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,

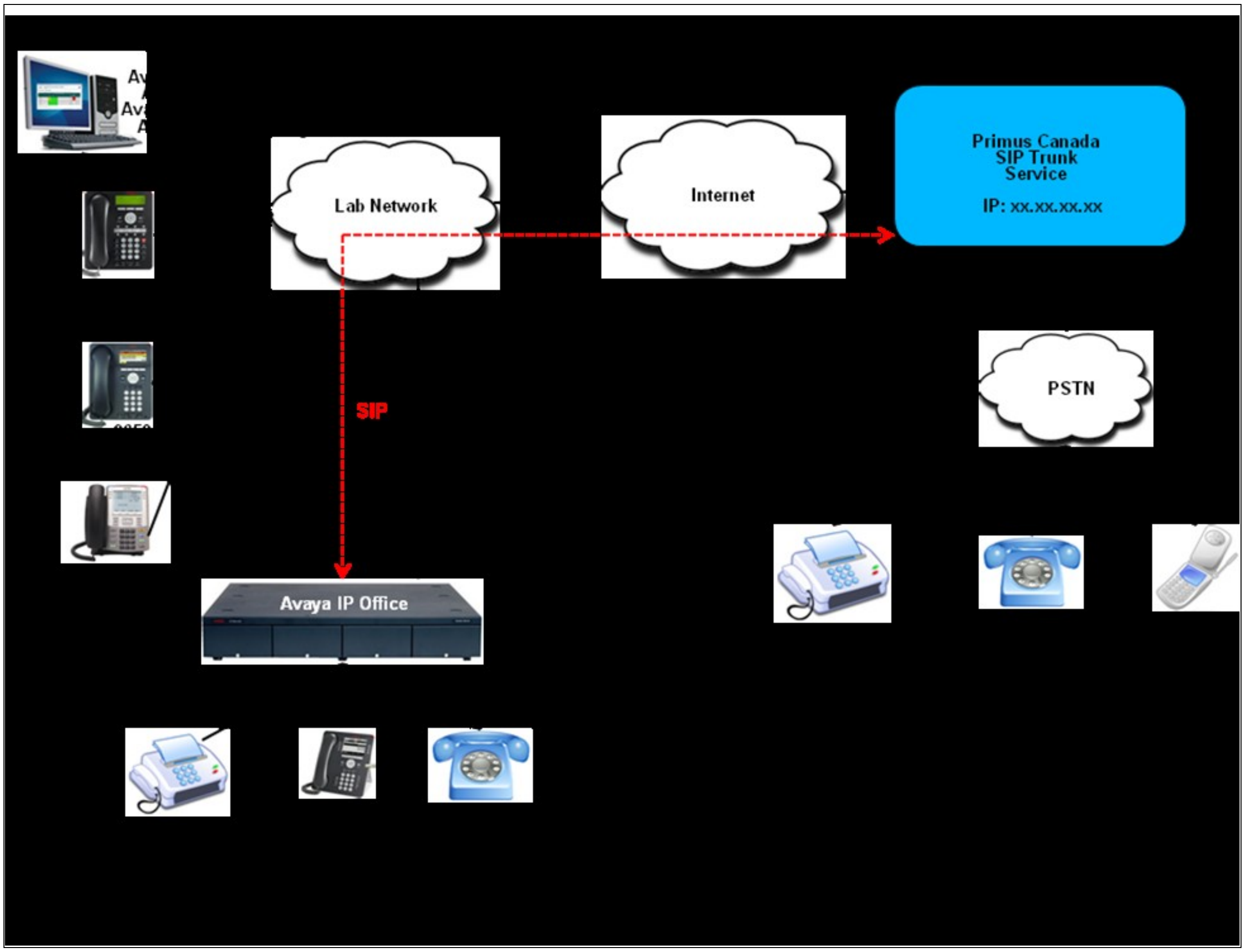


Figure 1: Avaya Interoperability Test Lab Configuration

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.

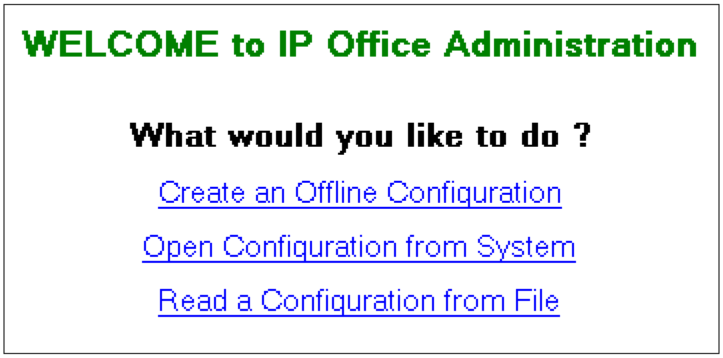
4. Equipment and Software Validated

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:

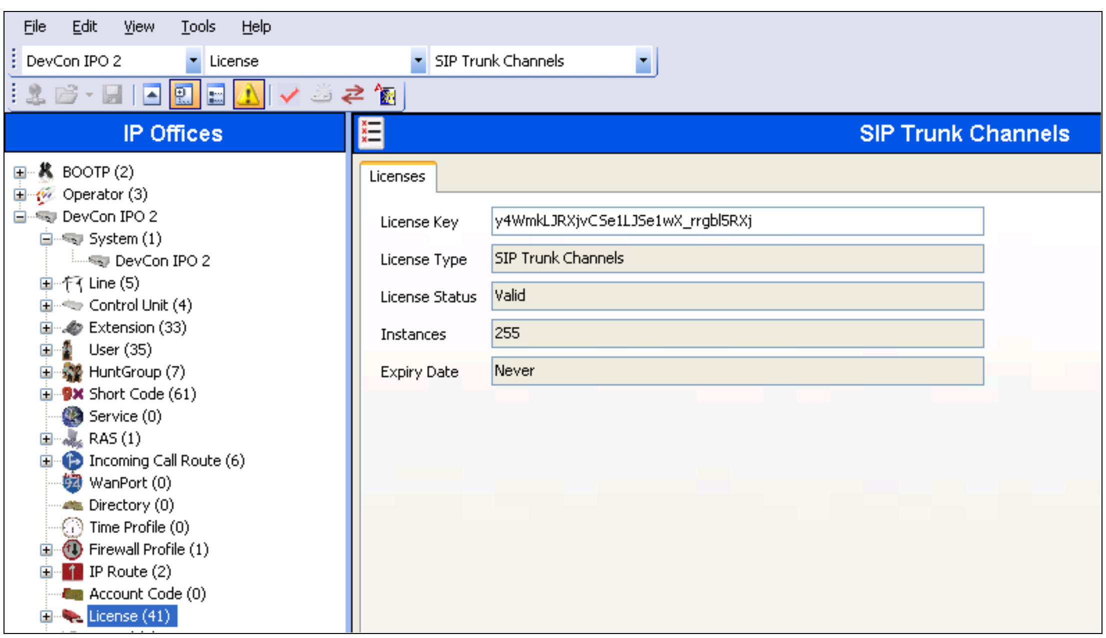


Navigate to **File → Open Configuration** (not shown), select the proper Avaya IP Office system from the pop-up window and log in with the appropriate credentials. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu 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.

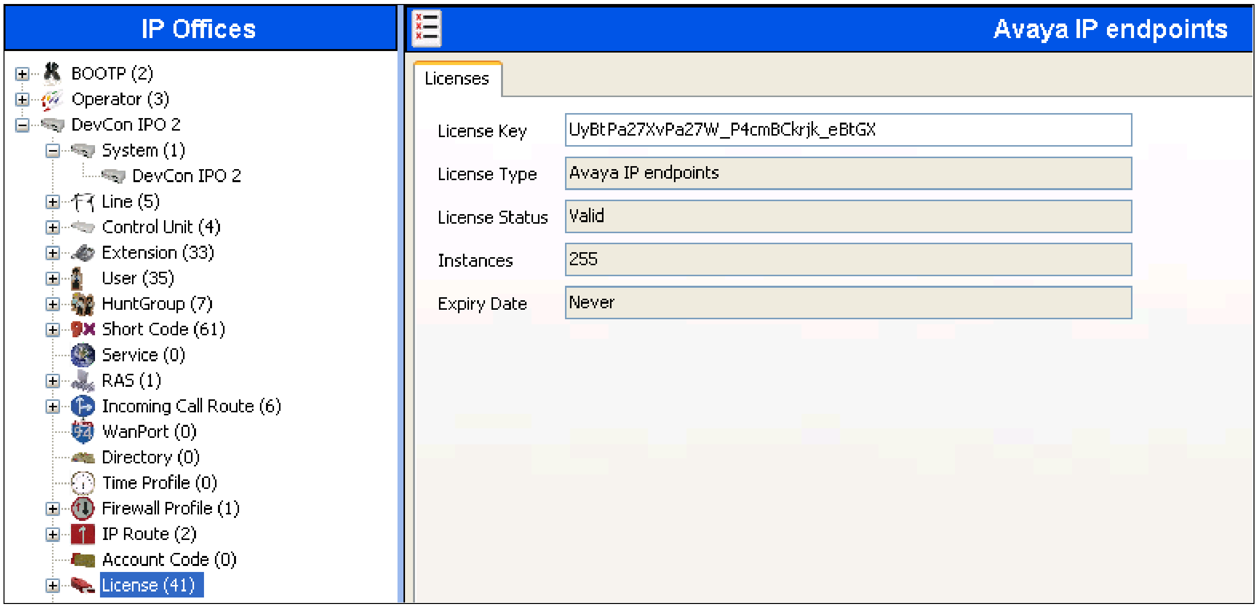
5.1. Licensing and Physical Hardware

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.

