Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking Using Verizon Business IP Contact Center VoIP Inbound and Avaya IP Office Release 10.1 – Issue 1.0

Abstract

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Contact Center VoIP Inbound service and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of Avaya IP Office Server Edition Release 10.1, an IP500 V2 Expansion System Release 10.1, Avaya Communicator for Windows, Avaya Communicator for Web, 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 10.1.

The Verizon Business IP Contact Center VoIP Inbound offer referenced within these Application Notes enables a business to receive inbound toll free calls via standards-based SIP trunks, without the need for additional TDM enterprise gateways or TDM cards and the associated maintenance costs.

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

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

1. Introduction

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Contact Center VoIP Inbound service (Verizon Business IPCC) and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office Server Edition Primary Release 10.1, an IP500 V2 Expansion System Release 10.1, Avaya Communicator for Windows, Avaya Communicator for Web, and Avaya SIP, H.323, digital, and analog endpoints. The single Server Edition Primary server provides IP Office Server Edition, Voicemail Pro, and Avaya one-X® Portal for IP Office.

These Application Notes complement previously published Application Notes by illustrating the configuration screens and Avaya testing of IP Office Release 10.1.

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

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

For more information on the Verizon Business IPCC service, visit http://www.verizonenterprise.com/products/business-communications/customer-contact-solutions/.

2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the Verizon Business IPCC service, as depicted in **Figure 1**. The Avaya IP Office was configured to use the commercially available SIP Trunking solution provided by the Verizon Business IPCC service. This allowed Avaya IP Office to receive inbound toll-free calls from the PSTN via the SIP protocol.

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

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

2.1. Interoperability Compliance Testing

The testing included executing the test cases detailed in **Section 10**, Reference [VZ-Test-Plan], which contains the Verizon Business IPCC Interoperability Lab Test Plan. To summarize, the testing included the following successful SIP trunk interoperability compliance testing:

- SIP OPTIONS monitoring of the health of the SIP trunk was verified. Verizon Business and IP Office can all monitor health using SIP OPTIONS.
- Proper recovery from induced failure conditions such as IP Office reboots, and IP network outages between Verizon and IP Office, of short and long durations.
- Incoming calls from the PSTN were routed to the toll-free numbers assigned by Verizon Business to the Avaya IP Office location. These incoming calls arrived via the SIP Line configured in **Section 5.4** and were answered by Avaya H.323 endpoints, Avaya SIP endpoints, Avaya digital endpoints, analog endpoints, Avaya Communicator for Windows, Avaya Communicator for Web, and Voicemail Pro.
- Proper disconnect when either party hangs up an active call.
- Proper disconnect when the PSTN caller abandons (i.e., hangs up) a toll-free call before the IP Office party has answered.
- Proper SIP 486 response and busy tone heard by the caller when a PSTN user calls a toll-free number directed to a busy IP Office user, an IP Office user with Do-not-disturb active, or an IP Office user that is logged out (i.e., assuming no redirection is configured for these conditions). Similarly, busy tone is heard when a PSTN user calls a toll-free number directed to a hunt group whose queue is “full” (i.e. if no redirection is configured for hunt group busy conditions, see **Section 5.6.2**).
- Proper termination of an inbound IP Toll Free call left in a ringing state for a relatively long duration.
- The display of caller ID on display-equipped Avaya IP Office endpoints was verified. The IP Office capability to use the caller ID received from Verizon to look up and display a name from a configurable directory was also exercised successfully.
- Privacy requests for inbound toll-free calls from the PSTN were verified. That is, when privacy is requested by a PSTN caller (e.g., dialing *67 from a mobile phone), the inbound toll-free call can be successfully completed to an IP Office telephone user while presenting a “WITHHELD” or anonymous display to an IP Office user (i.e., rather than the caller’s telephone number).
- Inbound toll-free long holding time call stability.
- Inbound fax using G.711 and T.38 were verified
- IP Office sends SIP 180 RINGING (no SDP in 180) for inbound calls and ring back tone is heard by the caller.

- Telephony features such as hold and resume, transfer of toll-free calls to other IP Office users, and conference of toll-free calls.
- Incoming voice calls using the G.729(a) and G.711 ULAW codecs, and proper protocol procedures related to media.
- DTMF transmission using RFC 2833. Successful IP Office Voicemail Pro menu navigation for incoming toll-free calls. Successful use of IP Office Mobile Call Control, where DTMF sequences can be performed remotely using the SIP Line.
- Incoming toll-free calls directed to the Hunt Groups configured in **Section 5.6.2** were verified. Incoming calls could be queued, queued with priority, and be answered by members of the hunt group as members become available.
- Incoming toll-free calls that were redirected by a SIP REFER message, generated by Avaya IP Office/Voicemail Pro, back to the Verizon Business IPCC service for redirection to an alternate destination.
- Outgoing calls from the Avaya IP Office location to the PSTN were routed via a SIP Line to the Verizon Business IP Trunk service described in reference [VZBIPT-IPO10.1]. As detailed in reference [VZBIPT-IPO10.1], these outgoing PSTN calls can be originated from Avaya H.323 endpoints, Avaya SIP endpoints, Avaya digital endpoints, and analog endpoints. The display of caller ID on display-equipped PSTN telephones was verified. In the context of inbound toll-free calls using Verizon Business IPCC, inbound toll-free calls arriving via the SIP Line configured in **Section 5.4** could be forwarded, transferred or twinned out the Verizon Business IP Trunk service SIP Line. Inbound toll-free calls from the Verizon Business IPCC SIP Line could also trigger mobile callback calls that use the Verizon Business IP Trunk service SIP Line.
- Call Forwarding of Verizon toll-free calls to PSTN destinations via the Verizon Business IP Trunk service documented in reference [VZBIPT-IPO10.1], presenting true calling party information to the mobile phone.
- Mobile twinning of Verizon toll-free calls to a mobile phone via the Verizon Business IP Trunk service documented in reference [VZBIPT-IPO10.1], presenting true calling party information to the mobile phone.
- Proper DiffServ markings for Avaya IP Office SIP signaling and RTP media.

2.2. Test Results

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

- In the reference configuration, redirecting an inbound Verizon Business IPCC toll-free call back out the Verizon Business IPCC service using the SIP REFER method was limited to SIP endpoints using blind transfer, and Voicemail Pro modules. It is not supported by H.323, digital, or analog endpoints. Other non-SIP endpoints utilized the Verizon Business IP Trunk service SIP Line to transfer a toll-free call out to the PSTN.

- **DiffServ markings:** For IP Office Server Edition, the IP header in SIP signaling packets sent from the IP Office server do not contain the DSCP values configured in IP Office Manager for Quality of Service policies (See **Section 5.2.2**). The IP headers in RTP media packets have the correct values. Also, this only affects Server Edition systems; the

IP headers in SIP signaling packets from IP 500V2 systems have the correct values. This anomaly is under investigated by IP Office product development (IPOFFICE-112012).

2.3. Support

2.3.1. Avaya

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

2.3.2. Verizon

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

3. Reference Configuration

Figure 1 illustrates a sample Avaya IP Office solution connected to the Verizon Business IPCC service. The Avaya equipment is located on a private IP subnet. An enterprise edge router provides access to the Verizon Business network via a Verizon Business T1 circuit. This circuit is provisioned for the Verizon Business PIP service.

In the sample configuration, the Avaya IP Office receives traffic from the Verizon Business IPCC service on port 5060 and sends traffic to port 5072, using UDP for network transport, as required by the Verizon Business IPCC service. Verizon provided five toll-free numbers associated with the IPCC service. These toll-free numbers were mapped to IP Office destinations via Incoming Call Routes as summarized in **Table 1**. The Avaya IP Office environment domain known to Verizon was *adevc.avaya.globalipcom.com*.

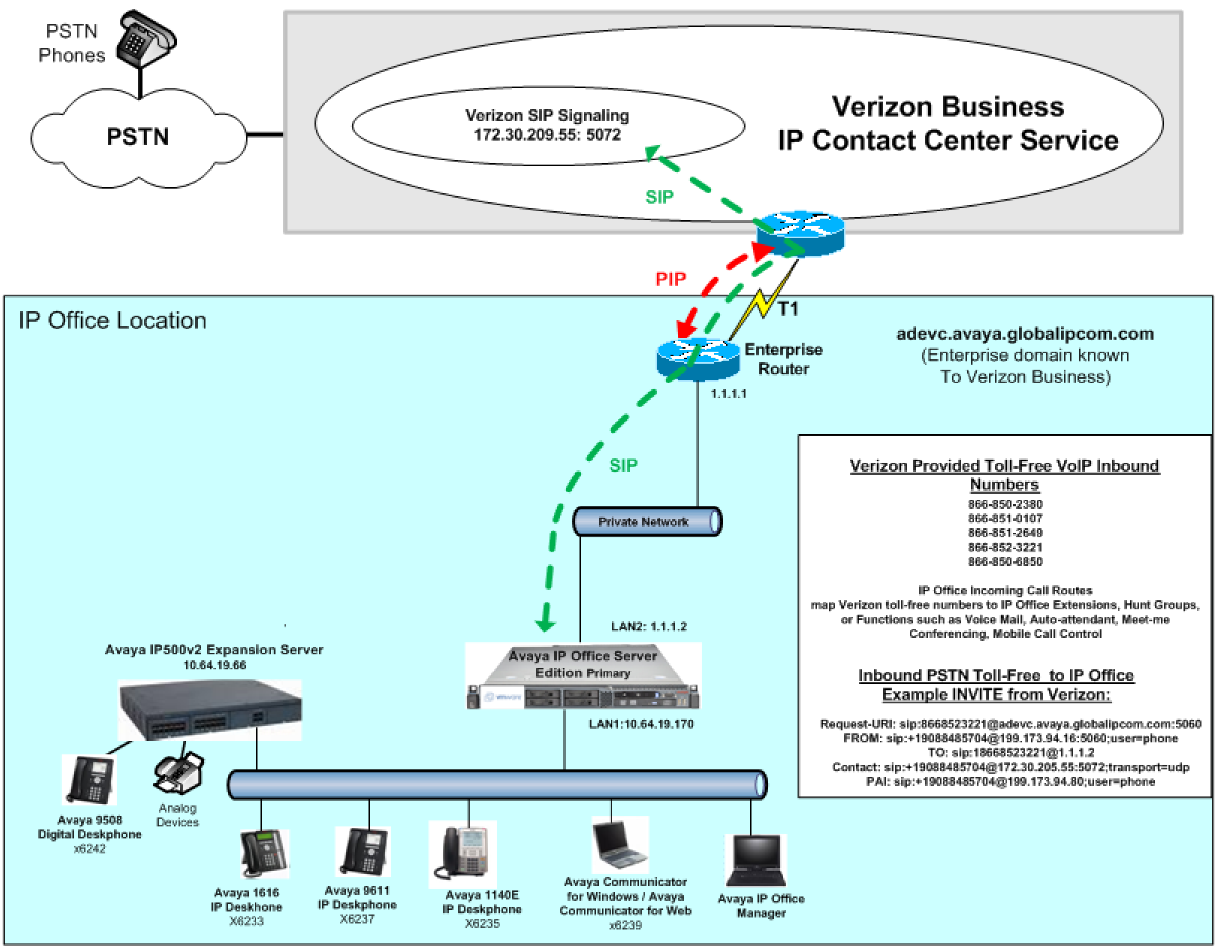


Figure 1: Avaya Interoperability Test Lab Configuration

The Verizon Business IP Trunk service used for outgoing calls, as described in **Section 2.1**, required different SIP line configuration parameters than what were needed for the Verizon Business IPCC service. A new SIP line was created in IP Office to support the Verizon Business IPCC service. This SIP line is separate from the SIP line previously created for Verizon Business IP Trunk service as described in reference [VZBIPT-IPO10.1]. Having separate SIP lines for each Verizon service will allow for unique parameters to be set on IP Office to accommodate the differences between the two services.

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

Verizon Provided Toll-Free Number	Configured Avaya IP Office Destination(s)	Notes
866-851-0107	x6235, x6239	Avaya 1140E, Avaya Communicator for Windows, and Avaya Communicator for Web.
866-851-2649	x6242, x6300	Digital Deskphone with Mobile Twinning and Mobile Call Control permission. Analog phone/fax.
866-850-2380	x6233, x6237	Avaya 9611 Deskphone Avaya 1616 Deskphone
866-850-6850	Voicemail Collect on Voicemail Pro	Allow external callers to access voice mail toll-free
866-850-6850	Inbound Mobile Call Control	Allow toll-free calls from pre-configured twinning numbers to access mobile call control
866-850-6850	Conference Bridge on Voicemail Pro	Allow external callers to access conference bridge toll-free
866-850-6850	Call Redirect	Allow external caller to be redirected to another number using SIP REFER method.
866-852-3221 (any caller)	“401 Call Center” Hunt Group (with default priority)	Hunt Group with queuing
866-852-3221 (specific callers)	“400 Overdue Account” Hunt Group	Show IP Office destination selection based on caller ID
866-852-3221 (specific priority callers)	“401 Call Center” Hunt Group (with High Priority)	Show IP Office priority queuing based on caller ID

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

4. Equipment and Software Validated

Table 3 shows the equipment and software used in the sample configuration.

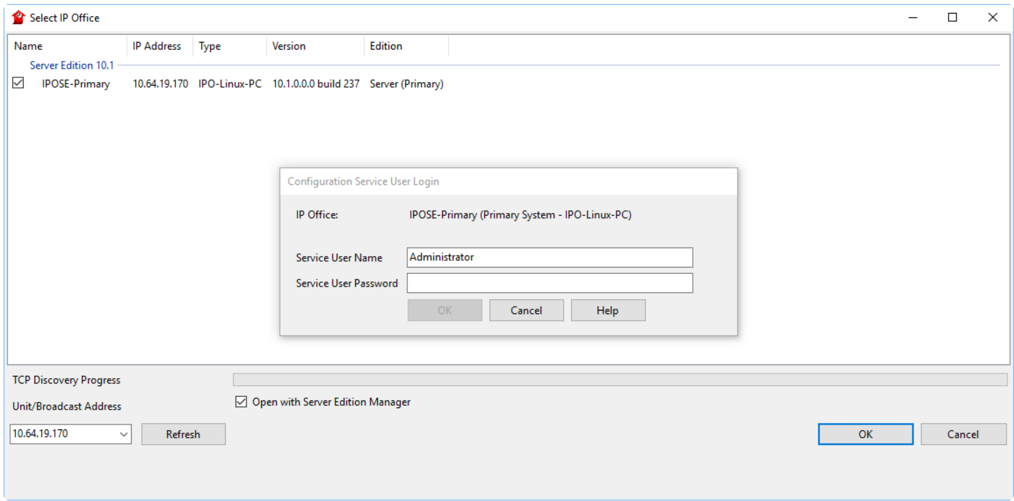
Equipment/Software	Release/Version
Avaya IP Office Server Edition (Primary Server) ▪ IP Office ▪ Voicemail Pro ▪ Avaya WebRTC Gateway ▪ Avaya one-X® Portal for IP Office	 Release 10.1.0.0.0 build 237 Release 10.1.0.0.0 build 241 Release 10.1.0.0.0 build 13 Release 10.1.0.0.0 build 305
Avaya IP Office IP500 V2 (Expansion System) ▪ Avaya IP Office TCM 8 ▪ Avaya IP Office COMBO6210/ATM4	 Release 10.1.0.0.0 Build 237 Release 10.1.0.0.0 Build 237
Avaya IP Office Manager	Release 10.1.0.0.0 Build 237
Avaya 9611SW IP Deskphone (H.323)	Release 6.6401
Avaya 1616 IP Deskphone (H.323)	Release 1.3.90
Avaya 1140E IP Deskphone (SIP)	Release 04.04.23
Avaya 9508 Digital Deskphone	Release 0.60
Avaya Communicator for Windows	Release 2.1.4.0
Avaya Communicator for Web	Release 1.0.17.2014
Analog Fax device	Ventafax 7.9

Table 2: Equipment and Software Tested

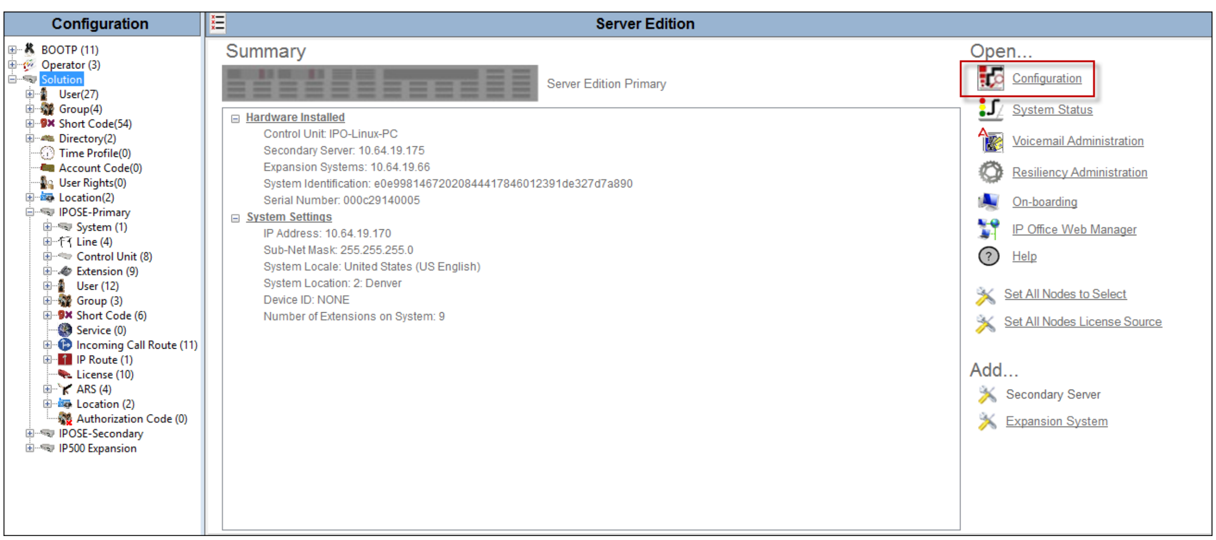
Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2, and also when deployed with all configurations of IP Office Server Edition. Note that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks.

