Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Windstream SIP Trunking with Avaya IP Office Release 8.0 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between Service Provider Windstream and Avaya IP Office Release 8.0.

Windstream SIP Trunking 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 costs 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 via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between Service Provider Windstream and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 8.0, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya SIP, H.323, digital, and analog endpoints.

The Windstream SIP Trunking service referenced within these Application Notes is designed for business customers. The service enables local and long distance PSTN calling via standards-based SIP trunks as an alternative to legacy analog or digital trunks, without the need for additional TDM enterprise gateways and the associated maintenance costs.

The Windstream SIP Trunking service does not require the enterprise IP PBX to register with the SIP network via SIP credentials. Customers are authenticated via IP authentication.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Avaya IP Office to connect to Windstream SIP Trunking service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality tests listed in **Section 2.1**.

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

A simulated enterprise site with Avaya IP Office was connected to Windstream SIP Trunking service. To verify SIP trunking interoperability, the following features and functionality were exercised during the interoperability compliance test:

- Response to SIP OPTIONS queries.
- Incoming PSTN calls to various phone types. Phone types included SIP, H.323, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the Service Provider.
- Outgoing PSTN calls from various phone types. Phone types included SIP, H.323, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the Service Provider.
- Inbound and outbound PSTN calls to/from the Avaya IP Office Softphone.
- Inbound and outbound long hold time call stability.
- Various call types including: local, long distance, international, outbound toll-free, operator service and directory assistance.
- G.711MU and G.729A codeccs.

- Caller number/ID presentation.
- Privacy requests (i.e., caller anonymity) and Caller ID restriction for inbound and outbound calls.
- DTMF transmission using RFC 2833.
- Voicemail navigation for inbound and outbound calls.
- Telephony features such as hold and resume, transfer, and conference.
- Off-net call forwarding
- Twinning to PSTN mobile phones on inbound calls.
- SIP REFER on transfer and forward

Items not supported or not tested included the following:

- Inbound toll-free and outbound emergency calls (911) are supported but were not tested as part of the compliance test.
- Operator (0) calls, Operator Assisted (0+10 digits) dialing and 411 / 1411 services are not supported and therefore were not tested.
- T.38 fax is not currently supported by Windstream.

2.2. Test Results

Interoperability testing of Windstream SIP Trunking was completed with successful results for all test cases with the exception of the observations/limitations described below.

- **SIP OPTIONS response:** IP Office sent SIP OPTIONS messages to the Windstream SIP gateway and Windstream returned a “501 Not Implemented” response. “200 OK" is the usual response though IP Office only requires a response, not a specific message.
- **Codec mis-match on outbound calls**: When incompatible codecs were configured, outbound calls completed but caller received dead air and then the call dropped and phone displayed “INCOMPATIBLE”. Inbound calls with mis-matched codecs did not go through and caller received an appropriate rejection message.
- **Fax**: Only G.711 Fax was tested.
- **Outbound call number restriction:** On outbound calls from IP Office when Caller-ID was restricted, Windstream did not allow the calls to go through. Inbound calls with restricted Caller-ID completed and the Caller-ID displayed was “Anonymous / Private”
- **Off-net call forwarding**: When an inbound call was placed to a station with forwarding enabled and the forwarding number was an off-net PSTN number, **RTP Keepalives** configuration on the IP Office LAN 2 interface must be enabled or the call would complete but receive dead-air. LAN2 is the interface typically used to connect to the SIP Service Provider.
- **Call display update**: Call display was not properly updated on PSTN phone to reflect the true connected party on calls that are transferred to the PSTN from the enterprise. After the call transfer was completed, the PSTN phone showed the party that initiated the transfer instead of the actual connected party.

- **IP Softphone**: To allow for proper transfer and forwarding capabilities, **Call Waiting On** and **Off hook Station** needed to be enabled in IP Office **User → Telephony → Call Settings** configuration page.
- **SIP REFER**: Windstream does not support SIP REFER at this time due to some Metaswitch limitations, but will support SIP REFER in a later release. Consequently, SIP REFER support was turned off on Avaya IP Office for the compliance test.

2.3. Support

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

For technical support on Windstream SIP Trunking, contact Windstream at http://www.windstream.com/ or 1-866-780-8639.

3. Reference Configuration

Figure 1 below illustrates the test configuration. The test configuration shows an enterprise site connected to Windstream SIP Trunking service through the public IP network. Located at the enterprise site is an Avaya IP Office 500v2 with the COMBO6210/ATM4 expansion card which provides connections for 6 digital stations, 2 analog stations, 4 analog trunks to the PSTN as well as 10-channel VCM (Voice Compression Module) for supporting VoIP codecs. The LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public IP network. Endpoints include an Avaya 9600 Series IP Telephone (with H.323 firmware), an Avaya 1120e Series IP Telephone (with SIP firmware), an Avaya 1416 Digital Telephone, an Avaya Analog Telephone and an Avaya IP Office Softphone. The site also has a Windows 2003 Server running Avaya Voicemail Pro for providing voice messaging service to the Avaya IP Office users. A separate Windows XP PC runs Avaya IP Office Manager to configure and administer the Avaya IP Office system. Mobility Twinning is configured for some of the Avaya IP Office users so that calls to these user's phones will also ring and can be answered at the configured mobile phones.

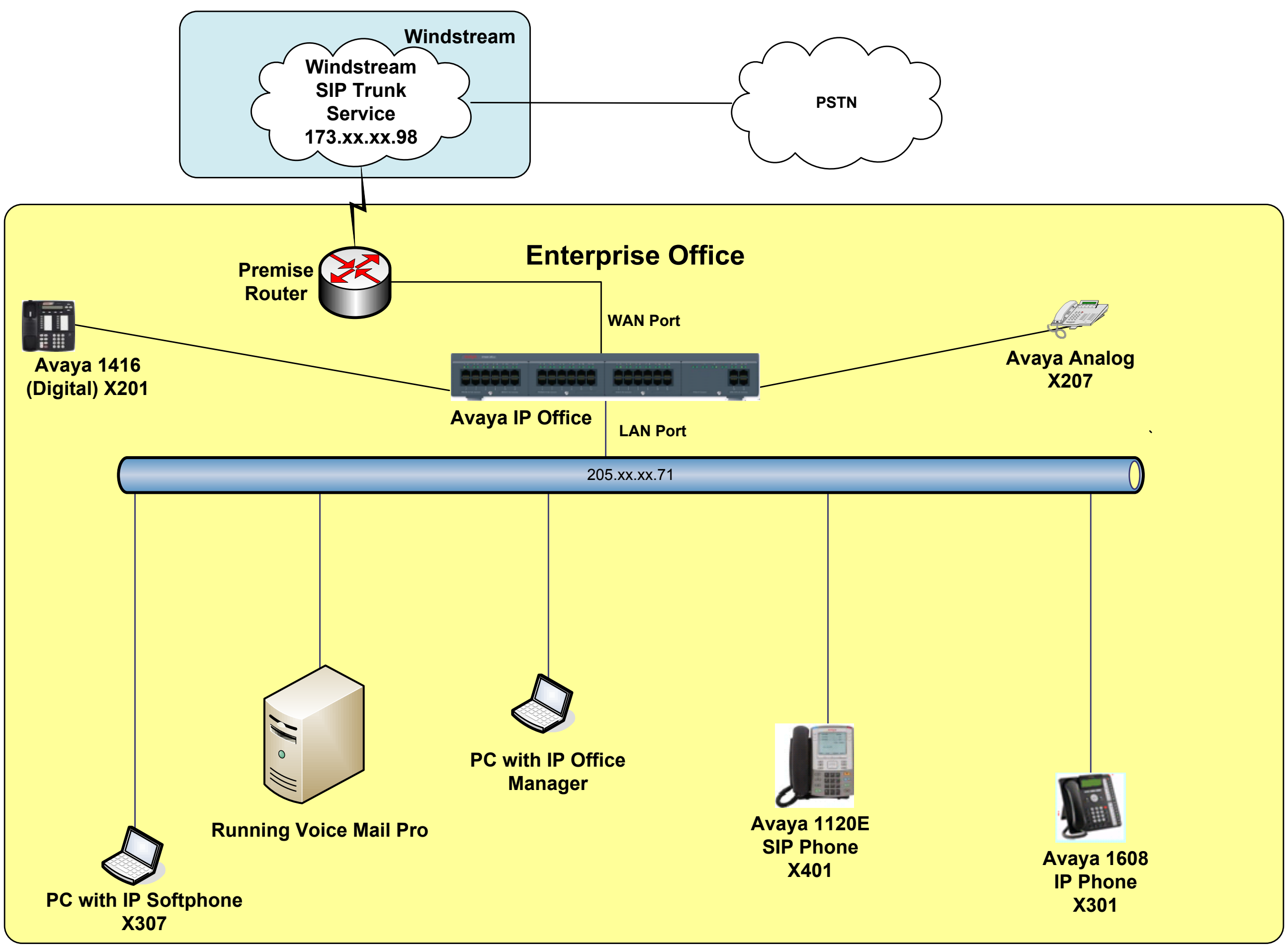


Figure 1: Test Configuration for Avaya IP Office with Windstream SIP Trunking Service

For security purposes, the real public IP addresses and PSTN routable phone numbers used in the compliance test are masked in these Application Notes.

For the purposes of the compliance test, Avaya IP Office users dialed a short code of 9 + N digits to send digits across the SIP trunk to Windstream. The short code of 9 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to Windstream. For calls within the North American Numbering Plan (NANP), the user would dial 11 (1 + 10) digits. Thus for these NANP calls, Avaya IP Office would send 11 digits in the Request URI and the To headers of an outbound SIP INVITE message. IP Office was configured to send 10 digits in the From and PAI headers. For inbound calls, Windstream SIP Trunking sent 10 digits in the Request URI, To, and From headers of inbound SIP INVITE messages.

Windstream uses the phone number in the From header of a SIP INVITE message to authenticate the calling party. Thus, a call will be rejected by the network unless the From header contains a number known to Windstream. This is especially important for calls inbound from the PSTN which are redirected back to the PSTN by call forwarding or twinning. For call forwarding, Avaya IP Office always sends the number of the forwarding phone in the From header. This is a number known to Windstream. As a result, the call display on the destination phone shows the forwarding party not the original caller. For twinning, this behavior can be slightly altered through configuration. See **Section 5.5** and **5.6** for details.

In an actual customer configuration, the enterprise site may also include additional network components between the Service Provider and Avaya IP Office such as a Session Border Controller or data firewall. A complete discussion of the configuration of these devices is beyond the scope of these Application Notes. However, it should be noted that SIP and RTP traffic between the Service Provider and Avaya IP Office must be allowed to pass through these devices.

4. Equipment and Software Validated

The following equipment and software/firmware were used for the sample configuration provided:

Avaya Telephony Components	
Equipment / Software	**Release / Version**
Avaya IP Office 500v2	8.0 (42)
Avaya IP Office COMBO6210/ATM4 Module	8.0 (42)
Avaya Voicemail Pro	8.0 (1009)
Avaya IP Office Manager	10.0 (42)
Avaya 1608 IP Telephone (H.323)	Avaya one-X Deskphone Value Edition 1.2.2
Avaya 1120E IP Telephone (SIP)	SIP1120 version 04.01.13.00
Avaya 1416 Digitial Telephone	N/A
Avaya Analog Telephone	N/A
Avaya IP Office Softphone	3.2.3.15 64595
Windstream SIP Trunking	
Equipment	**Release**
Metaswitch Adtran TA908E	7.30 SU 56 R10.1

5. Configure Avaya IP Office

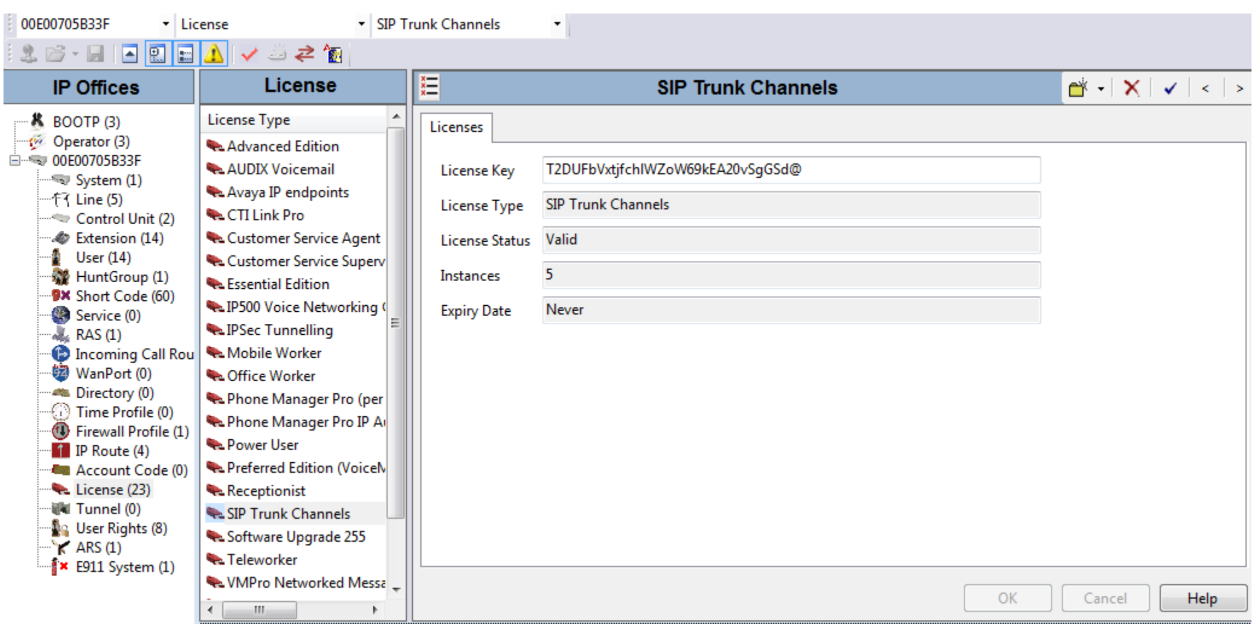
This section describes the Avaya IP Office configuration necessary to support connectivity to Windstream SIP Trunking service. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials. A management window will appear similar to the one shown in the next section. 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. These panes will be referenced throughout the Avaya IP Office configuration. Proper licensing as well as standard feature configurations that are not directly related to the interface with the Service Provider (such as LAN interface to the enterprise site and IP Office Softphone support) is assumed to be already in place.

