Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SoTel Systems SIPTrunking with Avaya IP Office R8.1- Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between SoTel Systems and Avaya IP Office R8.1.

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

SoTel Systems 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 SoTel Systems and Avaya IP Office R8.1.

In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 running Release 8.1 software, Avaya Voicemail Pro messaging application, SIP-based Avaya softphones (IP Office Softphone and Flare® Experience for Windows), and Avaya H.323 and SIP hard phones.

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

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 business site using Avaya IP Office to connect to SoTel Systems SIP Trunking service. This 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 interoperability compliance test.

- Response to SIP OPTIONS queries.
- Incoming PSTN calls to H.323 and SIP telephones at the business site. All inbound PSTN calls were routed to the business site across the SIP trunk from the service provider.
- Outgoing PSTN calls from H.323 and SIP telephones at the business site. All outbound PSTN calls were routed from the business site across the SIP trunk to the service provider
- Various call types including: local, long distance, outbound toll-free, international, and directory assistance.
- 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 to PSTN mobile phones on inbound calls.
- Use of SIP INVITE for call redirection to PSTN.

- Inbound and outbound long-duration calls stability.
- Inbound and outbound long holding time call stability.
- Response to incomplete call attempts and trunk errors.
- T.38 fax.

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.
- SoTel Systems SIP Trunking does not support Operator call (0) and Operator-Assisted call (0 + 10 digits).
- SoTel Systems SIP Trunking does not support use of SIP REFER message for call re-direction.

2.2. Test Results

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

- **OPTIONS –** SoTel Systems returned "405 Method not allowed" to OPTIONS from Avaya IP office. Though Avaya IP Office treated this message as a valid indication that the far end was alive, a normal "200 OK" response was appropriate and desirable.
- **Codec Mismatch –** When Avaya IP Office was configured with a codec unmatched by SoTel, outbound call INVITE would receive a "500 Internal Server Error" response. A more appropriate status message like "488 Unacceptable" was desirable.
- **Call Session Expiration –** Sotel Systems SIP Trunking did not originate session-refresh reINVITEs since it does not support session timer.
- **Inbound T.38 Fax –** At the end of a T.38 fax session, Avaya IP Office would issue an INVITE to SoTel to switch back to a voice codec (G.711MU or G.729A depending on the codec configuration on Avaya IP Office). Sotel responded with "200 OK" to the INVITE for switching back to G.711MU, but with "488 Not Acceptable Here" to the INVITE for switching back to G.729A. The fax call terminated properly and there was no perceivable user impact. Sotel Systems concluded after investigation that the 488 message was actually from the backbone carrier of the test circuit, and advised that this issue would not happen in production environments.

2.3. Support

For technical support on the SoTel Systems SIP Trunking service, contact SoTel Systems by

- Calling 877-697-6835
- Sending email via the Support link at www.sotelsystems.com

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 a business site connected to Sotel Systems SIP Trunking.

Located at the business site is an Avaya IP Office 500v2. Endpoints include various Avaya IP Telephones (with H.323 and SIP firmware) and SIP-based Avaya softphones (Avaya IP Office Softphone and Avaya Flare® Experience for Windows). The site also has a Windows PC running Avaya 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.

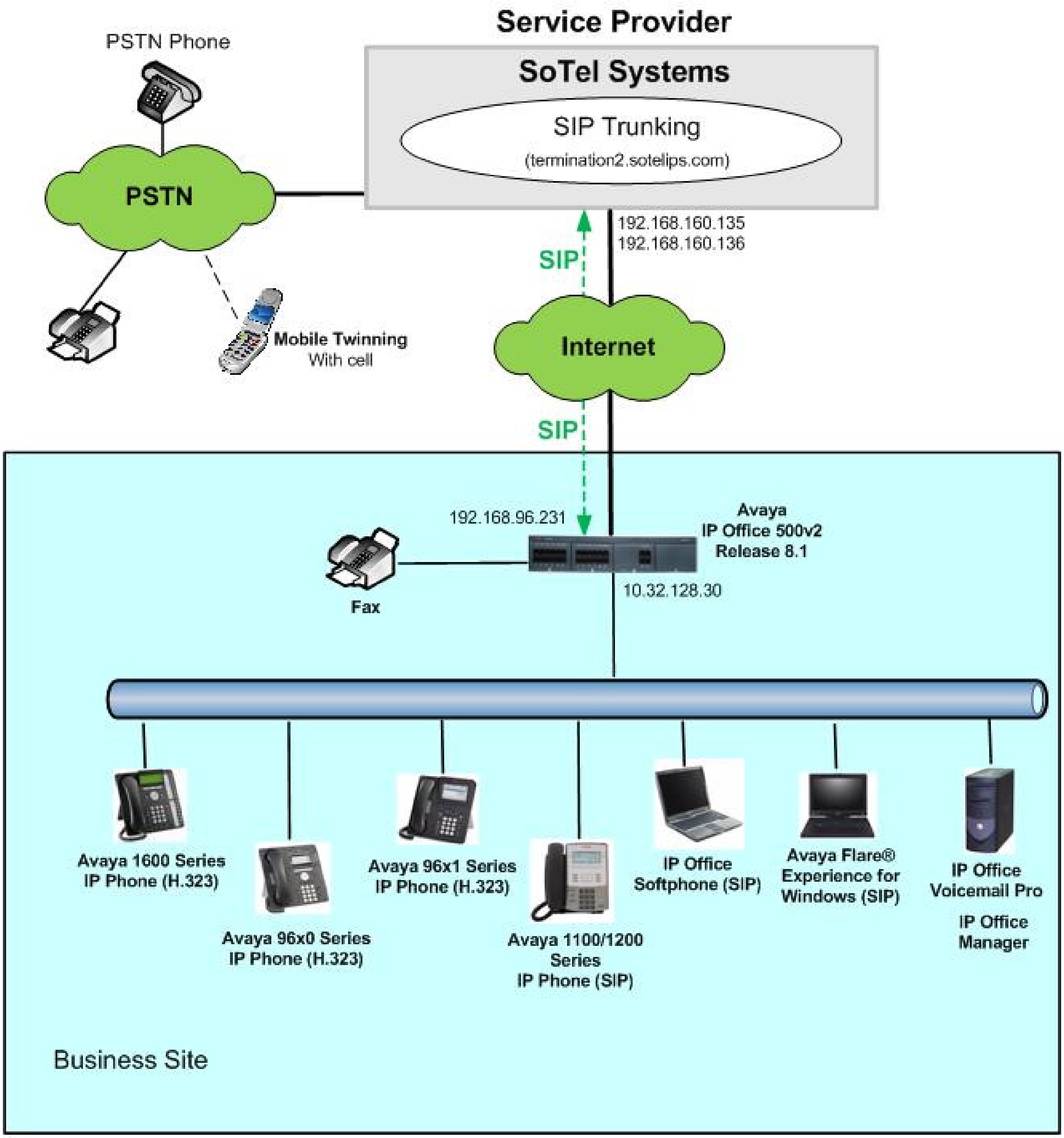


Figure 1: Test Configuration

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

SoTel Systems SIP Trunking is configured with two distinct border elements using separate IP addresses for connecting to the business sites. The inbound calls can be sent from either of these 2 border elements. For the compliance test, the Avaya IP Office was configured to receive inbound calls from either border element, but send outbound calls to the first border element at 192.168.160.135.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to send digits across the SIP trunk to SoTel Systems. The short code of 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 calls 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. SoTel Systems sent 11 digits in the Request URI and the To header of inbound SIP INVITE messages.

In an actual customer configuration, the business 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.

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 500v2	8.1 (69)
Avaya IP Office COMBO6210/ATM4 Module	8.1 (69)
Avaya IP Office Manager	10.1 (69)
Avaya Voicemail Pro	8.1 (9203)
Avaya 1600 Series IP Telephones (H.323)	Avaya one-X Deskphone 1.3
Avaya 9600 Series IP Telephones (H.323)	Avaya one-X Deskphone 3.1
Avaya 9611 Series IP Telephones (H.323)	Avaya one-X Deskphone 6.2
Avaya 1120E IP Telephone (SIP)	4.03.12.00
Avaya IP Office Softphone	3.2.3.48 67009
Avaya Flare® Experience for Windows	1.1.1.7
SoTel Systems Components	
Equipment / Software	**Release / Version**
ACME Net-Net 4500	SCX6.3.9 Patch 7 (Build 330)
Sonus GSX 9000	V09.00.00A203

5. Configure Avaya IP Office

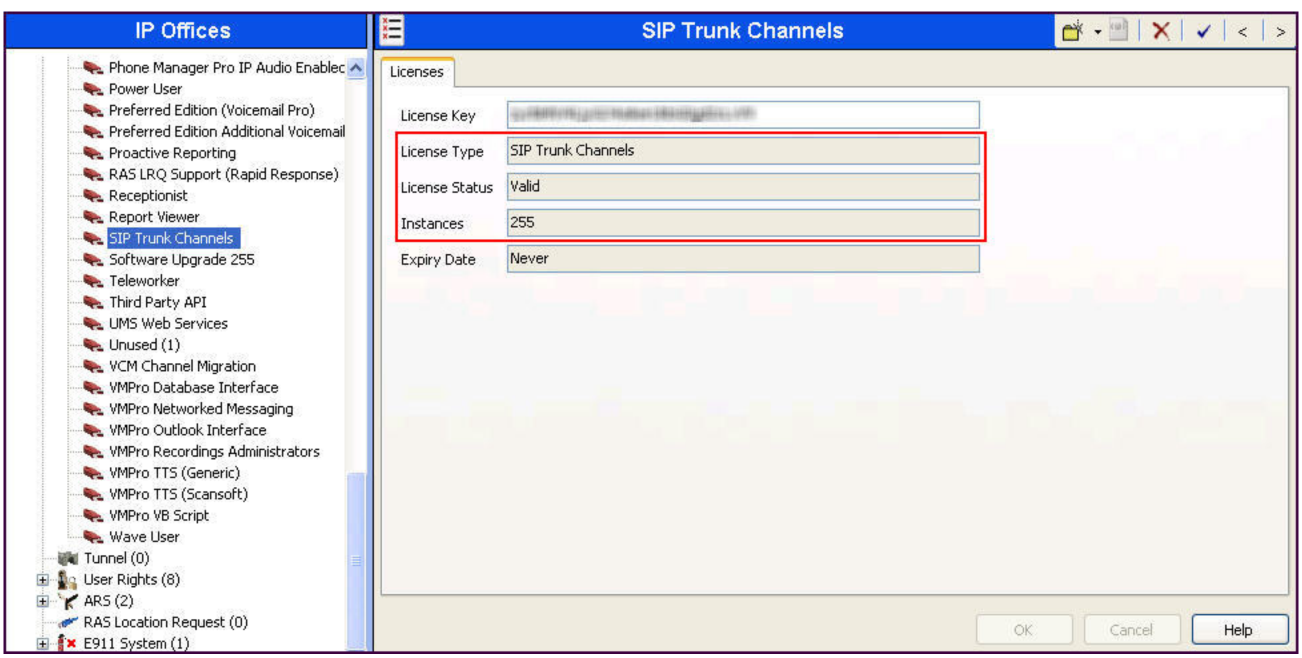
This section describes the Avaya IP Office configuration to support connectivity to the SoTel Systems SIP Trunking service. Avaya IP Office is configured through the Avaya IP Office Manager application. From the PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials. A management window similar to the one shown in the next section will appear. The appearance of the Avaya 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 and the Details pane on the right side. All licensing and feature configuration that is not directly related to the interface with the service provider (e.g., twinning and IP Office Softphone support, etc.) is assumed to already be in place.

