Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Bell Canada SIP Trunking with Avaya IP Office 9.0.4 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Bell Canada Session Initiation Protocol (SIP) Trunking with Avaya IP Office Release 9.0.4.

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

Bell Canada 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 Bell Canada (with load balancer) and Avaya IP Office solution. In the sample configuration, Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 9.0.4, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, SIP, digital, and analog endpoints.

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

The Bell Canada SIP Trunking service uses Digest Authentication for outbound calls from the enterprise, using challenge-response authentication for each call to the Bell Canada network based on a configured user name and password (provided by Bell Canada and configured in Avaya IP Office). This call authentication scheme as specified in SIP RFC 3261 provides security and integrity protection for SIP signaling.

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 Bell Canada SIP Trunking service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality tests listed in **Section 2.1**.

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

2.1. Interoperability Compliance Testing

A simulated enterprise site with Avaya IP Office was connected to Bell Canada 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, SIP, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types. Phone types included H.323, SIP, 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 mode.
- Off-net call forwarding.
- Twinning to mobile phones on inbound calls.

2.2. Test Results

Bell Canada SIP Trunking passed compliance testing.

Items not supported or not tested included the following:

- Inbound toll-free is supported but was not tested as part of the compliance test.
- T.38 Fax is not supported.
- Call redirection (Blind/Consultative transfer) using REFER is not supported.

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

- **OPTIONS from IP Office** – If the periodicity of sending OPTIONS to monitor the SIP trunk connectivity is configured on Avaya IP Office to be larger than the periodicity of OPTIONS from Bell Canada, IP Office would effectively cease to send OPTIONS to Bell Canada. This is expected behavior on Avaya IP Office since it resets the timer for sending OPTIONS starting from the most recently received OPTIONS from the network.

2.3. Support

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

For technical support on Bell Canada SIP Trunking, contact Bell Canada at http://www.bell.ca/enterprise/EntPrd_SIP_Trunking.page.

3. Reference Configuration

Figure 1 below, illustrates the test configuration. The test configuration shows an enterprise site connected to Bell Canada 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 and Avaya IP Office Softphone. A separate Windows XP PC runs Avaya IP Office Manager to configure and administer Avaya IP Office.

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

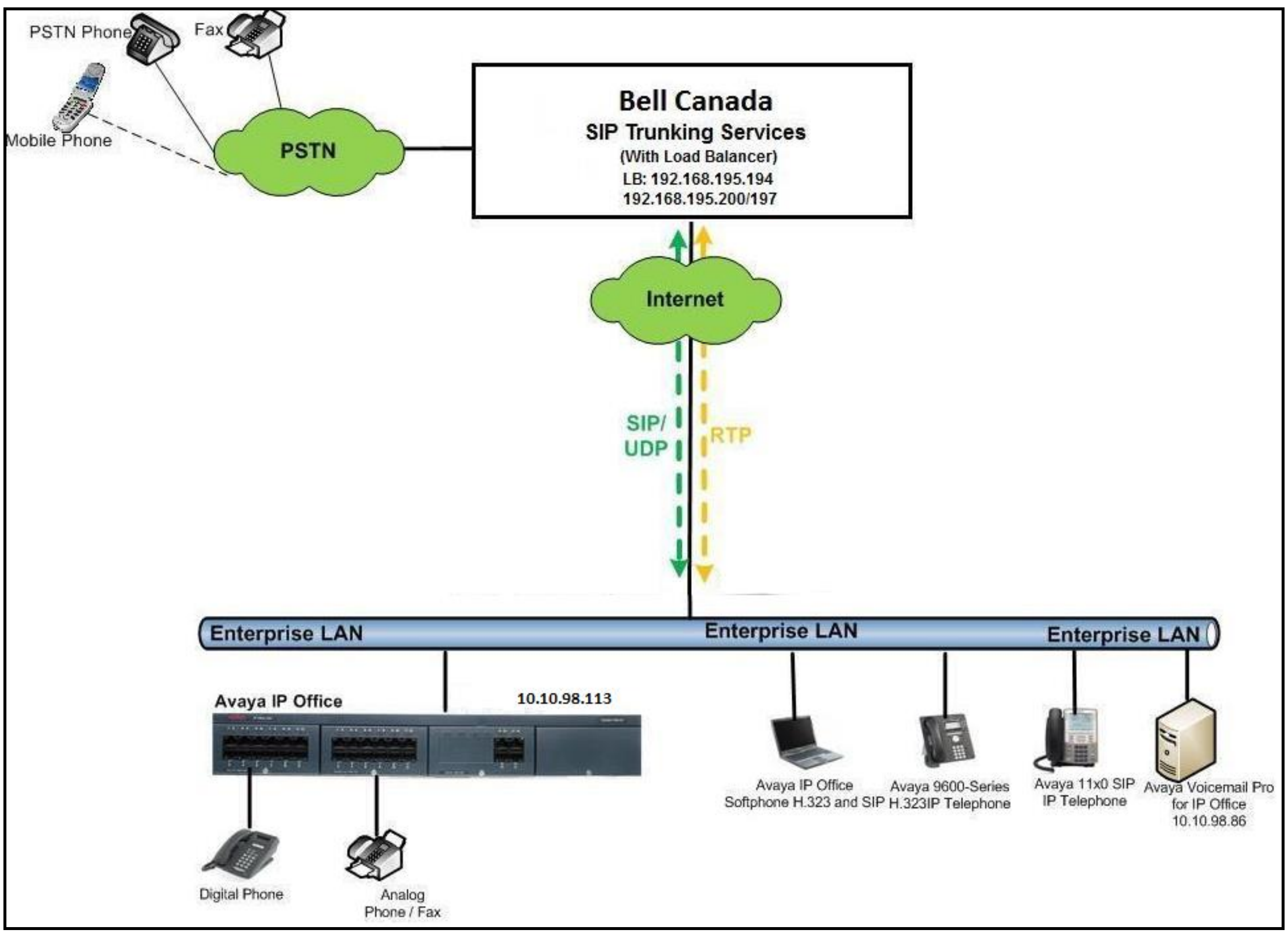


Figure 1: Test Configuration for Avaya IP Office with Bell Canada SIP Trunking Service

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 Bell Canada. The short code of 9 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to Bell Canada. 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, Bell Canada SIP Trunking sent 10 digits in the Request URI and the To field of inbound SIP INVITE messages.

Bell Canada uses the phone number in the From header of a SIP INVITE message to authenticate the calling party. Thus, a call will be rejected by the network unless the From header contains a number known to Bell Canada. This is especially important for calls inbound from the PSTN which are redirected back to the PSTN by call forwarding or twinning. For call forwarding, Avaya IP Office sends the number of the forwarding phone in the From header. This is a number known to Bell Canada. As a result, the call display on the destination phone shows the forwarding party not the original caller. For twinning, this behavior can be slightly altered through configuration. See **Sections 5.3** and **5.4** for details.

Note that the calling party authentication using the phone number by Bell Canada, as mentioned above, is in addition to the Digest Authentication by Bell Canada during call setup SIP signaling exchanges using a user name and password as configured in Avaya IP Office for all calls from the enterprise to Bell Canada.

Additionally, IP Office WAN interface is connecting to Bell’s load balancer over the internet SIP trunk for outbound call from the enterprise to PSTN via single IP address. For inbound from PSTN to enterprise, calls will coming in to enterprise via two IP addresses as shown in **Figure 1**.

