



DevConnect Program

Application Notes for Configuring Bell Canada SIP Trunking with Avaya IP Office Server Edition Release 11.1 - 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 Server Edition Release 11.1.

Bell Canada SIP Trunking Service 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.

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

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 Avaya DevConnect Program.

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between Bell Canada and Avaya IP Office Server Edition solution. In the sample configuration, the Avaya IP Office Server Edition solution consists of the Primary Server running the Avaya IP Office Server Edition Linux software Release 11.1, Avaya IP Office Server Edition Expansion System (IP500 V2), Avaya Voicemail Pro, WebRTC and one-X Portal services enabled, Avaya Communicator for Web, Avaya Workplace for Windows, Avaya H.323 and Avaya SIP Deskphones, digital and analog endpoints.

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

2. General Test Approach and Test Results

The general test approach was to simulate an enterprise site in the DevConnect Program by connecting IP Office to Bell Canada's SIP Trunking service across the public internet. The configuration 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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya and Bell Canada products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

2.1. Interoperability Compliance Testing

To verify SIP trunking interoperability, the following features and functionality were exercised during the interoperability compliance test:

- Incoming PSTN calls to various phone types. Phone types included H.323, SIP, digital, and analog phones 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 phones 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 from/to the Avaya Workplace for Windows (SIP)
- Inbound and outbound PSTN calls from/to the Avaya Communicator for Web (WebRTC) with basic telephony transfer feature
- Inbound and outbound long hold time call stability
- Various call types including: local, long distance, international call, outbound toll-free and outbound call to 411 service
- SIP transport UDP/RTP and Port 5060 between Bell Canada and the simulated Avaya enterprise site
- Codec G.711MU, 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
- SIP OPTIONS queries and responses
- SIP Authentication
- Voicemail navigation for inbound and outbound calls
- Telephony features such as hold and resume, transfer, and conference
- T.38 fax
- Off-net call forwarding
- Off-net call transfer
- Twinning to mobile phones on inbound calls
- Note: Bell Canada's Static/ Dynamic ONND (Outgoing Name & Number Display): In order to test this feature, Bell Canada requested to modify the header's parameters in the FROM/PAI/DIVERSION/CONTACT SIP headers, however Avaya IP Office did not have the functionality to modify the header's parameters. Therefore, this test was not applicable.

Items not supported or not tested included the following:

- Bell Canada does not support SIP Registration
- Bell Canada does not support TLS/SRTP
- Bell Canada supports inbound toll-free service in production, however it is not available in their test lab during the compliance testing
- Bell Canada supports outbound operator 0 call in production, however it is not available in their test lab during the compliance testing

- Bell Canada supports outbound 911 call in production, however it is not available in their test lab during the compliance testing

2.2. Test Results

Interoperability testing of Bell Canada was completed with successful results for all test cases.

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
<https://business.bell.ca/shop/enterprise/sip-trunking-service>.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the Bell Canada network through the public internet. 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.

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

- IP Office Server Edition Primary Server
- IP Office Voicemail Pro
- IP Office Server Edition Expansion System (IP500 V2)
- WebRTC and one-X Portal services
- Avaya 96x1 Series IP Deskphones (H.323)
- Avaya 11x0 Series IP Deskphones (SIP)
- Avaya J129 IP Deskphones (SIP)
- Avaya 1408 Digital phones
- Avaya Analog phones
- Avaya Communicator for Web
- Avaya Workplace for Windows (SIP)

The Primary Server consists of a Dell PowerEdge R640 server, running the Avaya IP Office Server Edition Linux software Release 11.1. Avaya Voicemail Pro runs as a service on the Primary Server. The LAN1 port of the Primary Server (Eth0) is connected to the enterprise LAN (Private network) while the LAN2 port is connected to the public network. The SIP trunk to the Bell Canada system is connected to LAN2 port of the Avaya IP Office Server Edition.

The optional Expansion System (IP500 V2) is used for the support of digital, analog, fax, and additional IP stations. It consists of an Avaya IP Office IP500V2 with the MOD DGTLS16 expansion module which provides connections for 16 digital stations, the PHONE 8 card which provides connections for 8 analog stations, as well as a 64-channel VCM (Voice Compression Module) for supporting VoIP codecs.

A separate Windows 10 Enterprise PC runs Avaya IP Office Server Edition Manager to configure and administer Avaya IP Office Server Edition system.

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

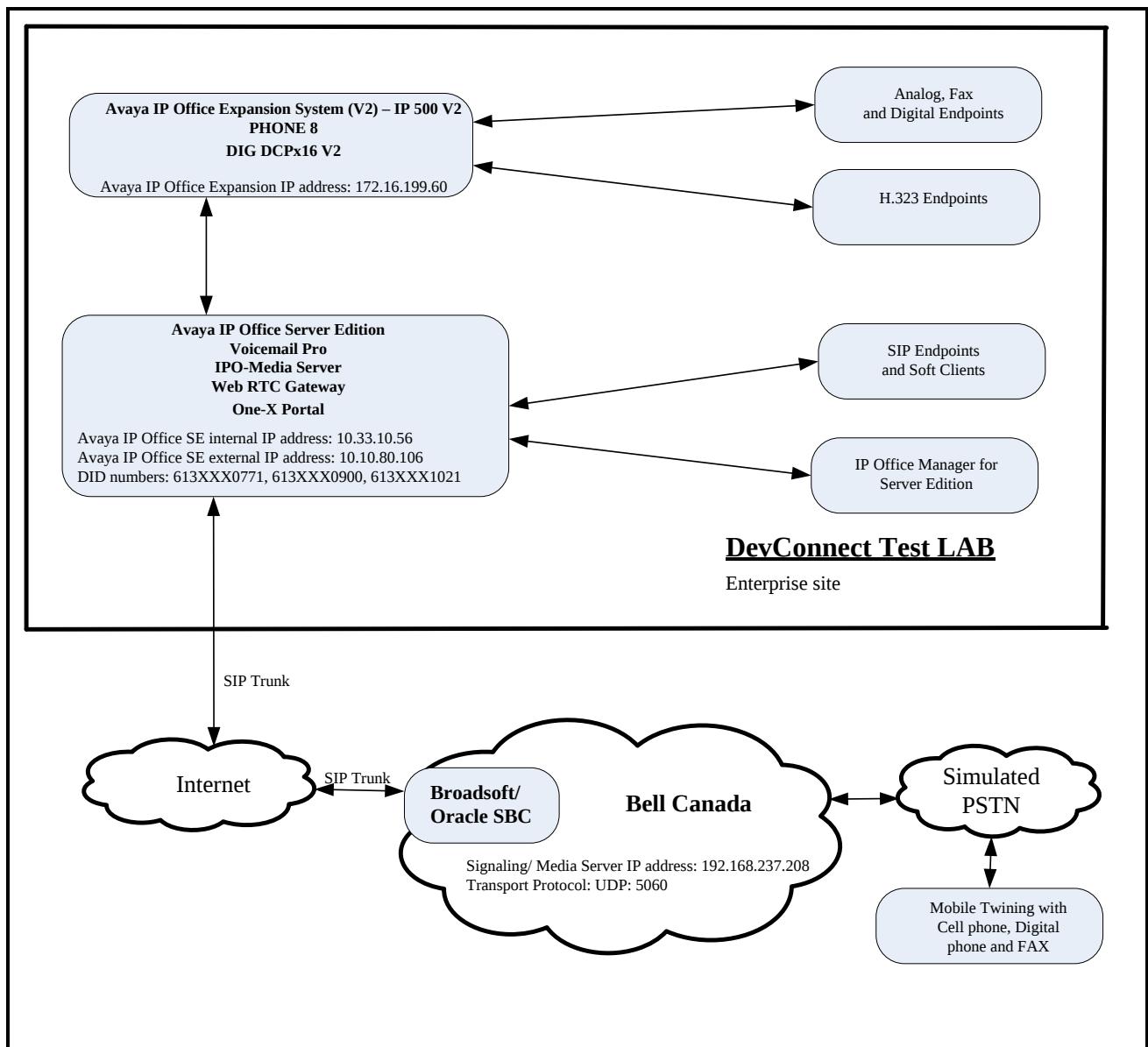


Figure 1 - Test Configuration for Avaya IP Office Server Edition with Bell Canada SIP Trunk Service

Inbound calls from the service provider via the SIP trunk arrive to the Server Edition Primary Server, where Incoming Call Routes are checked to determine the call destination. In the event that the destination of the incoming call is an endpoint in the Expansion System (IP500 V2), the call is sent via the Small Community Network (SCN) H.323 trunk (IP Office Line) to the expansion IP500V2 for routing to the final endpoint. This SCN H.323 trunk is automatically created during the initial process of addition of the Expansion System to the IP Office Server Edition solution.

Similarly, outbound calls from the enterprise to the PSTN are routed via the SIP trunk to the Bell Canada network. Calls originated from extensions registered to the Primary Server are routed directly to Bell Canada, while calls originated from extensions on the Expansion System are sent to the Primary Server via SCN H.323 trunk, before being routed to Bell Canada via the SIP trunk.

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 the compliance test, outbound calls to Canadian numbers within the North American Numbering Plan (NANP) were tested. The user would dial 11 (1 + 10) digits. 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, and it was configured to send 10 digits in the From field. For inbound calls, Bell Canada 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 Server Edition, such as a 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 SRTP traffic between the service provider and Avaya IP Office Server Edition 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:

Component	Version
Avaya	
Avaya IP Office Server Edition solution	
▪ Primary Server Dell PowerEdge R640 – IPO-Linux-PC	11.1.2.4.0 build 18
▪ IPO-Media Server	11.1.2.4.0 build 18
▪ Voicemail Pro	11.1.2.4.0 build 2
▪ Web RTC Gateway	11.1.2.3.0 build 2
▪ one-X Portal	11.1.2.4.0 build 3
▪ IP Office Manager for Server Edition	11.1.2.4.0 build 18
▪ IP Office Expansion System (V2) – IP 500 V2	11.1.2.4.0 build 18
▪ IP Office Analogue - PHONE 8	11.1.2.4.0 build 18
▪ IP Office Digital - DIG DCPx16 V2	11.1.2.4.0 build 18
Avaya 1140E IP Deskphone (SIP)	04.04.33
Avaya 9641G IP Deskphone (H323)	6.8.5.3.2
Avaya 9621G IP Deskphone (H323)	6.8.5.3.2
Avaya J129 IP Deskphone (SIP)	4.0.7.1.5
Avaya Communicator for Web	1.0.20.1722
Avaya Workplace Client for Windows	3.28.0.73
Avaya 1408D Digital Deskphone	R48
Avaya Analog Deskphone	N/A
VentaFax	7.10.258.664
Bell Canada	
Broadsoft SoftSwitch	22 sp1
Oracle SBC	7.4.0 m2p4

Table 1: Equipment and Software Tested

Note – Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2, and also when deployed with all configurations of IP Office Server Edition. IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks.

