Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking Using Verizon Business IP Contact Center VoIP Inbound and Avaya IP Office Release 10.0 with Avaya Session Border Controller for Enterprise Release 7.1 – 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 7.1, Avaya IP Office Server Edition Release 10.0, an IP500 V2 Expansion System Release 10.0, 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 10.0 and Avaya Session Border Controller for Enterprise Release 7.1.

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.

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

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

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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 (Verizon Business IPCC) 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 7.1, Avaya IP Office Server Edition Release 10.0, an IP500 V2 Expansion System Release 10.0, 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 10.0 and Avaya Session Border Controller for Enterprise Release 7.1.

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.

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2.1. Interoperability Compliance Testing

The testing included executing the test cases detailed in **Section 10**, Reference [VZ-Test-Plan], which contains the Verizon Business 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 Communicator, Avaya Communicator, 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 directed to a hunt group whose queue is "full" (i.e. if no redirection is configured for hunt group busy conditions, see **Section 5.6.2**).
- 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.

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- Incoming toll-free calls directed to the Hunt Groups configured in **Section 5.6.2** were verified. Incoming calls could be queued, queued with priority, and be answered by members of the hunt group as members become available.
- Outgoing calls from the Avaya IP Office location to the PSTN were routed via a SIP Line to the Verizon Business IP Trunk service described in reference [VZBIPT-IPO10SBC]. As detailed in reference [VZBIPT-IPO10SBC], these outgoing PSTN calls can be originated from Avaya H.323 telephones, Avaya SIP telephones, Avaya digital telephones, and analog endpoints. The display of caller ID on display-equipped PSTN telephones was verified. In the context of inbound toll-free calls using Verizon Business IPCC, inbound toll-free calls arriving via the SIP Line configured in **Section 5.4** could be forwarded or twinned out the Verizon Business IP Trunk service SIP Line. Inbound toll-free calls from the Verizon Business IPCC SIP Line could also trigger mobile callback calls that use the Verizon Business IP Trunk service SIP Line.
- Call Forwarding of Verizon toll-free calls to PSTN destinations via the Verizon Business IP Trunk service documented in reference [VZBIPT-IPO10SBC], presenting true calling party information to the mobile phone. See Section 2.2 for additional information.
- Mobile twinning of Verizon toll-free calls to a mobile phone via the Verizon Business IP Trunk service documented in reference [VZBIPT-IPO10SBC], presenting true calling party information to the mobile phone.
- Proper DiffServ markings for Avaya SBCE SIP signaling and RTP media.
- Avaya Remote Worker configuration via the Avaya SBCE.

2.2. Test Results

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

- In the reference configuration, transferring an inbound Verizon Business IPCC toll-free call back out the Verizon Business IPCC service using the SIP REFER method was limited to SIP endpoints using blind transfer, and Voicemail Pro modules. It is not supported by H.323, digital, or analog endpoints.
- When transferring an inbound Verizon Business IPCC toll-free call back out the Verizon Business IPCC service using the SIP REFER method, the original caller may hear camp-on tones in the event the call is not answered within approximately 30 seconds after the transfer completes. Once the call is answered, the two callers are connected together successfully with two-way audio. This anomaly appears to be caused by Verizon sending a SIP NOTIFY message with “100 Trying” after the SIP REFER, and not following up with a NOTIFY with “180 Ringing” or “183 Session Progress”. Without receiving an 18x message in the NOTIFY, IP Office does not consider the appropriate call timers to be satisfied and attempts to complete the transfer using SIP INVITE. This method is not allowed with the Verizon Business IPCC service, thus causing camp-on tones to occur. A Signaling Manipulation script was created on the Avaya SBCE to change the “100 Trying” within the SIP NOTIFY message to “180 Ringing” towards IP Office (**Section 0**). This satisfies the call timers, and allows the transfer to continue longer than 30 seconds without camp-on tones being heard.

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2.3. Support

2.3.1. Avaya

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

2.3.2. Verizon

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

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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*.

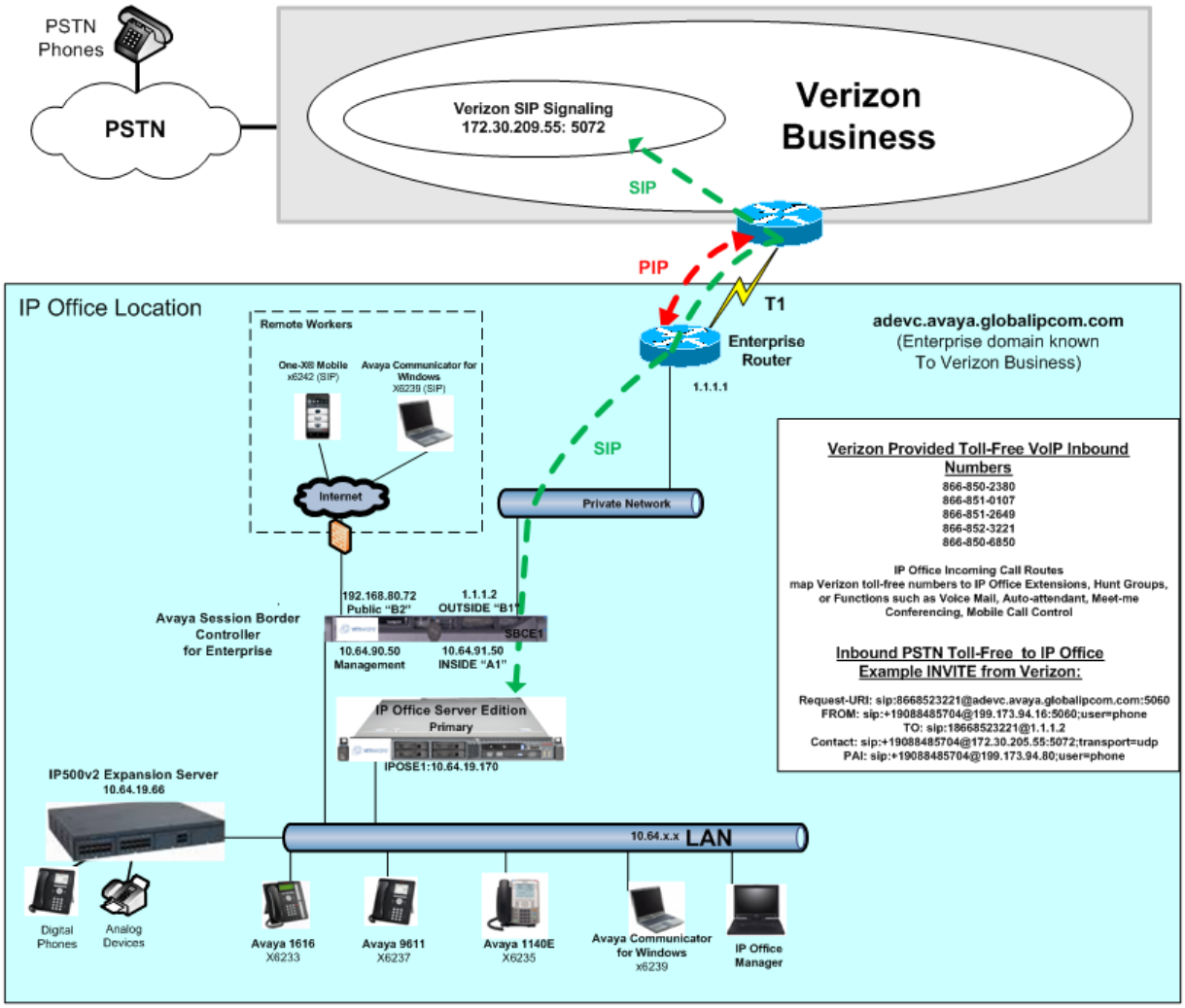

Figure 1: Avaya Interoperability Test Lab Configuration

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The Verizon Business IP Trunk service used for outgoing calls, as described in **Section 2.1**, required different SIP line configuration parameters than what were needed for the Verizon Business IPCC service. A new SIP line was created in IP Office towards Avaya SBCE to support the Verizon Business IPCC service. This SIP line is separate from the SIP line previously created towards Avaya SBCE for Verizon Business IP Trunk service as described in reference [VZBIPT-IPO10SBC]. Having separate SIP lines for each Verizon service will allow for unique parameters to be set on IP Office to accommodate the differences between the two services. In order to support two SIP lines going towards the same Avaya SBCE, the Avaya SBCE was configured with two internal IP addresses designated for SIP trunk traffic.

Additionally, the reference configuration included remote worker functionality. 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 Communicator for Windows (using TLS and SRTP)
- Avaya one-X® Mobile Preferred for IP Office on Android (using TLS and SRTP)

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 references [7] and [8].

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	X6235, x6239	Avaya 1140E, Avaya one-X® Mobile
866-850-2380	X6232, x6241, x6234	Digital Telephone with Mobile Twinning and Mobile Call Control permission.
866-851-2649	X6233, x6237	Avaya 9611 Telephone Avaya 1616 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

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Verizon Provided Toll-Free Number	Configured Avaya IP Office Destination(s)	Notes
866-850-6850	Conference Bridge on Voicemail Pro	Allow external callers to access conference bridge toll-free
866-852-3221 (any caller)	"401 Call Center" 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 Call Center" 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 3 shows the equipment and software used in the sample configuration.

Equipment	Software
Avaya Session Border Controller for Enterprise	Release 7.1.0.0-04-11122
Avaya IP Office Server Edition (Primary Server)	Release 10.0.0.0.0 Build 550
Avaya IP Office Server Edition (Secondary Server)	Release 10.0.0.0.0 Build 550
Avaya IP Office IP500 V2 (Expansion System)	Release 10.0.0.0.0 Build 550
Avaya IP Office Manager	Release 10.0.0.0.0 Build 550
Avaya 9611SW IP Telephone (H.323)	Release 6.2209
Avaya 1140E IP Telephone (SIP)	Release 04.04.18
Avaya 9508 Digital Telephone	Release 0.55
Avaya T7316E Digital Telephone	N/A
Avaya Communicator for Windows	Release 2.1.3.80
Avaya one-X® Mobile Preferred for IP Office on Android	Release 10.0.0.184

Table 2: Equipment and Software Tested

Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2, and also when deployed with all configurations of IP Office Server Edition. Note that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks.

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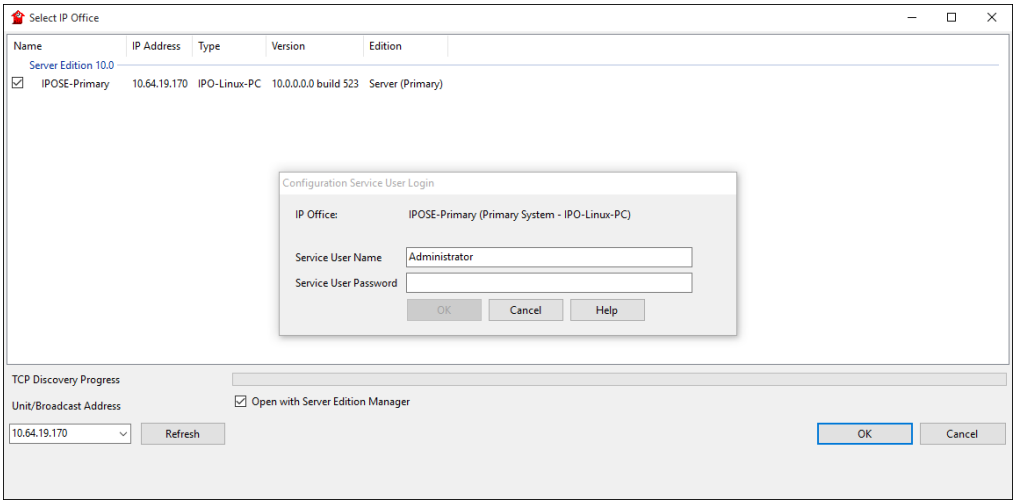
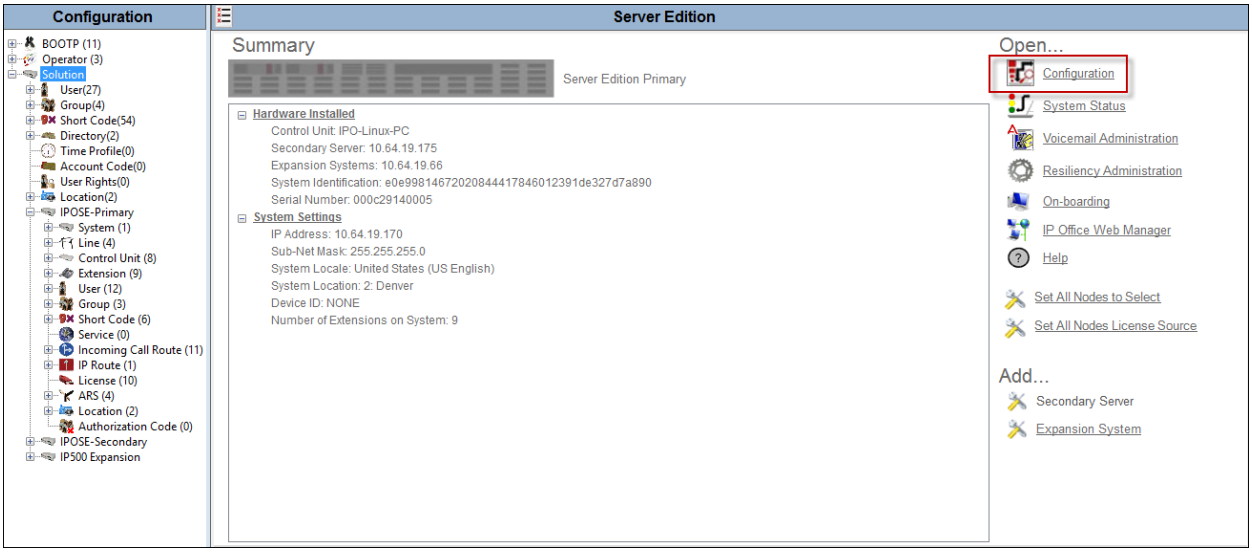
5. Avaya IP Office Configuration

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 → All Apps → IP Office → Manager** to launch the Manager application. Navigate to **File → Open Configuration** (not shown), select the proper Avaya IP Office system from the pop-up window, and log in using the appropriate credentials.



On Server Edition systems, the Solution View screen will appear, similar to the one shown below. If the left navigation pane does not immediately appear, click on the **Configuration** link as highlighted below. In the reference configuration, IP users registered to the Primary server and failover to the Secondary server. Digital and Analog users are configured on the Expansion System. A SIP trunk to the Primary SBCE for Verizon Business IPCC service is configured on the Primary server. Clicking the "plus" sign next to the Primary server system name, e.g., **IPOSE-Primary,** on the left navigation pane will expand the menu on this server.



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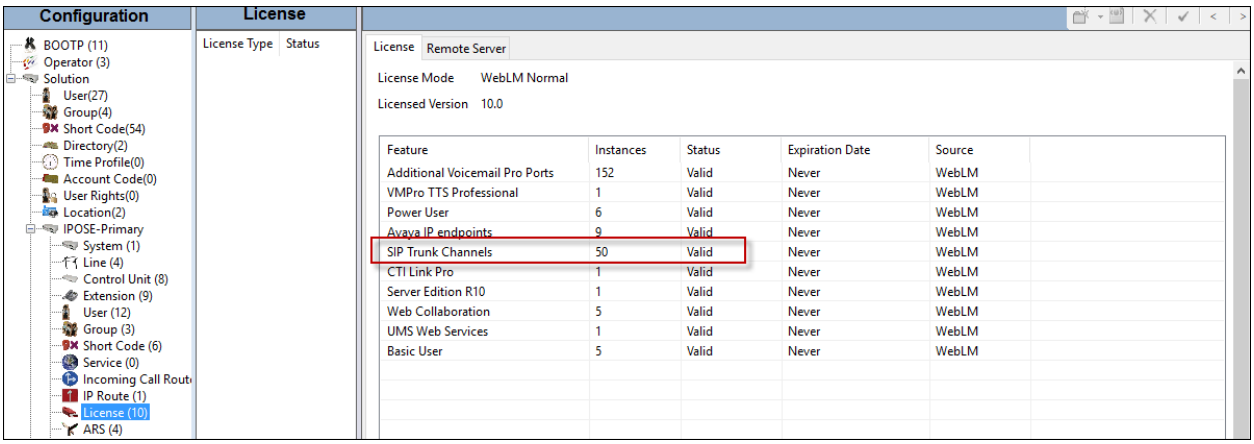
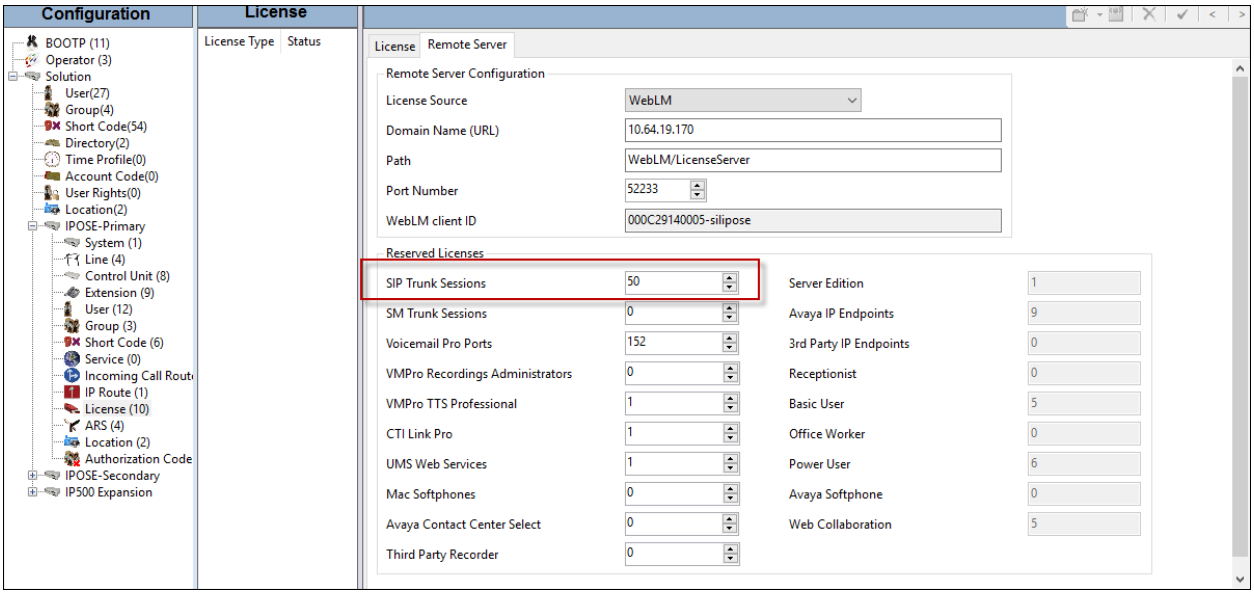
5.1. Licensing and Physical Hardware

In the sample configuration, **IPOSE-Primary** was used as the system name of the Primary Server, **IPOSE-Secondary** was used as the system name of the Secondary Server and **IP500 Expansion** was used as the system name of the Expansion System. All navigation described in the following sections (e.g., **License**) appears as submenus underneath the system name in the Navigation Pane.

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.





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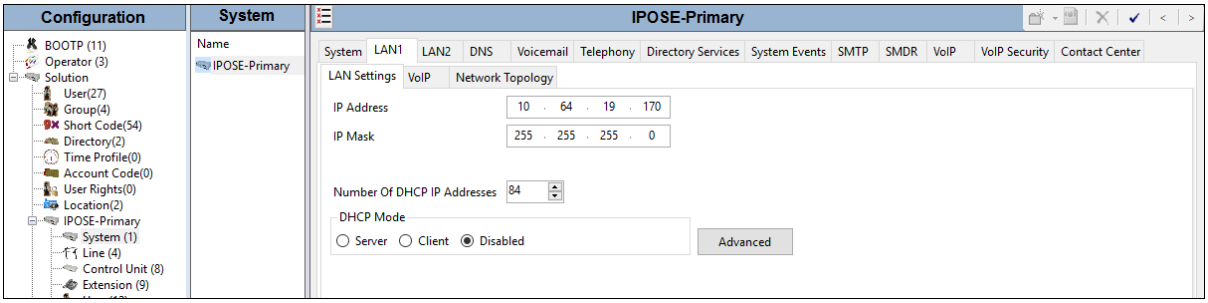
5.2. System Settings

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. For all of the following configuration sections, the **OK** button (not shown) must be selected in order for any changes to be saved.

5.2.1. LAN Settings

In the sample configuration, LAN1 is used to connect the Primary server 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 Primary server is **10.64.19.170**. 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** parameter 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** parameter is checked to allow Avaya 1140E, Avaya one-X® Mobile Preferred for IP Office, and Avaya Communicator usage. The **SIP Trunks Enable parameter** must be checked to enable the configuration of SIP trunks to Verizon Business. The **SIP Domain Name** and **SIP Registrar FQDN** may be set according to customer requirements.

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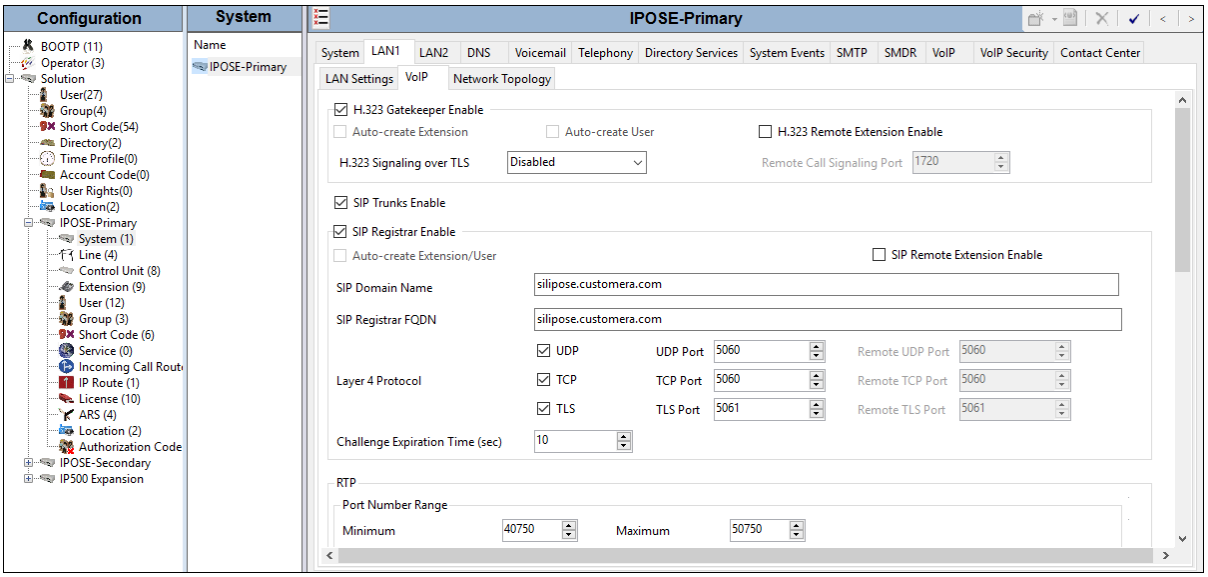
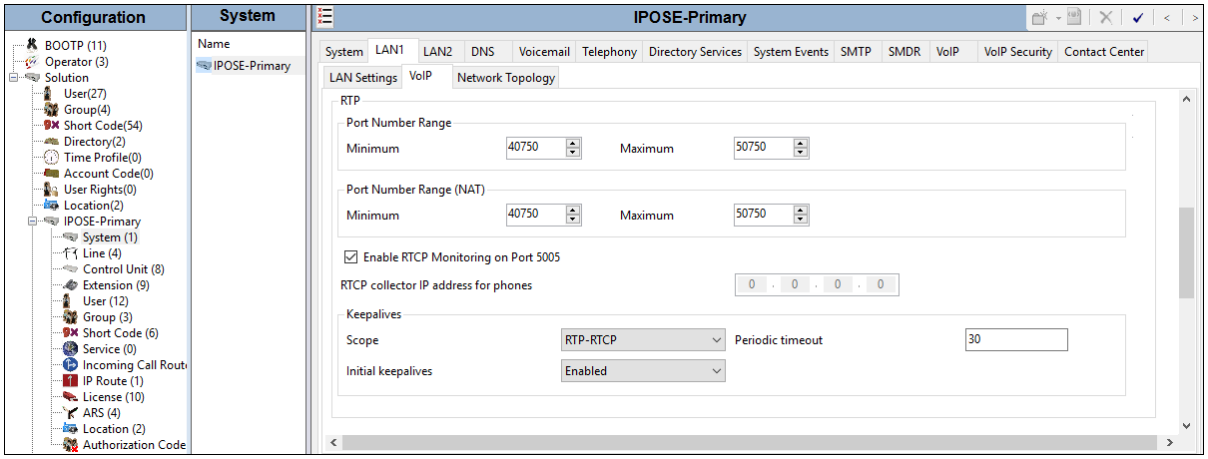
If desired, the **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media paths from Avaya SBCE to the Primary server. The defaults are used here.



Scroll down to the **Keepalives** section, and set the **Scope** to "**RTP-RTCP**". Set the periodic timeout to "**30**" and the **Initial keepalives** parameter to "**Enabled**". These settings will cause the Primary server to send RTP and RTCP keepalive packets starting at the time of initial connection and every 30 seconds thereafter if no other RTP or RTCP traffic is present. This facilitates the flow of media in cases where each end of the connection is waiting to see media from the other, as well as helping to keep ports open for the duration of the call.



Scrolling down, the Primary server 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

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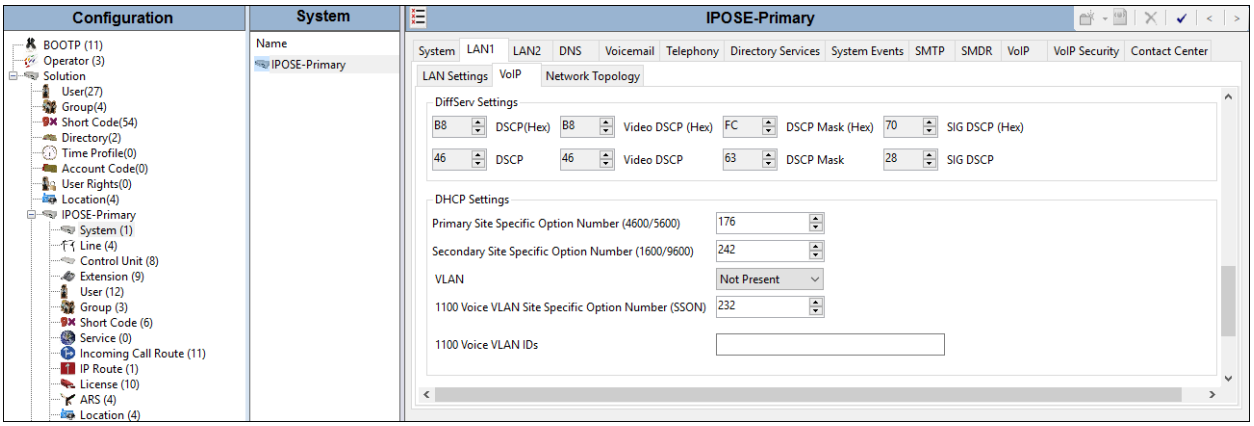
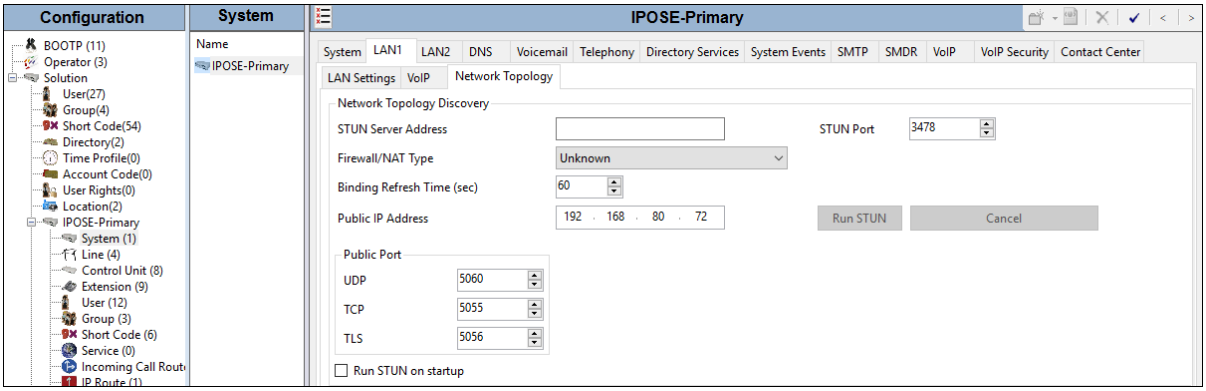
allow alignment with network routing policies, which are outside the scope of these Application Notes. Other parameters on this screen may be set according to customer requirements.



