Avaya Solution & Interoperability Test Lab

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

Abstract

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Trunk SIP Trunk Service Offer and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of Avaya Session Border Controller for Enterprise Release 7.1, Avaya IP Office Server Edition 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.

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

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 production Verizon Business IP Trunking 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 Trunk SIP Trunk Service Offer and an Avaya IP Office solution. The Verizon Business SIP trunk redundant architecture (2-CPE) is supported by dual Avaya Session Border Controllers for Enterprise (Avaya SBCE). The Verizon Business SIP Trunk redundant (2-CPE) architecture provides for redundant SIP trunk access between the Verizon Business IP Trunk service offer and the customer premises equipment (CPE). In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office Server Edition Primary Server, an Avaya IP Office Server Edition Secondary Server, an IP500 V2 Expansion System; Avaya Voicemail Pro; Avaya Communicator for Windows; Avaya SIP, H.323, digital, and analog endpoints.

Dual Avaya SBCEs are used as edge devices between the Avaya CPE and the Verizon Business network, and provide for Verizon Business 2-CPE redundancy. In addition, the Avaya SBCEs 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.

Note - The Verizon Business SIP Trunk Redundant (2-CPE) architecture is a service option and its use is not a requirement of the Verizon Business IP Trunk service offer.

Verizon Business and Avaya developed the SIP Trunk Redundant (2-CPE) architecture to ensure that SIP trunk calls can be automatically re-routed to bypass SIP trunk failures due to network or component outages. The 2-CPE architecture described in these Application Notes is based on a customer location having two Avaya SBCEs. One Avaya SBCE is designated as Primary and one as Secondary.

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

For more information on the Verizon Business IP Trunk service, including access alternatives, visit http://www.verizonenterprise.com/products/business-communications/voice-over-ip/.

2. General Test Approach and Test Results

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

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent

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to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

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

- Incoming calls from the PSTN were routed to the DID numbers assigned by Verizon Business to the Avaya IP Office location. These incoming PSTN calls arrived via the SIP Line and were answered by Avaya SIP telephones, Avaya H.323 telephones, Avaya digital telephones, analog telephones, analog fax machines, Avaya Communicator for Windows and Avaya Voicemail Pro. The display of caller ID on display-equipped Avaya IP Office telephones was verified.
- Incoming calls answered by members of Hunt Groups were verified.
- Outgoing calls from the Avaya IP Office location to the PSTN were routed via the SIP Line to Verizon Business. These outgoing PSTN calls were originated from Avaya SIP telephones, Avaya H.323 telephones, Avaya digital telephones, analog endpoints, Avaya Communicator for Windows and Avaya Voicemail Pro. The display of caller ID on display-equipped PSTN telephones was verified.
- Inbound / Outbound fax using G.711 and T.38 were verified.
- Proper disconnect when the caller abandoned a call before answer for both inbound and outbound calls.
- Proper disconnect when the IP Office party or the PSTN party terminated an active call.
- Proper busy tone heard when an IP Office user called a busy PSTN user, or a PSTN user called a busy IP Office user (i.e., if no redirection was configured for user busy conditions).
- Various outbound PSTN call types were tested including long distance, international, toll-free, operator assisted, and directory assistance calls.
- Requests for privacy (i.e., caller anonymity) for IP Office outbound calls to the PSTN were verified. That is, when privacy is requested by IP Office, outbound PSTN calls were successfully completed while withholding the caller ID from the displays of display-equipped PSTN telephones.
- Privacy requests for inbound calls from the PSTN to IP Office users were verified. That is, when privacy is requested by a PSTN caller, the inbound PSTN call was successfully completed to an IP Office user while presenting an "anonymous" display to the IP Office user.
- SIP OPTIONS monitoring of the health of the SIP trunk was verified. Both Verizon Business and IP Office were able to monitor SIP trunk health using SIP OPTIONS.
- IP Office outbound calls were placed with simple short codes as well as using ARS. Using ARS, the ability of IP Office to route-advance to an alternate route was exercised when the primary SIP line was not responding. The Line Group associated with the Verizon Business SIP Trunk was the primary line group chosen for a call, or an alternate line group was selected upon failure of a primary line.

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- Incoming and outgoing calls using the G.729A and G.711MU codecs.
- DTMF transmission (RFC 2833) with successful voice mail navigation using G.729A and G.711MU for incoming and outgoing calls. Successful navigation of a simple auto-attendant application configured on Avaya Voicemail Pro.
- Inbound and outbound long holding time call stability.
- Telephony features such as call waiting, hold, transfer, and conference.
- Attended call transfer using the SIP REFER method.
- Unattended or "blind" call transfer using the SIP REFER method.
- Inbound calls from Verizon IP Trunk service that were call forwarded back to PSTN destinations, presenting true calling party information to the PSTN phone, via Verizon IP Trunk service.
- Mobile twinning to a mobile phone, presenting true calling party information to the mobile phone. Outbound mobile call control was also verified successfully (e.g., using DTMF on a twinned call to place new calls and create a conference via a mobile phone).
- DiffServ markings in accordance with network requirements for Avaya SBCE SIP signaling and RTP media.
- Mobility Features such as Mobile Callback and Mobile Call Control.
- Automatic fail-over testing associated with the 2-CPE redundancy and IP Office Server Edition resiliency (i.e., calls automatically re-routed around component outages).
- 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.

- **SIP endpoint transfers:** When Refer based call transfers are performed, Verizon does not send NOTIFY SIP messages to Avaya IP Office to signal transfer completion. Some Avaya SIP endpoints (e.g., Avaya 1140E, and Avaya Communicator for Windows) require receipt of a NOTIFY when Refer based call transfers are performed. The IP Office SIP Line option, **Emulate NOTIFY for Refer** will send the necessary NOTIFY messages to these endpoints (see **Section 5.4.7**).

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

3. Reference Configuration

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

In the sample configuration, Avaya SBCEs receives traffic from the Verizon Business IP Trunk service on port 5060 and sends traffic to port 5071, using UDP for network transport, as required by the Verizon Business IP Trunk service. As shown in **Table 1**, the Verizon Business IP Trunk service provided Direct Inward Dial (DID) numbers. These DID numbers were mapped to IP Office destinations via Incoming Call Routes in the IP Office configuration.

Verizon Business used the Fully Qualified Domain Name (FQDN) *pcelban0001.avayalincroft.globalipcom.com*.
The Avaya CPE environment was assigned FQDN *adevc.avaya.globalipcom.com* by Verizon Business.

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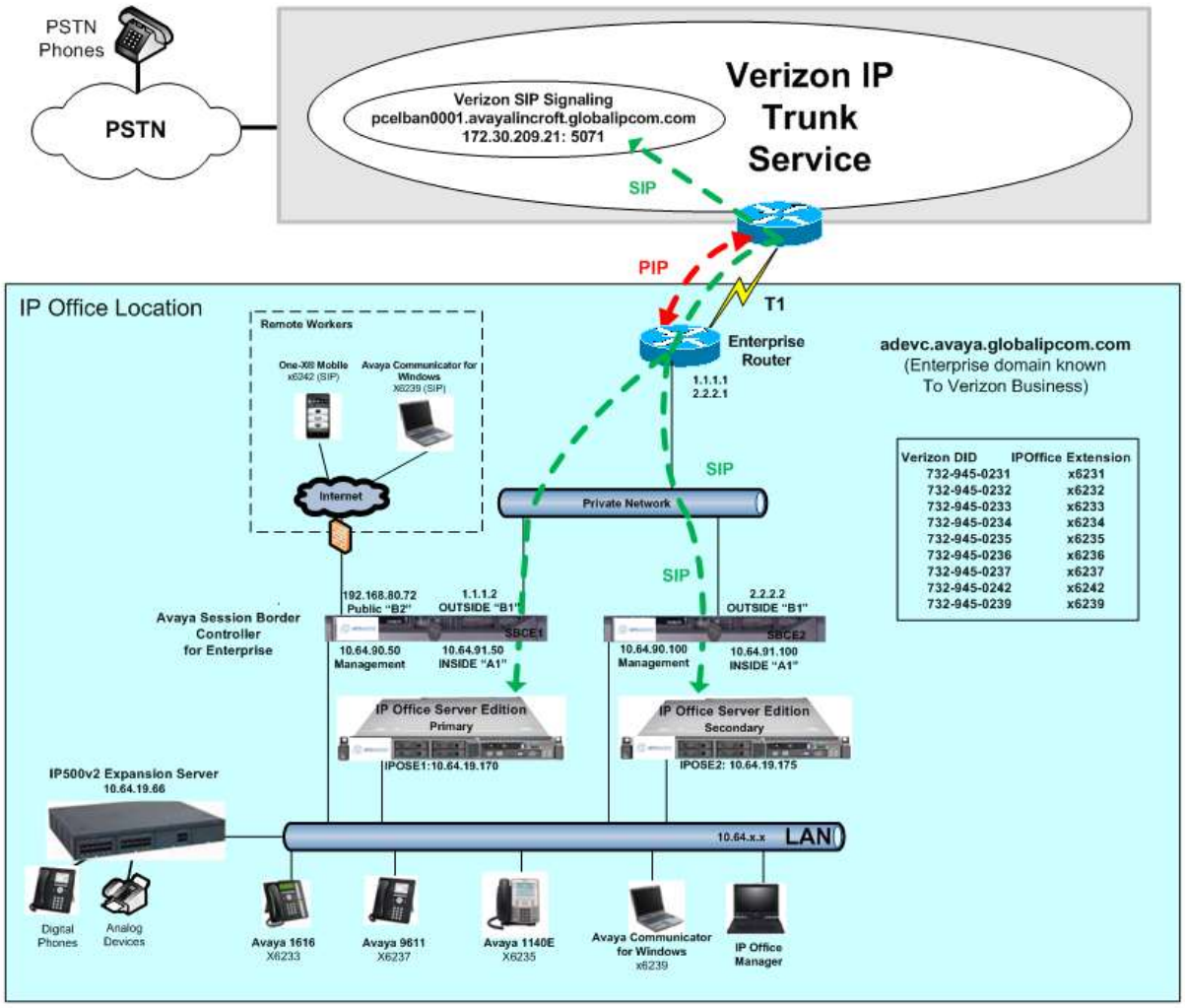

Figure 1: Avaya Interoperability Test Lab Configuration

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 referenceError! Reference source not found. **[7]**.

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4. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Equipment	**Software**
Avaya Session Border Controller for Enterprise	Release 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 6.6229
Avaya 1140E IP Telephone (SIP)	Release 04.04.23
Avaya 9508 Digital Telephone	Release 0.59
Avaya T7316E Digital Telephone	N/A
Avaya Communicator for Windows	Release 2.1.3.237
Avaya one-X® Mobile Preferred for IP Office on Android	Release 10.0.0.184

Table 1: 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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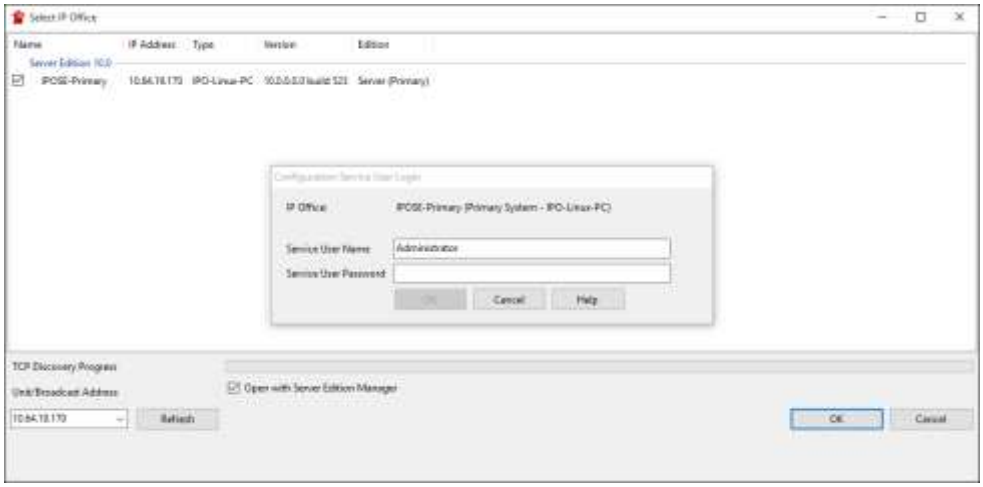
5. Avaya IP Office Primary Configuration

IP Office is configured via the IP Office Manager program. For more information on IP Office Manager, consult reference **[2]**. From the IP Office Manager PC, select **Start → Programs → IP Office → Manager** to launch the Manager application. 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 is configured on the Primary server, and a SIP trunk to the Secondary SBCE is configured on the Secondary 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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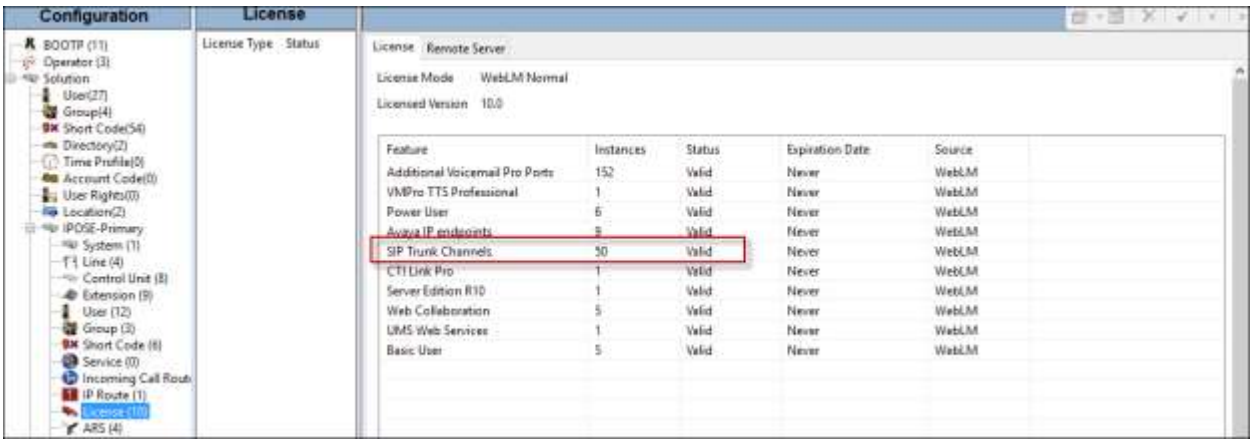
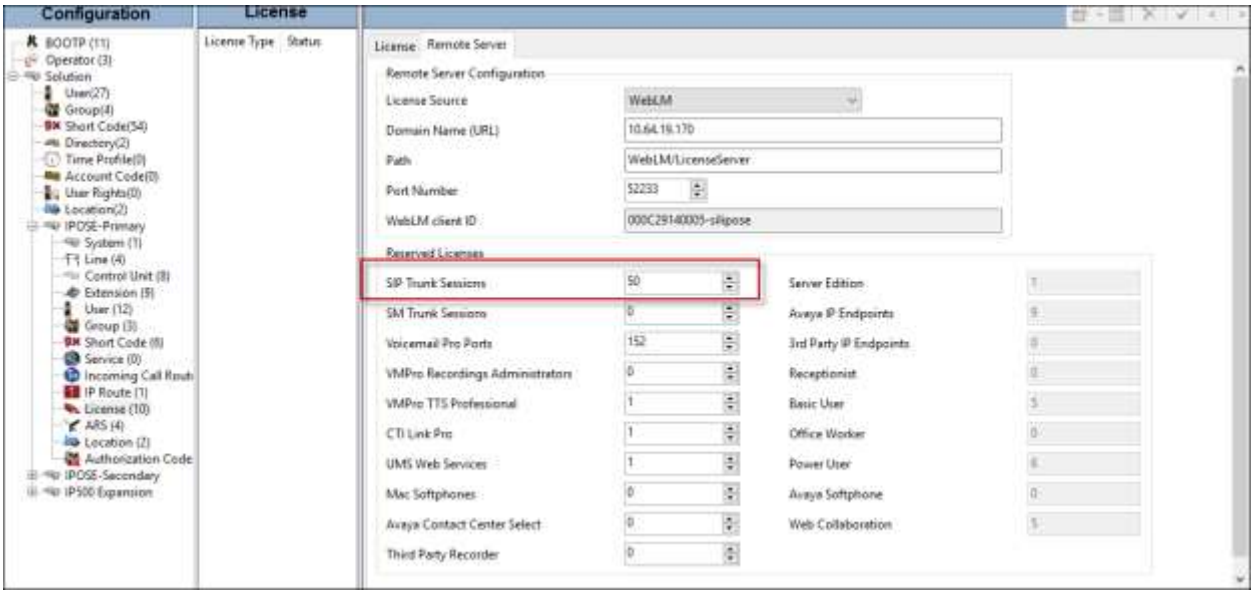
5.1. Licensing

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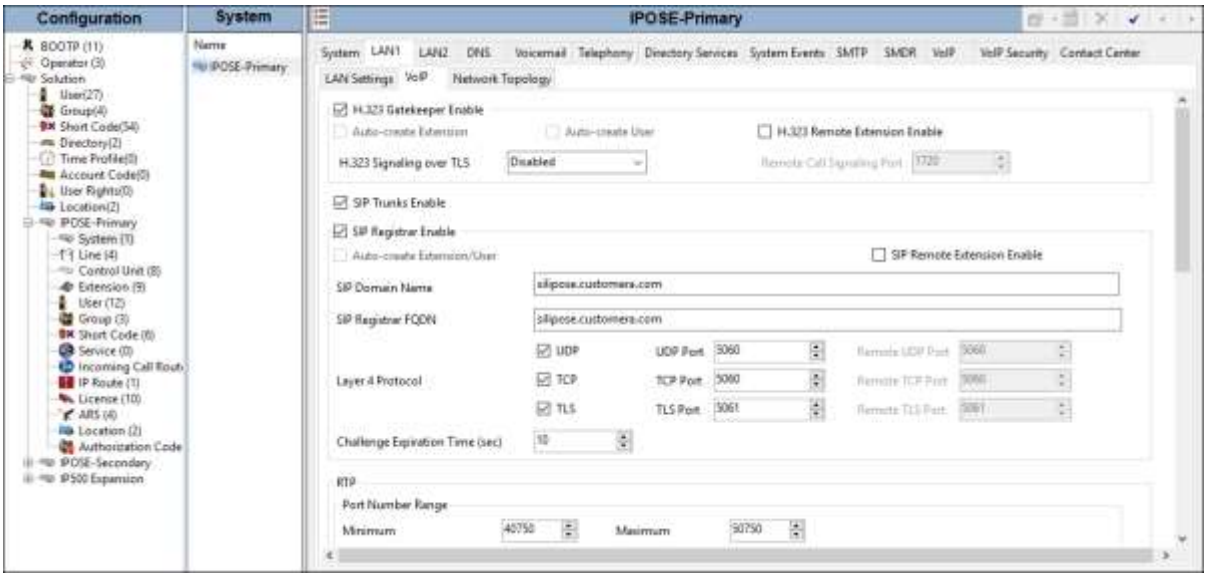
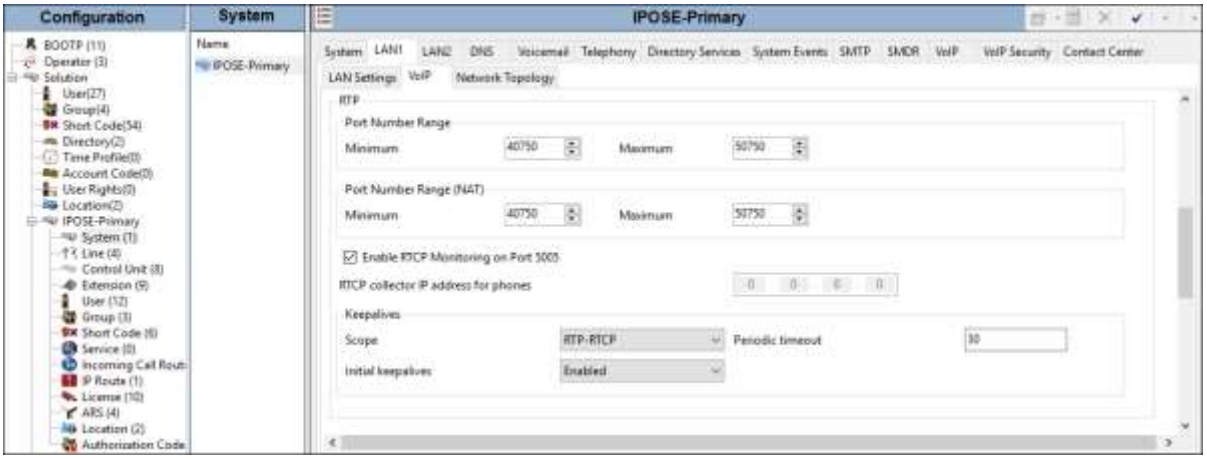
If desired, the **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media paths from Avaya SBCE to the Primary server. The defaults are used here.



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.



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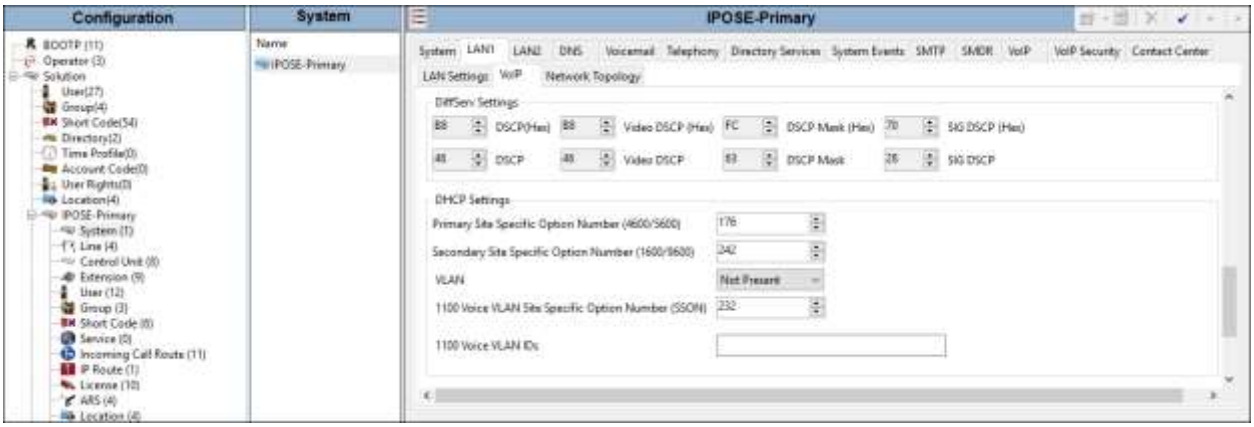
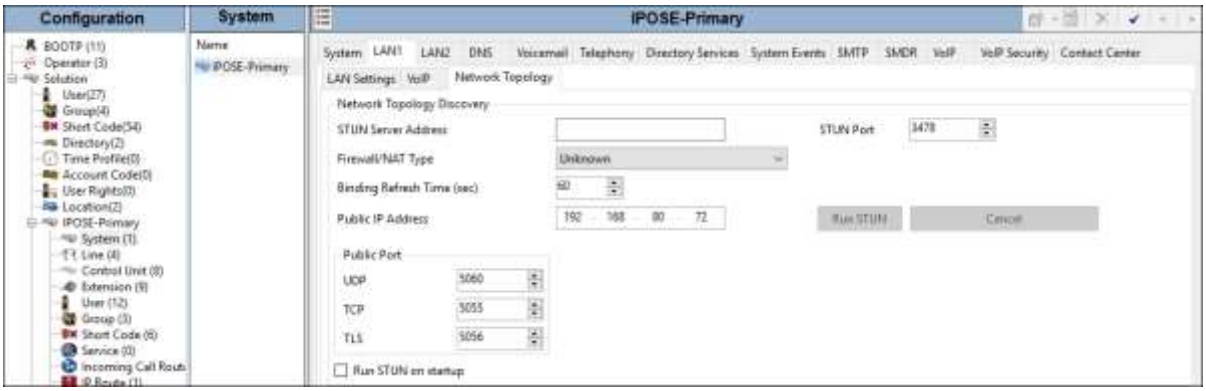
Scrolling down, the Primary server can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Service policies. In the sample configuration shown below, IP Office will mark SIP signaling with a value associated with "Assured Forwarding" 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 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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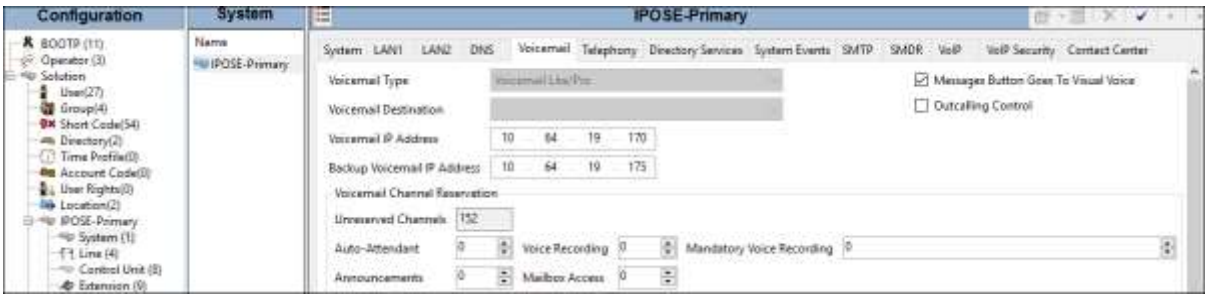
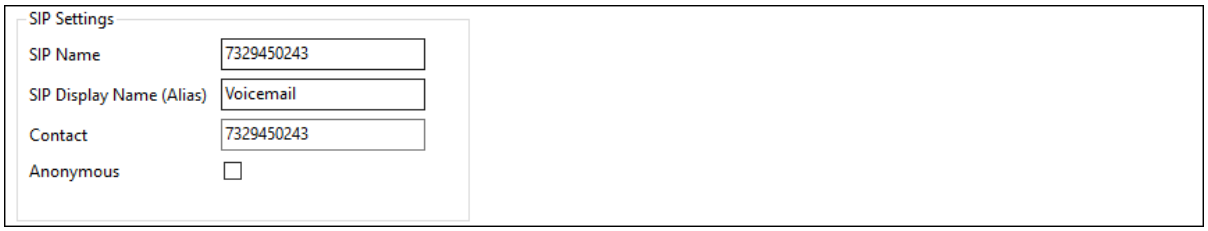
5.2.2. Voicemail Settings

To view or change voicemail settings, select the **Voicemail** tab as shown in the following screen. The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. The **Voicemail Type** in the sample configuration is "**Voicemail Lite/Pro**". The **Voicemail IP Address** in the sample configuration is "**10.64.19.170**", the IP address of the Primary server running the Voicemail Pro software. The **Backup Voicemail IP Address** is "**10.64.19.175**", the IP address of the Secondary server.



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



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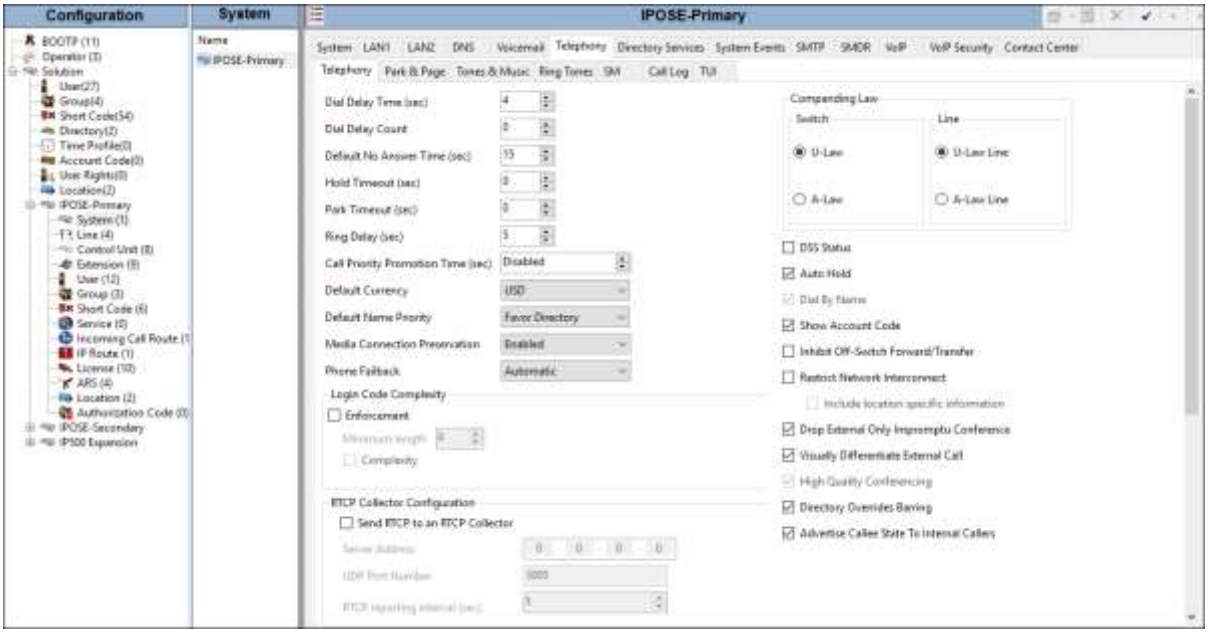
5.2.3. System Telephony Configuration

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



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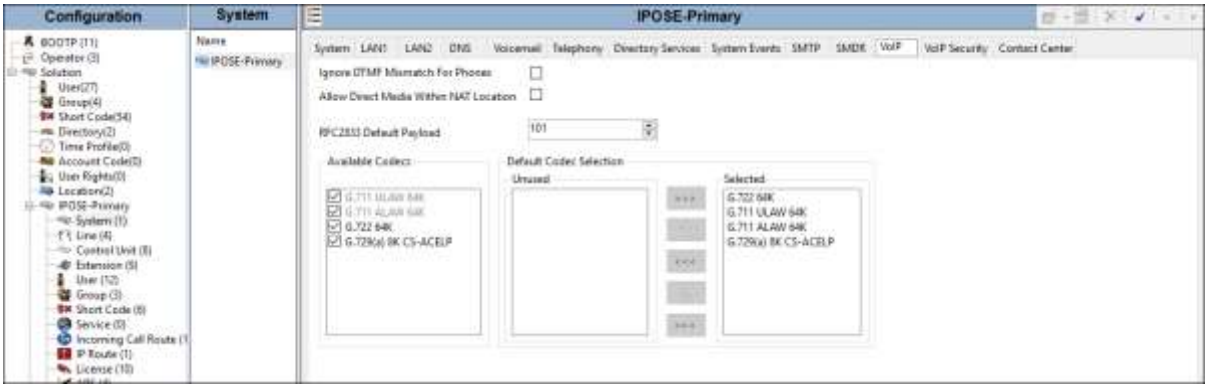
5.2.4. System Codecs Configuration

To view or change system codec settings, select the **VoIP** tab. On the left, observe the list of **Available Codecs**. In the example screen below, which is not intended to be prescriptive, the parameter 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 5.4.6**). 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

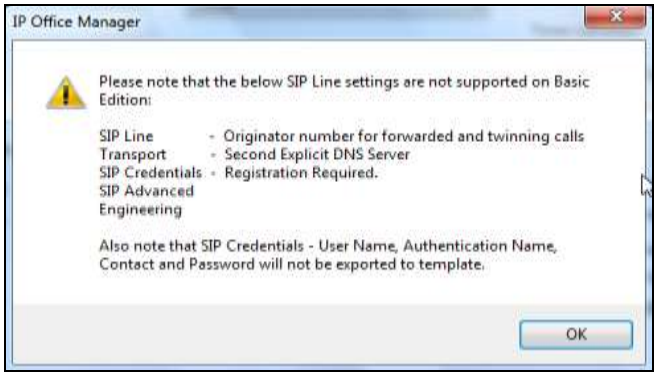
This section shows the configuration screens for the SIP Line in IP Office Release 10.0. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the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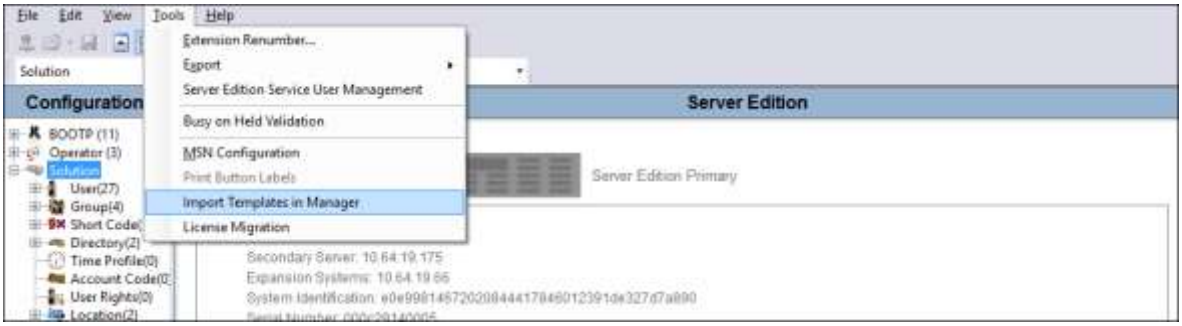
5.4.1. Importing a SIP Line Template

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.

