Avaya Solution & Interoperability Test Lab

Application Notes for Configuring XO Communications SIP Trunking with Avaya IP Office 9.0 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider XO Communications and Avaya IP Office 9.0.

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

XO Communications is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider XO Communications and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 9.0, Avaya embedded Voicemail, Avaya IP Office Softphone, Avaya H.323, Avaya SIP, digital and analog endpoints.

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

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Avaya IP Office connecting to XO Communications using SIP Trunking service.
This configuration (shown in **Figure 1**) was used to exercise the features and functionality tests listed in **Section 2.1**. Note: NAT devices added between Avaya IP Office and the XO Communications network should be transparent to the SIP signaling.

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

2.1. Interoperability Compliance Testing

A simulated enterprise site with Avaya IP Office was connected to XO Communications SIP Trunking service. To verify SIP trunking interoperability, the following features and functionality were exercised during the interoperability compliance test:

- Incoming PSTN calls to various phone types. Phone types included H.323, SIP, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types. Phone types included H.323, SIP, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls from/to the Avaya IP Office Softphone.
- Inbound and outbound long hold time call stability.
- Various call types including: local, long distance, international call, inbound toll-free, outbound toll-free, operator assisted call, 411 and 911 services.
- Codec G.729A and G.711U.
- Caller number/ID presentation.

- Privacy requests (i.e., caller anonymity) and Caller ID restriction for inbound and outbound calls.
- DTMF transmission using RFC 2833.
- Voicemail navigation for inbound and outbound calls.
- Telephony features such as hold and resume, transfer, and conference.
- Use of SIP REFER for call transfer to PSTN.
- FAX using T.38 Fallback or G.711 Pass Through.
- Off-net call forwarding (XO Communications supports Diversion Header).
- Twinning to mobile phones on inbound calls.

2.2. Test Results

Interoperability testing of XO Communications SIP Trunking was completed with successful results for all test cases.

2.3. Support

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

For technical support on the XO Communications system, please visit: http://www.xo.com

3. Reference Configuration

Figure 1 below illustrates the test configuration. The test configuration shows an enterprise site connected to XO Communications SIP Trunking service through the public IP network. For confidentiality and privacy purposes, actual public IP addresses used in this testing have been masked out and replaced with fictitious IP addresses throughout the document.

Located at the enterprise site is an Avaya IP Office 500v2 with the MOD DGTL STA16 expansion module which provides connections for 16 digital stations to the PSTN, and the extension PHONE 8 card which provides connections for 8 analog stations to the PSTN as well as 64-channel VCM (Voice Compression Module) for supporting VoIP codecs. The voicemail service is embedded on Avaya IP Office. The LAN2 port of Avaya IP Office is connected to the public IP network. Endpoints include an Avaya 9600 Series IP Telephone (with H.323 firmware), an Avaya 1100 Series IP Telephone (with SIP firmware), an Avaya 1408D Digital Telephones, an Avaya Analog Telephone and an Avaya IP Office Softphone H.323. A separate Windows XP PC runs Avaya IP Office Manager to configure and administer Avaya IP Office.

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

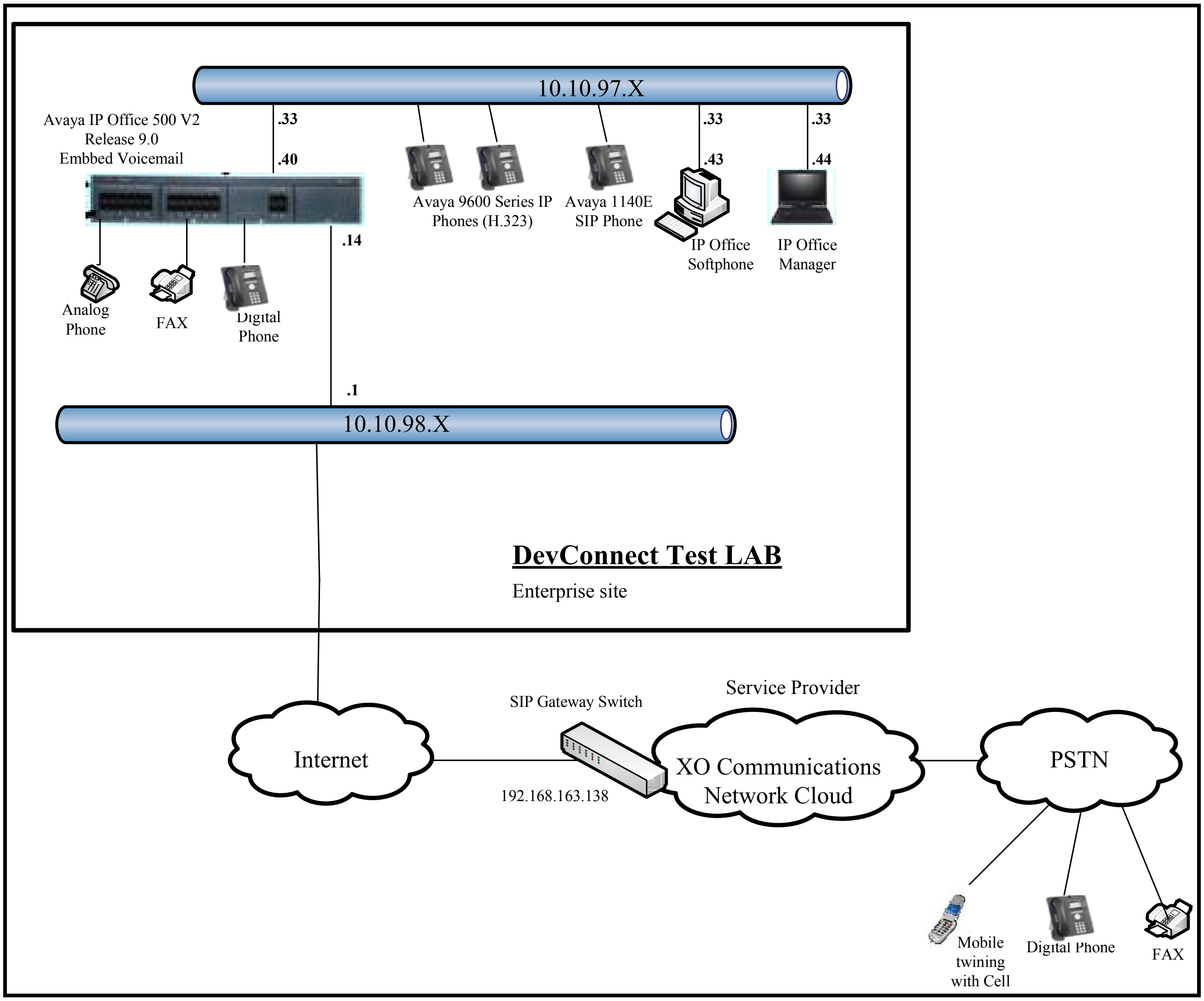


Figure 1: Test Configuration for Avaya IP Office with XO Communications SIP Trunking Service

For the purposes of the compliance test, Avaya IP Office users dialed a short code of 5 + N digits to send digits across the SIP trunk to XO Communications. The short code of 5 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to XO Communications. For calls within the North American Numbering Plan (NANP), the user would dial 11 (1 + 10) digits. Thus for these NANP calls, Avaya IP Office would send 11 digits in the Request URI and the To field of an outbound SIP INVITE message. It was configured to send 10 digits in the From field. For inbound calls, XO Communications SIP Trunking sent 10 digits in the Request URI and the To field of inbound SIP INVITE messages.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and 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/firmware were used for the sample configuration provided:

Avaya Telephony Components	
Equipment	**Release**
Avaya IP Office 500v2	9.0 (Build829)
Avaya IP Office DIG DCP*16 V2	9.0 (Build829)
Avaya IP Office Ext Card Phone 8	9.0 (Build829)
Avaya IP Office Manager	9R
Avaya 9641G IP Telephone (H.323)	2.2.09
Avaya 9640G IP Telephone (H.323)	3.1.04S
Avaya 1140E IP Telephone (SIP)	SIP1140e.04.03.12
Avaya Digital Telephones (1408D)	N/A
Avaya Symphony 2000 Analog Telephone	N/A
Avaya IP Office Softphone	3.2.3.49 68975
HP Officejet 4500 (fax)	N/A
XO Communications Components	
Equipment	**Release**
Broadsoft Softswitch	Rel_18.sp1_1.890
Media Gateway SONUS GSX9000	V07.03.01 F009

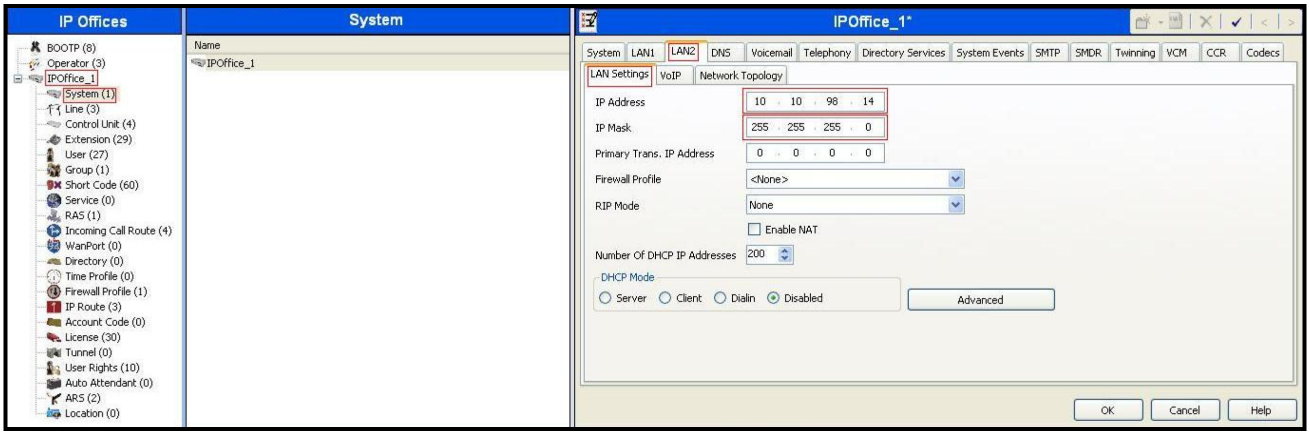
Note: Testing was performed with IP Office 500 v2 R9.0, but it also applies to IP Office Server Edition R9.0. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R9.0 to support analog or digital endpoints or trunks. IP Office Server Edition does not support TAPI Wave or Group Voicemail.

5. Configure IP Office

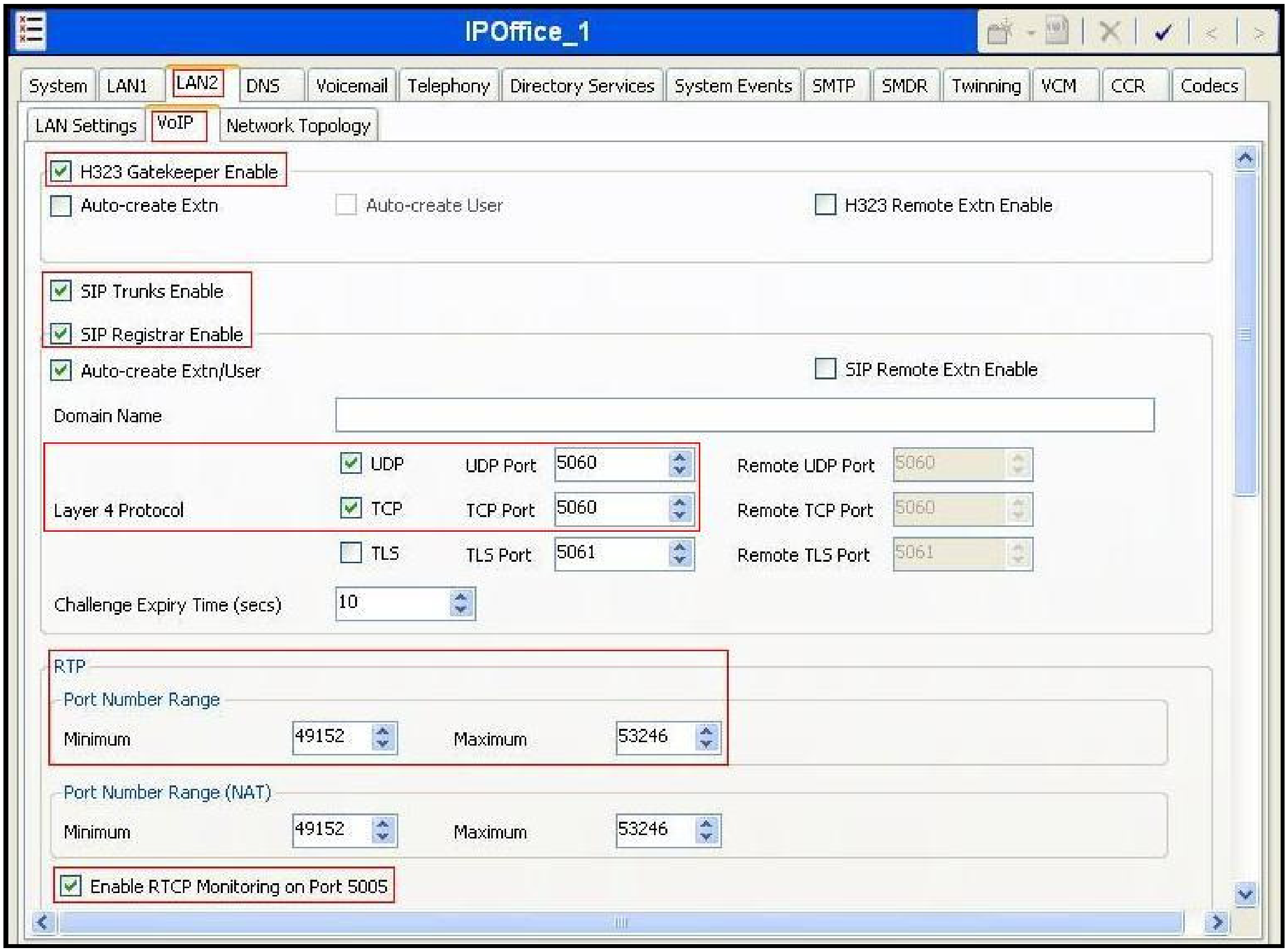
This section describes the Avaya IP Office configuration to support connectivity to XO Communications SIP Trunking service. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials. A management window will appear similar to the one shown in the next section. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details pane on the right side. These panes will be referenced throughout the Avaya IP Office configuration. Proper licensing as well as standard feature configurations that are not directly related to the interface with the service provider (such as the LAN interface to the enterprise site and IP Office Softphone support) is assumed to be already in place.

