



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Bell Canada SIP Trunking with Avaya IP Office 7.0 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider Bell Canada and Avaya IP Office 7.0.

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 and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 7.0, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, 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 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**.

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, 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, G.711A 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
- 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 and outbound emergency calls (911) are supported but were not tested as part of the compliance test.
- Faxing between the enterprise site and PSTN were not tested as part of the compliance test since Bell Canada currently does not support T.38 FoIP (Fax over IP) on its SIP Trunking production system.

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.
- **Codec Mismatch** – Bell Canada SIP Trunking service returns “486 Busy here” for outbound call from the enterprise to the PSTN analog phone instead of a more appropriate status code. For inbound calls from the PSTN to the enterprise, the IP Office treats this situation properly (returning “488 Not Acceptable Here”). Bell Canada has reproduced this problem in the lab and is working on a resolution.
- **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. SIP signaling trace showed that the enterprise IP Office did not send an UPDATE message to the network to update the call display of the PSTN phone in call transfer. This problem was reported to Avaya IP Office for further investigation.
- **DTMF tone transmission**: Bell Canada SIP Trunking service does not use out-of-band RTP events for DTMF tone transmissions as per RFC 2833 for inbound calls to the enterprise. Instead, DTMF tones were passed in-band using G.711MU on inbound calls. This configuration is “on purpose to avoid double conversion between in-band and RTP event when a call initiated from PSTN is forwarded back to PSTN by PBX” (quoted from Bell Canada support). Outbound calls from the enterprise use out-of-band DTMF tone transmission as configured on IP Office.

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.

Located at the enterprise site is an Avaya IP Office 500v2 with the COMBO6210/ATM4 expansion card which provides connections for 6 digital stations, 2 analog stations, 4 analog trunks to the PSTN as well as 10-channel VCM (Voice Compression Module) for supporting VoIP codecs. The LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public IP network. Endpoints include an Avaya 1600 Series IP Telephone (with H.323 firmware), an Avaya 9600 Series IP Telephone (with H.323 firmware), an Avaya 5410 and an Avaya 6408D Digital Telephones, an Avaya 6210 Analog Telephone and an Avaya IP Office Softphone. The site also has a Windows 2003 Server running Avaya Voicemail Pro for providing voice messaging service to the Avaya IP Office users. A separate Windows XP PC runs Avaya IP Office Manager to configure and administer the Avaya IP Office.

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.

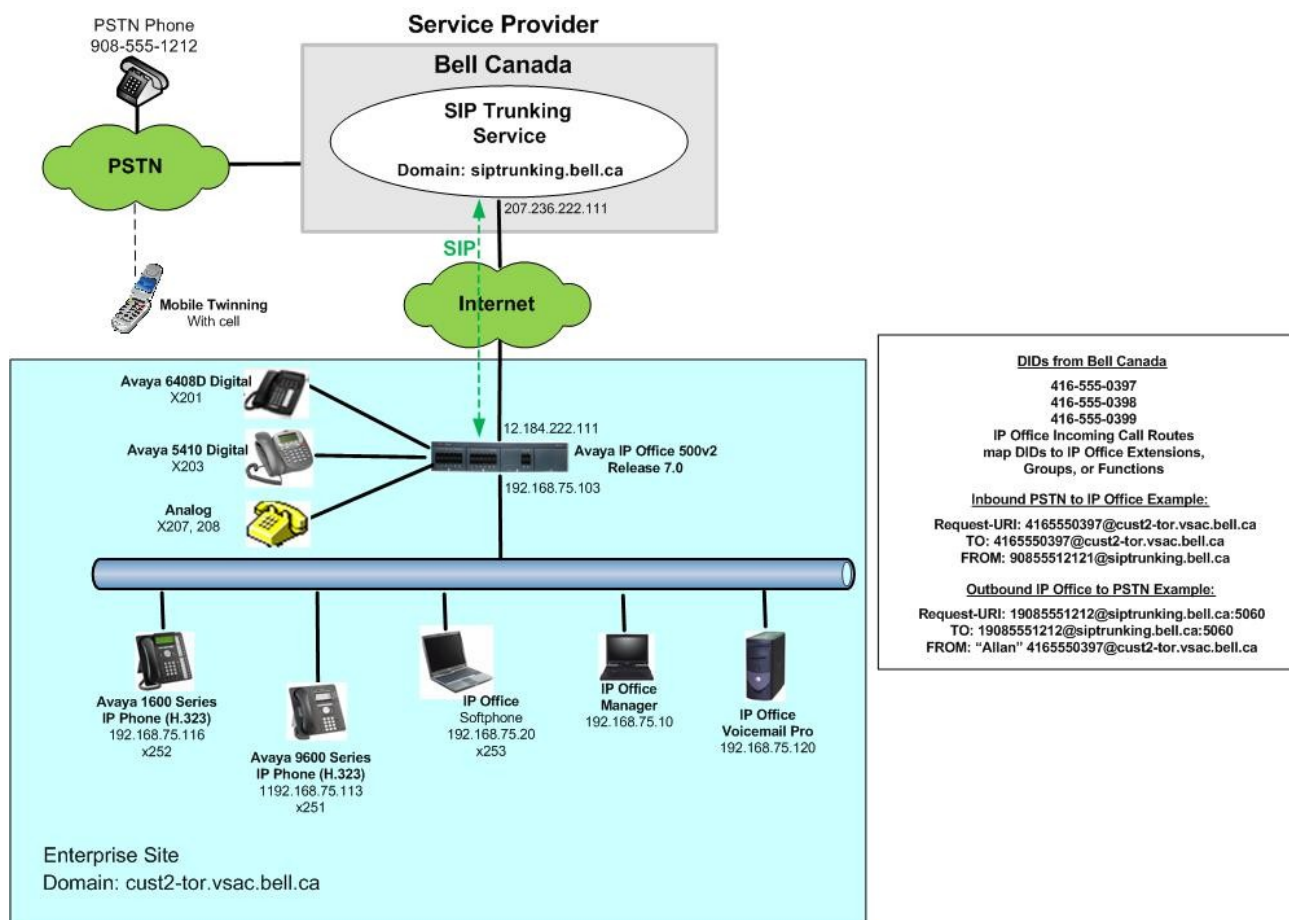


Figure 1: Test Configuration for Avaya IP Office with Bell Canada SIP Trunking Service

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

For the purposes of the compliance test, Avaya IP Office users dialed a short code of 9 + N digits to send digits across the SIP trunk to 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.

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

4. Equipment and Software Validated

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

Avaya Telephony Components	
Equipment	Release
Avaya IP Office 500v2	7.0 (5)
Avaya IP Office COMBO6210/ATM4 Module	7.0 (5)
Avaya Voicemail Pro	7.0.17
Avaya IP Office Manager	9.0 (5)
Avaya 1616 IP Telephone (H.323)	Avaya one-X Deskphone Value Edition 1.2.2
Avaya 9620 IP Telephone (H.323)	Avaya one-X Deskphone Edition 3.1
Avaya Digital Telephones (5410)	N/A
Avaya Digital Telephones (6408D)	N/A
Avaya 6210 Analog Telephone	N/A
Avaya IP Office Softphone	3.1.2.17 59616
Bell Canada SIP Trunking	
Equipment	Release
Acme Packet Net-Net 4250 SBC	Firmware SC6.2.0 MR-4 Patch 1 (Build 718)
BroadWorks Soft Switch	Rel16
Legacy Nortel CS2K Media Gateway	SN10 PVG/IW-SPM

