Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking Using Verizon Business IP Contact Center VoIP Inbound and Avaya IP Office Release 7 – Issue 1.1

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 Release 7 Preferred Edition, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, 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.

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

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1. Introduction

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Contact Center VoIP Inbound Service and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500 Release 7 Preferred Edition, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, digital, and analog endpoints.

Customers using Avaya IP Office with the Verizon Business IP Contact Center SIP Trunk 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 IP Contact Center 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 IP Contact Center service terminated via a PIP network connection, the solution validated in this document applies equally to IP Contact Center services delivered via IDA service terminations.

For more information on the Verizon Business IP Contact Center service, including access alternatives, visit http://www.verizonbusiness.com/products/contactcenter/ip/.

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 test cases detailed in Reference [VZ-Test-Plan]. To summarize, the testing included the following successful SIP trunk interoperability compliance testing:

- 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 digital telephones, analog telephones, Avaya IP Office Softphone, and Avaya IP Office Voicemail Pro voicemail, auto-attendant, and meet-me conferencing applications. The display of caller ID on display-equipped Avaya IP Office telephones was verified.
- 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, hear announcements and music while in queue, 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 [VZB-IPT-IPOR7]. As

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detailed in reference [VZB-IPT-IPOR7], these outgoing PSTN calls can be originated from Avaya H.323 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 IP Contact Center, inbound toll-free calls arriving via the SIP Line configured in Section 5.4 could be transferred, forwarded, or twinned out the Verizon IP Trunk Service SIP Line. Inbound toll-free calls from the Verizon IP Contact Center SIP Line could also trigger mobile callback calls that use the Verizon IP Trunk Service SIP Line.

- Proper disconnect when the PSTN caller abandons a call before answer for inbound toll-free calls.
- Proper disconnect when either party hangs up an active call.
- Proper busy tone heard when a PSTN user calls a toll-free number directed to a busy IP Office user (i.e., if no redirection is configured for user busy 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).
- 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 user while presenting a "WITHHELD" or anonymous display to the IP Office user.
- SIP OPTIONS monitoring of the health of the SIP trunk was verified. Both Verizon Business and IP Office can monitor health using SIP OPTIONS. The Avaya IP Office configurable control of SIP OPTIONS timing was exercised successfully.
- Calls using the G.729(a) and G.711 ULAW codecs.
- DTMF transmission using RFC 2833. Successful voice mail navigation for G.729a and G.711MU for incoming toll-free calls. Successful navigation of a simple auto-attendant application configured on IP Office Voicemail Pro. Successful entry of conference codes and PINs collected by Voicemail Pro for IP Office meet-me conferencing. Successful use of IP Office Mobile Call Control, where DTMF sequences can be performed remotely using the SIP Line.
- The "callback" feature of Avaya Voicemail Pro was tested successfully. When a call was made via a toll-free number, and a message was left for a voice mail subscriber with "callback" configured, an outbound call was placed to the subscriber's configured mobile telephone via the Verizon IP Trunk SIP Line described in [VZB-IPT-IPOR7]. Upon answer, Voicemail Pro announces the call and prompts the user to enter the "#" key to accept the call. The user has the opportunity to navigate the voicemail TUI via DTMF (e.g., to listen to the voice message that stimulated the callback).
- Inbound toll-free long holding time call stability (See Section 2.2. Although long SIP sessions are not refreshed, the media paths remained connected using the sample configuration)
- Telephony features such as call waiting, hold, transfer, and conference.
- Call Forwarding of Verizon toll-free calls to PSTN destinations via the Verizon IP Trunk service documented in reference [VZB-IPT-IPOR7], presenting true calling party information to the mobile phone

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- Mobile twinning of Verizon toll-free calls to a mobile phone via the Verizon IP Trunk service documented in reference [VZB-IPT-IPOR7], 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 receive busy tone.
- Mobile callback. A Verizon toll-free number can be associated with mobile callback. A configured mobile twinning PSTN caller may dial the toll-free number, hear ring back, and hang up. IP Office will call the user back, in this case via the Verizon IP Trunk service described in [VZB-IPT-IPOR7].
- Proper DiffServ markings for IP Office SIP signaling and RTP media

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. The Verizon IPCC Service does not support fax.

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

3. When using the IP Office Softphone, inbound PSTN calls from the Verizon Business IP Toll Free service to the IP Office Softphone may negotiate to the G.711MU codec, even if the SIP Line configuration lists G.729a first on the **VoIP** tab (as shown in Section 5.4). Specifically, if the IP Office Softphone user has logged in with the "IP Office: Default" profile, and the **Automatic Codec Preference** parameter on the **System → Telephony** tab is set to "G.711 ULAW 64K", an inbound call from the Verizon Business IP Trunk service to the IP Office Softphone will use G.711MU. The IP Office Softphone user can log in with the "IP Office: Low Bandwidth" profile to ensure use of G.729a for both inbound and outbound calls via the SIP Line to Verizon Business.

4. Although the Verizon Business IP Contact Center service supports transfer using the SIP REFER method, IP Office will not send REFER to Verizon in the verified configuration.

5. 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 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.

6. When a user on the PSTN hangs up an active call, Verizon 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 soon after

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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.

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

2.3. Support

2.3.1. Avaya

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

2.3.2. Verizon

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

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3. Reference Configuration

Figure 1 illustrates an example Avaya IP Office solution connected to the Verizon Business IP Contact Center SIP Trunk service. The Avaya equipment is located on a private IP subnet. An enterprise edge router provides access to the Verizon Business network via a Verizon Business T1 circuit. This circuit is provisioned for the Verizon Business Private IP (PIP) service. Companion Application Notes, found in reference [VZB-IPT-IPOR7], illustrate 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 IP Contact Center service were accessible via the same PIP connection.

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

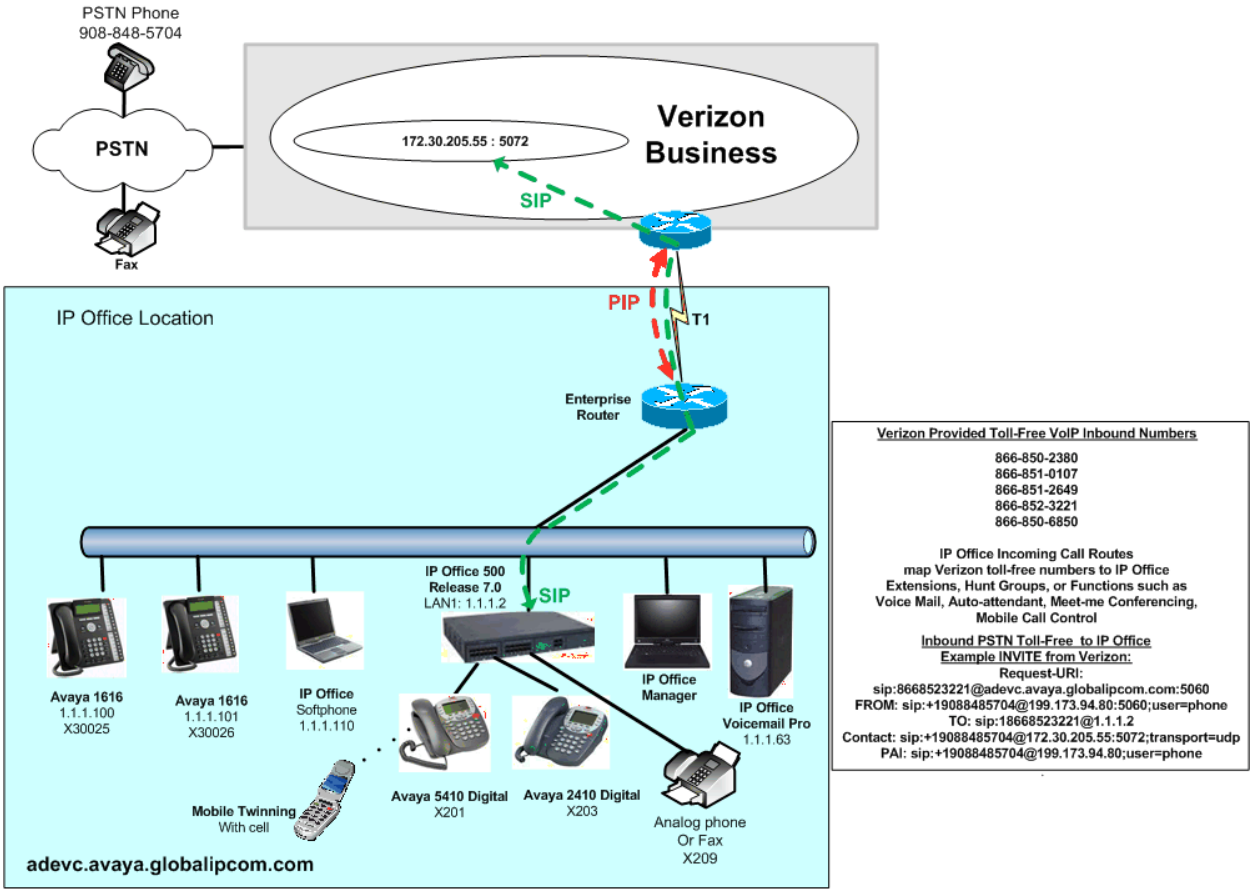

Figure 1: Avaya IP Office with Verizon IP Contact Center Service

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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 (e.g., 866-850-6850) was routed to different IP Office destinations (i.e., IP Office configuration changes were made to the Incoming Call Route for 866-850-6850 between successive tests).

Verizon Provided Toll-Free Number	Configured Avaya IP Office Destination	Notes
866-852-3221	x30026	Avaya 1616 Telephone, or Avaya IP Office Softphone logged in as x30026
866-850-2380	x201	Digital Telephone with Mobile Twinning and Mobile Call Control permission
866-851-2649	x30025	Avaya 1616 Telephone
866-850-6850	Voicemail Collect on Voicemail Pro	Allow external callers to access voice mail toll-free
866-850-6850	Automated Attendant on Voicemail Pro	Allow external callers to reach attendant toll-free, and transfer to IP Office destinations
866-850-6850	Meet-Me Conferencing prompts / announcements on Voicemail Pro	Collect conference ID and PIN, transfer to IP Office conference by ID
866-850-6850	Inbound Mobile Call Control	Allow toll-free calls from pre-configured twinning numbers to access mobile call control
866-850-6850	Mobile Callback	Allow toll-free calls from pre-configured twinning numbers to access mobile callback, IP Office calls caller back via Verizon IP Trunk Service
866-851-0107 (any caller)	“220 Sales” Hunt Group (with default priority)	“LongestWaiting” Hunt Group with queuing, announcements
866-851-0107 (specific callers)	“221 Overdue Account” Hunt Group	Show IP Office destination route selection based on caller
866-851-0107 (specific priority callers)	“220 Sales” Hunt Group (with High Priority)	Show IP Office priority queuing based on caller

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

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4. Equipment and Software Validated

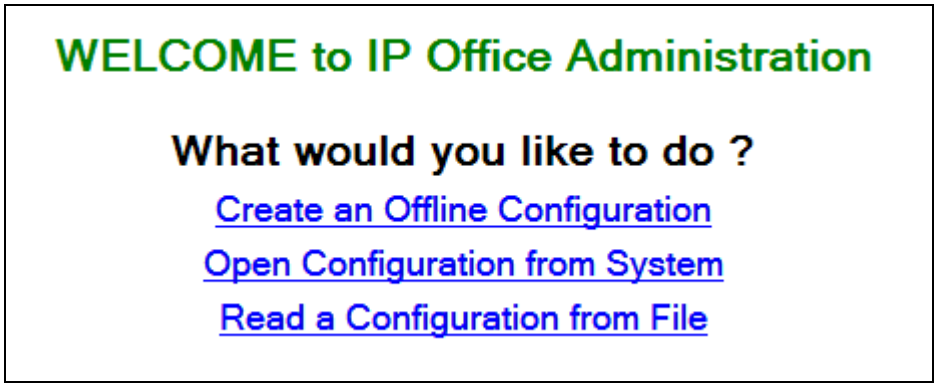
Table 2 shows the equipment and software used in the sample configuration.

Equipment	Software
Avaya IP Office 500	Release 7.0 (5)
Avaya IP Office Manager	Release 9.0 (5)
Avaya IP Office Voicemail Pro	Release 7.0 (17)
Avaya IP Office Voicemail Pro Client	Version 7.0 (17)
Avaya 1600-Series Telephones (H.323)	Release 1.3
Avaya 2400-Series and 5400-Series Digital Telephones	REL: 6.00
Avaya IP Office Softphone	Release 3.1.2.17 59616

Table 2: Equipment and Software Tested

5. Avaya IP Office Configuration

IP Office is configured via the IP Office Manager program. For more information on IP Office Manager, consult reference [IPO-MGR]. 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 will be displayed:



Open the IP Office configuration, either by reading the configuration from the IP Office server, or from file. The appearance of the IP Office Manager can be customized using the **View** menu. 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. Physical, Network, and Security Configuration

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

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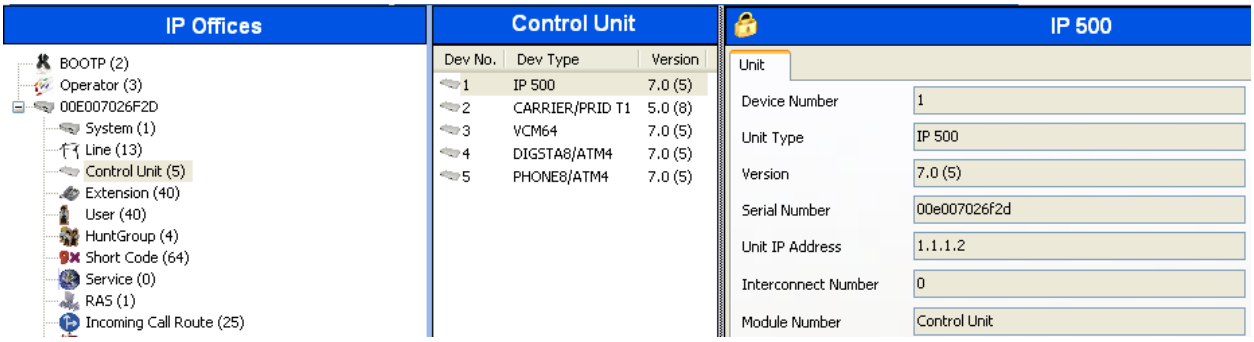
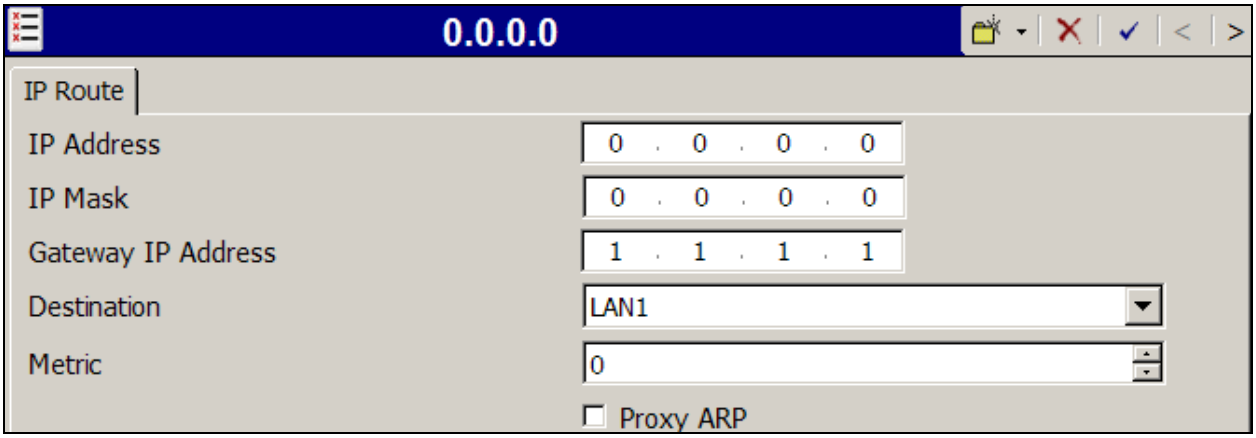
In the sample configuration, looking at the IP Office 500 from left to right, the first module is blank (i.e., no module is physically inserted). The next slots from left to right contain a VCM64, a Digital station module, and a "Phone8" analog module. The VCM64 is a Voice Compression Module supporting VoIP codecs. The Digital module allows connection of Avaya 5400-Series and Avaya 2400-Series Digital telephones. Referring to **Figure 1**, the Avaya 5410 telephone with extension 201 is connected to port 1 of the Digital module, and the Avaya 2410 telephone with extension 203 is connected to port 3 of the Digital module. The "Phone8" module allows connection of analog devices such as simple analog telephones or fax machines. In the testing of the sample configuration, analog telephones or a fax machine were connected to a port of the "Phone8" module.

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 **Control Unit** Group pane. In the screen below, **IP 500** is selected in the Group pane, revealing additional information about the IP 500 in the **IP 500** 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 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 default route using **Destination** LAN1.



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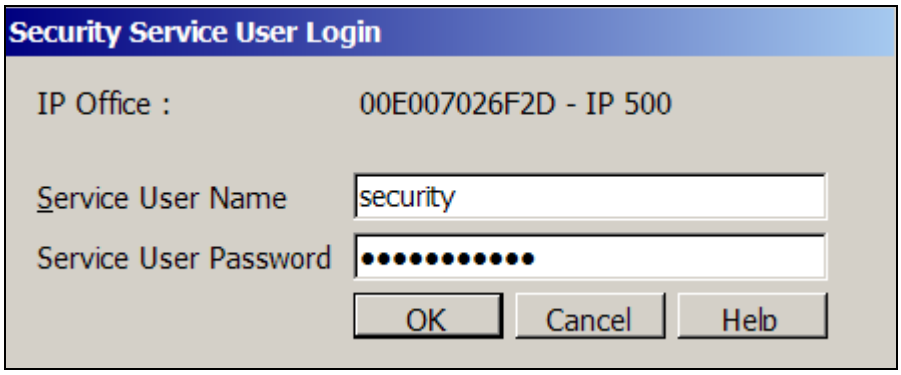
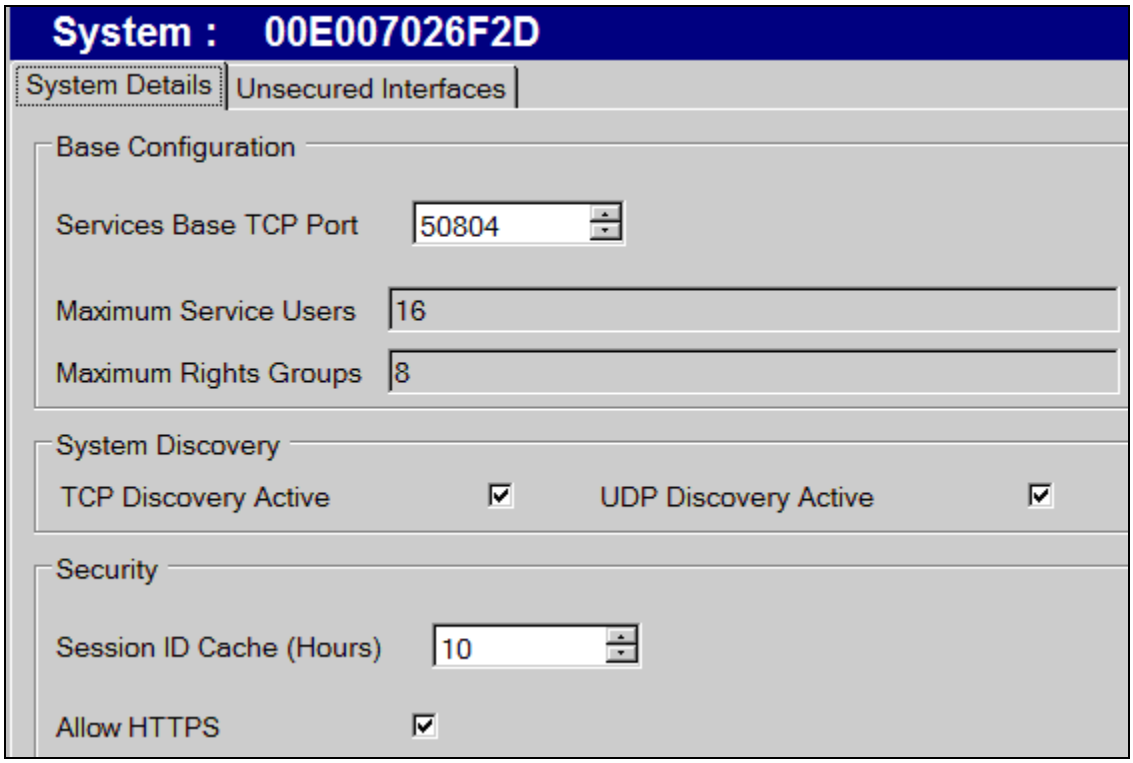
To facilitate use of Avaya IP Office Softphone, https was enabled in the sample configuration. To check whether https is enabled, navigate to **File → Advanced → Security Settings**. A screen such as the following is presented. Log in with the appropriate security credentials.



After logging in, select **System** from the Navigation pane and the appropriate IP Office system from the Group pane. In the Details pane, select the **System Details** tab. Verify that **Allow HTTPS** is checked. If not, check the box, click **OK**, and heed the on-screen prompts and warnings. Note that this action may be service disrupting.