5.1. LAN2 Settings

In the sample configuration, the **IPOffice_1** was used as the system name and the LAN2 port was used to connect to XO Communications SIP Trunking service. To access the LAN2 settings, first navigate to **System (1) → IPOffice_1** in the Navigation and Group Panes and then navigate to the **LAN2→ LAN Settings** tab in the Details Pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office LAN port. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements.

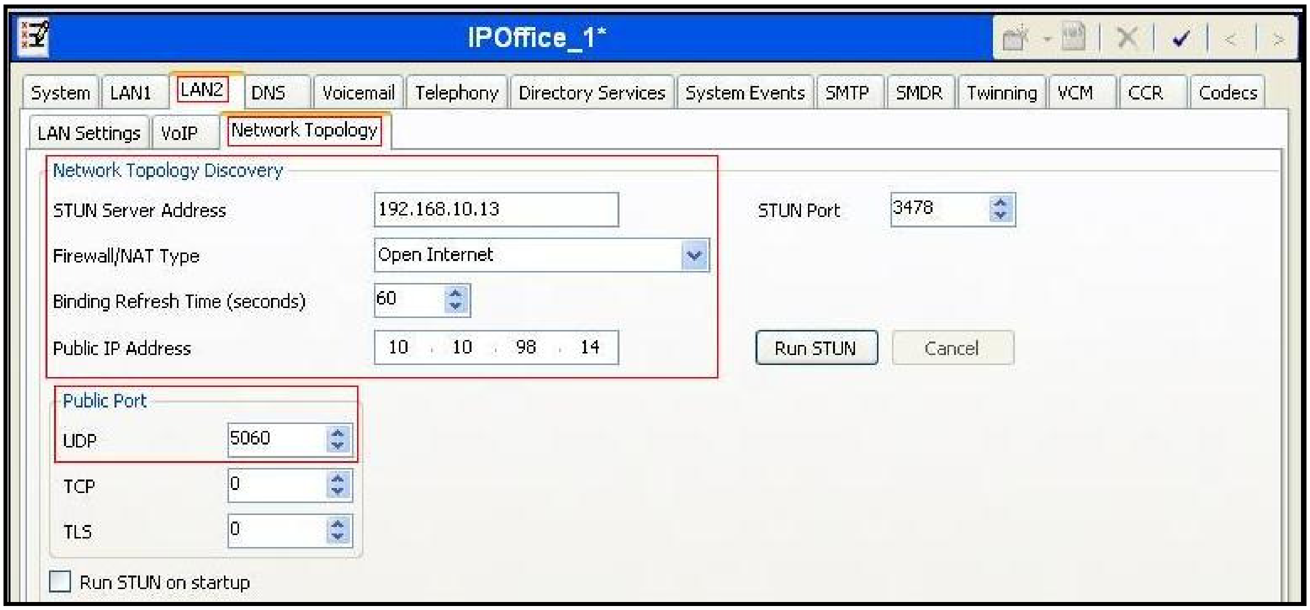


Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the 9600-Series IP Telephones used in the sample configuration. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to XO Communications. The **SIP Registrar Enable** box is checked to allow Avaya IP Office Softphone usage. The **Layer 4 Protocol** use **UDP** with port **5060** and **TCP** with port **5060**. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using **LAN2**. The **Enable RTCP Monitoring On Port 5005** is checked. The specific values used for the compliance test are shown in the example below. All other parameters should be set according to customer requirements**.**



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

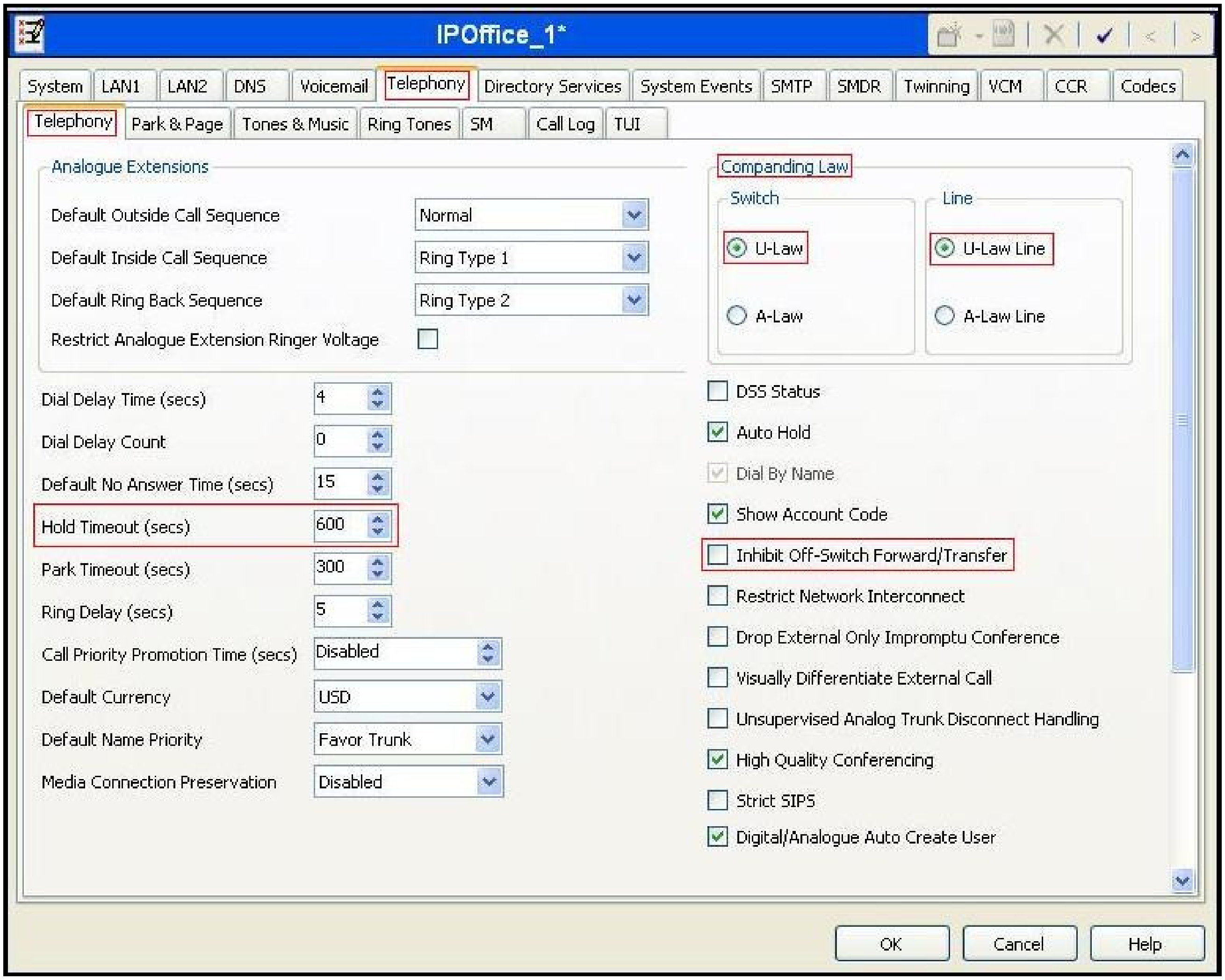
- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to **Open Internet**. With this configuration, STUN will not be used.
- Set the **Binding Refresh Time (seconds)** to **60**. This value is used as one input to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider. (Refer to **Section 5.10**)
- Set **Public IP Address** to the IP address of the Avaya IP Office LAN2 port.
- Set **Public Port** for **UDP** as **5060**.
- All other parameters should be set according to customer requirements.



In the compliance test, the LAN1 interface was used to connect the Avaya IP Office to the enterprise site IP network. The LAN1 interface configuration is not directly relevant to the interface with XO Communications SIP Trunking service, and therefore is not described in these Application Notes.

5.2. System Telephony Settings

Navigate to **System (1) → IPOffice_1** in the Navigation and Group Panes and then navigate to the **Telephony → Telephony** tab in the Details Pane. Choose the **Companding Law** typical for the enterprise location. For North America, **U-Law** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfers to the PSTN via the service provider across the SIP trunk. Set **Hold Timeout (secs)** to **600**.



5.3. System Codec Settings

Navigate to **System (1)** → **IPOffice_1** in the Navigation and Group Panes and then navigate to the **Codecs** tab in the Details Pane. Choose the **RFC2833 Default Payload** as IP Office default of **101**. Select codecs **G.711 ULAW 64K** and **G.729(a) 8K CS-ACELP** that XO Communications support.

5.4. Twinning Calling Party Settings

When using twinning, the calling party number displayed on the twinned phone is controlled by two parameters. These parameters only affect twinning and do not impact the messaging or operation of other redirected calls such as forwarded calls. The first parameter is the **Send original calling party information for Mobile Twinning** box on the **Twinning** tab, as shown below. The second parameter is the **Send Caller ID** parameter on the **SIP Line** form (shown in **Section 5.5**).

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

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

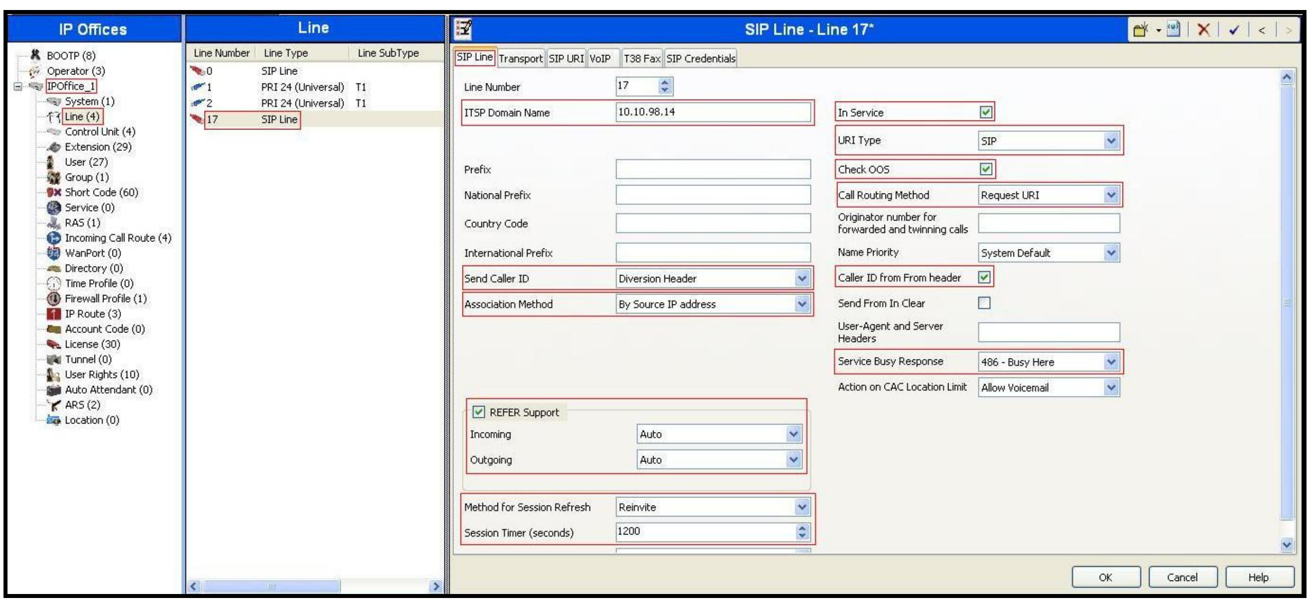
If this option is unchecked, the value sent in the SIP From header is determined by the setting of the second parameter mentioned above.