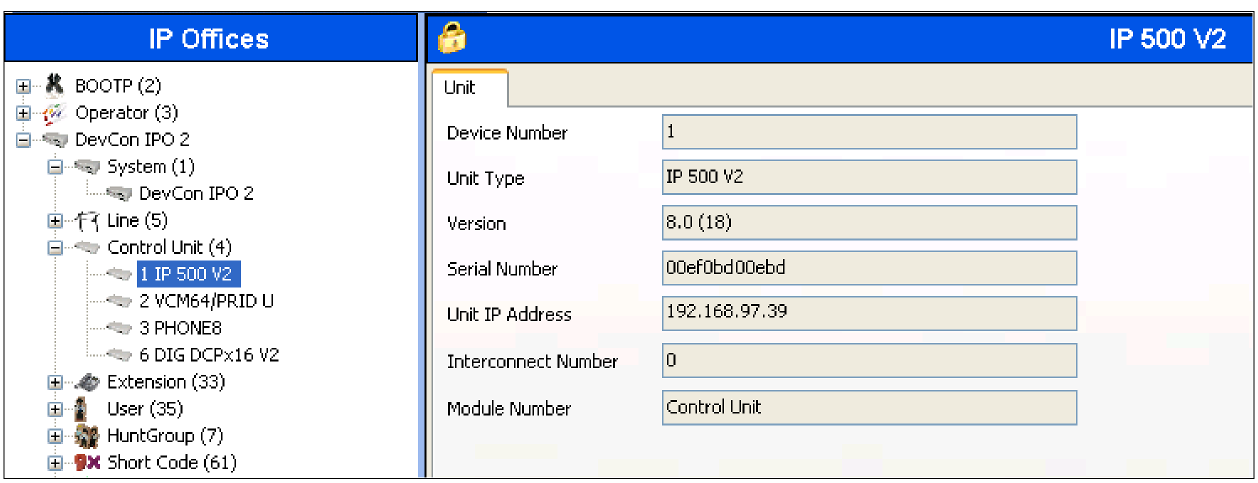


If Avaya IP Telephones will be used as is the case in these Application Notes, verify the Avaya IP endpoints license. Click **License** in the Navigation pane and **Avaya IP endpoints**. Confirm a valid license with sufficient **Instances** in the Details pane.



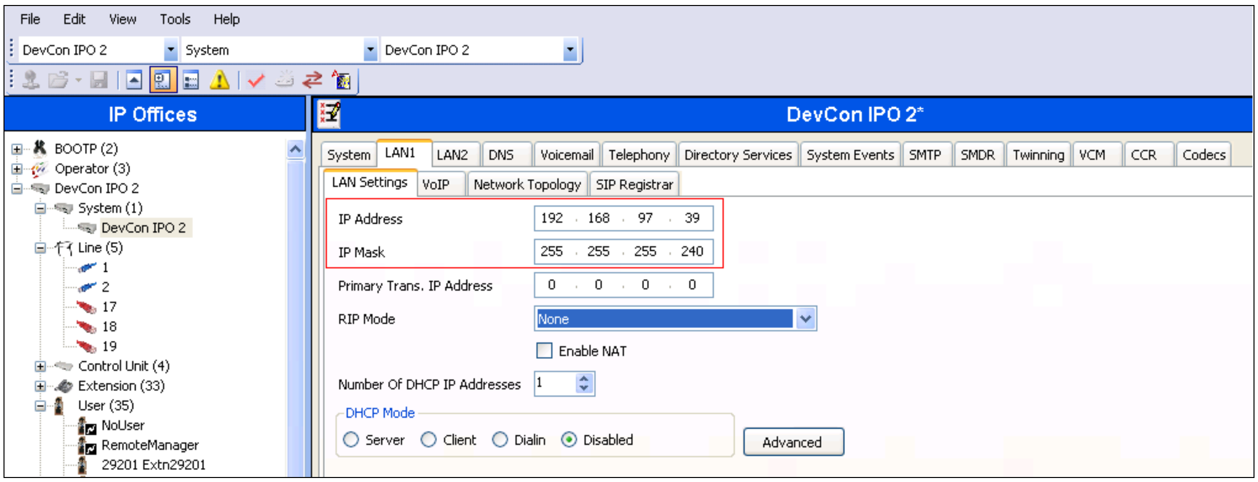
The VCM is a Voice Compression Module supporting VoIP codecs. An IP Office hardware configuration with a VCM component is necessary to support SIP trunking.

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.

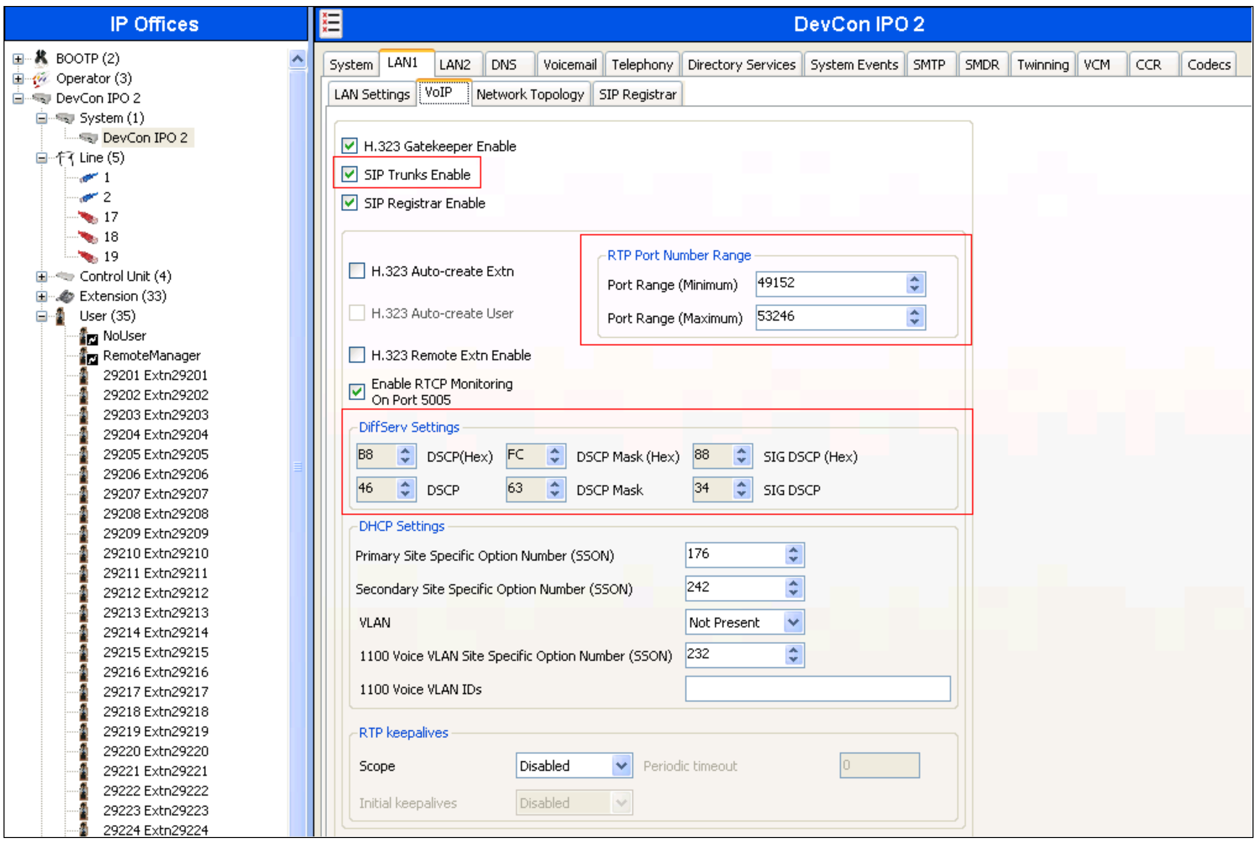


5.2. LAN1 Settings

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.

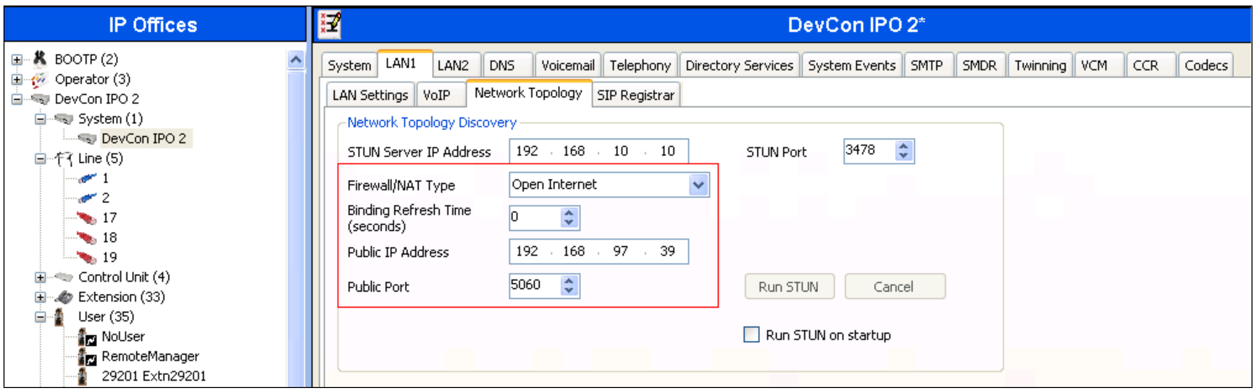


On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP 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.



