Avaya Solution & Interoperability Test Lab

Application Notes for SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service and Avaya IP Office Release 8.1 – Issue 1.0

Abstract

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Verizon Business IP 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 v2 Release 8.1 Essential Edition, Embedded Voicemail in Intuity Mode, Avaya IP Office Softphone, and Avaya H.323, SIP, 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 v2 Release 8.1 Essential Edition, Avaya Embedded Voicemail in Intuity Mode, Avaya IP Office Softphone, and Avaya H.323, SIP, 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. With the market growth of SIP trunk deployments in the SME segment, importing and using SIP trunk templates to reduce installation time and errors associated with programming, will become increasingly valuable to installers working with R8.1. See Appendix A for the Template used in this configuration.

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/

2. 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.

Testing was successful. Any limitations related to the overall configuration are noted in Section 2.2.

2.1. Interoperability Compliance Testing

The verification testing 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 SIP telephones, Avaya H.323 telephones, Avaya digital telephones, analog telephones, analog fax machines, Avaya IP Office Softphone, and Avaya IP Office Embedded 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 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 SIP phones, 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.
- Inbound / Outbound fax using G711 and T38 were 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 Embedded Voicemail.
- 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, presenting true calling party information to the PSTN phone, 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.
- Mobility Features such as Mobile Callback and Mobile Call Control

2.2. 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. **FAX:** A SIP Line on IP Office Release 8.1 can be configured to support T.38 fax or fax over G.711. T38 is a new offer from Verizon Business IP Trunk service and requires that the **Disable T30 ECM** be checked on the **SIP Line→T38 Fax** page as indicated in Section 5.4.4. During compliance testing, there were greater than expected fax failure rates when using Verizon’s IWSPM Media Gateway. Other Verizon media gateways used during testing were within the allowable threshold for fax failures. If the Verizon IWSPM Media Gateway is used, a separate analog POTS line is recommended for fax transmissions. Also, Verizon Business IP Trunk service will not perform the expected re-invite to T38 on an outbound fax, but instead will wait and expect IP Office to issue the re-invite to T38. Once the re-invite is issued, Verizon will send a 200 OK to acknowledge the T38. This will be transparent to the user.

2. **HOLD**: 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. **CODEC MISMATCH**: If there is not a matching codec configured on the **SIP Line → VoIP** tab to match the service provider, on placing a call the user will briefly hear ring back and then the phone will display **Number Busy**.

4. **SIP PHONE TRANSFER:** When the IP Office transferor of an outbound call to the PSTN via Verizon is a SIP device registered to IP Office (e.g., Avaya 1140E, Avaya 1220, or IP Softphone in the sample configuration), and the REFER transfer option is enabled on the SIP Line to Verizon, the transferor may briefly see the display “Transfer failed” after the final user operation, even if the transfer has actually succeeded. On the production circuit used for testing, Verizon did not send NOTIFY messages to IP Office to signal transfer completion. Internal tracking issue IPOFFICE-35823 has been created for this issue.

5. **OFF-NET TRANSFER with REFER**: When an IP Office extension tries to blind transfer a call from a PSTN extension to a PSTN extension, the transfer will complete but will not use a REFER, instead the re-invite method will be used. This is important to note because DSP resources will still be needed for the call that was transferred. This issue is under investigation and internal tracking issue IPOFFICE-31274 has been created with a fix included in Version 8.1.56.

6. **DNS-SRV:** Although Avaya IP Office supports DNS-SRV to a Verizon DNS server as verified in Section 7.2, IP Office does not automatically fail over outbound calls to alternate Verizon SIP destinations if the Verizon DNS returns multiple answers, and the first listed response is unavailable. This anomaly is under investigation by the IP Office product team as IPOFFICE-34076.

7. **Short Duration DTMF:** When interworking with Verizon media gateways that use the VSP3 DSPs, outbound short DTMF digit intervals from IP Office play out at 50ms. Other DSP types play out proper durations. Although Avaya IP Office complies with RFC 2833, the VSP3 requires the DTMF events to have duration field values of at least 20ms and the IP Office has been observed to have values lower than this requirement. This may result in some IVR applications not recognizing DTMF digits.

8. **Echo Cancellation:** Avaya IP Office is designed to bypass the echo canceller when a CED tone is detected to support data transmissions that require no echo cancellation. However, Group 3 facsimile and certain low speed voiceband data transmissions are adversely affected if echo cancelation is disabled. To prevent false bypass of the echo canceller IP Office should re-enable the echo caneller after a silence period of about 250ms. It was observed during testing that the IP Office bypassed the echo canceller after the CED tone, but did not re-enable it after the aforementioned silence period. This anomaly is under investigation by the IP Office product team as IPOFFICE-34077.

2.2.1. Avaya

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

2.2.2. Verizon

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

3. 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.

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.

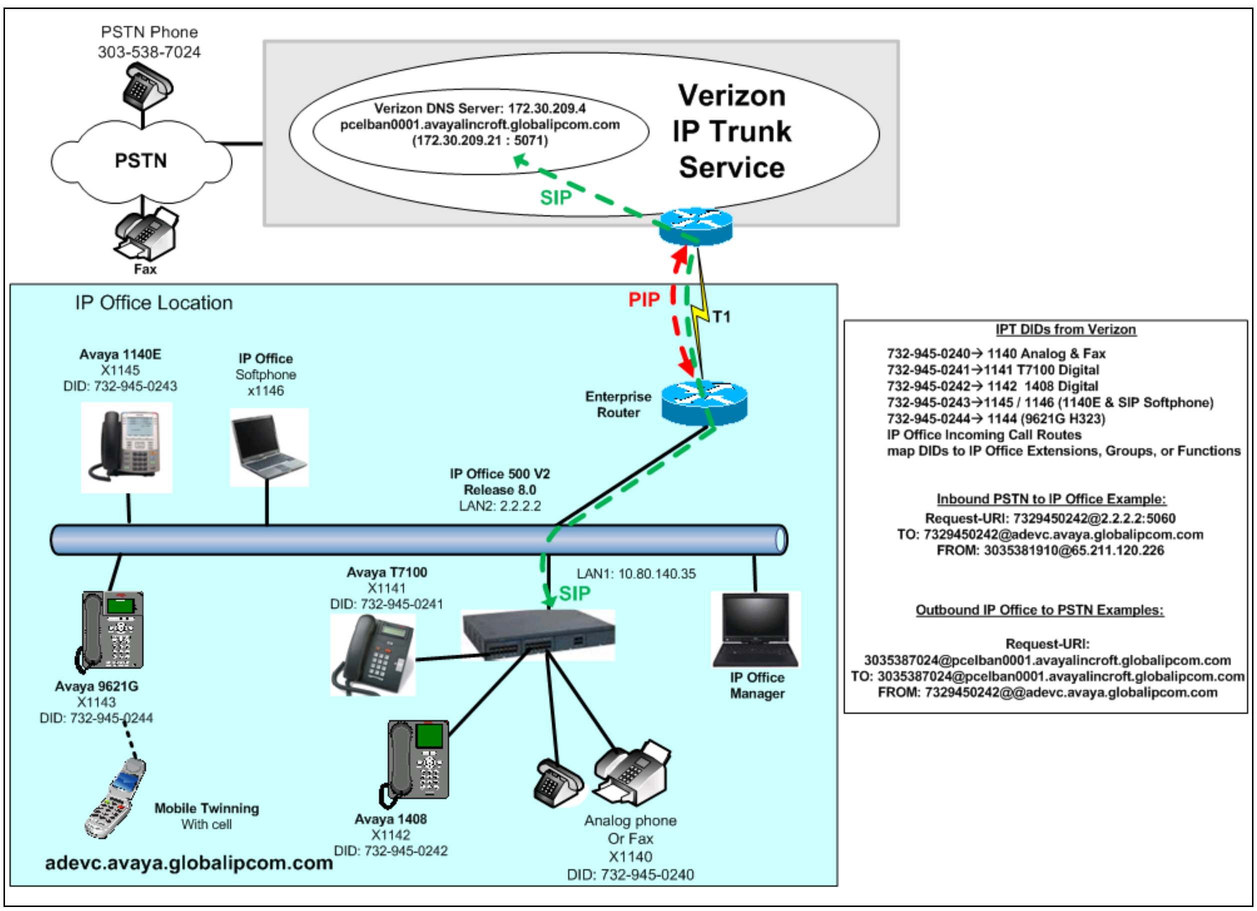


Figure 1: Avaya IP Office with Verizon IP Trunk SIP Trunk Service

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 5. Selected verifications are illustrated in Section 7.

Verizon Provided DID	Avaya IP Office Destination	Notes
732-945-0240	X 1140	Analog telephone or Fax machine
732-945-0241	X 1141	T7100 Digital Telephone
732-945-0242	X 1142	1408 Digital Telephone
732-945-0244	X 1144	Avaya H.323 - 9621G
732-945-0243	Hunt Group x 1145 & x1146	Avaya SIP 1140E & Avaya IP Office Softphone

Table 1: Verizon DID to IP Office Mappings

4. Equipment and Software Validated

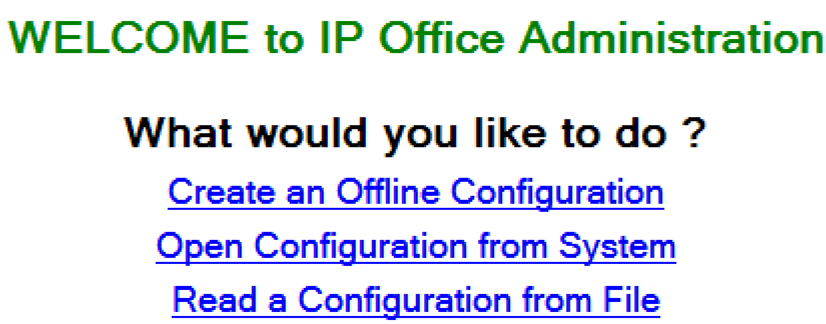
Table 2 shows the equipment and software used in the sample configuration.

Equipment	Software
Avaya IP Office 500 v2	Release 8.1 (43)
Avaya IP Office Manager	Release 10.1 (43)
Avaya 2500 Analog Telephone	N/A
Avaya 1408 Digital Telephone	N/A
Avaya T7100 Digital Telephone	N/A
Avaya 1600-Series Telephones (H.323)	Release 1.300B
Avaya 1140E SIP	04.03.09
Avaya IP Office Softphone	Release 3.2.3.15 64595
Okidata 2450 (analog fax)	N/A

Table 2: Equipment and Software Tested

5. Avaya IP Office Configuration

IP Office is configured via the IP Office Manager program. For more information on IP Office Manager, consult reference [2]. From the IP Office Manager PC, select **Start** → **Programs** → **IP Office** → **Manager** to launch the Manager application. A screen that includes the following in the center may be displayed:



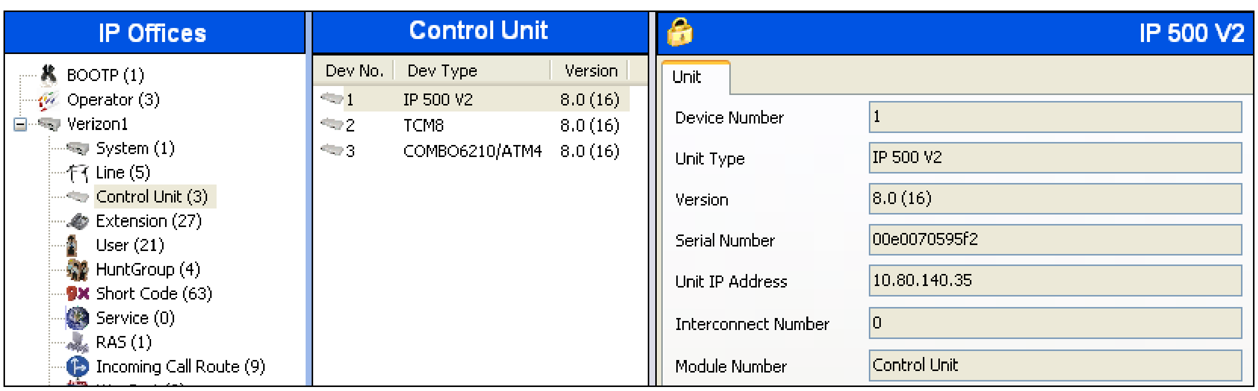
Open the IP Office configuration, either by reading the configuration from the IP Office server, or from file. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left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

5.1. Physical, Network, and Security Configuration

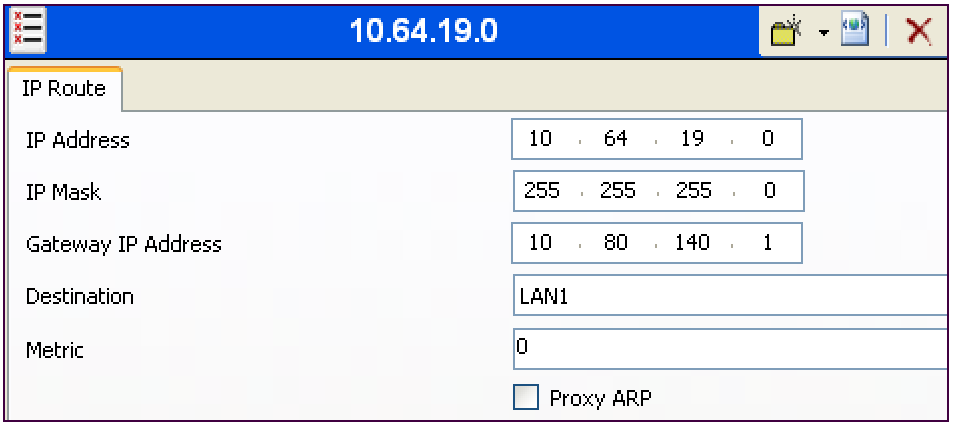
This section describes attributes of the sample configuration, but is not meant to be prescriptive. Consult reference [1] for more information on the topics in this section.

In the sample configuration, looking at the IP Office 500 from left to right, the first module is a TCM8 Digital Station module, the second card is a COMBO6210/ATM4 module and the third and fourth slots are blank. The TCM8 is used to add TCM RJ45 extension ports to an IP500 V2 control unit. It provides 8 RJ45 extension ports for supported M-Series and T-Series digital stations. It can also be used for 4100 and 7400 Series phone support by connection to a Digital Mobility Solution system. The COMBO6210/ATM4 is used to add a combination of ports to an IP500 V2 control unit and is not supported by IP500 control units. The module supports 10 voice compression channels. Codec support is G.711, G729a and G.723 with 64ms echo cancellation. G.722 is supported by IP Office Release 8.0 and higher. The “Combo” card will support 6 Digital Station ports for digital stations in slots 1-6 (except 3800, 4100, 4400, 7400, M and T-Series), 2 Analog Extension ports in slots 7-8, and 4 Analog Trunk ports in slots 9-12. Referring to **Figure 1**, the Avaya T7100 telephone with extension 1141 is connected to port 1 of the TCM8 module, and the Avaya 1408 telephone with extension 1142 is connected to port 1 of the “Combo” card. The analog extension or fax machine is connected to the “Combo” card on port 7.

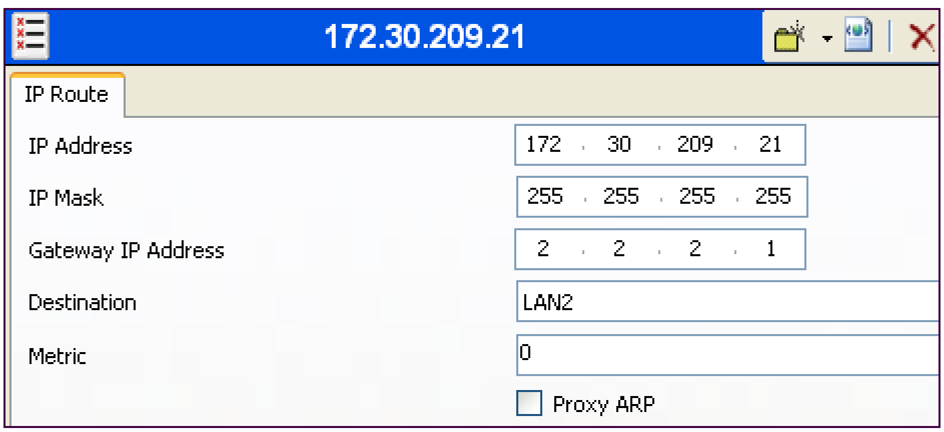
The following screen shows the modules in the IP Office used in the sample configuration. To access such a screen, select **Control Unit** in the Navigation pane. The modules appear in the Group pane. In the screen below, **IP 500 V2** is selected in the Group pane, revealing additional information about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



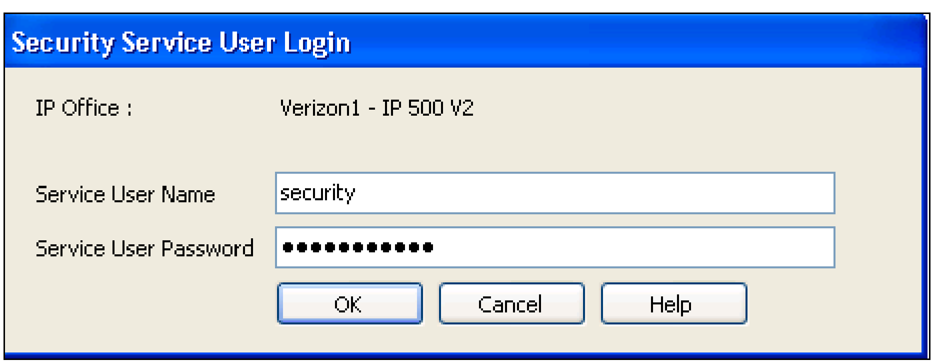
In the sample configuration, the IP Office LAN1 port is physically connected to the local area network switch at the IP Office customer site. The default gateway for this network is 10.80.140.1. To add an IP Route in IP Office, right-click **IP Route** from the Navigation pane, and select **New**. To view or edit an existing route, select **IP Route** from the Navigation pane, and select the appropriate route from the Group pane. The following screen shows the Details pane with the relevant route using **Destination** LAN1.



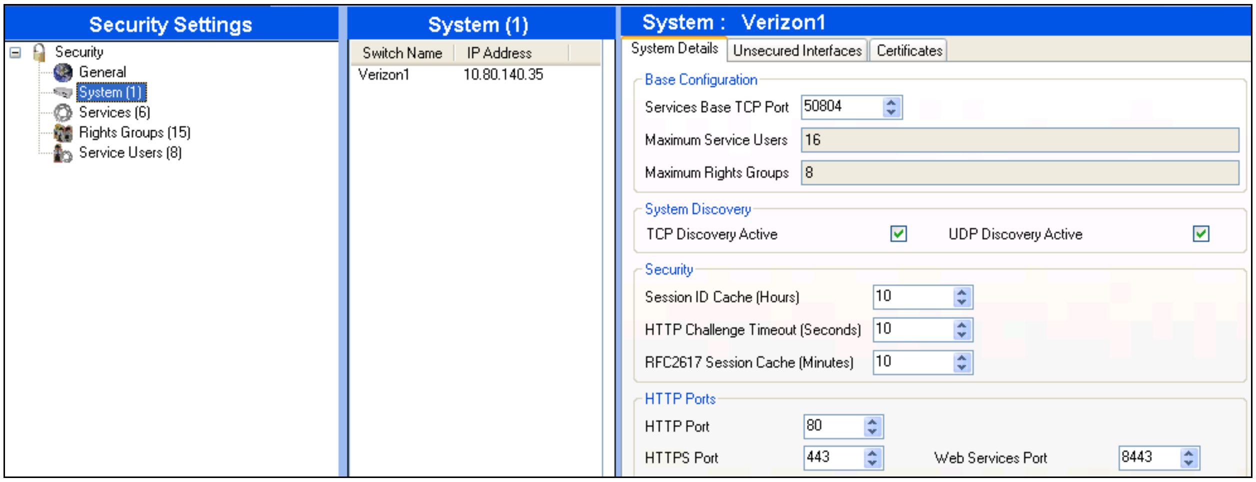
The IP Office LAN2 port is physically connected to the service provider and has a default gateway of 2.2.2.1. Right-click **IP Route** from the Navigation pane, and select **New** to add another route. The following screen shows the Details pane with the relevant route using **Destination** LAN2.



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



After logging in, select **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 the **HTTPS Port** is configured as intended, as shown below.

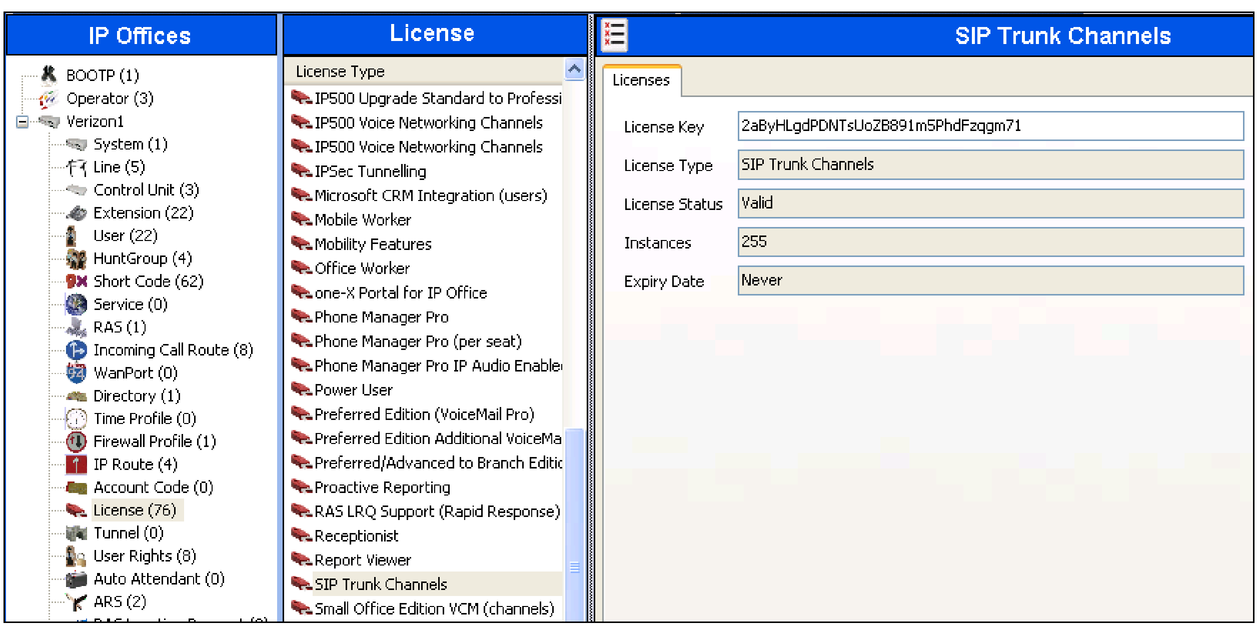


When complete, select **File → Configuration** to return to configuration activities.

