Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Frontier Communications SIP Trunking (Metaswitch) with Avaya IP Office R8.1- Issue 1.0

Abstract

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

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

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

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between Frontier Communications 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, Avaya H.323 and SIP hard phones, and SIP-based Avaya softphones (IP Office Softphone and Flare® Experience for Windows).

The Frontier Communications SIP Trunking service (Metaswitch) provides PSTN access via a SIP trunk between the business site and the Frontier Communications 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

The general test approach was to configure a simulated enterprise site using Avaya IP Office to connect to the Frontier Communications SIP Trunking service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality tests listed in **Section 2.1**.

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

2.1. Interoperability Compliance Testing

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

- SIP OPTIONS queries and responses.
- Incoming PSTN calls to H.323 and SIP telephones at the enterprise 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 enterprise site. All outbound PSTN calls were routed from the enterprise site across the SIP trunk to the service provider
- Various call types including: local, long distance, outbound toll-free, international, operator (0), 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 REFER message 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 busy or error conditions.
- Inbound 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.
- Frontier Communications SIP Trunking does not support Operator-Assisted call (0 + 10 digits).
- Frontier Communications SIP Trunking does not support T.38 for outbound fax calls.

2.2. Test Results

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

- **OPTIONS Response –** Frontier responded to OPTIONS from the enterprise site with "403 From: URI not recognized" instead of "200 OK". This is because Avaya IP Office specifies no user part of the URI in its OPTIONS request and headers (e.g., Request-Line: OPTIONS sip:135.10.96.231 SIP/2.0). This is a fixed configuration that cannot be changed on Avaya IP Office. Avaya IP Office treated the 403 response as a legitimate response from the far end for verifying active state of the SIP trunk connection.
- **OPTIONS During Prolonged Inbound Call Ringing –** After receiving "180 Ringing" from Avaya IP Office on inbound INVITE, Frontier ceased to send OPTIONS to the enterprise site for long duration of ringing. The OPTIONS would resume after the call was answered. Frontier opened a trouble ticket on this problem to its soft switch vendor during the prior test with Avaya IP Office Release 8.0.
- **Codec Lockdown on Outbound Calls –** When Avaya IP Office was configured with G.711MU and G.729A codecs in that preference order, Frontier responded to the outbound INVITE request with only G.711MU in the SDP as expected. However, when Avaya IP Office was configured with the same 2 codecs but with G.729A as the preferred codec, Frontier responded to the outbound INVITE request with both codecs in the SDP (with G.729A listed first) instead of selecting one from the INVITE SDP list. This behavior had no user impact. Calls were successful.
- **Call to Invalid PSTN Number –** Frontier returned "500 Internal Server Error" to the outbound call INVITE to an invalid PSTN number. A more appropriate status message like "404 Not Found" would be more appropriate.

- **REFER –** When REFER was used for off-net call transfer or forward, at the end of the off-net call re-direction when both legs of the call at Avaya IP Office had already been terminated, Frontier sent an INVITE to Avaya IP Office which responded with "481 Dialog/Transaction Does Not Exist". This INVITE and response sequence did not happen all the time, and when it happened, the sequence would repeat itself for about 30 to 45 seconds then went away. User experience was not negatively affected (i.e., the call was forwarded or transferred successfully). This same problem was experienced during the prior test with Avaya IP Office Release 8.0.

2.3. Support

For technical support on Frontier SIP Trunking, contact Frontier

- Use the Technical Support link for business customers at http://www.frontier.com, or
- Call the business customer support number at 877-462-8188 (for former Verizon customers) or 800-921-8102 (for other Frontier customers).

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 Frontier Communications SIP Trunking.

Located at the enterprise site is an Avaya IP Office 500V2 and Avaya endpoints. 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.

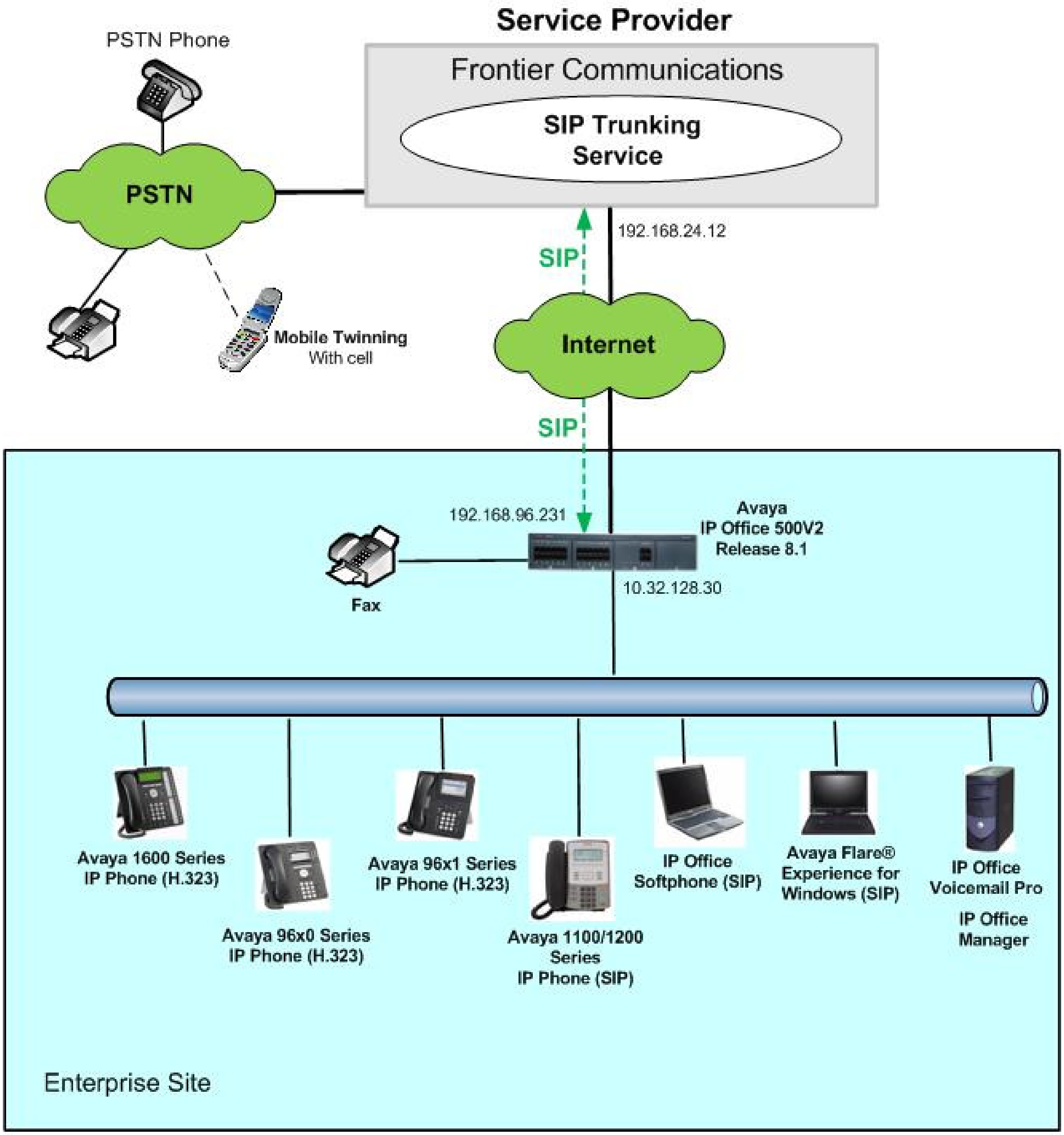


Figure 1: Test Configuration

For security purposes, any actual public IP addresses used in the compliance test were changed to 192.168.x.x throughout these Application Notes where the 3rd and 4th octets were retained from the real addresses.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to send digits across the SIP trunk to Frontier Communications. 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. Frontier Communications sent 10 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.

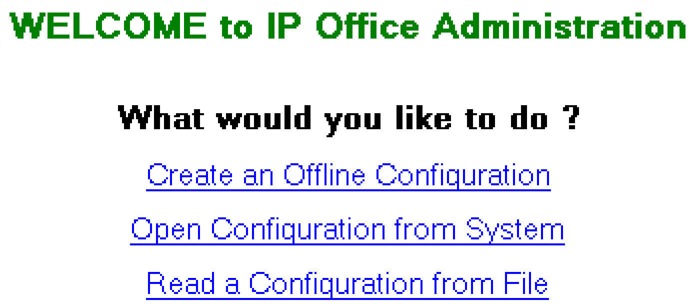
4. Equipment and Software Validated

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
Venta Fax PC Application	6.6.156.385
Frontier Communications Components	
Equipment / Software	**Release / Version**
ACME Net-Net 4000	6.2
MSW	8.1

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.