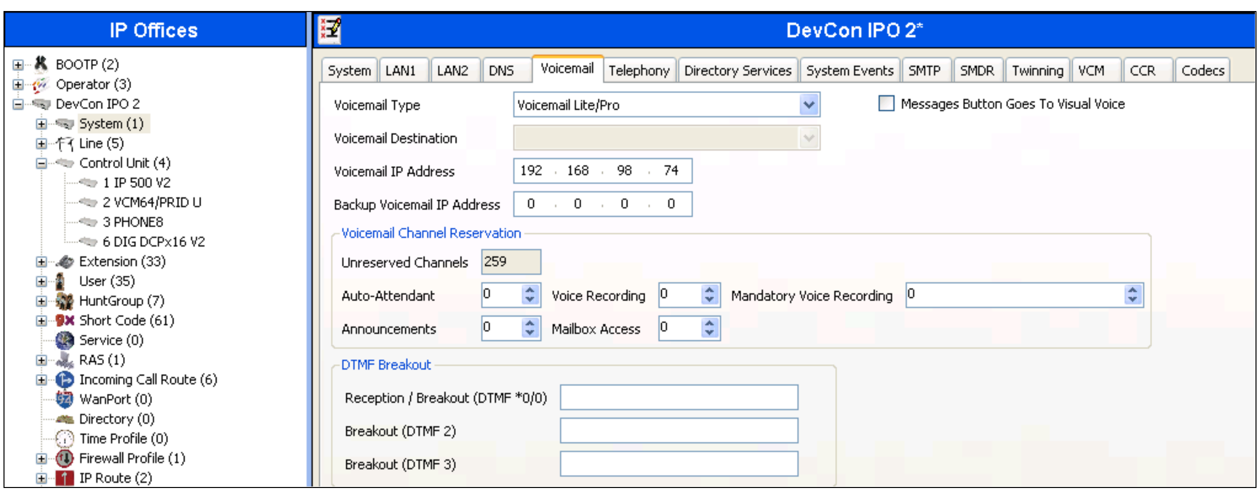
On the **Network Topology** tab in the Details Pane, configure the following parameters:

- 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.



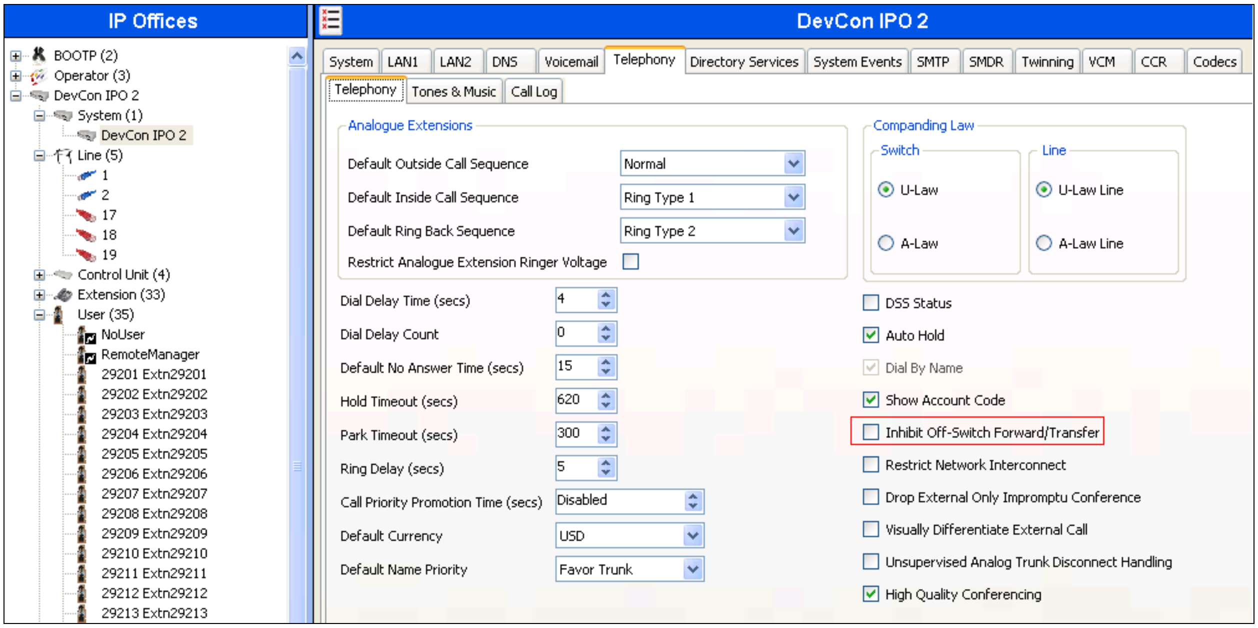
5.3. Voicemail Settings

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.



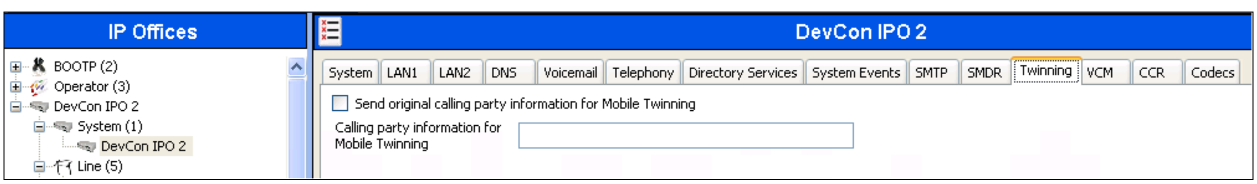
5.4. System Telephony Settings

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.



5.5. Twinning Calling Party Settings

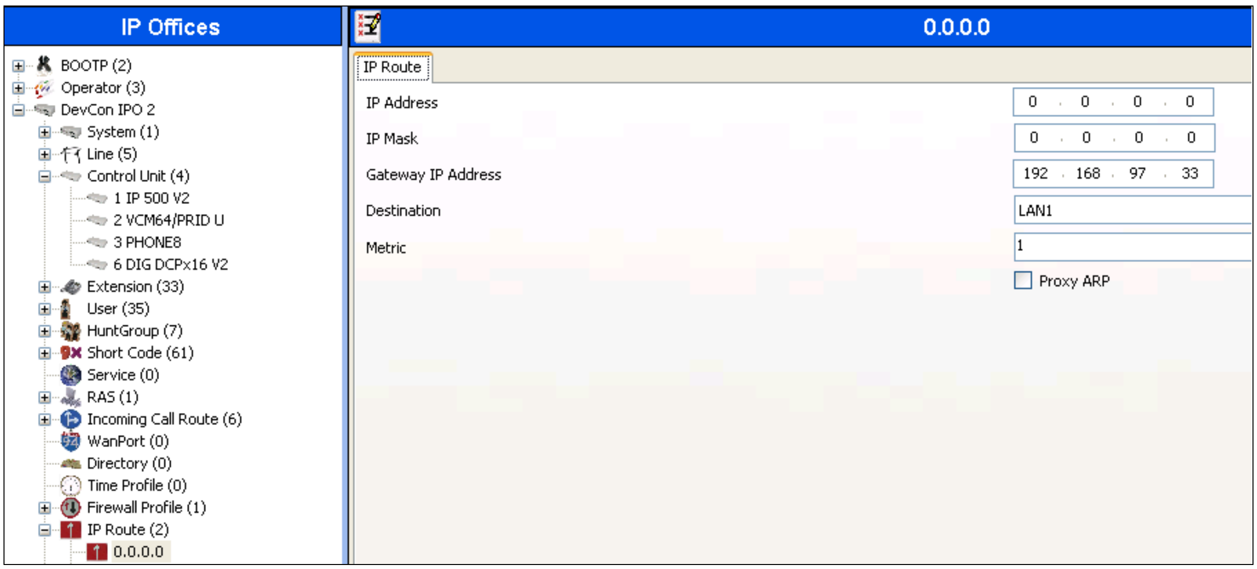
To view or change Twinning settings, select the **Twinning** tab as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning** is left blank.



5.6. IP Route

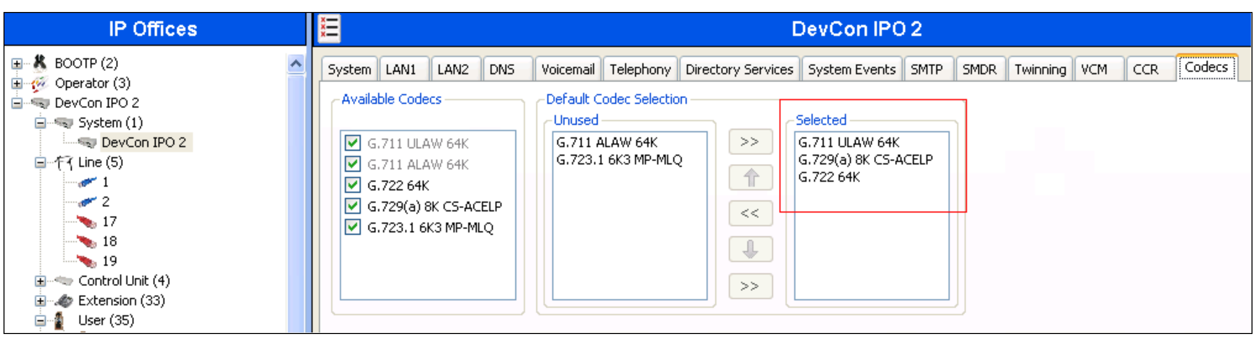
Navigate to **IP Route** in the left Navigation Pane, and then right-click to select **New** (not shown). Create a default route with the following parameters:

- 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.



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.