- For the compliance test, the **Send original calling party information for Mobile Twinning** box in the **System→Twinning** tab was unchecked. The value sent in the SIP From header is determined by the setting of the **Send Caller ID** parameter on the **SIP Line** form**.**

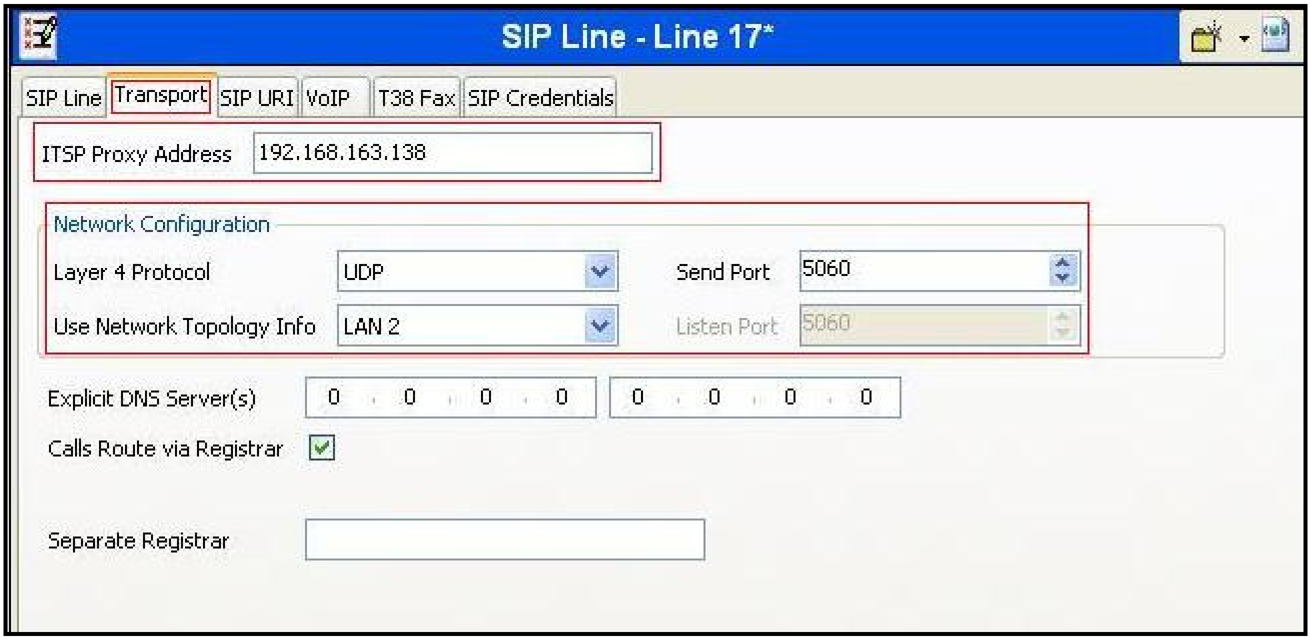
5.5. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and XO Communications SIP Trunking service. To create a SIP line, begin by navigating to **Line** in the left Navigation Pane, then right-click in the Group Pane and select **New → SIP Line** (Not Shown). On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

- Set **ITSP Domain Name** to the enterprise domain so that IP Office uses this domain as the host portion of the SIP URI in SIP headers such as the From header.
- Check the **In Service** box.
- Set **URI Type** to **SIP**.
- Check the **Check OOS** box. With this option selected, IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- Set **Call Routing Method** to **Request URI**.
- Check **Caller ID from From header** box.
- Set **Send Caller ID** to **Diversion Header**. For the compliance test, this parameter was used for call forwarding and it was used in Mobility Twinning since **Send original calling party information for Mobile Twinning** is not optioned in **Section 5.4**.
- Set **Association Method** to **By Source IP address**.
- Set **Service Busy Response** as **486 – Busy Here**.
- Check **REFER Support** to enable SIP REFER for call transfers. Select the default values of "**Auto**" for **Incoming** and **Outgoing** to use of SIP REFER effectively.
- Set **Method for Session Refresh** to **Reinvite** with **Session Timer (seconds)**: **1200**. If the Method for Session Refresh set to the default value "On Demand", then IP Office does not initiate Session Timer and only supports it if it is initiated by the other side.
- Default values may be used for all other parameters.

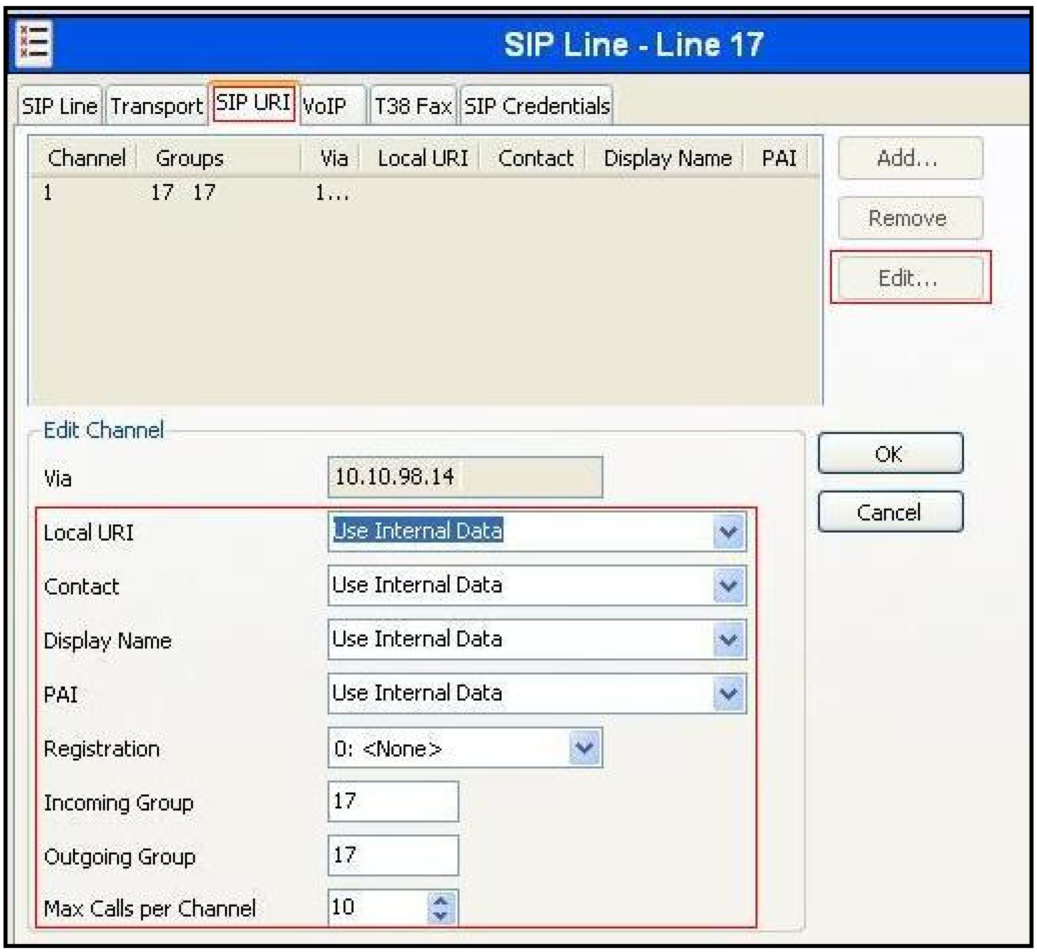


Select the **Transport** tab. The **ITSP Proxy Address** is set to the XO Communications SIP proxy gateway IP address provided by XO Communications. As shown in **Figure 1**, this IP address is **192.168.163.138**. In the **Network Configuration** area, **UDP** is selected as the **Layer 4 Protocol**, and the **Send Port** is set to the port number provided by XO Communications, in this case the well known SIP port of **5060** was used. The **Use Network Topology Info** parameter is set to **LAN 2**. This associates the SIP Line with the parameters in the **System → LAN2 → Network Topology** tab. Other parameters retain default values in the screen below.



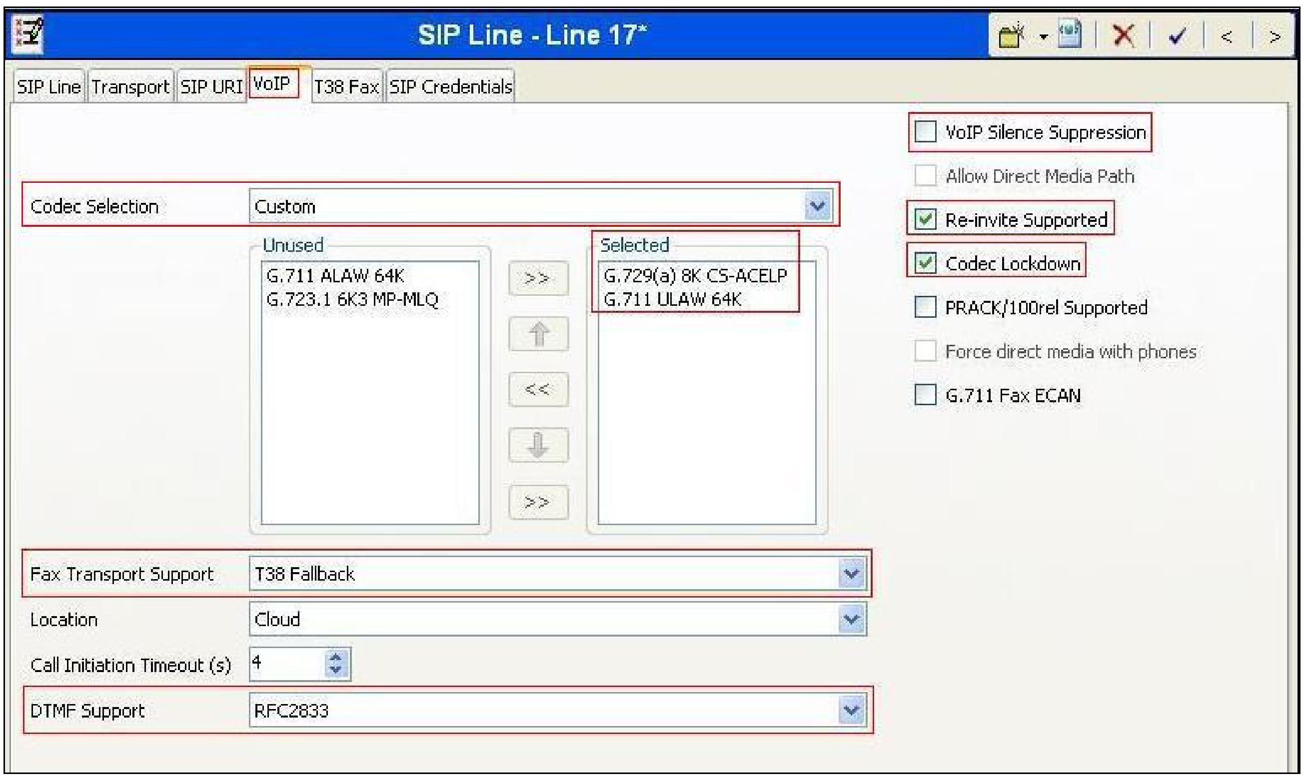
A SIP URI entry must be created to match each incoming number that Avaya IP Office will accept on this line. Select the **SIP URI** tab; click the **Add** button and the **New Channel** area will appear at the bottom of the pane (Not shown). To edit an existing entry, click an entry in the list at the top, and click the **Edit…** button. In the example screen below, a previously configured entry is edited. For the compliance test, a single SIP URI entry was created that matched any DID number assigned to an Avaya IP Office user. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact**, and **Display Name** to **Use Internal Data**. This setting allows calls on this line which SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.7**.
- Set **PAI** to **Use Internal Data**. With this setting IP Office will populate the SIP P-Asserted-Identity header on outgoing calls with the data set in the **SIP** tab of the call initiating **User** as shown in **Section 5.7**.
- Set **Registration** to **0: <None>** since XO SIP Trunking service does not require registration.
- Associate this line with an incoming line group in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. For the compliance test, a new incoming and outgoing group **17** was defined that only contains this line (line 17).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.



