



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 8.0 with AT&T Mobility in Puerto Rico SIP Trunk Service - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between the service provider AT&T Mobility in Puerto Rico and Avaya IP Office 8.0.

The AT&T Mobility in Puerto Rico SIP Trunk Service provides PSTN access via a SIP trunk between the enterprise and the AT&T network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between the service provider AT&T Mobility in Puerto Rico and an Avaya IP Office solution.

In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 8.0, Avaya Voicemail Pro, Avaya IP Office soft clients and Avaya desk phones, including SIP, H.323, digital, and analog endpoints.

The Avaya IP Office system can be configured to authenticate with the SIP service provider using either SIP Trunk Registration or Static IP Authentication. These Application Notes cover the configuration of IP Office using SIP Trunk Registration, and the use of the Digest Authentication method for the validation of the calls flowing from the Enterprise to the PSTN, via the AT&T Mobility in Puerto Rico SIP Trunk Service.

The AT&T Mobility SIP Trunk Service referenced within these Application Notes is designed for business customers. The service enables local and long distance PSTN calling via standards based SIP trunks as an alternative to legacy analog or digital trunks, without the need for additional TDM enterprise gateways and the associated maintenance costs.

In the following pages, and for brevity in these Application Notes, the service provider's name "AT&T Mobility in Puerto Rico" will be abbreviated and referred to as "AT&T Mobility" or just "AT&T".

2. General Test Approach and Test Results

A simulated enterprise site was configured in the test lab using Avaya IP Office, connected to the AT&T Mobility SIP Trunk Service by means of a SIP trunk over the public Internet. This scenario may differ from a real customer environment in Puerto Rico, in which a dedicated private network connection could be provided by AT&T to the customer site.

The 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 trunk registration to the network.
- Response to SIP OPTIONS queries.
- Incoming PSTN calls to various phone types. Phone types included SIP, H.323, digital and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Authentication of calls made from the Enterprise to the PSTN.
- Outgoing PSTN calls from various phone types. Phone types included SIP, H.323, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from soft clients.
- Various call types including: local, long distance, international, outbound toll-free, emergency (911) and directory assistance (411 and 611).
- Codecs G.729A and G.711MU.
- Caller ID presentation and Caller ID restriction
- DTMF transmission using RFC 2833
- Voicemail navigation for inbound and outbound calls
- User features such as hold and resume, transfer, and conference
- Off-net call forwarding and twinning
- Use of the SIP REFER method as an alternative approach to performing call forwarding and call transfer of inbound calls back to the PSTN.
- T.38 Faxing.

Items not supported or not tested included the following:

- Operator services such as dialing 0 or 0 + 10 digits are not supported in this offer by AT&T in Puerto Rico.

2.2. Test Results

Interoperability testing with AT&T was completed with successful results for all test cases with the exception of the observations/limitations described below:

- At the time of writing this App Notes, for incoming calls that are forwarded back to the PSTN via the SIP trunk, IP Office will automatically insert the IP address of its LAN2 interface in the host portion of the Diversion header sent to the service provider. Since AT&T required the use of domain names on requests arriving to their network, including Diversion headers, these calls were rejected. This does not happen with other types of calls using Diversion headers, like twinned calls, where the header is built based on the entry on the “**ITSP Domain Name**” field in the SIP Line tab. A workaround to this situation is not to use the Diversion header at all in the IP Office, setting the “**Send caller ID**” entry in the SIP Line tab to “**None**”. By setting this, calls that are forwarded from the IP Office back to the SIP trunk are allowed to complete, with the constraint that the caller ID shown in the receiving end of these forwarded calls will be that of the host extension in the IP Office, not the originator caller ID. This limitation will also affect twinned calls. An issue report was opened with IP Office Development for investigation.

2.3. Support

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

For technical support on the AT&T Mobility SIP Trunk Services offer, call the AT&T Mobility Network Operations Center in Puerto Rico at 787-717-9900.

