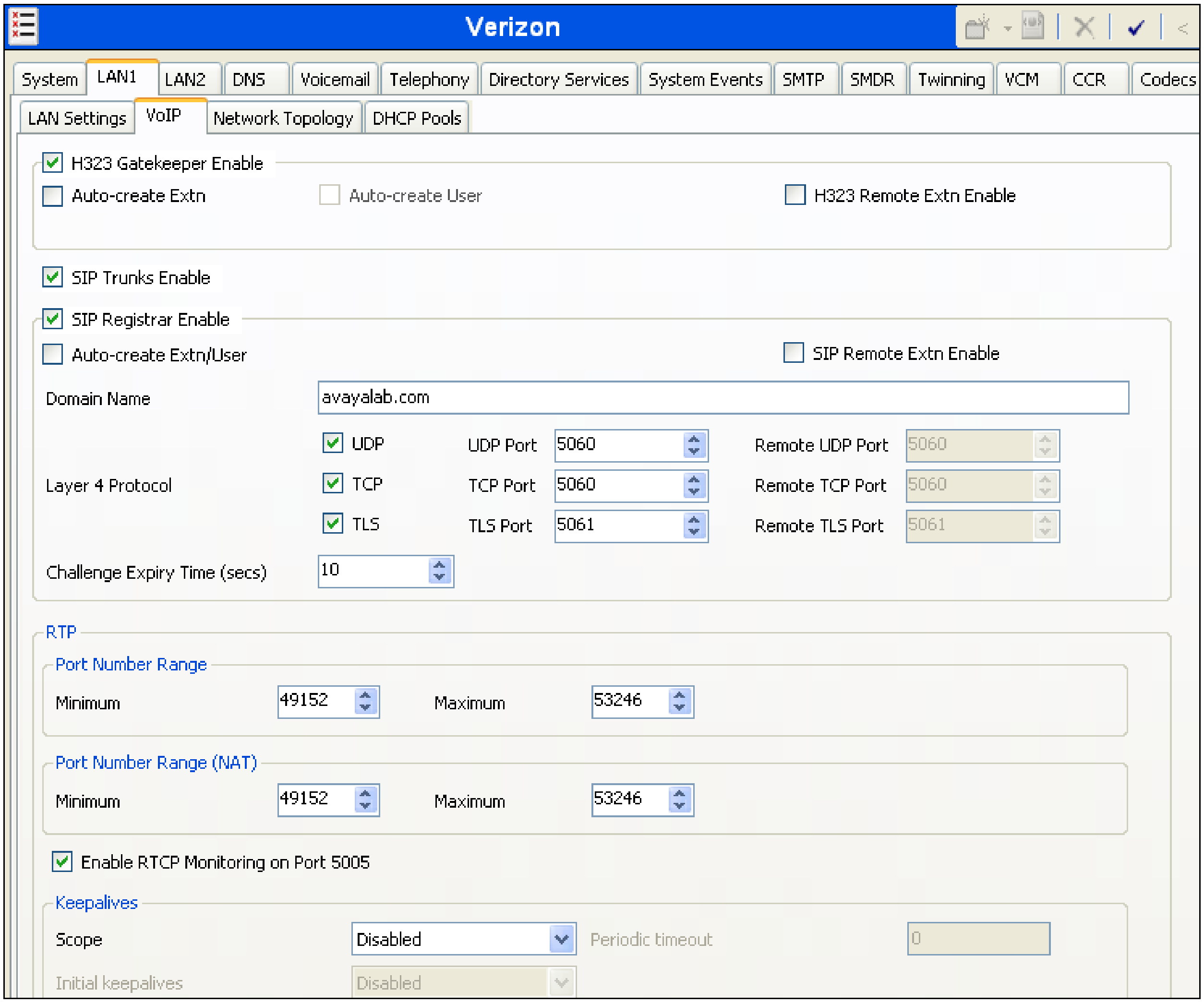
The IP500/IP500 V2 control units have 2 RJ45 Ethernet ports, physically marked as LAN and WAN. Within the system configuration, the physical LAN port is LAN1, the physical WAN port is LAN2.

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.

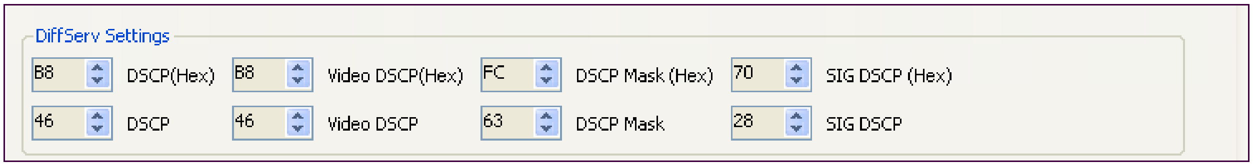


Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 1616 and 9611 used in the sample configuration. The **SIP Registrar Enable** box is checked to allow Avaya 1140E, 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.

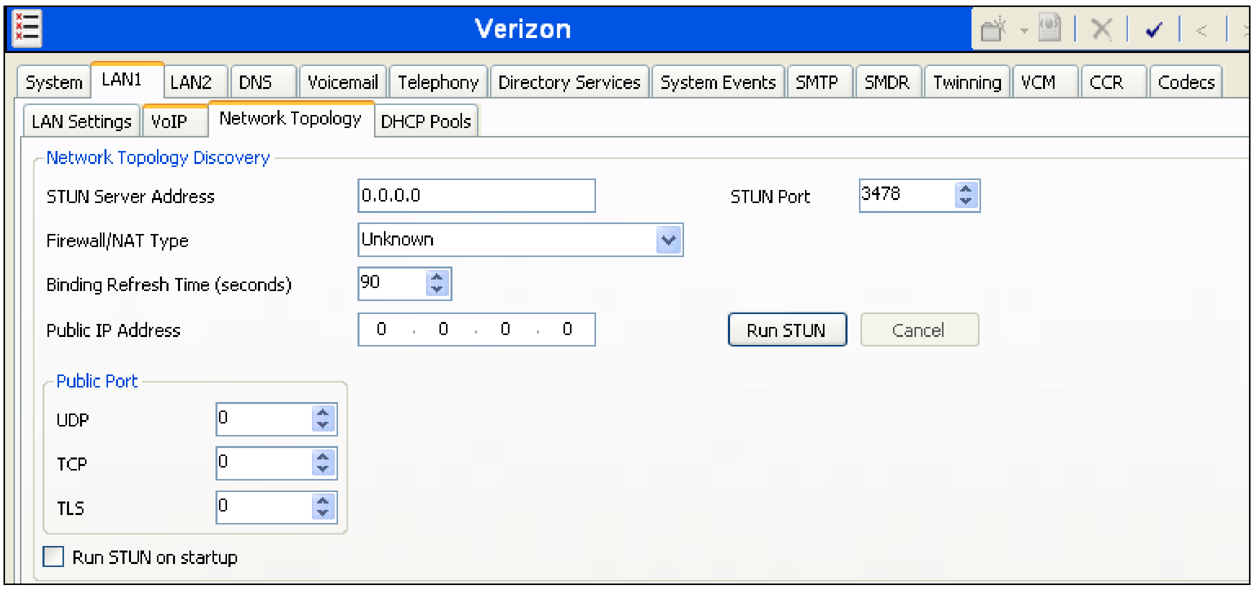
If desired, the **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media paths from Avaya SBCE to IP Office. The defaults are used here.



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 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.

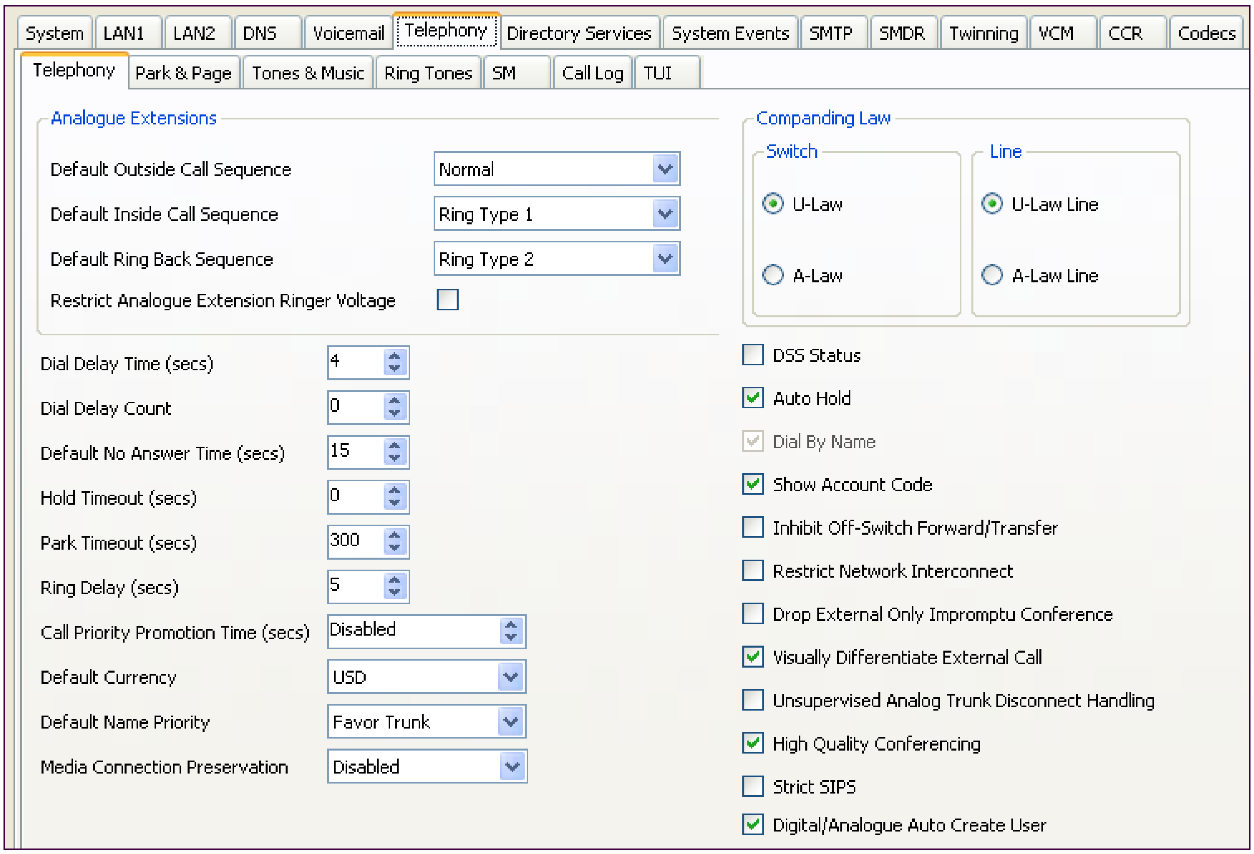


Select the **Network Topology** tab as shown in the following screen. In the sample configuration, the default settings are used and the **Use Network Topology Info** in the **SIP Line** is set to "**None**" in **Section 5.4.3**. The **Binding Refresh Time (seconds)** can be used to lower the SIP OPTIONS 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.



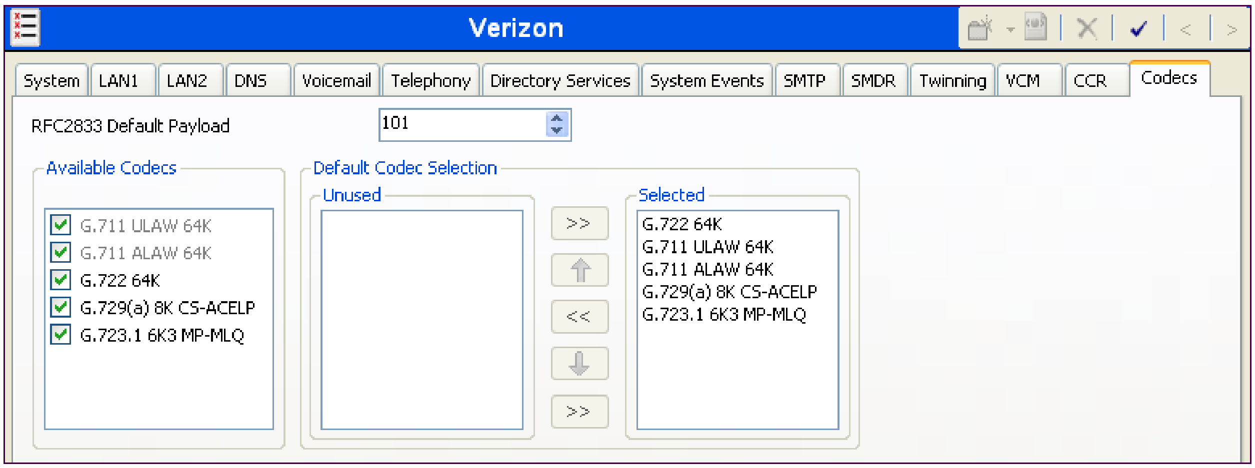
5.2.2. System Telephony Configuration

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.



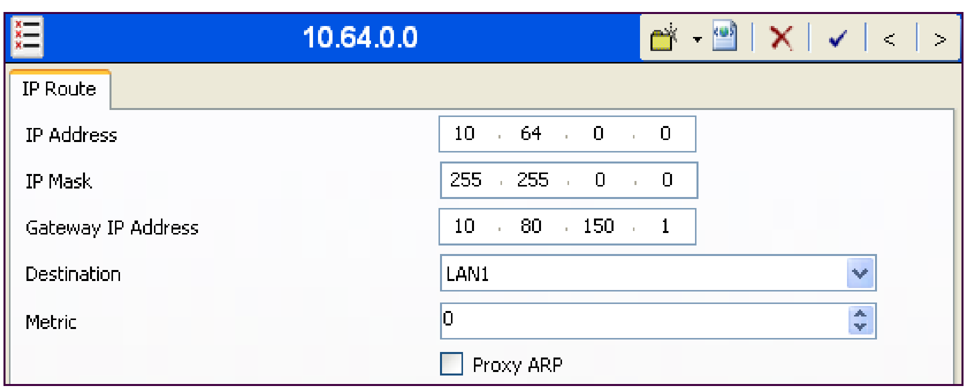
5.2.3. System Codecs Configuration

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 0**). 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.



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.



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:

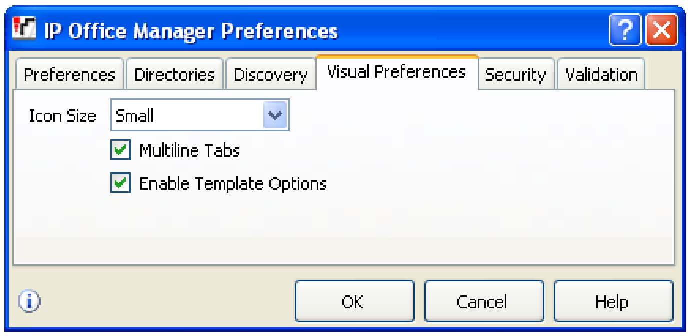
- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required

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**.

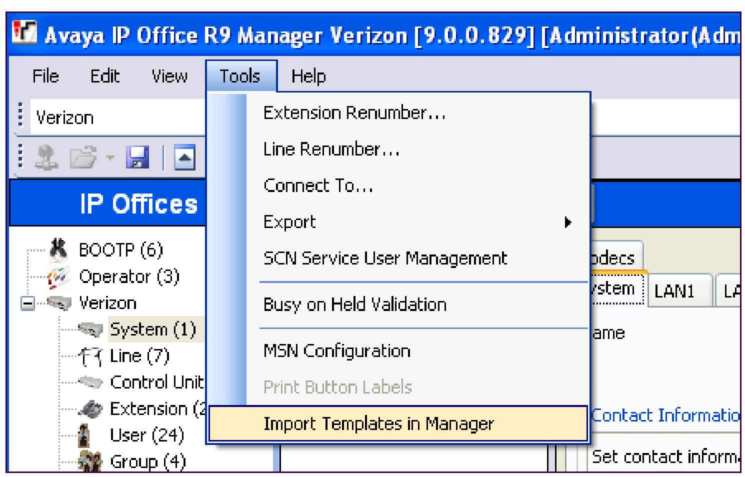
5.4.1. SIP Line From Template

1. Copy the template file to the computer where IP Office Manager is installed. Rename the template file to **US_VerizonIPCC-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**.

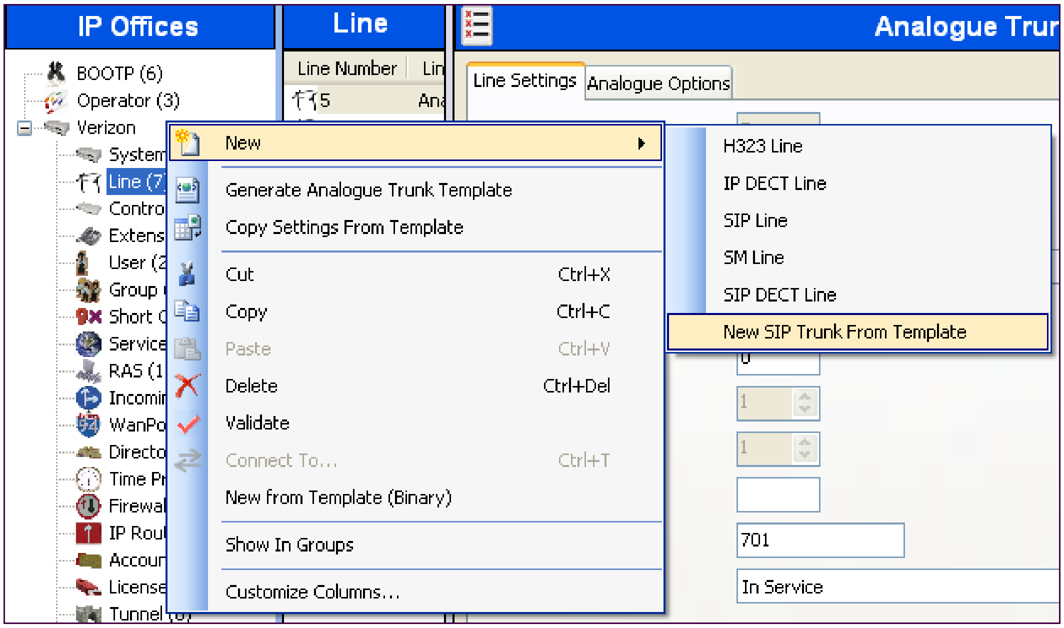


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default template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

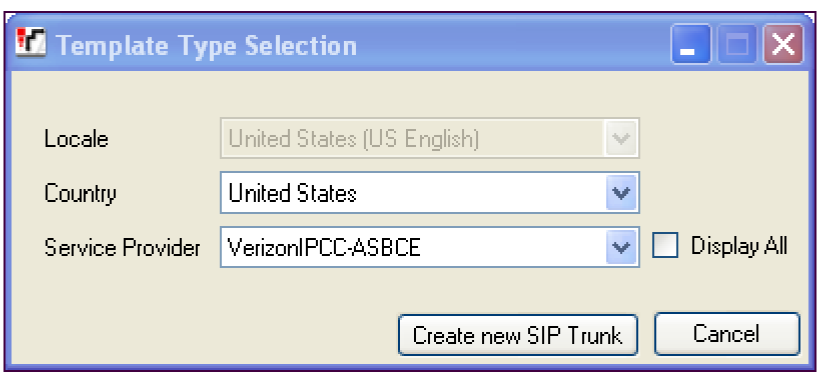


In the pop-up window (not shown) that appears, select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window (not shown) will appear stating success or failure. Click **OK** (not shown) to the IP Office template directory.

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 "**VerizonIPCC-ASBCE**" from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name (**US_VerizonIPCC-ASBCE_SIPTrunk.xml)** created in **Step 1**. Click **Create new SIP Trunk** to finish creating the trunk.



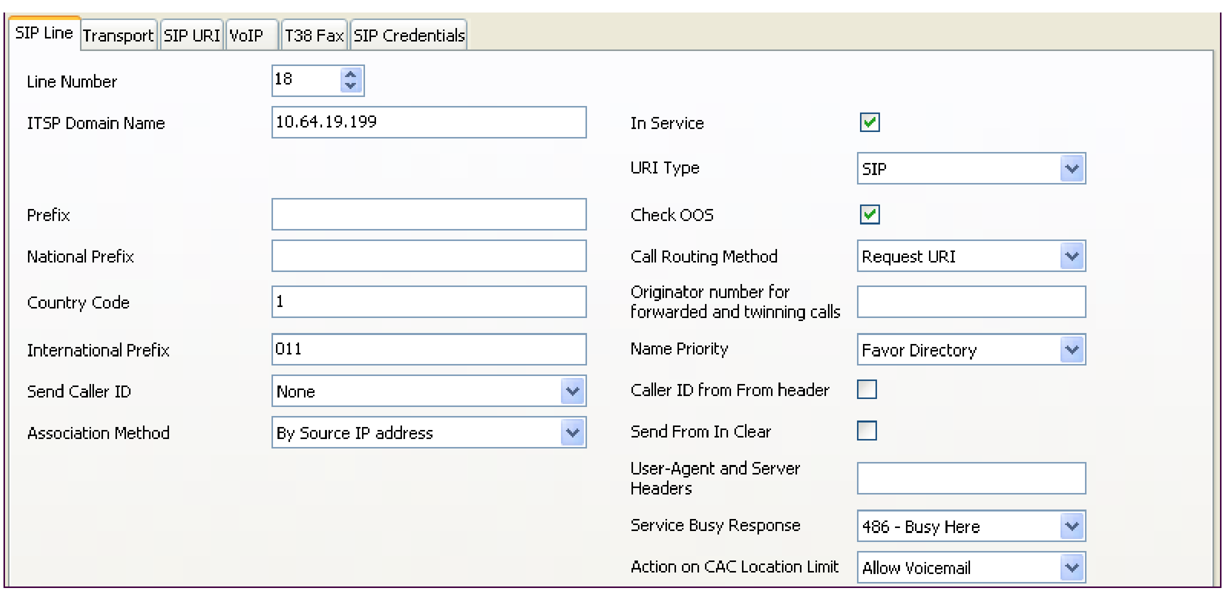
Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.4.2 – 5.4.5**.

5.4.2. SIP Line – SIP Line Tab

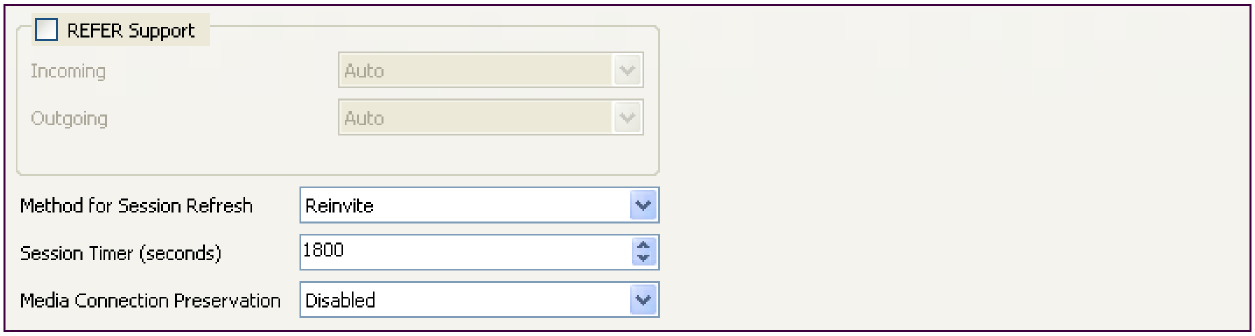
The **SIP Line** tab in the Details pane is shown below for Line Number 18, used for Avaya SBCE to the Verizon Business IPCC service. The **ITSP Domain Name** is configured with the inside IP address of the Avaya SBCE as shown in **Figure 1**. By default, the **In Service** and **Check OOS** boxes are 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 **Call Routing Method** can retain the default "Request URI" setting, or may be changed to "To Header", to match Incoming Call Routes based on the contents of the "To Header". In the sample configuration, the default "Request URI" setting was used. As can be observed in the sample INVITE header contents in **Figure 1** and **Section 8.1.2**, the Request-URI and the To header do not necessarily contain the same number. In the tested configuration, the Request-URI contained the toll-free number, and the "To" header contained 1 followed by the toll-free number.

In the sample configuration, the IP Office **Country Code** is set to 1. The "From" and "PAI" headers received from Verizon for calls from U.S. PSTN numbers contain "+1" before the calling PSTN number. By configuring the IP Office **Country Code** to 1, the caller ID display presented to IP Office users will be the PSTN number without any codes or prefixes. For example, a call from 3035387006 would display 3035387006. If the **Country Code** does not match the value following the "+" from Verizon, the IP Office user display would show the contents of the **International Prefix** field, followed by the value following the "+", followed by the PSTN number. For example, if the Country Code parameter were left blank, the IP Office user would see a display such as "01113035387006". Aside from display implications, if the **Country Code** is not configured, other patterns may also fail to match as expected, such as a match on the **Incoming CLI** field of the Incoming Call Route. See **Section 5.7.3** for configuration of incoming call routing based on the calling number.

