Avaya Solution & Interoperability Test Lab

Application Notes for Configuring ThinkTel SIP Trunking Service with Avaya IP Office R9.0.3 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Avaya IP Office Release 9.0.3 to inter-operate with the ThinkTel SIP Trunking Service.

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

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.

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 Avaya IP Office Release 9.0.3 to inter-operate with the ThinkTel SIP Trunking Service.

In the sample configuration, the Avaya IP Office solution consists of Avaya IP Office 500 V2 running Release 9.0 SP3 software, Avaya Preferred Edition (a.k.a Voicemail Pro) messaging application, Avaya H.323 and SIP hardphones, and SIP-based Avaya softphones.

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

For brevity, the remainder of this document mostly refers to the ThinkTel SIP Trunking Service simply as ThinkTel.

The terms "Service Provider" and "ThinkTel" will be used interchangeable throughout these Application Notes.

2. General Test Approach and Test Results

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.

The general test approach was to configure a simulated enterprise site using Avaya IP Office to connect to ThinkTel via the public Internet. The configuration shown in **Figure 1** was used to exercise the features and functionality tests listed in **Section 2.1**

2.1. Interoperability Compliance Testing

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

- SIP OPTIONS queries and responses.
- Incoming calls from the PSTN to H.323 and SIP telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing calls to the PSTN from H.323 and SIP telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Various call types including: local, long distance, outbound toll-free, international, and Local Directory Assistance (411) calls.
- G.711MU and G.729A codecs.
- Caller ID presentation and Caller ID restriction.
- DTMF transmission using RFC 2833.

- Voicemail access and navigation for inbound and outbound calls.
- Telephony supplementary features such as hold and resume, transfer, and conference.
- Off-net call forwarding and call transfer/conference.
- Twinning on inbound calls to PSTN mobile phones.
- Use of the SIP REFER method for transferring calls off-net to the PSTN.
- Inbound and outbound long-duration call stability.
- Inbound and outbound long hold time call stability.
- Response to incomplete call attempts and trunk busy or error conditions.
- T.38 and G.711 pass-through fax.

2.2. Test Results

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

- **OPTIONS Response –** On the test circuit used for the compliance test, ThinkTel responded to OPTIONS from IP Office with "401 Unauthorized" instead of "200 OK". For ThinkTel to respond to OPTIONS with "200 OK", it requires that the OPTIONS message contain an Authorization header, but Avaya IP Office cannot be configured to satisfy this requirement. However, IP Office treats any response to OPTIONS as an indication that the far end is active and responding, therefore will not take the SIP connection to the far end out of service. This OPTIONS response is listed here simply as an observation since it had no impact on the status of the SIP connection between IP Office and ThinkTel.
- **Digest Authentication –** Only the first outbound call after IP Office was rebooted would show the usual challenge/response exchange of Digest Authentication (1st outbound INVITE eliciting 401 message back as challenge, thereupon a 2nd outbound INVITE containing the required authentication data as response). After that, the outbound calls would contain an Authorization header specifying all the required authentication information; therefore the Digest Authentication handshake was obviated.
- **Codec Lockdown –** When the SDP of an outbound call INVITE contained the codec list of G.729A and G.711MU in that preference order, ThinkTel's call connect "200 OK" contained the same codecs in same order in the SDP instead of the single preferred codec (G.729A). However, When the SDP of outbound INVITE contained the codec list in the order of G.711MU and G.729A, ThinkTel's call connect "200 OK" contained the single preferred codec (G.711MU). This difference in codec lockdown behavior is listed here simply as an observation since there was no user impact.
- **Unsupported Codec –** When outbound call was configured to use a codec unsupported by ThinkTel, ThinkTel would return a "183 Session In Progress" message whose SDP contained G.711MU. IP Office would then terminate the call by issuing the CANCEL message. While this behavior was acceptable, it would be more desirable for ThinkTel to return an explicit status message, like "488 Not Acceptable Here" or "415 Media Type Missing" in response to the outbound INVITE. ThinkTel opened an enhancement ticket to its platform vendor for this issue.

- **REFER Signaling –** When the SIP REFER method was used for off-net call re-direction, ThinkTel would issue an INVITE towards the IP Office caller after issuing BYE's, upon accepting the REFER from the enterprise, to terminate the calls between the PSTN endpoints and the IP Office caller (so that IP Office would no longer anchor the calls after the call re-direction). This INVITE would elicit an IP Office response of "481 Dialog/Transaction Does Not Exist" since the call had already been terminated by ThinkTel. This untidy signaling exchange had no user impact; off-net call re-direction using REFER was successfully tested/verified.
- **Direct Media –** Avaya IP Office R9.0 offers a new Direct Media capability on IP Office 500 V2 that allows IP endpoints to send RTP media directly to each other rather than having all the media flow through the IP Office, using up VoIP and relay resources. This capability is not supported on the configuration described in these application notes where Avaya IP Office connects to the service provider network via a direct public Internet connection without using an enterprise Session Border Controller.

Items not supported or not tested include the following:

- Inbound toll-free and emergency calls (911) were not tested as part of the compliance test.
- ThinkTel does not support Operator (0) and Operator-Assisted (0 + 10-digits) calls.
- ThinkTel does not support SIP session timer refresh. ThinkTel did not initiate session refresh via SIP re-INVITE or UPDATE messages, and no message headers for session refresh handshake were contained in SIP messages from ThinkTel. In the compliance test, session refresh for active calls was initiated from IP Office.

2.3. Support

For technical support on ThinkTel SIP Trunking Service, contact ThinkTel technical support at:

- Phone: 1 (866) 928-4465
- Email: support@thinktel.ca
- Website: http://support.thinktel.ca/

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. Alternatively, in the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus.

3. Reference Configuration

Figure 1 illustrates the test configuration showing an enterprise site connected to the ThinkTel SIP Trunking Service.

Located at the enterprise site is an Avaya IP Office 500 V2 running the Release 9.0 SP3 software. Endpoints include various Avaya IP Telephones (with H.323 and SIP firmware) and SIP-based Avaya softphones (Avaya IP Office Video Softphone and Avaya Flare® Experience for Windows). The site also has a Windows PC running Avaya Preferred Edition (a.k.a. Voicemail Pro) for providing voice messaging service to the Avaya IP Office users, and Avaya IP Office Manager for administering 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.

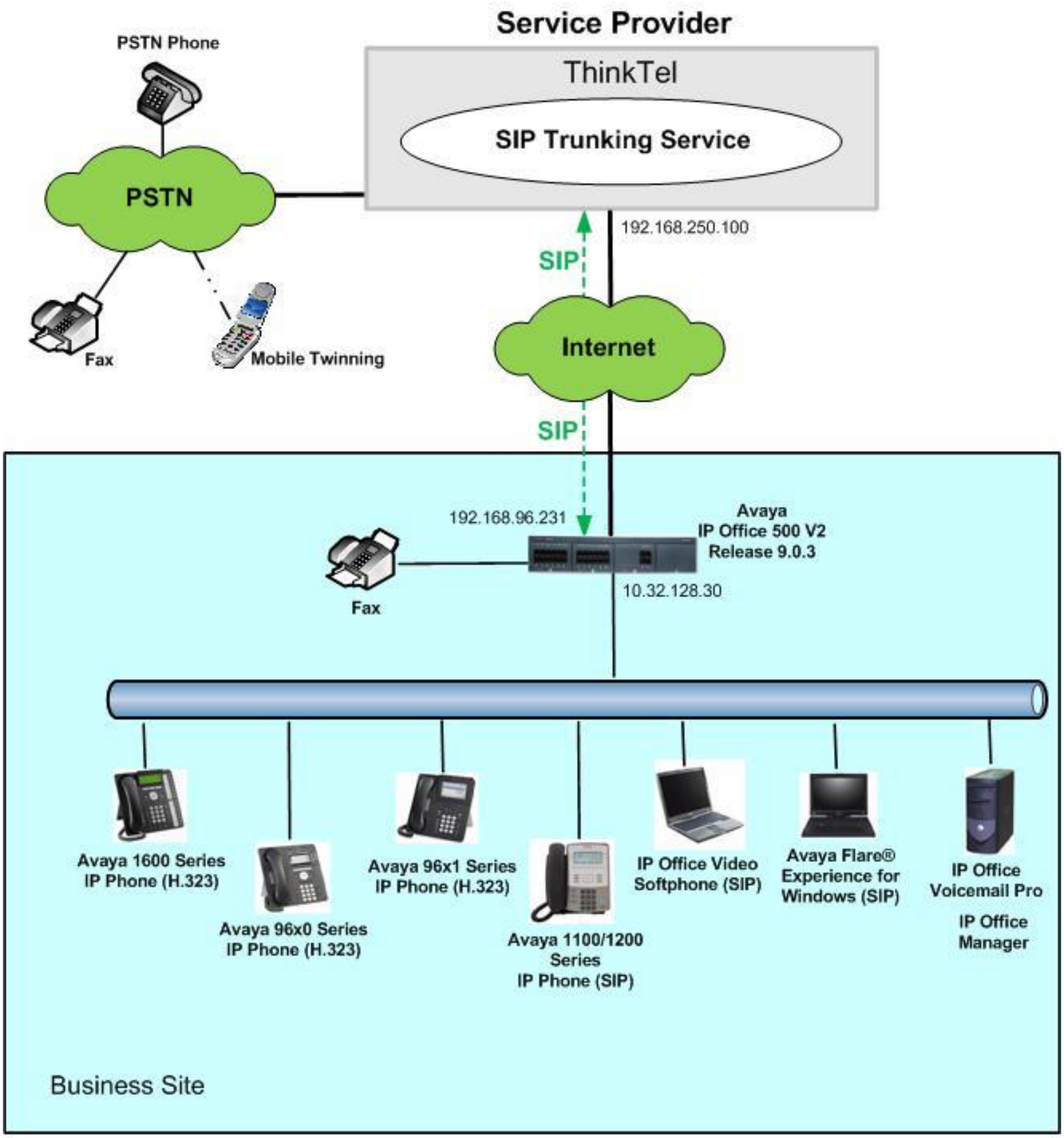


Figure 1: Avaya IP Office with ThinkTel SIP Trunking Service

For security purposes, any actual public IP addresses used in the compliance test were changed to 192.168.x.x throughout these Application Notes.

During compliance testing, Avaya IP Office users dialed a short code of 8 or 9, plus N digits to send digits across the SIP trunk to ThinkTel. The short code (8 or 9) was stripped off by Avaya IP Office but the remaining N digits were sent to the service provider network. For calls within the North American Numbering Plan (NANP), the user dialed 11 (1 + 10) digits for long distance and local calls. Thus, for these NANP calls, Avaya IP Office sent 11 digits in the Request URI and the To header of an outbound SIP INVITE message. ThinkTel sent 10 digits in the Request URI and the To header of inbound SIP INVITE messages.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and 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 Avaya IP Office must be allowed to pass through these devices.

The administration of the Avaya Preferred Edition (Voicemail Pro) messaging service and endpoints on Avaya IP Office is standard. Since these configuration tasks are not directly related to the inter-operation with the ThinkTel SIP Trunking Service, they are not included in these Application Notes.

4. Equipment and Software Validated

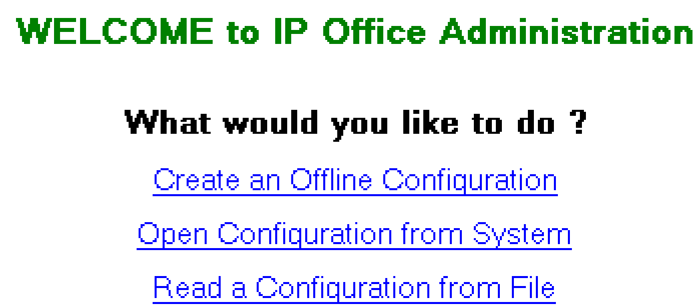
The following equipment and software/firmware were used for the sample configuration provided:

Avaya Telephony Components	
Equipment / Software	**Release / Version**
Avaya IP Office 500 V2	9.0.3.0 build 941
Avaya IP Office COMBO6210/ATM4 Module	9.0.3.0 build 941
Avaya IP Office Manager	9.0.3.0 build 941
Avaya Preferred Edition (a.k.a Voicemail Pro)	9.0.3.0 build 212
Avaya 1616 IP Telephones (H.323)	Avaya one-X Deskphone 1.3 SP5
Avaya 9611G IP Telephones (H.323)	Avaya one-X Deskphone 6.3.1.16_V452
Avaya 9630G IP Telephones (H.323)	Avaya one-X Deskphone 3.212A
Avaya 1120E IP Telephone (SIP)	4.04.14.00
Avaya IP Office Video Softphone (Windows)	3.2.3.49 68975
Avaya Flare® Experience for Windows	1.1.4.23
ThinkTel Components	
Equipment / Software	**Release / Version**
Metaswitch	7.4
Opensips Session Border Controller	1.6.2

Testing was performed with IP Office 500 V2, but this testing also applies to Avaya IP Office Server Edition running the same version of software. Note that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks.

5. Configure Avaya IP Office

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



Select **Open Configuration from System**. If the above screen does not appear, the configuration may be alternatively opened by navigating to **File** → **Open Configuration** at the top of the Avaya IP Office Manager window. Select the proper Avaya IP Office system from the pop-up window and log in with the appropriate credentials.