4. Equipment and Software Validated

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

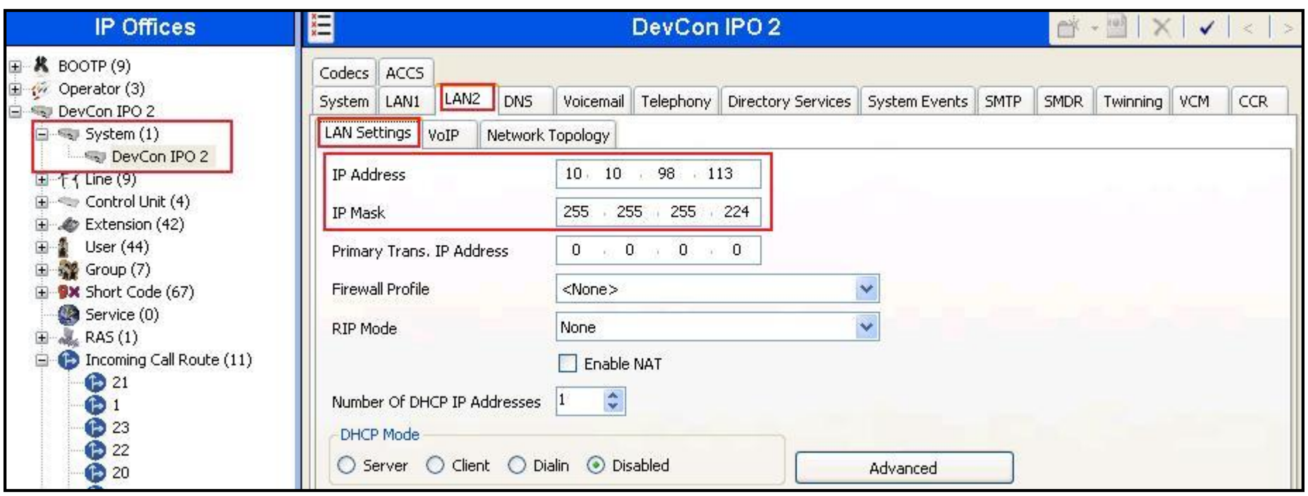
Avaya Telephony Components	
Equipment	**Release**
Avaya IP Office 500v2	9.0.4.00.965
Avaya IP Office Manager	9.0.4.00.965
Avaya Voicemail Pro for IP Office	9.0.4.00.965
Avaya 11x0 IP Telephone (SIP)	SIP11x0e04.03.12.00
Avaya 9630G IP Telephone (H.323)	Avaya one-X® Deskphone Edition S3.2
Avaya IP Office Softphone	3.2.3.20 64770
Avaya Digital Telephone (9508)	N/A
Avaya Symphony 2000 Analog Telephone	N/A
Bell Canada SIP Trunking Service Components	
Component	Release
F5 Load Balancer	11
Oracle ACME Packet Net-Net 4500	6.3.7 MR-3 Patch 1
BroadSoft Broadworks	18
Legacy Nortel CS2K Media Gateway	SN10 PVG/IW-SPM

5. Configure IP Office

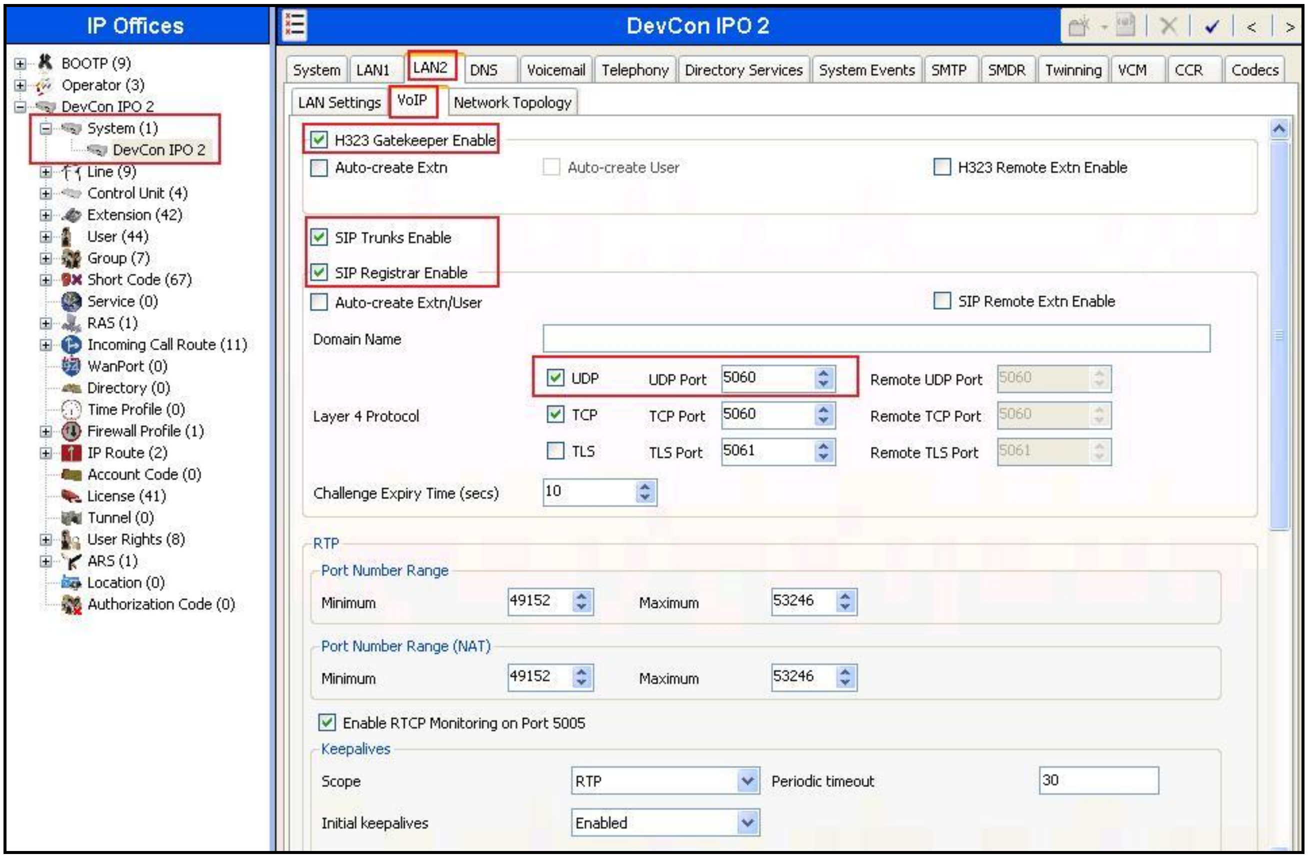
This section describes Avaya IP Office configuration to support connectivity to Bell Canada SIP Trunking service. Avaya IP Office is configured through Avaya IP Office Manager PC application. From a PC running Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch 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 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 Avaya IP Office to the public network. The LAN1 settings correspond to the WAN port on 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. Click **OK**.