5.1. Licensing and Physical Hardware

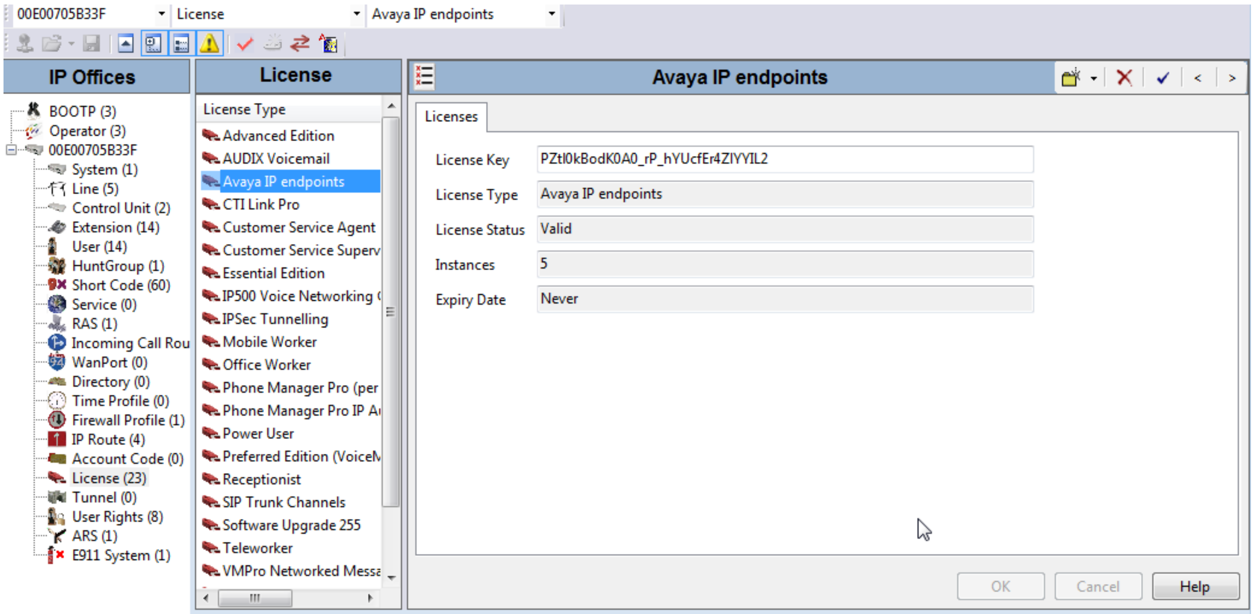
The Licensing and Physical Hardware Section is not directly related to the compliance test. This section was included for completeness and reference only.

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

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



If Avaya IP Telephones will be used as is the case in these Application Notes, verify the Avaya IP endpoints license. Click **License** in the Navigation pane and **Avaya IP endpoints** in the Group pane. Confirm a valid license with sufficient “Instances” in the Details pane.



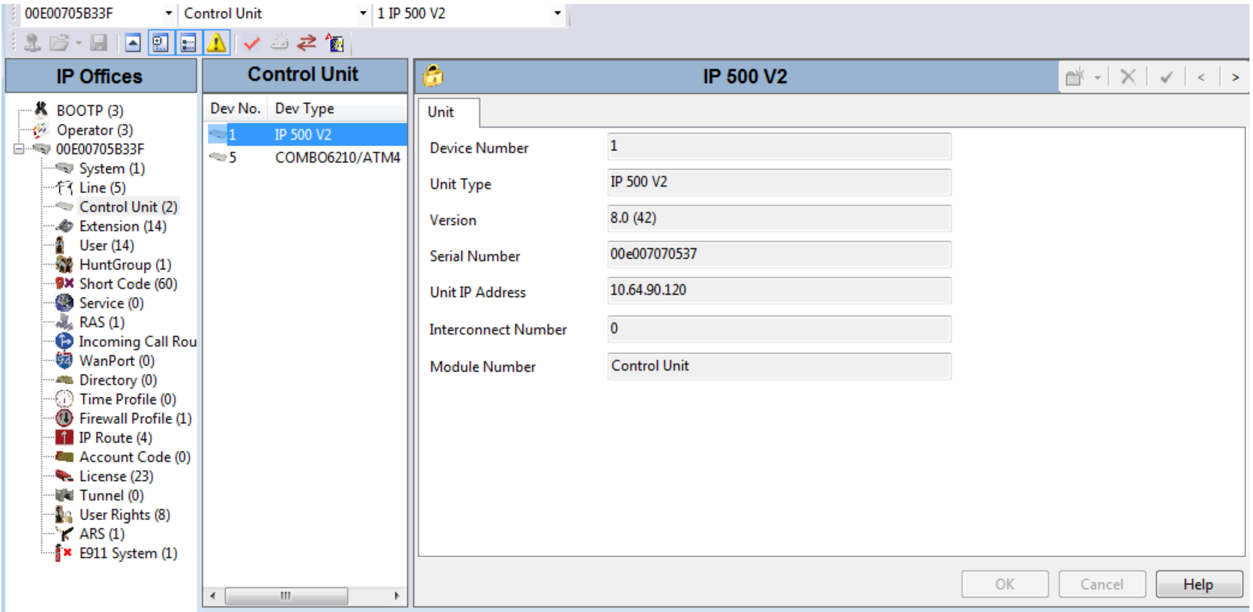
5.2. Physical, System and Network Settings

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

5.2.1. Physical Configuration and Settings

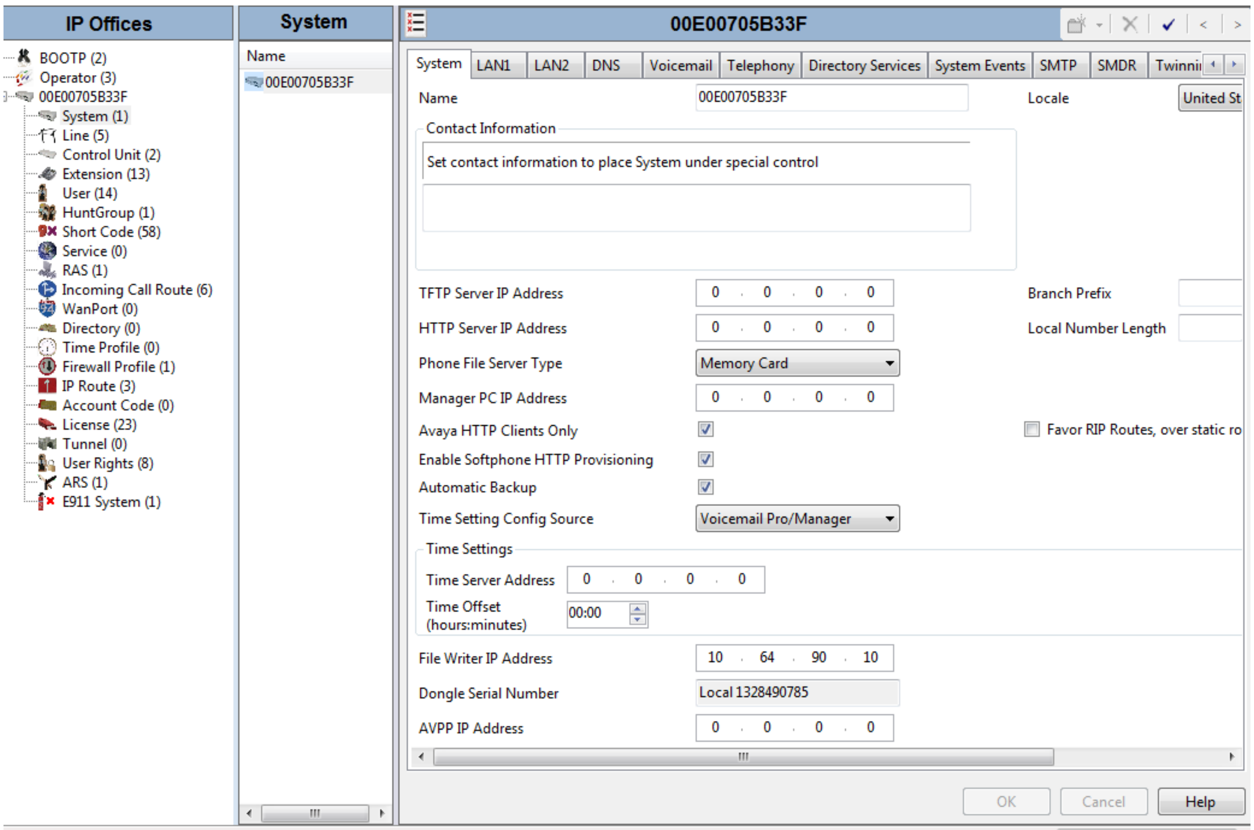
In the sample configuration, looking at the IP Office 500 v2 from left to right, the fifth slot from left to right contains the COMBO 6210/ ATM4 card, an analog and digital combo card. Slot two through four was blank (i.e., no module is physically inserted). The COMBO6210/ATM4 module allows connection of analog and Avaya digital endpoints. In the testing of the sample configuration, an analog telephone or a fax machine is connected to port 7 of the module and an Avaya 1416 digital station is connected to port 1.

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.



5.2.2. System Settings

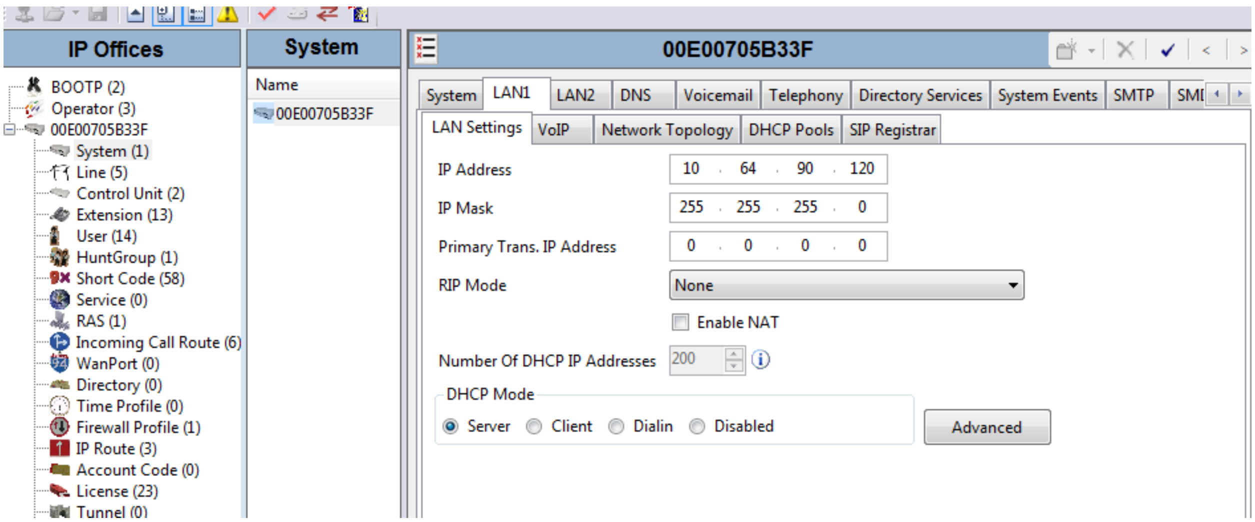
Select **System** from the Navigation pane to access the main system menu. The **Name** field can be used for a descriptive name of the system such as **Windstream.** The MAC address ***00E00705B33F*** was used as the system name in this configuration. The **Avaya HTTP Clients Only** and **Enable SoftPhone HTTP Provisioning** boxes are checked to facilitate Avaya IP Office Softphone usage.



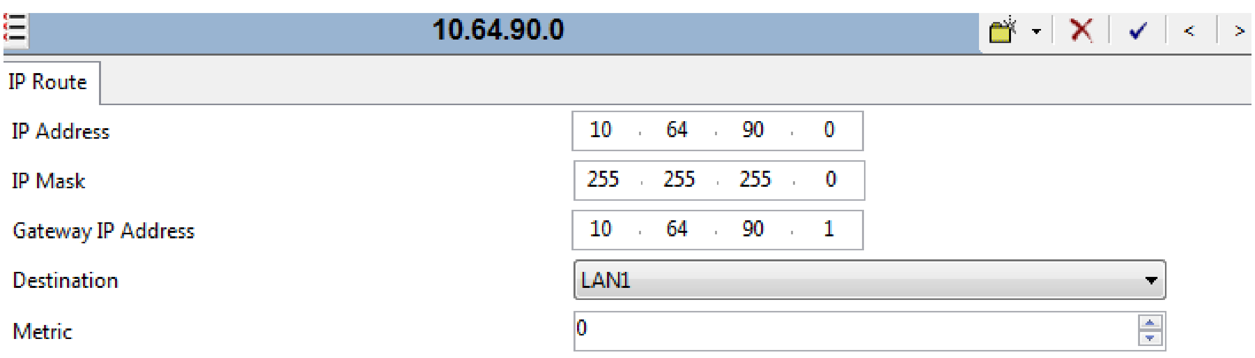
In the compliance test, the LAN1 interface was used to connect the Avaya IP Office to the enterprise site IP network. The LAN1 interface configuration is not directly relevant to the interface with Windstream SIP Trunking service, therefore, this section was included for completeness and reference only.