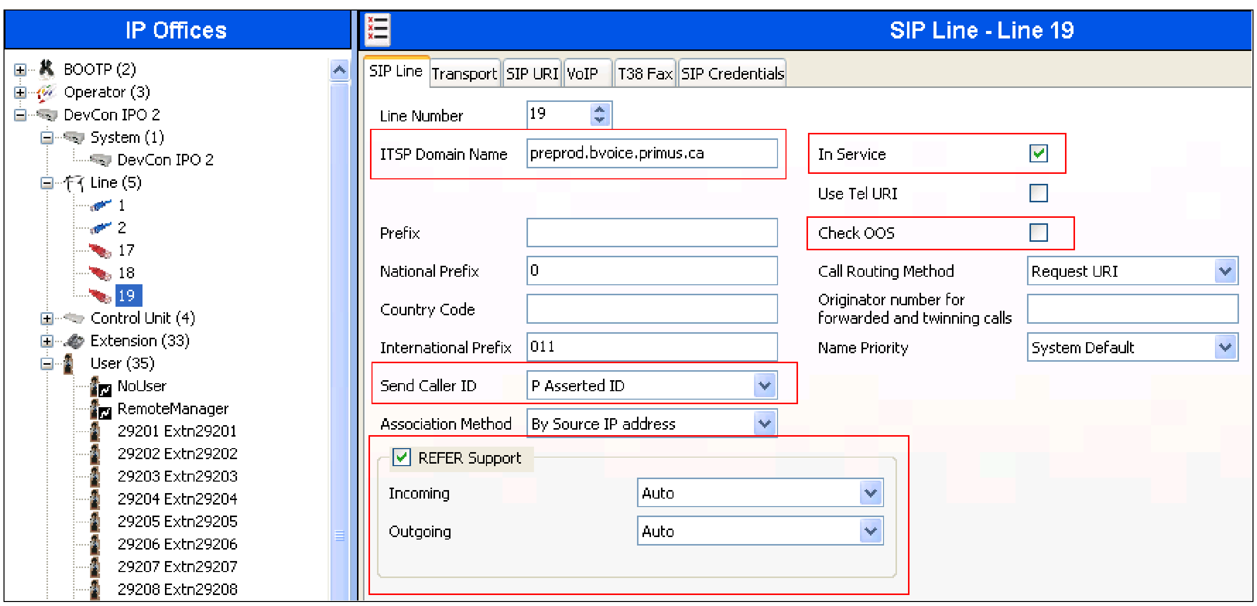
5.8. SIP Line

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

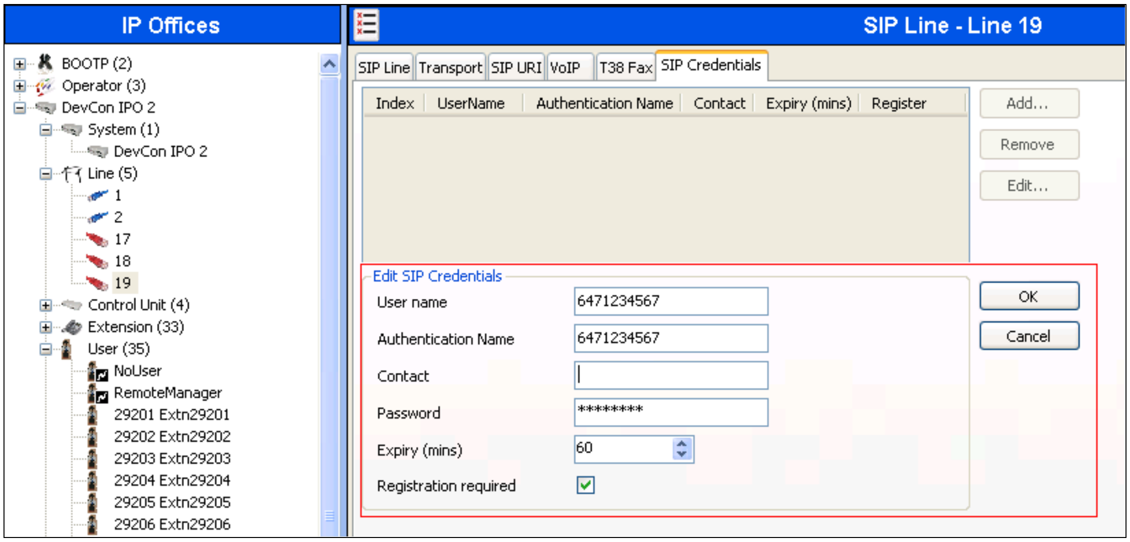
On the **SIP Line** tab in the Details Pane, configure the parameters as shown below.

- 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.



5.8.2. SIP Line – SIP Credentials

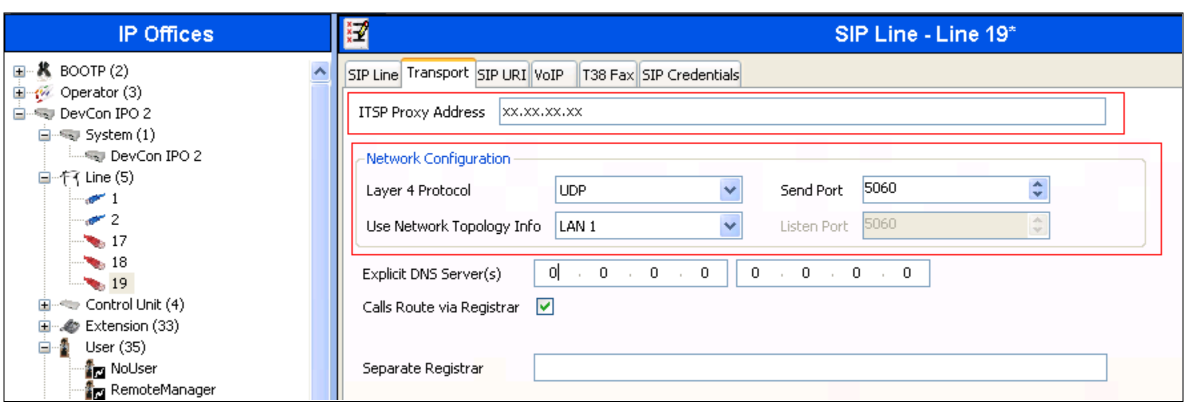
Primus requires trunk registration using credentials. In the screen below, from the **SIP Credentials** tab add the **User name**, **Authentication Name** and **Password** provided by the partner. Check the **Registration required** box. Retain default values for the remaining fields.



5.8.3. SIP Line - Transport Tab

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

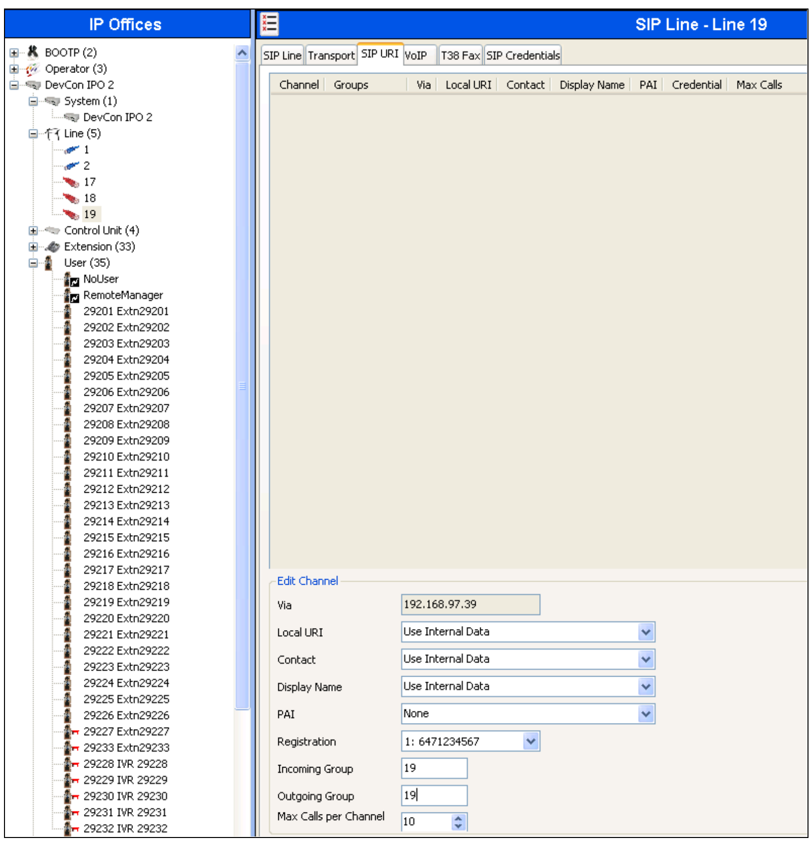
- Set **ITSP Proxy Address** to the IP address of the Primus SIP proxy.
- Set **Layer 4 Protocol** to ***UDP***.
- Set **Use Network Topology Info** to the network port configured in **Section 5.2**.
- Set the **Send Port** to **5060**.
- Default values may be used for all other parameters.



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.