3. Reference Configuration

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

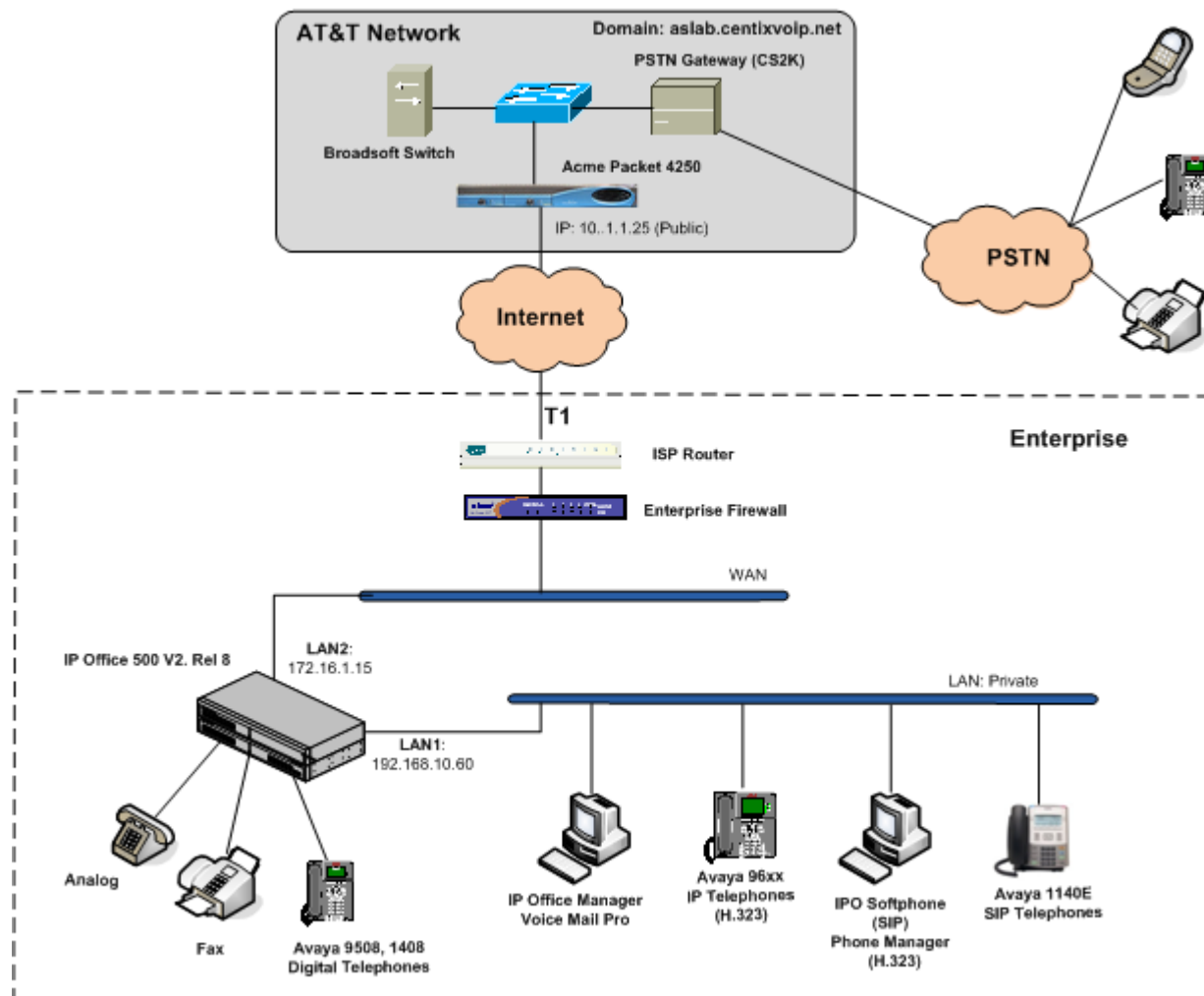


Figure 1: Test Configuration

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

The enterprise site contains the Avaya IP Office 500v2 with analog and digital extension expansion modules, as well as a VCM64 (Voice Compression Module) for supporting VoIP codecs. The LAN1 port of Avaya IP Office is connected to the enterprise LAN while the LAN2 port is connected to the public IP network. Endpoints include Avaya 9600 Series IP Telephones (with H.323 firmware), Avaya 1140E IP Telephones (with SIP firmware), Avaya 1408 and

9508D Digital Telephones, analog telephones, Officejet 4500 fax machines and PCs running Avaya IP Office Softphone and Avaya IP Office Phone Manager as SIP and H.323 soft clients, respectively. The site also has a Windows XP PC running Avaya IP Office Manager to configure and administer the Avaya IP Office system, and Avaya Voicemail Pro providing voice messaging service to the Avaya IP Office users. Mobile Twinning is configured for some of the Avaya IP Office users so that calls to these users' extensions will also ring and can be answered at the configured mobile phones.

In an actual customer configuration, the enterprise site may include additional network components between the service provider and the Avaya IP Office system, such as a session border controller or data firewall. A complete discussion of the configuration of these devices is beyond the scope of these Application Notes, however, it should be noted that SIP and RTP traffic between the service provider and the Avaya IP Office system must be allowed to pass through these devices.

Since Puerto Rico is a country member of the North American Numbering Plan (NANP), the users dialed 10 digits for local calls, and 11 (1 + 10) or 10 digits for other calls between the NANP.