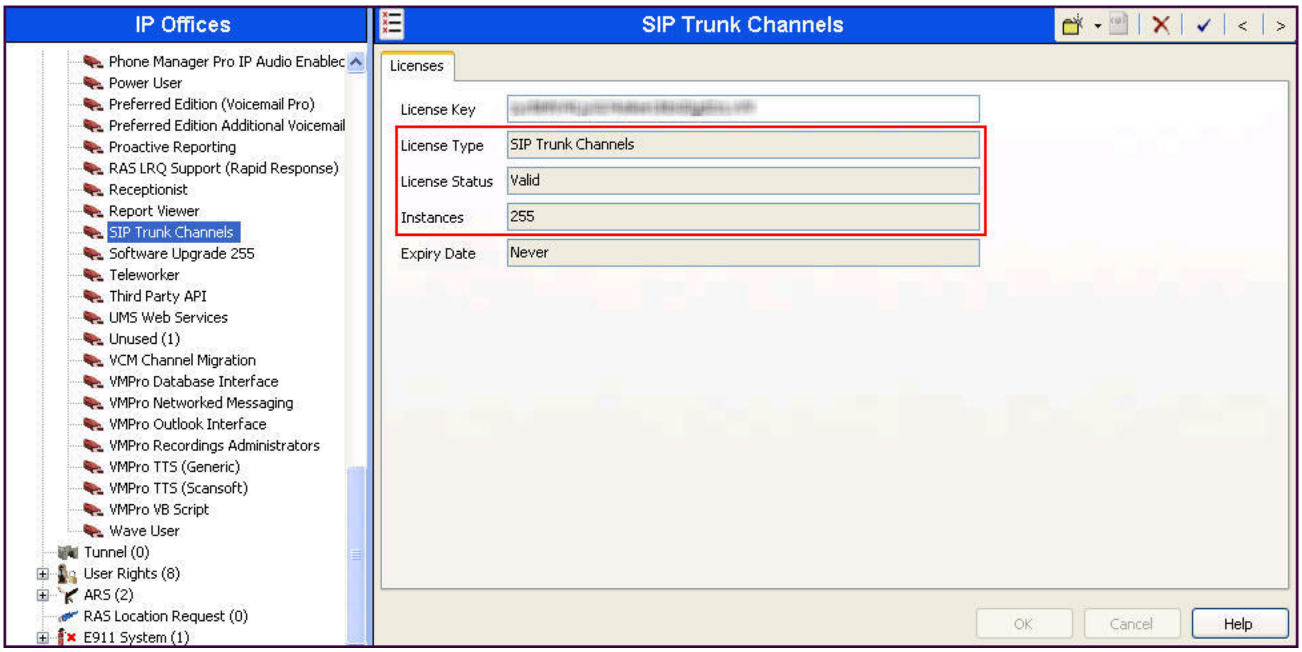
The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this document, the **View** menu was configured to show the Navigation pane on the left side, omit the Group pane in the center, and show the Details pane on the right side. Since the Group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

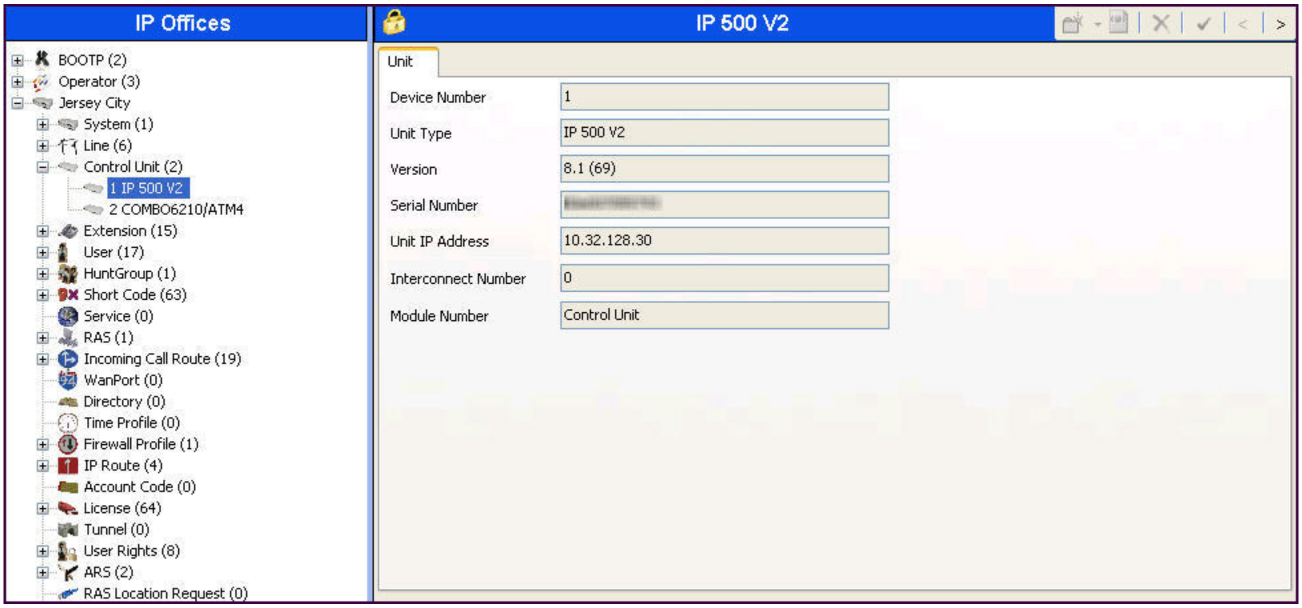
The configuration and features described in these Application Notes require Avaya IP Office to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

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**.

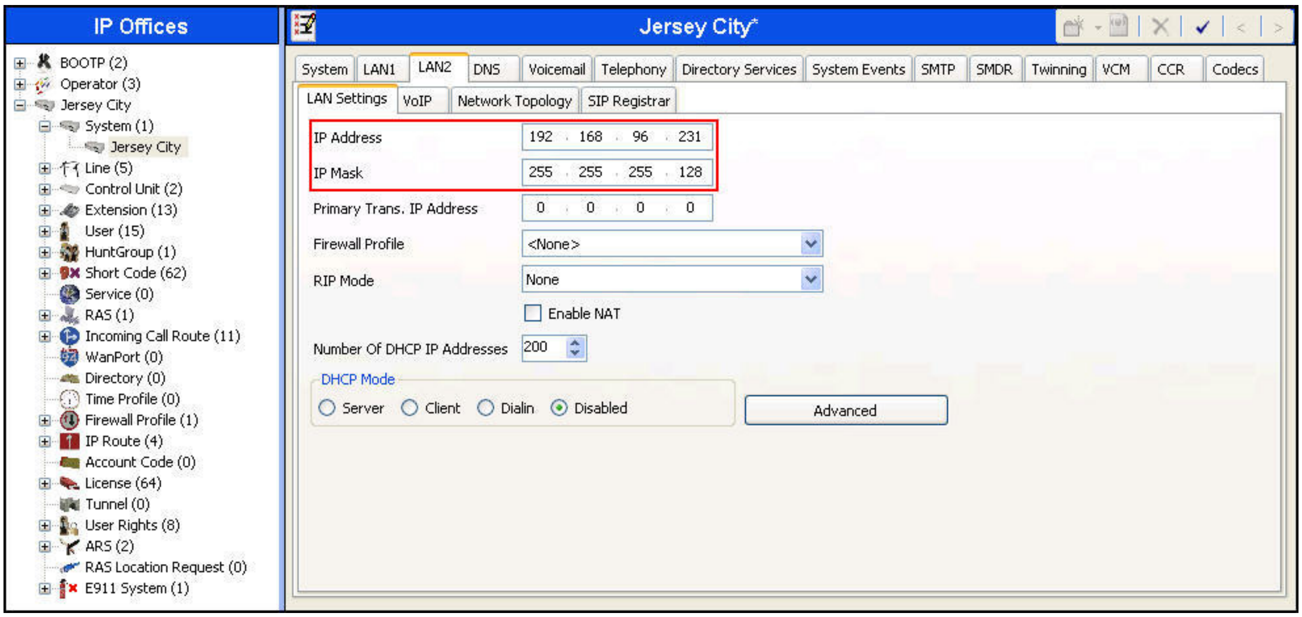


5.2. System

This section describes the steps necessary to configure system settings.

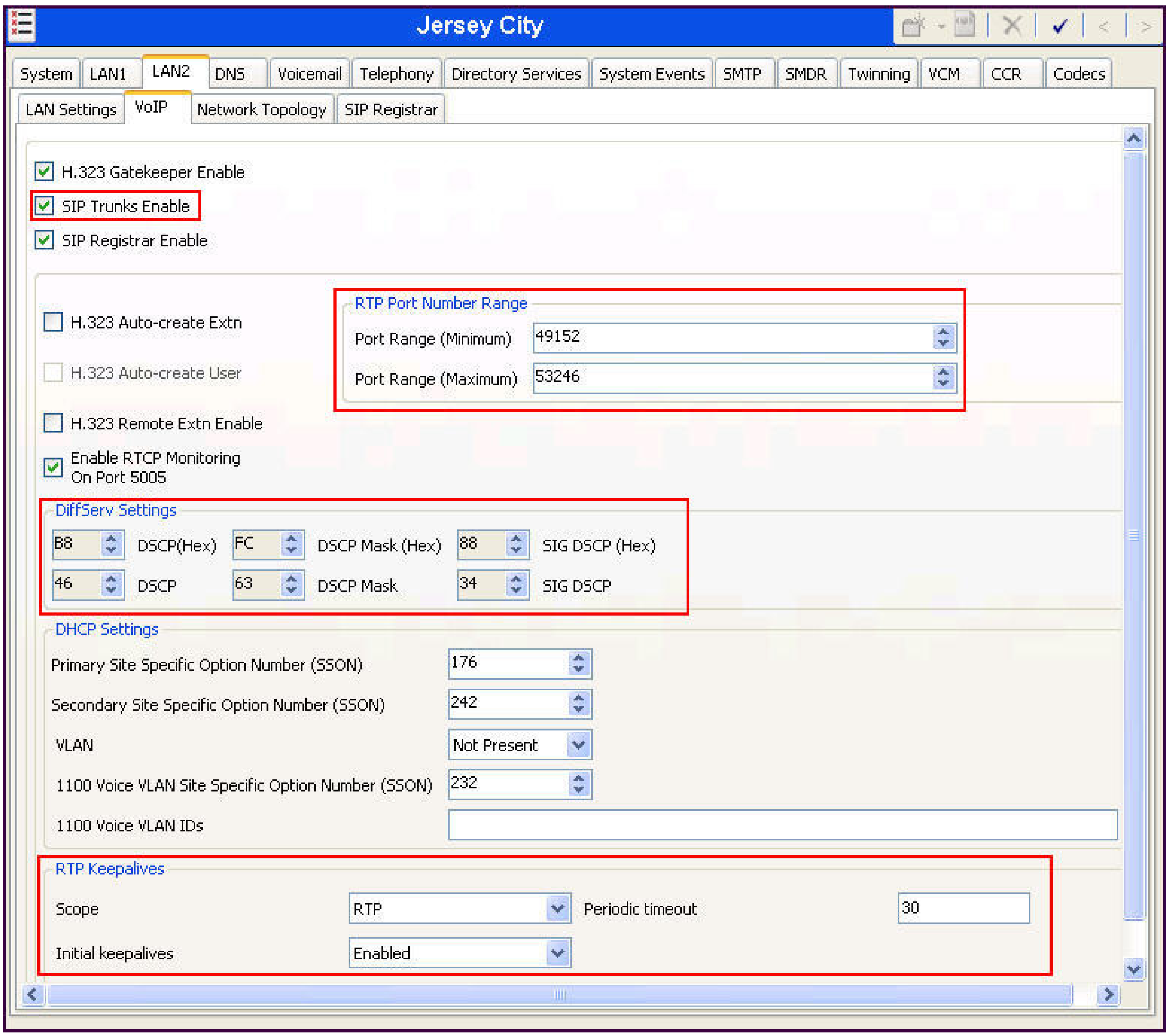
5.2.1. LAN2 Tab

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 **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.