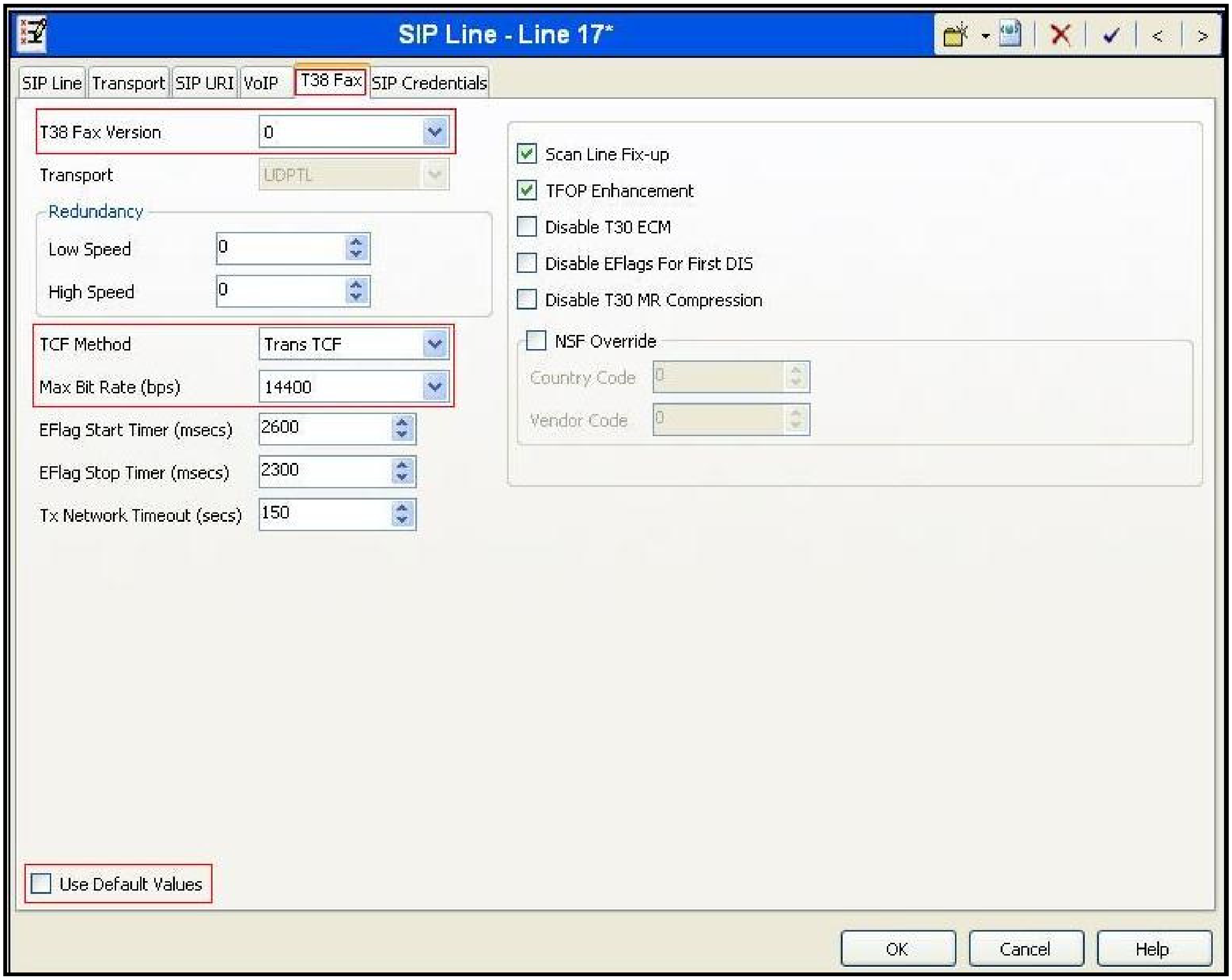
Select the **VoIP** tab to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- The **Codec Selection** can be selected by choosing **Custom** from the pull-down menu, allowing an explicit ordered list of codecs to be specified. Selecting **G.729(a) 8K CS – ACELP** and **G.711 ULAW 64K** codecs causes Avaya IP Office to include these codecs, which are supported by the XO Communications SIP Trunking service, in the Session Description Protocol (SDP) offer, in that order.
- Uncheck the **VoIP Silence Suppression** box.
- Check the **Re-invite Supported** box.
- Check **Codec Lockdown**.
- Set **Fax Transport Support** to **T38 Fallback** or **G.711** from the pull-down menu. XO Communications support both Fax T.38 Fallback and Fax G.711 Pass Through.
- Set the **DTMF Support** to **RFC2833** from the pull-down menu. 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**.**



Select the **T38 Fax** tab to set the T38 Fax parameters of the SIP line as shown below:

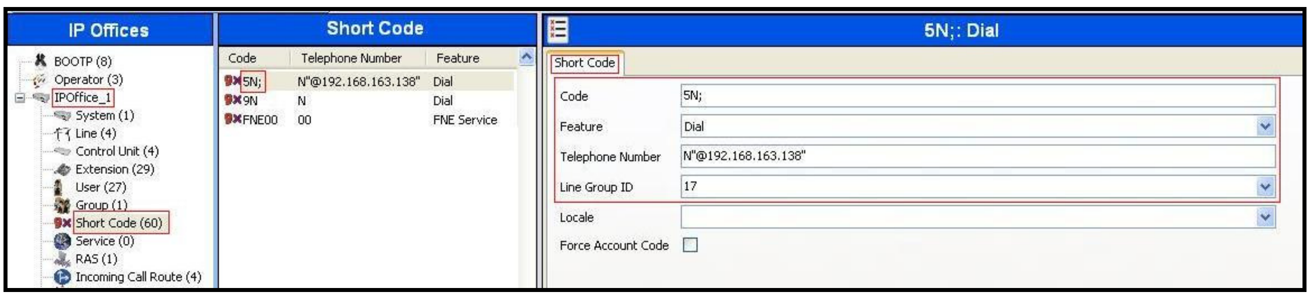
- Uncheck **Use Default Values** at the bottom of the screen.
- Set **T38 Fax Version** to **0**. XO SIP Trunking supports T.38 Fax version 0.
- Set **TCF Method** to **Trans TCF**.
- Set **Max Bit Rate (bps)** to **14400**, the highest fax bit rate that XO supports for T.38 faxing.
- Default values may be used for all other parameters.



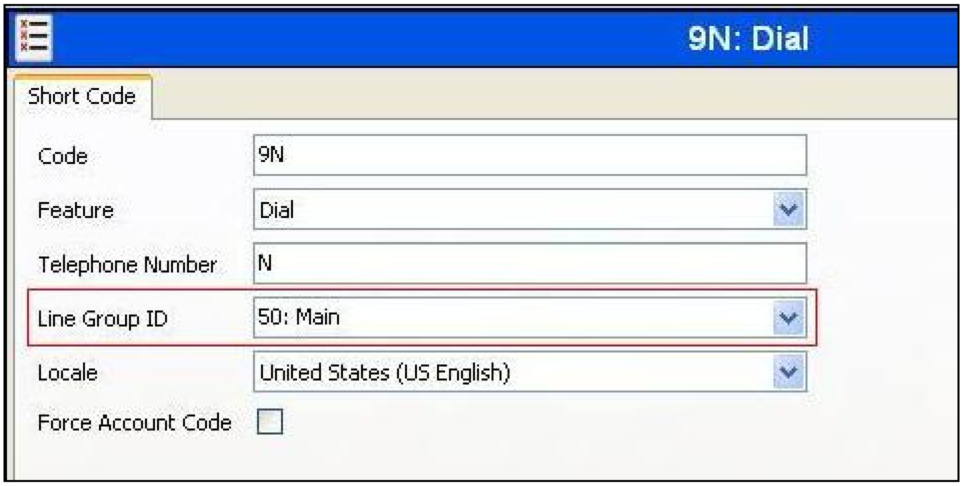
5.6. Short Code

Define a short code to route outbound traffic to the SIP line. To create a short code, select **Short Code** in the left Navigation Pane, then right-click in the Group Pane and select **New** (Not Shown). On the **Short Code** tab in the Details Pane, configure the parameters for the new short code to be created. The screen below shows the details of the previously administered "5N;" short code used in the test configuration.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **5N;**, this short code will be invoked when the user dials 5 followed by any number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N"@192.168.163.138"**. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. The value **N** represents the number dialed by the user. The host part following the "@" is the domain of the service provider network.
- Set the **Line Group ID** to the **Outgoing Group 17** defined on the **SIP URI** tab on the **SIP Line** in **Section 5.5**. This short code will use this line group when placing the outbound call.

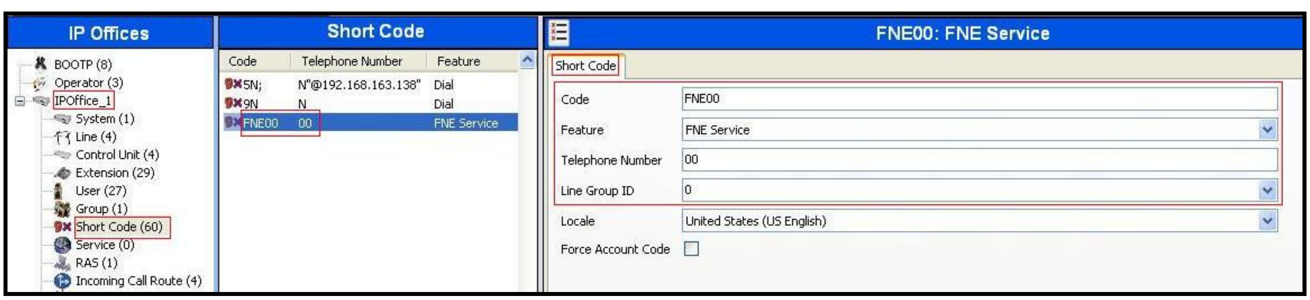


The simple “5N;” short codes illustrated above does not provide a means of alternate routing if the configured SIP Line is out of service or temporarily not responding. When alternate routing options and/or more customized analysis of the digits following the short code are desired, the Automatic Route Selection (ARS) feature may be used. In the following example screen, the short code **9N** is illustrated for access to ARS. When the Avaya IP Office user dials 9 plus any number N, rather than being directed to a specific **Line Group ID**, the call is directed to **Line Group ID 50: Main**, configurable via ARS. See **Section 5.9** for example ARS route configuration for **50: Main** as well as a backup route.



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

- For **Code** field, enter FNE feature code as **FNE00** for **Dial Tone**.
- Set **Feature** to **FNE Service**.
- Set **Telephone Number** to **00**.
- Set **Line Group ID** to **0**.
- Default values may be used for other parameters.

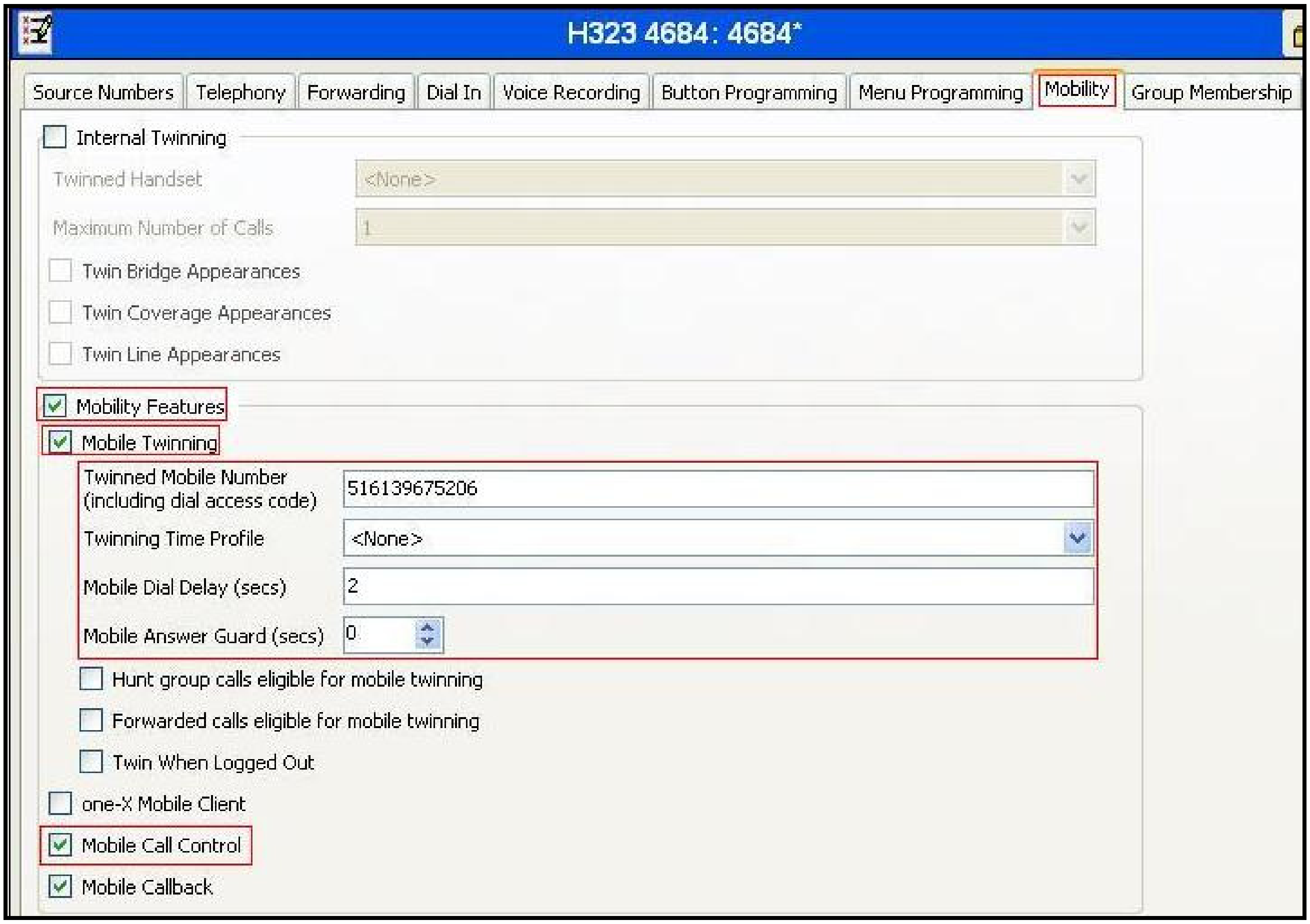


5.7. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.5**. To configure these settings, first select **User** in the left Navigation Pane, then select the name of the user to be modified in the center Group Pane. In the example below, the name of the user is **H323 4684**. Select the **SIP** tab in the Details Pane.