Select the **Network Topology** tab as shown in the following screen. The **Firewall/NAT Type** is set to "**Unknown**" in the sample configuration.



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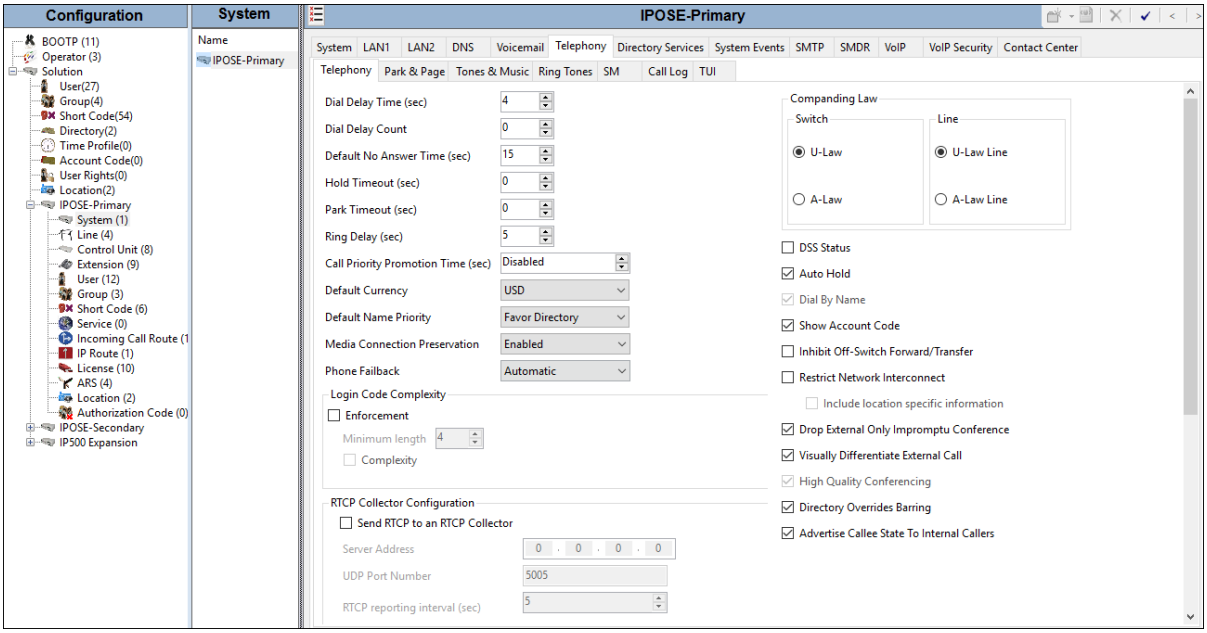
5.2.2. System Telephony Configuration

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 IP Contact Center service, 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.



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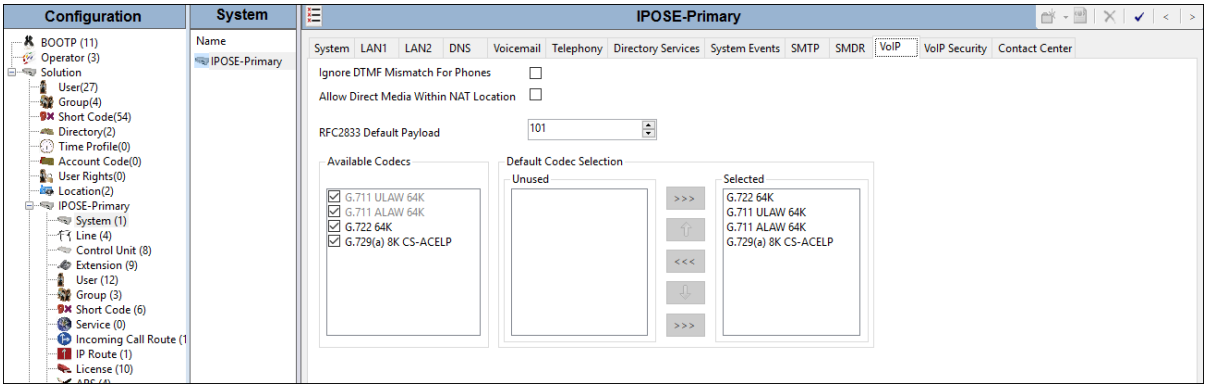
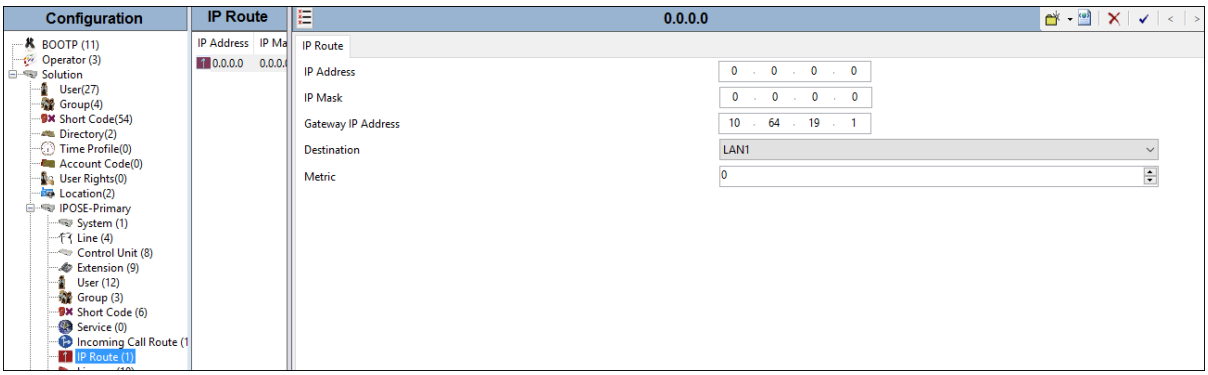
5.2.3. System Codecs Configuration

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 5.4**). 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 set to "**101**", the value preferred by Verizon Business.



5.3. IP Route

In the sample configuration, the Primary server LAN1 port is physically connected to the local area network switch at the IP Office customer site. The default gateway for this network is 10.64.19.1. 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 the Primary server, right-click **IP Route** from the Navigation pane, and select **New** (not shown). 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**".



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5.4. SIP Line

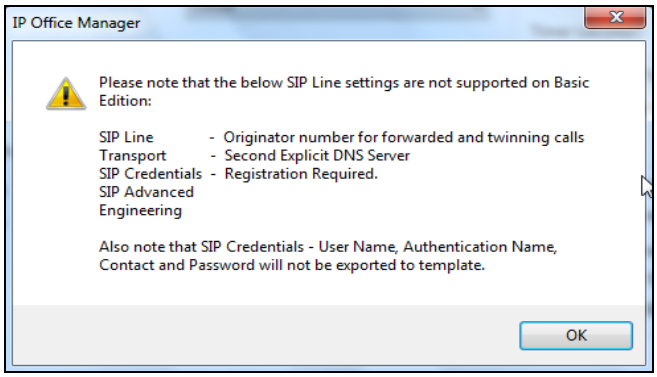
This section shows the configuration screens for the SIP Line in IP Office Release 10.0. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.4.2** 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 5.4.3 – 5.4.7**.

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



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 5.4.3 – 5.4.7**.

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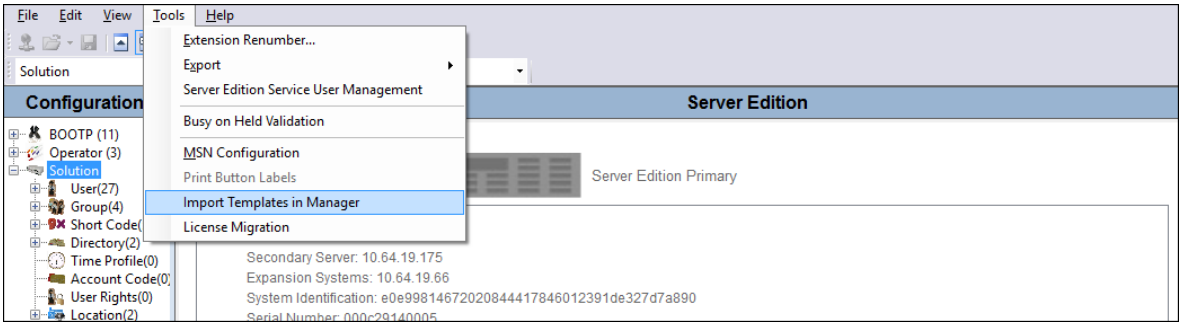
5.4.1. Importing a SIP Line Template

Note – DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500v2) and IP Office Server Edition systems. Alternatively, binary templates may be generated. However, binary templates include all the configuration parameters of the Trunk, including sensitive customer specific information. Therefore, binary templates should only be used for cloning trunks within a specific customer's environment.

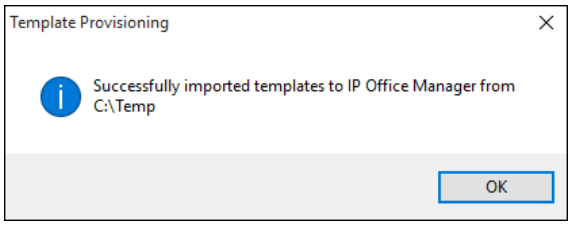
1. Copy a previously created template file to a location (e.g., *\temp*) on the same computer where IP Office Manager is installed.

2. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**.



3. A folder browser will open (not shown). Select the directory used in **step 1** to store the template(s) (e.g., *\temp*). In the reference configuration, template file **Verizon SBCE.xml** was imported. The template files are automatically copied into the IP Office default template location, **C:\Program Files\Avaya\IP Office\Manager\Templates**.

4. After the import is complete, a final import status pop-up window will open stating success or failure.



5.4.2. Creating a SIP Trunk from an XML Template

1. To create the SIP Trunk from a template, right-click on **Line** in the Navigation Pane, and hover over **New from Template**. A list of templates will appear, select the XML template name from **Section 5.4.1**

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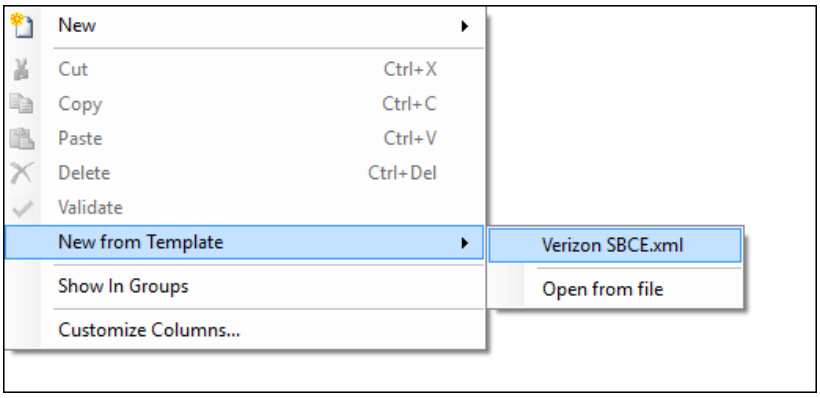
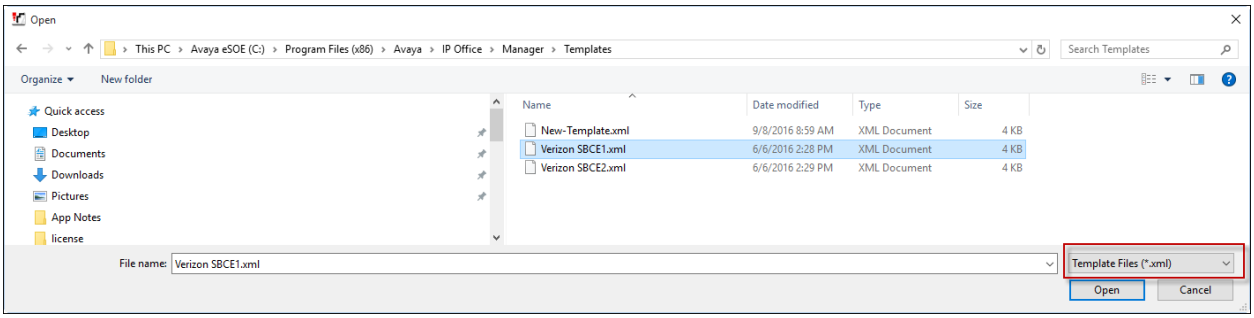


If the template is not listed, select **Open from file**, and navigate to **C:\Program Files\Avaya\IP Office\Manager\Templates**. Select ***.xml** as the file type, find the template, and click **Open**.



The newly created SIP Line will appear in the Navigation pane (e.g., SIP Line **9**).

Line Number	Line Type	Line SubType
1	IP Office Line	WebSocket Server SCN
3	IP Office Line	WebSocket Server SCN
9	SIP Line	

Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.4.3 –5.4.7**.

5.4.3. SIP Line – SIP Line Tab

The **SIP Line** tab in the Details pane is shown below for Line Number 17, 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**.

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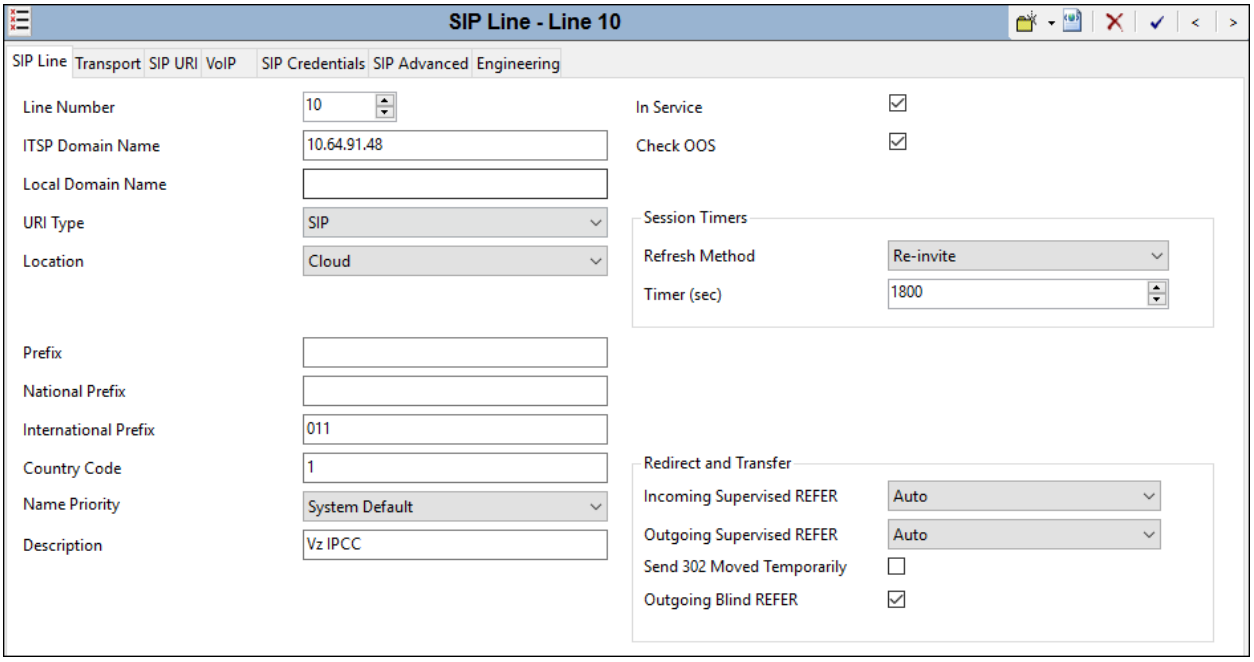
In the sample configuration, the IP Office **Country Code** is set to 1. The "From" and "PAI" headers received from Verizon for calls from U.S. PSTN numbers contain "+1" before the calling PSTN number. By configuring the IP Office **Country Code** to 1, the caller ID display presented to IP Office users 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 5.8.3** for configuration of incoming call routing based on the calling number.

Under **Session Timers**, the **Refresh Method** is set to "**Reinvite**" and the **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.

Under **Redirect and Transfer**, The default automatic determination of **Incoming Supervised REFER** and **Outgoing Supervised REFER** is "**Auto**". Verizon Business IPCC service does not support supervised REFER, and with this setting, IP Office will not send a REFER for supervised transfers. The **Send 302 Moved Temporarily** setting is unchecked, as Verizon does not support receiving a 302 Moved Temporarily message. Optionally, the **Outgoing Blind REFER** box can be checked to enable use of REFER for blind transfers. In the sample configuration, this parameter is checked. See **Section 2.2** for limitations.



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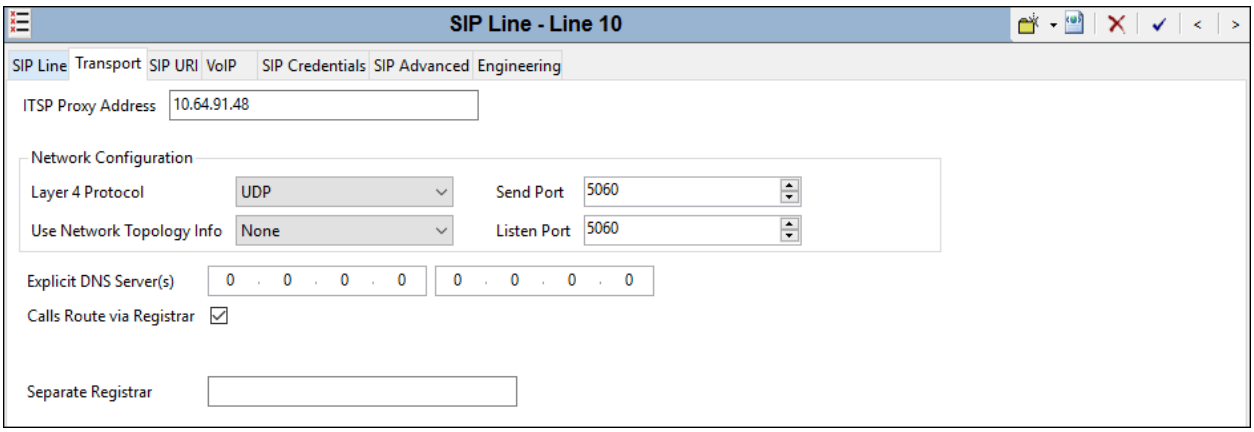
5.4.4. SIP Line – Transport Tab

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, "**UDP**" 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**".



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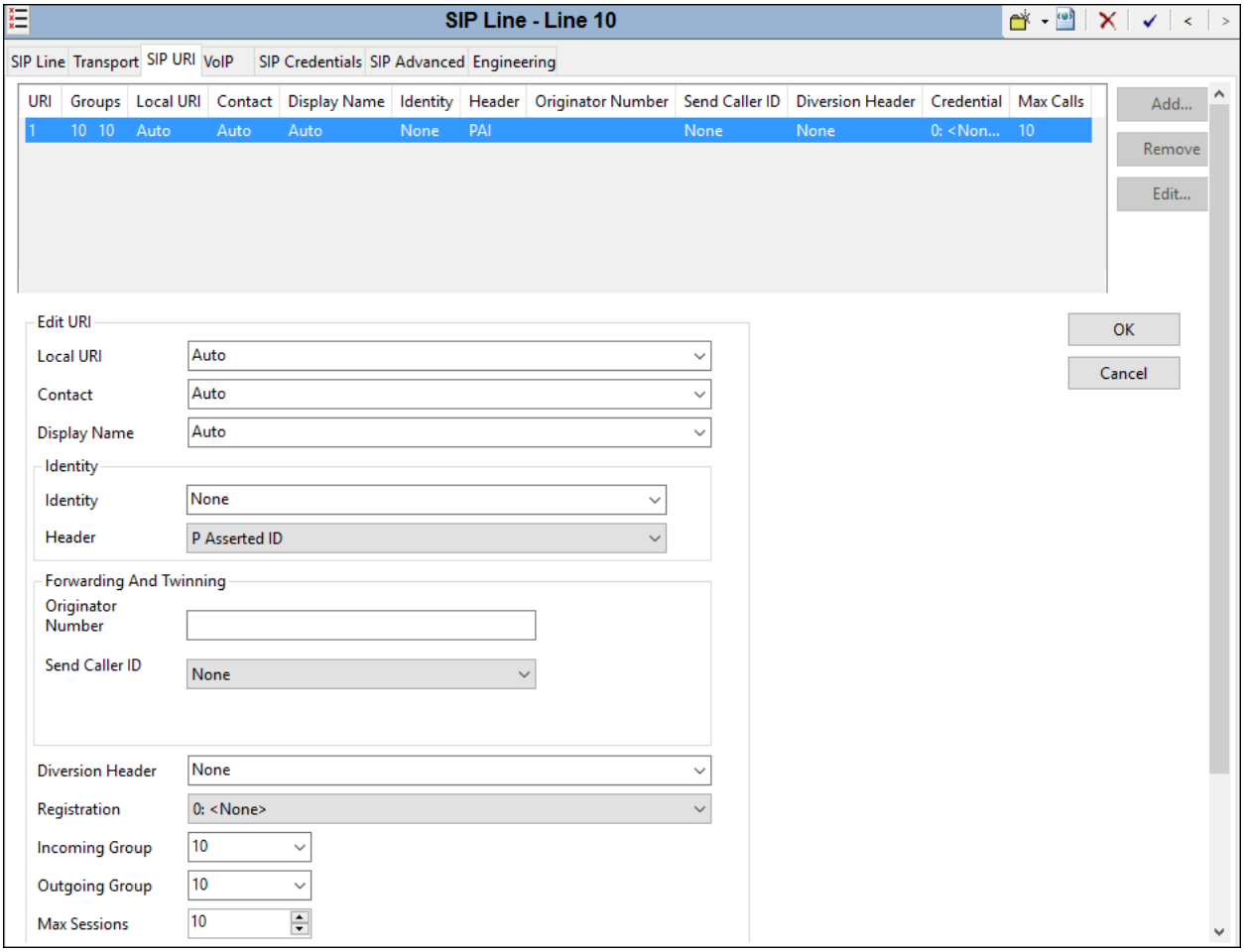
5.4.5. SIP Line – SIP URI Tab

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 URI area will be opened.

In the sample configuration, "**Auto**" is selected for the **Local URI**, **Contact** and **Display Name**. This setting replaces the wildcard "*****" used in previous releases. The **Identity** parameter is set to "**None**", and the **Header** parameter is set to "**P Asserted ID".** 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 "**10**", 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, also set here to "**10**", is relevant when using the SIP REFER method to transfer an inbound toll-free call back out the Verizon Business IPCC service. Click **OK**.



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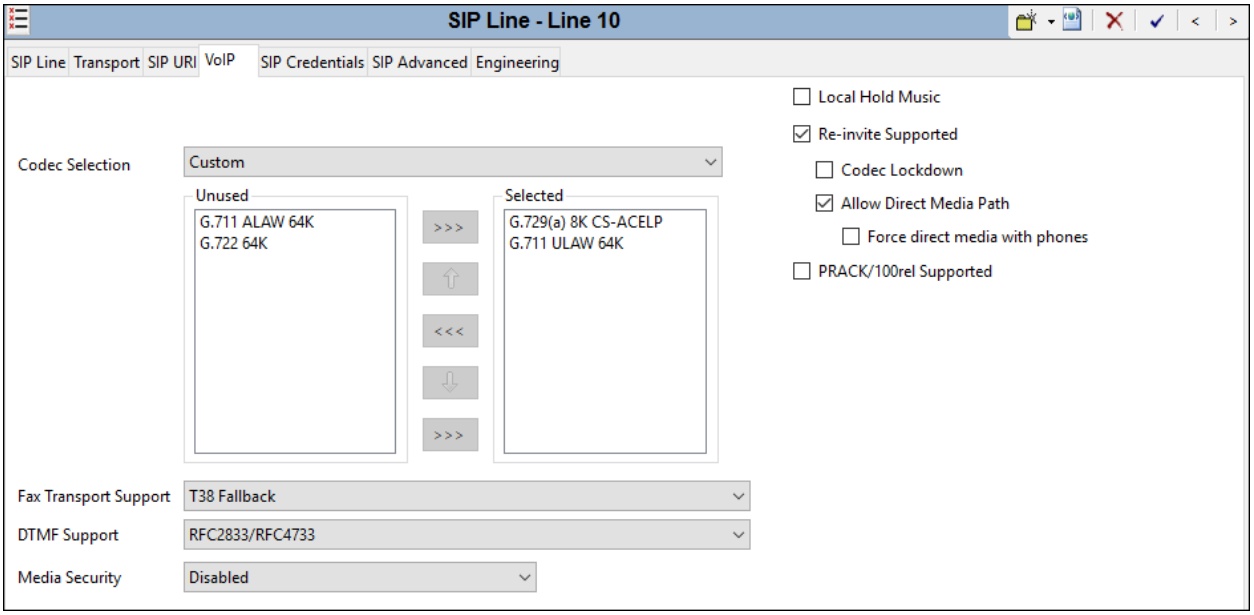
5.4.6. SIP Line – VoIP Tab

Select the **VoIP** tab. The **Codec Selection** drop-down parameter **System Default** (default) will match the codecs set in the system wide Default Selection list (**System → Codecs**). In the sample configuration, "**Custom**" is selected and codecs preferred by Verizon are included (i.e., G729(a) 8K CS-ACELP and G.711 ULAW 64K). This will cause IP Office to include G.729a and G.711MU in the Session Description Protocol (SDP) offer, in that order. Set the **Fax Transport Support** drop-down to "**T38 Fallback**". This enables T.38 to be used if supported and will fall-back to G.711 if not. The **DTMF Support** parameter can remain set to the default value "**RFC2833/RFC4733**". The **Media Security** parameter can retain its default value of "**Disabled**" as Verizon does not support media encryption. The **Re-invite Supported** parameter can be checked to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk. The **Allow Direct Media Path** parameter can be checked to allow for direct media between IP endpoints and the internal interface of the Avaya SBCE, freeing up DSP resources on the Primary server. The **PRACK/100rel Supported** parameter can be checked to enable support for the PRACK (Provisional Reliable Acknowledgement) message on SIP trunks.

For PSTN originations, Verizon preferred the G.729a codec in the SDP, while also allowing the G.711MU codec. During 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.



