



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Primus SIP Trunking with Avaya IP Office Release 8.0 – Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between Primus and Avaya IP Office Release 8.0.

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

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted in the Avaya Solutions and Interoperability Test Lab, utilizing Primus SIP Trunk Services.

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between Primus and Avaya IP Office Release 8.0.

The Primus SIP Trunking service referenced within these Application Notes is positioned for customers that have an IP-PBX or IP-based network equipment with SIP functionality, but need a form of IP transport and local services to complete their solution.

Primus SIP Trunking will enable delivery of originating traffic and termination of local, long-distance and toll-free traffic across a single broadband connection. A SIP signaling interface will be enabled to the Customer Premises Equipment (CPE).

2. General Test Approach and Test Results

The general test approach was to connect a simulated enterprise site to the Primus SIP Trunking service via the public Internet and exercise the features and functionality listed in **Section 2.1**. The simulated enterprise site was comprised of Avaya IP Office and various Avaya endpoints.

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

2.1. Interoperability Compliance Testing

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

- Response to SIP REGISTER queries
- Incoming PSTN calls to various phone types including H.323, SIP, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types including H.323, SIP, digital, and analog telephones at the enterprise. All outgoing PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Inbound and outbound PSTN calls to/from Avaya IP Office Softphone.
- Various call types including: local, long distance, outbound toll-free and local directory assistance.
- Codec G.711MU, G.729A and G722.
- T38 Fax.
- Caller ID presentation and Caller ID restriction.

- DTMF transmission using RFC 2833.
- Voicemail navigation using DTMF for inbound and outbound calls.
- User features such as hold and resume, transfer, and conference.
- Off-net call forwarding and twinning.

2.2. Test Results

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

- **OPTIONS** – Primus does not support OPTIONS messages and therefore this was not turned on in IP Office.
- **T.38 Fax** – Primus supports T38 fax; however when IP Office was configured to use the same, both inbound and outbound faxes used G711 instead of T38. This is a known issue in IP Office 8.0.18. Upgraded the system to 8.0.43 and the issue is still seen. Avaya IP Office design is aware of the issue and investigation is in progress at the time of writing this document.
- **Codec** – Primus only supports G729, G711MU and G722.
- **Inbound Toll Free** – Primus does not offer this service and therefore requested not to test this feature.
- **Emergency Calls** – Primus had not setup address/location and therefore while testing emergency calls, 911 agents were able to see the calling number only.

2.3. Support

For technical support on Primus SIP Trunking, contact Primus using the Customer Service links at <http://businesssupport.primus.ca/>.

3. Reference Configuration

Figure 1 illustrates the sample configuration used for the DevConnect compliance testing. The sample configuration shows an enterprise site connected to Primus SIP Trunking.

Located at the enterprise site is an Avaya IP Office 500. The LAN1 port of Avaya IP Office is connected to Lab Network which is connected to the Public Network. Endpoints include an Avaya 1608 IP Telephone (with H.323 firmware), an Avaya 1140E IP Telephone (with SIP firmware), an Avaya 9650 IP Telephone (with H.323 firmware), an Avaya 9508 Digital Telephone, a Fax machine and a traditional Analog Telephone. The site also has a Windows XP Professional 2002 SP3 Server running Avaya Voicemail Pro for voicemail, Avaya IP Office Manager to configure the Avaya IP Office, Avaya Phone Manager and Avaya IP Office Softphone,

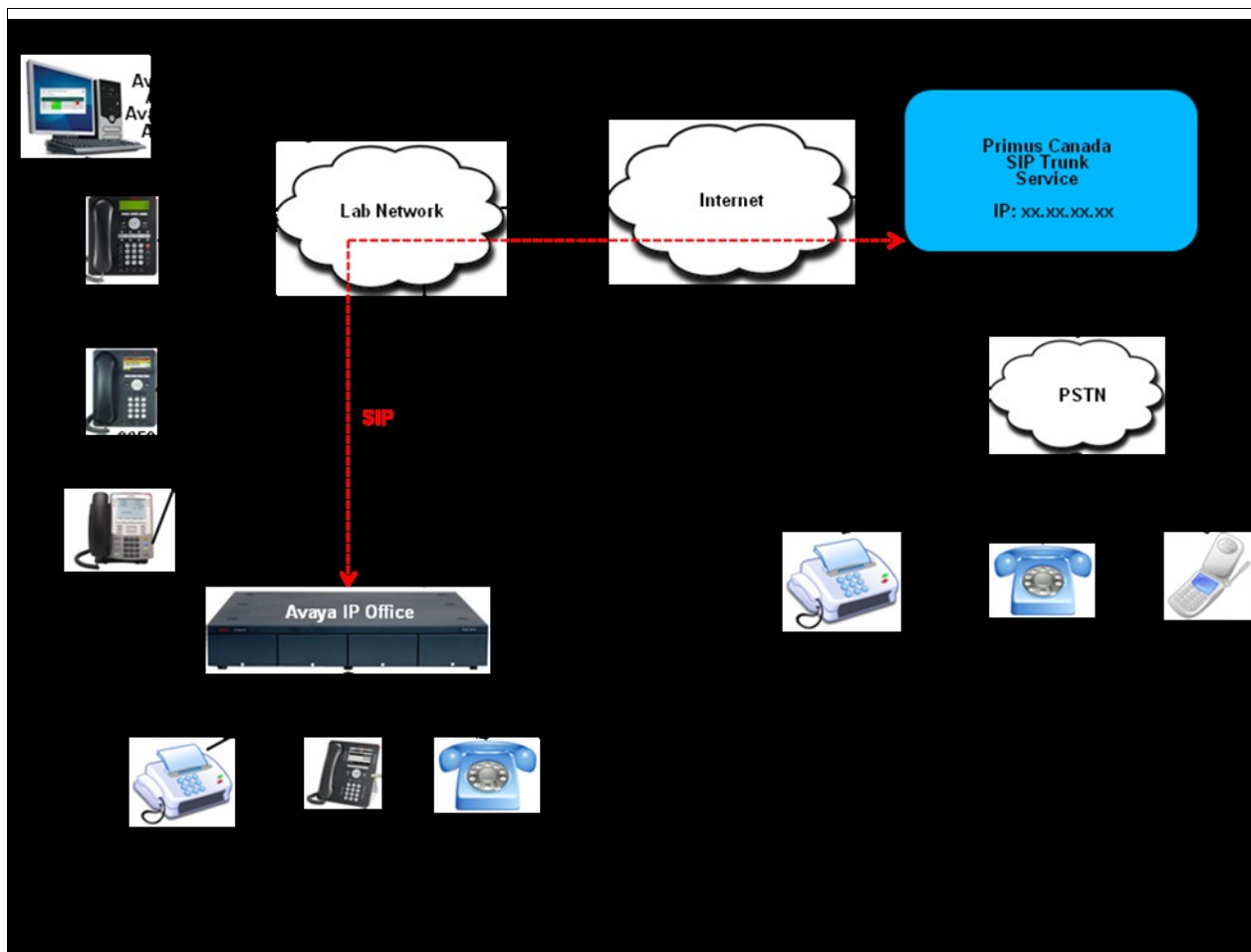


Figure 1: Avaya Interoperability Test Lab Configuration

For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead, public IP addresses have been replaced with private addresses and all phone numbers have been replaced with numbers that cannot be routed.

