Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 8.1 with Telecommunications Services of Trinidad and Tobago SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between the service provider Telecommunications Services of Trinidad and Tobago and Avaya IP Office 8.1.

During the interoperability testing, Avaya IP Office was able to interoperate with the Telecommunications Services of Trinidad and Tobago Soft switch via SIP trunking. This test was performed to verify SIP trunk features including basic call, call forward (all calls, busy, no answer), call transfer (blind and consult), conference, and voice mail. The calls were placed in both directions using various endpoint types.

Telecommunications Services of Trinidad and Tobago 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 the service provider Telecommunications Services of Trinidad and Tobago and an Avaya IP Office solution.

In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 8.1, Avaya Voicemail Pro, Avaya IP Office soft clients and Avaya desk phones, including SIP, H.323, digital, and analog endpoints.

The Telecommunications Services of Trinidad and Tobago SIP Trunk Service referenced within these Application Notes is designed for business customers in Trinidad and Tobago. Customers using this service with the Avaya IP Office solution are able to place and receive PSTN calls via a broadband WAN connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as analog and/or ISDN-PRI trunks. This approach generally results in lower cost for the enterprise. Telecommunications Services of Trinidad and Tobago will be referred to as **TSTT** here after.

The Avaya IP Office system can be configured to authenticate with the SIP service provider using either SIP Trunk Registration or Static IP Authentication. These Application Notes cover the configuration of IP Office using Static IP Authentication with the service provider.

2. General Test Approach and Test Results

A simulated enterprise site was configured in the test lab using Avaya IP Office, connected to the TSTT SIP Trunk Service via a SIP trunk over the public Internet. This scenario may differ from a real customer environment, in which a dedicated private network connection could be provided by TSTT to the actual customer site.

The configuration shown in **Section 3 - Figure 1** was used to exercise the features and functionality tests listed in **Section 2.1**.

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

Testing was performed using IP Office 500v2 R8.1, but results also apply to IP Office Server Edition R8.1. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R8.1 to support analog or digital endpoints and trunks.

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 OPTIONS queries.
- Incoming PSTN calls to various phone types. Phone types included SIP, H.323, digital and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types. Phone types included SIP, H.323, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from soft clients.
- Various call types including: long distance, international, outbound toll-free, etc.
- Codec's G.711MU and G.729A (For Codec G.729A Test Results refer to **Section 2.2**).
- Caller ID presentation and Caller ID restriction.
- DTMF transmission using RFC 2833.
- Voicemail navigation 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 with TSTT was completed successfully with the exception of the observations/limitations described below:

- **SIP REFER** – On PSTN calls to or from IP Office that are transferred back to the PSTN on the SIP trunk, TSTT responds with a "202 Accepted" to the REFER message sent by IP Office, but the call between the two PSTN endpoints drops and the PSTN phone receives re-order tone. REFER needs to be disabled in IP Office for the Call Transfer to complete successfully; otherwise the call transfer will fail. The implication is that IP Office SIP trunk channels are not released after the call transfer is completed and two (2) trunk channels will remain connected/busy for the duration of the call.
- **T.38 or G.711 Pass-Through fax calls** – With IP Office **Fax Transport Support** set as **T.38** or **T.38 Fallback** on the **SIP Line/VoIP**, on outbound calls (IPO→PSTN) TSTT did not send a re-INVITE to switch from G.711 to T.38. TSTT's recommendation is **not** to use T.38 fax transport, only G.711 fax Pass-through. With IP Office **Fax Transport Support** set as **G.711** on the **SIP Line/VoIP**, fax calls were unsuccessful, thus **T.38 or G.711** fax transports **are not** recommended for this solution.
- **Codec G.729A** – TSTT supports codecs G.711MU and G.729A, but during the testing TSTT was rejecting calls with G.729A codec offer with **488 Invalid Media Type**. This issue is under investigation by TSTT.

2.3. Support

For technical support on the TSTT SIP Trunk Service offer, visit http://tstt.co.tt/

3. Reference Configuration

Figure 1 below illustrates the test configuration. It shows the enterprise site connected to the TSTT SIP Trunk Service through the public IP network.

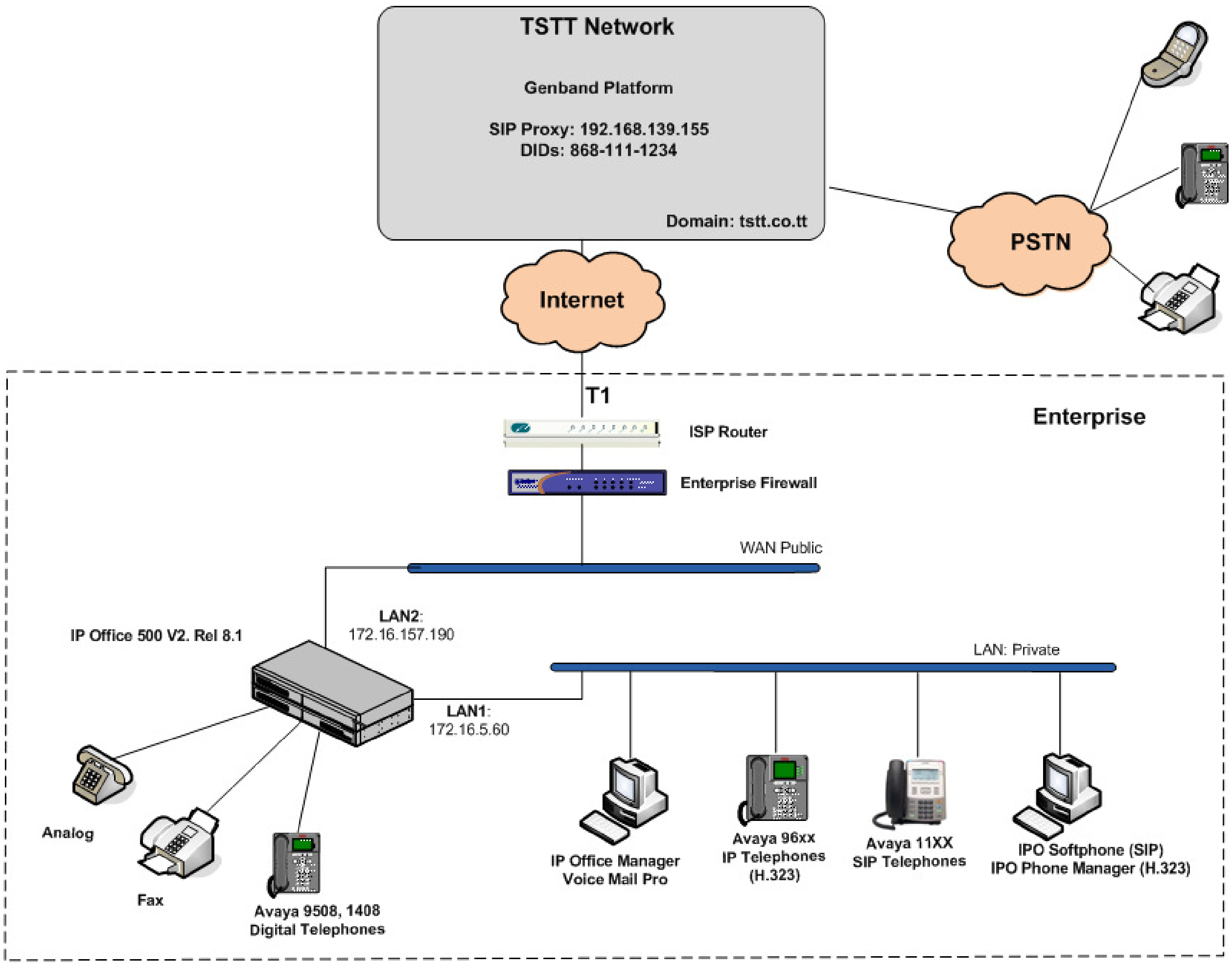


Figure 1: Test Configuration

Note that for security purposes, all public IP addresses and phone numbers shown throughout these Application Notes have been edited so that the actual IP addresses of the network elements and public PSTN numbers are not revealed.

The enterprise site contains the Avaya IP Office 500v2 with analog and digital extension expansion modules, as well as a VCM64 (Voice Compression Module) for supporting VoIP codecs. The LAN1 port of Avaya IP Office is connected to the enterprise LAN while the LAN2 port is connected to the public IP network. Endpoints include Avaya 9600 Series IP Telephones (with H.323 firmware), Avaya 1140E IP Telephones (with SIP firmware), Avaya 1408 and 9508D Digital Telephones, analog telephones, Officejet 4500 fax machines and PCs running Avaya IP Office Softphone configured as SIP soft clients and Avaya Phone Manager configured as H.323 soft clients. The site also has a Windows XP PC running Avaya IP Office Manager to configure and administer the Avaya IP Office system, and Avaya Voicemail Pro providing voice messaging service to the Avaya IP Office users. Mobile Twinning is configured for some of the Avaya IP Office users so that calls to these users' extensions will also ring and can be answered at the configured mobile phones.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to make calls across the SIP trunk to TSTT. The short code 9 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to the network. Since Trinidad and Tobago is a country member of the North American Numbering Plan (NANP), the users dialed 10 digits for local calls, including the area code, and 11 (1 + 10) digits for other calls between the NANP.

In an actual customer configuration, the enterprise site may include additional network components between the service provider and the Avaya IP Office system, 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 all SIP and RTP traffic between the service provider and the Avaya IP Office system 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	Release / Version
Avaya	
Avaya IP Office 500v2	8.1 (69)
Avaya IP Office Digital Expansion Module DCPx16	10.1 (69)
Avaya IP Office Manager	10.1 (69)
Avaya IP Office Voicemail Pro	8.1.9203.0
Avaya 9620 IP Telephone (H.323)	Avaya one-X Deskphone Edition S3.2
Avaya 1140E IP Telephone (SIP)	04.03.12.00
Avaya Digital Telephone 1408	32
Avaya Digital Telephone 9508	0.45
Avaya IP Office Softphone (SIP)	3.2.3.48.67009
Telecommunications Services of Trinidad and Tobago SIP Trunk Service	
Genband Softswitch	CVM 13

5. Configure IP Office

This section describes the Avaya IP Office configuration necessary to support connectivity to the TSTT SIP Trunk Service. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. 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. A management window will appear similar to the one shown in the next section.

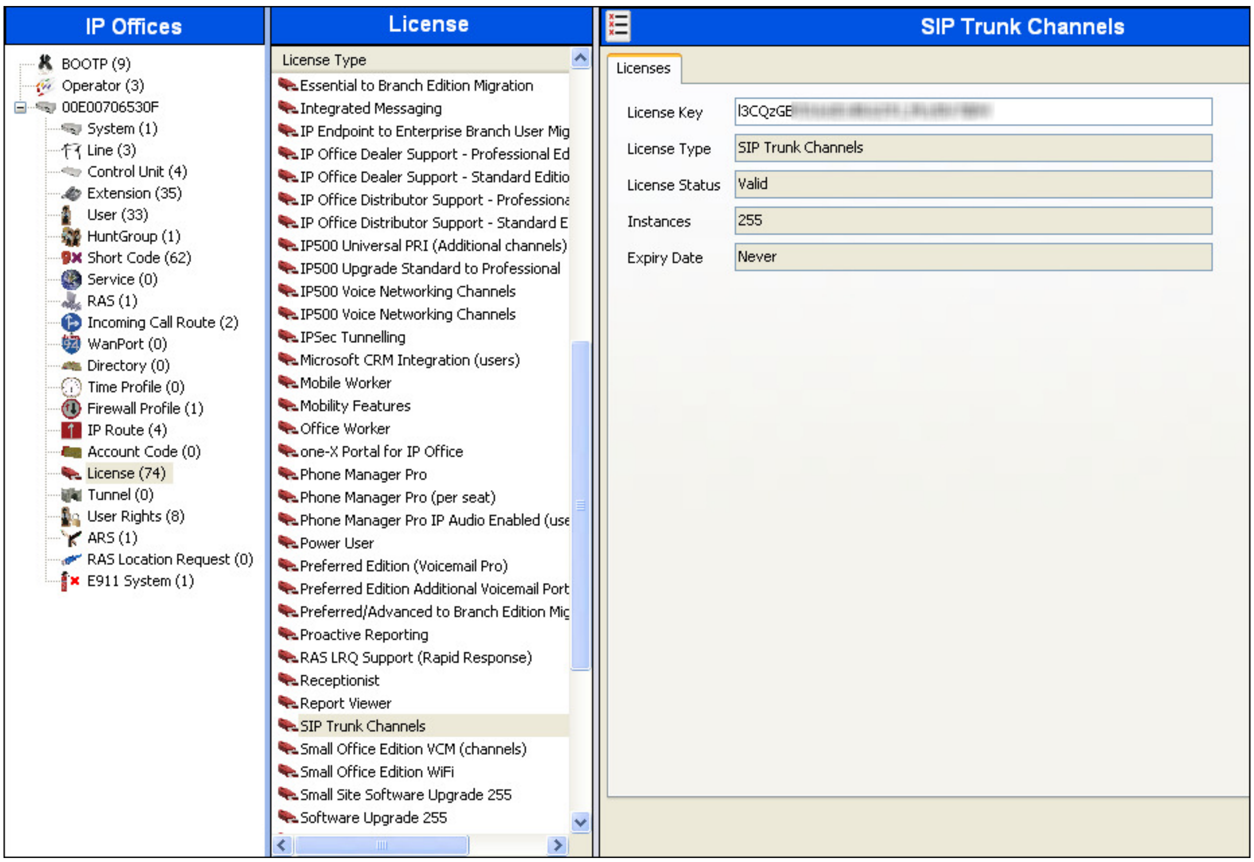
The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center and the Details pane on the right side. These panes will be referenced throughout the Avaya IP Office configuration.

Standard feature configurations that are not directly related to the interfacing with the service provider (such as LAN interface to the enterprise site, Twinning and IP Office Softphone support) are assumed to be already in place, and they are not part of these Application Notes.

