Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Windstream SIP Trunking (Metaswitch Platform) with Avaya IP Office Release 8.1 – Issue 1.0

Abstract

These Application Notes describes the steps to configure Session Initiation Protocol (SIP) Trunking between Windstream Metaswitch and Avaya IP Office Release 8.1.

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 cost for the enterprise.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted in the Avaya Solutions and Interoperability Test Lab, utilizing Windstream SIP Trunk Services.

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between Windstream Metaswitch and Avaya IP Office Release 8.1.

The Windstream SIP Trunking service referenced within these Application Notes is positioned for customers that have an IP-PBX or IP-based network equipment with SIP functionality, but need a form of IP transport and local services to complete their solution.

Windstream SIP Trunking will enable delivery of origination and termination of local, long-distance and toll-free traffic across a single broadband connection. A SIP signaling interface will be enabled to the Customer Premises Equipment (CPE).

2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the Windstream SIP Trunking service 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:

- Response to SIP OPTIONS queries
- Incoming PSTN calls to various phone types including H.323, SIP, 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 to various phone types including H.323, SIP, digital, and analog telephones at the enterprise. All outgoing PSTN calls were routed to the enterprise across the SIP trunk from the service provider
- Inbound and outbound PSTN calls to/from Avaya IP Office Softphone
- Various call types including: local, long distance, outbound toll-free, and local directory assistance
- Codec G.711MU
- G.711 Fax

- Caller ID presentation and Caller ID restriction
- DTMF transmission using RFC 2833
- Voicemail navigation using DTMF for inbound and outbound calls
- User features such as hold and resume, transfer, and conference
- Off-net call forwarding and twinning

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.

- **T.38 Fax** – The use of T.38 Fax did not pass compliance testing. Windstream returns a "488 Not Acceptable Here" response to the SIP INVITE from Avaya IP Office with T.38 parameters. Thus, the use of T.38 Fax is not recommended with this solution. Fax calls were successful during testing using the fax transport method of "G.711" on the SIP Line. See **Section 5.7.4**. This transport method is intended to improve fax success rates by making fax-aware provisions for calls known to be fax calls, such as disabling the digital signal processing appropriate for voice calls.
- **Call Transfer** – When an H.323 enterprise extension blind transferred a call with a PSTN phone (either inbound or outbound) off-net back to PSTN, Windstream responded to REFER from the enterprise with "403 Refer in bad call state" instead of "202 Accepted". User experience was not negatively affected (i.e., the call was transferred successfully). Consultative transfer of similar call to PSTN worked properly (Windstream responded with "202 Accepted" to REFER from the enterprise).
- **SIP Blind Transfer –** When a SIP enterprise extension transferred a call with a PSTN phone (either inbound or outbound) off-net back to PSTN, IP Office does not send a REFER message to Windstream. User experience was not negatively affected (i.e., the call was transferred successfully). This observation is under investigation by IP Office product development (IPOFFICE- 31274).
- **One-X® Portal for IP Office –** When an outbound call to a PSTN phone is blind transferred to another PSTN phone using the One-X Portal client, the FROM header in the INVITE contains the wrong caller ID and Windstream responds with "403 From: URI not recognized" causing the transfer to fail. A recommended workaround is to perform a consultative transfer. This observation is under investigation by IP Office product development (IPOFFICE- 31275).

2.3. Support

For technical support on Windstream SIP Trunking, contact Windstream using the Customer Service links at www.windstream.com.

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. 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 Windstream SIP Trunking.

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 9630 IP Telephone (with H.323 firmware), an Avaya 9621 IP Telephone (with H.323 firmware), an Avaya IP Office Phone Manager, an Avaya IP Office Softphone, an Avaya 9508 Digital Telephone, an Avaya T7316E and an Avaya 6210 Analog Telephone. The site also has a Windows 2003 Server running Avaya Voicemail Pro for voicemail, one-X® Portal for Windows and running Avaya IP Office Manager to configure the Avaya IP Office.

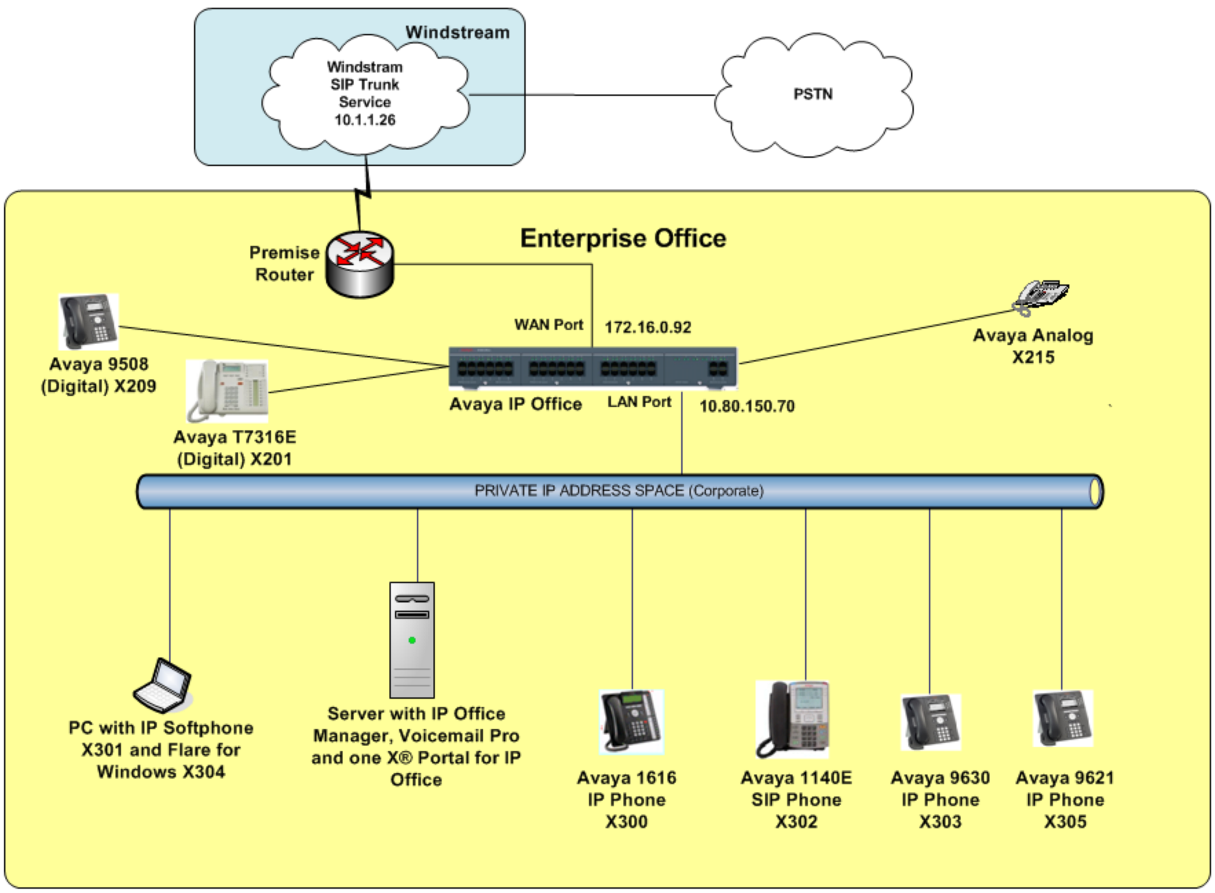


Figure 1: Avaya Interoperability Test Lab Configuration

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.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and the 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 the Avaya IP Office must be allowed to pass through these devices.

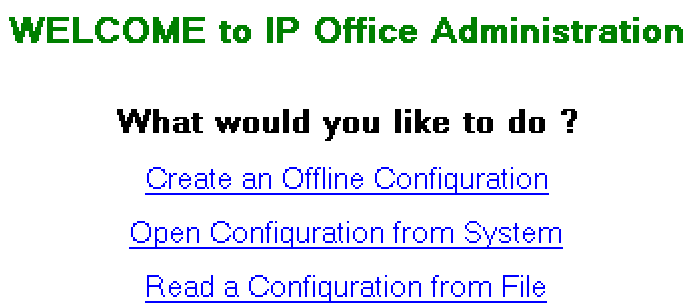
4. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Equipment	**Software**
Avaya IP Office 500 V2	Release 8.1 (43)
Avaya Voicemail Pro	Release 8.1 (810)
Avaya IP Office Manager	Release 10.1 (43)
Avaya 1616SW IP Telephone (H.323)	Release 1.301S
Avaya 9630SW IP Telephone (H.323)	Release 3.104S
Avaya 9621SW IP Telephone (H.323)	Release 6.2119
Avaya 1140E IP Telephone (SIP)	Release 04.03.12
Avaya 9508 Digital Telephone	Release 0.39
Avaya IP Office Softphone	Release 3.2.3.20
Avaya Flare Communicator for Windows	Release 1.0.0
IP Office one-X® Portal	Release 8.1.76
Windstream SIP Trunking Solution Components	
Component	**Release**
Metaswitch	7.03.00 SU 56

5. Configure Avaya IP Office

Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the Avaya IP Office Manager PC, select **Start → Programs → IP Office → Manager** to launch the application. A screen that includes the following in the center may be displayed:

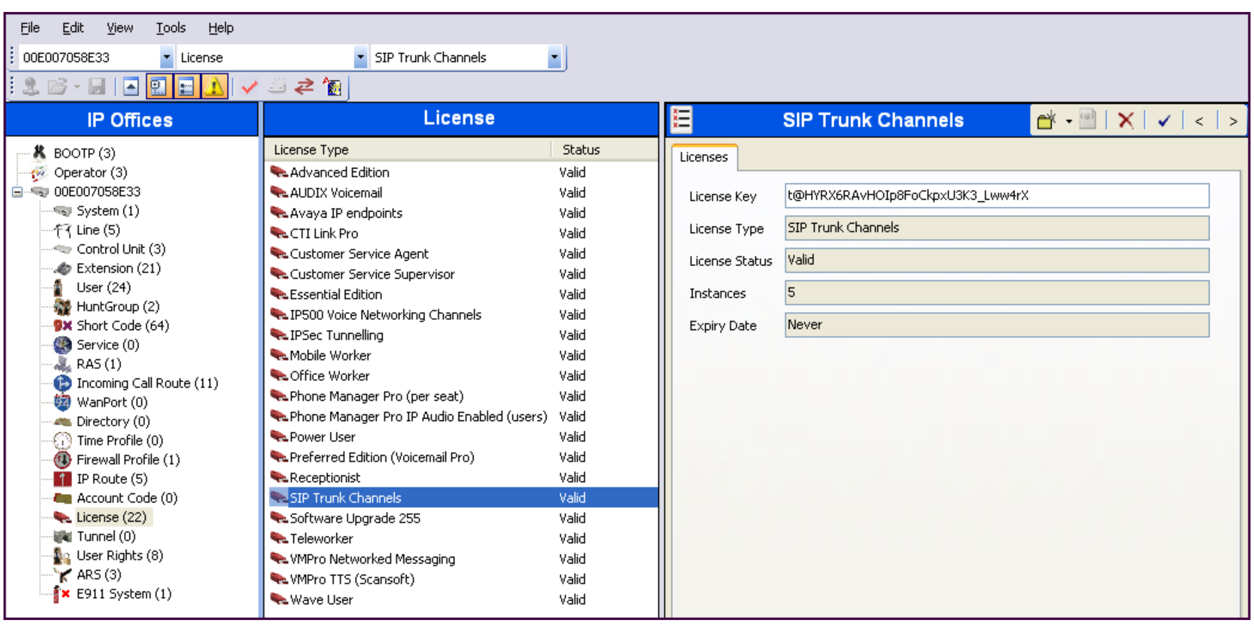


Navigate to **File → Open Configuration**, select the proper Avaya IP Office system from the pop-up window and log in with the appropriate credentials. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details pane on the right side. These panes will be referenced throughout the Avaya IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the service provider (such as twinning and IP Office Softphone support) is assumed to already be in place.

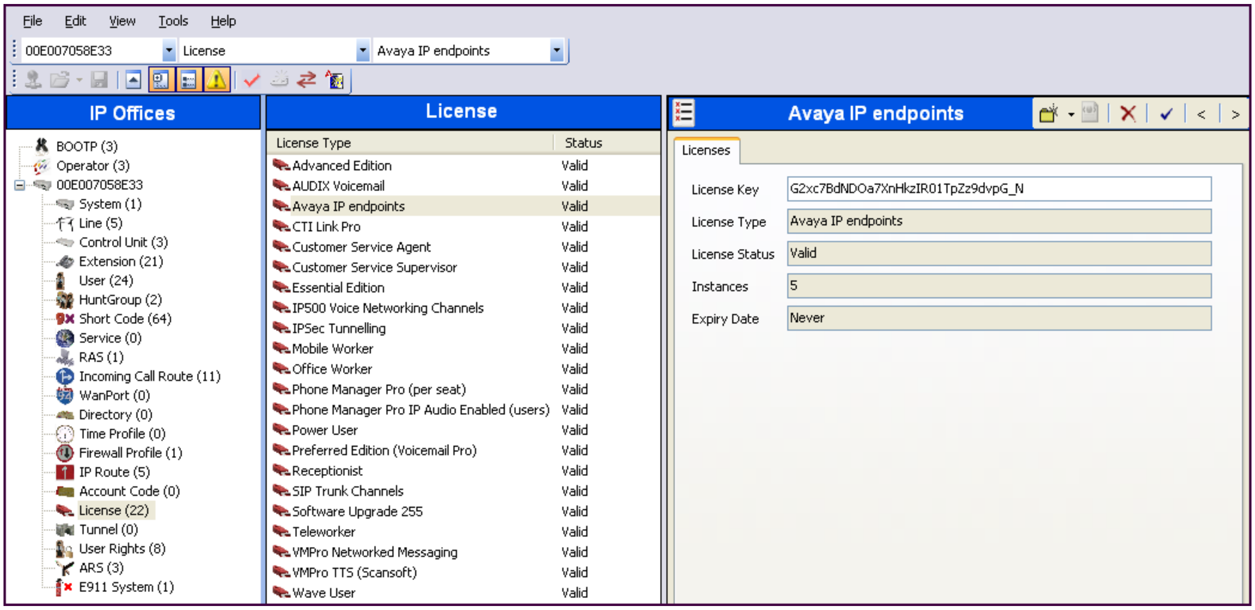
5.1. Licensing and Physical Hardware

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 and **SIP Trunk Channels** in the Group pane. Confirm a valid license with sufficient **Instances** (trunk channels) in the Details pane.