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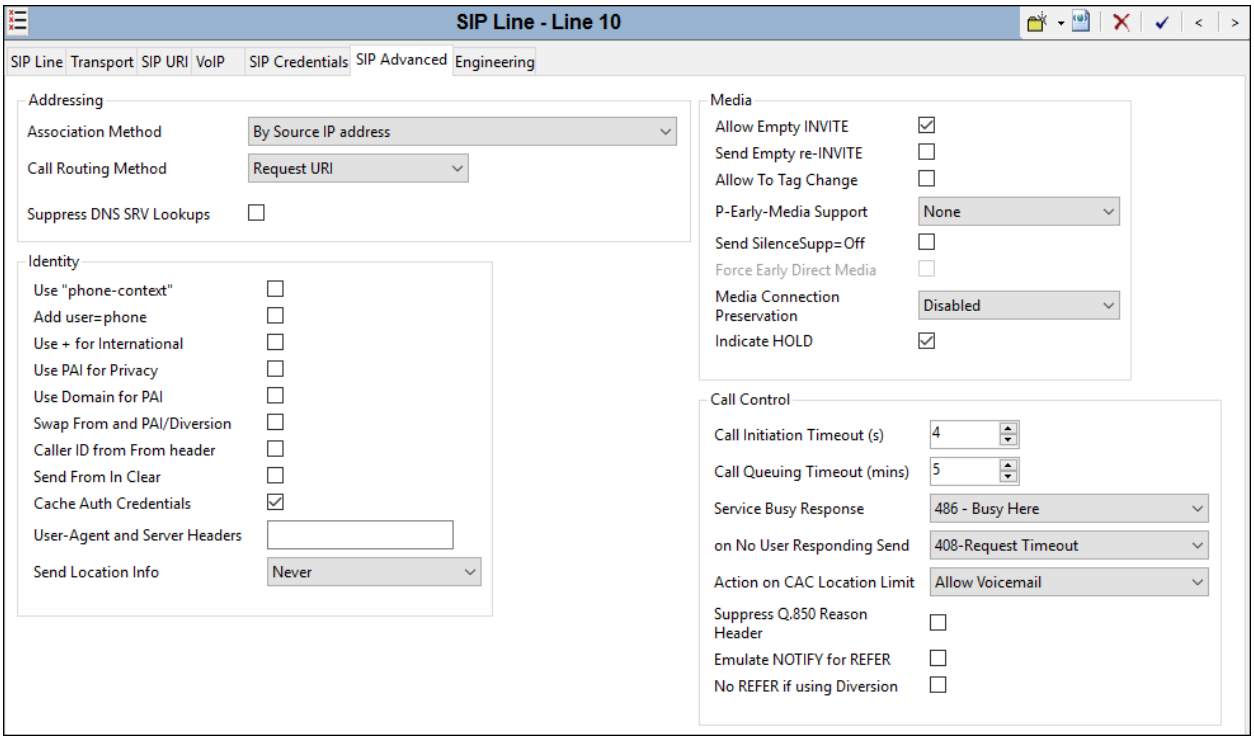
5.4.7. SIP Line – SIP Advanced Tab

Select the **SIP Advanced** tab. In the **Media** area, the **Allow Empty INVITE** box is checked to allow IP Office to be the recipient of a Verizon Business IPCC enhanced transfer where the initial INVITE may not have SDP information. The **Indicate HOLD** box is checked to have IP Office signal to Verizon when a call is placed on/off hold. Other parameters may be left at default values.



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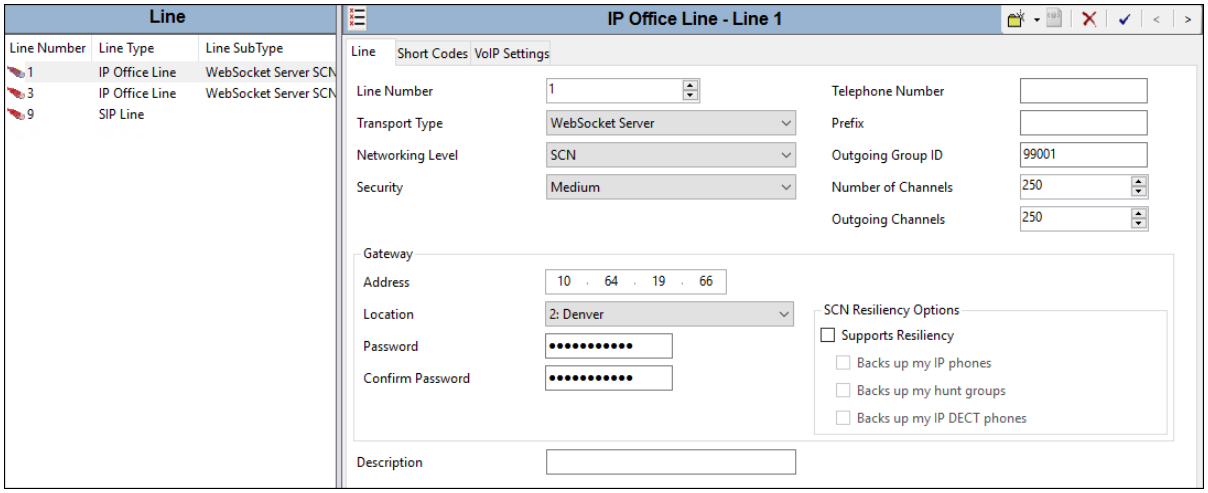
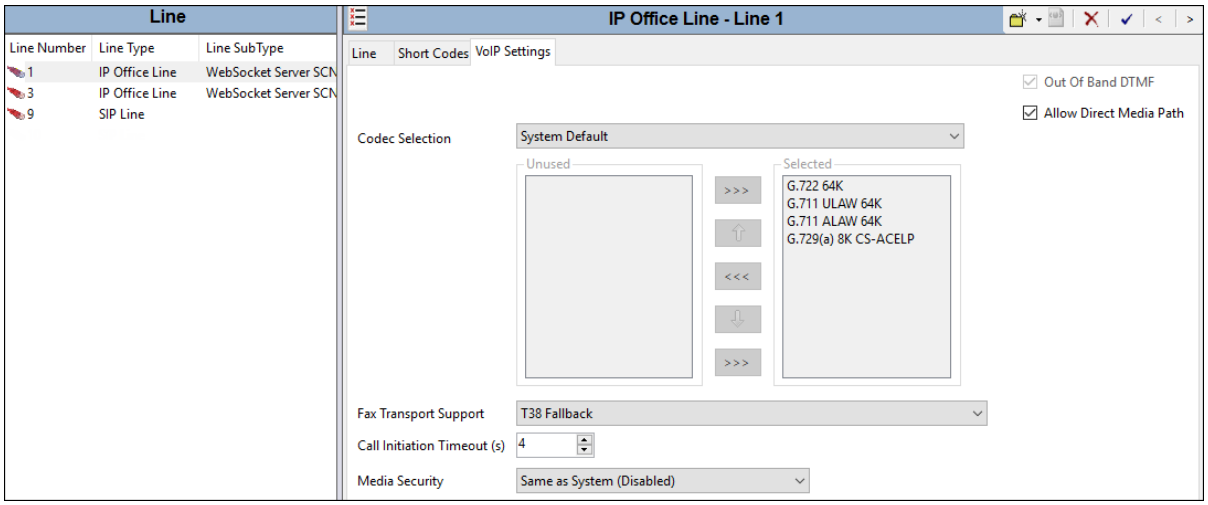
5.5. IP Office Line

IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution. To edit an existing IP Office Line, select **Line** in the Navigation pane, and select the appropriate Line to be configured in the Group pane. Below is the IP Office Line to the Expansion System.



Select the **VoIP Settings** tab. In the reference configuration, a fax machine is connected to one of the analog ports on the Expansion System. To accommodate T.38 fax, set the **Fax Transport Support** drop-down to "**T38 Fallback**" on the **VoIP Settings** tab. The **Allow Direct media Path** parameter is checked to allow RTP and T.38 packets to route directly to the internal IP address of the Avaya SBCE.



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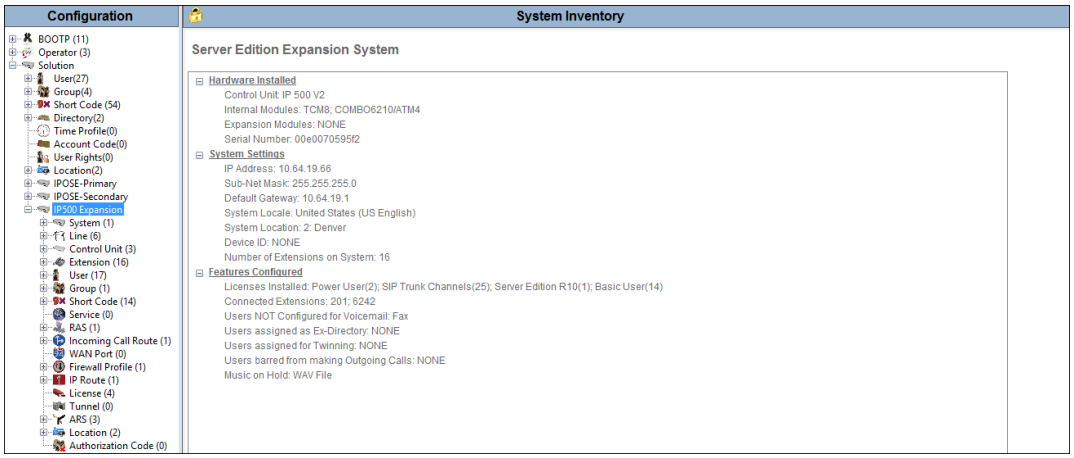
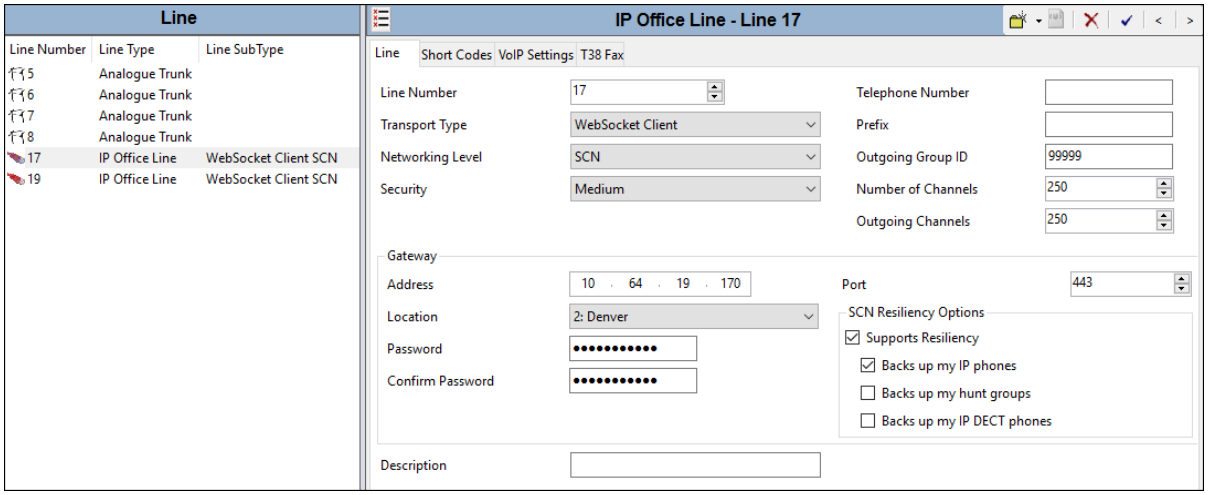
Similarly, modify the IP Office Line on the Expansion System. Click the "plus" sign next to **IP500 Expansion** on the left navigation pane to expand the menu on this server.



Select **Line** in the Navigation pane, and select the appropriate Line to be configured in the Group pane. Below is the IP Office Line to the Primary server.



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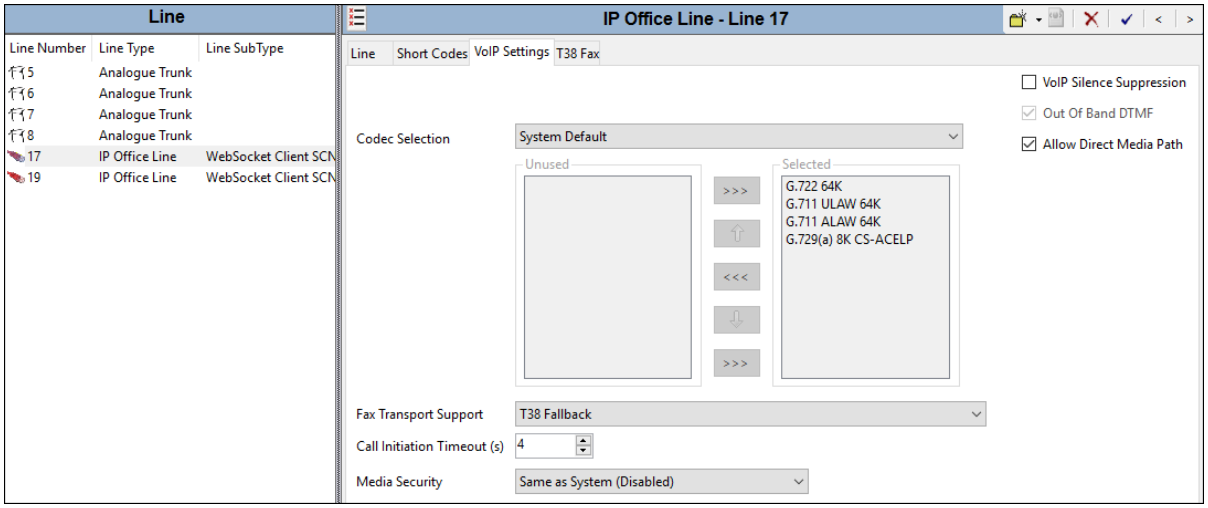
Select the **VoIP Settings** tab to set the **Fax Transport Supported** to "**T38 Fallback**".



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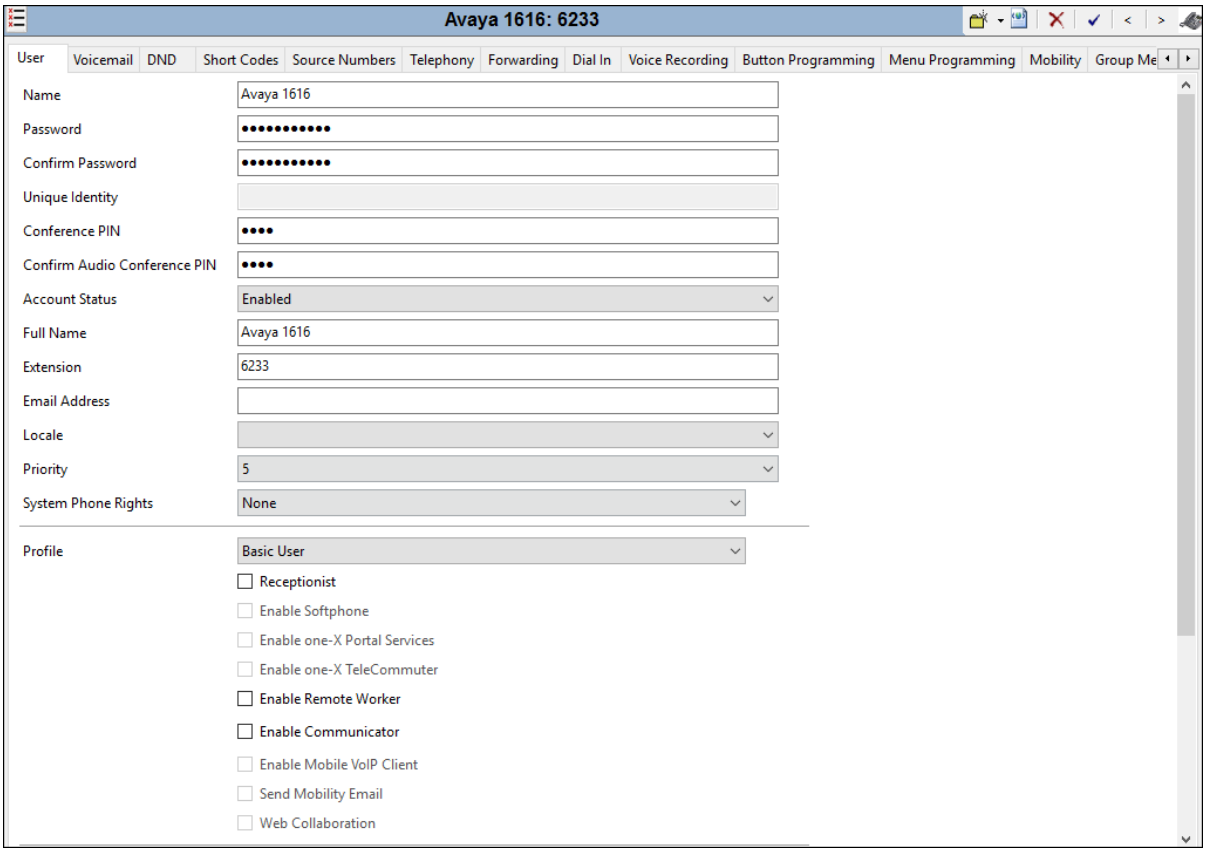
5.6. Users, Extensions, and Hunt Groups

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.

5.6.1. H.323 User 6233

The following screen shows the **User** tab for User 6233. As shown in **Figure 1**, this user corresponds to the Avaya 1616 H.323 endpoint.



The following screen shows the **SIP** tab for User 6233. In the sample configuration, the **SIP Name** and **Contact** parameters are configured with a Verizon Business IP Trunk DID number for the user, 7329450233. As shown in [VZBIPT-IPO10SBC], 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.

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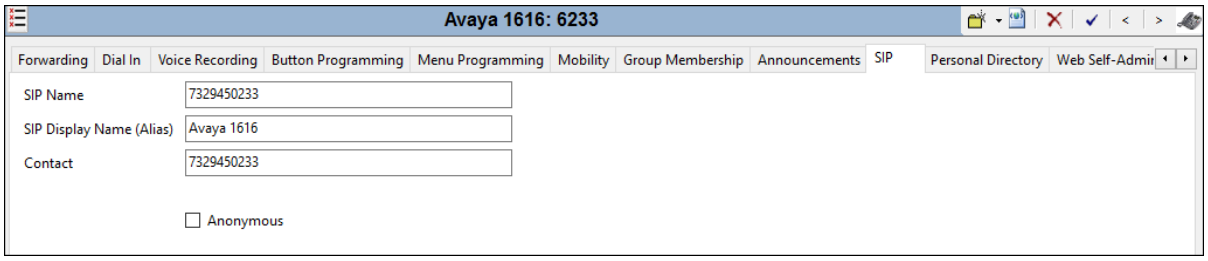
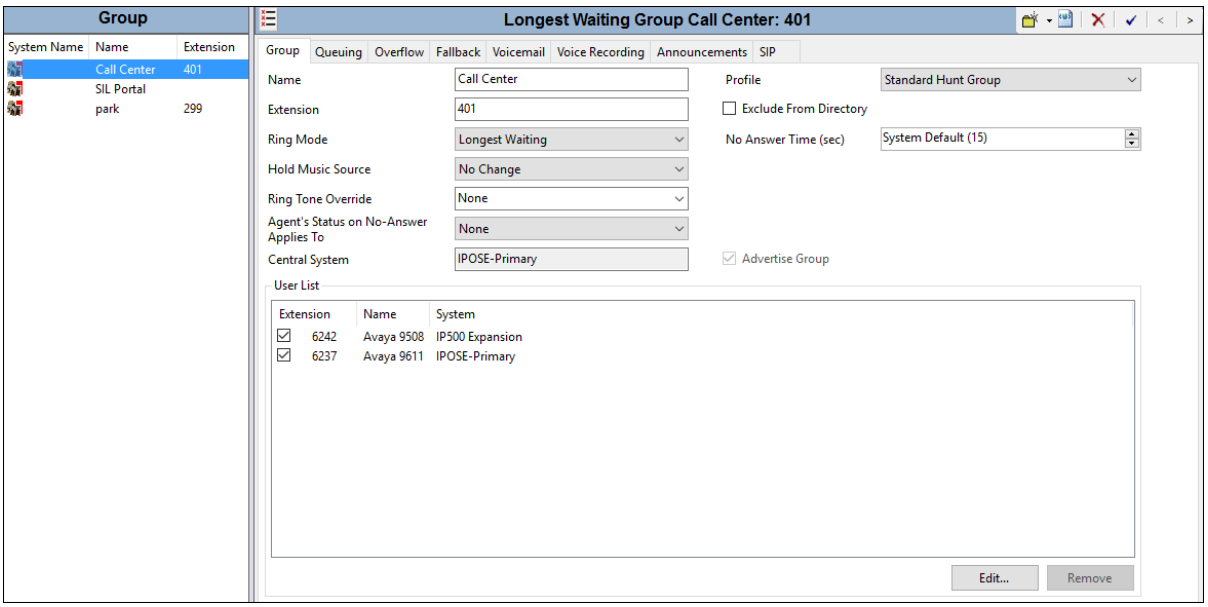


5.6.2. Hunt Groups

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** "**Call Center**". 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**.



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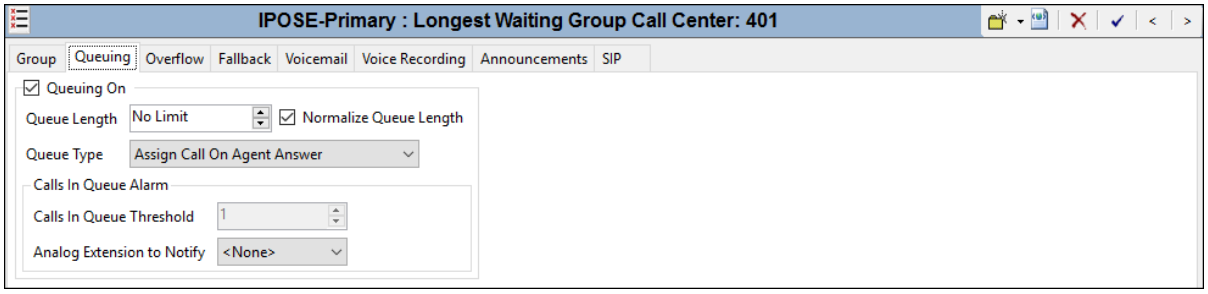
The following screen shows the **Queuing** tab for hunt group 401. In the sample configuration, the hunt group was configured to allow queuing so that incoming Verizon toll-free calls could be queued when all the members of the hunt group were busy on calls. In the testing associated with these Application 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 5.8.3**) 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.

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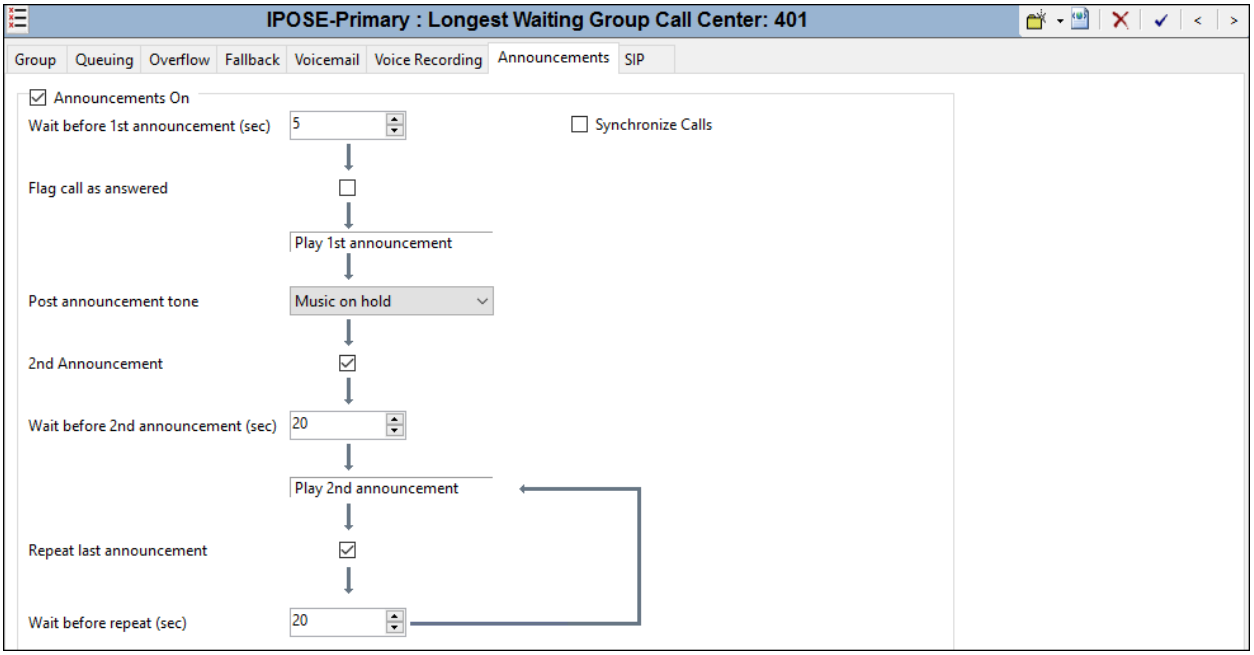
IP Office supports priority for queuing. For example, if low priority calls are waiting in queue, a higher priority call entering queue can be moved to the front of the queue and serviced before lower priority callers. For an inbound SIP trunk call, the priority can be specified on the Incoming Call Route as shown in **Section 5.8.3**.



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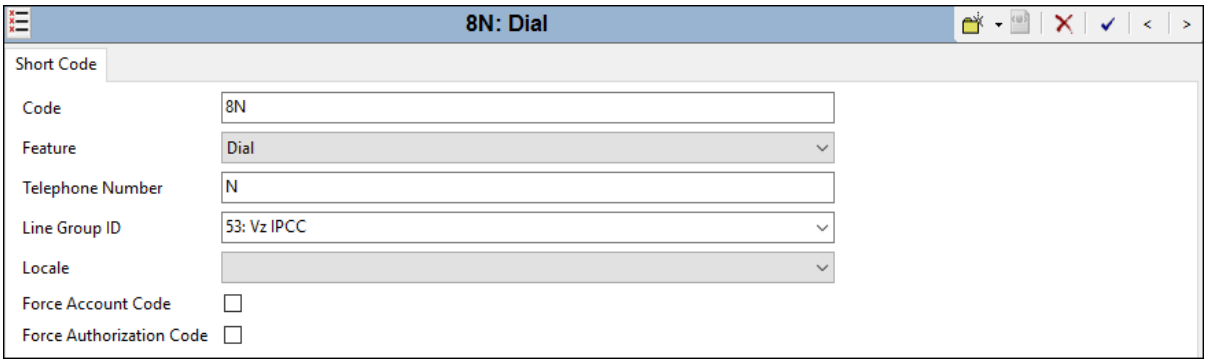
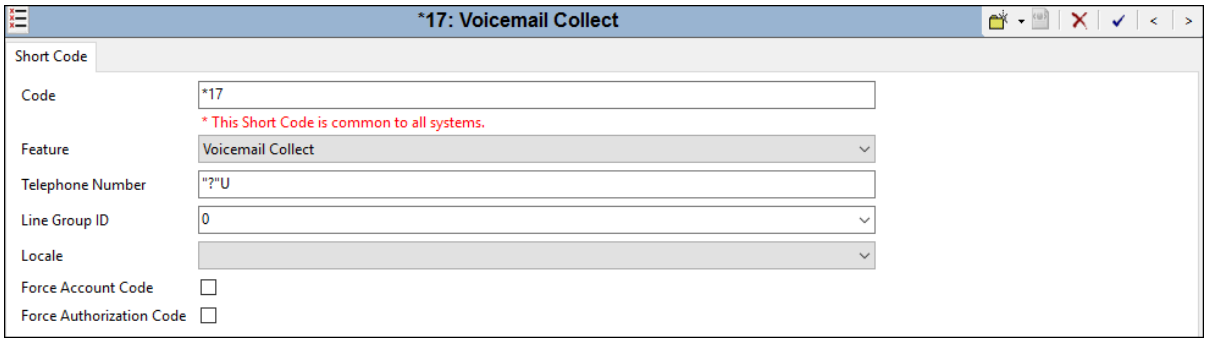
5.7. Short Codes

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.

Verizon Business IPCC service allows for blind transfers out their service using the SIP REFER Method. In the screen shown below, the short code "**8N**" is illustrated. The **Code** parameter is set to "**8N**". The **Feature** parameter is set to "**Dial**". The **Telephone Number** parameter is set to "**N**". The **Telephone Number** field is used to construct the Request URI and To Header in the outgoing SIP INVITE message. The **Line Group ID** parameter is set to "**53: VzIPCC**", configurable via ARS. See **Section 5.10** for example ARS route configuration for "53: VzIPCC.



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 5.8** for configuration of Incoming Call Routes.