The configuration screens in this section only show 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

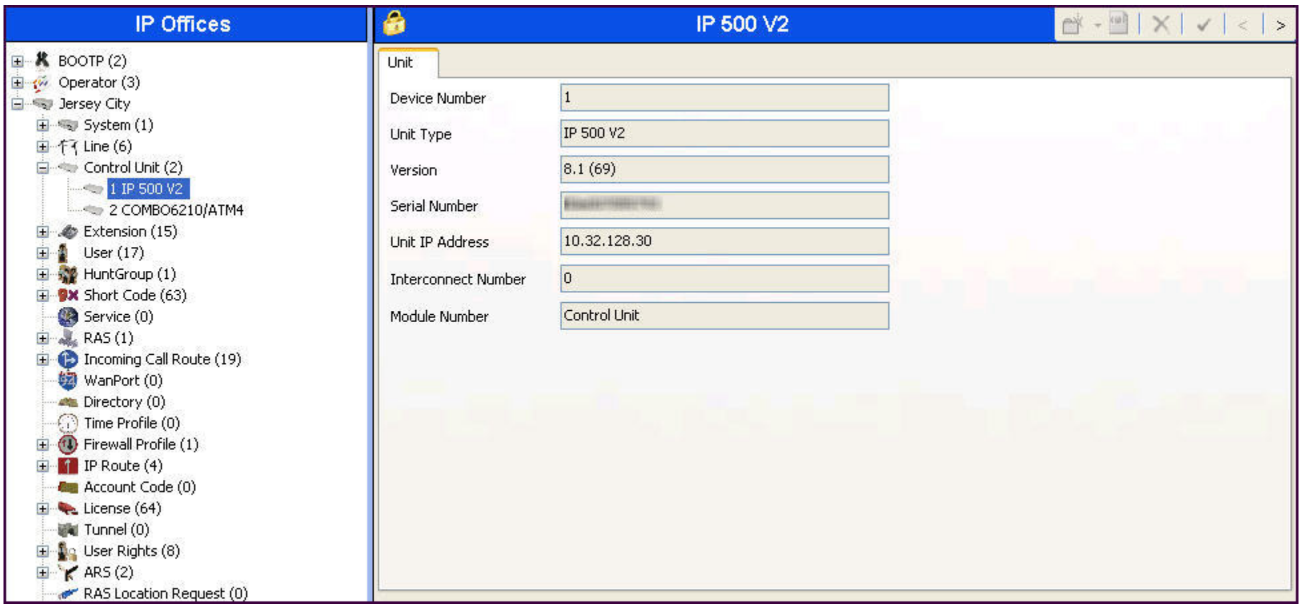
The configuration and features described in these Application Notes require Avaya IP Office to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

To verify that there is a SIP Trunk Channels License with sufficient capacity; click **License → SIP Trunk Channels** in the Navigation pane. Confirm a valid license with sufficient **Instances** (trunk channels) in the Details pane.



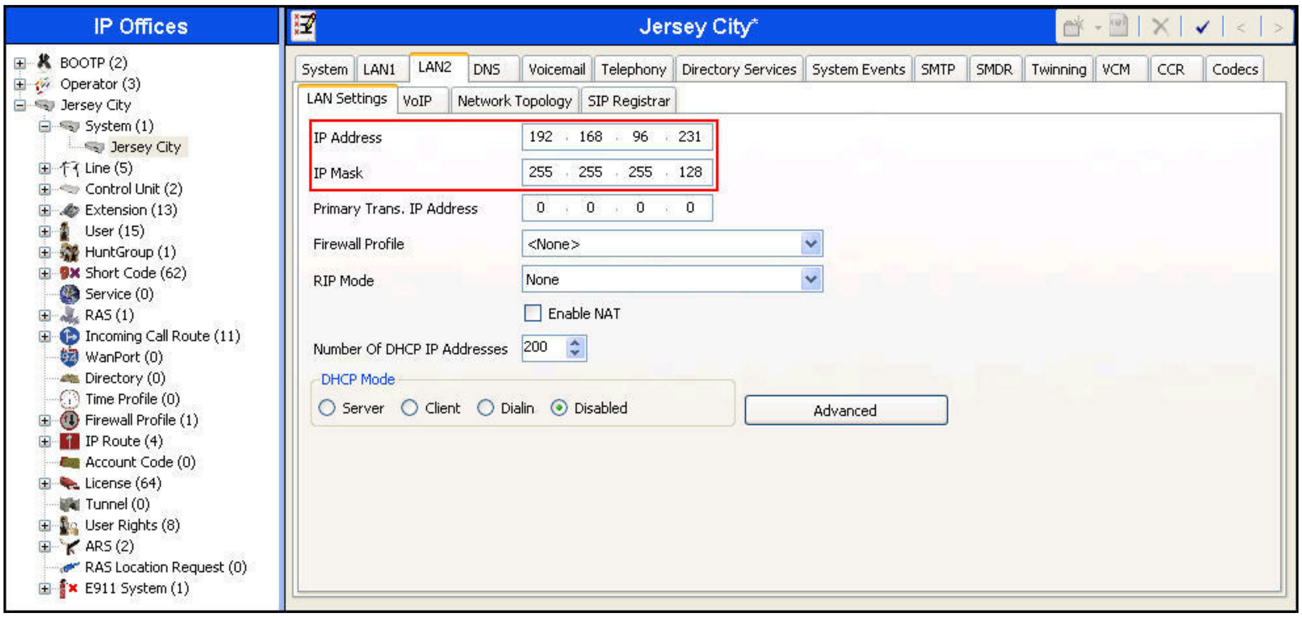
To view the physical hardware comprising the Avaya IP Office system, expand the components under the **Control Unit** in the Navigation pane. In the sample configuration, the second component listed is a Combination Card. This module has 6 digital stations ports, two analog extension ports, 4 analog 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 following screen shows the details of the **IP 500 V2**.



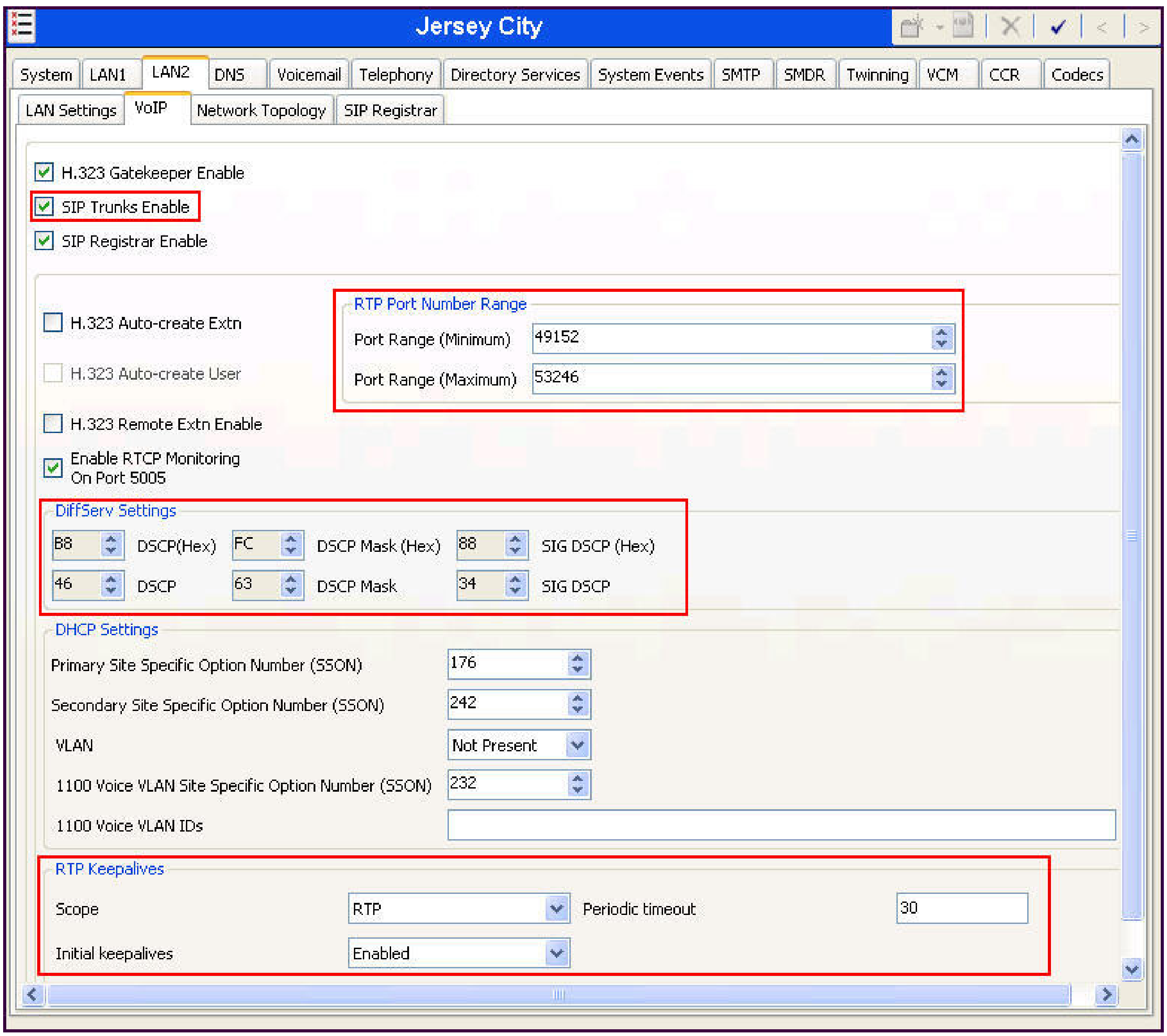
5.2. LAN2 Settings

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 **Jersey City → 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.



