Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Windstream SIP Trunking Service (Sonus Platform) with Avaya IP Office Release 9.0 – Issue 1.0

Abstract

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between the Windstream (Sonus Platform) and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500 V2 Release 9.0, Avaya Voicemail Pro, and Avaya SIP, H.323, digital, and analog endpoints.

The Windstream SIP Trunking Service provides PSTN access via a SIP trunk between the enterprise and the Windstream network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

Windstream is a member of the Avaya DevConnect Service Provider program. 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.

1. Introduction

These Application Notes describe a sample configuration using Session Initiation Protocol (SIP) trunking between Windstream (Sonus platform) and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500 V2 Release 9.0, Avaya Voicemail Pro, and Avaya SIP, H.323, digital, and analog endpoints.

Customers using Avaya IP Office with the Windstream 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.

2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the Windstream SIP Trunking service as a non-registering device via the public Internet and exercise the features and functionality listed in **Section 2.1**. The simulated enterprise site was comprised of Avaya IP Office and various Avaya endpoints.

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 Windstream 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, Avaya Flare® Experience, and Avaya Voicemail Pro. The display of caller ID on display-equipped Avaya IP Office telephones was verified.
- Outgoing calls from the Avaya IP Office location to the PSTN were routed via the SIP Line to Windstream. These outgoing PSTN calls were originated from Avaya SIP phones, Avaya H.323 telephones, Avaya digital telephones, analog endpoints, Avaya IP Office Softphone, Avaya Flare® Experience, and Avaya Voicemail Pro. The display of caller ID on display-equipped PSTN telephones was verified.
- Inbound / Outbound fax calls 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, 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.
- Incoming and outgoing calls using the G.729A and G.711MU codecs.
- DTMF transmission (RFC 2833) with successful voice mail navigation 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.
- Inbound calls from Windstream SIP Trunk service that were call forwarded back to PSTN destinations, presenting true calling party information to the PSTN phone, via Windstream.
- 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.

Items not supported or not tested included the following:

- Inbound toll-free and emergency calls (911) are supported but were not tested as part of the compliance test.
- Operator (0) calls are not supported by Windstream.
- Network Call Redirection using the SIP REFER method is not supported by Windstream.

2.2. Test Results

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

- **T.38 Fax –** When the fax call is initiated with G.729A codec as the first choice on an outbound fax call, Windstream will send a re-Invite to G.711MU first before the call is negotiated to T.38. The G.711MU codec needs to an available codec on IP Office for the fax call to properly negotiate to T.38. See **Section 5.4.5**.

  On an inbound fax call, IP Office will send a re-Invite with the original voice codec when the fax transmission is complete. If the original voice codec is G.729A, Windstream will return a "488 Not Acceptable Here" error. This does not have a negative effect on fax calls because it happens after the fax has been sent at the end of the call.

2.3. Support

Contact information for technical support on the Windstream SIP Trunking service:

- Web: http://www.windstreambusiness.com/customer/support
- Telephone: (866) 445-5882

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. Alternatively, in the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus

3. Reference Configuration

Figure 1 illustrates the sample configuration used for the DevConnect compliance testing. The sample configuration shows an enterprise site connected to the Windstream SIP Trunk service.

Located at the enterprise site is an Avaya IP Office 500 V2. The LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public network. Endpoints include an Avaya 1616 IP Telephone (with H.323 firmware), an Avaya 1140E IP Telephone (with SIP firmware), an Avaya 9611 IP Telephone (with H.323 firmware), and an Avaya 6210 Analog Telephone. The site also has an IP Office Application Server running the Voicemail Pro for voicemail and one-X® Portal.

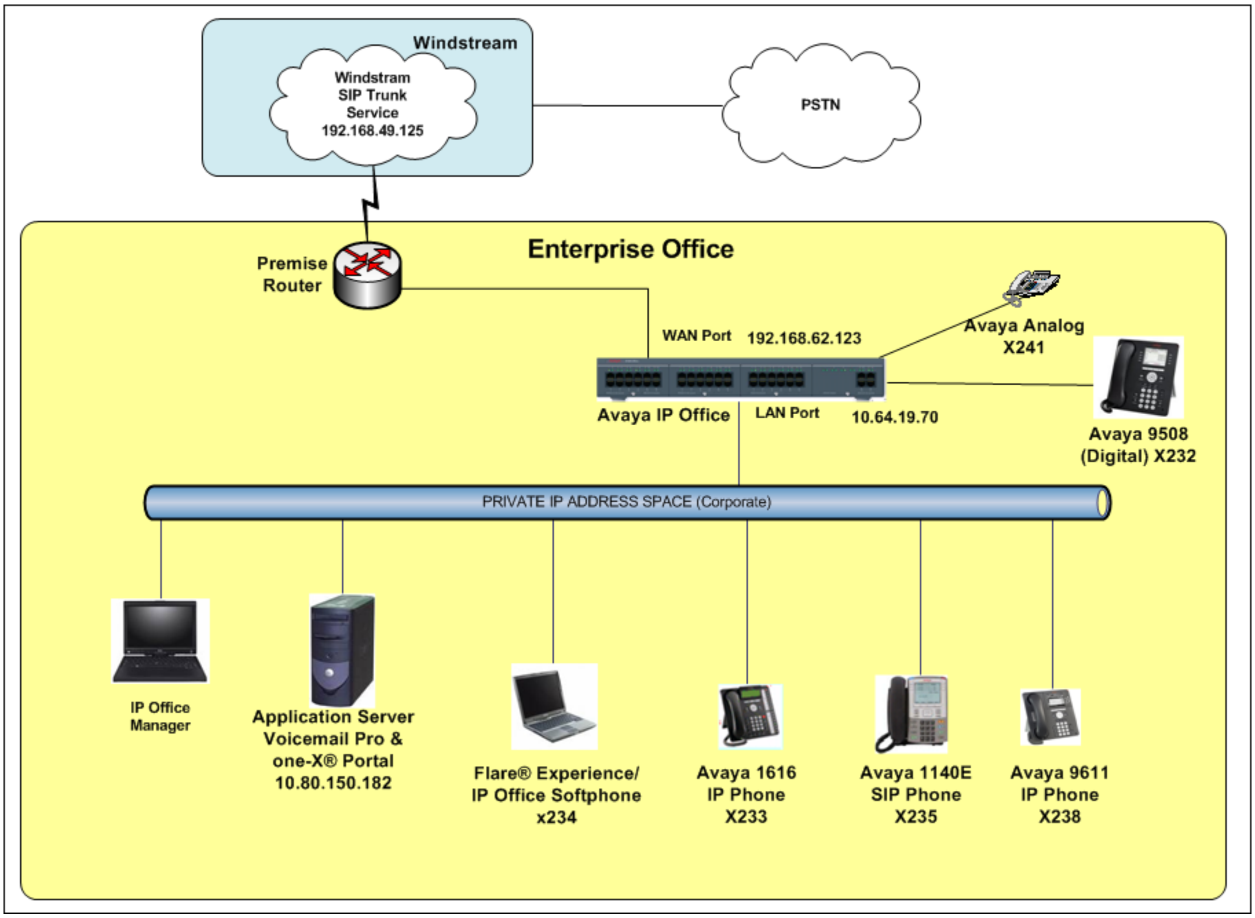


Figure 1: Avaya Interoperability Test Lab Configuration

For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead, public IP addresses have been replaced with private addresses and all phone numbers have been replaced with numbers that cannot be routed.

4. Equipment and Software Validated

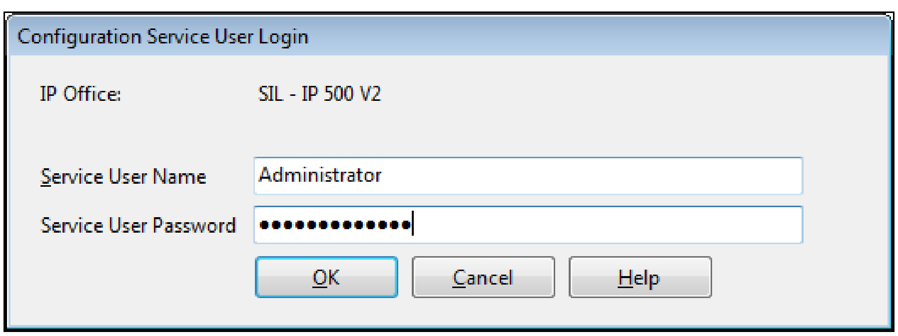
Table 2 shows the equipment and software used in the sample configuration

Avaya IP Telephony Solution Components	
Equipment	**Software**
Avaya IP Office 500 V2	Release 9.0.200.860
Avaya Application Server	Release 9.0.200-860
Avaya IP Office Manager	Release 9.0.2.0 Build 860
Avaya 1616SW IP Telephone (H.323)	Release 1.343A
Avaya 9611SW IP Telephone (H.323)	Release 6.2209
Avaya 1140E IP Telephone (SIP)	Release 04.04.10
Avaya 9508 Digital Telephone	Release 0.45
Avaya IP Office Softphone	Release 3.2.3.49
Avaya Flare® Experience for Windows	Release 1.1.4.23
Windstream Components	
Equipment	**Software**
Sonus	Release 9.1.0

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

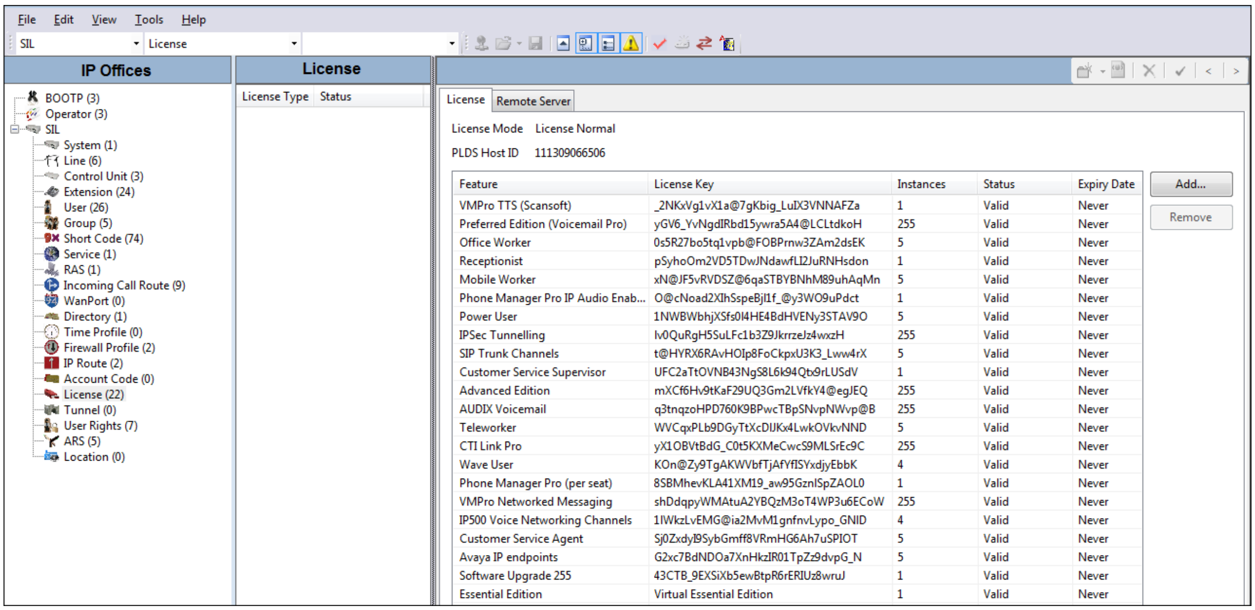


Log in with the appropriate configuration credentials. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details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

5.1. Licensing and Physical Hardware

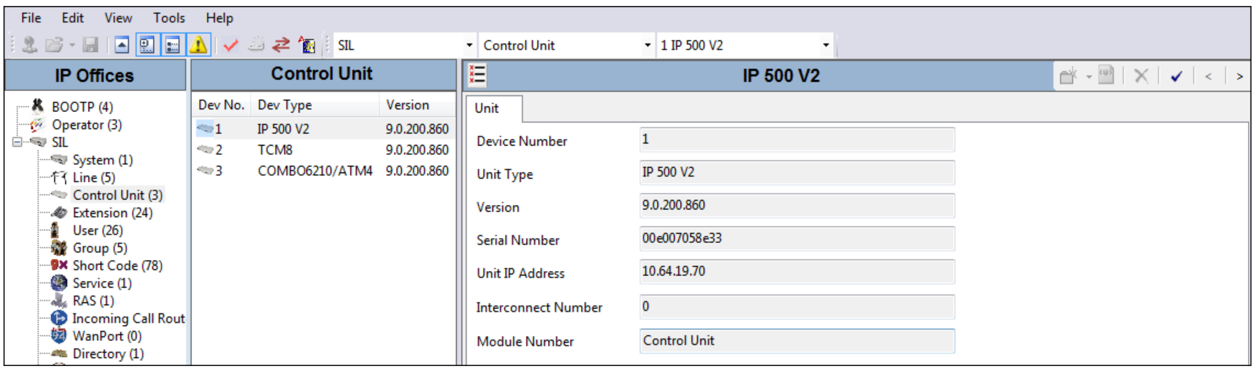
The configuration and features described in these Application Notes require the IP Office system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

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 Combination Card. This module has 6 Digital Stations ports, 2 Analog Station ports, 4 Analog Trunk ports and 10 VCM channels. The VCM is a Voice Compression Module supporting VoIP codecs. An IP Office hardware configuration with a VCM component is necessary to support SIP trunking.