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

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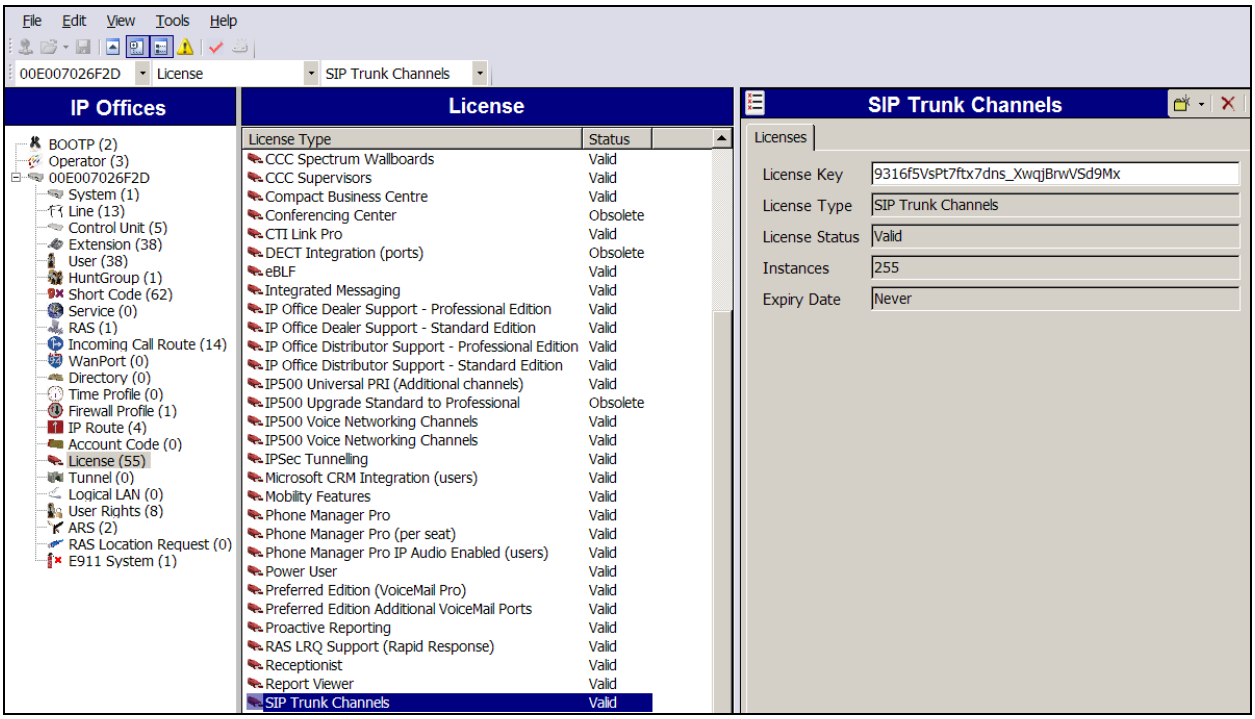
5.2. Licensing

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.

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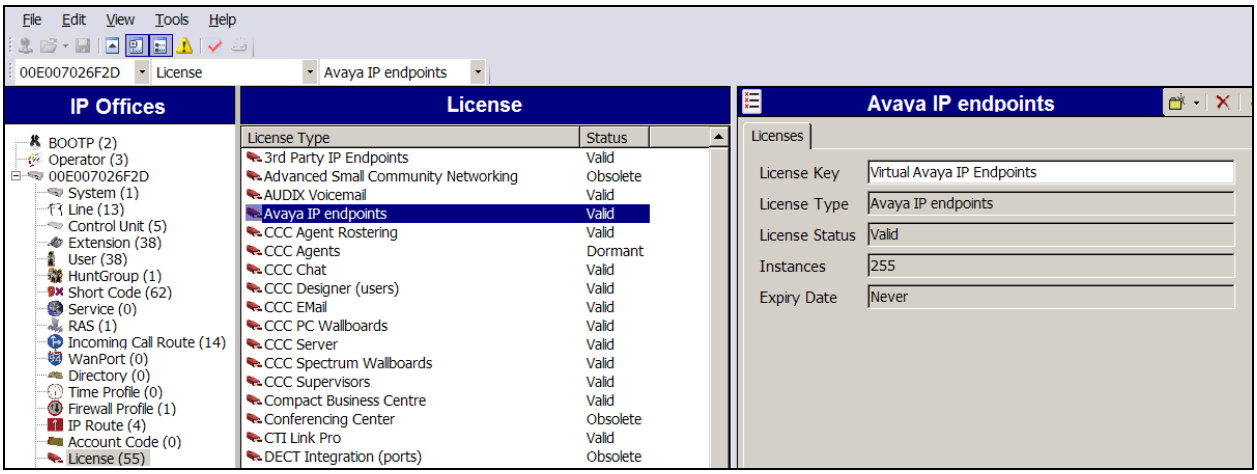
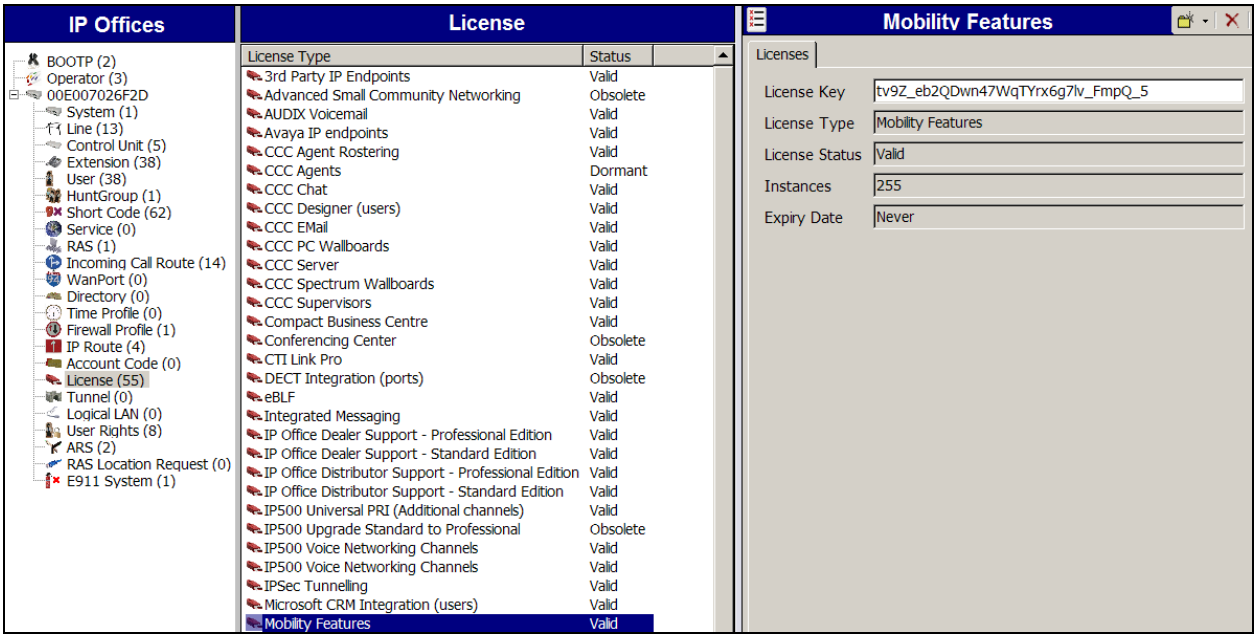


A similar process can be used to check the license status for other desired features. For example, the following screen shows the availability of a valid license for Mobility features. In the sample configuration, various mobility features including Mobile Twinning are used.



The following screen shows the availability of a valid license for **Power User** features. In the sample configuration, the user with extension 30026 will be configured as a "Power User" and will be capable of using the Avaya IP Office Softphone.

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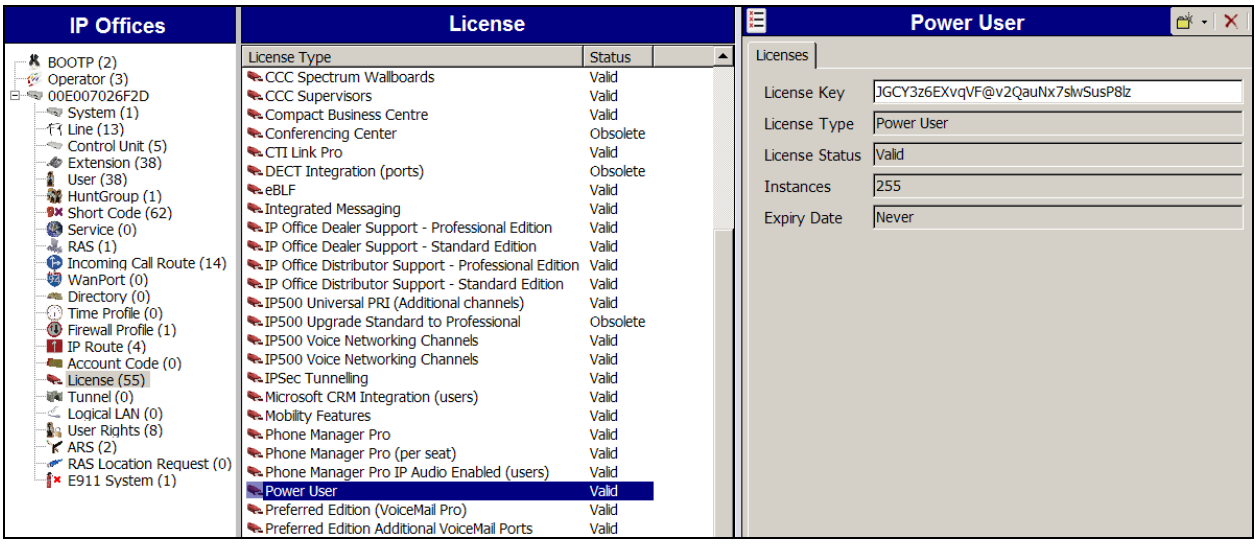

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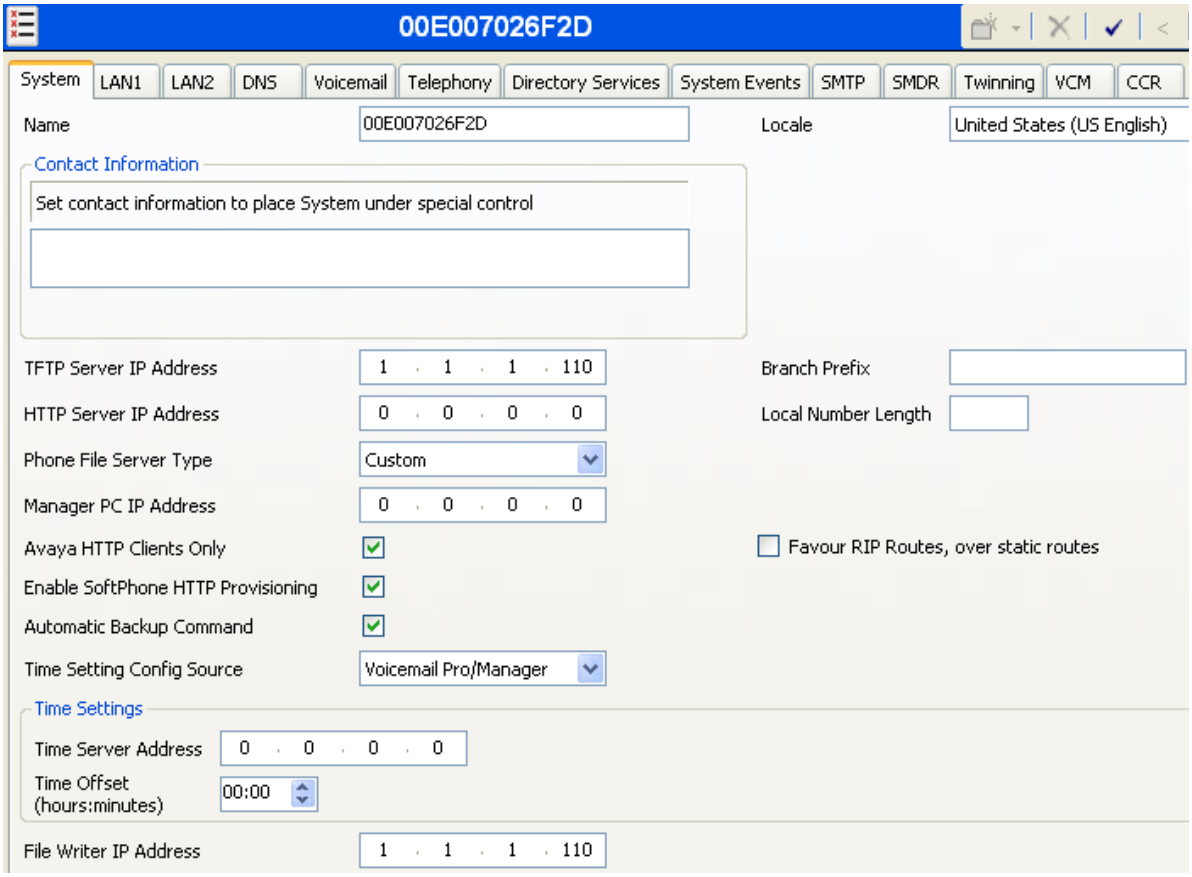
5.3. System Settings

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.

5.3.1. 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, the MAC address is used as the name. The **Avaya HTTP Clients Only** and **Enable SoftPhone HTTP Provisioning** boxes are checked to facilitate Avaya IP Office Softphone usage.



5.3.2. LAN Settings

In the sample configuration, LAN1 was used to connect the IP Office to the enterprise network. Other LAN choices (e.g., LAN2) may also be used. 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, known to Verizon Business, is 1.1.1.2. Other parameters on this screen may be set according to customer requirements.

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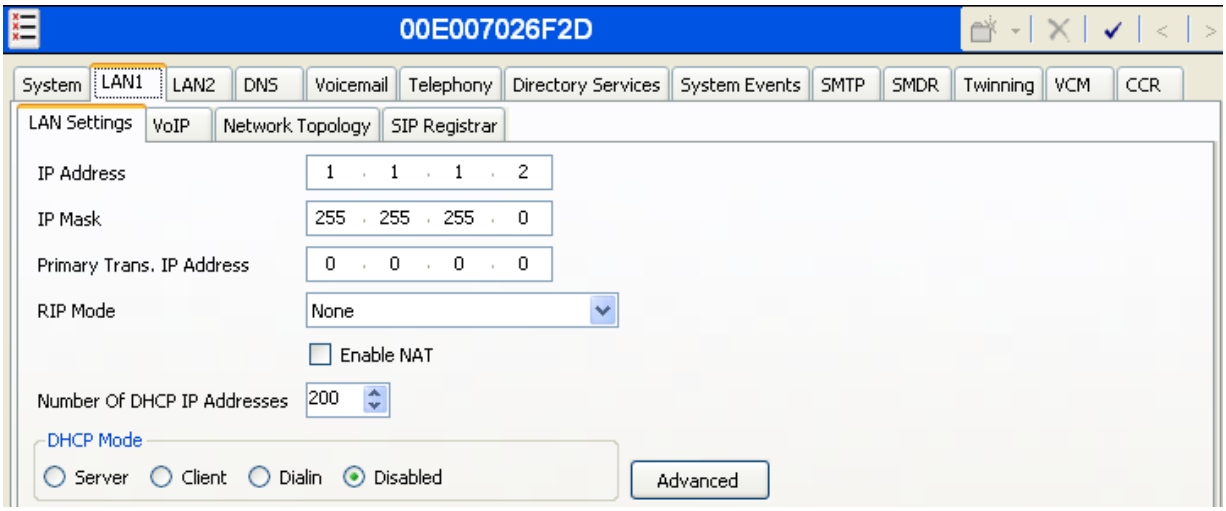

Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 1600-Series Telephones used in the sample configuration. The **SIP Trunks Enable** box must be checked to enable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

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. That is, for toll-free SIP trunk calls from Verizon Business, the SIP protocol exchanges will result in Verizon Business sending RTP media to IP Office using a UDP port in the configurable range shown below.

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 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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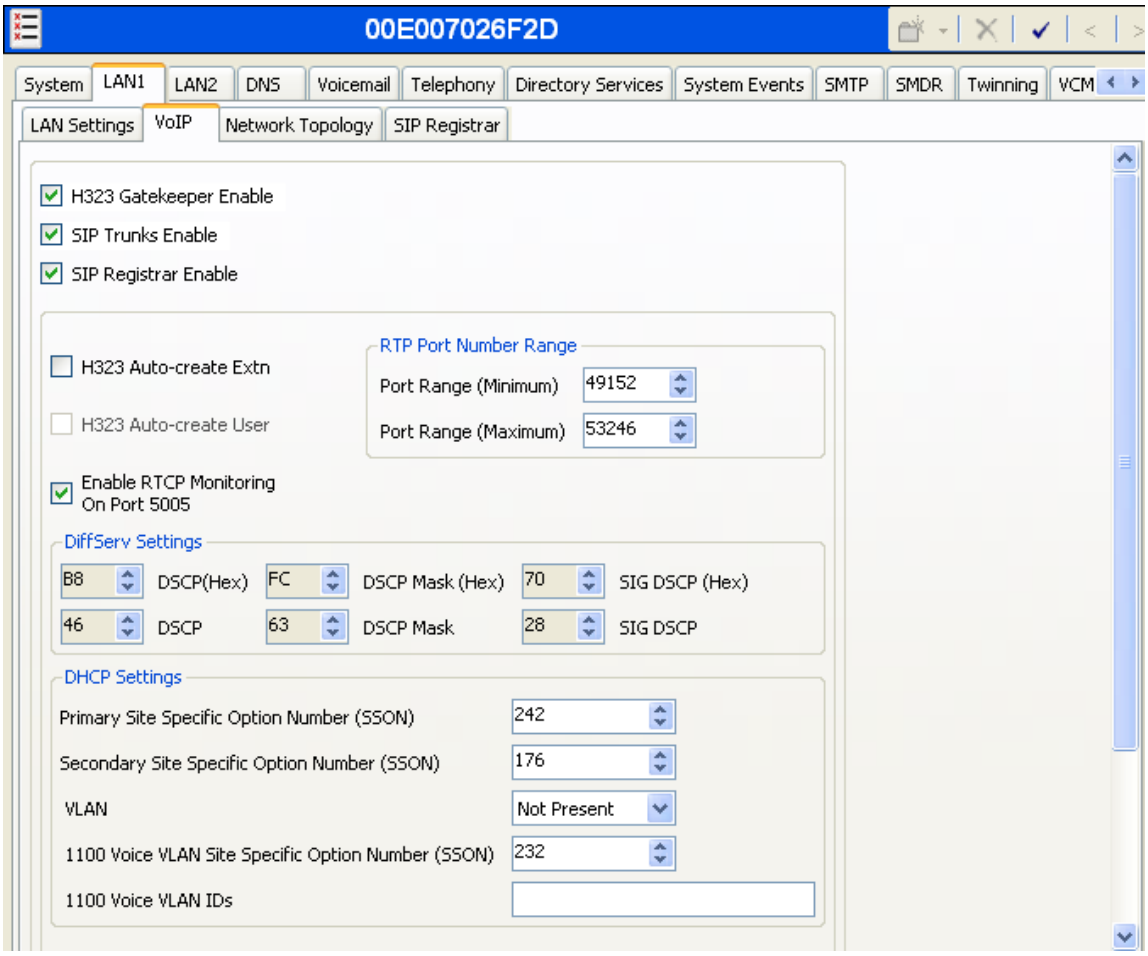

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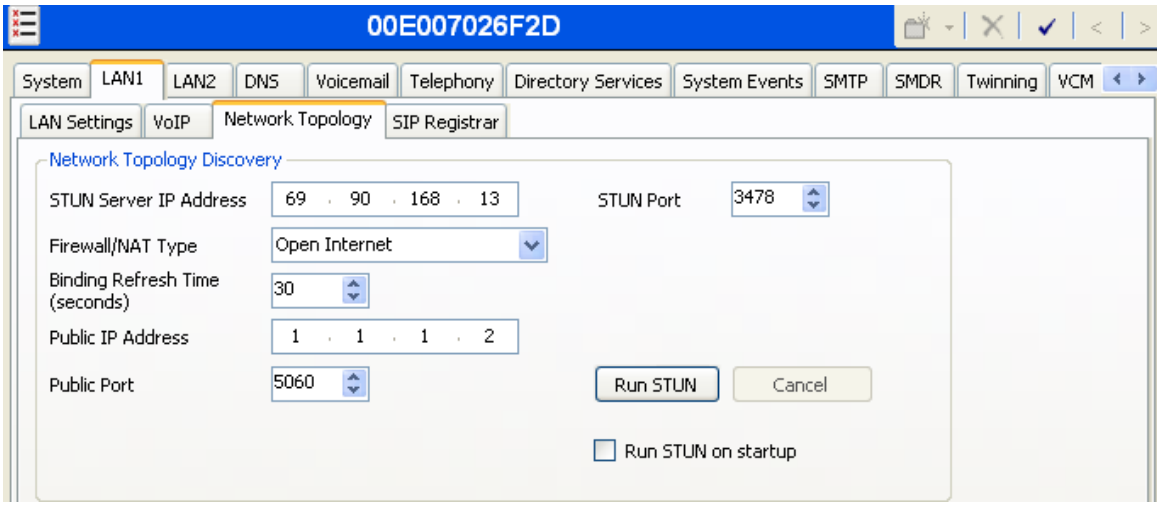
Select the **Network Topology** tab as shown in the following screen. For **Public IP Address**, enter the Avaya IP Office LAN1 IP address. Set the **Public Port** to 5060. In the sample configuration, the **Firewall/NAT Type** is set to "Open Internet". With this configuration, STUN will not be used. During the testing, the **Binding Refresh Time** was varied (e.g., 30 seconds, 90 seconds to test SIP OPTIONS timing). Click the **OK** button.



Note: The **Firewall/NAT Type** parameter may need to be different, depending on the type of firewall or Network Address Translation device used at the customer premise.