The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this document, the **View** menu was configured to show the Navigation Pane on the left side, omit the Group Pane in the center, and show the Details Pane on the right side. Since the Group Pane has been omitted, its content is shown as submenus in the Navigation Pane. These panes (Navigation and Details) will be referenced throughout the Avaya IP Office configuration.

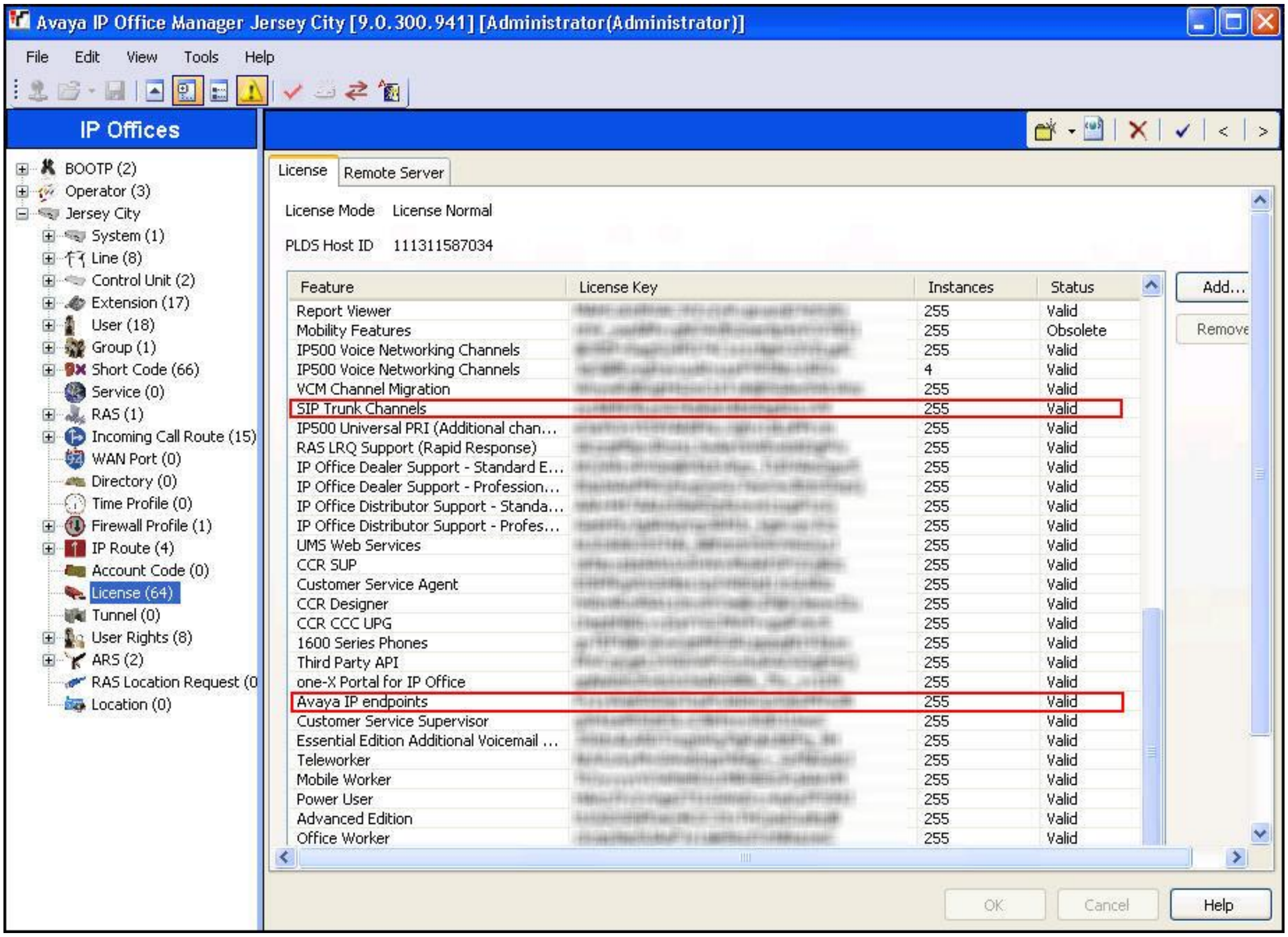
All licensing and feature configuration that is not directly related to the interface with the service provider (such as twinning and IP Office Softphone support) is assumed to already be in place.

In the sample configuration, **Jersey City** was used as the system name. All navigation described in the following sections (e.g., **License** → **SIP Trunk Channels)** appears as submenus underneath the system name **Jersey City** in the Navigation Pane. The configuration screens highlight values/settings configured for the compliance test. Defaults were used for other values and may be customized based upon requirements in the field.

5.1. Licensing and Physical Hardware

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

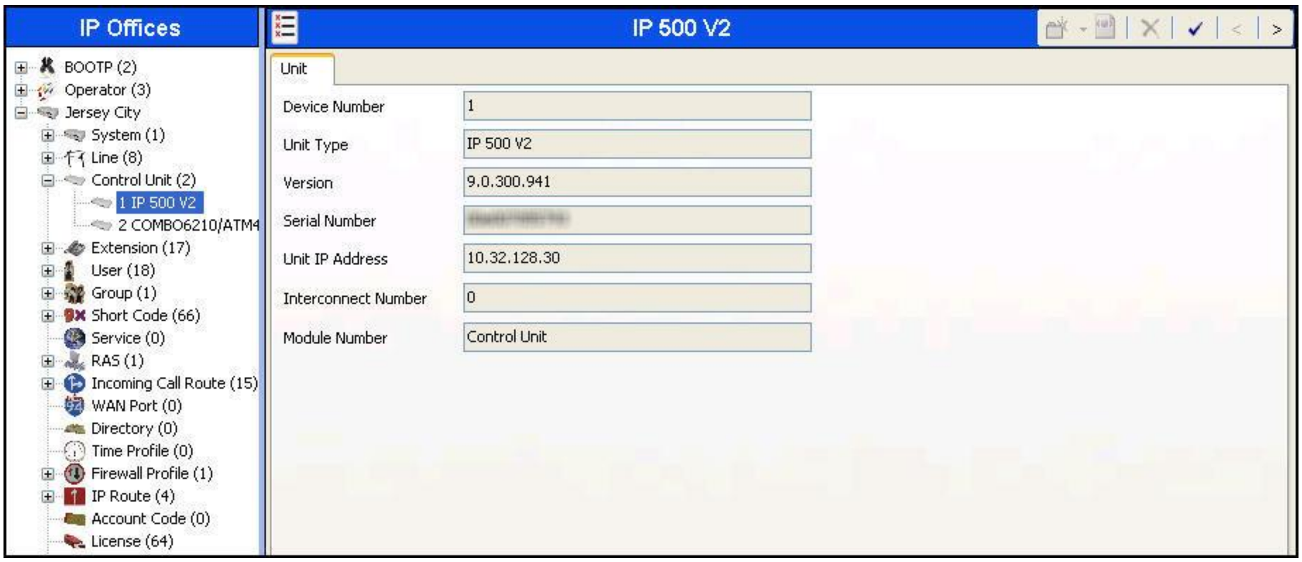
To verify that there is a **SIP Trunk Channels** License with sufficient capacity; click **License** in the Navigation Pane. Confirm a valid license with sufficient **Instances** (trunk channels) in the Details Pane. The screen below also shows the valid license for **Avaya IP endpoints**.



To view the physical hardware comprising the Avaya IP Office system, expand the components under the **Control Unit** in the Navigation Pane. In the sample configuration, the second component listed is a Combination Card. This module has 6 digital station ports, two analog extension ports, 4 analog trunk ports and 10 VCM channels. The VCM is a Voice Compression Module supporting VoIP codecs. An Avaya IP Office hardware configuration with a VCM component is necessary to support SIP trunking.

To view the details of the component, select the component in the Navigation Pane.

The screen below shows the details of the IP 500 V2:



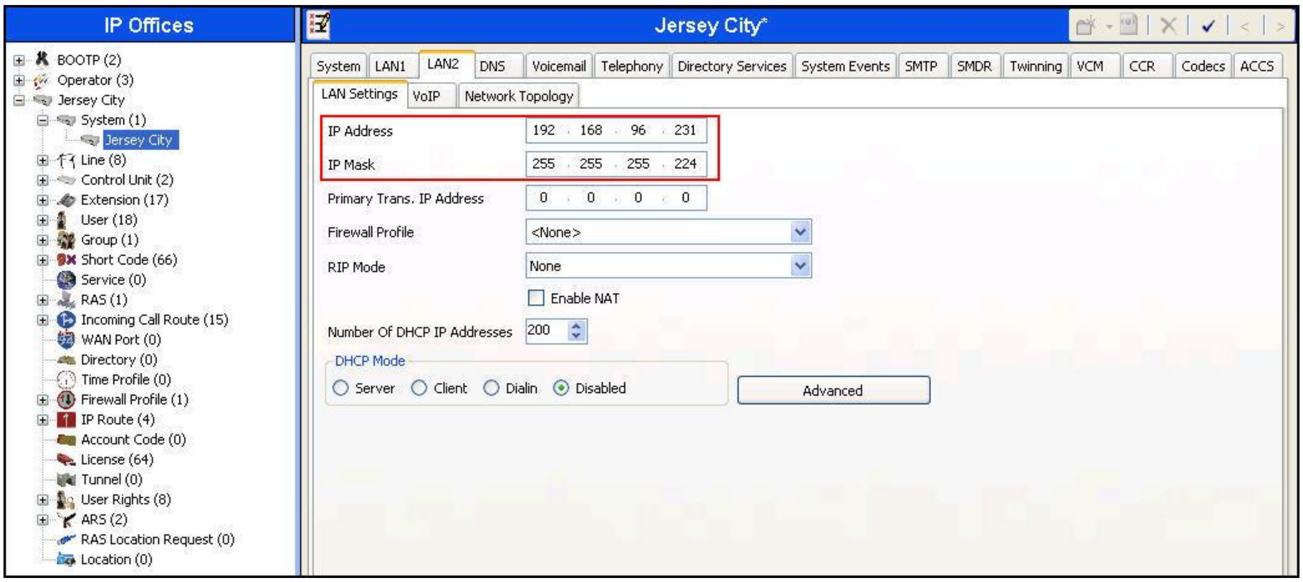
The screen below shows the details of the Combination Card:

5.2. System

This section configures the necessary system settings

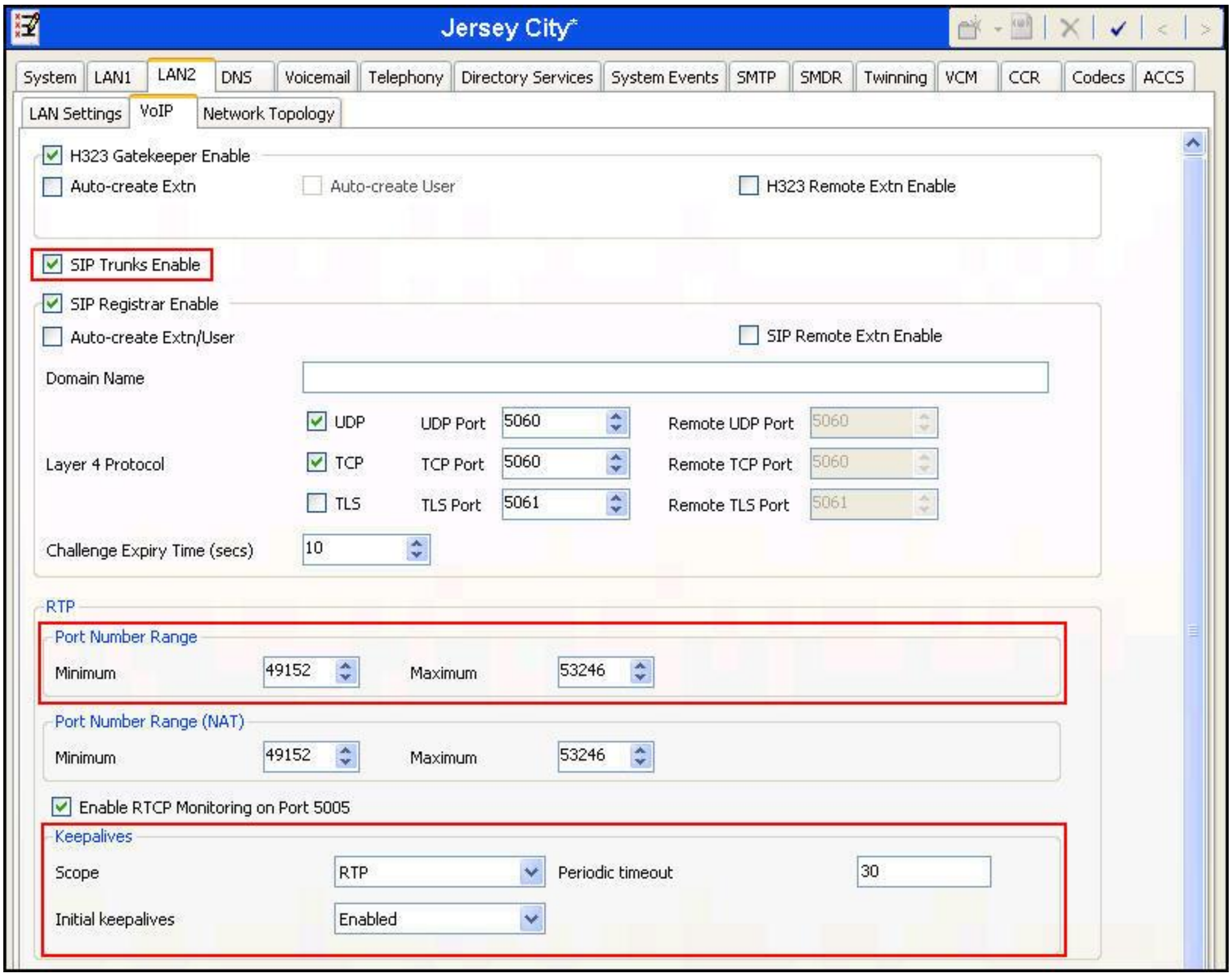
5.2.1. System - LAN2 Tab

In the sample configuration, ***Jersey City*** was used as the system name and the WAN port (LAN2 port) was used to connect the Avaya IP Office to the public network. The LAN2 settings correspond to the WAN interface on Avaya IP Office. To access the LAN2 settings, first navigate to **System → Jersey City** in the Navigation Pane 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.

