Avaya Solution & Interoperability Test Lab

Application Notes for SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service and Avaya IP Office Release 9.1 – Issue 1.1

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 9.1 Preferred Edition, Avaya Voicemail Pro, Avaya Communicator for Windows, and Avaya SIP, H.323, digital, and analog endpoints.

These Application Notes complement previously published Application Notes by illustrating the configuration screens and Avaya testing of IP Office Release 9.1.

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 Solution & Interoperability Test Lab, utilizing a Verizon Business Private IP (PIP) circuit connection to the production Verizon Business IP Trunking service.

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as the observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

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 9.1 Preferred Edition, Avaya Voicemail Pro, Avaya Communicator for Windows, and Avaya SIP, H.323, digital, and analog endpoints.

These Application Notes complement previously published Application Notes by illustrating the configuration screens and Avaya testing of IP Office Release 9.1.

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 also applies to IP Trunk services delivered via IDA service terminations.

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

2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site 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.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

To verify SIP trunking interoperability, the following features and functionality were covered during the interoperability compliance test:

- 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 Communicator for Windows, and Avaya Voicemail Pro. The display of caller ID on display-equipped Avaya IP Office telephones was verified.
- Incoming calls answered by members of circular 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, Avaya Communicator for Windows and Avaya Voicemail Pro. The display of caller ID on display-equipped PSTN telephones was verified.
- Inbound / Outbound fax using G.711 and T.38 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 SIP trunk health using SIP OPTIONS.
- 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 was selected upon failure of a primary line.
- Incoming and outgoing calls using the G.729A and G.711MU 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 Avaya Voicemail Pro.
- Inbound and outbound long holding time call stability.
- Telephony features such as call waiting, hold, transfer, and conference.
- Attended call transfer using the SIP REFER method.
- Unattended, or “blind” call transfer using the SIP REFER method.

- 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).
- DiffServ markings in accordance with network requirements for IP Office SIP signaling and RTP media.
- Mobility Features such as Mobile Callback and Mobile Call Control.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results. The following observations were noted.

1. **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 expected behavior of IP Office, and transparent to the users on the call.

2. **SIP endpoint RFC2833:** Although Avaya IP Office can specify the RFC2833 Telephone Event 101 to use for Analog/Digital and H.323 sets, (see **Section 5.2.6**), it was found that the Avaya Communicator for Windows uses Telephone Event 120. No issues were found during testing as a result of this behavior.

3. **SIP endpoint transfers:** When Refer based call transfers are performed, Verizon does not send NOTIFY SIP messages to Avaya IP Office to signal transfer completion. Some Avaya SIP endpoints (e.g., Avaya 1140E, and Avaya Communicator for Windows) require receipt of a NOTIFY when Refer based call transfers are performed. Setting the IP Office SIP Line option, **Emulate NOTIFY for Refer**, will send the necessary NOTIFY messages to these endpoints (see **Section 5.4.8**).

2.3. Support

2.3.1. Avaya

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

2.3.2. Verizon

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

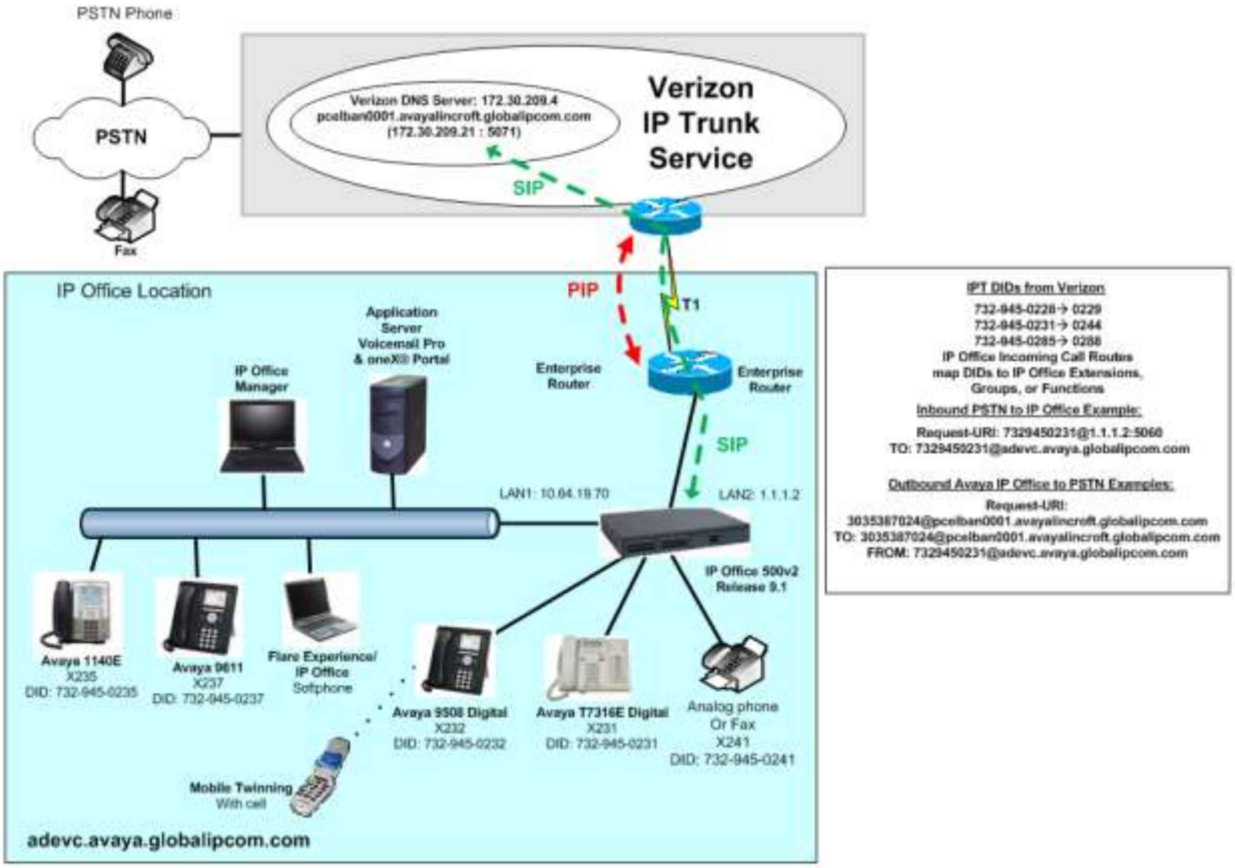


Figure 1: Avaya Interoperability Test Lab Configuration

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**.

Verizon Provided DID	Avaya IP Office Destination	Notes
732-945-0231	X 231	T7316E Digital Telephone
732-945-0232	X 232	9508 Digital Telephone
732-945-0234	X 234	Avaya Communicator for Windows
732-945-0235	X 235	Avaya SIP 1140E
732-945-0237	X 237	Avaya H.323 - 9611
732-945-0239	Voicemail	
732-945-0240	Short Code: FNE31	FNE Service 31 (Mobile Call Control)
732-945-0241	X 241	Analog telephone or Fax machine
732-945-0242	X 401 Hunt Group	Rotary Ring Mode to all Users

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

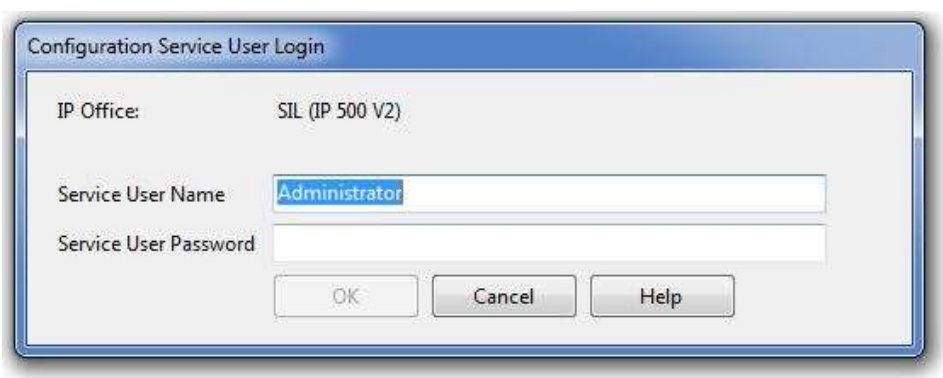
Avaya IP Telephony Solution Components	
Equipment	**Software**
Avaya IP Office 500 V2	Release 9.1.100.10
Avaya Application Server	Release 9.1.1.0.10
Avaya IP Office Manager	Release 9.1.1.0 Build 10
Avaya 9611SW IP Telephone (H.323)	Release 6.2209
Avaya 1140E IP Telephone (SIP)	Release 04.04.18
Avaya 9508 Digital Telephone	Release 0.55
Avaya T7316E Digital Telephone	N/A
Avaya Communicator for Windows	Release 2.0.3.30

Table 2: Equipment and Software Tested

Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2, and also when deployed with all configurations of IP Office Server Edition without T.38 Fax service (T.38 fax is not supported on IP Office Server Edition). Note that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks.

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. Provided that the IP Office system is accessible to IP Office Manager, the following will be displayed in the center of the opening screen:

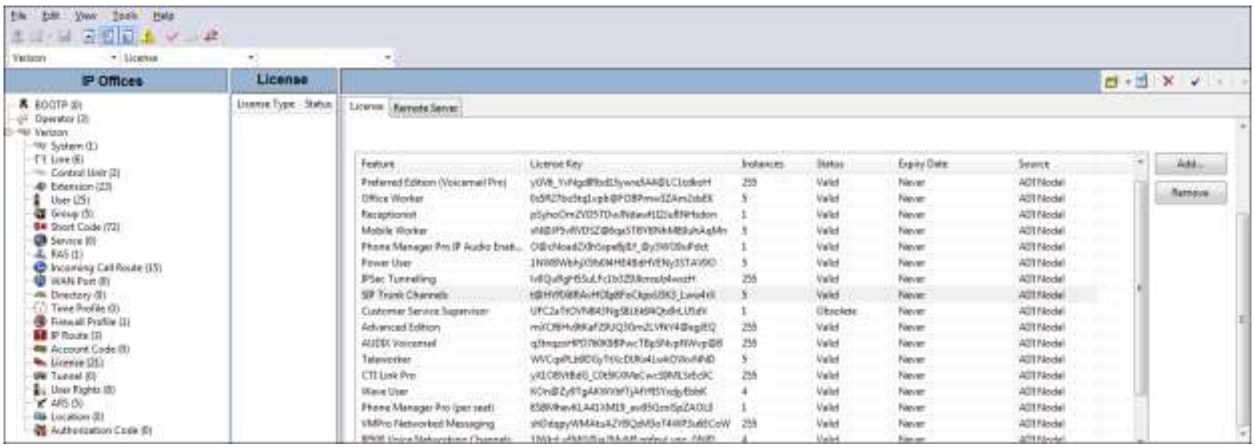


Log in with the appropriate configuration credentials. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details pane on the right side.

5.1. Licensing and Physical Hardware

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. Confirm a valid **SIP Trunk Channels** license with sufficient **Instances** (trunk channels). If Avaya IP Telephones will be used as is the case in these Application Notes, verify the **Avaya IP endpoints** license.



In the sample configuration, looking at the IP Office 500 V2 from left to right, the first module is a TCM 8 Digital Station Module. This module supports BCM / Norstar T-Series and M-Series telephones. The second module is a COMBO6210/ATM4 module. This module is used to add a combination of ports to an IP500 V2 control unit and is not supported by IP500 control units. The module supports 10 voice compression channels. Codec support is G.711, 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 T7315E telephone with extension 231 is connected to port 1 of the TCM8 module, and the Avaya 9508 telephone with extension 232 is connected to port 1 of the “Combo” card. The analog extension or fax machine is connected to the “Combo” card on port 7.

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

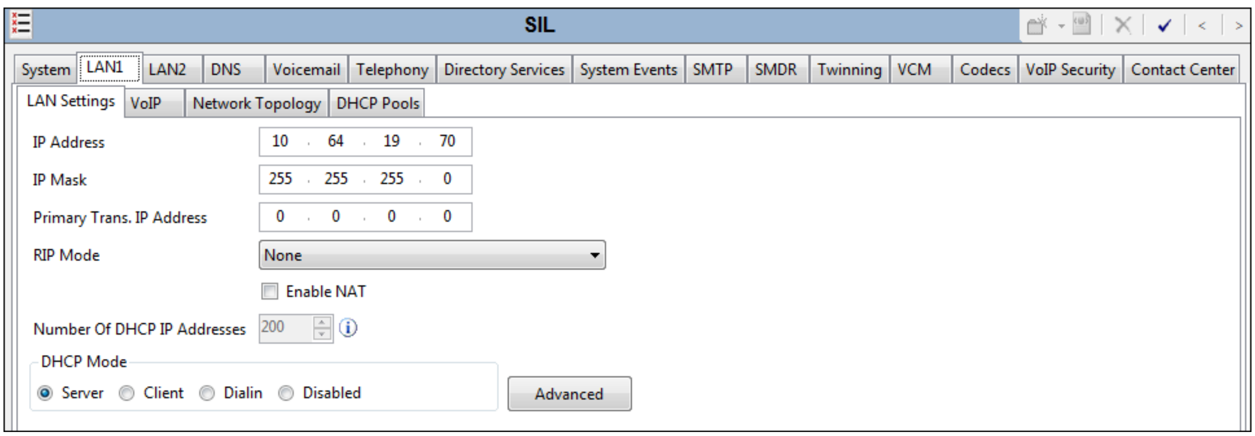
5.2. System Settings

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. For all of the following configuration sections, the **OK** button (not shown) must be selected in order for any changes to be saved.

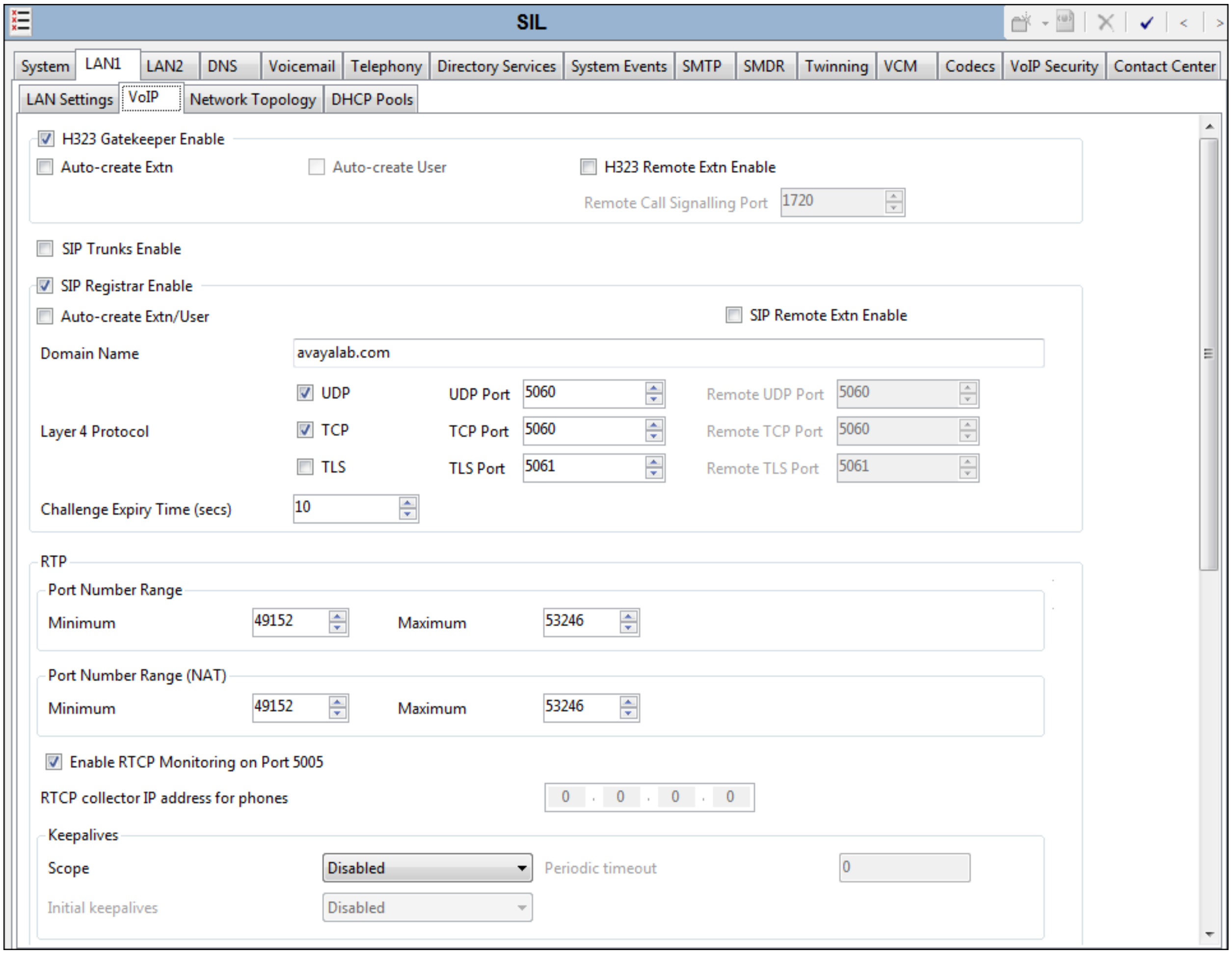
5.2.1. LAN 1 Settings

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

In the sample configuration, LAN1 is 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 10.64.19.70. **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.