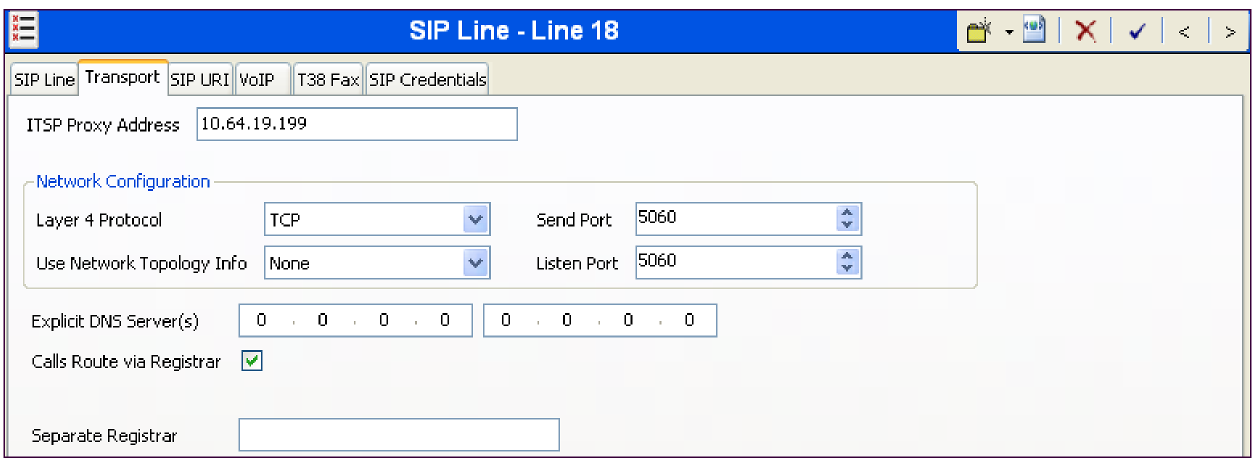


Towards the bottom of the screen, **REFER Support** is unchecked as the REFER method provided by IP Office is not supported by Verizon IPCC. The **Method for Session Refresh** parameter and the related **Session Timer (seconds)** parameter are new with IP Office Release 9.0. The **Method for Session 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". Click **OK** (not shown).



5.4.3. SIP Line - Transport Tab

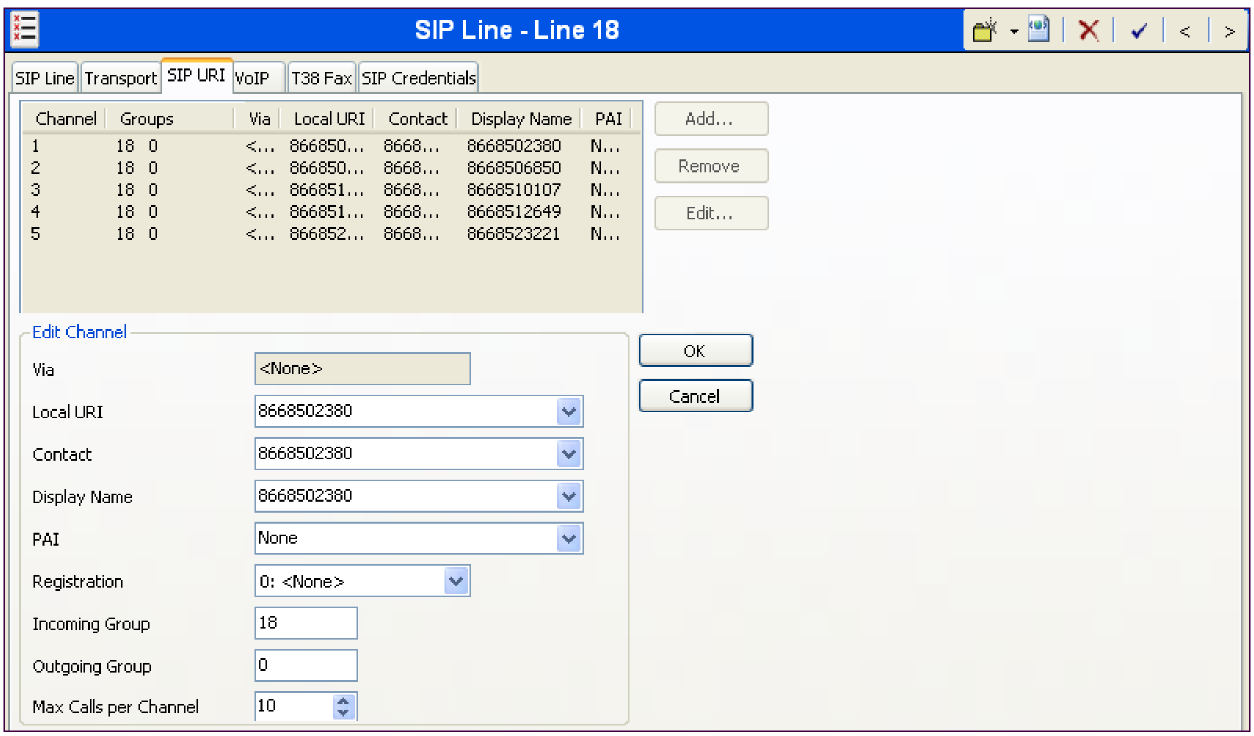
Select the **Transport** tab. The **ITSP Proxy Address** is set to the inside IP address of the Avaya SBCE as shown in **Figure 1**. In the **Network Configuration** area, "**TCP**" is selected as the **Layer 4 Protocol**. The **Send Port** and **Listen Port** can retain the default value 5060. The **Use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



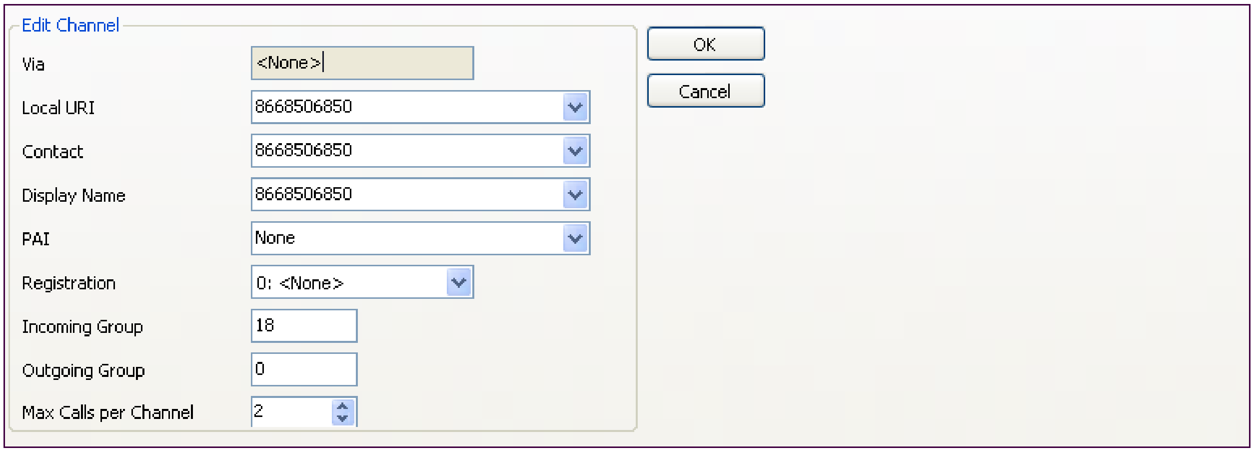
5.4.4. SIP Line - SIP URI Tab

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 sample configuration, each of the Verizon-provided toll free numbers are entered as a SIP URI, with the specific number entered in the **Local URI**, **Contact**, and **Display Name** fields. The **PAI** parameter is left at the default value of "**None**". The **Registration** parameter is set to the default "**0: <None>**" since Verizon Business IPCC 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 **0**, is not relevant in that Verizon Business IPCC service is inbound only. Click **OK**.

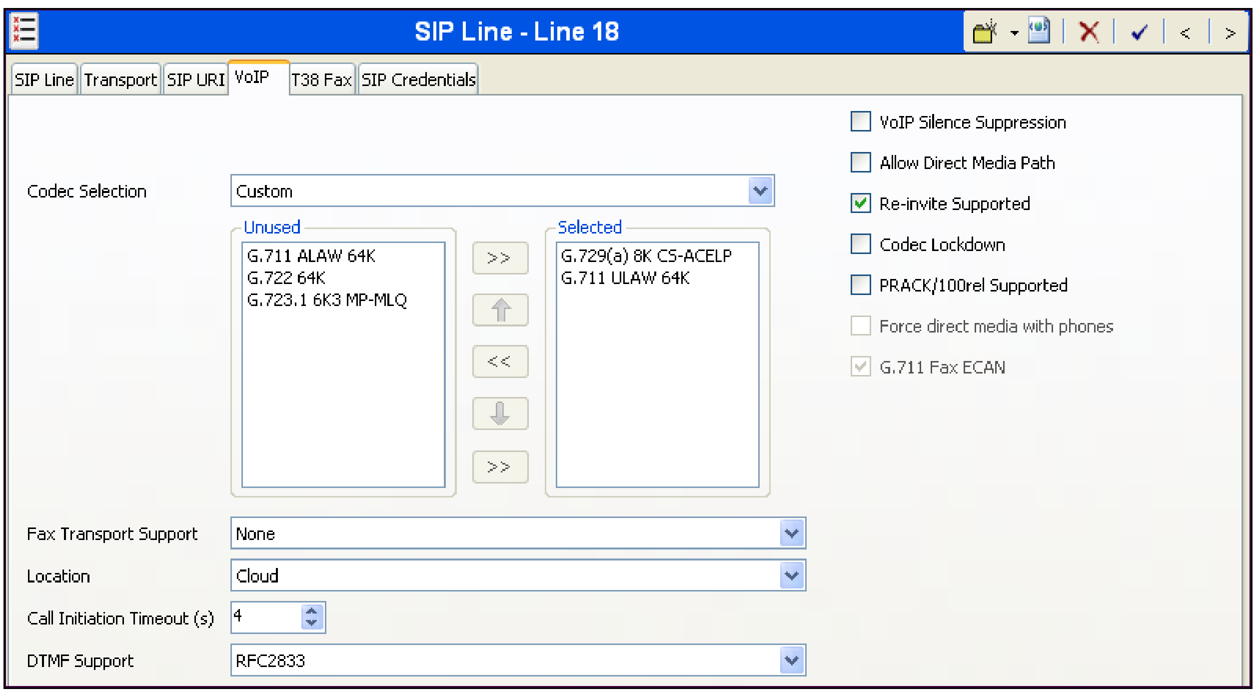


IP Office allows the number of simultaneous calls to a specific SIP URI to be managed using the **Max Calls per Channel** field. In the following screen, note that the **Max Calls per Channel** field has been changed from the default 10 to 2. With this configuration, two simultaneous calls to the number 866-850-6850 will be allowed. Once two calls are active, and a third call is attempted to 866-850-6850, IP Office will return a SIP 4xx response. Calls to other toll-free numbers using this same SIP Line are unaffected by the Max Calls per Channel for a different URI. Therefore, this approach could be used to control the maximum number of calls to each of the specific toll-free numbers, preventing a surge of calls to a given toll-free number from monopolizing the available call handling capacity of the access line or IP Office resources. An alternative means to restrict the number of simultaneous calls to a toll-free number that terminates on a hunt group would be to limit the queue size of the destination hunt group. If a non-priority call arrives to IP Office to a hunt group with a fixed size queue, and the queue is full, and there is no voice mail for the hunt group, IP Office returns a 486 Busy Here. See **Section 5.5.4** for hunt group configuration.



5.4.5. SIP Line - VoIP Tab

Select the **VoIP** tab. In the sample configuration, the **Codec Selection** was configured using the "**Custom**" option, allowing an explicit ordered list of codecs to be specified, different from the system default (see **Section 5.2.3**). The arrow buttons can be used such that **G.729(a) 8K CS-ACELP** and **G.711 ULAW 64K** codecs are listed in the **Selected** column. This configures IP Office to support either G.729a or G.711MU for this SIP Line. 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. Verizon preferred the G.729A codec in SDP, while also allowing the G.711MU codec. The IP Office configuration shown below matches these Verizon preferences. In the course of testing, the IP Office configuration was varied such that G.711MU was the preferred or only codec listed, and G.711MU calls were also successfully verified. The **PRACK/100rel Supported** parameter should be left at the default unchecked value. Since the Verizon Business IP Contact Center service does not support fax, the **Fax Transport Support** parameter is set to "**None**", and the **T38 Fax** tab need not be visited. Since the Verizon Business IPCC service does not require registration, the **SIP Credentials** tab need not be visited. Click **OK** (not shown).

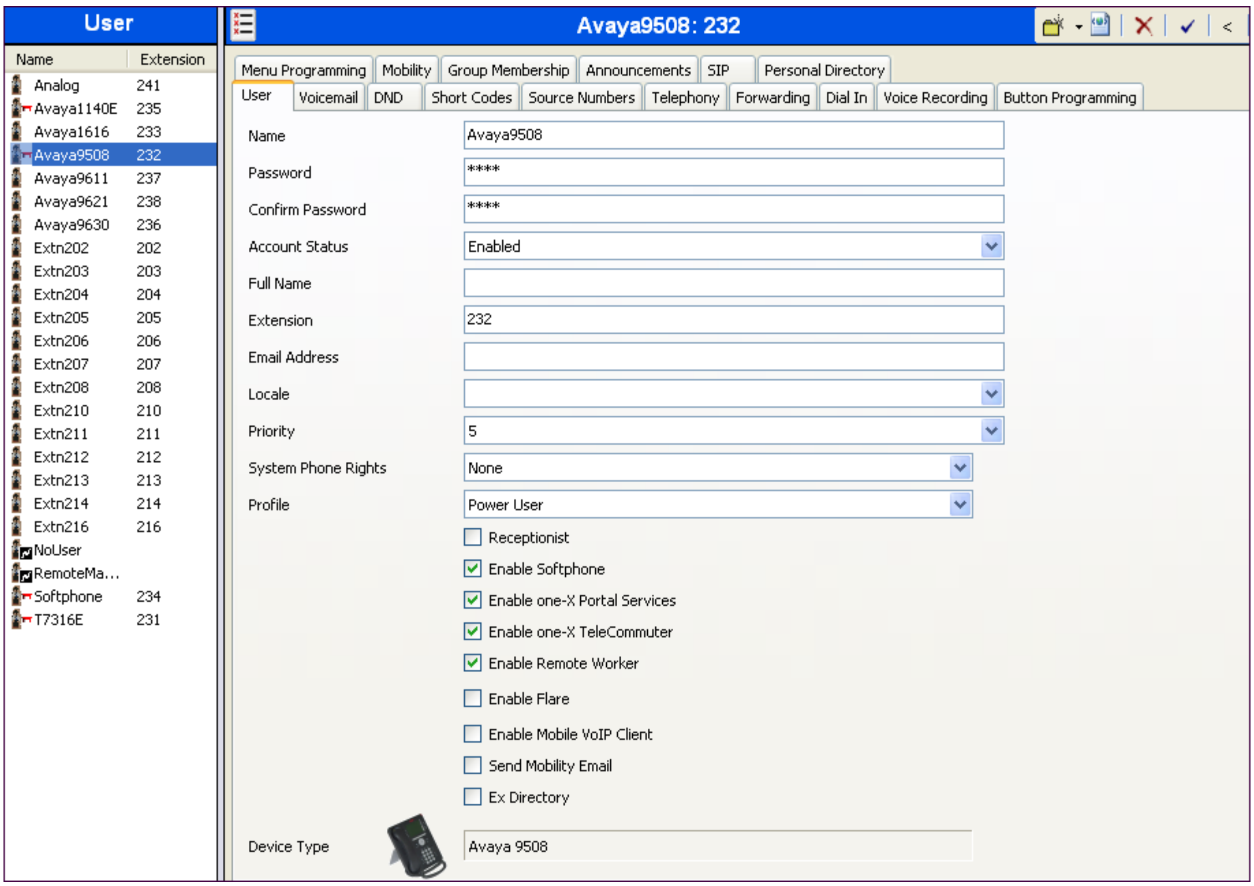


5.5. Users, Extensions, and Hunt Groups

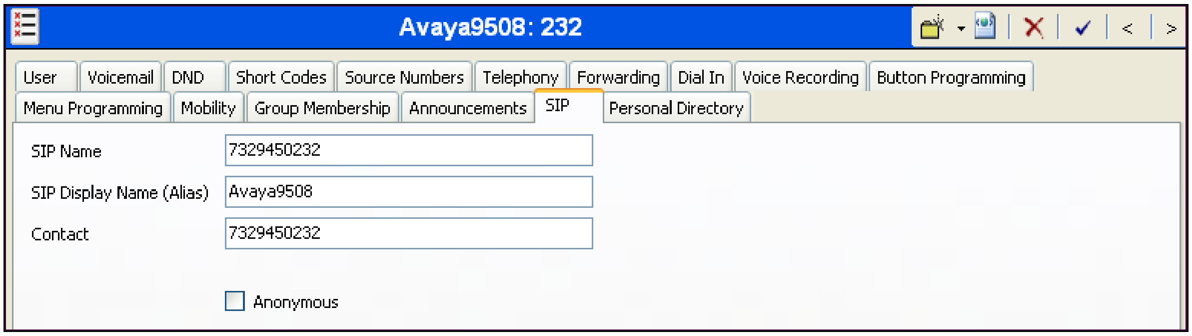
In this section, examples of IP Office Users, Extensions, and Hunt Groups will be illustrated. In the interests of brevity, not all users and extensions shown in **Figure 1** will be presented, since the configuration can be easily extrapolated to other users. To add a User, right click on **User** in 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.

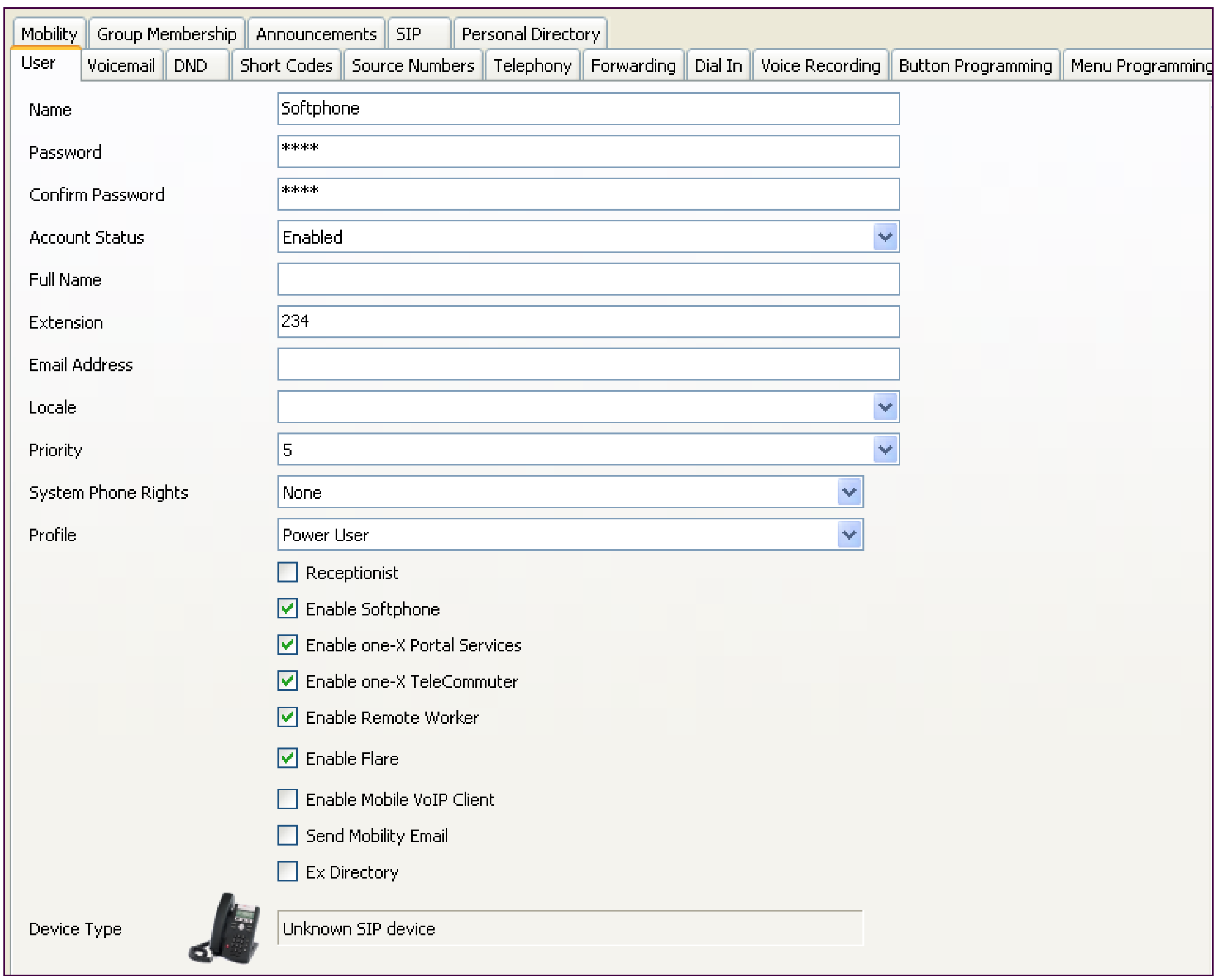


The following screen shows the **SIP** tab for User 232. In the sample configuration, the **SIP Name** and **Contact** parameters are configured with a Verizon Business IP Trunk DID number for the user, 7329450232. As shown in [VZBIPT-IPO9SBC], these parameters configure the user part of the SIP URI in the From header for outgoing SIP trunk calls, and allow matching of the SIP URI for incoming calls from Verizon IP Trunk service, 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.

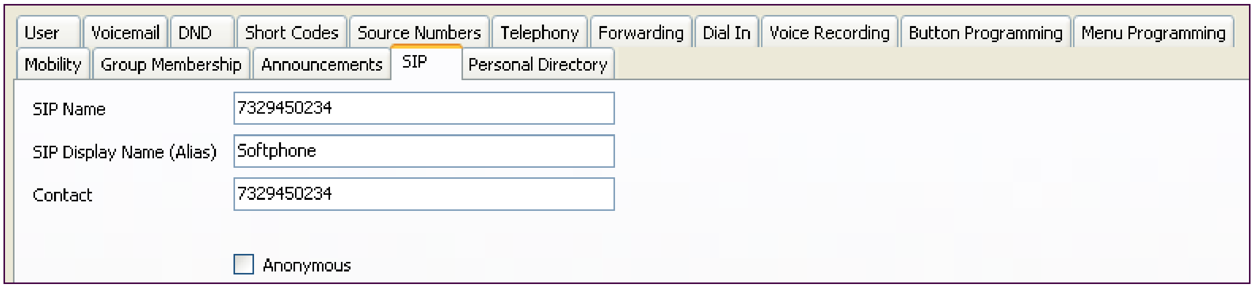


5.5.2. Avaya Flare Experience User 234 with IP Office Softphone Privileges

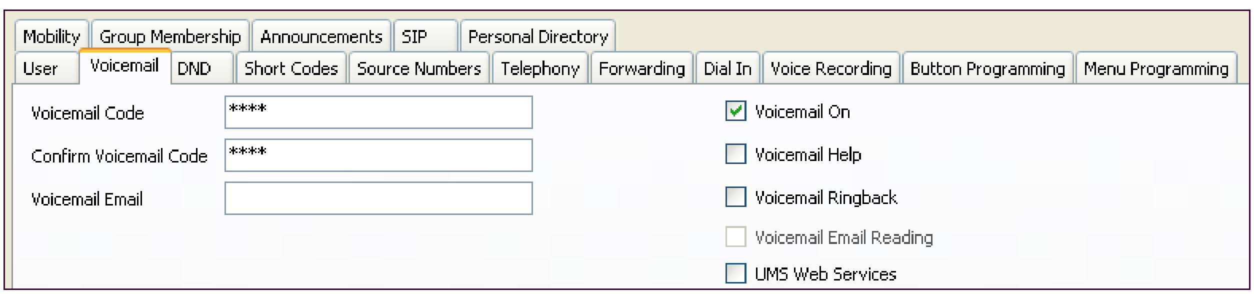
The following screen shows the **User** tab for User 234. This user corresponds to a user that will be granted "Power User", Flare features and Avaya IP Office Softphone features. The **Profile** parameter is set to "Power User". The **Enable Softphone** and **Enable Flare** boxes are checked.