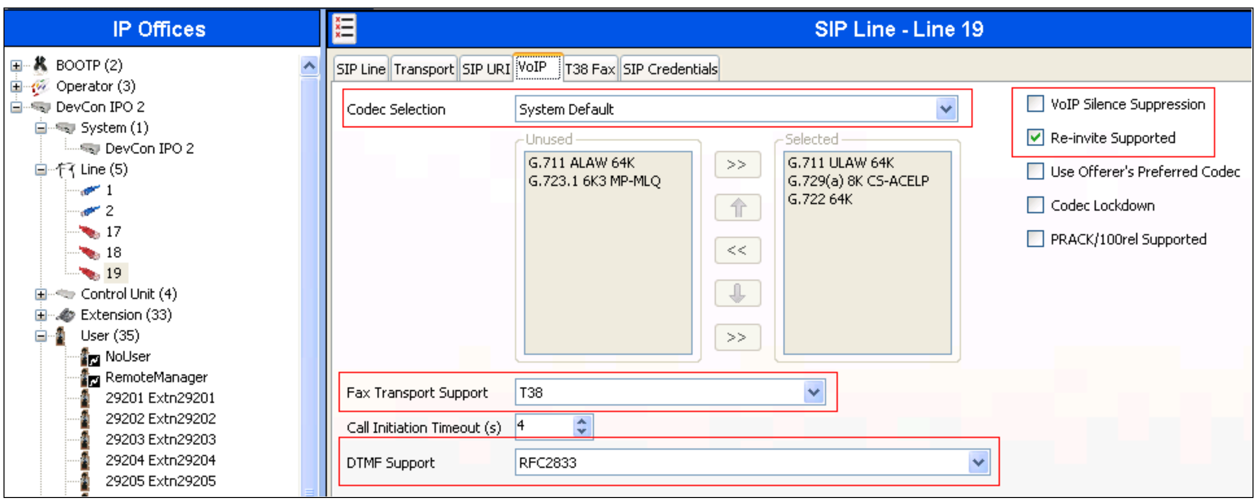


5.8.5. SIP Line - VoIP Tab

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

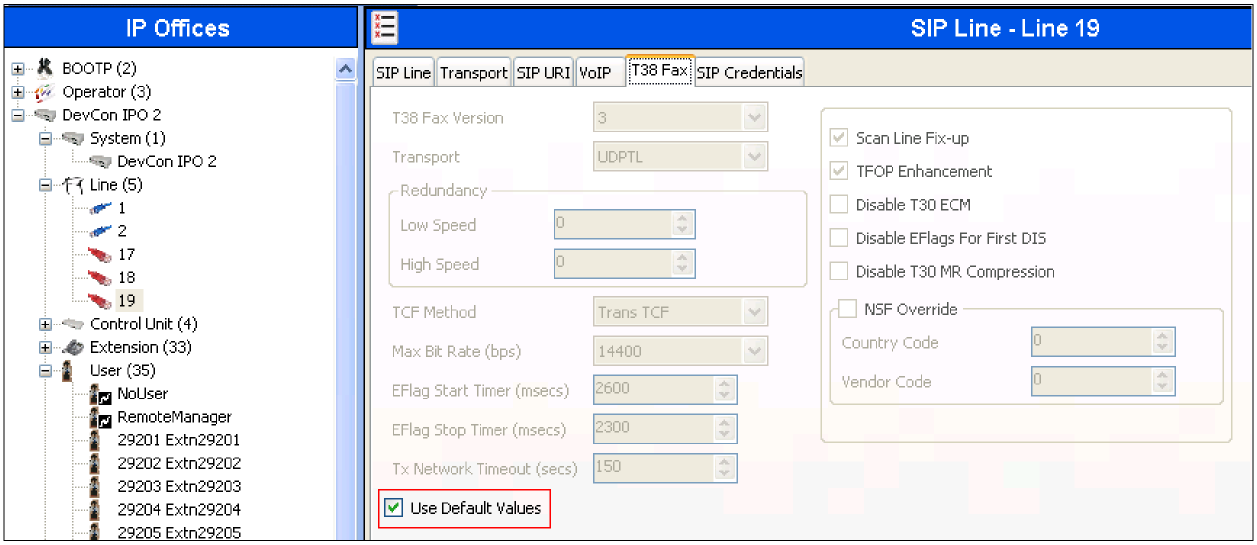
- Set the **Codec Selection** field to ***System Default*** to allow the service provider supported codec which were already configured in **Section 5.7**.
- Uncheck the **VoIP Silence Suppression** box.
- Check the **Re-invite Supported** box.
- Set the **Fax Transport Support** to ***T38***. See **Section 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).



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.

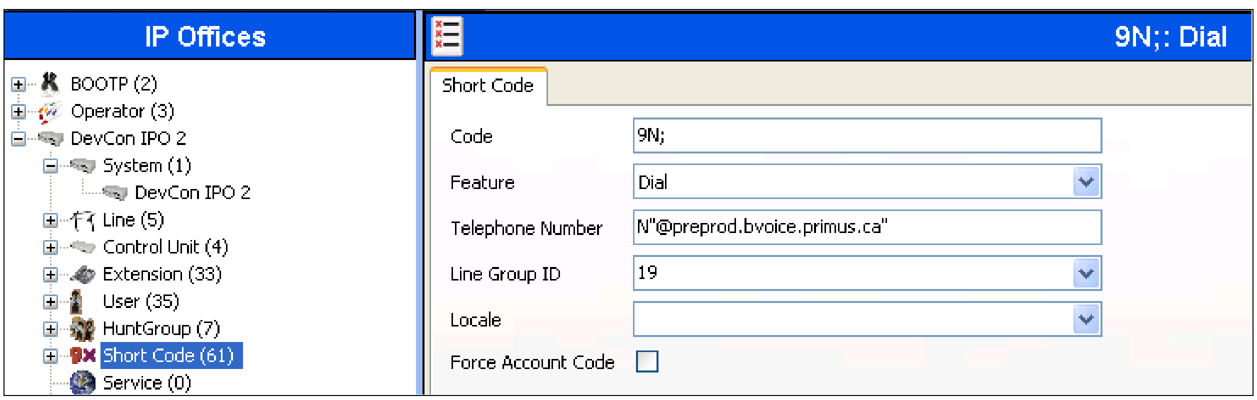


5.9. Short Codes

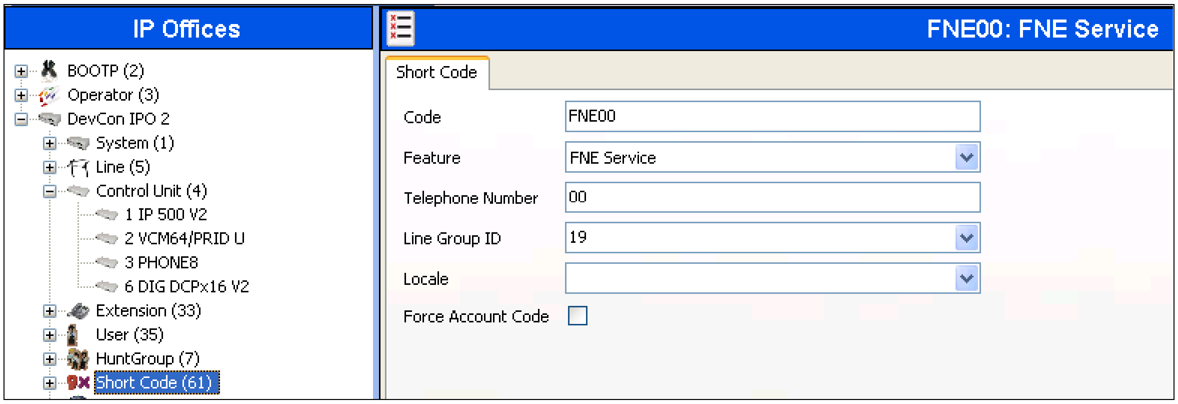
Define a short code to route outbound traffic to the SIP line. To create a short code, right-click on **Short Code** in the Navigation Pane and select **New** (not shown). On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

- 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).

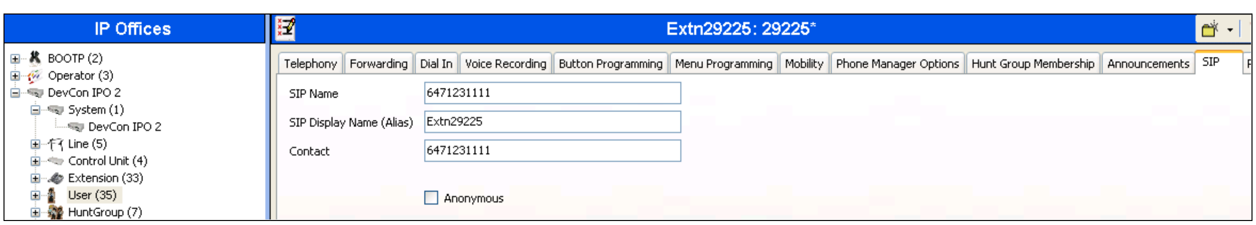


The following screen illustrates a short code that acts like a feature access code rather than a means to access a SIP Line. In this case, the **Code** ***FNE00*** is defined for **Feature** ***FNE Service*** to **Telephone Number** ***00*** (Mobile Call Control). This short code will be used as means 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.

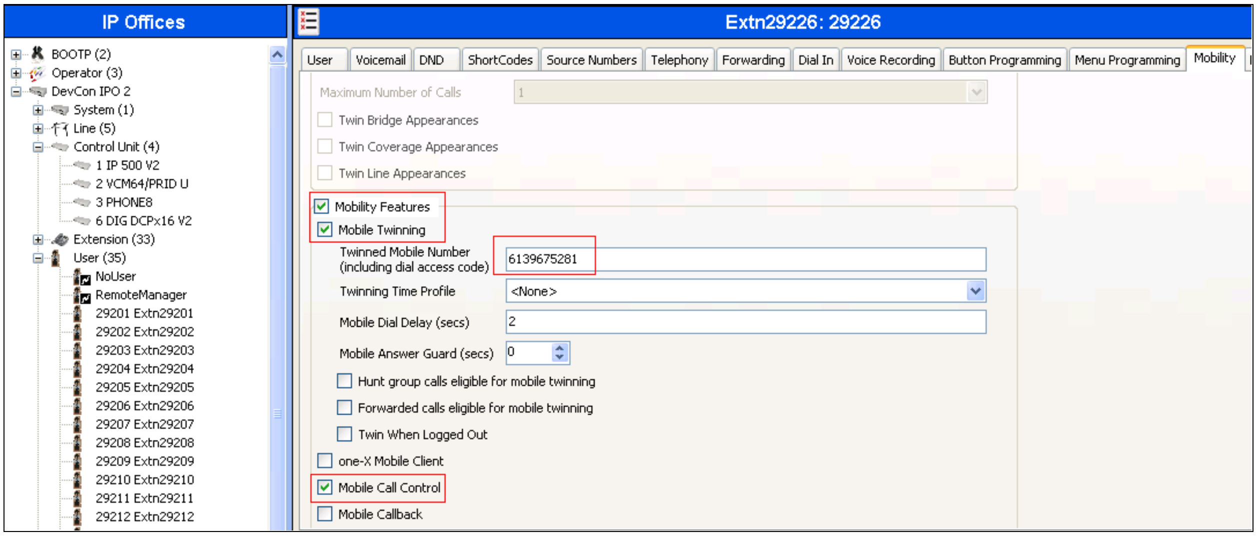


5.10. User

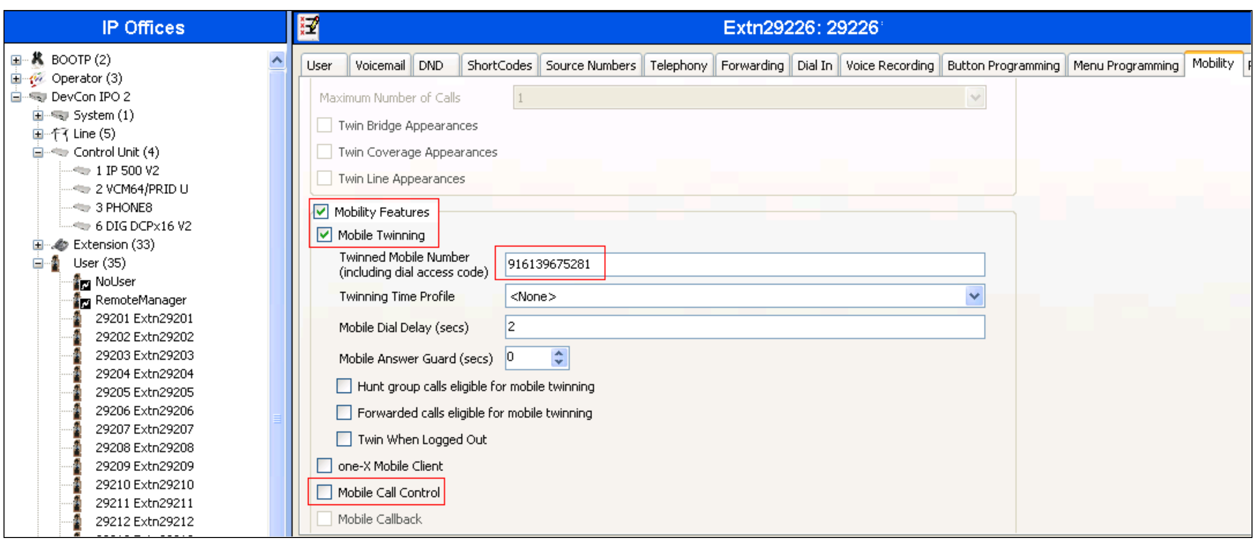
Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.88**. To configure these settings, first navigate to **User** in the Navigation Pane, and then click on the user to be modified. Select the **SIP** tab in the Details Pane. The 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 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.