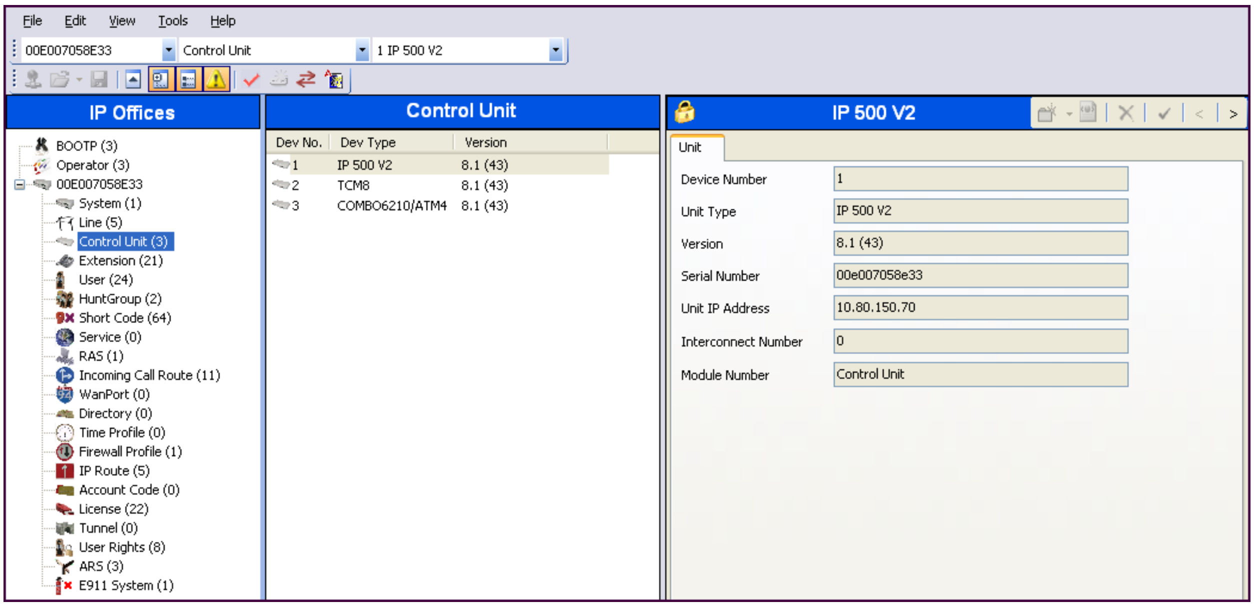


If Avaya IP Telephones will be used as is the case in these Application Notes, verify the Avaya IP endpoints license. Click **License** in the Navigation pane and **Avaya IP endpoints** in the Group pane. Confirm a valid license with sufficient **Instances** in the Details pane.



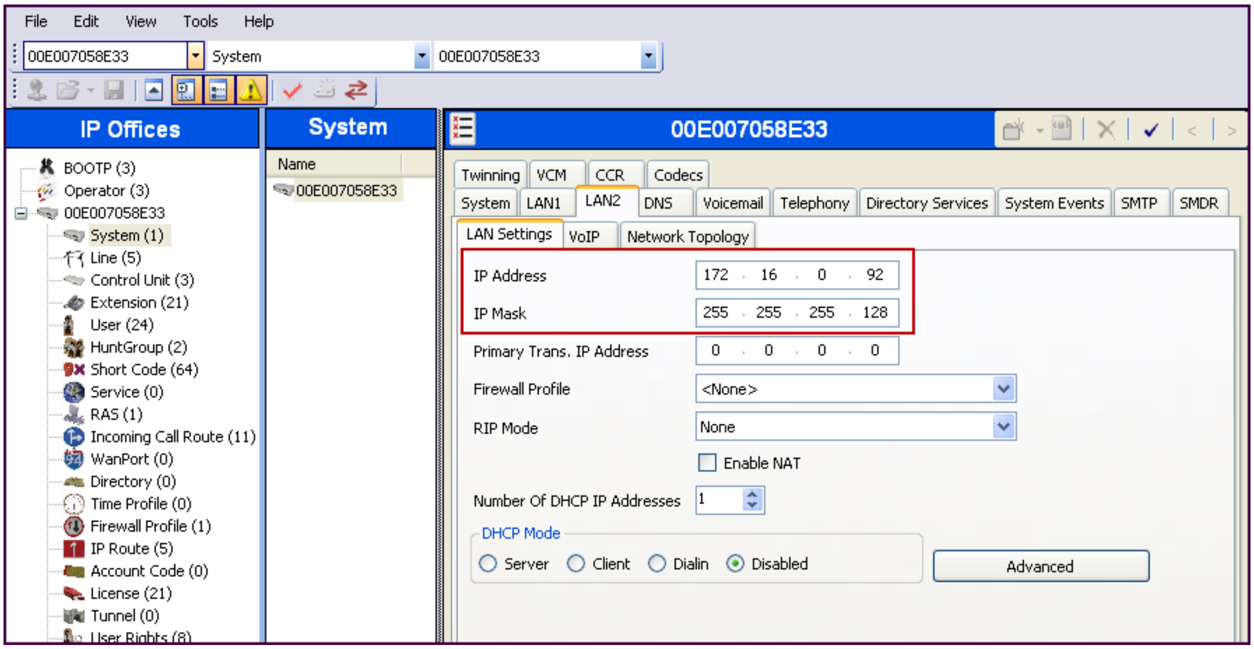
In the sample configuration, looking at the IP Office 500 from left to right, the first module is a TCM 8. This module supports BCM / Norstar T-Series and M-Series telephones. The second module is a Combination Card. This module has 6 Digital Stations ports, 2 Analog Station ports, 4 Analog Trunk ports 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.

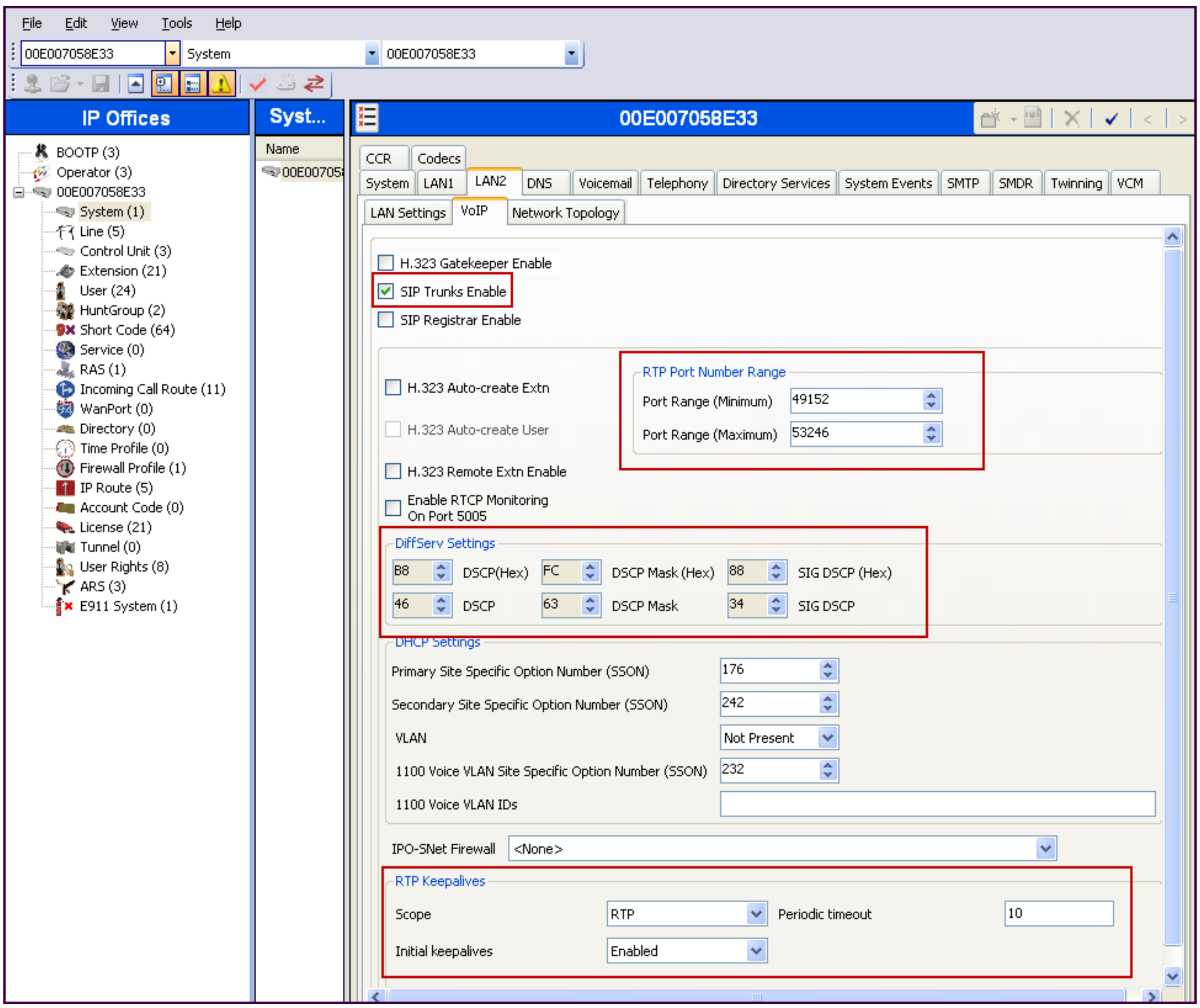


5.2. LAN2 Settings

In the sample configuration the WAN port 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 500. To access the LAN2 settings, first navigate to **System** in the Navigation Pane 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.

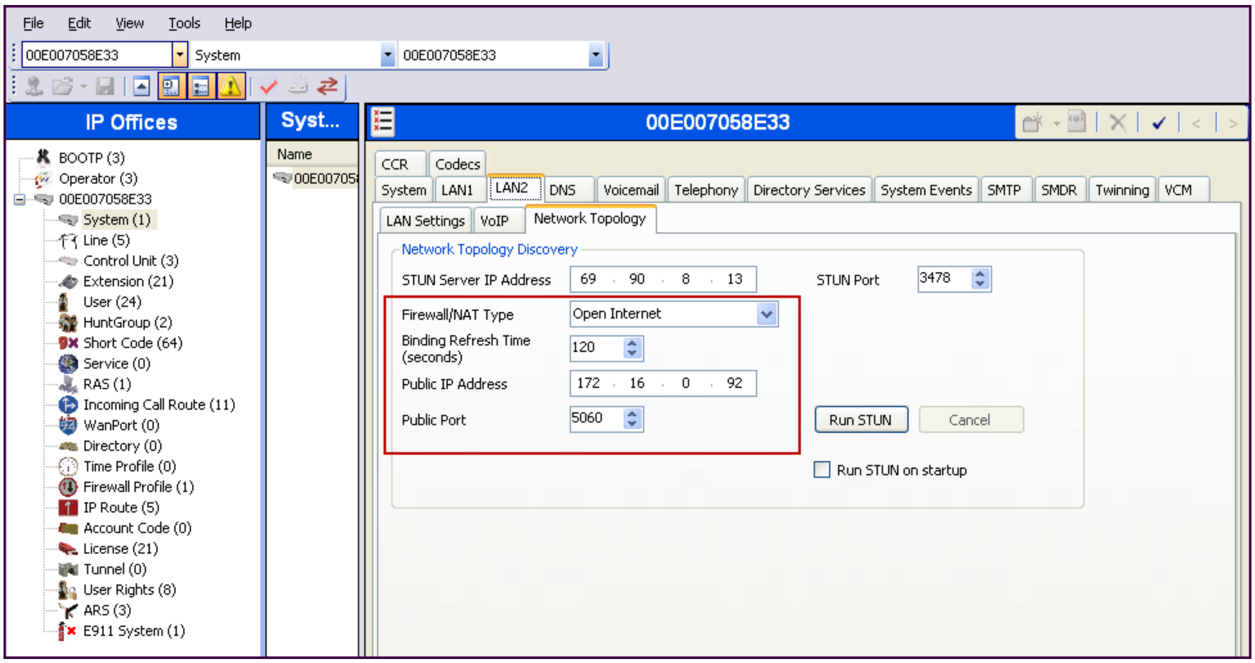


On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. Under **RTP Keepalives** set the **Scope** to ***RTP***, the **Initial keepalives** to ***Enabled*** and the **Periodic timeout** to ***10***. Enabling this will prevent the loss of speech path on calls forwarded across the SIP trunk. These settings instruct Avaya IP Office to sent RTP keepalive packets every 10 seconds from the establishment of the connection. This will start media flowing from the far-end endpoint in those cases where the far-end endpoint is waiting to receive media before it starts to send media of its own. 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. The specific values used for the compliance test are shown in the example below and are also the default values. All other parameters should be set according to customer requirements.



