Avaya Solution & Interoperability Test Lab

Application Notes for Configuring ThinkTel SIP Trunk Service with Avaya IP Office 10.1 - Issue 1.0

Abstract

These Application Notes describe the procedure for configuring ThinkTel Session Initiation Protocol (SIP) Trunking with Avaya IP Office.

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

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

ThinkTel 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 procedure for configuring Session Initiation Protocol (SIP) Trunking between service provider ThinkTel and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office Server Edition Release 10.1, Avaya Voicemail Pro, Avaya Communicator for Windows, Avaya Communicator for Web and Avaya H.323, SIP, digital, and analog endpoints.

The ThinkTel 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 to connect to ThinkTel SIP Trunking service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality tests listed in **Section 2.1**.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. . The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution. Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

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

2.1. Interoperability Compliance Testing

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

- Response to SIP OPTIONS queries.
- 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 long holding time call stability.
- Various call types including: local, long distance, international, outbound toll-free, operator service and directory assistance.
- Codec G.711U and G.729.
- 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.
- Fax T.38.
- Off-net call forwarding.
- Twinning to mobile phones on inbound calls.
- Avaya Communicator for Web Client (WebRTC).
- Avaya Communicator for Windows.

Note: Avaya Communicator for Web client (WebRTC) was tested as part of this solution. The configuration necessary to support Avaya Communicator for Web client is beyond the scope of these Application Notes and is not included in these Application Notes. For these configuration details, see **Reference [5]**.

Items, that are not supported, include the following:

- Call Redirection using REFER.
- Operator (0) and operator assist (0 + 10 digits) calls

2.2. Test Results

Interoperability testing of ThinkTel with the Avaya SIP-enabled enterprise solution was completed with successful results for all test cases with the exception of the observations and limitations described below:

- **OPTIONS** – ThinkTel responded to SIP OPTIONS from Avaya with 401 unauthorized error code, but it did not affect the operation of the SIP trunk.
- **Fax T.38** – When performing outbound T.38 fax, Avaya system sent re-INVITE to switch to fax t.38 codec, ThinkTel system responded with error code “488 Not Acceptable Here”. Fax t.38 does work and completed. There is no user impacted. Issue has been resolved by ThinkTel.

2.3. Support

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

For technical support on ThinkTel SIP Trunking, contact ThinkTel at http://www.thinktel.ca.

3. Reference Configuration

Figure 1 below illustrates the test configuration. The test configuration shows an enterprise site connected to ThinkTel 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 Server Edition Primary with the Avaya IP Office 500 V2 Expansion System which provides connections for 16 digital stations 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 LAN port of Avaya IP Office is connected to the enterprise LAN while the WAN port is connected to the public IP network via Avaya SBCE. Endpoints include an Avaya 9600 Series IP Telephone (with H.323 firmware), Avaya 11x0 Series IP Telephone (with SIP firmware), an Avaya 9508 Digital Telephone, an Avaya Symphony 2000 Analog Telephone and an Avaya IP Office Softphones. A separate Windows PC runs Avaya IP Office Manager to configure and administer the Avaya IP Office.

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

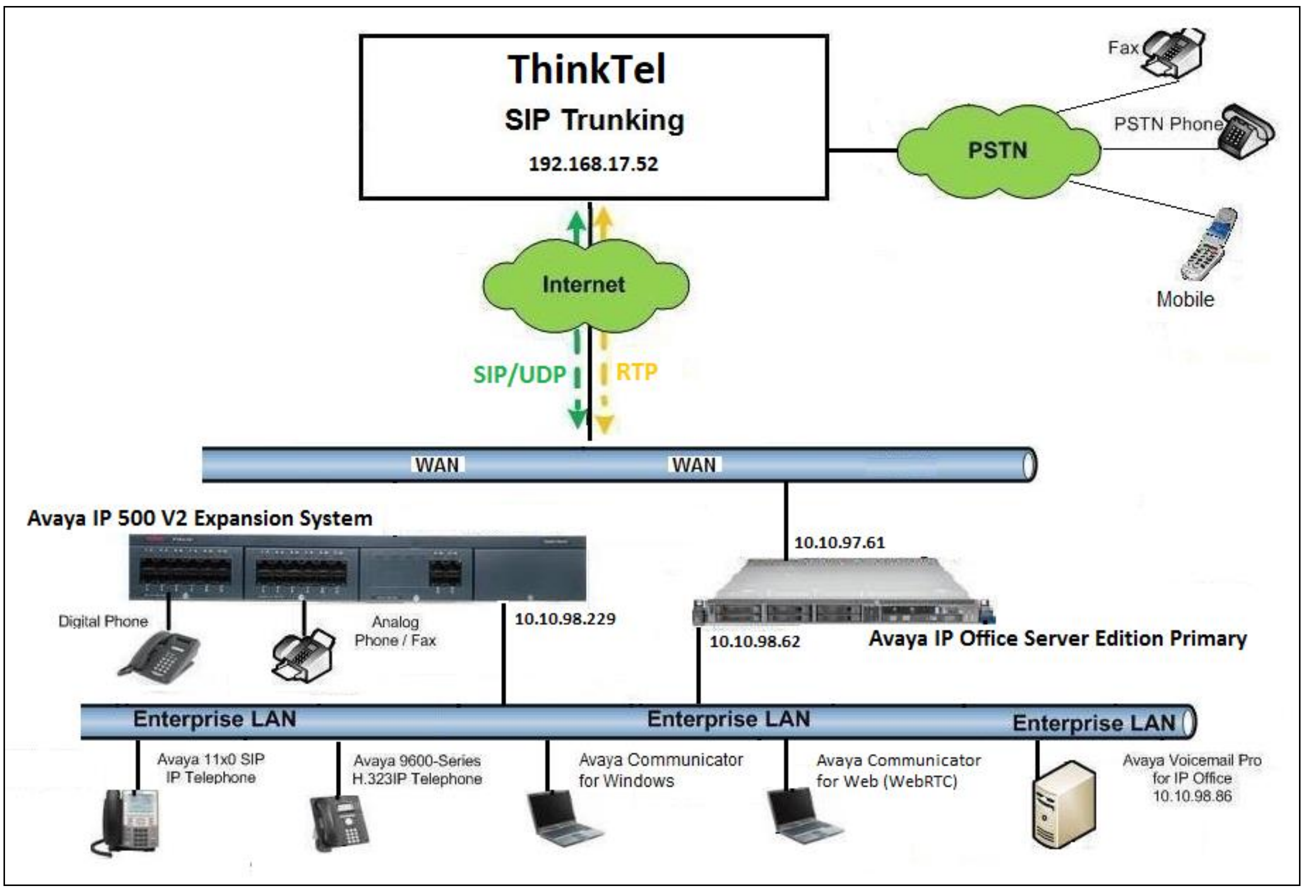


Figure 1: Test Configuration for Avaya IP Office with ThinkTel SIP Trunking Service

For the purposes of the compliance test, Avaya IP Office users dialed a short code of 6 + N digits to send digits across the SIP trunk to ThinkTel. The short code of 6 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to ThinkTel and no digit manipulation programing was required. 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. Avaya IP Office was configured to send 10 digits in the From field. ThinkTel SIP Trunking would send 10 digits in the Request URI and the To field of inbound SIP INVITE messages.

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 Server Edition	10.1.0.0.0 build 237
Avaya IP Office 500v2 (Expansion)	10.1.0.0.0 build 237
Avaya IP Office Manager	10.1.0.0.0 build 237
Avaya Voicemail Pro for IP Office	10.1.0.0.0 build 237
Avaya 11x0 IP Telephone (SIP)	SIP11x0e04.04.23.00
Avaya 9621G IP Telephone (H.323)	6.6401
Avaya Communicator for Windows	2.1.4.84
Avaya Communicator for Web (WebRTC)	1.0.17.1725
Avaya Digital Telephone (9508)	0.45
Avaya Symphony 2000 Analog Telephone	N/A
ThinkTel SIP Trunking Service Components	
Component	Release
Metaswitch (SIP Server)	8.1
Proxy Server Opensips	1.11.5

Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with all configurations of IP Office Server Edition.

5. Configure IP Office

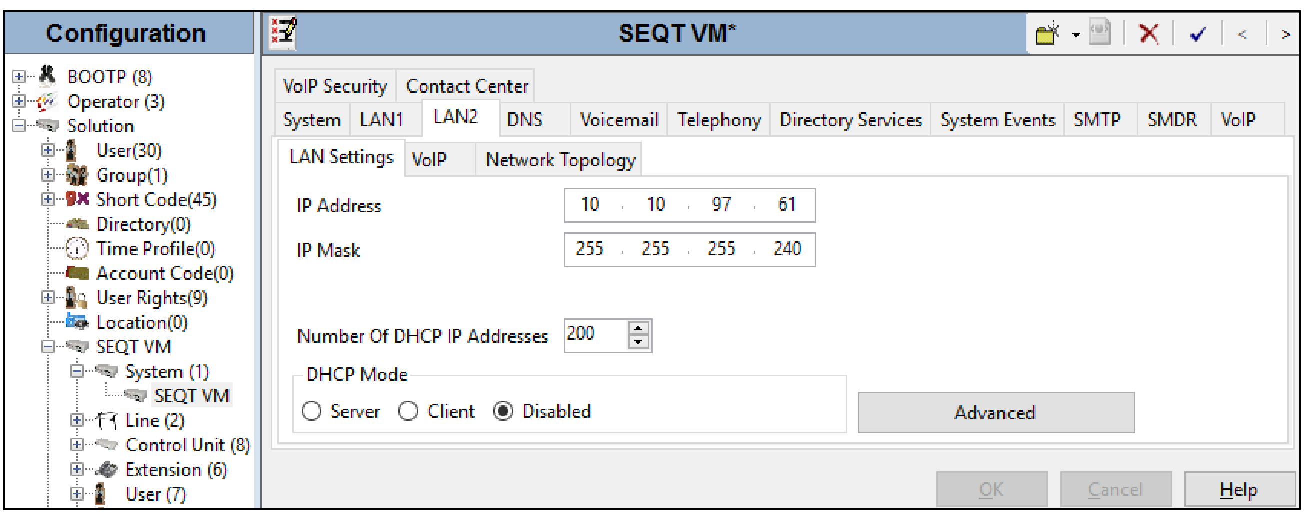
This section describes the Avaya IP Office configuration to support connectivity to ThinkTel SIP Trunking service. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** 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 also be customized using the **View** menu. In some 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. Some of these panes will be referenced in 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 LAN interface to the enterprise site) is assumed to be already in place.

5.1. LAN Settings

In the sample configuration, the **SEQT VM** was used as the system name and the LAN2 connects the Avaya IP Office to the WAN.