5. Avaya IP Office Primary Configuration

This section illustrates relevant aspects of the Primary server used in the verification of these Application Notes. The Primary server is configured via the IP Office Manager program. For more information on IP Office Manager, consult reference **[2]**. From the IP Office Manager PC, select **Start → All Apps → IP Office → Manager** to launch the Manager application. Navigate to **File → Open Configuration** (not shown), select the proper Avaya IP Office system from the pop-up window, and log in using the appropriate credentials.



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

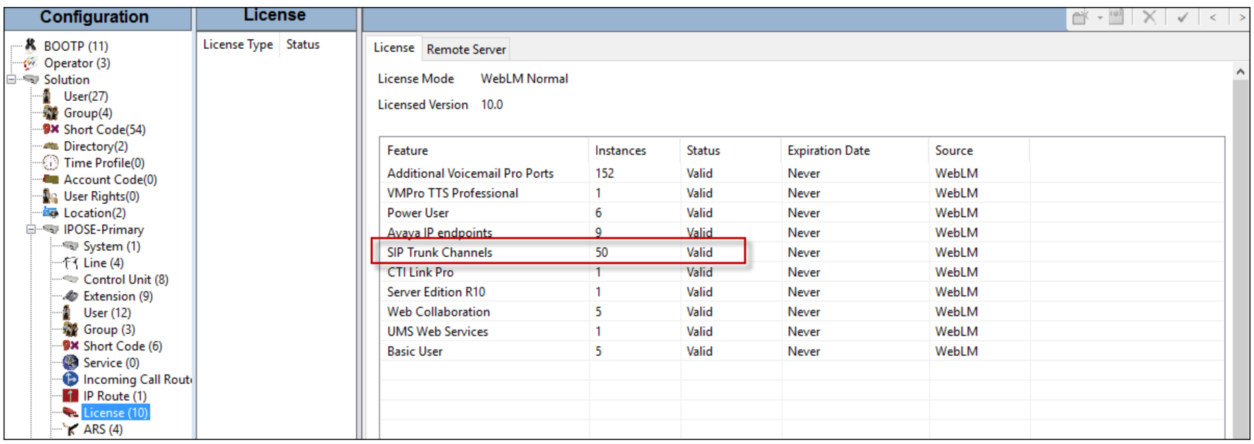


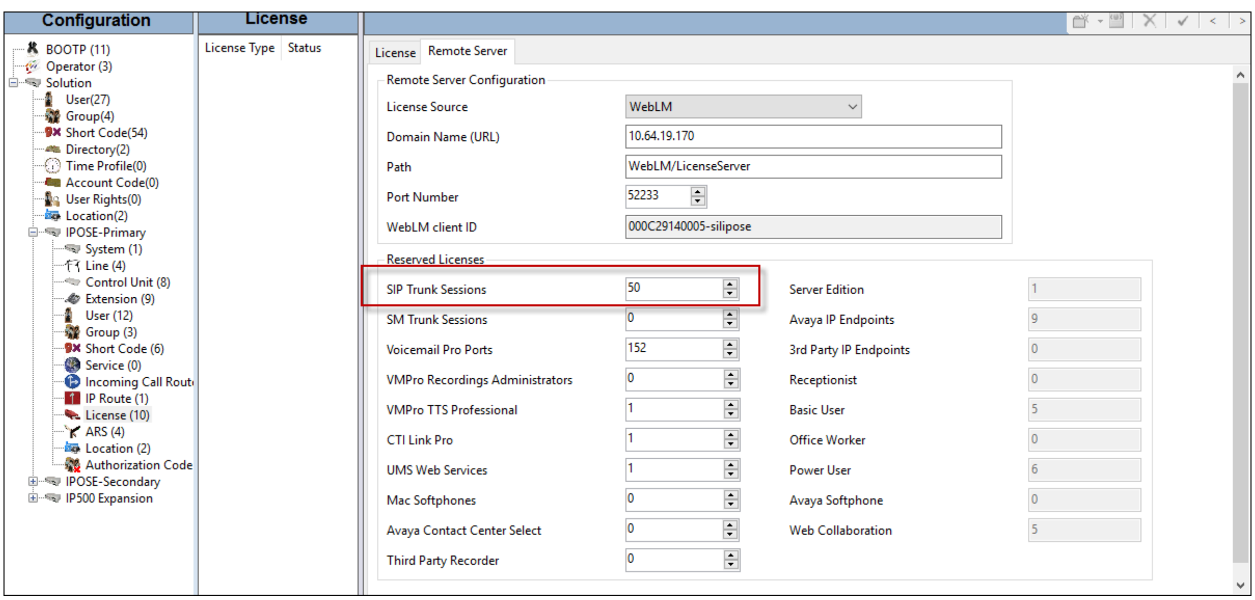
5.1. Licensing and Physical Hardware

In the sample configuration, **IPOSE-Primary** was used as the system name of the Primary 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 Deskphones will be used as is the case in these Application Notes, verify the **Avaya IP endpoints** license.



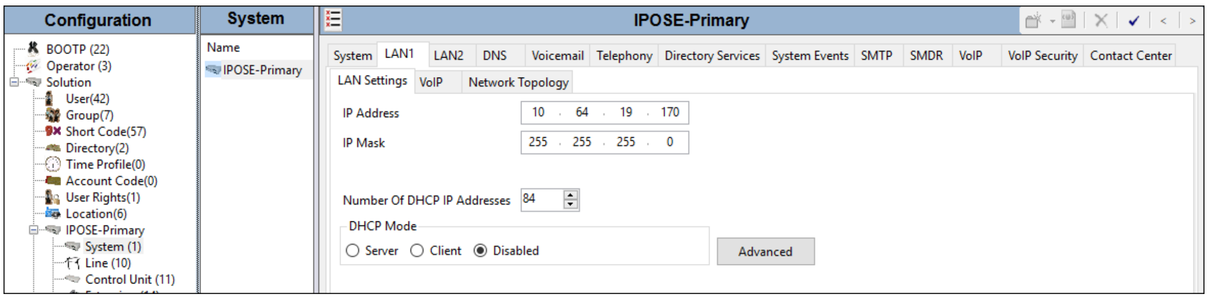


5.2. System Settings

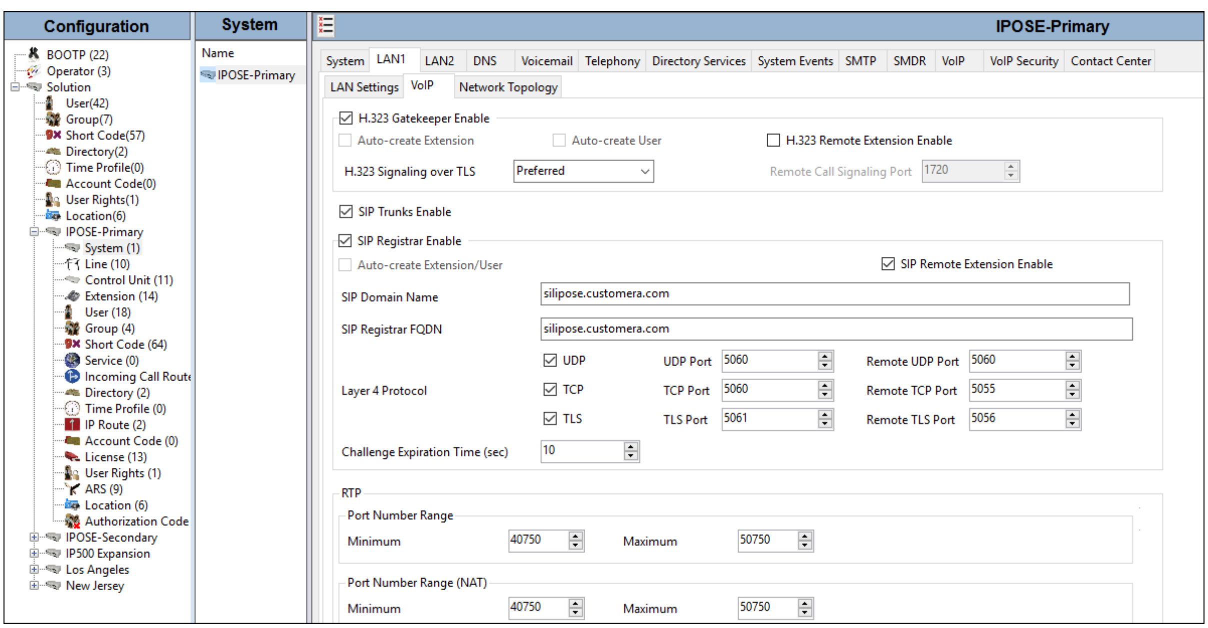
This section illustrates the configuration of system settings. Select **System** in the Navigation pane to configure these settings. The subsection order corresponds to a left to right navigation of the tabs in the Details pane for System settings. For all of the following configuration sections, the **OK** button (not shown) must be selected in order for any changes to be saved.

5.2.1. LAN 1 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.

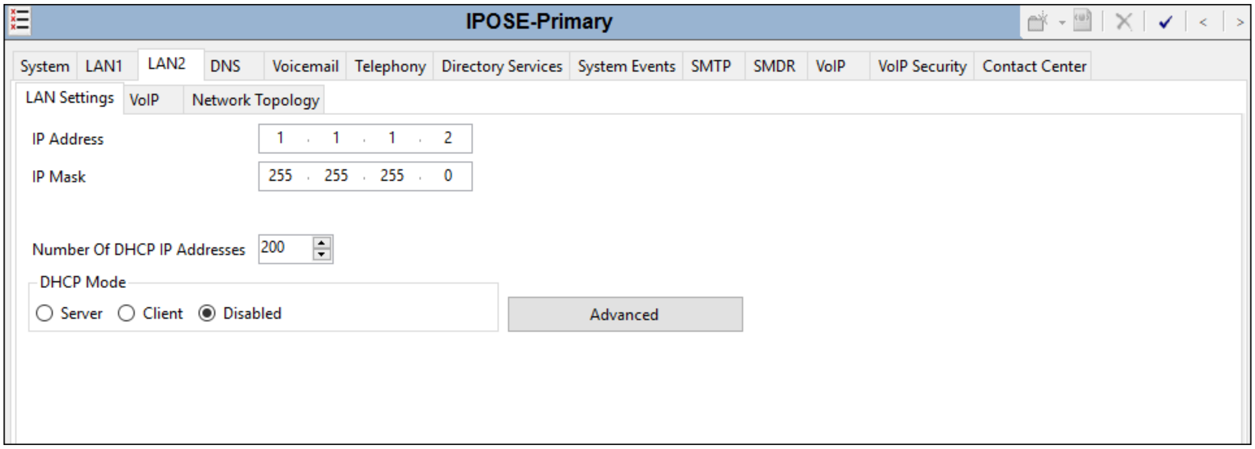


Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** parameter is checked to allow the use of Avaya IP Deskphones using the H.323 protocol, such as the Avaya 1616 and 9611 used in the sample configuration. The **H.323 Signaling over TLS** is set to "**Preferred**" to allow TLS signaling for H.323 endpoint that support it. The **SIP Registrar Enable** parameter is checked to allow Avaya 1140E and Avaya Communicator usage. The **SIP Domain Name** and **SIP Registrar FQDN** may be set according to customer requirements.



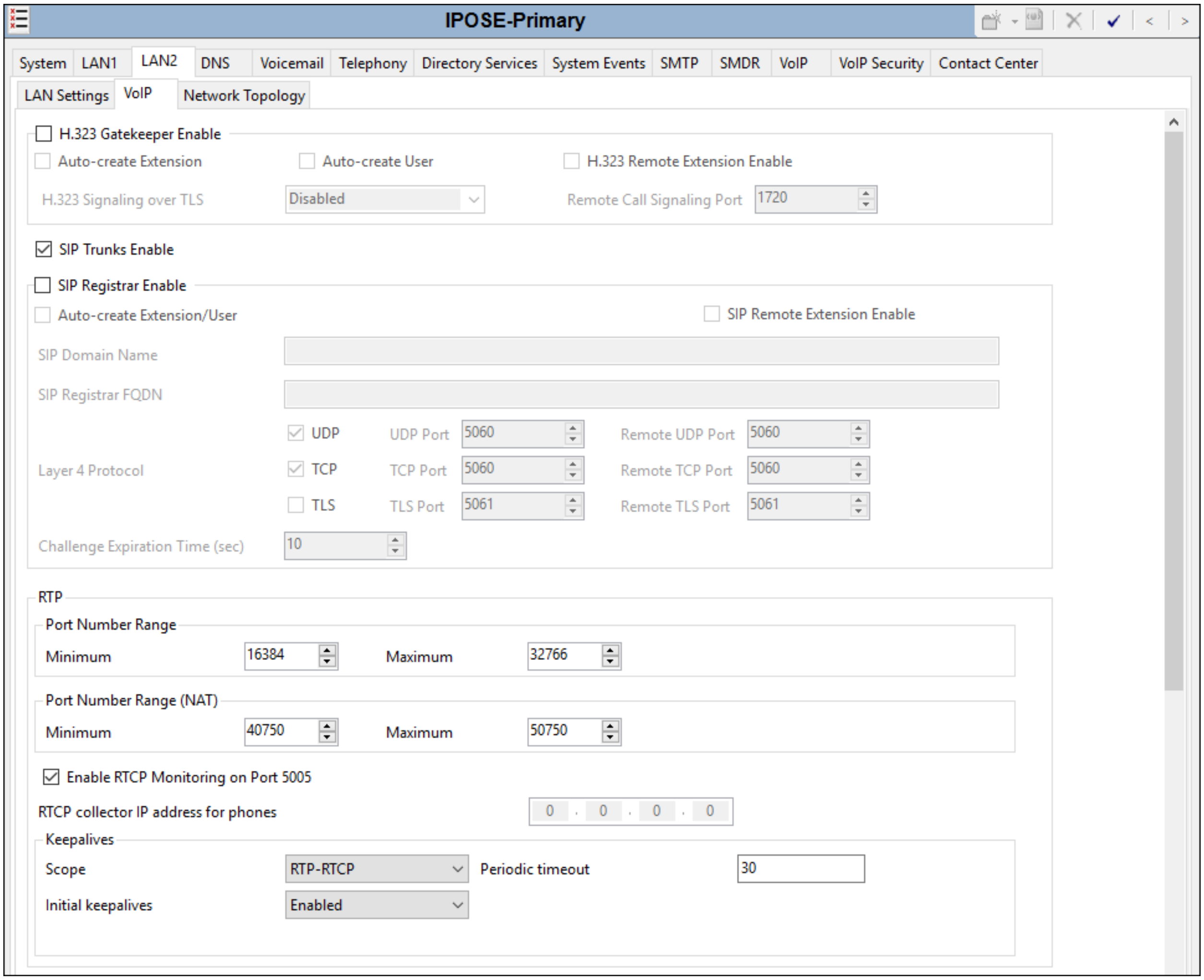
5.2.2. LAN 2 Settings

In the sample configuration, LAN2 is used to connect the IP Office to the Verizon PIP network. To view or configure the **IP Address** of LAN2, select the **LAN2** tab followed by the **LAN Settings** tab. As shown in **Figure 1**, the IP address of the IP Office, known to Verizon, is "**1.1.1.2**". **DHCP Mode** is set to "**Disabled**" since DHCP is unnecessary towards Verizon. Other parameters on this screen may be set according to customer requirements.

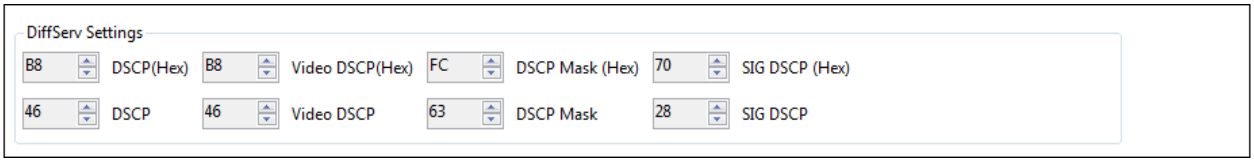


Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** and **SIP Registrar Enable** boxes are unchecked since IP telephones will not be registering on this link. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Verizon Business.

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

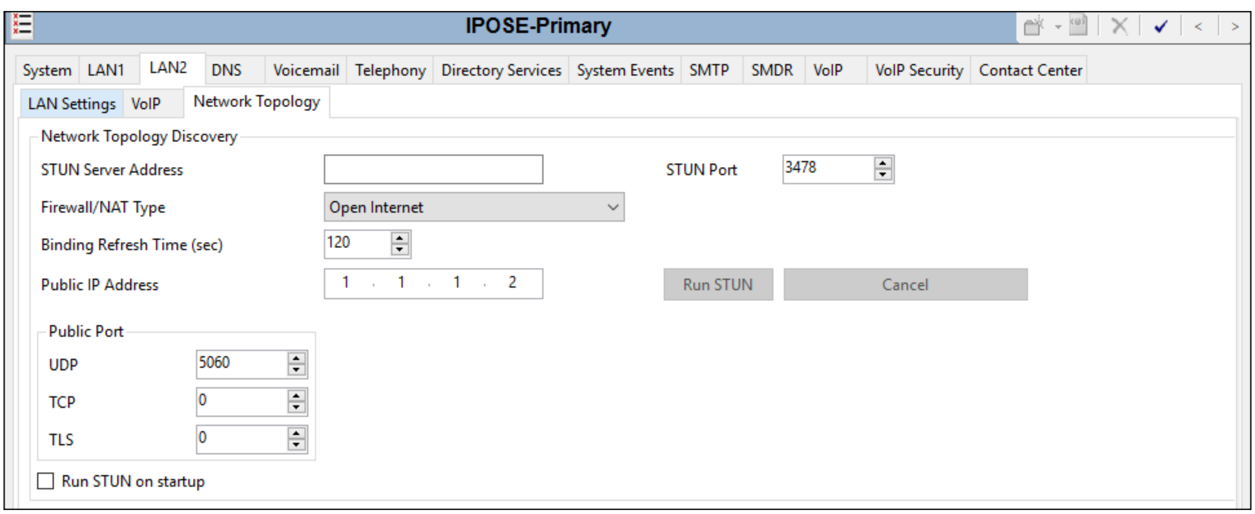


Scrolling down, IP Office can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Service policies. In the sample configuration shown below, IP Office will mark SIP signaling with a value associated with “Assured Forwarding” using DSCP decimal 28 (**SIG DSCP** parameter). IP Office will mark the RTP media with a value associated with “Expedited Forwarding” using DSCP decimal 46 (**DSCP** parameter). See **Section 2.2** for limitations with IP Office Server Edition. This screen enables flexibility in IP Office DiffServ markings (RFC 2474) to allow alignment with network routing policies, which are outside the scope of these Application Notes. Other parameters on this screen may be set according to customer requirements.