5.2.3. LAN1 Settings

Under **System → LAN1 → LAN Settings**, the **IP Address** of the LAN Interface is set, along with the **IP Mask** of the subnet and the **DHCP Mode**. Here the LAN1 IP Address is **10.64.90.120** with a mask of **255.255.255.0** and the **DHCP Mode** is set to **Server**. The ⓘ indicates that Advanced DHCP settings are being used and by placing the mouse cursor over the ⓘ button you can see the notice. The **DHCP Pools** tab will display the DHCP configuration (not shown).

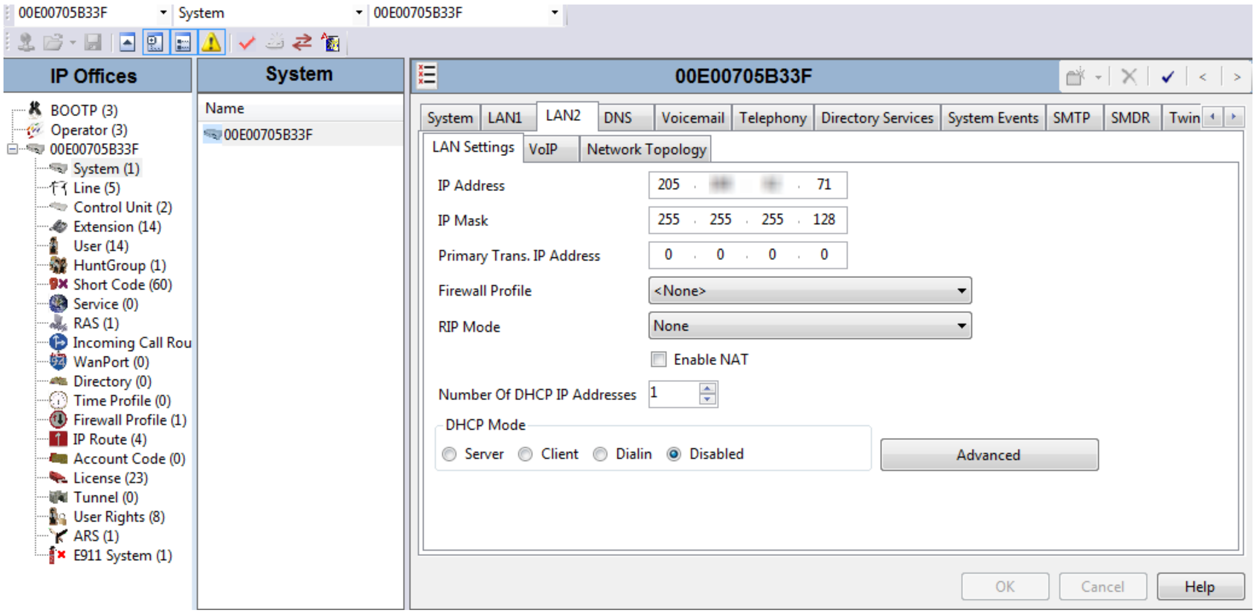


In the sample configuration, the IP Office LAN1 port is physically connected to the local area network switch at the IP Office customer site. The default gateway for this network is 10.64.90.1. To add an IP Route in IP Office, right-click **IP Route** from the Navigation pane, and select **New.** To view or edit an existing route, select **IP Route** from the Navigation pane, and select the appropriate route from the Group pane. The following screen shows the Details pane with the relevant default route using **Destination** LAN1.

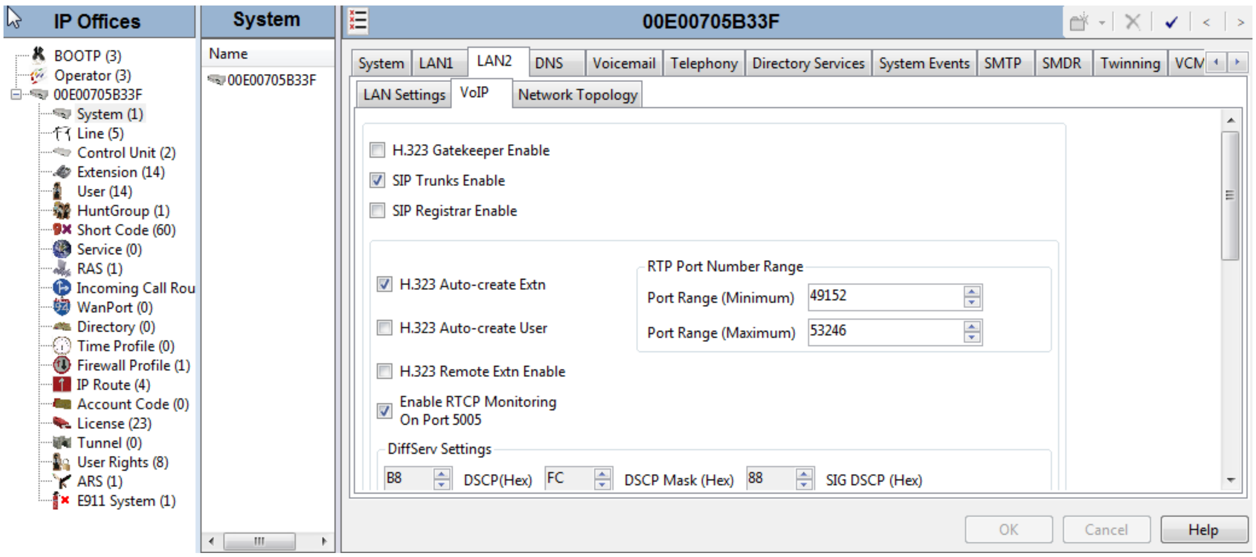


5.2.4. LAN2 Settings

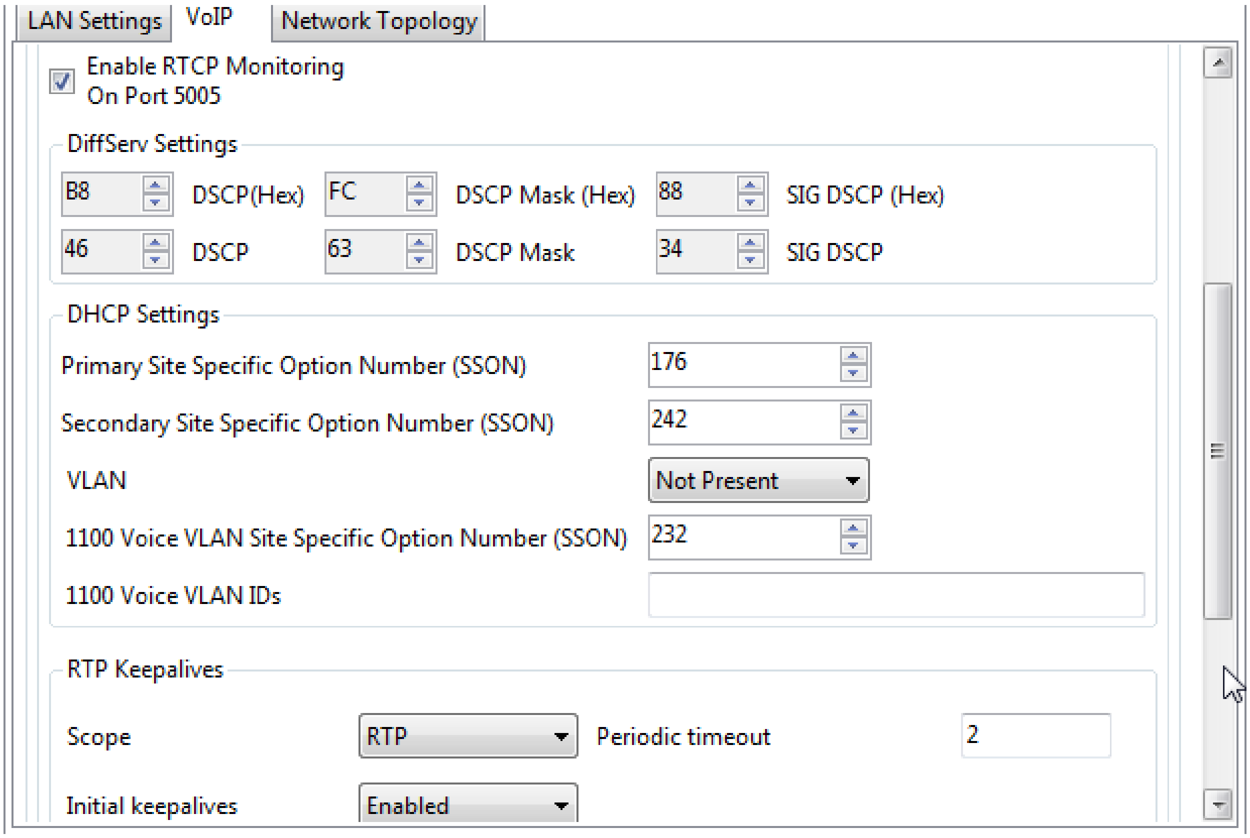
In the sample configuration, the MAC address ***00E00705B33F*** was used as the system name and the WAN port (LAN2) was used to connect the Avaya IP Office to the public network. The LAN2 settings correspond to the WAN port on the Avaya IP Office system. To access the LAN2 settings, first navigate to **System (1) → 00E00705B33F** in the Navigation and Group Panes respectively, and then navigate to the **LAN2→ LAN Settings** tab in the Details Pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office WAN port. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements. Click "**OK**" when finished and save the configuration.



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. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2. Avaya IP Office can also 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

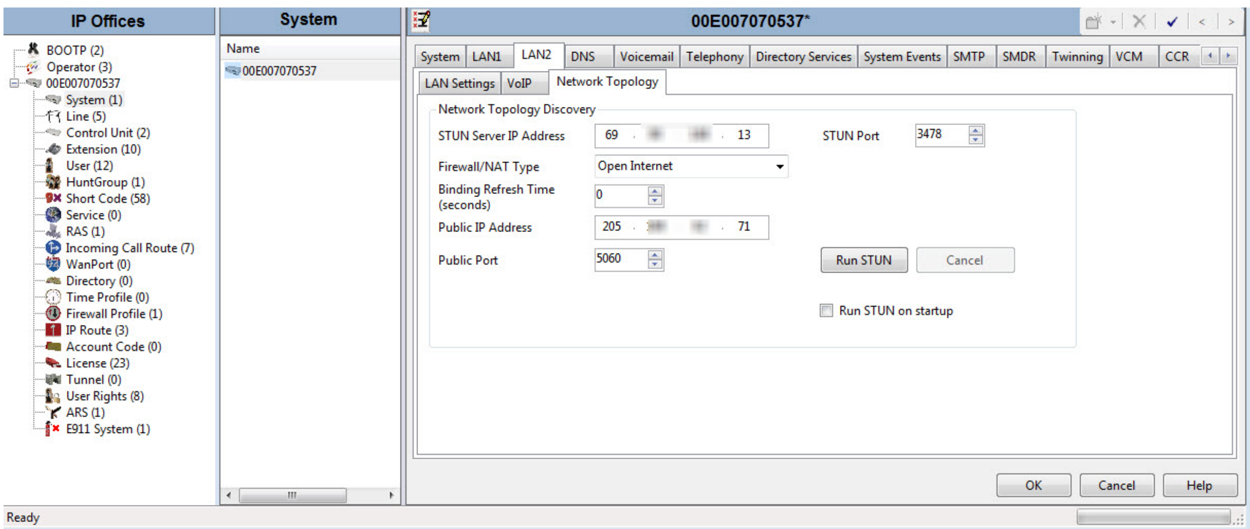


Below is the bottom half of the SIP Line VoIP screen. To prevent possible loss of audio path during some off-net call forward scenarios, it is recommended to set the following fields under **RTP Keepalives**: set **Scope** to ***RTP***, set **Initial keepalives** to ***Enable*** and enter an appropriate **Periodic timeout** value from ***1*** to ***180*** seconds. The specific values used for the compliance test are shown in the example below. All other parameters should be set according to customer requirements.