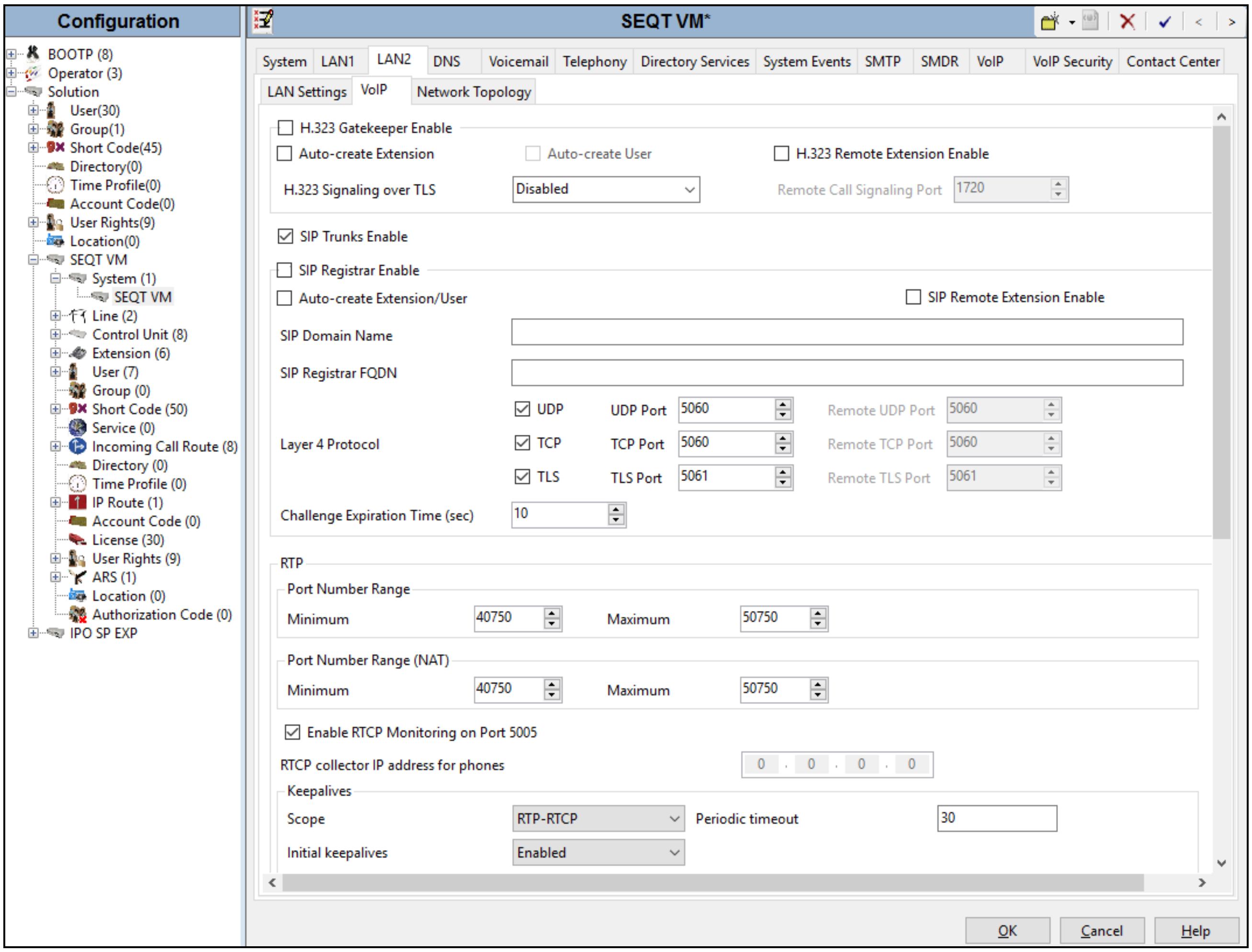
To access the LAN settings, first navigate to **System (1) → SEQT VM** in the **Navigation** and then navigate to the **LAN2 → LAN Settings** tab in the **Details** pane.

- Set the **IP Address** field to the IP address assigned to the 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.
- Click **OK**.



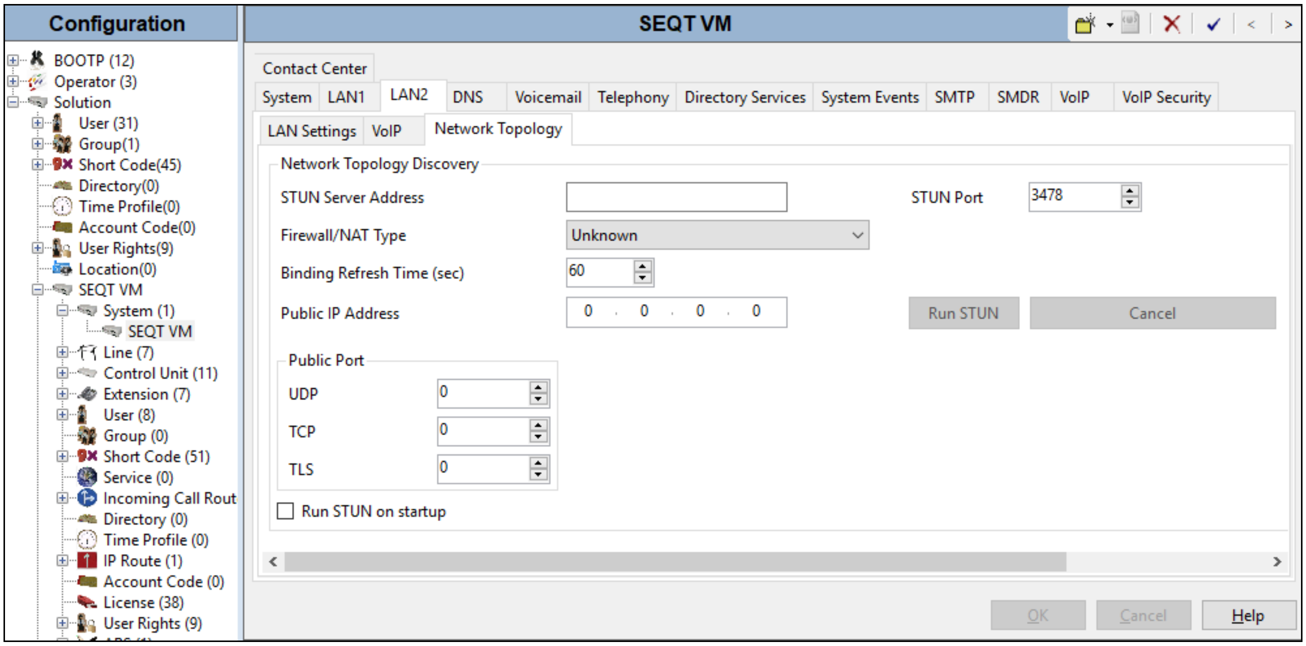
Select the **VoIP** tab as shown in the following screen.

- The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to ThinkTel.
- The **Layer 4 Protocol,** check the **UDP**, **TCP** and **TLS** boxes. Then set **UDP** and **TCP Ports** to ***5060***, and **TLS port** to ***5061***.
- **Enable RTCP Monitoring on Port 5005** and **Keepalives** should be set as shown in capture below.
- All other parameters should be set according to customer requirements**.**
- Click **OK**.



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 ***Unknown***. With this configuration, **STUN** will not be used.
- Set **Binding Refresh Time (seconds)** to ***60***. This value is used as one input to determine the frequency at which IP Office will send SIP OPTIONS messages to the service provider.
- All other parameters should be set according to customer requirements.
- Click **OK**.

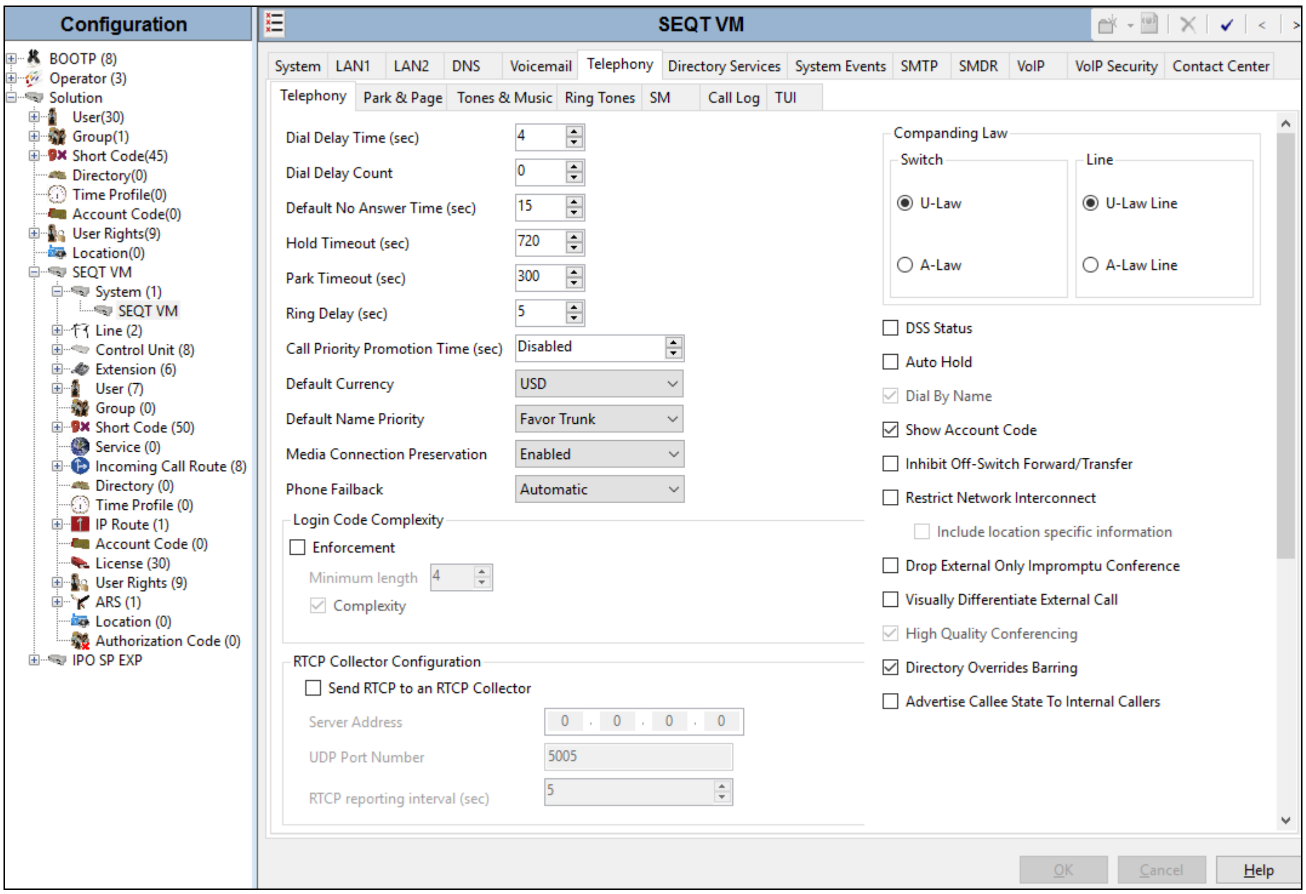


In the compliance test, the LAN1 interface was used to connect the Avaya IP Office to the enterprise site IP network. The LAN1 interface configuration is not directly relevant to the interface with ThinkTel SIP Trunking service, and therefore is not described in these Application Notes.