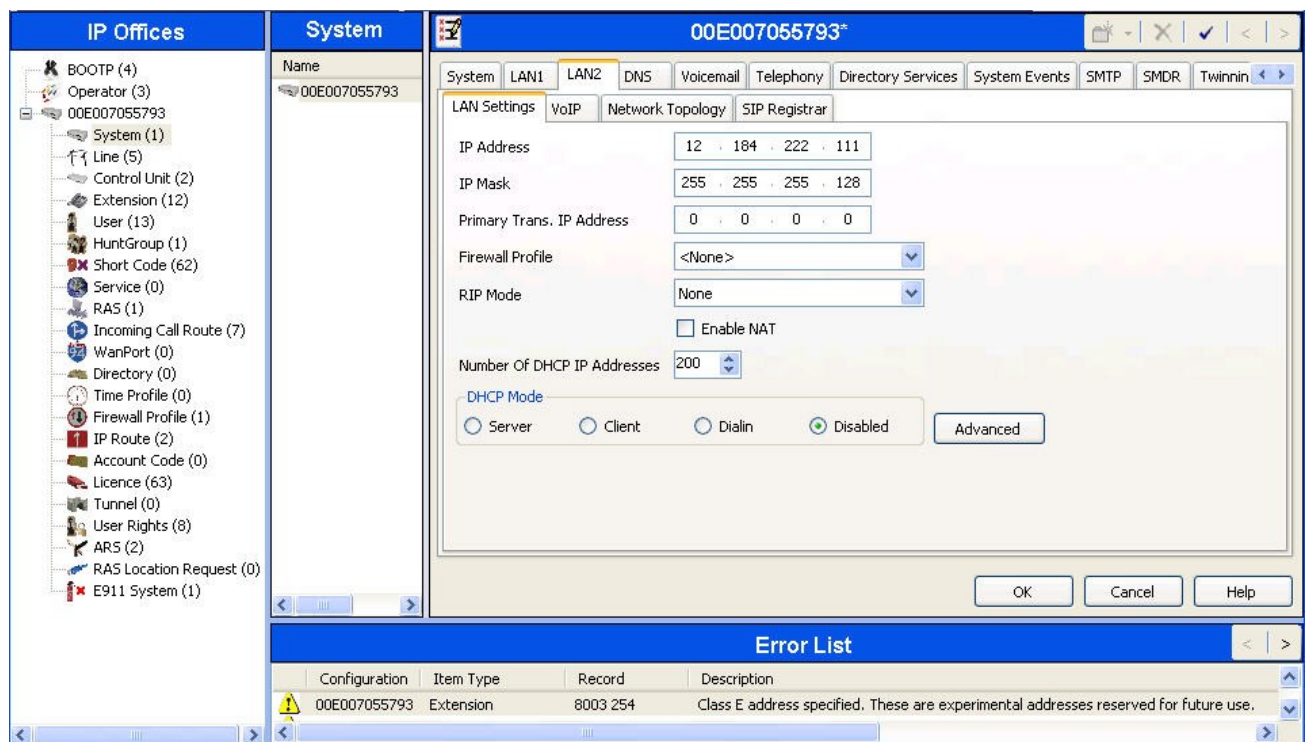
5. Configure IP Office

This section describes the Avaya IP Office configuration to support connectivity to Bell Canada 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. LAN2 Settings

In the sample configuration, the MAC address **00E007055793** 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 LAN2 settings, first navigate to **System (1) → 00E007055793** 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.



The screenshot displays the Avaya IP Office configuration interface. On the left, a tree view shows the system hierarchy under 'IP Offices', including 'BOOTP (4)', 'Operator (3)', and 'System (1)'. The main panel is titled '00E007055793*' and shows the 'LAN2' configuration tab. The 'IP Address' is set to 12.184.222.111 and the 'IP Mask' is 255.255.255.128. Other settings include 'Primary Trans. IP Address' (0.0.0.0), 'Firewall Profile' (<None>), 'RIP Mode' (None), and 'Number Of DHCP IP Addresses' (200). The 'DHCP Mode' is set to 'Disabled'. At the bottom, an 'Error List' table shows a warning for the configuration item.

Configuration	Item Type	Record	Description
00E007055793	Extension	8003 254	Class E address specified. These are experimental addresses reserved for future use.

Select the **VoIP** tab as shown in the following screen. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Bell Canada. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 1600-Series and the 9600-Series IP Telephones used in the sample configuration. 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. Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling. The specific values used for the compliance test are shown in the example below. All other parameters should be set according to customer requirements.

00E007055793

System LAN1 LAN2 DNS Voicemail Telephony Directory Services System Events SMTP SMDR Twinning

LAN Settings VoIP Network Topology SIP Registrar

☒ H323 Gatekeeper Enable

☒ SIP Trunks Enable

☒ SIP Registrar Enable

☐ H323 Auto-create Extn

☐ H323 Auto-create User

☒ Enable RTCP Monitoring On Port 5005

RTP Port Number Range

Port Range (Minimum) 49152

Port Range (Maximum) 53246

DiffServ Settings

B8 DSCP(Hex) FC DSCP Mask (Hex) 88 SIG DSCP (Hex)

46 DSCP 63 DSCP Mask 34 SIG DSCP

DHCP Settings

Primary Site Specific Option Number (SSON) 176

Secondary Site Specific Option Number (SSON) 242

OK Cancel Help

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 the Avaya IP Office WAN port.
- All other parameters should be set according to customer requirements.

The screenshot shows the Avaya IP Office configuration window with the title bar '00E007055793*'. The 'LAN2' tab is selected in the top navigation bar. Below it, the 'Network Topology' sub-tab is active. The 'Network Topology Discovery' section contains the following settings:

- STUN Server IP Address: 69 . 90 . 168 . 13
- STUN Port: 3478
- Firewall/NAT Type: Open Internet (selected from a dropdown menu)
- Binding Refresh Time (seconds): 60
- Public IP Address: 12 . 184 . 222 . 111
- Public Port: 5060

At the bottom of the section are two buttons: 'Run STUN' and 'Cancel'. Below these is a checkbox labeled 'Run STUN on startup' which is currently unchecked. At the very bottom of the window are three buttons: 'OK', 'Cancel', and 'Help'.

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 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. Set the **Automatic Codec Preference** for the default codec to be used for intra-enterprise traffic. 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.

00E007055793

System LAN1 LAN2 DNS Voicemail **Telephony** Directory Services System Events SMTP SMDR Twinning VCM

Telephony Tones & Music Call Log

Analogue Extensions

Default Outside Call Sequence Normal

Default Inside Call Sequence Ring Type 1

Default Ring Back Sequence Ring Type 2

Restrict Analogue Extension Ringer Voltage ☐

Dial Delay Time (secs) 4

Dial Delay Count 0

Default No Answer Time (secs) 15

Hold Timeout (secs) 120

Park Timeout (secs) 300

Ring Delay (secs) 5

Call Priority Promotion Time (secs) Disabled

Default Currency USD

Automatic Codec Preference G.711 ULAW 64K

Companding Law

Switch

☒ ULAW

☐ ALAW

Line

☒ ULAW Line

☐ ALAW Line

☐ DSS Status

☒ Auto Hold

☒ Dial By Name

☒ Show Account Code

☐ Inhibit Off-Switch Forward/Transfer

☐ Restrict Network Interconnect

☐ Drop External Only Impromptu Conference

☐ Visually Differentiate External Call

OK Cancel Help

5.3. Twinning Calling Party Settings

When using twinning, the calling party number displayed on the twinned phone is controlled by two parameters. The first parameter is the **Send original calling party information for Mobile Twinning** box on the **System→Twinning** tab. The second parameter is the **Send Caller ID** parameter on the **SIP Line** form (shown in **Section 5.4**).

If **Send original calling party information for Mobile Twinning** on the **System→Twinning** tab is optioned, the setting of the second parameter is ignored and Avaya IP Office will send the following in the SIP From Header:

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

The above behavior in Avaya IP Office Release 7 is the same as in Avaya IP Office Release 6 and was tested and verified in the compliance test.

