Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking Using Verizon Business IP Contact Center VoIP Inbound and Avaya IP Office Release 8.1 – 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 8.1 Essential Edition, Avaya Voicemail Pro, Avaya IP Office Softphone, 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 8.1 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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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 8.1 Essential Edition, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, SIP, digital, and analog endpoints.

IP Office Release 8.1 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/

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.

1.1. Interoperability Compliance Testing

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

- SIP OPTIONS monitoring of the health of the SIP trunk was verified. Verizon Business 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

telephones, Avaya digital telephones, analog telephones, Avaya IP Office Softphone, and Avaya IP Office Voicemail Pro.
- Proper disconnect when either party hangs up an active call
- Proper disconnect when the PSTN caller abandons (i.e., hangs up) a toll-free call before the IP Office party has answered.
- Proper SIP 486 response and busy tone heard by the caller when a PSTN user calls a toll-free number directed to a busy IP Office user, an IP Office user with Do-not-disturb active, or an IP Office user that is logged out (i.e., assuming no redirection is configured for these conditions). Similarly, busy tone is heard when a PSTN user calls a toll-free number whose "SIP URI Max Calls per Channel" has been reached (see Section 5.4). 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 5.5.4).
- 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 (See Section 2.2. Although long SIP sessions are not refreshed, the media paths remain connected.)
- 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 5.5.4 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-IPO81]. As detailed in reference [VZBIPT-IPO81], these outgoing PSTN calls can be originated from Avaya H.323 telephones, Avaya SIP telephones, Avaya digital telephones, analog endpoints, and Avaya IP Office Softphone. 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-IPO81], presenting true calling party information to the mobile phone. See Section 2.2 for additional information.
- Mobile twinning of Verizon toll-free calls to a mobile phone via the Verizon Business IP Trunk service documented in reference [VZBIPT-IPO81], 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 5.6. 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

1.2. 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. The Verizon Business IPCC service does not support fax.

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

3. Although the Verizon Business IPCC service supports transfer using the SIP REFER method and IP Office supports sending REFER, IP Office will not send REFER to Verizon in the verified configuration.

4. The SIP protocol allows sessions to be refreshed for calls that remain active for some time. In the tested configuration, neither Verizon nor IP Office send SIP re-INVITE or UPDATE messages to refresh a session. In the tested configuration, this is transparent to the users that are party to the call in that the media paths remain established.

5. When a user on the PSTN hangs up an active call, Verizon Business IPCC will send an INVITE with SDP containing 0.0.0.0 before sending the BYE to clear the call. IP Office

processes the INVITE with SDP containing 0.0.0.0 as a request to hold the call, and then processes the BYE to disconnect the call. If the IP Office user is still listening after the PSTN user hangs up, the IP Office user may very briefly hear music on hold from IP Office before the BYE is processed and the call appearance is idled.

6. IP Office does not support the receipt of an initial INVITE that does not contain SDP. Therefore, IP Office does not support the Verizon Business IPCC "enhanced transfer" service, which sends an initial INVITE without SDP to the transfer-to site of an enhanced transfer.

1.3. Support

1.3.1. Avaya

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

1.3.2. Verizon

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

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-IPO81] 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*.

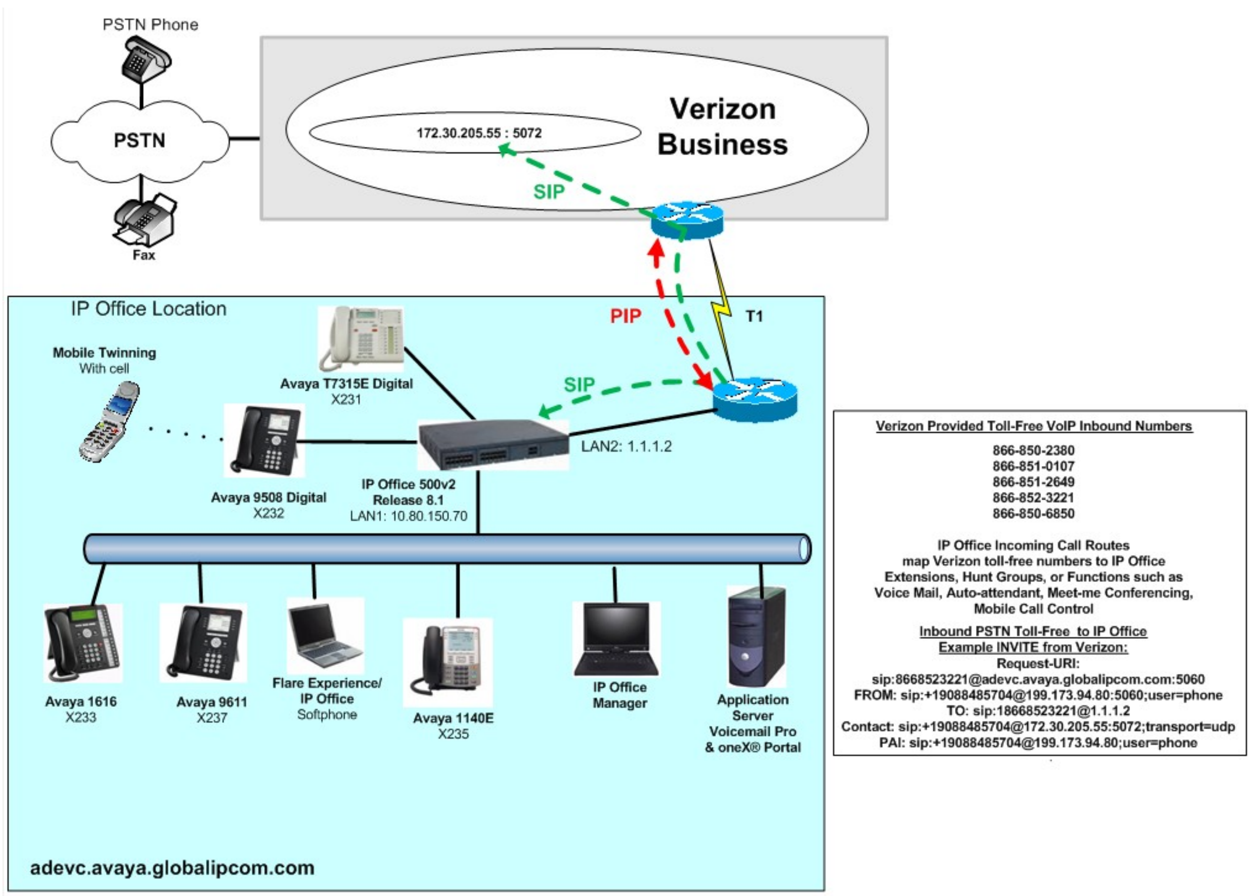


Figure 1: Avaya IP Office with Verizon IP Contact Center Service

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 Flare Experience capabilities.
866-851-2649	x233, x237	Avaya 1616 Telephone, Avaya 9611 Telephone
866-850-6850	Voicemail Collect on Voicemail Pro	Allow external callers to access voice mail toll-free
866-850-6850	Inbound Mobile Call Control	Allow toll-free calls from pre-configured twinning numbers to access mobile call control
866-850-6850	Conference Bridge on Voicemail Pro	Allow external callers to access conference bridge toll-free
866-852-3221 (any caller)	"401 Sales" Hunt Group (with default priority)	Hunt Group with queuing
866-852-3221 (specific callers)	"400 Overdue Account" Hunt Group	Show IP Office destination selection based on caller ID
866-852-3221 (specific priority callers)	"401 Sales" Hunt Group (with High Priority)	Show IP Office priority queuing based on caller ID

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

Equipment and Software Validated

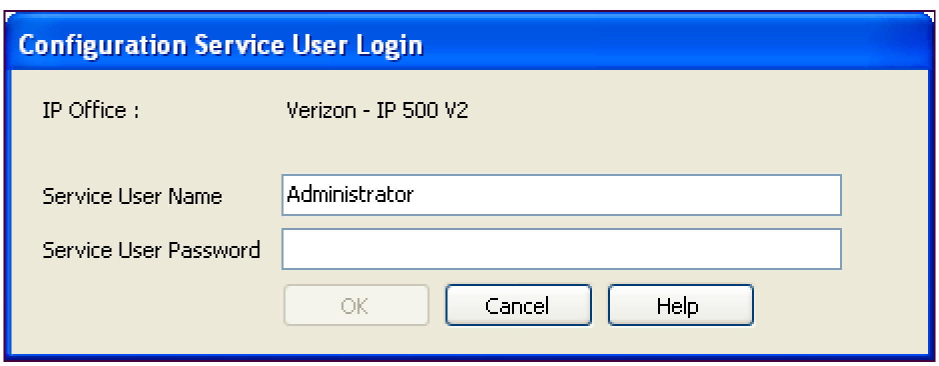
Table 2 shows the equipment and software used in the sample configuration.

Equipment	Software
Avaya IP Office 500 v2	Release 8.1 (65)
Avaya IP Office Manager	Release 10.1 (65)
Avaya Application Server	8.1.20-3
Avaya 2500 Analog Telephone	N/A
Avaya 9508 Digital Telephone	N/A
Avaya T7315E Digital Telephone	N/A
Avaya 1616 IP Telephone (H.323)	Release 1.302B
Avaya 9611 IP Telephone (H.323)	Release 6.2209
Avaya 1140E SIP	04.03.12
Avaya IP Office Softphone	Release 3.2.3.20 64770
Avaya Flare Experience	1.1.0.5

Table 2: Equipment and Software Tested

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:



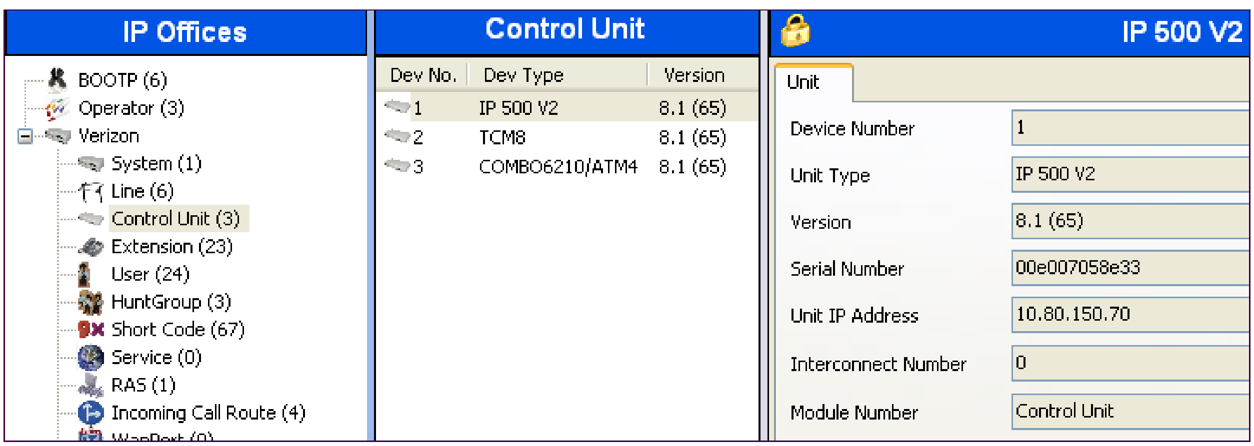
Log in with the appropriate configuration credentials. The appearance of the IP Office Manager can be customized using the **View** menu (not shown). In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and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

1.4. Physical, Network, and Security Configuration

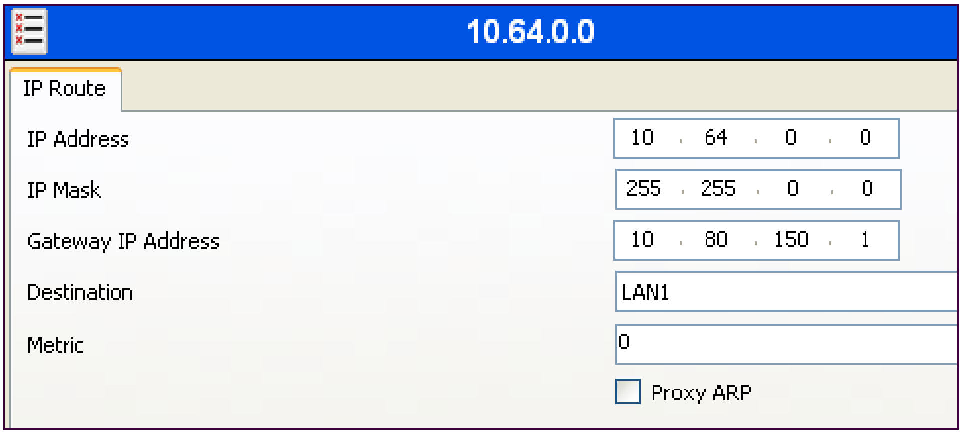
This section describes attributes of the sample configuration, but is not meant to be prescriptive. Consult reference [1] for more information on the topics in this section.

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, G729Aand 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

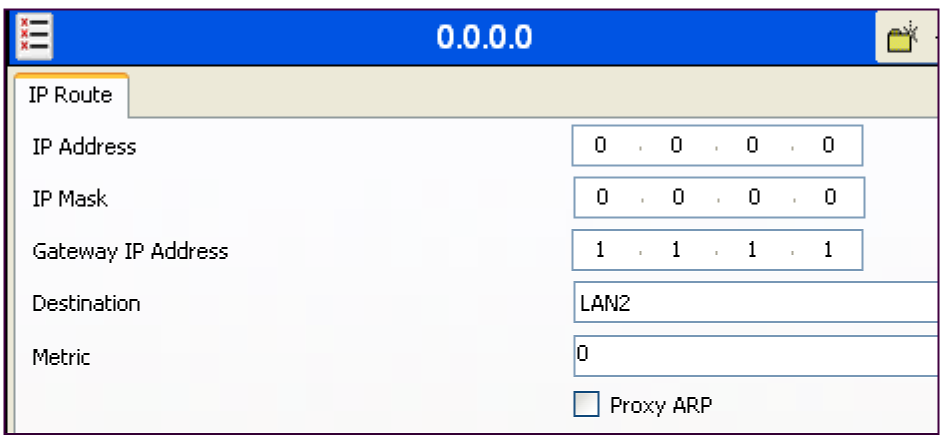
The following screen shows the modules in the IP Office used in the sample configuration. To access such a screen, select **Control Unit** in the Navigation pane. The modules appear in the Group pane. In the screen below, **IP 500 V2** is selected in the Group pane, revealing additional information about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



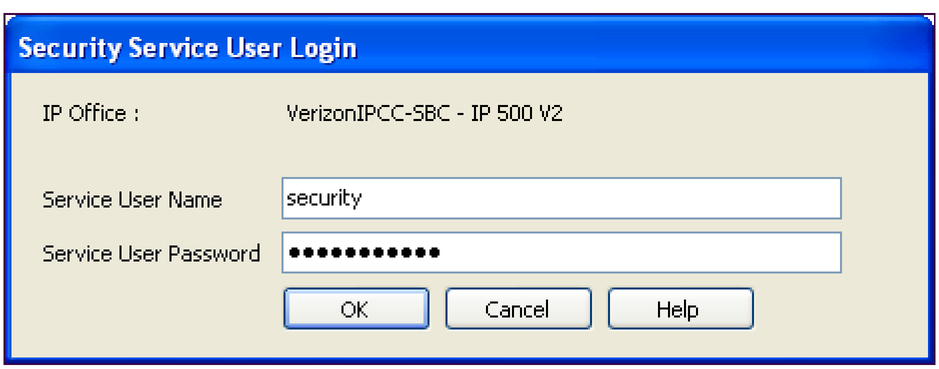
In the sample configuration, the IP Office LAN1 port is physically connected to the local area network switch at the IP Office customer site. The default gateway for this network is 10.80.150.1. To add an IP Route in IP Office, right-click **IP Route** from the Navigation pane, and select **New**. To view or edit an existing route, select **IP Route** from the Navigation pane, and select the appropriate route from the Group pane. The following screen shows the Details pane with the relevant route using **Destination** LAN1.



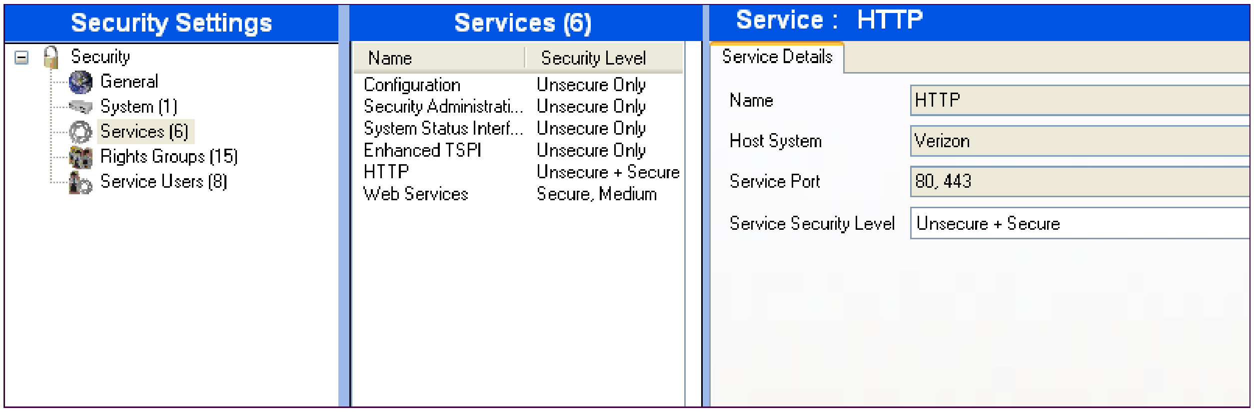
The IP Office LAN2 port is physically connected to the service provider and has a default gateway of 1.1.1.1. Right-click **IP Route** from the Navigation pane, and select **New** to add another route. The following screen shows the Details pane with the relevant route using **Destination** LAN2



To facilitate use of Avaya IP Office Softphone, https was enabled in the sample configuration. To check whether https is enabled, navigate to **File → Advanced → Security Settings**. A screen such as the following is presented. Log in with the appropriate security credentials.



After logging in, select **Services** from the Navigation pane and **HTTP** from the Group pane. In the Details pane, verify the **Service Security Level** is configured as intended, as shown below.

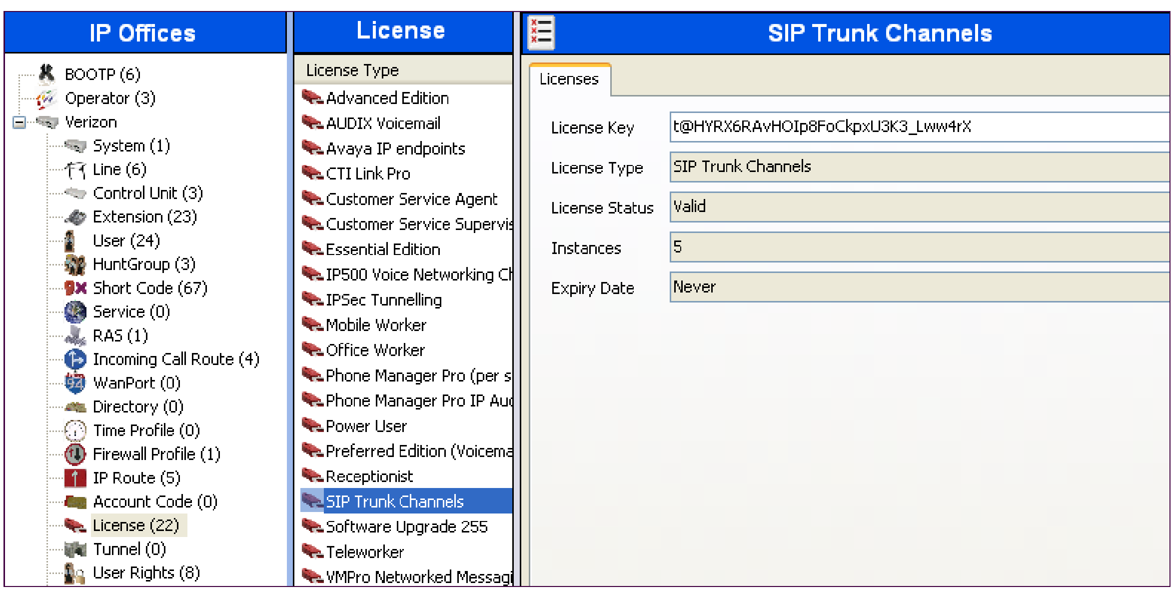


When complete, select **File → Configuration** to return to configuration activities.