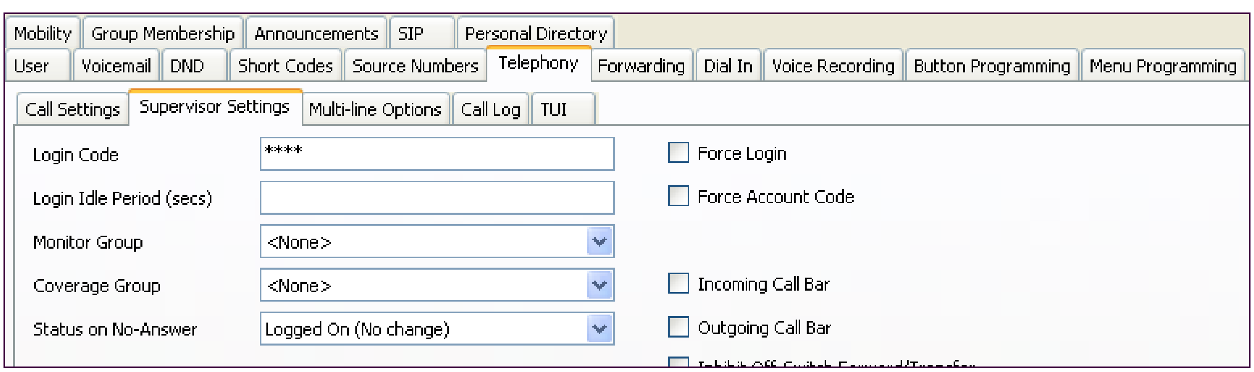
Like the user with extension 232, the **SIP** tab for the user with extension 234 is configured with a **SIP Name** and **Contact** specifying the user's Verizon Business DID number using the Verizon Business IP Trunk service, as detailed in [VZBIPT-IPO9SBC].



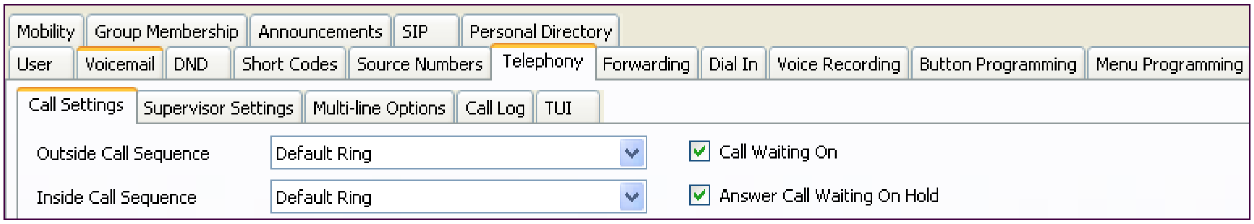
The following screen shows the Voicemail tab for the user with extension 234. The **Voicemail On** box is checked, and a voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters. In the verification of these Application Notes, incoming calls from the Verizon Business IPCC service to this user were redirected to Voicemail Pro after no answer. Voicemail messages were recorded and retrieved successfully. Voice mail navigation and retrieval were performed locally and from PSTN telephones using the Verizon Business IPCC toll-free numbers, to test DTMF using RFC 2833, and to test assignment of a Verizon-provided toll free number to the "Voicemail Collect" feature (e.g., via the *17 short code shown in **Section 5.6**).



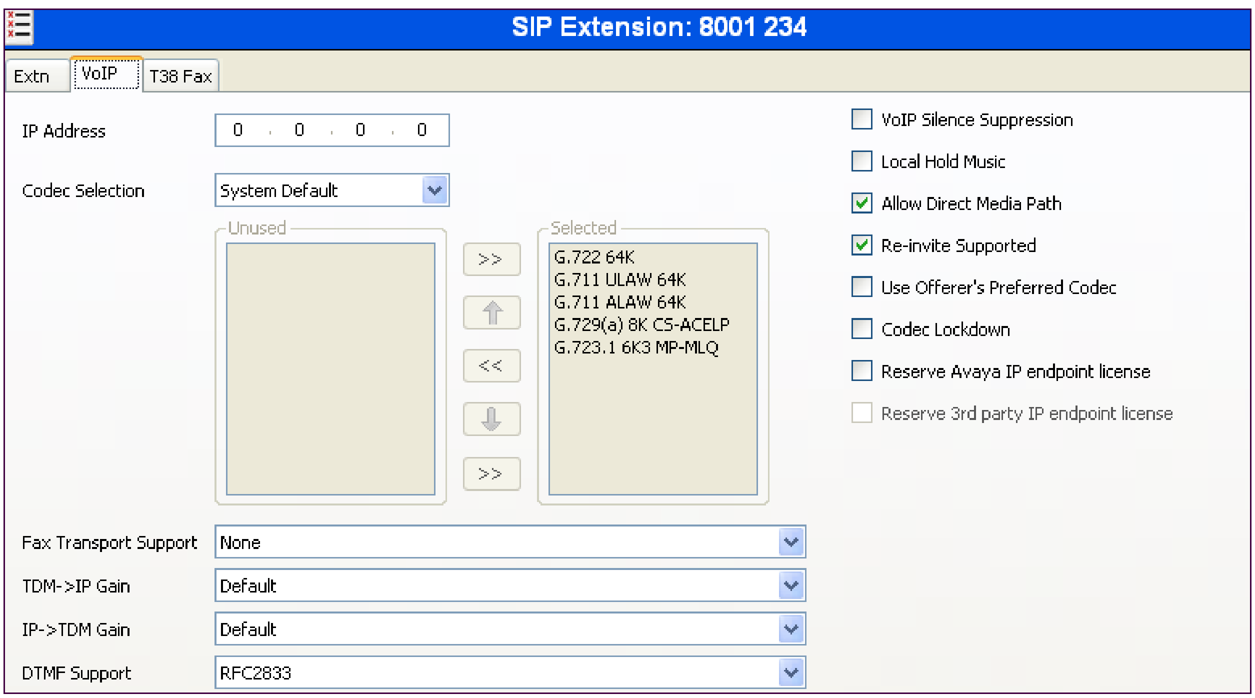
Select the **Telephony** tab. Then select the **Supervisor Settings** tab as shown below. The **Login Code** will be used by the Flare Experience and IP Office Softphone user as the login password.



Select the **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow an IP Office Softphone logged in as this extension to have multiple call appearances (e.g., necessary for call transfer from IP Office Softphone).



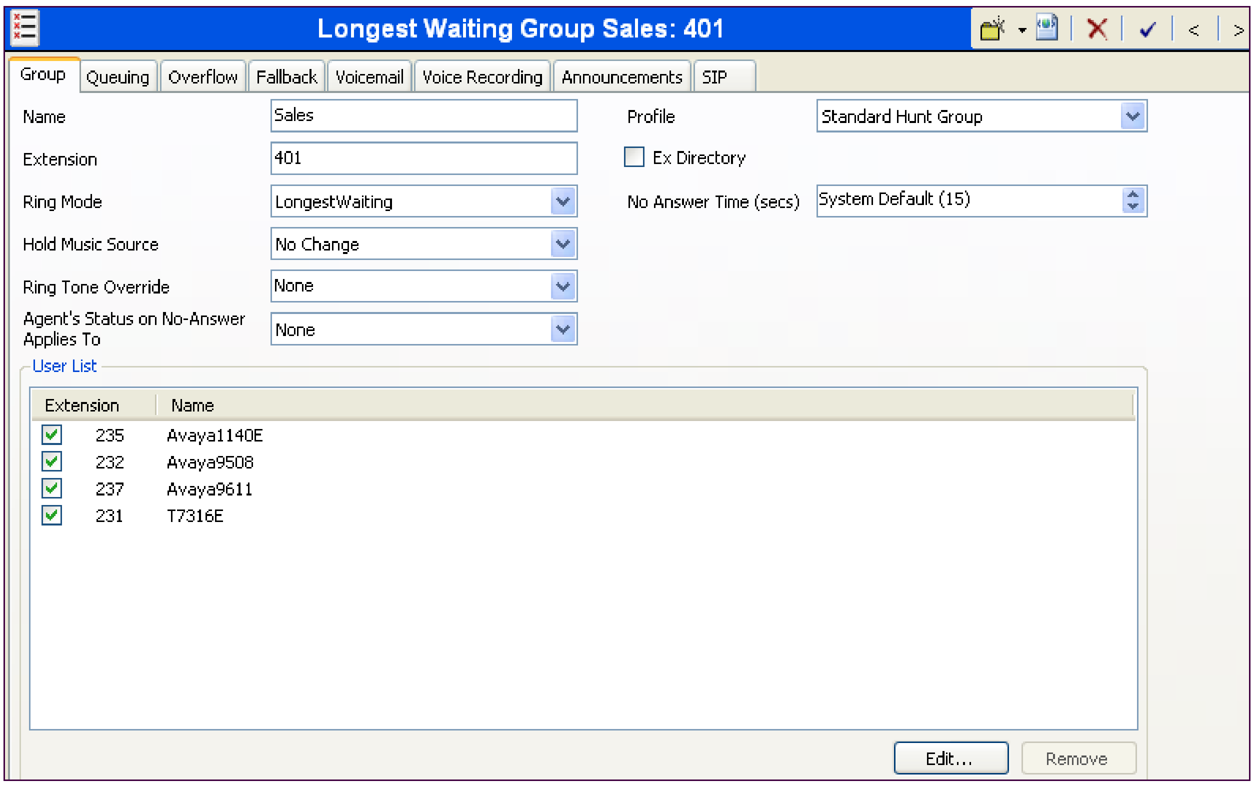
The following screen shows the Extension information for this user, simply to illustrate the **VoIP** tab available for a SIP Telephone. To view, select **Extension** (not shown) from the Navigation pane, and the appropriate extension from the Group pane. Select **VoIP** in the Details pane. The new **Codec Selection** parameter may retain the default setting "System Default" to follow the system configuration shown in **Section 5.2.3**Error! Reference source not found.. 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.



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 **HuntGroup** (not shown) from the Navigation pane, and select **New**. To view or edit an existing hunt group, select **HuntGroup** from the Navigation pane, and the appropriate hunt group from the Group pane.

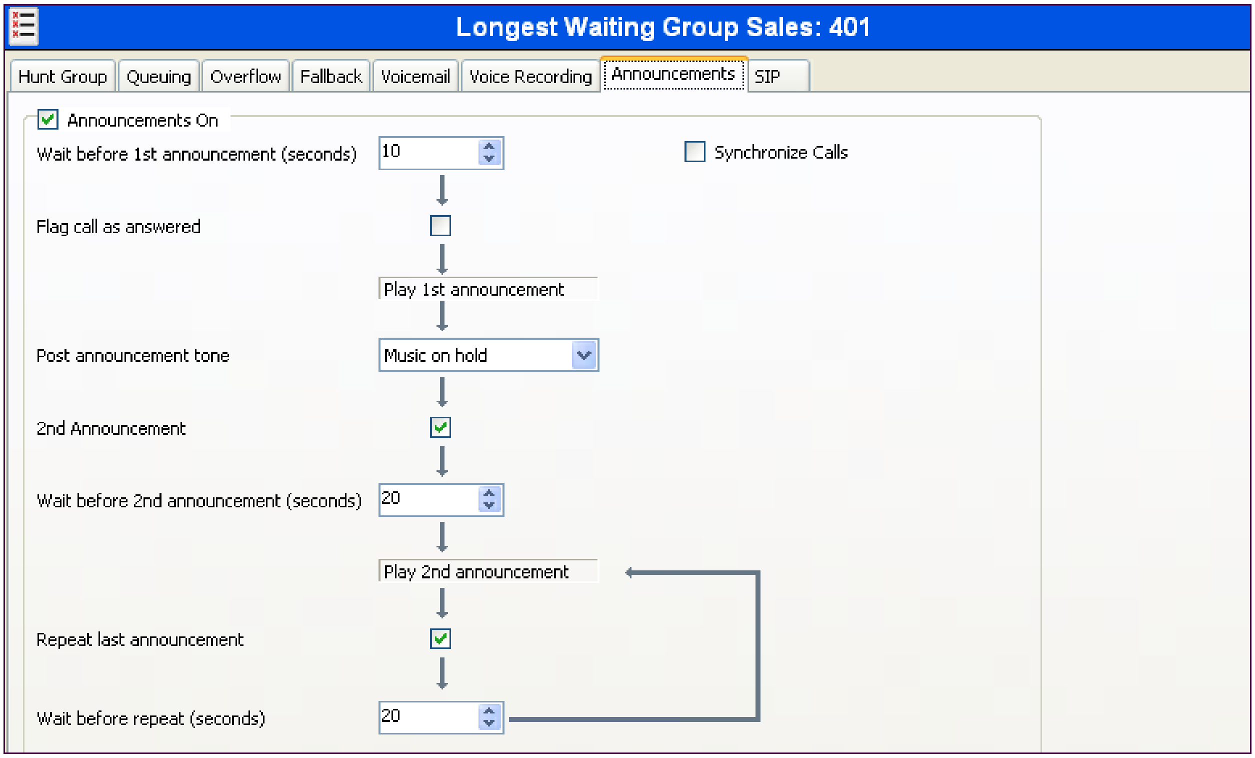
The following screen shows the **Hunt Group** tab for a hunt group with Extension 401 and **Name** “Sales”. This hunt group was configured to contain various telephones from **Figure 1**. The **Ring Mode** was set to “**LongestWaiting**” (i.e., “longest waiting”, most idle user receives next call). Click the **Edit** button to change the **User List**.



The following screen shows the **Queuing** tab for hunt group 401. In the sample configuration, the hunt group was configured to allow queuing so that incoming Verizon toll-free calls could be queued when all the members of the hunt group were busy on calls. In the testing associated with these Application Notes, the **Queue Length** was varied using both "No Limit" and specifically sized queues. For example, if the **Queue Length** is configured to 2, and if two calls are already in queue, a third call to the Verizon toll-free number corresponding to this hunt group will get busy tone because IP Office will send a 486 Busy Here (i.e., if there is no Voicemail for the hunt group). As another example, if the **Queue Length** has a fixed limit of 2, and if two calls are already in queue, a third call to the Verizon toll-free number destined for this hunt group from a priority caller (see **Section 5.7.3**) will be queued ahead of non-priority callers, temporarily expanding the queue.

The following screen shows the **Announcements** tab for hunt group 401. In the sample configuration, when a call arrives when all members of the hunt group are busy on calls, the caller will first hear ring back tone. If a member of the hunt group does not become available after 10 seconds, the call will be answered by IP Office (i.e., 200 OK will be sent to Verizon), and the toll-free caller will hear a first announcement. Note that the **Flag call as answered** box is relevant for reporting applications, but does not change the fact that IP Office will answer the call when the first announcement is played. If the call is still not answered after the first announcement completes, the caller will hear music, a repeating second announcement, music, and so on until the call is answered by a member of the hunt group, or answered by voicemail for the hunt group (if configured). If a member of the hunt group becomes available while the caller is listening to ring back, music, or an announcement, the call is de-queued and delivered to the available member.

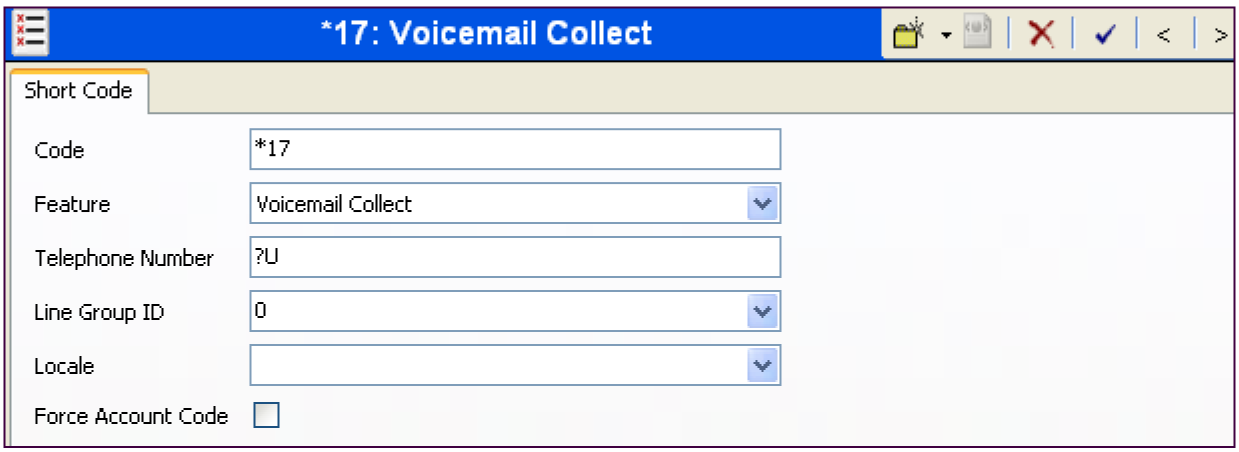
IP Office supports priority for queuing. For example, if low priority calls are waiting in queue, a higher priority call entering queue can be moved to the front of the queue and serviced before lower priority callers. For an inbound SIP trunk call, the priority can be specified on the Incoming Call Route as shown in **Section 5.7.3**.



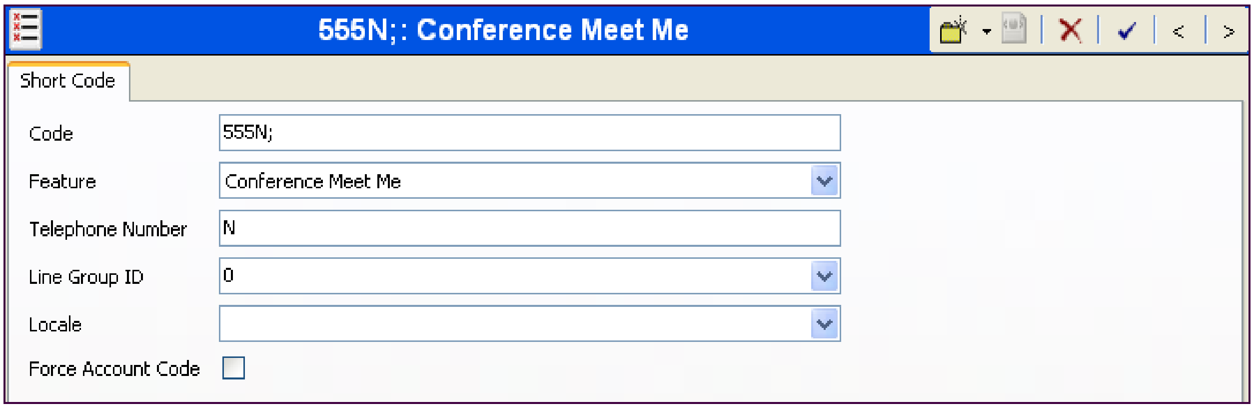
5.6. Short Codes

In this section, various examples of IP Office short codes will be illustrated. To add a short code, right click on **Short Code** (not shown) 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.

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** "***17**" is defined for **Feature** "**Voicemail Collect**". This short code will be used as one means to allow a Verizon toll-free number to be programmed to route directly to voice messaging, via inclusion of this short code as the destination of an Incoming Call Route. See **Section 5.7** for configuration of Incoming Call Routes.



The following screen illustrates another short code. In this case, the **Code** "**555N;**" is defined for **Feature** "**Conference Meet Me**" and **Telephone Number** "**N**". In the verification of these Application Notes, this short code was used in conjunction with a Voicemail Pro module named "MeetMeConf". Although the Voicemail Pro configuration is beyond the scope of these Application Notes, the module enabled a PSTN caller to dial a Verizon toll-free number, be prompted to enter a conference ID and PIN by Voicemail Pro, and then be transferred to the appropriate meet-me conference based on the ID entered by the caller. Local IP Office callers could also dial 555xxx to join the corresponding conference with ID xxx.

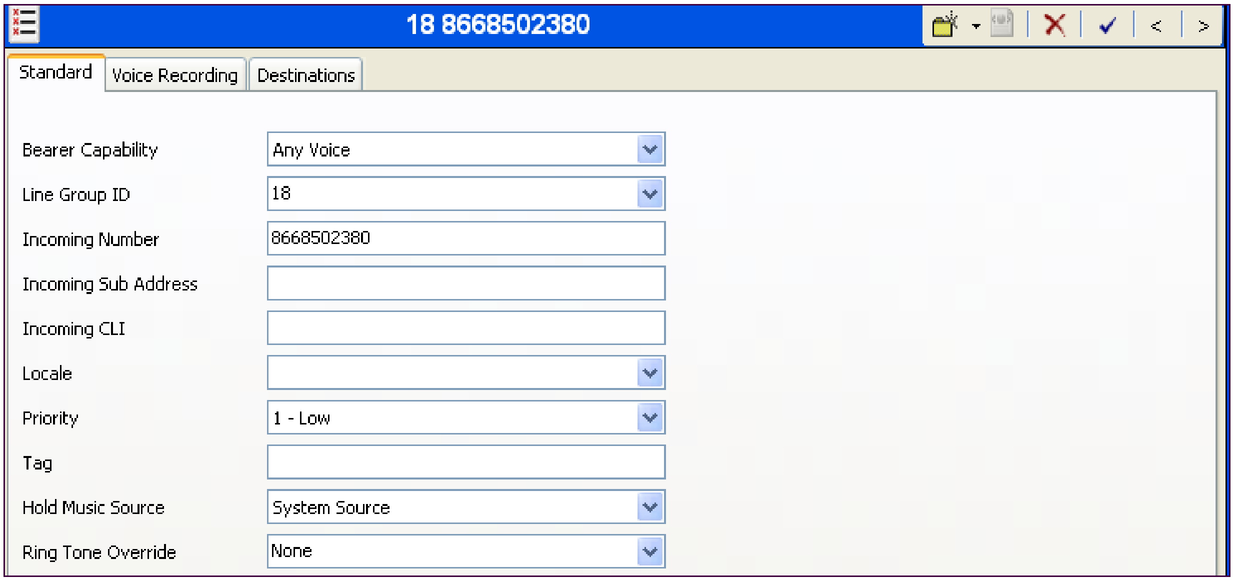


5.7. Incoming Call Routes

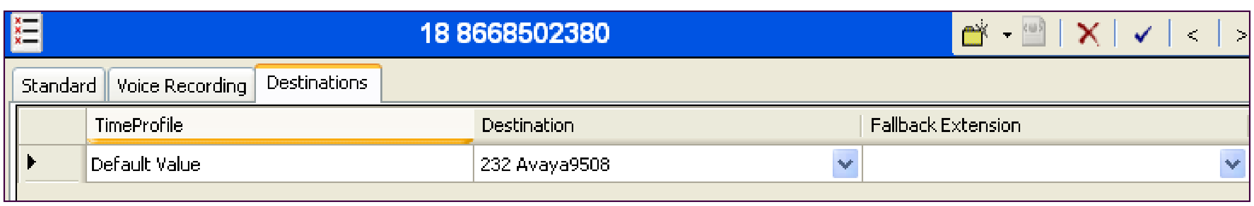
In this section, IP Office Incoming Call Routes are illustrated. Each Incoming Call Route will map a Verizon Business toll-free number to a destination user, group, or function on IP Office. In some cases, the destination will be chosen based on the combination of the toll-free number and the caller id of the caller. Example mappings are summarized in **Table 1** in **Section** 0. To add an incoming call route, right click on **Incoming Call Route** (not shown) 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.

5.7.1. Incoming Call Route to a Specific Telephone Extension

In the screen shown below, the incoming call route for **Incoming Number** "8668502380" is illustrated. The **Line Group Id** is "18", matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to the Verizon Business IPCC service, in **Section 5.4.4**.



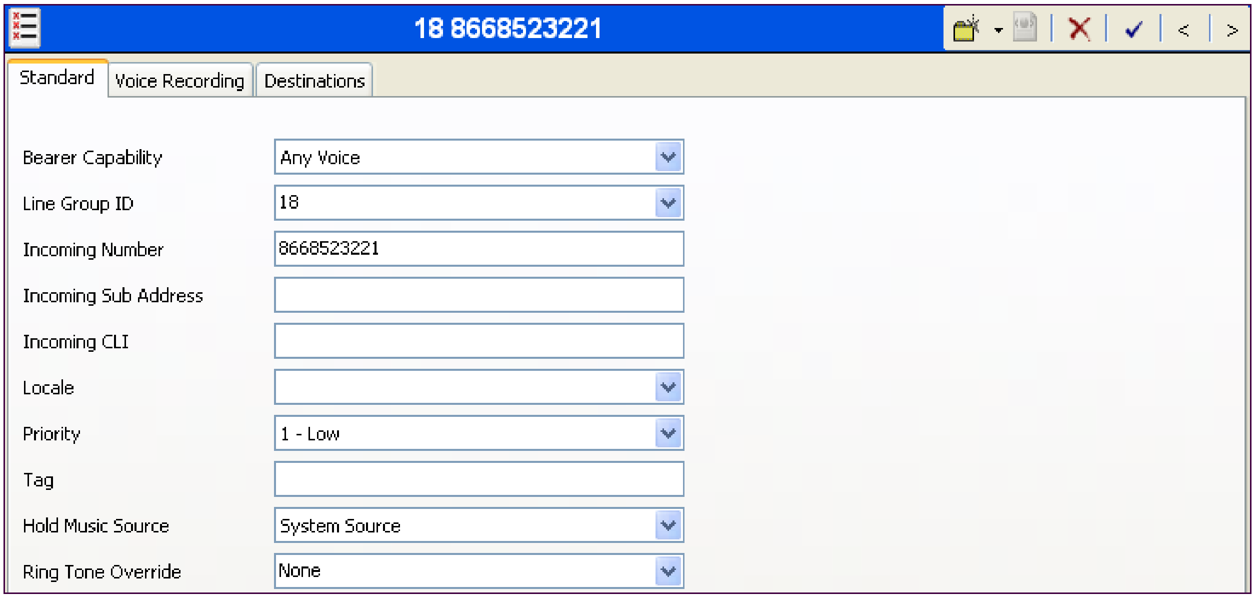
Select the **Destinations** tab. From the **Destination** drop-down, select an extension to receive the call when a PSTN user dials 8668502380. As shown in **Table 1**, 8668502380 is the number associated with IP Office user extension 232. (The **Destination** was changed in the course of testing to associate different destinations with the toll-free numbers.)