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.



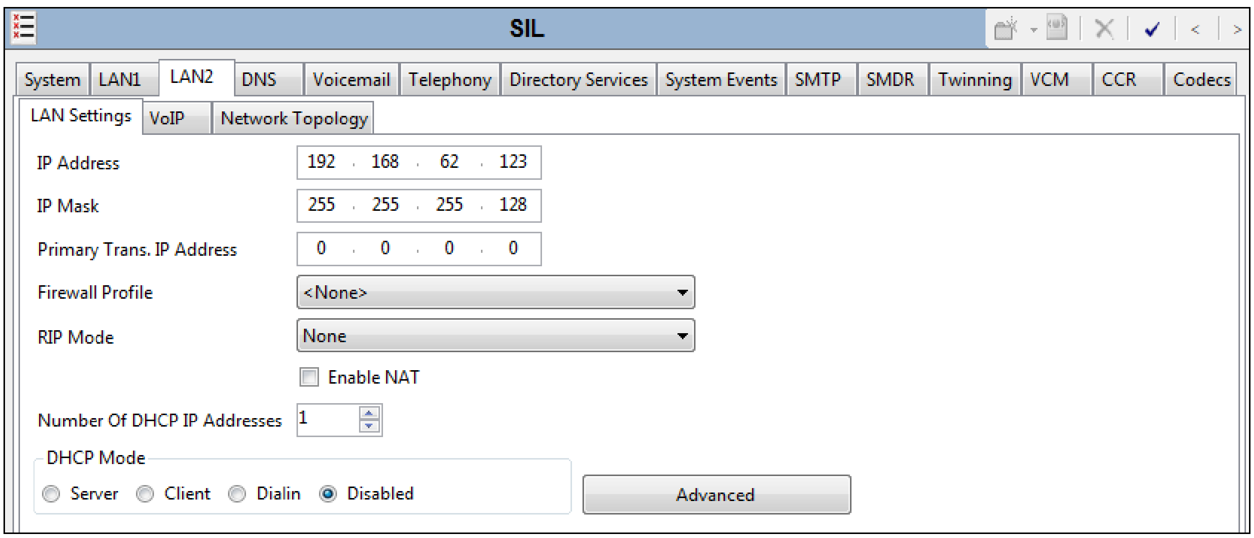
5.2. System Settings

This section illustrates the configuration of system settings. The settings presented here simply illustrate the sample configuration and are not intended to be prescriptive. 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.

5.2.1. LAN 2 Settings

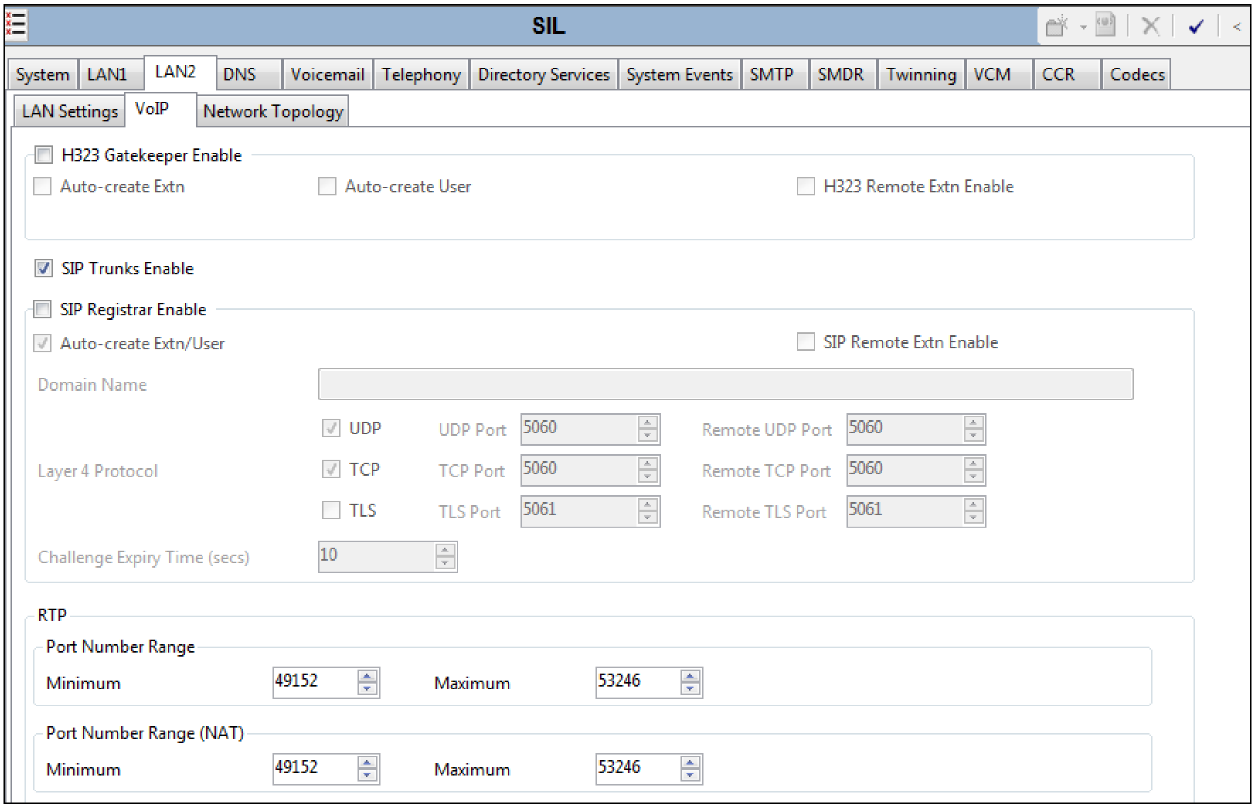
The IP500/IP500 V2 control units have 2 RJ45 Ethernet ports, physically marked as LAN and WAN. Within the system configuration, the physical LAN port is LAN1, the physical WAN port is LAN2.

In the sample configuration, LAN2 is used to connect the IP Office to the public network. To view or configure the **IP Address** of LAN2, select the **LAN2** tab followed by the **LAN Settings** tab. Set the **IP Address** field to the IP address assigned to the Avaya IP Office LAN2 port. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements.

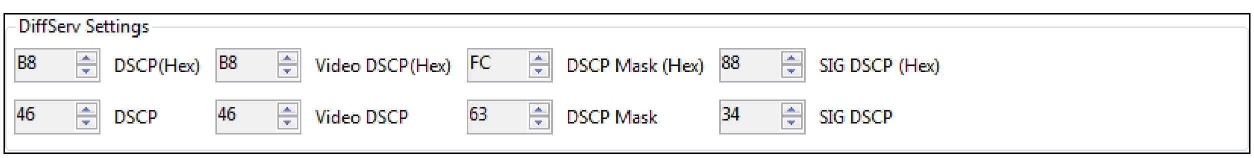


Select the **VoIP** tab as shown in the following screen. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Windstream.

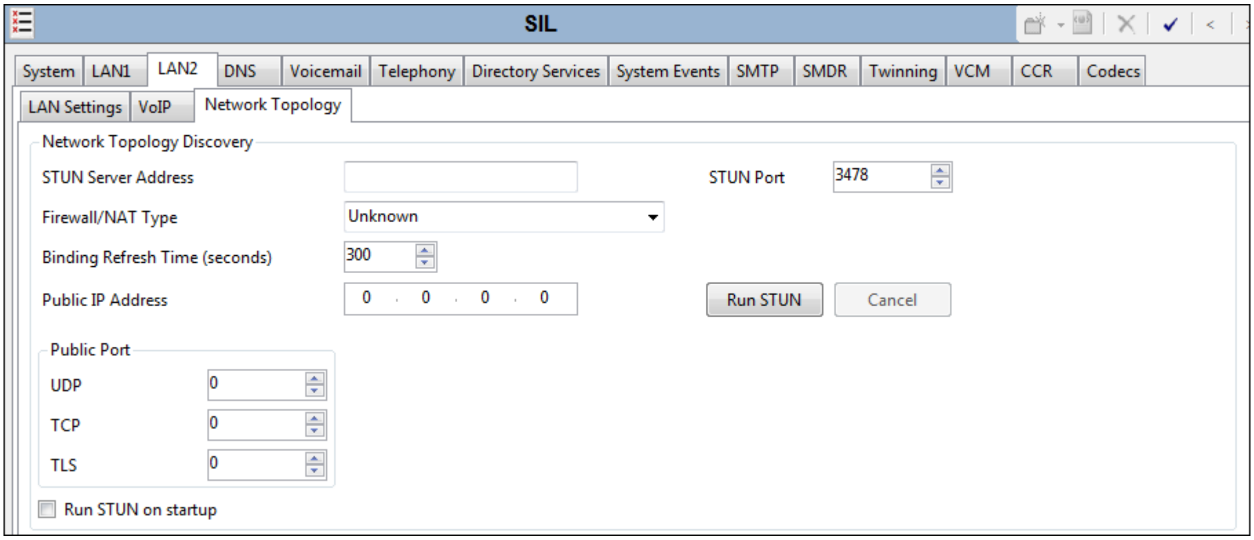
If desired, the **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media paths from Windstream to IP Office. The defaults are used here.



Scroll down to the **DiffServ Settings** section. Avaya IP Office can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling. The specific values used for the compliance test are shown in the screen below and are also the default values. For a customer installation, if the default values are not sufficient, appropriate values should be provided by the customer.

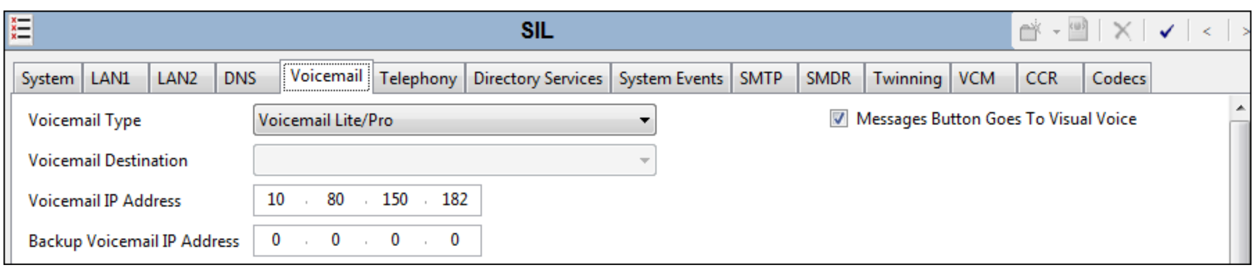


Select the **Network Topology** tab as shown in the following screen. In the sample configuration, the default settings are used and the **Use Network Topology Info** in the **SIP Line** is set to "None" in **Section 5.4.3**.