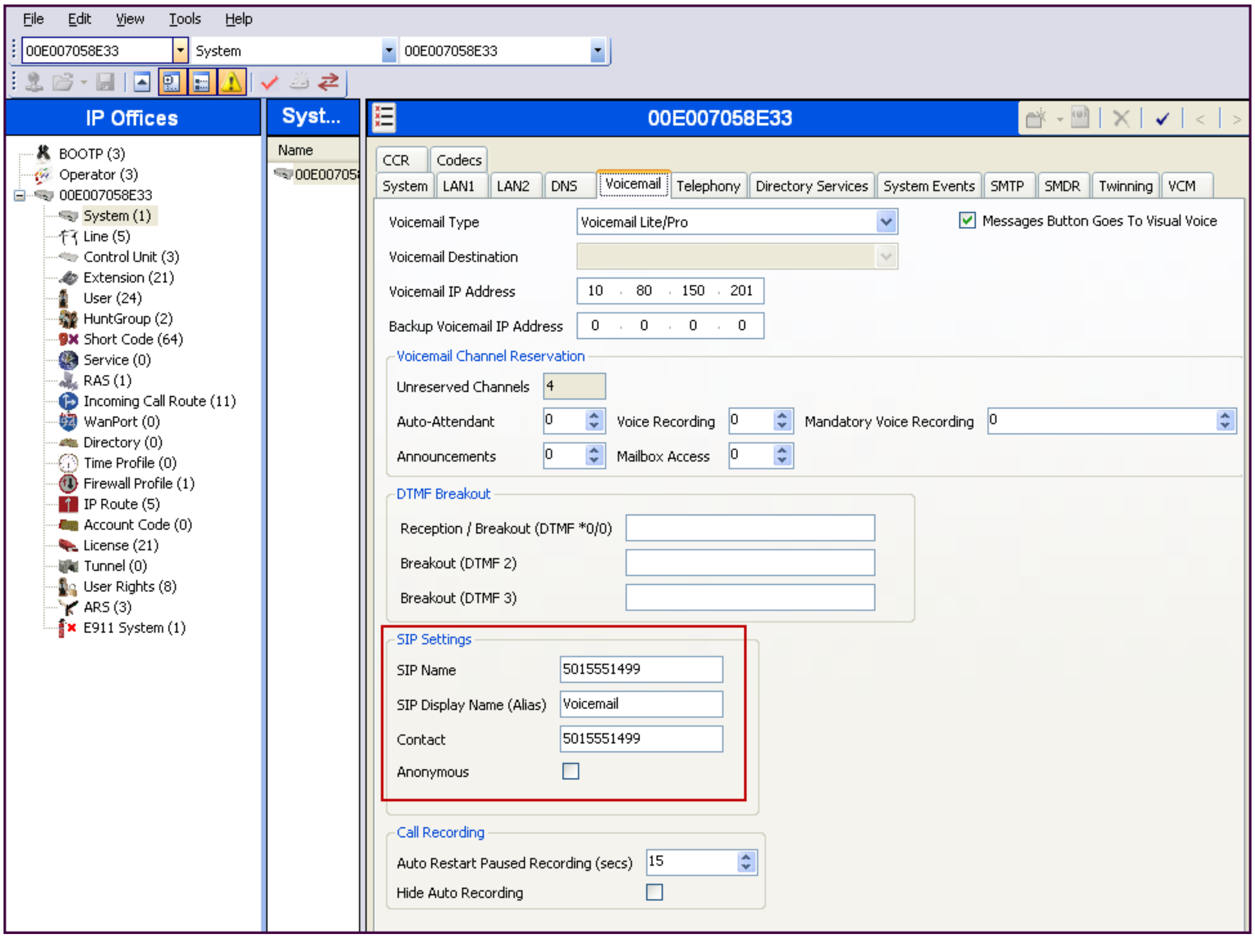
On the **Network Topology** tab in the Details Pane, configure the following parameters:

- 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***.
- Set **Binding Refresh Time (seconds)** to ***120***. This value is used to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port.
- Set the **Public Port** to ***5060***.
- All other parameters should be set according to customer requirements.



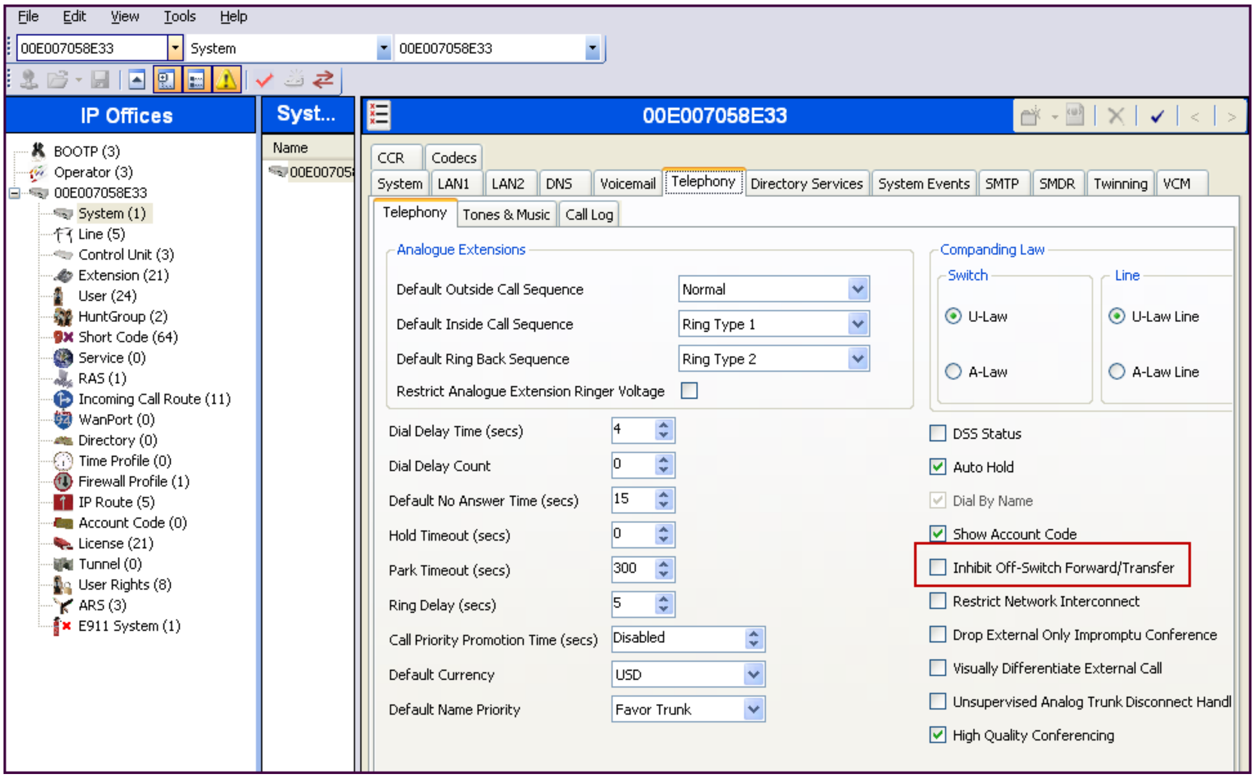
5.3. Voicemail Settings

On the **Voicemail** tab in the Details Pane, configure the **SIP Settings** section. 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 from Voicemail (e.g., Outcalling). 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. Uncheck the **Anonymous** box to allow Voicemail Caller ID information to the network.



5.4. System Telephony Settings

On the **Telephony** tab in the Details Pane, 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. If for security reasons incoming calls should not be allowed to transfer back to the PSTN then leave this setting checked.



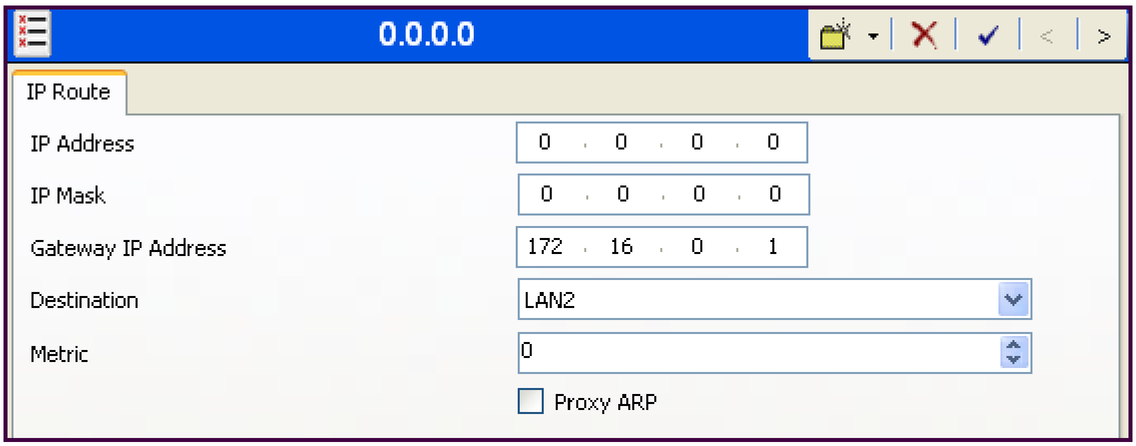
5.5. Twinning Calling Party Settings

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. This will allow the **Send Caller ID** setting in **Section 5.7.1** to control the calling party information for Mobile Twinning and forwarded calls.

5.6. IP Route

Navigate to **IP Route** in the left Navigation Pane, and then right-click on the Group Pane to select **New** (not shown). Create a default route with the following parameters:

- Set **IP Address** and **IP Mask** to ***0.0.0.0***.
- Set **Gateway IP Address** to the IP Address of the default router to reach Windstream.
- Set **Destination** to ***LAN2*** from the pull-down menu.



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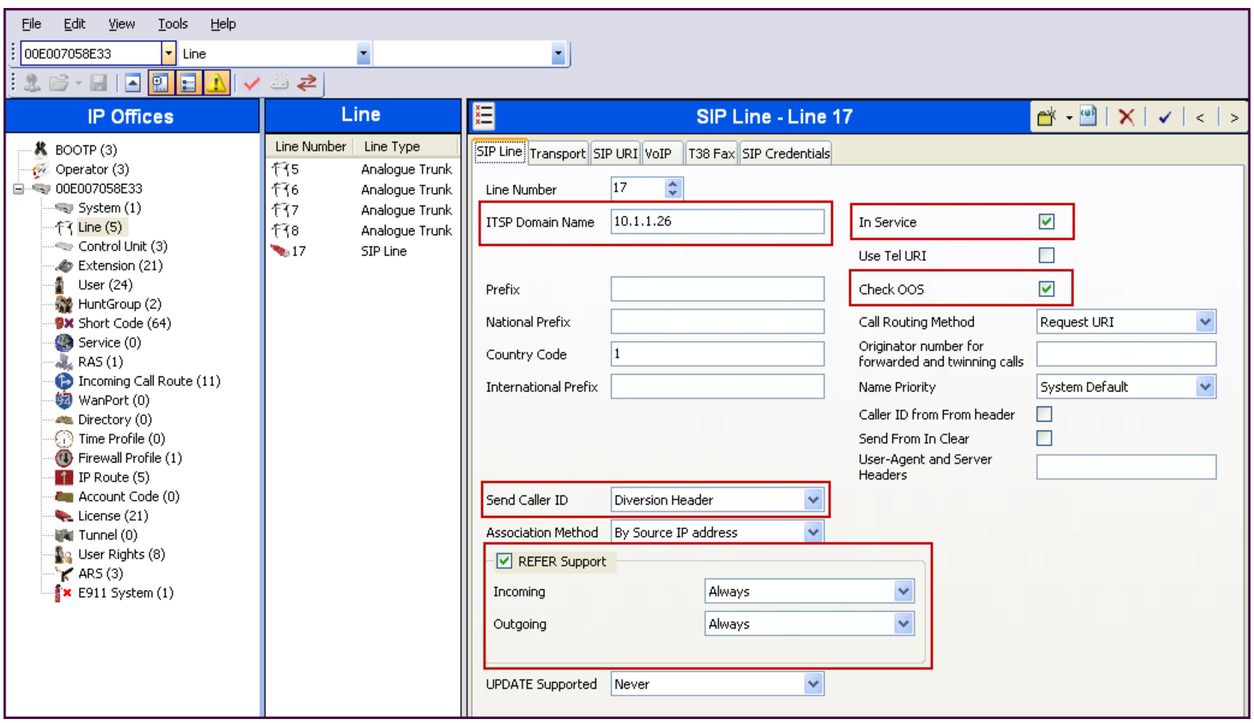
5.7. SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and Windstream SIP Trunking. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click in the Group Pane and select **New** → **SIP Line**.

5.7.1. SIP Line – SIP Line Tab

On the **SIP Line** tab in the Details Pane, configure the parameters as shown below.