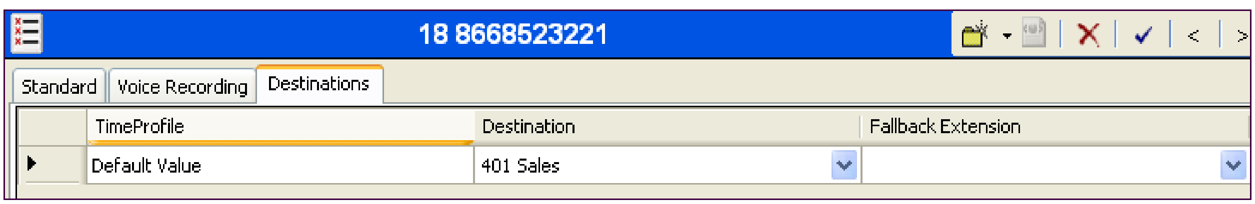
Incoming Call Routes for other direct mappings of toll-free numbers to IP Office users are not presented here, but are configured in the same fashion.

5.7.2. Incoming Call Routes to a Hunt Group by Dialed Toll-Free Number

In the screen shown below, an incoming call route for **Incoming Number** “8668523221” 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**.



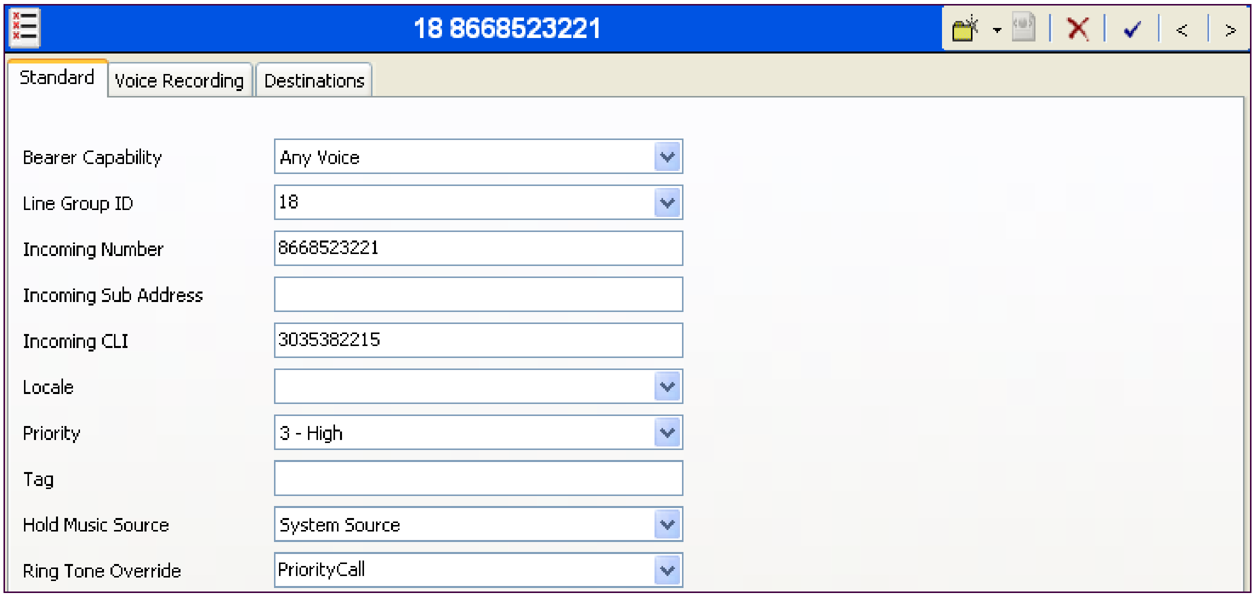
Select the **Destinations** tab. From the **Destination** drop-down, select the destination to receive the call when an arbitrary PSTN user dials 8668523221. As shown in **Table 1**, 8668510107 is the toll-free number associated with IP Office hunt group extension 401, the “Sales” hunt group.



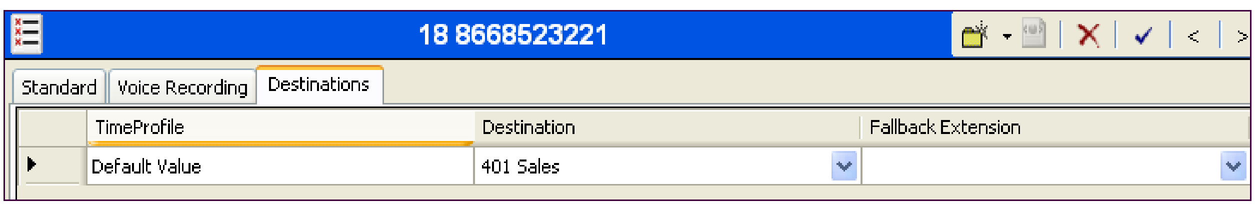
5.7.3. Incoming Call Routes Based on Calling Party Number

This section presents a simple example showing that IP Office can use the calling party number to distinguish call priority or call destination, for calls to the same toll-free number. Although the matching shown here is based on the full calling number, partial matching is also possible (e.g., to match a calling area code for a targeted geographic treatment).

In the screen shown below, the incoming call route for **Incoming Number** “8668523221” and **Incoming CLI** “3035387022” 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**. Note that the **Incoming Number** is the same as the toll-free number configured in the previous section. This route will be used for calls to the toll-free number specifically from a caller with caller ID “3035387022”. In this case, to allow this caller to be treated with priority when calling in for “Sales”, the **Priority** field is set to “**3 – High**”.

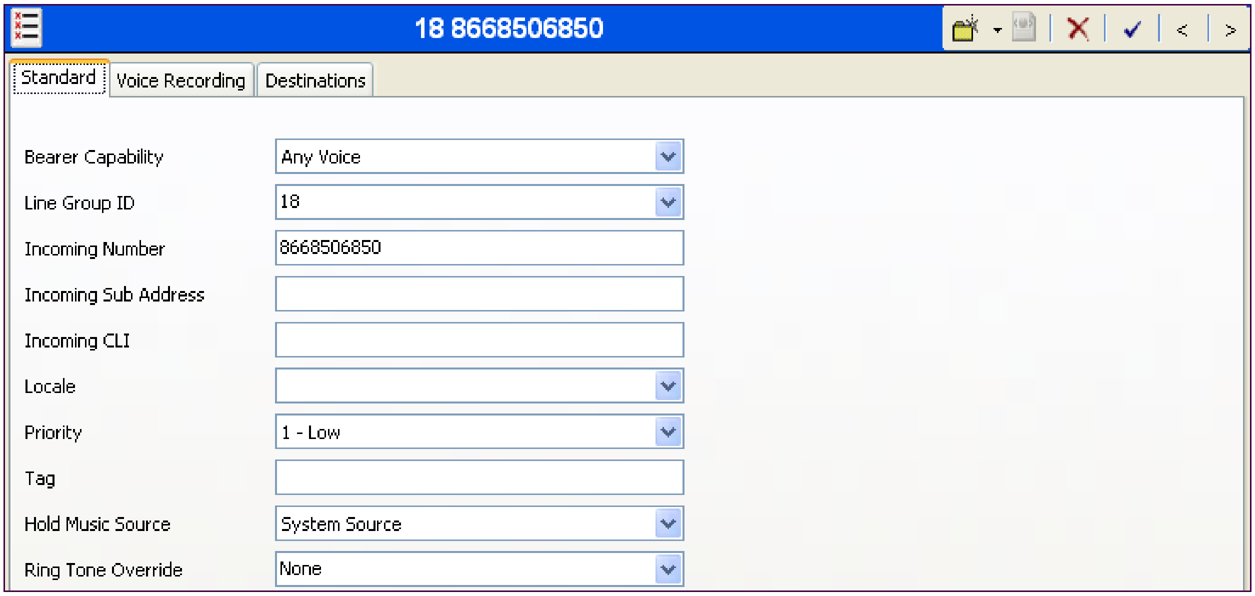


Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when PSTN user 3035387022 dials 8668523221. In this case, the **Destination** is also the hunt group “**401 Sales**”, but since high priority has been configured via the **Standard** tab, incoming calls from this caller will move to the front of the queue, and be serviced before calls waiting in queue from other non-priority callers.



5.7.4. Incoming Call Routes to Various IP Office Features

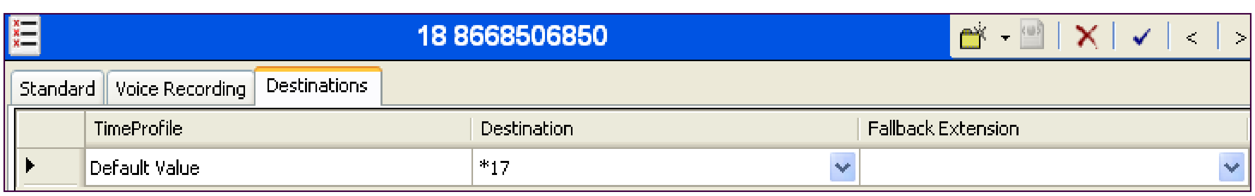
In the sample configuration, the incoming call route for **Incoming Number** "8668506850" was varied to test different destination features, such as Voice Mail, Mobile Call Control, and Conference Bridge, as shown in **Table 1** in **Section 0**. The screen showing the **Standard** tab for this toll-free number is shown below.



When configuring an Incoming Call Route, the **Destination** field can be manually configured with a number such as a short code, or certain keywords available from the drop-down list. At different times during testing, the **Destinations** tab for 8668506850 was configured to contain the following destinations:

- "***17**" (short code "Voicemail Collect", as shown in **Section 5.6**). With this destination, an incoming call to 8668506850 will be delivered directly to voice mail, allowing the caller to log-in to voice mail and access messages.
- "**VM:MeetMeConf**" With this destination, an incoming call to 8668506850 will be delivered directly to the Voicemail Pro Module "MeetMeConf" created for use as a conference bridge.

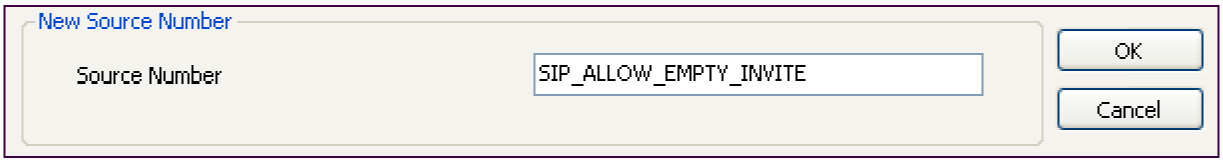
An example screen showing the short code configured for Voicemail Collect is shown below.



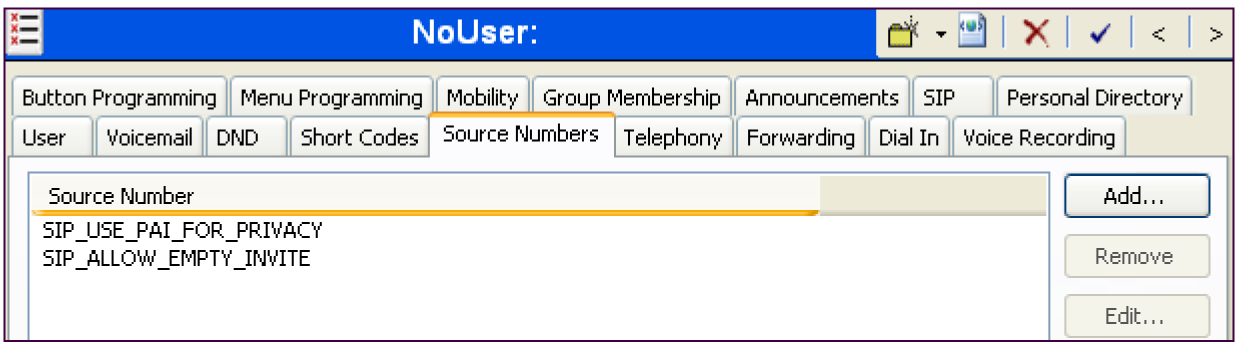
5.8. SIP INVITE Without SDP

The Verizon IPCC “enhanced transfer” service sends an initial INVITE without SDP to the transfer-to site of an enhanced transfer. To allow the IP Office to be the recipient of this type of transferred call, a NoUser Source Number can be configured as follows.

From the Navigation pane, select **User**. From the Group pane, scroll down past the configured users and select the user named **NoUser**. From the NoUser Details pane, select the tab **Source Numbers**. Press the **Add…** button to the right of the list of any previously configured Source Numbers. In the **Source Number** field shown below, type **SIP_ALLOW_EMPTY_INVITE**.



The source number **SIP_ALLOW_EMPTY_INVITE** should now appear in the list of Source Numbers as shown below.

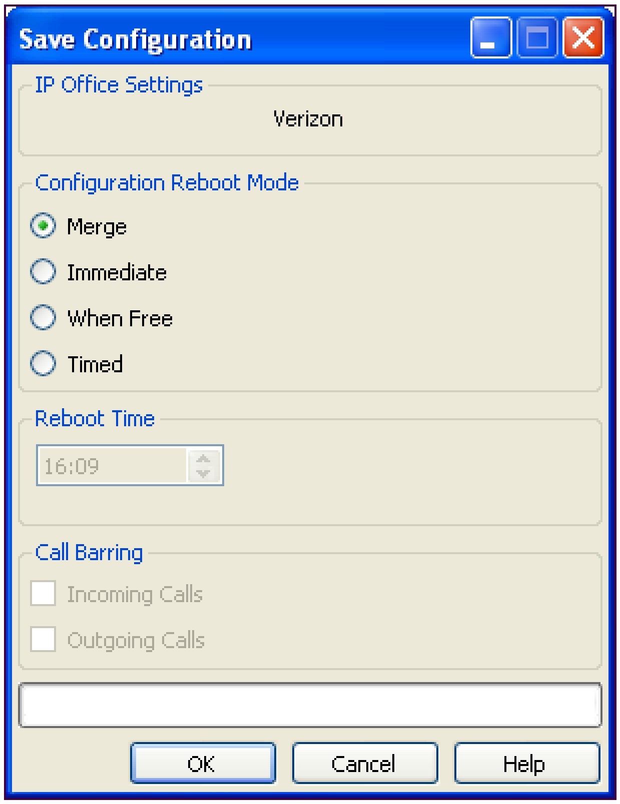


With this configuration, IP Office can be the recipient of a Verizon IPCC enhanced transfer where the initial INVITE will not have SDP information

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.

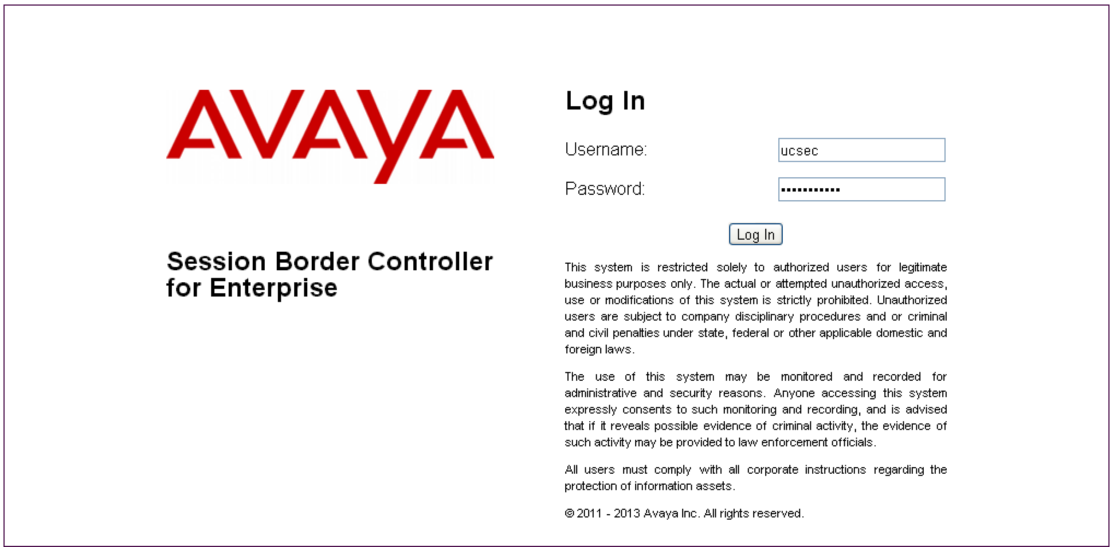


6. Configure Avaya Session Border Controller for Enterprise

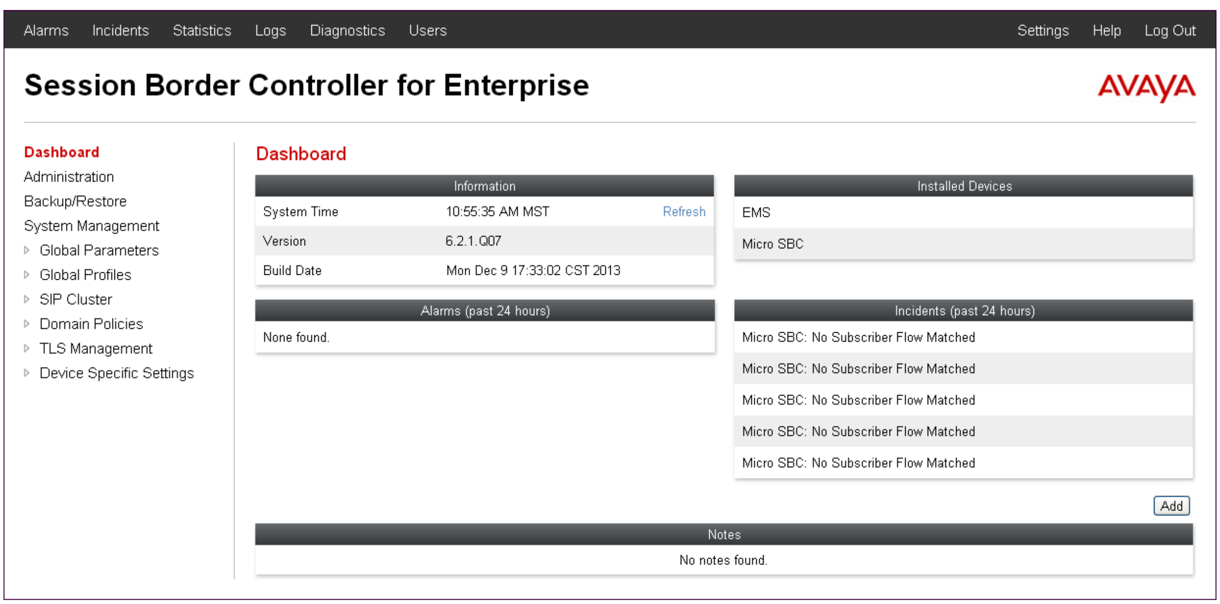
This section covers the configuration of the Avaya SBCE. It is assumed that the Avaya SBCE software has already been installed.

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.

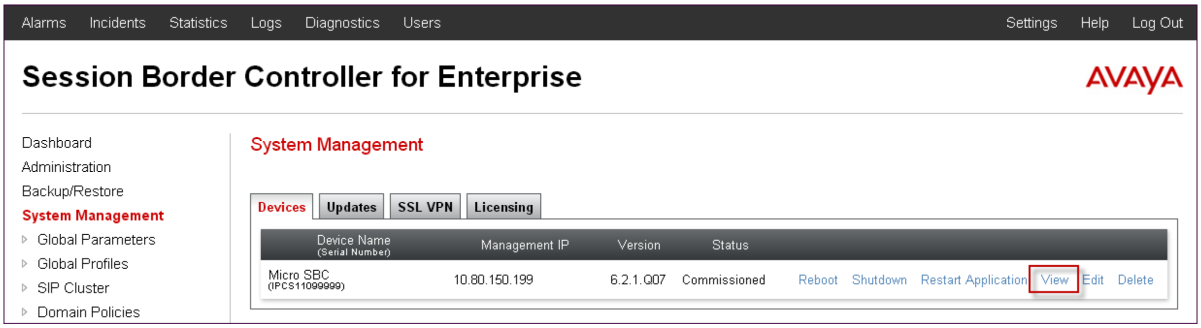
Log in with the appropriate credentials. Click **Log In**.



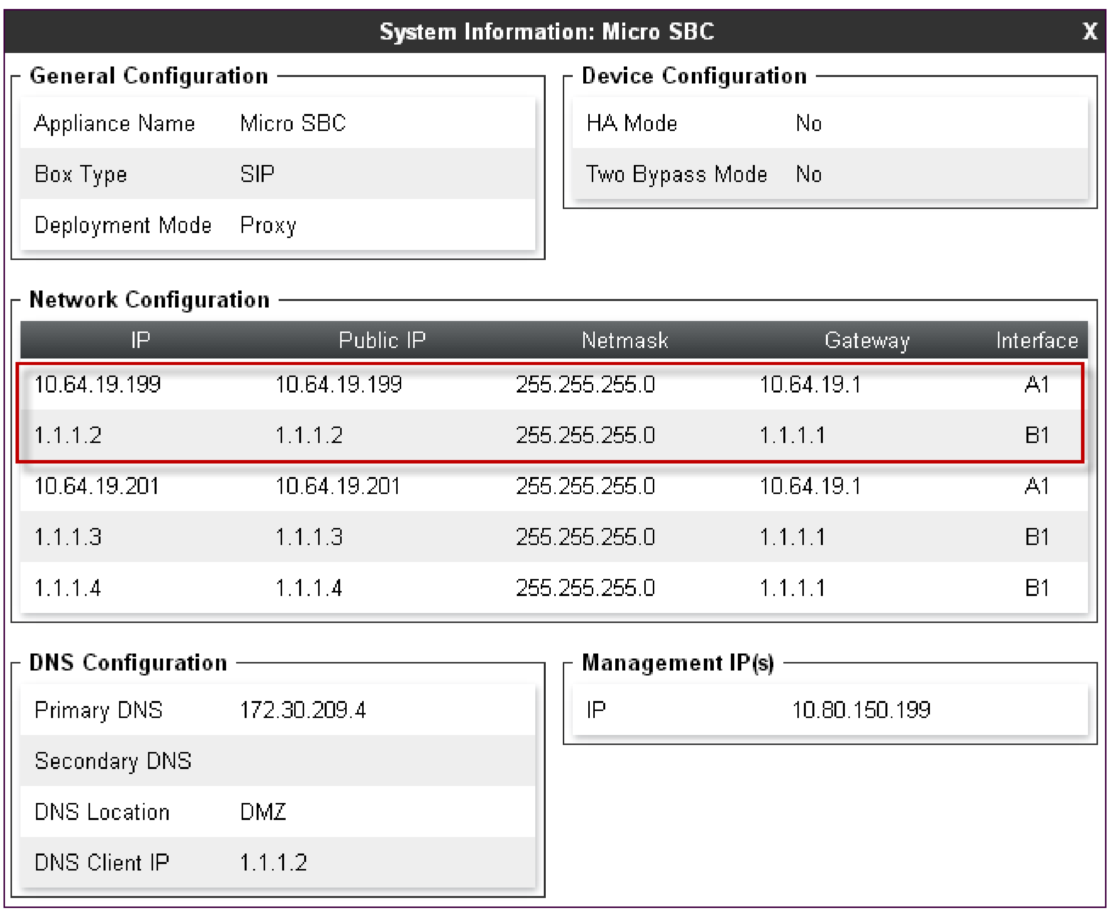
The Dashboard for the Avaya SBCE will appear.



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.

