Avaya Solution & Interoperability Test Lab

Application Notes for Windstream SIP Trunking Service using Broadsoft Platform with Avaya IP Office 8.1 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between Windstream SIP Trunking Service using Broadsoft Platform and Avaya IP Office 8.1.

Windstream SIP Trunking Service provides PSTN access via a SIP Trunk between the enterprise and Windstream networks as an alternative to legacy analog or ISDN-PRI trunks. This approach generally results in lower cost for the enterprise.

Windstream is a member of the Avaya DevConnect Service Provider Program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted 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 Windstream SIP Trunking Service using Broadsoft Platform (Windstream) and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of Avaya IP Office (IP Office) 500v2 Release 8.1 and various Avaya endpoints.

Windstream SIP Trunking Service using Broadsoft Platform (Windstream) referenced within these Application Notes is designed for business customers. The service enables PSTN calling via a broadband WAN connection using SIP protocol. This converged network solution is a cost effective alternative to traditional PSTN trunks such as analog and/or ISDN-PRI.

To establish the SIP Trunk, Windstream requires IP Office to implement dynamic registration. IP Office will send a SIP REGISTER request with a configured user name. This credential is provided by Windstream and configured on IP Office. If the credential is verified, Windstream will respond with a SIP 200 OK message to bring up the SIP Trunk.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using IP Office to connect to Windstream. This configuration (shown in **Figure 1**) was used to exercise the feature and functionalities 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

To verify Windstream SIP Trunking interoperability, the following features and functionalities were exercised during the compliance testing:

- Incoming PSTN calls to various phone types including H.323, SIP, digital and analog telephones at the enterprise. All incoming calls from PSTN were routed to the enterprise across the SIP Trunk from the service provider.
- Outgoing PSTN calls from various phone types including SIP, H.323, digital and analog telephone at the enterprise. All outgoing calls to PSTN were routed from the enterprise across the SIP Trunk to the service provider.
- Incoming and outgoing PSTN calls to/from Avaya IP Office Softphone using both SIP and H.323 protocols.
- Dialing plans including local, long distance, international, outgoing toll-free, operator, operator assisted, local directory assistance 411 calls, etc.
- Calling Party Name presentation and Calling Party Name restriction (using the "PPI" header).
- Proper codec negotiation of G.711MU and G.729 codecs.

- Proper media transmission using G.711MU and G.729 codecs.
- Proper early media transmissions G.711MU and G.729 codecs.
- Incoming and outgoing fax over IP with G.711MU codec.
- DTMF tone transmissions as out-of-band RTP events per RFC 2833.
- Voicemail navigation for incoming and outgoing calls.
- Telephony features such as hold and resume, call transfer, call forward and conferencing.
- Off-net call transfer using re-INVITE method.
- Off-net call forward using Diversion method.
- Twinning the incoming calls to mobile phones using Diversion method.
- Dynamic SIP Trunk registration using REGISTER method.
- Response to OPTIONS heartbeat.
- Session Timer refresh per RFC 4028.
- Response to incomplete call attempts and trunk errors.

Items that are not supported by Windstream or not part of the compliance testing, are listed as follows:

- Inbound toll-free and outgoing emergency (E911) calls are supported but were not tested as part of the compliance testing because the necessary configuration was not available during testing.
- Reliability of Provisional Responses (RFC 3262) is not supported.
- Fax over IP with T.38 codec is not supported.
- Off-net call forward with History-Info method is not supported.

2.2 Test Results

Interoperability testing of Windstream with the Avaya SIP-enabled enterprise solution was successfully completed with the exception of the observations/limitations described below.

1. **Calling Party Name of the originating PSTN party of incoming calls is not consistently delivered to IP Office**. In the “From” header of the incoming INVITE, sometimes Windstream did not deliver Calling Party Name. At other times, it delivered Calling Party Name as “Toronto ON”, “Anonymous” or “Unavailable”. This issue has been encountered in the test environment, but Windstream claims it can deliver Calling Party Name reliably in production environments.

2. **Calling Party Name of outgoing calls is not delivered to PSTN**. In the “From” header of the outgoing INVITE, IP Office sends Calling Party Name to Windstream. However, Windstream did not deliver it to the PSTN. This issue has been encountered in the test environment, but Windstream claims it can deliver Calling Party Name reliably in production environments.

3. **Calling Party Name and Number are not updated if IP Office off-net redirects (by transferring or forwarding) an incoming or outgoing PSTN call to internal station**. Before and after completing the local redirection call to internal station, IP Office did not send UPDATE or re-INVITE signaling to update the call display to PSTN party. This issue has been encountered in the test environment, but it has been confirmed to be supported by Windstream on the production network with extra cost.

4. **Calling Party Number is not updated if IP Office off-net redirects (by transferring or forwarding) an incoming or outgoing PSTN call back to PSTN**. Before completing the off-net call redirection, IP Office did not send UPDATE or re-INVITE signaling to update the call display to PSTN parties. This is a known behavior of IP Office with no available resolution at this time. This issue has low user impact, it is listed here simply as an observation. **Note**: even if the Calling Party Name and Number update feature is supported by enterprise system, the customer still needs to order additional features Enterprise Trunking feature from Windstream for this to work.

5. **Calling Party Number is now corrected if IP Office off-net call forward an incoming or outgoing PSTN call back to PSTN**. In order to perform off-net call forwarding, IP Office sent initial INVITE on the 2nd leg with the "Diversion" containing a subscribed DID number to support call authentication done by service provider. The same DID number has also been sent in the "P-Asserted-Identity" header, this caused the forwarded PSTN party to unexpectedly display DID number of IP Office station instead of displaying Calling Party Number of the originating PSTN party in the "From" header. This issue has been corrected by removing the support for the "P-Asserted-Identity" header on the SIP Trunk configuration on IP Office. As a result, IP Office does not send the "P-Asserted-Identity" header in the signaling to service provider. This workaround assists service provider transmitting proper display info which is presented in the "From" header. For more information, refer to **Section 5.5**. This is a known behavior of IP Office with no available resolution at this time. This issue has low user impact, it is listed here simply as an observation. **Note**: The "P-Asserted-Identity" header is recommended by Windstream to be disabled on the SIP Trunk. By disabling the "P-Asserted-Identity" header, IP Office will use the "P-Preferred-Identity" header for outgoing private calls to provide necessary information for call authentication done by service provider and the calls appeared to work well.

6. **Calling Party Number is not properly displayed before SIP station completes blind transferring of an outgoing PSTN call to H.323 station**. Before the H.323 station answers the blind transferred call, it displays "external" instead of displaying Calling Party Number of the called PSTN party. The issue did not happen when using the H.323 station to perform the blind transfer. This is a known behavior of IP Office with no available resolution at this time. This issue has low user impact, it is listed here simply as an observation.

7. **Off-net call forward unconditional or busy calls fail if the forwarded-to PSTN party takes longer than 8 seconds to respond**. The issue has been seen on cellular PSTN phone. IP Office relied on the response time of the cellular PSTN phone on the 2nd leg to transmit ringback to the original calling PSTN party on the 1st leg. If the response time exceeded 8 seconds, Windstream terminated the call by sending CANCEL request on the 1st call leg. This failed the scenarios. The issue does not happen if the cellular PSTN phone has a response time under 8 second limit. The same call scenarios are successful on the regular PSTN phone (instead of the cellular PSTN party). Windstream is recommended to increase Session Provisioning Timer to correct the issue. This issue is acknowledged as a known behavior of Windstream SIP Trunking Service using Broadsoft Platform with no resolution available at this time. It is listed here as a limitation.

8. **Off-net call transfer is now working properly using re-INVITE method**. If REFER method was supported on the SIP Trunk, SIP station displayed "transfer failed" even after it successfully completed an off-net call transfer call back to PSTN. The issue occurred because Windstream did not send a NOTIFY in responding to the REFER from IP Office. As a workaround, REFER support was disabled during the compliance testing, this forced IP Office to use re-INVITE for off-net call transfer and the calls appeared to work well. This issue is acknowledged as a known behavior of Windstream SIP Trunking Service using Broadsoft Platform with no resolution available at this time. This issue has low user impact, it is listed here simply as an observation.

9. **Performing a warm restart on IP Office causes the SIP Trunk to go out of service temporarily**. It has been observed that when performing a warm restart on IP Office e.g., to apply a configuration change, IP Office sent REGISTER/ Expires: 0 to de-register then follow by another REGISTER/ Expires: 300 to re-register back. Windstream unexpectedly responded 200 OK to REGISTER/ Expires: 300 before 200 OK to REGISTER/ Expires: 0. After this event, IP Office assumed the REGISTRATION was successfully complete and the SIP Trunk was now active. However, by sending 200 OK in incorrect order, Windstream tagged the SIP Trunk as inactive. It resulted all incoming and outgoing calls could not be successfully complete. Until the REGISTRATION expired in 300 seconds, IP Office sent another REGISTER and it helped to clear up the in-active state of the SIP Trunk at service provider side then Windstream started to process incoming and outgoing calls as expected.

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 Service, please contact Windstream technical support at:

- Phone: 1 (866) 990-3282
- Website: http://www.windstreambusiness.com/support/customer-support

3. Reference Configuration

Figure 1 below illustrates the test configuration. It shows an enterprise site connected to the Windstream network through the internet.

For confidentiality and privacy purposes, the actual public IP addresses and PSTN routable phone numbers used in the certification testing have been replaced with fictitious parameters throughout the Application Notes.

The Avaya components used to create the simulated customer site included:

- Avaya IP Office v500
- Avaya Voicemail Pro for IP Office
- Avaya 9600 Series H.323 IP Telephones
- Avaya 11x0 Series SIP IP Telephones
- Avaya IP Office Softphones (SIP and H.323 modes)
- Avaya 1408D Digital Telephones

- Avaya Symphony 2000 Analog Telephones

Located at the enterprise site is the Avaya IP Office 500v2 with the MOD DGTL STA16 expansion to provide connection for 16 digital stations, the PHONE 8 module to provide connection for 8 analog stations and the 64-channel Voice Compression Module (VCM) for supporting VoIP codec. IP Office has the LAN port that connects to the enterprise networks and the WAN port that connects to Windstream network via the Internet.

Mobility Twinning is configured for some IP Office users so that incoming calls to these user phones can also be delivered to the configured mobile phones.

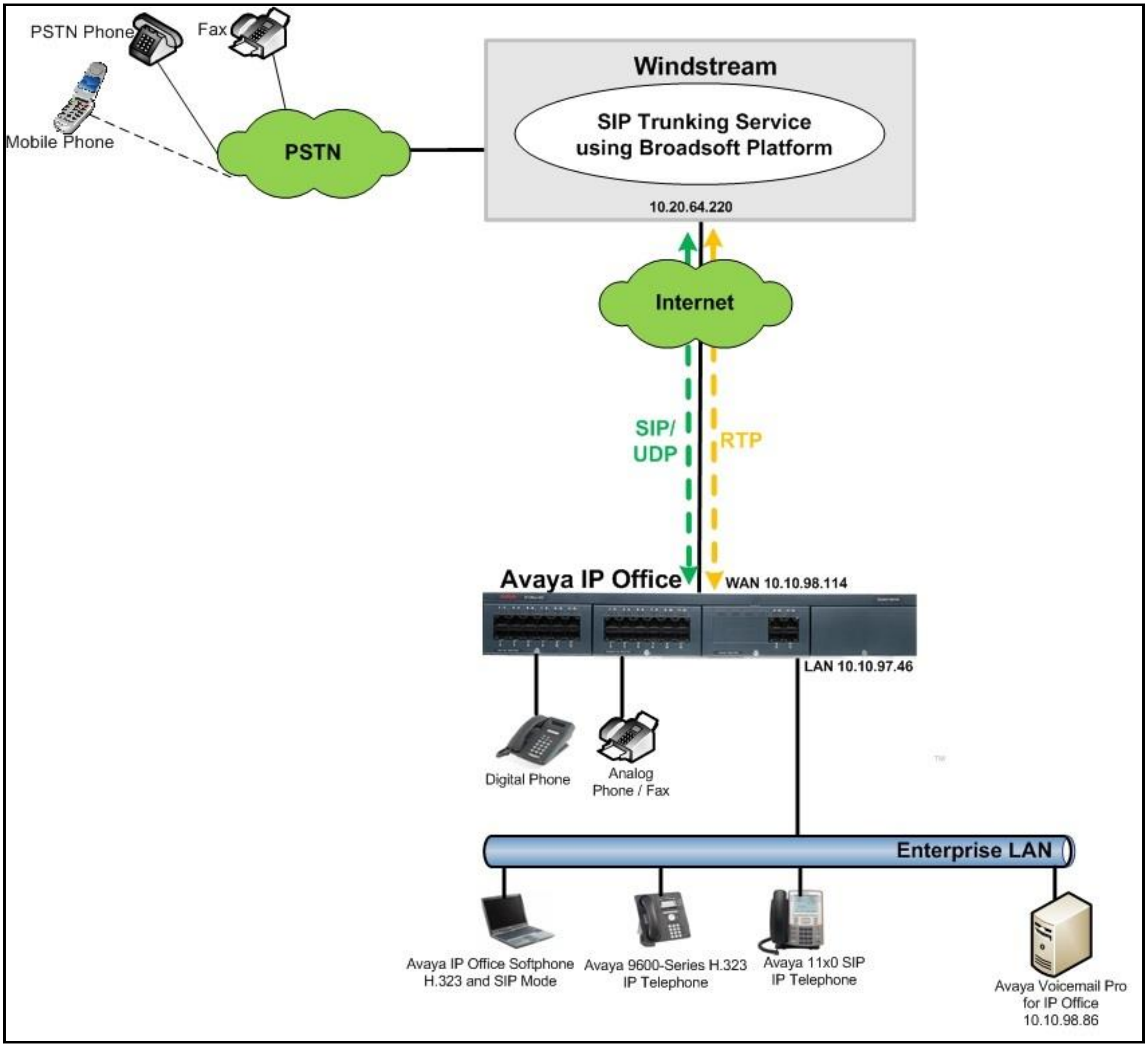


Figure 1: **Avaya IP Telephony Network Connecting to Windstream SIP Trunking Service**

For the compliance testing, Windstream provided the service provider public SIP domain as its Central Office (CO) IP address 10.20.64.220 and the enterprise public SIP domain as IP Office WAN IP address 10.10.98.114. These public SIP domains will be used for the public SIP traffic between IP Office and Windstream. The transport protocol between IP Office and Windstream across the public network is UDP.

For outgoing calls, IP Office sent 11 digits in the destination headers, e.g. “Request-URI” and “To”, and sent 10 digits in the source headers e.g. “From”, “Contact”, and “P-Asserted-Identity”. For incoming calls, Windstream sent 10 digits in both destination and source headers.