4. Equipment and Software Validated

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

| Component | Version |
|---|--------------------------------------|
| Avaya | |
| Avaya IP Office 500v2 | 8.0 (42) |
| Avaya IP Office Digital Expansion Module DCPx16 | 10.0 (42) |
| Avaya IP Office Manager | 10.0 (42) |
| Avaya IP Office Voicemail Pro | 8.0(1009) |
| Avaya 9620 IP Telephone (H.323) | Avaya one-X Deskphone Edition 3.1 |
| Avaya 9640 IP Telephone (H.323) | Avaya one-X Deskphone Edition 3.1 |
| Avaya 1140E IP Telephone (SIP) | 04.03.09.00 |
| Avaya Digital Telephone 1408 | N/A |
| Avaya Digital Phone 9508 | N/A |
| Avaya IP Office Softphone | 3.1.2.17_59616 |
| Avaya IP Office Phone Manager | 4.2.39 |
| AT&T Puerto Rico SIP Trunking | |
| Acme-Packet Net-Net 4250 SBC | Firmware SC6.1.0 MR-9 GA (Build 938) |
| BroadWorks Soft Switch | R17 |
| Nortel CS2K PSTN Gateway | CVM11 |

5. Configure IP Office

This section describes the Avaya IP Office configuration necessary to support connectivity to the AT&T Mobility SIP Trunk Service. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials. A management window will appear similar to the one shown in the next section.

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

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

5.1. Licensing

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

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

The screenshot displays the Avaya IP Office Manager interface with three panes: Navigation, Group, and Details.

- Navigation Pane (Left):** Shows a tree structure of IP Office components. The 'License' component is selected and highlighted in blue.
- Group Pane (Center):** Displays a list of license types. 'SIP Trunk Channels' is selected and highlighted in blue.
- Details Pane (Right):** Shows the configuration details for the selected license.

| SIP Trunk Channels | |
|--------------------|----------------------------|
| License Key | XnMtHe6b9DXL42v [REDACTED] |
| License Type | SIP Trunk Channels |
| License Status | Valid |
| Instances | 255 |
| Expiry Date | Never |

5.2. LAN2 Settings

In the sample configuration, **IP500V2 Main** was used as the system name, and the WAN port was used to connect the Avaya IP Office to the public network. The LAN2 settings correspond to the WAN port on the Avaya IP Office. To access the LAN2 settings, first navigate to **System (1) → IP500V2 Main** in the Navigation and Group panes and then navigate to the **LAN2 → LAN Settings** tab in the Details pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office WAN port. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements.

The screenshot displays the configuration interface for 'IP500V2 Main'. On the left, a navigation pane shows a tree structure with 'IP500V2 Main' selected, leading to 'System (1)'. The main pane has tabs for 'System', 'LAN1', 'LAN2', 'DNS', 'Voicemail', 'Telephony', 'Directory Services', 'System Events', 'SMTP', and 'SMDR'. The 'LAN2' tab is active, showing 'LAN Settings'. Fields include: IP Address (172.16.1.15), IP Mask (255.255.255.192), Primary Trans. IP Address (0.0.0.0), Firewall Profile (<None>), RIP Mode (None), Enable NAT (unchecked), and Number Of DHCP IP Addresses (200). At the bottom, DHCP Mode is set to 'Disabled'.

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

The screenshot shows the 'VoIP' tab in the 'IP500V2 Main' configuration. It features checkboxes for 'H.323 Gatekeeper Enable' (unchecked), 'SIP Trunks Enable' (checked), and 'SIP Registrar Enable' (unchecked). Below these are options for 'H.323 Auto-create Extn' (unchecked), 'H.323 Auto-create User' (disabled), 'H.323 Remote Extn Enable' (unchecked), and 'Enable RTCP Monitoring On Port 5005' (checked). A section for 'RTP Port Number Range' contains two input fields: 'Port Range (Minimum)' set to 49152 and 'Port Range (Maximum)' set to 53246.

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

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

The screenshot displays the configuration interface for an Avaya IP Office, organized into three main sections:

- DiffServ Settings:** This section contains six spinners for configuring DSCP values. The first row shows 'DSCP(Hex)' set to 'B8', 'DSCP Mask (Hex)' set to 'FC', and 'SIG DSCP (Hex)' set to '88'. The second row shows 'DSCP' set to '46', 'DSCP Mask' set to '63', and 'SIG DSCP' set to '34'.
- DHCP Settings:** This section includes four spinners: 'Primary Site Specific Option Number (SSON)' set to '176', 'Secondary Site Specific Option Number (SSON)' set to '242', '1100 Voice VLAN Site Specific Option Number (SSON)' set to '232', and '1100 Voice VLAN IDs' which is an empty text field. A 'VLAN' dropdown menu is set to 'Not Present'.
- RTP Keepalives:** This section contains two dropdown menus. The 'Scope' dropdown is set to 'RTP', and the 'Initial keepalives' dropdown is set to 'Enabled'. A 'Periodic timeout' spinner is set to '0'.