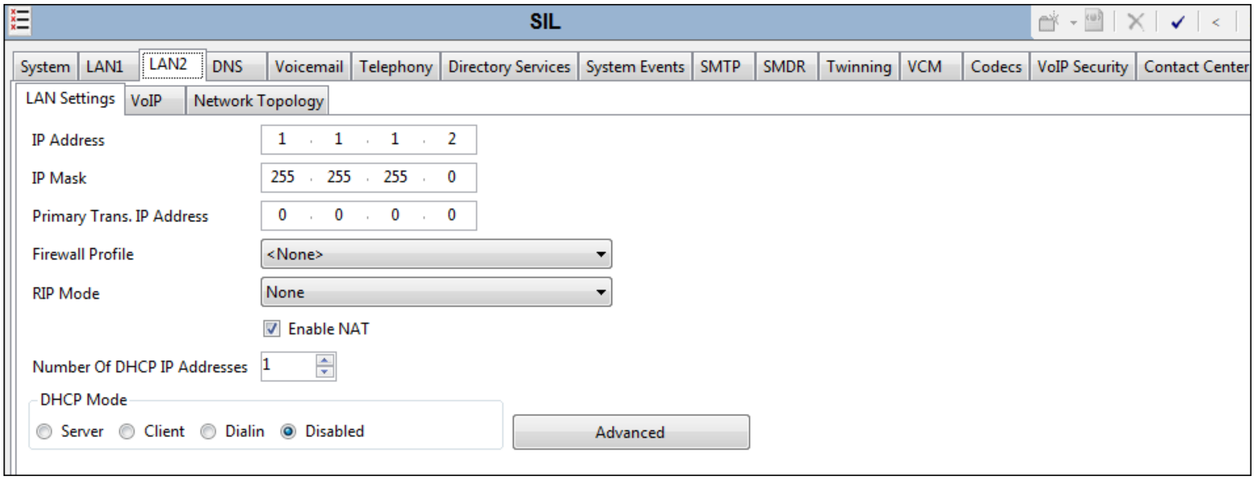


Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 9600-Series Telephones used in the sample configuration. The **SIP Registrar Enable** box is checked to allow the use of Avaya 1140E and Avaya Communicator for Windows.



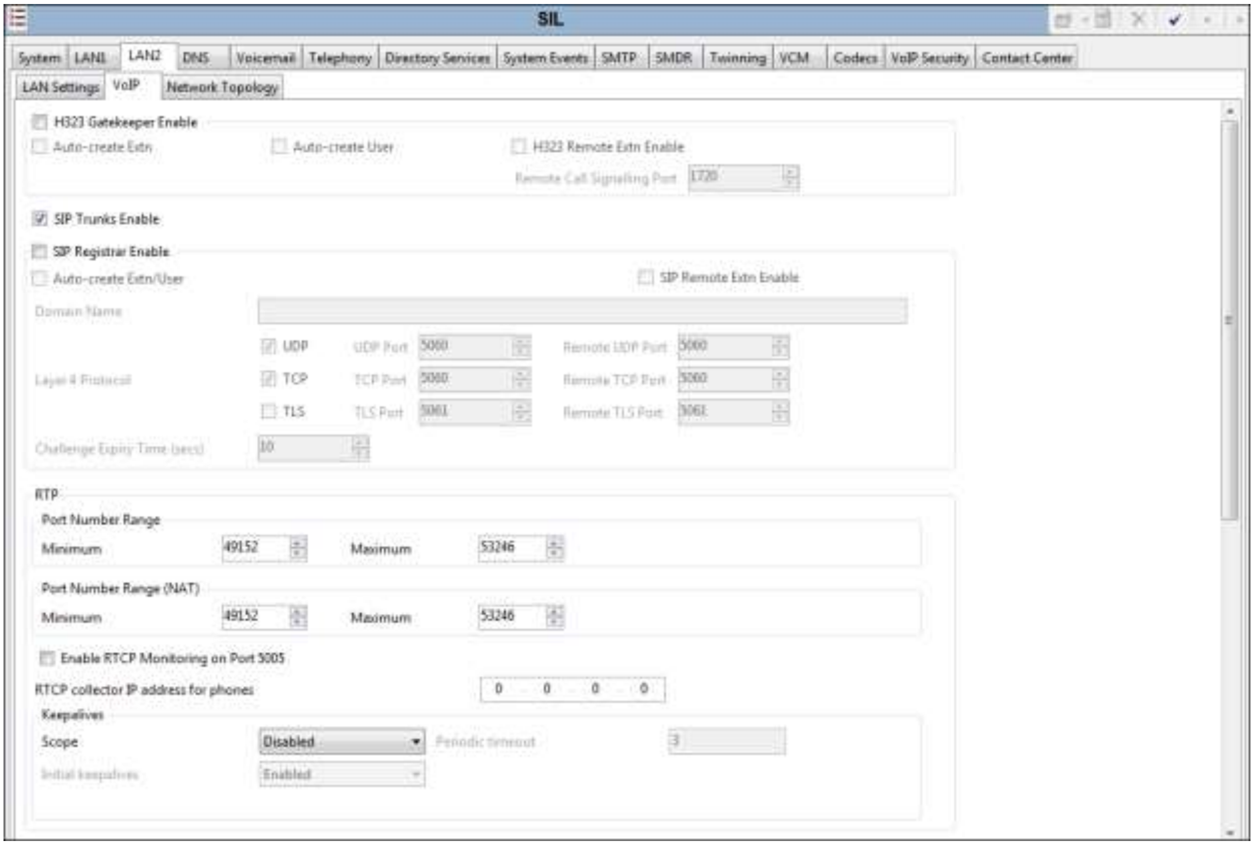
5.2.2. LAN 2 Settings

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

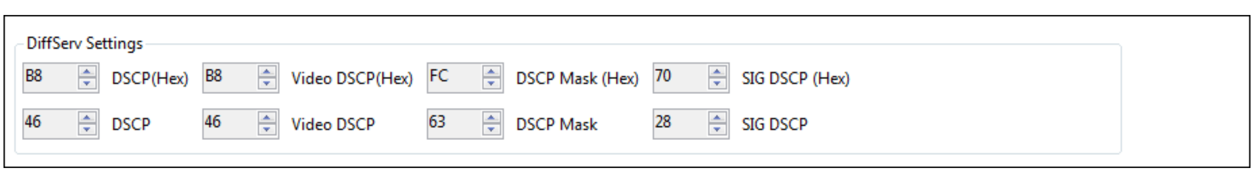


Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** and **SIP Registrar Enable** boxes are unchecked since IP telephones will not be registering on this link. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Verizon Business.

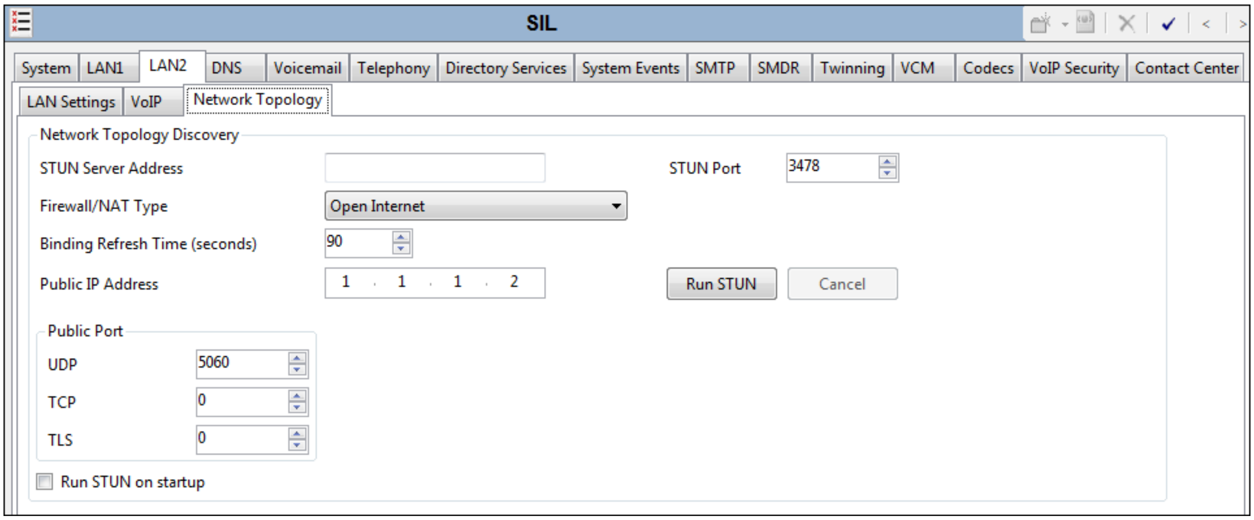
If desired, the **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media paths from Verizon Business to IP Office. The defaults are used here.



Scrolling down, 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.

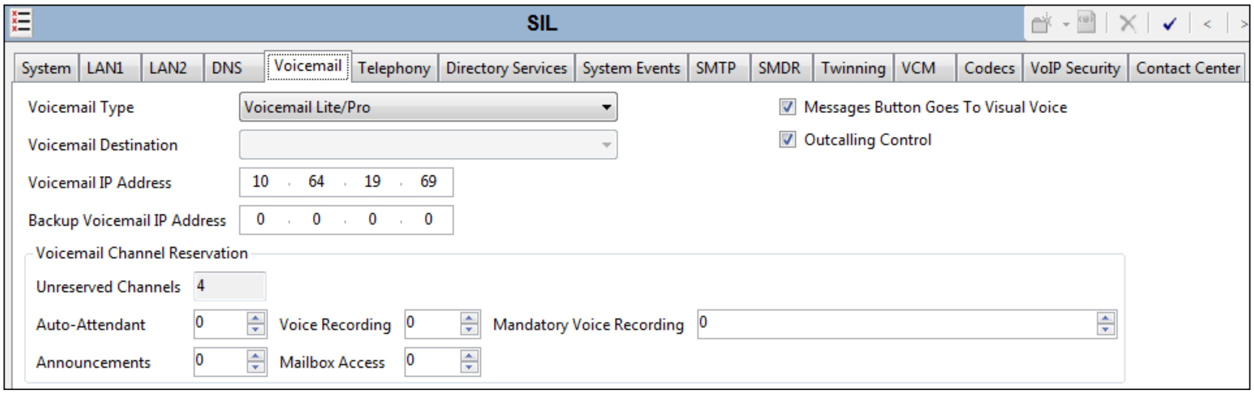


Select the **Network Topology** tab as shown in the following screen. The **Firewall/NAT Type** is set to **Open Internet** in the sample configuration. Note that the **Firewall/NAT Type** parameter may need to be set differently, depending on the type of firewall or Network Address Translation device used at the customer premise. The **Binding Refresh Time (seconds)** can be used to lower the SIP OPTIONS timing from the default of 300 seconds. During the testing, the Binding Refresh Time was varied (e.g., 90 seconds, 120 seconds) to test SIP OPTIONS timing. The **Public IP Address** is set to the IP address known to Verizon. In the sample configuration, this is 1.1.1.2. The **UDP Public Port** is set to "5060".



5.2.3. Voicemail Settings

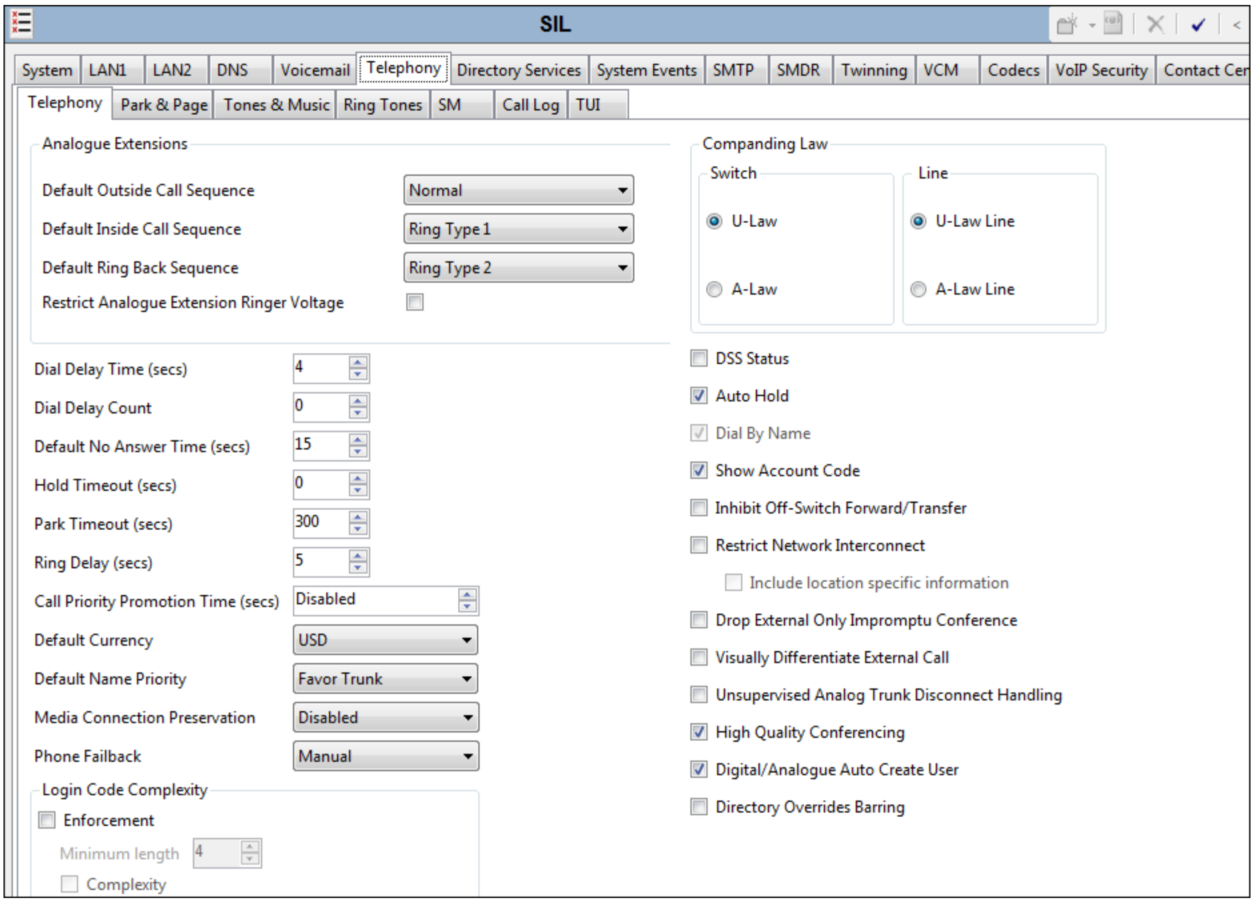
To view or change voicemail settings, select the **Voicemail** tab as shown in the following screen. The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. The **Voicemail Type** in the sample configuration is “Voicemail Lite/Pro”. Other Voicemail types may be used. The Voicemail IP Address in the sample configuration is 10.64.19.69, the IP address of the IP Office Application Server 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 5.5**).