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

4. Equipment and Software Validated

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

Equipment	Software
Avaya IP Office 500v2	8.0 (18)
Avaya IP Office Manager	10.0.18
Avaya IP Office Voicemail Pro	8.0 (80029)
Avaya 9650 IP Telephone (H.323)	3.186a
Avaya 1608 IP Telephones (H.323)	1.300B
Avaya 1140 SIP Telephones	04.03.09.00
Avaya 9508 Digital Telephone	N/A
Avaya Analog Telephone	N/A
Avaya IP Office Softphone	3.2.3.15_64595
Avaya IP Office Phone Manager Lite	4.2.39
Service Provider	Software
SBC : Genband S3	6.0.3.15
Broadsoft Broadworks	R17 SP4

5. Configure Avaya IP Office

Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the Avaya IP Office Manager PC, select **Start** → **Programs** → **IP Office** → **Manager** to launch the application. A screen that includes the following in the center may be displayed:

WELCOME to IP Office Administration

What would you like to do ?

[Create an Offline Configuration](#)

[Open Configuration from System](#)

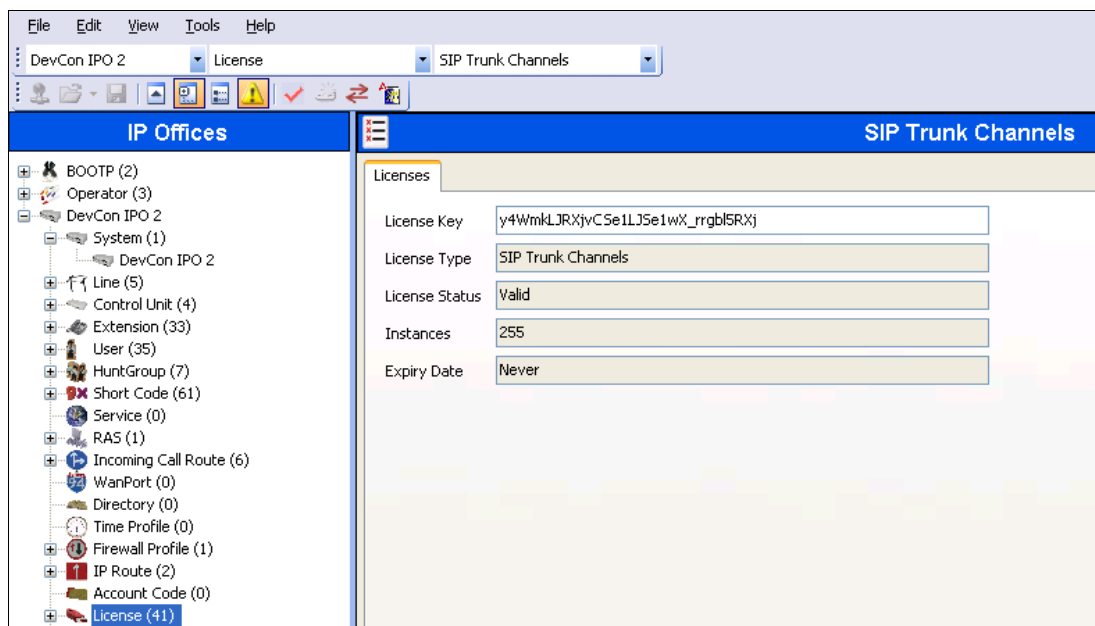
[Read a Configuration from File](#)

Navigate to **File** → **Open Configuration** (not shown), select the proper Avaya IP Office system from the pop-up window and log in with the appropriate credentials. 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 and the Details pane on the right side. These panes will be referenced throughout the Avaya IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the service provider (such as twinning and IP Office Softphone support) is assumed to already be in place.

5.1. Licensing and Physical Hardware

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

To verify that there is a SIP Trunk Channels License with sufficient capacity; click **License** in the Navigation pane and **SIP Trunk Channels**. Confirm a valid license with sufficient **Instances** (trunk channels) in the Details pane.



If Avaya IP Telephones will be used as is the case in these Application Notes, verify the Avaya IP endpoints license. Click **License** in the Navigation pane and **Avaya IP endpoints**. Confirm a valid license with sufficient **Instances** in the Details pane.

IP Offices		Avaya IP endpoints											
<ul style="list-style-type: none"> BOOTP (2) Operator (3) DevCon IPO 2 <ul style="list-style-type: none"> System (1) DevCon IPO 2 Line (5) Control Unit (4) Extension (33) User (35) HuntGroup (7) Short Code (61) Service (0) RAS (1) Incoming Call Route (6) WanPort (0) Directory (0) Time Profile (0) Firewall Profile (1) IP Route (2) Account Code (0) License (41) 	<div>Licenses</div> <table> <tr> <td>License Key</td> <td>UyBtPa27XvPa27W_P4cmBCkrjk_eBtGX</td> </tr> <tr> <td>License Type</td> <td>Avaya IP endpoints</td> </tr> <tr> <td>License Status</td> <td>Valid</td> </tr> <tr> <td>Instances</td> <td>255</td> </tr> <tr> <td>Expiry Date</td> <td>Never</td> </tr> </table>			License Key	UyBtPa27XvPa27W_P4cmBCkrjk_eBtGX	License Type	Avaya IP endpoints	License Status	Valid	Instances	255	Expiry Date	Never
License Key	UyBtPa27XvPa27W_P4cmBCkrjk_eBtGX												
License Type	Avaya IP endpoints												
License Status	Valid												
Instances	255												
Expiry Date	Never												

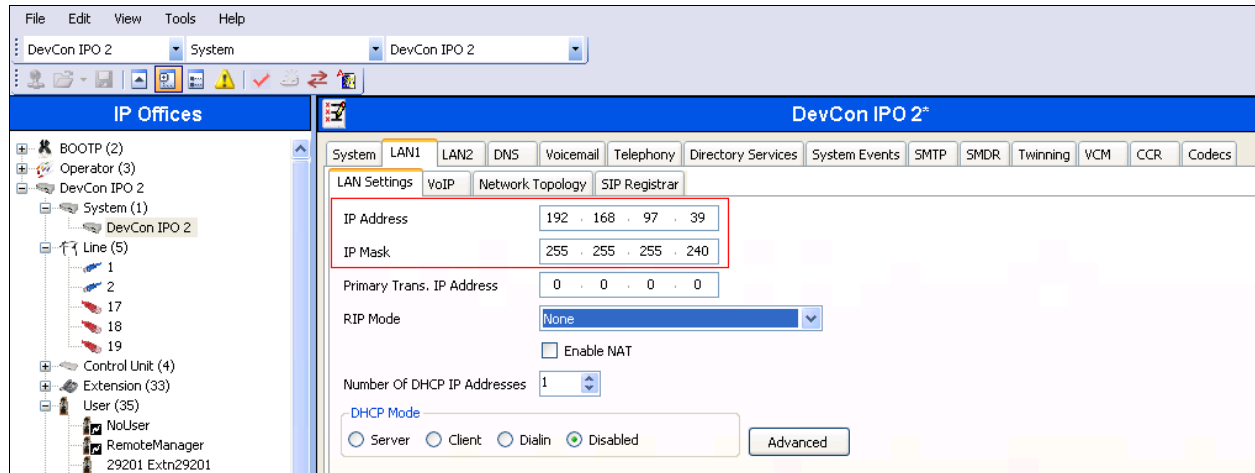
The VCM is a Voice Compression Module supporting VoIP codecs. An IP Office hardware configuration with a VCM component is necessary to support SIP trunking.

The following screen shows the modules in the IP Office used in the sample configuration. To access such a screen, select **Control Unit** in the Navigation pane to show the modules. In the screen below, **IP 500 V2** is selected, revealing additional information about the IP 500 V2 in the Details pane.