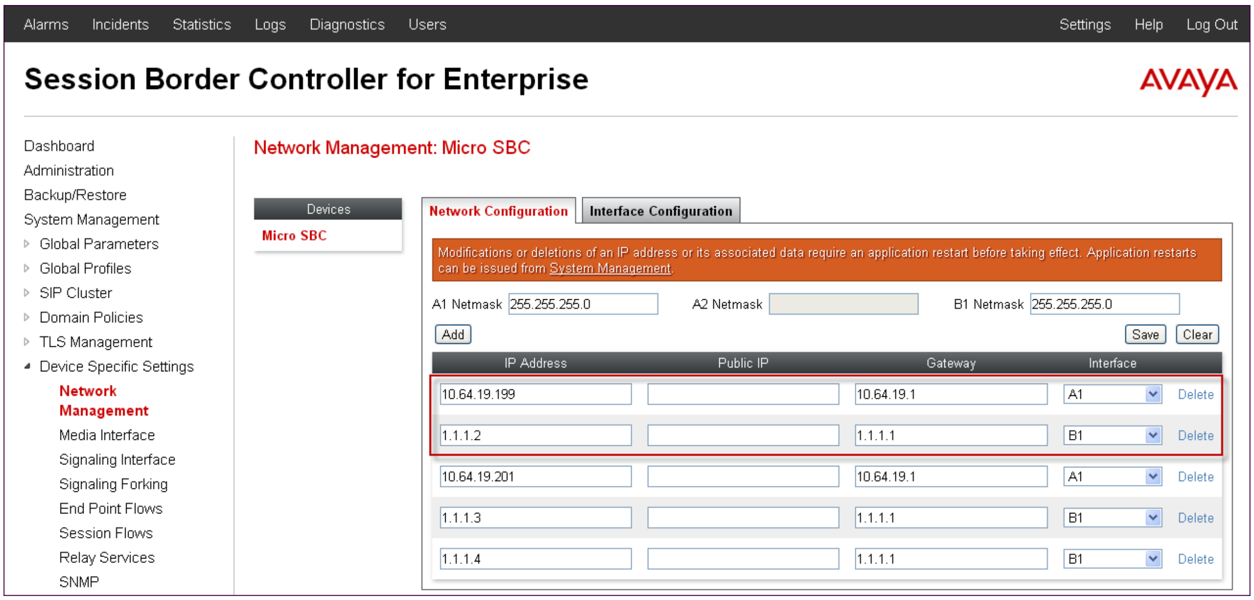


The **System Information** screen shows the **Network Configuration**, **DNS Configuration** and **Management IP(s)** information provided during installation and corresponds to **Figure 1**. The highlighted **A1** and **B1** IP addresses are the ones relevant to the configuration of the SIP trunk to 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.

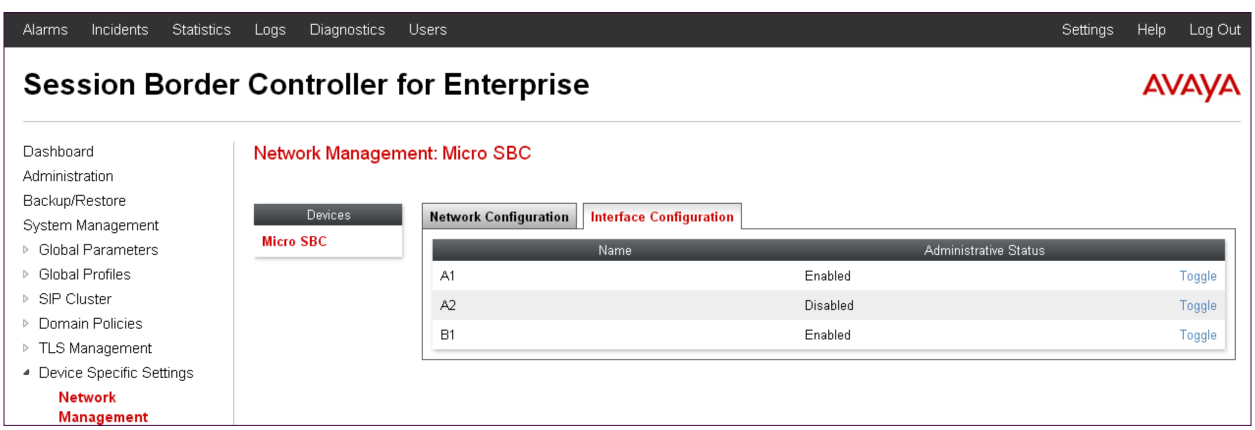


6.1. Network Management

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.

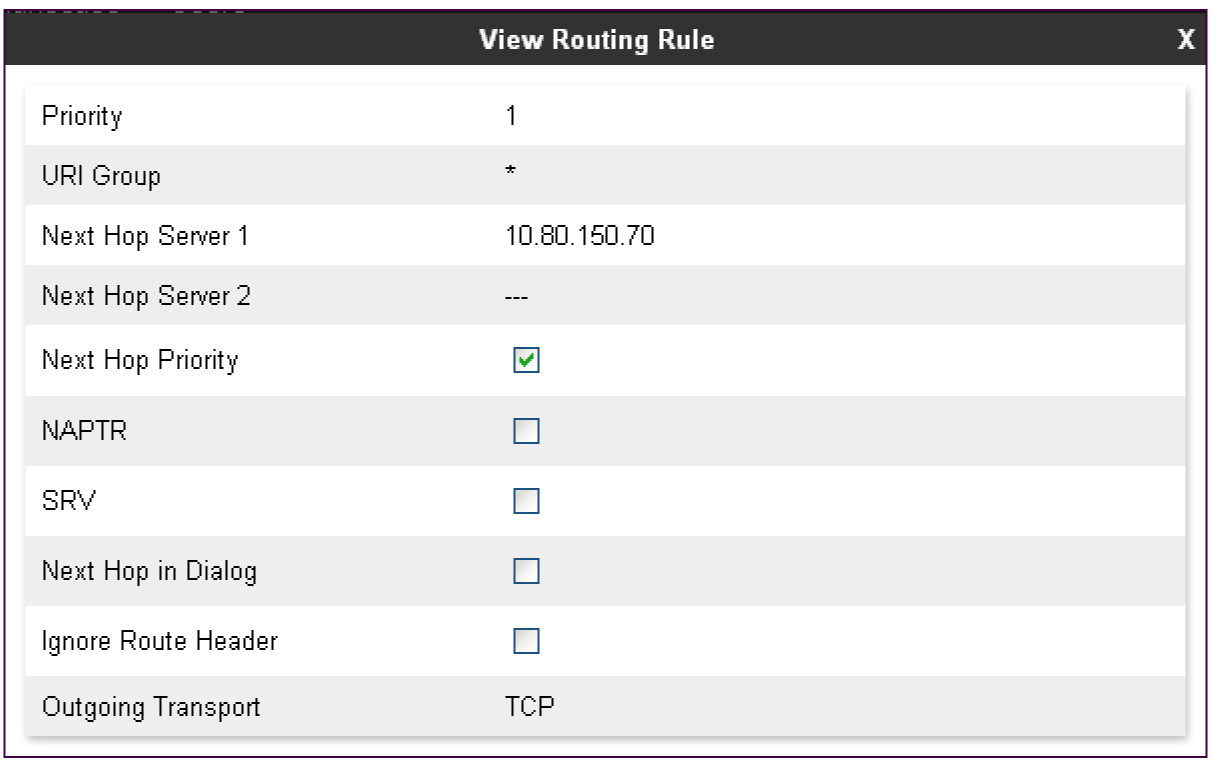


6.2. Routing Profile

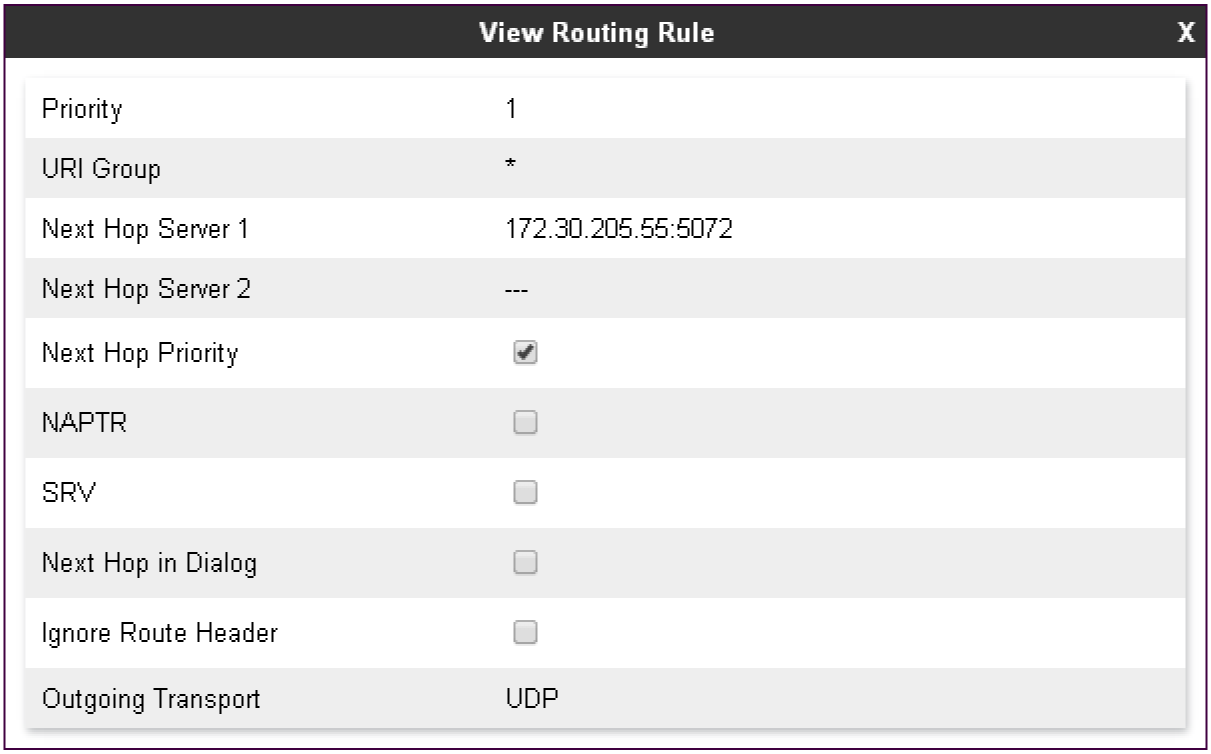
Routing profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and thereby ascertain which security features will be applied to those packets. Parameters defined by Routing Profiles include packet transport 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 Contact Center SIP 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 Routing Profile "**IP Office**" created for 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**. Leave the **Routing Priority based on Next Hop Server** box checked and enter "**TCP**" for the **Outgoing Transport field** matching the **Layer 4 Protocol** set in IP Office **SIP Line →Transport** in **Section 5.4.3**.



The following screen shows Routing Profile "**Vz_IPCC**" created for Verizon. The **Next Hop Server 1** is the IP address and port of the Verizon SIP signaling interface. It is only necessary to include the port after the IP address when it is not the default SIP port. Choose **UDP** for **Outgoing Transport**.



View Routing Rule

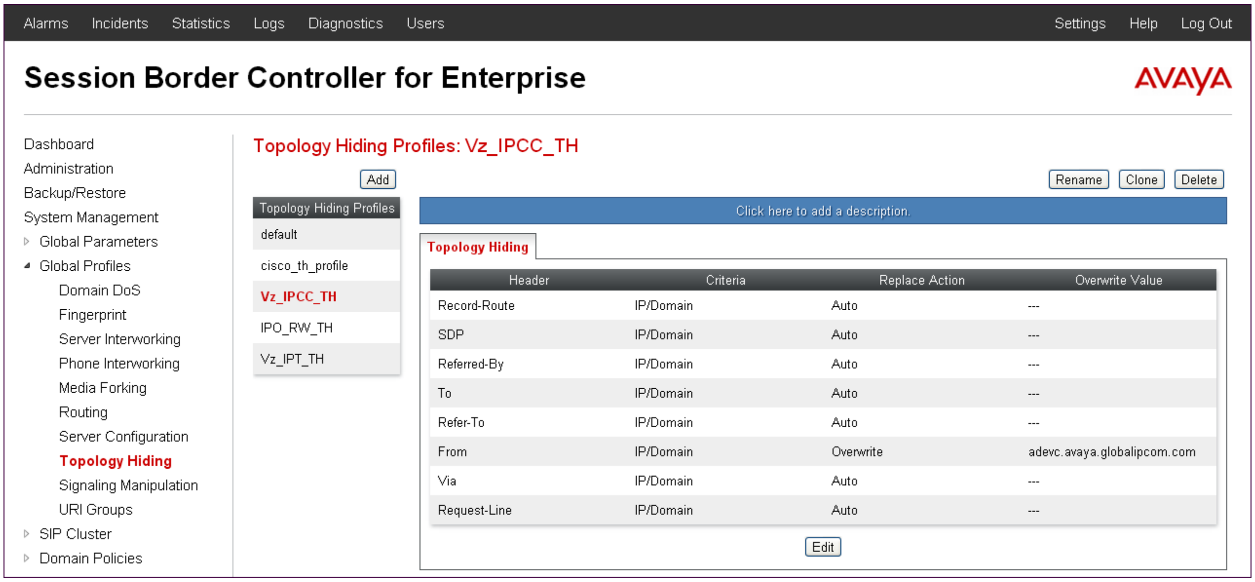
Priority	1
URI Group	*
Next Hop Server 1	172.30.205.55:5072
Next Hop Server 2	---
Next Hop Priority	☑
NAPTR	☐
SRV	☐
Next Hop in Dialog	☐
Ignore Route Header	☐
Outgoing Transport	UDP

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, or the service provider SIP Proxy depending on the header, and the **Overwrite** will use the value in the **Overwrite Value**. In the example shown, "**Vz_IPCC_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 IPCC service.



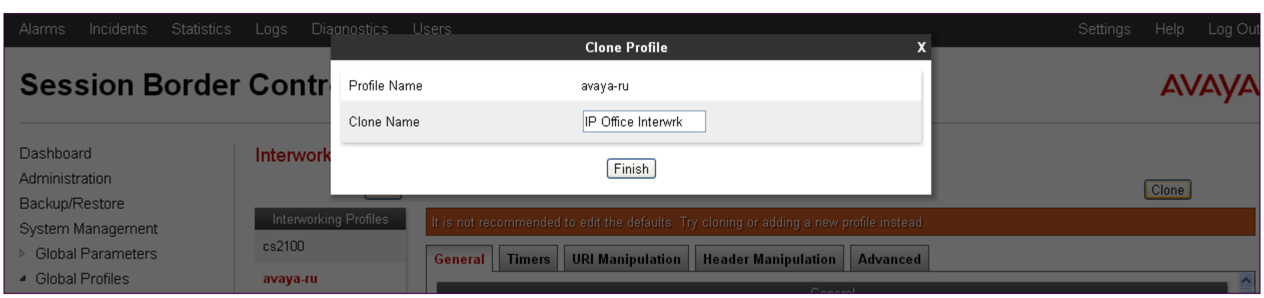
6.4. Server Interworking Profile

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.

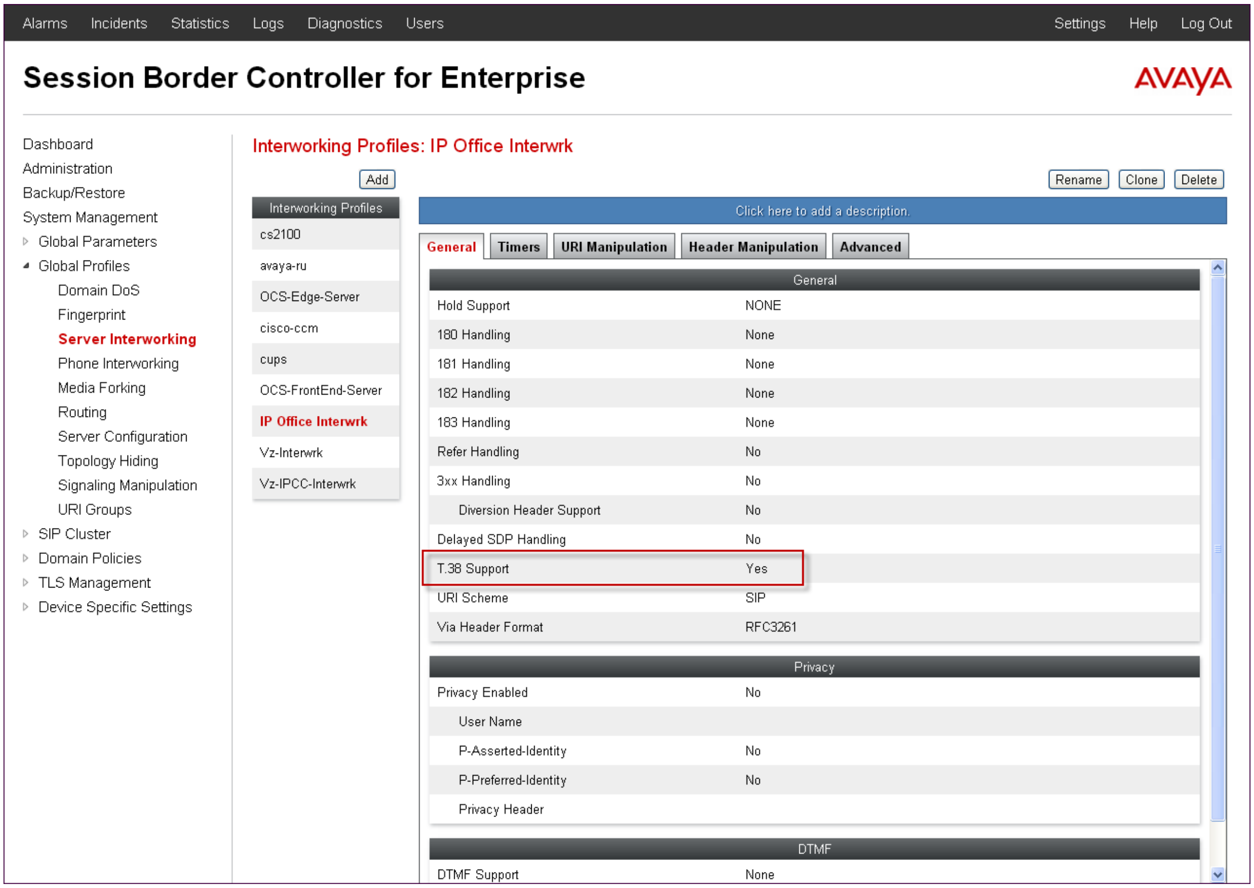
In the sample configuration, separate Server Interworking Profiles were created for IP Office and Verizon Business IPCC service.

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

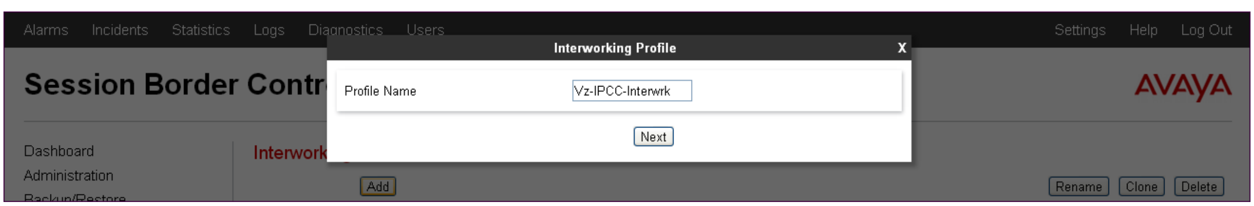


The following screen shows the "**IP Office Interwrk**" profile used in the sample configuration, with **T.38 Support** set to "**Yes**". Default values can be used for all other fields.

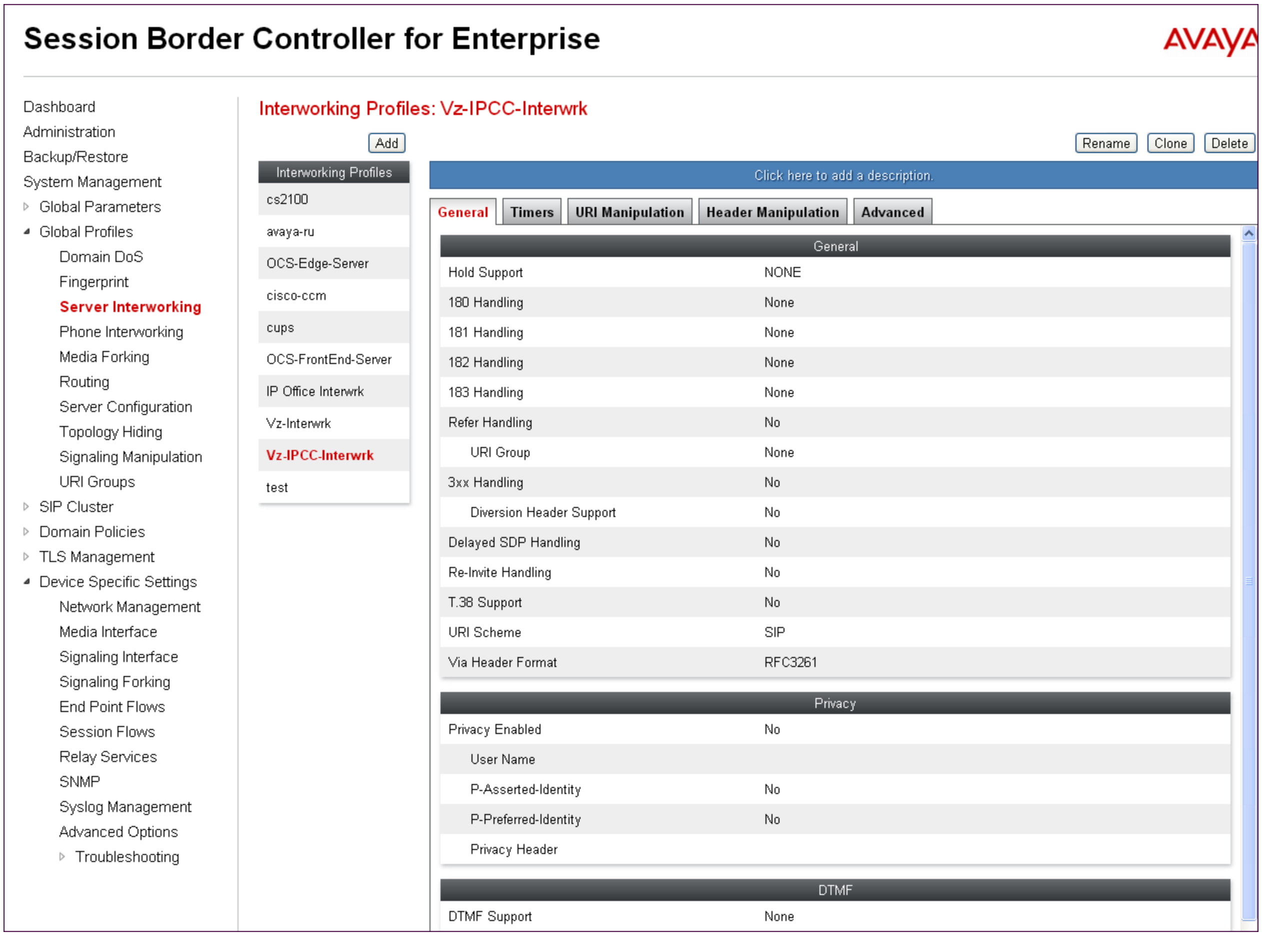


6.4.2. Server Interworking Profile – Verizon

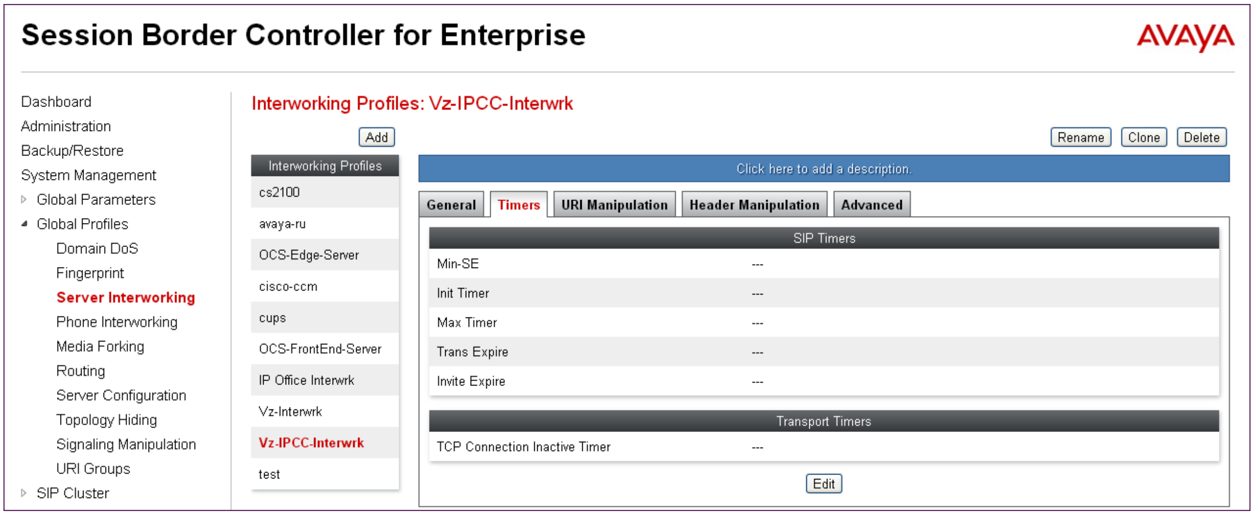
To create a new Server Interworking Profile for Verizon, navigate to **Global Profiles → Server Interworking** and click **Add** as shown below. Enter a **Profile Name** and click **Next**.