5.2. System Telephony Settings

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 transfer to the PSTN via the service provider across the SIP trunk.
- Uncheck the **Drop External Only Impromptu Conference** box to allow the host of the conference leaving the active call without forcing other parties of the conference.
- Other parameters are left at default.
- Click **OK**.



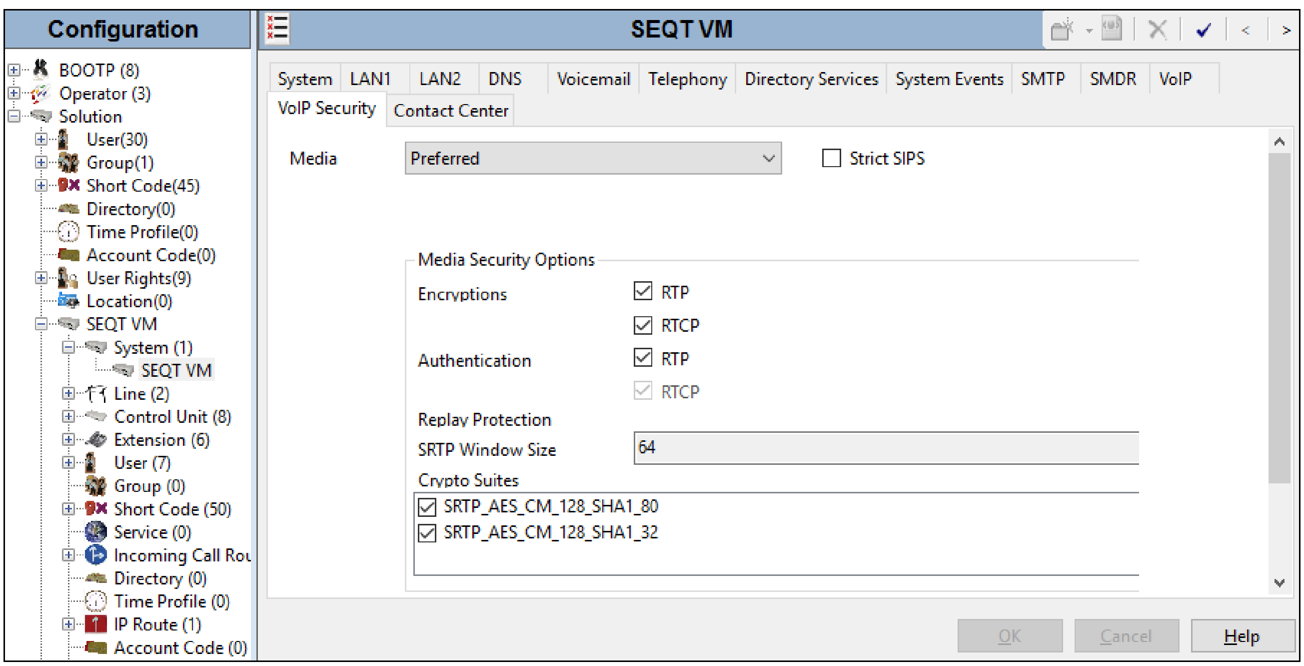
5.3. VoIP Security Settings

When enabling SRTP on the system, the recommended setting is Preferred. In this scenario, IP Office uses SRTP if supported by the other end, and otherwise uses RTP. If the Enforced setting is used, and SRTP is not supported by the other end, the call is not established.

Individual SIP lines and extensions have media security settings that can override system level settings. This can be used for special cases where the trunk or extension setting must be different from the system settings.

In the compliance testing, **Preferred** is set at system, trunk and extension levels to allow the system to fall back to non-secure media in case there is an issue with SRTP. This would help to avoid a blackout situation within the enterprise network. In some specific deployments, if supported, **Enforce** is set at the trunk level to ensure the secured communication over the public internet using both signaling (TLS) and media (SRTP). Navigate to **System → VoIP Security** tab and configure as follow:

- Select ***Preferred*** for **Media Security.** The system attempts to use secure media first and if unsuccessful, falls back to non-secure media within Avaya IP Office system.
- Check **RTCP** check-box.
- Other parameters are left as default.
- Click **OK**.



5.4. Administer SIP Line

A SIP line is needed to establish the SIP connection between IP Office and ThinkTel SIP Trunking service. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.4.1** to create the SIP Line from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:

- IP addresses.
- SIP Credentials (if applicable).
- SIP URI entries.
- Setting of the **Use Network Topology Info** field on the Transport tab.

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Section 5.4.2**.

Also, the following SIP Line settings are not supported on Basic Edition:

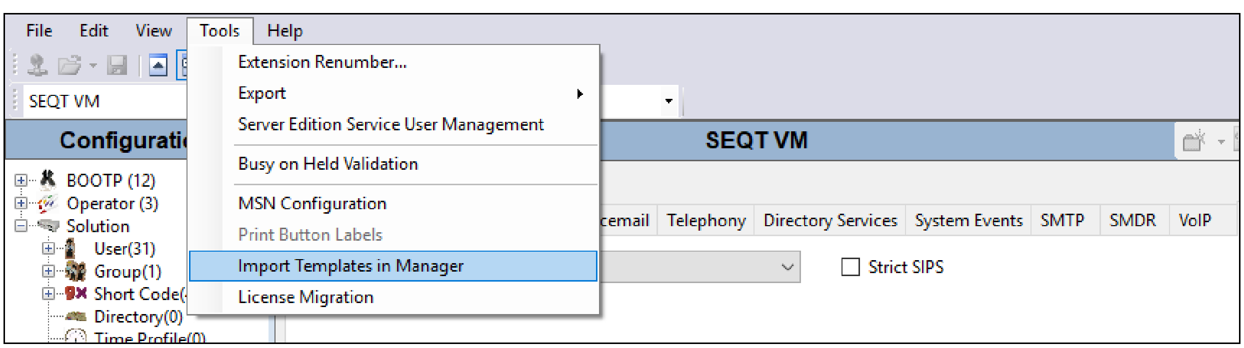
- SIP Line – Originator number for forwarded and twinning calls.
- Transport – Second Explicit DNS Server.
- SIP Credentials – Registration Required.

Alternatively, a SIP Line can be created manually. To do so right-click **Line** in the Navigation Pane and select **New → SIP Line**, then follow the steps outlined in **Section 5.4.2**.

5.4.1. Create SIP Line from Template

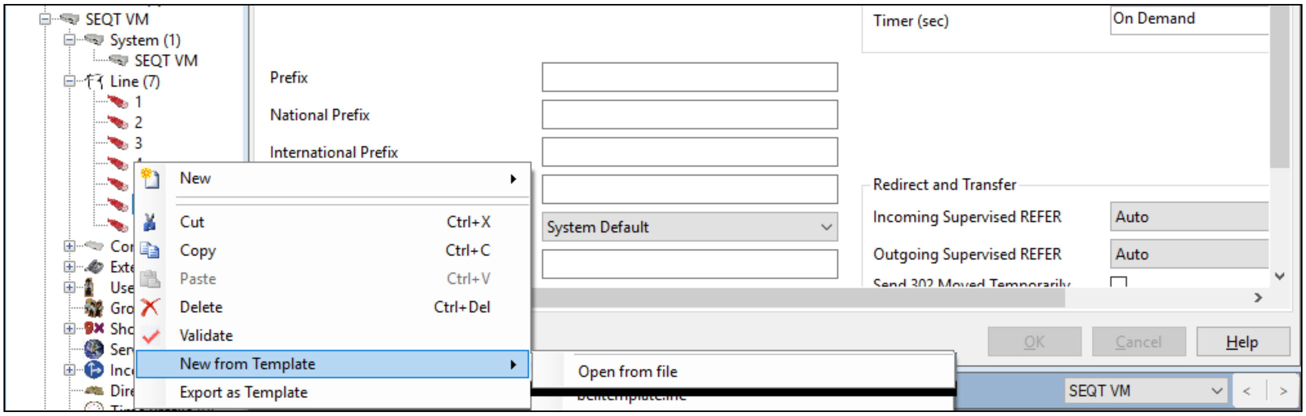
1. Copy the template file to the computer where IP Office Manager is installed. The template file is **TTELIPO101.xml**. The file name is important in locating the proper template file in **Step 4**.

2. Import the template into IP Office Manager.
   From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory. The default template location is **C:\Program Files\Avaya\IP Office\Manager\Templates**.

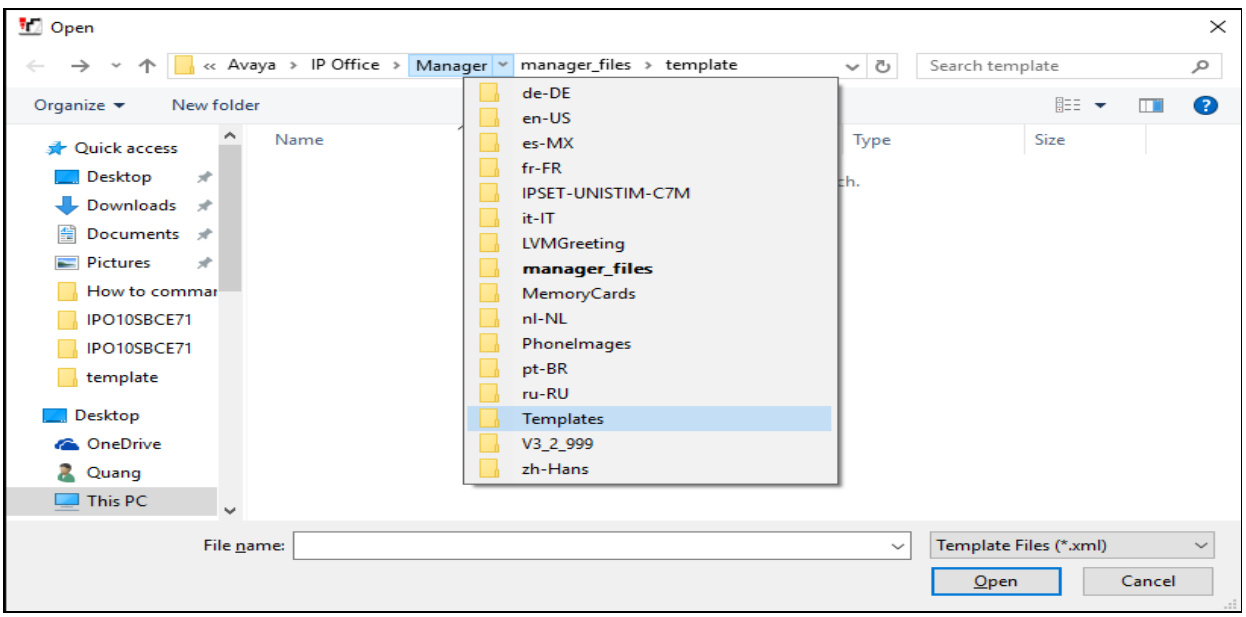


In the pop-up window that appears (not shown), select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window will appear (not shown) stating success or failure. Then click **OK** (not shown) to continue. If preferred, this step may be skipped if the template file is copied directly to the IP Office template directory.

