



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Frontier SIP Trunking with Avaya IP Office 9.0 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider Frontier (Metaswitch platform) and Avaya IP Office 9.0.

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

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

Frontier is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider Frontier and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office (IPO) solution consists of an Avaya IP Office 500v2 Release 9.0, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, digital, and analog endpoints.

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

2. General Test Approach and Test Results

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

Testing was performed with IP Office 500 v2 R9.0, but it also applies to IP Office Server Edition R9.0. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R9.0 to support analog or digital endpoints or trunks.

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

2.1. Interoperability Compliance Testing

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

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

- Caller number/ID presentation.
- Privacy requests (i.e., caller anonymity) and Caller ID restriction for inbound and outbound calls.
- DTMF transmission using RFC 2833.
- Voicemail navigation for inbound and outbound calls.
- Telephony features such as hold and resume, transfer, and conference.
- Use of SIP REFER for call transfer to PSTN
- FAX G.711 Pass Through and T.38.
- Off-net call forwarding.
- Twinning to mobile phones on inbound calls.

2.2. Test Results

Frontier SIP Trunking passed compliance testing.

Items not supported or not tested included the following:

- Inbound toll-free and outbound emergency calls (911) are supported but were not tested as part of the compliance test.
- T.38 Fax is not fully supported.

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

- **Incorrect Call Display on PSTN Phone** – Call display was not properly updated on PSTN phone involved in a call transfer. After the call transfer was completed, the PSTN phone did not display the actual connected party but instead showed the party that initiated the transfer. However, the call transfer is completed with 2 way voice path.
- **Consultative Transfer using REFER on SIP Desk-phone** – Scenario of Inbound and Outbound calls to transfer the calls from one PSTN to another PSTN phone. After press “Transfer” button to complete the call transfer on the SIP Desk-phone, 11XX series, the phone displayed “Transfer failed”. The 11XX desk-phone is still active and in off-hook state. Hanging up the phone will put the phone back on-hook state. There is no impact on user experience, both PSTN phones have 2 ways speech path, PSTN originator and PSTN transferee.
- **Call Display on PSTN Phone on PSTN Hold and Resume** – Call display was not properly updated on PSTN phone involved in PSTN Hold and Resume operation. After the inbound call from PSTN to Avaya IP Office, the call was put on hold by PSTN phone, when the call was resumed, the PSTN phone displayed trunk number instead of the calling party ID.
- **T.38 Faxing** – Frontier supports outbound T.38 faxing only for local calls. Outbound long-distance T.38 faxing failed: Frontier responded to T.38 re-INVITE from Avaya IP Office with "488 Not Acceptable Here". Inbound T.38 faxing worked properly. Plain G.711u faxing worked fine for outbound faxing. During compliance testing, a workaround configuration (T.38 with Fallback) was administered on Avaya IP Office (see **Section 5.4.2**) so that inbound faxing used T.38; outbound faxing used T.38 for local faxing call and G.711u for long-distance faxing call.

2.3. Support

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

For technical support on Frontier SIP Trunking service, contact Frontier at <https://frontier.com/enterprise>

3. Reference Configuration

Figure 1 below illustrates the test configuration. The test configuration shows an enterprise site connected to Frontier SIP Trunking service through the public IP network. For confidentiality and privacy purposes, actual public IP addresses used in this testing have been masked out and replaced with fictitious IP addresses throughout the document.

Located at the enterprise site is an Avaya IP Office 500v2 with the MOD DGTL STA16 expansion which provides connections for 16 digital stations to the PSTN. The extension PHONE 8 card which provides connections for 8 analog stations to the PSTN as well as 64-channel VCM (Voice Compression Module) for supporting VoIP codecs. The LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public IP network. Endpoints include an Avaya 9600 Series IP Telephone (with H.323 firmware), an Avaya 9508 Digital Telephone, an Avaya Symphony 2000 Analog Telephone, Avaya IP Office Softphone and 11XX series SIP phone. A separate Windows XP PC runs Avaya IP Office Manager to configure and administer the Avaya IP Office.

Mobility Twinning is configured for some of the Avaya IP Office users so that calls to these user phones will also ring and can be answered at the configured mobile phones.

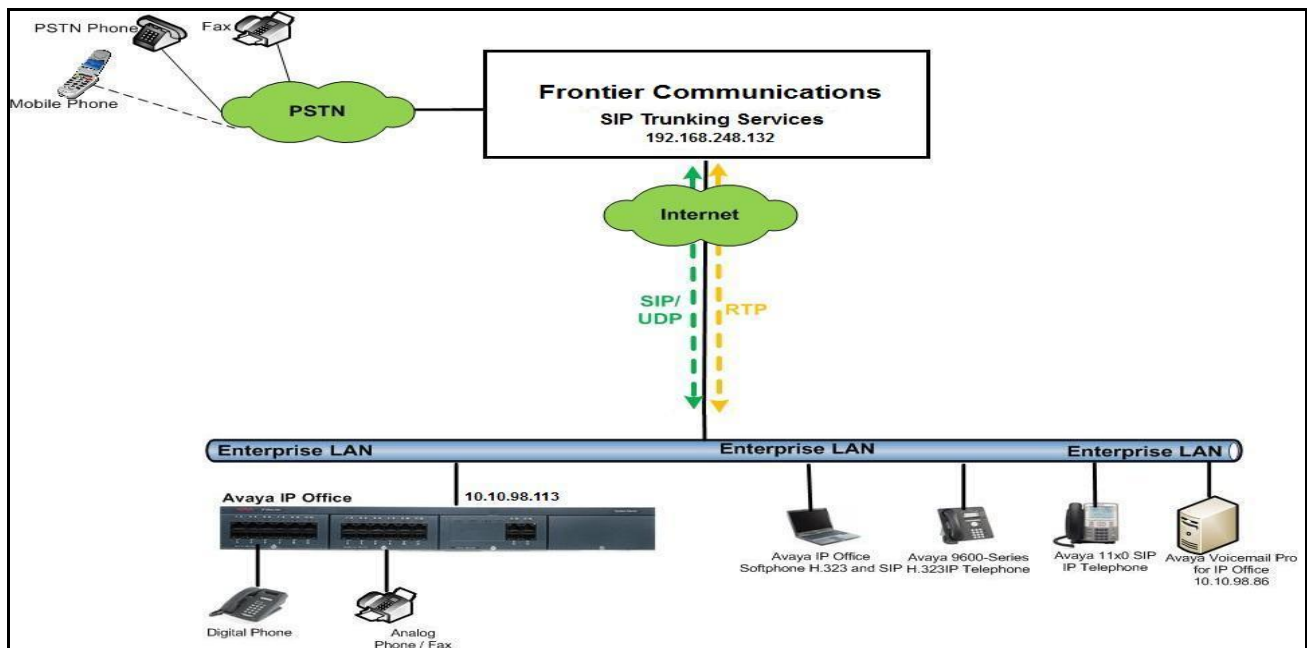


Figure 1: Test Configuration for Avaya IP Office with Frontier SIP Trunking Service

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

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/firmware were used for the sample configuration provided:

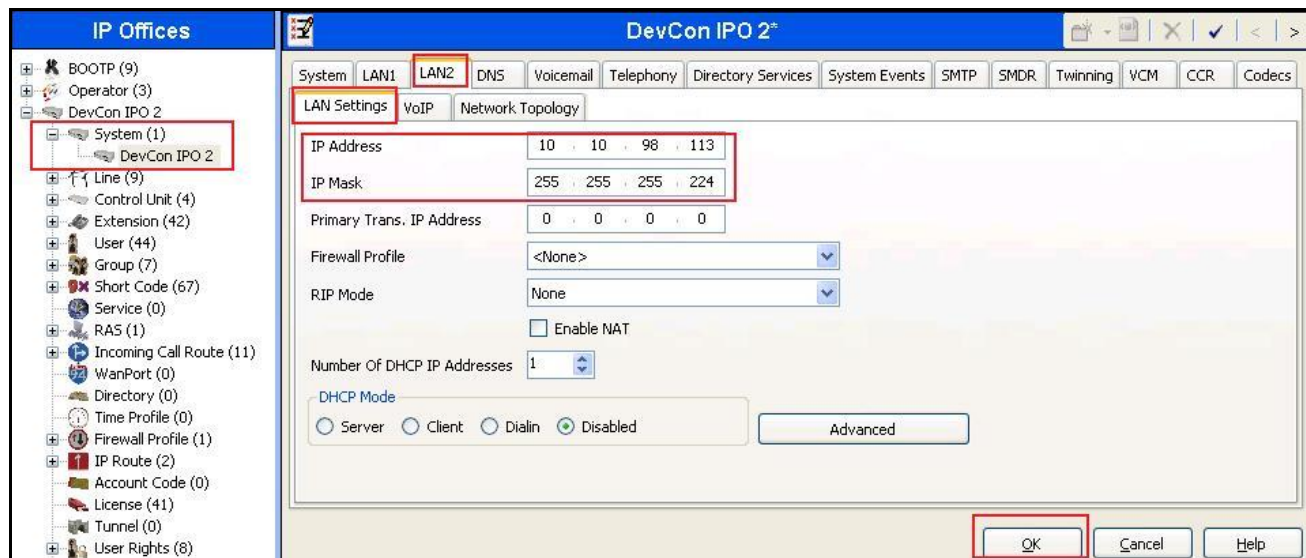
Avaya Telephony Components	
Equipment	Release
Avaya IP Office 500v2	9.0.300.941
Avaya IP Office Manager	9.0.300.941
Avaya Voicemail Pro for IP Office	9.0.300.941
Avaya 11x0 IP Telephone (SIP)	SIP11x0e04.03.12.00
Avaya 9630G IP Telephone (H.323)	Avaya one-X® Deskphone Edition S3.2
Avaya IP Office Softphone	3.2.3.20 64770
Avaya Digital Telephone (9508)	N/A
Avaya Symphony 2000 Analog Telephone	N/A
Frontier SIP Trunking Service Components	
Component	Release
Acme Packet Net-Net 4000 SBC	6.2
Metaswitch	8.1

5. Configure IP Office

This section describes the Avaya IP Office configuration to support connectivity to Frontier SIP Trunking service. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials. A management window will appear similar to the one shown in the next section. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details pane on the right side. These panes will be referenced throughout the Avaya IP Office configuration. Proper licensing as well as standard feature configurations that are not directly related to the interface with the service provider (such as LAN interface to the enterprise site and IP Office Softphone support) is assumed to be already in place.

5.1. LAN Settings

In the sample configuration, the **DevCon IPO2** was used as the system name and 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. To access the LAN settings, first navigate to **System (1) → DevCon IPO2** in the Navigation and Group Panes 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. Then click **OK** to submit.



Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such the 9600-Series IP Telephones used in the sample configuration. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Frontier. The **SIP Registrar Enable** box is checked to allow Avaya IP Office Softphone usage. 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**. The specific values used for the compliance test are shown in the example below. All other parameters should be set according to customer requirements. Then click **OK** to submit.

The screenshot shows the DevCon IPO 2 configuration window. The left sidebar displays a tree view with 'System (1)' and 'DevCon IPO 2' selected. The main area has tabs for System, LAN1, LAN2 (selected), DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, Twinning, VCM, CCR, and Codecs. The LAN2 tab has sub-tabs for LAN Settings, VoIP (selected), and Network Topology. The VoIP sub-tab contains several sections:

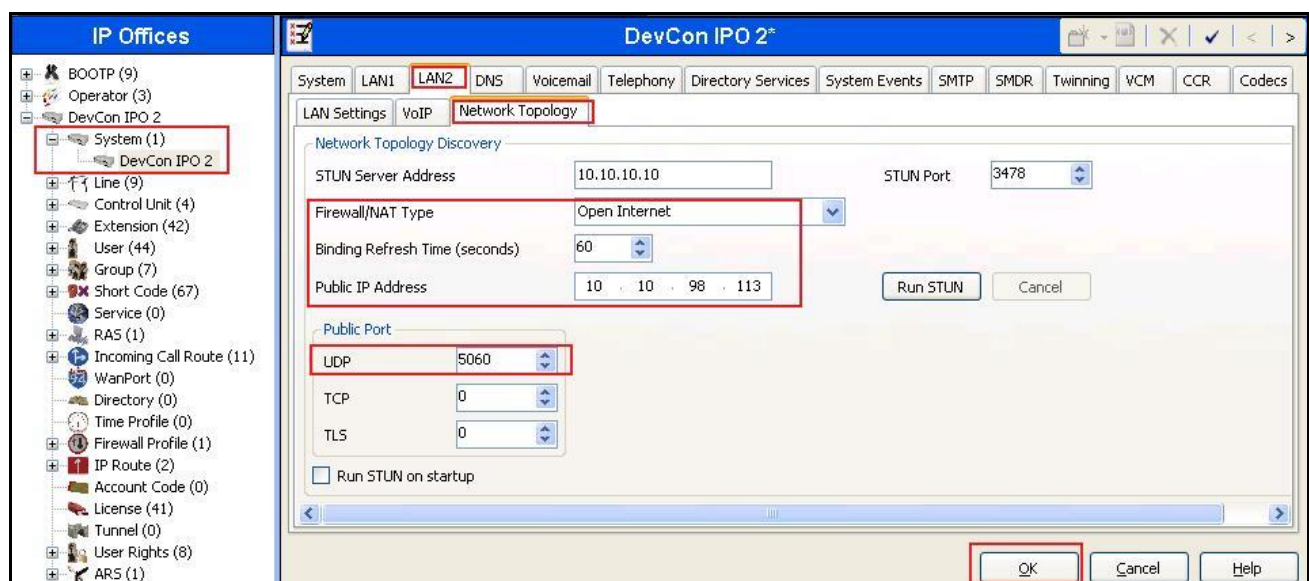
- H323 Gatekeeper Enable** (checked)
- SIP Trunks Enable** (checked)
- SIP Registrar Enable** (checked)
- Domain Name** (empty)
- Layer 4 Protocol** (UDP, TCP, TLS)
- Challenge Expiry Time (secs)** (10)
- RTP** (Port Number Range, Port Number Range (NAT))
- Keepalives** (Scope: RTP, Periodic timeout: 30, Initial keepalives: Enabled)

The OK button is highlighted with a red box.

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

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to **Open Internet**. With this configuration, STUN will not be used.
- Set **Binding Refresh Time (seconds)** to **60**. This value is used as one input to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider. See **Section 5.9** or complete details.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port. **Public Port, UDP** is set to **5060**
- All other parameters should be set according to customer requirements.