5.1. Licensing

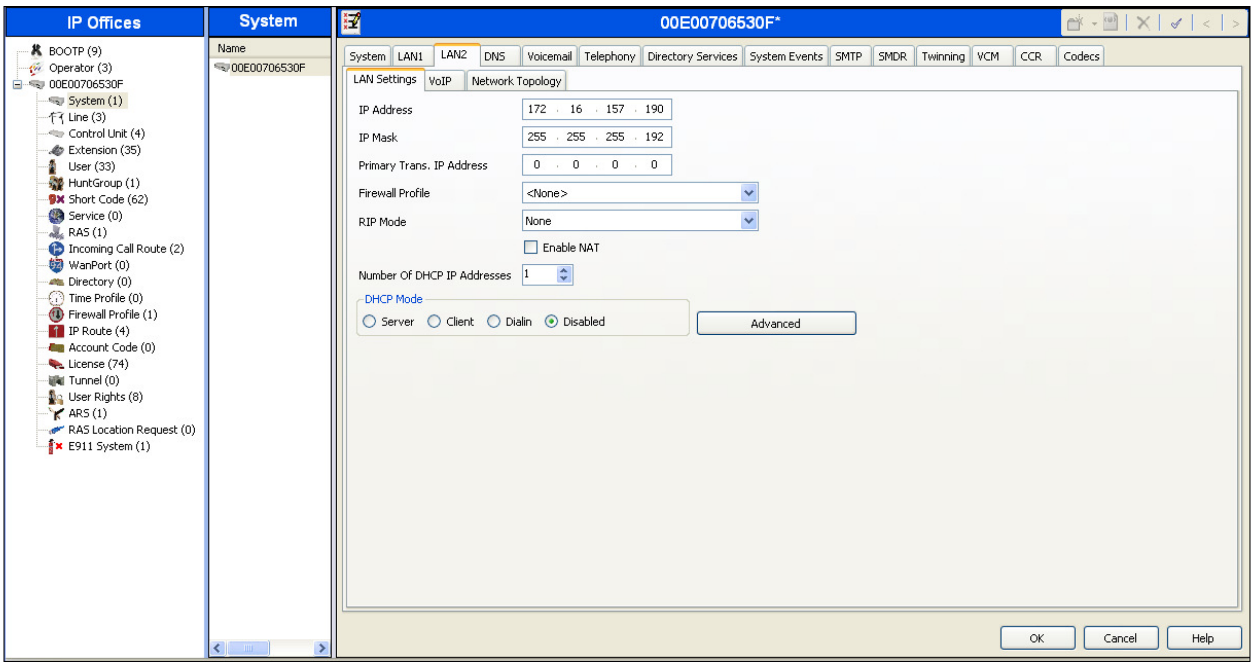
The configuration and features described in these Application Notes require the IP Office system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

To verify that there is a SIP Trunk Channels License with sufficient capacity; click **License** in the Navigation pane and **SIP Trunk Channels** in the Group pane. Confirm that there is a valid license with sufficient "Instances" (trunk channels) in the Details pane. Note that the actual License Key in the screen below was hidden for security purposes.



5.2. LAN2 Settings

As shown in the following screen, the Name field can be used for a descriptive name of the system. In the sample configuration, the MAC address **00E00706530F** was used as the system name, and the WAN port was used to connect the Avaya IP Office to the public network. The LAN2 settings correspond to the WAN port on the Avaya IP Office. To access the LAN2 settings, first navigate to **System (1) → 00E00706530F** in the Navigation and Group panes and then navigate to the **LAN2 → LAN Settings** tab in the Details pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office WAN port. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements.

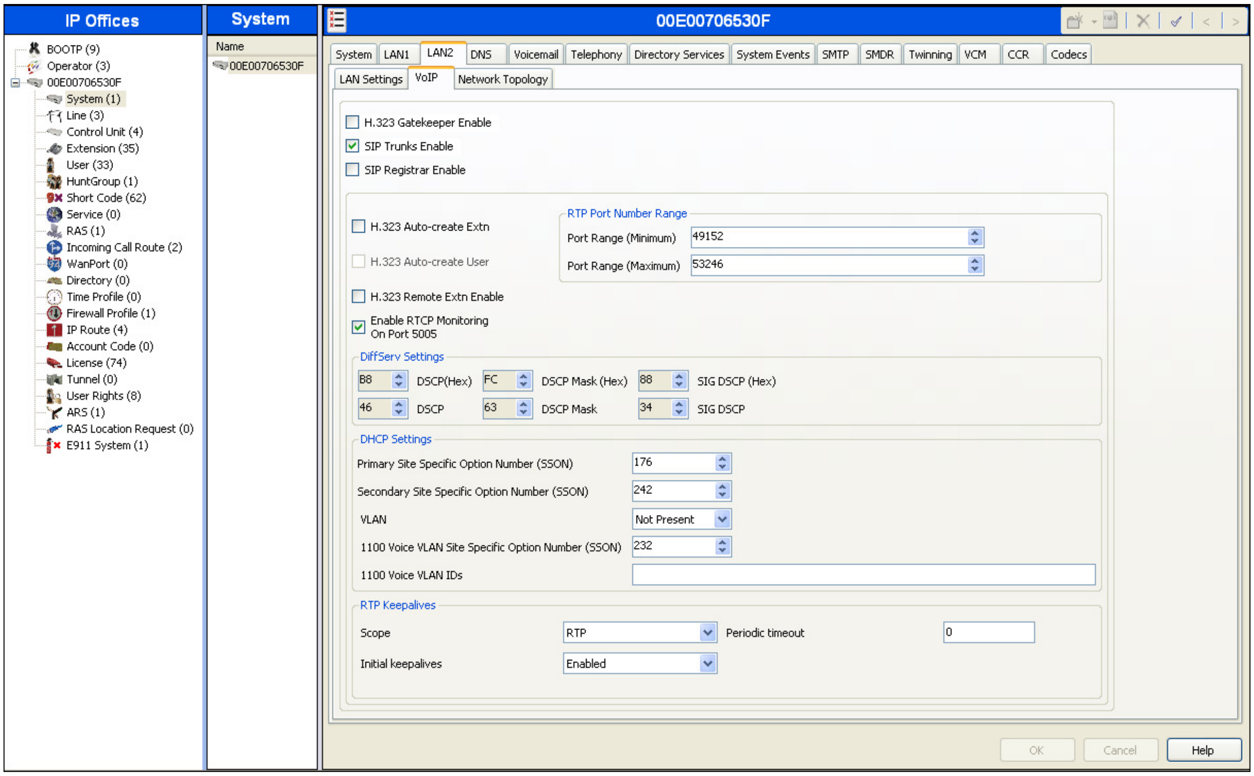


On the **VoIP** tab in the Details pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The **RTP Port Number Range** can be customized to a specific range of receive ports for RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to ports in the configured range for calls using LAN2.

Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling. The specific values used for the compliance test are shown in the example below.

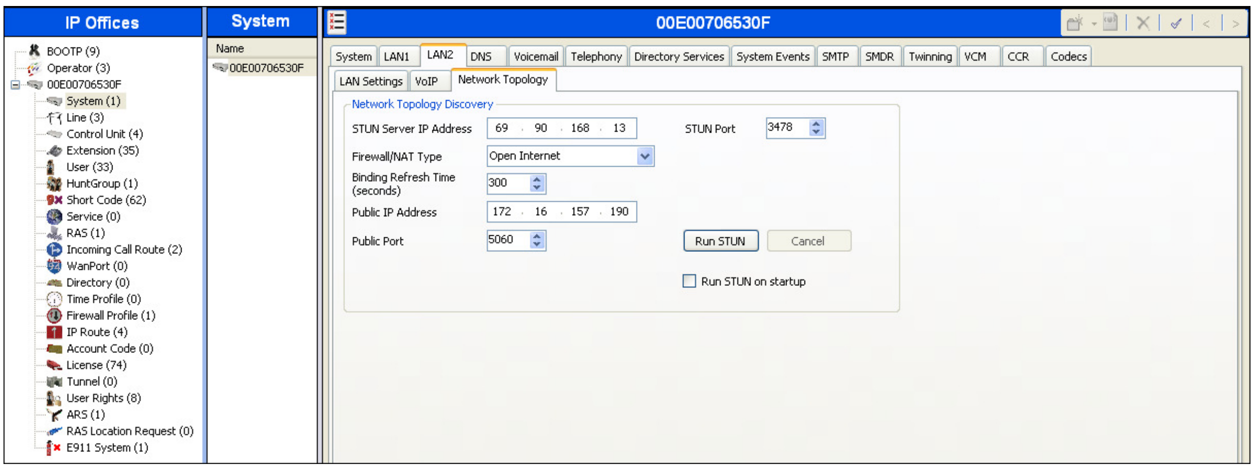
In the **RTP Keepalives** section at the bottom of the page, set the **Scope** field to **RTP**, and **Initial keepalives** to **Enabled**. This will cause the Avaya IP Office to send RTP keepalive packets at the beginning of the calls. To avoid problems of media deadlock that can occur with certain types of forwarded calls that are routed from the Avaya IP Office back to the network, over the same SIP trunk.

All other parameters should be set according to customer requirements.



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

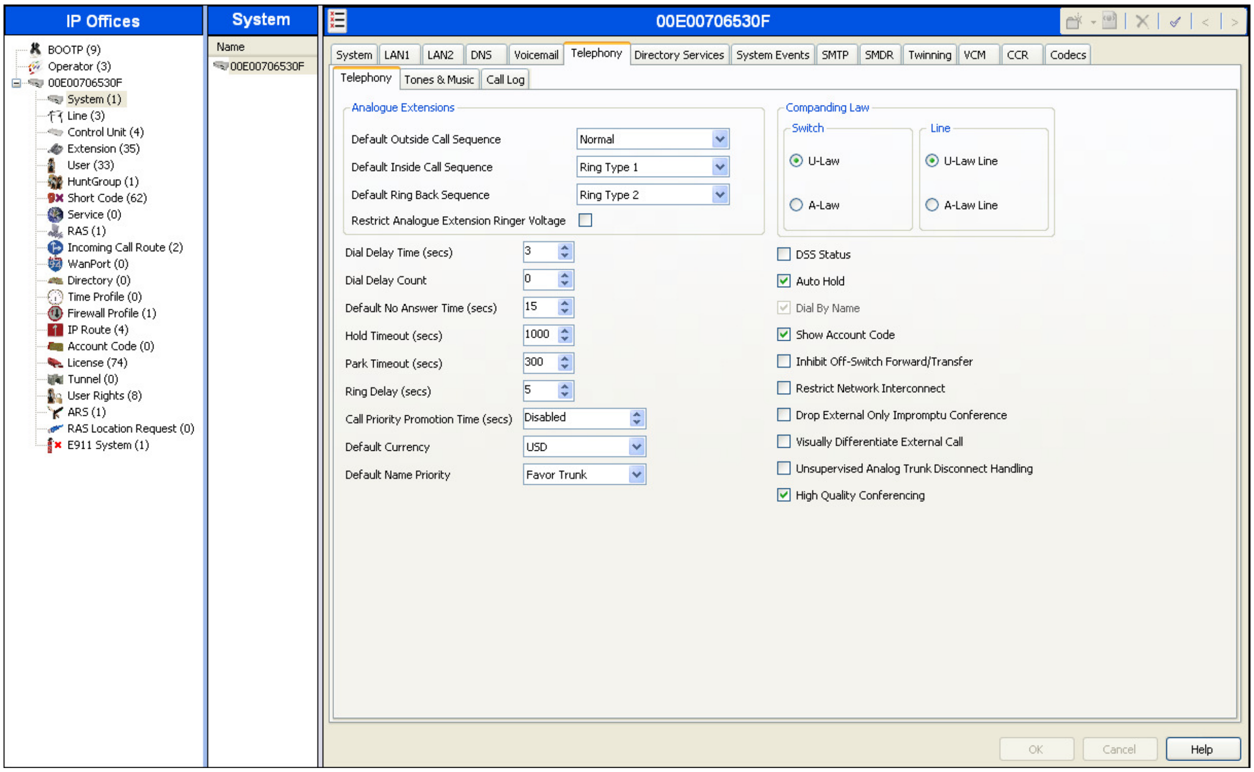
- Select the **Firewall/NAT Type** from the pull-down menu to the option that matches the network configuration. Since no network address translation (NAT) was used in the compliance test, the parameter was set to **Open Internet**. With this configuration, the **STUN Server IP Address** and **STUN Port** are not used.
- Set **Binding Refresh Time (seconds)** to **300 (or every 5 minutes)**.This value is used to determine the frequency at which Avaya IP Office will send SIP OPTION messages to the service provider.
- Set **Public IP Address** to the IP address that was set for LAN2.
- Set **Public Port** to **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 the TSTT SIP Trunk Service, and therefore is not described in these Application Notes.

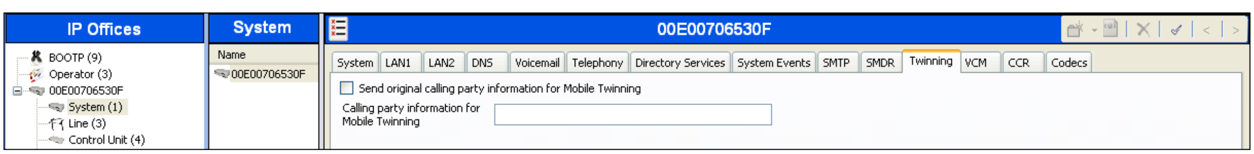
5.3. System Telephony Settings

Navigate to the **Telephony → Telephony** Tab in the Details Pane. Choose the **Companding Law** typical for the enterprise location, **U-Law** was used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfers to the PSTN via the SIP trunk to the service provider.