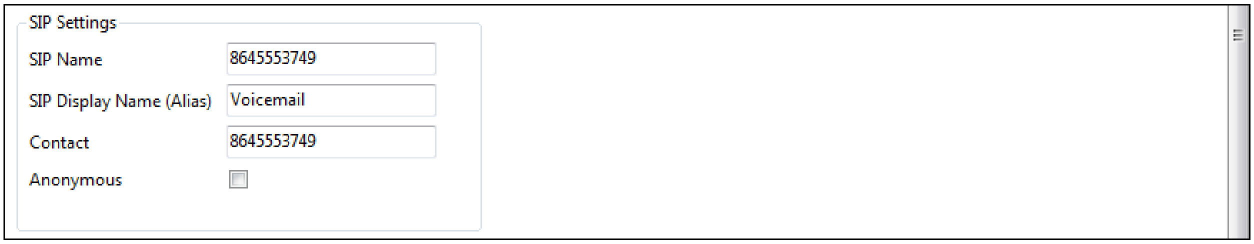


5.2.2. Voicemail Settings

To view or change voicemail settings, select the **Voicemail** tab as shown in the following screen. 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 10.80.150.182, the IP address of the IP Office Application Server running the Voicemail Pro software, as shown in **Figure 1**.

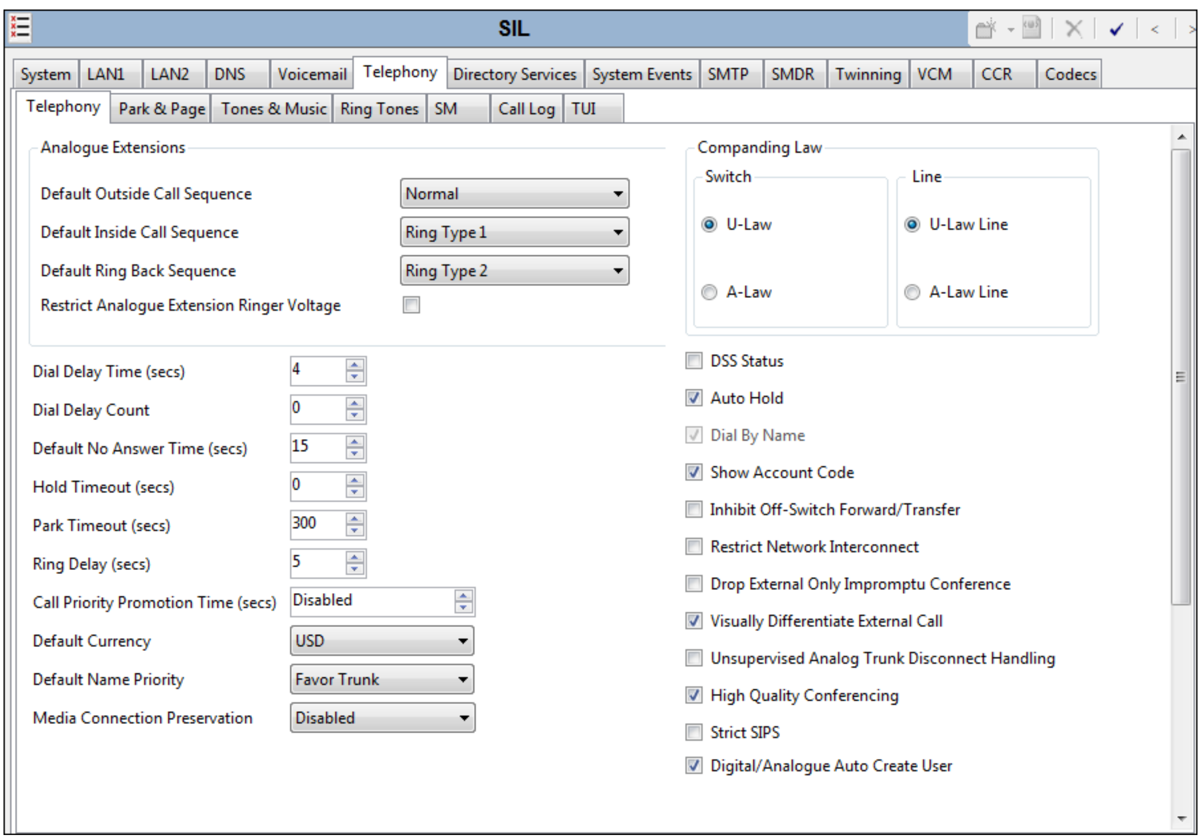


In the sample configuration, the “Callback” application of Avaya Voicemail Pro was used to allow Voicemail Pro to call out via the SIP Line to Windstream when a message is left in a voice mailbox. The **SIP Settings** shown in the screen below enable IP Office to populate the SIP headers for an outbound “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**).



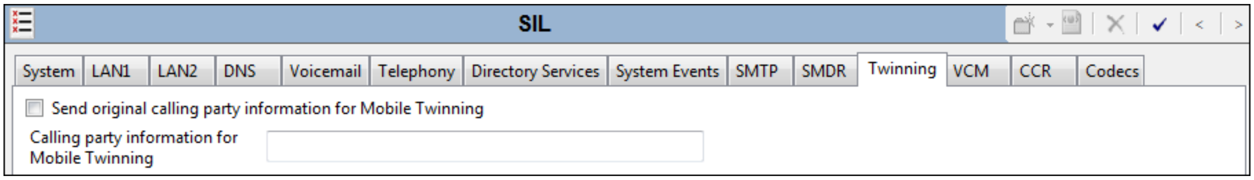
5.2.3. System Telephony Configuration

To view or change telephony settings, select the **Telephony** tab and **Telephony** sub-tab as shown in the following screen. 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 Windstream service can be tested. That is, a call can arrive to IP Office via Windstream, and be forwarded or transferred back to the PSTN with the outbound leg of the call using the Windstream SIP 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.



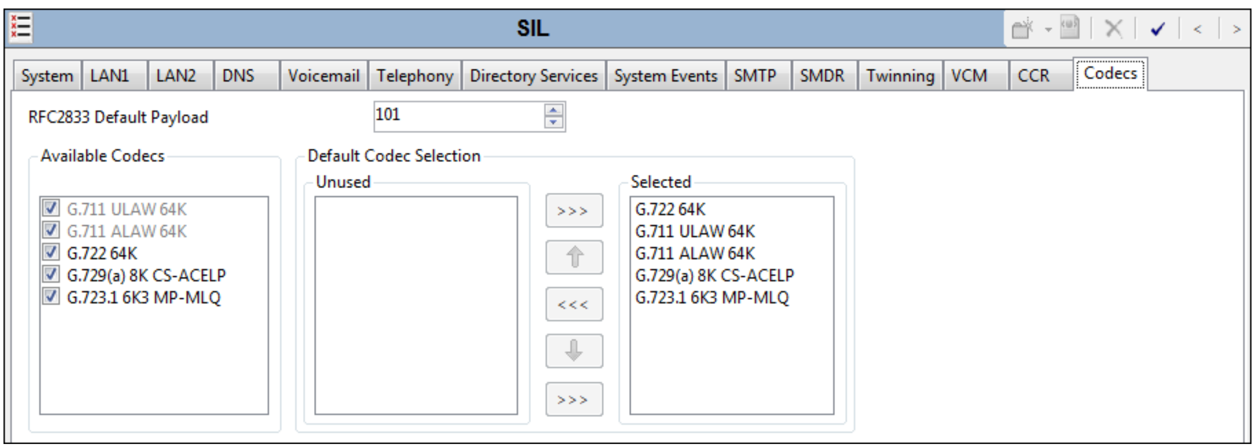
5.2.4. System Twinning Configuration

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 5.4.2**), 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 Windstream SIP Trunk service.



5.2.5. System Codecs Configuration

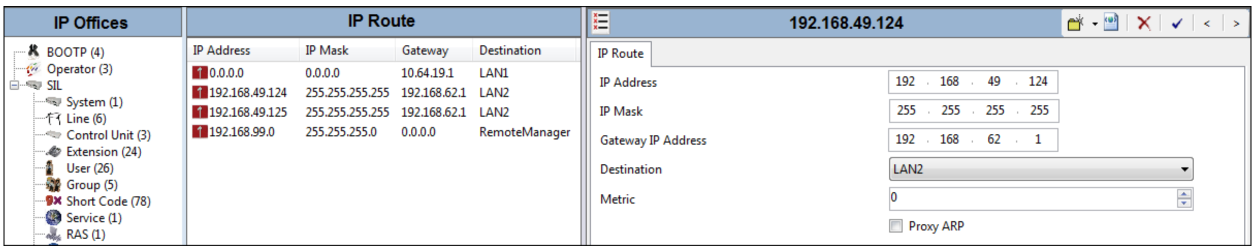
To view or change system codec settings, select the **Codecs** tab. On the left, observe the list of **Available Codecs**. In the example screen below, the box next to each codec is checked, making all the codecs available in other screens where codec configuration may be performed (such as the SIP Line in **Section 5.4.5**). 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. The **RFC2833 Default Payload** parameter is new in IP Office 9.0. Set the payload parameter to "101", the value preferred by Windstream.



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 service provider and has a default gateway of 192.168.62.1. 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. 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 5.4.1** 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 5.4.2 – 5.4.6**.

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

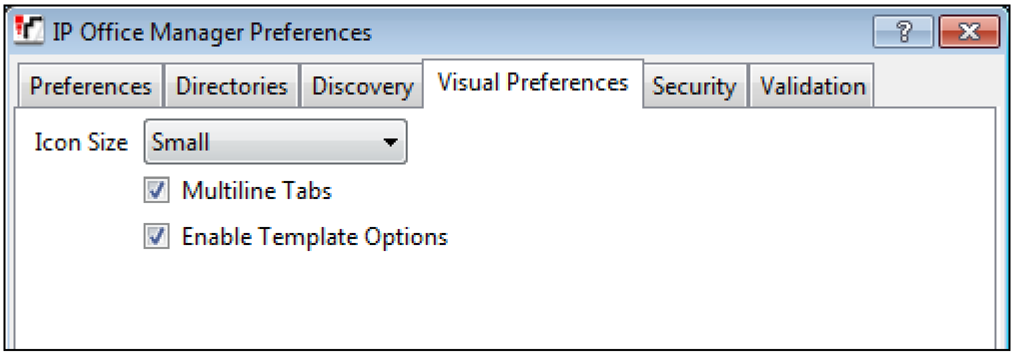
- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required

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 5.4.2 – 5.4.6**.

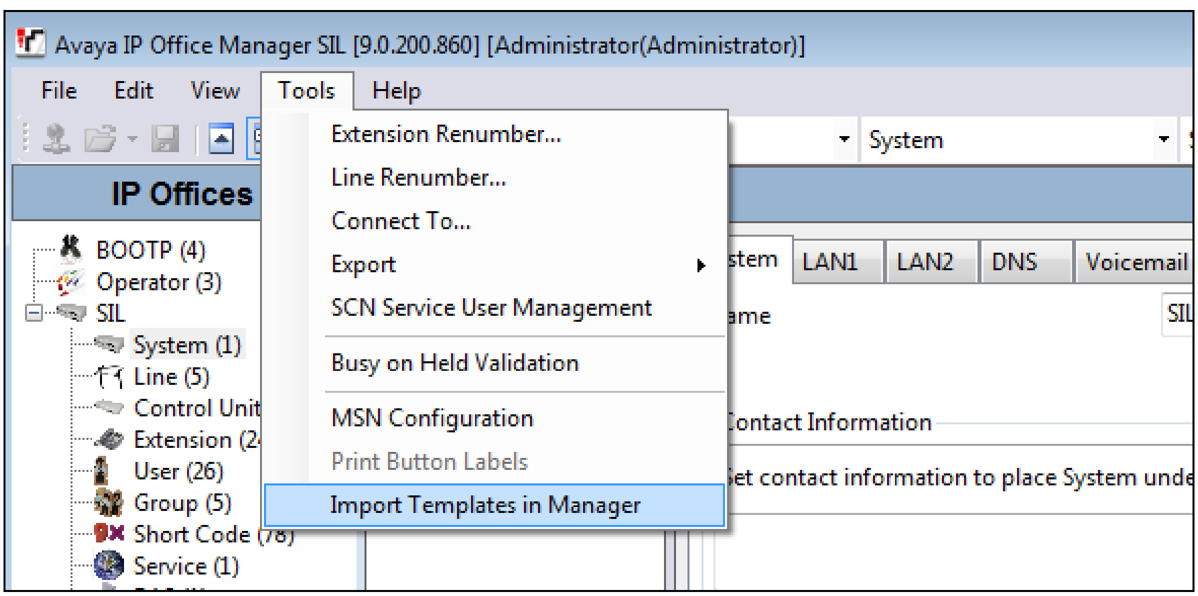
5.4.1. SIP Line From Template

1. Copy the template file to the computer where IP Office Manager is installed. Verify the template file is **US_WindstreamSonus.xml**. The file name is important in locating the proper template file in **Step 5**.