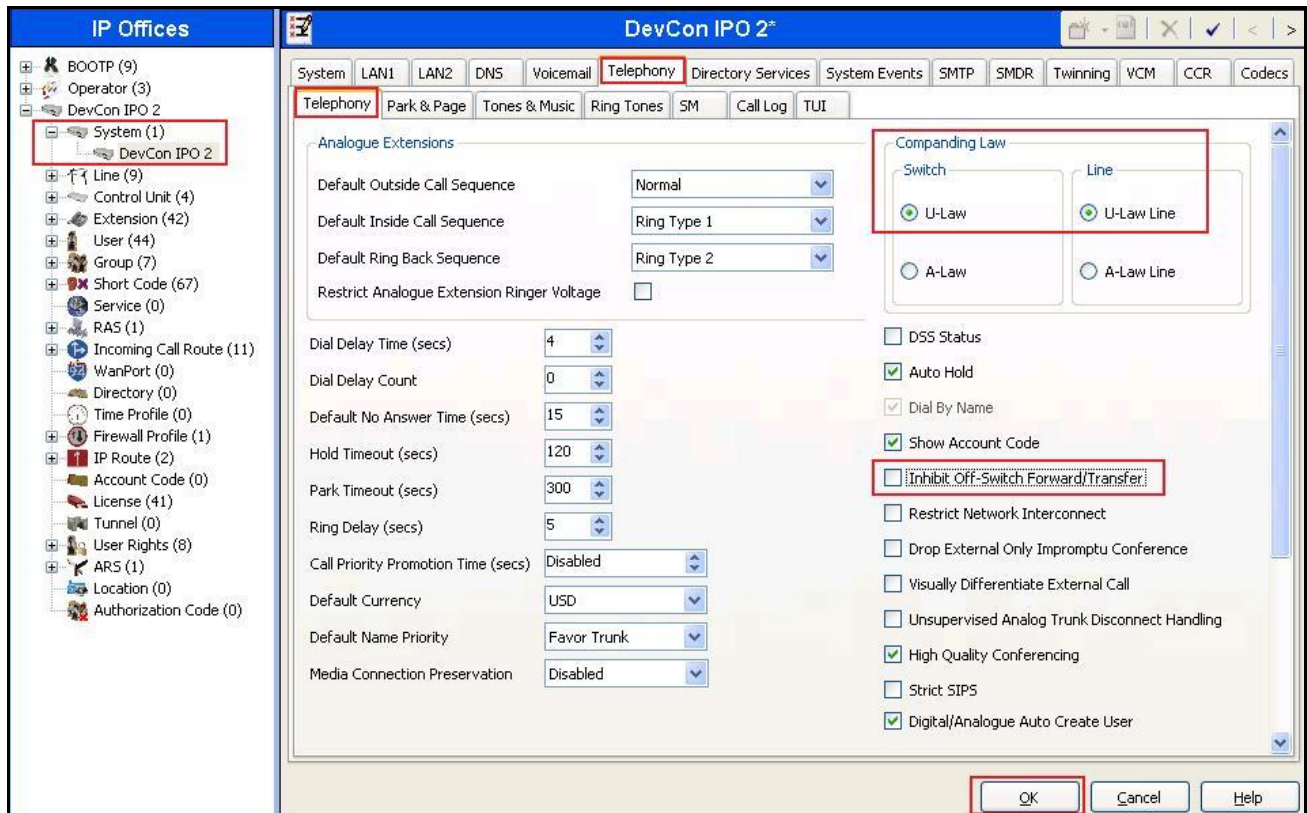
Click **OK** to submit.



In the compliance test, the **LAN1** interface was used to connect the Avaya IP Office to the enterprise site IP network. The **LAN1** interface configuration is not directly relevant to the interface with Frontier SIP Trunking service, and therefore is not described in these Application Notes.

5.2. System Telephony Settings

Navigate to the **Telephony** → **Telephony** Tab in the Details Pane. Choose the **Companding Law** typical for the enterprise location. For North America, **U-Law** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the service provider across the SIP trunk. Then click **OK**.



5.3. Twinning Calling Party Settings

When using twinning, the calling party number displayed on the twinned phone is controlled by two parameters. These parameters only affect twinning and do not impact the messaging or operation of other redirected calls such as forwarded calls. The first parameter is the **Send original calling party information for Mobile Twinning** box on the **System→Twinning** tab. The second parameter is the **Send Caller ID** parameter on the **SIP Line** form (shown in **Section 5.4**).

For the compliance testing, the **Send original calling party information for Mobile Twinning** as shown below was unchecked. This setting allows the **Send Caller ID** parameter that is set to **Diversion Header** in **Section 5.4.2** to be used. IP Office will send the 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 the PSTN to a twinned phone, IP Office sends Calling Party Number of the originating PSTN party.
- Click **OK** (not shown).



5.4. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and Frontier SIP Trunking service. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.4.1** to create the SIP Line from the template.

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

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

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.4.2**.

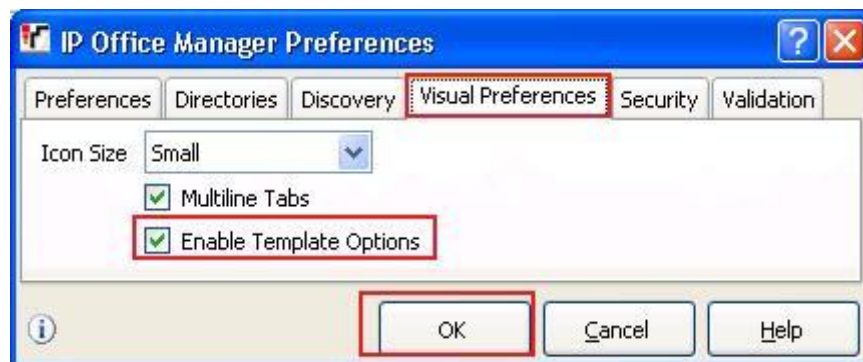
Also, the following SIP Line settings are not supported on Basic Edition:

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

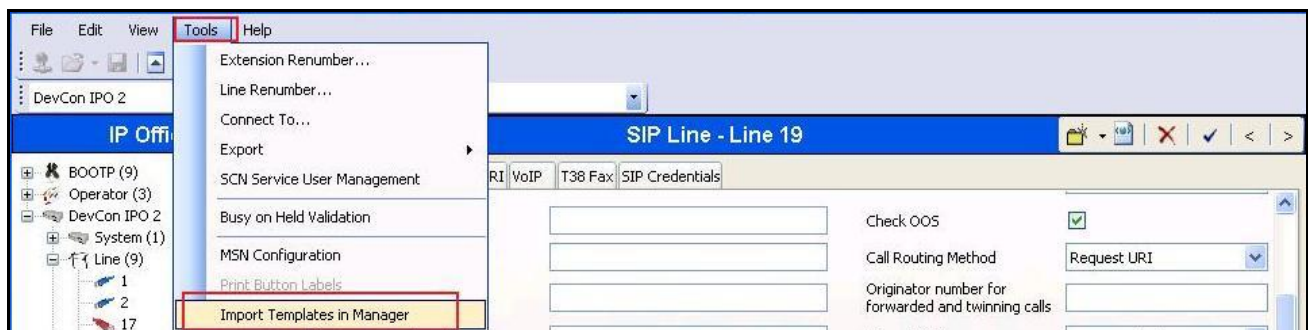
Alternatively, a SIP Line can be created manually. To do so right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.4.2**.