5.4. Twinning Calling Party Settings

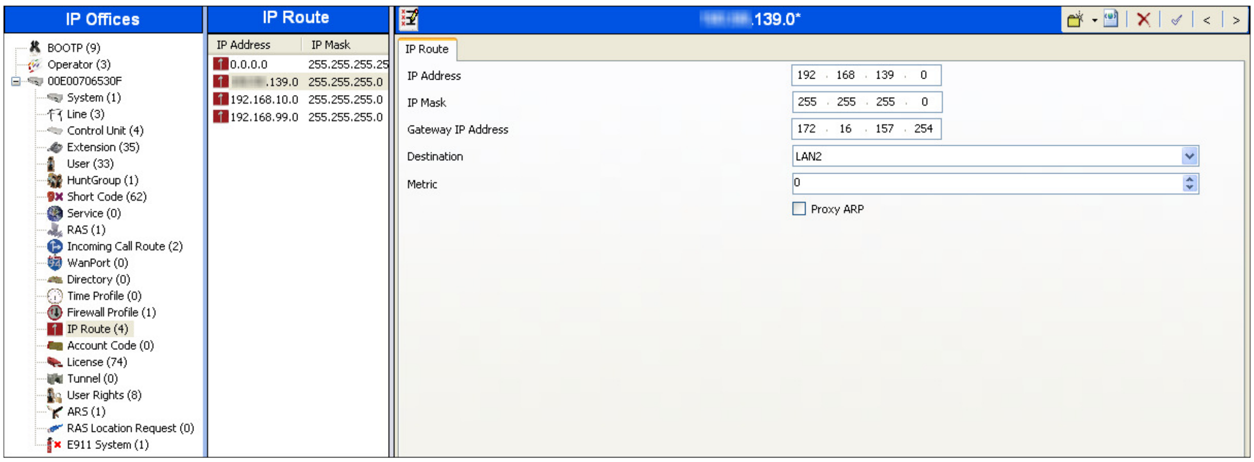
Navigate to the **Twinning** tab on the Details Pane. Uncheck the **Send original calling party information for Mobile Twinning** box. This will allow the Caller ID for Twinning to be controlled by the setting on the SIP Line (**Section 5.6**). This setting also impacts the Caller ID for call forwarding.



5.5. IP Route

Create an IP route to specify the IP address of the gateway or router where the IP Office needs to send the packets, in order to reach the subnet where the SIP proxy is located on the TSTT network. On the left navigation pane, right-click on **IP Route** and select **New**.

- Set the **IP Address** and **IP Mask** of the remote TSTT SIP Proxy subnet.
- Set **Gateway IP Address** to the IP Address of the router used to reach the external network. For the test configuration, this was the IP address of the local ISP router.
- Set **Destination** to **LAN2** from the pull-down menu.



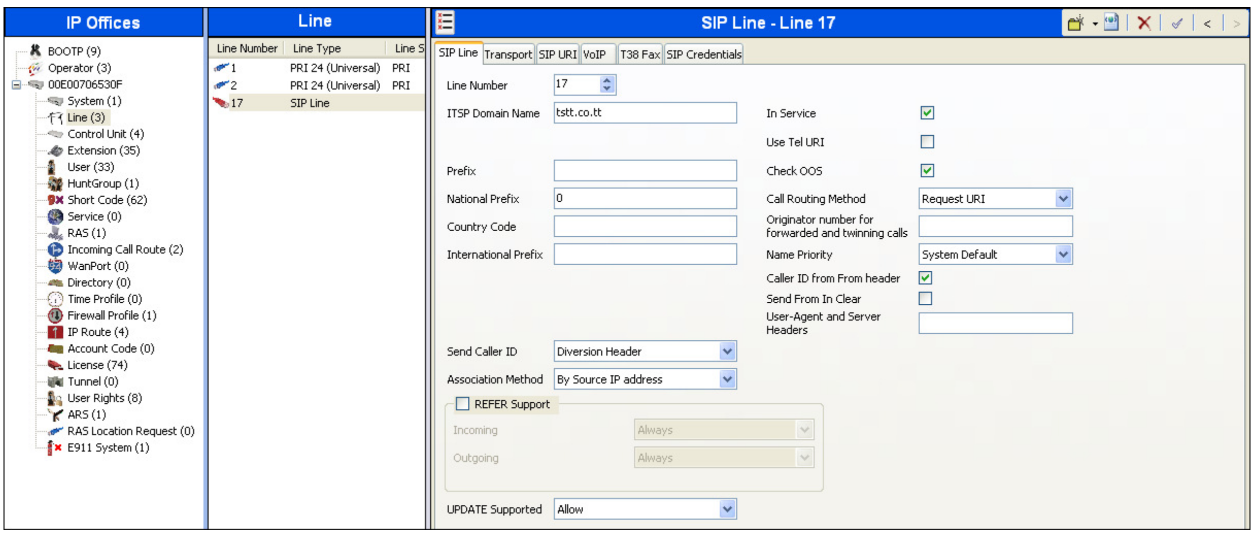
5.6. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the TSTT SIP Trunk Service. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New→ SIP Line**.

5.6.1. SIP Line Tab

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

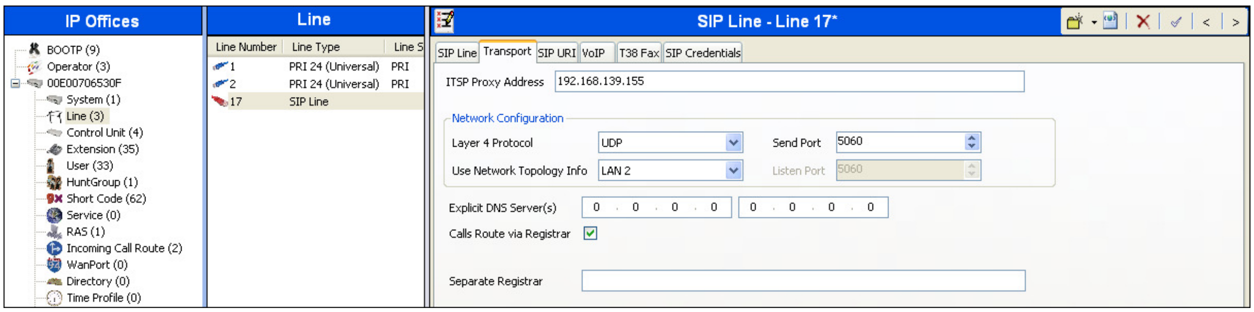
- Set **ITSP Domain Name** to the domain of the Service Provider, Avaya IP Office uses this domain on the host portion of SIP URI in SIP headers such as the From header.
- Check the **In Service** box.
- Check the **Check OOS** box. With this option selected, IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- Set **Call Routing Method** to **Request URI**.
- Check the **Caller ID from From header** box.
- Set **Send Caller ID** to **Diversion Header.**
- Uncheck the **REFER support** box. IP Office will not send REFER headers for calls that are transferred back to the PSTN. See **Section 2.2** for more information.
- Set **UPDATE Supported** to **Allow**.
- Default values may be used for all other parameters.



5.6.2. Transport Tab

Select the **Transport** tab and set the following:

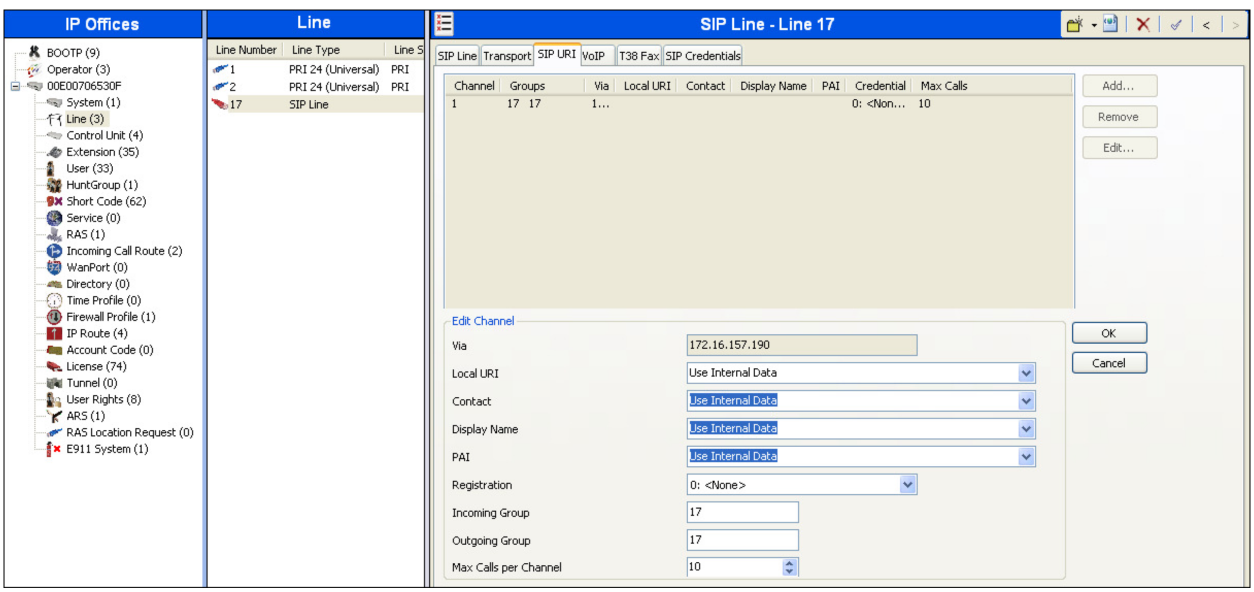
- Set the **ITSP Proxy Address** to the IP address of the TSTT proxy server.
- Set the **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to **LAN2** as configured in **Section 5.2**.
- Set the **Send Port** to **5060**.
- Default values may be used for all other parameters.



5.6.3. SIP URI Tab

A SIP URI entry needs to 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 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 was 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:

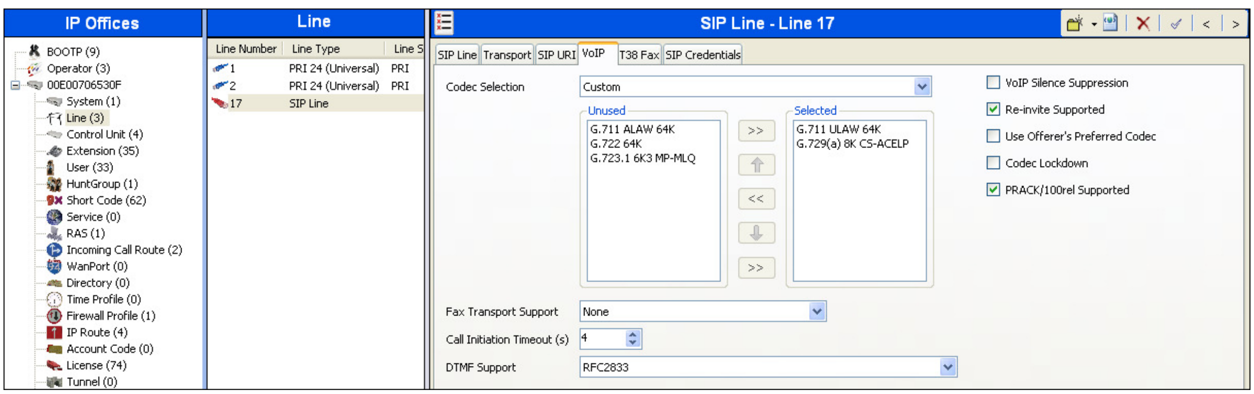
- Set **Local URI**, **Contact** and **Display Name** to **Use Internal Data**. This setting allows calls on this line whose SIP URI match the number set in the **SIP** tab of any User as shown in **Section 5.8**.
- Associate this line with an incoming line group by entering a line group number in the **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. 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.



5.6.4. VoIP Tab

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

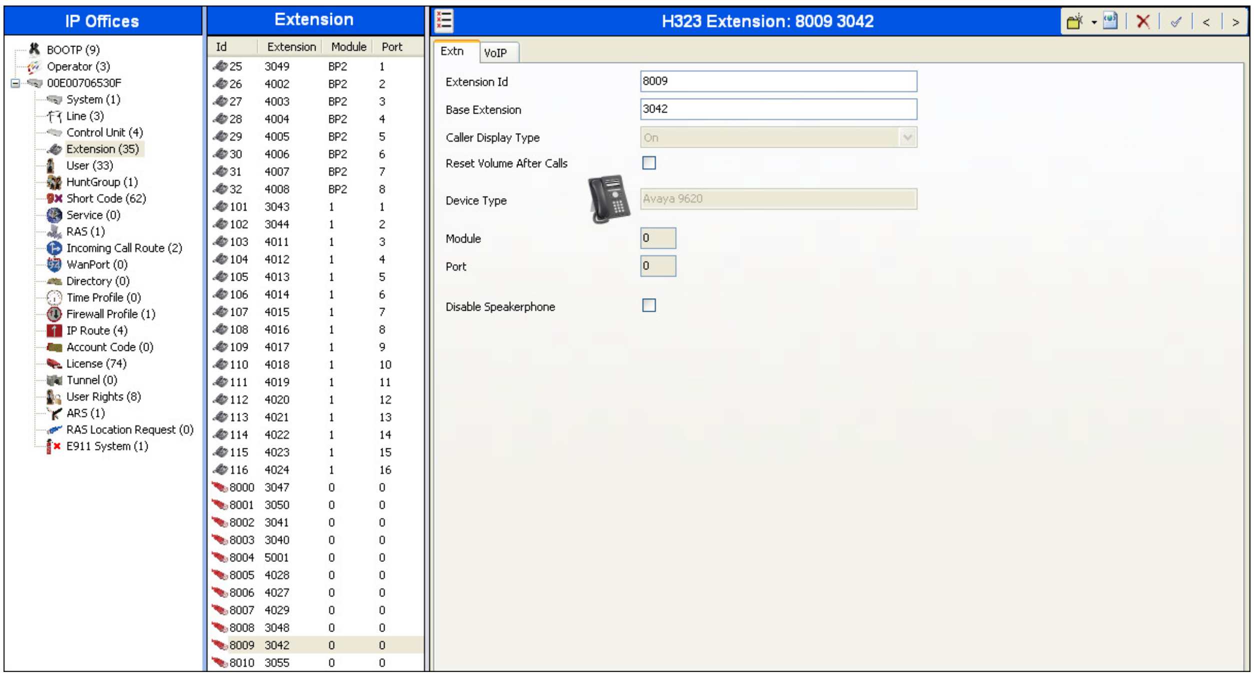
- In the sample configuration, the **Codec Selection** was configured using the **Custom** option, allowing an explicit ordered list of codecs to be specified. The buttons allow setting the specific order of preference for the codecs to be used on the line, as shown. TSTT supports codec's G.711MU and G.729A, during the testing TSTT was rejecting calls with G.729A codec offers with **488 Invalid Media Type**, as described in **Section 2.2**
- Set **Fax Transport Support** to **None**. **T.38 or G.711** fax transports **are not** recommended for this solution, as described in **Section 2.2**.
- Set the **DTMF Support** field to **RFC2833.** This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Check the **Re-invite Supported** box to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk.
- Check the **PRACK/100rel Supported** box, to advertise the support for provisional responses and Early Media to TSTT.
- Default values may be used for all other parameters.



