Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service Offer and Avaya IP Office Release 6 – Issue 1.0

Abstract

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Trunk SIP Trunk Service Offer and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500 Release 6 Preferred Edition, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, digital, and analog endpoints.

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

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

Table of Contents

1. Introduction

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Trunk SIP Trunk Service Offer and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500 Release 6 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 Trunk SIP Trunk service are able to place and receive PSTN calls via the SIP protocol. The converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI.

Verizon Business IP Trunk service offer 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 Trunk service terminated via a PIP network connection, the solution validated in this document applies equally to IP Trunk services delivered via IDA service terminations.

For more information on the Verizon Business IP Trunking service, including access alternatives, visit http://www.verizonbusiness.com/us/products/voip/trunking/

Table 1 shows the mapping of Verizon-provided DID numbers to IP Office users, groups, or functions. The associated IP Office configuration is shown in Section 3. Selected verifications are illustrated in Section 5.

Verizon Provided DID	Avaya IP Office Destination	Notes
732-945-0228	Auto-Attendant on Voicemail Pro	See Section 3.6
732-945-0229	x30026	Avaya 1616 Telephone, or Avaya IP Office Softphone logged in as x30026
732-945-0231	x201	Digital Telephone with Mobile Twinning Active
732-945-0232	x203	Digital Telephone
732-945-0233	x209	Analog telephone or Fax machine, see Section 1.1
732-945-0234	Voicemail Collect on Voicemail Pro	See Section 3.6, also used for Voicemail Pro Callback
732-945-0235	x30025	Avaya 1616 IP Telephone
732-945-0236	"200 Main" Hunt Group	Collective Ring Hunt Group
732-945-0237	30200 Hunt Group	Sequential Ring Hunt Group

Table 1: Verizon DID to IP Office Mappings

Figure 1 illustrates an example Avaya IP Office solution connected to the Verizon Business IP Trunk SIP Trunk service. The Avaya equipment is located on a private IP subnet. An enterprise edge router provides access to the Verizon Business IP Trunk service network via a Verizon Business T1 circuit. This circuit is provisioned for the Verizon Business Private IP (PIP) service. The Verizon network configuration is identical to the configuration described in reference [CM-VZIPT].

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

Verizon Business used FQDN *pcelban0001.avayalincroft.globalipcom.com*. The Avaya IP Office environment was assigned FQDN *adevc.avaya.globalipcom.com* by Verizon Business.

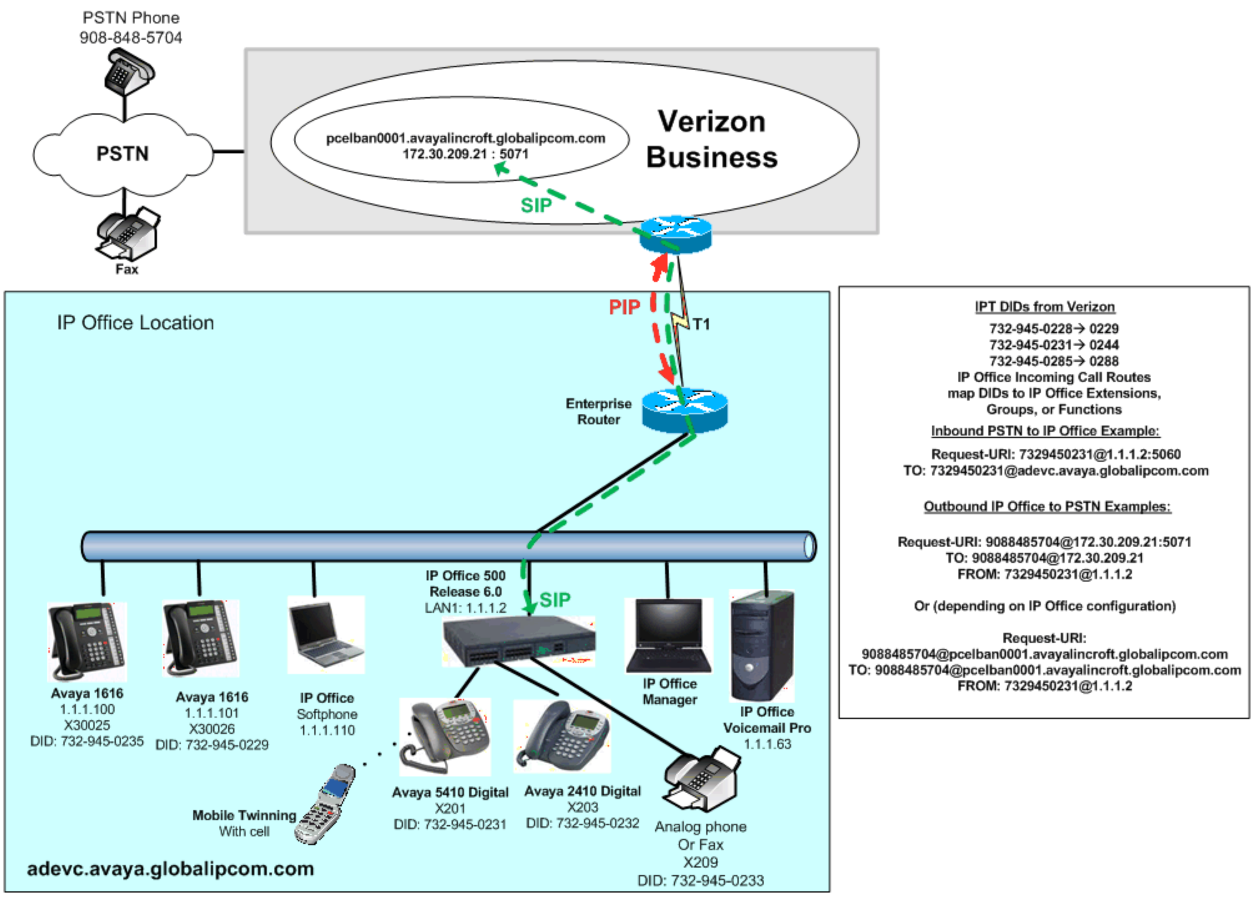


Figure 1: Avaya IP Office with Verizon IP Trunk SIP Trunk Service

1.1. Known Limitations

Interoperability testing of the sample configuration was completed with successful results, with the successful verifications summarized in Section 5. The following observations were noted:

1. For **Compression Mode**, either **G.729a 8K CS-ACELP** or **G.711 ULAW 64K** can be selected for voice calls. Note that T.38 fax is not supported by the Verizon IP Trunk SIP Trunk Service, and T.38 fax is the only fax method supported using SIP Lines on IP Office. Although the Verizon Business IP Trunk Service does not support T.38 fax, and T.38 fax is the only fax method supported by IP Office for SIP Lines, calls were nevertheless made to and from an analog fax machine connected to IP Office. Fax calls may succeed using G.711 but cannot be guaranteed. Although not supported, if fax is to be attempted, **G.711 ULAW 64K** can be selected as the only allowed codec on the SIP Line. Alternatively, if G.729a is desired for voice calls and G.711 for fax calls, the **Re-Invite Supported** option for the SIP Line can be checked to allow re-negotiation to G.711 for a call involving a fax machine that begins at G.729a.

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 that are party to the call.

3. With the configuration described in these Application Notes, when a call uses the IP Office Mobile Twinning feature, and the call is delivered via the Verizon Business IP Trunk service to a PSTN telephone (e.g., a mobile telephone), the calling number displayed on the PSTN telephone (e.g., mobile phone) is the actual caller's calling party number. However, for other types of call diversion (e.g., call forwarding), the calling number displayed on the diverted-to PSTN telephone is the DID associated with the Avaya IP Office telephone (e.g., the forwarding user), and not the actual caller's calling party number. IP Office will populate the Diversion header for Mobile Twinning, but not for other forms of diversion such as call forwarding.

4. When using the IP Office Softphone, inbound PSTN calls from the Verizon Business IP Trunk 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 3.4.3). 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.

5. Although ARS alternate routing was tested successfully, there is no capability to control the timing of the alternate routing. IP Office will wait a short fixed amount of time (e.g., roughly 5 seconds) for a response after sending INVITE to a primary SIP Line. If no response is received, route advance to a configured alternate route occurs after

approximately 5 seconds. The IP Office product team expects to provide configurable control of this timing in a future release.

6. For inbound PSTN calls to IP Office, if Display information is provided by Verizon in the To Header, this display information is sent by IP Office in the 200 OK as the User portion of the Contact in the Contact header, rather than the DID number configured for the answering user. In the sample configuration, this does not cause a user-perceivable problem, but this is an anomaly that the IP Office product team will correct in a future release.

7. IP Office Release 6 does not support DNS SRV records. Static provisioning of the Verizon Business SIP IP Address and port are shown in these Application Notes.

2. Equipment and Software Validated

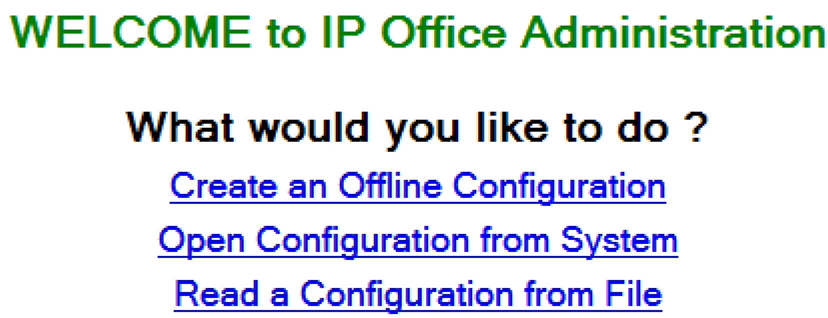
Table 2 shows the equipment and software used in the sample configuration.

Equipment	Software
Avaya IP Office 500	Release 6.0 (6.0.8) (Preferred Edition)
Avaya IP Office Manager	Release 8.0 (8.0.8) (Preferred Edition)
Avaya IP Office Voicemail Pro	Release 6.0.16
Avaya IP Office Voicemail Pro Client	Version 6.0 (16)
Avaya 1600-Series Telephones (H.323)	Release 1.2.2
Avaya 2400-Series and 5400-Series Digital Telephones	REL: 6.00 (downloaded from IP Office)
Avaya IP Office Softphone	Release 3.0 (56516)
Brother Intellifax 1360 (analog fax)	N/A

Table 2: Equipment and Software Tested

3. 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:



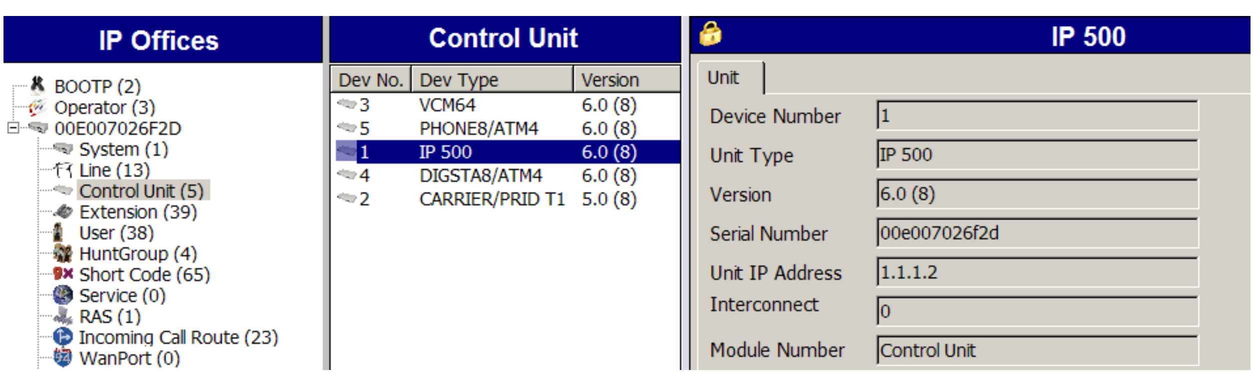
Open the IP Office configuration, either by reading the configuration from the IP Office server, or from file. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details pane on the right side.

3.1. Physical, Network, and Security Configuration

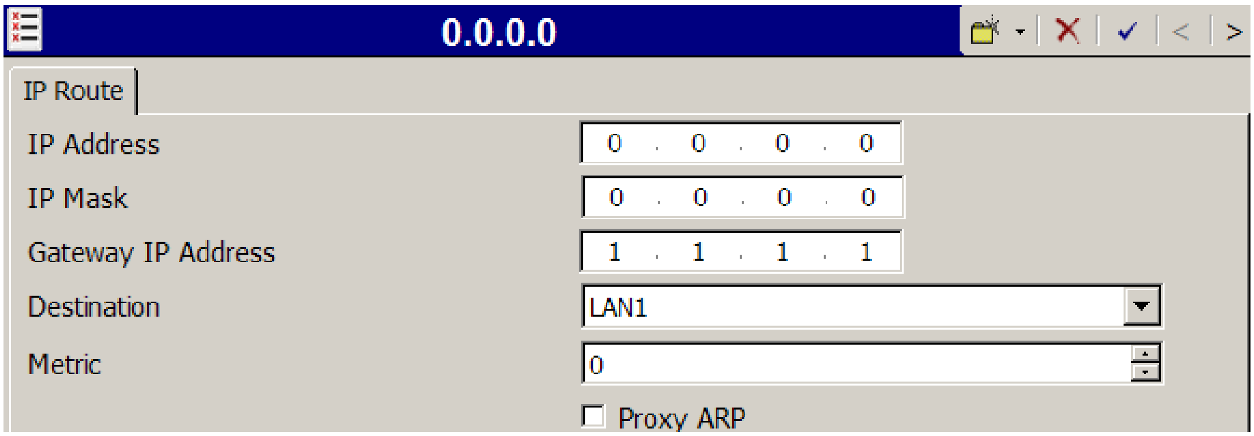
This section describes attributes of the sample configuration, but is not meant to be prescriptive. Consult reference [IPO-INSTALL] for more information on the topics in this section.

In the sample configuration, looking at the IP Office 500 from left to right, the first module is blank (i.e., no module is 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, an analog telephone or a fax machine is connected to port 1 of the "Phone8" module.

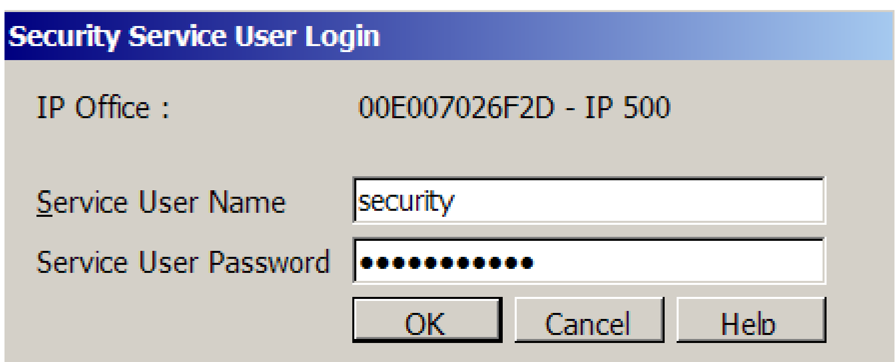
The following screen shows the modules in the IP Office used in the sample configuration. To access such a screen, select **Control Unit** in the Navigation pane. The modules appear in the Group pane. In the screen below, **IP 500** is selected in the Group pane, revealing additional information about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



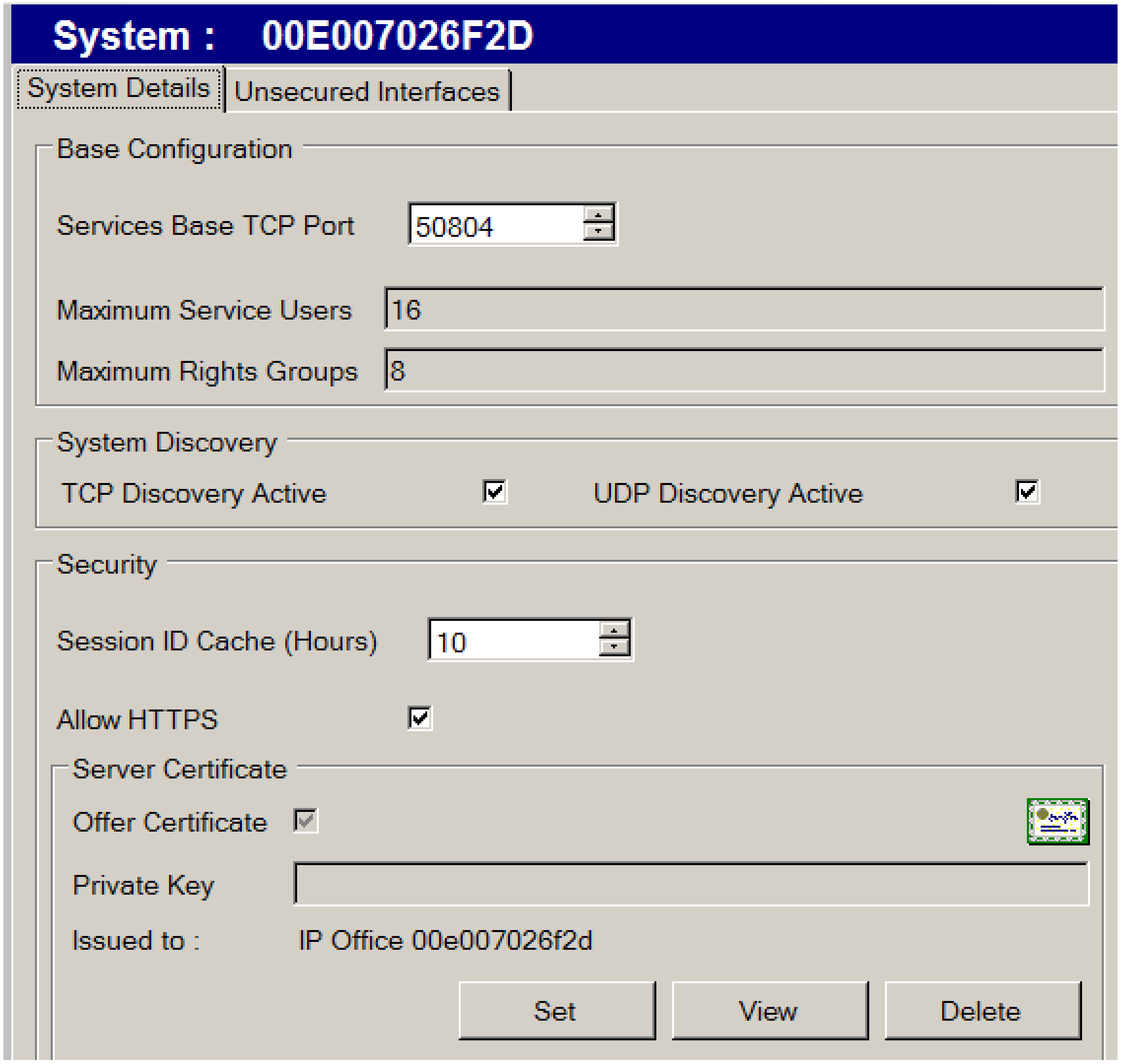
In the sample configuration, the IP Office LAN1 port is physically connected to the local area network switch at the IP Office customer site. The default gateway for this network is 1.1.1.1. To add an IP Route in IP Office, right-click **IP Route** from the Navigation pane, and select **New.** To view 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.



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.

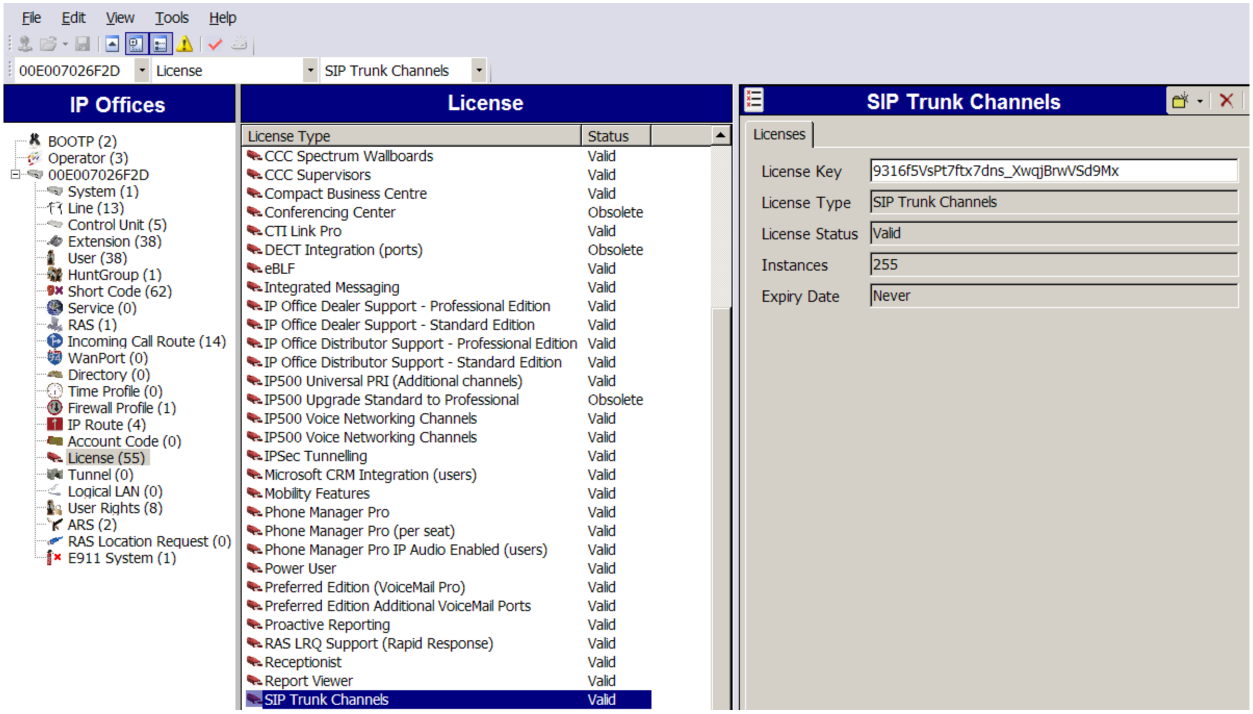


When complete, select **File → Configuration** to return to configuration activities.

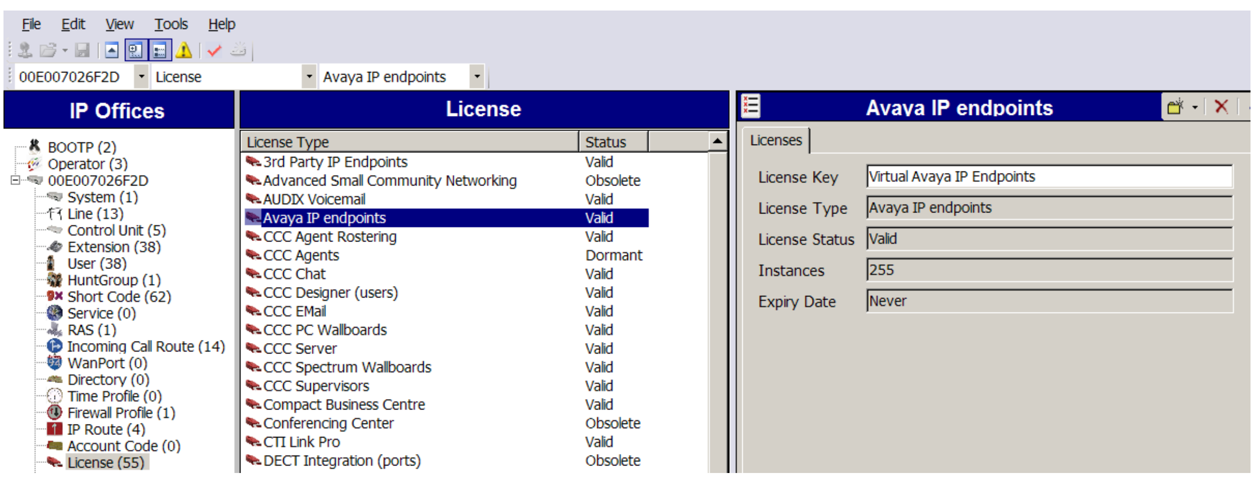
3.2. Licensing

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.

