AVAYA

Avaya Solution & Interoperability Test Lab

Application Notes for SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service and Avaya IP Office Release 6.1, Using REFER and DNS SRV – 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 6.1 Preferred Edition, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, digital, and analog endpoints.

These Application Notes complement previously published Application Notes by illustrating the configuration and verification of new capabilities of IP Office Release 6.1, including transfer using SIP REFER to Verizon, and use of DNS SRV to determine the Verizon Business SIP signaling information from a Verizon DNS Server. **Although these new IP Office Release 6.1 capabilities have not been independently certified by Verizon labs, these Application Notes can be used to facilitate customer engagements via the Verizon field trial process for either or both new features.**

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.

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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 6.1 Preferred Edition, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, digital, and analog endpoints.

These Application Notes are based on the previously published Application Notes in reference [JRR-IPOR6] which covered IP Office Release 6. These Application Notes complement previously published Application Notes by illustrating the configuration and verification of new capabilities of IP Office Release 6.1, including transfer using SIP REFER to Verizon, and use of DNS SRV to determine the Verizon Business SIP signaling information from a Verizon DNS Server. Although these new IP Office Release 6.1 capabilities have not been independently certified by Verizon labs, these Application Notes can be used to facilitate customer engagements via the Verizon field trial process for either or both new features. REFER-based transfer can allow the trunks to the IP Office enterprise location to be released when a call is transferred, resulting in a connection that no longer uses IP Office resources. The use of DNS SRV obviates the need to statically configure the Verizon SIP Signaling IP Address and port information in IP Office, thus allowing dynamic, automatic updates to SIP signaling to occur without manual intervention by the IP Office administrator, if Verizon network conditions change.

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/

1.1. Interoperability Compliance Testing

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.

This document supplements reference [JRR-IPOR6], which included detailed illustrations of the verification of representative calls using the IP Office System Status application, the IP Office System Monitor application, and Wireshark.

The focus of the verification testing associated with this version of the Application Notes was the use of SIP REFER messaging for call transfer scenarios, and the use of DNS SRV rather than static provisioning of the Verizon SIP signaling information. Note that the REFER testing and the DNS SRV testing are not dependent on one another. That is, IP Office can use REFER for call transfer without using DNS SRV, and IP office can use DNS SRV without using REFER for call transfer. These Application Notes are intended to facilitate Verizon field trial testing of either or both capabilities.

1.2. Support

1.2.1. Avaya

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

1.2.2. Verizon

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

1.3. Known Limitations

Interoperability testing of the sample configuration was completed with successful results, with the successful verifications detailed in Section 7. 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 on the call.

3. 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 4.4.4). 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.

2. Reference Configuration

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 configuration is identical to the configuration used in reference [JRR-IPOR6].

In the sample configuration, IP Office receives traffic from the Verizon Business IP Trunk service on port 5060. IP Office uses DNS SRV, using UDP for transport, to determine the IP Address and port to be used to send SIP signaling to Verizon. In the sample configuration, the DNS process will result in SIP signaling being sent to IP Address 172.30.209.21 and port 5071, but these values are not statically configured in IP Office. 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 in the IP Office configuration.

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

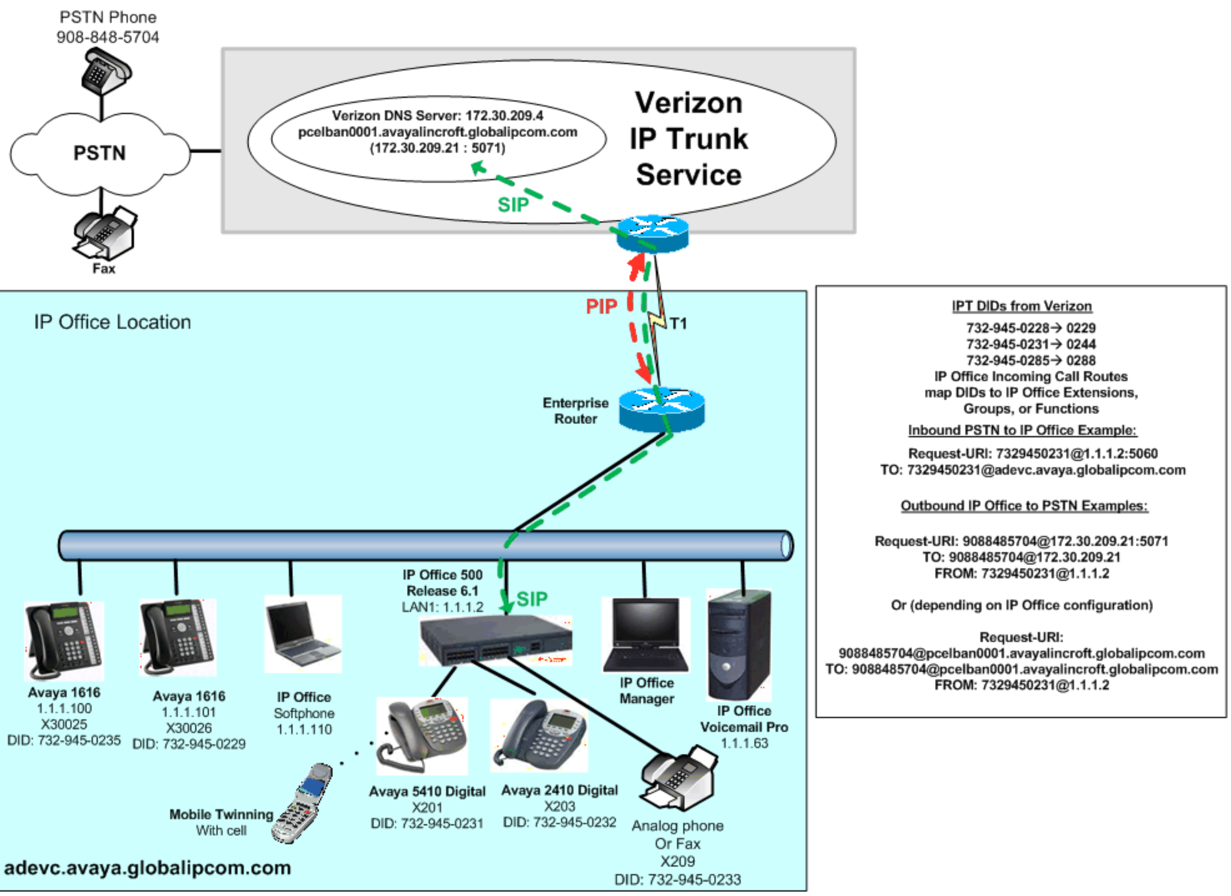


Figure 1: Avaya IP Office with Verizon IP Trunk SIP Trunk Service

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 4. Selected verifications are illustrated in Section 7.

Verizon Provided DID	Avaya IP Office Destination	Notes
732-945-0228	Auto-Attendant on Voicemail Pro	See Section 4.6
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 1.3
732-945-0234	Voicemail Collect on Voicemail Pro	See Section 4.6, 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

3. Equipment and Software Validated

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

Equipment	Software
Avaya IP Office 500	Release 6.1 (6.1.5) (Preferred Edition)
Avaya IP Office Manager	Release 8.1 (8.1.5) (Preferred Edition)
Avaya IP Office Voicemail Pro	Release 6.1.16
Avaya IP Office Voicemail Pro Client	Version 6.1 (16)
Avaya 1600-Series Telephones (H.323)	Release 1.3
Avaya 2400-Series and 5400-Series Digital Telephones	REL: 6.00 (downloaded from IP Office)
Avaya IP Office Softphone	Release 3.1.2.17 59616
Brother Intellifax 1360 (analog fax)	N/A

Table 2: Equipment and Software Tested

4. Avaya IP Office Configuration

The Avaya IP Office configuration shown in this section is effectively the same as the configuration shown in reference [JRR-IPOR61]. The only substantive configuration screen differences are shown in Section 4.4 for the SIP Line.

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 may be displayed:

WELCOME to IP Office Administration

What would you like to do ?

Create an Offline Configuration

Open Configuration from System

Read a Configuration from File

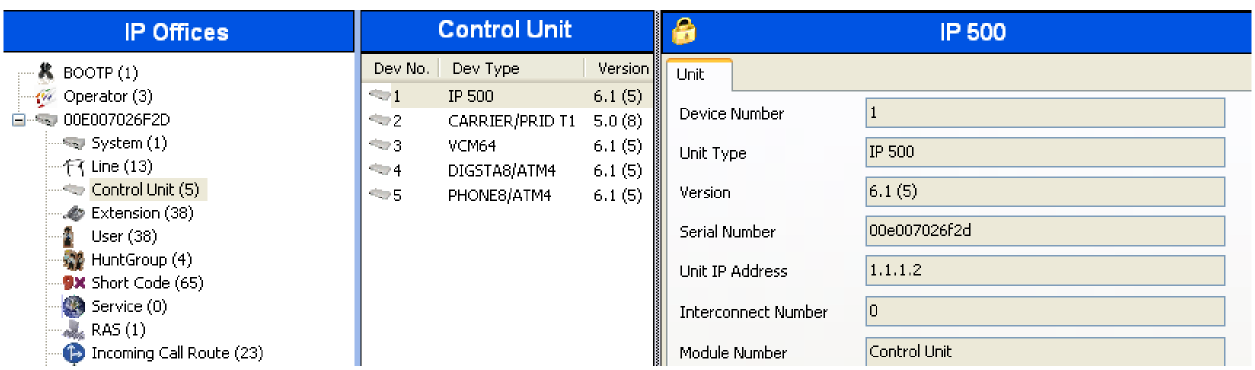
Open the IP Office configuration, either by reading the configuration from the IP Office server, or from file. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left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

4.1. Physical, Network, and Security Configuration

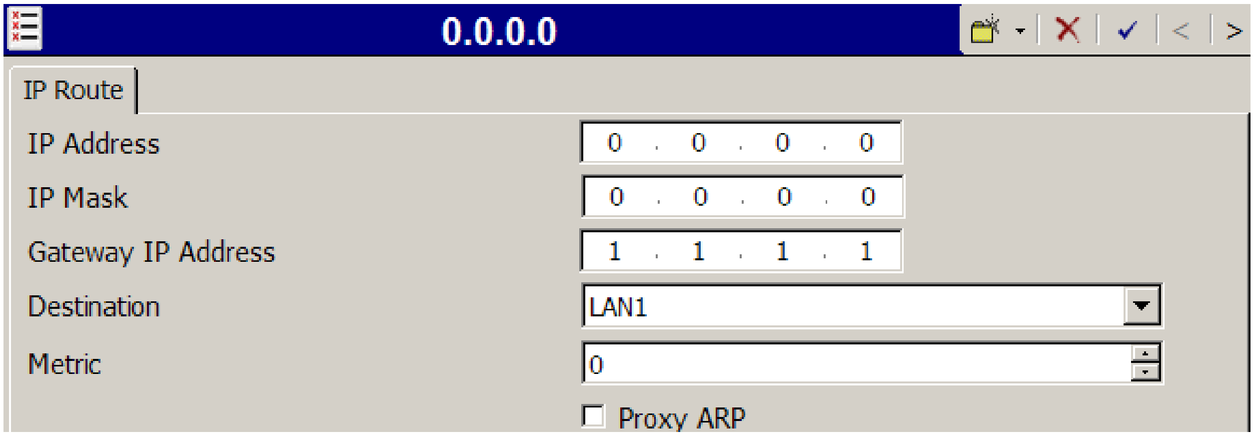
This section describes attributes of the sample configuration, but is not meant to be prescriptive. Consult reference [IPO-INSTALL] for more information on the topics in this section.

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.

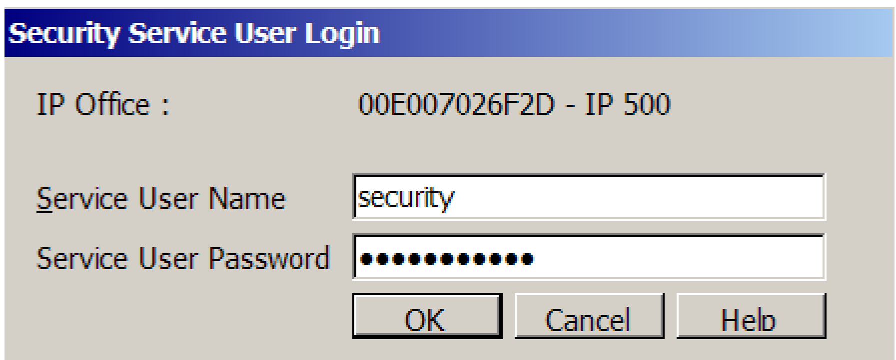
The following screen shows the modules in the IP Office used in the sample configuration. To access such a screen, select **Control Unit** in the Navigation pane. The modules appear in the Group pane. In the screen below, **IP 500** is selected in the Group pane, revealing additional information about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



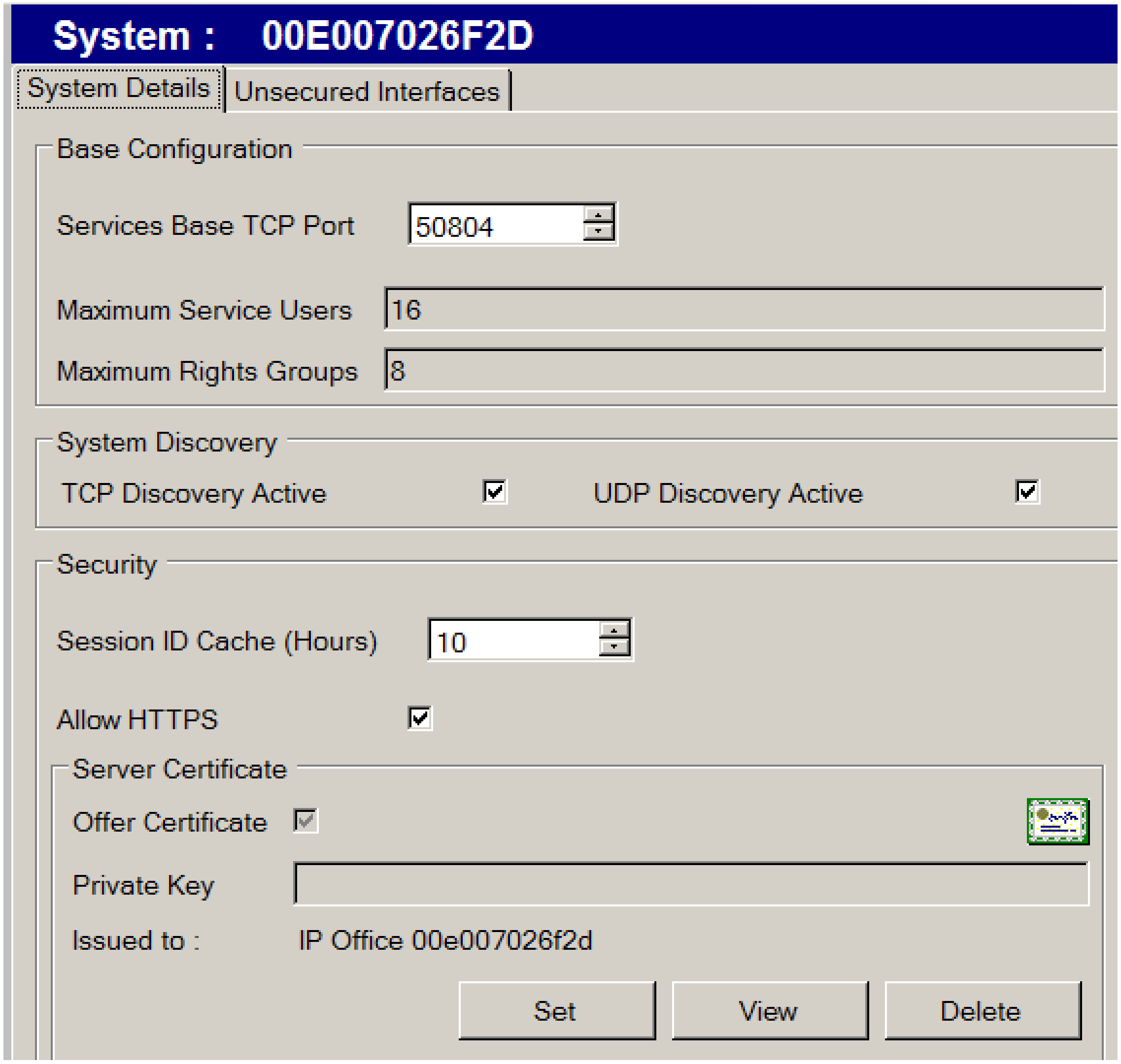
In the sample configuration, the IP Office LAN1 port is physically connected to the local area network switch at the IP Office customer site. The default gateway for this network is 1.1.1.1. To add an IP Route in IP Office, right-click **IP Route** from the Navigation pane, and select **New.** To view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

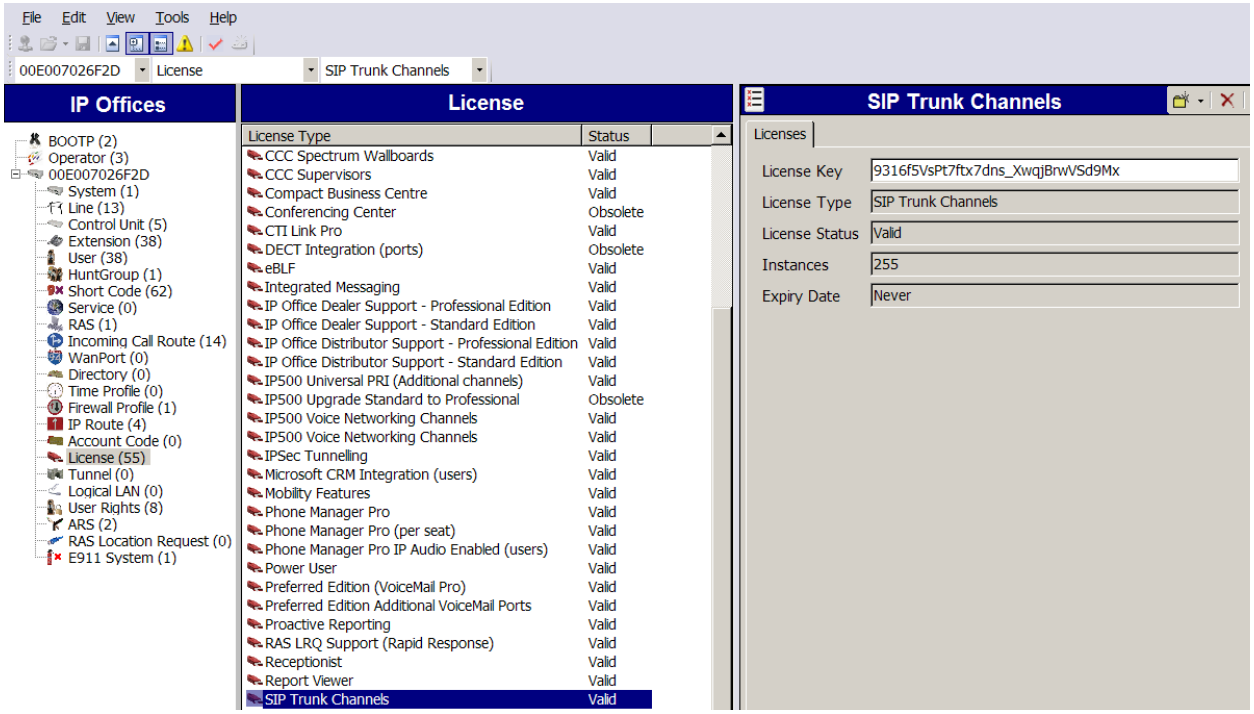


When complete, select **File → Configuration** to return to configuration activities.