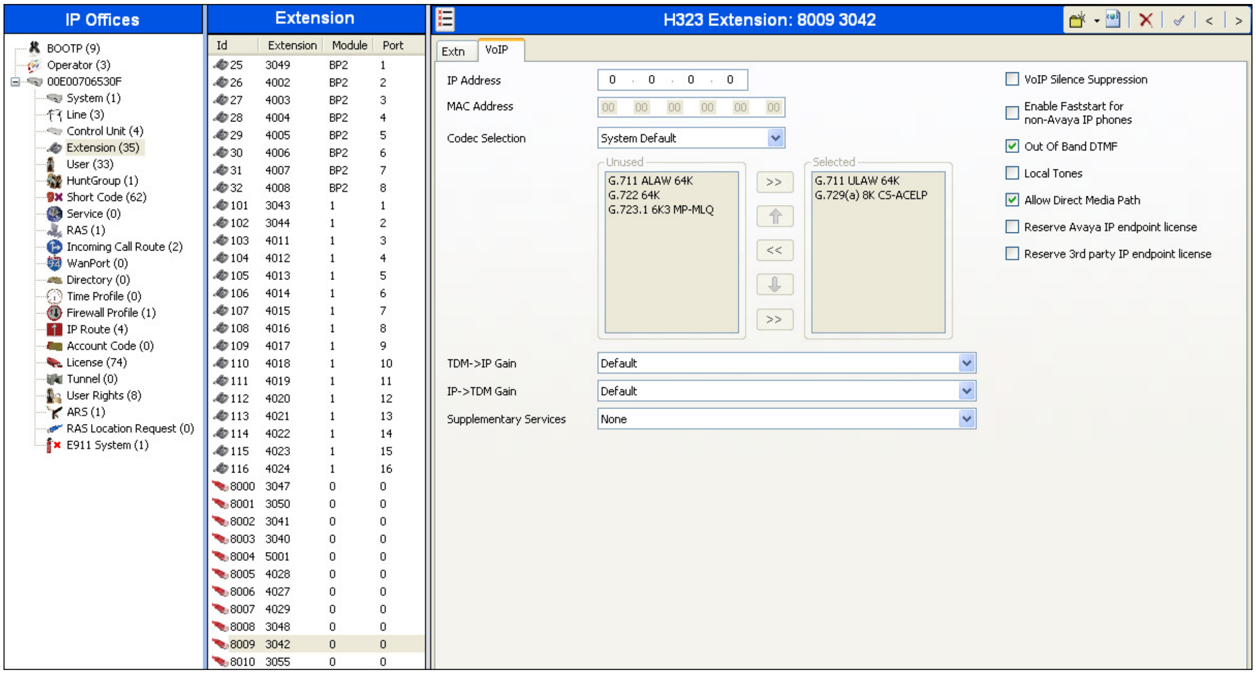
5.7. Extension

In this section, an example of an Avaya IP Office Extension will be illustrated. In the interests of brevity, not all users and extensions shown in **Figure 1** will be presented, since the configuration can be easily extrapolated to other users and extensions. To add an Extension, right click on **Extension** then select **New → Select H323 or SIP.**

Select the **Extn** tab. Following is an example of extension 3042; this extension corresponds to a H.323 extension.

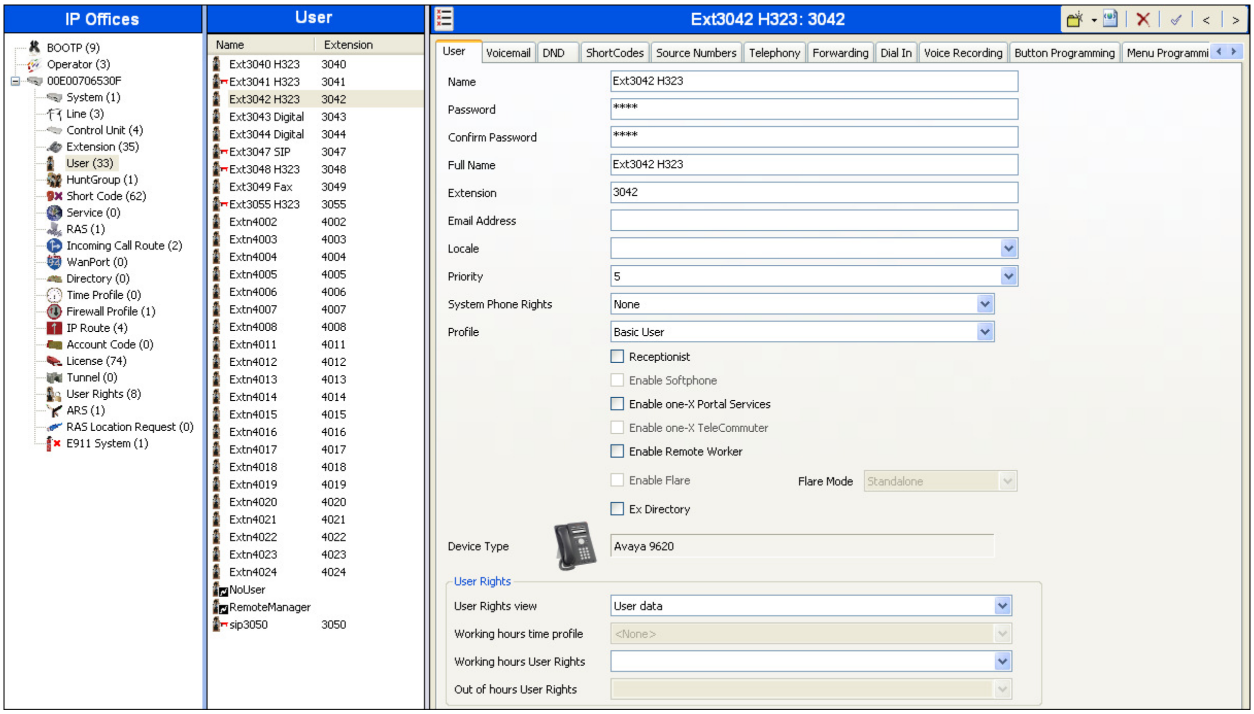


Select the **VOIP** tab. Use default values on VoIP tab. Following is an example for Extension 3042; this extension corresponds to a H.323 extension.



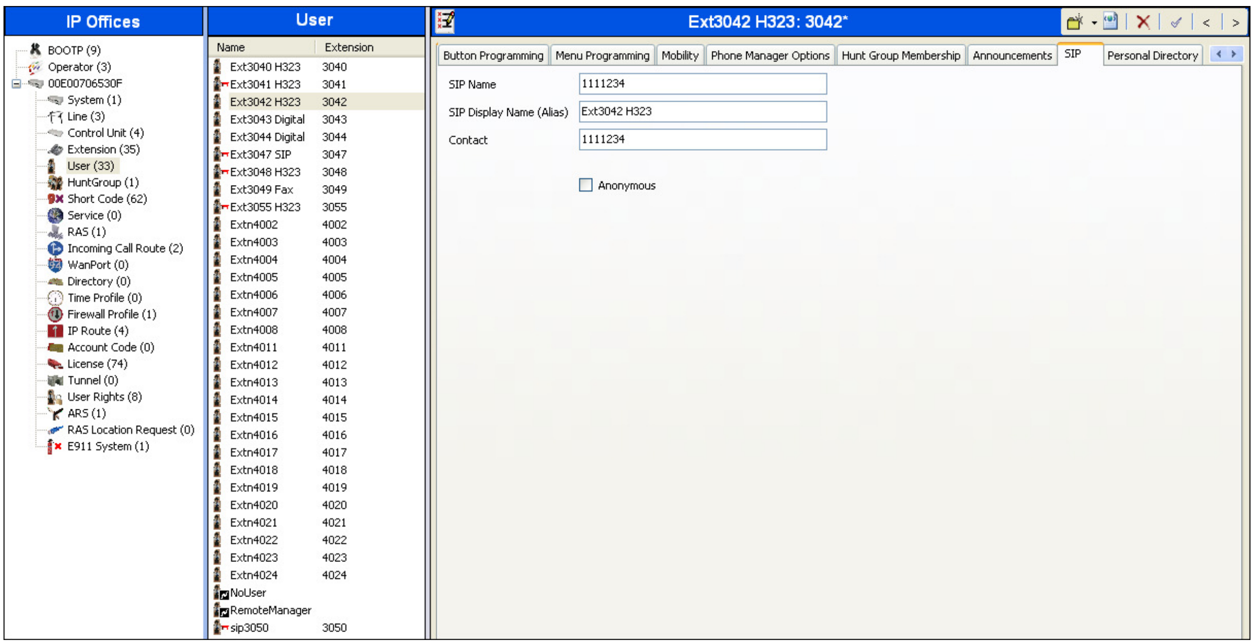
5.8. Users

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.6**. To configure these settings, first navigate to **User** in the left Navigation Pane, and 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 **Ext3042 H323**.

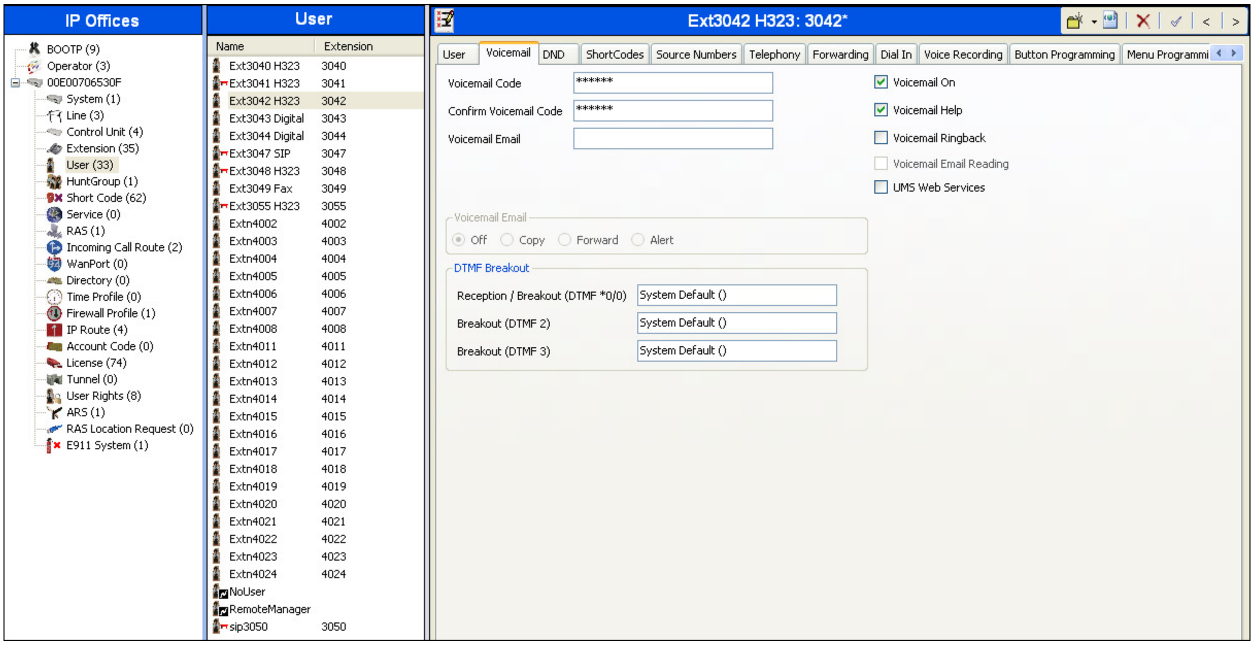


Select the **SIP** tab. The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From and Contact headers for outgoing SIP trunk calls. In addition, these settings are used to match against the SIP URI of incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.6**). The example below shows the settings for user “Ext3042 H323”. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise by TSTT. In the example, DID number **1111234** was used. Only the last seven digits of the DID were assigned since TSTT only sends seven digits without the area code (868). The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name.

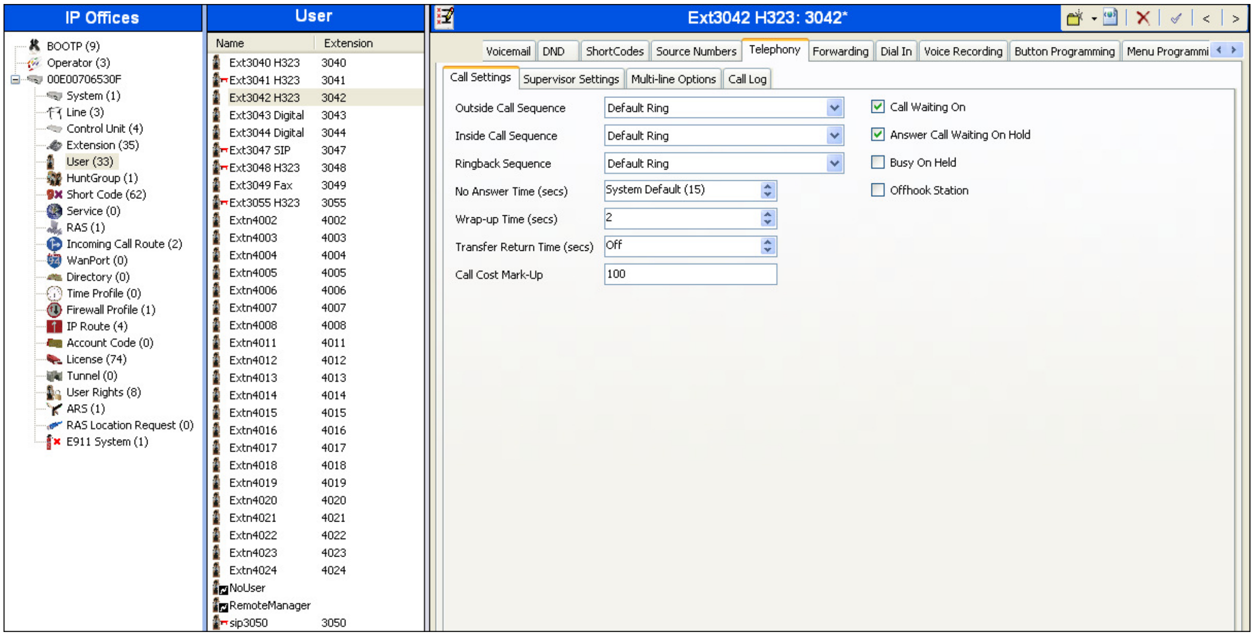
If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user’s information from the network.