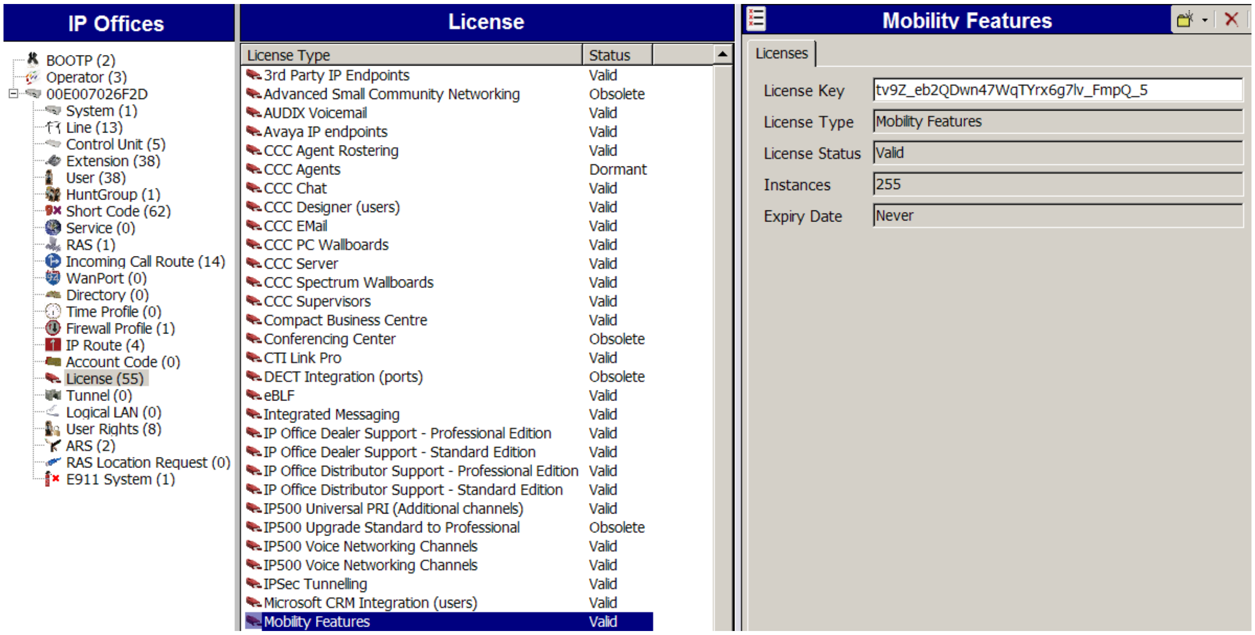
To verify that there is a SIP Trunk Channels License with sufficient capacity, click **License** in the Navigation pane and **SIP Trunk Channels** in the Group pane. Confirm a valid license with sufficient "Instances" (trunk channels) in the Details pane.



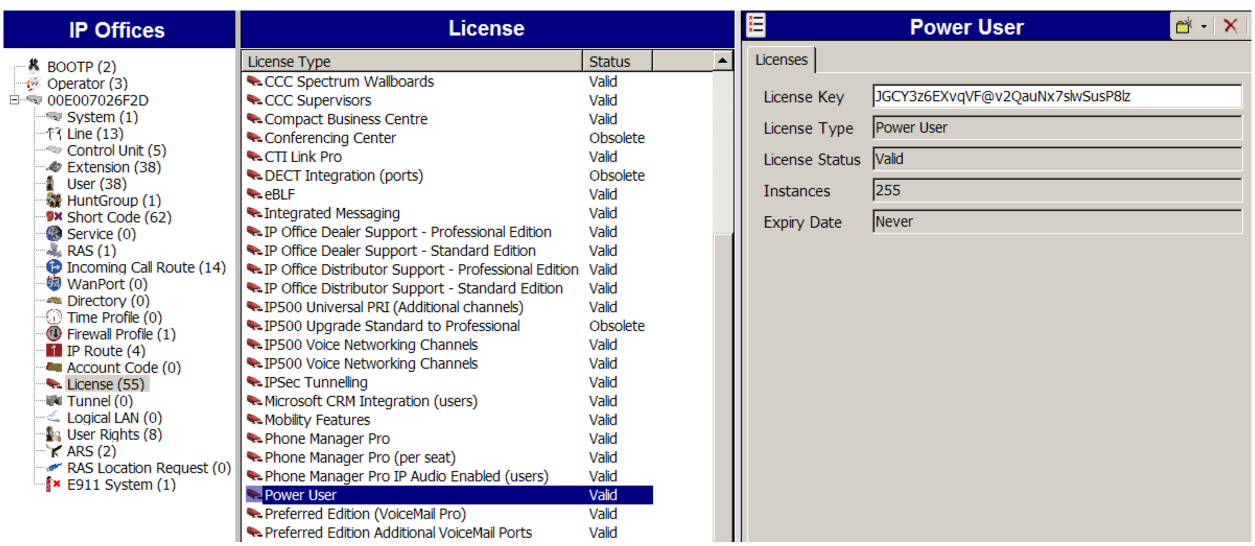
If Avaya IP Telephones will be used, verify the Avaya IP endpoints license. Click **License** in the Navigation pane and **Avaya IP endpoints** in the Group pane. Confirm a valid license with sufficient "Instances" in the Details pane.



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.

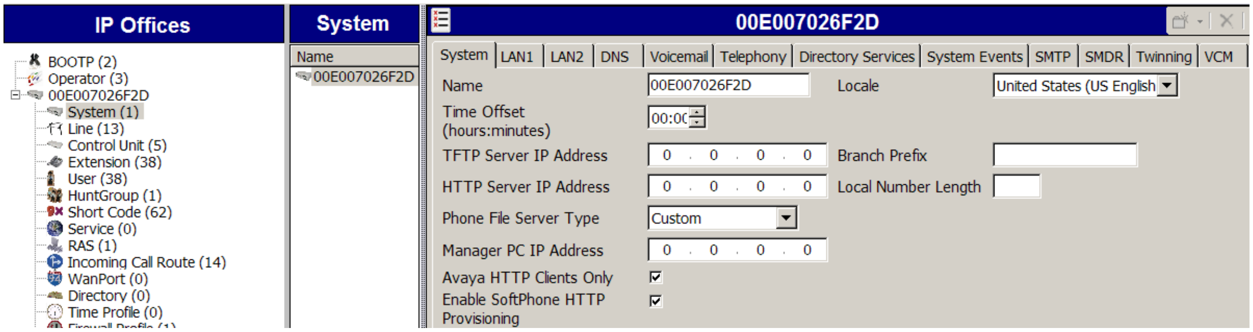


3.3. System Settings

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.

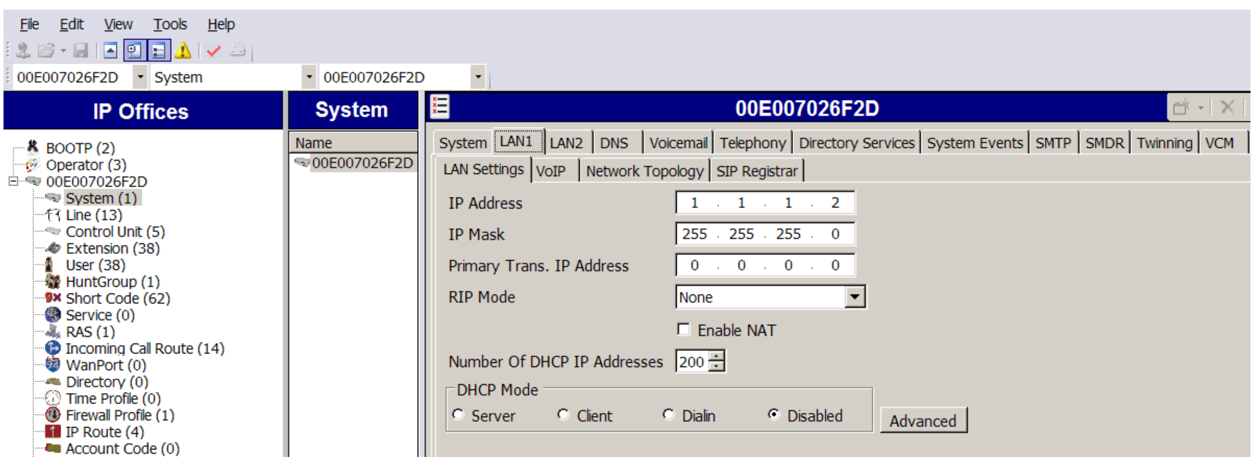
3.3.1. System Tab

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.



3.3.2. LAN Settings

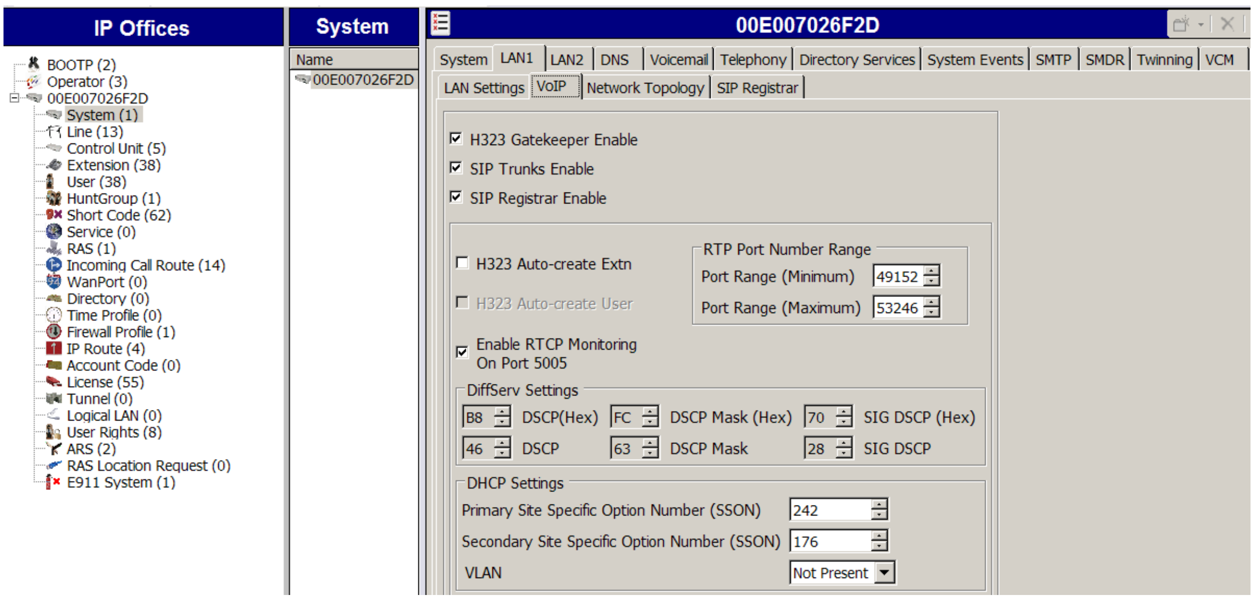
In the sample configuration, LAN1 was used to connect the IP Office to the enterprise network. Other LAN choices (e.g., LAN2) may also be used. To view or configure the **IP Address** of LAN1, select the **LAN1** tab followed by the **LAN Settings** tab. As shown in **Figure 1**, the IP Address of the IP Office, known to Verizon Business, is 1.1.1.2. Other parameters on this screen may be set according to customer requirements.



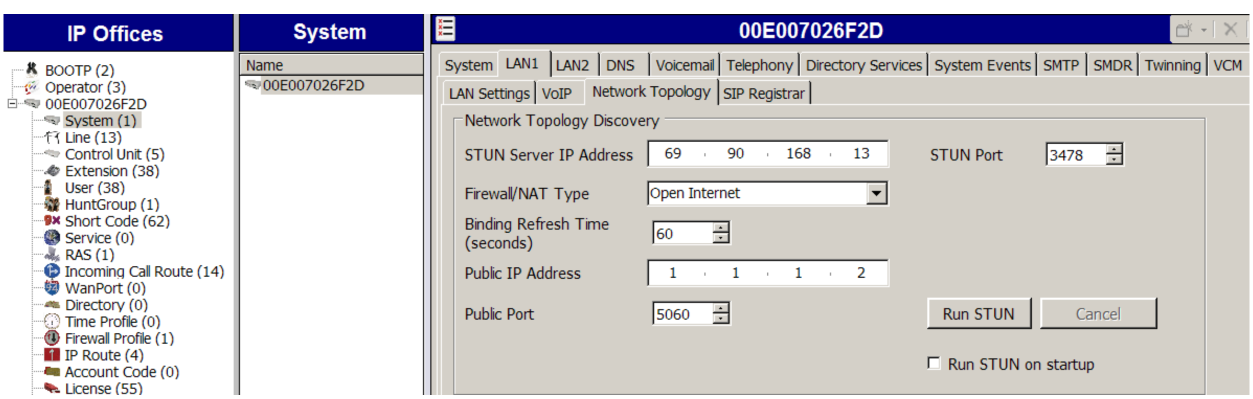
Select the **VoIP** tab as shown in the following screen. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Verizon Business. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 1600-Series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



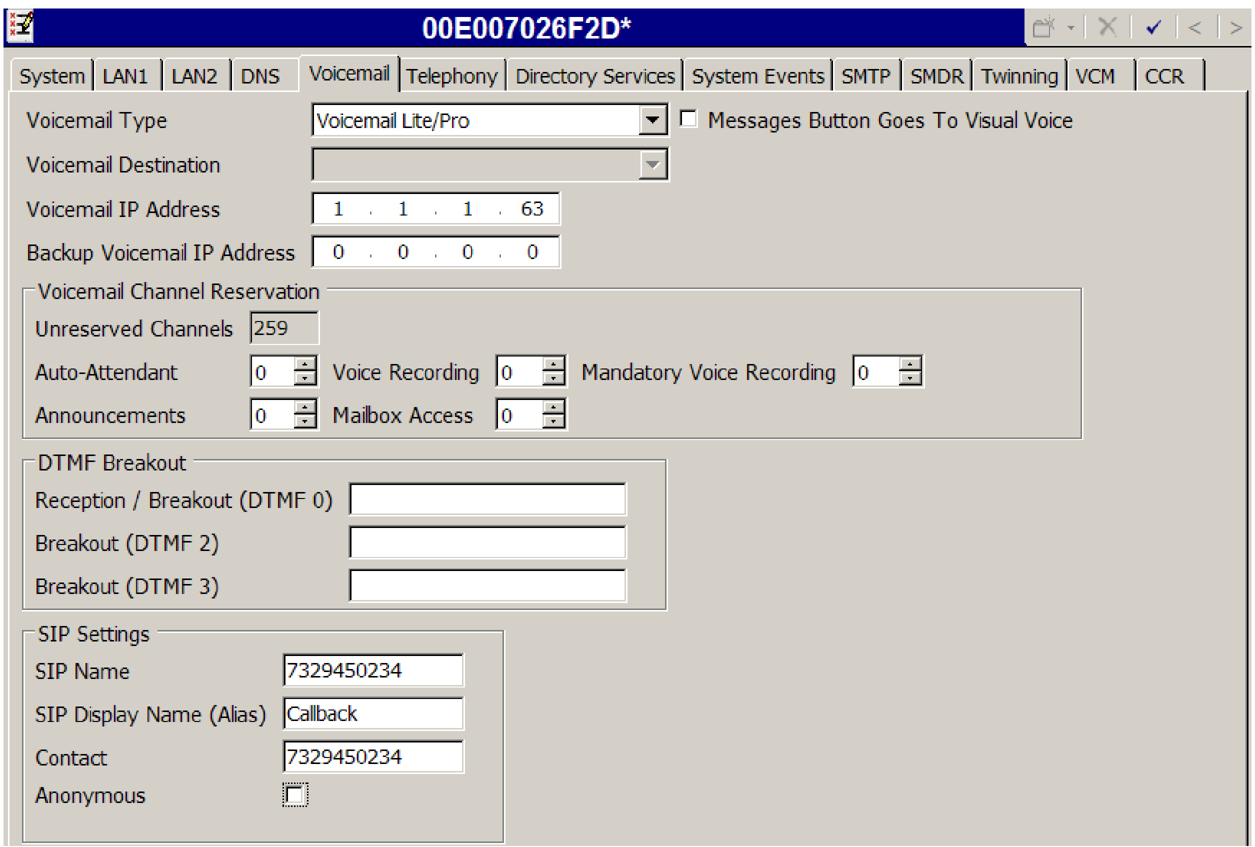
Select the **Network Topology** tab as shown in the following screen. For **Public IP Address**, enter the Avaya IP Office LAN1 IP address. Set the **Public Port** to 5060. In the sample configuration, the **Firewall/NAT Type** is set to “Open Internet”, and the **Binding Refresh Time** is set to 60 seconds. Later, the SIP Line will be configured to use the network topology information for LAN1. 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.

3.3.3. Voicemail

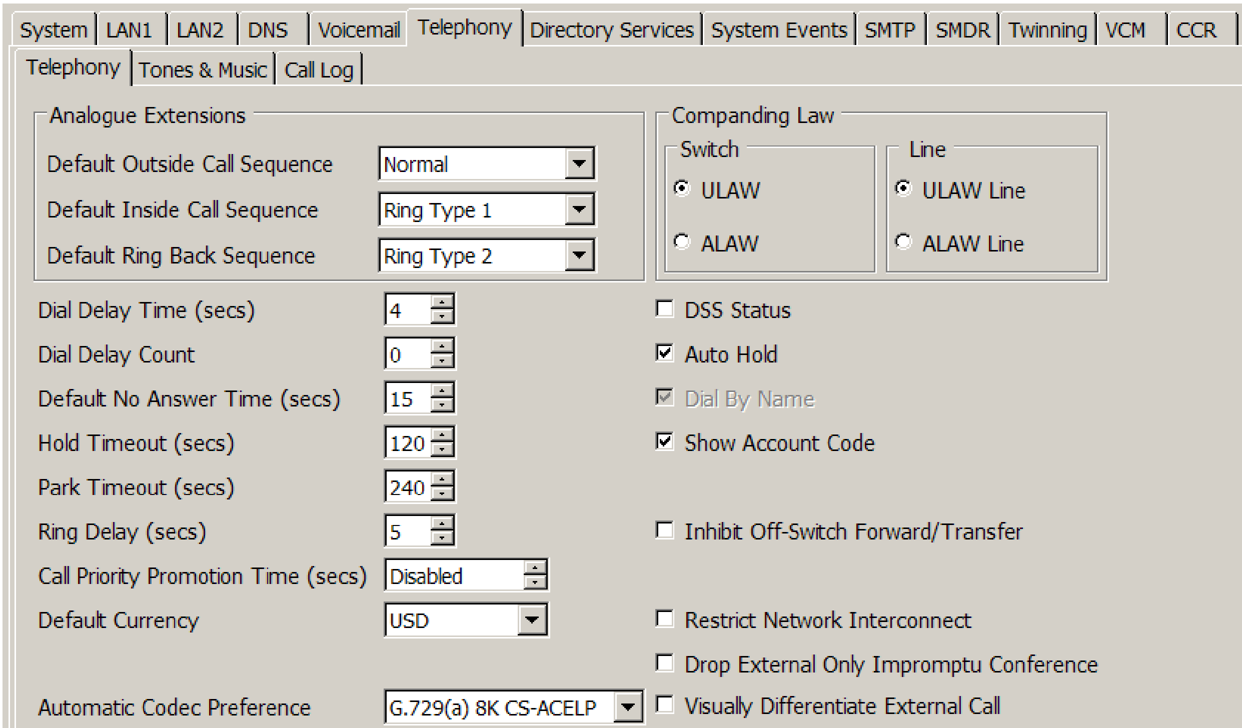
To view or change voicemail settings, select the **Voicemail** tab as shown in the following screen. The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. The **Voicemail Type** in the sample configuration is "Voicemail Lite/Pro". Other Voicemail 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**. In the sample configuration, the "Callback" application of Avaya Voicemail Pro was used to allow Voicemail Pro to call out via the SIP Line to Verizon Business when a message is left in a voice mailbox. The **SIP Settings** shown in the screen below enable IP Office to populate the SIP headers for an outbound "callback" call from Voicemail Pro, similar to the way the fields with these same names apply to calls made from telephone users (e.g., see Section 3.5). Other parameters on this screen may be set according to customer requirements.



3.3.4. System Telephony Configuration

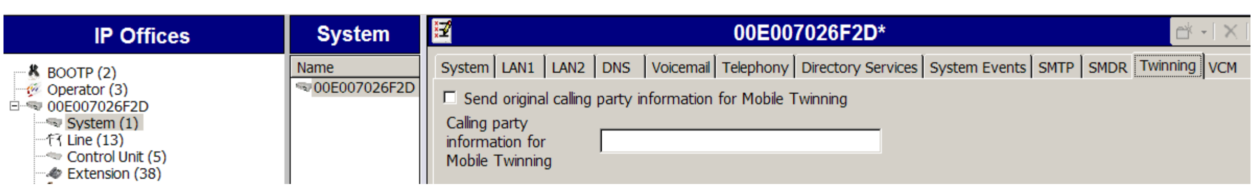
To view or change telephony settings, select the **Telephony** tab and **Telephony** sub-tab as shown in the following screen. The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. In the sample configuration, the **Inhibit Off-Switch Forward/Transfer** box is unchecked so that call forwarding and call transfer to

PSTN destinations via the Verizon Business IP Trunk service can be tested. 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.



3.3.5. 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 (Section 3.4), 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. An example Wireshark trace for a call using Diversion Header with Mobile Twinning is shown in Section 5.



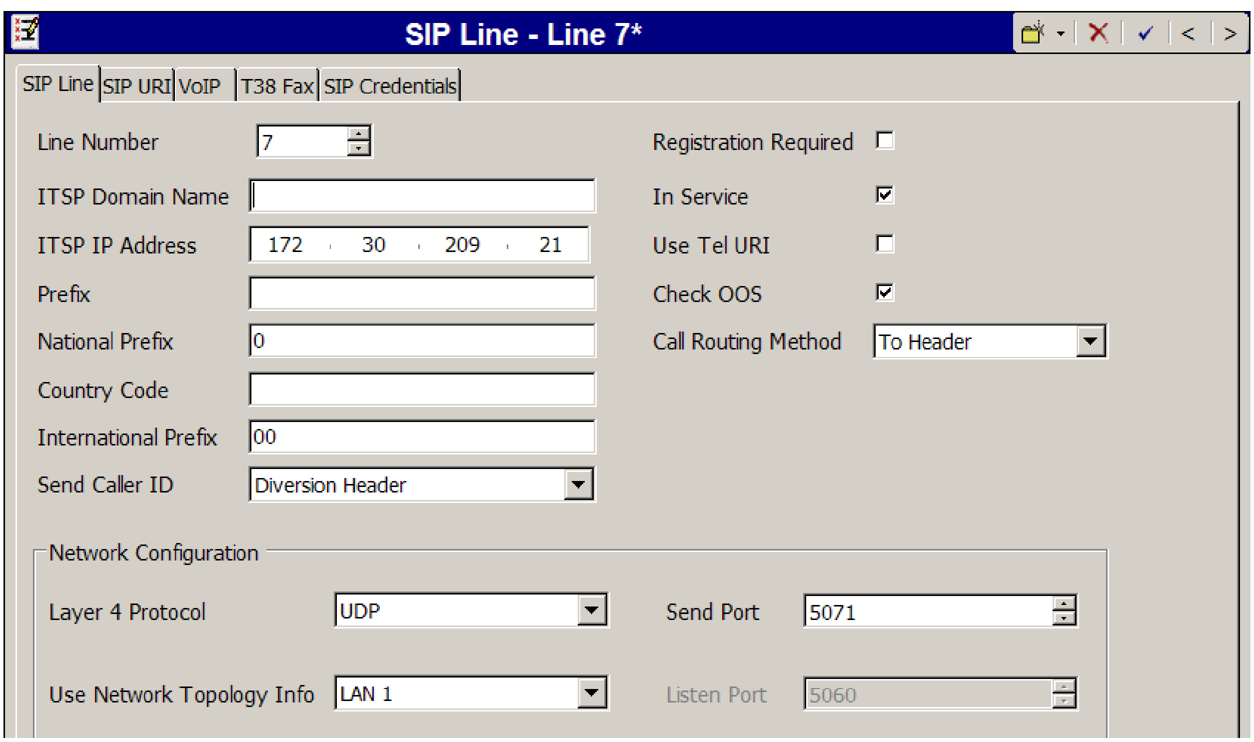
3.4. SIP Line

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.