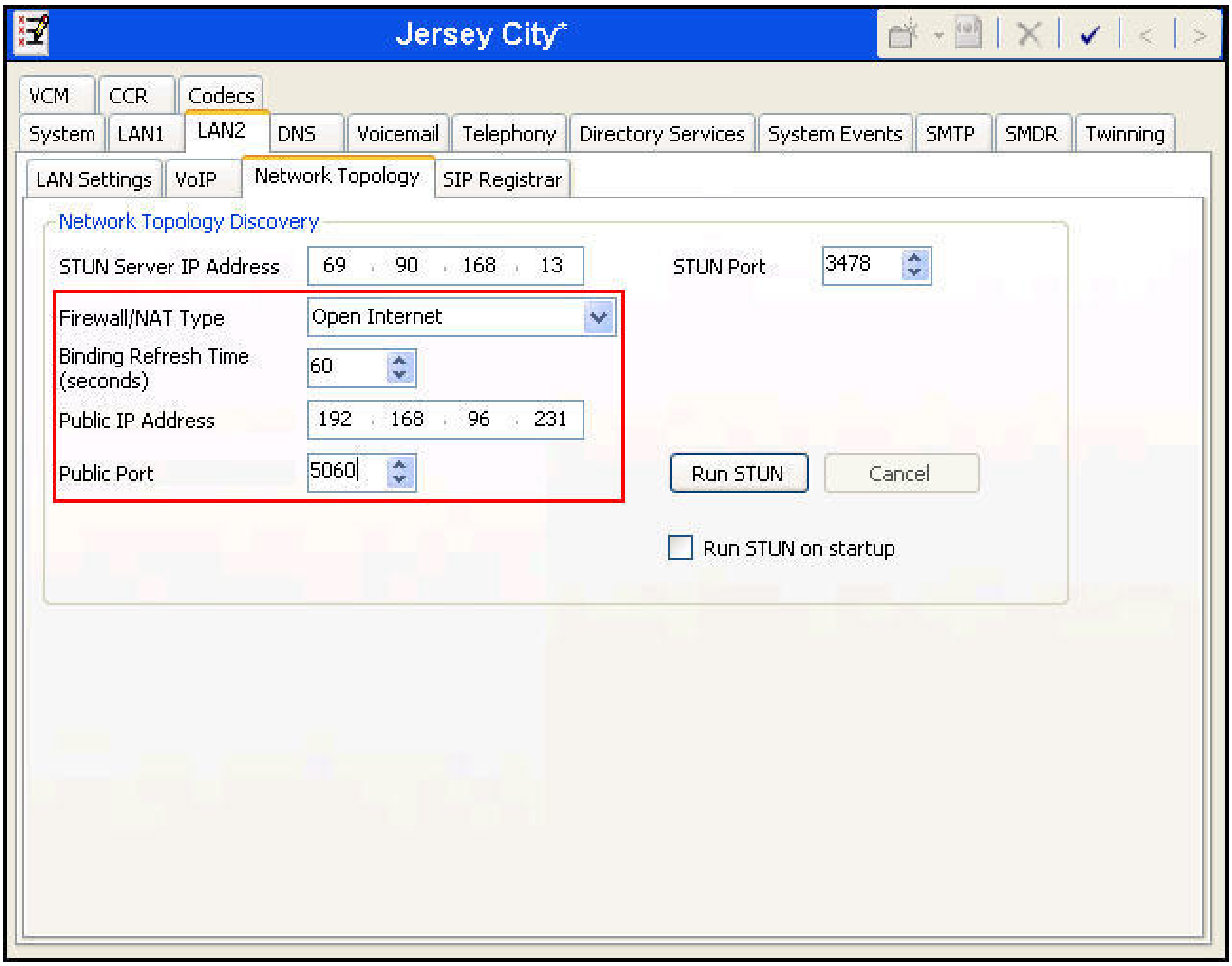
On the **VoIP** tab of LAN2 in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent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

In the **RTP Keepalives** section. Select ***RTP*** for **Scope**; select ***Enabled*** for **Initial keepalives**; enter ***30*** for **Periodic timeout**. These settings direct IP Office to send artificial RTP packets toward the service provider at the start of the call to prevent audio loss in certain off-net call redirection scenarios. This configuration was necessary because the service provider expected IP Office endpoint to send RTP packets first even though there was no IP Office media endpoint involved in this call situation since the call had been re-directed back to PSTN.



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

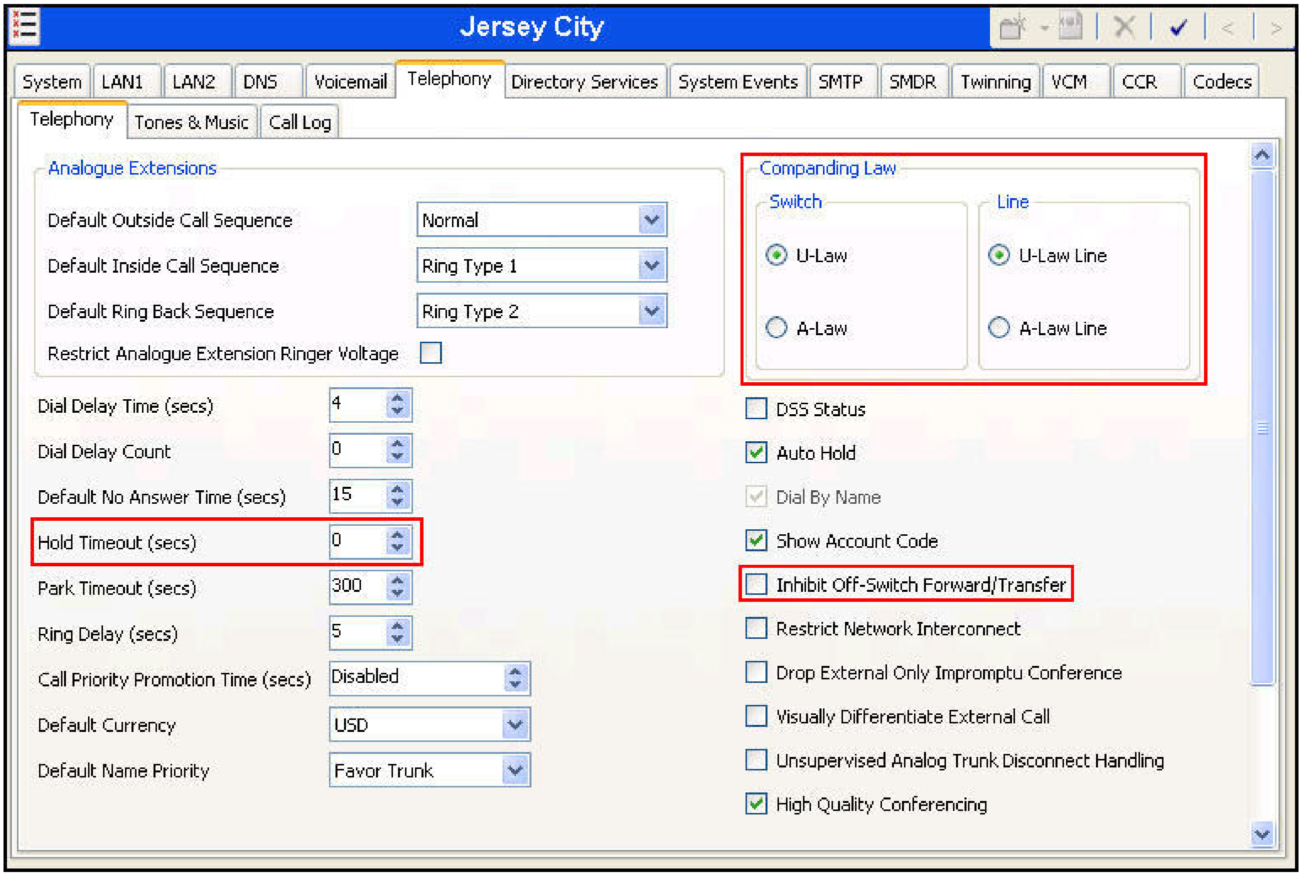
- Select **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to ***Open Internet***. With the ***Open Internet*** setting, **STUN Server IP 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 5.10** for complete details.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port.
- Set **Public Port** to ***5060***.



During the 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 SoTel Systems, and therefore is not described in these Application Notes.

5.3. System Telephony Settings

Navigate to the **Telephony → Telephony** tab on the Details Pane. Choose the **Companding Law** typical for the business site. For the compliance test, ***ULAW*** 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. Enter or select ***0*** for **Hold Timeout (secs)** so that calls on hold will not time out.



5.4. Twinning Calling Party Settings

Navigate to the **Twinning** Tab on the Details Pane. For the compliance test, the **Send original calling party information for Mobile Twinning** box was checked as shown below.

If this box is checked, Avaya IP Office will send the following in the SIP From Header of the INVITE message for twinning a call to a PSTN mobile phone. The value in the From header determines what gets displayed as the calling party number:

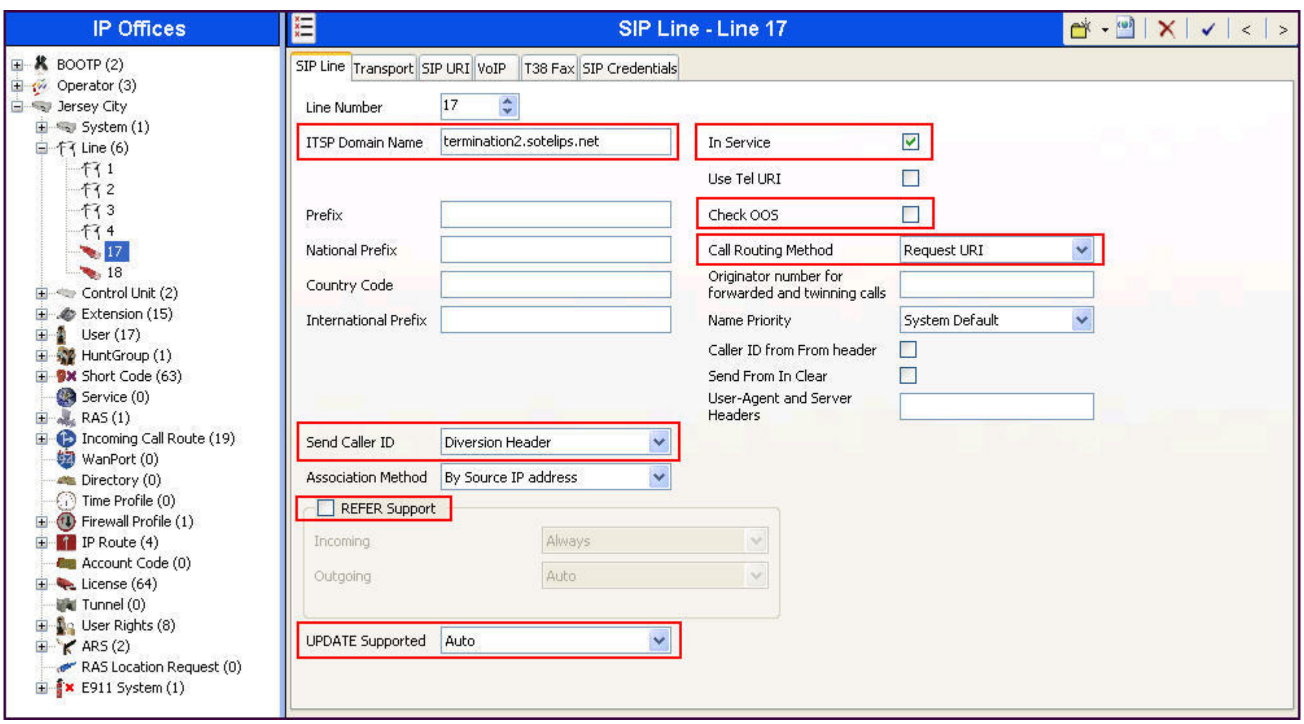
- On calls from an internal extension to another internal phone with twinning enabled, Avaya IP Office will send the calling party number of the originating extension (i.e., DID number assigned to this extension).
- On calls from the PSTN to an internal phone with twinning enabled, Avaya IP Office will send the originating PSTN calling party number.

5.5. Administer SIP Line (First)

A SIP line is needed to establish the SIP connection between Avaya IP Office and SoTel Systems SIP Trunking. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New → SIP Line** (not shown). On the **SIP Line** tab in the Details Pane, configure the parameters as shown below.

- Set the **ITSP Domain Name** to the domain name for IP-authenticated SIP accounts as provided by Sotel Systems.
- Check the **In Service** box.
- Uncheck the **Check OOS** box. SoTel Systems SIP Trunking responds to the SIP OPTIONS messages sent by Avaya IP Office with "405 Method not allowed" instead of the expected "200 OK" (as noted in **Section 2.2**). Thus, this setting will prevent Avaya IP Office from taking the SIP trunk 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 Avaya IP Office will include the Diversion Header for calls that are forwarded out the SIP Line to the PSTN.
- Uncheck **REFER Support**. SoTel Systems SIP Trunking does not support use of REFER for call re-direction.
- Set **UPDATE Supported** to ***Auto***. With this setting Avaya IP Office will send UPDATE messages for session refresh if the other party supports UPDATE.

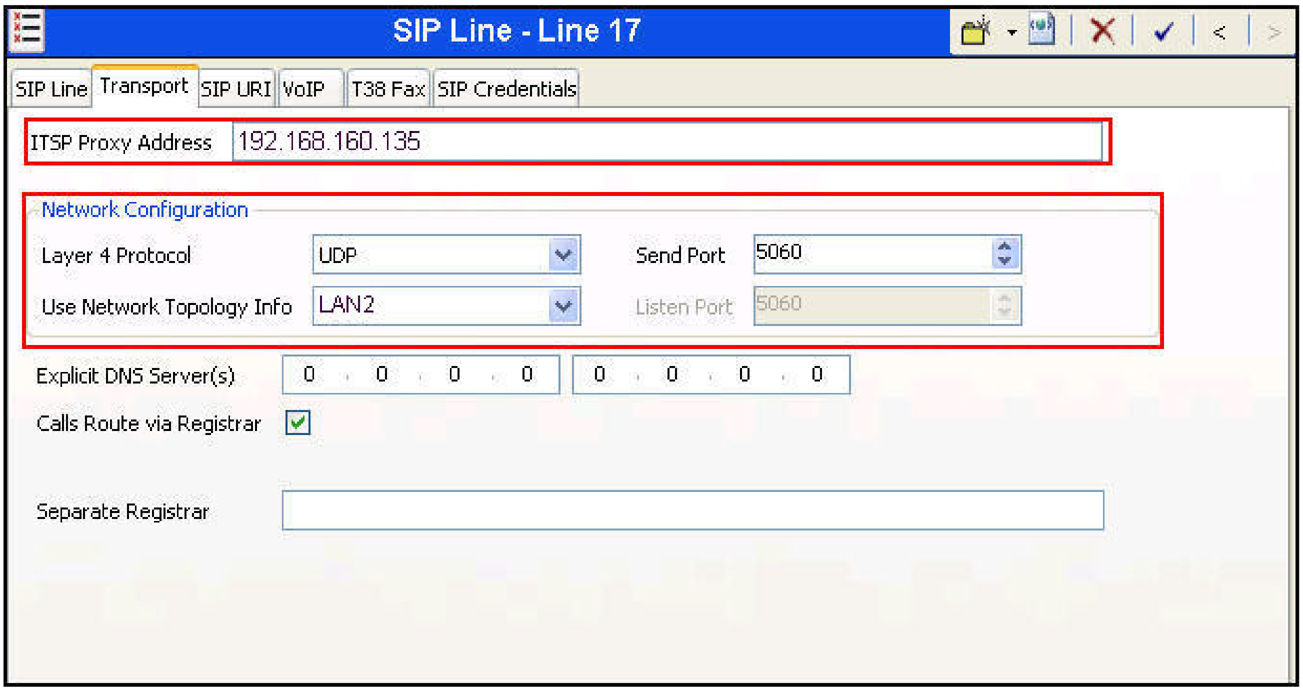
Default values may be used for all other parameters.



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 SoTel Systems.
- 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***.

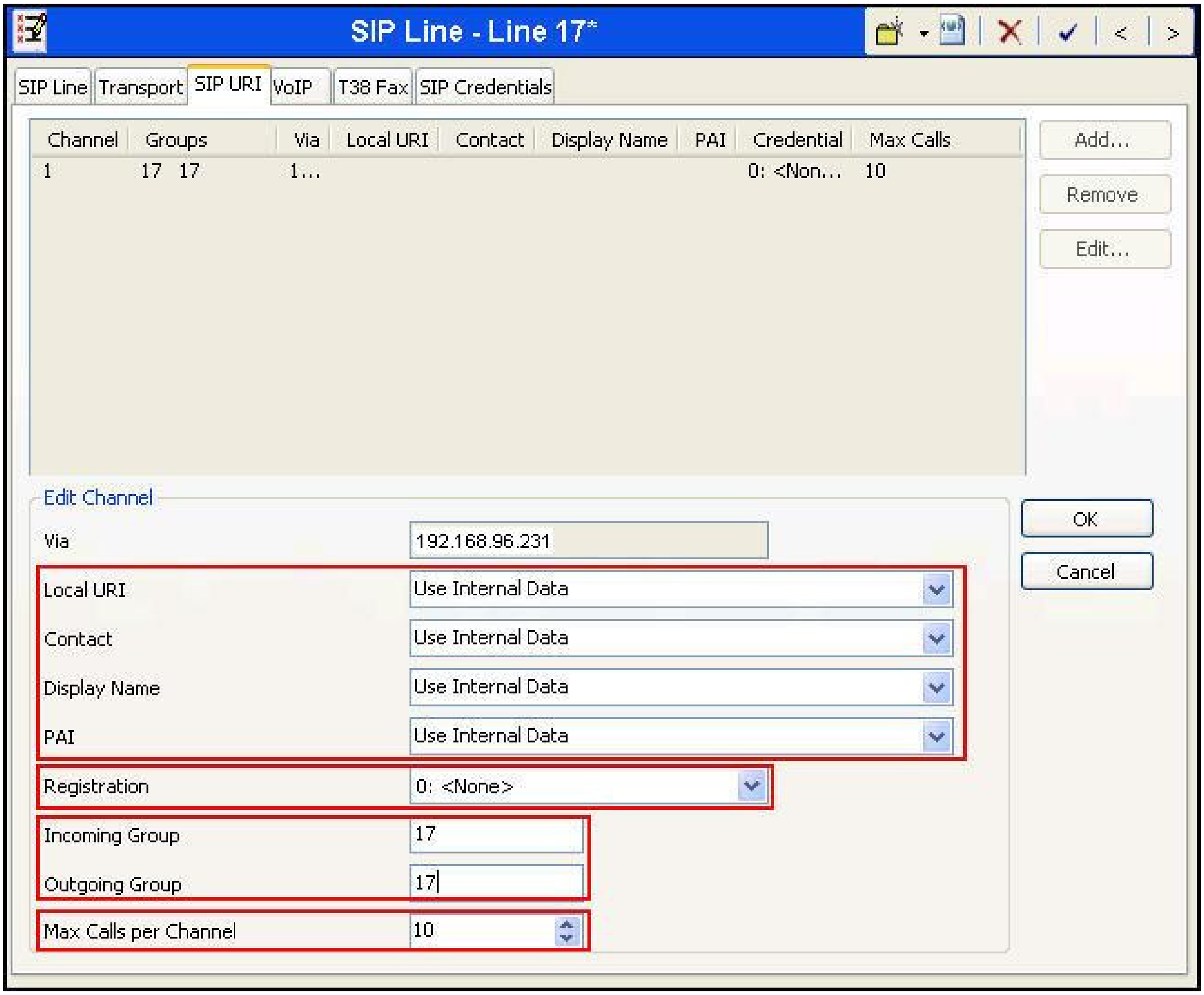
Default values may be used for all other parameters.



Select the **SIP URI** tab to create a SIP URI entry or edit an existing 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 that matched any DID number assigned to an Avaya IP Office user. The following screen shows the edit window on a previously configured entry for the compliance test.