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. Verify that the box is checked next to **Enable Template Options**. Click **OK**.

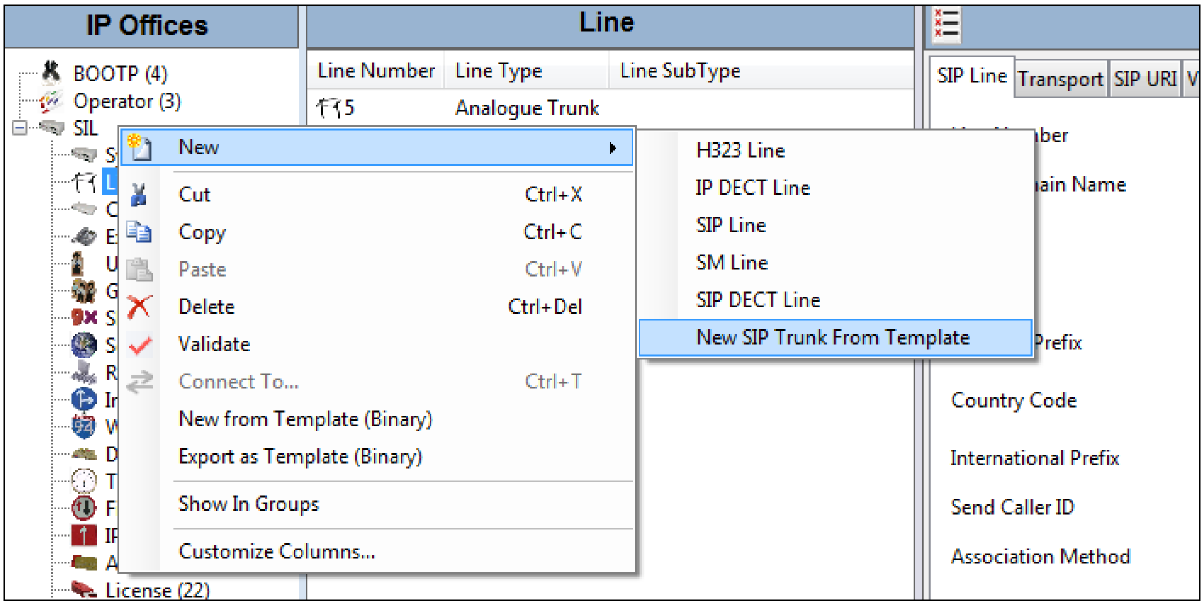


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default template location is **C:\Program Files\Avaya\IP Office\Manager\Templates**.

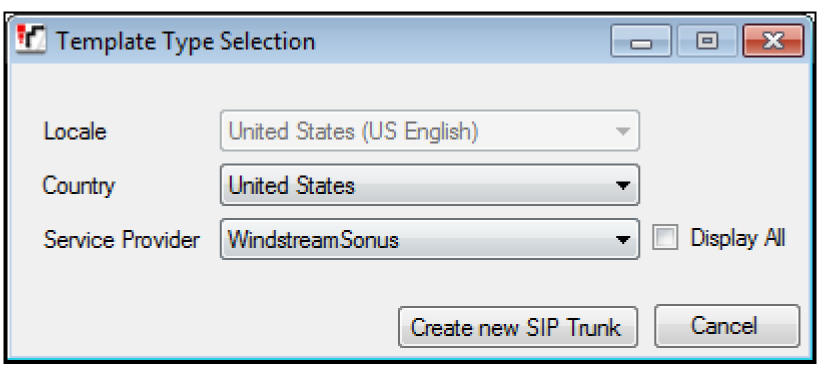


In the pop-up window (not shown) that appears, select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window (not shown) will appear stating success or failure. Click **OK** (not shown) to continue. If preferred, this step may be skipped if the template file is copied directly to the IP Office template directory.

4. To create the SIP Trunk from the template, right-click on **Line** in the Navigation Pane, then navigate to **New → New SIP Trunk From Template**.



5. In the subsequent Template Type Selection pop-up window, select "United States" from the **Country** pull-down menu and select "WindstreamSonus" from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name (**US_WindstreamSonus.xml**) created in **Step 1**. Click **Create new SIP Trunk** to finish creating the trunk.



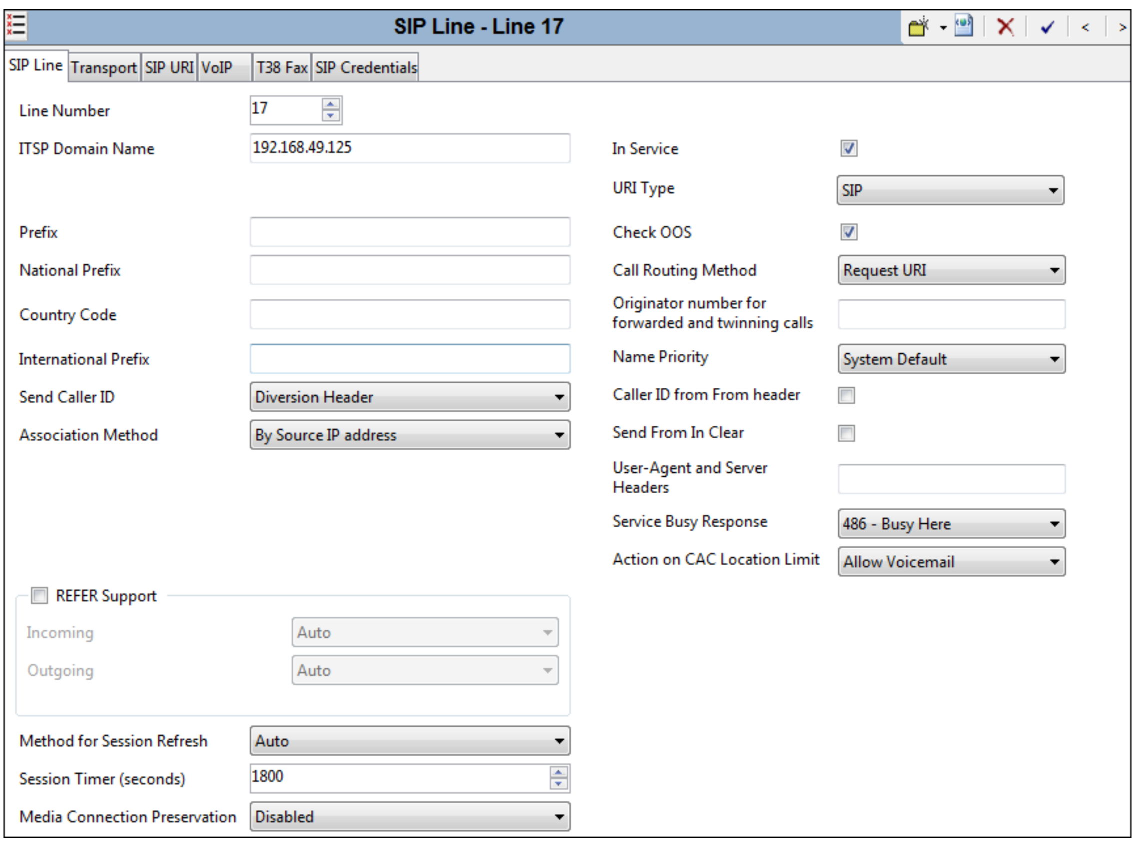
Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.4.2 – 5.4.6**.

5.4.2. SIP Line – SIP Line Tab

The **SIP Line** tab in the Details pane is shown below for Line Number 17, used for the Windstream SIP Trunk service. The **ITSP Domain Name** is configured with the IP address supplied by Windstream. 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.

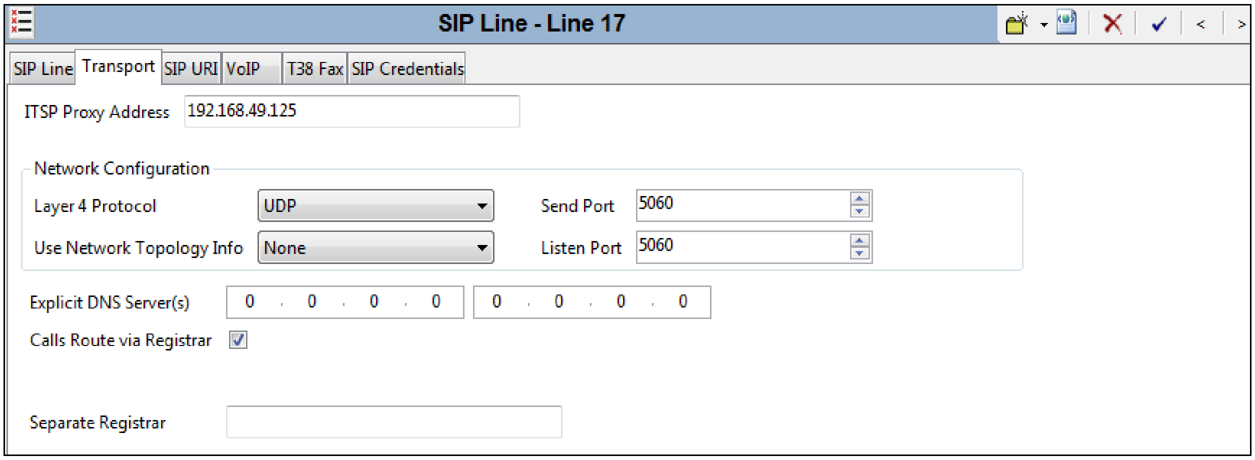
The **Send Caller ID** parameter is set to "Diversion Header". With this setting and the related configuration in **Section 5.2.5**, IP Office will include the Diversion Header for calls that are directed via Mobile Twinning out the SIP Line to Windstream. The Diversion Header will contain the number associated with the Twinning user, allowing Windstream 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 Windstream. **REFER Support** is uncheck as Windstream does not support REFER.

The **Method for Session Refresh** parameter and the related **Session Timer (seconds)** parameter are new with IP Office Release 9.0. The **Method for Session Refresh** is set to "Auto" and the **Session Timer (seconds)** is set to "1800". With this configuration, IP Office will send UPDATE messages every 15 minutes (half of the set value) to keep the active session alive. If UPDATE is not supported, re-INVITE messages are sent. The **Media Connection Preservation** parameter retains the default setting "Disabled"



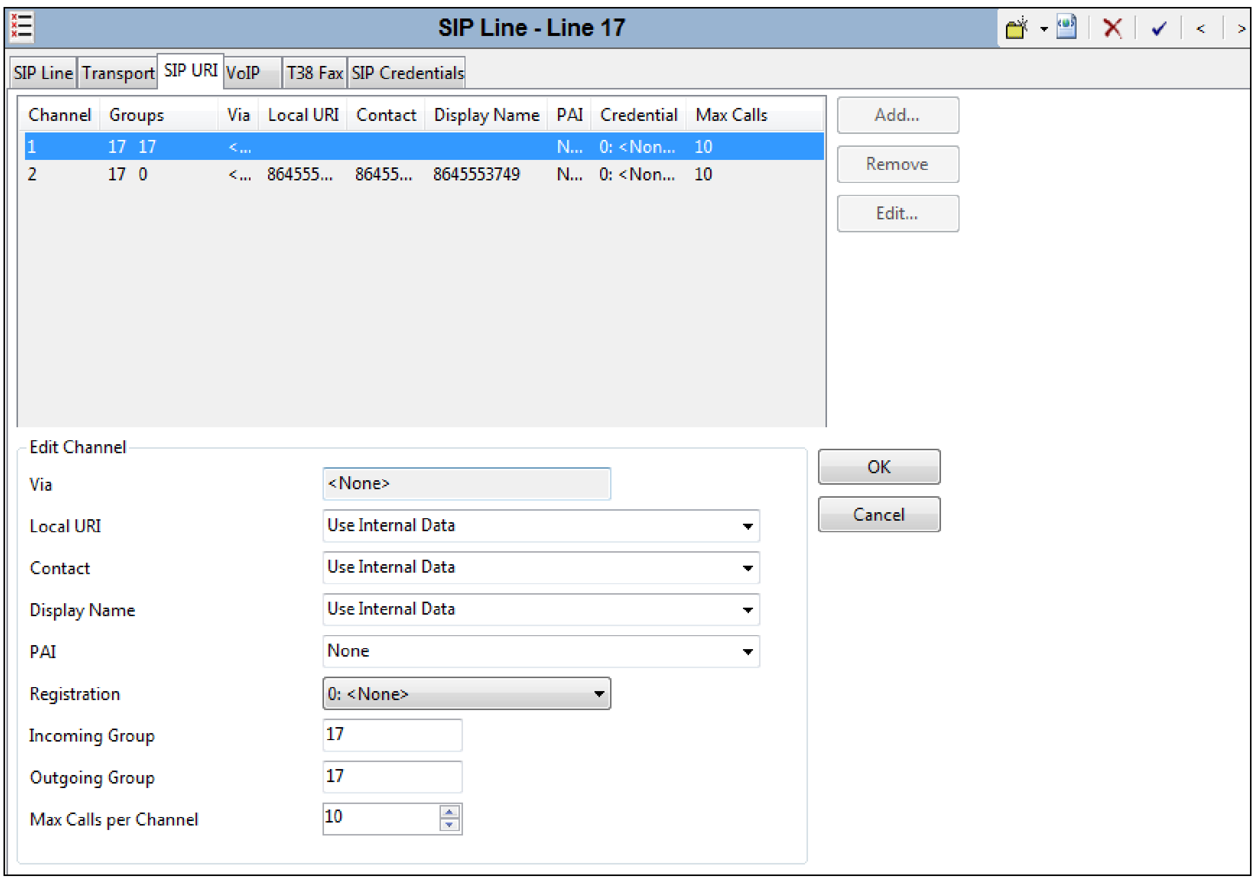
5.4.3. SIP Line - Transport Tab

Select the **Transport** tab. The **ITSP Proxy Address** is set to the IP address supplied by Windstream. In the **Network Configuration** area, "UDP" is selected as the **Layer 4 Protocol**. The **Send Port** and **Listen Port** can retain the default value 5060. The **Use Network Topology Info** parameter is set to "None".