5. Configure Avaya IP Office Server Edition Solution

This section describes the Avaya IP Office Server Edition solution configuration necessary to support connectivity to the Bell Canada. It is assumed that the initial installation and provisioning of the Server Edition Primary Server and Expansion System has been previously completed and therefore is not covered in these Application Notes. For information on these installation tasks, refer to the Additional References **Section 9**.

This section describes the Avaya IP Office Server Edition configuration to support connectivity to Bell Canada system. Avaya IP Office Server Edition is configured through the Avaya IP Office Server Edition Manager PC application. From a PC running the Avaya IP Office Server Edition Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office Server Edition system from the pop-up window. Log in using appropriate credentials.

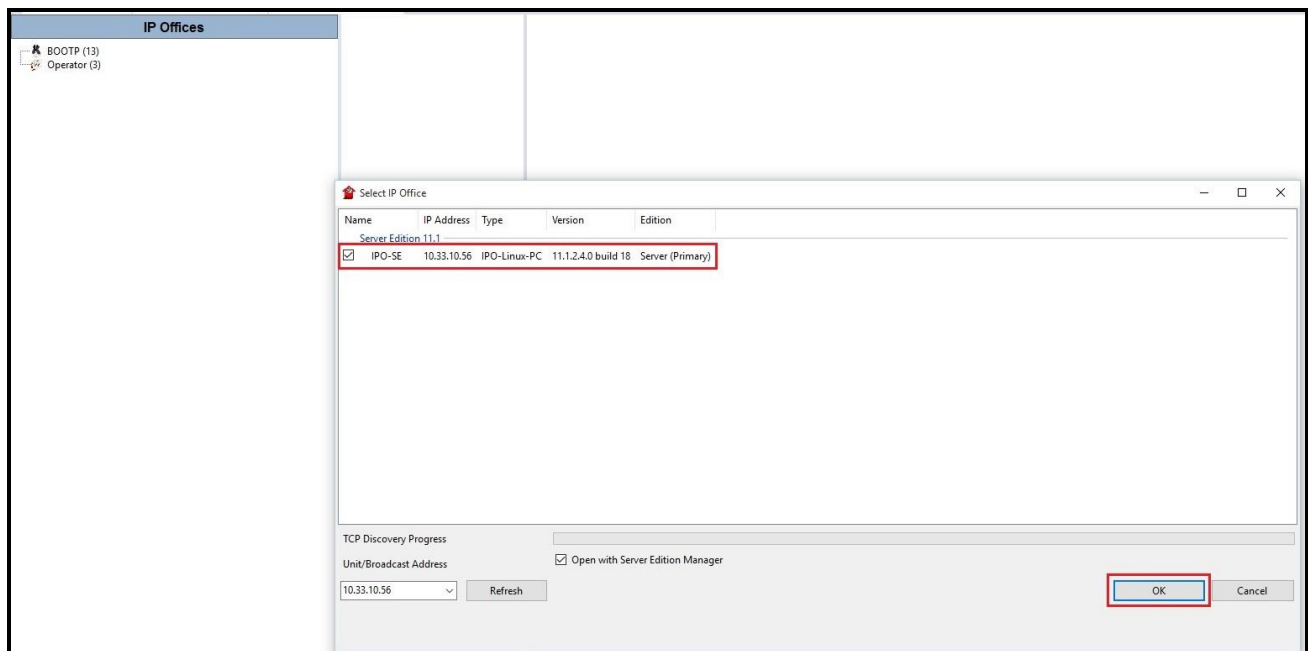


Figure 2 – Avaya IP Office Server Edition Selection

The appearance of the Avaya IP Office Server Edition Manager can be customized using the **View** menu. In the screens presented in this section, it includes the system inventory of the servers and links for administration and configuration tasks.

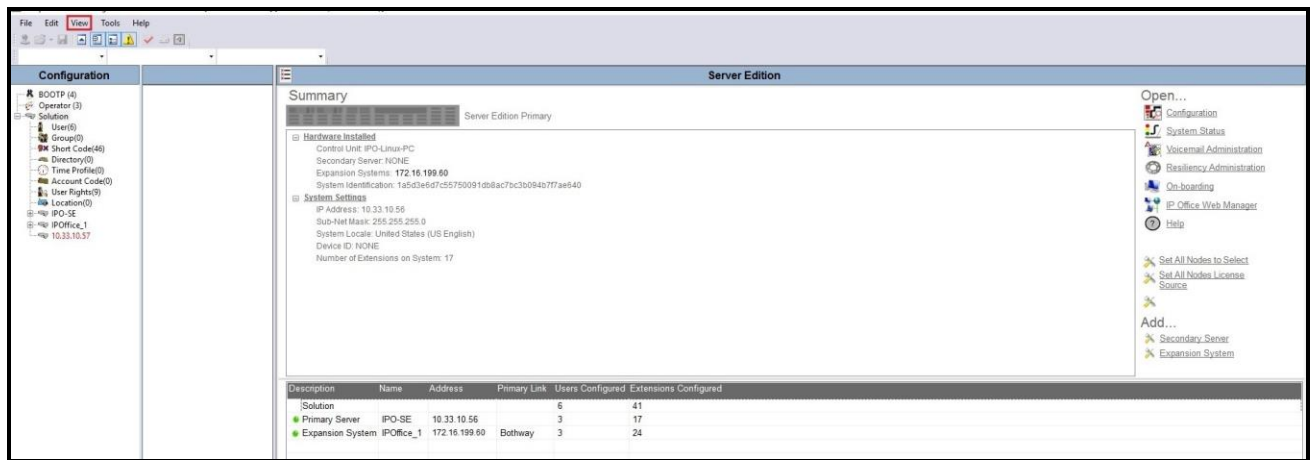


Figure 3 – Avaya IP Office Server Edition View Menu

The configuration and features described in these Application Notes require the Avaya IP Office Server Edition system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

Configuration	License																																																																																																									
- BOOTP (4) - Operator (3) - Solution - User(10) - Group(0) - Short Code(46) - Directory(0) - Time Profile(0) - Account Code(0) - User Rights(9) - Location(0) - IPOSEE - System (1) - Line (3) - Control Unit (8) - Extension (6) - User (7) - Group (0) - Short Code (2) - Service (0) - Incoming Call Route (7) - IP Route (2) - License(7) - MRS (1) - Location (0)	License Type Status License Remote Server License Mode WebLM Normal Licensed Version 11.0 <table border="1"> <thead> <tr> <th>Feature</th><th>Instances</th><th>Status</th><th>Expiration Date</th><th>Source</th></tr> </thead> <tbody> <tr><td>Additional Voicemail Pro Ports</td><td>6</td><td>Valid</td><td>Never</td><td>WebLM</td></tr> <tr><td>Power User</td><td>4</td><td>Valid</td><td>Never</td><td>WebLM</td></tr> <tr><td>Avaya IP endpoints</td><td>6</td><td>Valid</td><td>Never</td><td>WebLM</td></tr> <tr style="outline: 2px solid red;"><td>SIP Trunk Channels</td><td>50</td><td>Valid</td><td>Never</td><td>WebLM</td></tr> <tr><td>Server Edition</td><td>1</td><td>Valid</td><td>Never</td><td>WebLM</td></tr> <tr><td>SM Trunk Channels</td><td>2</td><td>Valid</td><td>Never</td><td>WebLM</td></tr> <tr><td>UMS Web Services</td><td>2</td><td>Valid</td><td>Never</td><td>WebLM</td></tr> <tr><td> </td><td> </td><td> </td><td> </td><td> </td></tr> <tr><td> </td><td> </td><td> </td><td> </td><td> </td></tr> <tr><td> </td><td> </td><td> </td><td> </td><td> </td></tr> <tr><td> </td><td> </td><td> </td><td> </td><td> </td></tr> <tr><td> </td><td> </td><td> </td><td> </td><td> </td></tr> <tr><td> </td><td> </td><td> </td><td> </td><td> </td></tr> <tr><td> </td><td> </td><td> </td><td> </td><td> </td></tr> <tr><td> </td><td> </td><td> </td><td> </td><td> </td></tr> <tr><td> </td><td> </td><td> </td><td> </td><td> </td></tr> <tr><td> </td><td> </td><td> </td><td> </td><td> </td></tr> <tr><td> </td><td> </td><td> </td><td> </td><td> </td></tr> <tr><td> </td><td> </td><td> </td><td> </td><td> </td></tr> <tr><td> </td><td> </td><td> </td><td> </td><td> </td></tr> </tbody> </table>	Feature	Instances	Status	Expiration Date	Source	Additional Voicemail Pro Ports	6	Valid	Never	WebLM	Power User	4	Valid	Never	WebLM	Avaya IP endpoints	6	Valid	Never	WebLM	SIP Trunk Channels	50	Valid	Never	WebLM	Server Edition	1	Valid	Never	WebLM	SM Trunk Channels	2	Valid	Never	WebLM	UMS Web Services	2	Valid	Never	WebLM																																																																	
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5.2. System Settings

5.2.1. System – LAN Tab

To configure the LAN2 settings on the Primary Server, complete the following steps. Navigate to **IPO-SE → System (1)** 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 Server Edition LAN2 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** to submit the change.

The screenshot shows the Avaya IP Office configuration interface. On the left is a 'Configuration' tree with various system components. The 'System' tab is active, showing 'IPO-SE'. The 'LAN2' port is selected in the 'System' list. The 'LAN Settings' tab is active, displaying the following fields:

System	LAN1	LAN2	DNS	Voicemail	Telephony	Directory Services	System Events	SMTP	SMDR	VoIP	Contact Center	Avaya Cloud Services	Avaya Push Notification Service
LAN Settings													
IP Address		10 . 10 . 80 . 106											
IP Mask		255 . 255 . 255 . 128											
Number Of DHCP IP Addresses		22											
DHCP Mode		☐ Server ☐ Client ☒ Disabled											
Advanced													
OK Cancel Help													

Figure 5 - Avaya IP Office Primary Server LAN2 Settings

The **VoIP** tab as shown in the screenshot below was configured with following settings:

- Check the **H323 Gatekeeper Enable** to allow Avaya IP Deskphones/Softphones using the H.323 protocol to register
- Check the **SIP Trunks Enable** to enable the configuration of SIP Trunk connecting to Bell Canada system
- Verify **Keepalives** to select **Scope** as **RTP-RTCP** with **Periodic timeout 60** and select **Initial keepalives** as **Enabled**
- All other parameters should be set according to customer requirements
- Click **OK** to submit the changes

The screenshot displays the 'LAN2' configuration page in the 'IPO-SE*' application. The 'VoIP' tab is selected under 'LAN Settings'. The 'H323 Gatekeeper Enable' checkbox is checked. Below it, 'Auto-create Extension' and 'Auto-create User' are unchecked, while 'H.323 Remote Extension Enable' is checked. 'H.323 Signaling over TLS' is set to 'Disabled' and 'Remote Call Signaling Port' is 1720. The 'SIP Trunks Enable' checkbox is also checked. Under 'SIP Registrar Enable', 'Auto-create Extension/User' is unchecked, 'SIP Remote Extension Enable' is unchecked, and 'Allowed SIP User Agents' is set to 'Block blacklist only'. The 'SIP Domain Name' and 'SIP Registrar FQDN' fields are empty. For 'Layer 4 Protocol', 'UDP', 'TCP', and 'TLS' are all checked. Their respective ports are: UDP Port 5060, Remote UDP Port 5060, TCP Port 5060, Remote TCP Port 5060, TLS Port 5061, and Remote TLS Port 5061. 'Challenge Expiration Time (sec)' is 10. In the 'RTP' section, 'Port Number Range' has Minimum 40750 and Maximum 50750. 'Port Number Range (NAT)' also has Minimum 40750 and Maximum 50750. 'Enable RTCP Monitoring on Port 5005' is checked, and 'RTCP collector IP address for phones' is 0.0.0.0.0. The 'Keepalives' section is highlighted with a red box, showing 'Scope' as 'RTP-RTCP', 'Periodic timeout' as 60, and 'Initial keepalives' as 'Enabled'. The 'OK' button at the bottom is also highlighted with a red box.