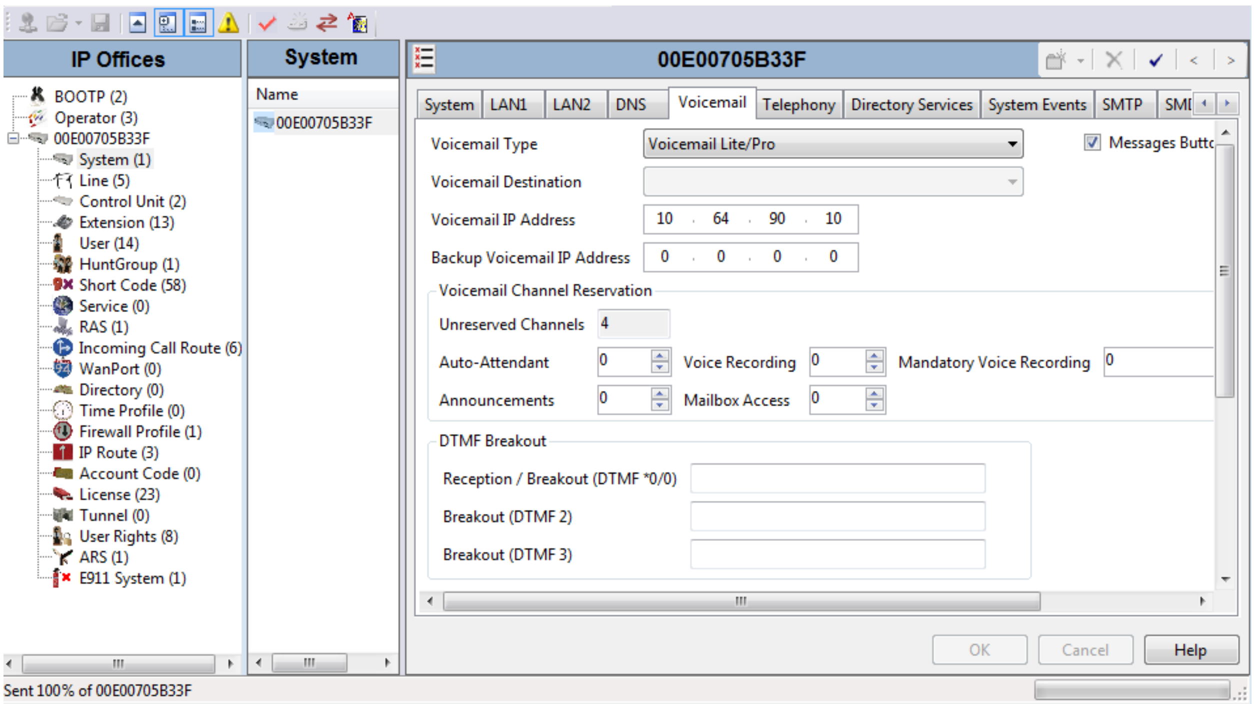
On the **Network Topology** tab in the Details Pane, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to ***Open Internet***. With this configuration, STUN will not be used.
- Set **Binding Refresh Time (seconds)** to ***0***. This value is used as one input to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the Service Provider. Windstream responded to OPTIONS messages with "501 Not Implemented", so this was set to ***0***, which means that **OPTIONS** messages are sent only every 5 minutes.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port.
- All other parameters should be set according to customer requirements.
- Click "**OK**" when finished and save the configuration.



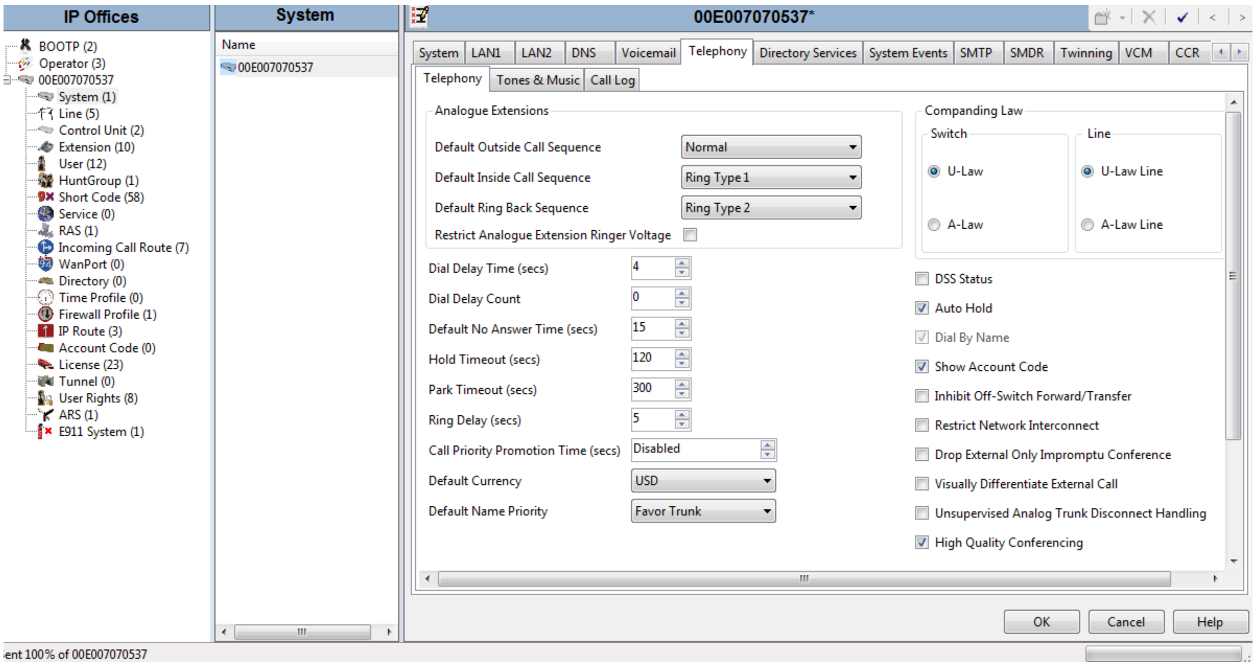
5.3. Voicemail

In the sample configuration, **Voicemail Lite/Pro** was used. To change voicemail settings, navigate to **System→Voicemail** as shown below.



5.4. System Telephony Settings

Navigate to the **Telephony → Telephony** tab in the Details Pane. Choose the **Companding Law** typical for the enterprise location. For North America, ***ULAW*** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the Service Provider across the SIP trunk.

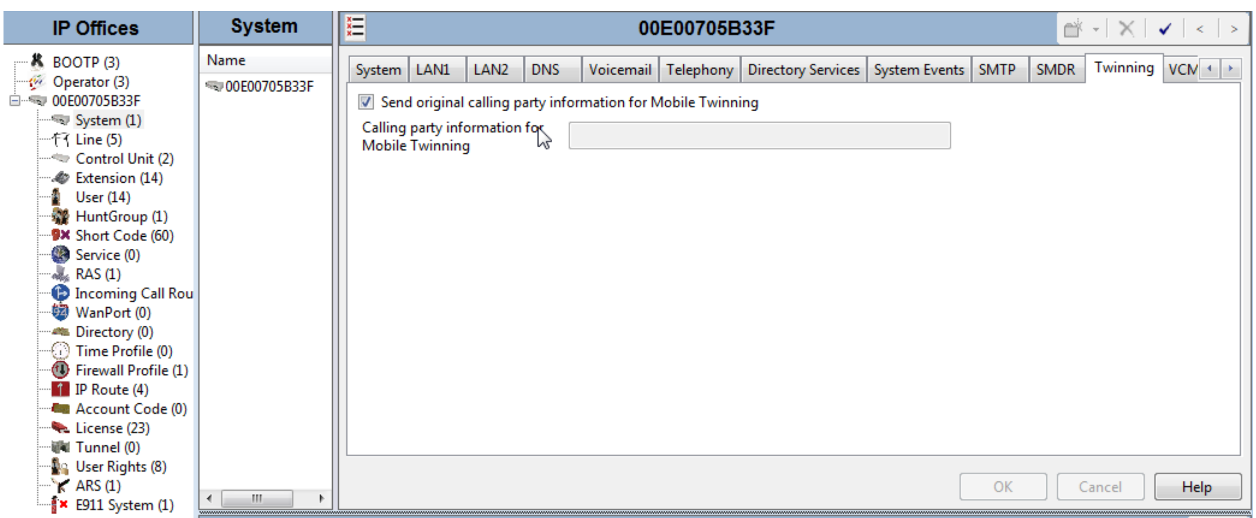


5.5. Twinning Calling Party Settings

When using twinning, the calling party number displayed on the twinned phone is controlled by two parameters. The first parameter is the **Send original calling party information for Mobile Twinning** box on the **System→Twinning** tab. The second parameter is the **Send Caller ID** parameter on the **SIP Line** form (shown in **Section 5.6**). If **Send original calling party information** for Mobile Twinning on the **System→Twinning** tab is set, the setting of the second parameter is ignored and Avaya IP Office will send the following in the SIP From Header:

- On calls from an internal extension to a twinned phone, Avaya IP Office will send the calling party number of the originating extension.
- On calls from the PSTN to a twinned phone, Avaya IP Office will send the calling party number of the host phone associated with the twinned destination (instead of the number of the originating caller).

The above behavior in Avaya IP Office Release 8 is the same as in Avaya IP Office Release 7 and was tested and verified in the compliance test. Avaya IP Office Release 8 also provides an alternative method of sending caller ID through SIP Diversion header (configured via unchecking **Send original calling party information for Mobile Twinning** here then selecting ***Diversion Header*** for the **Send Caller ID** parameter on the **SIP Line** form in **Section 5.6**). This alternative configuration could provide more accurate caller ID information if the Service Provider supports the SIP Diversion header (not tested in the compliance test). For the compliance test, the **Send original calling party information for Mobile Twinning** box in the **System→Twinning** tab was checked which overrides any setting of the **Send Caller ID** parameter on the **SIP Line** form.



5.6. Administer SIP Line

This section shows the configuration screens for the **SIP Line configuration** in IP Office Release 8. Since IP Office Release 8 introduced new SIP Line parameters and re-oriented existing parameters, this section has the most substantive changes in these Application Notes.

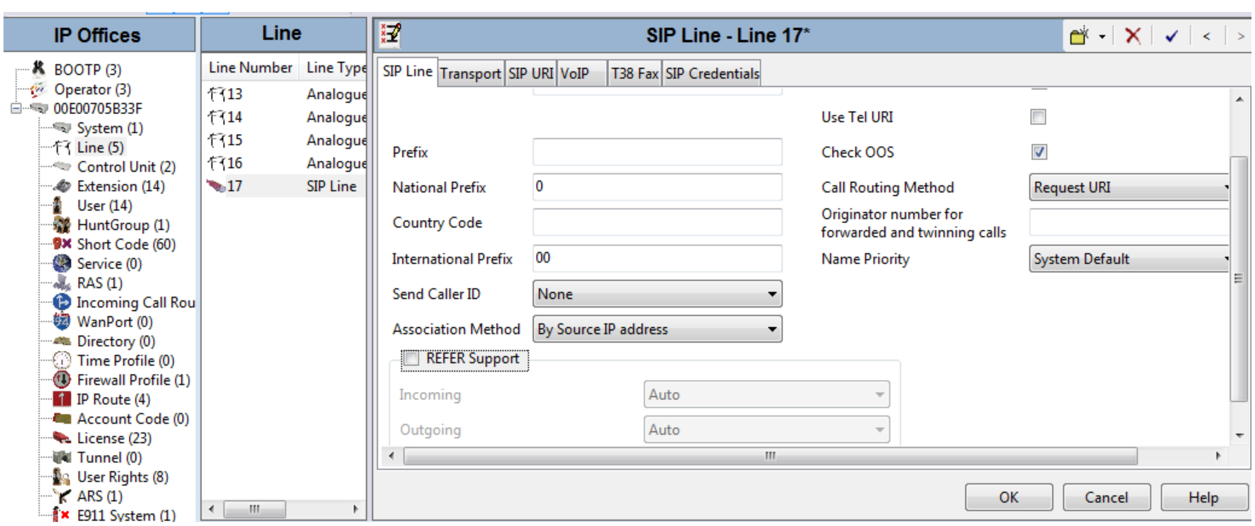
5.6.1. SIP Line – SIP Line Tab

A SIP line is needed to establish the SIP connection between Avaya IP Office and Windstream SIP Trunking service. To create a SIP line, begin by navigating to **Line** in the left Navigation Pane, then right-click in the Group Pane and select **New → SIP Line**. On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

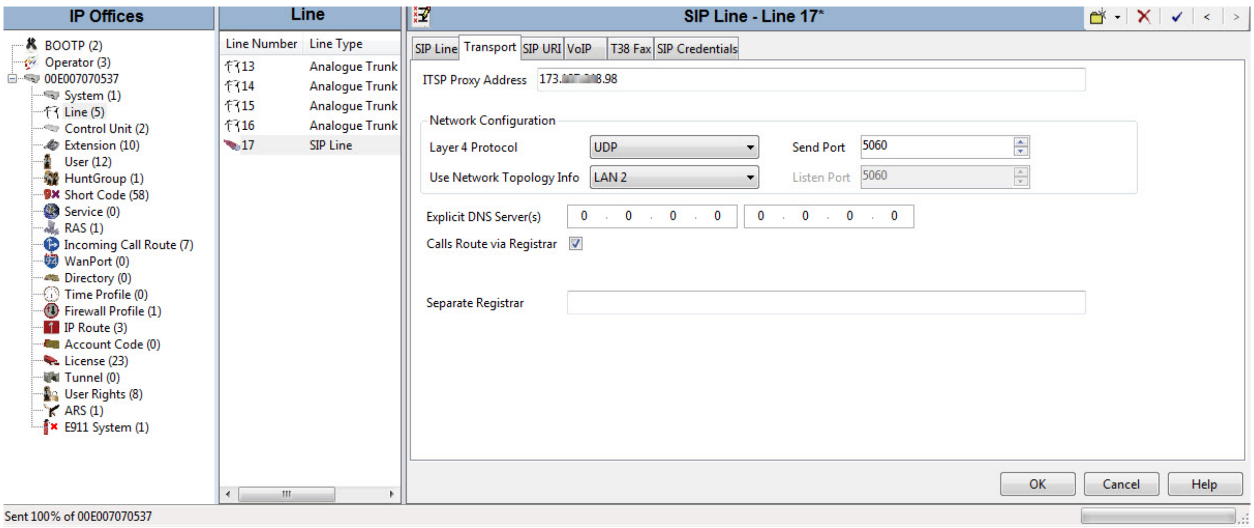
- Set **ITSP Domain Name** to the enterprise domain, or the IP address of the IP Office WAN interface, so that IP Office uses this domain / IP address as the host portion of SIP URI in SIP headers such as the From header.
- Set **Send Caller ID** to ***None***. For the compliance test, this parameter was ignored since **Send original calling party information for Mobile Twinning** is optioned in **Section 5.5**.
- Check the **In Service** box.
- Check the **Check OOS** box. With this option selected, Avaya IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- Default values may be used for all other parameters.