In an actual customer configuration, the enterprise site may also include additional network components between service provider and 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 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:

Avaya Telephony Components	
Equipment/Software	**Release/Version**
Avaya IP Office 500v2	8.1 (69)
Avaya IP Office DIG DCP*16 V2	8.1 (69)
Avaya IP Office Ext Card Phone 8	8.1
Avaya IP Office Manager	10.1 (69)
Avaya Voicemail Pro for IP Office	8.1.9203.0
Avaya 9640 IP Telephone (H.323)	Avaya one-X® Deskphone Edition 6.0.1
Avaya 11x0 IP Telephone (SIP)	SIP11x0e04.03.12.00
Avaya IP Office Softphone	3.2.3.48 67009
Avaya Digital Telephones (1408D)	N/A
Avaya Symphony 2000 Analog Telephone	N/A

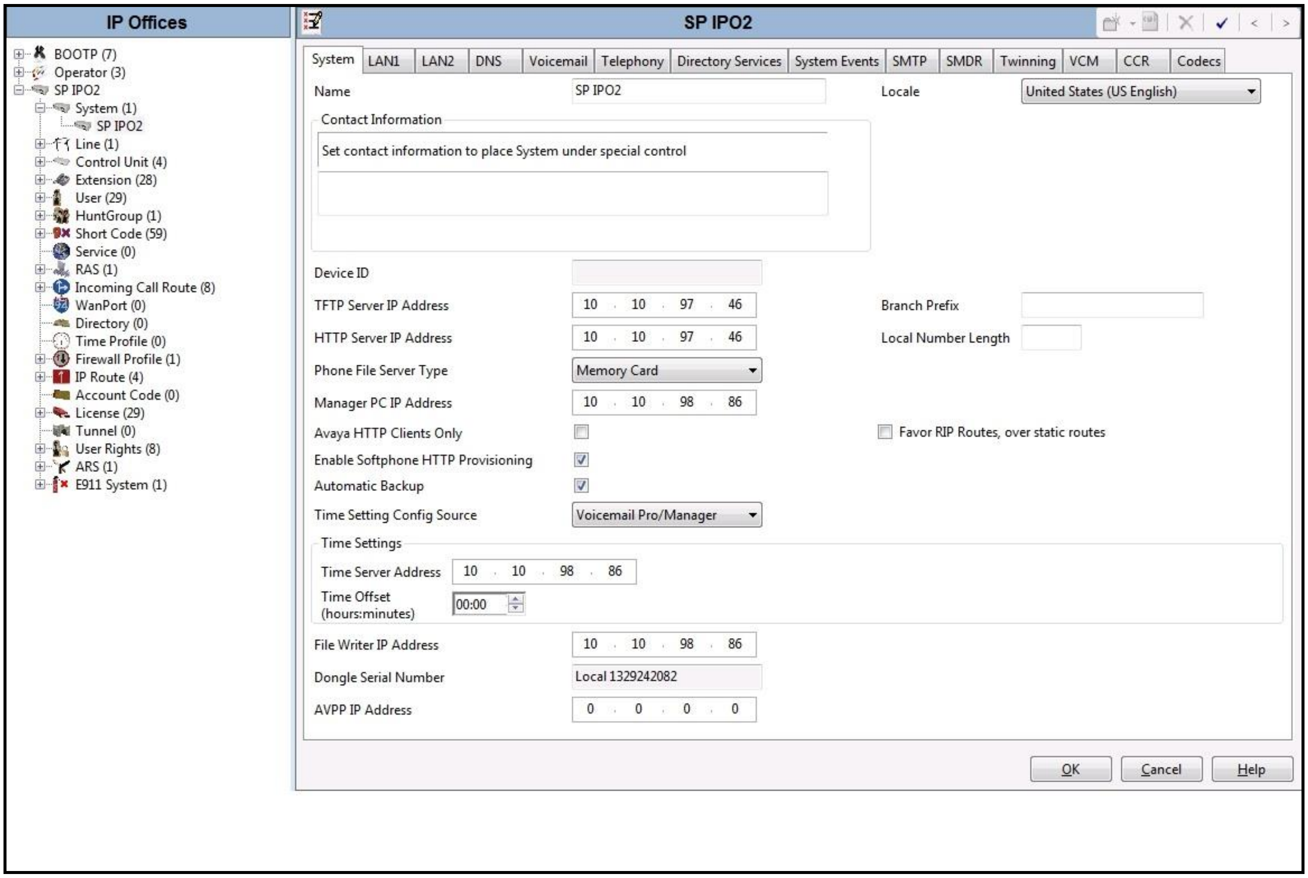
Windstream SIP Trunking Service Components	
Equipment/Software	**Release/Version**
Broadsoft	17sp4
Acme Packet Net-Net 4250 Session Border Controller	6.2.0 patch 3

Testing was performed with IP Office 500v2 R8.1, but it also applies to IP Office Server Edition R8.1. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R8.1 to support analog or digital endpoints or trunks.

5. Configure IP Office

This section describes IP Office configuration required to interwork with Windstream. It was configured through Avaya IP Office Manager (IP Office Manager) which is a PC application. On the

PC, select **Start → Programs → IP Office → Manager** to launch IP Office Manager. Navigate to **File → Open Configuration**, select a proper IP Office system from the pop-up window and log in with the appropriate credentials. A management window will appear as shown below. The appearance of IP Office Manager can be customized using the **View** menu (not shown). In the screenshots presented in this section, the **View** menu was configured to show Navigation Pane on the left side and Details Pane on the right side. These panes will be referenced throughout these Application Notes.

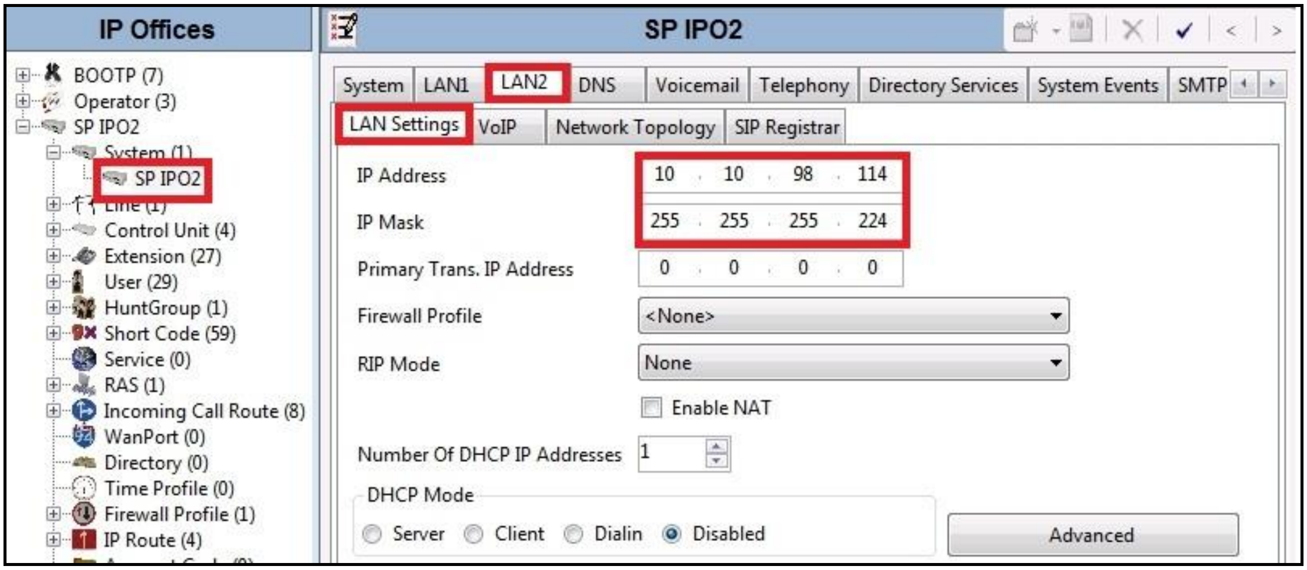


These Application Notes assume the basic installation and configuration have already been completed and are not discussed here. For further information on IP Office, please consult references in **Section 9**.

5.1 LAN

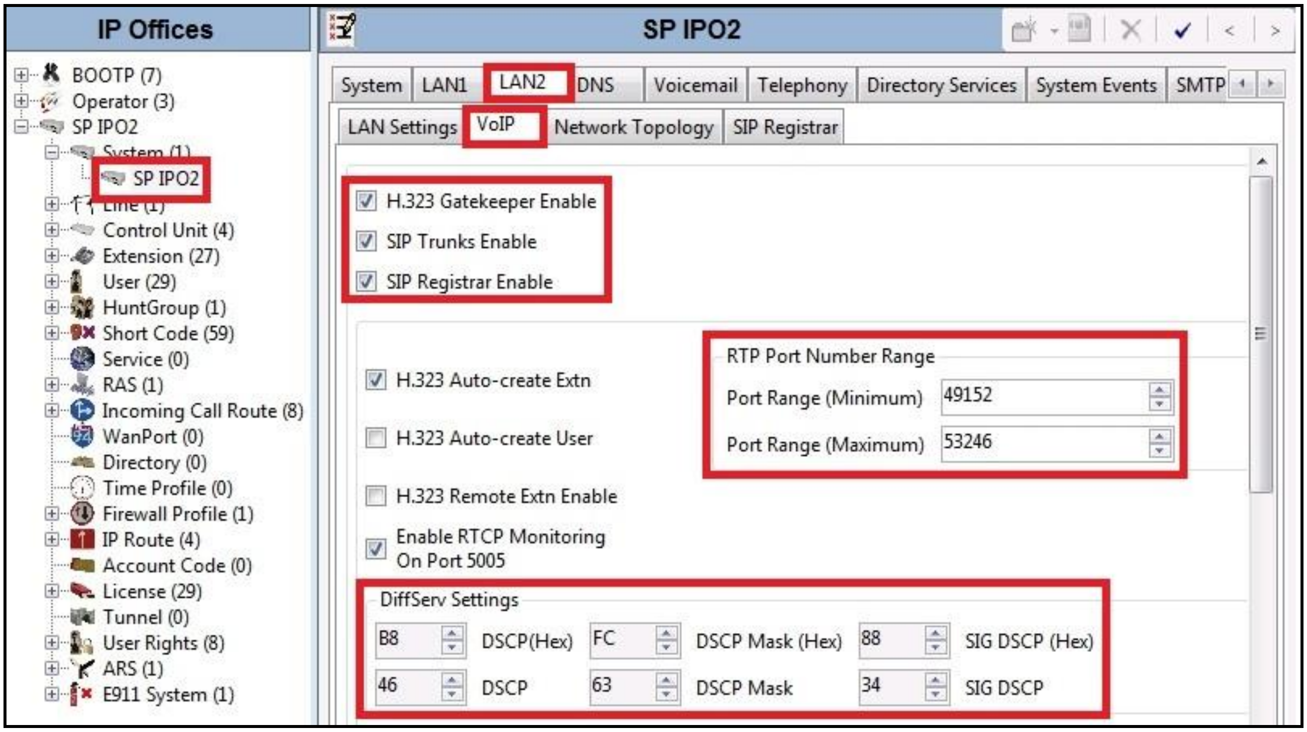
In the sample configuration, IP Office was configured with system name **SP IPO2** and the WAN port (**LAN2**) was used to connect to Windstream network via the internet. To access the **LAN2** settings, navigate to **System (1) → SP IPO2** in the Navigation Pane then in the Details Pane navigate to **LAN2→ LAN Settings** tab. The LAN2 settings for the compliance testing are shown in the screenshot below with following configurations.

- Set the **IP Address** field to the WAN IP address, e.g. 10.10.98.114.
- Set the **IP Mask** field to the subnet mask of the public network, e.g. 255.255.255.224.
- All other parameters should be set according to customer requirements.
- Click OK to commit (not shown) then press Ctrl + S to save.



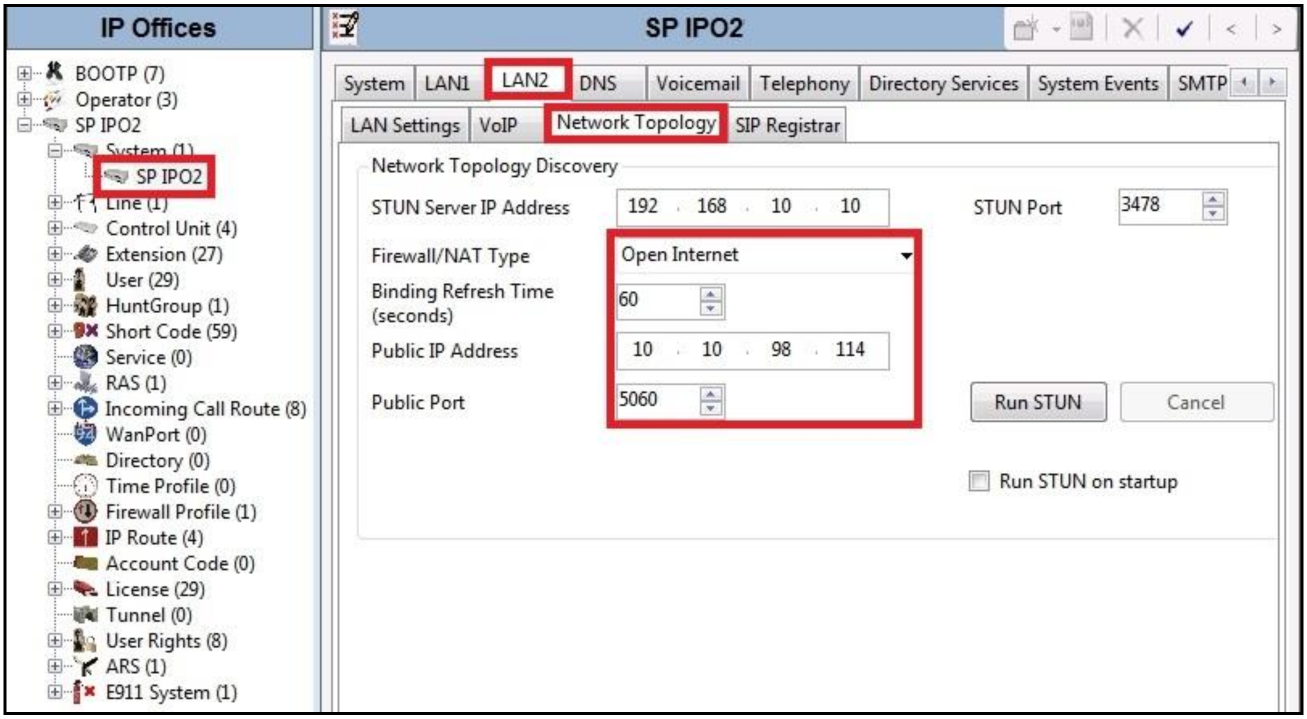
On **VoIP** tab as shown in the screenshot below, configure with following settings.

- Check **H323 Gatekeeper Enable** to allow Avaya IP Telephones/Softphones using H.323 protocol to register.
- Check **SIP Trunks Enable** to enable the configuration of SIP Trunk connecting to Windstream.
- Check **SIP Registrar Enable** to allow Avaya IP Telephones/Softphones to register using SIP protocol.
- Verify **RTP Port Number Range** settings for a specific range for RTP. The **Port Range (Minimum)** and **Port Range (Maximum)** values were kept as default.
- Verify **DiffServ Settings** were kept as default for Differentiated Services Code Point (DSCP) parameter in IP packet headers to support Quality of Services policies for both signaling and media. **DSCP** and **SIG DSCP** fields are the values defined for media and signaling appropriately.
- All other parameters should be set according to customer requirements.
- Click OK to commit (not shown) then press Ctrl + S to save.



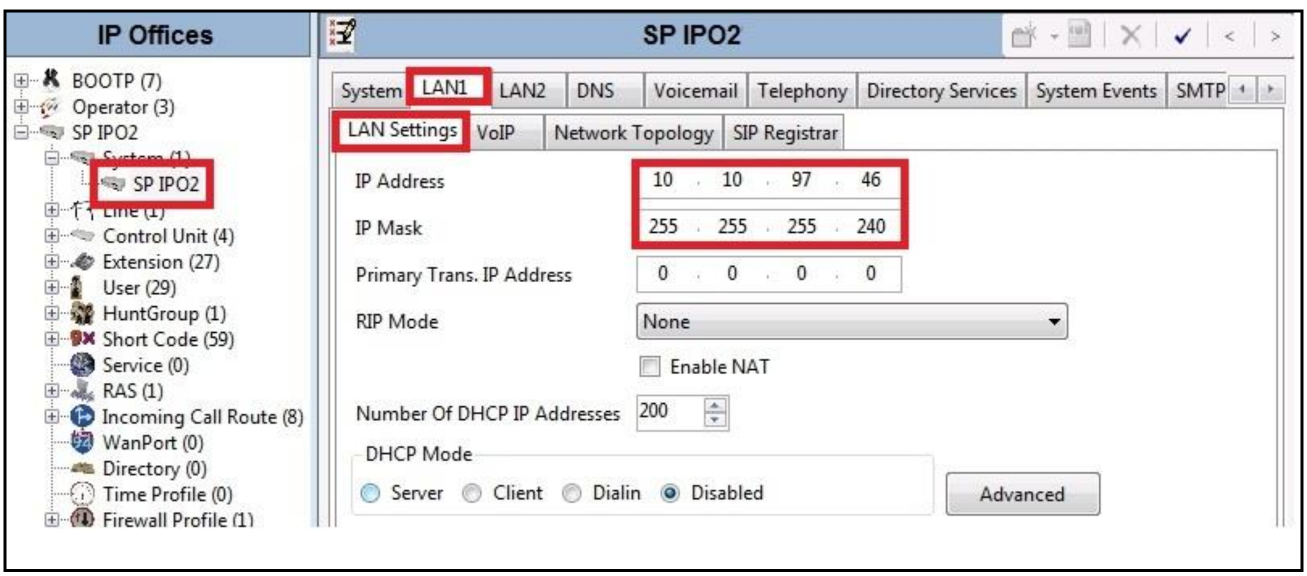
Under **Network Topology** tab in the Details Pane, configure the following parameters:

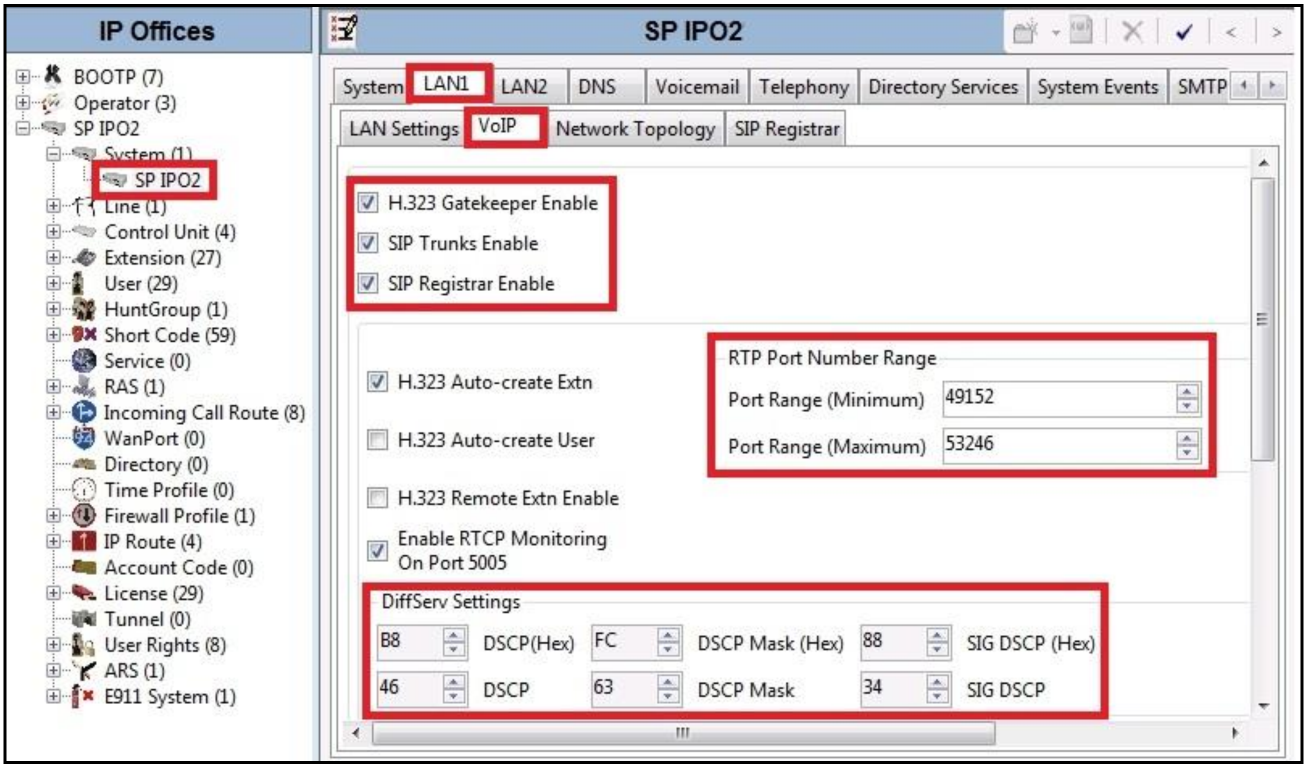
- Select **Firewall/NAT Type** from the pull-down menu that matches the network configuration. In the compliance testing, it was set to **Open Internet**. With this configuration, even the default STUN settings were populated but they will not be used.
- Set **Binding Refresh Time (seconds)** to **60**. This value was used as one input to determine the frequency that IP Office will send OPTIONS heartbeat to service provider.
- Set **Public IP Address** to the IP Office WAN IP address e.g., **10.10.98.114**.
- Set **Public Port** to default SIP port i.e., **5060**.
- All other parameters should be set according to customer requirements.
- Click OK to commit (not shown) then press Ctrl + S to save.

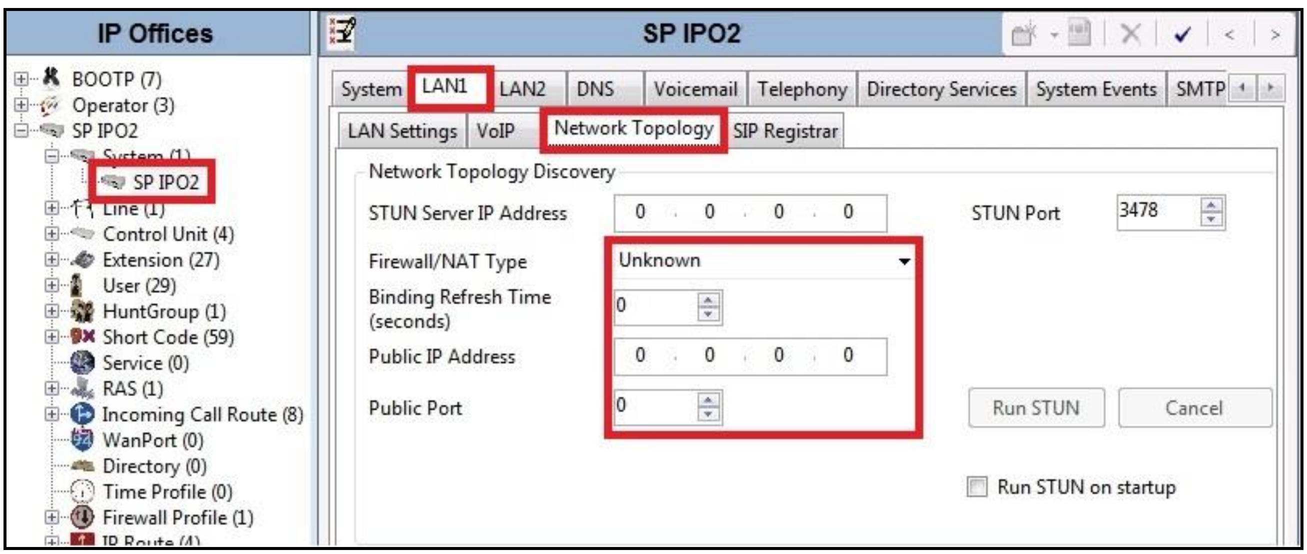


In the compliance testing, IP Office used **LAN1** interface to connect to the enterprise networks. The **LAN1** settings were similarly configured with the following exceptions of **LAN Settings → IP Address** field was set to the private IP address **10.10.97.46** and **Network Topology → Firewall/ NAT Type** field was set to **Unknown** and **Binding Refresh Time (seconds)**, **Public IP Address** and **Public Port** fields were kept disabled as default.

Following screenshots show LAN1 settings for **LAN Settings**, **VoIP** and **Network Topology** tabs.



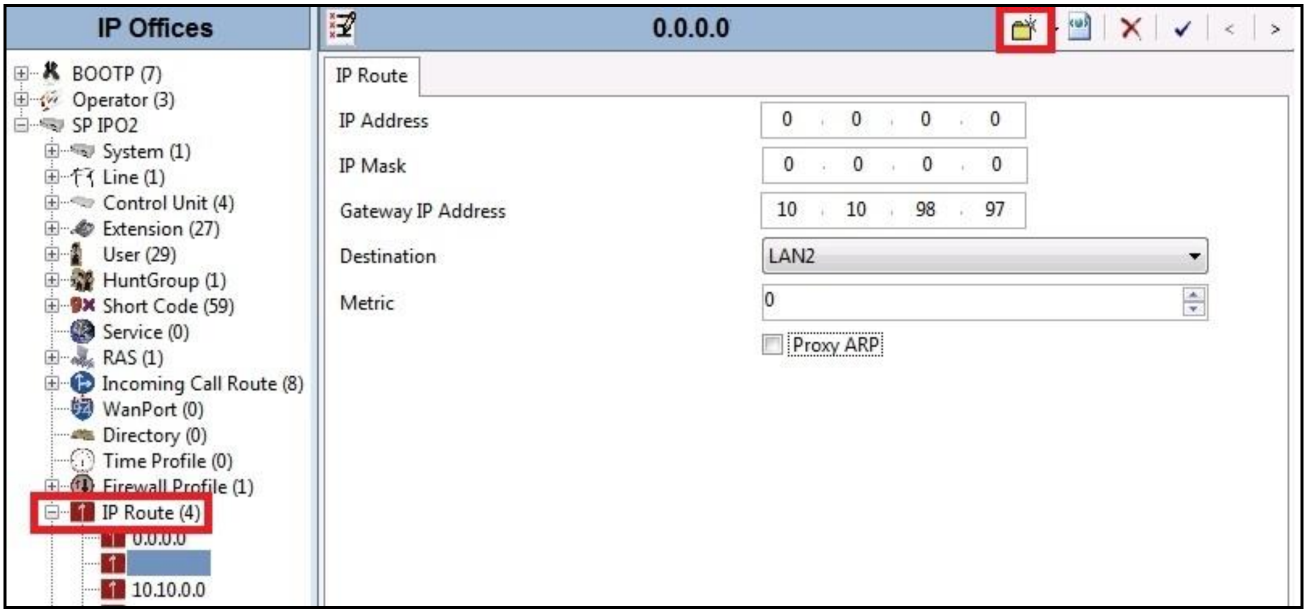




5.2 IP Route

IP Route settings include a public route on LAN2 (WAN) connecting to Windstream and private route on LAN1 connecting to the private enterprise networks.

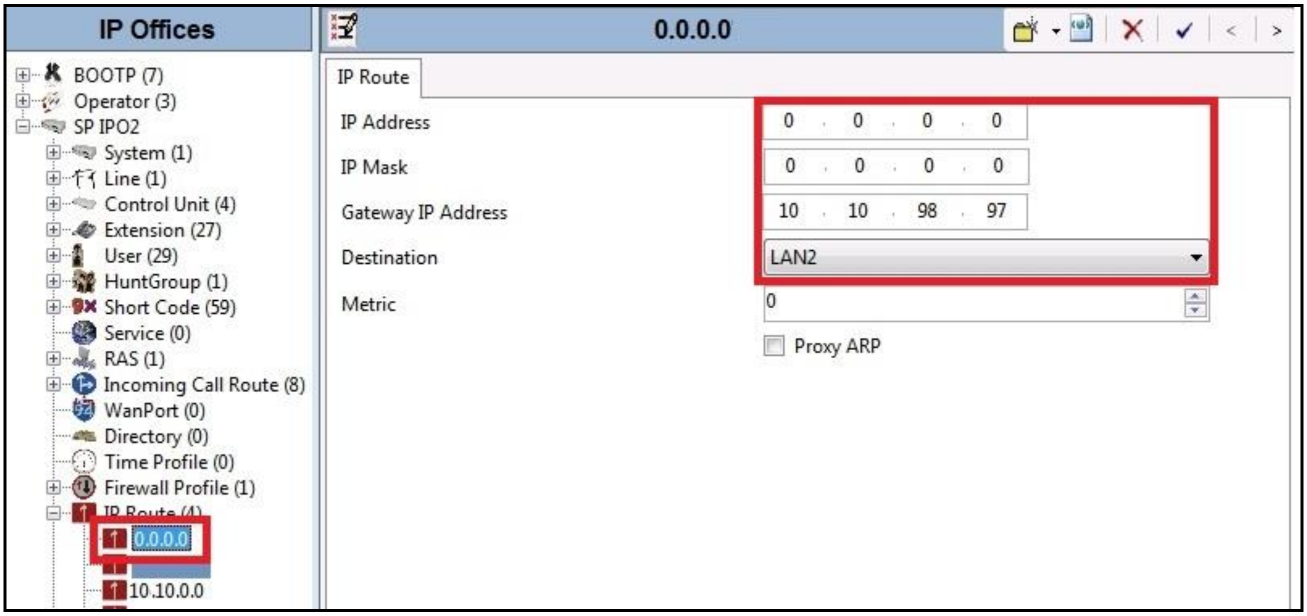
To create IP Route, select **IP Route** in the **Navigation Pane** then click “Create a New Record” icon as shown in the screenshot below.



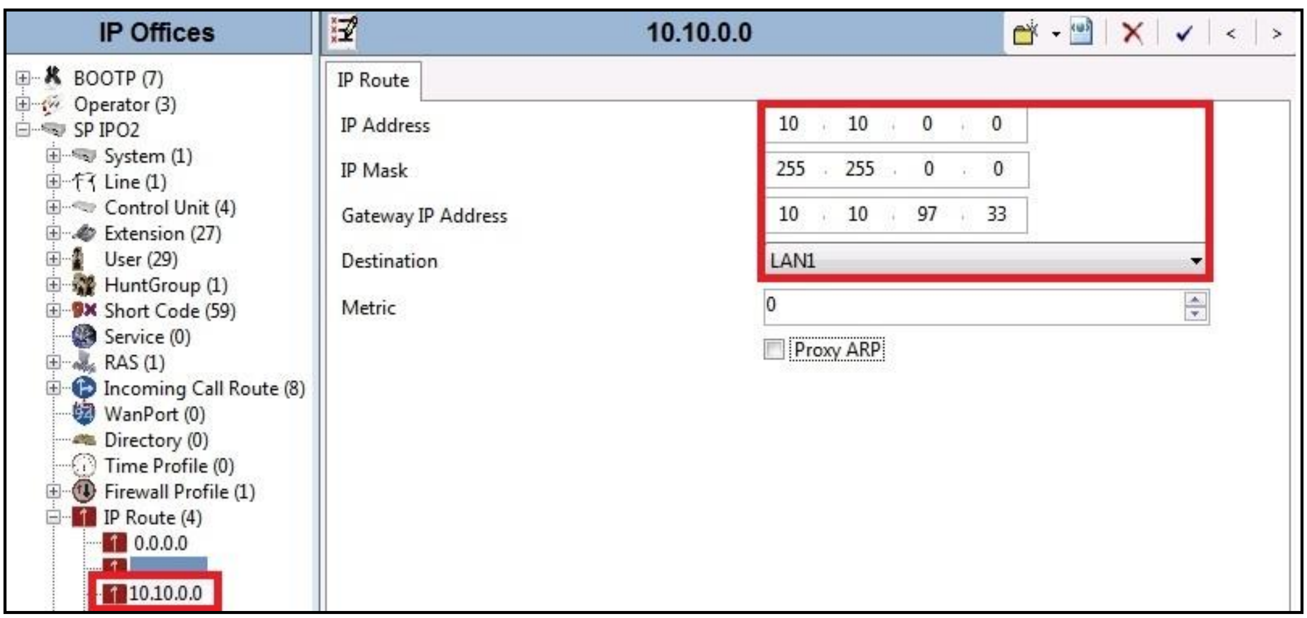
The IP Route was configured using following settings.

- Set **IP Address** field to the address of destination network.
- Set **IP Mask** field to the subnet mask of destination network.
- Set **Gateway IP Address** field to the IP address of enterprise gateway that routes traffic to destination network.
- Set **Destination** field to **LAN2** for the public route and **LAN1** for the private route.
- All other parameters should be set according to customer requirements.
- Click OK to commit (not shown) then press Ctrl + S to save.

The public route was created on LAN2 is shown below. LAN2 was planned to be used as default network interface of IP Office. It was used for SIP and RTP traffics to Windstream. Therefore, it was assigned for the default network address 0.0.0.0 and subnet mask 0.0.0.0. The default gateway was set to IP address 10.10.98.97 which is an internal gateway on the enterprise network that connects to LAN2.



The following screenshot shows the private route that was created on LAN1. LAN1 was planned to be used for SIP and RTP traffic to IP telephones over the private enterprise network. Therefore, it was assigned to the network address 10.10.0.0 and subnet mask 255.255.0.0. The default gateway was set to IP address 10.10.97.33 which is an internal gateway on the enterprise network that connects to LAN1.

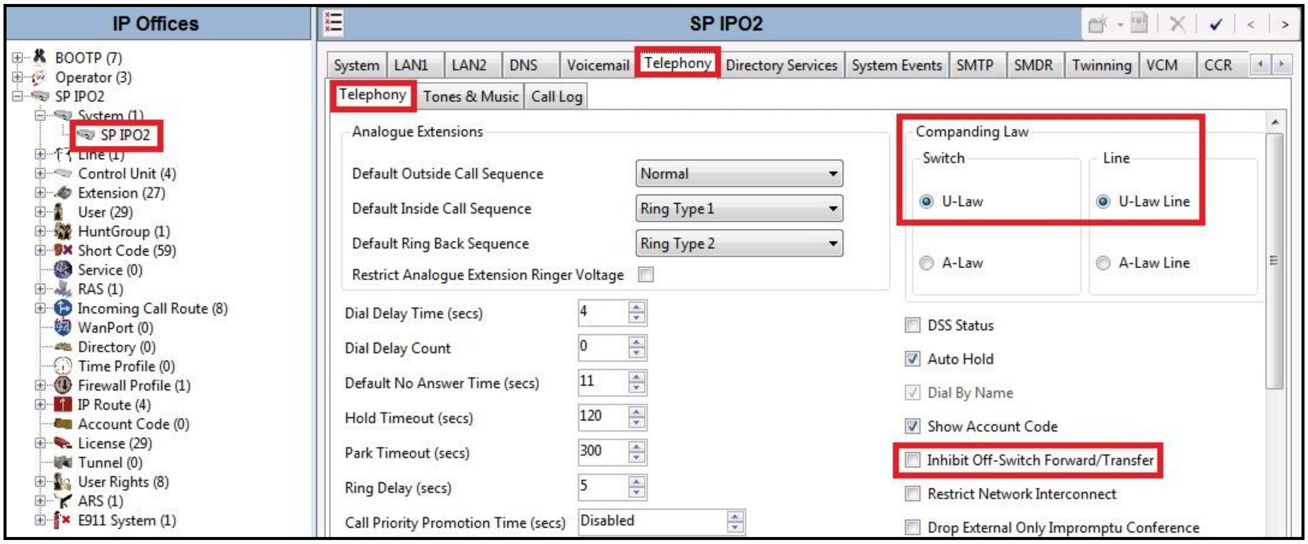


5.3 System Telephony and Codecs

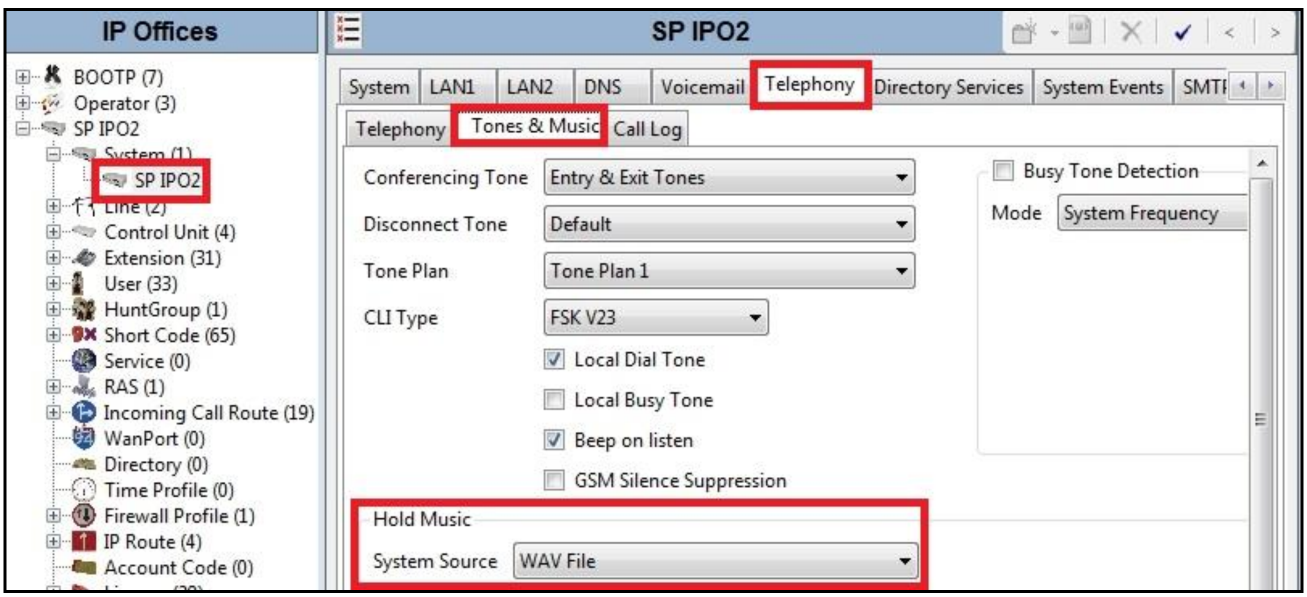
Navigate to **System (1) → SP IPO2** in the Navigation Pane and then select **Telephony → Telephony** tab in the Details Pane.