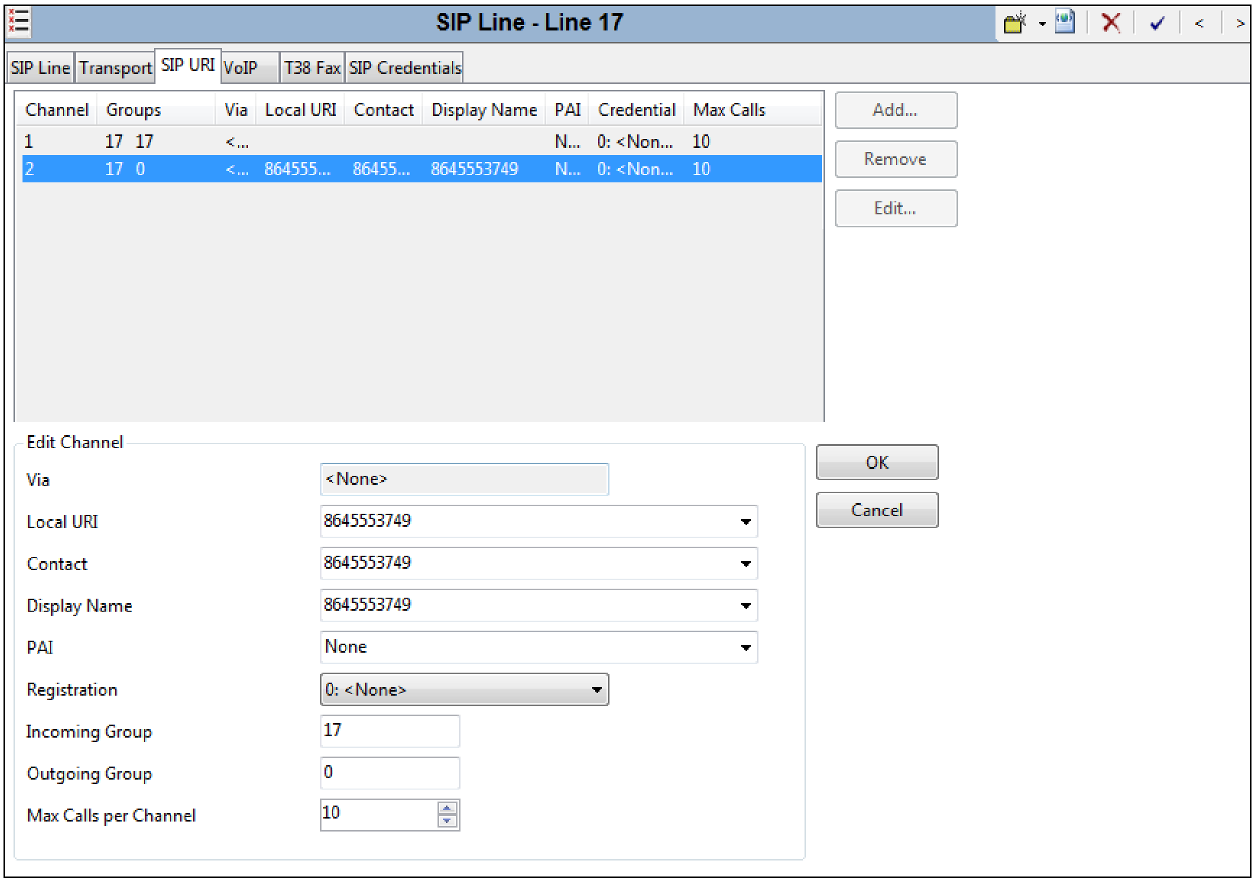


5.4.4. SIP Line - SIP URI Tab

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 Windstream 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 Windstream 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. Click **OK**.

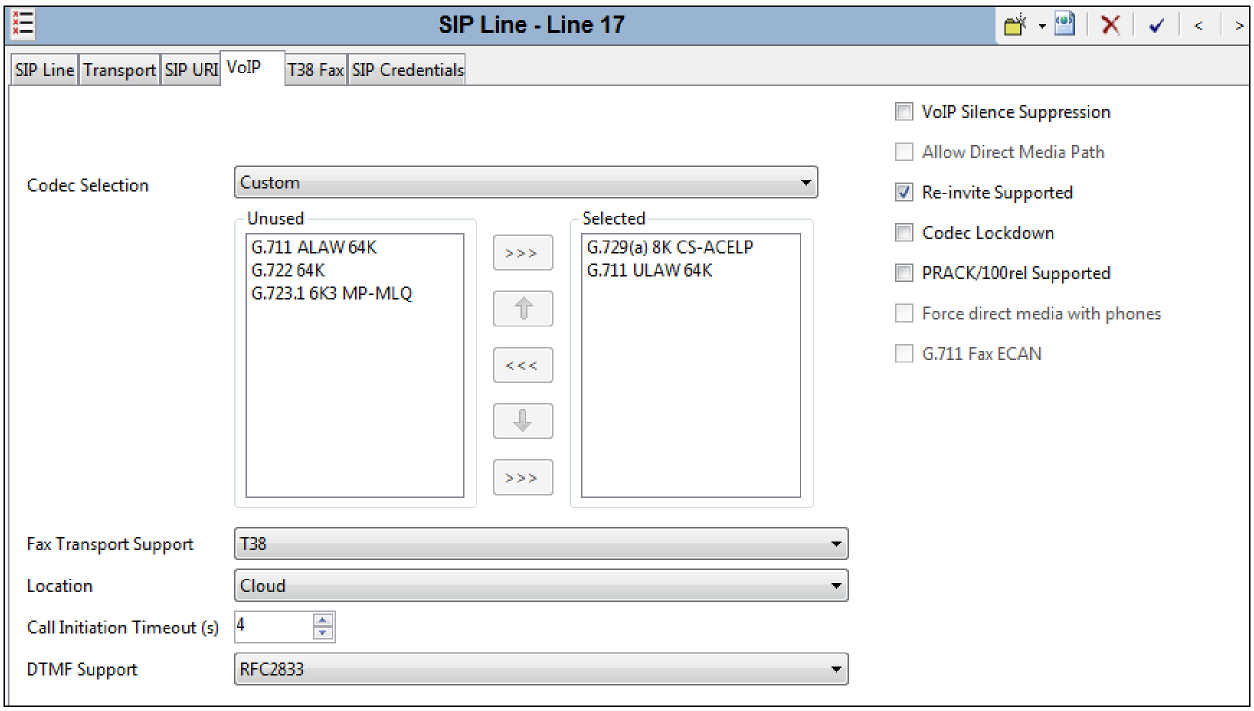


In the sample configuration, the single SIP URI shown above is sufficient to allow incoming calls for Windstream DID numbers destined for specific IP Office users or IP Office hunt groups. The calls are accepted by IP Office since the incoming number will match the SIP Name configured for the user or hunt group that is the destination for the call. Channel 2 displays a service number, 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 number 864-555-3749 will be assigned as a service number in the Incoming Call Routes in **Section 5.7**.



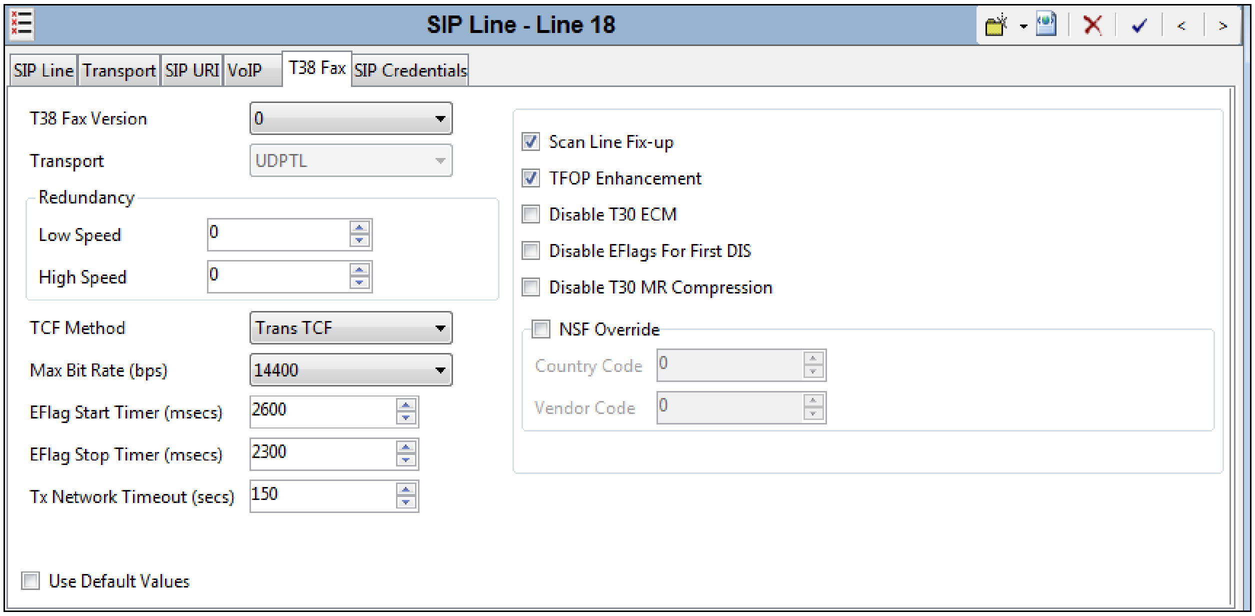
5.4.5. SIP Line - VoIP Tab

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 Windstream are included (i.e., G729(a) 8K CS-ACELP and G.711 ULAW 64K). This will cause IP Office to include G.729a and G.711MU in the Session Description Protocol (SDP) offer, in that order. Set the **Fax Transport Support** drop-down to "T38". This enables T.38 fax relay to be used when fax tones are detected during the call. 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 and the G.711 ULAW codec should be an available choice if the SIP Line will be used for fax. For PSTN originations, Windstream preferred the G.729 codec in the SDP, while also allowing the G.711MU codec. Click **OK** (not shown).