All other parameters should be set according to customer requirements.

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

- Select the **Firewall/NAT Type** from the pull-down menu to the option that matches the network configuration. 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 this configuration, the **STUN Server IP Address** and **STUN Port** are not used.
- Set **Binding Refresh Time (seconds)** to **180**. This value is used to determine the frequency at which Avaya IP Office will send SIP OPTION messages to the service provider.
- Set **Public IP Address** to the IP address that was set for LAN2.
- Set **Public Port** to **5060**.
- All other parameters should be set according to customer requirements

The screenshot shows the 'IP500V2 Main' configuration window. The 'Network Topology' tab is selected. Under 'Network Topology Discovery', the following settings are visible: STUN Server IP Address (69 . 90 . 168 . 13), STUN Port (3478), Firewall/NAT Type (Open Internet), Binding Refresh Time (seconds) (180), Public IP Address (172 . 16 . 1 . 15), and Public Port (5060). There are 'Run STUN' and 'Cancel' buttons, and a checkbox for 'Run STUN on startup' which is currently unchecked.

For the compliance test, the LAN1 interface was used to connect the Avaya IP Office to the enterprise site IP network. The LAN1 interface configuration is not directly relevant to the interface with the AT&T SIP Trunk Service, and therefore is not described in these Application Notes.

5.3. Voicemail

On the **Voicemail** tab in the Details pane, configure the parameters on the **SIP Settings** section. This is necessary if remote notification features such as Callback and Outcalling are enabled, and outbound calls from the IP Office Voicemail system are to be routed on the SIP trunk. The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From and Contact headers for outgoing SIP trunk calls from the Voicemail. They were set to one of the DID numbers assigned by AT&T. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. Uncheck the **Anonymous** box. All other values used for the compliance test are shown in the screen below.

The screenshot shows the IP500V2 Main configuration window with the Voicemail tab selected. The left pane shows a tree view of the system configuration, including IP Offices, System, and various services. The main pane displays the Voicemail configuration settings.

IP500V2 Main

System | LAN1 | LAN2 | DNS | Voicemail | Telephony | Directory Services | System Events | SMTP | SMDR | Twinning | VCM | CCR

Voicemail Type: Voicemail Lite/Pro ☐ Messages Button Goes To Visual Voice

Voicemail Destination: [Dropdown]

Voicemail IP Address: 192 . 168 . 10 . 150

Backup Voicemail IP Address: 0 . 0 . 0 . 0

Voicemail Channel Reservation

Unreserved Channels: 259

Auto-Attendant: 0 Voice Recording: 0 Mandatory Voice Recording: 0

Announcements: 0 Mailbox Access: 0

DTMF Breakout

Reception / Breakout (DTMF *0/0): [Text Box]

Breakout (DTMF 2): [Text Box]

Breakout (DTMF 3): [Text Box]

SIP Settings

SIP Name: 7871111234

SIP Display Name (Alias): Voice Mail

Contact: 7871111234

Anonymous: ☐

Call Recording

Auto Restart Paused Recording (secs): 15

Hide Auto Recording: ☐

5.4. System Telephony Settings

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

The screenshot shows the 'IP500V2 Main' configuration window with the 'Telephony' tab selected. The left pane shows a tree view of the system hierarchy. The main area is divided into two sections: 'Analogue Extensions' and 'Companding Law'. The 'Analogue Extensions' section includes settings for Default Outside Call Sequence (Normal), Default Inside Call Sequence (Ring Type 1), Default Ring Back Sequence (Ring Type 2), and Restrict Analogue Extension Ringer Voltage (unchecked). The 'Companding Law' section has two sub-sections: 'Switch' and 'Line'. Both have radio buttons for 'U-Law' (selected) and 'A-Law'. Other settings include DSS Status (unchecked), Auto Hold (checked), Dial By Name (checked), Show Account Code (checked), Inhibit Off-Switch Forward/Transfer (unchecked), Restrict Network Interconnect (unchecked), Drop External Only Impromptu Conference (unchecked), Visually Differentiate External Call (unchecked), Unsupervised Analog Trunk Disconnect Handling (unchecked), and High Quality Conferencing (checked).