The System Telephony settings are shown in the screenshot below with following configurations.

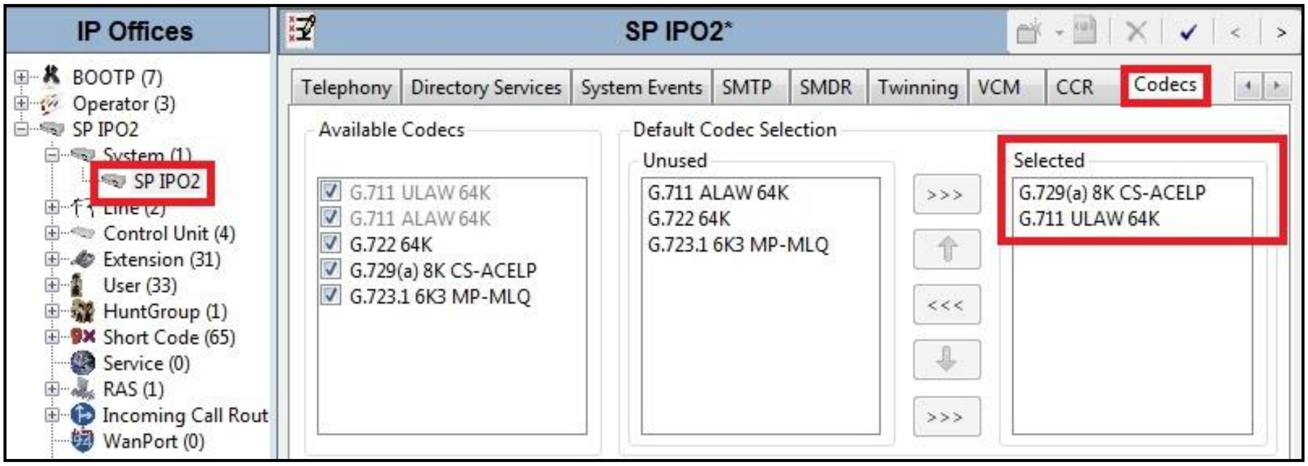
- Choose **Companding Law** for the enterprise geographic location. For North America, **U-LAW** is used for both **Switch** and **Line**.
- Uncheck **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to PSTN via the service provider SIP trunk.
- Click OK to commit (not shown) then press Ctrl + S to save.



Under **Tones & Music** tab as shown below, **Hold Music** was configured with **System Source** to use **WAV File** which is an uploaded medium to provide Music on Hold on the SIP Trunk.



For **Codecs** settings, navigate to the **System (1) → SP IPO2** in the Navigation Pane and then select **Codecs**. The **Codecs** settings are shown in the screenshot below with G.729 and G.711MU were selected in prioritized order. In the compliance testing, Windstream supported G.729 as the first choice and G.711MU as the second choice for RTP traffic.



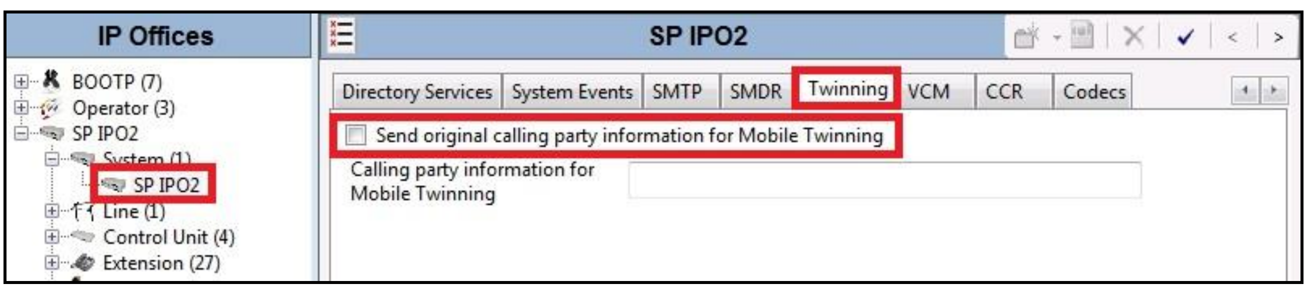
Click OK to commit (not shown) then press Ctrl + S to save.

5.4 Twinning Calling Party

When using twinning, Calling Party Number displayed on twinned phone is controlled by two parameters. The first parameter is **Send original calling party information for Mobile Twinning** box on **System→Twinning** tab. The second parameter is **Send Caller ID** parameter on **SIP Line** form as shown in **Section 5.5**.

For the compliance testing, **Send original calling party information for Mobile Twinning** was unchecked as shown below. This setting allows **Send Caller ID** parameter that was set to **Diversion Header** (see **Section 5.5**) to be used. IP Office will send following in the "From" header:

- On calls from an internal extension to a twinned phone, IP Office sends Calling Party Number of the originating extension.
- On calls from PSTN to a twinned phone, IP Office sends Calling Party Number of the originating PSTN party.



5.5 Administer SIP Line

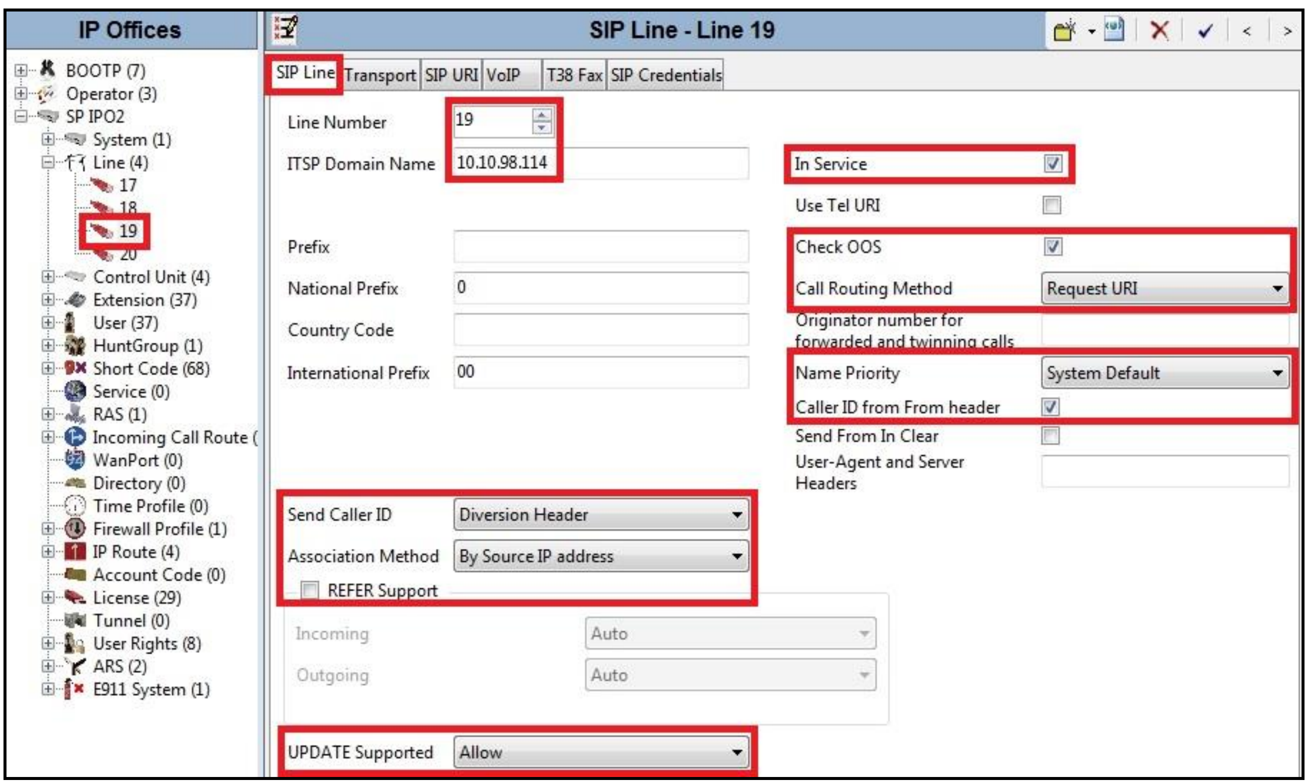
A SIP Line is needed to establish the SIP Trunk between IP Office and Windstream.

To create a SIP Line, navigate to **Line** in the left Navigation Pane and then select **New → SIP Line** (not shown).

5.5.1 Administer SIP Line Settings

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

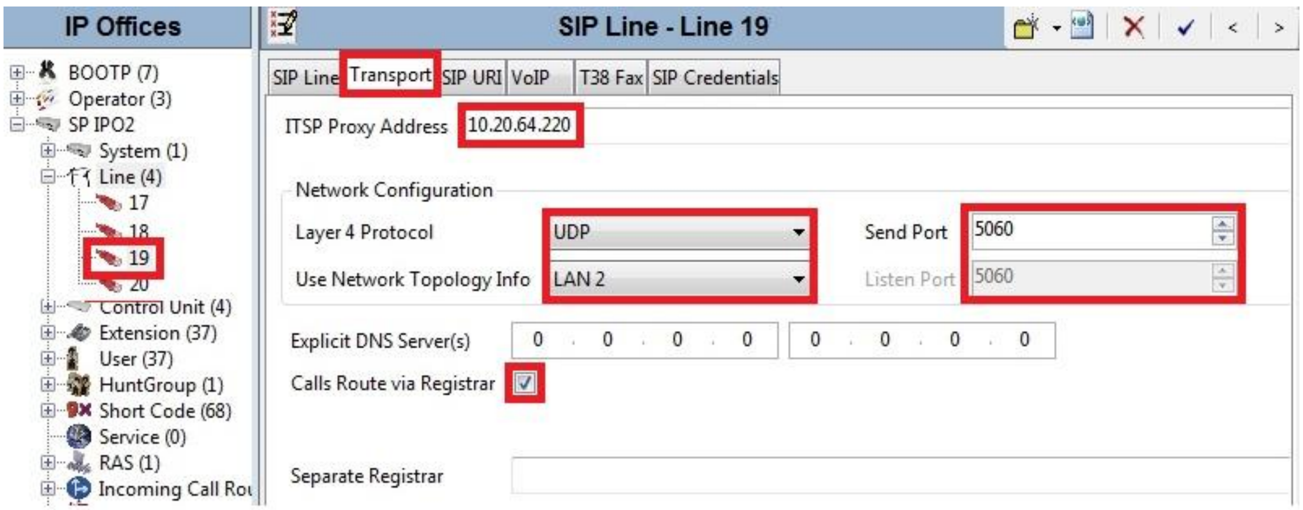
- Set **Line Number** field to an unassigned number, e.g. 19.
- Set **ITSP Domain Name** field to FQDN or IP address that will be used as the enterprise SIP domain so that IP Office will use this domain as the URI-Host of the “From”, “P-Asserted-Identity” or “Diversion” header. In the compliance testing, Windstream required the public IP address of IP Office 10.10.98.114 be used as the URI Host.
- Set **Send Caller ID** field to **Diversion Header**. For the compliance testing, this parameter was used for Caller ID since **Send original calling party information for Mobile Twinning** was unchecked as shown in **Section 5.4**.
- Set **Association Method** field to **By Source IP address**. This setting allows IP Office to apply the configuration for public SIP Trunk to incoming and outgoing calls from/ to Windstream.
- Uncheck **REFER Support** field to allow IP Office to use re-INVITE to for call transfer, for more detail information, refer to **Section 2.2**, observation **8**.
- Set **UPDATE Supported** field to **Allow** to enable the use of UPDATE method.
- Check **In Service** box.
- Check **Check OOS** box. With this option selected, IP Office will periodically send OPTIONS/ heartbeat to check for the status of the SIP Trunk.
- Set **Call Routing Method** field to **Request URI**.
- Set **Name Priority** field to **System Default**.
- Check **Call ID from From header** box.
- Default values may be used for all other parameters.



5.5.2 Administer Transport Settings

Select **Transport** tab then configure following parameters as shown in the screenshot below.

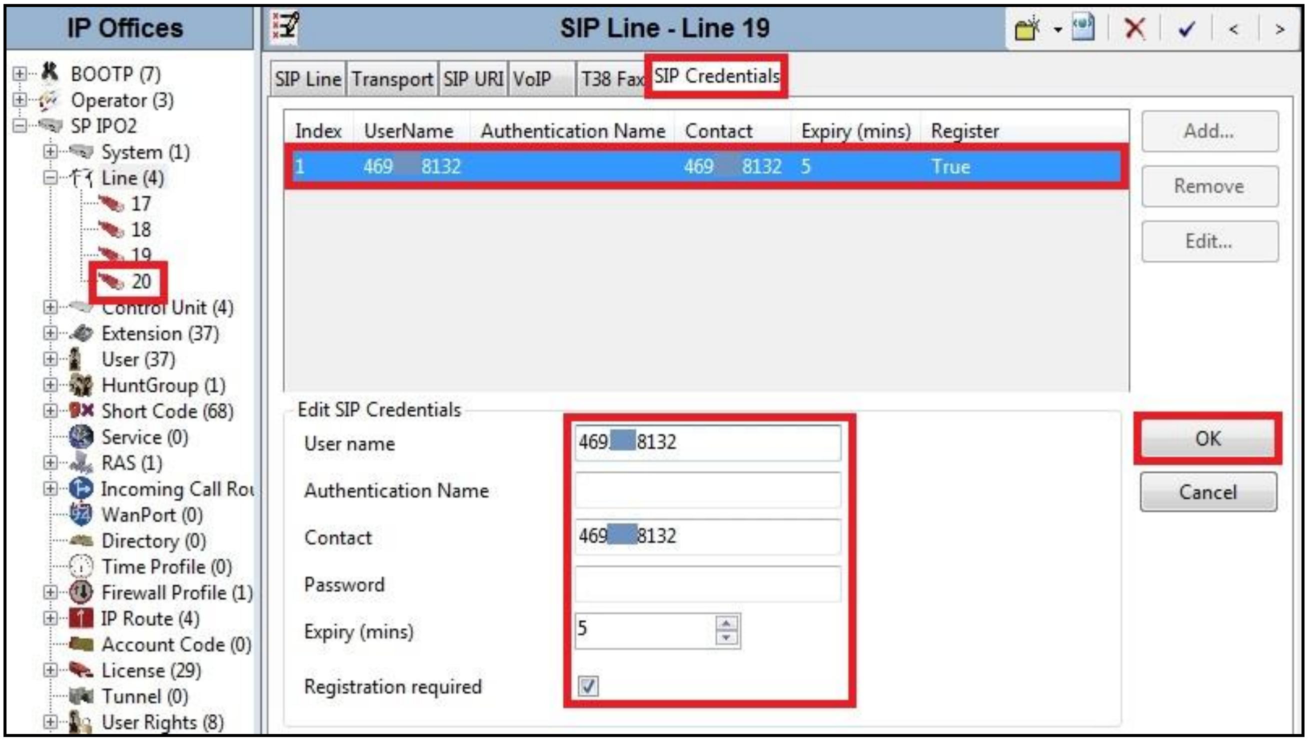
- **ITSP Proxy Address** was set to the Windstream SIP Proxy IP Address. As shown in **Figure 1**, this IP Address is 10.20.64.220.
- In **Network Configuration** area, **UDP** was selected as **Layer 4 Protocol** and **Send Port** was set to port number **5060** which was provided by Windstream.
- **Use Network Topology Info** parameter was set to **LAN 2**. This associates the SIP Line 19 with the parameters configured on **System → LAN2 → Network Topology** tab.
- **Calls Route via Registrar** was checked. In this certification testing, IP Office sent REGISTER to dynamically establish the SIP Trunk. All calls will be routed on the connection set by Registration method.
- Other parameters retain default values in the screenshot below.