5.4.1. Create SIP Line From Template

1. Copy the template file to the computer where IP Office Manager is installed. Rename the template file to **FTSIPTIPO9.xml**. The file name is important in locating the proper template file in **Step 5**.
2. Verify that template options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the IP Office Manager Preferences window that appears, select the Visual Preferences tab. Verify that the box is checked next to **Enable Template Options**. Click **OK**.

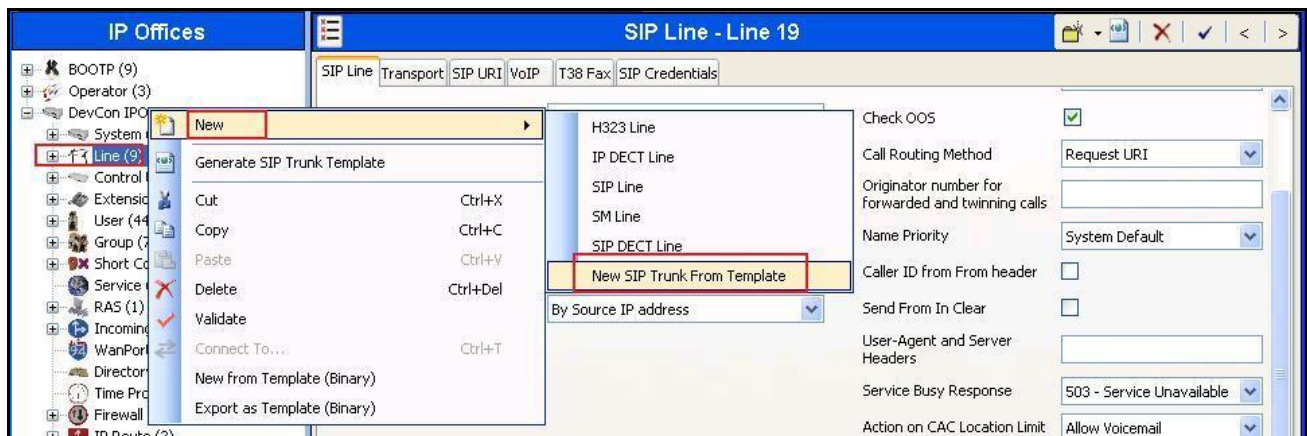


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default template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



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

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



5. In the subsequent Template Type Selection pop-up window, select **United States** from the **Country** pull-down menu and select **Frontier** from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name (**FTSIPTIPO9.xml**) created in **Step 1**. Click **Create new SIP Trunk** to finish creating the trunk.



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

5.4.2. Create SIP Line Manually

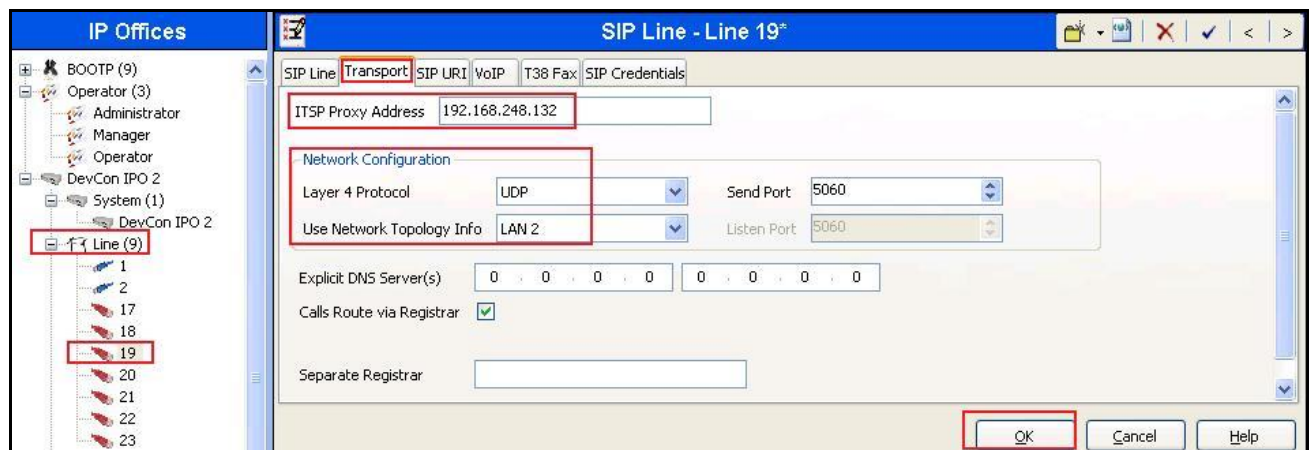
To create a SIP line, begin by navigating to **Line** in the left Navigation Pane, then right-click in the Group Pane and select **New → SIP Line** (not shown). On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

- Set **ITSP Domain Name** to the enterprise domain so that IP Office uses this domain as the host portion of SIP URI in SIP headers such as the From header.
- Set **Send Caller ID** to **Diversion Header**.
- Check the **In Service** box.
- Check the **Check OOS** box. With this option selected, IP Office will use the SIP OPTIONS method to periodically check the SIP Line
- **URI Type** is set to **SIP**.
- Default values may be used for all other parameters.

The area of the screen entitled **REFER Support** is used to enable/disable SIP REFER for call transfers. The default values of “Auto” for **Incoming** and **Outgoing** effectively disable use of SIP REFER. To enable SIP REFER, select “**Always**” from the drop-down menu for **Incoming** and **Outgoing**. In the compliance test, both configurations were successfully tested to transfer a call between a PSTN phone and an enterprise phone to a second PSTN phone. Then click **OK**.

The screenshot displays the 'SIP Line - Line 19*' configuration window. The left navigation pane shows the 'IP Offices' tree with 'Line (9)' selected. The main configuration area is divided into several sections. The 'SIP Line' tab is active, showing fields for 'Line Number' (19), 'ITSP Domain Name' (192.168.248.132), 'Prefix', 'National Prefix' (0), 'Country Code', 'International Prefix' (00), 'Send Caller ID' (Diversion Header), and 'Association Method' (By Source IP address). The 'In Service' checkbox is checked, and 'URI Type' is set to 'SIP'. The 'Check OOS' checkbox is also checked. The 'Call Routing Method' is set to 'Request URI'. The 'Originator number for forwarded and twinning calls' field is empty. The 'Name Priority' is set to 'System Default'. The 'Caller ID from From header' checkbox is unchecked. The 'Send From In Clear' checkbox is unchecked. The 'User-Agent and Server Headers' field is empty. The 'Service Busy Response' is set to '486 - Busy Here'. The 'Action on CAC Location Limit' is set to 'Allow Voicemail'. The 'REFER Support' section is expanded, showing 'Incoming' and 'Outgoing' both set to 'Auto'. The 'Method for Session Refresh' is set to 'Auto'. The 'Session Timer (seconds)' is set to 'On Demand'. The 'Media Connection Preservation' is set to 'Disabled'. The 'OK', 'Cancel', and 'Help' buttons are at the bottom right.