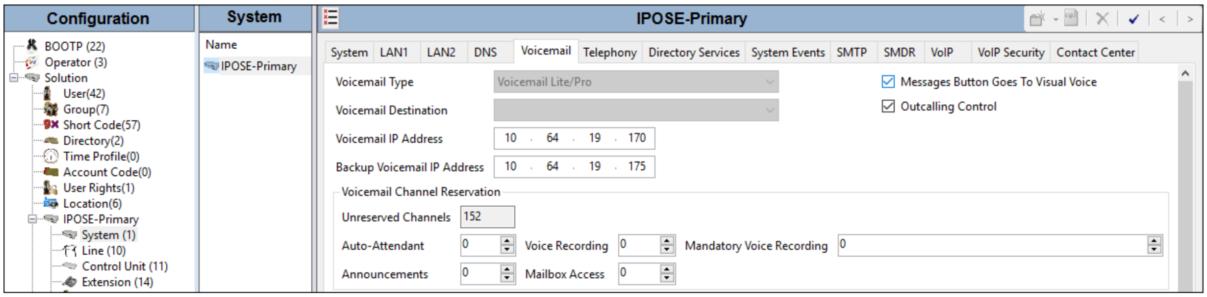
Select the **Network Topology** tab as shown in the following screen. In the reference configuration, it was not necessary to configure the Network Topology tab for the purposes of SIP trunking. If a NAT is used between Avaya IP Office and AT&T Boarder Element, then the SIP Line Use Network Topology Info field in **Section 5.4.4** should be set to the LAN2 interface used by the trunk and the Network Topology tab needs to be configured with the details of the NAT device. The Network Topology tab is summarized here for completeness

The **Firewall/NAT Type** is set to “**Open Internet**” in the sample configuration. Note that the **Firewall/NAT Type** parameter may need to be set differently, depending on the type of firewall or Network Address Translation device used at the customer premise. The **Binding Refresh Time (sec)** specifies how often IP Office will issue a SIP OPTIONS message to check the SIP trunk connection status to Verizon Business. In the reference configuration, **120** is specified. The **Public IP Address** is set to the IP address known to Verizon. In the sample configuration, this is “**1.1.1.2**”. The **UDP Public Port** is set to “**5060**”.

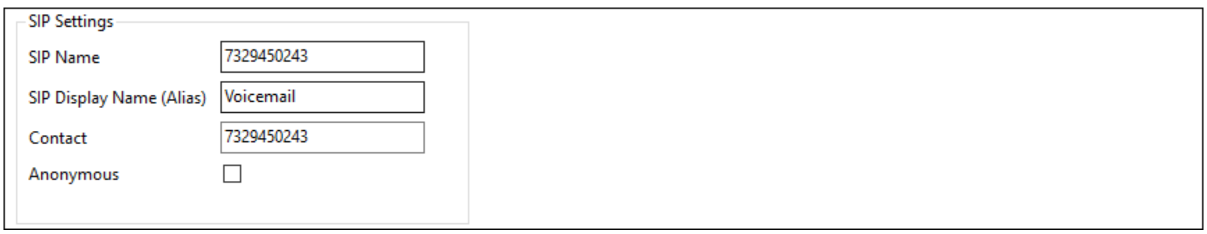


5.2.3. Voicemail

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.

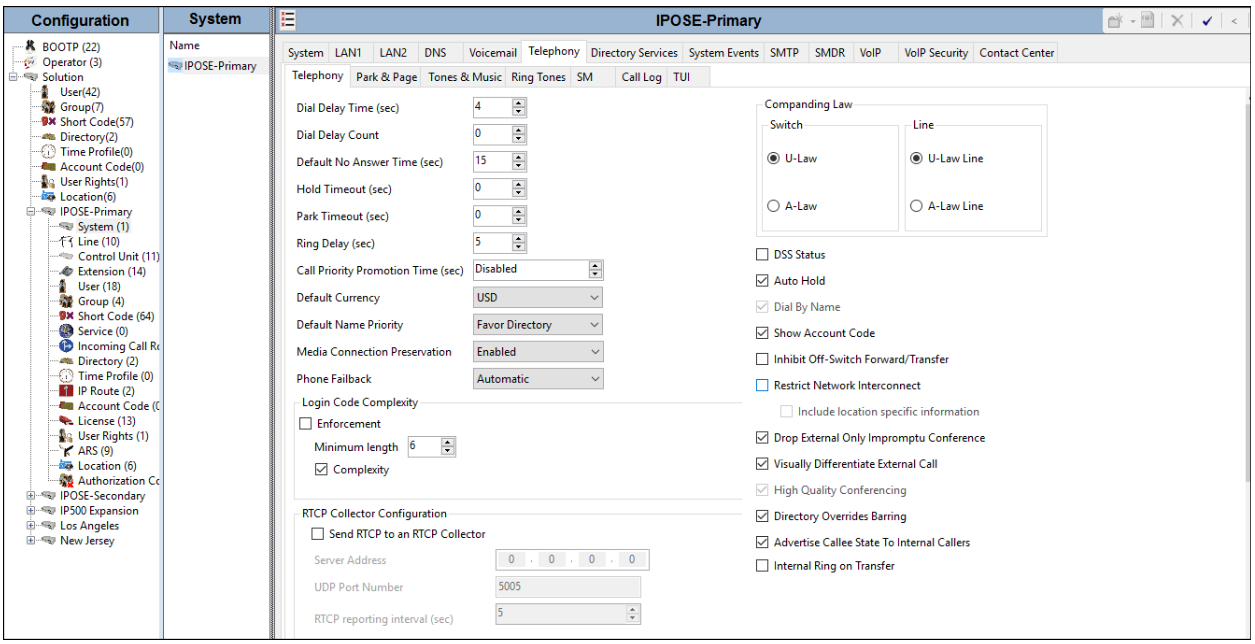


As described in [VZBIPT-IPO10.1], the "Callback" application of Avaya Voicemail Pro was used to allow Voicemail Pro to call out via the SIP Line to Verizon Business IP Trunk service when a message is left in a voice mailbox.



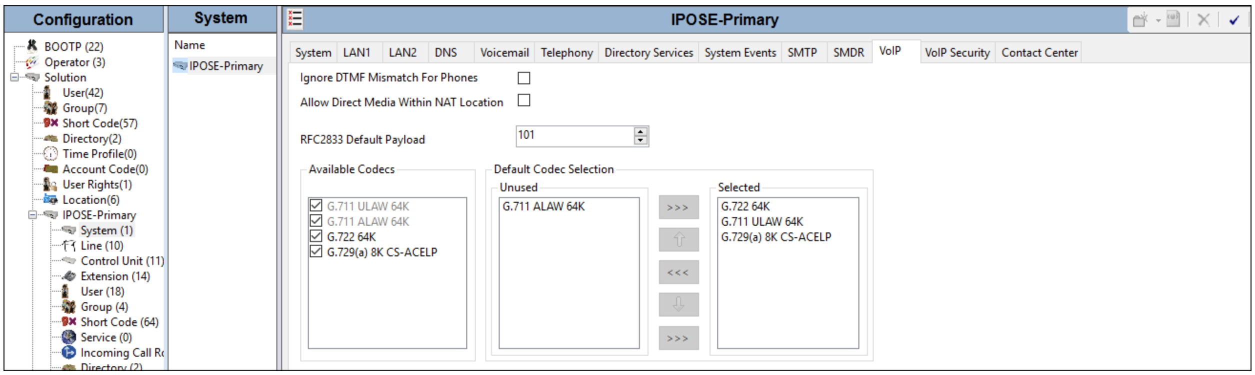
5.2.4. System Telephony Configuration

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



5.2.5. System Codecs Configuration

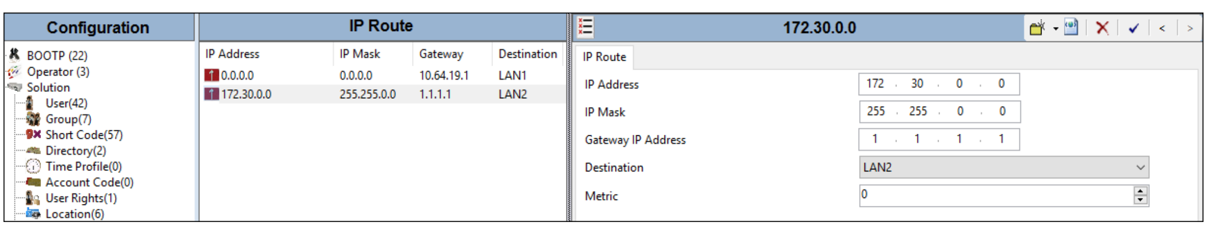
To view or change system codec settings, select the **VoIP** tab. On the left, observe the list of **Available Codecs**. In the example screen below, which is not intended to be prescriptive, the box next to each codec is checked, making all the codecs available in other screens where codec configuration may be performed (such as the SIP Line in **Section 5.4**). The **Default Codec Selection** area enables the codec preference order to be configured on a system-wide basis, using the up, down, left, and right arrows. By default, all IP (SIP and H.323) lines and extensions will assume the system default codec selection, unless configured otherwise for the specific line or extension. The **RFC2833 Default Payload** parameter is set to "**101**", the value preferred by Verizon Business.



5.3. IP Route

In the sample configuration, the IP Office LAN1 port is physically connected to the local area network switch at the IP Office customer site. The default gateway for this network is "**10.64.19.1**".

The IP Office LAN2 port is physically connected to the Verizon PIP network and has a default gateway of "**1.1.1.1**". To add an IP Route in IP Office, right-click **IP Route** from the Navigation pane, and select **New**. To view or edit an existing route, select **IP Route** from the Navigation pane, and select the appropriate route from the Group pane. The following screen shows the Details pane with the relevant route using **Destination** "**LAN2**".



5.4. SIP Line

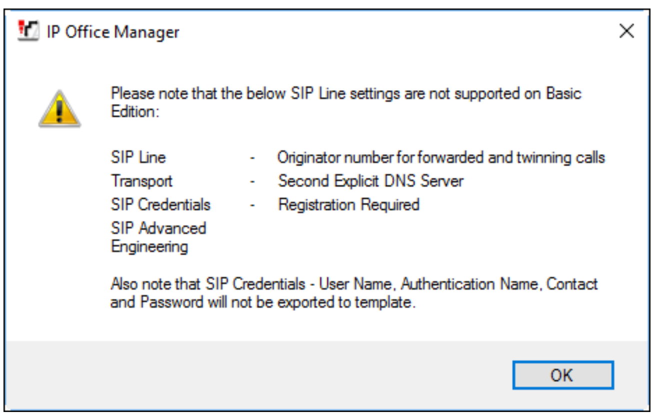
This section shows the configuration screens for the SIP Line in IP Office Release 10.1. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the 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:



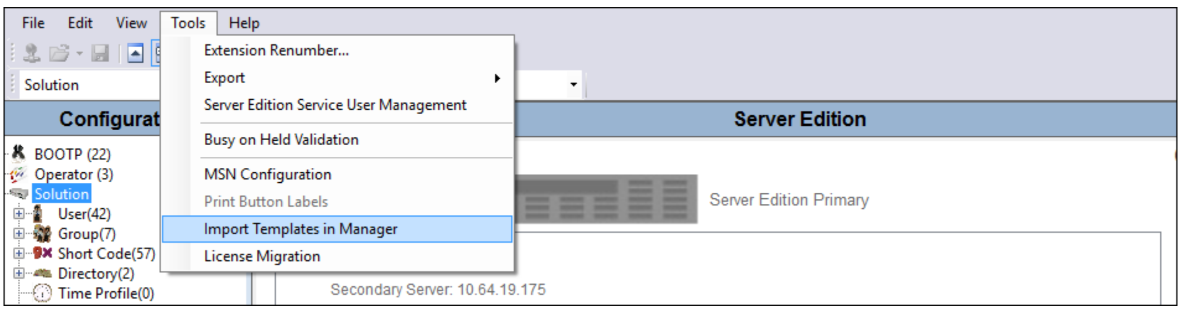
Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.4.3 – 5.4.7**.

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.

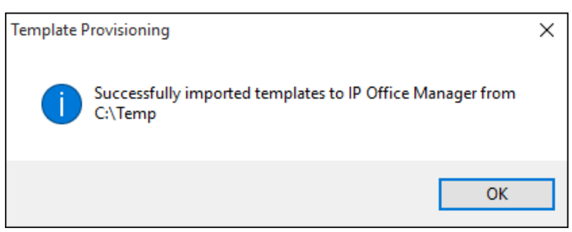
1. Copy a previously created template file to a location (e.g., *temp*) on the same computer where IP Office Manager is installed.

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



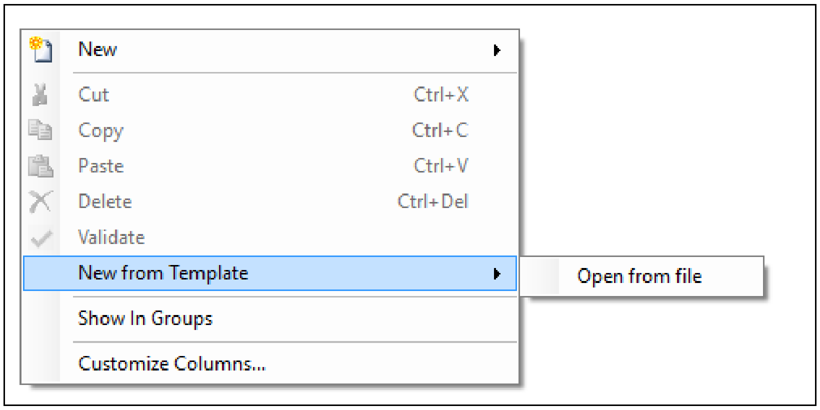
3. A folder browser will open (not shown). Select the directory used in **step 1** to store the template(s) (e.g., *temp*). In the reference configuration, template file **New-Template.xml** was imported. The template files are automatically copied into the IP Office default template location, **C:\Program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

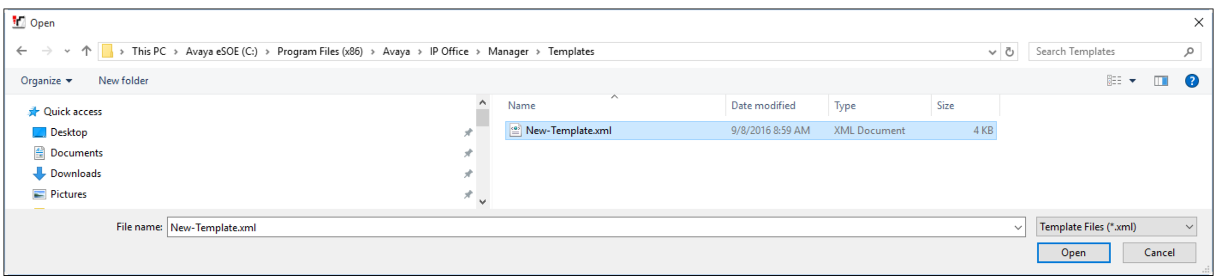


5.4.2. Creating a SIP Trunk from an XML Template

1. To create the SIP Trunk from a template, right-click on **Line** in the Navigation pane, and hover over **New from Template**, and select **Open from file**.



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 **2**).

Line Number	Line Type	Line SubType
1	IP Office Line	WebSocket Server SCN
3	IP Office Line	WebSocket Server SCN
2	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**.

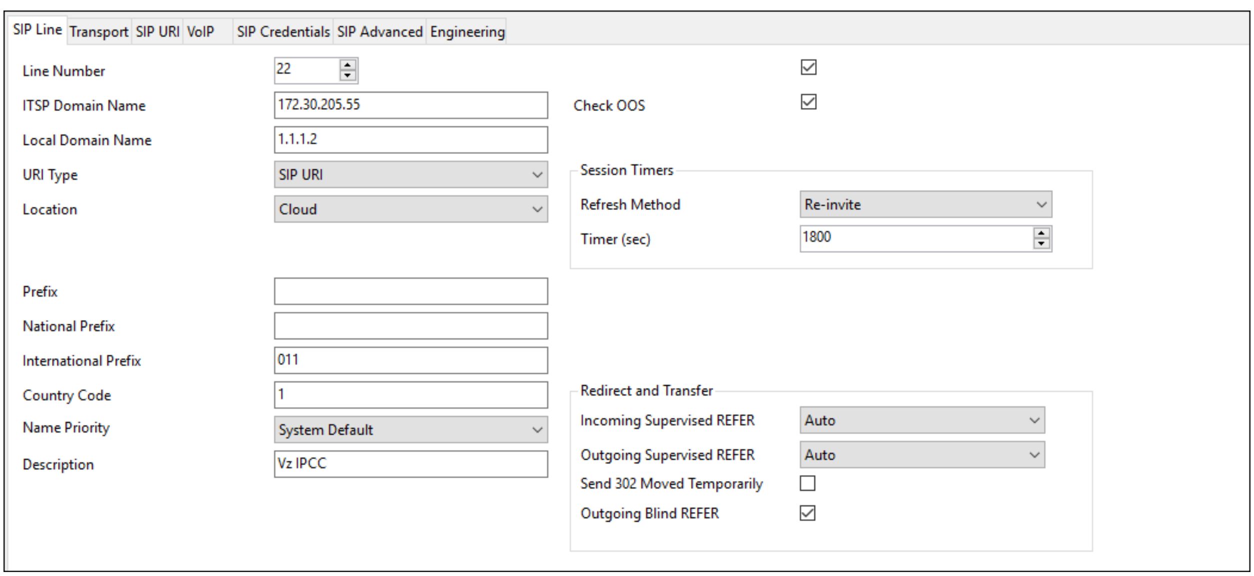
5.4.3. SIP Line – SIP Line Tab

The **SIP Line** tab in the Details pane is shown below for **Line Number 22**, used for the Verizon Business IPCC service. The **ITSP Domain Name** is configured to the IP address supplied by Verizon (“**172.30.205.55**”). The **Local Domain** is set to the IP address of the Avaya IP Office LAN2 SIP trunking interface (e.g., “**1.1.1.2**”). 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 LAN2, as shown in **Section 5.2.2**.