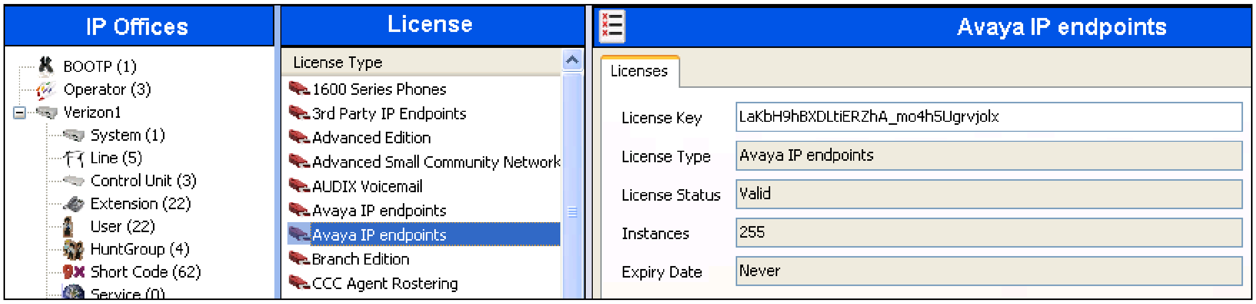
5.2. Licensing

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

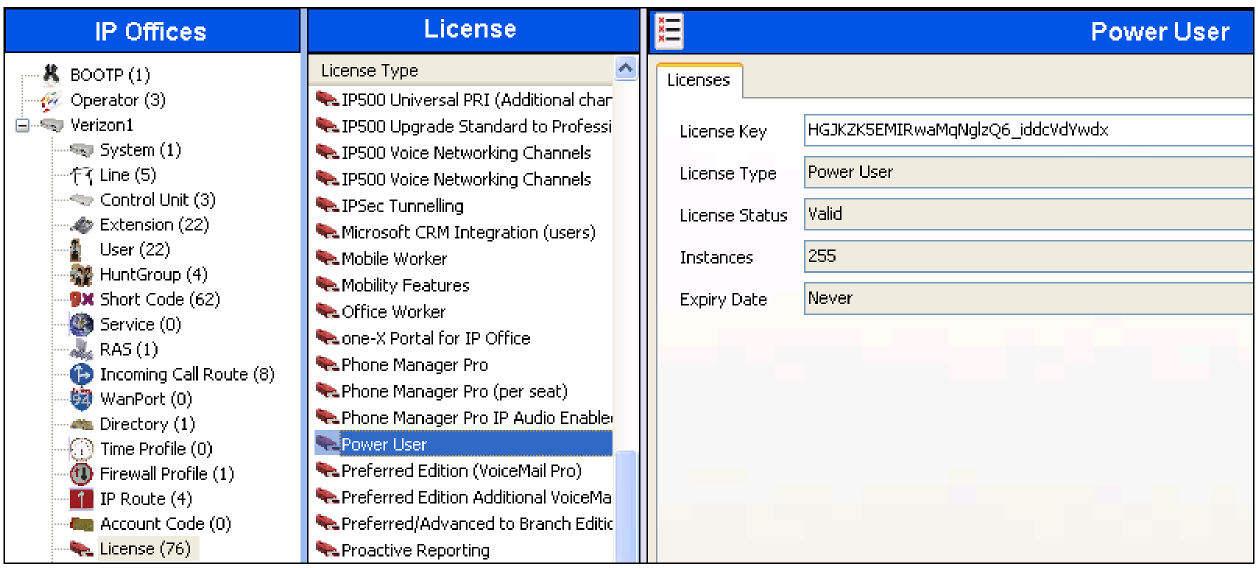
To verify that there is a SIP Trunk Channels License with sufficient capacity, click **License** in the Navigation pane and **SIP Trunk Channels** in the Group pane. Confirm a valid license with sufficient "Instances" (trunk channels) in the Details pane.



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



The following screen shows the availability of a valid license for **Power User** features. In the sample configuration, the user with extension 1146 will be configured as a "Power User" and will be capable of using the Avaya IP Office Softphone.

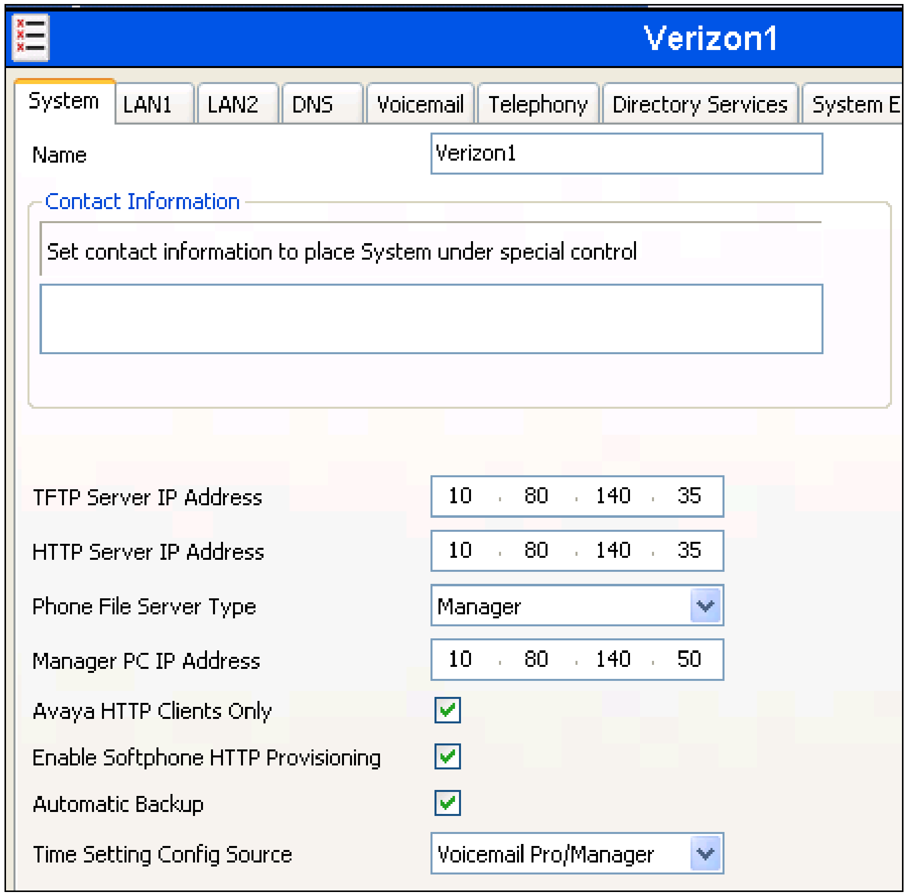


5.3. System Settings

This section illustrates the configuration of system settings. Select **System** in the Navigation pane to configure these settings. The subsection order corresponds to a left to right navigation of the tabs in the Details pane for System settings.

5.3.1. System Tab

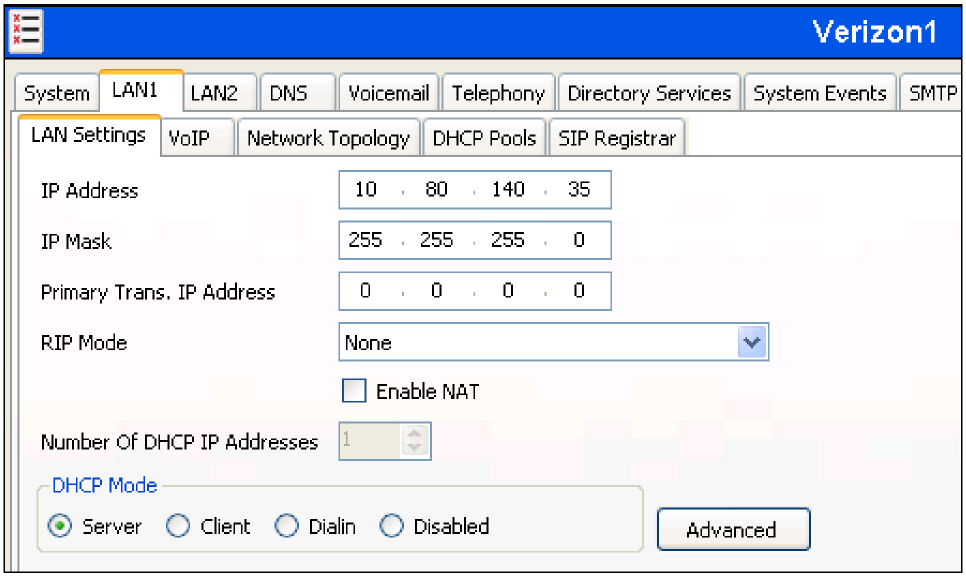
With the proper system name selected in the Group pane, select the **System** tab in the Details pane. The following screen shows a portion of the **System** tab. The **Name** field can be used for a descriptive name of the system. In this case, Verizon1 is used as the name. The **Avaya HTTP Clients Only** and **Enable SoftPhone HTTP Provisioning** boxes are checked to facilitate Avaya IP Office Softphone usage.



5.3.2. LAN 1 Settings

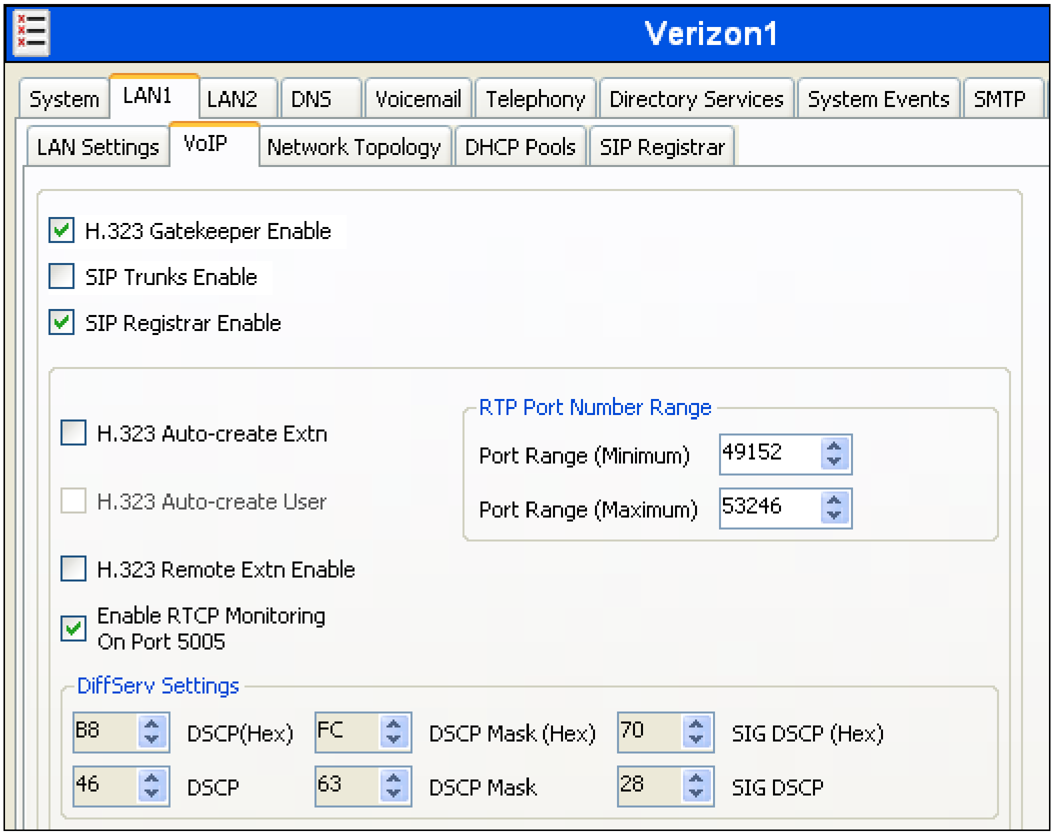
The IP500/IP500 V2 control units have 2 RJ45 Ethernet ports, physically marked as LAN and WAN. These form a full-duplex managed layer-3 switch. Within the system configuration, the physical LAN port is LAN1, the physical WAN port is LAN2.

In the sample configuration, LAN1 was used to connect the IP Office to the enterprise network. To view or configure the **IP Address** of LAN1, select the **LAN1** tab followed by the **LAN Settings** tab. As shown in **Figure 1**, the IP Address of the IP Office is 10.80.140.35. **DHCP Mode** is also set to **Server** so that IP phones will get an IP Address from the IP Office Server. Other parameters on this screen may be set according to customer requirements.

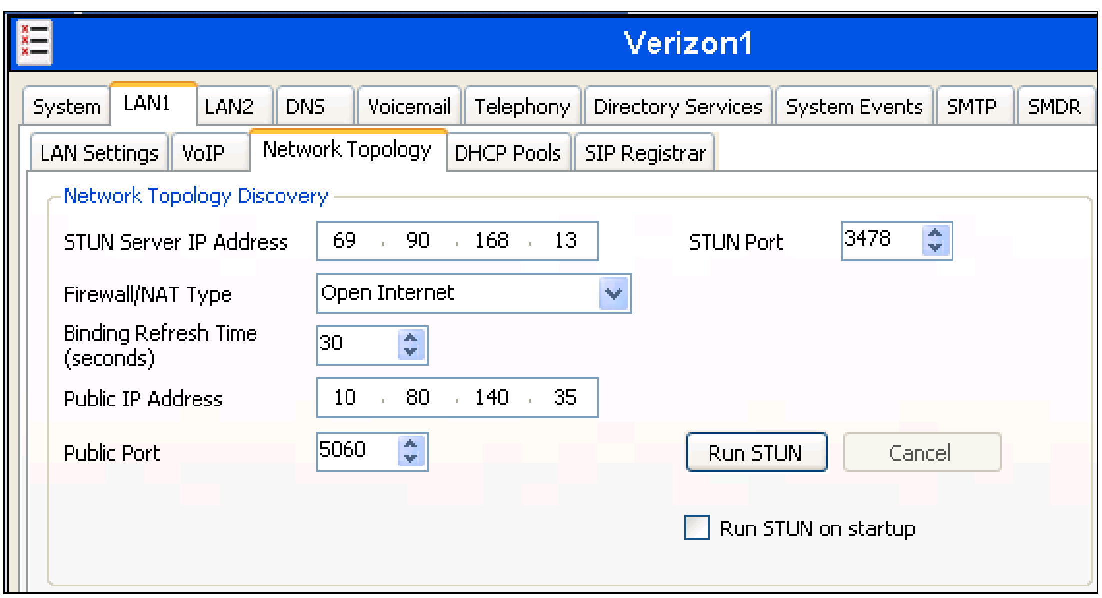


Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 1600-Series Telephones used in the sample configuration. The **SIP Registrar Enable** box is checked to allow Avaya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

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

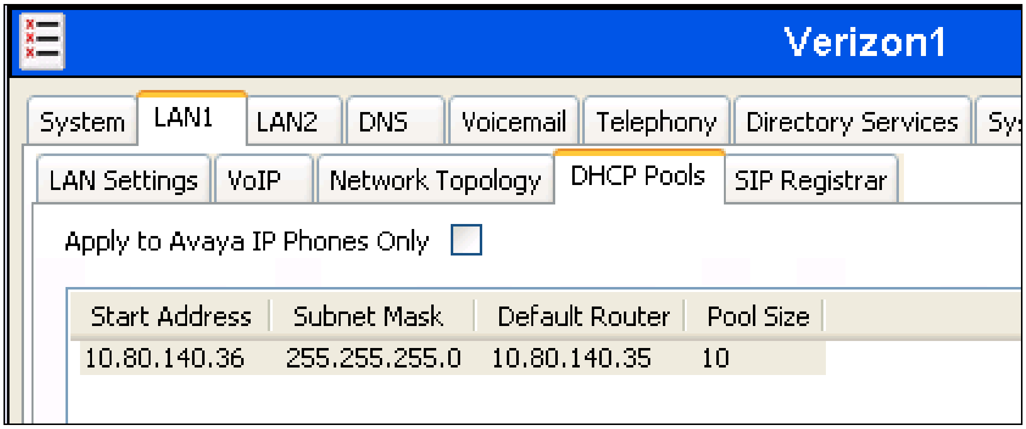


Select the **Network Topology** tab as shown in the following screen. For **Public IP Address**, enter the Avaya IP Office LAN1 IP address. Set the **Public Port** to 5060. In the sample configuration, the **Firewall/NAT Type** is set to "Open Internet". With this configuration, STUN will not be used. Click the **OK** button.

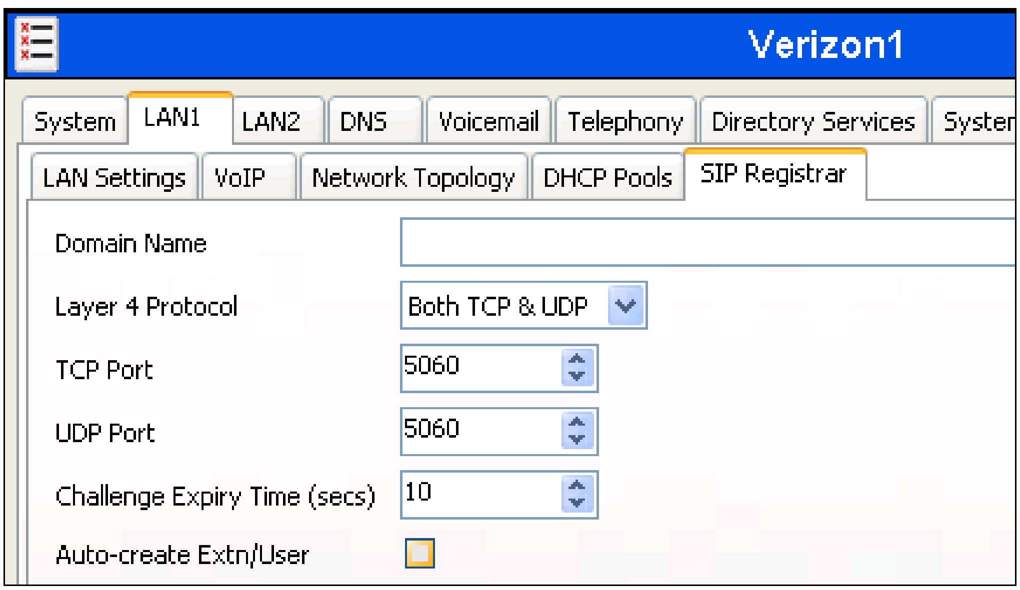


Note: The **Firewall/NAT Type** parameter may need to be different, depending on the type of firewall or Network Address Translation device used at the customer premise.

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



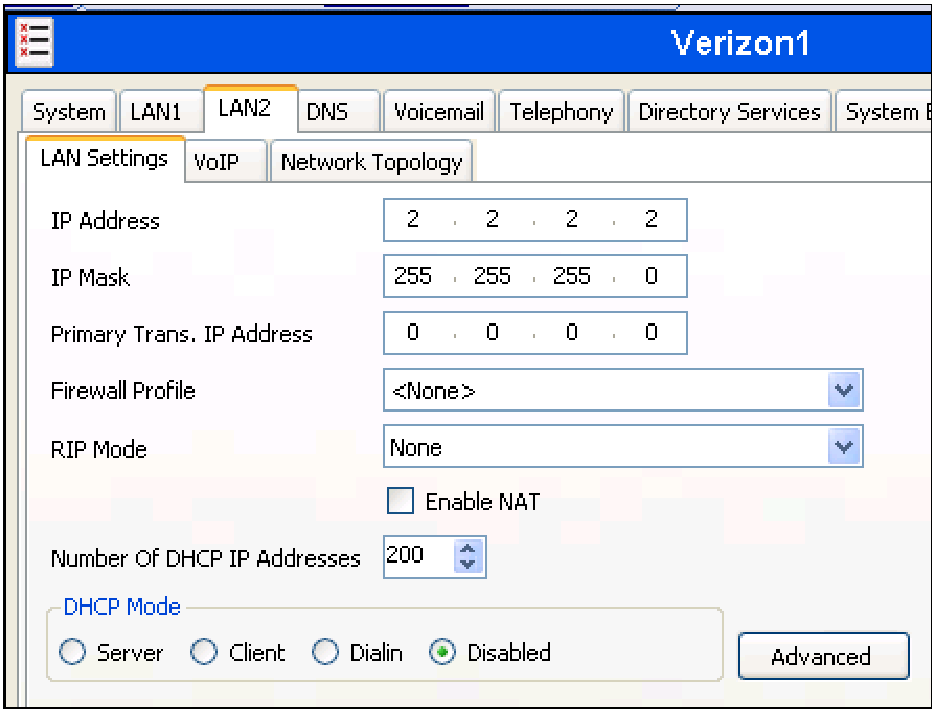
Optionally, select the **SIP Registrar** tab. The following screen shows the settings used in the sample configuration.



5.3.3. LAN 2 Settings

In the sample configuration, LAN2 was used to connect the IP Office to the Verizon network. To view or configure the **IP Address** of LAN2, select the **LAN2** tab followed by the **LAN Settings** tab. As shown in **Figure 1**, the IP Address of the IP Office, known to Verizon, is 2.2.2.2. **DHCP**

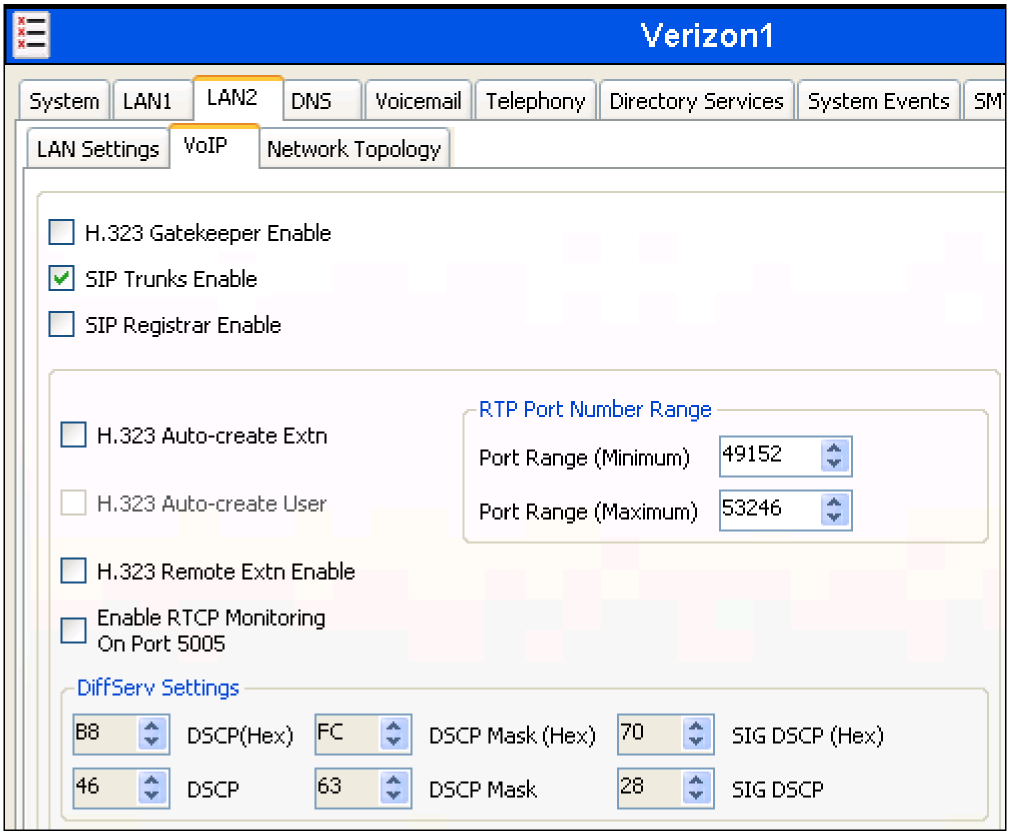
Mode is set to **Disabled** since DHCP is unnecessary towards Verizon. Other parameters on this screen may be set according to customer requirements.