Figure 6 - Avaya IP Office Primary Server LAN2 VoIP

To configure the LAN1 settings tab for the Expansion System, navigate to **Solution → IPOffice_1 → System (1)** in the Navigation and Group Panes and then navigate to the **LAN1 → LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields should be populated with the values assigned during the Expansion System initial installation process. Verify the configuration or modify the values if needed. While DHCP was disabled during the compliance test, this parameter should be set according to customer requirements. Other settings were left at their default values. Click **OK** to submit the change.

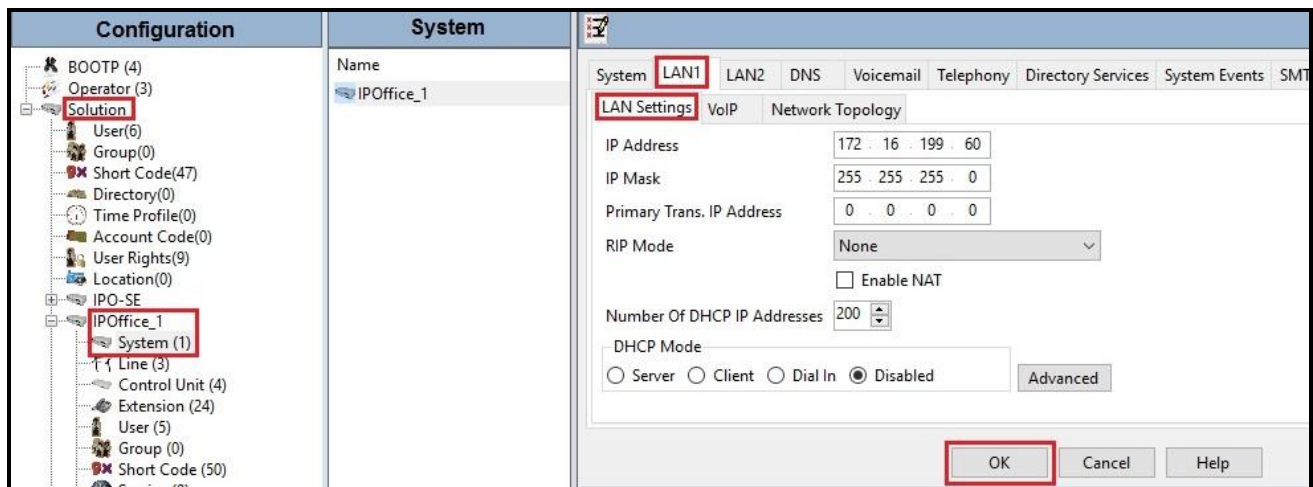


Figure 7 - Avaya IP Office Expansion Server Settings

The **VoIP** tab for LAN1 in the Expansion System (not shown) can be configured using the same values previously described for the **VoIP** tab in the Primary Server.

5.2.2. System – Telephony Tab

Navigate to **Solution → IPO-SE → System (1)** in the Navigation and Group Panes (not shown) and then navigate to the **Telephony → Telephony** tab in the Details Pane. Choose the **Companding Law** typical for the enterprise location. For North American area, **U-Law** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfers to the PSTN via the service provider across the SIP trunk. The Hold Timeout (sec) field controls how long calls remain on hold before being alerted to the user and should be set based on the customer's requirement. Set **Default Name Priority** to **Favor Trunk** to have IP Office display the name provided in the Caller ID from the SIP trunk. Defaults were used for all other settings. Click **OK** to submit the changes.

The screenshot shows the 'IPO-SE' configuration window with the 'Telephony' tab selected. The 'Telephony' sub-tab is also selected. The 'Companding Law' section is expanded, showing 'U-Law' selected for both 'Switch' and 'Line'. The 'Inhibit Off-Switch Forward/Transfer' checkbox is unchecked. The 'Hold Timeout (sec)' is set to 3600. The 'Default Name Priority' is set to 'Favor Trunk'. The 'OK' button is highlighted.

Field	Value
Dial Delay Time (sec)	4
Dial Delay Count	0
Default No Answer Time (sec)	15
Hold Timeout (sec)	3600
Park Timeout (sec)	300
Ring Delay (sec)	5
Call Priority Promotion Time (sec)	Disabled
Default Currency	USD
Default Name Priority	Favor Trunk
Media Connection Preservation	Enabled
Phone Failback	Automatic
Login Code Complexity	Enforcement
Minimum length	6
Complexity	Complexity
RTCP Collector Configuration	Send RTCP to an RTCP Collector
Server Address	0 . 0 . 0 . 0
UDP Port Number	5005
RTCP reporting interval (sec)	5

Figure 8 - Avaya IP Office Primary Server Telephony

Navigate to **Solution → IPOOffice_1 → System (1)** (not shown) and repeat the steps above to configure the **Telephony** settings for the Expansion System.

5.2.3. System – VoIP Tab

Navigate to **Solution → IPO-SE → System (1)** in the Navigation and Group Panes and then navigate to the **VoIP** tab in the Details Pane. Leave the **RFC2833 Default Payload** as the default value of **101**. The buttons between the two lists can be used to move codecs between the **Unused** and **Selected** lists, and to change the order of the codecs in the **Selected** codecs list. By default, all IP lines and phones (SIP and H.323) will use the system default codec selection shown here, unless configured otherwise for a specific line or extension. The example below shows the codecs used for IP phones (SIP and H.323), the system's default codecs and order were used. Click **OK** (Not shown) to submit the changes.

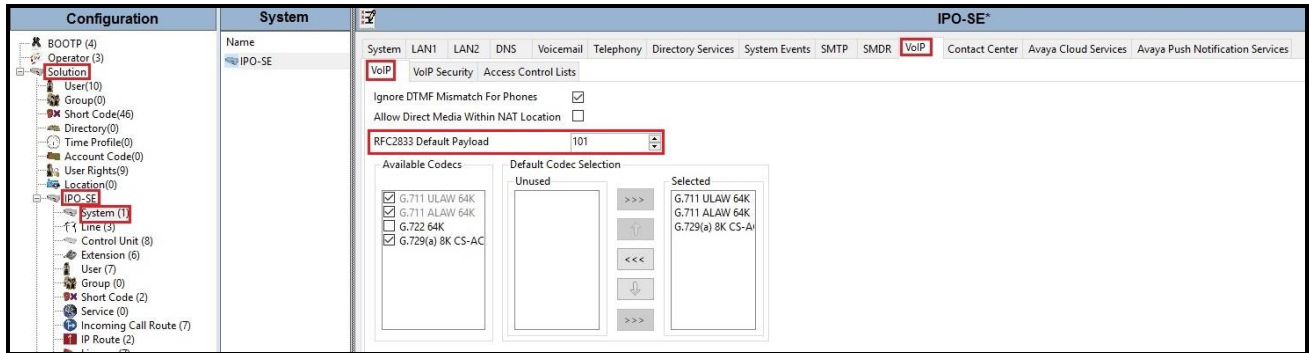


Figure 9 - Avaya IP Office Primary Server VoIP

Note: The codec selections defined under this section (VoIP – VoIP tab) are the codecs selected for the IP phones/extensions. The codec selections defined under **Section 5.4.2** (SIP Line – VoIP tab) are the codecs selected for the SIP Line (Trunk).

5.3. IP Route

Create an IP route to specify the IP address of the gateway or router where the IP Office needs to send the packets in order to route calls to Bell Canada.

To create an IP route for the Primary system, navigate to **Solution → IPO-SE → IP Route**, right-click on **IP Route** and select **New** (Not shown). The values used during the compliance test are shown below:

- Set the **IP Address** and **IP Mask** to **0.0.0.0** to make this the default route
- Set **Gateway IP Address** to the IP address of the gateway/router used to route calls to the public network, e.g., **10.10.80.1**
- Set **Destination** to **LAN2** from the pull-down menu
- Click **OK** to commit