The following screen illustrates another short code. In this case, the **Code** "**551N;**" 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

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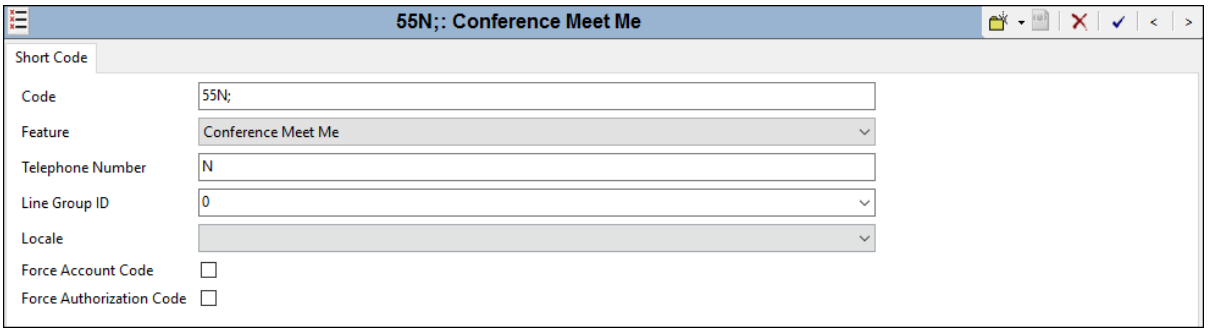
Application Notes, the module enabled a PSTN caller to dial a Verizon toll-free number, be prompted to enter a conference ID and PIN by Voicemail Pro, and then be transferred to the appropriate meet-me conference based on the ID entered by the caller. Local IP Office callers could also dial 556xxx to join the corresponding conference ID.

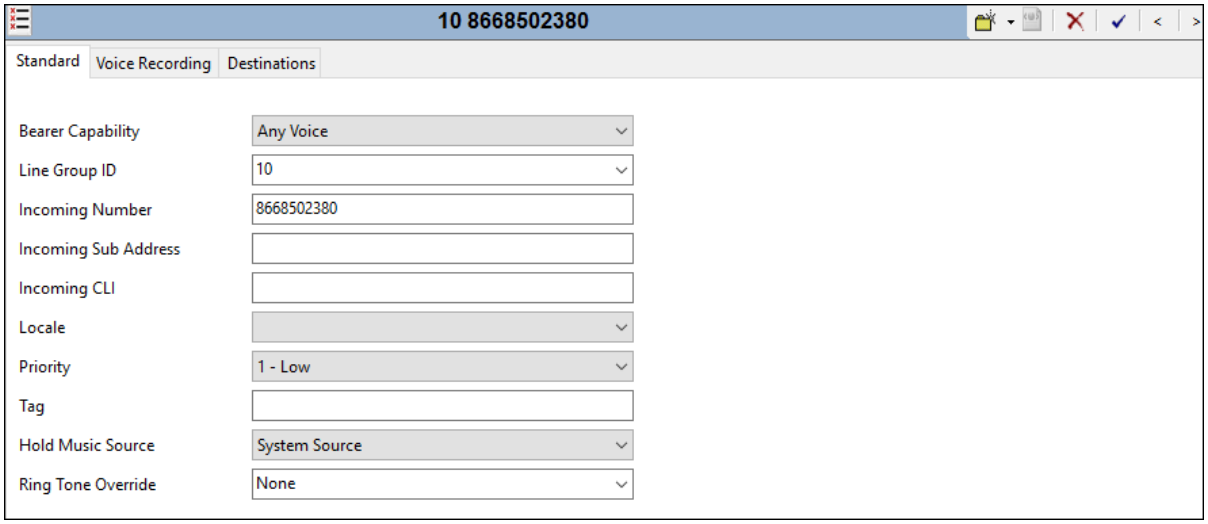


5.8. Incoming Call Routes

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 3**. 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.

5.8.1. 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 "**10**", matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to the Verizon Business IPCC service, in **Section 5.4.5**.



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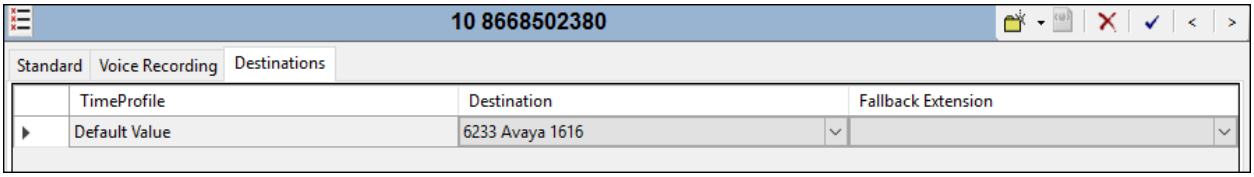
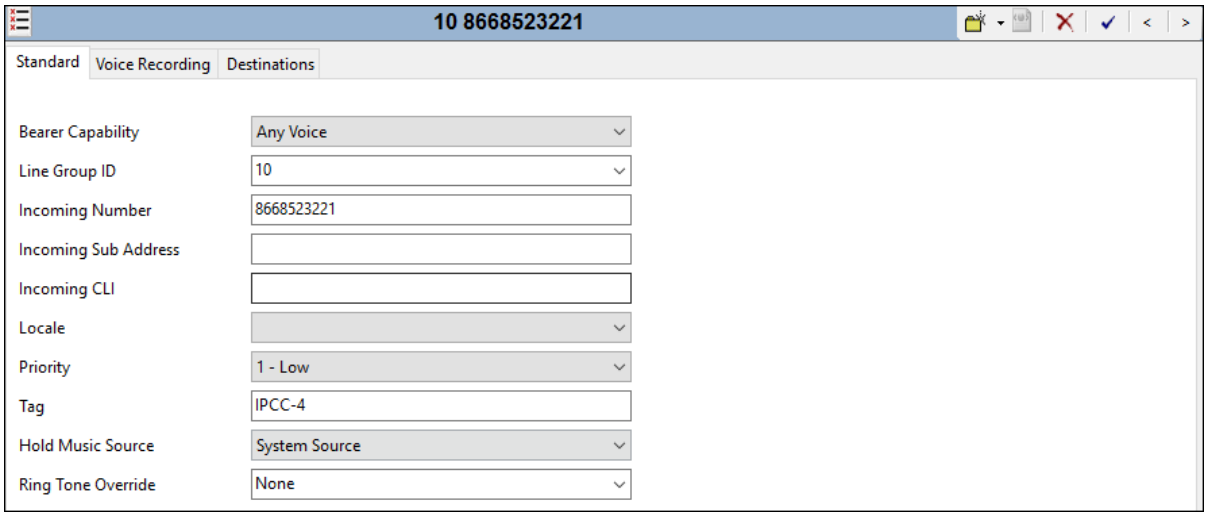
Select the **Destinations** tab. From the **Destination** drop-down, select an extension to receive the call when a PSTN user dials 8668502380. As shown in **Table 1**, 8668502380 is the number associated with IP Office user extension 6233. (The **Destination** was changed in the course of testing to associate different destinations with the toll-free numbers.)



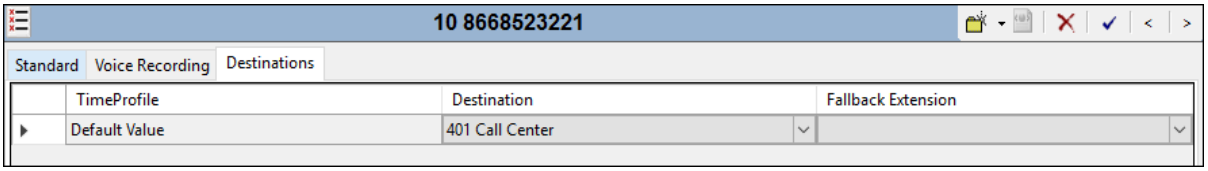
Incoming Call Routes for other direct mappings of toll-free numbers to IP Office users are not presented here, but are configured in the same fashion.

5.8.2. 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 "**10**", matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in **Section 5.4.5**. Optionally, the **Tag** parameter can be populated with a descriptive name that will associate the call with this incoming call route.



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**, 8668523221 is the toll-free number associated with IP Office hunt group extension 401, the "Call Center" hunt group.



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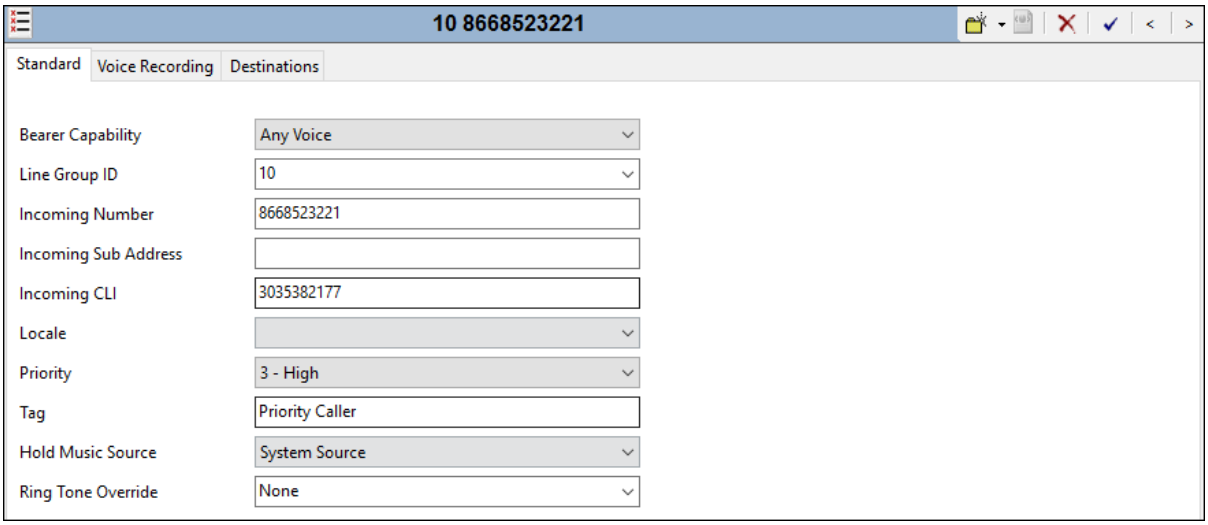
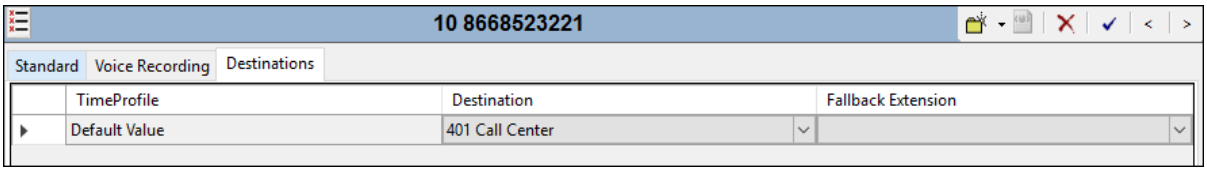
5.8.3. Incoming Call Routes Based on Calling Party Number

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** "**3035382177**" is illustrated. The **Line Group Id** is "**10**", 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 "3035382177". In this case, to allow this caller to be treated with priority when calling in, the **Priority** field is set to "**3 – High**". Optionally, the **Tag** parameter can be populated with a descriptive name that will associate the call with this incoming call route.



Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when PSTN user 3035382177 dials 8668523221. In this case, the **Destination** is also the hunt group "**401 Call Center**", 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.



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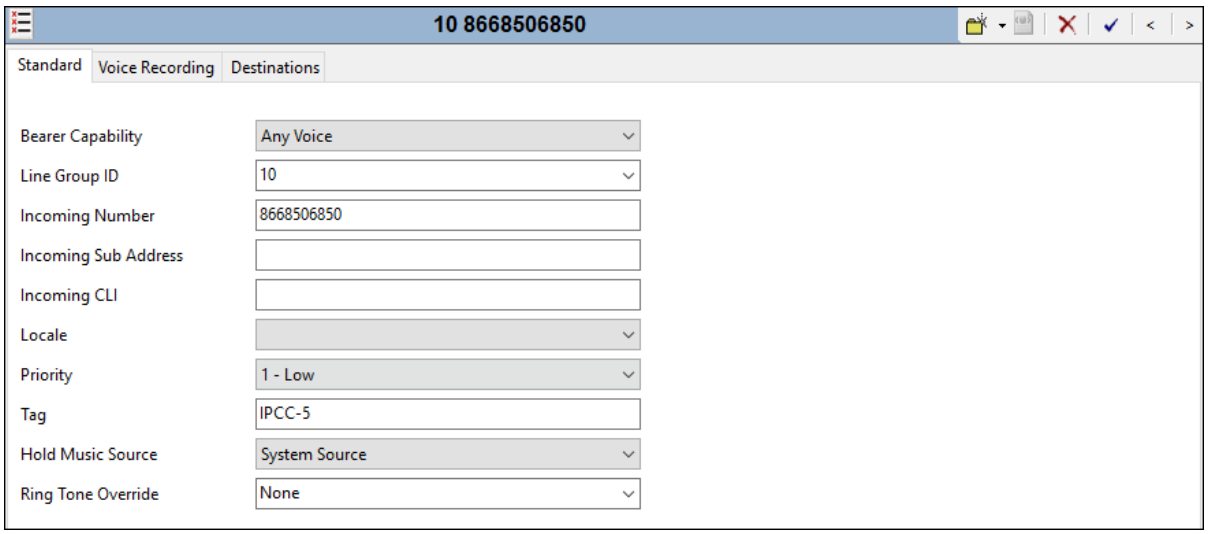
5.8.4. Incoming Call Routes to Various IP Office Features

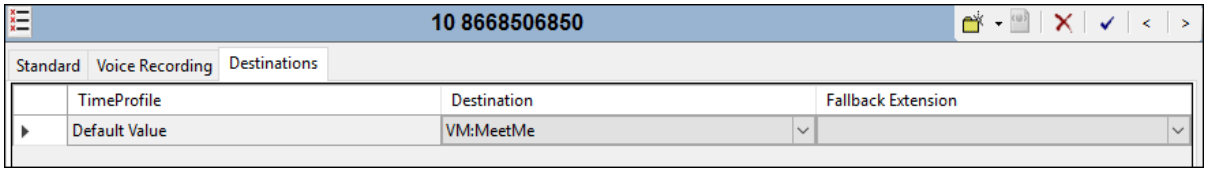
In the sample configuration, the incoming call route for **Incoming Number** "**8668506850**" was varied to test different destination features, such as Voice Mail, Mobile Call Control, Refer Call Redirection, and Conference Bridge, as shown in **Table 1** in **Section 3**. The screen showing the **Standard** 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 5.7**). 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:MeetMe**" With this destination, an incoming call to 8668506850 will be delivered directly to the Voicemail Pro Module "MeetMe" created for use as a conference bridge.
- "**VM:Refer**" With this destination, an incoming call to 8668506850 will be delivered directly to the Voicemail Pro Module "MeetMe" created for use as a Refer Call Redirection example.

An example screen showing the short code configured for a Voicemail Pro Module is shown below.



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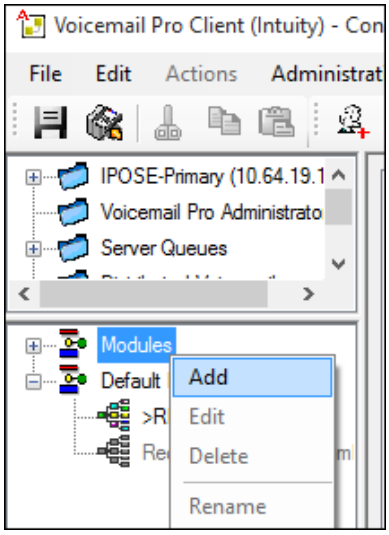
5.9. Voicemail Pro Refer Module

Note - While Avaya Voicemail Pro provisioning and programming is beyond the scope of this document, a sample Module is described below.

The Refer Module is provisioned to play an announcement to the caller, and then generate a Refer (without Replaces) back to the Verizon Business IPCC service. This is accomplished via the Voicemail Pro Client interface.

From the IP Office Manager PC, select **Start → All Apps → IP Office → Voicemail Pro Client** to launch the Voicemail Pro Client interface. Navigate to **File → Login**, select the proper Voicemail Pro system, and log in using the appropriate credentials (not shown). Create a **Start Point** by right clicking on **Modules** and selecting **Add**.



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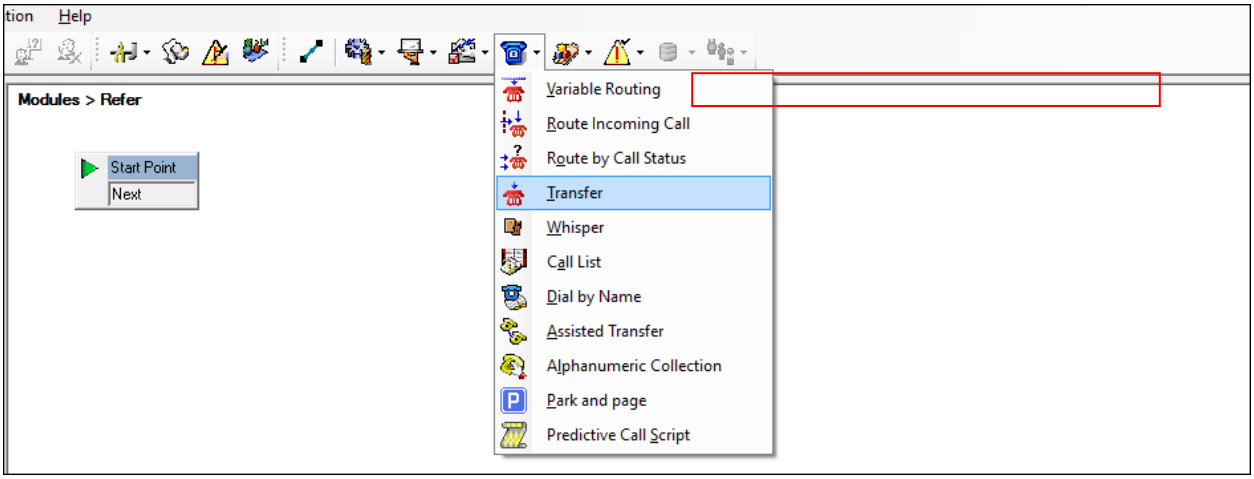
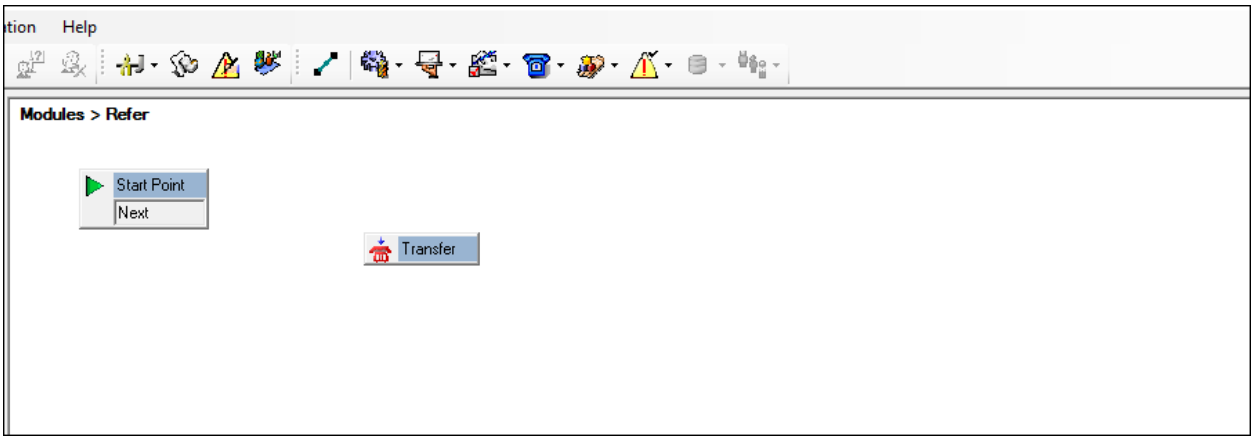
Enter a name (e.g., **Refer**) and click on **OK** (not shown). The new Module "Refer" will appear, and a **Start Point** icon will appear in the work area.

Click on the **Telephony Actions** icon , select the **Transfer** icon , and click on the work area to place the **Transfer** icon in the work area.





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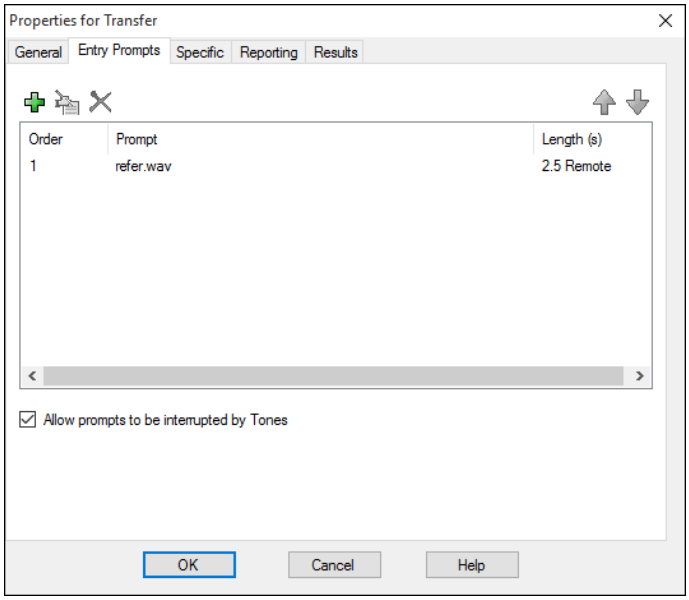
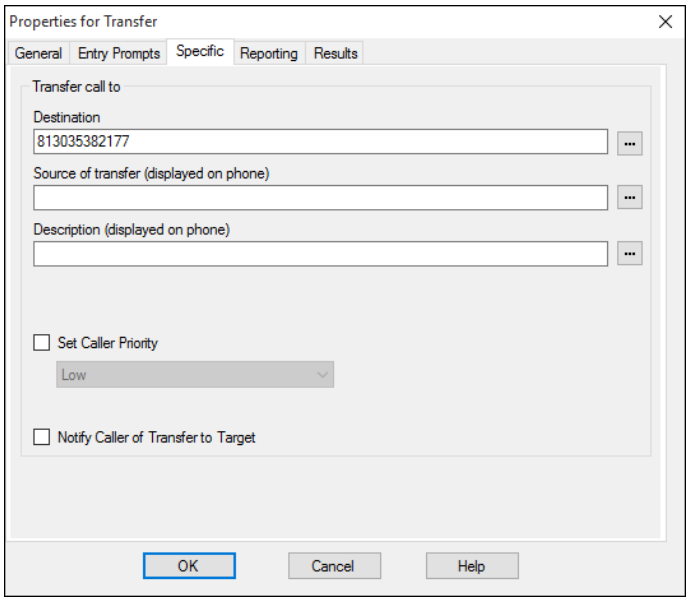
Double click on the **Transfer** icon, select the **Entry Prompts** tab and select or create an announcement to be played to the caller prior to the Refer (e.g., **refer.wav**). To modify an existing recording, double click on the .wav file and rerecord. If no .wav files exist, click on the icon to open the .wav file editor.



On the **Specific** tab enter the destination, including the outbound Short Code (e.g., **813035382177**). Click on **OK**.



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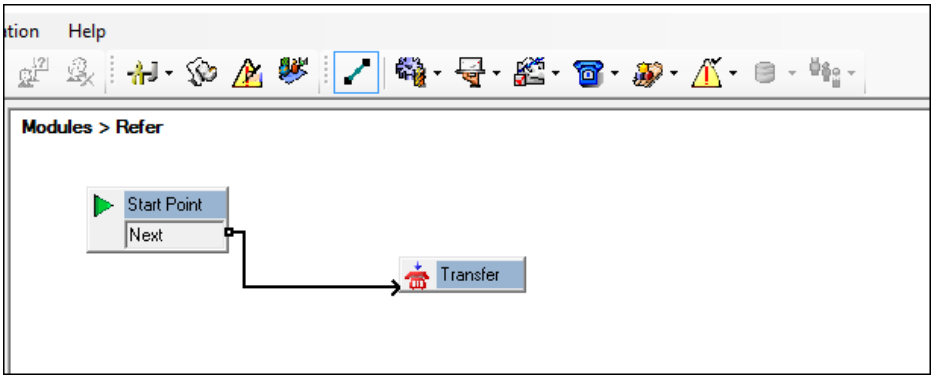
From the options bar, select the **Connector** icon and drag a connecting flow line from the **Start Point** box to the **Transfer** box.



From the top menu select **File → Save & Make Live**, or select the icon.

When the associated Verizon Business Toll-Free number is received, IP Office will send the call to Voicemail Pro (see **Section 5.8.4**). The caller will hear the announcement (e.g., **refer.wav**), and Voicemail Pro/Avaya IP Office will send a Refer back to the Verizon Business IPCC service, specifying "**13035382177**" in the Refer-To header. The Verizon Business IPCC service will then send a new Invite to the 1-303-538-2177 destination.

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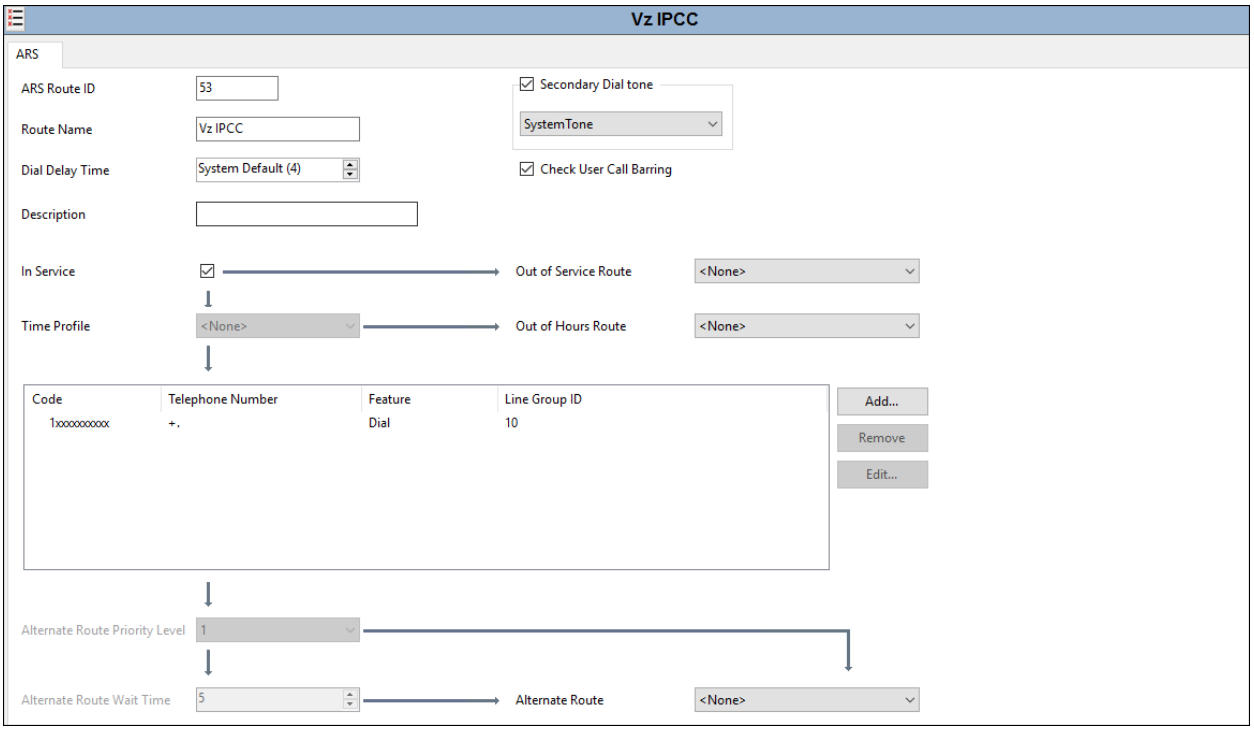
5.10. Alternate Route Selection (ARS)

Alternate Route Selection (ARS) is used to route outbound traffic to the SIP line. To define a new ARS route, right-click **ARS** in the Navigation pane and select **New**. In the Details pane that appears, a collection of matching patterns (similar to short codes) can be entered to route calls as shown below.

To add a new ARS route, right-click **ARS** in the Navigation pane, and select **New**. To view or edit an existing ARS route, select **ARS** in the Navigation pane, and select the appropriate route name in the Group pane.