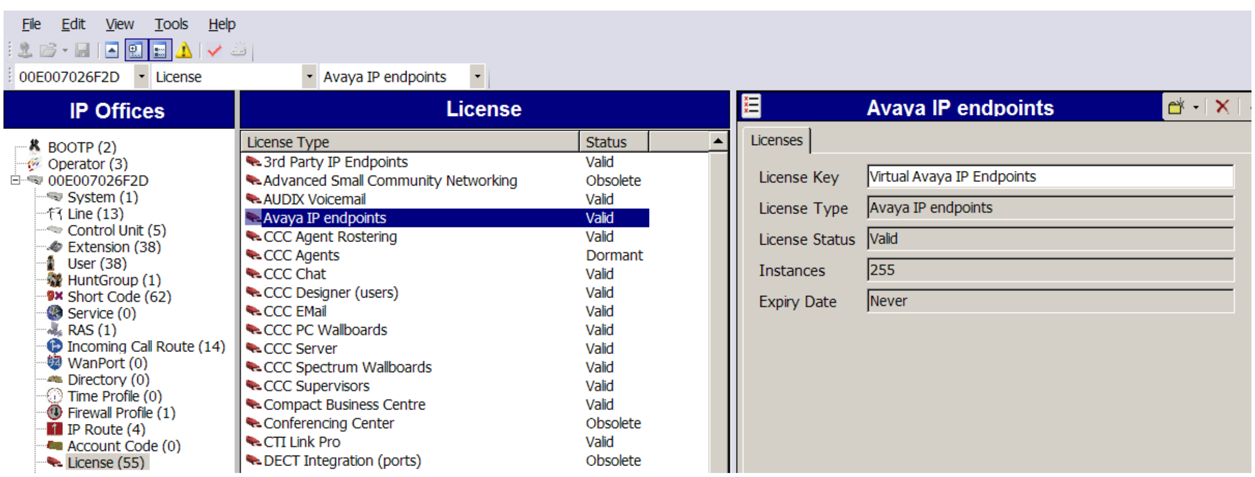
4.2. Licensing

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.

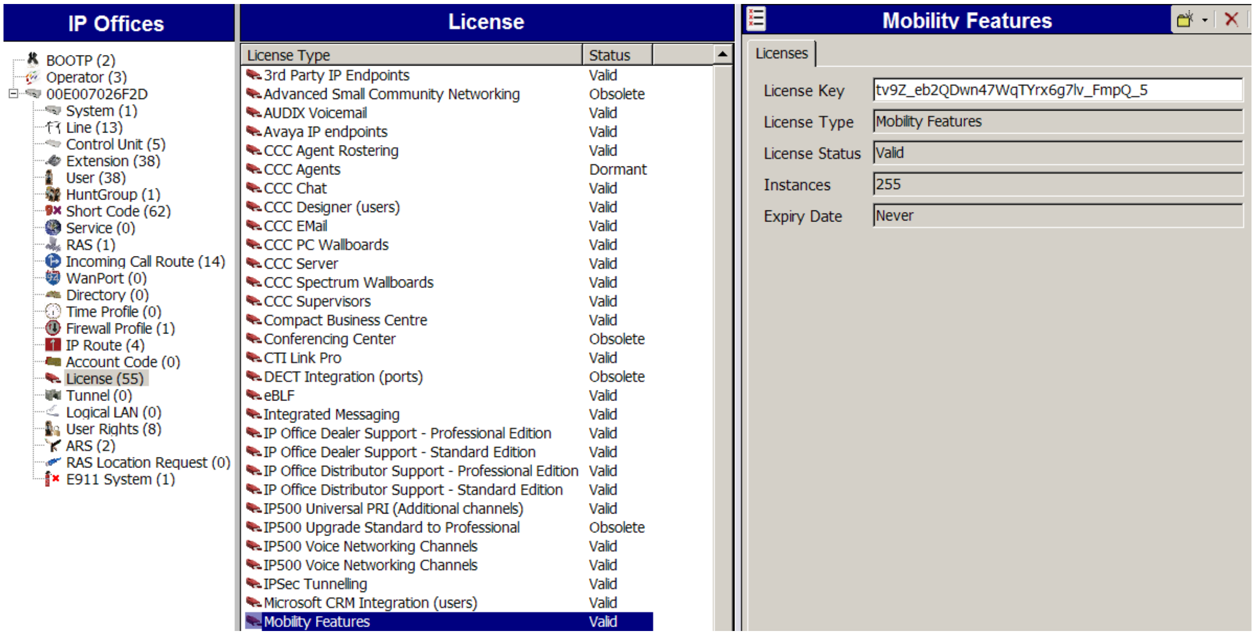
To verify that there is a SIP Trunk Channels License with sufficient capacity, click **License** in the Navigation pane and **SIP Trunk Channels** in the Group pane. Confirm a valid license with sufficient "Instances" (trunk channels) in the Details pane.



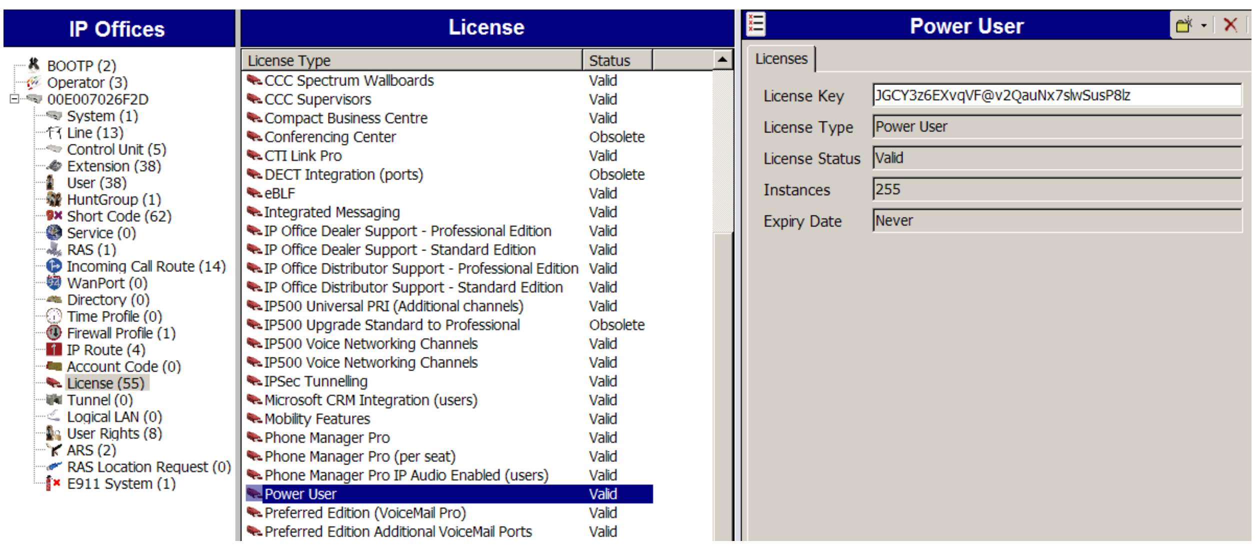
If Avaya IP Telephones will be used, verify the Avaya IP endpoints license. Click **License** in the Navigation pane and **Avaya IP endpoints** in the Group pane. Confirm a valid license with sufficient "Instances" in the Details pane.



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.

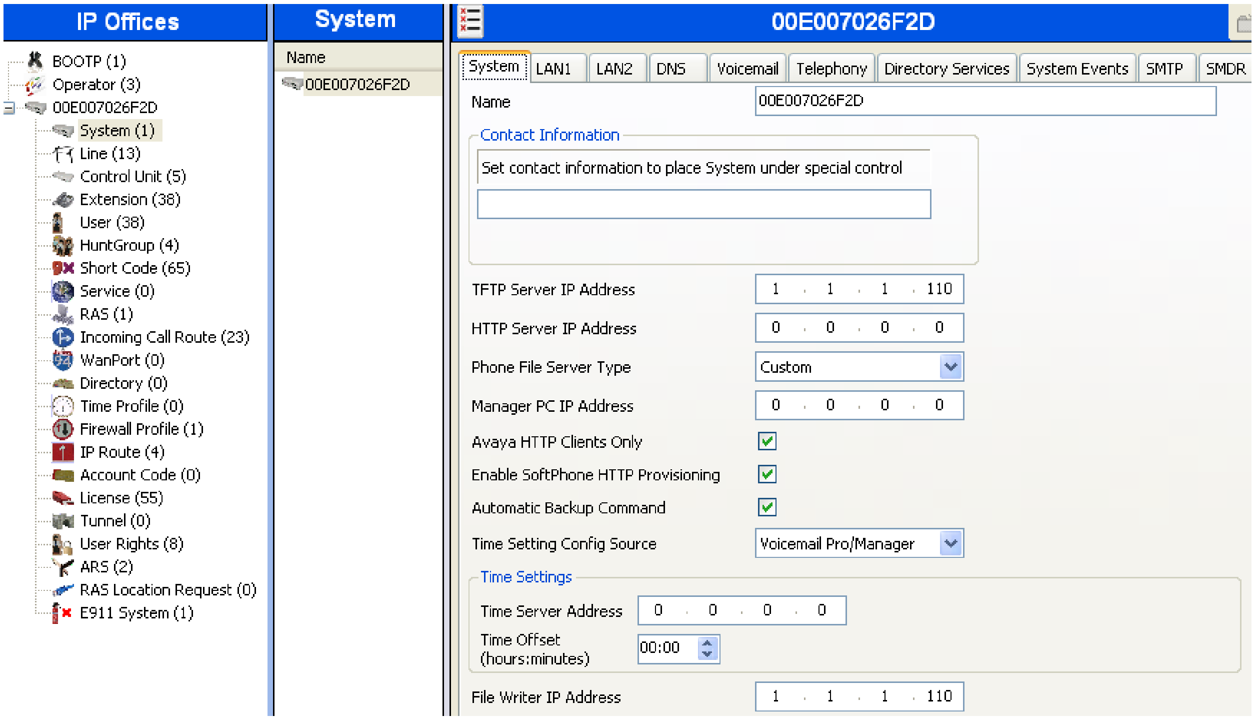


4.3. System Settings

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.

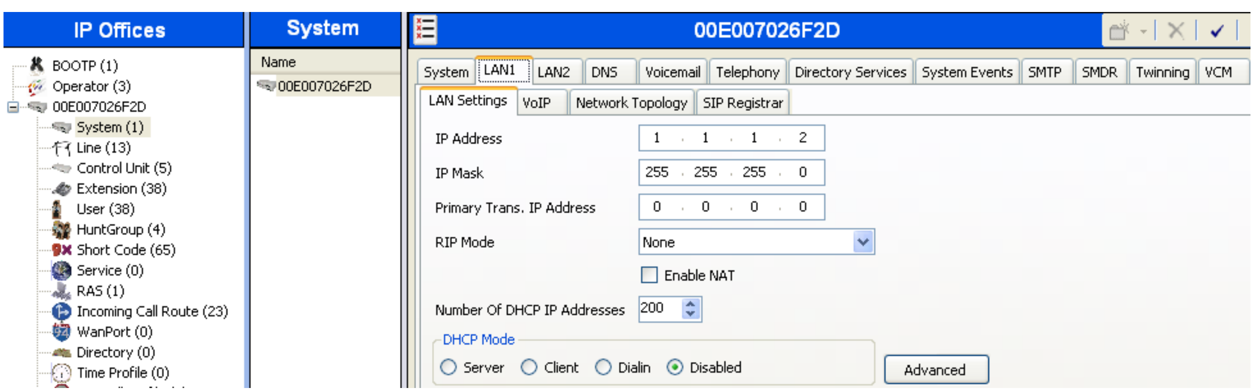
4.3.1. System Tab

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.



4.3.2. LAN Settings

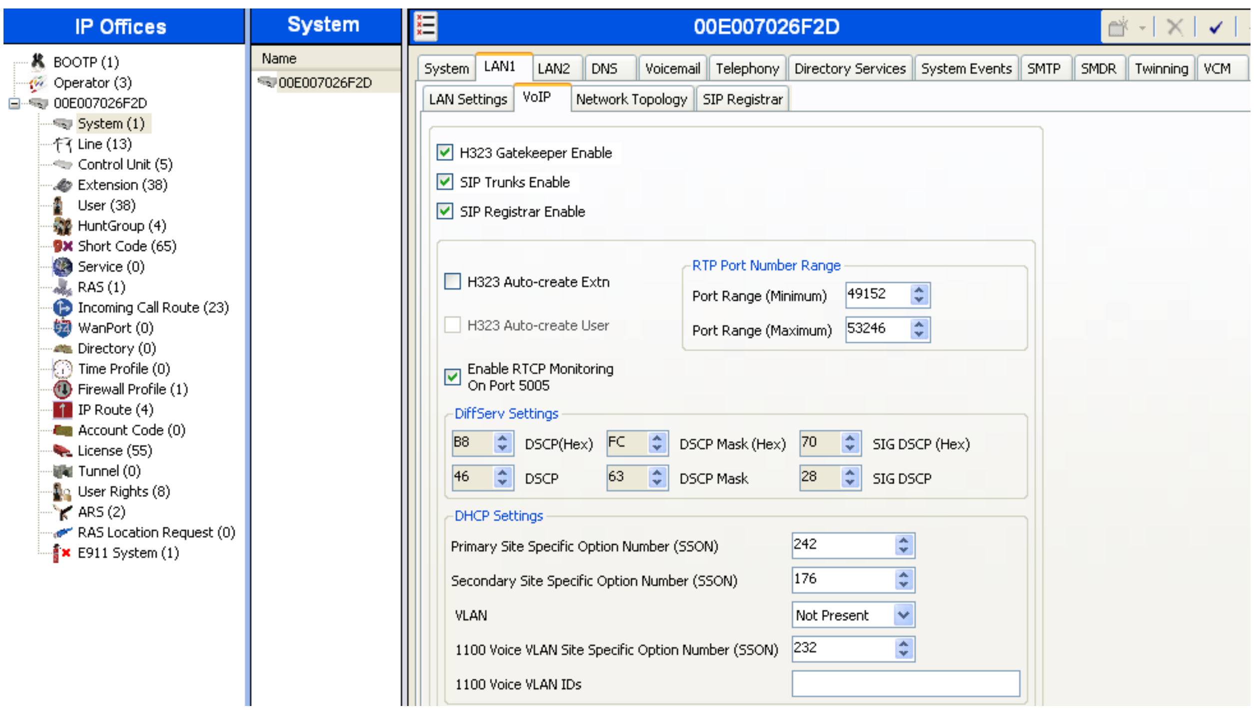
In the sample configuration, LAN1 was used to connect the IP Office to the enterprise network. Other LAN choices (e.g., LAN2) may also be used. To view or configure the **IP Address** of LAN1, select the **LAN1** tab followed by the **LAN Settings** tab. As shown in **Figure 1**, the IP Address of the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



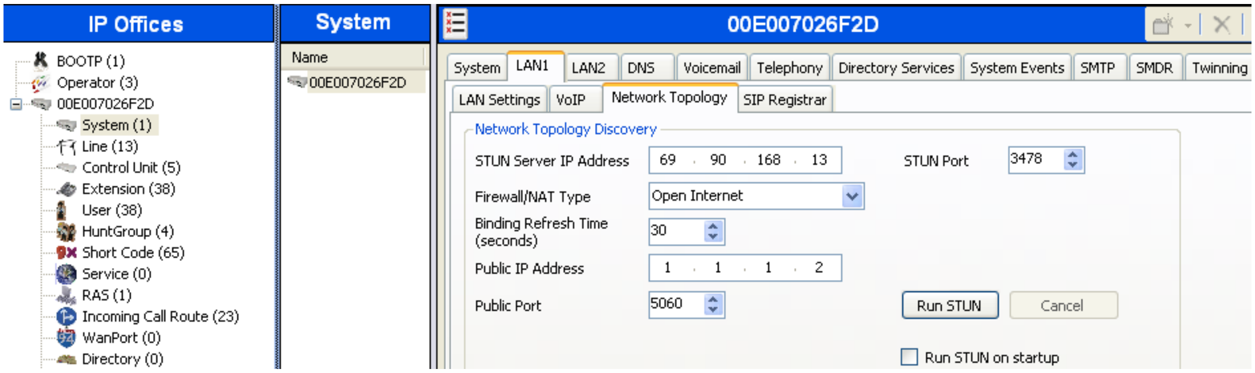
Select the **VoIP** tab as shown in the following screen. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Verizon Business. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 1600-Series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

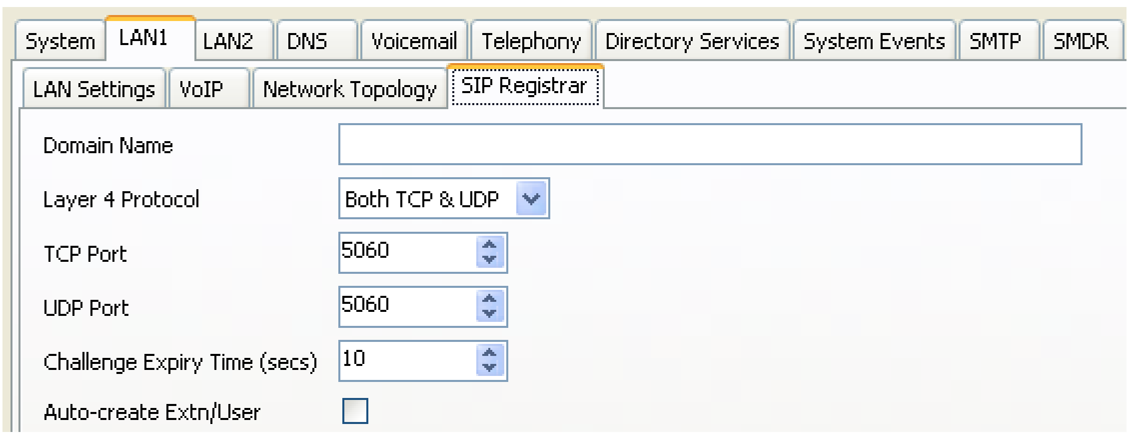


Select the **Network Topology** tab as shown in the following screen. For **Public IP Address**, enter the Avaya IP Office LAN1 IP address. Set the **Public Port** to 5060. In the sample configuration, the **Firewall/NAT Type** is set to "Open Internet". With this configuration, STUN will not be used.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



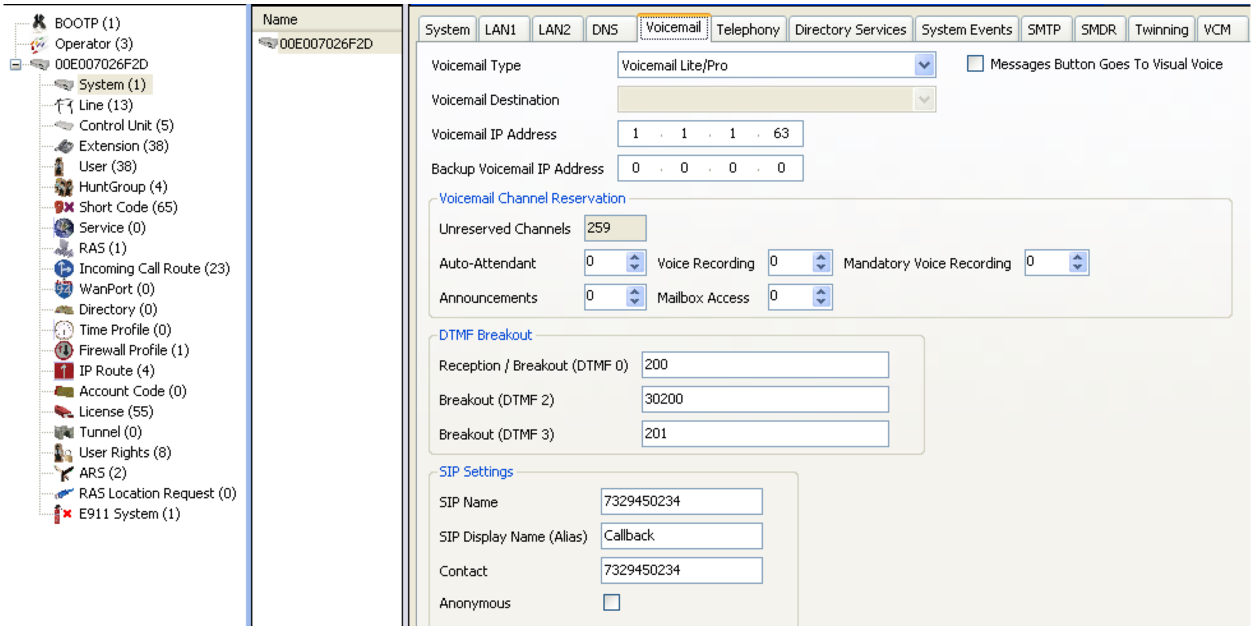
Note: The **Firewall/NAT Type** parameter may need to be different, depending on the type of firewall or Network Address Translation device used at the customer premise.