3.4.1. SIP Line - SIP Line Tab

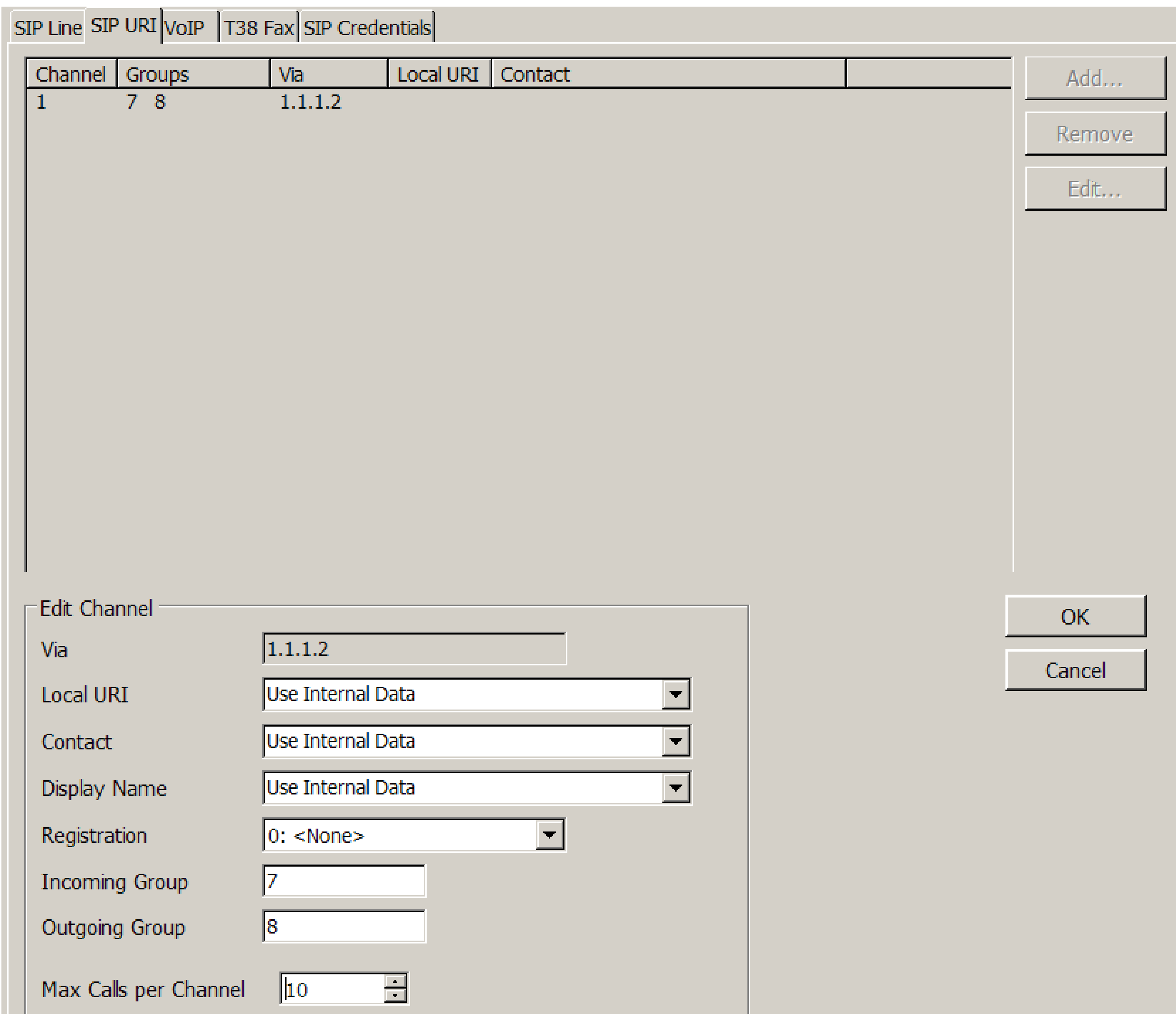
The **SIP Line** tab in the Details pane is shown below for Line Number 7, used for the Verizon Business IP Trunk service. The **ITSP IP Address** is set to the IP Address provided by Verizon Business. As shown in **Figure 1**, this IP Address is 172.30.209.21. In the **Network Configuration** area, 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 5071 in the sample configuration. Note that IP Office does not support DNS SRV for lookup of the SIP IP Address and port. 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. By default, **Check OOS** is 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, which has been set to 60 seconds as shown in Section 3.3.2. See Section 3.10 for additional information related to configuring the periodicity of SIP OPTIONS.

The **Send Caller ID** parameter is set to "Diversion Header". With this setting and the related configuration in Section 3.3.5 and Section 3.9, IP Office will include the Diversion Header for calls that are directed via Mobile Twinning out the SIP Line to Verizon. The Diversion Header will contain the number associated with the Twinning user, allowing Verizon to admit the call, and the From Header will be populated with the true calling party identity, allowing the twinning destination (e.g., mobile phone) to see the true caller id. The **Call Routing Method** can retain the default "Request URI" setting, or may be changed to "To Header" as shown, to match Incoming Call Routes based on the contents of the "To Header". In the sample configuration, both approaches were tested successfully. Click **OK** (not shown).



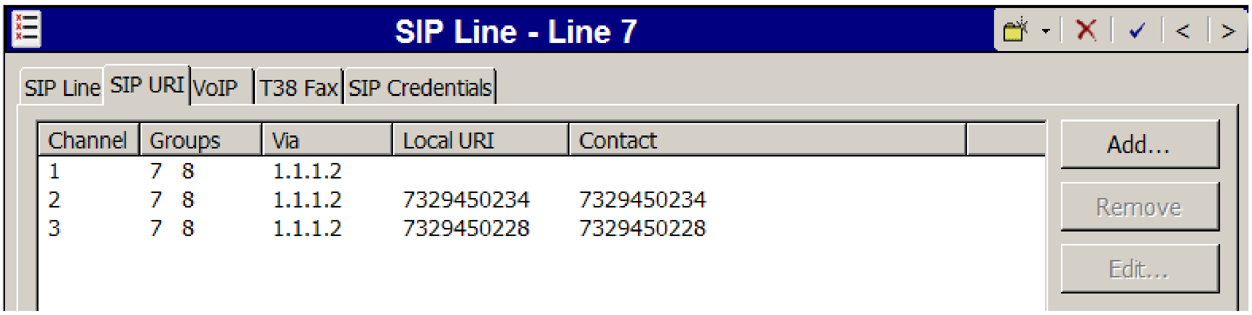
3.4.2. SIP Line - SIP URI Tab

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 example screen below, a previously configured entry is edited. "Use Internal Data" is selected for the **Local URI**, **Contact**, and **Display Name**. Information configured on the SIP Tab for individual users will be used to populate the SIP headers. The **Registration** parameter is set to the default "0: <None>" since Verizon Business IP Trunk service does not require registration. The **Incoming Group** parameter, set here to 7, will be referenced when configuring Incoming Call Routes to map inbound SIP trunk calls to IP Office destinations in Section 3.7. The **Outgoing Group** parameter, set here to 8, will be used for routing outbound calls to Verizon via the Short Codes (Section 3.6) or ARS configuration (Section 3.8)**.** The **Max Calls per Channel** parameter, configured here to 10, sets the maximum number of simultaneous calls that can use the URI before IP Office returns busy to any further calls. Click **OK**.



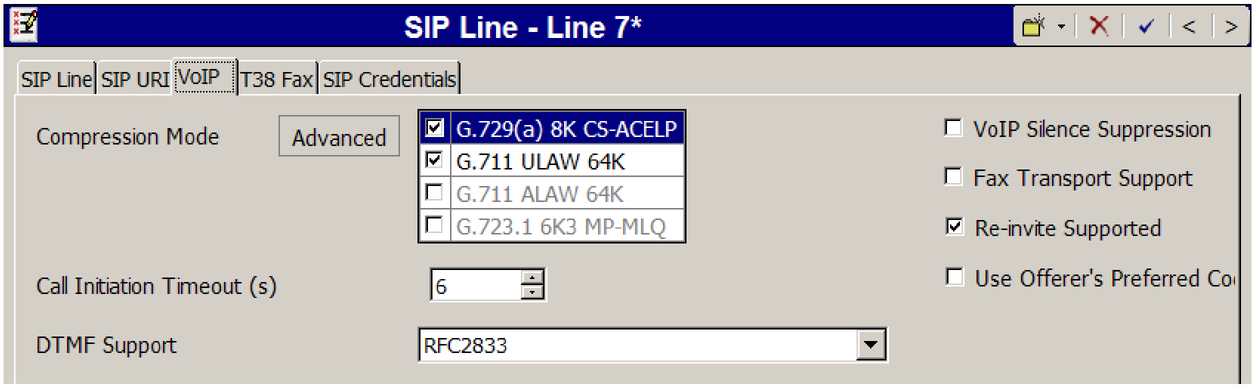
In the sample configuration, the single SIP URI shown above was sufficient to allow incoming calls for Verizon DID numbers destined for specific IP Office users or IP Office hunt groups. The calls are accepted by IP Office since the incoming number will match the SIP Name configured for the user or hunt group that is the destination for the call. For service numbers, such as a DID number routed directly to voicemail, or a DID number routed to an auto-attendant service on Voicemail Pro, the DID numbers that IP Office should admit can be entered into the **Local URI** and **Contact** fields instead of "Use Internal Data".

The following shows the **SIP URI** tab for SIP Line 7 after the SIP URIs corresponding to Voice mail (732-945-0234) and Auto-attendant (732-945-0228) have been added.



3.4.3. 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 cause IP Office to include both G.729a and G.711MU in the Session Description Protocol (SDP), in that order. 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 setting. In the sample configuration, Verizon also 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 Verizon Business IP Trunk service does not support T.38 fax, the **Fax Transport Support** parameter is not checked, and the **T38 Fax** tab need not be visited. Since the Verizon Business IP Trunk service does not require registration, the **SIP Credentials** tab need not be visited. Click **OK** (not shown).

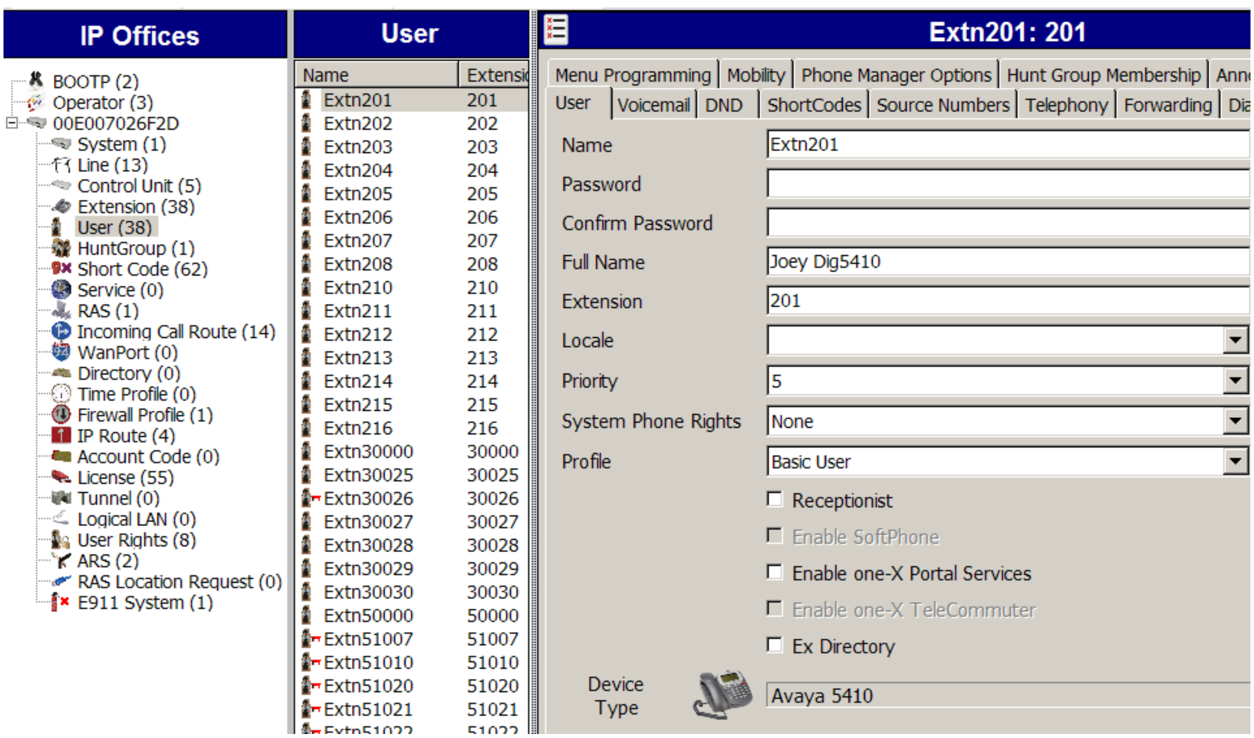


3.5. Users, Extensions, and Hunt Groups

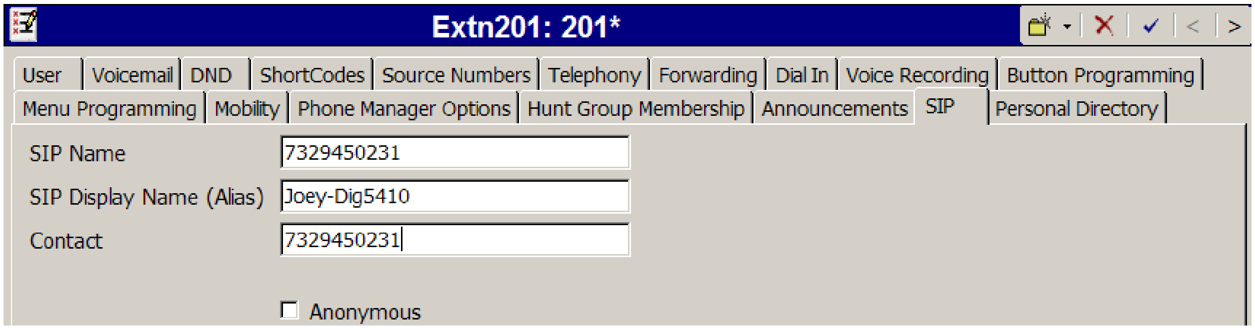
In this section, examples of IP Office Users, Extensions, and Hunt Groups will be illustrated. In the interests of brevity, not all users and extensions shown in **Figure 1** will be presented, since the configuration can be easily extrapolated to other users. To add a User, right click on **User** in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

3.5.1. User 201

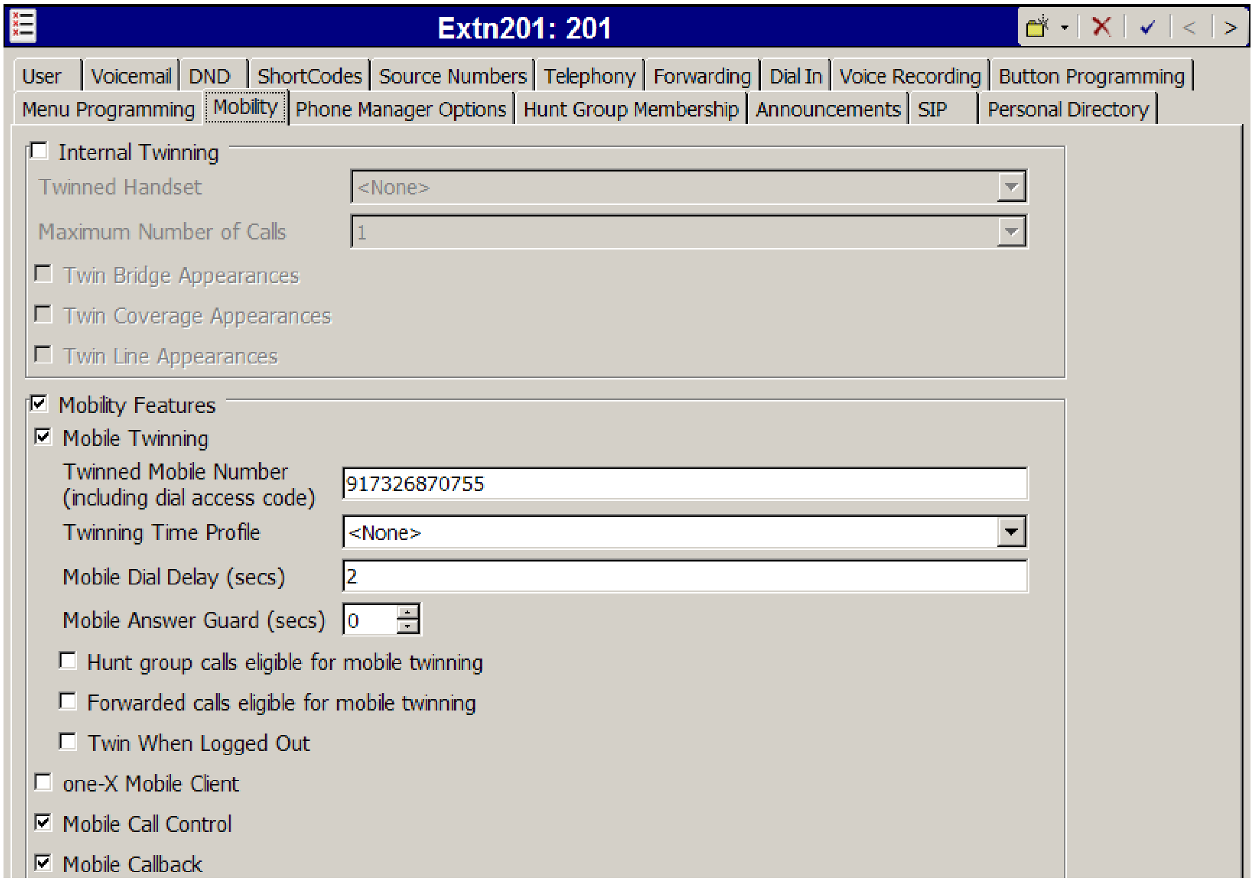
The following screen shows the **User** tab for User 201. As shown in **Figure 1**, this user corresponds to the digital telephone 5410.



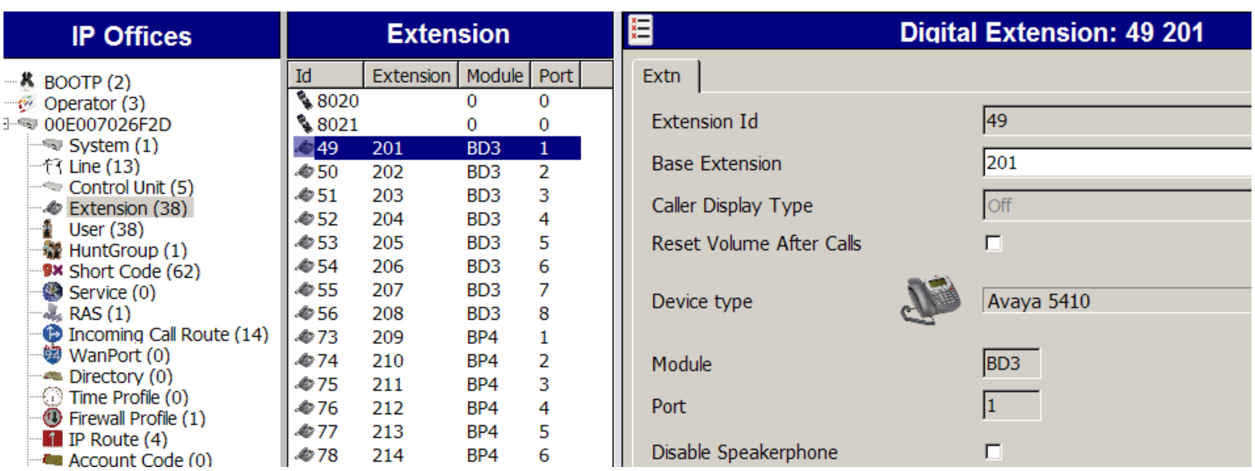
The following screen shows the **SIP** tab for User 201. The **SIP Name** and **Contact** parameters are configured with the DID number of the user, 7329450231. These parameters configure the user part of the SIP URI in the From header for outgoing SIP trunk calls, and allow matching of the SIP URI for incoming calls, without having to enter this number as an explicit SIP URI for the SIP Line. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network. See Section 3.6 for a method of using a short code (rather than static user provisioning) to place an anonymous call.



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.

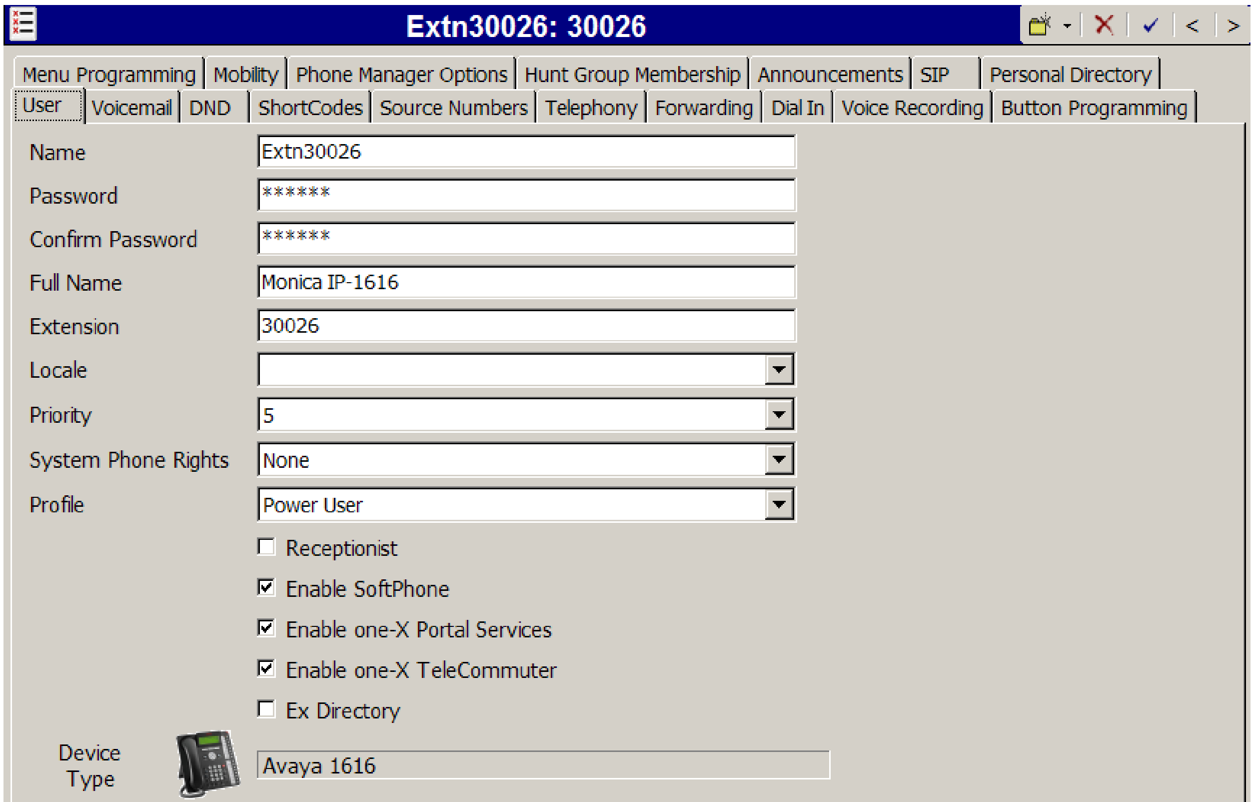


The following screen shows the Extension information for this user. To view, select Extension from the Navigation pane, and the appropriate extension from the Group pane. As stated in Section 3.1, the Avaya 5410 telephone user with extension 201 is connected to port 1 of the digital module.