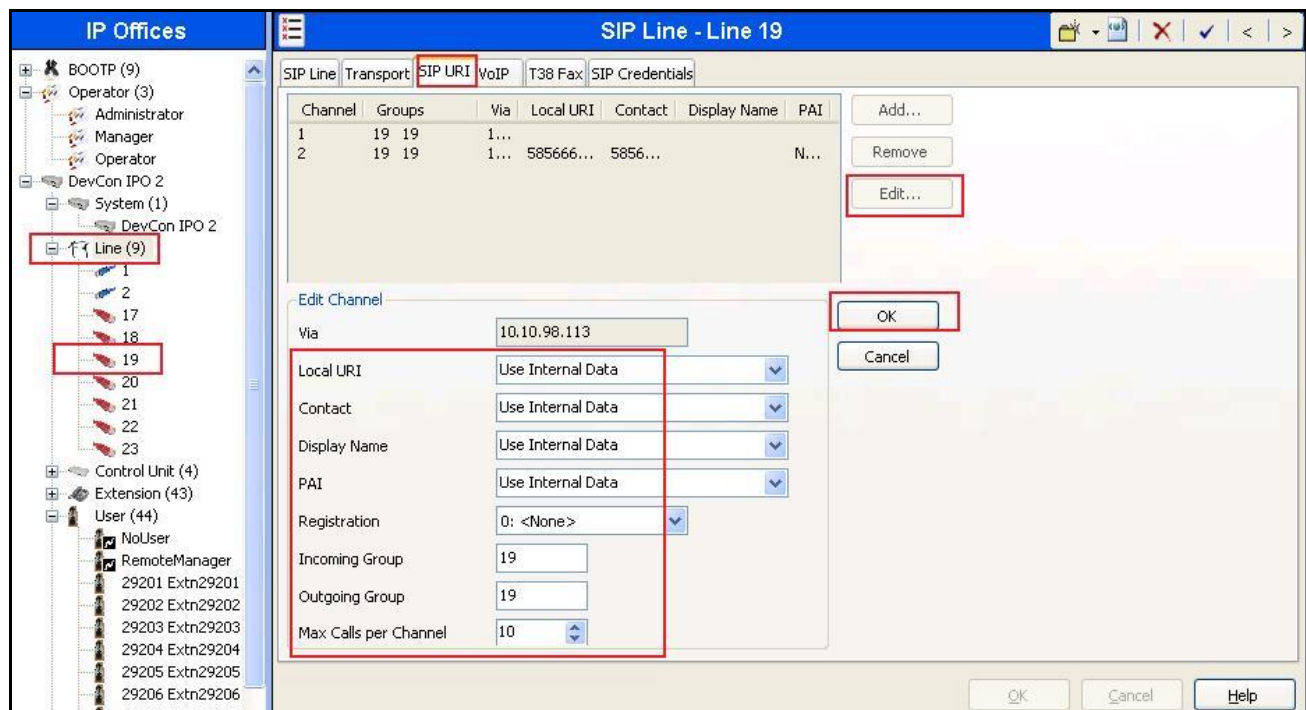
Select the **Transport** tab. The **ITSP Proxy Address** is set to Frontier SIP Proxy IP Address provided by Frontier. As shown in **Figure 1**, this IP Address is **192.168.248.132**. In the **Network Configuration** area, **UDP** is selected as the **Layer 4 Protocol**, and the **Send Port** is set to the port number provided by Frontier. The **Use Network Topology Info** parameter is set to **LAN 2**. This associates the SIP Line with the parameters in the **System → LAN2 → Network Topology** tab. Other parameters retain default values in the screen below. Then click **OK**.



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

- Set **Local URI**, **Contact** and **Display Name** to *Use Internal Data*. This setting allows calls on this line which SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.6**.
- Set **PAI** to *Use Internal Data*. With this setting IP Office will populate the SIP P-Asserted-Identity header on outgoing calls with the data set in the **SIP** tab of the call initiating **User** as shown in **Section 5.6**.
- Associate this line with an incoming line group in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. For the compliance test, a new incoming and outgoing group **19** was defined that only contains this line (line 19).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Click **OK**.

SIP URI entry for Channel 1



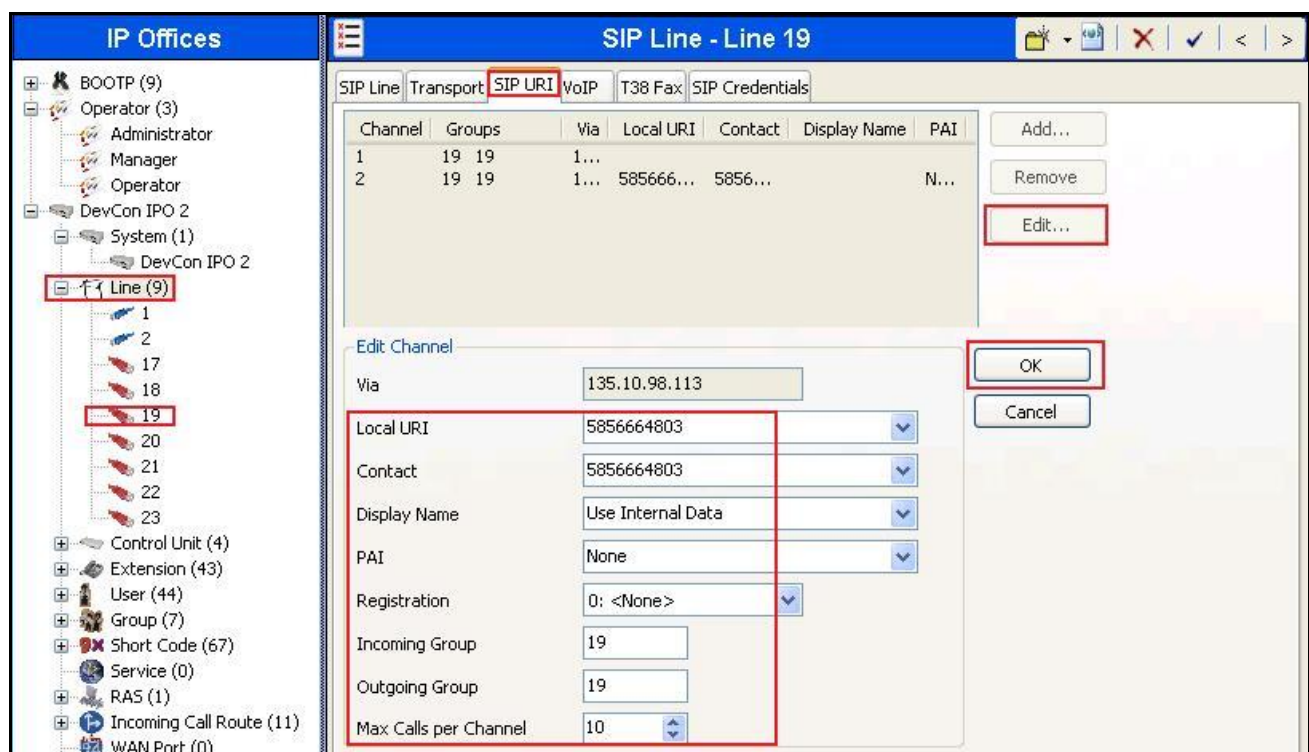
SIP URI entry **Channel 2** was similarly created for incoming calls appropriately to pre-define DID numbers **5856664803** to access to Feature Name Extension 00 (FNE00). The Short Codes for FNE00 was defined in **Section 5.5** to provide Dial Tone and Mobile Callback for mobility extension.