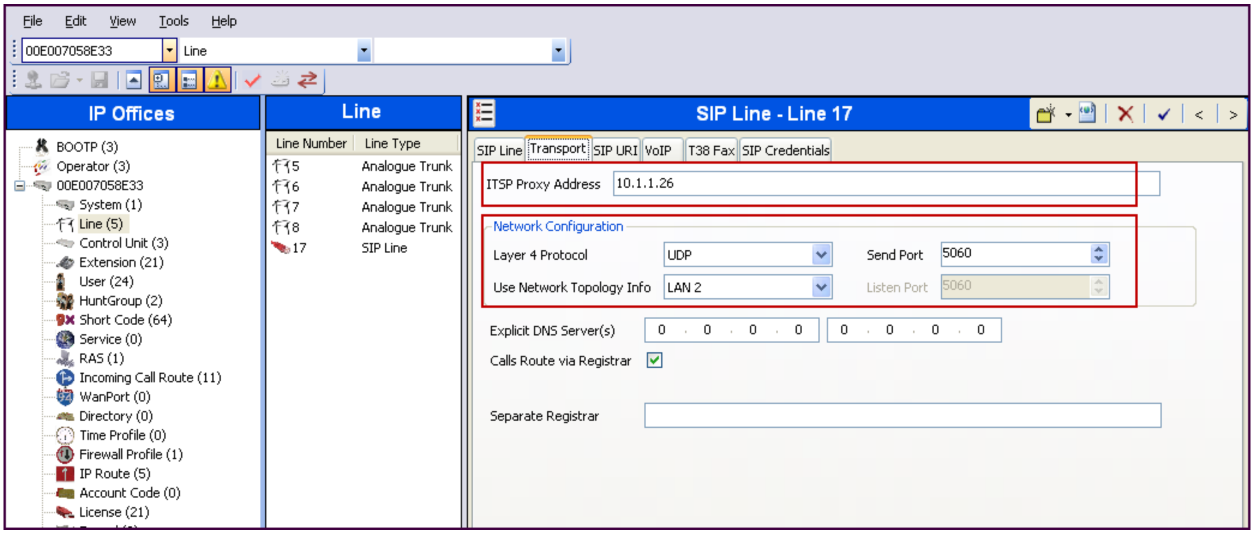
- Set **ITSP Domain Name** to the IP address of the Windstream SIP proxy.
- Set **Send Caller ID** to ***Diversion Header***. With this setting and the related configuration in **Section 5.5**, IP Office will include the Diversion Header for calls that are directed via Mobile Twinning out the SIP Line to Windstream. It will also include the Diversion Header for calls that are call forwarded out the SIP Line.
- Check **REFER Support**. Set the **Incoming** and **Outgoing** fields to ***Always***. See **Section 2.2** for limitations using REFER.
- Check the **In Service** box. This makes the trunk available to incoming and outgoing calls.
- Check the **Check OOS** box. IP Office will use the SIP OPTIONS method to periodically check the SIP Line. The time between SIP OPTIONS sent by IP Office will use the **Binding Refresh Time** for LAN2, as shown in **Section 5.2**.
- Default values may be used for all other parameters.



5.7.2. SIP Line - Transport Tab

Select the **Transport** tab. This tab was first introduced in Release 6.1. Some information configured in this tab had been under the **SIP Line** tab in Release 6.0. Set the parameters as shown below.

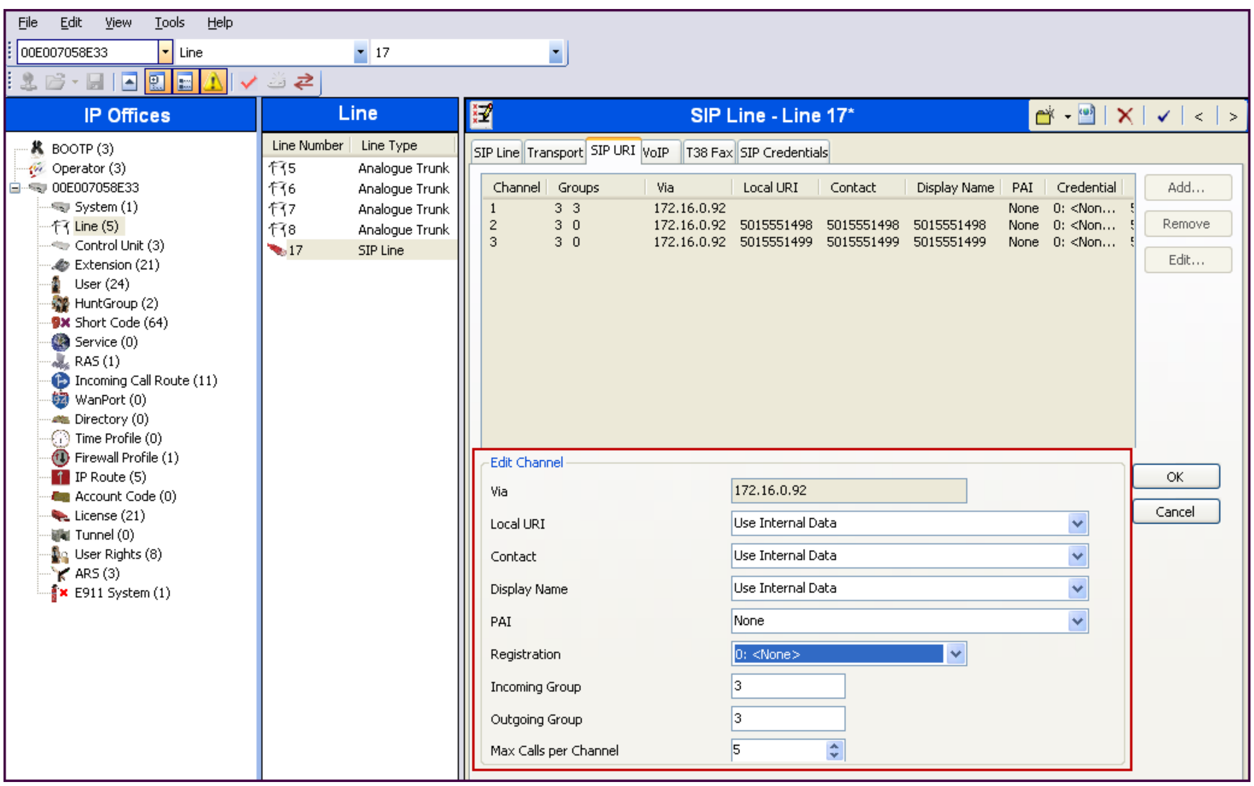
- Set **ITSP Proxy Address** to the IP address of the Windstream SIP proxy.
- Set **Layer 4 Protocol** to ***UDP***.
- Set **Use Network Topology Info** to the network port configured in **Section 5.2**.
- Set the **Send Port** to **5060**.
- Default values may be used for all other parameters.



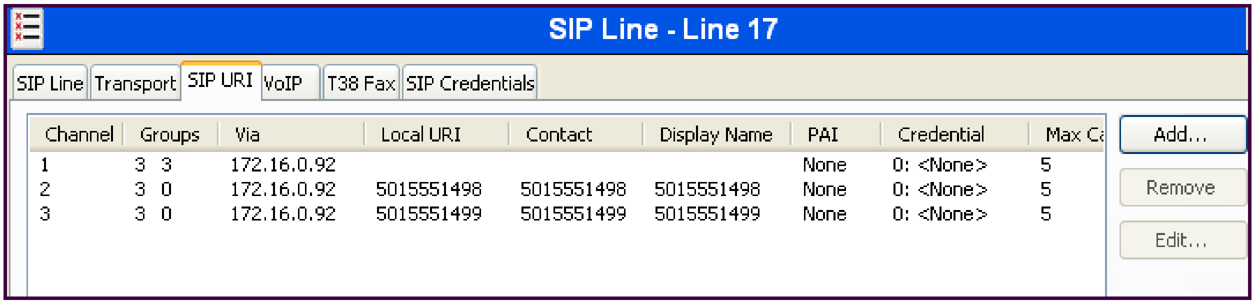
5.7.3. SIP Line - SIP URI Tab

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, 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. 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 whose SIP URI matches the number set in the **SIP** tab of any User as shown in **Section 5.9**.
- Associate this line with an incoming line group by entering a line group number 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. The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group ***3*** was defined that only contains this line (line 17).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.



In the sample configuration, the single SIP URI previously created shown below as **Channel 1** was 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. **Channels 2** and **3** display service numbers, such as a DID number routed directly to voicemail or DID used for Mobile Call Control. DID numbers that IP Office should admit can be entered into the Local URI and Contact fields instead of ***Use Internal Data***. The numbers ***501-555-1498*** and ***501-555-1499*** will be assigned as service numbers in the Incoming Call Routes in **Section 5.10**.



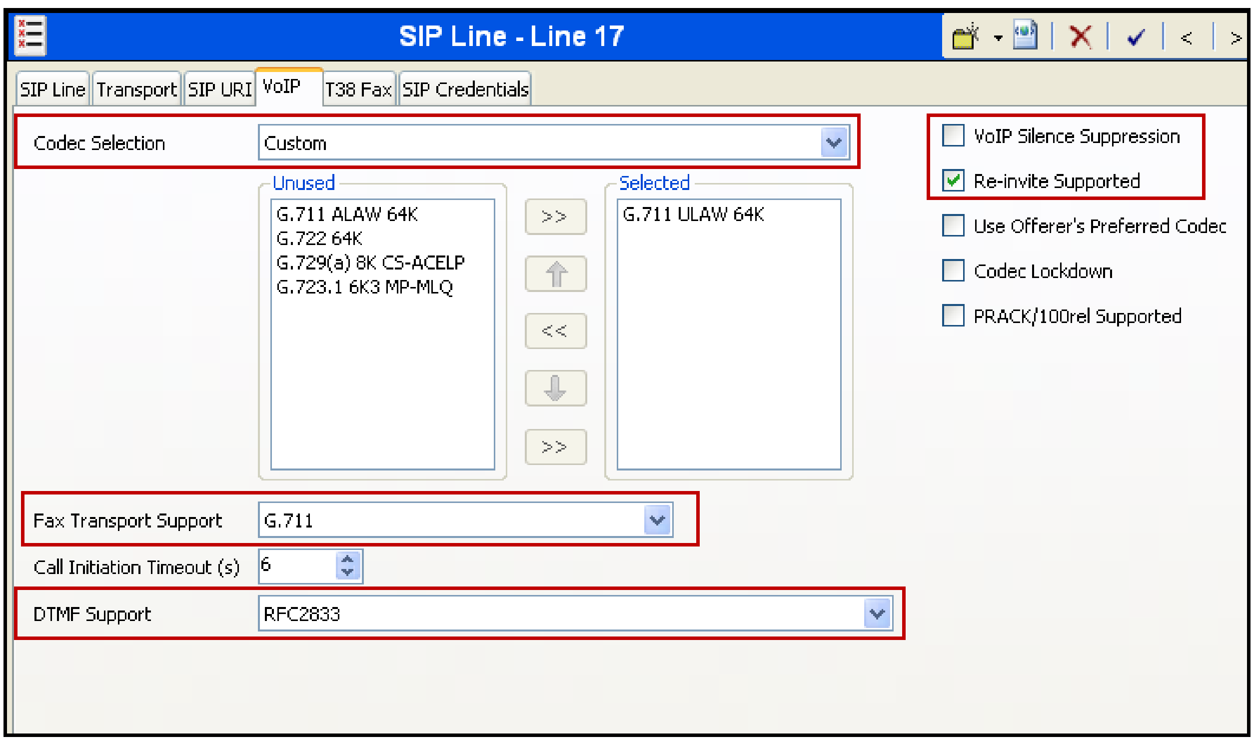
Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Ca
1	3 3	172.16.0.92				None	0: <None>	5
2	3 0	172.16.0.92	5015551498	5015551498	5015551498	None	0: <None>	5
3	3 0	172.16.0.92	5015551499	5015551499	5015551499	None	0: <None>	5

5.7.4. SIP Line - VoIP Tab

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

- Set the **Codec Selection** field to ***Custom*** to allow the specific codec selection to be different from the system default. To modify, click on a codec and use the arrow keys to move it to the **Selected** column and to change the order of preference. For compliance testing, ***G.711ULAW*** was used.
- Uncheck the **VoIP Silence Suppression** box.
- Check the **Re-invite Supported** box.
- Set the **Fax Transport Support** to ***G.711***. See **Section 2.2** for additional fax considerations.
- Set the **DTMF Support** field to ***RFC2833***. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Default values may be used for all other parameters.

Click the **OK** button at the bottom of the page (not shown).