The following screen shows an example ARS configuration for the route named "**VzIPCC**". Verizon Business IPCC service allows for blind transfers using the SIP REFER Method. The sequence of **X**s used in the **Code** column of the entries to specify the exact number of digits to be expected following the access. The entry below shows that for calls to area codes in the North American Numbering Plan, the user dials 8, followed by 11 digits. The **Telephone Number** is set to "**+.**". This prepends a plus sign (+) to the beginning of the number dialed, denoting a global E.164 number. This is the format preferred by Verizon Business IPCC service for the destination number specified in the Refer-To header. The **Line Group ID** is set to "**10**" matching the number of the **Outgoing Group** configured on the **SIP URI** tab of SIP Line 10 to Verizon Business (**Section 5.4.5**).



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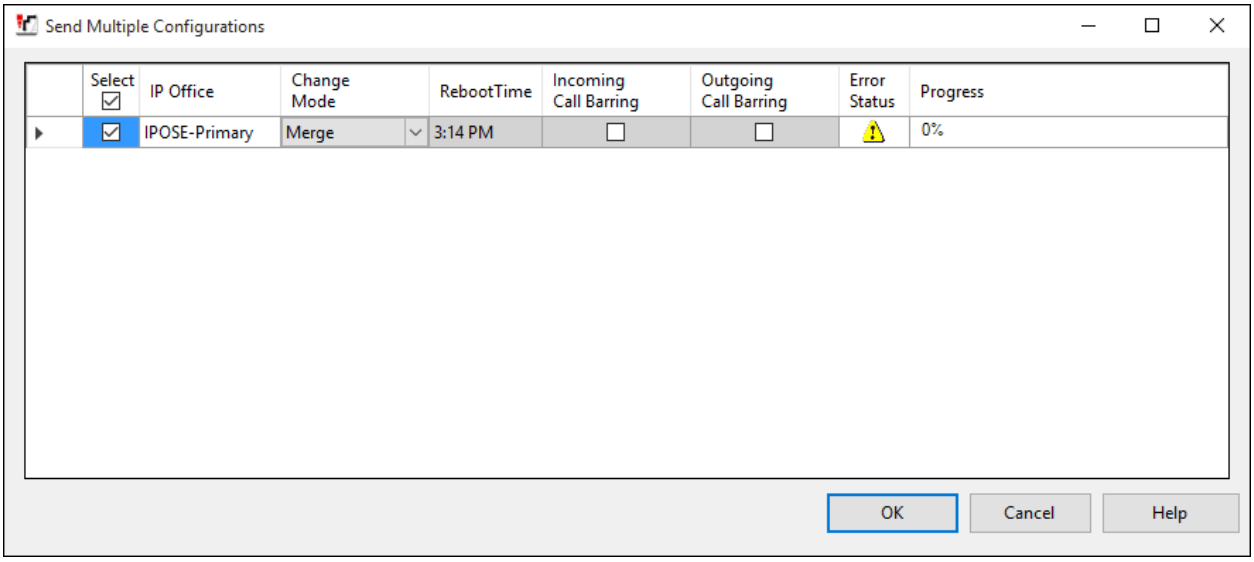
5.11. Save Configuration

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 for the **Change Mode**, 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.



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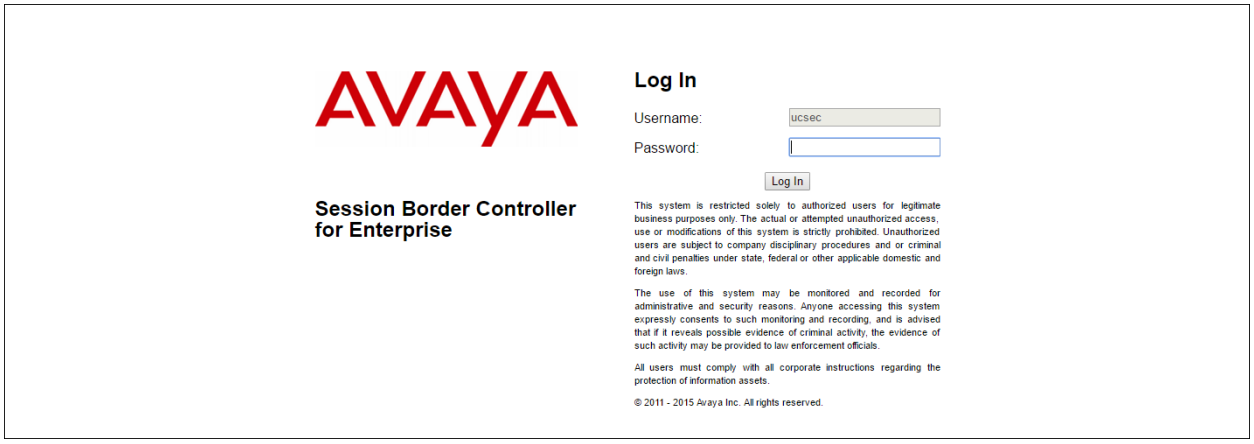
6. Configure Avaya Session Border Controller for Enterprise

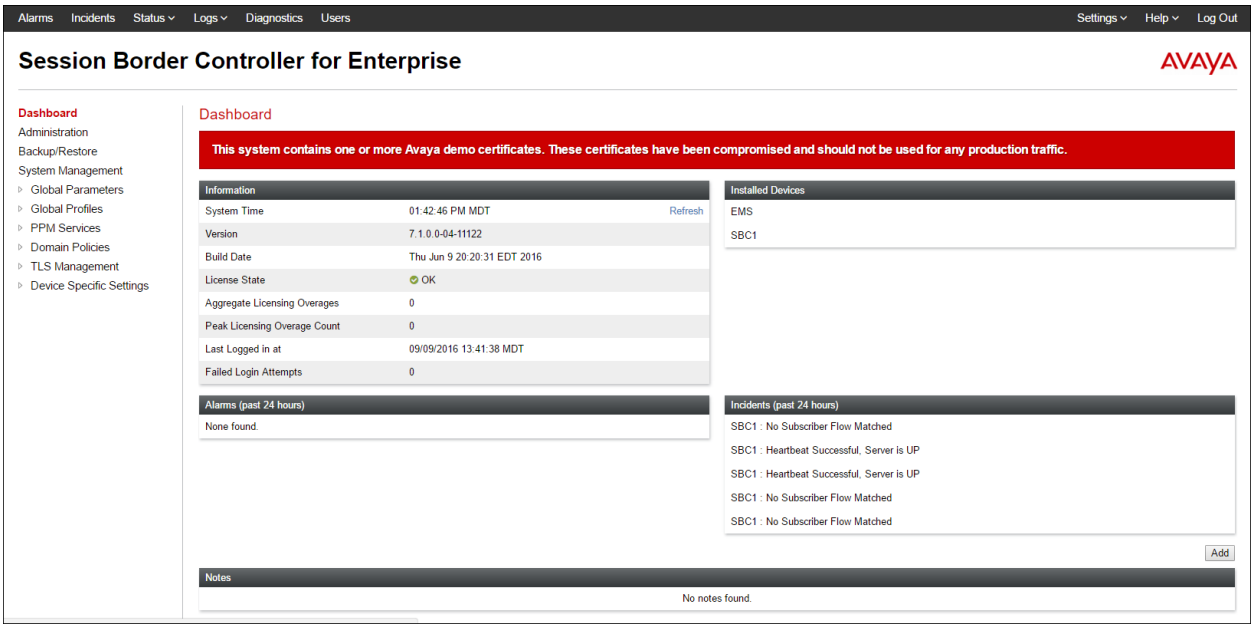
This section covers the configuration of the Avaya SBCE. It is assumed that the Avaya SBCE software has already been installed.

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



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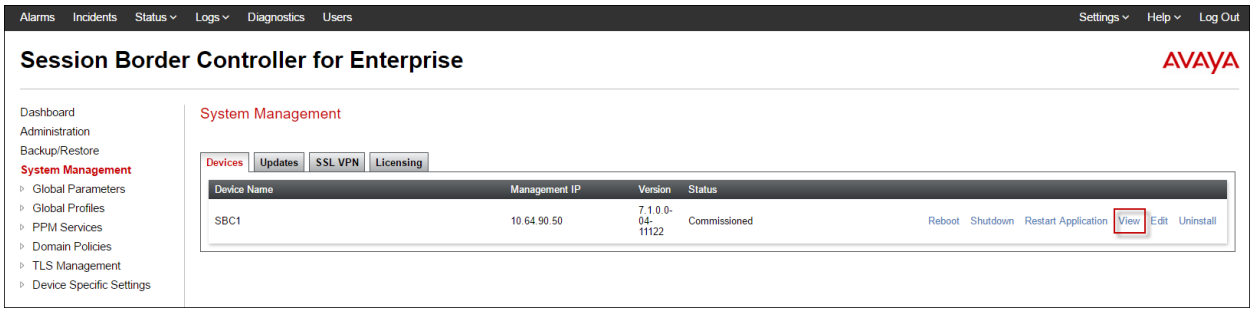
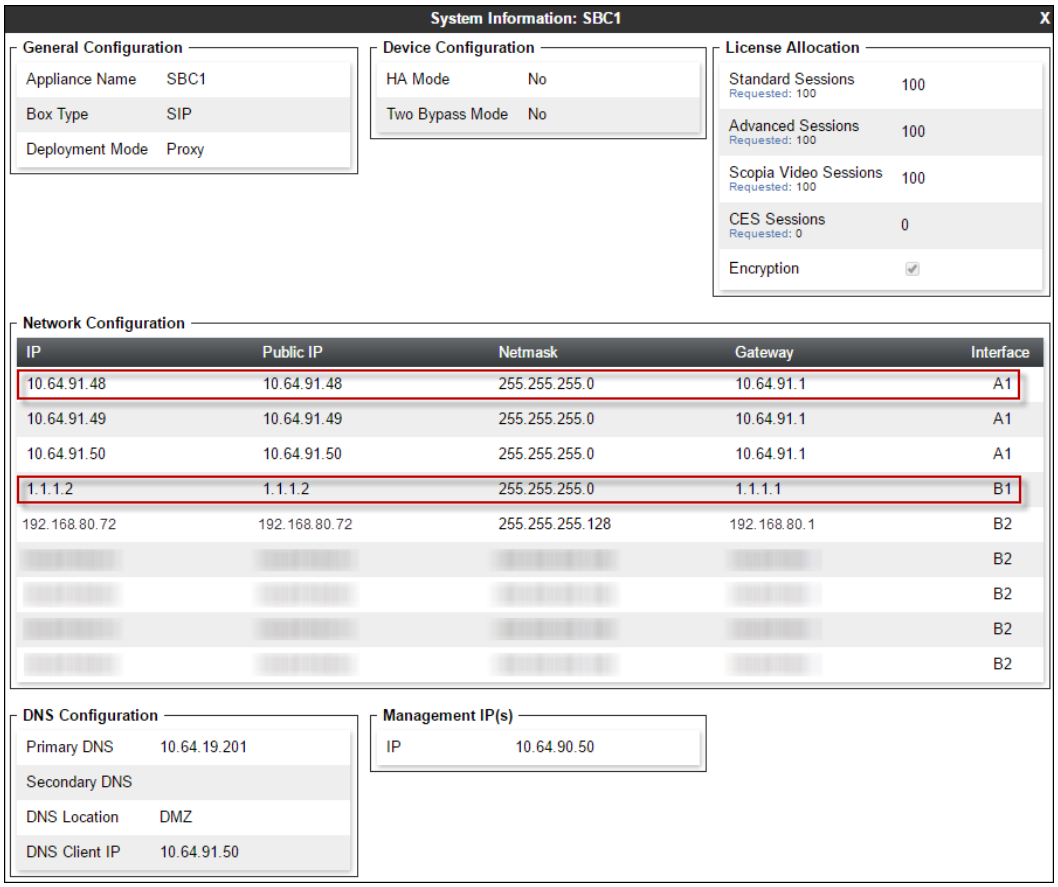
To view system information that was configured during installation, navigate to **System Management**. A list of installed devices is shown in the right pane. In the case of the sample configuration, a single device named "**SBC1**" is shown. To view the configuration of this device, click **View** as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



The **System Information** screen shows the **Network Configuration**, **DNS Configuration** and **Management IP(s)** information provided during installation and corresponds to **Figure 1**. The highlighted **A1** and **A2** IP addresses are the ones relevant to the configuration of the SIP trunk to Verizon. Other IP addresses assigned to these interfaces and interface **B1** on the screen below are used to support remote workers and are not the focus of these Application Notes.



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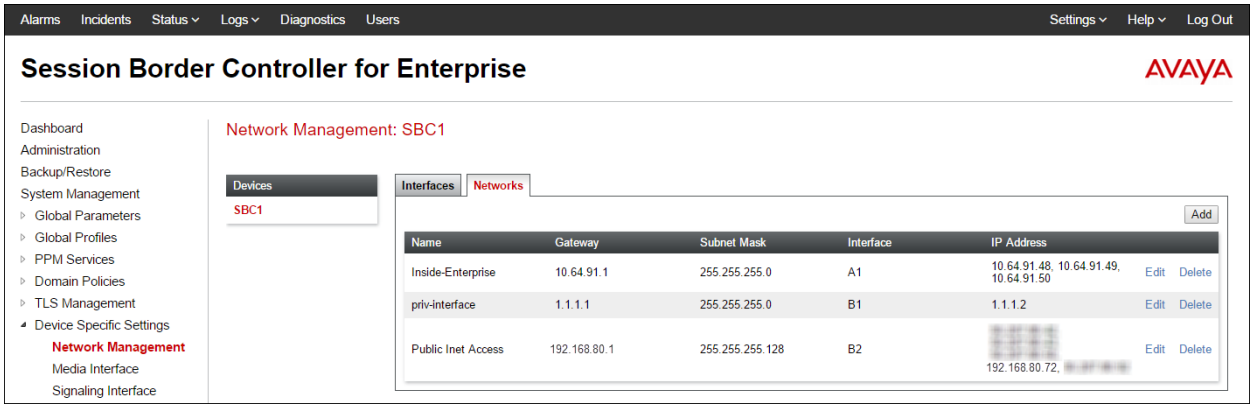
6.1. Network Management

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 public facing interface used for remote workers is **B2**.

The following Avaya SBCE IP addresses and associated interfaces were used in the sample configuration:

- **A1**: **10.64.91.48** – IP address configured for Verizon Business IPCC VoIP Inbound Service to IP Office.
- **A1: 10.64.91.49** – IP address configured for Remote Workers access to the enterprise private network. This address is relevant to the Remote Workers functionality and is not discussed in this document.
- **A1**: **10.64.91.50** – IP address configured for Verizon Business IP Trunk SIP Trunk Service to IP Office. This address is used to connect IP Office to the Verizon Business IP Trunk service as illustrated in [VZBIPT-IPO10SBC].
- **B1: 1.1.1.2** – IP address configured for the Verizon Private IP service. This address is known to Verizon and is associated with the FQDN *adevc.avaya.globalipcom.com.*
- **B2:192.168.80.72** - IP address configured for Remote Worker traffic. Remote Worker SIP endpoints will use this "public" address to established connections to the IP Office through the Avaya SBCE for registration and telephony functions. This address is relevant to the Remote Workers functionality and is not discussed in this document.



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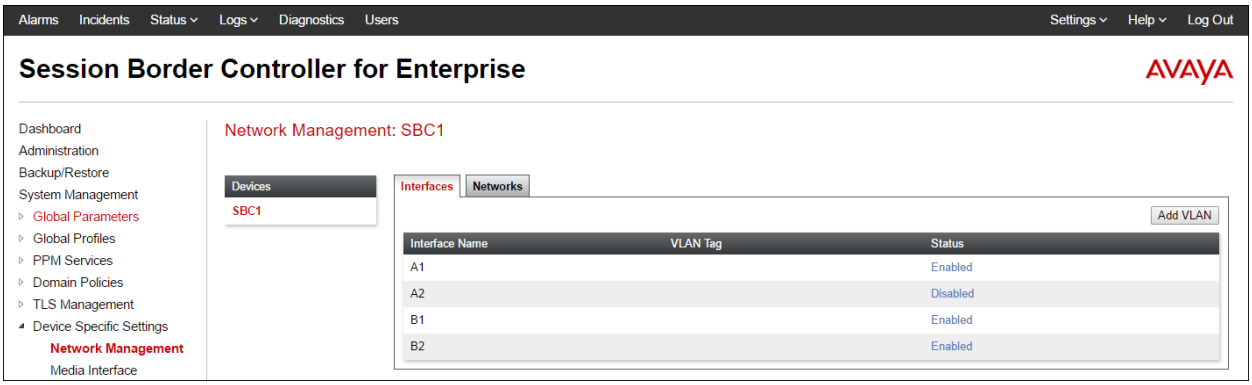
The following screen shows interface **A1**, **B1**, and **B2** are **Enabled**. To enable an interface click the corresponding **Disabled** Status link to change it to **Enabled**.



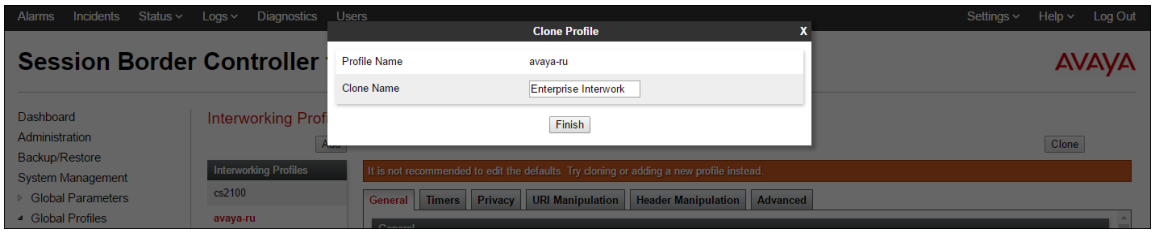
6.2. Server Interworking Profile

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.

6.2.1. 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 **Finish** to continue.



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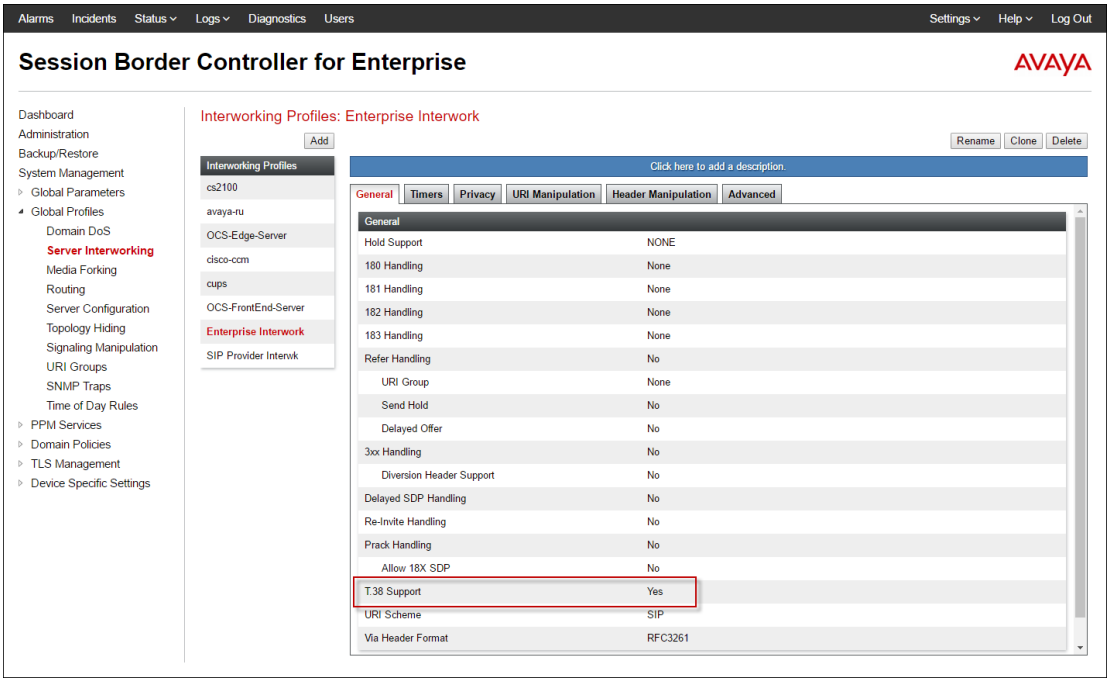
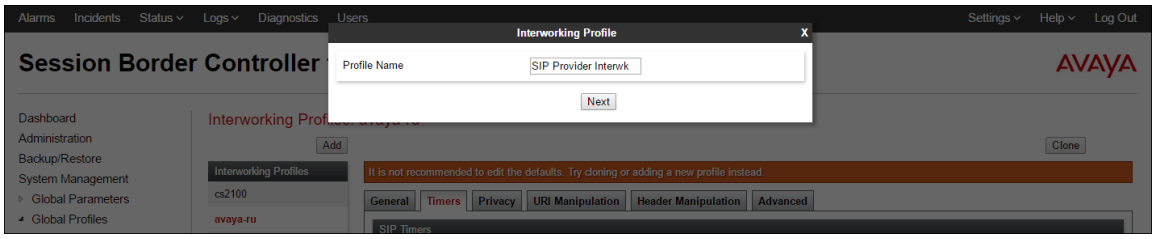
The following screen shows the "**Enterprise Interwork**" profile used in the sample configuration, with **T.38 Support** set to "**Yes**". To modify the profile, scroll down to the bottom of the screen and click **Edit.** Select the **T.38 Support** parameter and then click **Next** and then **Finish**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



6.2.2. 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**.



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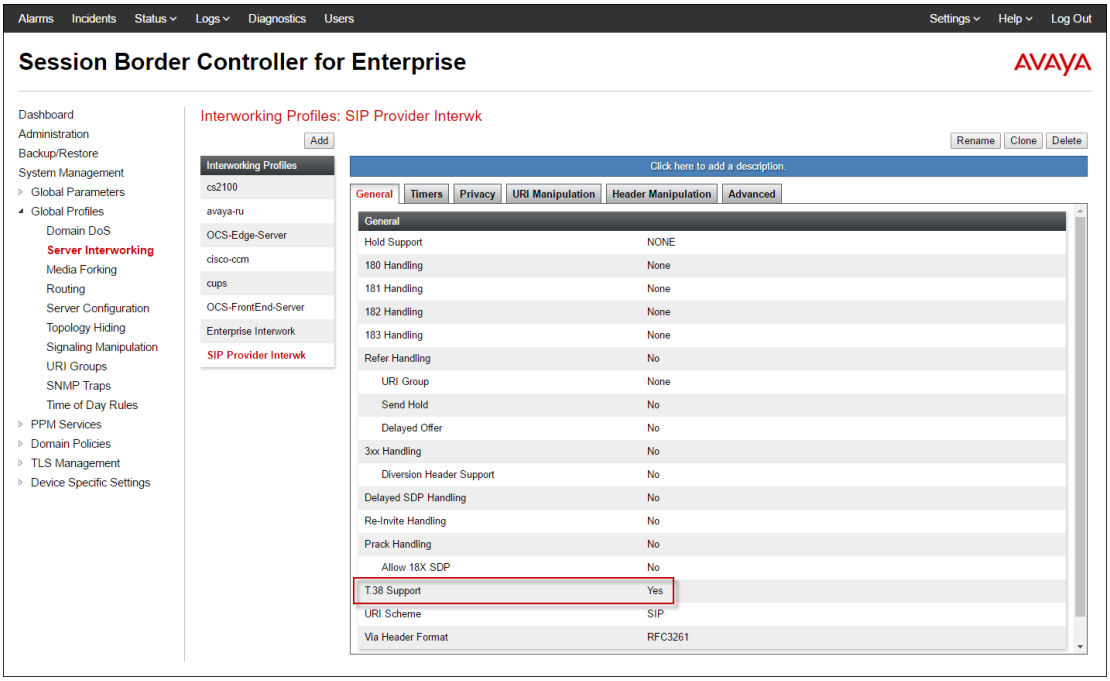
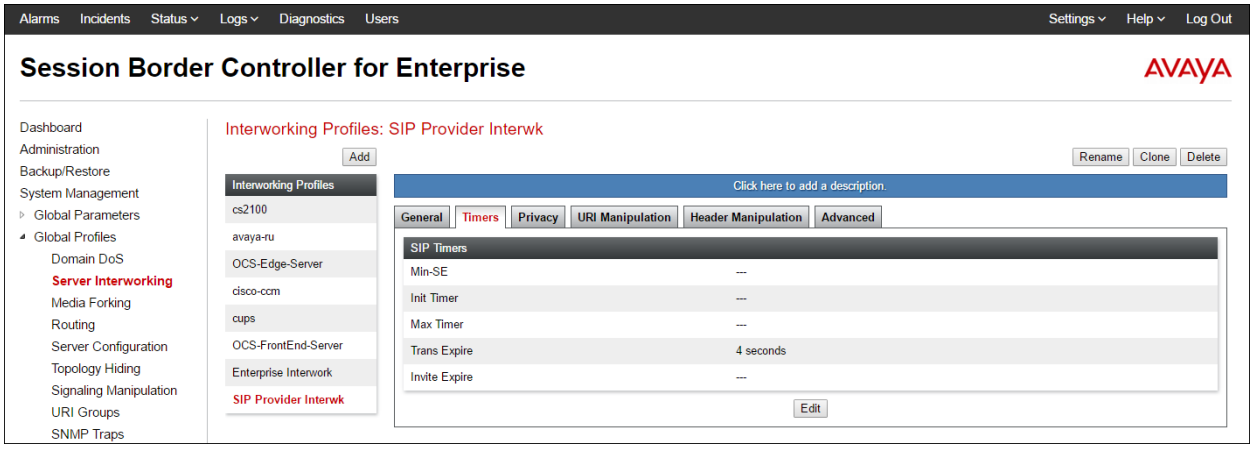
The following screens show the "**SIP Provider Interwrk**" profile used in the sample configuration. On the **General** tab, default values are used with the exception of **T.38 Support** set to "**Yes**".



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



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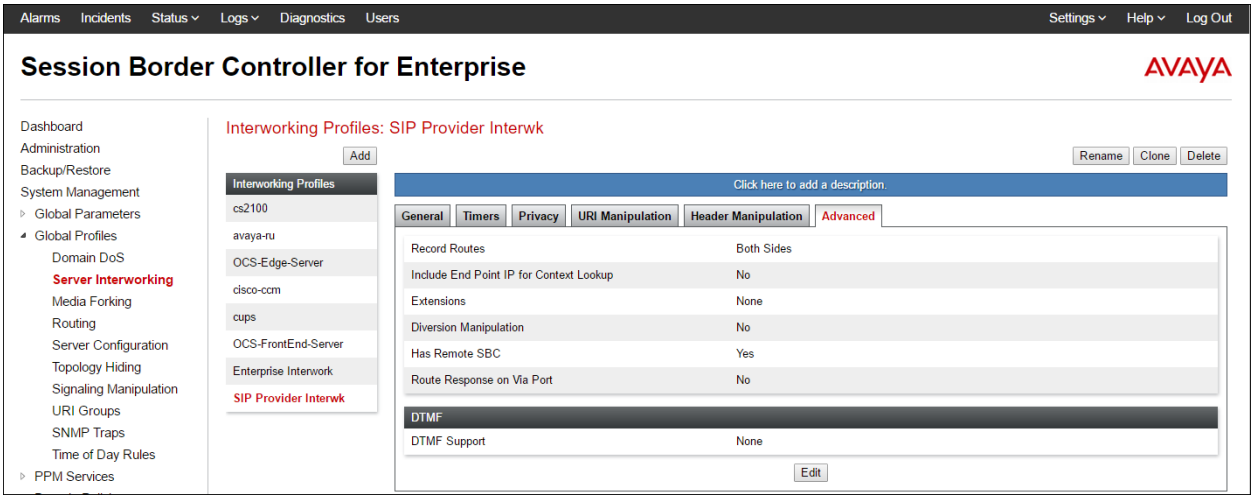
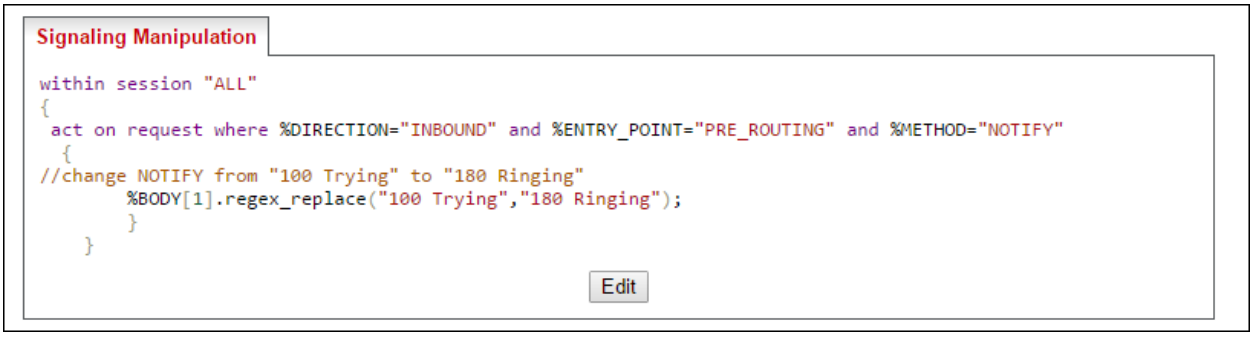
Click **Next** to accept default parameters for the **Privacy**, **URI Manipulation**, and **Header Manipulation** tabs (not shown) and advance to the Advanced area. **Record Routes** is set to "**Both Sides**". Default values can be used for all other fields.