Channel 2 as shown in the screenshot below was configured with following parameters.

- Set the **Local URI** and **Contact** fields to pre-define DID number **5856664803** appropriately for **Channel 2**.
- Associate **Incoming Group** and **Outgoing Group** to SIP Line **19**.
- Set the **Max Calls per Channel** field to **10**.
- Other parameters retain default values.

Click **OK**.

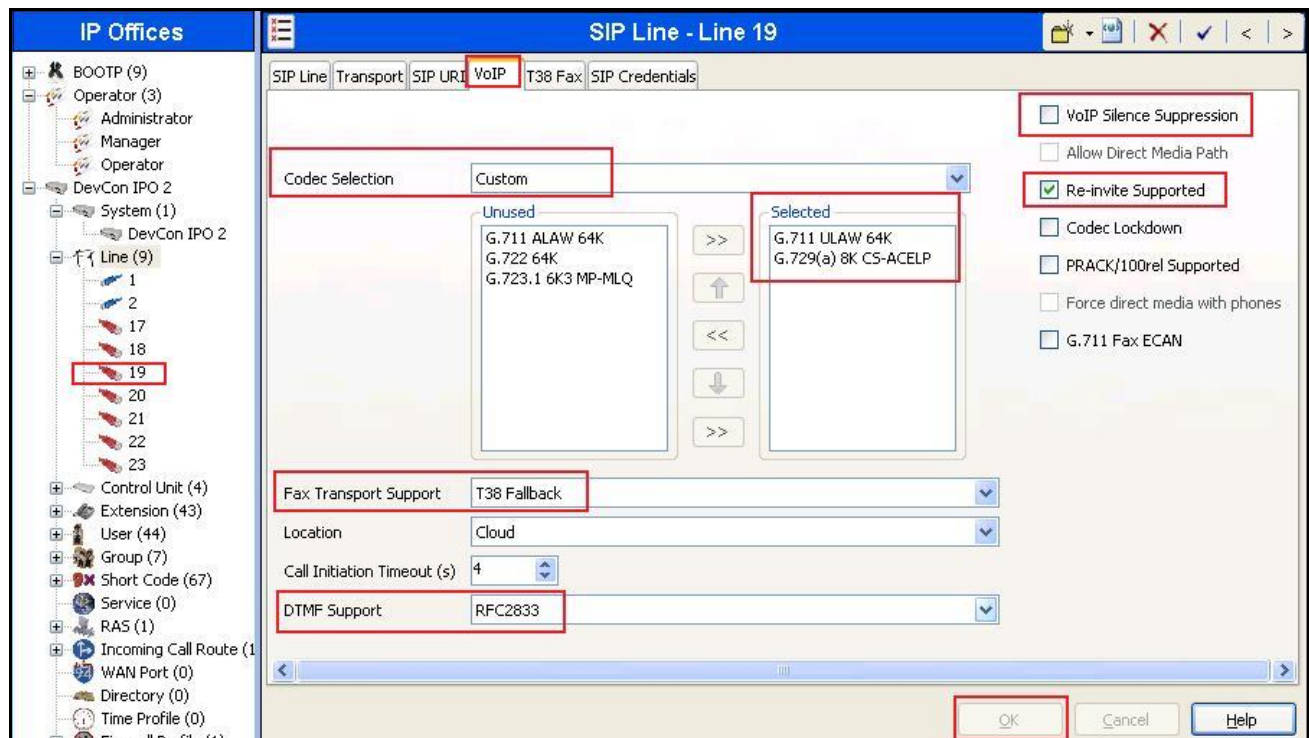
SIP URI entry for **Channel 2**



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

- The **Codec Selection** can be selected by choosing **Custom** from the pull-down menu, allowing an explicit ordered list of codecs to be specified. Select **G.711 ULAW 64K** as first choice and **G.729(a) 8K CS-ACELP** as second choice of codecs, cause Avaya IP Office to include these codes, supported by the Frontier SIP Trunking service, in the Session Description Protocol (SDP) offer, in that order.
- Set **Fax Transport Support** to **T.38 Fallback** from the pull-down menu.
- Set the **DTMF Support** field to **RFC2833** from the pull-down menu. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Uncheck the **VoIP Silence Suppression** box. By unchecking the **VoIP Silence Suppression** box, calls can be established with the G.729 codec but without silence suppression.
- Check the **Re-invite Supported** box.
- Check the **Re-invite Supported** box.
- Default values may be used for all other parameters.

Click **OK**.



5.5. Short Code

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

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **9N;**, this short code will be invoked when the user dials 9 followed by any number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N”@192.168.248.132”**. 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 host part following the “@” is the domain of the service provider network.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.4**. This short code will use this line group when placing the outbound call.
- Set **Locale** to **United States (US English)**.

Click **OK** to commit.

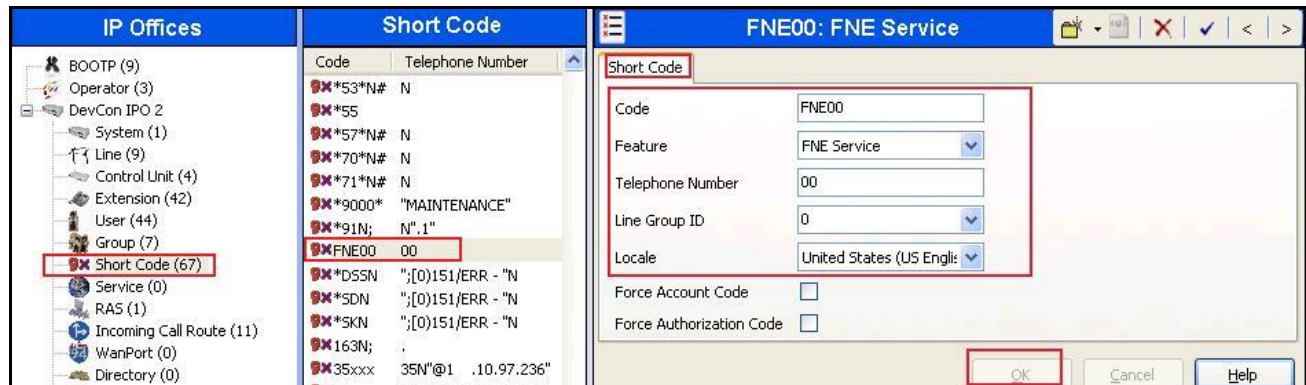
The screenshot displays the Avaya SIP Office configuration interface. On the left, the 'IP Offices' tree shows the 'Short Code (67)' item selected. The main area is divided into two panes. The left pane, titled 'Short Code', contains a table with columns 'Code' and 'Telephone Number'. The right pane, titled '9N;: Dial', shows the configuration for the selected short code. The configuration fields are as follows:

Field	Value
Code	9N;
Feature	Dial
Telephone Number	N”@192.168.248.132”
Line Group ID	19
Locale	United States (US English)
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

At the bottom of the configuration window, there are three buttons: 'OK', 'Cancel', and 'Help'. The 'OK' button is highlighted with a red box.

For incoming calls from mobility extension to FNE features hosted by IP Office to provide **Dial Tone** functionality, Short Code **FNE00** was created. The **FNE00** was configured with the following parameters.