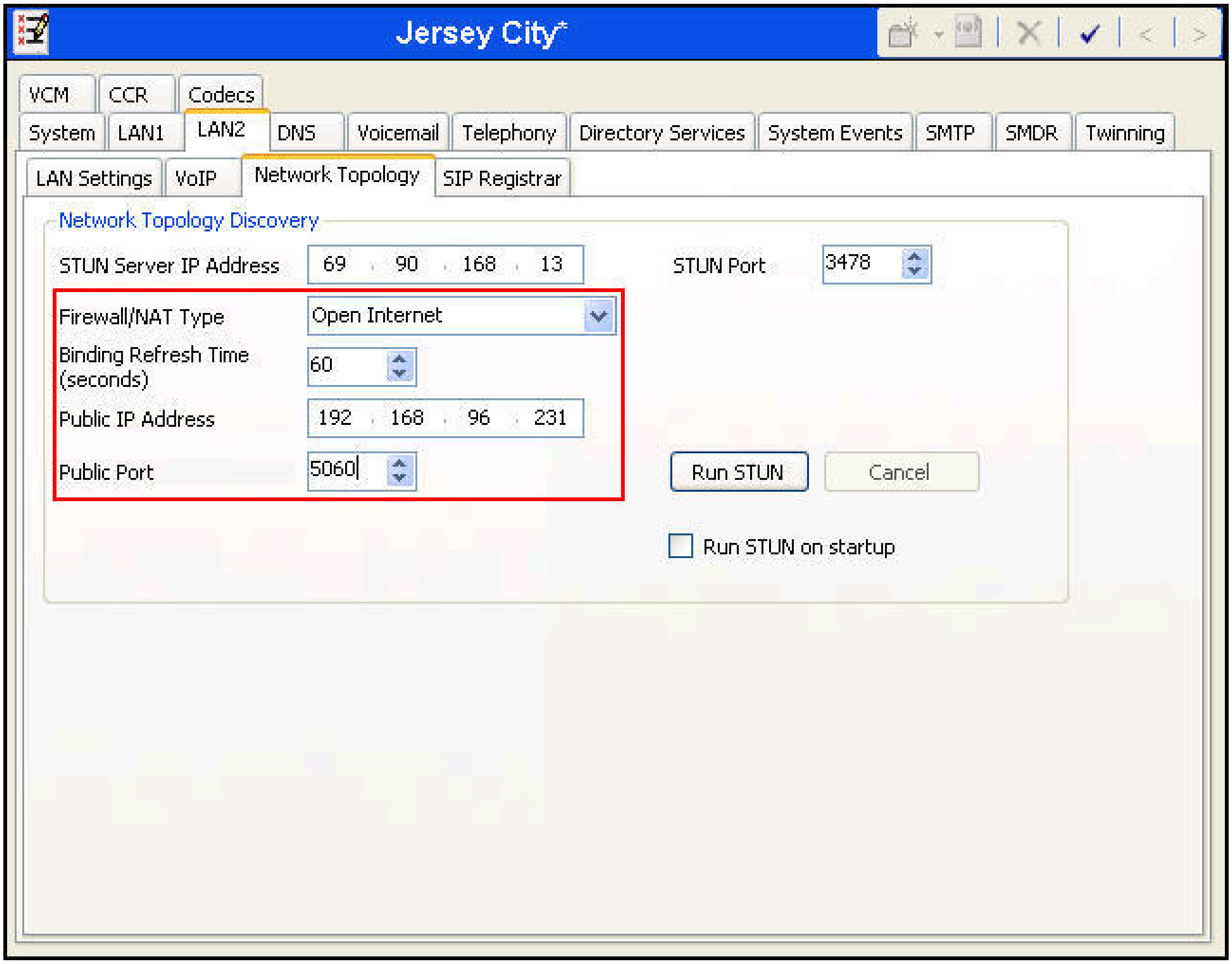
On the **VoIP** tab of LAN2 in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent 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 Avaya 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. Some service providers expect the IP Office endpoint to send RTP packets first even though there is no IP Office media endpoint involved in this call situation since the call has been re-directed back to PSTN.



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

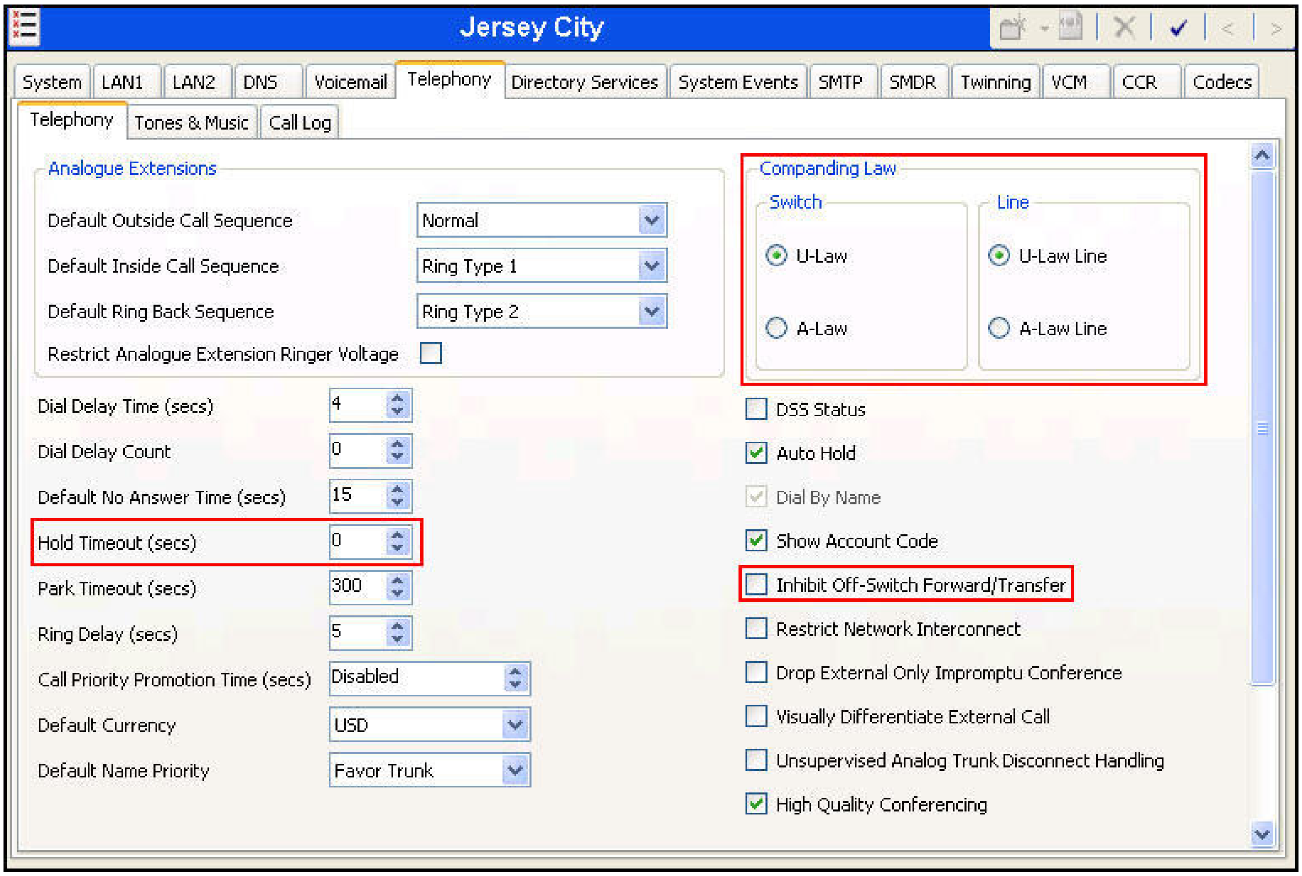
- Select ***Open Internet*** for **Firewall/NAT Type** from the pull-down menu. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**. With the ***Open Internet*** setting, **STUN Server IP Address** is not used.
- Set **Binding Refresh Time (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 5.8** 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 Frontier Communications, and therefore is not described in these Application Notes.

5.2.2. 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.