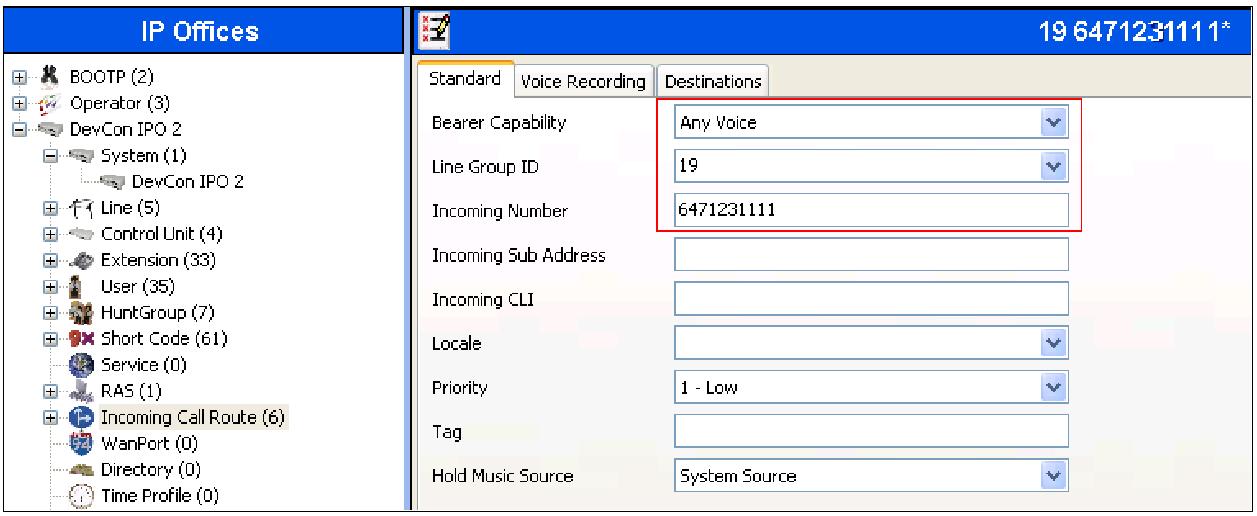
The screen below has the same configuration as the screen above except the **Mobile Call Control** box is not checked and the **Twinned Mobile Number** field is ***916139675281***. This is the configuration used during compliance testing for the twinning feature only.



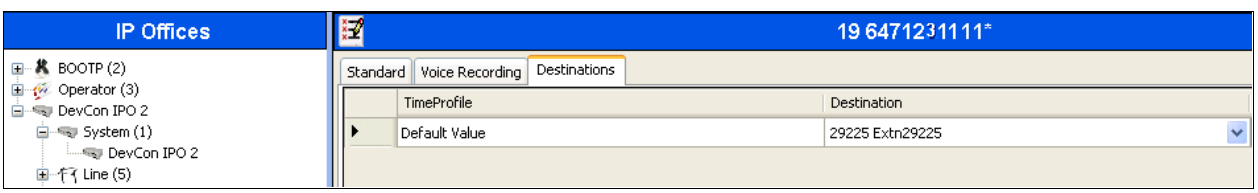
5.11. Incoming Call Route

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.

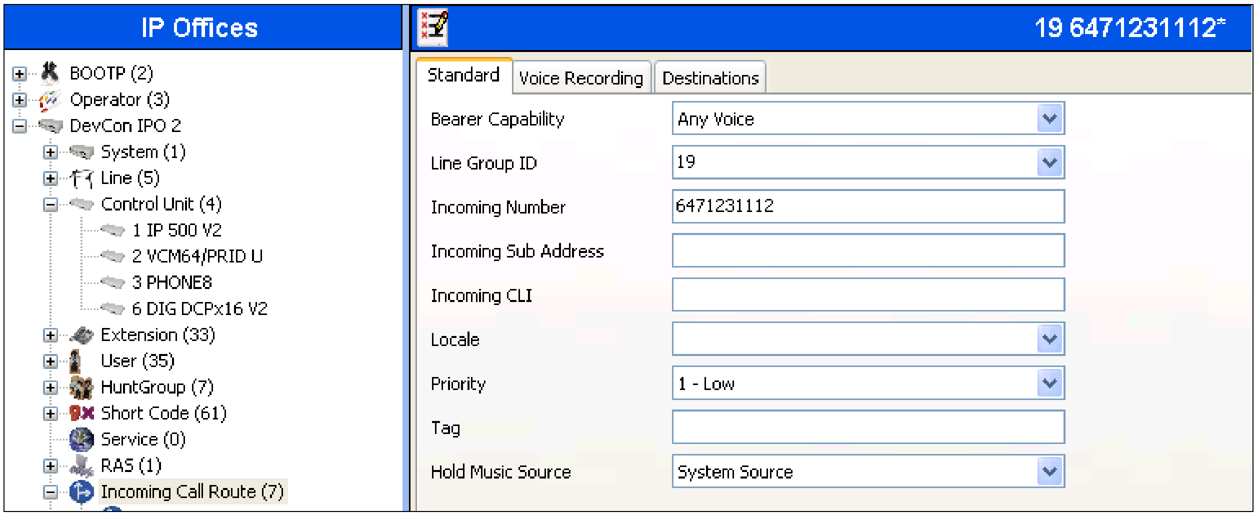


On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. Click the **OK** button (not shown). In this example, incoming calls to 6471231111 on line 19 are routed to extension 29225.

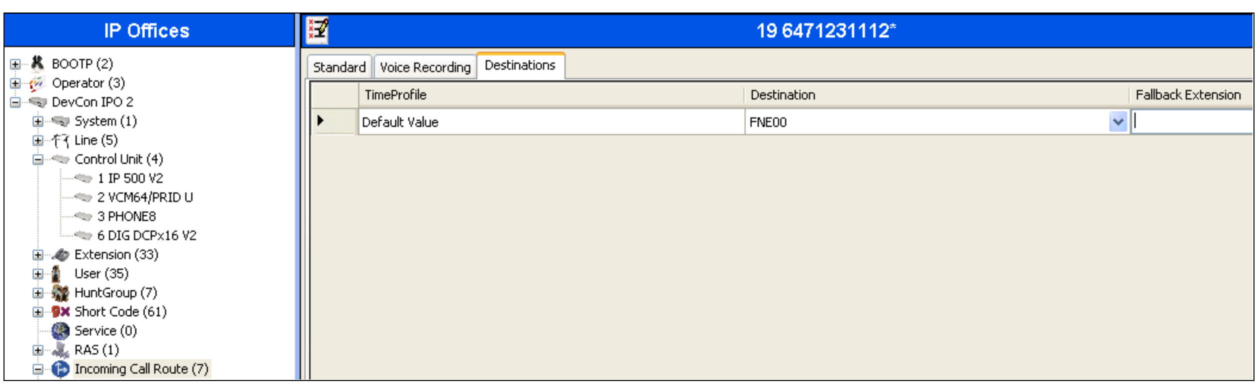


Incoming Call Routes for other direct mappings of DID numbers to IP Office users listed in **Figure 1** are omitted here, but can be configured in the same fashion.

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**.



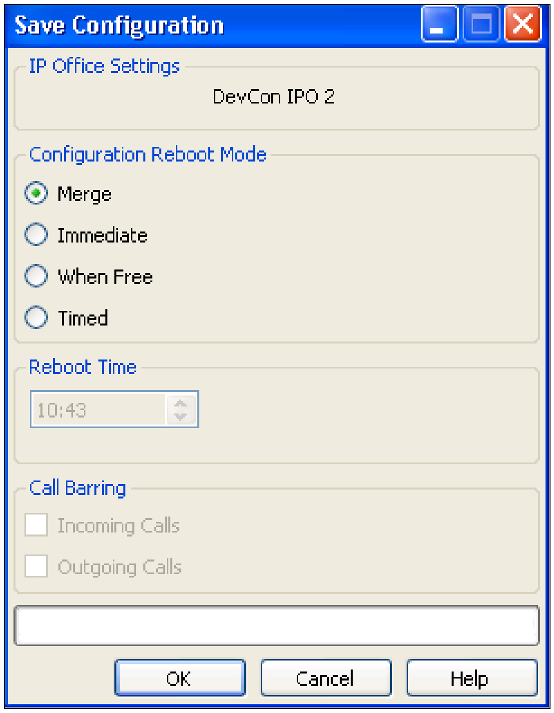
When configuring an Incoming Call Route, the **Destination** field can be manually configured with a number such as a short code, or certain keywords available from the pull-down menu. For example, the following **Destinations** tab for an incoming call route contains the **Destination** ***FNE00*** 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.