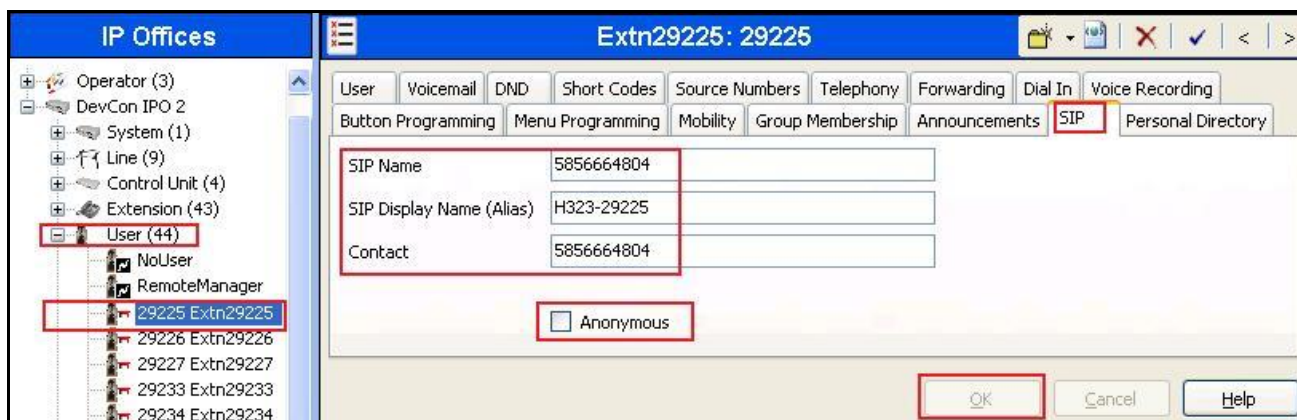
- In the **Code** field, enter the FNE feature code as **FNE00** for **Dial Tone**.
- Set the **Feature** field to **FNE Service**.
- Set the **Telephone Number** field to **00** for **FNE00**.
- Set the **Line Group ID** field to **0**.
- Set **Locale** to **United States (US English)**.
- Retain default values for other fields.
- Click **OK**.



5.6. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.4**. To configure these settings, first select **User** in the left Navigation Pane, then select the name of the user to be modified in the center Group Pane. In the example below, the name of the user is "H323-29225". 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 header for outgoing SIP trunk calls. They also allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.4**). The example below shows the settings for user H323-29225. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise from Frontier. 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 **OK** to commit.



One of the H.323 IP Phones at the enterprise site uses the Mobile Twinning feature. The following screen shows the **Mobility** tab for User H323-29225. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case 96139675281. Other options can be set according to customer requirements. Click **OK**.



5.7. Incoming Call Route

An incoming call route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by the service provider. To create an incoming call route, select **Incoming Call Route** in the left Navigation Pane, then right-click in the center Group Pane and select **New** (not shown). 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 5.4**.
- Set the **Incoming Number** to the incoming number on which this route should match.
- Set **Locale** to **United States (US English)**
- Default values can be used for all other fields.
- Click **OK**.

The screenshot shows the 'Incoming Call Route' configuration window for the number '19 5856664804'. The 'Standard' tab is active. The left pane shows the 'IP Offices' hierarchy with 'Incoming Call Route (11)' selected. The main area contains the following fields:

Bearer Capacity	Any Voice
Line Group ID	19
Incoming Number	5856664804
Incoming Sub Address	
Incoming CLI	
Locale	United States (US English)
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

Buttons at the bottom: OK, Cancel, Help.

On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to 585-666-4804 on line 19 are routed to extension **29225**. Click **OK**.

The screenshot shows the 'Incoming Call Route' configuration window for the number '19 5856664804', now on the 'Destinations' tab. The left pane is the same. The main area shows a table with one row:

TimeProfile	Destination	Fallback Extension
Default Value	29225 Extn29225	

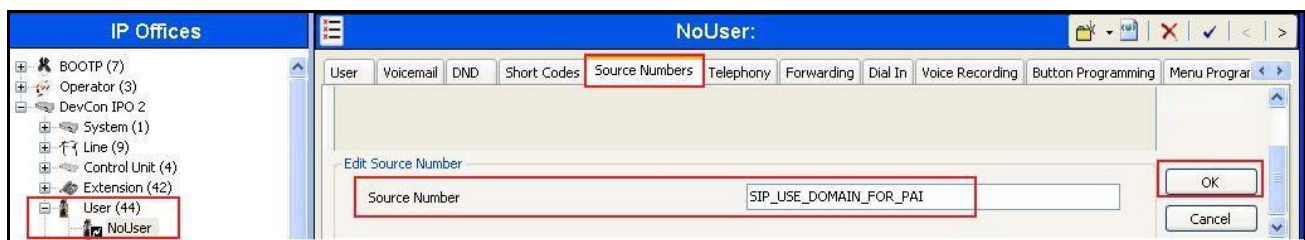
Buttons at the bottom: OK, Cancel, Help.

5.8. Privacy/Anonymous Calls

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

To configure Avaya IP Office to use PAI for privacy calls, navigate to **User → NoUser** in the Navigation / Group Panes. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button (not shown).

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



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



5.9. SIP Options

Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.1** and the **SIP_OPTIONS_PERIOD** parameter (in minutes) that can be set on the **Source Number** tab of the **noUser** user. The OPTIONS period is determined in the following manner:

- If no **SIP_OPTIONS_PERIOD** parameter is defined and the **Binding Refresh Time** is **0**, then the default value of 300 seconds is used.
- To establish a period less than 300 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 300 seconds. The OPTIONS message period will be equal to the **Binding Refresh Time**.
- To establish a period greater than 300 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 300 seconds. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD**.

To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User → noUser** (not shown) in the Navigation / Group Panes. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button (not shown). At the bottom of the Details Pane, the **Source Number** field will appear. Enter **SIP_OPTIONS_PERIOD=X**, where **X** is the desired value in minutes. Click the **OK** button (not shown).

In this testing, **SIP_OPTIONS_PERIOD** parameter was not defined in order for IPO to send SIP options messages periodically every **60** seconds, which is the **Binding Refresh Time** value set in **Section 5.1**.

5.10. Save Configuration

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

6. Frontier SIP Trunking Configuration

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

- IP address of the Frontier SIP proxy.
- Supported codecs
- DID numbers
- IP addresses and port numbers used for signaling or media through any security devices.

7. Verification Steps

The following steps may be used to verify the configuration:

- Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application from **Start → Programs → IP Office → System Status** on the PC where Avaya IP Office Manager was installed. Select the SIP line of interest from the left pane. On the **Status** tab in the right pane, verify that the **Current State** is *Idle* for each channel (assuming no active calls at present time).

The screenshot shows the Avaya IP Office System Status application. The left pane lists various system components, with 'Line: 19' selected under 'Trunks (9)'. The main pane displays the 'SIP Trunk Summary' for Line 19. The summary includes the following information:

- Peer Domain Name: 192.168.248.132
- Resolved Address: 192.168.248.132
- Line Number: 19
- Number of Administered Channels: 20
- Number of Channels in Use: 0
- Administered Compression: G711 Mu, G729 A
- Silence Suppression: Off
- Layer 4 Protocol: UDP
- SIP Trunk Channel Licenses: Unlimited
- SIP Trunk Channel Licenses in Use: 0
- SIP Device Features: REFER (Incoming and Outgoing), UPDATE (Incoming and Outgoing)

A green circle indicates 0% utilization. Below the summary is a table showing the status of the channels:

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 Packe...
1			Idle	01:02:57											
2			Idle	01:02:57											
3			Idle	01:02:57											
4			Idle	01:02:57											

At the bottom of the main pane, there are buttons for 'Trace', 'Trace All', 'Pause', 'Ping', 'Call Details', 'Print...', and 'Save As...'. The status bar at the bottom right shows '4:18:22 PM' and 'Online'.