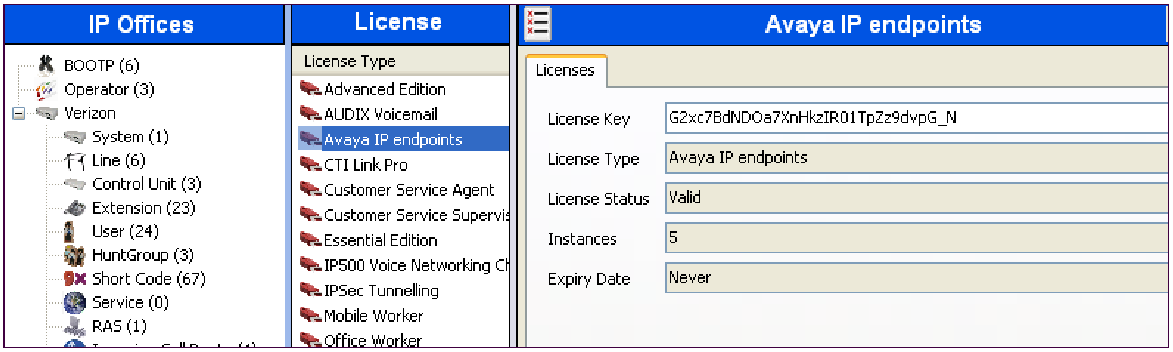
1.5. Licensing

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.

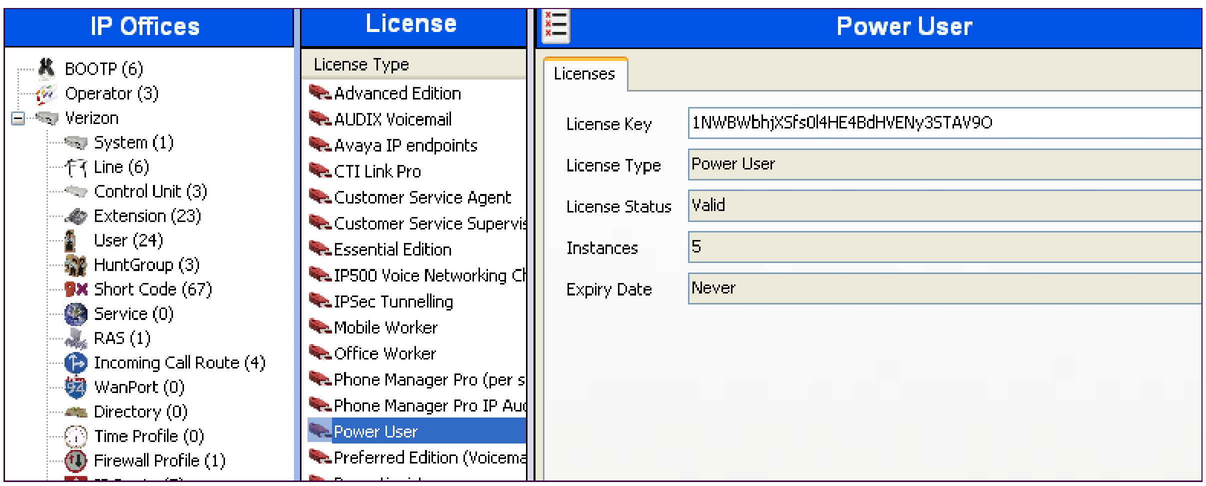
To verify that there is a SIP Trunk Channels License with sufficient capacity, click **License** in the Navigation pane and **SIP Trunk Channels** in the Group pane. Confirm a valid license with sufficient "Instances" (trunk channels) in the Details pane.



If Avaya IP Telephones will be used, verify the Avaya IP endpoints license. Click **License** in the Navigation pane and **Avaya IP endpoints** in the Group pane. Confirm a valid license with sufficient "Instances" in the Details pane.



A similar process can be used to check the license status for other desired features. For example, the following screen shows the availability of a valid license for **Power User** features. In the sample configuration, the user with extension 234 will be configured as a "Power User" and will be capable of using the Avaya IP Office Softphone.

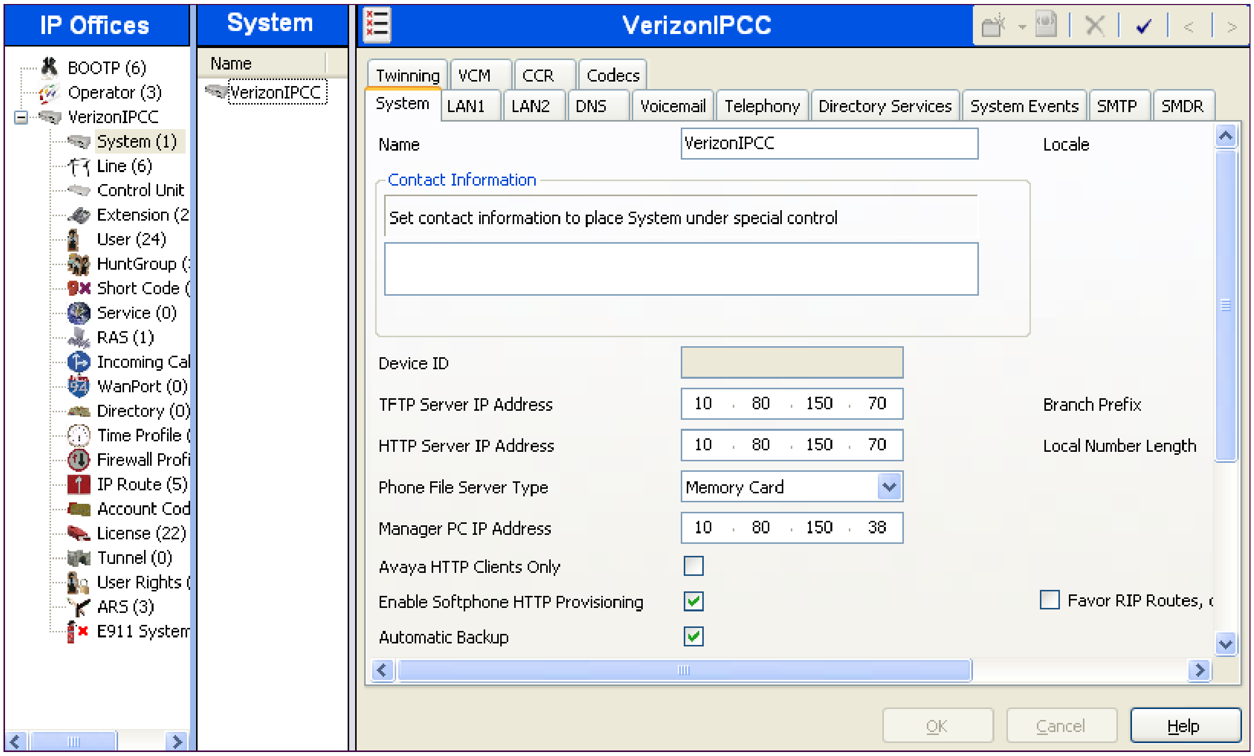


1.6. System Settings

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.

1.6.1. System Tab

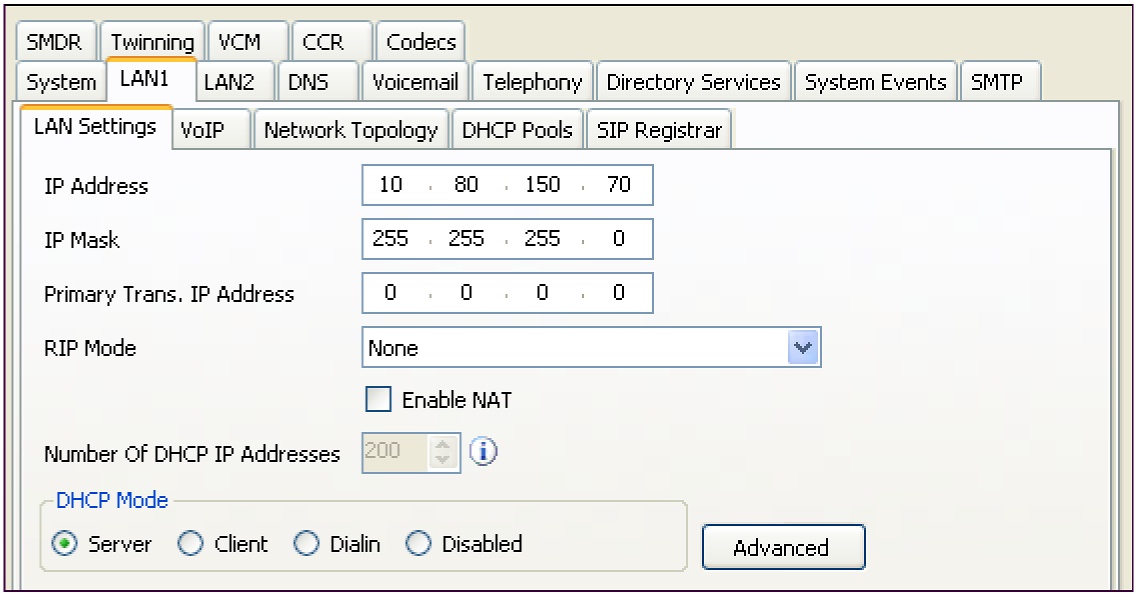
With the proper system name selected in the Group pane, select the **System** tab in the Details pane. The following screen shows a portion of the **System** tab. The **Name** field can be used for a descriptive name of the system. In this case, "VerizonIPCC" is used as the name. The **Enable SoftPhone HTTP Provisioning** box is checked to facilitate Avaya IP Office Softphone usage.



1.6.2. LAN 1 Settings

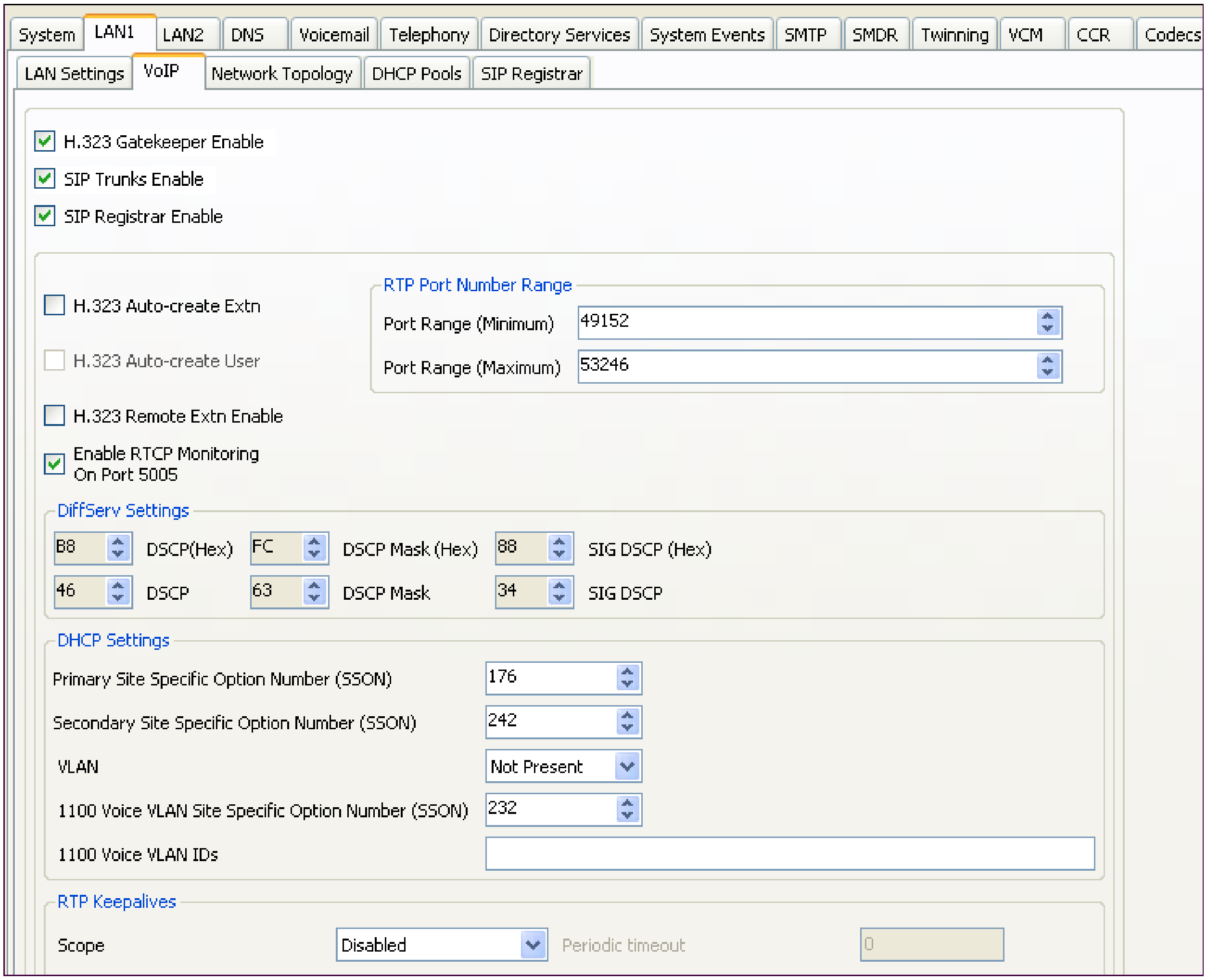
The IP500/IP500 V2 control units have 2 RJ45 Ethernet ports, physically marked as LAN and WAN. Within the system configuration, the physical LAN port is LAN1, the physical WAN port is LAN2.

In the sample configuration, LAN1 was used to connect the IP Office to the enterprise network. To view or configure the **IP Address** of LAN1, select the **LAN1** tab followed by the **LAN Settings** tab. As shown in **Figure 1**, the IP Address of the IP Office is 10.80.150.70. Other parameters on this screen may be set according to customer requirements. In the example screen, the **DHCP Mode** was set to “Server” to allow IP Office to facilitate provisioning for the IP Telephones in the sample configuration.

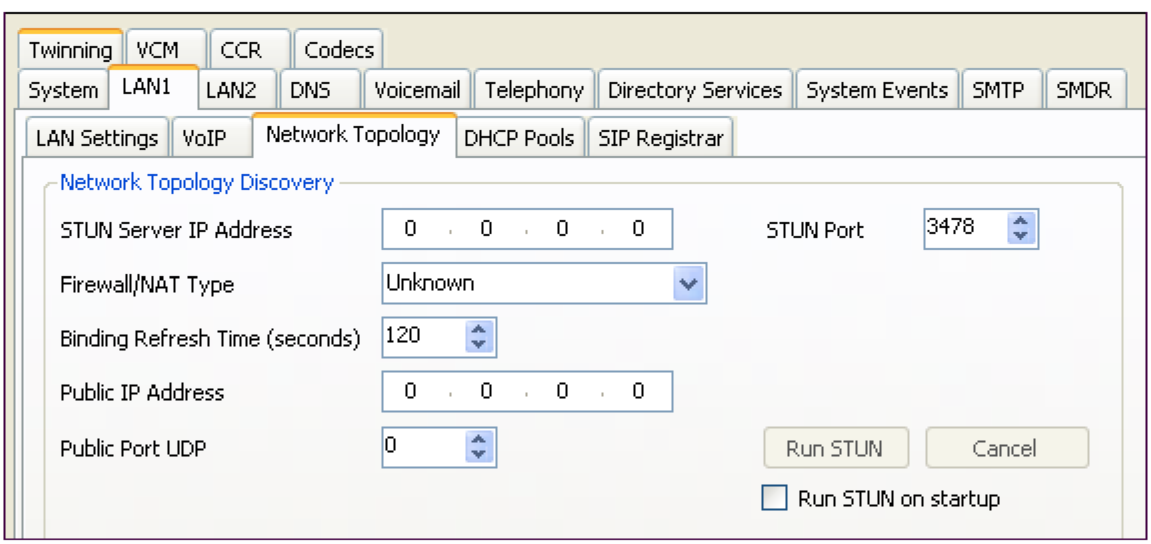


Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 1600-Series and 9600-Series Telephones used in the sample configuration. The **SIP Registrar Enable** box is checked to allow Avaya 1140E, Avaya Flare Experience, and Avaya IP Office Softphone usage. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Verizon Business

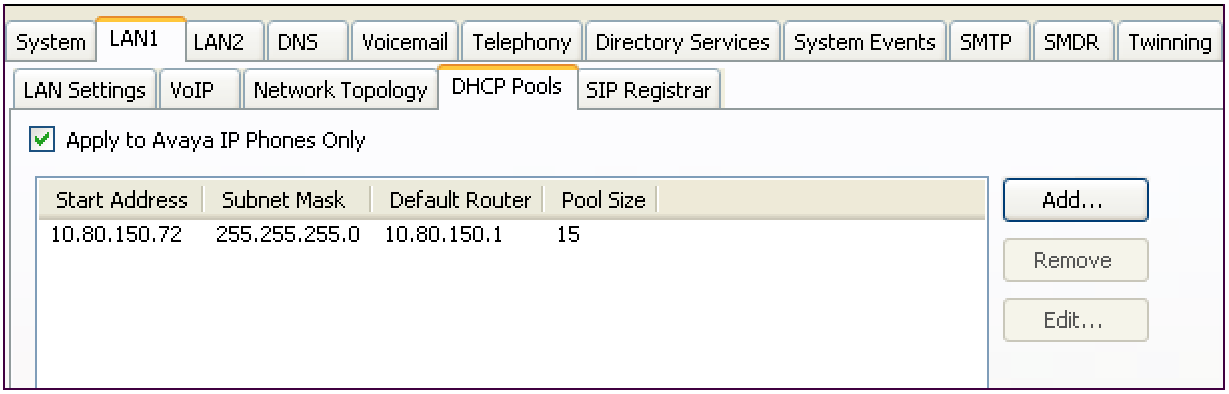
RTP Port Number: For each VoIP call, a receive port for incoming Real Time Protocol (RTP) traffic is selected from a defined range of possible ports, using the even numbers in that range. The Real Time Control Protocol (RTCP) traffic for the same call uses the RTP port number plus 1 (i.e., the odd numbers). For control units and Avaya H.323 IP phones, the default port range used is 49152 to 53246. On some installations, it may be a requirement to change or restrict the port range used. It is recommended that only port numbers between 49152 and 65535 are used, that being the range defined by the Internet Assigned Numbers Authority (IANA) for dynamic usage. **Port Range (minimum):** Default = 49152. Range = 1024 to 64510. This sets the lower limit for the RTP port numbers used by the system. **Port Range (maximum):** Default = 53246. Range = 2048 to 65534. This sets the upper limit for the RTP port numbers used by the system. The gap between the minimum and the maximum must be at least 1024.