The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From header for outgoing SIP trunk calls. They also allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.5**). The example below shows the settings for user **H323 4684**. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise provided by XO Communications. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network.

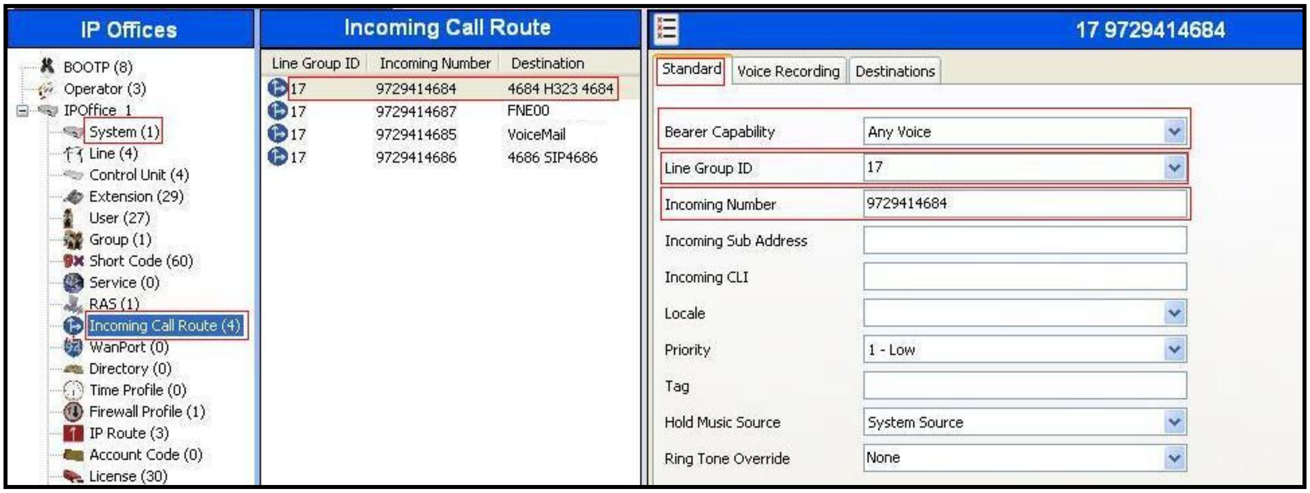
One of the H.323 IP Phones at the enterprise site uses the Mobile Twinning feature. The following screen shows the **Mobility** tab for **User H323 4684**. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case **516139675206**. Check **Mobile Call Control** to allow incoming calls from mobility extension to access FNE00 (see **Section 5.6**). Other options can be set according to customer requirements.



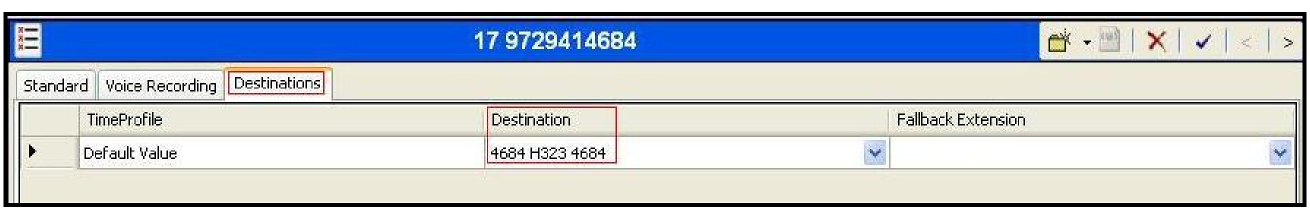
5.8. Incoming Call Route

An Incoming Call Route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by service provider. To create an incoming call route, select **Incoming Call Route** in the left Navigation Pane, then right-click in the center Group Pane and select **New** (Not Shown). On the **Standard** tab of the Details Pane, enter the parameters as shown below:

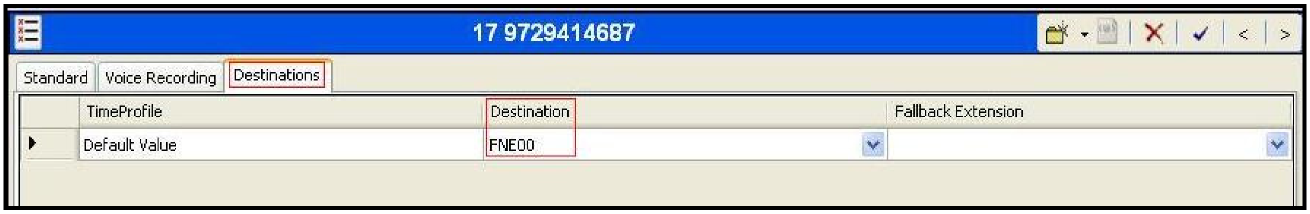
- Set the **Bearer Capability** to **Any Voice**.
- Set the **Line Group ID** to the **Incoming Group 17** defined on the **SIP URI** tab on the **SIP Line** in **Section 5.5**.
- Set the **Incoming Number** to the incoming number on which this route should match.
- Default values can be used for all other fields.



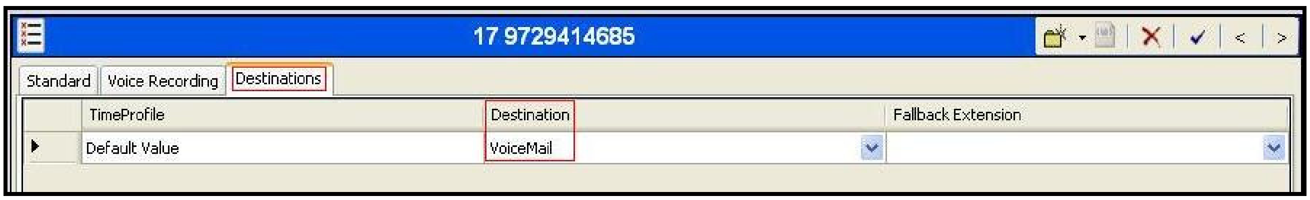
On the **Destination** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to **9729414684** on line 17 are routed to extension **4684 H323 4684**.



The incoming calls to DID number **9729414687** were configured to access **FNE00**. The **Destination** was appropriately defined as **FNE00** as below screenshot:



The incoming calls to DID number **9729414685** were configured to access **VoiceMail**. The **Destination** was appropriately defined as **VoiceMail** as below screenshot:



5.9. ARS and Alternate Routing

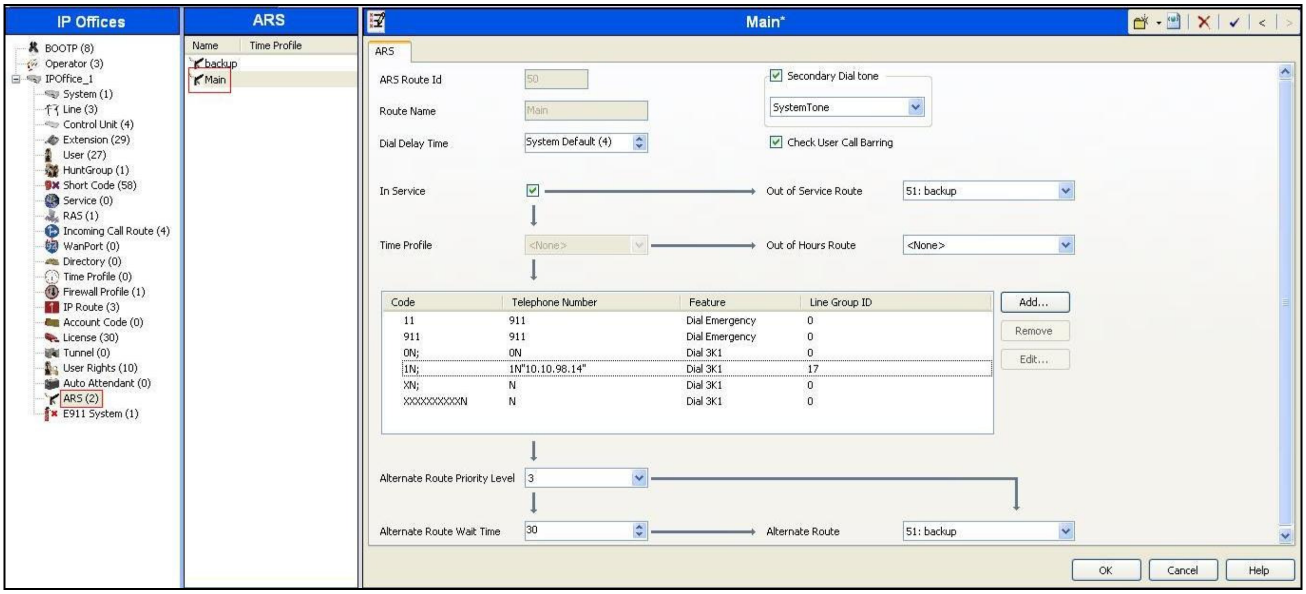
While detailed coverage of Automatic Route Selection (ARS) is beyond the scope of these Application Notes, this section includes basic ARS screen illustrations and considerations. ARS is shown here mainly to illustrate alternate routing should the SIP Line be out of service or temporarily not responding.

Optionally, ARS can be used rather than the simple **5N;** short code approach documented in **Section 5.6**. With ARS, a secondary dial tone can be provided after the access code, time-based routing criteria can be introduced, and alternate routing can be specified so that a call can re-route automatically if the primary route or outgoing line group is not available. Although not shown in this section, ARS also facilitates more specific dialed telephone number matching, enabling immediate routing and alternate treatment for different types of numbers following the access code. For example, if all 1+10 digit calls following an access code should use the SIP Line preferentially, but other local or service numbers following the access code should prefer a different outgoing line group, ARS can be used to distinguish these call behaviors.

To add a new ARS route, right-click **ARS** in the Navigation pane, and select **New** (Not Shown). To view or edit an existing ARS route, select **ARS** in the Navigation pane, and select the appropriate route name in the Group pane.