Optionally, select the **SIP Registrar** tab. The following screen shows the settings used in the sample configuration.



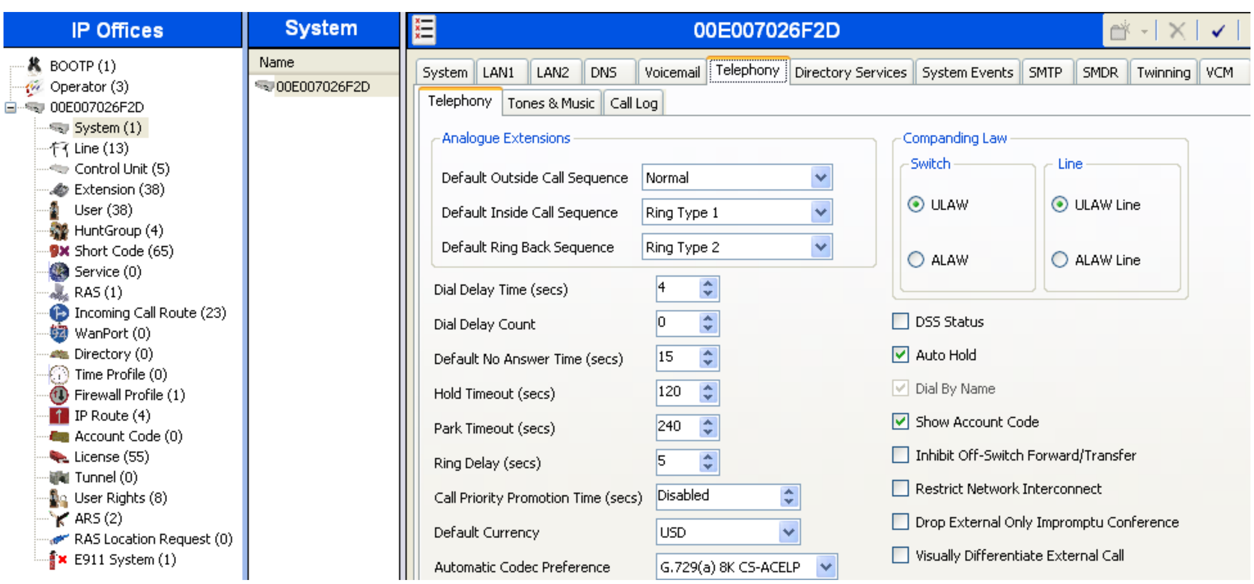
4.3.3. Voicemail

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 4.5). Other parameters on this screen may be set according to customer requirements.



4.3.4. System Telephony Configuration

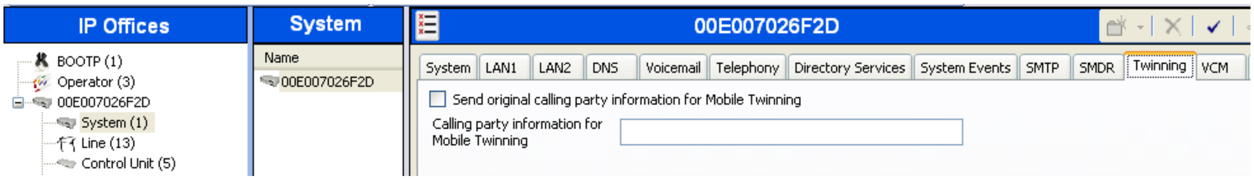
To view or change telephony settings, select the **Telephony** tab and **Telephony** sub-tab as shown in the following screen. The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. In the sample configuration, the **Inhibit Off-Switch Forward/Transfer**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



4.3.5. 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 4.4), the true identity of a PSTN 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.



4.4. SIP Line

This section shows the configuration screens for the SIP Line in IP Office Release 6.1. Since IP Office Release 6.1 introduced new SIP Line parameters and re-oriented existing parameters, this section has the most substantive changes in these Application Notes, compared to the configuration documented in references [JRR-IPOR6] and [JRR-IPOR61].

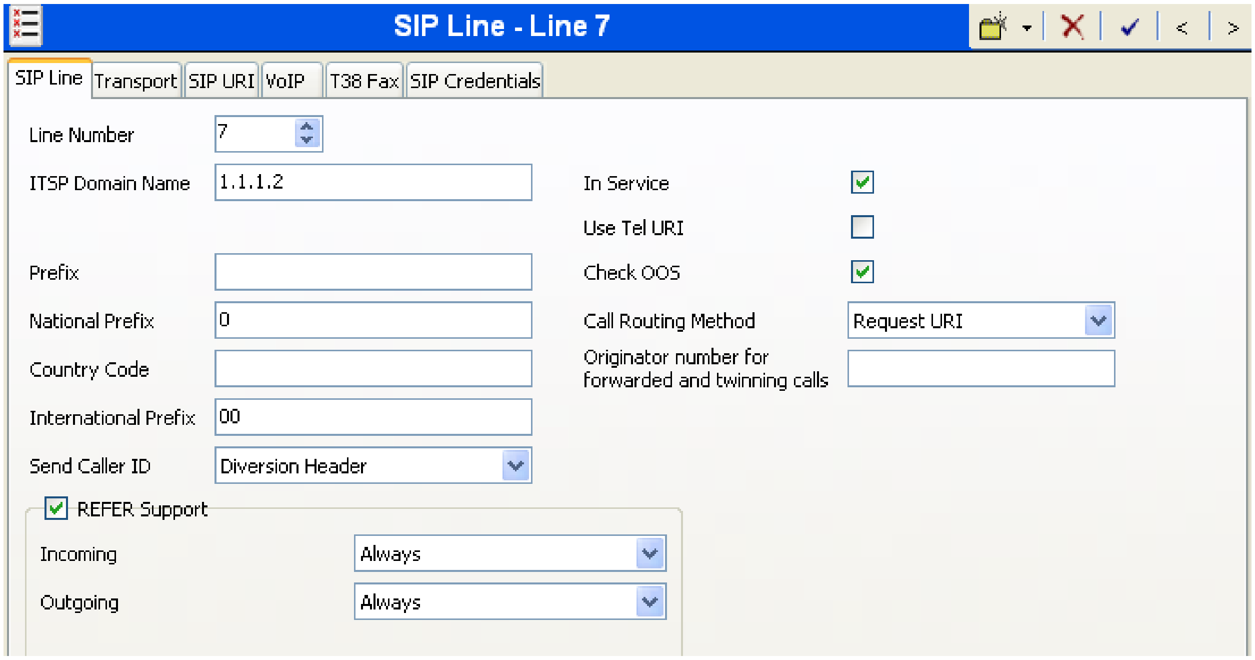
To add a new SIP Line, right click on **Line** in the Navigation pane, and select **New → SIP Line**. A new Line Number will be assigned automatically. To edit an existing SIP Line, click **Line** in the Navigation pane, and the SIP Line to be configured in the Group pane.

4.4.1. 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 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 and Diversion 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 4.3.2. See Section 4.10 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 4.3.5, 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", to match Incoming Call Routes based on the contents of the "To Header". Click **OK** (not shown).

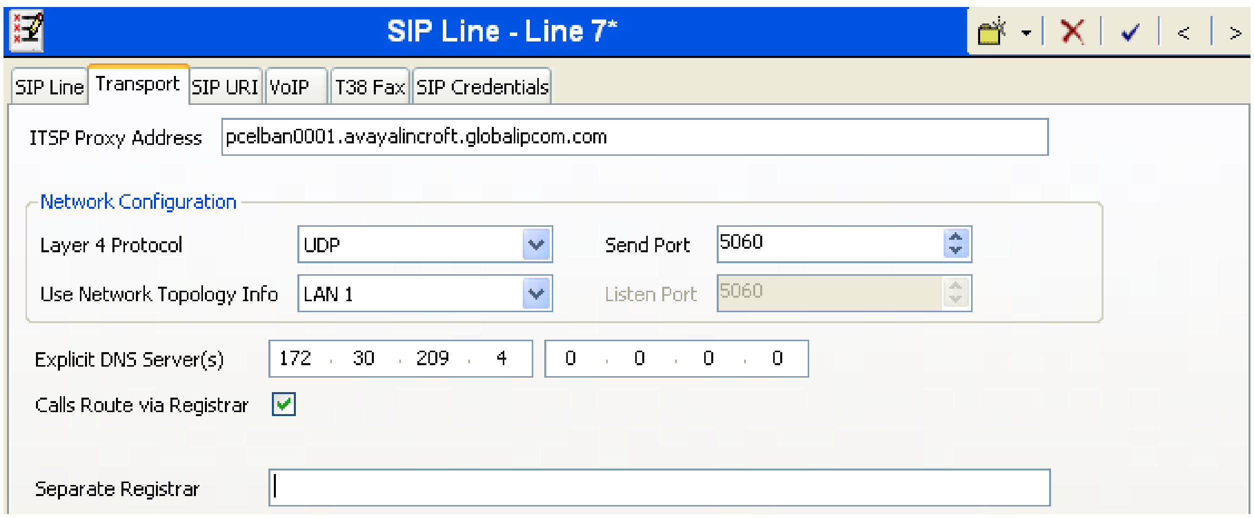
The area of the screen entitled **REFER Support** is new for IP Office Release 6.1. In the following screen, “Always” has been selected from the drop-down menu for the **Incoming** and **Outgoing** parameters, to enable use of SIP REFER in the sample configuration.



4.4.2. SIP Line - Transport Tab

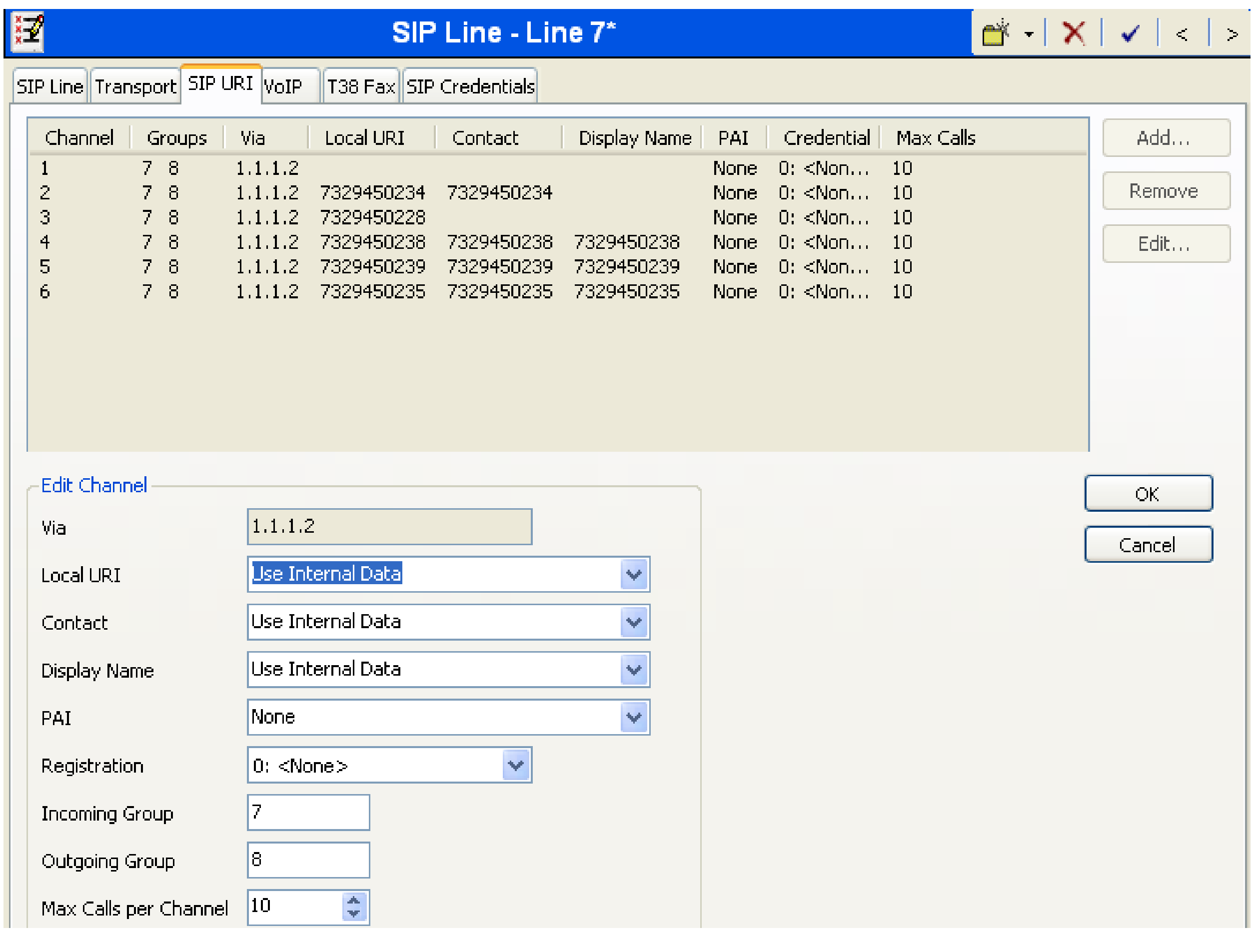
Select the **Transport** tab. This tab is new in Release 6.1. Some information configured in this tab had been under the **SIP Line** tab in Release 6.0.

The **ITSP Proxy Address** is set to the Verizon domain provided by Verizon Business. As shown in **Figure 1**, this domain is “pcelban0001.avayalincroft.globalipcom.com”. Optionally, the domain configured in the ITSP Proxy Address field can be suffixed with a number. For example, “pcelban0001.avayalincroft.globalipcom.com(4)” may be entered. In the **Network Configuration** area, UDP is selected as the **Layer 4 Protocol**. Since DNS SRV will be used, the **Send Port** can retain the default value 5060. The port to which IP Office sends SIP messages will be determined via the DNS procedures. 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. The **Explicit DNS Server(s)** is configured with the DNS Server IP address provided by Verizon Business, which is 172.30.209.4 in the sample configuration.



4.4.3. SIP Line - SIP URI Tab

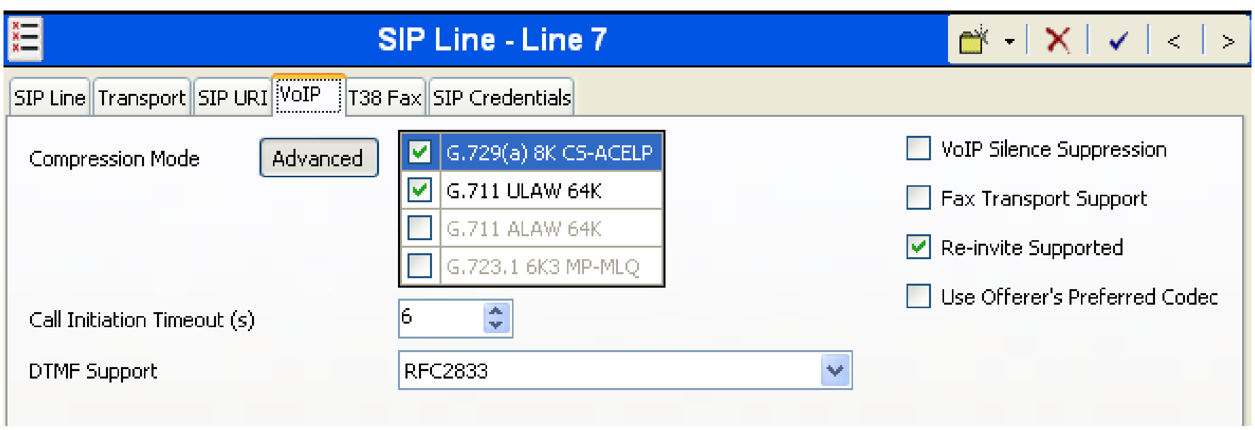
Select the **SIP URI** tab. To add a new SIP URI, click the **Add…** button. In the bottom of the screen, a New Channel area will be opened. To edit an existing entry, click an entry in the list at the top, and click the **Edit…** button. In the bottom of the screen, the Edit Channel area will be opened. 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 **PAI** parameter is new for 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 will behave like IP Office Release 6.0 with respect to the SIP P-Asserted-Identity header (e.g., IP Office will not include a PAI header for an outbound call unless privacy is asserted). 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 4.7. The **Outgoing Group** parameter, set here to 8, will be used for routing outbound calls to Verizon via the Short Codes (Section 4.6) or ARS configuration (Section 4.8)**.** 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".

4.4.4. 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 cause IP Office to include both G.729a and G.711MU in the Session Description Protocol (SDP) offer, 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).