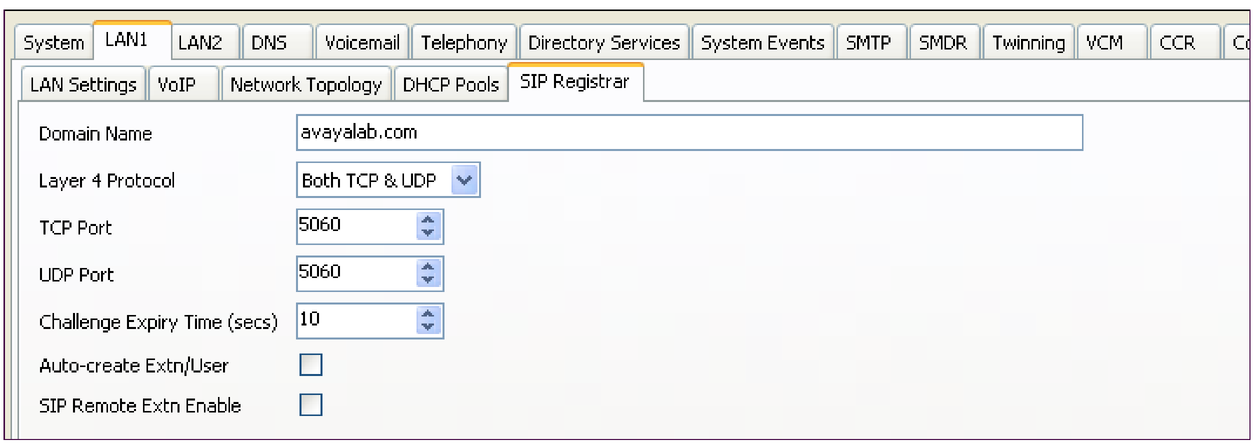
Select the **Network Topology** tab as shown in the following screen. In the sample configuration, the default settings were used.



If using IP Office as a DHCP server and DHCP Server mode has been selected from the **LAN1 → Lan Settings** Tab, click the **DHCP Pools** tab. Although beyond the intended scope of these Application Notes, the following screen is shown as a simple example.

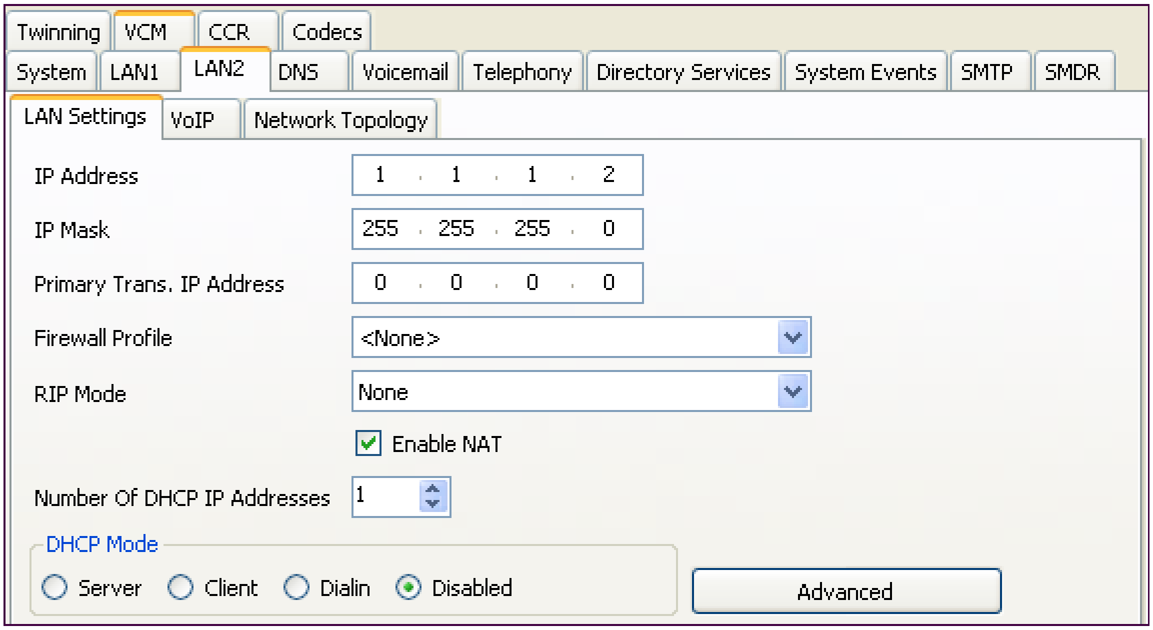


Optionally, select the **SIP Registrar** tab. The following screen shows the settings used in the sample configuration. The **Domain Name** has been set to the customer premises equipment domain "avayalab.com". If the **Domain Name** is left at the default blank setting, SIP registrations may use the IP Office LAN 1 IP Address. All other parameters shown are default values.



1.6.3. LAN 2 Settings

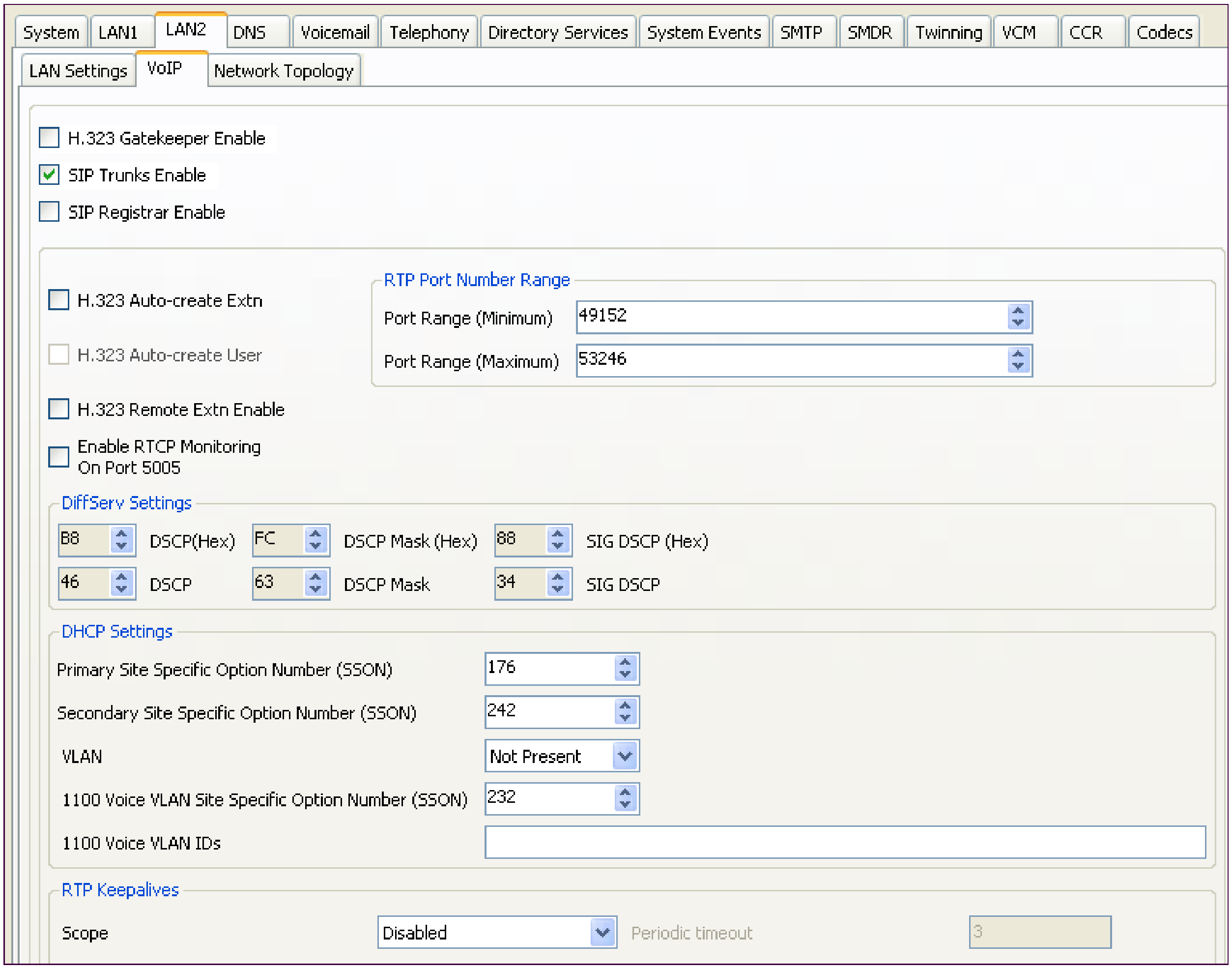
In the sample configuration, LAN2 was used to connect the IP Office to the Verizon network. To view or configure the **IP Address** of LAN2, select the **LAN2** tab followed by the **LAN Settings** tab. As shown in **Figure 1**, the IP Address of the IP Office, known to Verizon, is 1.1.1.2. **DHCP Mode**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



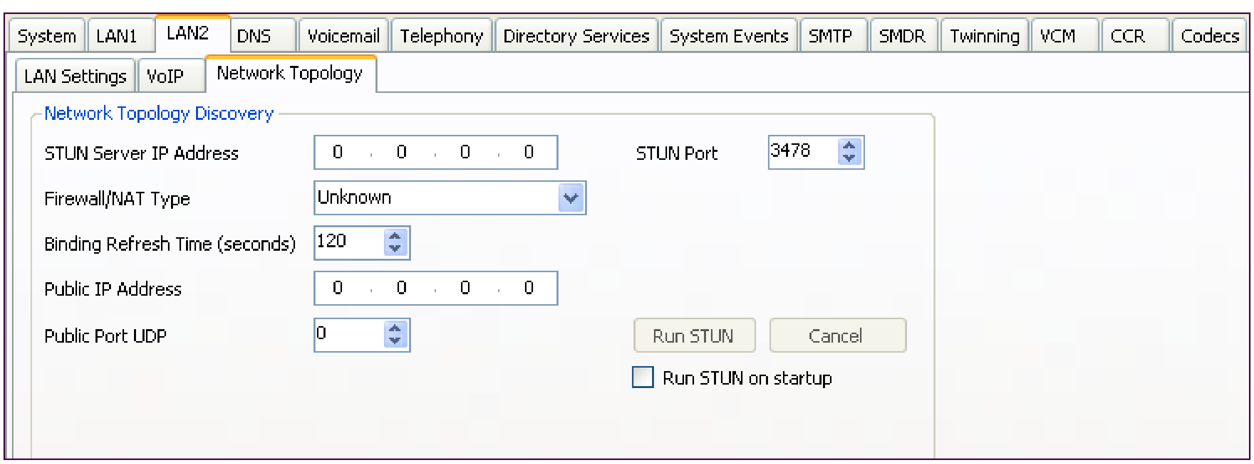
Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** and **SIP Registrar Enable** boxes are unchecked since IP telephones will not be registering on this link. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Verizon Business.

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. See Section 5.3.2 for more information on these RTP settings.

If desired, 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 34 (**SIG DSCP** parameter). IP Office will mark the RTP media with a value associated with “Expedited Forwarding” using DSCP decimal 46 (**DSCP** parameter). This screen enables flexibility in IP Office DiffServ markings (RFC 2474) to allow alignment with network routing policies, which are outside the scope of these Application Notes. Other parameters on this screen may be set according to customer requirements.



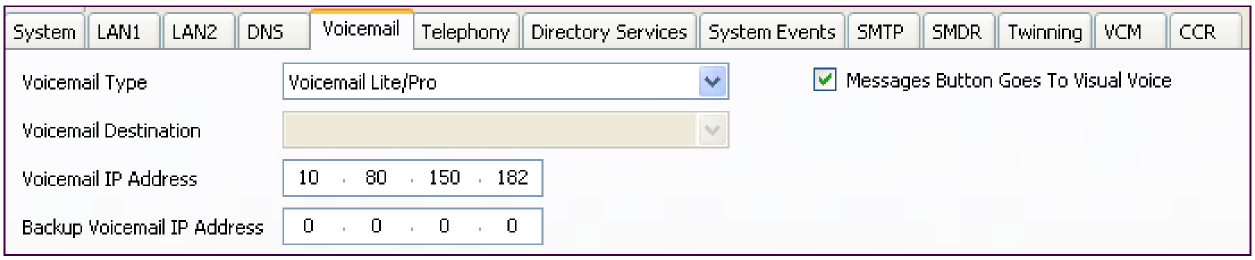
Select the **Network Topology** tab as shown in the following screen. In the sample configuration, the default settings were used and the **Use Network Topology Info** in the **SIP Line** was set to "None" in Section 5.4.2. The **Binding Refresh Time (seconds)** can still be used to lower the SIP OPTIONS timing from the default of 300 seconds. During the testing, the Binding Refresh Time was varied (e.g., 60 seconds, 120 seconds to test SIP OPTIONS timing).



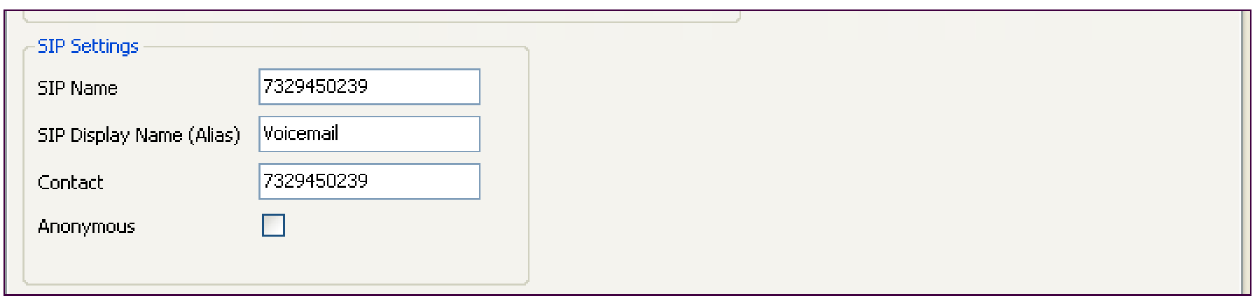
Since **SIP Registrar Enable** was unchecked on the VOIP tab, the SIP Registrar Tab is not present on LAN2.

1.6.4. 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 10.80.150.182, the IP Address of the PC running the Voicemail Pro software, as shown in **Figure 1**.



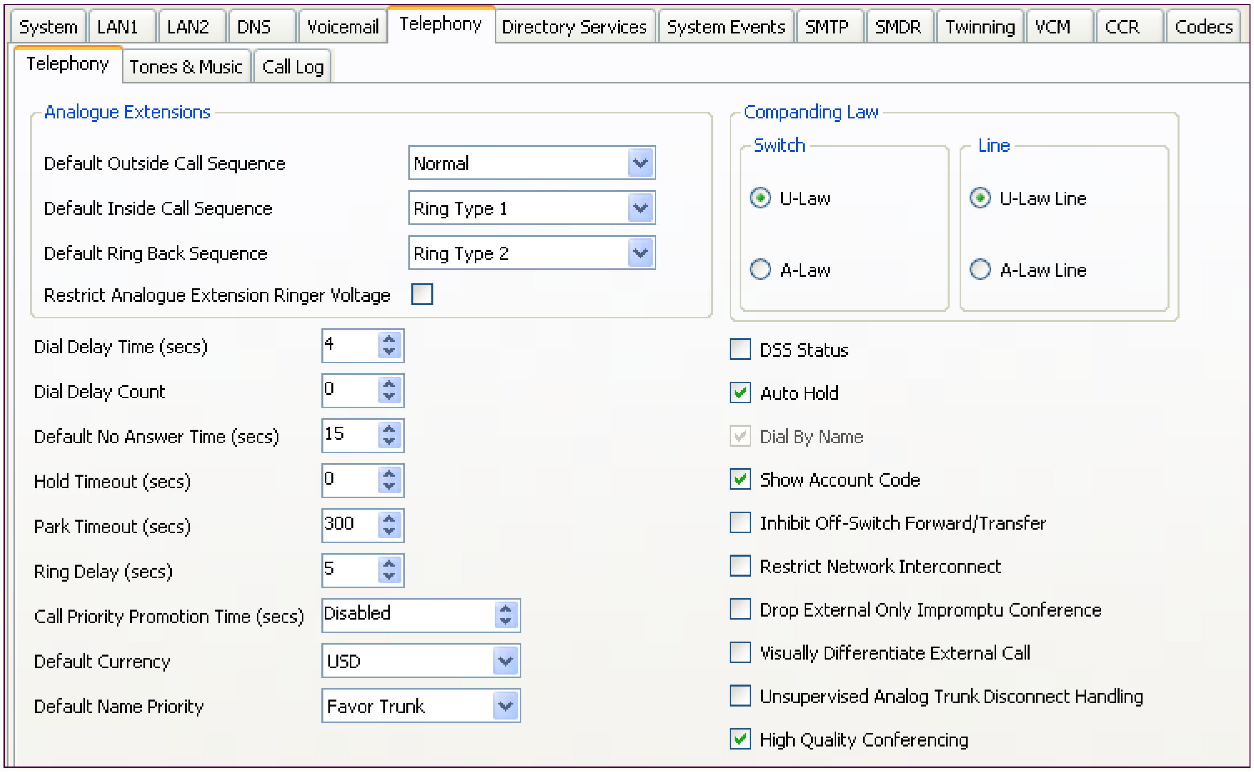
As described in [VZBIPT-IPO81], the “Callback” application of Avaya Voicemail Pro was used to allow Voicemail Pro to call out via the SIP Line to Verizon Business IP Trunk service when a message is left in a voice mailbox.



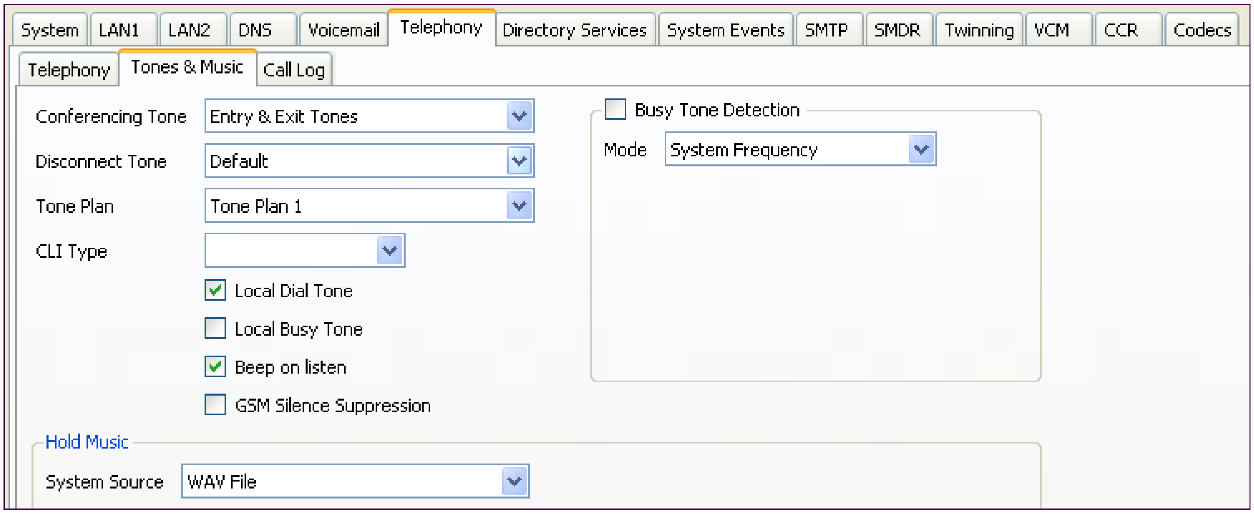
1.6.5. System Telephony Configuration

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 IPCC service, and be forwarded or transferred back to the PSTN with the outbound leg of the call using the Verizon Business 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.