6.3. Signaling Manipulation

For the compliance test, signaling manipulation script "**VzIPCC script**" was created for the Verizon Business IPCC server. The script contains a manipulation to convert the "100 Trying" from within the SIP NOTIFY message to "180 Ringing". This is necessary for blind call transfers using the REFER method. See **Section 2.2** for details. The script instructions to perform these manipulations are shown below and in **Appendix A**.



6.4. Server Configuration

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

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

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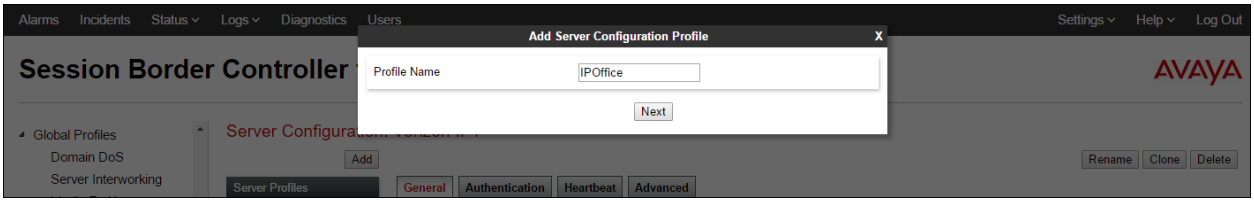
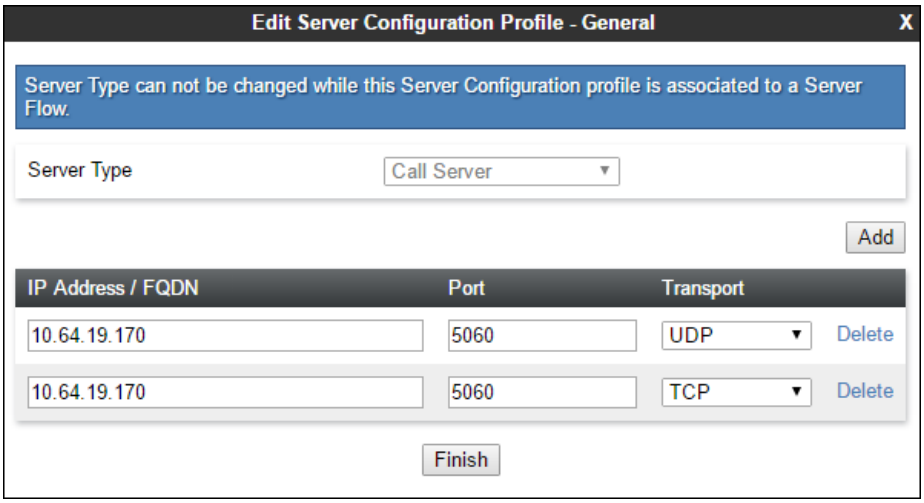
6.4.1. Server Configuration – IP Office

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 "**IPOffice**". In the **General** parameters, the **Server Type** is set to "**Call Server**". In the **IP Address / FQDN** field, the IP Address of the Primary Server LAN 1 interface in the sample configuration is entered. This IP address is 10.64.19.170. Under **Port**, "**5060**" is entered, and the **Transport** parameter is set to "**UDP**". If remote workers will also be configured using the same Avaya SBCE, click the **Add** button to create a second connection to IP Office using 5060 with the TCP transport. Remote worker SIP traffic will use TCP between Avaya SBCE and IP Office. If adding the profile, click **Next** (not shown) to proceed. If editing an existing profile, click **Finish**.



Default values can be used on the **Authentication** tab, click **Next** (not shown) to proceed to the **Heartbeats** tab. The Avaya SBCE can be configured to source "heartbeats" in the form of PINGs or SIP OPTIONS towards IP Office. When remote workers are configured, IP Office may not respond to SIP OPTIONS from the SBCE IP address designated for remote workers. Therefore PING will be used instead.

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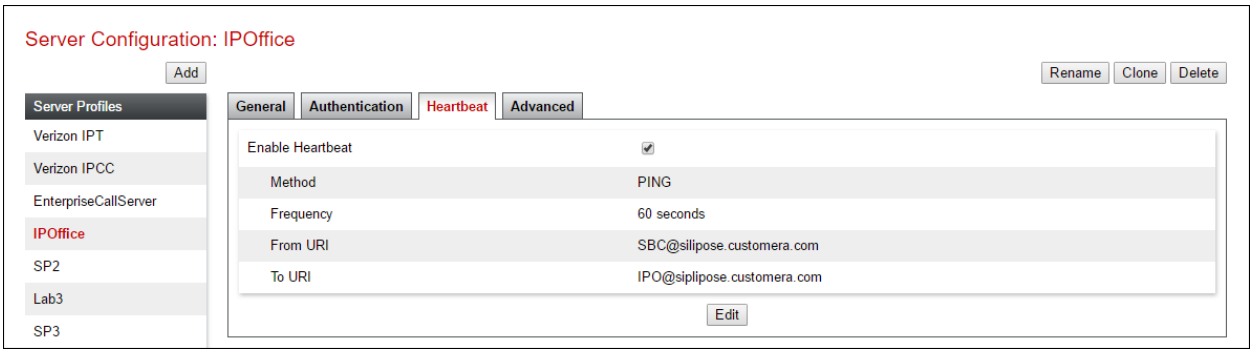
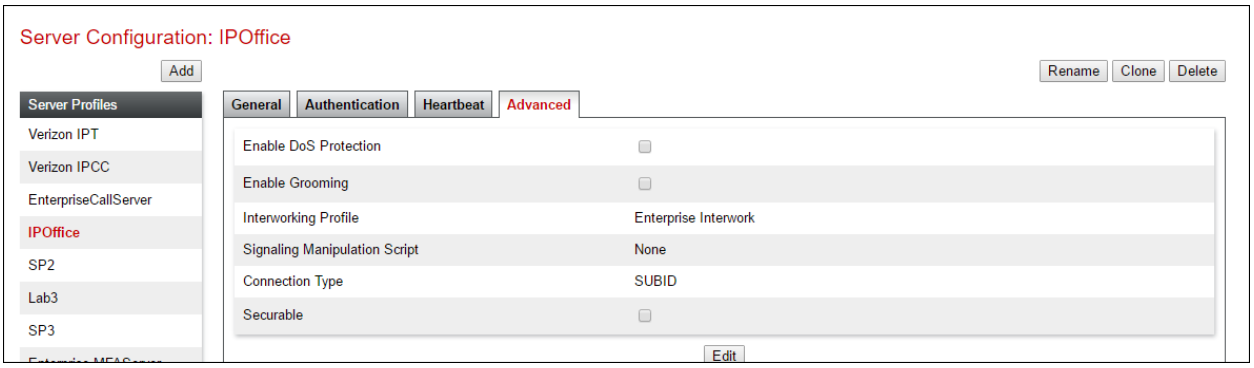
Select "**PING**" from the **Method** drop-down menu. Select the desired frequency that the SBCE will source PINGs towards IP Office.

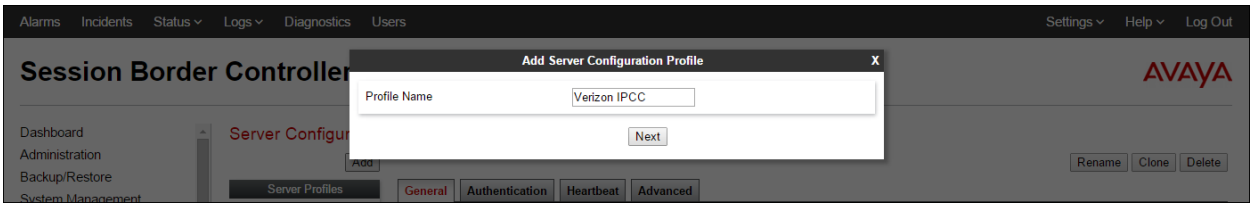


On the **Advanced** tab, the **Interworking Profile** is set to "**Enterprise Interwork**" created in **Section 6.2.1** for IP Office. In the sample configuration, this server configuration is also used for remote workers via TCP. **Enable Grooming** allows the same TCP connection to be used for all SIP messages from this device. However, IP Office uses different TCP connections to each endpoint, therefore **Enable Grooming** is unchecked.



6.4.2. 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**.



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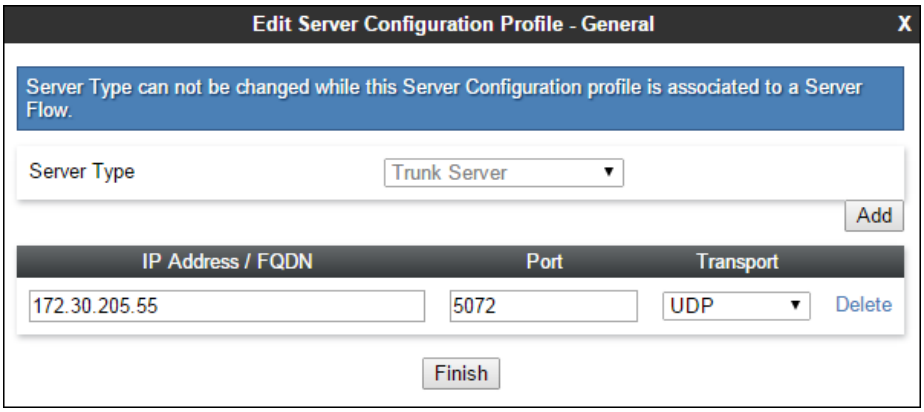
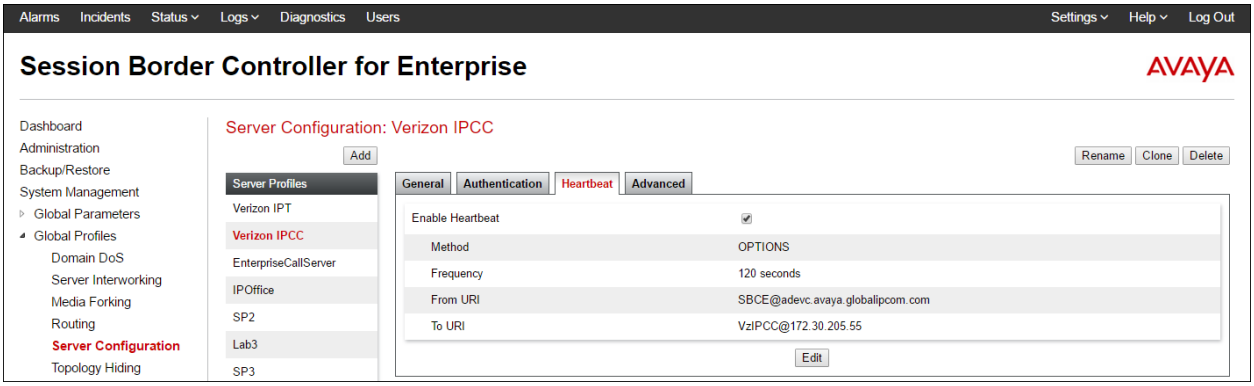
The following screens illustrate the Server Configuration for the Profile name "**Verizon IPCC**". In the **General** parameters, the **Server Type** is set to "**Trunk Server**". In the **IP Address / FQDN** field, the Verizon-provided IP address is entered. This is "**172.30.205.55**". Under **Port**, "**5072**" is entered, and the **Transport** parameter is set to "**UDP**". If adding the profile, click **Next** (not shown) to proceed. If editing an existing profile, click **Finish**.



Default values can be used on the **Authentication** tab, click **Next** (not shown) to proceed to the **Heartbeats** tab. The Avaya SBCE can be configured to source "heartbeats" in the form of SIP OPTIONS towards Verizon. This configuration is optional. Independent of whether the Avaya SBCE is configured to source SIP OPTIONS towards Verizon, Verizon will receive OPTIONS from the IP Office site as a result of the **Check OOS** parameter being enabled on IP Office (see **Section 5.4.3**). When IP Office sends SIP OPTIONS to the inside IP Address of the Avaya SBCE, the Avaya SBCE will send SIP OPTIONS to Verizon Business. When Verizon Business responds, the Avaya SBCE will pass the response to IP Office.

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



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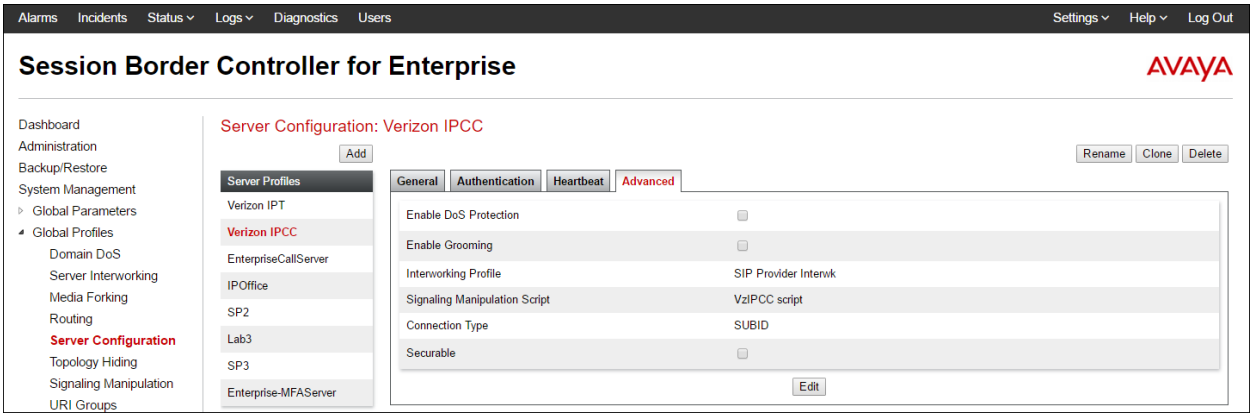
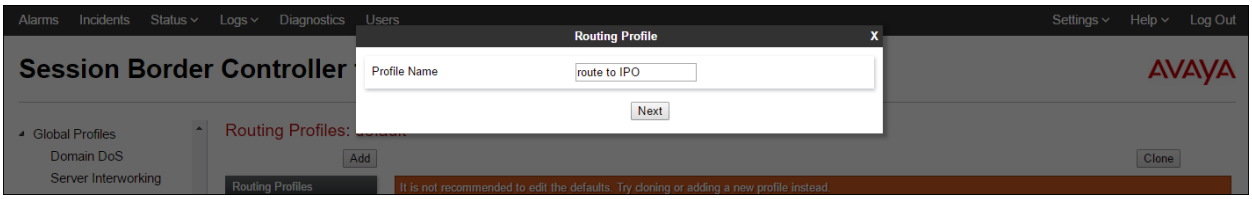
On the **Advanced** tab, **Enable Grooming** is not used for UDP connections and left unchecked. The **Interworking Profile** is set to "**SIP Provider Interwk**" created in **Section 6.2.2** for Verizon. The **Signaling Manipulation Script** is set to the script created for Verizon Business IPCC in **Section 0**.



6.5. Routing Profile

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

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



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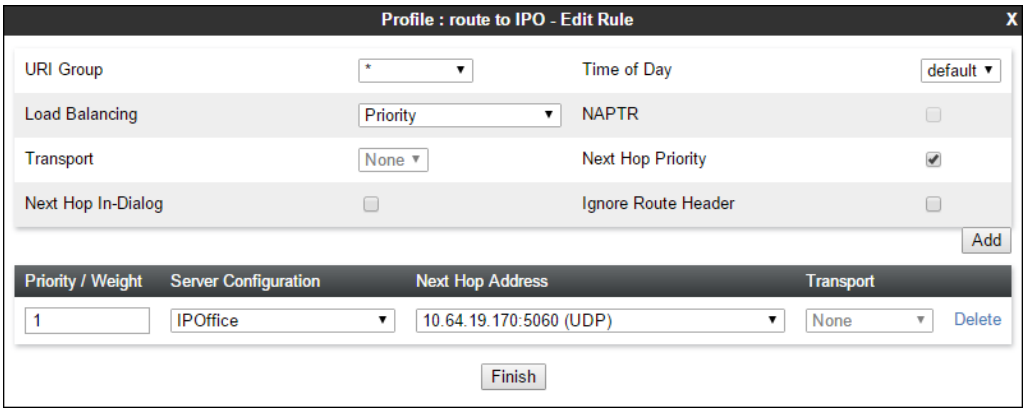
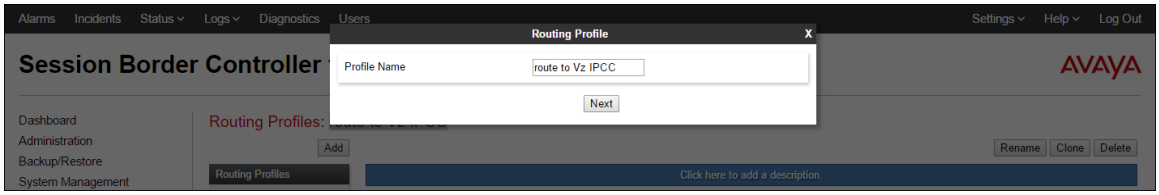
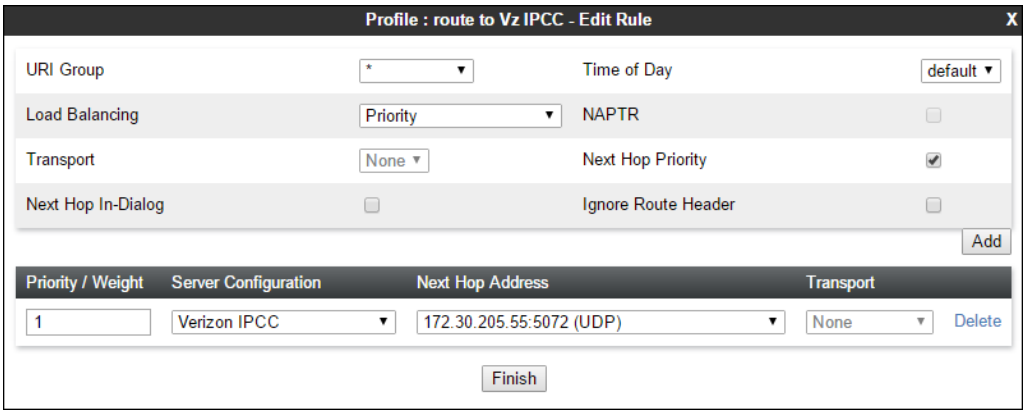
The following screen shows the Routing Profile "**route to IPO**" created in the sample configuration. The parameters in the top portion of the profile are left at their default settings. The **Priority / Weight** parameter is set to "**1**", and the IP Office **Server Configuration**, created in **Section 6.4.1**, is selected from the drop-down menu. The **Next Hop Address** is automatically selected with one of the values from the IP Office Server Configuration, and **Transport** becomes greyed out. Select the UDP entry from the drop-down menu for the **Next Hop Address**, and select **Finish**.



Similarly add a Routing Profile to the Verizon Business IPCC service.



The following screen shows the Routing Profile "**route to Vz IPCC**" created in the sample configuration. The parameters in the top portion of the profile are left at their default settings. The **Priority / Weight** parameter is set to "**1**", and the Verizon **Server Configuration**, created in **Section 6.4.2**, is selected from the drop-down menu. The **Next Hop Address** is automatically selected with the values from the Server Configuration, and the Transport becomes greyed out. Click **Finish**.



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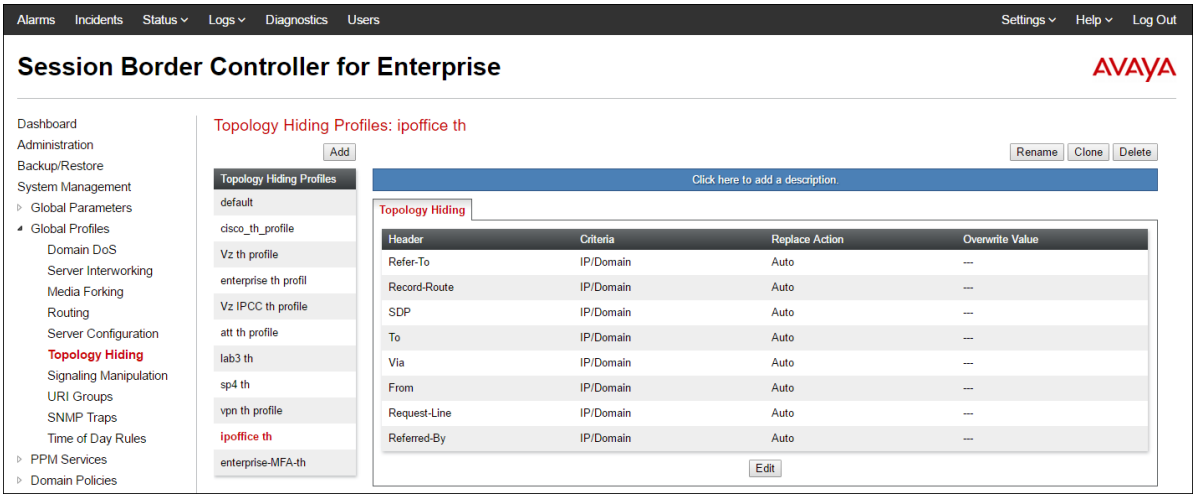
6.6. Topology Hiding Profile

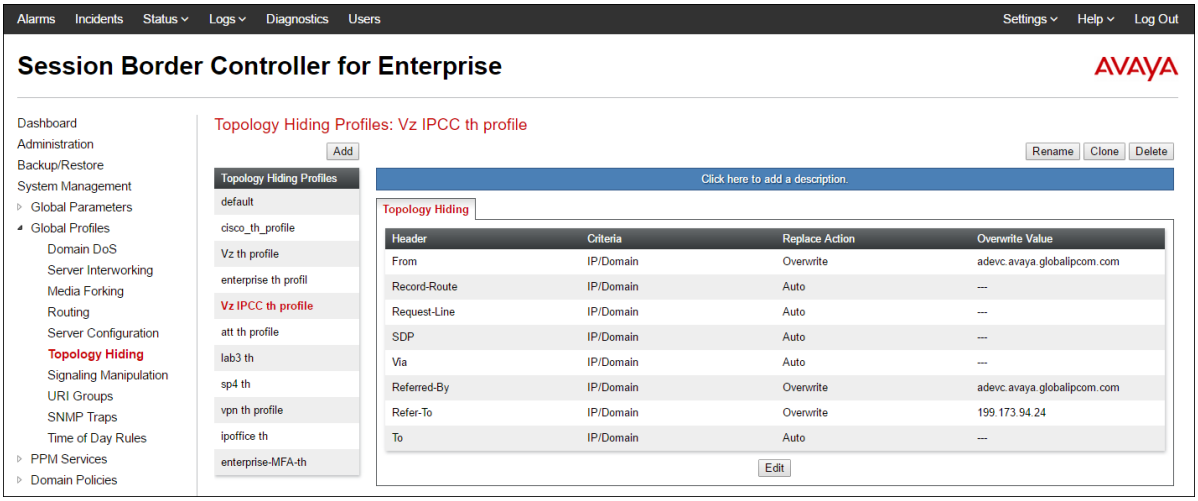
Click the **Add** button to add a new profile, or select an existing topology hiding profile to edit. In the sample configuration, the "**default**" profile was cloned for IP Office, and cloned and modified for Verizon.

In the **Replace Action** column an action of **Auto** will replace the header field with the IP address of the Avaya SBCE interface and the **Overwrite** will use the value in the **Overwrite Value**.

In the example shown, "**ipoffice th**" was cloned from the default profile and will later be applied in the direction of IP Office.



In the example shown, "**Vz th profile**" was cloned from the default profile and will later be applied in the direction of Verizon. **Overwrite** is selected for the From and Referred-By headers and domain of "**adevc.avaya.globalipcom.com**" is inserted. This is the IP Office domain known to Verizon Business IPCC service. The Refer-To header's domain is overwritten with the IP address presented in the original INVITE from Verizon's IP-IVR service. If the IP-IVR service is not used, the Refer-To header can retain the default **Replace Action** of "**Auto**".



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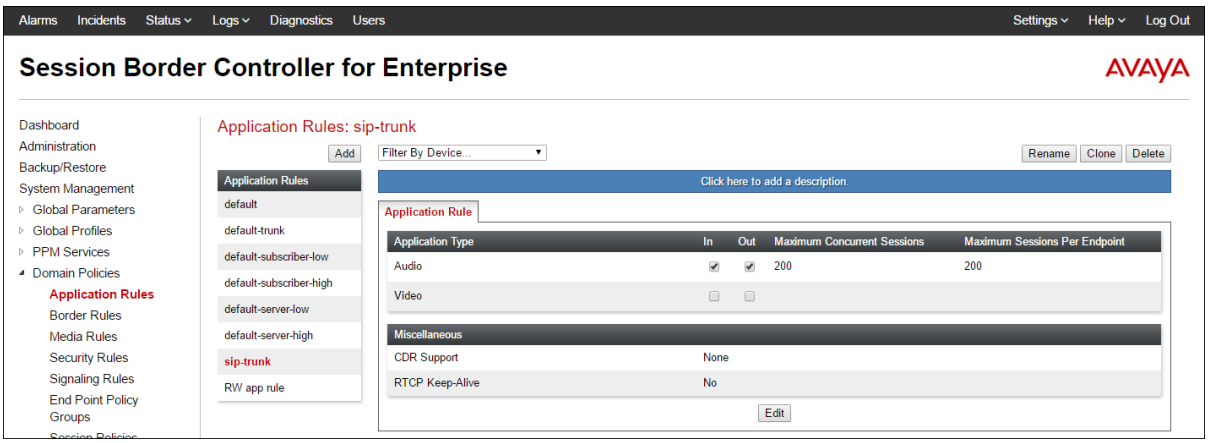
6.7. Application Rule

Application Rules define which types of SIP-based Unified Communications (UC) applications the Avaya SBCE security device will protect: voice, video, and/or Instant Messaging (IM). In addition, 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. Click the **Add** button to add a new profile, or select an existing topology hiding profile to edit. In the sample configuration, the "**sip-trunk**" profile was created for IP Office and Verizon. In an actual customer installation, set the **Maximum Concurrent Sessions** for the **Audio** application to a value slightly larger than the licensed sessions. For example, if licensed for 150 session set the values to "**200**". The **Maximum Session Per Endpoint** should match the **Maximum Concurrent Sessions**.



6.8. Media Rule

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

Select **Domain Policies → Media Rules** from the left-side menu as shown below. In the sample configuration, the default media rule "**default-low-med**" was cloned for IP Office, "**IPO SIPTrk Med Rule**", and Verizon Business IP Trunk, "**Vz SIPTrk Med Rule**". With the "**default-low-med"** rule chosen, click **Clone**. Enter a descriptive name for the new rule and click **Finish** (not shown).