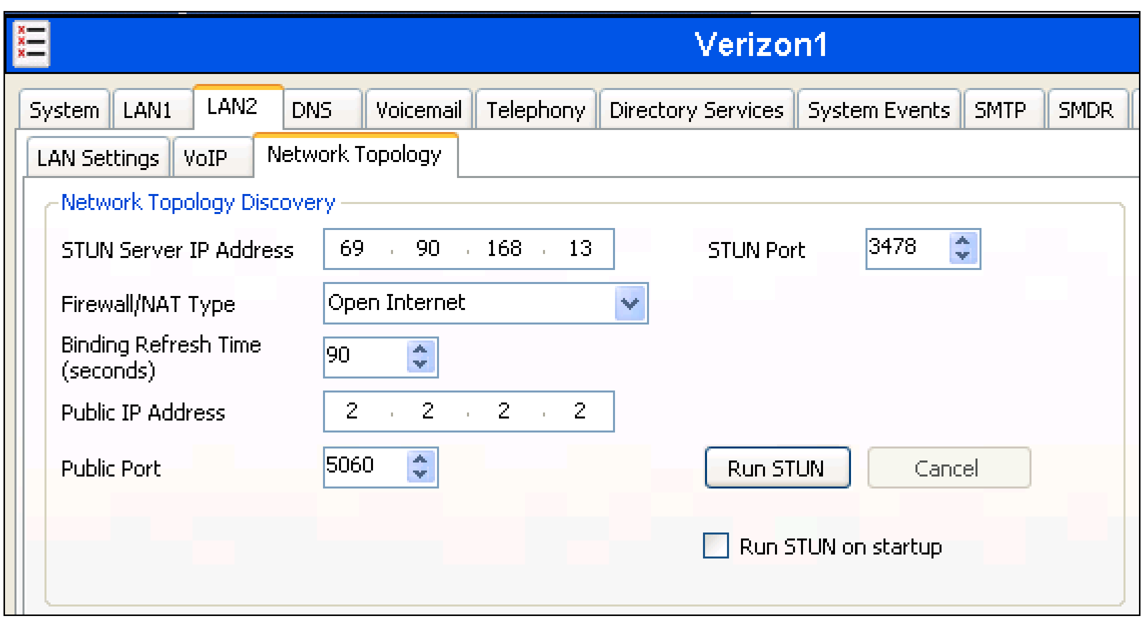
Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** and **SIP Registrar Enable** boxes are unchecked since IP telephones will not be registering on this link. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Verizon Business.

If desired, the **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media paths from Verizon Business to IP Office. The defaults are used here. See Section 5.3.2 for more information on these RTP settings.

If desired, IP Office can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Service policies. In the sample configuration shown below, IP Office will mark SIP signaling with a value associated with "Assured Forwarding" using DSCP decimal 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.



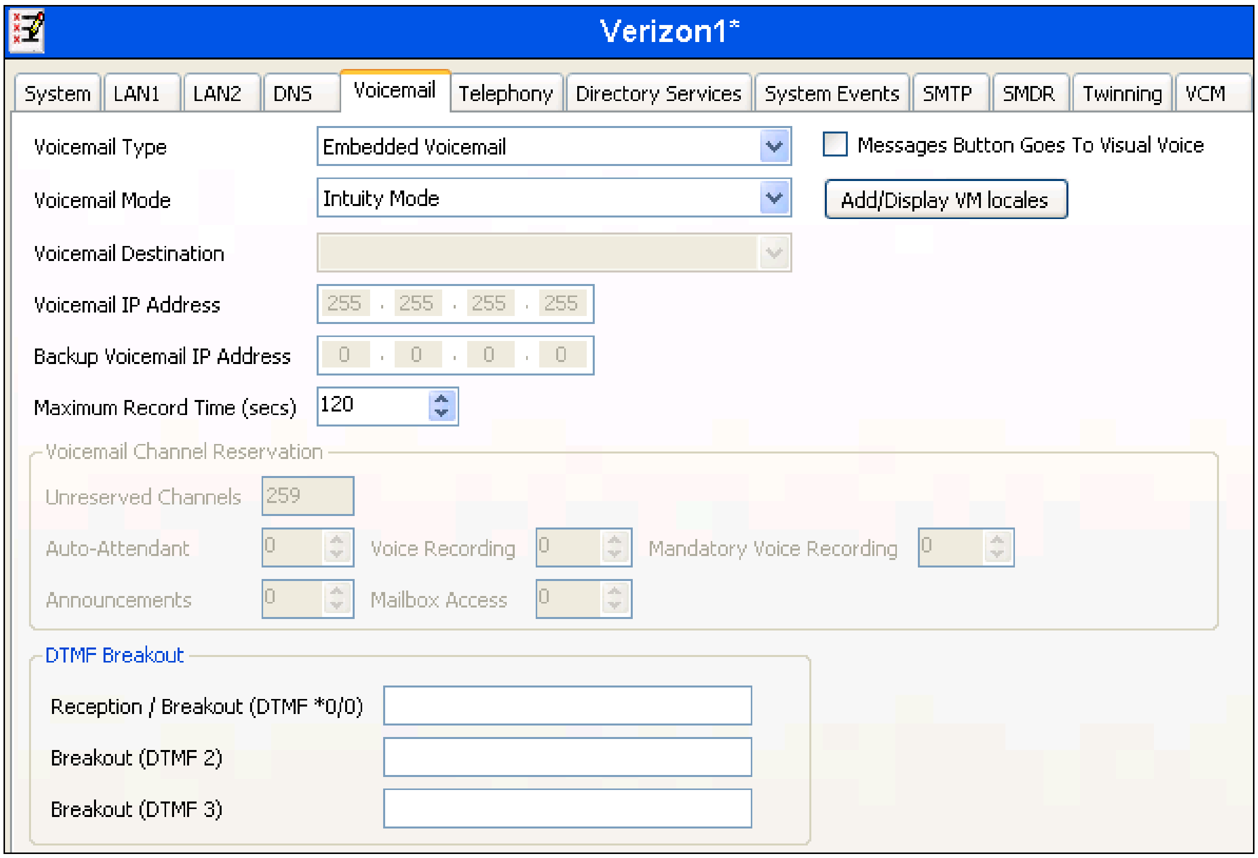
Select the **Network Topology** tab as shown in the following screen. The **Binding Refresh Time** can be configured to vary SIP OPTIONS timing. For **Public IP Address**, enter the Avaya IP Office LAN2 IP address. Set the **Public Port** to 5060.



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

5.3.4. Voicemail

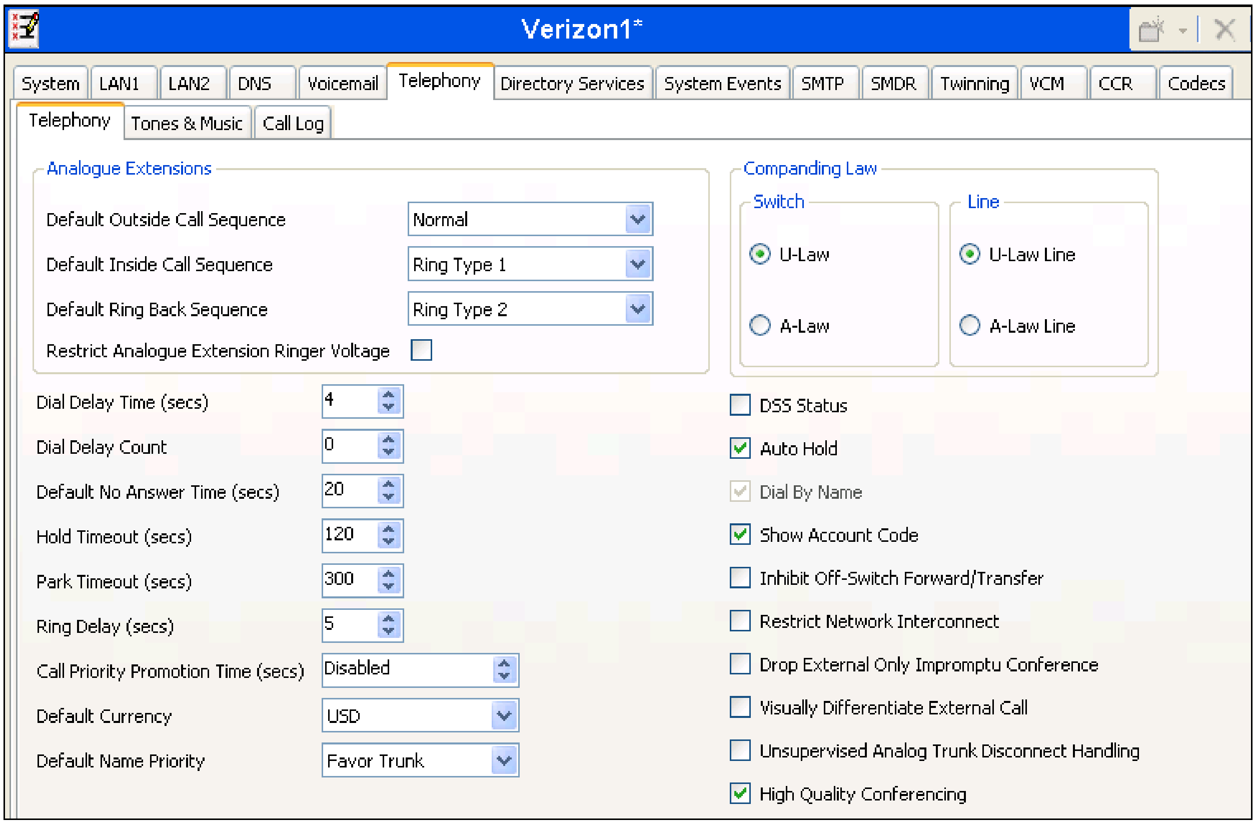
To view or change voicemail settings, select the **Voicemail** tab as shown in the following screen. The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. The **Voicemail Type** in the sample configuration is "Embedded Voicemail" in "Intuity Mode". Other Voicemail types may be used. Other parameters on this screen may be set according to customer requirements.



5.3.5. System Telephony Configuration

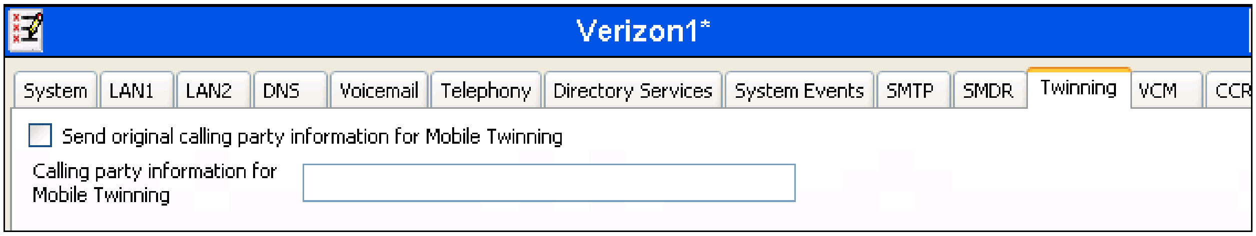
To view or change telephony settings, select the **Telephony** tab and **Telephony** sub-tab as shown in the following screen. The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. In the sample configuration, the **Inhibit Off-Switch Forward/Transfer** box is unchecked so that call forwarding and call transfer to PSTN destinations via the Verizon Business IP Trunk service can be tested. That is, a call can arrive to IP Office via the Verizon Business IP Trunk, and be forwarded or transferred back to the PSTN with the outbound leg of the call using the Verizon IP Trunk service. The **Companding Law** parameters are set to "ULAW" as is typical in North American locales. Other parameters on this screen may be set according to customer requirements.

The **Default Name Priority** is a new field in IP Office Release 8 and can be relevant to SIP Trunking. The option to "Favor Trunk" or "Favor Directory" can be set system-wide using the screen below, or set uniquely for each line. With the option to "Favor Directory", IP Office will prefer to display names found in a personal or system directory over those arriving from the far-end, if there is a directory match to the caller ID. This capability will be illustrated further in the context of the SIP Line to Verizon. A user's personal directory example is shown in Section 5.5.2.



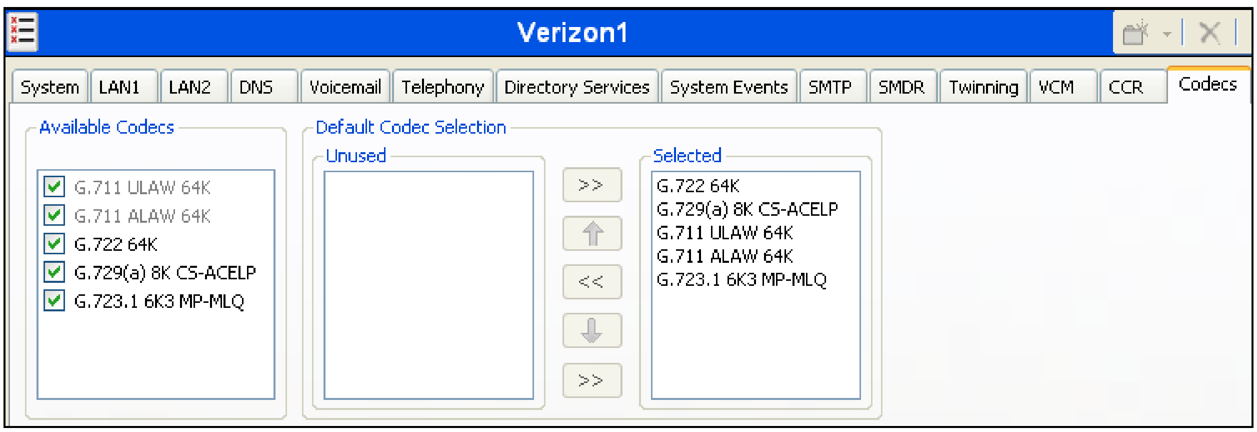
5.3.6. System Twinning Configuration

To view or change Twinning settings, select the **Twinning** tab as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning** is left blank. With this configuration, and related configuration of "Diversion header" on the SIP Line (Section 5.4), 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.



5.3.7. System Codecs Configuration (New in IP Office Release 8)

The **System → Codecs** tab is new in IP Office Release 8. On the left, observe the list of **Available Codecs**. In the example screen below, which is not intended to be prescriptive, the box next to each codec is checked, making all the codecs available in other screens where codec configuration may be performed (such as the SIP Line in Section 5.4). The **Default Codec Selection** area enables the codec preference order to be configured on a system-wide basis, using the up, down, left, and right arrows. By default, all IP (SIP and H.323) lines and extensions will assume the system default codec selection, unless configured otherwise for the specific line or extension.



5.4. SIP Line

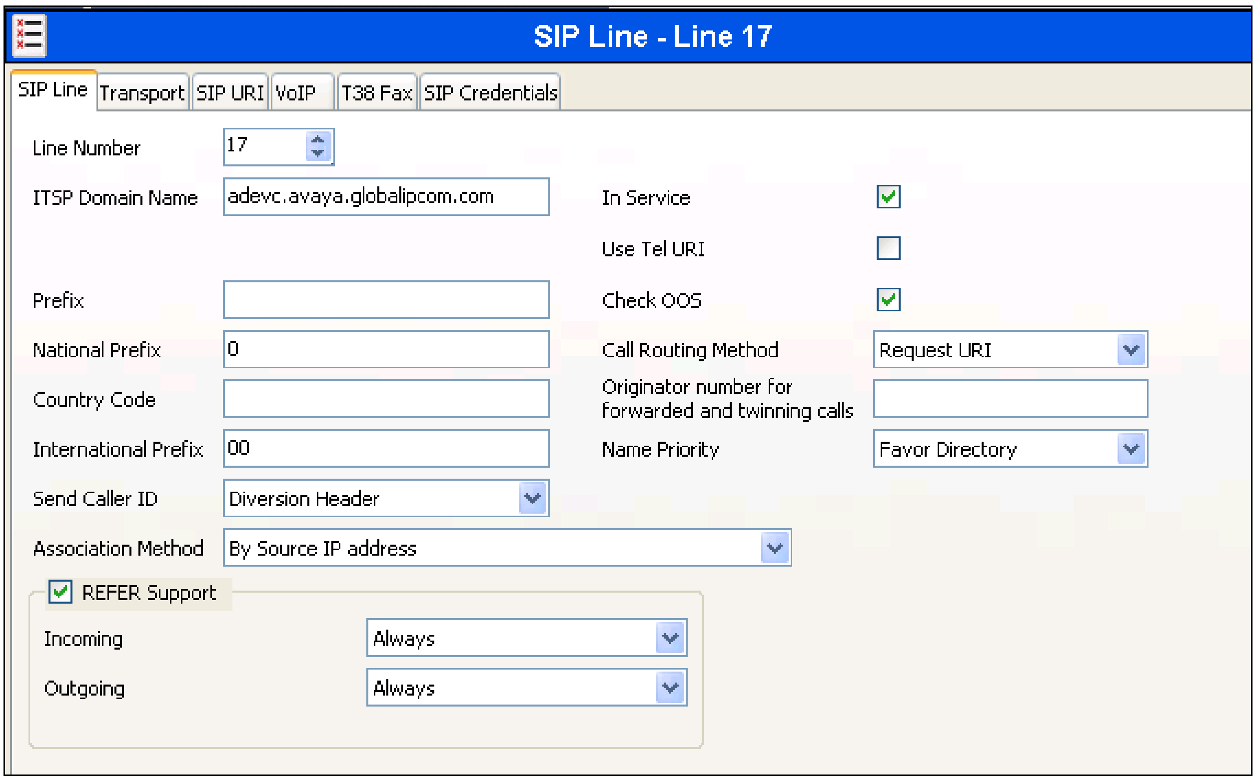
The **SIP Line** tab in the Details pane is shown below for Line Number 17, used for the Verizon Business IP Trunk. The **ITSP Domain Name** can be configured to the IP Office LAN2 address (2.2.2.2) or in this case to the domain supplied by Verizon (advec.avaya.globalipcom.com). By default, the **In Service** and **Check OOS** boxes are checked.

The **Call Routing Method** can retain the default "Request URI" setting, or may be changed to "To Header", to match Incoming Call Routes based on the contents of the "To Header". In the sample configuration, the default "Request URI" setting was used.

The area of the screen entitled **REFER Support** was introduced in IP Office Release 6.1. The default automatic determination of REFER support is "Auto". Alternatively, the default can be overridden with "Never" to explicitly disable use of REFER, or "Always" to explicitly enable use of REFER. The **Association Method** parameter was introduced in IP Office Release 7.0, and the screen below shows the default value, which is sufficient in the sample configuration.

The various alternatives for the **Association Method** may be useful when multiple SIP Trunks with different SIP domains resolve to a single IP Address. The default option associates incoming requests with SIP Lines by comparing the source IP Address and port of the incoming message against the configured far-end of the SIP Line.

The **Name Priority** parameter is new in IP Office Release 8.0. The **Name Priority** parameter can retain the default “System Default” setting, or can be configured to “Favor Trunk” or “Favor Directory” as shown in the sample screen below. “System Default” will use the setting displayed on the System → Telephony→ Telephony Tab. The “Favor Directory” setting enables IP Office to match the caller’s telephone number against available system or personal directories, and display the name obtained from a match in the directory, if any, rather than name information received in the SIP signaling from Verizon. See Section 5.5.2. Click **OK** (not shown).

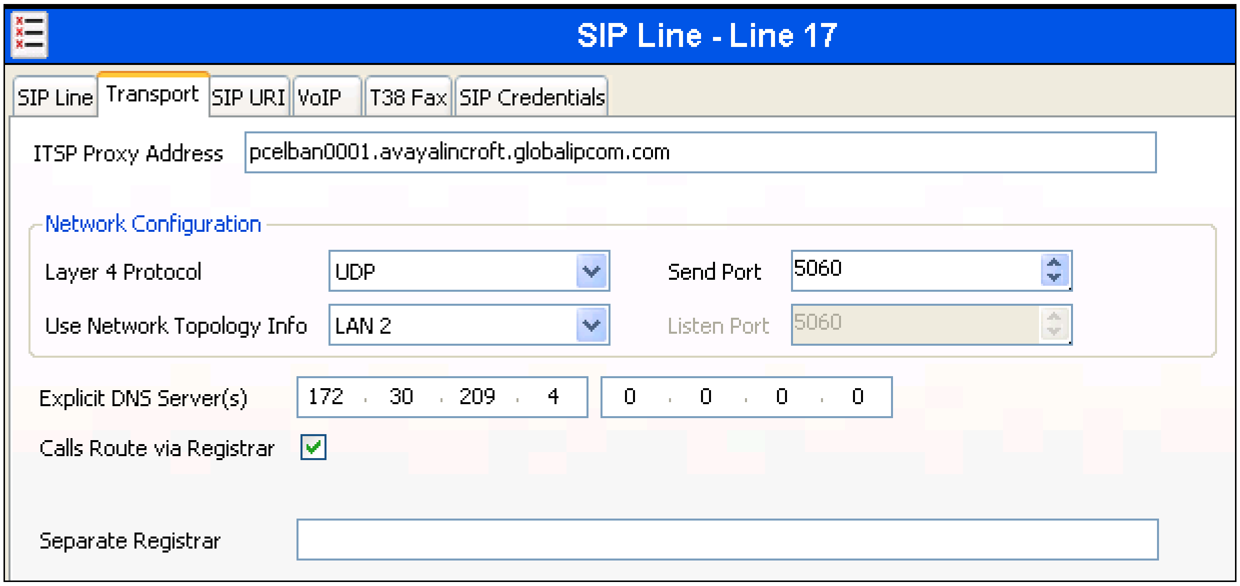


5.4.1. SIP Line - Transport Tab

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

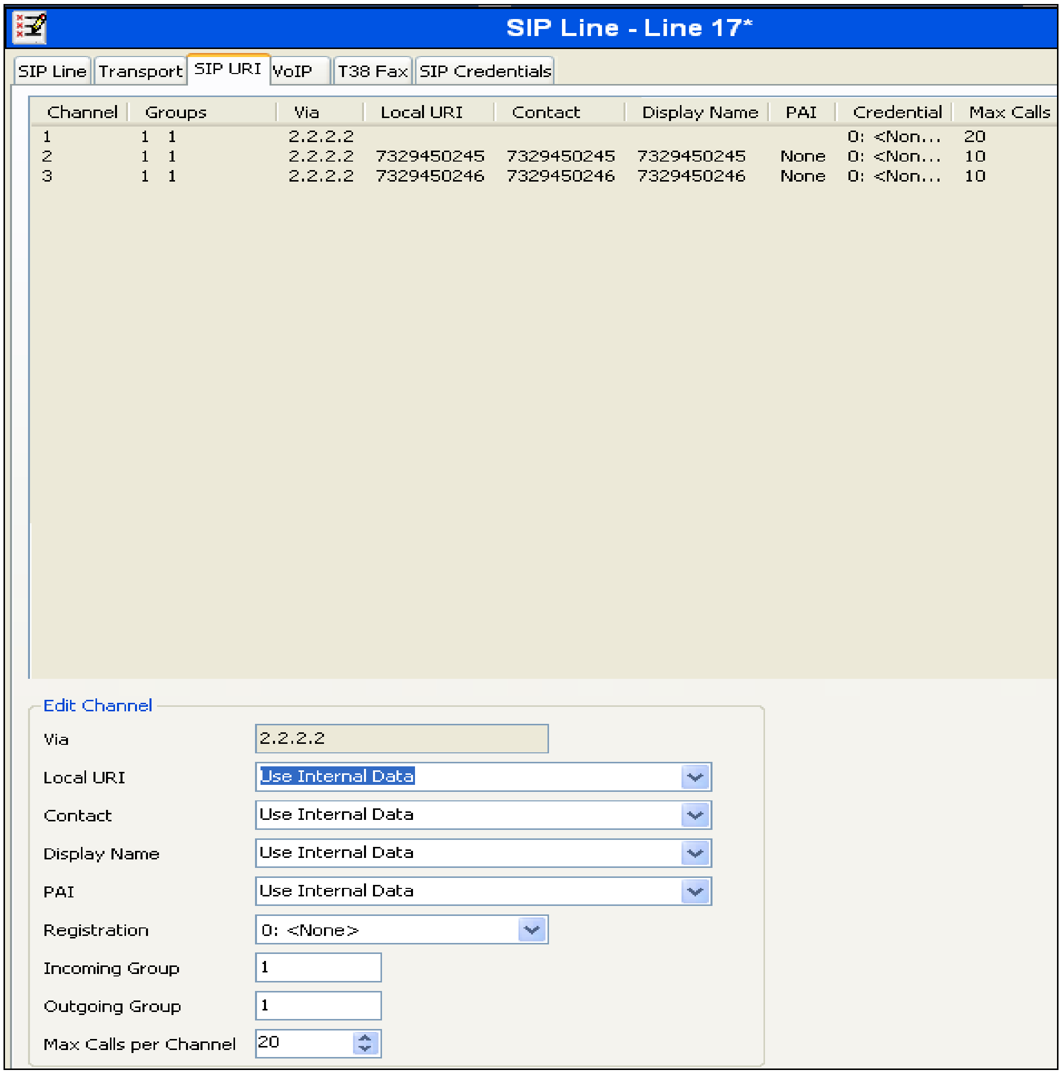
The **ITSP Proxy Address** is set to the Verizon domain provided by Verizon Business. As shown in **Figure 1**, this domain is “pcelban0001.avayalincroft.globalipcom.com”. In the **Network Configuration** area, UDP is selected as the **Layer 4 Protocol**. Since DNS SRV will be used, the **Send Port** 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 2”. This associates the SIP Line with the parameters in the **System → LAN2 → 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.