In the sample configuration, the IP Office **Country Code** is set to 1. The From and PAI headers received from Verizon for calls from U.S. PSTN numbers contain "+1" before the calling PSTN number. By configuring the IP Office **Country Code** to 1, the caller ID display presented to IP Office users will be the PSTN number without any codes or prefixes. For example, a call from 3035387006 would display 3035387006. If the **Country Code** does not match the value following the "+" from Verizon, the IP Office user display would show the contents of the **International Prefix** field, followed by the value following the "+", followed by the PSTN number. For example, if the **Country Code** parameter were left blank, the IP Office user would see a display such as "01113035387006". Aside from display implications, if the **Country Code** is not configured, other patterns may also fail to match as expected, such as a match on the **Incoming CLI** field of the Incoming Call Route. See **Section 5.8.3** for configuration of incoming call routing based on the calling number.

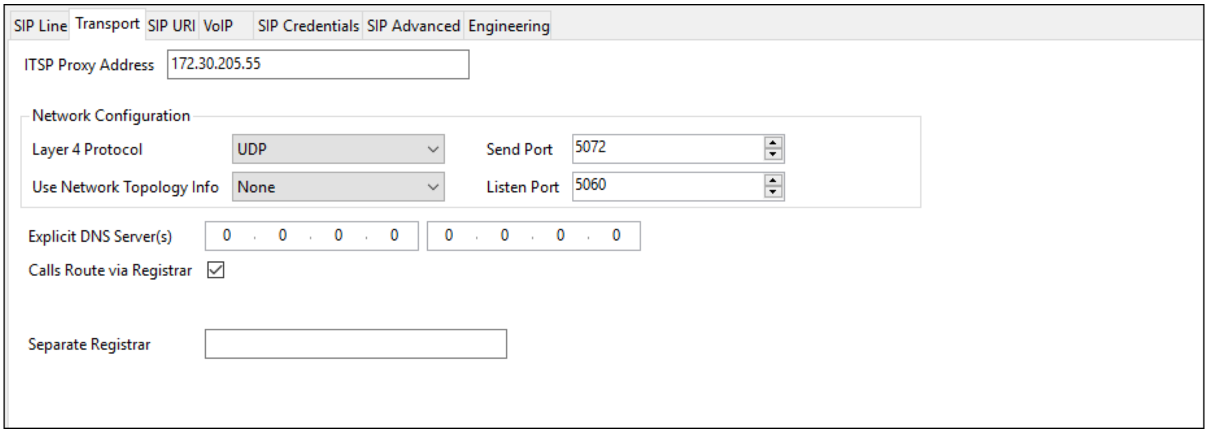
Under **Session Timers**, the **Refresh Method** is set to "**Re-invite**" and the **Timer (seconds)** is set to "**1800**". With this configuration, IP Office will send re-INVITEs every 15 minutes (half of the set value) to keep the active session alive.

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



5.4.4. SIP Line – Transport Tab

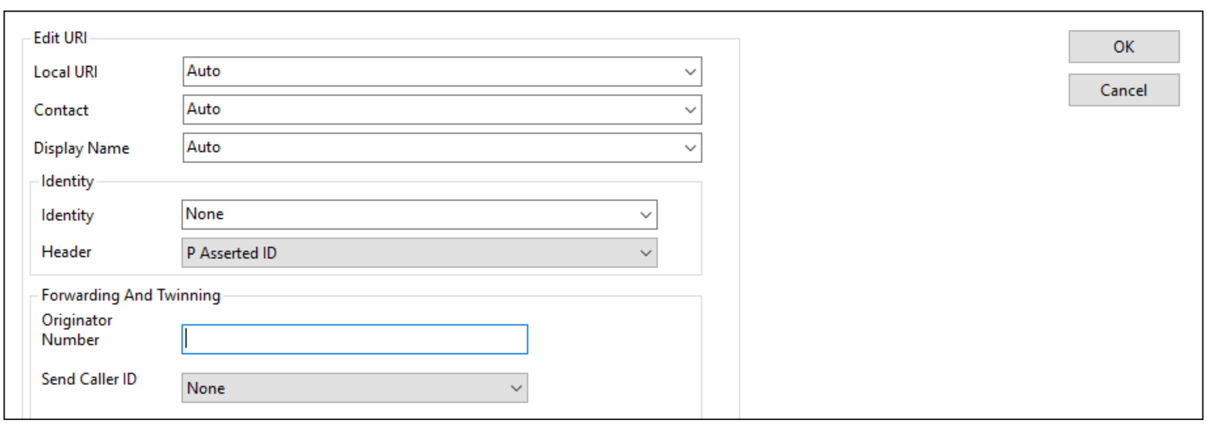
Select the **Transport** tab. The **ITSP Proxy Address** is set to the IP address supplied by Verizon ("**172.30.205.55**"). In the **Network Configuration** area, "**UDP**" is selected as the **Layer 4 Protocol**. The **Send Port** is set to the port number provided by Verizon Business. As shown in **Figure 1**, this port is "**5072**" in the sample configuration. The **Use Network Topology Info** parameter is set to "**None**".



5.4.5. SIP Line – SIP URI Tab

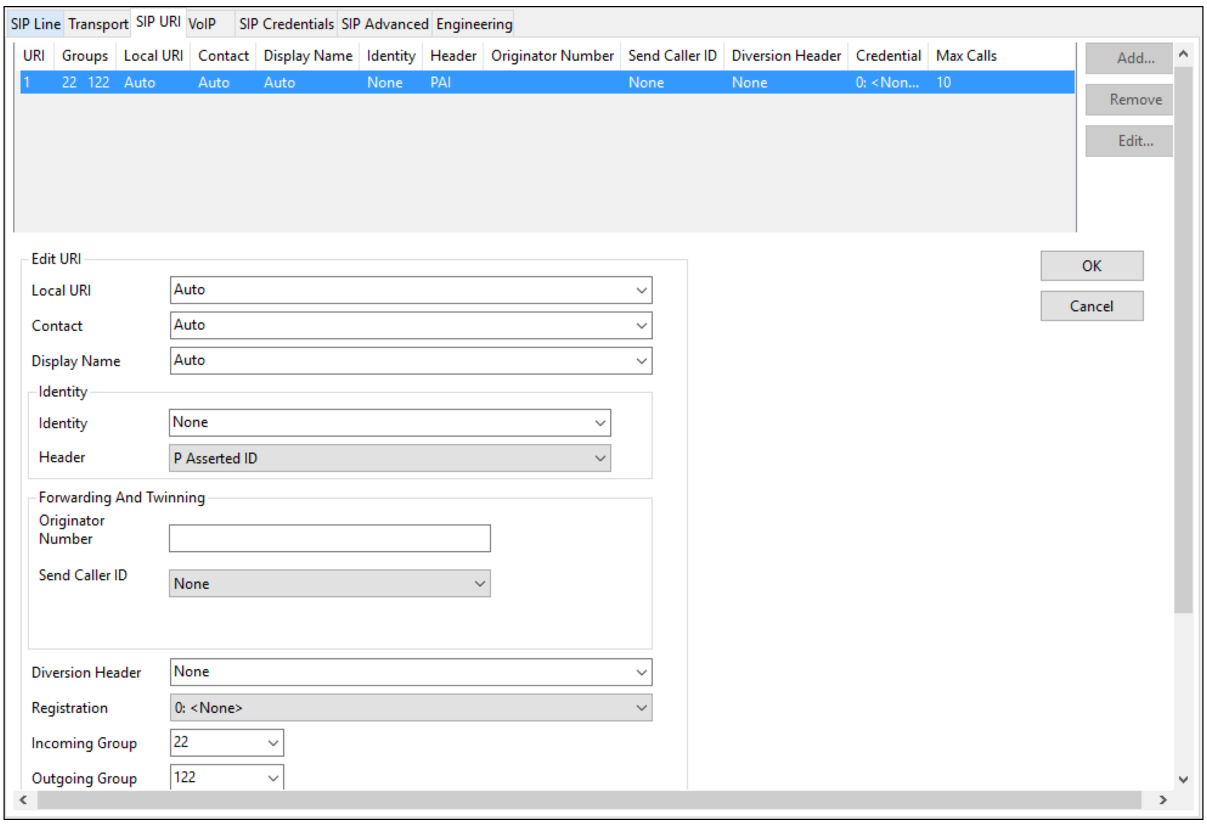
Select the **SIP URI** tab. To add a new SIP URI, click the **Add…** button. In the bottom of the screen, a New URI area will be opened. To edit an existing entry, click an entry in the list at the top, and click the **Edit…** button. In the bottom of the screen, the **Edit URI** area will be opened.

In the sample configuration, "**Auto**" is selected for the **Local URI**, **Contact** and **Display Name**. The **Identity** parameter is set to "**None**", and the **Header** parameter is set to "**P Asserted ID**". The **Forwarding And Twinning** section can remain default.



The **Diversion Header** is set to “**None**”. The **Registration** parameter is set to the default “**0: <None>**” since Verizon Business IPCC service does not require registration. The **Incoming Group** parameter, set here to “**22**”, 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 “**122**”, is relevant when using the SIP REFER method to transfer an inbound toll-free call back out the Verizon Business IPCC service. Click **OK** (not shown).

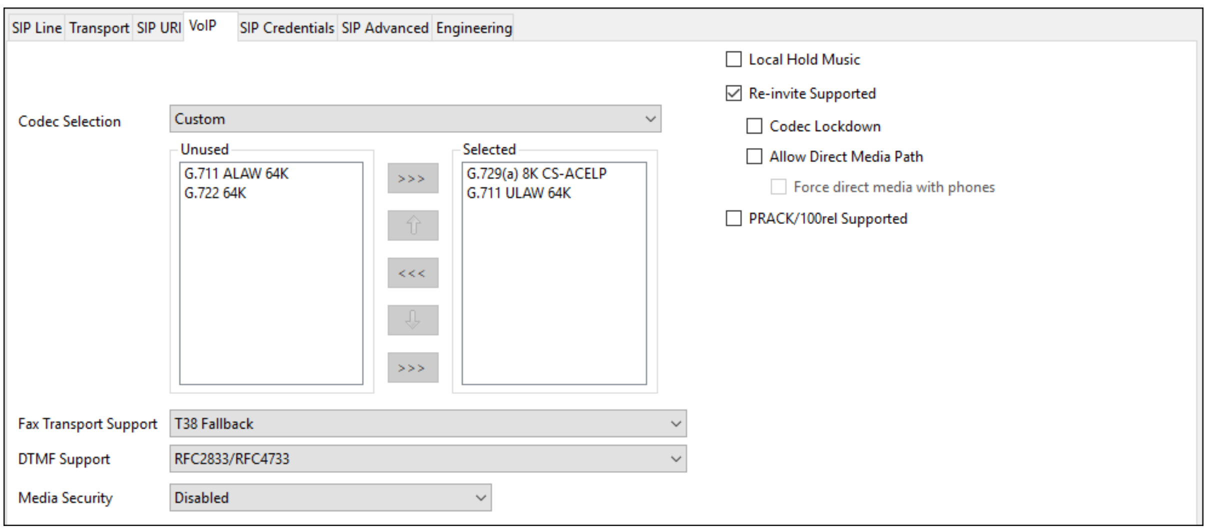
The complete URI entry is shown below.



5.4.6. SIP Line – VoIP Tab

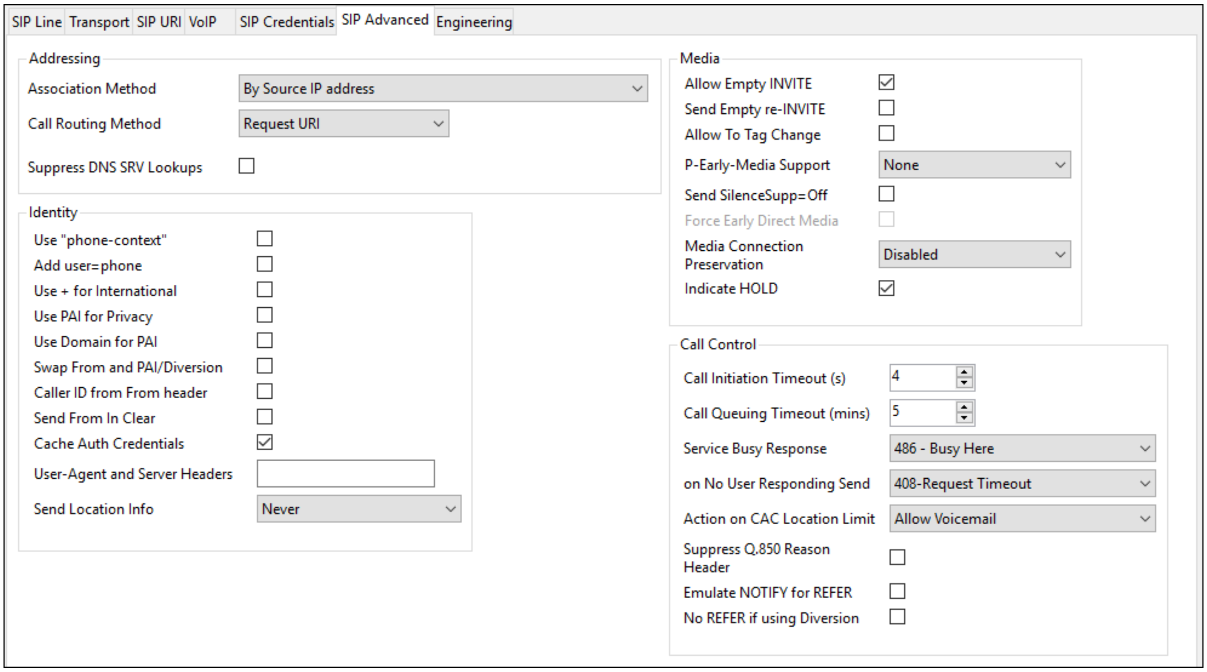
Select the **VoIP** tab. The **Codec Selection** drop-down parameter "**System Default**" (default) will match the codecs set in the system wide Default Selection list (**System → Codecs**). In the sample configuration, "**Custom**" is selected and codecs preferred by Verizon are included (i.e., G729(a) 8K CS-ACELP and G.711 ULAW 64K). This will cause IP Office to include G.729a and G.711MU in the Session Description Protocol (SDP) offer, in that order. 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. 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.

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.



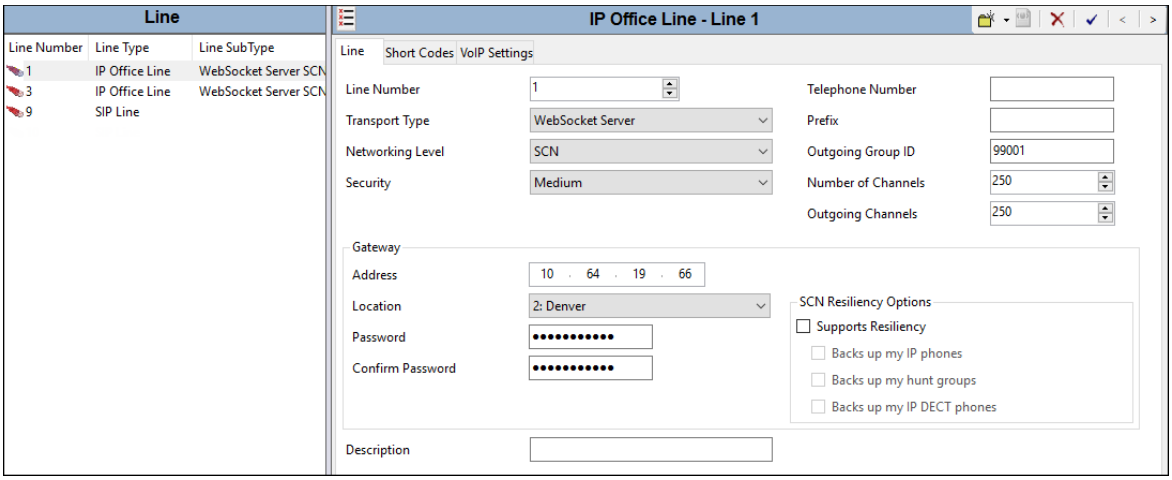
5.4.7. SIP Line – SIP Advanced Tab

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

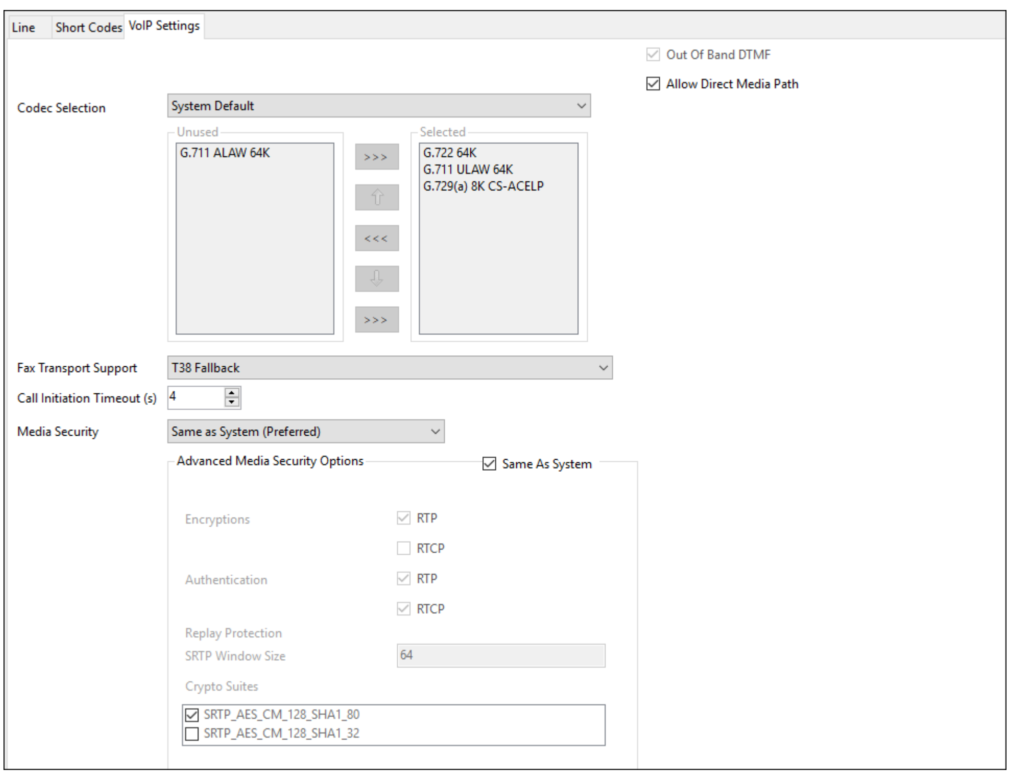


5.5. IP Office Line

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



Select the **VoIP Settings** tab. In the sample 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.