IP Offices		IP 500 V2															
<ul style="list-style-type: none"> BOOTP (2) Operator (3) DevCon IPO 2 <ul style="list-style-type: none"> System (1) DevCon IPO 2 Line (5) Control Unit (4) <ul style="list-style-type: none"> 1 IP 500 V2 2 VCM64/PRID U 3 PHONE8 6 DIG DCPx16 V2 Extension (33) User (35) HuntGroup (7) Short Code (61) 	<div>Unit</div> <table> <tr> <td>Device Number</td> <td>1</td> </tr> <tr> <td>Unit Type</td> <td>IP 500 V2</td> </tr> <tr> <td>Version</td> <td>8.0 (18)</td> </tr> <tr> <td>Serial Number</td> <td>00ef0bd00ebd</td> </tr> <tr> <td>Unit IP Address</td> <td>192.168.97.39</td> </tr> <tr> <td>Interconnect Number</td> <td>0</td> </tr> <tr> <td>Module Number</td> <td>Control Unit</td> </tr> </table>			Device Number	1	Unit Type	IP 500 V2	Version	8.0 (18)	Serial Number	00ef0bd00ebd	Unit IP Address	192.168.97.39	Interconnect Number	0	Module Number	Control Unit
Device Number	1																
Unit Type	IP 500 V2																
Version	8.0 (18)																
Serial Number	00ef0bd00ebd																
Unit IP Address	192.168.97.39																
Interconnect Number	0																
Module Number	Control Unit																

5.2. LAN1 Settings

In the sample configuration the LAN port was used to connect the Avaya IP Office to the public network. The LAN1 settings correspond to the LAN port on the Avaya IP Office 500. To access the LAN1 settings, first navigate to **System** in the Navigation Pane and then navigate to the **LAN1 → LAN Settings** tab in the Details Pane. Set the **IP Address** and **IP Mask** field to the IP address and mask assigned to the Avaya IP Office LAN port. All other parameters should be set according to customer requirements.



On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. 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 LAN1. During compliance testing the number range were left at default values. 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 and are also the default values. All other parameters should be set according to customer requirements.

The screenshot displays the Avaya IP Office configuration interface for 'DevCon IPO 2'. The left pane shows a tree view of the system hierarchy, including 'IP Offices', 'System (1)', 'Line (5)', 'Control Unit (4)', 'Extension (33)', and 'User (35)'. The right pane shows the 'VoIP' configuration tab. The 'SIP Trunks Enable' checkbox is checked. The 'RTP Port Number Range' is set to 49152 (Minimum) and 53246 (Maximum). The 'DiffServ Settings' section shows DSCP values of 88 and 46, and SIG DSCP values of FC and 63. The 'DHCP Settings' section shows various option numbers and VLAN settings. The 'RTP keepalives' section shows the scope as Disabled.

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**.
- Set **Binding Refresh Time (seconds)** to **0**. This value is used to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider.
- Set **Public IP Address** to the IP address of the Avaya IP Office LAN port.
- Set the **Public Port** to **5060**.
- All other parameters should be set according to customer requirements.

The screenshot shows the 'DevCon IPO 2*' configuration window with the 'Network Topology' tab selected. The 'Network Topology Discovery' section contains the following settings:

- STUN Server IP Address: 192 . 168 . 10 . 10
- STUN Port: 3478
- Firewall/NAT Type: Open Internet (selected from a dropdown menu)
- Binding Refresh Time (seconds): 0
- Public IP Address: 192 . 168 . 97 . 39
- Public Port: 5060
- Run STUN: Run STUN (button), Cancel (button)
- Run STUN on startup: ☐

5.3. Voicemail Settings

On the **Voicemail** tab in the Details Pane, select **Voicemail Lite/Pro** from the drop down for the **Voicemail Type** field. Configure the IP address of the server where the Voicemail is installed in the **Voicemail IP Address** field. Retain default values for the rest of the fields.

The screenshot shows the 'DevCon IPO 2*' configuration window with the 'Voicemail' tab selected. The settings are as follows:

- Voicemail Type: Voicemail Lite/Pro (selected from a dropdown menu)
- Messages Button Goes To Visual Voice: ☐
- Voicemail Destination: (empty dropdown menu)
- Voicemail IP Address: 192 . 168 . 98 . 74
- Backup Voicemail IP Address: 0 . 0 . 0 . 0
- Voicemail Channel Reservation:
 - Unreserved Channels: 259
 - Auto-Attendant: 0
 - Voice Recording: 0
 - Mandatory Voice Recording: 0
 - Announcements: 0
 - Mailbox Access: 0
- DTMF Breakout:
 - Reception / Breakout (DTMF *0/0): (empty text field)
 - Breakout (DTMF 2): (empty text field)
 - Breakout (DTMF 3): (empty text field)

5.4. System Telephony Settings

On the **Telephony** tab in the Details Pane, 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. If for security reasons incoming calls should not be allowed to transfer back to the PSTN then leave this setting checked.

The screenshot shows the 'DevCon IPO 2' configuration window with the 'Telephony' tab selected. The left pane shows a tree view of the system hierarchy. The main area contains various settings for analogue extensions and companding law. The 'Inhibit Off-Switch Forward/Transfer' checkbox is highlighted with a red box.

Setting	Value
Default Outside Call Sequence	Normal
Default Inside Call Sequence	Ring Type 1
Default Ring Back Sequence	Ring Type 2
Restrict Analogue Extension Ringer Voltage	<input type="checkbox"/>
Dial Delay Time (secs)	4
Dial Delay Count	0
Default No Answer Time (secs)	15
Hold Timeout (secs)	620
Park Timeout (secs)	300
Ring Delay (secs)	5
Call Priority Promotion Time (secs)	Disabled
Default Currency	USD
Default Name Priority	Favor Trunk
U-Law	<input checked="" type="radio"/>
A-Law	<input type="radio"/>
U-Law Line	<input checked="" type="radio"/>
A-Law Line	<input type="radio"/>
DSS Status	<input type="checkbox"/>
Auto Hold	<input checked="" type="checkbox"/>
Dial By Name	<input checked="" type="checkbox"/>
Show Account Code	<input checked="" type="checkbox"/>
Inhibit Off-Switch Forward/Transfer	<input type="checkbox"/>
Restrict Network Interconnect	<input type="checkbox"/>
Drop External Only Impromptu Conference	<input type="checkbox"/>
Visually Differentiate External Call	<input type="checkbox"/>
Unsupervised Analog Trunk Disconnect Handling	<input type="checkbox"/>
High Quality Conferencing	<input checked="" type="checkbox"/>

5.5. Twinning Calling Party Settings

To view or change Twinning settings, select the **Twining** tab as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning** is left blank.

The screenshot shows the 'DevCon IPO 2' configuration window with the 'Twining' tab selected. The left pane shows the same tree view. The main area contains settings for mobile twinning.

Setting	Value
Send original calling party information for Mobile Twinning	<input type="checkbox"/>
Calling party information for Mobile Twinning	

5.6. IP Route

Navigate to **IP Route** in the left Navigation Pane, and then right-click to select **New** (not shown). Create a default route with the following parameters:

- Set **IP Address** and **IP Mask** to **0.0.0.0**.
- Set **Gateway IP Address** to the IP Address of the default router to reach LAN1.
- Set **Destination** to **LAN1** from the pull-down menu.

The screenshot shows the 'IP Route' configuration window. The left navigation pane lists various system components, with 'IP Route (2)' selected. The main configuration area on the right is titled '0.0.0.0' and contains the following fields:

- IP Address:** 0 . 0 . 0 . 0
- IP Mask:** 0 . 0 . 0 . 0
- Gateway IP Address:** 192 . 168 . 97 . 33
- Destination:** LAN1
- Metric:** 1
- Proxy ARP:** ☐

5.7. Codecs

Since Primus supports G711MU, G729 and G722, only these codecs were selected from the codec selection as shown in the screen below.