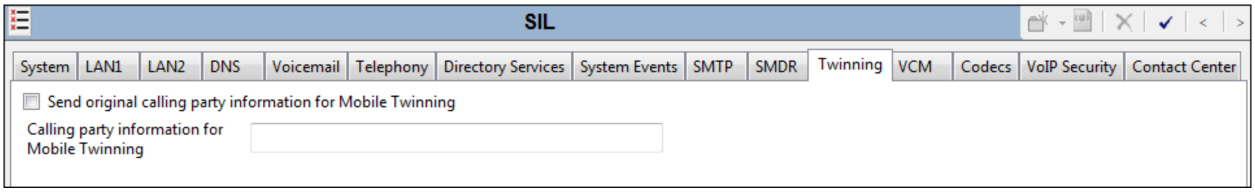
5.2.4. System Telephony Configuration

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



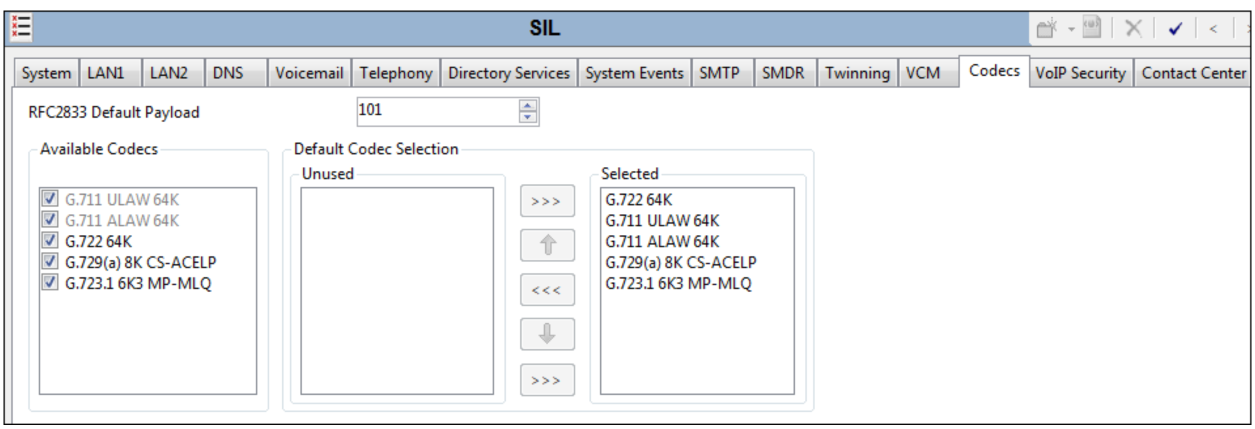
5.2.5. System Twinning Configuration

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 5.4.3**), 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.



5.2.6. System Codecs Configuration

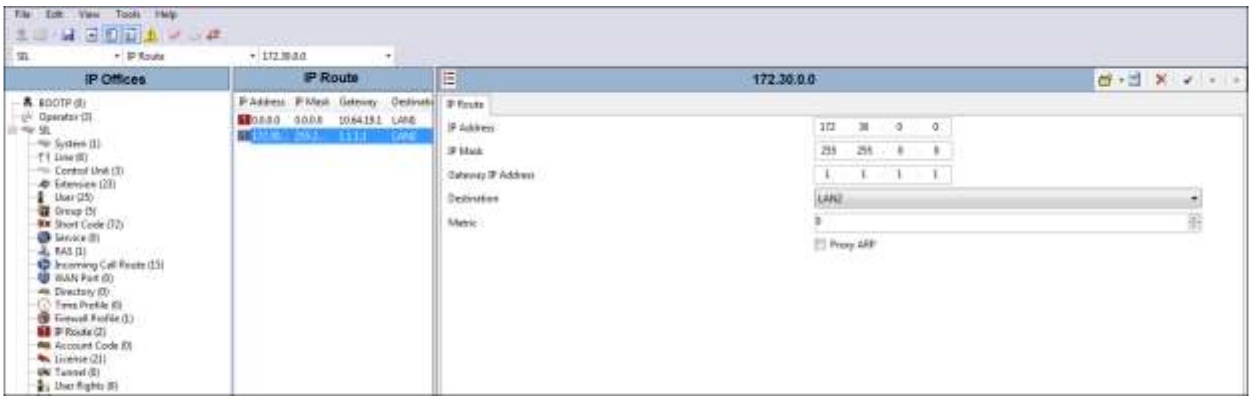
To view or change system codec settings, select the **Codecs** tab. On the left, observe the list of **Available Codecs**. In the example screen below, which is not intended to be prescriptive, the box next to each codec is checked, making all the codecs available in other screens where codec configuration may be performed (such as the SIP Line in **Section 5.4.6**). 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. Set the **RFC2833 Default Payload** parameter to "101", the value preferred by Verizon Business.



5.3. IP Route

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 10.64.19.1.

The IP Office LAN2 port is physically connected to the Verizon PIP network and has a default gateway of 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 route using **Destination** "LAN2".



5.4. SIP Line

This section shows the configuration screens for the SIP Line in IP Office Release 9.1. Since IP Office Release 9.1 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 [IPOR81] and [IPOR9].

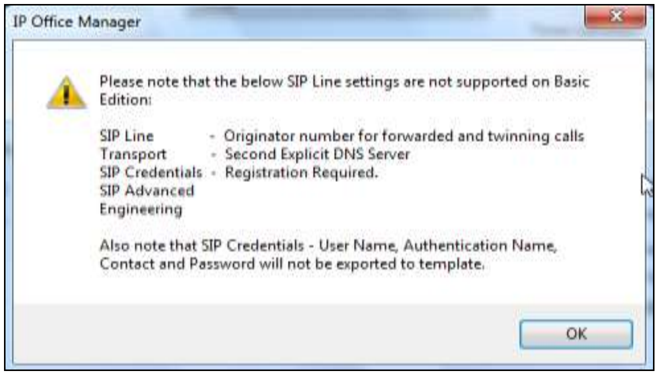
The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.4.2** to create the SIP Line from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:

- IP addresses
- SIP Credentials (if applicable)
- SIP URI entries
- Setting of the **Use Network Topology Info** field on the Transport tab.

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.4.3 – 5.4.8**.

In addition, the following SIP Line settings are not supported on Basic Edition:



Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.4.3 – 5.4.8**.

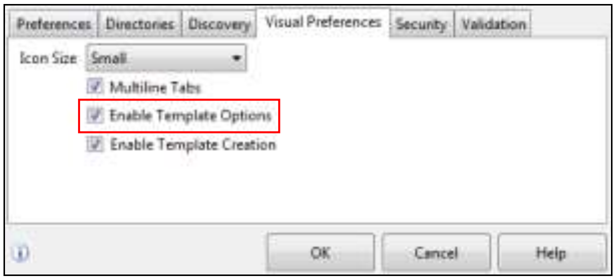
5.4.1. Importing a SIP Line Template

Note – DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500v2) and IP Office Server Edition systems. Alternatively, binary templates may be generated. However, binary templates include all the configuration parameters of the Trunk, including sensitive customer specific information. Therefore, binary templates should only be used for cloning trunks within a specific customer's environment.

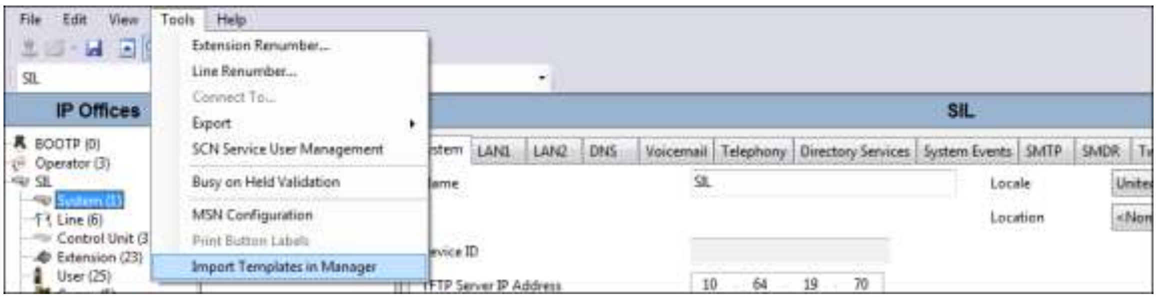
1. Copy a previously created template file to a location (e.g., *temp*) on the same computer where IP Office Manager is installed. By default, the template file name will have the format **AF_<*user supplied text*>_SIPTrunk.xml**, where the **<*user supplied text*>** portion is entered during template file creation.

   Note – If necessary, the **<*user supplied text*>** portion of the template file name may be modified, however the **AF_<*user supplied text*>_SIPTrunk.xml** format of the file name must be maintained. For example, an original template file **AF_*TEST* _SIPTrunk.xml** could be changed to **AF_*Test1*_SIPTrunk.xml**. The template file name is selected in **Section 5.4.2** to create a new SIP Line.

2. Verify that Template Options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the IP Office Manager Preferences window that appears, select the **Visual Preferences** tab. Check the box next to **Enable Template Options**. Click **OK**.

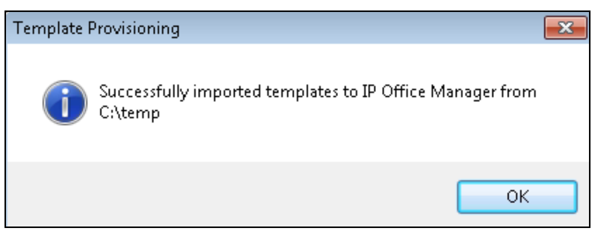


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**.



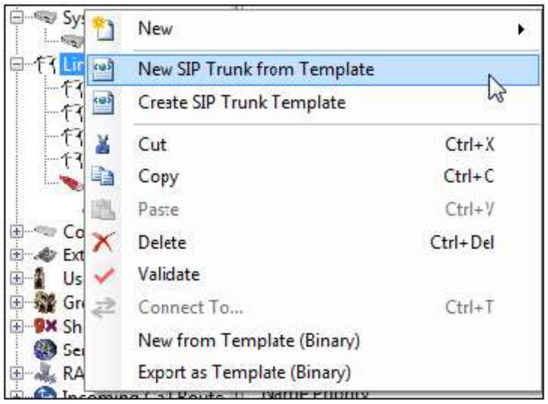
4. A folder browser will open (not shown). Select the directory used in **step 1** to store the template(s) (e.g., *temp*). In the reference configuration, template file **AF_VerizonIPT_SIPTrunk.xml** was imported. The template files are automatically copied into the IP Office default template location, **C:\Program Files\Avaya\IP Office\Manager\Templates**.

5. After the import is complete, a final import status pop-up window will open stating success or failure. Click **OK**.



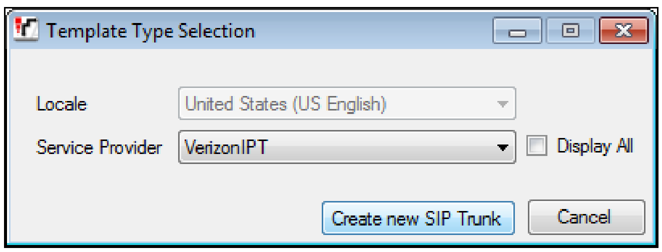
5.4.2. Creating a SIP Trunk from an XML Template

1. To create the SIP Trunk from a template, right-click on **Line** in the Navigation Pane, and select **New SIP Trunk from Template**.

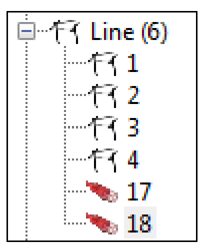


2. In the subsequent **Template Type Selection** pop-up window, from the **Service Provider** pull-down menu, select the XML template name from **Section 0**. Click **Create new SIP Trunk**.

Note – The drop down menu will display the **<*user supplied text*>** part of the template file name (see **Section 0**). If the **Display All** box is checked, then the full template file name is displayed.



The newly created SIP Line will appear in the Navigation pane (e.g., SIP Line **18**).



Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.4.3 – 5.4.8**.

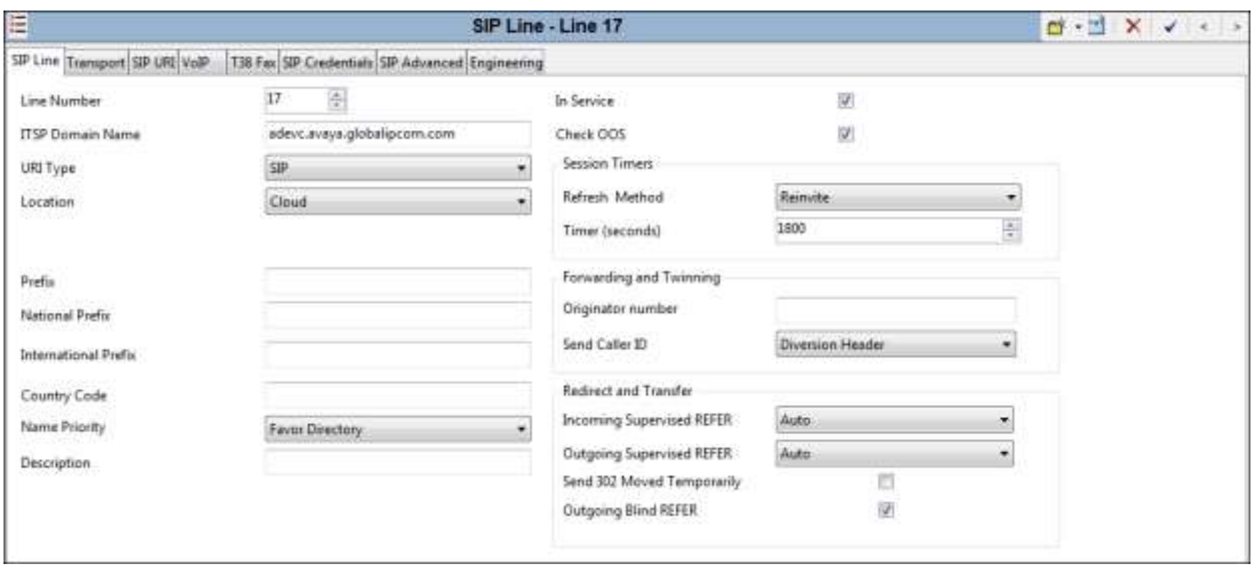
5.4.3. SIP Line – SIP Line Tab

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** is configured to the domain supplied by Verizon (advec.avaya.globalipcom.com). 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 LAN2, as shown in **Section 5.2.2**.