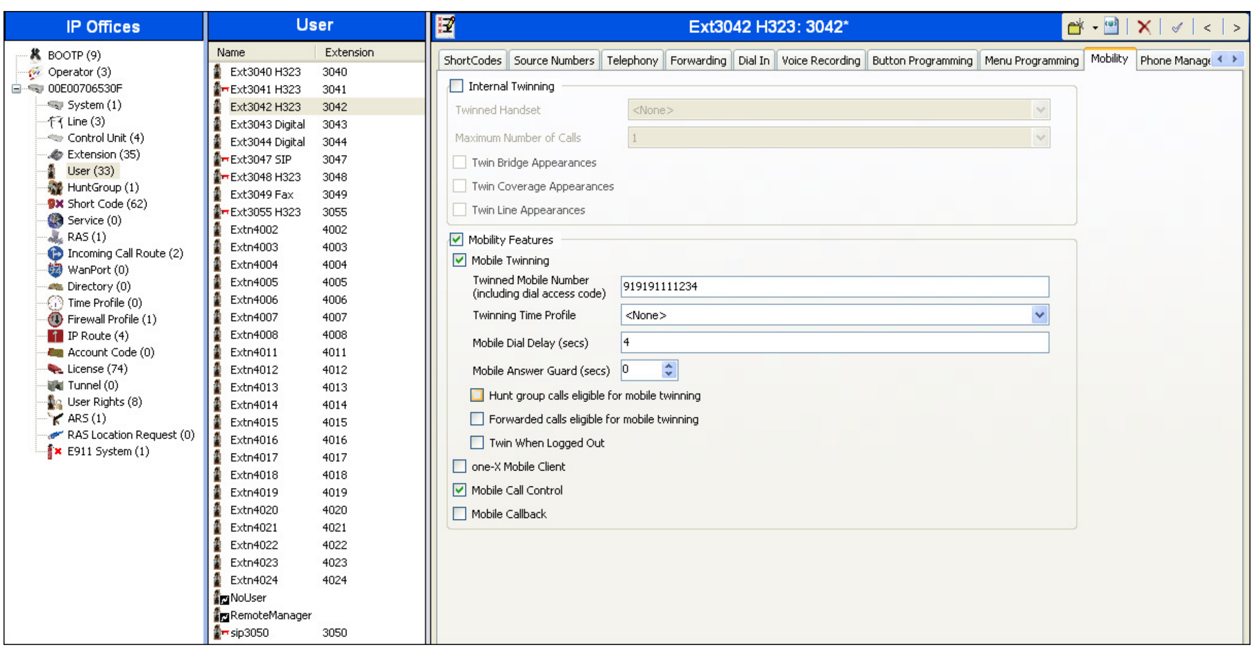
Select the **Voice Mail** tab. The following screen shows the **Voicemail** tab for the user with extension 3042. The **Voicemail On** box is checked. Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters. In the verification of these Application Notes, incoming calls from TSTT SIP Trunk to this user were redirected to Voicemail Pro after no answer. Voicemail messages were recorded and retrieved successfully. Voice mail navigation and retrieval was performed locally and from PSTN telephones to test DTMF using RFC 2833.



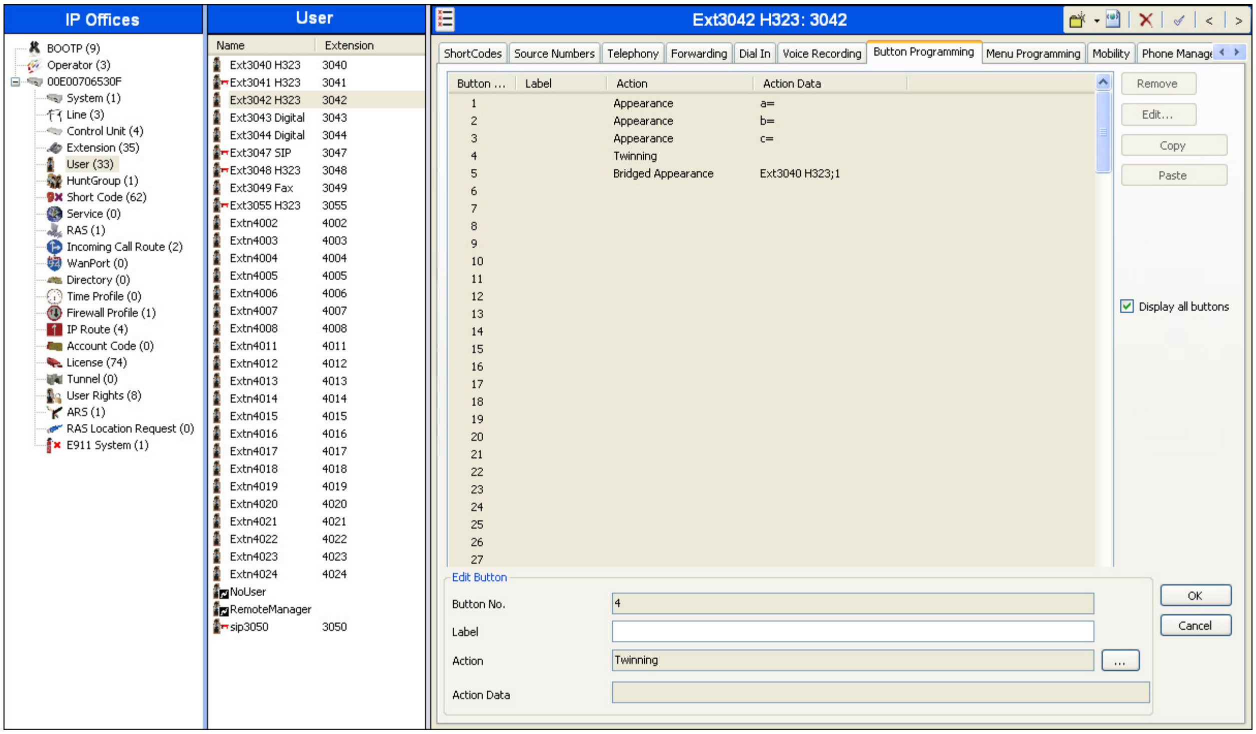
Select the **Telephony** tab, then **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow an Avaya IP Office phone logged in as this extension to have multiple call appearances. Note: **Call Waiting On** is necessary for call transfer.



Select the **Mobility** tab. In the sample configuration user 3042 was one of the users configured to test the Mobile Twinning feature. The following screen shows the **Mobility** tab for User 3042. 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 telephone, in this case **919191111234**. Other options can be set according to customer requirements.



To program a key on the telephone to turn Mobile Twinning on or off, select the **Button Programming** tab on the user, then select the button to program to turn Mobile Twinning on and off, click on **Edit → Emulation → Twinning**. In the sample below, button **4** was programmed to turn Mobile Twinning on and off on user 3042.



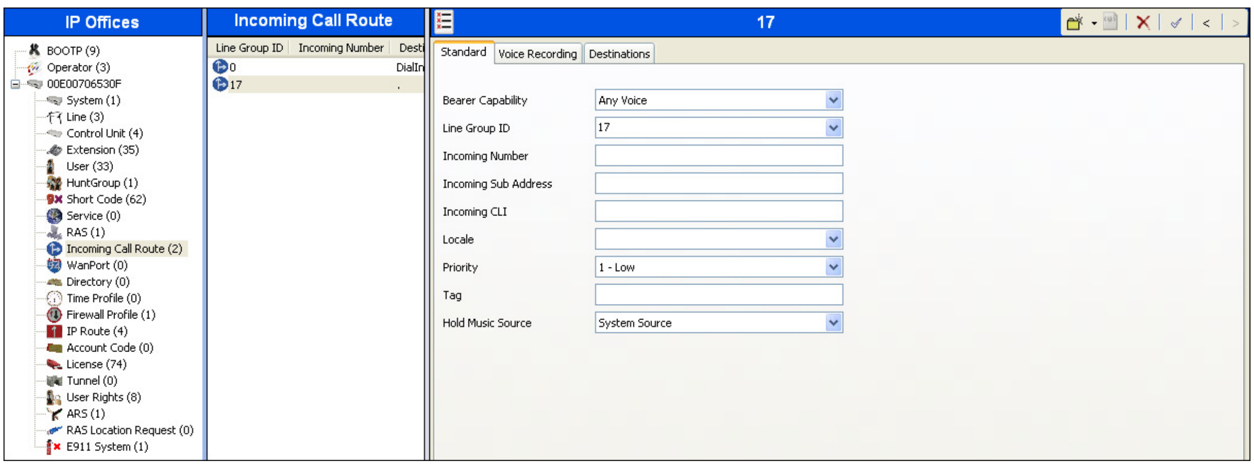
5.9. Incoming Call Route

An incoming call route maps inbound DID numbers on a specific line to internal extensions, hunt groups, short codes, etc, within the IP Office system. Incoming call routes should be defined for each DID number assigned by the service provider.

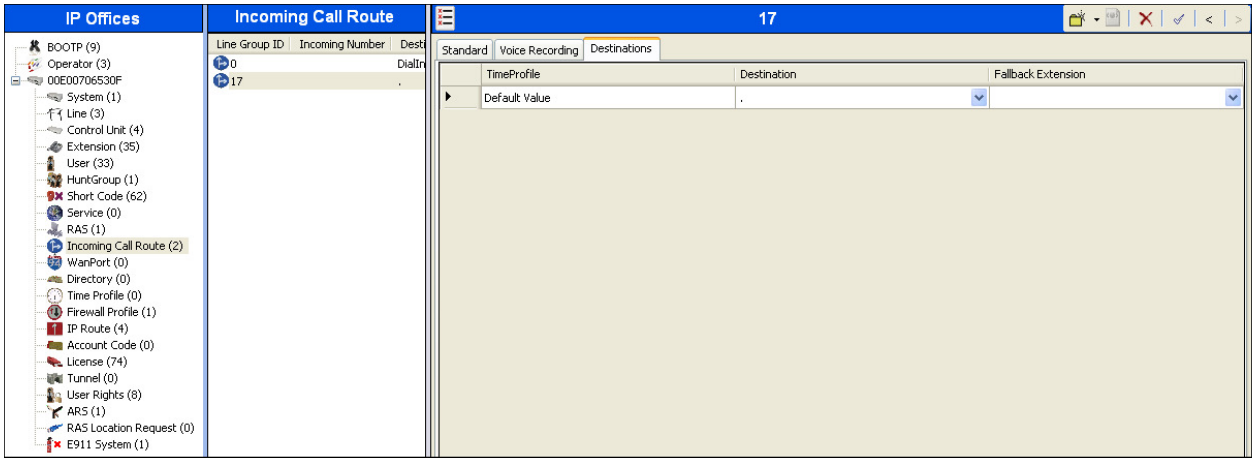
In a scenario similar to the one used for the compliance test, only one incoming route is needed, which allows any incoming number arriving on the SIP trunk to reach any predefined extension in IP Office. The routing decision for the call is based on the parameters previously configured for **Call Routing Method** and **SIP URI (Section 5.6)** and the users **SIP Name** and **Contact**, already populated with the assigned TSTT DID numbers **(Section 5.8)**

From the left Navigation Pane, right-click on **Incoming Call Route** and select **New.**
On the Details Pane, under the **Standard** tab, set the parameters as show bellow:

- Set **Bearer Capacity** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.6**.
- Default values may be used for all other parameters.



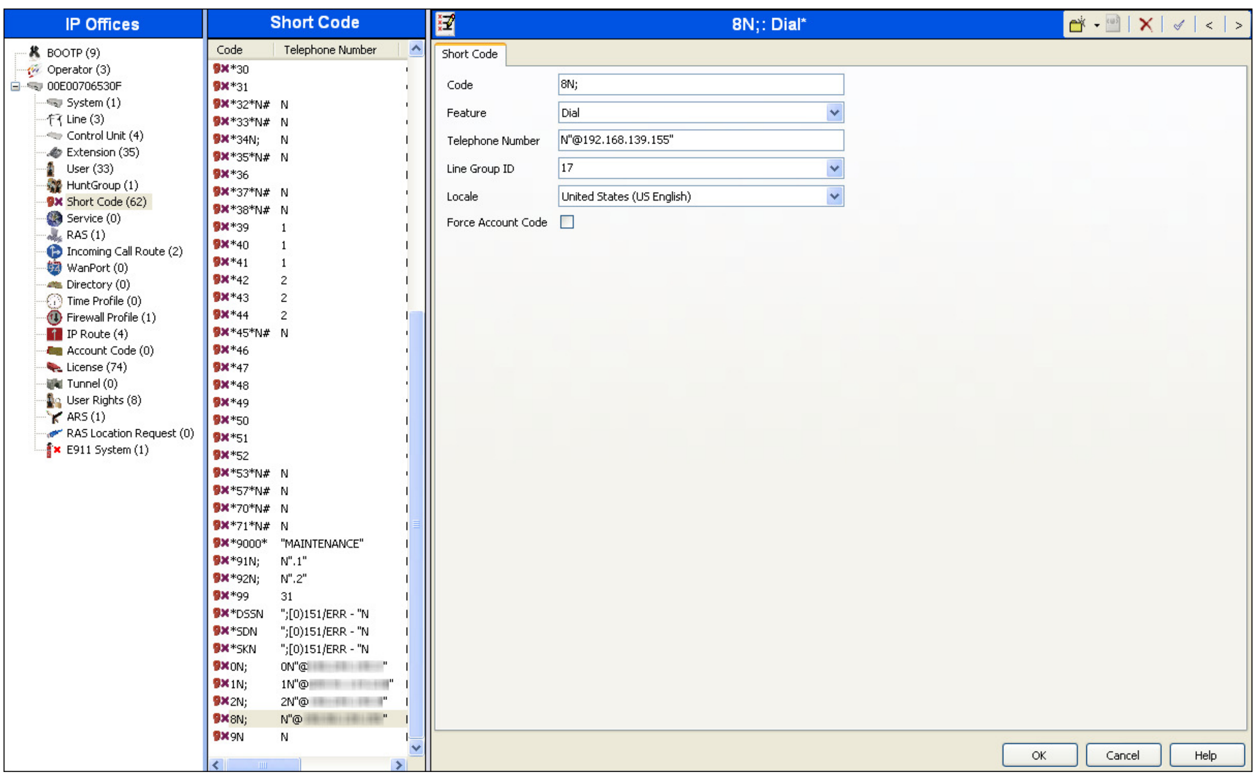
- Under the **Destinations** tab, enter "**.**" for the **Default Value**. This setting will allow the call to be routed to any destination with a value on its **SIP Name** field, entered on the **SIP** tab of that **User,** which matches the number present on the user part of the incoming Request URI.