3. To create the SIP Trunk from the template, right-click on **Line** in the Navigation Pane, then navigate to **New → New from Template**. Select the imported file.



4. On Open pop-up windows, Navigate to **Manager → Templates**, make sure **Template File (.xml)** is the file type selected. Then select the file **TTELIPO101.xml**. Click **Open** and **OK** (not shown).

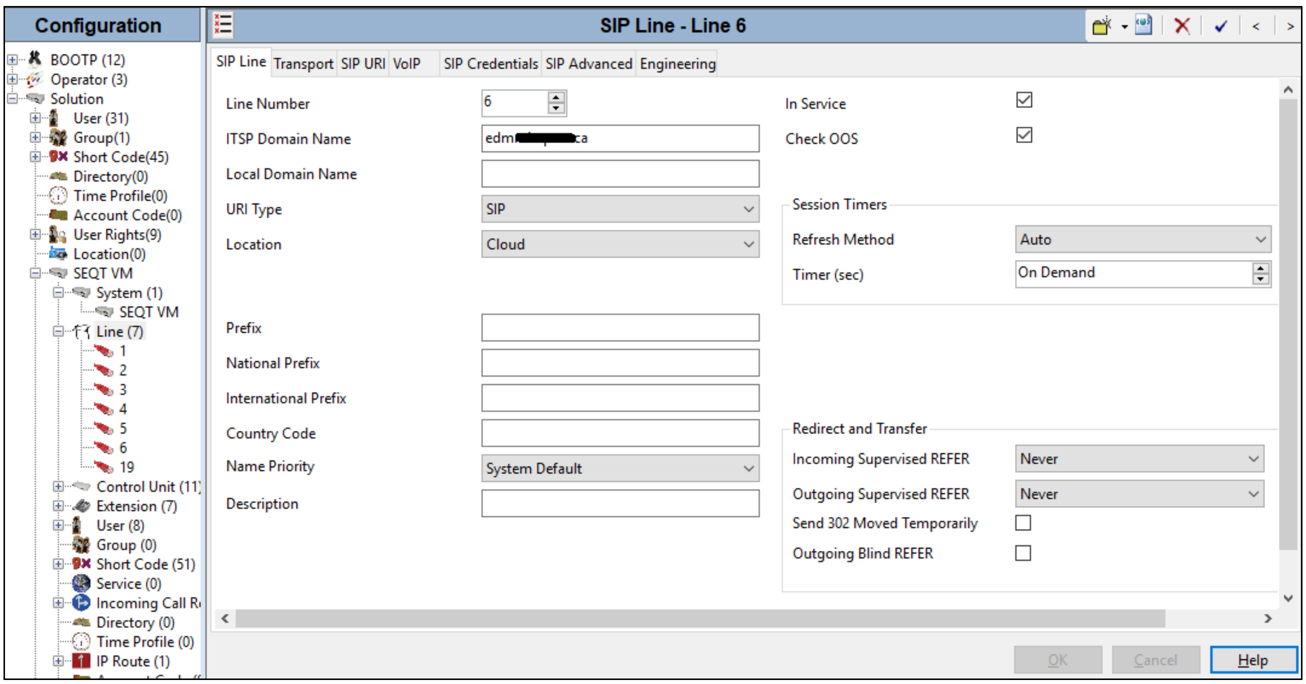


5. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Section 5.4.2**.

5.4.2. Create SIP Line Manually

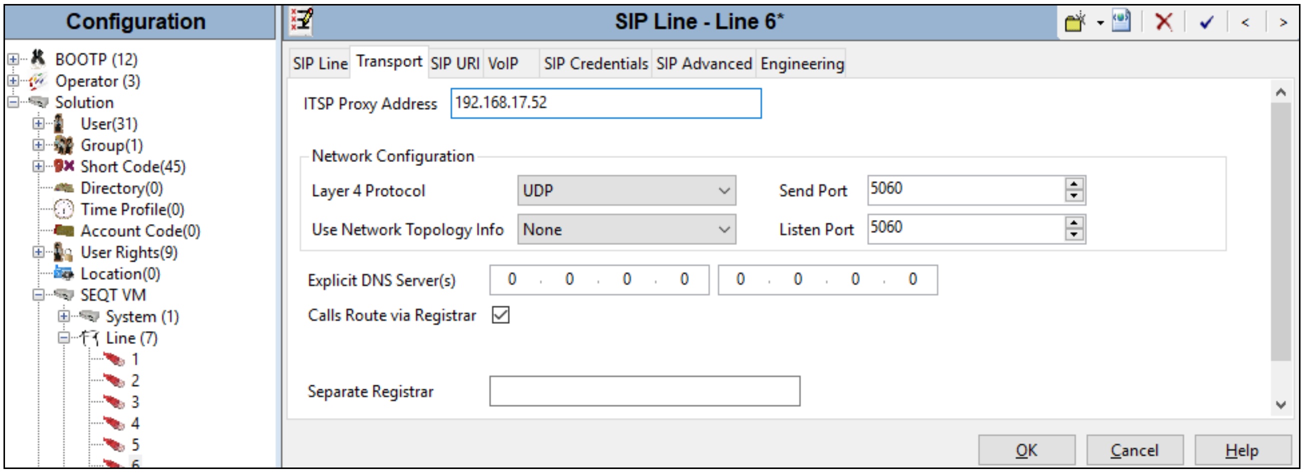
To create a SIP line, begin by navigating to **Line** in the left **Navigation** pane and then right click to select **New → SIP Line**. On the **SIP Line** tab in the **Details** pane, configure the parameters as shown below:

- Set **ITSP Domain Name** to the ITSP domain so that IP Office uses this domain as 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.
- **Incoming Supervised REFER** is set to ***Never*** to allow IP Office to support call transfer using re-INVITE only.
- **Outgoing Supervised REFER** is set to ***Never*** to allow IP Office to support call transfer using re-INVITE only.
- Other parameters are set as default values.
- Click **OK**.



Select the **Transport** tab and enter the following information.

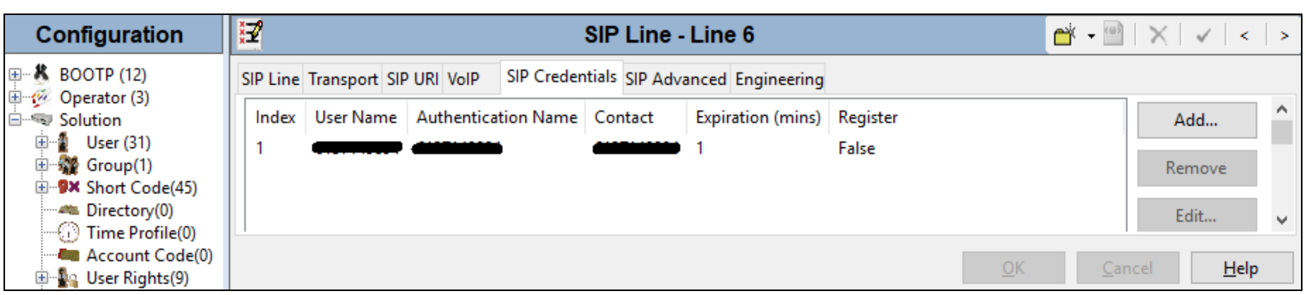
- The **ITSP Proxy Address** is set to connect to the service provider trunk IP address.
- **Layer 4 Protocol** is set to ***UDP***.
- **Send Port** is set to the port number of IP Office, ***5060***.
- **Use Network Topology Info** parameter is set to ***None***.
- Other parameters retain default values in the screen below.
- Click **OK**.



A SIP Credentials entry must be created for SIP trunking registration and Digest Authentication that are used by ThinkTel SIP trunking service to authenticate calls from the enterprise to the PSTN. To create a SIP Credentials entry, first select the **SIP Credentials** tab. Click the **Add** button and the **New SIP Credentials** 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 bottom of the screen, the Edit SIP Credentials area will be opened. Screen capture below shows an example of newly created SIP credential. The entry was created with the parameters shown below:

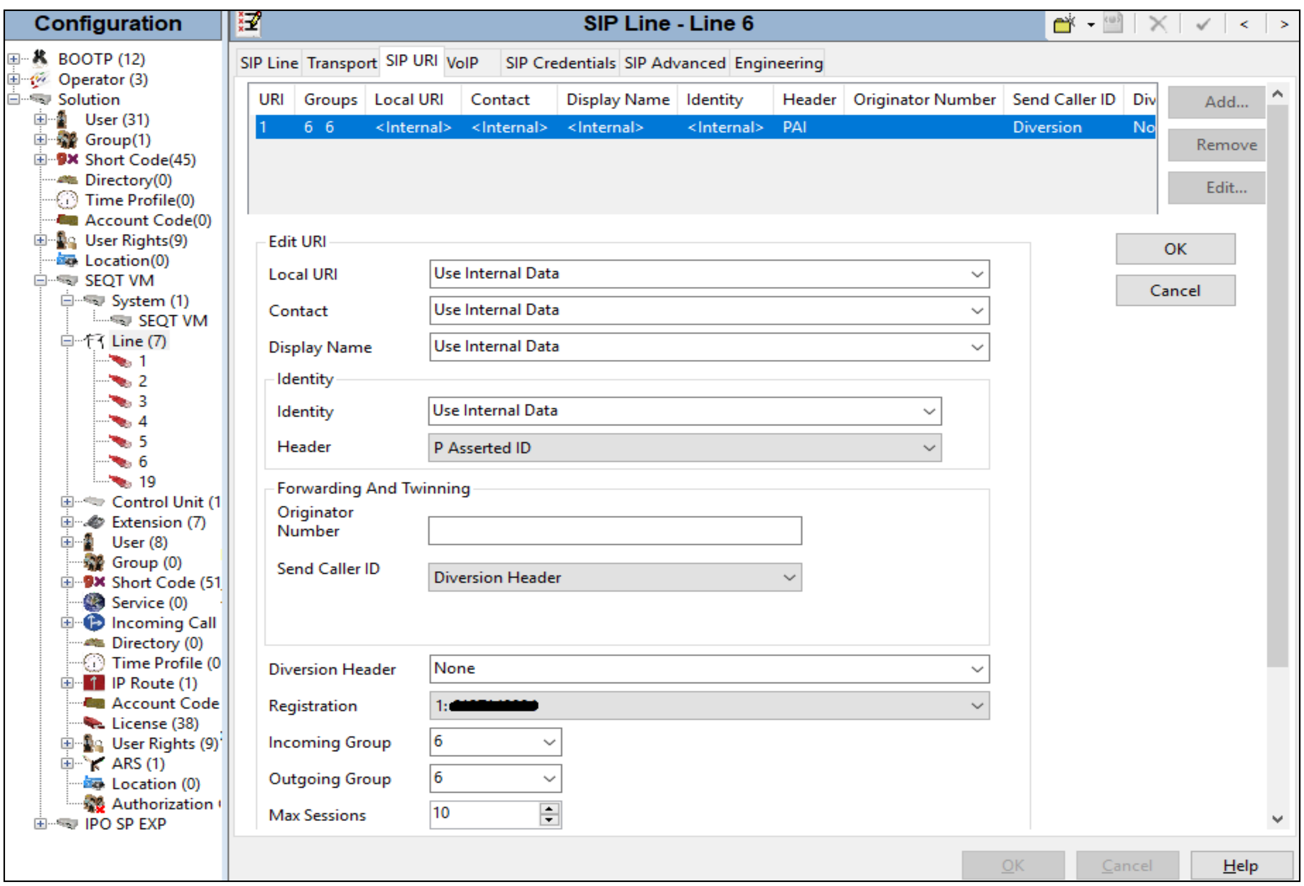
- Set **User name, Authentication Name** and **Contact** to the value provided by the service provider.
- Set **Password** and **Confirmed Password** to the value provided by the service provider.
- The **Expiration (mins)** is set to ***1***.
- Ensure **Registration required** check-box is un-check.