Under **Forwarding and Twinning**, the **Send Caller ID** parameter is set to “Diversion Header”. With this setting and the related configuration in **Section 5.2.5**, 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. 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. IP Office will also include the Diversion header for calls that are call forwarded out the SIP Line to Verizon.

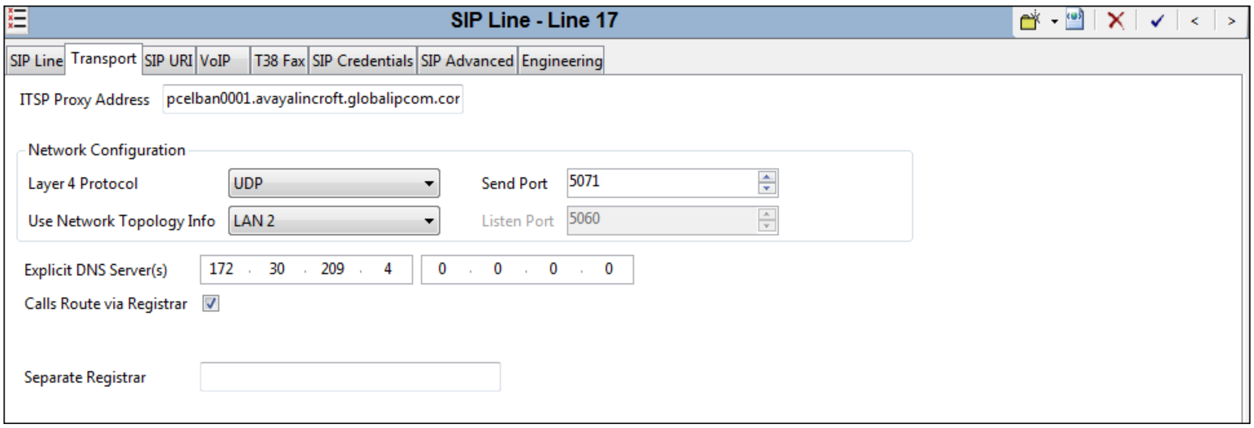
Under **Session Timers**, the **Refresh Method** is set to “Reinvite” and the **Timer (seconds)** is set to “1800”. With this configuration, IP Office will send re-INVITEs every 15 minutes (half of the set value) to keep the active session alive.

Under **Redirect and Transfer**, The default automatic determination of **Incoming Supervised REFER** and **Outgoing Supervised REFER** is “Auto”. Alternatively, the default can be overridden with “Never” to explicitly disable use of supervised REFER, or “Always” to explicitly enable use of supervised REFER. The **Send 302 Moved Temporarily** setting is unchecked, as Verizon does not support receiving a 302 Moved Temporarily message. Verizon does support the REFER method for blind transfers, so the **Outgoing Blind REFER** box can be checked to enable this feature on IP Office. With this feature enabled, IP Office will not send an INVITE to initiate the transfer, instead only send a REFER on the current active call. This instructs Verizon to perform the transfer by initiating the new call and release the current call with IP Office. This is optional for Verizon Business IP Trunk service, in that if the Supervised REFER settings were set to “Always”, IP Office will still send a REFER for blind transfers, but will first send an INVITE for a new call to initiate the transfer.



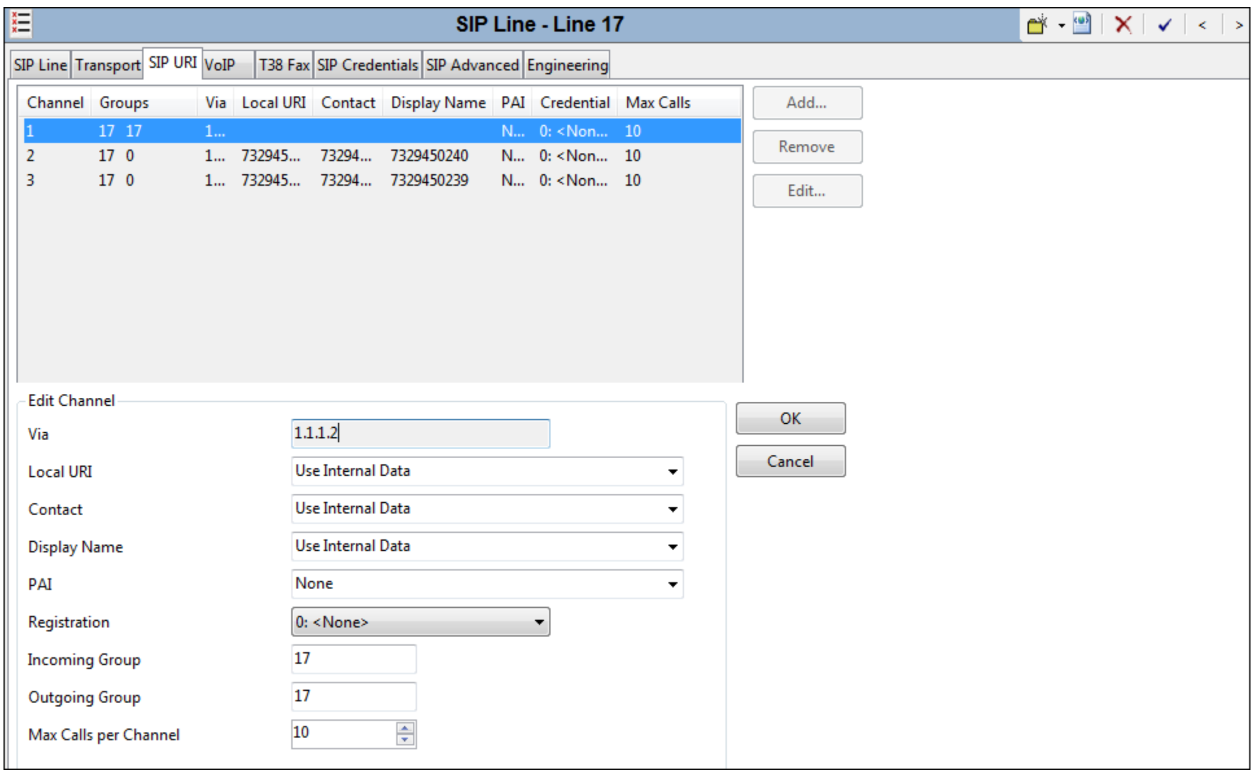
5.4.4. SIP Line - Transport Tab

Select the **Transport** tab. 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 IP address provided by Verizon Business (i.e., 172.30.209.21) may be used in place of this domain. In the **Network Configuration** area, "UDP" is selected as the **Layer 4 Protocol** and the **Send Port** is set to the port number provided by Verizon Business. As shown in **Figure 1**, the port is 5071 in the sample configuration. The **Use Network Topology Info** parameter is set to "LAN 2", the LAN port connected to the Verizon PIP network. 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.



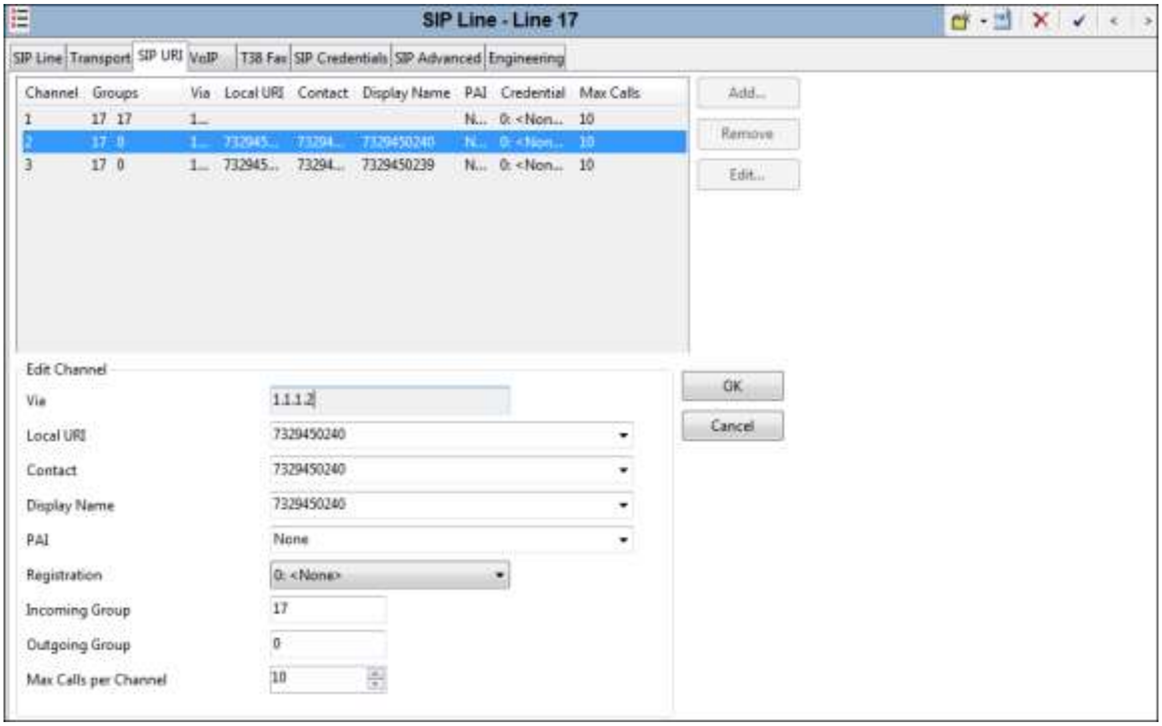
5.4.5. SIP Line - SIP URI Tab

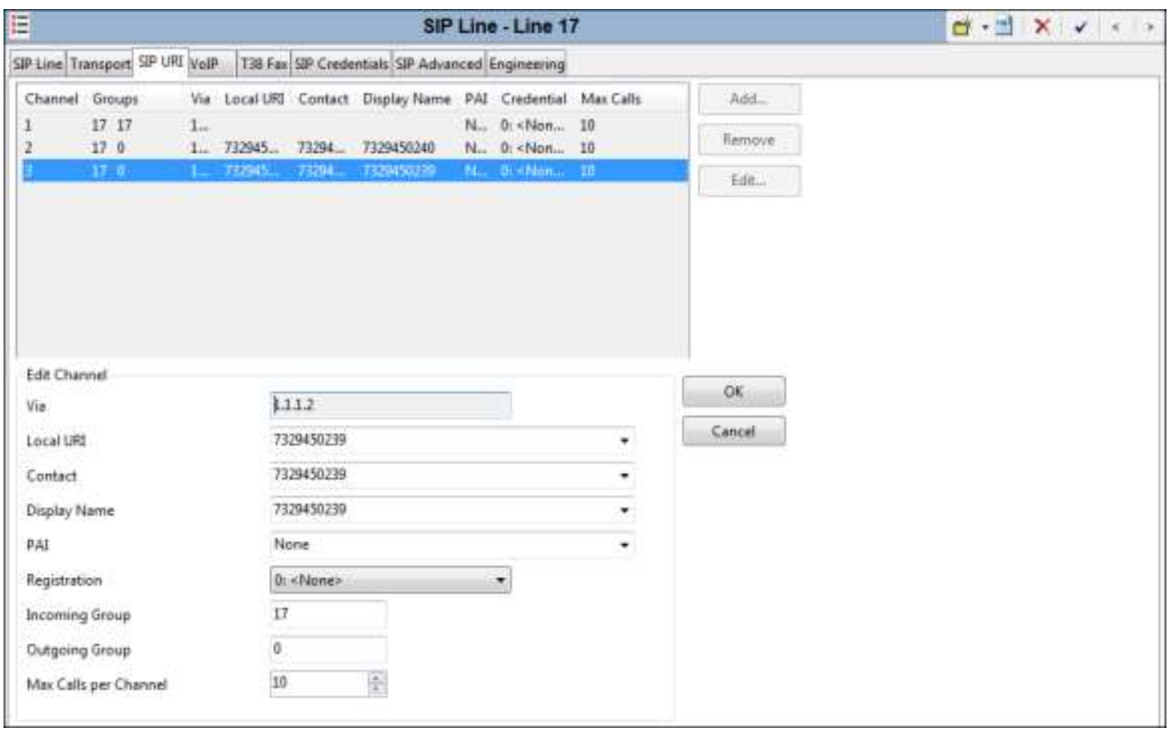
Select the **SIP URI** tab. To add a new SIP URI, click the **Add…** button. In the bottom of the screen, a New Channel area will be opened. To edit an existing entry, click an entry in the list at the top, and click the **Edit…** button. In the bottom of the screen, the Edit Channel area will be opened. 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 set to "None". 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 17, 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 17, 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 10, sets the maximum number of simultaneous calls that can use the URI before IP Office returns busy to any further calls.



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 732-945-0239 and 732-945-0240 will be assigned as service numbers in the Incoming Call Routes in **Section 5.7**.