IP Offices
BOOTP (9)
Operator (3)
00E00706530F
System (1)
Line (3)
Control Unit (4)
Extension (35)
User (33)
HuntGroup (1)
Short Code (62)
Service (0)
RAS (1)
Incoming Call Route (2)
WanPort (0)
Directory (0)
Time Profile (0)
Firewall Profile (1)
IP Route (4)
Account Code (0)
License (74)
Tunnel (0)
User Rights (8)
ARS (1)
RAS Location Request (0)
E911 System (1)
Incoming Call Route
Line Group ID
Incoming Number
0
17
17
Standard
Voice Recording
Destinations
TimeProfile
Destination
Fallback Extension
Default Value

5.10. Short Code

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**. On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

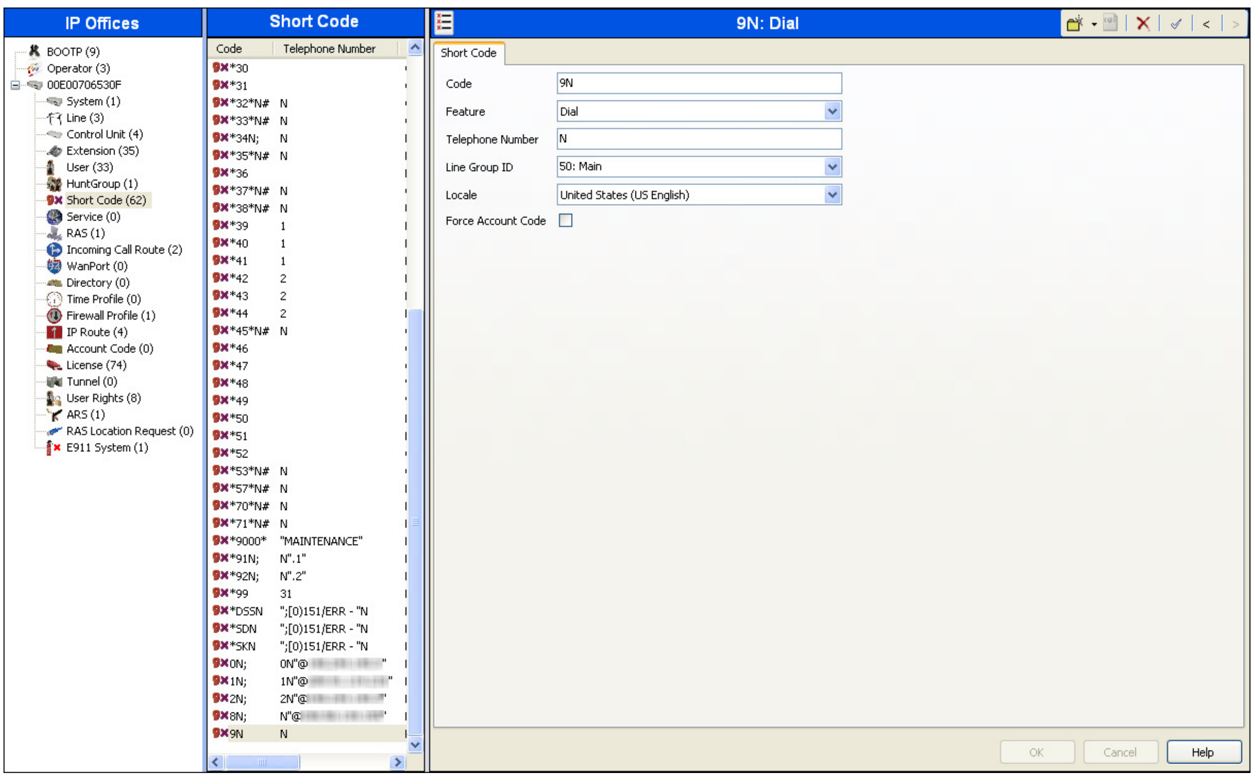
- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **8N;**. This short code will be invoked when the user dials 8 followed by any number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N"@192.168.139.155"**. 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 IP address 192.168.139.155 represents the IP address of the TSTT 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 5.6.3**. This short code will use this line group when placing outbound calls.
- Default values may be used for all other parameters.



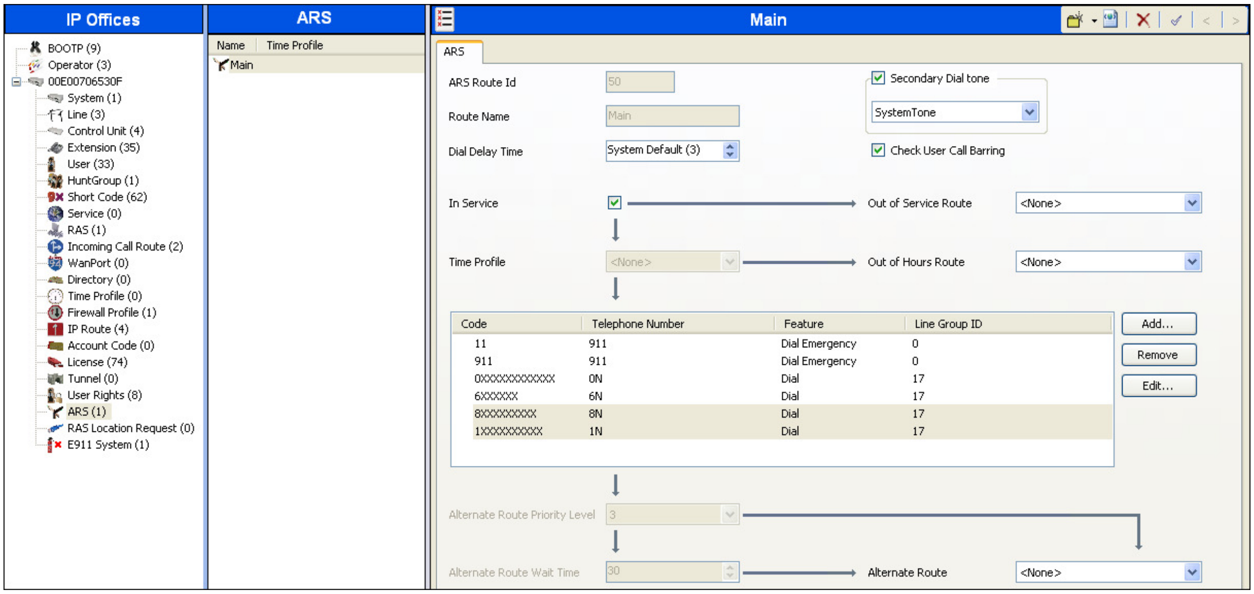
5.11. Automatic Route Selection

Optionally, Automatic Route Selection (ARS) can be used rather than the simple short code approach described above. With ARS, secondary dial tone can be provided after the access code. Other features like time-based routing criteria 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. ARS also facilitates a more granular treatment for different types of calls, and permits a more specific matching of the telephone number dialed following the access code. While detailed coverage of ARS is beyond the scope of these Application Notes, and alternate routing was not used in the reference configuration, this section includes some basic screen illustrations of the ARS settings used during the compliance test.

To create a short code to be used for ARS, right-click on **Short Code** in the Navigation Pane and select **New**. The screen below shows the short code **9N** created. Note that the semi-colon is not used here. In this case, when the Avaya IP Office user dials **9** plus any number **N**, instead of being directed to a specific Line Group ID, the call is directed to **Line Group 50: Main**, which is configurable via ARS.



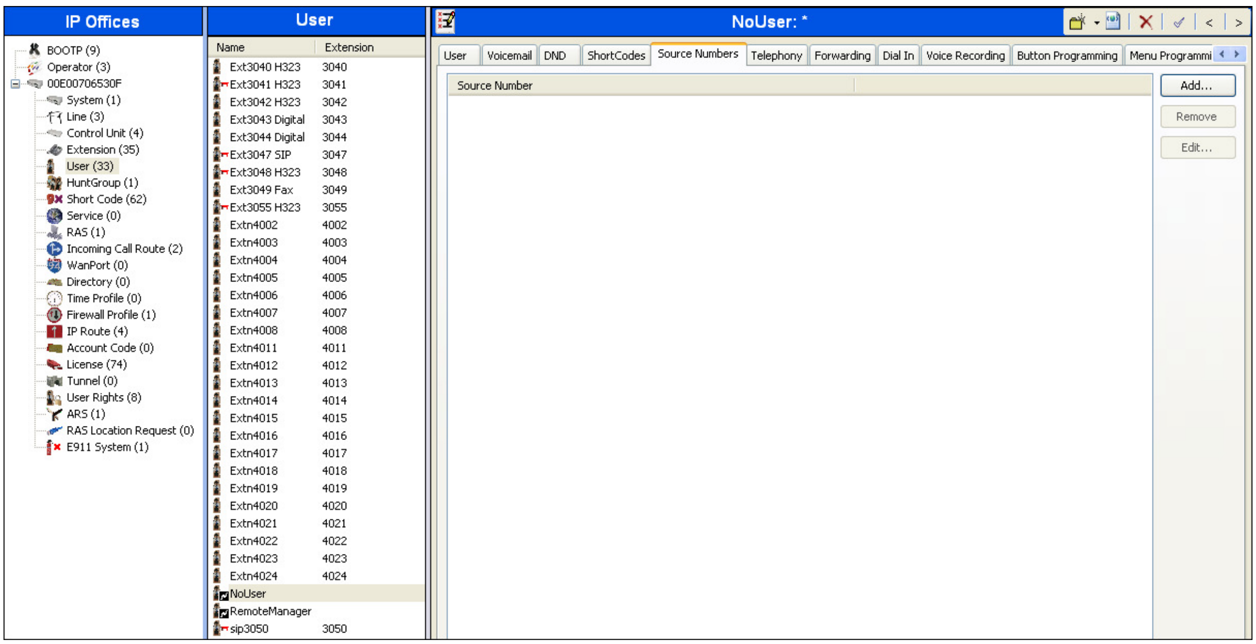
The following screen shows the example ARS configuration for the route **Main**. Note the sequence of **X**s used in the **Code** column of the entries to specify the exact number of digits to be expected, following the access code and the first digit on the string. This type of setting results in a much quicker response in the delivery of the call by the IP Office. The example below shows that for local calls, the user dialed 9, then 10 digit numbers starting with an 8. For calls to other area codes in the North American Numbering Plan, the user dialed 9, followed by 11 digits, starting with a 1.



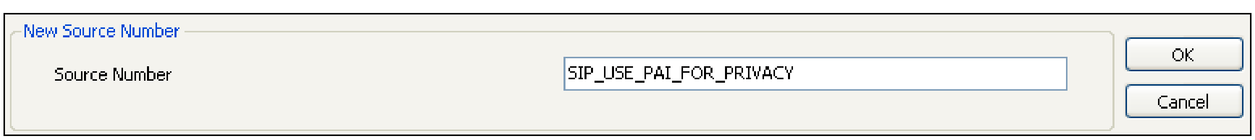
5.12. Privacy/Anonymous Calls

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

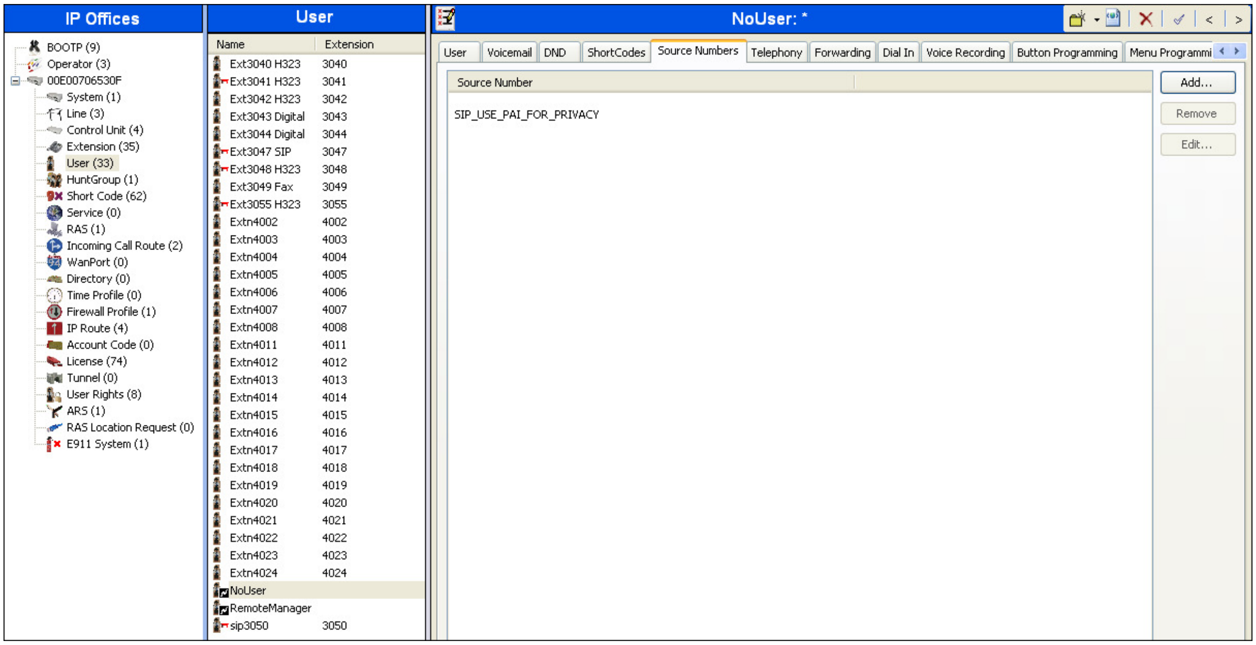
To configure Avaya IP Office to use PAI for privacy calls, navigate to **User → NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



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



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

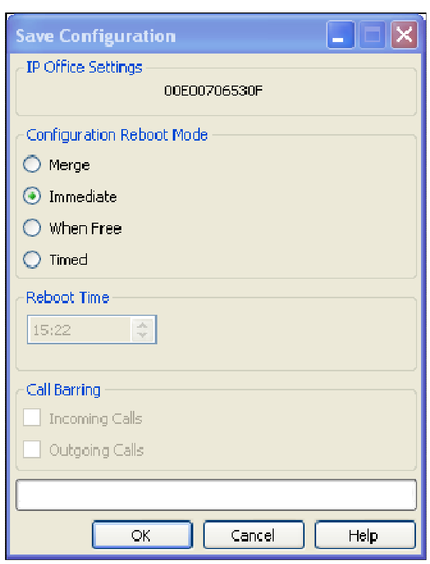


5.13. Save Configuration

When desired, send the configuration changes made in Avaya IP Office Manager to the Avaya IP Office server in order for the changes to take effect.