Avaya IP Office Release 7 also provides an alternative method of sending caller ID through SIP Diversion header (configured via unchecking **Send original calling party information for Mobile Twinning** here then selecting **Diversion Header** for the **Send Caller ID** parameter on the **SIP Line** form in **Section 5.4**). This alternative configuration could provide more accurate caller ID information if the service provider supports the SIP Diversion header (not tested in the compliance test).

For the compliance test, the **Send original calling party information for Mobile Twinning** box in the **System→Twinning** tab was checked which overrides any setting of the **Send Caller ID** parameter on the **SIP Line** form.



The screenshot shows the 'System→Twinning' configuration window in Avaya IP Office. The window title is '00E007055793*'. The 'System' tab is selected, and the 'Send original calling party information for Mobile Twinning' checkbox is checked. Below it is a text field for 'Calling party information for Mobile Twinning'. The window has OK, Cancel, and Help buttons at the bottom.

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. 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 **None**. 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, both configurations were successfully tested to transfer a call between a PSTN phone and an enterprise phone to a second PSTN phone.

The screenshot displays the Avaya IP Office configuration interface. The left pane shows the 'IP Offices' tree with 'Line (5)' selected. The middle pane shows the 'Line' list with 'Line 17' selected. The right pane shows the 'SIP Line - Line 17' configuration tab. The configuration includes the following fields and settings:

Field	Value	Field	Value
Line Number	17	In Service	<input checked="" type="checkbox"/>
ITSP Domain Name	cust2-tor.vtac.bell.ca	Use Tel URI	<input type="checkbox"/>
Prefix		Check OOS	<input checked="" type="checkbox"/>
National Prefix		Call Routing Method	Request URI
Country Code		Originator number for forwarded and twinning calls	
International Prefix			
Send Caller ID	None		
Association Method	By Source IP address		
REFER Support			
Incoming	Always		
Outgoing	Always		

Buttons at the bottom: OK, Cancel, Help.

Select the **Transport** tab. The **ITSP Proxy Address** is set to the Bell Canada SIP Proxy IP Address provided by Bell Canada. As shown in **Figure 1**, this IP Address is **207.236.222.111**. 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 Bell Canada. 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.

The screenshot shows the 'SIP Line - Line 17*' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' is set to '207.236.222.111'. In the 'Network Configuration' section, 'Layer 4 Protocol' is set to 'UDP', 'Send Port' is '5060', and 'Use Network Topology Info' is set to 'LAN 2'. 'Listen Port' is also '5060'. 'Explicit DNS Server(s)' are set to '0 . 0 . 0 . 0' and '0 . 0 . 0 . 0'. 'Calls Route via Registrar' is checked. 'Separate Registrar' is empty. The window has 'OK', 'Cancel', and 'Help' buttons at the bottom.

Tab	ITSP Proxy Address	Layer 4 Protocol	Send Port	Use Network Topology Info	Listen Port	Explicit DNS Server(s)	Calls Route via Registrar	Separate Registrar
Transport	207.236.222.111	UDP	5060	LAN 2	5060	0 . 0 . 0 . 0 0 . 0 . 0 . 0	<input checked="" type="checkbox"/>	

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.
- Uncheck the **Registration required** option. Bell Canada does not require registration for Digest Authentication.

The screenshot shows the 'SIP Line - Line 17' configuration window. The 'SIP Credentials' tab is selected. The table below lists the existing credentials:

Index	UserName	Authentication Name	Contact	Expiry	Register
1	4167751396	4167751396		60	False

The 'Edit SIP Credentials' dialog box is open, showing the following fields:

- User name: 4167751396
- Authentication Name: 4167751396
- Contact: (empty)
- Password: (masked with asterisks)
- Expiry: 60
- Registration required: ☐

Buttons for 'Add...', 'Remove', 'OK', 'Cancel', and 'Help' are visible.

A SIP URI entry must be created to match each incoming number that Avaya IP Office will accept on this line. Select the **SIP URI** tab, then click the **Add** button and the **New Channel** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit...** button. In the example screen below, a previously configured entry is edited. 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 whose SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.6**.
- Set **PAI** to **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**.
- 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 **17** was defined that only contains this line (line 17).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

The screenshot shows the 'SIP Line - Line 17' configuration window. The 'SIP URI' tab is selected. At the top, there are tabs for 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. Below these is a table with columns: Channel, Groups, Via, Local URI, Contact, Display Name, PAI, Credential, and Max Calls. The table contains one entry: Channel 1, Groups 17 17, Via 1..., Local URI 1: 4167..., and Max Calls 10. To the right of the table is an 'Add...' button. Below the table is an 'Edit Channel' dialog box. The dialog box has fields for: Via (12.184.222.111), Local URI (Use Internal Data), Contact (Use Internal Data), Display Name (Use Internal Data), PAI (Use Internal Data), Registration (1: 4167751396), Incoming Group (17), Outgoing Group (17), and Max Calls per Channel (10). To the right of the dialog box are 'OK' and 'Cancel' buttons. At the bottom of the main window are 'OK', 'Cancel', and 'Help' buttons.

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	17 17	1...	1: 4167...					10

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

- The **Compression Mode** was configured using the **Advanced** button, allowing an explicit ordered list of codecs to be specified. Check marks next to **G.711 ULAW 64K**, **G.711 ALAW 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 the **DTMF Support** field to **RFC2833**. 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.
- Select **None** for **Fax Transport Support**. T.38 faxing is not currently supported by Bell Canada. If G.711 pass-through faxing is supported by the service provider, **G.711** should be selected here (not tested in the compliance test).
- Check the **Re-invite Supported** box.
- Default values may be used for all other parameters.

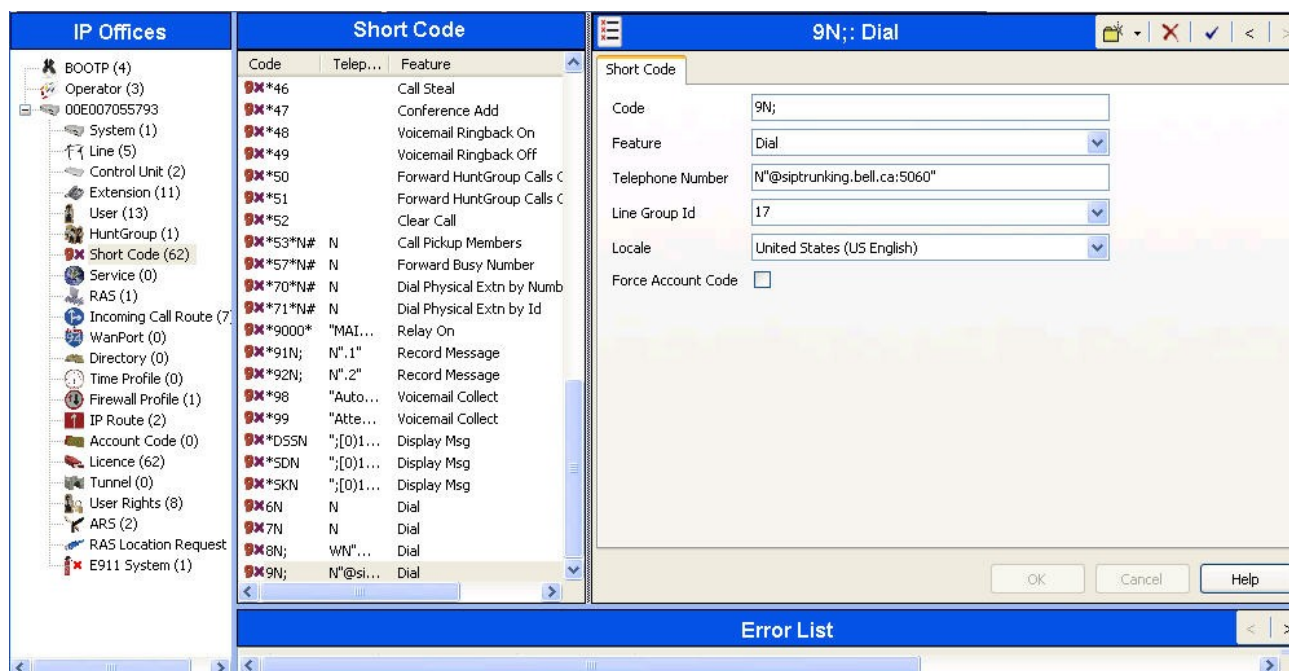
The screenshot shows the 'SIP Line - Line 17' configuration window with the 'VoIP' tab selected. The 'Compression Mode' is set to 'Advanced', showing a list of codecs: G.711 ULAW 64K (checked), G.711 ALAW 64K (checked), G.729(a) 8K CS-ACELP (checked), and G.723.1 6K3 MP-MLQ (unchecked). The 'Fax Transport Support' is set to 'None'. The 'Call Initiation Timeout (s)' is set to 4. The 'DTMF Support' is set to 'RFC2833'. On the right, 'VoIP Silence Suppression' is unchecked, 'Re-invite Supported' is checked, 'Use Offerer's Preferred Code' is unchecked, and 'Codec Lockdown' is unchecked. The window has 'OK', 'Cancel', and 'Help' buttons at the bottom.

