Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking Using Verizon Business IP Contact Center VoIP Inbound and Avaya IP Office Release 9.1 – Issue 1.0

Abstract

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Contact Center VoIP Inbound SIP Trunk Service and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500 V2 Release 9.1 Preferred Edition, Avaya Voicemail Pro, Avaya Communicator for Windows, and Avaya H.323, SIP, digital, and analog endpoints.

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.

IP Office Release 9.1 has not been independently certified by Verizon labs. These Application Notes can be used to facilitate customer engagements via the Verizon field trial process, pending Verizon labs independent certification.

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

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1. Introduction

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Contact Center (Verizon Business IPCC) VoIP Inbound Service and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500 V2 Release 9.1 Preferred Edition, Avaya Voicemail Pro, Avaya Communicator for Windows, and Avaya H.323, SIP, digital, and analog endpoints.

IP Office Release 9.1 has not been independently certified by Verizon labs. These Application Notes can be used to facilitate customer engagements via the Verizon field trial process, pending Verizon labs independent certification.

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 IP Contact Center services delivered via IDA service terminations.

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

2. General Test Approach and Test Results

The Avaya IP Office location was connected to the Verizon Business IPCC service, as depicted in **Figure 1**. Avaya IP Office was configured to use the commercially available IP Toll Free VoIP Inbound solution. This allowed Avaya IP Office to receive inbound toll-free calls from the PSTN via the SIP protocol.

2.1. Interoperability Compliance Testing

The testing included executing the test cases detailed in 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 IPCC and IP Office can both monitor health using SIP OPTIONS.
- Proper recovery from induced failure conditions such as IP Office reboots, and long and short duration IP network outages between Verizon and IP Office
- Incoming calls from the PSTN were routed to the toll-free numbers assigned by Verizon Business to the Avaya IP Office location. These incoming calls arrived via the SIP Line configured in **Section 5.4** and were answered by Avaya H.323 telephones, Avaya SIP

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telephones, Avaya digital telephones, analog telephones, Avaya Communicator for Windows, and Avaya IP Office Voicemail Pro.

- Proper disconnect when either party hangs up an active call
- Proper disconnect when the PSTN caller abandons (i.e., hangs up) a toll-free call before the IP Office party has answered.
- Proper SIP 486 response and busy tone heard by the caller when a PSTN user calls a toll-free number directed to a busy IP Office user, an IP Office user with Do-not-disturb active, or an IP Office user that is logged out (i.e., assuming no redirection is configured for these conditions). Similarly, busy tone is heard when a PSTN user calls a toll-free number whose "SIP URI Max Calls per Channel" has been reached (see **Section 5.4**). Similarly, busy tone is heard when a PSTN user calls a toll-free number directed to a hunt group whose queue is "full" (i.e. if no redirection is configured for hunt group busy conditions, see **Section 5.5.3**).
- Proper termination of an inbound IP Toll Free call left in a ringing state for a relatively long duration
- The display of caller ID on display-equipped Avaya IP Office telephones was verified. The IP Office capability to use the caller ID received from Verizon to look up and display a name from a configurable directory was also exercised successfully.
- Privacy requests for inbound toll-free calls from the PSTN were verified. That is, when privacy is requested by a PSTN caller (e.g., dialing *67 from a mobile phone), the inbound toll-free call can be successfully completed to an IP Office telephone user while presenting a "WITHHELD" or anonymous display to an IP Office user (i.e., rather than the caller's telephone number).
- Inbound toll-free long holding time call stability
- IP Office complies with RFC 3261 SIP Methods
- IP Office can use UDP for SIP transport with Verizon IPCC
- IP Office can use a configured UDP port for SIP signaling with Verizon IPCC
- IP Office accepts the full SIP headers sent by Verizon IPCC
- IP Office sends SIP 180 RINGING (no SDP in 180) for inbound calls and ring back tone is heard by the caller.
- IP Office does not return a SIP 302 to Verizon IPCC
- 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.729A and G.711 ULAW codecs, and proper protocol procedures related to media
- DTMF transmission using RFC 2833. Successful IP Office Voicemail Pro menu navigation for incoming toll-free calls. Successful use of IP Office Mobile Call Control, where DTMF sequences can be performed remotely using the SIP Line.
- Incoming toll-free calls directed to the Hunt Groups configured in **Section 5.5.3** were verified. Incoming calls could be queued, queued with priority, and be answered by members of the hunt group as members become available.
- 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-IPO91]. As detailed in reference [VZBIPT-IPO91], these outgoing PSTN calls can be originated from

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Avaya H.323 telephones, Avaya SIP telephones, Avaya digital telephones, analog endpoints, and Avaya Communicator for Windows. The display of caller ID on display-equipped PSTN telephones was verified. In the context of inbound toll-free calls using Verizon Business IPCC, inbound toll-free calls arriving via the SIP Line configured in **Section 5.4** could be forwarded or twinned out the Verizon Business IP Trunk service SIP Line. Inbound toll-free calls from the Verizon Business IPCC SIP Line could also trigger mobile callback calls that use the Verizon Business IP Trunk service SIP Line.

- Call Forwarding of Verizon toll-free calls to PSTN destinations via the Verizon Business IP Trunk service documented in reference [VZBIPT-IPO91], 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-IPO91], presenting true calling party information to the mobile phone.
- Inbound mobile call control, mapping a Verizon toll-free number to the mobile call control feature, as shown in **Section 5.7.4**. That is, a configured mobile twinning PSTN caller may dial a Verizon toll-free number, receive dial tone from IP Office, and place calls using IP Office, as if the user were calling from their IP Office telephone. Calls to the same toll-free number from calling numbers that are not configured in IP Office for mobile call control receive busy tone.
- DiffServ markings for Avaya IP Office SIP signaling and RTP media consistent with network capability for optimum routing of VoIP.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results as described in **Section 2.1**. The following observations may be noteworthy:

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

2. During testing, blind transfers were made from an inbound toll-free call from Verizon Business IPCC service using the SIP REFER method back out the Verizon Business IPCC service. After receiving the REFER, Verizon would send a “202 Accepted”, followed by a NOTIFY with “100 Trying”, then another NOTIFY with “200 OK”, once the call was answered. Typically an active response to a referenced call would be to receive in a SIP NOTIFY message with “180 Ringing”, “183 Session Progress” or “200 OK”. If IP Office does not receive one of the aforementioned active responses within ten seconds, it considers the REFER a failure, and will attempt to send the transfer using the SIP INVITE method. Verizon does not send an active response to the referenced call until it is answered. IP Office is expecting a response while the referenced call is ringing. Therefore, if it cannot be guaranteed that the far end will answer the call quickly, for example transferring to an IVR, then the blind transfer using the SIP REFER method is not recommended. To ensure the SIP REFER method is not used, disable the **Outgoing Blind REFER** parameter on the SIP Line as shown in **Section 5.4.3**.

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2.3. Support

2.3.1. Avaya

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

2.3.2. Verizon

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

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3. Reference Configuration

Figure 1 illustrates an example Avaya IP Office solution connected to the Verizon Business IPCC SIP Trunk service. The Avaya equipment is located on a private IP subnet. An enterprise edge router provides access to the Verizon Business network via a Verizon Business T1 circuit. This circuit is provisioned for the Verizon Business Private IP (PIP) service. Reference [VZBIPT-IPO91] illustrates IP Office interoperability with the Verizon Business IP Trunk service. In the verification testing associated with these Application Notes, both the Verizon IP Trunk service and the Verizon Business IPCC service were accessible via the same PIP connection.

In the sample configuration, 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 IP Contact Center service. These toll-free numbers were mapped to IP Office destinations via Incoming Call Routes as summarized in **Table 1**. The Avaya IP Office environment domain known to Verizon was *adevc.avaya.globalipcom.com*.

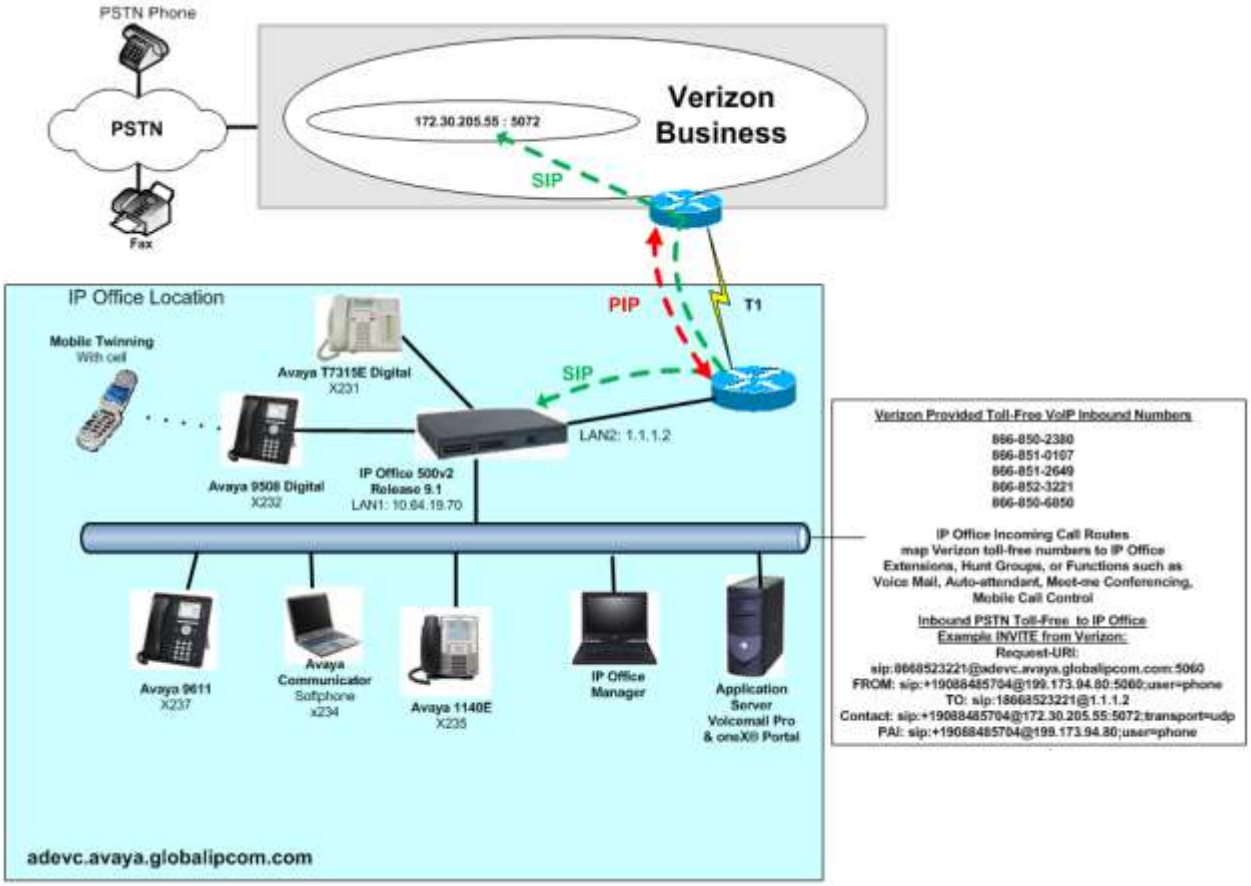

Figure 1: Avaya IP Office with Verizon IP Contact Center Service

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

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

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4. Equipment and Software Validated

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

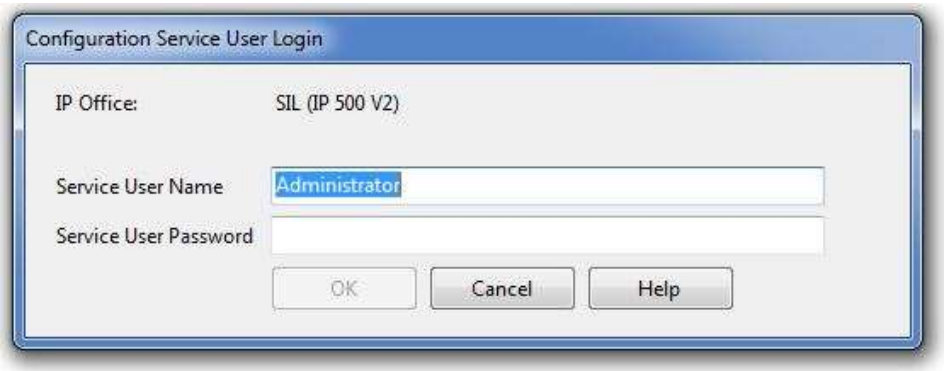
Equipment	Software
Avaya IP Office 500 V2	Release 9.1.100.10
Avaya IP Office Manager	Release 9.1.1.0 Build 10
Avaya Application Server	Release 9.1.1.0.10
Avaya 2500 Analog Telephone	N/A
Avaya 9508 Digital Telephone	N/A
Avaya T7315E Digital Telephone	N/A
Avaya 9611 IP Telephone (H.323)	Release 6.2209
Avaya 1140E SIP	Release 04.04.18
Avaya Communicator for Windows	Release 2.0.3.30

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 without T.38 Fax service (T.38 fax is not supported on 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 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. A screen that includes the following in the center may be displayed:



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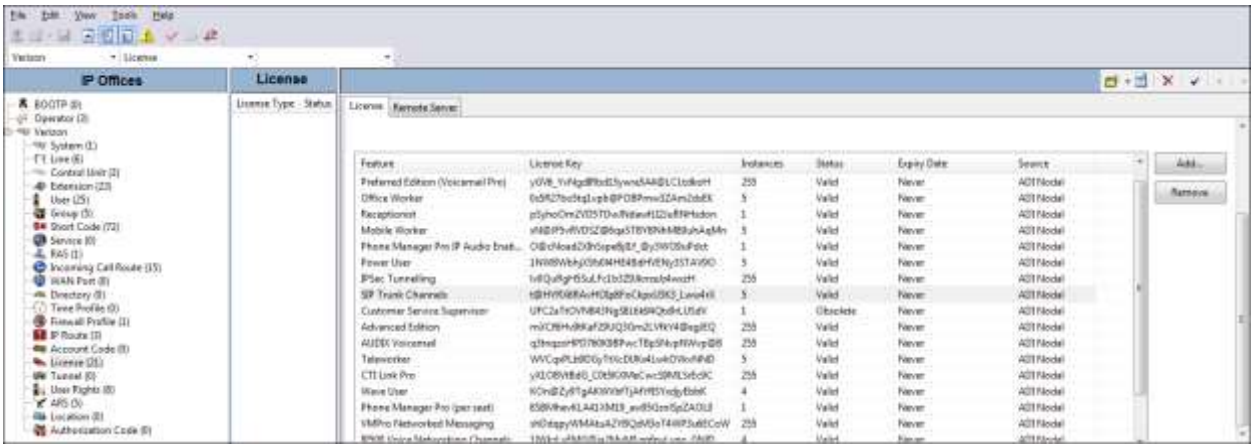
Log in with the appropriate configuration credentials. The appearance of the IP Office Manager can be customized using the **View** menu (not shown). In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details pane on the right side.

5.1. Licensing and Physical Hardware

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

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



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

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

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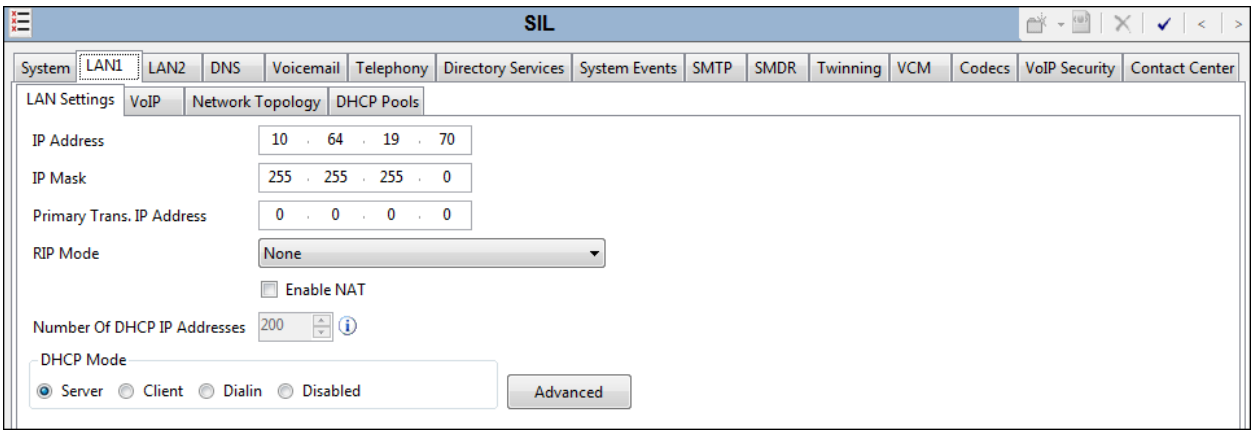
5.2. System Settings

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.