5.5. Twinning Calling Party Settings

Navigate to the **Twinning** tab on the Details Pane. Uncheck the **Send original calling party information for Mobile Twinning** box. This will allow the Caller ID for Twinning to be controlled by the setting on the SIP Line (**Section 5.8**). This setting also impacts the Caller ID for call forwarding.

The screenshot shows the 'IP500V2 Main' configuration window with the 'Twinning' tab selected. The left pane shows the system hierarchy. The main area has a single setting: 'Send original calling party information for Mobile Twinning' (unchecked). Below this is a text field labeled 'Calling party information for Mobile Twinning'.

5.6. System's Default Codec Selection

Select the **Codecs** tab in the Details Pane. The list of **Available Codecs** shows all the codecs supported by the system, and those selected as usable. The **Default Codec Selection** area enables the codec preference order to be configured on a system-wide basis. The buttons between the two lists can be used to move codecs between the **Unused** and the **Selected** lists, and to change the order of the codecs in the **Selected** codecs list. By default, all IP (SIP and H.323) lines and extensions will use this system default codec selection, unless configured otherwise for a specific line or extension.

The screenshot shows the 'IP500V2 Main' configuration window with the 'Codecs' tab selected. The 'Available Codecs' list on the left contains five entries, all of which are checked: G.711 ULAW 64K, G.711 ALAW 64K, G.722 64K, G.729(a) 8K CS-ACELP, and G.723.1 6K3 MP-MLQ. The 'Default Codec Selection' area in the center has an empty 'Unused' list and a 'Selected' list containing the same five codecs. Between the lists are five buttons: '>>', an up arrow, '<<', a down arrow, and '>>'. The top navigation bar includes tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, Twinning, WCM, CCR, and Codecs.

5.7. IP Route

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

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

The screenshot shows the 'IP Route' configuration window. The title bar displays '10.1.1.0'. The 'IP Address' field is set to '10 . 1 . 1 . 0', the 'IP Mask' field is set to '255 . 255 . 255 . 0', and the 'Gateway IP Address' field is set to '172 . 16 . 1 . 254'. The 'Destination' dropdown menu is set to 'LAN2', and the 'Metric' field is set to '0'. There is an unchecked checkbox for 'Proxy ARP' at the bottom.

5.8. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the AT&T SIP Trunk Service. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New**→ **SIP Line**. On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

- Set the **ITSP Domain Name** to the domain known and expected by the Service Provider. IP Office will use this domain as the host portion of the SIP URI in SIP headers of the messages sent to the network.
- Check the **In Service** box.
- Check the **Check OOS** box. With this option selected, IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- Set **Call Routing Method** to **Request URI**.
- Set **Send Caller ID** to **None**, as the workaround to the Diversion header problem described in **Section 2.2**.
- Check the **REFER support** box. Select **Always** for both **Incoming** and **Outgoing** to enable the IP Office to send REFER headers for transferred and forwarded calls that are routed back to the PSTN.
- Default values may be used for all other parameters.