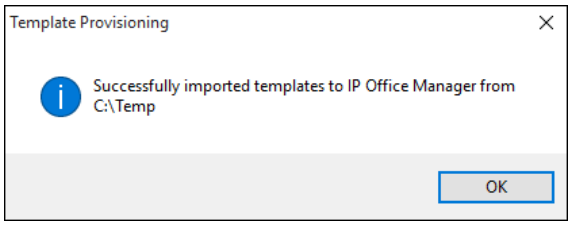
1. Copy a previously created template file to a location (e.g., *\temp*) on the same computer where IP Office Manager is installed.

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 SBCE1.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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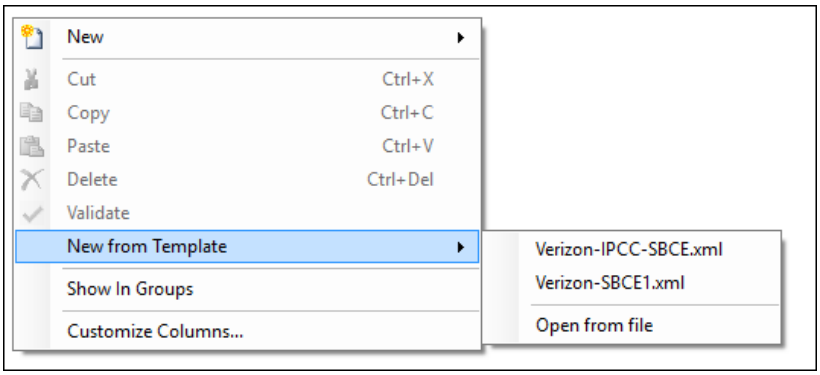

If the template is not listed, select **Open from file**, and navigate to **C:\Program Files\Avaya\IP Office\Manager\Templates**. Select ***.xml** as the file type, find the template, and click **Open**.

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 9, used for Avaya SBCE to the Verizon Business IP Trunk service. The **ITSP Domain Name** is configured with the inside IP address of the Avaya SBCE as shown in **Figure 1**. The **Local Domain** is left blank. 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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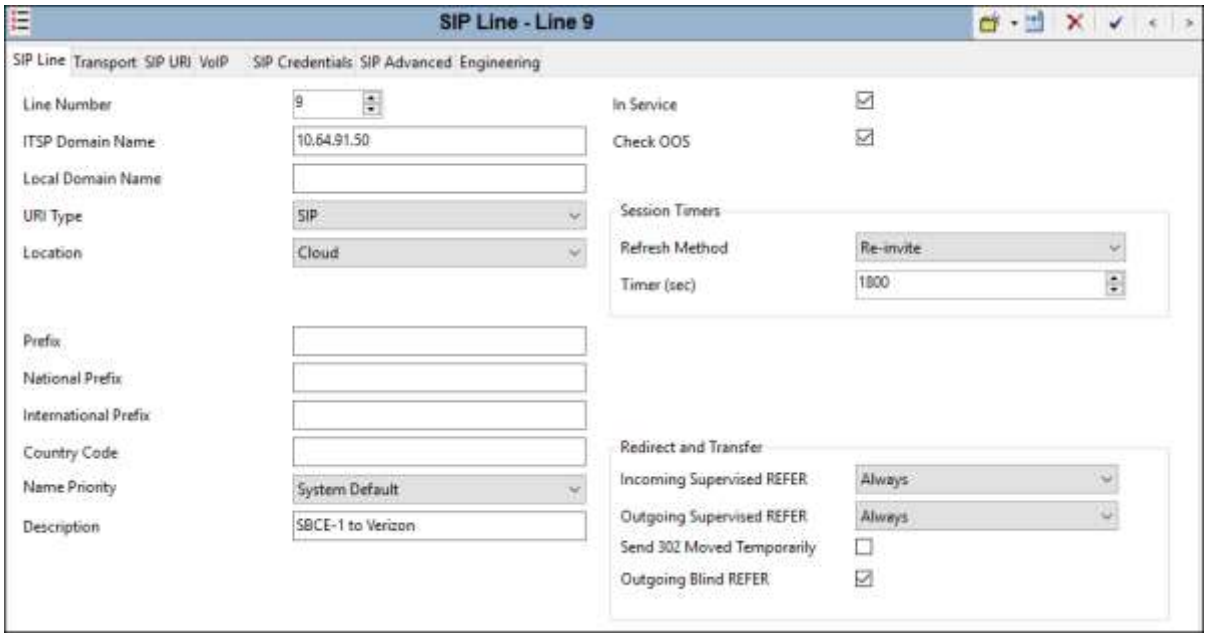
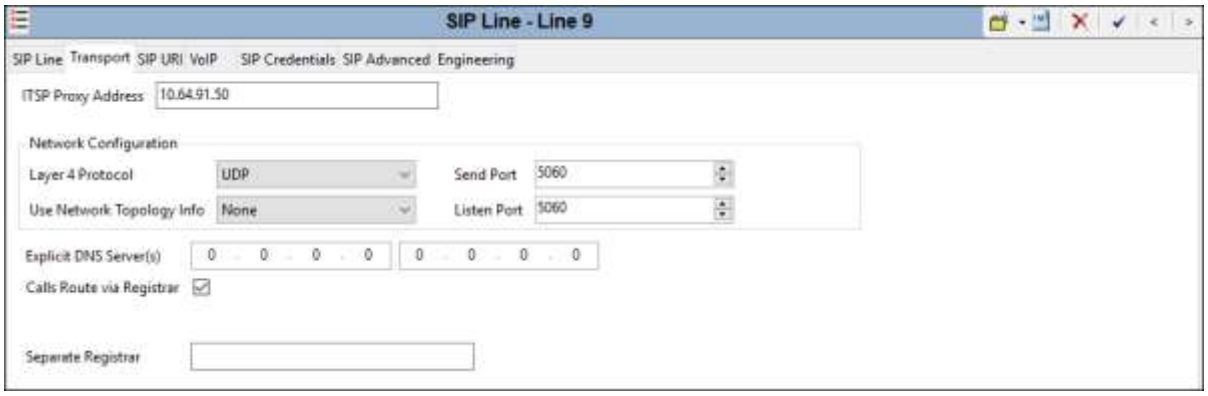
Under **Session Timers**, the **Refresh Method** is set to "**Reinvite**" and the **Timer (seconds)** is set to "**1800**". With this configuration, IP Office will send re-INVITEs every 15 minutes (half of the set value) to keep the active session alive.

Under **Redirect and Transfer**, the default automatic determination of **Incoming Supervised REFER** and **Outgoing Supervised REFER** is "**Auto**". Alternatively, the default can be overridden with "**Never**" to explicitly disable use of supervised REFER, or "**Always**" to explicitly enable use of supervised REFER, as shown below. The **Send 302 Moved Temporarily** setting is unchecked, as Verizon does not support receiving a 302 Moved Temporarily message. Optionally, the **Outgoing Blind REFER** parameter can be checked to enable use of REFER for blind transfers.



5.4.4. SIP Line - Transport Tab

Select the **Transport** tab. The **ITSP Proxy Address** is set to the inside IP address of the primary 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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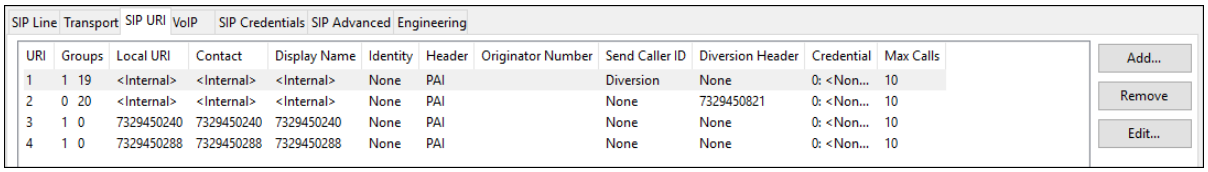
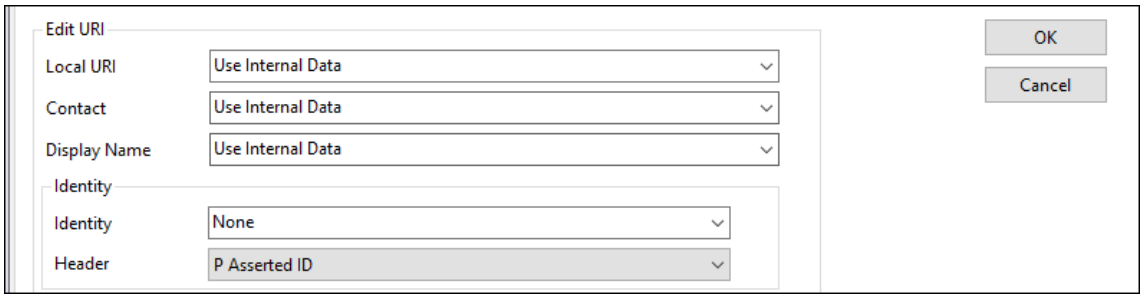
5.4.5. SIP Line - SIP URI Tab

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.

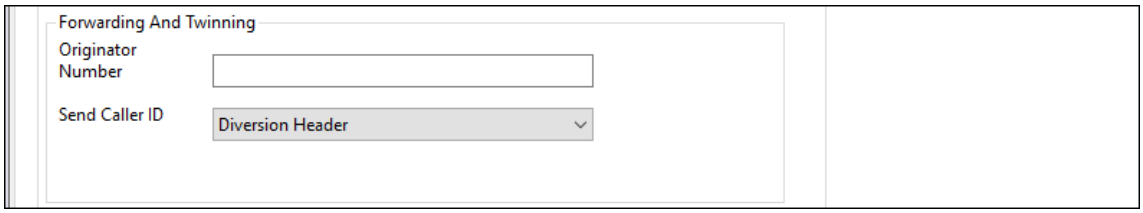


URI	Groups	Local URI	Contact	Display Name	Identity	Header	Originator Number	Send Caller ID	Diversion Header	Credential	Max Calls
1	1 19	<Internal>	<Internal>	<Internal>	None	PAI		Diversion	None	0: <Non...	10
2	0 20	<Internal>	<Internal>	<Internal>	None	PAI		None	7329450821	0: <Non...	10
3	1 0	7329450240	7329450240	7329450240	None	PAI		None	None	0: <Non...	10
4	1 0	7329450288	7329450288	7329450288	None	PAI		None	None	0: <Non...	10

In the example screen below, a previously configured entry is edited. "**Use Internal Data**" is selected for the **Local URI**, **Contact** and **Display Name**. Information configured on the SIP Tab for individual users will be used to populate the SIP headers. The **Identity** parameter is set to "**None**", and the **Header** parameter is set to "**P Asserted ID"**.



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



The **Diversion Header** is new in IP Office 10.0, and can be configured with a value to allow IP Office to include a Diversion Header in the original outbound SIP INVITE message. This can be used to send a known billing number to accompany a branch office number, toll-free number, or extension number in the From Header to present as the caller id of an outbound call. An example of this would be the use of Verizon's Unscreened ANI service offer. For this URI entry the

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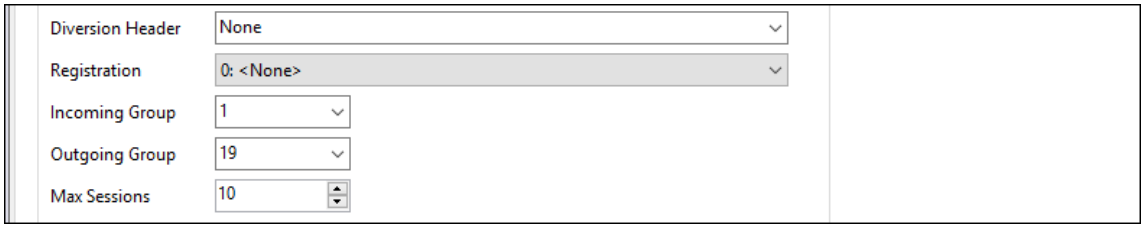
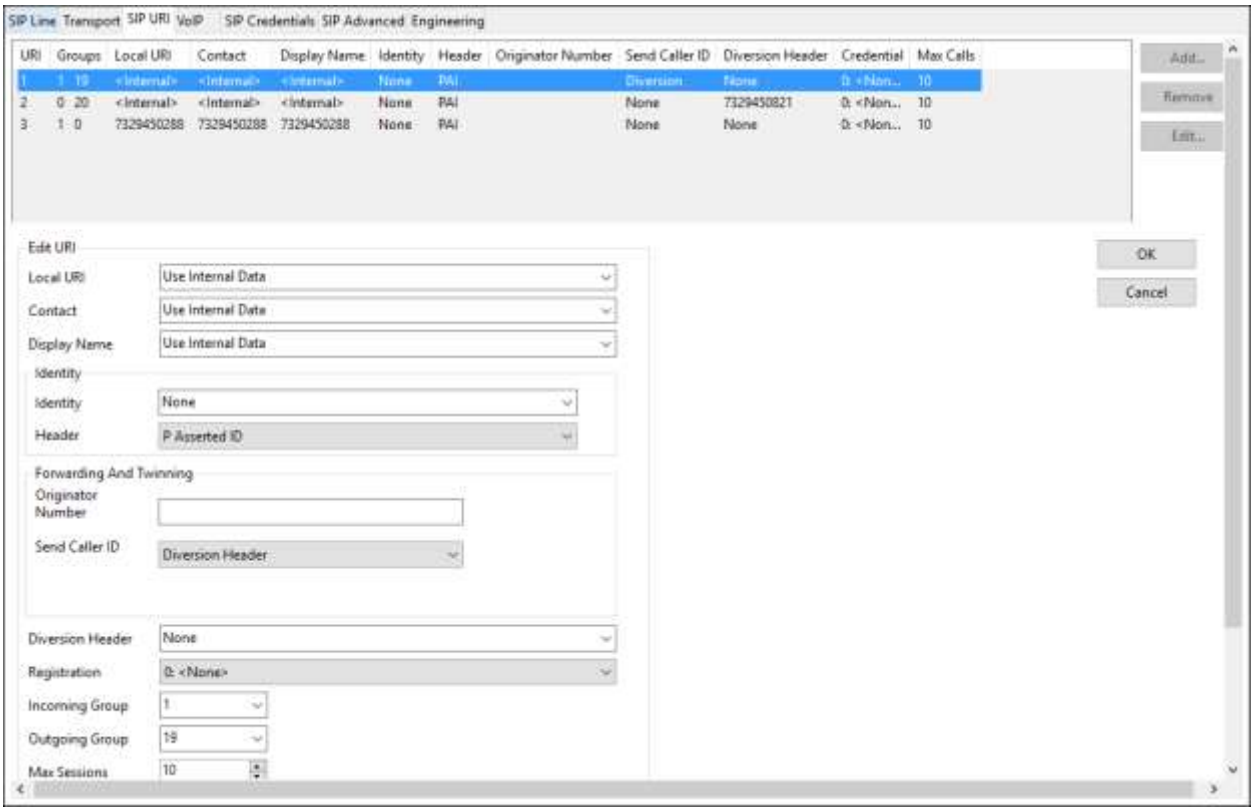
Diversion Header field is set to "**None**". The **Registration** parameter is set to the default "**0: <None>**" since Verizon Business IP Trunk service does not require registration. The **Incoming Group** parameter, set here to "**1**", will be referenced when configuring Incoming Call Routes to map inbound SIP trunk calls to IP Office destinations in **Section 5.8**. The **Outgoing Group** parameter, set here to "**19**", will be used for routing outbound calls to Verizon via the Short Codes (**Section 5.7**). The **Max Sessions** parameter, configured here to "**10**", sets the maximum number of simultaneous calls that can use the URI before IP Office returns busy to any further calls.



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



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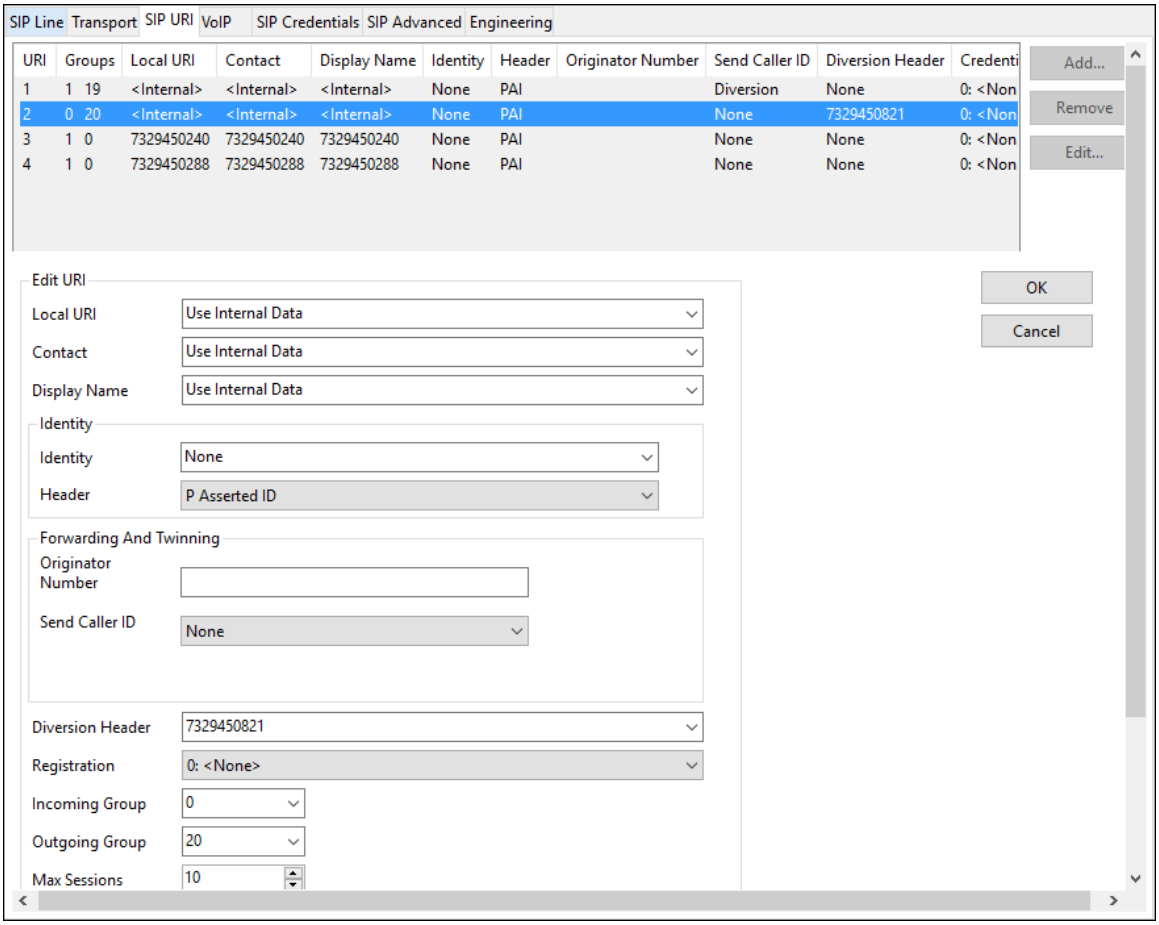
The following screen shows an example configuration for Verizon's Unscreened ANI feature. This optional configuration allows customers to send an "unscreened" ANI to Verizon's network which is then displayed to the called party as Caller ID. An "unscreened" ANI can be any telephone number that the customer passes through Verizon's network for Caller ID display purposes only. If this feature is enabled on the Verizon IP Trunk services, Verizon will designate one of the assigned telephone numbers as a "Screened Telephone Number" for each unique location. Verizon will use this Screened Telephone Number to determine call origination for billing, call routing, and E911.

The Screened Telephone Number (STN) provided by Verizon for this test is 732-945-0821. Typically, customers would have one or more STN; one for every location. A central Primary server could be used to pass multiple STNs to Verizon based on the Outgoing Group selected. The STN would then be entered in the **Diversion Header** field as shown below.



URIs 3 and 4 display service numbers, such as a DID number routed directly to voicemail or DID used for Mobile Call Control. DID numbers that IP Office should admit can be entered into the **Local URI** and **Contact** fields instead of "**Use Internal Data**". The numbers 732-945-0240 and 732-945-0288 will be assigned as service numbers in the Incoming Call Routes in **Section 5.8**.

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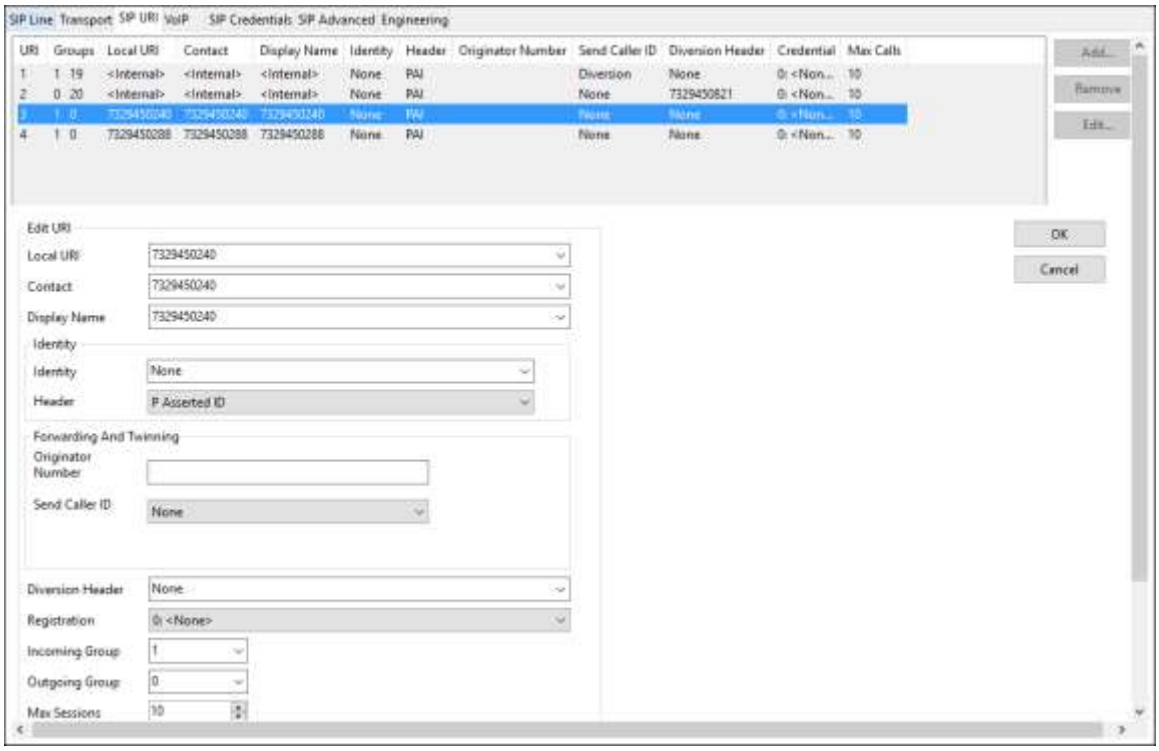

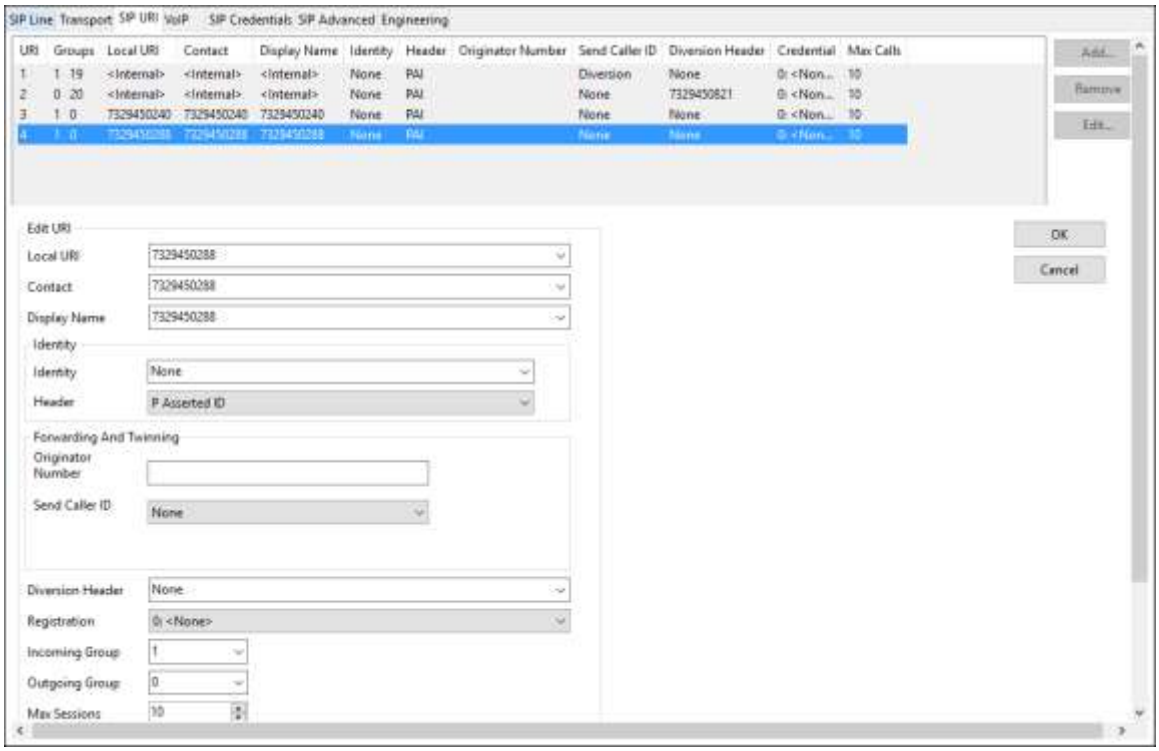

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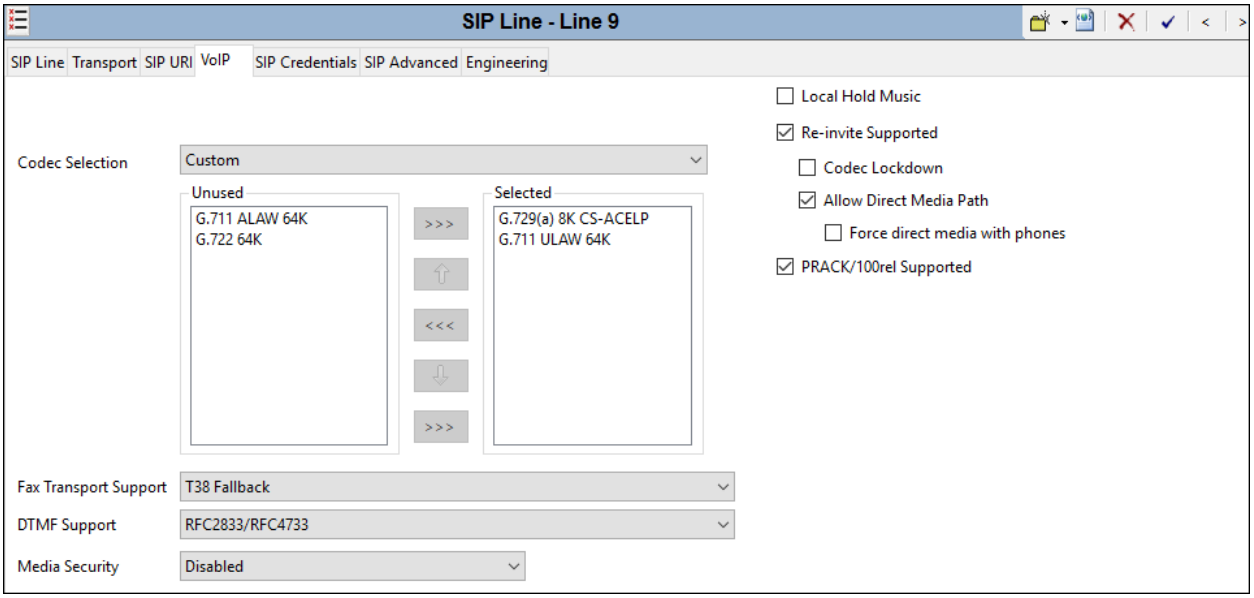
5.4.6. SIP Line - VoIP Tab

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, and 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. Depending on the customer network, 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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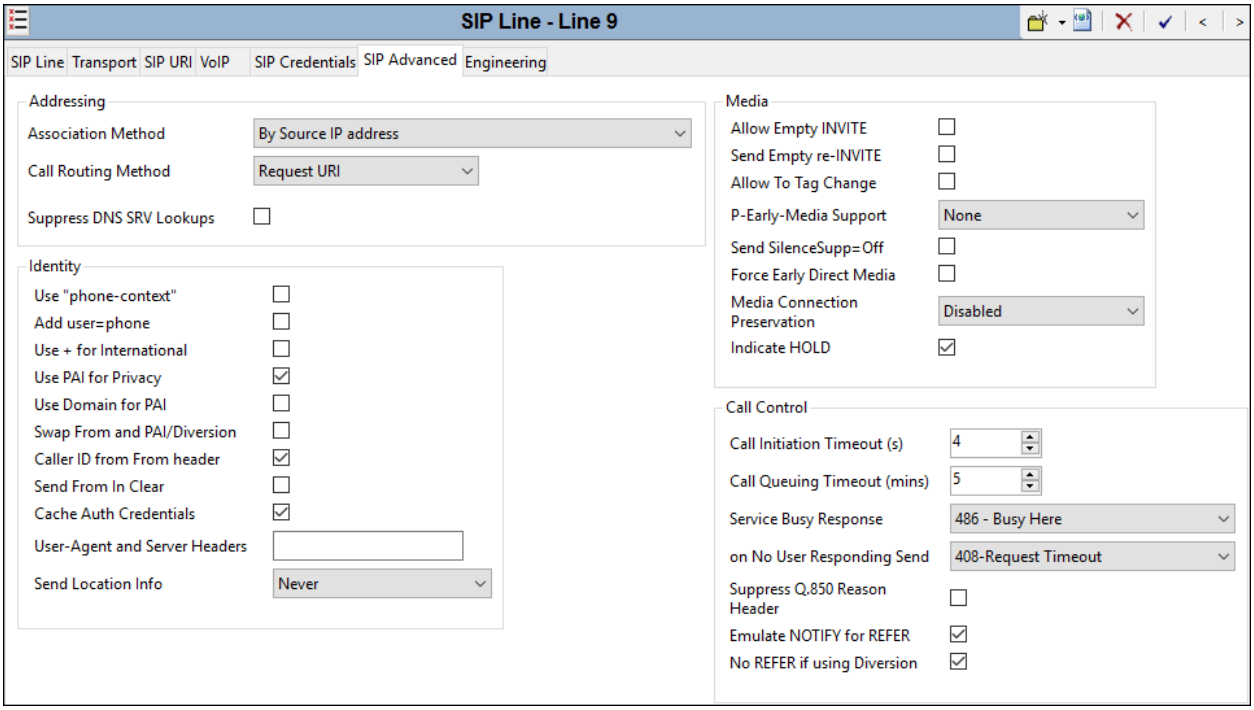
5.4.7. SIP Line – SIP Advanced Tab

Select the **SIP Advanced** tab. In the **Identity** area, the **Use PAI for Privacy** parameter is checked to include the caller's DID number in the P-Asserted-Identity (PAI) SIP header for a privacy requested call. This PAI SIP header is required by Verizon Business to admit an otherwise anonymous caller to the network. The **Caller ID from From header** parameter is checked to have IP Office use the Caller ID information in the From SIP header rather than the PAI or Contact SIP header for inbound calls. This will allow the Caller Name presented in the From SIP header by Verizon Business to also be included in the Caller ID.