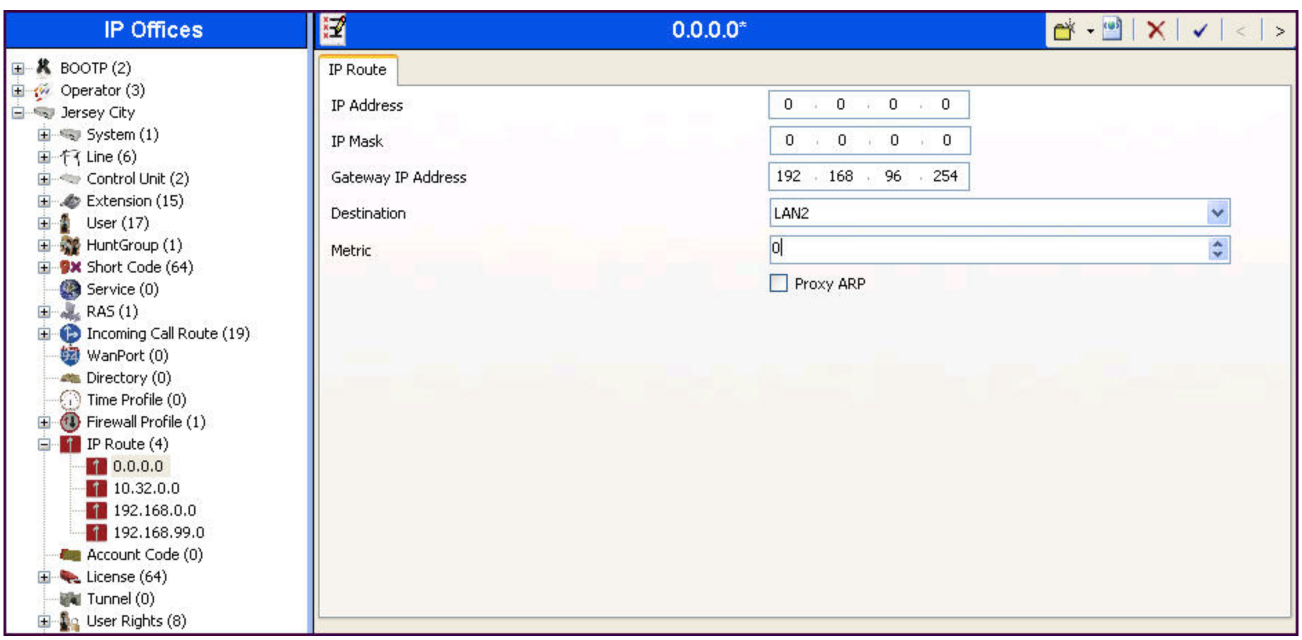
5.2.3. Twinning Calling Party Settings

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.

5.3. 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.** Create/verify a default route with the following parameters:

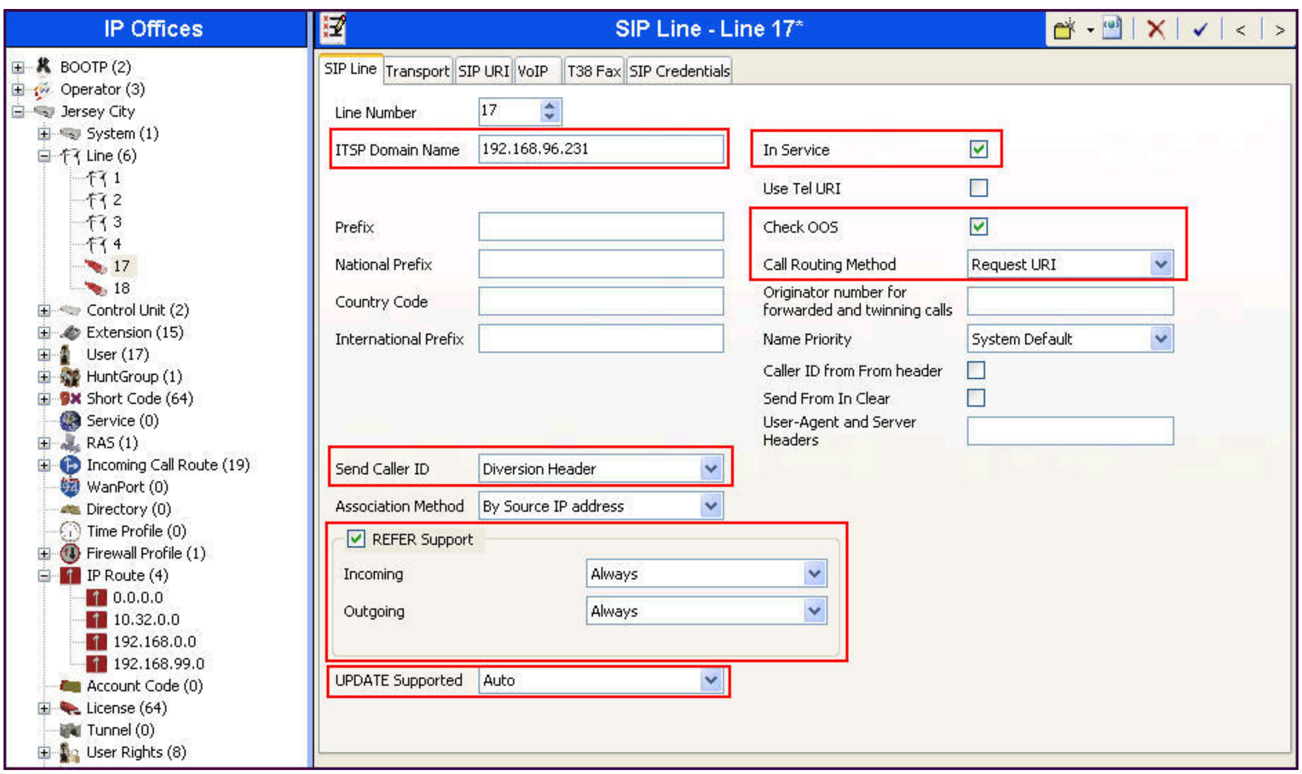
- Set **IP Address** and **IP Mask** to **0.0.0.0**.
- Set **Gateway IP Address** to the IP address of the gateway for the public internet WAN network.
- Set **Destination** to **LAN2** from the drop-down list.



5.4. Administer SIP Line

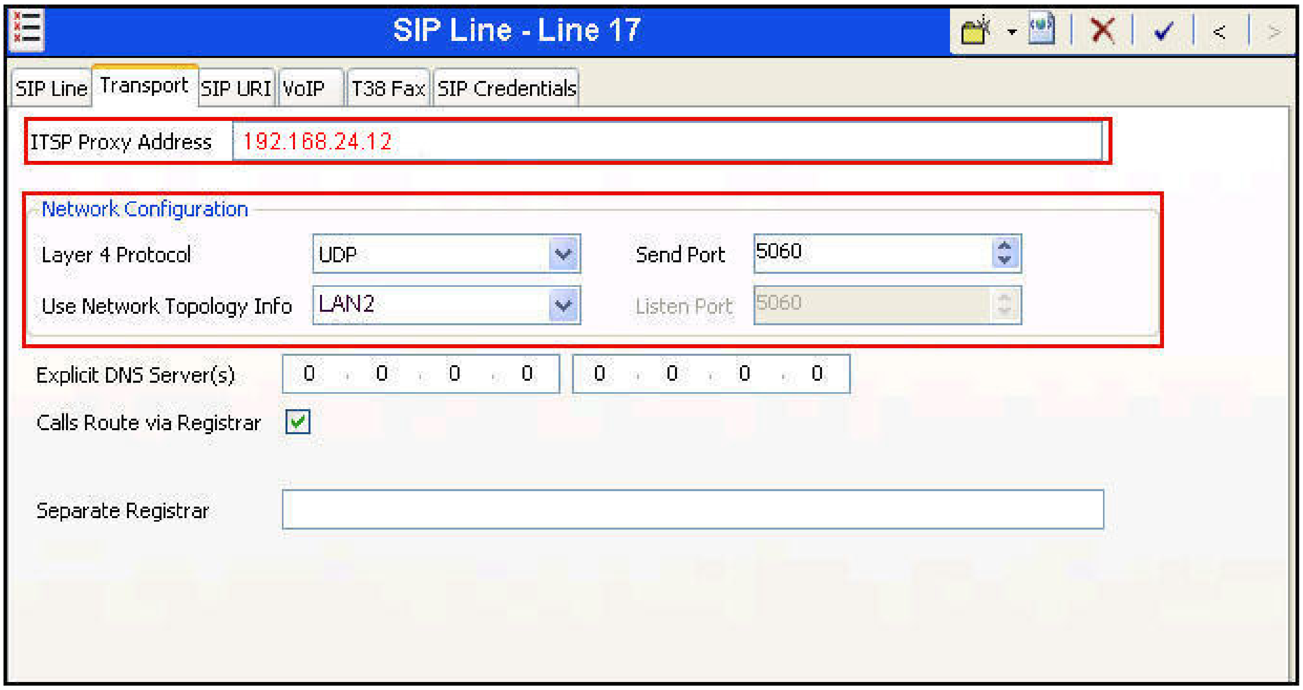
A SIP line is needed to establish the SIP connection between Avaya IP Office and Frontier Communications SIP Trunking. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New → SIP Line** (not shown). On the **SIP Line** tab in the Details Pane, configure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

- Set **ITSP Domain Name** to the LAN2 IP address (***192.168.96.231***) so that Avaya IP Office uses this IP as the URI host portion in SIP headers such as From and Diversion..
- Check the **In Service** box.
- Check **OOS** box. Avaya IP Office will use the SIP OPTIONS method to periodically check the SIP Line. The time between SIP OPTIONS sent by IP Office will use the **Binding Refresh Time** for LAN2, as shown in **Section 5.2.1**.
- 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 5.2.3**, Avaya IP Office will include the Diversion Header for calls that are directed via Mobile Twinning out the SIP Line to Frontier. It will also include Diversion Header for calls that are call forwarded out the SIP Line.
- Check **REFER Support**; then select ***Always*** from the drop-down menu for **Incoming** and **Outgoing**. Frontier Communications SIP Trunking supports use of REFER for off-net call forward and transfer.
- Set **UPDATE Supported** to ***Auto***. With this setting Avaya IP Office will send UPDATE messages for session refresh if the other party supports UPDATE.