Click **OK.**



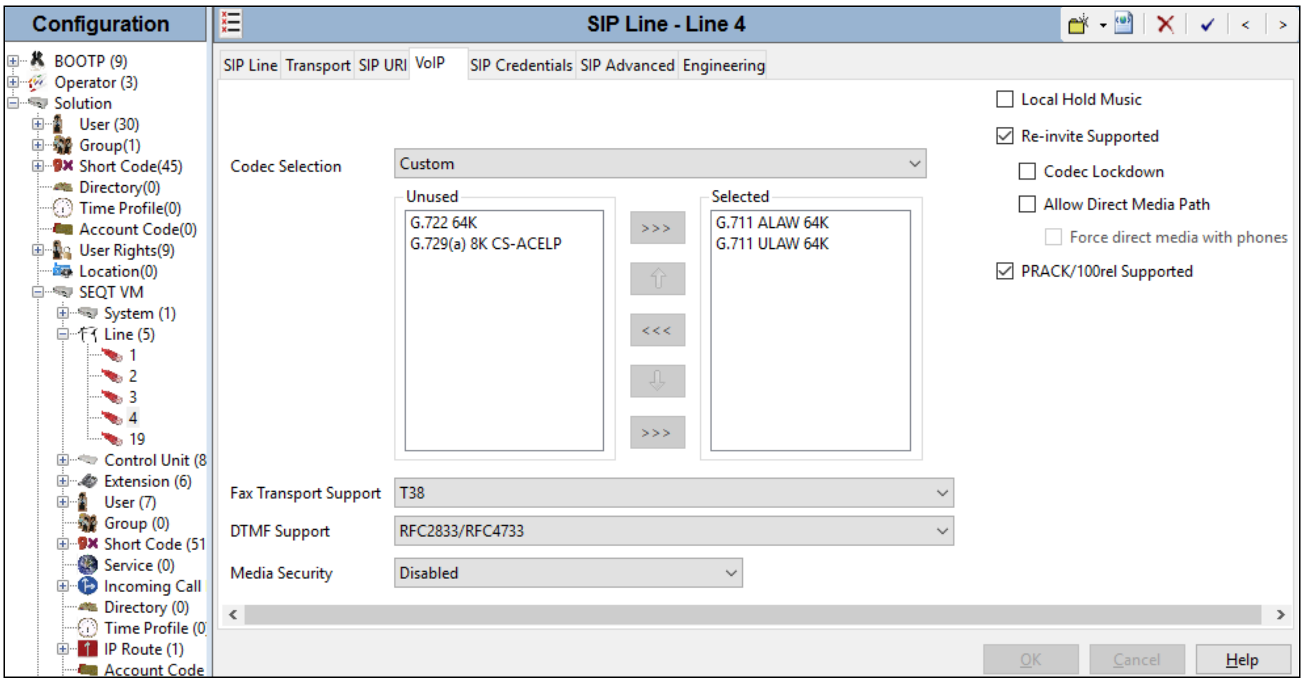
A **SIP URI** entry **1** is created to match incoming numbers that IP Office will accept on this line. Select the **SIP URI** tab, click **Add** button and then **New URI** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit…** button. In the example screen below, a previously configured entry is edited. For the compliance test, a single SIP URI entry was created that matched any DID number assigned to an 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 to use the number that IP Office is using to make the call as the From field. The number will be aligned with IP Office internal number schema.
- Set **Identity** to ***Use Internal Data*** and **Header** to ***P Asserted ID*** for **Identity**.
- Set **Sent Caller ID** to ***Diversion Header*** for **Forward and Twinning**.
- Set **Diversion Header** to ***None***.
- For **Registration**, select the account credentials previously configured on the line's **SIP Credentials** tab.
- 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 ***6*** was defined that only contains this line (line 4).
- Set **Max Sessions** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Other parameters retain default values and or set according customer requirements.
- Click **OK**.



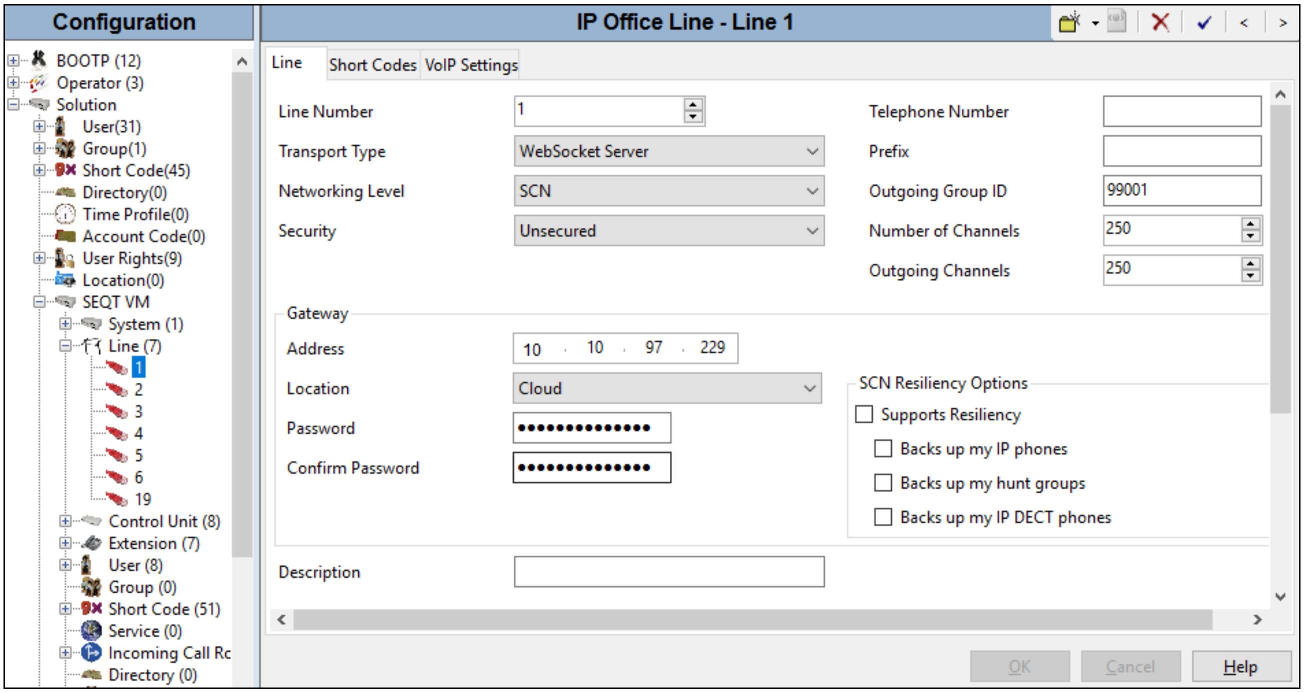
Select the **VoIP** tab to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- The **Codec Selection** can be selected by choosing ***Custom*** from the pull-down menu, allowing an explicit ordered list of codecs to be specified.
- Select ***G.711 ULAW*** and ***G.729(a) 8K CS-ACELP*** codecs supported by the ThinkTel SIP Trunking service, in the Session Description Protocol (SDP) offer.
- Set **Fax Transport Support** to ***T38*** from the pull-down.
- Set the **DTMF Support** field to ***RFC2833/RFC4733*** from the pull-down menu. This directs IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Check the **Re-invite Supported** box.
- Check the **PRACK/100rel Supported** box.
- Default values may be used for all other parameters.
- Click **OK**.