The **Default Name Priority** is a new field in IP Office Release 8 and can be relevant to SIP Trunking. The option to “Favor Trunk” or “Favor Directory” can be set system-wide using the screen below, or set uniquely for each line. With the option to "Favor Directory", IP Office will prefer to display names found in a personal or system directory over those arriving from the far-end, if there is a directory match to the caller ID. This capability will be illustrated further in the context of the SIP Line to Verizon. A user's personal directory example is shown in Section 5.5.1.

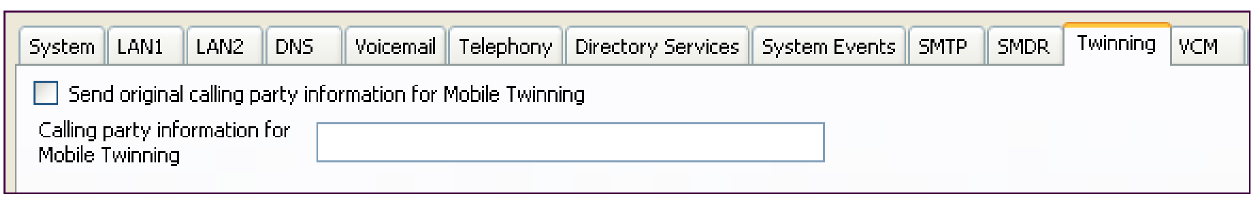


To view or change settings associated with tones or music, select the **Telephony** tab and **Tones & Music** sub-tab as shown in the following screen. The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. In the sample configuration, music on hold was provided via a WAV file from IP Office. For conferences, entry tone and exit tones are provided.



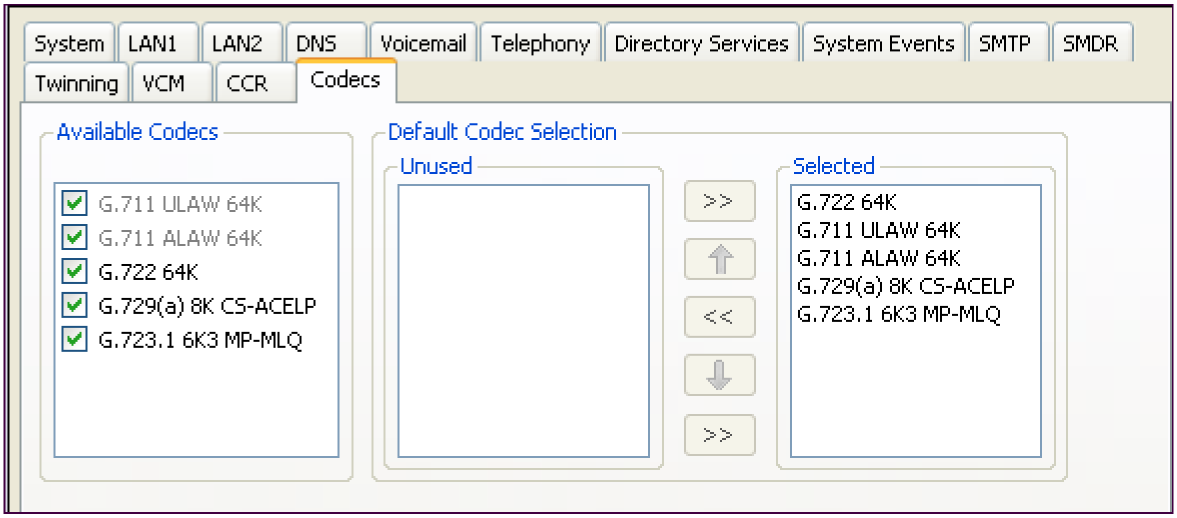
1.6.6. System Twinning Configuration

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-IPO81]), the true identity of the caller can be presented to the twinning destination (e.g., a user's mobile phone) 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.



1.6.7. System Codecs Configuration

The **System → Codecs** tab was introduced in IP Office Release 8. 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.



1.7. SIP Line

This section shows the configuration screens for the SIP Line in IP Office Release 8.1. The Appendix in Section 11 contains an example SIP Trunk template file that was generated from the SIP Line configured in this section.

To add a new SIP Line, right click on **Line** in the Navigation pane, and select **New → SIP Line**. A new Line Number will be assigned automatically. To edit an existing SIP Line, click **Line** in the Navigation pane, and the SIP Line to be configured in the Group pane.

1.7.1. SIP Line - SIP Line Tab

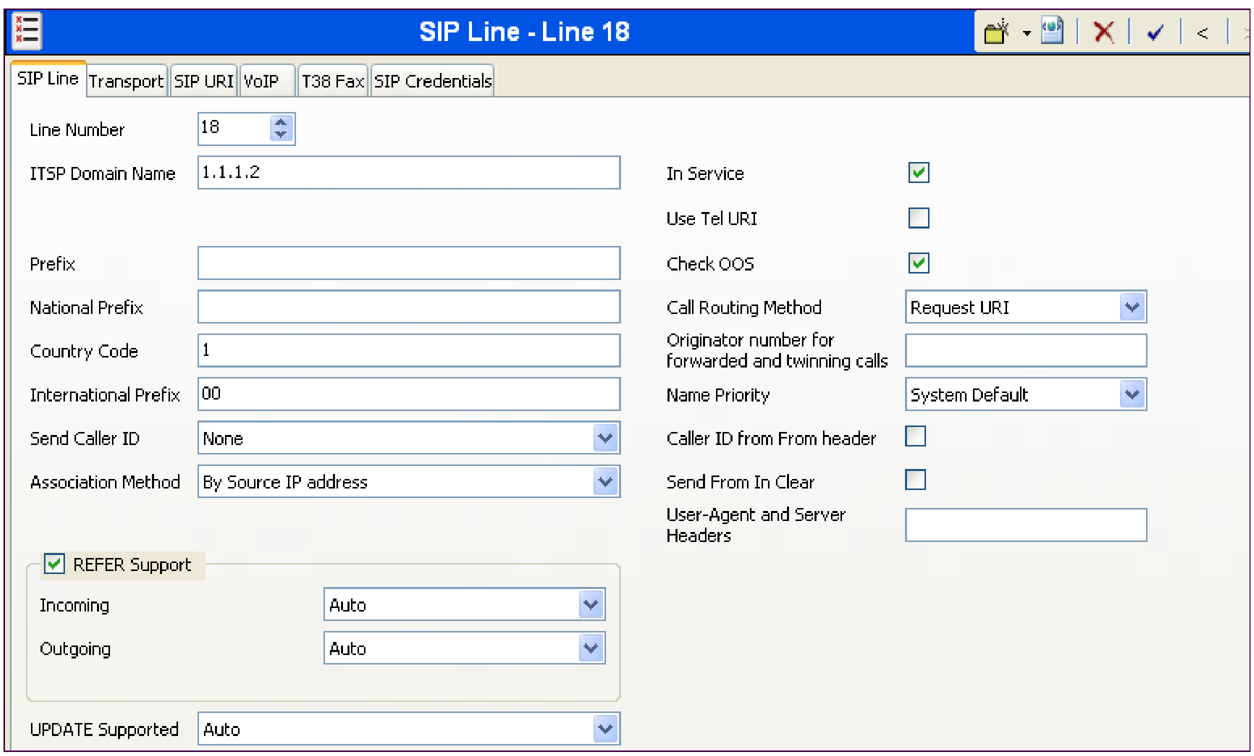
The **SIP Line** tab in the Details pane is shown below for Line Number 18, used for the Verizon Business IP Contact Center service. The **ITSP Domain Name** is configured to the IP Office LAN2 address (1.1.1.2). By default, the **In Service** and **Check OOS** boxes are checked.

The **Call Routing Method** can retain the default "Request URI" setting, or may be changed to "To Header", to match Incoming Call Routes based on the contents of the "To Header". In the sample configuration, the default "Request URI" setting was used. As can be observed in the sample INVITE header contents in **Figure 1** and Section 7.2, the Request-URI and the To header do not necessarily contain the same number. In the tested configuration, the Request-URI contained the toll-free number, and the "To" header contained 1 followed by the toll-free number.

In the sample configuration, the IP Office **Country Code** was 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 “0013035387006”. 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.

The area of the screen entitled **REFER Support** was introduced in IP Office Release 6.1. In the following screen, the default automatic determination of REFER support is shown. Alternatively, the default can be overridden with “Never” to explicitly disable use of REFER, or “Always” to explicitly enable use of REFER. The **Association Method** parameter was introduced in IP Office Release 7.0, and the screen below shows the default value, which is sufficient in the sample configuration. The various alternatives for the **Association Method** may be useful when multiple SIP Trunks with different SIP domains resolve to a single IP Address. The default option associates incoming requests with SIP Lines by comparing the source IP Address and port of the incoming message against the configured far-end of the SIP Line.

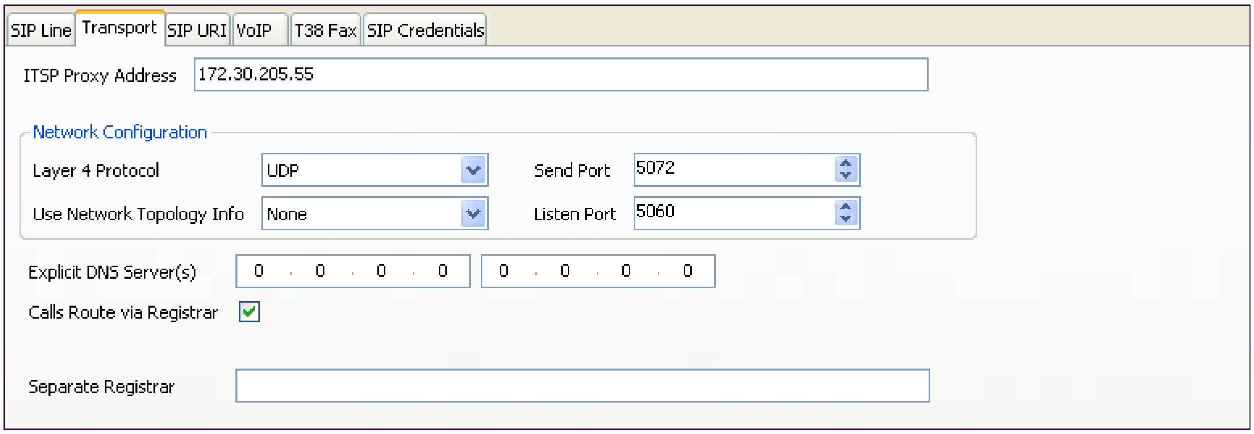
The **Name Priority** parameter was introduced in IP Office Release 8.0. The **Name Priority** parameter can retain the default “System Default” setting, or can be configured to “Favor Trunk” or “Favor Directory” as shown in the sample screen below. “System Default” will use the setting displayed on the System → Telephony→ Telephony Tab. The “Favor Directory” setting enables IP Office to match the caller’s telephone number against available system or personal directories, and display the name obtained from a match in the directory, if any, rather than name information received in the SIP signaling from Verizon. Click **OK** (not shown).



1.7.2. SIP Line - Transport Tab

Select the **Transport** tab. This tab was introduced in Release 6.1. Some information configured in this tab had been under the **SIP Line** tab in Release 6.0.

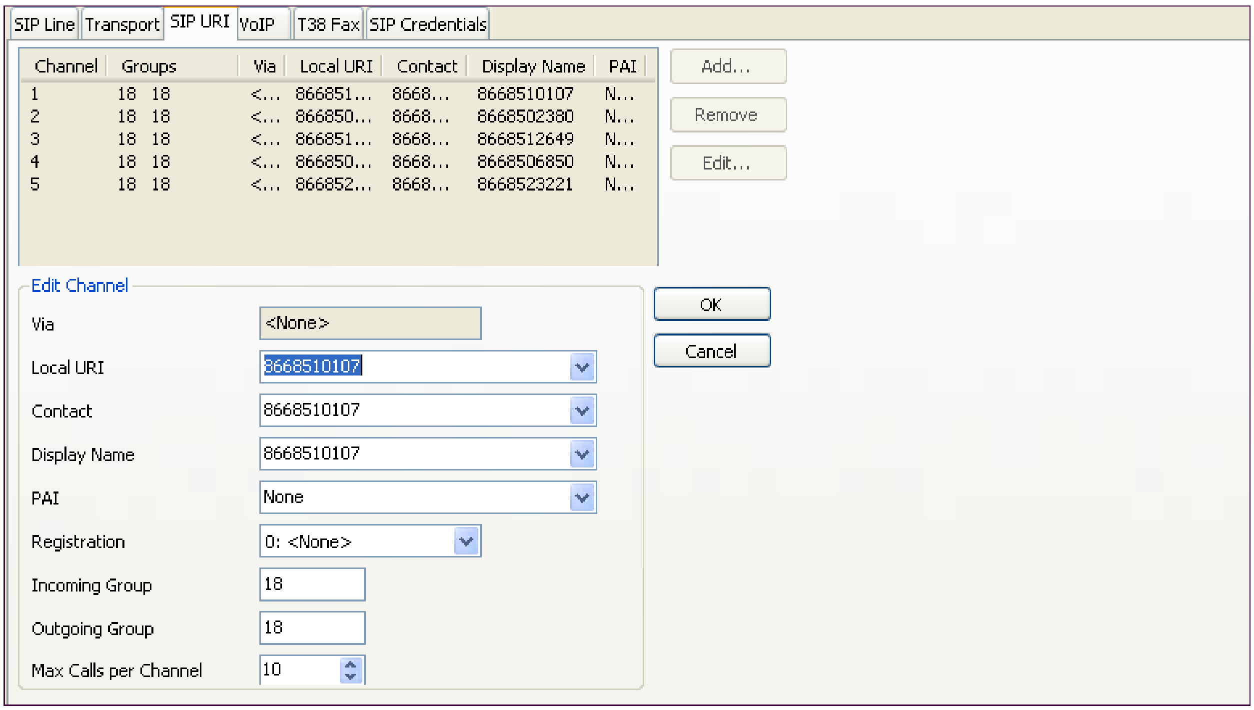
The **ITSP Proxy Address** is set to the IP Address provided by Verizon Business. As shown in **Figure 1**, this IP Address is 172.30.205.55. In the **Network Configuration** area, UDP is selected as the **Layer 4 Protocol**, and the **Send Port** is set to the port number provided by Verizon Business.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



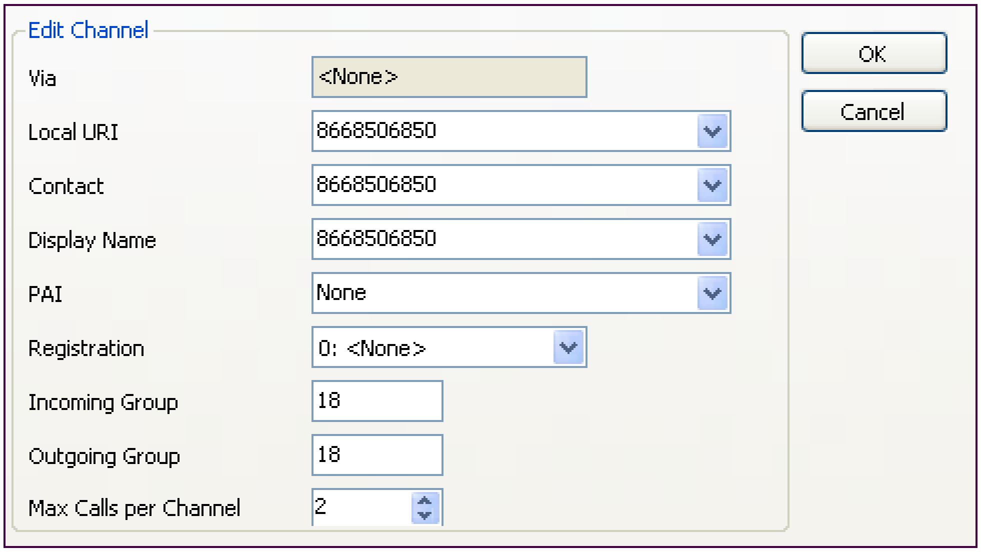
1.7.3. SIP Line - SIP URI Tab

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 was introduced in IP Office Release 6.1, and the value "None" is shown selected from the drop-down menu. The **Registration** parameter is set to the default "0: <None>" since Verizon Business IP Contact Center service does not require registration. The **Incoming Group** parameter, set here to 18, will be referenced when configuring Incoming Call Routes to map inbound SIP trunk calls to IP Office destinations in Section 5.7. The **Outgoing Group** parameter, also set here to 18, 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**.

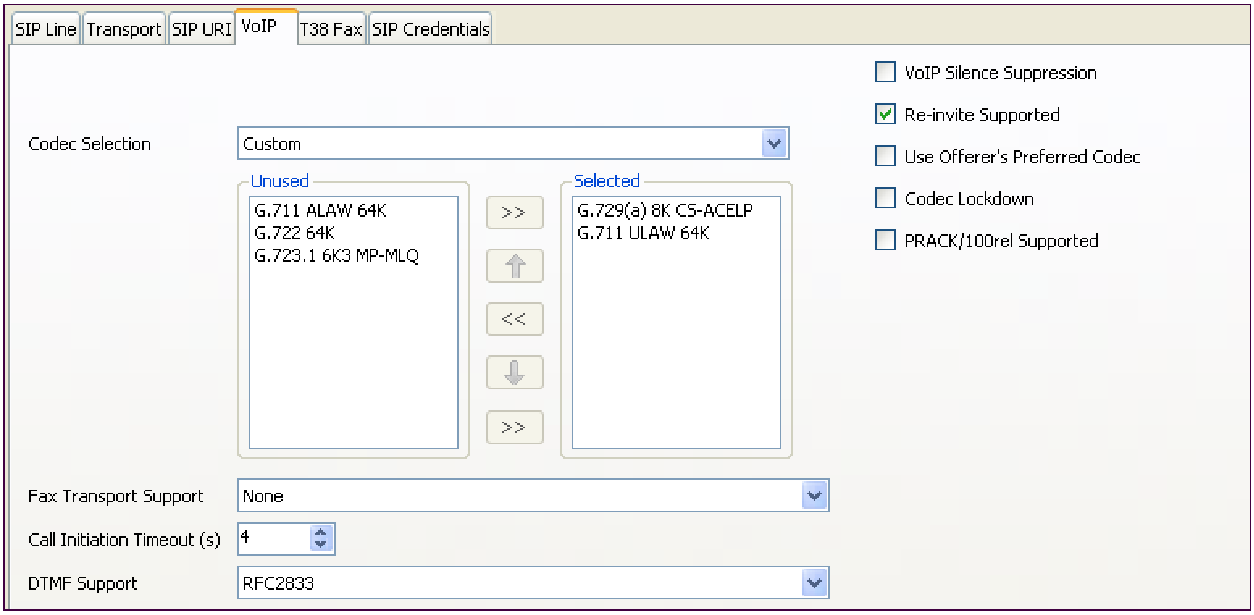


IP Office allows the number of simultaneous calls to a specific SIP URI to be managed using the **Max Calls per Channel** field. In the following screen, note that the **Max Calls per Channel** field has been changed from the default 10 to 2. With this configuration, two simultaneous calls to the number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



1.7.4. SIP Line - VoIP Tab

Select the **VoIP** tab. In the sample configuration, the **Codec Selection** was configured using the "Custom" option, allowing an explicit ordered list of codecs to be specified, different from the system default (see Section 5.3.6). The arrow buttons can be used such that **G.729(a) 8K CS-ACELP** and **G.711 ULAW 64K** codecs are listed in the **Selected** column. This configures IP Office to support either G.729a or G.711MU for this SIP Line. The **DTMF Support** parameter can remain set to the default value "RFC2833". The **Re-invite Supported** parameter can be checked to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk. The **Use Offerer's Preferred Codec** parameter can be left at the default unchecked value, or may be checked. In the sample configuration, Verizon preferred the G.729A codec in SDP, while also allowing the G.711MU codec. The IP Office configuration shown below matches these Verizon preferences. In the course of testing, the IP Office configuration was varied such that G.711MU was the preferred or only codec listed, and G.711MU calls were also successfully verified. The **PRACK/100rel Supported** parameter was introduced in IP Office Release 8, and should be left at the default unchecked value. Since the Verizon Business IP Contact Center service does not support fax, the **Fax Transport Support** parameter is set to "None", and the **T38 Fax** tab need not be visited. Since the Verizon Business IPCC service does not require registration, the **SIP Credentials** tab need not be visited. Click **OK** (not shown).