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

Once the configuration is validated, a screen similar to the following will appear, with either the **Merge** or the **Immediate** radio button chosen based on the nature of the configuration changes made since the last save. Note that clicking OK may cause a service disruption due to system reboot. Click OK if desired.



6. Telecommunications Services of Trinidad and Tobago SIP Trunking Configuration

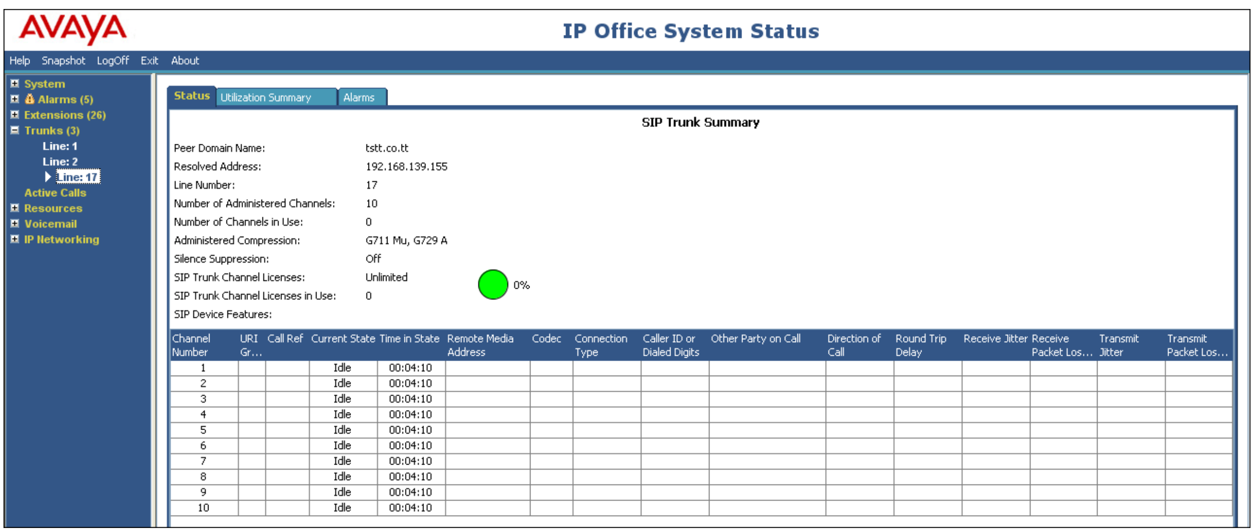
TSTT is responsible for the configuration of the SIP Trunk Service. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise. TSTT will provide the customer the necessary information to configure the Avaya IP Office SIP trunk connection, including:

- IP address of the TSTT SIP Proxy server.
- Supported codecs and order of preference.
- 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.

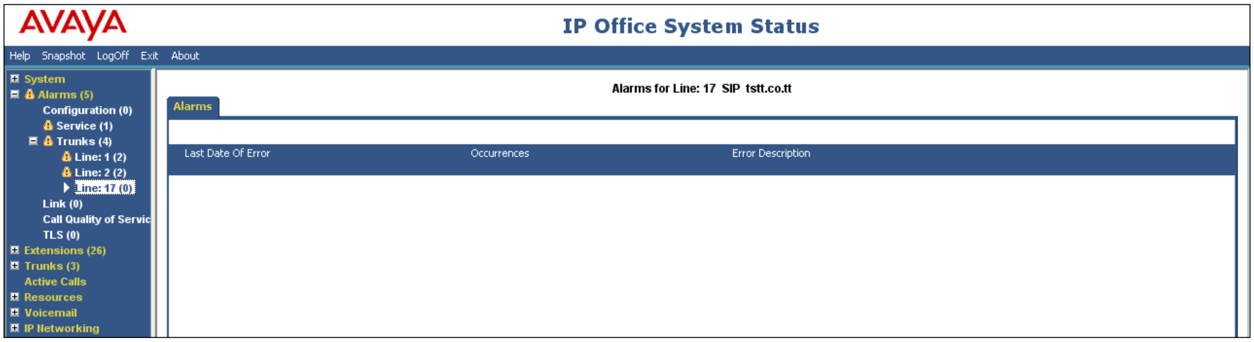
7. Verification Steps

The following steps may be used to verify the configuration:

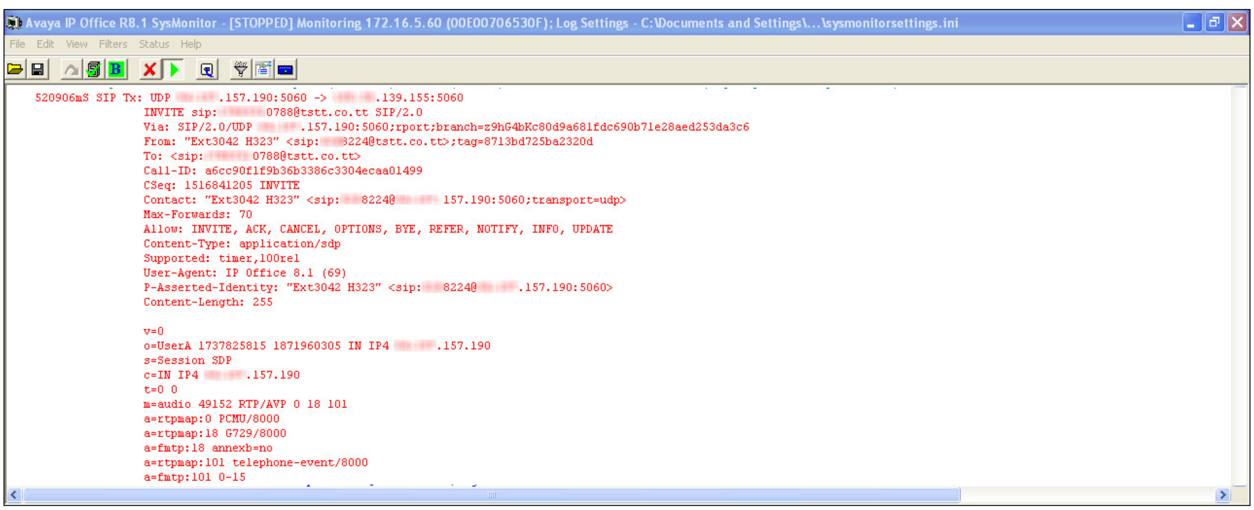
- Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application from **Start →Programs →IP Office →System Status** on the PC where Avaya IP Office Manager was installed (not shown). Log in using the appropriate credentials and select the SIP line of interest from the left pane. On the **Status** tab in the right pane, verify that the **Current State** is **Idle** for each channel (assuming there are no active calls).



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



- Verify that a phone connected to Avaya IP Office can successfully place a call to the PSTN with two-way audio.
- Verify that a phone connected to the PSTN can successfully place a call to an Avaya IP Office phone with two-way audio.
- The Avaya IP Office Monitor application can also be used to monitor and troubleshoot SIP signaling messaging between TSTT and IP Office. Launch the application from **Start →Programs →IP Office →Monitor** on the PC where Avaya IP Office Manager was installed. The sample screen below shows part of the messages on an outbound call.



8. Conclusion

These Application Notes describe the procedures required to configure SIP trunk connectivity between Avaya IP Office 8.1 and the Telecommunications Services of Trinidad and Tobago SIP Trunk Service, as shown in **Figure 1**.

Interoperability testing was completed successfully with the observations/limitations noted in **Section 2.2.**

9. Additional References

[1] IP Office 8.1, Installing IP500/IP500 V2 Document Number 15-601042 - Issue 27n, 22 August 2013, https://downloads.avaya.com/css/P8/documents/100162521

[2] IP Office Manager R8.1 FP1, Document Number 15-601011 - Issue 29v 0, Friday, August 30, 2013, https://downloads.avaya.com/css/P8/documents/100162522

[3] IP Office Release 8.1 Implementing Voicemail Pro, Document Number 15-601064 – Issue 8b, Tuesday, December 11, 2012, https://downloads.avaya.com/css/P8/documents/100163127

[4] IP Office 8.1 Using System Status, Document Number 15-601758 – Issue 07a, 24 May, 2013 https://downloads.avaya.com/css/P8/documents/100150298

[5] Avaya IP Office Knowledgebase, http://marketingtools.avaya.com/knowledgebase

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

Product documentation for the TSTT SIP Trunk Service is available from Telecommunications Services of Trinidad and Tobago.

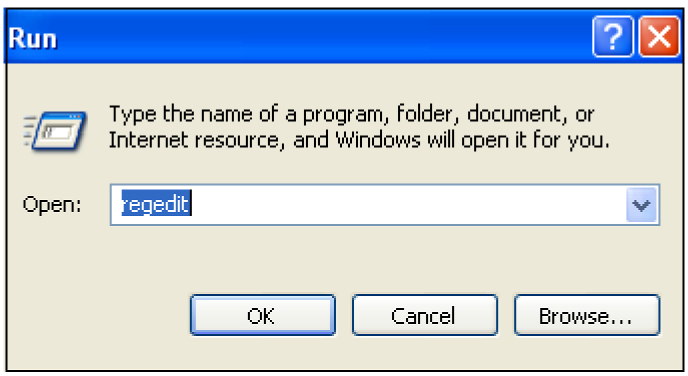
Appendix: SIP Line Template

Avaya IP Office Release 8.1 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.

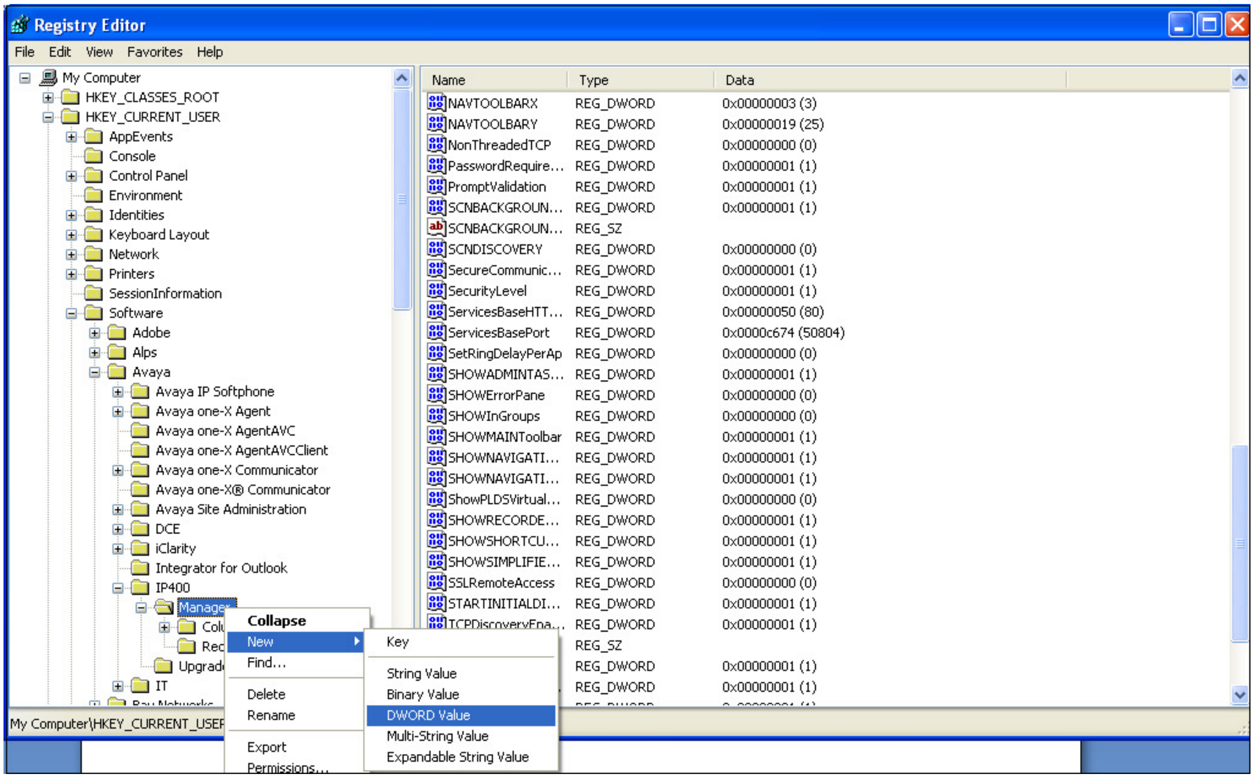
Not all of the configuration information is included in the SIP Line Template, therefore, it is critical that the SIP Line configuration be verified/updated after a template has been imported, and additional configuration be supplemented using **Section 5.6** in these Application Notes as a reference.

To create a SIP Line Template from the configuration described in these Application Notes, configure the parameters as described below.