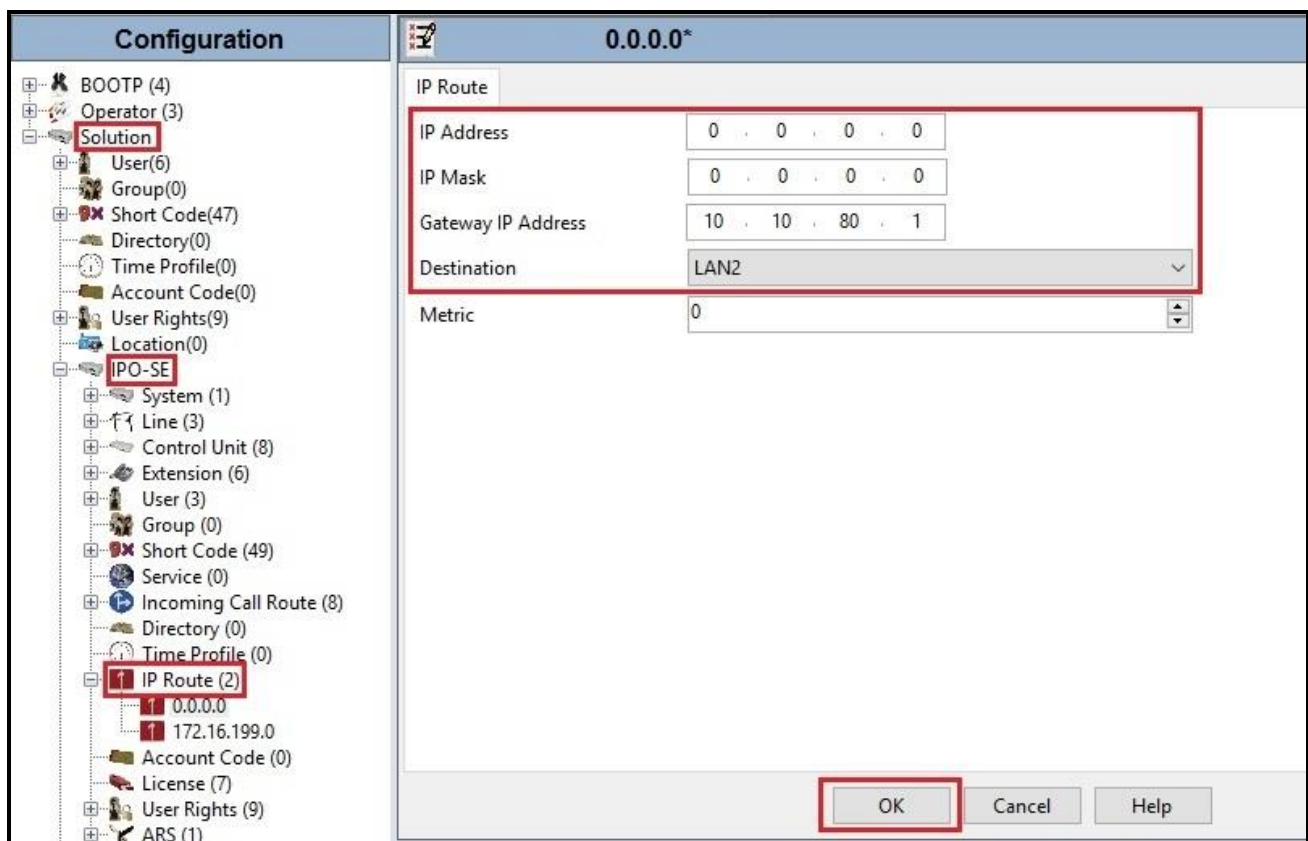


Figure 10 - Avaya IP Office Primary Server IP Route

To create an IP route for the Expansion system, navigate to **Solution → IPOffice_1 → IP Route**, right-click on **IP Route** and select **New** (Not shown). The values used during the compliance test are shown below:

- Set the **IP Address** and **IP Mask** to **0.0.0.0** to make this the default route

- Set **Gateway IP Address** to the IP address of the gateway/router used to route calls to the private network, e.g., **172.16.199.1**
- Set **Destination** to **LAN1** from the pull-down menu
- Click **OK** to commit

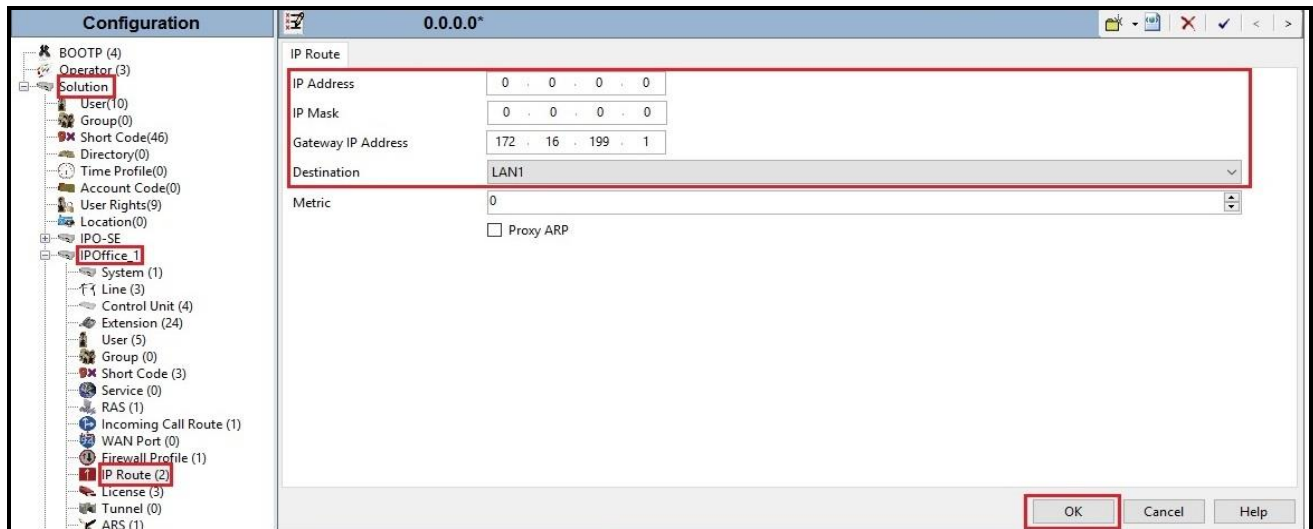


Figure 11 - Avaya IP Office Expansion Server IP Route

5.4. Administer SIP Line

A SIP Line is needed to establish the SIP connection between Avaya IP Office Server Edition and Bell Canada system. 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 Avaya IP Office Server Edition 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 **Section 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
- SIP Advanced Engineering

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 **Section 5.4.2**.

For the compliance test, SIP Line 17 was used as trunk for both outgoing and incoming calls.

5.4.1. Create SIP Line from an XML Template

SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500 V2) and IP Office Server Edition systems. Alternatively, binary templates may be generated. However, binary templates include all the configuration parameters of the Trunk, including sensitive customer specific information. Therefore, binary templates should only be used for cloning trunks within a specific customer's environment

Create a new folder in a location where Avaya IP Office Server Edition Manager is installed (e.g., C:\Bell Canada\Template). Copy the template file to this folder and rename the template file to **BC-IPO11_1.xml** (for SIP Line 17).

Create the SIP Trunk from the template, from the Primary server, right-click on **Line** in the Navigation Pane, then navigate to **New from Template** → **Open from file**.

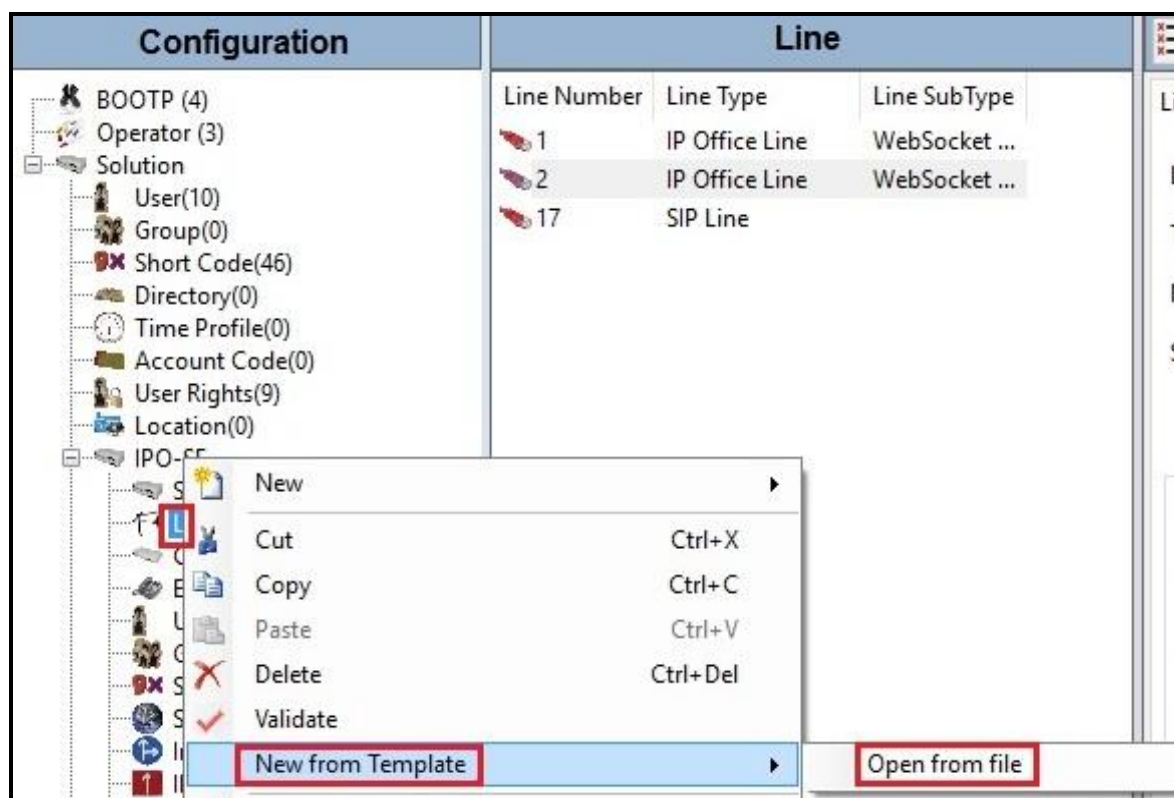


Figure 12 – Create SIP Line from an XML Template

Select the **Template Files (*.xml)** and select the copied template at folder (e.g., C:\Bell Canada\Template). Click **Open** button to create a SIP line from template.