5.4.6. SIP Line- T38 Fax

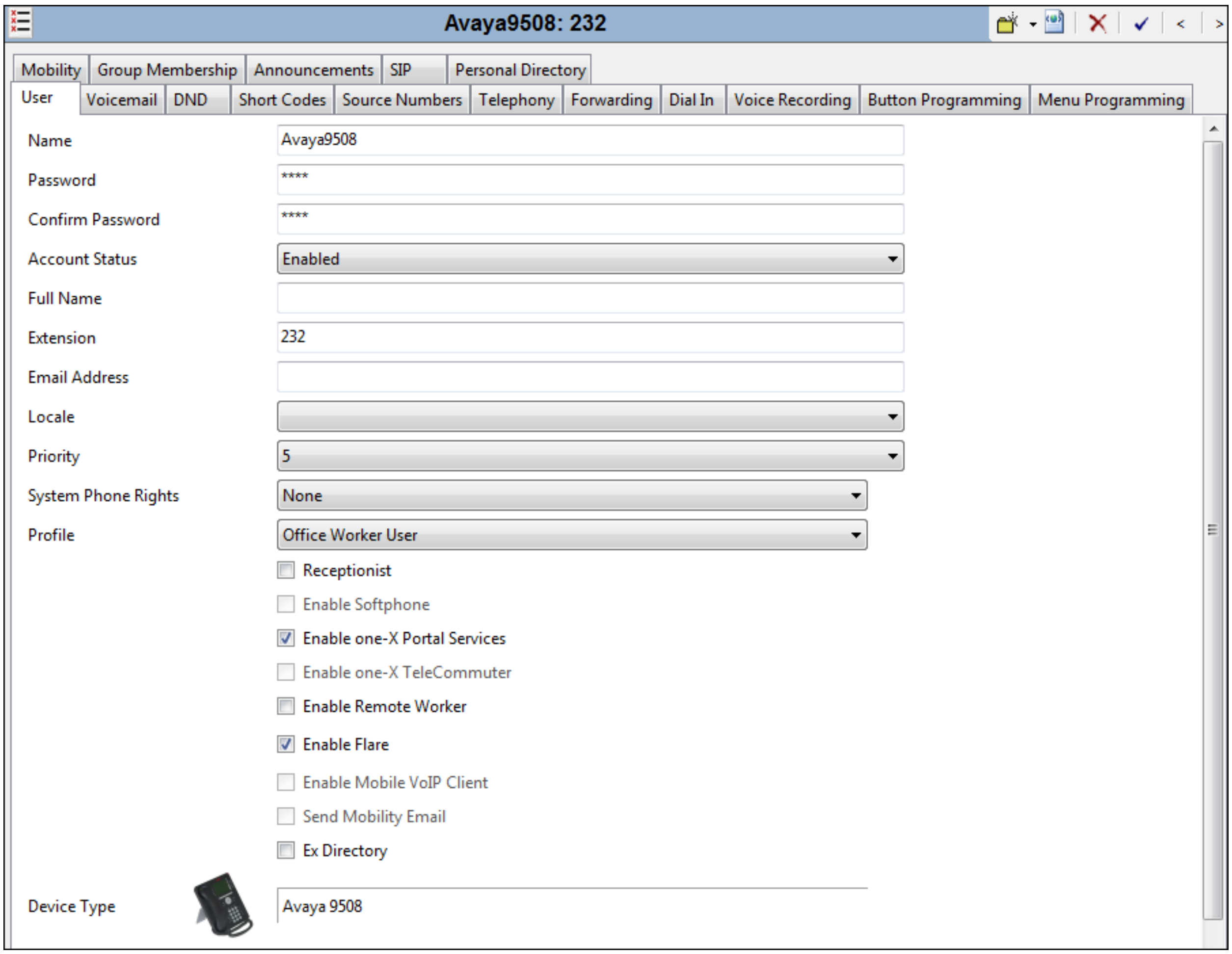
The settings on this tab are only accessible if **Re-invite Supported** is checked and a value for **Fax Transport Support** other than "None" are selected on the **VoIP** tab. Fax relay is only supported on IP500/IP500 V2 systems with an IP500 VCM card. Uncheck **Use Default Values** at the bottom of the screen and set **T38 Fax Version** to "0". This is the version supported by Windstream.



5.5. Users

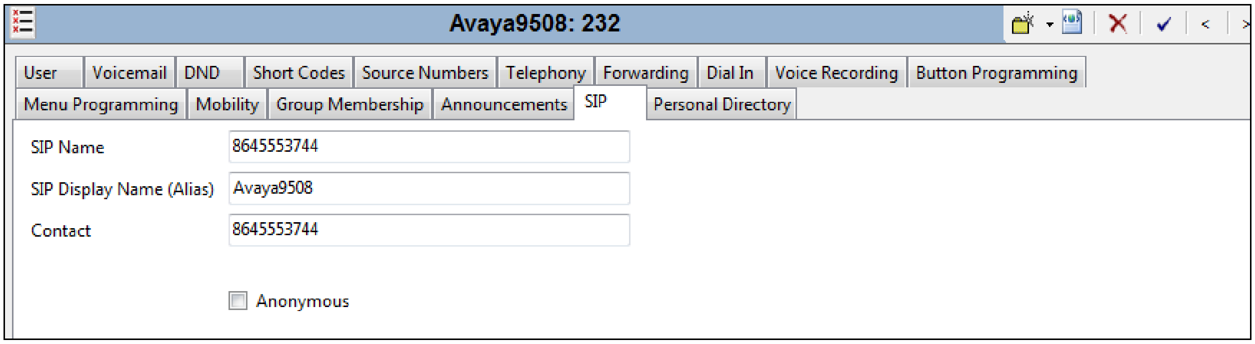
In this section, an example of an IP Office User will be illustrated. In the interests of brevity, not all users shown in **Figure 1** will be presented, since the configuration can be easily extrapolated to other users. To add a User, right click on **User** in the Navigation pane, and select **New**.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

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

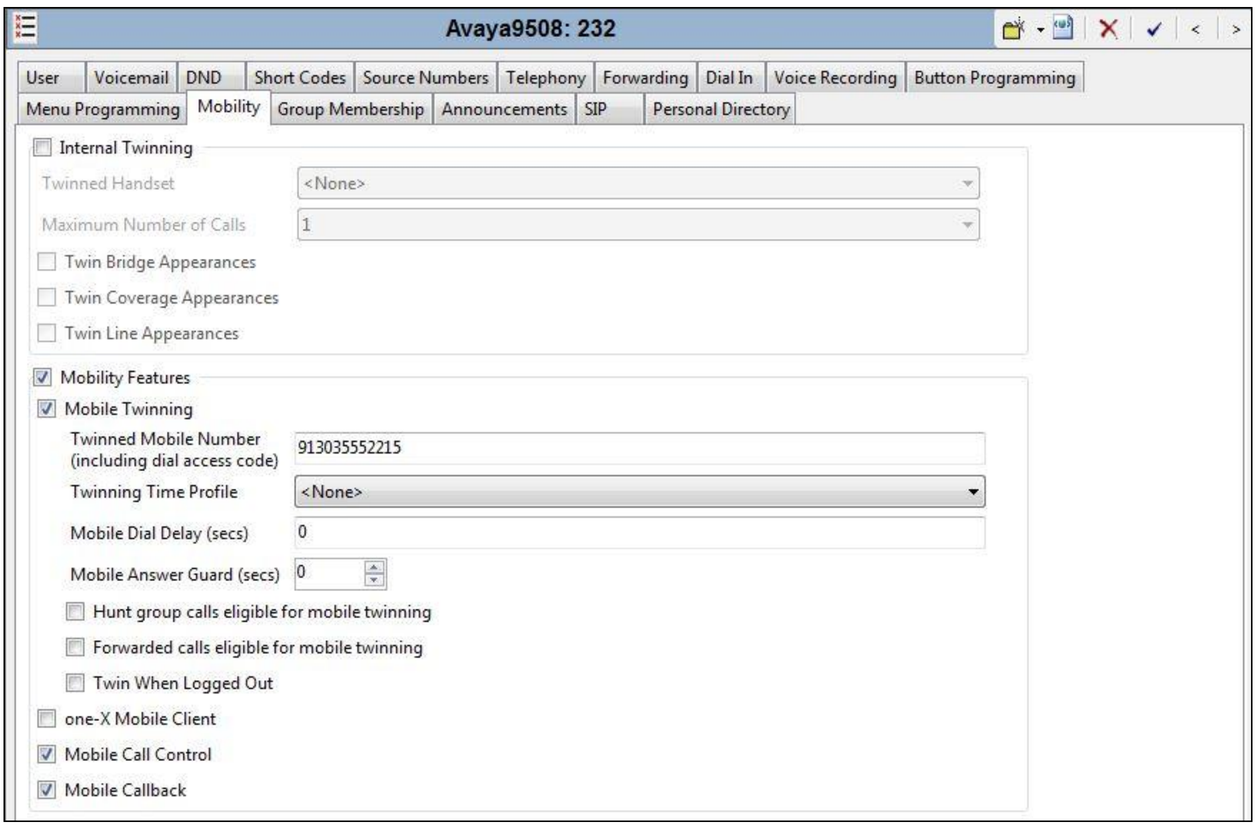


The following screen shows the **SIP** tab for User 232. The **SIP Name** and **Contact** parameters are configured with the DID number of the user, 864-555-3744. These parameters configure the user part of the SIP URI in the From header for outgoing SIP trunk calls, and allow matching of the SIP URI 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, in this case 913035552215. Other options can be set according to customer requirements.

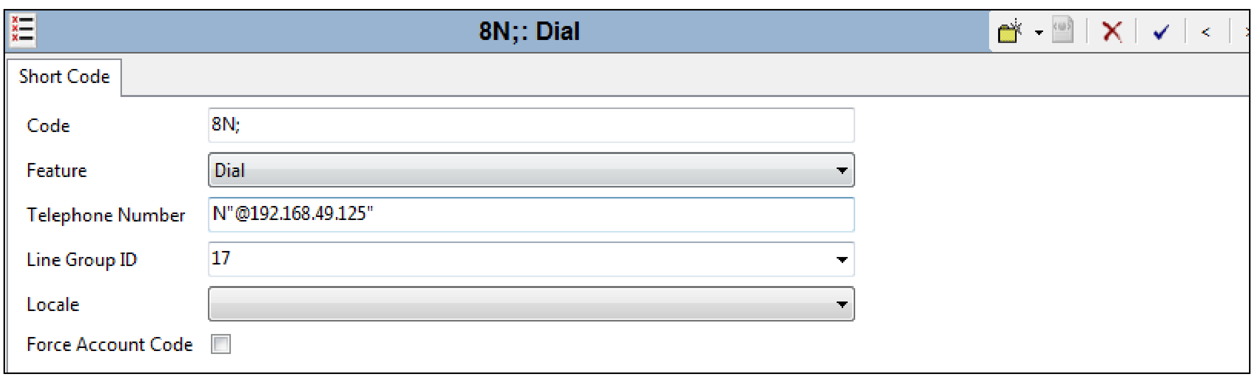


5.6. Short Codes

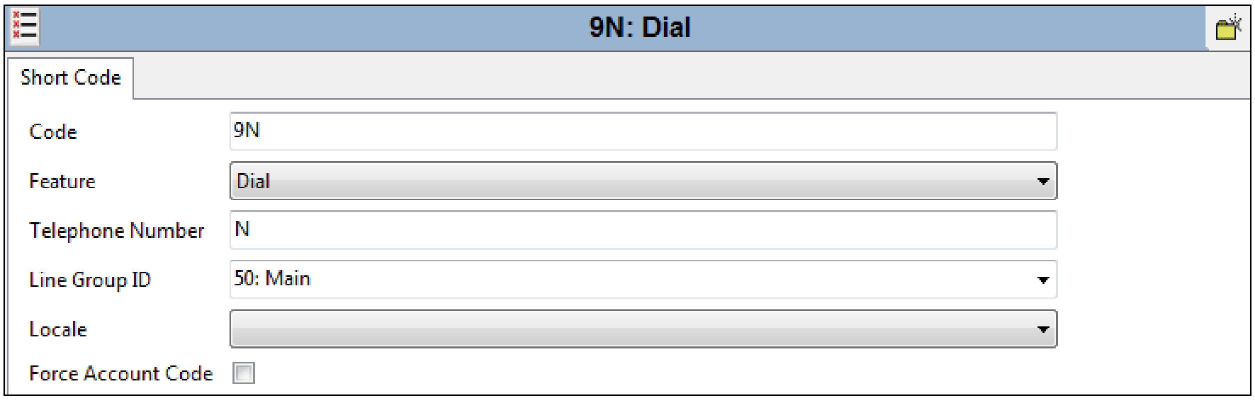
In this section, various examples of IP Office short codes will be illustrated. To add a short code, right click on **Short Code** in the Navigation pane, and select **New**. To edit an existing short code, click **Short Code** in the Navigation pane, and the short code to be configured in the Group 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"@192.168.49.125". The **Telephone Number** field is used to construct the Request URI and To Header in the outgoing SIP INVITE message. The value "N" represents the number dialed by the user. The IP address 192.168.49.125 represents the IP address of the Windstream SIP Trunk service. 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 Windstream (**Section 5.4.4**).