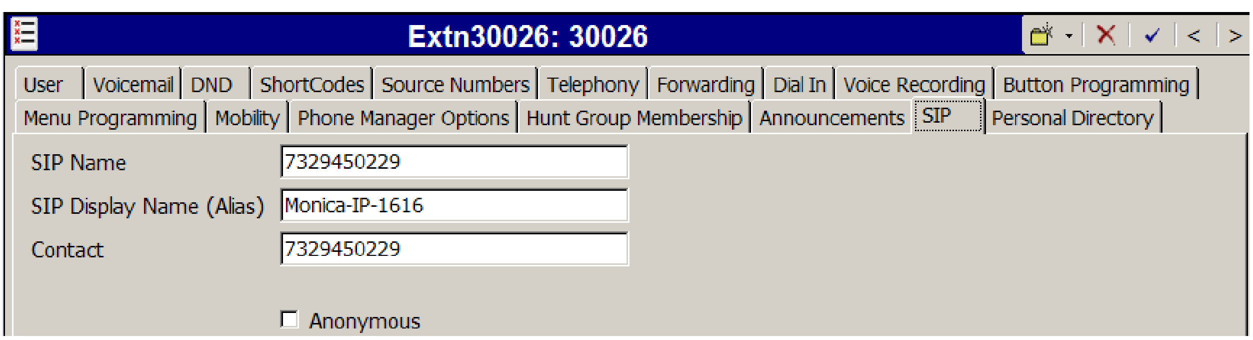


3.5.2. User 30026

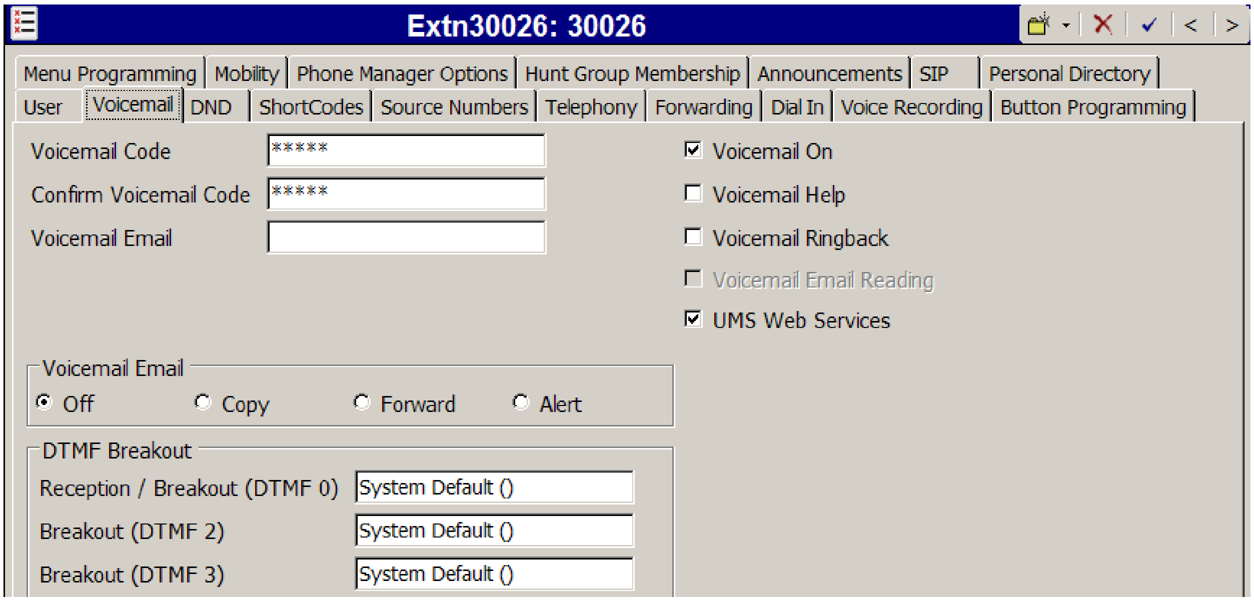
The following screen shows the **User** tab for User 30026. This user corresponds to an Avaya 1616 IP Telephone that will be granted “Power User” and Avaya IP Office Softphone features. The **Profile** parameter is set to “Power User”. The **Enable Softphone** box is checked, along with other advanced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



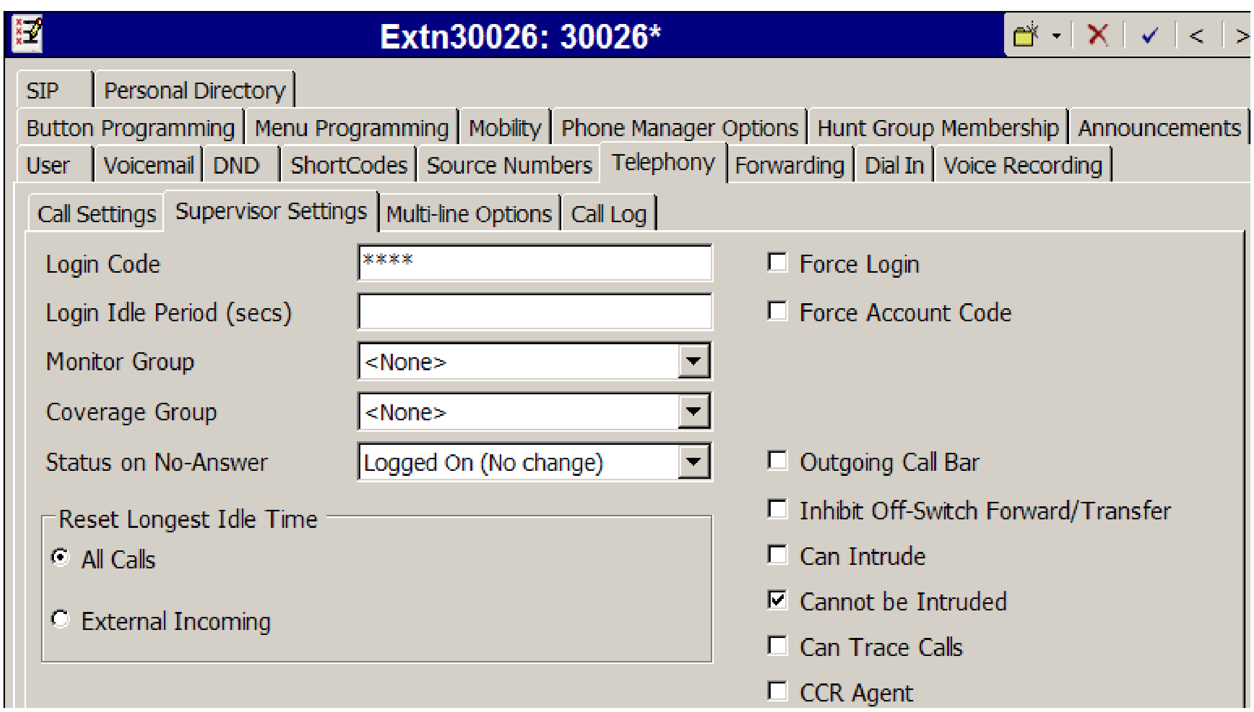
Like the user with extension 201, the **SIP** tab for the user with extension 30026 is configured with a **SIP Name** and **Contact** specifying the user’s Verizon Business DID number.



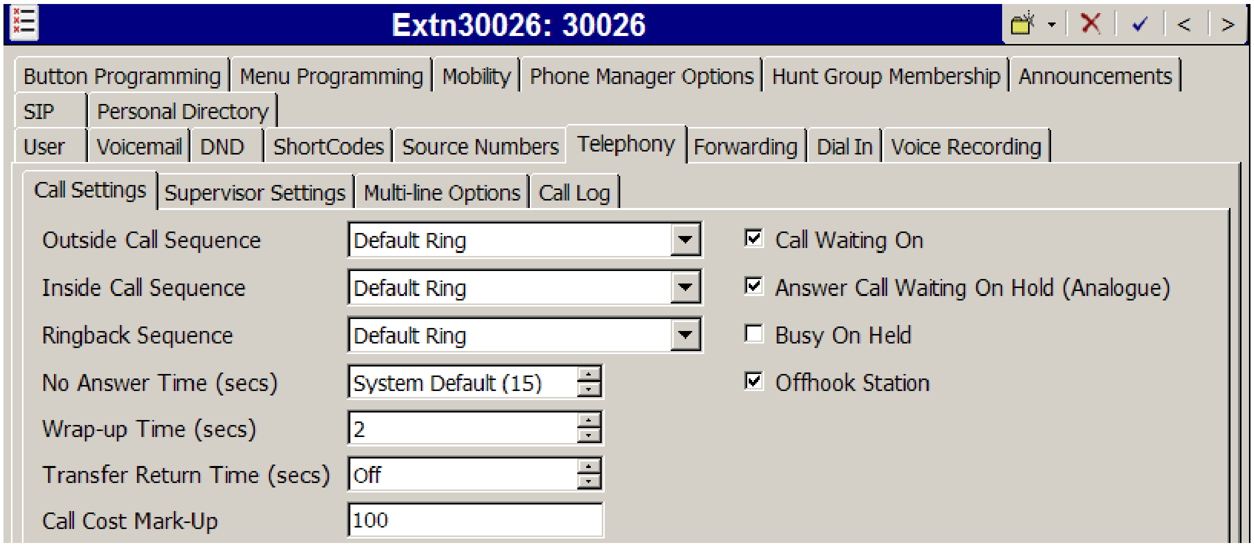
The following screen shows the **Voicemail** tab for the user with extension 30026. The **Voicemail On** box is checked, and a voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters. In the verification of these Application Notes, incoming calls from the Verizon Business IP Trunk 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, to test DTMF using RFC 2833, and to test assignment of a Verizon DID number to the "Voicemail Collect" feature (e.g., via the *17 short code shown in Section 3.6).



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

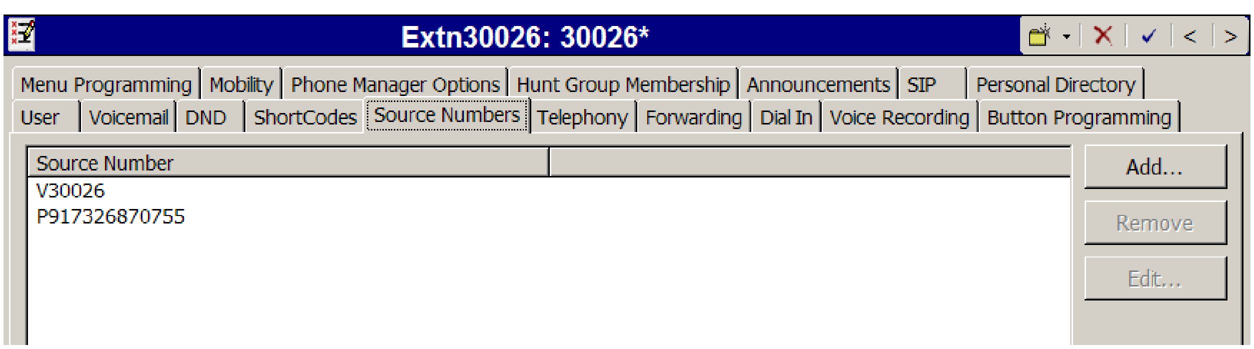


Select the **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow an IP Office Softphone logged in as this extension to have multiple call appearances (e.g., necessary for call transfer).

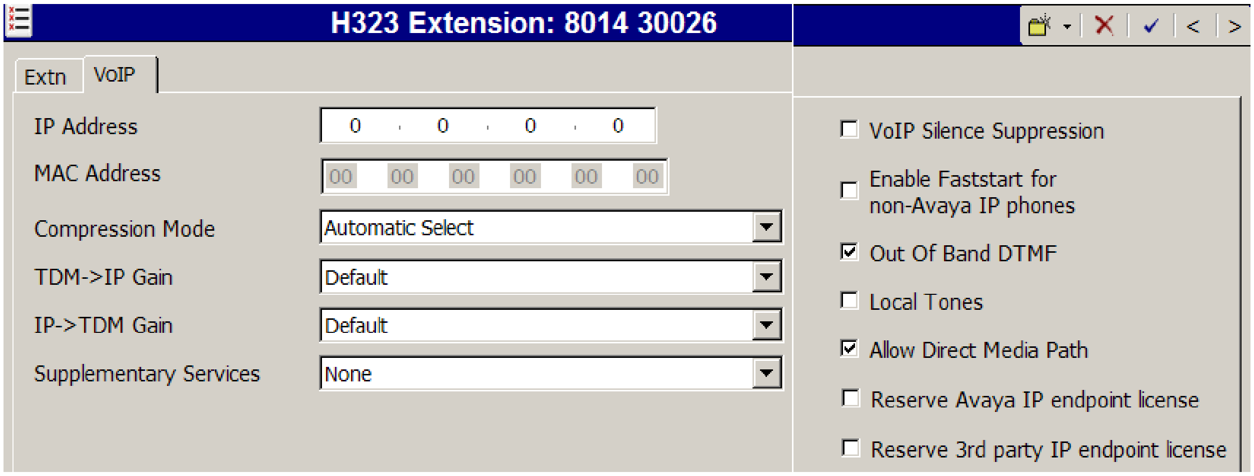


The following screen shows the **Source Numbers** tab for the user with extension 30026. Although the Voicemail Pro configuration is beyond the scope of these Application Notes, the "Callback" feature has been enabled on Voicemail Pro for this user, and the Source Number "P917326870755" has been previously added. With this configuration, when a message is left in this user's Voicemail Pro mailbox, a "callback" call will be initiated to "917326870755". The callback call will be sent to Verizon via SIP Line 7, 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 3.3.3. It is possible (and more typical) for the end user to configure callback numbers via the Voicemail Pro Telephony User Interface, rather than the administrator configuring a callback number via the "P" Source Number in IP Office, as shown below.

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**.



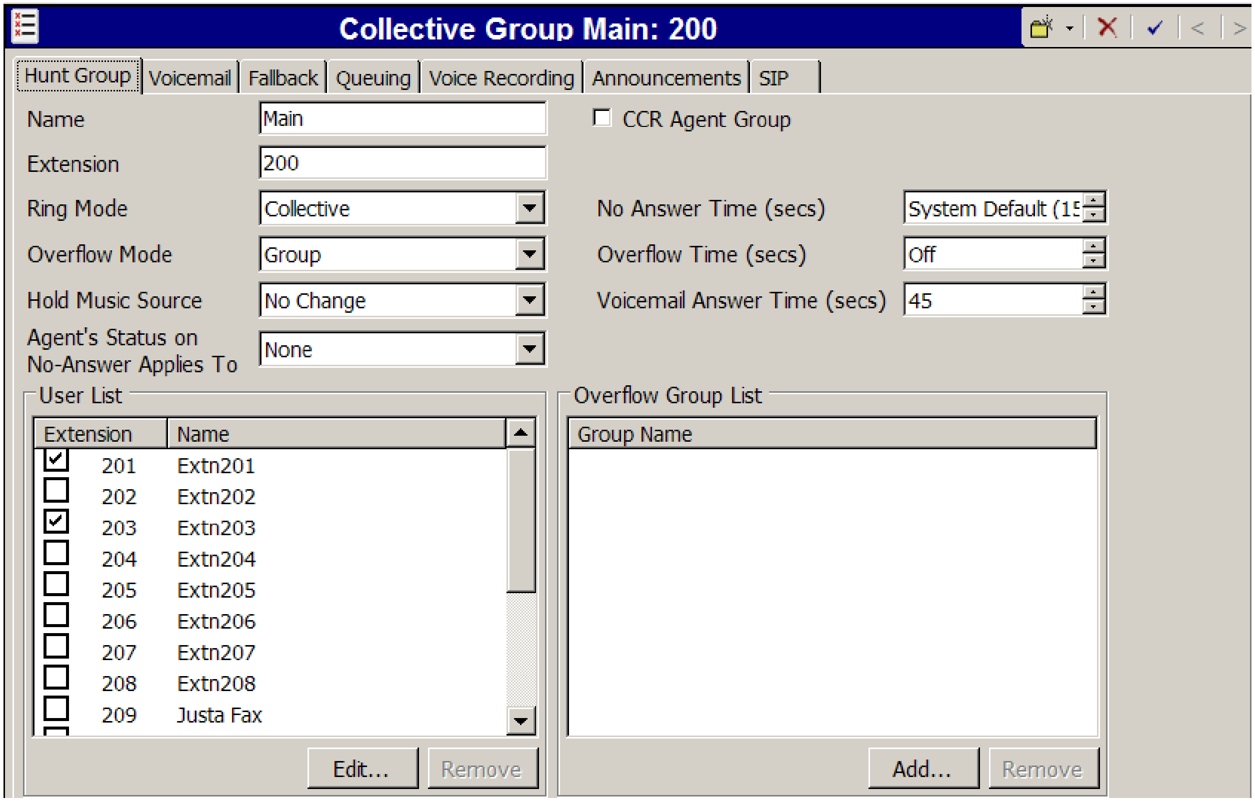
The following screen shows the Extension information for this user, simply to illustrate the **VoIP** tab available for an IP Telephone. To view, select **Extension** from the Navigation pane, and the appropriate extension from the Group pane. Select **VoIP** in the Details pane.



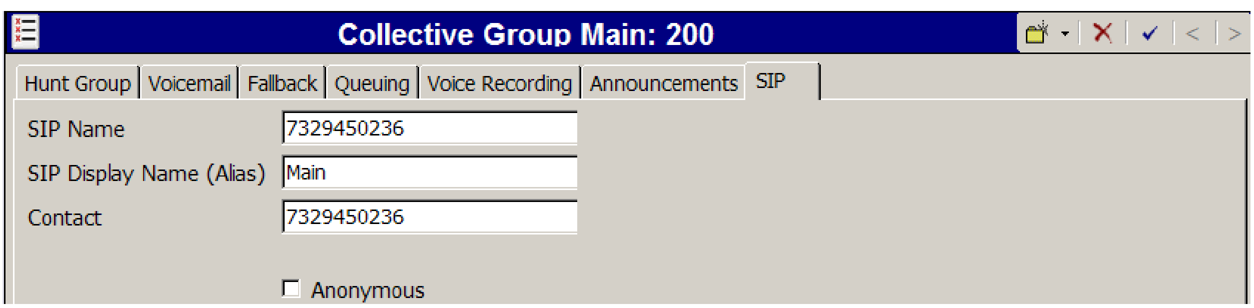
3.5.3. Hunt Groups

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.

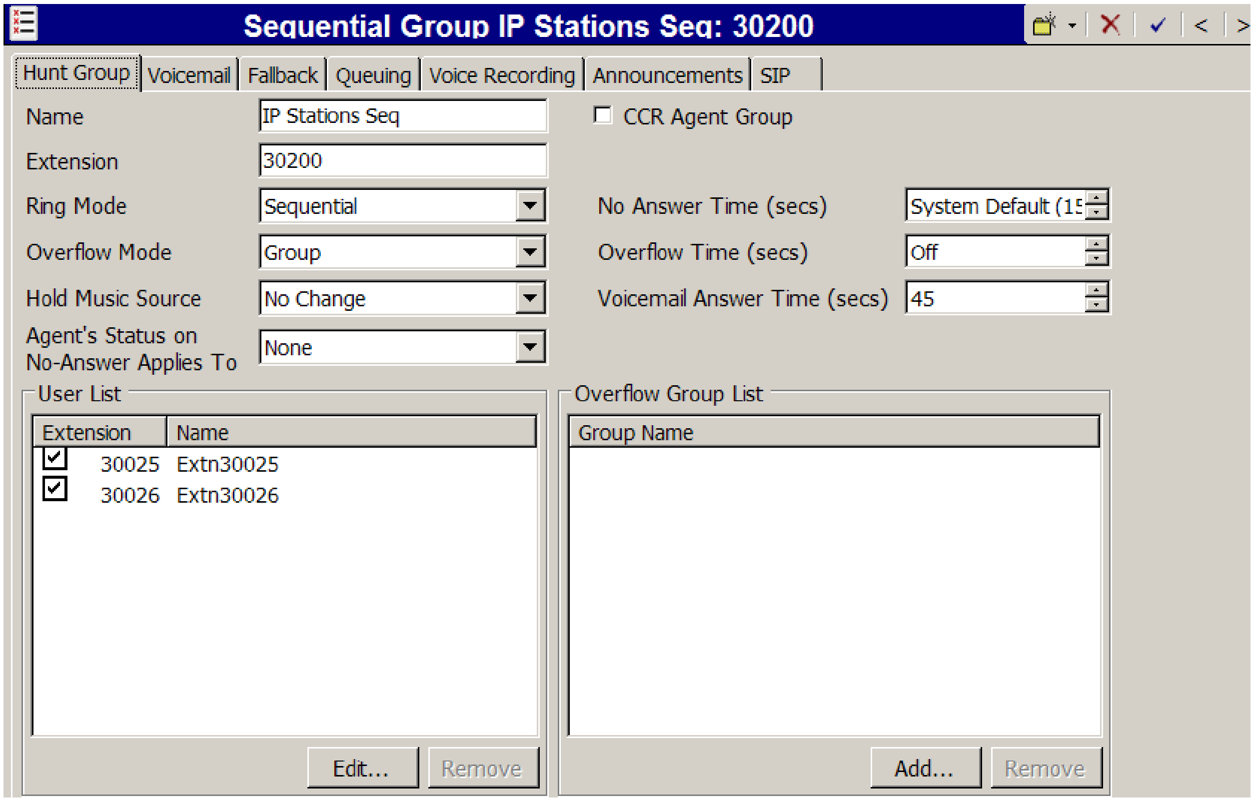
The following screen shows the **Hunt Group** tab for hunt group 200. This hunt group was configured to contain the two digital telephones x201 and x203 in **Figure 1**. These telephones will both ring when the hunt group number is called, due to the **Ring Mode** setting "Collective". Click the **Edit** button to change the **User List**.



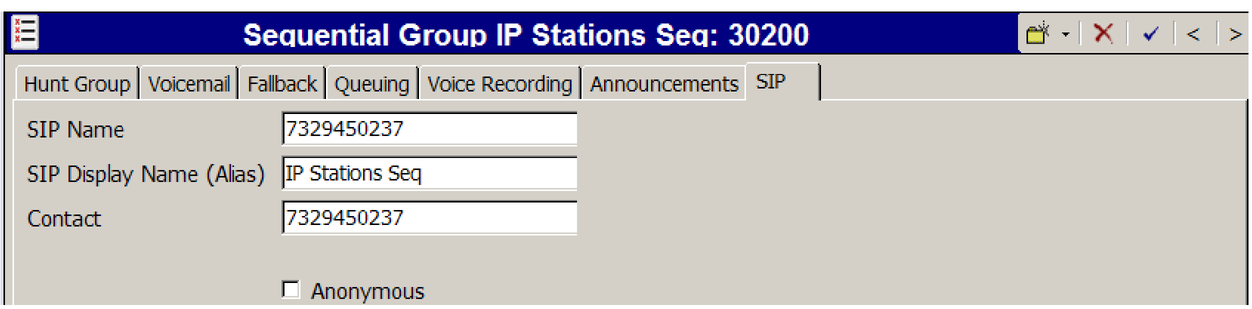
The following screen shows the **SIP** tab for hunt group 200. The **SIP Name** and **Contact** are configured with Verizon DID 7329450236. Later, in Section 3.7, an Incoming Call Route will map 7329450236 to this hunt group.



The following screen shows the **Hunt Group** tab for another hunt group 30200. This hunt group was configured to contain the two IP telephones x30025 and x30026 in **Figure 1**. These telephones will ring sequentially when the hunt group number is called, due to the **Ring Mode** setting "Sequential". That is, extension 30025 will ring first. If unanswered, extension 30026 will ring, and so on (for larger groups).



The following screen shows the **SIP** tab for hunt group 30200. The **SIP Name** and **Contact** are configured with Verizon DID 7329450237. Later, in Section 3.7, an incoming call route will map this same Verizon DID number to this hunt group.