| IP Offices | Line | SIP Line - Line 17 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
|--|---|--------------------|-----------|---------|-----------------|---|-------|----|--------|-----|-------|-----|-------|-----|-------|-----|-------|-----|-------|-----|-------|-----|-------|-----|-------|-----|-------|-----|-------|-----|-------|-----|-------|-----|-------|-----|-------|-----|-------|-----|-------|---|----------|-----------|---------|------|---------|-----------------|-------------|----|--|--|--|--|------------------|----------------------|--|--|--|--|------------|-------------------------------------|--|--|--|--|-------------|--------------------------|--|--|--|--|-----------|-------------------------------------|--|--|--|--|---------------------|-------------|--|--|--|--|--|--|--|--|--|--|---------------|----------------|--|--|--|--|--------|--|--|--|--|--|-----------------|---|--|--|--|--|--------------|--|--|--|--|--|----------------------|----|--|--|--|--|----------------|------|--|--|--|--|--------------------|----------------------|--|--|--|--|---------------|-------------------------------------|--|--|--|--|----------|--------|--|--|--|--|----------|--------|--|--|--|--|
| <ul style="list-style-type: none"> BOOTP (4) Operator (3) IP500V2 Main System (1) Line (19) Control Unit (5) Extension (47) User (47) HuntGroup (1) Short Code (63) Service (0) RAS (1) Incoming Call Route WanPort (0) Directory (0) Time Profile (0) Firewall Profile (1) IP Route (3) Account Code (0) License (74) Tunnel (0) User Rights (8) ARS (1) | <table border="1"> <thead> <tr> <th>Line Number</th> <th>Line Type</th> </tr> </thead> <tbody> <tr> <td>1</td> <td>PRI 2</td> </tr> <tr> <td>2</td> <td>PRI 2</td> </tr> <tr> <td>17</td> <td>SIP Li</td> </tr> <tr> <td>201</td> <td>Analo</td> </tr> <tr> <td>202</td> <td>Analo</td> </tr> <tr> <td>203</td> <td>Analo</td> </tr> <tr> <td>204</td> <td>Analo</td> </tr> <tr> <td>205</td> <td>Analo</td> </tr> <tr> <td>206</td> <td>Analo</td> </tr> <tr> <td>207</td> <td>Analo</td> </tr> <tr> <td>208</td> <td>Analo</td> </tr> <tr> <td>209</td> <td>Analo</td> </tr> <tr> <td>210</td> <td>Analo</td> </tr> <tr> <td>211</td> <td>Analo</td> </tr> <tr> <td>212</td> <td>Analo</td> </tr> <tr> <td>213</td> <td>Analo</td> </tr> <tr> <td>214</td> <td>Analo</td> </tr> <tr> <td>215</td> <td>Analo</td> </tr> <tr> <td>216</td> <td>Analo</td> </tr> </tbody> </table> | Line Number | Line Type | 1 | PRI 2 | 2 | PRI 2 | 17 | SIP Li | 201 | Analo | 202 | Analo | 203 | Analo | 204 | Analo | 205 | Analo | 206 | Analo | 207 | Analo | 208 | Analo | 209 | Analo | 210 | Analo | 211 | Analo | 212 | Analo | 213 | Analo | 214 | Analo | 215 | Analo | 216 | Analo | <table border="1"> <thead> <tr> <th>SIP Line</th> <th>Transport</th> <th>SIP URI</th> <th>VoIP</th> <th>T38 Fax</th> <th>SIP Credentials</th> </tr> </thead> <tbody> <tr> <td>Line Number</td> <td>17</td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>ITSP Domain Name</td> <td>aslab.centixvoip.net</td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>In Service</td> <td><input checked="" type="checkbox"/></td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>Use Tel URI</td> <td><input type="checkbox"/></td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>Check OOS</td> <td><input checked="" type="checkbox"/></td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>Call Routing Method</td> <td>Request URI</td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>Originator number for forwarded and twinning calls</td> <td></td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>Name Priority</td> <td>System Default</td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>Prefix</td> <td></td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>National Prefix</td> <td>0</td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>Country Code</td> <td></td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>International Prefix</td> <td>00</td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>Send Caller ID</td> <td>None</td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>Association Method</td> <td>By Source IP address</td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>REFER Support</td> <td><input checked="" type="checkbox"/></td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>Incoming</td> <td>Always</td> <td></td> <td></td> <td></td> <td></td> </tr> <tr> <td>Outgoing</td> <td>Always</td> <td></td> <td></td> <td></td> <td></td> </tr> </tbody> </table> | SIP Line | Transport | SIP URI | VoIP | T38 Fax | SIP Credentials | Line Number | 17 | | | | | ITSP Domain Name | aslab.centixvoip.net | | | | | In Service | <input checked="" type="checkbox"/> | | | | | Use Tel URI | <input type="checkbox"/> | | | | | Check OOS | <input checked="" type="checkbox"/> | | | | | Call Routing Method | Request URI | | | | | Originator number for forwarded and twinning calls | | | | | | Name Priority | System Default | | | | | Prefix | | | | | | National Prefix | 0 | | | | | Country Code | | | | | | International Prefix | 00 | | | | | Send Caller ID | None | | | | | Association Method | By Source IP address | | | | | REFER Support | <input checked="" type="checkbox"/> | | | | | Incoming | Always | | | | | Outgoing | Always | | | | |
| Line Number | Line Type | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 1 | PRI 2 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 2 | PRI 2 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 17 | SIP Li | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 201 | Analo | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 202 | Analo | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 203 | Analo | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 204 | Analo | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 205 | Analo | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 206 | Analo | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 207 | Analo | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 208 | Analo | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 209 | Analo | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 210 | Analo | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 211 | Analo | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 212 | Analo | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 213 | Analo | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 214 | Analo | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 215 | Analo | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 216 | Analo | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| SIP Line | Transport | SIP URI | VoIP | T38 Fax | SIP Credentials | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Line Number | 17 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| ITSP Domain Name | aslab.centixvoip.net | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| In Service | <input checked="" type="checkbox"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Use Tel URI | <input type="checkbox"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Check OOS | <input checked="" type="checkbox"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Call Routing Method | Request URI | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Originator number for forwarded and twinning calls | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Name Priority | System Default | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Prefix | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| National Prefix | 0 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Country Code | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| International Prefix | 00 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Send Caller ID | None | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Association Method | By Source IP address | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| REFER Support | <input checked="" type="checkbox"/> | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Incoming | Always | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Outgoing | Always | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |

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

- Set the **ITSP Proxy Address** to the IP address of the AT&T proxy server.
- Set the **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to **LAN2** as configured in **Section 5.1**.
- Set the **Send Port** to **5060**.
- Default values may be used for all other parameters.