The screenshot shows the 'DevCon IPO 2' configuration window, specifically the 'Codecs' tab. The left navigation pane shows 'DevCon IPO 2' selected. The main configuration area is divided into three sections:

- Available Codecs:** A list of codecs with checkboxes, including G.711 ULAW 64K, G.711 ALAW 64K, G.722 64K, G.729(a) 8K CS-ACELP, and G.723.1 6K3 MP-MLQ.
- Default Codec Selection:** A list of codecs, including G.711 ALAW 64K and G.723.1 6K3 MP-MLQ.
- Selected:** A list of selected codecs, including G.711 ULAW 64K, G.729(a) 8K CS-ACELP, and G.722 64K.

5.8. SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and Primus SIP Trunking. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New** → **SIP Line** (not shown).

5.8.1. SIP Line – SIP Line Tab

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

- Set **ITSP Domain Name** to the domain name of Primus SIP as provided by the partner.
- Set **Send Caller ID** to **P Asserted ID**. This field allows selection of the value the SIP Line should use for the original calling party ID when routing twinned calls.
- Check **REFER Support**.
- Check the **In Service** box. This makes the trunk available to incoming and outgoing calls.
- Uncheck the **Check OOS** box. The Options feature is not supported see **Section 2.2** for details. Default values may be used for all other parameters.

The screenshot displays the Avaya IP Office configuration interface. On the left is a navigation pane titled "IP Offices" containing a tree view of system components: BOOTP (2), Operator (3), DevCon IPO 2, System (1), DevCon IPO 2, Line (5) with sub-items 1, 2, 17, 18, and 19, Control Unit (4), Extension (33), User (35), NoUser, RemoteManager, and a list of extensions from 29201 to 29208. The main pane is titled "SIP Line - Line 19" and contains several tabs: SIP Line, Transport, SIP URI, VoIP, T38 Fax, and SIP Credentials. The "SIP Line" tab is selected, showing the following configuration fields: Line Number (19), ITSP Domain Name (preprod.bvoice.primus.ca), In Service (checked), Use Tel URI (unchecked), Check OOS (unchecked), Prefix, National Prefix (0), Country Code, International Prefix (011), Send Caller ID (P Asserted ID), Association Method (By Source IP address), and a REFER Support section with Incoming and Outgoing dropdowns set to Auto. Red rectangular boxes highlight the ITSP Domain Name, In Service, Check OOS, Send Caller ID, and the entire REFER Support section.

5.8.2. SIP Line – SIP Credentials

Primus requires trunk registration using credentials. In the screen below, from the **SIP Credentials** tab add the **User name**, **Authentication Name** and **Password** provided by the partner. Check the **Registration required** box. Retain default values for the remaining fields.

The screenshot shows the 'SIP Line - Line 19' configuration window. The 'SIP Credentials' tab is selected. A table lists SIP credentials with columns: Index, UserName, Authentication Name, Contact, Expiry (mins), and Register. Below the table, the 'Edit SIP Credentials' dialog is open, showing fields for User name, Authentication Name, Contact, Password, Expiry (mins), and a checked 'Registration required' checkbox. The 'OK' and 'Cancel' buttons are visible.

5.8.3. SIP Line - Transport Tab

Select the **Transport** tab. Set the parameters as shown below.

- Set **ITSP Proxy Address** to the IP address of the Primus SIP proxy.
- Set **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to the network port configured in **Section 5.2**.
- Set the **Send Port** to **5060**.
- Default values may be used for all other parameters.

The screenshot shows the 'SIP Line - Line 19*' configuration window. The 'Transport' tab is selected. The 'ITSP Proxy Address' field is highlighted with a red box. Below it, the 'Network Configuration' section is highlighted with a red box, showing 'Layer 4 Protocol' set to 'UDP', 'Send Port' set to '5060', 'Use Network Topology Info' set to 'LAN 1', and 'Listen Port' set to '5060'. Other fields like 'Explicit DNS Server(s)', 'Calls Route via Registrar', and 'Separate Registrar' are also visible.

5.8.4. SIP Line - SIP URI Tab

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, and then click the **Add** button and the **New Channel** area will appear at the bottom of the pane. To add an entry, click on the **ADD** button. In the example screen below, a previously configured entry is shown. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact** and **Display Name** to *Use Internal Data*. This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any User as shown in **Section 5.1010**.
- From the **Registration** drop down menu, select the value that was configured in **Section 5.8.2**.
- Associate this line with an incoming line group by entering a line group number 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. The outgoing line group number is used in defining short codes for routing outbound traffic to this line.
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

The screenshot displays the Avaya IP Office configuration interface. On the left, a tree view shows the hierarchy: IP Offices, BOOTP (2), Operator (3), DevCon IPO 2, System (1), DevCon IPO 2, Line (5), Control Unit (4), Extension (33), and User (35). The 'Line (5)' folder is expanded, showing lines 1, 2, 17, 18, and 19. Line 19 is selected. On the right, the 'SIP Line - Line 19' configuration pane is shown. The 'SIP URI' tab is active. Below the tab, a table lists the configured SIP URI entries. The table has columns: Channel, Groups, Via, Local URI, Contact, Display Name, PAI, Credential, and Max Calls. The table is currently empty. Below the table, the 'Edit Channel' section is visible, showing the following configuration: Via: 192.168.97.39, Local URI: Use Internal Data, Contact: Use Internal Data, Display Name: Use Internal Data, PAI: None, Registration: 1: 6471234567, Incoming Group: 19, Outgoing Group: 19, and Max Calls per Channel: 10.

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
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Edit Channel

Via: 192.168.97.39

Local URI: Use Internal Data

Contact: Use Internal Data

Display Name: Use Internal Data

PAI: None

Registration: 1: 6471234567

Incoming Group: 19

Outgoing Group: 19

Max Calls per Channel: 10

5.8.5. SIP Line - VoIP Tab

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

- Set the **Codec Selection** field to **System Default** to allow the service provider supported codec which were already configured in **Section 5.7**.
- Uncheck the **VoIP Silence Suppression** box.
- Check the **Re-invite Supported** box.
- Set the **Fax Transport Support** to **T38**. See **Section 2.2** for additional fax considerations.
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Default values may be used for all other parameters.

Click the **OK** button at the bottom of the page (not shown).

The screenshot displays the Avaya IP Office configuration interface. On the left, a tree view shows the hierarchy: IP Offices, BOOTP (2), Operator (3), DevCon IPO 2, System (1), DevCon IPO 2, Line (5) with sub-items 1, 2, 17, 18, 19, Control Unit (4), Extension (33), and User (35) with sub-items NoUser, RemoteManager, and several extension numbers. The main panel is titled 'SIP Line - Line 19' and has tabs for SIP Line, Transport, SIP URI, VoIP, T38 Fax, and SIP Credentials. The 'VoIP' tab is active. It contains several configuration fields and checkboxes. The 'Codec Selection' dropdown is set to 'System Default'. Below it, two lists of codecs are shown: 'Unused' (G.711 ALAW 64K, G.723.1 6K3 MP-MLQ) and 'Selected' (G.711 ULAW 64K, G.729(a) 8K CS-ACELP, G.722 64K). The 'Fax Transport Support' dropdown is set to 'T38'. The 'Call Initiation Timeout (s)' is set to '4'. The 'DTMF Support' dropdown is set to 'RFC2833'. On the right, a group of checkboxes includes 'VoIP Silence Suppression' (unchecked), 'Re-invite Supported' (checked), 'Use Offerer's Preferred Codec' (unchecked), 'Codec Lockdown' (unchecked), and 'PRACK/100rel Supported' (unchecked).