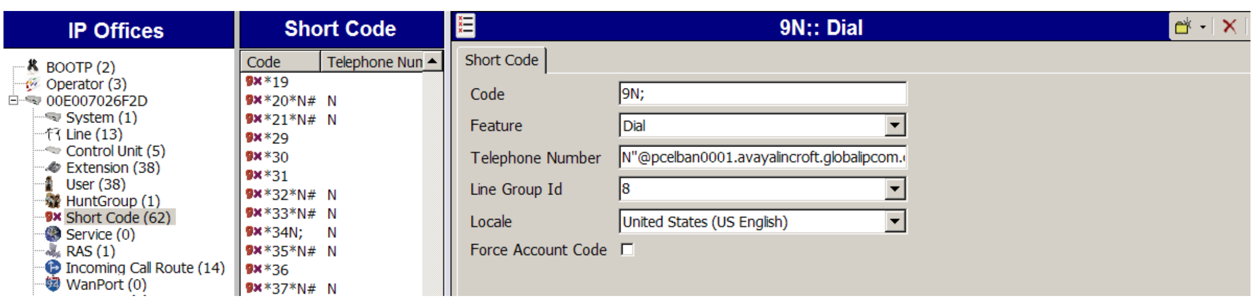


3.6. Short Codes

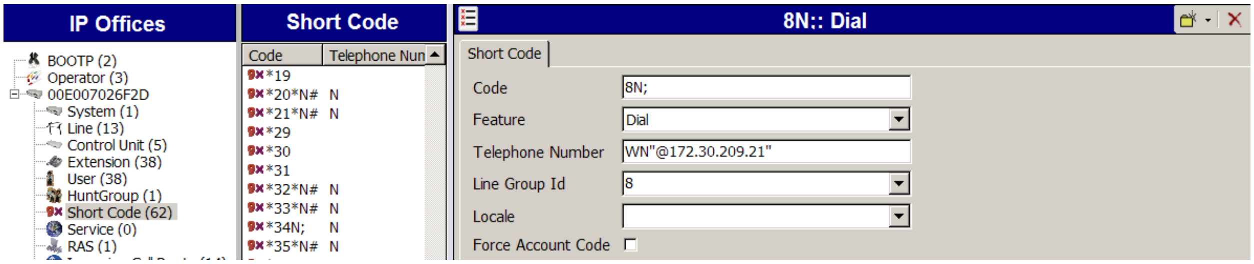
In this section, various examples of IP Office short codes will be illustrated. To add a short code, right click on **Short Code** in the Navigation pane, and select **New.** To edit an existing short code, click **Short Code** in the Navigation pane, and the short code to be configured in the Group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

In the screen shown below, the short code "9N;" is illustrated. 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**"** with the text string beginning with @ in quotes**.** Below, the Verizon provided domain shown in **Figure 1** is configured. 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 (Section 3.4)**.**

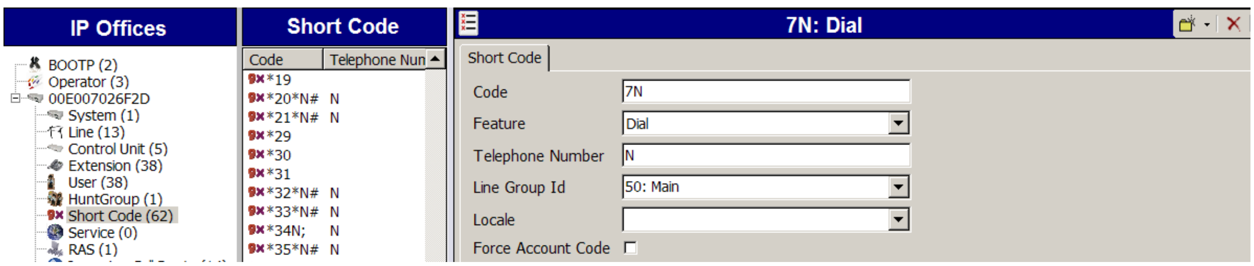
This simple short code will allow an IP Office user to dial the digit 9 followed by any telephone number, symbolized by the letter **N**, to reach the SIP Line to Verizon business. **N** can be any number such as a 10-digit number, a 1+10 digit number, a toll free number, directory assistance (e.g., 411), etc. This short code approach has the virtue of simplicity, but does not provide for alternate routing or an awareness of end of user dialing. When a users dial 9 plus the number, IP Office must wait for an end of dialing timeout before sending the SIP INVITE to Verizon Business. Click the **OK** button (not shown).



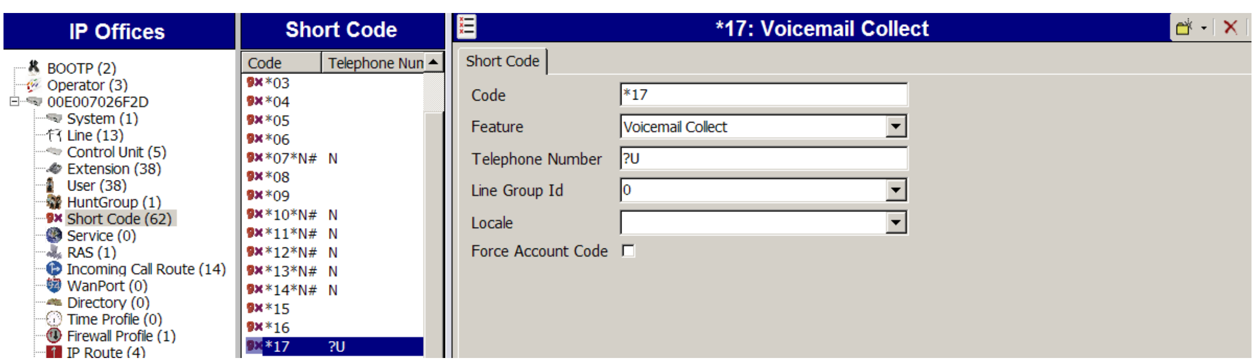
Optionally, add or edit a short code that can be used to access the SIP Line anonymously. In the screen shown below, the short code "8N;" is illustrated. This short code is similar to the "9N;" short code except that the Verizon IP Address rather than the domain is entered in the **Telephone Number** field. This is done for variety; either method can be used. The Telephone Number field begins with the letter **W,** which means "withhold the outgoing calling line identification". In the case of the SIP Line to Verizon documented in these Application Notes, when a user dials 8 plus the number, IP Office will include the user's telephone number in the P-Asserted-Identity (PAI) header along with "Privacy: Id". Verizon will allow the call due to the presence of a valid DID in the PAI header, but will prevent presentation of the caller id to the called PSTN destination. See Section 5 for an example verification with Wireshark trace.



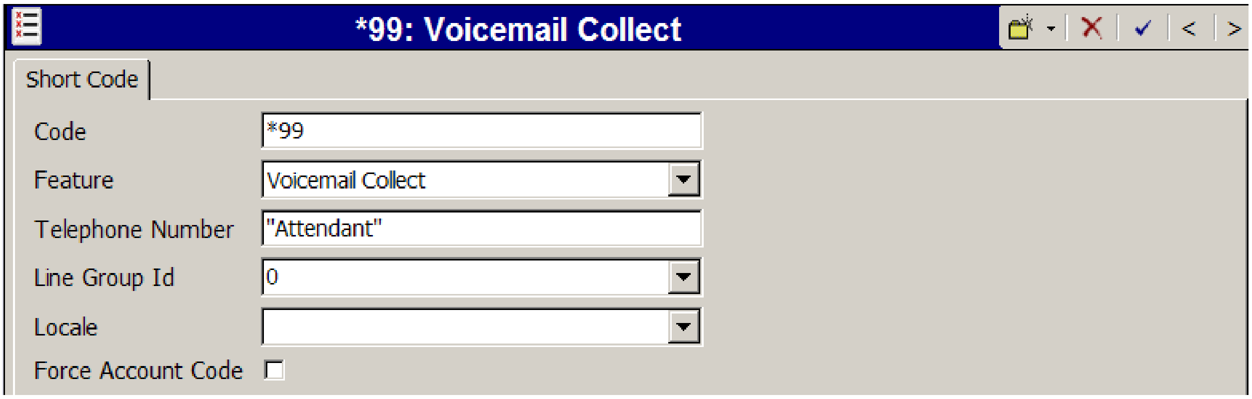
The simple "9N;" and "8N;" short codes illustrated previously do not provide a means of alternate routing if the primary Verizon SIP line is out of service or temporarily not responding. When alternate routing options and/or more customized analysis of the digits following the short code are desired, the Automatic Route Selection (ARS) feature may be used. In the following example screen, the short code 7N is illustrated for access to ARS. When the IP Office user dials 7 plus any number N, rather than being directed to a specific **Line Group Id**, the call is directed to **Line Group ID** "50: Main", configurable via ARS. See Section 3.8 for example ARS route configuration for "50: Main" as well as a backup route.



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 DID 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 3.7.



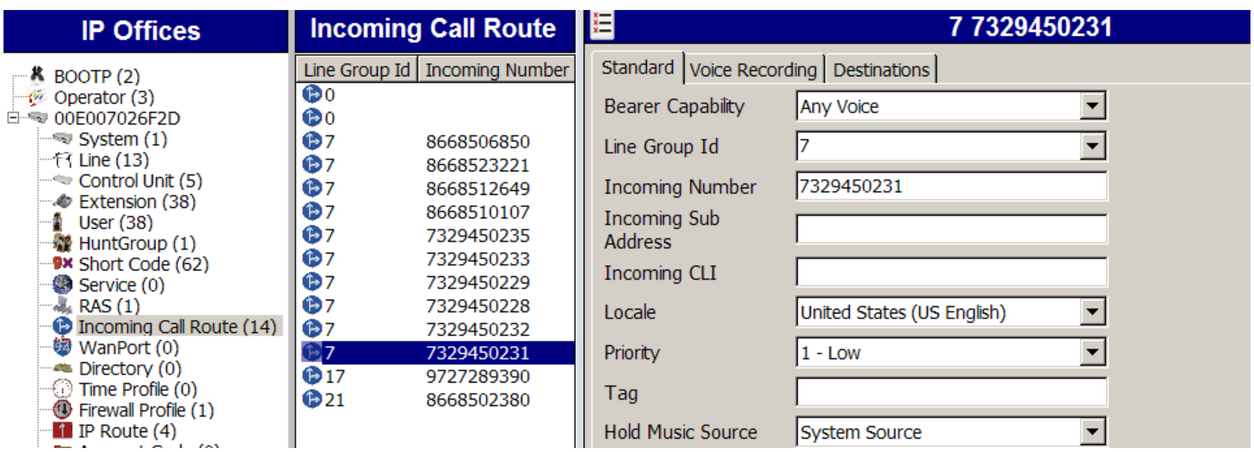
The following screen illustrates another short code that acts like a feature access code rather than a means to access a SIP Line. In this case, the **Code** "*99" is defined for **Feature** "Voicemail Collect" and Telephone Number "Attendant". This short code will be used as a means to allow a Verizon DID to be programmed to route directly to a Voicemail Pro "Attendant" (auto-attendant) application, via inclusion of this short code as the destination of an Incoming Call Route. See Section 3.7.



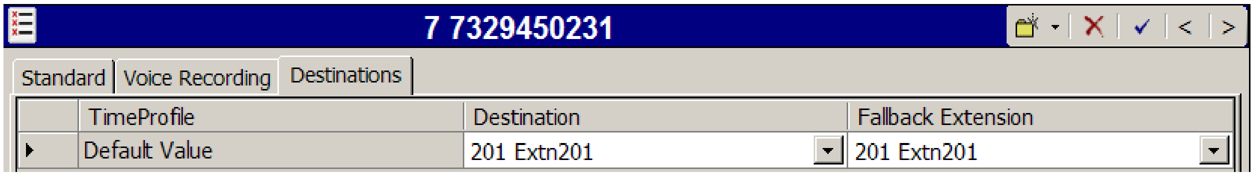
3.7. Incoming Call Routes

In this section, IP Office Incoming Call Routes are illustrated. Each Incoming Call Route will map a specific Verizon Business DID number to a destination user, group, or function on IP Office. To add an incoming call route, right click on **Incoming Call Route** in the Navigation pane, and select **New.** To edit an existing incoming call route, select **Incoming Call Route** in the Navigation pane, and the appropriate incoming call route to be configured in the Group pane.

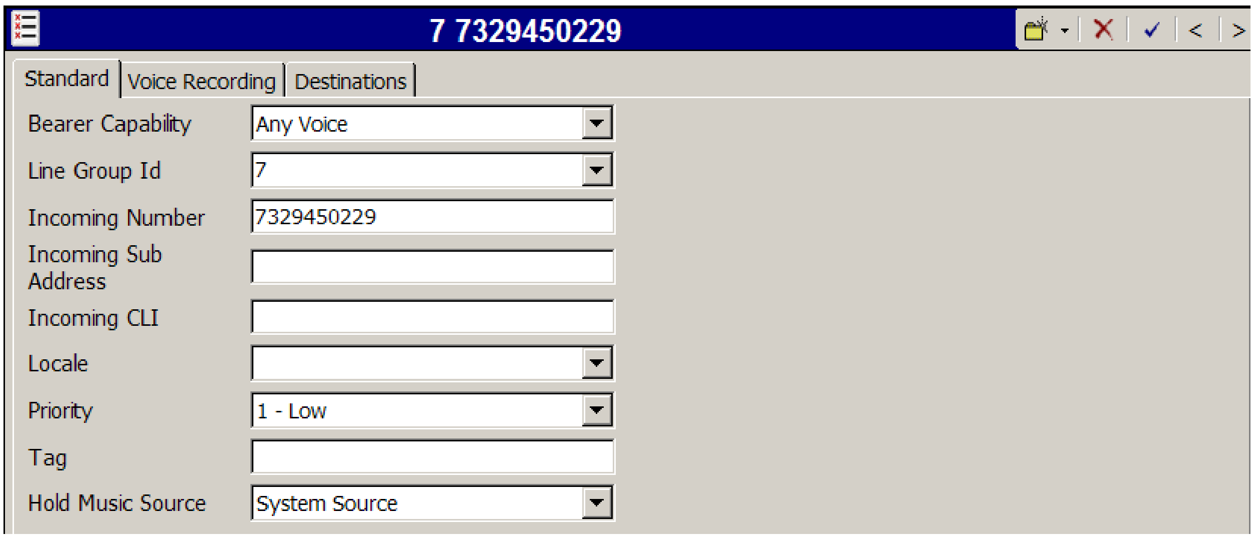
In the screen shown below, the incoming call route for **Incoming Number** "7329450231" is illustrated. The **Line Group Id** is 7, matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to Verizon Business in Section 3.4.



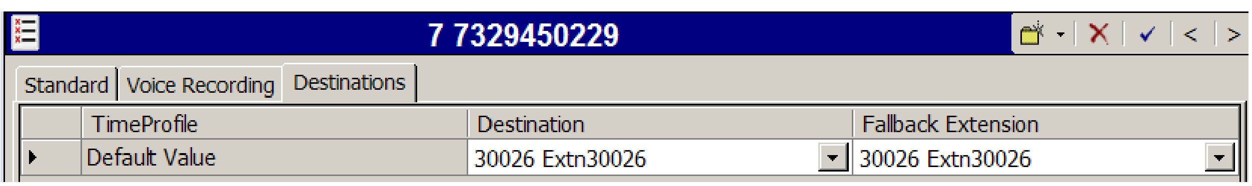
Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when a PSTN user dials 7329450231. As shown in **Table 1**, 7329450231 is the DID number associated with IP Office user extension 201.



In the screen shown below, the incoming call route for **Incoming Number** "7329450229" is illustrated. The **Line Group Id** is 7, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in Section 3.4.

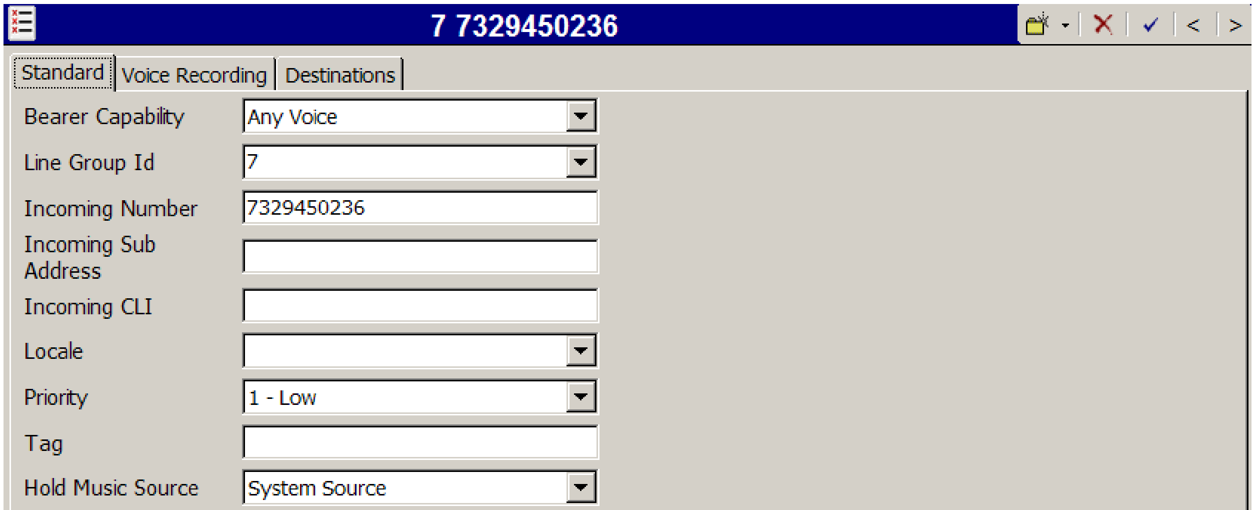


Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when a PSTN user dials 7329450229. As shown in **Table 1**, 7329450229 is the DID number associated with IP Office user extension 30026.

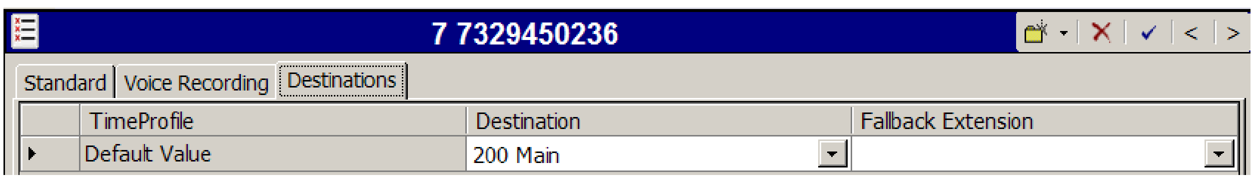


Incoming Call Routes for other direct mappings of DID numbers to IP Office users listed in **Table 1** are omitted here, but can be configured in the same fashion.

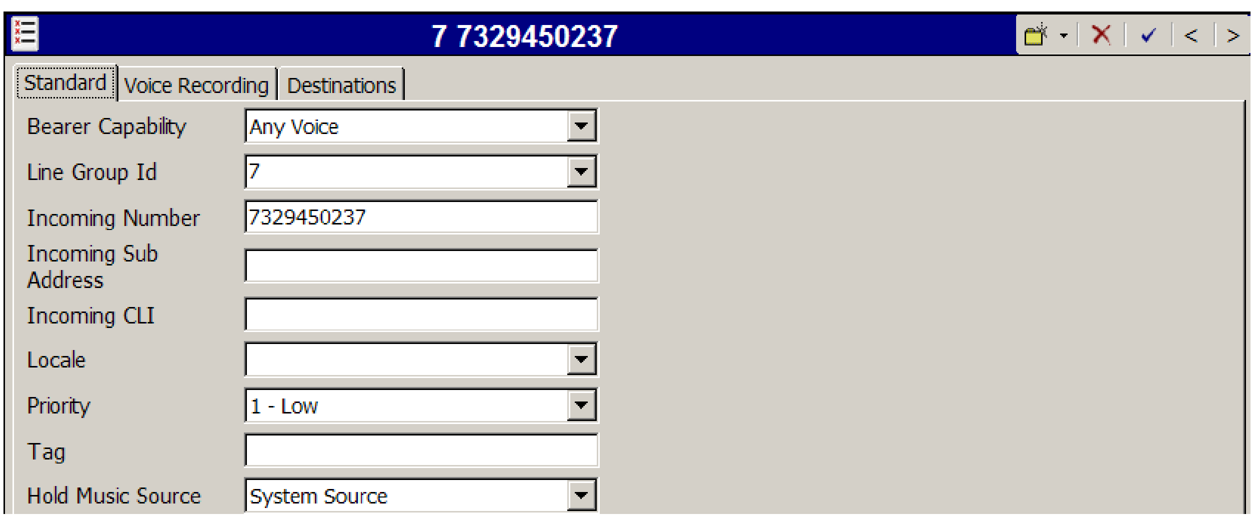
In the screen shown below, the incoming call route for **Incoming Number** "7329450236" is illustrated. The **Line Group Id** is 7, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in Section 3.4.



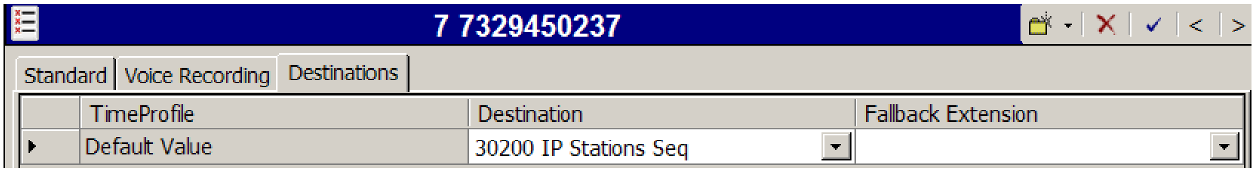
Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when a PSTN user dials 7329450236. In this case, the destination is the hunt group "200 Main" whose configuration is shown in Section 3.5.3.



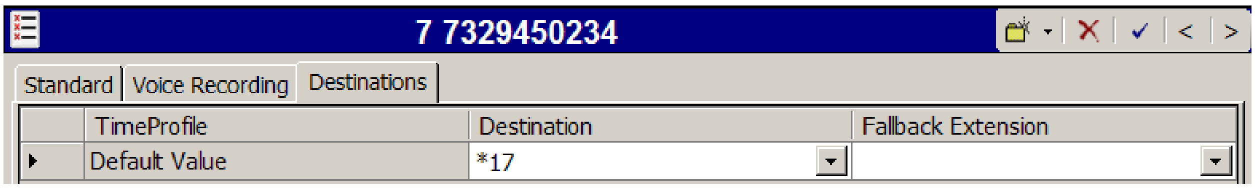
In the screen shown below, the incoming call route for **Incoming Number** "7329450237" is illustrated. The **Line Group Id** is 7, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in Section 3.4.



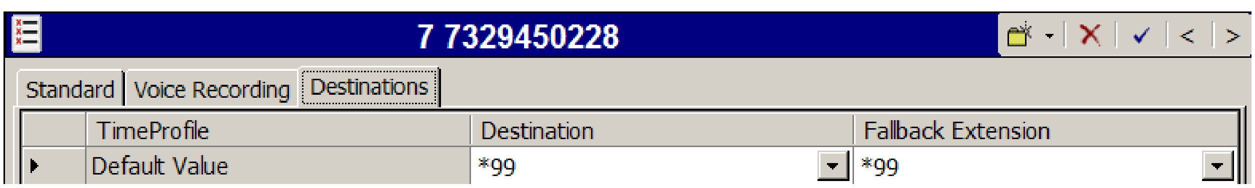
Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when a PSTN user dials 7329450237. In this case, the destination is the hunt group "30200 IP Stations Seq" whose configuration is shown in Section 3.5.3.



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. For example, the following **Destinations** tab for an incoming call route contains the **Destination** "*17" entered manually. The dial string "*17" is the short code for "Voicemail Collect", as shown in Section 3.6. An incoming call to 732-945-0234 will be delivered directly to voice mail, allowing the caller to log-in to voicemail and access messages.



Similar, the following **Destinations** tab for an incoming call route contains the **Destination** "*99" entered manually. The dial string "*99" is the short code for accessing the "Attendant" application on Voicemail Pro, as shown in Section 3.6. An incoming call to 732-945-0228 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 to IP Office users. The configuration of the "Attendant" application on Voicemail Pro is outside the intended scope of these Application Notes.