In the Media area, the **Indicate HOLD** parameter is checked to have IP Office send an INVITE with media attribute "sendonly", indicating the call was placed on hold. This is the preferred behavior for Verizon Business to indicate placing a call on hold.

In the **Call Control** area, the **Emulate NOTIFY for Refer** parameter is check. This is required for SIP endpoints that perform Refer based transfers across the SIP line. See **Section 2.2** for more details. The **No Refer if using Diversion** parameter is check to prevent IP Office from using the SIP REFER method on call forwarded scenarios that use a Diversion SIP header. Verizon does not support this type of refer, and would respond with a "603 Decline" SIP message.



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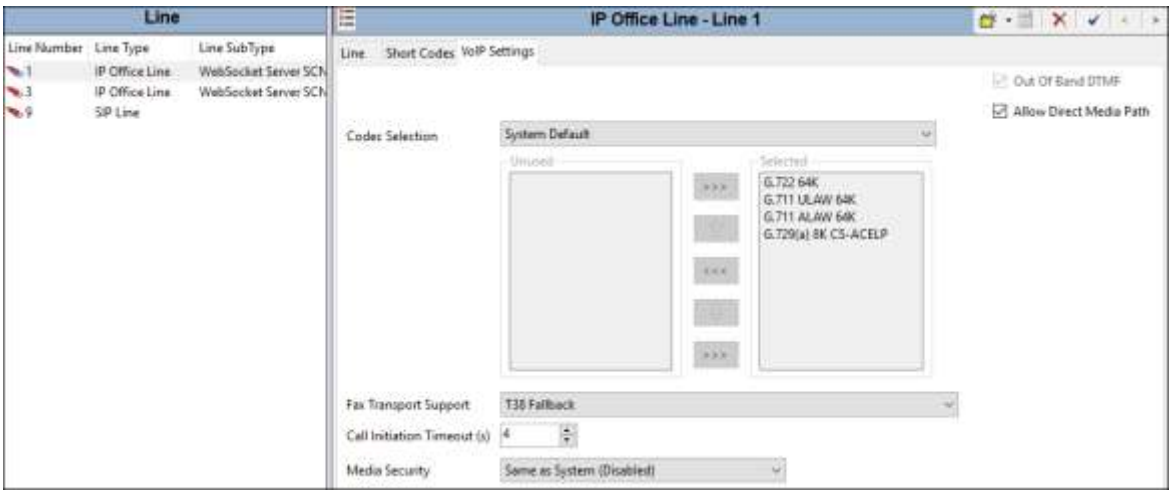
Note – An IP Office user whose calling line identification is not typically withheld from the network can request privacy in the sample configuration by dialing the short code *67 to access the SIP Line, as described in **Section 5.7**. Certain Avaya telephones can also request privacy, without dialing a unique short code, using **Features → Call Settings → Withhold Number**. The **Withhold Number** parameter may be set to "On" (i.e., for privacy). Specific users may be configured to always withhold calling line identification by checking the **Anonymous** field in the **SIP** tab for the user (**Section 5.6.1**).

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. Below is the IP Office Line to the Expansion System.

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 from the Expansion System to route directly to the internal IP address of the Avaya SBCE.



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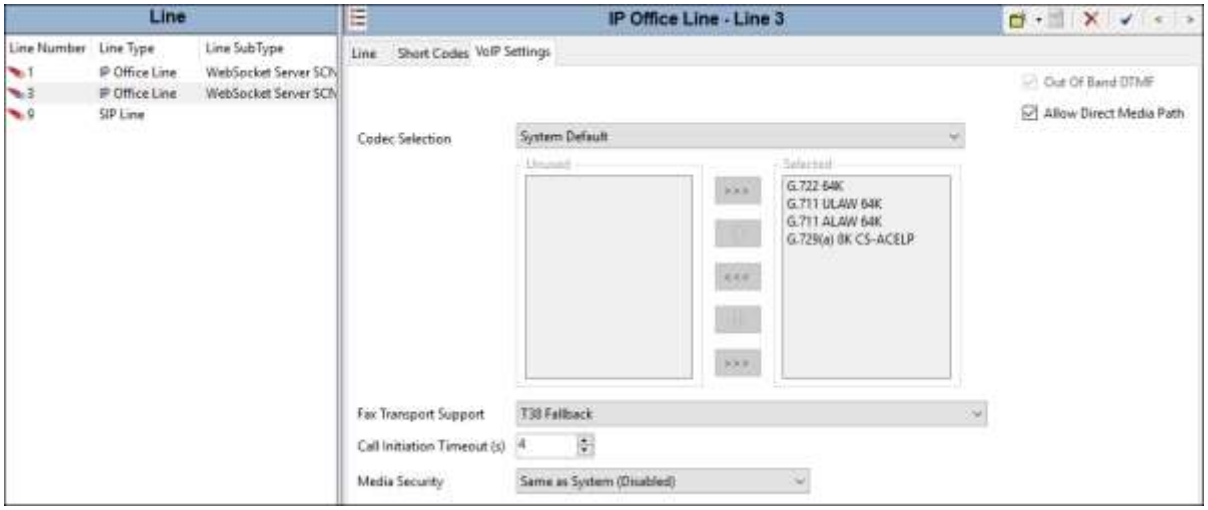
Below is the IP Office line to the Secondary Server. The **Outgoing Group ID** will be used in the ARS form to route calls to the secondary Avaya SBCE.



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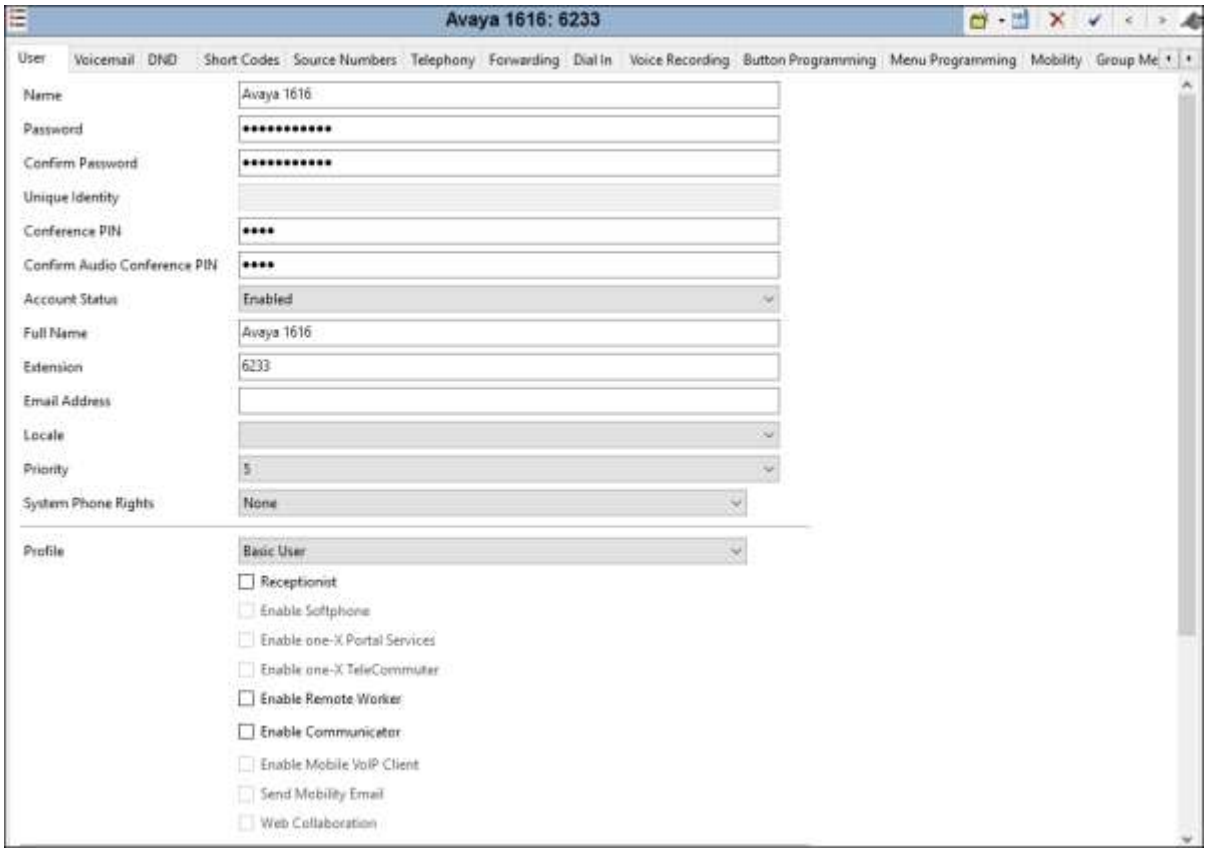
5.6. Users, Extensions, and Hunt Groups

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

5.6.1. 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.



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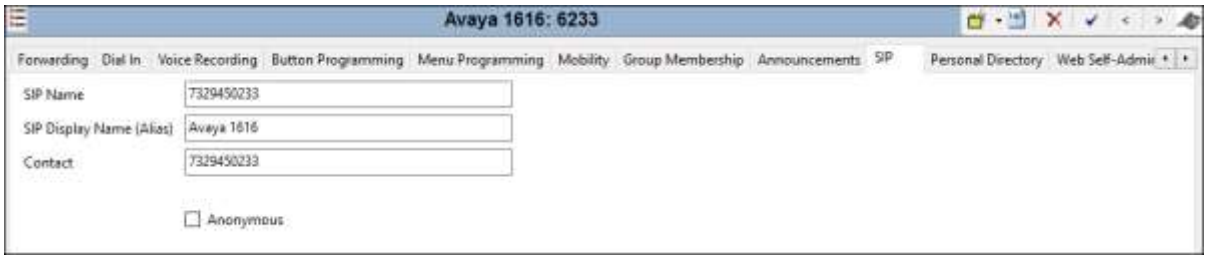
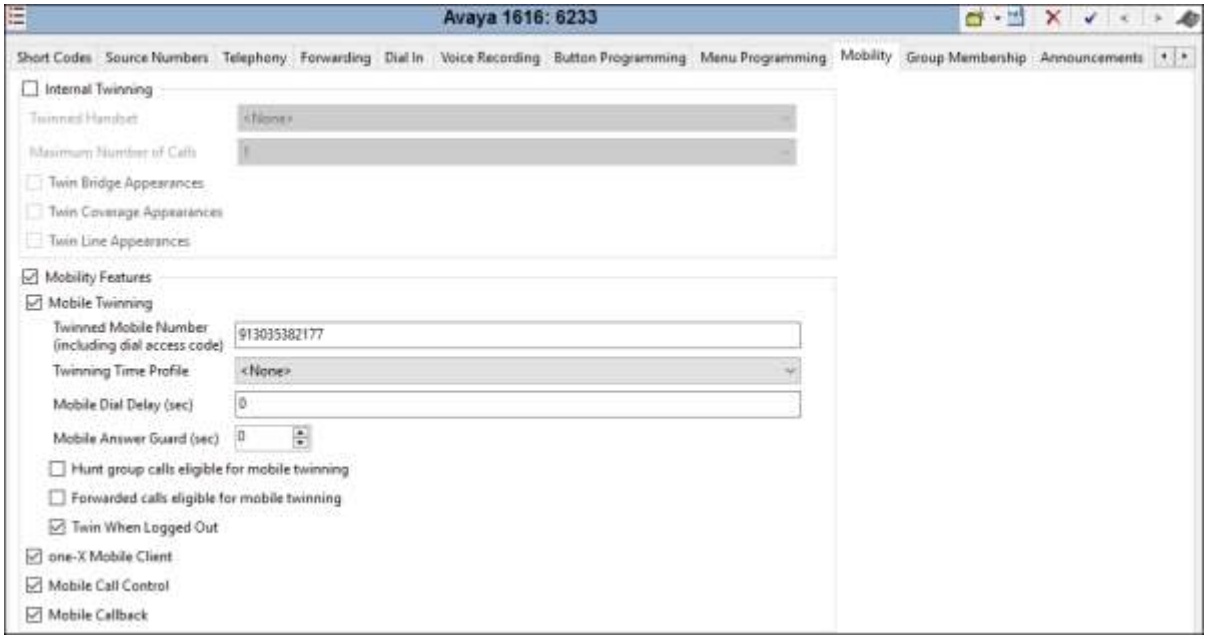
The following screen shows the **SIP** tab for user 6233. The **SIP Name** and **Contact** parameters are configured with the DID number of the user, 732-945-0233. These parameters configure the user part of the SIP URI in the From header for outgoing SIP trunk calls, and allow matching of the SIP URI for incoming calls, without having to enter this number as an explicit SIP URI for the SIP Line. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** parameter may be checked to withhold the user's information from the network. See **Section 5.7** for a method of using a short code (rather than static user provisioning) to place an anonymous call.



The following screen shows the **Mobility** tab for user 6233. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, including the dial access code for ARS, in this case 913035382177. Other options can be set according to customer requirements.



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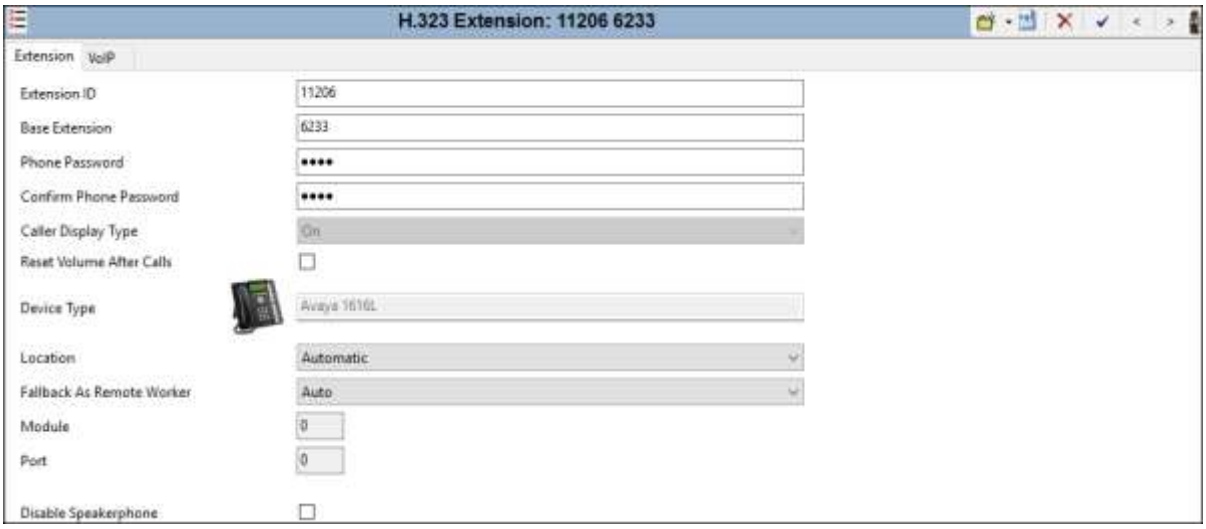
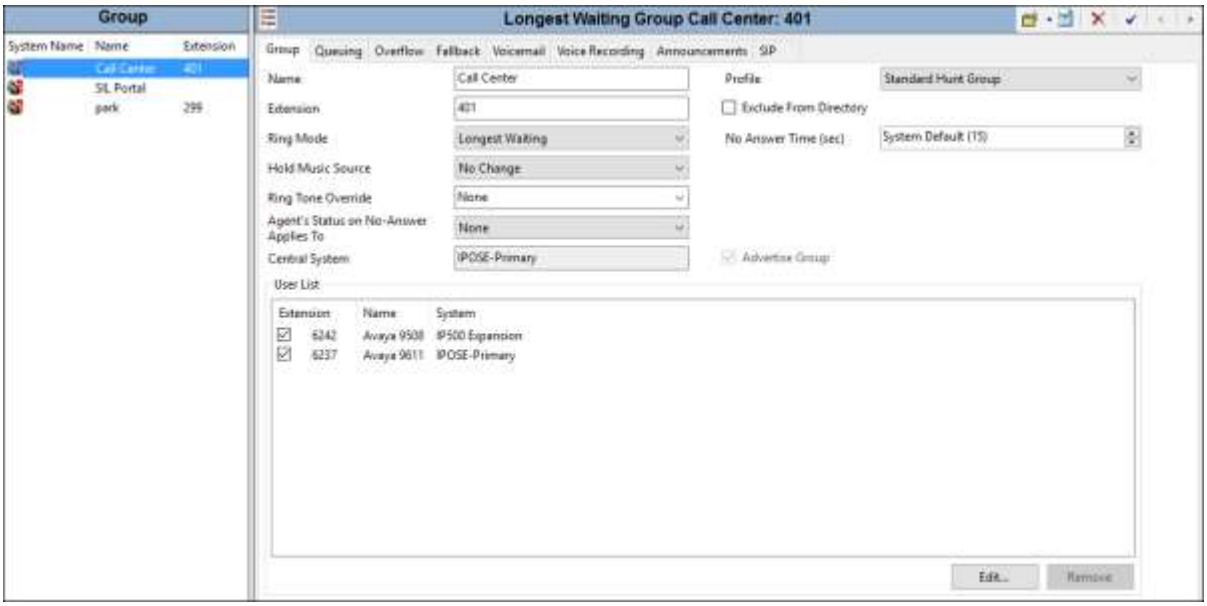
The following screen shows the Extension information for this user. To view, select **Extension** from the Navigation pane, and the appropriate extension from the Group pane.



5.6.2. 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 **Group** from the Navigation pane, and select **New**. To view or edit an existing hunt group, select **Group** from the Navigation pane, and the appropriate hunt group from the Group pane.

The following screen shows the **Group** tab for hunt group 401. The telephone extensions in the **User List** are rung based the extension that has been unused for the longest period, due to the **Ring Mode** setting "**Longest Waiting**" (i.e., "longest waiting", most idle user receives next call). Click the **Edit** button to change the **User List**.



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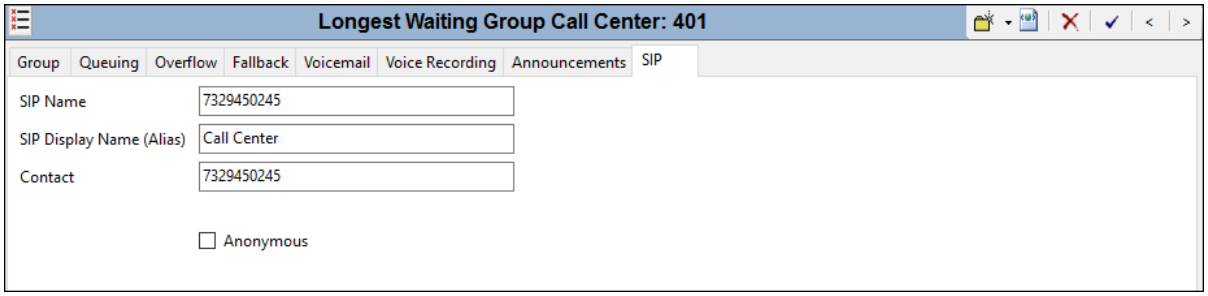
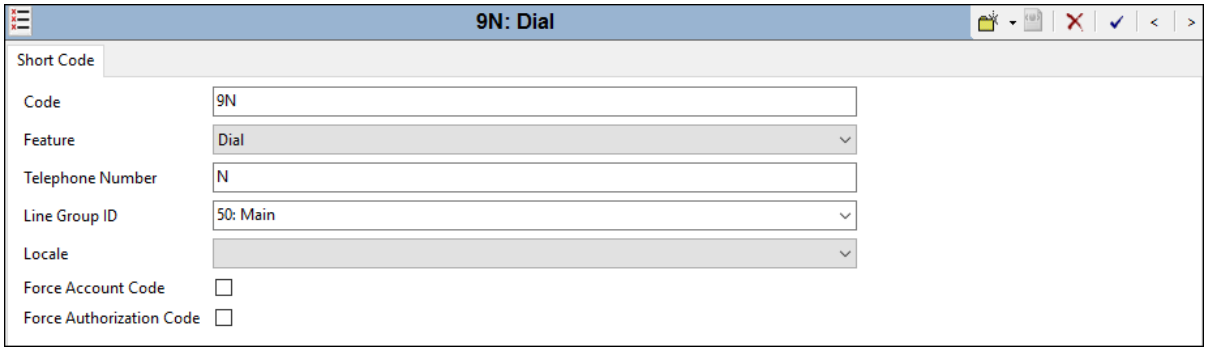
The following screen shows the **SIP** tab for hunt group 401. The **SIP Name** and **Contact** are configured with Verizon DID 7329450245. Later, in **Section 5.8**, an Incoming Call Route will map 7329450245 to this hunt group based on the information entered on this tab.



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** in the Navigation pane, and select **New**. To edit an existing short code, click **Short Code** in the Navigation pane, and the short code to be configured in the Group pane.

In the screen shown below, the short code "**9N**" is illustrated. The **Code** parameter is set to "**9N**". The **Feature** parameter is set to "**Dial**". The **Telephone Number** parameter is set to "**N**". The **Telephone Number** field is used to construct the Request URI and To Header in the outgoing SIP INVITE message. The **Line Group ID** parameter is set to "**50: Main**", configurable via ARS. See **Section 5.9** for example ARS route configuration for "50: Main" as well as an alternate route.



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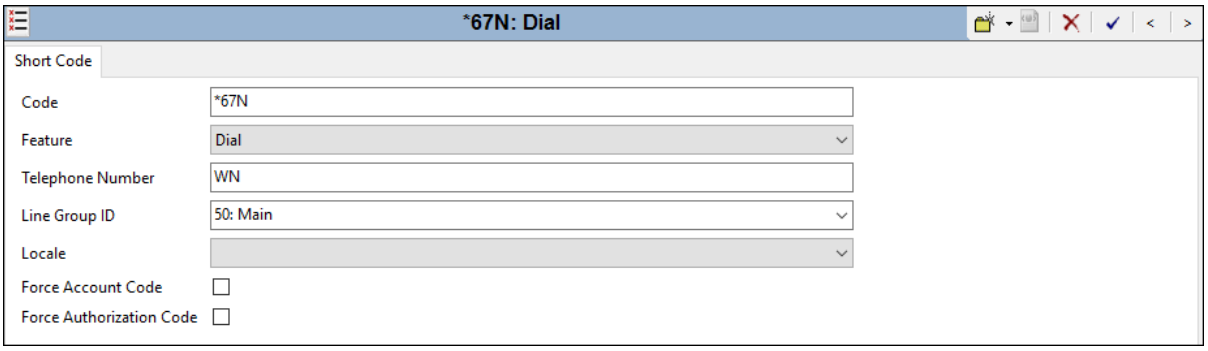
Optionally, add or edit a short code that can be used to access the SIP Line anonymously. In the screen shown below, the short code "***67N**" is illustrated. This short code is similar to the "9N" short code except that the **Telephone Number** field begins with the letter "W", which means "withhold the outgoing calling line identification". In the case of the SIP Line connecting to Verizon documented in these Application Notes, when a user dials *67 plus any number "N", IP Office will include the user's telephone number in the P-Asserted-Identity (PAI) header (see **Section 5.4.7**) along with "Privacy: Id". Verizon will allow the call due to the presence of a valid DID in the PAI header, but will prevent presentation of the caller id to the called PSTN destination.



The following screen illustrates a solution level short code, common to all servers, that acts like a feature access code rather than a means to access a SIP Line. In this case, the **Code** "**FNE31**" is defined for **Feature** "**FNE Service**" to **Telephone Number** "**31**" (Mobile Call Control). This short code will be used as means to allow a Verizon DID to be programmed to route directly to this feature, via inclusion of this short code as the destination of an Incoming Call Route. See **Section 5.8**. This feature is used to provide dial tone to twinned mobile devices (e.g., cell phone) directly from IP Office; once dial tone is received the user can perform dialing actions including making calls and activating Short Codes.

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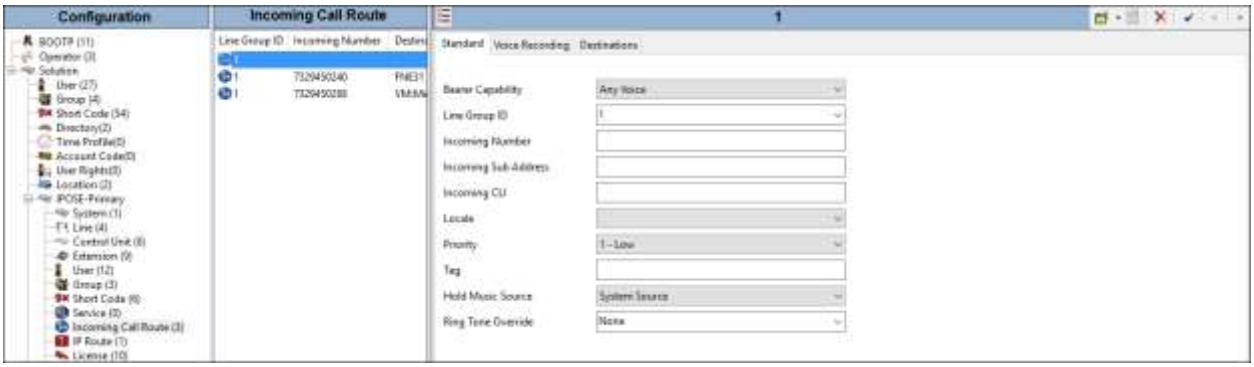
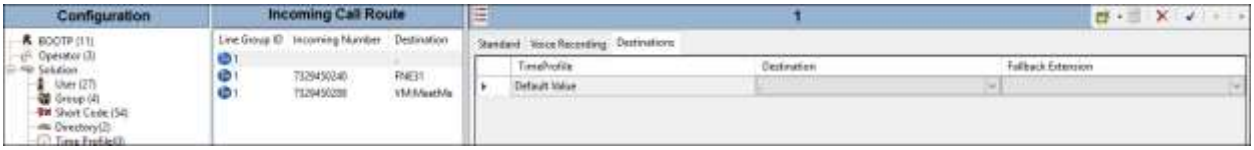
5.8. Incoming Call Routes

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

In the screen shown below, a simple incoming call route is illustrated. The **Line Group Id** is "**1**", matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to Verizon Business in **Section 5.4.5**. The **Incoming Number** field is left blank to match all details of the number in the To header.



The following **Destinations** tab for the incoming call route contains the **Destination** "**.**" entered manually. This will match the **Incoming Number** field as the destination and route the call based on the information in the SIP tab for the user or hunt group as illustrated in **Section 5.6**. For example, a call to 732-945-0232 will be routed to user 232, because this user has 7329450232 configured for the **SIP Name** and **Contact** parameters.



In the following screen, the incoming call route for **Incoming Number** "**7329450240**" is illustrated. The **Line Group Id** is "**1**", matching the Incoming Group field configured in the **SIP URI** tab for the SIP Line to Verizon Business in **Section 5.4.5**.

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The following **Destinations** tab for the incoming call route contains the **Destination** "**FNE31**" entered manually. The name "FNE31" is the short code for accessing the "Mobile Call Control" application configured in **Section 5.7**, and 732-945-0240 was configured in **Section 5.4.5** on the SIP URI tab as an incoming number. An incoming call to 732-945-0240 will be delivered directly to internal dial tone from the IP Office, allowing the caller to perform dialing actions including making calls and activating Short Codes. The incoming caller ID must match the Twinned Mobile Number entered in the User Mobility tab in **Section 5.6.1**; otherwise the IP Office responds with a "486 Busy Here" and the caller will hear a busy tone.