5.5.3 Administer SIP Credential Settings

A SIP Credentials entry must be created for Registration and Digest Authentication used by Windstream to authenticate signaling from the enterprise. To create a SIP Credentials entry, first select **SIP Credentials** tab. Click **Add** button and **New Channel** area will appear at the bottom of the pane (not shown). To edit an existing entry, click an entry in the list at the top, and click **Edit…** button. In the bottom of the screenshot, **Edit SIP Credentials** area will be opened. In the example screenshot below, a previously configured entry is edited. The entry was created with the parameters shown below:

- Set **User name** field to the value provided by the service provider, e.g. 469XXX8132.
- Set **Contact** field to the value provided by the service provider, e.g. 469XXX8132.
- Set **Authentication Name** and **Password** fields for Digest Authentication to blank because it was not required by Windstream.
- Set **Expiry (mins)** is set to **5**.
- Check the **Registration required** field option to allow IP Office to send REGISTER to Windstream.



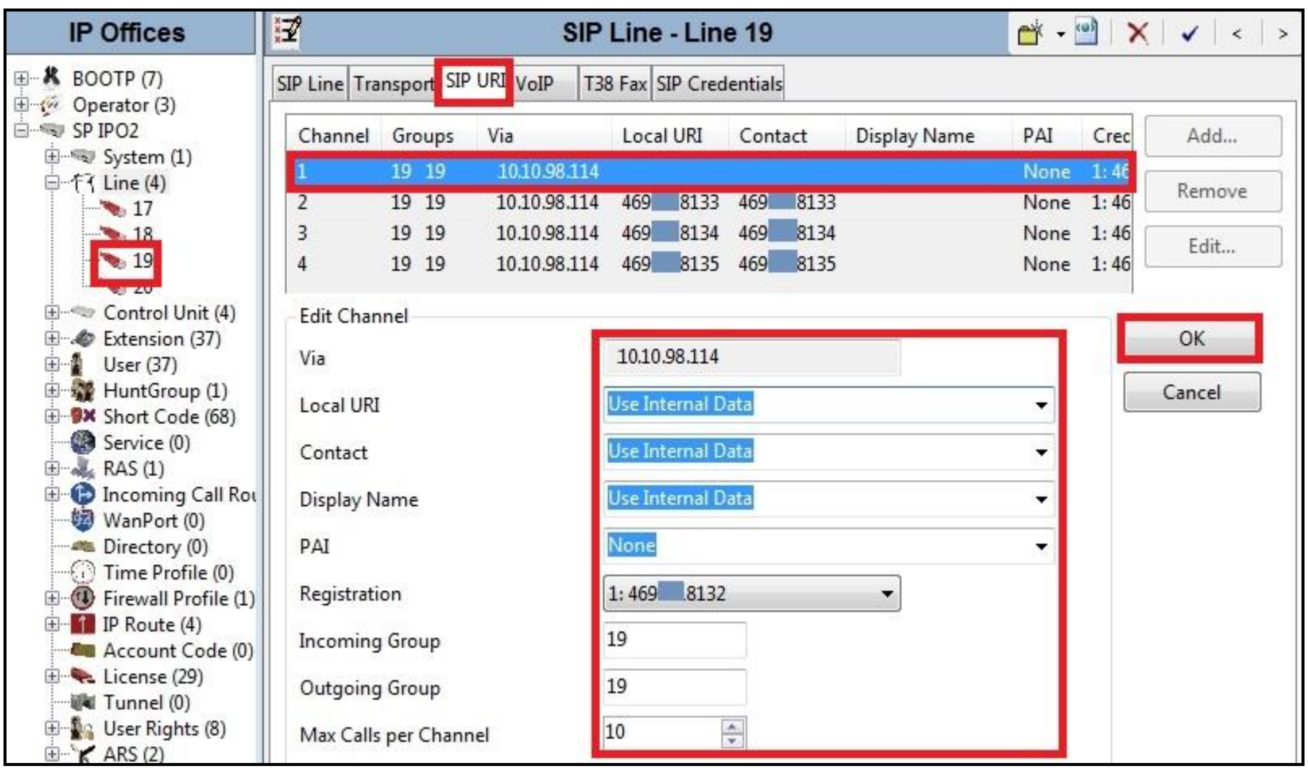
Note: The Expiry was set to an amount that is small enough for IP Office to reduce the downtime on the SIP Trunk when IP Office performed a warm restart. It has been observed that when performing a warm restart on IP Office e.g., to apply a configuration change, IP Office sent REGISTER/ Expires: 0 to de-register then follow by another REGISTER/ Expires: 300 to re-register back. Windstream unexpectedly responded 200OK to REGISTER/ Expires: 300 before 200OK to REGISTER/ Expires: 0. After this event, IP Office assumed the REGISTRATION was successfully complete and the SIP Trunk was now active. However, by sending 200OK in incorrect order, Windstream interpreted that IP Office requested to de-register and the SIP Trunk was now inactive. It resulted all incoming and outgoing calls could not be successfully complete. Until the REGISTRATION expired in 300 seconds, IP Office sent another REGISTER and it helped to clear up the in-active state of the SIP Trunk at service provider side then Windstream started to process incoming and outgoing calls as expected. For detailed information, refer to **Section 2.2**, observation **9**.

5.5.4 Administer SIP URI Settings

A SIP URI entry must be created to match Calling Party Number for incoming calls or to present Calling Party Number for outgoing calls on this line. Select **SIP URI** tab, click **Add** button then **New Channel** area will appear at the bottom of the pane (not shown). To edit an existing entry, click an entry in the list at the top and click **Edit…** button. In the example screenshot below, a previously configured entry is edited.

For the compliance testing, SIP URI entry with **Channel 1** was created for incoming and outgoing calls. Its parameters were shown below:

- Set **Local URI**, **Contact** and **Display Name** fields to **Internal Data**. This setting uses Calling Party Number defined under **SIP** tab of **User** as shown in **Section 5.7** for public SIP calls.
- Set **PAI** field to **None** to disable PAI on the SIP Trunk. For more information, refer to **Section 2.2**, observation **5**.
- For **Registration** field, select the account credentials previously configured on the **SIP Credentials** tab.
- Associate this line with an incoming line group in **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 **Outgoing Group** field. For the compliance testing, a new incoming and outgoing group **19** was defined.
- Set **Max Calls per Channel** field to 10 which is the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

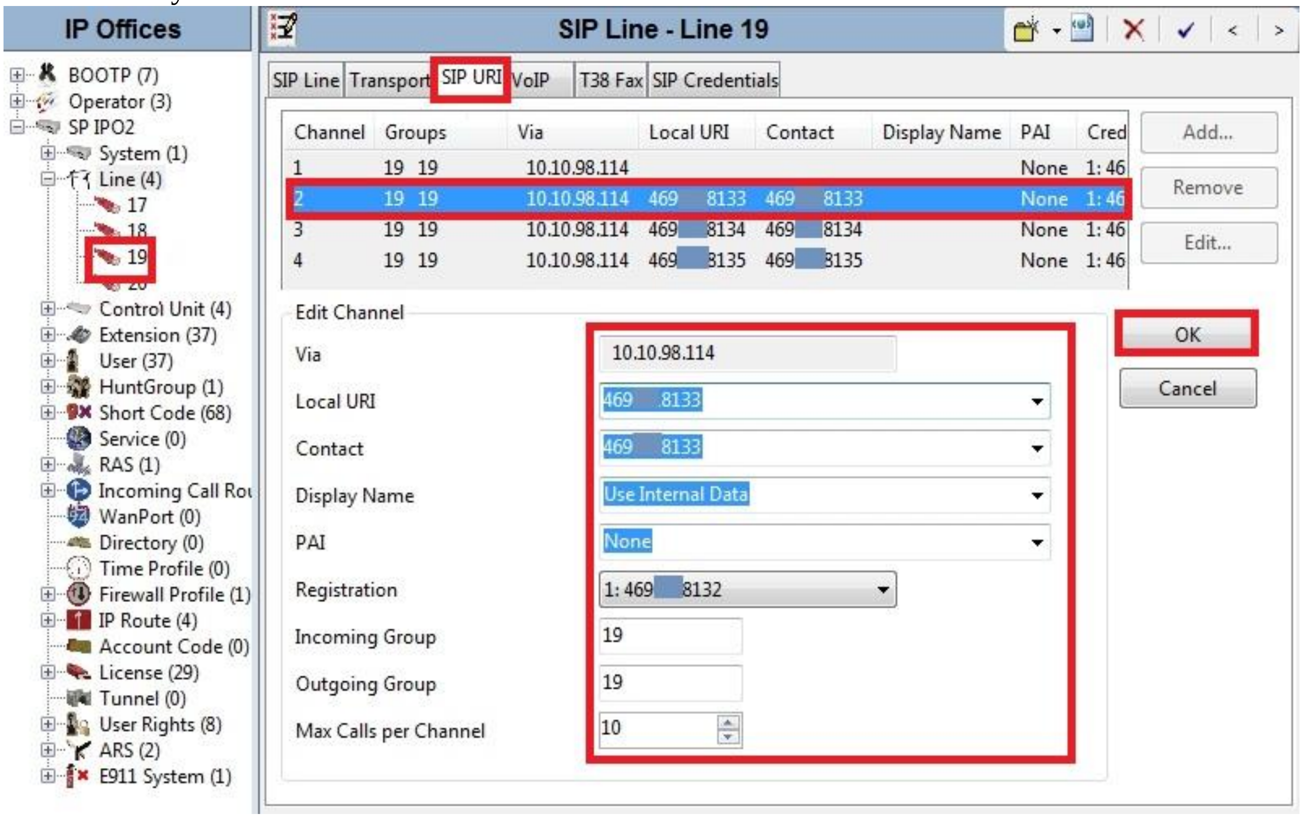


SIP URI entries with **Channel 2**, **Channel 3** and **Channel 4** were similarly created for incoming calls appropriately to pre-define DID numbers **469XXX8133**, **469XXX8134** and **469XXX8135** for accessing to Feature Name Extension 00 (FNE00), Feature Name Extension 33 (FNE33), and VoiceMail. The Short Codes for FNE00 and FNE33 were defined in **Section** Error! Reference source not found. to provide Dial Tone and Mobile Callback for mobility extension.

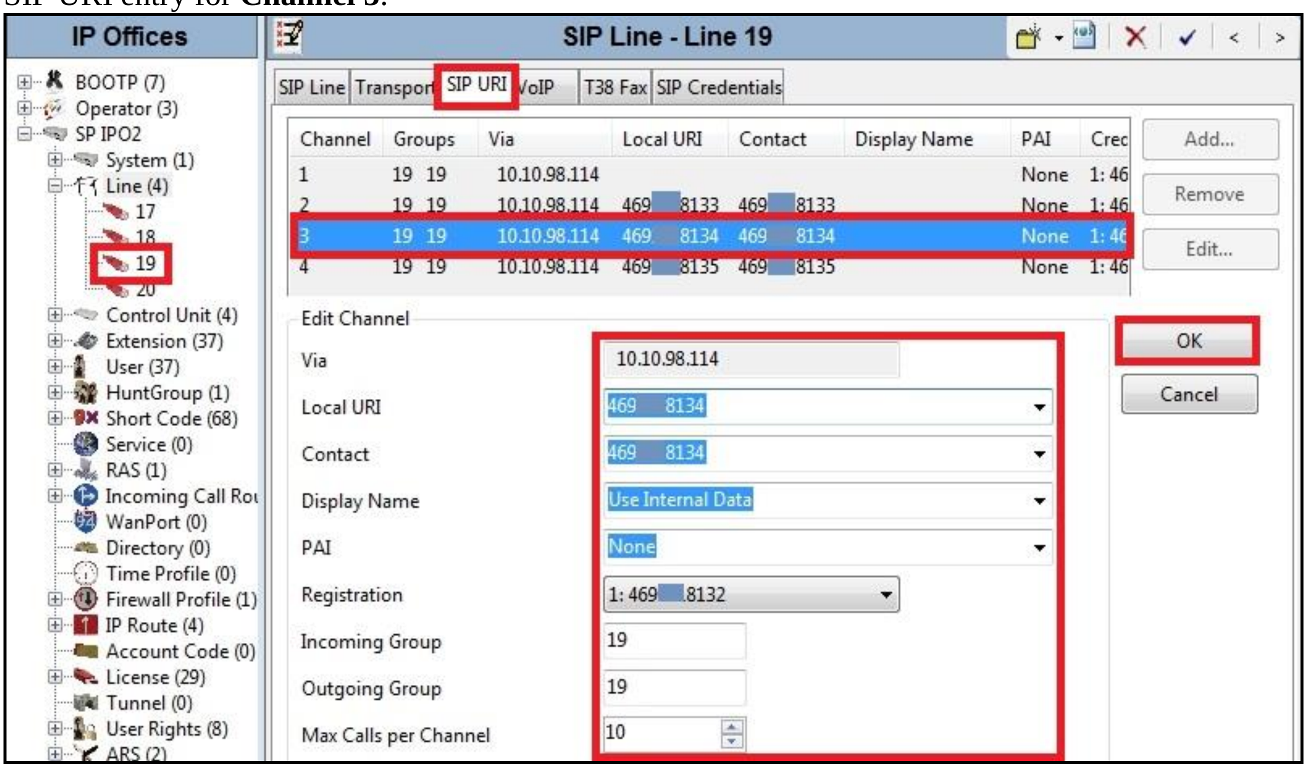
The SIP URI **Channel 2**, **Channel 3** and **Channel 4** as shown in the screenshot below, were configured with following parameters.

- Set **Local URI** and **Contact** fields to pre-define DID number **469XXX8133**, **469XXX8134** and **469XXX8135** appropriately for **Channel 2**, **Channel 3** and **Channel 4**.
- Associate **Incoming Group** and **Outgoing Group** to SIP Line 19.
- Set **Max Calls per Channel** field to **10**.
- Other parameters retain default values.
- Click OK to commit.