5.4.2. SIP Line - SIP URI Tab

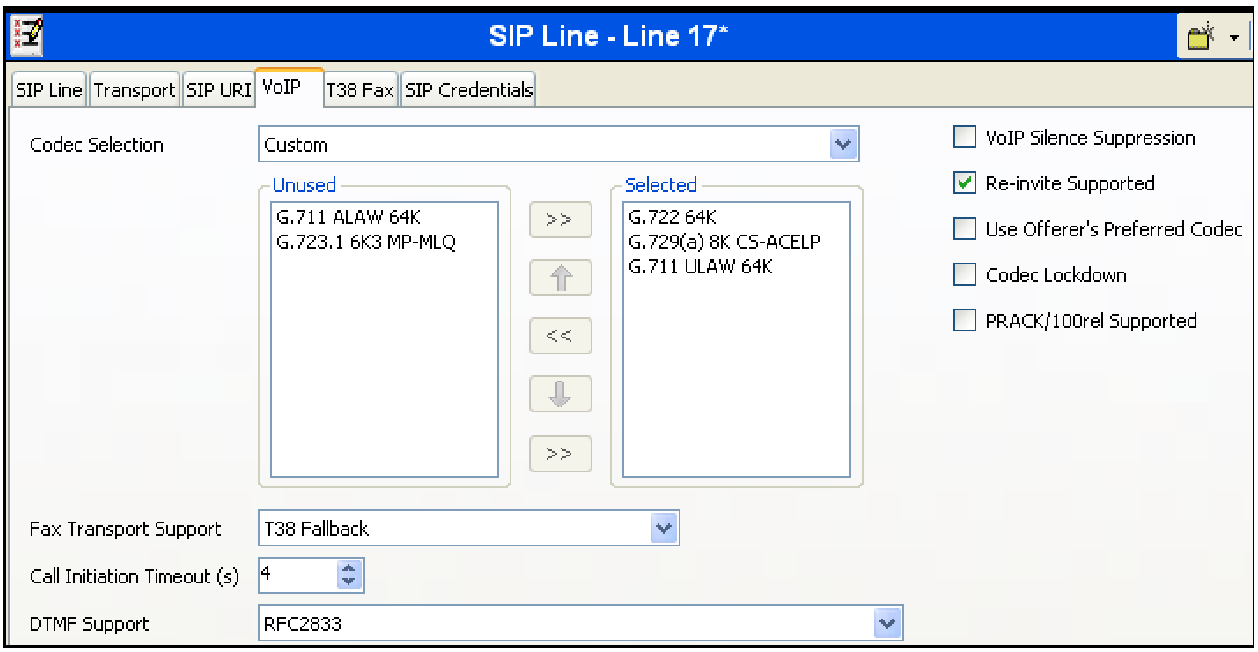
Select the **SIP URI** tab. To add a new SIP URI, click the **Add…** button. In the bottom of the screen, a New Channel area will be opened. To edit an existing entry, click an entry in the list at the top, and click the **Edit…**button. In the bottom of the screen, the Edit Channel area will be opened. 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 **PAI** parameter was introduced in IP Office Release 6.1, and the value "Use Internal Data" is shown selected from the drop-down menu. This inserts the P-Asserted-Identity (PAI) header, to assert the identity of users in outgoing SIP requests or response messages, when Privacy is requested. With PAI set to "none", IP Office Release 6.1 and 8.1 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 1, will be referenced when configuring Incoming Call Routes to map inbound SIP trunk calls to IP Office destinations in Section 5.7. The **Outgoing Group** parameter, set here to 1, will be used for routing outbound calls to Verizon via the Short Codes (Section 5.6) or ARS configuration (Section 5.8). The **Max Calls per Channel** parameter, configured here to 20, 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. Channels 2 and 3 display service numbers, such as a DID number routed directly to voicemail or DID used for Mobile Call Control. DID numbers that IP Office should admit can be entered into the **Local URI** and **Contact** fields instead of “Use Internal Data”. The numbers 732-945-0245 and 732-945-0246 will be assigned as service numbers in the Incoming Call Routes in Section 5.7.

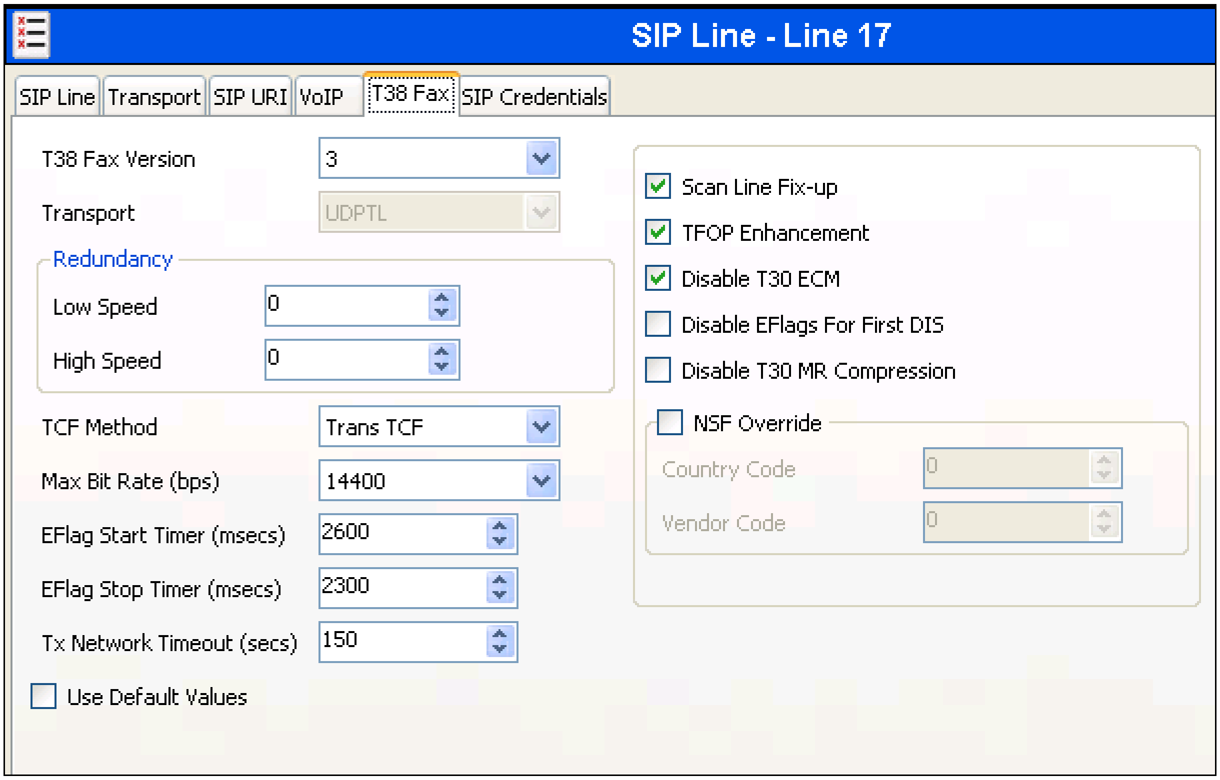
5.4.3. SIP Line - VoIP Tab

Select the **VoIP** tab. The **Codec Selection** drop-down box → **System Default** (default) when selected will match the codecs set in the system wide Default Selection list (**System** → **Codecs**). In the sample configuration, **Custom** was selected and codecs preferred by Verizon were included as well as the newly supported G.722 codec (i.e., **G.722 64K**, **G729(a) 8K CS-ACELP** and **G.711 ULAW 64K**). This will cause IP Office to include G.722, G.729a and G.711MU in the Session Description Protocol (SDP) offer, in that order. Set the **Fax Transport Support** drop-down to "T38 Fallback". This enables T.38 to be used if supported and will fall-back to G.711 if not. If using T.38 fax, the **T38 Fax** tab must be visited and the **Disable T30 ECM** option checked or fax failures using T38 may occur (See Section 5.4.4 and Section 2.2 for further information). 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 **Re-invite Supported** parameter should be checked if the SIP Line will be used for fax. For PSTN originations, Verizon preferred the G.729a codec in SDP, while also allowing the G.711MU codec. However, if an originator is at a SIP connected location and offers G.722, Verizon will preserve this offer and allow G.722 to be negotiated and used end to end. During 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 require registration, the **SIP Credentials** tab need not be visited. The **Codec Lockdown** parameter was new in IP Office Release 7 and may retain the default un-checked value. Click **OK** (not shown).



5.4.4. T38 Fax

The settings on this tab are only accessible if **Re-invite Supported** is checked and a value for **Fax Transport Support** other than **"None"** are selected on the **VoIP** tab. Fax relay is only supported on IP500/IP500 V2 systems with an IP500 VCM card. The **Disable T30 ECM** must be checked or fax errors may be experienced when using T38 Fax (See 1.3 for further information). When selected, it disables the T.30 Error Correction Mode used for fax transmission. All other values are left at default.

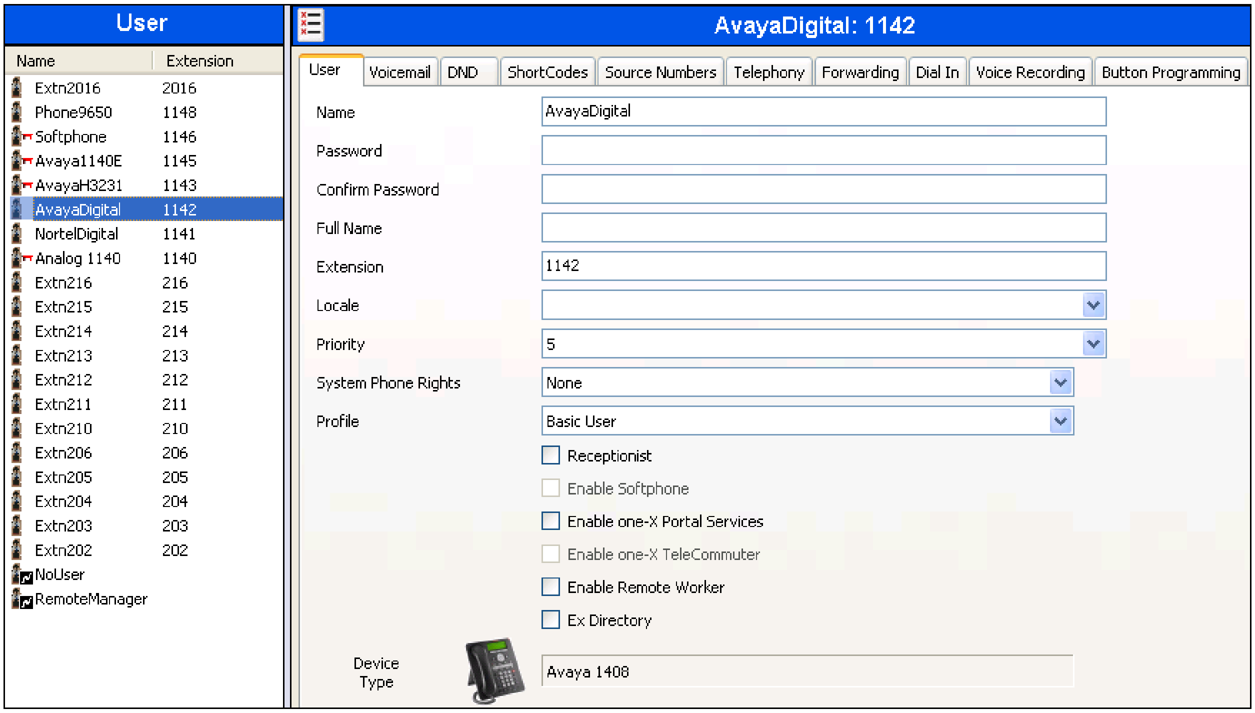


5.5. Users, Extensions, and Hunt Groups

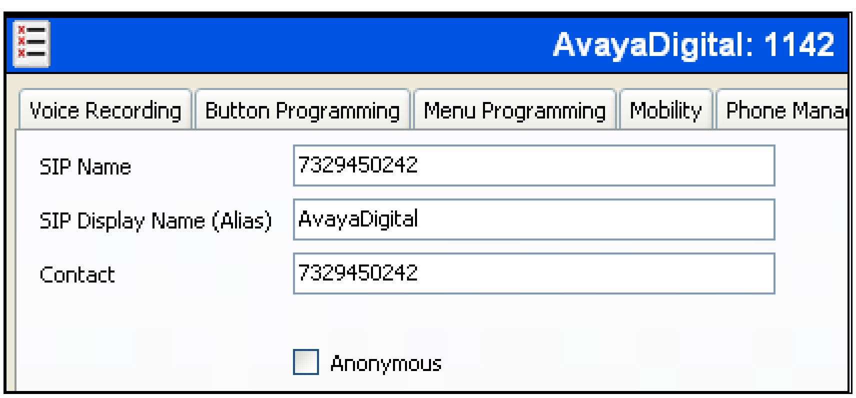
In this section, examples of IP Office Users, Extensions, and Hunt Groups will be illustrated. In the interests of brevity, not all users and extensions shown in **Figure 1** will be presented, since the configuration can be easily extrapolated to other users. To add a User, right click on **User** in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

5.5.1. Digital User 1142

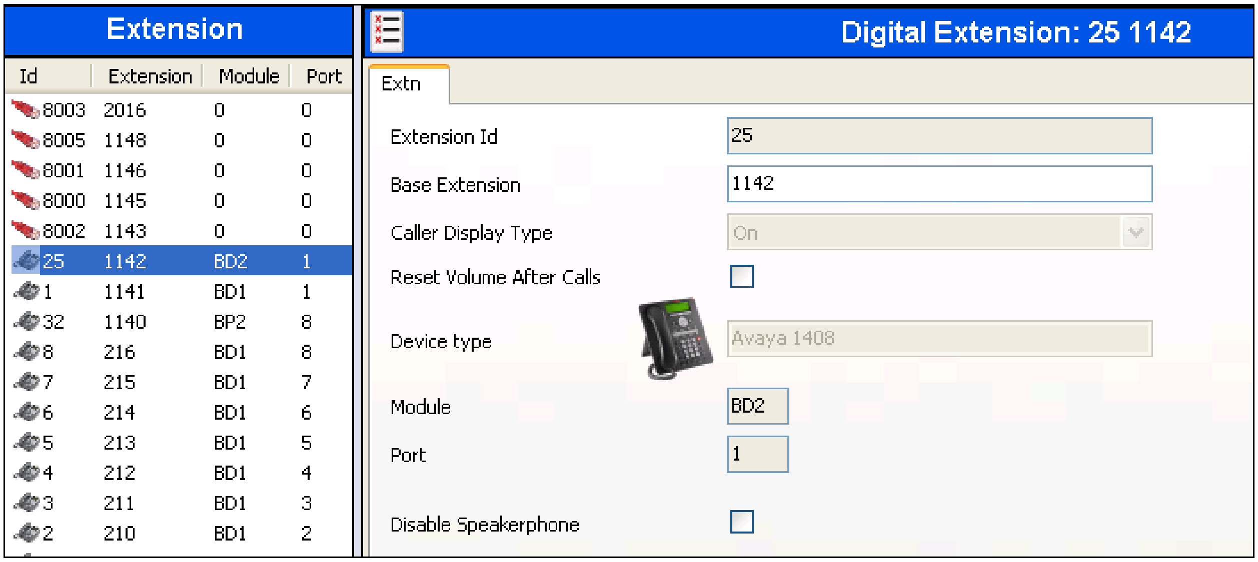
The following screen shows the **User** tab for User 1142. As shown in **Figure 1**, this user corresponds to the Avaya Digital 1408.



The following screen shows the **SIP** tab for User 1142. The **SIP Name** and **Contact** parameters are configured with the DID number of the user, 732-945-0242. 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 5.6 for a method of using a short code (rather than static user provisioning) to place an anonymous call.



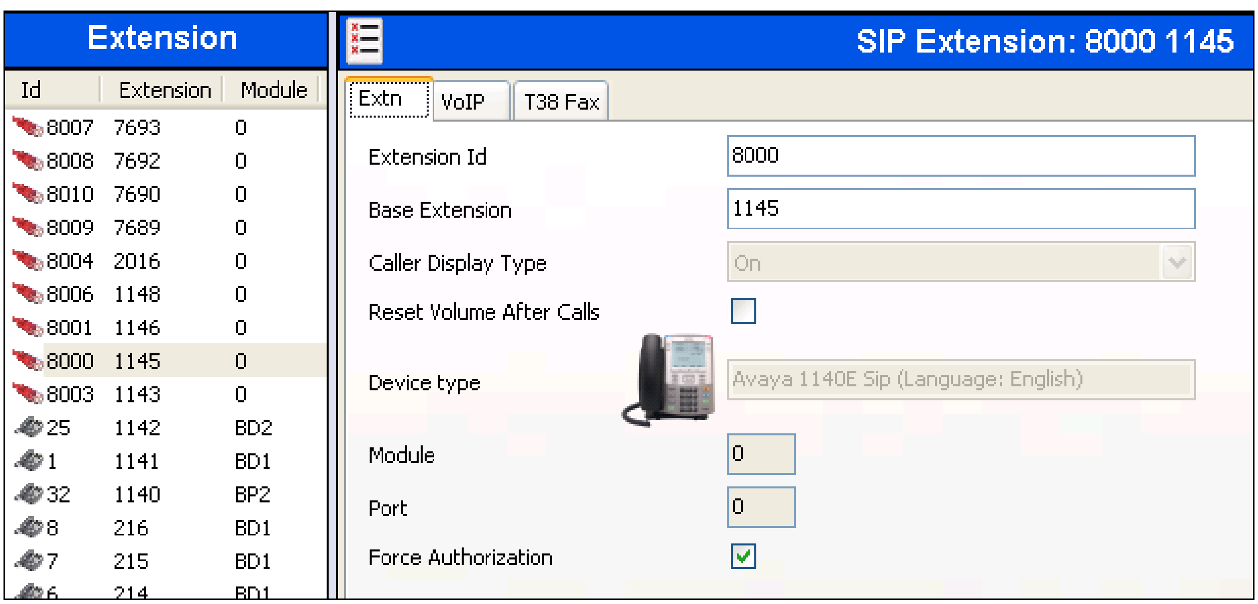
The following screen shows the Extension information for this user. To view, select Extension from the Navigation pane, and the appropriate extension from the Group pane.



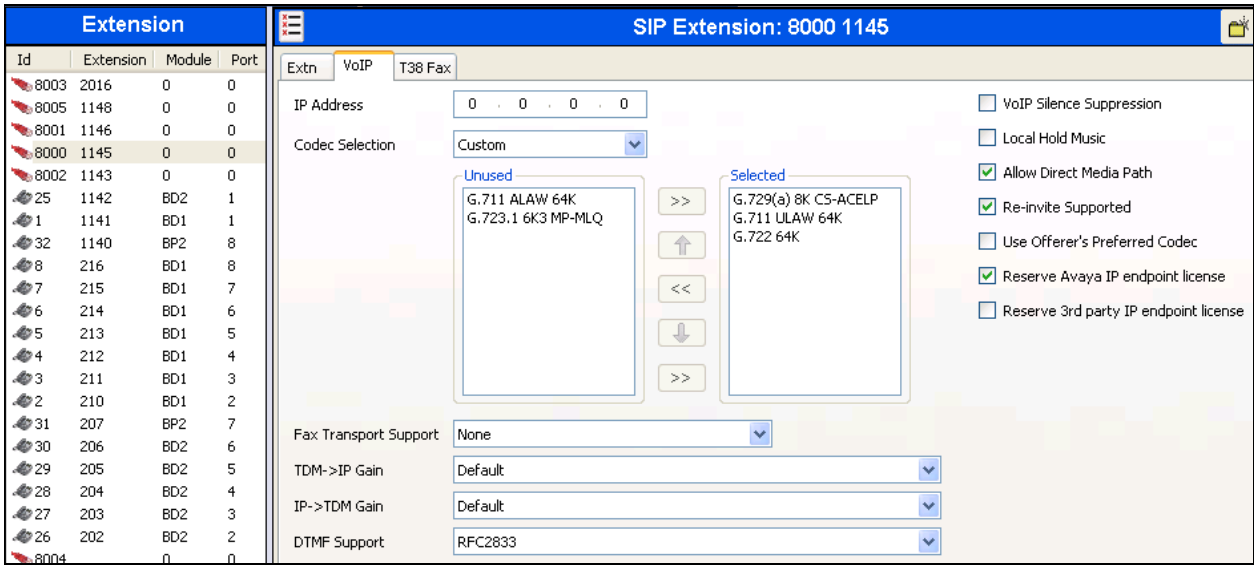
5.5.2. SIP Telephone Users (Avaya 1140E, Avaya 1220)

The process of adding the Avaya 1140E and Avaya 1220 SIP Telephones to the configuration is illustrated in Reference [VZB-IPT-IPOR7]. This section will summarize aspects of the completed configuration for the Avaya 1140E only. The configuration of the Avaya 1220 is similar.