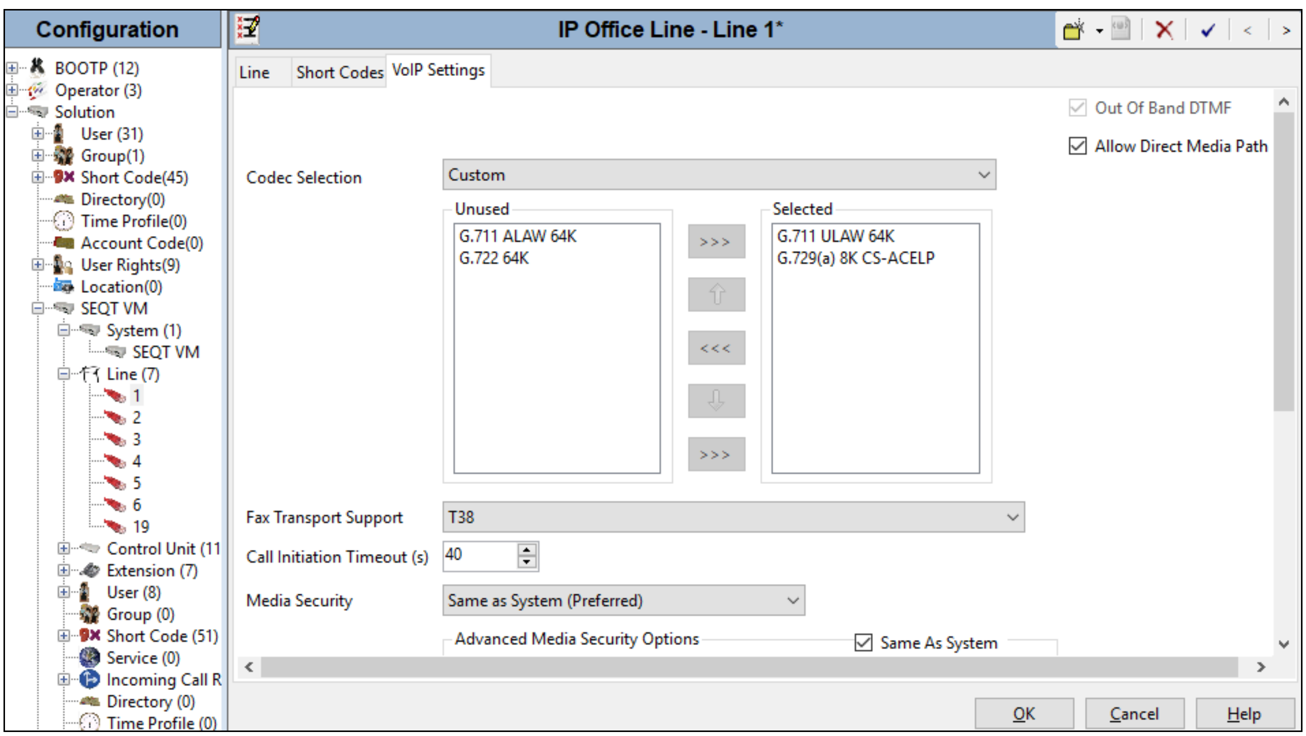


5.5. IP Office Line Server Edition

The IP Office line on Server Edition is created below.

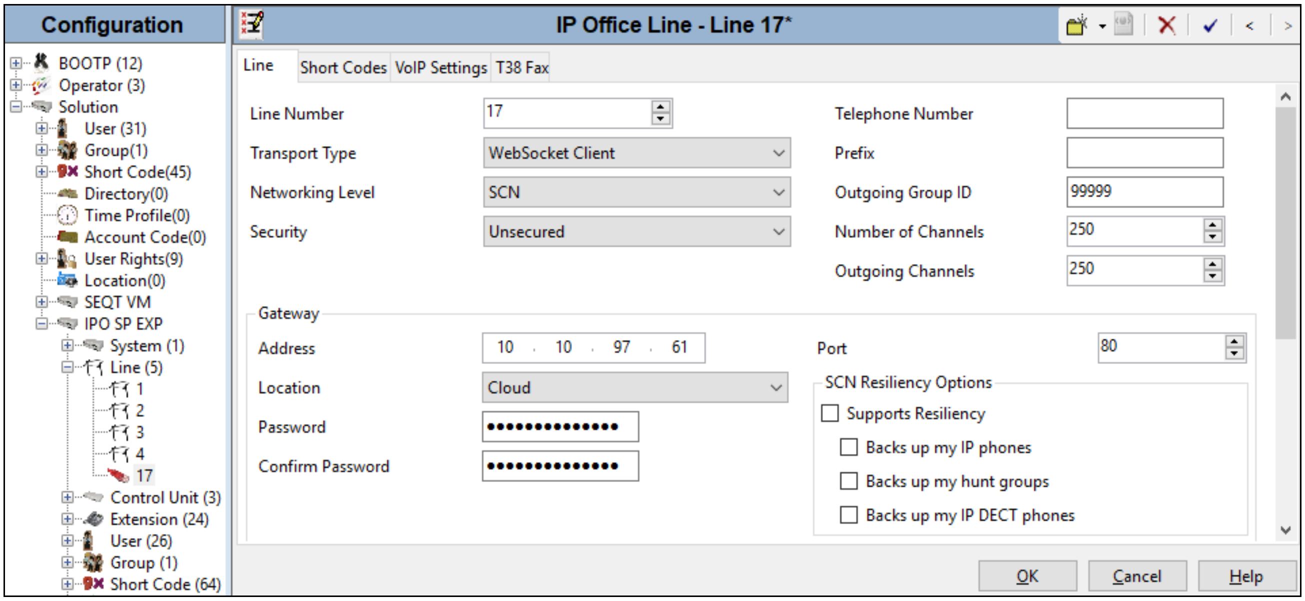


VoIP Settings tab is required to set for **Fax Transport Support** as ***T.38*** as the SIP trunk to service provider.



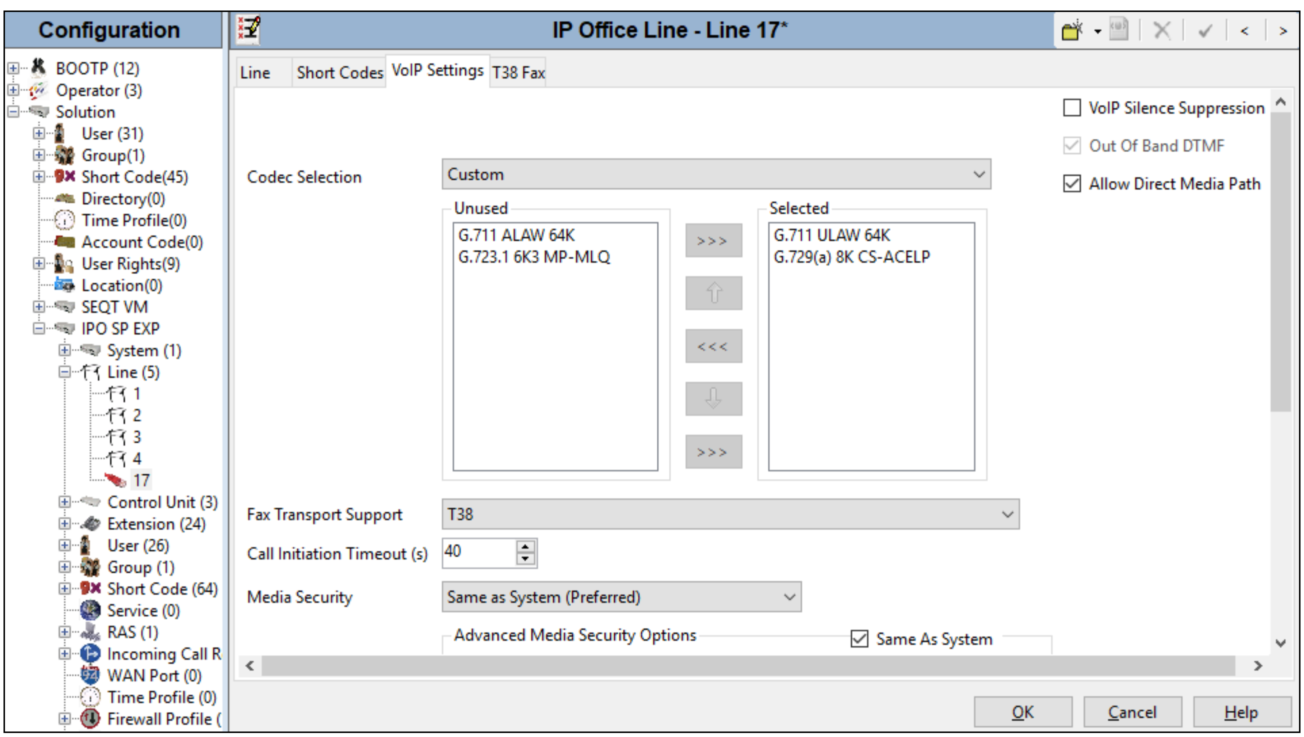
5.6. IP Office Line Secondary Server

The IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution. Below is the IP Office Line to the Primary server.

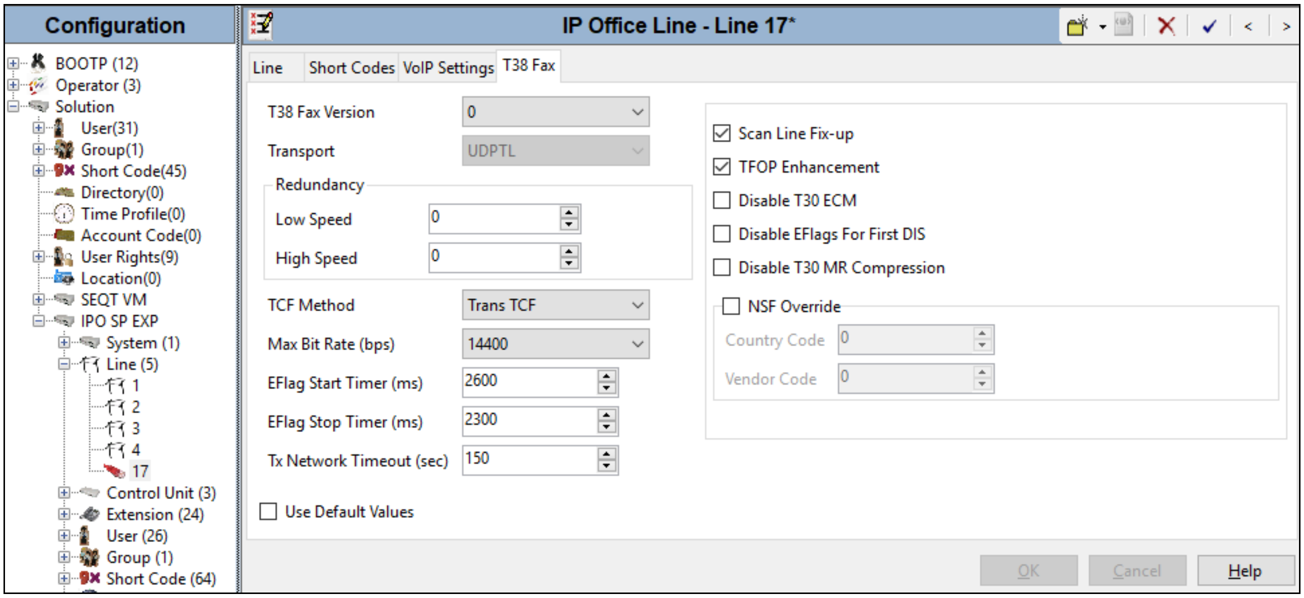


In this testing configuration, a fax machine is connected to one of the analog ports on the Expansion System. To accommodate T.38 pass-through fax, select the **VoIP Settings** tab and configure the following:

- Select ***T.38*** for **Fax Transport Support**.