5.8.6. SIP Line – T38 Fax Tab

In the **T38 Fax** tab the **Use Default Values** box was checked for this compliance testing. User can uncheck this box and change the various values of the fields in this tab as required.

The screenshot displays the configuration interface for a SIP Line, specifically Line 19. The left sidebar shows a hierarchical tree of network elements, including IP Offices, BOOTP, Operator, DevCon IPO, System, DevCon IPO, Line (5), Control Unit, Extension, and User. The main panel is titled 'SIP Line - Line 19' and contains several tabs: SIP Line, Transport, SIP URI, VoIP, T38 Fax, and SIP Credentials. The 'T38 Fax' tab is currently selected. It contains various configuration fields and checkboxes. The 'Use Default Values' checkbox is checked and highlighted with a red box. Other fields include T38 Fax Version (3), Transport (UDPTL), Redundancy (Low Speed and High Speed, both set to 0), TCF Method (Trans TCF), Max Bit Rate (bps) (14400), EFlag Start Timer (msecs) (2600), EFlag Stop Timer (msecs) (2300), and Tx Network Timeout (secs) (150). On the right side, there are checkboxes for Scan Line Fix-up, TFOP Enhancement, Disable T30 ECM, Disable EFlags For First DIS, and Disable T30 MR Compression. Below these is an NSF Override section with Country Code and Vendor Code fields, both set to 0.

Field	Value
T38 Fax Version	3
Transport	UDPTL
Redundancy - Low Speed	0
Redundancy - High Speed	0
TCF Method	Trans TCF
Max Bit Rate (bps)	14400
EFlag Start Timer (msecs)	2600
EFlag Stop Timer (msecs)	2300
Tx Network Timeout (secs)	150
Use Default Values	<input checked="" type="checkbox"/>
Scan Line Fix-up	<input checked="" type="checkbox"/>
TFOP Enhancement	<input checked="" type="checkbox"/>
Disable T30 ECM	<input type="checkbox"/>
Disable EFlags For First DIS	<input type="checkbox"/>
Disable T30 MR Compression	<input type="checkbox"/>
NSF Override - Country Code	0
NSF Override - Vendor Code	0

5.9. Short Codes

Define a short code to route outbound traffic to the SIP line. To create a short code, right-click on **Short Code** in the Navigation Pane and select **New** (not shown). On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

- 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"@preprod.bvoice.primus.ca"**. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. The value **N** represents the number dialed by the user. The address **preprod.bvoice.primus.ca** represents the address of the Primus SIP proxy.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.8.44**. This short code will use this line group when placing the outbound call.

Click the **OK** button (not shown).

The screenshot displays the Avaya SIP Office configuration interface. On the left is the 'IP Offices' navigation pane, which includes a tree view with nodes such as BOOTP (2), Operator (3), DevCon IPO 2, System (1), DevCon IPO 2, Line (5), Control Unit (4), Extension (33), User (35), HuntGroup (7), Short Code (61), and Service (0). The 'Short Code (61)' node is selected. On the right is the 'Short Code' configuration pane, titled '9N;; Dial'. It contains the following fields and values:

Field	Value
Code	9N;
Feature	Dial
Telephone Number	N"@preprod.bvoice.primus.ca"
Line Group ID	19
Locale	
Force Account Code	<input type="checkbox"/>

The following screen illustrates a short code that acts like a feature access code rather than a means to access a SIP Line. In this case, the **Code FNE00** is defined for **Feature FNE Service** to **Telephone Number 00** (Mobile Call Control). This short code will be used as means to allow a Primus DID to be programmed to route directly to this feature, via inclusion of this short code as the destination of an Incoming Call Route. See **Section 5.111**. This feature is used to provide dial tone to twinned mobile devices (e.g., cell phone) directly from IP Office; once dial tone is received the user can perform dialing actions including making calls and activating Short Codes.

The screenshot shows the 'IP Offices' configuration window. On the left is a tree view with 'Short Code (61)' selected. The main pane is titled 'FNE00: FNE Service' and contains the following fields:

- Short Code** (tab):
- Code**: FNE00
- Feature**: FNE Service (dropdown)
- Telephone Number**: 00
- Line Group ID**: 19 (dropdown)
- Locale**: (dropdown)
- Force Account Code**: ☐

5.10. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.88**. To configure these settings, first navigate to **User** in the Navigation Pane, and then click on the user to be modified. 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 and Contact headers for outgoing SIP trunk calls and allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line. See **Section 5.8.44**. The example below shows the settings for User 29225. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise from Primus. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network. Click the **OK** button (not shown).

The screenshot shows the 'IP Offices' configuration window with 'User (35)' selected. The main pane is titled 'Extn29225: 29225*' and has tabs for 'Telephony', 'Forwarding', 'Dial In', 'Voice Recording', 'Button Programming', 'Menu Programming', 'Mobility', 'Phone Manager Options', 'Hunt Group Membership', 'Announcements', and 'SIP'. The 'SIP' tab is active, showing the following fields:

- SIP Name**: 6471231111
- SIP Display Name (Alias)**: Extn29225
- Contact**: 6471231111
- Anonymous**: ☐

The following screen shows the **Mobility** tab for User 29226. The **Mobility Features**, **Mobile Twinning** and **Mobile Call Control** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone over the SIP trunk, in this case **6139675281**. Other options can be set according to customer requirements. This is the configuration used during compliance testing for the FNE00 feature.

The screen below has the same configuration as the screen above except the **Mobile Call Control** box is not checked and the **Twinned Mobile Number** field is **916139675281**. This is the configuration used during compliance testing for the twinning feature only.

5.11. 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, right-click **Incoming Call Routes** in the Navigation Pane and select **New**. 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 line group of the SIP line defined in **Section 5.8.44**.
- Set the **Incoming Number** to the incoming number on which this route should match. Matching is right to left.
- Default values can be used for all other fields.

The screenshot shows the 'Incoming Call Route' configuration window with the 'Standard' tab selected. The window title is '19 6471231111*'. The left navigation pane shows a tree structure under 'IP Offices' with 'Incoming Call Route (6)' selected. The main area contains the following fields:

Field	Value
Bearer Capability	Any Voice
Line Group ID	19
Incoming Number	6471231111
Incoming Sub Address	
Incoming CLI	
Locale	
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. Click the **OK** button (not shown). In this example, incoming calls to 6471231111 on line 19 are routed to extension 29225.

The screenshot shows the 'Incoming Call Route' configuration window with the 'Destinations' tab selected. The window title is '19 6471231111*'. The left navigation pane shows a tree structure under 'IP Offices' with 'Incoming Call Route (6)' selected. The main area contains the following table:

TimeProfile	Destination
Default Value	29225 Extn29225

Incoming Call Routes for other direct mappings of DID numbers to IP Office users listed in **Figure 1** are omitted here, but can be configured in the same fashion.

In the screen shown below, the incoming call route for **Incoming Number 6471231112** is illustrated. The **Line Group Id** is **19**, matching the Incoming Group field configured in the SIP URI tab in **Section 5.8.4**.

The screenshot displays the 'IP Offices' configuration window. On the left is a tree view of the system hierarchy, including BOOTP, Operator, DevCon IPO, System, Line, Control Unit, Extension, User, HuntGroup, Short Code, Service, RAS, and Incoming Call Route. The 'Incoming Call Route (7)' is selected. The main panel shows the configuration for the incoming call route for the number '19 6471231112'. The 'Standard' tab is active, showing fields for Bearer Capability (Any Voice), Line Group ID (19), Incoming Number (6471231112), Incoming Sub Address, Incoming CLI, Locale, Priority (1 - Low), Tag, and Hold Music Source (System Source).