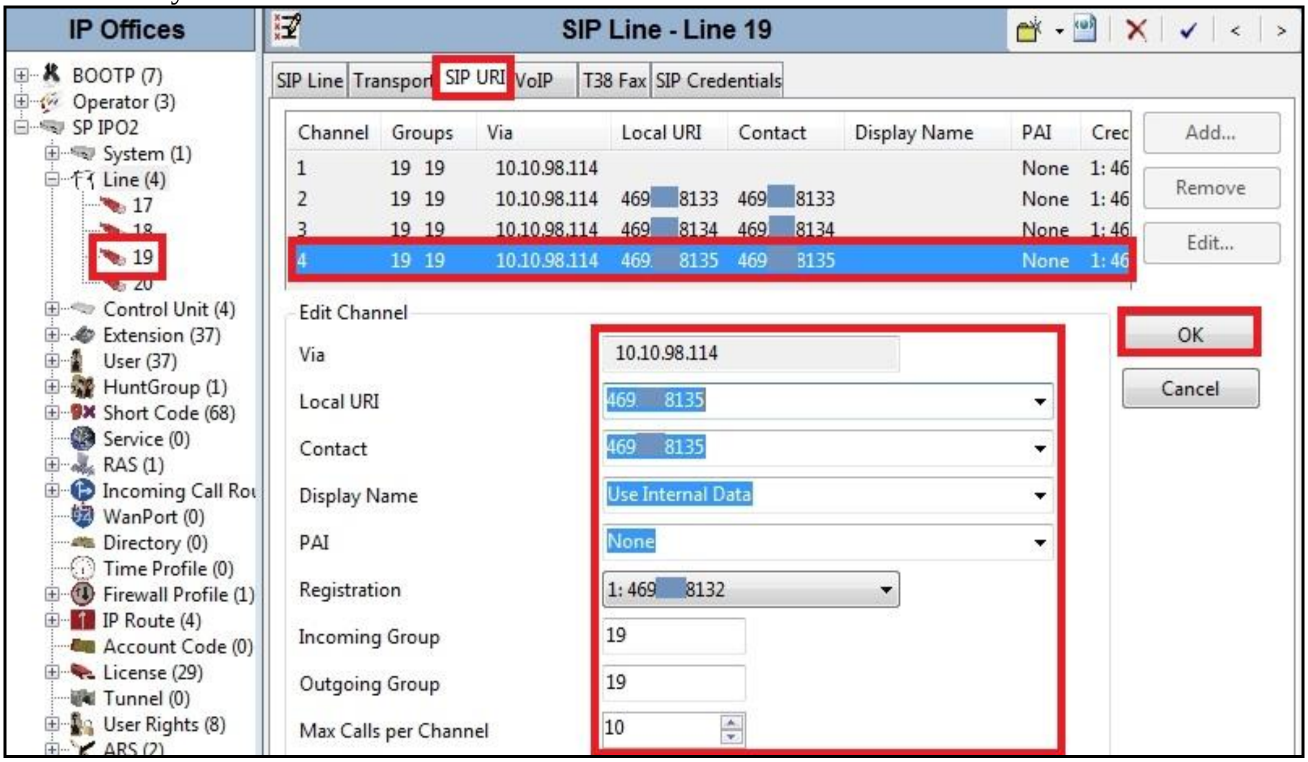
SIP URI entry for **Channel 2**:



SIP URI entry for **Channel 3**:



SIP URI entry for **Channel 4**:

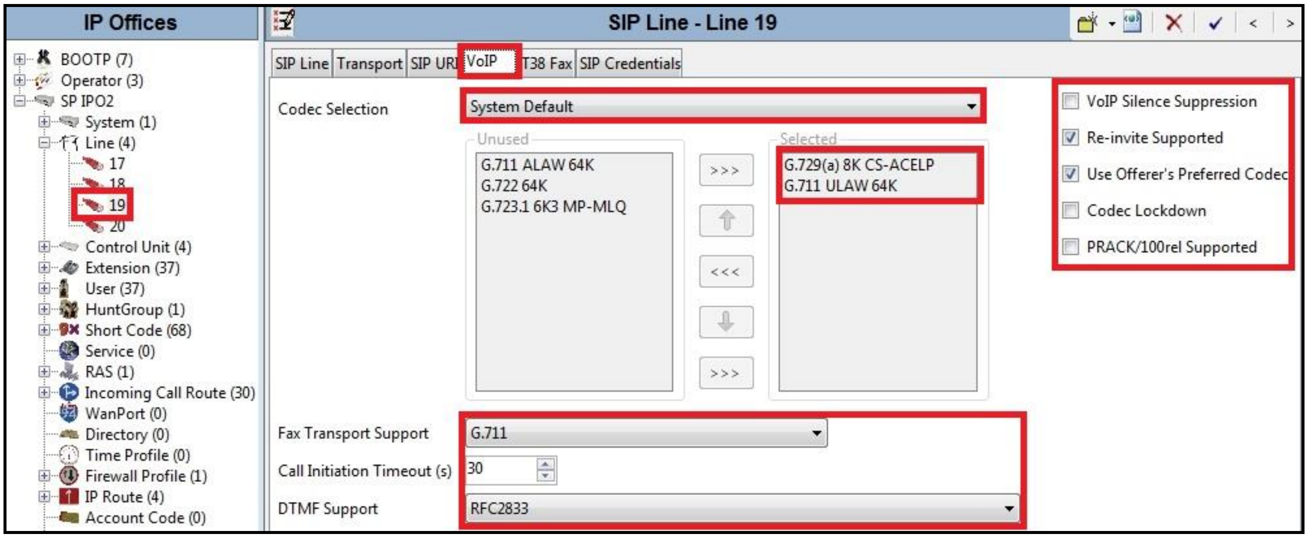


Click OK to commit (not shown) then press Ctrl + S to save.

5.5.5 Administer VoIP Settings

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

- **Codec Selection** can be selected by choosing **System Default** from the pull-down menu to use the System Codecs as defined in **Section 5.3**. The codec order was configured as **G.729(a) 8K CS-ACELP** and **G.711 ULAW 64K** which were supported by Windstream. IP Office includes these codes in the right prioritized order in the Session Description Protocol (SDP) offer or answer defined for RTP traffic.
- Set **Fax Transport Support** to **G.711** from the pull-down menu.
- Set **Call Initiation Timeout (s)** to **30** seconds to allow a long enough duration for a public call to be established over the SIP Trunk.
- Set **DTMF Support** to **RFC2833** from the pull-down menu. This directs IP Office to send out-of-band DTMF tones using RTP events per RFC 2833.
- Uncheck **VoIP Silence Suppression** box. By unchecking **VoIP Silence Suppression** box, calls can be established with the G.729 codec but without silence suppression.
- Check **Re-invite Supported** box.
- Check **Use Offerer's Preferred Codec** box.
- Uncheck **Codec Lockdown** box.
- Uncheck **PRACK/100rel** because it was not supported by Windstream.
- Default values may be used for all other parameters.
- Click OK to commit (not shown) then press Ctrl + S to save.

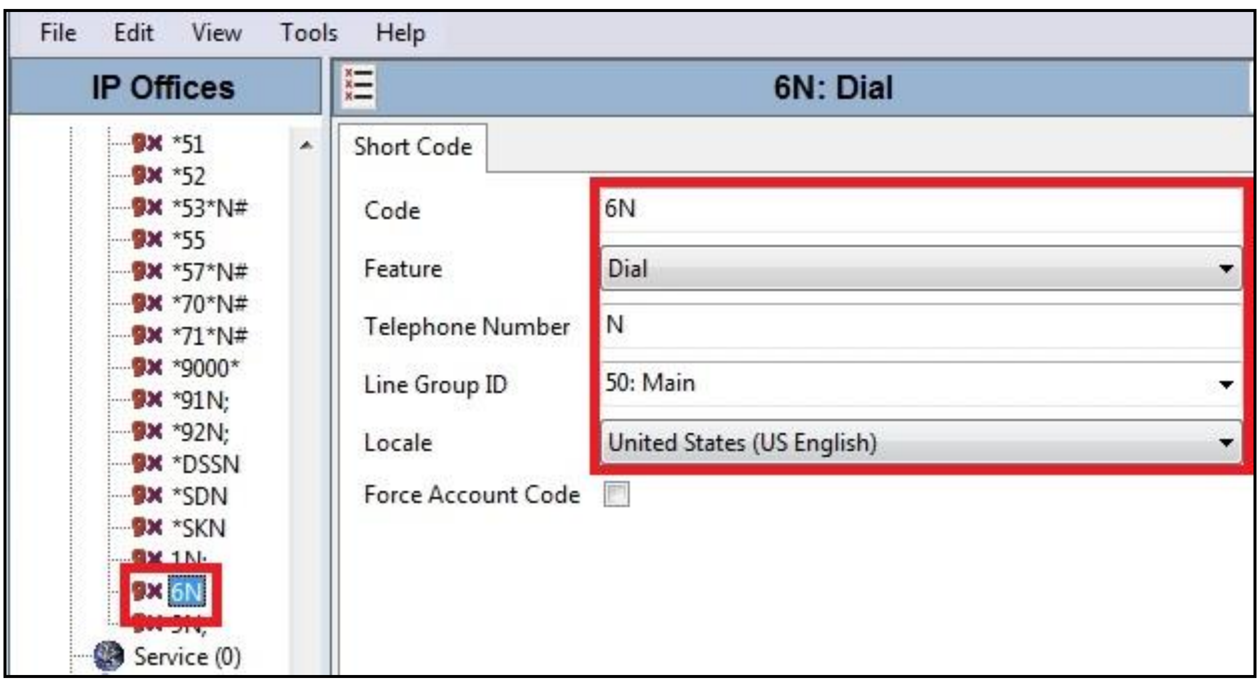


5.6 Short Code

Define a short code to route outgoing traffic to the SIP Line. To create a short code, select **Short Code** in the left Navigation Pane and then right-click and select **New** (not shown). On **Short Code** tab in the Details Pane, configure parameters for the new short code to be created. The screenshot below shows the details of the previously administered short code "98N;" used in the test configuration.

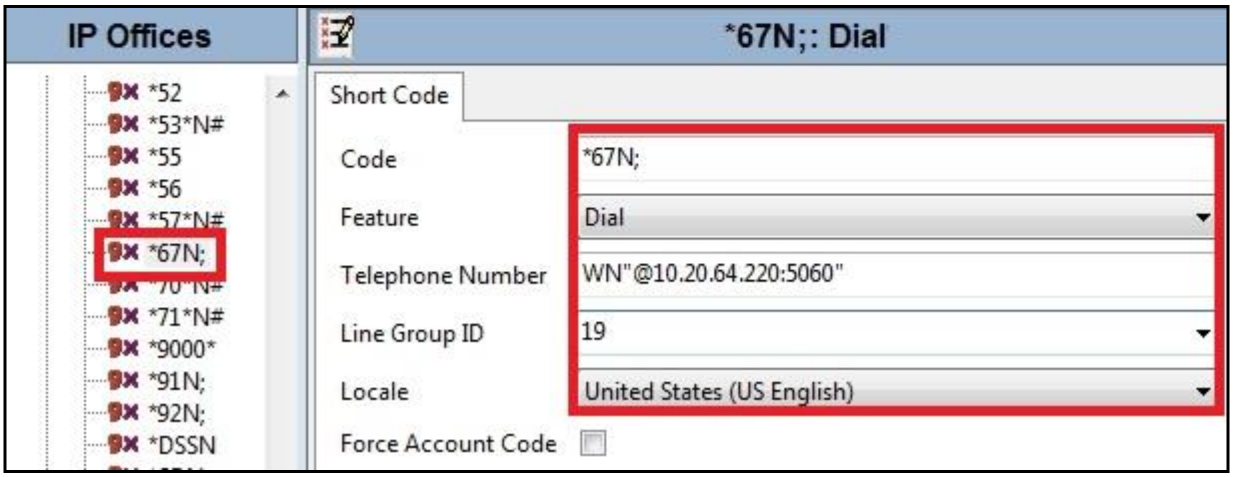
- In **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **98N;**, This short code will be invoked when the user dials 98 followed by any number.
- Set **Feature** field to **Dial**. This is the feature that the short code will invoke.
- Set **Telephone Number** field to **N"@10.20.64.220:5060"**. This field was used to construct the "Request URI" and "To" headers for outgoing calls. The value **N** represents the number dialed by the user. The host part following by "@" is the domain of service provider networks.
- Set **Line Group ID** field to the outgoing line group number **19** defined on **SIP URI** tab for **SIP Line** in **Section 5.5.1**. This short code uses this line group when placing outgoing calls.
- Set **Locale** to **United State (US English)**.

The **98N;** short codes illustrated above do not provide a mean 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 screenshot, the short code 6N is illustrated for accessing to ARS. When the 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 5.9** for example ARS route configuration for **50: Main** as well as a backup route.



For private outgoing calls, Short Code ***67N;** was created as shown in the screenshot below. The digits ***67** was used as a prefix that IP Office user will dial to access to the SIP Trunk for private outgoing calls to PSTN. This causes the called PSTN party not to display Calling Party Name and Number associated with IP Office user.

- In **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, it is ***67N;**. This short code will be invoked when the user dials ***67** followed by any number.
- Set **Feature** to **Dial**. This is the feature that the short code will invoke.
- Set **Telephone Number** to **WN"@10.20.64.220:5060"**. This field is used to construct the "Request URI" and "To" headers for private outgoing calls. The value **W** directs IP Office to mask the "From" header with **anonymous** to block Calling Party Name and Calling Party Number. The value **N** represents the number dialed by the user. The host part following the "@" is the service provider SIP domain.
- Set **Line Group ID** field to **19** which is the outgoing line group number defined on **SIP URI** tab on the **SIP Line** in **Section 5.5.1**. This short code will use this line group when placing private outgoing calls.
- Set **Locale** to **United State (US English)**.

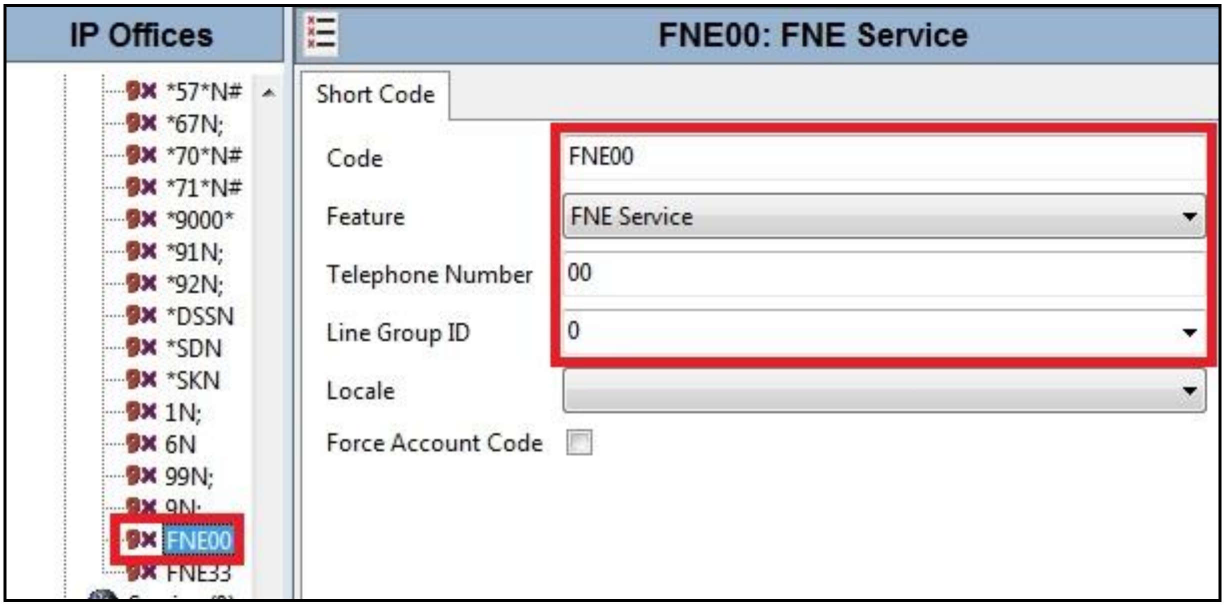


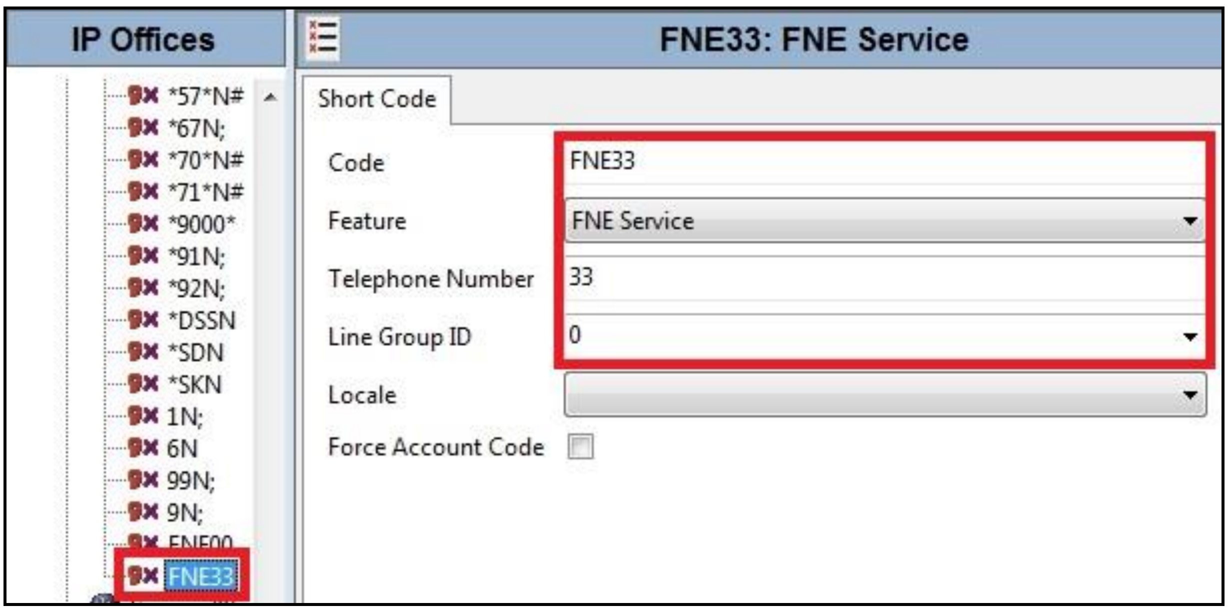
Note: For outgoing private calls, IP Office send "P-Preferred-Identity" header for call authentication purpose. For more information, refer to **Section 2.12**, observation **5**.

For incoming calls from mobility extension to FNE features hosted by IP Office to provide **Dial Tone** or **Mobilecallback** functionalities, Short Code **FNE00** and **FNE33** were created. The **FNE00** and **FNE33** were configured with following parameters.