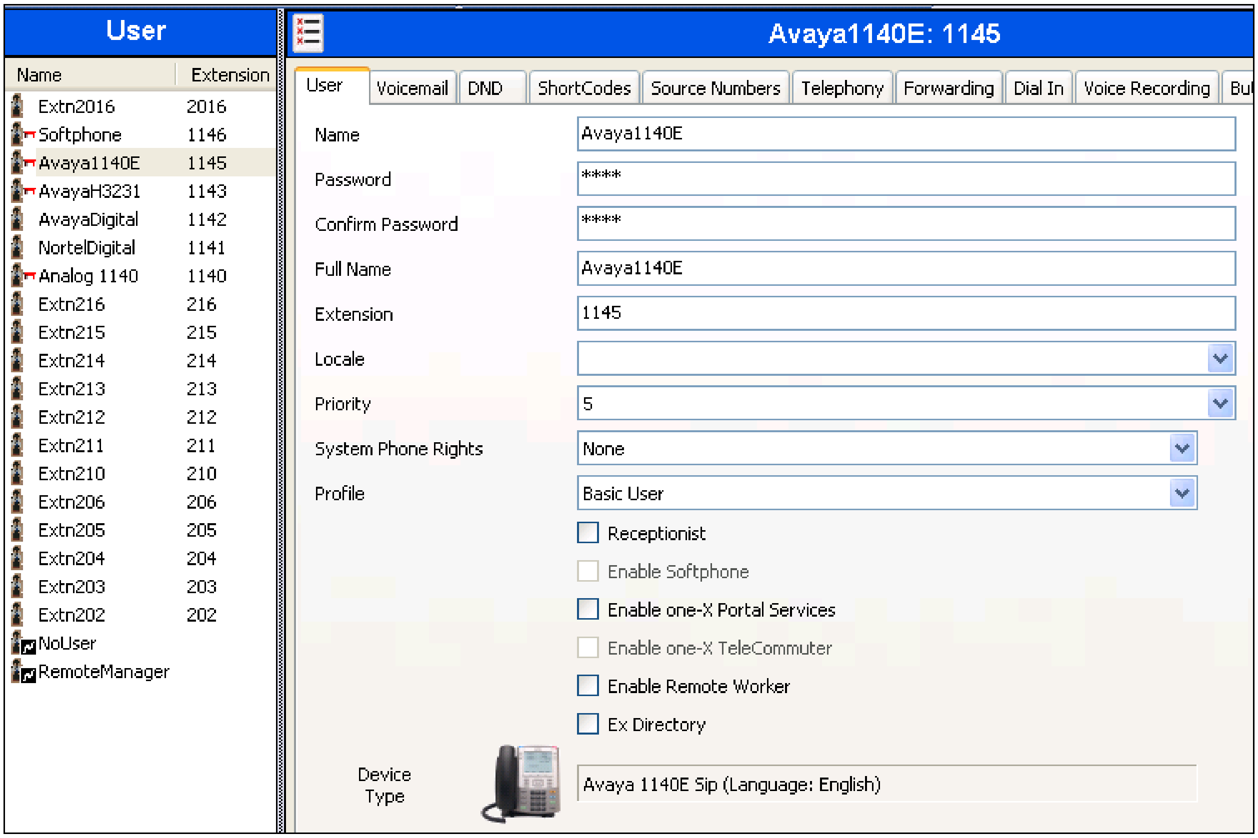
A new SIP extension may be added by right-clicking on **Extension** in the Navigation pane and selecting **New SIP Extension**. Alternatively, an existing SIP extension may be selected in the group pane. The following screen shows the **Extn** tab for the extension corresponding to an Avaya 1140E. The **Base Extension** field is populated with 1145, the extension assigned to the Avaya 1140E. Ensure the **Force Authorization** box is checked.



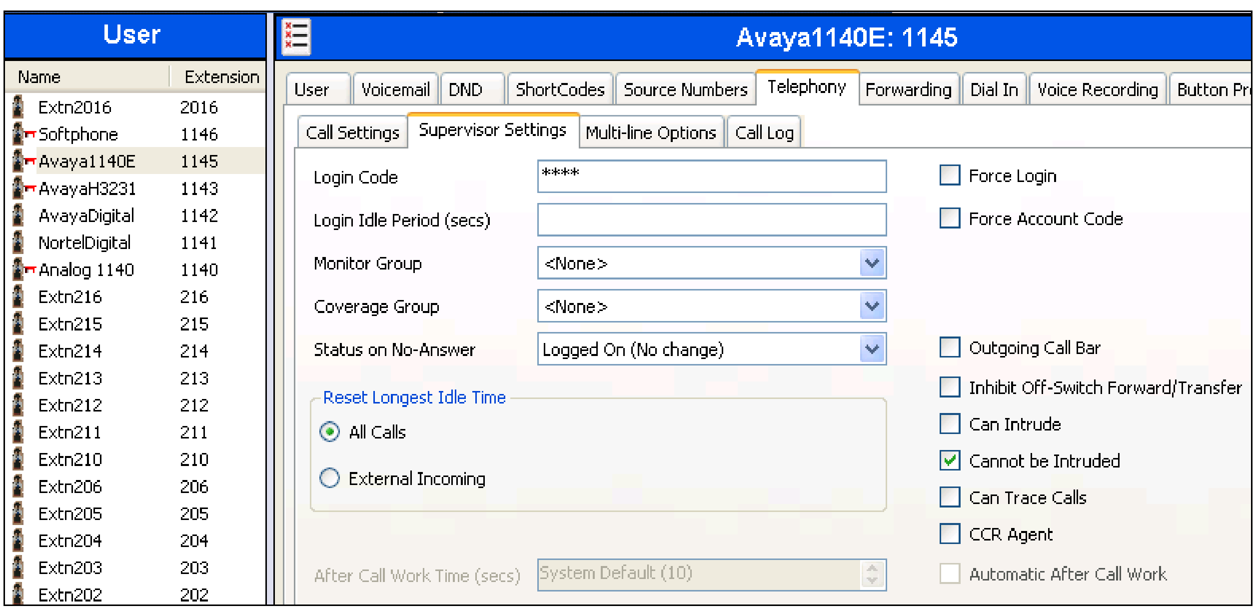
The following screen shows the **VoIP** tab for the extension. The **IP Address** field may be left blank. Check the **Reserve Avaya IP endpoint license** box. The new **Codec Selection** parameter may retain the default setting “System Default” to follow the system configuration shown in Section 5.4.3. Alternatively, “Custom” may be selected to allow the codecs to be configured for this extension, using the arrow keys to select and order the codecs. Other fields may retain default values.



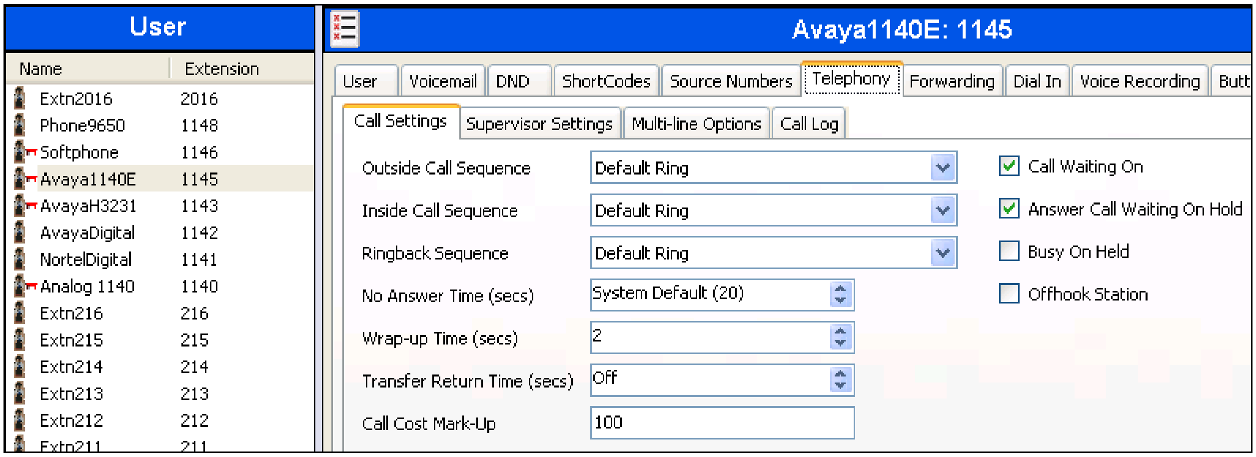
The following screen shows the **User** tab for User 1145 corresponding to an Avaya 1140E. The **Extension** parameter is populated with extension 1145.



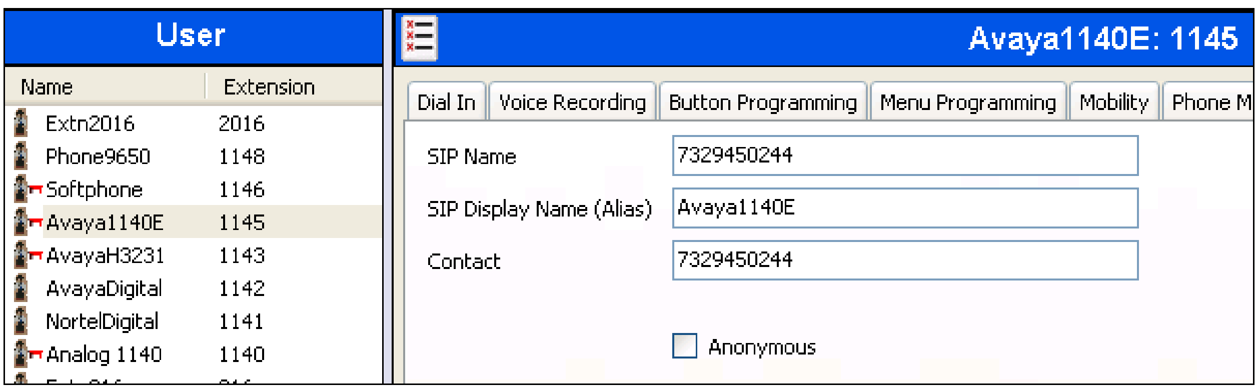
Select the **Telephony** tab. Then select the **Supervisor Settings** tab as shown below. The **Login Code** will be used by the Avaya 1140E telephone user as the login password.



Remaining in the **Telephony** tab for the user, select the **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow multiple call appearances and transfer operations.



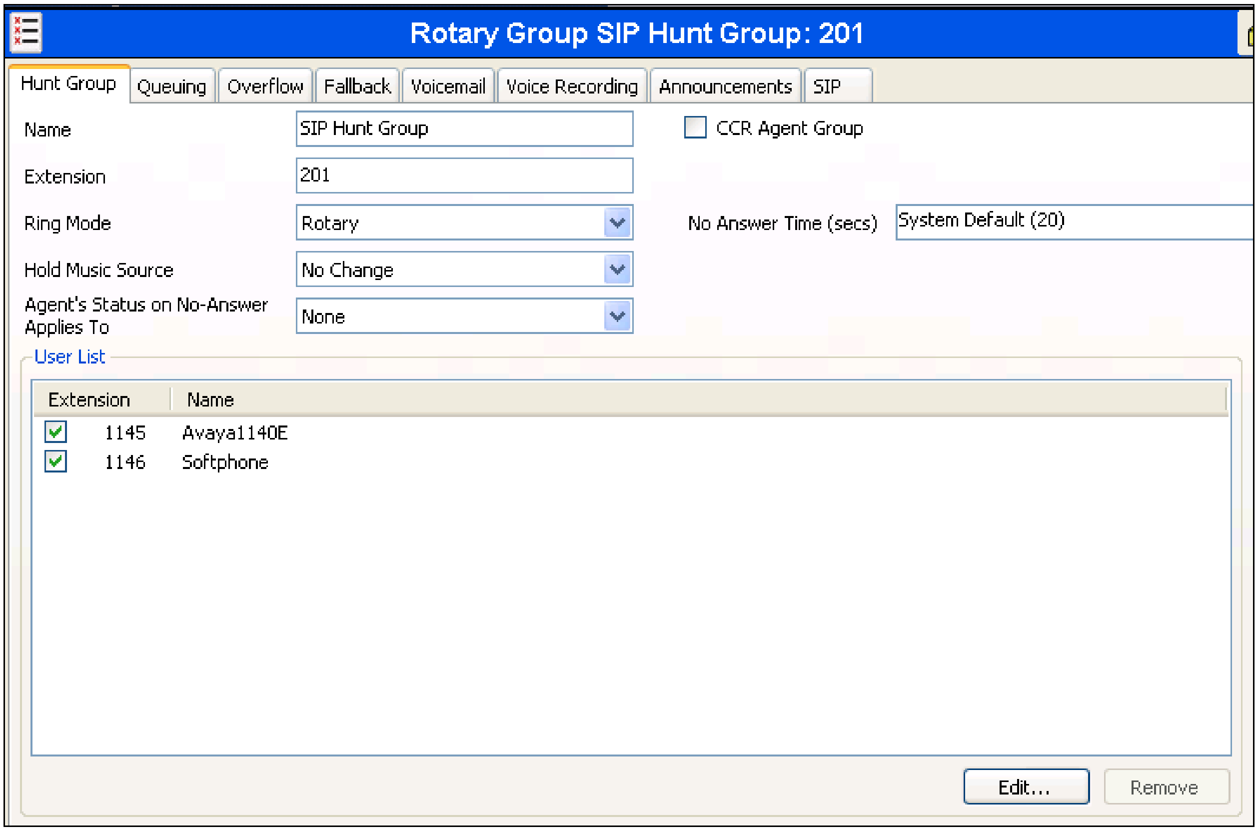
Like other users previously illustrated, the **SIP** tab for the user with extension 1145 is configured with a **SIP Name** and **Contact** specifying the user's Verizon IP Trunk service DID number.



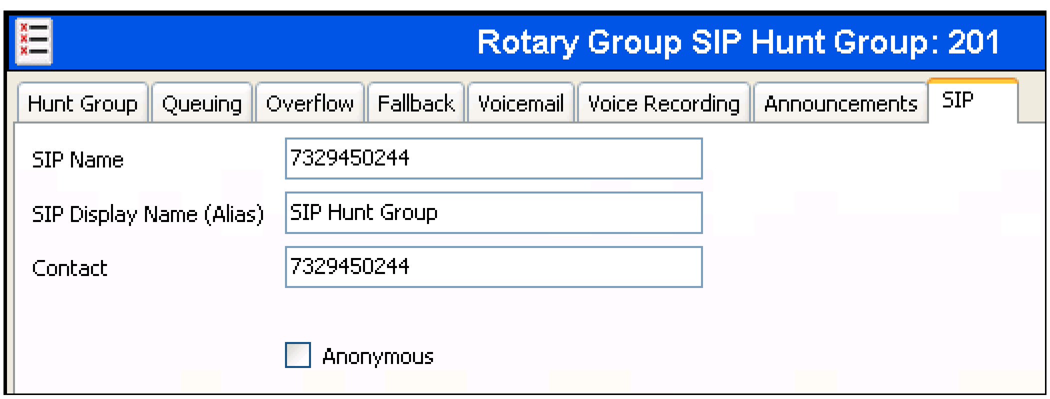
5.5.3. Hunt Groups

During the verification of these Application Notes, users could also receive incoming calls as members of a hunt group. To configure a new hunt group, right-click **HuntGroup** from the Navigation pane, and select **New**. To view or edit an existing hunt group, select **HuntGroup** from the Navigation pane, and the appropriate hunt group from the Group pane.

The following screen shows the **Hunt Group** tab for hunt group 201. This hunt group was configured to contain the two SIP telephones x1145(1140E) and x1146(Softphone) in **Figure 1**. These telephones extensions are rung in order, one after the other. However, the last extension used is remembered. The next call received rings the next extension in the list, due to the **Ring Mode** setting "Rotary" (previously called Circular). Click the **Edit** button to change the **User List**.



The following screen shows the **SIP** tab for hunt group 201. The **SIP Name** and **Contact** are configured with Verizon DID 7329450244. Later, in Section 5.7, an Incoming Call Route will map 7329450244 to this hunt group.

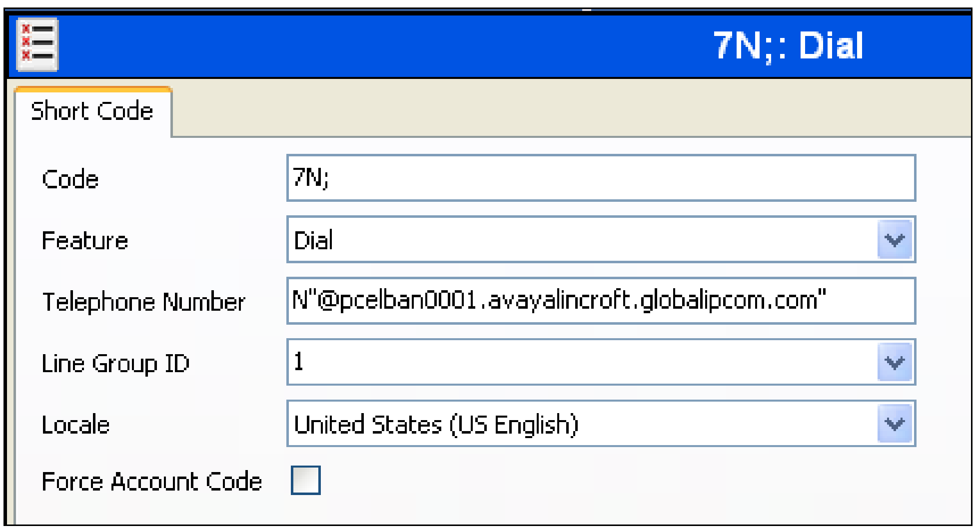


5.6. Short Codes

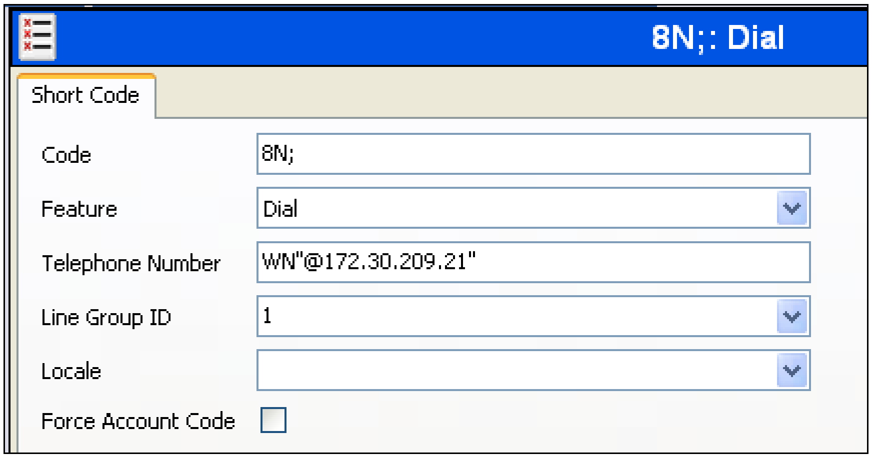
In this section, various examples of IP Office short codes will be illustrated. To add a short code, right click on **Short Code** in the Navigation pane, and select **New**. To edit an existing short code, click **Short Code** in the Navigation pane, and the short code to be configured in the Group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

In the screen shown below, the short code "7N;" is illustrated. The **Code** parameter is set to "7N;". The **Feature** parameter is set to "Dial". The **Telephone Number** parameter is set to N"@Domain Name or IP Address of Verizon Business IP Trunk Service" 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 1, matching the number of the **Outgoing Group** configured on the **SIP URI** tab of SIP Line 17 to Verizon Business (Section 5.4)**.**

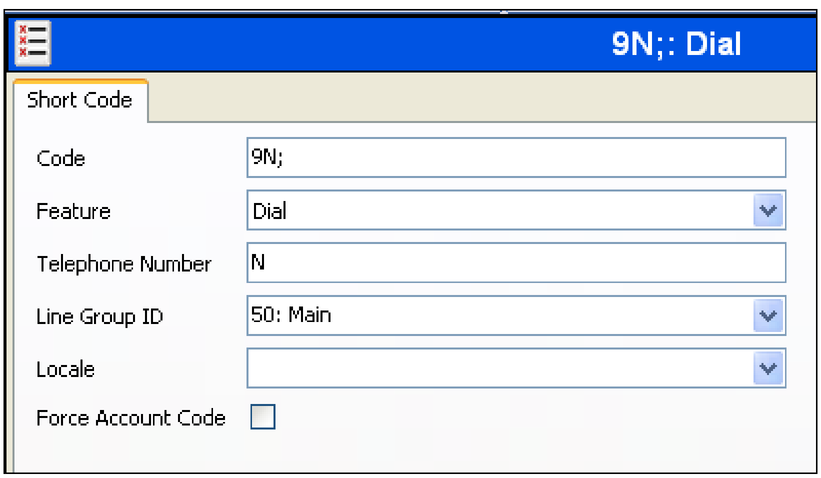
This simple short code will allow an IP Office user to dial the digit 7 followed by any telephone number, symbolized by the letter **N**, to reach the SIP Line to Verizon business. **N** can be any number such as a 10-digit number, a 1+10 digit number, a toll free number, directory assistance (e.g., 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 7 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).



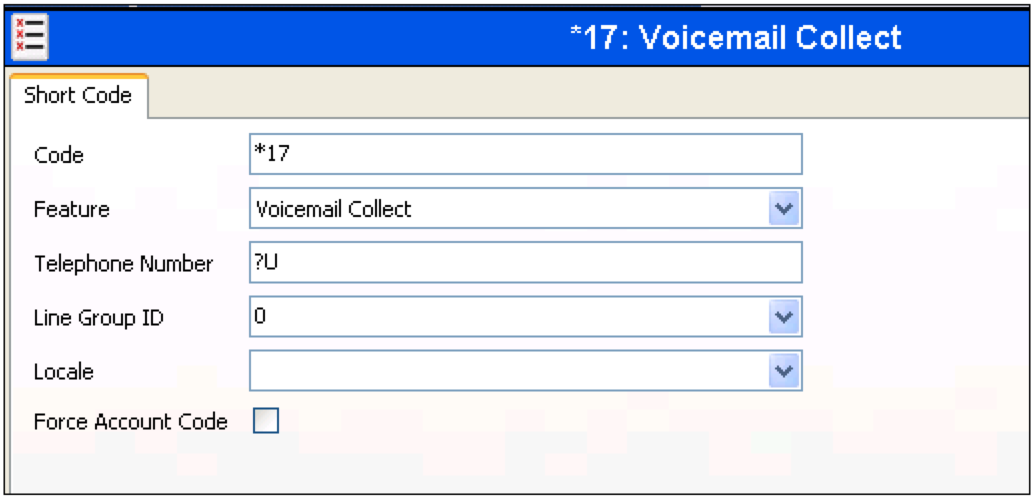
Optionally, add or edit a short code that can be used to access the SIP Line anonymously. In the screen shown below, the short code "8N;" is illustrated. This short code is similar to the "7N;" short code except that the Verizon IP Address rather than the domain is entered in the **Telephone Number** 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.