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

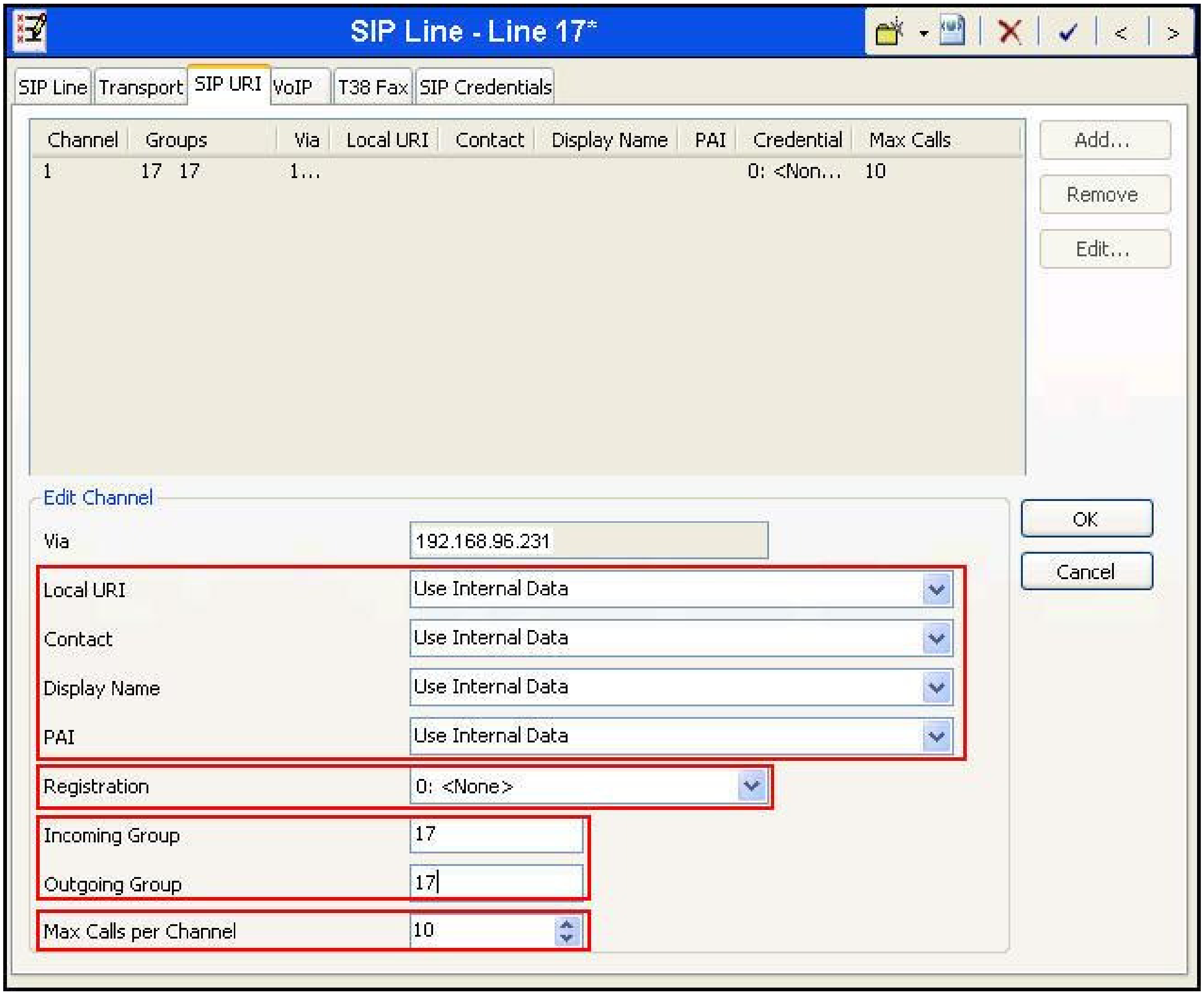
- Set the **ITSP Proxy Address** to the IP address of the service provider SIP Trunking access interface provided by Frontier Communications.
- Set the **Layer 4 Protocol** to ***UDP***.
- Set **Use Network Topology Info** to ***LAN2*** as configured in **Section 5.2.1**.
- Set the **Send Port** to ***5060***.



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.

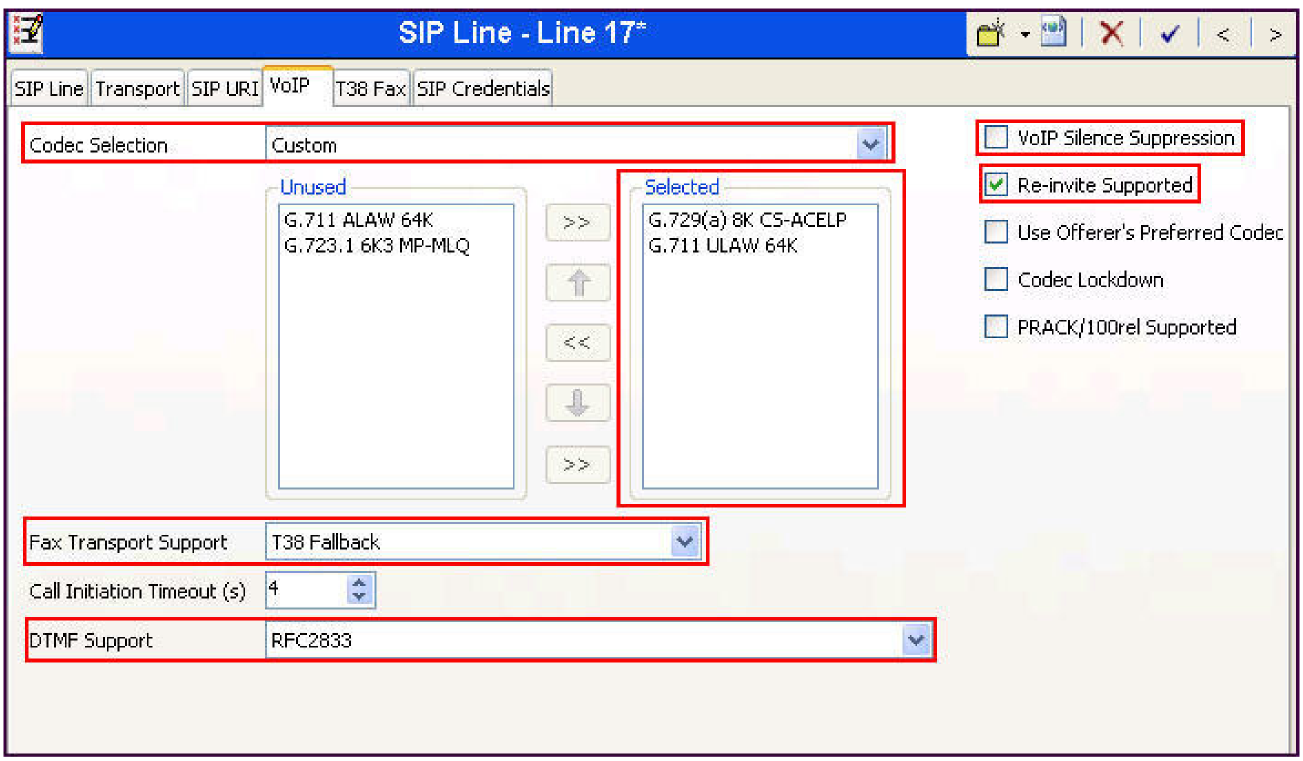
- Set **Local URI** to ***Use Internal Data***. This setting allows calls on this line whose SIP URI matches the **SIP Name** set on the **SIP** tab of any **User** as shown in **Section 5.8**.
- Set **Contact** and **Display Name** to ***Use Internal Data***. This setting will cause the Contact and Display Name data to be set from the corresponding fields on the **SIP** tab of the individual **User** as shown in **Section 5.68**.
- 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 5.68**.
- For **Registration**, select ***0:<None>*** from the pull-down menu. The test circuit used for the compliance test 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.



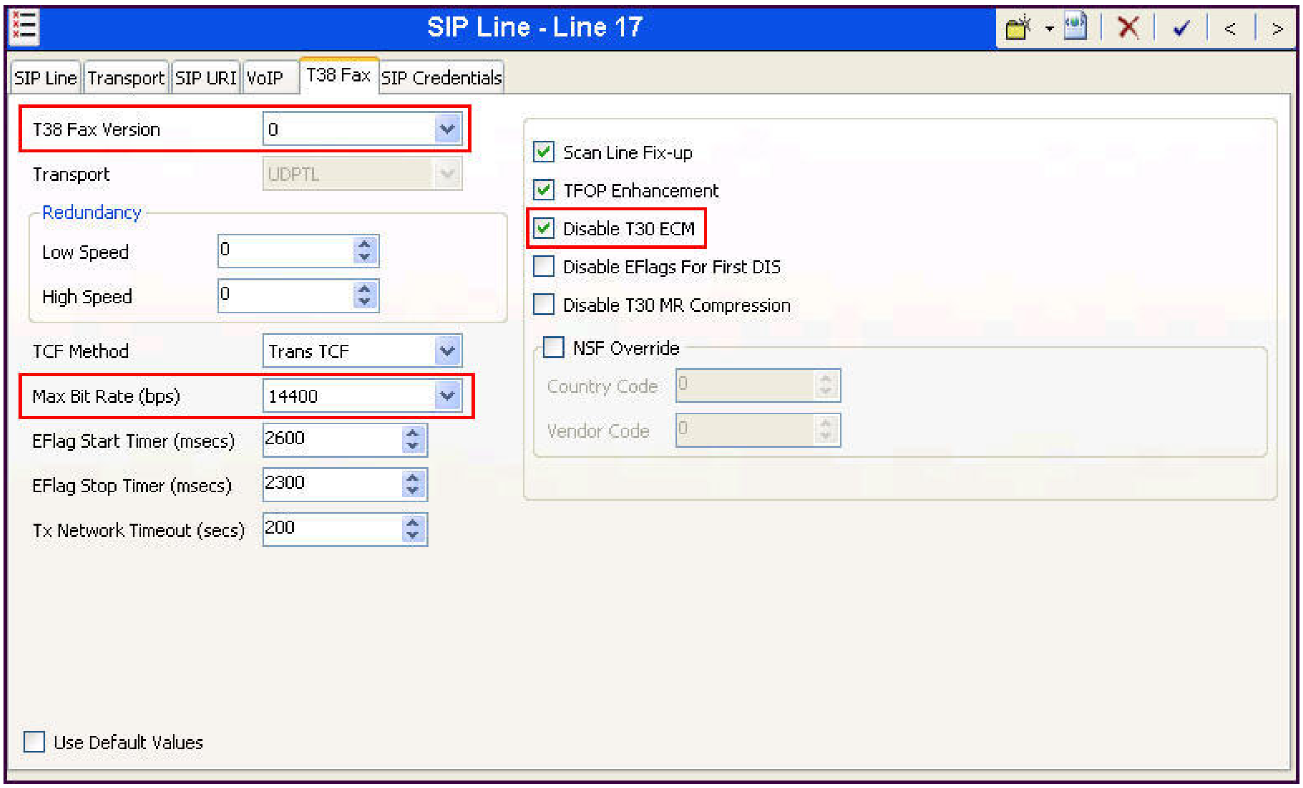
Select the **VoIP** tab, to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below.