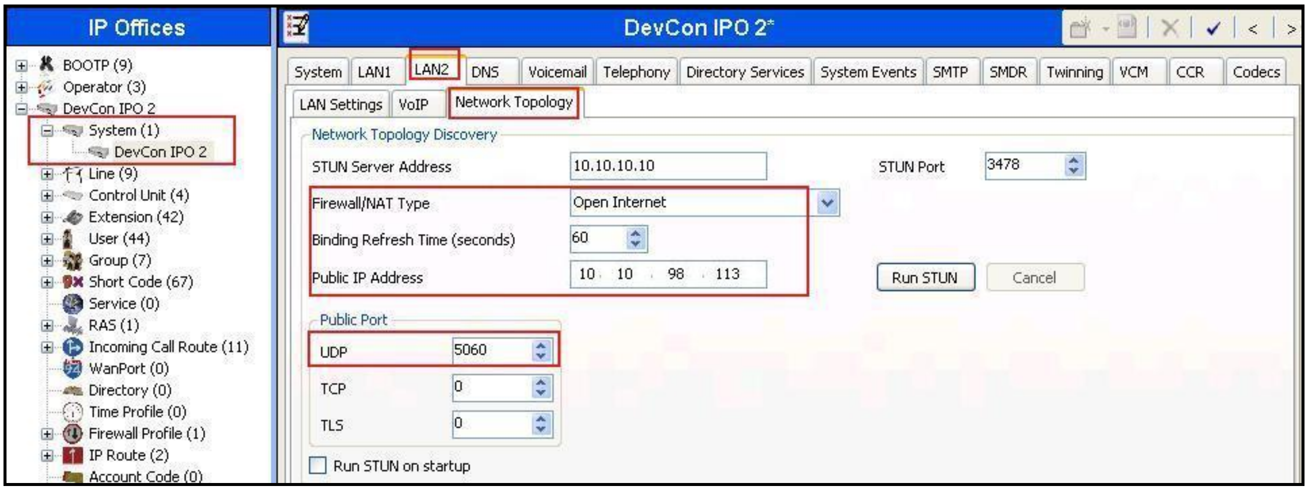


Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such the 9600-Series IP Telephones used in the sample configuration. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Bell Canada. 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**.** Click **OK.**



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

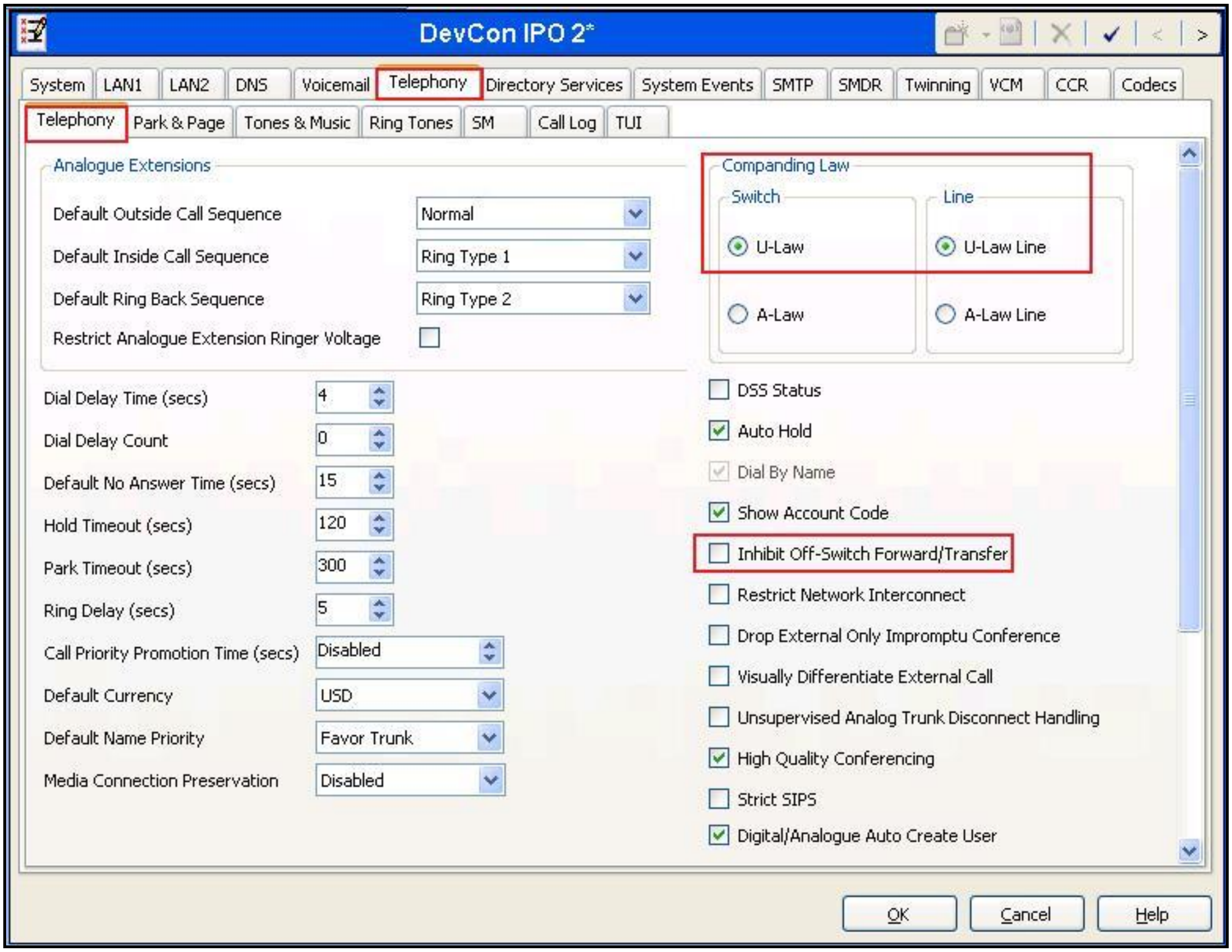
- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to ***Open Internet***. With this configuration, STUN will not 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 5.10** for complete details.
- Set **Public IP Address** to the IP address of Avaya IP Office WAN port. **Public Port** is set to **5060.**
- All other parameters should be set according to customer requirements.
- Click **OK**.



In the compliance test, the LAN1 interface was used to connect Avaya IP Office to the enterprise site IP network. The LAN1 interface configuration is not directly relevant to the interface with Bell Canada 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, ***ULAW*** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the service provider across the SIP trunk. 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 affects 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 **Send Caller ID** parameter that was set to **Diversion** 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**.

5.4. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and Bell Canada 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:

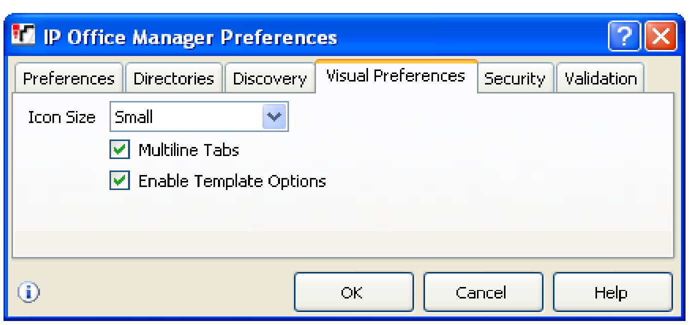
- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required

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 **CA_Bell Canada_SIPTrunk.xml**. The file name is important in locating the proper template file in **Step 5**.

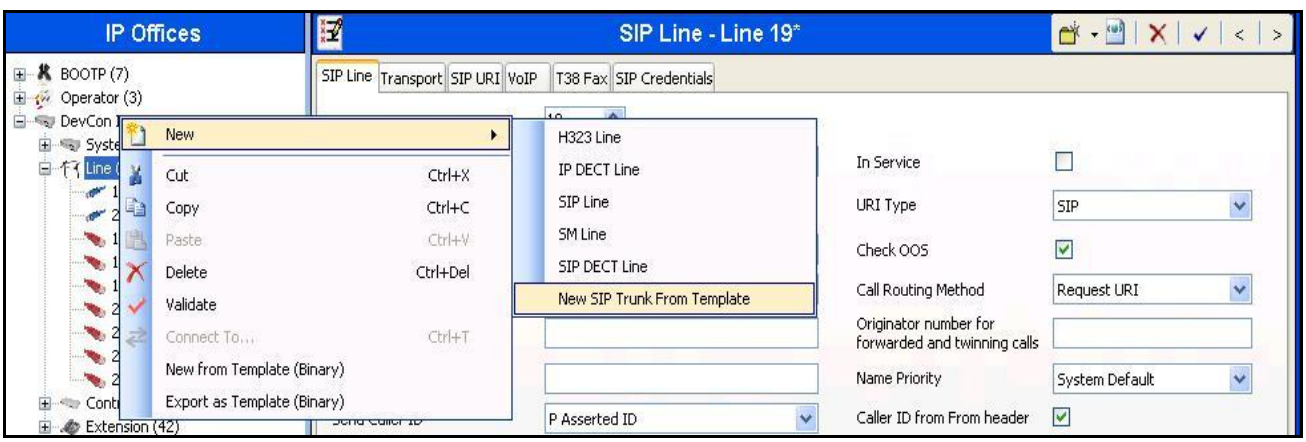
2. Verify that template options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the IP Office Manager Preferences window that appears, select the Visual Preferences tab. Verify that the box is checked next to **Enable Template Options**. Click **OK**.