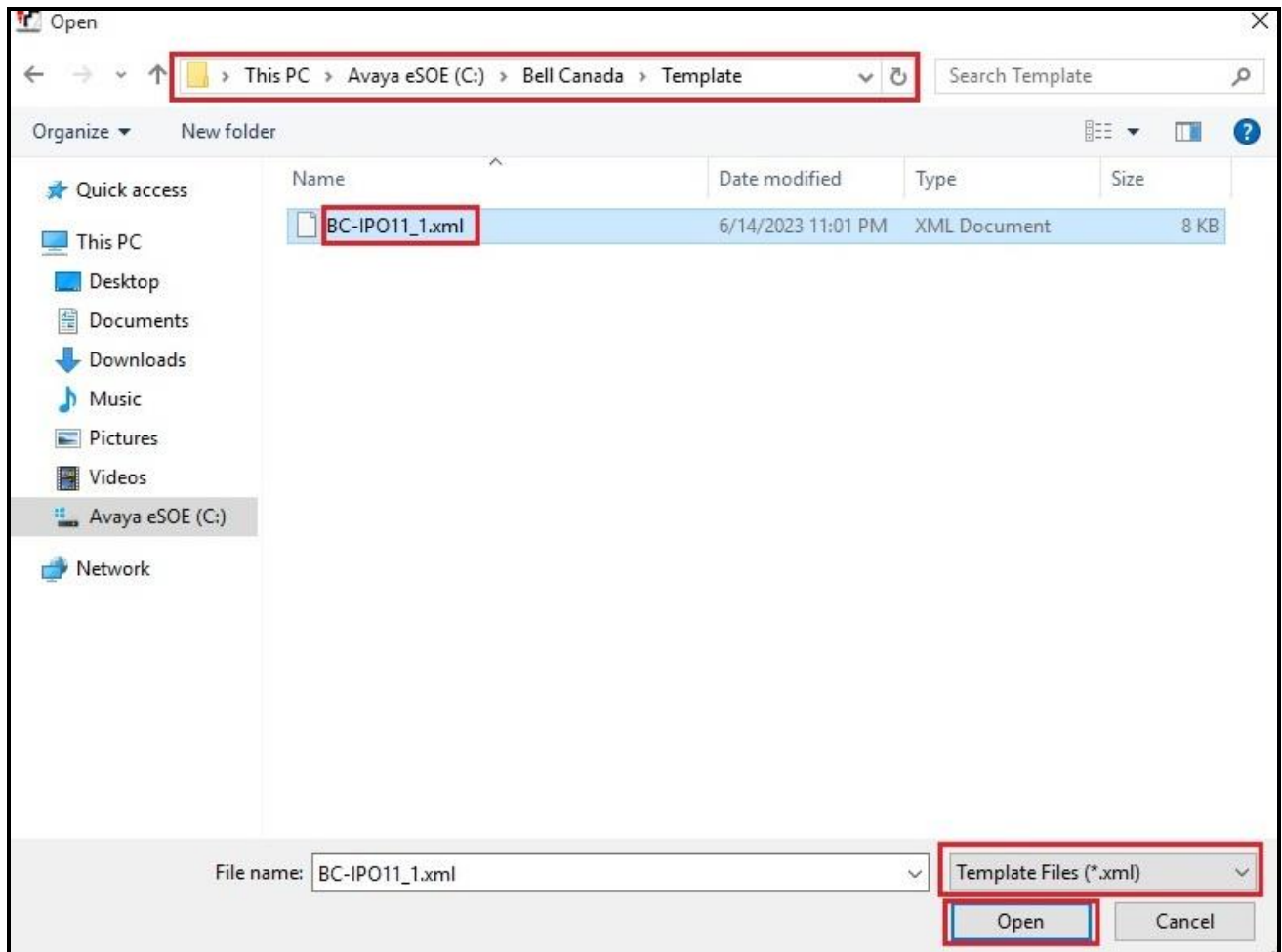


Figure 13 – Create SIP Line from directory

A pop-up window below will appear stating success (or failure). Then click **OK** to continue.



Figure 14 – Create SIP Line from Template successfully

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

5.4.2. Create SIP Line Manually

To create a SIP line, begin by navigating to **Line** in the left Navigation Pane, then right-click in the Group Pane and select **New** → **SIP Line** (not shown).

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

- Select available **Line Number: 17**
- Set **ITSP Domain Name** to the Bell Canada domain for production service. This field is used to specify the default host part of the SIP URI in the To and R-URI fields for outgoing calls
- Leave **Local Domain Name** to **vendor10.lab.internetvoice.ca**
- Check the **In Service** and **Check OOS** boxes
- Set **URI Type** to **SIP URI**
- For **Session Timers**, set **Refresh Method** to **Auto** with **Timer (sec)** to **On Demand**
- Set **Name Priority** to **Favor Trunk**. As described in **Section 5.2.2**, the **Default Name Priority** parameter may retain the default **Favor Trunk** setting or can be configured to **Favor Directory**. As shown below, the default **Favor Trunk** setting was used in the reference configuration
- For **Redirect and Transfer**, set **Incoming Supervised REFER** and **Outgoing Supervised Auto**. Note: Avaya IP Office uses the Allow header of the OPTIONS response to determine whether the endpoint supports REFER. In this case, Bell Canada responded without Allow: REFER. Therefore, Avaya IP Office does not send REFER if AUTO is configured.
- Default values may be used for all other parameters
- Click **OK** to commit then press **Ctrl + S** to save

The screenshot shows the Avaya IP Office configuration interface. The left pane shows the Configuration tree with 'Line 17' selected. The middle pane shows the 'Line' details for 'Line 17'. The right pane shows the 'SIP Line - Line 17' configuration. The configuration includes the following fields and settings:

- Line Number:** 17
- ITSP Domain Name:** siptrunking.bell.ca
- Local Domain Name:** vendor10.lab.internetvoice.ca
- URI Type:** SIP URI
- Location:** Cloud
- Prefix:** (empty)
- National Prefix:** (empty)
- International Prefix:** (empty)
- Country Code:** (empty)
- Name Priority:** Favor Trunk
- Description:** (empty)
- In Service:** ☒
- Check OOS:** ☒
- Session Timers:**
 - Refresh Method:** Auto
 - Timer (sec):** On Demand
- Redirect and Transfer:**
 - Incoming Supervised REFER:** Auto
 - Outgoing Supervised REFER:** Auto
- Send 302 Moved Temporarily:** ☐
- Outgoing Blind REFER:** ☐

The OK button is highlighted with a red box.

Figure 15 – SIP Line Configuration

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

- The **ITSP Proxy Address** was set to the IP addresses of Bell Canada signaling servers: **192.168.237.208** as shown in **Figure 1**. This is the SIP Proxy address used for outgoing SIP calls
- In the **Network Configuration** area, **UDP** was selected as the **Layer 4 Protocol** and the **Send Port** was set to **5060**
- The **Use Network Topology Info** parameter was set to **None**. The **Listen Port** was set to **5060**. Note: For the compliance testing, the **Use Network Topology Info** field was set to **None**, since no NAT was using in the test configuration. In addition, it was not necessary to configure the **System → LAN2 → Network Topology** tab for the purposes of SIP trunking. If a NAT is used between Avaya IP Office and the other end of the trunk, then the **Use Network Topology Info** field should be set to the LAN interface (**LAN2**) used by the trunk and the **System → LAN2 → Network Topology** tab needs to be configured with the details of the NAT device
- The **Calls Route via Registrar** was unchecked as Bell Canada did not support the dynamic Registration on the SIP Trunk
- Other parameters retain default values
- Click **OK** to commit then press Ctrl + S to save

The screenshot shows the 'SIP Line - Line 17*' configuration window. The 'Transport' tab is active. The 'ITSP Proxy Address' is set to 192.168.237.208. The 'Network Configuration' section shows 'Layer 4 Protocol' as UDP, 'Send Port' as 5060, 'Use Network Topology Info' as None, and 'Listen Port' as 5060. The 'Calls Route via Registrar' checkbox is unchecked. The 'Explicit DNS Server(s)' are set to 0.0.0.0. The 'Separate Registrar' field is empty. The 'OK' button is highlighted.

Figure 16 – SIP Line Transport Configuration

On the **SIP Credentials** tab in the Details Pane, click **Add** button to configure the parameters as shown below:

- **User name:** VEND10_613XXX0771_01A (Bell Canada provided this information)
- **Authentication Name:** VEND10_613XXX0771_01A (Bell Canada provided this information)
- **Contact:** VEND10_613XXX0771_01A (Bell Canada provided this information)
- **Password:** ***** (Bell Canada provided this information)
- **Confirm Password:** ***** (Bell Canada provided this information)
- **Expiration (mins):** 60
- Uncheck **Registration required** option
- Other parameters retain default values
- Click **OK** to commit then press Ctrl + S to save

SIP Line - Line 17*

SIP Line | Transport | Call Details | VoIP | **SIP Credentials** | SIP Advanced | Engineering

dex	User Name	Authentication Name	Contact	Expiration (mins)	Register
	VEND10_613XXX0771_01A	VEND10_613XXX0771_01A	VEND10_613XXX0771_01A	60	False

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

Edit SIP Credentials

User name: VEND10_613XXX0771_01A

Authentication Name: VEND10_613XXX0771_01A

Contact: VEND10_613XXX0771_01A

Password: *****

Confirm Password: *****

Expiration (mins): 60

Registration required: ☐

Buttons: OK, Cancel

Figure 17 – SIP Line SIP Credentials Configuration

The SIP URI entry must be created to match any DID number assigned to an Avaya IP Office user and Avaya IP Office will route the calls on this SIP line. Select the **Call Details** tab; click the **Add** button and the **New Channel** area will appear at the bottom of the pane (not shown). To edit an existing entry, click an entry in the list at the top, and click **Edit...** button. In the example screen below, a previously configured entry is edited

A 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:

- Associate this SIP line with an incoming line group in the **Incoming Group** field and an outgoing line group in the **Outgoing Group** field. This line group number will be used in defining incoming and outgoing call routes for this line. For the compliance test, a new line group **17** was defined that only contains this line (line 17)
- Select **Credentials** as **1:VEND10_613XXX0771_01A**
- Set **Max Sessions** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern
- Check **P Asserted ID** and **Diversion Header** options
- Set the **Local URI**, **Contact**, **P Asserted ID** and **Diversion Header** fields to the values shown in the screenshot below
- Click **OK** to submit the changes

SIP Line - Line 17

SIP Line: Transport **Call Details** VoIP SIP Credentials: SIP Advanced Engineering

SIP URIs

URI Groups Credential Local URI Contact P Asserted ID P Preferred ID Diversion Header Remote Party ID

SIP Line - 17 | Call Details | SIP URI

New URI

Incoming Group: 17 Max Sessions: 10

Outgoing Group: 17

Credentials: 1:VEND10_613XXX0771_0

Display		Content	
Local URI	Auto	Auto	
Contact	Auto	Auto	
P Asserted ID	☒ Auto	Auto	
P Preferred ID	☐ None	None	
Diversion Header	☒ Auto	Auto	
Remote Party ID	☐ None	None	

Field meaning

Outgoing Calls		Forwarding/Twinning		Incoming Calls	
Caller		Caller		Called	
Caller		Caller		Called	
Caller		Caller		Called	
None		None		None	
None		Caller		None	
None		None		None	

OK Cancel Help

Figure 18 – SIP Line Call Details Configuration

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. The **G.711 ULAW 64K** and **G.729(a) 8K CS-ACELP** codecs are selected. Avaya IP Office Server Edition supports these codecs, which are sent to the Bell Canada, in the Session Description Protocol (SDP) offer
- Check the **Re-invite Supported** box
- Set **Fax Transport Support** to **T38 Fallback** from the pull-down menu
- Set the **DTMF Support** to **RFC2833/RFC4733** from the pull-down menu. This directs Avaya IP Office Server Edition to send DTMF tones using RTP events messages as defined in RFC2833 and RFC4733
- Default values may be used for all other parameters
- Click **OK** to submit the changes

The screenshot shows the 'SIP Line - Line 17*' configuration window. The 'VoIP' tab is active. The 'Codec Selection' is set to 'Custom'. The 'Unused' list contains 'G.711 ALAW 64K'. The 'Selected' list contains 'G.711 ULAW 64K' and 'G.729(a) 8K CS-ACELP'. The 'Re-invite Supported' checkbox is checked. The 'Fax Transport Support' is set to 'T38 Fallback' and 'DTMF Support' is set to 'RFC2833/RFC4733'. The 'Media Security' is set to 'Disabled'. The 'OK' button is highlighted.

Figure 19 – SIP Line VoIP Configuration

5.5. IP Office Line in Primary System

In IP Office Server Edition systems, IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution. To edit an existing IP Office Line, select **Line** in the Navigation pane, and select the appropriate line to be configured in the Group pane.