The area of the screen entitled **REFER Support** is used to enable/disable SIP REFER for call transfers. If **REFER Support** is left unchecked, or the fields for **Incoming** and **Outgoing** are set to ***Auto***, this will effectively disable the use of SIP REFER. To enable SIP REFER, check the **REFER Support** box and select ***Always*** from the drop-down menu for **Incoming** and **Outgoing**.

For the compliance test, **REFER Support** was disabled.

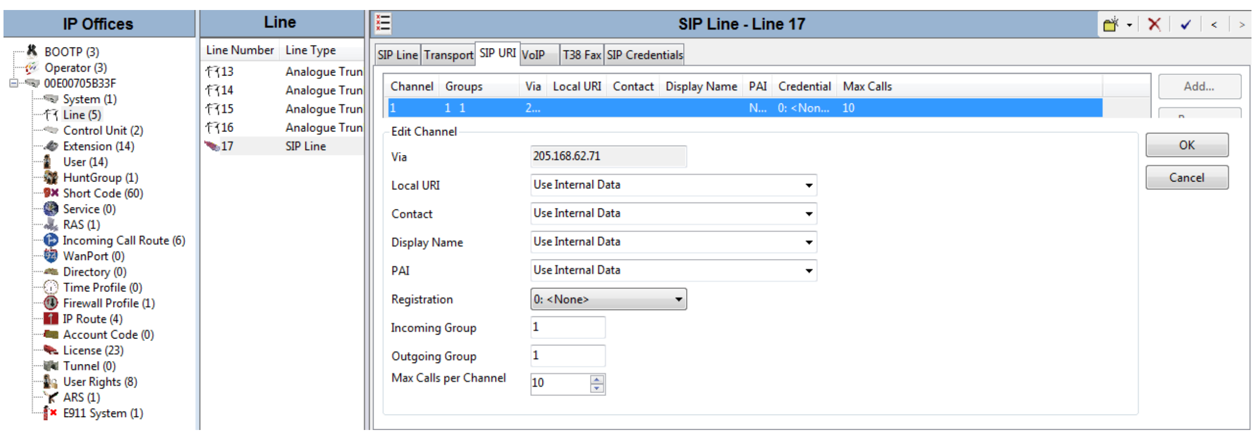


Select the **Transport** tab. The **ITSP Proxy Address** is set to the Windstream SIP Proxy IP Address provided by Windstream. As shown in **Figure 1**, this IP Address is **173.*XXX.XXX.98***. In the **Network Configuration** area, ***UDP*** is selected as the **Layer 4 Protocol**, and the **Send Port** is set to the port number provided by Windstream. The **Use Network Topology Info** parameter is set to ***LAN 2***. This associates the SIP Line with the parameters in the **System → LAN2 → Network Topology** tab. Other parameters retain default values in the screen below.



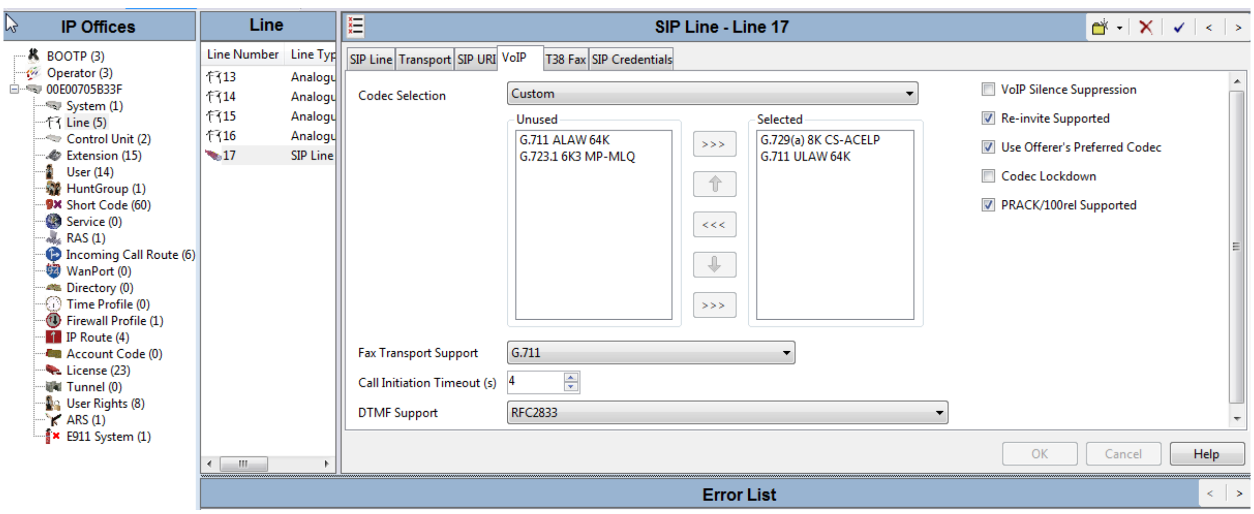
A SIP URI entry must be created to match each incoming number that Avaya IP Office will accept on this line. Select the **SIP URI** tab, and then click the **Add** button and the **New Channel** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit…** button. In the example screen below, a previously configured entry is edited. For the compliance test, a single SIP URI entry was created that matched any DID number assigned to an Avaya IP Office user. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact** and **Display Name** to ***Use Internal Data***. This setting allows calls on this line for SIP URIs that match the number set in the **SIP** tab of any **User** as shown in **Section 5.8**.
- Set **PAI** to ***Use Internal Data***. With this setting Avaya IP Office will populate the SIP P-Asserted-Identity header on outgoing calls with the data set in the **SIP** tab of the **User** initiating the call as shown in **Section 5.8**.
- For **Registration**, select ***None*** since Windstream doesn't require Registration. If Registration was required, then the account credentials that would have been configured on the line's **SIP Credentials** tab would be entered here.
- Associate this line with an incoming line group in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. For the compliance test, a new incoming and outgoing line group ***1*** was defined.
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.



Select the **VoIP** tab to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

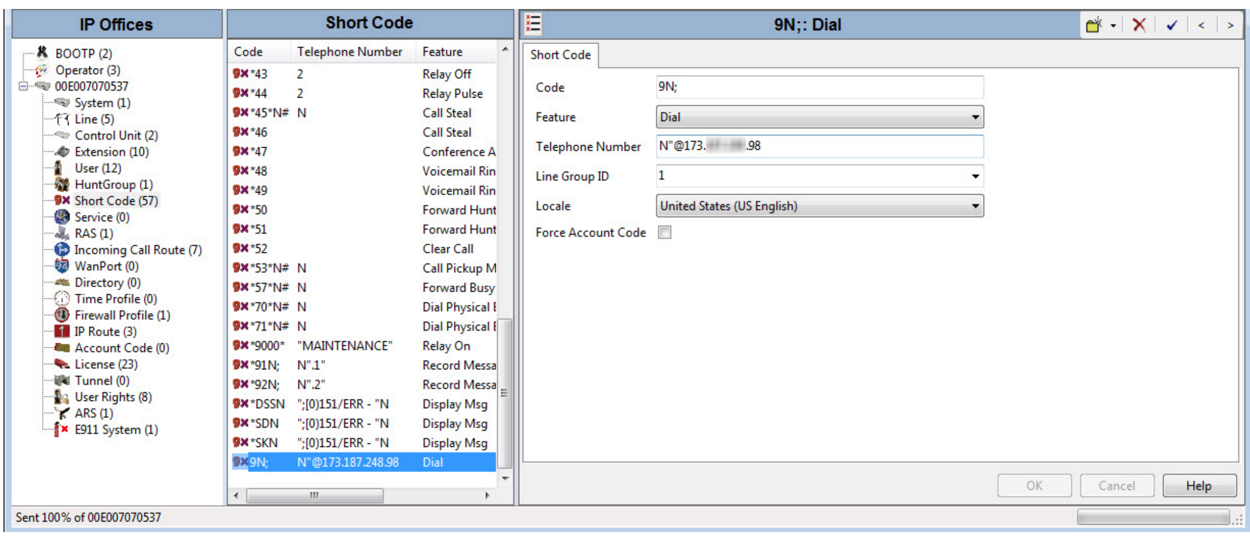
- The **Codec Selection Mode** was configured using the **Arrow** button, allowing an explicit ordered list of codecs to be specified. Highlight the Codes in the **Unused** box and move them over to the **Selected** box using the arrow button. Selecting ***G.711 ULAW and G.729 (a) 8K CS-ACELP codecs*** cause Avaya IP Office to include these codecs, supported by the Windstream SIP Trunking service, in the Session Description Protocol (SDP) offer, in that order.
- Set the **DTMF Support** field to ***RFC2833***. This directs Avaya IP Office to send DTMF tones using RTP event messages as defined in RFC2833.
- Uncheck the **VoIP Silence Suppression** box. By unchecking the **VoIP Silence Suppression** box, calls can be established with the G.729 codec but without silence suppression.
- Check the **Re-invite Supported** box.
- Check the **PRACK/100rel Supported** box.
- Since T.38 is not supported by Windstream, select ***G.711*** for **Fax Transport Support**.
- Default values may be used for all other parameters.



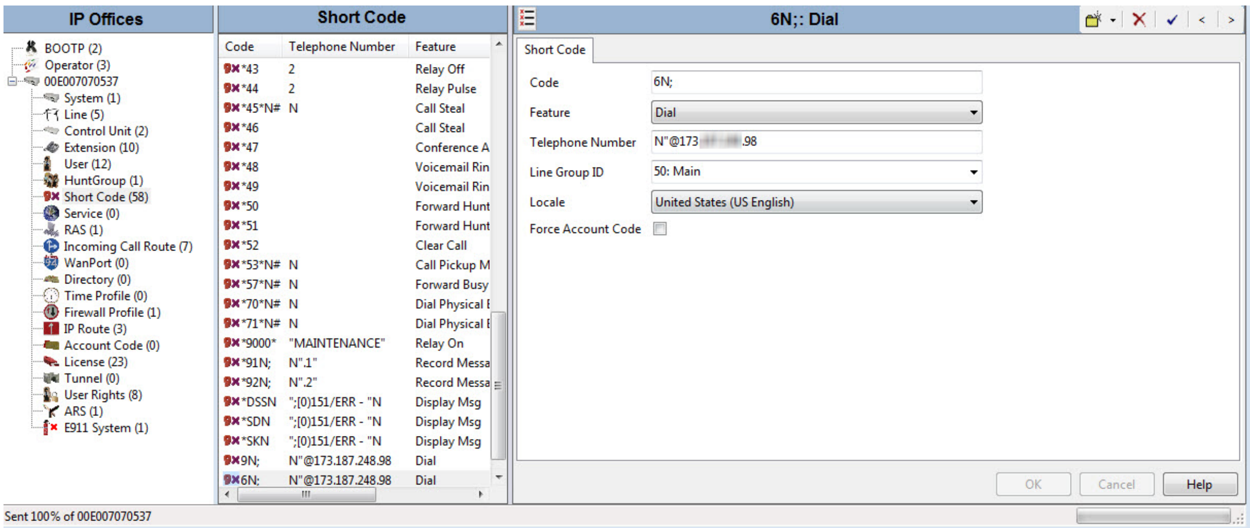
5.7. Short Code

Define a short code to route outbound traffic to the SIP line. To create a short code, select **Short Code** in the left Navigation Pane, then right-click in the Group Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters for the new short code to be created. The screen below shows the details of the previously administered "9N;" short code used in the test configuration.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, ***9N;*** This short code will be invoked when the user dials 9 followed by any number.
- Set **Feature** to ***Dial***. This is the action that the short code will perform.
- Set **Telephone Number** to ***N"@173.XXX.XXX.98"***. This 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 host part following the "@" is the domain or IP address of the Service Provider network.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.6**. This short code will use this line group when placing the outbound call.