Field	Value
Compression Mode	Advanced
Fax Transport Support	None
Call Initiation Timeout (s)	4
DTMF Support	RFC2833
VoIP Silence Suppression	<input type="checkbox"/>
Re-invite Supported	<input checked="" type="checkbox"/>
Use Offerer's Preferred Code	<input type="checkbox"/>
Codec Lockdown	<input type="checkbox"/>

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"@siptrunking.bell.ca:5060"**. 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.



The simple “9N;” short codes illustrated above does not provide a means of alternate routing if the configured SIP Line is out of service or temporarily not responding. When alternate routing options and/or more customized analysis of the digits following the short code are desired, the Automatic Route Selection (ARS) feature may be used. In the following example screen, the short code 6N is illustrated for access to ARS. When the Avaya IP Office user dials 6 plus any number N, rather than being directed to a specific **Line Group Id**, the call is directed to **Line Group ID “50: Main”**, configurable via ARS. See **Section 5.8** for example ARS route configuration for “50: Main” as well as a backup route.

6N: Dial

Short Code

Code: 6N

Feature: Dial

Telephone Number: N

Line Group Id: 50: Main

Locale:

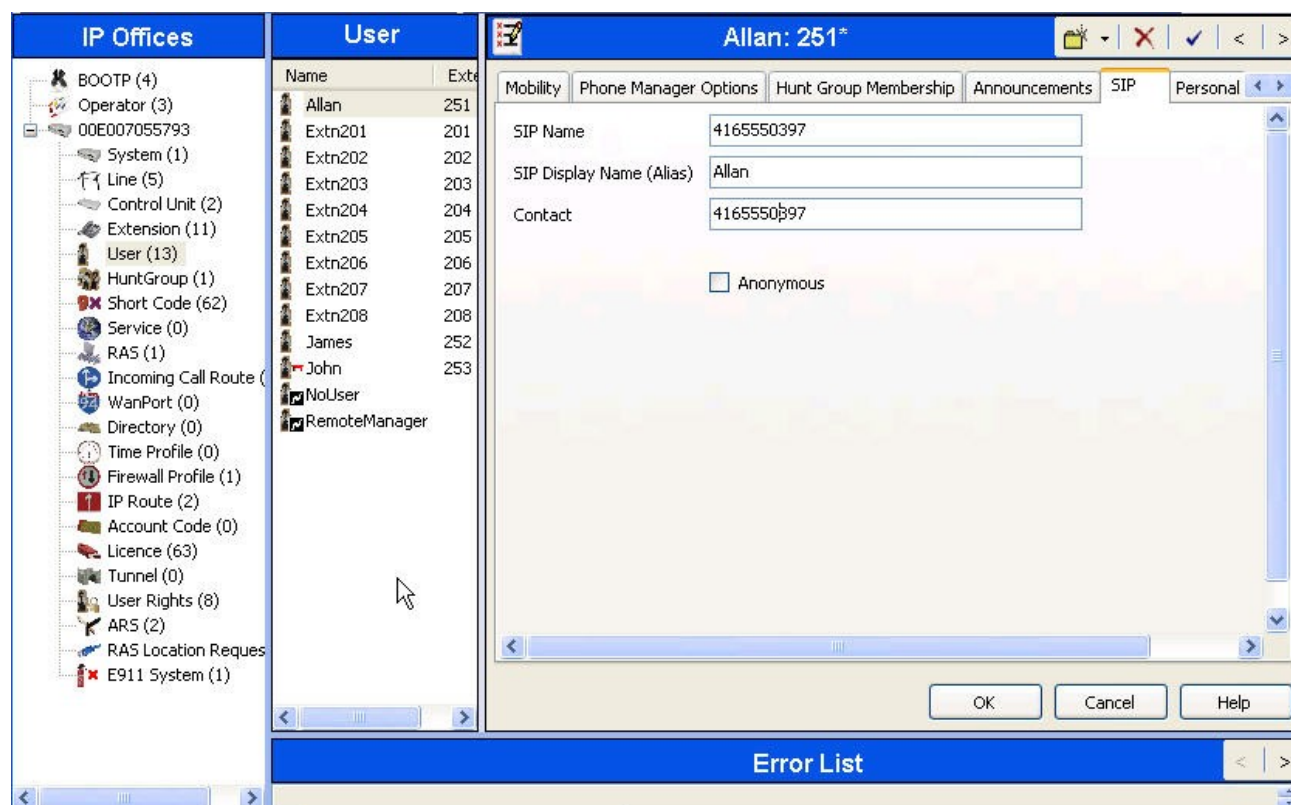
Force Account Code: ☐

OK Cancel Help

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 “Allan”. 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 Allan. 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 Allan. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone. Other options can be set according to customer requirements.

The screenshot shows a configuration window titled "Allan: 251*" with a blue header bar. Below the header is a tabbed interface with the following tabs: "Menu Programming", "Mobility" (selected), "Phone Manager Options", "Hunt Group Membership", "Announcements", "SIP", and "Personal Directory".

The "Mobility" tab is active, showing two main sections:

- Internal Twinning**: This section is collapsed. It contains a "Twinned Handset" dropdown menu (set to "<None>"), a "Maximum Number of Calls" dropdown menu (set to "1"), and three unchecked checkboxes: "Twin Bridge Appearances", "Twin Coverage Appearances", and "Twin Line Appearances".
- Mobility Features**: This section is expanded and has a checked checkbox. It contains a sub-section **Mobile Twinning** which is also checked. This sub-section includes:
 - "Twinned Mobile Number (including dial access code)": A text field containing "917325551212".
 - "Twinning Time Profile": A dropdown menu set to "<None>".
 - "Mobile Dial Delay (secs)": A text field containing "2".
 - "Mobile Answer Guard (secs)": A spinner box set to "0".
 - Four unchecked checkboxes: "Hunt group calls eligible for mobile twinning", "Forwarded calls eligible for mobile twinning", "Twin When Logged Out", and "one-X Mobile Client".
 - Two more unchecked checkboxes: "Mobile Call Control" and "Mobile Callback".

At the bottom of the window are three buttons: "OK", "Cancel", and "Help".

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.
- Default values can be used for all other fields.

Line Group Id	Incoming Number	Destination
0		200 Main
0		DialIn
1		203 Extn203
17	4165550397	251 Allan
17	4165550398	207 Extn207
17	4165550399	253 John

Standard	Voice Recording	Destinations
Bearer Capability: Any Voice		
Line Group Id: 17		
Incoming Number: 4165550397		
Incoming Sub Address:		
Incoming CLI:		
Locale: United States (US English)		
Priority: 1 - Low		
Tag:		
Hold Music Source: System Source		

On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to 416-555-0397 on line 17 are routed to user Allan at extension 251.