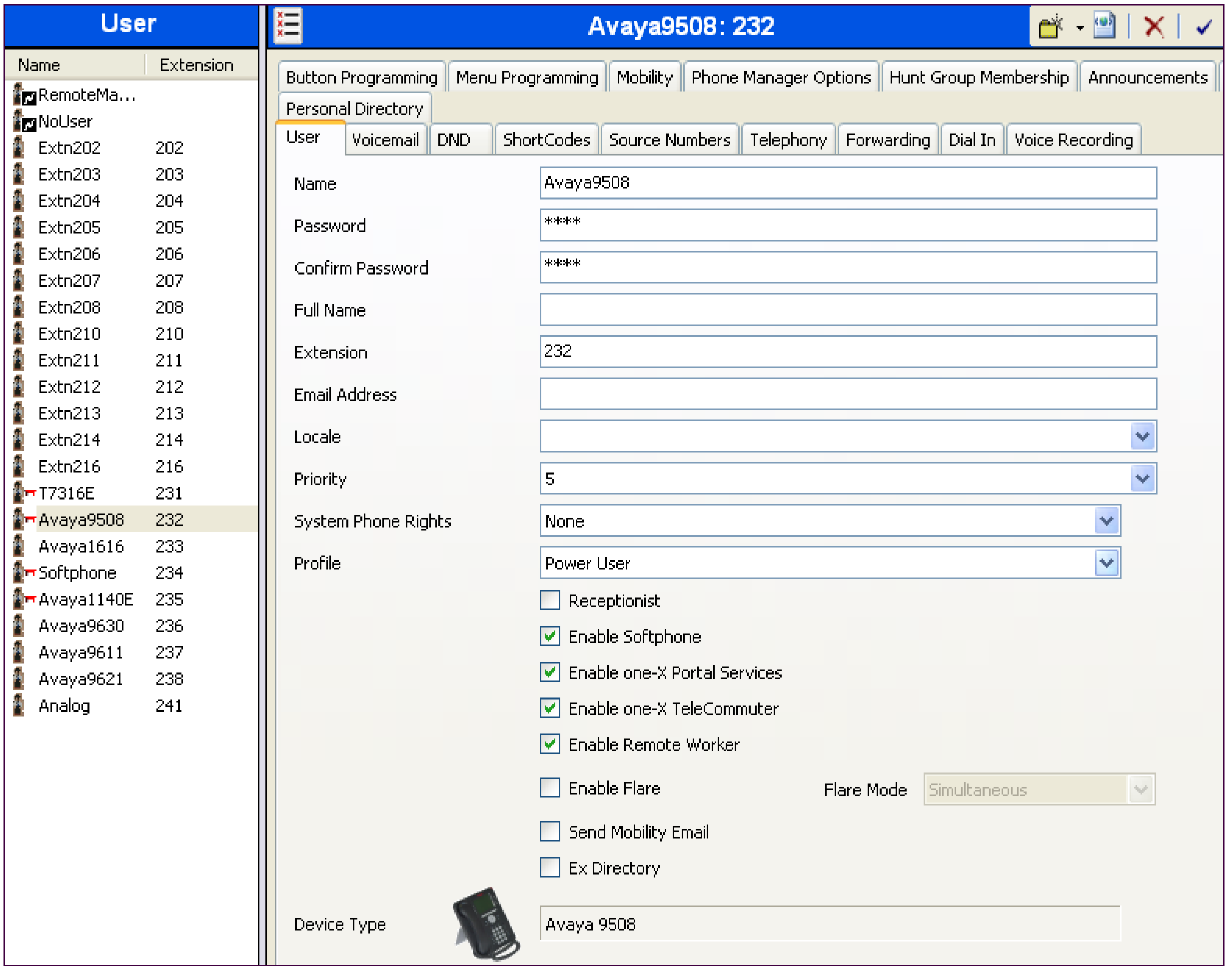


1.8. Users, Extensions, and Hunt Groups

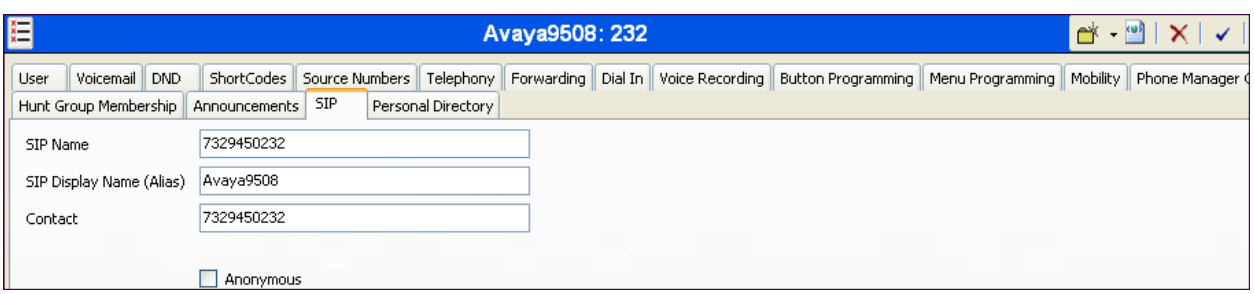
In this section, examples of IP Office Users, Extensions, and Hunt Groups will be illustrated. In the interests of brevity, not all users and extensions shown in **Figure 1** will be presented, since the configuration can be easily extrapolated to other users. To add a User, right click on **User** in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

1.8.1. Digital User 232

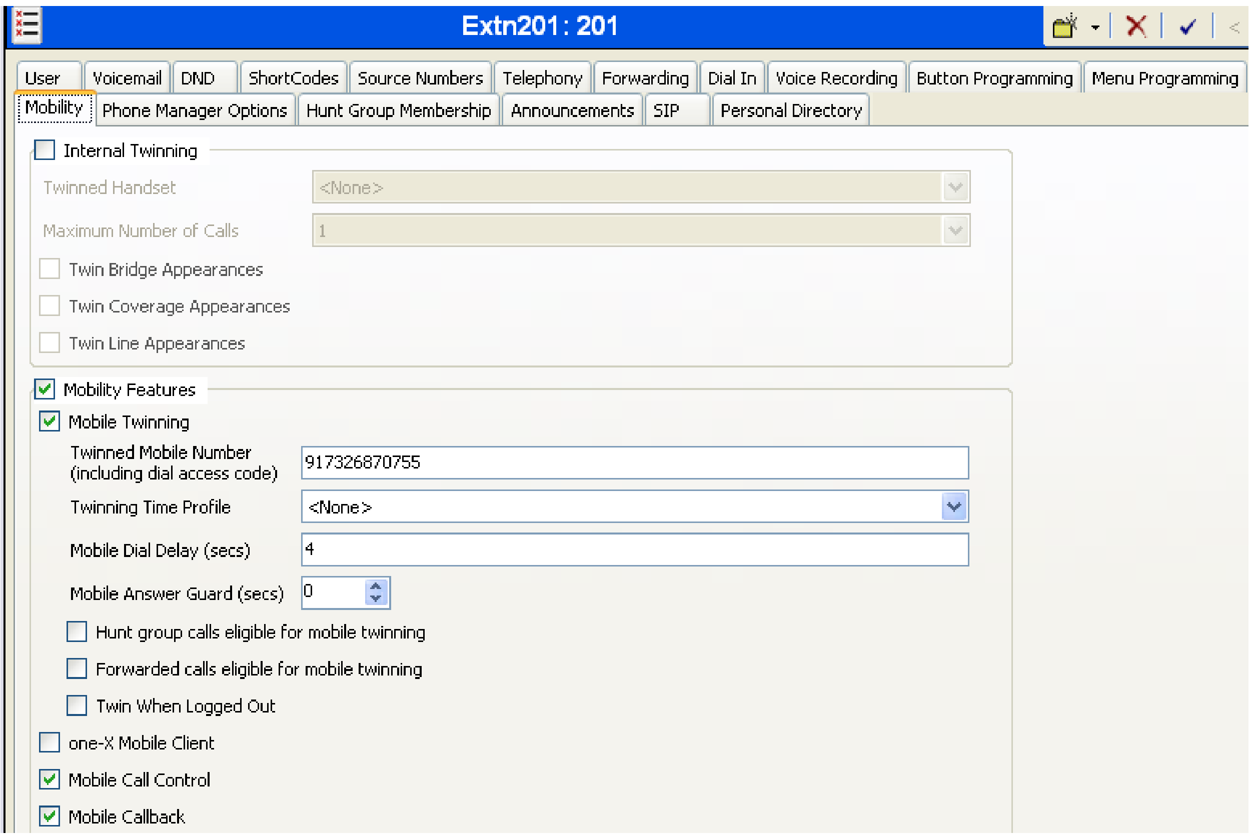
The following screen shows the **User** tab for User 232. As shown in **Figure 1**, this user corresponds to the Avaya Digital 9508.



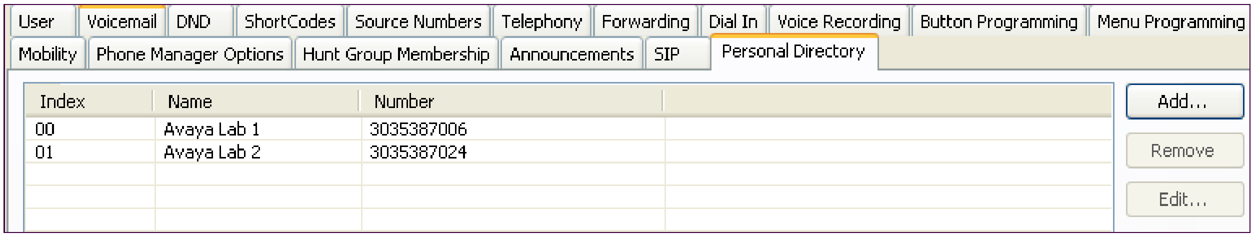
The following screen shows the **SIP** tab for User 232. In the sample configuration, the **SIP Name** and **Contact** parameters are configured with a Verizon Business IP Trunk DID number for the user, 7329450232. As shown in [VZBIPT-IPO81], these parameters configure the user part of the SIP URI in 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.



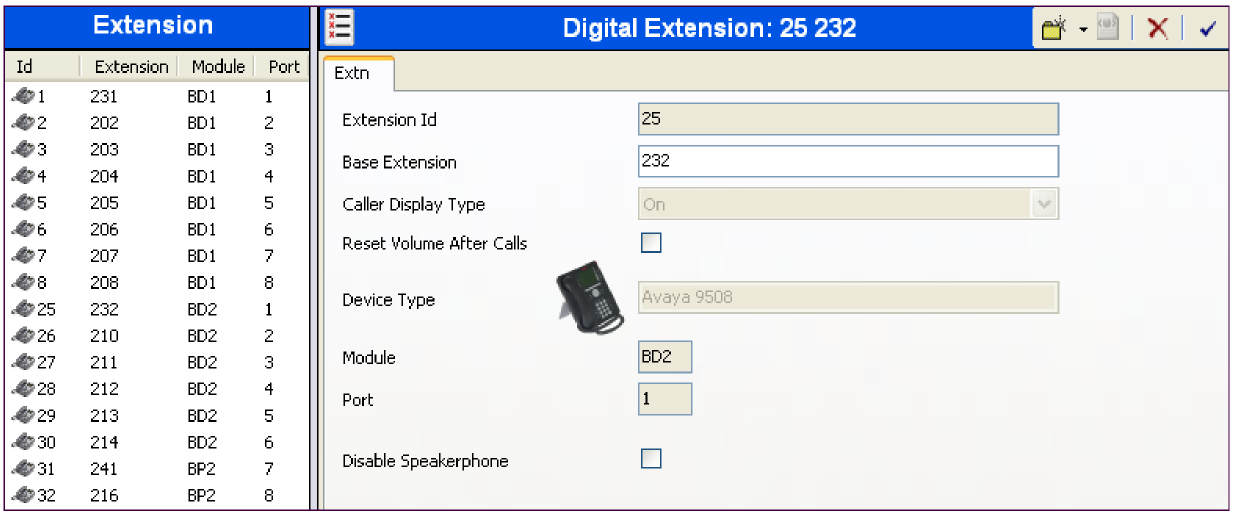
From **Figure 1**, note that user 232 will use the Mobile Twinning feature. The following screen shows the **Mobility** tab for User 232. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case 913035387024. 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-IPO81].



As described in Section 5.3.4, names can be entered in directories to allow IP Office to match the caller ID for incoming calls and display the names from the directory. The following screen shows the **Personal Directory** tab for user 232. With the configuration shown below and on the SIP Line in Section 5.4.1 (where "Favor Directory" is selected), if user 232 receives an inbound Verizon IP Toll Free call from the telephone number 13035387006, the phone will display the name "Avaya Lab 1" (along with the number).

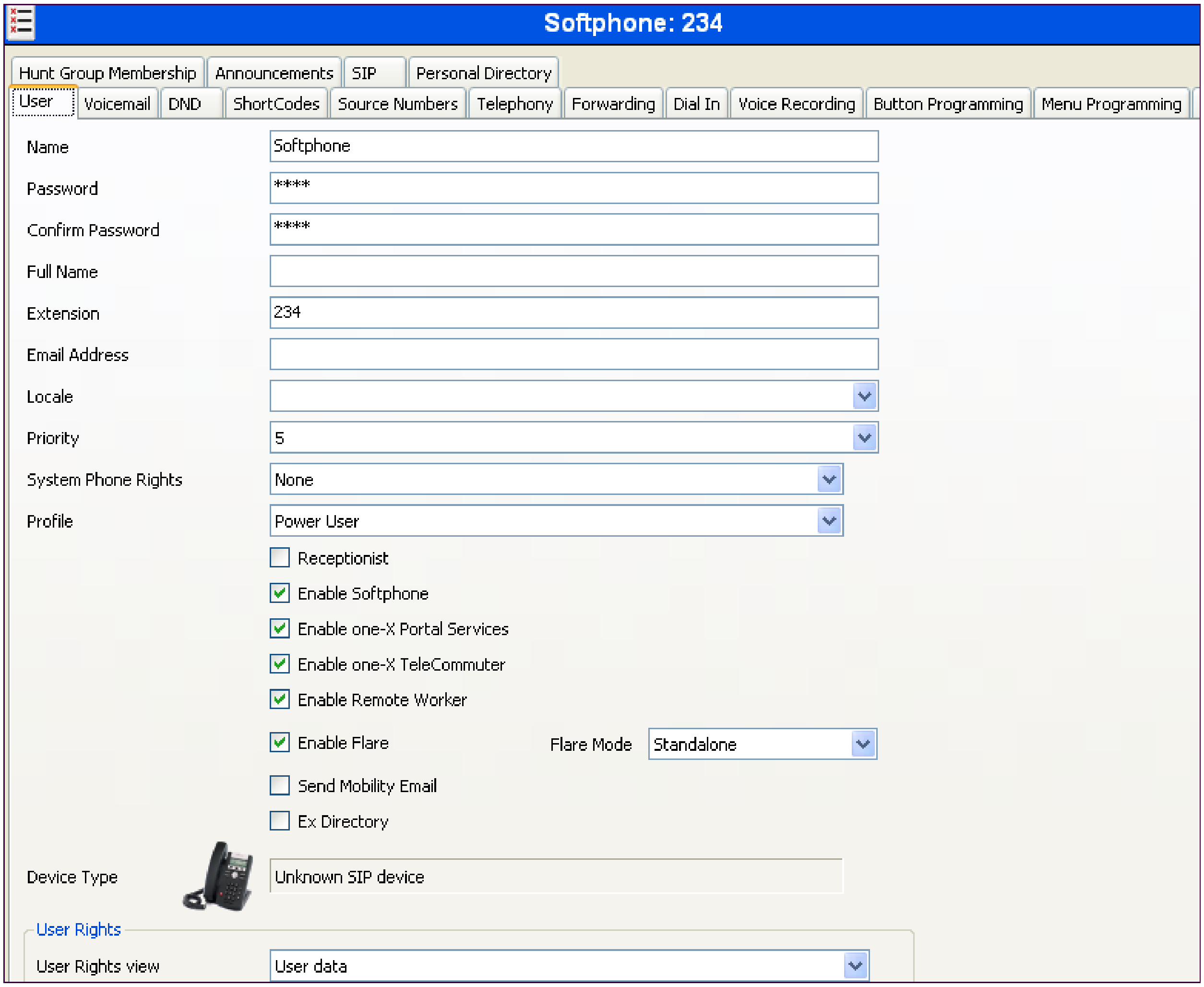


The following screen shows the Extension information for this user. To view, select **Extension** from the Navigation pane, and the appropriate extension from the Group pane.