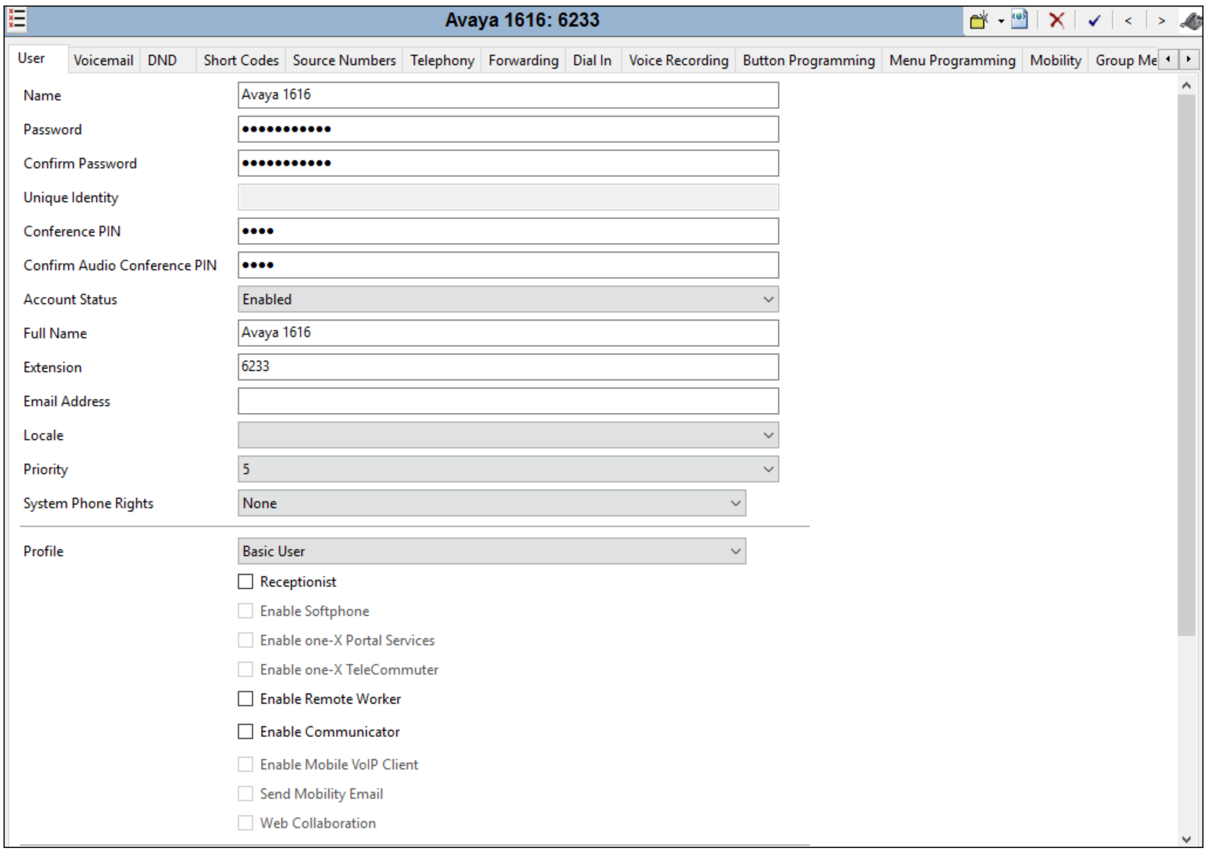


5.6. Users, Extensions, and Hunt Groups

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

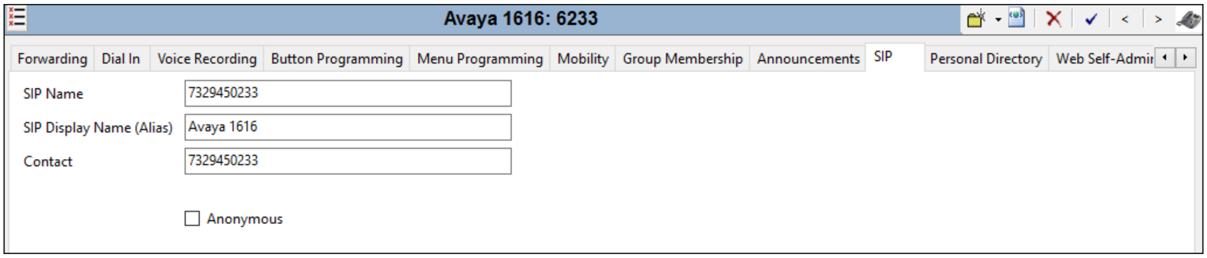
5.6.1. H.323 User 6233

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



The following screen shows the **SIP** tab for User 6233. In the sample configuration, the **SIP Name** and **Contact** parameters are configured with a Verizon Business IP Trunk DID number for the user, 7329450233. As shown in [VZBIPT-IPO10.1], these parameters configure the user part of the SIP URI in the From header for outgoing SIP trunk calls, and allow matching of the SIP URI for incoming calls from Verizon IP Trunk service, without having to enter this number as an explicit SIP URI for the SIP Line. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network. Since the user already has a local DID number configured, matching of the SIP URI for incoming toll-free calls is accomplished by setting the SIP Line **Local URI** and

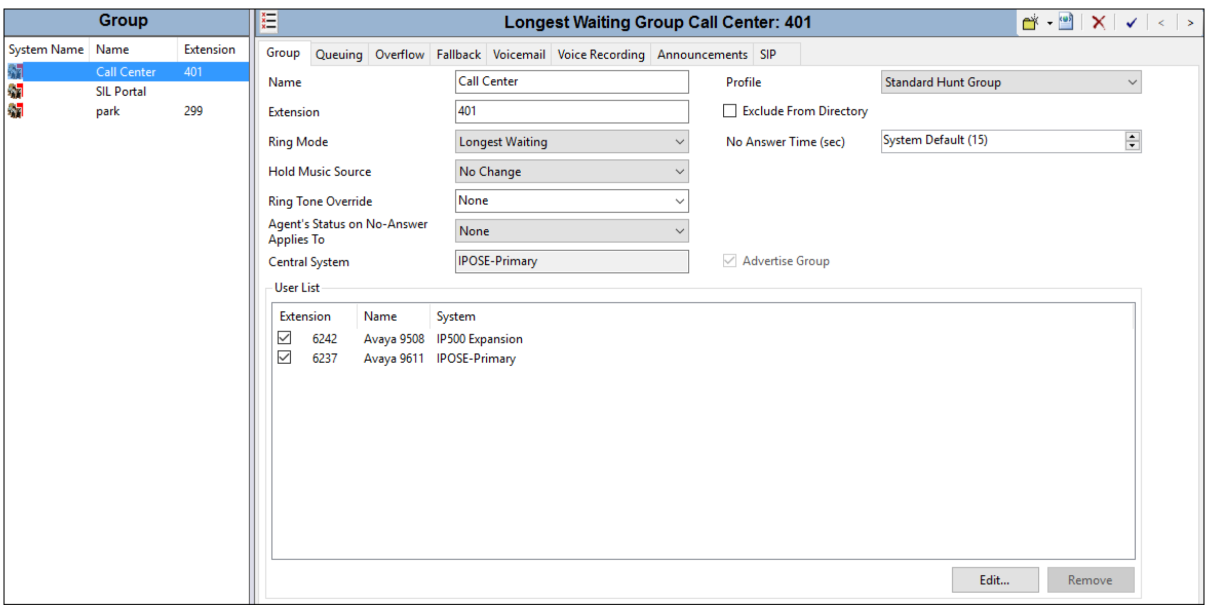
Contact fields to "**Auto**" as shown in **Section 5.4.5**, and specifying a unique incoming call route in **Section 5.8.1**.



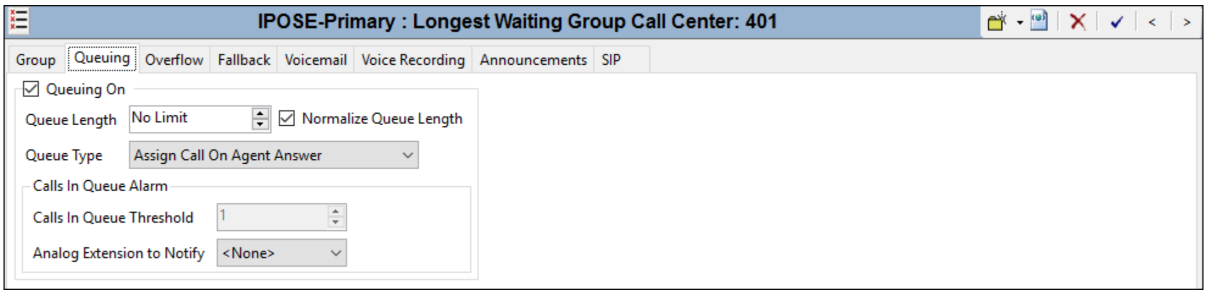
5.6.2. Hunt Groups

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** (not shown) 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 **Hunt Group** tab for a hunt group with **Extension** "**401**" and **Name** "**Call Center**". This hunt group was configured to contain various endpoints from **Figure 1**. The **Ring Mode** was set to "**LongestWaiting**" (i.e., most idle user to receive the next call). Click the **Edit** button to change the **User List**.

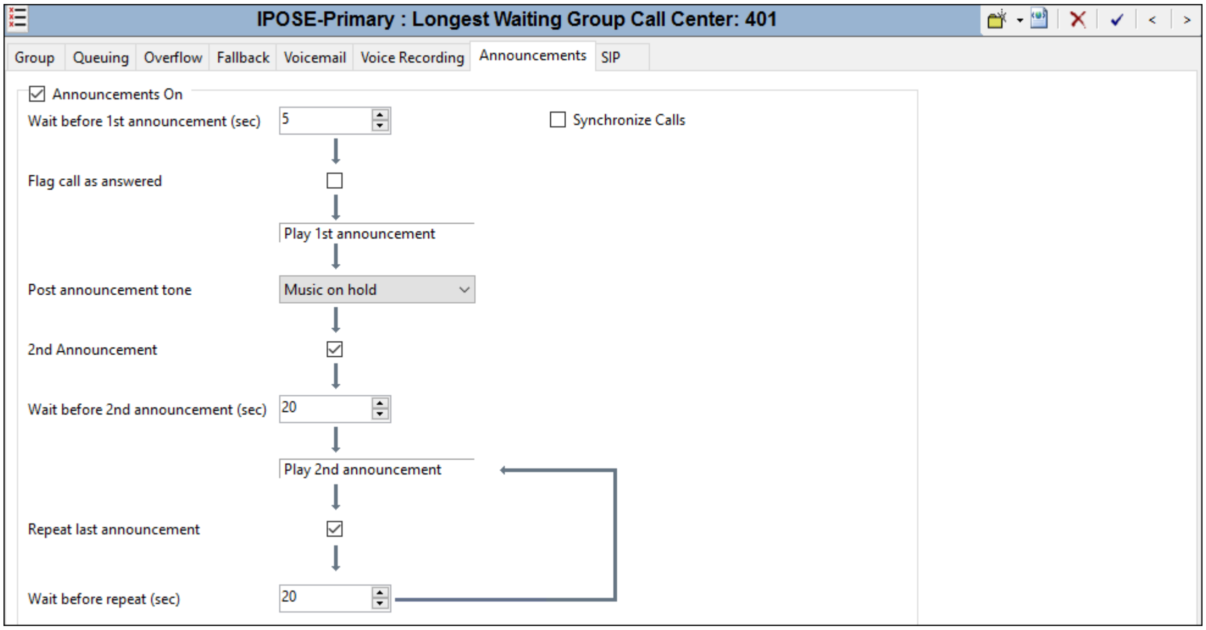


The following screen shows the **Queuing** tab for hunt group 401. In the sample configuration, the hunt group was configured to allow queuing so that incoming Verizon toll-free calls could be queued when all the members of the hunt group were busy on calls. In the testing associated with these Application Notes, the **Queue Length** was varied using both "No Limit" and specifically sized queues. For example, if the **Queue Length** is configured to 2, and if two calls are already in queue, a third call to the Verizon toll-free number corresponding to this hunt group will get busy tone because IP Office will send a 486 Busy Here (i.e., if there is no Voicemail for the hunt group). As another example, if the **Queue Length** has a fixed limit of 2, and if two calls are already in queue, a third call to the Verizon toll-free number destined for this hunt group from a priority caller (see **Section 5.8.3**) will be queued ahead of non-priority callers, temporarily expanding the queue.



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

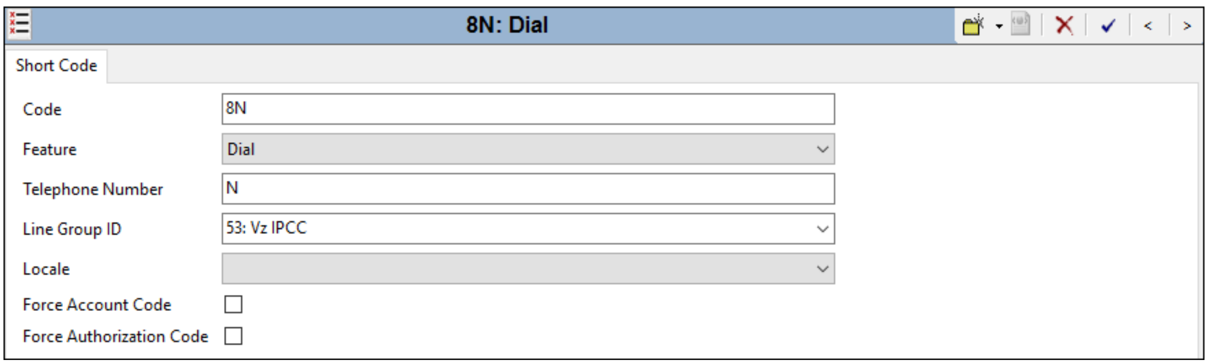
IP Office supports priority for queuing. For example, if low priority calls are waiting in queue, a higher priority call entering queue can be moved to the front of the queue and serviced before lower priority callers. For an inbound SIP trunk call, the priority can be specified on the Incoming Call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



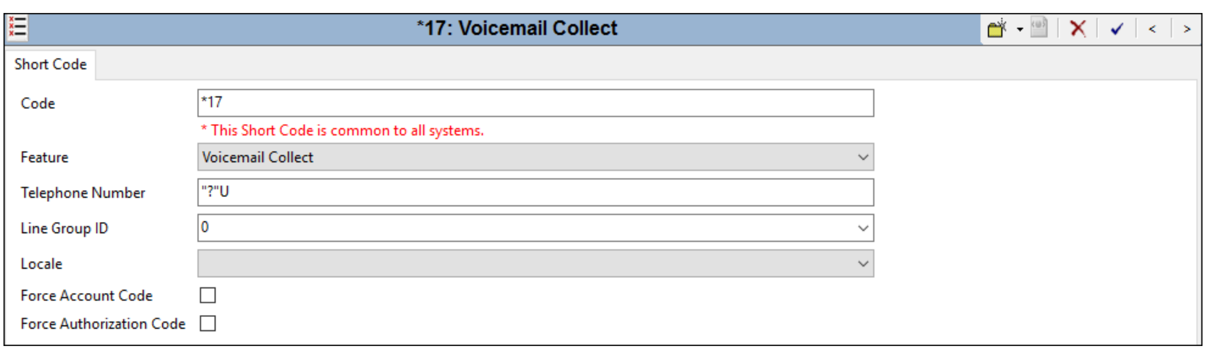
5.7. Short Codes

In this section, various examples of IP Office short codes will be illustrated. To add a short code, right click on **Short Code** (not shown) in the Navigation pane, and select **New**. To edit an existing short code, click **Short Code** in the Navigation pane, and the short code to be configured in the Group pane.

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

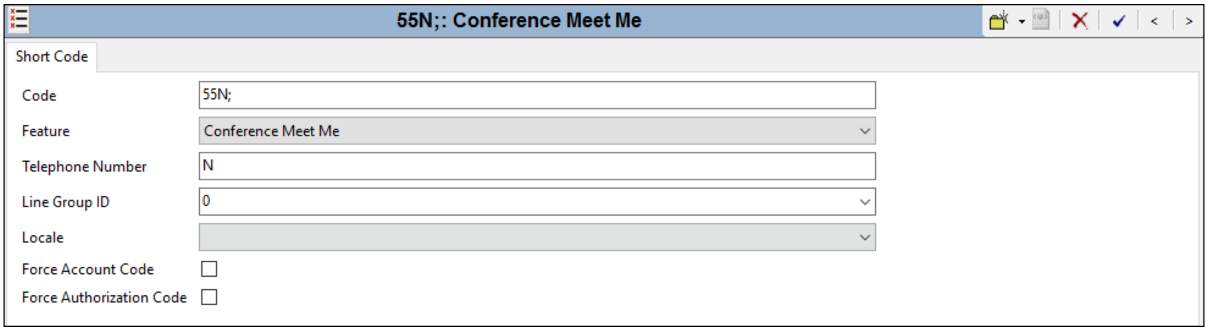


The following screen illustrates a short code that acts like a feature access code rather than a means to access a SIP Line. In this case, the **Code** "***17**" is defined for **Feature** "**Voicemail Collect**". This short code will be used as one means to allow a Verizon toll-free number to be programmed to route directly to voice messaging, via inclusion of this short code as the destination of an Incoming Call Route. See **Section 5.8** for configuration of Incoming Call Routes.



The following screen illustrates another short code. In this case, the **Code** "**55N;**" is defined for **Feature** "**Conference Meet Me**" and **Telephone Number** "**N**". In the verification of these Application Notes, this short code was used in conjunction with a Voicemail Pro module named

“MeetMeConf”. Although the Voicemail Pro configuration is beyond the scope of these Application Notes, the module enabled a PSTN caller to dial a Verizon toll-free number, be prompted by Voicemail Pro to enter a conference ID and PIN, and then be transferred to the appropriate meet-me conference based on the ID entered by the caller. Local IP Office callers could also dial 55xxxx to join the corresponding conference ID.