5.2.1. LAN 1 Settings

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

In the sample configuration, LAN1 is used to connect the IP Office to the enterprise network. To view or configure the **IP Address** of LAN1, select the **LAN1** tab followed by the **LAN Settings** tab. As shown in **Figure 1**, the IP Address of the IP Office is 10.64.19.70. **DHCP Mode** is also set to **Server** so that IP phones will get an IP Address from the IP Office Server. Other parameters on this screen may be set according to customer requirements.



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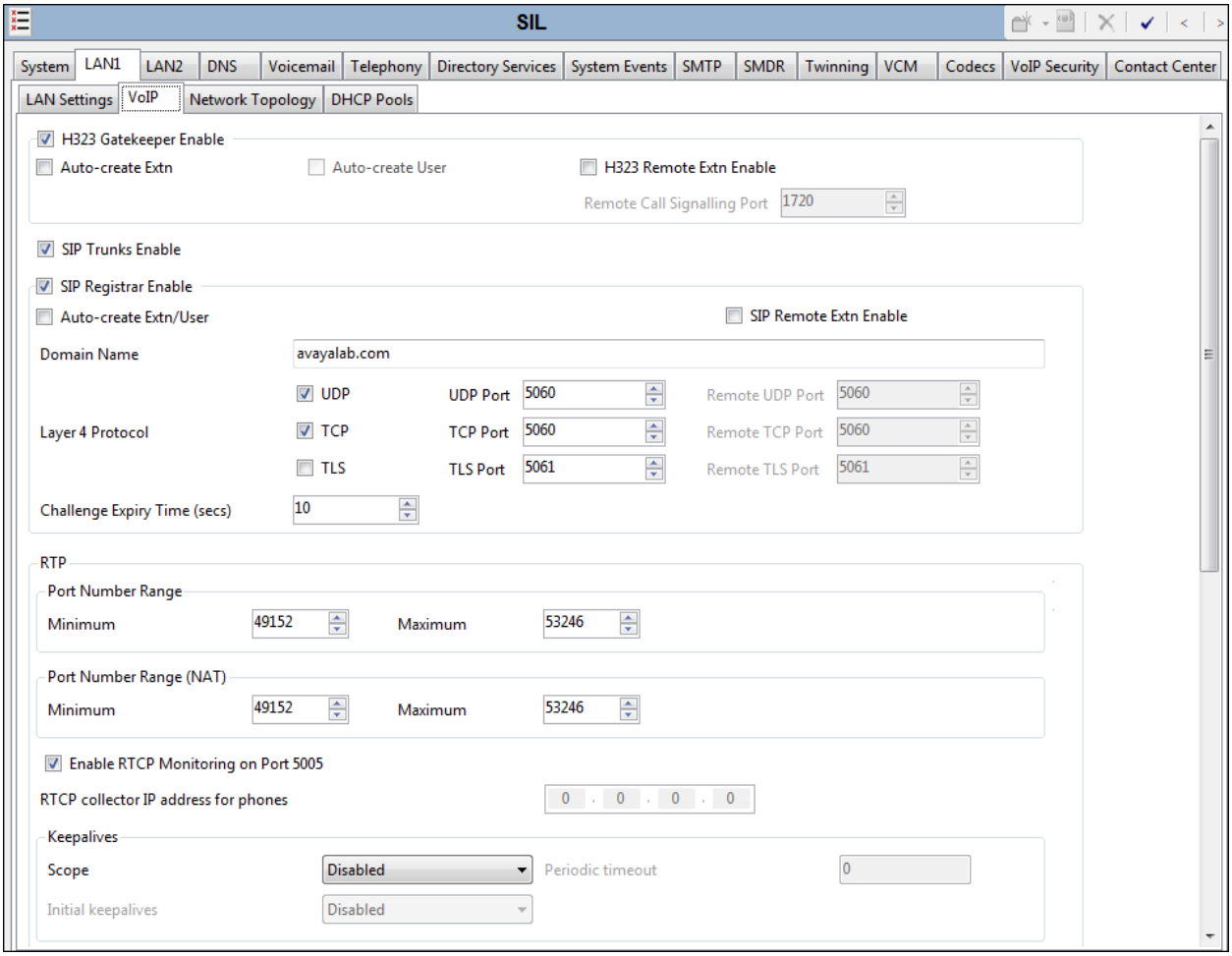
Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 9611 Telephone used in the sample configuration. The **SIP Registrar Enable** box is checked to allow Avaya 1140E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



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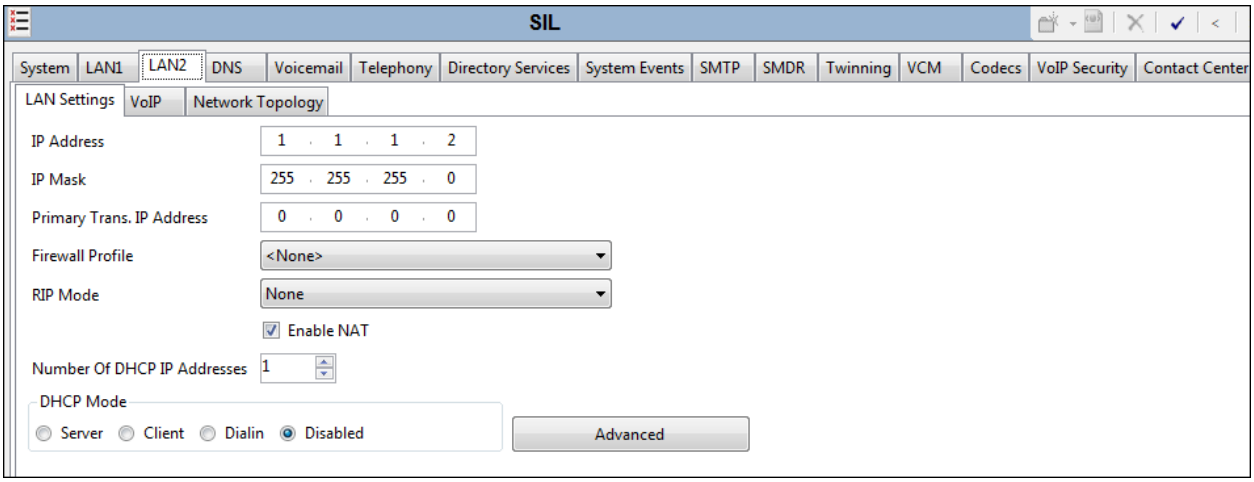
5.2.2. LAN 2 Settings

In the sample configuration, LAN2 is used to connect the IP Office to the Verizon 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.



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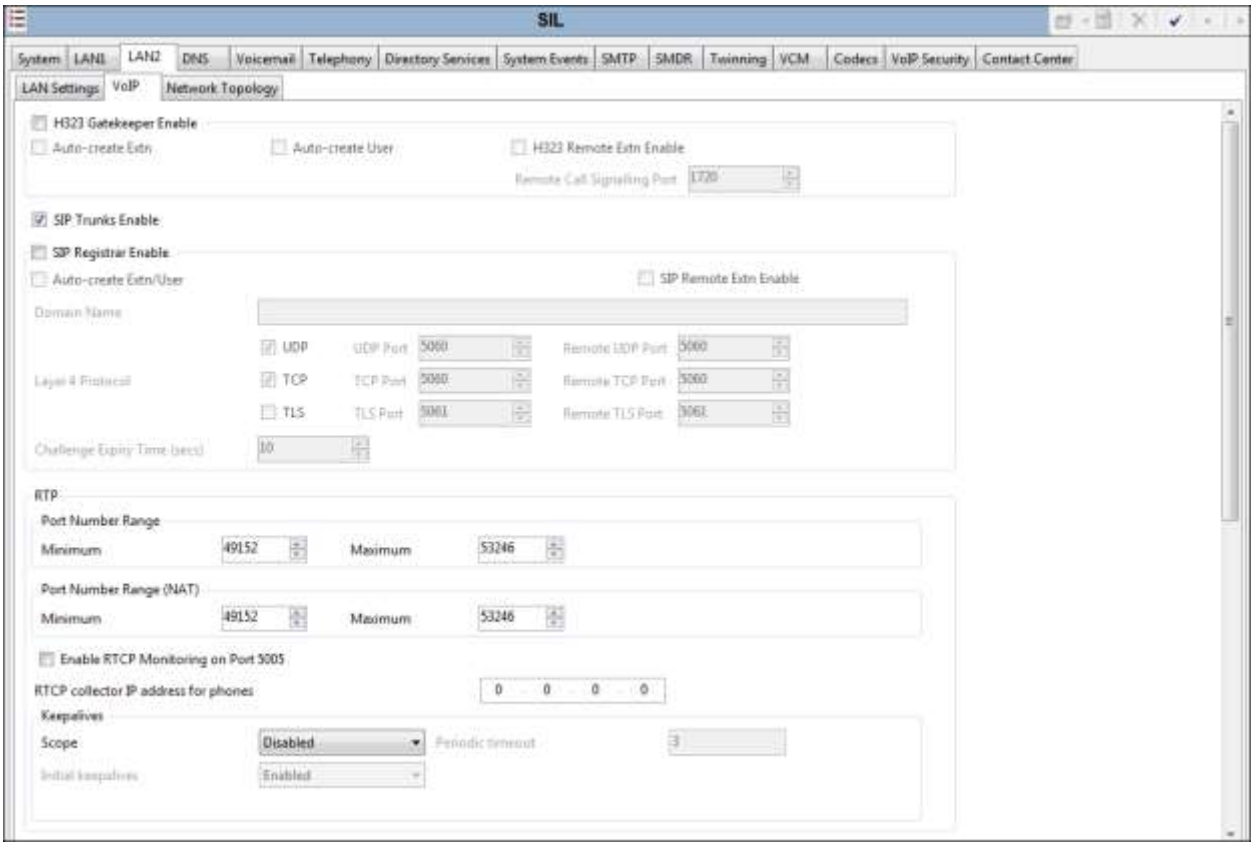
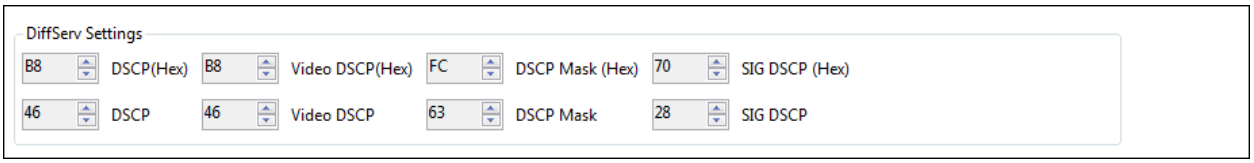
Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** and **SIP Registrar Enable** boxes are unchecked since IP telephones will not be registering on this link. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Verizon Business.

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). 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.



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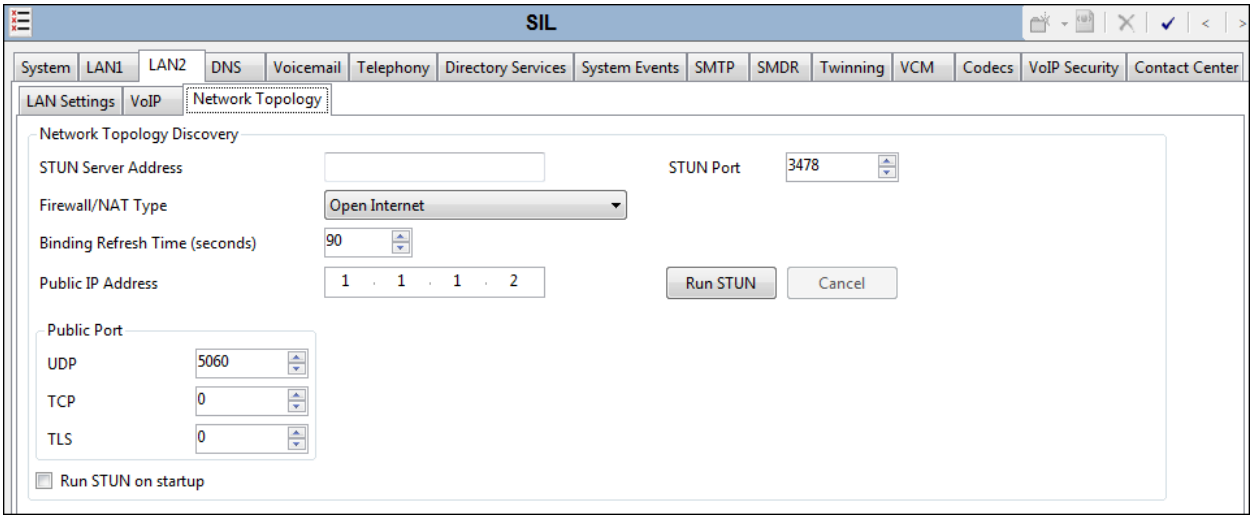
Select the **Network Topology** tab as shown in the following screen. The **Firewall/NAT Type** is set to **Open Internet** in the sample configuration. Note that the **Firewall/NAT Type** parameter may need to be set differently, depending on the type of firewall or Network Address Translation device used at the customer premise. The **Binding Refresh Time (seconds)** can be used to lower the SIP OPTIONS timing from the default of 300 seconds. During the testing, the Binding Refresh Time was varied (e.g., 90 seconds, 120 seconds) to test SIP OPTIONS timing. 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".



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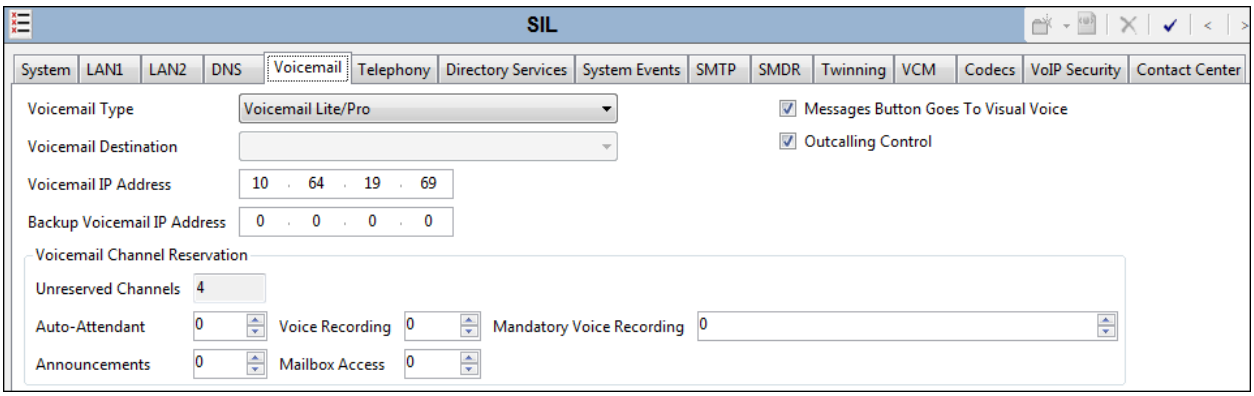
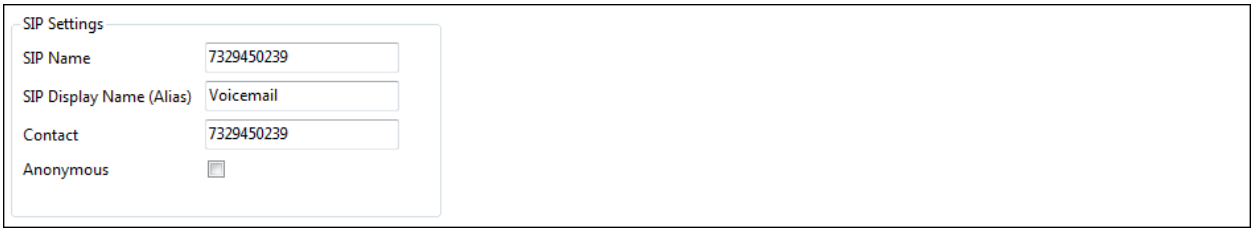
5.2.3. Voicemail

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". Other Voicemail types may be used. The Voicemail IP Address in the sample configuration is 10.64.19.69, the IP address of the IP Office Application Server running the Voicemail Pro software, as shown in **Figure 1**.



As described in [VZBIPT-IPO91], 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.