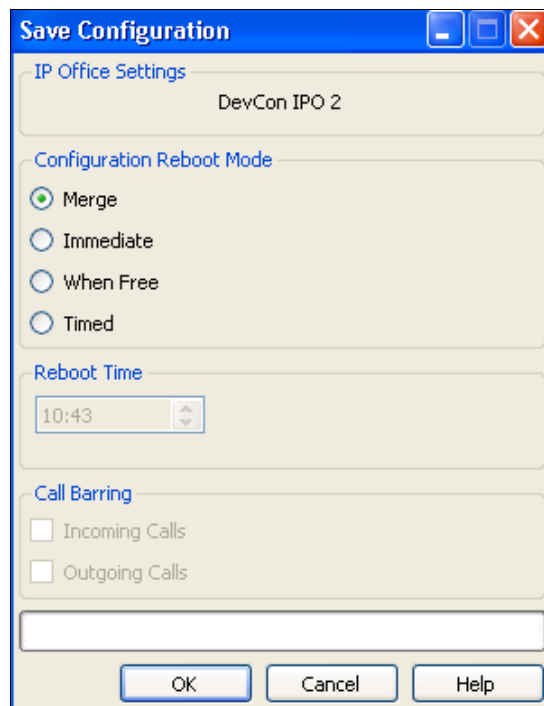
When configuring an Incoming Call Route, the **Destination** field can be manually configured with a number such as a short code, or certain keywords available from the pull-down menu. For example, the following **Destinations** tab for an incoming call route contains the **Destination FNE00** entered manually. **FNE00** is the short code for **FNE Service**, as shown in **Section 5.99**. An incoming call to 647-123-1112 will be delivered directly to internal dial tone from the IP Office, allowing the caller to perform dialing actions including making calls and activating Short Codes. The incoming caller ID must match the Twinned Mobile Number entered in the User Mobility tab (**Section 5.1010**); otherwise the IP Office responds with a 486 Busy Here and the caller will hear a busy tone.

The screenshot displays the 'IP Offices' configuration window, showing the 'Destinations' tab for the incoming call route for the number '19 6471231112'. The 'Destinations' tab is active, showing a table with columns for TimeProfile, Destination, and Fallback Extension. The table contains one row: 'Default Value' for TimeProfile, 'FNE00' for Destination, and a dropdown menu for Fallback Extension.

5.12. 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.

The following will appear, with either **Merge** or **Immediate** selected, based on the nature of the configuration changes made since the last save. Note that clicking **OK** may cause a service disruption. Click **OK** if desired.



6. Primus SIP Trunking Configuration

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

- IP address of the Primus SIP proxy
- Supported codec
- DID numbers
- All IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices.

This section provides verification steps that may be performed in the field to verify that the solution is configured properly.

6.1. System Status

The System Status application is used to monitor and troubleshoot IP Office. Use the System Status application to verify the state of the SIP trunk. System Status can be accessed from **Start** → **Programs** → **IP Office** → **System Status**. Or by opening an Internet browser and type the URL: `http://ipaddress` where *ipaddress* is the IP address of the Avaya IP Office LAN1 interface. Click on **System Status** to launch the application.



The following screen shows an example **Logon** screen. Enter the IP Office IP address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.

AVAYA IP Office System Status

Help About

Online Offline

Logon

Control Unit IP Address: 192.168.97.39

Services Base TCP Port: 50804

User Name: Administrator

Password:

☒ Auto reconnect

Logon

Select the SIP line under **Trunks** from the left pane. On the **Status** tab in the right pane, verify the **Current State** is **Idle** for each channel.

AVAYA IP Office System Status

Help Snapshot LogOff Exit About

System

- Alarms (7)
- Extensions (27)
- Trunks (5)
 - Line: 1
 - Line: 2
 - Line: 17
 - Line: 18
 - Line: 19
- Active Calls
- Resources
- Voicemail
- IP Networking

Status Utilization Summary Alarms Registration

SIP Trunk Summary

Peer Domain Name: preprod.bvoice.primus.ca

Resolved Address: xx.xx.xx.xx

Line Number: 19

Number of Administered Channels: 10

Number of Channels in Use: 0

Administered Compression: G711 Mu, G729 A, G722

Silence Suppression: Off

SIP Trunk Channel Licenses: Unlimited

SIP Trunk Channel Licenses in Use: 0 0%

SIP Device Features: REFER (Incoming and Outgoing), UPDATE (Incoming and Outgoing)

Channel Number	URI Gr...	Call Ref	Current State	Time in State	Remote Media Address	Codec	Connection Type	Caller ID or Dialed Digits	Other Party on Call	Direction of Call
1			Idle	2 days 15:...						
2			Idle	2 days 15:...						
3			Idle	2 days 15:...						
4			Idle	2 days 15:...						
5			Idle	2 days 15:...						
6			Idle	2 days 15:...						
7			Idle	2 days 15:...						
8			Idle	2 days 15:...						
9			Idle	2 days 15:...						
10			Idle	2 days 15:...						

Select the **Registration** tab and verify that status of the SIP line is registered. Select the **Alarms** tab (not shown) and confirm there are no alarms for the SIP line.

The screenshot shows the AVAYA IP Office System Status web interface. The left sidebar contains navigation links: System, Alarms (7), Extensions (27), Trunks (5), Active Calls, Resources, Voicemail, and IP Networking. The main content area has tabs for Status, Utilization Summary, Alarms, and Registration. The Registration tab is active, displaying a table titled 'Registration Status'.

Index	User Name	Status	Retry Time
1	6471234567	Registered	6/4/2012 3:51:18 PM

6.2. Monitor

The Monitor application can also be used to monitor and troubleshoot IP Office. Monitor can be accessed from **Start → Programs → IP Office → Monitor**. The application allows the monitored information to be customized. To customize, select the button that is third from the right in the screen below, or select **Filters → Trace Options**.

The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, the **SIP Rx** and **SIP Tx** boxes are checked. All SIP messages will appear in the trace with the color blue. To customize the color, right-click on **SIP Rx** or **SIP Tx** and select the desired color.

The screenshot shows the 'All Settings' dialog box with the 'SIP' tab selected. The 'Events' section has 'Sip' checked with a 'High' priority. The 'Packets' section has 'Sip Rx' and 'Sip Tx' checked, with 'hex' selected for both. The 'IP Filter' field is empty.

T1		VPN		WAN		SCN	
ATM	Call	DTE	EConf	Frame Relay	GDD	H.323	Interface
ISDN	Key/Lamp	Directory	Media	PPP	R2	Routing	Services
						SIP	System
<p>Events</p> <p><input checked="" type="checkbox"/> Sip High <input type="checkbox"/> STUN</p> <p>Packets</p> <p><input type="checkbox"/> SIP Reg/Opt Rx <input type="checkbox"/> SIP Misc Rx</p> <p><input type="checkbox"/> SIP Reg/Opt Tx <input type="checkbox"/> SIP Misc Tx</p> <p><input type="checkbox"/> SIP Call Rx <input type="checkbox"/> Cm Notify Rx</p> <p><input type="checkbox"/> SIP Call Tx <input type="checkbox"/> Cm Notify Tx</p> <p><input checked="" type="checkbox"/> Sip Rx <input type="checkbox"/> hex IP Filter (nnn.nnn.nnn.nnn)</p> <p><input checked="" type="checkbox"/> Sip Tx <input type="checkbox"/> hex</p>							
<p>Default All Clear All Tab Clear All Tab Set All OK Cancel</p> <p>Save File Load File Select File</p>							