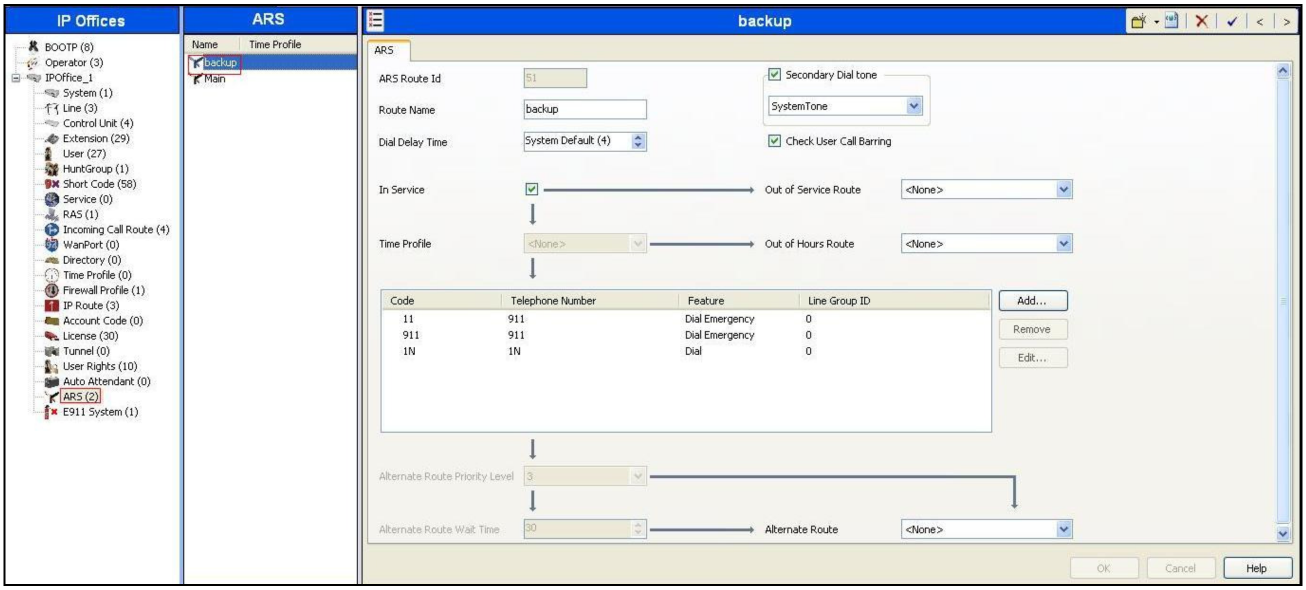
The following screen shows an example ARS configuration for the route named **Main**. The **In Service** parameter refers to the ARS form itself, not the Line Groups that may be referenced in the form. If the **In Service** box is un-checked, calls are routed to the ARS route name specified in the **Out of Service Route** parameter. IP Office short codes may also be defined to allow an ARS route to be disabled or enabled from a telephone. The configurable provisioning of an Out of Service

Route, and the means to manually activate the Out of Service Route can be helpful for scheduled maintenance or other known service-affecting events for the primary route.



Assuming the primary route is in-service, the number passed from the short code used to access ARS (e.g., **9N** in **Section 5.6**) can be further analyzed to direct the call to a specific Line Group ID. Per the example screen above, if the user dialed 9-1-613-967-5205, the call would be directed to Line Group 17, the SIP Line configured and described in these Application Notes. If Line Group 17 cannot be used, the call can automatically route to the route name configured in the **Additional Route** parameter in the lower right of the screen. Since alternate routing can be considered a privilege not available to all callers, IP Office can control access to the alternate route by comparing the calling user's priority to the value in the **Alternate Route Priority Level** field.

The following screen shows an example ARS configuration for the route named **backup** with ARS Route Id 51. Continuing the example, if the user dialed 9-1-613-967-5205, and the call could not be routed via the primary route **50: Main** described above, the call will be delivered to this **backup** route. Per the configuration shown below, the call will be delivered to Line Group 1, using an analog/PRI trunk connecting the Avaya IP Office to the PSTN as a backup connection. In this case, the originally dialed number (sans the short code 9) will be dialed as is through the analog/PRI trunk to the PSTN.



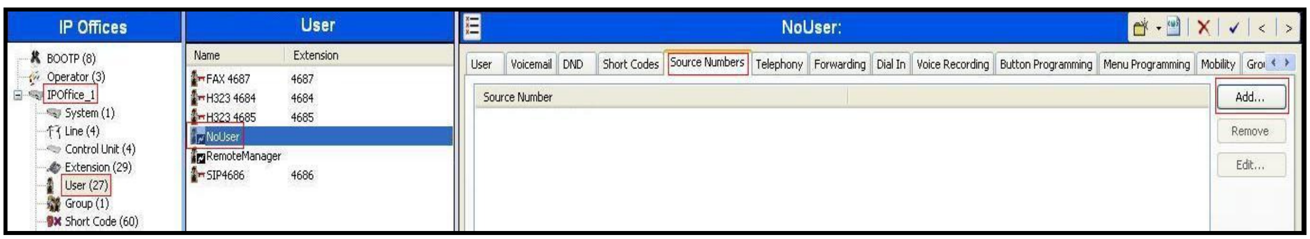
In the testing associated with the configuration, calls were successfully delivered to SIP Line 17 via the primary ARS route **50: Main** or to the analog/PRI trunk via the backup ARS route shown above. When the primary route experiences a network outage, Avaya IP Office successfully routed the call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

5.10. SIP OPTIONS

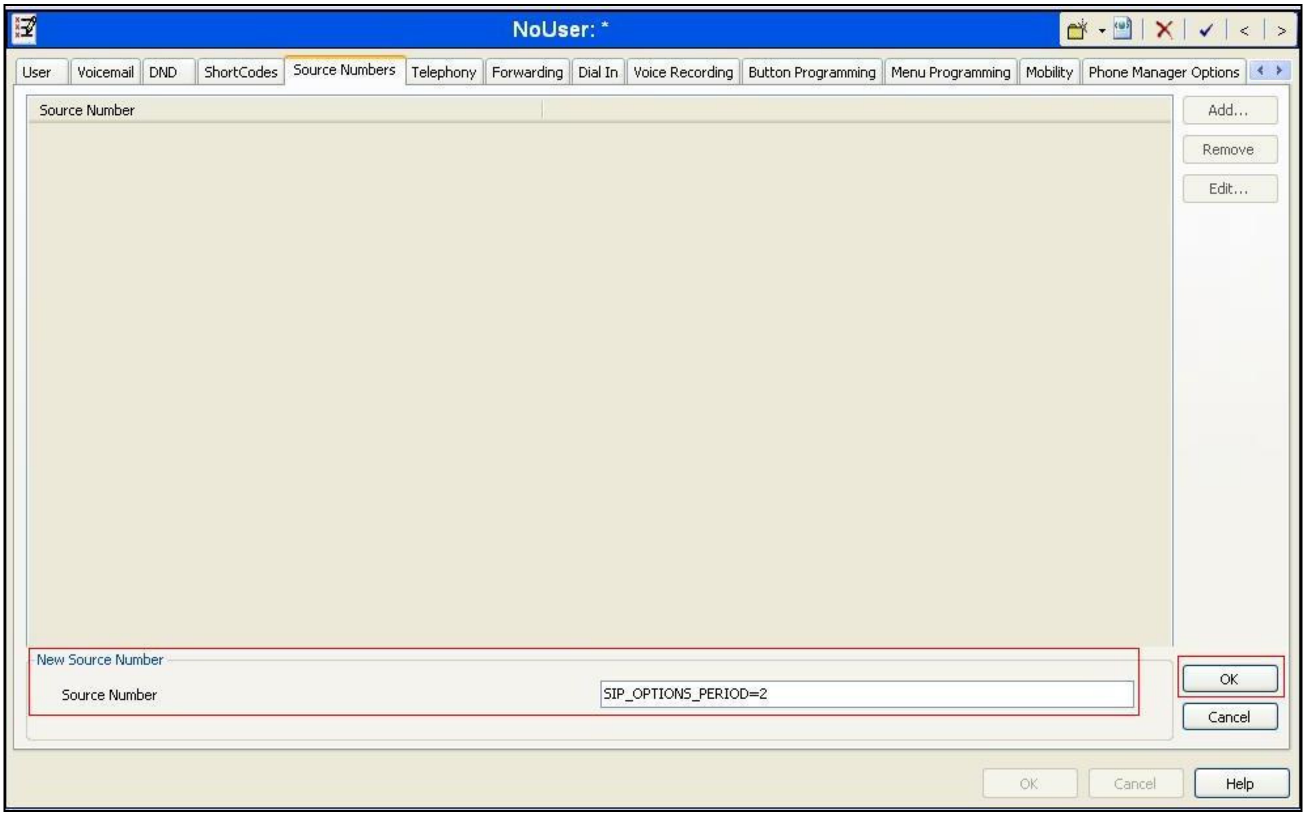
Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.1** and the **SIP_OPTIONS_PERIOD** parameter (in minutes) that can be set on the **Source Numbers** tab of the **NoUser** user. The OPTIONS period is determined in the following manner:

- If no **SIP_OPTIONS_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 44 seconds is used.
- To establish a period less than 42 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 42 secs. The OPTIONS message period will be equal to the **Binding Refresh Time**.
- To establish a period greater than 42 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 42 secs. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD**.

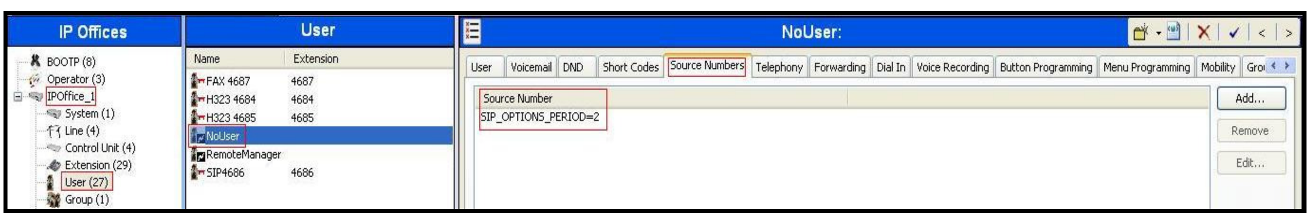
To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User → NoUser** in the Navigation / Group Panes. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



At the bottom of the Details Pane, the **New Source Number** field will appear. Enter **SIP_OPTIONS_PERIOD=X**, where **X** is the desired value in minutes. Click **OK**.



The **SIP_OPTIONS_PERIOD** parameter will appear in the list of Source Numbers as shown below. For the compliance test, an OPTIONS period of 1 minute was desired. The **Binding Refresh Time** was set to **60** seconds (1 minute) in **Section 5.1**. The **SIP_OPTIONS_PERIOD** was set to **2** minutes. Avaya IP Office chose the OPTIONS period as the smaller of these two values (1 minute).



5.11. Save Configuration

Navigate to **File → Save Configuration** in the menu bar at the top of the screen to save the configuration performed in the preceding sections.

6. XO Communications SIP Trunking Configuration

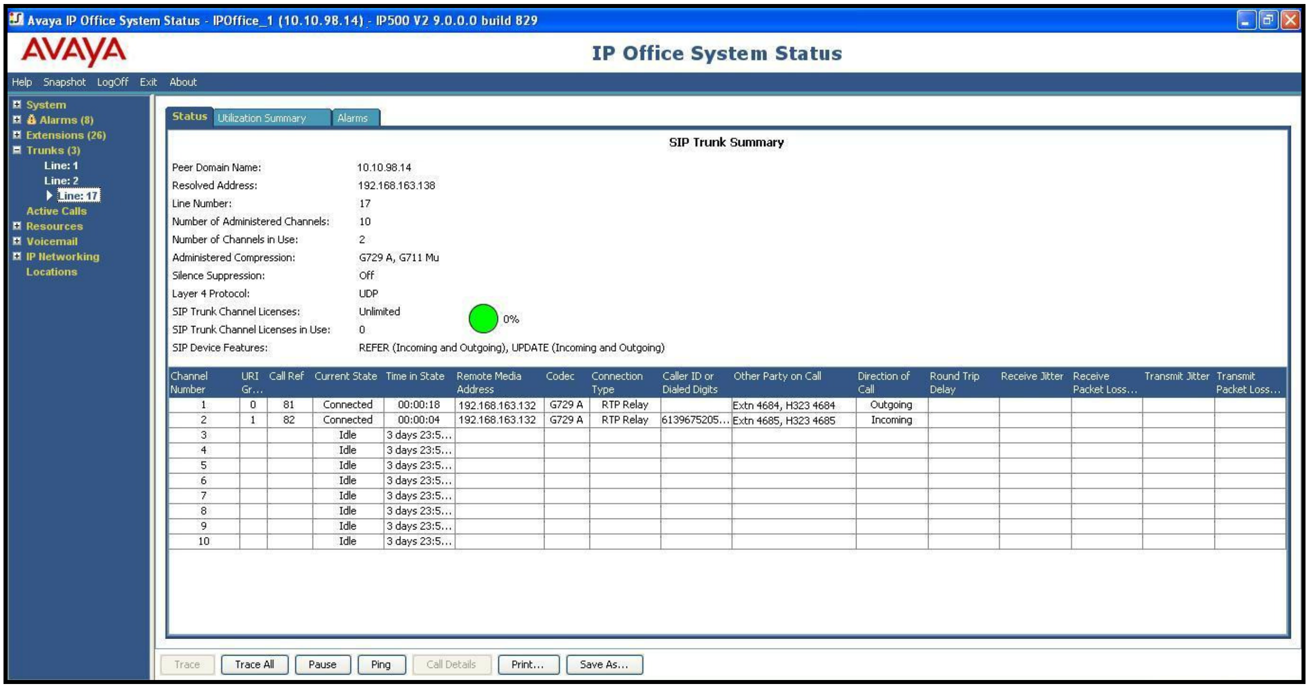
XO Communications is responsible for the configuration of XO Communications SIP Trunking service. The customer must provide the IP address used to reach the Avaya IP Office at the enterprise. XO Communications will provide the customer the necessary information to configure the SIP connection between Avaya IP Office and XO Communications. The provided information from XO Communications includes:

- Fully Qualified Domain Name, IP address and port number used for signaling or media through any security.
- DID numbers.
- XO Communications SIP trunking Specification.