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

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

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



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

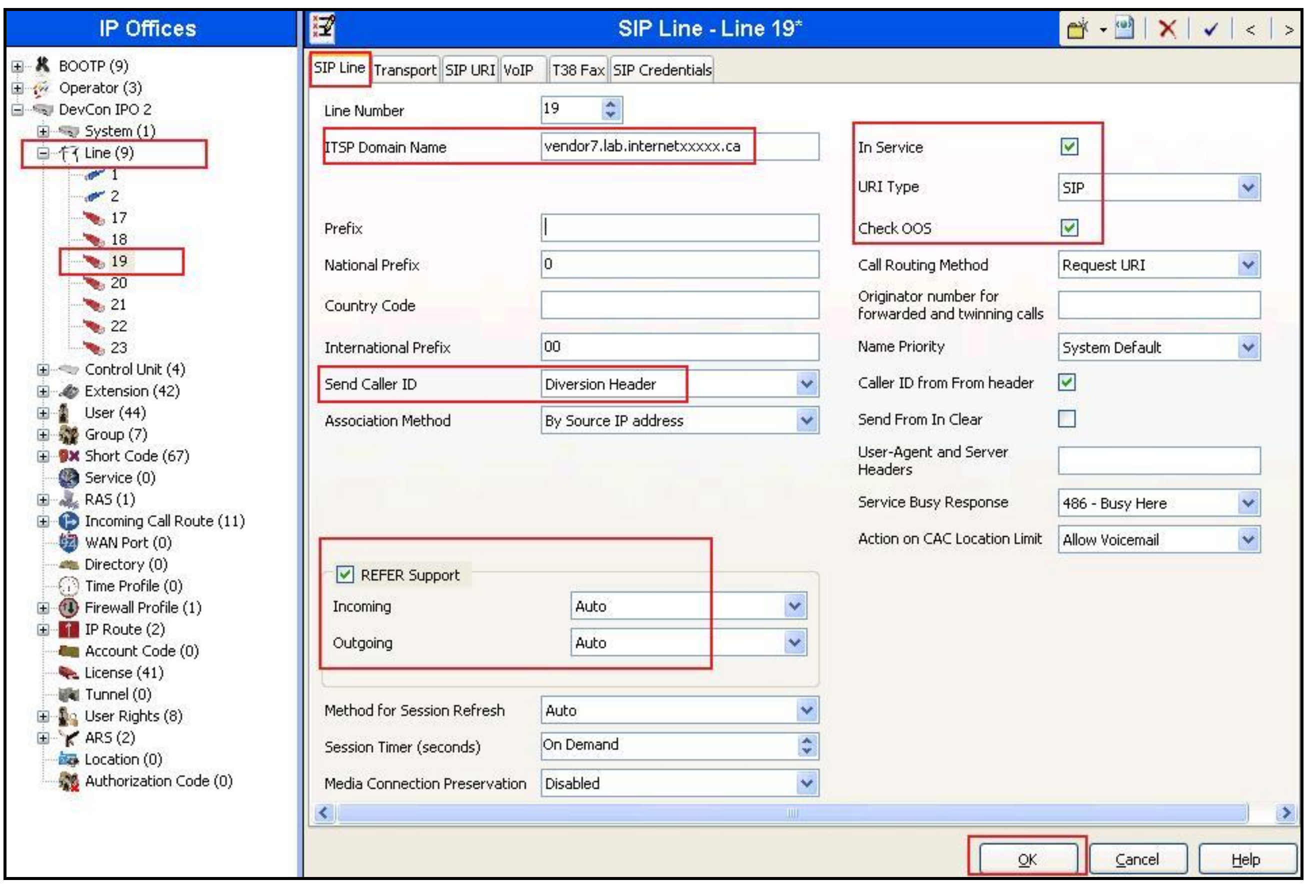
6. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.4.2.**

5.4.2. Create SIP Line Manually

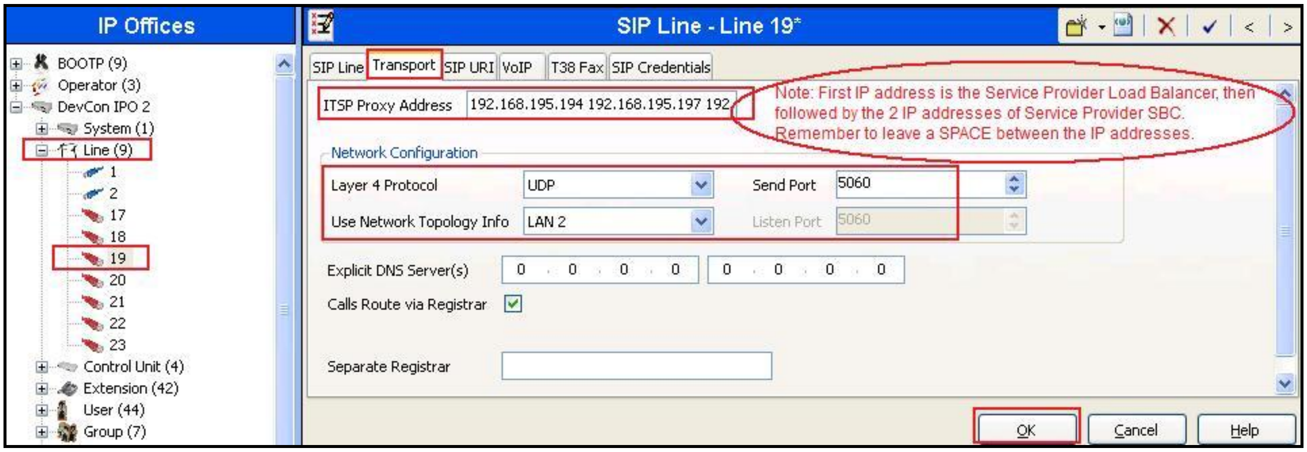
To create a SIP line, begin by navigating to **Line** in the left Navigation Pane, then right-click in the Group Pane and select **New → SIP Line**. On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

- 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***. For the compliance test, this parameter was ignored since **Send original calling party information for Mobile Twinning** is optioned in **Section 5.3**.
- 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.
- 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, REFER method is not supported. Therefore, the value "***Auto***" is set for **Incoming** and **Outgoing** parameters. Click **OK**.