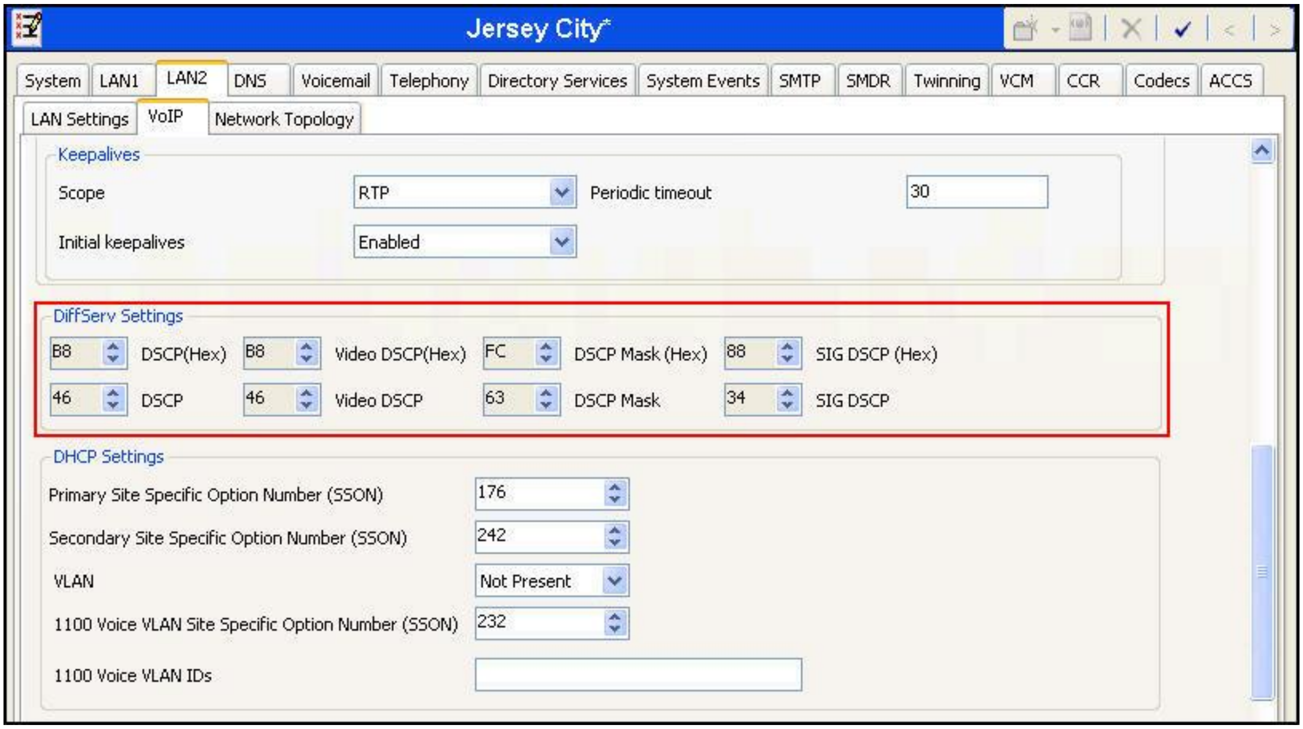


On the **VoIP** tab of LAN2 in the Details Pane, configure the following parameters:

- Check the **SIP Trunks Enable** box to enable the configuration of SIP trunks.
- In the **RTP** section, the **RTP Port Number Range** can be customized to a specific range of receiving ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a port in the configurable range for calls using LAN2.
- In the **Keepalives** section, select ***RTP*** for **Scope**; select ***Enabled*** for **Initial keepalives**; enter ***30*** for **Periodic timeout**. These settings direct IP Office to send a RTP keepalive packet starting at the time of initial connection and every 30 seconds thereafter if no other RTP traffic is present. This facilitates the flow of media in cases where each end of the connection is waiting for media from the other, as well as helping to keep firewall ports open for the duration of the call.



Scroll down to the **DiffServ Settings** section. Avaya IP Office can 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 screen below and are also the default values. For a customer installation, if the default values are not sufficient, appropriate values should be provided by the customer.



On the **Network Topology** tab of LAN2 in the Details Pane, configure the following parameters:

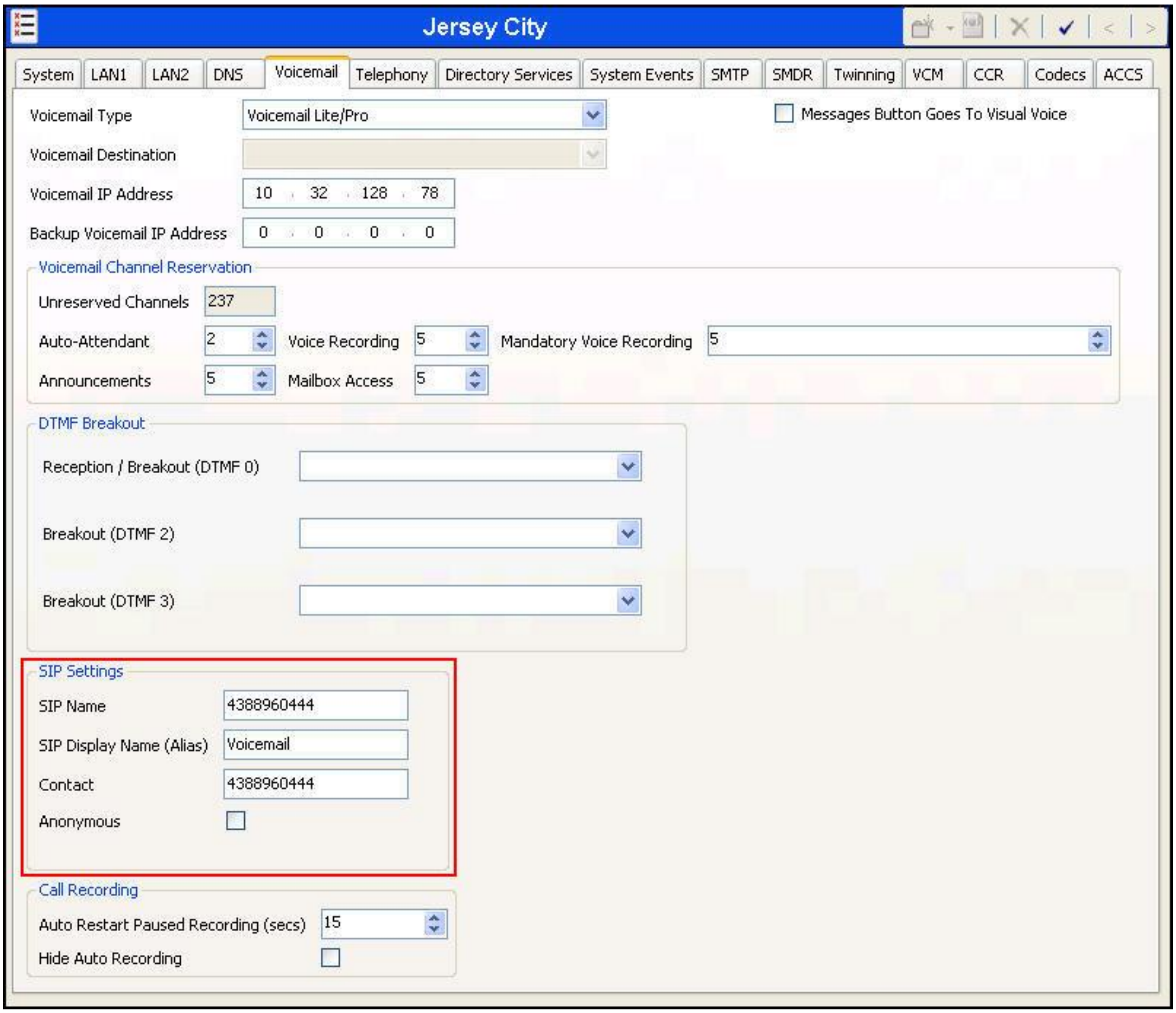
- Select **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to ***Open Internet***. With the ***Open Internet*** setting, **STUN Server Address** is not used.
- Set **Binding Refresh Time (seconds)** to a desired value. 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. See **Section 5.9** for complete details.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port.
- Set **Public Port** to ***5060*** for **UDP**.

During compliance testing, 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, and therefore is not described in these Application Notes.

5.2.2. System - Voicemail Tab

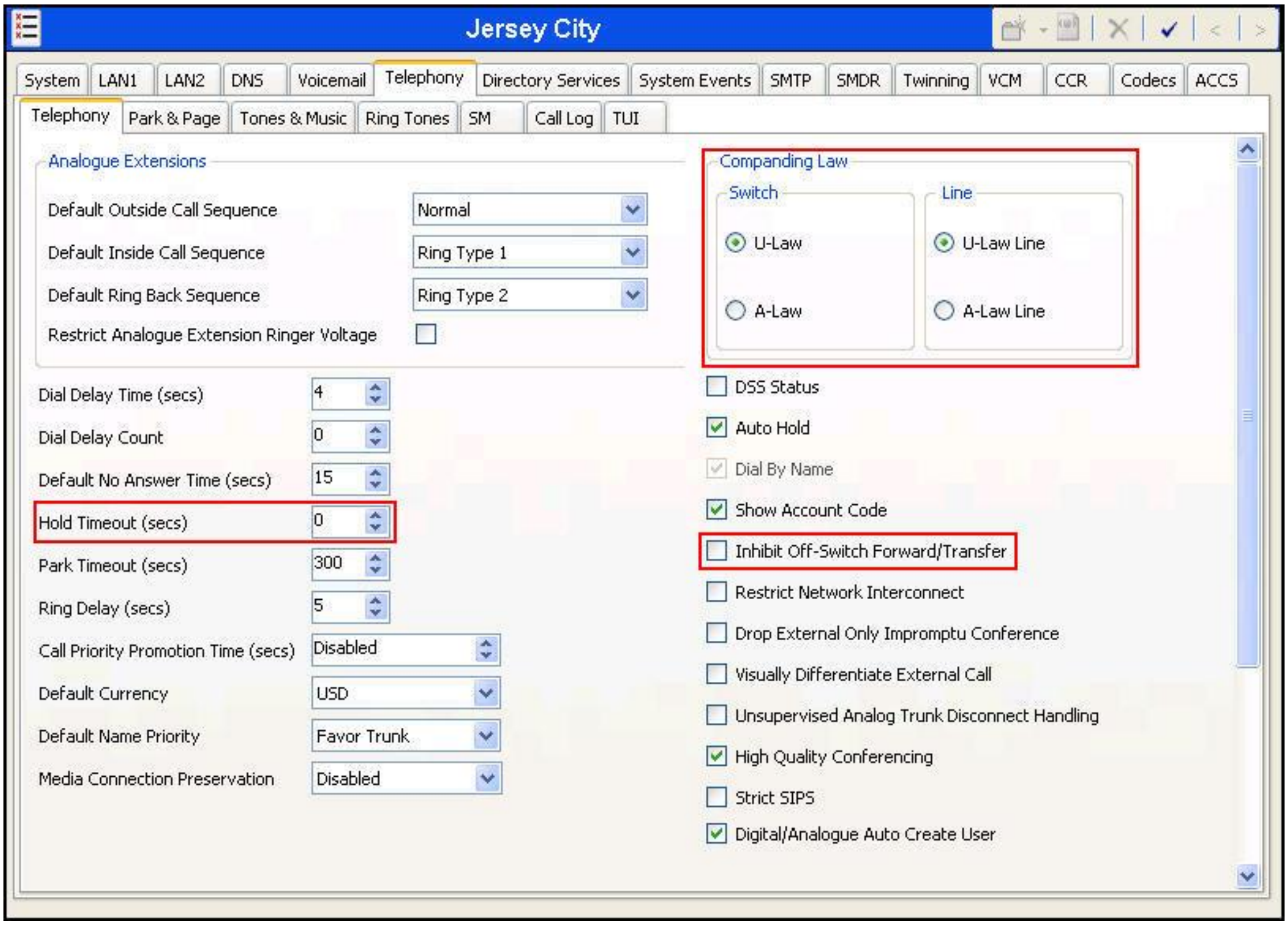
In the **Voicemail** tab of the Details Pane, configure the **SIP Settings** section. The **SIP Name** and **Contact** are set to one of the DID numbers provided by ThinkTel. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. Uncheck the **Anonymous** box to allow the Voicemail Caller ID information to be sent to the network.