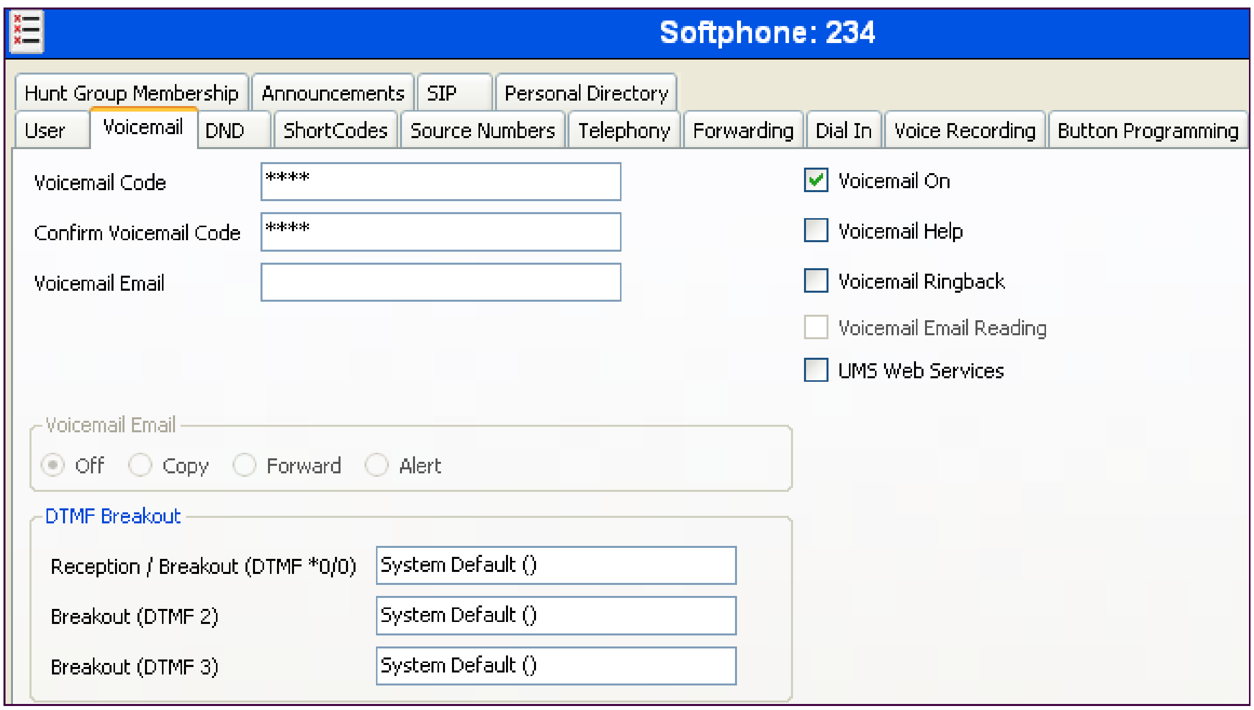
1.8.2. Avaya Flare Experience User 234 with IP Office Softphone Privileges

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

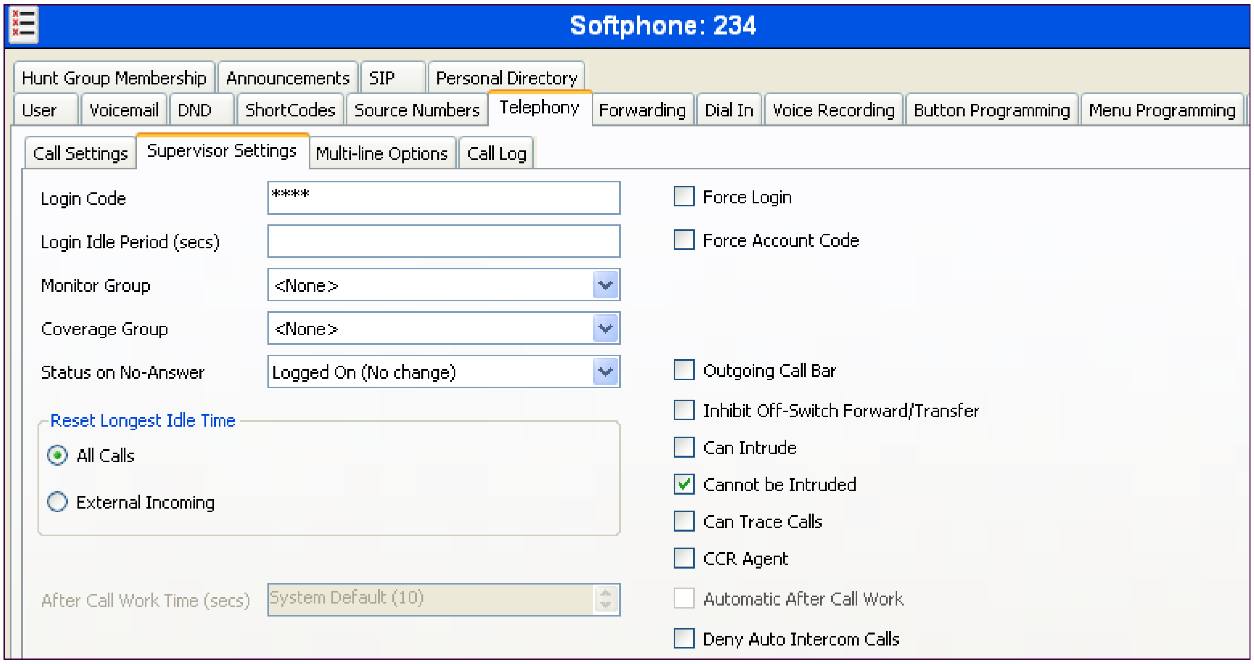


Like the user with extension 232, the **SIP** tab for the user with extension 234 is configured with a **SIP Name** and **Contact** specifying the user’s Verizon Business DID number using the Verizon Business IP Trunk service, as detailed in [VZBIPT-IPO81].

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 IP Contact Center service to this user were redirected to Voicemail Pro after no answer. Voicemail messages were recorded and retrieved successfully. Voice mail navigation and retrieval were performed locally and from PSTN telephones using the Verizon Business IPCC toll-free numbers, to test DTMF using RFC 2833, and to test assignment of a Verizon-provided toll free number to the "Voicemail Collect" feature (e.g., via the *17 short code shown in Section 5.6).

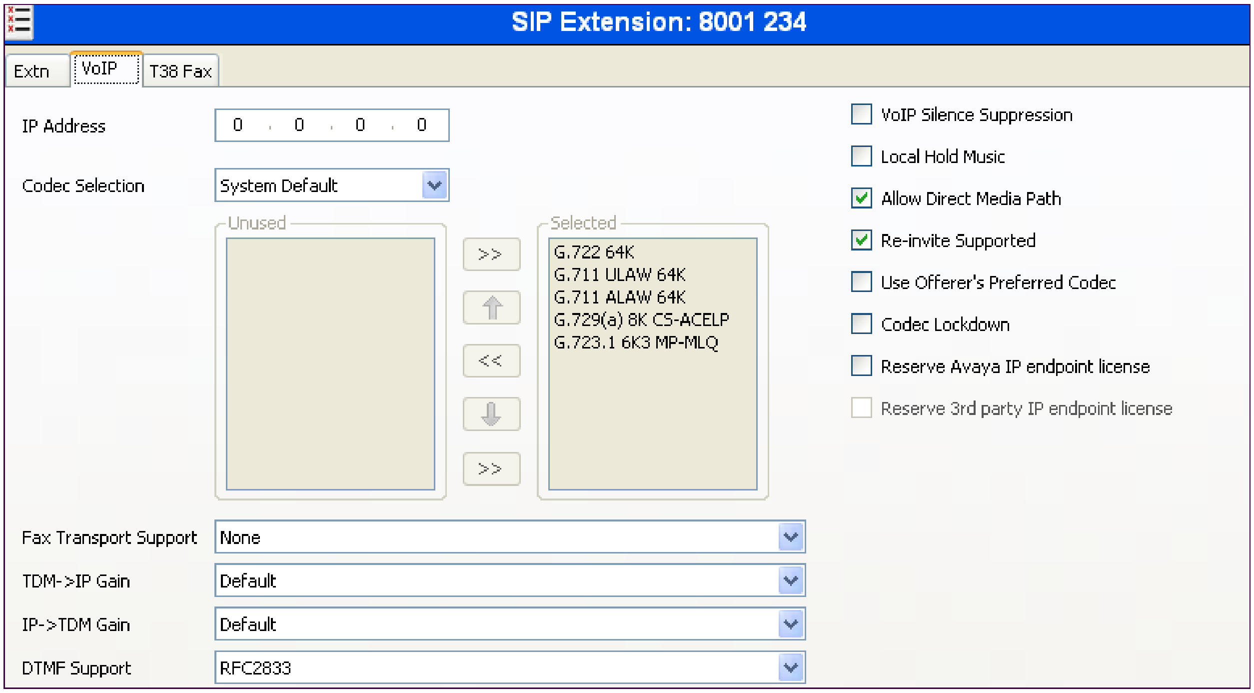


Select the **Telephony** tab. Then select the **Supervisor Settings** tab as shown below. The **Login Code** will be used by the Flare Experience and IP Office Softphone user as the login password.



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

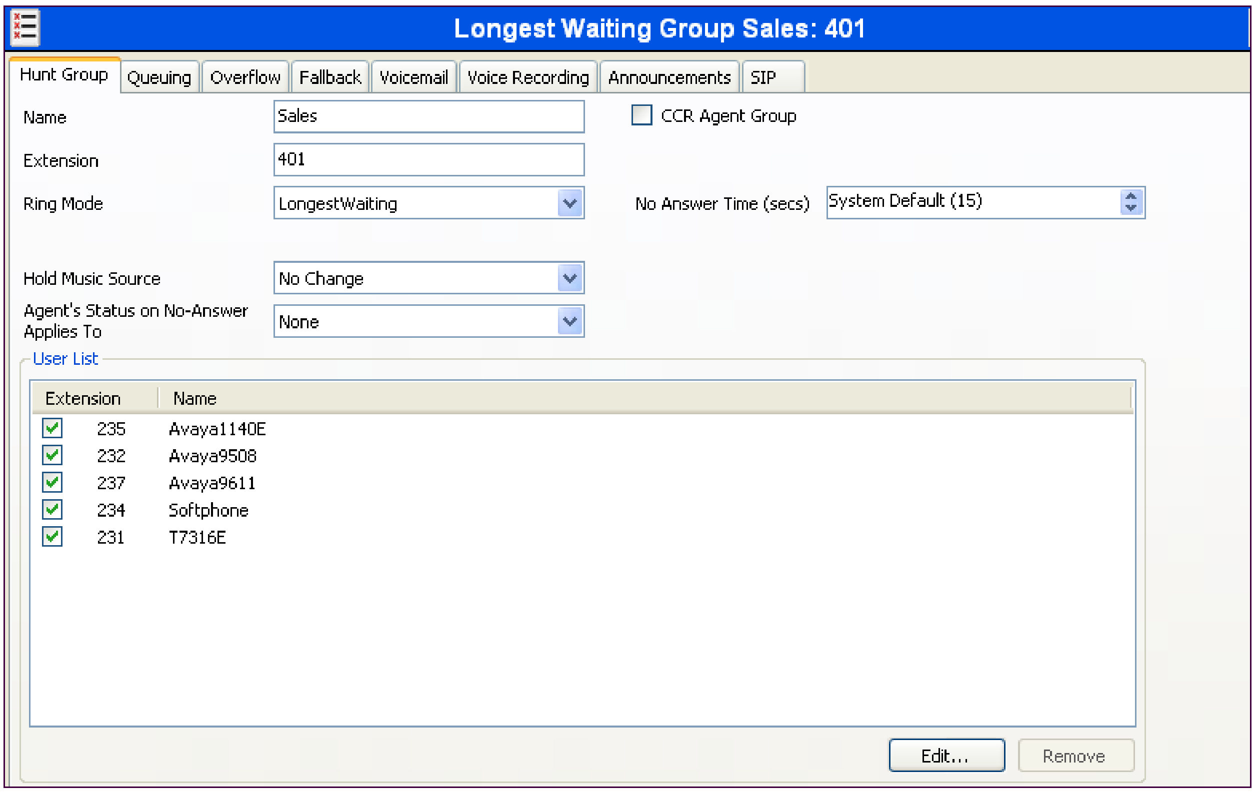
The following screen shows the Extension information for this user, simply to illustrate the **VoIP** tab available for a SIP Telephone. To view, select **Extension** from the Navigation pane, and the appropriate extension from the Group pane. Select **VoIP** in the Details pane. The new **Codec Selection** parameter may retain the default setting "System Default" to follow the system configuration shown in Section 5.3.6. 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.


SIP Extension: 8001 234
Extn
VoIP
T38 Fax
IP Address
0 . 0 . 0 . 0
Codec Selection
System Default
Unused
Selected
G.722 64K
G.711 ULAW 64K
G.711 ALAW 64K
G.729(a) 8K CS-ACELP
G.723.1 6K3 MP-MLQ
VoIP Silence Suppression
Local Hold Music
Allow Direct Media Path
Re-invite Supported
Use Offerer's Preferred Codec
Codec Lockdown
Reserve Avaya IP endpoint license
Reserve 3rd party IP endpoint license
Fax Transport Support
None
TDM->IP Gain
Default
IP->TDM Gain
Default
DTMF Support
RFC2833

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

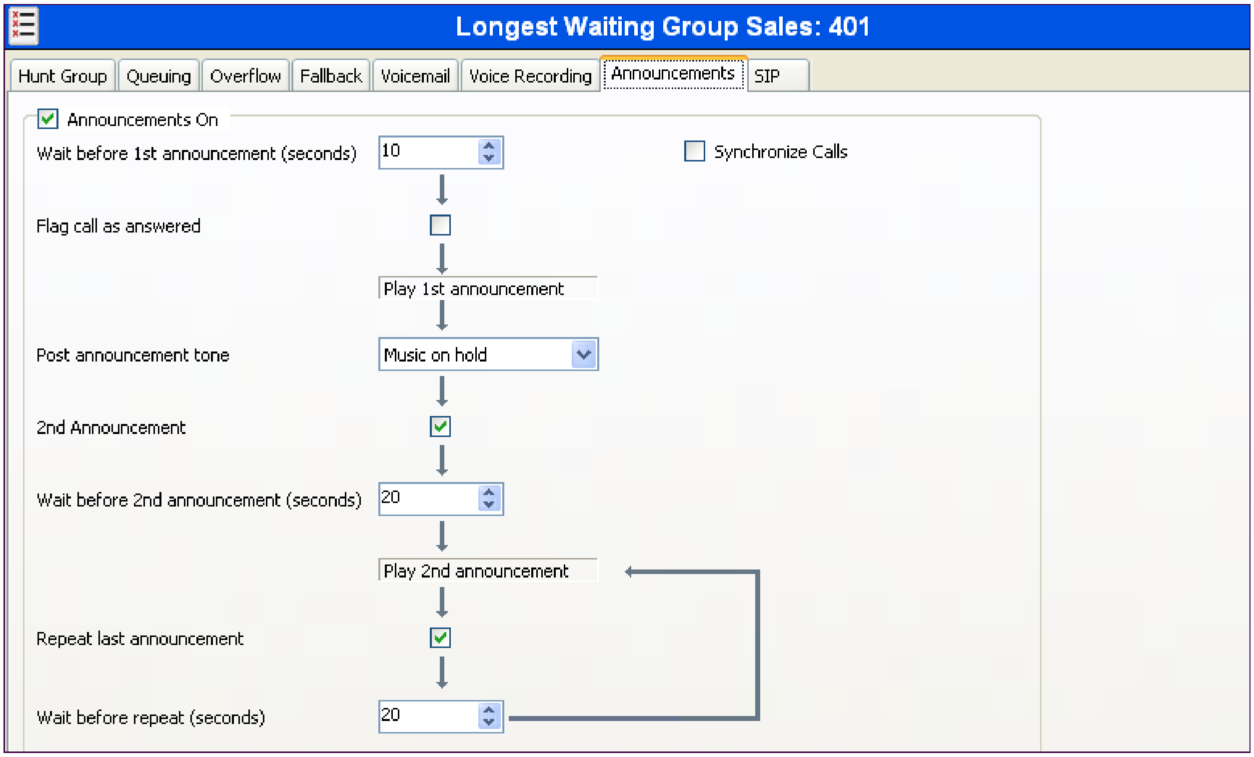
The following screen shows the **Hunt Group** tab for a hunt group with Extension 401 and **Name** "Sales". This hunt group was configured to contain various telephones from **Figure 1**. The **Ring Mode** was set to "LongestWaiting" (i.e., "longest waiting", most idle user receives next call). Click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



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

The following screen shows the **Announcements** tab for hunt group 220. In the sample configuration, when a call arrives when all members of the hunt group are busy on calls, the caller will first hear ring back tone. If a member of the hunt group does not become available after 10 seconds, the call will be answered by IP Office (i.e., 200 OK will be sent to Verizon), and the toll-free caller will hear a first announcement. Note that the **Flag call as answered** box is relevant for reporting applications, but does not change the fact that IP Office will answer the call when the first announcement is played. If the call is still not answered after the first announcement completes, the caller will hear music, a repeating second announcement, music, and so on until the call is answered by a member of the hunt group, or answered by voicemail for the hunt group (if configured). If a member of the hunt group becomes available while the caller is listening to ring back, music, or an announcement, the call is de-queued and delivered to the available member.

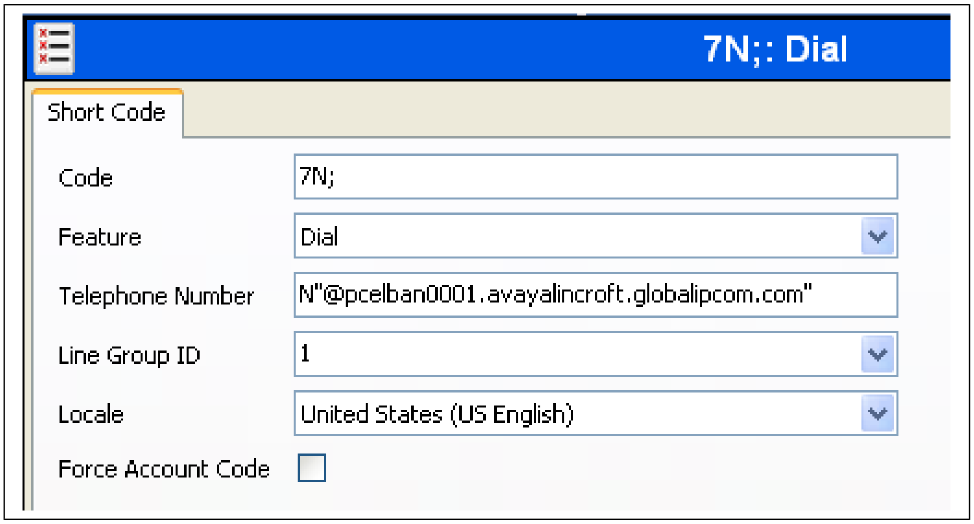
IP Office supports priority for queuing. For example, if low priority calls are waiting in queue, a higher priority call entering queue can be moved to the front of the queue and serviced before lower priority callers. For an inbound SIP trunk call, the priority can be specified on the Incoming Call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



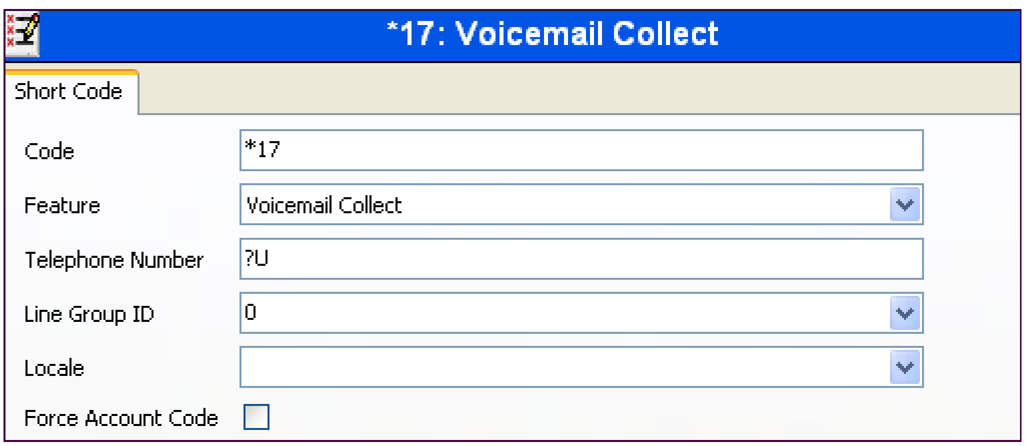
1.9. Short Codes

In this section, various examples of IP Office short codes will be illustrated. To add a short code, right click on **Short Code** in the Navigation pane, and select **New**. To edit an existing short code, click **Short Code** in the Navigation pane, and the short code to be configured in the Group pane.