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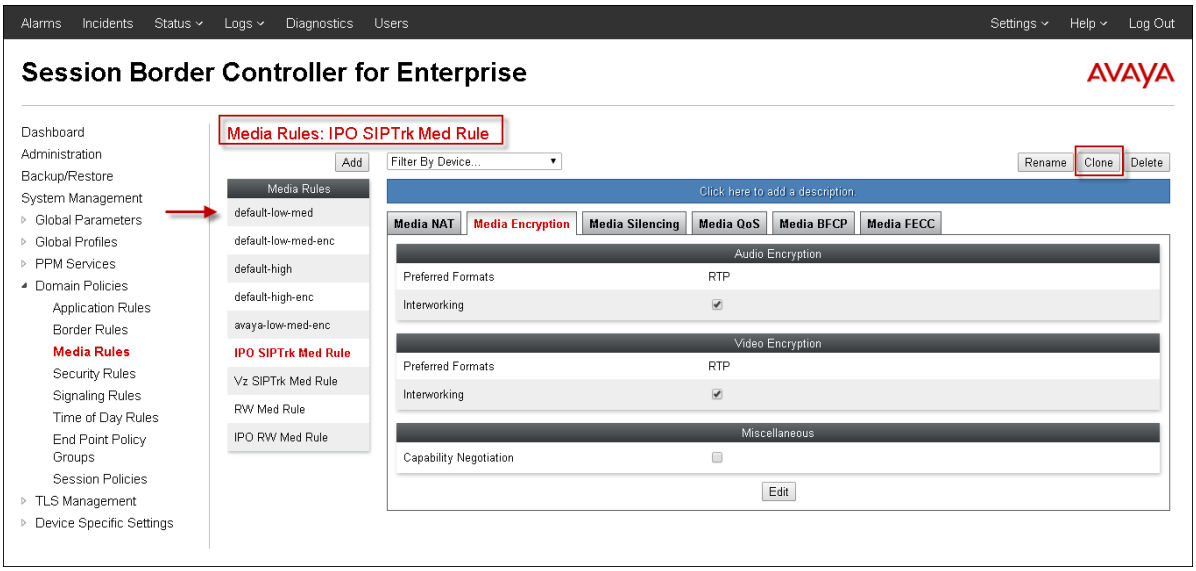
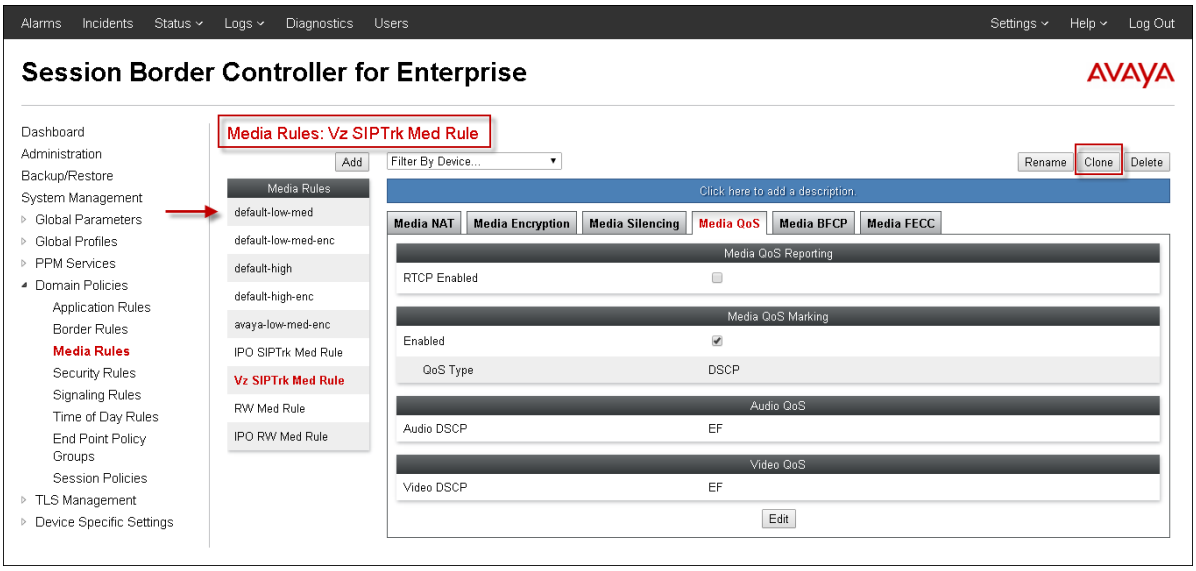
In the sample configuration, signaling rule "**IPO SIPTrk Sig Rule**" was used for IP Office as shown below.



The Verizon Media Rule, "**Vz SIPTrk Med Rule**" with the DSCP values "**EF**" for expedited forwarding (default value) for **Media QoS**.



6.9. Signaling Rule

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

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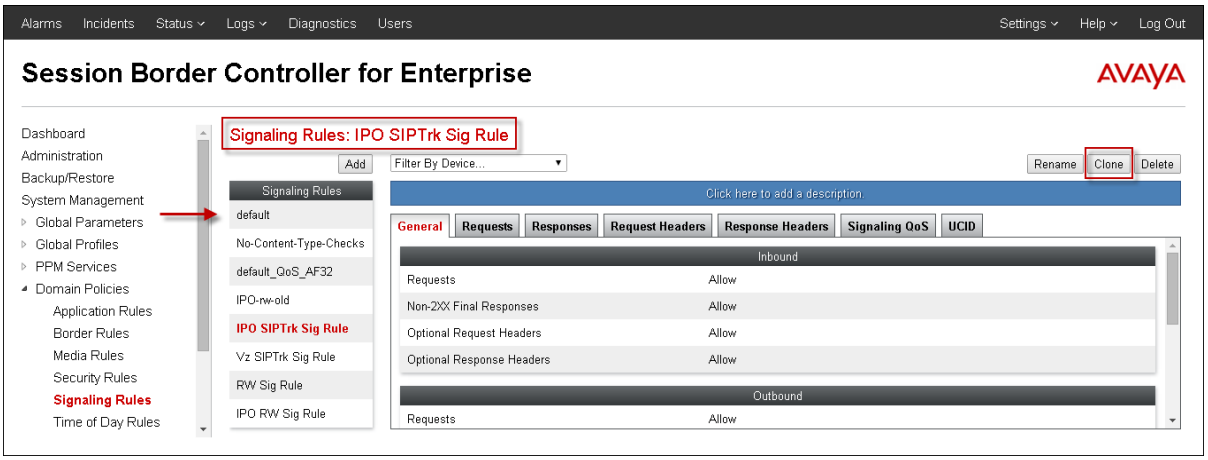
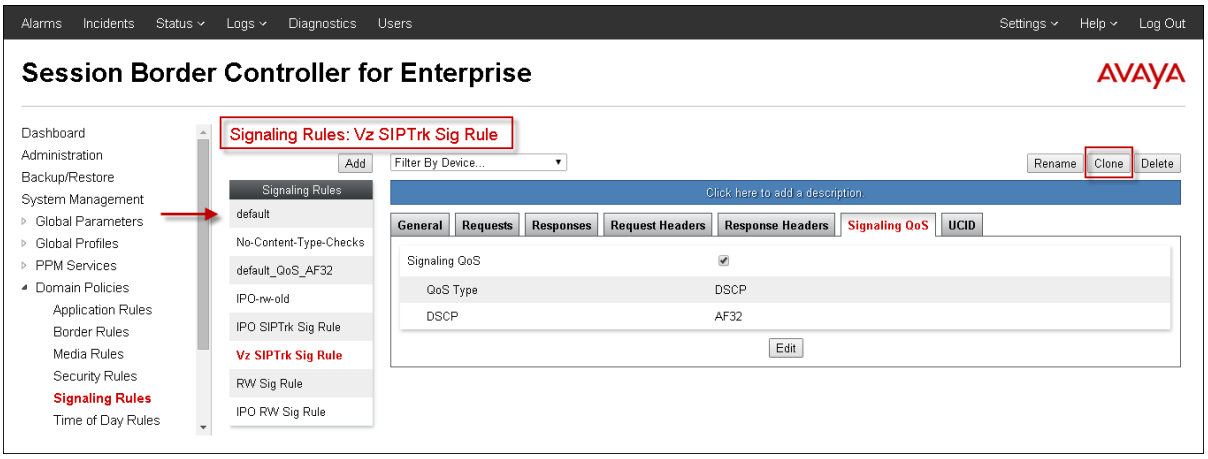
Clone and modify the **default** signaling rule to add the proper quality of service to the SIP signaling. To clone a signaling rule, navigate to **Domain Policies → Signaling Rules**. With the "**default**" rule chosen, click **Clone**. Enter a descriptive name for the new rule and click **Finish** (not shown). In the sample configuration, signaling rule "**IPO SIPTrk Sig Rule**" is unchanged from the default rule.



Signaling rule "**Vz SIPTrk Sig Rule**" was also cloned from the default rule and used for Verizon. The DSCP value "**AF32**" for assured forwarding is changed from the default settings for **Signaling QoS** as shown below.



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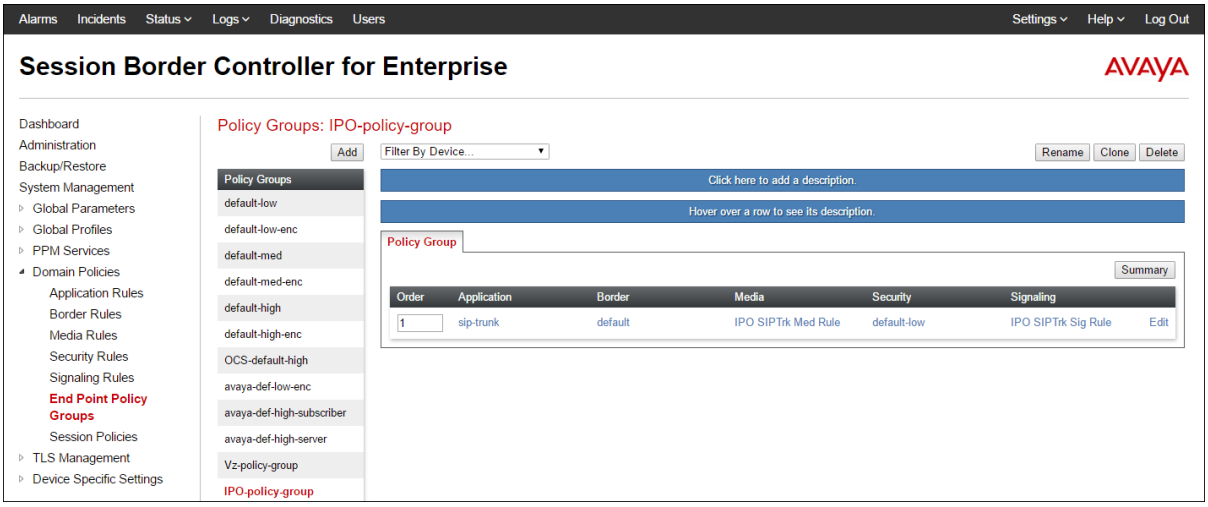
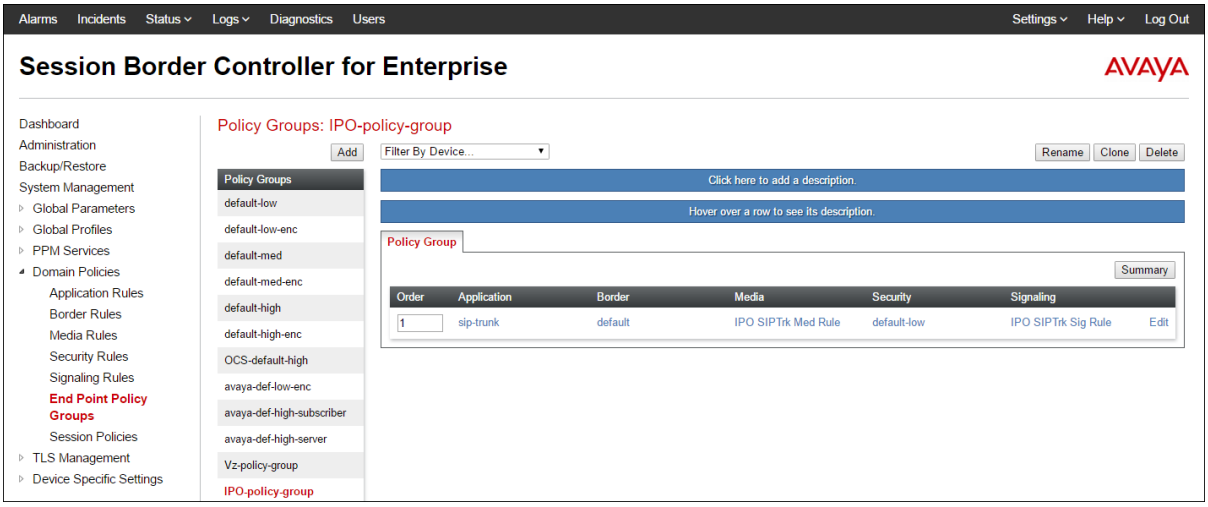
6.10. Endpoint Policy Groups

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 6.13**.

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 "**IPO-policy-group**" created for IP Office. The details of the non-default rules chosen are shown in previous sections.



The following screen shows the "**Vz-policy-group**" created for Verizon Business IPCC service. The details of the non-default rules chosen are shown in previous sections.



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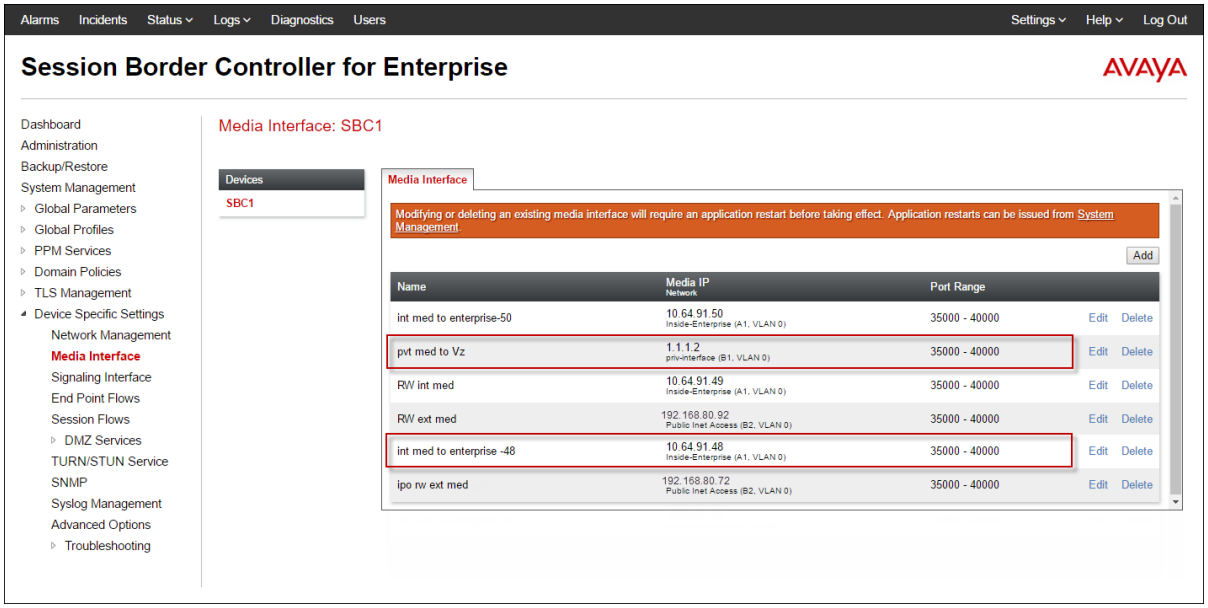
6.11. Media Interface

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



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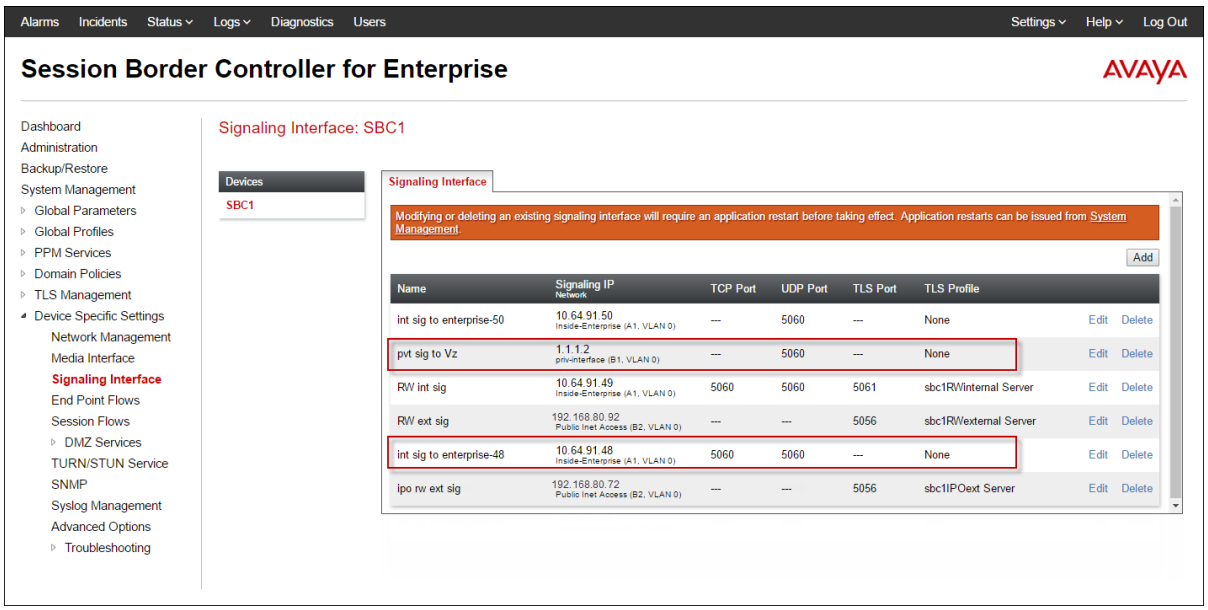
6.12. Signaling Interface

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



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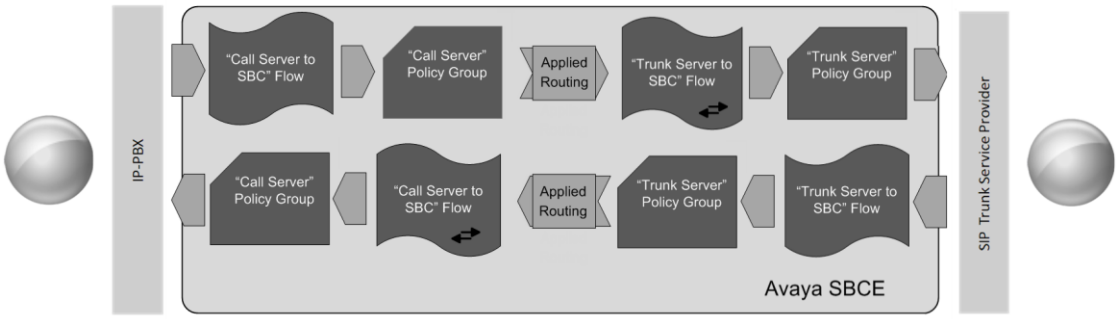
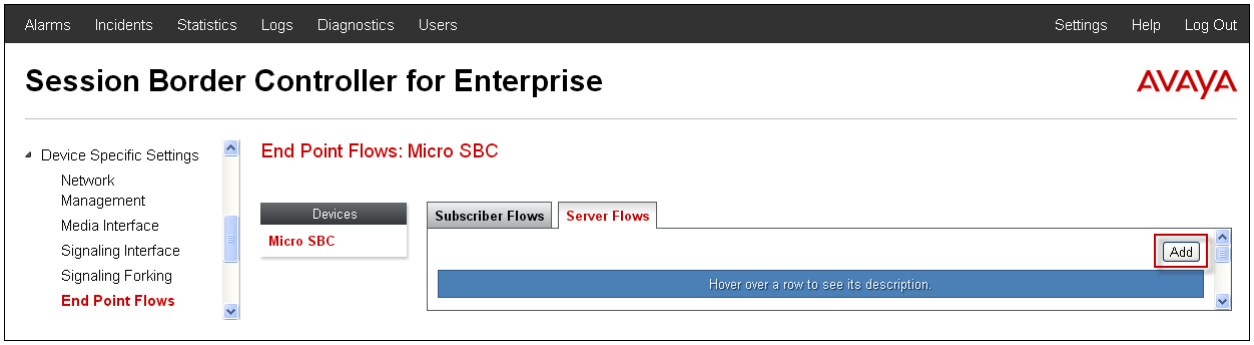
6.13. End Point Flows - Server Flow

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



Create a Server Flow for IP Office and Verizon Business IPCC 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.



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

View Flow: Verizon IPCC Flow

Criteria		Profile	
Flow Name	Verizon IPCC Flow	Signaling Interface	pvt sig to Vz
Server Configuration	Verizon IPCC	Media Interface	pvt med to Vz
URI Group	*	End Point Policy Group	Vz-policy-group
Transport	*	Routing Profile	route to IPO
Remote Subnet	*	Topology Hiding Profile	Vz IPCC th profile
Received Interface	int sig to enterprise-48	Signaling Manipulation Script	None
		Remote Branch Office	Any

Once again, select the **Server Flows** tab and click **Add**. The following screen shows the flow named "**IPO SIP Trunk to Vz IPCC**" viewed from the sample configuration. This flow uses the interfaces, polices, and profiles defined in previous sections. In addition, the **Remote Subnet** is configured with the Verizon-provided IP address for IPCC service, along with "/32", i.e., "**172.30.205.55/32**". Using "/32" for the subnet mask indicates that a single IP address, not a range of addresses, is the criteria to match for this particular IP Office flow.

View Flow: IPO SIP Trunk to Vz IPCC

Criteria		Profile	
Flow Name	IPO SIP Trunk to Vz IPCC	Signaling Interface	int sig to enterprise-48
Server Configuration	IPOffice	Media Interface	int med to enterprise -48
URI Group	*	End Point Policy Group	IPO-policy-group
Transport	*	Routing Profile	route to Vz IPCC
Remote Subnet	172.30.205.55/32	Topology Hiding Profile	ipoffice th
Received Interface	pvt sig to Vz	Signaling Manipulation Script	None
		Remote Branch Office	Any

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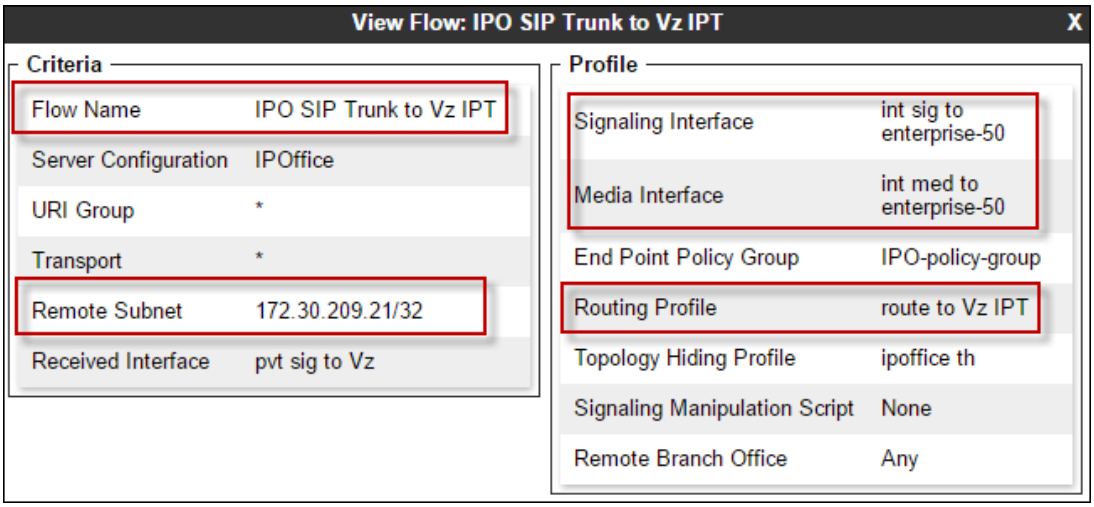
For illustration, the following screen shows the flow named "**IPO SIP Trunk to Vz IPT**" viewed from the sample configuration. This flow was originally created for use with Verizon Business IP Trunk service as shown in [VZBIPT-IPO10SBC]. Similar to the "**IPO SIP Trunk to Vz IPCC**" flow shown above, the **Remote Subnet** is also configured. However, it is configured with the Verizon-provided IP address for IP Trunk service, rather than the IPCC service. This is shown here to illustrate how the Avaya SBCE can send SIP traffic out on different internal signaling and media interfaces based on the **Criteria** specified in the flow. In the sample configuration provided, a Verizon Business IP Trunk SIP call from IP address 172.30.209.21 will use internal IP address 10.64.91.50 towards IP Office, while a Verizon Business IPCC SIP call from IP address 172.30.205.55 will use internal IP address 10.64.91.48. This will allow IP Office to have separate unique SIP Line configurations to the Avaya SBCE for each Verizon service.



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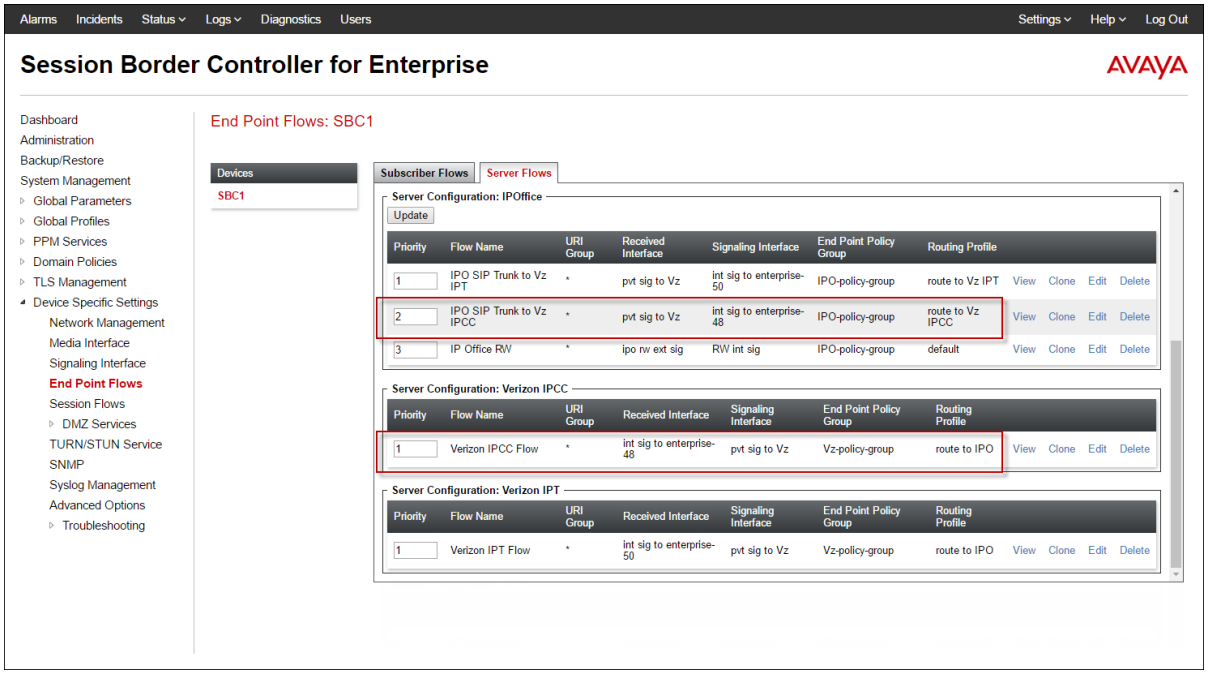
The following screen summarizes the Server Flows configured in the sample configuration. The different flows under the **IP Office** Server Configuration will match based on the **Criteria** of each flow as shown above.



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7. Verizon Business Configuration

Information regarding Verizon Business IP Contact Center service offer can be found by contacting a Verizon Business sales representative, or by visiting http://www.verizonbusiness.com/Products/communications/contact-center/

The configuration described in these Application Notes was located in the Avaya Solutions and Interoperability Lab. The Verizon Business IP Contact Center service was accessed via a Verizon Private IP (PIP) T1 connection as described in Section 1. Verizon Business provided the necessary service provisioning, which included the domain *adevc.avaya.globalipcom.com* for the Avaya IP Office location.

For service provisioning, Verizon will require the customer IP address of the Avaya Session Border Controller for Enterprise. Verizon provided the following information for the compliance testing: the IP address and port used by the Verizon SBC, and the toll-free numbers shown in **Figure 1** and **Table 1**. This information was used to complete the Avaya Session Border Controller for Enterprise shown in **Section 6** and the IP Office configuration shown in **Section 5**.