As an example, the following shows a portion of the monitoring window for an outbound call from extension 29225, whose DID is 647-123-1111, calling out to the PSTN via the Primus IP Trunking Service. The telephone user dialed is 1-613-967-5280. The IP address of Primus SIP trunking is being hidden for security reasons.

```
383110774mS SIP Tx: UDP 192.168.97.39:5060 ->xx.xx.xx.xx :5060
INVITE sip:16139675280@preprod.bvoice.primus.ca SIP/2.0
Via: SIP/2.0/UDP 192.168.97.39:5060;rport;branch=z9hG4bK42051c3403e602d06671b1ceda2b0fd8
From: "Extn29225" <sip:6471231111@preprod.bvoice.primus.ca>;tag=544edc71fe73ca7b
To: <sip:16139675280@preprod.bvoice.primus.ca>
Call-ID: 14b458852d6f81cf9b50063798ecff0d@192.168.97.39
CSeq: 1874326810 INVITE
Contact: "Extn29225" <sip:6471231111@192.168.97.39:5060;transport=udp>
Max-Forwards: 70
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, UPDATE
Content-Type: application/sdp
Supported: timer
Content-Length: 275

v=0
o=UserA 3784094907 3802922234 IN IP4 192.168.97.39
s=Session SDP
c=IN IP4 192.168.97.39
t=0 0
m=audio 49154 RTP/AVP 0 18 9 101
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:9 G722/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

7. Conclusion

These Application Notes describe the configuration necessary to connect Avaya IP Office 8.0 to Primus SIP Trunking service. Primus SIP Trunking is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. It provides a flexible, cost-saving alternative to traditional hardwired telephony trunks. Primus SIP Trunking passed compliance testing. Please refer to **Section 2.2** for any exceptions.

8. Additional References

This section references documentation relevant to these Application Notes. In general, Avaya product documentation is available at <http://support.avaya.com>.

[1] IP Office 8.0 IP Office Installation, Document Number 15-601042, Issue 25b, March 08, 2012

[2] IP Office Release 8.0 Manager 10.0, Document Number 15-601011, Issue 28h, March 28 2012

[3] IP Office System Status Application, Issue 06b, November 12, 2011 Document Number 15-601758

[4] IP Office Release 8.0 Administering Voicemail Pro, Document Number 15-601063, Issue 27b, April 06 2012

[5] IP Office System Monitor, Document Number 15-601019, Issue 02b

Additional IP Office documentation can be found at:

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

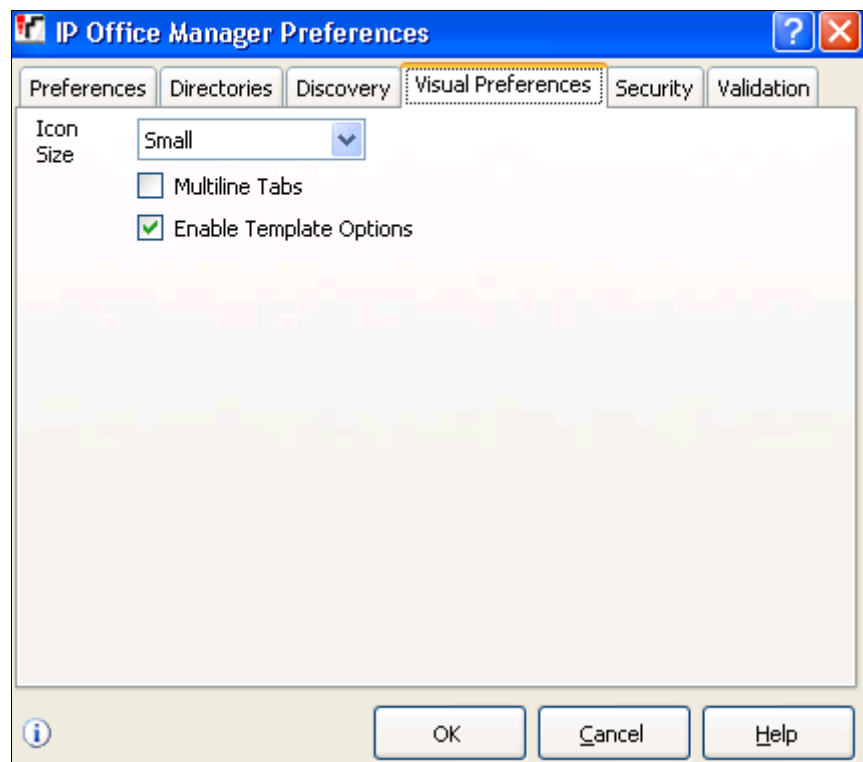
9. Appendix - SIP Line Template

IP Office Release 8.0 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 5.8** in these Application Notes as a reference.

9.1. Configure IP Office Manager for Template Creation

To enable IP Office to create a SIP Trunk template, configure as follows on the desktop where the IP Office Manager is installed:

1. Navigate to **File → Preferences** on the IP Office Manager and select the **Visual Preferences** tab. Check the **Enable Template Options** box as shown below.

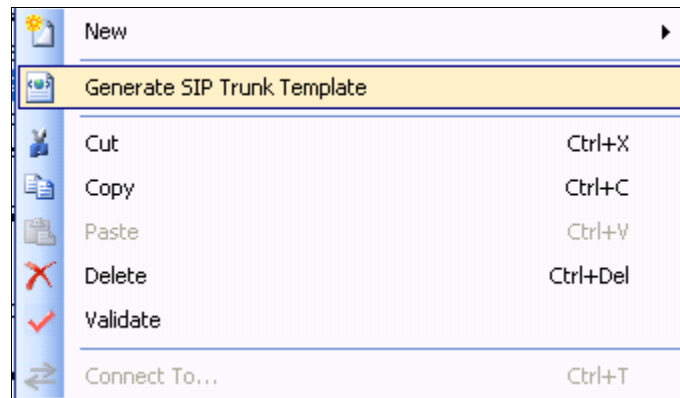


2. Run **regedit** on the desktop and navigate to **HKEY_CURRENT_USER/Software/Avaya/IP400/Manager** and add a **DWORD** value **TemplateProvisioning** and set its value to **1**. Reboot the server hosting the IP Office Manager.

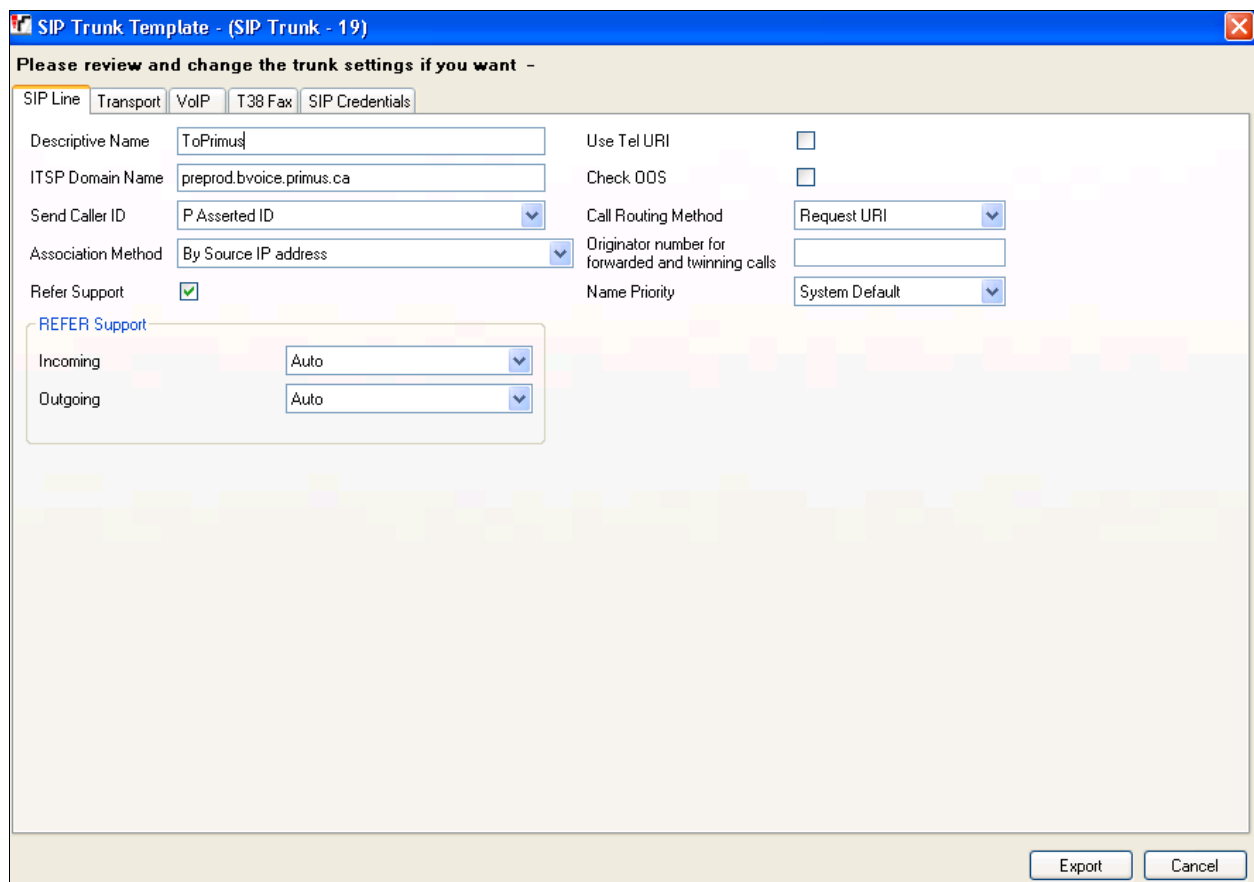
9.2. Generate a SIP Trunk Template

To generate a SIP Trunk template from an existing SIP trunk, execute the following steps:

1. Select the SIP trunk under line and right click on the SIP line number for which the SIP trunk template is to be generated and then click **Generate SIP Trunk Template**.



2. In the SIP Trunk Template screen shown below, enter a template name in the **Descriptive Name** field and click **Export**.

A screenshot of the 'SIP Trunk Template - (SIP Trunk - 19)' dialog box. The dialog has a title bar with a close button. Below the title bar is a tabbed interface with tabs for 'SIP Line', 'Transport', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'SIP Line' tab is selected. The main area contains several fields and checkboxes. The 'Descriptive Name' field is filled with 'ToPrimus'. The 'ITSP Domain Name' field is filled with 'preprod.bvoice.primus.ca'. The 'Send Caller ID' dropdown is set to 'P Asserted ID'. The 'Association Method' dropdown is set to 'By Source IP address'. The 'Refer Support' checkbox is checked. The 'Use Tel URI' checkbox is unchecked. The 'Check OOS' checkbox is unchecked. The 'Call Routing Method' dropdown is set to 'Request URI'. The 'Originator number for forwarded and twinning calls' field is empty. The 'Name Priority' dropdown is set to 'System Default'. There is a 'REFER Support' section with 'Incoming' and 'Outgoing' dropdowns, both set to 'Auto'. At the bottom right are 'Export' and 'Cancel' buttons.

3. In the **Template Type Selection** screen, enter **Country** and **Service Provider** and click **Generate Template**.

The screenshot shows a window titled "Template Type Selection". Inside, there are three dropdown menus. The first is labeled "Locale" and has "United States (US English)" selected. The second is labeled "Country" and is currently empty. The third is labeled "Service Provider" and has "Primus" selected. At the bottom right of the window are two buttons: "Generate Template" and "Cancel".

A popup screen shows up (not shown) asking where the template is to be stored. This section shows an example SIP Trunk Template generated from the configuration presented in this document.