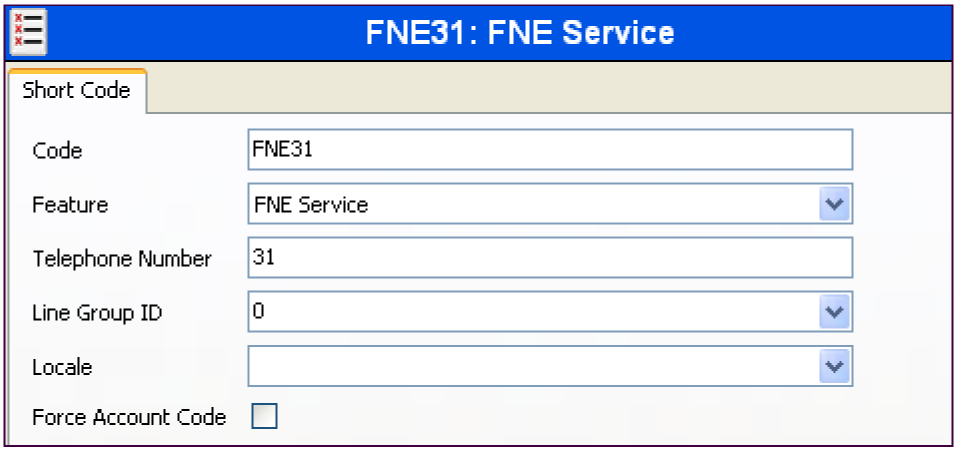
In the screen shown below, the short code "7N;" is repeated from references [VZBIPT-IPO81]. The **Code** parameter is set to "7N;". The **Feature** parameter is set to "Dial". The **Telephone Number** parameter is set to N"@Domain Name or IP Address of Verizon Business IP Trunk Service". The **Telephone 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 1, matching the number of the **Outgoing Group** configured on the **SIP URI** tab of SIP Line 17 to Verizon Business documented in [VZBIPT-IPO81]. 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-IPO81].



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 **1.10**. 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.

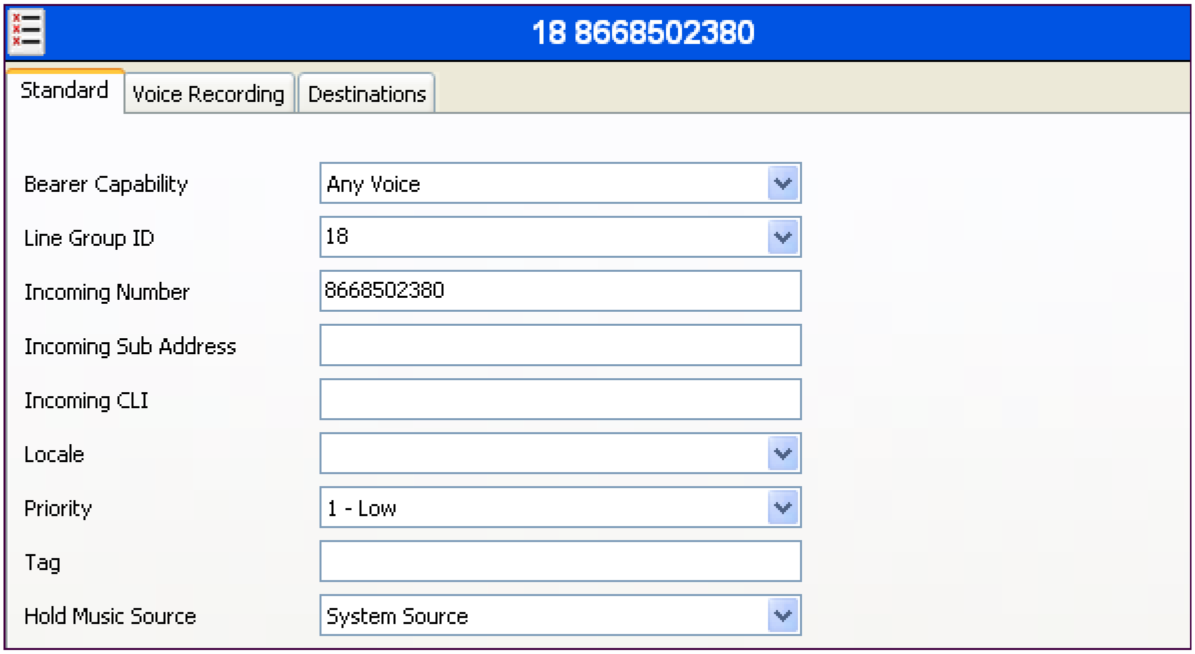


1.10. Incoming Call Routes

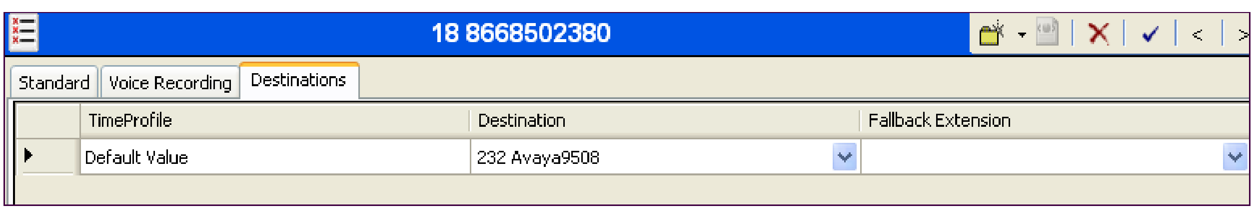
In this section, IP Office Incoming Call Routes are illustrated. Each Incoming Call Route will map a Verizon Business toll-free number to a destination user, group, or function on IP Office. In some cases, the destination will be chosen based on the combination of the toll-free number and the caller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

1.10.1. Incoming Call Route to a Specific Telephone Extension

In the screen shown below, the incoming call route for **Incoming Number** "8668502380" is illustrated. The **Line Group Id** is 18, matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to the Verizon Business IPCC service, in Section 5.4.



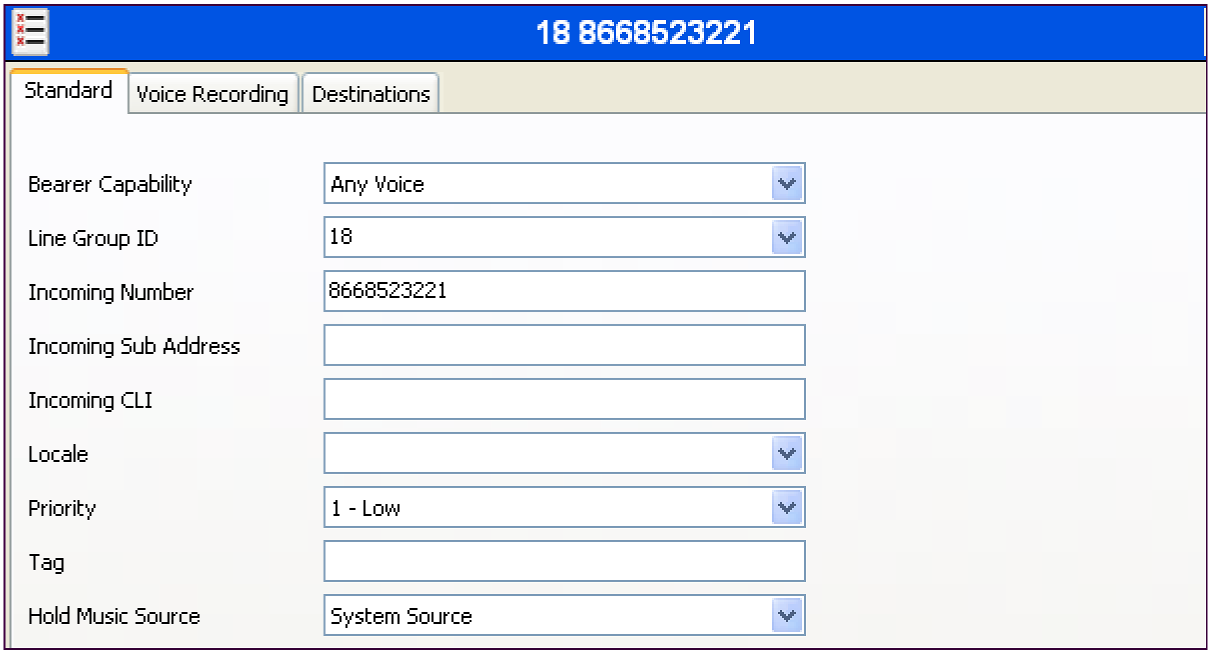
Select the **Destinations** tab. From the **Destination** drop-down, select an extension to receive the call when a PSTN user dials 8668502380. As shown in **Table 1**, 8668502380 is the number associated with IP Office user extension 232. (The **Destination** was changed in the course of testing to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



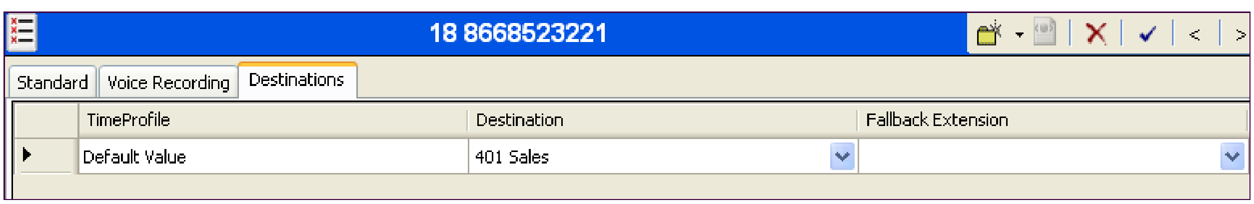
Incoming Call Routes for other direct mappings of toll-free numbers to IP Office users are not presented here, but are configured in the same fashion.

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

In the screen shown below, an incoming call route for **Incoming Number** "8668523221" is illustrated. The **Line Group Id** is 18, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in Section 5.4.



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

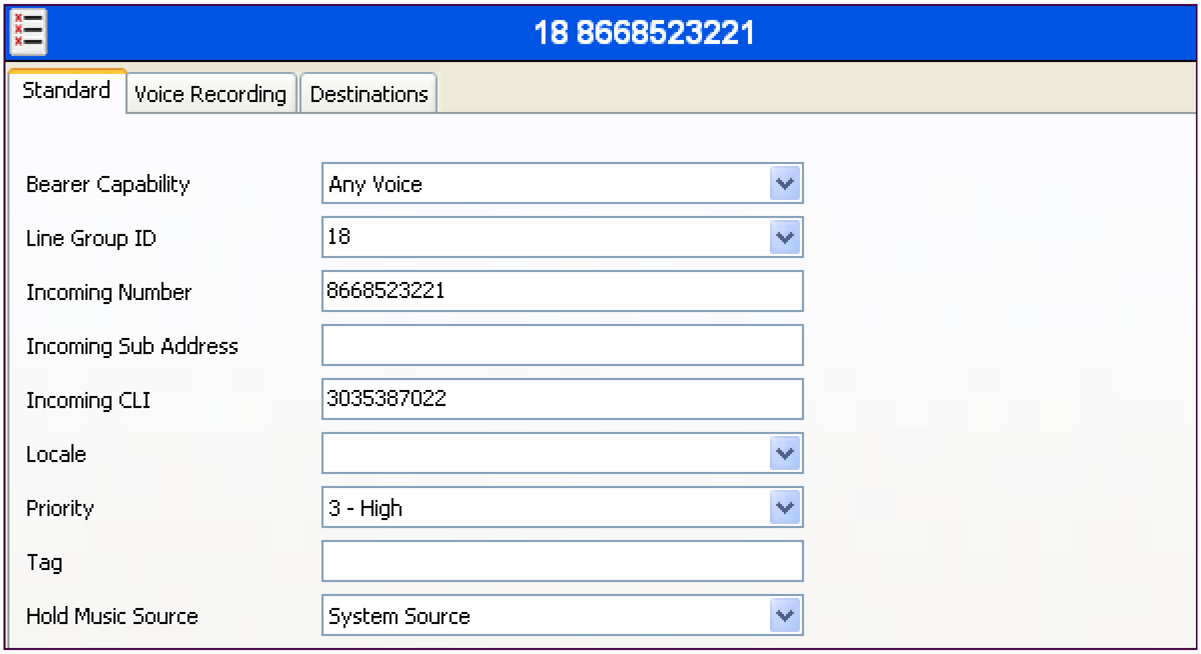


1.10.3. Incoming Call Routes Based on Calling Party Number

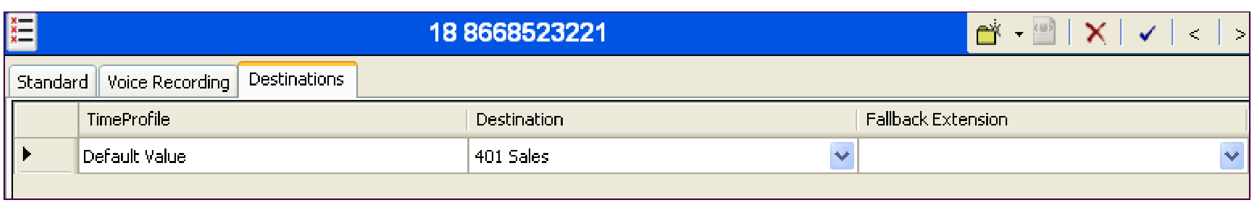
This section presents simple examples showing that IP Office can use the calling party number to distinguish call priority or call destination, for calls to the same toll-free number. While the matching shown here is based on the full calling number, partial matching is also possible (e.g., to match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

In the screen shown below, the incoming call route for **Incoming Number** "8668523221" and **Incoming CLI** "3035387022" is illustrated. The **Line Group Id** is 18, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in Section 5.4.

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

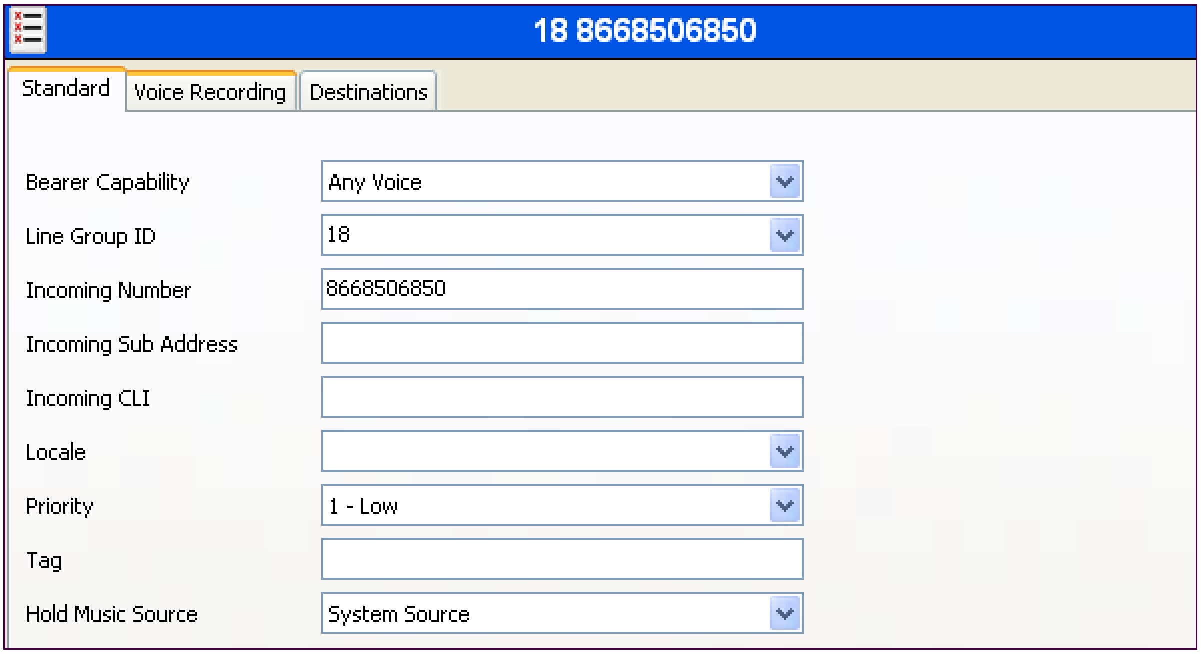


Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when PSTN user 13035387022 dials 8668523221. In this case, the **Destination** is also the hunt group "401 Sales", but since high priority has been configured via the **Standard** tab, incoming calls from this caller will move to the front of the queue, and be serviced before calls waiting in queue from other non-priority callers.



1.10.4. Incoming Call Routes to Various IP Office Features

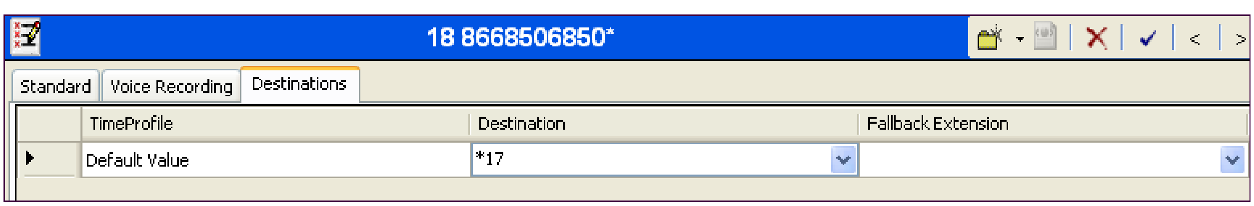
In the sample configuration, the incoming call route for **Incoming Number** "8668506850" was varied to test different destination features, such as Voice Mail, Mobile Call Control, and Conference Bridge, as shown in **Table 1** in **Section 3**. The screen showing the **Standard** tab for this toll-free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

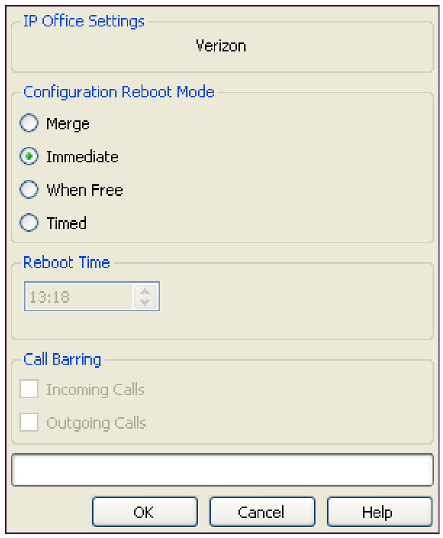
An example screen showing the short code configured for Voicemail Collect is shown below.



1.11. Saving Configuration Changes to IP Office

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

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



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.