TimeProfile	Destination	Fallback Extension
Default Value	251 Allan	

5.8. ARS and Alternate Routing

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

Optionally, ARS can be used rather than the simple “9N;” short code approach documented in **Section 5.5**. With ARS, a secondary dial tone can be provided after the access code, time-based routing criteria can be introduced, and alternate routing can be specified so that a call can re-route automatically if the primary route or outgoing line group is not available. Although not shown in this section, ARS also facilitates more specific dialed telephone number matching, enabling immediate routing and alternate treatment for different types of numbers following the access code. For example, if all 1+10 digit calls following an access code should use the SIP Line preferentially, but other local or service numbers following the access code should prefer a different outgoing line group, ARS can be used to distinguish the call behaviors.

To add a new ARS route, right-click **ARS** in the Navigation pane, and select **New**. To view or edit an existing ARS route, select **ARS** in the Navigation pane, and select the appropriate route name in the Group pane.

The following screen shows an example ARS configuration for the route named “Main”. The **In Service** parameter refers to the ARS form itself, not the Line Groups that may be referenced in the form. If the **In Service** box is un-checked, calls are routed to the ARS route name specified in the **Out of Service Route** parameter. IP Office short codes may also be defined to allow an ARS route to be disabled or enabled from a telephone. The configurable provisioning of an Out of Service Route, and the means to manually activate the Out of Service Route can be helpful for scheduled maintenance or other known service-affecting events for the primary route.

IP Offices

- BOOTP (4)
- Operator (3)
- 00E007055793
- System (1)
- Line (5)
- Control Unit (2)
- Extension (11)
- User (13)
- HuntGroup (1)
- Short Code (62)
- Service (0)
- RAS (1)
- Incoming Call Route (6)
- WanPort (0)
- Directory (0)
- Time Profile (0)
- Firewall Profile (1)
- IP Route (2)
- Account Code (0)
- Licence (62)
- Tunnel (0)
- User Rights (8)
- RAS Location Request (0)
- E911 System (1)

ARS

Name: backup
Time Profile: Main

Main

ARS Route Id: 50
Route Name: Main
Dial Delay Time: System Default (4)
In Service: ☒
Time Profile: <None>
Secondary Dial tone: ☒ SystemTone
Check User Call Barring: ☒
Out of Service Route: 51: backup
Out of Hours Route: <None>

Code	Telephone Number	Feature	Line Group Id
11	911	Dial Emergency	0
911	911	Dial Emergency	0
0N;	0N	Dial 3K1	0
1N;	1N"@siptrunking.bell.ca:5060"	Dial 3K1	17
XN;	N	Dial 3K1	0
XXXXXXXXXXN	N	Dial 3K1	0

Alternate Route Priority Level: 3
Alternate Route Wait Time: 30
Additional Route: 51: backup

Buttons: Add..., Remove, Edit..., OK, Cancel, Help

Assuming the primary route is in-service, the number passed from the short code used to access ARS (e.g., 6N in **Section 5.5**) can be further analyzed to direct the call to a specific Line Group ID. Per the example screen above, if the user dialed 6-1-908-555-1212, the call would be directed to Line Group 17, the SIP Line configured and described in these Application Notes. If Line Group 17 cannot be used, the call can automatically route to the route name configured in the **Additional Route** parameter in the lower right of the screen. Since alternate routing can be considered a privilege not available to all callers, IP Office can control access to the alternate route by comparing the calling user's priority to the value in the **Alternate Route Priority Level** field.

The following screen shows an example ARS configuration for the route named “backup” with ARS Route ID 51. Continuing the example, if the user dialed 6-1-908-555-1212, and the call could not be routed via the primary route “50: Main” described above, the call will be delivered to this “backup” route. Per the configuration shown below, the call will be delivered to Line Group 1, using an analog trunk connecting Avaya IP Office to the PSTN as a backup connection. In this case, the originally dialed number (sans the short code 6) will be dialed as is through the analog trunk to the PSTN. Additional codes (e.g., 411, 0+10, etc.) can be added to the ARS route by pressing the **Add...** button to the right of the list of previously configured codes.

ARS backup

ARS Route Id: 51

Route Name: backup

Dial Delay Time: System Default (4)

☐ Secondary Dial tone

☐ Check User Call Barring

In Service: ☒ Out of Service Route: <None>

Time Profile: <None> Out of Hours Route: <None>

Code	Telephone Number	Feature	Line Group Id
11	911	Dial Emergency	0
911	911	Dial Emergency	0
1N;	1N	Dial	1

Buttons: Add..., Remove, Edit...

Alternate Route Priority Level: 3

Alternate Route Wait Time: 30

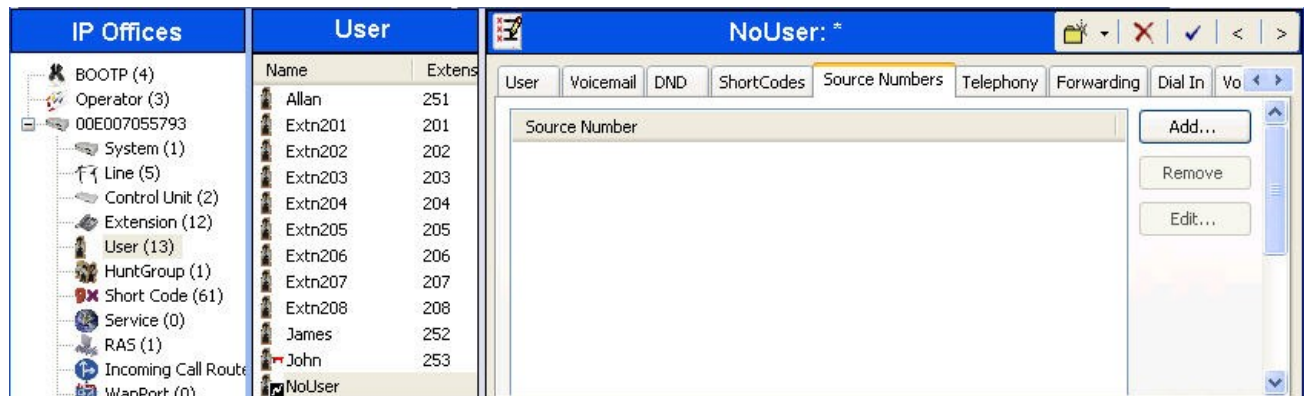
Additional Route: <None>

In the testing associated with the configuration, calls were successfully delivered to SIP Line 17 via the primary ARS route “50: Main” or to the analog trunk via the backup ARS route shown above. When the primary route experienced a network outage, Avaya IP Office successfully routed the call via the backup route.