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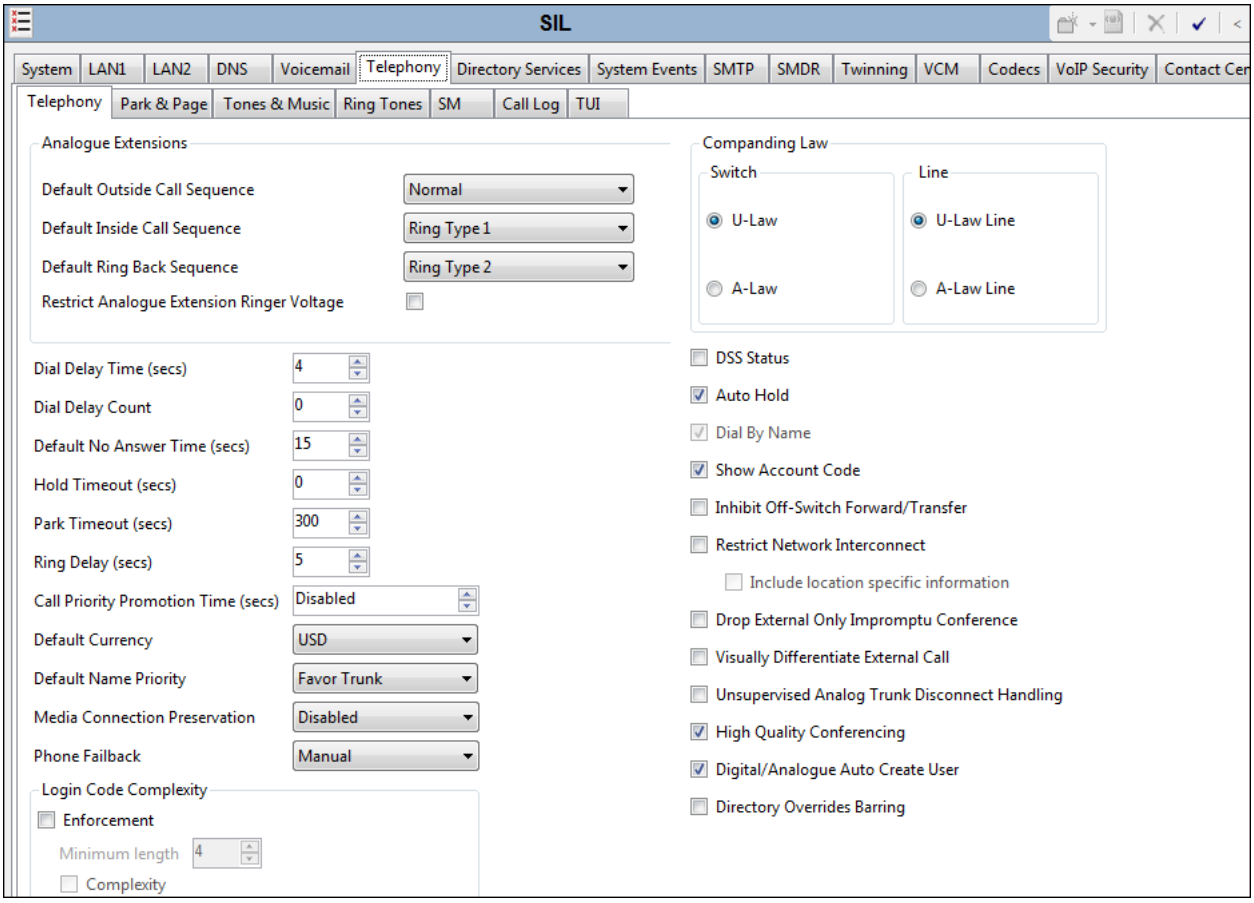
5.2.4. System Telephony Configuration

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 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 "ULAW" as is typical in North American locales. Other parameters on this screen may be set according to customer requirements.



5.2.5. System Twinning Configuration

To view or change Twinning settings, select the **Twinning** tab as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning** is left blank. With this configuration, and related configuration of "Diversion header" on the SIP Line to Verizon Business IP Trunk service (Section 5.4.1 of reference [VZBIPT-IPO91]), the true identity of the caller can be presented to the twinning destination (e.g., a user's mobile phone)

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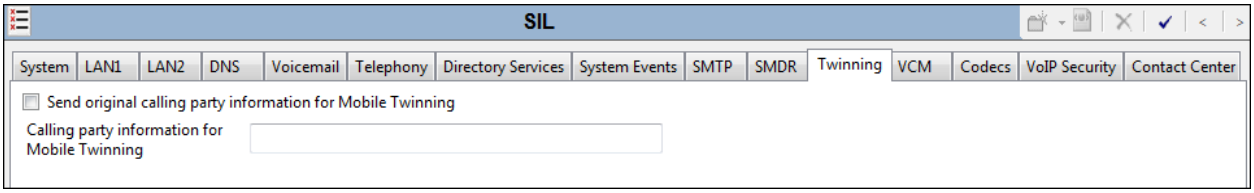
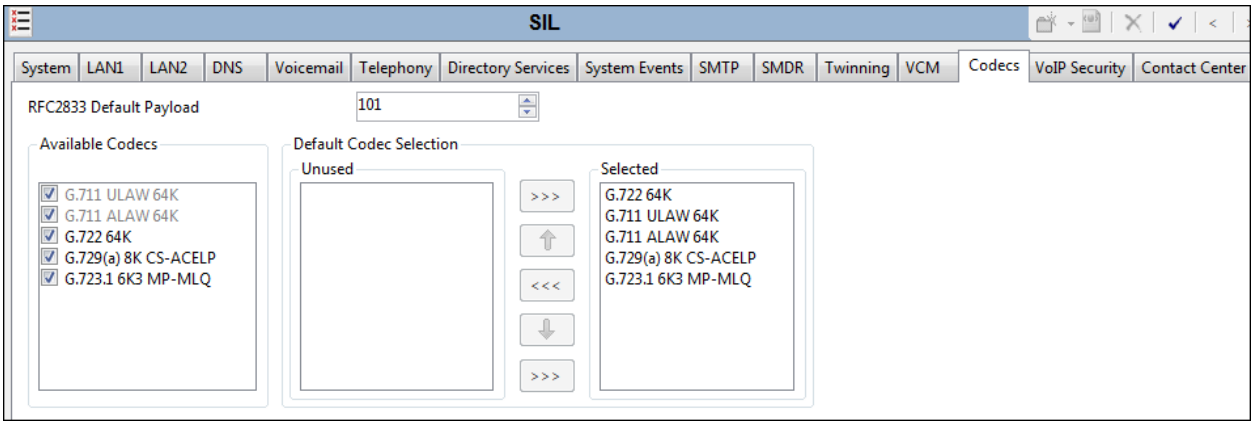
when a call is twinned out via the Verizon Business IP Trunk service. That is, a call can arrive via a Verizon Business IPCC service toll-free number, and be twinned out to a mobile telephone using the Verizon Business IP Trunk service, with the twinned mobile phone seeing the identity of the caller that dialed the Verizon toll-free number.



5.2.6. System Codecs Configuration

To view or change system codec settings, select the **Codecs** tab. On the left, observe the list of **Available Codecs**. In the example screen below, which is not intended to be prescriptive, the box next to each codec is checked, making all the codecs available in other screens where codec configuration may be performed (such as the SIP Line in **Section 5.4.6**). 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. Set the **RFC2833 Default Payload** parameter to "101", the value preferred by Verizon Business.



5.3. IP Route

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

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".

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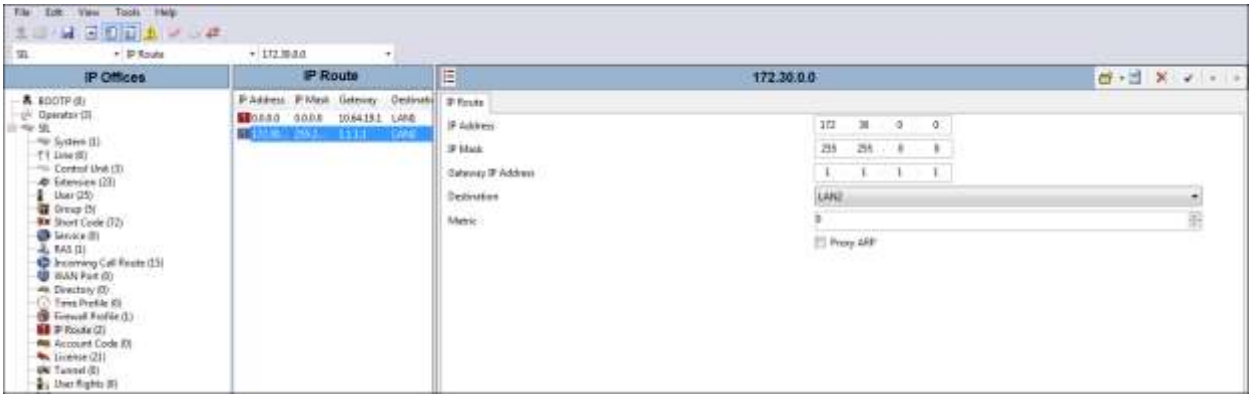


5.4. SIP Line

This section shows the configuration screens for the SIP Line in IP Office Release 9.1. Since IP Office Release 9.1 introduced new SIP Line parameters and re-oriented existing parameters, this section has the most substantive changes in these Application Notes, compared to the configuration documented in references [VZB-IPCCIPOR9].

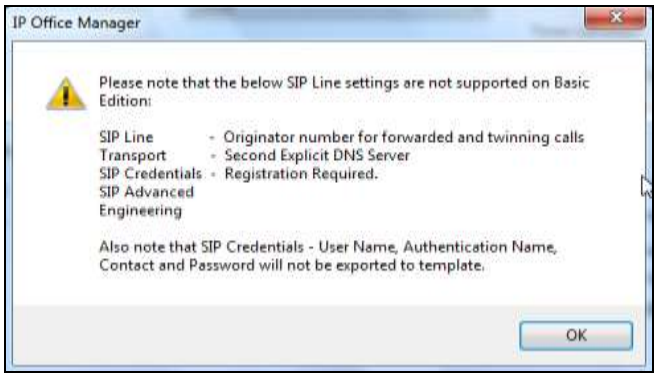
The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.4.2** to create the SIP Line from the template.

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 5.4.2– 5.4.8**.

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



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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 – 5.4.8**.

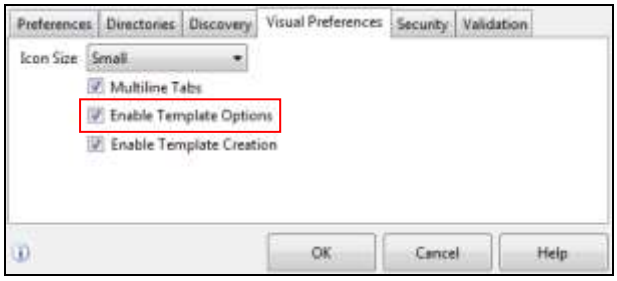
5.4.1. Importing a SIP Line Template

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 Preferred Edition (IP500 V2) 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. By default, the template file name will have the format **AF_<*user supplied text*>_SIPTrunk.xml**, where the **<*user supplied text*>** portion is entered during template file creation.

   Note – If necessary, the **<*user supplied text*>** portion of the template file name may be modified, however the **AF_<*user supplied text*>_SIPTrunk.xml** format of the file name must be maintained. For example, an original template file **AF_*TEST*_SIPTrunk.xml** could be changed to **AF_*Test1*_SIPTrunk.xml**. The template file name is selected in **Section 5.4.2** to create a new SIP Line.

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



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

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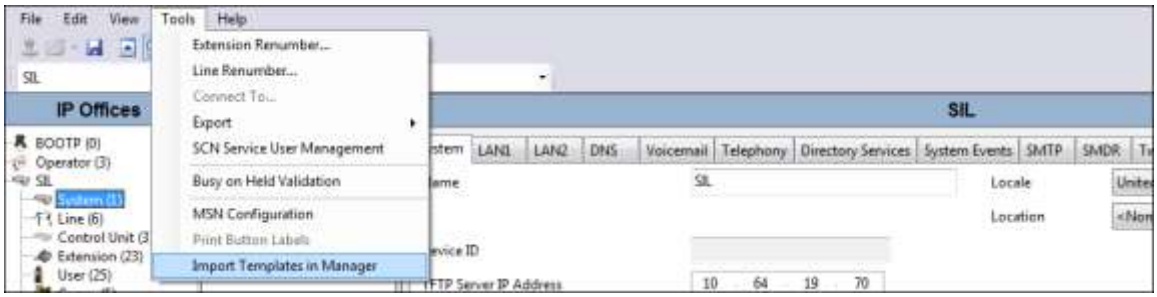
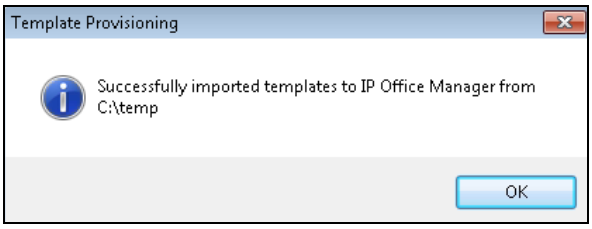


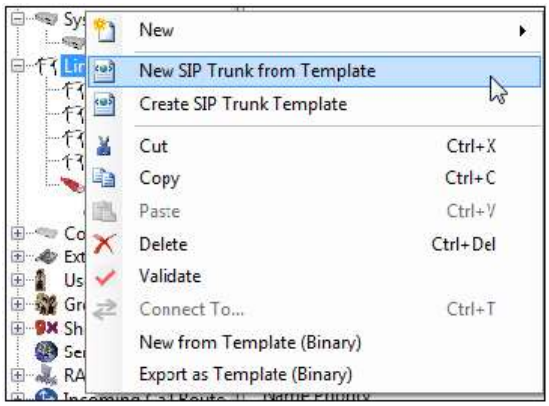
4. A folder browser will open (not shown). Select the directory used in **step 1** to store the template(s) (e.g., *\temp*). In the reference configuration, template file **AF_Verizon IPCC_ SIPTrunk.xml** was imported. The template files are automatically copied into the IP Office default template location,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

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



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 select **New SIP Trunk from Template**.



2. In the subsequent **Template Type Selection** pop-up window, from the **Service Provider** pull-down menu, select the XML template name from **Section 5.4.1**. Click **Create new SIP Trunk**.

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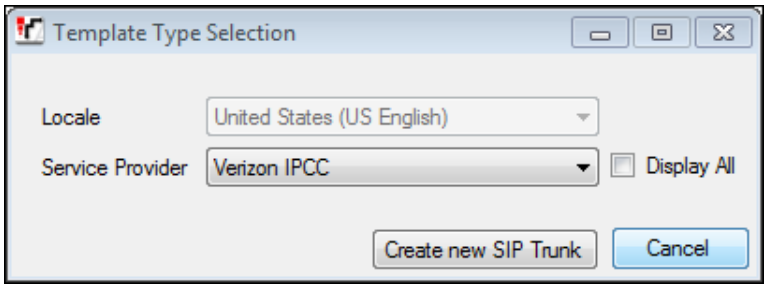
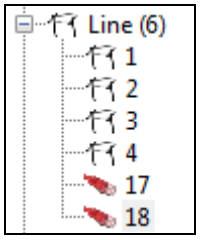
Note – The drop down menu will display the ***<user supplied text>*** part of the template file name (see **Section 5.4.1**). If the **Display All** box is checked, then the full template file name is displayed.



The newly created SIP Line will appear in the Navigation pane (e.g., SIP Line **18**).



Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.4.3 – 5.4.8**.

5.4.3. SIP Line - SIP Line Tab

The **SIP Line** tab in the Details pane is shown below for Line Number 21, used for the Verizon Business IPCC service. The **ITSP Domain Name** is configured to the domain supplied by Verizon (advec.avaya.globalipcom.com). 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 +13035387006 would display 3035387006. If the **Country Code** does not match the value following the "+" from Verizon, the IP Office user display would show the contents of the **International Prefix** field, followed by the value following the "+", followed by the PSTN number. For example, if the Country Code parameter were left blank, the IP Office user would see a display such as "01113035387006". Aside from display implications, if the **Country Code** is not configured, other patterns may also fail to match as expected, such as a match on the **Incoming CLI** field of the Incoming Call Route. See **Section 5.7.3** for configuration of incoming call routing based on the calling number.

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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. Verify the **Send 302 Moved Temporarily** setting is unchecked, as Verizon does not support receiving a 302 Moved Temporarily message.