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

5.9. ARS and Alternate Routing

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 **Main**. 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 first entry below shows that for calls to area codes in the North American Numbering Plan, the user dials 9, followed by 10 digits. The list of codes defined below is simply an example and not intended to be prescriptive. Other dialing codes may be appropriate for different customer networks. The **Line Group ID** is set to "**19**" matching the number of the **Outgoing Group** configured on the **SIP URI** tab of SIP Line 9 to Verizon Business (**Section 5.4.5**).

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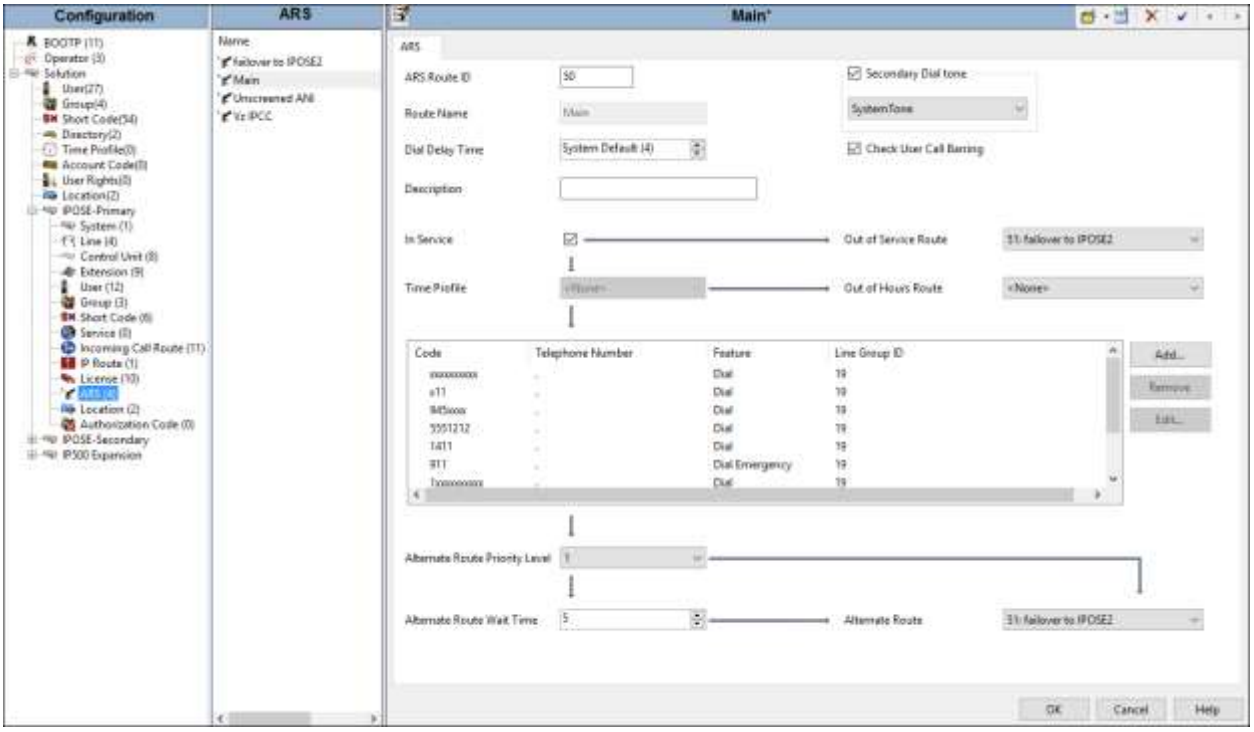

Assuming the primary route is in-service, the number passed from the short code used to access ARS (e.g., 9N in **Section 5.7**) can be further analyzed to direct the call to a specific Line Group ID. Per the example screen above, if the user dialed 9-303-538-1000, the call would be directed to Line Group 19. If Line Group 19 cannot be used, the call can automatically route to the route name configured in the **Alternate Route** parameter in the lower right of the screen. Since alternate routing can be considered a privilege not available to all callers, IP Office can control access to the alternate route by comparing the calling user's priority to the value in the **Alternate Route Priority Level** field.

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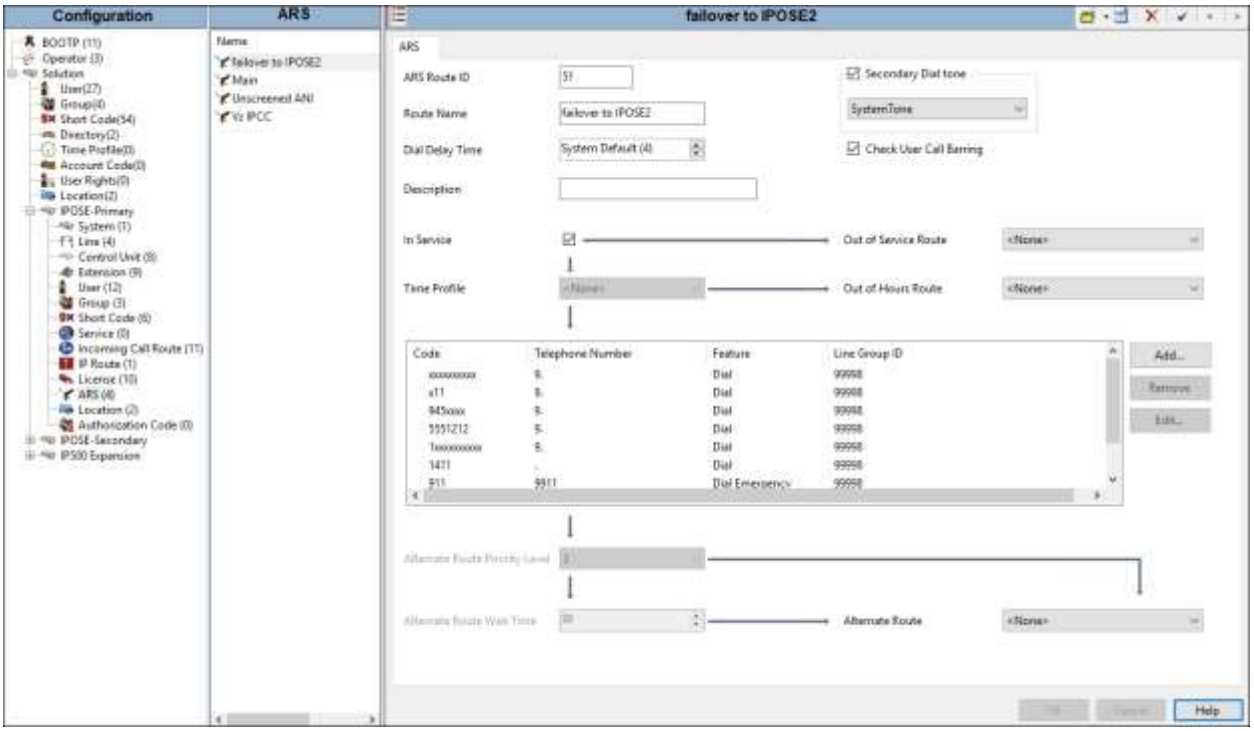
The following screen shows an example ARS configuration for the route named "**failover to IPOSE2**", **ARS Route ID** "**51**". Continuing the example, if the user dialed 9-303-538-1000, and the call could not be routed via the primary route "50: Main" described above, the call will be delivered to this "alternate" route. Per the configuration shown below, the call will be delivered to Line Group 99998 using the IP Office Line to the Secondary server (**Section 5.5**). The telephone number presented would be prefixed with "9" so the call can be analyzed using the ARS short code created on the Secondary server.



If a primary route experiences a network outage such that no response is received to an outbound INVITE, IP Office successfully routes the call via the alternate route. The user receives an audible tone when the re-routing occurs and may briefly see "Waiting for Line" on the display.

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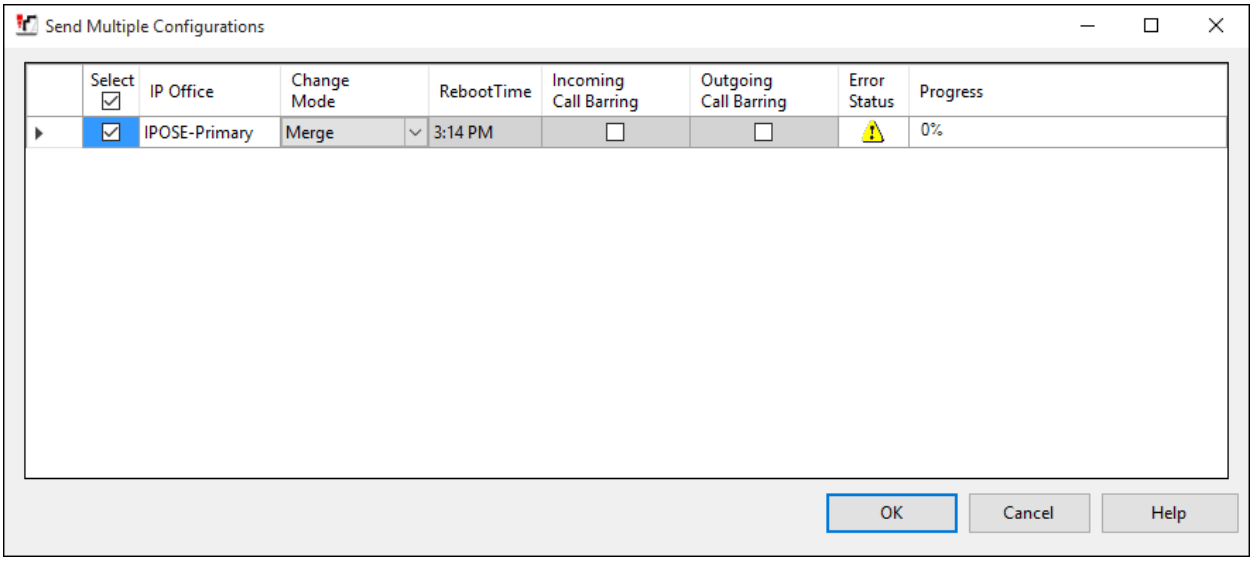
5.10. Save Configuration

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.



6. Avaya IP Office Secondary Configuration

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. Clicking the "plus" sign next to **IPOSE-Secondary** on the left navigation pane will expand the menu on this server.

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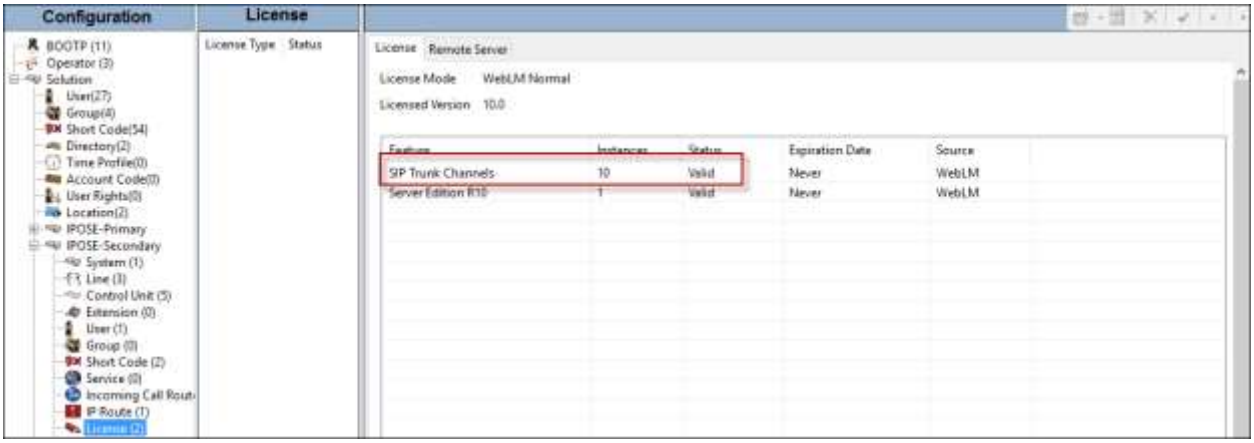
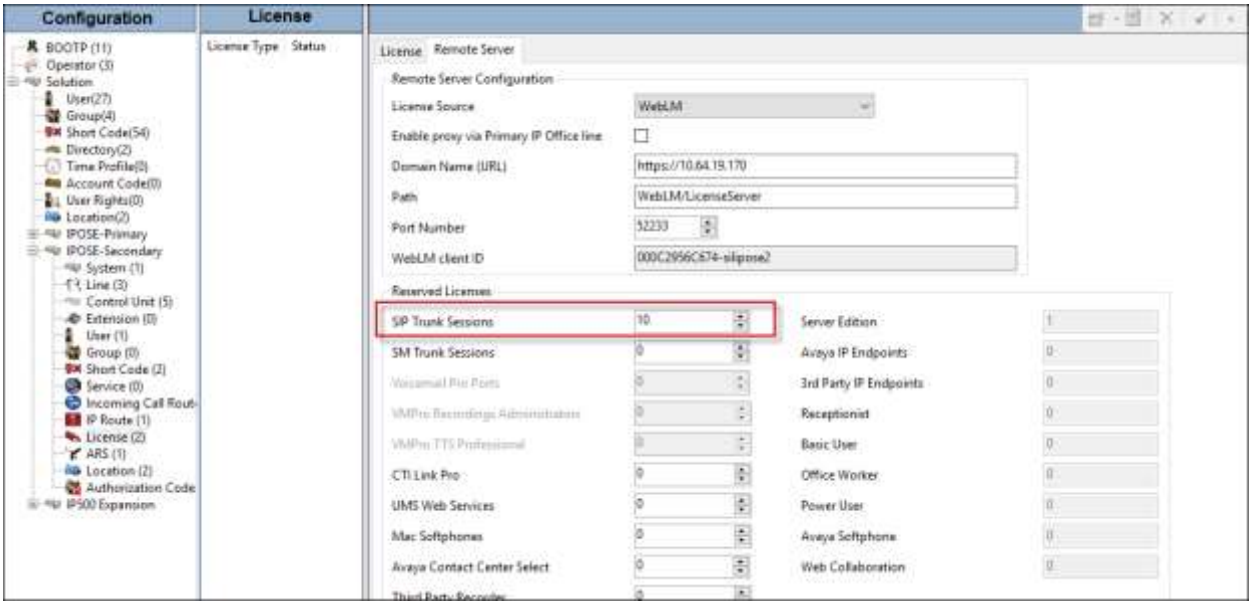
6.1. Licensing

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).





6.2. 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.

6.2.1. LAN Settings

In the sample configuration, LAN1 is used to connect the Secondary server to the enterprise network. To view or configure the **IP Address** of LAN1, select the **LAN1** tab followed by the

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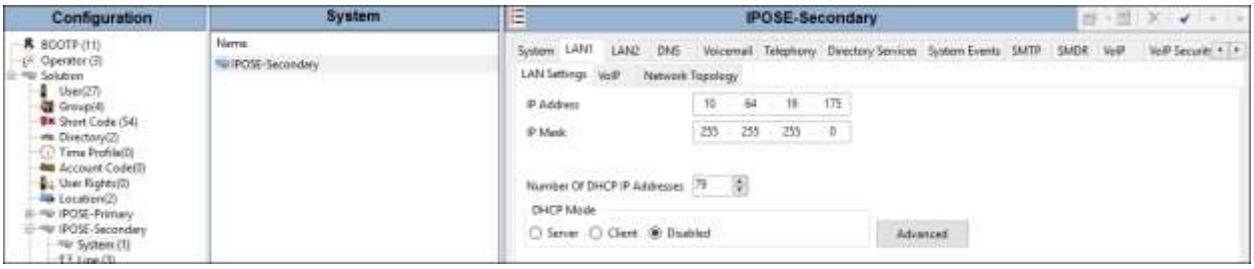
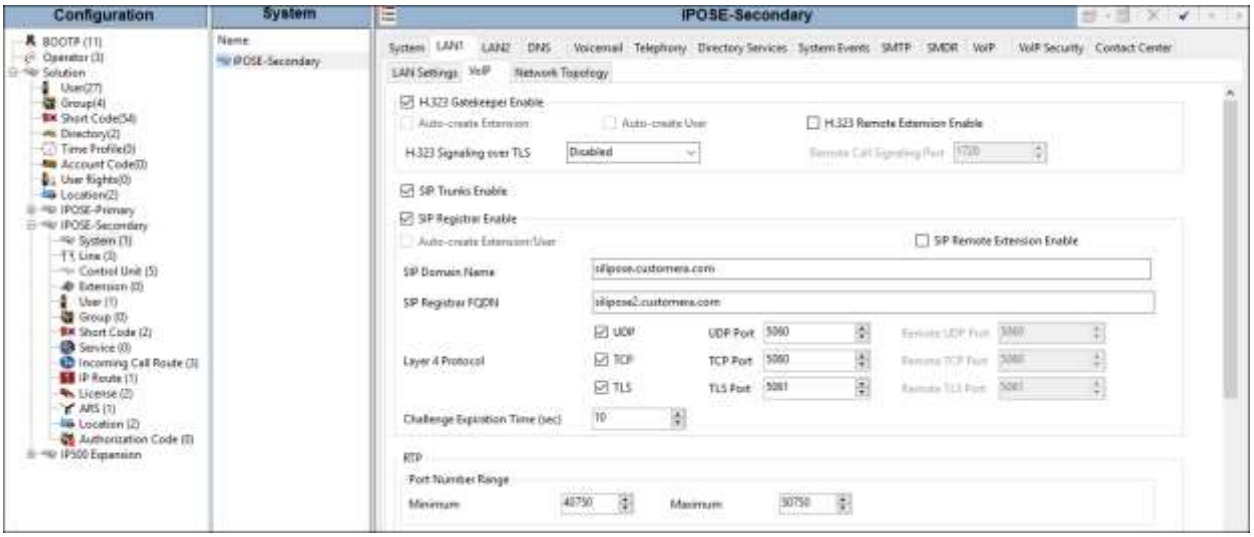
LAN Settings tab. As shown in **Figure 1**, the IP Address of the IP Office is **10.64.19.175**. Other parameters on this screen may be set according to customer requirements.



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.

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 Secondary server. The defaults are used here.



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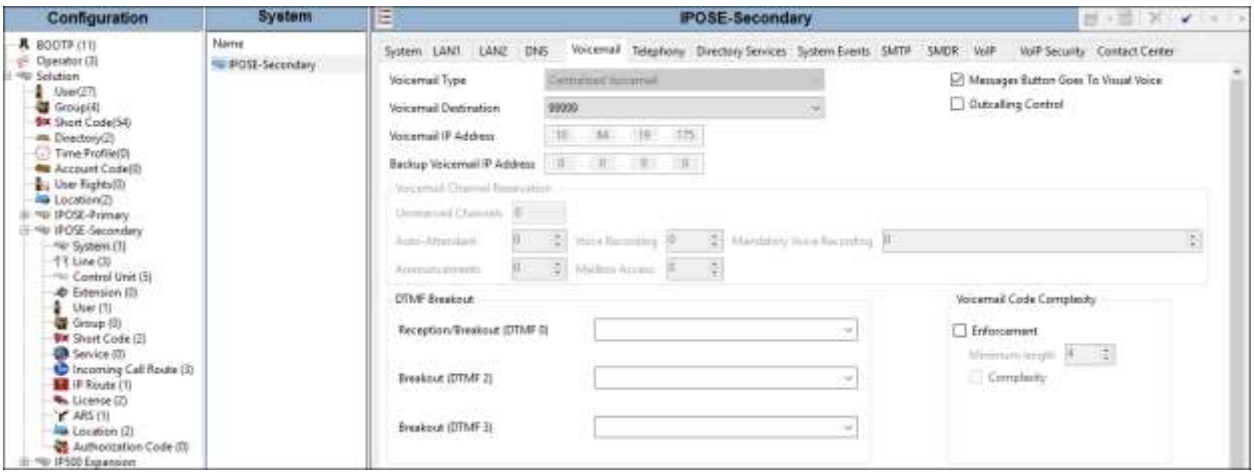
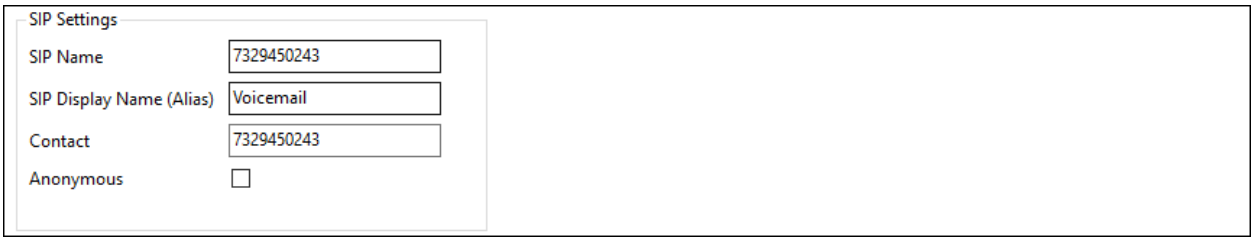
6.2.2. Voicemail Settings

To view or change voicemail settings, select the **Voicemail** tab as shown in the following screen. The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. The **Voicemail Type** in the sample configuration is "**Centralized Voicemail**". The **Voicemail Destination** is "**99999**", the Line Group of the IP Office Line to the Primary server.



Similar to the Primary server configuration, the **SIP Settings** shown in the screen below enable the Secondary server to populate the SIP headers for an outbound "callback" call from Voicemail Pro, similar to the way the fields with these same names apply to calls made from telephone users (e.g., see **Section 5.5**).



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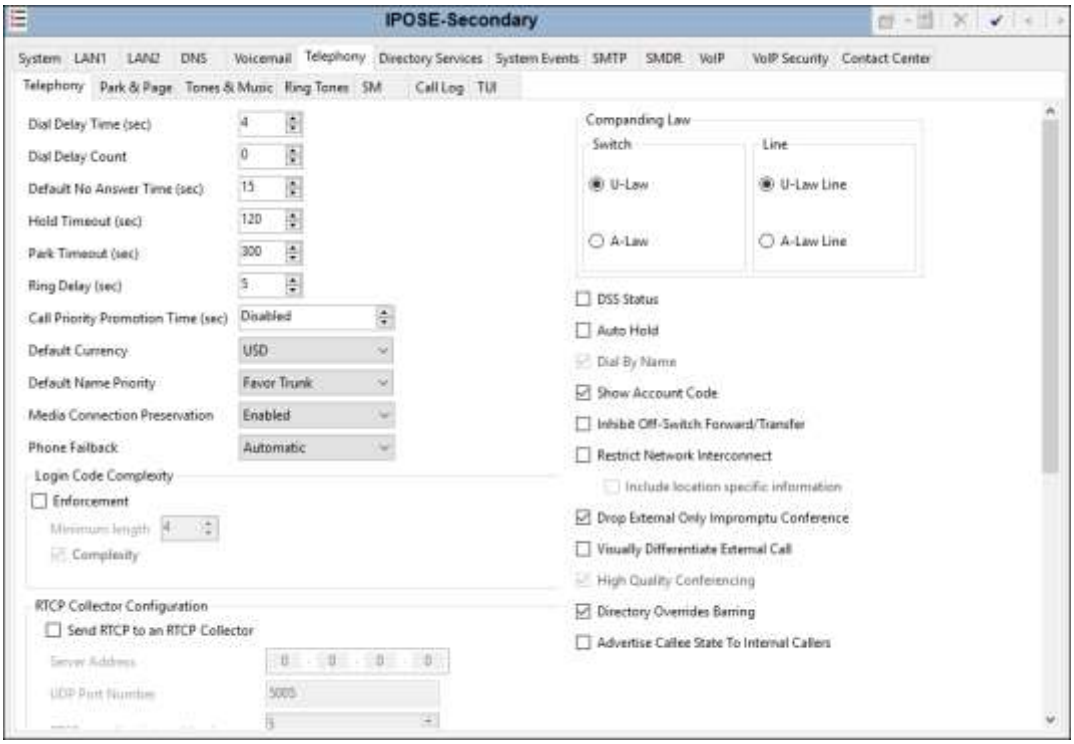
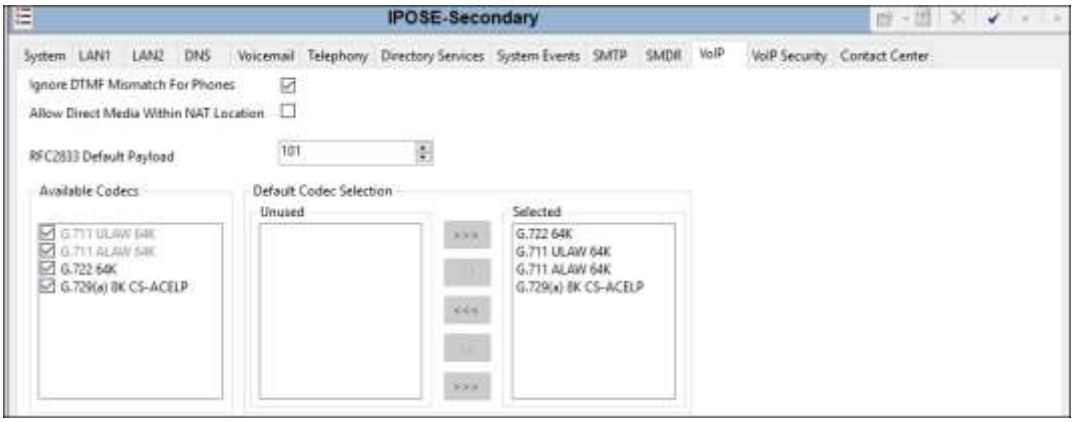
6.2.3. System Telephony Configuration

Configuration is the same as the Primary server, as shown in **Section 5.2.3**.



6.2.4. System Codecs Configuration

Configuration is the same as the Primary server, as shown in **Section 5.2.4**.



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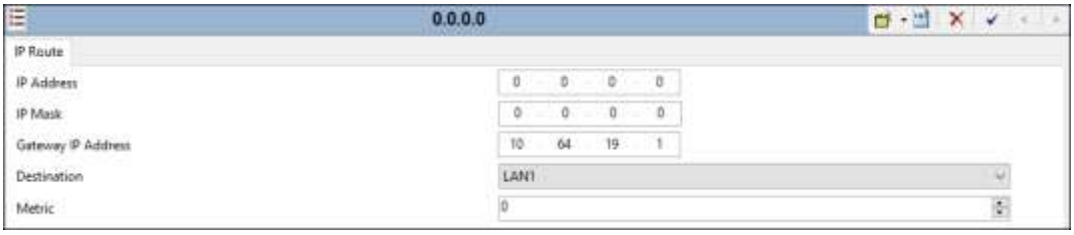
6.3. IP Route

Configuration is the same as the Primary server, as shown in **Section 5.3**.

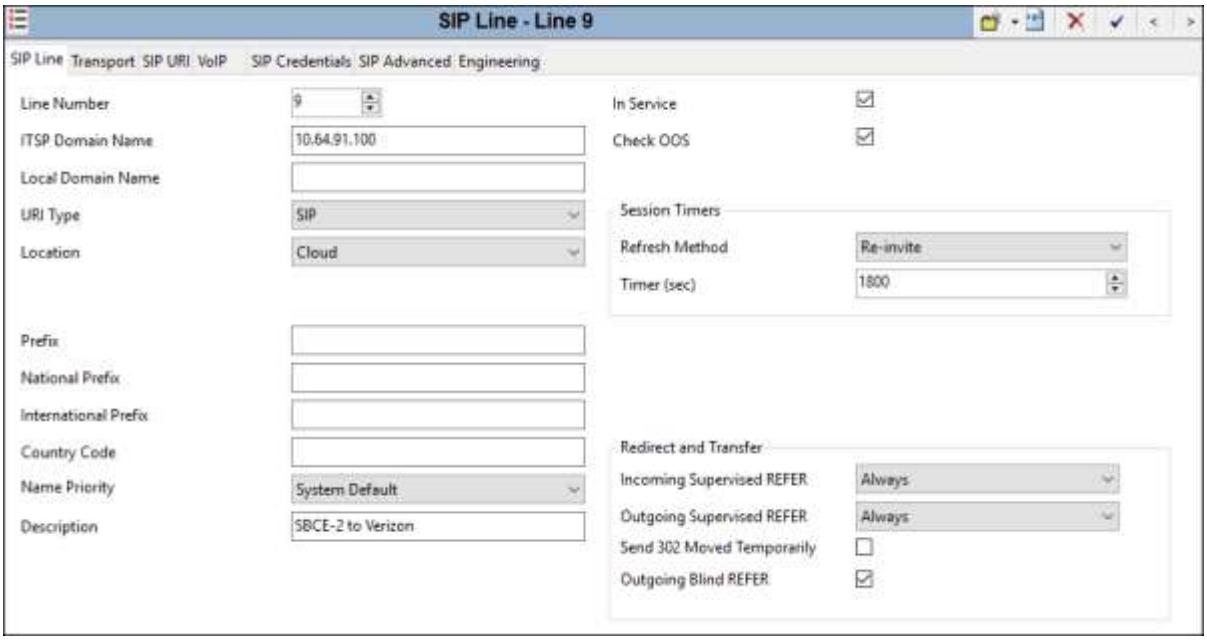


6.4. SIP Line

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.

6.4.1. SIP Line – SIP Line Tab

Same as the Primary server, with the exception of the **ITSP Domain Name** is set to the secondary Avaya SBCE internal IP address.