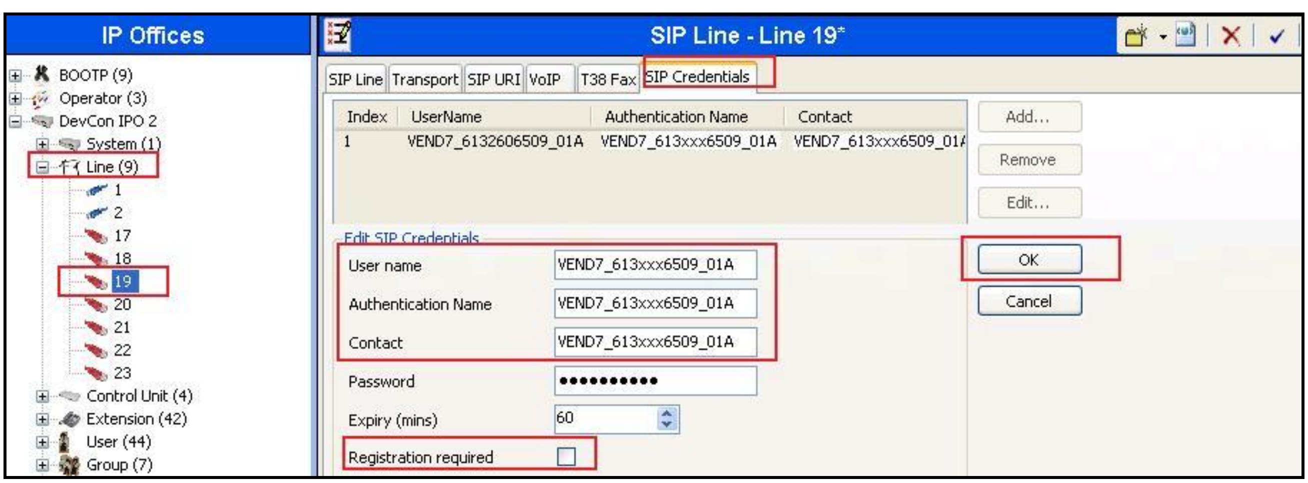


Select the **Transport** tab. The **ITSP Proxy Address** is set to internal provided IP Address of Bell Load Balancer and other 2 IP addresses of Bell's SBC. In the **Network Configuration** area, ***UDP*** is selected as the **Layer 4 Protocol**, and the **Send Port** is set to the port number of Avaya IP Office. 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. Click **OK**.



A SIP Credentials entry must be created for Digest Authentication used by Bell Canada SIP trunking service to authenticate calls from the enterprise to the PSTN. To create a SIP Credentials entry, first select the **SIP Credentials** tab. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit…** button. In the bottom of the screen, the Edit Channel area will be opened. In the example screen below, a previously configured entry is edited. The entry was created with the parameters shown below:

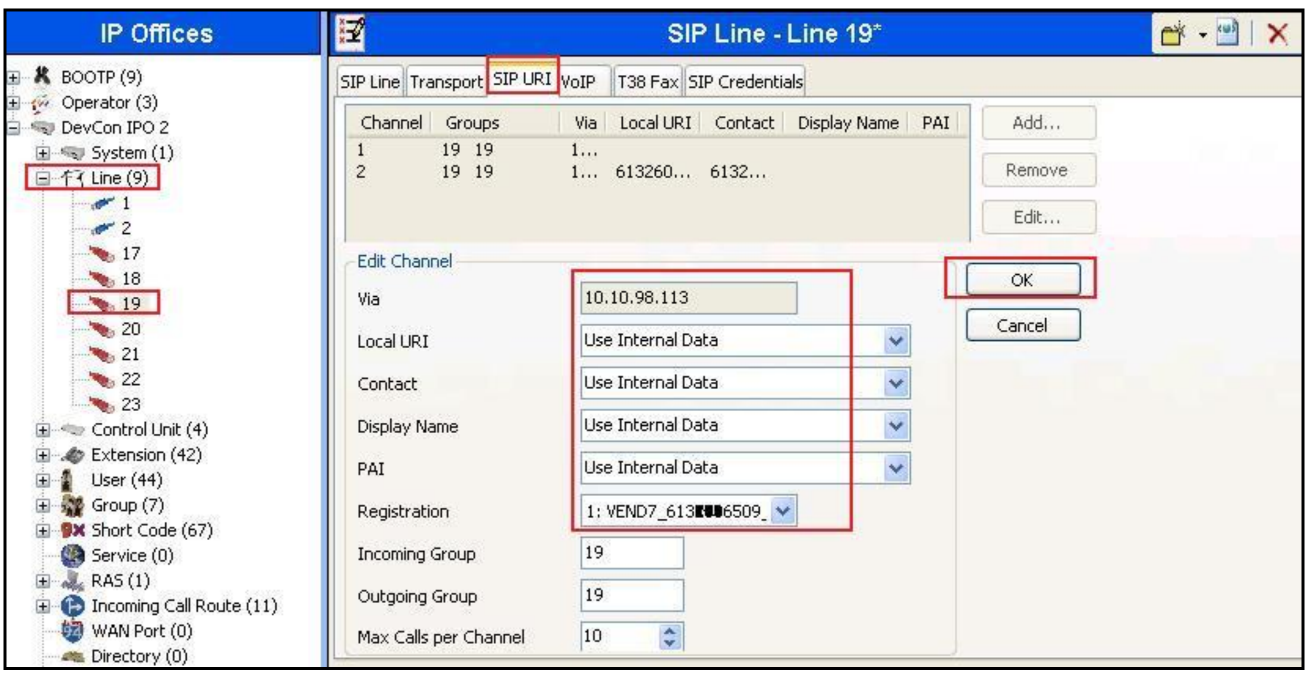
- Set **User name** and **Authentication Name** to the value provided by the service provider.
- Set **Password** to the value provided by the service provider. **Expiry (mins)** is set to **60**
- Uncheck the **Registration required** option. Bell Canada does not require registration for Digest Authentication.
- 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 ***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**.
- For **Registration**, select the account credentials previously configured on the line's **SIP Credentials** tab.
- 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**.

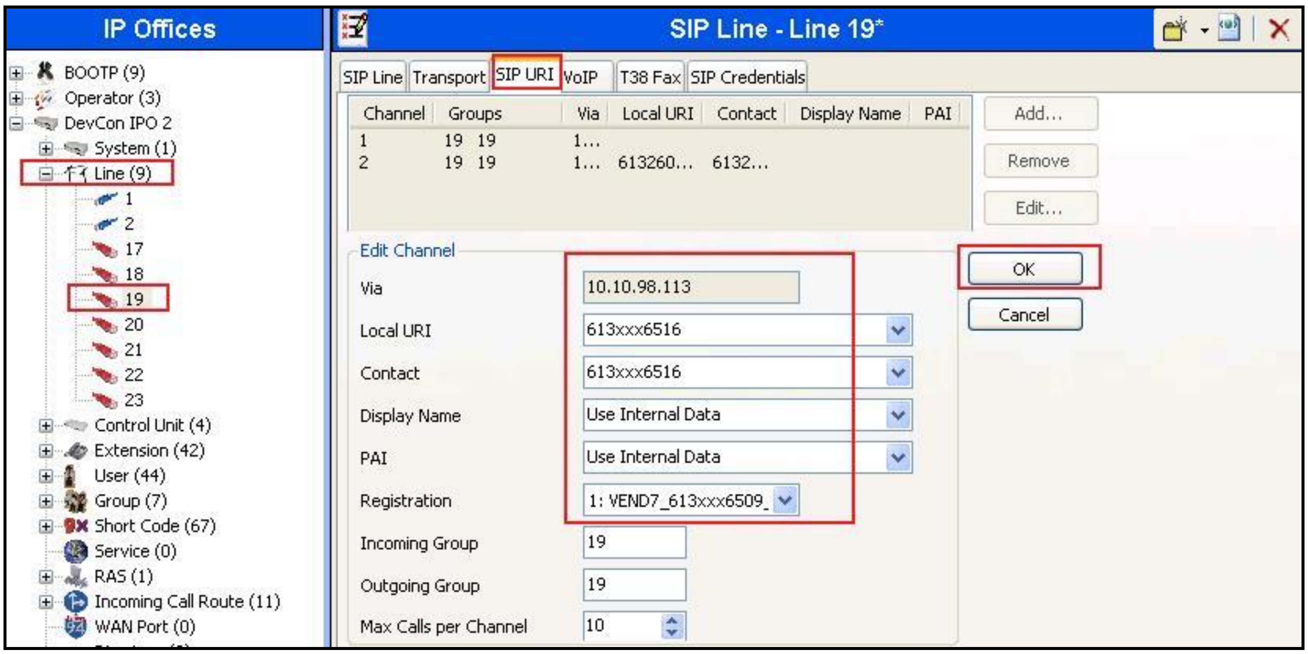
SIP URI entry for **Channel 1**



SIP URI entry **Channel 2** was similarly created for incoming calls appropriately to pre-define DID numbers **613xxx6516** 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. The **Channel 2,** as shown in the screenshot below, was configured with following parameters.