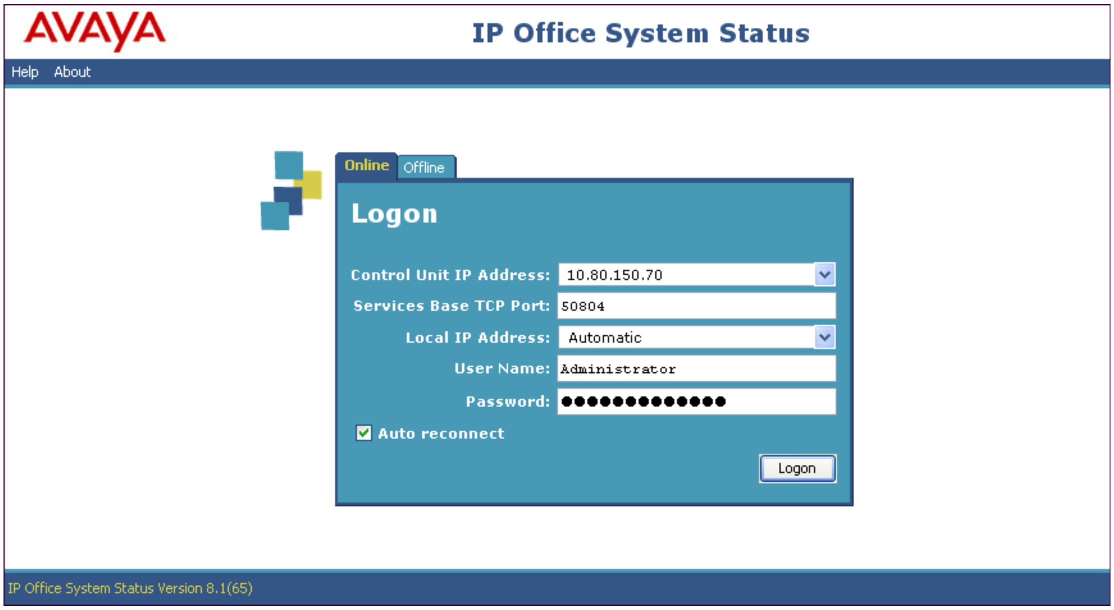
Verification

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

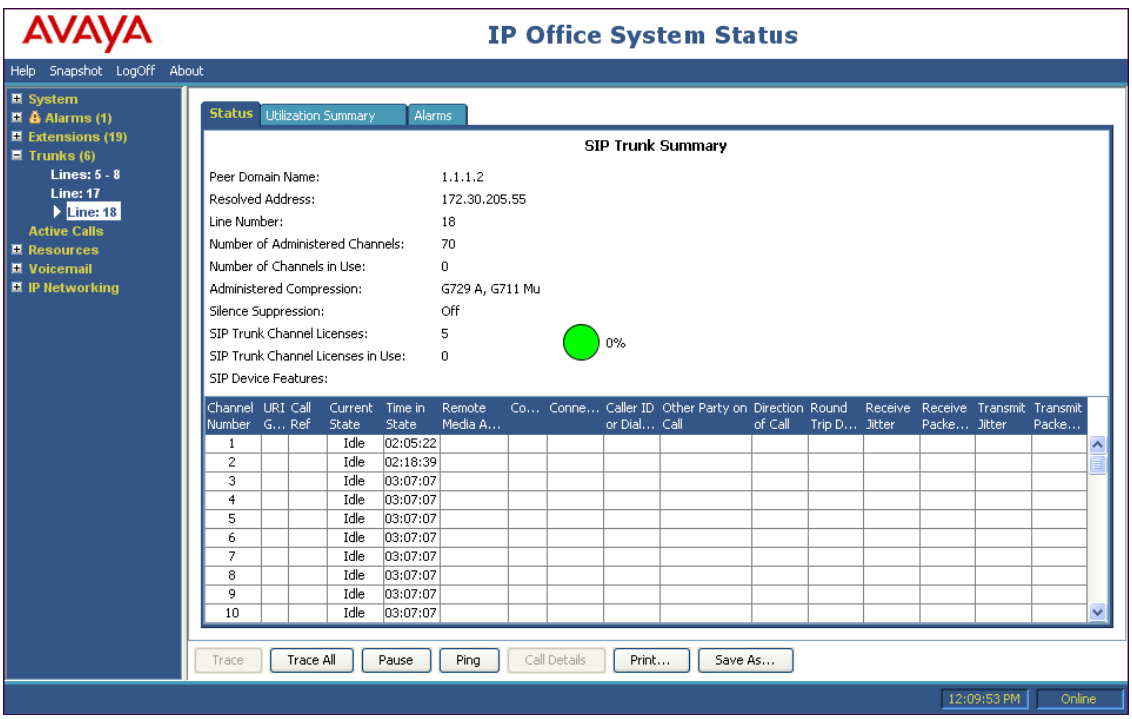
1.11.1. System Status

The System Status application is used to monitor and troubleshoot IP Office. Use the System Status application to verify the state of the SIP trunk. System Status can be accessed from **Start → Programs → IP Office → System Status**. Or by opening an Internet browser and type the URL: http://*ipaddress* where *ipaddress* is the IP address of the Avaya IP Office LAN1 interface. See reference [4] for more information. Click on **System Status** to launch the application.

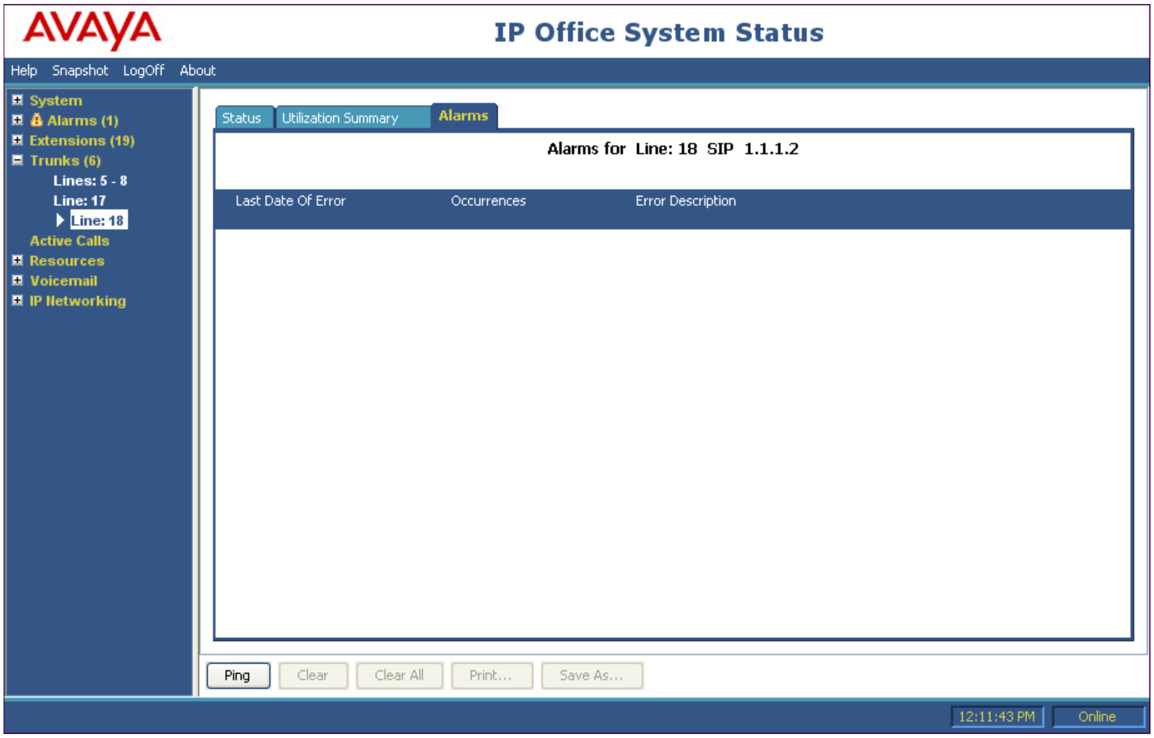
The following screen shows an example **Logon** screen. Enter the IP Office IP address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.



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.



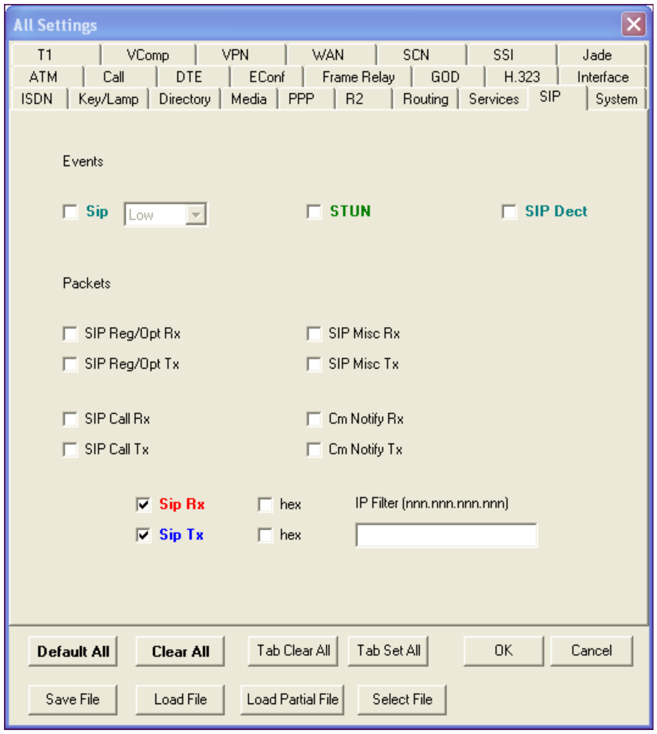
Select the **Alarms** tab and verify that no alarms are active on the SIP line.



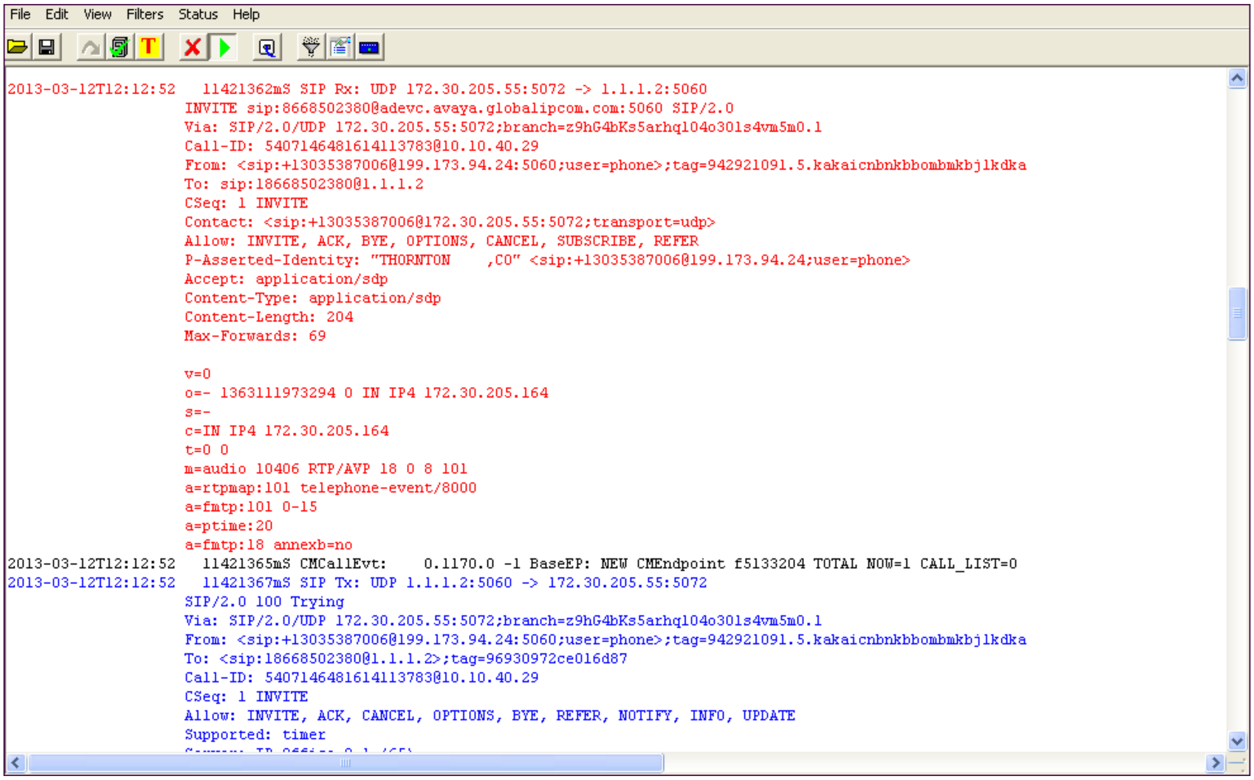
1.11.2. System Monitor

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**. The application allows the monitored information to be customized. To customize, select the button that is third from the right in the screen below, or select **Filters →Trace Options**.

The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, the **SIP Rx** and **SIP Tx** boxes are checked. All SIP messages will appear in the trace with the color blue. To customize the color, right-click on **SIP Rx** or **SIP Tx** and select the desired color.


All Settings
T1
VComp
VPN
WAN
SCN
SSI
Jade
ATM
Call
DTE
EConf
Frame Relay
GOD
H.323
Interface
ISDN
Key/Lamp
Directory
Media
PPP
R2
Routing
Services
SIP
System
Events
Sip
Low
STUN
SIP Dect
Packets
SIP Reg/Opt Rx
SIP Misc Rx
SIP Reg/Opt Tx
SIP Misc Tx
SIP Call Rx
Cm Notify Rx
SIP Call Tx
Cm Notify Tx
Sip Rx
hex
IP Filter (nnn.nnn.nnn.nnn)
Sip Tx
hex
Default All
Clear All
Tab Clear All
Tab Set All
OK
Cancel
Save File
Load File
Load Partial File
Select File

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.



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 8.1 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 8.1 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.

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 8.1 Installation Manual,* Document Number 15-601042, August 2012

[2] *IP Office Manager Manual 10.0,* Document Number 15-601011, August 2012

[3] *IP Office Release 8.1 Implementing Voicemail Pro*, Document Number 15-601064, June 2012

[4] *IP Office System Status Application*, Document Number 15-601758, November 2011

[5] *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 8.0 with Verizon IPCC Service Suite.
[VZB-IPCCIPOR8FT] Application Notes for Configuring SIP Trunking using Verizon Business IP Contact Center VoIP Inbound and Avaya IP Office Release 8, Issue 1.0

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 8.1 with Verizon IP Trunk Service Suite.
[VZBIPT-IPO81] Application Notes for SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service and Avaya IP Office Release 8.1– Issue 1.1

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

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

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

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

Appendix A: SIP Line Template

Avaya IP Office Release 8.1 supports a SIP Line Template (in xml format) that can be created from an existing configuration and imported into a new installation to simplify configuration procedures as well as to reduce potential configuration errors.

Note that not all of the configuration information, particularly items relevant to a specific installation environment, is included in the SIP Line Template. Therefore, it is critical that the SIP Line configuration be verified/updated after a template has been imported and additional configuration be supplemented using Section 5.4 in these Application Notes as a reference.

The SIP Line Template created from the configuration as documented in these Application Notes is as follows:

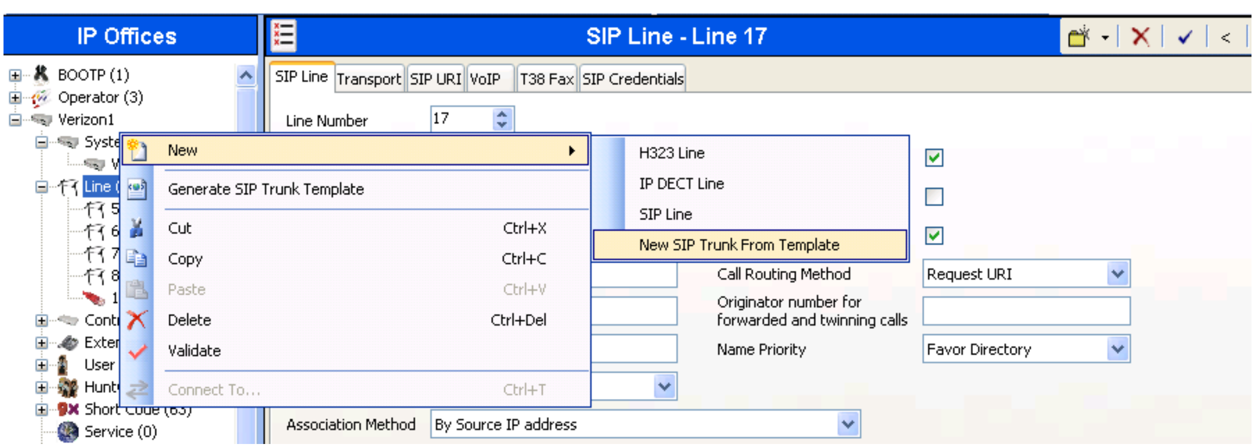

```xml
<?xml version="1.0" encoding="utf-8"?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20130312</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>VerizonIPCC_IPO81</DescriptiveName>
  <ITSPDomainName>1.1.1.2</ITSPDomainName>
  <SendCallerID>CallerIDNone</SendCallerID>
  <ReferSupport>true</ReferSupport>
  <ReferSupportIncoming>2</ReferSupportIncoming>
  <ReferSupportOutgoing>2</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>true</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>SystemDefault</LineNamePriority>
  <UpdateSupport>UpdateAuto</UpdateSupport>
  <UserAgentServerHeader />
  <CallerIDfromFromheader>false</CallerIDfromFromheader>
  <PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
  <ITSPProxy>172.30.205.55</ITSPProxy>
  <LayerFourProtocol>SipUDP</LayerFourProtocol>
  <SendPort>5072</SendPort>
  <ListenPort>5060</ListenPort>
  <DNSServerOne>0.0.0.0</DNSServerOne>
  <DNSServerTwo>0.0.0.0</DNSServerTwo>
  <CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>
  <SeparateRegistrar />
  <CompressionMode>AUTOSELECT</CompressionMode>
  <UseAdvVoiceCodecPrefs>true</UseAdvVoiceCodecPrefs>
  <AdvCodecPref>G.729(a) 8K CS-ACELP,G.711 ULAW 64K</AdvCodecPref>
  <CallInitiationTimeout>4</CallInitiationTimeout>
  <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
```

```
  <VoipSilenceSupression>false</VoipSilenceSupression>
  <ReinviteSupported>true</ReinviteSupported>
  <FaxTransportSupport>FOIP_NONE</FaxTransportSupport>
  <UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
  <CodecLockdown>false</CodecLockdown>
  <Rel100Supported>false</Rel100Supported>
  <T38FaxVersion>3</T38FaxVersion>
  <Transport>UDPTL</Transport>
  <LowSpeed>0</LowSpeed>
  <HighSpeed>0</HighSpeed>
  <TCFMethod>Trans_TCF</TCFMethod>
  <MaxBitRate>FaxRate_14400</MaxBitRate>
  <EflagStartTimer>2600</EflagStartTimer>
  <EflagStopTimer>2300</EflagStopTimer>
  <UseDefaultValues>true</UseDefaultValues>
  <ScanLineFixup>true</ScanLineFixup>
  <TFOPEnhancement>true</TFOPEnhancement>
  <DisableT30ECM>false</DisableT30ECM>
  <DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
  <DisableT30MRCompression>false</DisableT30MRCompression>
  <NSFOverride>false</NSFOverride>
</Template>
```

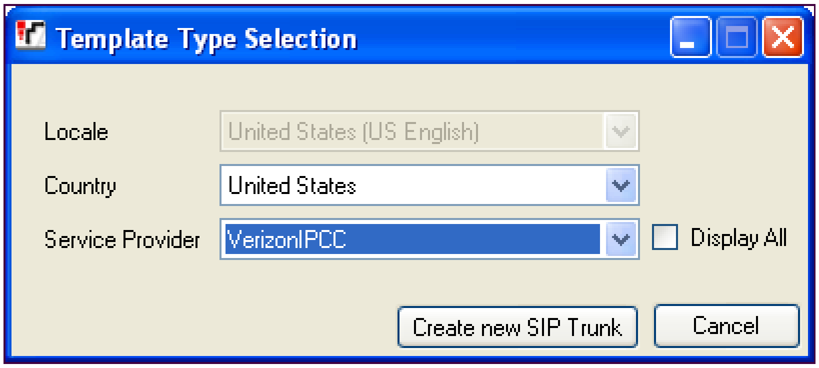

To import the above template into a new installation:

1. On the PC where IP Office Manager was installed, copy and paste the above template into a text document named **US_VerizonIPCC_SIPTrunk.xml**. Move the .xml file to the IP Office Manager template directory (C:\Program Files\Avaya\IP Office\Manager\Templates). It may be necessary to create this directory.

2. Import the template into an IP Office installation by creating a new SIP Line as shown in the screenshot below. In the Navigation Pane on the left, right-click on **Line** then navigate to **New → New SIP Trunk From Template**:



1. Verify that "United States" is automatically populated for **Country** and "VerizonIPCC" is automatically populated for **Service Provider** in the resulting Template Type Selection screen as shown below. Click **Create new SIP Trunk** to finish the importing process.



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.