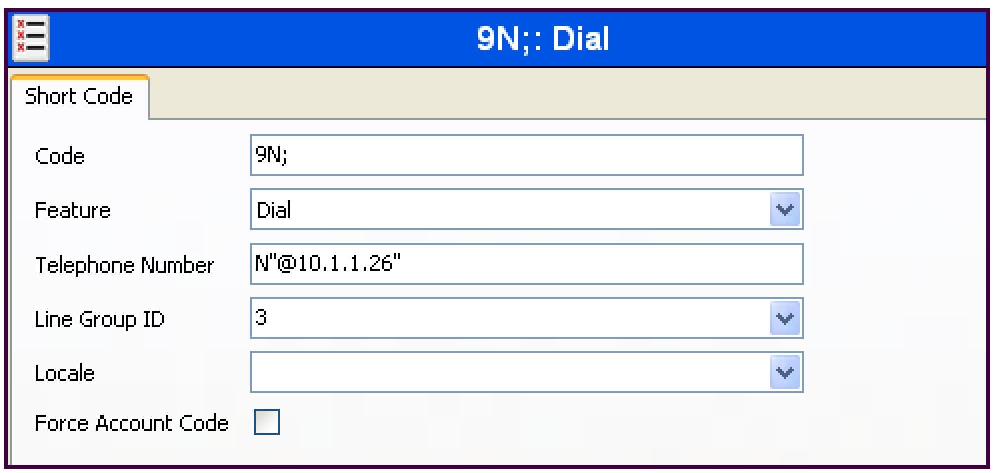


5.8. Short Codes

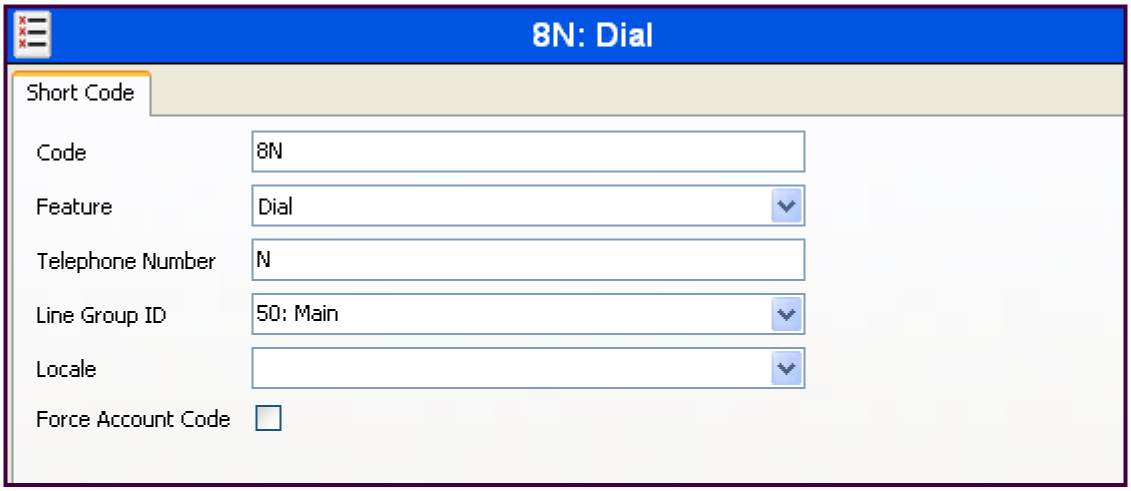
Define a short code to route outbound traffic to the SIP line. To create a short code, right-click on **Short Code** in the Navigation Pane and select **New** (not shown). On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

- 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"@10.1.1.26"***. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. The value **N** represents the number dialed by the user. The IP address 10.1.1.26 represents the IP address of the Windstream SIP proxy.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.7.3**. This short code will use this line group when placing the outbound call.

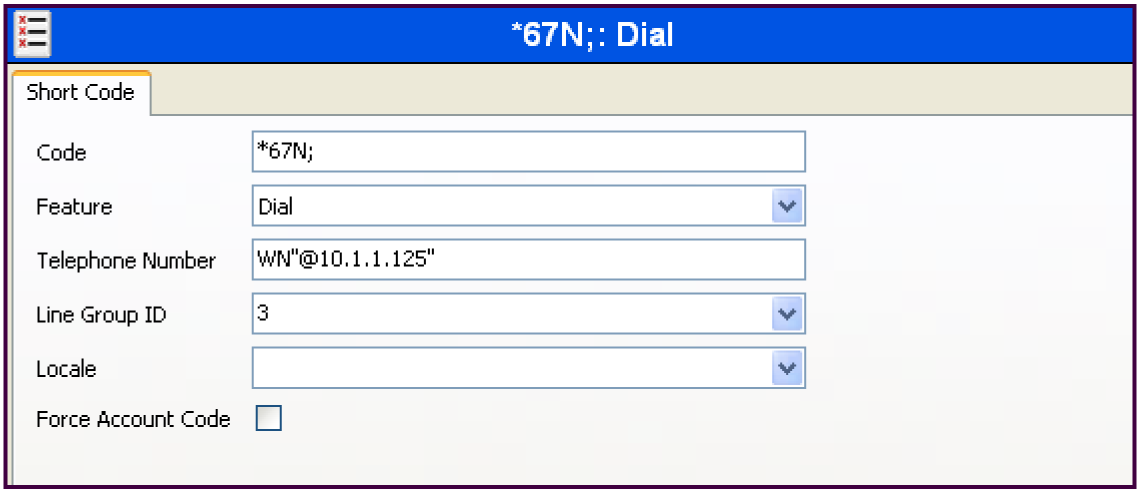
Click the OK button (not shown).



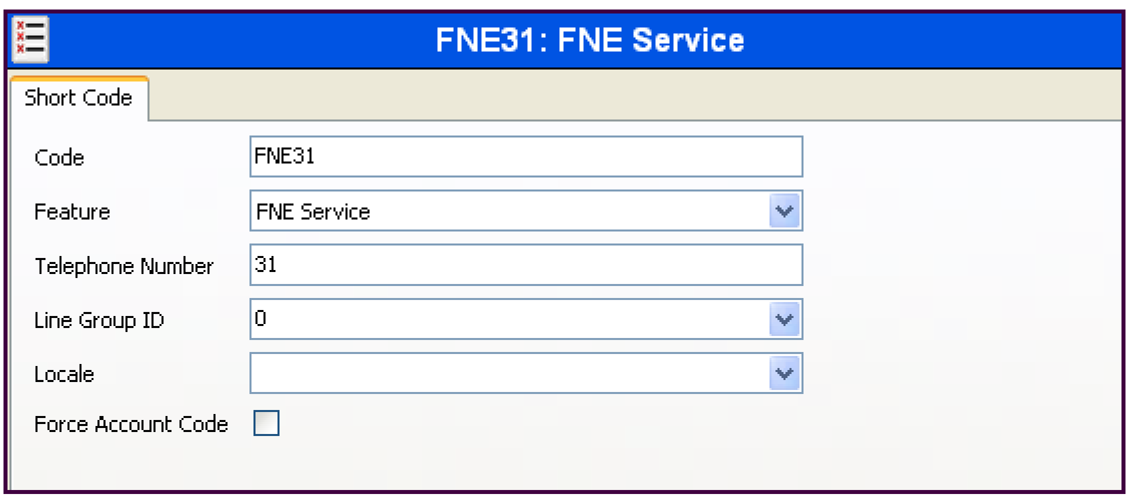
The simple **9N;** short code previously illustrated does not provide a means of alternate routing if the configured Windstream SIP Line is out of service or temporarily not responding. When alternate routing options and/or more customized analysis of the digits following the short code are desired, the Automatic Route Selection (ARS) feature may be used. In the following example screen, the short code ***8N*** is illustrated for access to ARS. When the Avaya IP Office user dials 8 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 5.11** 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 **9N;** 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 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-Asserted-Identity (PAI) header along with "Privacy: Id". Windstream will allow the call due to the presence of a valid DID in the PAI header, but will prevent presentation of the caller id to the called PSTN destination.

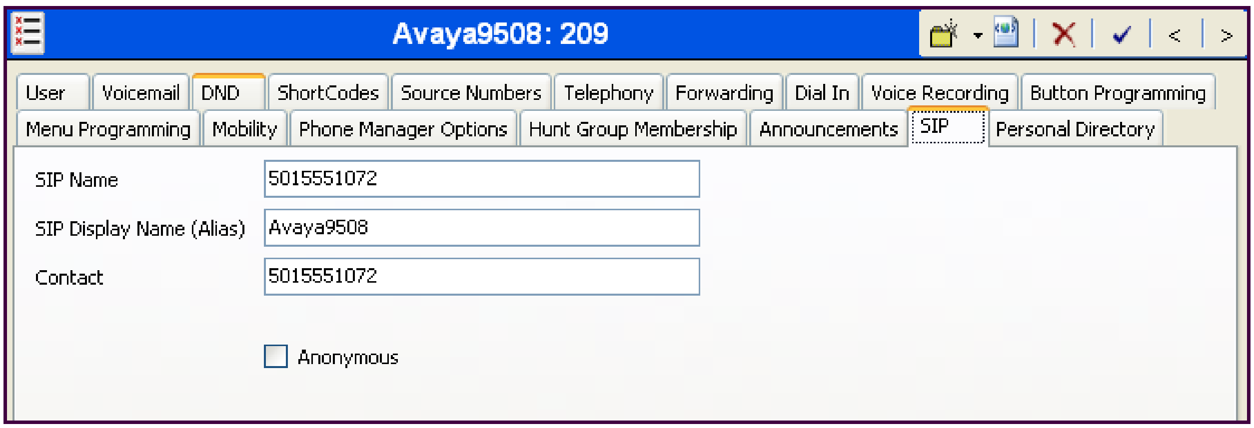


The following screen illustrates a short code that acts like a feature access code rather than a means to access a SIP Line. In this case, the **Code *FNE31*** is defined for **Feature *FNE Service*** to **Telephone Number *31*** (Mobile Call Control). This short code will be used as means to allow a 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 5.10**. 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.

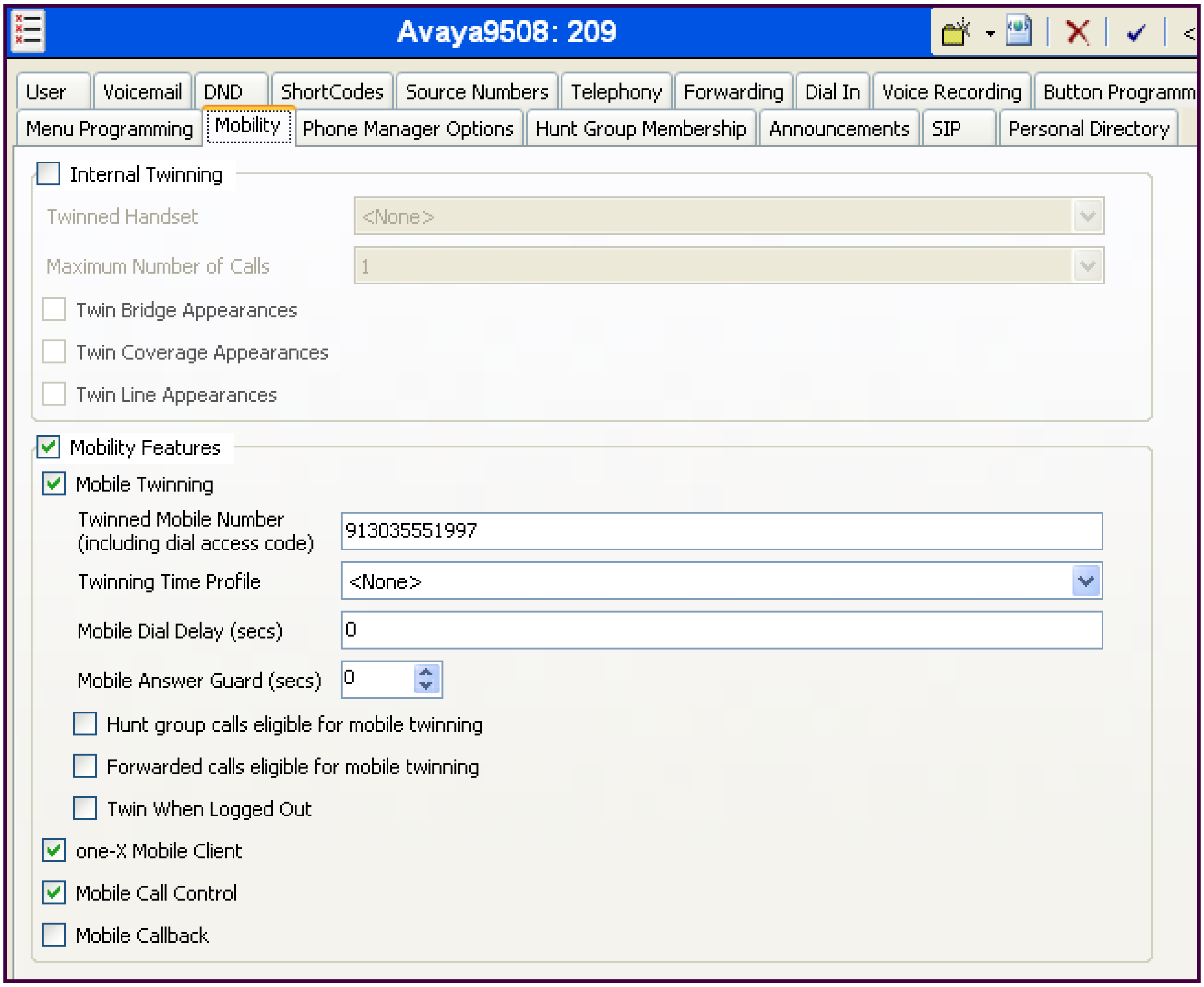


5.9. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.7**. To configure these settings, first navigate to **User** in the Navigation Pane, and then click on the user in the Group Pane to be modified. 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 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. See **Section 5.7.3**. The example below shows the settings for User 209. 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. Click the **OK** button (not shown).



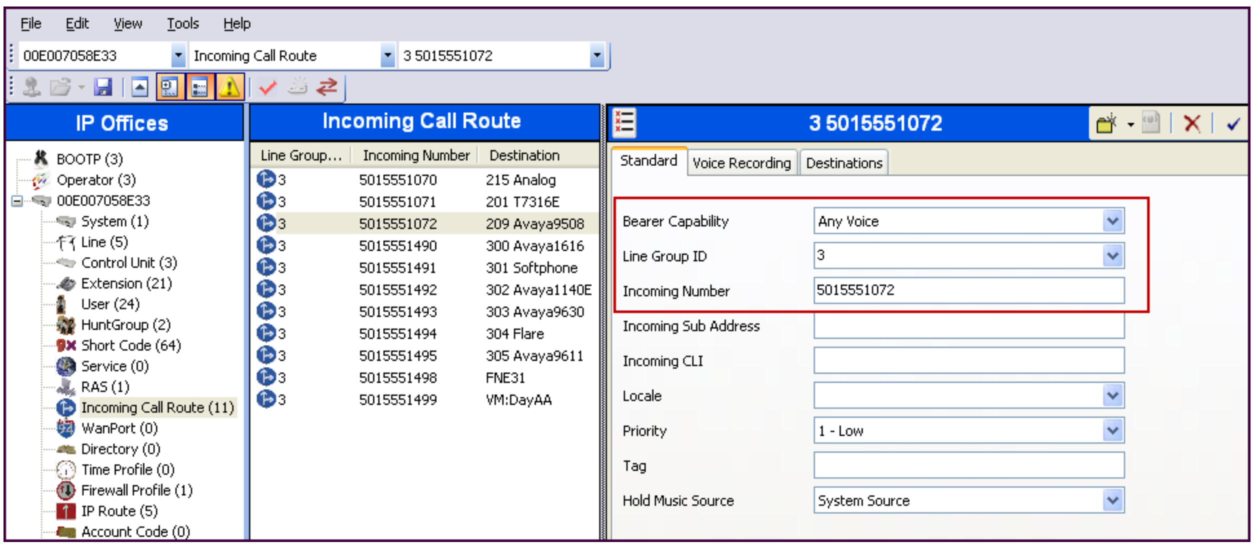
The following screen shows the **Mobility** tab for User 209. The **Mobility Features** and **Mobile Twinning boxes** are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone over the SIP trunk, in this case ***913035551997***. Other options can be set according to customer requirements.