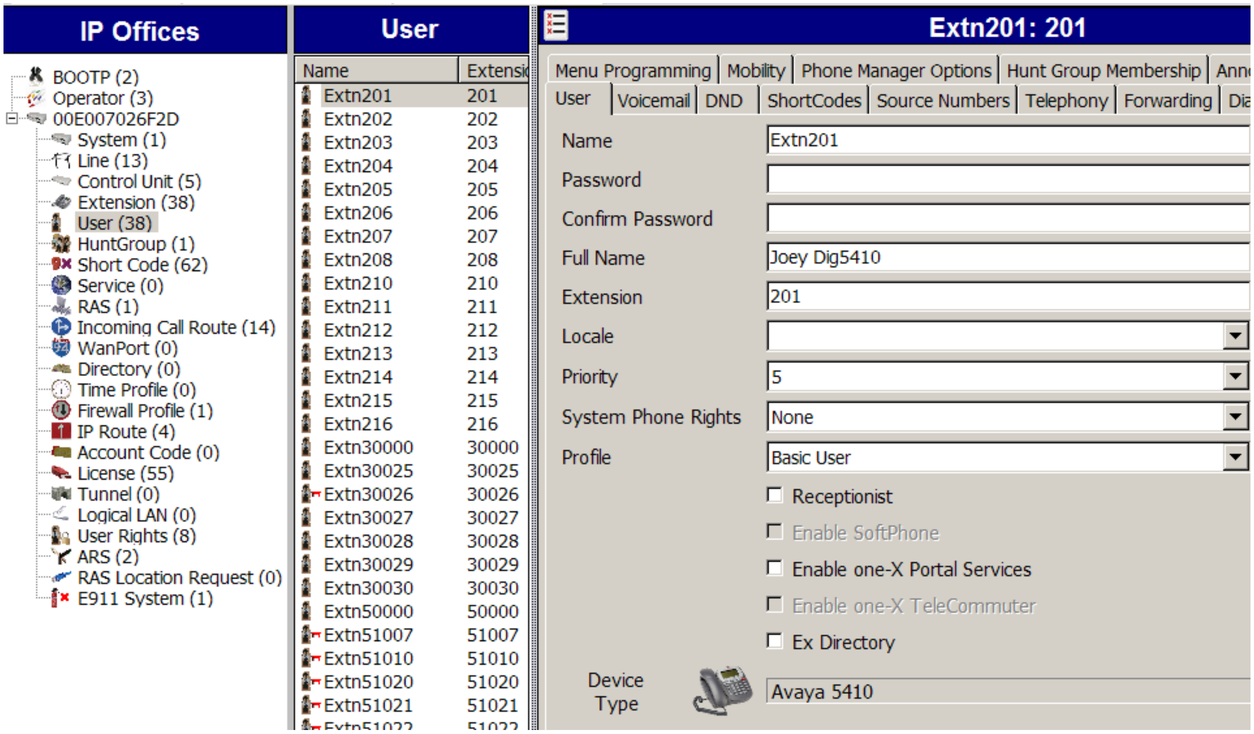


4.5. Users, Extensions, and Hunt Groups

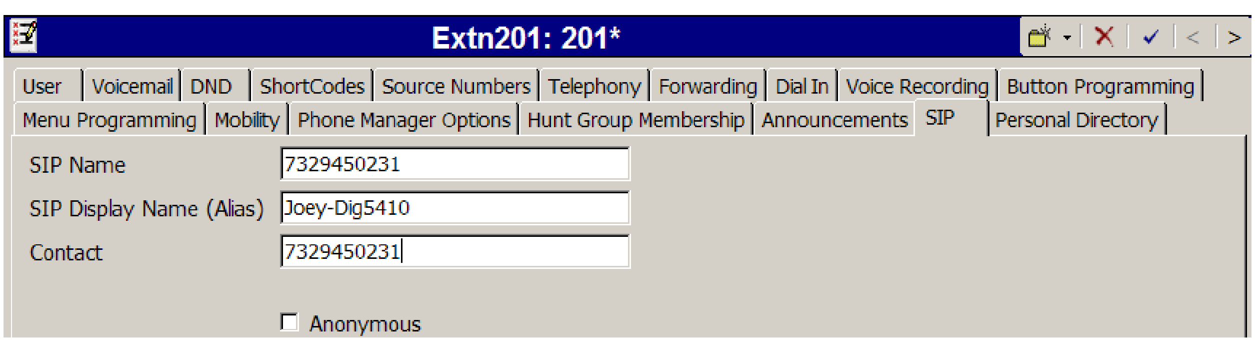
In this section, examples of IP Office Users, Extensions, and Hunt Groups will be illustrated. In the interests of brevity, not all users and extensions shown in **Figure 1** will be presented, since the configuration can be easily extrapolated to other users. To add a User, right click on **User** in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

4.5.1. User 201

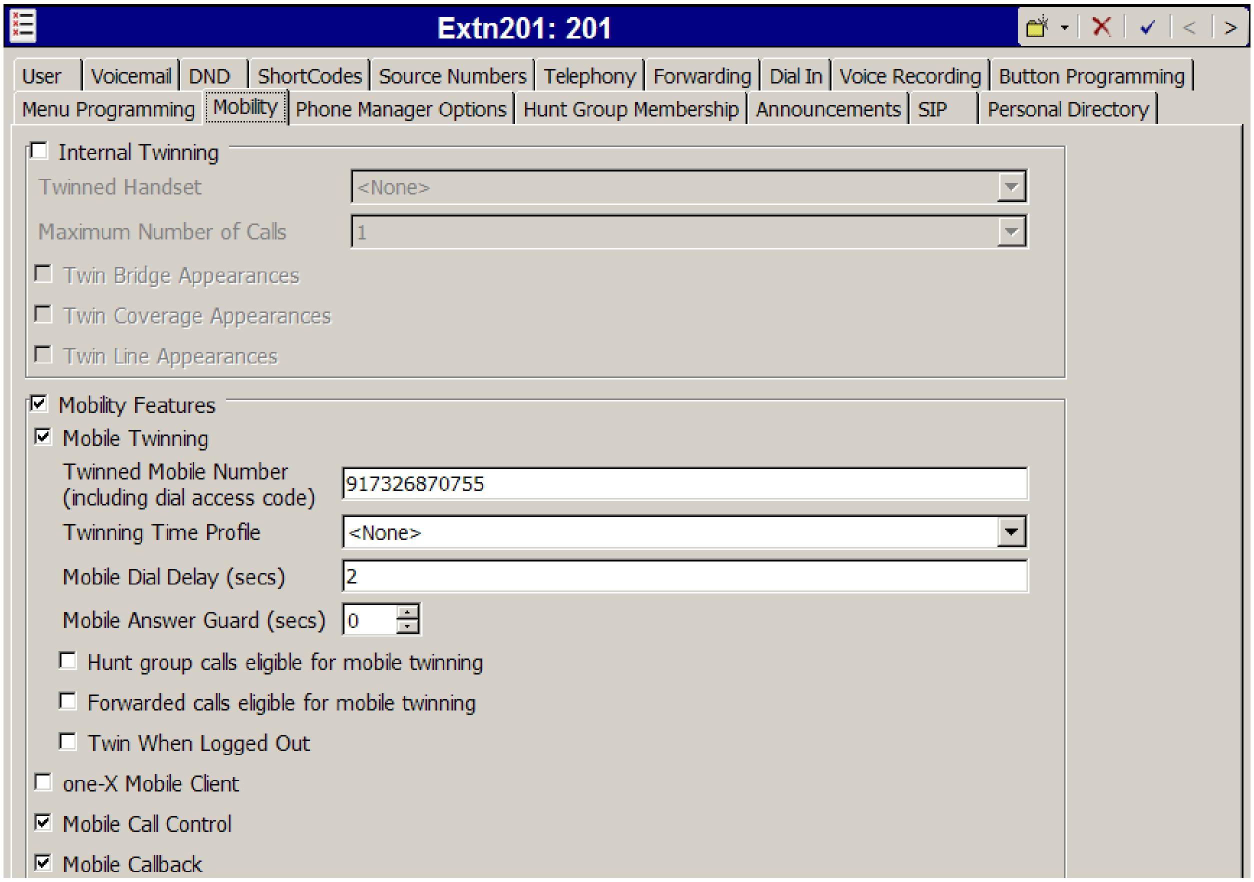
The following screen shows the **User** tab for User 201. As shown in **Figure 1**, this user corresponds to the digital telephone 5410.



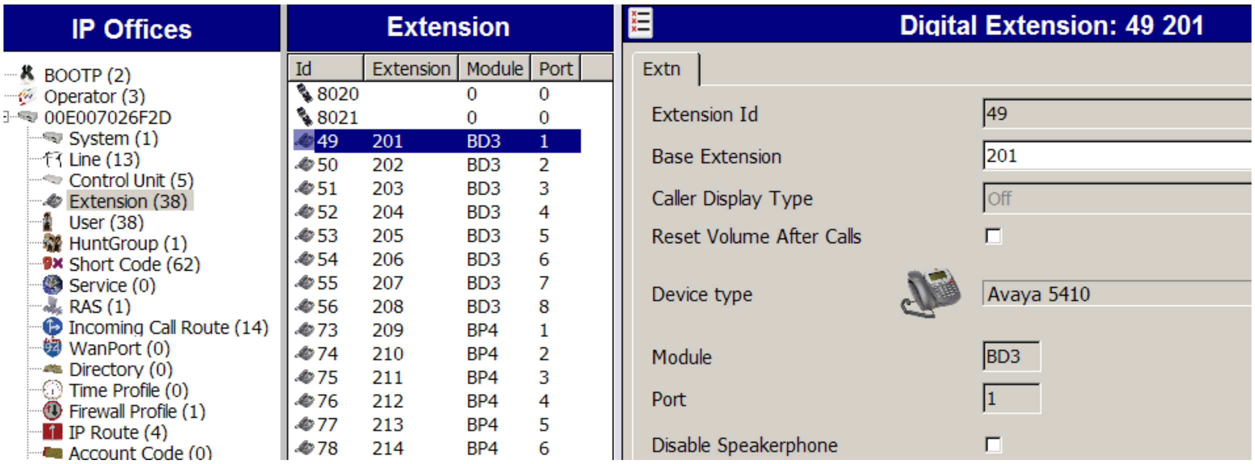
The following screen shows the **SIP** tab for User 201. The **SIP Name** and **Contact** parameters are configured with the DID number of the user, 7329450231. These parameters configure the user part of the SIP URI in the From header for outgoing SIP trunk calls, and allow matching of the SIP URI for 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 4.6 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.

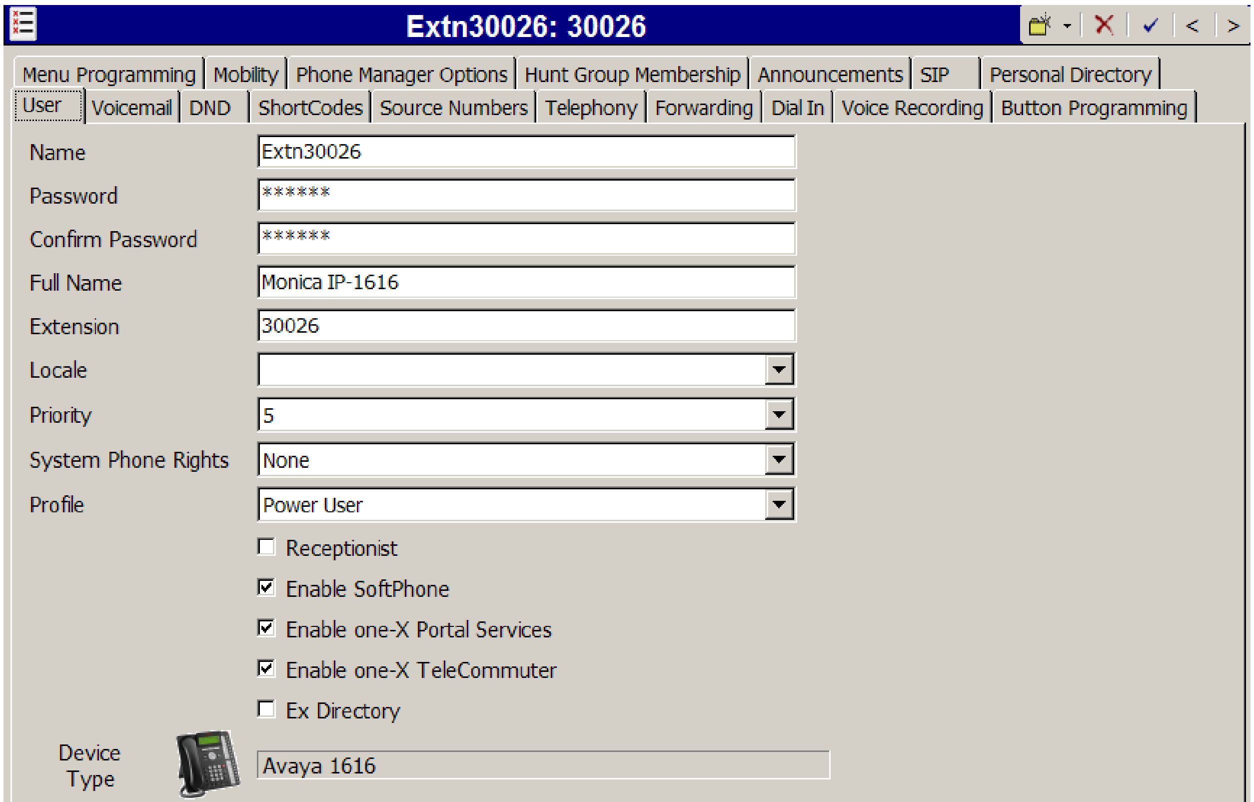


The following screen shows the Extension information for this user. To view, select Extension from the Navigation pane, and the appropriate extension from the Group pane. As stated in Section 4.1, the Avaya 5410 telephone user with extension 201 is connected to port 1 of the digital module.

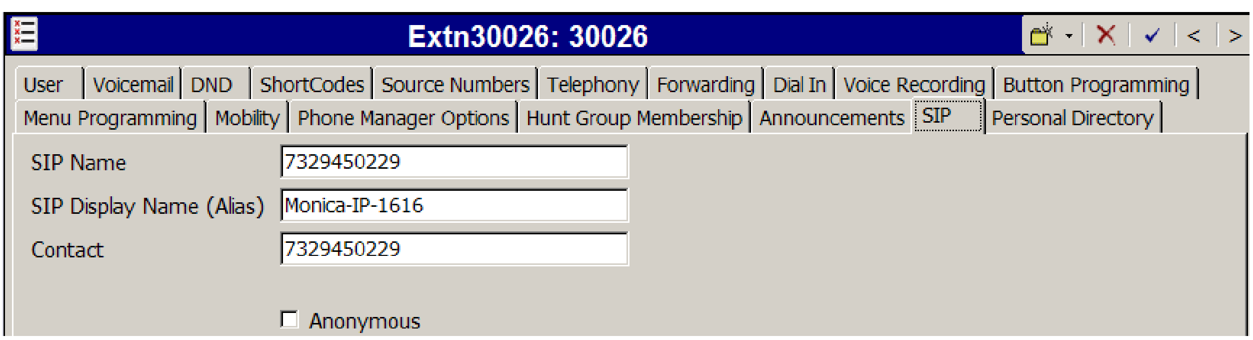


4.5.2. User 30026

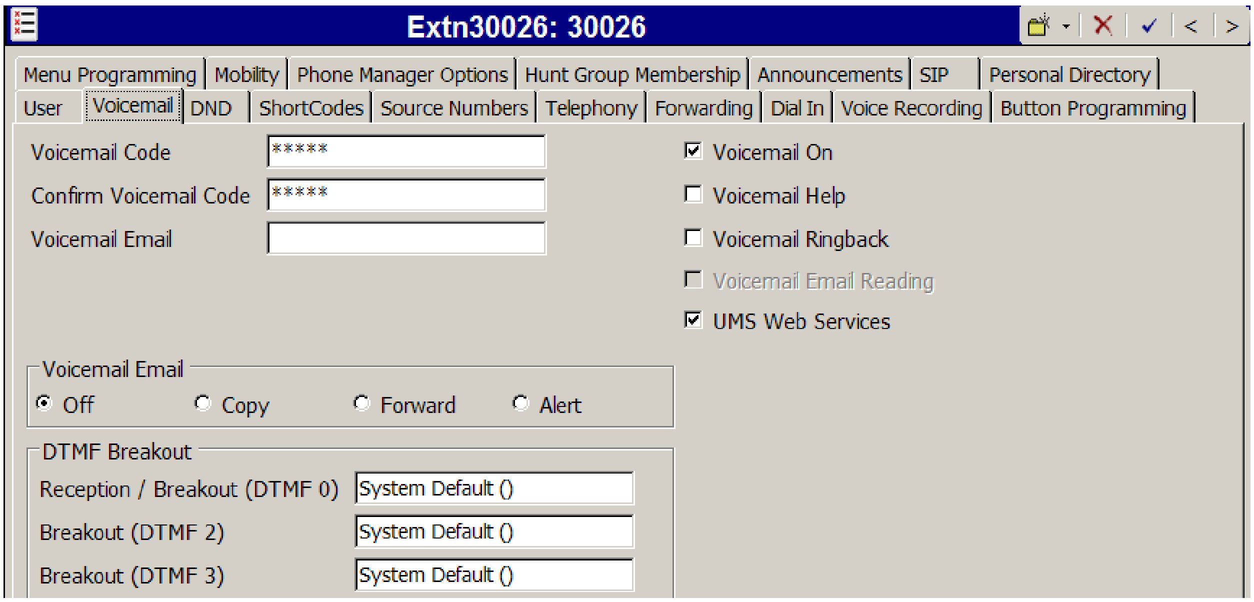
The following screen shows the **User** tab for User 30026. This user corresponds to an Avaya 1616 IP Telephone that will be granted “Power User” and Avaya IP Office Softphone features. The **Profile** parameter is set to “Power User”. The **Enable Softphone** box is checked, along with other advanced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



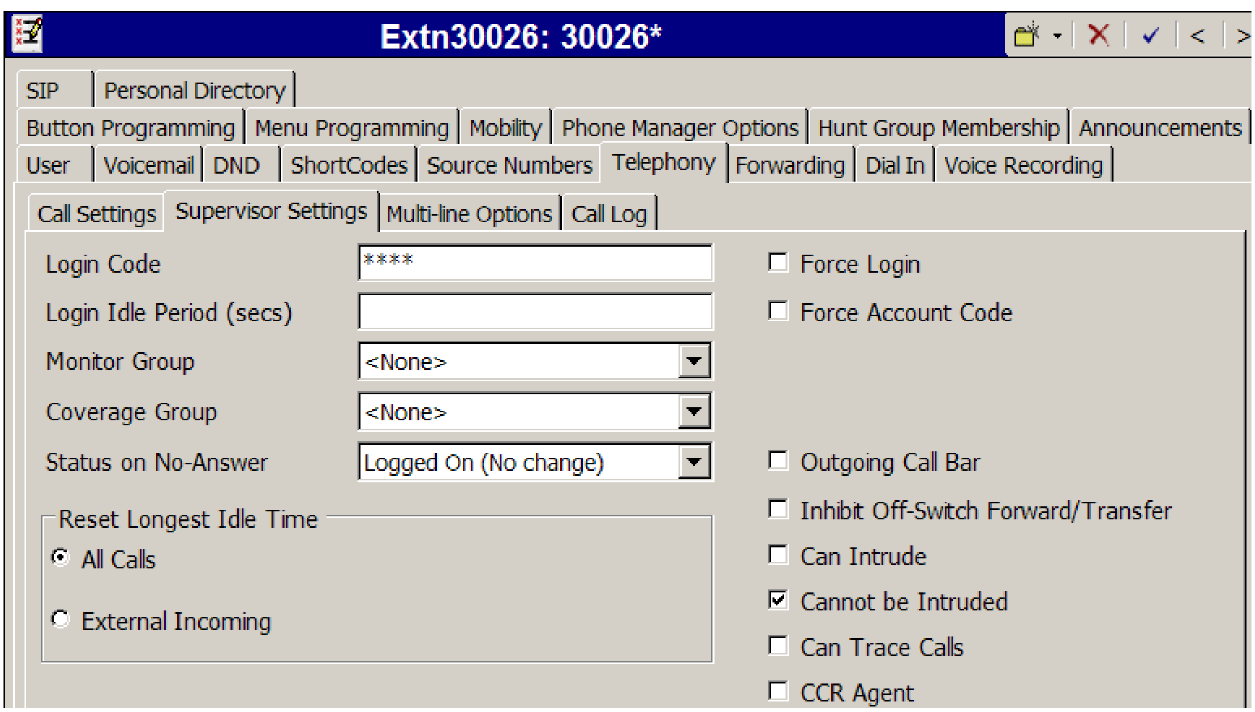
Like the user with extension 201, the **SIP** tab for the user with extension 30026 is configured with a **SIP Name** and **Contact** specifying the user’s Verizon Business DID number.