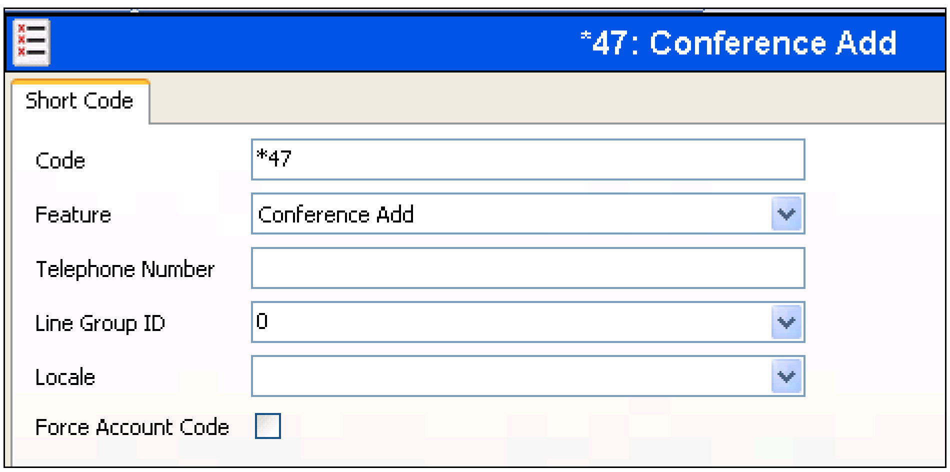
The simple "7N;" and "8N;" short codes illustrated previously do not provide a means of alternate routing if the primary Verizon SIP line is out of service or temporarily not responding. When alternate routing options and/or more customized analysis of the digits following the short code are desired, the Automatic Route Selection (ARS) feature may be used. In the following example screen, the short code 9N is illustrated for access to ARS. When the IP Office user dials 9 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 5.8 for example ARS route configuration for "50: Main" as well as a backup route.



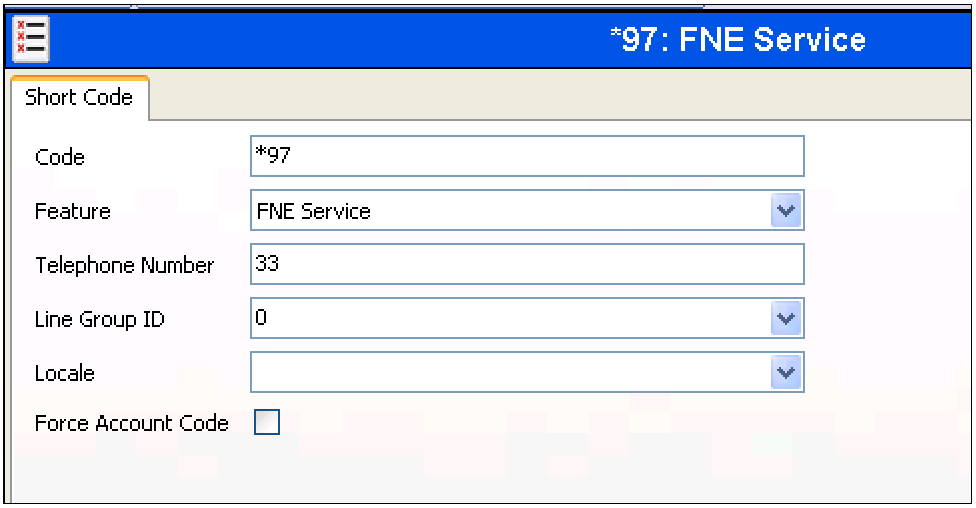
The following screen illustrates a short code that acts like a feature access code rather than a means to access a SIP Line. In this case, the **Code** "*17" is defined for **Feature** "Voicemail Collect". This short code will be used as one means to allow a Verizon DID to be programmed to route directly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



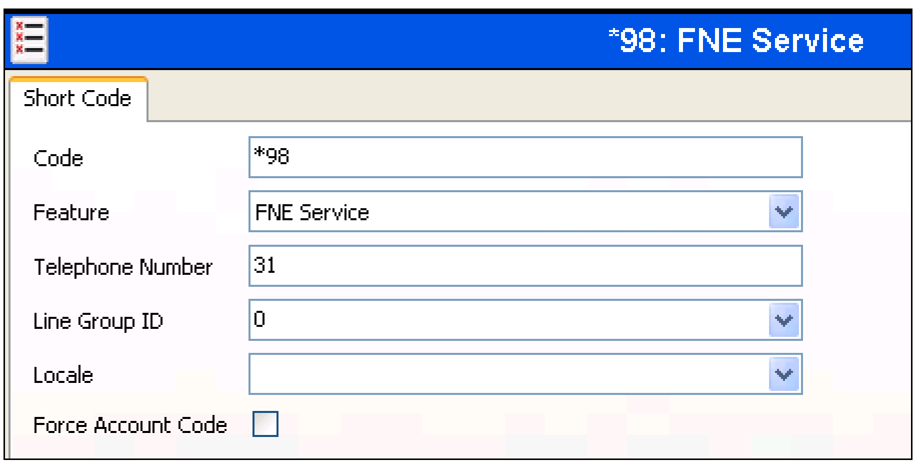
The following screen illustrates another short code. In this case, the **Code** "*47" is defined for **Feature** "Conference Add". In the verification of these Application Notes, "*47" was used by mobile telephones to create a conference via a DTMF sequence using the IP Office Mobile Call Control feature.



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



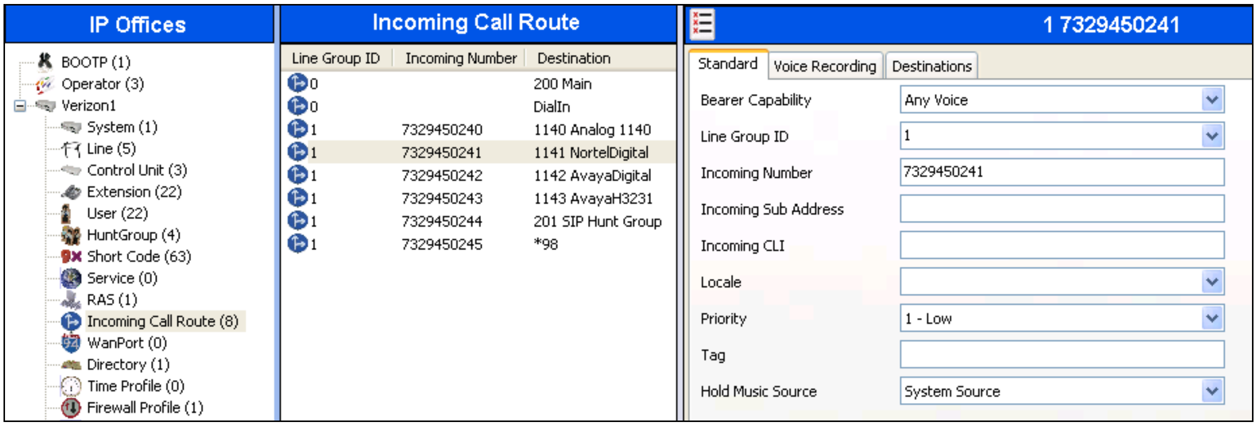
The following screen illustrates another short code. In this case, the **Code** "*98" is defined for **Feature** "FNE Service" and **Telephone Number** "31" for "Mobile Call Control". In the verification of these Application Notes, "*98" was used as the destination of an Incoming Call Route for a Verizon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



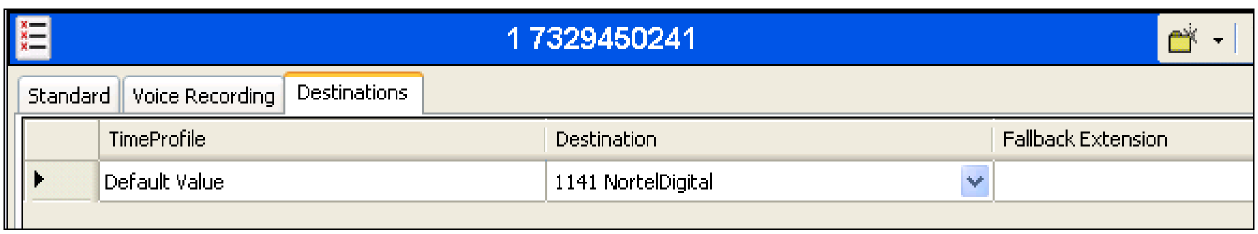
5.7. Incoming Call Routes

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** “7329450241” is illustrated. The **Line Group Id** is 1, matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to Verizon Business in Section 5.4.2.

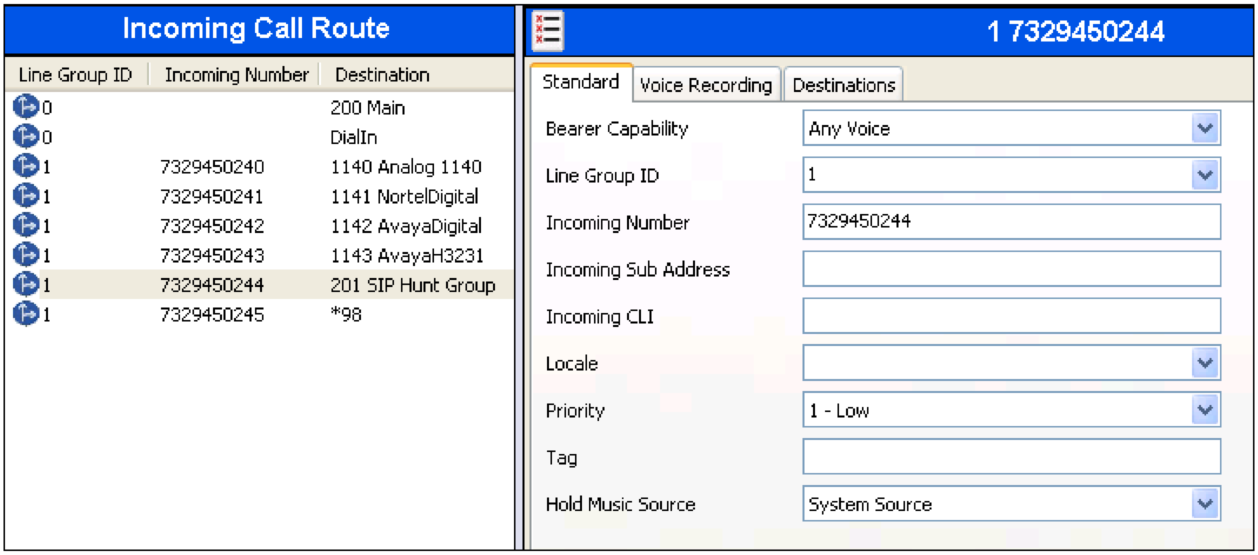


Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when a PSTN user dials 7329450241. As shown in **Table 1**, 7329450241 is the DID number associated with IP Office user extension 1141.

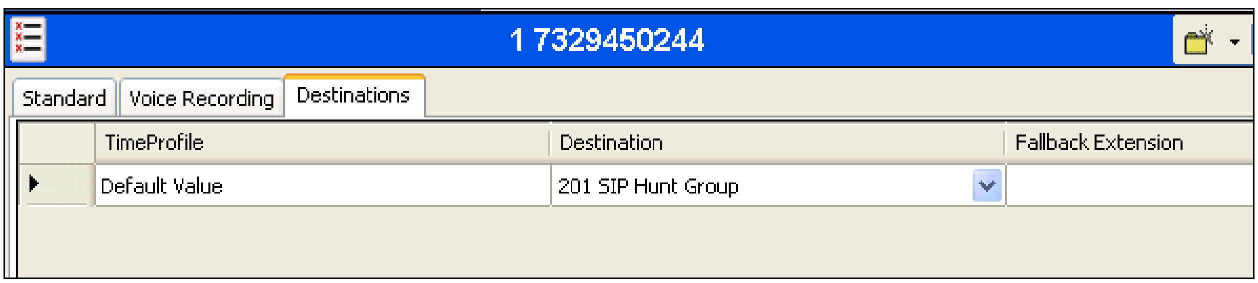


Incoming Call Routes for other direct mappings of DID numbers to IP Office users listed in **Table 1** are omitted here, but can be configured in the same fashion.

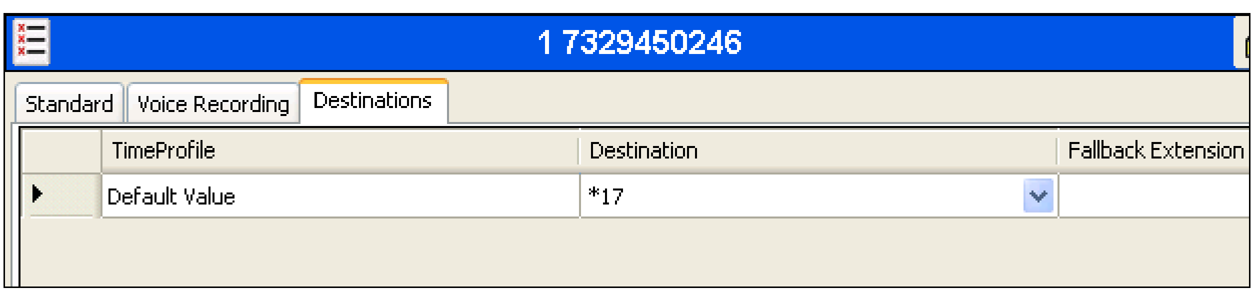
In the screen shown below, the incoming call route for **Incoming Number** “7329450244” is illustrated. The **Line Group Id** is 1, matching the Incoming Group field configured in the SIP URI tab for the SIP Line to Verizon Business in Section 5.4.2.



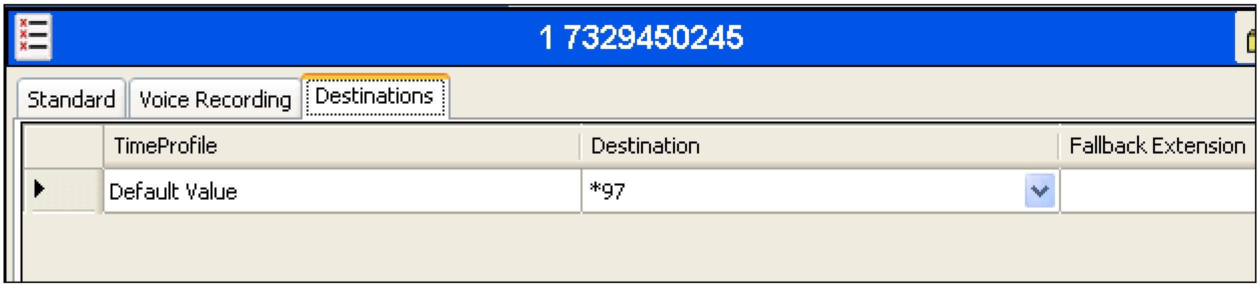
Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when a PSTN user dials 7329450244. In this case, the destination is the hunt group “201 SIP Hunt Group” whose configuration is shown in Section 5.5.4



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 5.6. An incoming call to 732-945-0246 will be delivered directly to voice mail, allowing the caller to log-in to voicemail and access messages. 732-945-0246 was previously defined in the SIP URI tab as a service number, refer to Section 5.4.2.



Similar, the following **Destinations** tab for an incoming call route contains the **Destination** "*97" entered manually. The dial string "*97" is the short code for accessing the "Mobile Call Back" application and 732-945-0245 was configured in Section 5.4.2 on the SIP URI tab as an incoming number. This enables DID calls to 732-945-0245 from a configured twinning destination to be dialed, and then hung up by the caller while hearing ring back. IP Office would then call the caller back using the Verizon IP Trunk Service.



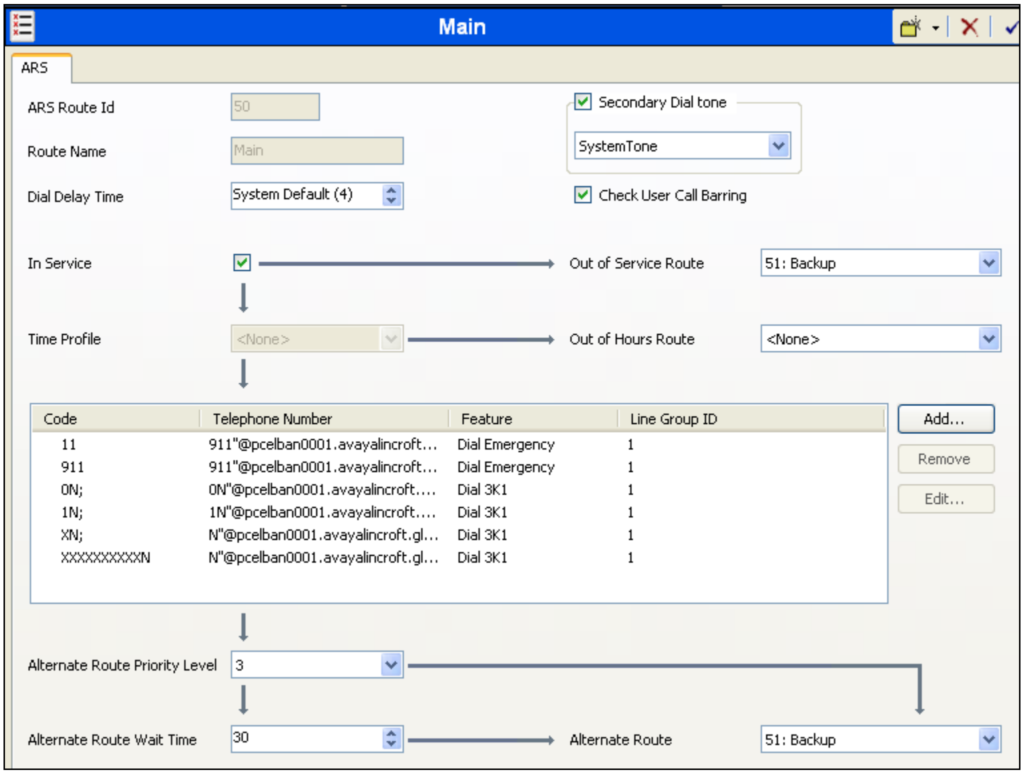
5.8. ARS and Alternate Routing

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 "7N;" short code approach documented in Section 5.6. 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.

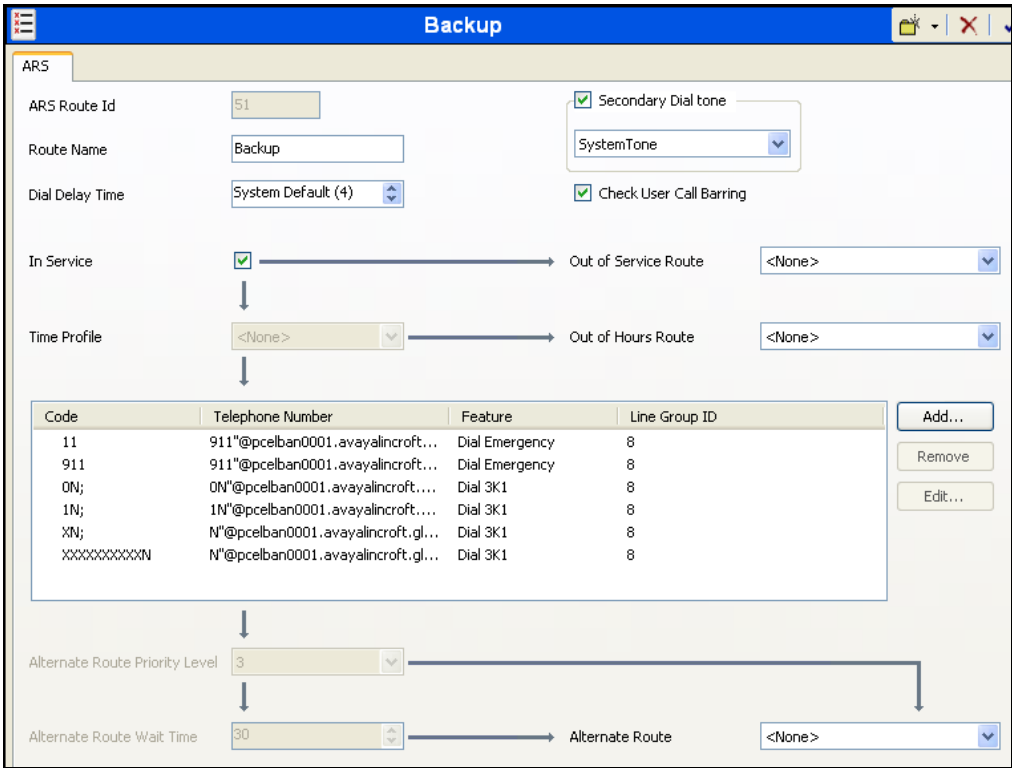
To add a new ARS route, right-click **ARS** in the Navigation pane, and select **New**. To view or edit an existing ARS route, select **ARS** in the Navigation pane, and select the appropriate route name in the Group pane.

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., 9N in Section 5.6) can be further analyzed to direct the call to a specific Line Group ID. Per the example screen above, if the user dialed 9-1-303-538-1000, the call would be directed to Line Group 1. If Line Group 1 cannot be used, the call can automatically route to the route name configured in the **Alternate 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 9-1-303-538-1000, 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, another SIP Line that exists in the configuration that is not described in these Application Notes.



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.

5.9. Privacy / Anonymous Calls

There are multiple methods for a user to withhold outgoing identification:

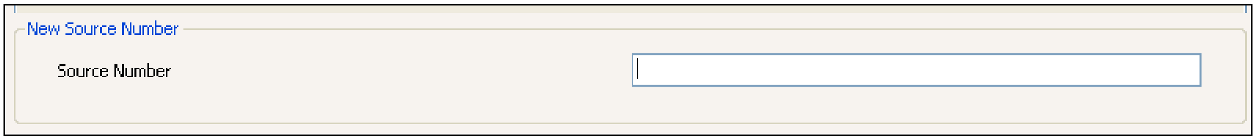
- Dialing the short code 8 to access the SIP Line. (Section 5.6)
- Specific users may be configured to always withhold calling line identification by checking the **Anonymous** field in the **SIP** tab for the user (Section 5.5).
- The Avaya 1600-Series IP Telephones can also request privacy for a specific call, without dialing a unique short code, using **Features → Call Settings → Withhold Number,** on the phone itself.