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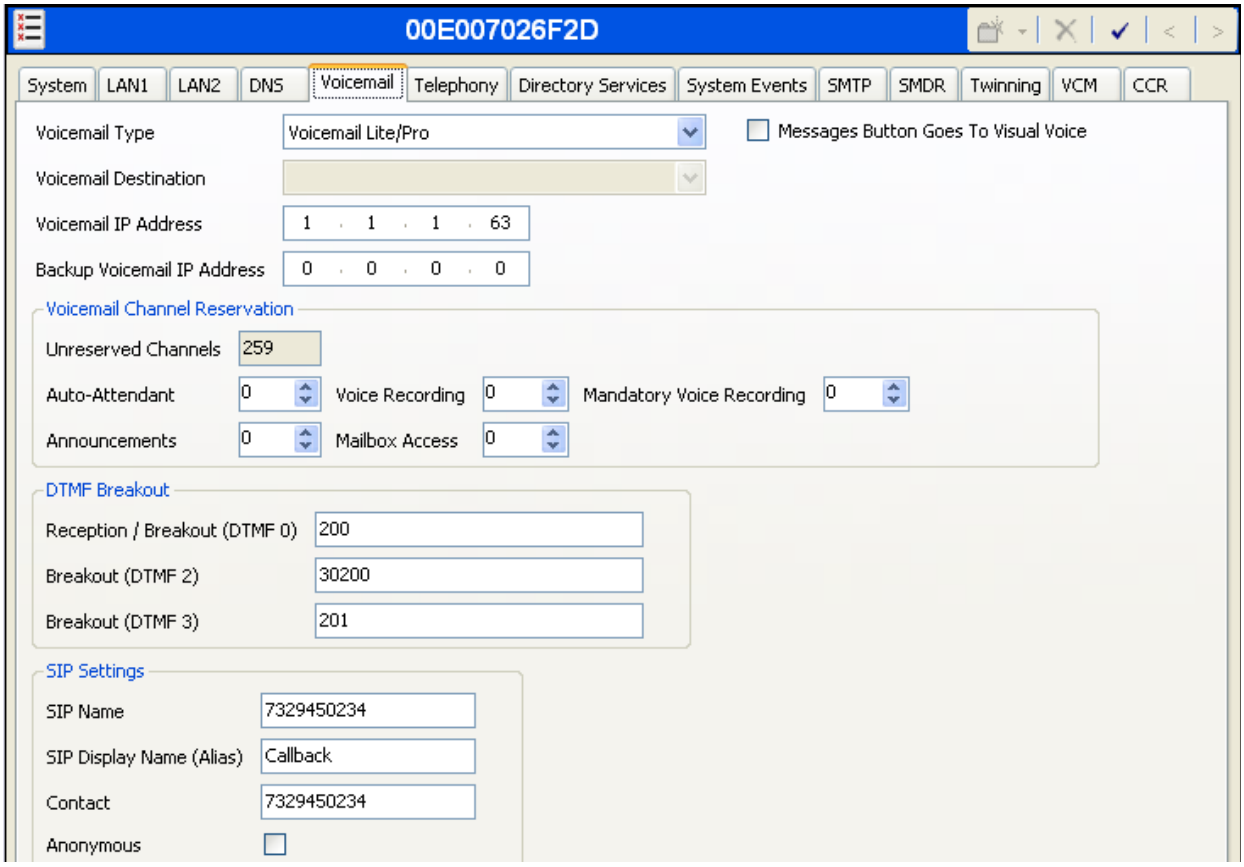
5.3.3. Voicemail

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 1.1.1.63, the IP Address of the PC running the Voicemail Pro software, as shown in **Figure 1**. As described in [VZB-IPT-IPOR6], the "Callback" application of Avaya Voicemail Pro was used to allow Voicemail Pro to call out via the Verizon Business IP Trunk service when a message is left in a voice mailbox.



5.3.4. 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 IP Contact Center service, and be forwarded or transferred back to the PSTN with the outbound leg of the call using the Verizon IP Trunk service. The **Companding Law** parameters are set to "ULAW" as is typical in North American locales. Other parameters on this screen may be set according to customer requirements.

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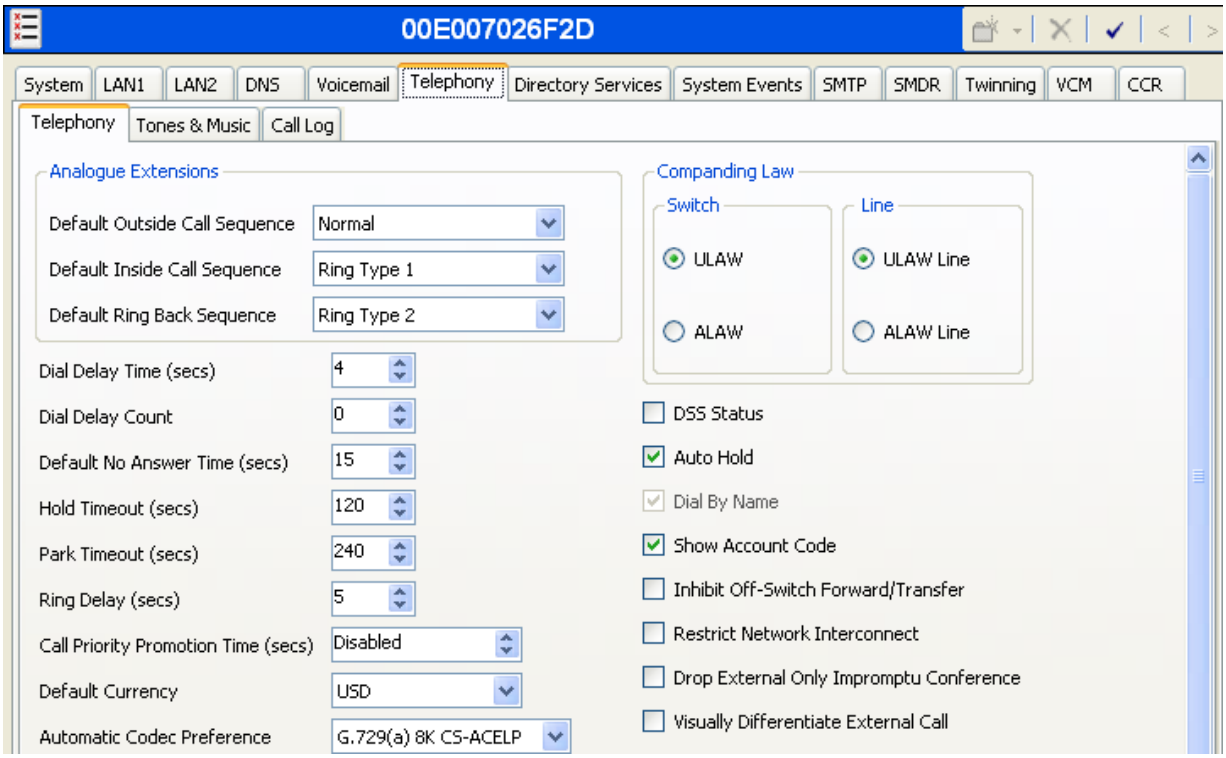
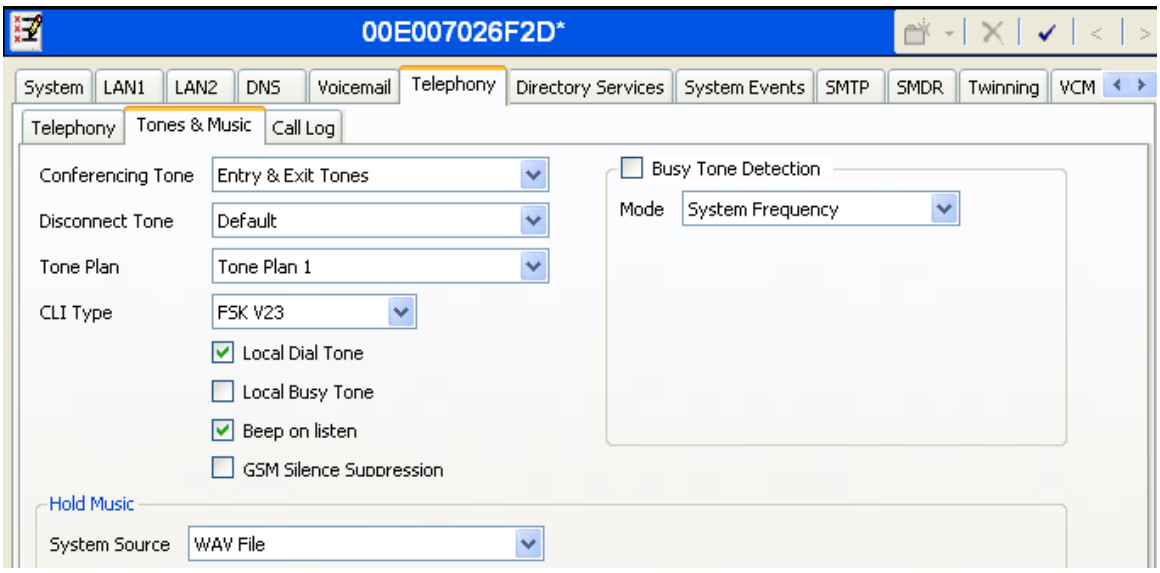


To view or change settings associated with tones or music, select the **Telephony** tab and **Tones & Music** sub-tab as shown in the following screen. The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. In the sample configuration, music on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



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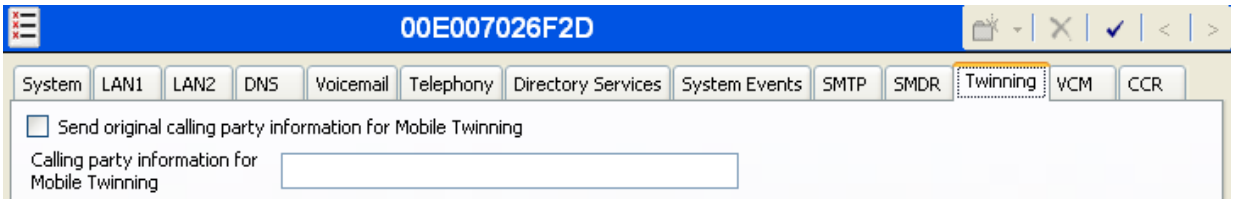
5.3.5. System Twinning Configuration

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 3.4 of reference [VZB-IPT-IPOR6]), 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 IP Contact Center service toll-free number, and be twinned out to a mobile telephone using the Verizon IP Trunk service, with the twinned mobile phone seeing the identity of the caller that dialed the Verizon toll-free number.



5.4. SIP Line

This section shows the configuration screens for the SIP Line in IP Office Release 7. Since IP Office Release 7 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 reference [JRR-IPOR6]. The Appendix in Section 10 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.

5.4.1. 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 LAN1 address (1.1.1.2) so that IP Office uses 1.1.1.2 as the host portion of SIP headers such as the From header. By default, the **In Service** and **Check OOS** boxes are checked. In the sample configuration, IP Office will use the SIP OPTIONS method to periodically check the SIP Line. The time between SIP OPTIONS sent by IP Office will use the **Binding Refresh Time** for LAN1, as shown in Section 5.3.2. See Section 5.8 for additional information related to configuring the periodicity of SIP OPTIONS.

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

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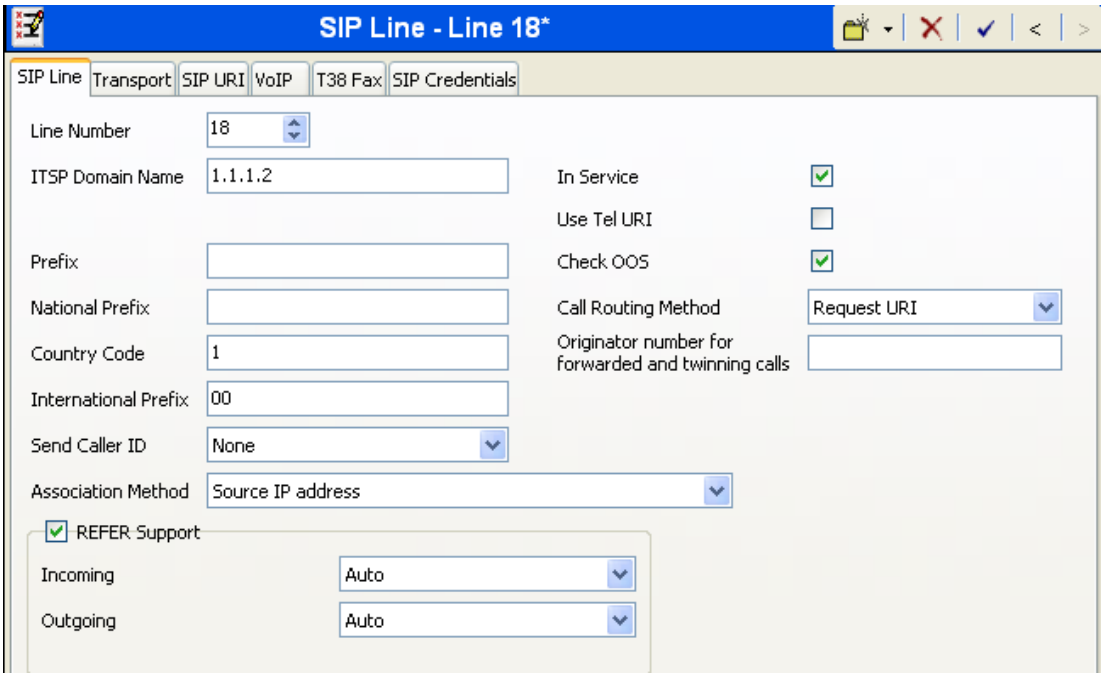
contained the toll-free number, and the "To" header contained 1 followed by the toll-free number.

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 9088485704 would display 9088485704. 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 "0019088485704". 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. Click **OK** (not shown).



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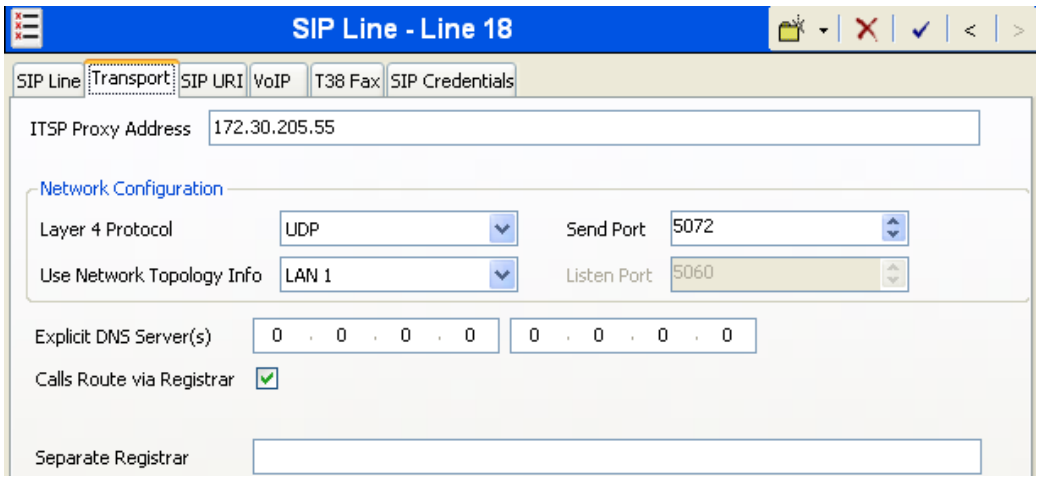
5.4.2. SIP Line - Transport Tab

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 "LAN 1". This associates the SIP Line with the parameters in the **System → LAN1 → Network Topology** tab.



5.4.3. 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.

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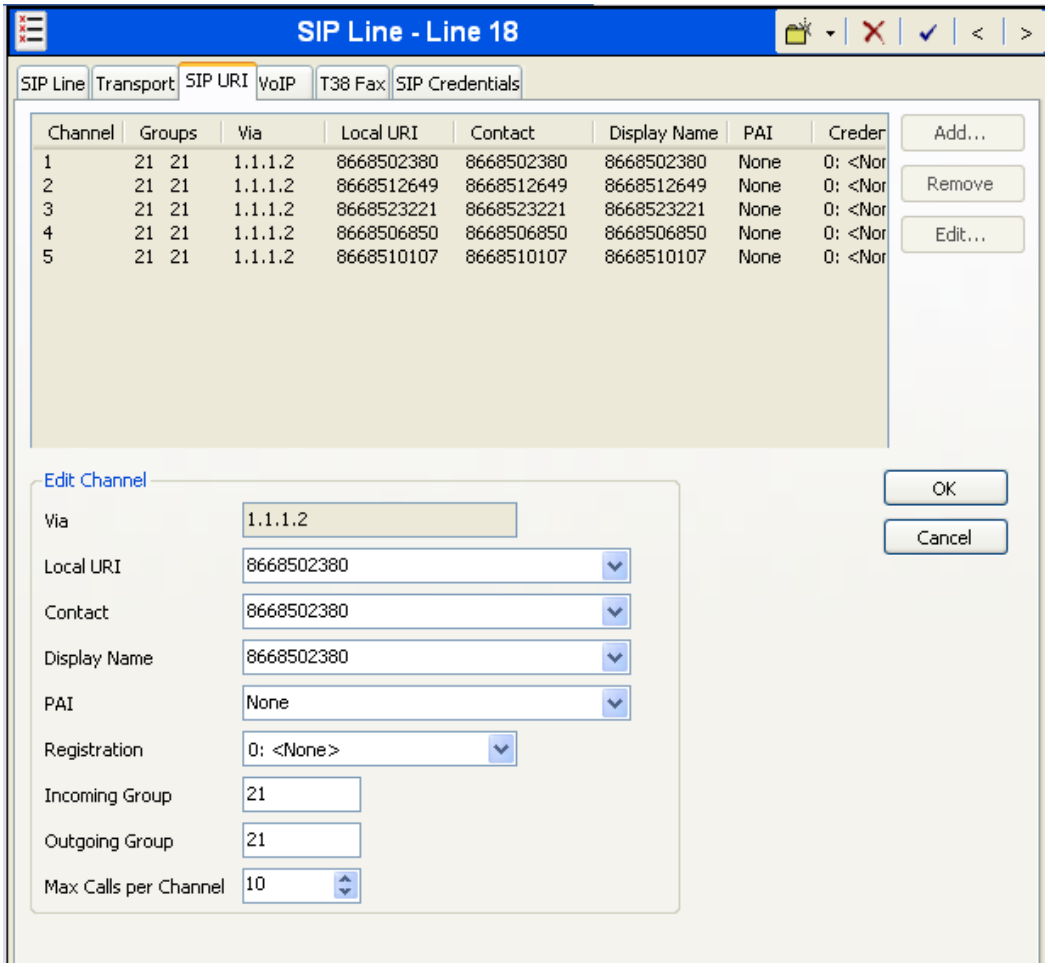
In the sample configuration, each of the Verizon-provided toll free numbers are entered as a SIP URI, with the specific number entered in the **Local URI**, **Contact**, and **Display Name** fields. The **PAI** parameter was introduced in IP Office Release 6.1, and the value "None" is shown selected from the drop-down menu. With PAI set to "none", IP Office Release 6.1 and 7.0 will behave like IP Office Release 6.0 with respect to the SIP P-Asserted-Identity header. 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 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 IP Contact Center 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 404 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

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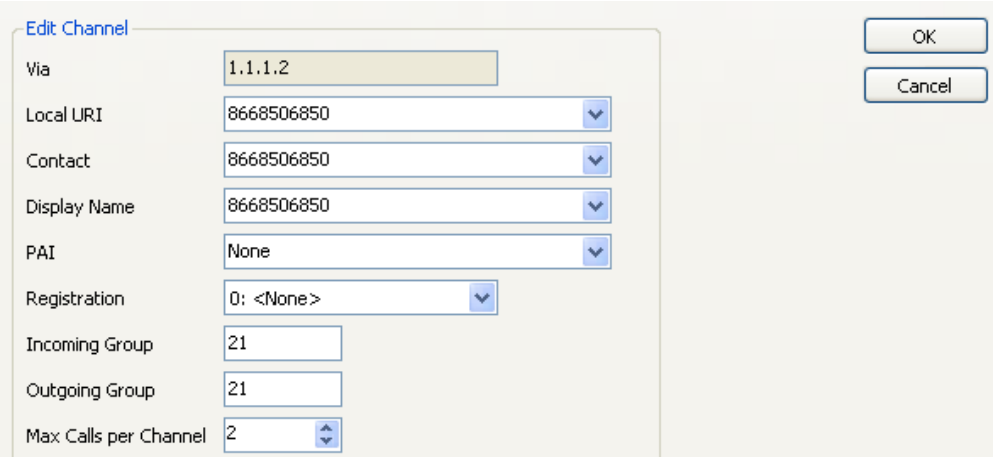
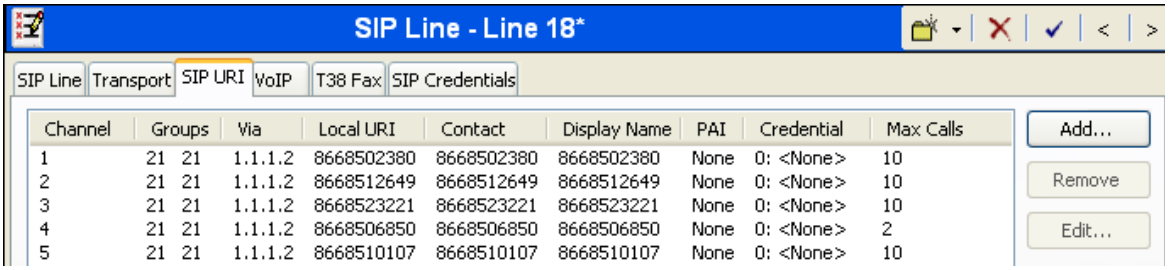
the specific toll-free numbers, preventing a surge of calls to a given toll-free number from monopolizing the available call handling capacity of the access line or IP Office resources. An alternative means to restrict the number of simultaneous calls to a toll-free number that terminates on a hunt group 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.



The following shows a portion of the **SIP URI** tab for SIP Line 18 after the SIP URIs corresponding to each of the Verizon-provided toll-free numbers have been added.