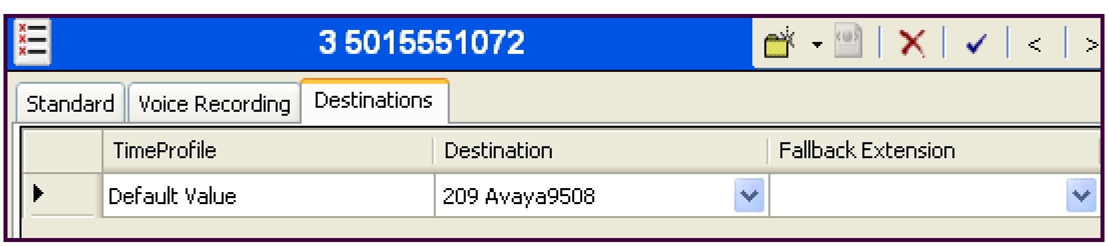
5.10. Incoming Call Route

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, right-click **Incoming Call Routes** in the Navigation 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 5.7.3**.
- Set the **Incoming Number** to the incoming number on which this route should match. Matching is right to left.
- Default values can be used for all other fields.

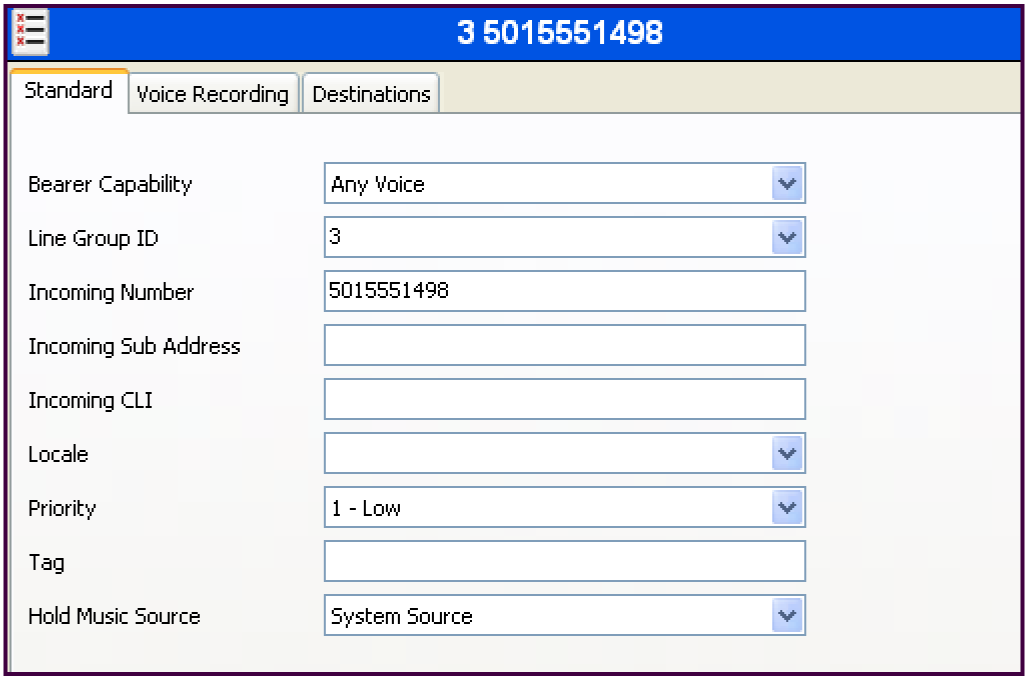


On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. Click the **OK** button (not shown). In this example, incoming calls to 501-555-1072 on line 3 are routed to extension 209.

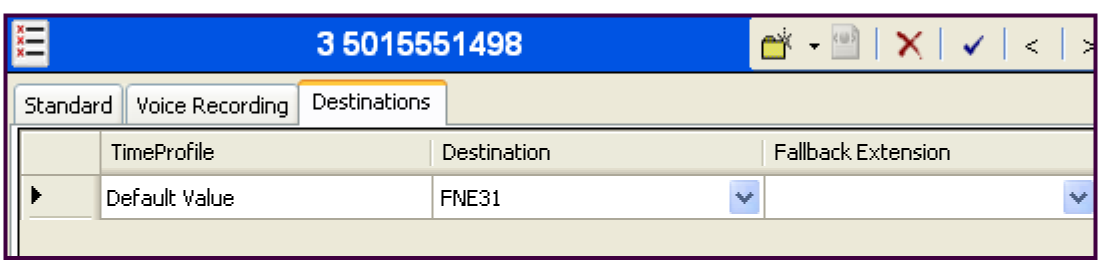


Incoming Call Routes for other direct mappings of DID numbers to IP Office users listed in **Figure 1** are omitted here, but can be configured in the same fashion.

In the screen shown below, the incoming call route for **Incoming Number *5015551498*** is illustrated. The **Line Group Id** is ***3***, matching the Incoming Group field configured in the SIP URI tab in **Section 5.7.3**.



When configuring an Incoming Call Route, the **Destination** field can be manually configured with a number such as a short code, or certain keywords available from the pull-down menu. For example, the following **Destinations** tab for an incoming call route contains the **Destination *FNE31*** entered manually. ***FNE31*** is the short code for ***FNE Service***, as shown in **Section 5.8**. An incoming call to 501-555-1498 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 5.9**); otherwise the IP Office responds with a 486 Busy Here and the caller will hear a busy tone.



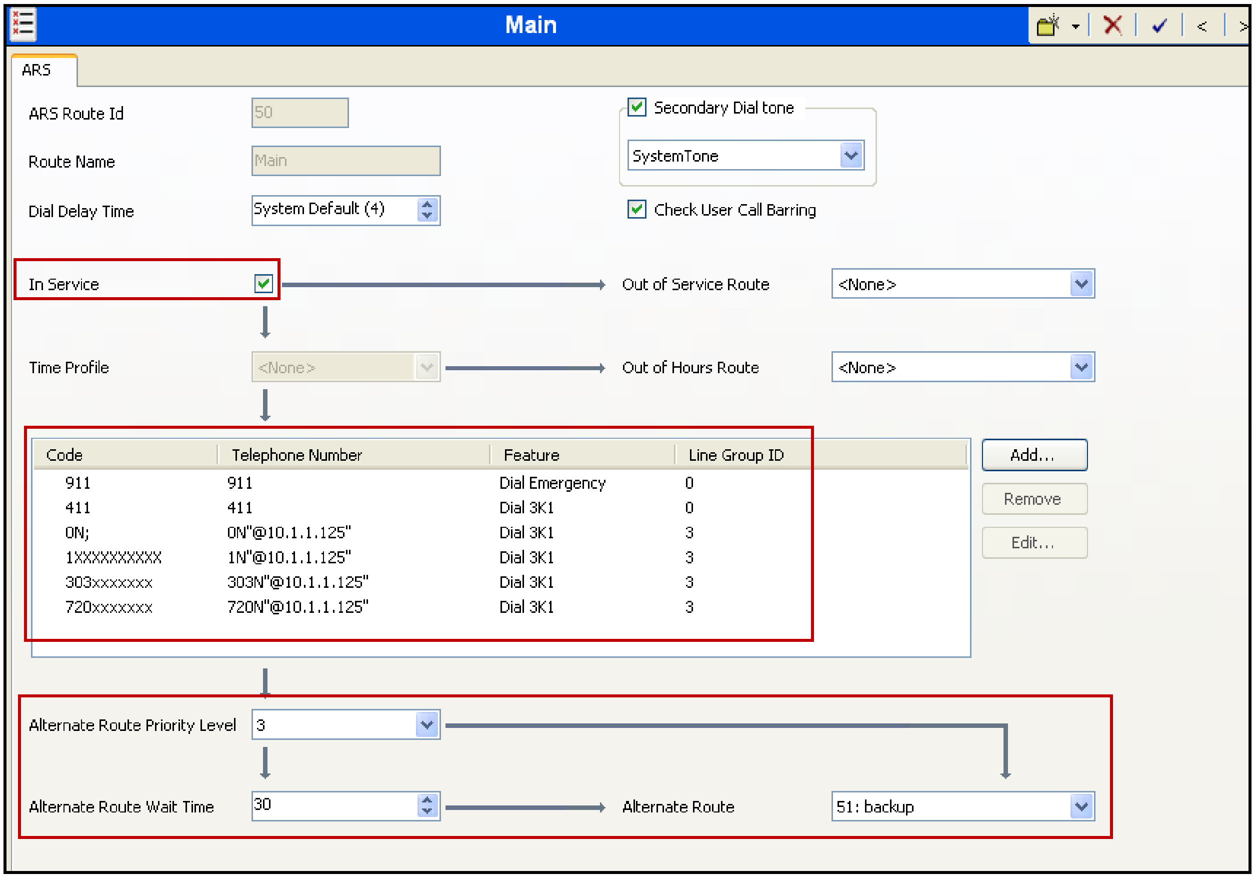
5.11. ARS and Alternate Routing

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

Optionally, Automatic Route Selection (ARS) can be used rather than the simple **9N;** short code approach documented in **Section 5.8**. 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 local and long distance calls should use the SIP Line preferentially, but service numbers should prefer a different outgoing line group, ARS can be used to distinguish the call behaviors.

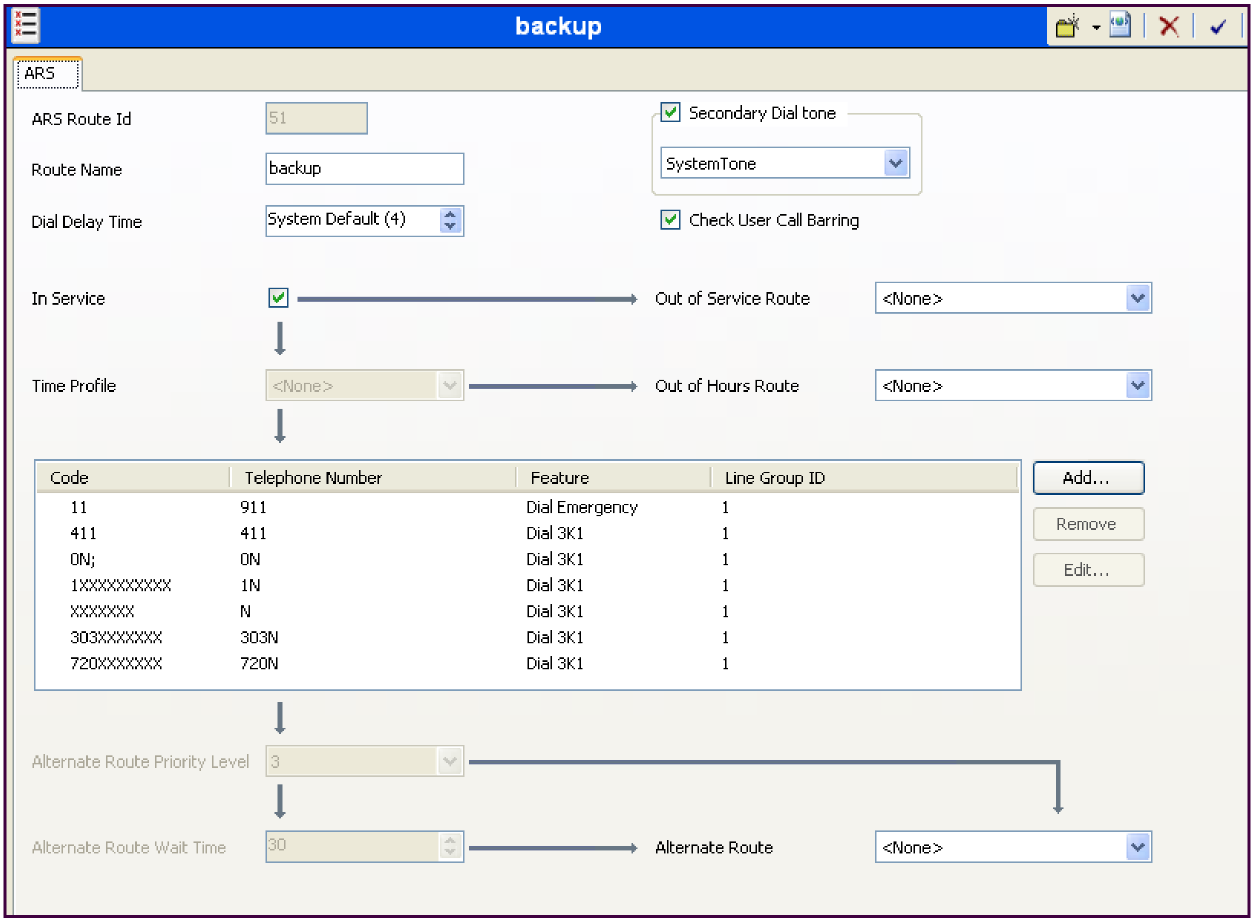
To add a new ARS route, right-click **ARS** in the Navigation pane and select **New**. To view or edit an existing ARS route, select ARS in the Navigation pane and select the appropriate route name in the Group pane (not shown).