To verify the IP Office line connecting the Primary System to the Expansion System, select **Line** on the navigation pane of Primary System and select the IP Office Line on the Group pane (line 2 on the screen below). Make note of the **Outgoing Group ID 99999** on the Details pane. The **Address of Gateway** is Avaya IP Office Expansion System LAN1 IP address **172.16.199.60**.

The screenshot displays the Avaya IP Office configuration interface. On the left, the 'Configuration' pane shows a tree view with 'Line' selected under 'IP Office Line'. The 'Line' pane shows a table with columns 'Line Number', 'Line Type', and 'Line SubType'. Line 2 is selected, showing 'IP Office Line' and 'WebSocket ...'. The 'IP Office Line - Line 2' configuration page is shown on the right. The 'Line' tab is active, displaying 'Line Number 2'. The 'Transport Type' is 'WebSocket Server', 'Networking Level' is 'SCN', and 'Security' is 'Medium'. The 'Gateway' section shows 'Address 172.16.199.60' and 'Location Cloud'. The 'Outgoing Group ID' is set to 99999. The 'SCN Resiliency Options' section is visible with checkboxes for 'Supports Resiliency', 'Backs up my IP phones', 'Backs up my hunt groups', and 'Backs up my IP DECT phones'.

Figure 20 – IP Office Line for Primary System

To verify the **VoIP Settings** of the IP Office line connecting the Primary System to the Expansion System, select **VoIP Settings** tab. The selected codecs are **G.711 ULAW 64K** and **G.729(a) 8K CS-ACELP**. Select **Fax Transport Support** to **T38 Fallback** (This setting should be as same as the VoIP settings in SIP line of Primary System and the VoIP settings in IP Office Line of Expansion System). Default values may be used for all other parameters. Click **OK** to submit the changes.

The screenshot displays the 'IP Office Line - Line 2*' configuration window, specifically the 'VoIP Settings' tab. The window is divided into several sections. At the top, there are tabs for 'Line', 'Short Codes', and 'VoIP Settings', with 'VoIP Settings' being the active tab. Below the tabs, there are two checkboxes: 'Out Of Band DTMF' and 'Allow Direct Media Path', both of which are checked. The main section is titled 'Codec Selection' and features a 'Custom' dropdown menu. Below this, there are two lists: 'Unused' and 'Selected'. The 'Unused' list contains 'G.711 ALAW 64K'. The 'Selected' list contains 'G.711 ULAW 64K' and 'G.729(a) 8K CS-ACELP'. Between these lists are four buttons: '>>>', '<<<', '<<<', and '>>>'. Below the codec lists, there are three more settings: 'Fax Transport Support' set to 'T38 Fallback', 'DTMF Support' set to 'RFC2833/RFC4733', and 'Media Security' set to 'Disabled'. At the bottom right, there are three buttons: 'OK', 'Cancel', and 'Help'. The 'OK' button is highlighted with a red box.

Figure 21 – IP Office Line for Primary System VoIP Settings

5.6. IP Office Line in Expansion System

To verify the IP Office line connecting the Expansion System to the Primary System, select Expansion Line on the navigation pane and select the IP Office Line on the Group pane (line 17 on the screen below). Make note of the **Outgoing Group ID 99999** on the Details pane. The **Address of Gateway** is Avaya IP Office Server Edition LAN1 IP address **10.33.10.56**.

The screenshot displays the Avaya IP Office configuration interface. On the left, the 'Configuration' pane shows a tree structure with 'IP Office 1' selected. The 'Line' pane shows a table with three lines: Line 1 (PRI 24 (Universal) PRI), Line 2 (PRI 24 (Universal) PRI), and Line 17 (IP Office Line WebSocket ...). The 'Details' pane for 'IP Office Line - Line 17' is shown on the right. It contains the following fields:

Line	Short Codes	VoIP Settings	T38 Fax
Line Number	17	Transport Type: WebSocket Client	Telephone Number: []
Networking Level	SCN	Security: Medium	Prefix: []
Gateway	Address: 10 . 33 . 10 . 56	Location: Cloud	Outgoing Group ID: 99999
	Password: []		Number of Channels: 250
	Confirm Password: []		Outgoing Channels: 250
			Port: 443
			SCN Resiliency Options: ☐ Supports Resiliency ☐ Backs up my IP phones ☐ Backs up my hunt groups ☐ Backs up my IP DECT phones
Description	[]		

At the bottom right, there are 'OK', 'Cancel', and 'Help' buttons.

Figure 22 – IP Office Line for Expansion System

To verify the **VoIP Settings** of the IP Office line connecting the Expansion System to the Primary Server, select **VoIP Settings** tab. The selected codecs are **G.711 ULAW 64K** and **G.729(a) 8K CS-ACELP** as System Default. Select **Fax Transport Support** to **T38 Fallback** (This setting should be as same as the VoIP settings in SIP line and IP Office Line of Primary System). Default values may be used for all other parameters. Click **OK** to submit the changes.

The screenshot displays the 'IP Office Line - Line 17*' configuration window. The 'VoIP Settings' tab is selected. The 'Codec Selection' section shows a 'Custom' dropdown. Below it, the 'Unused' list contains 'G.711 ALAW 64K', and the 'Selected' list contains 'G.711 ULAW 64K' and 'G.729(a) 8K CS-ACELP'. The 'Fax Transport Support' dropdown is set to 'T38 Fallback'. The 'DTMF Support' dropdown is set to 'RFC2833/RFC4733'. The 'Media Security' dropdown is set to 'Disabled'. On the right, there are checkboxes for 'VoIP Silence Suppression' (unchecked), 'Out Of Band DTMF' (checked), and 'Allow Direct Media Path' (checked). At the bottom right, the 'OK' button is highlighted with a red box.

Figure 23 – IP Office Line for Expansion Server VoIP Settings

To verify the **T38 Fax** of the IP Office line connecting the Expansion System to the Primary Server, select **T38 Fax** tab (Note: The T38 Fax tab is only active when Fax Transport Support is selected as T38 Fallback on VoIP Settings tab). Uncheck the **Use Default Values** at the bottom of the screen. Set the **T.38 Fax Version** to **0**. Default values may be used for all other parameters. Click the **OK** to submit the changes.

The screenshot shows the 'IP Office Line - Line 17*' configuration window with the 'T38 Fax' tab selected. The 'T38 Fax Version' is set to '0'. The 'Transport' is set to 'UDPTL'. The 'Redundancy' section shows 'Low Speed' and 'High Speed' both set to '0'. The 'TCF Method' is set to 'Trans TCF'. The 'Max Bit Rate (bps)' is set to '14400'. The 'EFlag Start Timer (ms)' is set to '2600'. The 'EFlag Stop Timer (ms)' is set to '2300'. The 'Tx Network Timeout (sec)' is set to '150'. The 'Use Default Values' checkbox is unchecked. The 'OK' button is highlighted.

Parameter	Value
T38 Fax Version	0
Transport	UDPTL
Low Speed	0
High Speed	0
TCF Method	Trans TCF
Max Bit Rate (bps)	14400
EFlag Start Timer (ms)	2600
EFlag Stop Timer (ms)	2300
Tx Network Timeout (sec)	150
Use Default Values	☐

Figure 24 – IP Office Line for Expansion Server T38 Fax

5.7. Outbound Short Code

Define a short code to route outbound traffic on the SIP line to Bell Canada. To create a short code, select **Short Code** in the left Navigation Pane, then right-click in the Group Pane and select **New** (not shown). 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 for Primary System used in the test configuration.

Navigate to **Solution → IPO-SE → Short Code**, right-click on **Short Code** and select **New**.

- 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. The value N represents the number dialed by the user. Note: Use the specific W in front of N for restricting all outbound calls
- Set the **Line Group ID** to the **Outgoing Group 17** defined on the **Call Details** tab on the **SIP Line** in **Section 5.4.2**. This short code will use this line group when placing the outbound call
- Set the **Locale** to **United States (US English)**
- Default values may be used for all other parameters
- Click **OK** to submit the changes

The screenshot displays the Avaya DevConnect configuration window. On the left, the 'Configuration' pane shows a tree structure with 'IPO-SE' selected. The 'Short Code' pane in the center lists various short codes, with '9N;' highlighted. The right pane shows the configuration details for the selected short code, titled '9N;: Dial'. The fields are: Code (9N;), Feature (Dial), Telephone Number (N), Line Group ID (17), and Locale (United States (US English)). There are checkboxes for 'Force Account Code' and 'Force Authorization Code', both of which are unchecked. At the bottom right, there are 'OK', 'Cancel', and 'Help' buttons. An 'Error List' pane is visible at the bottom.

Figure 25 – Short Code 9N for Primary Server

The screen below shows the details of the previously administered “9N;” short code for Expansion System used in the test configuration.

Navigate to **Solution → IPOffice_1 → Short Code**, right-click on **Short Code** and select **New**

- 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 (using Avaya analog or digital phones) dials 9 followed by any number
- Set **Feature** to **Dial**. This is the action that the short code will perform
- Set **Telephone Number** to 9N

- Set the **Line Group ID** to **99999** defined on the **Outgoing Group ID** of the IP Office line connecting the Expansion System to the Primary System. This short code will use this line group when placing the outbound call via Avaya IP Office Server Edition Primary Server
- Default values may be used for all other parameters
- Click **OK** to submit the changes

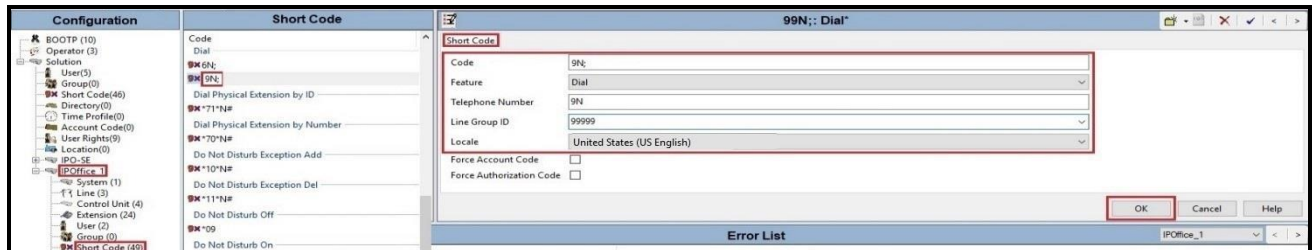


Figure 26 – Short Code 9N for Expansion System

The feature of incoming calls from mobility extension to idle-appearance FNE (Feature Name Extension) is hosted by Avaya IP Office Server Edition. The Short Code **FNE00** was configured with following parameters:

- For **Code** field, enter FNE feature code as **FNE00** for dial tone
- Set **Feature** to **FNE Service**
- Set **Telephone Number** to **00**
- Set **Line Group ID** to **0**
- Set the **Locale** to **United States (US English)**
- Default values may be used for other parameters
- Click **OK** to submit the changes

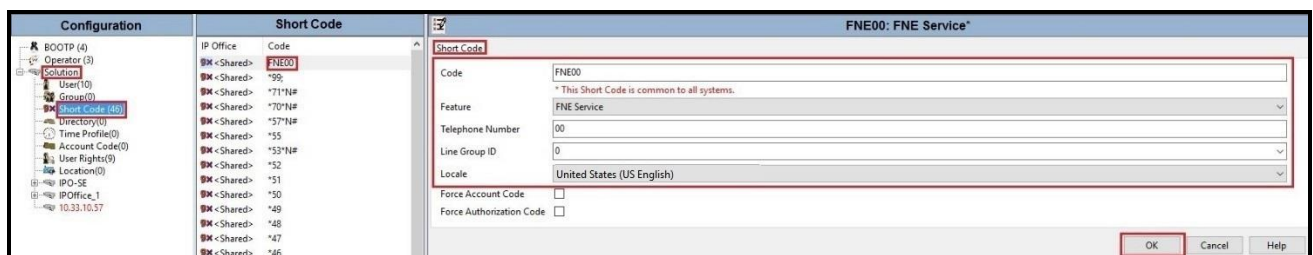


Figure 27 – Short Code FNE

5.8. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP Line defined in **Section 5.4.2**. 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 **613XXX0771**. Select the **User** tab in the Details pane.