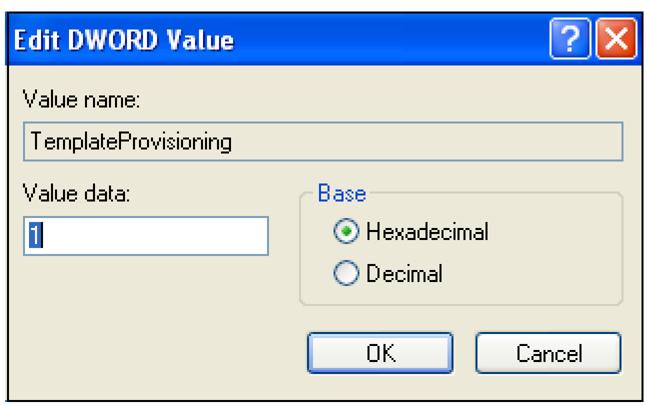
Use the Windows Registry Editor on the PC where Avaya IP Office Manager is installed to add a new **TemplateProvisioning** registry entry. Select **Start → Run**. Enter **regedit** in the **Open** box.



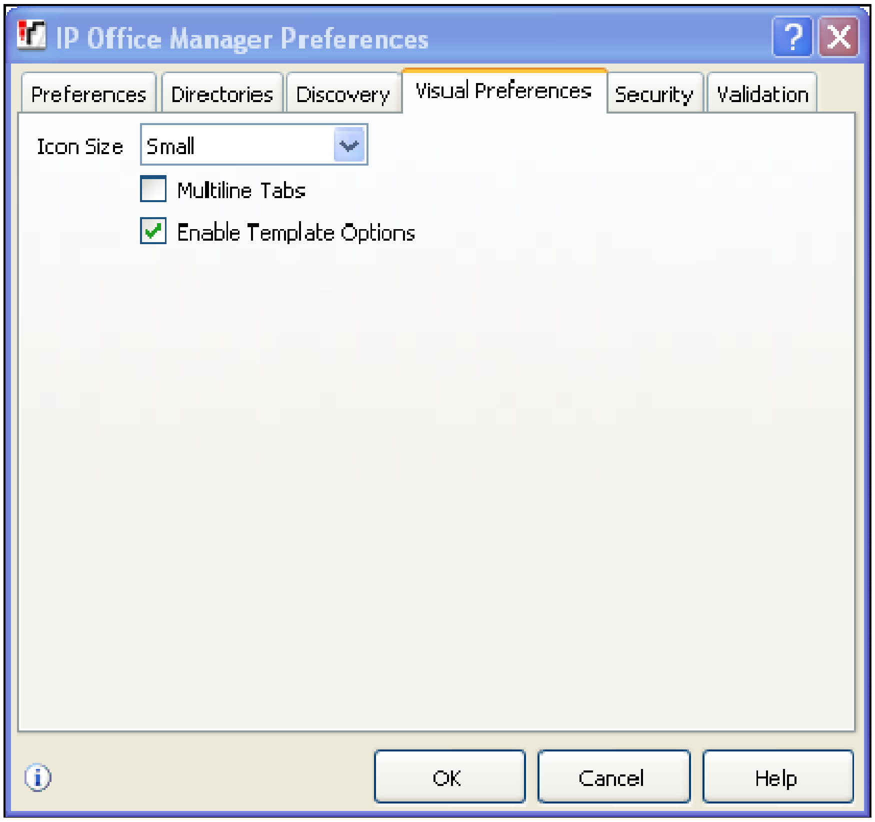
On the Registry Editor, navigate to **HKEY_CURRENT_USER → Software → Avaya →IP400**. Right click on **Manager** and select **New → DWORD Value**.



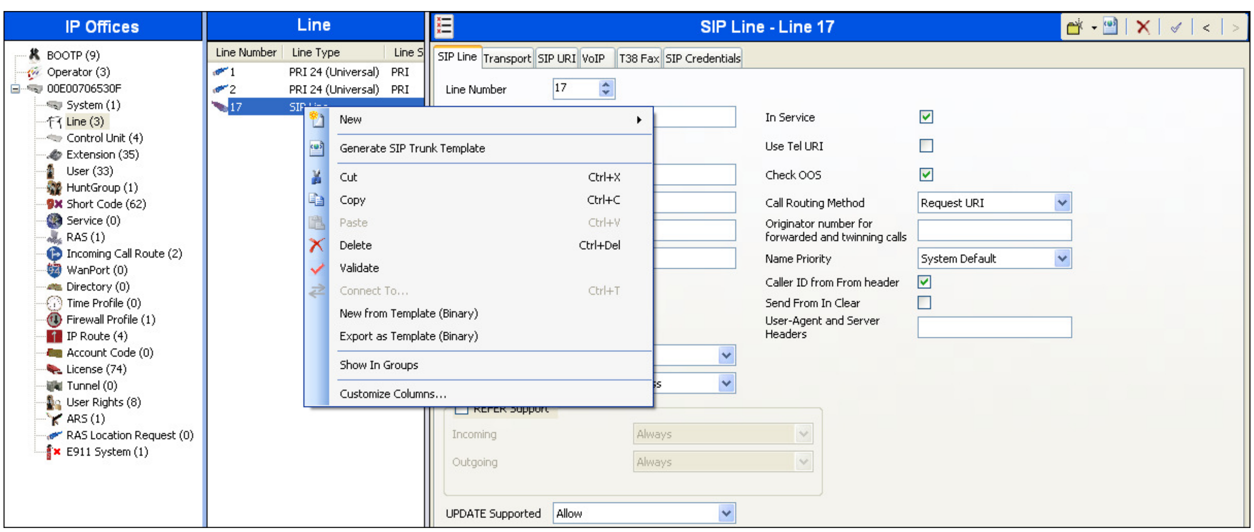
Right click the newly created entry and rename it to **TemplateProvisioning**. Double click the entry and change the value under **Value Data** from "**0**" to "**1**". Restart the PC.



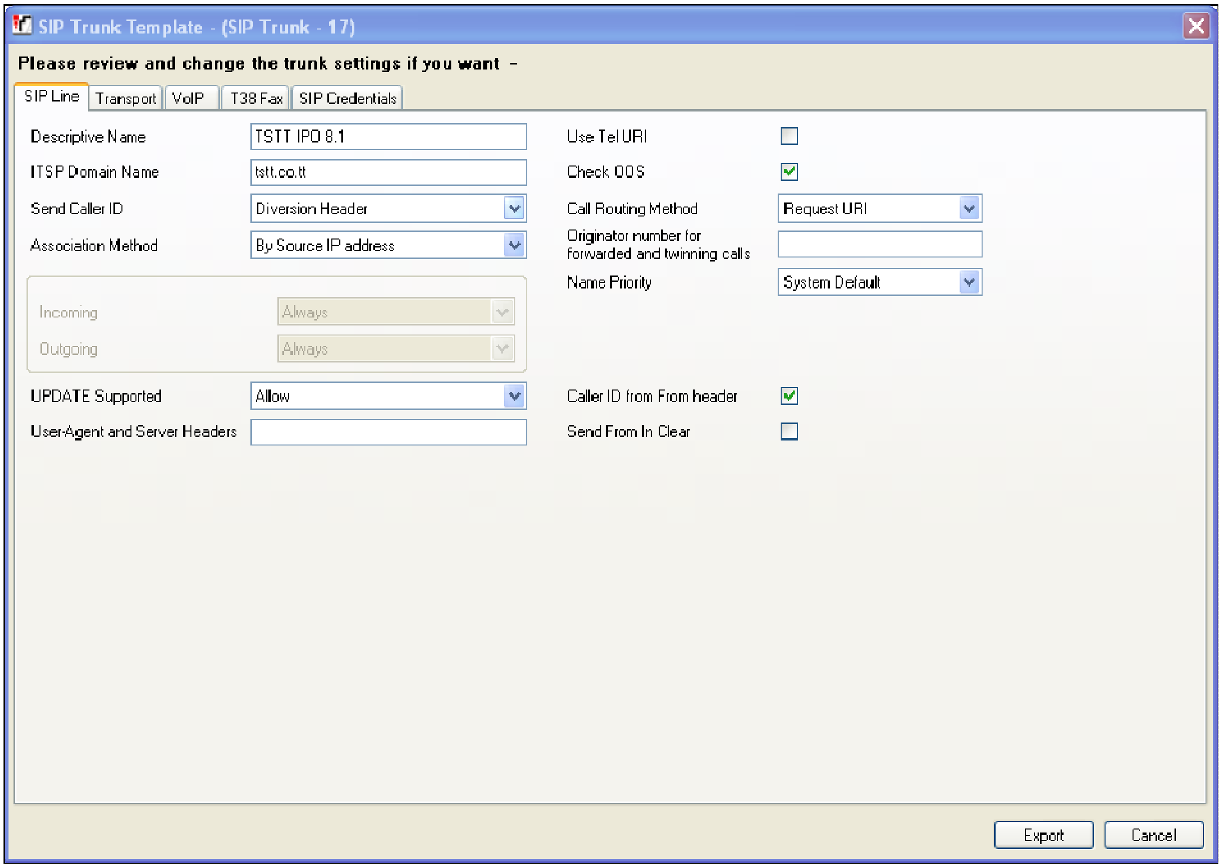
To enable template support in the IP Office Manager, select **File**, then **Preferences**. On the **Visual Preferences** tab, check the **Enable Template Options** box.



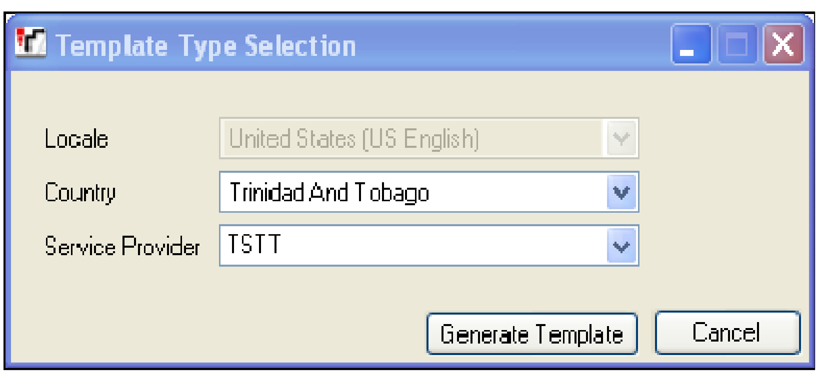
To create a SIP Line Template from the configuration, on the left Navigation Pane, right click the Sip Line (**17**), and select **Generate SIP Trunk Template**.



The trunk's settings are displayed as configured in **Section 5.6.** 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 following is the exported SIP Line Template file **TT_TSTT_SIPTrunk.xml**:

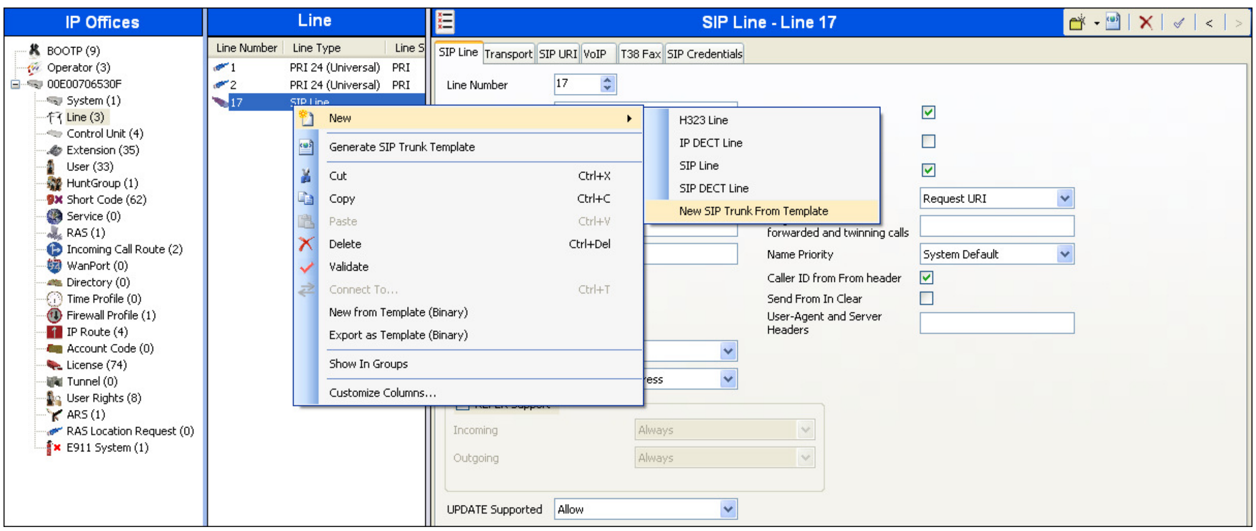
```
<?xml version="1.0" encoding="utf-8" ?>
<Template xmlns="urn:SIPTrunk-schema">
```

```
<TemplateType>SIPTrunk</TemplateType>
<Version>20130913</Version>
<SystemLocale>enu</SystemLocale>
<DescriptiveName>TSTT IPO 8.1</DescriptiveName>
<ITSPDomainName>tstt.co.tt</ITSPDomainName>
<SendCallerID>CallerIDDIV</SendCallerID>
<ReferSupport>false</ReferSupport>
<ReferSupportIncoming>1</ReferSupportIncoming>
<ReferSupportOutgoing>1</ReferSupportOutgoing>
<RegistrationRequired>false</RegistrationRequired>
<UseTelURI>false</UseTelURI>
<CheckOOS>true</CheckOOS>
<CallRoutingMethod>1</CallRoutingMethod>
<OriginatorNumber />
<AssociationMethod>SourceIP</AssociationMethod>
<LineNamePriority>SystemDefault</LineNamePriority>
<UpdateSupport>UpdateAllow</UpdateSupport>
<UserAgentServerHeader />
<CallerIDfromFromheader>true</CallerIDfromFromheader>
<PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
<ITSPProxy>192.168.139.155</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.711 ULAW 64K,G.729(a) 8K CS-ACELP</AdvCodecPref>
<CallInitiationTimeout>4</CallInitiationTimeout>
<DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
<VoipSilenceSupression>false</VoipSilenceSupression>
<ReinviteSupported>true</ReinviteSupported>
<FaxTransportSupport>FOIP_NONE</FaxTransportSupport>
<UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
<CodecLockdown>false</CodecLockdown>
<Rel100Supported>true</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>
  </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.

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.