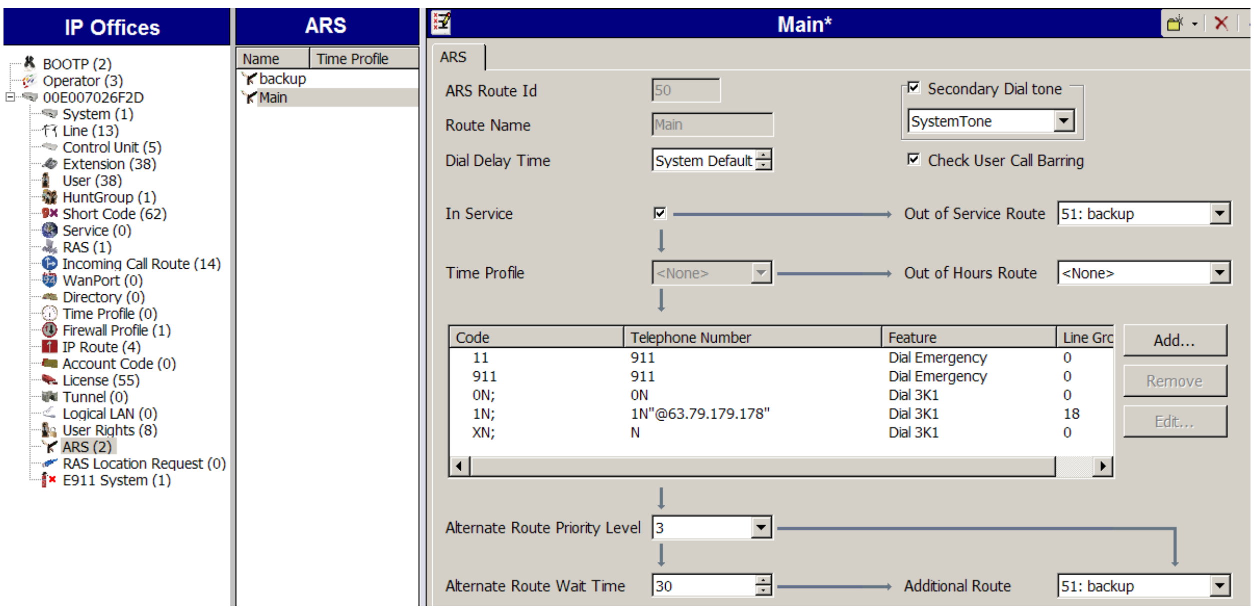
3.8. ARS and Alternate Routing

While detailed coverage of ARS is beyond the scope of these Application Notes, this section includes basic ARS screen illustrations and considerations. ARS is illustrated here mainly to illustrate alternate routing should the SIP Line be out of service or temporarily not responding.

Optionally, Automatic Route Selection (ARS) can be used rather than the simple "9N;" short code approach documented in Section 3.6. With ARS, secondary dial tone can be provided after the access code, time-based routing criteria can be introduced, and alternate routing can be specified so that a call can re-route automatically if the primary route or outgoing line group is not available. Although not shown in this section, ARS also facilitates more specific dialed telephone number matching, enabling immediate routing and alternate treatment for different types of numbers following the access code. For example, if all 1+10 digit calls following an access code should use the SIP Line preferentially, but other local or service numbers following the access code should prefer a different outgoing line group, ARS can be used to distinguish the call behaviors.

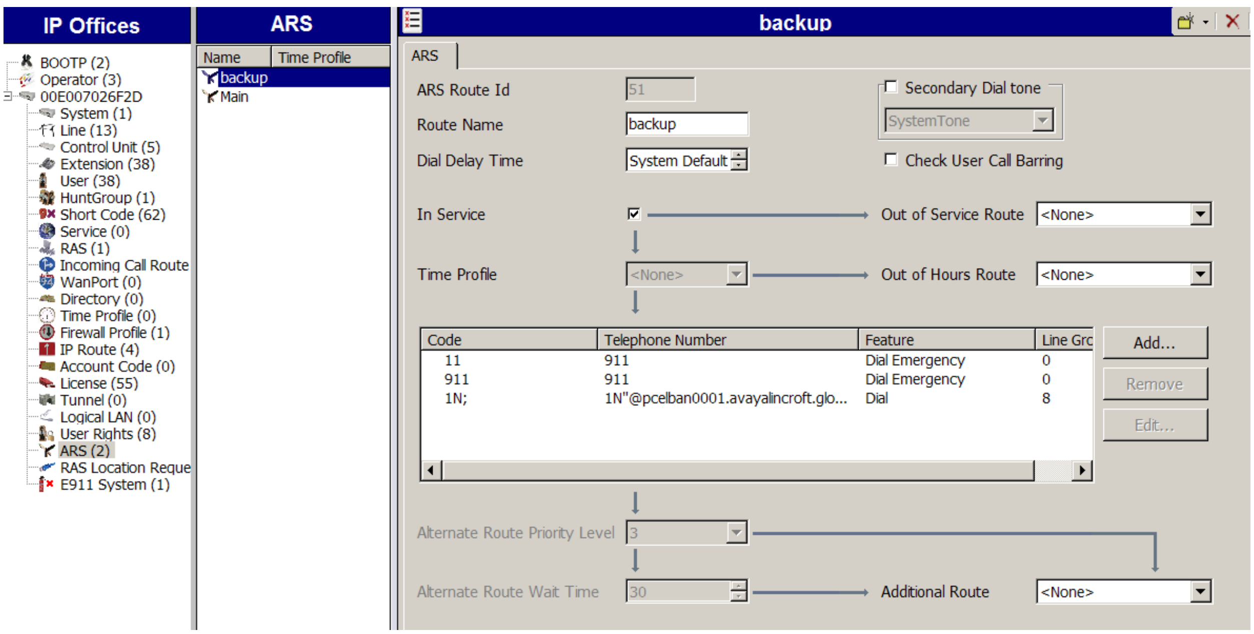
To add a new ARS route, right-click **ARS** in the Navigation pane, and select **New**. To view or edit an existing ARS route, select **ARS** in the Navigation pane, and select the appropriate route name in the Group pane.

The following screen shows an example ARS configuration for the route named "Main". The **In Service** parameter refers to the ARS form itself, not the Line Groups that may be referenced in the form. If the **In Service** box is un-checked, calls are routed to the ARS route name specified in the **Out of Service Route** parameter. IP Office short codes may also be defined to allow an ARS route to be disabled or enabled from a telephone. The configurable provisioning of an Out of Service Route, and the means to manually activate the Out of Service Route can be helpful for scheduled maintenance or other known service-affecting events for the primary route.



Assuming the primary route is in-service, the number passed from the short code used to access ARS (e.g., 7N in Section 3.6) can be further analyzed to direct the call to a specific Line Group ID. Per the example screen above, if the user dialed 7-1-908-848-5704, the call would be directed to Line Group 18, another SIP Line that exists in the configuration that is not described in these Application Notes. If Line Group 18 cannot be used, the call can automatically route to the route name configured in the **Additional Route** parameter in the lower right of the screen. Since alternate routing can be considered a privilege not available to all callers, IP Office can control access to the alternate route by comparing the calling user's priority to the value in the **Alternate Route Priority Level** field.

The following screen shows an example ARS configuration for the route named "backup", ARS Route ID 51. Continuing the example, if the user dialed 7-1-908-848-5704, and the call could not be routed via the primary route "50: Main" described above, the call will be delivered to this "backup" route. Per the configuration shown below, the call will be delivered to Line Group 8, using the SIP Line to Verizon Business IP Trunk service described in these Application Notes. The configuration of the **Code**, **Telephone Number**, **Feature**, and **Line Group ID** for an ARS route is similar to the configuration already shown for short codes in Section 3.6. In this case, the originally dialed number (sans the short code 7) is delivered in the Request URI along with the Verizon FQDN (i.e., the contents of the INVITE sent to Verizon are the same as the 9-1-908-848-5704 short code 9 approach from Section 3.6). Additional codes (e.g., 411, 0+10, etc.) can be added to the ARS route by pressing the **Add…** button to the right of the list of previously configured codes.



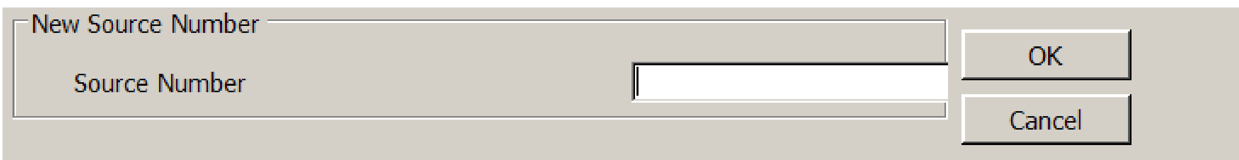
In the testing associated with the configuration, calls were successfully delivered to SIP Line 8 via both the primary ARS route "50: Main" (via changes to "50: Main") as well as the backup ARS route shown above. If a primary route experiences a network outage such that no response is received to an outbound INVITE, IP Office successfully routes the call via the backup route. The user receives an audible tone when the re-routing occurs and may briefly see "Waiting for Line" on the display. The testing verified that the INVITE was sent to the primary route, and the call re-routed upon timeout. The call was made right after a failure of the primary route was induced, so IP Office had not yet marked the SIP Line out of service as a result of no response to SIP OPTIONS. Testing also verified that calls can be delivered to Verizon via the alternate route when the primary route was manually marked out-of-service, or known to be out-of-service due to prior failure of SIP OPTIONS.

3.9. Privacy / Anonymous Calls

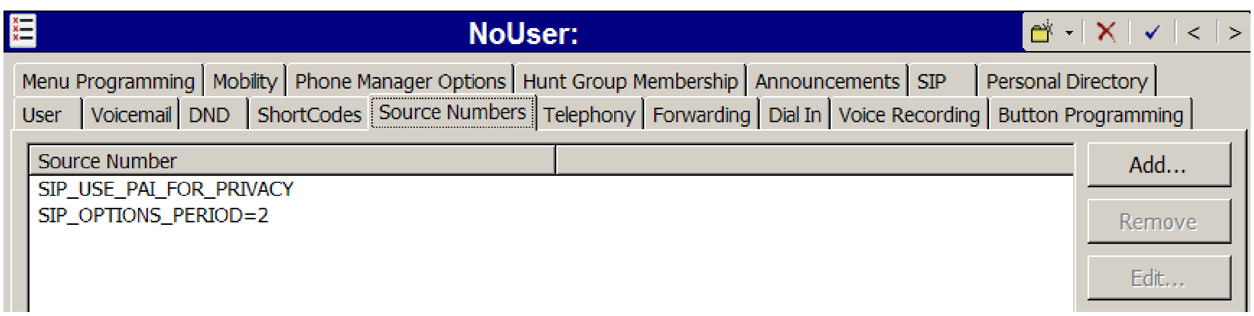
As described in Section 3.6, an IP Office user whose calling line identification is not typically withheld from the network can request privacy in the sample configuration by dialing the short code 8 to access the SIP Line. The Avaya 1600-Series IP Telephones can also request privacy for a specific call, without dialing a unique short code, using Features → Call Settings → Withhold Number. Specific users may be configured to always withhold calling line identification by checking the **Anonymous** field in the **SIP** tab for the user (Section 3.5).

To configure IP Office to include the caller's DID number in the P-Asserted-Identity SIP header, required by Verizon Business to admit an otherwise anonymous caller to the network, the following procedure may be used.

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_USE_PAI_FOR_PRIVACY**. Click **OK**.



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

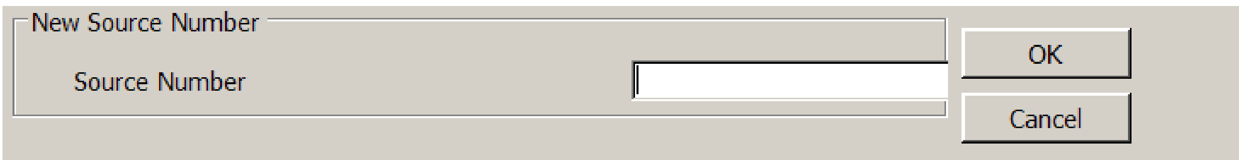


3.10. SIP Options Frequency

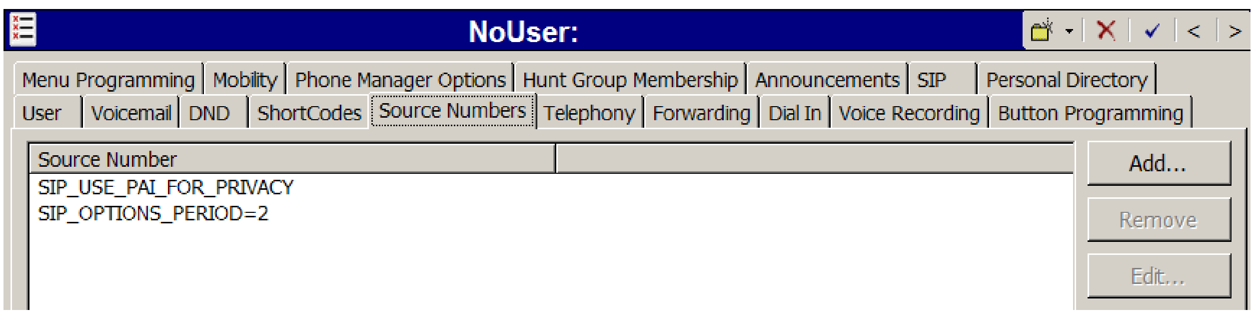
In Section 3.4, the SIP Line to Verizon Business is shown with the **Check OOS** box checked. In the sample configuration, IP Office periodically checks the health of the SIP Line by sending a SIP OPTIONS message. If there is no response, IP Office can mark the trunk out of service. Although ARS as shown in Section 3.8 can include alternate routes to complete calls even if the far-end is not responding, IP Office must wait for the outbound INVITE to timeout before route advance. Once the SIP OPTIONS maintenance recognizes that the SIP Line is out-of-service, new calls will no longer be delayed before route advance. Also, once the problem with the SIP Line is resolved, the SIP OPTIONS maintenance will automatically bring the link back to the in-service state.

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 3.3.2 and Section 3.4

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 Interval**. In the sample configuration, the value used for X was 2 minutes. Click **OK**.



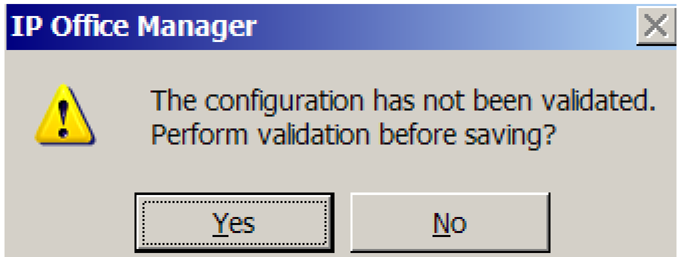
The source number **SIP_OPTIONS_PERIOD=2** should now appear in the list of Source Numbers as shown below.



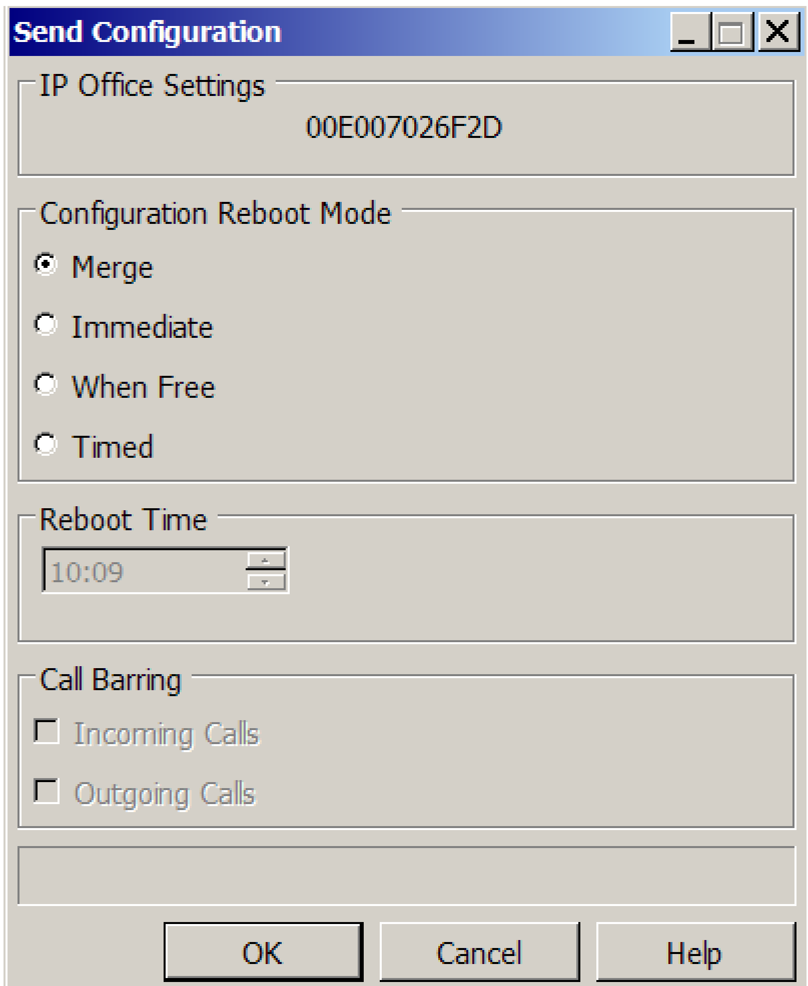
With this configuration, Binding Refresh Intervals of 30 seconds and 60 seconds were tested successfully. That is, IP Office sourced SIP OPTIONS every 30 or 60 seconds, depending on the value configured in the Binding Refresh Interval, since the Binding Refresh Interval was less than the value configured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

3.11. 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.



4. Verizon Business Configuration

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

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

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

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

For service provisioning, Verizon will require the customer IP address used to reach the Avaya 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 Direct Inward Dialed (DID) numbers shown in **Figure 1** and **Table 1**. This information was used to complete the Avaya IP Office configuration shown in Section 3.

5. Verifications

The Avaya IP Office location was connected to the Verizon Business IP Trunk Service, as depicted in **Figure 1.** Avaya IP Office was configured to use the commercially available SIP Trunking solution provided by the Verizon Business IP Trunk SIP Trunk Service. This allowed Avaya IP Office users to make calls to the PSTN and receive calls from the PSTN via the Verizon Business IP Trunk SIP Trunk Service.

5.1. Verification Summary

This section summarizes the verification testing associated with these Application Notes. Successful SIP trunk interoperability compliance testing included the following:

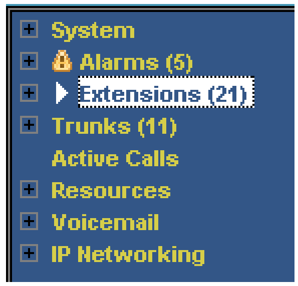
- Incoming calls from the PSTN were routed to the DID numbers assigned by Verizon Business to the Avaya IP Office location. These incoming PSTN calls arrived via the SIP Line and were answered by Avaya H.323 telephones, Avaya digital telephones, analog telephones, Avaya IP Office Softphone, and Avaya IP Office Voicemail Pro voicemail and auto-attendant applications. The display of caller ID on display-equipped Avaya IP Office telephones was verified.
- Incoming calls answered by members of collective and sequential Hunt Groups were verified.
- Outgoing calls from the Avaya IP Office location to the PSTN were routed via the SIP Line to Verizon Business. These outgoing PSTN calls were originated from Avaya 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.
- Proper disconnect when the caller abandons a call before answer for both inbound and outbound calls.
- Proper disconnect when the IP Office party or the PSTN party hang-up an active call

- Proper busy tone heard when an IP Office user calls a busy PSTN user, or a PSTN user calls a busy IP Office user (i.e., if no redirection is configured for user busy conditions)
- Various outbound PSTN call types were tested including long distance, international, toll-free, operator assisted, and directory assistance calls.
- Requests for privacy (i.e., caller anonymity) for IP Office outbound calls to the PSTN were verified. That is, when privacy is requested by IP Office, outbound PSTN calls can be successfully completed while withholding the caller ID from the displays of display-equipped PSTN telephones.
- Privacy requests for inbound calls from the PSTN to IP Office users were verified. That is, when privacy is requested by a PSTN caller, the inbound PSTN call can be successfully completed to an IP Office user while presenting an “anonymous” display to the IP Office user.
- SIP OPTIONS monitoring of the health of the SIP trunk was verified. Both Verizon Business and IP Office can monitor health using SIP OPTIONS. The Avaya IP Office configurable control of SIP OPTIONS timing was exercised successfully.
- IP Office outbound calls were placed with simple short codes as well as using ARS. Using ARS, the ability of IP Office to route-advance to an alternate route when the primary SIP line is not responding was exercised. The Line Group associated with the Verizon Business SIP Line can be the primary line group chosen for a call, or an alternate line group selected upon failure of a primary line.
- Incoming and outgoing calls using the G.729(a) and G.711 ULAW codecs.
- DTMF transmission using RFC 2833 with successful voice mail navigation for G.729a and G.711MU for incoming and outgoing calls. Successful navigation of a simple auto-attendant application configured on IP Office Voicemail Pro.
- The “callback” feature of Avaya Voicemail Pro was tested successfully. When 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 SIP Line to Verizon Business. 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 and outbound long holding time call stability
- Telephony features such as call waiting, hold, transfer, and conference.
- Call Forwarding to PSTN destinations (see Section 1.1)
- Mobile twinning to a mobile phone, presenting true calling party information to the mobile phone. Outbound mobile call control was also verified successfully (e.g., using DTMF on a twinned call to place new calls and create a conference via a mobile phone).
- Proper DiffServ markings for IP Office SIP signaling and RTP media

5.2. 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 Navigation pane allows various views into the system status. The following screen shows an example Extension Summary report, accessible by selecting **Extensions** from the navigation pane as shown below.



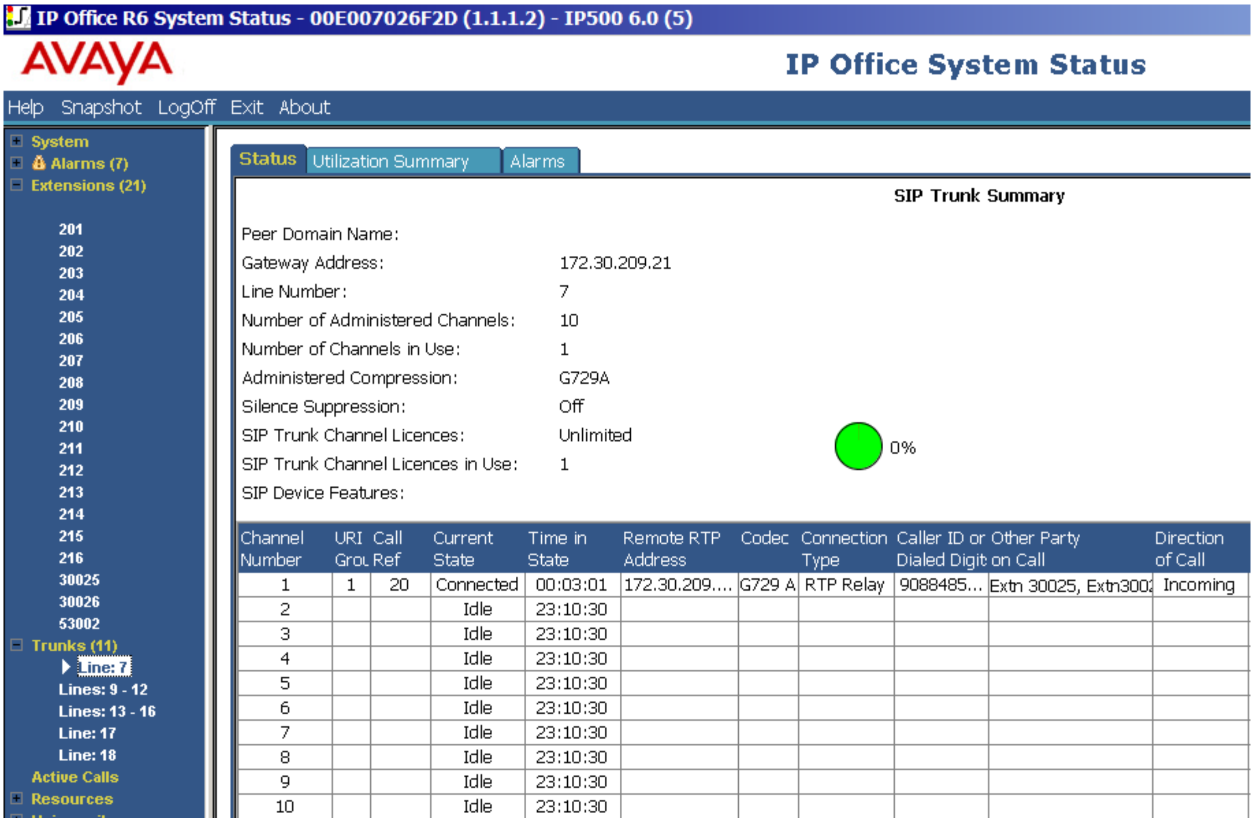
At the time this summary report was captured, the IP Office Softphone was logged in as user 30026, causing the Avaya 1616 telephone that had been logged in as 30026 to enter "NoUser" mode (as expected). This can be observed in the last two rows below. In the final row, note that the Telephone Type for the Avaya IP Office Softphone logged in as Current User Extension 30026 is shown as "SIP Softphone", with an arbitrary Home Extension Number.