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., 8N in **Section 5.8**) can be further analyzed to direct the call to a specific Line Group ID per the example screen above. If the user dialed 8-1-303-555-1997, the call would be directed to Line Group 3, configured in **Section 5.7.4**. If Line Group 3 cannot be used, the call can automatically route to the route name configured in the **Alternate Route** parameter in the lower right of the screen. Since alternate routing can be considered a privilege not available to all callers, IP Office can control access to the alternate route by comparing the calling user's priority to the value in the **Alternate Route Priority Level** field.

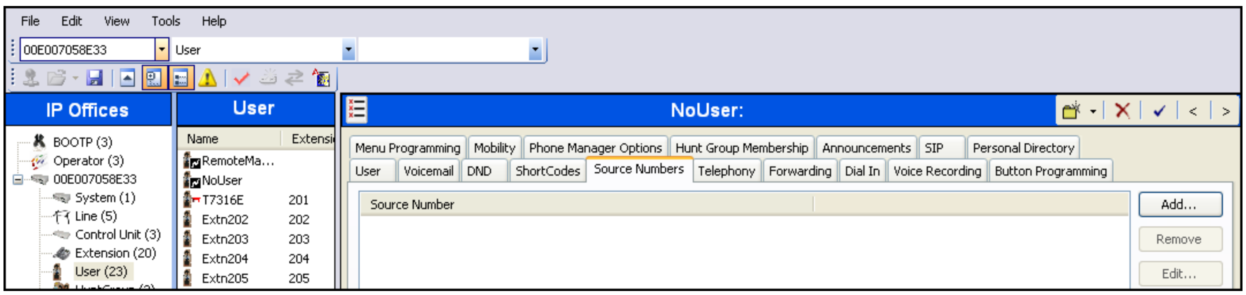
The following screen shows an example ARS configuration for the route named ***backup***, **ARS Route ID *51***. Continuing the example, if the user dialed 8-1-303-555-1997, and the call could not be routed via the primary route **50: Main** described above, the call will be delivered to this **Alternate Route**. Per the configuration shown below, the call will be delivered to **Line Group ID *1***, 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 5.8**.



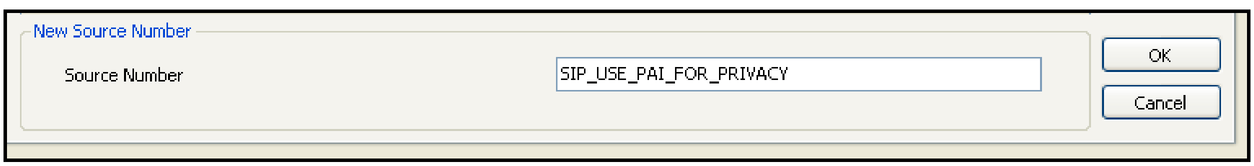
5.12. Privacy/Anonymous Calls

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. By default, Avaya IP Office will use PPI for privacy. For the compliance test, PAI was used for the purposes of privacy.

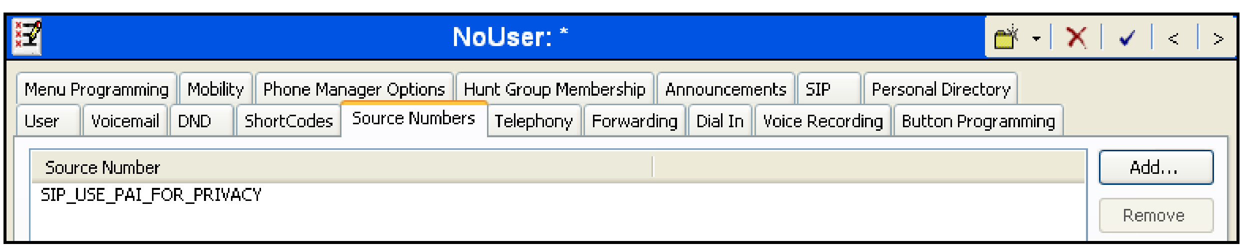
To configure Avaya IP Office to use PAI for privacy calls, navigate to **User** in the Navigation Pane, then **NoUser** in the Group Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



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



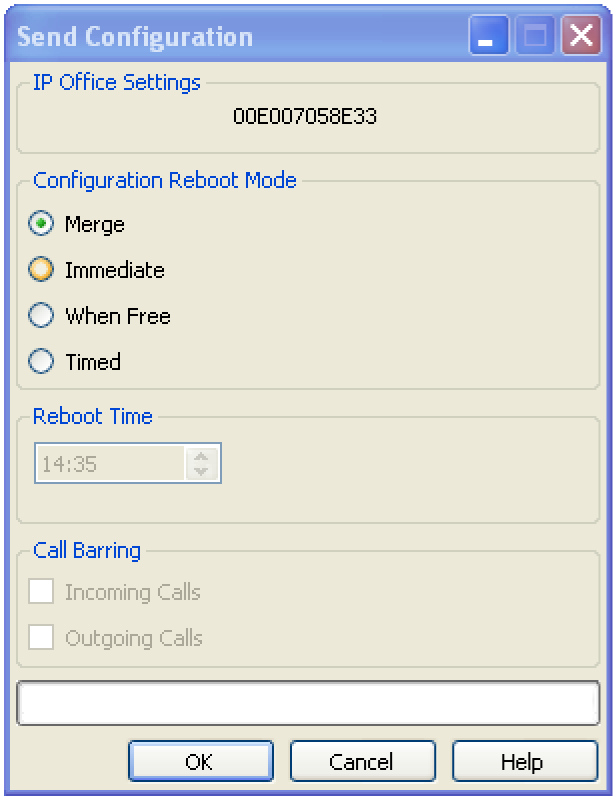
The **SIP_USE_PAI_FOR_PRIVACY** parameter will appear in the list of Source Numbers as shown below. Click **OK** at the bottom of the screen (not shown).



5.13. 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 SIP Trunking Configuration

Windstream is responsible for the configuration of Windstream SIP Trunking. The customer will need to provide the IP address 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 including:

- IP address of the Windstream SIP proxy
- Supported codecs
- DID numbers
- All IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices

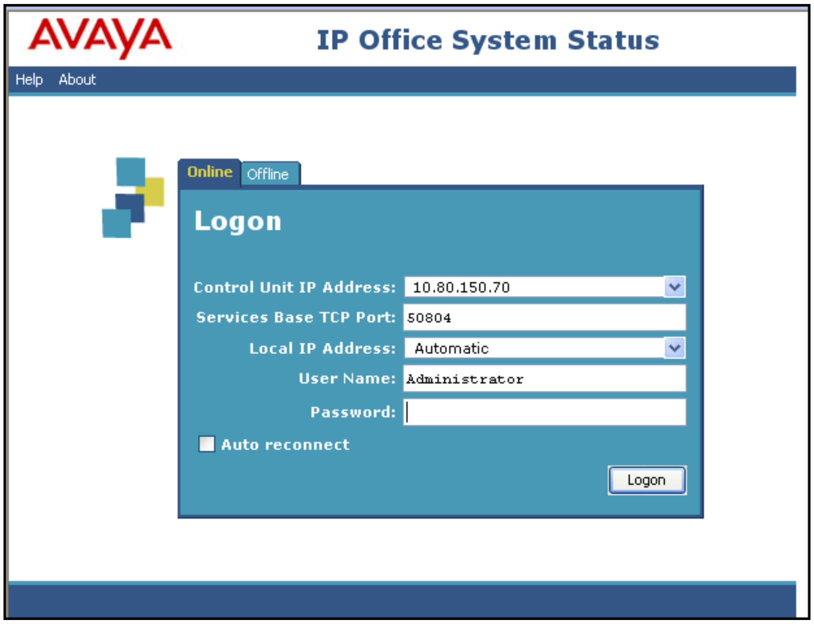
7. Verification Steps

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

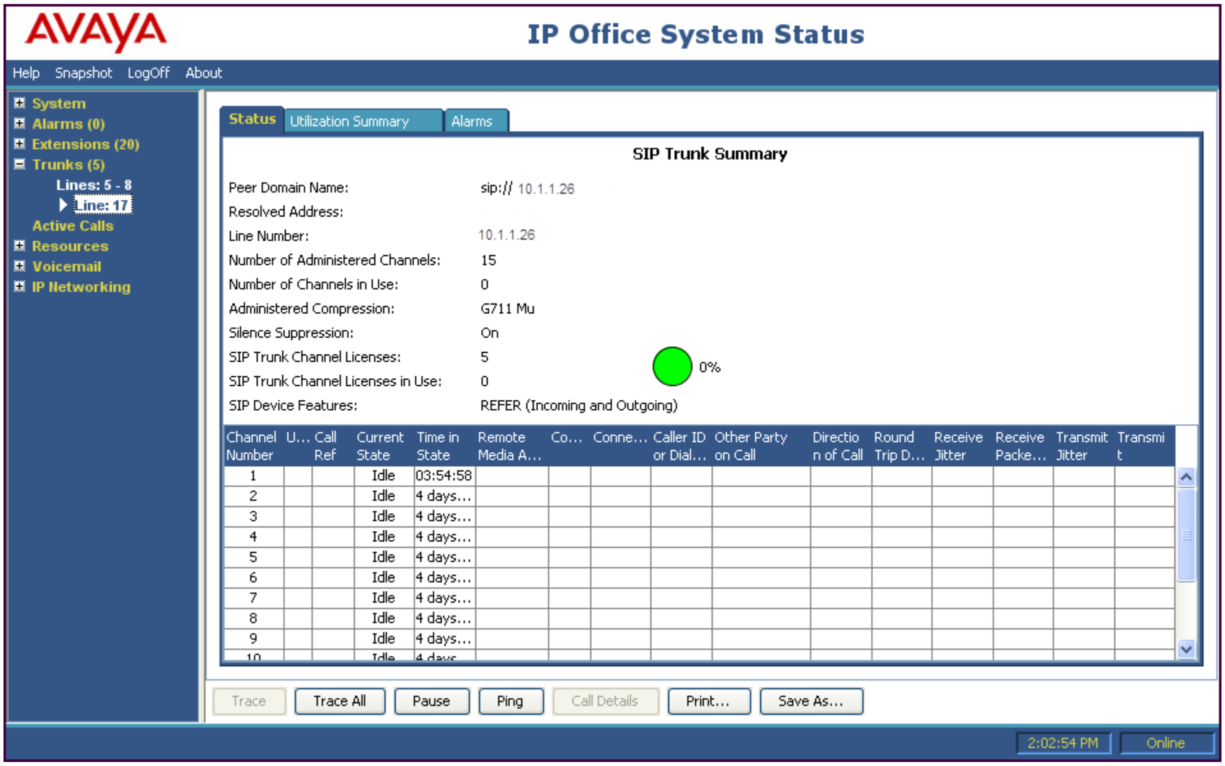
7.1. System Status

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

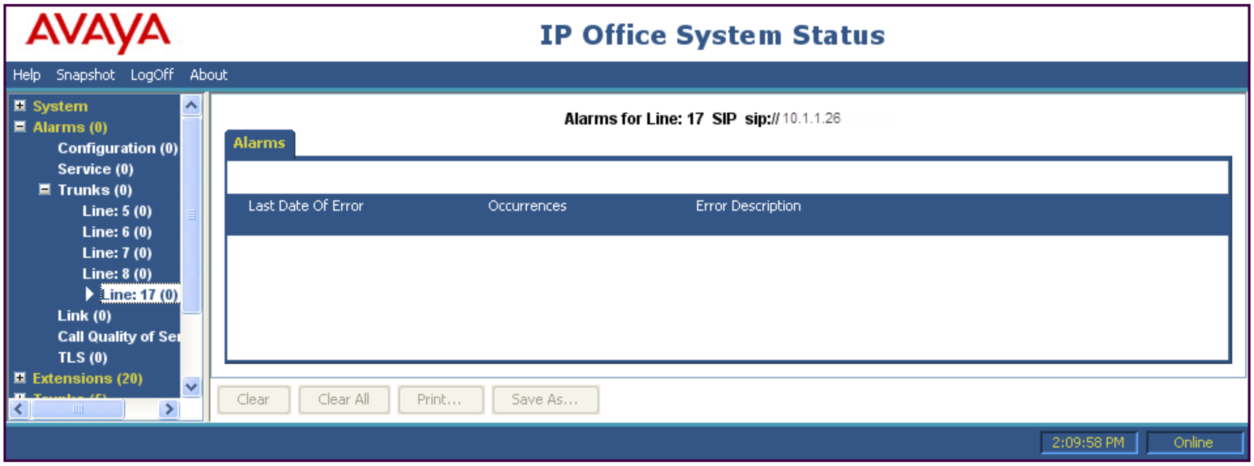
The following screen shows an example **Logon** screen. Enter the IP Office IP address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.