Note: When **Auto** is selected for the **Local URI**, **Contact** and **Diversion Header** parameters (See **Section 5.4.2 - Call Detail** tab), the information in the Incoming Call Route (See **Section 5.9**) is used to populate the SIP From and Contact headers for outbound calls.

The screenshot displays the Avaya DevConnect configuration interface. On the left is a 'Configuration' tree with 'User' selected under 'IPO-SE'. The center pane shows a list of users, with '613XXX0771' highlighted. The right pane shows the 'User' configuration tab for '613XXX0771: 613XXX0771'. The configuration includes fields for Name, Password, Confirm Password, Unique Identity, Conference PIN, Confirm Audio, Conference PIN, Account Status (set to 'Enabled'), Full Name, Extension, Email Address, Locale (set to 'United States (US English)'), Priority (set to '\$'), System Phone Rights (set to 'None'), and Profile (set to 'Power User'). Below these are checkboxes for various services: Receptionist, Enable Softphone, Enable one-X Portal Services, Enable one-X TeleCommuter, Enable Remote Worker, Enable Desktop/Tablet VoIP client, Enable Mobile VoIP Client, Enable MS Teams Client, and Send Mobility Email.

Configuration	User	613XXX0771: 613XXX0771
BOOTP (13) Operator (3) Solution User (6) Group(0) Short Code(47) Directory(0) Time Profile(0) Account Code(0) User Rights(9) Location(0) IPO-SE System (1) Line (1) Control Unit (9) Extension (24) User (7) Group (0) Short Code (49) Service (0) Incoming Call Route (5) Directory (0) Time Profile (0) IP Route (3) Account Code (0) License (30) User Rights (9) Auto Attendant (0) ARS (1) Conference (0) Location (0) Authorization Code (0)	Name 613XXX0771 613XXX0900 613XXX1021 NoUser	User Voicemail DND Short Codes Source Numbers Telephony Forwarding Dial In Voice Recording Button Programming Menu Programming
		Name 613XXX0771 Password •••••••• Confirm Password •••••••• Unique Identity Conference PIN Confirm Audio Conference PIN Account Status Enabled Full Name 613XXX0771 Extension 613XXX0771 Email Address Locale United States (US English) Priority \$ System Phone Rights None Profile Power User ☐ Receptionist ☒ Enable Softphone ☒ Enable one-X Portal Services ☒ Enable one-X TeleCommuter ☒ Enable Remote Worker ☒ Enable Desktop/Tablet VoIP client ☒ Enable Mobile VoIP Client ☒ Enable MS Teams Client ☐ Send Mobility Email

Figure 28 – User Configuration – User Tab

To configure the restricted outbound call for a user by using specific W in the Short Code, 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 **613XXX0771**. Select the **Short Codes** tab in the Details pane.

- 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 **WN**. The value **N** represents the number dialed by the user. Note: Use the specific **W** in front of **N** for restricting outbound calls for a user
- Set the **Line Group ID** to the **Outgoing Group 17** defined on the **Call Details** tab on the **SIP Line** in **Section 5.4.2**. This short code will use this line group when placing the outbound call
- Set the **Locale** to **United States (US English)**
- Default values may be used for all other parameters
- Click **OK** to submit the changes

The screenshot shows the 'Short Codes' tab for user '613XXX0771'. The 'New Short Code' section is highlighted with a red box. It contains the following fields:

Code	Telephone Number	Feature	Line Group ID
9N;	WN	Dial	17

Below these fields, the 'Locale' is set to 'United States (US English)'. There are also checkboxes for 'Force Account Code' and 'Force Authorization Code', both of which are currently unchecked. The 'OK' button is highlighted with a red box.

Figure 29 – User Configuration – Short Code tab

One of the H.323 IP Deskphones at the enterprise site uses the Mobile Twinning feature. The following screen shows the **Mobility** tab for User **613XXX0771**. 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 **01613XXX5096**. Check **Mobile Call Control** to allow incoming calls from mobility extension to access FNE00 (Defined in **Section 5.7**). Other options can be set according to customer requirements.

The screenshot displays the Avaya User Configuration interface for User **613XXX0771: 613XXX0771**. The **Mobility** tab is selected, and the **Mobility Features** and **Mobile Twinning** sections are highlighted with red boxes.

Mobility Features:

- ☒ **Mobility Features**
- ☒ **Mobile Twinning**
- ☐ **Fallback Twinning**
- Twinned Mobile Number (including dial access code):** 91613XXX5096
- Twinned Time Profile:** <None>
- Mobile Dial Delay (sec):** 2
- Mobile Answer Guard (sec):** 0
- ☐ **Hunt group calls eligible for mobile twinning**
- ☐ **Forwarded calls eligible for mobile twinning**
- ☐ **Twin When Logged Out**
- ☐ **one-X Mobile Client**
- ☒ **Mobile Call Control**
- ☐ **Mobile Callback**

Other settings visible in the Mobility tab:

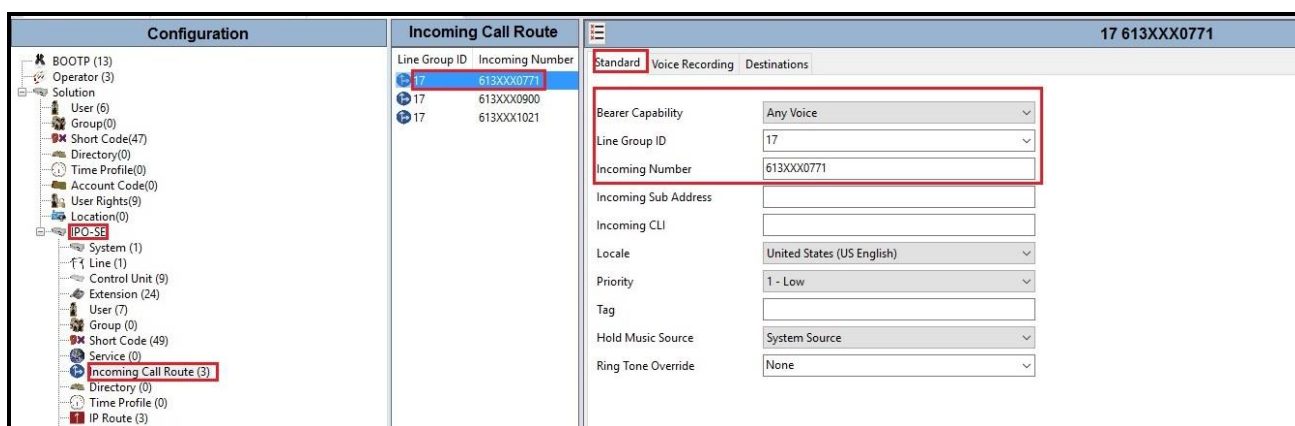
- Simultaneous:** Coverage Delay (secs) 0, MS Teams URI
- ☐ **Internal Twinning**
 - Twinned Handset:** <None>
 - Maximum Number of Calls:** 1
 - ☐ **Twin Bridge Appearances**
 - ☐ **Twin Coverage Appearances**
 - ☐ **Twin Line Appearances**

Figure 30 – Mobility Configuration for User

5.9. 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 service provider. To create an incoming call route, select **Incoming Call Route** in the left Navigation Pane, then right-click in the center Group Pane and select **New** (not shown). On the **Standard** tab of the Details Pane, enter the parameters as shown below:

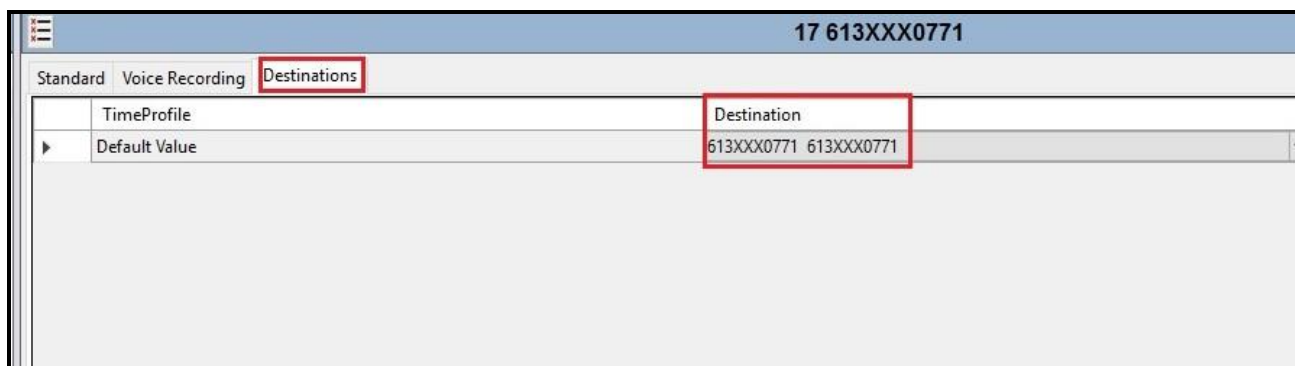
- Set the **Bearer Capability** to **Any Voice**
- Set the **Line Group ID** to the **Incoming Group 17** defined on the **Call Details** tab on the **SIP Line** in **Section 5.4.2**
- Set the **Incoming Number** to the incoming DID number on which this route should match
- Default values can be used for all other fields