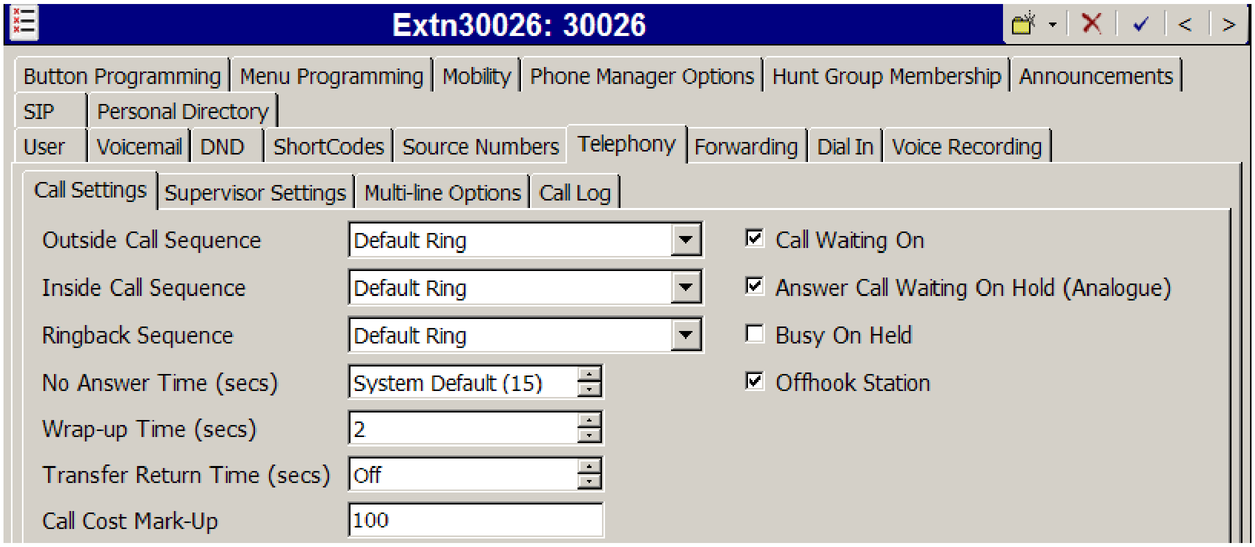
The following screen shows the **Voicemail** tab for the user with extension 30026. The **Voicemail On** box is checked, and a voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters. In the verification of these Application Notes, incoming calls from 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 4.6).



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

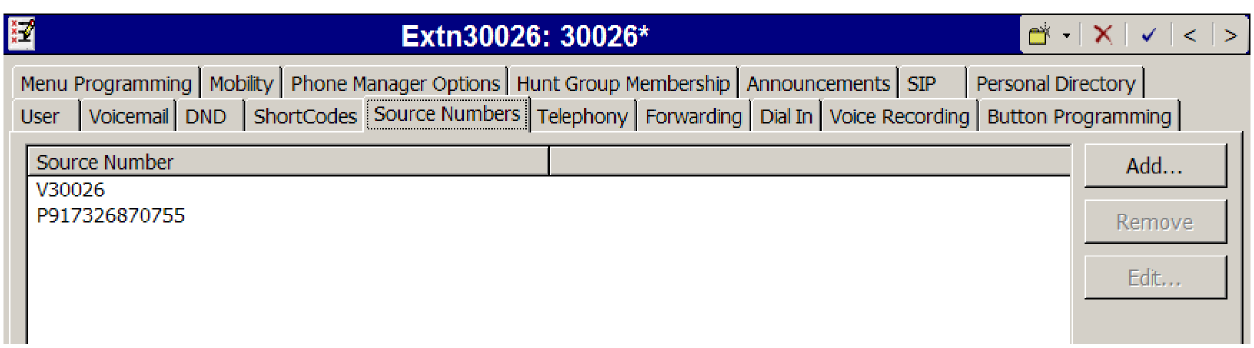


Select the **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow an IP Office Softphone logged in as this extension to have multiple call appearances (e.g., necessary for call transfer).

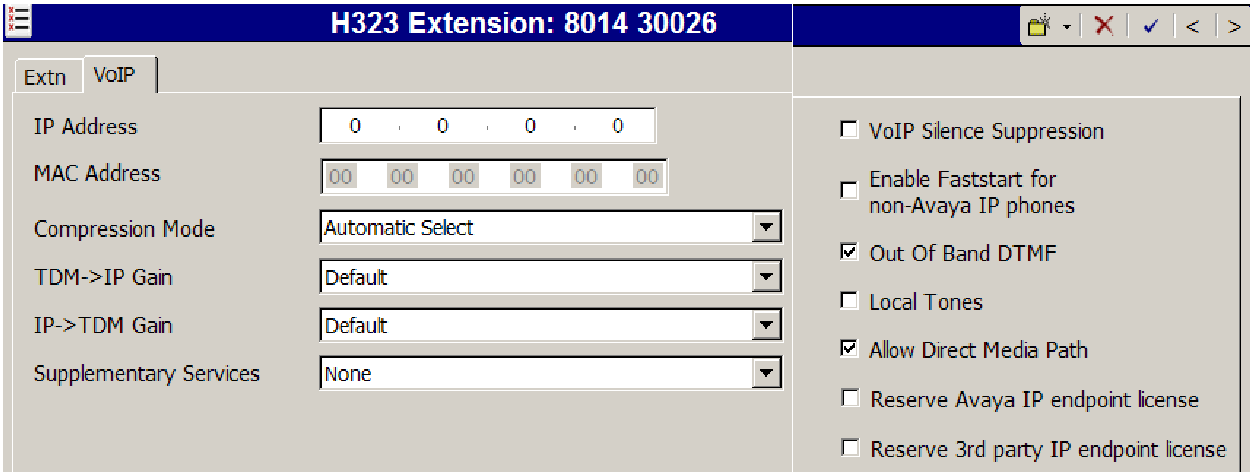


The following screen shows the **Source Numbers** tab for the user with extension 30026. Although the Voicemail Pro configuration is beyond the scope of these Application Notes, the "Callback" feature has been enabled on Voicemail Pro for this user, and the Source Number "P917326870755" has been previously 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 4.3.3. 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**.



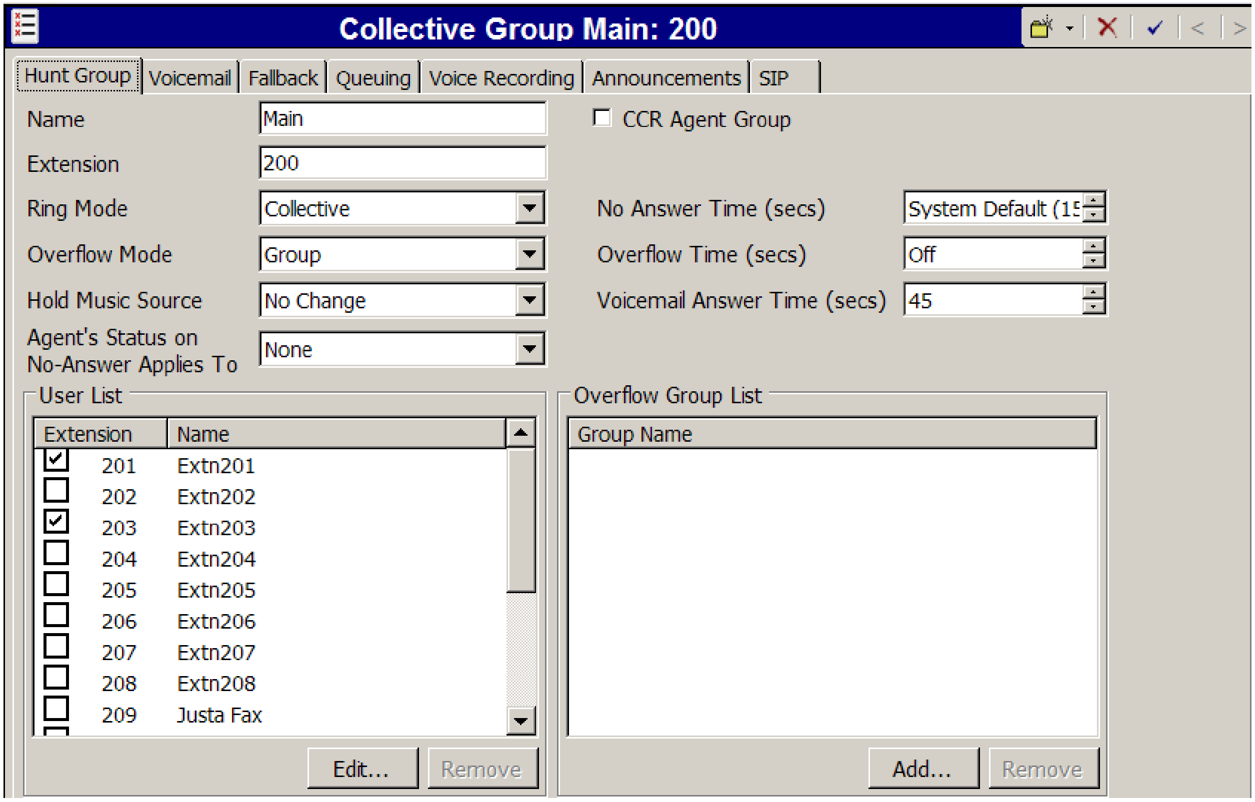
The following screen shows the Extension information for this user, simply to illustrate the **VoIP** tab available for an IP Telephone. To view, select **Extension** from the Navigation pane, and the appropriate extension from the Group pane. Select **VoIP** in the Details pane.



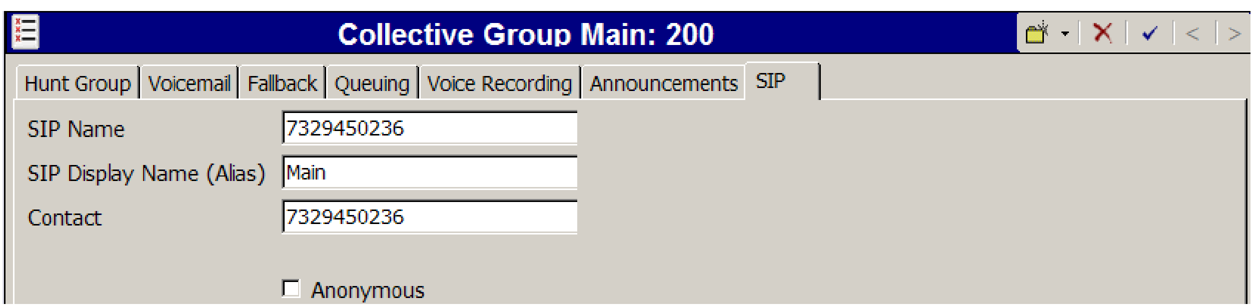
4.5.3. Hunt Groups

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.

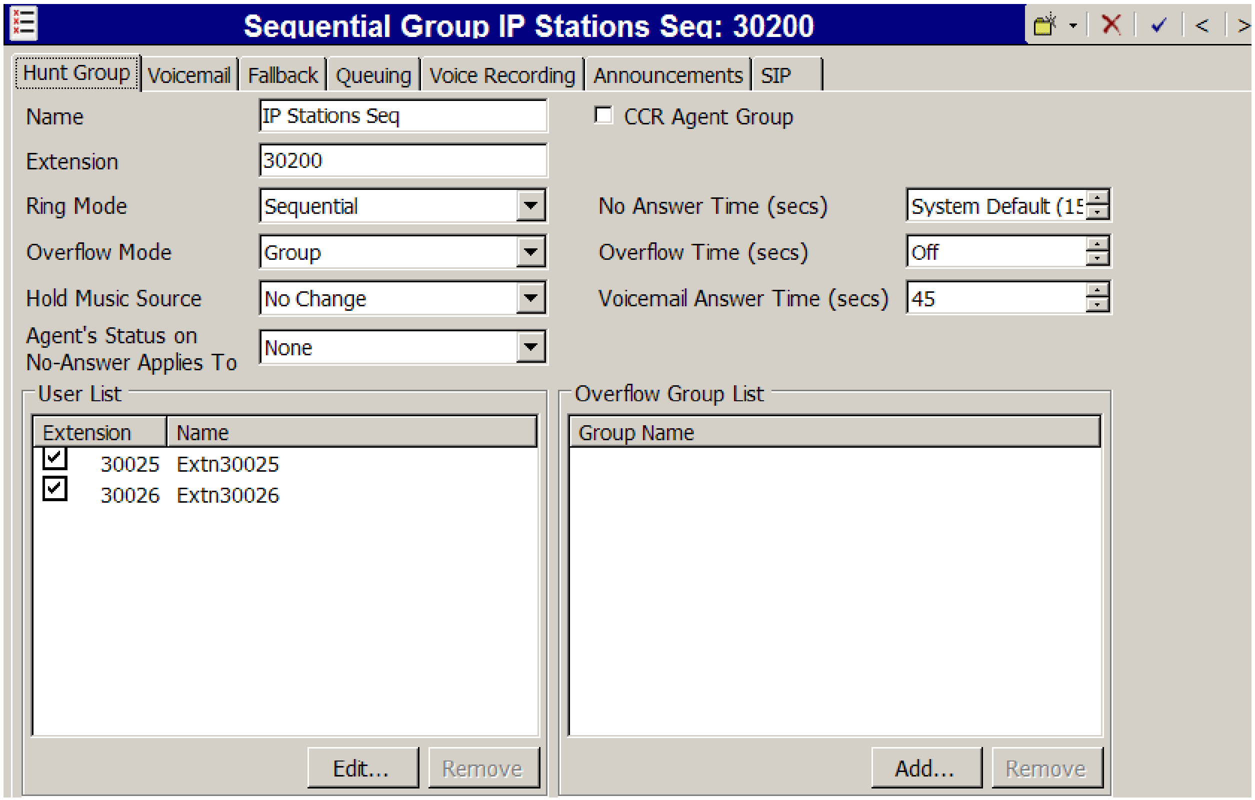
The following screen shows the **Hunt Group** tab for hunt group 200. This hunt group was configured to contain the two digital telephones x201 and x203 in **Figure 1**. These telephones will both ring when the hunt group number is called, due to the **Ring Mode** setting “Collective”. Click the **Edit**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



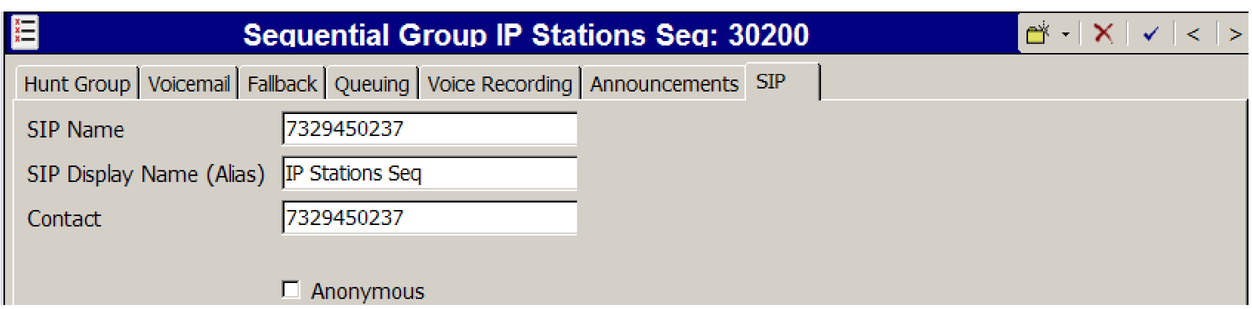
The following screen shows the **SIP** tab for hunt group 200. The **SIP Name** and **Contact** are configured with Verizon DID 7329450236. Later, in Section 4.7, an Incoming Call Route will map 7329450236 to this hunt group.



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 4.7, an incoming call route will map this same Verizon DID number to this hunt group.

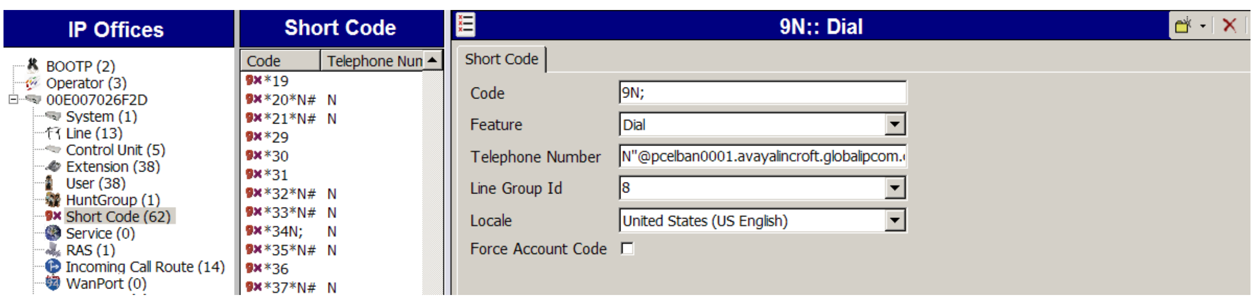


4.6. Short Codes

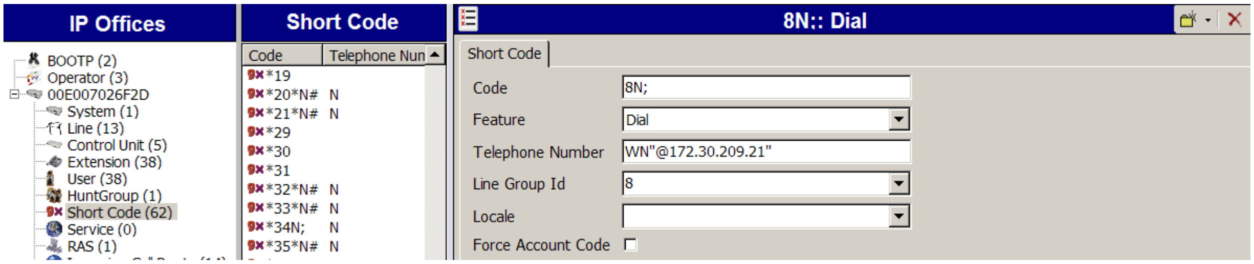
In this section, various examples of IP Office short codes will be illustrated. To add a short code, right click on **Short Code** in the Navigation pane, and select **New.** To edit an existing short code, click **Short Code** in the Navigation pane, and the short code to be configured in the Group 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 4.4)**.**

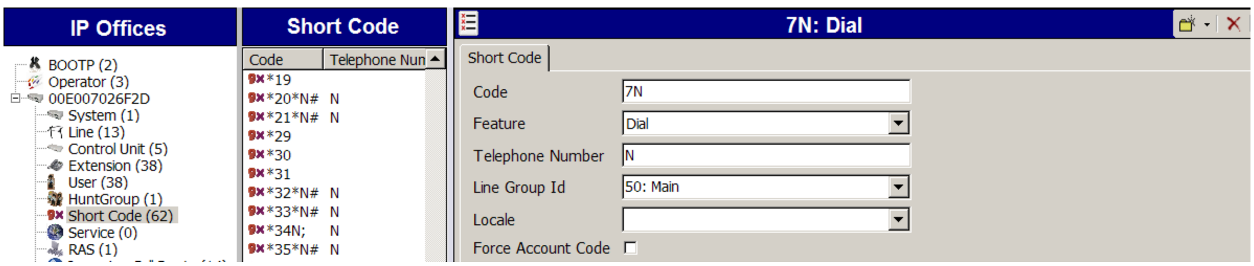
This simple short code will allow an IP Office user to dial the digit 9 followed by any telephone number, symbolized by the letter **N**, to reach the SIP Line to Verizon business. **N** can be any number such as a 10-digit number, a 1+10 digit number, a toll free number, directory assistance (e.g., 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).



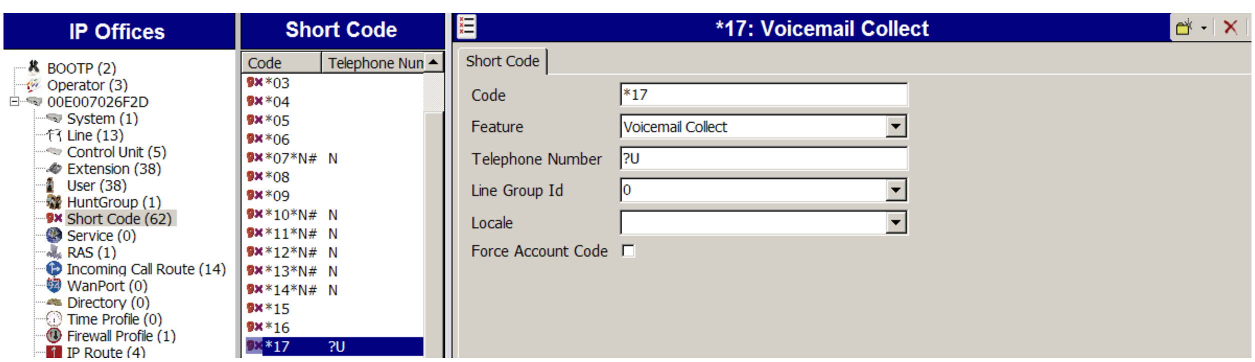
Optionally, add or edit a short code that can be used to access the SIP Line anonymously. In the screen shown below, the short code “8N;” is illustrated. This short code is similar to the “9N;” short code except that the Verizon IP Address rather than the domain is entered in the **Telephone Number** 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.