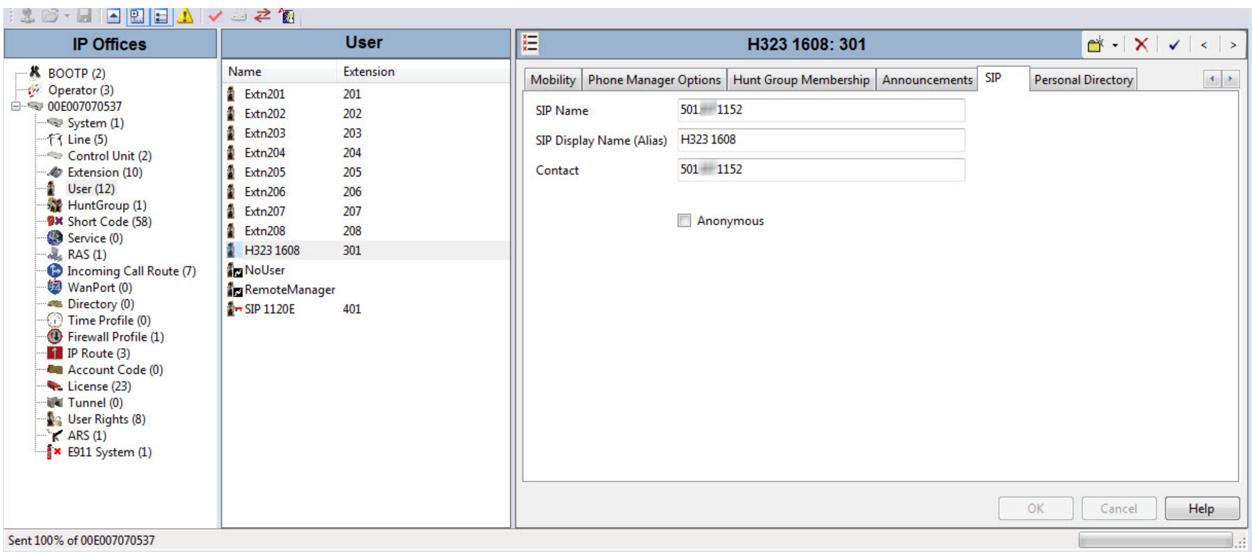
The simple "9N;" short code illustrated above does not provide a means of alternate routing if the configured SIP Line is out of service or temporarily not responding. When alternate routing options and/or more customized analysis of the digits following the short code are desired, the Automatic Route Selection (ARS) feature may be used. In the following example screen, the short code 6N is illustrated for access to ARS. When an Avaya IP Office user dials 6 plus any number N, rather than being directed to a specific **Line Group Id**, the call is directed to **Line Group ID** "50: Main", configurable via ARS. See **Section 5.10** for example ARS route configuration for "50: Main" as well as a backup route.



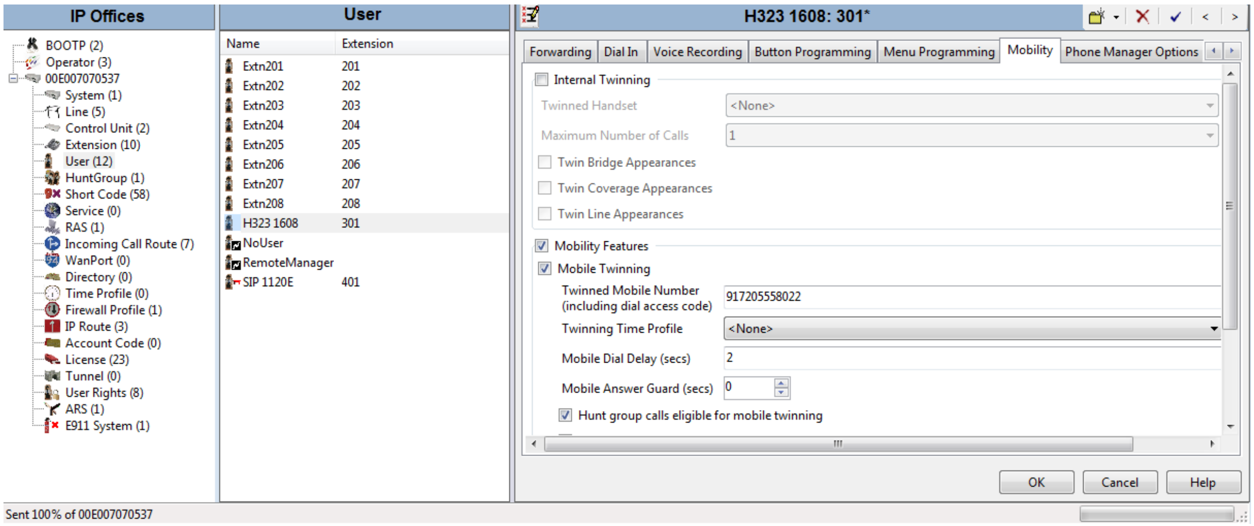
5.8. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.6**. To configure these settings, first select **User** in the left Navigation Pane, then select the name of the user to be modified in the center Group Pane. In the example below, the name of the user is “H323 1608”. Select the **SIP** tab in the Details Pane.

The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From and Contact headers for outgoing SIP trunk calls. They also allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.6**). The example below shows the settings for user “H323 1608”. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise from Windstream. 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.



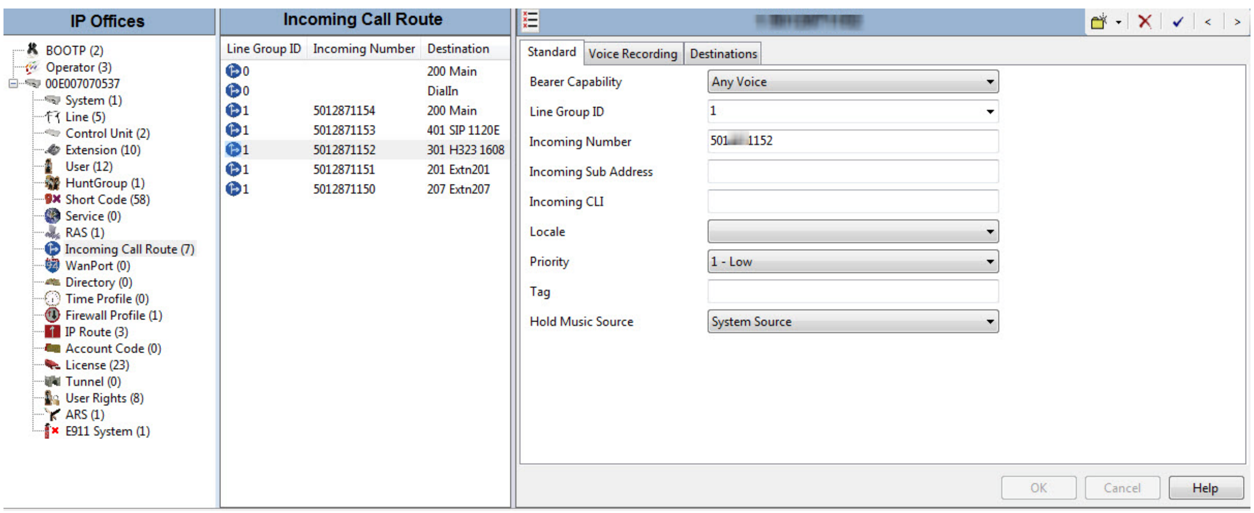
One of the H.323 IP Phones at the enterprise site uses the Mobile Twinning feature.   The following screen shows the **Mobility** tab for User "H323 1608". The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone. Other options can be set according to customer requirements.



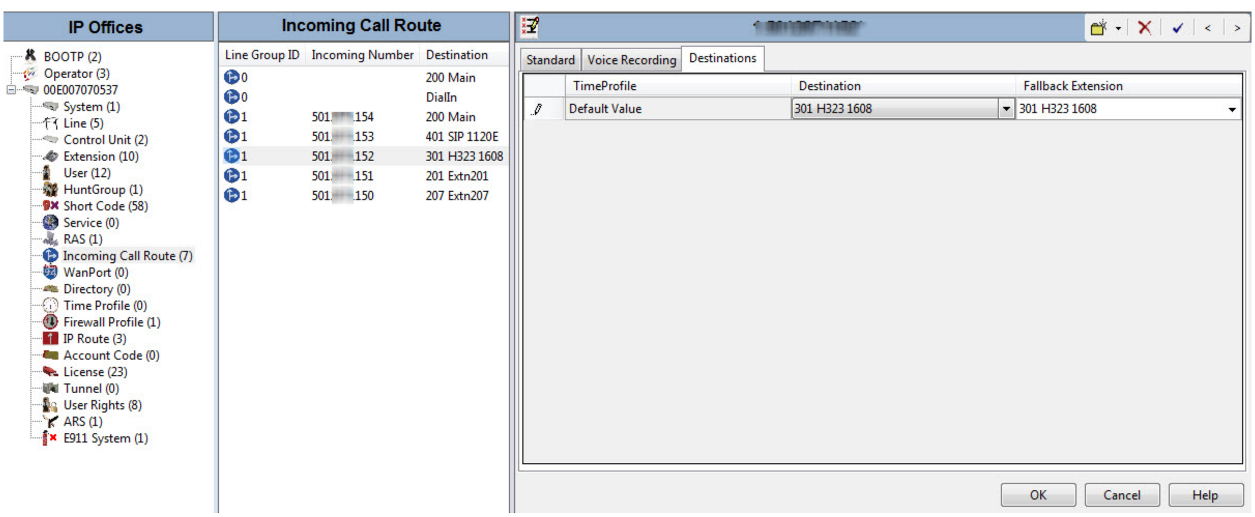
5.9. Incoming Call Route

An incoming call route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by the Service Provider. To create an incoming call route, select **Incoming Call Route** in the left Navigation Pane, then right-click in the center Group Pane and select **New**. On the **Standard** tab of the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capacity** to ***Any Voice***.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.6**.
- Set the **Incoming Number** to the incoming number on which this route should match.
- Default values can be used for all other fields.



On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to 501-XXX-X152 on line Group 1 are routed to the Avaya user "H.323 1608" at extension 301.



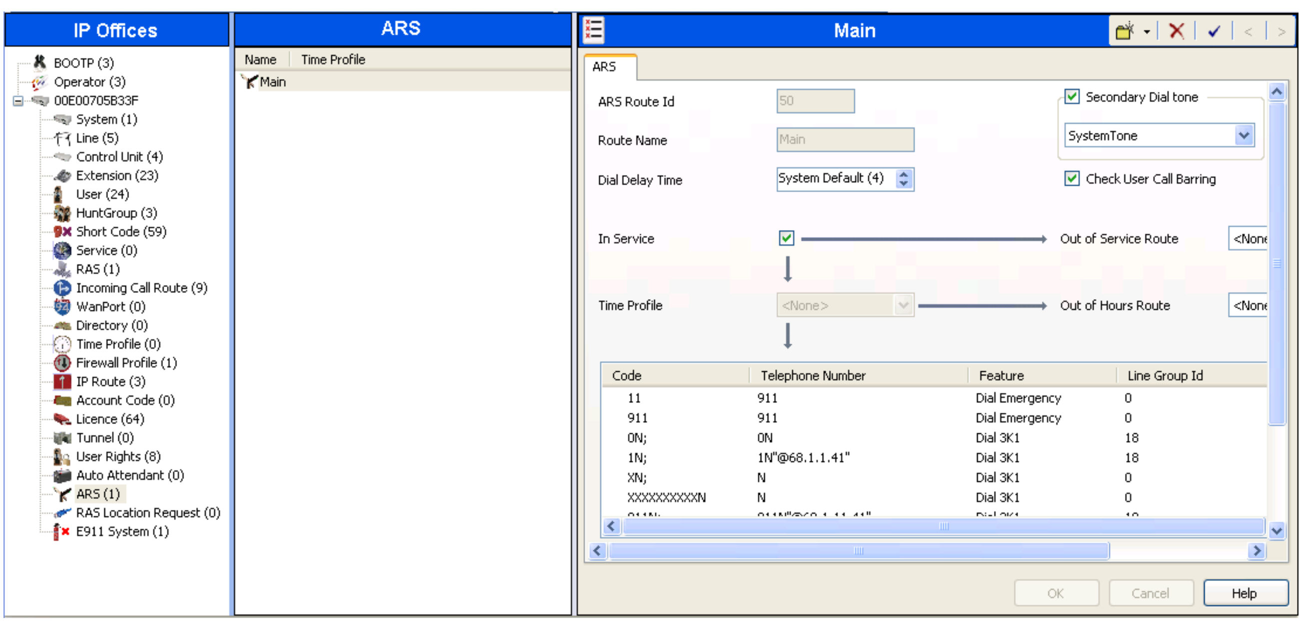
5.10. ARS and Alternate Routing

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

Optionally, ARS can be used rather than the simple "9N;" short code approach documented in **Section 5.7**. With ARS, a secondary dial tone can be provided after the access code, time-based routing criteria can be introduced, and alternate routing can be specified so that a call can re-route automatically if the primary route or outgoing line group is not available. Although not shown in this section, ARS also facilitates more specific dialed telephone number matching, enabling immediate routing and alternate treatment for different types of numbers following the access code. For example, if all 1+10 digit calls following an access code should use the SIP Line preferentially, but other local or service numbers following the access code should prefer a different outgoing line group, ARS can be used to distinguish between these 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.