Extension Summary

You can get more information about an extension by double-clicking the Home Extension Number.

Home Extension Number	Current User Extension	Current User Name	Module/ Slot/ IP Address	Port Number/ MAC Address	Telephone Type
		NoUser	IP DECT module		DECT IP
		NoUser	IP DECT module		DECT IP
201	201	Extn201	Slot: 3	1	5410
202			Slot: 3	2	unplugged
203	203	Extn203	Slot: 3	3	2410
204			Slot: 3	4	unplugged
205			Slot: 3	5	unplugged
206			Slot: 3	6	unplugged
207			Slot: 3	7	unplugged
208			Slot: 3	8	unplugged
209	209	Justa Fax	Slot: 4	1	POT (CLI Off)
210	210	Extn210	Slot: 4	2	POT (CLI Off)
211	211	Extn211	Slot: 4	3	POT (CLI Off)
212	212	Extn212	Slot: 4	4	POT (CLI Off)
213	213	Extn213	Slot: 4	5	POT (CLI Off)
214	214	Extn214	Slot: 4	6	POT (CLI Off)
215	215	Extn215	Slot: 4	7	POT (CLI Off)
216	216	Extn216	Slot: 4	8	POT (CLI Off)
30025	30025	Extn30025	1.1.1.100	00-07-3B-BF-22-0D	1616
30026		NoUser	1.1.1.101	00-07-3B-BF-22-0C	1616
53005	30026	Extn30026	1.1.1.110		SIP SoftPhone

The following screen shows an example **Status** tab for **Trunks → Line 7**, the SIP Line to Verizon. In this case, the screen was captured while there was an incoming call active to extension 30025, an Avaya 1616 IP Telephone. As can be observed, the call used Codec G.729a and Connection Type RTP relay.



The following screen shows an example **Extension Status** tab for **Extensions → 30025**. In this case, the screen was captured while there was an active incoming PSTN SIP Trunk call from PSTN user 908-848-5704 to extension 30025. Again, it can be observed that the call used G.729a and RTP relay. The Remote RTP Address (172.30.209.132) is in the Verizon network (i.e., signaled by Verizon in the SIP SDP).

Extension Status

Extension Number:	30025
IP address:	1.1.1.100
MAC address:	00-07-3B-BF-22-0D
Firmware Version:	1.2200
Gatekeeper:	Primary
Telephone Type:	1616
Current User Extension Number:	30025
Current User Name:	Extn30025
Forwarding:	Off
Twinning:	Off
Do Not Disturb:	Off
Message Waiting:	Off
Number of New Messages:	0
Phone Manager Type:	None
Licensed:	Yes
License Reserved:	No
Last Date and Time License Allocated:	2/15/2004 3:02:43 AM
Packet Loss Fraction:	0%
Jitter:	17.4ms
Round Trip Delay:	0ms
Connection Type:	RTP Relay
Codec:	G729 A
Remote RTP Address:	172.30.209.132

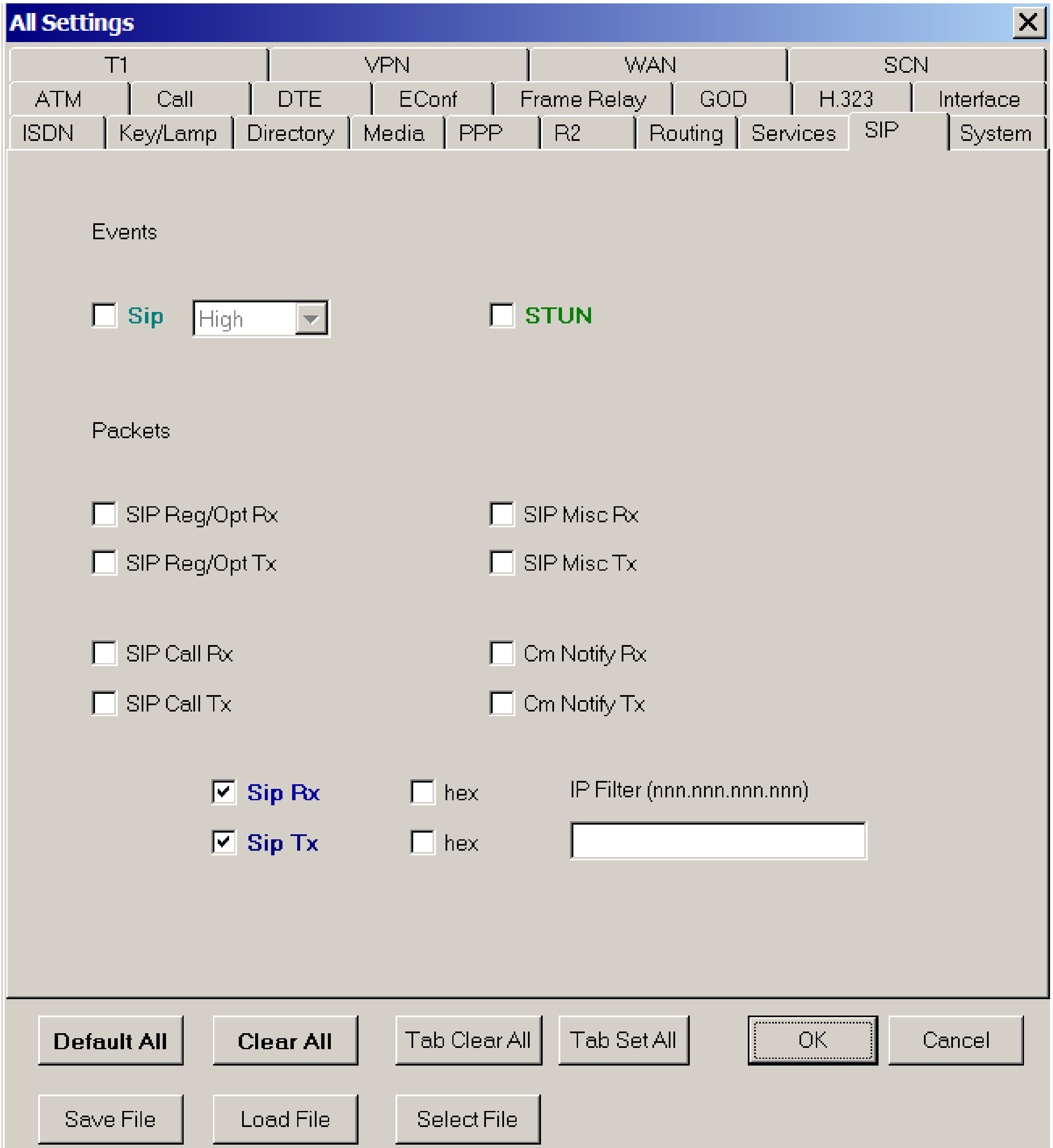
Button Number	Button Type	Call Ref	Current State	Time in State	Calling Number or Called Number	Direction	Other F
1	CA	20	Connected	00:01:10	9088485704@65.211....	Incoming	Line: 7
2	CA		Idle				
3	CA		Idle				

5.3. System Monitor Application

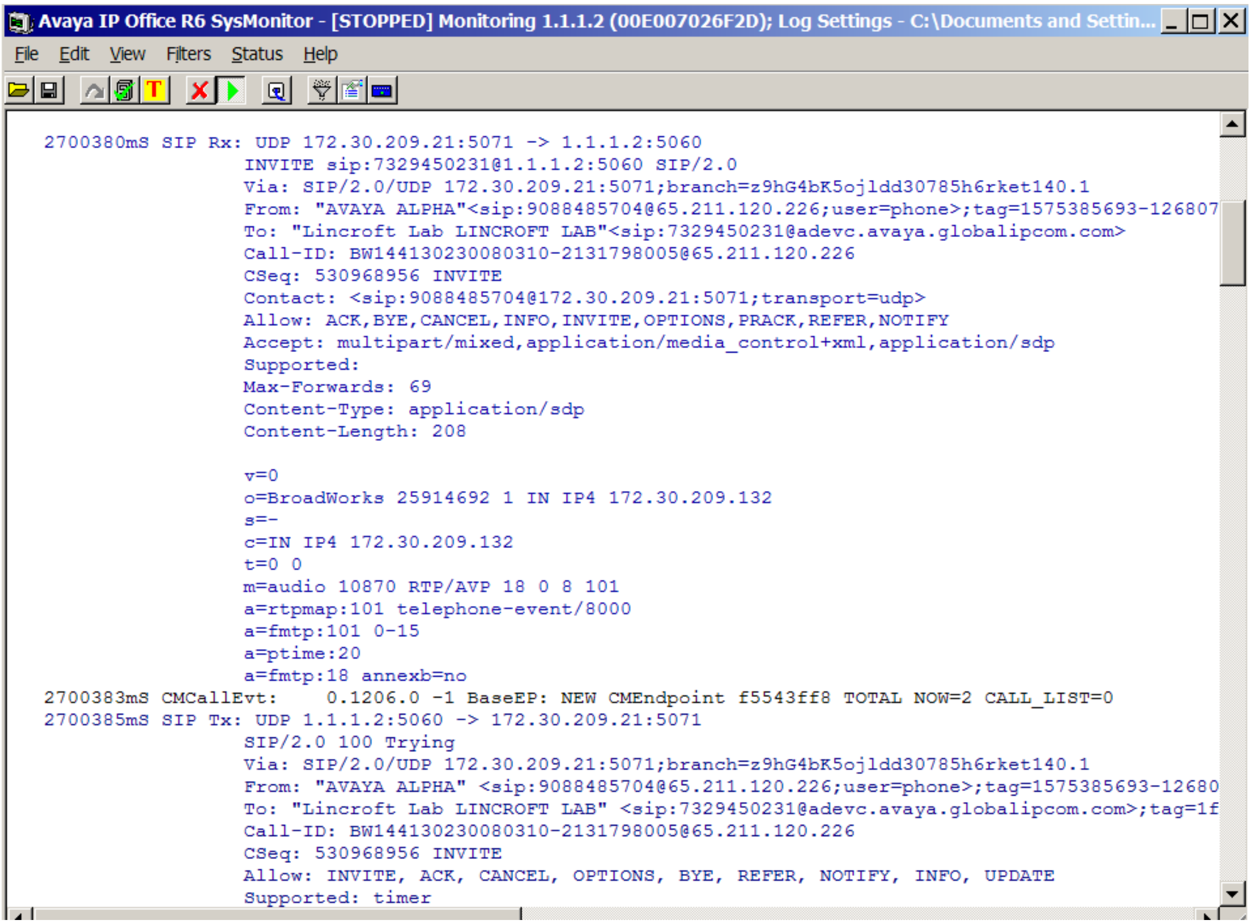
The System Monitor application can also be used to monitor or troubleshoot. The System Monitor application can typically be accessed from **Start → Programs → IP Office → Monitor**. See reference [IPO-MON] for more information.

The application allows the monitored information to be customized. To customize, select the button shown within the red rectangle below, or **Filters → Trace Options**.

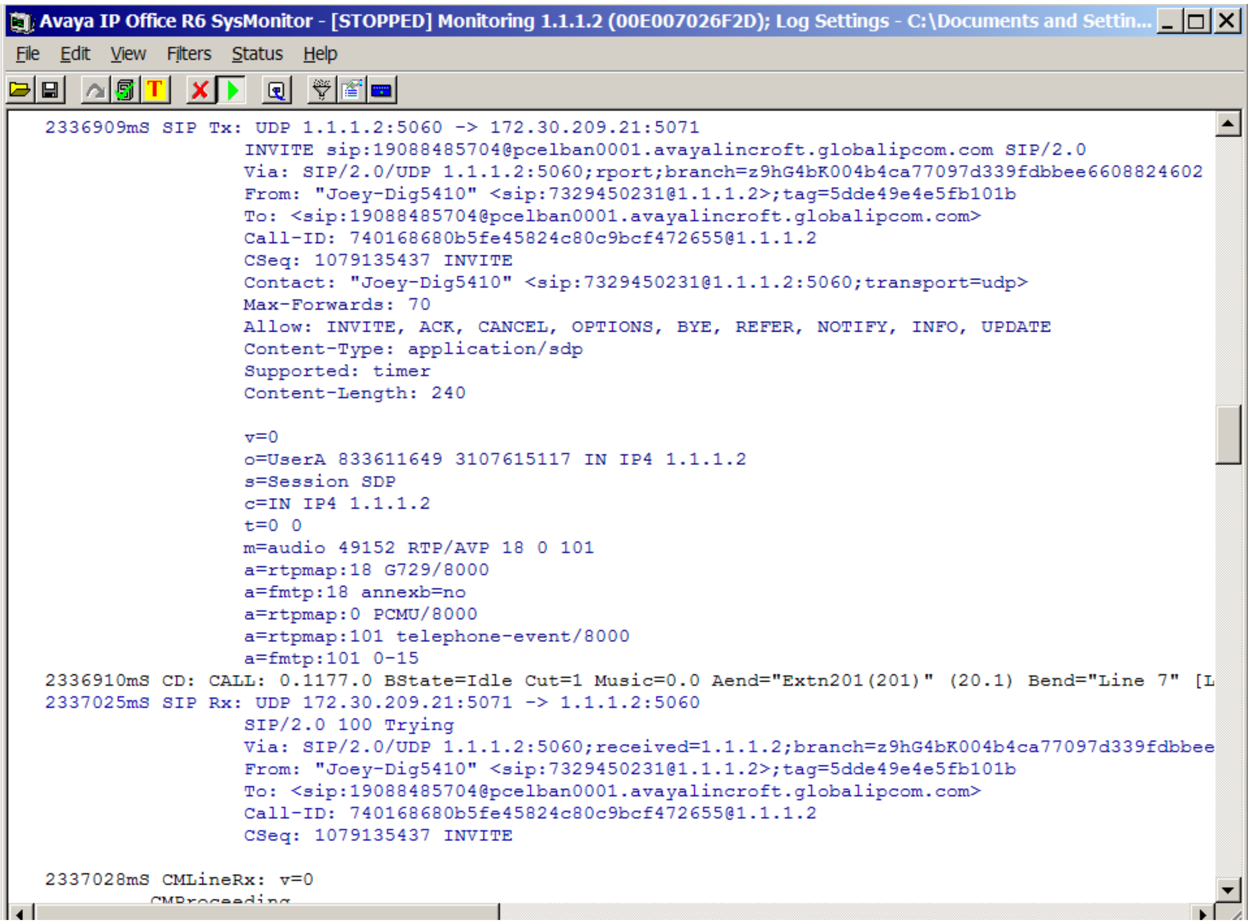
The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, all SIP messages will appear in the trace with the color blue. To customize the color, right-click on **SIP Rx** or **SIP Tx** and select the desired color.



The following shows a portion of an example SIP trace for an incoming call. This example window shows the inbound SIP INVITE from Verizon, and the 100 Trying returned by IP Office. In this case, the call was from PSTN user 908-848-5704 to the Verizon DID 732-945-0231.



The following shows a portion of an example trace for an outbound call. This example window shows the outbound SIP INVITE to Verizon, and the 100 Trying returned by Verizon. In this case, IP Office user 201 is calling PSTN user 1-908-848-5704. IP Office user 201 corresponds to Verizon DID 732-945-0231, which can be observed in the From header.



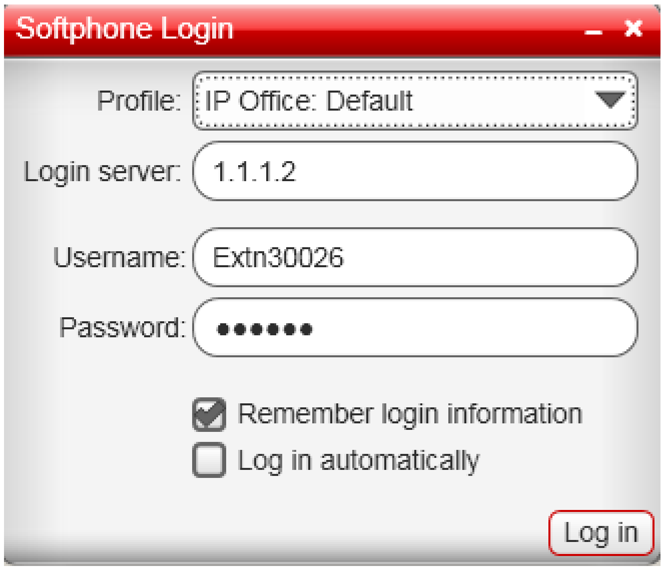

```
2336909mS SIP Tx: UDP 1.1.1.2:5060 -> 172.30.209.21:5071
                 INVITE sip:19088485704@pcelban0001.avayalincroft.globalipcom.com SIP/2.0
                 Via: SIP/2.0/UDP 1.1.1.2:5060;rport;branch=z9hG4bK004b4ca77097d339fdbbee6608824602
                 From: "Joey-Dig5410" <sip:7329450231@1.1.1.2>;tag=5dde49e4e5fb101b
                 To: <sip:19088485704@pcelban0001.avayalincroft.globalipcom.com>
                 Call-ID: 740168680b5fe45824c80c9bcf472655@1.1.1.2
                 CSeq: 1079135437 INVITE
                 Contact: "Joey-Dig5410" <sip:7329450231@1.1.1.2:5060;transport=udp>
                 Max-Forwards: 70
                 Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
                 Content-Type: application/sdp
                 Supported: timer
                 Content-Length: 240

                 v=0
                 o=UserA 833611649 3107615117 IN IP4 1.1.1.2
                 s=Session SDP
                 c=IN IP4 1.1.1.2
                 t=0 0
                 m=audio 49152 RTP/AVP 18 0 101
                 a=rtpmap:18 G729/8000
                 a=fmtp:18 annexb=no
                 a=rtpmap:0 PCMU/8000
                 a=rtpmap:101 telephone-event/8000
                 a=fmtp:101 0-15
2336910mS CD: CALL: 0.1177.0 BState=Idle Cut=1 Music=0.0 Aend="Extn201(201)" (20.1) Bend="Line 7" [L
2337025mS SIP Rx: UDP 172.30.209.21:5071 -> 1.1.1.2:5060
                 SIP/2.0 100 Trying
                 Via: SIP/2.0/UDP 1.1.1.2:5060;received=1.1.1.2;branch=z9hG4bK004b4ca77097d339fdbbee
                 From: "Joey-Dig5410" <sip:7329450231@1.1.1.2>;tag=5dde49e4e5fb101b
                 To: <sip:19088485704@pcelban0001.avayalincroft.globalipcom.com>
                 Call-ID: 740168680b5fe45824c80c9bcf472655@1.1.1.2
                 CSeq: 1079135437 INVITE

2337028mS CMLineRx: v=0
         CMProceeding
```

5.4. IP Office Softphone Application

The following screen shows the IP Office Softphone Login screen. In this case, the **Profile** "IP Office: Default" is chosen. The **Login server** is the IP Address of IP Office as configured in Section 3.1. In the **Username** and **Password** fields, enter the values configured for the **Name** and **Password** fields in the IP Office **User** tab, as shown in Section 3.5.

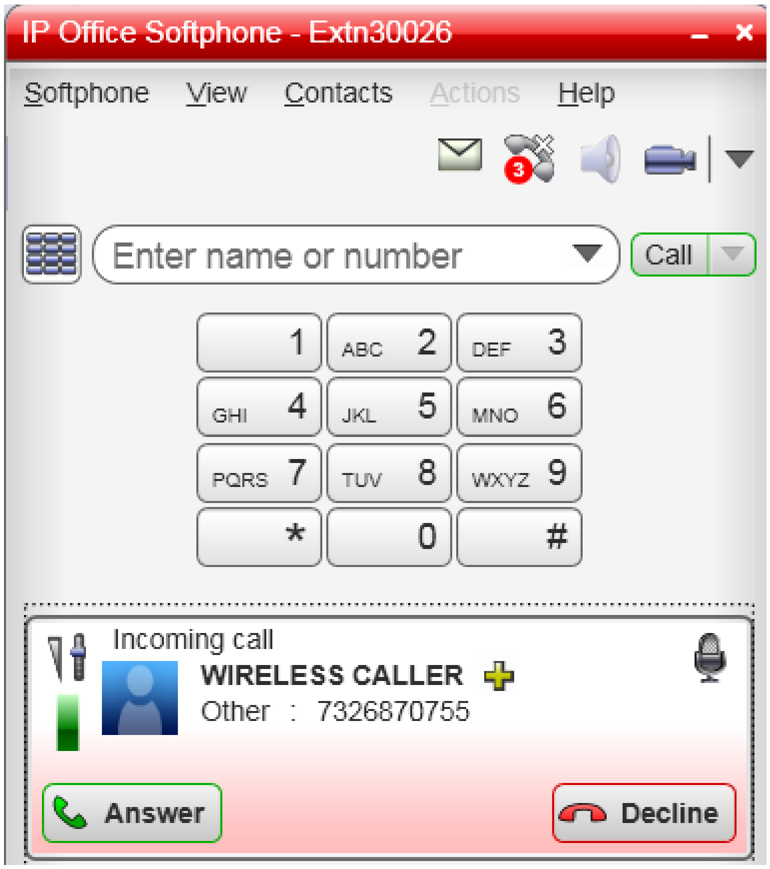


The following screen shows the IP Office Softphone Login screen. In this case, the **Profile** "IP Office: Low Bandwidth" is chosen. As noted in Section 1.1, if use of G.729a is desired for both inbound and outbound calls using the SIP Line configured in Section 3.4, the "IP Office: Low Bandwidth" option can be chosen when logging in to the IP Office Softphone.

The following screen capture shows the IP Office Softphone interface on an outbound PSTN call. In this case, the IP Office Softphone user dialed the PSTN user 908-848-5704 via the “9N;” short code illustrated in Section 3.6.

The following screen capture shows a portion of the IP Office Softphone interface as an inbound PSTN call is ringing the user. In this case, a mobile phone (732-687-0755) on the PSTN dialed 732-945-0229, and the Incoming Call Route illustrated in Section 3.7 sent the call to user 30026.

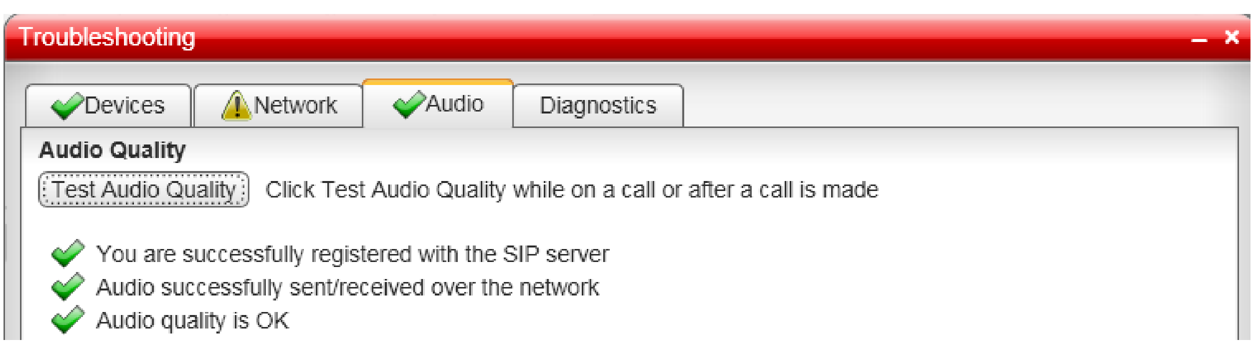
The IP Softphone user is currently logged in as 30026 and receives the call. If the **Answer** button is pressed, the call is answered successfully. If the **Decline** button is pressed, the call is redirected to the user's voice mail greeting, since this user has voice mail. The words "WIRELESS CALLER" appearing on the display were included in the Display Information in the From header of the inbound INVITE from Verizon.