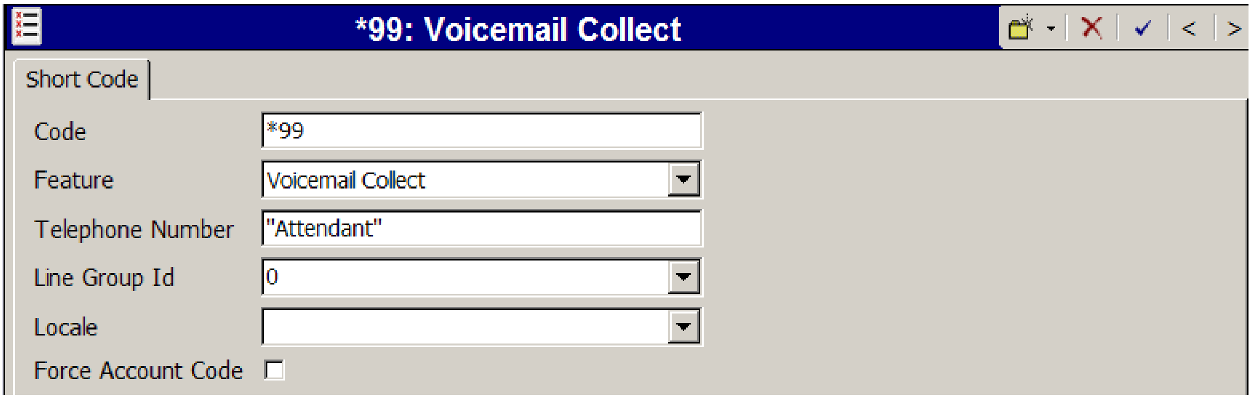
The simple "9N;" and "8N;" short codes illustrated previously do not provide a means of alternate routing if the primary Verizon SIP line is out of service or temporarily not responding. When alternate routing options and/or more customized analysis of the digits following the short code are desired, 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 4.8 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 4.7.



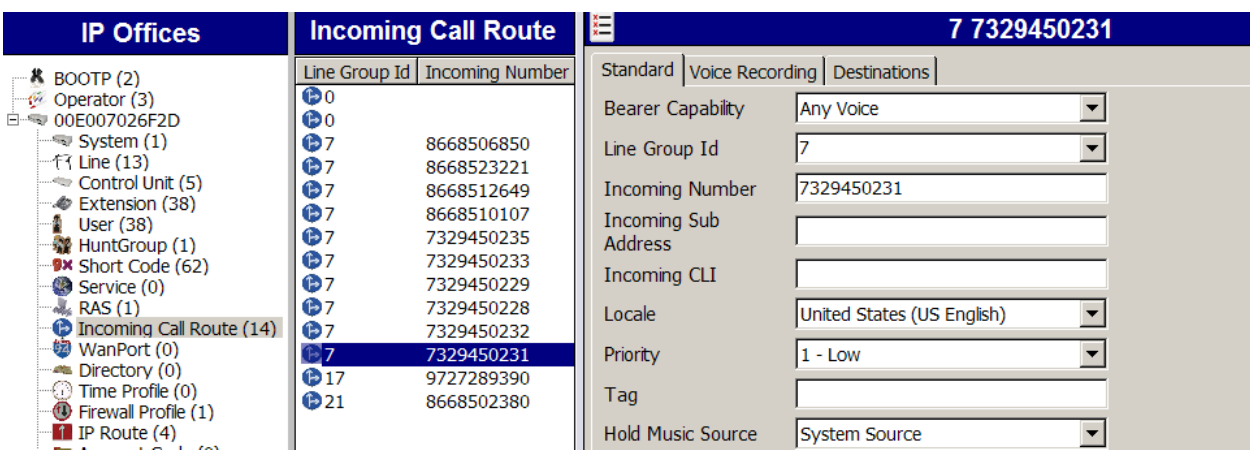
The following screen illustrates another short code that acts like a feature access code rather than a means to access a SIP Line. In this case, the **Code** "*99" is defined for **Feature** "Voicemail Collect" and Telephone Number "Attendant". This short code will be used as a means to allow a Verizon 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 4.7.



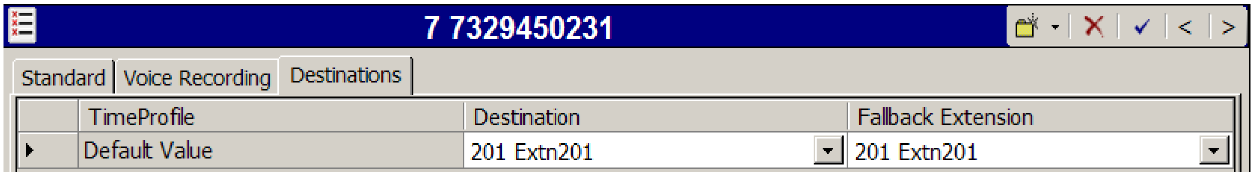
4.7. Incoming Call Routes

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.

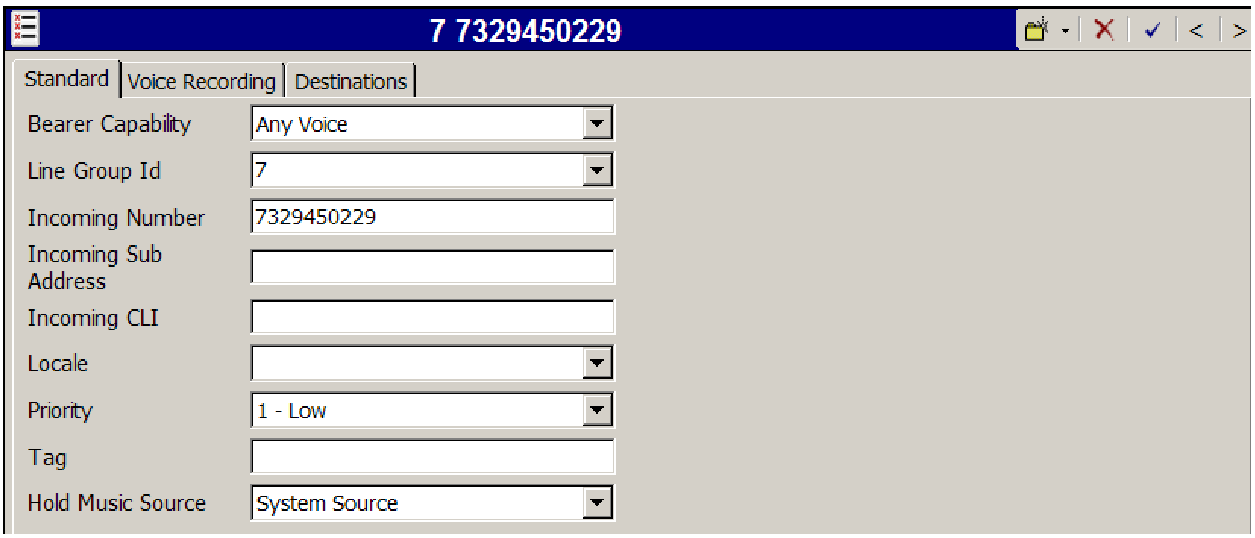
In the screen shown below, the incoming call route for **Incoming Number** "7329450231" is illustrated. The **Line Group Id** is 7, matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to Verizon Business in Section 4.4.



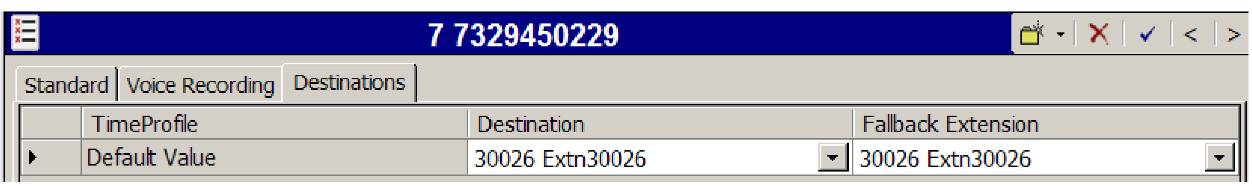
Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when a PSTN user dials 7329450231. As shown in **Table 1**, 7329450231 is the DID number associated with IP Office user extension 201.



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 4.4.

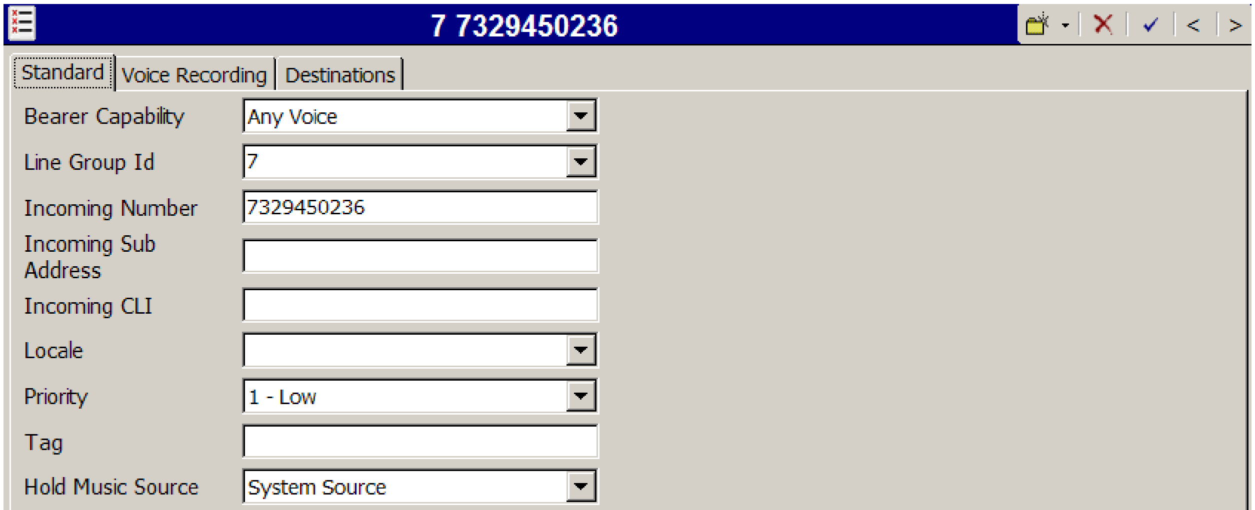


Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when a PSTN user dials 7329450229. As shown in **Table 1**, 7329450229 is the DID number associated with IP Office user extension 30026.

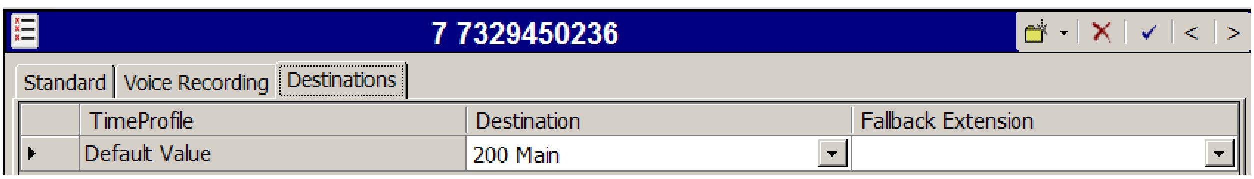


Incoming Call Routes for other direct mappings of DID numbers to IP Office users listed in **Table 1** are omitted here, but can be configured in the same fashion.

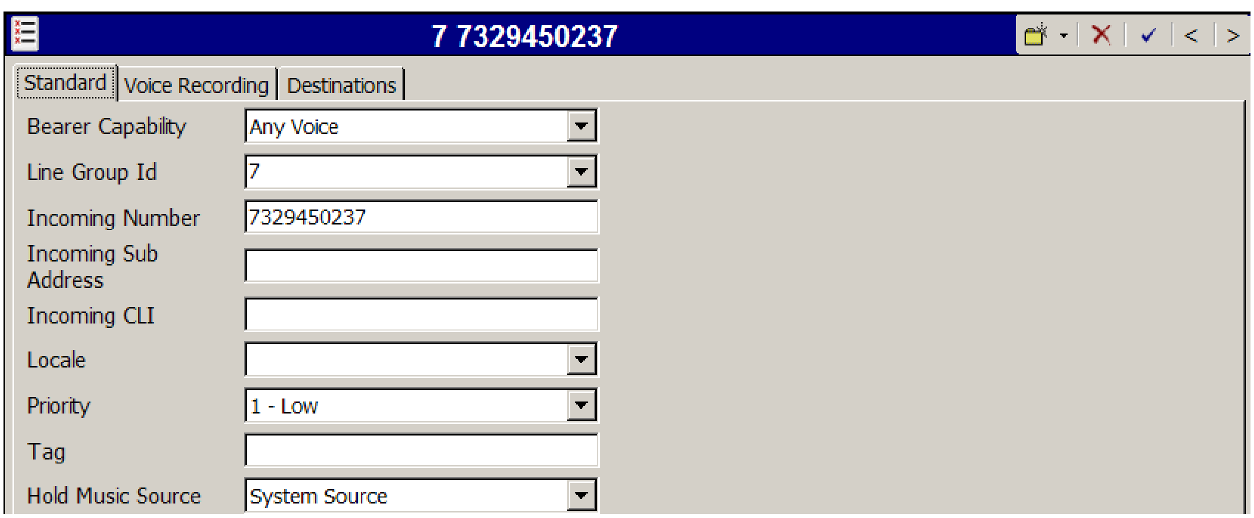
In the screen shown below, the incoming call route for **Incoming Number** "7329450236" is illustrated. The **Line Group Id** is 7, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in Section 4.4.



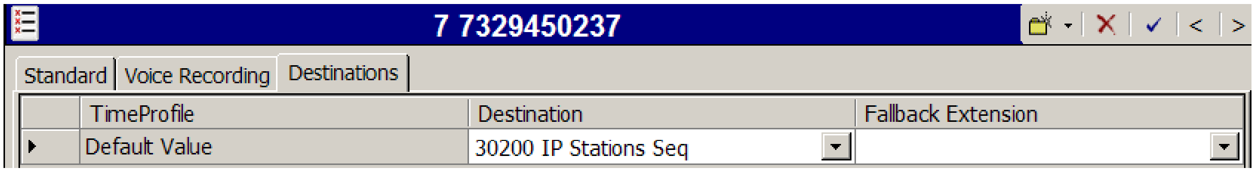
Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when a PSTN user dials 7329450236. In this case, the destination is the hunt group "200 Main" whose configuration is shown in Section 4.5.3.



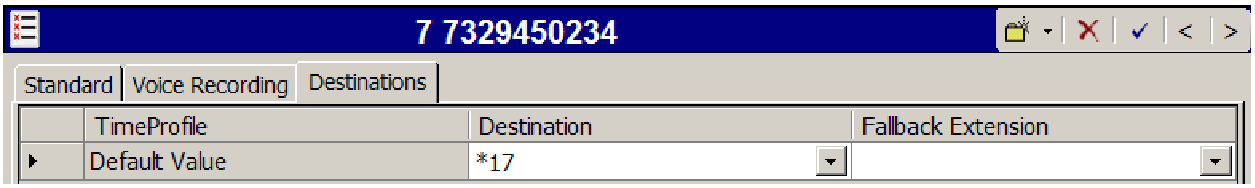
In the screen shown below, the incoming call route for **Incoming Number** "7329450237" is illustrated. The **Line Group Id** is 7, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in Section 4.4.



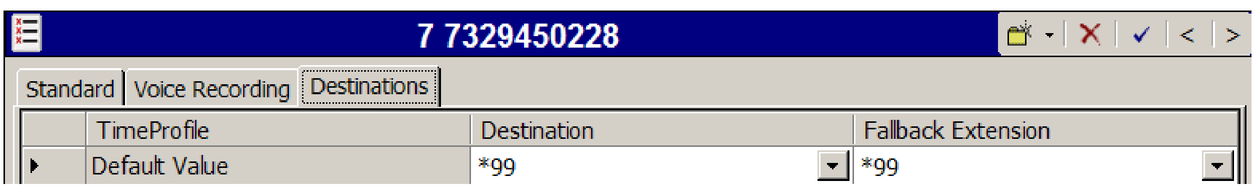
Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when a PSTN user dials 7329450237. In this case, the destination is the hunt group "30200 IP Stations Seq" whose configuration is shown in Section 4.5.3.



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 4.6. 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 4.6. 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.



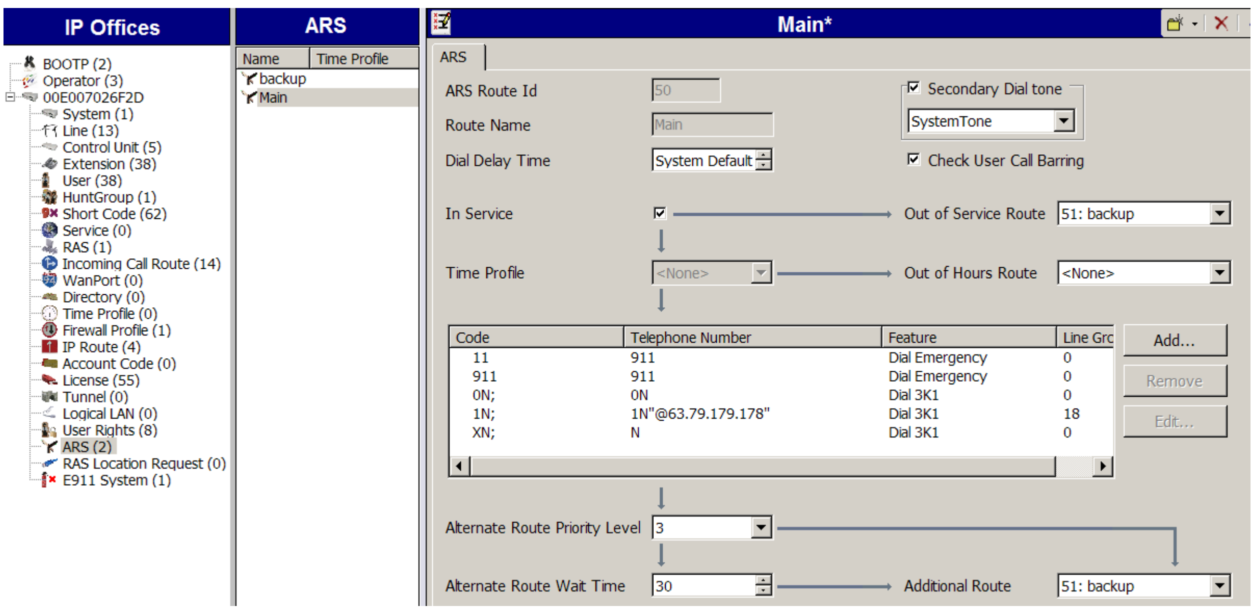
4.8. ARS and Alternate Routing

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 4.6. 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.