Note the selection for **Voicemail Type** and the IP address setting for **Voicemail IP Address**. These are for configuring Voicemail Pro as the voice messaging service for Avaya IP Office users (part of the standard IP Office setup beyond the scope of these Application Notes).



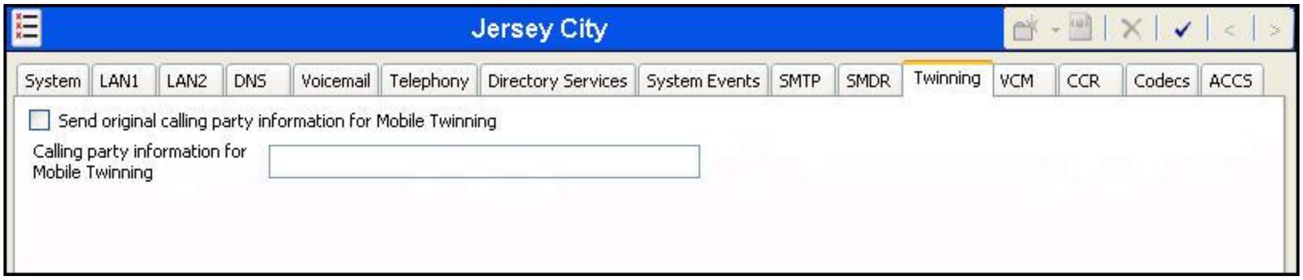
5.2.3. System - Telephony Tab

Navigate to the **Telephony → Telephony** tab in the Details Pane. Enter or select ***0*** for **Hold Timeout (secs)** so that calls on hold will not time out. Choose the **Companding Law** typical for the enterprise site. For the compliance test, ***U-LAW*** was 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 per customer business policies. Note that this configuration might pose a security issue (Toll Fraud). Customers should exercise caution with this configuration.



5.2.4. System - Twinning Tab

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



5.2.5. System – Codecs Tab

In the **Codecs** tab of the Details Pane, select or enter ***101*** for **RFC2833 Default Payload**. This setting was recommended by ThinkTel for use with out-band DTMF tone transmissions.

5.3. IP Route

Navigate to **IP Route → 0.0.0.0** in the left Navigation Pane if a default route already exists. Otherwise, to create the default route, right-click on **IP Route** and select **New** (not shown)**.** Create and verify a default route with the following parameters:

- Set **IP Address** and **IP Mask** to ***0.0.0.0***.
- Set **Gateway IP Address** to the IP address of the gateway for the public internet network.
- Set **Destination** to ***LAN2*** from the drop-down list.

5.4. Administer SIP Line

A SIP Line is needed to establish the SIP connection between Avaya IP Office and ThinkTel. 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 SIP Line **Transport** tab.

Therefore, it is important that the SIP Line configuration be reviewed and updated after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.4.2** through **5.4.6**.

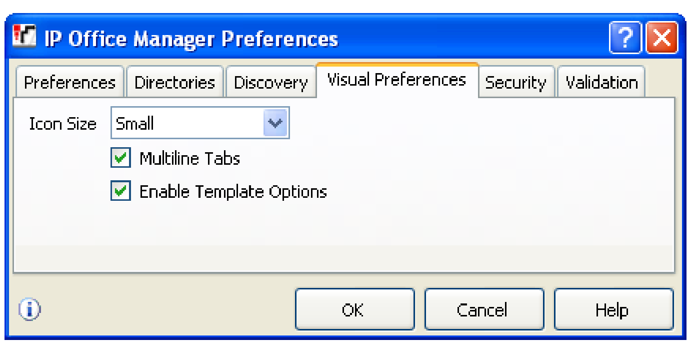
Also, the following SIP Line settings are not supported on Avaya IP Office Basic Edition:

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

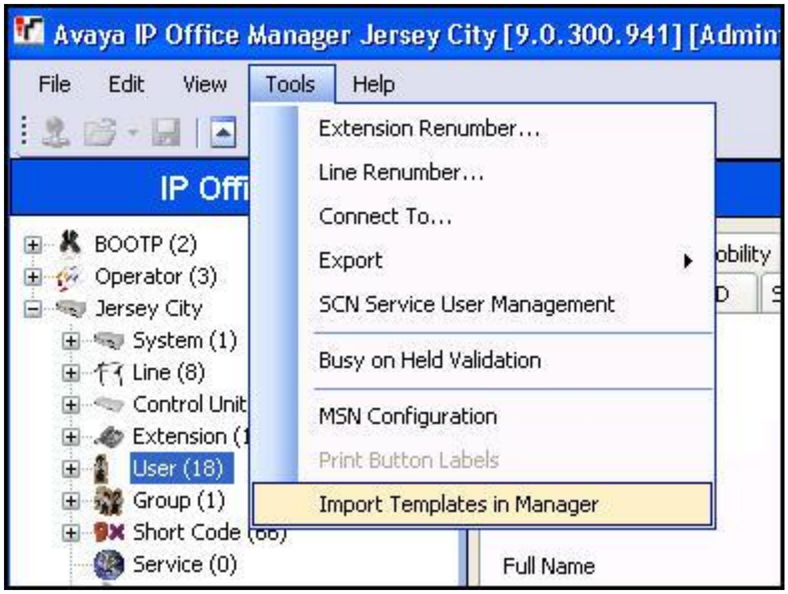
5.4.1. Create SIP Line From Template

1. Copy the template file to the computer where IP Office Manager is installed. Rename the template file to **CA_ThinkTel_SIPTrunk.xml**. The file name is important in locating the proper template file in **Step 5**.

2. Verify that template options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the IP Office Manager Preferences window that appears, select the **Visual Preferences** tab. Verify that the option box is checked next to **Enable Template Options**. Click **OK**.



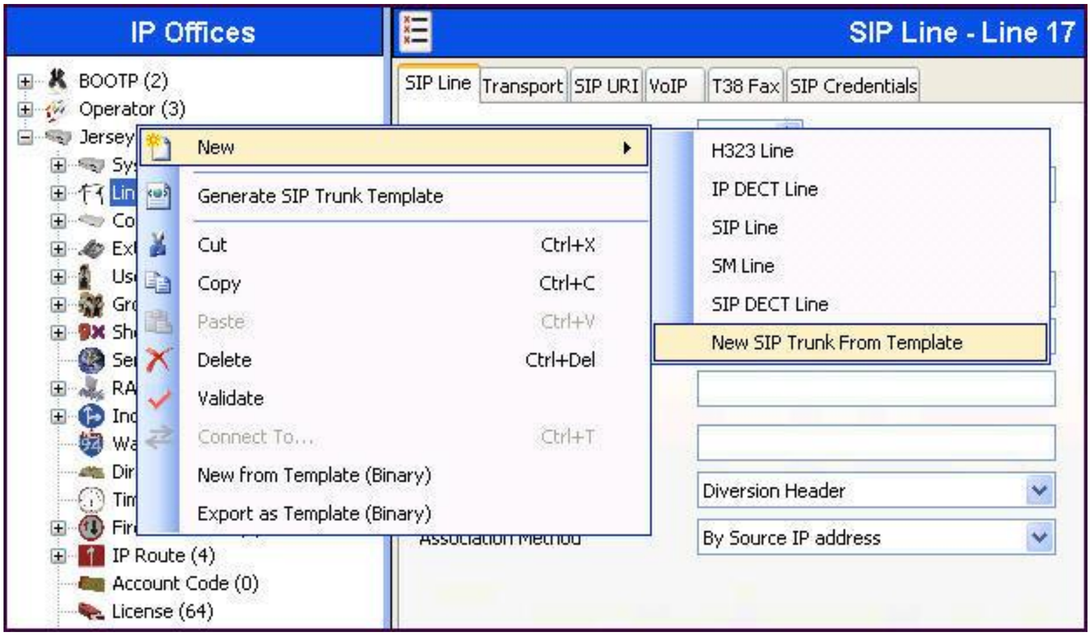
3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default template location is **C:\Program Files\Avaya\IP Office\Manager\Templates**.



In the pop-up window (not shown) that appears, select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window (not shown) will appear stating success or failure. 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.

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



5. In the subsequent **Template Type Selection** pop-up window, select ***United States*** from the **Country** drop-down list and select ***ThinkTel*** from the **Service Provider** drop-down list as shown below. These values correspond to parts of the file name (**CA_ThinkTel_SIPTrunk.xml)** created in **Step 1**. Click **Create new SIP Trunk** to finish creating the trunk.

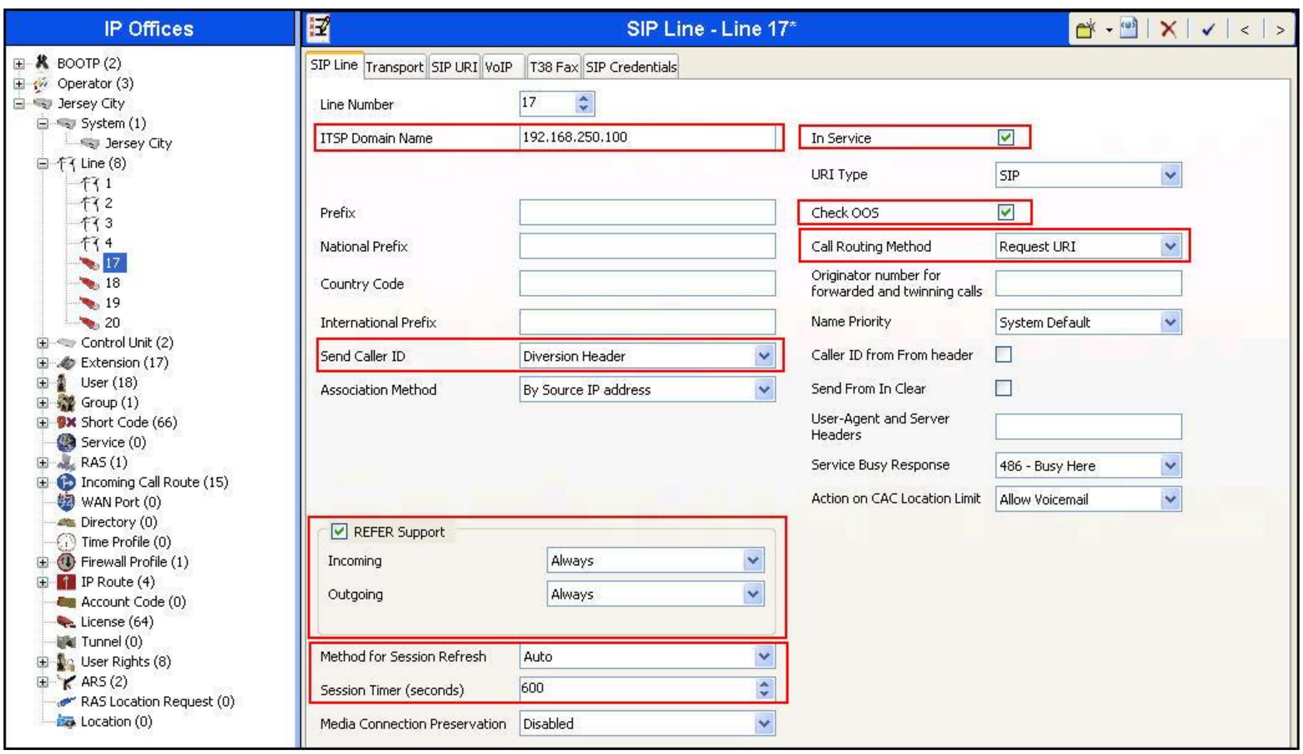
Note that the newly created SIP Line may not immediately appear in the Navigation pane until the configuration was saved, closed and reopened in IP Office Manager.

6. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.4.2** through **5.4.6**.