Verizon Business, however, requires IP Office to include the caller's DID number in the P-Asserted-Identity SIP header to admit an otherwise anonymous caller to the network. You can set the PAI with two different procedures:

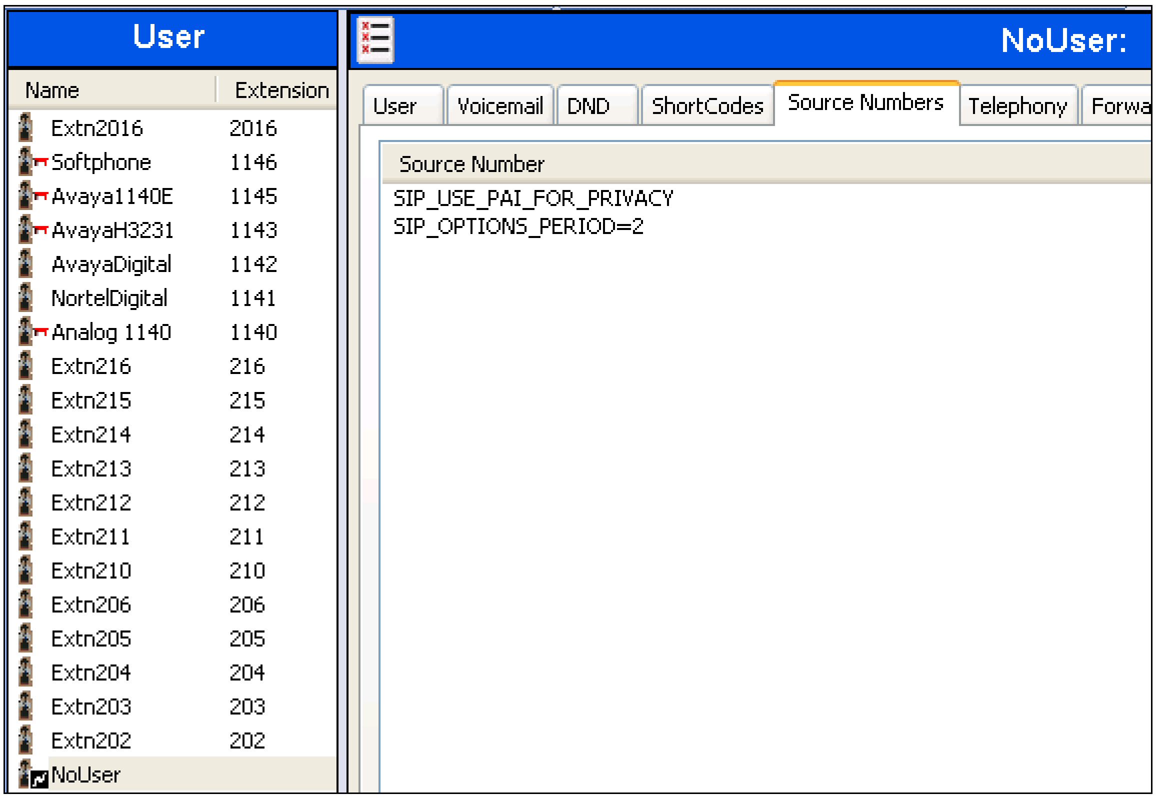
- "Use Internal Data" in the PAI parameter on the SIP Line as shown in Section 5.4.2
- Alternatively, perform the following:

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.

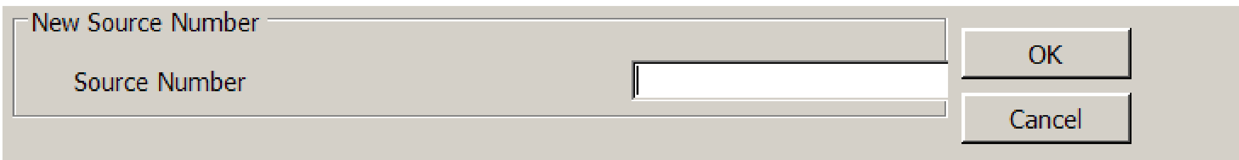


5.10. SIP Options Frequency

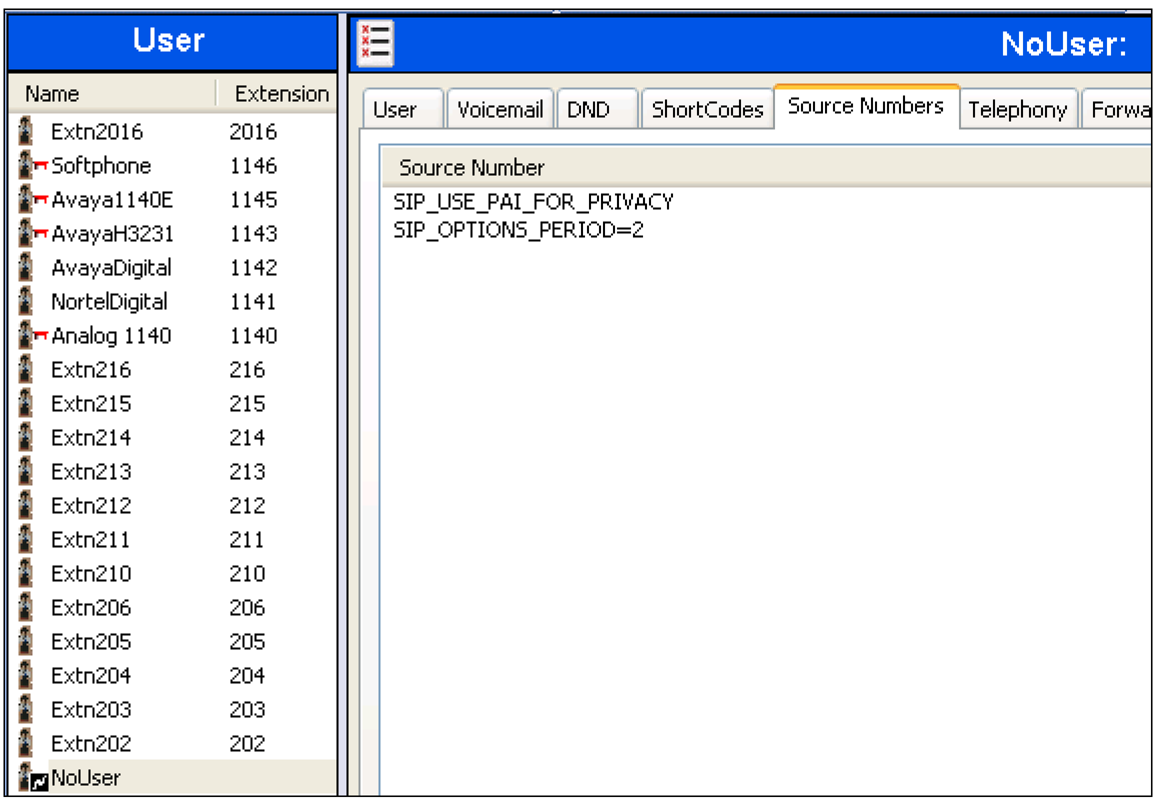
In the sample configuration, IP Office periodically checks the health of the SIP Line by sending a SIP OPTIONS message. If there is no response, IP Office can mark the trunk out of service. Although ARS as shown in Section 5.8 can include alternate routes to complete calls even if the far-end is not 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 5.3 and Section 5.4.

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

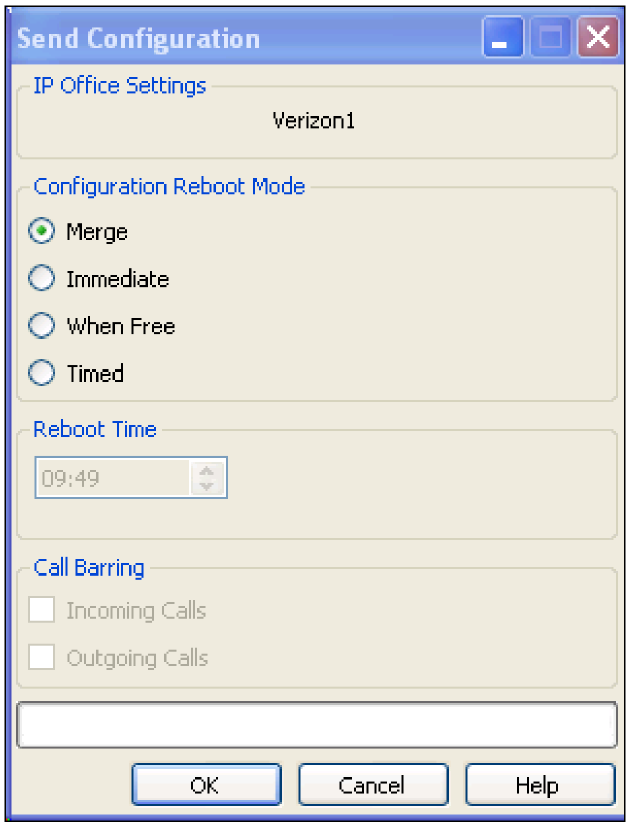


With this configuration, Binding Refresh Intervals of 30 seconds and 90 seconds were tested successfully. That is, IP Office sourced SIP OPTIONS every 30 or 90 seconds, depending on the value configured in the Binding Refresh Interval, since the Binding Refresh Interval was less than the value configured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

5.11. 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.

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.



6. 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, DNS server information, 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 5.

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

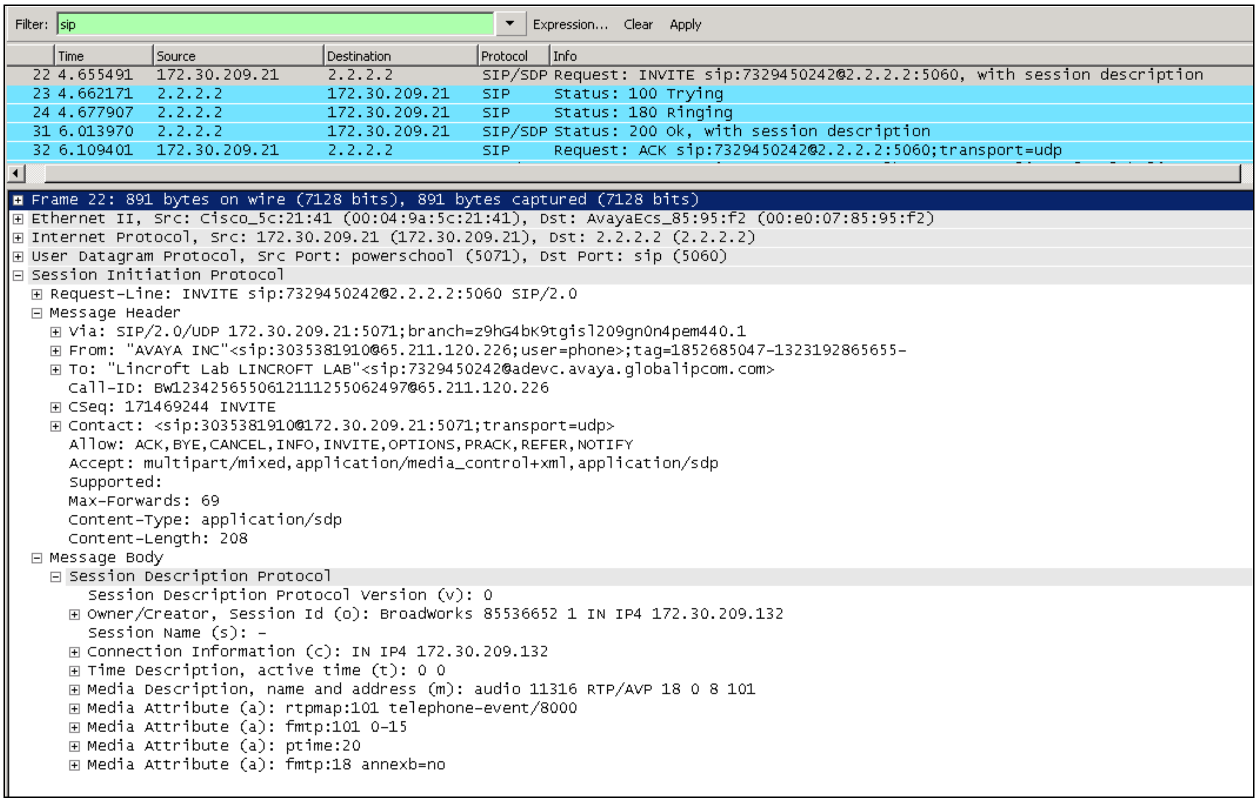
The following scenarios will result in IP Office sending REFER to the Verizon network. Each scenario was tested successfully.

- 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

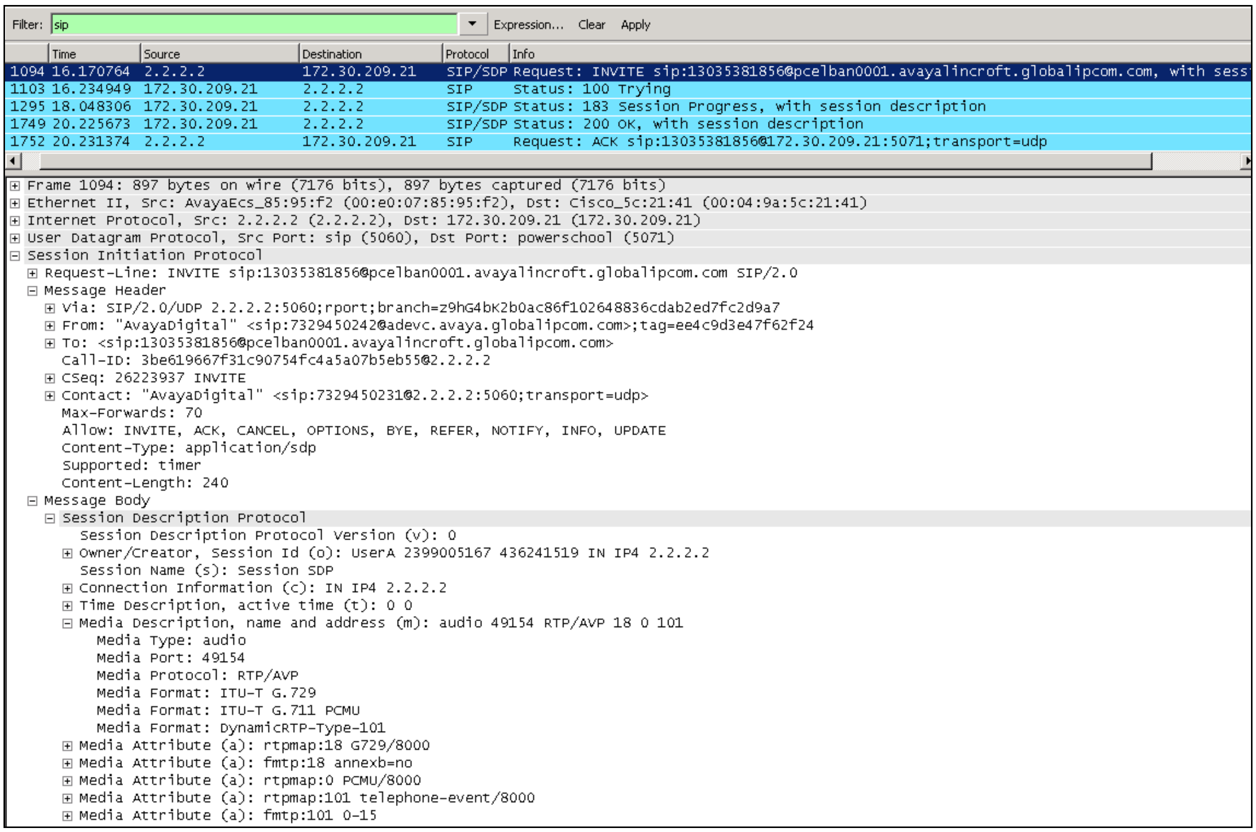
This section illustrates the SIP signaling for an inbound Verizon IP Trunk call that is transferred back to the PSTN by an IP Office user. IP Office will use SIP REFER.

The following screen shows the portion of the trace until the point where the IP Office user answers the inbound call. Frame 22 is selected and expanded to show the contents of the INVITE message from Verizon. In frame 31, IP Office answers the call with a 200 OK.

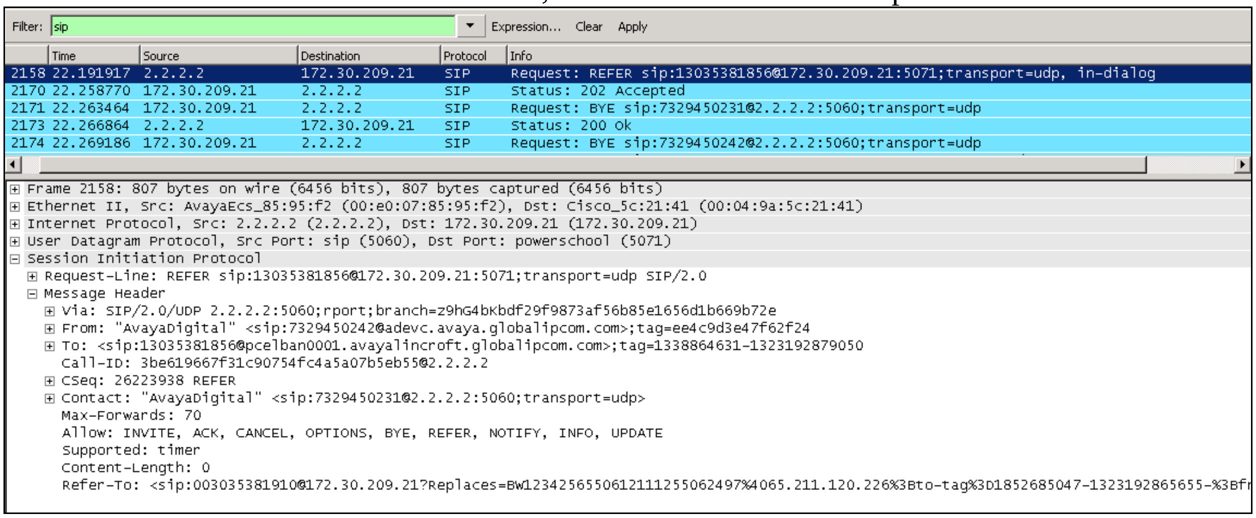


After answering the inbound call, the IP Office user presses the transfer button, and dials the number 9-1-303-538-1856 which is routed by IP Office to the Verizon IP Trunk service. In this “attended transfer” example, the IP Office user performing the transfer waits for the outbound call to be answered before pressing the transfer button a second time to complete the transfer.

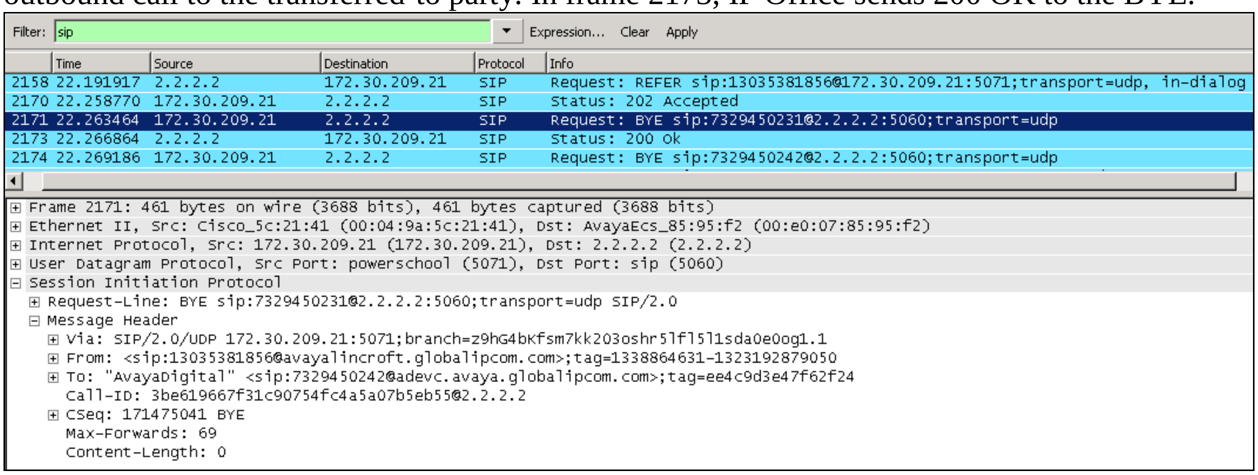
Scrolling down in the same trace in the screen below, frame 1094 is selected to show the contents of an outbound INVITE sent by IP Office to the Verizon IP Trunk service. In frame 1749, Verizon sends the 200 OK when the called party answers the call. NOTE: This Wireshark was taken before the addition of the G.722 codec into the configuration.



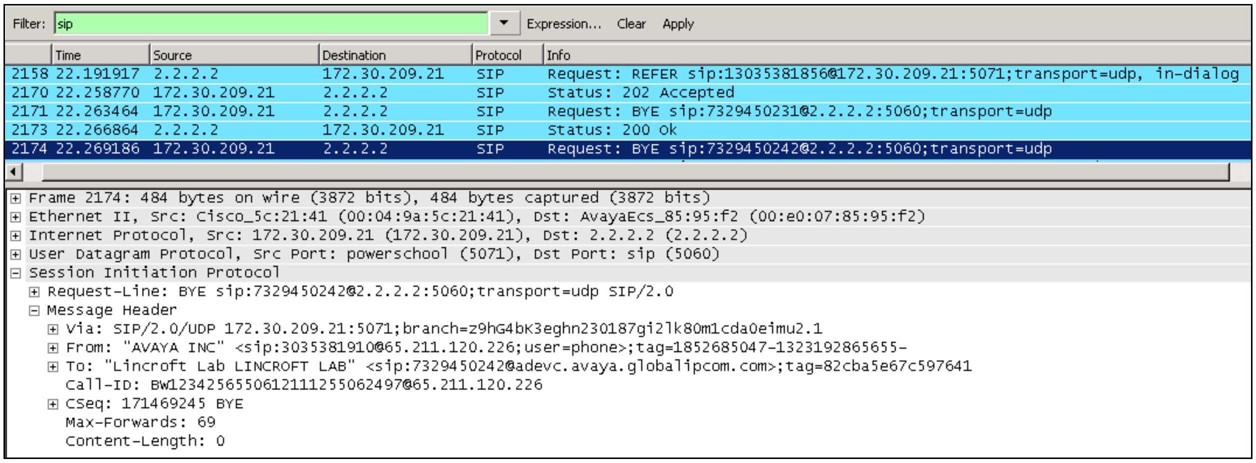
In this example, after the called party answers, the IP Office user presses the transfer button a second time to complete the transfer. Scrolling down in the same trace in the screen below, frame 2158 is selected to show the contents of a REFER message sent by IP Office to the Verizon IP Trunk service. 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 2170, Verizon sends a 202 Accepted for the REFER.



In frame 2171 expanded below, Verizon sends a BYE for the Call-ID associated with the outbound call to the transferred-to party. In frame 2173, IP Office sends 200 OK to the BYE.



In frame 2174 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 (303-538-1910) is speaking with the transferred-to destination (303-538-1856).