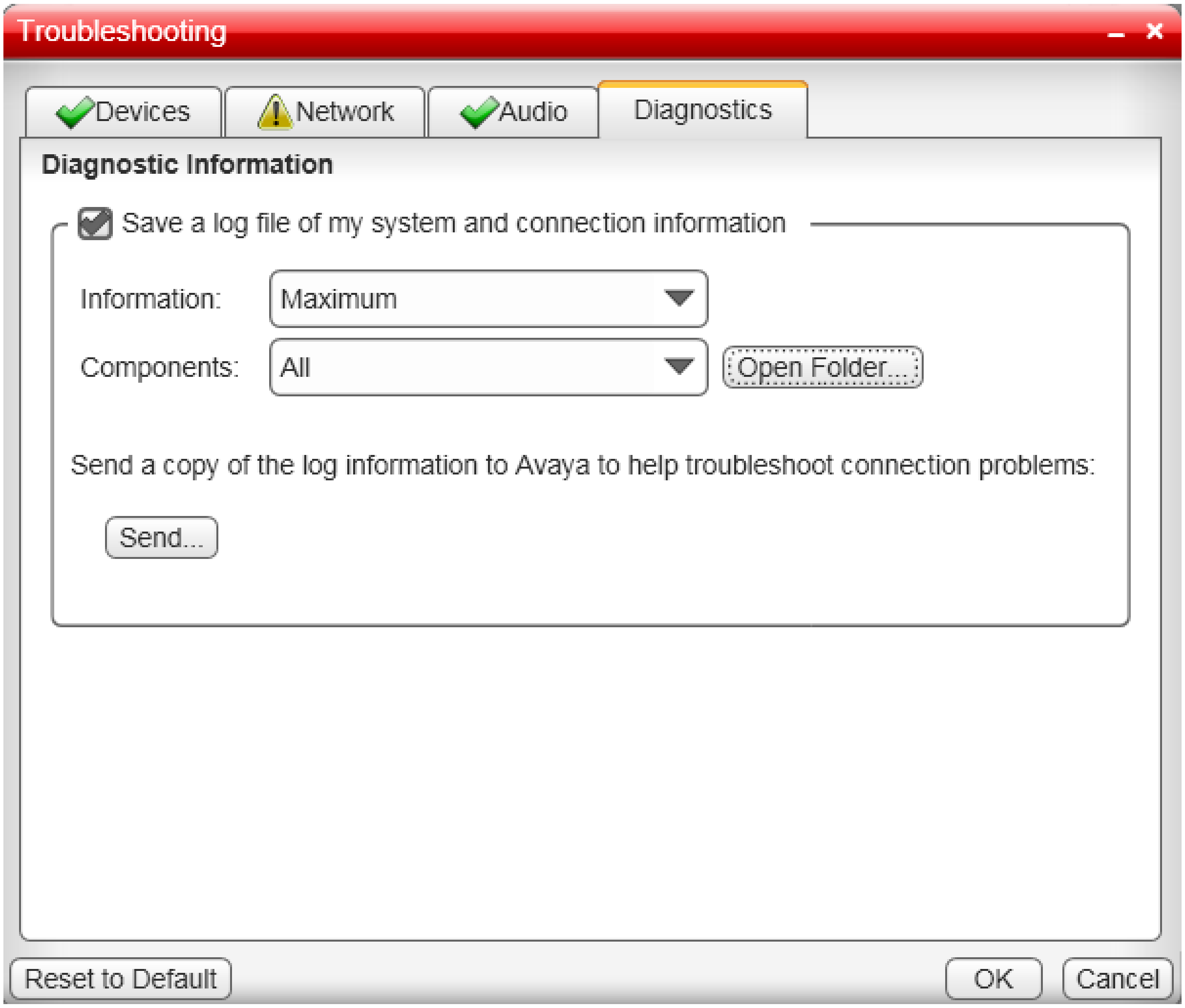
The following screen capture shows a portion of the IP Office Softphone interface as an inbound “private / anonymous” PSTN call is alerting the user. In this case, the same mobile phone (732-687-0755) on the PSTN dialed *67-732-945-0229, and the Incoming Call Route illustrated in Section 3.7 sent the call to user 30026. For this mobile telephone, dialing *67 before the destination number is a request for privacy (i.e., a request to prevent the presentation of the calling party identity to the called party). The IP Softphone user is currently logged in as 30026 and receives the call. Observe the anonymous display.

The following screen capture shows a portion of the IP Office Softphone interface after an inbound PSTN call has been answered. In this case, a PSTN phone (908-848-5704) dialed 732-945-0229, and the Incoming Call Route illustrated in Section 3.7 sent the call to user 30026. The IP Softphone user is currently logged in as 30026 answers the call. The words "AVAYA ALPHA" appearing on the display were included in the From header of the inbound INVITE from Verizon.

The IP Office Softphone includes integrated diagnostics capabilities. For example, the following screen can be viewed by selecting **Help → Troubleshooting** and selecting the **Audio** tab.



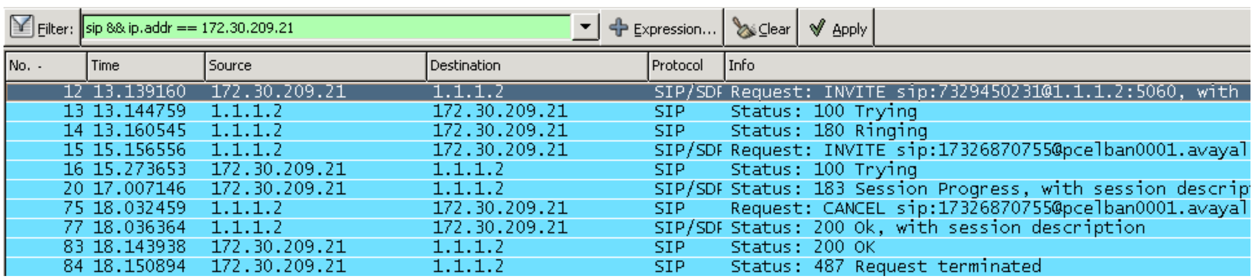
The following screen can be viewed by selecting the **Diagnostics** tab from the Troubleshooting screen shown above. The **Information** and **Components** drop-down menus allow customization of the level and type of troubleshooting information to be gathered. The diagnostic information can be written to a file using the **Save a log file of my system and connection information** check box, and the log files can be accessed via the **Open Folder** button.



5.5. Mobile Twinning Application

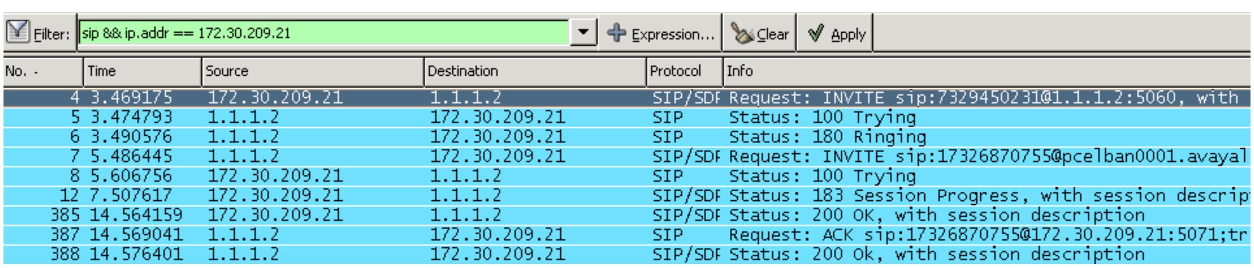
As shown in Section 3.5, user 201 is configured for Mobile Twinning. When an incoming call rings to user 201, the call is "twinned out" to the user's mobile phone, configured as 9-1-732-687-0755. The **Mobile Delay Time** on the Mobility tab for the user was set to 2 seconds. Therefore, the outbound INVITE to Verizon will be sent after 2 seconds of ringing the IP Office user. If the IP Office user answers before the Mobile Delay Time, the call will not be twinned out to Verizon. If the IP Office user answers after the INVITE message has been sent to Verizon, and before the mobile device answers, IP Office will cancel the outbound call.

The following portion of a filtered Wireshark trace illustrates the behavior for an incoming PSTN call from 908-848-5704 to 732-945-0231 (user 201) that is answered by the IP Office user after the INVITE has been sent to Verizon. In frame 12, the INVITE arrives from Verizon for DID 7329450231, whose IP Office Incoming Call Route maps to destination 201. In frame 15 two seconds later, IP Office sends an INVITE to Verizon to the PSTN number configured as the mobile telephone for user 201. In frame 75, IP Office sends a CANCEL because the local IP Office user has answered. In frame 77, the IP Office sends the 200 OK answering the incoming call.



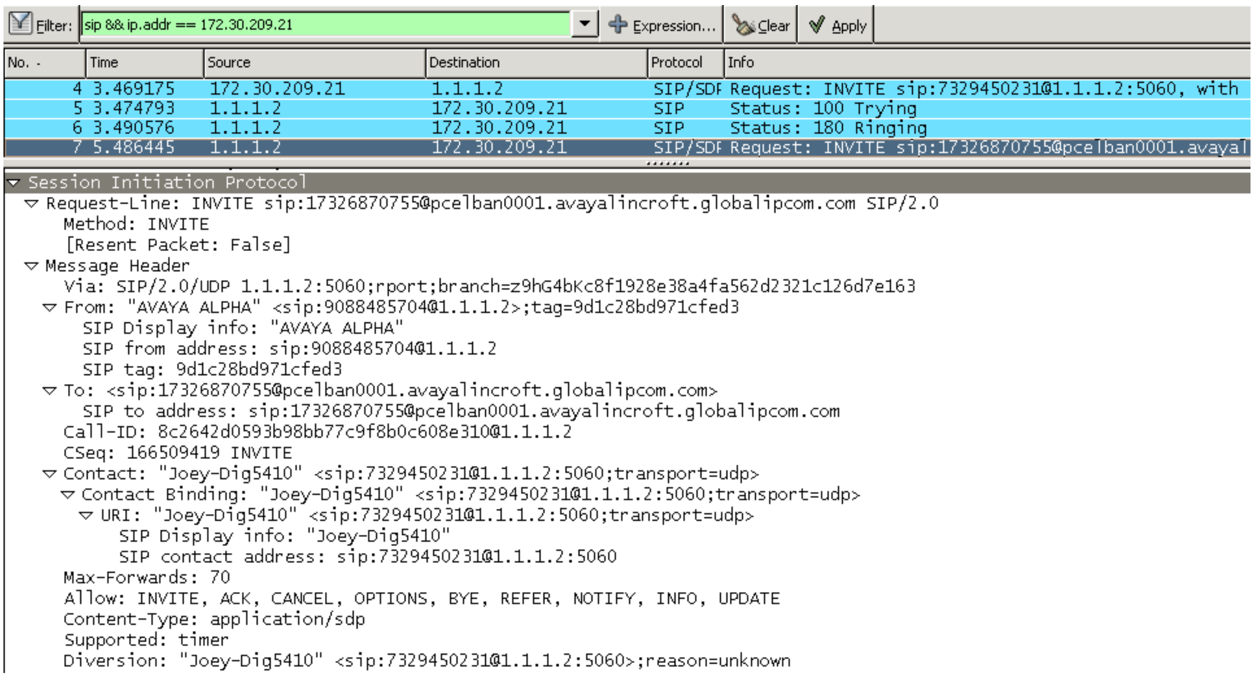
No.	Time	Source	Destination	Protocol	Info
12	13.139160	172.30.209.21	1.1.1.2	SIP/SDF	Request: INVITE sip:7329450231@1.1.1.2:5060, with
13	13.144759	1.1.1.2	172.30.209.21	SIP	Status: 100 Trying
14	13.160545	1.1.1.2	172.30.209.21	SIP	Status: 180 Ringing
15	15.156556	1.1.1.2	172.30.209.21	SIP/SDF	Request: INVITE sip:17326870755@pcelban0001.avayal
16	15.273653	172.30.209.21	1.1.1.2	SIP	Status: 100 Trying
20	17.007146	172.30.209.21	1.1.1.2	SIP/SDF	Status: 183 Session Progress, with session descrip
75	18.032459	1.1.1.2	172.30.209.21	SIP	Request: CANCEL sip:17326870755@pcelban0001.avayal
77	18.036364	1.1.1.2	172.30.209.21	SIP/SDF	Status: 200 Ok, with session description
83	18.143938	172.30.209.21	1.1.1.2	SIP	Status: 200 OK
84	18.150894	172.30.209.21	1.1.1.2	SIP	Status: 487 Request terminated

The following portion of a filtered Wireshark trace illustrates the behavior for an incoming PSTN call from 908-848-5704 to user 201 that is answered by the mobile telephone. In frame 4, the INVITE arrives from Verizon for DID 7329450231. In frame 7 two seconds later, IP Office sends an INVITE to Verizon to the PSTN number configured as the mobile telephone for user 201. In frame 385, Verizon sends a 200 OK when the mobile telephone user answers the call. In frame 388, IP Office sends the 200 OK answering the incoming call.



No.	Time	Source	Destination	Protocol	Info
4	3.469175	172.30.209.21	1.1.1.2	SIP/SDF	Request: INVITE sip:7329450231@1.1.1.2:5060, with
5	3.474793	1.1.1.2	172.30.209.21	SIP	Status: 100 Trying
6	3.490576	1.1.1.2	172.30.209.21	SIP	Status: 180 Ringing
7	5.486445	1.1.1.2	172.30.209.21	SIP/SDF	Request: INVITE sip:17326870755@pcelban0001.avayal
8	5.606756	172.30.209.21	1.1.1.2	SIP	Status: 100 Trying
12	7.507617	172.30.209.21	1.1.1.2	SIP/SDF	Status: 183 Session Progress, with session descrip
385	14.564159	172.30.209.21	1.1.1.2	SIP/SDF	Status: 200 OK, with session description
387	14.569041	1.1.1.2	172.30.209.21	SIP	Request: ACK sip:17326870755@172.30.209.21:5071;tr
388	14.576401	1.1.1.2	172.30.209.21	SIP/SDF	Status: 200 Ok, with session description

This same trace can be further analyzed to reveal the Diversion information included in the outbound INVITE to Verizon for the twinned outbound call leg. The following screen shows frame 7 expanded to show the SIP information. Note that the From header contains the original calling party information, allowing the mobile telephone receiving the twinned call to see the true caller's identity. As can be observed in the last line below, the Diversion header inserted by IP Office includes the DID associated with the IP Office twinning user, 7329450231. This allows Verizon to admit the outbound twinned call to the network.



5.6. Outbound Anonymous / Private Calls

As shown in Section 3.6, a user may dial using the short code 8 to request privacy. In the filtered Wireshark trace shown below, user 201 dialed 8-1-908-848-5704. The INVITE sent by IP Office in frame 7 is expanded. Observe that IP Office includes “Privacy: Id”, with the P-Asserted-Identity (PAI) Header containing the DID of the user. Verizon completes the call to the called PSTN user, and the called telephone does not see the caller id information. Section 3.4 and Section 3.9 show the configuration required to cause IP Office Release 6 to include the PAI Header, as required by Verizon for an anonymous call.

Filter: sip && ip.addr == 172.30.209.21 | Expression... | Clear | Apply

No.	Time	Source	Destination	Protocol	Info
7	8.399423	1.1.1.2	172.30.209.21	SIP/SDF	Request: INVITE sip:19088485704@172.30.209.21, wit
8	8.515158	172.30.209.21	1.1.1.2	SIP	Status: 100 Trying
16	10.031815	172.30.209.21	1.1.1.2	SIP/SDF	Status: 183 Session Progress, with session descrip
221	12.135034	172.30.209.21	1.1.1.2	SIP/SDF	Status: 200 OK, with session description
223	12.140778	1.1.1.2	172.30.209.21	SIP	Request: ACK sip:19088485704@172.30.209.21:5071;tr

```
Session Initiation Protocol
  Request-Line: INVITE sip:19088485704@172.30.209.21 SIP/2.0
    Method: INVITE
    [Resent Packet: False]
  Message Header
    Via: SIP/2.0/UDP 1.1.1.2:5060;rport;branch=z9hG4bKd37695b09bb413cb70d1e1e7cc84d82c
    From: "Anonymous" <sip:restricted@1.1.1.2>;tag=63c4ae39aa68e1ab
      SIP Display info: "Anonymous"
      SIP from address: sip:restricted@1.1.1.2
      SIP tag: 63c4ae39aa68e1ab
    To: <sip:19088485704@172.30.209.21>
      SIP to address: sip:19088485704@172.30.209.21
    Call-ID: d1f43241f6a19f4523ae2097bb7307ac@1.1.1.2
    CSeq: 1299433042 INVITE
    Contact: <sip:anonymous@1.1.1.2:5060;transport=udp>
      Contact Binding: <sip:anonymous@1.1.1.2:5060;transport=udp>
        URI: <sip:anonymous@1.1.1.2:5060;transport=udp>
          SIP contact address: sip:anonymous@1.1.1.2:5060
    Max-Forwards: 70
    Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
    Content-Type: application/sdp
    Supported: timer
    Privacy: id
    P-Asserted-Identity: "Unavailable" <sip:7329450231@1.1.1.2:5060>
```

5.7. SIP OPTIONS

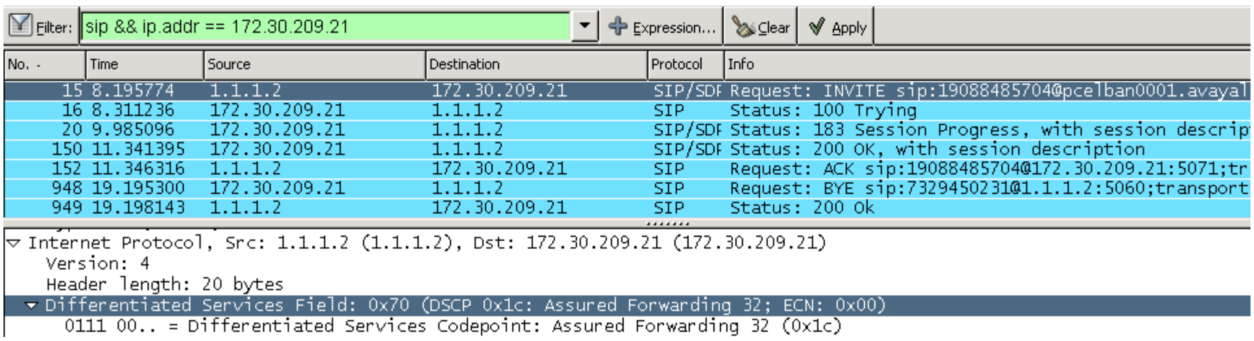
IP Office can be configured to maintain a SIP Line using SIP OPTIONS. Sections 3.3, 3.4, and 3.10 illustrate the pertinent configuration. In the filtered Wireshark trace shown below, SIP OPTIONS and 200 OK responses can be observed. In frame 37, IP Office sends OPTIONS to Verizon, and Verizon responds with 200 OK in frame 40. Coincidentally, Verizon sent SIP OPTIONS to IP Office at about the same time, with the OPTIONS from Verizon shown in frame 38 and the 200 OK from IP Office in frame 39. Frame 150 shows another OPTIONS sent by IP Office, 60 seconds after the OPTIONS in frame 37. (At the time of this trace, the Binding Refresh Time for LAN1 was set to 60 seconds.)

Filter: sip && ip.addr == 172.30.209.21 | Expression... | Clear | Apply

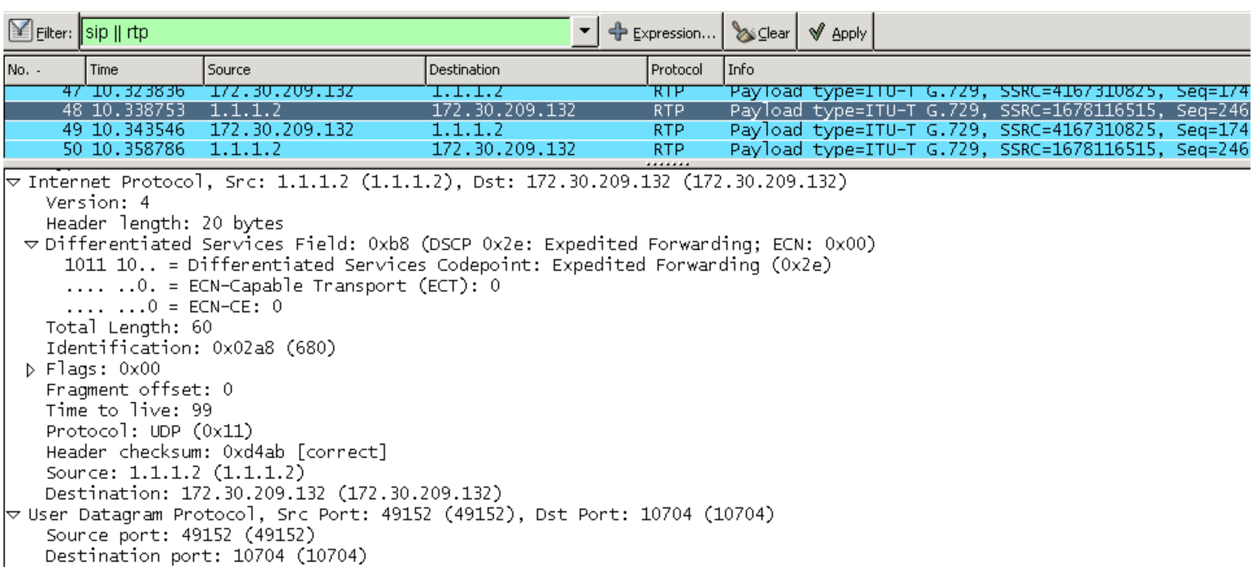
No.	Time	Source	Destination	Protocol	Info
37	10.761024	1.1.1.2	172.30.209.21	SIP	Request: OPTIONS sip:Unknown@172.30.209.21
38	10.871717	172.30.209.21	1.1.1.2	SIP	Request: OPTIONS sip:1.1.1.2:5060
39	10.876520	1.1.1.2	172.30.209.21	SIP/SDF	Status: 200 Ok, with session description
40	10.879783	172.30.209.21	1.1.1.2	SIP	Status: 200 OK
150	70.872142	1.1.1.2	172.30.209.21	SIP	Request: OPTIONS sip:Unknown@172.30.209.21
151	70.988908	172.30.209.21	1.1.1.2	SIP	Status: 200 OK

5.8. DiffServ Markings by IP Office

As shown in Section 3.3, IP Office can mark the Differentiated Services Field in the IP Header to configurable values for SIP signaling and RTP. For example, the following portion of a filtered Wireshark trace shows the SIP INVITE message sent by IP Office. As can be observed, the Differentiated Services Field in the IP Header is decoded as "Assured Forwarding 32", as configured in Section 3.3.2 for SIP signaling. Other values may be used.



The following portion of a filtered Wireshark trace shows an example RTP packet sent by IP Office. As can be observed, the Differentiated Services Field in the IP Header is decoded as "Expedited Forwarding", as configured in Section 3.3.2 for RTP. Other values may be used. It can also be observed that the IP Office UDP port for RTP (49152) is within the range configured in Section 3.3.2. Note that the IP Office IP Address 1.1.1.2 is always the source for the RTP from IP Office towards Verizon, regardless of whether the telephone device at the customer location is analog, digital, H.323, or SIP.



6. Support

6.1. Avaya

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

6.2. Verizon

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

7. 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 can be successfully combined with a Verizon Business IP Trunk SIP trunk service connection to create an end-to-end SIP Telephony business solution. Through following the example configurations provided in this document, customers using Avaya IP Office can now connect to the PSTN via a Verizon Business IP Trunk SIP Trunk service connection, thus eliminating the costs of analog or digital trunk connections previously required to access the PSTN. Utilizing this solution, IP Office customers can now 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.

8. References

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

[IPO-INSTALL] IP Office 6.0 Installation Manual, Issue 21f, March 1 2010
Document Number 15-601042
http://support.avaya.com/css/P8/documents/100073460

[IPO-MGR] IP Office Release 6.0 Manager 8.0, Issue 24h, February 20, 2010
Document Number 15-601011
http://support.avaya.com/elmodocs2/ip_office/R4.2/Newissuesept08/eng/manager_en.pdf

[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 6.0 Voicemail Pro, Issue 22b, January 16, 2010
Document Number 15-601063

http://support.avaya.com/css/P8/documents/100073435

[IPO-MON] IP Office System Monitor, Issue 02b, November 28, 2008
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/

[CM-VZIPT] Application Notes for Avaya Aura™ Communication Manager 5.2, Avaya Aura™ Session Manager 1.1, and Acme Packet 3800 Net-Net Session Director integration with Verizon Business IP Trunk SIP trunk service offer – Issue 1.3
https://devconnect.avaya.com/public/download/dyn/AvayaSM_VzB_IPT.pdf

[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

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.