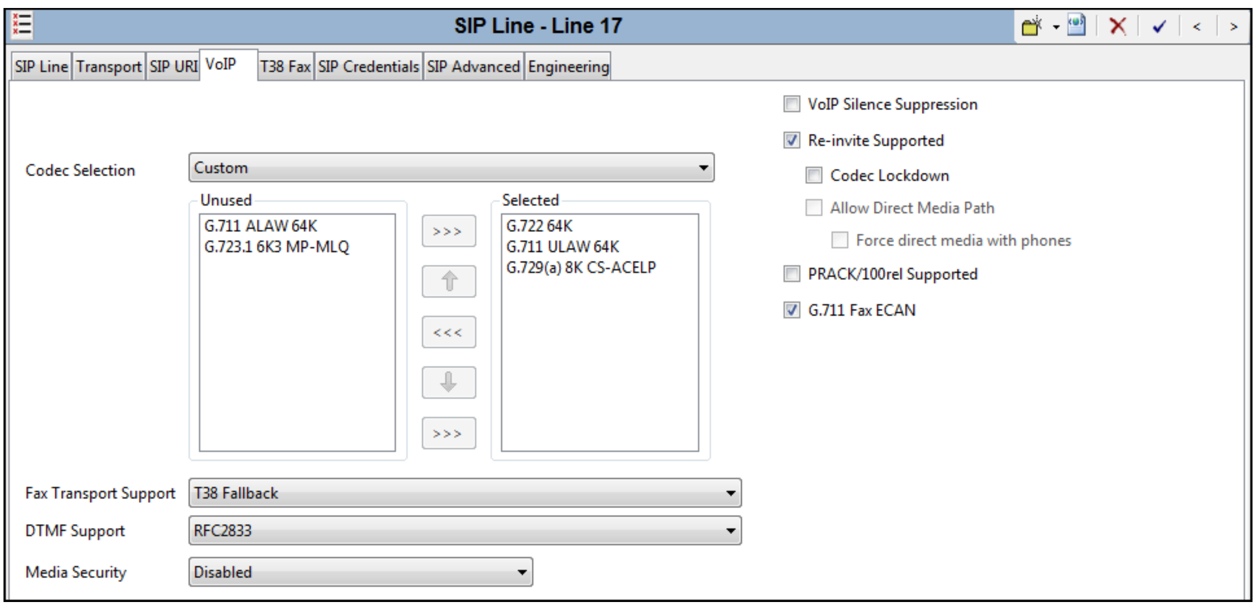




5.4.6. SIP Line - VoIP Tab

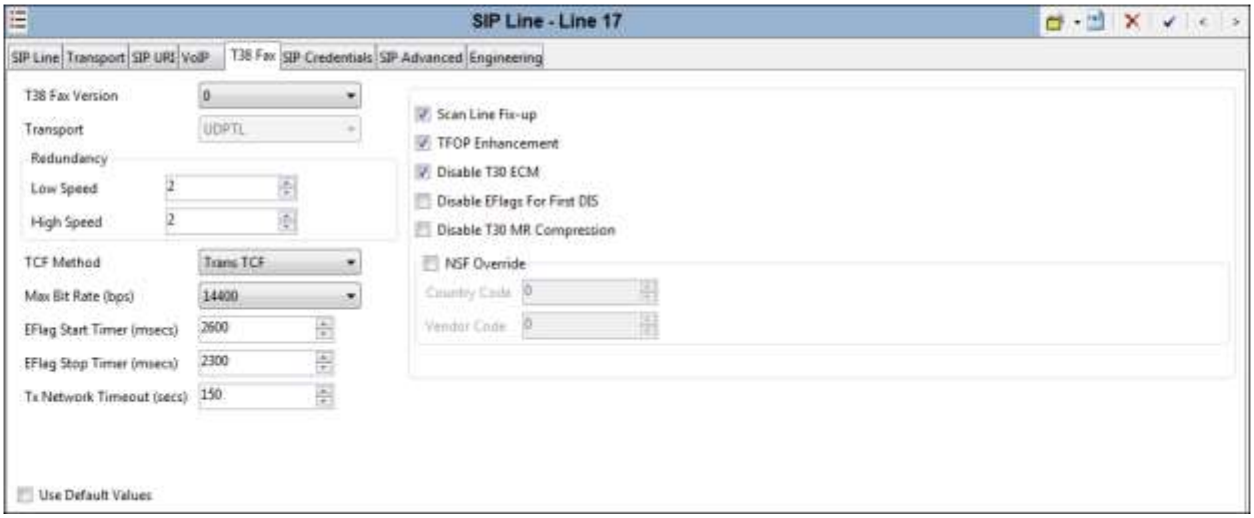
Select the **VoIP** tab. The **Codec Selection** drop-down box **System Default** (default) will match the codecs set in the system wide Default Selection list (**System → Codecs**). In the sample configuration, **Custom** is selected and codecs preferred by Verizon are included (i.e., G.722 64K, G.711 ULAW 64K, and G729(a) 8K CS-ACELP). This will cause IP Office to include G.722, G.711MU and G.729a 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. For PSTN originations, Verizon preferred the G.729a codec in the SDP, while also allowing the G.711MU codec. However, if an originator is at another Verizon 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.729a was the preferred or only codec listed, and G.729a calls were also successfully verified.

If the SIP Line will be used for any fax calls, the **G.711 Fax ECAN** parameter should be checked and **G.711 ULAW 64K** set as the first codec preference. Also, 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.



5.4.7. SIP Line – T38 Fax Tab

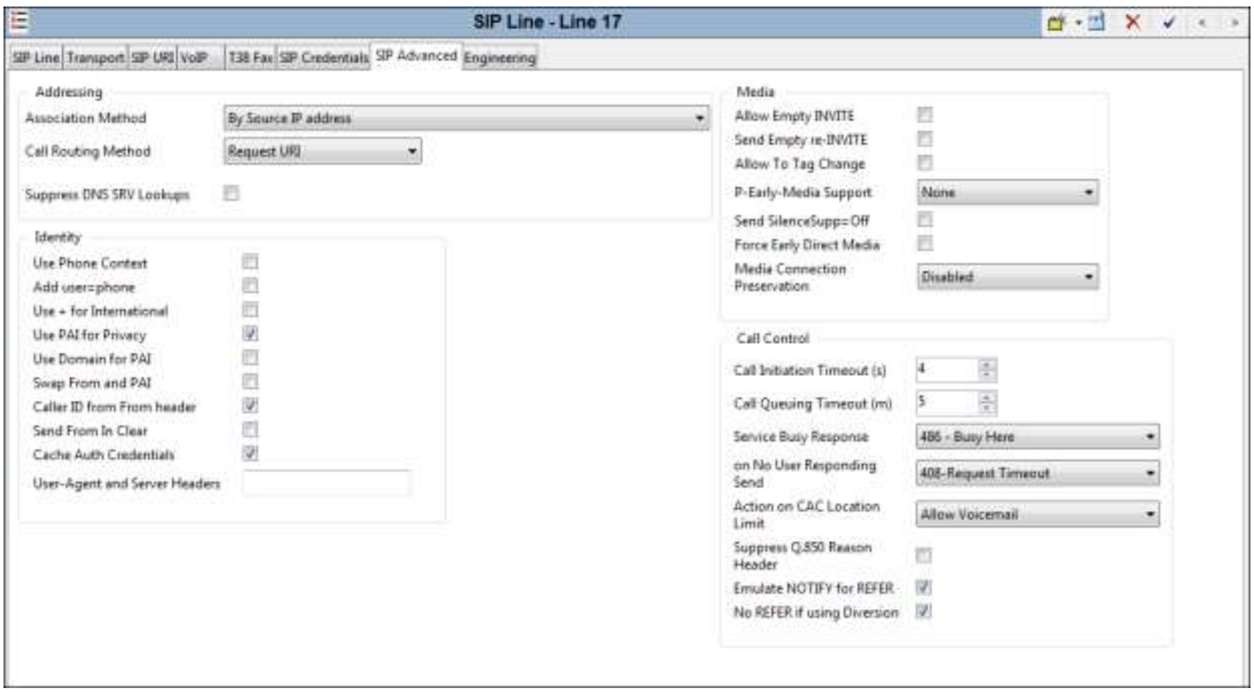
The settings on this tab are only accessible if **Re-invite Supported** is checked and a value for **Fax Transport Support** other than "None" are selected on the VoIP tab. Fax relay is only supported on IP500 V2 systems with an IP500 VCM card. The **Disable T30 ECM** must be checked or fax errors may be experienced when using T38 Fax. When selected, it disables the T.30 Error Correction Mode used for fax transmission. The **T38 Fax Version** is set to "0". In the **Redundancy** area, the **Low Speed** and **High Speed** parameters are set to "2". All other values are left at default. Since the Verizon Business IP Trunk service does not require registration, the **SIP Credentials** tab need not be visited.



5.4.8. SIP Line – SIP Advanced Tab

Select the **SIP Advanced** tab. In the **Identity** area, the **Use PAI for Privacy** box is checked to include the caller's DID number in the P-Asserted-Identity (PAI) SIP header for a privacy requested call. This PAI SIP header is required by Verizon Business to admit an otherwise anonymous caller to the network. The **Caller ID from From header** box is checked to have IP Office use the Caller ID information in the From SIP header rather than the PAI or Contact SIP header for inbound calls. This will allow the Caller Name presented in the From SIP header by Verizon Business to also be included in the Caller ID.

In the **Call Control** area, the **Emulate NOTIFY for Refer** box is check. This is required for SIP endpoints that perform Refer based transfers across the SIP line. See **Section 2.2** for more details. The **No Refer if using Diversion** box is check to prevent IP Office from using the SIP REFER method on call forwarded scenarios that use a Diversion SIP header. Verizon does not support this type of refer, and would respond with a "603 Decline" SIP message.



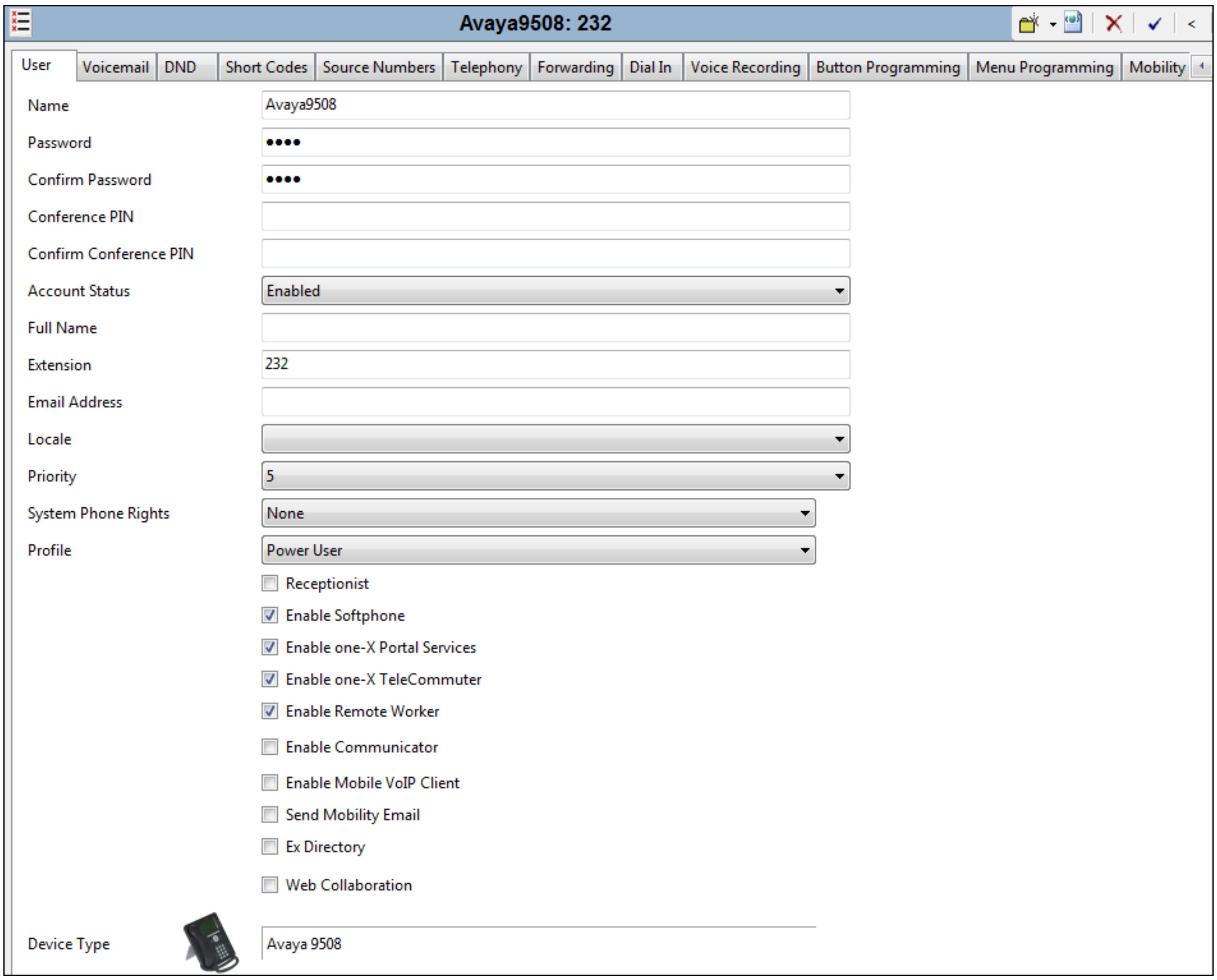
Note – An IP Office user whose calling line identification is not typically withheld from the network can request privacy in the sample configuration by dialing the short code *67 to access the SIP Line, as described in **Section 5.6**. Certain Avaya telephones can also request privacy, without dialing a unique short code, using **Features → Call Settings → Withhold Number**. The **Withhold Number** parameter may be set to "On" (i.e., for privacy). Specific users may be configured to always withhold calling line identification by checking the **Anonymous** field in the **SIP** tab for the user (**Section 5.5.1**).

5.5. Users, Extensions, and Hunt Groups

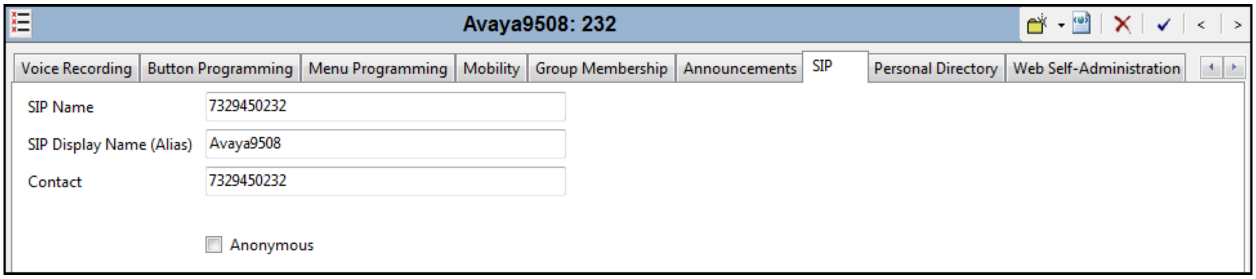
In this section, examples of IP Office Users, Extensions, and Groups will be illustrated. In the interests of brevity, not all users and extensions shown in **Figure 1** will be presented, since the configuration can be easily extrapolated to other users. To add a User, right click on **User** in the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

5.5.1. User 232 (Digital)

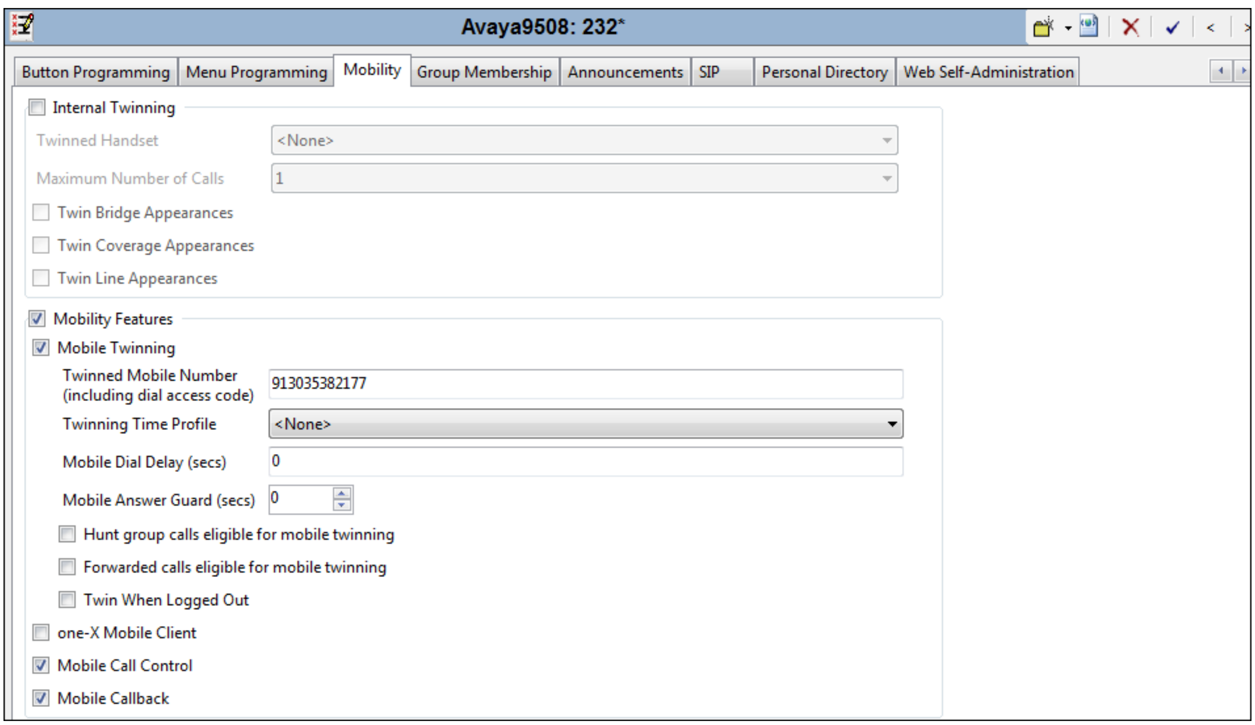
The following screen shows the **User** tab for User 232. As shown in **Figure 1**, this user corresponds to the Avaya Digital 9508.