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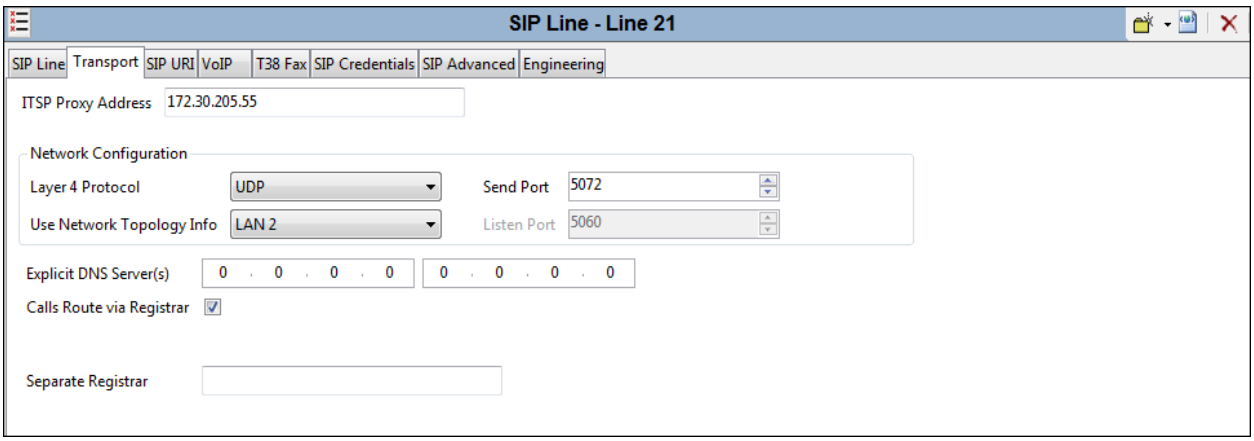
5.4.4. SIP Line - Transport Tab

Select the **Transport** tab. The **ITSP Proxy Address** is set to the IP Address provided by Verizon Business. As shown in **Figure 1**, this IP Address is 172.30.205.55. In the **Network Configuration** area, UDP is selected as the **Layer 4 Protocol**, and 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 "LAN 2", the LAN port connected to the Verizon PIP network.



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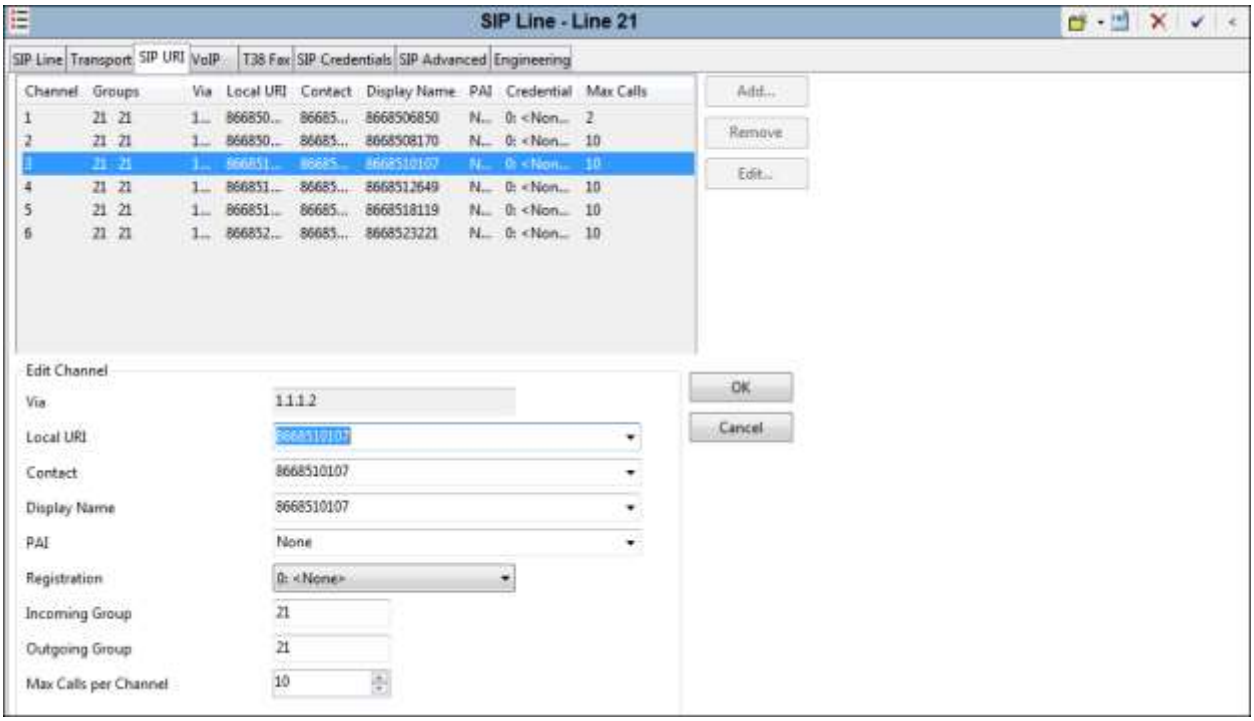
5.4.5. SIP Line - SIP URI Tab

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 Channel area will be opened.

In the sample configuration, each of the Verizon-provided toll free numbers are entered as a SIP URI, with the specific number entered in the **Local URI**, **Contact**, and **Display Name** fields. The **PAI** parameter is 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 21, will be referenced when configuring Incoming Call Routes to map inbound SIP trunk calls to IP Office destinations in **Section 5.7**. The **Outgoing Group** parameter, also set here to 21, is not relevant in that this SIP Line will not be chosen for outbound calls, since the Verizon Business IPCC service will only be used for inbound toll-free calls. Click **OK**.



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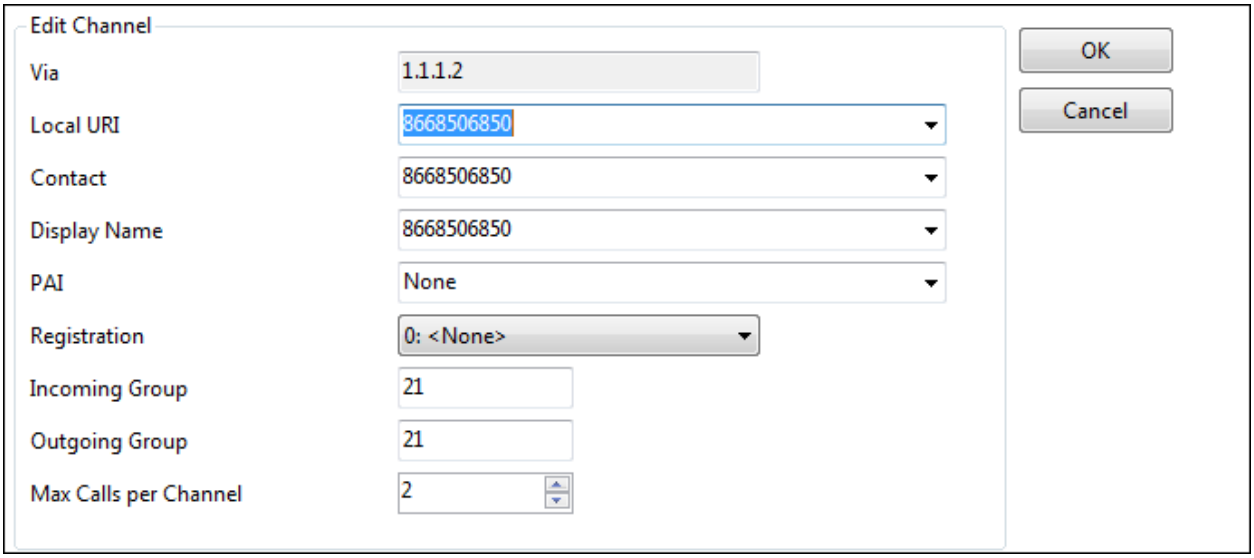
IP Office allows the number of simultaneous calls to a specific SIP URI to be managed using the **Max Calls per Channel** field. In the following screen, note that the **Max Calls per Channel** field has been changed from the default 10 to 2. With this configuration, two simultaneous calls to the number 866-850-6850 will be allowed. Once two calls are active, and a third call is attempted to 866-850-6850, IP Office will return a SIP 4xx response. Calls to other toll-free numbers using this same SIP Line are unaffected by the Max Calls per Channel for a different URI. Therefore, this approach could be used to control the maximum number of calls to each of the specific toll-free numbers, preventing a surge of calls to a given toll-free number from monopolizing the available call handling capacity of the access line or IP Office resources. An alternative means to restrict the number of simultaneous calls to a toll-free number that terminates on a hunt group would be to limit the queue size of the destination hunt group. If a non-priority call arrives to IP Office to a hunt group with a fixed size queue, and the queue is full, and there is no voice mail for the hunt group, IP Office returns a 486 Busy Here. See **Section 5.5.3** for hunt group configuration.



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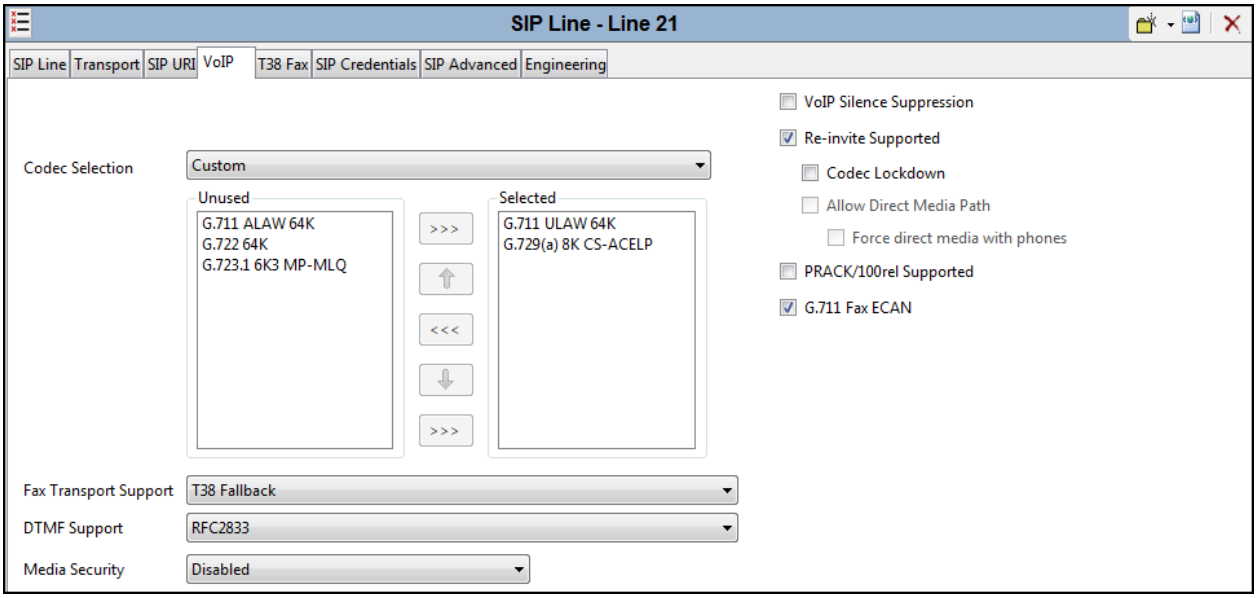
5.4.6. SIP Line - VoIP Tab

Select the **VoIP** tab. In the sample configuration, the **Codec Selection** is configured using the "Custom" option, allowing an explicit ordered list of codecs to be specified, different from the system default (see **Section 5.2.6**). The arrow buttons can be used such that **G.711 ULAW 64K** and **G.729(a) 8K CS-ACELP** codecs are listed in the **Selected** column. This configures IP Office to support either G.711MU or G.729a for this SIP Line. The **DTMF Support** parameter can remain set to the default value "RFC2833". The **Re-invite Supported** parameter can be checked to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk. 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.729a was the preferred or only codec listed, and G.729a calls were also successfully verified.

If the SIP Line will be used for fax, the **G.711 Fax ECAN** parameter should be checked and **G.711 ULAW 64K** set as the first codec preference. Also, 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. Click **OK** (not shown).



5.4.7. SIP Line – T38 Fax Tab

The settings on this tab are only accessible if **Re-invite Supported** is checked and a value for **Fax Transport Support** other than "None" are selected on the VoIP tab. Fax relay is only supported on IP500 V2 systems with an IP500 VCM card. To modify the values on this tab, uncheck the **Use Default Values** parameter at the bottom of the form. The **Disable T30 ECM** must be checked or fax errors may be experienced when using T38 Fax. When selected, it disables the T.30 Error Correction Mode used for fax transmission. The **T38 Fax Version** is set to "0". In the **Redundancy** area, **Low Speed** and **High Speed** are set to "2". All other values are

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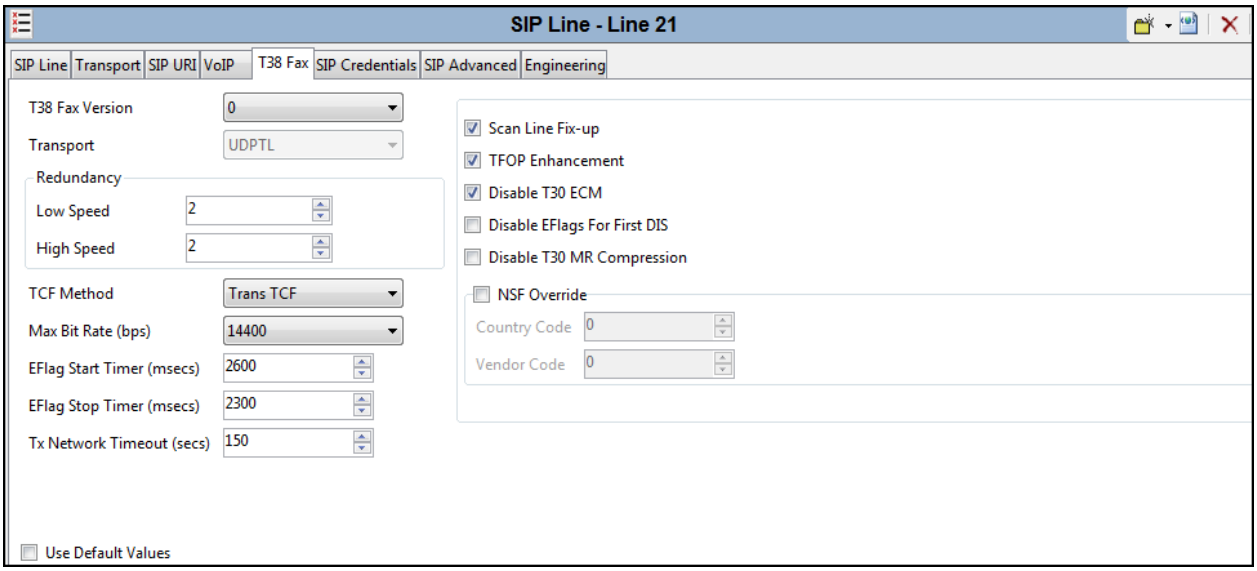
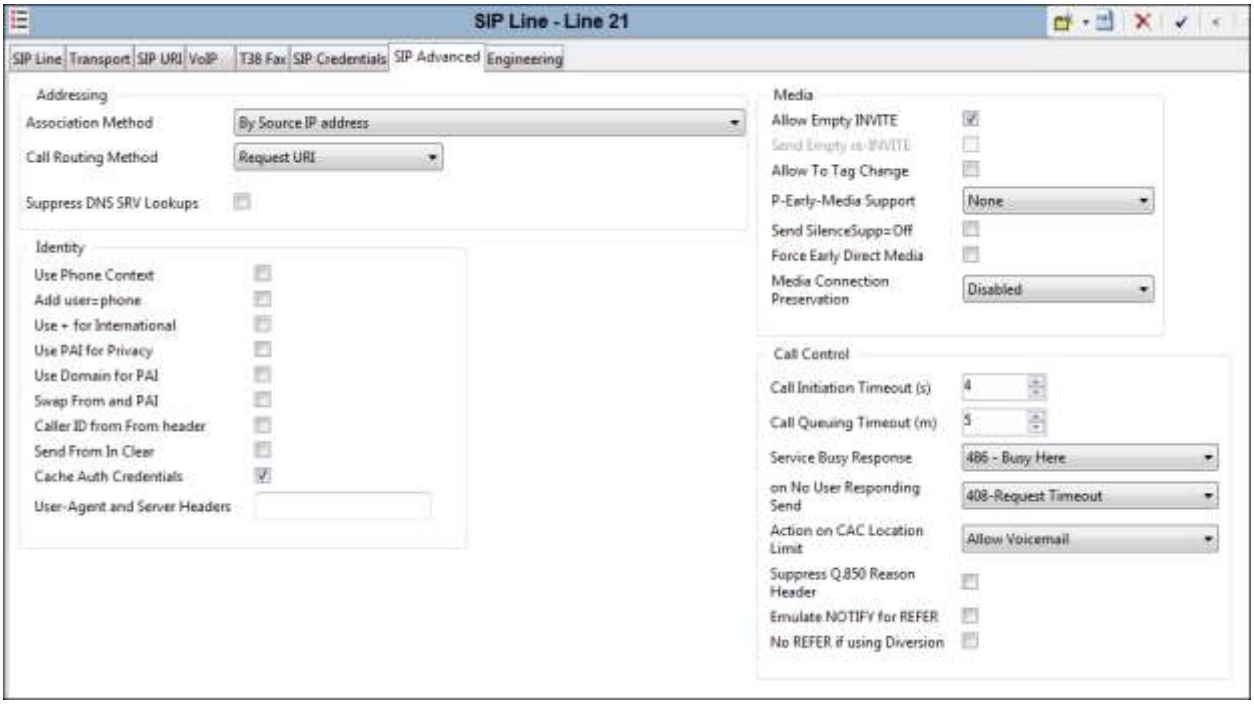
left at default. Since the Verizon Business IPCC service does not require registration, the **SIP Credentials** tab need not be visited. Click **OK** (not shown)



5.4.8. 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. Other parameters may be left at default values.