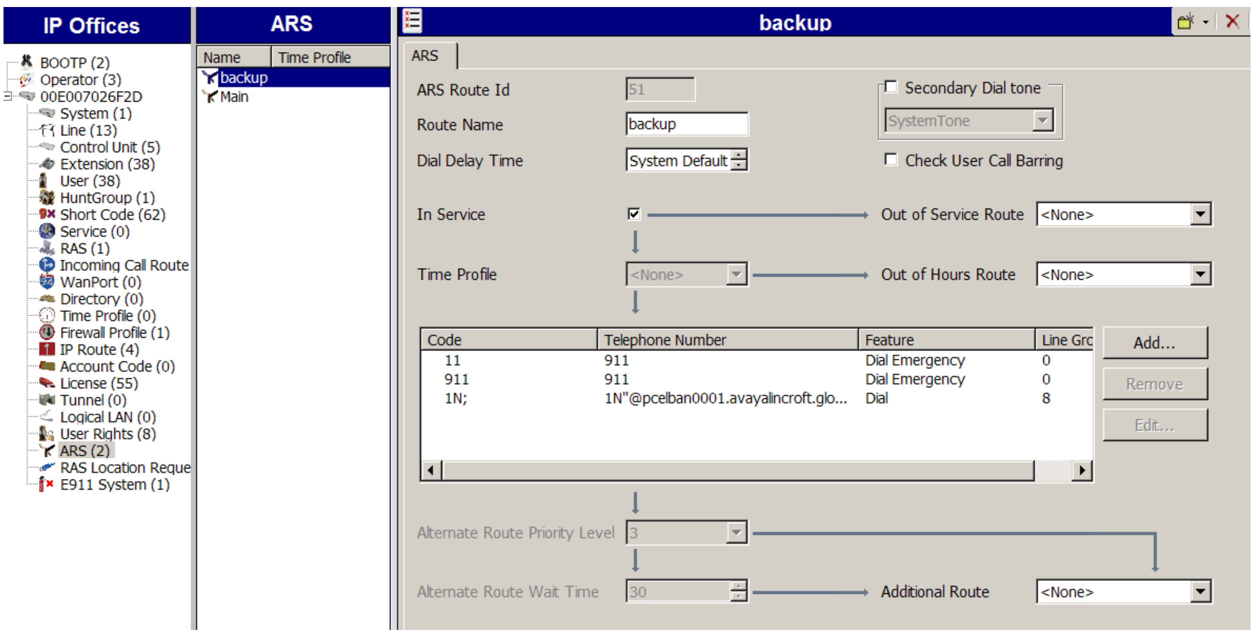
To add a new ARS route, right-click **ARS** in the Navigation pane, and select **New**. To view or edit an existing ARS route, select **ARS** in the Navigation pane, and select the appropriate route name in the Group pane.

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 4.6) 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 4.6. 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 4.6). 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.



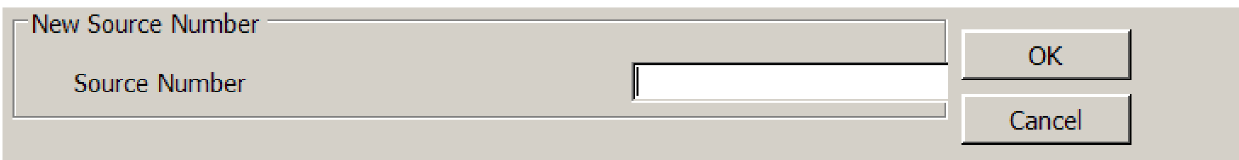
In the testing associated with the configuration, calls were successfully delivered to SIP Line 8 via both the primary ARS route "50: Main" (via changes to "50: Main") as well as the backup ARS route shown above. If a primary route experiences a network outage such that no response is received to an 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.

4.9. Privacy / Anonymous Calls

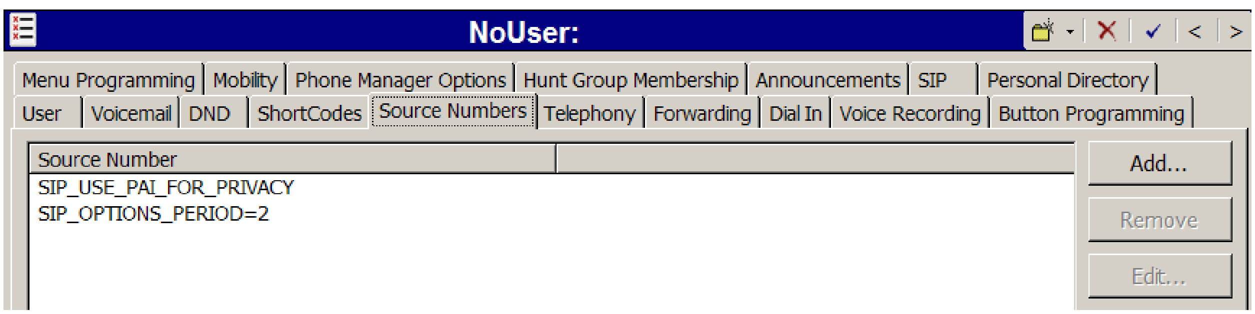
As described in Section 4.6, an IP Office user whose calling line identification is not typically withheld from the network can request privacy in the sample configuration by dialing the short code 8 to access the SIP Line. The Avaya 1600-Series IP Telephones can also request privacy for a specific call, 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 4.5).

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.

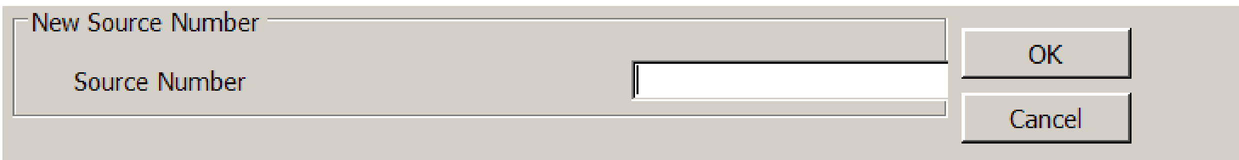


4.10. SIP Options Frequency

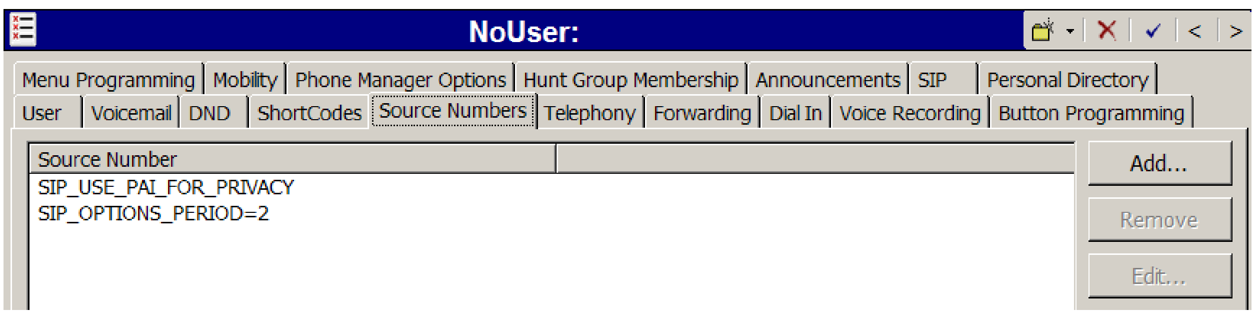
In Section 4.4, the SIP Line to Verizon Business is shown with the **Check OOS** box checked. In the sample configuration, IP Office periodically checks the health of the SIP Line by sending a SIP OPTIONS message. If there is no response, IP Office can mark the trunk out of service. Although ARS as shown in Section 4.8 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 4.3.2 and Section 4.4.

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**.



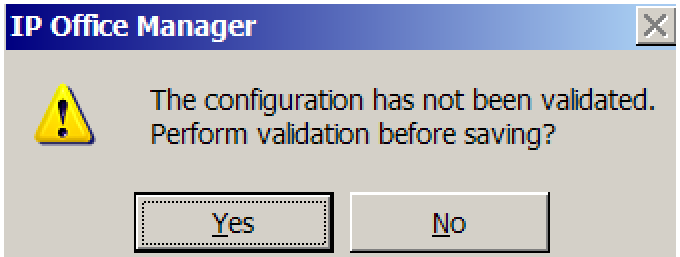
The source number **SIP_OPTIONS_PERIOD=2** should now appear in the list of Source Numbers as shown below.



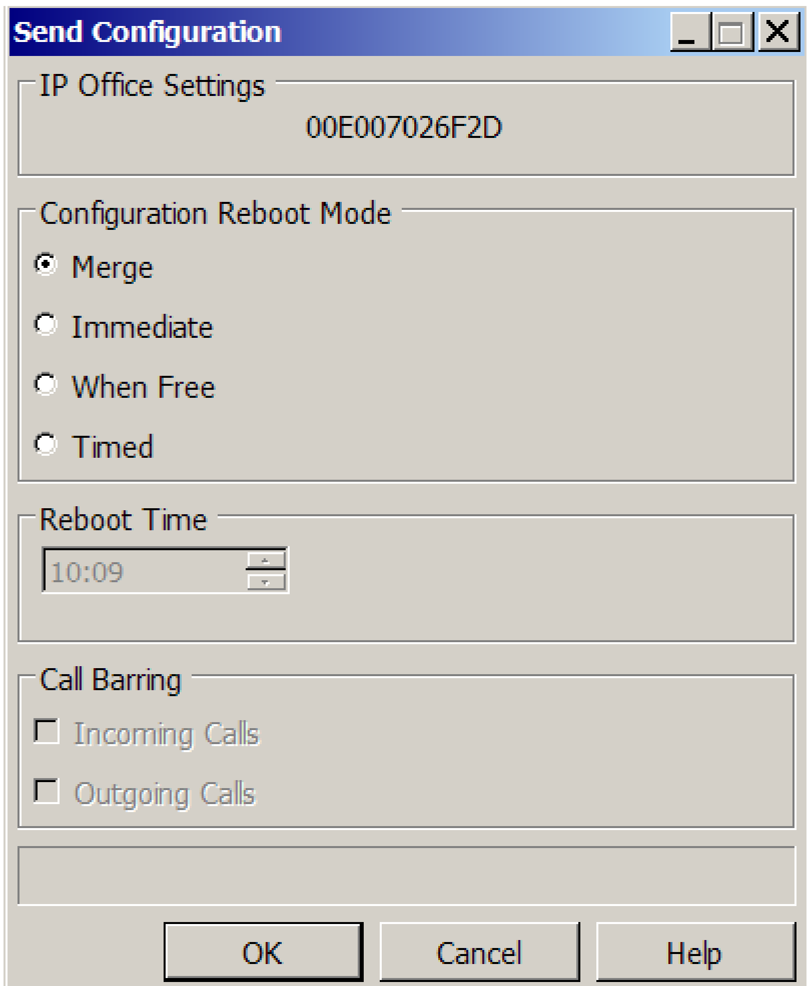
With this configuration, Binding Refresh Intervals of 30 seconds and 90 seconds were tested successfully. That is, IP Office sourced SIP OPTIONS every 30 or 90 seconds, depending on the value configured in the Binding Refresh Interval, since the Binding Refresh Interval was less than the value configured via the SIP_OPTIONS_PERIOD source number.

4.11. 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 "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.



5. 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 4.

6. General Test Approach and Results

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.

The focus of the verification testing associated with this version of the IP Office with Verizon IP Trunk Application Notes was the use of SIP REFER messaging for call transfer scenarios, and the use of DNS SRV to Verizon, rather than static provisioning of the Verizon SIP signaling information. Note that the REFER testing and the DNS SRV testing are not dependent on one another. That is, IP Office can use REFER for call transfer without using DNS SRV, and IP office can use DNS SRV without using REFER for call transfer.

Testing was successful. Any limitations related to the overall configuration are noted in Section 1.3.

7. Verifications

This section summarizes and provides detailed illustrations of the verification of REFER and DNS SRV with Verizon IP Trunk Service.

7.1. REFER Testing

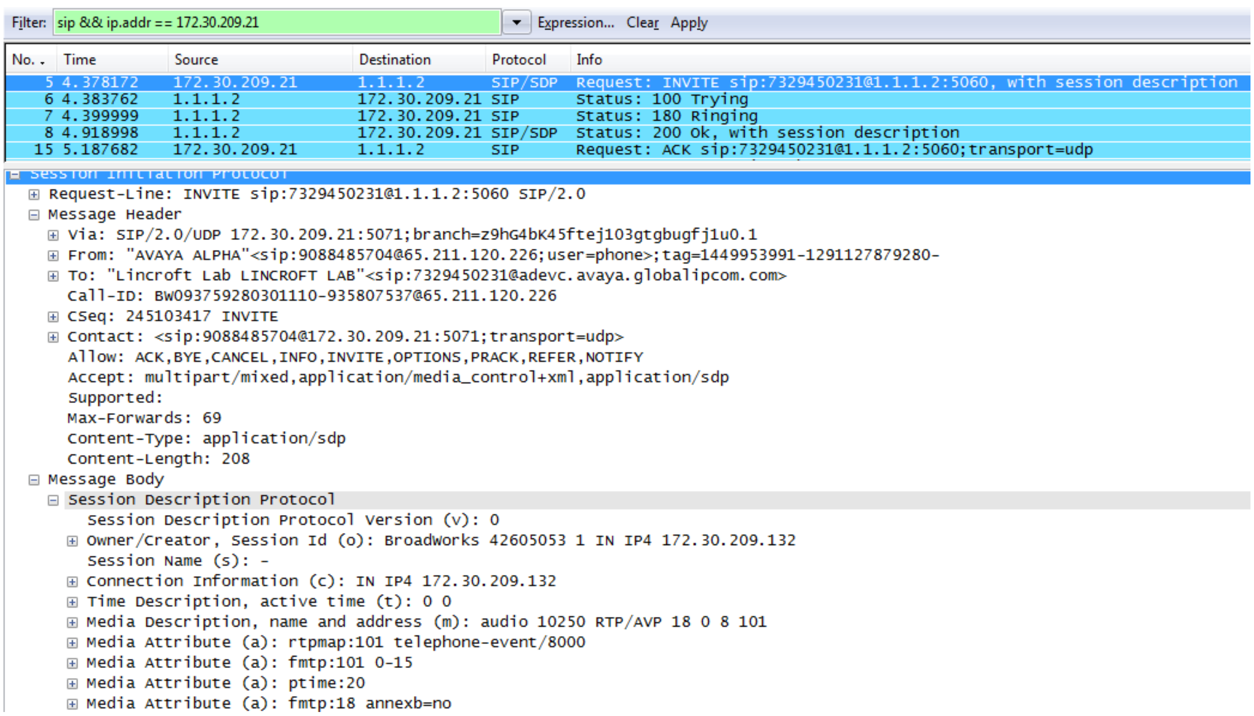
The following scenarios will result in IP Office sending REFER to the Verizon network. Each scenario was tested successfully.

- PSTN user makes call to Verizon IP Trunk DID and IP Office user answers. IP Office user performs an attended transfer of the inbound call to a PSTN destination using the Verizon IP Trunk service. In this context, an attended transfer implies that the destination of the outbound call answers before the IP Office user completes the transfer. In terms of SIP signaling, this means that IP Office sends the REFER after the 200 OK is received from Verizon. This scenario is illustrated with Wireshark in Section 7.1.1.
- IP Office user makes an outbound call to the PSTN via the Verizon IP Trunk service. The IP Office user then performs an attended transfer of the call to another PSTN destination using the Verizon IP Trunk service. In this context, an attended transfer implies that the destination of the outbound call answers before the IP Office user completes the transfer. In terms of SIP signaling, this means that IP Office sends the REFER after the 200 OK is received from Verizon.
- PSTN user makes call to Verizon IP Trunk DID and IP Office user answers. The IP Office user performs an unattended transfer of the inbound call to a PSTN destination using the Verizon IP Trunk service. In this context, an unattended transfer implies that the destination of the outbound call does not answer before the IP Office user completes the transfer. In terms of SIP signaling, this means that IP Office sends the REFER after a message such as 183 Session Progress is received from Verizon but before a 200 OK is received from Verizon.
- IP Office user makes an outbound call to the PSTN via the Verizon IP Trunk service. The IP Office user performs an unattended transfer of the call to another PSTN destination using the Verizon IP Trunk service. In this context, an unattended transfer implies that the destination of the outbound call does not answer before the IP Office user completes the transfer. In terms of SIP signaling, this means that IP Office sends the REFER after a message such as 183 Session Progress is received from Verizon but before a 200 OK is received from Verizon.

7.1.1. Wireshark Trace Illustration for REFER-Transfer

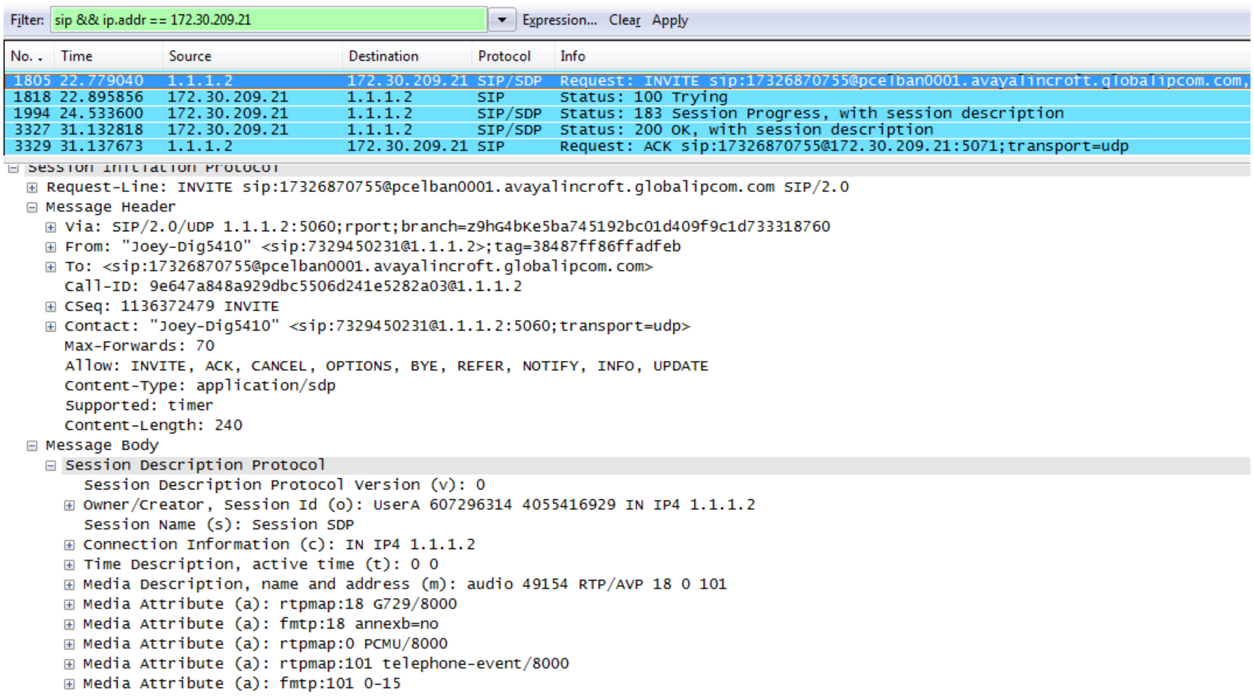
This section illustrates the SIP signaling for an inbound Verizon IP Trunk call that is transferred back to the PSTN by an IP Office user. IP Office will use SIP REFER.

The following screen shows the portion of the trace until the point where the IP Office user answers the inbound call. Frame 5 is selected and expanded to show the contents of the INVITE message from Verizon. In frame 8, IP Office answers the call with a 200 OK.