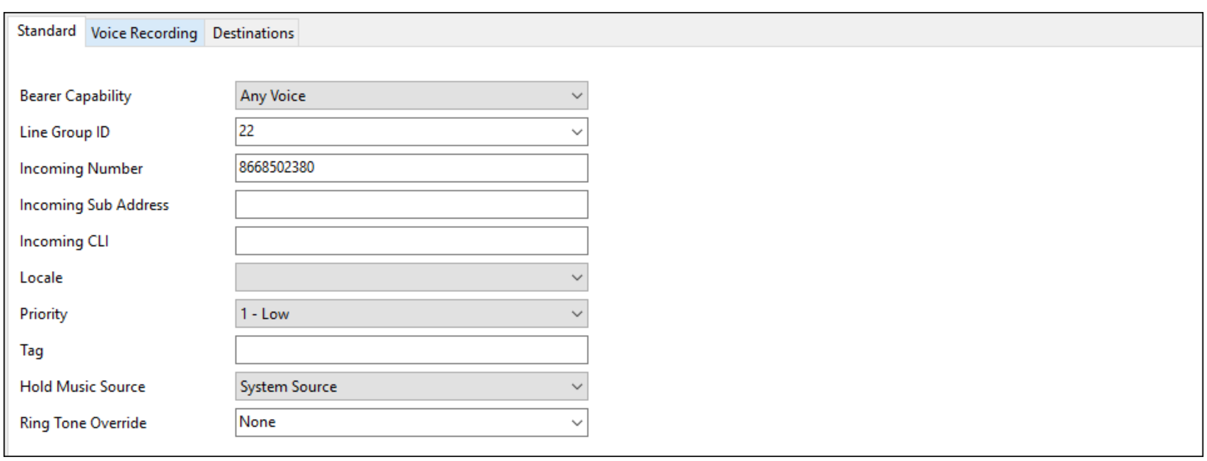


5.8. Incoming Call Routes

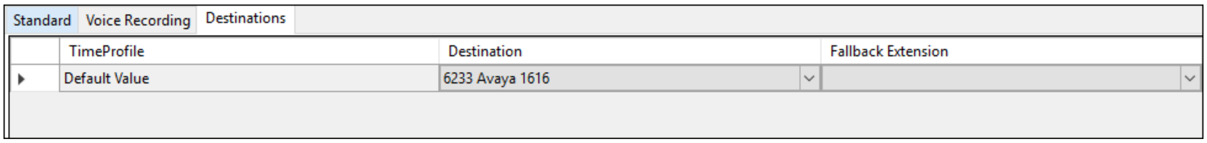
In this section, IP Office Incoming Call Routes are illustrated. Each Incoming Call Route will map a Verizon Business toll-free number to a destination user, group, or function on IP Office. In some cases, the destination will be chosen based on the combination of the toll-free number and the caller ID of the caller. Sample mappings are summarized in **Table 1** in **Section 3**. To add an incoming call route, right click on **Incoming Call Route** (not shown) in the Navigation pane, and select **New**. To edit an existing incoming call route, select **Incoming Call Route** in the Navigation pane, and the appropriate incoming call route to be configured in the Group pane.

5.8.1. Incoming Call Route to a Specific Telephone Extension

In the screen shown below, the incoming call route for **Incoming Number** “**8668502380**” is illustrated. The **Line Group ID** is “**22**”, matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to the Verizon Business IPCC service, in **Section** Error! Reference source not found..



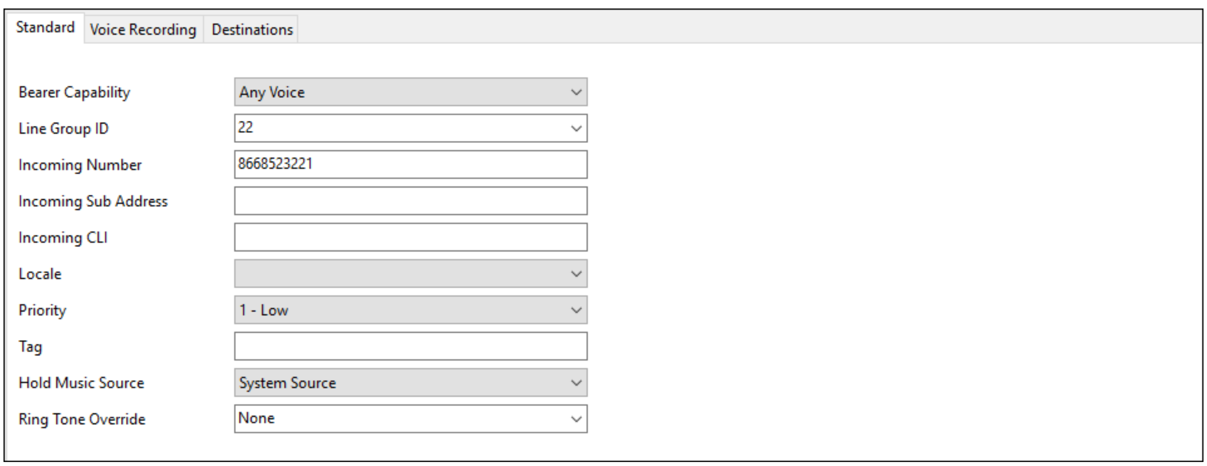
Select the **Destinations** tab. From the **Destination** drop-down, select an extension to receive the call when a PSTN user dials 8668502380. As shown in **Table 1**, 8668502380 is the number associated with IP Office user extension 6233. (The **Destination** was changed in the course of testing to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



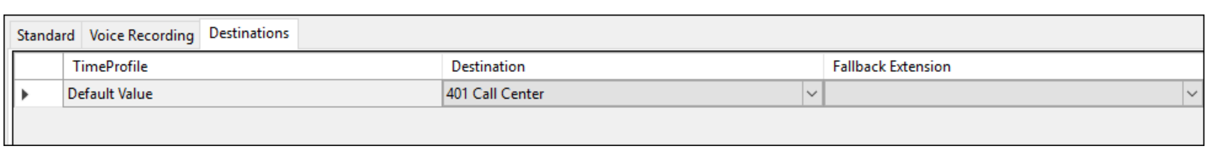
Incoming Call Routes for other direct mappings of toll-free numbers to IP Office users are not presented here, but are configured in the same fashion.

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

In the screen shown below, an incoming call route for **Incoming Number** "**8668523221**" is illustrated. The **Line Group ID** is "**22**", matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in **Section** Error! Reference source not found.. Optionally, the **Tag** parameter can be populated with a descriptive name that will associate the call with this incoming call route.



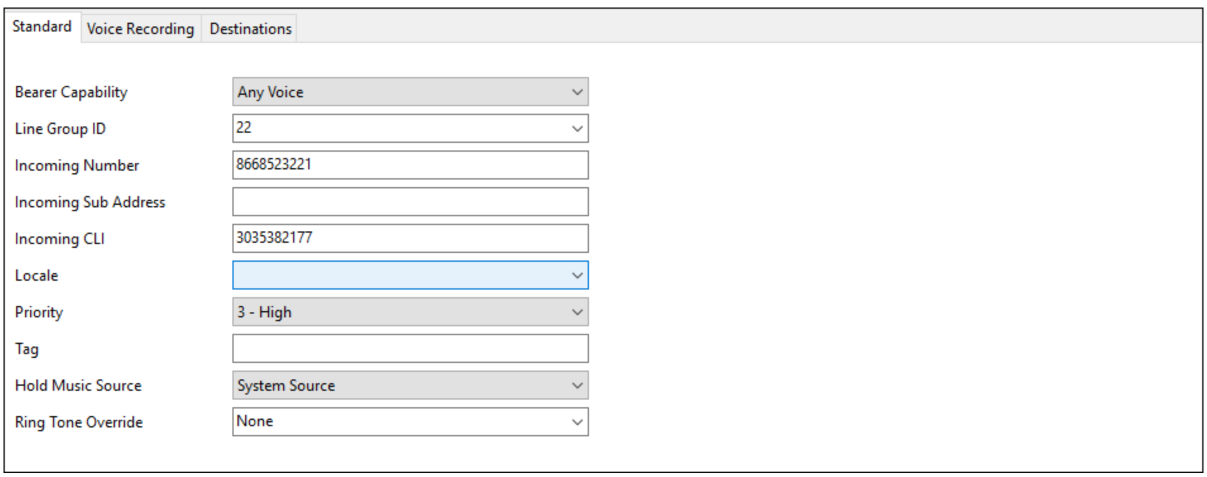
Select the **Destinations** tab. From the **Destination** drop-down, select the destination to receive the call when an arbitrary PSTN user dials 8668523221. As shown in **Table 1**, 8668523221 is the toll-free number associated with IP Office hunt group extension 401, the "Call Center" hunt group.



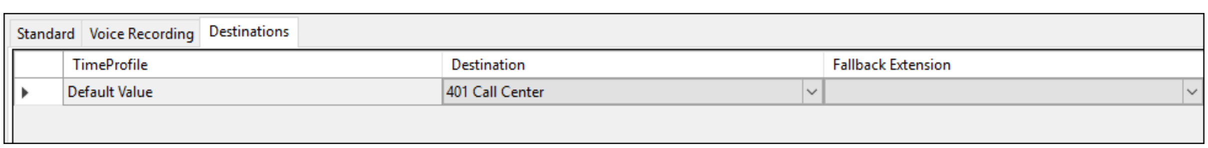
5.8.3. Incoming Call Routes Based on Calling Party Number

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

In the screen shown below, the incoming call route for **Incoming Number** "**8668523221**" and **Incoming CLI** "**3035382177**" is illustrated. The **Line Group ID** is "**22**", matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in **Section 5.4.5**. Note that the **Incoming Number** is the same as the toll-free number configured in the previous section. This route will be used for calls to the toll-free number specifically from a caller with caller ID "**3035382177**". In this case, to allow this caller to be treated with priority when calling in, the **Priority** field is set to "**3 – High**". Optionally, the **Tag** parameter can be populated with a descriptive name that will associate the call with this incoming call route.

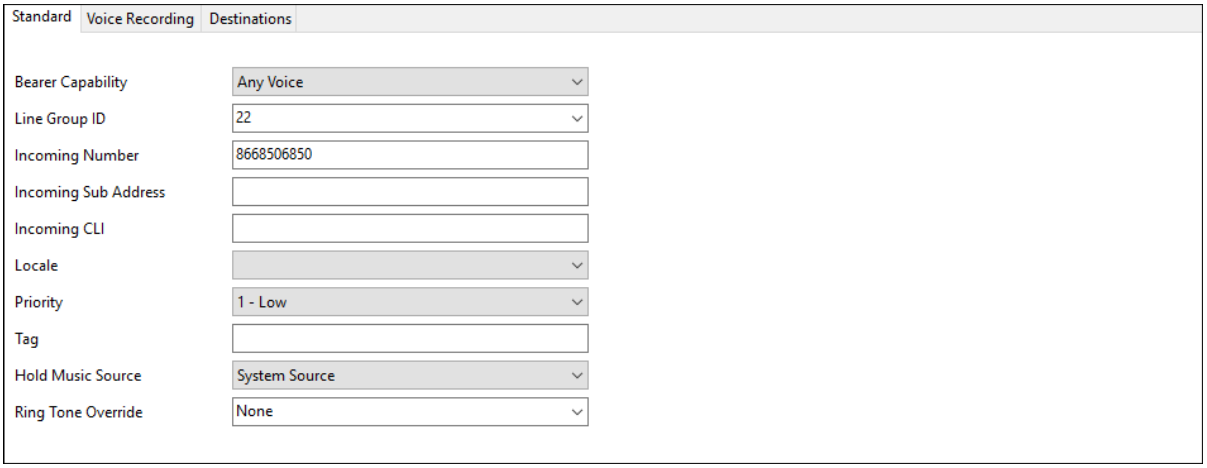


Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when PSTN user 3035382177 dials 8668523221. In this case, the **Destination** is also the hunt group "**401 Call Center**", but since high priority has been configured via the **Standard** tab, incoming calls from this caller will move to the front of the queue, and be serviced before calls waiting in queue from other non-priority callers.



5.8.4. Incoming Call Routes to Various IP Office Features

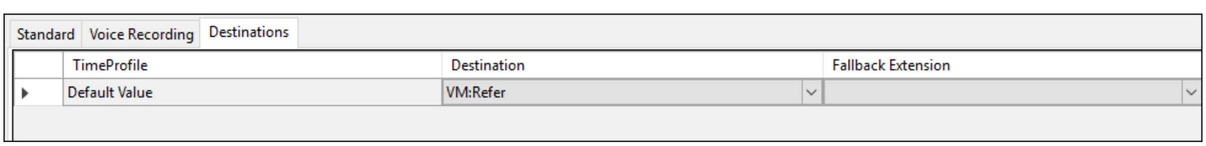
In the sample configuration, the incoming call route for **Incoming Number** "**8668506850**" was varied to test different destination features, such as Voice Mail, Mobile Call Control, Refer Call Redirection, and Conference Bridge, as shown in **Table 1** in **Section 3**. The screen showing the **Standard** tab for this toll-free number is shown below.



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

- "***17**" (short code "Voicemail Collect", as shown in **Section 5.7**). With this destination, an incoming call to 8668506850 will be delivered directly to voice mail, allowing the caller to log-in to voice mail and access messages.
- "**VM:MeetMe**" With this destination, an incoming call to 8668506850 will be delivered directly to the Voicemail Pro Module "MeetMe" created for use as a conference bridge.
- "**VM:Refer**" With this destination, an incoming call to 8668506850 will be delivered directly to the Voicemail Pro Module "Refer" created for use as a Refer Call Redirection example.

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

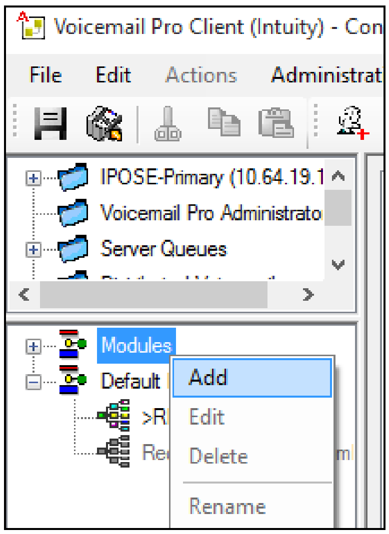


5.9. Voicemail Pro Refer Module

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

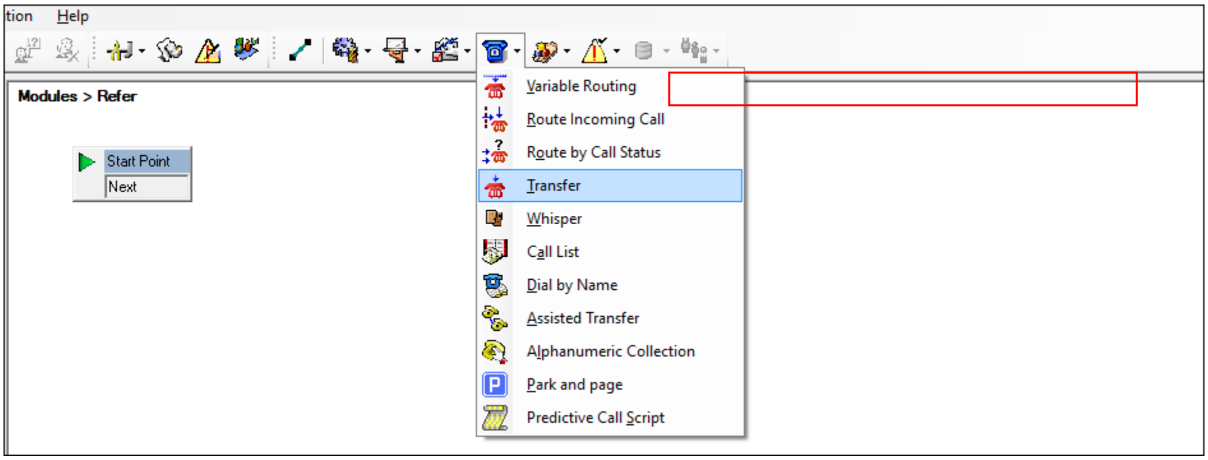
The Refer Module is provisioned to play an announcement to the caller, and then generate a Refer (without Replaces) back to the Verizon Business IPCC service. This is accomplished via the Voicemail Pro Client interface.

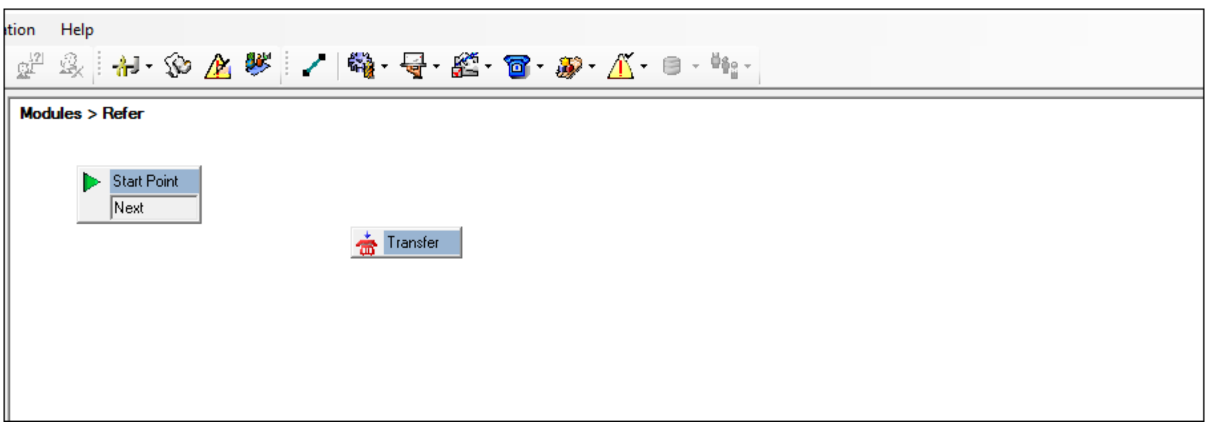
From the IP Office Manager PC, select **Start → All Apps → IP Office → Voicemail Pro Client** to launch the Voicemail Pro Client interface. Navigate to **File → Login**, select the proper Voicemail Pro system, and log in using the appropriate credentials (not shown). Create a **Start Point** by right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



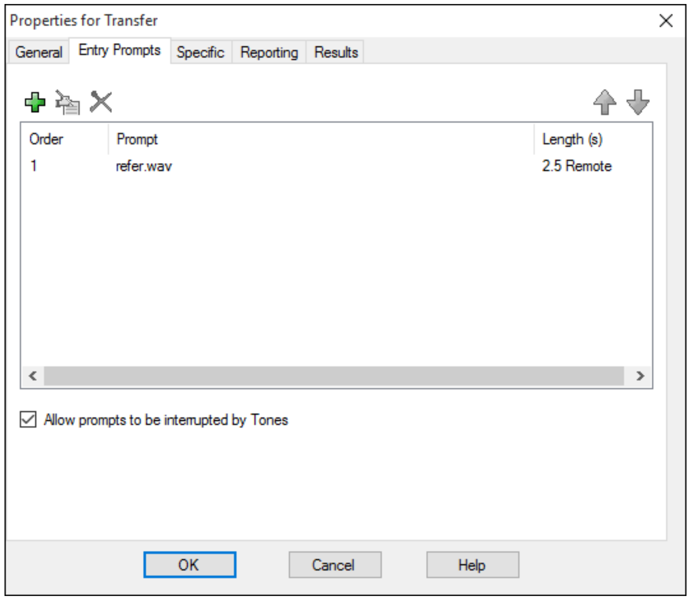
Enter a name (e.g., **Refer**) and click on **OK** (not shown). The new Module "**Refer**" will appear, and a **Start Point** icon will appear in the work area.