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

The screenshot shows the Avaya IP Office System Status application with the 'Alarms' tab selected. The main pane displays 'Alarms for Line: 19 SIP 192.168.248.132'. Below this, there is a table with the following columns:

Last Date Of Error	Occurrences	Error Description
--------------------	-------------	-------------------

At the bottom of the main pane, there are buttons for 'Ping', 'Clear', 'Clear All', 'Print...', and 'Save As...'. The status bar at the bottom right shows '9:54:40 AM' and 'Online'.

- Verify that a phone connected to PSTN can successfully place a call to the Avaya IP Office with two-way audio.
- Verify that a phone connected to Avaya IP Office can successfully place a call to the PSTN with two-way audio.
- Using a network sniffing tool e.g. Wireshark to monitor the SIP signalling between the enterprise and Frontier. The sniffer traces are captured at the public interface of the Avaya IPO.

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

- Incoming INVITE request from Frontier.

```
INVITE sip:5856664804@135.10.98.113:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 173.84.248.132:5060;branch=z9hG4bKfjna8h10copgcp0j23s0.1
Allow-Events: message-summary, refer, dialog, line-seize, presence, call-info, as-
feature-event
Max-Forwards: 69
Call-ID: C06B24E6@10.160.139.4
From:
<sip:01101116139675258@10.160.139.4:5060>;tag=10.160.139.4+1+1d4804+2708a7d1;isup-
oli=00
To: <sip:5856664804@135.10.98.113>
CSeq: 32904857 INVITE
Expires: 180
Organization: MetaSwitch
Supported: resource-priority, 100rel
Content-Length: 140
Content-Type: application/sdp
Contact: <sip:01101116139675258@173.84.248.132:5060;transport=udp>;isup-oli=00
Alert-Info: <http://www.notused.com>;info=alert-internal
P-Asserted-Identity: <sip:01101116139675258@10.160.139.4:5060>

v=0
o=- 2955450815 2955450815 IN IP4 173.84.248.132
s=-
c=IN IP4 173.84.248.132
t=0 0
m=audio 57206 RTP/AVP 0
a=ptime:20
a=sendrecv
```

- Outgoing 200OK response from the enterprise.

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 173.84.248.132:5060;branch=z9hG4bKfjna8h10copgcp0j23s0.1
From:
<sip:01101116139675258@10.160.139.4:5060>;tag=10.160.139.4+1+1d4804+2708a7d1;isup-
oli=00
Call-ID: C06B24E6@10.160.139.4
CSeq: 32904857 INVITE
Contact: "H323-29225" <sip:5856664804@135.10.98.113:5060;transport=udp>
Allow: INVITE,ACK,CANCEL,OPTIONS,BYE,INFO,REFER,NOTIFY,UPDATE
Supported: timer
Server: IP Office 9.0.3.0 build 941
To: <sip:5856664804@135.10.98.113>;tag=b8a18b11dedd4b5b
Content-Type: application/sdp
Content-Length: 148

v=0
o=UserA 344782133 965829690 IN IP4 135.10.98.113
s=Session SDP
c=IN IP4 135.10.98.113
t=0 0
m=audio 49154 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

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

- Outgoing INVITE request from the enterprise.

```
INVITE sip:6139675281@173.84.248.132 SIP/2.0
Via: SIP/2.0/UDP
135.10.98.113:5060;rport;branch=z9hG4bK9c9d07ded7923bbde42fd1069c6f1941
From: "H323-29225" <sip:5856664804@173.84.248.132>;tag=91a269887e293b20
To: <sip:6139675281@173.84.248.132>
Call-ID: a24ab5298a5bd54e54eb80f7f7dd4888
CSeq: 742880383 INVITE
Contact: "H323-29225" <sip:5856664804@135.10.98.113:5060;transport=udp>
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,OPTIONS,BYE,INFO,REFER,NOTIFY,UPDATE
Content-Type: application/sdp
Supported: timer
User-Agent: IP Office 9.0.3.0 build 941
P-Asserted-Identity: "H323-29225" <sip:5856664804@173.84.248.132:5060>
Content-Length: 275

v=0
o=UserA 48792568 2664751453 IN IP4 135.10.98.113
s=Session SDP
c=IN IP4 135.10.98.113
t=0 0
m=audio 49154 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

- Incoming 200OK response from Frontier.

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
135.10.98.113:5060;received=135.10.98.113;rport;branch=z9hG4bK9c9d07ded7923bbde42fd106
9c6f1941
From: "H323-29225" <sip:5856664804@173.84.248.132>;tag=91a269887e293b20
To: <sip:6139675281@173.84.248.132>;tag=10.160.139.4+1+752110+44aa4a92
Call-ID: a24ab5298a5bd54e54eb80f7f7dd4888
CSeq: 742880383 INVITE
Server: DC-SIP/2.0
Organization: MetaSwitch
Allow-Events: message-summary, refer, dialog, line-seize, presence, call-info, as-
feature-event
Supported: resource-priority, 100rel
Allow: INVITE, ACK, CANCEL, BYE, REGISTER, OPTIONS, PRACK, UPDATE, SUBSCRIBE, NOTIFY,
REFER, INFO, PUBLISH
Accept-Encoding: identity
Accept: application/sdp, application/simple-message-summary, message/sipfrag,
application/isup, application/x-simple-call-service-info, multipart/mixed,
application/broadsoft, application/vq-rtpcpxr, application/media_control+xml,
application/dtmf-relay, text/plain, application/x-as-feature-event+xml
Contact: <sip:6139675281@173.84.248.132:5060;transport=udp>
Content-Length: 179
Content-Type: application/sdp

v=0
o=- 2958339575 2958339575 IN IP4 173.84.248.132
s=-
c=IN IP4 173.84.248.132
t=0 0
m=audio 57392 RTP/AVP 0 101
a=rtpmap:101 telephone-event/8000
a=ptime:20
a=sendrecv
```

8. Conclusion

Interoperability testing of the sample configuration was completed with successful results for all test cases with the exception of the observations/limitations described in **Section 2.2**.

The Frontier SIP Trunking passed compliance testing. These Application Notes describe the procedures required to configure the SIP connection between Avaya IP Office and the Frontier SIP Trunking service as shown in **Figure 1**.

9. Additional References

- [1] IP Office 9.0 Installation, Document number 15-601042 Issue 28, 11 October 2013
- [2] IP Office 9.0 Manager 9.0, Document number 15-601011 Issue 9.01, 09 September 2013
- [3] IP Office 9.0 Administering Voicemail Pro, Document number 15-601063 Issue 9.0 Release 1.0, September 2013
- [4] IP Office Embedded Voicemail User Guide (IP Office Mode), Document number 15-604067 Issue 9.0, 10 September 2013

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

<http://marketingtools.avaya.com/knowledgebase/>

Product documentation for Frontier SIP Trunking is available from Frontier.

<https://frontier.com/enterprise>

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