Configuration	Incoming Call Route	17 613XXX0771
BOOTP (13) Operator (3) Solution User (6) Group (0) Short Code (47) Directory (0) Time Profile (0) Account Code (0) User Rights (9) Location (0) IPO-SE System (1) Line (1) Control Unit (9) Extension (24) User (7) Group (0) Short Code (49) Service (0) Incoming Call Route (3) Directory (0) Time Profile (0) IP Route (3)	Line Group ID Incoming Number 17 613XXX0771 17 613XXX0900 17 613XXX1021	Standard Voice Recording Destinations Bearer Capability Any Voice Line Group ID 17 Incoming Number 613XXX0771 Incoming Sub Address Incoming CLI Locale United States (US English) Priority 1 - Low Tag Hold Music Source System Source Ring Tone Override None

Figure 31 – Incoming Call Route Configuration

On the **Destination** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to **613XXX0771** on line 17 are routed to **Destination 613XXX0771 613XXX0771** as below screenshot:



17 613XXX0771		
Standard	Voice Recording	Destinations
TimeProfile	Destination	
Default Value	613XXX0771 613XXX0771	

Figure 32 – Incoming Call Route for Destination 613XXX0771

For Feature Name Extension Service testing purpose, the incoming calls to DID number **613XXX0900** were configured to access **FNE00**. The **Destination** was appropriately defined as **FNE00** as below screenshot:

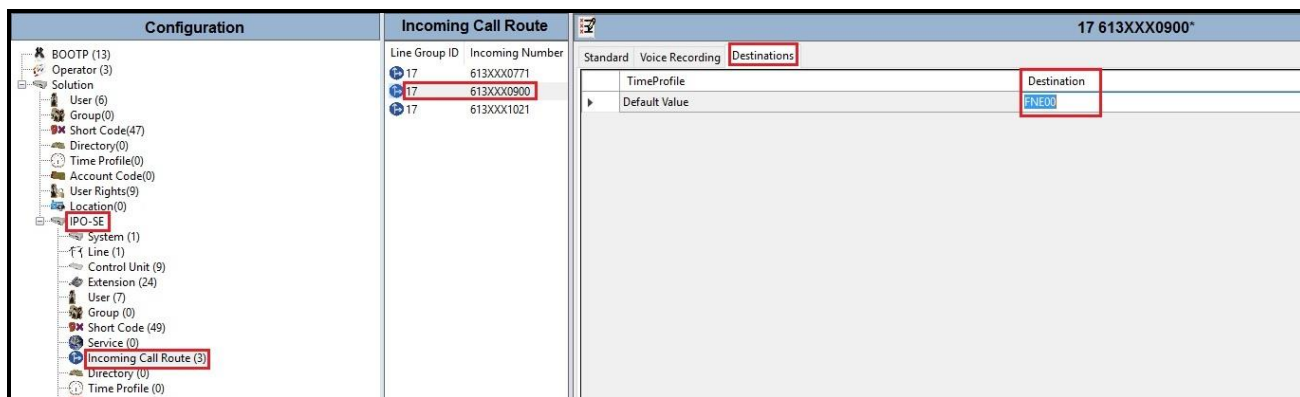


Figure 33 – Incoming Call Route for Destination FNE

For Voice Mail testing purpose, the incoming calls to DID number **613XXX1021** were configured to access **VoiceMail**. The **Destination** was appropriately defined as **VoiceMail** as below screenshot:

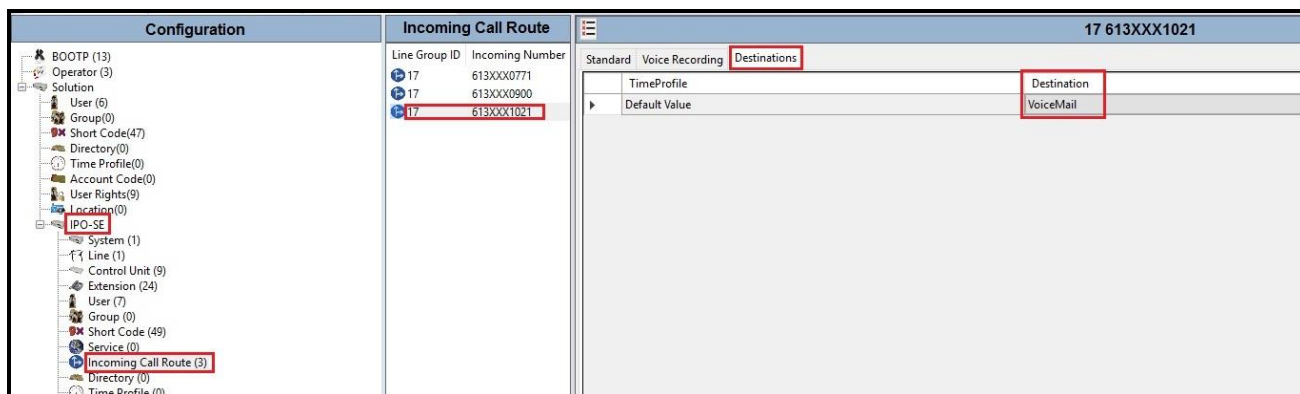


Figure 34 – Incoming Call Route for Destination VoiceMail

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 Trunk Configuration

Bell Canada is responsible for the configuration of Bell Canada SIP Trunking Service. The customer must provide the IP address used to reach the Avaya IP Office Server Edition LAN2 port at the enterprise. Bell Canada will provide the customer necessary information to configure the SIP connection between Avaya IP Office Server Edition and Bell Canada. The provided information from Bell Canada includes:

- SIP Proxy IP address and port number used for signaling and media
- DID numbers
- Bell Canada SIP Trunk Specification

7. Verification Steps

The following steps may be used to verify the configuration:

- Use the Avaya IP Office Monitor application to monitor the active SIP call traces between the enterprise and Bell Canada. Launch the application from **Start → All apps → IP Office → Monitor** on the PC where Avaya IP Office Server Edition Manager was installed. Click start/ stop buttons to capture the SIP call traces.

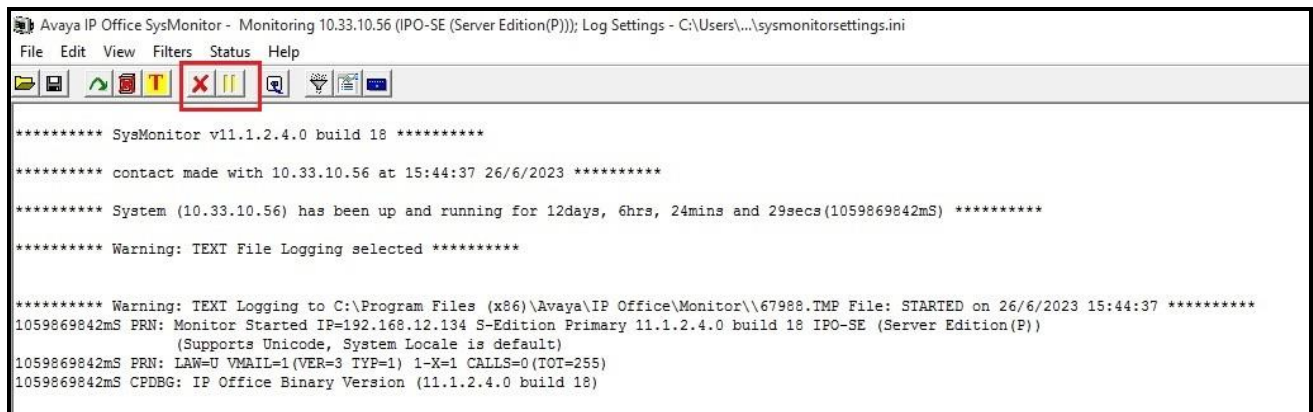


Figure 35 – SIP Trace Monitor

- Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application from **Start → All apps → IP Office → System Status** on the PC where Avaya IP Office Server Edition 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** for each channel (The below screen shot showed 2 active calls at present time)

Channel Number	URI Group	Call Ref	Current State	Time in State	Remote Media Address	Codec	Connection Type	Caller ID or Dialed Other Party on Call	Direction of Call	Round Trip Delay	Receive Jitter	Receive Packet Loss Fraction	Transmit Jitter	Transmit Packet Loss Fraction
1	1	197	Connected	00:00:18	192.168.237.208	G711MU	RTP Relay	Ext#130000771, 6130000771	Outgoing	0ms	0ms	0%		
2	1	198	Connected	00:00:06	192.168.237.208	G711MU	RTP Relay	Ext#130000900, 6130000900	Incoming	0ms	0ms	0%		
3			Idle	4 days 01:54:30										
4			Idle	4 days 22:41:11										
5			Idle	4 days 22:41:11										
6			Idle	4 days 22:41:11										
7			Idle	4 days 22:41:11										
8			Idle	4 days 22:41:11										
9			Idle	4 days 22:41:11										

Figure 36 – SIP Trunk status

- Use the Avaya IP Office System Status application to verify that no alarms are active on the SIP line. Launch the application from **Start → All apps → IP Office → System Status** on the PC where Avaya IP Office Manager was installed. Select **Alarm → Trunks** to verify that no alarms are active on the SIP line



The screenshot shows the Avaya IP Office System Status application. The left sidebar has a tree view with 'System' selected, and 'Alarms (0)' expanded. Under 'Alarms', 'Configuration (0)' is selected, and 'Service (0)' is expanded. Under 'Service', 'Trunks (0)' is selected, and 'Lines: 2 (0)' is expanded. The main area displays a table titled 'Select a line to display the alarm information'.

Line	Module / Slot / Type	Port Number / Address / Domain	Alarms
2	IP Office	172.16.199.60	0
17	SIP	sip://192.168.237.208	0

Figure 37 – SIP Trunk alarm

- Verify that a phone connected to the PSTN can successfully place a call to Avaya IP Office Server Edition with two-way audio
- Verify that a phone connected to Avaya IP Office Server Edition can successfully place a call to the PSTN with two-way audio
- Use a network sniffing tool e.g., Wireshark to monitor the SIP signaling between the enterprise and Bell Canada. The sniffer traces are captured at the LAN2 port interface of the Avaya IP Office Server Edition

8. Conclusion

Bell Canada passed compliance testing excepting the limitation in **Section 2.2**. These Application Notes describe the procedures required to configure the SIP connections between Avaya IP Office Server Edition and the Bell Canada system as shown in **Figure 1**.

9. Additional References

- [1] *Avaya IP Office Technical Bulletin 236 / General Availability (GA)- IP Office Release 11.1.2 Service Pack 4, Issue 2, 08th February 2023*
- [2] *Deploying IP Office Server Edition and Application Servers, Release 11.1 FP2, Issue 26, January 2023*
- [3] *Deploying Avaya IP Office Servers as Virtual Machines, Release 11.1.2.4, Issue 13, January 2023*
- [4] *IP Office Platform 11.1, Deploying an IP Office 500 V2/V2A in IP Office Basic Edition Mode, Issue 38e, Monday, February 28, 2022*
- [5] *Administering Avaya IP Office using Manager, Release 11.1.2.4, Issue 43, March 2023.*
Deploying Avaya Session Border Controller for Enterprise on a Virtualized Environment Platform, Release 10.1.x, Issue 1, December 2021

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

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