5.4.2. SIP Line – SIP Line Tab

In the **SIP Line** tab of the Details Pane, configure the parameters as shown below:

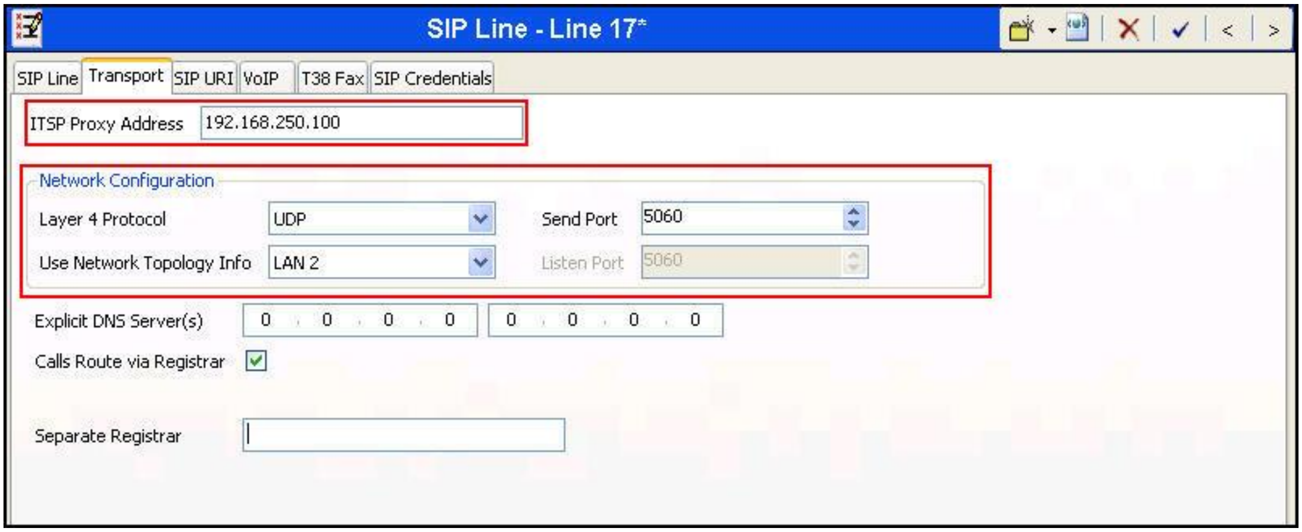
- Set **ITSP Domain Name** to the IP address of the service provider SIP Trunking access interface (provided by ThinkTel) which the SIP Line is connecting to.
- Check the **In Service** box.
- Check **OOS** box. Avaya IP Office will check the SIP OPTIONS response from the far end to determine whether to take the SIP Line out of service.
- Set **Call Routing Method** to ***Request URI***. Avaya IP Office will route inbound calls based on the number in the Request URI.
- Set **Send Caller ID** to ***Diversion Header***. With this setting and the related configuration in **Section 5.2.4**, Avaya IP Office will include the Diversion Header for calls that are redirected via Mobile Twinning out the SIP Line to the PSTN. It will also include the Diversion Header for calls that are forwarded out the SIP Line.
- Check **REFER Support**, then select ***Always*** for **Incoming** and **Outgoing**. ThinkTel supports using REFER message for off-net call re-direction as in call transfer.
- Set **Method for Session Refresh** to ***Auto***. With this setting Avaya IP Office will send UPDATE messages for session refresh if the other party supports UPDATE. If UPDATE is not supported, re-INVITE messages are sent.
- Set **Session Timer (seconds)** to a desired value. Avaya IP Office will send out session refresh UPDATE or re-INVITE at the specified intervals (half of the specified value)



5.4.3. SIP Line – Transport Tab

Navigate to the **Transport** tab and set the following:

- Set the **ITSP Proxy Address** to the IP address of the service provider SIP Trunking access interface (provided by ThinkTel) which the SIP Line is connecting to.
- Set the **Layer 4 Protocol** to ***UDP***.
- Set **Use Network Topology Info** to ***LAN2*** as configured in **Section 5.2.1**.
- Set the **Send Port** to ***5060***.

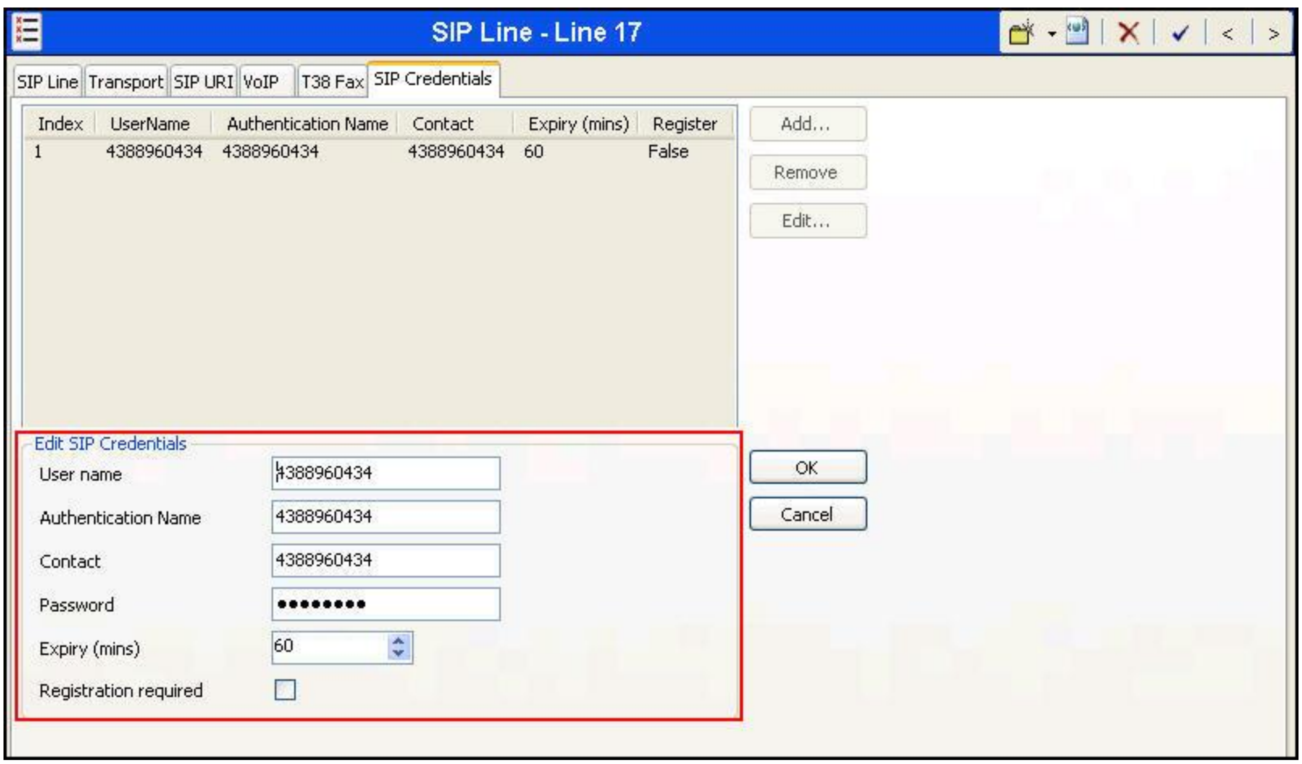


5.4.4. SIP Line – SIP Credentials Tab

SIP Credentials are used to register the SIP Trunk with a service provider that requires SIP Registration. SIP Credentials are also used to provide the required information for Digest Authentication of outbound calls. SIP Credentials are unique per customer and therefore customers must contact the service provider to obtain the proper registration credentials for their deployment.

Select the **SIP Credentials** 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. The screen below shows a sample entry being edited.

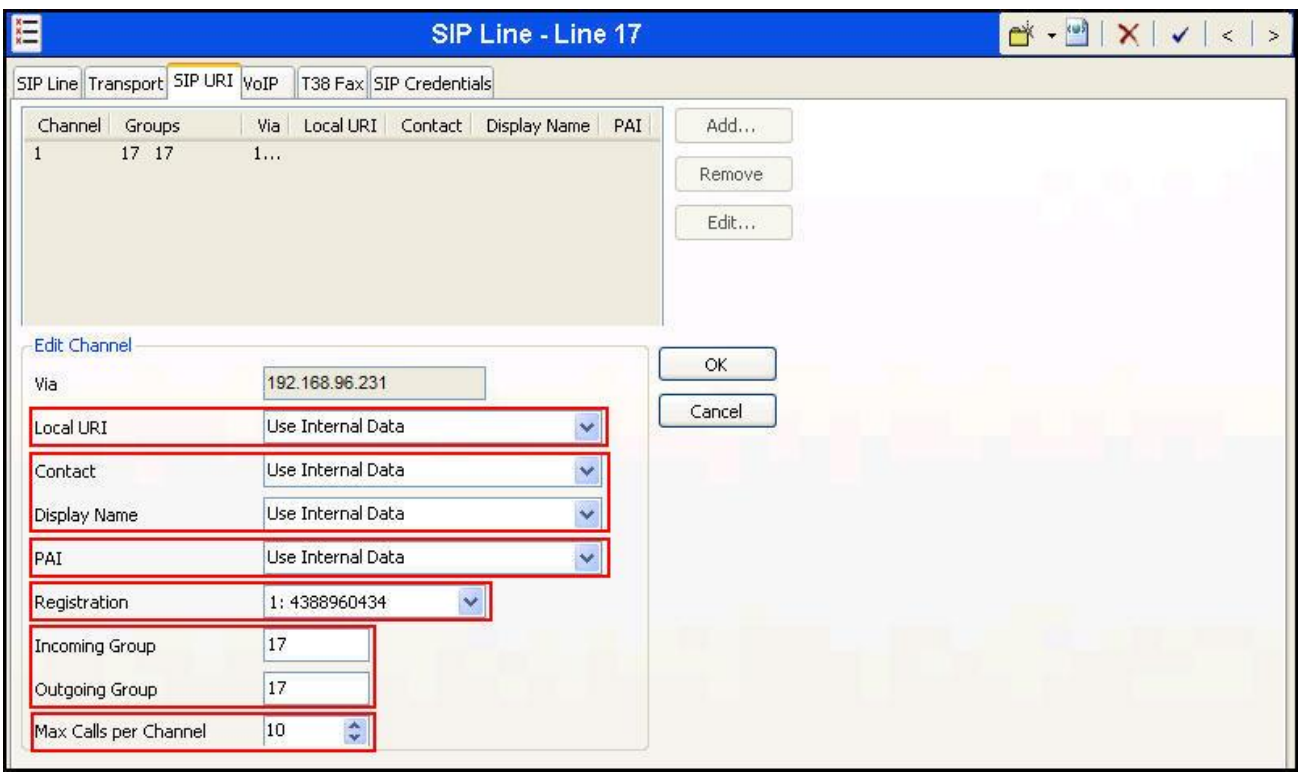
- Set **User name** and **Authentication Name** to the registration string provided by the service provider. This is generally a 10-digit telephone number as shown below.
- Set the **Contact** field to the value provided by the service provider.
- In the **Password** field, enter the password provided by the service provider.
- In the **Expiry (mins)** field, enter the time in minutes.
- Check the **Registration required** field if Registration is required for the SIP Trunking customer account, otherwise leave it unchecked. For the compliance test, ThinkTel did not require trunk registration.



5.4.5. SIP Line – SIP URI Tab

Select the **SIP URI** tab to create or edit a SIP URI entry. A SIP URI entry matches each incoming number that Avaya IP Office will accept on this line. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane. For the compliance test, a single SIP URI entry was created to match any DID number assigned to Avaya IP Office users. The following screen shows the edit window on a previously configured entry for the compliance test.

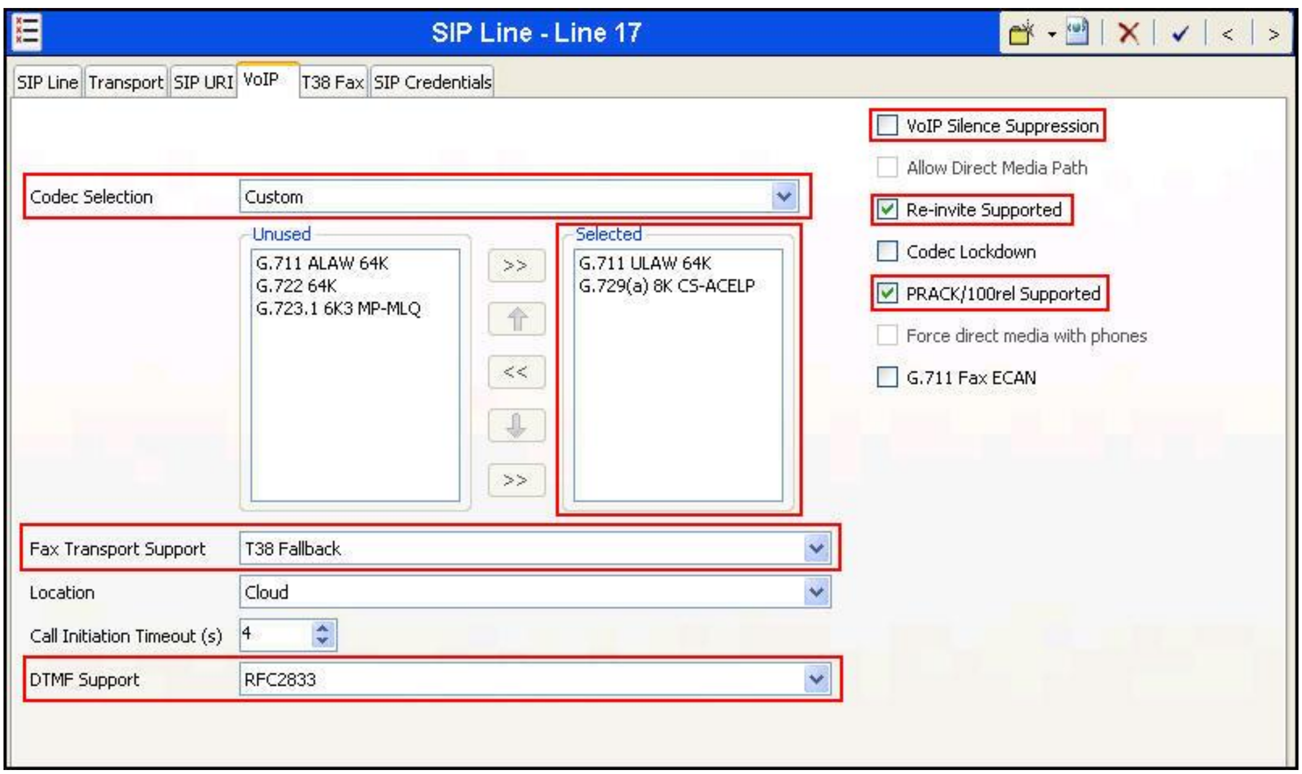
- Set **Local URI** to ***Use Internal Data***. This setting allows calls on this line whose SIP URI matches the **SIP Name** set on the **SIP** tab of any **User** as shown in **Section 5.6**.
- Set **Contact** and **Display Name** to ***Use Internal Data***. This setting will cause the Contact and Display Name data for outbound messages to be set from the corresponding fields on the **SIP** tab of the individual **User** as shown in **Section 5.66**.
- Set **PAI** to ***Use Internal Data***. This setting directs Avaya IP Office to send the PAI header (P-Asserted-Identity) when appropriate. The PAI header will be populated from the data set in the **SIP** tab of the call initiating **User** as shown in **Section 5.66**.
- Select the **Registration** value that was configured in **Section 5.4.4**.
- Associate this line with an incoming line group by entering 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. For the compliance test, ***17*** was specified for both the incoming and the outgoing group. Note that this group number can be different than the SIP Line number.
- Set **Max Calls per Channel** to the number of simultaneous SIP calls allowed.