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


IP Office System Status
Help Snapshot LogOff About
System
Alarms (0)
Extensions (20)
Trunks (5)
Lines: 5 - 8
Line: 17
Active Calls
Resources
Voicemail
IP Networking
Status
Utilization Summary
Alarms
SIP Trunk Summary
Peer Domain Name: sip:// 10.1.1.26
Resolved Address:
Line Number: 10.1.1.26
Number of Administered Channels: 15
Number of Channels in Use: 0
Administered Compression: G711 Mu
Silence Suppression: On
SIP Trunk Channel Licenses: 5
SIP Trunk Channel Licenses in Use: 0
0%
SIP Device Features: REFER (Incoming and Outgoing)
Channel Number
U...
Call Ref
Current State
Time in State
Remote Media A...
Co...
Conne...
Caller ID or Dial...
Other Party on Call
Direction of Call
Round Trip D...
Receive Jitter
Receive Packe...
Transmit Jitter
Transmit
1 Idle 03:54:58
2 Idle 4 days...
3 Idle 4 days...
4 Idle 4 days...
5 Idle 4 days...
6 Idle 4 days...
7 Idle 4 days...
8 Idle 4 days...
9 Idle 4 days...
Trace
Trace All
Pause
Ping
Call Details
Print...
Save As...
2:02:54 PM
Online

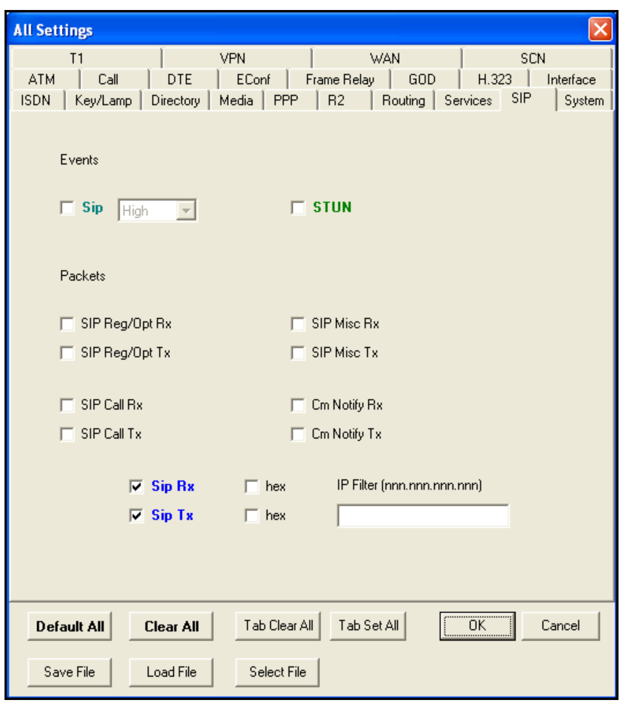
Select the **Alarms** tab and verify that no alarms are active on the SIP line.



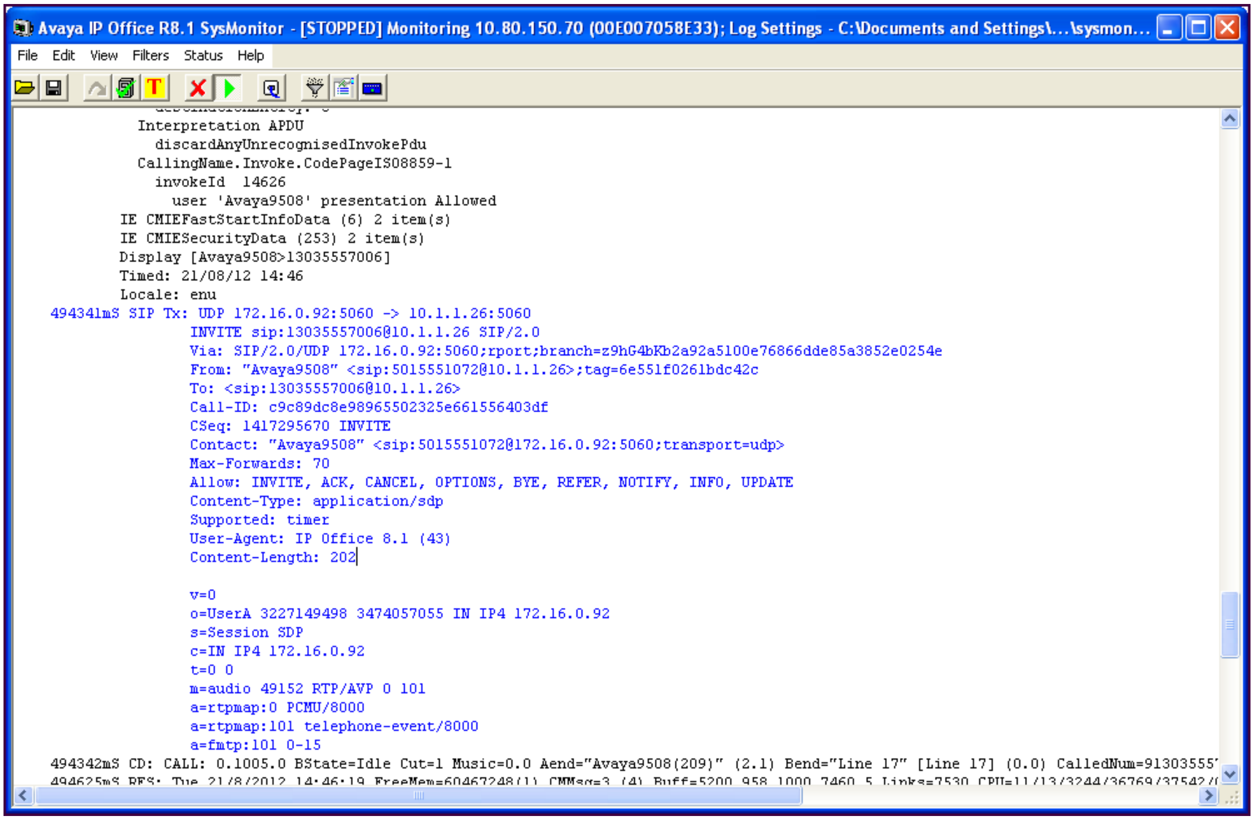
7.2. Monitor

The Monitor application can also be used to monitor and troubleshoot IP Office. Monitor can be accessed from **Start** → **Programs** → **IP Office** → **Monitor**. The application allows the monitored information to be customized. To customize, select the button that is third from the right in the screen below, or select **Filters** →**Trace Options**.

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



As an example, the following shows a portion of the monitoring window for an outbound call from extension 209, whose DID is 5015551072, calling out to the PSTN via the Windstream IP Trunking Service. The telephone user dialed 9-1-303-555-7006.



8. Conclusion

These Application Notes describe the configuration necessary to connect Avaya IP Office 8.1 to Windstream Metaswitch SIP Trunking service. Windstream SIP Trunking is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. It provides a flexible, cost-saving alternative to traditional hardwired telephony trunks. Windstream SIP Trunking passed compliance testing. Please refer to **Section 2.2** for any exceptions.

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 8.1 IP Office Installation, Document Number 15-601042, Issue 26h, August 17, 2012

[2] IP Office Release 8.1 Manager 10.1, Document Number 15-601011, Issue 29o, August 03, 2012

[3] IP Office System Status Application, Issue 06b, November 12, 2011 Document Number 15-601758

[4] IP Office Release 8.1 Implementing Voicemail Pro, Document Number 15-601064, Issue 03a, June 12 2012

[5] IP Office System Monitor, Document Number 15-601019, Issue 02b

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

10. Appendix - SIP Line Template

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

Note that not all of the configuration information is included in the SIP Line Template, particularly items relevant to a specific installation 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 5.7** in these Application Notes as a reference.

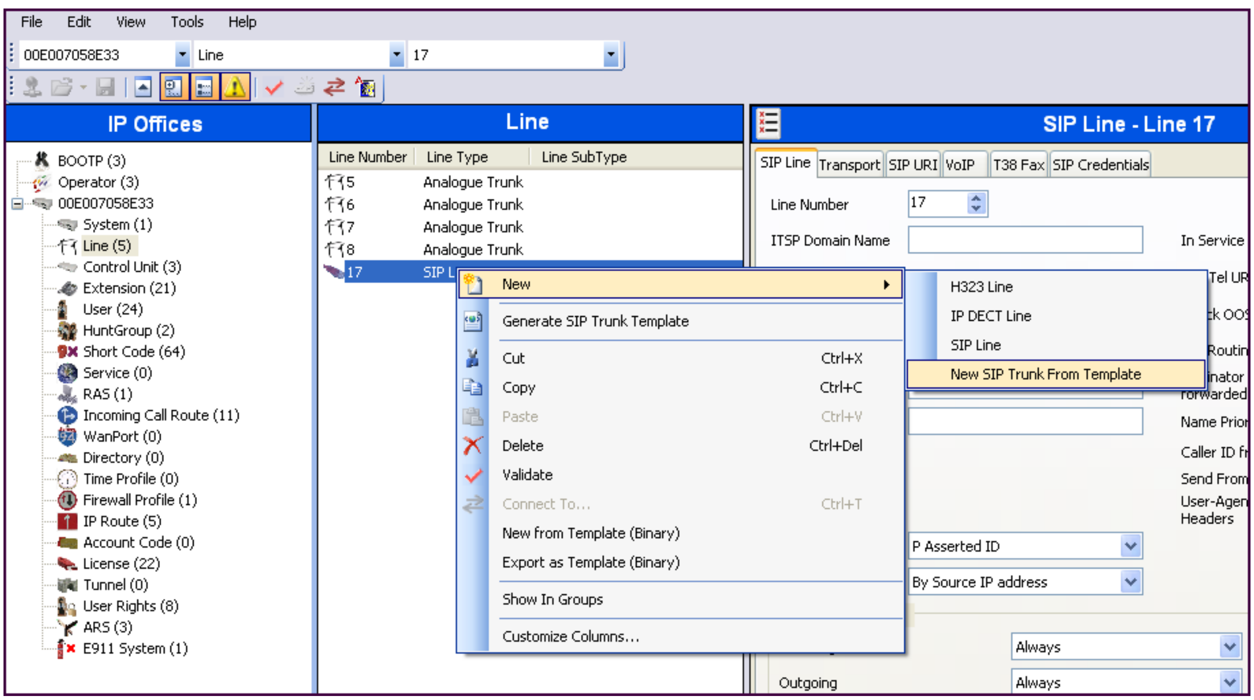
The SIP Line Template created from the configuration as documented in these Application Notes is as follows:


```
<?xml version="1.0" encoding="utf-8"?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20120817</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>Windstream Metaswitch IPO81</DescriptiveName>
  <ITSPDomainName>10.1.1.26</ITSPDomainName>
  <SendCallerID>CallerIDDIV</SendCallerID>
  <ReferSupport>true</ReferSupport>
  <ReferSupportIncoming>1</ReferSupportIncoming>
  <ReferSupportOutgoing>1</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>true</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>SystemDefault</LineNamePriority>
  <UpdateSupport>UpdateNever</UpdateSupport>
  <UserAgentServerHeader />
  <CallerIDfromFromheader>false</CallerIDfromFromheader>
  <PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
  <ITSPProxy>10.1.1.26</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.711 ULAW 64K</AdvCodecPref>
  <CallInitiationTimeout>6</CallInitiationTimeout>
  <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
  <VoipSilenceSupression>false</VoipSilenceSupression>
  <ReinviteSupported>true</ReinviteSupported>
  <FaxTransportSupport>FOIP_G711</FaxTransportSupport>
  <UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
  <CodecLockdown>false</CodecLockdown>
  <Rel100Supported>false</Rel100Supported>
```

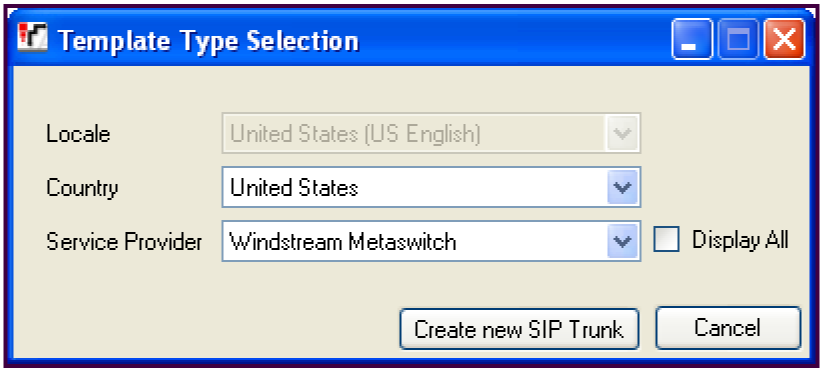
```
  <T38FaxVersion>0</T38FaxVersion>
  <Transport>UDPTL</Transport>
  <LowSpeed>0</LowSpeed>
  <HighSpeed>0</HighSpeed>
  <TCFMethod>Trans_TCF</TCFMethod>
  <MaxBitRate>FaxRate_9600</MaxBitRate>
  <EflagStartTimer>2600</EflagStartTimer>
  <EflagStopTimer>2300</EflagStopTimer>
  <UseDefaultValues>false</UseDefaultValues>
  <ScanLineFixup>true</ScanLineFixup>
  <TFOPEnhancement>true</TFOPEnhancement>
  <DisableT30ECM>false</DisableT30ECM>
  <DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
  <DisableT30MRCompression>false</DisableT30MRCompression>
  <NSFOverride>false</NSFOverride>
</Template>
```


1. On the PC where IP Office Manager was installed, copy and paste the above template into a text document named **US_Windstream Metaswitch_SIPTrunk.xml** (the document must be named EXACTLY as show). Move the .xml file to the IP Office Manager template directory (C:\Program Files\Avaya\IP Office\Manager\Templates). It may be necessary to create the directory if it does not already exist.

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



3. Verify that ***United States*** is automatically populated for **Country** and ***Windstream Metaswitch*** 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.