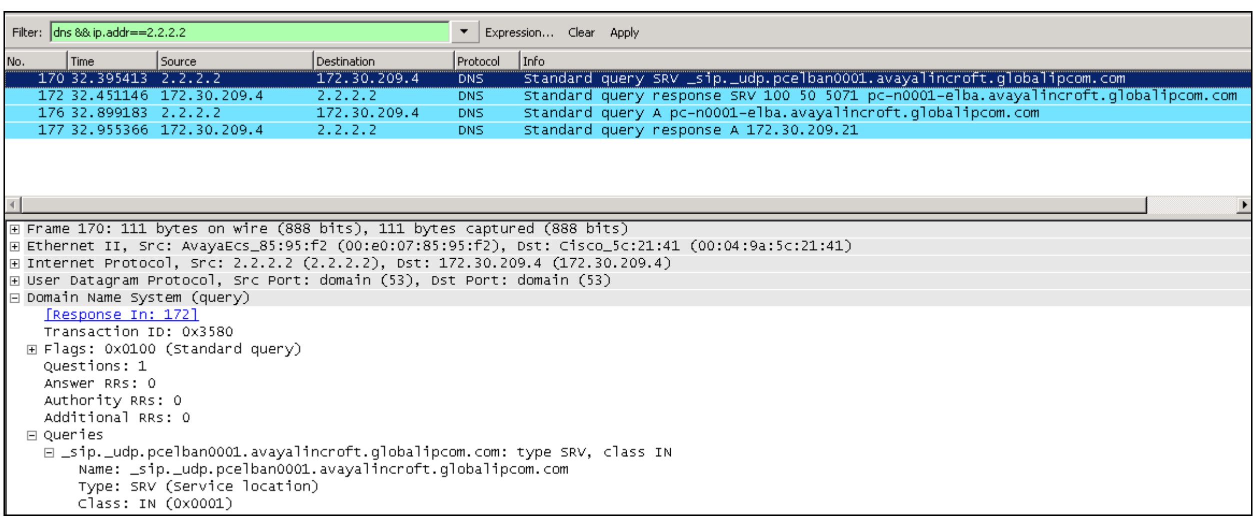


7.2. DNS SRV Testing

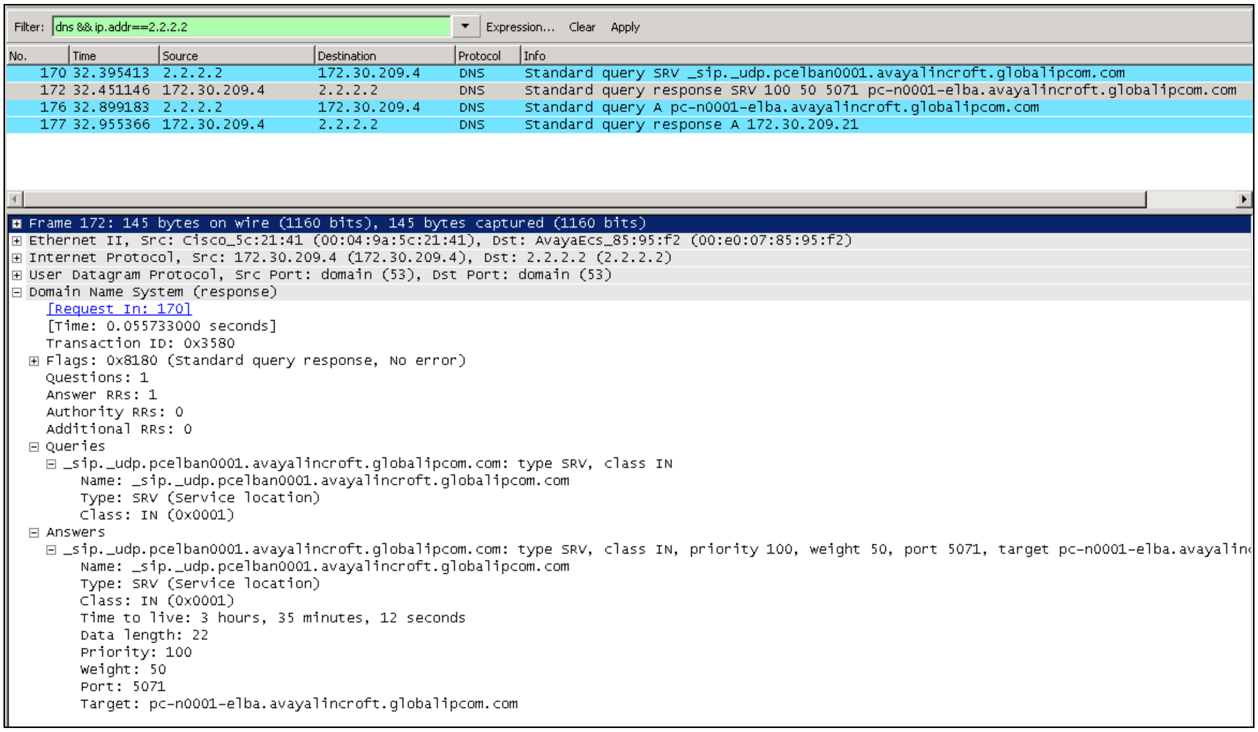
The IP Office capability to determine the Verizon SIP signaling address and port using DNS procedures was tested using the production Verizon PIP circuit. Rather than statically configure IP Office with the Verizon IP Address and SIP signaling port, as was the case shown in reference [JRR-IPOR6], IP 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

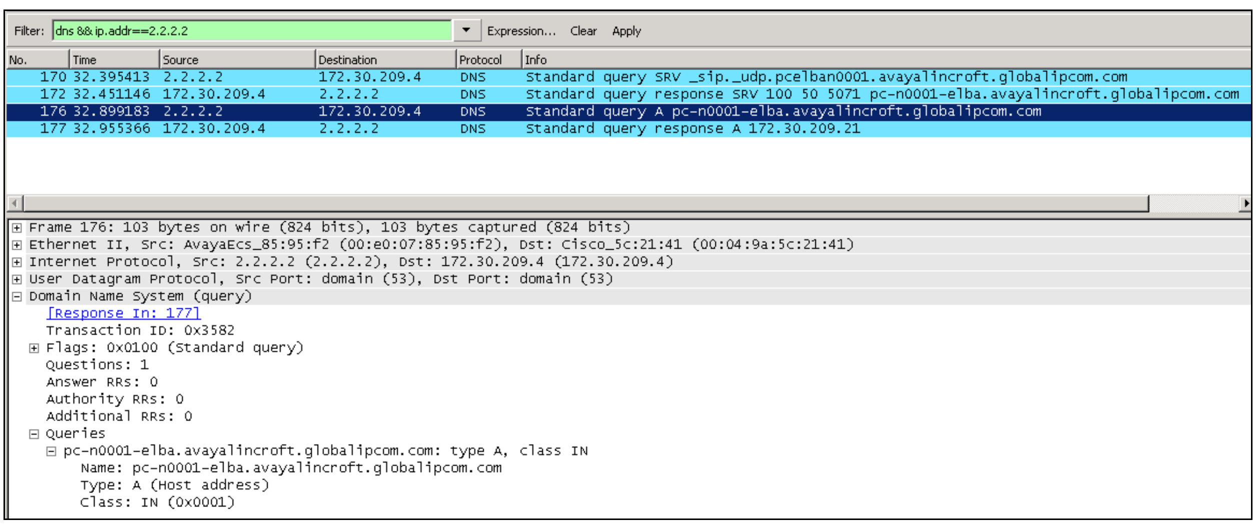
This section illustrates the DNS signaling used when the SIP Line in IP Office is configured to use DNS SRV. Please reference Section 5.4.1 of these Application Notes for the relevant configuration. In the filtered Wireshark trace shown below, IP Office (2.2.2.2) sends a DNS SRV query to the Verizon DNS server (172.302.209.4) configured in IP Office for the SIP Line. Frame 170 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 172 is highlighted and expanded in the following screen. Note that the “Answer” contains Target “pc-n0001-elba.avayalincroft.globalipcom.com” and port 5071.

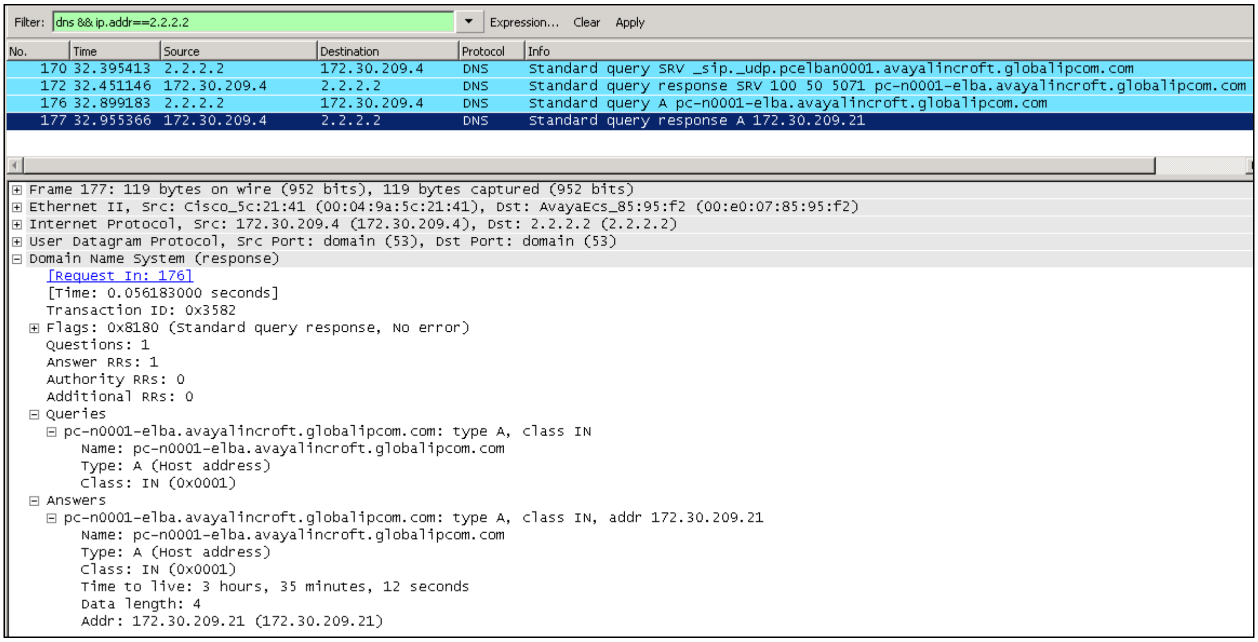


Frame 176 is expanded below to illustrate the IP Office DNS A-query to determine the IP Address associated with the name “pc-n0001-elba.avayalincroft.globalipcom.com” (i.e., the “Target” returned by Verizon as shown in the prior screen).



Frame 177 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

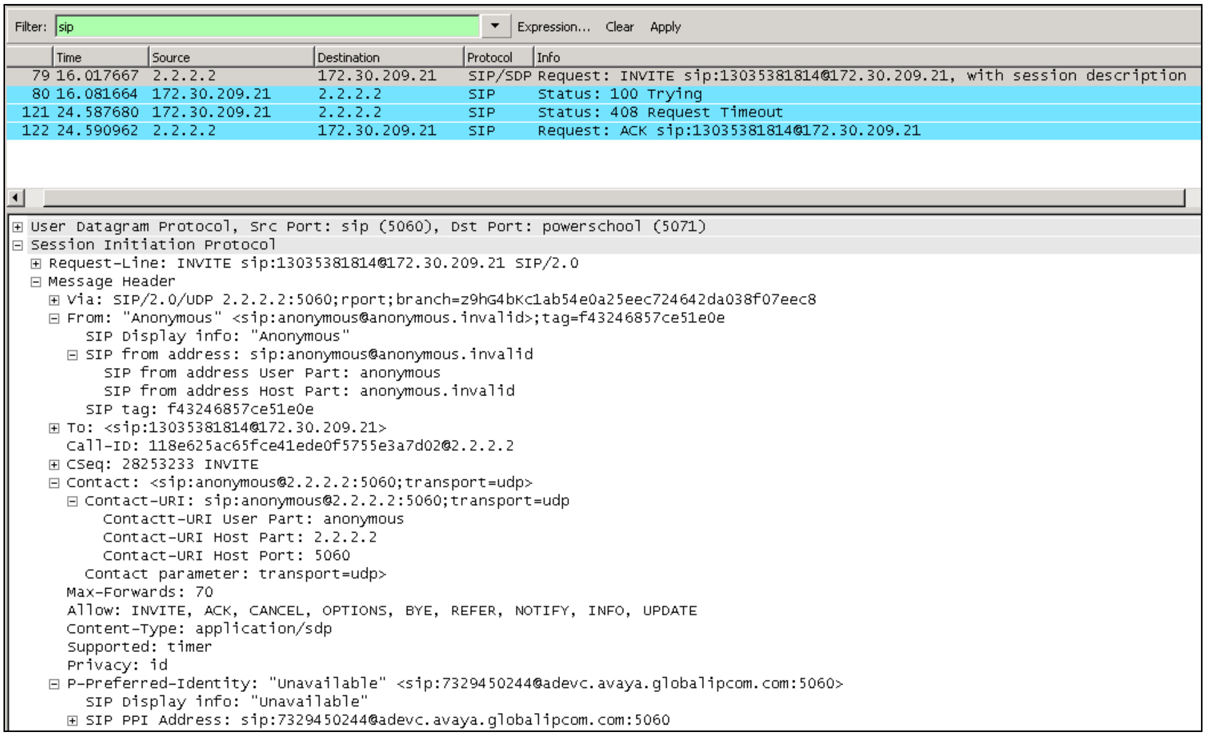
(172.30.209.21) and SIP signaling port (5071) used by Verizon IP Trunk service on the production circuit, without any static provisioning of this information within IP Office.



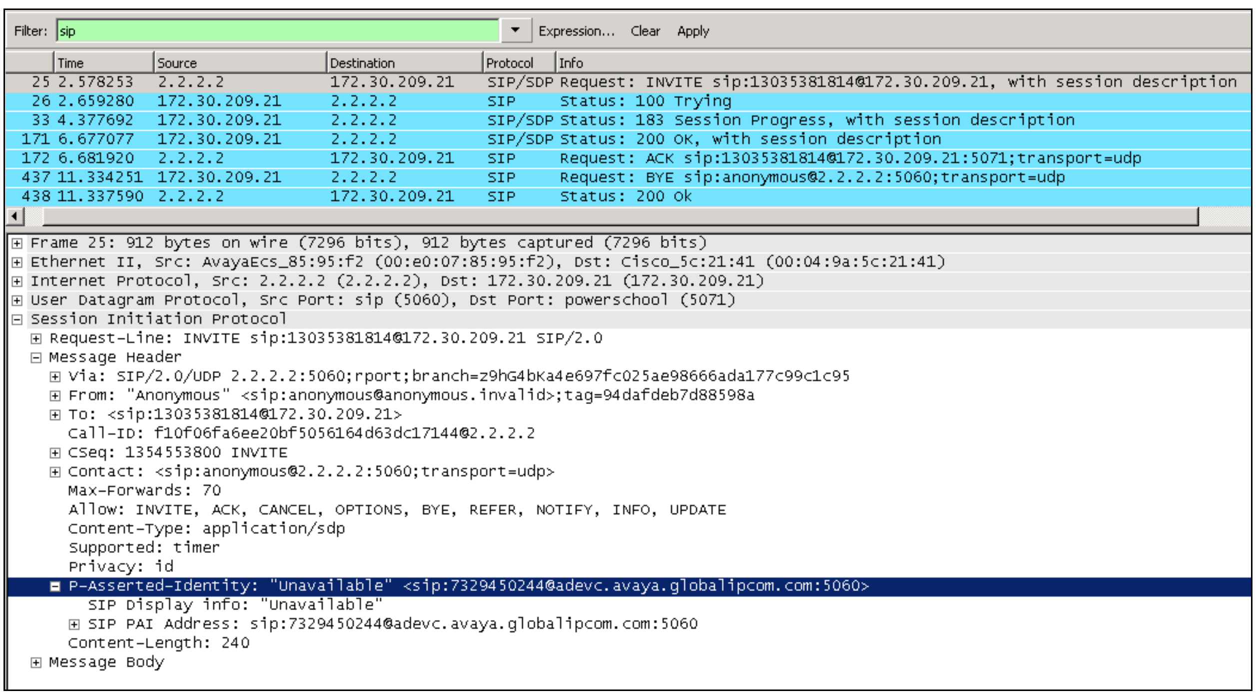
7.3. Wireshark Privacy Headers Verification

Section 5.9 outlined the options for user privacy. Calls wanting to restrict originating caller information will have their calls rejected from the Verizon network unless a valid P-Asserted-Identity field with a valid DID is presented to the network.

In the following Wireshark, the PAI has not been set. The user 732-945-0244 places a privacy call by dialing 8-1-303-538-1814. The P-Preferred Identity is sent, but not the P-Asserted-Identity. The caller receives a 408 Request Timeout from the Network and the call does not complete.



After the PAI is set on the **SIP LINE → SIP URI** tab to "Use Internal Data" or the No User Source Number with SIP_USE_PAI_FOR_PRIVACY, the call is again placed with 8-1-303-538-1814 and the P-Asserted-Identity is now included in the outgoing INVITE and the call is placed successfully.



8. Conclusion

IP Office is a highly modular IP telephone system designed to meet the needs of home offices, standalone businesses, and networked branch and head offices for small and medium enterprises. These Application Notes demonstrated how IP Office Release 8.1 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.

9. References

[1] IP Office 8.1 Installation Manual, Document Number 15-601042, August 2012
[2] IP Office Manager Manual 10.0, Document Number 15-601011, August 2012
[3] IP Office Release 8.1 Implementing Voicemail Pro, Document Number 15-601064, June, 2012
[4] IP Office System Status Application, Document Number 15-601758, November 2011
[5] Avaya IP Office Knowledgebase, http://marketingtools.avaya.com/knowledgebase

Product documentation for Avaya products may be found at http://support.avaya.com.

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 7.0 with Verizon IP Trunk Service Suite.

[VZB-IPT-IPOR7] Application Notes for Configuring SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service Offer and Avaya IP Office Release 7 – Issue 1.1
https://devconnect.avaya.com/public/download/dyn/VZBIPT-IPO7FT.pdf

10. Appendix A: SIP Line Template

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

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

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

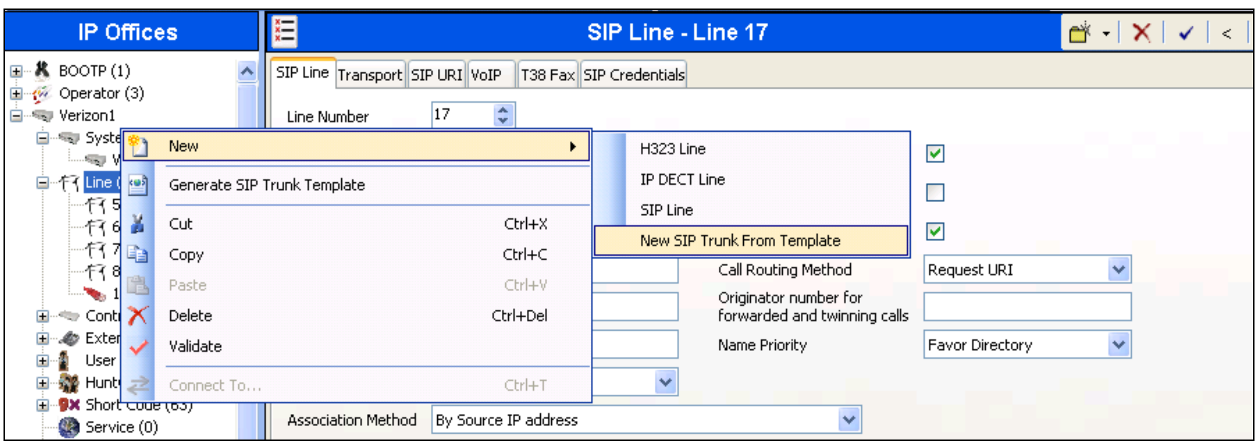

```
<?xml version="1.0" encoding="UTF-8"?>
-<Template xmlns="urn:SIPTrunk-schema"> <TemplateType>SIPTrunk</TemplateType>
<Version>20121109</Version> <SystemLocale>enu</SystemLocale>
<DescriptiveName>Vz_8.1</DescriptiveName>
<ITSPDomainName>adevc.avaya.globalipcom.com</ITSPDomainName>
<SendCallerID>CallerIDDIV</SendCallerID> <ReferSupport>true</ReferSupport>
<ReferSupportIncoming>1</ReferSupportIncoming>
<ReferSupportOutgoing>1</ReferSupportOutgoing>
<RegistrationRequired>false</RegistrationRequired> <UseTelURI>false</UseTelURI>
<CheckOOS>true</CheckOOS> <CallRoutingMethod>1</CallRoutingMethod>
<OriginatorNumber/> <AssociationMethod>SourceIP</AssociationMethod>
<LineNamePriority>FavourDirectory</LineNamePriority>
<UpdateSupport>UpdateNever</UpdateSupport> <UserAgentServerHeader/>
<CallerIDfromFromheader>false</CallerIDfromFromheader>
<PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
<ITSPProxy>pcelban0001.avayalincroft.globalipcom.com</ITSPProxy>
<LayerFourProtocol>SipUDP</LayerFourProtocol> <SendPort>5060</SendPort>
<ListenPort>5060</ListenPort> <DNSServerOne>172.30.209.4</DNSServerOne>
<DNSServerTwo>0.0.0.0</DNSServerTwo>
<CallsRouteViaRegistrar>true</CallsRouteViaRegistrar> <SeparateRegistrar/>
<CompressionMode>AUTOSELECT</CompressionMode>
<UseAdvVoiceCodecPrefs>true</UseAdvVoiceCodecPrefs> <AdvCodecPref>G.729(a) 8K
CS-ACELP,G.711 ULAW 64K,G.711 ALAW 64K</AdvCodecPref>
<CallInitiationTimeout>4</CallInitiationTimeout>
<DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
<VoipSilenceSupression>false</VoipSilenceSupression>
<ReinviteSupported>true</ReinviteSupported>
<FaxTransportSupport>FOIP_T38</FaxTransportSupport>
<UseOffererPrefferedCodec>true</UseOffererPrefferedCodec>
<CodecLockdown>false</CodecLockdown> <Rel100Supported>false</Rel100Supported>
<T38FaxVersion>3</T38FaxVersion> <Transport>UDPTL</Transport>
<LowSpeed>0</LowSpeed> <HighSpeed>0</HighSpeed>
```

<TCFMethod>Trans_TCF</TCFMethod> <MaxBitRate>FaxRate_14400</MaxBitRate>
<EflagStartTimer>2600</EflagStartTimer> <EflagStopTimer>2300</EflagStopTimer>
<UseDefaultValues>false</UseDefaultValues> <ScanLineFixup>true</ScanLineFixup>
<TFOPEnhancement>true</TFOPEnhancement> <DisableT30ECM>true</DisableT30ECM>
<DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
<DisableT30MRCompression>false</DisableT30MRCompression>
<NSFOverride>false</NSFOverride> </Template>

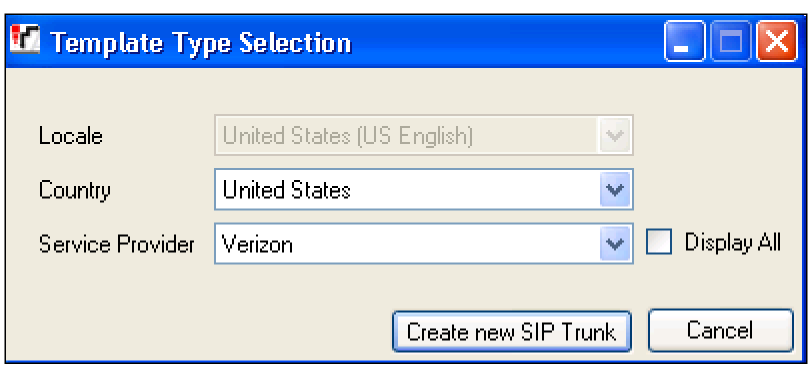
To import the above template into a new installation:

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

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



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



Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.