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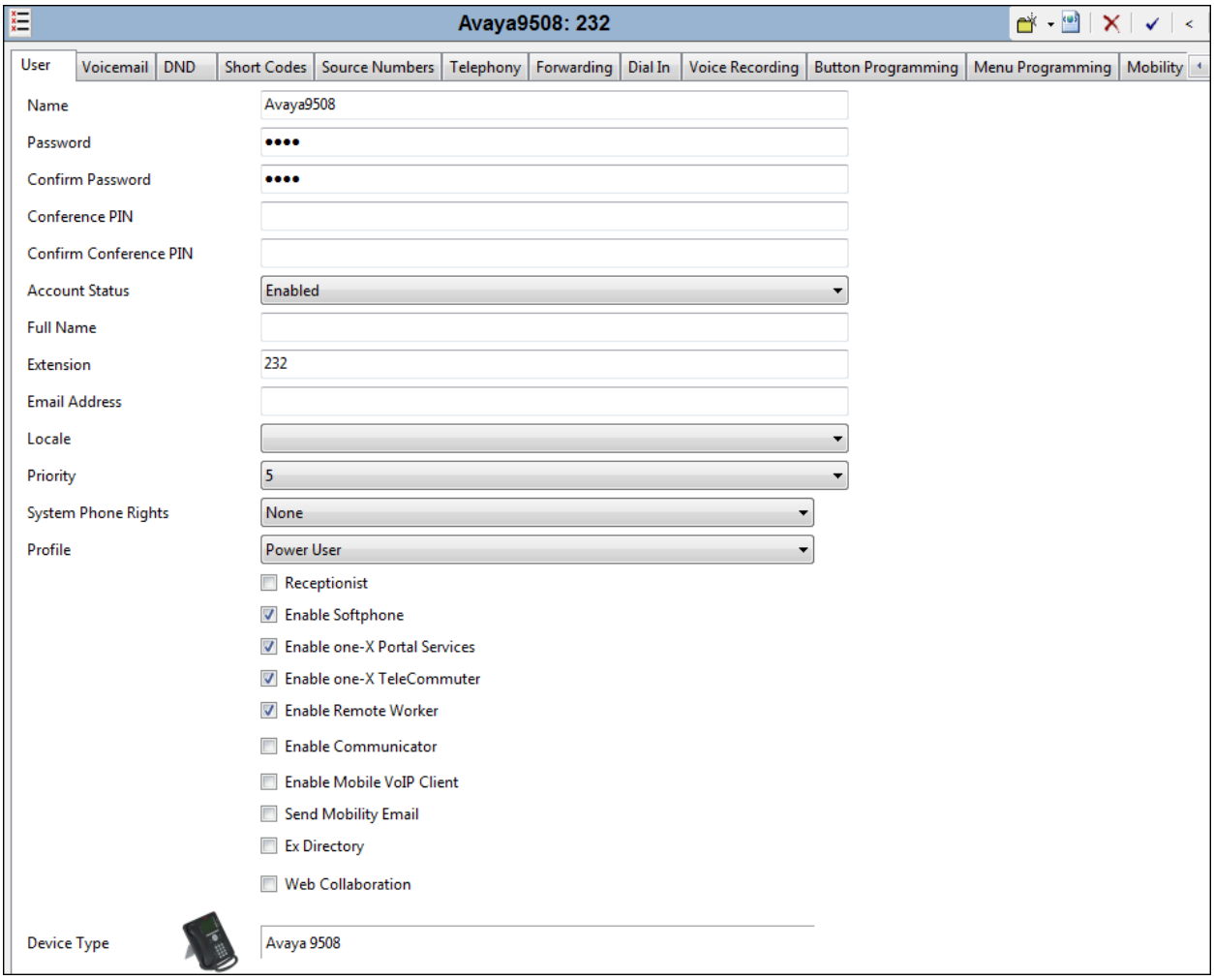
5.5. Users, Extensions, and Hunt Groups

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

5.5.1. User 232 (Digital)

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



The following screen shows the **SIP** tab for User 232. In the sample configuration, the **SIP Name** and **Contact** parameters are configured with a Verizon Business IP Trunk DID number for the user, 7329450232. As shown in [VZBIPT-IPO91], 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

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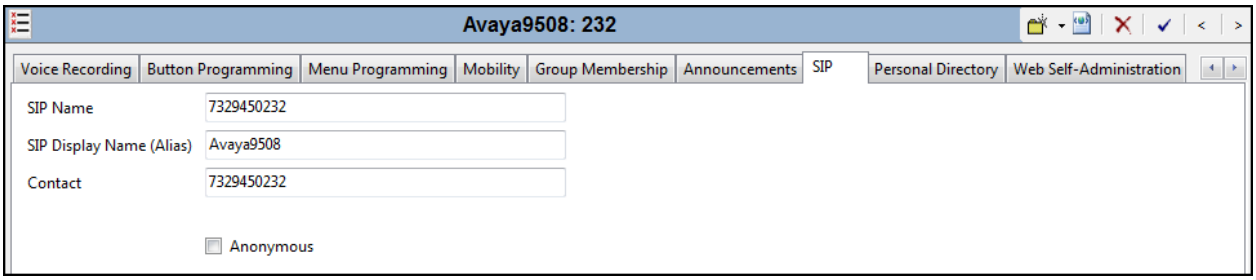
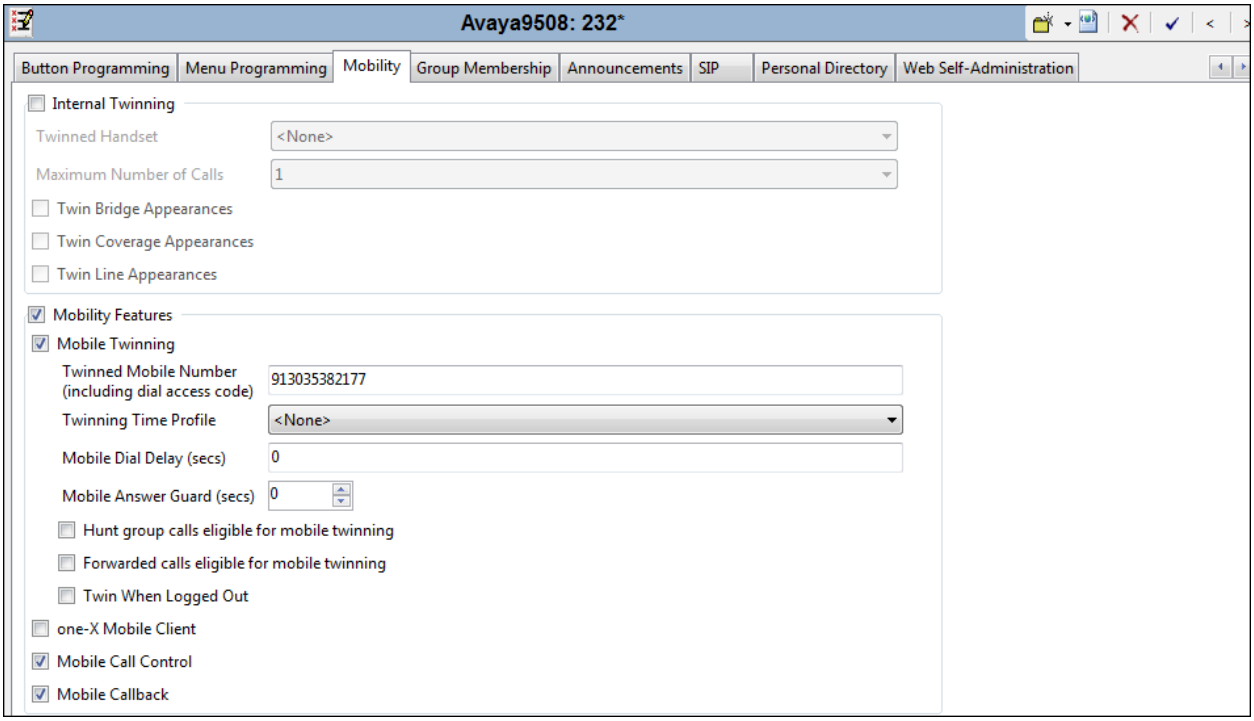
as an explicit SIP URI for the SIP Line. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network.



From **Figure 1**, note that user 232 will use the Mobile Twinning feature. The following screen shows the **Mobility** tab for User 232. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case 913035382177. Other options can be set according to customer requirements. In the sample configuration, the **Mobile Call Control** and **Mobile Callback** boxes were checked, and both mobile call control feature and mobile callback were tested using a Verizon-provided Toll Free number. In the case of mobile callback, a Verizon provided toll-free number was used to call in to IP Office and hang up. The mobile callback outbound leg used the Verizon Business IP Trunk service provisioned in [VZBIPT-IPO91].



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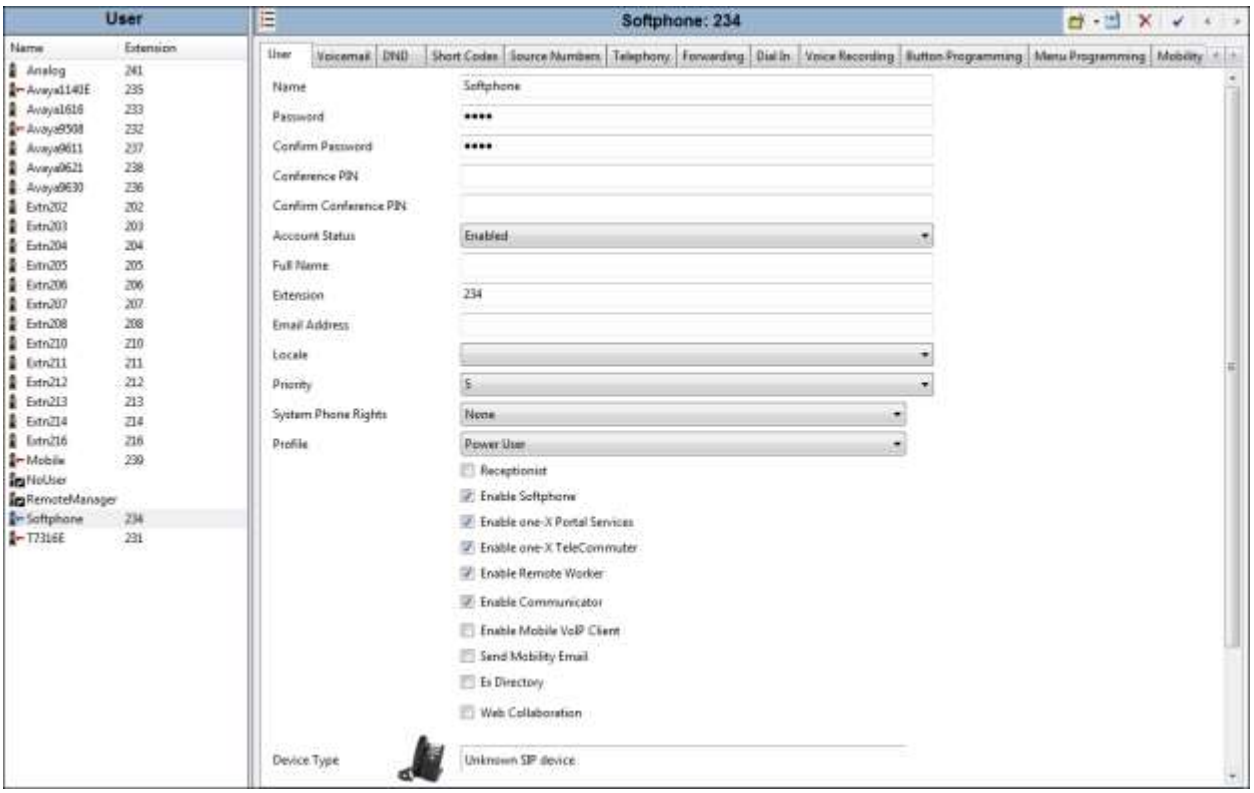
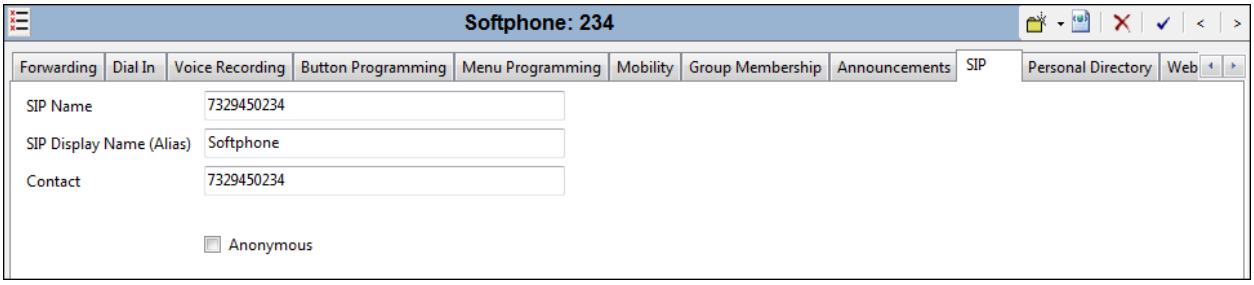
5.5.2. User 234 (Avaya Communicator for Windows)

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



Like the user with extension 232, the **SIP** tab for the user with extension 234 is configured with a **SIP Name** and **Contact** specifying the user's Verizon Business DID number using the Verizon Business IP Trunk service, as detailed in [VZBIPT-IPO91].



The following screen shows the Voicemail tab for the user with extension 234. The **Voicemail On** box is checked, and a voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters. In the verification of these Application Notes, incoming calls from the Verizon Business IPCC service to this user were redirected to Voicemail Pro after no answer. Voicemail messages were recorded and retrieved successfully. Voice mail navigation