The following screen shows the **SIP** tab for User 232. The **SIP Name** and **Contact** parameters are configured with the DID number of the user, 732-945-0232. 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.



From **Figure 1**, note that user 232 will use the Mobile Twinning feature. The following screen shows the **Mobility** tab for User 232. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, including the dial access code for ARS, in this case 913035382177. Other options can be set according to customer requirements.

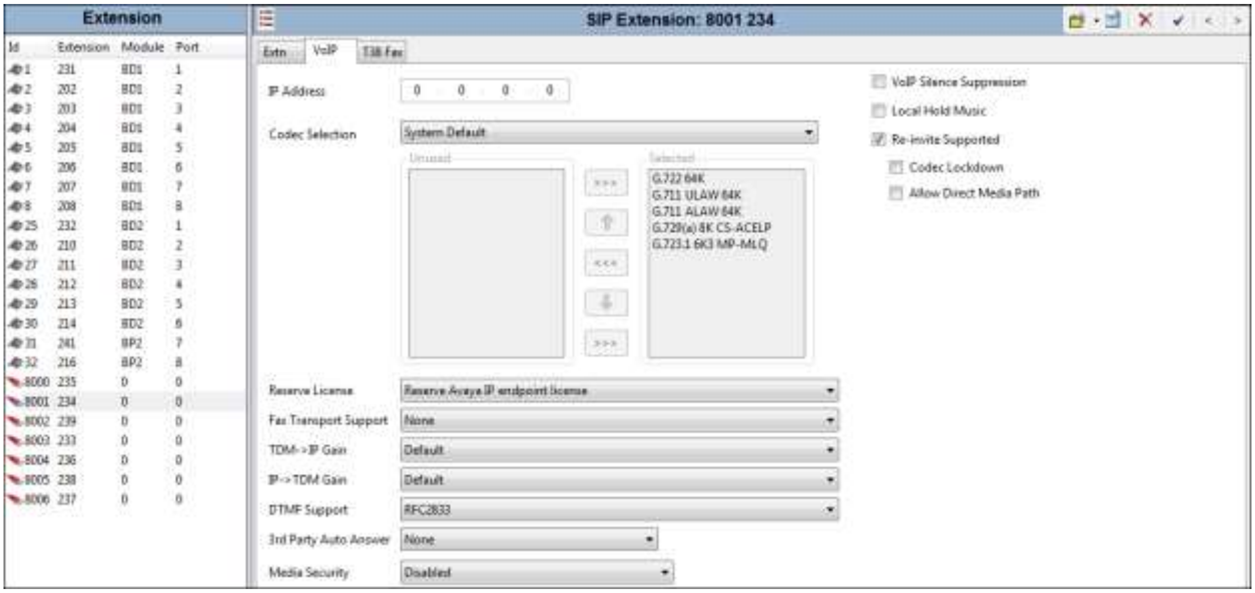


The following screen shows the Extension information for this user. To view, select **Extension** from the Navigation pane, and the appropriate extension from the Group pane.

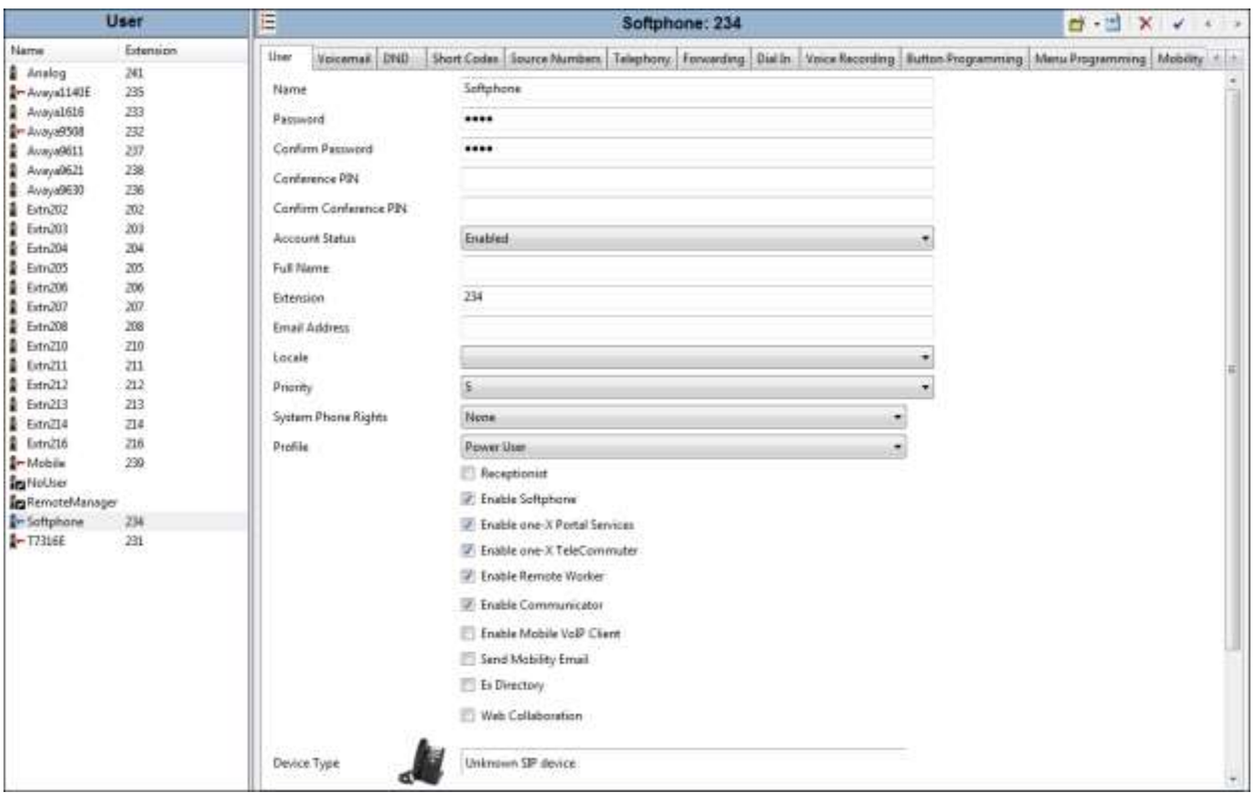
5.5.2. User 234 (Avaya Communicator for Windows)

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 Communicator for Windows. The **Base Extension** field is populated with 234, the extension assigned to the Avaya Communicator for Windows. Ensure the **Force Authorization** box is checked.

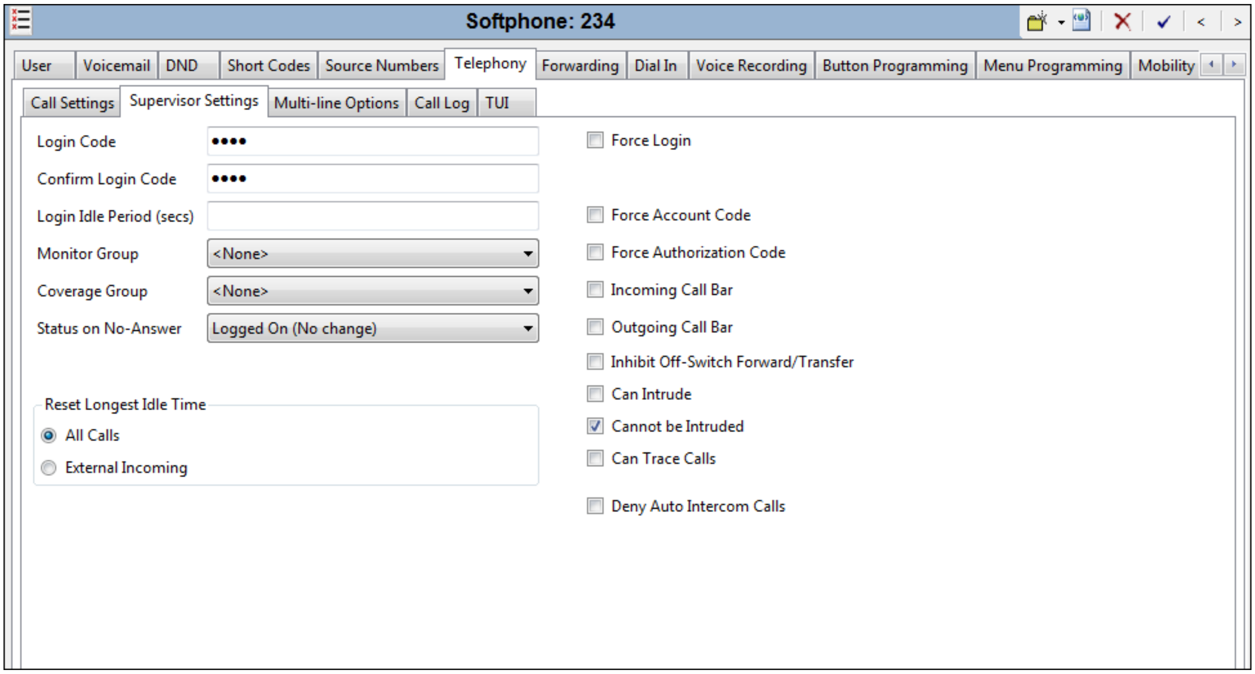
The following screen shows the **VoIP** tab for the extension. The **IP Address** field may be left blank. For the **Reserve License** parameter, select "Reserve Avaya IP endpoint license" from the drop-down box. The **Codec Selection** parameter may retain the default setting "System Default" to follow the system configuration shown in **Section 5.2.6**. Alternatively, "Custom" may be selected to allow the codecs to be configured for this extension, using the arrow keys to select and order the codecs. Other fields may retain default values.



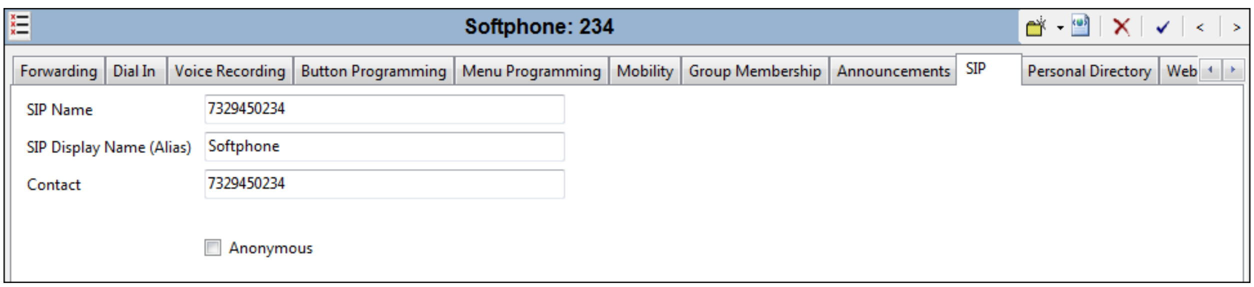
The following screen shows the **User** tab for User 234 corresponding to an Avaya Communicator for Windows. The **Extension** parameter is populated with extension 234. In the sample configuration, the **Profile** is set to “Power User”, with **Enable Softphone**, and **Enable Communicator** checked.



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



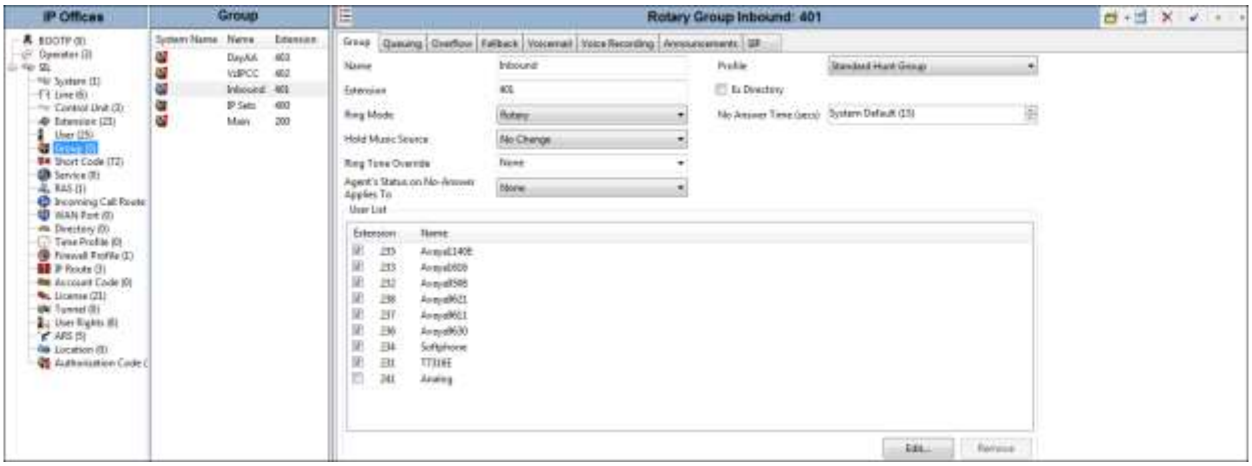
Like other users previously illustrated, the **SIP** tab for the user with extension 234 is configured with a **SIP Name** and **Contact** specifying the user's Verizon IP Trunk service DID number.



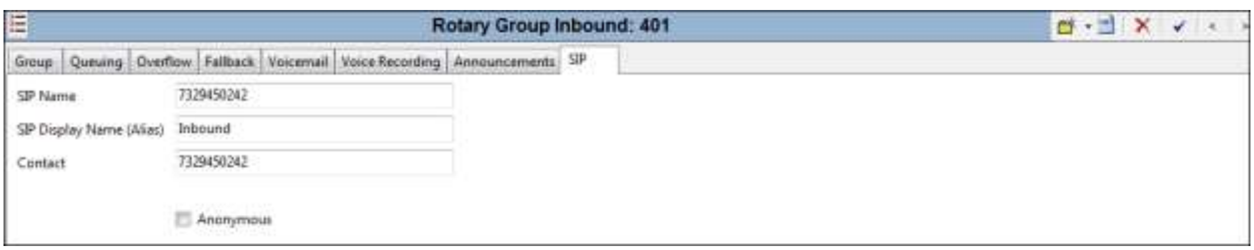
5.5.3. Hunt Groups

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 **Group** 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 **Group** tab for hunt group 401. The telephone extensions in the **User List** 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". Click the **Edit** button to change the **User List**.



The following screen shows the **SIP** tab for hunt group 401. The **SIP Name** and **Contact** are configured with Verizon DID 7329450242. Later, in **Section 5.7**, an Incoming Call Route will map 7329450242 to this hunt group based on the information entered on this tab.