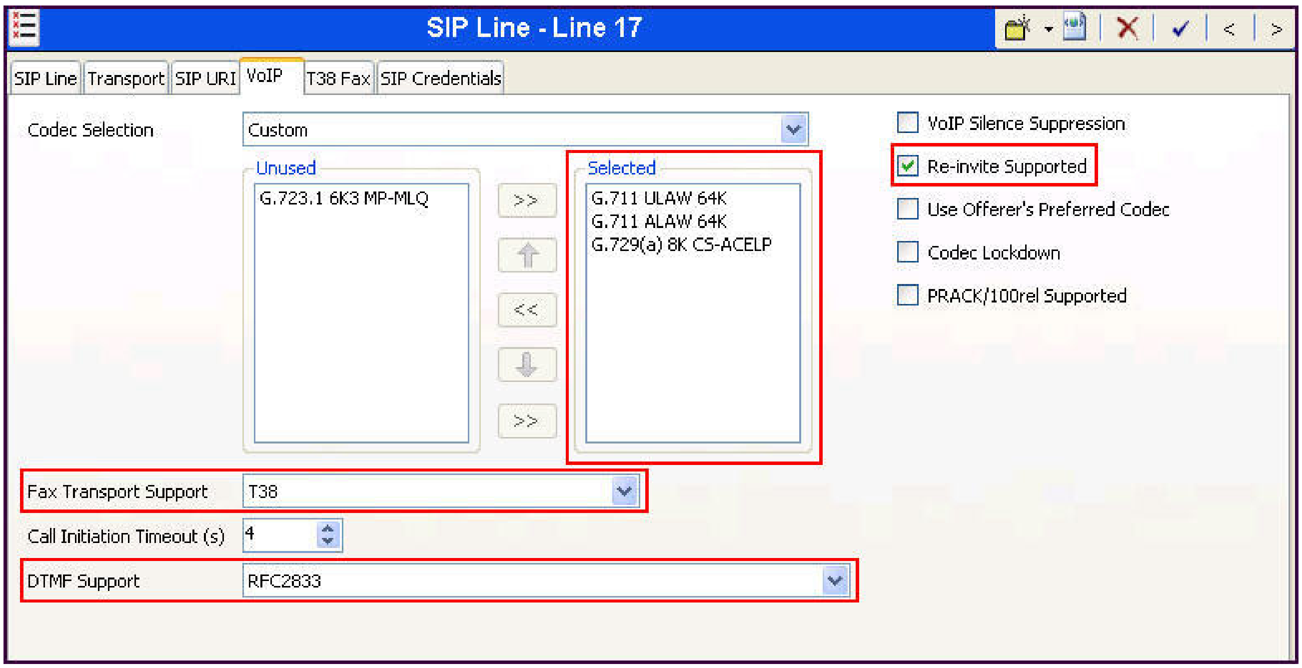
- Set **Local URI** to ***Use Internal Data***. This setting allows calls on this line whose SIP URI matches the **SIP Name** set on the **SIP** tab of any **User** as shown in **Section 5.8**.
- Set **Contact** and **Display Name** to ***Use Internal Data***. This setting will cause the Contact and Display Name data to be set from the corresponding fields on the **SIP** tab of the individual **User** as shown in **Section 5.88**.
- Set **PAI** to ***Use Internal Data***. This setting directs Avaya IP Office to send the PAI (P-Asserted-Identity) header 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 5.88**.
- For **Registration**, select ***0:<None>*** from the pull-down menu. The compliance test used an IP-authenticated SIP account that did not require trunk registration.

- 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, the incoming and outgoing group ***17*** was specified (Note: In the sample configuration, the line group number happened to be identical to the SIP Line number, but these two numbers do not need to be the same).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls allowed using this SIP URI pattern.



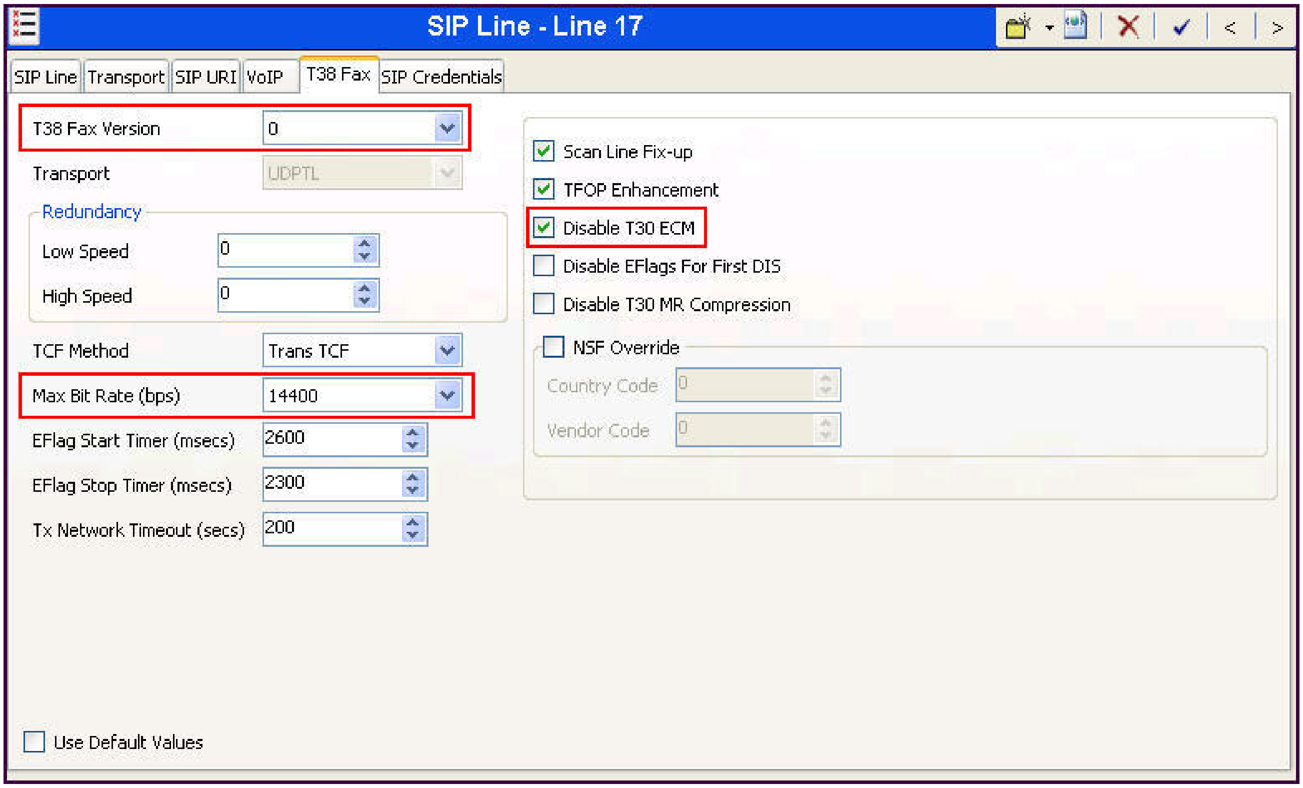
Select the **VoIP** tab, to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below.

- Set the **Codec Selection** to ***Custom***.
- Choose ***G.711 ULAW 64K***, ***G.711 ALALW 64K*** and ***G.729(a) 8K CS-ACELP*** from the **Unused** box and move these codec selections to the **Selected** box. Use the up and down arrows to order the selected codecs as shown below to match the codec preference order used by SoTel Systems IP Trunking.
- Check the **Re-invite Supported** box.
- Set **Fax Transport Support** to ***T38***.
- Set the **DTMF Support** field to ***RFC2833***. This directs Avaya IP Office to send DTMF tones using RTP event messages as defined in RFC2833.



Select the **T38 Fax** tab to set the Fax over Internet Protocol parameters of the SIP line. Set the parameters as shown below.

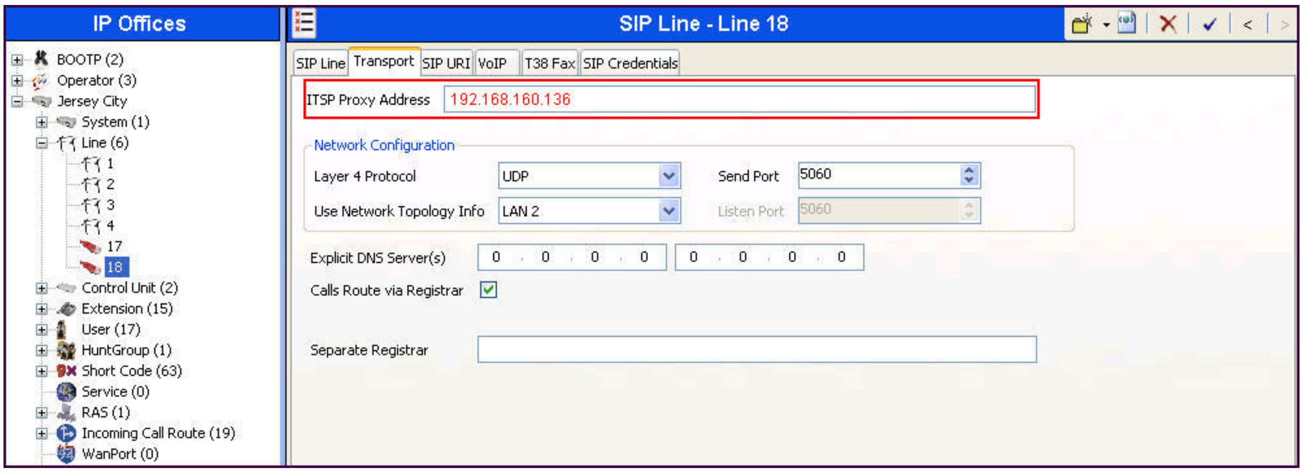
- Uncheck **Use Default Values** at the bottom of the screen.
- Set **T38 Fax Version** to ***0***. SoTel Systems SIP Trunking supports T.38 fax version 0.
- Set **Max Bit Rate (bps)** to 14400, the highest fax bit rate that Avaya IP Office supports for T.38 faxing.
- Check the **Disable T30 ECM** option.
- Default values may be used for all other parameters.