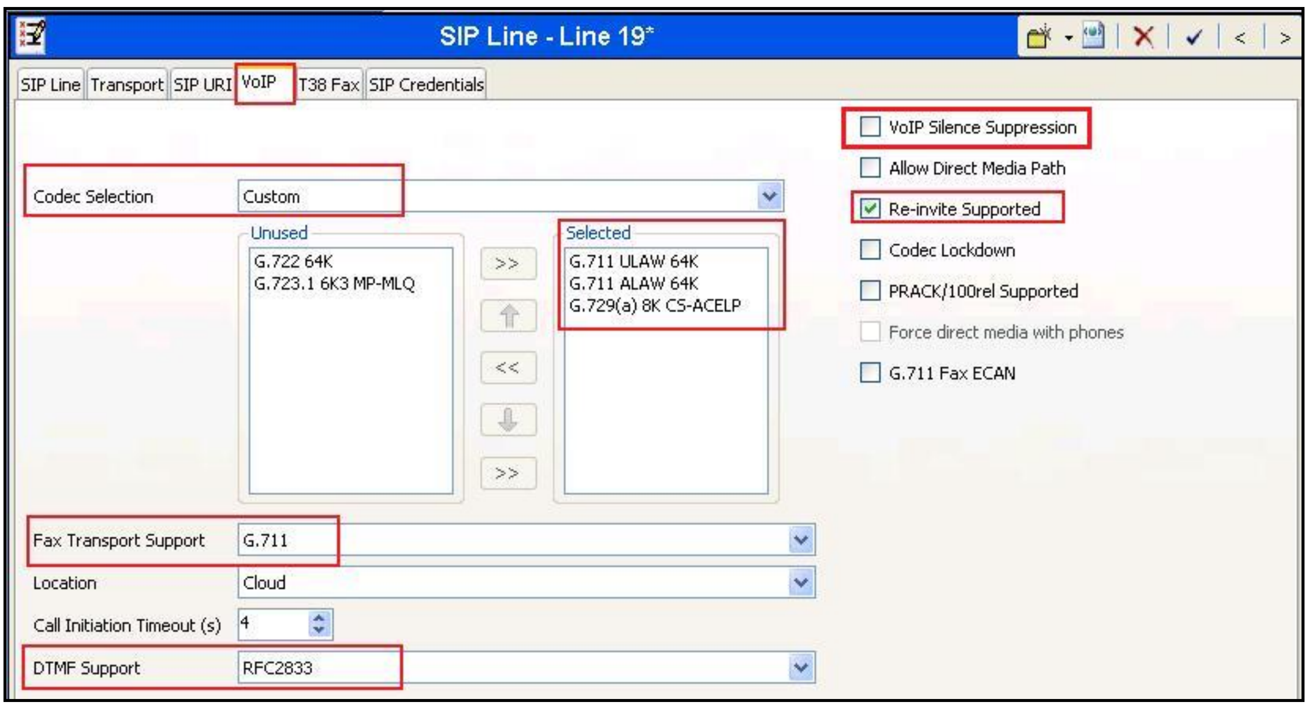
- Set the **Local URI** and **Contact** fields to pre-define DID number **613xxx6516** appropriately for **Channel 2**.
- Associate **Incoming Group** and **Outgoing Group** to SIP Line 19.
- Set the **Max Calls per Channel** field to **10**.
- Other parameters retain default values.
- Click **OK**.

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:

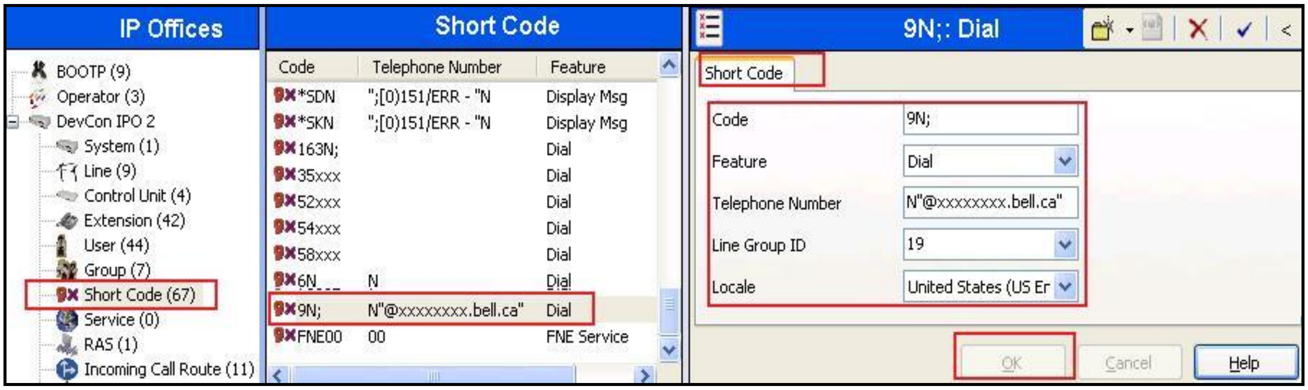
- The **Codec Selection** can be selected by choosing **Custom** from the pull-down menu, allowing an explicit ordered list of codecs to be specified. Select ***G.711 ULAW 64K***and ***G.729(a) 8K CS-ACELP*** codecs cause Avaya IP Office to include these codes, supported by the Bell Canada SIP Trunking service, in the Session Description Protocol (SDP) offer, in that order.
- Set **Fax Transport Support** to **G711** from the pull-down menu (T.38 faxing is not currently supported by Bell Canada).
- 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.
- Default values may be used for all other parameters**.**
- Click **OK.**



5.5. Short Code

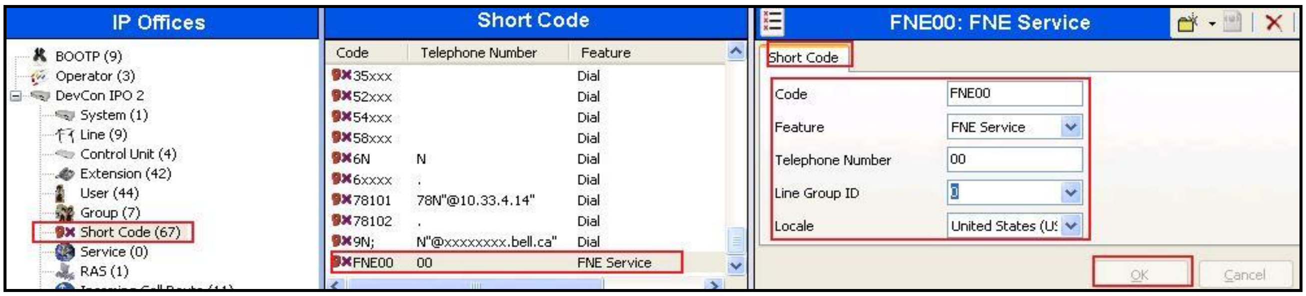
Define a short code to route outbound traffic to the SIP line. To create a short code, select **Short Code** in the left Navigation Pane, then right-click in the Group Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters for the new short code to be created. 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"@xxxxxxxx.bell.ca"***. 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 State (US English)**.
- Click **OK**.