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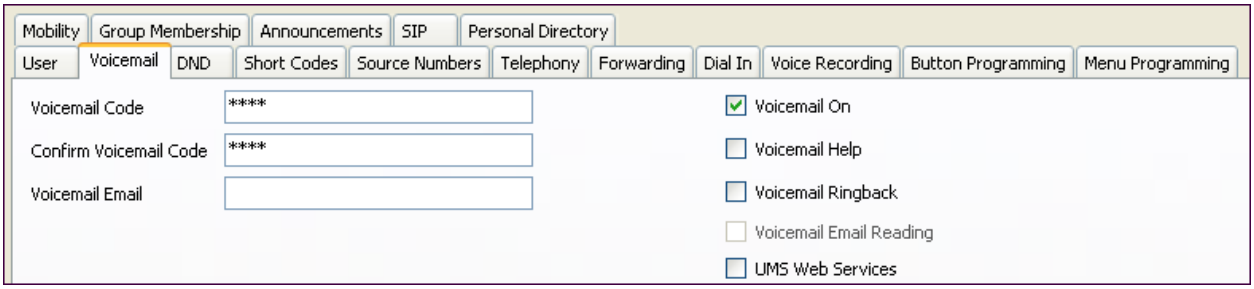
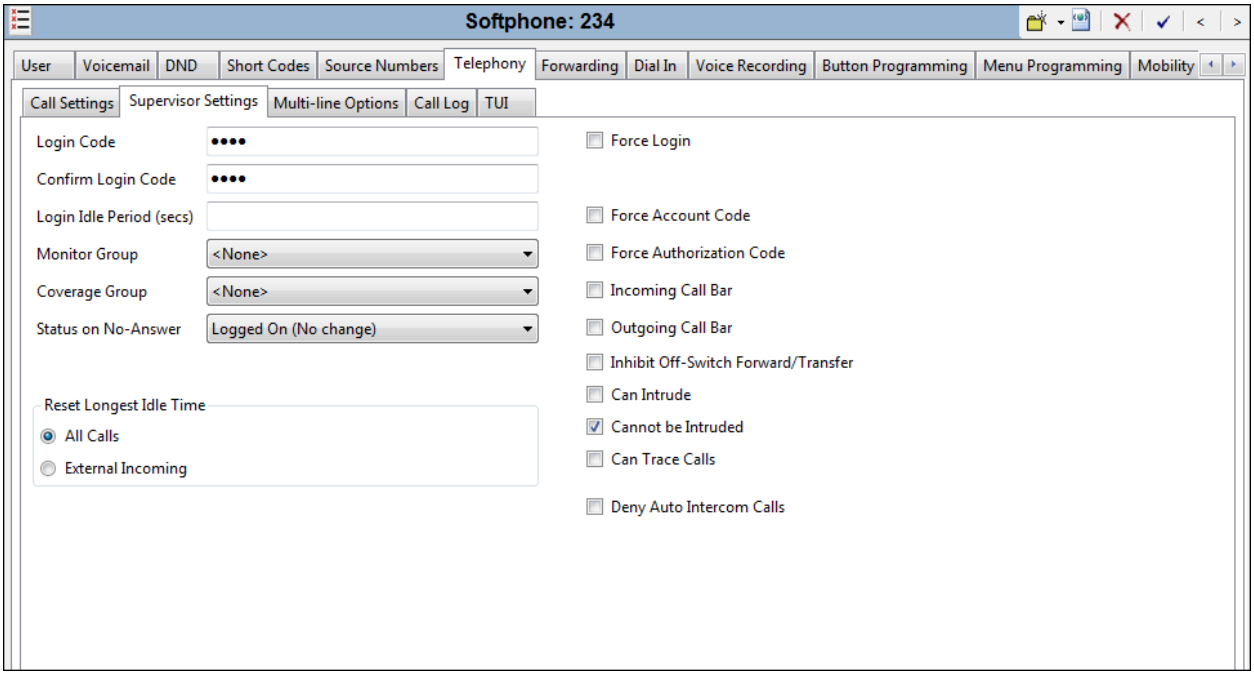
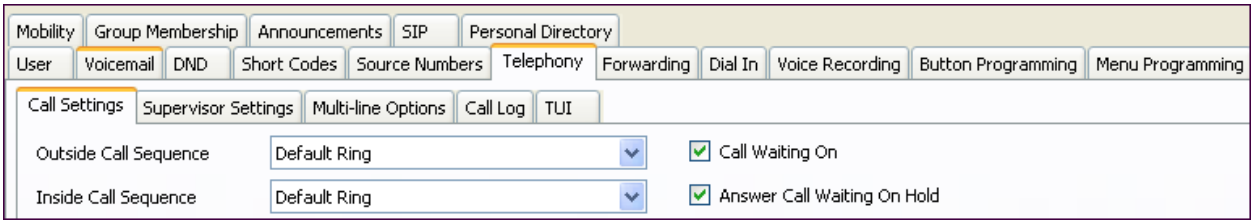
and retrieval were performed locally and from PSTN telephones using the Verizon Business IPCC toll-free numbers, to test DTMF using RFC 2833, and to test assignment of a Verizon-provided toll free number to the "Voicemail Collect" feature (e.g., via the *17 short code shown in **Section 5.6**).



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



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



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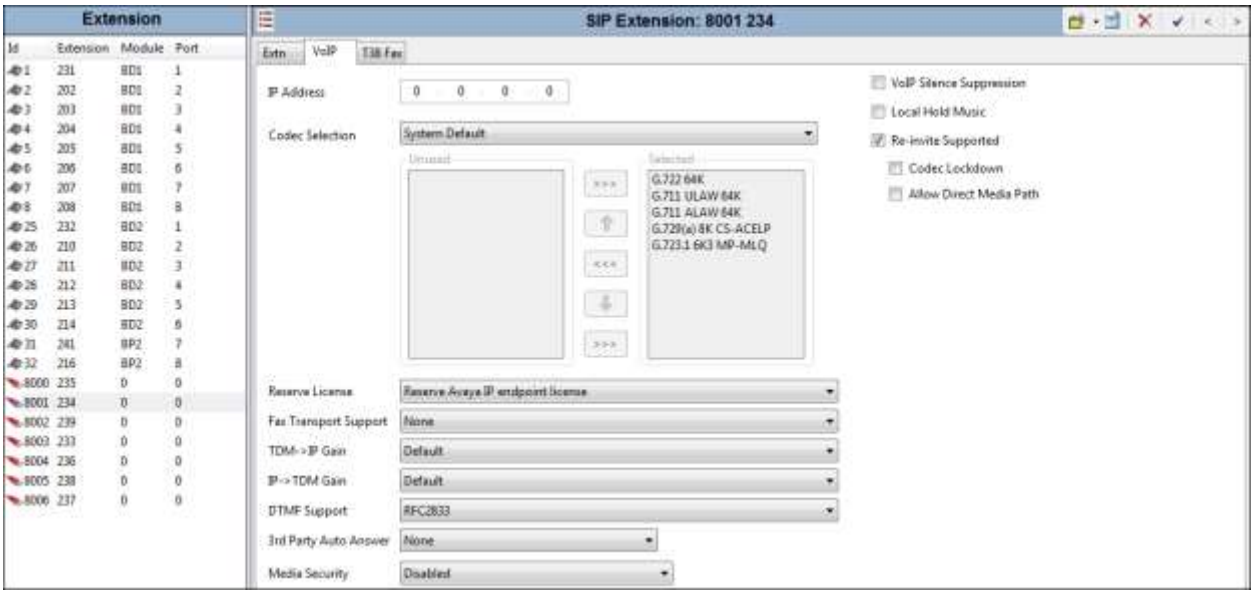
The following screen shows the Extension information for this user, simply to illustrate the **VoIP** tab available for a SIP Telephone. To view, select **Extension** (not shown) from the Navigation pane, and the appropriate extension from the Group pane. Select **VoIP** in the Details pane. The new **Codec Selection** parameter may retain the default setting "System Default" to follow the system configuration shown in **Section 5.2.6**. Alternatively, "Custom" may be selected to allow the codecs to be configured for this extension, using the arrow keys to select and order the codecs.



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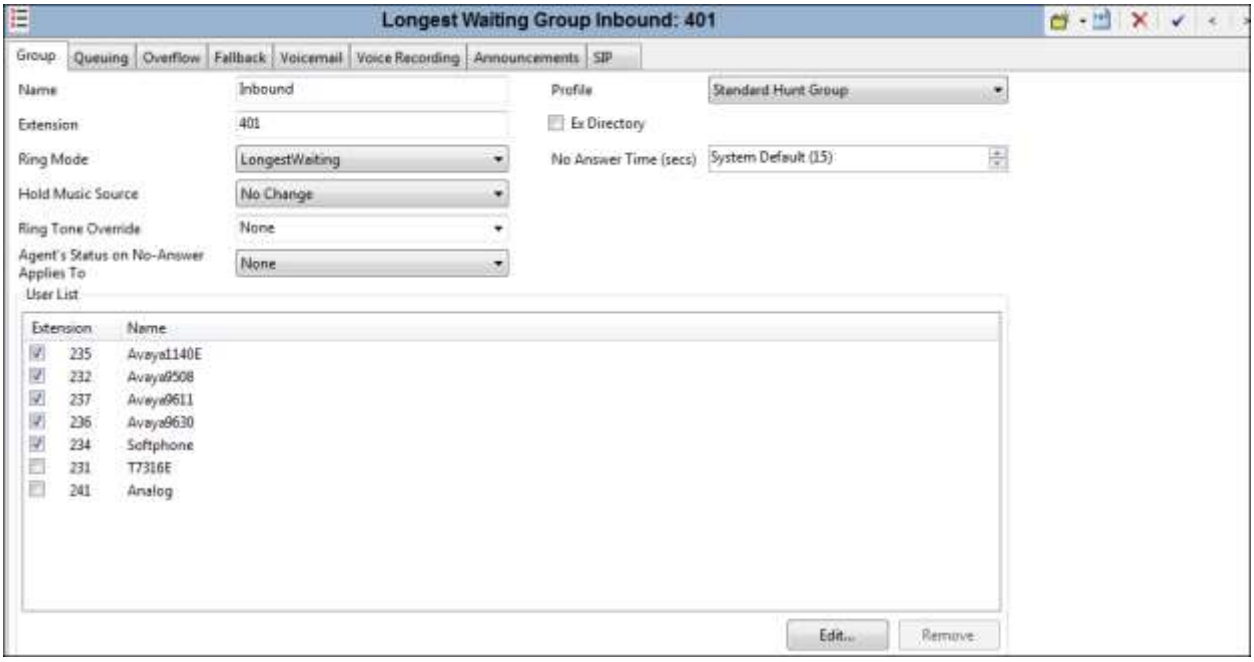
5.5.3. Hunt Groups

During the verification of these Application Notes, users could also receive incoming calls as members of a hunt group. To configure a new hunt group, right-click **HuntGroup** (not shown) from the Navigation pane, and select **New**. To view or edit an existing hunt group, select **HuntGroup** from the Navigation pane, and the appropriate hunt group from the Group pane.

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



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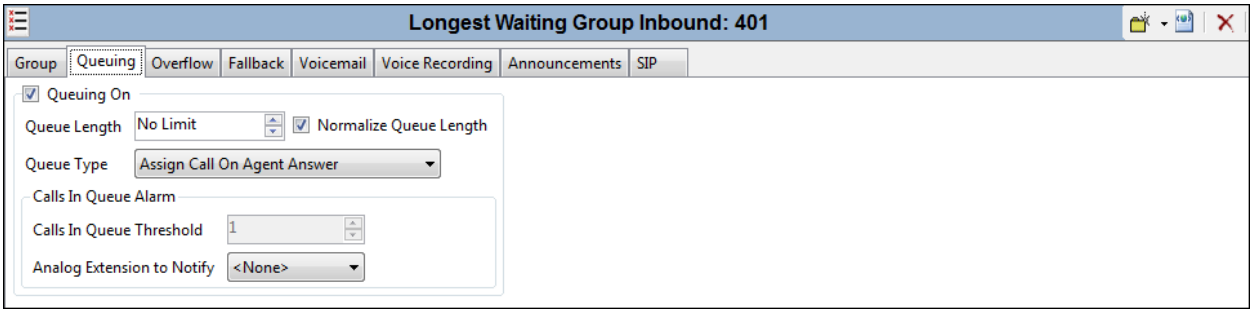
The following screen shows the **Queuing** tab for hunt group 401. In the sample configuration, the hunt group is configured to allow queuing so that incoming Verizon toll-free calls could be queued when all the members of the hunt group were busy on calls. In the testing associated with these Application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



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

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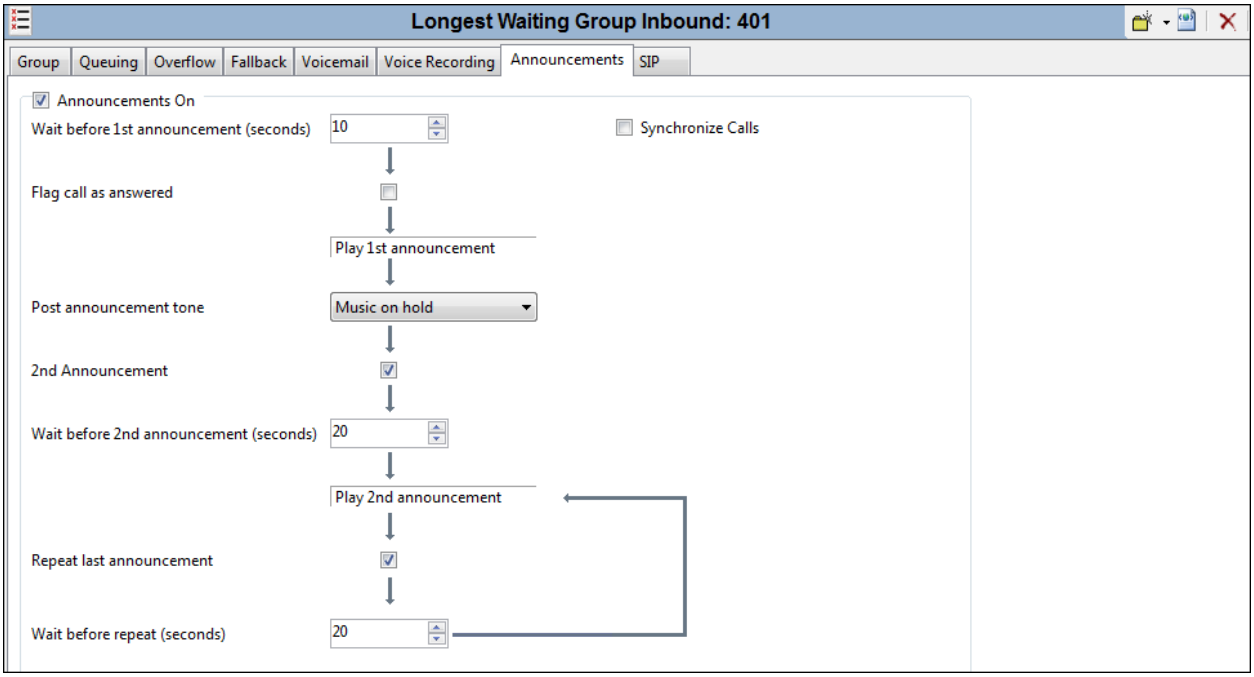
IP Office supports priority for queuing. For example, if low priority calls are waiting in queue, a higher priority call entering queue can be moved to the front of the queue and serviced before lower priority callers. For an inbound SIP trunk call, the priority can be specified on the Incoming Call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



5.6. Short Codes

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

In the screen shown below, the short code "8N;" is repeated from references [VZBIPT-IPO91]. The **Code** parameter is set to "8N;". The **Feature** parameter is set to "Dial". The **Telephone Number** parameter is set to N"@Domain Name or IP Address of Verizon Business IP Trunk Service". 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 17, matching the number of the **Outgoing Group** configured on the **SIP URI** tab of SIP Line 17 to Verizon Business documented in [VZBIPT-IPO91]. Although Verizon Business IPCC service, the focus of these Application Notes, is used for inbound toll-free numbers, inbound toll-free calls can be twinned, forwarded, or transferred back to the PSTN via the Verizon Business IP Trunk SIP Line. In addition, inbound IPCC toll-free calls used to access the IP Office mobile call back feature can have the call back occur using the Verizon Business IP Trunk SIP Line. For more information on outbound calls, short codes, and ARS, see reference [VZBIPT-IPO91].

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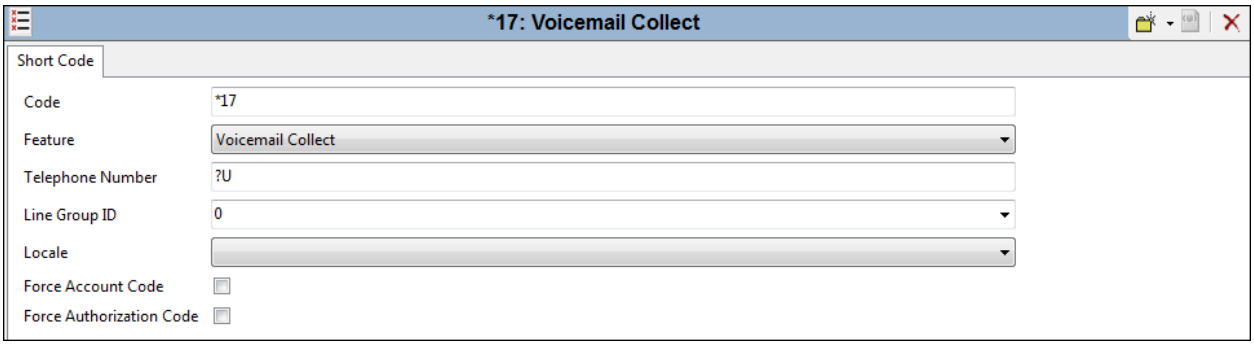
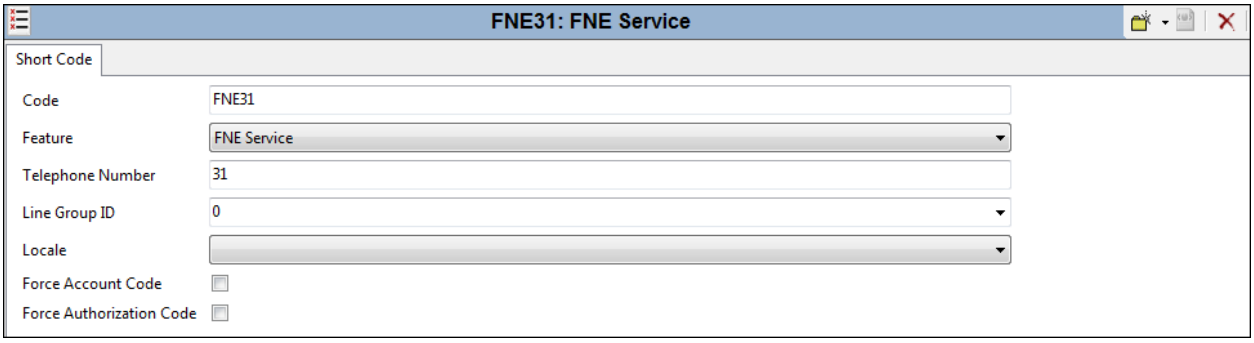
The following screen illustrates a short code that acts like a feature access code rather than a means to access a SIP Line. In this case, the **Code** "*17" is defined for **Feature** "Voicemail Collect". This short code will be used as one means to allow a Verizon toll-free number to be programmed to route directly to voice messaging, via inclusion of this short code as the destination of an Incoming Call Route. See **Section 5.7** for configuration of Incoming Call Routes.