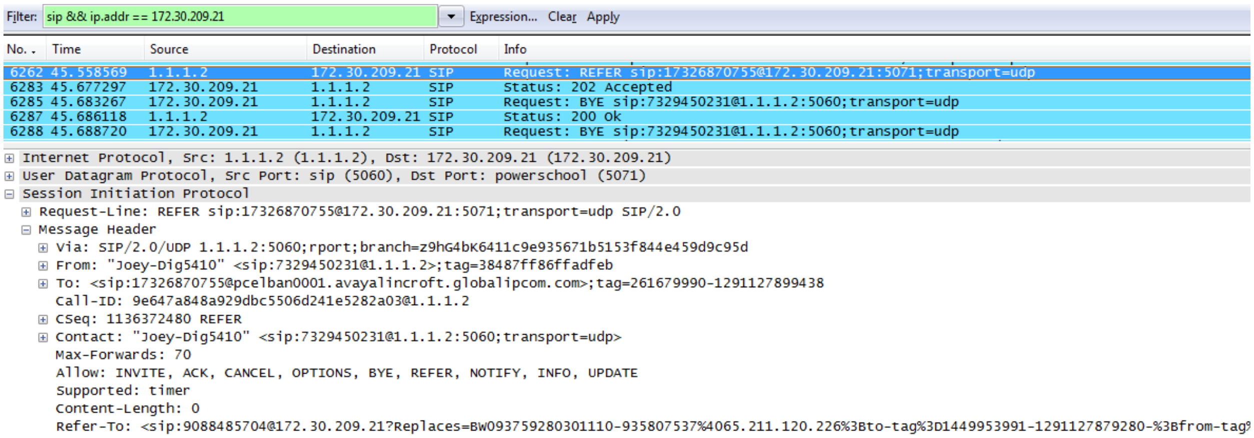


After answering the inbound call, the IP Office user presses the transfer button, and dials the number 9-1-732-687-0755 which is routed by IP Office to the Verizon IP Trunk service. In this “attended transfer” example, the IP Office user performing the transfer waits for the outbound call to be answered before pressing the transfer button a second time to complete the transfer.

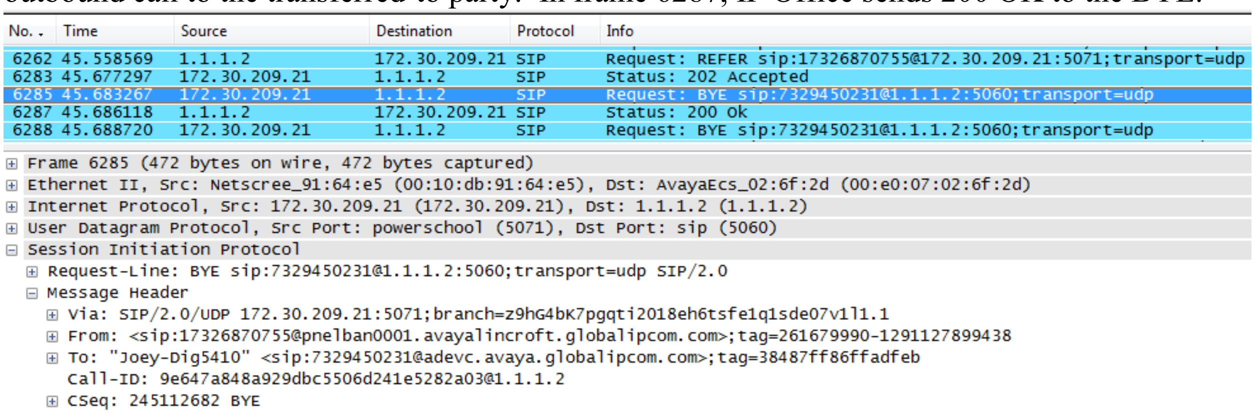
Scrolling down in the same trace in the screen below, frame 1805 is selected to show the contents of an outbound INVITE sent by IP Office to the Verizon IP Trunk service. In frame 3327, Verizon sends the 200 OK when the called party answers the call.



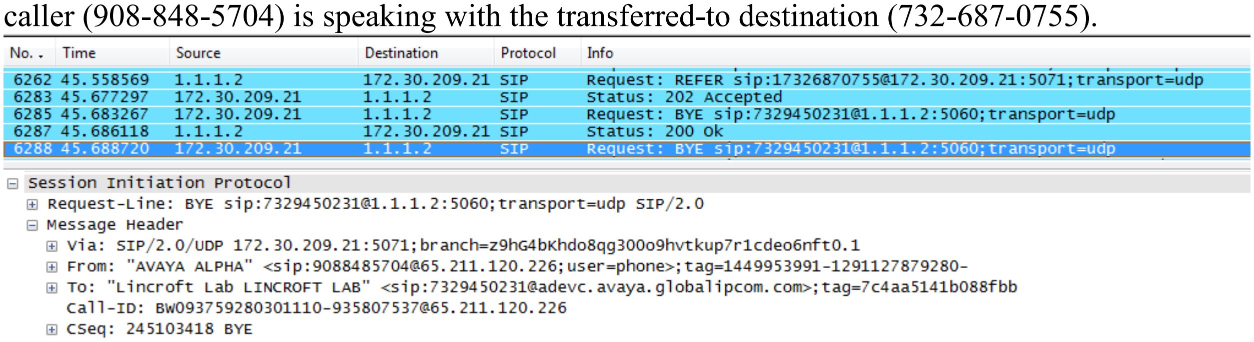
In this example, after the called party answers, the IP Office user presses the transfer button a second time to complete the transfer. Scrolling down in the same trace in the screen below, frame 6262 is selected to show the contents of a REFER message sent by IP Office to the Verizon IP Trunk service. The REFER contains the Call-ID of the outbound call from IP Office to Verizon, and the Refer-To header contains the Call-ID, to-tag, and from-tag, associated with the original inbound call from Verizon. In frame 6283, Verizon sends a 202 Accepted for the REFER.



In frame 6285 expanded below, Verizon sends a BYE for the Call-ID associated with the outbound call to the transferred-to party. In frame 6287, IP Office sends 200 OK to the BYE.



In frame 6288 expanded below, Verizon sends a BYE for the Call-ID associated with the original inbound call from Verizon to IP Office. The two IP Office trunks are cleared, and the PSTN caller (908-848-5704) is speaking with the transferred-to destination (732-687-0755).

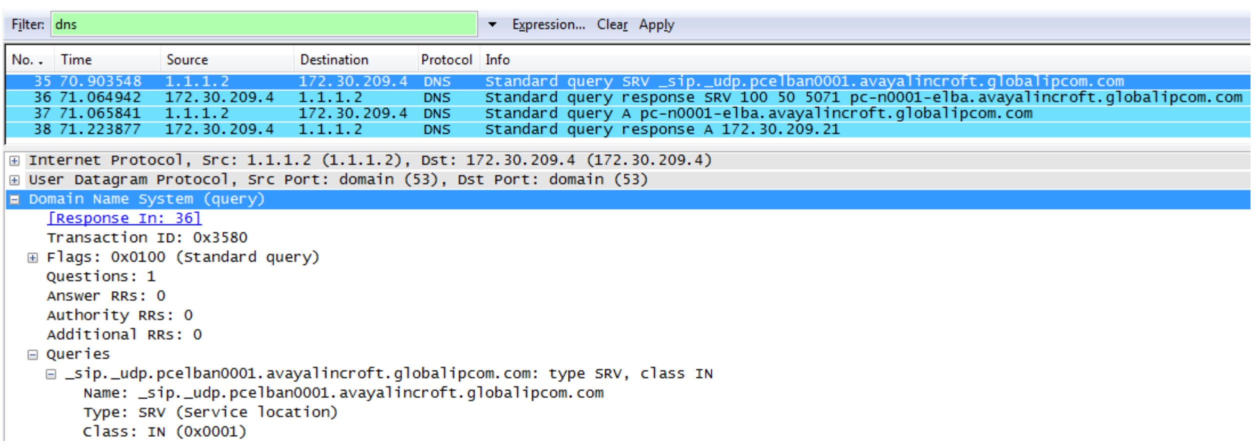


7.2. DNS SRV Testing

The new IP Office Release 6.1 capability to determine the Verizon SIP signaling address and port using DNS procedures was tested using the production Verizon PIP circuit. Rather than statically configure IP Office with the Verizon IP Address and SIP signaling port, as was the case shown in reference [JRR-IPOR6], IP Office determined the Verizon IP Address and signaling port dynamically using DNS. On the production circuit used for testing, Verizon responded with one "answer".

7.2.1. Wireshark Trace Illustration for DNS SRV

This section illustrates the DNS signaling used when the SIP Line in IP Office is configured to use DNS SRV. Please reference Section 4.4.2 of these Application Notes for the relevant configuration. In the filtered Wireshark trace shown below, IP Office (1.1.1.2) sends a DNS SRV query to the Verizon DNS server (172.302.209.4) configured in IP Office for the SIP Line. Frame 35 is highlighted and expanded. Note that the query contains "_sip._udp.pcelban0001.avayalincroft.globalipcom.com" because the IP Office SIP Line has been configured with "pcelban0001.avayalincroft.globalipcom.com" as the Verizon domain, using UDP for transport.



The Verizon DNS response in frame 36 is highlighted and expanded in the following screen. Note that the "Answer" contains Target "pc-n0001-elba.avayalincroft.globalipcom.com" and port 5071.

No.	Time	Source	Destination	Protocol	Info
35	70.903548	1.1.1.2	172.30.209.4	DNS	Standard query SRV _sip._udp.pcelban0001.avayalincroft.globalipcom.com
36	71.064942	172.30.209.4	1.1.1.2	DNS	Standard query response SRV 100 50 5071 pc-n0001-elba.avayalincroft.globalipcom.com
37	71.065841	1.1.1.2	172.30.209.4	DNS	Standard query A pc-n0001-elba.avayalincroft.globalipcom.com
38	71.223877	172.30.209.4	1.1.1.2	DNS	Standard query response A 172.30.209.21

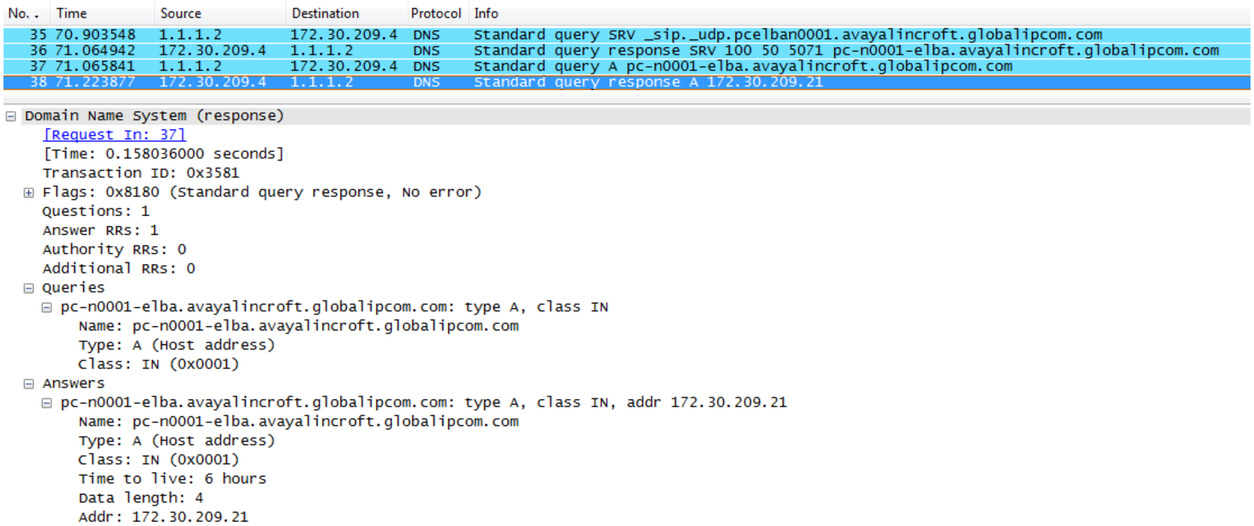
```
⊞ User Datagram Protocol, Src Port: domain (53), Dst Port: domain (53)
⊟ Domain Name System (response)
    [Request In: 35]
    [Time: 0.161394000 seconds]
    Transaction ID: 0x3580
  ⊞ Flags: 0x8180 (Standard query response, No error)
    Questions: 1
    Answer RRs: 1
    Authority RRs: 0
    Additional RRs: 0
  ⊟ Queries
    ⊟ _sip._udp.pcelban0001.avayalincroft.globalipcom.com: type SRV, class IN
        Name: _sip._udp.pcelban0001.avayalincroft.globalipcom.com
        Type: SRV (Service location)
        Class: IN (0x0001)
  ⊟ Answers
    ⊟ _sip._udp.pcelban0001.avayalincroft.globalipcom.com: type SRV, class IN, priority 100, weight 50, port 5071, target pc-n0001-elba.a
        Name: _sip._udp.pcelban0001.avayalincroft.globalipcom.com
        Type: SRV (Service location)
        Class: IN (0x0001)
        Time to live: 6 hours
        Data length: 22
        Priority: 100
        Weight: 50
        Port: 5071
        Target: pc-n0001-elba.avayalincroft.globalipcom.com
```

Frame 37 is expanded below to illustrate the IP Office DNS A-query to determine the IP Address associated with the name "pc-n0001-elba.avayalincroft.globalipcom.com" (i.e., the "Target" returned by Verizon as shown in the prior screen).

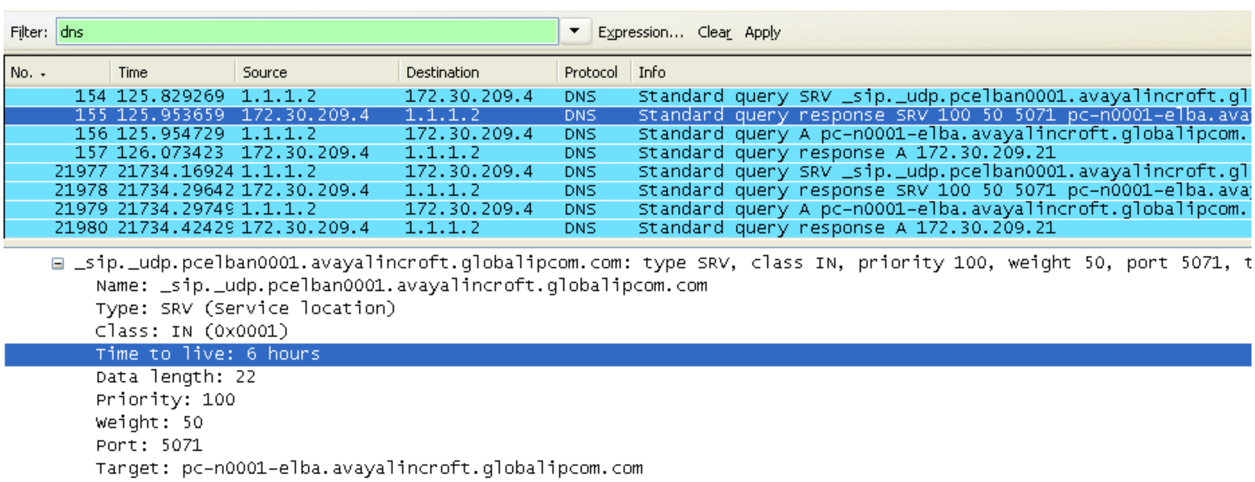
No.	Time	Source	Destination	Protocol	Info
35	70.903548	1.1.1.2	172.30.209.4	DNS	Standard query SRV _sip._udp.pcelban0001.avayalincroft.globalipcom.com
36	71.064942	172.30.209.4	1.1.1.2	DNS	Standard query response SRV 100 50 5071 pc-n0001-elba.avayalincroft.globalipcom.com
37	71.065841	1.1.1.2	172.30.209.4	DNS	Standard query A pc-n0001-elba.avayalincroft.globalipcom.com
38	71.223877	172.30.209.4	1.1.1.2	DNS	Standard query response A 172.30.209.21

```
⊞ User Datagram Protocol, Src Port: domain (53), Dst Port: domain (53)
⊟ Domain Name System (query)
    [Response In: 38]
    Transaction ID: 0x3581
  ⊞ Flags: 0x0100 (Standard query)
    Questions: 1
    Answer RRs: 0
    Authority RRs: 0
    Additional RRs: 0
  ⊟ Queries
    ⊟ pc-n0001-elba.avayalincroft.globalipcom.com: type A, class IN
        Name: pc-n0001-elba.avayalincroft.globalipcom.com
        Type: A (Host address)
        Class: IN (0x0001)
```

Frame 38 is expanded below to illustrate the Verizon "answer" to the IP Office DNS A-query. Note that the IP address returned is 172.30.209.21. IP Office has now determined the IP Address (172.30.209.21) and SIP signaling port (5071) used by Verizon IP Trunk service on the production circuit, without any static provisioning of this information within IP Office.



In the following screen from a different trace left running for more than six hours and filtered to show only dns, frame 155 is highlighted and expanded below to show that the "Time to live" parameter returned by Verizon in the DNS SRV answer is 6 hours. Note that frame 21977, where IP Office sends another DNS SRV query, occurs roughly six hours after frame 155. That is, rounding the times shown in the trace below to seconds, time 21734 minus time 126 equals 21608 seconds. 21608 divided by 3600 seconds per hour = 6.00 hours.



7.3. Verification Summary

See Section 5 of reference [JRR-IPOR6] for detailed illustrations of the verification of representative calls using the IP Office System Status application, the IP Office System Monitor application, and Wireshark. The verification testing from reference [JRR-IPOR6] included the following successful SIP trunk interoperability compliance testing:

- 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 abandoned a call before answer for both inbound and outbound calls.
- Proper disconnect when the IP Office party or the PSTN party terminated an active call.
- Proper busy tone heard when an IP Office user called a busy PSTN user, or a PSTN user called a busy IP Office user (i.e., if no redirection was 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 were 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 was 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 were able to 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 was exercised when the primary SIP line was not responding. The Line Group associated with the Verizon Business SIP Line was 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 (RFC 2833) with successful voice mail navigation using 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 announced the call and prompted the user to enter the "#" key to accept the call. The user had 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.
- Inbound calls from Verizon IP Trunk Service that were call forwarded back to PSTN destinations via Verizon IP Trunk Service.
- 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.

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 can be successfully combined with a Verizon Business IP Trunk SIP trunk service connection to create an end-to-end SIP Telephony business solution. By following the example configurations provided in this document, customers using Avaya IP Office can 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 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.

As noted in the Introduction, the focus of this version of the IP Office with Verizon IP Trunk Application Notes is the use of two newly configurable IP Office Release 6.1 SIP trunk features: REFER-based transfer and DNS-SRV with Verizon IP Trunk Service. Although these new features have not been independently certified by Verizon labs, these Application Notes may be used to facilitate field trial testing of either or both of these new capabilities of IP Office Release 6.1 with Verizon IP Trunk service.

9. 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 6.1 Installation Manual, Issue 22g, November 17 2010
Document Number 15-601042
https://support.avaya.com/css/P8/documents/100119958

[IPO-MGR] IP Office Release 6.1 Manager 8.1, Issue 25i, November 23, 2010
Document Number 15-601011
https://support.avaya.com/css/P8/documents/100119917

[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 6.1 Voicemail Pro Installation and Maintenance, Issue 23c, November 5, 2010
Document Number 15-601063
https://support.avaya.com/css/P8/documents/100119901

[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/

The following Application Notes formed the base configuration for this document. The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 6.0.
[JRR-IPOR6] Application Notes for Configuring SIP Trunk 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

The following Application Notes provide the configuration that will enable IP Office Release 6.1 to perform like IP Office Release 6.0. That is, in reference [JRR-IPOR61], the newly configurable SIP features of IP Office Release 6.1, such as SIP REFER and DNS SRV, are disabled. 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 Trunk SIP Trunk Service Offer and Avaya IP Office Release 6.1, Issue 1.0

[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.