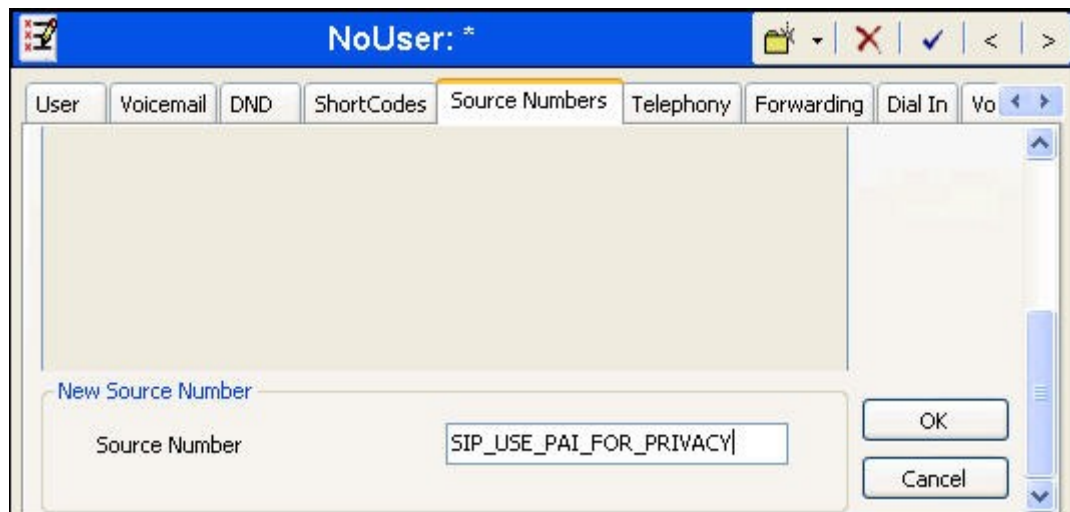
5.9. Privacy/Anonymous Calls

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

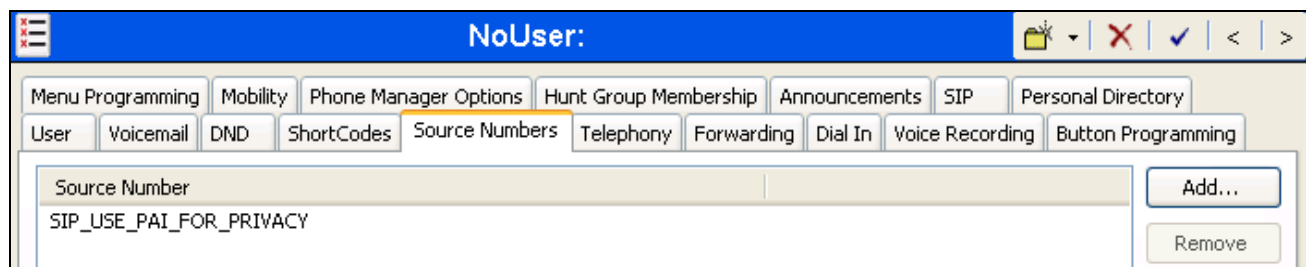
To configure Avaya IP Office to use PAI for privacy calls, navigate to **User** → **noUser** in the Navigation / Group Panes. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



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



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

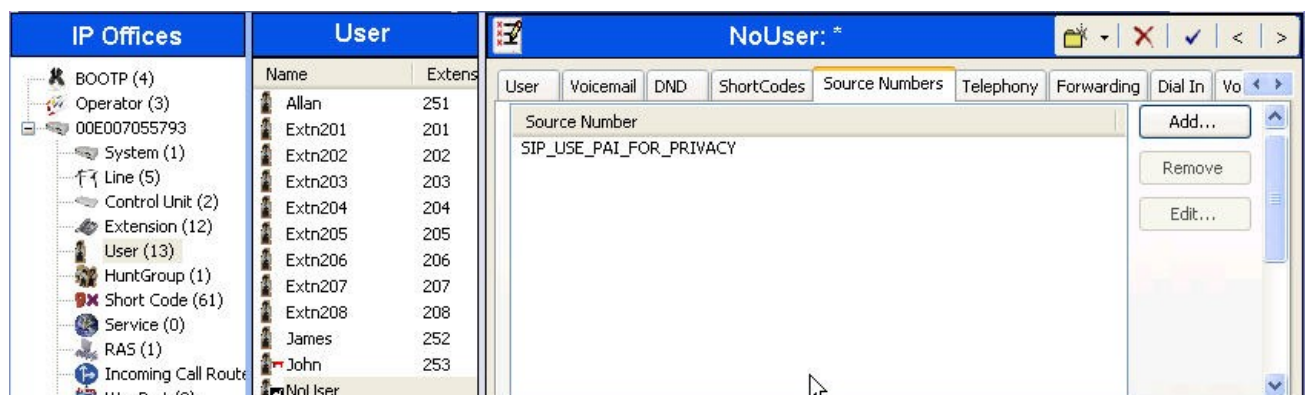


5.10. SIP Options

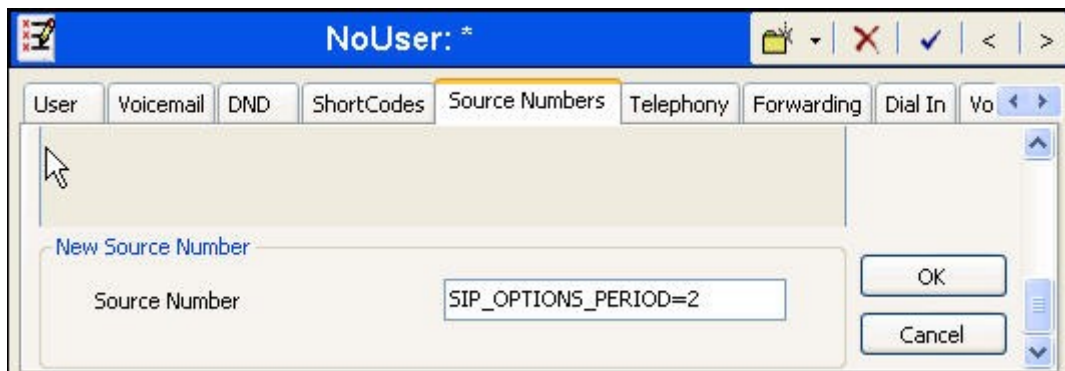
Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.1** and the **SIP_OPTIONS_PERIOD** 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**.

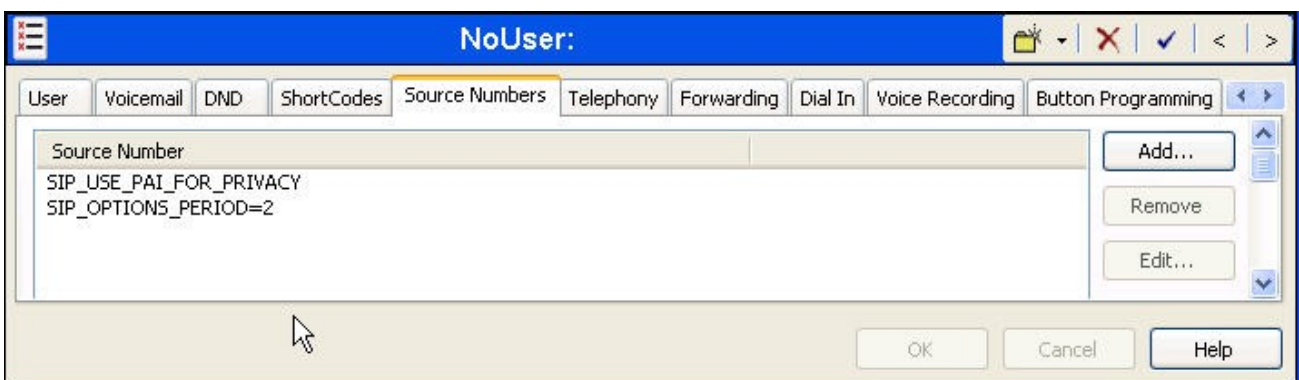
To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User → noUser** in the Navigation / Group Panes. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



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).