5.4.4. SIP Line - VoIP Tab

Select the **VoIP** tab. In the sample configuration, the **Compression Mode** was configured using the **Advanced** button, allowing an explicit ordered list of codecs to be specified. Place a check mark next to the **G.729(a) 8K CS-ACELP** and **G.711 ULAW 64K** codecs to indicate IP Office can 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. Since the

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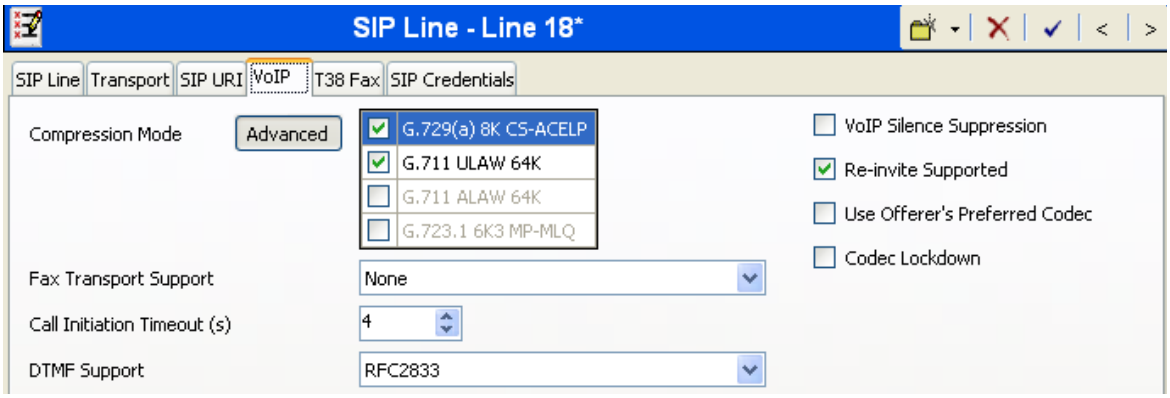
Verizon Business IP Contact Center service does not support fax, the **Fax Transport Support** parameter is set to "None", and the **T38 Fax** tab need not be visited. Since the Verizon Business IP Contact Center service does not require registration, the **SIP Credentials** tab need not be visited. Click **OK** (not shown).



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 201

The following screen shows the **User** tab for User 201. As shown in **Figure 1**, this user corresponds to the digital telephone 5410.

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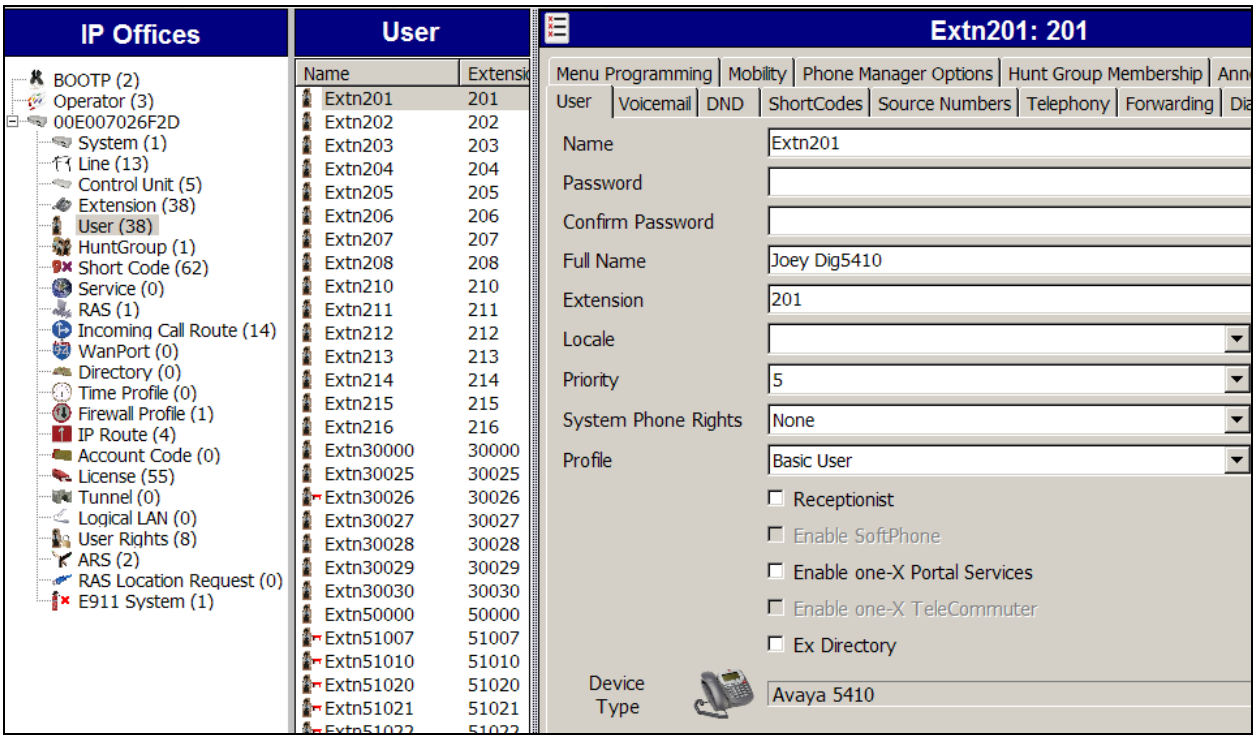
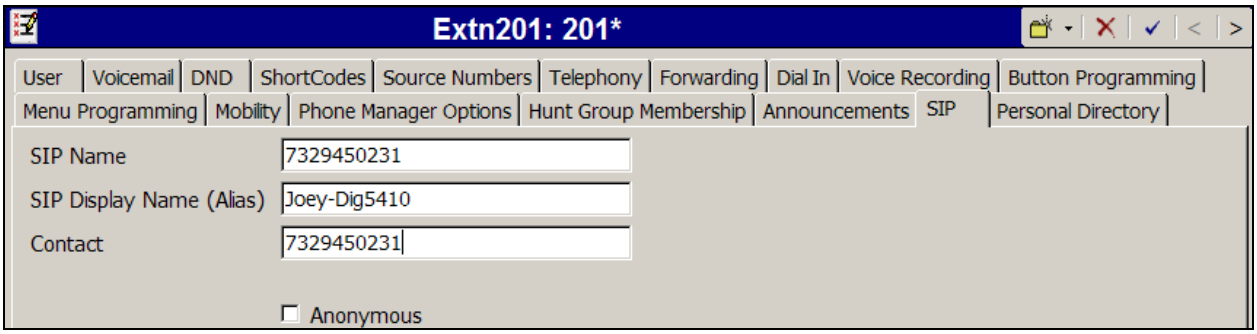


The following screen shows the **SIP** tab for User 201. The **SIP Name** and **Contact** parameters are configured with a Verizon IP Trunk DID number for the user, 7329450231. As shown in [VZB-IPT-IPOR6], these parameters configure the user part of the SIP URI in the From header for outgoing SIP trunk 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 201 will use the Mobile Twinning feature. The following screen shows the **Mobility** tab for User 201. 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 917326870755. 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

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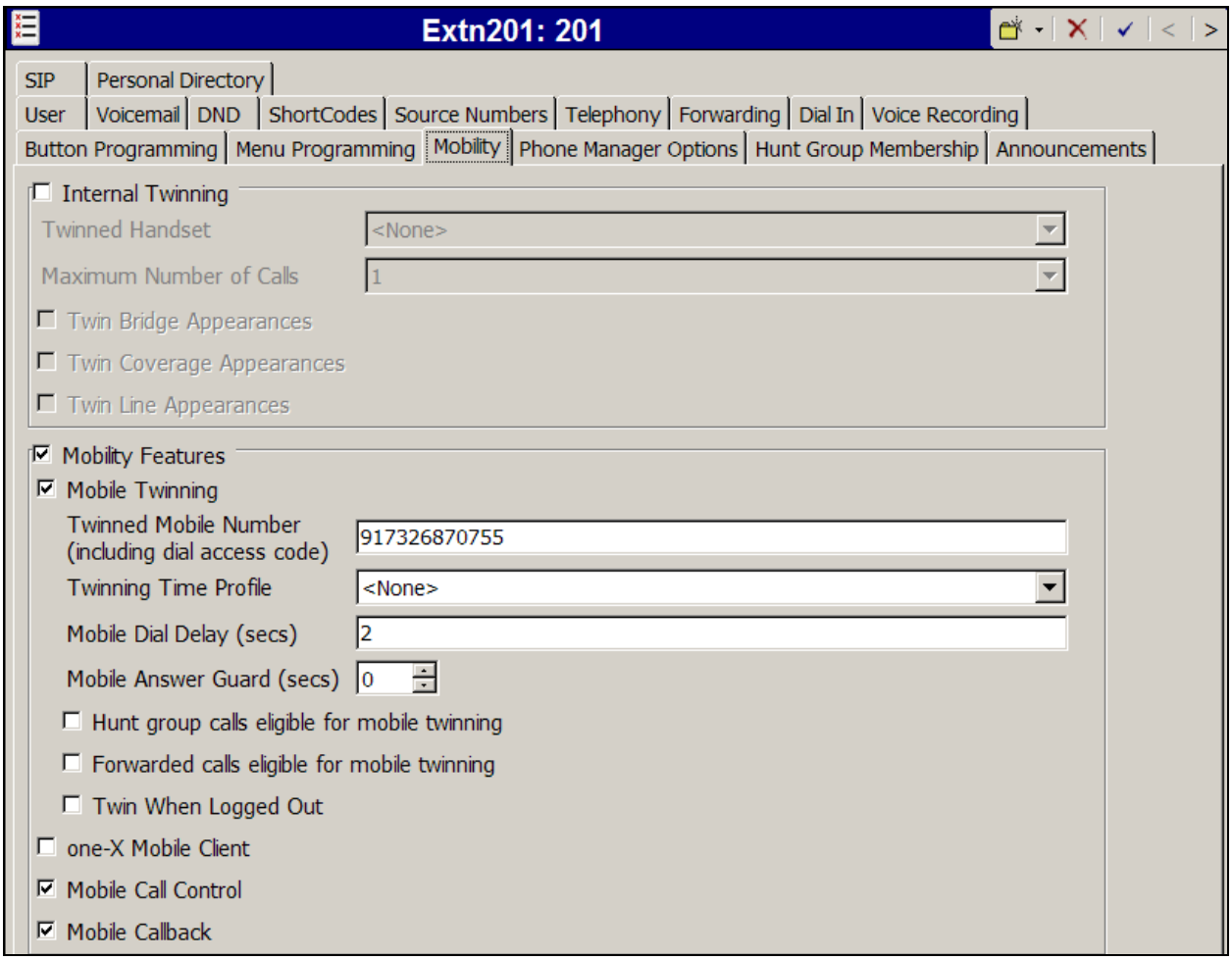
tested using a Verizon-provided Toll Free number. In the case of mobile callback, a Verizon provided toll-free number was used to call in to IP Office and hang up. The mobile callback outbound leg used the Verizon IP Trunk service provisioned in [VZB-IPT-IPOR6].



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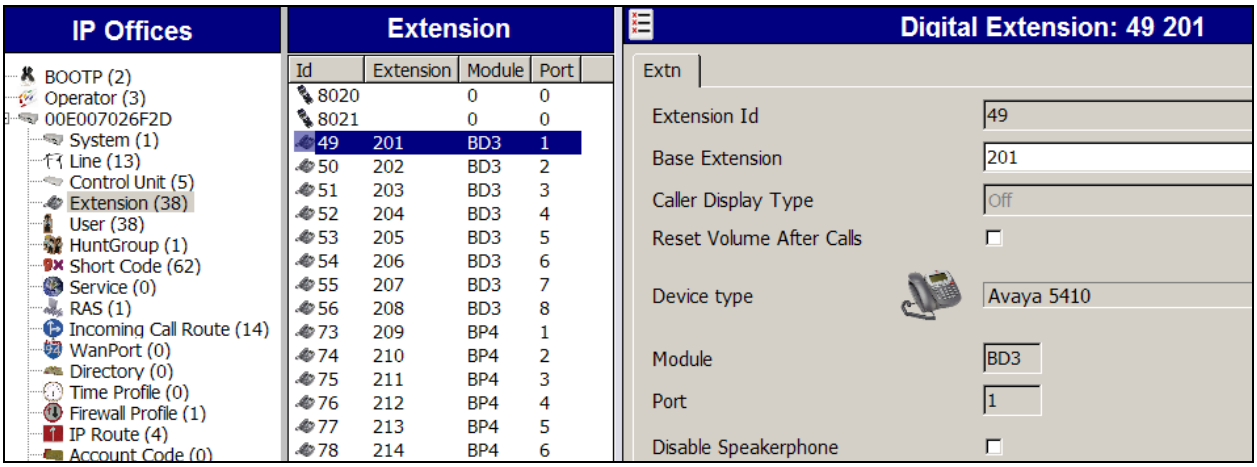
The following screen shows the Extension information for this user. To view, select **Extension** from the Navigation pane, and the appropriate extension from the Group pane. As stated in Section 5.1, the Avaya 5410 telephone user with extension 201 is connected to port 1 of the digital module.



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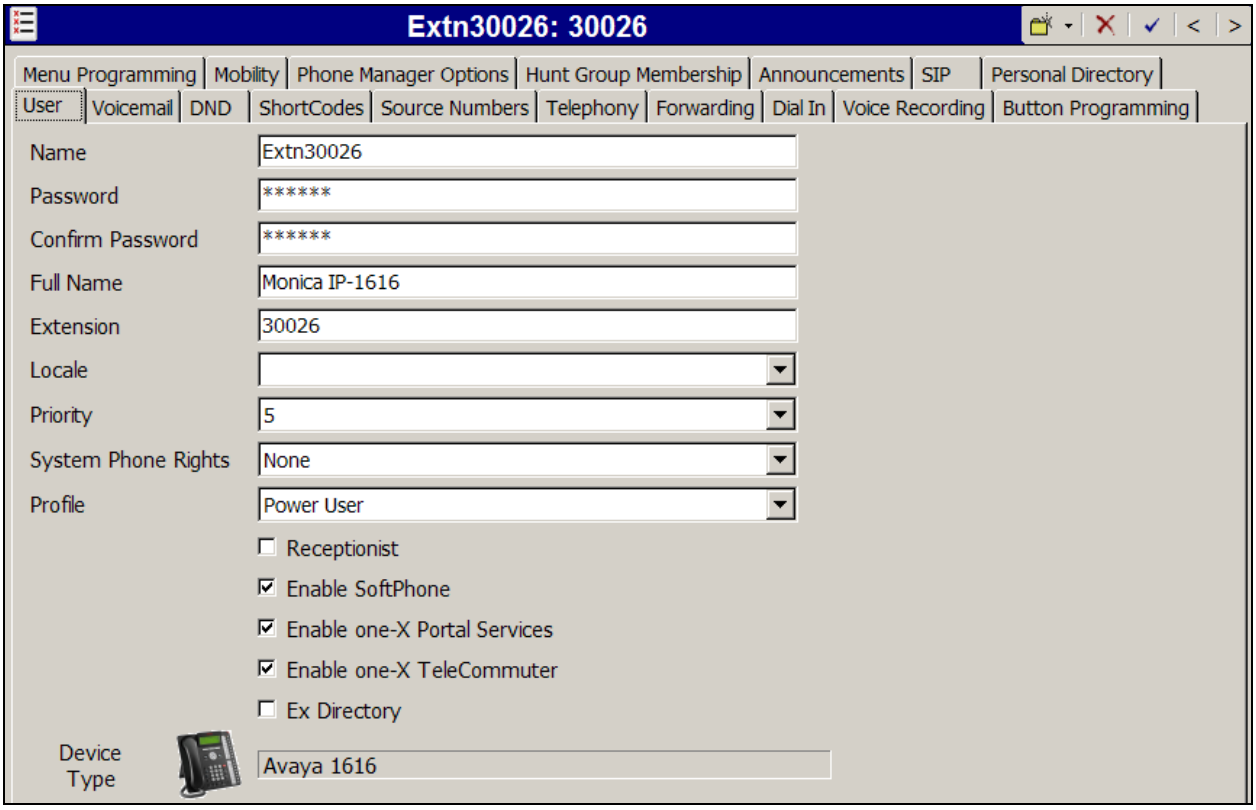
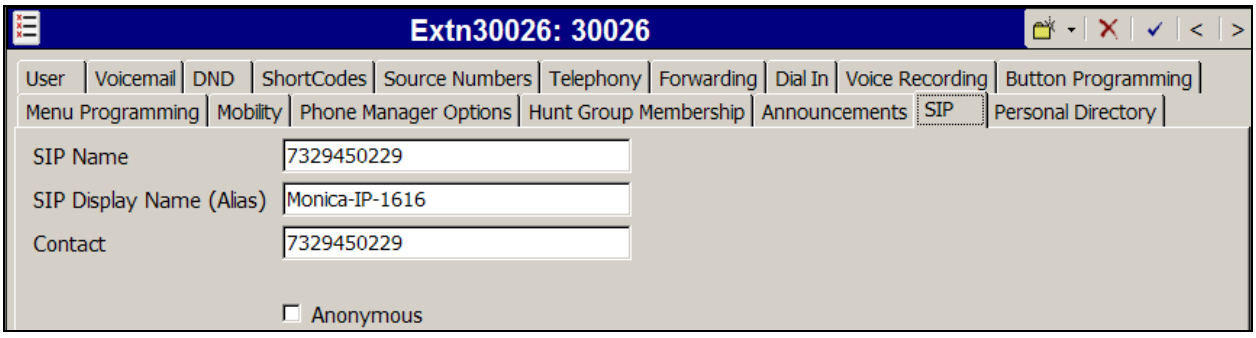
5.5.2. User 30026

The following screen shows the **User** tab for User 30026. This user corresponds to an Avaya 1616 IP Telephone that will be granted "Power User" and Avaya IP Office Softphone features. The **Profile** parameter is set to "Power User". The **Enable Softphone** box is checked, along with other advanced capabilities.



Like the user with extension 201, the **SIP** tab for the user with extension 30026 is configured with a **SIP Name** and **Contact** specifying the user's Verizon Business DID number using the Verizon IP Trunk Service, as detailed in [VZB-IPT-IPOR6].



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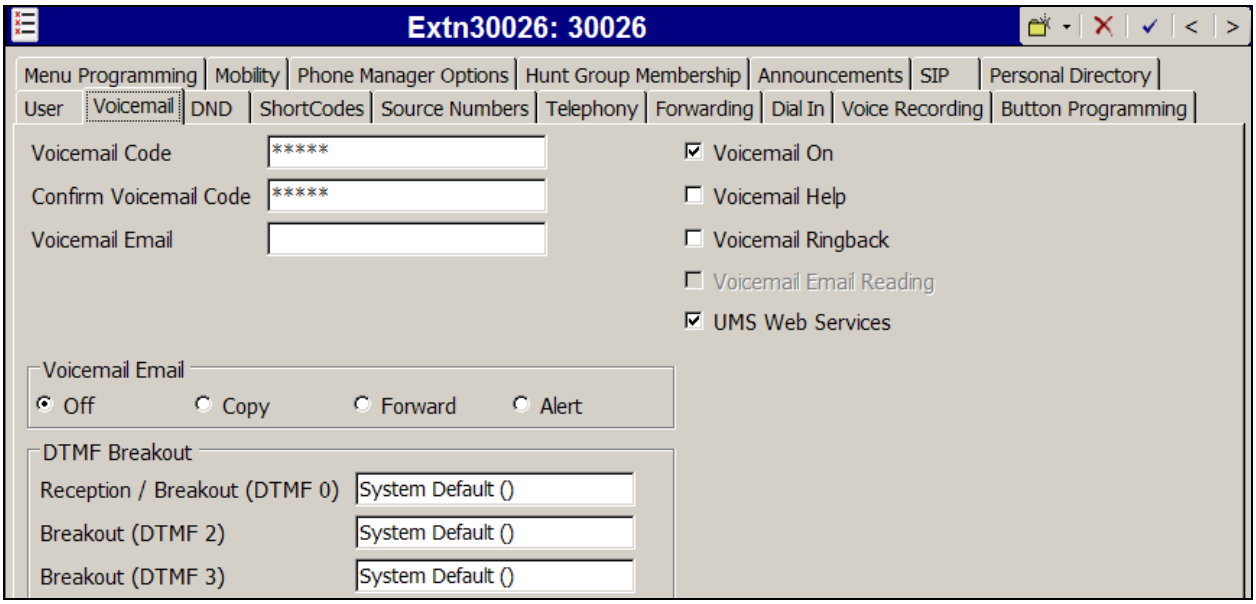
The following screen shows the Voicemail tab for the user with extension 30026. The **Voicemail On** box is checked, and a voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters. In the verification of these Application Notes, incoming calls from the 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 IP Contact Center 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).