- Set the **Codec Selection** to ***Custom***.
- Choose ***G.729(a) 8K CS-ACELP*** and ***G.711 ULAW 64K*** from the **Unused** box and move these 2 selections to the **Selected** box. Use the down/up arrows to order the 2 selected codecs as shown. These 2 codecs are supported by Frontier SIP Trunking.
- 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.
- Check the **Re-invite Supported** option box.
- Select ***T38 Fallback*** for **Fax Transport Support**. ***T38 Fallback*** was selected so that if T.38 is not supported by the remote end, Avaya IP Office will fall back to G.711 pass-through faxing. This configuration was necessary since outbound T.38 faxing was not supported by Frontier SIP Trunking. Note, however, that faxing using G.711 pass-through is generally not guaranteed to be successful.



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

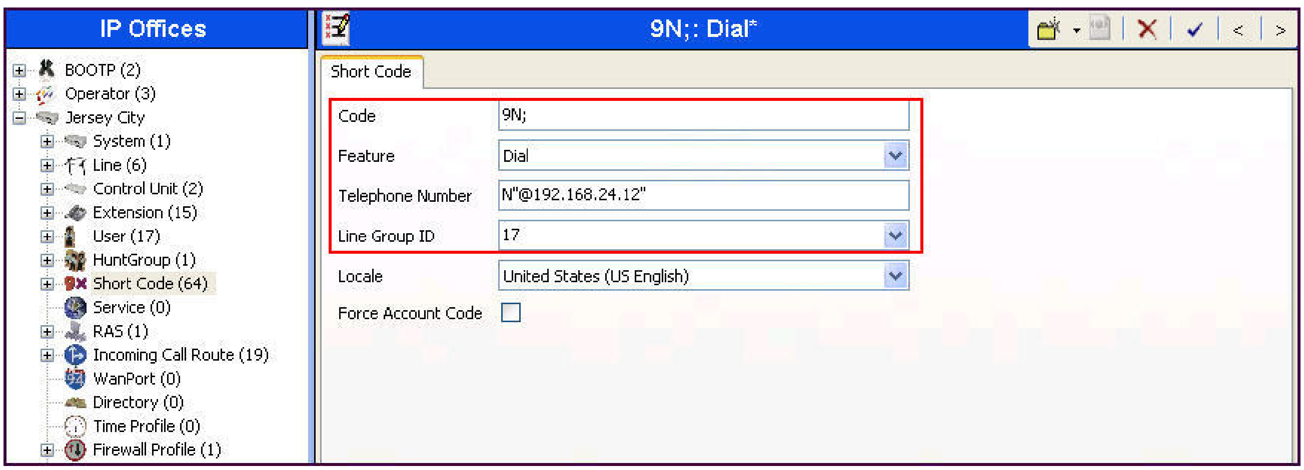
- Uncheck **Use Default Values** at the bottom of the screen.
- Set **T38 Fax Version** to ***0***. Frontier Communications SIP Trunking supports T.38 fax version 0.
- Set **Max Bit Rate (bps)** to ***14400***, the highest fax bit rate that Avaya IP Office supports for T.38 faxing.
- Check the **Disable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



5.5. 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"@192.168.24.12"***. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. The value ***N*** represents the number dialed by the user. The IP address following the @ sign is the IP address of the Frontier SIP proxy.
- Set the **Line Group Id** to the **Outgoing Group** number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.4**. This short code will use this line group when placing the outbound calls.

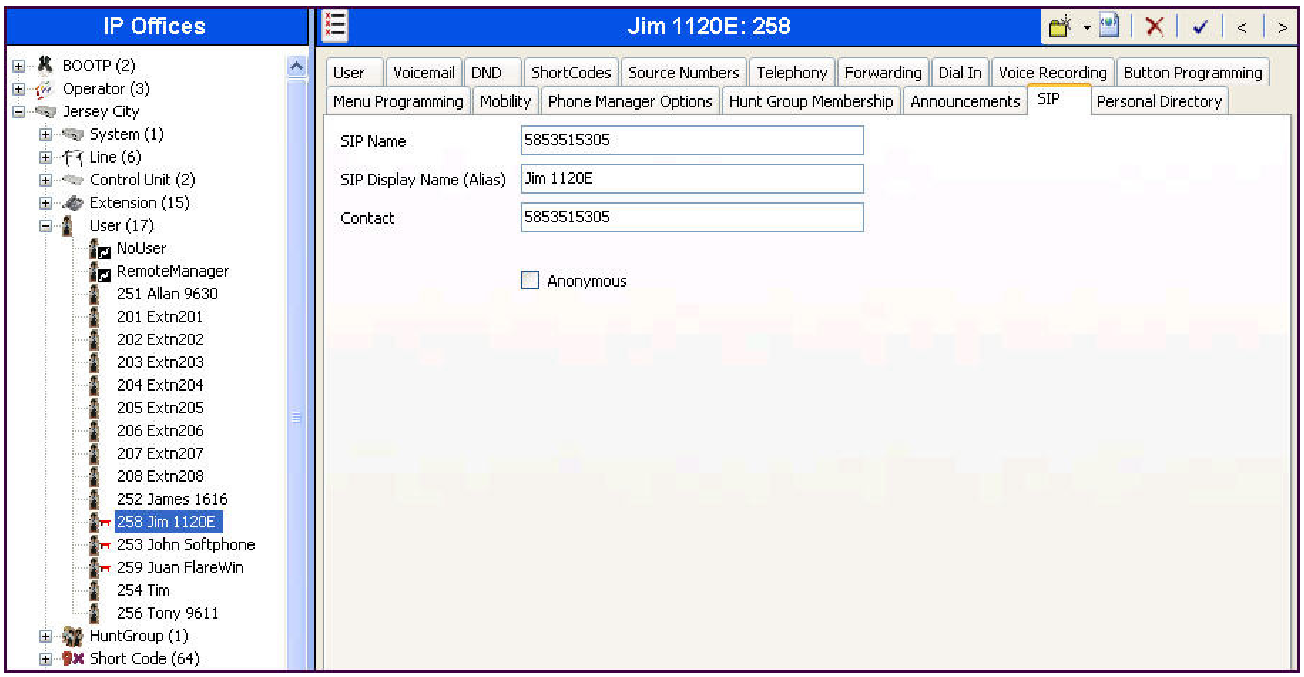


Optionally, add or edit a short code that can be used to access the SIP Line anonymously. In the screen shown below, the short code ***67N*;** is illustrated. This short code is similar to the **9N;** short code except that the **Telephone Number** field begins with the letter ***W***, which means "withhold 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 in the P-Asserted-Identity (PAI) header and would include the Privacy: Id header. Frontier would allow the call due to the presence of a valid DID in the PAI header, but 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 Lines defined in **Section 5.4**. 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 Frontier Communications. 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.

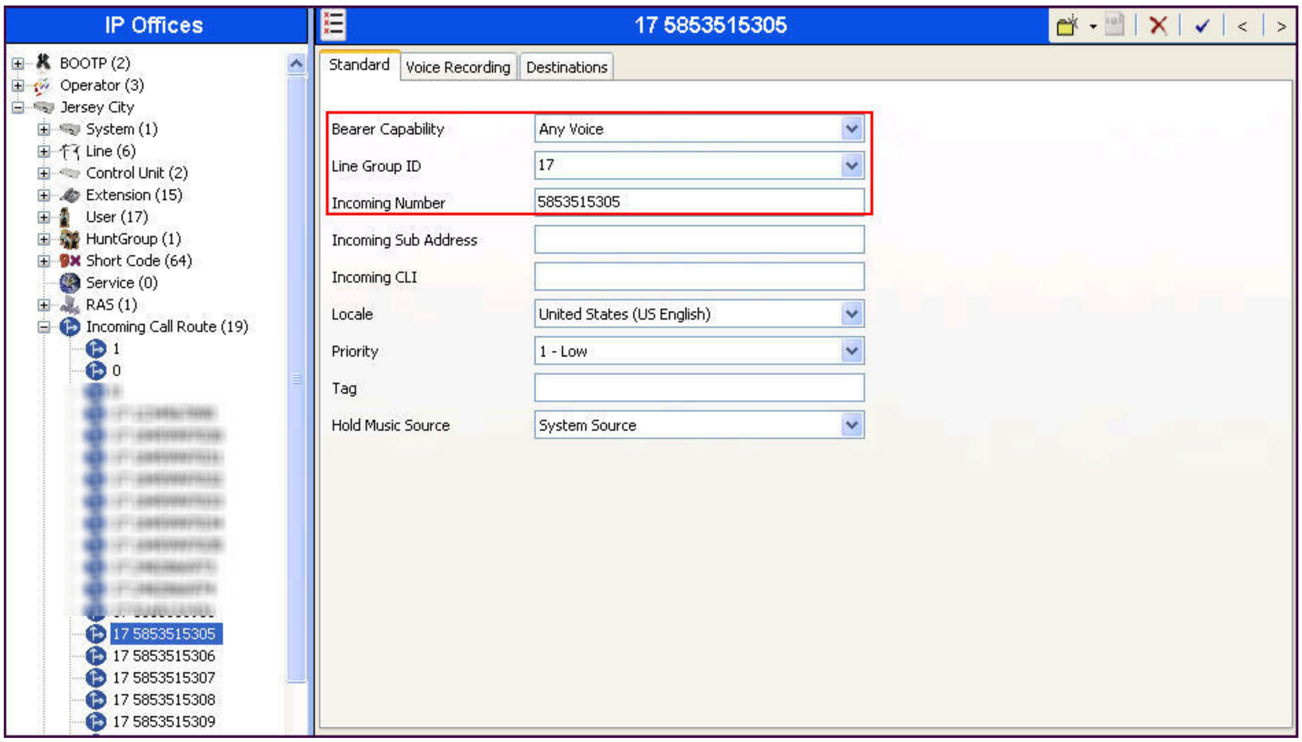
If outbound calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network.