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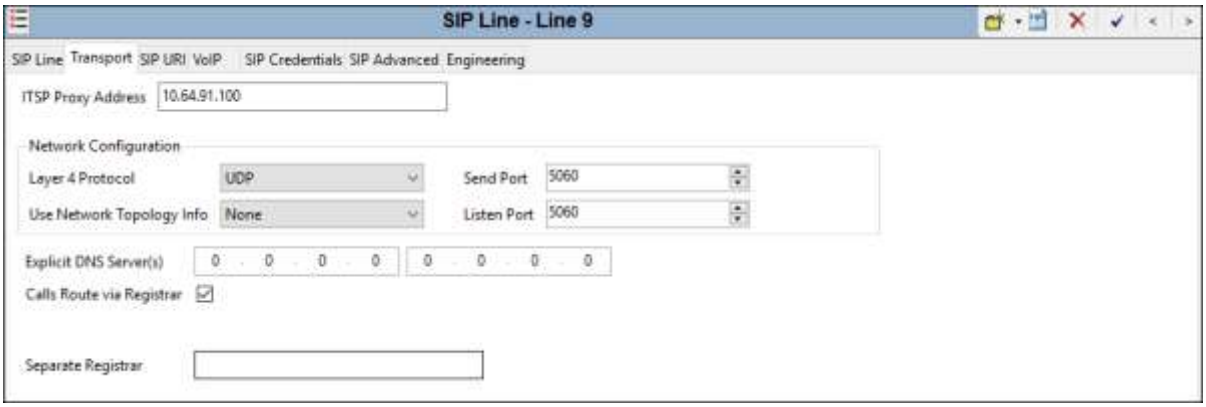
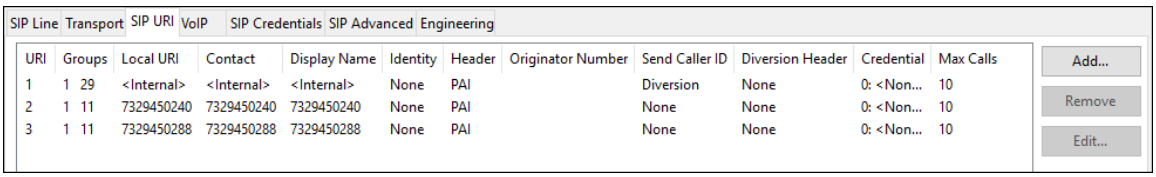
6.4.2. SIP Line - Transport Tab

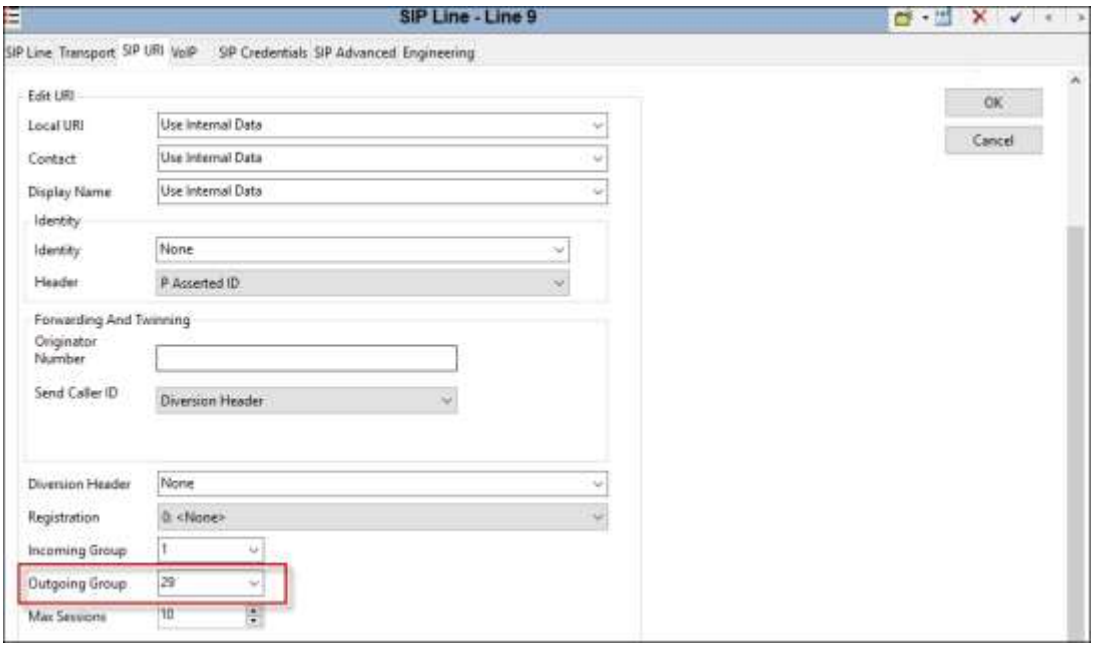
Select the **Transport** tab. The **ITSP Proxy Address** is set to the inside IP address of the secondary Avaya SBCE as shown in **Figure 1**. Remaining field are the same as the Primary server.



6.4.3. SIP Line - SIP URI Tab

Create URI entries similar to the Primary server. Outgoing Group IDs cannot be shared between systems, so the **Outgoing Group** is set to "**29**" for the first entry. URIs 2 and 3 will not be used for outbound calling, so an unused number is assigned for the **Outgoing Group**.





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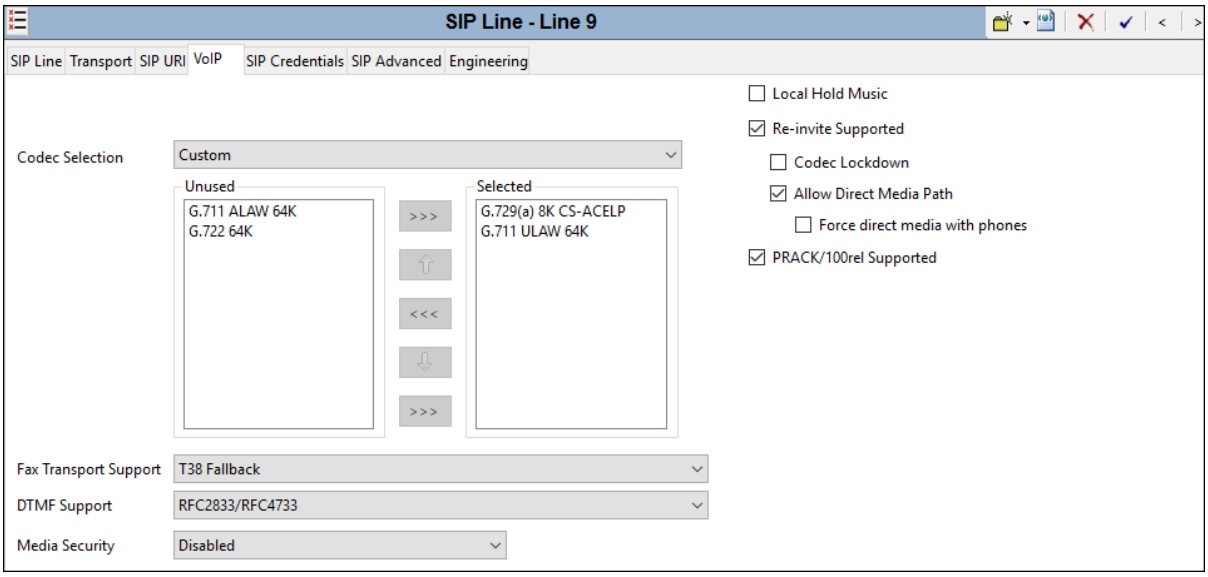
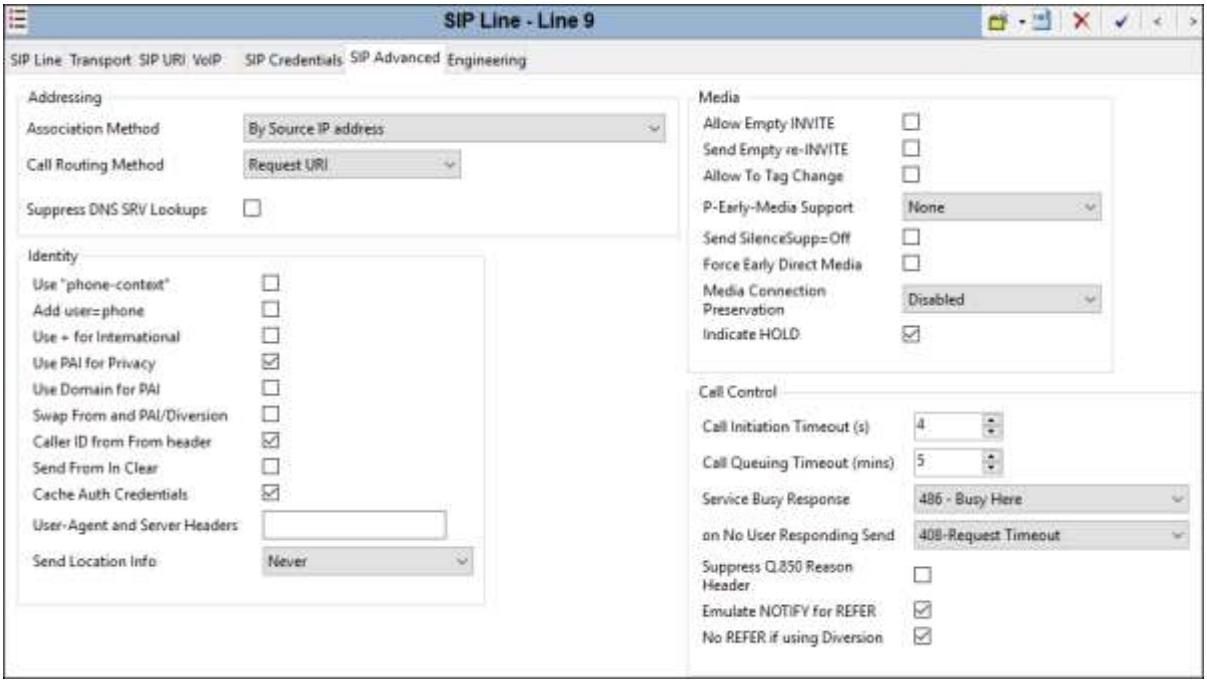
6.4.4. SIP Line - VoIP Tab

Configuration is the same as the Primary server, as shown in **Section 5.4.6**.



6.4.5. SIP Line – SIP Advanced Tab

Configuration is the same as the Primary server, as shown in **Section 5.4.7**.



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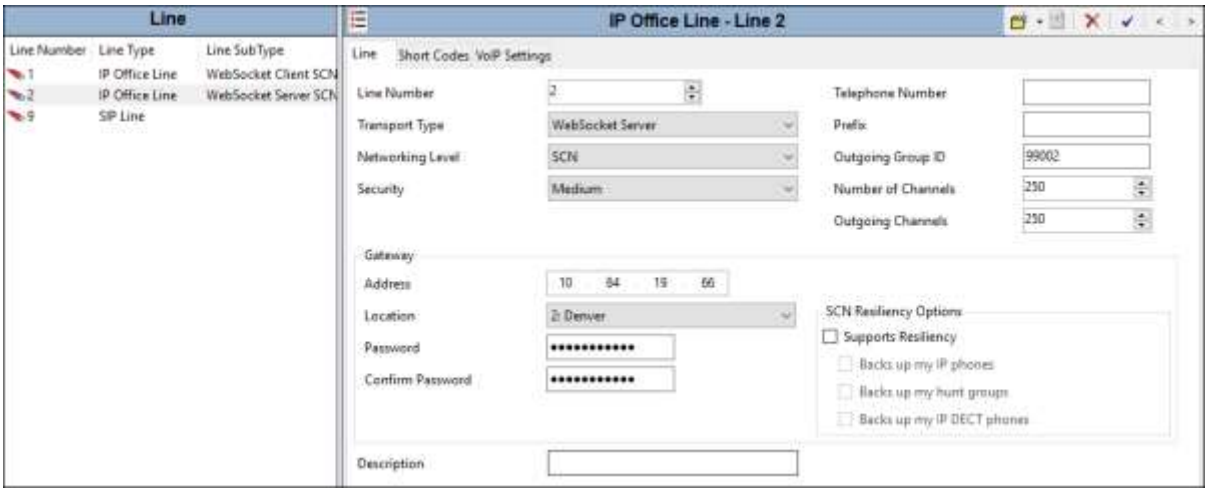
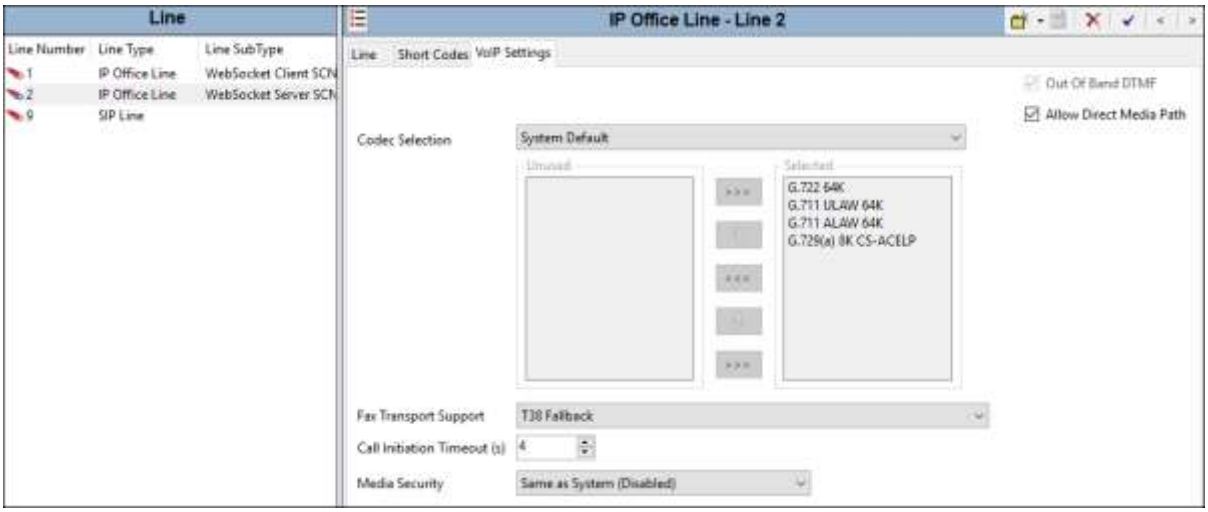
6.5. IP Office Line

The IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution.



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.



6.6. Users

In the reference configuration, there are no users created on the Secondary server. In the event of a failure of the Primary server, users will automatically register to the Secondary server as guest users.

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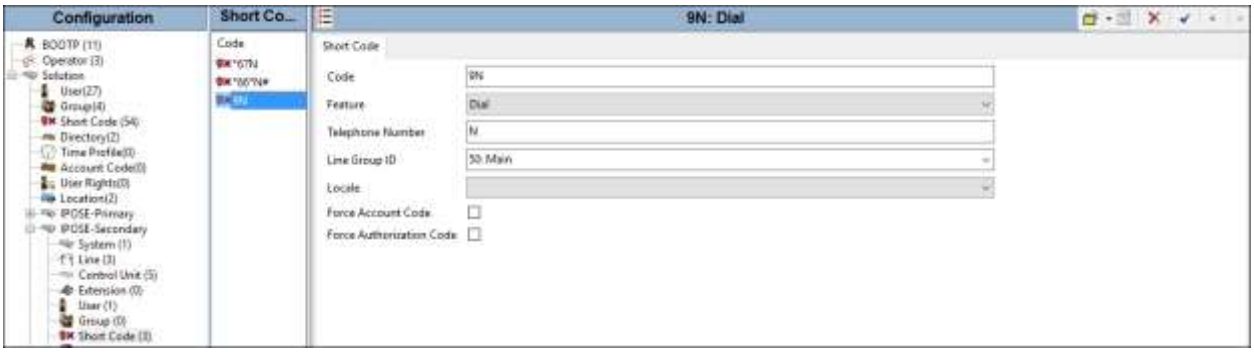
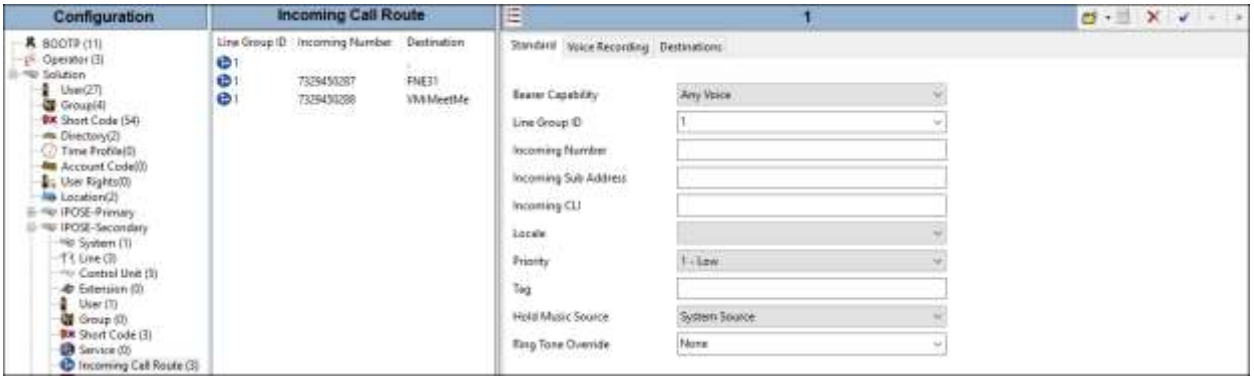
6.7. Short Codes

Similar to the configuration of the Primary server in **Section 5.7**, create a Short Code to access ARS.



6.8. Incoming Call Routes

Similar to the configuration of the Primary server in **Section 5.8**, create Incoming Call Routes for **Line Group ID** "**1**".



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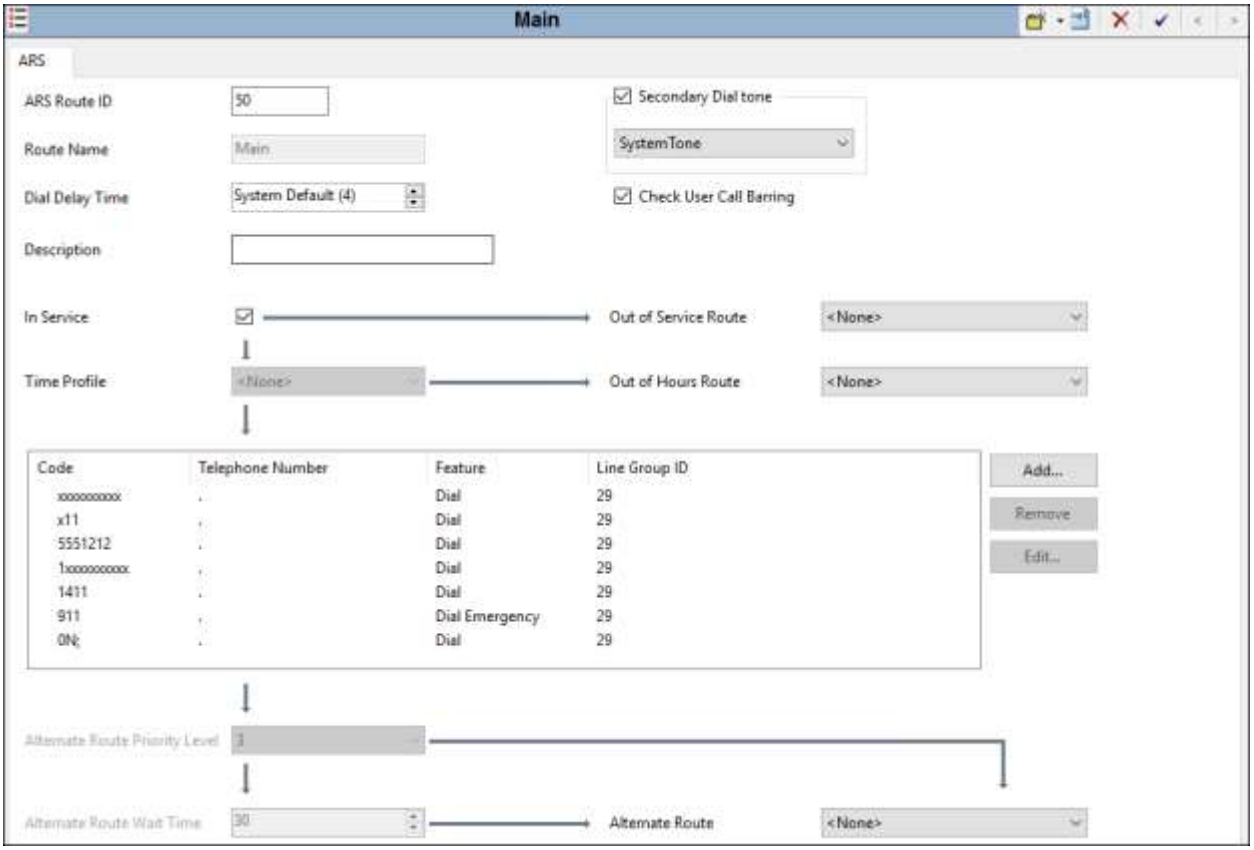
6.9. ARS

The following screen shows an example ARS configuration for the route named **Main** on the Secondary server. The **Line Group ID** is set to "**29**" matching the number of the **Outgoing Group** configured on the **SIP URI** tab of SIP Line 9 to Verizon Business (**Section 6.4.3**).



6.10. 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.

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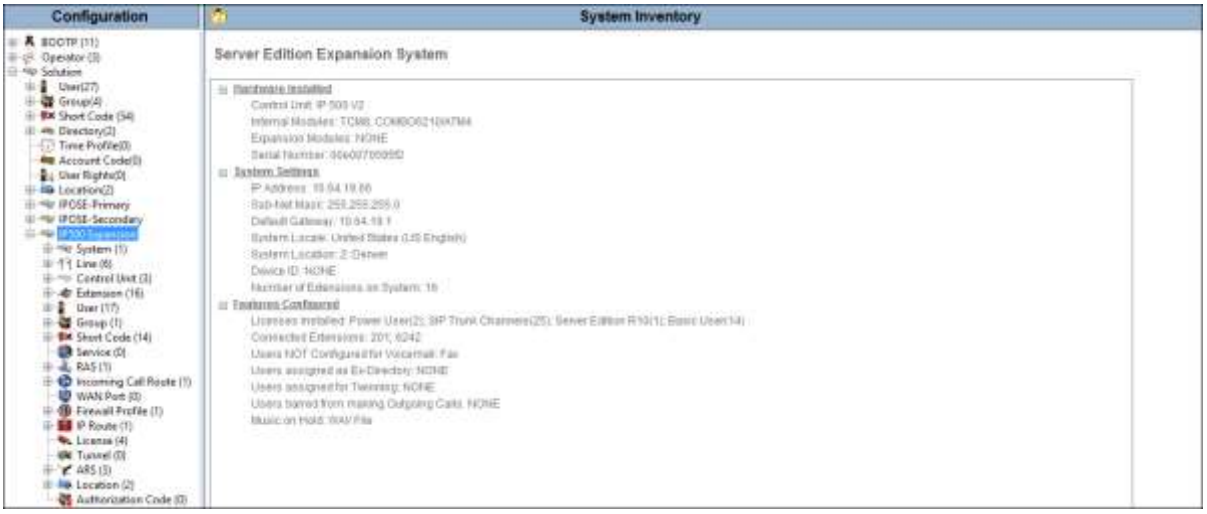
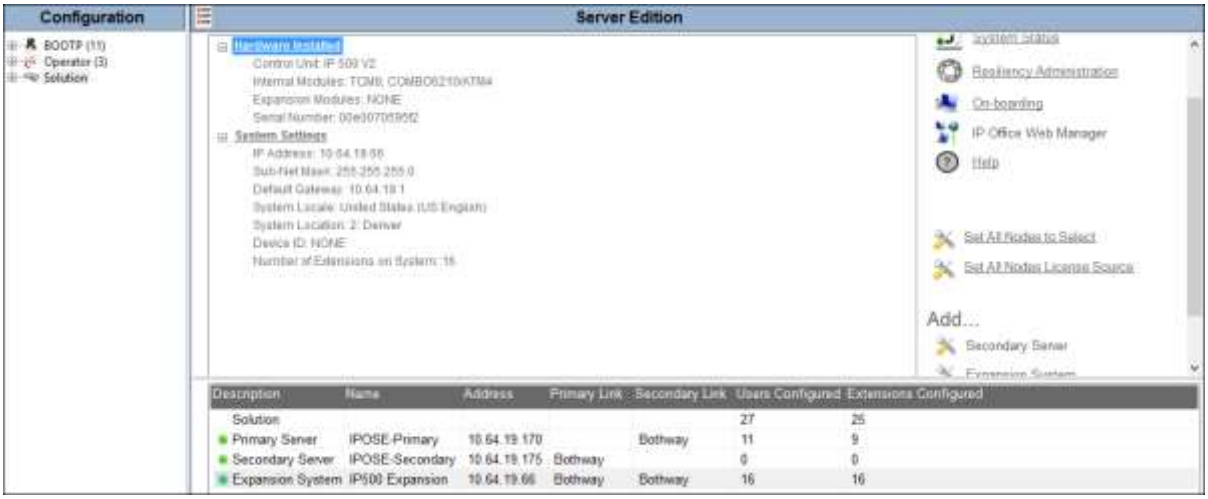
7. Avaya IP Office Expansion Configuration

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. Clicking the "plus" sign next to **IP500 Expansion** on the left navigation pane will expand the menu on this server.



7.1. Physical Hardware

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



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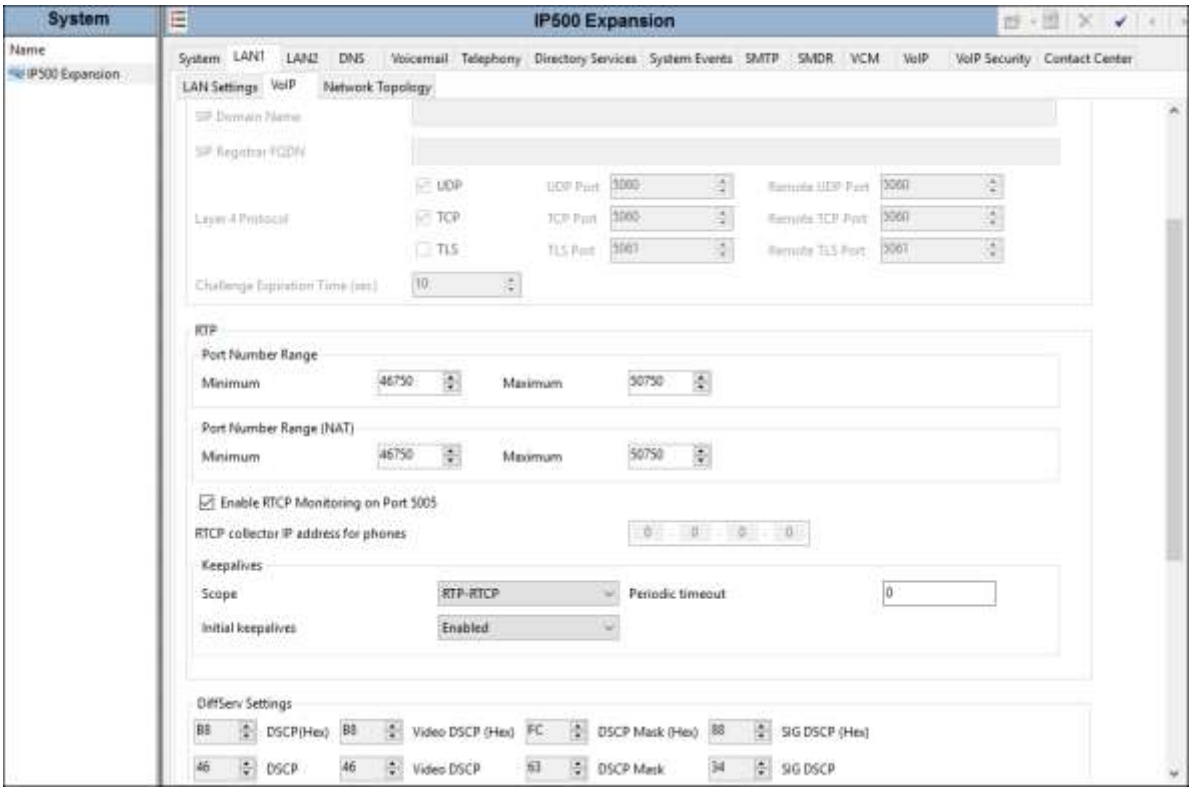
7.2. System Settings

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.

7.2.1. LAN Settings

In the sample configuration, LAN1 is used to connect the Expansion System 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 Expansion System is **10.64.19.66**. Other parameters on this screen may be set according to customer requirements.

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



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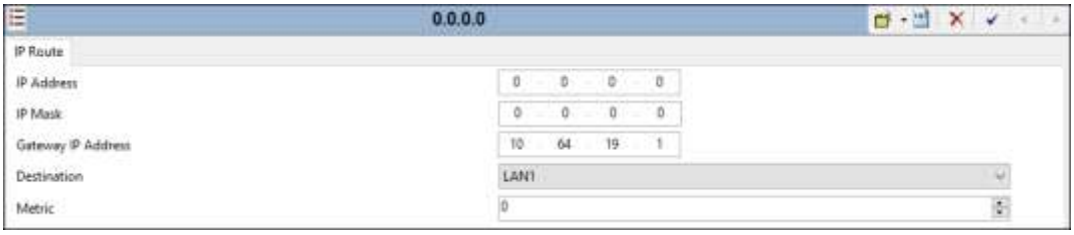
7.3. IP Route

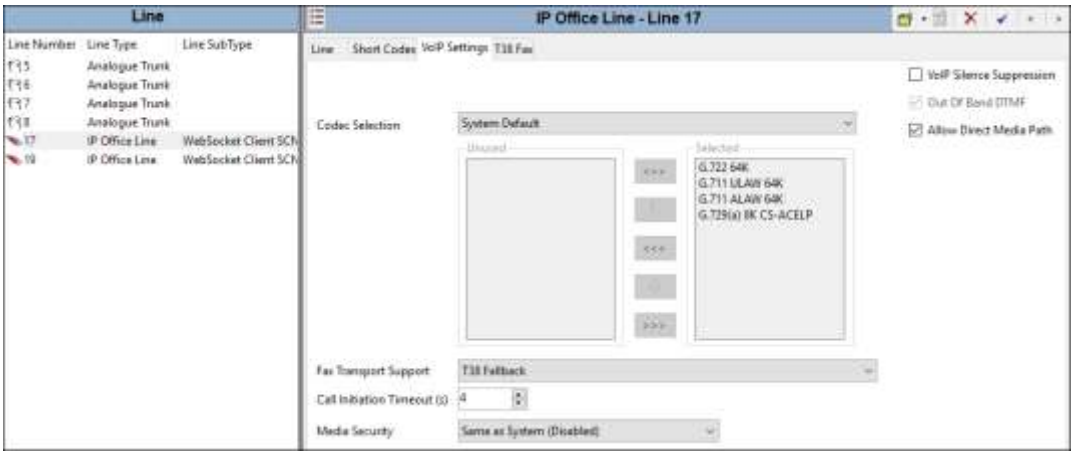
Configuration is the same as the Primary server, as shown in **Section 5.3**.



7.4. IP Office Line

The IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution. Below is the IP Office Line to the Primary server.