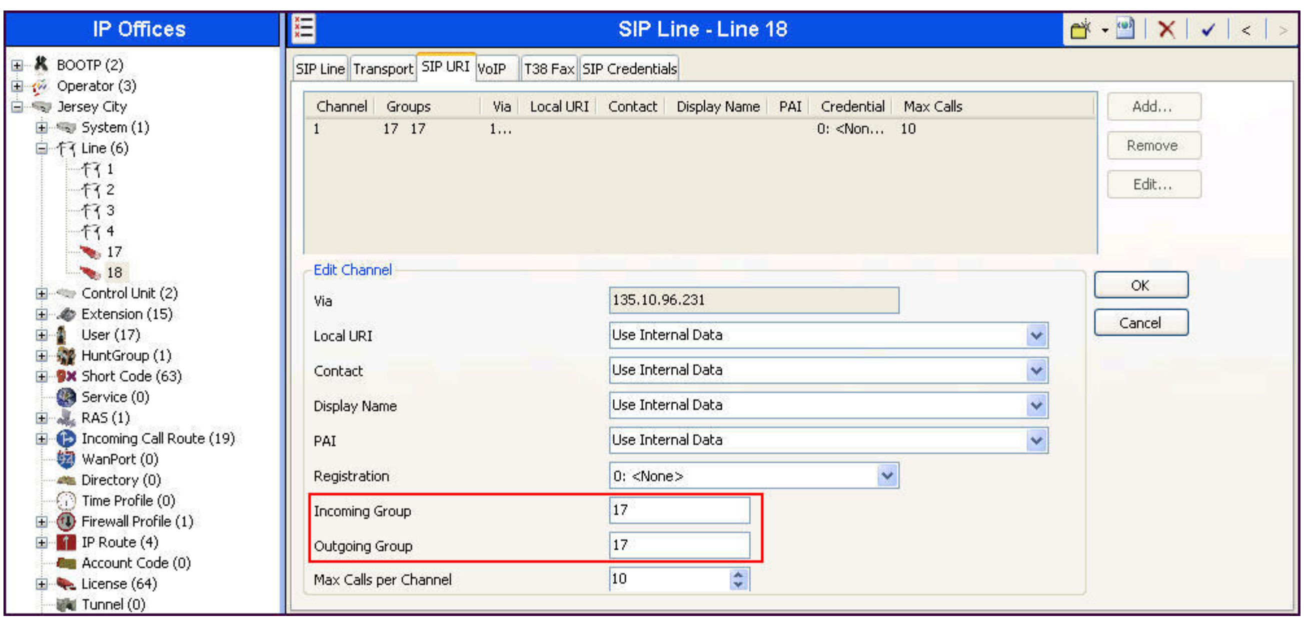
5.6. Administer SIP Line (Second)

As noted in **Section 3**, SoTel Systems SIP Trunking uses two border elements for connecting to the business sites. A second SIP Line is required to be configured containing the second border element's IP address. Other than the new line number and the **ITSP Proxy Address** in the 2nd SIP Line's **Transport** tab, all other administration for the SIP Line is identical to the previously administered SIP Line in **Section 5.5**.

Shown below is the **Transport** tab for the 2nd SIP Line (line number 18). Note the **ITSP Proxy Address** setting different than the setting for the first SIP Line (line number 17) in **Section 5.5**.



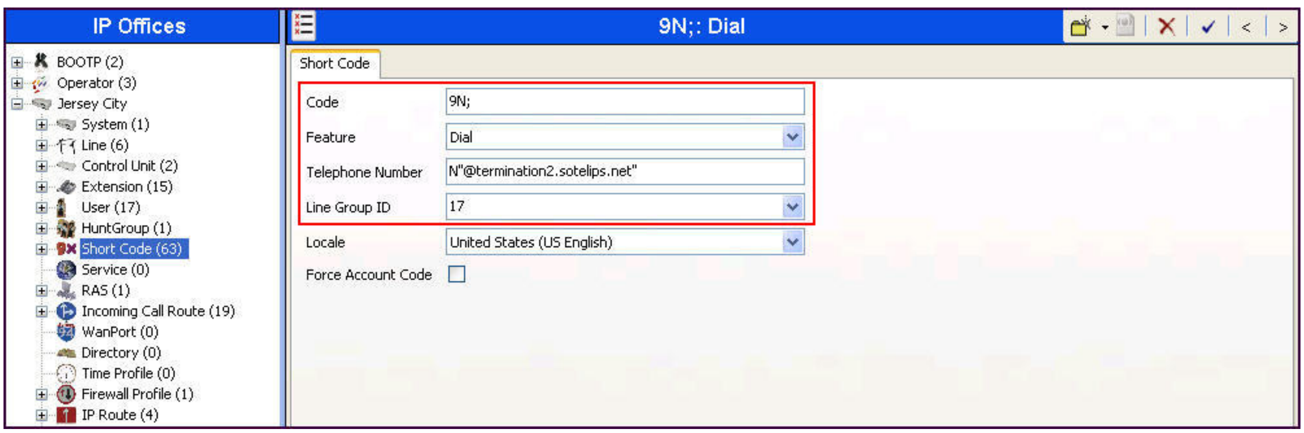
Shown below is the **SIP URI** configuration for the 2nd SIP Line. Note the settings for **Incoming Group** and **Outgoing Group** identical to those for the 1st SIP Line in **Section 5.5**.



5.7. 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:

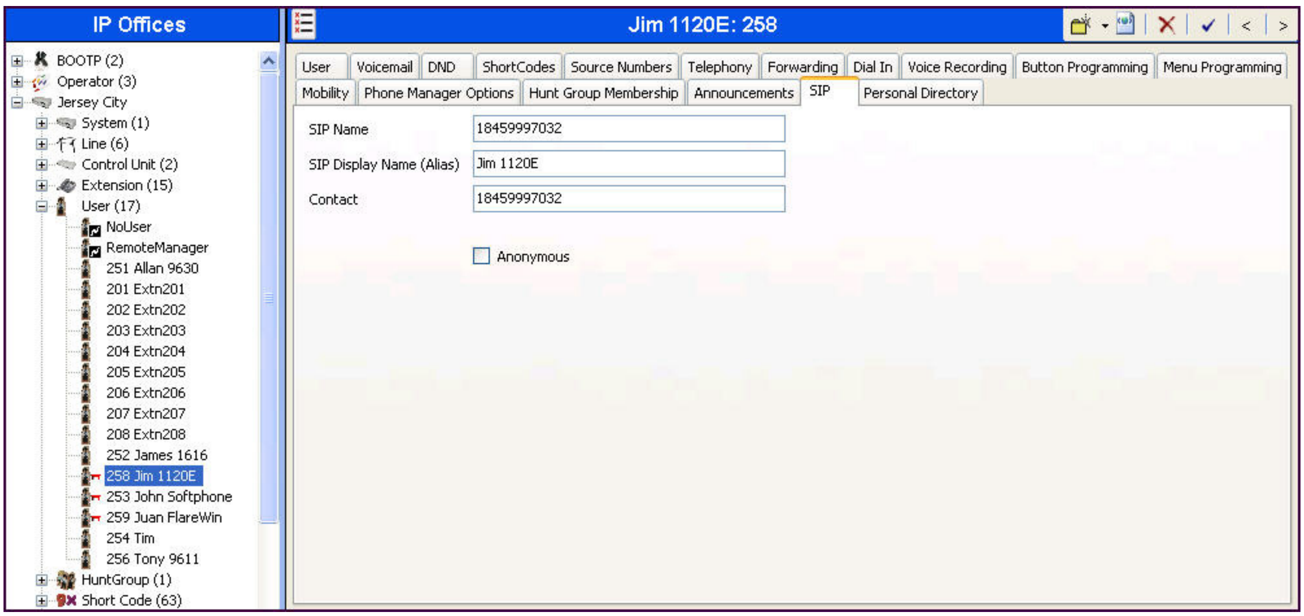
- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. The ***9N;*** short code, used for the compliance test, will be invoked when the user dials 9 followed by any number.
- Set **Feature** to ***Dial***. This is the action that the short code will perform.
- Set **Telephone Number** to ***N"@termination2.sotelips.net"***. 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 domain for the business site's IP-authenticated SIP account (provided by SoTel Systems) follows the @ sign in the above expression.
- Set the **Line Group Id** to the **Outgoing Group** number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.5**. This short code will use this line group when placing the outbound call.



5.8. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP Lines defined in **Section 5.5** and **Section 5.6**. 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 ***Jim 1120E***. Select the **SIP** tab in the Details Pane. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise by SoTel Systems. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. 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 calls.

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.

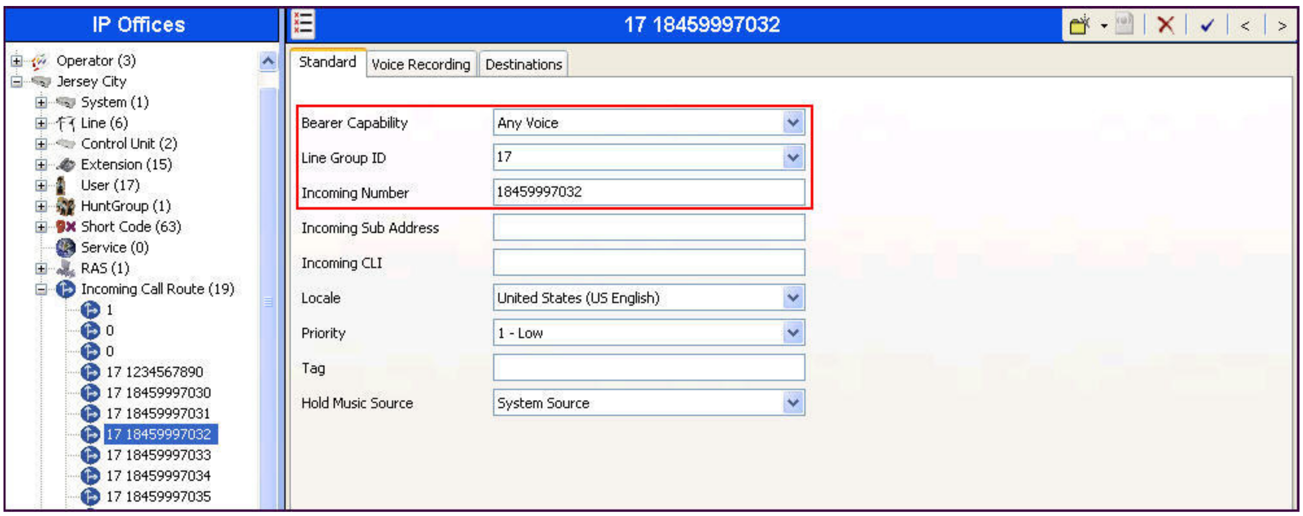


5.9. Incoming Call Route

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, right-click **Incoming Call Routes** in the Navigation Pane and select **New** (not shown). 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 Group** of the SIP Line defined in **Section 5.5**.
- Set the **Incoming Number** to the incoming DID number on which this route should match. Matching is right to left.

Default values can be used for all other fields.