5.4.6. SIP Line – VoIP Tab

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

- Set the **Codec Selection** to ***Custom***.
- Choose ***G.711 ULAW 64K*** and ***G.729(a) 8K CS-ACELP*** from the **Unused** box and move these two selections to the **Selected** box. Use the up and down arrows in the middle to order these 2 codes. The G.711MU and G.729A codecs are supported by ThinkTel. G.711MU was configured as the preferred codec for the compliance test as shown below.
- Select ***T38 Fallback*** for **Fax Transport Support** so that Avaya IP Office uses T.38 for sending and receiving faxes on this SIP line. If the called destination does not support T.38, the system will send a re-INVITE to change the transport method to G.711 (for falling back to G.711 pass-through fax).
- Set the **DTMF Support** field to ***RFC2833***. This directs Avaya IP Office to send DTMF tones as out-band RTP events as per RFC2833.
- Uncheck the **VoIP Silence Suppression** option box.
- Verify that **Allow Direct Media Path** is disabled (see the **Direct Media** item in the observation/limitation list in **Section 2.2**).
- Check the **Re-invite Supported** option box.
- Check the **PRACK/100rel Supported** option box. This setting enables support by Avaya IP Office for the PRACK (Provisional Reliable Acknowledgement) message on SIP trunks.

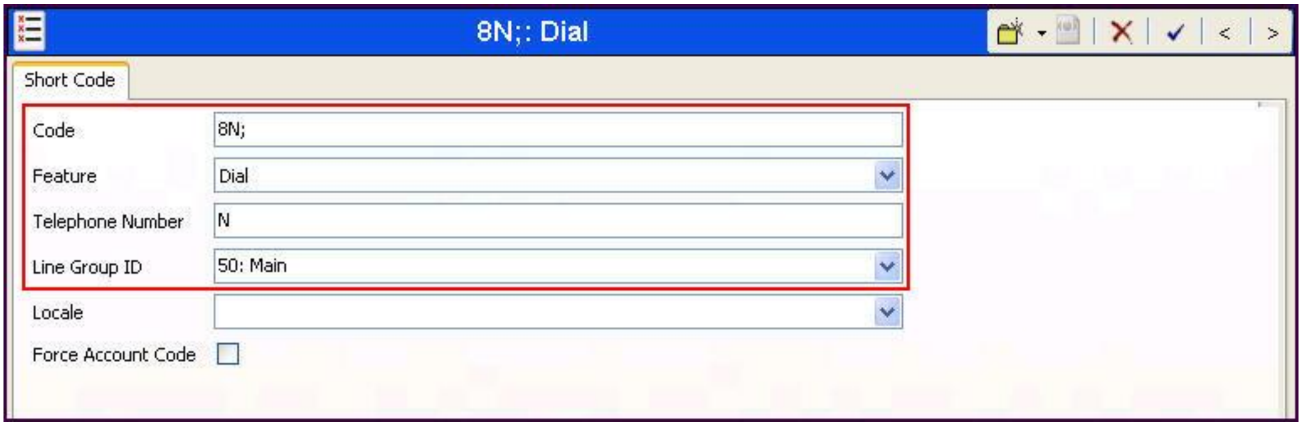


5.5. Short Code

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

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. The ***9N;*** short code, used for the compliance test, will be invoked when the user dials 9 followed by any number.
- Set **Feature** to ***Dial***. This is the action that the short code will perform.
- Set **Telephone Number** to ***N"@192.168.250.100"***. This field is used to construct the Request URI and the To header in the outgoing SIP INVITE message. The value ***N*** represents the number dialed by the user. The IP address following the **@** sign is the IP address of the ThinkTel access interface towards which the outbound call is sent.
- Set the **Line Group Id** to the **Outgoing Group** number defined on the **SIP URI** tab on the SIP Line in **Section 5.4.5**. This short code will use this line group when placing the outbound calls.

The simple **9N;** short code 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 dialed digits following the short code are desired, the Automatic Route Selection (ARS) feature may be used. In the screen below, the short code ***8N;*** is illustrated for access to ARS. When the Avaya IP Office user dials 8 plus any number ***N***, rather than being directed to a specific **Line Group Id**, the call is directed to ***50: Main***, configurable via ARS. See **Section 5.8** for example ARS route configuration.

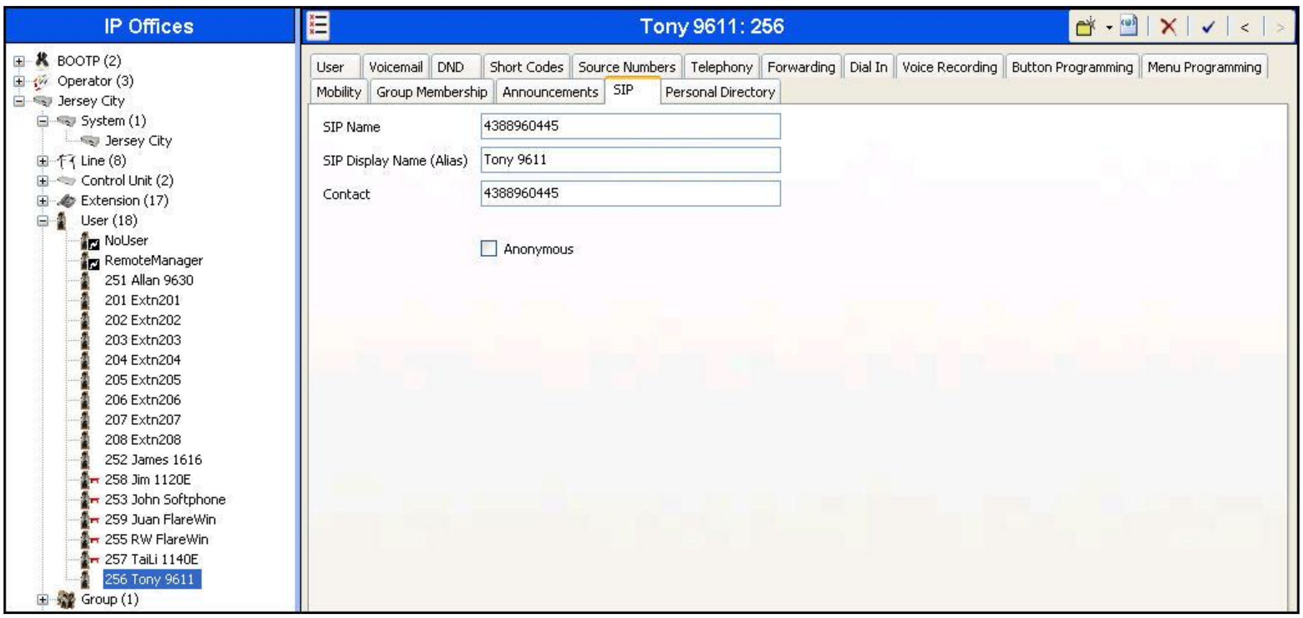


Optionally, add or edit a short code that can be used to access the SIP Line anonymously. In the screen shown below, the short code ****67N;*** is illustrated. This short code is similar to the **9N;** short code except that the **Telephone Number** field begins with the letter ***W***, which means "withhold the outgoing calling line identification". In the case of the compliance test, when a user dialed *67 plus the number, Avaya IP Office would include the user's telephone number (DID number assigned to the user) in the **P-Asserted-Identity** (PAI) header and would include the **Privacy: id** header in the outbound INVITE message. Consequently ThinkTel would prevent presentation of the caller id to the called PSTN destination.

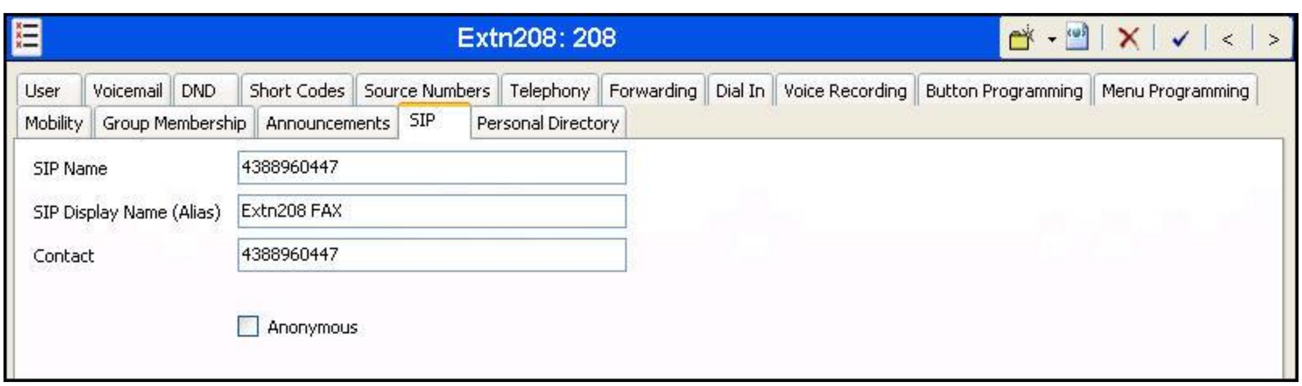
5.6. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP Line. To configure these settings, first navigate to **User→Name** in the Navigation Pane, where **Name** is the name of the user to be modified. In the example below, the name of the user is ***Tony 9611*** at extension 256. Select the **SIP** tab in the Details Pane. The **SIP Name** and **Contact** are set to one of the DID numbers provided by ThinkTel. The **SIP Display Name (Alias)** can optionally be configured with a descriptive text string. The value entered for the **Contact** field will be used in the Contact header for outgoing SIP INVITE to the service provider. The value entered for the **SIP Name** is used as the user part of the SIP URI in the From header for outgoing SIP trunk call INVITE.

If outbound 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 (or alternatively use the ***67N;** short code as defined in **Section 5.5**).



The following screen shows the similar SIP settings for an analog extension user for fax:

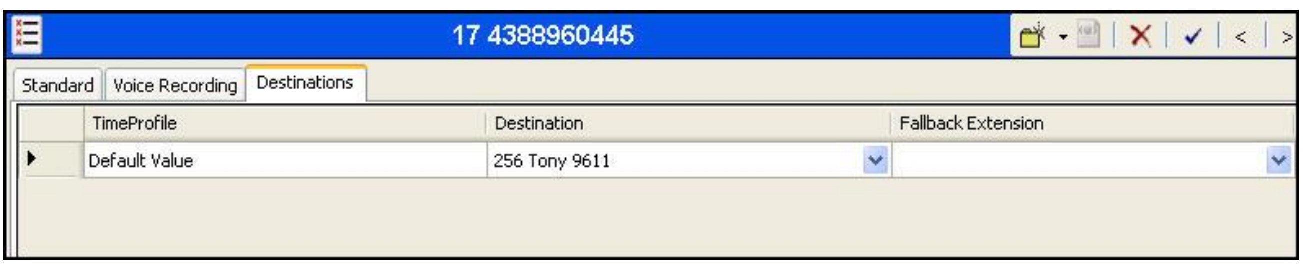


5.7. Incoming Call Route

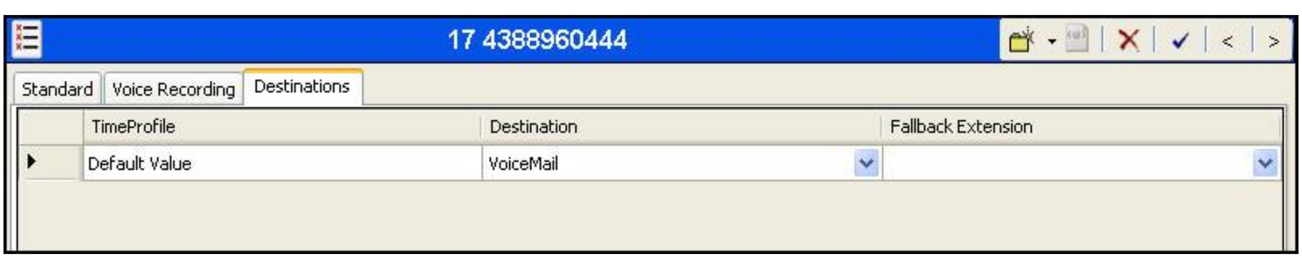
An incoming call route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by the service provider. To create an incoming call route, right-click **Incoming Call Route** in the Navigation Pane and select **New** (not shown).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

- Set the **Bearer Capacity** to ***Any Voice***.
- Set the **Line Group Id** to the **Incoming Group** of the SIP Line defined in **Section 5.4.5**.
- Set the **Incoming Number** to the incoming DID number on which this route should match.