The screenshot shows the 'SIP Line - Line 17' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' is set to '10.1.1.25'. Under 'Network Configuration', 'Layer 4 Protocol' is set to 'UDP', 'Send Port' is '5060', 'Use Network Topology Info' is 'LAN 2', and 'Listen Port' is '5060'. 'Explicit DNS Server(s)' are set to '0 . 0 . 0 . 0' and '0 . 0 . 0 . 0'. 'Calls Route via Registrar' is checked. 'Separate Registrar' is empty.

| SIP Line - Line 17 | | | | | |
|-------------------------------|-----------|-------------------------------------|------|-------------------|-----------------|
| SIP Line | Transport | SIP URI | VoIP | T38 Fax | SIP Credentials |
| ITSP Proxy Address: 10.1.1.25 | | | | | |
| Network Configuration | | | | | |
| Layer 4 Protocol | | UDP | | Send Port: 5060 | |
| Use Network Topology Info | | LAN 2 | | Listen Port: 5060 | |
| Explicit DNS Server(s) | | 0 . 0 . 0 . 0 | | 0 . 0 . 0 . 0 | |
| Calls Route via Registrar | | <input checked="" type="checkbox"/> | | | |
| Separate Registrar | | | | | |

SIP Credentials must be created for the Digest Authentication scheme used by AT&T for the SIP trunk registration and the authentication of calls made from the enterprise to the PSTN.

Every outbound REGISTER or INVITE request sent from IP Office is initially rejected by AT&T, with an error code “401 Unauthorized”. This message from the service provider also contains a challenge or “nonce”, which in conjunction with the SIP credentials previously supplied by AT&T are used by IP Office to generate a new request by means of an MD5 hash algorithm. This request, containing a newly generated Authorization header with the proper authentication credentials and response to the “nonce”, is sent to AT&T and is then accepted and processed accordingly.

To create a SIP Credentials entry, first select the **SIP Credentials** tab. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane. For the compliance test, a single SIP credential entry was created with the parameters shown below:

- Set **User name**, **Authentication Name** and **Password** to the values provided by the service provider.
- Leave the **Registration required** option checked, since the SIP trunk will need to register with the AT&T SIP Trunk Service.
- Use the default value of **60** minutes, in the **Expiry** field. The actual registration expiration time is negotiated and agreed upon as part of the registration exchange.

The screenshot shows the 'SIP Line - Line 17*' configuration window. The 'SIP Credentials' tab is selected. A table lists the credentials, and an 'Edit SIP Credentials' dialog box is open at the bottom.

| Index | UserName | Authentication Name | Contact | Expiry (mins) | Register |
|-------|------------|---------------------|---------|---------------|----------|
| 1 | 7871111235 | 7871111235 | | 60 | True |

Buttons: Add..., Remove, Edit...

Edit SIP Credentials

User name: 7871111235
Authentication Name: 7871111235
Contact:
Password:
Expiry (mins): 60
Registration required: ☒

Buttons: OK, Cancel

A SIP URI entry needs to be created to match each incoming number that Avaya IP Office will accept on this line. To create a SIP URI entry, first select the **SIP URI** tab. 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 number assigned to an Avaya IP Office user. The entry was created with the parameters shown below.

- Set **Local URI**, **Contact** and **Display Name** to *Use Internal Data*. Set **PAI** to **None**.
- For **Registration**, select **1:7871111235** from the pull down menu. This matches the SIP credentials previously created on the **SIP Credentials** tab.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group **17** was defined that only contains this line (line 17).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.
- Default values may be used for all other parameters.

| Channel | Groups | Via | Local URI | Contact | Display Name | PAI | Credential | Max Calls |
|---------|--------|------|-----------|---------|--------------|------|------------|-----------|
| 1 | 17 17 | 6... | | | | N... | 1: 7871... | 10 |

Edit Channel

Via: 172.16.1.15

Local URI: Use Internal Data

Contact: Use Internal Data

Display Name: Use Internal Data

PAI: None

Registration: 1: 7871111235

Incoming Group: 17

Outgoing Group: 17

Max Calls per Channel: 10

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

- In the sample configuration, the **Codec Selection** was configured using the **Custom** option, allowing an explicit ordered list of codecs to be specified, different from the system default defined in **Section 5.6**. The buttons allow to set an explicit list of codecs to be used on the line, in that specific order of preference.
- For **Fax Transport Support**, select **T38 Fallback**. With this setting, outgoing fax calls will use T.38 transport, but under certain error conditions a re-invite is sent to connect at G711 pass-through. Incoming audio calls that detect fax tones also initiate fax transport using T38 Fallback.
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Uncheck the **VoIP Silence Suppression** box.
- Check the **Re-invite Supported** box to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk.
- Check the **PRACK/100rel Supported** box, to advertise the support for provisional responses and Early Media to AT&T.
- Default values may be used for all other parameters.