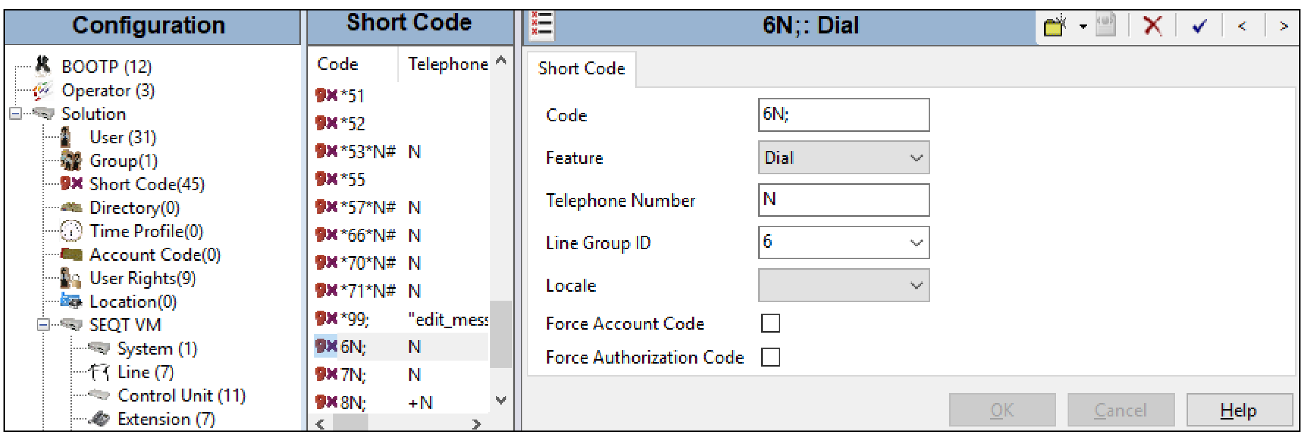
On the **T38 Fax** tab, make sure to uncheck the **Use Default Value** checkbox. Then select ***0*** from **T38 Fax Version** drop down menu as service provider support version 0.



5.7. Short Code

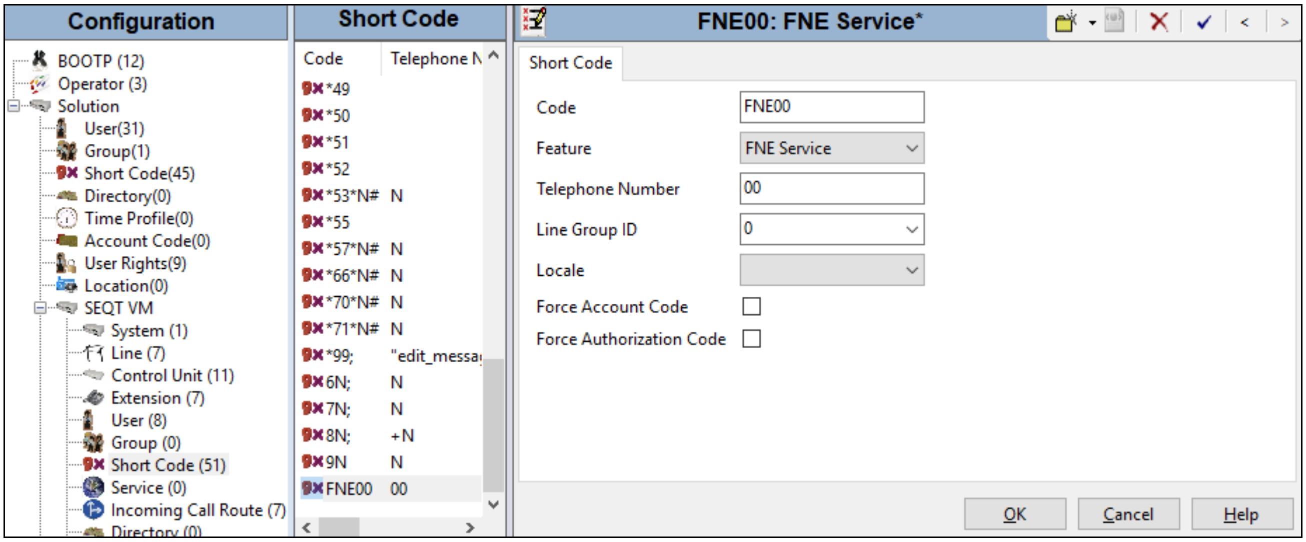
Define a short code to route outbound traffic to the SIP line. To create a short code, select **Short Code** in the left **Navigation** pane, then right-click in the Group Pane and select **New**. On the **Short Code** tab in the **Details** pane, configure the parameters for the new short code to be created. The screen below shows the details of the previously administered "6N;" 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, ***6N;*** short code will be invoked when the user dials 6 followed by any number.
- Set **Feature** to ***Dial***. This is the action that the short code will perform.
- Set **Telephone Number** to the value shown in the capture below. 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.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.4**. This short code will use this line group when placing the outbound call.
- Others parameters are at default values.
- Click **OK**.



For incoming calls from mobility extension to Feature Name/Number Extension (FNE) features hosted by IP Office to provide dial tone functionality, Short Code **FNE00** was created. The FNE00 was configured with the following parameters.

- In the **Code** field, enter the FNE feature code as ***FNE00*** for dial tone.
- Set the **Feature** field to ***FNE Service***.
- Set the **Telephone Number** field to ***00***.
- Set the **Line Group ID** field to ***0***.
- Retain default values for other fields.
- Click **OK**.

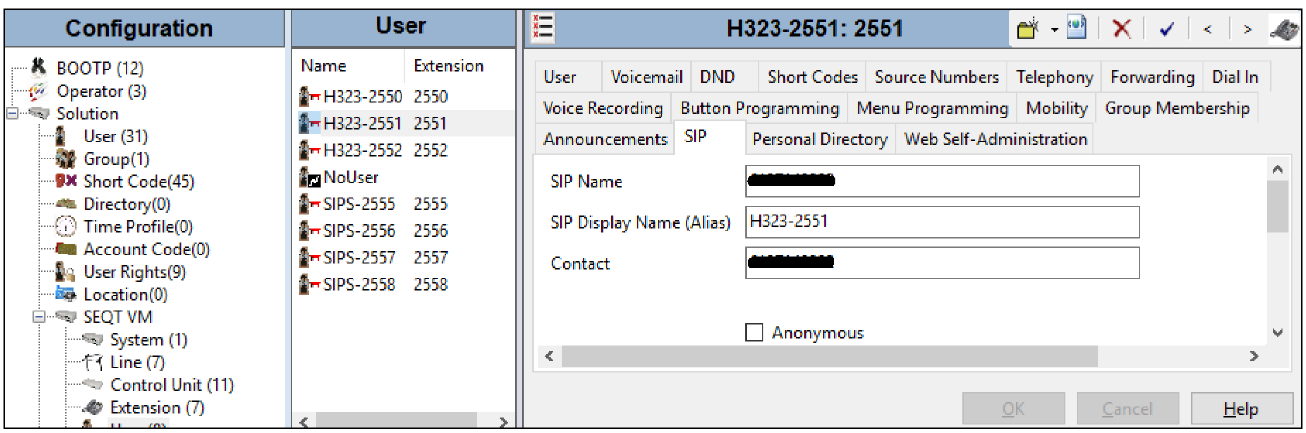


5.8. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.4**. 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 “H323-2551”. Select the **SIP** tab (**Appendix** shows how to get this SIP tab) in the **Details** pane.

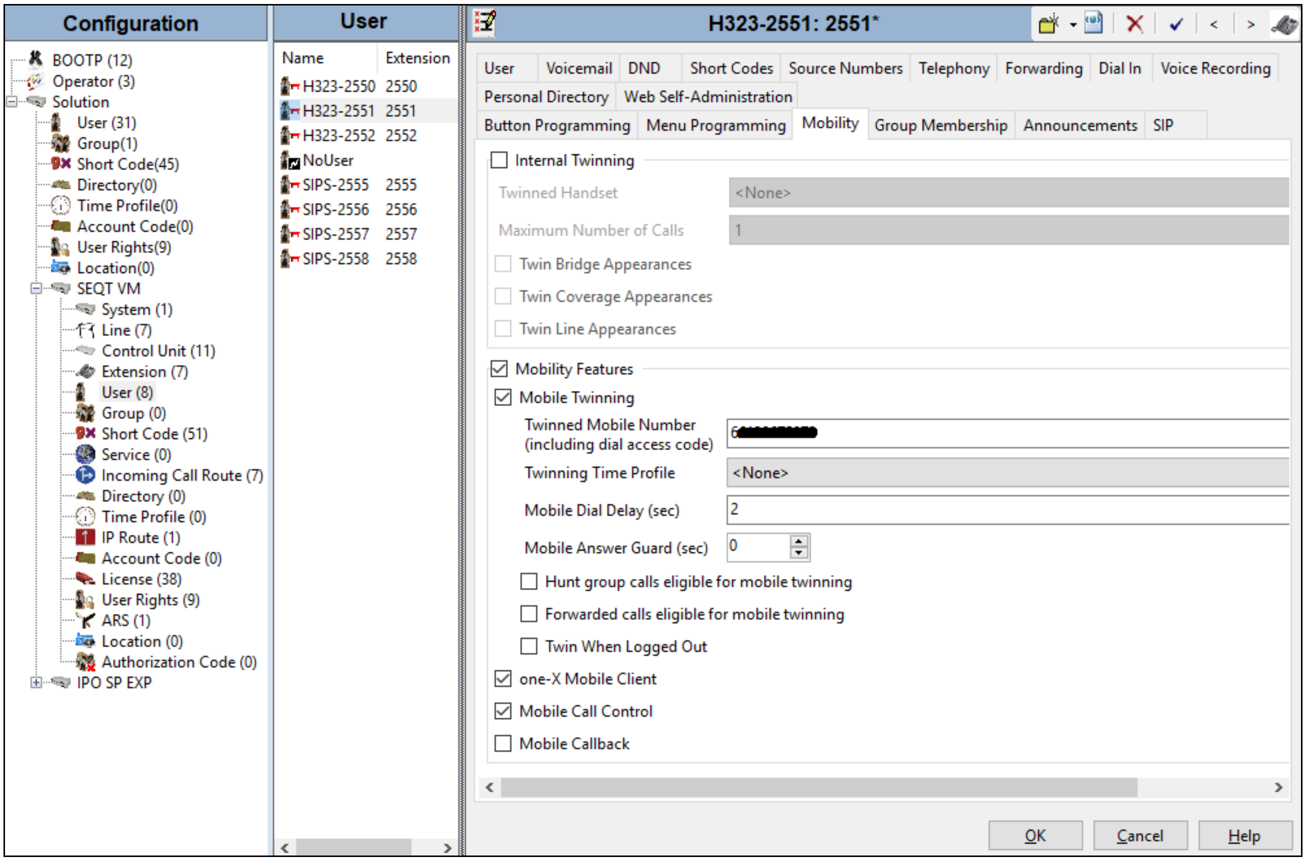
The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From header for outgoing SIP trunk calls. They also allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.4**). The example below shows the settings for user H323-2551.

- The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise from service provider.
- The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name.
- If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user’s information from the network.
- Click **OK**.



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 H323-2551.