On the **Destinations** tab, select the destination from the pull-down list of the **Destination** field. In this example, incoming calls to the DID number 14388960445 on Incoming Group 17 are to be routed to the user "Tony 9611" at extension 256.



The screen below shows calls routed to IP Office Voicemail Pro for message retrieval.



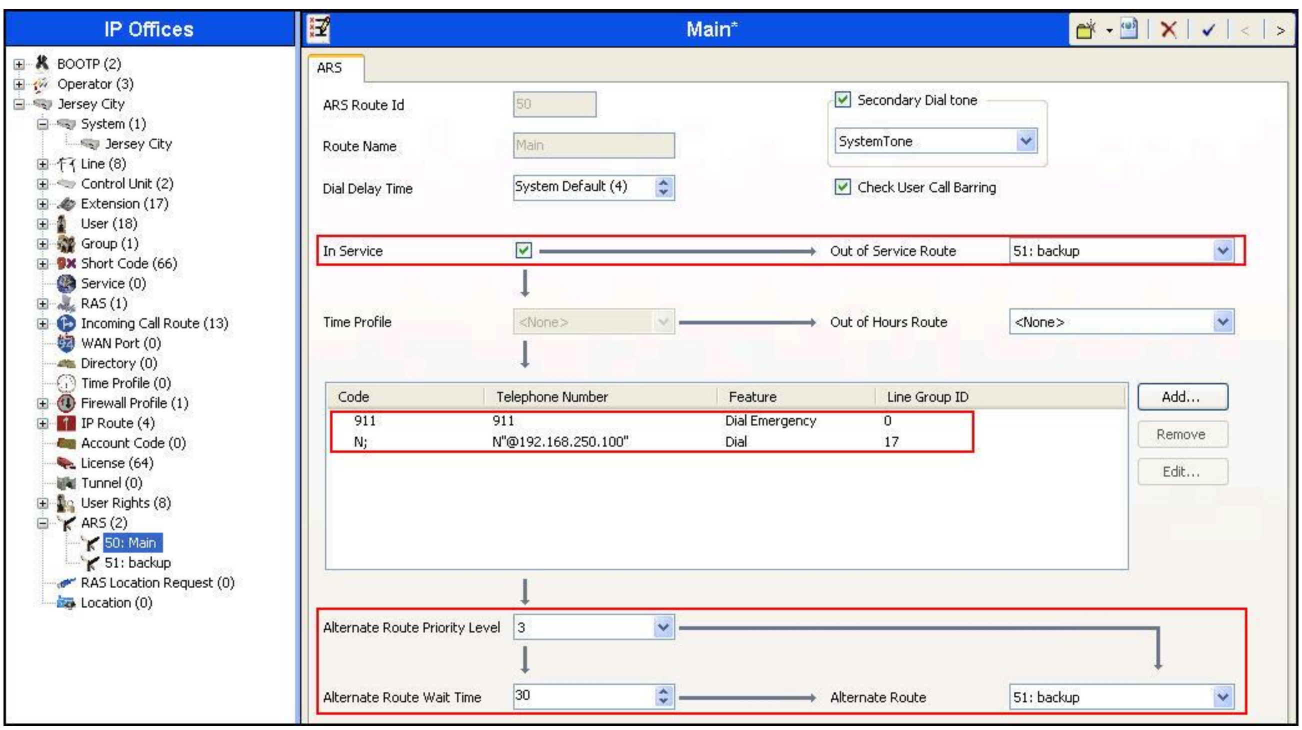
5.8. ARS and Alternate Routing

While detailed coverage of Automatic Route Selection (ARS) is beyond the scope of these Application Notes, this section includes basic ARS screen illustration 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 to supplement or replace the simple **9N;** short code approach documented in **Section 5.5**. With ARS, 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. 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 local and long distance calls should use the SIP Line, but service numbers should prefer a different outgoing line group, ARS can be used to distinguish between the two call patterns.

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, expand ARS in the Navigation Pane and select a route name.

The following screen shows an example ARS configuration for the route named ***50: 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.



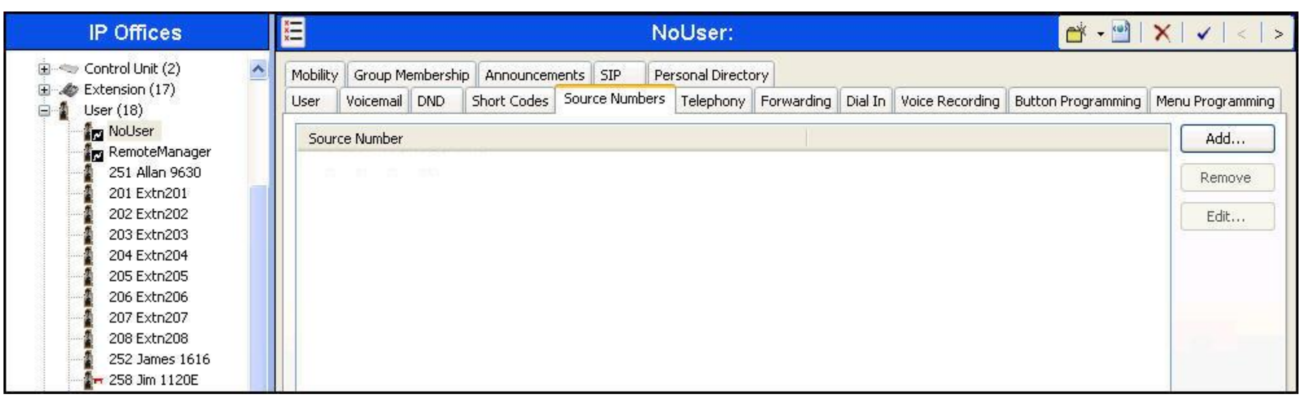
Assuming the primary route is in-service, the number passed from the short code used to access ARS (e.g., **8N;** in **Section 5.5**) can be further analyzed to direct the call to a specific Line Group ID. Per the example screen above, if the user dialed 8-911, the call would be directed to Line Group 0 to be sent out to the local area emergency response center (note that a short code 911 can also be configured to send the emergency call out when the user simply dials 911). If the user dialed 8 + any other number, the call would be directed to Line Group 17 as configured in **Section 5.4.5**. If the primary route cannot be used, the call can automatically route to the route name specified in the **Alternate Route** field in the lower right of the screen (***51: Backup***). Since alternate routing is considered a privilege not available to all callers, IP Office can control access to the alternate route by comparing the calling user's priority, configured in the **User** tab of individual users, to the value in the **Alternate Route Priority Level** field.

5.9. SIP Options

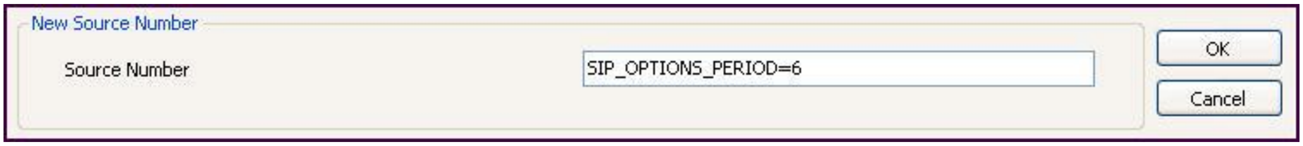
Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. By default, Avaya IP Office Release 9.0 sends out OPTIONS every 300 seconds. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.2.1** and the **SIP_OPTIONS_PERIOD** parameter (in minutes) that can be set on the **Source Number** tab of the **noUser** user. The OPTIONS period is determined in the following manner:

- To use the default value, set **Binding Refresh Time** to 300. OPTIONS will be sent at the 300 second frequency.
- To establish a period of less than 300 seconds, do not define the **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 300 seconds. The OPTIONS message period will be equal to the **Binding Refresh Time** setting.
- To establish a period greater than 300 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 300 seconds. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD** settings.

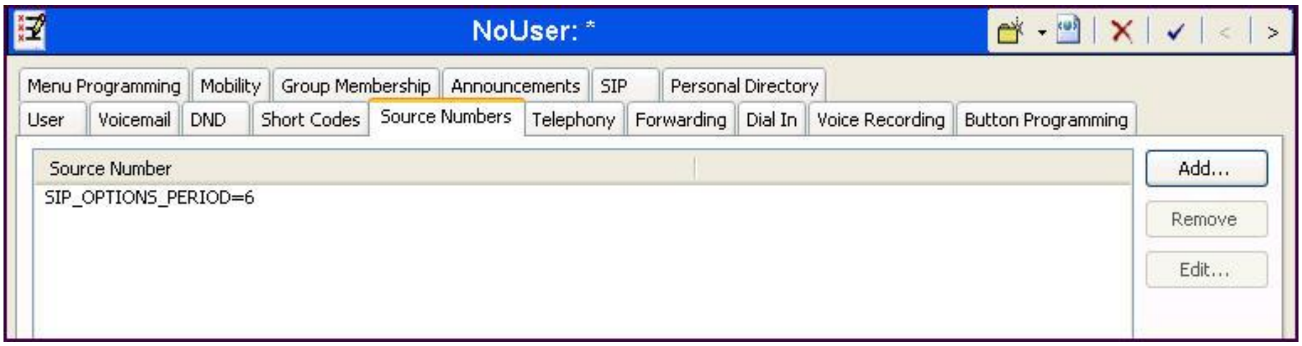
To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User → noUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



At the bottom of the Details Pane, the **Source Number** field will appear. Enter ***SIP_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. Click **OK** at the bottom of the screen (not shown).



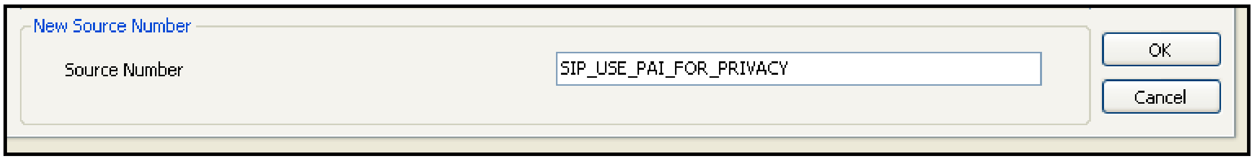
For the compliance test, an OPTIONS period of 3 minutes was desired. The **Binding Refresh Time** was set to ***180*** seconds in **Section 5.2.1**. Thus, there was no need to define **SIP_OPTIONS_PERIOD**.

5.10. Privacy/Anonymous Calls

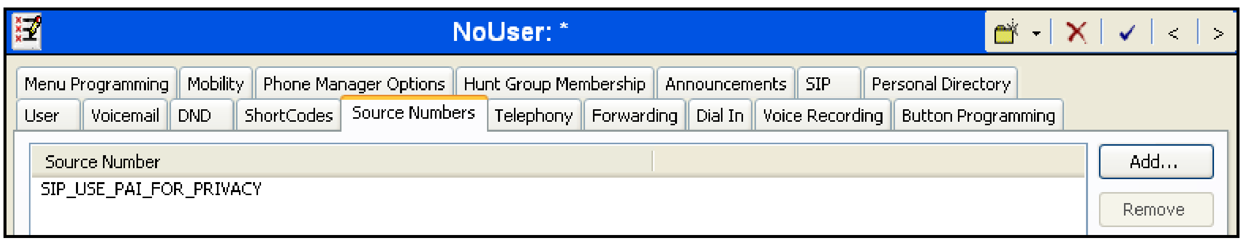
For outbound calls with privacy (anonymous) enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with "anonymous". Avaya IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the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

To configure Avaya IP Office to use PAI for privacy calls, select **NoUser** under **User** in the Navigation Pane, then select the **Source Numbers** tab in the Details Pane as shown in the first screen in **Section 5.9**. 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**.