- For **Code** field, enter FNE feature code as **FNE00** for **Dial Tone** or **FNE33** for **Mobile Callback**.
- Set **Feature** field to **FNE Service**.
- Set **Telephone Number** field to **00** for **FNE00** or **33** for **FNE33**.
- Set **Line Group ID** field to **0**.
- Retain default values for other fields.

Following screenshots illustrate **FNE00** and **FNE33** configurations.



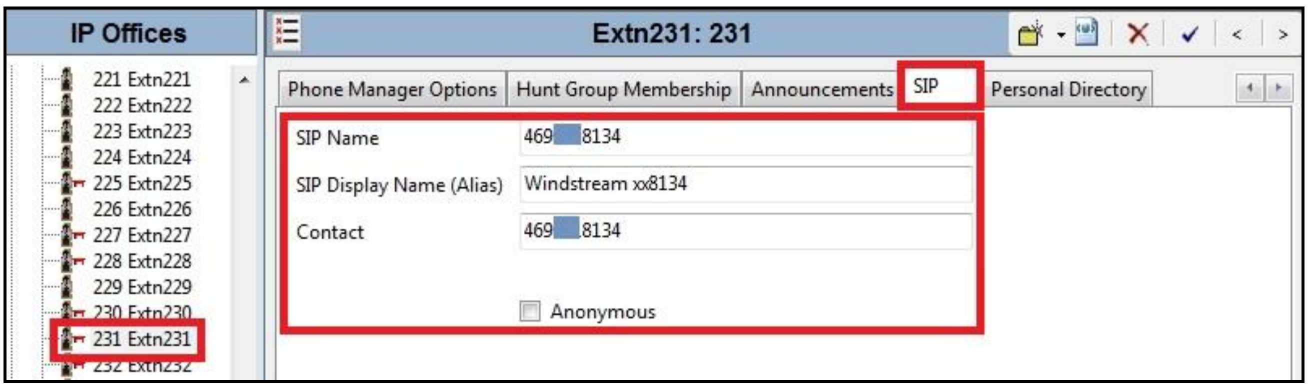


When complete, click OK to commit (not shown) then press Ctrl + S to save.

5.7 User

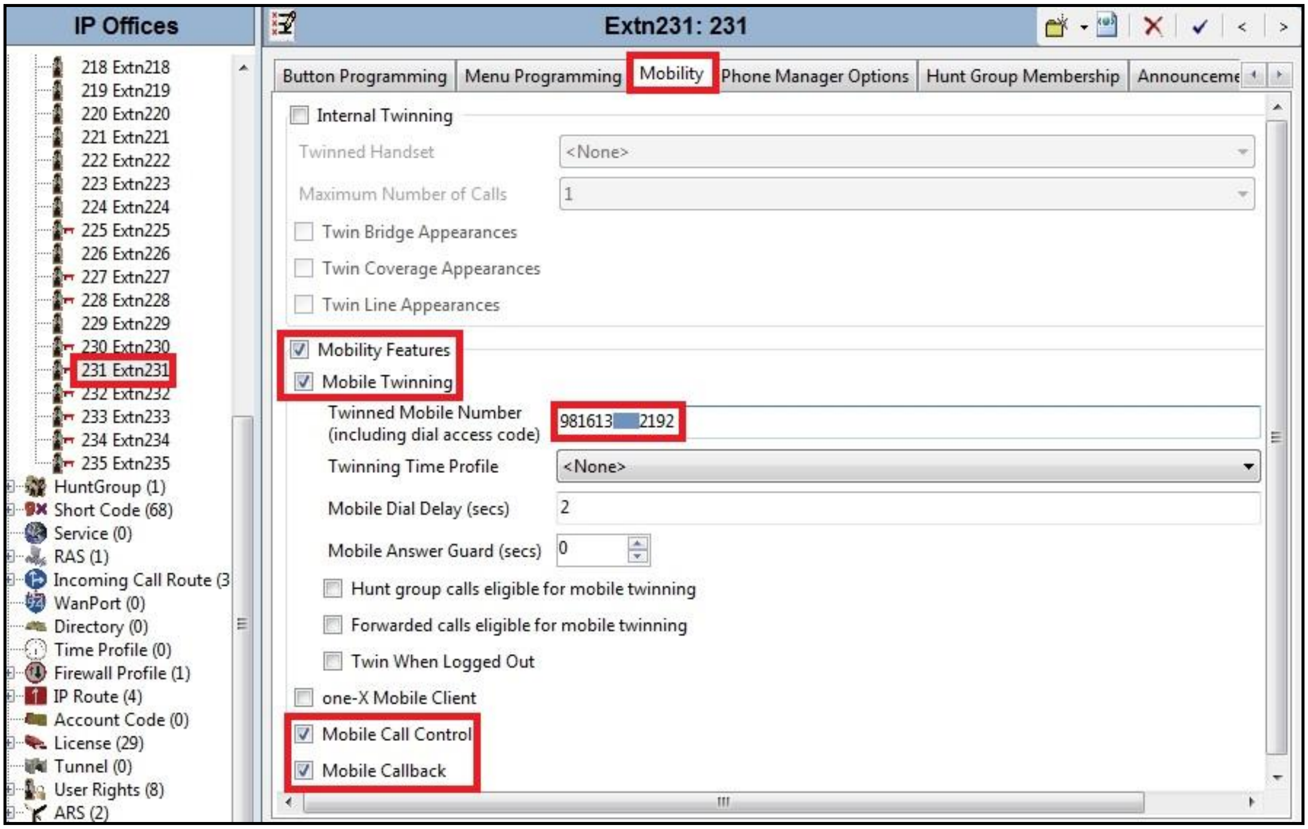
Configure SIP parameters for each user that will be placing and receiving calls via the SIP Line as defined in **Section 5.5**. To configure these settings, first select **User** in the left Navigation Pane and then select the name of the user to be modified. In the example below, with the user **Exnt231** selected, select **SIP** tab in the Details Pane.

The values entered for **SIP Name** and **Contact** fields were used as URI-User in the "From" header for outgoing calls. They also allow matching of URI-User for incoming calls without having to enter this number as an explicit SIP URI for the SIP Line (see **Section 5.5**). **SIP Name** and **Contact** fields were set to one of the DID numbers assigned to the enterprise by Windstream. **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 **Anonymous** box may be checked to withhold the user information from the networks. **Note**: For outgoing private calls, IP Office send "P-Preferred-Identity" header for call authentication purpose. For more information, refer to **Section 2.12**, observation **5**.



Mobile Twinning feature may be enabled on the user to allow incoming calls to simultaneously alert the desk phone and the mobile phone. The following screenshot shows **Mobility** tab was configured with following paramters:

- **Mobility Features** and **Mobile Twinning** boxes were checked.
- **Twinned Mobile Number** was configured with the number to reach the twinned mobile telephone, in this case it was **981613XXX2192** including digit 98 as the dial access code and 1613XXX2192 as the mobility extension.
- Check **Mobile Call Control** to allow incoming calls from mobility extension to access FNE00 (see **Section 5.6**).
- Check **Mobile Callback** to allow IP Office to call back mobility extension to provide dial tone responding to incoming calls from mobility extension to access FNE33 (see **Section 5.6**).
- Other options can be set according to customer requirements.

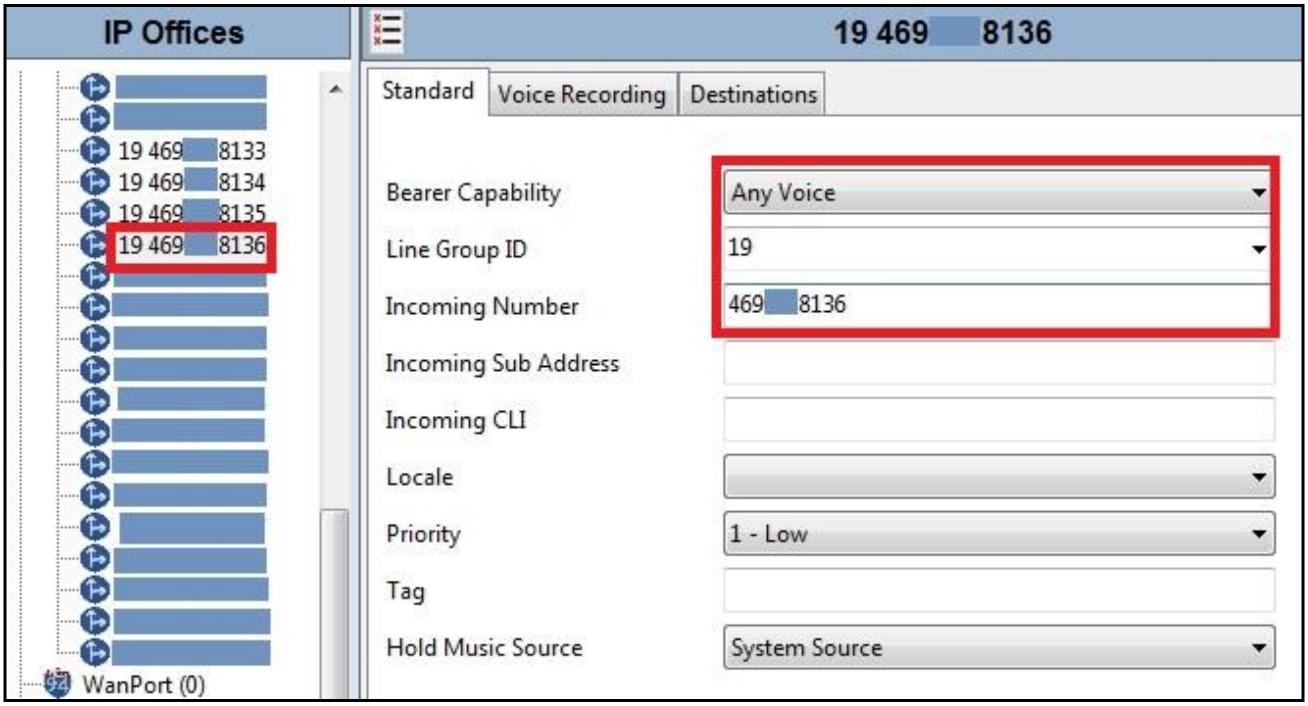


5.8 Incoming Call Route

An Incoming Call Route maps an incoming call on a specific SIP Line to an internal extension. This procedure should be repeated for each DID number provided by service provider. To create an Incoming Call Route, right click on **Incoming Call Route** in the left Navigation Pane and select **New** (not shown). On **Standard** tab of the Details Pane, enter the parameters as shown below:

- Set **Bearer Capability** field to **Any Voice**.
- Set **Line Group ID** field to SIP Line group 19 as defined in **Section 5.5**.
- Set **Incoming Number** field to the DID number that associate to the internal extension.
- Default values can be used for all other fields.

The screenshot below shows Incoming Call Route **19 469XXX8136** configured to receive an incoming call to DID number **469XXX8136** then alert local station **231**.



On **Destinations** tab, select destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to **469XXX8136** on SIP Line 19 are routed to local station **231 Extn231**.

Following screenshots show Incoming Call Routes to receive incoming calls to DID numbers **469XXX8133**, **469XXX8134** and **469XXX8135** that were similarly configured to access **FNE00**, **FNE33** and **VoiceMail**. The **Destinations** were appropriately defined as **FNE00**, **FNE33** and **VoiceMail**. **Note**: FNE00 and FNE33 were entered manually by selecting **Destination** as **DialIn** (not shown) then input the appropriate FNE feature code.

When complete, click OK to commit (not shown) then press Ctrl + S to save.

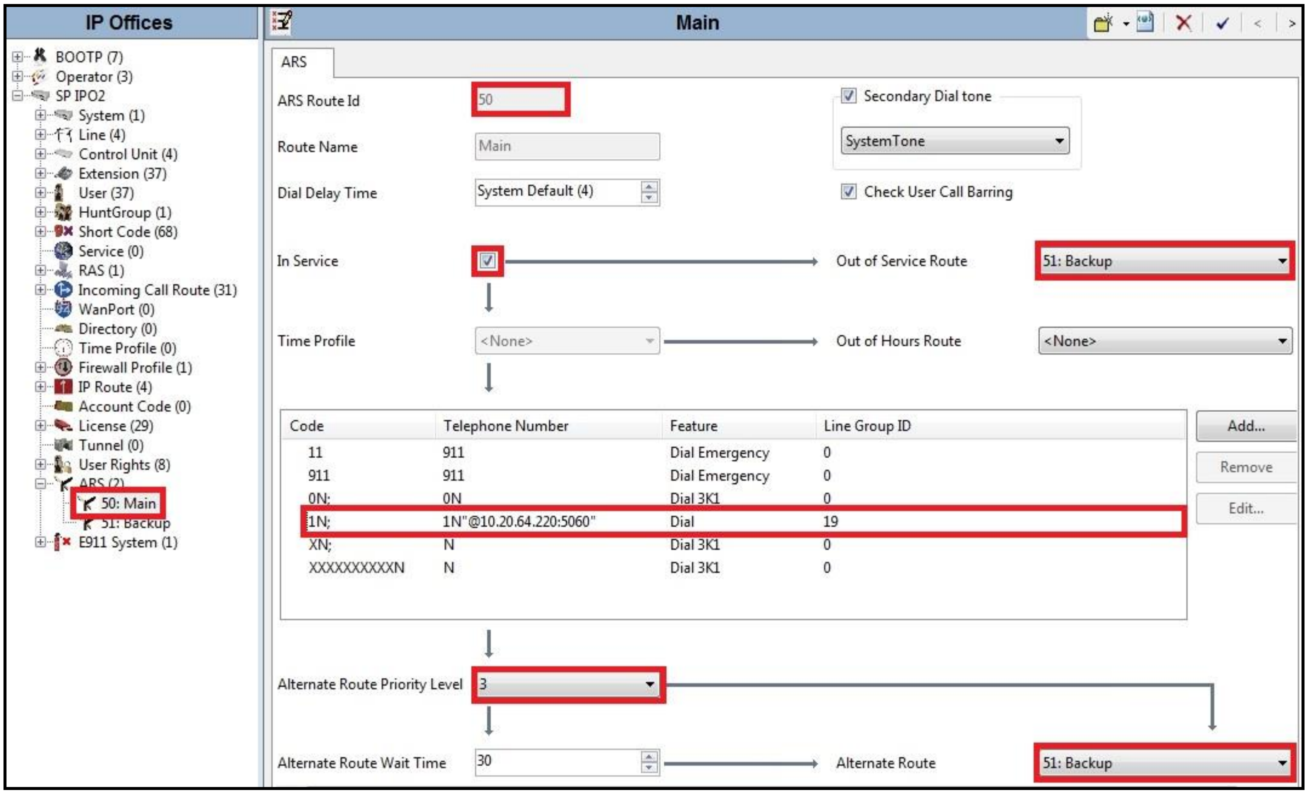
5.9 ARS and Alternate Routing

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

Optionally, ARS can be used rather than the simple **98N;** short code approach as documented in **Section 5.6**. 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 be rerouted automatically if the primary route or outgoing line group is not available. Although not shown in this section, ARS also facilitates more specific dialed telephone number matching, enabling immediate routing and alternate treatment for different types of numbers following the access code. For example, if all 1+10 digit calls following an access code should use the SIP Line preferentially, but other local or service numbers following the access code should prefer a different outgoing line group, ARS can be used to distinguish the call behaviors.

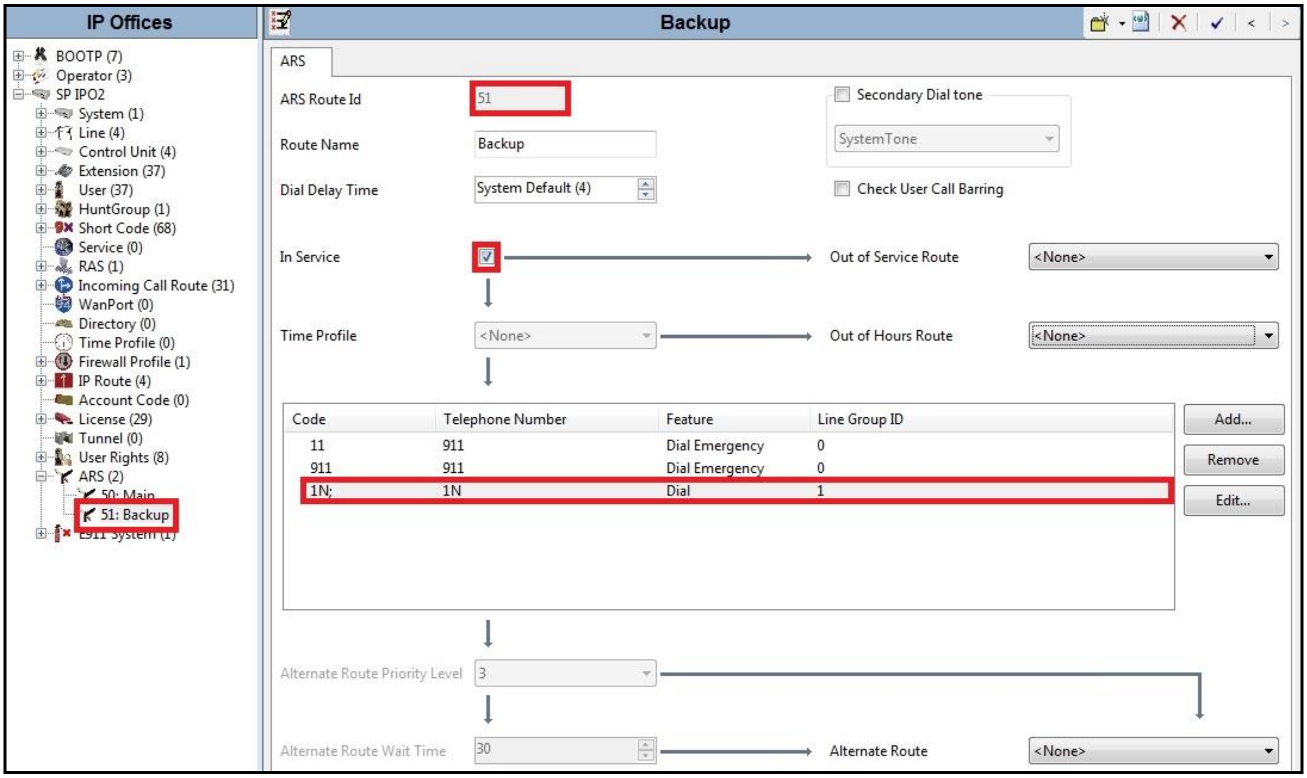
A new ARS can be created by right-click **ARS** in the Navigation pane then select **New** (not shown). To view or edit an existing ARS route, select **ARS** in the Navigation pane then select the appropriate route name.