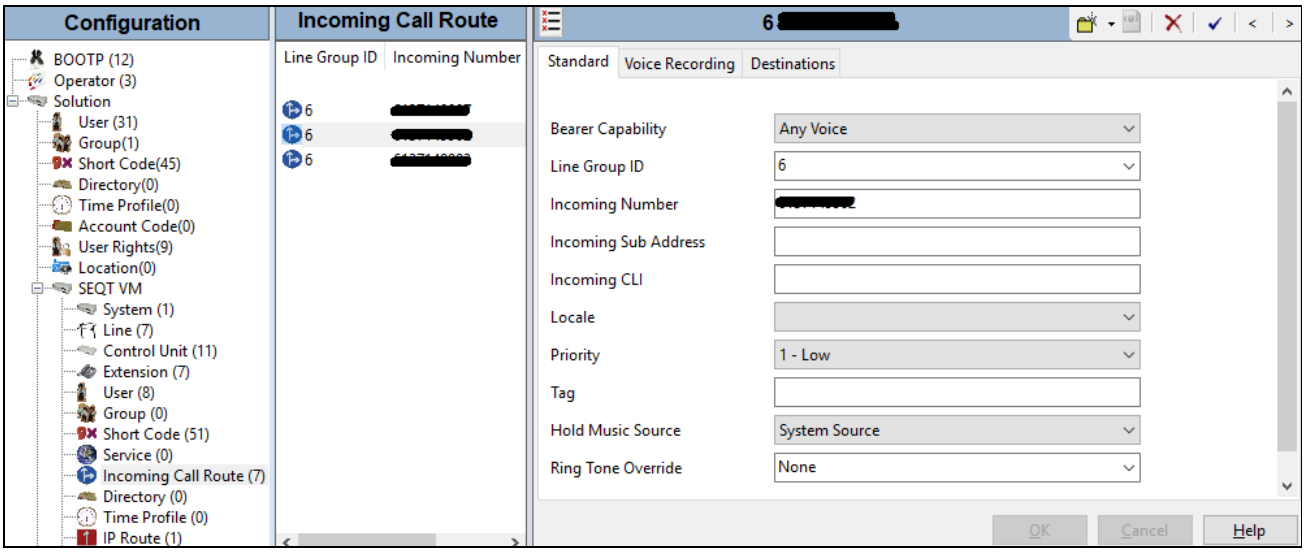
- The **Mobility Features** and **Mobile Twinning** boxes are checked.
- The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone.
- Other options can be set according to customer requirements.
- Click **OK**.



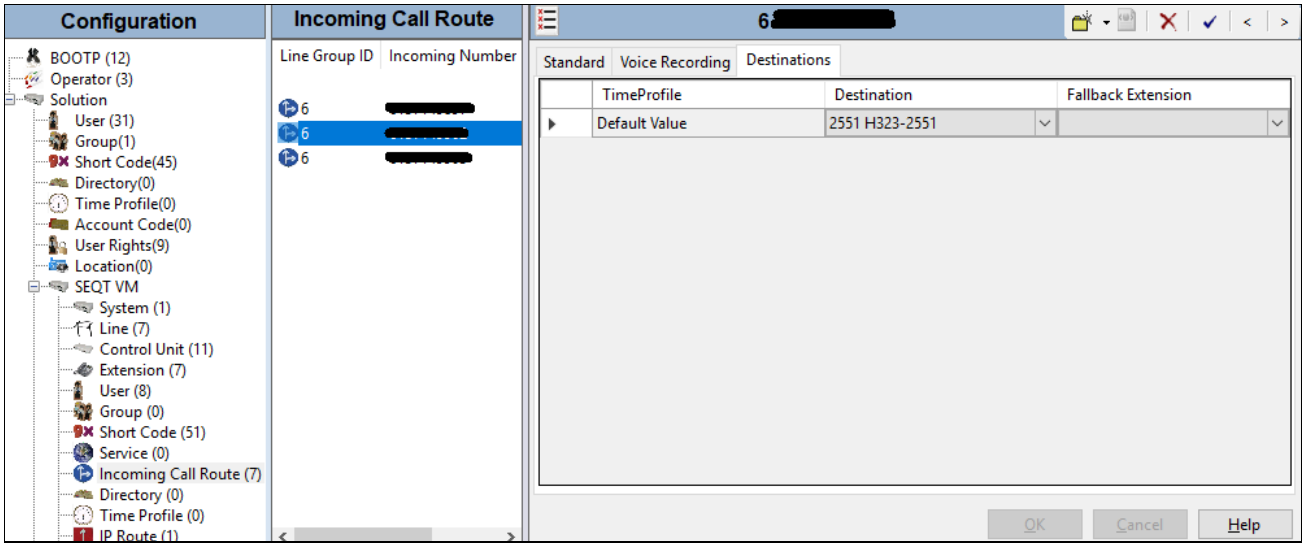
5.9. Incoming Call Route

An incoming call route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by the 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**. On the **Standard** tab of the **Details** pane, enter the parameters as shown below:

- Set the **Bearer Capacity** to ***Any Voice***.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.4**.
- Set the **Incoming Number** to the incoming number on which this route should match.
- Default values can be used for all other fields.
- Click **OK**.



On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to ***613XXXXXXX*** on line 4 are routed to extension ***2551***. Click **OK**.



5.10. Save Configuration

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

6. ThinkTel SIP Trunking Configuration

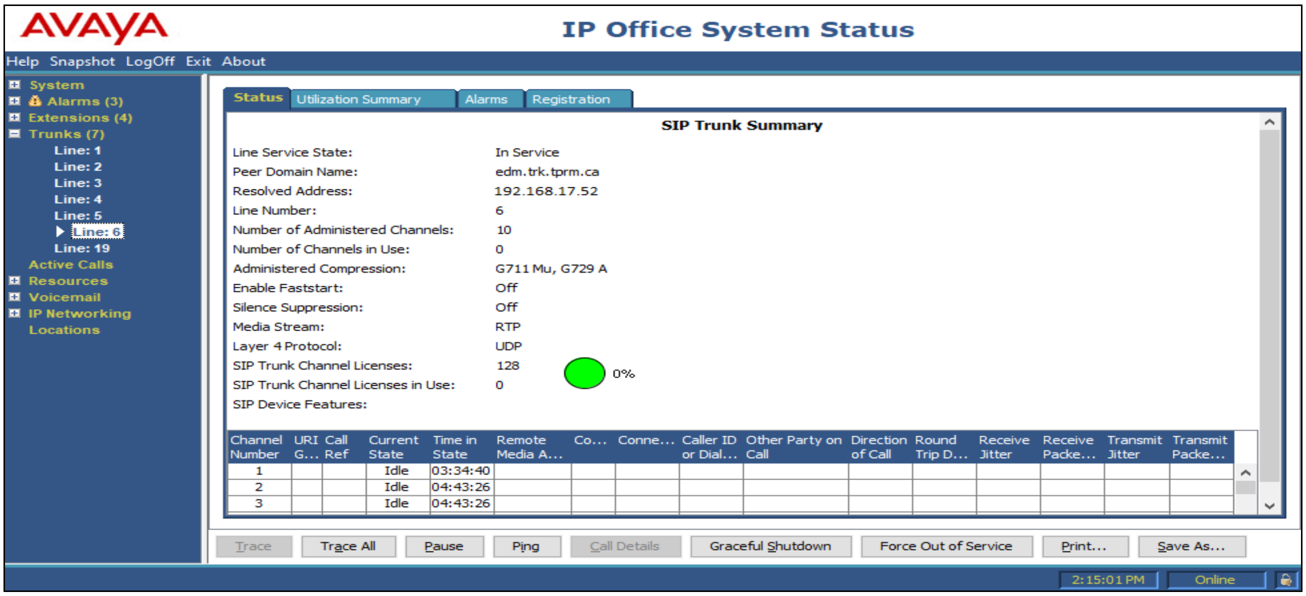
ThinkTel is responsible for the configuration of ThinkTel SIP Trunking service. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise. ThinkTel will provide the customer the necessary information to configure the Avaya IP Office SIP connection to ThinkTel.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

- IP address of the ThinkTel SIP proxy.
- Supported codecs.
- DID numbers.
- IP addresses and port numbers used for signaling or media through any security devices.

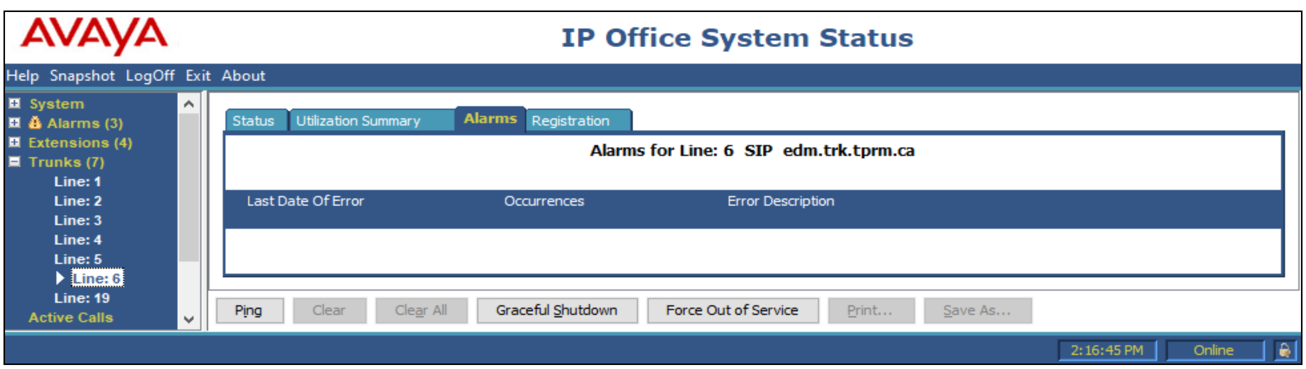
7. Verification Steps

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. 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 no active calls at present time).



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



- Verify that a phone connected to PSTN can successfully place a call to the 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.
- Using a network sniffing tool, e.g., Wireshark, to monitor the SIP signalling between the enterprise and ThinkTel. The sniffer traces are captured at the public interface of the Avaya IP Office.

8. Conclusion

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

9. Additional References

[1] *Administering Avaya IP Office Platform with Manager*, Release 10.0, August 2016.

[2] *Avaya IP Office™ Platform Server Edition Reference Configuration, Release 10.0, Issue 04.AD, August 2016.*

[3] *Deploying IP Office™ Platform Server Edition Solution, Release 10.0, August 2016.*

[4] *IP Office™ Platform, Using a Voicemail Pro IP Office Mode Mailbox, Issue 10D, May 2016.*

[5] *Using Avaya Communicator for Web*, Release 1, Issue 1.0.6, May 2016.

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/

Product documentation for ThinkTel SIP Trunking is available from ThinkTel.

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.