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8. Verifications

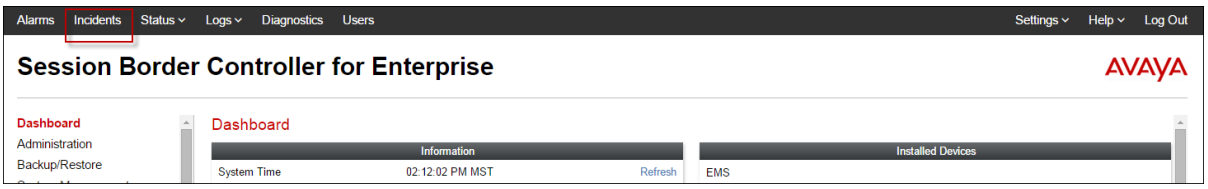
This section provides example verifications of the Avaya configuration with Verizon Business Private IP (PIP) Trunk service.

8.1. Avaya SBCE

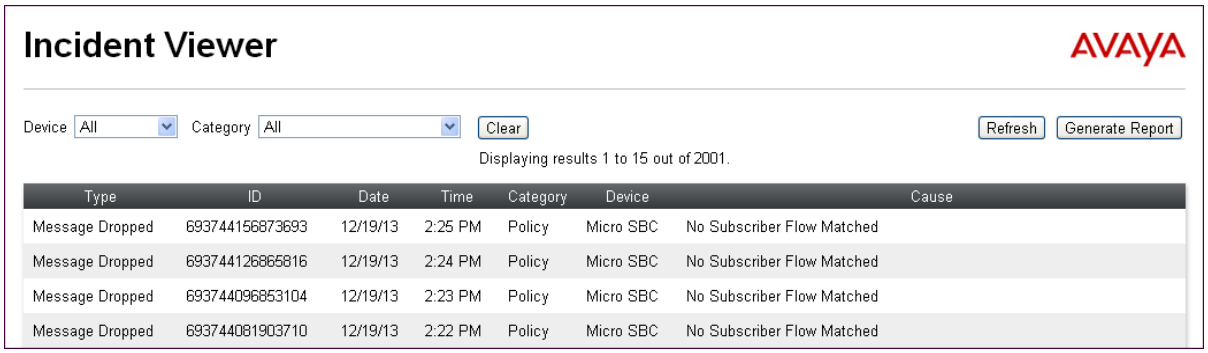
This section provides verification steps that may be performed with the Avaya SBCE.

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. Server Status

The **Server Status** can be access from the Avaya SBCE Dashboard by selecting the **Status** menu, and then **Server Status**.

A pop-up window will appear with the **Status** of "**UP**" for the Verizon Business IPCC Server Profile. The **Server Profile** will only list servers with Server Configuration settings that have Heartbeats enabled, see **Section 6.4.2**.

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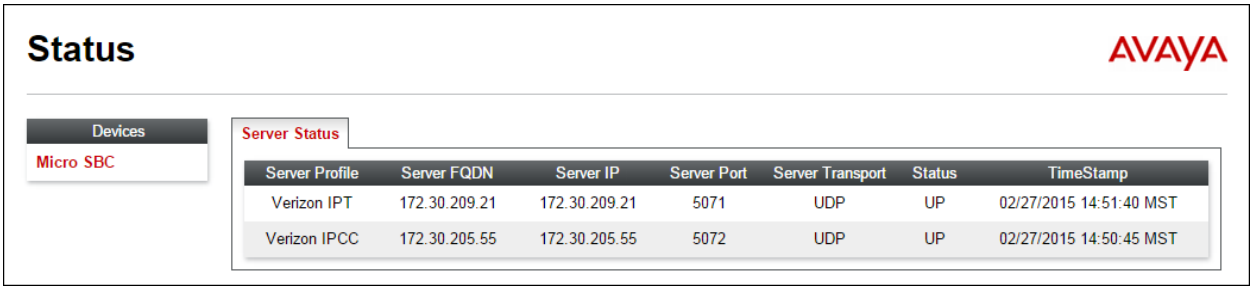
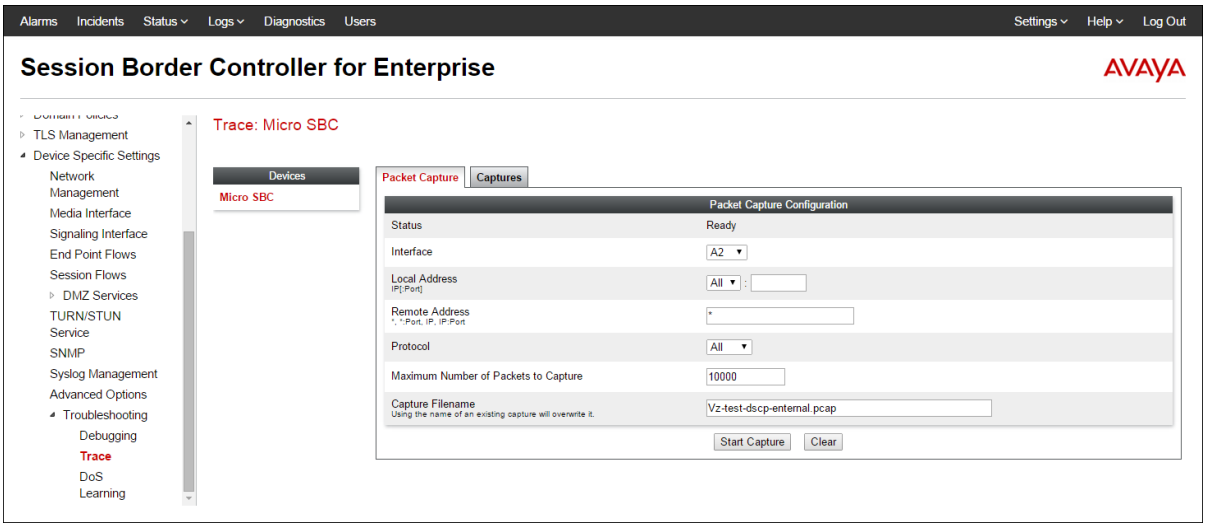
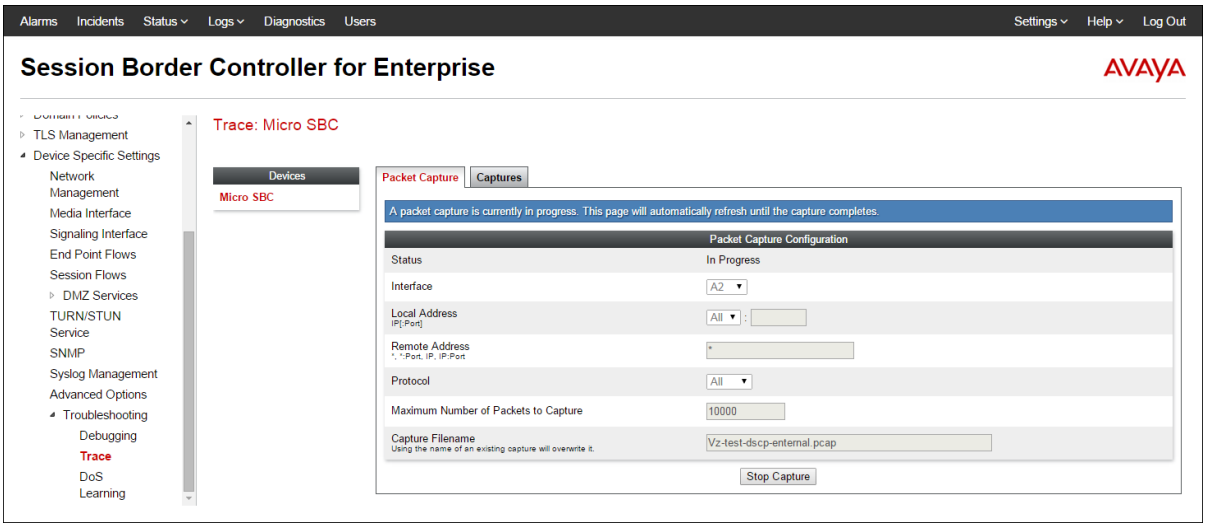


8.1.3. Tracing

To take a call trace, navigate to **Device Specific Settings → Troubleshooting → Trace** and select the **Packet Capture** tab. Then 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.



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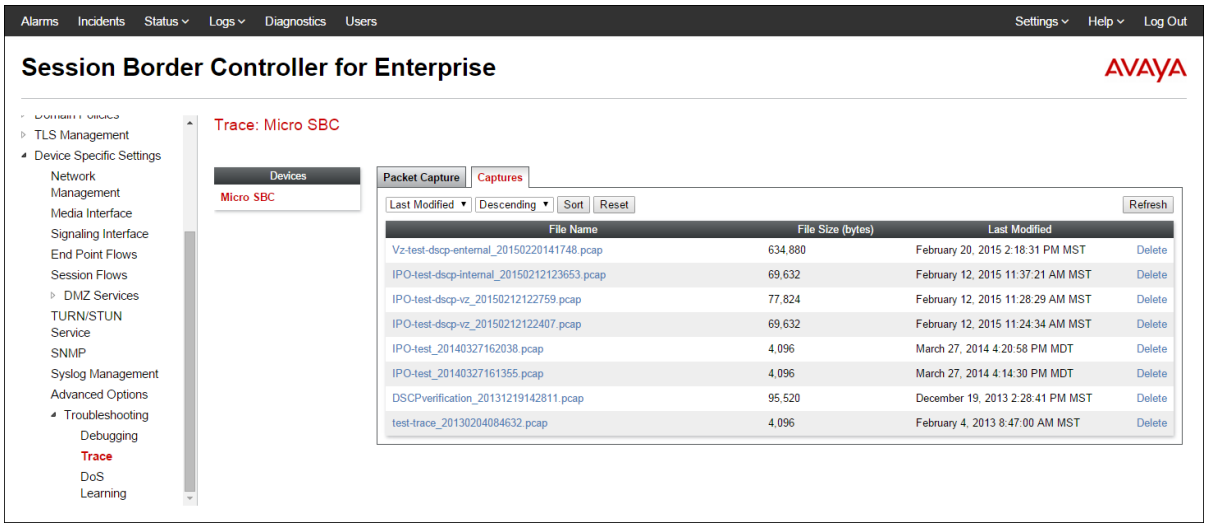
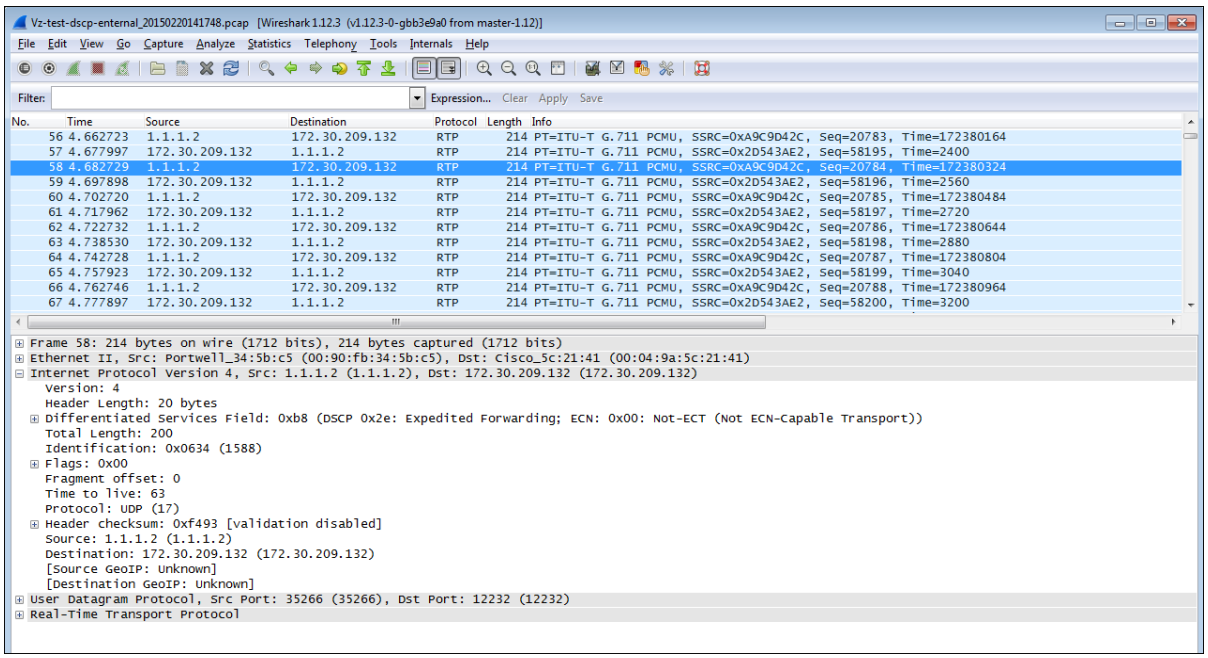
Select the **Captures** tab to view the files created during the packet capture.



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



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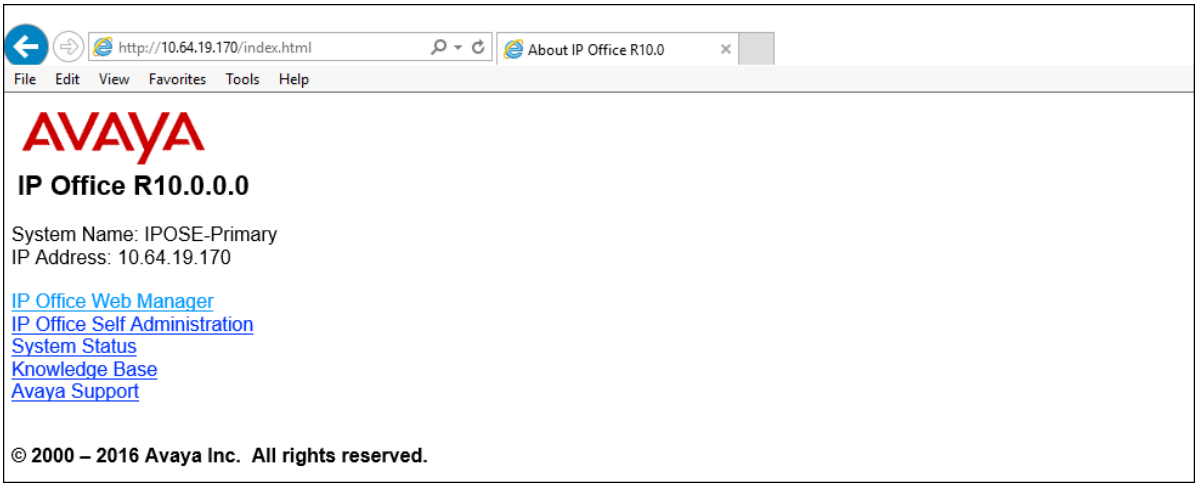
8.2. IP Office

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

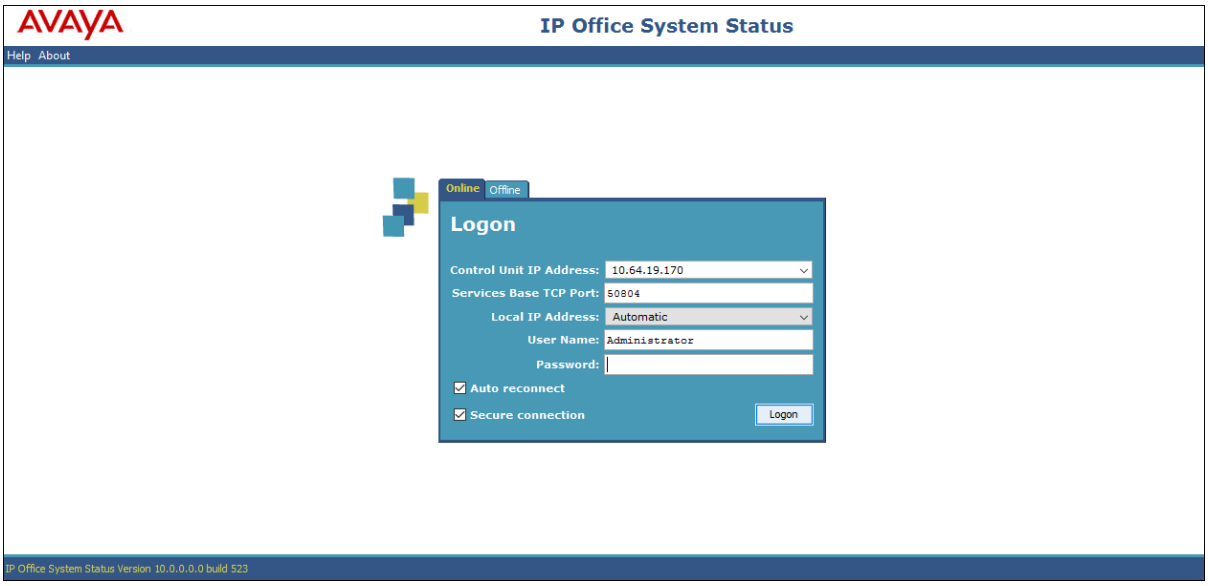
8.2.1. System Status

8.2.2. System Status

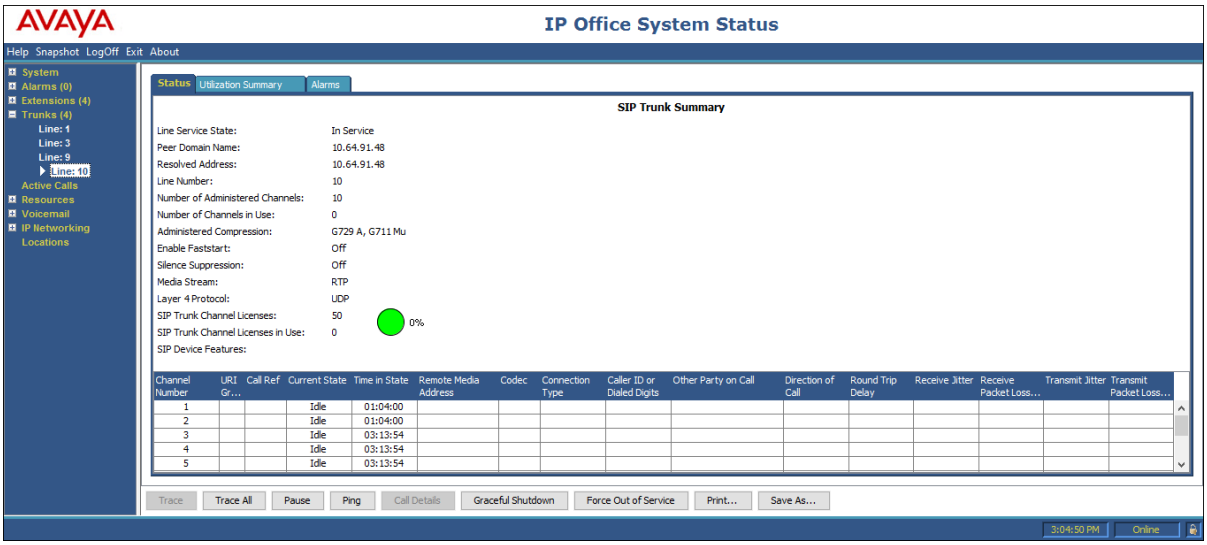
The System Status application is used to monitor and troubleshoot IP Office. Use the System Status application to verify the state of the SIP trunk. System Status can be accessed from **Start → Programs → IP Office → System Status**. Or by opening an Internet browser and type the URL: http://*ipaddress*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



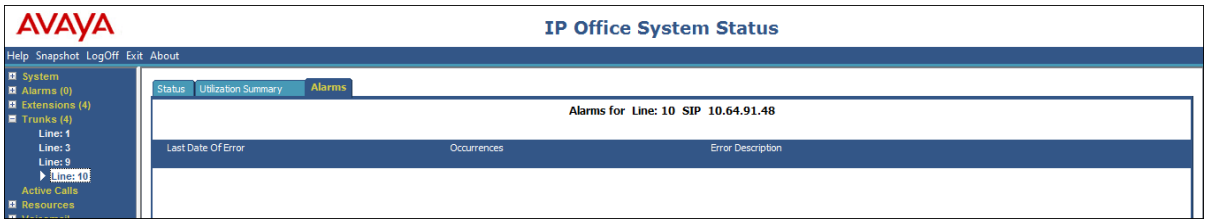
The following screen shows an example **Logon** screen. Enter the IP Office IP address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.



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



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



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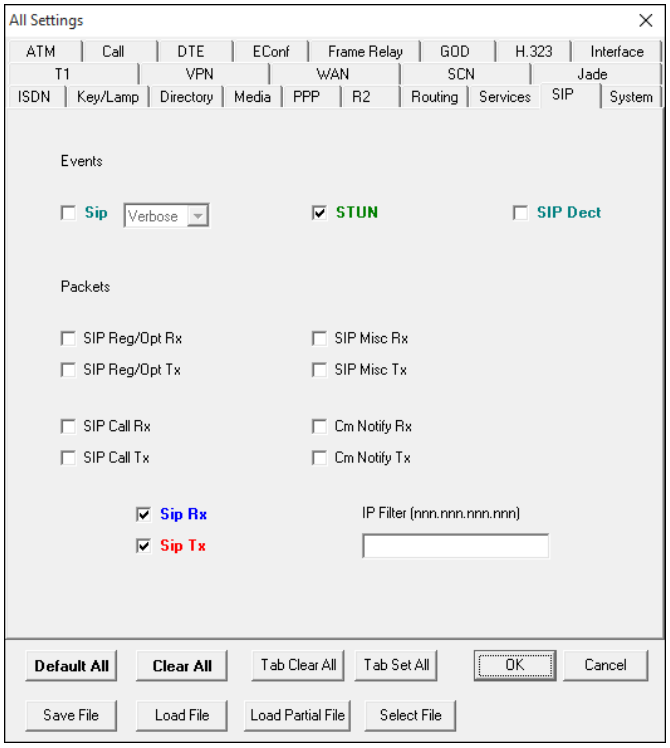
8.2.3. Monitor

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



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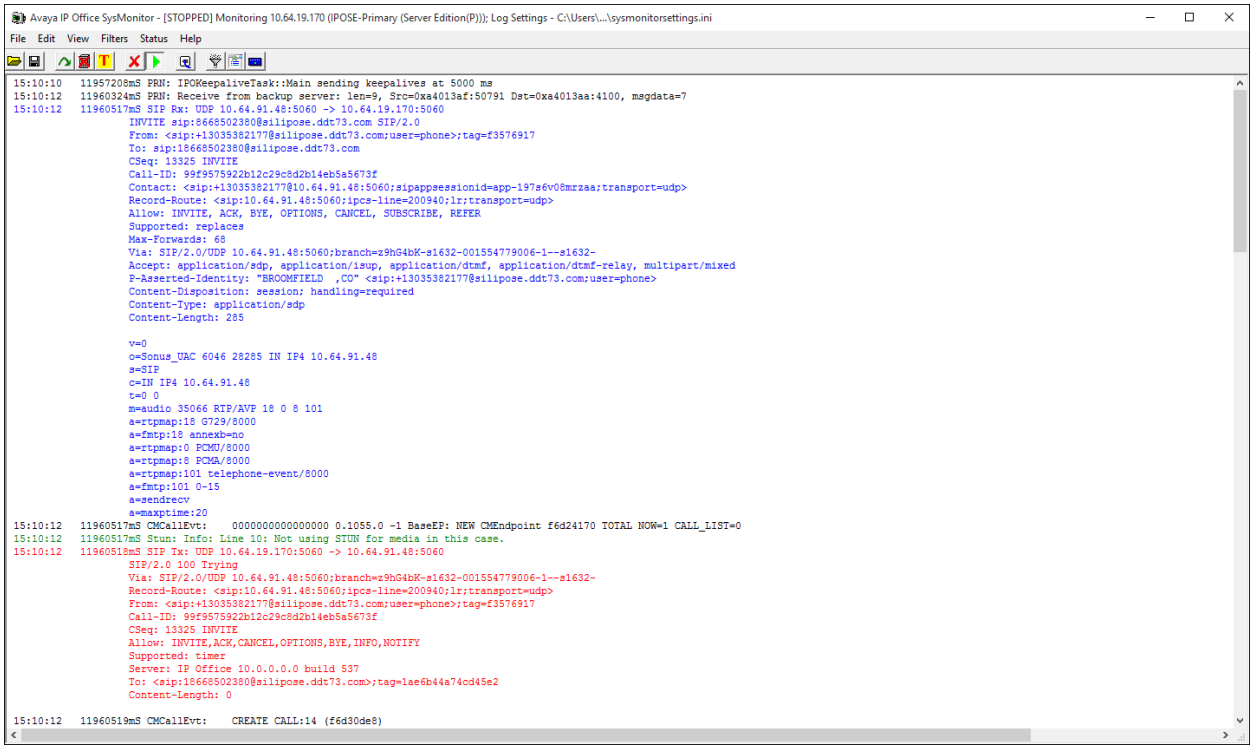
As an example, the following shows a portion of the monitoring window for an inbound call to Verizon IP Toll Free number 1-866-850-2380. Details of the SIP INVITE message sent by Verizon are shown below. This information matches the configuration in these Application Notes and is not intended to be prescriptive. 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.



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9. Conclusion

IP Office is a highly modular IP telephone system designed to meet the needs of home offices, standalone businesses, and networked offices for small and medium enterprises.

These Application Notes demonstrated how IP Office Release 10.0 with Avaya Session Border Controller for Enterprise Release 7.1 can be successfully combined with a Verizon Business IP Contact Center VoIP Inbound Service connection to enable a business to receive toll-free calls. Utilizing this solution, IP Office customers can leverage the operational efficiencies and cost savings associated with SIP trunking while gaining the advanced technical features provided through the marriage of best of breed technologies from Avaya and Verizon

10. Additional References

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

[1] *IP Office ™ Platform 10.0, Deploying Avaya IP Office ™ Platform IP500 V2*, Document Number 15-601042, July 2016

[2] *Administering Avaya IP Office ™ Platform with Manager*, February 2016

[3] *IP Office ™ Platform 10.0, Installing and Maintaining the Avaya IP Office ™ Platform Application Server*, Document Number 15-601011 Issue 10f, July 2016

[4] *IP Office ™ Platform 10.0, Deploying Avaya IP Office ™ Platform Servers as Virtual Machines*, Document Number 15-601011 Issue 03d, July 2016

[5] *IP Office ™ Platform 10.0, Using Avaya IP Office ™ System Status*, Document Number 15-601758, July 2016

[6] *Administering Avaya Communicator for Android, iPad, and Windows*, February 2016

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

[8] *Application Notes for configuring Avaya IP Office 9.0 and Avaya Session Border Controller for Enterprise 6.3 to support Remote Workers– Issue 1.0*, Feb 2015

[9] *Deploying Avaya Session Border Controller for Enterprise*, June 2016

[10] *Administering Avaya Session Border Controller*, June 2016

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

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The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 9.1 with Avaya Session Border Controller for Enterprise Release 6.3 with Verizon Business IPCC Service Suite.
[VZBIPCCIPO91SBC] Application Notes for Configuring SIP Trunking using Verizon Business IP Contact Center VoIP Inbound and Avaya IP Office Release 9.1 with Avaya Session Border Controller for Enterprise 6.3, Issue 1.1

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 10.0 and Avaya Session Border Controller for Enterprise with Verizon IP Trunk Service Suite.
[VZBIPT-IPO10SBC] Application Notes for SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service and Avaya IP Office Release 10.0 with Avaya Session Border Controller for Enterprise Release 7.1, Issue 1.0

[RFC-3261] RFC 3261 *SIP: Session Initiation Protocol* http://www.ietf.org/rfc/rfc3261.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 v1.6
- [VZ-Spec] Verizon Business IPCC Trunk Interface Network Interface Specification, Document Version 2.2.1.9

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11. Appendix A: Verizon Business IPCC Script

```
// Signaling Manipulations created for interoperability
within session "ALL"
{
 act on request where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING" and
%METHOD="NOTIFY"
  {
//change NOTIFY from "100 Trying" to "180 Ringing"
        %BODY[1].regex_replace("100 Trying","180 Ringing");
        }
    }
```


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