The following screenshot shows an example configuration for ARS **50:Main**. The **In Service** parameter refers to the ARS form itself. If the **In Service** box is unchecked, 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 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 5.6**) can be further analyzed to direct the call to a specific Line Group ID. Per the sample screenshot above, if the user dialed 61613XXX2192, the call would be directed to Line Group 19 which is the SIP Line configured and described in these Application Notes. If the Line Group 19 cannot be used, the call can automatically be routed to the **Alternate Route Priority Level 3** as shown in the screenshot. **Note**: Alternate routing can be considered a privilege not available to all callers. IP Office can control access to the alternate route by comparing the priority of the calling users to the value in **Alternate Route Priority Level** field.

The following screenshot shows an example ARS configuration for the route ARS **51:Backup**. Continuing from the prior example, if the user dialed 61613XXX2192 and the call could not be routed via the primary route **50: Main** as described above, the call will be delivered to the alternate route **51:Backup**. Per the configuration shown below, the call will be delivered to Line Group 1, using an analog trunk connecting IP Office to PSTN as a backup connection. In this case, the original dialed number (sans the short code 6) will be dialed as is through the analog/PRI trunk to the PSTN. Additional codes (e.g., 411, 0+10, etc.) can be added to ARS route by selecting **Add…** button to the right of the list of previously configured codes (not shown).



5.10 Save Configuration

Navigate to **File → Save Configuration** in the menu bar at the top of the screenshot to save the configuration performed in the preceding sections (not shown).

6. Windstream SIP Trunking Service Configuration

Windstream is responsible for the configuration of Windstream SIP Trunking Service. Windstream will provide the customer the necessary information to configure the Avaya IP Office SIP Trunk. The provided information from Windstream includes:

- IP address of the Windstream SIP proxy.
- Credential for Registration.
- DID numbers.
- Supported codecs.
- A customer specific SIP signaling reference.

The sample configuration between the enterprise and Windstream for the compliance testing is a dynamic using SIP Registration method as described in **Section 5.5**.

7. Verification and Troubleshooting

This section provides verification steps that may be performed in the field to verify that the solution is configured properly. This section also provides a list of useful troubleshooting tips that can be used to troubleshoot the solution.

7.1 Verification Steps

Following activities were made to each test scenario:

- Verify that endpoints at the enterprise site can place and receive calls to PSTN and that the call remains active for more than 35 seconds.
- Verify that user on both PSTN and the enterprise sides can end an active call by hanging up.

7.2 Protocol Traces

Following SIP message headers were inspected using sniffer trace analysis tool:

- Request-URI: Verify proper request number and SIP domain.
- From: Verify proper display name and display number.
- To: Verify proper display name and display number.
- P-Preferred-Identity: Verify proper display name and display number.
- Privacy: Verify privacy masking with “id”.
- Diversion: Verify proper display name and display number.

Following attributes in SIP message body were inspected using sniffer trace analysis tool:

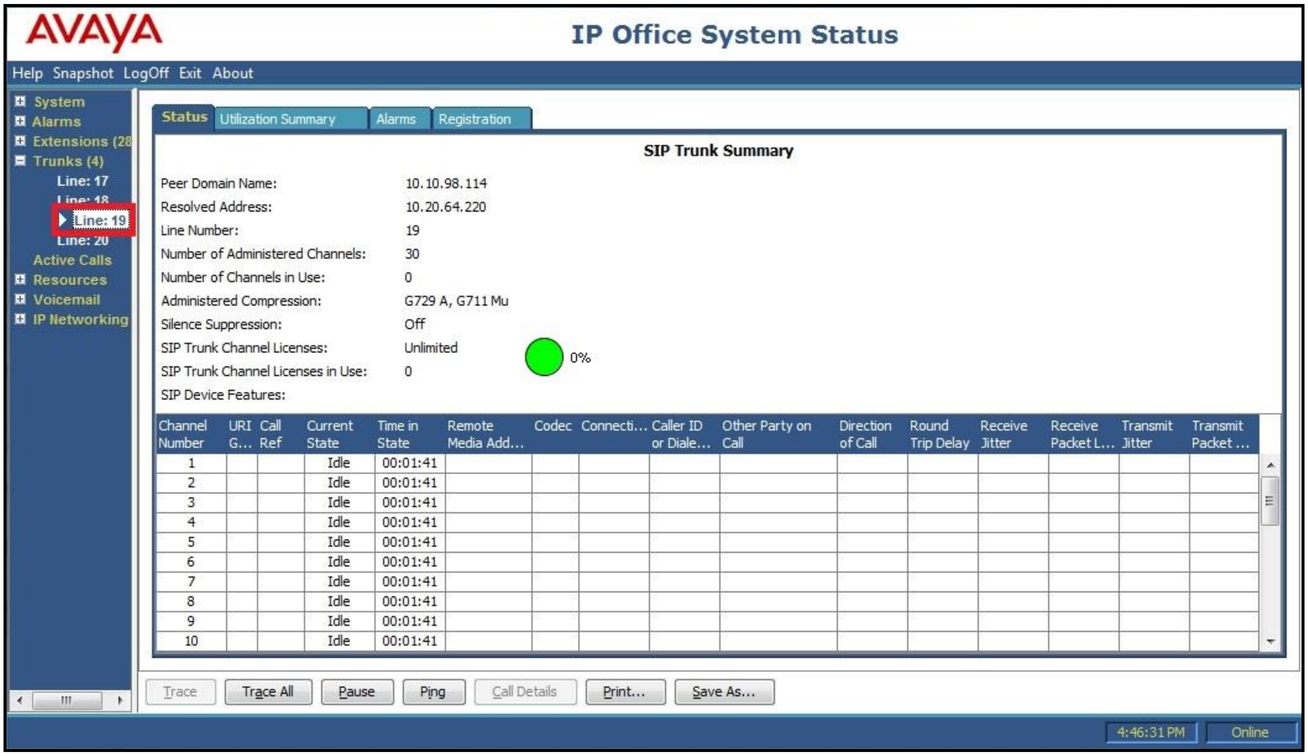
- Connection Information (c line): Verify correct IP addresses of near and far endpoints.
- Time Description (t line): Verify correct session timeout value of near and far endpoints.
- Media Description (m line): Verify correct audio port, codec, DTMF event description.
- Media Attribute (a line): Verify correct audio port, codec, ptime, send/ receive ability, DTMF event.

7.3 Troubleshooting

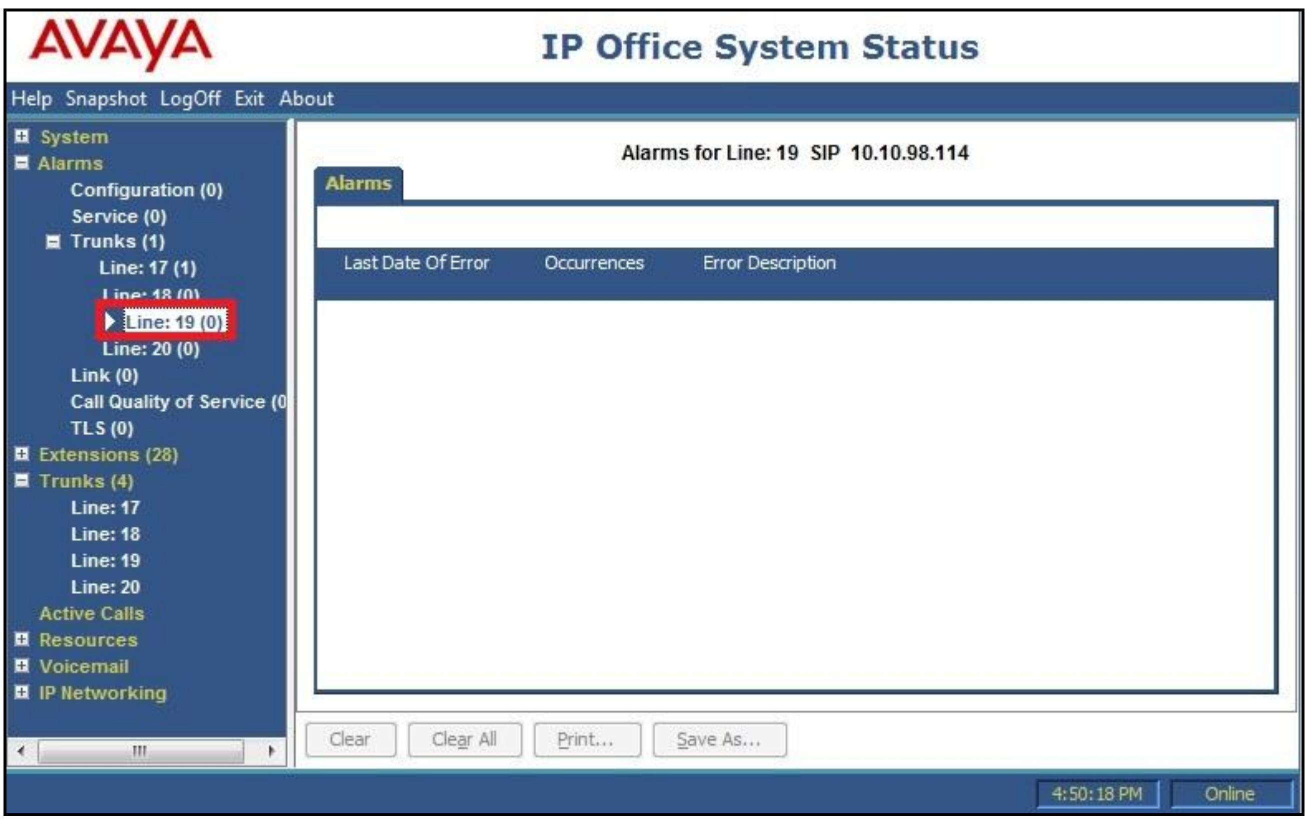
7.3.1 IP Office System Status

Following steps may be used to verify the configuration:

- Use 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 IP Office Manager is 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 **Alarms** tab and verify that no alarms are active on the SIP Line.



7.3.2 Sniffer Traces Analysis

Using a network sniffing tool (e.g., Wireshark) to monitor the SIP signaling between the enterprise and Windstream. The sniffer traces are captured at the public interface of IP Office.

Following screenshots show an example incoming call from Windstream to the enterprise.

- Incoming INVITE request from Windstream.

```
INVITE sip:469XXX8133@10.10.98.114:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 10.20.64.220:5060;branch=z9hG4bKo2dbdr009olh8nskt441.1
From: "Unavailable"<sip:1613XXX5203@10.20.64.220;user=phone;broadworks=BWWESTSIGIS-
1ecpqqcalh9ba>;tag=130052695-1373577854828-
To: "469XXX8133 469XXX8133"<sip:469XXX8133@10.20.64.220;interopis=interopis-
krbt95tpijjh8>
Call-ID: BW2124148281107133932840200@64.199.51.199
CSeq: 667901879 INVITE
Contact: <sip:1613XXX5203@10.20.64.220:5060;broadworks=BWWESTSIGIS-
o6i7c69dv2579;transport=udp>
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY
Accept: application/media_control+xml,application/sdp,multipart/mixed
Supported: timer
Min-SE: 60
Max-Forwards: 47
Content-Type: application/sdp
Content-Length: 283

v=0
o=BroadWorks 412718 1 IN IP4 10.20.64.220
s=-
c=IN IP4 10.20.64.220
t=0 0
m=audio 39090 RTP/AVP 18 0 8 101
a=sendrecv
a=ptime:20
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=fmtp:18 annexb=no
```

- 200 OK response from the enterprise.

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.20.64.220:5060;branch=z9hG4bKo2dbdr009olh8nskt441.1
From: "Unavailable" <sip:1613XXX5203@10.20.64.220;user=phone;broadworks=BWWESTSIGIS-
1ecpqqcalh9ba>;tag=130052695-1373577854828-
To: "469XXX8133 469XXX8133" <sip:469XXX8133@10.20.64.220;interopis=interopis-
krbt95tpijjh8>;tag=34fbf31d33fae30c
Call-ID: BW2124148281107133932840200@64.199.51.199
CSeq: 667901879 INVITE
Contact: "Windstream x8133" <sip:469XXX8133@10.10.98.114:5060;transport=udp>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
Supported: timer
Server: IP Office 8.1 (69)
Min-SE: 90
Content-Type: application/sdp
Content-Length: 227
```

```
v=0
o=UserA 401568664 303158339 IN IP4 10.10.98.114
s=Session SDP
c=IN IP4 10.10.98.114
t=0 0
m=audio 49152 RTP/AVP 18 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

Following screenshots show an example outgoing call from the enterprise to Windstream.

- Outgoing INVITE request from the enterprise.

```
INVITE sip:1613XXX5204@10.20.64.220:5060 SIP/2.0
Via: SIP/2.0/UDP
10.10.98.114:5060;rport;branch=z9hG4bK087bf22127e2d68c357da43b80f9aa17
From: "Windstream xx8134" <sip:469XXX8134@10.10.98.114>;tag=611bddf3a282a586
To: <sip:1613XXX5204@10.20.64.220:5060>
Call-ID: 0b1b3ee84d631c74fbb24d5e80215cfc
CSeq: 236560110 INVITE
Contact: "Windstream xx8134" <sip:469XXX8134@10.10.98.114:5060;transport=udp>
Max-Forwards: 70
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
Content-Type: application/sdp
Supported: timer
User-Agent: IP Office 8.1 (69)
Content-Length: 253

v=0
o=UserA 2635351851 3228128835 IN IP4 10.10.98.114
s=Session SDP
c=IN IP4 10.10.98.114
t=0 0
m=audio 49152 RTP/AVP 18 0 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

- Incoming 200 OK response from Windstream.

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
10.10.98.114:5060;branch=z9hG4bK087bf22127e2d68c357da43b80f9aa17;rport=5060
From: "Windstream xx8134" <sip:469XXX8134@10.10.98.114>;tag=611bddf3a282a586
To: <sip:1613XXX5204@10.20.64.220:5060>;tag=554238335-1373578033674
Call-ID: 0b1b3ee84d631c74fbb24d5e80215cfc
CSeq: 236560110 INVITE
Supported: timer
Contact: <sip:1613XXX5204@10.20.64.220:5060;broadworks=BWWESTSIGIS-
o6i7c69dv2579;transport=udp>
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY
Accept: application/media_control+xml,application/sdp
Content-Type: application/sdp
Content-Length: 235

v=0
```

```
o=BroadWorks 412723 1 IN IP4 10.20.64.220
s=-
c=IN IP4 10.20.64.220
t=0 0
m=audio 39082 RTP/AVP 18 101
a=sendrecv
a=ptime:20
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=fmtp:18 annexb=no
```

8. Conclusion

These Application Notes describe the configuration necessary to connect Avaya IP Office Release 8.1 to Windstream SIP Trunking Service using Broadsoft Platform.

All of the test cases have been executed. Despite the number of observations seen during testing as noted in **Section 2.2**, the test results met the objectives outlined in **Section 2.1**. The Windstream SIP Trunking Service using Broadsoft Platform is considered **compliant** with Avaya IP Office Release 8.1.

9. References

[1] *IP Office 8.1 Installation*, Document Number 15-601042, Issue 26j, 19 Sep 2012.
[2] *IP Office 8.1 Manager 10.1*, Document Number 15-601011, Issue 29o, 03 Aug 2012.
[3] *IP Office 8.1 Administering Voicemail Pro*, Document Number 15-601063, Issue 27b, 05 June 2012.

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 Service using Broadsoft Platform is available from Windstream.

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.