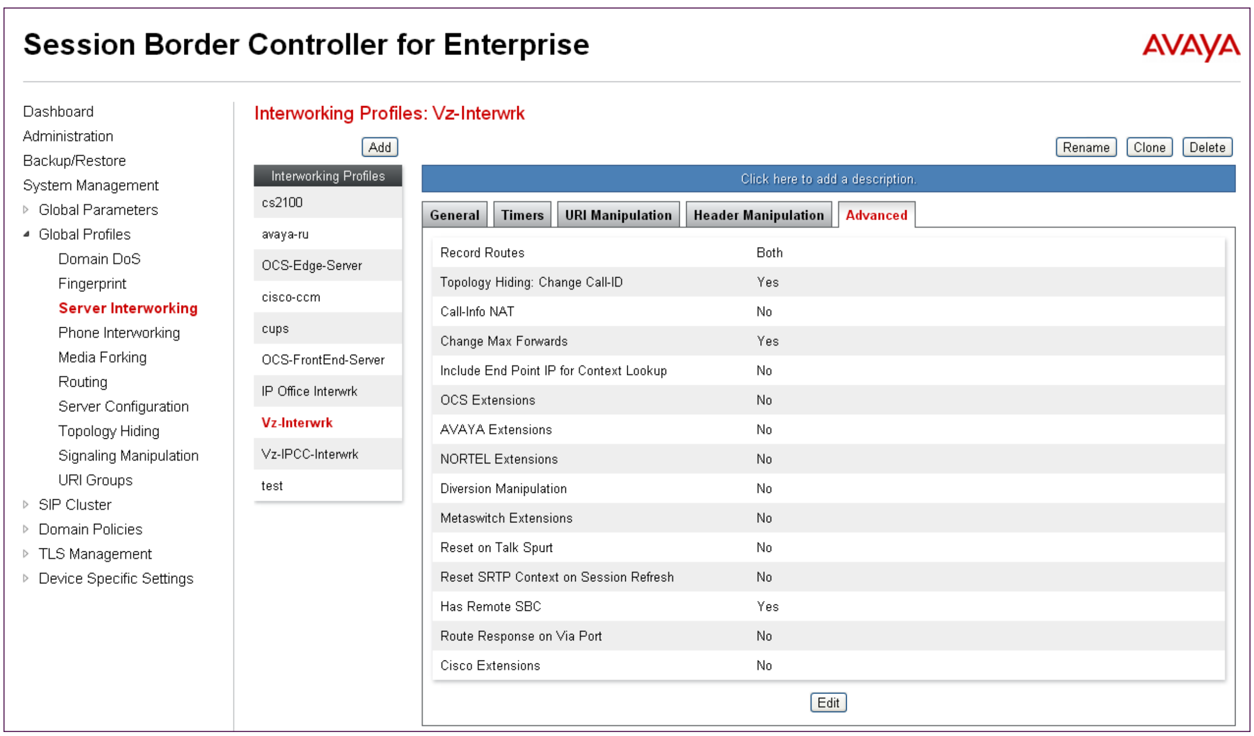
The following screens show the "**Vz-IPCC-Interwrk**" profile used in the sample configuration. On the **General** tab, default values can be used for all fields.


Session Border Controller for Enterprise
AVAYA
Dashboard
Administration
Backup/Restore
System Management
Global Parameters
Global Profiles
Domain DoS
Fingerprint
Server Interworking
Phone Interworking
Media Forking
Routing
Server Configuration
Topology Hiding
Signaling Manipulation
URI Groups
SIP Cluster
Domain Policies
TLS Management
Device Specific Settings
Network Management
Media Interface
Signaling Interface
Signaling Forking
End Point Flows
Session Flows
Relay Services
SNMP
Syslog Management
Advanced Options
Troubleshooting
Interworking Profiles: Vz-IPCC-Interwrk
Add
Rename
Clone
Delete
Interworking Profiles
cs2100
avaya-ru
OCS-Edge-Server
cisco-ccm
cups
OCS-FrontEnd-Server
IP Office Interwrk
Vz-Interwrk
Vz-IPCC-Interwrk
test
Click here to add a description.
General
Timers
URI Manipulation
Header Manipulation
Advanced
General
Hold Support NONE
180 Handling None
181 Handling None
182 Handling None
183 Handling None
Refer Handling No
URI Group None
3xx Handling No
Diversion Header Support No
Delayed SDP Handling No
Re-Invite Handling No
T.38 Support No
URI Scheme SIP
Via Header Format RFC3261
Privacy
Privacy Enabled No
User Name
P-Asserted-Identity No
P-Preferred-Identity No
Privacy Header
DTMF
DTMF Support None

Timers tab:



Advanced tab:



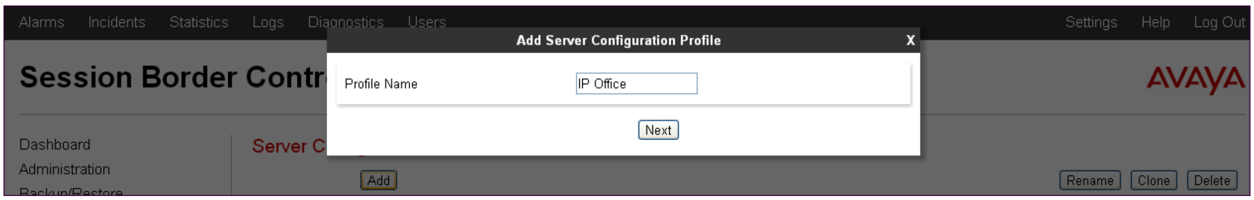
6.5. Server Configuration

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.

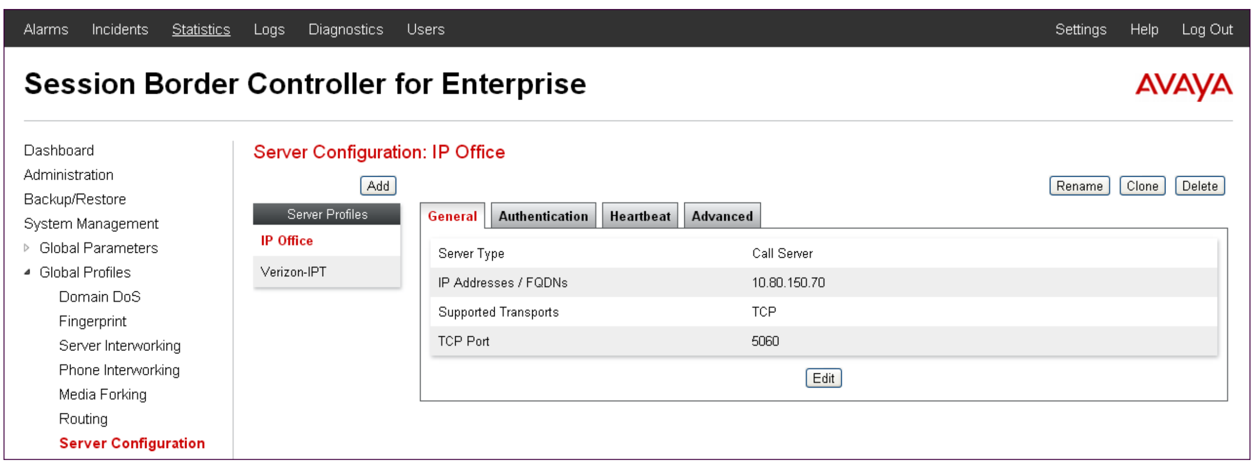
In the sample configuration, separate Server Configurations were created for IP Office and Verizon Business IPCC service.

6.5.1. Server Configuration – IP Office

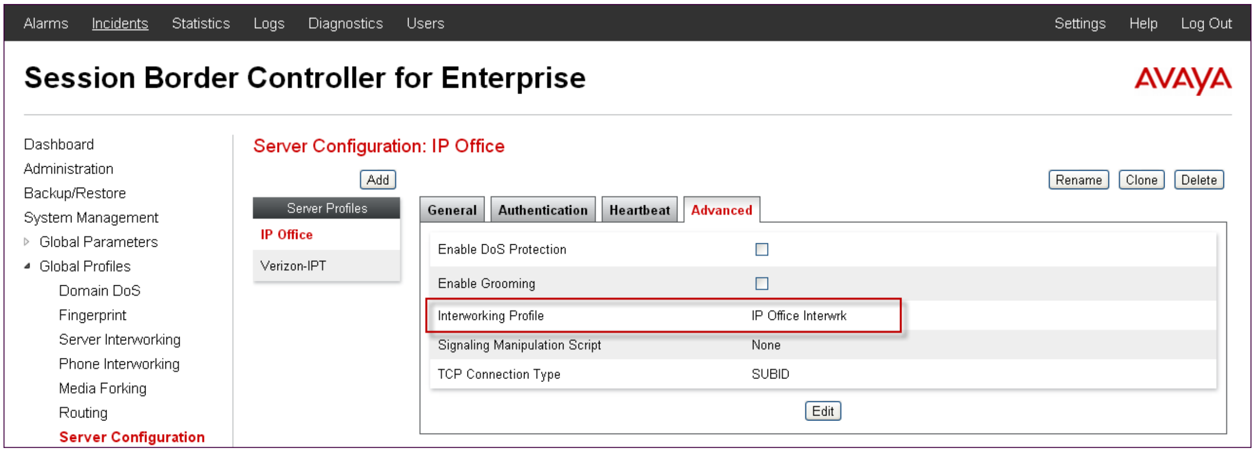
To add a Server Configuration Profile for IP Office, navigate to **Global Profiles → Server Configuration** and click **Add**. Enter a descriptive name for the new profile and click **Next**.



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**".

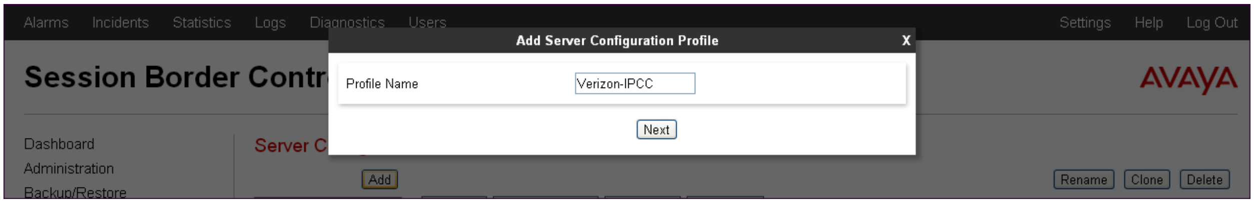


Default values can be used on the **Authentication** and **Heartbeat** tabs (not shown). On the **Advanced** tab, the **Interworking Profile** is set to "**IP Office Interwrk**" created in **Section 6.4.1** for IP Office. If remote workers will not be used, **Enable Grooming** can be selected to allow 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.

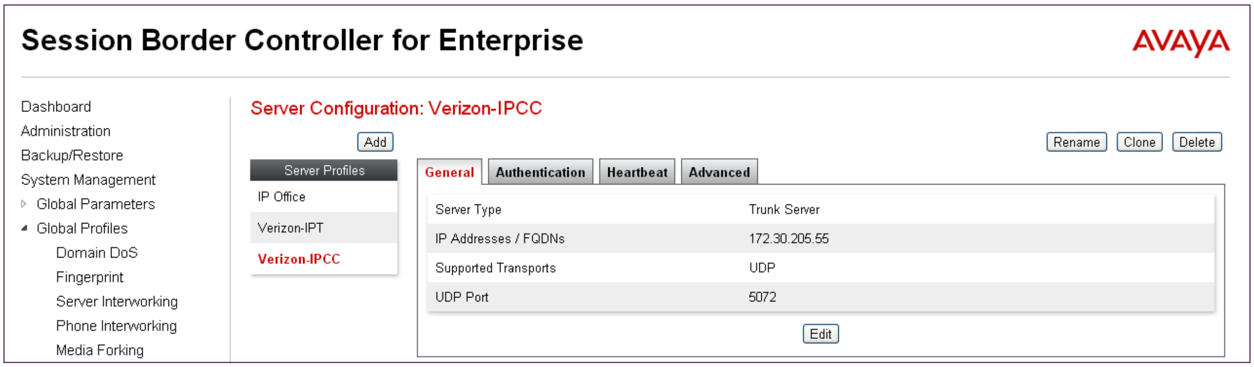


6.5.2. Server Configuration - Verizon

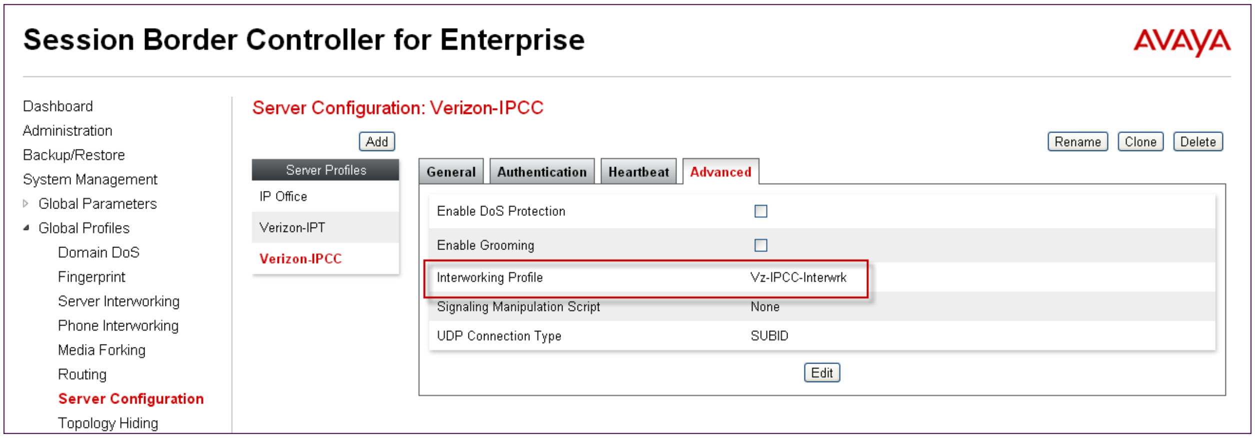
To add a Server Configuration Profile for Verizon, navigate to **Global Profiles → Server Configuration** and click **Add**. Enter a descriptive name for the new profile and click **Next**.



The following screens illustrate the Server Configuration for the Profile name "**Verizon-IPCC**". In the **General** parameters, the **Server Type** is set to "**Trunk Server**". In the **IP Addresses / Supported FQDNs** area, the Verizon-provided IP address is entered. In the sample configuration this is "**172.30.205.55**". In the **Supported Transports** area, **UDP** is selected, and the **UDP Port** is set to "**5072**".



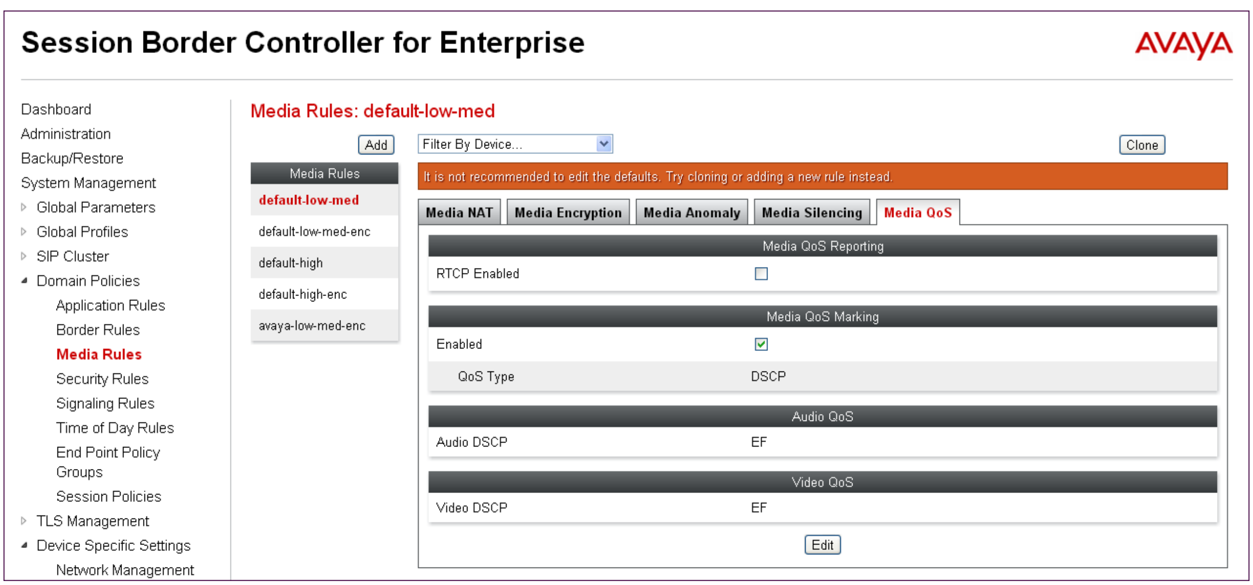
Default values can be used on the **Authentication** and **Heartbeat** tabs (not shown). On the **Advanced** tab, **Enable Grooming** is not used for UDP connections and left unchecked. The **Interworking Profile** is set to “**Vz-IPCC-Interwrk**” created in **Section 6.4.2** for Verizon.



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.

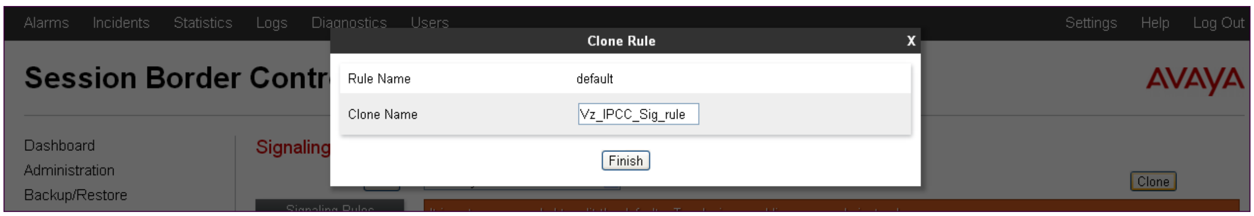
Select **Domain Policies → Media Rules** from the left-side menu as shown below. In the sample configuration, a single default media rule “**default-low-med**” was used with the DSCP values “**EF**” for expedited forwarding (default value) for **Media QoS** as shown below.



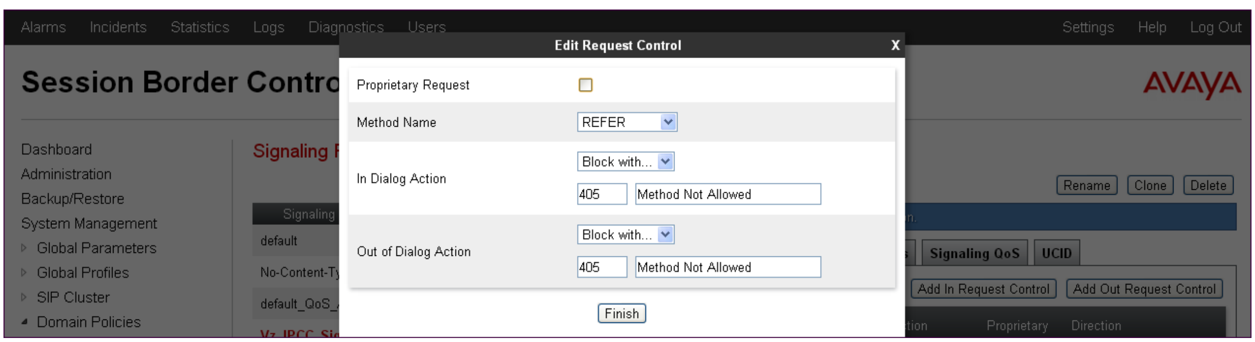
6.7. Signaling Rule

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.

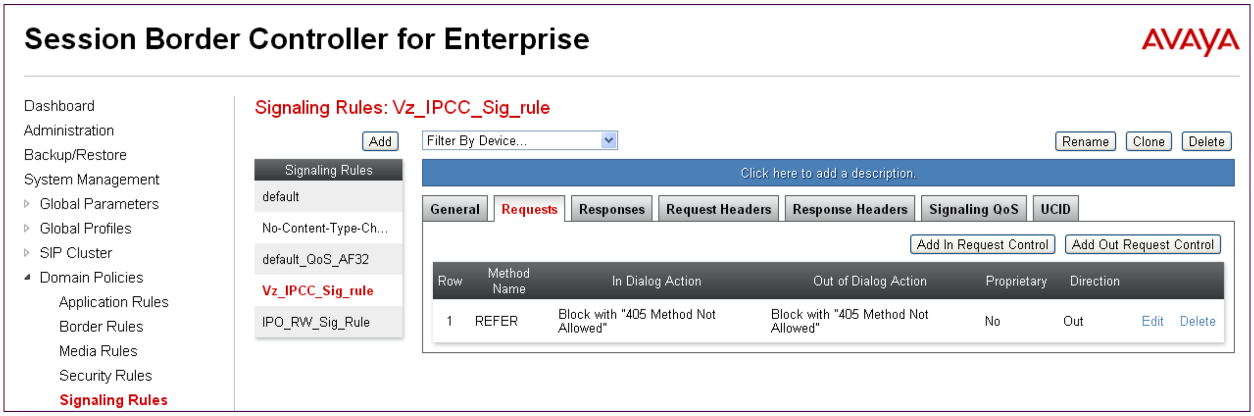
Clone and modify the default signaling rule to block REFER messages from IP Office to Verizon IPCC and to add the proper quality of service to the SIP signaling. To clone a signaling rule, navigate to **Domain Policies → Signaling Rules**. With the **default** rule chosen, click **Clone** (not shown).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



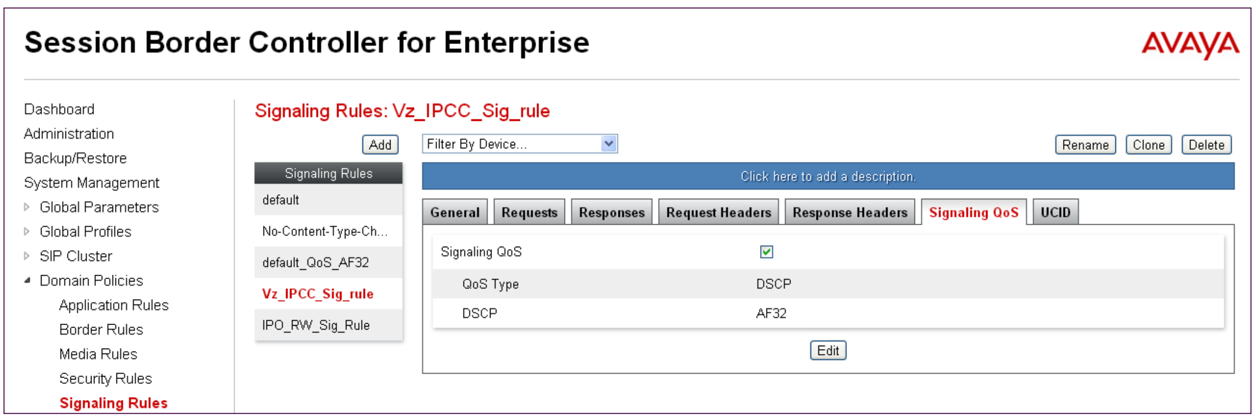
Select the **Requests** tab, and select the **Add Out Request Control** button (not shown). Select "**REFER**" as the **Method Name**. In the **In Dialog Action** and **Out of Dialog Action**, select "**Blocks with…**" and type "**405**" and "**Method Not Allowed**" as shown below. The intent is to have the SBC return a "405 Method Not Allowed" response whenever a REFER is sent to Verizon IPCC from IP Office. See **Section 2.2** for additional information.



Once complete, the Request Headers tab appears as follows.