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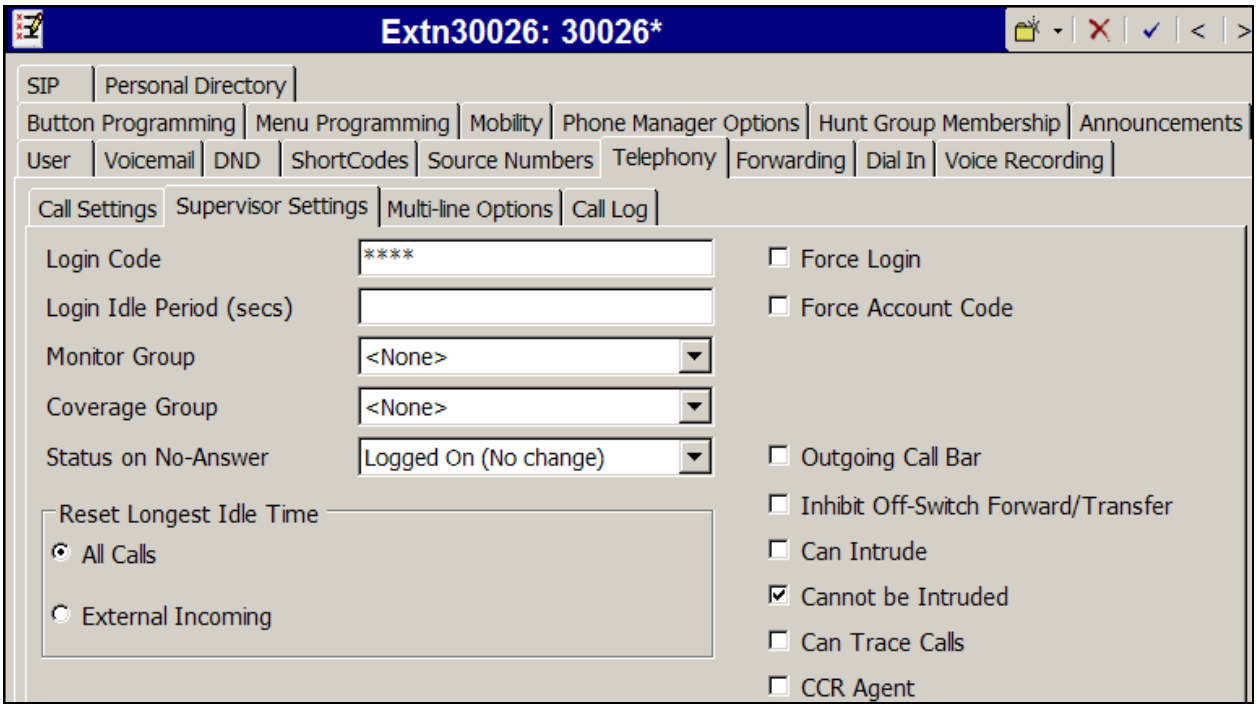
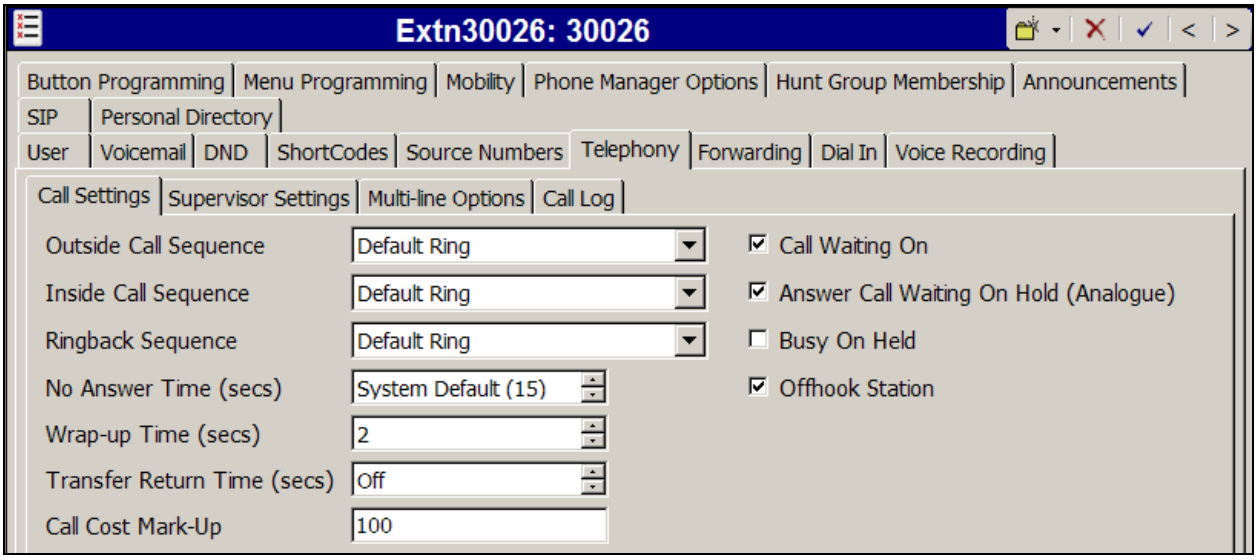
Select the **Telephony** tab followed by the **Supervisor Settings** tab as shown below. To allow hot desking, enter a **Login Code**.



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).



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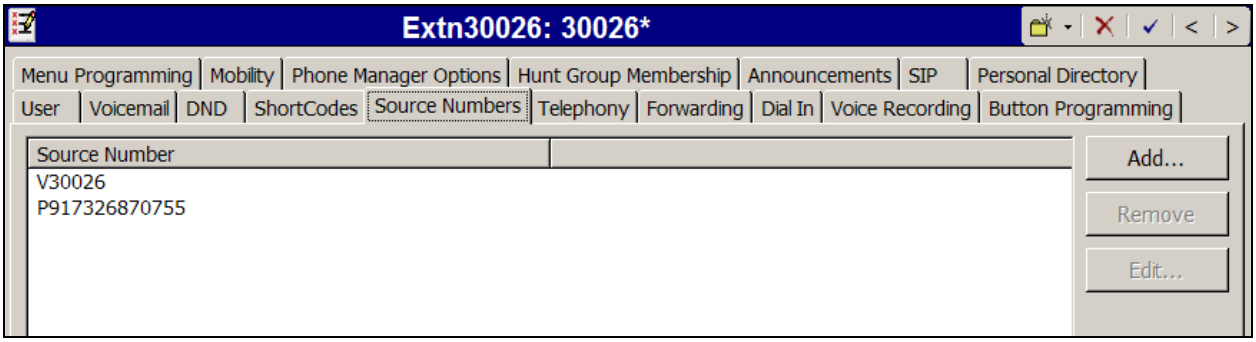
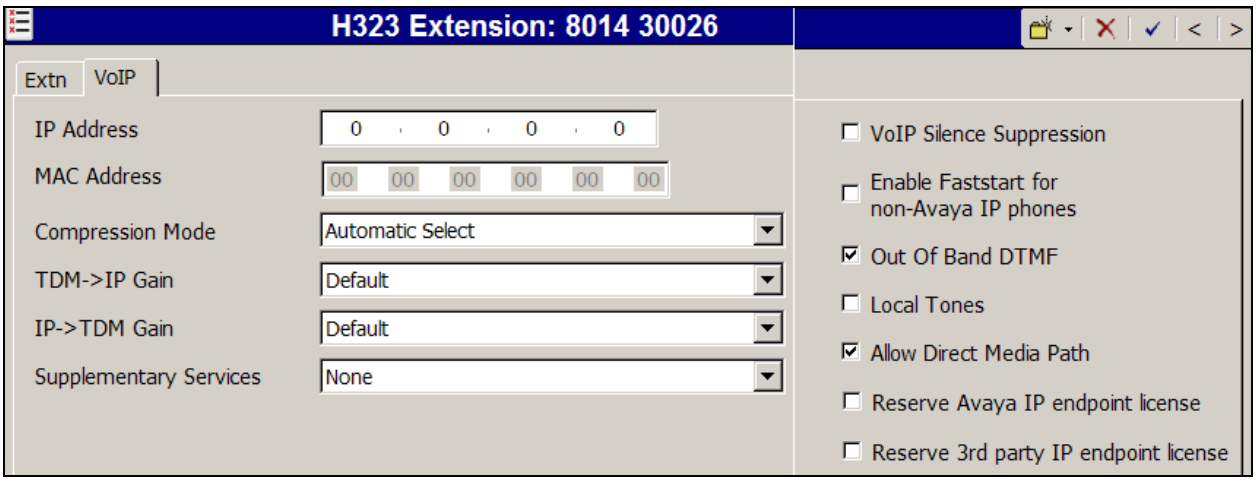
One method of configuring the number that Voicemail Pro will "call back" when a voice message is left for a user is shown below. In the approach shown here, the number is configured by the administrator using IP Office Manager. Note that callback numbers and options may also be configured by the end user using the Telephone User Interface of Voicemail Pro (i.e., enter "6" from the initial activity menu). The following screen shows the **Source Numbers** tab for the user with extension 30026. Although the Voicemail Pro configuration is beyond the scope of these Application Notes, the "Callback" feature has been enabled on Voicemail Pro, and the Source Number "P917326870755" has been added. With this configuration, when a message is left in this user's Voicemail Pro mailbox, a "callback" call will be initiated to "917326870755". As detailed in [VZB-IPT-IPOR6], the callback call will be sent to the Verizon Business IP Trunk service, and the From and Contact headers in the SIP INVITE will be populated with the information configured in the **System → Voicemail** tab shown in Section 5.3.3.

To add a new Source Number, Press the **Add…** button to the right of the list of any previously configured Source Numbers. To edit an existing Source Number, select the Source Number from the list, and click **Edit…** When finished, click **OK**.



The following screen shows the Extension information for this user, simply to illustrate the **VoIP** tab available for an IP Telephone. To view, select **Extension** from the Navigation pane, and the appropriate extension from the Group pane. Select **VoIP** in the Details pane.



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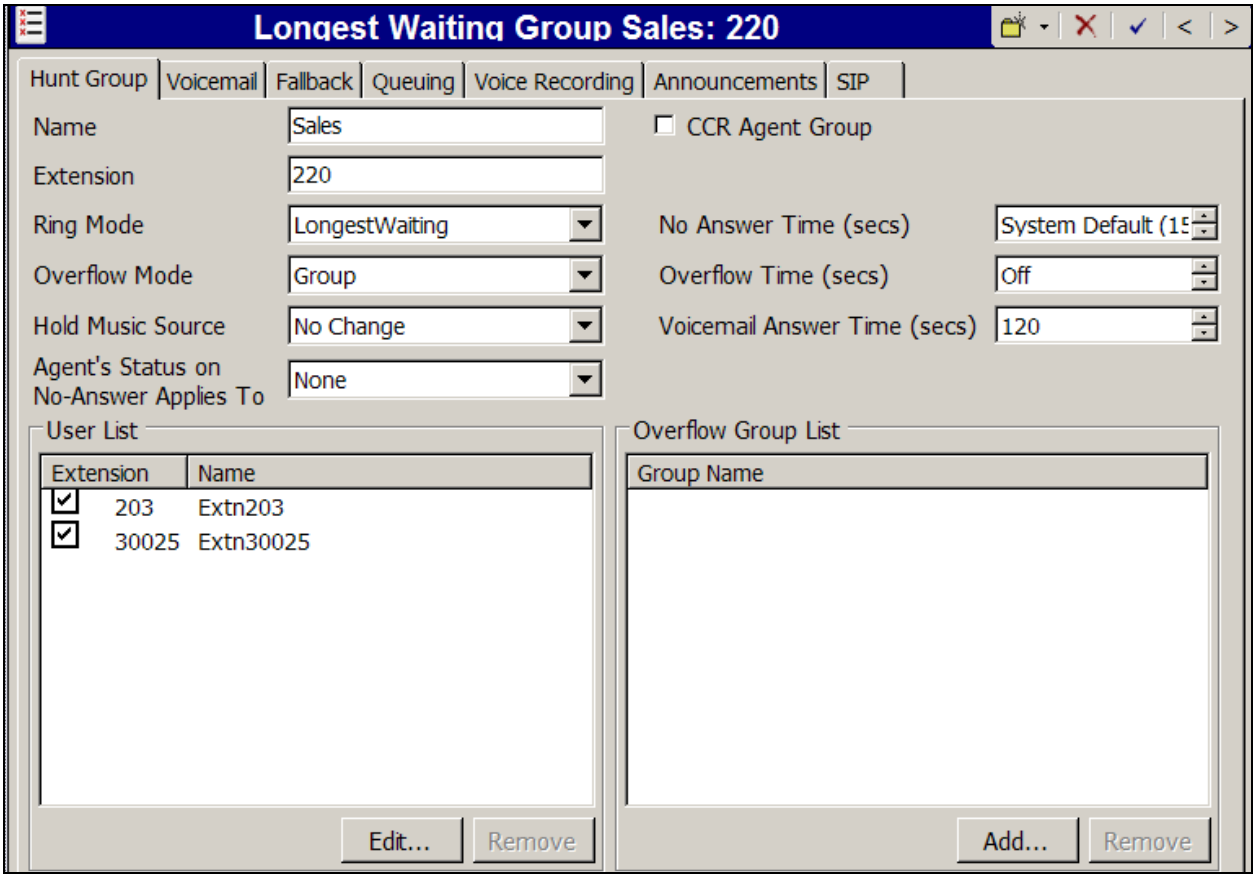
5.5.3. Hunt Groups

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** 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 220 and **Name** "Sales". This hunt group was configured to contain the two telephones x203 and x30025 in **Figure 1**. The **Ring Mode** was set to "LongestWaiting" (i.e., "longest waiting", most idle user receives next call). Click the **Edit** button to change the **User List**.



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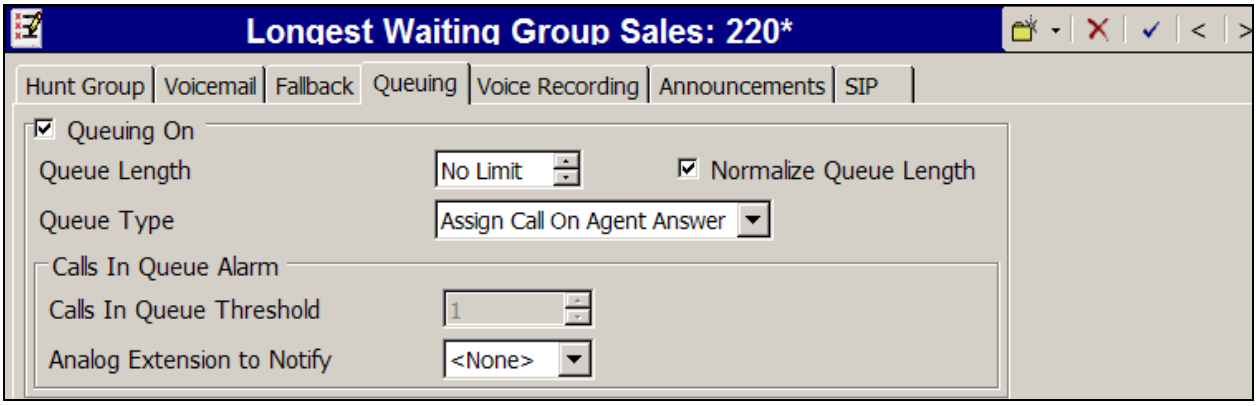
The following screen shows the **Queuing** tab for hunt group 220. In the sample configuration, the hunt group was configured to allow queuing so that incoming Verizon toll-free calls could be queued when all the members of the hunt group were busy on calls. In the testing associated with these Application 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.



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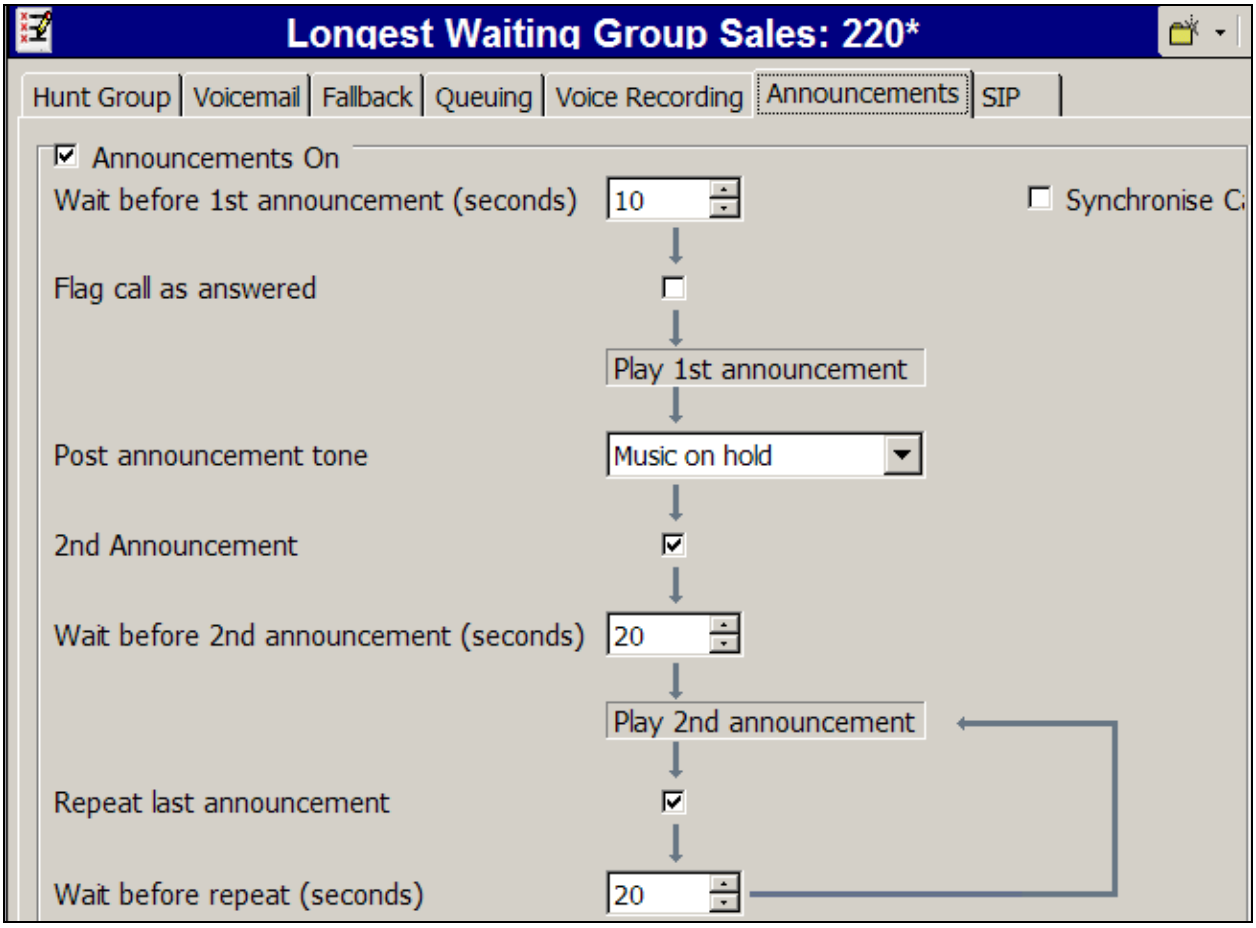
The following screen shows the **Announcements** tab for hunt group 220. In the sample configuration, when a call arrives when all members of the hunt group are busy on calls, the caller will first hear ring back tone. If a member of the hunt group does not become available after 10 seconds, the call 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.



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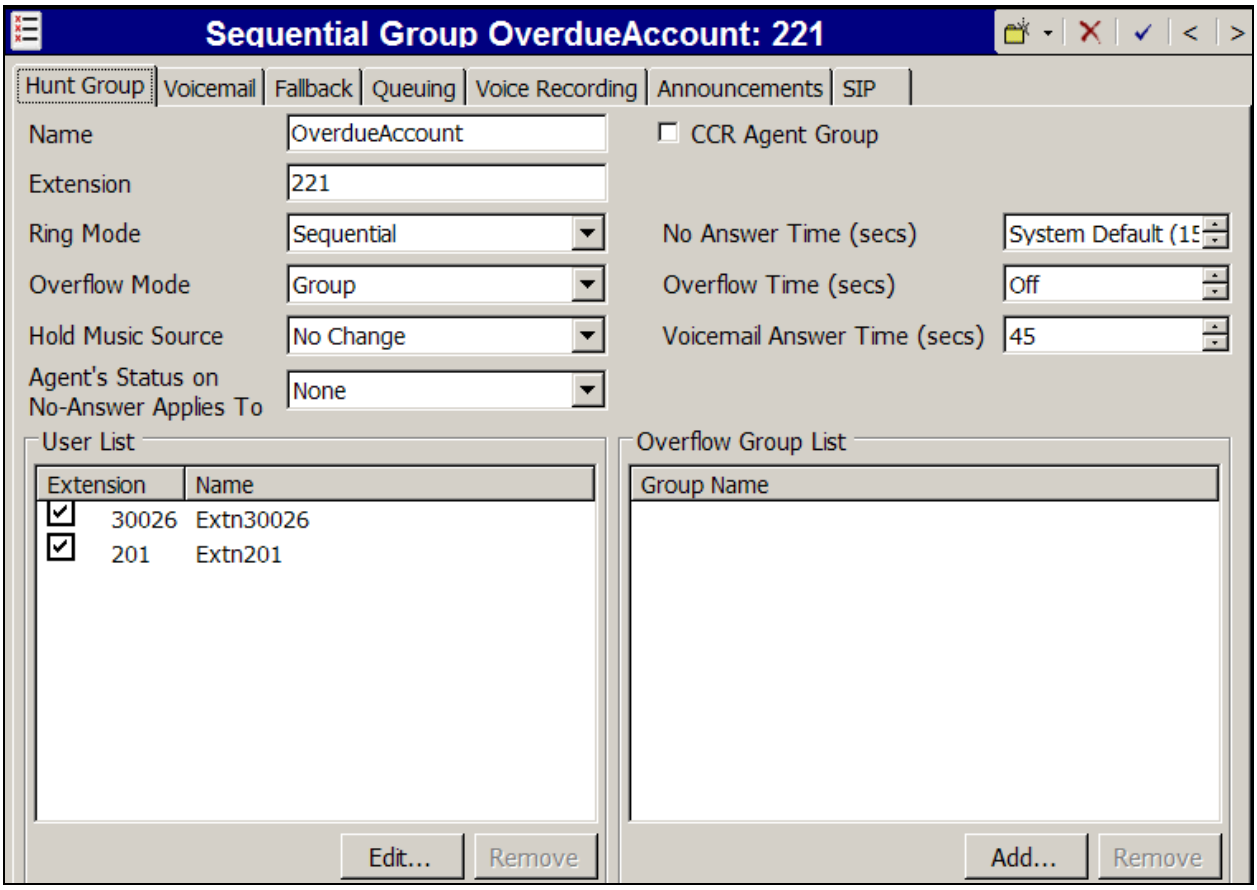
The following screen shows the **Hunt Group** tab for another hunt group 221. This hunt group was configured to contain the two telephones x201 and x30026. In the sample configuration, this hunt group was used to illustrate the ability of IP Office to route calls made to the same toll-free number to different destinations, depending on the caller id. Using the example in Section 5.7.3, calls to the toll-free number corresponding to the "Sales" hunt group from a specific caller with an "overdue account" was sent to the "Overdue Accounts" hunt group rather than the Sales hunt group.