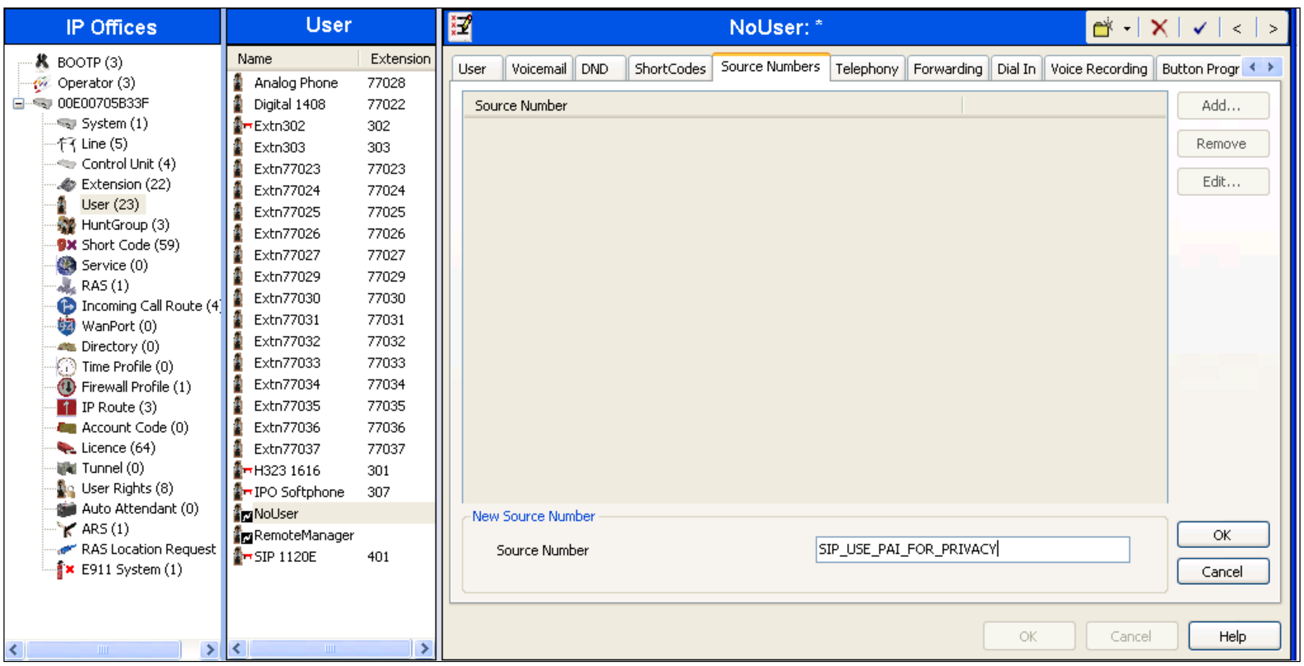
Assuming the primary route is in-service, the number passed from the short code used to access ARS (e.g., 6N in **Section 5.7**) can be further analyzed to direct the call to a specific Line Group ID. Per the example screen above, if the user dialed 6-1-303-555-1234, the call would be directed to Line Group 18. If Line Group 18 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 (not shown). Since alternate routing can be considered a privilege not available to all callers, Avaya IP Office can control access to the alternate route by comparing the calling user's priority, set in the User configuration, to the value in the **Alternate Route Priority Level** field (not shown).

5.10. Privacy/Anonymous Calls

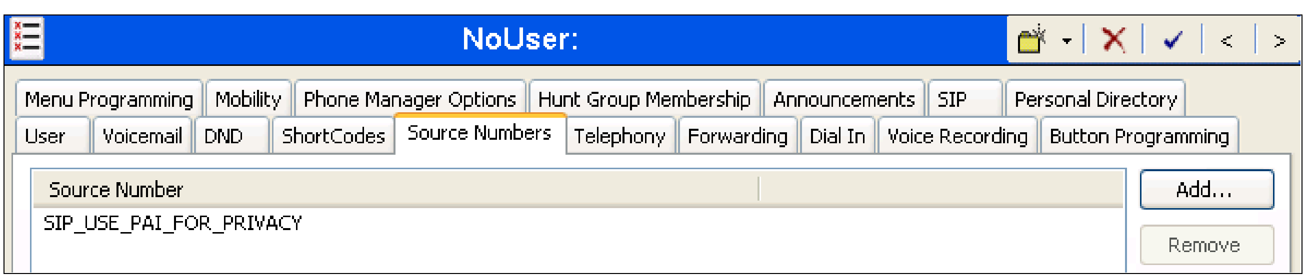
For outbound calls with privacy (anonymous) enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with "restricted" and "anonymous" respectively. Avaya IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing. For the compliance test, PAI was used for the purposes of privacy.

To configure Avaya IP Office to use PAI for privacy calls, navigate to **User → NoUser** in the Navigation / Group Panes. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.

At the bottom of the Details Pane, the **Source Number** field will appear. Enter ***SIP_USE_PAI_FOR_PRIVACY***. Click **OK**.



The **SIP_USE_PAI_FOR_PRIVACY** parameter will appear in the list of Source Numbers as shown below.

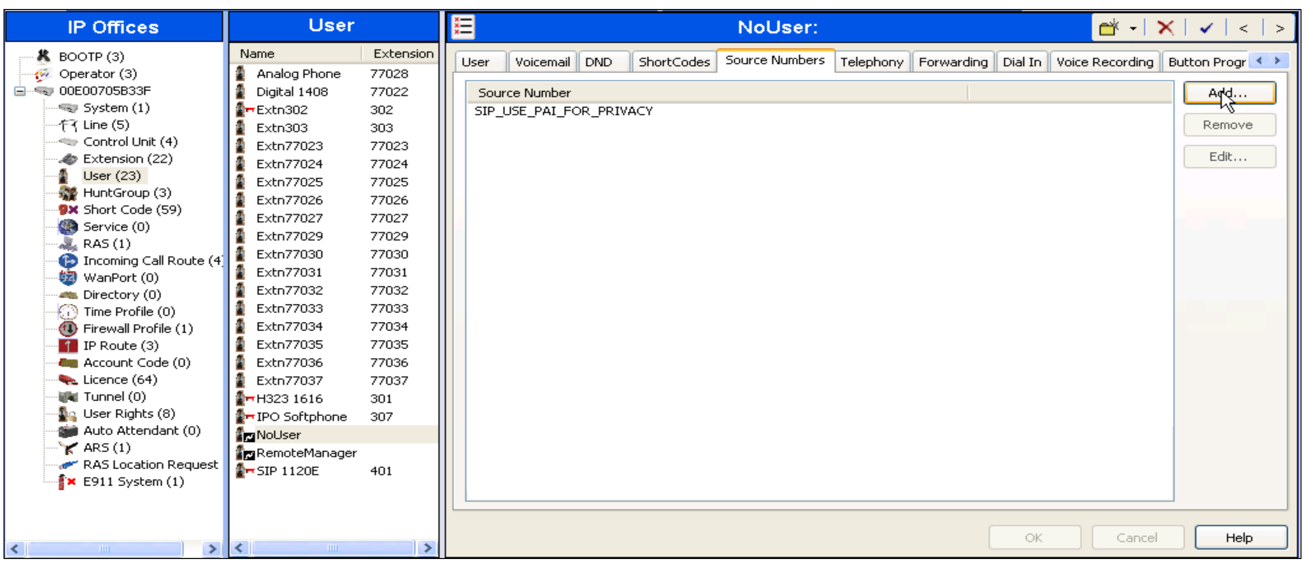


5.11. SIP Options

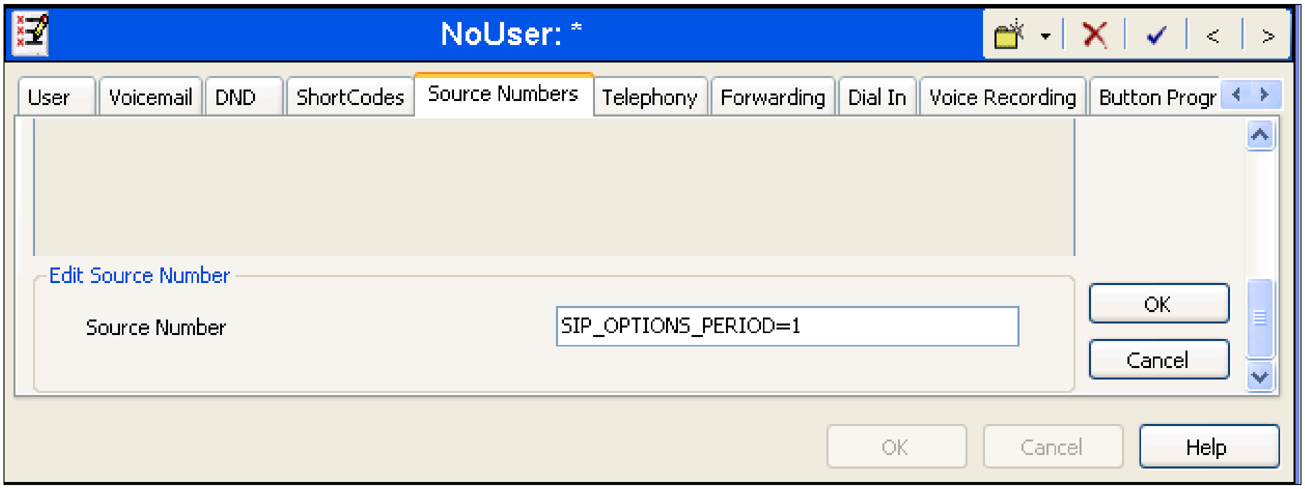
Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. By default, the Avaya IP Office Release 8.0 sends out OPTIONS every 300 seconds (changed from previous releases). The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.2** and the **SIP_OPTIONS_PERIOD** parameter (in minutes) that can be set on the **Source Number** tab of the **NoUser** user. The OPTIONS period is determined in the following manner:

- To use the default value, set Binding Refresh = 0 or 300. OPTIONS will be sent at the 300 second frequency.
- To establish a period of less than 300 seconds, do not define the **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 300 seconds. The OPTIONS message period will be equal to the **Binding Refresh Time setting**.
- To establish a period greater than 300 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 300 seconds. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD**.

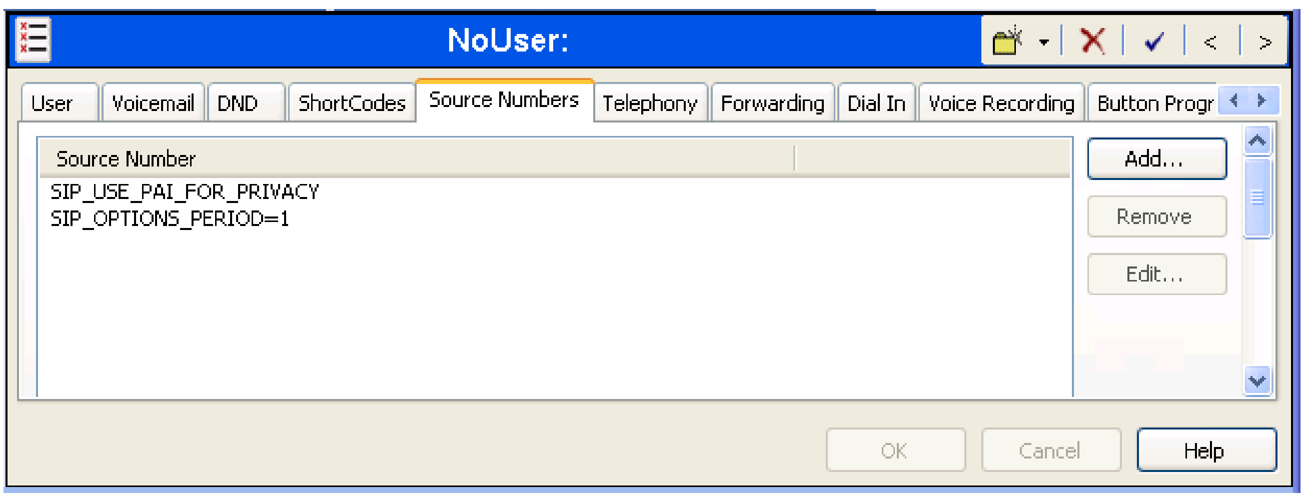
To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User → noUser** in the Navigation / Group Panes. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



At the bottom of the Details Pane, the **Source Number** field will appear. Enter **SIP_OPTIONS_PERIOD=*X***, where ***X*** is the desired value in minutes. Click **OK**.



The **SIP_OPTIONS_PERIOD** parameter will appear in the list of Source Numbers as shown below. For the compliance test, the **Binding Refresh Time** was set to ***0*** in **Section 5.2**, therefore Avaya IP Office sends out OPTIONS at the frequency of 300 seconds, or every 5 minutes. The **SIP_OPTIONS_PERIOD** parameter does not need to be defined in this case. It is documented here for reference and completeness.



5.12. 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.