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



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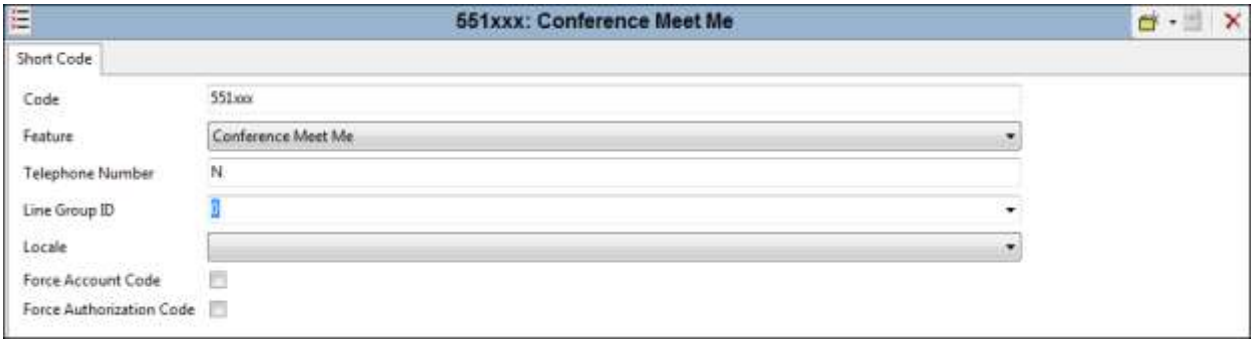
The following screen illustrates another short code. In this case, the **Code** “551xxx” is defined for **Feature** “Conference Meet Me” and **Telephone Number** “N”. In the verification of these Application Notes, this short code was used in conjunction with a Voicemail Pro module named “MeetMeConf”. Although the Voicemail Pro configuration is beyond the scope of these Application Notes, the module enabled a PSTN caller to dial a Verizon toll-free number, be prompted to enter a conference ID and PIN by Voicemail Pro, and then be transferred to the appropriate meet-me conference based on the ID entered by the caller. Local IP Office callers could also dial 551xxx to join the corresponding conference with ID xxx.



5.7. Incoming Call Routes

In this section, IP Office Incoming Call Routes are illustrated. Each Incoming Call Route will map a Verizon Business toll-free number to a destination user, group, or function on IP Office. In some cases, the destination will be chosen based on the combination of the toll-free number and the caller id of the caller. Example mappings are summarized in **Table 1** in **Section 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.

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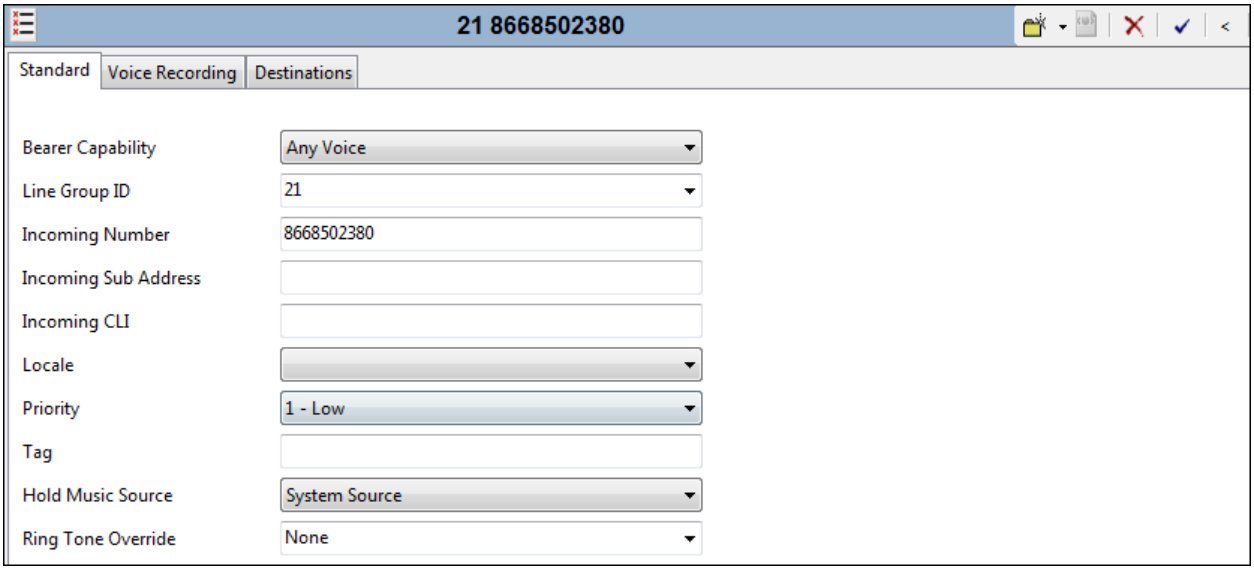
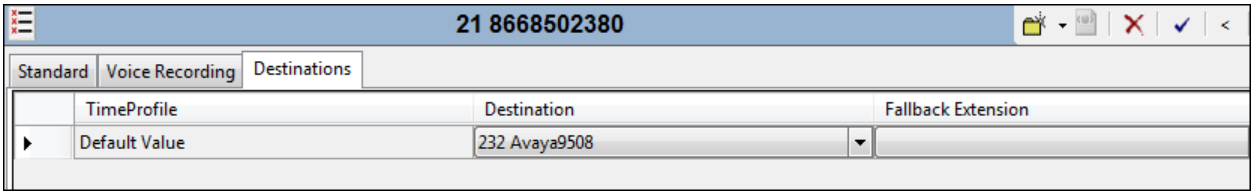
5.7.1. Incoming Call Route to a Specific Telephone Extension

In the screen shown below, the incoming call route for **Incoming Number** "8668502380" is illustrated. The **Line Group Id** is 21, matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to the Verizon Business IPCC service, in **Section 5.4.5**.



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



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

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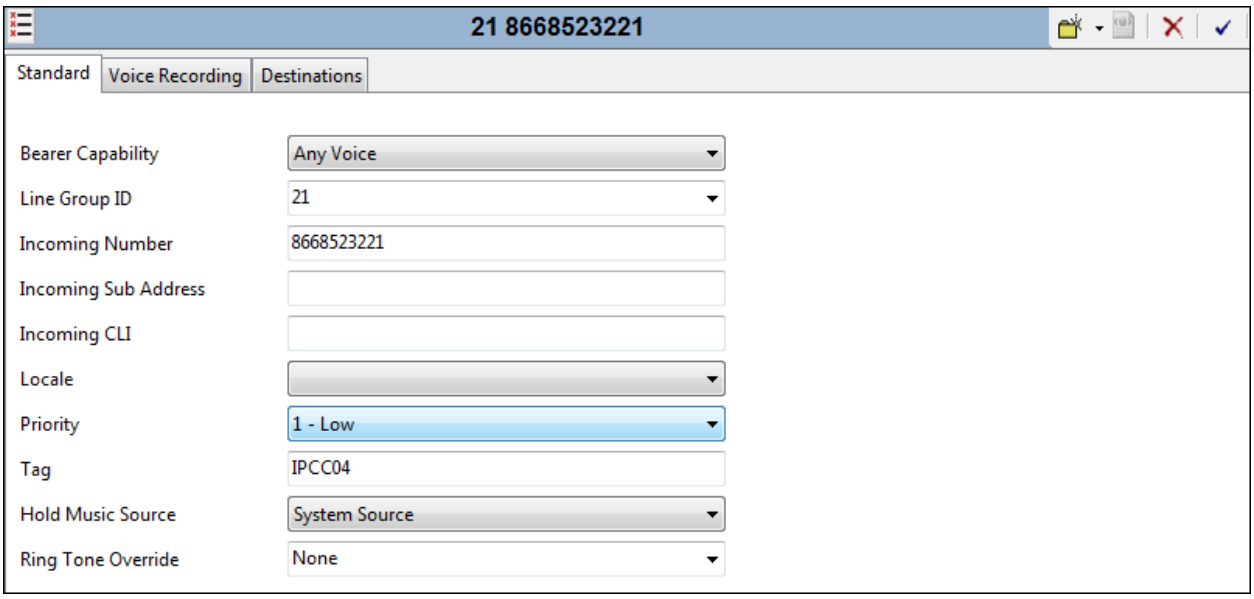
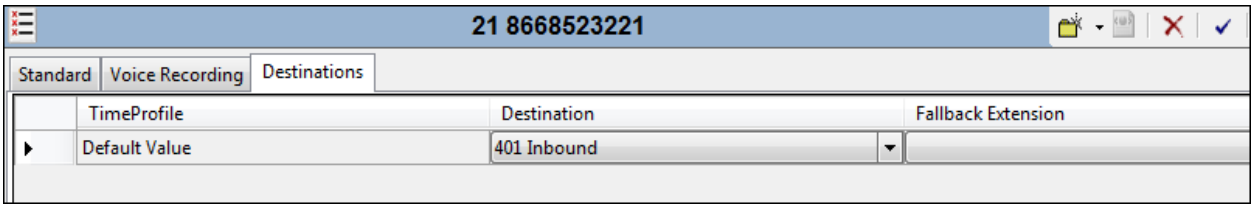
5.7.2. Incoming Call Routes to a Hunt Group by Dialed Toll-Free Number

In the screen shown below, an incoming call route for **Incoming Number** "8668523221" is illustrated. The **Line Group Id** is 21, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in **Section 5.4.5**.



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



5.7.3. Incoming Call Routes Based on Calling Party Number

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

In the screen shown below, the incoming call route for **Incoming Number** "8668523221" and **Incoming CLI** "3035382177" is illustrated. The **Line Group Id** is 21, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in **Section 5.4**. Note that the **Incoming Number** is the same as the toll-free number configured in the previous section. This route will be used for calls to the toll-free number specifically from a caller with

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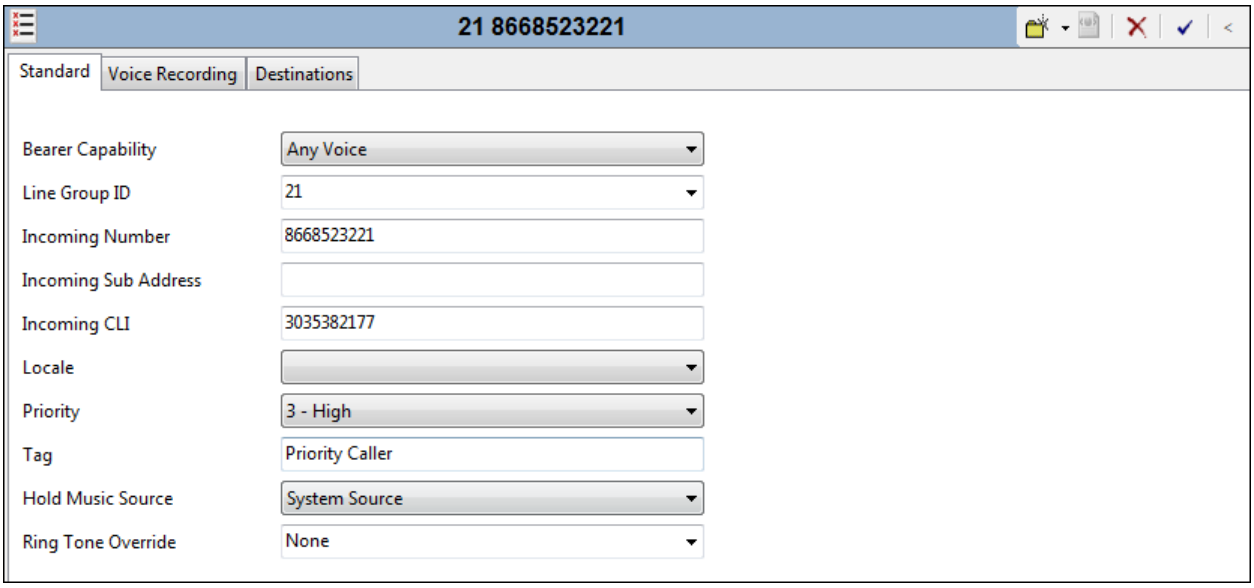
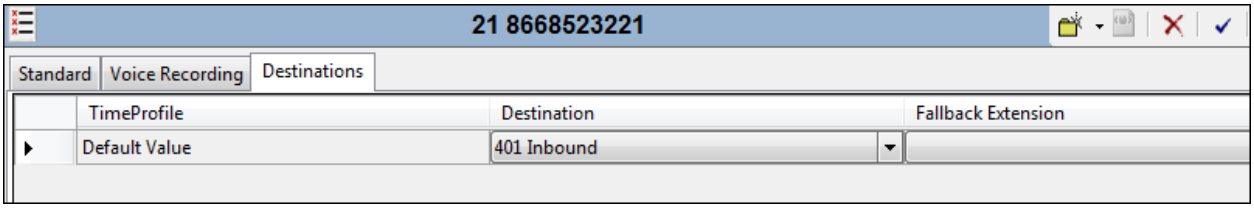
caller ID “3035382177”. In this case, to allow this caller to be treated with priority when calling in for “401 Inbound”, the **Priority** field is set to “3 – High”.



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



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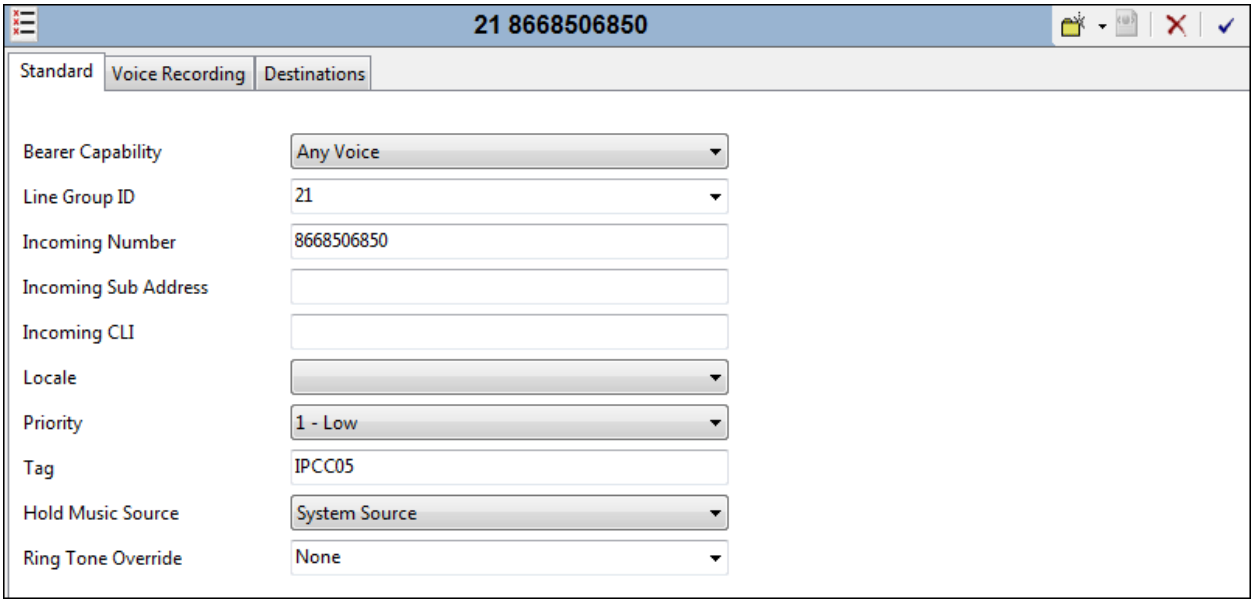
5.7.4. Incoming Call Routes to Various IP Office Features

In the sample configuration, the incoming call route for **Incoming Number** "8668506850" was varied to test different destination features, such as Voice Mail, Mobile Call Control, and Conference Bridge, as shown in **Table 1** in **Section 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 5.6**). With this destination, an incoming call to 8668506850 will be delivered directly to voice mail, allowing the caller to log-in to voice mail and access messages.
- "FNE31" (short code for accessing the Mobile Call Control feature directly, as shown in **Section 5.6**) With this destination, an incoming call to 8668506850 from configured mobile callers will be provided dial tone to make calls from the mobile phone as if the user were using their IP Office extension.
- "VM:MeetMeConf" With this destination, an incoming call to 8668506850 will be delivered directly to the Voicemail Pro Module "MeetMeConf" created for use as a conference bridge.