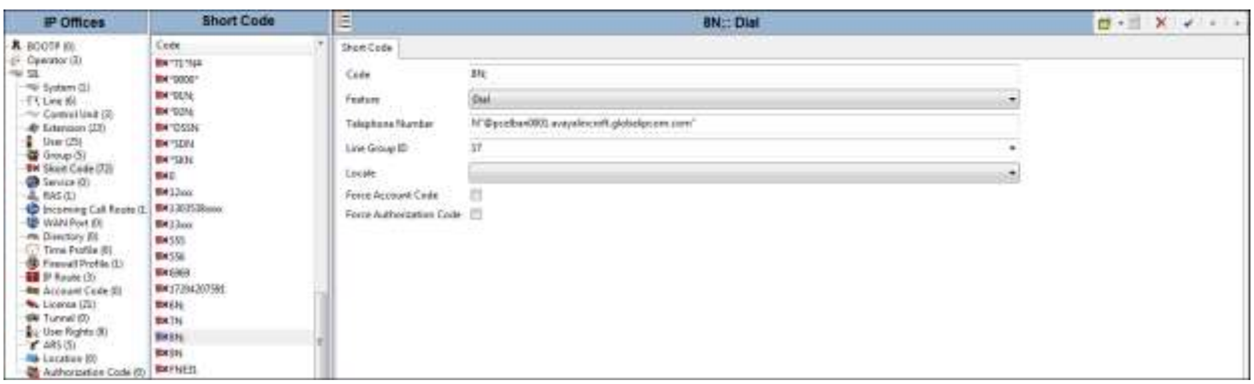


5.6. Short Codes

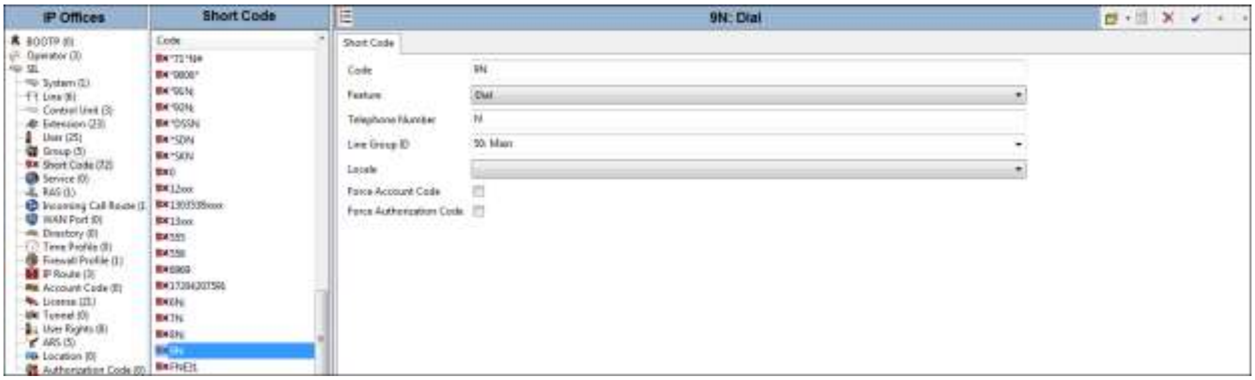
In this section, various examples of IP Office short codes will be illustrated. To add a short code, right click on **Short Code** in the Navigation pane, and select **New.** To edit an existing short code, click **Short Code** in the Navigation pane, and the short code to be configured in the Group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

In the screen shown below, the short code “8N;” is illustrated. The **Code** parameter is set to “8N;”. 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 “17”, matching the number of the **Outgoing Group** configured on the **SIP URI** tab of SIP Line 17 to Verizon Business (**Section 5.4.5**).

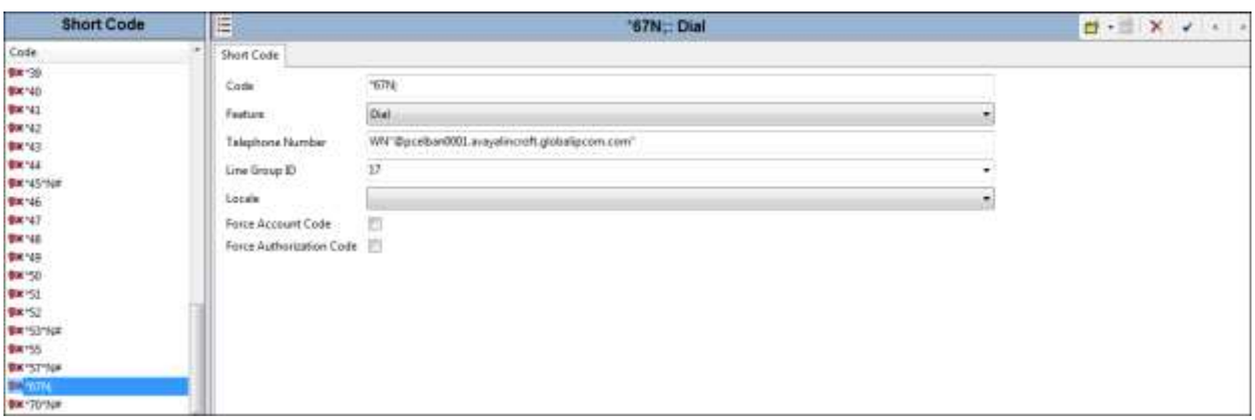
This simple short code will allow an IP Office user to dial the digit 8 followed by any telephone number, symbolized by the letter N, to reach the SIP Line to Verizon business. “N” can be any number such as a 10-digit number, a 1+10 digit number, a toll free number, directory assistance (e.g., 411), etc. This short code approach has the virtue of simplicity, but does not provide for alternate routing or an awareness of the end of a dialed digit string. When users dial 8 plus the number, IP Office must wait for an end of dialing timeout before sending the SIP INVITE to Verizon Business.



The simple “8N;” short code previously illustrated does not provide a means of alternate routing if the configured 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 Avaya IP Office user dials 9 plus any number “N”, rather than being directed to a specific **Line Group Id**, the call is directed to “50: Main”, configurable via ARS. See **Section 5.8** for example ARS route configuration for “50: Main” as well as a backup route.



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 “*67N;” is illustrated. This short code is similar to the “8N;” short code except that the **Telephone Number** field begins with the letter “W”, which means “withhold the outgoing calling line identification”. In the case of the SIP Line connecting to Verizon documented in these Application Notes, when a user dials *67 plus any number “N”, IP Office will include the user’s telephone number in the P-Asserted-Identity (PAI) header (see **Section 5.4.8**) 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.



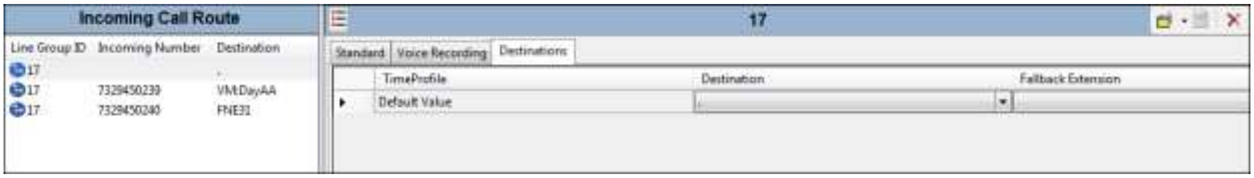
The following screen illustrates a short code that acts like a feature access code rather than a means to access a SIP Line. In this case, the **Code** “FNE31” is defined for **Feature** “FNE Service” to **Telephone Number** “31” (Mobile Call Control). This short code will be used as means to allow a Verizon DID to be programmed to route directly to this feature, via inclusion of this short code as the destination of an Incoming Call Route. See **Section 5.7**. This feature is used to provide dial tone to twinned mobile devices (e.g., cell phone) directly from IP Office; once dial tone is received the user can perform dialing actions including making calls and activating Short Codes.

5.7. Incoming Call Routes

In this section, IP Office Incoming Call Routes are illustrated. 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.

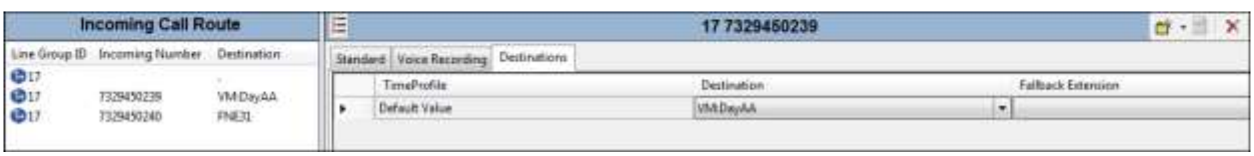
In the screen shown below, a simple incoming call route is illustrated. The **Line Group Id** is 17, matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to Verizon Business in **Section 5.4.5**. The **Incoming Number** field is left blank to match all details of the To header.

The following **Destinations** tab for the incoming call route contains the **Destination** "." entered manually. This will match the **Incoming Number** field as the Destination and route the call based on the information in the SIP tab for the User or hunt group as illustrated in **Section 5.5**. For example, a call to 732-945-0232 will be routed to user 232, because this user has 7329450232 configured for the **SIP Name** and **Contact** parameters.

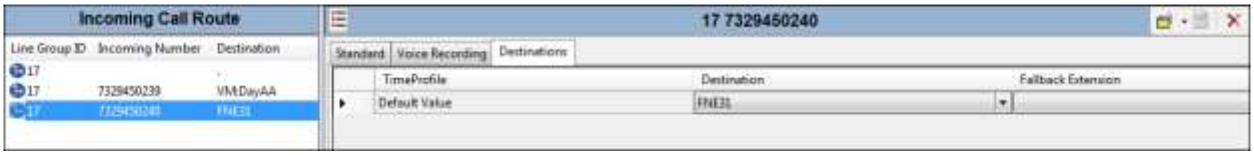


In the screen shown below, the incoming call route for **Incoming Number** "7329450239" is illustrated. The **Line Group Id** is "17", matching the Incoming Group field configured in the **SIP URI** tab for the SIP Line to Verizon Business in **Section 5.4.5**.

The following **Destinations** tab for the incoming call route contains the **Destination** "VM:DayAA" entered manually. An incoming call to 732-945-0239 will be delivered directed to the Voicemail Pro Module "DayAA", for the daytime automated attendant.



Similarly, the following **Destinations** tab for an incoming call route contains the **Destination** “FNE31” entered manually. The name “FNE31” is the short code for accessing the “Mobile Call Control” application configured in **Section 5.6**, and 732-945-0240 was configured in **Section 5.4.5** on the SIP URI tab as an incoming number. An incoming call to 732-945-0240 will be delivered directly to internal dial tone from the IP Office, allowing the caller to perform dialing actions including making calls and activating Short Codes. The incoming caller ID must match the Twinned Mobile Number entered in the User Mobility tab in **Section 5.5.1**; otherwise the IP Office responds with a 486 Busy Here and the caller will hear a busy tone.



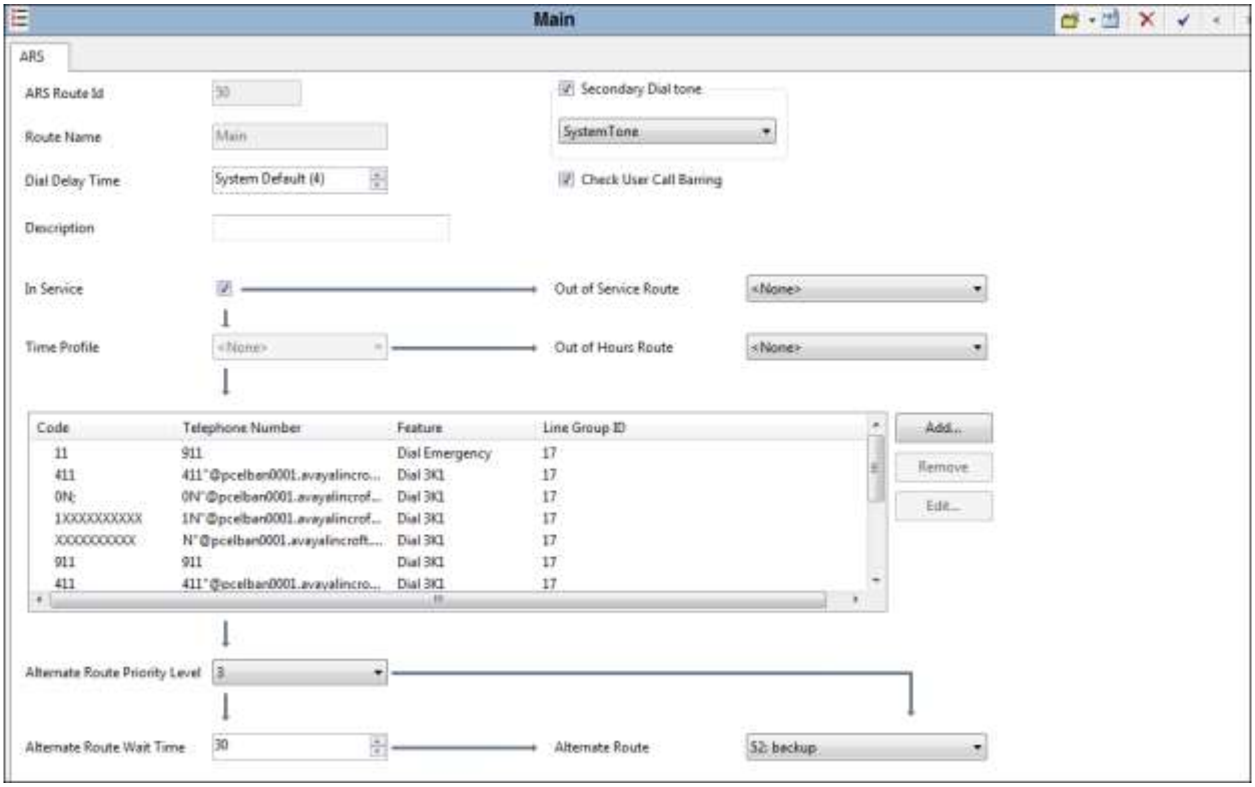
5.8. ARS and Alternate Routing

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 demonstrate 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 “8N;” 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. 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.

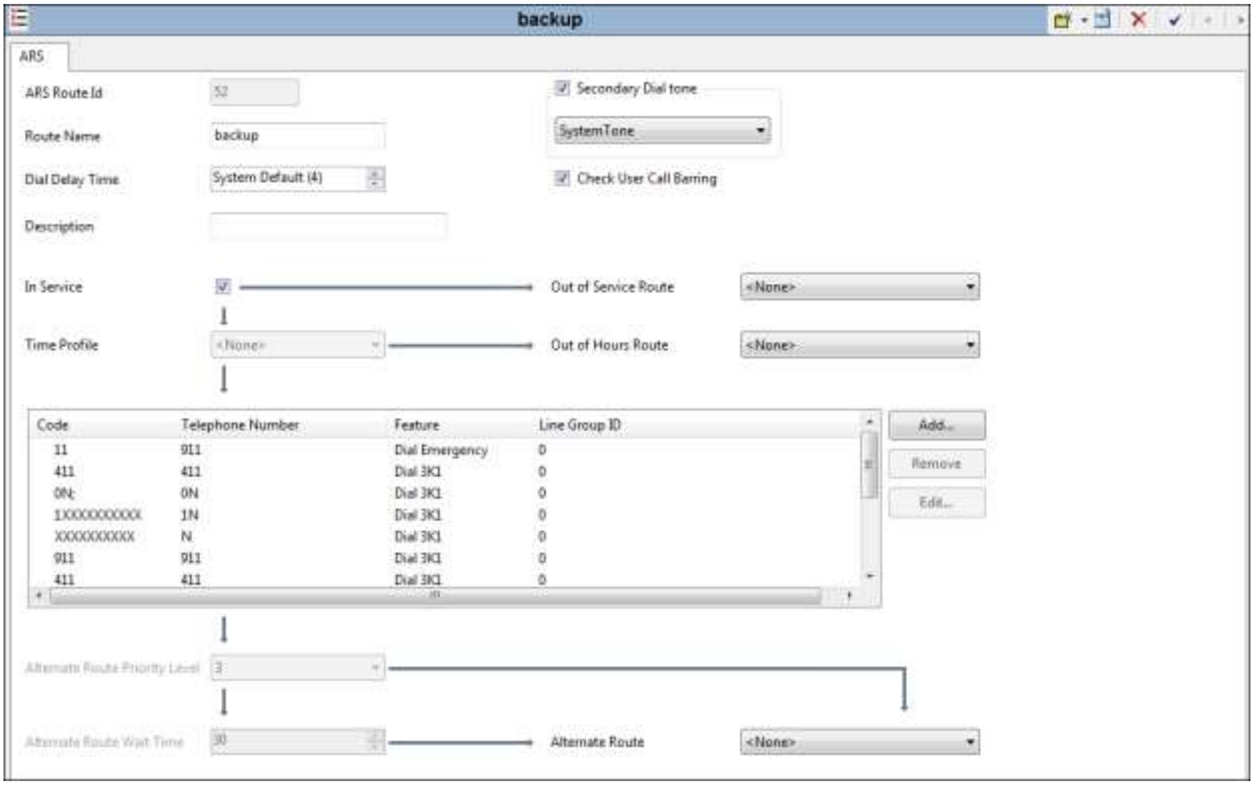
The following screen shows an example ARS configuration for the route named “Main”. The **In Service** parameter refers to the ARS form itself, not the Line Groups that may be referenced in the form. If the **In Service** box is un-checked, calls are routed to the ARS route name specified in the **Out 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 17. If Line Group 17 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** 52. 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 0 using the analog lines. 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 5.6**.