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.

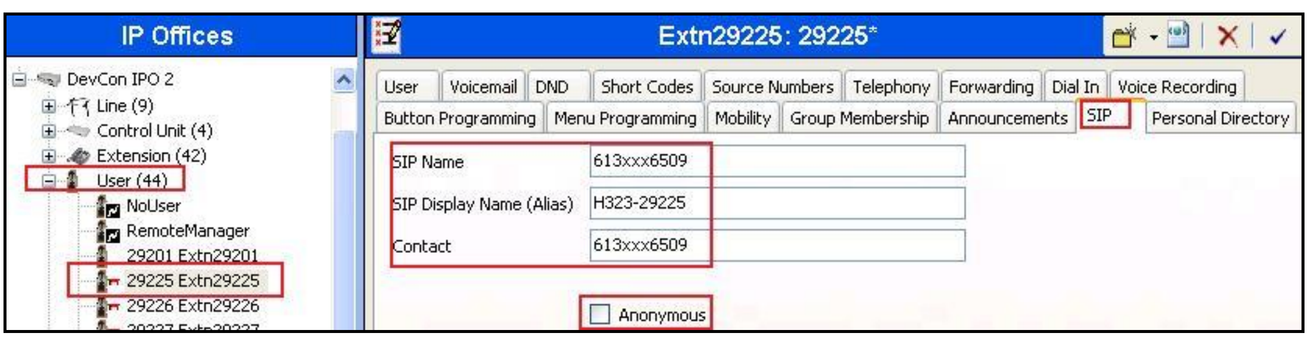
- In the **Code** field, enter the FNE feature code as **FNE00** for **Dial Tone**.
- Set the **Feature** field to **FNE Service**.
- Set the **Telephone Number** field to **00** for **FNE00**.
- Set the **Line Group ID** field to **0**.
- Retain default values for other fields.
- Click **OK**.



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 Bell Canada. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user’s information from the network.

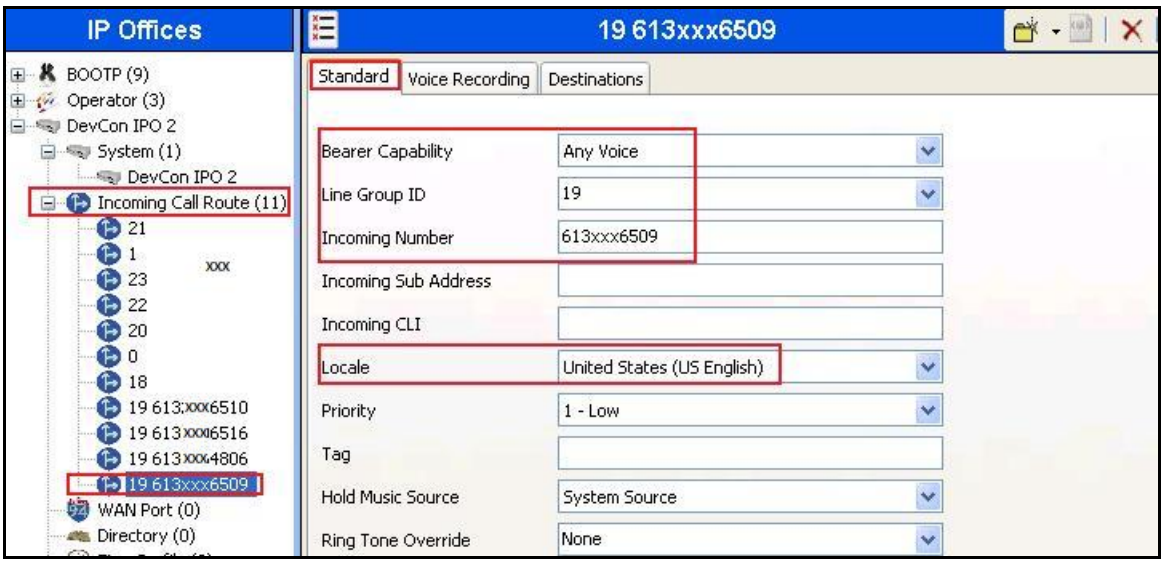


One of the H.323 IP Phones at the enterprise site uses the Mobile Twinning feature. The following screen shows the **Mobility** tab for User H323-29225. The **Mobility Features** and **Mobile Twinning** boxes are checked**.** The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case 916139675279. Other options can be set according to customer requirements.

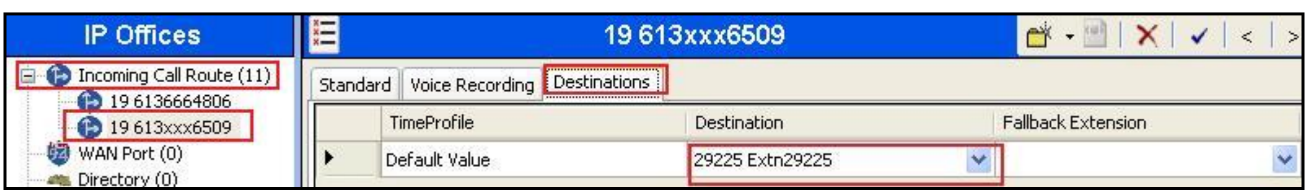
5.7. Incoming Call Route

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**. 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 State (US English)**
- Default values can be used for all other fields.



On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to 613xxx6509 on line 19 are routed to extension **29225**.

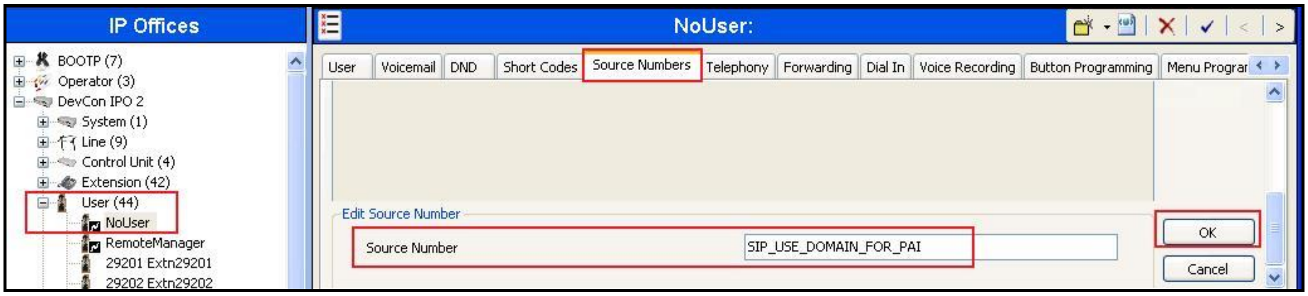


5.8. Privacy/Anonymous Calls

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_PAI***. Click **OK**.



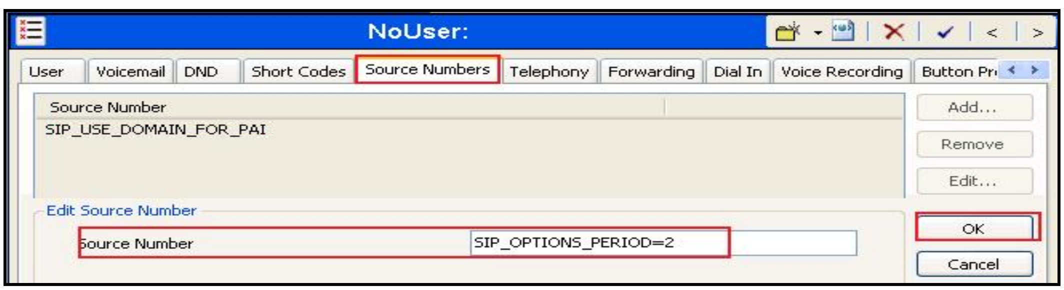
The **SIP_USE_DOMAIN_FOR_PAI** parameter will appear in the list of Source Numbers as shown below.