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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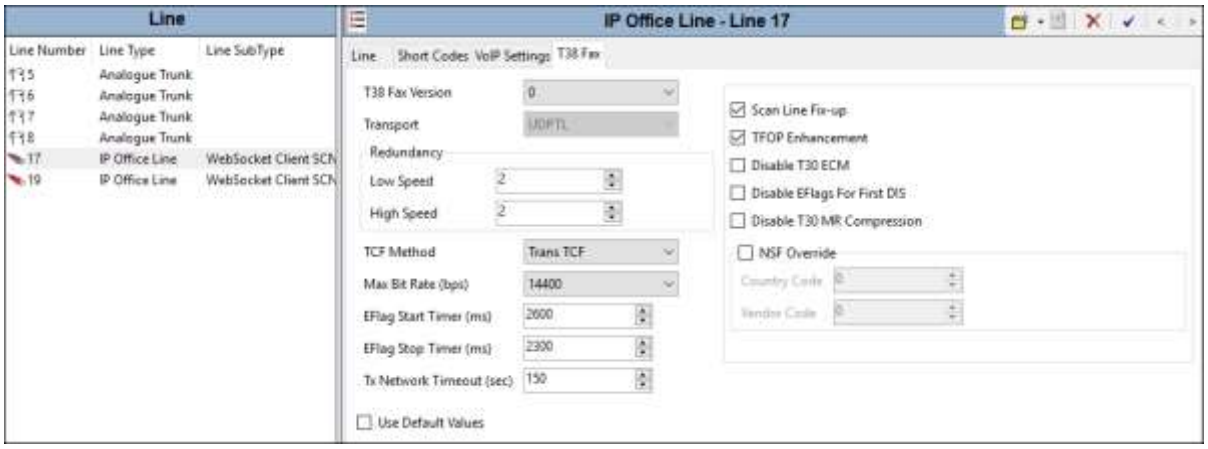
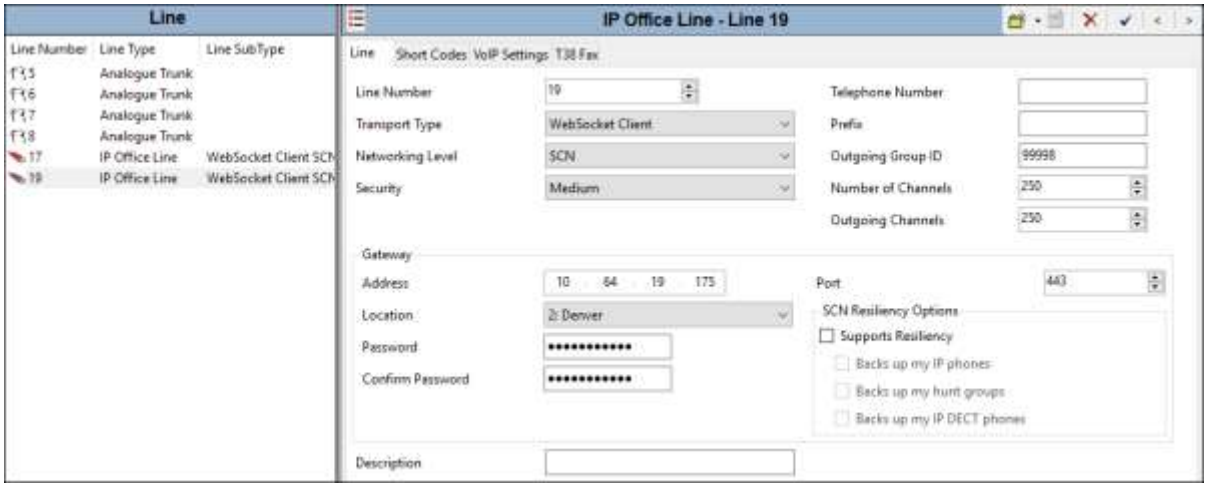
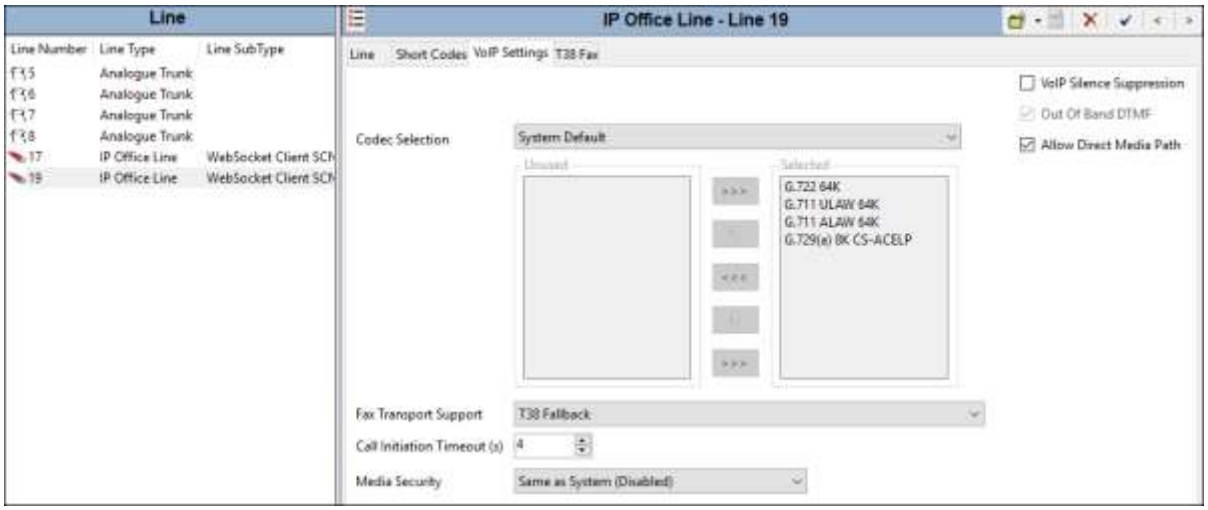
Select the **T38 Fax** tab. The **T38 Fax Version** is set to "**0**". In the **Redundancy** area, the **Low Speed** and **High Speed** parameters are set to "**2**". All other values are left at default.



The following screens illustrate the IP Office Line to the Secondary server:





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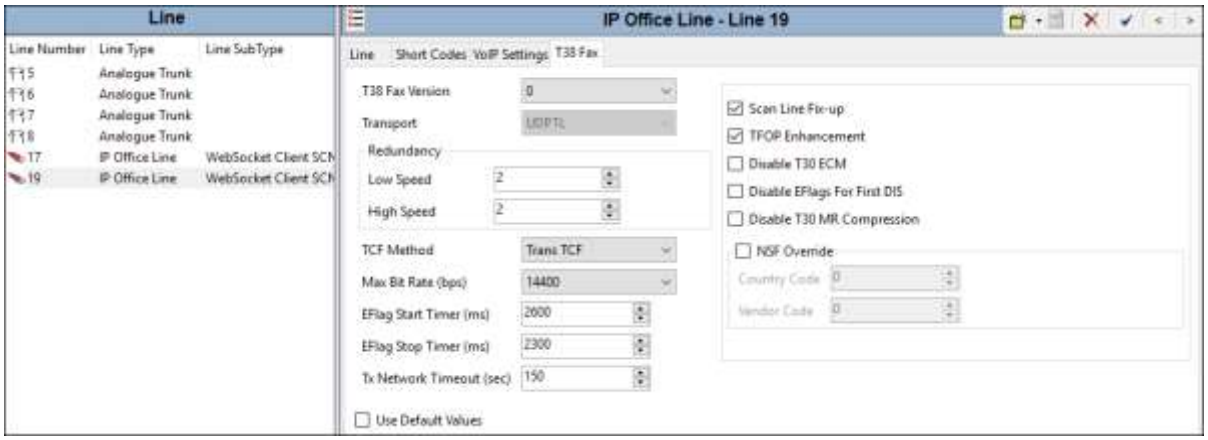
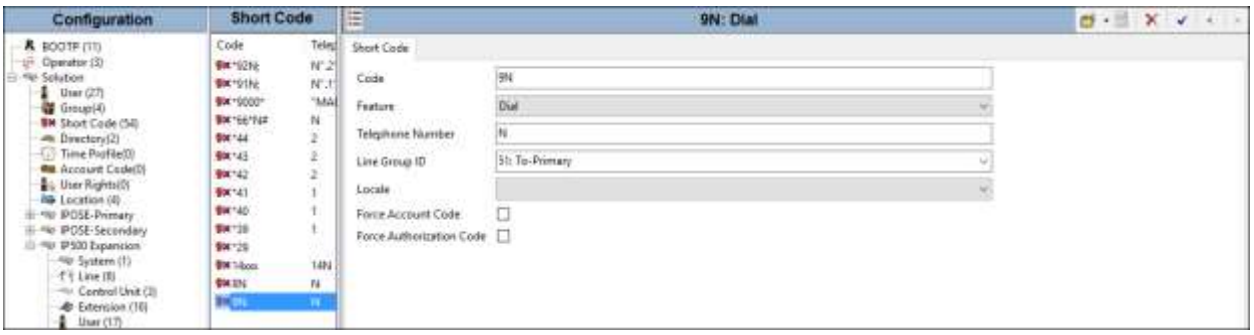


7.5. Short Codes

Similar to the configuration of the Primary server in **Section 5.7**, create a Short Code to access ARS. In the reference configuration, the **Line Group ID** is set to an ARS route illustrated in the next section.



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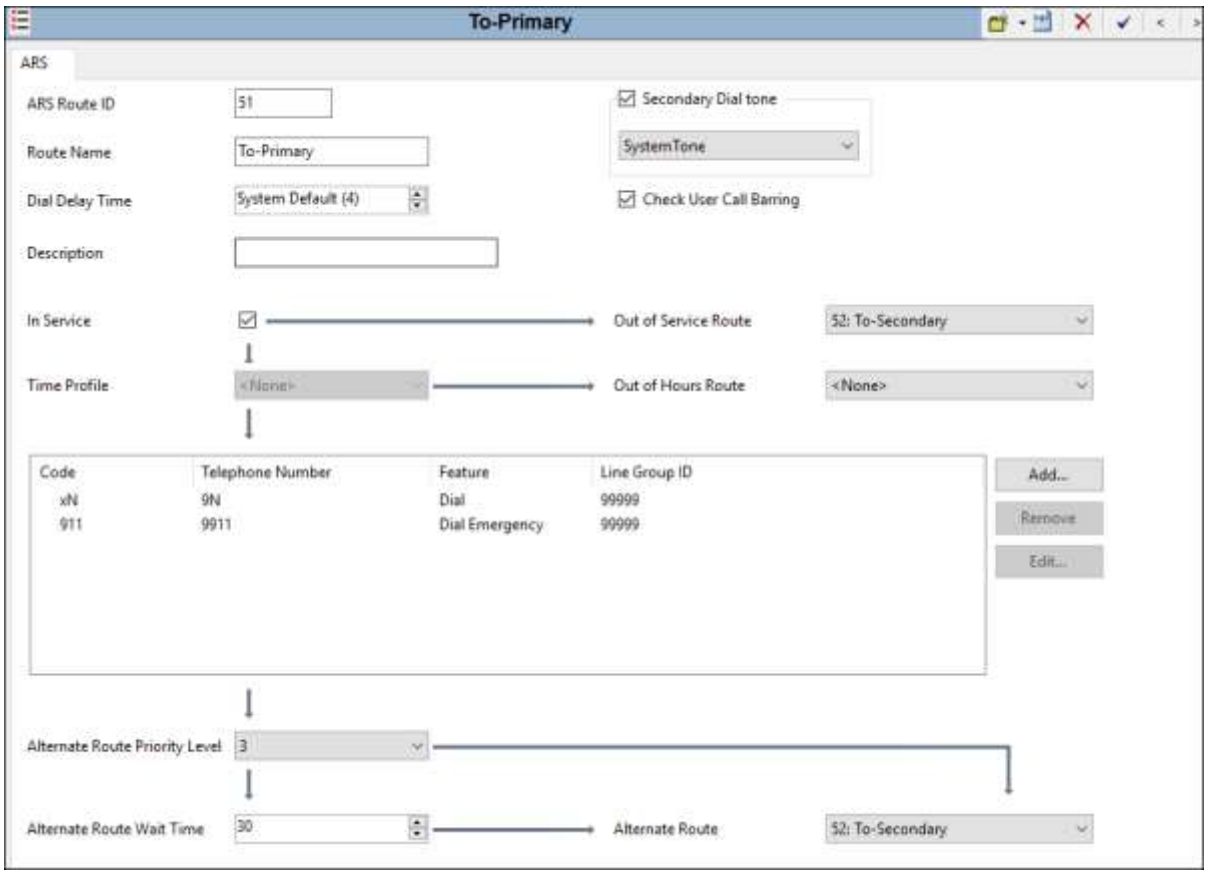
7.6. ARS

The following screen shows an example ARS configuration for the route named "**To-Primary**" on the Expansion System. The **Line Group ID** is set to "**99999**" matching the number of the **Outgoing Group** configured on the IP Office Line 17 to the Primary server (**Section 7.4**). The **Alternate Route** is set to "**To-Secondary**" to route calls to the Secondary server in the event the Primary server is unavailable.



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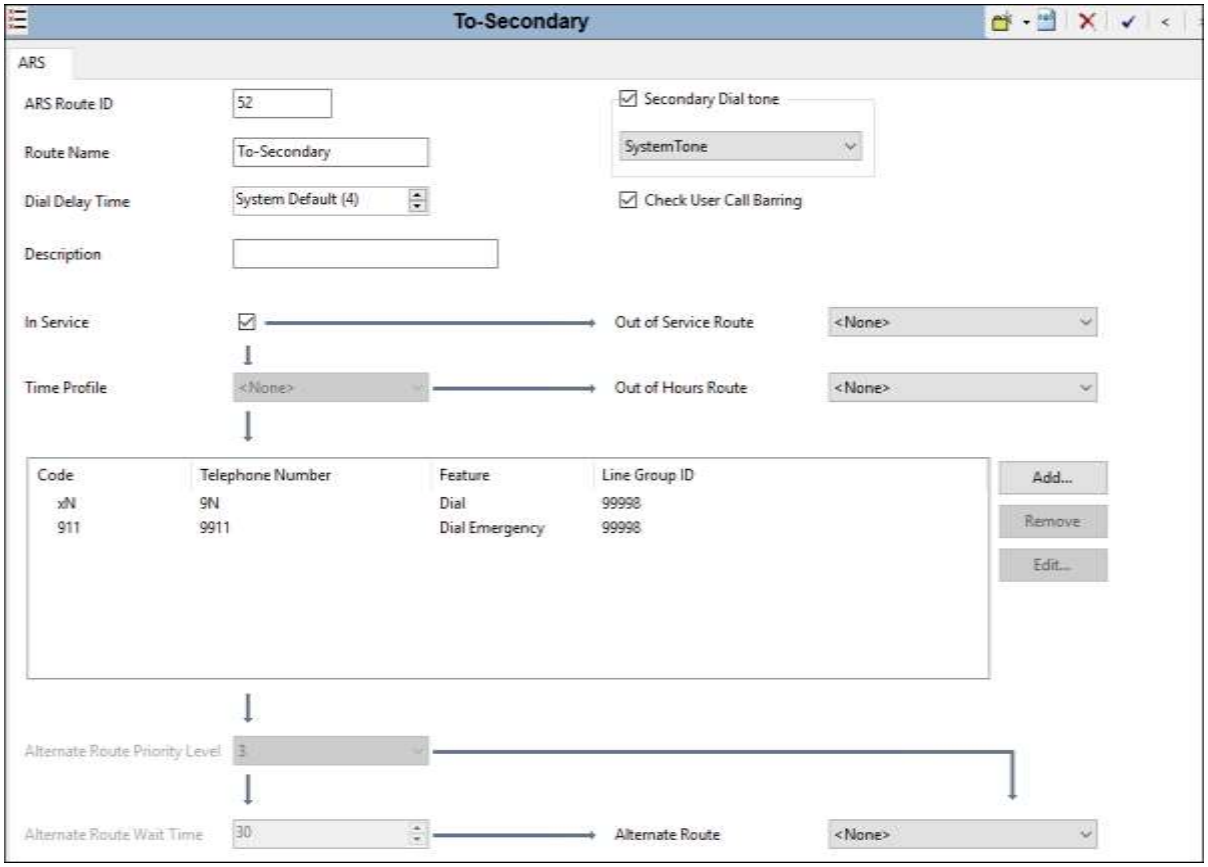
The following screen shows an example ARS configuration for the route named "**To-Secondary**" on the Expansion System. The **Line Group ID** is set to "**99998**" matching the number of the **Outgoing Group** configured on the IP Office Line 19 to the Secondary server (**Section 7.4**).



7.7. 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.

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8. Configure Avaya Session Border Controller for Enterprise

In the sample configuration, dual Avaya SBCEs are used as edge devices between the CPE and Verizon Business.

This section covers the configuration of the Avaya SBCE. It is assumed that the initial provisioning of the Avaya SBCE, including the assignment of the management interface IP Address and license installation have already been completed; hence these tasks are not covered in these Application Notes. For more information on the installation and provisioning of the Avaya SBCE consult the Avaya SBCE documentation in the **Additional References** section.

As described in **Section 1**, Verizon Business IP Trunking supports a redundant (2-CPE) architecture that provides for redundant SIP trunk access between the Verizon Business IP Trunk service offer and the SIP trunk architecture customer premises equipment (CPE). In the reference configuration two Avaya SBCEs were used to provide the 2-CPE redundant access.

Note – The following Sections describe the provisioning of the Primary Avaya SBCE. The configuration of the Secondary Avaya SBCE is identical unless otherwise noted (e.g., IP addressing).

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**.

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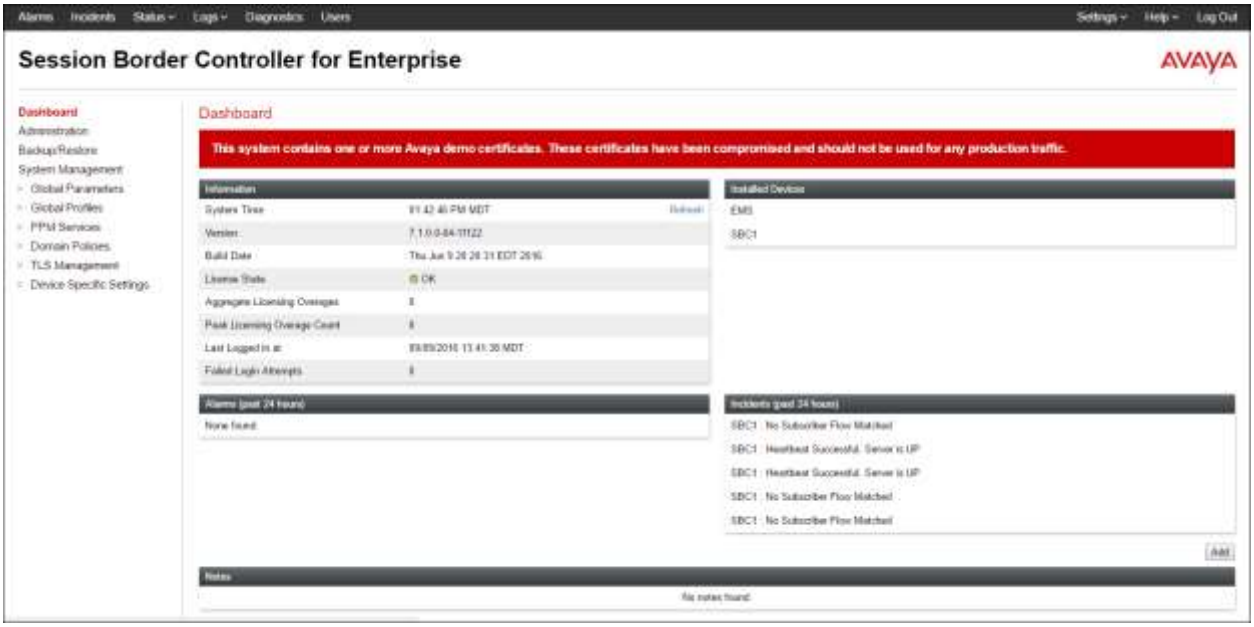
The main page of the Avaya SBCE will appear. Note that the installed software version is displayed. Verify that the **License State** is **"OK**". The SBCE will only operate for a short time without a valid license. Contact your Avaya representative to obtain a license.



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.

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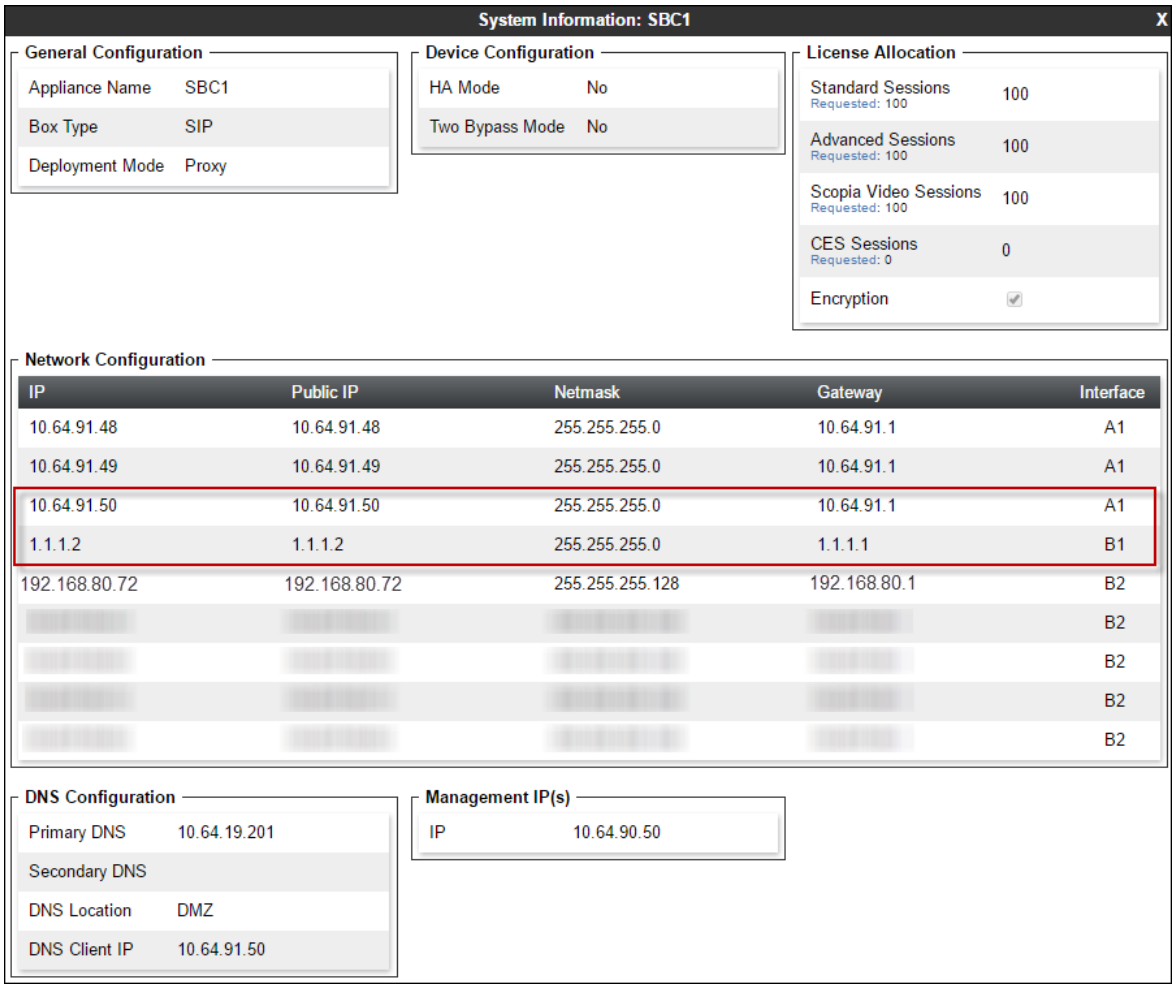
The **System Information** screen shows the **Network Configuration**, **DNS Configuration** and **Management IP(s)** information provided during installation and corresponds to **Figure 1**. The highlighted **A1** and **B1** IP addresses are the ones relevant to the configuration of the SIP trunk to Verizon. Other IP addresses assigned to these interfaces and interface **B2** on the screen below are used to support remote workers and are not the focus of these Application Notes.



8.1. 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**.

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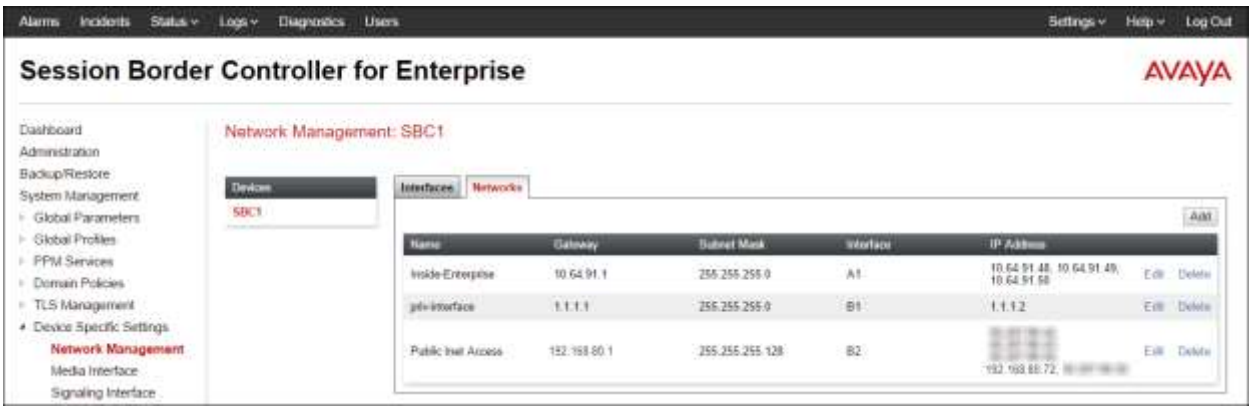
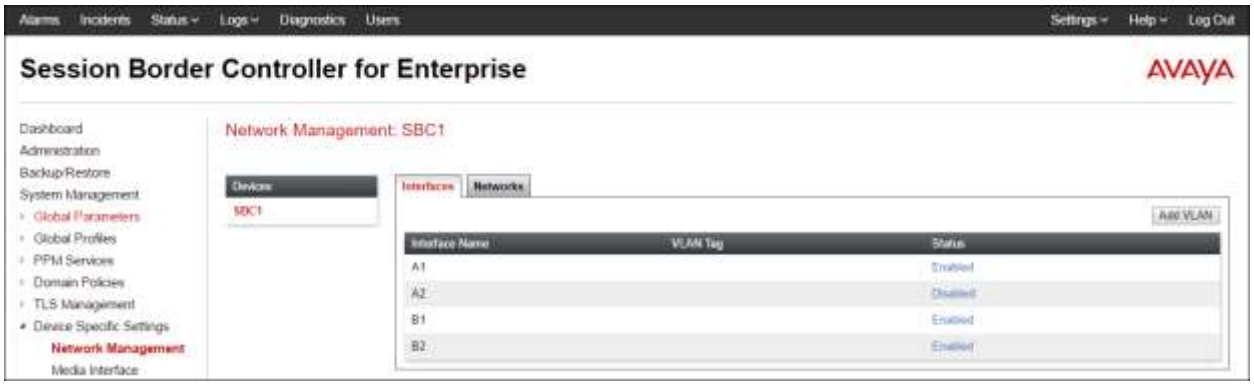
The following Avaya SBCE IP addresses and associated interfaces were used in the sample configuration:

- **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*. See **Section 3**.
- **A1**: **10.64.91.50** – IP address configured for Verizon Business IP Trunk SIP Trunk Service to IP Office.
- **A1**: **10.64.91.48** – IP address configured for Verizon Business IPCC VoIP Inbound Service to IP Office. This address is used to connect IP Office to the Verizon Business IPCC service as illustrated in [VZIPCC-IPO10SBC].
- **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.
- **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.



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**.



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8.2. Server Interworking Profile

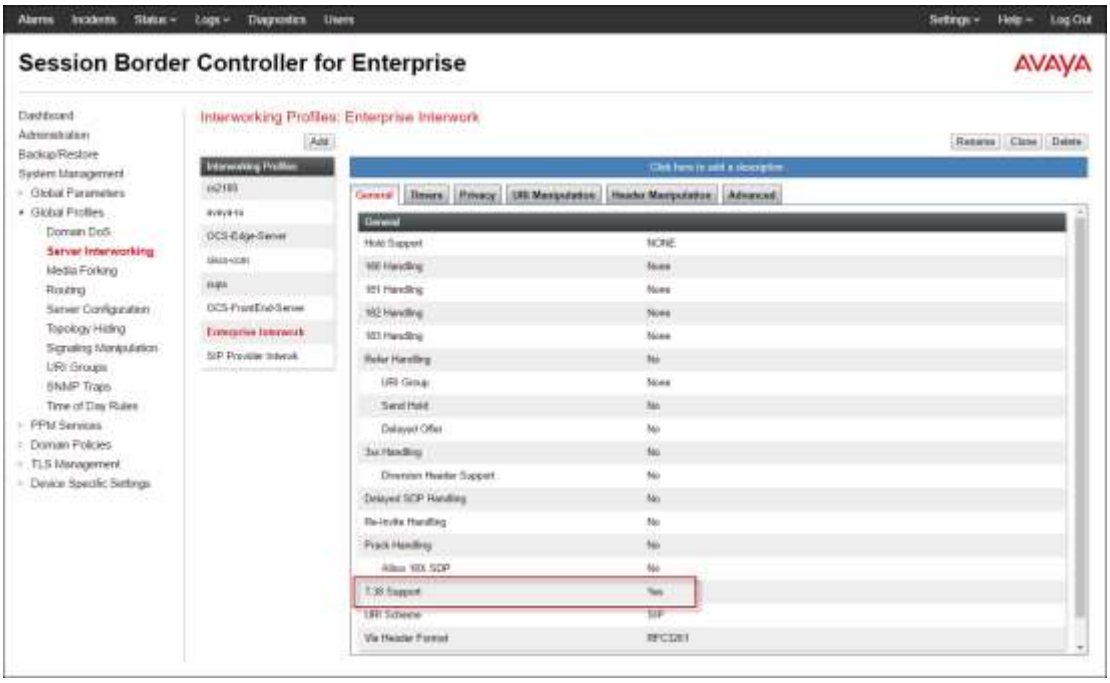
The Server Internetworking profile includes parameters to make the Avaya SBCE function in an enterprise VoIP network using different implementations of the SIP protocol. There are default profiles available that may be used as is, or modified, or new profiles can be configured as described below.

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

8.2.1. 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.

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.