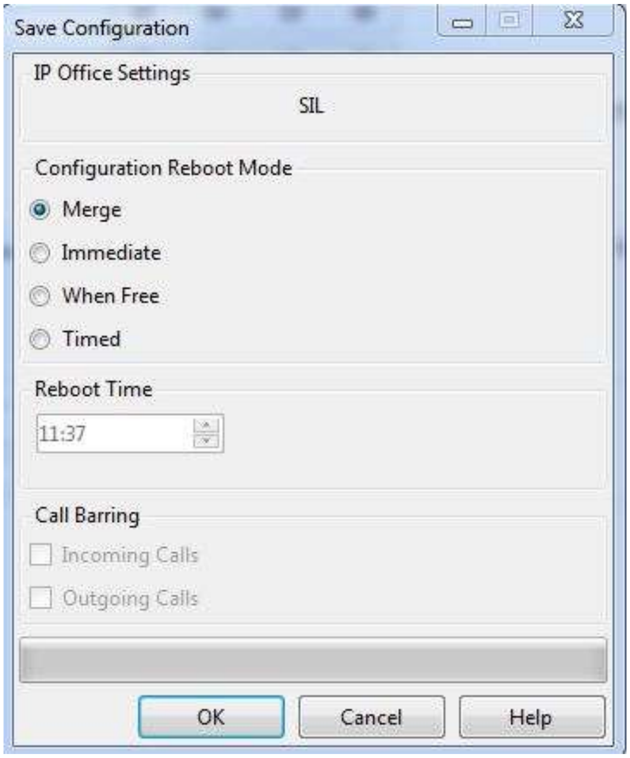


If a primary route experiences a network outage such that no response is received to an outbound INVITE, IP Office successfully routes the call via the backup route. The user receives an audible tone when the re-routing occurs and may briefly see "Waiting for Line" on the display.

5.9. Save Configuration

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

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



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. Verification Steps

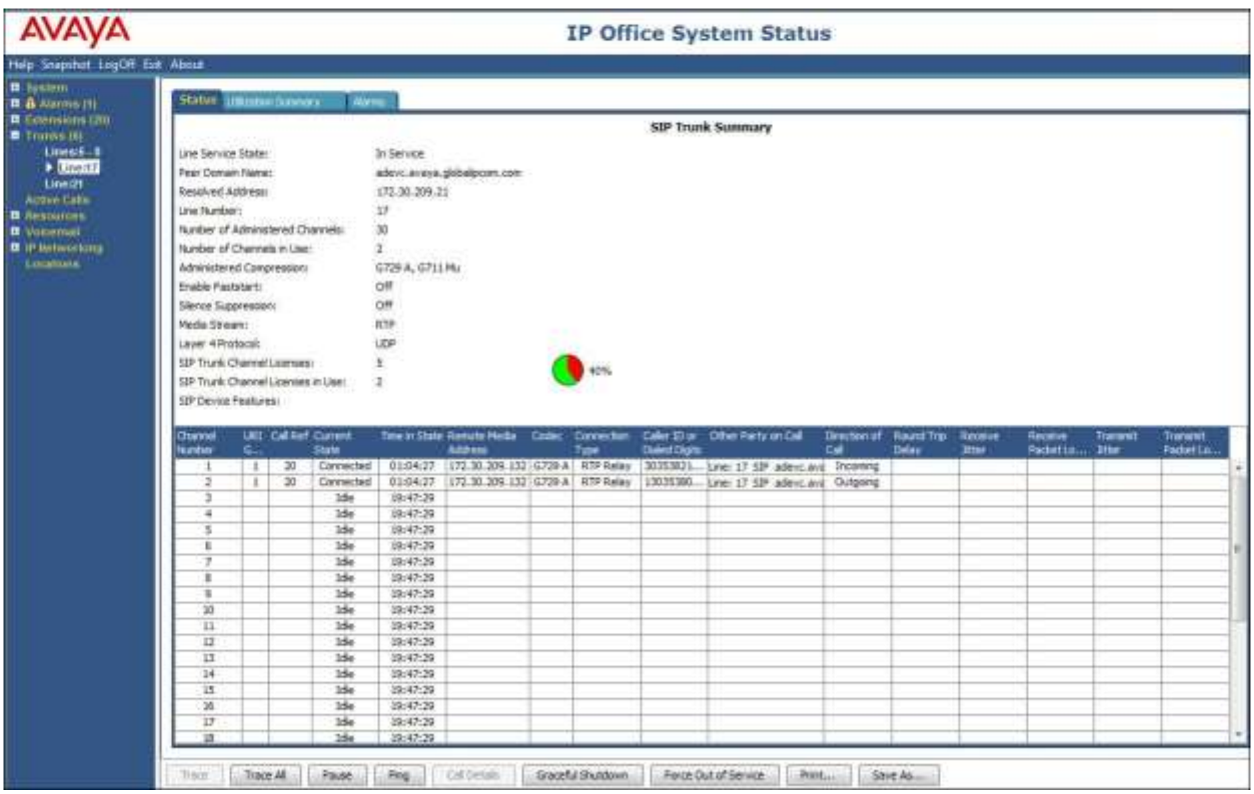
This section provides verification steps that may be performed in the field to verify that the solution is configured properly.

7.1. System Status

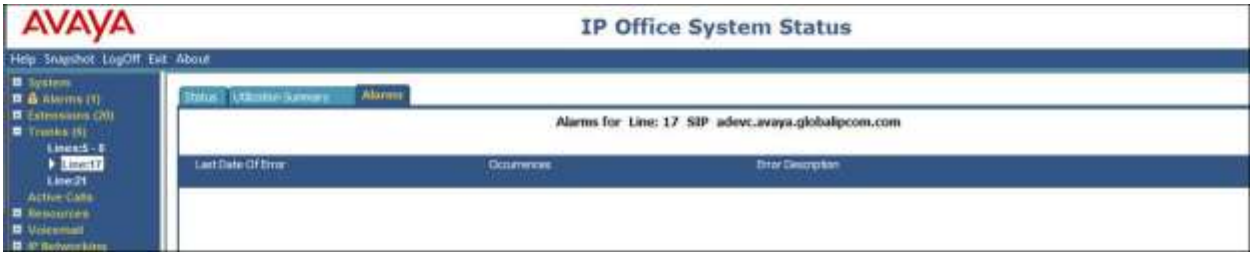
The System Status application is used to monitor and troubleshoot IP Office. Use the System Status application to verify the state of the SIP trunk. System Status can be accessed from **Start → Programs → IP Office → System Status**. Or by opening an Internet browser and type the URL: http://*ipaddress*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

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

Select the SIP line under **Trunks** from the left pane. On the **Status** tab in the right pane, verify the **Current Stat**e is ***Idle*** for each channel.



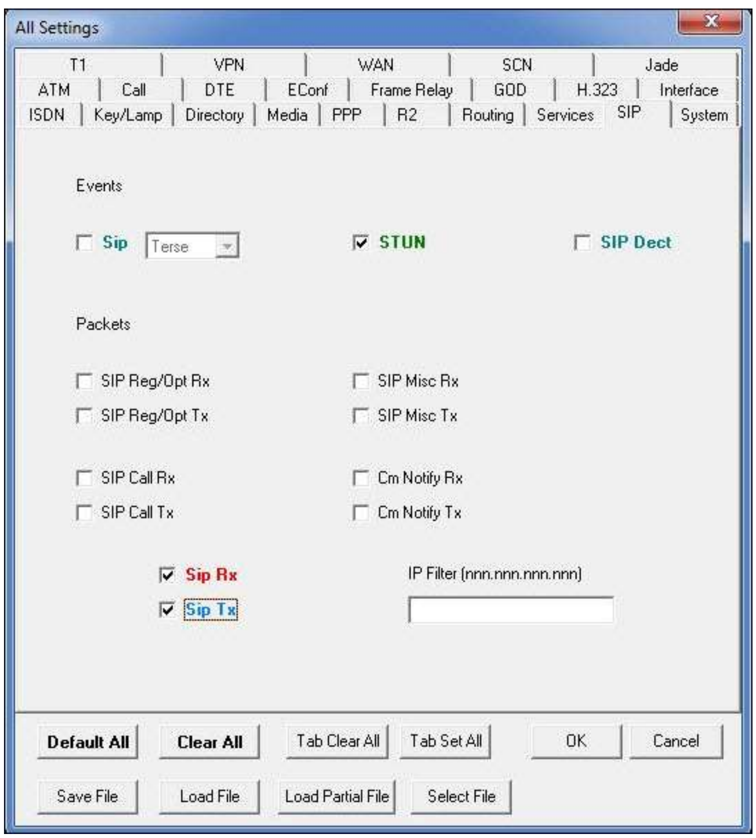
Select the **Alarms** tab and verify that no alarms are active on the SIP line.



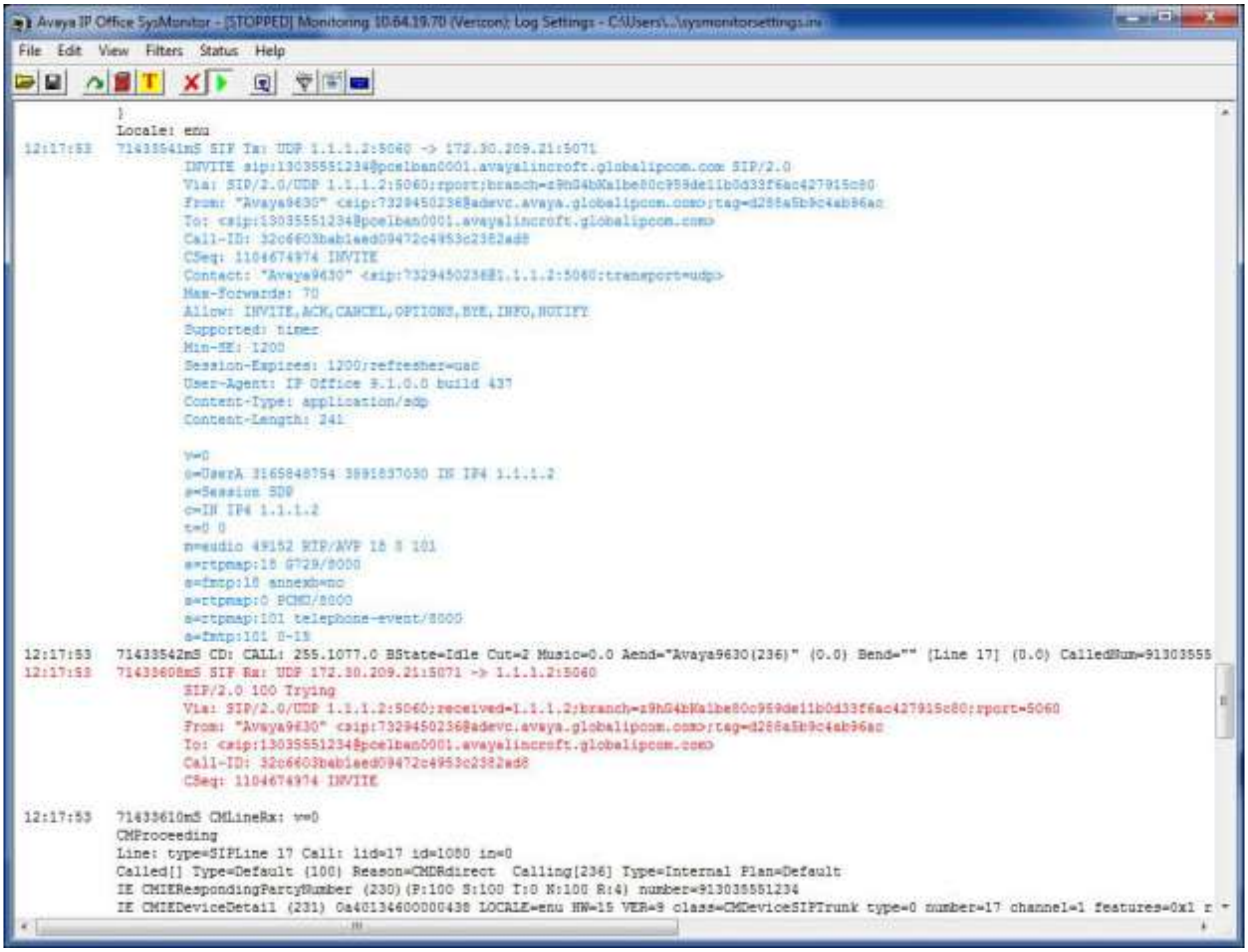
7.2. Monitor

The Monitor application can also be used to monitor and troubleshoot IP Office. Monitor can be accessed from **Start** → **Programs** → **IP Office** → **Monitor**. The application allows the monitored information to be customized. To customize, select **Filters** → **Trace Options**.

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



As an example, the following shows a portion of the monitoring window for an outbound call from extension 232, whose DID is 732-945-0232, calling out to the PSTN via the Verizon Business IP Trunk Service. The telephone user dialed 9-1-303-555-1234.



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 9.1 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. Additional References

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

[1] *IP Office ™ Platform 9.1, Deploying Avaya IP Office ™ Platform IP500 V2,* Document Number 15-601042, January 2015

[2] *Administering Avaya IP Office ™ Platform with Manager*, January 2015

[3] *IP Office ™ Platform 9.1, Installing and Maintaining the Avaya IP Office ™ Platform Application Server*, Document Number 15-601011 Issue 10d, March 2015

[4] *IP Office ™ Platform 9.1, Deploying Avaya IP Office ™ Platform Servers as Virtual Machines,* Document Number 15-601011 Issue 03c, January 2015

[5] *IP Office ™ Platform 9.1, Using Avaya IP Office ™ System Status*, Document Number 15-601758, October 2014

[6] *Administering Avaya Communicator on IP Office,* December 2014

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

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 9.0.

[IPOR9] Application Notes for Configuring SIP Trunk using Verizon Business IP Trunk SIP Trunk Service Offer and Avaya IP Office Release 9.0, Issue 1.1

The Application Notes referenced below correspond to the formal compliance testing by Avaya and Verizon Business for IP Office Release 8.1.

[IPOR81] Application Notes for SIP Trunking Using Verizon Business IP Trunk SIP Trunk Service and Avaya IP Office Release 8.1 – 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.