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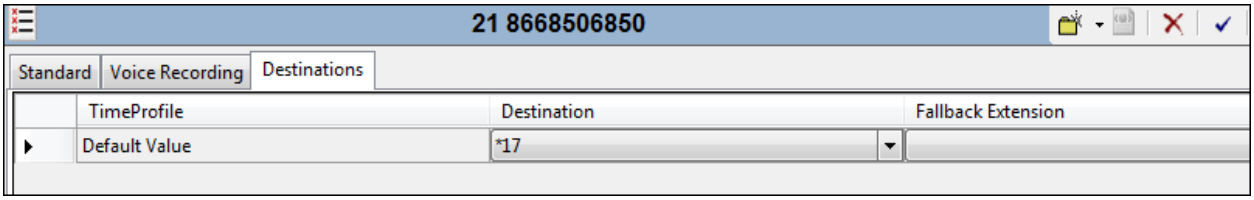
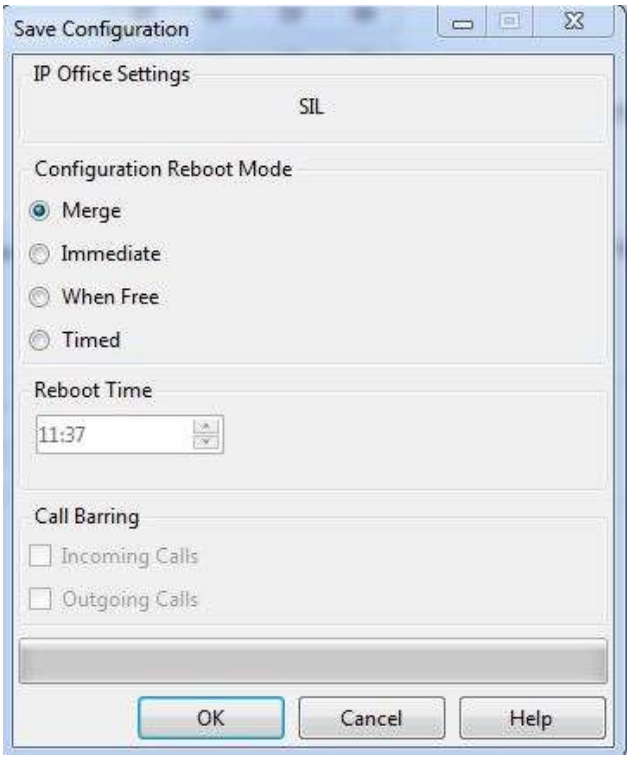
An example screen showing the short code configured for Voicemail Collect is shown below.



5.8. Saving Configuration Changes to IP Office

Navigate to **File → Save Configuration** (not shown) in the menu bar at the top of the screen to save the configuration performed in the preceding sections.

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



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6. Verizon Business Configuration

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

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

For service provisioning, Verizon will require the customer IP address of the Avaya 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**.

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7. Verification

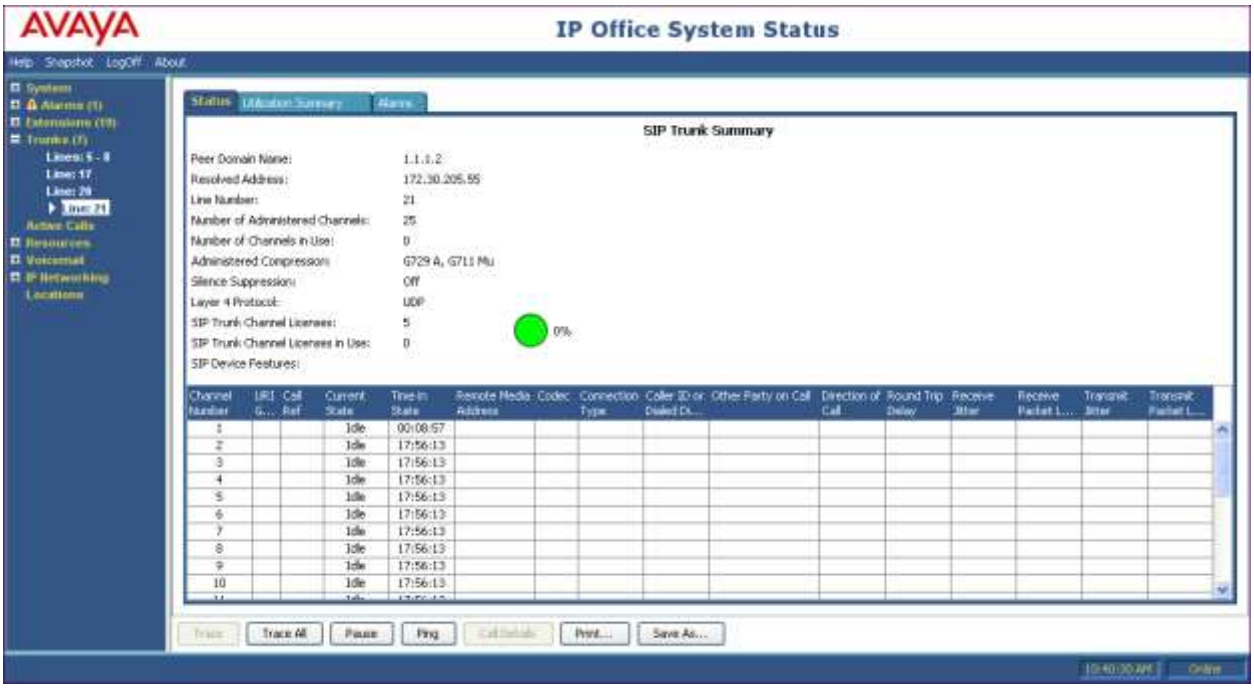
This section illustrates means to verify the configuration besides simply making the types of calls detailed in **Section 2.1**.

7.1.1. 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** (not shown). 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. See reference [4] for more information. Click on **System Status** to launch the application.

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

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



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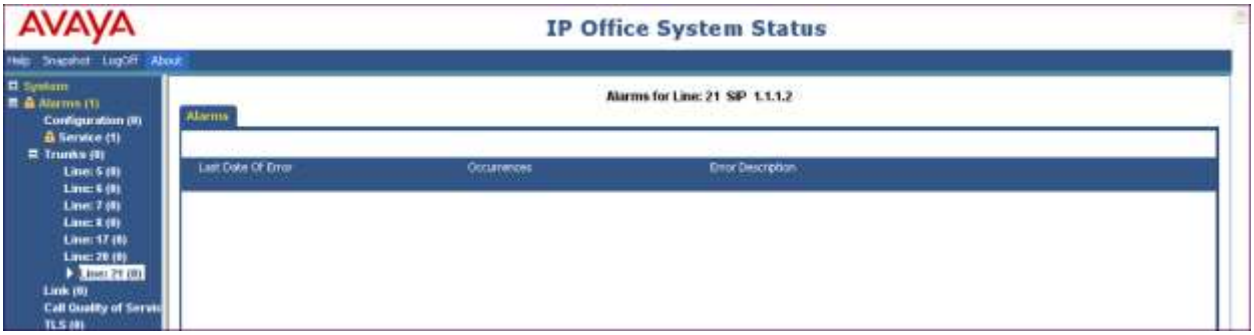
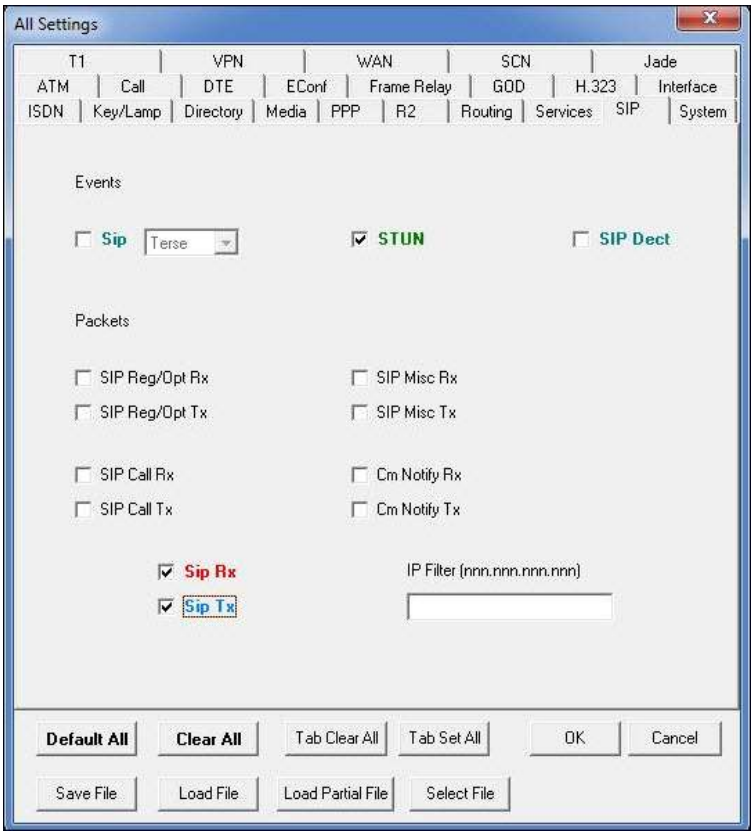
Select the **Alarms** tab and verify that no alarms are active on the SIP line.



7.1.2. System Monitor

The System Monitor application can also be used to monitor and troubleshoot IP Office. System Monitor can be accessed from **Start → Programs → IP Office → Monitor** (not shown). The application allows the monitored information to be customized. To customize, select the button that is third from the right in the screen below, or select **Filters →Trace Options** (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. SIP messages received will appear in the trace with the color red, while SIP messages sent will appear 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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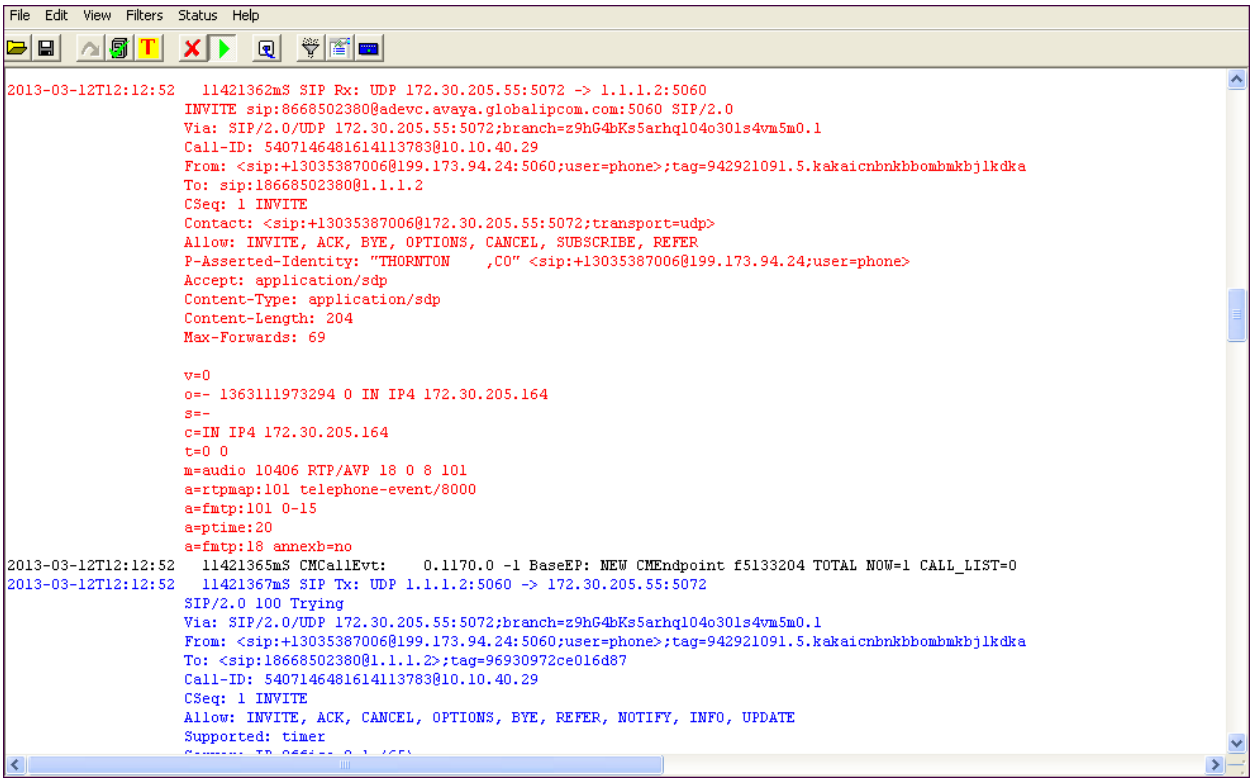
The following screen shows a portion of the monitor trace of an inbound call. As can be observed, PSTN caller 303-538-7006 dialed Verizon IP Toll Free number 1-866-850-2380. Details of the SIP INVITE message sent by Verizon are shown below. This information matches the configuration in these Application 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.



```
File  Edit  View  Filters  Status  Help

2013-03-12T12:12:52   11421362mS SIP Rx: UDP 172.30.205.55:5072 -> 1.1.1.2:5060
                      INVITE sip:8668502380@adevc.avaya.globalipcom.com:5060 SIP/2.0
                      Via: SIP/2.0/UDP 172.30.205.55:5072;branch=z9hG4bKs5arhql04o30ls4vm5m0.1
                      Call-ID: 5407146481614113783@10.10.40.29
                      From: <sip:+13035387006@199.173.94.24:5060;user=phone>;tag=942921091.5.kakaicnbnkbbombmkbjlkdka
                      To: sip:18668502380@1.1.1.2
                      CSeq: 1 INVITE
                      Contact: <sip:+13035387006@172.30.205.55:5072;transport=udp>
                      Allow: INVITE, ACK, BYE, OPTIONS, CANCEL, SUBSCRIBE, REFER
                      P-Asserted-Identity: "THORNTON    ,CO" <sip:+13035387006@199.173.94.24;user=phone>
                      Accept: application/sdp
                      Content-Type: application/sdp
                      Content-Length: 204
                      Max-Forwards: 69

                      v=0
                      o=- 13631111973294 0 IN IP4 172.30.205.164
                      s=-
                      c=IN IP4 172.30.205.164
                      t=0 0
                      m=audio 10406 RTP/AVP 18 0 8 101
                      a=rtpmap:101 telephone-event/8000
                      a=fmtp:101 0-15
                      a=ptime:20
                      a=fmtp:18 annexb=no
2013-03-12T12:12:52   11421365mS CMCallEvt:    0.1170.0 -1 BaseEP: NEW CMEndpoint f5133204 TOTAL NOW=1 CALL_LIST=0
2013-03-12T12:12:52   11421367mS SIP Tx: UDP 1.1.1.2:5060 -> 172.30.205.55:5072
                      SIP/2.0 100 Trying
                      Via: SIP/2.0/UDP 172.30.205.55:5072;branch=z9hG4bKs5arhql04o30ls4vm5m0.1
                      From: <sip:+13035387006@199.173.94.24:5060;user=phone>;tag=942921091.5.kakaicnbnkbbombmkbjlkdka
                      To: <sip:18668502380@1.1.1.2>;tag=96930972ce016d87
                      Call-ID: 5407146481614113783@10.10.40.29
                      CSeq: 1 INVITE
                      Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
                      Supported: timer
```

8. 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 9.1 can be successfully combined with a Verizon Business IP Contact Center SIP trunk 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.

IP Office Release 9.1 has not been independently certified by Verizon labs. These Application Notes can be used to facilitate customer engagements via the Verizon field trial process, pending Verizon labs independent certification.

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9. 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 9.1, Deploying Avaya IP Office ™ Platform IP500 V2*, Document Number 15-601042, January 2015

[2] *Administering Avaya IP Office ™ Platform with Manager*, January 2015

[3] *IP Office ™ Platform 9.1, Installing and Maintaining the Avaya IP Office ™ Platform Application Server*, Document Number 15-601011 Issue 10d, March 2015

[4] *IP Office ™ Platform 9.1, Deploying Avaya IP Office ™ Platform Servers as Virtual Machines*, Document Number 15-601011 Issue 03c, January 2015

[5] *IP Office ™ Platform 9.1, Using Avaya IP Office ™ System Status*, Document Number 15-601758, October 2014

[6] *Administering Avaya Communicator on IP Office*, December 2014

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

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

[9] *Avaya IP Office Knowledgebase*, 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 9.0 with Verizon Business IPCC Service Suite.
[VZB-IPCCIPOR9] Application Notes for Configuring SIP Trunking using Verizon Business IP Contact Center VoIP Inbound and Avaya IP Office Release 9, Issue 1.0

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 9.1 with Verizon IP Trunk Service Suite.
[VZBIPT-IPO91] Application Notes for SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service and Avaya IP Office Release 9.1– Issue 1.0

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