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:

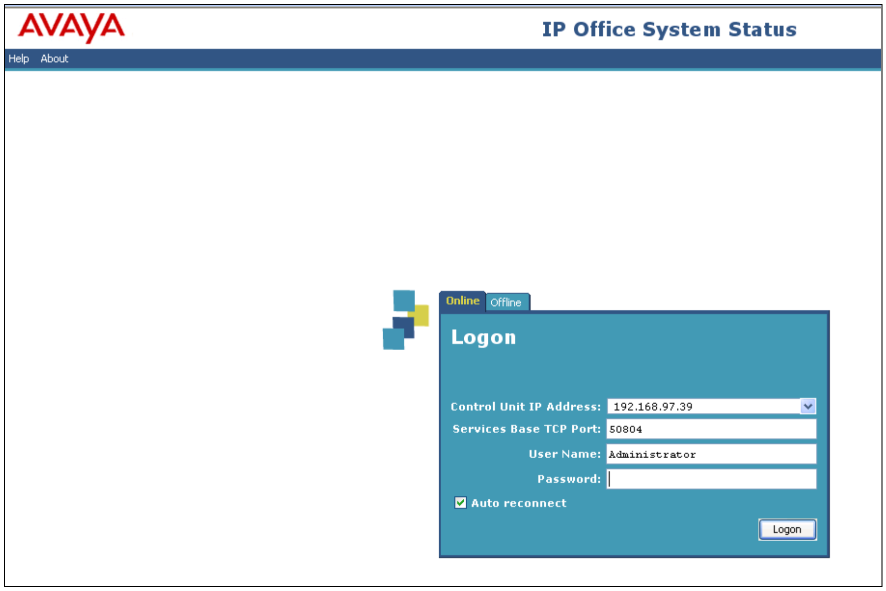
- IP address of the Primus SIP proxy
- Supported codec
- DID numbers
- All IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices.

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

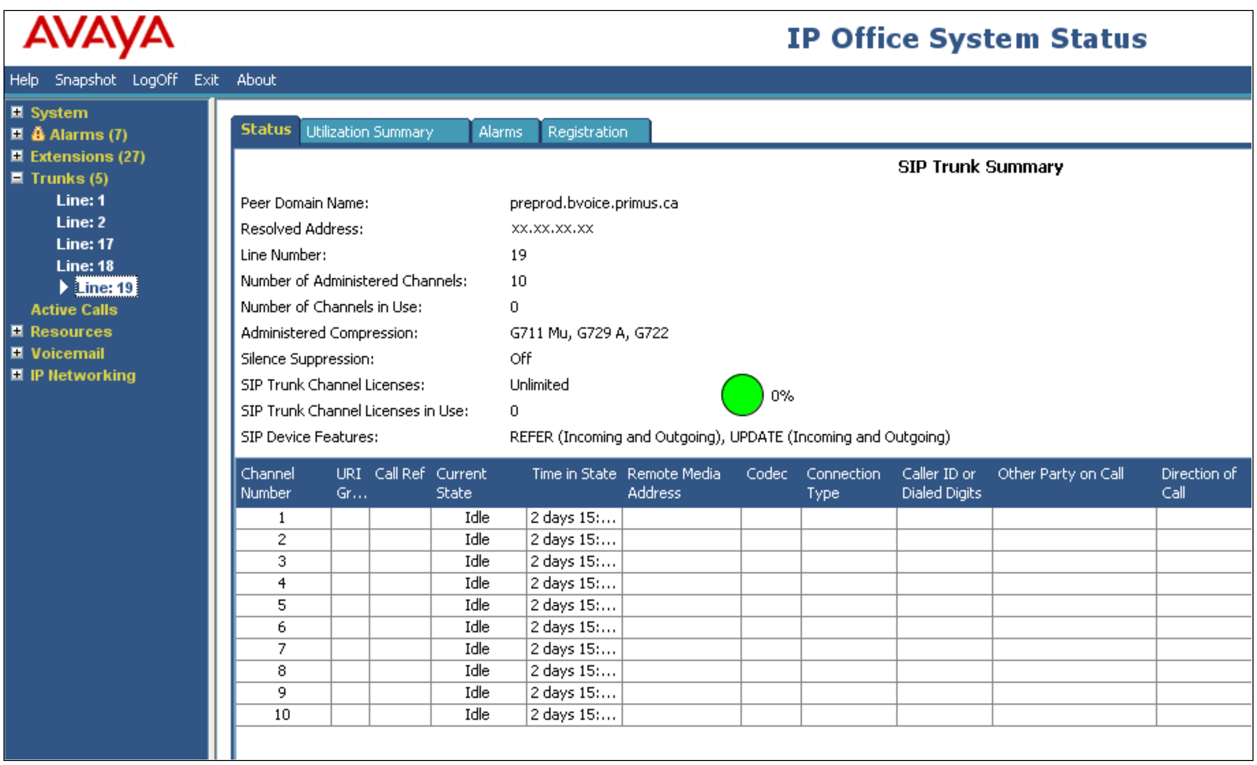
6.1. System Status

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.

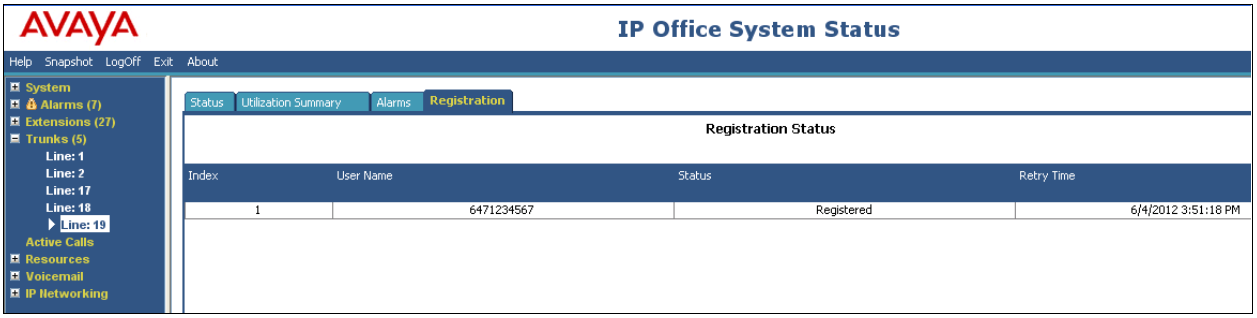
The following screen shows an example **Logon** screen. Enter the IP Office IP address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.



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



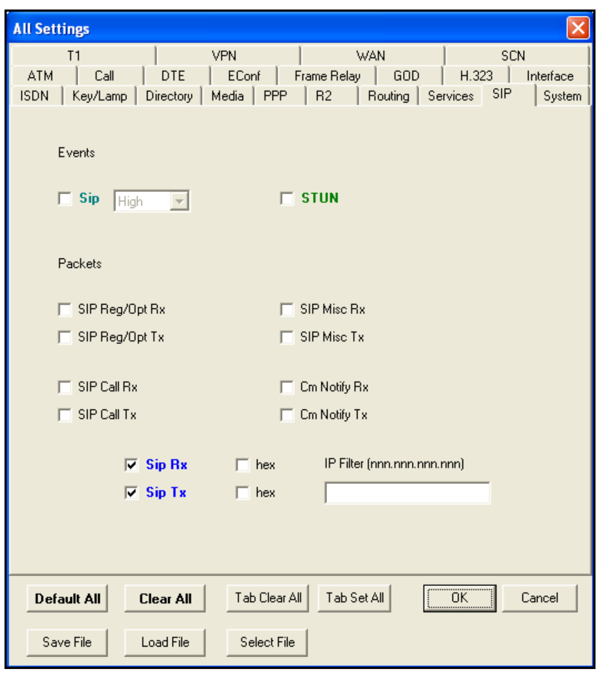
Select the **Registration** tab and verify that status of the SIP line is registered. Select the **Alarms** tab (not shown) and confirm there are no alarms for the SIP line.



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.



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/

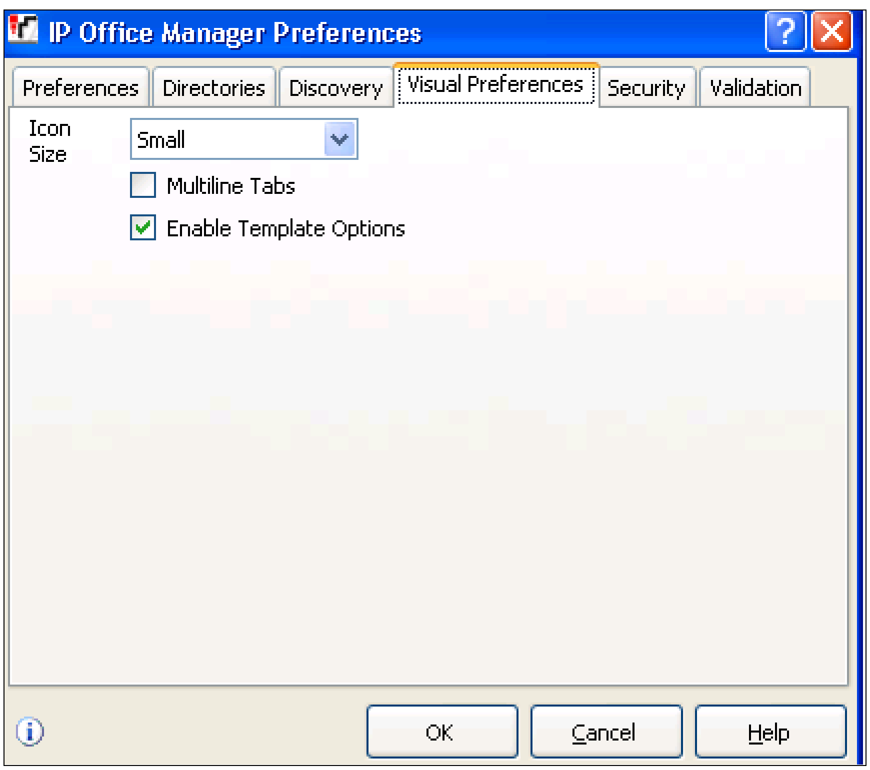
9. Appendix - SIP Line Template

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.