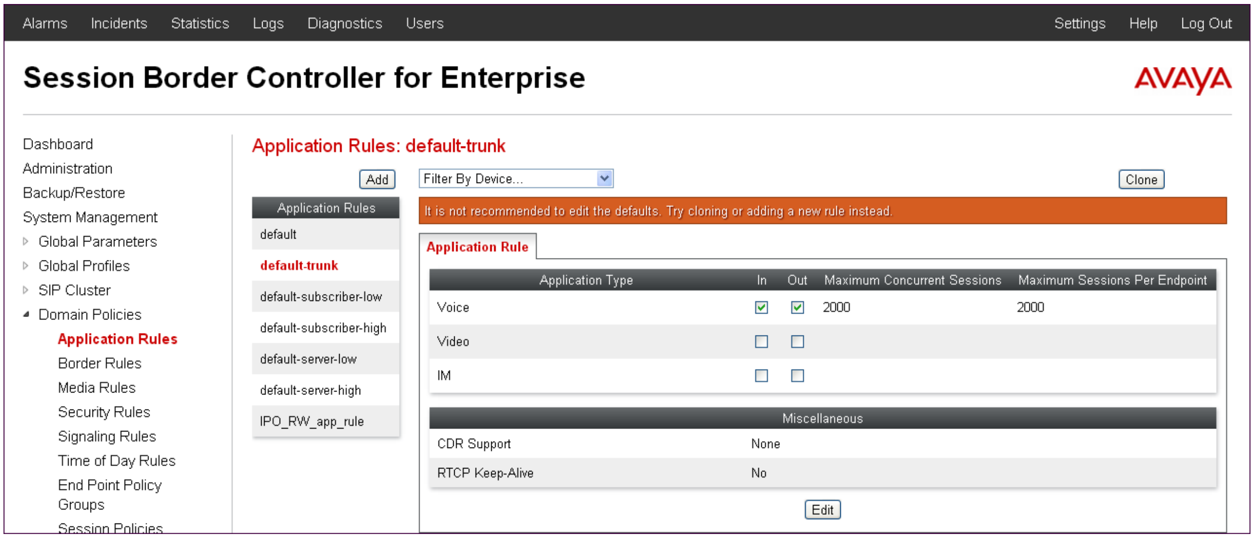
The following screen shows the **Signaling QoS** set with the DSCP value “**AF32**” for assured forwarding.



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.

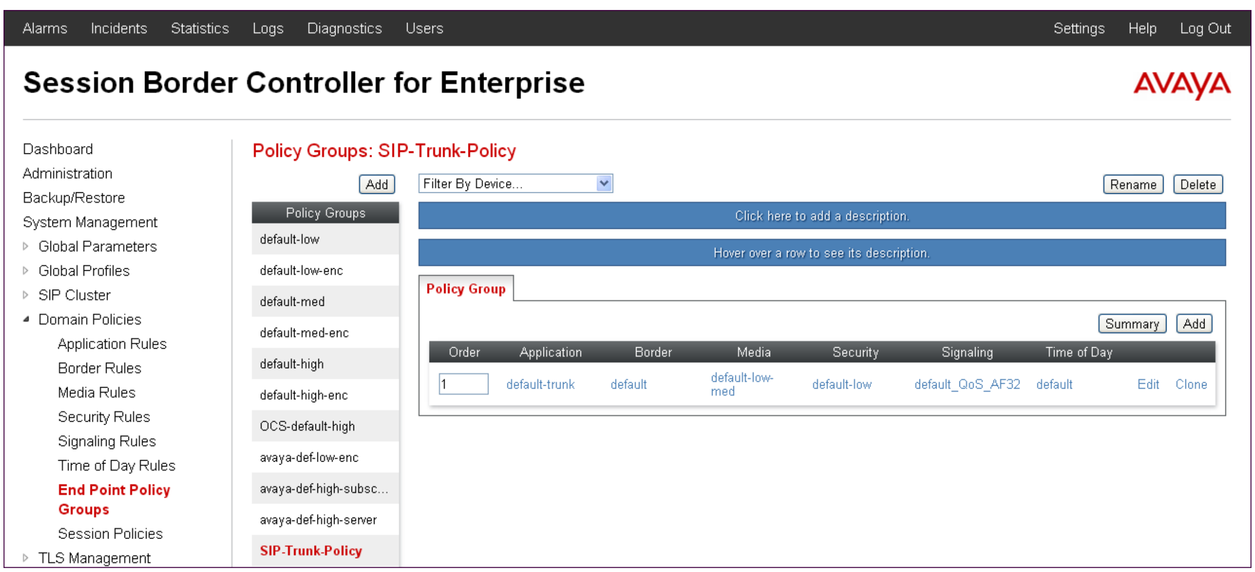
Select **Domain Policies → Application Rules** from the left-side menu as shown below. In the sample configuration, a single default application rule “**default-trunk**” is used and will be applied to the Endpoint Policy Group in the next section.



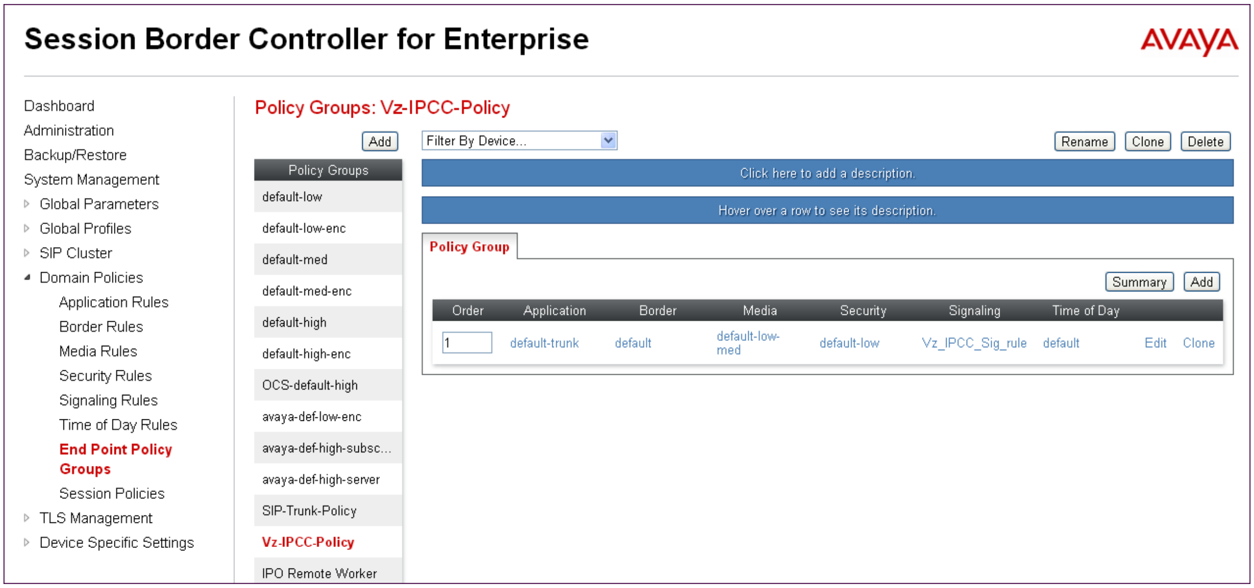
6.9. Endpoint Policy Groups

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.



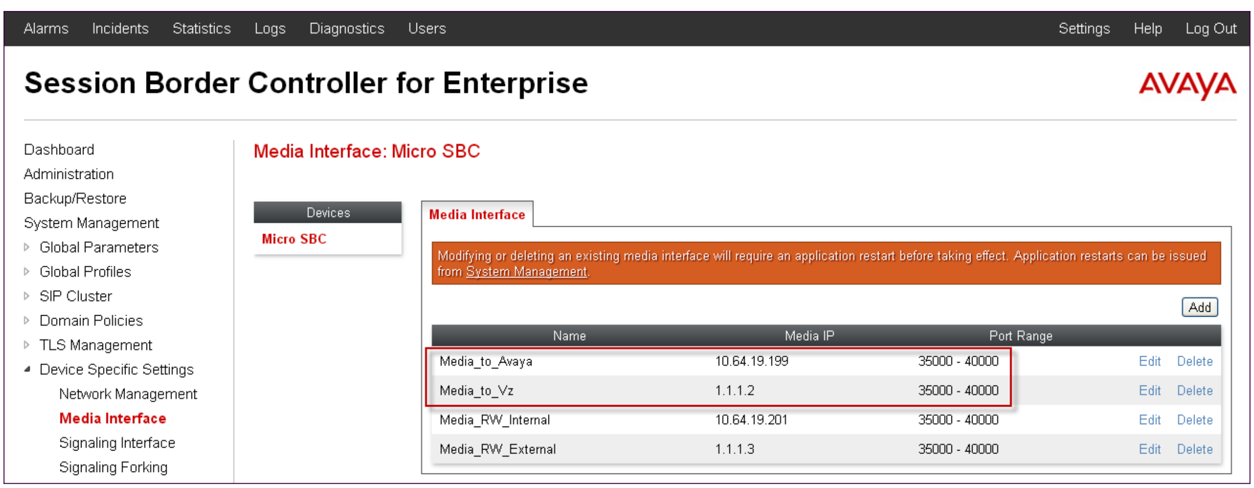
Similarly, a separate profile named "**Vz-IPCC-Policy**" was created for Verizon Business IP Contact Center SIP Trunk service using defaults selected for all fields, with the exception of **Application** set to "**default-trunk**" and **Signaling** set to "**Vz_IPCC_Sig_rule**" as shown below.



6.10. Media Interface

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

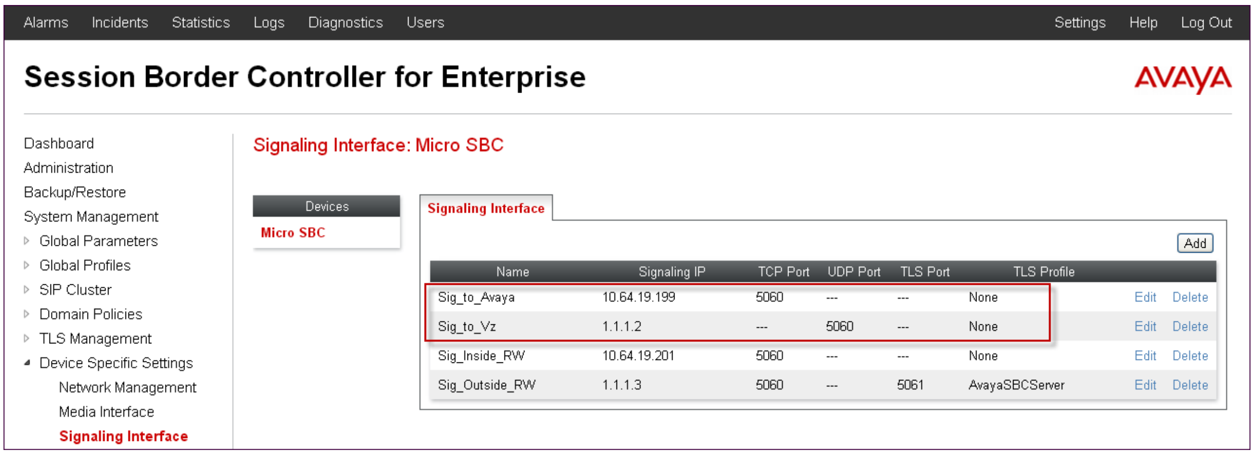
To create a new Media Interface, navigate to **Device Specific Settings → Media Interface** and click **Add**. The following screen shows the media interfaces defined for the sample configuration.



6.11. Signaling Interface

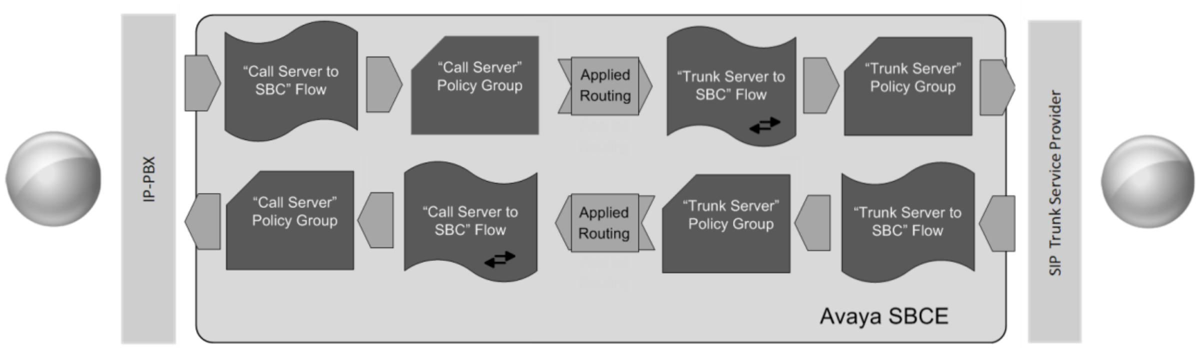
The Signaling Interface screen is where the SIP signaling ports are defined. Avaya SBCE will listen for SIP requests on the defined ports. Create a signaling interface for the inside and outside IP interfaces.

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.

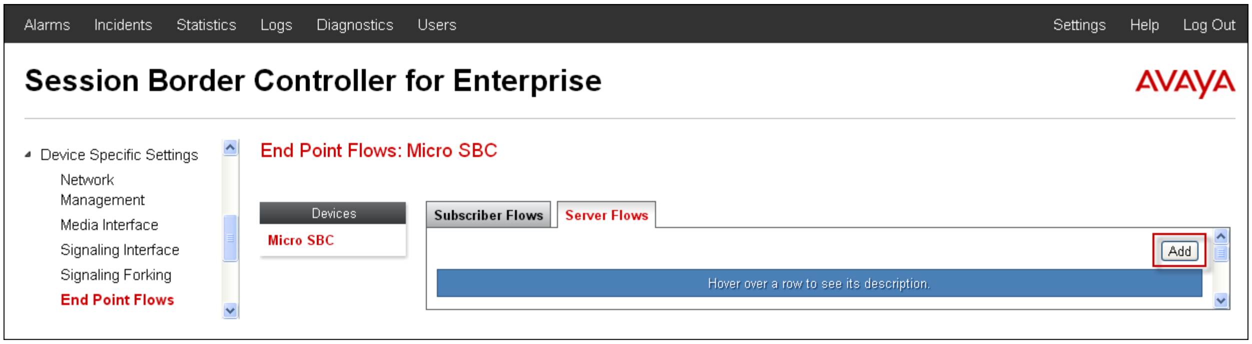


6.12. End Point Flows - Server Flow

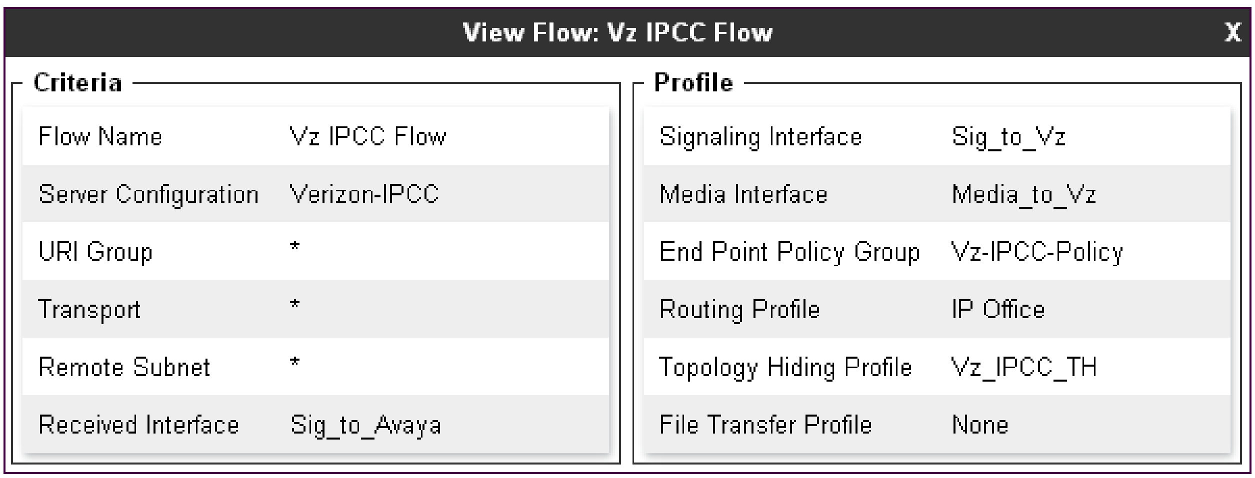
When a packet is received by Avaya SBCE, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to a policy which contains several rules concerning processing, privileges, authentication, routing, etc. Once routing 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.



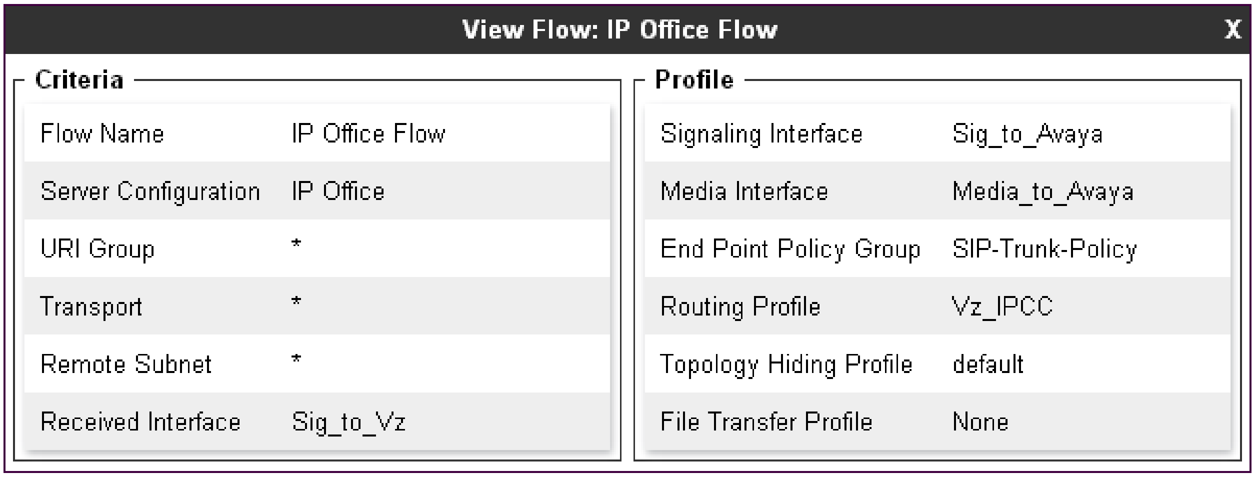
Create a Server Flow for IP Office and Verizon Business IP Trunk service. To create a Server Flow, navigate to **Device Specific Settings → End Point Flows**. Select the **Server Flows** tab and click **Add** as highlighted below.



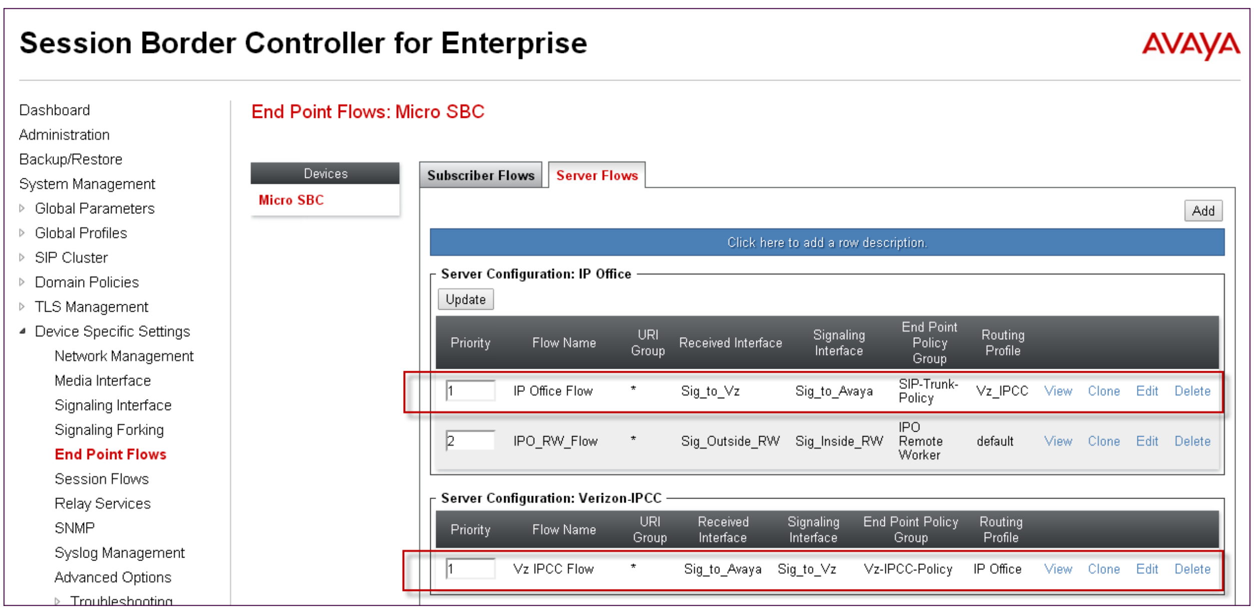
The following screen show the flow named "**Verizon IPCC Flow**" viewed from the sample configuration. This flow uses the interfaces, polices, and profiles defined in previous sections.



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



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



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.

8. Verifications

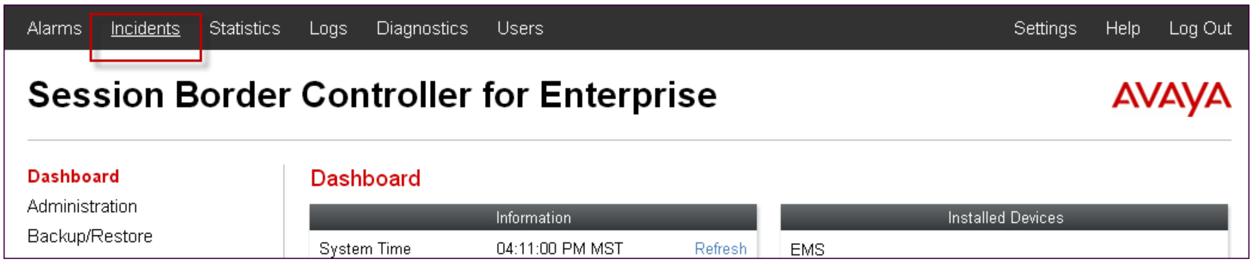
This section provides example verifications of the Avaya configuration with Verizon Business Private IP (PIP) Trunk service.

8.1. Avaya SBCE

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

8.1.1. Incidents

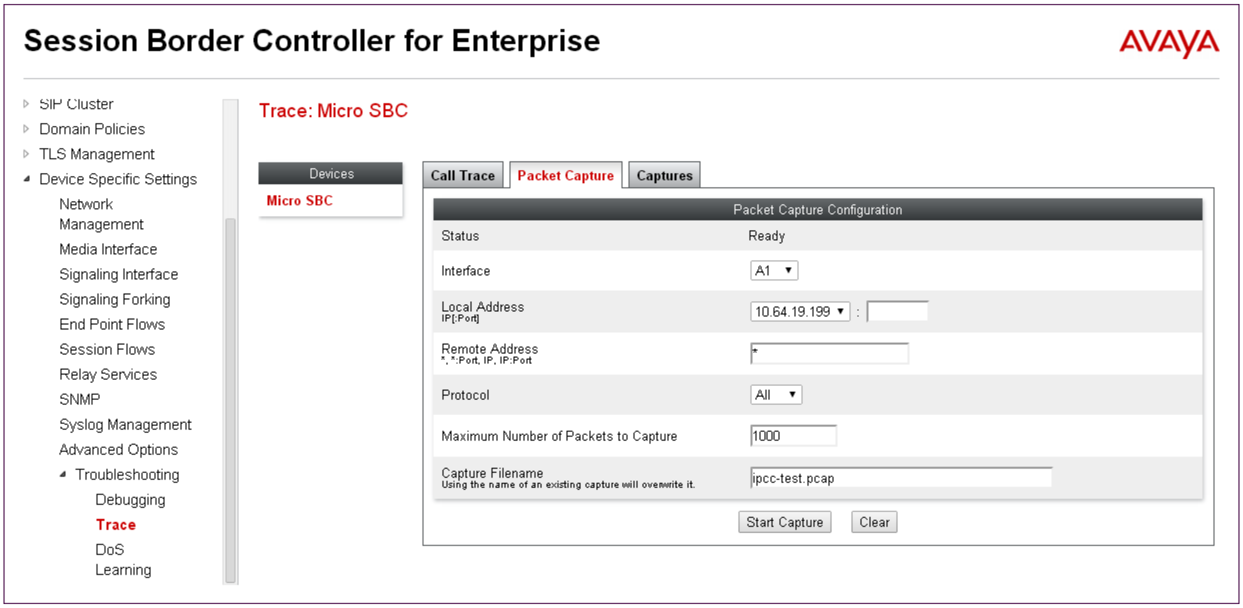
The Incident Viewer can be accessed from the Avaya SBCE Dashboard as highlighted in the screen shot below.