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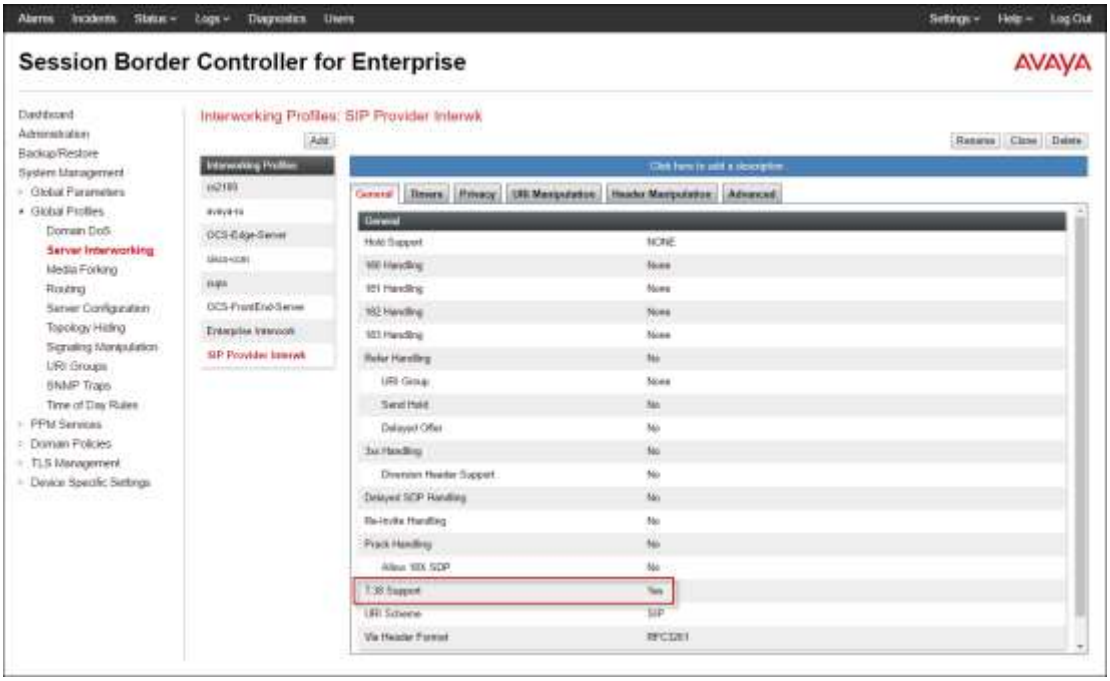
8.2.2. Server Interworking Profile – Verizon

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**.

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**".



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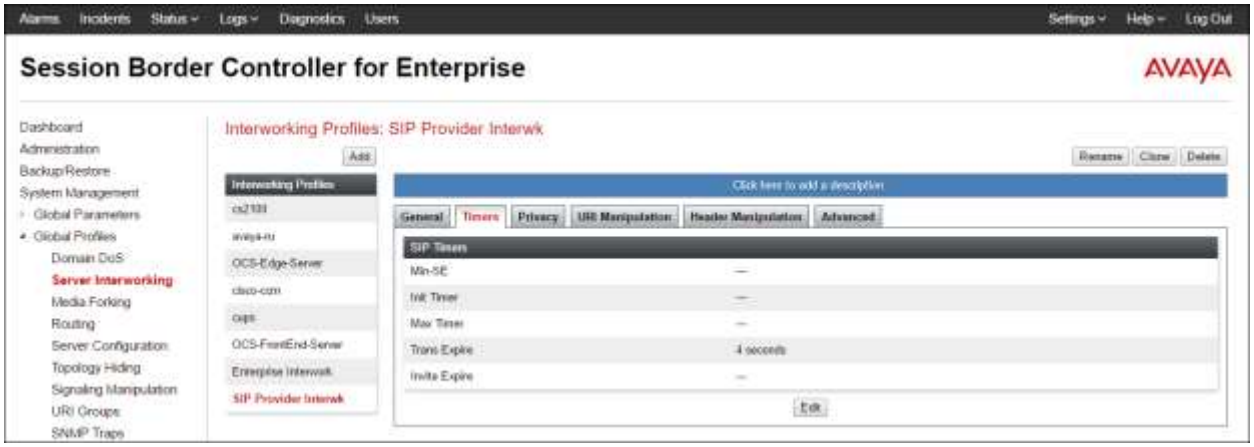
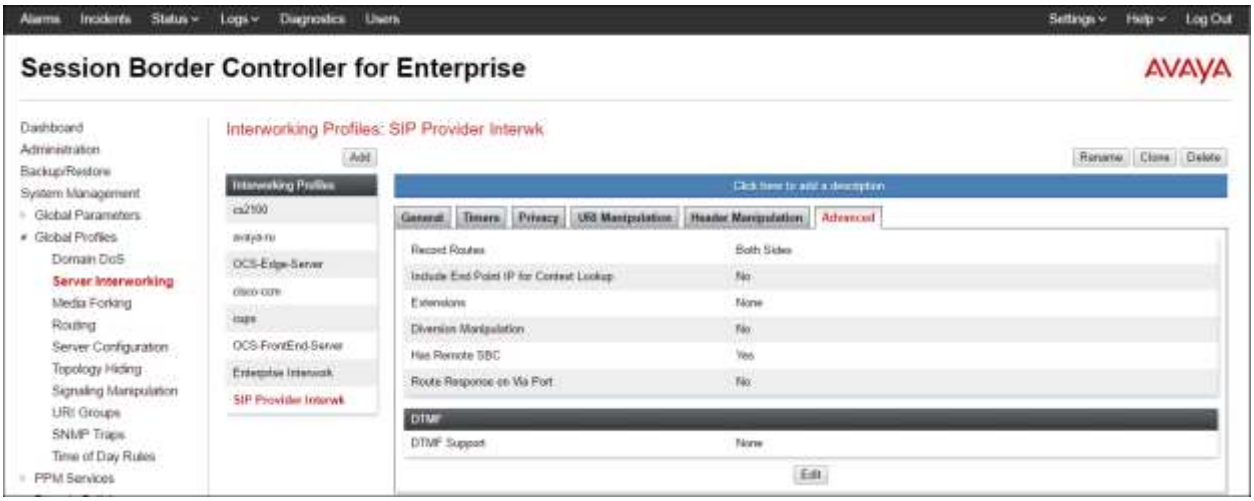
The **Timers** tab shows the values used for compliance testing for the **Trans Expire** field. The **Trans Expire** timer sets the allotted time the Avaya SBCE will try the first primary server before trying the secondary server.



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.



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8.3. Server Configuration

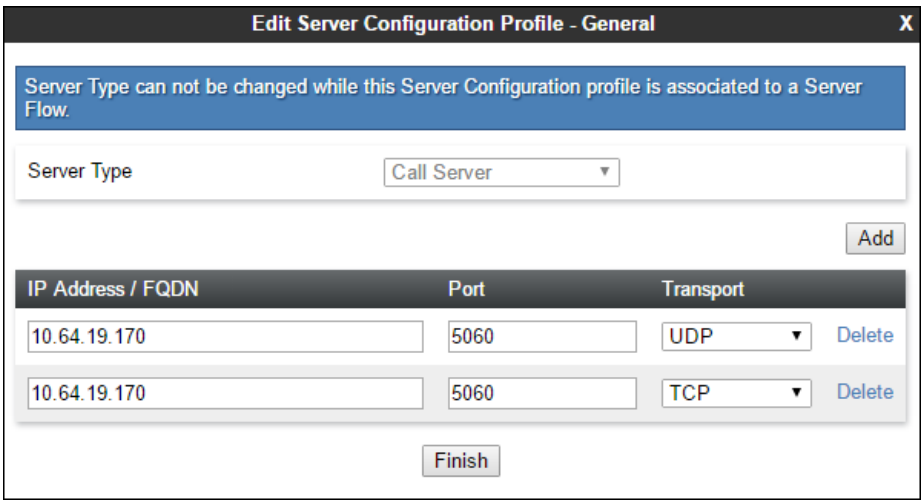
The **Server Configuration** contains parameters to configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters, DoS security statistics, and trusted domains.

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

8.3.1. 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**.



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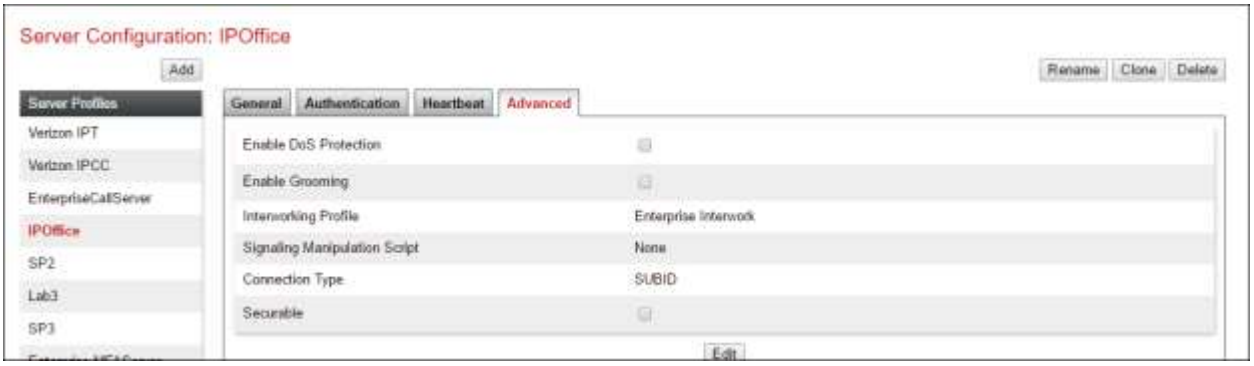
Default values can be used on the **Authentication** tab, click **Next** (not shown) to proceed to the **Heartbeats** tab. The Avaya SBCE can be configured to source "heartbeats" in the form of PINGs or SIP OPTIONS towards IP Office. When remote workers are configured, IP Office may not respond to SIP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

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 8.2.1** 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.



8.3.2. 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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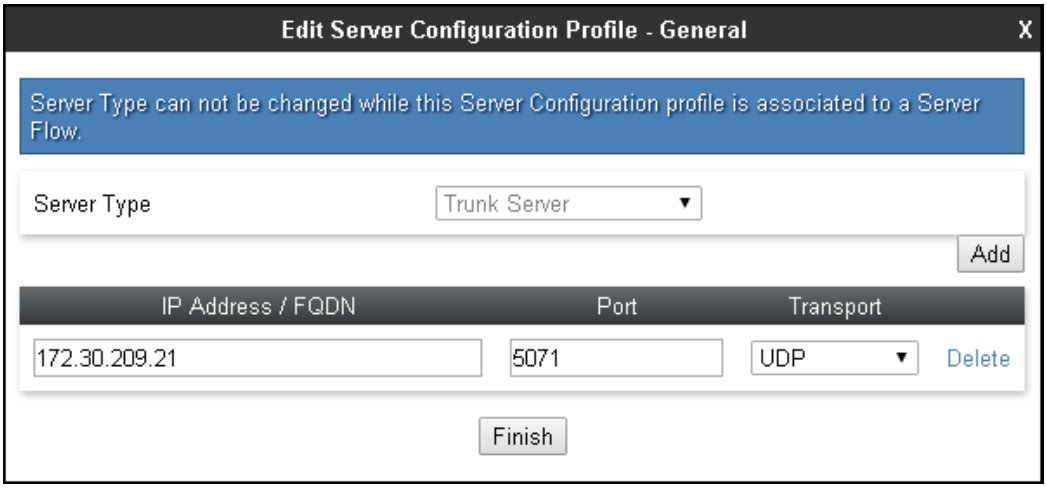
The following screens illustrate the Server Configuration for the Profile name "**Verizon IPT**". In the **General** parameters, the **Server Type** is set to "**Trunk Server**". In the **IP Address / FQDN** field, the Verizon-provided IP address is entered. This is "**172.30.209.21**". Under **Port**, "**5071**" 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 private IP Address of the Avaya SBCE, the Avaya SBCE will send SIP OPTIONS to Verizon. When Verizon 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.

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On the **Advanced** tab, **Enable Grooming** is not used for UDP connections and is left unchecked. The **Interworking Profile** is set to "**SIP Provider Interwk**" created in **Section 8.2.2** for Verizon.

8.4. Routing Profile

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

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

The following screen shows the Routing Profile "**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 8.3.1**, 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**.

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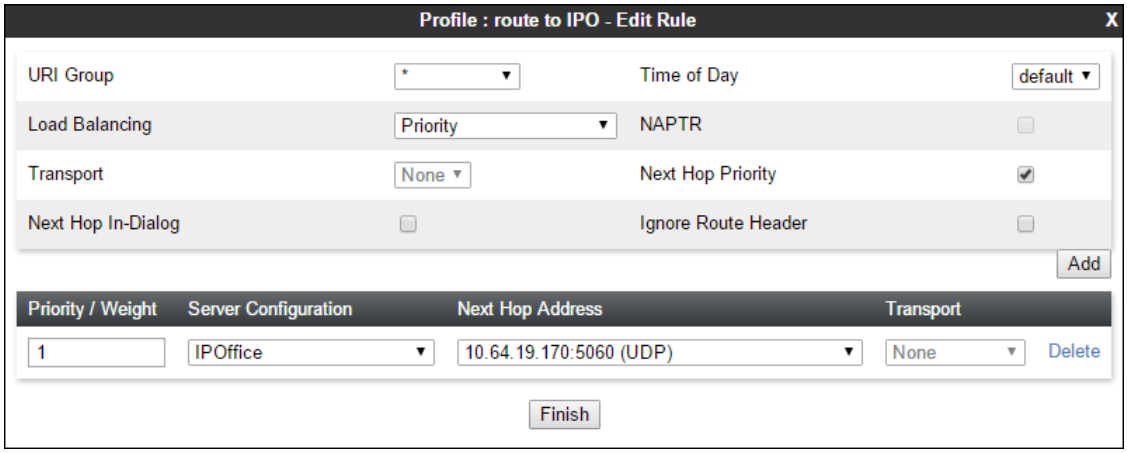
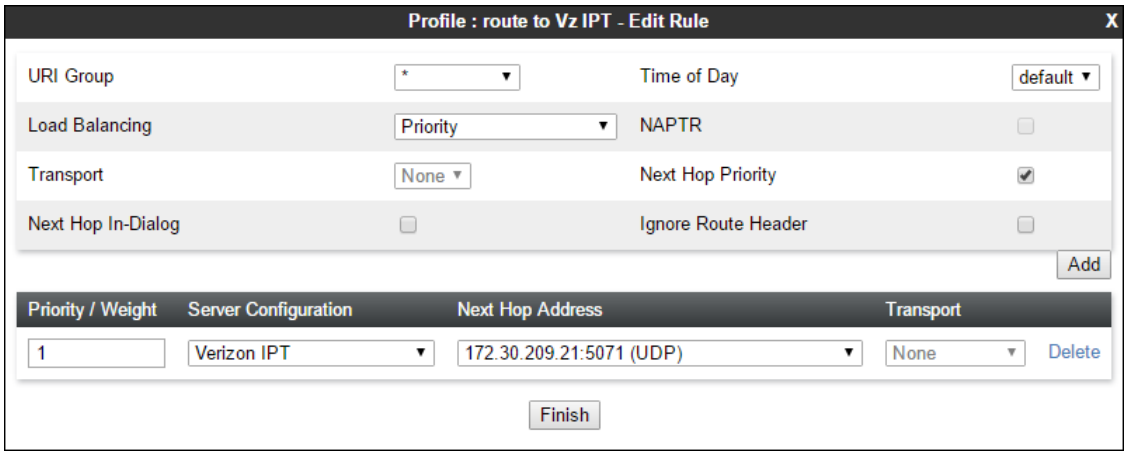


Similarly add a Routing Profile to Verizon Business IP Trunk.

The following screen shows the Routing Profile "**route to Vz IPT**" 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 8.3.2**, is selected from the drop-down menu. The **Next Hop Address** is automatically selected with the values from the Server Configuration, and **Transport** becomes greyed out. Click **Finish**.



8.5. Topology Hiding Profile

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

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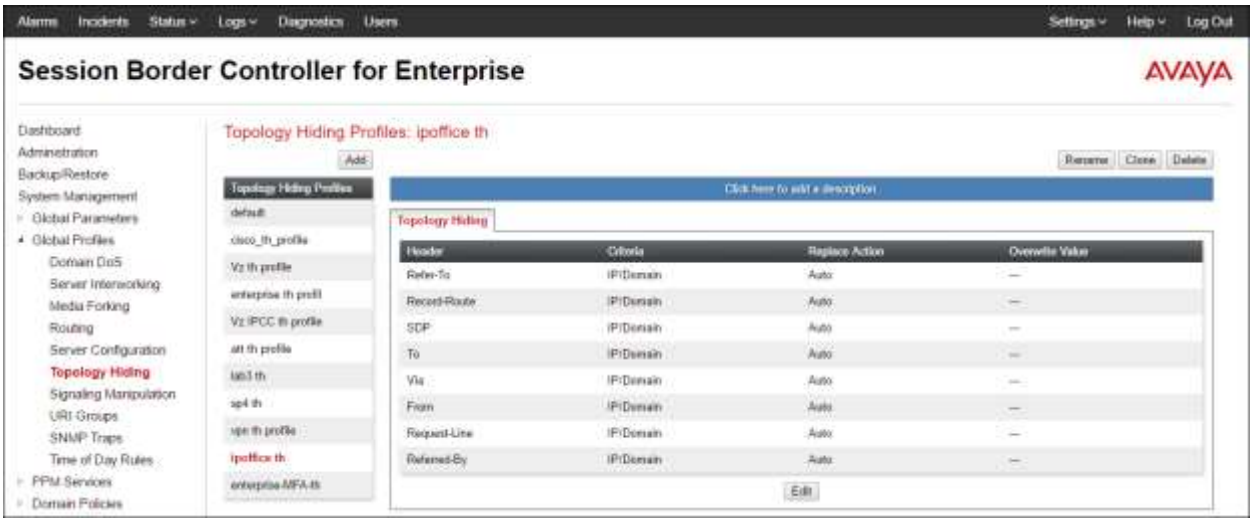
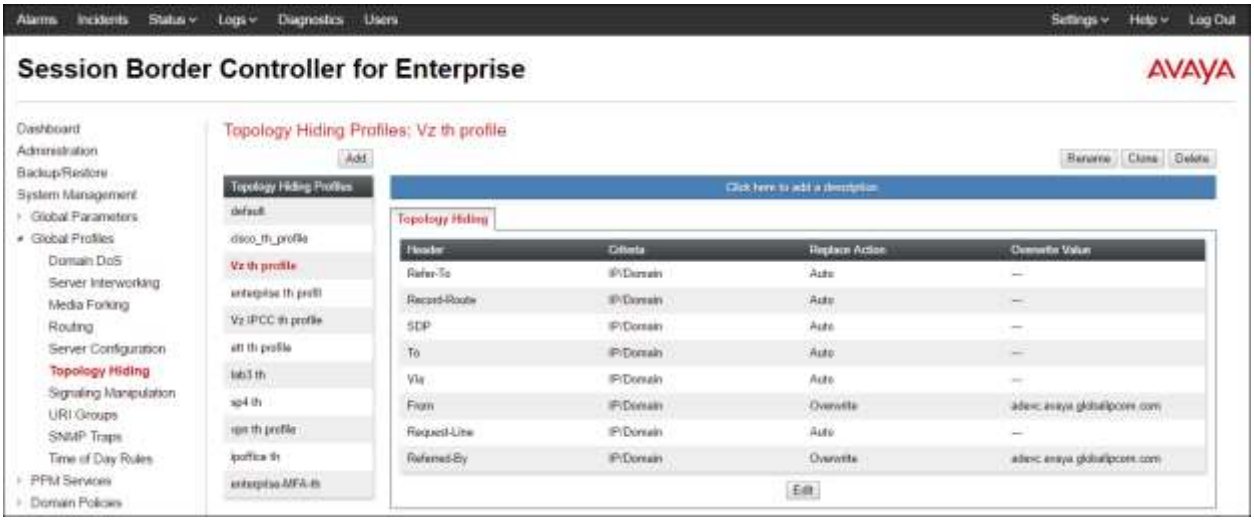
Click the **Add** button to add a new profile, or select an existing topology hiding profile to edit. In the sample configuration, the "**default**" profile was cloned for IP Office, and cloned and modified for Verizon.

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 IP Trunk service.



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8.6. 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 Business. 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**.

8.7. 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).

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

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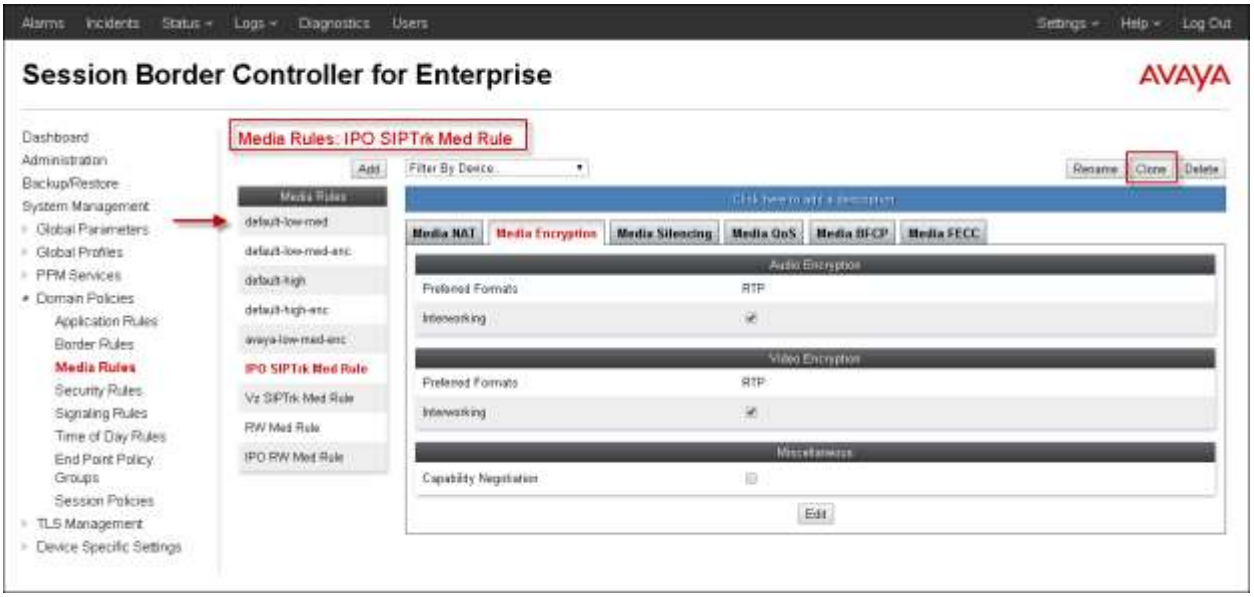
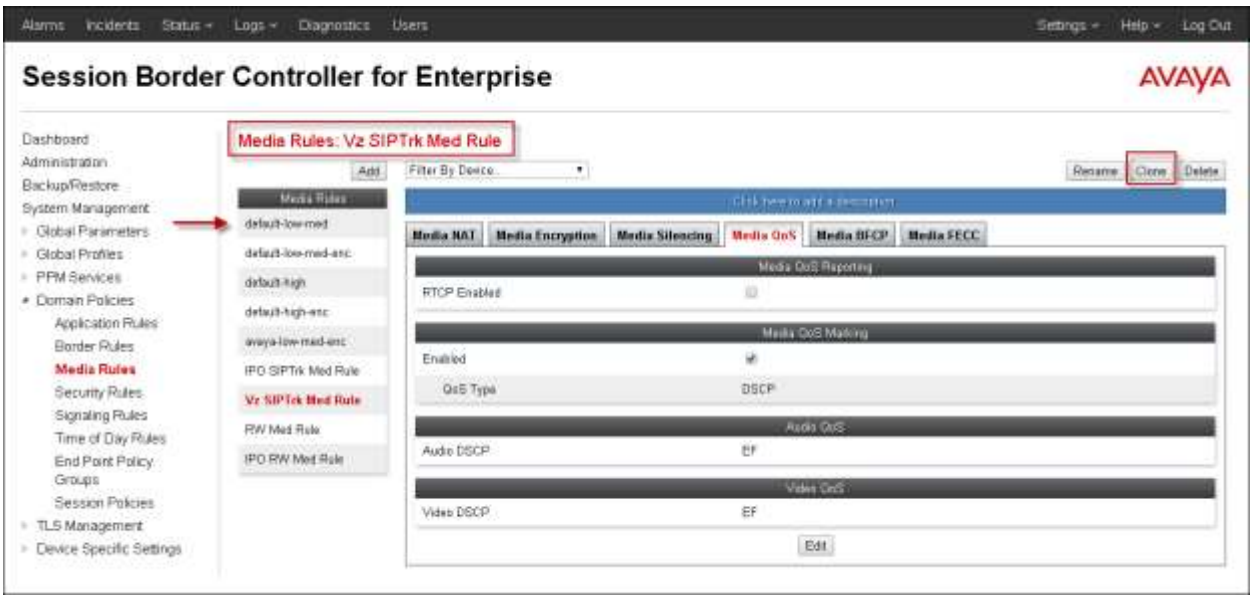


The Verizon Media Rule, "**Vz SIPTrk Med Rule**" with the DSCP values "**EF**" for expedited forwarding (default value) for **Media QoS**.



8.8. 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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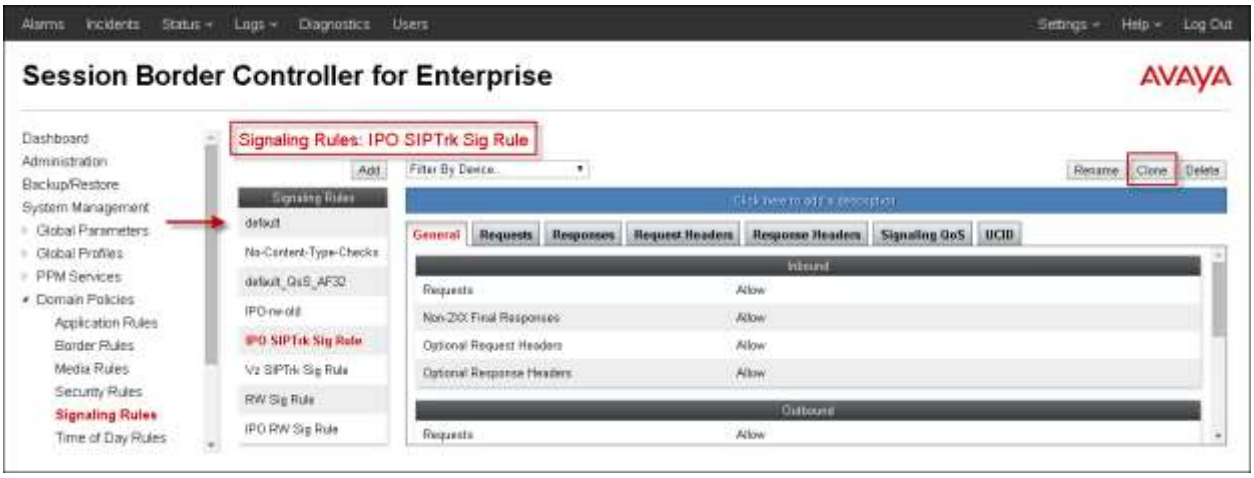
Clone and modify the **default** signaling rule to add the proper quality of service to the SIP signaling. To clone a signaling rule, navigate to **Domain Policies → Signaling Rules**. With the "**default"** rule chosen, click **Clone**. Enter a descriptive name for the new rule and click **Finish** (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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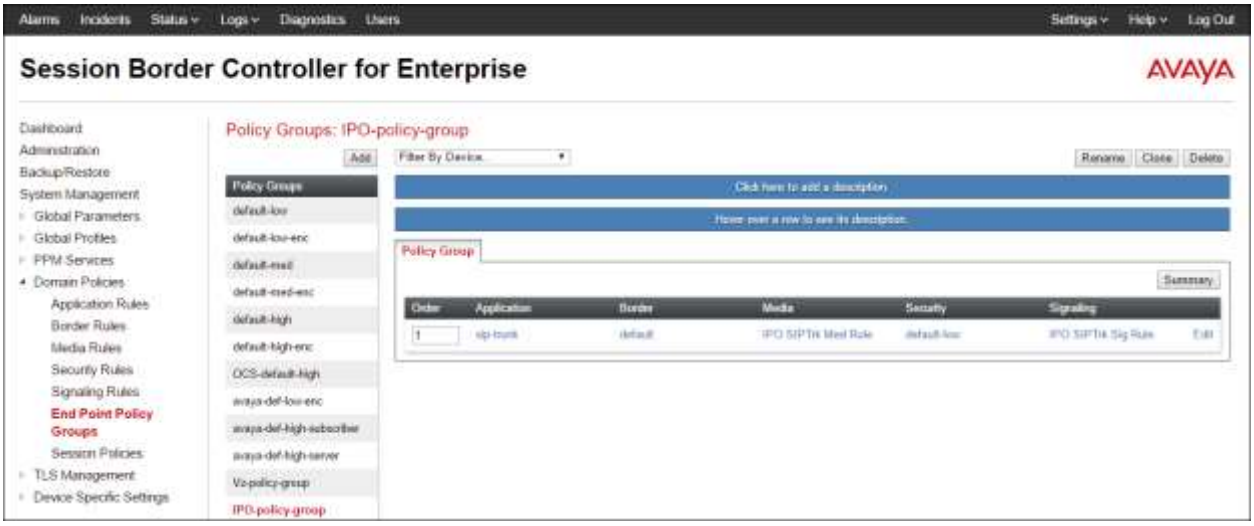
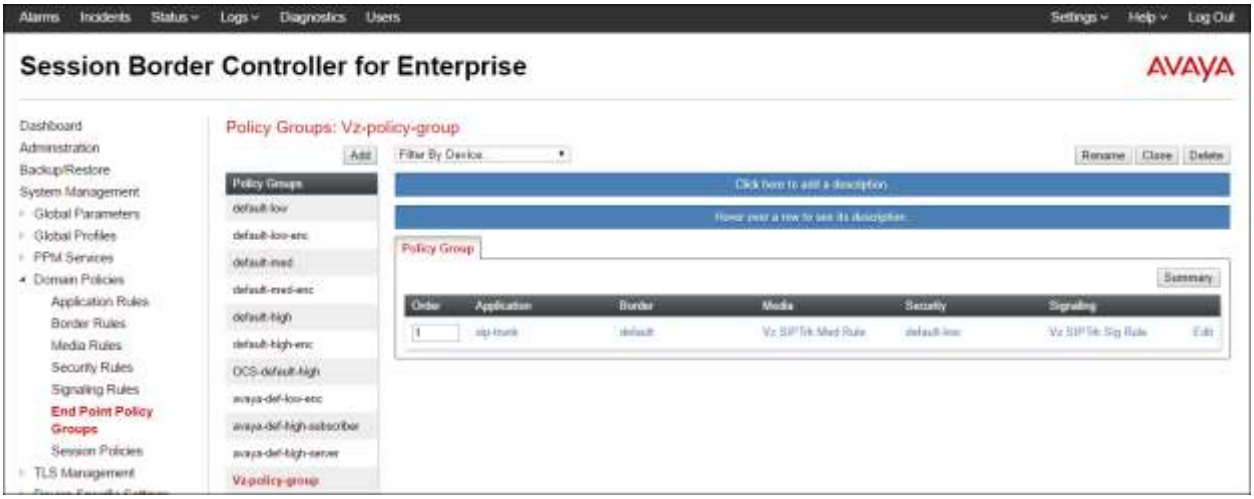
8.9. Endpoint Policy Groups

The rules created within the Domain Policy section are assigned to an Endpoint Policy Group. The Endpoint Policy Group is then applied to a Server Flow in **Section 8.12**.