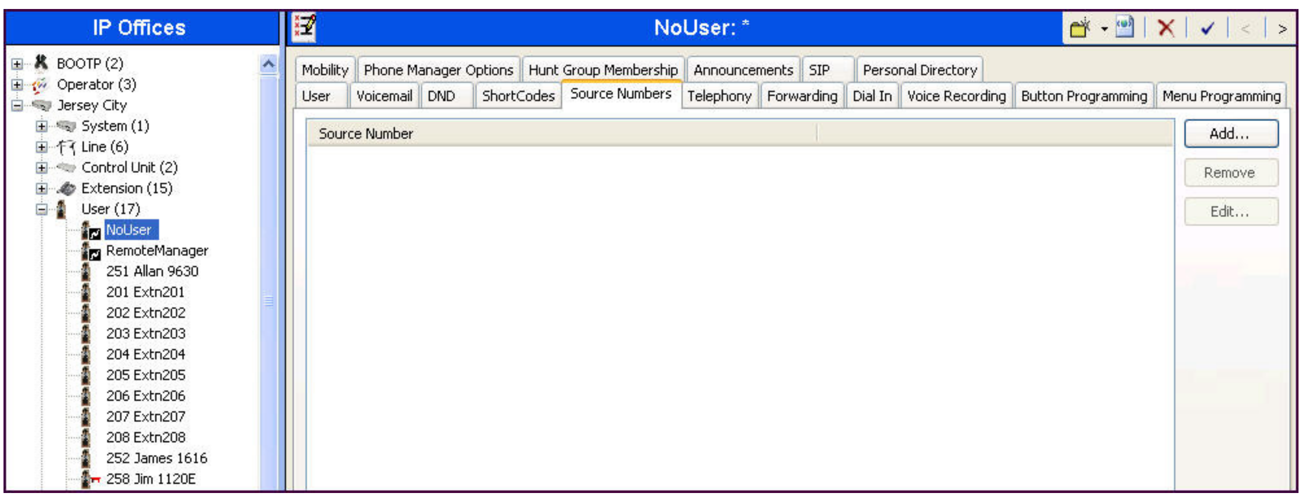
On the **Destinations** tab, select the destination from the pull-down list of the **Destination** field. In this example, incoming calls to 18459997032 on Incoming Group 17 (but the call can be received on either SIP Line 17 or SIP Line 18 since both lines have the **Incoming Group** set to 17 – see **Sections 5.5** and **5.6**) are to be routed to the user "Jim 1120E" at extension 258.

5.10. SIP Options

Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. By default, Avaya IP Office Release 8.1 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** 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:

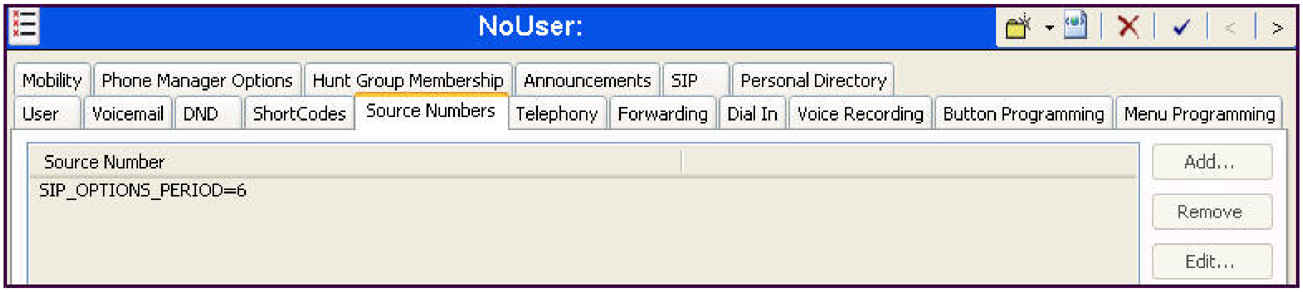
- To use the default value, set Binding Refresh = 0 or 300. OPTIONS will be sent at the 300 second frequency.
- To establish a period of less than 300 seconds, do not define the **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 300 seconds. The OPTIONS message 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**.

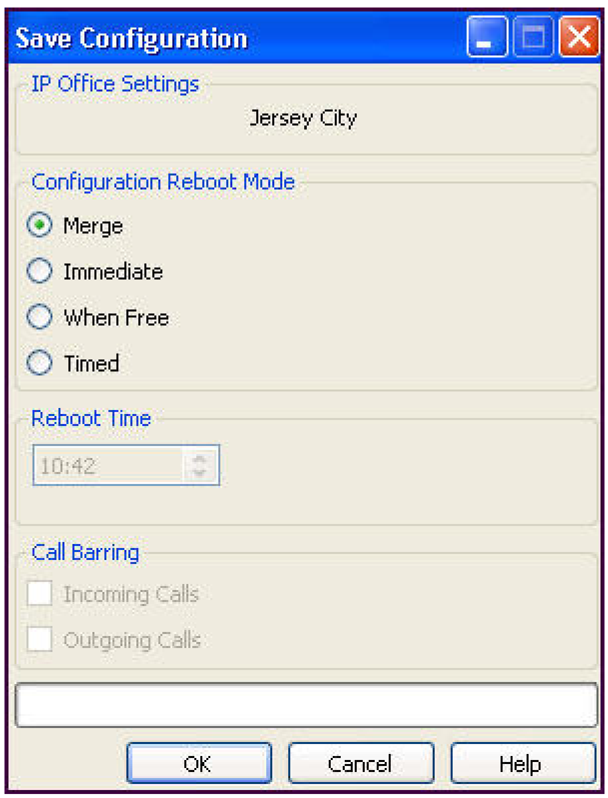
The **SIP_OPTIONS_PERIOD** parameter will appear in the list of Source Numbers as shown below. For the compliance test, an OPTIONS period of 60 seconds was desired. The **Binding Refresh Time** was set to ***60*** seconds in **Section 5.2**. There was no need to define **SIP_OPTIONS_PERIOD**.



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 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. SoTel Systems SIP Trunking Configuration

SoTel Systems is responsible for the configuration of its SIP Trunking service. The customer will need to provide the IP address used to reach the Avaya IP Office at the business site. SoTel Systems will provide the customer the necessary information to configure the Avaya IP Office SIP connection to SoTel Systems SIP Trunking including:

- Network edge IP addresses of the SoTel Systems SIP Trunking service.
- Transport and port for the SoTel Systems SIP Trunking connection to the Avaya IP Office at the business site.
- DID numbers to assign to users at the business site.
- Supported codecs and their preference order.

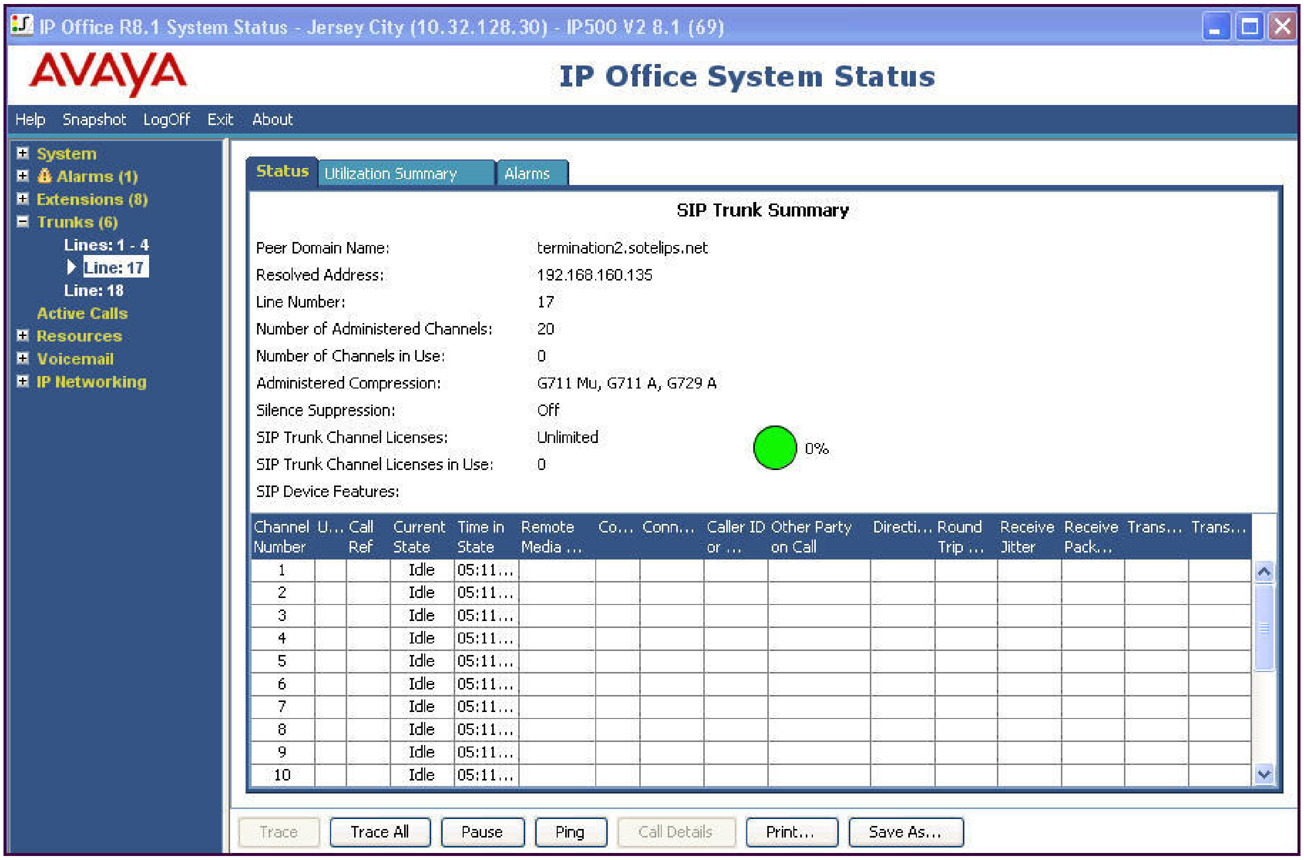
SoTel Systems provides connection information at https://sotelips.net/support/index.php?/Knowledgebase/Article/View/17/0/connecting-to-sotel-ip-services---sip-trunking

SoTel Systems provides SIP Interoperability information at https://sotelips.net/support/index.php?/Knowledgebase/Article/View/38/0/sip-interoperability

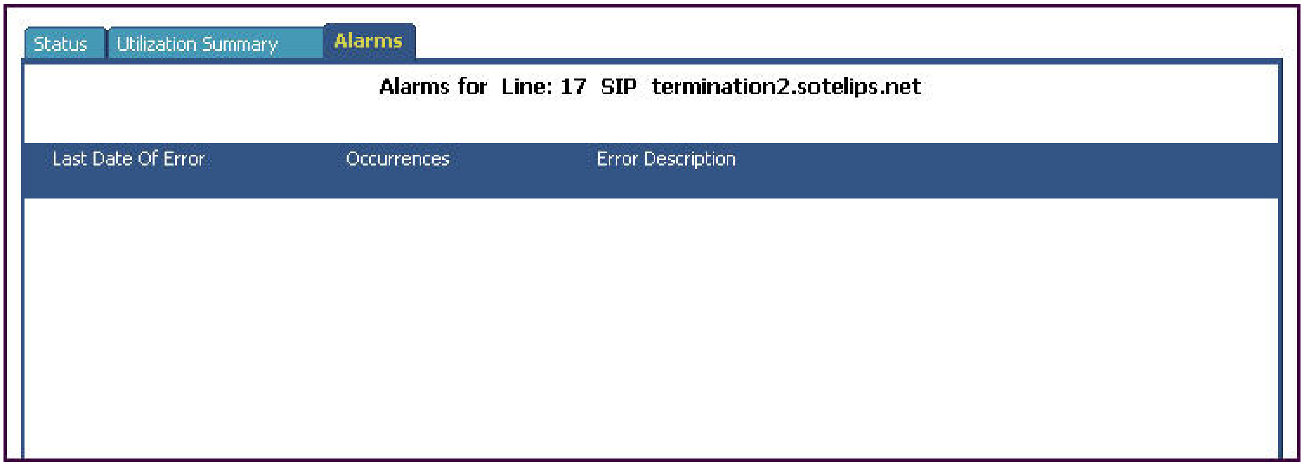
7. Verification Steps

The following steps may be used to verify the configuration:

- Use the Avaya IP Office System Status application to verify the SIP connection state. Launch the application from **Start → Programs → IP Office → System Status** on the Avaya IP Office Manager PC. 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 if no active call is currently in session (as shown below); or some channels are taken by active calls with **Current Sate** shown as ***Connected*** (not shown).