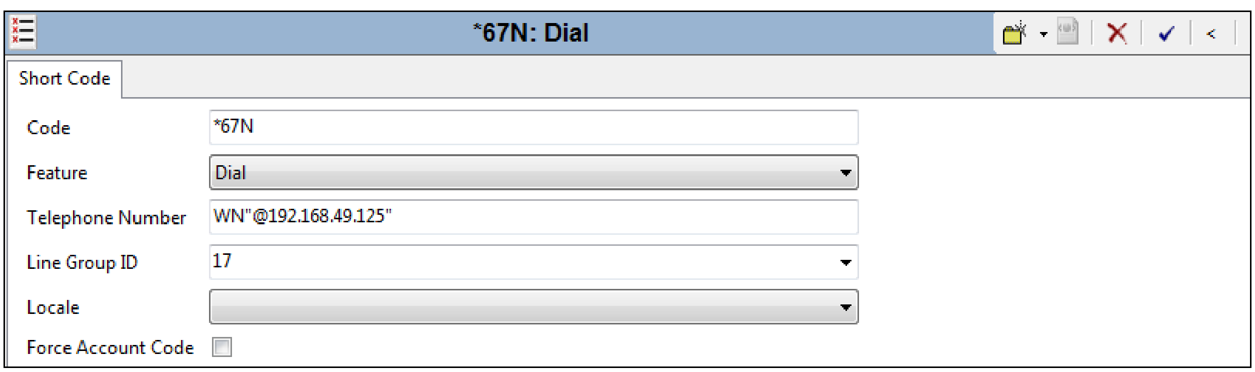
This simple short code will allow an IP Office user to dial the digit 8 followed by any telephone number, symbolized by the letter N, to reach the SIP Line to Windstream. "N" can be any number such as a 1+10 digit number, a toll free number, directory assistance (e.g., 411), etc. This short code approach 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 Windstream.



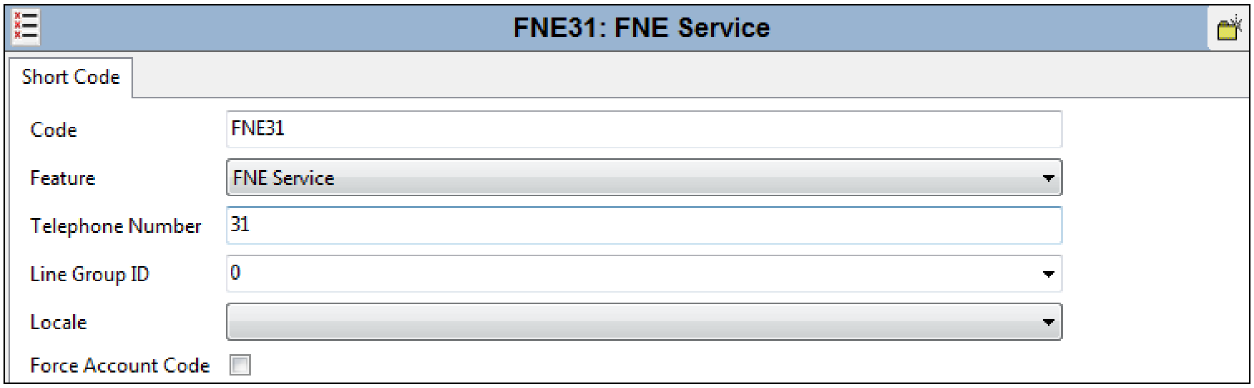
The simple "8N;" short code previously illustrated does not provide a means of alternate routing if the configured SIP Line is out of service or temporarily not responding. When alternate routing options and/or more customized analysis of the digits following the short code are desired, the Automatic 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 Windstream 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-Preferred-Identity header along with "Privacy: id". With these headers, Windstream will prevent presentation of the caller id to the called PSTN destination.



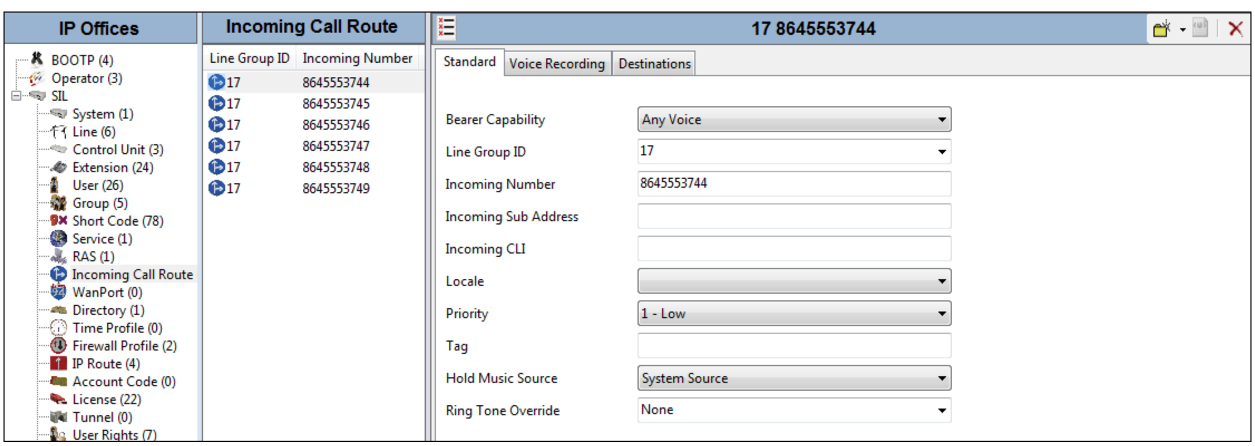
The following screen illustrates a short code that acts like a feature access code rather than a means to access a SIP Line. In this case, the **Code** “FNE31” is defined for **Feature** “FNE Service” to **Telephone Number** “31” (Mobile Call Control). This short code will be used as means to allow a Windstream 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.



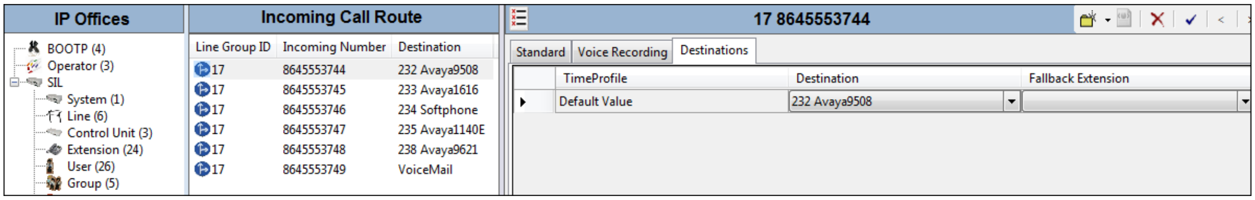
5.7. Incoming Call Routes

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, the incoming call route for **Incoming Number** “8645553744” is illustrated. The **Line Group Id** is “17”, matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Line to Windstream in **Section 5.4.4**.

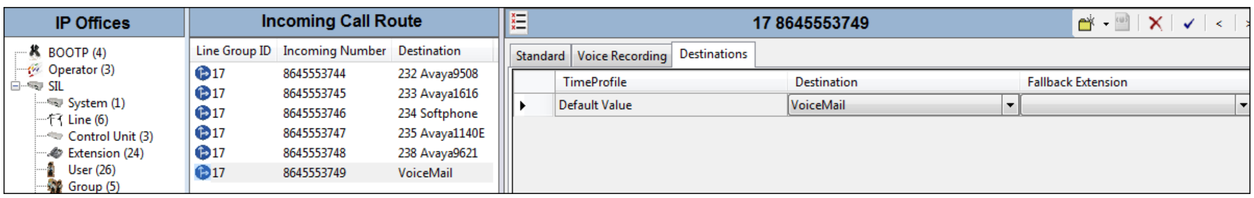


Select the **Destinations** tab. From the **Destination** drop-down, select the extension to receive the call when a PSTN user dials 864-555-3744.

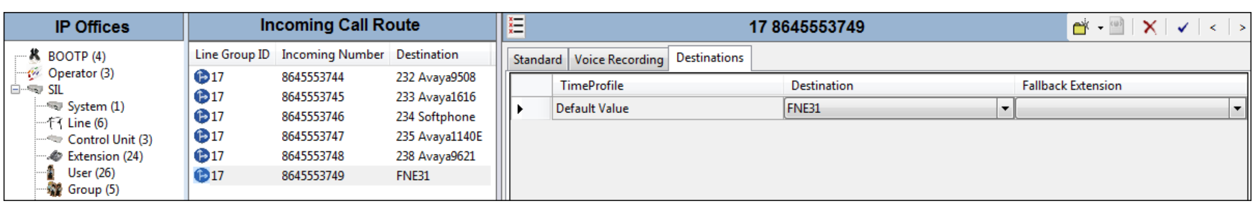


Incoming Call Routes for other direct mappings of DID numbers to IP Office users listed in **Figure 1** are omitted here, but can be configured in the same fashion.

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 pull-down menu. For example, the following **Destinations** tab for an incoming call route contains the **Destination** "VoiceMail" entered manually. An incoming call to 864-555-3749 will be delivered directly to voicemail, allowing the caller to log-in to voicemail and access messages.



At different times during testing, the destination for 864-555-3749 was changed to test the IP Office Mobile Call Control feature. The following **Destinations** tab for the incoming call route contains the **Destination** "FNE31" entered manually. With this destination, an incoming call to 864-555-3749 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 (**Section 5.5**); otherwise the IP Office responds with a 486 Busy Here and the caller will hear a busy tone.



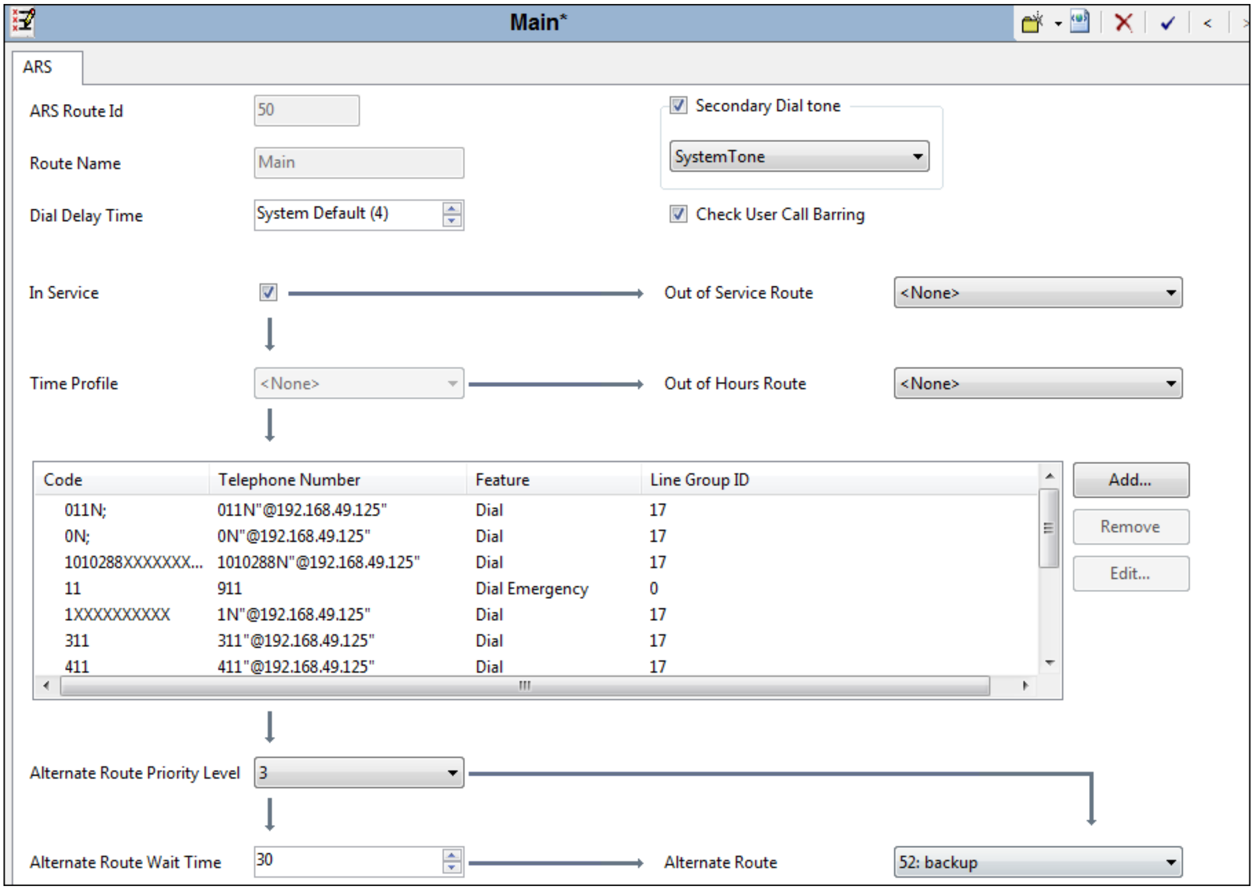
5.8. ARS and Alternate Routing

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 certain service numbers following the access code should prefer a different outgoing line group, ARS can be used to distinguish the call behaviors.