5.7. 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 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 5.4**.
- Set the **Incoming Number** to the incoming DID number on which this route should match. Matching is right to left.



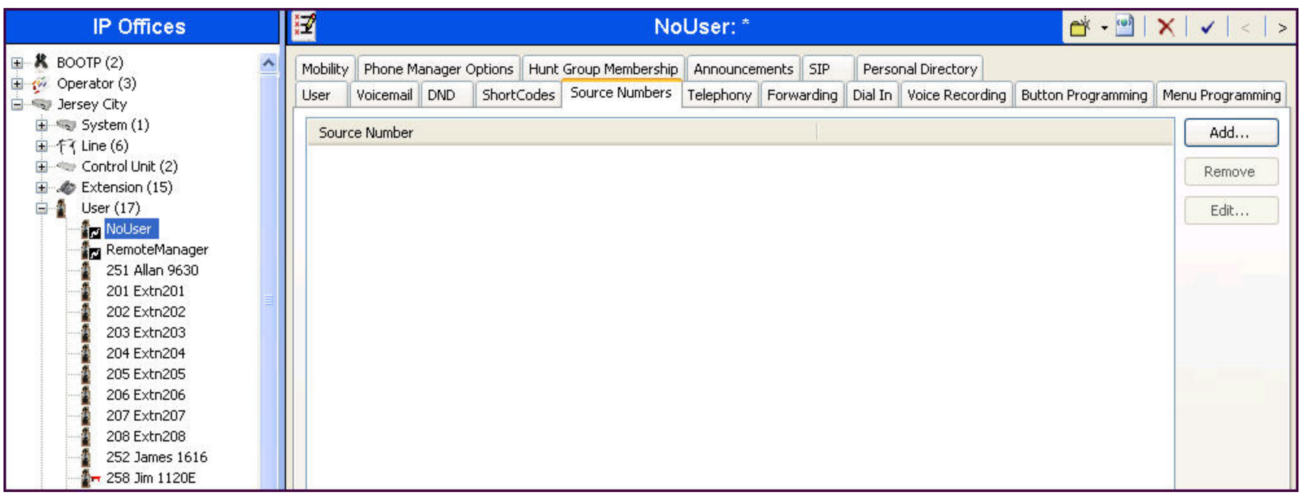
On the **Destinations** tab, select the destination from the pull-down list of the **Destination** field. In this example, incoming calls to 5853515305 on Incoming Group 17 are to be routed to the user "Jim 1120E" at extension 258.

5.8. 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:

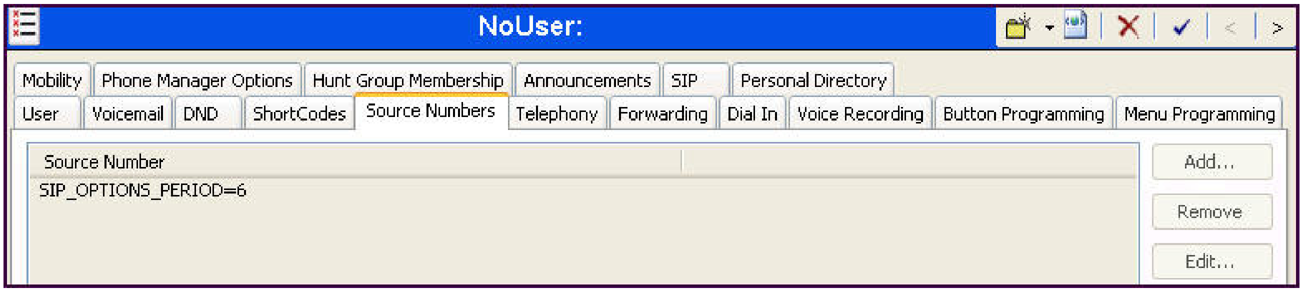
- To use the default value, set Binding Refresh = 0 or 300. OPTIONS will be sent at the 300 second frequency.
- To establish a period of less than 300 seconds, do not define the **SIP_OPTIONS_PERIOD** parameter. Instead, set the **Binding Refresh Time** to a value less than 300 seconds. The OPTIONS message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

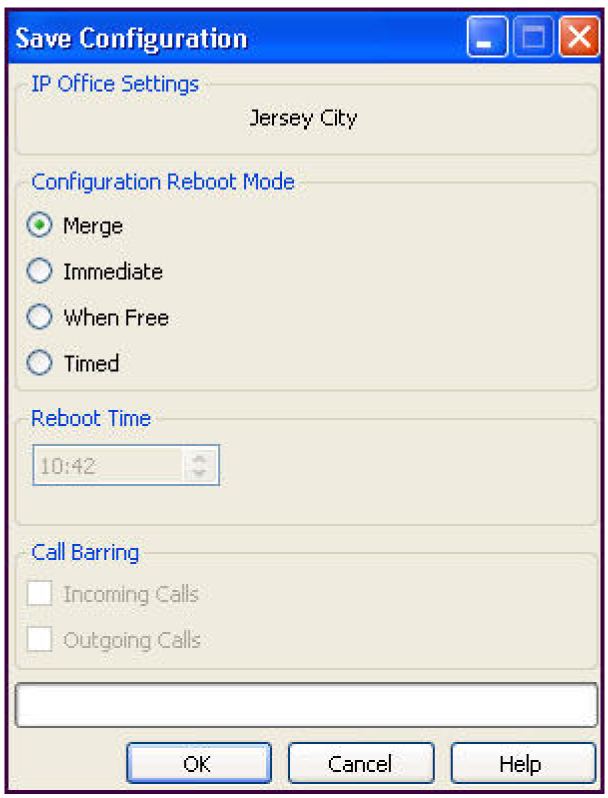
The **SIP_OPTIONS_PERIOD** parameter will appear in the list of Source Numbers as shown below. For the compliance test, an OPTIONS period of 60 seconds was desired. The **Binding Refresh Time** was set to ***60*** seconds in **Section 5.2**. There was no need to define **SIP_OPTIONS_PERIOD**.



5.9. 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. Frontier Communications SIP Trunking Configuration

Frontier Communications 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. Frontier Communications will provide the customer the necessary information to configure the Avaya IP Office SIP connection to Frontier Communications SIP Trunking including:

- Network edge IP addresses of the Frontier Communications SIP Trunking service.
- Transport and port for the Frontier Communications 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.