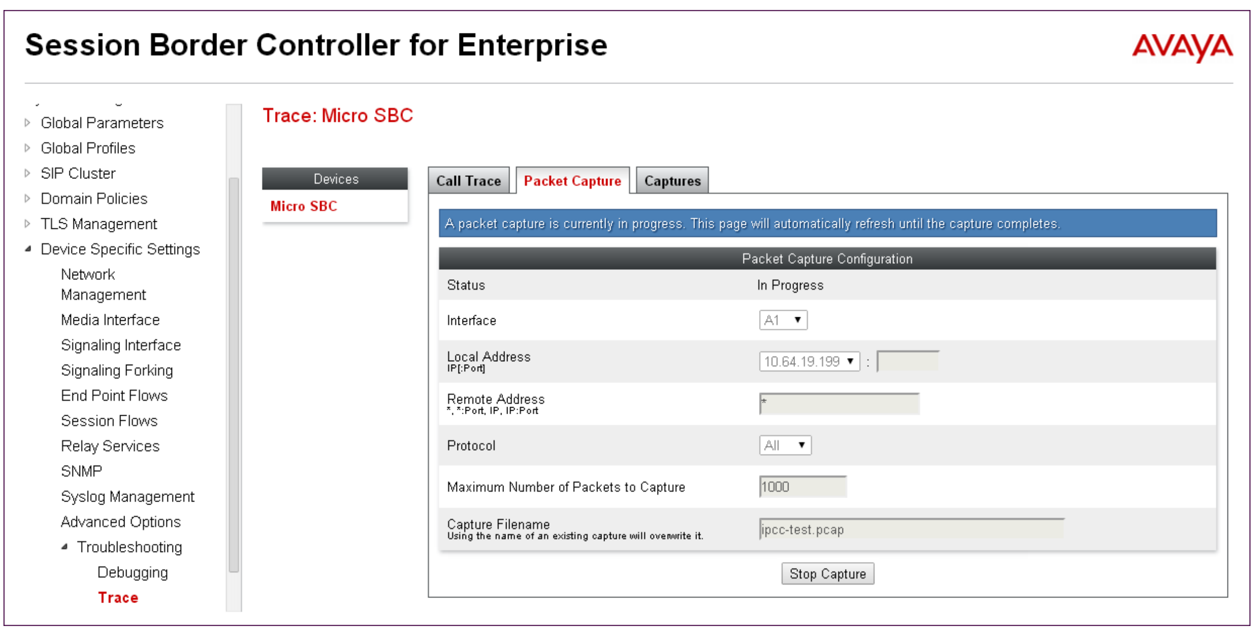
Use the Incident Viewer to verify Server Heartbeat and to troubleshoot routing failures.

8.1.2. Tracing

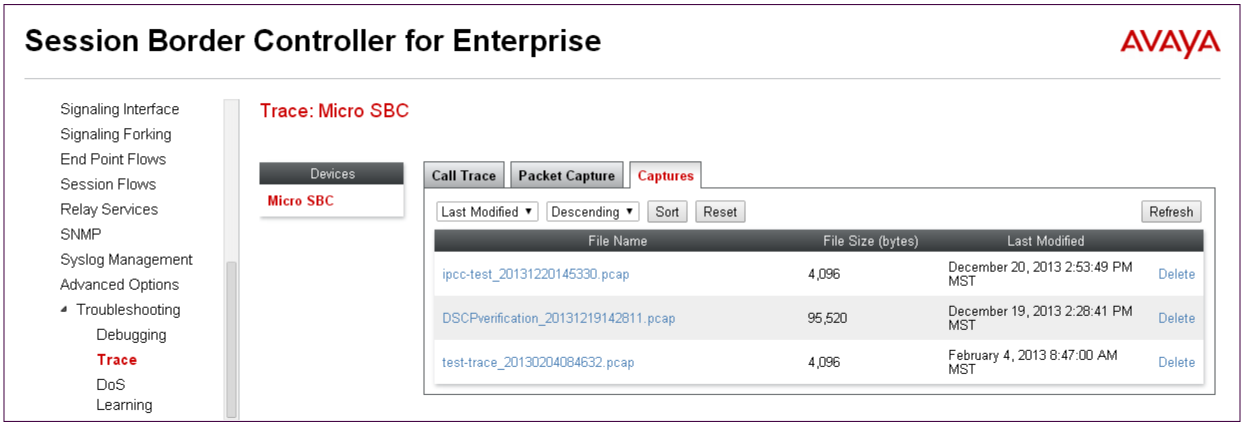
To take a call trace, navigate to **Device Specific Settings → Trace** and select the **Packet Capture** tab. Populate the fields for the capture parameters and click **Start Capture** as shown below.



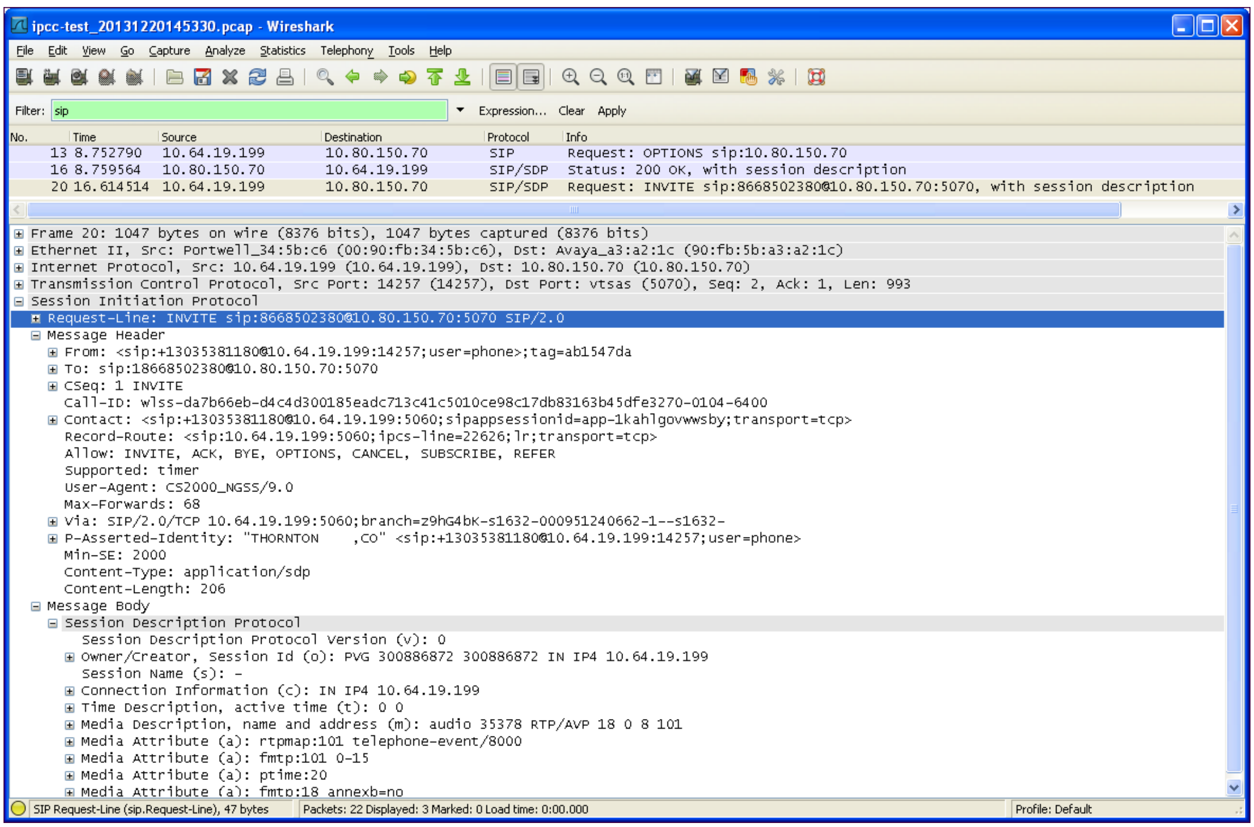
When tracing has reached the desired number of packets the trace will stop automatically, or alternatively, hit the **Stop Capture** button at the bottom.



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



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



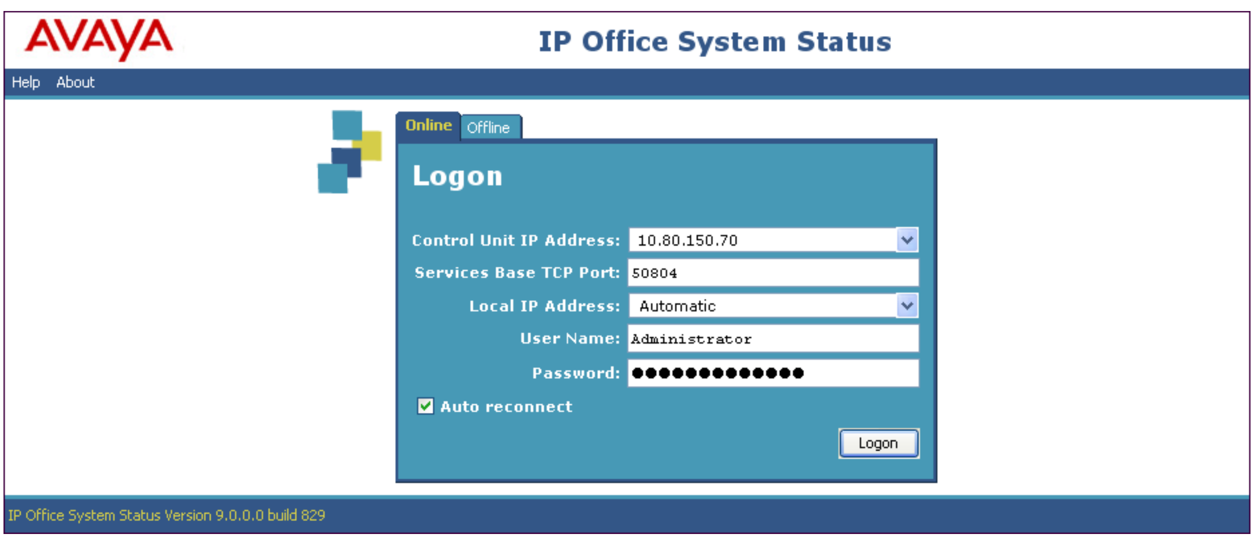
8.2. IP Office

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

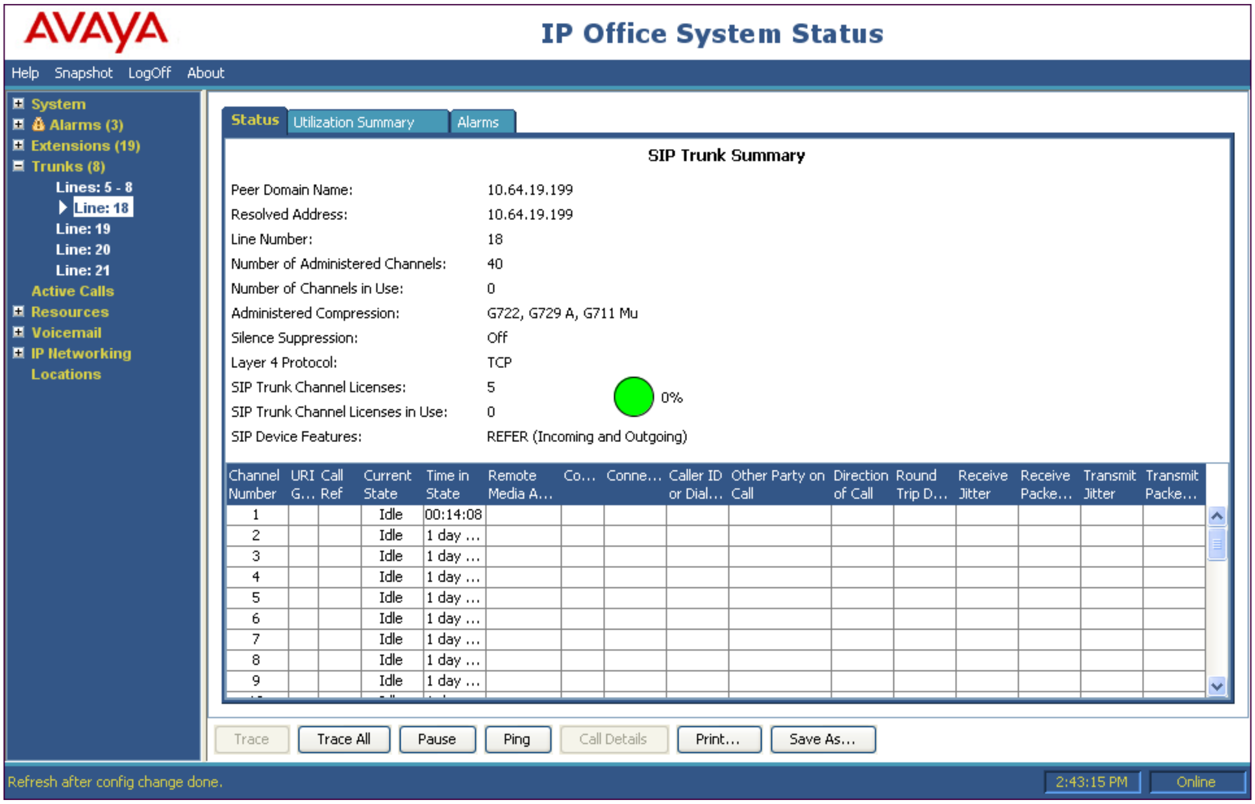
8.2.1. System Status

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.

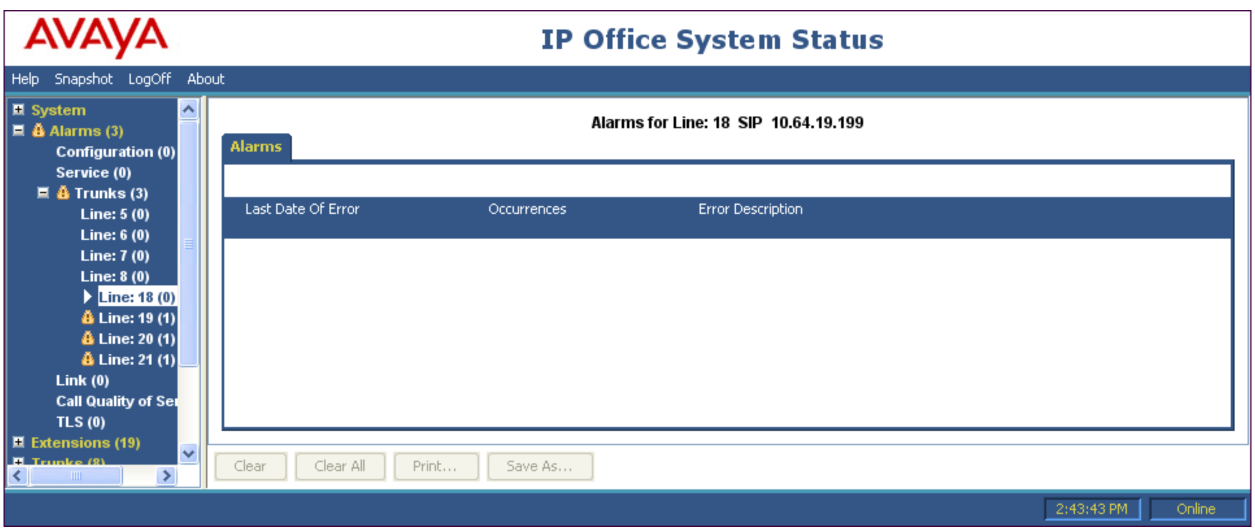
The following screen shows an example **Logon** screen. Enter the IP Office IP address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.



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



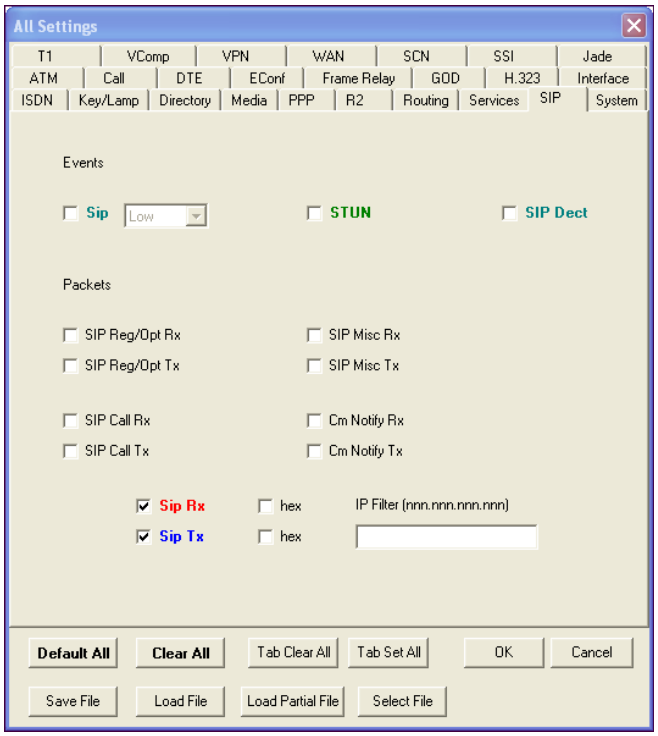
Select the **Alarms** tab and verify that no alarms are active on the SIP line.



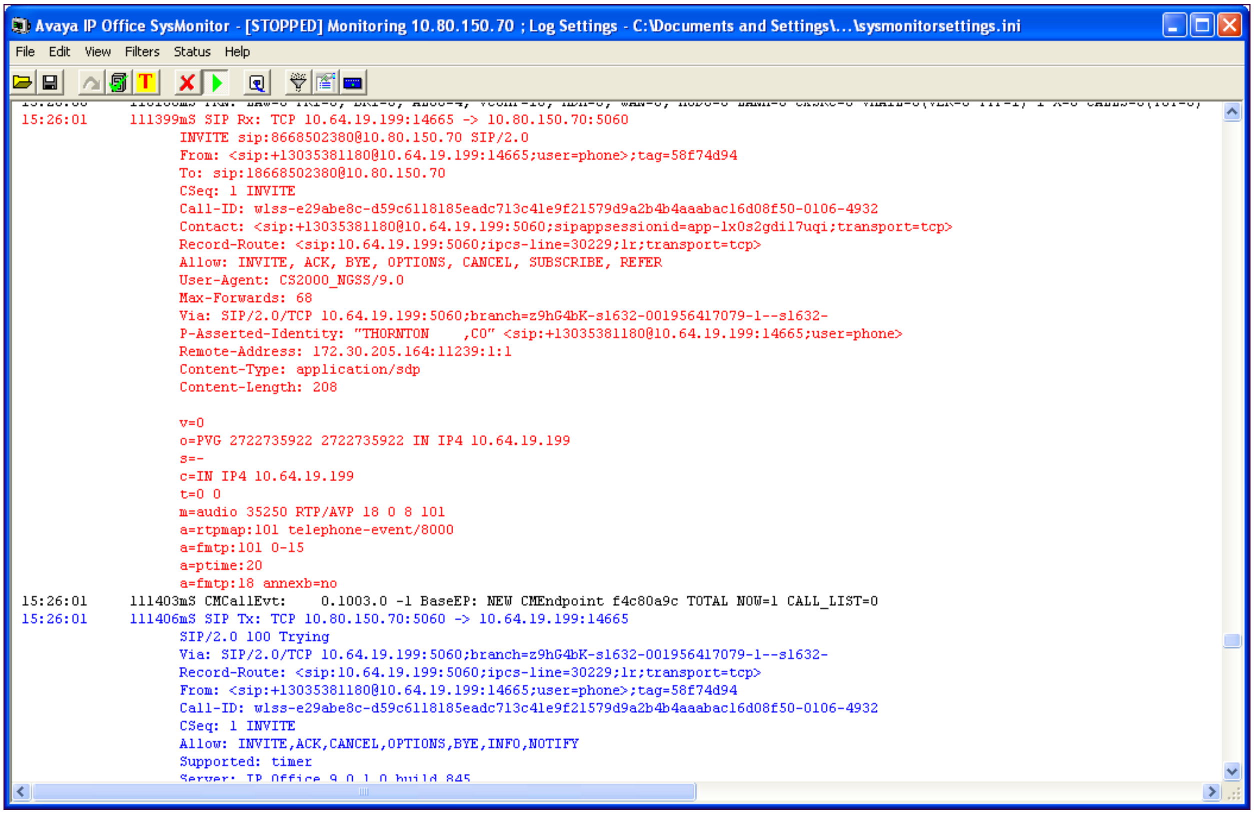
8.2.2. Monitor

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 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. SIP messages will appear in the trace with the color red for Rx and blue for Tx. To customize the color, right-click on **SIP Rx** or **SIP Tx** and select the desired color.



As an example, the following shows a portion of the monitoring window for an inbound call to Verizon IP Toll Free number 1-866-850-2380. Details of the SIP INVITE message sent by Verizon are shown below. This information matches the configuration in these Application Notes and is not intended to be prescriptive. The intent is to illustrate the INVITE sent by Verizon in the sample configuration, along with the means to retrieve this type of trace information from IP Office.



```
Avaya IP Office SysMonitor - [STOPPED] Monitoring 10.80.150.70 ; Log Settings - C:\Documents and Settings\...\sysmonitorsettings.ini
File  Edit  View  Filters  Status  Help

15:26:01     111399mS SIP Rx: TCP 10.64.19.199:14665 -> 10.80.150.70:5060
                    INVITE sip:8668502380@10.80.150.70 SIP/2.0
                    From: <sip:+13035381180@10.64.19.199:14665;user=phone>;tag=58f74d94
                    To: sip:18668502380@10.80.150.70
                    CSeq: 1 INVITE
                    Call-ID: wlss-e29abe8c-d59c6118185eadc713c41e9f21579d9a2b4b4aaabac16d08f50-0106-4932
                    Contact: <sip:+13035381180@10.64.19.199:5060;sipappsessionid=app-1x0s2gdil7uqi;transport=tcp>
                    Record-Route: <sip:10.64.19.199:5060;ipcs-line=30229;lr;transport=tcp>
                    Allow: INVITE, ACK, BYE, OPTIONS, CANCEL, SUBSCRIBE, REFER
                    User-Agent: CS2000_NGSS/9.0
                    Max-Forwards: 68
                    Via: SIP/2.0/TCP 10.64.19.199:5060;branch=z9hG4bK-s1632-001956417079-1--s1632-
                    P-Asserted-Identity: "THORNTON    ,CO" <sip:+13035381180@10.64.19.199:14665;user=phone>
                    Remote-Address: 172.30.205.164:11239:1:1
                    Content-Type: application/sdp
                    Content-Length: 208

                    v=0
                    o=PVG 2722735922 2722735922 IN IP4 10.64.19.199
                    s=-
                    c=IN IP4 10.64.19.199
                    t=0 0
                    m=audio 35250 RTP/AVP 18 0 8 101
                    a=rtpmap:101 telephone-event/8000
                    a=fmtp:101 0-15
                    a=ptime:20
                    a=fmtp:18 annexb=no
15:26:01     111403mS CMCallEvt:    0.1003.0 -1 BaseEP: NEW CMEndpoint f4c80a9c TOTAL NOW=1 CALL_LIST=0
15:26:01     111406mS SIP Tx: TCP 10.80.150.70:5060 -> 10.64.19.199:14665
                    SIP/2.0 100 Trying
                    Via: SIP/2.0/TCP 10.64.19.199:5060;branch=z9hG4bK-s1632-001956417079-1--s1632-
                    Record-Route: <sip:10.64.19.199:5060;ipcs-line=30229;lr;transport=tcp>
                    From: <sip:+13035381180@10.64.19.199:14665;user=phone>;tag=58f74d94
                    Call-ID: wlss-e29abe8c-d59c6118185eadc713c41e9f21579d9a2b4b4aaabac16d08f50-0106-4932
                    CSeq: 1 INVITE
                    Allow: INVITE,ACK,CANCEL,OPTIONS,BYE,INFO,NOTIFY
                    Supported: timer
                    Server: IP Office 9.0.1.0 build 845
```

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.

IP Office Release 9.0 with Avaya Session Border Controller for Enterprise Release 6.2 has not been independently certified by Verizon labs. These Application Notes can be used to facilitate customer engagements via the Verizon field trial process, pending Verizon labs independent certification.

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 for Enterprise*, January 2014

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

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 8.1 with Avaya Session Border Controller for Enterprise Release 6.2 with Verizon IPCC Service Suite.
[VZBIPCCIPO81SBC] Application Notes for Configuring SIP Trunking using Verizon Business IP Contact Center VoIP Inbound and Avaya IP Office Release 8 with Avaya Session Border Controller for Enterprise, Issue 1.0

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 9.0 and Avaya Session Border Controller for Enterprise with Verizon IP Trunk Service Suite.
[VZBIPT-IPO9SBC] 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 1.0

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

[RFC-2833] RFC 2833 *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals* http://www.ietf.org/rfc/rfc2833.txt

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

- [VZ-Test-Plan] Test Suite for CPE IP Trunking Interoperability v1.3
- [VZ-Spec] Verizon Business IPCC Trunk Interface Network Interface Specification, Document Version 2.2.1.9

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