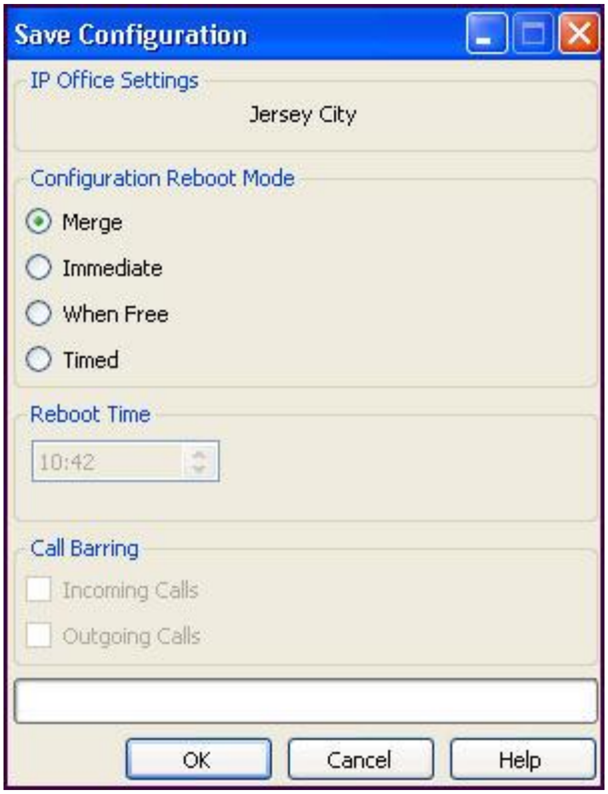
The **SIP_USE_PAI_FOR_PRIVACY** parameter will appear in the list of Source Numbers as shown below. Click **OK** at the bottom of the screen (not shown).



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.

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



6. ThinkTel SIP Trunking Configuration

ThinkTel is responsible for the configuration of the ThinkTel SIP Trunking Service. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise site (i.e., the IP address of the LAN2 port on the Avaya IP Office). ThinkTel will provide the customer the necessary information to configure the Avaya IP Office including:

- Access interface IP address of the ThinkTel SIP Trunking Service.
- Transport and port for the SIP connection to the enterprise site.
- DID numbers to assign to users at the enterprise.
- Supported codecs and their preference order.

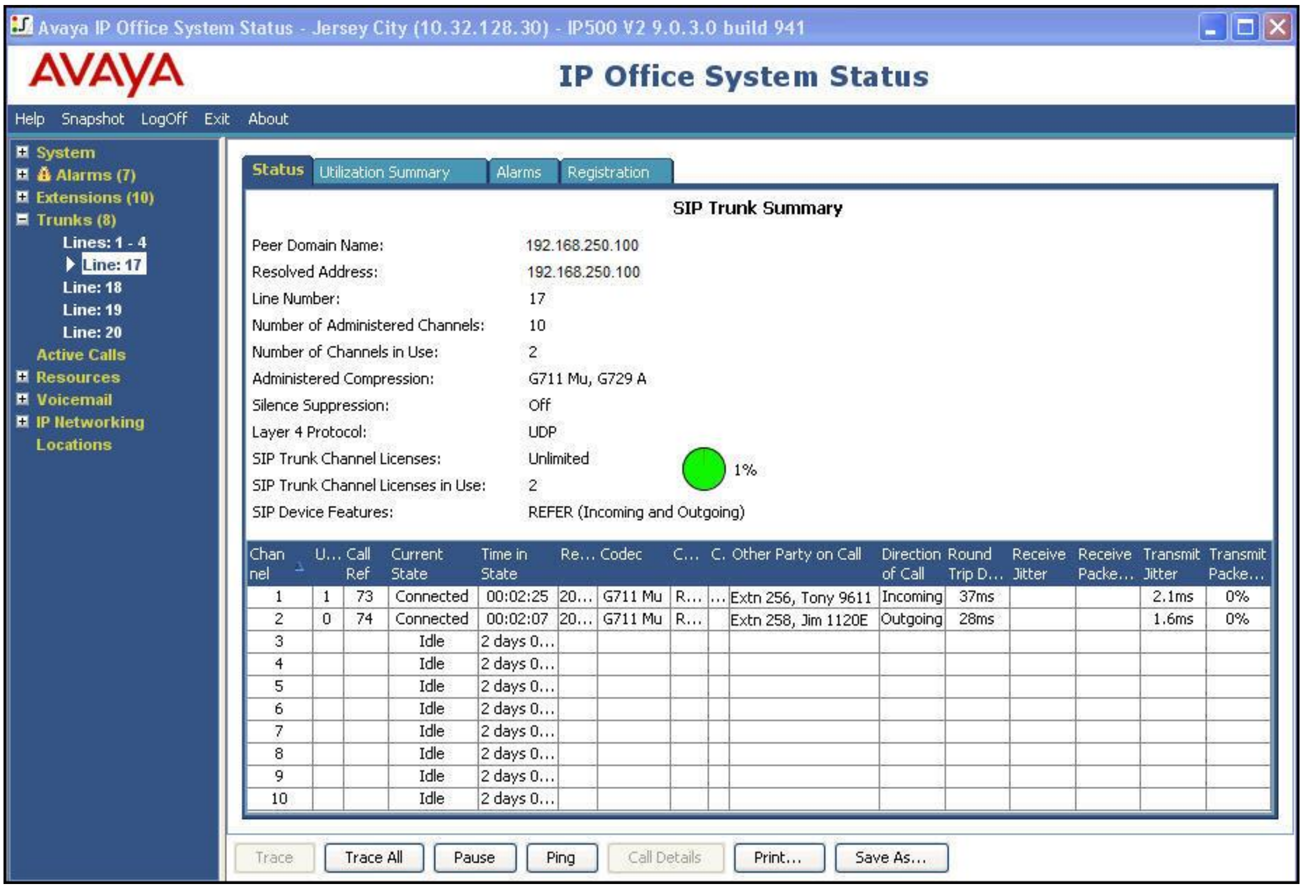
7. Verification Steps

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

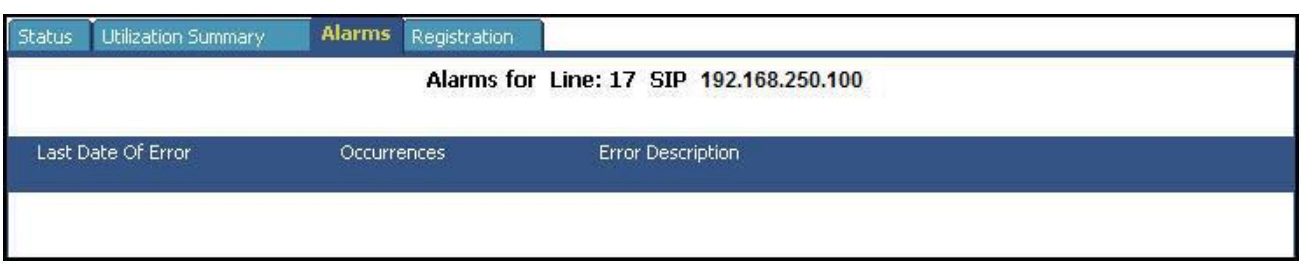
7.1. System Status

Use the Avaya IP Office System Status application to check the SIP Line channels state and alarms:

- Launch the application from **Start → Programs → IP Office → System Status** on the Avaya IP Office Manager PC. Select the SIP Line under **Trunks** from the left pane. In the **Status** tab in the right pane, verify the **Current State** is ***Idle*** for channels not taken by active calls; the state should be ***Connected*** for the channels engaged in active calls.

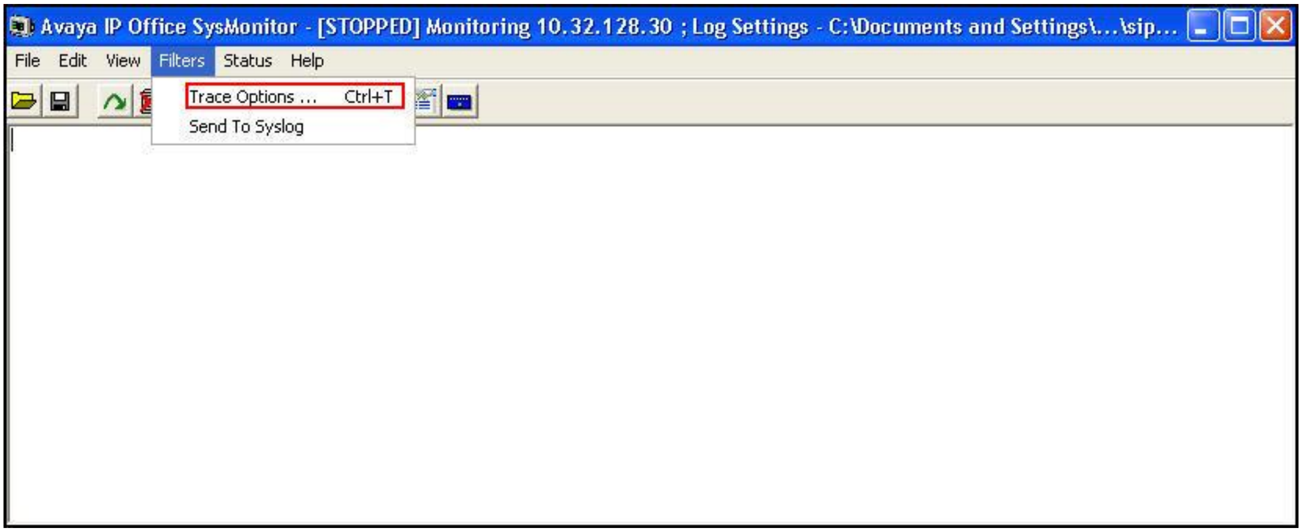


- Select the **Alarms** tab and verify that no alarms are active on the SIP Line.

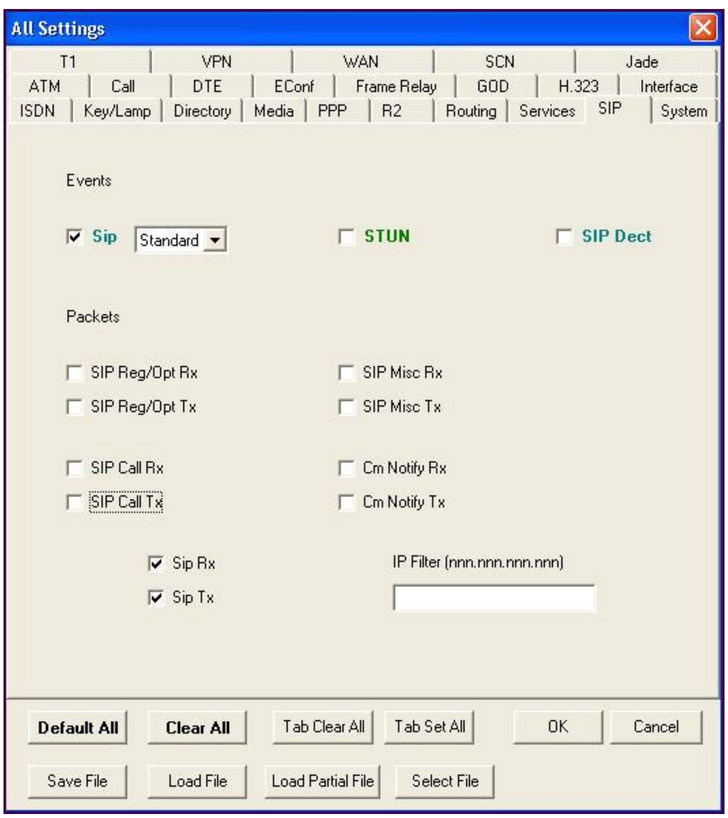


7.2. Monitor

The Monitor application can be used to monitor and troubleshoot Avaya IP Office. Monitor can be accessed from **Start → Programs → IP Office → Monitor** on the Avaya IP Office Manager PC. The application allows monitored information to be customized. To customize, select **Filters → Trace Options…** as shown below:



The following screen shows the **SIP** tab of trace options. In this example, **Standard** SIP Events and the **SIP Rx** and **SIP Tx** boxes are checked.



8. Conclusion

The ThinkTel SIP Trunking Service passed compliance testing with Avaya IP Office Release 9.0.3. These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office Release 9.0.3 and the ThinkTel SIP Trunking Service as shown in **Figure 1**. Test results and observations are noted in **Section 2.2**.

9. Additional References

[1] *IP Office Documentation Library,* Release 9.0, Documentation number 15-604278 Issue 1, September 2013.

[2] *IP Office 9.0.3 Product Description,* Documentation number 15-601041 Issue 27.03.0, May 2014.

[3] *Avaya IP Office 9.0.3 Installing IP500/IP500 V2,* Document number 15-601042 Issue 29f, June 2014.

[4] *Avaya IP Office 9.0.3 Administering Voicemail Pro,* Document number 15-601063 Issue 09d, July 2014.

[5] *Avaya IP Office Manager Release 9.0.3,* Document number 15-601011 Issue 9.0.3, May 2014.

[6] *Avaya IP Office 9.0 Using System Status*, Document number 15-601758 Issue 09c, August 2013.

[7] *Avaya IP Office 9.0 Using IP Office System Monitor*, Document Number 15-601019, Issue 05e, February 2014.

[8] *Avaya IP Office 9.0 H.323 Telephone Installation*, Document Number 15-601046, Issue 19c, February 2014.

[9] *Avaya IP Office 9.0 SIP Extension Installation*, Issue 3c, August 2013.

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

Product documentation for ThinkTel SIP Trunking Service 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.