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

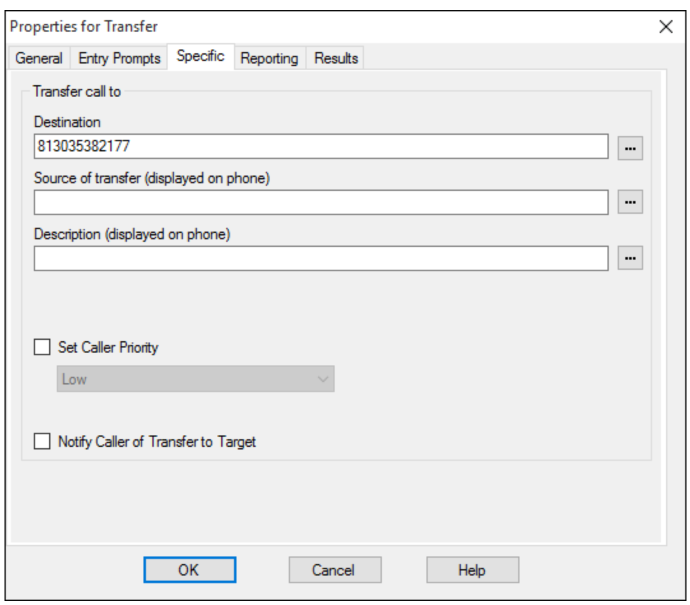




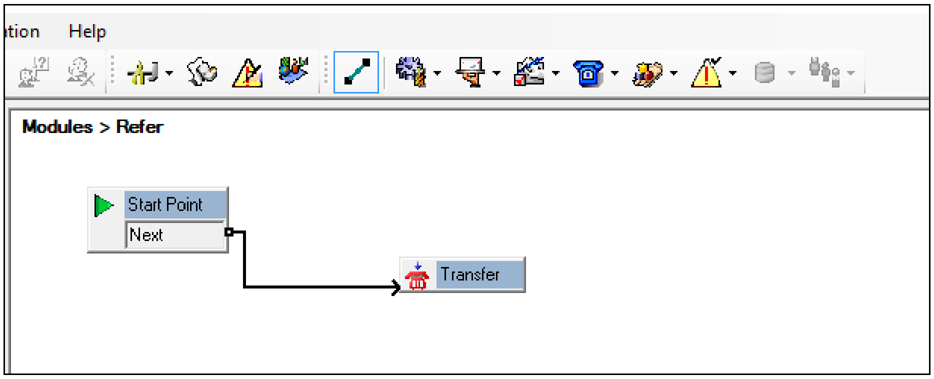
Double click on the **Transfer** icon, select the **Entry Prompts** tab and select or create an announcement to be played to the caller prior to the Refer (e.g., **refer.wav**). To modify an existing recording, double click on the .wav file and rerecord. If no .wav files exist, click on the icon to open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



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



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



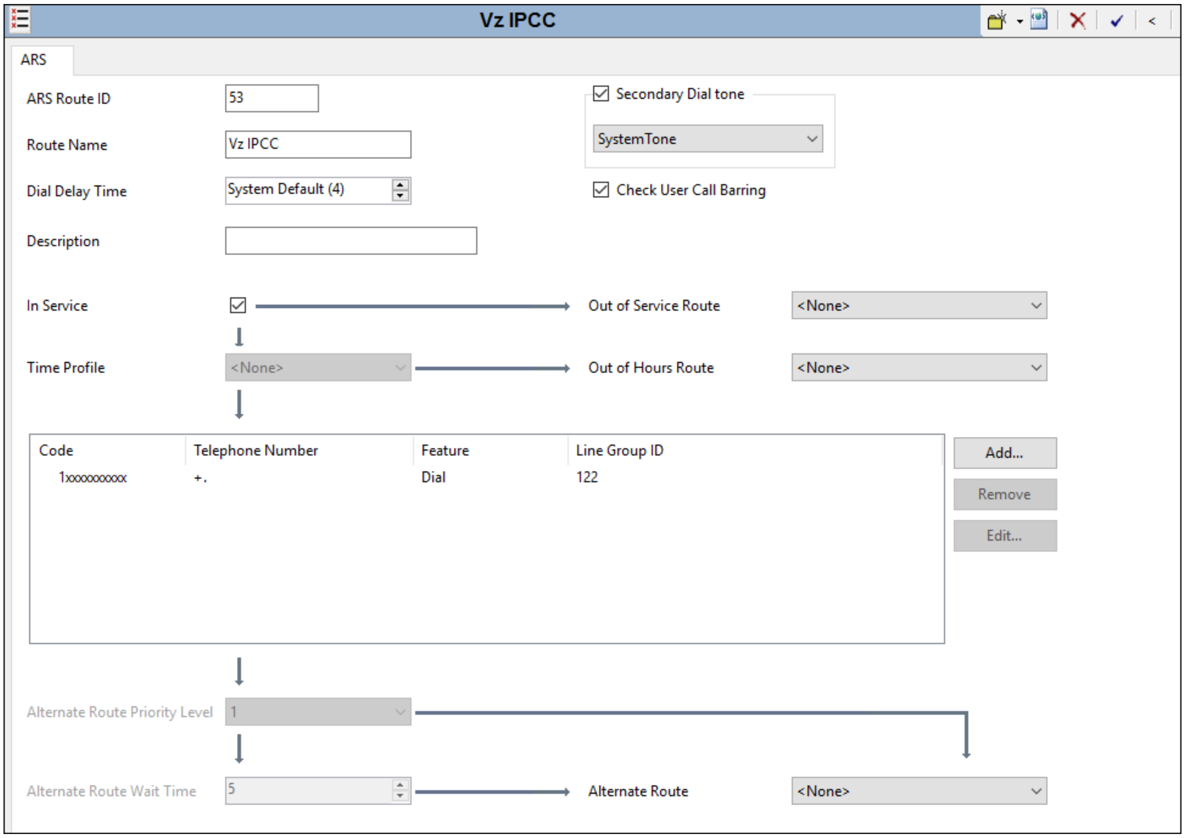
From the top menu select **File → Save & Make Live**, or select the icon.

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

5.10. Alternate Route Selection (ARS)

Alternate Route Selection (ARS) is used to route outbound traffic to the SIP line. 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. In the Details pane that appears, a collection of matching patterns (similar to short codes) can be entered to route calls as shown below.

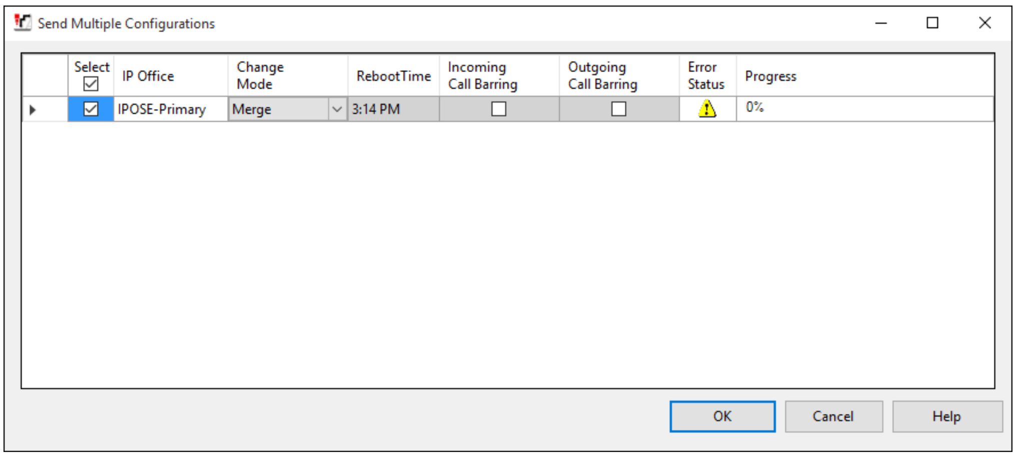
The following screen shows a sample ARS configuration for the route named "**VzIPCC**". Verizon Business IPCC service allows for blind transfers using the SIP REFER method. The sequence of **X**s used in the **Code** column specify the exact number of digits to be expected following the access. The entry below shows that for calls to area codes in the North American Numbering Plan, the user dials 8 (stripped by the short code configuration), followed by 11 digits. The **Telephone Number** is set to "**+.**". This prepends a plus sign (+) to the beginning of the number dialed, denoting a global E.164 number. This is the format preferred by Verizon Business IPCC service for the destination number specified in the Refer-To header. The **Line Group ID** is set to "**122**" matching the number of the **Outgoing Group** configured on the **SIP URI** tab of SIP Line 22 to Verizon Business (**Section** Error! Reference source not found.).



5.11. Save Configuration

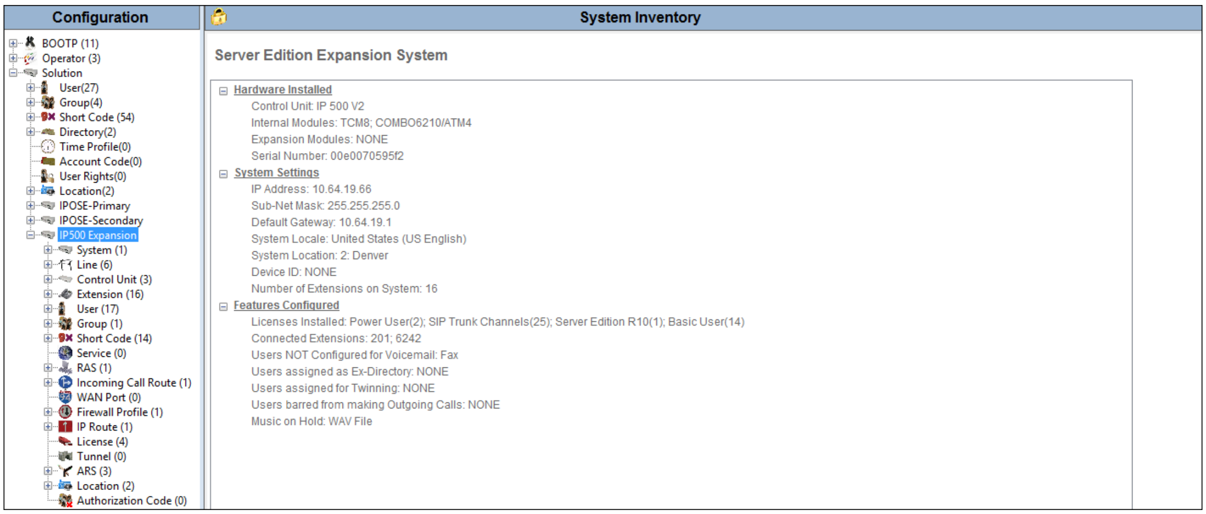
Navigate to **File → Save Configuration** in the menu bar at the top of the screen to save the configuration performed in the preceding sections.

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



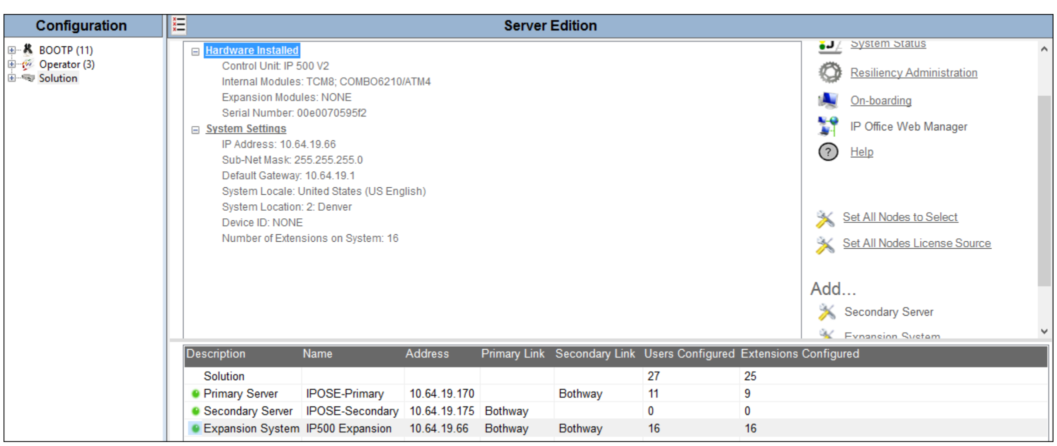
6. Avaya IP Office Expansion Configuration

The section illustrates relevant aspects of the Expansion System used in the verification of these Application Notes. The Expansion System is configured by logging in to the Primary server. Navigate to **File → Open Configuration** (not shown), select the proper Primary server 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.



6.1. 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.

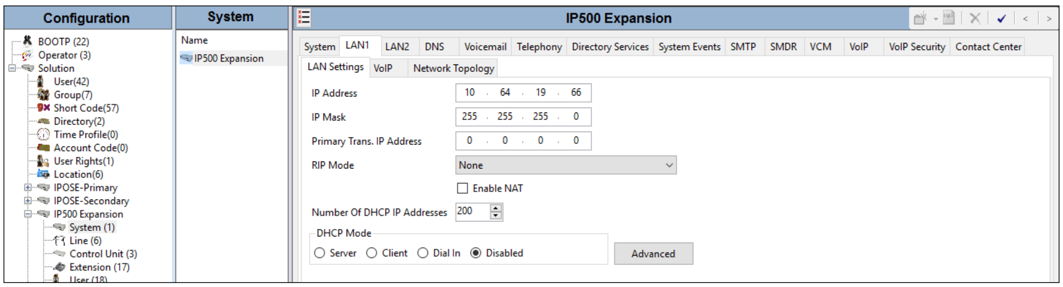


6.2. System Settings

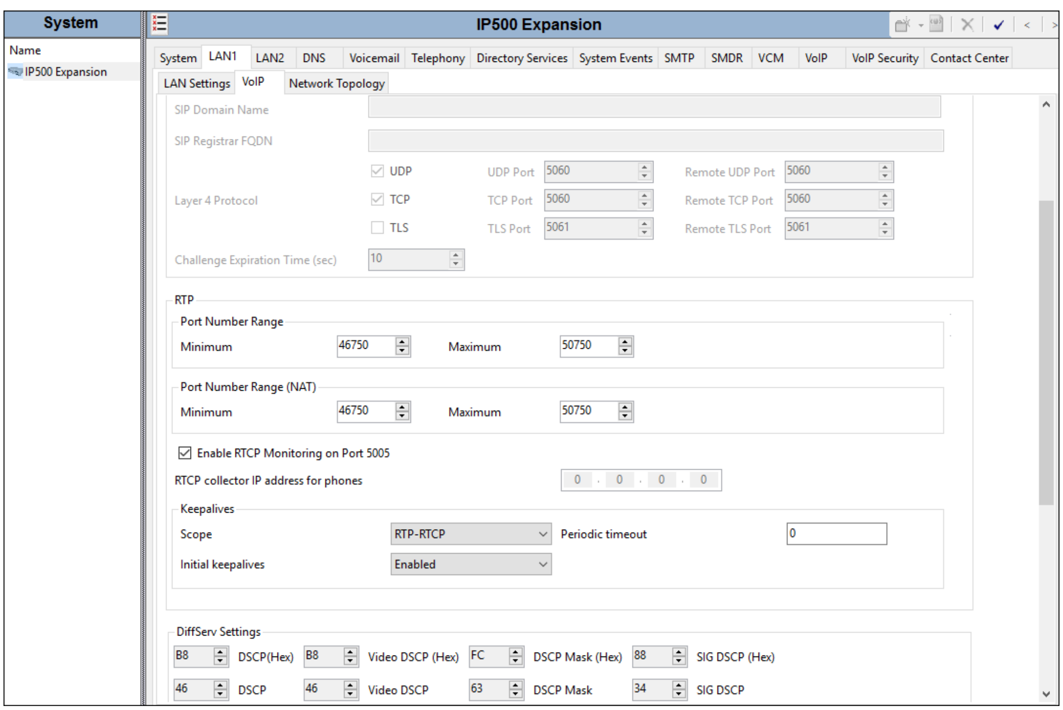
This section illustrates the configuration of system settings. Select **System** in the Navigation pane to configure these settings. The subsection order corresponds to a left to right navigation of the tabs in the Details pane for System settings. For all of the following configuration sections, the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

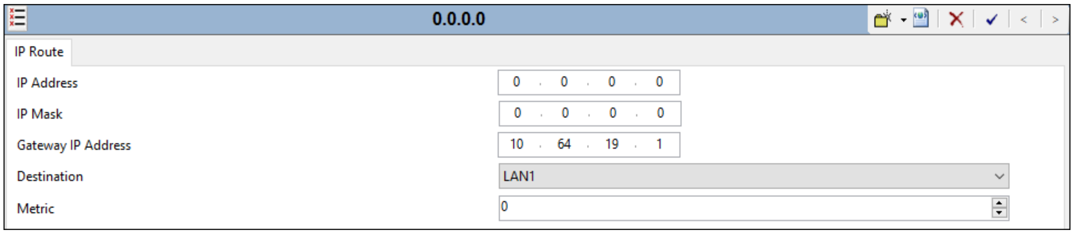


Select the **VoIP** tab as shown in the following screen. If desired, the **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media paths from Verizon to IP Office. The defaults are used here.