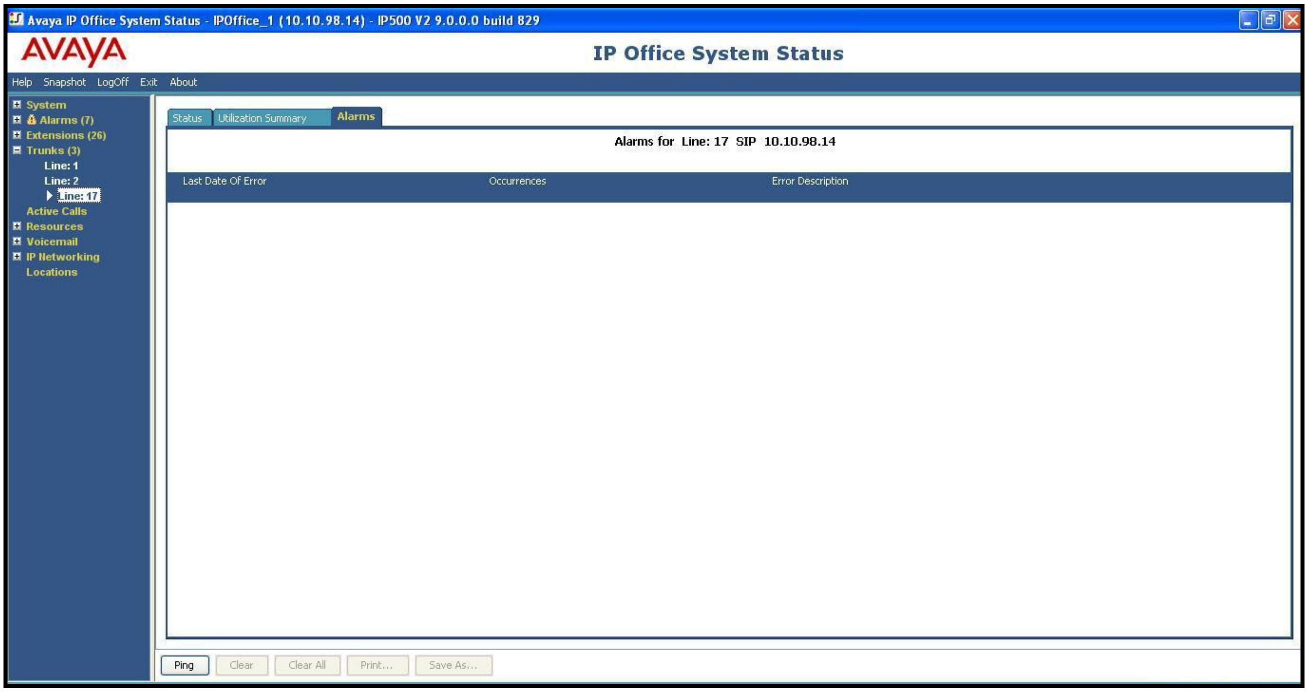
7. Verification Steps

The following steps may be used to verify the configuration:

- Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application from **Start → Programs → IP Office → System Status** on the PC where Avaya IP Office Manager was installed. Select the SIP line of interest from the left pane. On the **Status** tab in the right pane, verify that the **Current State** for each channel (The below screen shot showed an active call at present time).



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



- Verify that a phone connected to the PSTN can successfully place a call to Avaya IP Office with two-way audio.
- Verify that a phone connected to Avaya IP Office can successfully place a call to the PSTN with two-way audio.

8. Conclusion

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

9. Additional References

[1] Avaya IP Office Document Library Release 9.0, Document number 15-604278 Issue 1, September 2013

[2] Avaya IP Office 9.0 Installing IP500/IP500V2, Document number 15-601042 Issue 28g, 11 October 2013

[3] Avaya IP Office Manager Release 9.0, Document number 15-601011 Issue 9.01, 09 September, 2013

[4] Avaya IP Office 9.0 Softphone User Guide (Windows), Issue 07b, 19 August 2013

[5] Avaya IP Office Embedded Voicemail User Guide (IP Office Mode), Document number 15-604067 Issue 12f, 19 August 2013

Product documentation for Avaya products may be found at: http://support.avaya.com. Additional IP Office documentation can be found at:
http://marketingtools.avaya.com/knowledgebase/ipoffice/general/rss2html.php?XMLFILE=manuals.xml&TEMPLATE=pdf_feed_template.html

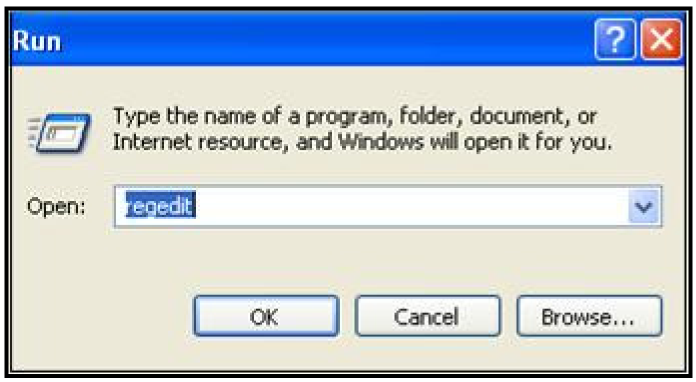
Product documentation for XO Communications SIP Trunking may be found at: http://www.xo.com

Appendix: SIP Line Template

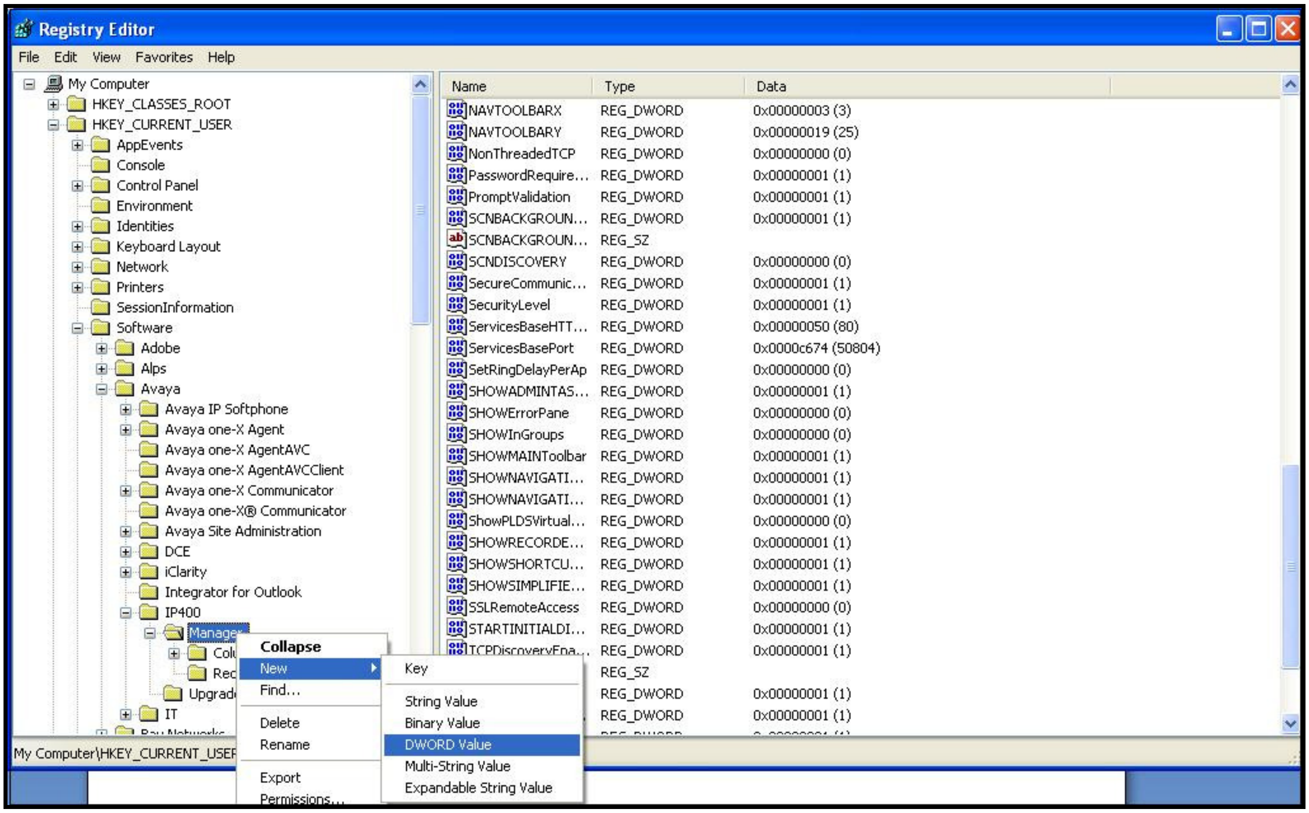
Avaya IP Office Release 9.0 supports a SIP Line Template (in xml format) that can be created from an existing configuration and imported into a new installation to simplify configuration procedures as well as to reduce potential configuration errors.

Note that not all of the configuration information, particularly items relevant to specific installation environment (e.g., the IP address assigned to the WAN interface of the IP Office, the service SIP Proxy IP address provided by XO Communications, etc.), 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 5.5** in these Application Notes as a reference.

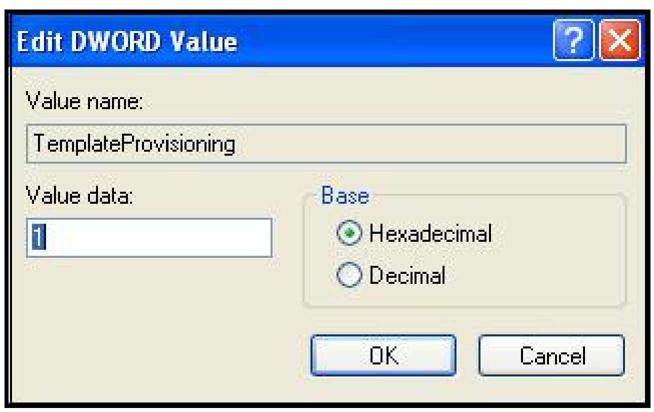
Use the Windows Registry Editor on the PC where Avaya IP Office Manager is installed to add a new **TemplateProvisioning** registry entry. Select **Start → Run**. Enter **regedit** in the **Open** box



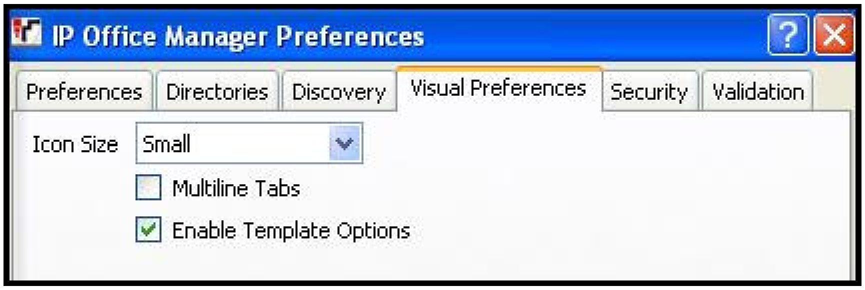
On the Registry Editor, navigate to **HKEY_CURRENT_USER → Software → Avaya →IP400**. Right click on **Manager** and select **New → DWORD Value**.