5.11. 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

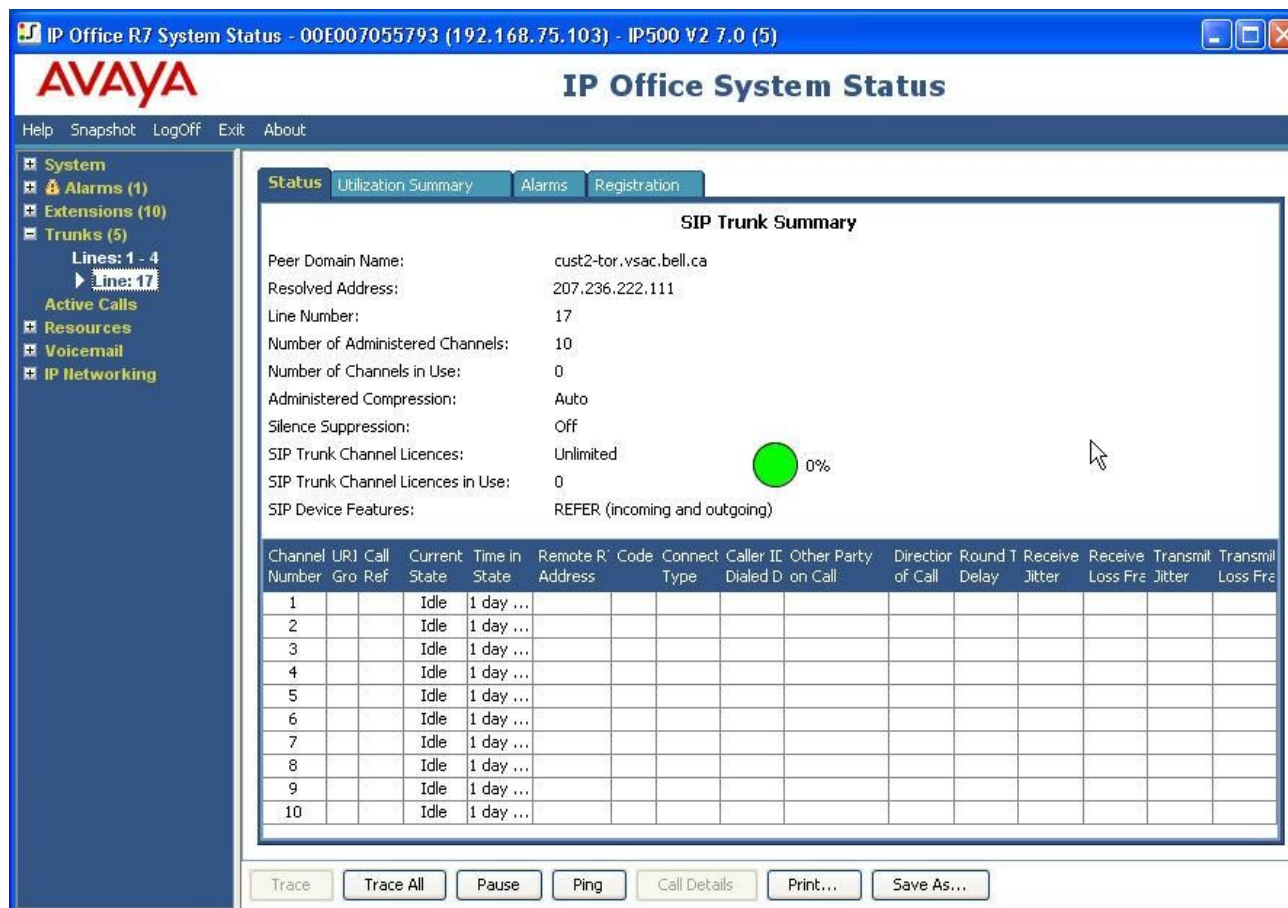
Bell Canada is responsible for the configuration of Bell Canada SIP Trunking service. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise. Bell Canada will provide the customer the necessary information to configure the Avaya IP Office SIP connection to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

- IP address of the Bell Canada SIP proxy.
- Bell Canada SIP domain
- 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).



IP Office R7 System Status - 00E007055793 (192.168.75.103) - IP500 V2 7.0 (5)

AVAYA IP Office System Status

Help Snapshot LogOff Exit About

- System
- Alarms (1)
- Extensions (10)
- Trunks (5)
 - Lines: 1 - 4
 - Line: 17
- Active Calls
- Resources
- Voicemail
- IP Networking

Status Utilization Summary Alarms Registration

SIP Trunk Summary

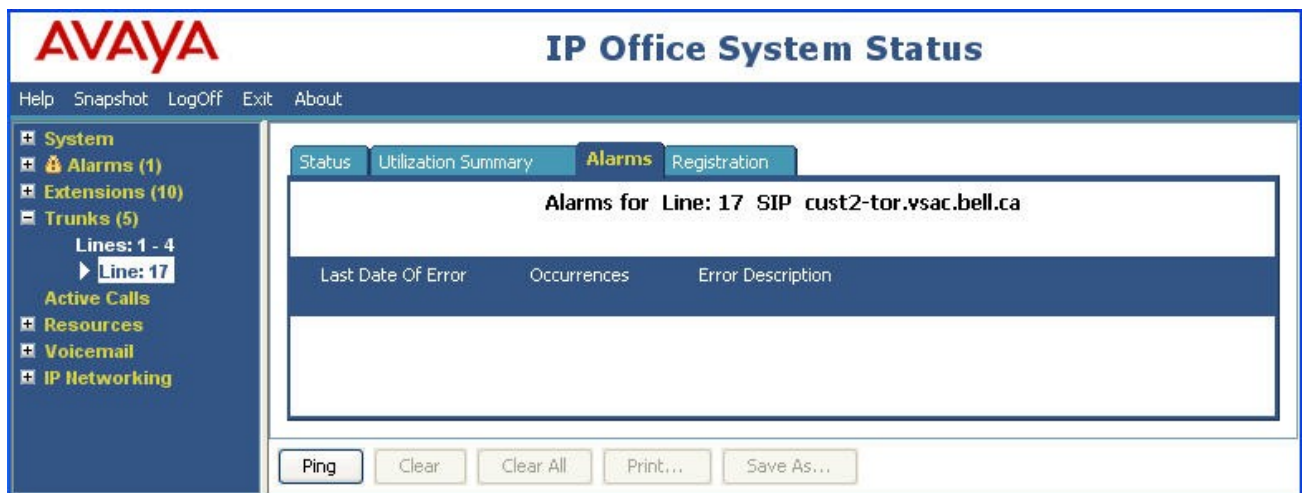
Peer Domain Name: cust2-tor.vtac.bell.ca
Resolved Address: 207.236.222.111
Line Number: 17
Number of Administered Channels: 10
Number of Channels in Use: 0
Administered Compression: Auto
Silence Suppression: Off
SIP Trunk Channel Licences: Unlimited
SIP Trunk Channel Licences in Use: 0
SIP Device Features: REFER (incoming and outgoing)

0%

Channel Number	URI	Call Gro Ref	Current State	Time in State	Remote R' Address	Code	Connect Type	Caller ID	Other Party	Direction of Call	Round T Delay	Receive Jitter	Receive Loss Fre	Transmit Jitter	Transmit Loss Fre
1			Idle	1 day ...											
2			Idle	1 day ...											
3			Idle	1 day ...											
4			Idle	1 day ...											
5			Idle	1 day ...											
6			Idle	1 day ...											
7			Idle	1 day ...											
8			Idle	1 day ...											
9			Idle	1 day ...											
10			Idle	1 day ...											

Trace Trace All Pause Ping Call Details Print... Save As...

- 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 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), monitor the SIP signaling messages between Bell Canada and Avaya IP Office on an outbound call from the enterprise to the PSTN, and verify the SIP signaling message exchanges for Digest Authentication:

1. Bell Canada SIP Trunking service returned a “401 Unauthorized” status message to the initial INVITE from the Avaya IP Office. The 401 message contained a **WWW-Authenticate** Header posing challenge for Digest Authentication.

Example of **WWW-Authenticate** Header:

```
WWW-Authenticate: DIGEST
qop="auth", nonce="BroadWorksXgmdj14a7T987am0BW", realm="siptrunking.bell.ca", algorithm=MD5
```

2. Avaya IP Office ACKed the above 401 message, then presented the Digest Authentication response by sending a second INVITE that contained an **Authorization** Header supplying the information for successful Digest Authentication. Note the username as configured in **Section 5.4 (SIP Credentials tab)**.