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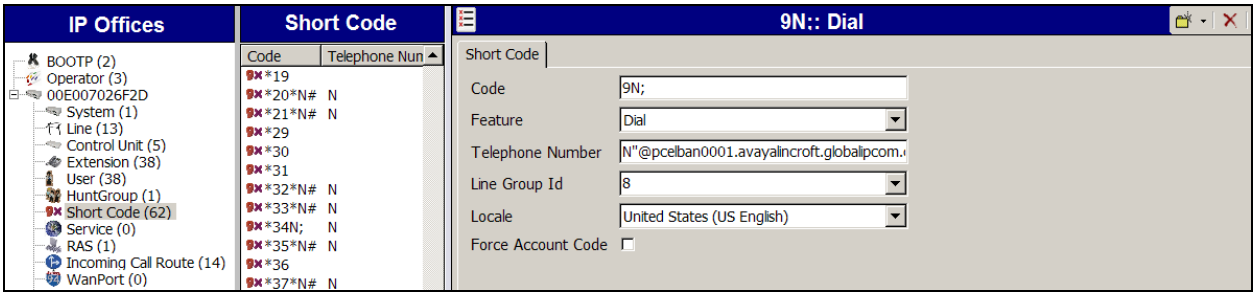
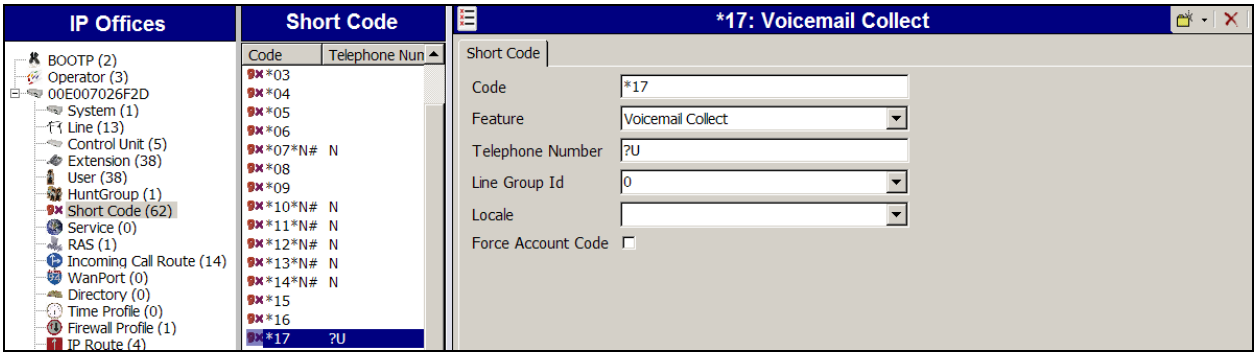
5.6. Short Codes

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

In the screen shown below, the short code "9N;" is repeated from Section 3.6 of reference [VZB-IPT-IPOR6]. The **Code** parameter is set to "9N;**"**. 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 8, matching the number of the **Outgoing Group** configured on the **SIP URI** tab of SIP Line 7 to Verizon Business documented in [VZB-IPT-IPOR6]**.** Although Verizon IP Contact Center, 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 IP Trunk SIP Line. In addition, inbound IP Contact Center toll-free calls used to access the IP Office mobile call back feature can have the call back occur using the Verizon IP Trunk SIP Line. For more information on outbound calls, short codes, and ARS, see reference [VZB-IPT-IPOR6]**.**



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.



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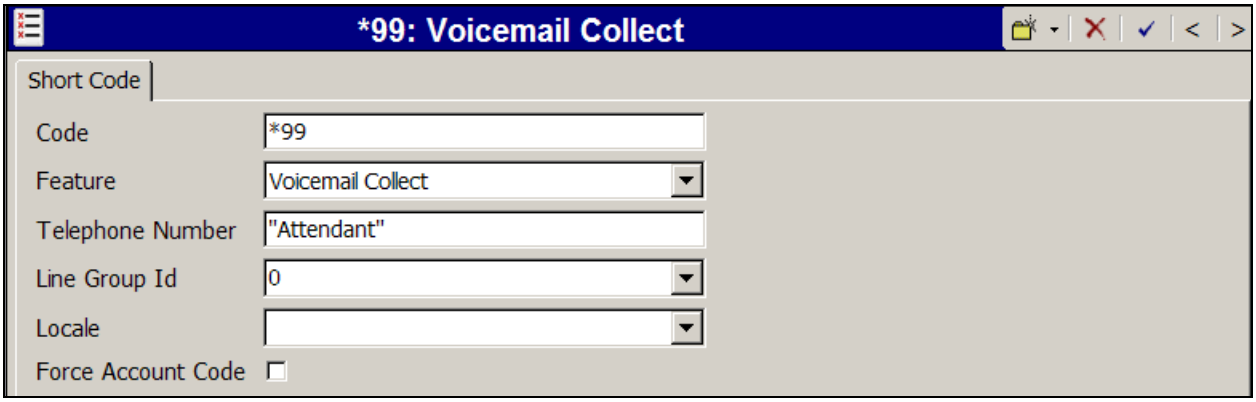
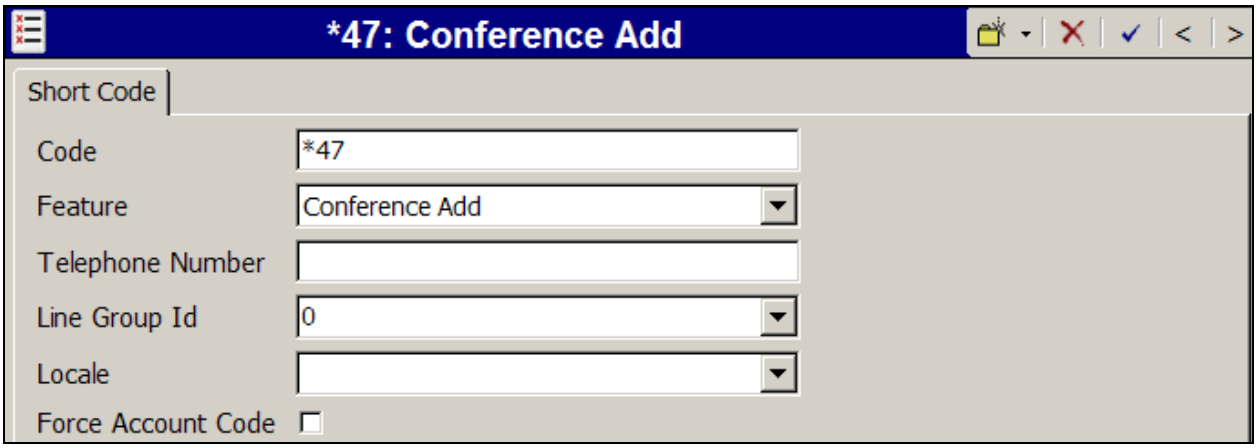
The following screen illustrates another short code. In this case, the **Code** "*99" is defined for **Feature** "Voicemail Collect" and Telephone Number "Attendant". This short code will be used as a means to allow a Verizon toll-free number to be programmed to route directly to a Voicemail Pro "Attendant" (auto-attendant) module, via inclusion of this short code as the destination of an Incoming Call Route. See Section 5.7.



The following screen illustrates another short code. In this case, the **Code** "*47" is defined for **Feature** "Conference Add". In the verification of these Application Notes, "*47" was used by mobile telephones to create a conference via a DTMF sequence using the IP Office mobile call control feature.



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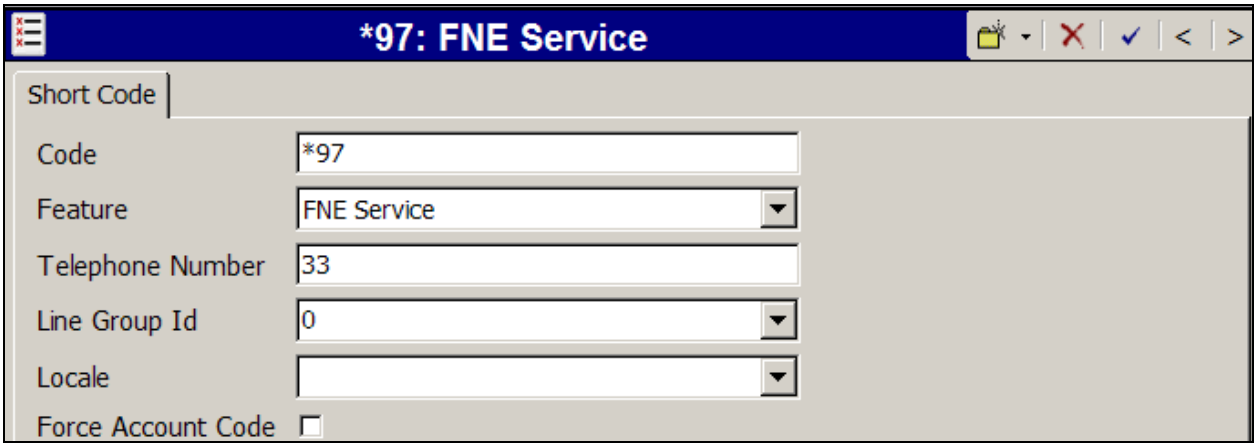
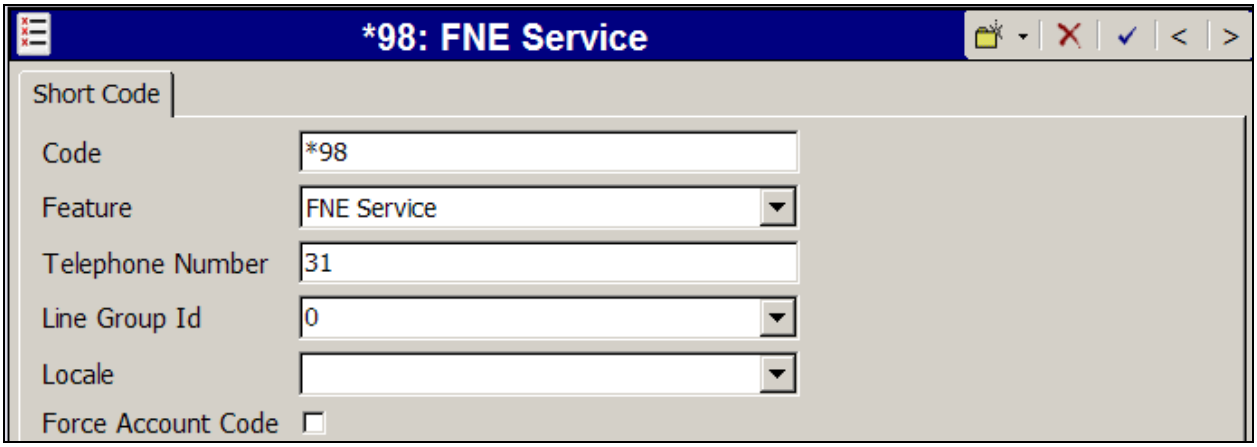
The following screen illustrates another short code. In this case, the **Code** "*97" is defined for **Feature** "FNE Service" and **Telephone Number** "33" for "Mobile Callback". In the verification of these Application Notes, "*97" was used as the destination of an Incoming Call Route for a Verizon toll-free number. This enabled toll-free calls from a configured twinning destination to be dialed, and then hung up by the toll-free caller while hearing ring back. IP Office would then call back the caller using the Verizon IP Trunk Service.



The following screen illustrates another short code. In this case, the **Code** "*98" is defined for **Feature** "FNE Service" and **Telephone Number** "31" for "Mobile Call Control". In the verification of these Application Notes, "*98" was used as the destination of an Incoming Call Route for a Verizon toll-free number. This enabled toll-free access to mobile call control from configured twinning destinations, allowing the mobile user to make calls as if the calls were made from the user's IP Office extension in the office.



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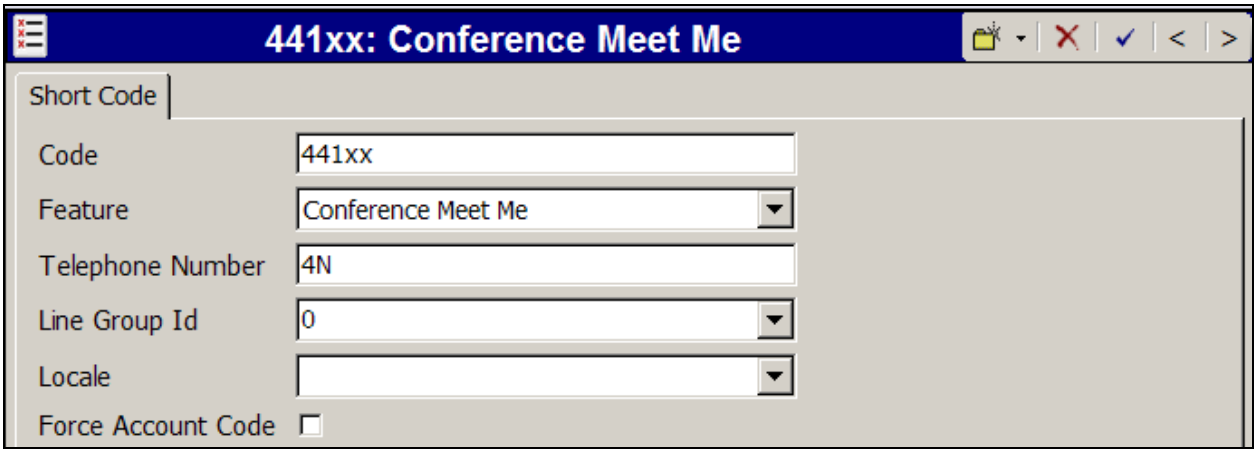
The following screen illustrates another short code. In this case, the **Code** "441xx" is defined for **Feature** "Conference Meet Me" and **Telephone Number** "4N". In the verification of these Application Notes, this short code was used in conjunction with a Voicemail Pro module named "MeetMeConf". 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 441xx to join the corresponding conference with ID 4xx.



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. The 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.

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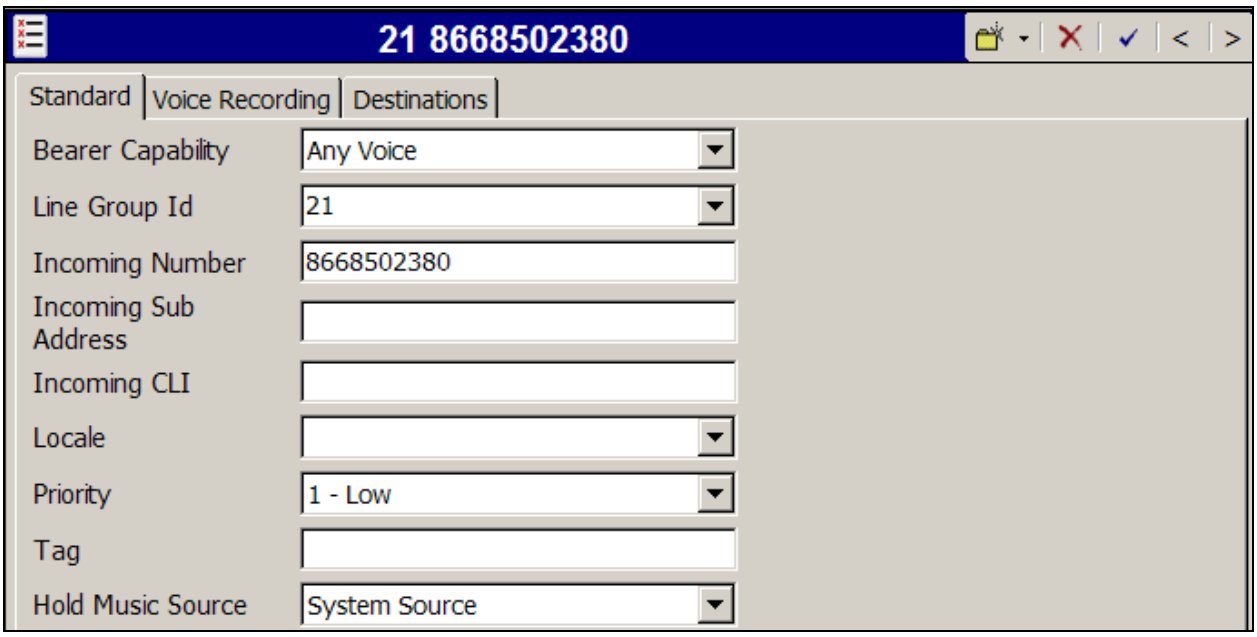
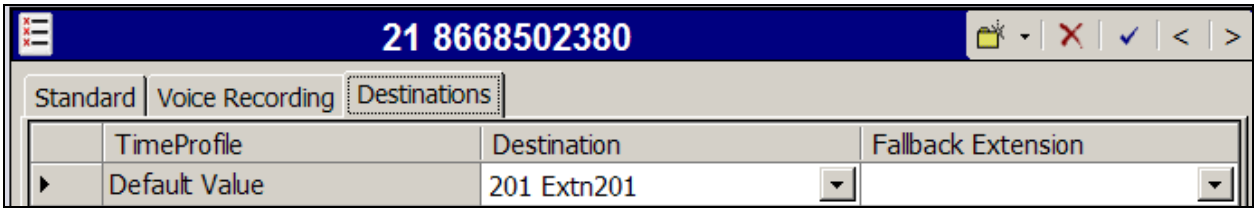
5.7.1. Incoming Call Route to a Specific Telephone Extension

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 IP Contact Center Service, in Section 5.4.



Select the **Destinations** tab. From the **Destination** drop-down, select the 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 201.



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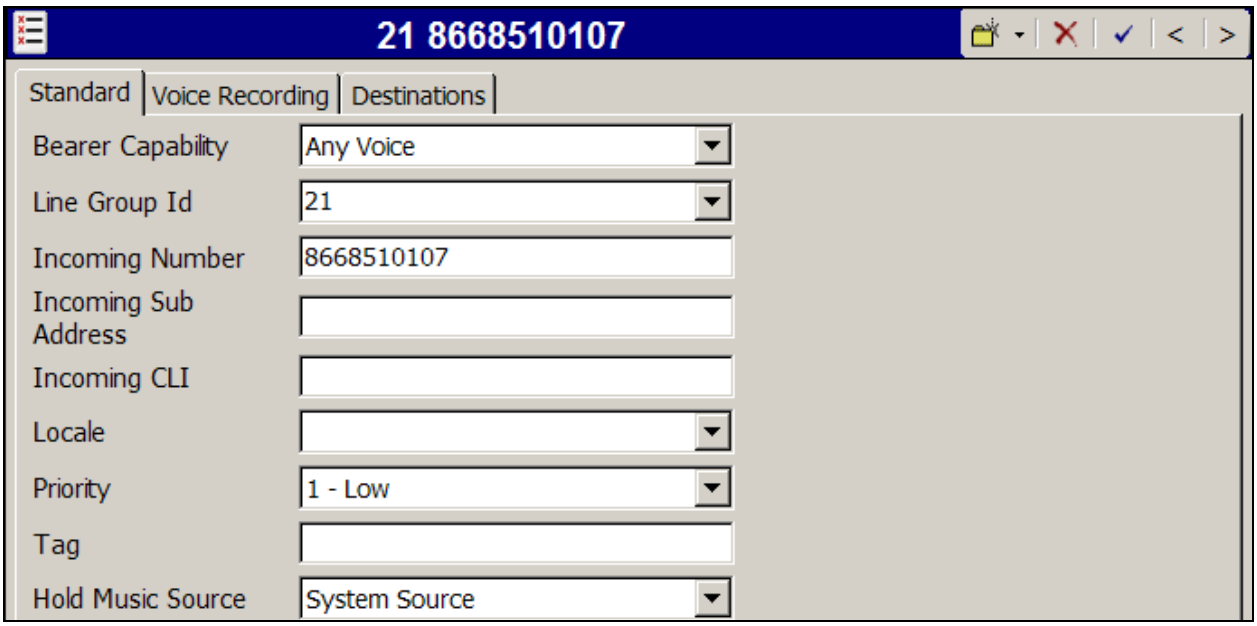
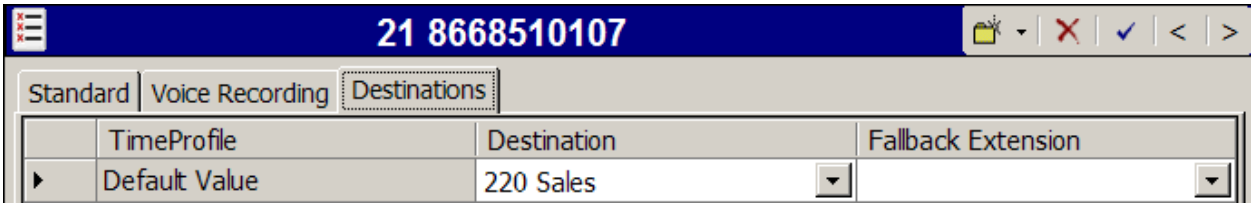
5.7.2. Incoming Call Routes to a Hunt Group by Dialed Toll-Free Number

In the screen shown below, an incoming call route for **Incoming Number** "8668510107" 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.



Select the **Destinations** tab. From the **Destination** drop-down, select the destination to receive the call when an arbitrary PSTN user dials 8668510107. As shown in **Table 1**, 8668510107 is the toll-free number associated with IP Office hunt group extension 220, the "Sales" hunt group.