- 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 the Avaya IP Office with two-way audio.

8. Conclusion

The SoTel Systems SIP Trunking service passed compliance testing. These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office R8.1 and SoTel Systems SIP Trunking as shown in **Figure 1**. Test results and observations are noted in **Section 2.2**.

9. Additional References

[1] *IP Office Release 8.1 FP1 Product Description,* Documentation number 15-601041 Issue 26.N, April 2013.

[2] *Avaya IP Office 8.1 Installing IP500/IP500 V2,* Document number15-601042 Issue 27m, July 2013.

[3] *Avaya IP Office 8.1 Implementing Voicemail Pro,* Document number15-601064 Issue 8b, December 2012.

[4] *Avaya IP Office 8.1 FP1 Manager 10.1,* Document number15-601011 Issue 29u, April 2013.

[5] *Avaya IP Office 8.1 Using System Status Application,* Document number15-601758 Issue 07a, May 2013.

Product documentation for Avaya products may be found at http://support.avaya.com.
Product documentation for the SoTel Systems SIP Trunking is available from SoTel Systems. See **Section 2.3** on how to contact SoTel Systems.

Appendix: SIP Line Template

From Release 8.0 on, Avaya IP Office supports SIP Line Template (in xml format) that can be created from an existing configuration and imported into a new installation to simplify configuration procedures as well as to reduce potential configuration errors.

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 5.5** in these Application Notes as a reference.

The SIP Line Template created for **SIP Line 17** from the configuration as documented in these Application Notes is as follows:


```
<?xml version="1.0" encoding="utf-8"?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20130910</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>SoTel SIP Trunking</DescriptiveName>
  <ITSPDomainName>termination2.sotelips.net</ITSPDomainName>
  <SendCallerID>CallerIDNone</SendCallerID>
  <ReferSupport>false</ReferSupport>
  <ReferSupportIncoming>1</ReferSupportIncoming>
  <ReferSupportOutgoing>2</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>false</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>SystemDefault</LineNamePriority>
  <UpdateSupport>UpdateAuto</UpdateSupport>
  <UserAgentServerHeader />
  <CallerIDfromFromheader>false</CallerIDfromFromheader>
  <PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
  <ITSPProxy>192.168.160.135</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.711 ALAW 64K,G.729(a) 8K CS-
ACELP</AdvCodecPref>
  <CallInitiationTimeout>4</CallInitiationTimeout>
  <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
  <VoipSilenceSupression>false</VoipSilenceSupression>
  <ReinviteSupported>true</ReinviteSupported>
  <FaxTransportSupport>FOIP_T38</FaxTransportSupport>
  <UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
```

```
  <CodecLockdown>false</CodecLockdown>
  <Rel100Supported>false</Rel100Supported>
  <T38FaxVersion>0</T38FaxVersion>
  <Transport>UDPTL</Transport>
  <LowSpeed>0</LowSpeed>
  <HighSpeed>0</HighSpeed>
  <TCFMethod>Trans_TCF</TCFMethod>
  <MaxBitRate>FaxRate_14400</MaxBitRate>
  <EflagStartTimer>2600</EflagStartTimer>
  <EflagStopTimer>2300</EflagStopTimer>
  <UseDefaultValues>false</UseDefaultValues>
  <ScanLineFixup>true</ScanLineFixup>
  <TFOPEnhancement>true</TFOPEnhancement>
  <DisableT30ECM>true</DisableT30ECM>
  <DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
  <DisableT30MRCompression>false</DisableT30MRCompression>
  <NSFOverride>false</NSFOverride>
</Template>
```


The SIP Line Template created for **SIP Line 18** from the configuration as documented in these Application Notes is as follows:

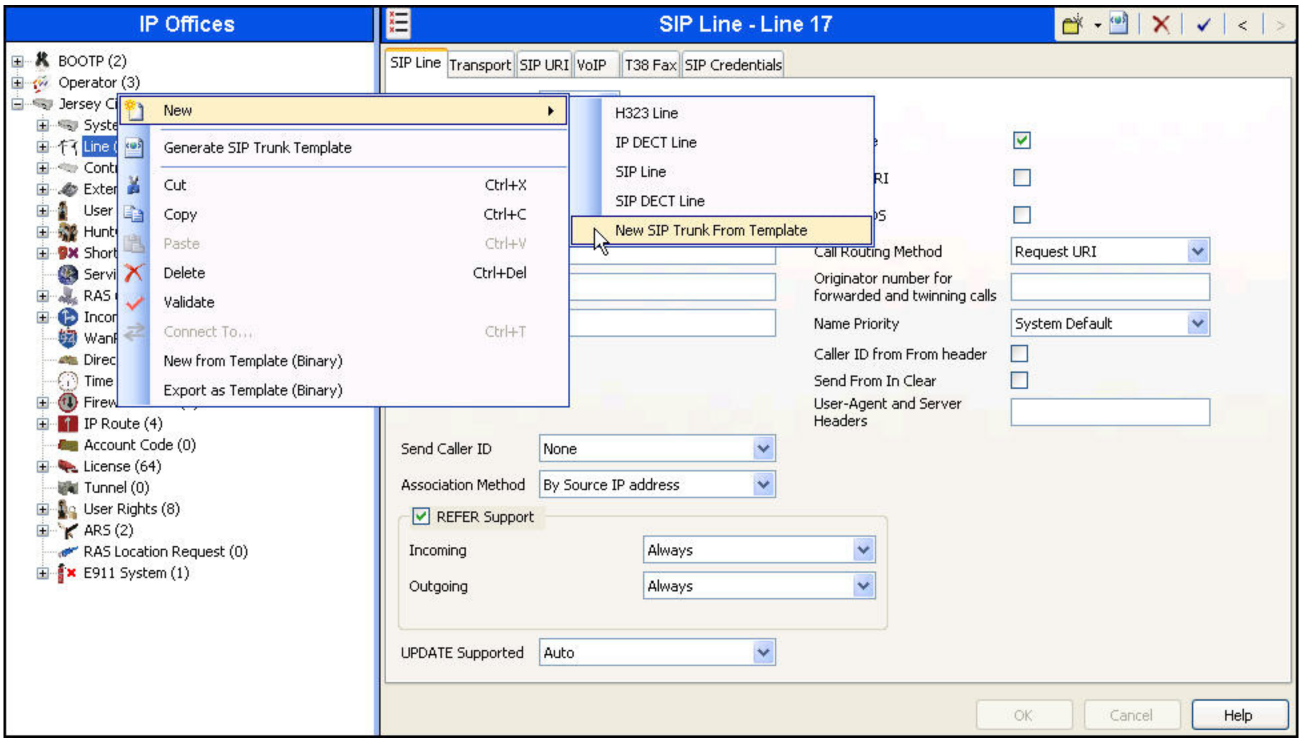

```
<?xml version="1.0" encoding="utf-8"?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20130910</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>SoTel SIP Trunking</DescriptiveName>
  <ITSPDomainName>termination2.sotelips.net</ITSPDomainName>
  <SendCallerID>CallerIDNone</SendCallerID>
  <ReferSupport>false</ReferSupport>
  <ReferSupportIncoming>2</ReferSupportIncoming>
  <ReferSupportOutgoing>2</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>false</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>SystemDefault</LineNamePriority>
  <UpdateSupport>UpdateAuto</UpdateSupport>
  <UserAgentServerHeader />
  <CallerIDfromFromheader>false</CallerIDfromFromheader>
  <PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
  <ITSPProxy>192.168.160.136</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.711 ALAW 64K,G.729(a) 8K CS-
ACELP</AdvCodecPref>
```

```
  <CallInitiationTimeout>4</CallInitiationTimeout>
  <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
  <VoipSilenceSupression>false</VoipSilenceSupression>
  <ReinviteSupported>true</ReinviteSupported>
  <FaxTransportSupport>FOIP_T38</FaxTransportSupport>
  <UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
  <CodecLockdown>false</CodecLockdown>
  <Rel100Supported>false</Rel100Supported>
  <T38FaxVersion>0</T38FaxVersion>
  <Transport>UDPTL</Transport>
  <LowSpeed>0</LowSpeed>
  <HighSpeed>0</HighSpeed>
  <TCFMethod>Trans_TCF</TCFMethod>
  <MaxBitRate>FaxRate_14400</MaxBitRate>
  <EflagStartTimer>2600</EflagStartTimer>
  <EflagStopTimer>2300</EflagStopTimer>
  <UseDefaultValues>false</UseDefaultValues>
  <ScanLineFixup>true</ScanLineFixup>
  <TFOPEnhancement>true</TFOPEnhancement>
  <DisableT30ECM>false</DisableT30ECM>
  <DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
  <DisableT30MRCompression>false</DisableT30MRCompression>
  <NSFOverride>false</NSFOverride>
</Template>
```

To import a SIP Line template into a new installation:

1. On the PC where IP Office Manager is installed, copy and paste one of the above templates into a text document named **US_SoTel Systems_SIPTrunk.xml**. Move the .xml file to the IP Office Manager template directory (default location is C:\Program Files\Avaya\IP Office\Manager\Templates).

2. Import the template into an IP Office installation 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**:



3. In the resulting **Template Type Selection** screen, verify that ***United States*** is automatically populated for **Country** and ***SoTel Systems*** is automatically populated for **Service Provider** as shown below. Click **Create new SIP Trunk** to finish the importing 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.