Example of **Authorization** Header:

```
Authorization: Digest
username="4167751396", realm="siptrunking.bell.ca", nonce="BroadWorksXgmdj14a7T987am0BW", response="89f1e3b7c2d52ad7d13fb58c34aee1e3", uri="sip:19085551212@siptrunking.bell.ca:5060", algorithm=MD5, qop=auth, nc=00000001, cnonce="f8c60f3798082a2a6915"
```

3. Bell Canda SIP Trunking service returned “100 Trying” and subsequent 18X call ringing or session progress messages signaling normal call progression.

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 7.0 IP Office Standard Version Installation*, Document number 15-601042, March 2011.
- [2] *IP Office Release 7.0 Manager 9.0*, Document number 15-601011, April 2011.
- [3] *IP Office Release 7.0 Voicemail Pro Installation and Maintenance*, Document Number 15-601063, March 2011.
- [4] *IP Office Release 6.0 System Status Application*, Document number 15-601758, February 2010.
- [5] *IP Office System Monitor*, Document Number 15-601019, November 28, 2008

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

Product documentation for Bell Canada SIP Trunking is available from Bell Canada.

Appendix: SIP Line Template

Avaya IP Office Release 7.0 supports SIP Line Template (in xml format) that can be created from an existing configuration and imported into a new installation to simplify configuration procedures as well as to reduce potential configuration errors.

Note that not all of the configuration information, particularly items relevant to specific installation environment, is included in the SIP Line Template. Therefore it is critical that the SIP Line configuration be verified/updated after a template has been imported and additional configuration be supplemented using **Section 5.4** in these Application Notes as a reference.

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

```
<?xml version="1.0" encoding="utf-8"?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20110419</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>SIP Line Template for Bell Canada</DescriptiveName>
  <ITSPDomainName>cust2-tor.vsac.bell.ca</ITSPDomainName>
  <SendCallerID>CallerIDNone</SendCallerID>
  <ReferSupport>true</ReferSupport>
  <ReferSupportIncoming>1</ReferSupportIncoming>
  <ReferSupportOutgoing>1</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>true</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <ITSPProxy>207.236.222.111</ITSPProxy>
  <LayerFourProtocol>SipUDP</LayerFourProtocol>
  <SendPort>5060</SendPort>
  <ListenPort>5060</ListenPort>
  <DNSServerOne>0.0.0.0</DNSServerOne>
  <DNSServerTwo>0.0.0.0</DNSServerTwo>
  <CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>
  <SeparateRegistrar />
  <CompressionMode>AUTOSELECT</CompressionMode>
  <CallInitiationTimeout>4</CallInitiationTimeout>
  <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
  <VoipSilenceSupression>false</VoipSilenceSupression>
  <ReinviteSupported>true</ReinviteSupported>
  <FaxTransportSupport>FOIP_NONE</FaxTransportSupport>
  <UseOffererPreferredCodec>false</UseOffererPreferredCodec>
  <CodecLockdown>false</CodecLockdown>
  <T38FaxVersion>3</T38FaxVersion>
  <Transport>UDPTL</Transport>
  <LowSpeed>0</LowSpeed>
```

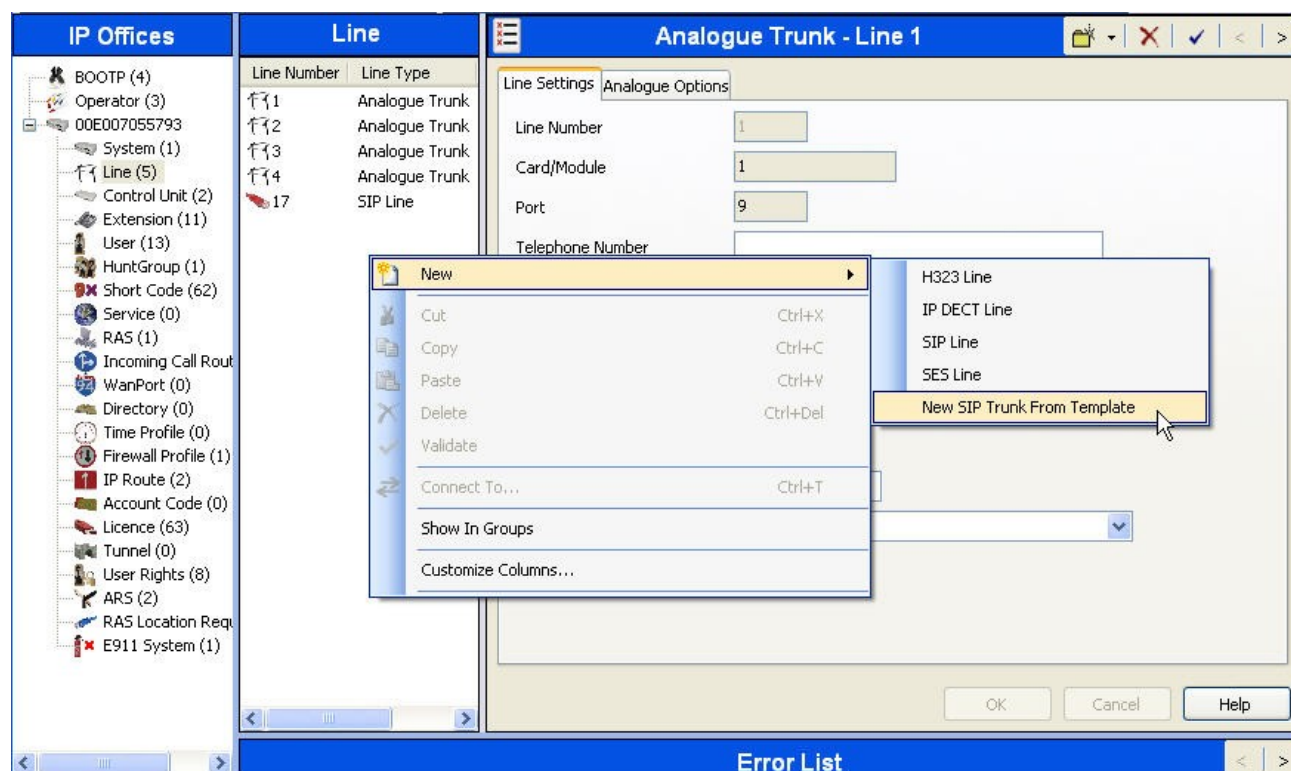
```

<HighSpeed>0</HighSpeed>
<TCFMethod>Trans_TCF</TCFMethod>
<MaxBitRate>FaxRate_14400</MaxBitRate>
<EflagStartTimer>2600</EflagStartTimer>
<EflagStopTimer>2300</EflagStopTimer>
<UseDefaultValues>true</UseDefaultValues>
<ScanLineFixup>true</ScanLineFixup>
<TFOPENenhancement>true</TFOPENenhancement>
<DisableT30ECM>false</DisableT30ECM>
<DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
<DisableT30MRCompression>false</DisableT30MRCompression>
<NSFOVERRIDE>false</NSFOVERRIDE>
<SIPCredentials>
  <Expiry>60</Expiry>
  <RegistrationRequired>false</RegistrationRequired>
</SIPCredentials>
</Template>

```

To import the above template into a new installation:

1. Copy and paste the above template into a text document named **US_Bell Canada_SIPTrunk.xml** on the PC where IP Office Manager was installed.
2. Import the template into an IP Office installation by creating a new SIP Line as shown in the screenshot below (right clicking **Line** in the left navigation pane):



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



Template Type Selection

Locale: United States (US English)

Country: United States

Service Provider: Bell Canada

☐ Display All

Create new SIP Trunk Cancel

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