5.7.3. 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** "8668510107" and **Incoming CLI** "7326870755" 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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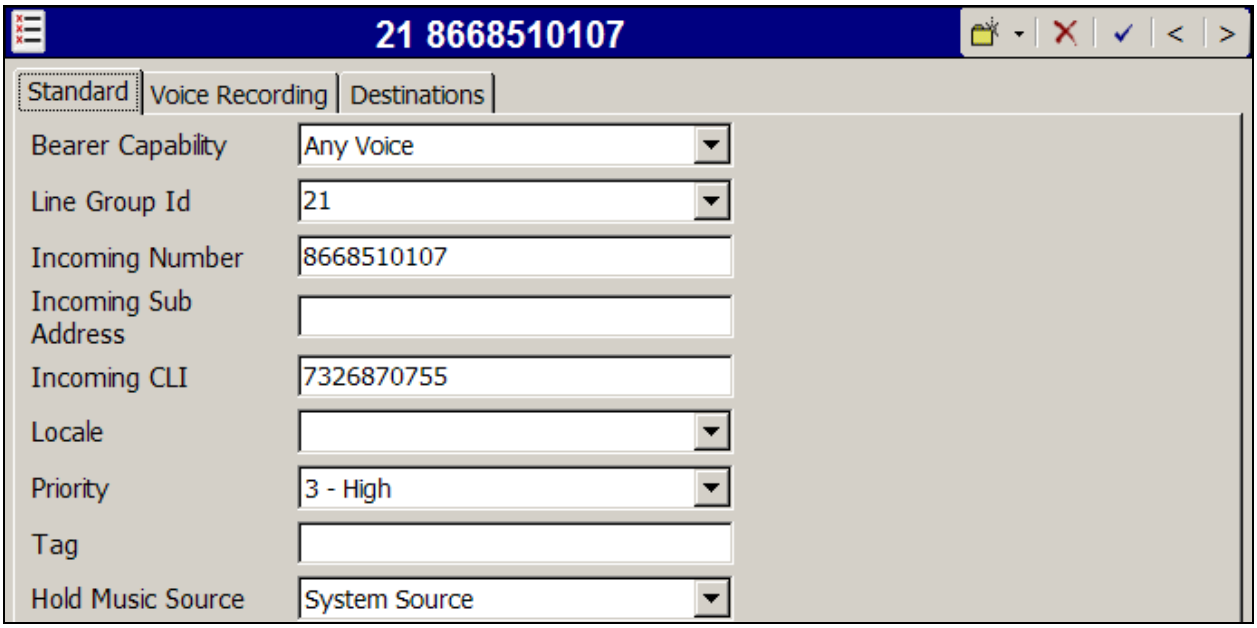
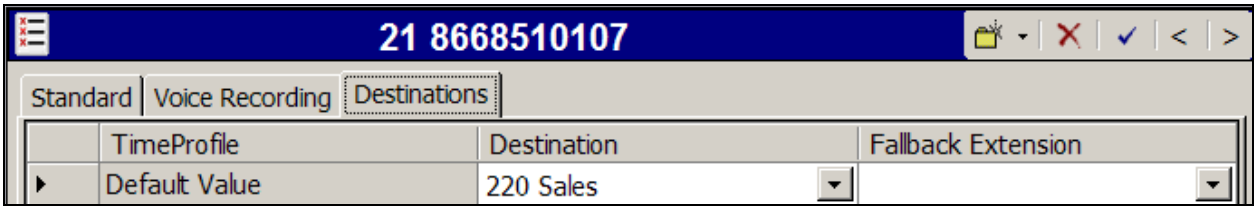
caller ID “7326870755”. In this case, to allow this caller to be treated with priority when calling in for “Sales”, the **Priority** field is set to “3 – High”.



Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when PSTN user 7326870755 dials 8668510107. In this case, the **Destination** is also the hunt group “220 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.



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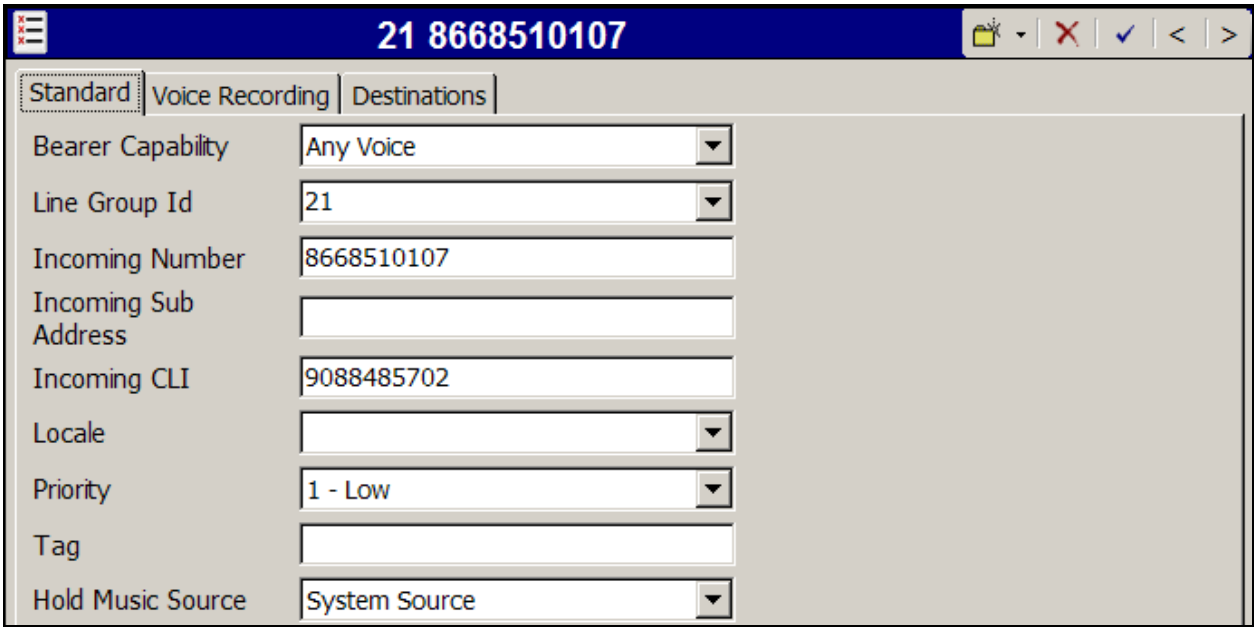
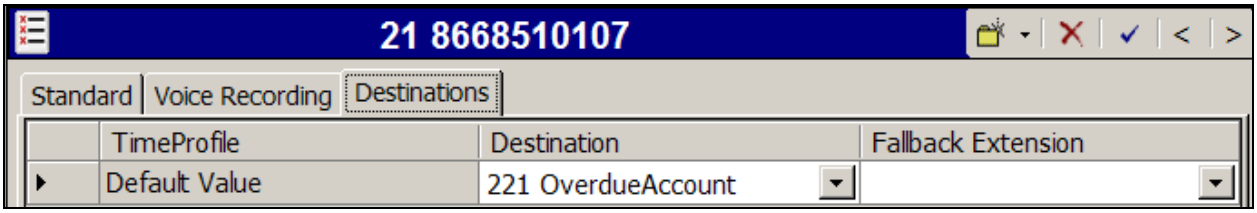
In the screen shown below, the incoming call route for **Incoming Number** "8668510107" and **Incoming CLI** "9088485702" is illustrated. The **Line Group Id** is 21, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in Section 5.4. Note that the **Incoming Number** is the same as the toll-free number configured above. This route will be used for calls to the toll-free number specifically from a caller with caller ID "9088485702".



In this case, assume that the business receiving the toll-free call knows that based on prior dealings, this caller should not be routed to the Sales hunt group, but rather to a different hunt group associated with overdue accounts.



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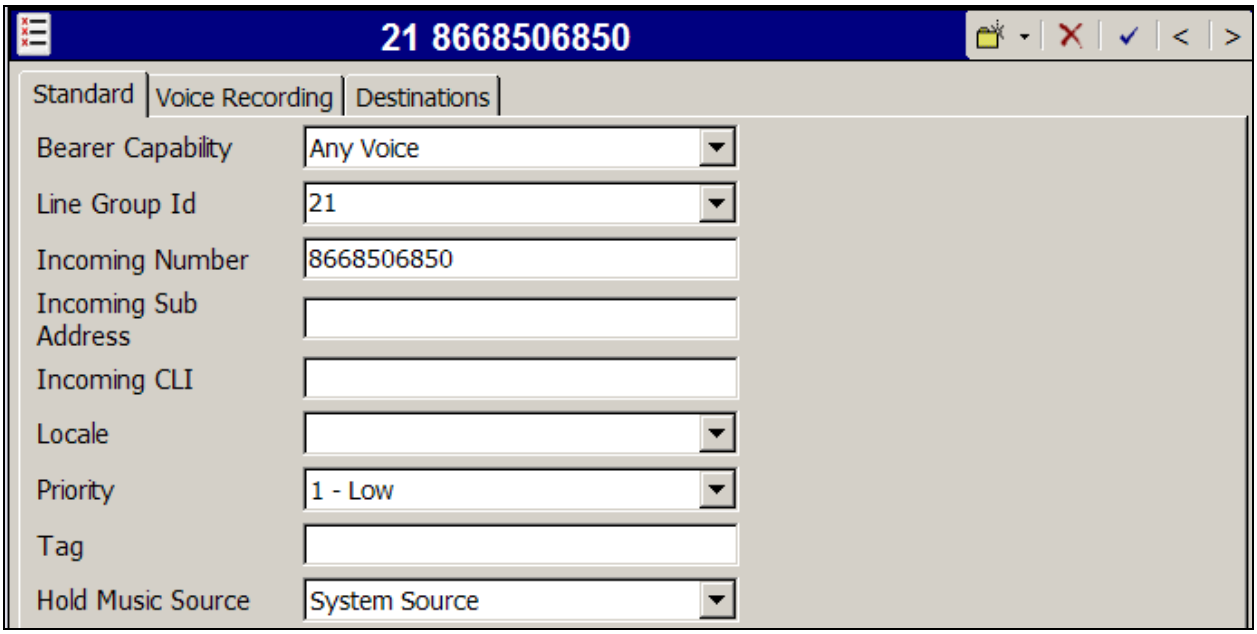
5.7.4. Incoming Call Routes to Various IP Office Features

In the sample configuration, the incoming call route for **Incoming Number** "8668506850" was varied to test different destination features, such as Voice Mail, Meet-Me Conferencing, Mobile Call Control, and Mobile Call Back, 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.
- *99" (short code for accessing an "Attendant" module on Voicemail Pro, as shown in Section 5.6). With this destination, an incoming call to 8668506850 will be delivered directly to the Voicemail Pro "Attendant" application, which will allow the caller to be prompted with announcements, navigate via DTMF, and transfer the call to IP Office users.
- "*97" (short code for accessing the Mobile Call Back feature, as shown in Section 5.6) With this destination, an incoming call to 8668506850 from configured mobile callers will be provided ring back tone. When the caller hangs up, IP Office will call the user back and provide dial tone for access to IP Office mobile call control features upon answer.

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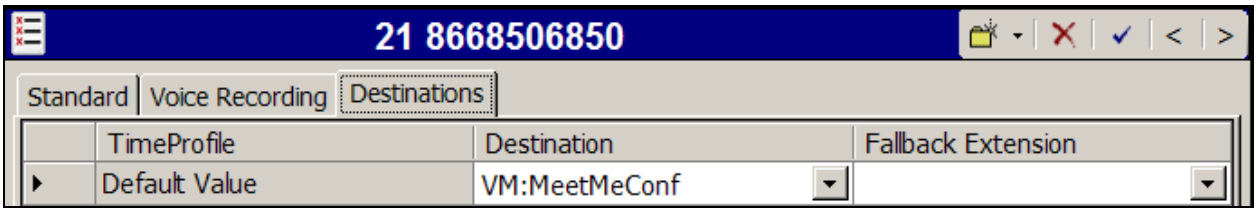
- "*98" (short code for accessing the Mobile Call Control feature directly, as shown in Section 5.6) With this destination, an incoming call to 8668506850 from configured mobile callers will be provided dial tone to make calls from the mobile phone as if the user were using their IP Office extension.
- "VM:MeetMeConf" as shown below. With this destination, an incoming call to 8668506850 is delivered to a Voicemail Pro module named "MeetMeConf" that prompts the user via a recorded announcement to enter a conference ID and password/PIN. Upon successful validation of the conference ID and password/PIN, Voicemail Pro transfers the caller to the appropriate conference ID on IP Office.

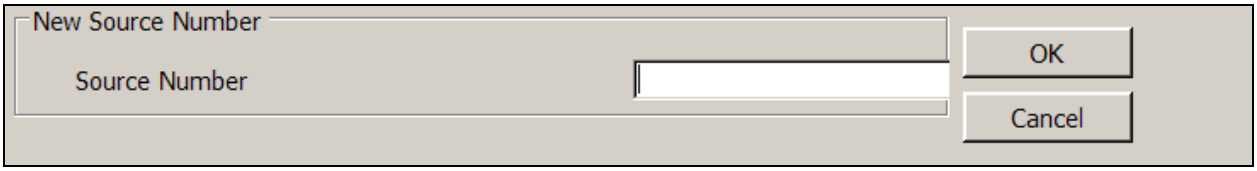


5.8. SIP Options Frequency

In Section 5.4.1, the SIP Line to Verizon Business is shown with the **Check OOS** box checked. In the sample configuration, IP Office periodically checks the health of the SIP Line by sending a SIP OPTIONS message. If there is no response, IP Office can mark the trunk out of service.

If a customer wishes to control how often SIP OPTIONS messages are sent by IP Office, a NoUser Source Number can be configured as follows. This configuration complements the configuration presented in Section 5.3.2 and Section 5.4.

From the Navigation pane, select **User**. From the Group pane, scroll down past the configured users and select the user named **NoUser**. From the NoUser Details pane, select the tab **Source Numbers**. Press the **Add…** button to the right of the list of any previously configured Source Numbers. In the **Source Number** field shown below, type **SIP_OPTIONS_PERIOD=X.** X is a value (in minutes) representing a longer time than the interval configured (in seconds) in the **Binding Refresh Time**. In the sample configuration, the value used for X was 2 minutes. Click **OK**.



The source number **SIP_OPTIONS_PERIOD=2** should now appear in the list of Source Numbers as shown below.

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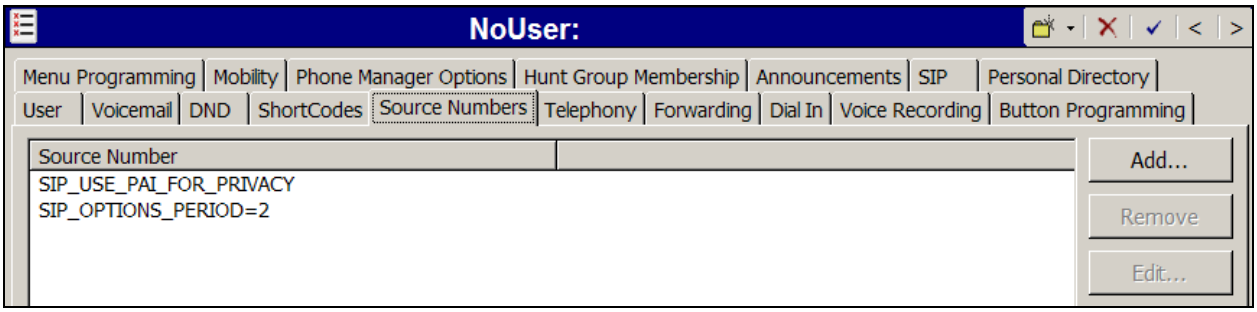
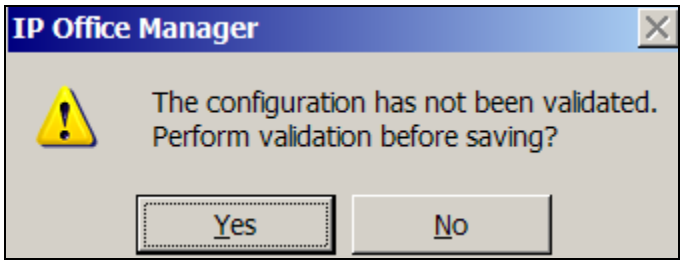


With this configuration, Binding Refresh Times of 30 seconds and 60 seconds were tested successfully. That is, IP Office sourced SIP OPTIONS every 30 or 60 seconds, depending on the value configured in the Binding Refresh Time, since the Binding Refresh Time was less than the value configured via the SIP_OPTIONS_PERIOD source number.

5.9. Saving Configuration Changes to IP Office

When desired, send the configuration changes made in IP Office Manager to the IP Office server, to cause the changes to take effect. Click the "floppy disk" icon that is the third icon from the left (i.e., common "save" icon with mouse-over help "Save Configuration File"). Click **Yes** to validate the configuration, if prompted.



Once the configuration is validated, a screen similar to 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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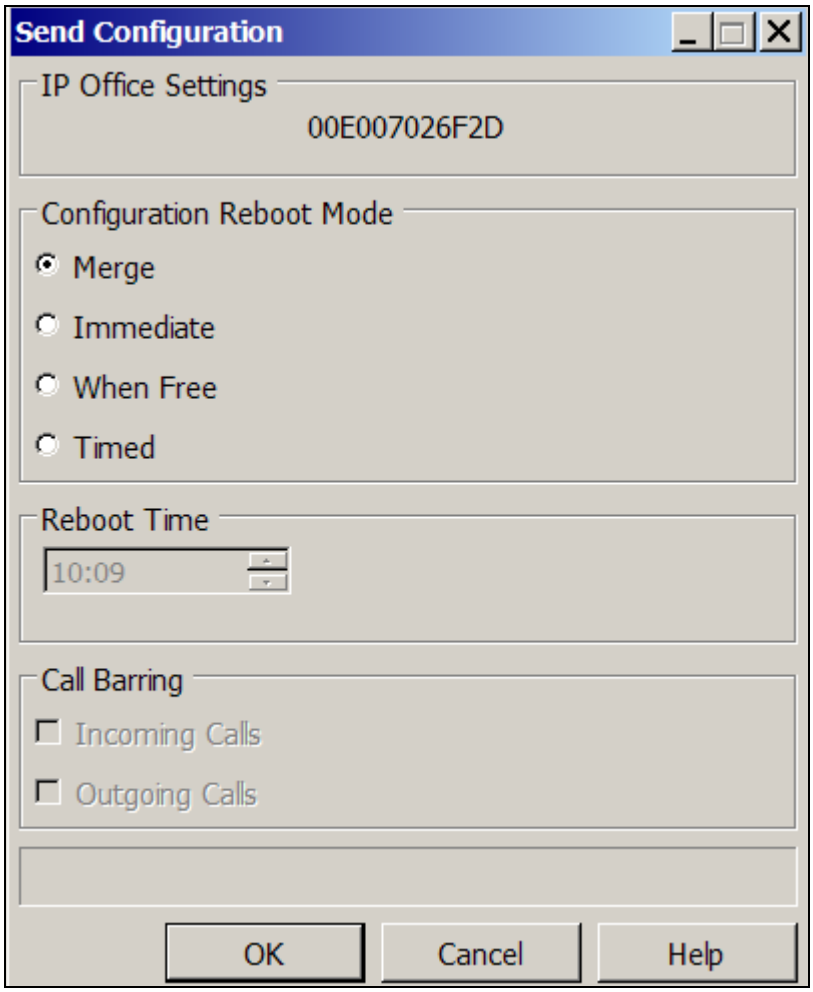

6. Verizon Business Configuration

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/contactcenter/ip/

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 Fully Qualified Domain Names (FQDN) *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 server. Verizon provided the following information for the compliance testing: the IP address and port used by the Verizon SIP SBC, and the toll-free numbers shown in **Figure 1** and **Table 1**. This information was used to complete the Avaya IP Office configuration shown in Section 5.

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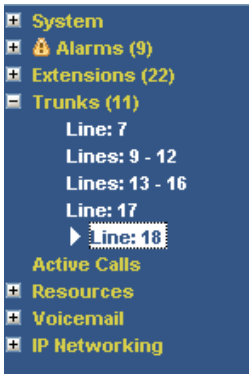
7. Verification Steps

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

7.1. System Status Application

The System Status application can be used to monitor or troubleshoot IP Office. The System Status application can typically be accessed from **Start → Programs → IP Office → System Status**. See reference [IPO-SYSSTAT] for more information. 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**.

After logging in, select **Trunks → Line: 18** from the left navigation menu. (SIP Line 18 is configured in Section 5.4 to communicate with the Verizon IPCC Service.)