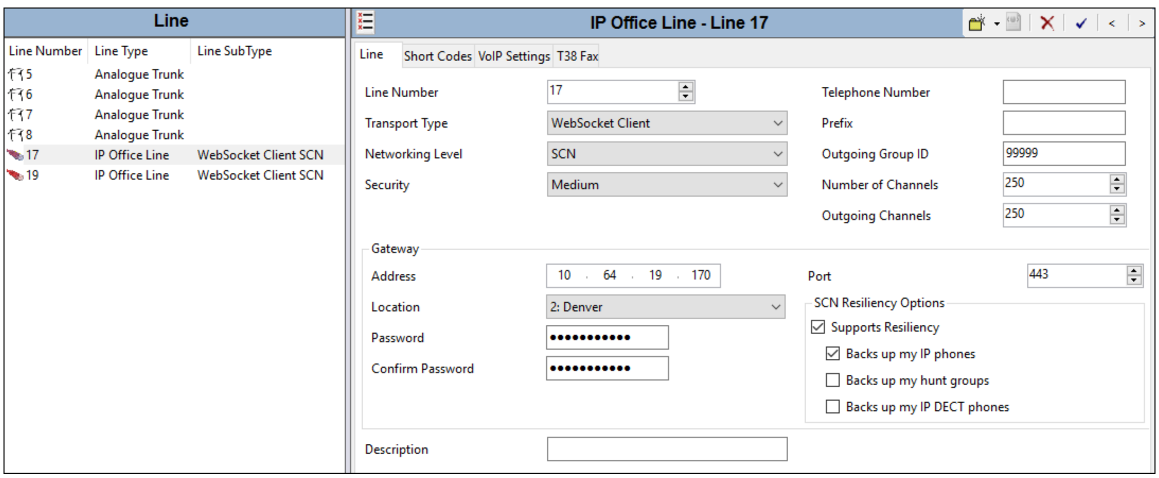
6.3. IP Route

Configuration is similar to the Primary server, as shown in **Section 5.3**. In the sample configuration, the Expansion System 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**".

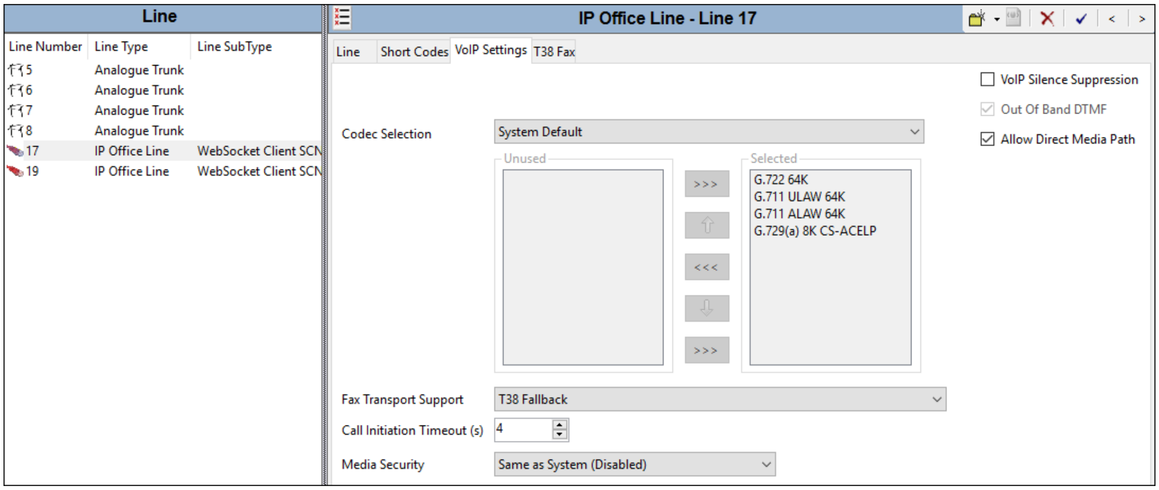


6.4. IP Office Line

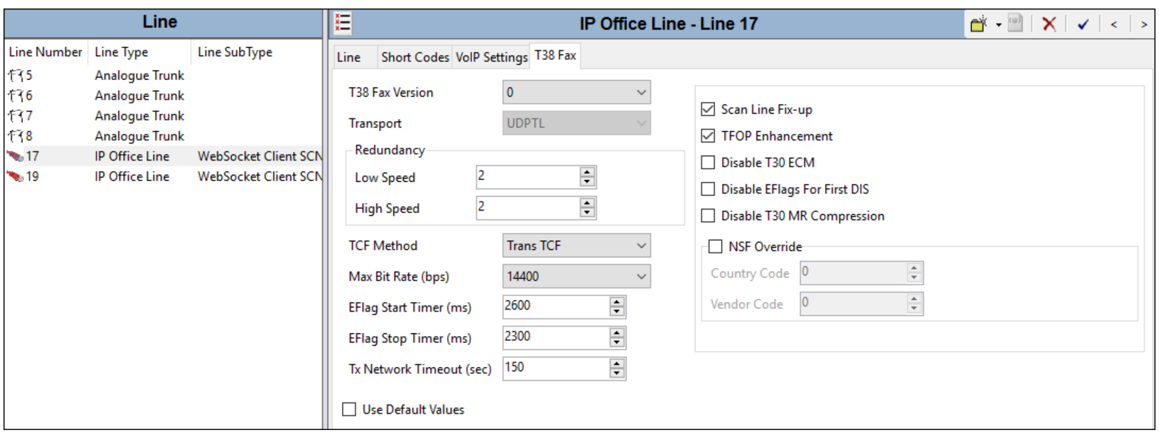
The IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution. Below is the IP Office Line to the Primary server.



In the sample 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.

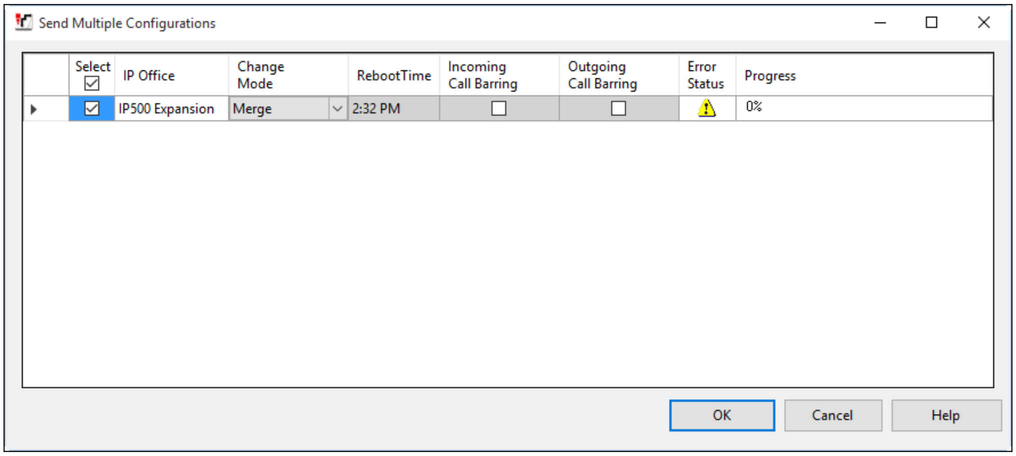


Select the **T38 Fax** tab. The **T38 Fax Version** is set to "**0**". In the **Redundancy** area, the **Low Speed** and **High Speed** parameters are set to "**2**". All other values are left at default.



6.5. 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.



7. Verizon Business Configuration

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

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

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

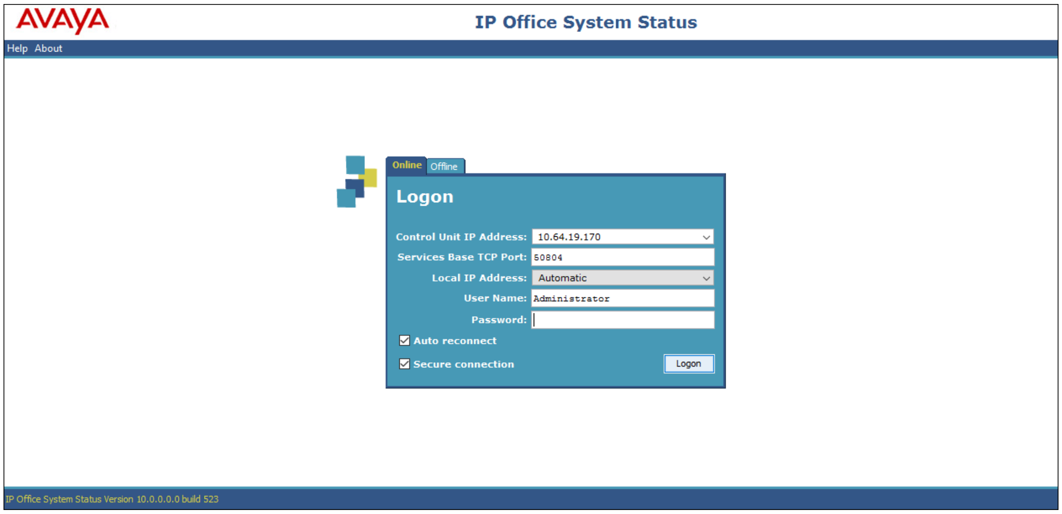
8. Verifications

This section provides sample verifications of the Avaya configuration with Verizon Business IPCC service.

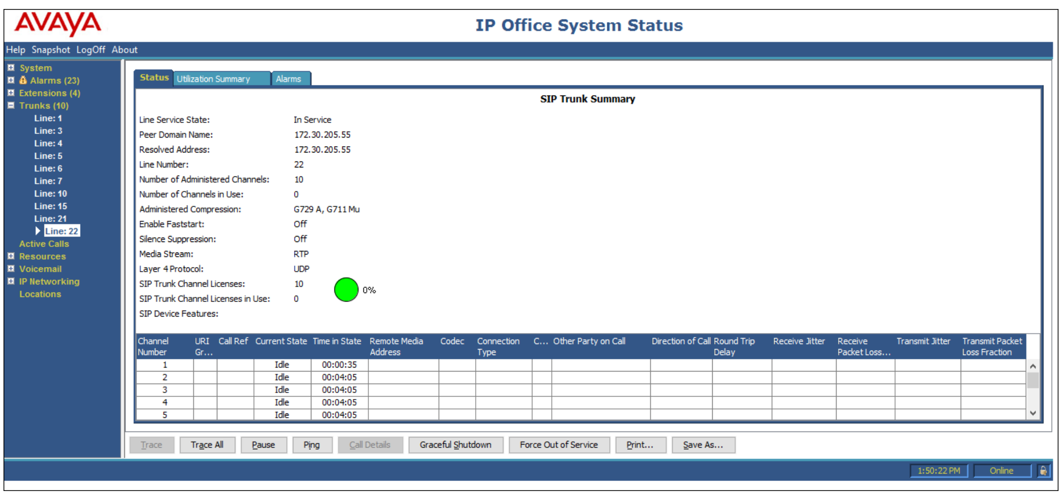
8.1. System Status

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 Server Edition Primary Server LAN1 interface. Click on **System Status** to launch the application.

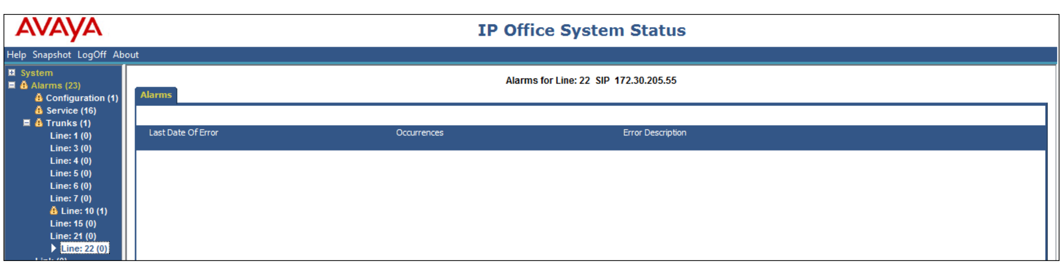
The following screen shows a sample **Logon** screen. Enter the Primary server IP address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.



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



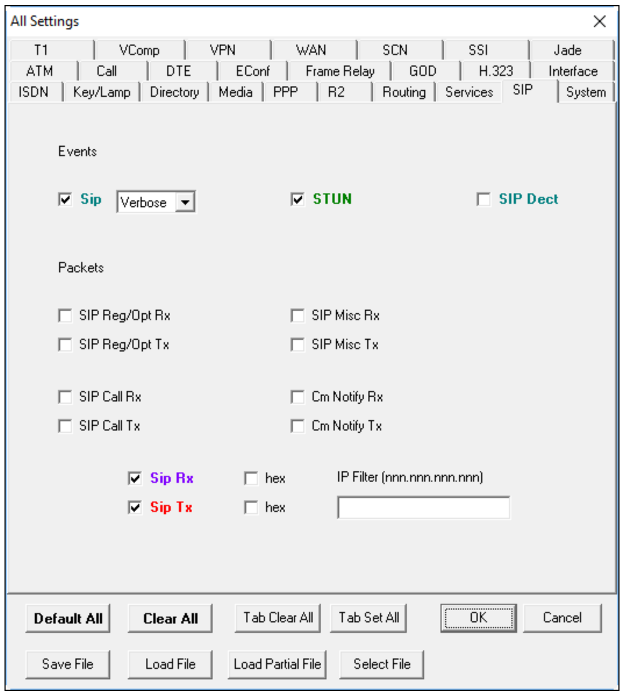
Select the **Alarms** tab and verify that no alarms are active on the SIP line.



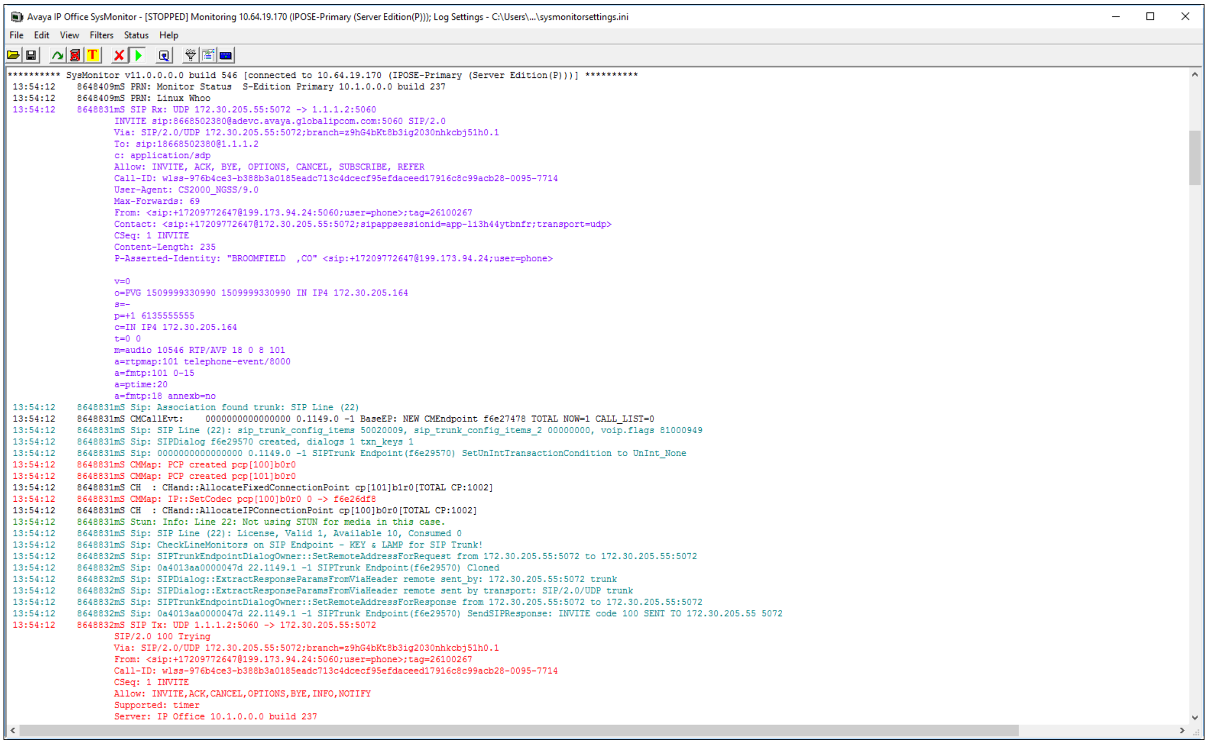
8.2. Monitor

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** (not shown).

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.



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



9. Conclusion

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

These Application Notes demonstrated how IP Office Release 10.1 can be successfully combined with a Verizon Business IPCC service connection to enable a business to receive toll-free calls. Utilizing this solution, IP Office customers can leverage the operational efficiencies and cost savings associated with SIP trunking while gaining the advanced technical features provided through the marriage of best of breed technologies from Avaya and Verizon

10. Additional References

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

[1] *Deploying IP Office™ Platform Server Edition Solution*, Release 10.1, June 2017

[2] *Administering Avaya IP Office™ Platform with Manager*, Release 10.1, June 2017

[3] *Administering Avaya IP Office™ Platform with Web Manager*, Release 10.1, June 2017

[4] *IP Office™ Platform 10.1*, *Deploying Avaya IP Office™ Platform IP500 V2*, September 2017

[5] *IP Office™ Platform 10.1, Using Avaya IP Office™ System Status*, July 2017

[6] *IP Office™ Platform 10.1, 1608/1616 Phone User Guide*, Document Number 15-601040 Issue 10d, July 2017

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

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

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 10.0 with Verizon Business IPCC Service Suite.
[VZBIPCC-IPO10] Application Notes for Configuring SIP Trunking using Verizon Business IP Contact Center VoIP Inbound and Avaya IP Office Release 10.0, Issue 1.0

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 10.1 with Verizon IP Trunk Service Suite.
[VZBIPT-IPO10.1] Application Notes for SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service and Avaya IP Office Release 10.1, Issue 1.0

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

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

- [VZ-Test-Plan] Test Suite for CPE IP Trunking Interoperability v1.6
- [VZ-Spec] Verizon Business IPCC Trunk Interface Network Interface Specification, Document Version 2.2.1.9

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