To create a new policy group, navigate to **Domain Policies → Endpoint Policy Groups** and click on **Add** as shown below. The following screen shows the "**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 IP Trunk service. The details of the non-default rules chosen are shown in previous sections.



8.10. Media Interface

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

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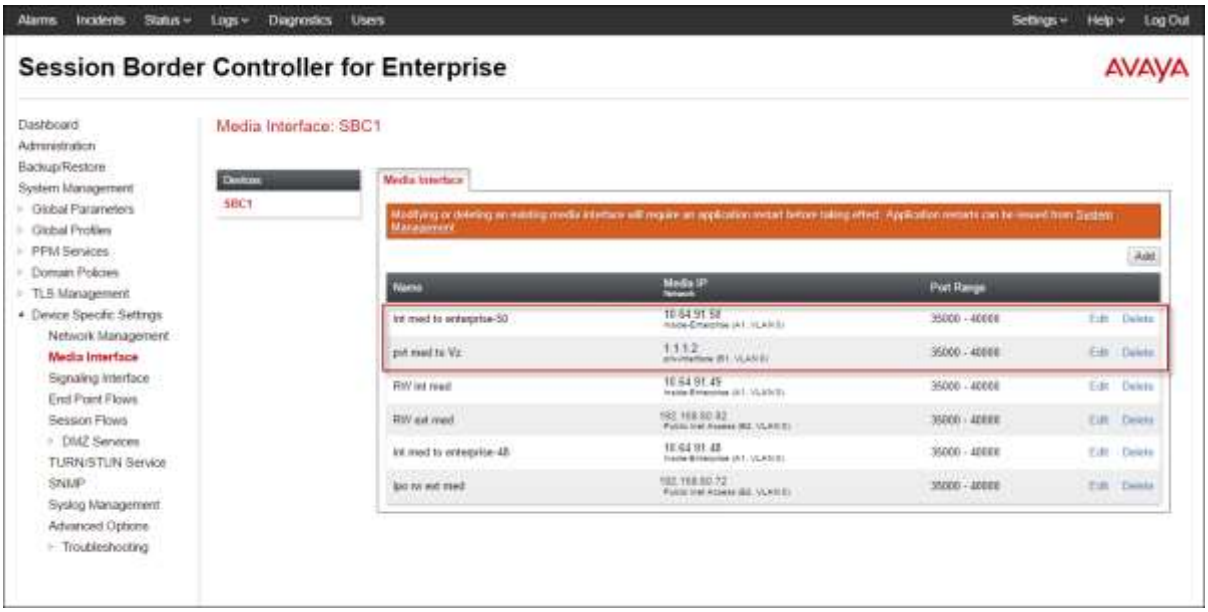
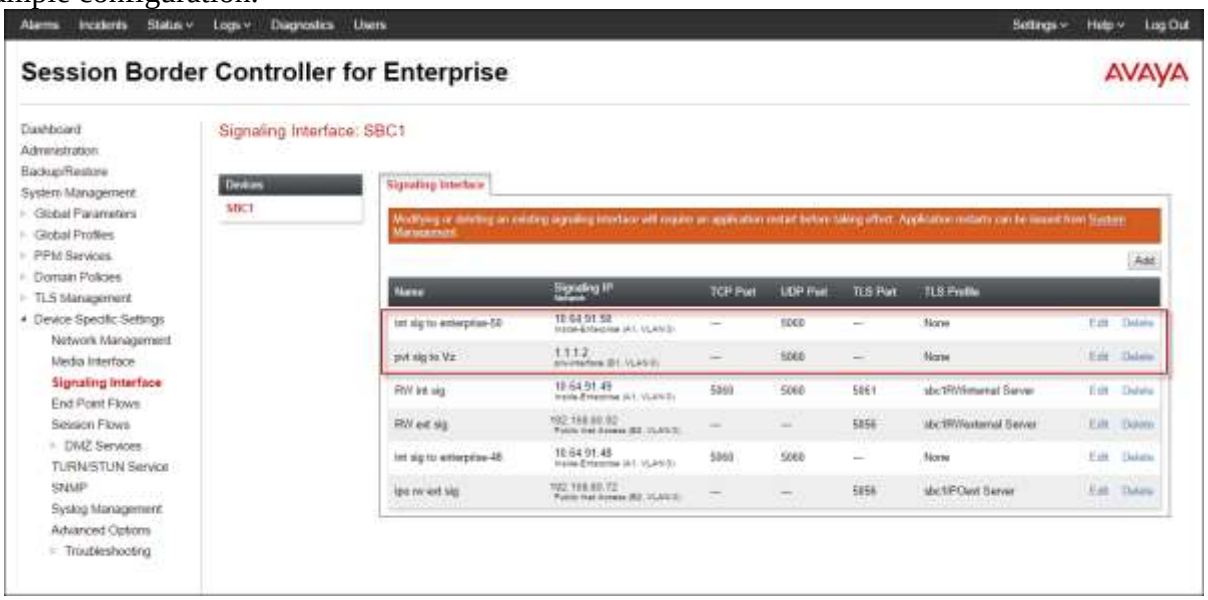
To create a new Media Interface, navigate to **Device Specific Settings → Media Interface** and click **Add**. The following screen shows the media interfaces defined for the sample configuration.



8.11. Signaling Interface

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

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



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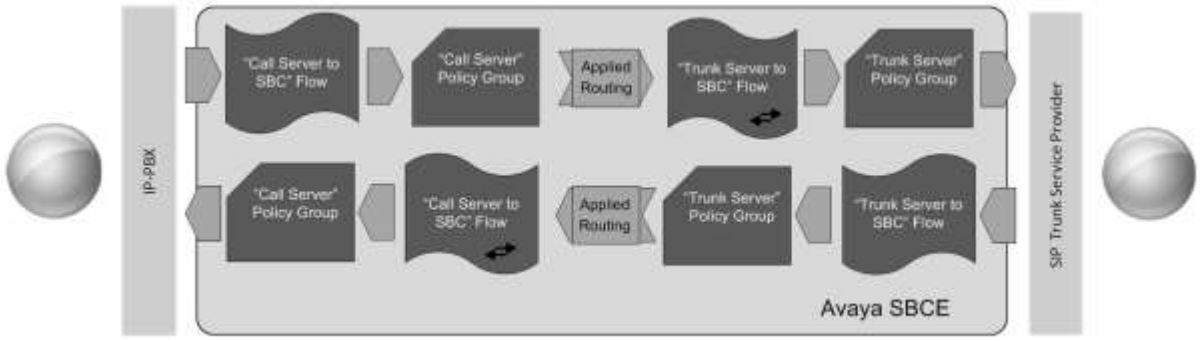
8.12. End Point Flows - Server Flow

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



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

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

View Flow: Verizon IPT Flow

Criteria		Profile	
Flow Name	Verizon IPT Flow	Signaling Interface	pvt sig to Vz
Server Configuration	Verizon IPT	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 th profile
Received Interface	int sig to enterprise-50	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 IPT**" 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 IP Trunk service, along with "/32", i.e., "**172.30.209.21/32**". 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 IPT

Criteria		Profile	
Flow Name	IPO SIP Trunk to Vz IPT	Signaling Interface	int sig to enterprise-50
Server Configuration	IPOffice	Media Interface	int med to enterprise-50
URI Group	*	End Point Policy Group	IPO-policy-group
Transport	*	Routing Profile	route to Vz IPT
Remote Subnet	172.30.209.21/32	Topology Hiding Profile	ipoffice th
Received Interface	pvt sig to Vz	Signaling Manipulation Script	None
		Remote Branch Office	Any

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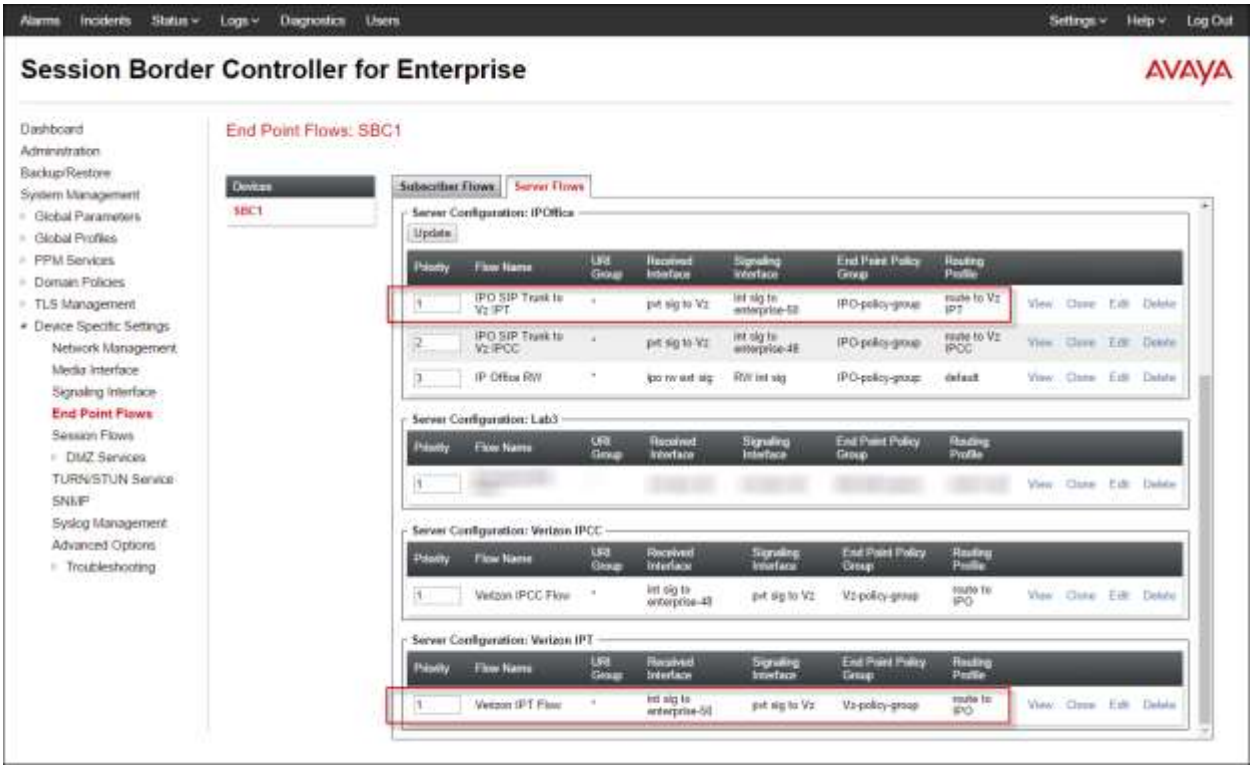
The following screen summarizes the Server Flows configured in the sample configuration. The highlighted flows are the ones relevant to the configuration of the SIP trunk to Verizon Business IP Trunk.



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9. Verizon Business Configuration

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

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

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

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

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

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10. Verifications

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

10.1. Avaya SBCE

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

10.1.1. Incidents

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

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

Type	ID	Date	Time	Category	Device	Cause
Server Heartbeat	732486352784939	6/3/16	10:51 AM	Policy	SBC1	Heartbeat Successful, Server is UP
Server Heartbeat	732486352784497	6/3/16	10:51 AM	Policy	SBC1	Heartbeat Successful, Server is UP
Server Heartbeat	732486352752785	6/3/16	10:51 AM	Policy	SBC1	Heartbeat Successful, Server is UP
Server Heartbeat	732486352752361	6/3/16	10:51 AM	Policy	SBC1	Heartbeat Successful, Server is UP

10.1.2. Server Status

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

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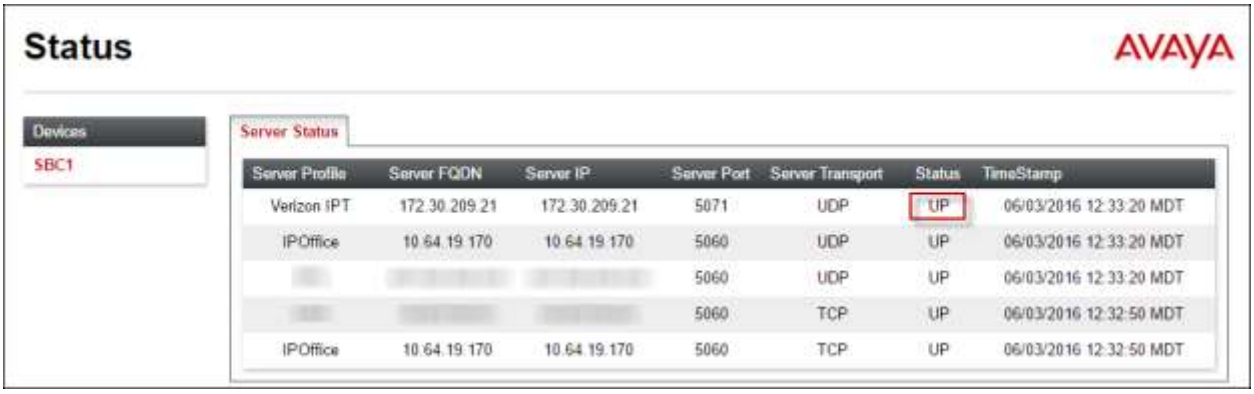
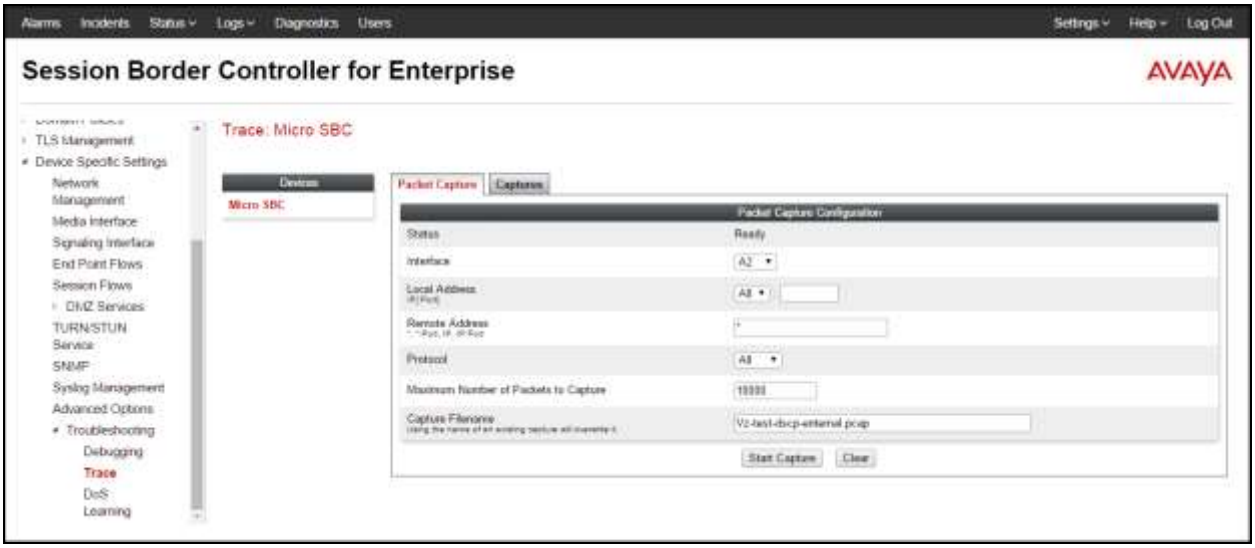
A pop-up window will appear with the **Status** of "**UP**" for the Verizon Business IP Trunk. The **Server Profile** will only list servers with Server Configuration settings that have Heartbeats enabled, see **Section 8.3.2**.



10.1.3. Tracing

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



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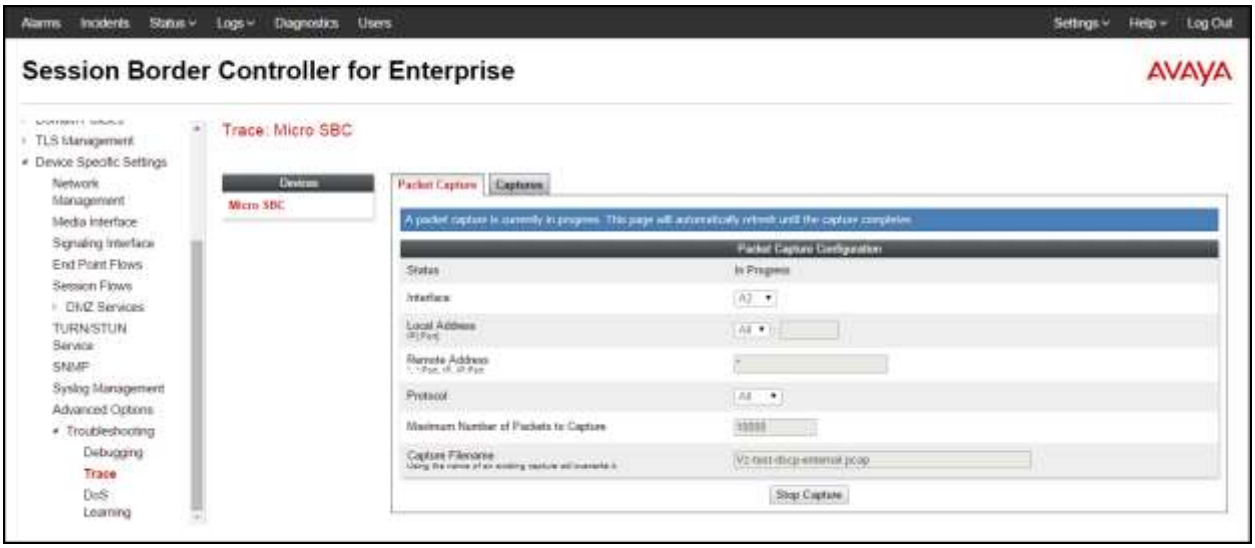
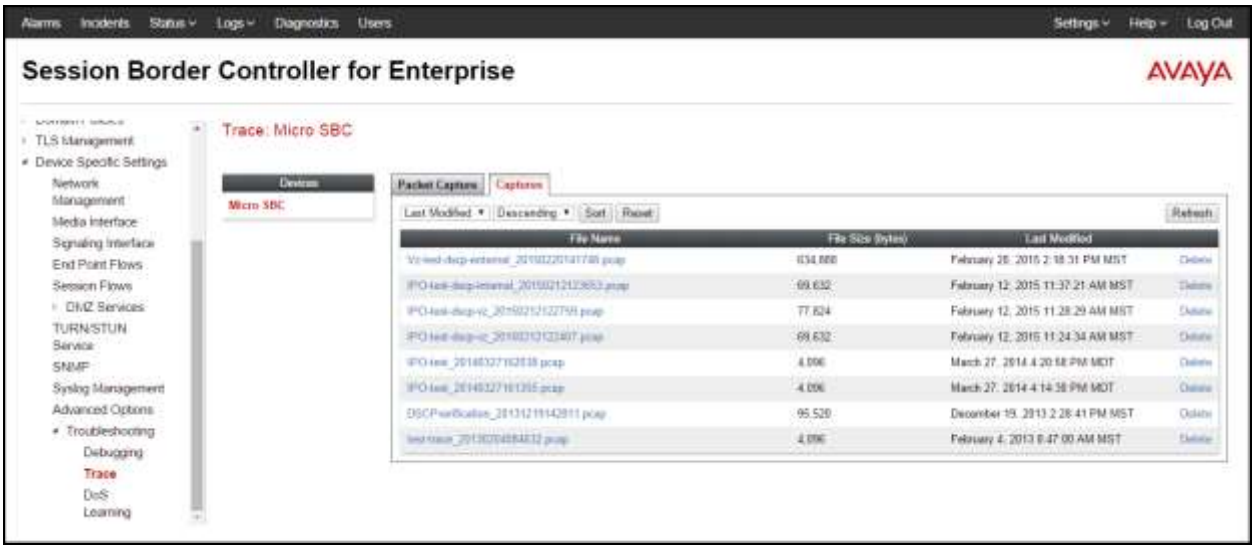
When tracing has reached the desired number of packets the trace will stop automatically, or alternatively, hit the **Stop Capture** button at the bottom.



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



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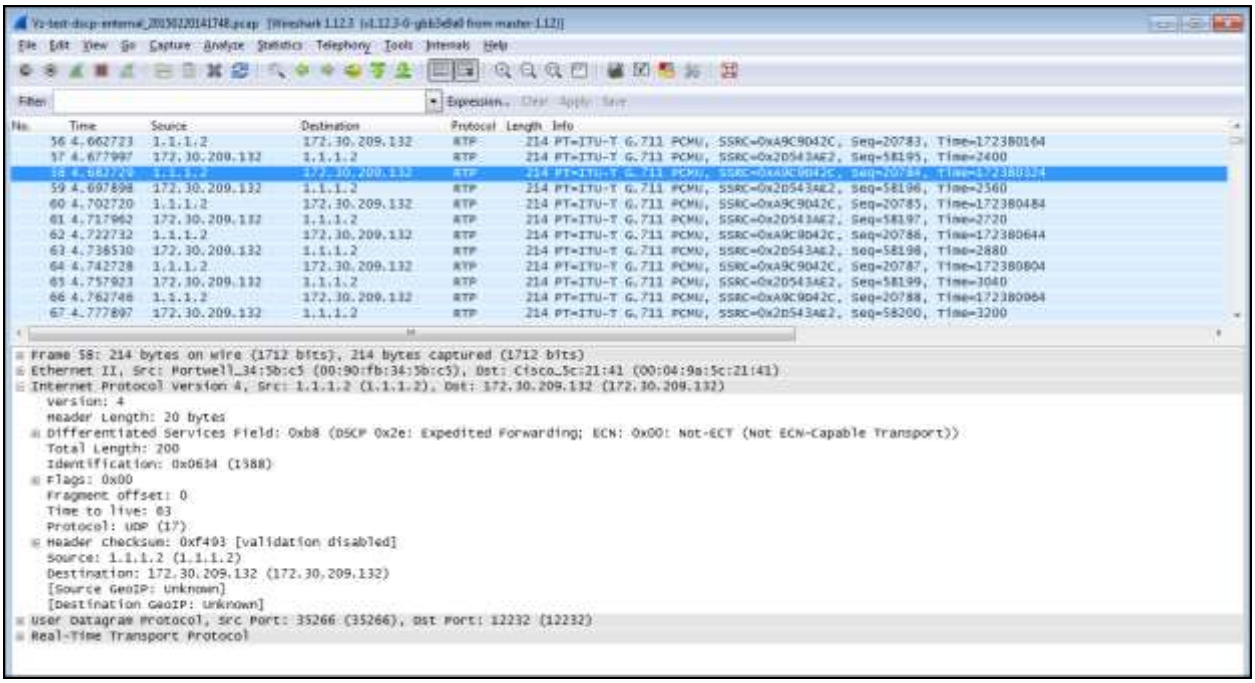
The packet capture file can be downloaded and then viewed using a Network Protocol Analyzer like WireShark.



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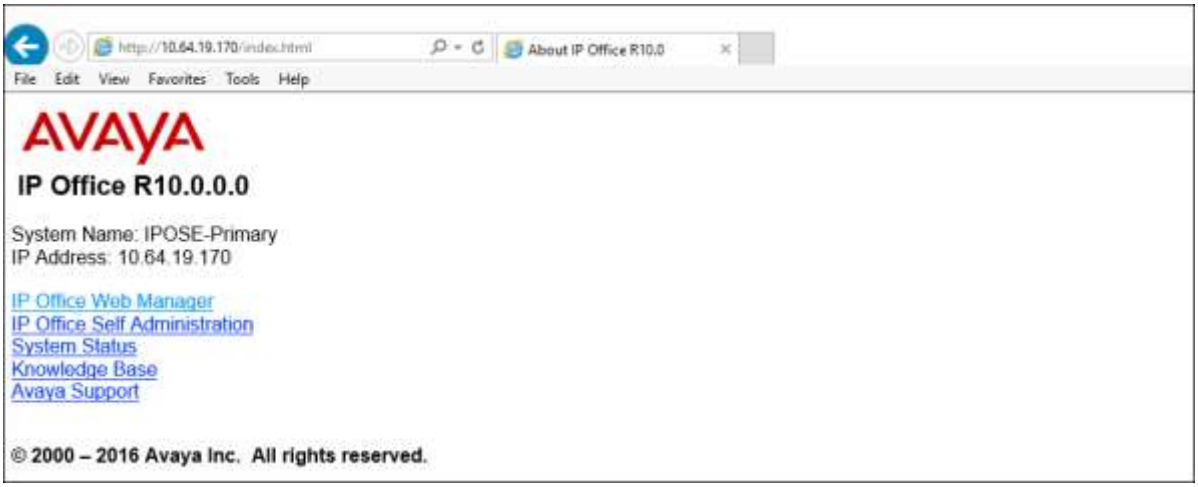
10.2. IP Office

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

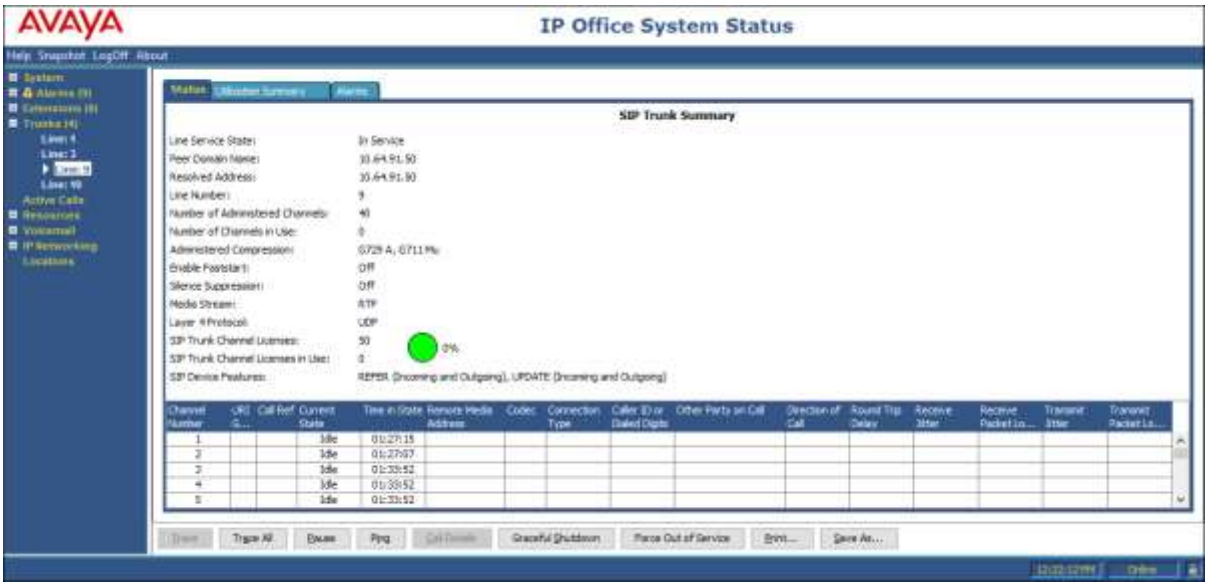
10.2.1. System Status

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



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

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



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

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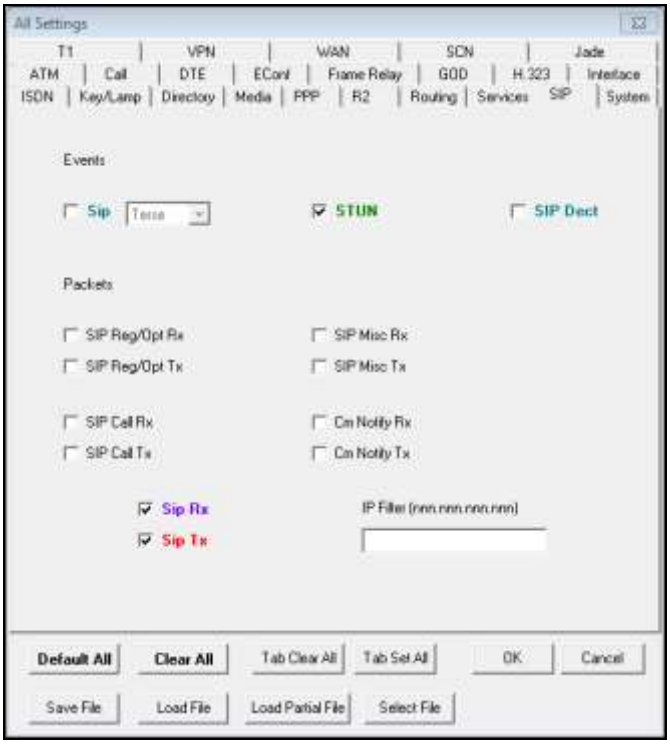
10.2.2. Monitor

The Monitor application can also be used to monitor and troubleshoot IP Office. Monitor can be accessed from **Start → Programs → IP Office → Monitor**. The application allows the monitored information to be customized. To customize, select **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 outbound call from extension 6242, whose DID is 732-945-0242, calling out to the PSTN via the Verizon Business IP Trunk service. The telephone user dialed 9-1-303-538-2177.

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11. Conclusion

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

These Application Notes demonstrated how IP Office Release 10.0 with Avaya Session Border Controller for Enterprise Release 7.1 can be successfully combined with a Verizon Business IP Trunk SIP Trunk Service connection to create an end-to-end SIP Telephony business solution. By following the example configurations provided in this document, customers using Avaya IP Office and Avaya SBCE can connect to the PSTN via a Verizon Business IP Trunk SIP Trunk service connection, thus eliminating the costs of analog or digital trunk connections previously required to access the PSTN. Utilizing this solution, IP Office customers can leverage the operational efficiencies and cost savings associated with SIP trunking while gaining the advanced technical features provided through the marriage of best of breed technologies from Avaya and Verizon.

12. 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 Servers as Virtual Machines*, Document Number 15-601011 Issue 03j, 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 10d, July 2016

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

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

[6] *Administering Avaya Communicator on IP Office*, February 2016

[7] *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

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

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

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

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

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 10.0 with Avaya Session Border Controller for Enterprise Release 7.1 with Verizon Business IPCC Service Suite.

[VZIPCC-IPO10SBC] 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

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