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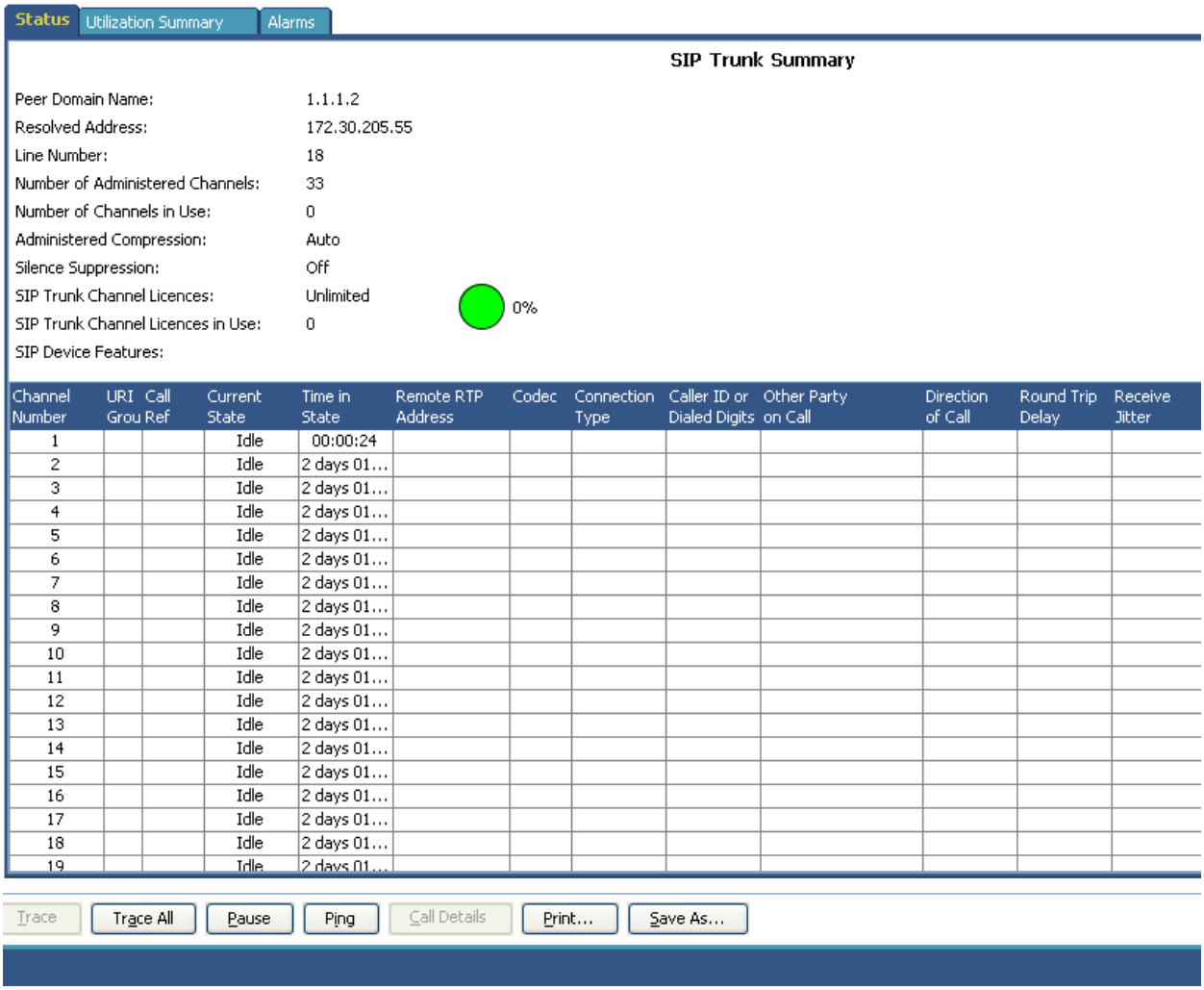
A screen such as the following is displayed. In the lower left, the **Trace All** button may be pressed to display tracing information as calls are made using this SIP Line.



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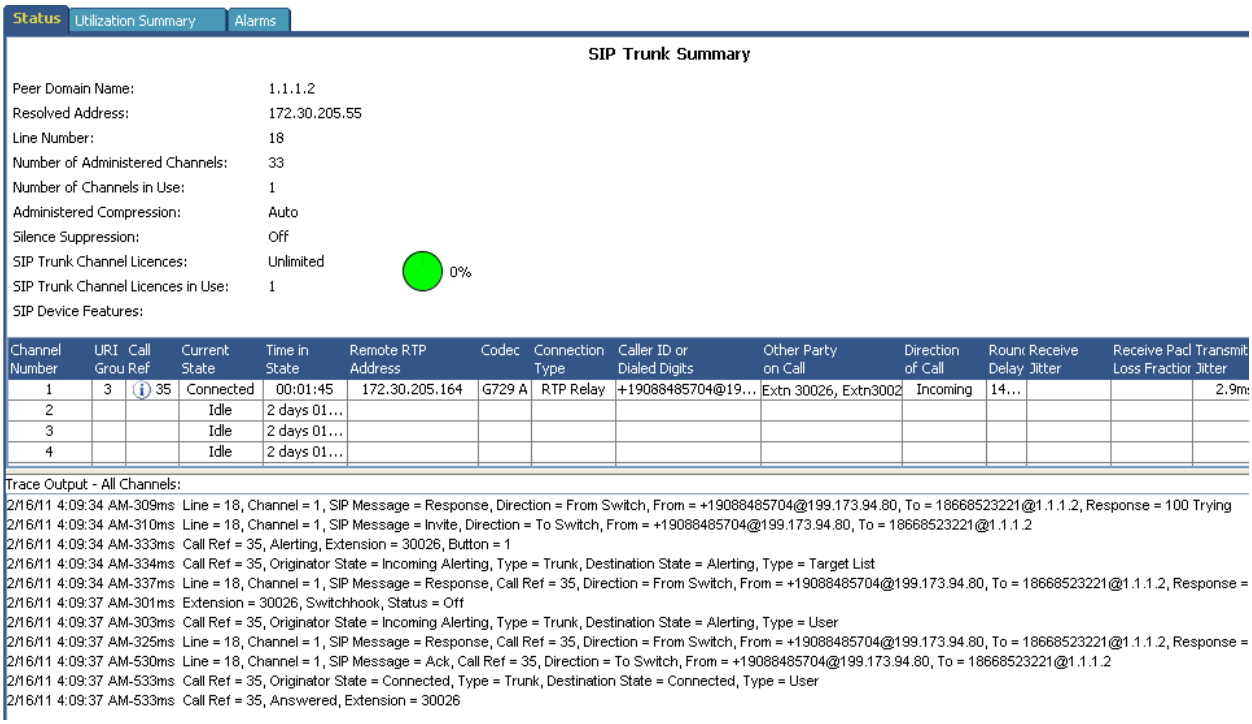
The following screen shows an example inbound call where PSTN caller 908-848-5704 called the Verizon IP Toll Free number 1-866-852-3221. The call is answered by the H.323 telephone with extension 30026. The lower portion of the screen reveals trace details that may be helpful when troubleshooting. Note that since this call was answered by an IP Telephone, the **Connection Type** shows "RTP Relay".



Status | Utilization Summary | Alarms

SIP Trunk Summary

Peer Domain Name:	1.1.1.2
Resolved Address:	172.30.205.55
Line Number:	18
Number of Administered Channels:	33
Number of Channels in Use:	1
Administered Compression:	Auto
Silence Suppression:	Off
SIP Trunk Channel Licences:	Unlimited
SIP Trunk Channel Licences in Use:	1
SIP Device Features:	

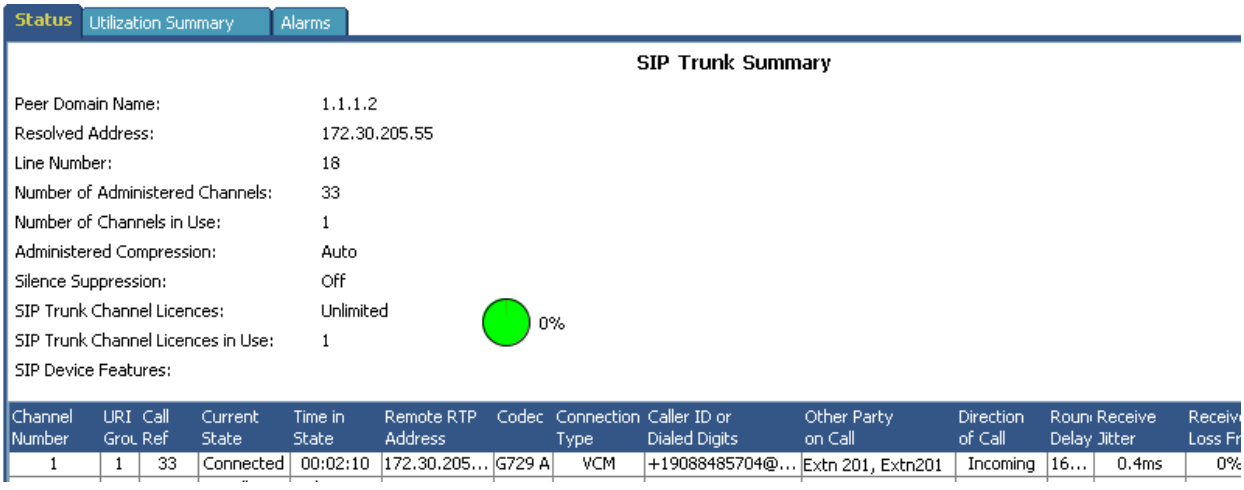
0%

Channel Number	URI Grou	Call Ref	Current State	Time in State	Remote RTP Address	Codec	Connection Type	Caller ID or Dialed Digits	Other Party on Call	Direction of Call	Roun Delay	Receive Jitter	Receive Pacl Loss Fractior	Transmit Jitter
1	3	35	Connected	00:01:45	172.30.205.164	G729 A	RTP Relay	+19088485704@19...	Extn 30026, Extn3002	Incoming	14...			2.9ms
2			Idle	2 days 01...										
3			Idle	2 days 01...										
4			Idle	2 days 01...										

Trace Output - All Channels:

2/16/11 4:09:34 AM-309ms Line = 18, Channel = 1, SIP Message = Response, Direction = From Switch, From = +19088485704@199.173.94.80, To = 18668523221@1.1.1.2, Response = 100 Trying
2/16/11 4:09:34 AM-310ms Line = 18, Channel = 1, SIP Message = Invite, Direction = To Switch, From = +19088485704@199.173.94.80, To = 18668523221@1.1.1.2
2/16/11 4:09:34 AM-333ms Call Ref = 35, Alerting, Extension = 30026, Button = 1
2/16/11 4:09:34 AM-334ms Call Ref = 35, Originator State = Incoming Alerting, Type = Trunk, Destination State = Alerting, Type = Target List
2/16/11 4:09:34 AM-337ms Line = 18, Channel = 1, SIP Message = Response, Call Ref = 35, Direction = From Switch, From = +19088485704@199.173.94.80, To = 18668523221@1.1.1.2, Response =
2/16/11 4:09:37 AM-301ms Extension = 30026, Switchhook, Status = Off
2/16/11 4:09:37 AM-303ms Call Ref = 35, Originator State = Incoming Alerting, Type = Trunk, Destination State = Alerting, Type = User
2/16/11 4:09:37 AM-325ms Line = 18, Channel = 1, SIP Message = Response, Call Ref = 35, Direction = From Switch, From = +19088485704@199.173.94.80, To = 18668523221@1.1.1.2, Response =
2/16/11 4:09:37 AM-530ms Line = 18, Channel = 1, SIP Message = Ack, Call Ref = 35, Direction = To Switch, From = +19088485704@199.173.94.80, To = 18668523221@1.1.1.2
2/16/11 4:09:37 AM-533ms Call Ref = 35, Originator State = Connected, Type = Trunk, Destination State = Connected, Type = User
2/16/11 4:09:37 AM-533ms Call Ref = 35, Answered, Extension = 30026

The following screen shows an example inbound call where PSTN caller 908-848-5704 called a Verizon IP Toll Free number when the call was answered by a digital telephone. Note that since this call was answered by a digital Telephone, IP Office compression resources must be used, and the **Connection Type** shows "VCM".



Status | Utilization Summary | Alarms

SIP Trunk Summary

Peer Domain Name:	1.1.1.2
Resolved Address:	172.30.205.55
Line Number:	18
Number of Administered Channels:	33
Number of Channels in Use:	1
Administered Compression:	Auto
Silence Suppression:	Off
SIP Trunk Channel Licences:	Unlimited
SIP Trunk Channel Licences in Use:	1
SIP Device Features:	

0%

Channel Number	URI Grou	Call Ref	Current State	Time in State	Remote RTP Address	Codec	Connection Type	Caller ID or Dialed Digits	Other Party on Call	Direction of Call	Roun Delay	Receive Jitter	Receiv Loss Fr
1	1	33	Connected	00:02:10	172.30.205...	G729 A	VCM	+19088485704@...	Extn 201, Extn201	Incoming	16...	0.4ms	0%

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7.2. System Monitor Application

The System Monitor application can also be used to monitor or troubleshoot. The System Monitor application can typically be accessed from **Start → Programs → IP Office → Monitor**. See reference [IPO-MON] for more information.

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**.

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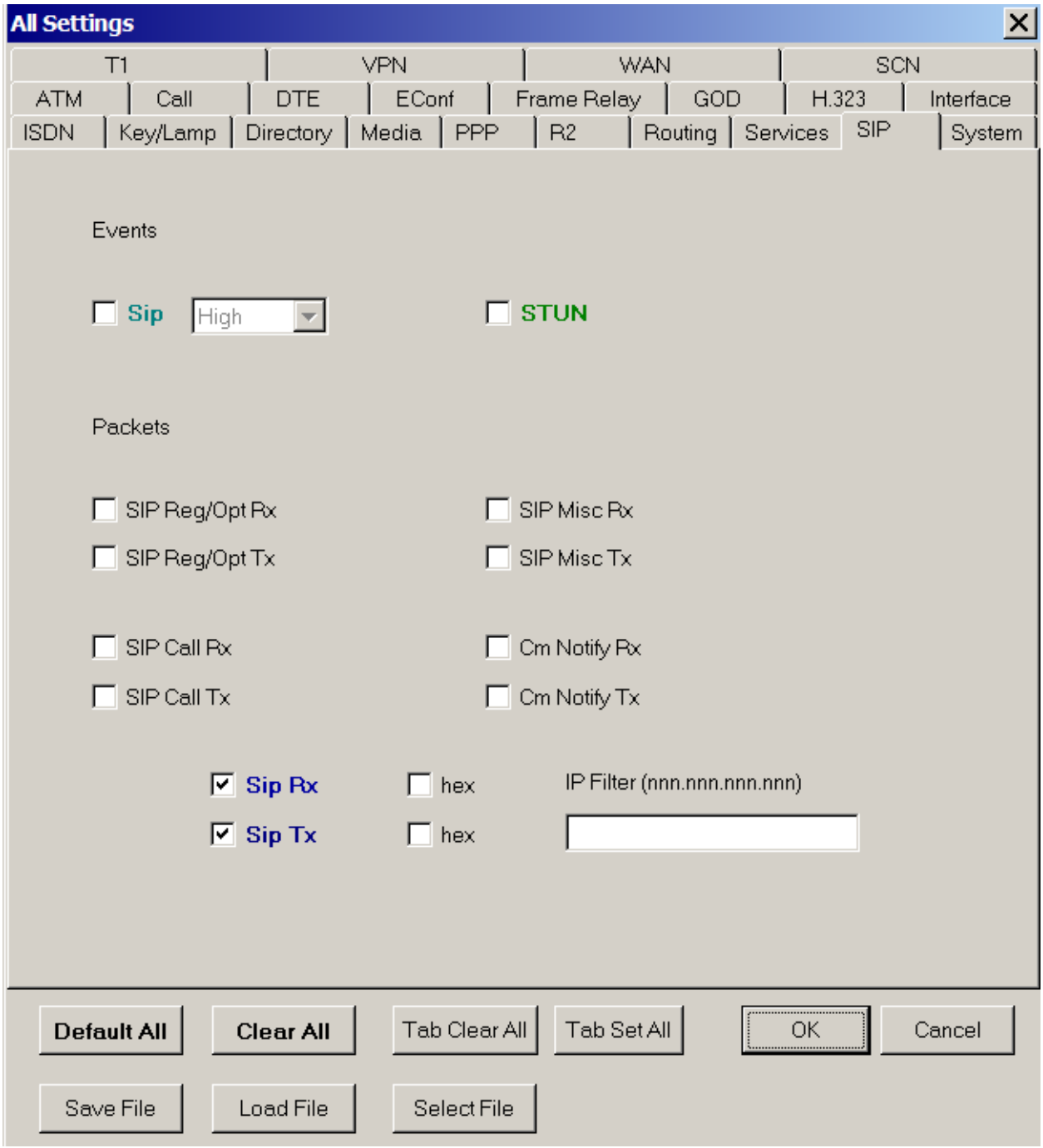
The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, the **SIP Rx** and **SIP Tx** boxes are checked. All SIP messages will appear in the trace with the color blue. To customize the color, right-click on **SIP Rx** or **SIP Tx** and select the desired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



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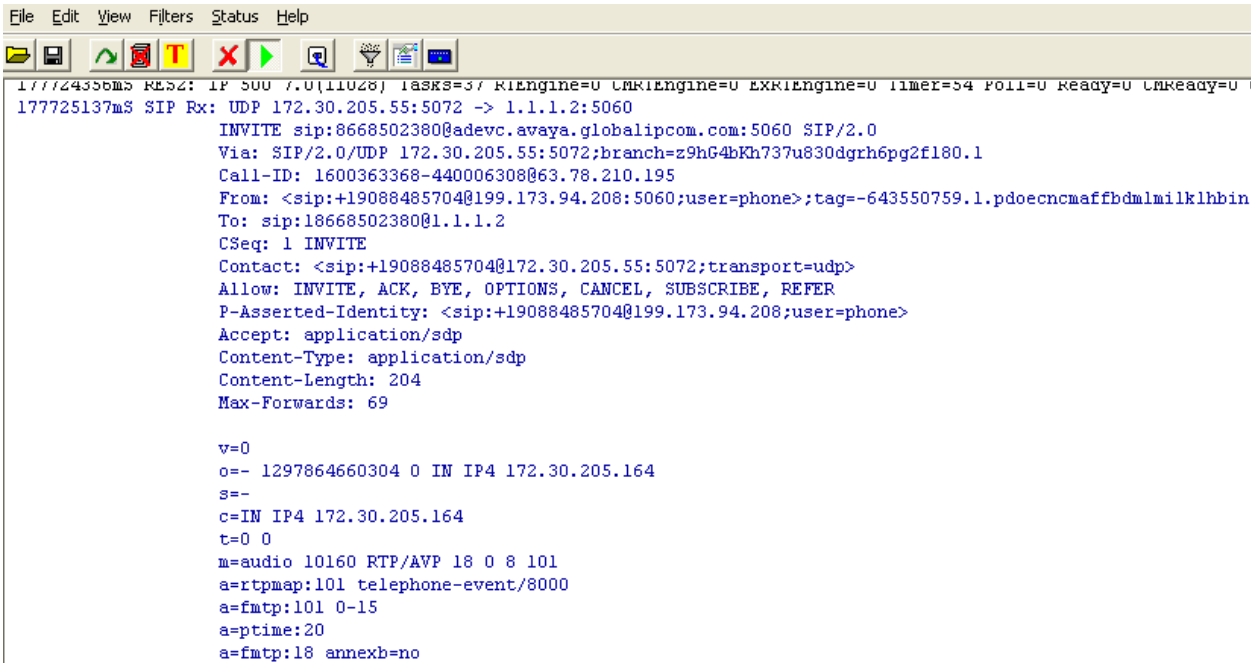
The following screen shows a portion of the monitor trace of an inbound call. As can be observed, PSTN caller 908-848-5704 dialed Verizon IP Toll Free number 1-866-850-2380. Details of the SIP INVITE message sent by Verizon are shown below. This information matches the configuration in these Application 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.



8. Conclusion

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

These Application Notes demonstrated how IP Office Release 7 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.

9. Additional References

This section references documentation relevant to these Application Notes. This section will be updated with Release 7 documentation references when these documents are posted to the Avaya support site. In general, Avaya product documentation is available at http://support.avaya.com

[IPO-INSTALL] IP Office 7.0 Installation Manual, Document Number 15-601042, Issue 23e, March 16, 2011
http://support.avaya.com/css/P8/documents/100129376

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[IPO-MGR] IP Office Release 7.0 Manager 9.0, Document Number 15-601011, Issue 26d, March 23, 2011
http://support.avaya.com/css/P8/documents/100129398

[IPO-SYSSTAT] IP Office Release 6.0 System Status Application, Issue 05a, February 12, 2010 Document Number 15-601758
http://support.avaya.com/css/P8/documents/100073300

[IPO-VMPRO] IP Office Release 7.0 Voicemail Pro Installation and Maintenance, Document Number 15-601063, Issue 25c, March 17, 2011
http://support.avaya.com/css/P8/documents/100129332

[IPO-MON] IP Office System Monitor, Document Number 15-601019
http://support.avaya.com/css/P8/documents/100073350

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

Avaya IP Office Video Softphone documentation can be found here:
http://marketingtools.avaya.com/knowledgebase/businesspartner/ipoffice/mergedProjects/softphoneuser/

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 6.0 with Verizon IPCC Service Suite.
[JRR-IPOR6] Application Notes for Configuring SIP Trunk using Verizon Business IP Contact Center VoIP Inbound and Avaya IP Office Release 6, Issue 1.0
https://devconnect.avaya.com/public/download/dyn/VZB-IPCC-IPOR6.pdf

The following Application Notes provide the configuration that will enable IP Office Release 6.1 to perform like IP Office Release 6.0. The Application Notes referenced below were written to support Verizon certification of IP Office Release 6.1 based on formal compliance testing of IP Office Release 6.0.
[JRR-IPOR61] Application Notes for Configuring SIP Trunk using Verizon Business IP Contact Center VoIP Inbound and Avaya IP Office Release 6.1, Issue 1.0
https://devconnect.avaya.com/public/download/dyn/VZB-IPCC-IPOR61.pdf

[VZB-IPT-IPOR6] Application Notes for Configuring SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service Offer and Avaya IP Office Release 6 –Issue 1.0
https://support.avaya.com/css/P8/documents/100082703

[VZB-IPT-IPOR7] Application Notes for Configuring SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service Offer and Avaya IP Office Release 7 – Issue 0.4

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

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[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] Verizon Business IPCC Interoperability Lab Test Plan, Revision 1.7
- [VZ-Spec] Verizon Business IPCC Trunk Interface Network Interface Specification, Document Version 2.2.1.9

10. Appendix – Example SIP Trunk Template

IP Office Release 7.0 introduces the capability to generate and use templates to facilitate system configuration. This section shows an example SIP Trunk Template generated from the configuration presented in this document.

```
<?xml version="1.0" encoding="utf-8"?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20110328</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>Verizon-IPCC-IP-Toll-Free</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>
  <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>
  <CallInitiationTimeout>4</CallInitiationTimeout>
  <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
  <VoipSilenceSupression>false</VoipSilenceSupression>
  <ReinviteSupported>true</ReinviteSupported>
  <FaxTransportSupport>FOIP_NONE</FaxTransportSupport>
  <UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
  <CodecLockdown>false</CodecLockdown>
  <T38FaxVersion>3</T38FaxVersion>
  <Transport>UDPTL</Transport>
```


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```
  <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>
```


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