5.9. SIP Options

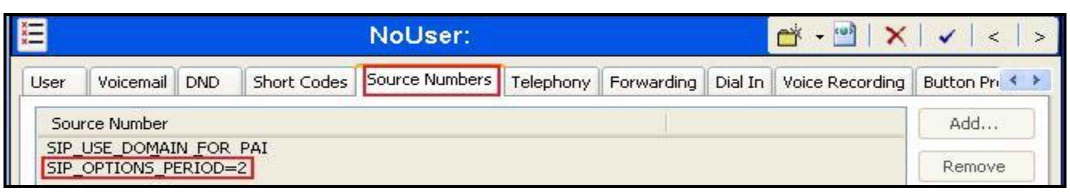
Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.1** and the **SIP_OPTIONS_PERIOD** 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 44 seconds is used.
- To establish a period less than 42 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 42 secs. The OPTIONS message period will be equal to the **Binding Refresh Time**.
- To establish a period greater than 42 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 42 secs. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD**.

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



The **SIP_OPTIONS_PERIOD** parameter will appear in the list of Source Numbers as shown below. For the compliance test, an OPTIONS period of 1 minute was desired. The **Binding Refresh Time** was set to ***60*** seconds (1 minute) in **Section 5.1**. The **SIP_OPTIONS_PERIOD** was set to ***2*** minutes. Avaya IP Office chose the OPTIONS period as the smaller of these two values (1 minute). Click the **OK** button (not shown).



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. Bell Canada SIP Trunking Configuration

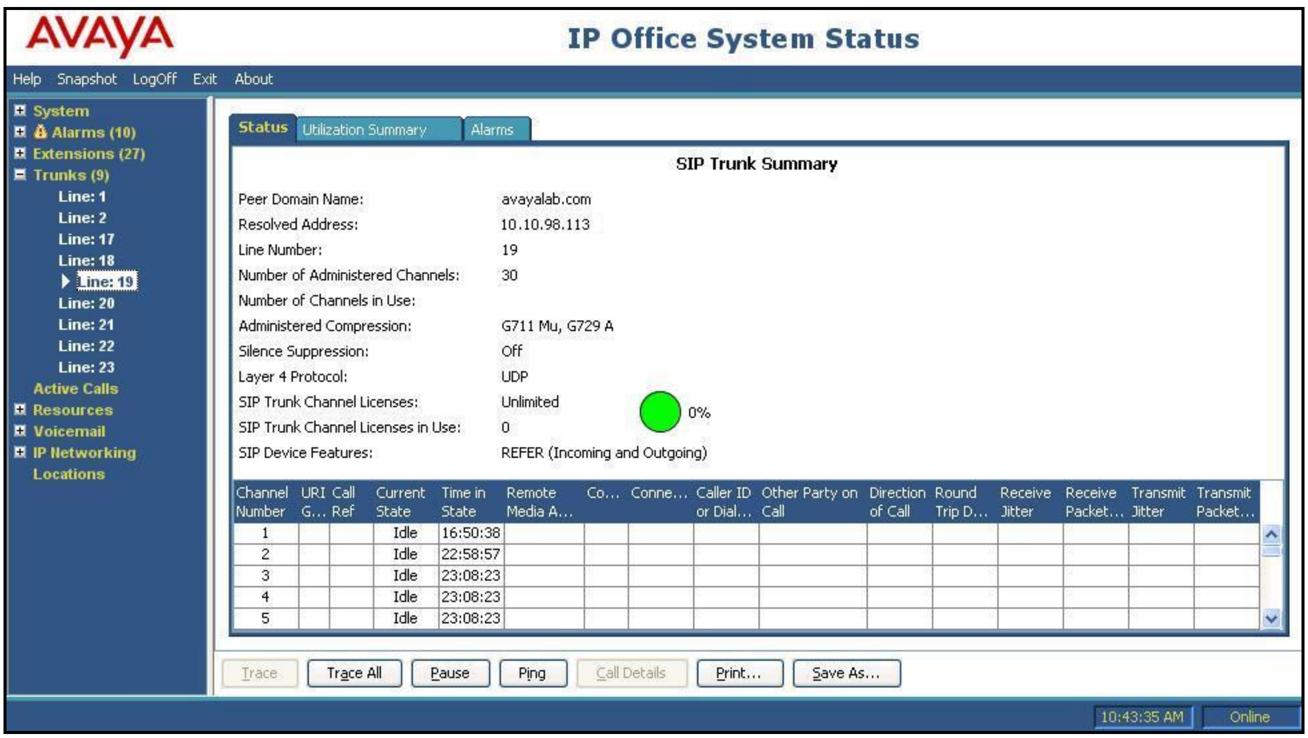
Bell Canada is responsible for the configuration of Bell Canada SIP Trunking service. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise. Bell Canada will provide the customer the necessary information to configure the Avaya IP Office SIP connection to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

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

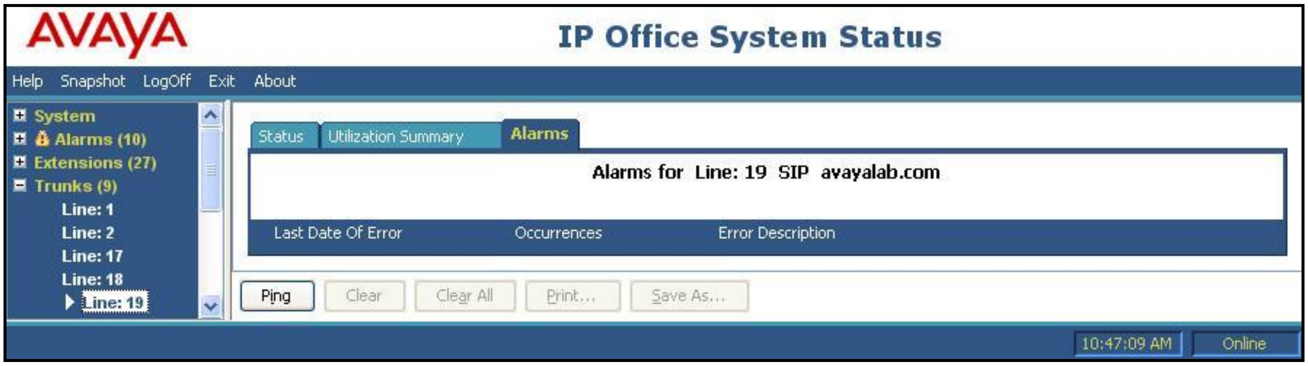
7. Verification Steps

The following steps may be used to verify the configuration:

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



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



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

8. Conclusion

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

9. Additional References

[1] *IP Office 9.0.3 Installation and Maintenance, Document number 15-601042*, Issue 09p, November 2014
[2] *IP Office 9.0.3 Manager 9.0, Document number 15-601011*, Issue 9.0.3, 08 May 2014
[3] *IP Office 9.0.3 Administering Voicemail Pro, Document number 15-601063*, Issue 9.0c, 24 April 2014
[4] *IP Office Embedded Voicemail User Guide (IP Office Mode), Document number 15-604067,* Issue 13a, 13 February 2014

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 Bell Canada SIP Trunking is available from Bell Canada.

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.