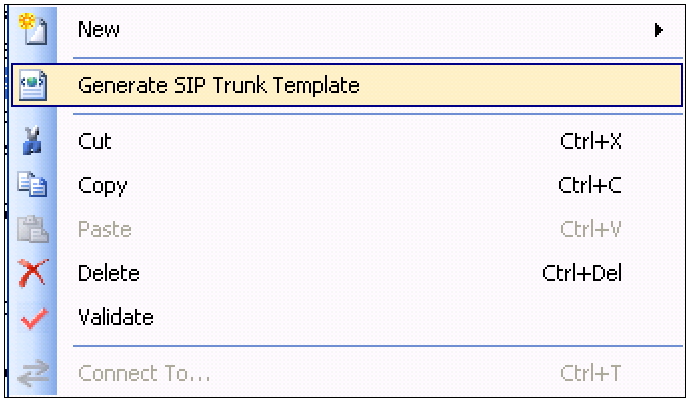


2. Run **regedit** on the desktop and navigate to **HKEY_CURRENT_USER/Software/Avaya/IP400/Manager** and add a **DWORD** value **TemplateProvisioning** and set its value to **1**. Reboot the server hosting the IP Office Manager.

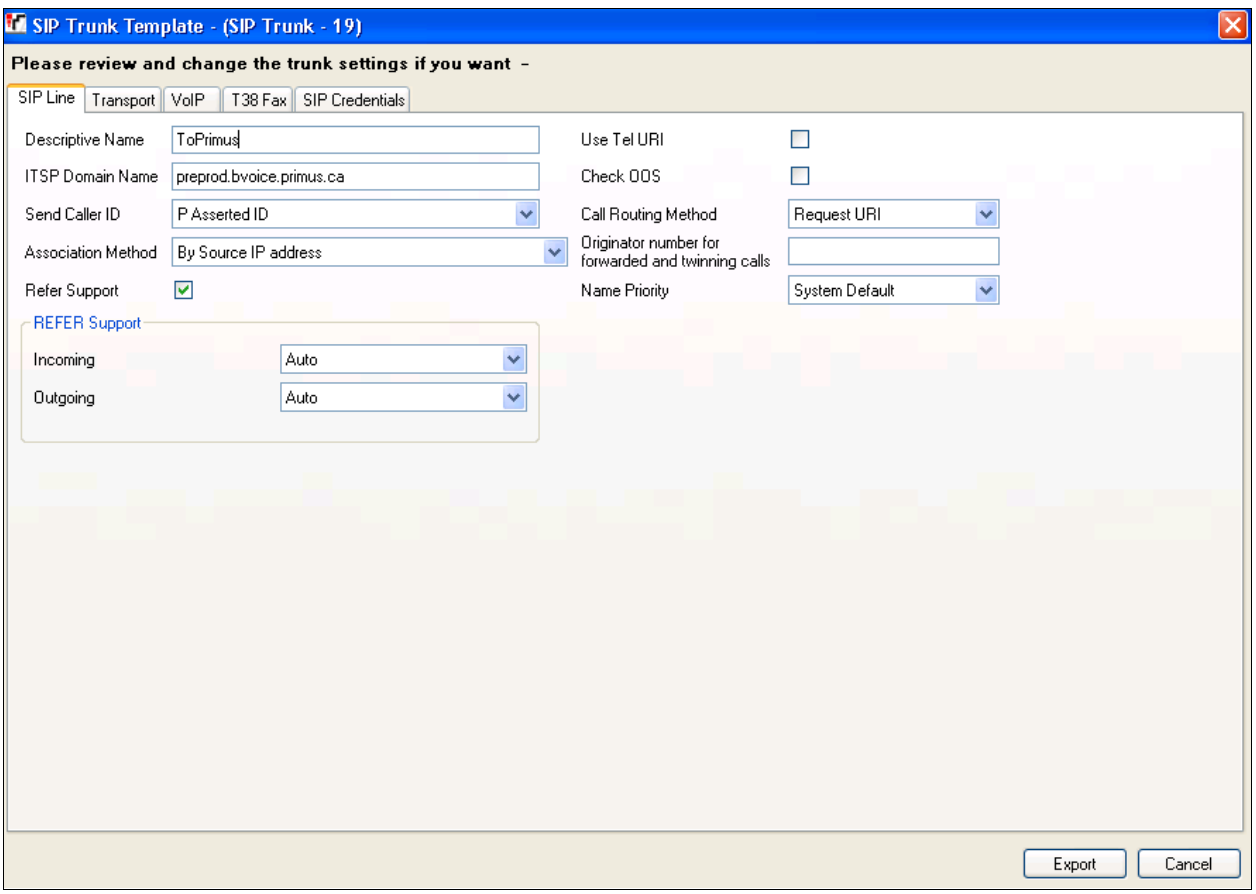
9.2. Generate a SIP Trunk Template

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

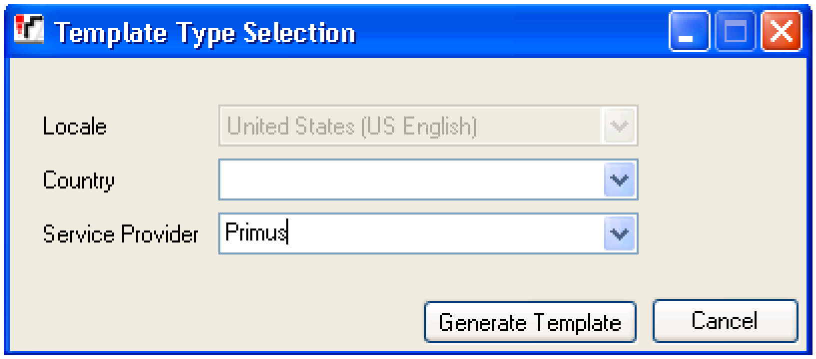
1. Select the SIP trunk under line and right click on the SIP line number for which the SIP trunk template is to be generated and then click **Generate SIP Trunk Template**.



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



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



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.

```
&lt;?xml version="1.0" encoding="utf-8" ?&gt;
- &lt;Template xmlns="urn:SIPTrunk-schema"&gt;
&lt;TemplateType&gt;SIPTrunk&lt;/TemplateType&gt;
&lt;Version&gt;20120604&lt;/Version&gt;
&lt;SystemLocale&gt;enu&lt;/SystemLocale&gt;
&lt;DescriptiveName&gt;ToPrimus&lt;/DescriptiveName&gt;
&lt;ITSPDomainName&gt;preprod.bvoice.primus.ca&lt;/ITSPDomainName&gt;
&lt;SendCallerID&gt;CallerIDPAID&lt;/SendCallerID&gt;
&lt;ReferSupport&gt;true&lt;/ReferSupport&gt;
&lt;ReferSupportIncoming&gt;2&lt;/ReferSupportIncoming&gt;
&lt;ReferSupportOutgoing&gt;2&lt;/ReferSupportOutgoing&gt;
&lt;RegistrationRequired&gt;false&lt;/RegistrationRequired&gt;
&lt;UseTelURI&gt;false&lt;/UseTelURI&gt;
&lt;CheckOOS&gt;false&lt;/CheckOOS&gt;
&lt;CallRoutingMethod&gt;1&lt;/CallRoutingMethod&gt;
&lt;OriginatorNumber /&gt;
&lt;AssociationMethod&gt;SourceIP&lt;/AssociationMethod&gt;
&lt;LineNamePriority&gt;SystemDefault&lt;/LineNamePriority&gt;
&lt;ITSPProxy&gt;xx.xx.xx.xx&lt;/ITSPProxy&gt; (note here that the service provider IP address is
  hidden for security reasons)
&lt;LayerFourProtocol&gt;SipUDP&lt;/LayerFourProtocol&gt;
&lt;SendPort&gt;5060&lt;/SendPort&gt;
&lt;ListenPort&gt;5060&lt;/ListenPort&gt;
&lt;DNSServerOne&gt;0.0.0.0&lt;/DNSServerOne&gt;
&lt;DNSServerTwo&gt;0.0.0.0&lt;/DNSServerTwo&gt;
&lt;CallsRouteViaRegistrar&gt;true&lt;/CallsRouteViaRegistrar&gt;
&lt;SeparateRegistrar /&gt;
&lt;CompressionMode&gt;AUTOSELECT&lt;/CompressionMode&gt;
&lt;UseAdvVoiceCodecPrefs&gt;false&lt;/UseAdvVoiceCodecPrefs&gt;
```

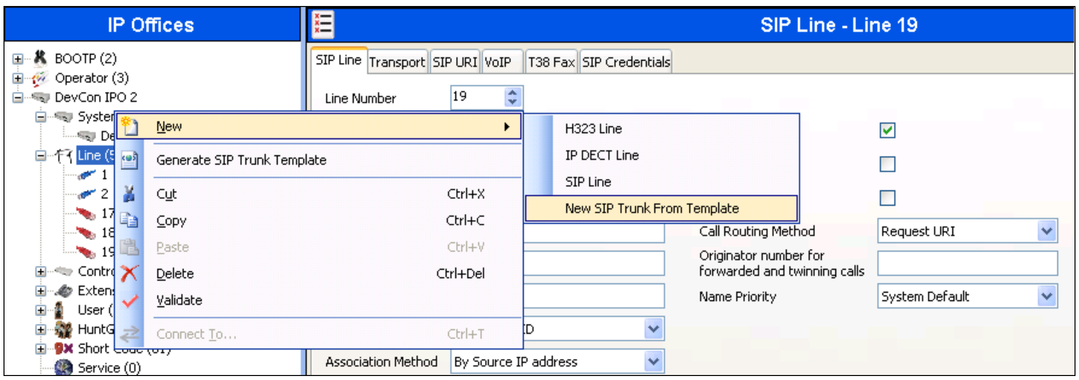
```
<CallInitiationTimeout>4</CallInitiationTimeout>
<DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
<VoipSilenceSupression>false</VoipSilenceSupression>
<ReinviteSupported>true</ReinviteSupported>
<FaxTransportSupport>FOIP_T38</FaxTransportSupport>
<UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
<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**.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.