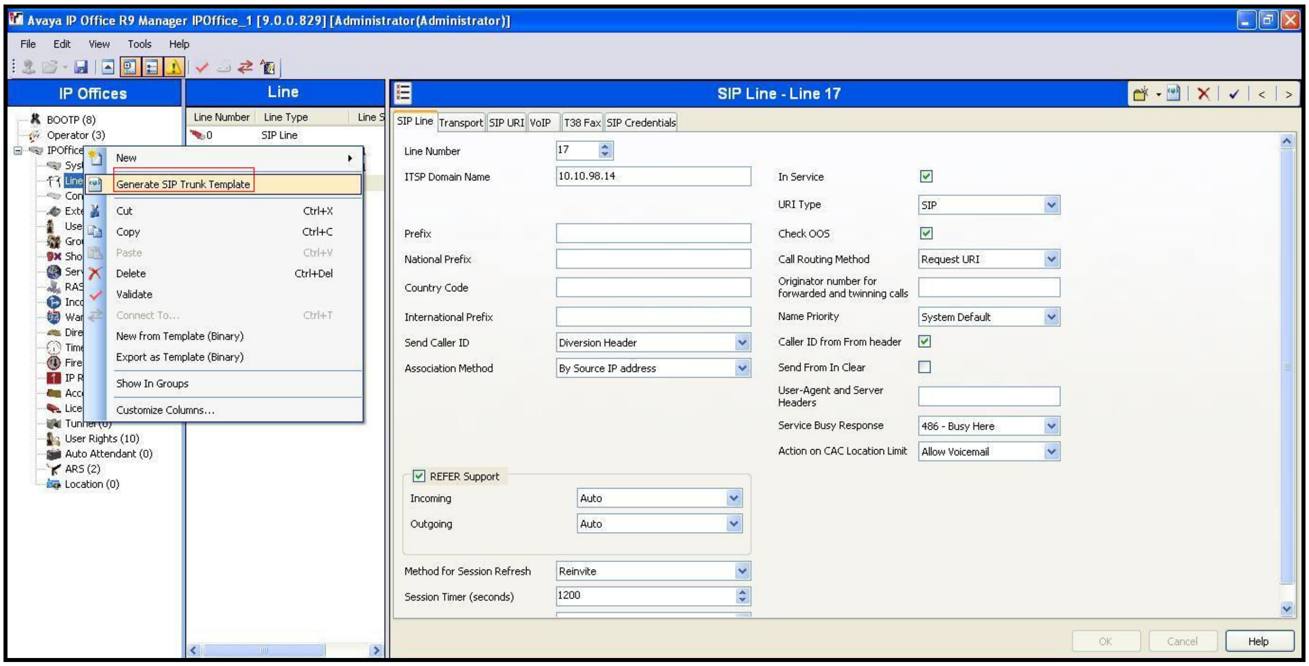
Right click the newly created entry and rename it to **TemplateProvisioning**. Double click the entry and change the value under **Value Data** from "**0**" to "**1**". Restart the PC.



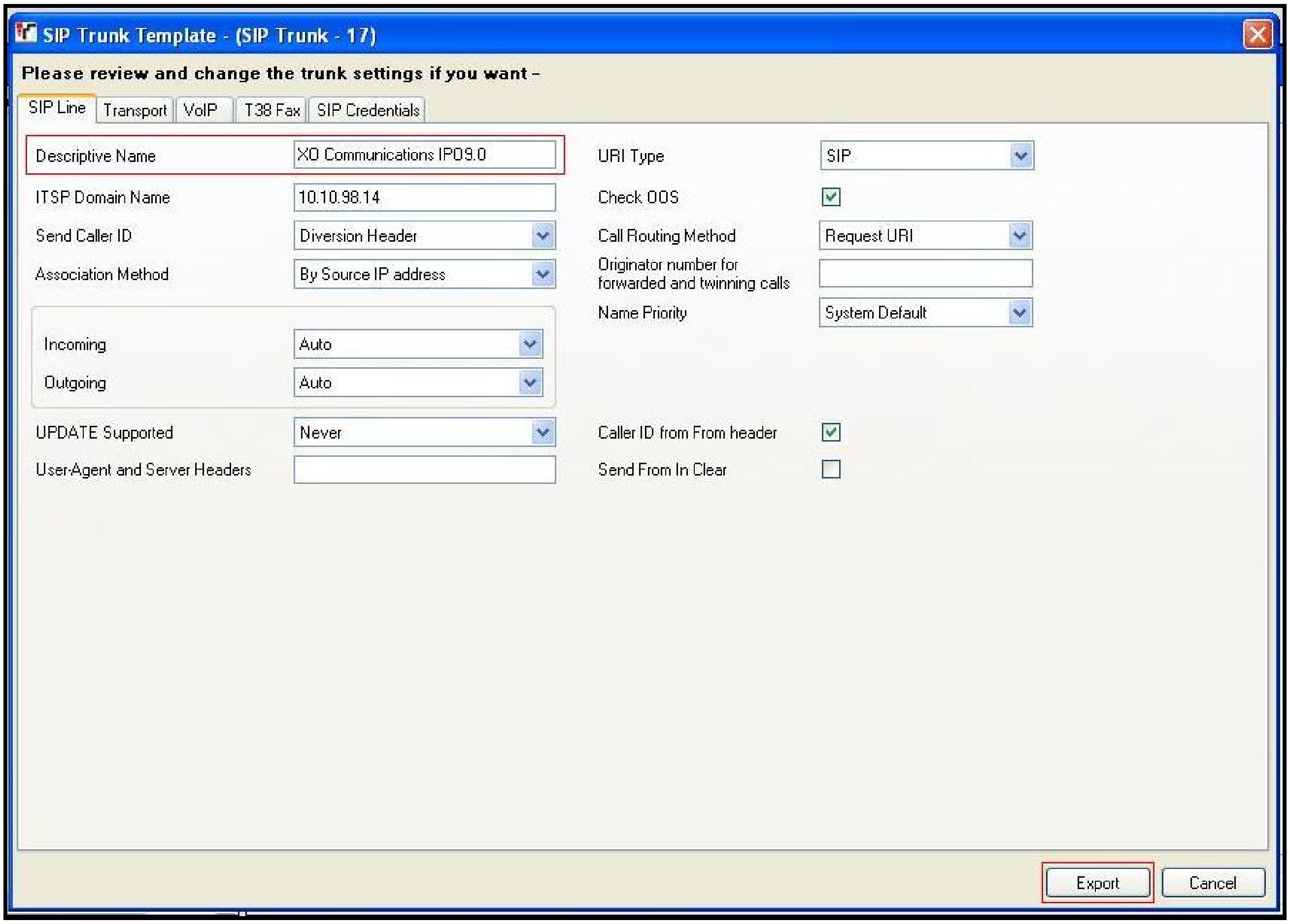
To enable template support in the IP Office Manager, select **File**, then **Preferences**. On the **Visual Preferences** tab, check the **Enable Template Options** box.



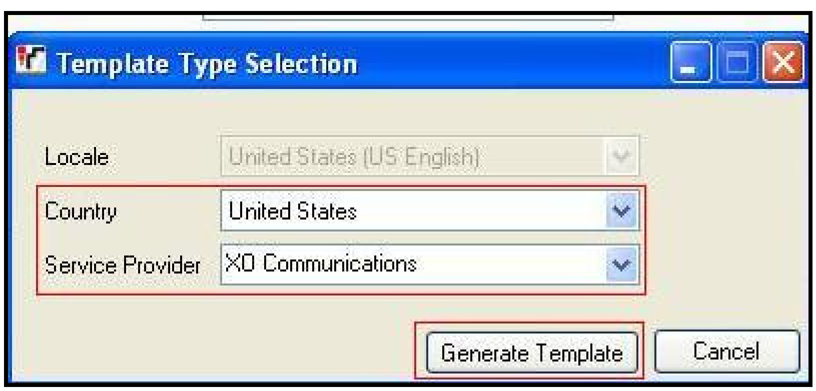
To create a SIP Line Template from the configuration, on the left Navigation Pane, right click the Sip Line (**17**), and select **Generate SIP Trunk Template**.



The trunk's settings are displayed as configured in **Section 5.5.** Enter a descriptive name for the template, adjust the settings if required, and then click on **Export**.



On the next screen, **Template Type Selection**, select the **Country,** enter the name for the **Service Provider**, and click **Generate Template**.



The warning for Generate Template as following, click **OK** to generate the template.

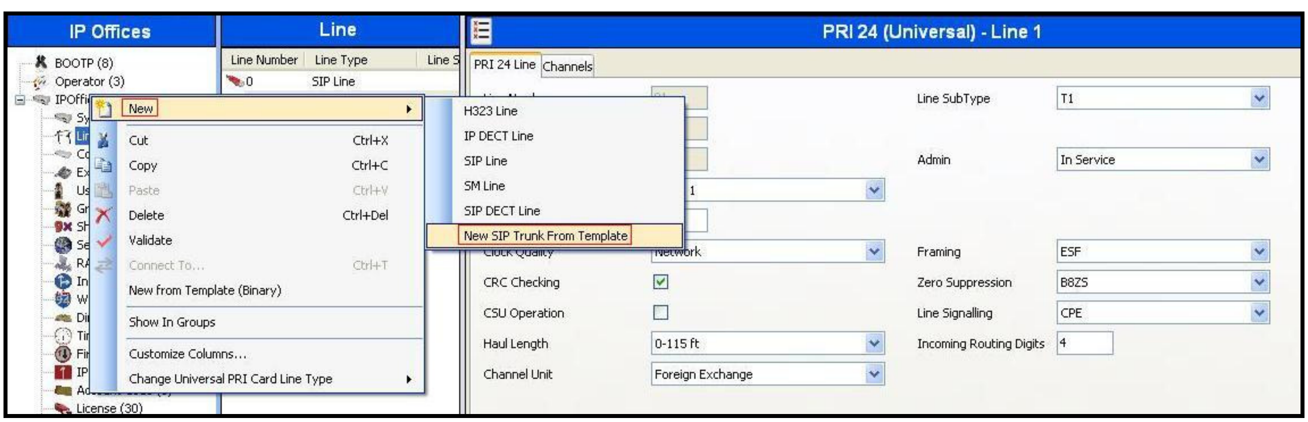
The following is the exported SIP Line Template file **US_XO Communications_SIPTrunk.xml**:


```xml
<?xml version="1.0" encoding="utf-8" ?>
<Template xmlns="urn:SIPTrunk-schema">
<TemplateType>SIPTrunk</TemplateType>
<Version>20131018</Version>
<SystemLocale>enu</SystemLocale>
<DescriptiveName>XO Communications IPO9.0</DescriptiveName>
<ITSPDomainName>10.10.98.14</ITSPDomainName>
<SendCallerID>CallerIDDIV</SendCallerID>
<ReferSupport>true</ReferSupport>
<ReferSupportIncoming>2</ReferSupportIncoming>
<ReferSupportOutgoing>2</ReferSupportOutgoing>
<RegistrationRequired>false</RegistrationRequired>
<UseTelURI>false</UseTelURI>
<CheckOOS>true</CheckOOS>
<CallRoutingMethod>1</CallRoutingMethod>
<OriginatorNumber />
<AssociationMethod>SourceIP</AssociationMethod>
<LineNamePriority>SystemDefault</LineNamePriority>
<UpdateSupport>UpdateNever</UpdateSupport>
<URIType>SIPURI</URIType>
<UserAgentServerHeader />
<CallerIDfromFromheader>true</CallerIDfromFromheader>
<PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
<ITSPProxy>192.168.163.138</ITSPProxy>
<LayerFourProtocol>SipUDP</LayerFourProtocol>
<SendPort>5060</SendPort>
<ListenPort>5060</ListenPort>
<DNSServerOne>0.0.0.0</DNSServerOne>
<DNSServerTwo>0.0.0.0</DNSServerTwo>
<CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>
<SeparateRegistrar />
<CompressionMode>AUTOSELECT</CompressionMode>
<UseAdvVoiceCodecPrefs>true</UseAdvVoiceCodecPrefs>
```

<AdvCodecPref>**G.729(a) 8K CS-ACELP,G.711 ULAW 64K**</AdvCodecPref>
<CallInitiationTimeout>**4**</CallInitiationTimeout>
<DTMFSupport>**DTMF_SUPPORT_RFC2833**</DTMFSupport>
<VoipSilenceSupression>**false**</VoipSilenceSupression>
<ReinviteSupported>**true**</ReinviteSupported>
<FaxTransportSupport>**FOIP_T38FB**</FaxTransportSupport>
<UseOffererPrefferedCodec>**false**</UseOffererPrefferedCodec>
<CodecLockdown>**false**</CodecLockdown>
<Rel100Supported>**false**</Rel100Supported>
<T38FaxVersion>**0**</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>**false**</UseDefaultValues>
<ScanLineFixup>**true**</ScanLineFixup>
<TFOPEnhancement>**true**</TFOPEnhancement>
<DisableT30ECM>**false**</DisableT30ECM>
<DisableEflagsForFirstDIS>**false**</DisableEflagsForFirstDIS>
<DisableT30MRCompression>**false**</DisableT30MRCompression>
<NSFOverride>**false**</NSFOverride>
</Template>

To import the template into a new IP Office system, copy and paste the exported xml template file into the Templates directory (C:\Program Files\Avaya\IP Office\Manager\Templates) on the PC where IP Office Manager for the new system is running.

Next, import the template into the new IP Office system by creating a new SIP Line as shown in the screenshot below. In the Navigation Pane on the left, right-click on **Line** then navigate to **New, New SIP Trunk From Template**:



On the next screen, **Template Type Selection**, verify that the information in the **Country** and **Service Provider** fields is correct. If more than one template is present, use the drop-down menus to select the required template. Click **Create new SIP Trunk** to finish the process.


Template Type Selection
Locale
United States (US English)
Country
United States
Service Provider
XO Communications
Display All
Create new SIP Trunk
Cancel

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.