```
<?xml version="1.0" encoding="utf-8" ?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20120604</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>ToPrimus</DescriptiveName>
  <ITSPDomainName>preprod.bvoice.primus.ca</ITSPDomainName>
  <SendCallerID>CallerIDPAID</SendCallerID>
  <ReferSupport>true</ReferSupport>
  <ReferSupportIncoming>2</ReferSupportIncoming>
  <ReferSupportOutgoing>2</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>false</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>SystemDefault</LineNamePriority>
  <ITSPProxy>xx.xx.xx.xx</ITSPProxy> (note here that the service provider IP address is
    hidden for security reasons)
  <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>
  <UseAdvVoiceCodecPrefs>false</UseAdvVoiceCodecPrefs>
```

```

<CallInitiationTimeout>4</CallInitiationTimeout>
<DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
<VoipSilenceSupression>false</VoipSilenceSupression>
<ReinviteSupported>true</ReinviteSupported>
<FaxTransportSupport>FOIP_T38</FaxTransportSupport>
<UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
<CodecLockdown>false</CodecLockdown>
<Rel100Supported>false</Rel100Supported>
<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>
<TFOPEnhancement>true</TFOPEnhancement>
<DisableT30ECM>false</DisableT30ECM>
<DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
<DisableT30MRCompression>false</DisableT30MRCompression>
<NSFOVERRIDE>false</NSFOVERRIDE>
- <SIPCredentials>
  <Expiry>60</Expiry>
  <RegistrationRequired>true</RegistrationRequired>
  </SIPCredentials>
</Template>

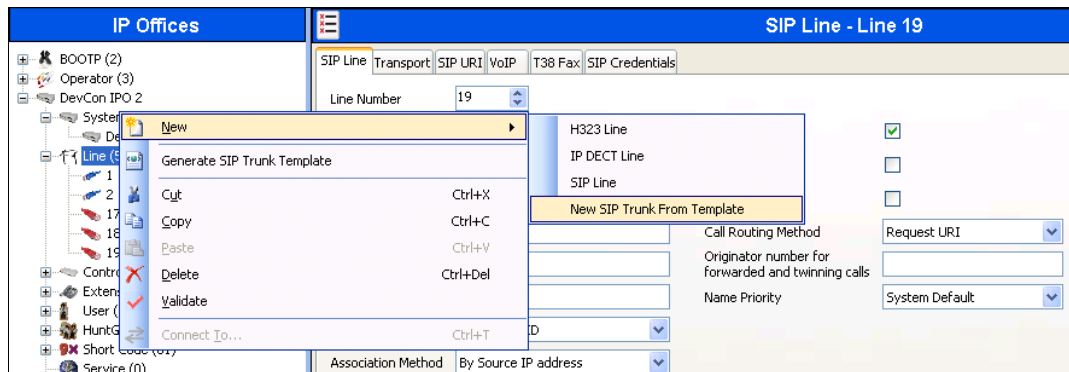
```


9.3. Create SIP Trunk from Template

To create a SIP Trunk from template shown above, execute the following steps:

On the PC where IP Office Manager was installed, copy and paste the above template into a text document named **US_Primus_SIPTrunk.xml** (the file must be named EXACTLY as show). Move the .xml file to the IP Office Manager template directory (C:\Program Files\Avaya\IP Office\Manager\Templates).

Right click on **Line**, select **New** and click **New SIP Trunk From Template**.



In the **Template Type Selection** screen displayed, verify that **Country** and **Service Provider** fields are auto populated with the information configured in **Section 9.2**. Click **Create new SIP Trunk**.



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