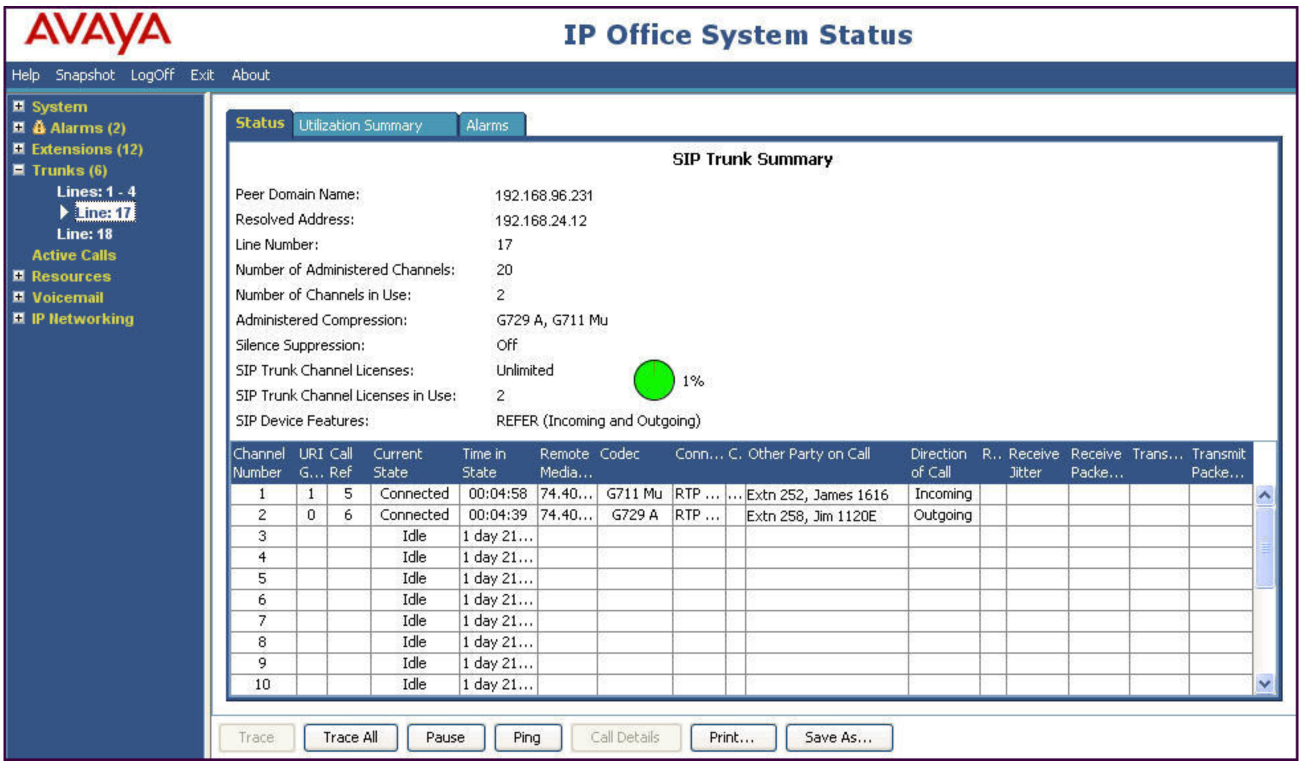
7. Verification Steps

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

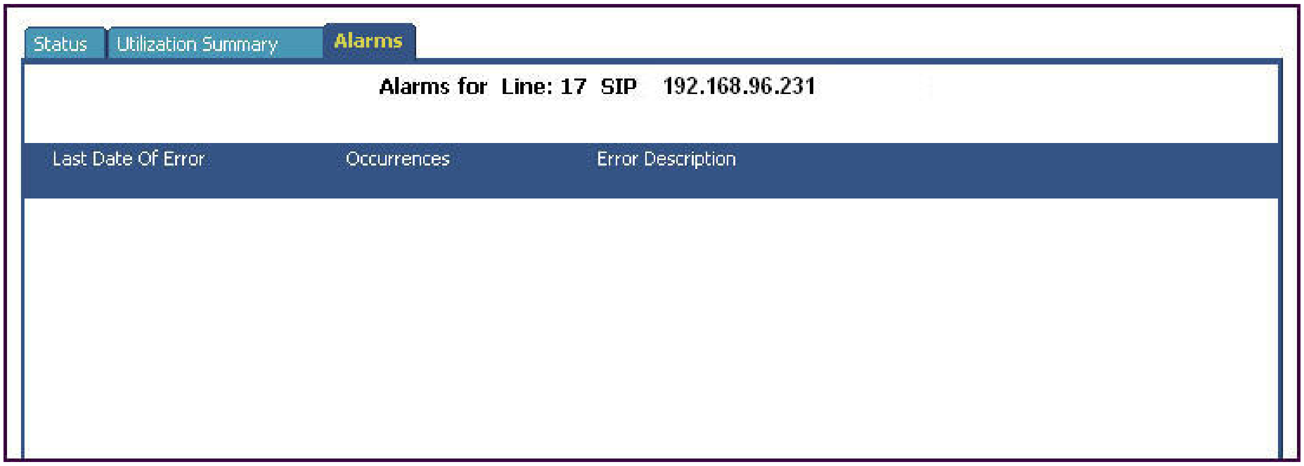
7.1. System Status

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 under **Trunks** from the left pane. On the **Status** tab in the right pane, verify the **Current Stat**e is ***Idle*** for channels where no active calls are currently in session; the state should be ***Connected*** for channels taken by active calls.



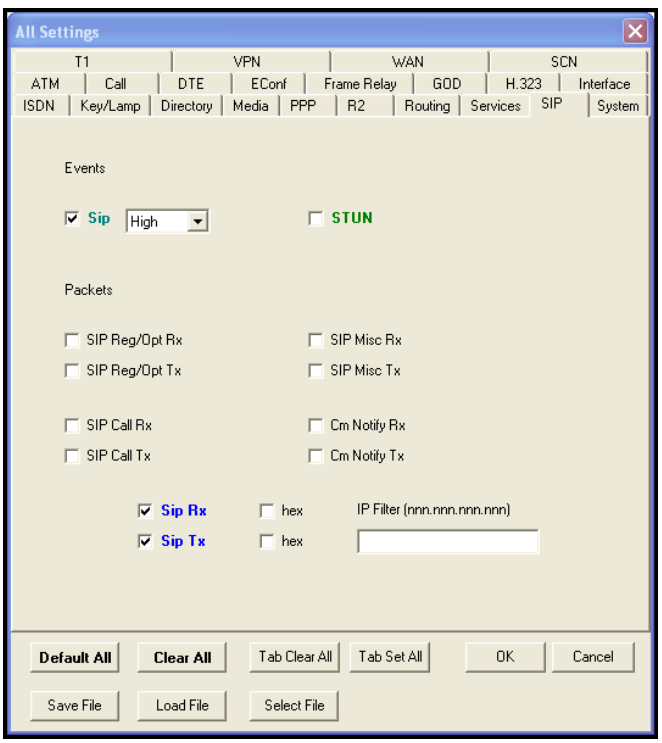
- Select the **Alarms** tab and verify that no alarms are active on the SIP line.



7.2. Monitor

The Monitor application can also be used to monitor and troubleshoot Avaya IP Office. Monitor can be accessed from **Start → Programs → IP Office → Monitor**. The application allows the monitored information to be customized. To customize, select **Filters → Trace Options**.

The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, the **SIP Rx** and **SIP Tx** boxes are checked.



8. Conclusion

The Frontier Communications SIP Trunking service passed compliance testing. These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office R8.1 and Frontier Communications 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.

[6] *IP Office System Monitor*, Document Number 15-601019, Issue 03c, March 1, 2013.

Additional IP Office documentation can be found at:
http://marketingtools.avaya.com/knowledgebase/.

Product documentation for Frontier Communications SIP Trunking is available from Frontier Communications. See **Section 2.3** on for contact information for Frontier Communications.

Appendix: SIP Line Template

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.

Note that not all of the configuration information, particularly items relevant to a specific installation environment, 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 5.4** 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:

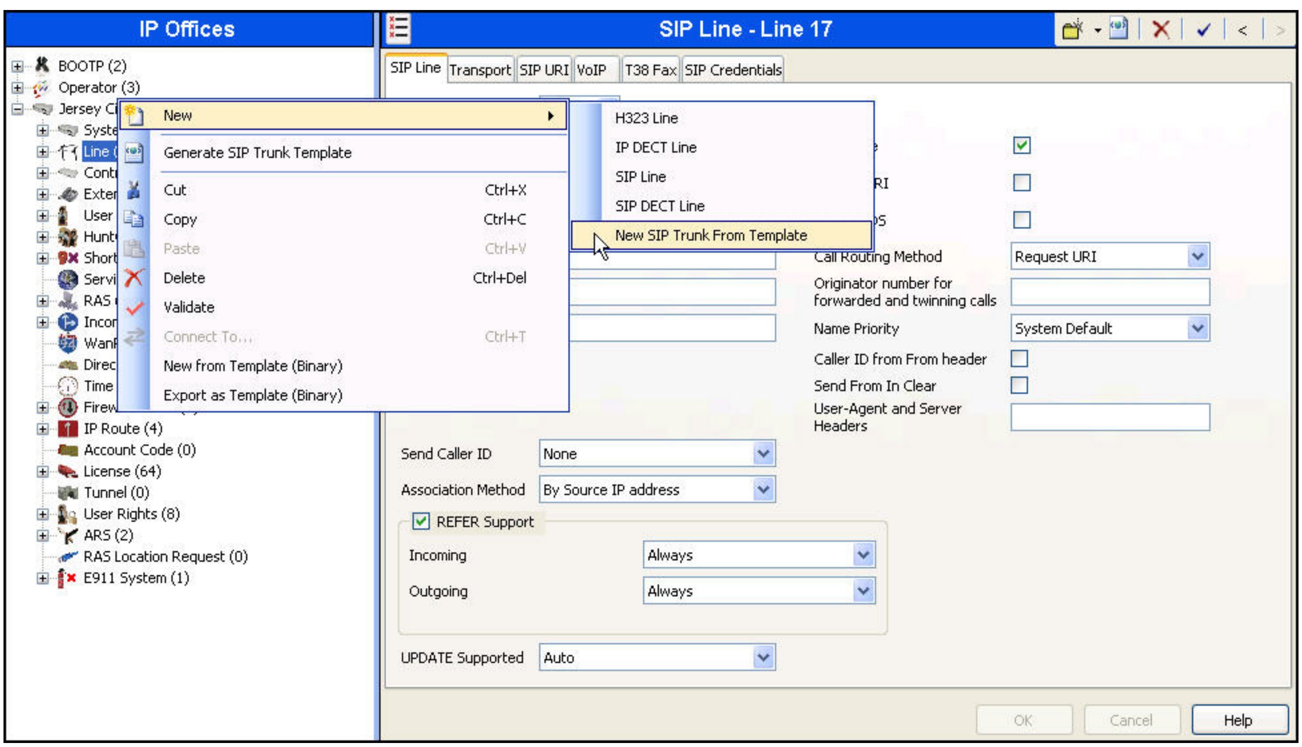

```xml
<?xml version="1.0" encoding="utf-8"?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20130920</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>Frontier SIP Trunking</DescriptiveName>
  <ITSPDomainName>192.168.96.231</ITSPDomainName>
  <SendCallerID>CallerIDDIV</SendCallerID>
  <ReferSupport>true</ReferSupport>
  <ReferSupportIncoming>1</ReferSupportIncoming>
  <ReferSupportOutgoing>1</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>true</CheckOOS>
  <CallRoutingMethod>1</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>SystemDefault</LineNamePriority>
  <UpdateSupport>UpdateAuto</UpdateSupport>
  <UserAgentServerHeader />
  <CallerIDfromFromheader>false</CallerIDfromFromheader>
  <PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
  <ITSPProxy>192.168.24.12</ITSPProxy>
  <LayerFourProtocol>SipUDP</LayerFourProtocol>
  <SendPort>5060</SendPort>
  <ListenPort>5060</ListenPort>
  <DNSServerOne>0.0.0.0</DNSServerOne>
  <DNSServerTwo>0.0.0.0</DNSServerTwo>
  <CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>
  <SeparateRegistrar />
  <CompressionMode>AUTOSELECT</CompressionMode>
  <UseAdvVoiceCodecPrefs>true</UseAdvVoiceCodecPrefs>
  <AdvCodecPref>G.729(a) 8K CS-ACELP,G.711 ULAW 64K</AdvCodecPref>
  <CallInitiationTimeout>4</CallInitiationTimeout>
  <DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
  <VoipSilenceSupression>false</VoipSilenceSupression>
  <ReinviteSupported>true</ReinviteSupported>
  <FaxTransportSupport>FOIP_T38FB</FaxTransportSupport>
  <UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
  <CodecLockdown>false</CodecLockdown>
```

```
  <Rel100Supported>false</Rel100Supported>
  <T38FaxVersion>0</T38FaxVersion>
  <Transport>UDPTL</Transport>
  <LowSpeed>0</LowSpeed>
  <HighSpeed>0</HighSpeed>
  <TCFMethod>Trans_TCF</TCFMethod>
  <MaxBitRate>FaxRate_14400</MaxBitRate>
  <EflagStartTimer>2600</EflagStartTimer>
  <EflagStopTimer>2300</EflagStopTimer>
  <UseDefaultValues>false</UseDefaultValues>
  <ScanLineFixup>true</ScanLineFixup>
  <TFOPEnhancement>true</TFOPEnhancement>
  <DisableT30ECM>true</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 the above template into a text document named **US_FrontierCommunications_SIPTrunk.xml**. Place the .xml file to the Avaya 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 ***Frontier Communications*** 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.