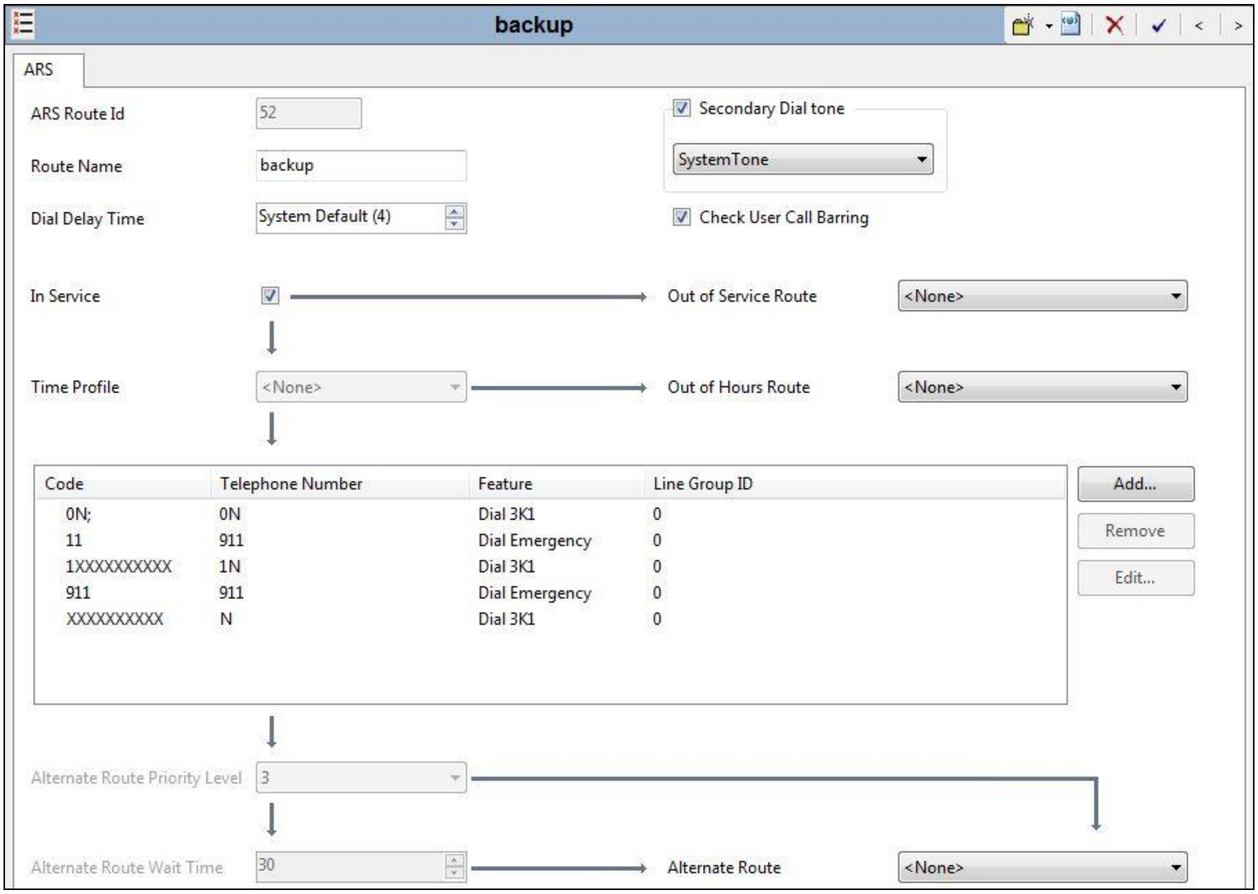
To add a new ARS route, right-click **ARS** in the Navigation pane, and select **New**. To view or edit an existing ARS route, select **ARS** in the Navigation pane, and select the appropriate route name in the Group pane.

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 9-1-303-555-1234, 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 9-1-303-555-1234, 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**.

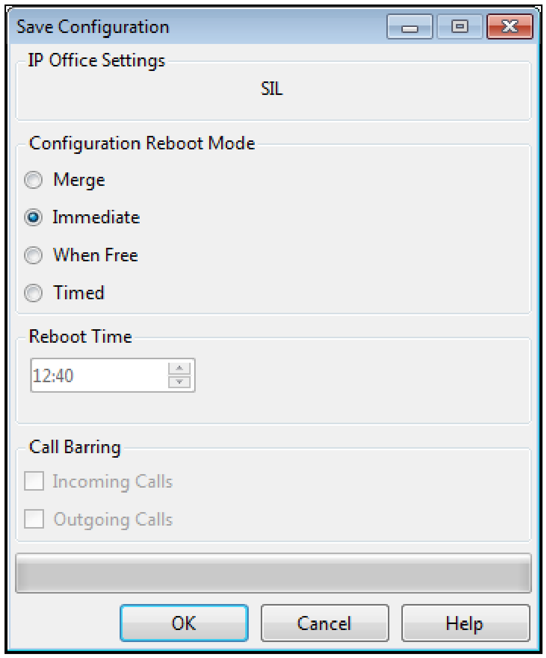


If a primary route experiences a network outage such that no response is received to an outbound INVITE, IP Office successfully routes the call via the backup route. The user receives an audible tone when the re-routing occurs and may briefly see “Waiting for Line” on the display.

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. Windstream Configuration

Windstream is responsible for the configuration of its SIP Trunking Service. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise site (i.e., the IP address of the public interface on the Avaya IP Office). Windstream will provide the customer the necessary information to configure the Avaya IP Office including:

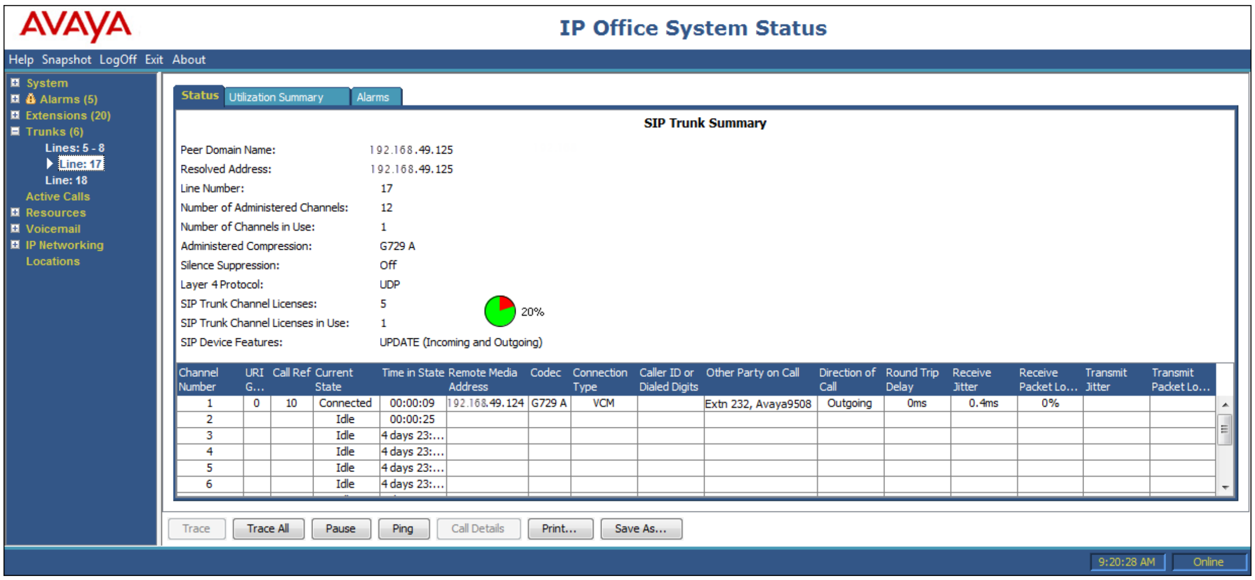
- Network edge IP addresses of the Windstream SIP Trunking Service
- Transport and port for the Windstream SIP Trunking connection to the Avaya IP Office at the enterprise
- DID numbers to assign to users at the enterprise
- Supported codecs and their preference order

7. Verification Steps

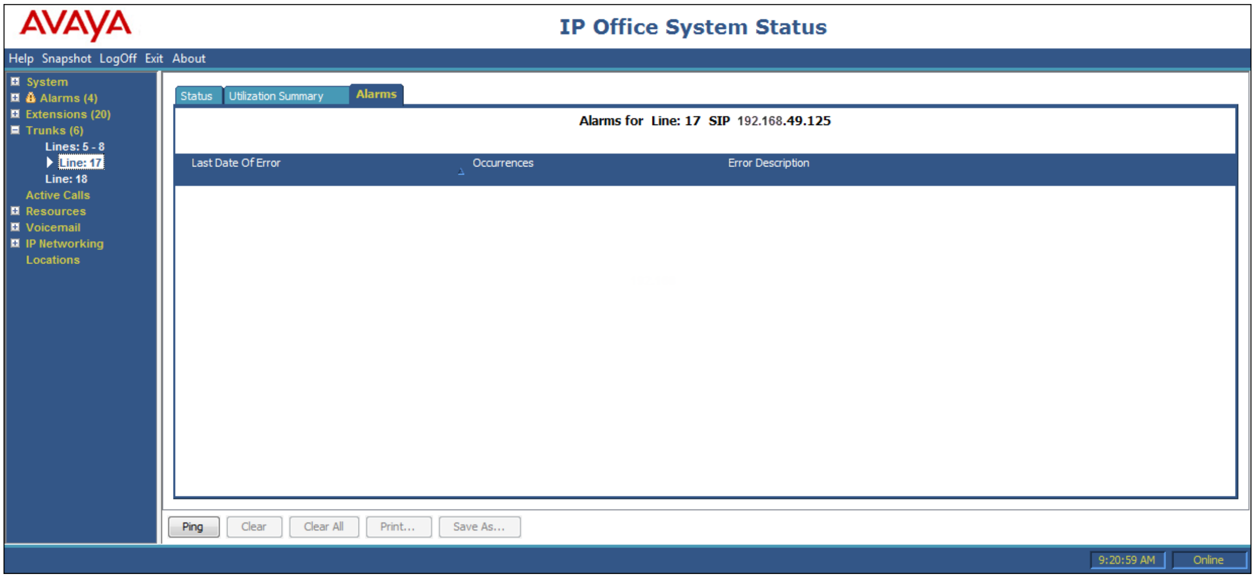
This section provides example verifications of the Avaya configuration with Windstream SIP Trunk service.

The following steps may be used to verify the configuration:

- Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application from **Start → Programs → IP Office → System Status** on the PC where Avaya IP Office Manager was installed. Select the SIP line of interest from the left pane. On the **Status** tab in the right pane, verify that the Current Stat for each channel (The following screen shot shows an active call).



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



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 9.0 can be successfully combined with a Windstream 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 Windstream 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 Windstream.

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 9.0 Installing IP500/IP 500 V2,* Document Number 15-601042, November 2013

[2] *IP Office Manager*, Document Number 15-601011, November 2013

[3] *IP Office Application server 9.0 Installation and Maintenance*, August 2013

[4] *IP Office 9.0 Using System Status*, Document Number 15-601758, August 2013

[5] *Administering Avaya Flare® Experience for iPad Devices and Windows,* September 2013

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

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

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.