6. Windstream SIP Trunking Configuration

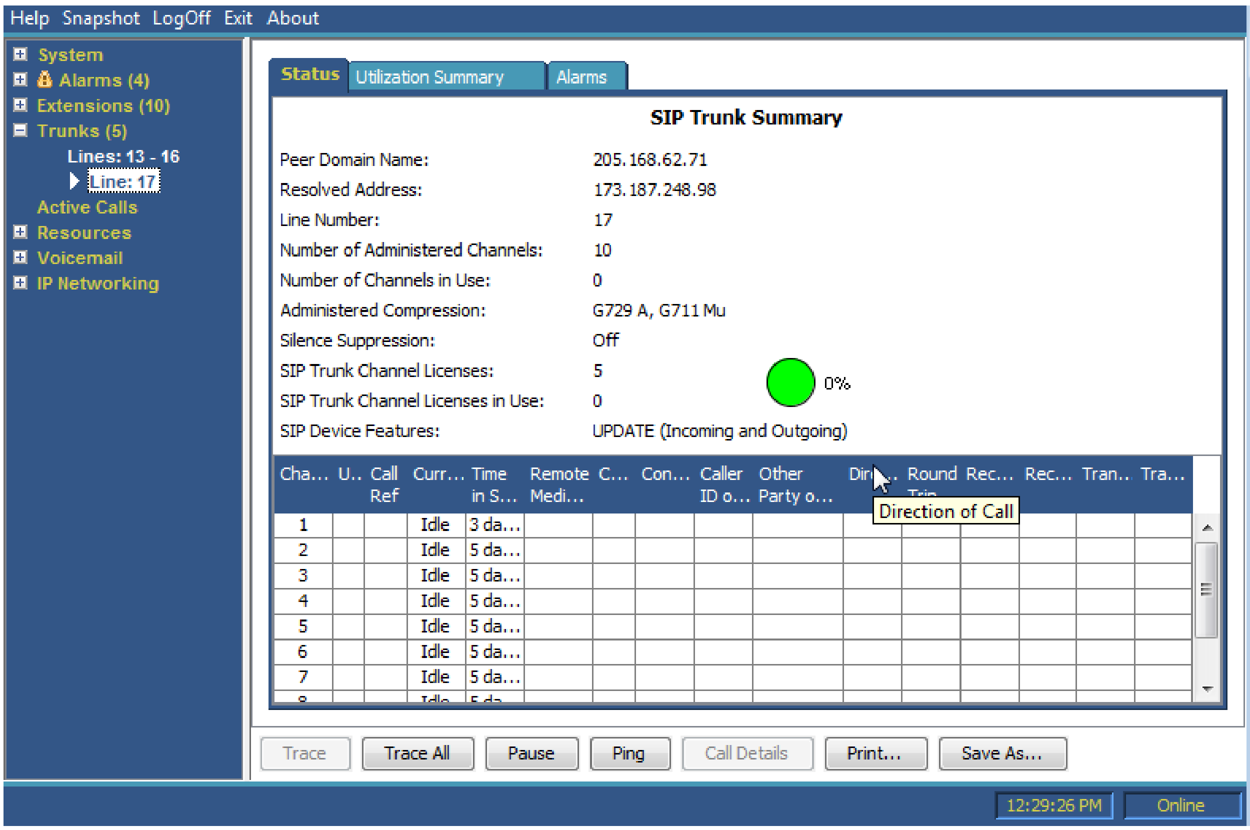
Windstream is responsible for the configuration of Windstream SIP Trunking service. The customer will need to provide the IP address and /or FQDN used to reach the Avaya IP Office at the enterprise. Windstream will provide the customer the necessary information to configure the Avaya IP Office SIP connection to Windstream. The provided information from Windstream includes:

- IP address of the Windstream SIP proxy
- Windstream SIP domain
- Supported codecs
- DID numbers
- IP addresses, port numbers and transport protocol used for signaling or media through any security devices

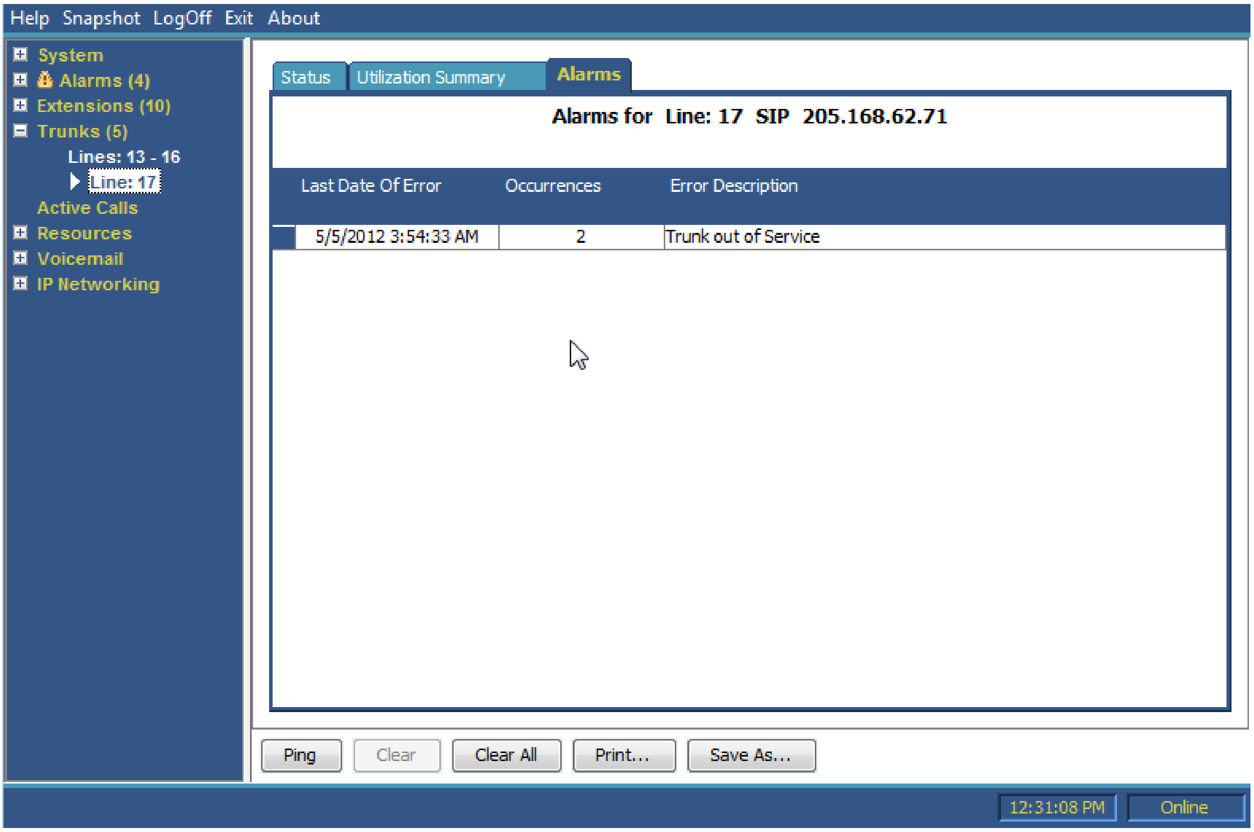
7. Verification Steps

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 State** is ***Idle*** for each channel (assuming no active calls at present time).



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



- Verify that a phone connected to the PSTN can successfully place a call to Avaya IP Office with two-way audio.
- Verify that a phone connected to Avaya IP Office can successfully place a call to the PSTN with two-way audio.
- Using a network sniffing tool (e.g., Wireshark), monitor the SIP signaling messages between Windstream and Avaya IP Office.

8. Conclusion

Windstream SIP Trunking passed compliance testing. These Application Notes describe the procedures required to configure the SIP trunk connection between Avaya IP Office and the Windstream SIP Trunking service as shown in **Figure 1**. Please refer to **Section 2.2** above for tTest results and any limitations that were observed.

9. Additional References

This section references the documentation relevant to these Application Notes.

[1] *IP Office 8.0 IP Office Standard Version Installation,* Document number15-601042, Issue 25b, March 2012.
[2] *IP Office Release 8.0 Manager 10.0,* Document number15-601011, Issue 28k, April 2012.
[3] *IP Office Release 8.0 Implementing Voicemail Pro*, Document Number 15-601064, Issue 02b, April 2012.
[4] *IP Office Release 8.0 Administering Voicemail Pro*, Document Number 15-601063, Issue 27b, April 2012.
[5] *IP Office System Status Application,* Document number15-601758, Issue 06b, November 2011.
[6] *IP Office System Monitor*, Document Number 15-601019, Issue 02b, November 28, 2008.

Product documentation for Avaya products may be found at http://support.avaya.com. Additional IP Office documentation can be found at:
http://marketingtools.avaya.com/knowledgebase/

Product documentation for Windstream SIP Trunking is available from Windstream.

Appendix: SIP Line Template

Avaya IP Office Release 8.0 supports SIP Line Template (in xml format) that can be created from an existing configuration and imported into a new installation to simplify configuration procedures as well as to reduce potential configuration errors.

Note that not all of the configuration information included in the SIP Line Template below, particularly items relevant to specific installation (e.g., the IP address assigned to the WAN interface of Avaya IP Office, the SIP Proxy IP address of the service provider, etc.), is proper for customer environment. Therefore it is critical that the SIP Line configuration be verified/updated after a template has been imported, and additional configuration be supplemented using **Section 5.6** in these Application Notes as a reference.

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

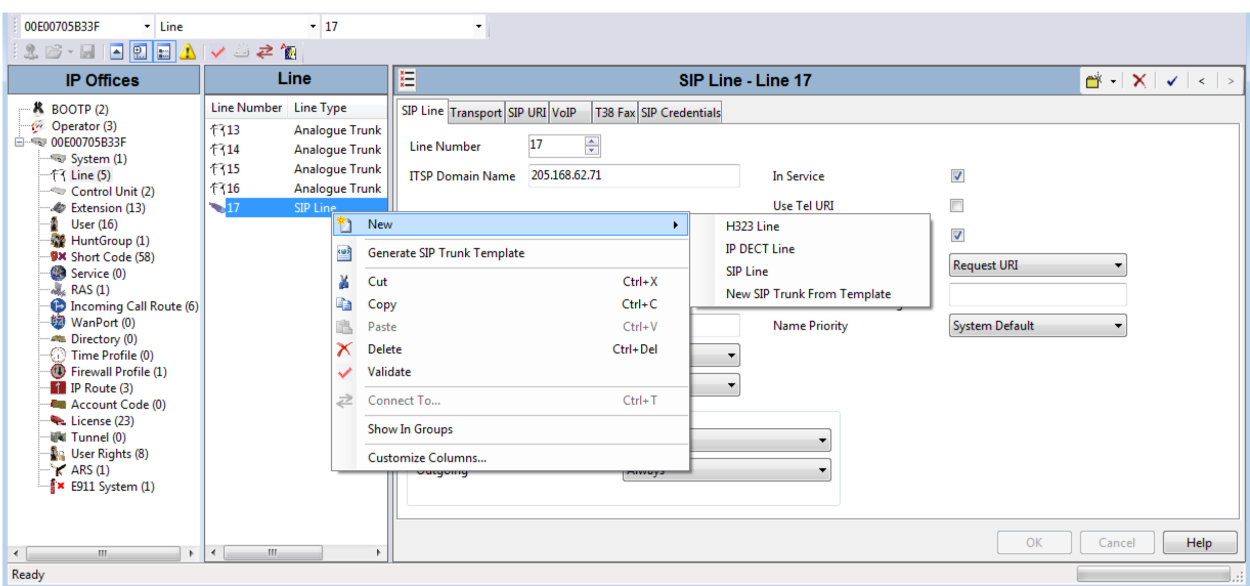

```
<?xml version="1.0" encoding="utf-8" ?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20120511</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>Winstream</DescriptiveName>
  <ITSPDomainName>205.XXX.XX.71</ITSPDomainName>
  <SendCallerID>CallerIDNone</SendCallerID>
  <ReferSupport>true</ReferSupport>
  <ReferSupportIncoming>0</ReferSupportIncoming>
  <ReferSupportOutgoing>0</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>true</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>SystemDefault</LineNamePriority>
  <ITSPProxy>173.XXX.XXX.98</ITSPProxy>
  <LayerFourProtocol>SipUDP</LayerFourProtocol>
  <SendPort>5060</SendPort>
  <ListenPort>5060</ListenPort>
  <DNSServerOne>0.0.0.0</DNSServerOne>
  <DNSServerTwo>0.0.0.0</DNSServerTwo>
  <CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>
  <SeparateRegistrar />
  <CompressionMode>AUTOSELECT</CompressionMode>
  <UseAdvVoiceCodecPrefs>true</UseAdvVoiceCodecPrefs>
  <AdvCodecPref>G.729(a) 8K CS-ACELP,G.711 ULAW 64K</AdvCodecPref>
  <CallInitiationTimeout>4</CallInitiationTimeout>
  <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
  <VoipSilenceSupression>false</VoipSilenceSupression>
```

```
    <ReinviteSupported>true</ReinviteSupported>
    <FaxTransportSupport>FOIP_G711</FaxTransportSupport>
    <UseOffererPrefferedCodec>true</UseOffererPrefferedCodec>
    <CodecLockdown>false</CodecLockdown>
    <Rel100Supported>true</Rel100Supported>
    <T38FaxVersion>3</T38FaxVersion>
    <Transport>UDPTL</Transport>
    <LowSpeed>0</LowSpeed>
    <HighSpeed>0</HighSpeed>
    <TCFMethod>Trans_TCF</TCFMethod>
    <MaxBitRate>FaxRate_14400</MaxBitRate>
    <EflagStartTimer>2600</EflagStartTimer>
    <EflagStopTimer>2300</EflagStopTimer>
    <UseDefaultValues>true</UseDefaultValues>
    <ScanLineFixup>true</ScanLineFixup>
    <TFOPEnhancement>true</TFOPEnhancement>
    <DisableT30ECM>false</DisableT30ECM>
    <DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
    <DisableT30MRCompression>false</DisableT30MRCompression>
    <NSFOverride>false</NSFOverride>
</Template>
```

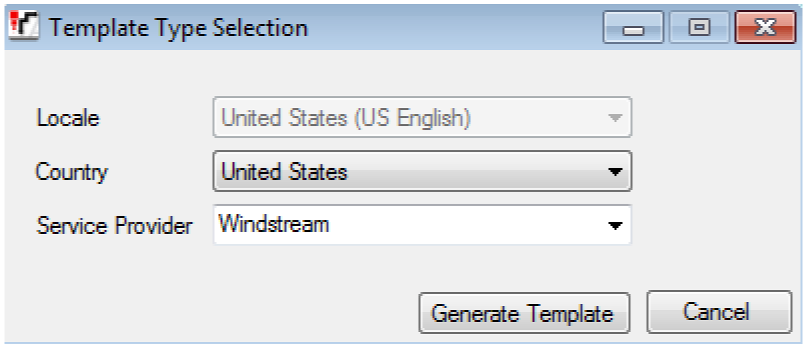
To import the above template into a new installation:

1. Copy and paste the above template into a text document named **`US_Windstream _SIPTrunk.xml`** on the PC where Avaya IP Office Manager was installed. Move the .xml file to the Avaya IP Office Manager template directory (C:\Program Files\Avaya\IP Office\Manager\Templates).

2. Import the template into an IP Office installation by creating a new SIP Line as shown in the screenshot below (right clicking **Line** in the left navigation pane):



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



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