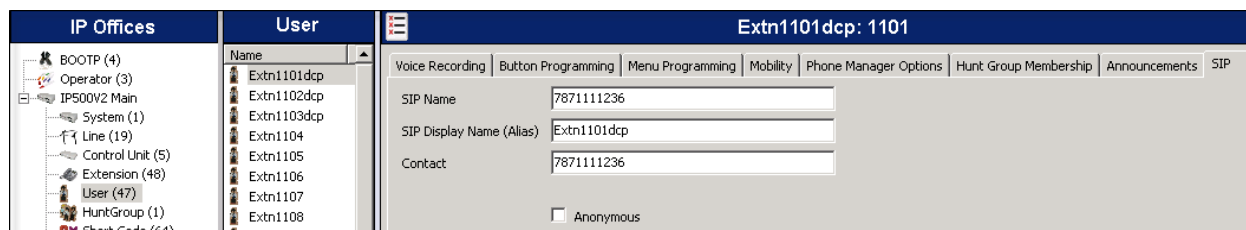
The screenshot shows the 'SIP Line - Line 17' configuration window with the 'VoIP' tab selected. The 'Codec Selection' dropdown is set to 'Custom'. Below it, there are two lists: 'Unused' and 'Selected'. The 'Unused' list contains 'G.711 ALAW 64K' and 'G.723.1 6K3 MP-MLQ'. The 'Selected' list contains 'G.729(a) 8K CS-ACELP' and 'G.711 ULAW 64K'. Between the lists are buttons for moving items: '>>', '<<', '<', and '>'. To the right of the lists are checkboxes for 'VoIP Silence Suppression' (unchecked), 'Re-invite Supported' (checked), 'Use Offerer's Preferred Codec' (unchecked), 'Codec Lockdown' (unchecked), and 'PRACK/100rel Supported' (checked). Below the codec lists, the 'Fax Transport Support' dropdown is set to 'T38 Fallback'. The 'Call Initiation Timeout (s)' is set to '4'. The 'DTMF Support' dropdown is set to 'RFC2833'.

Since default values were used on the **T38 Fax** tab, this tab will not be visited.

5.9. Users

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.8**. To configure these settings, first navigate to **User** in the left Navigation Pane, and then select the name of the user to be modified in the center Group Pane. In the example below, the name of the user is *Extn 1101dcp*. Select the **SIP** tab in the Details Pane.

The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From and Contact headers for outgoing SIP trunk calls. In addition, these settings are used to match against the SIP URI of incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.8**). The example below shows the settings for user “Extn1101dcp”. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise by AT&T. In the example, the DID number **7871111236** was used. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name.



5.10. Incoming Call Route

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

In a scenario like the one used for the compliance test, only one incoming route is needed, which allows any incoming number arriving on the SIP trunk to reach any predefined extension in IP Office. The routing decision for the call is based on the parameters configured for **Call Routing Method** and **SIP URI (Section 5.8)** and the users **SIP Name**, already populated with the assigned AT&T DID numbers (**Section 5.9**)

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

- Set **Bearer Capacity** to *Any Voice*.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.8**.
- Default values may be used for all other parameters.

| IP Offices | Incoming C... | 17 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
|--|--|---------------|---------|---|--|---|--|----|--|----|--|--|----------|-----------------|--------------|-------------------|--|-----------|---------------|--|----|-----------------|--|--|----------------------|--|--|--------------|--|--|--------|--|--|----------|--|---------|-----|--|--|-------------------|--|---------------|
| <ul style="list-style-type: none"> BOOTP (4) Operator (3) IP500V2 Main <ul style="list-style-type: none"> System (1) Line (19) Control Unit (5) Extension (47) User (47) HuntGroup (1) Short Code (63) Service (0) RAS (1) Incoming Call Rou WanPort (0) Directory (0) Time Profile (0) Firewall Profile (1) | <table border="1"> <thead> <tr> <th>Line Group ID</th> <th>Incomin</th> </tr> </thead> <tbody> <tr><td>0</td><td></td></tr> <tr><td>0</td><td></td></tr> <tr><td>17</td><td></td></tr> <tr><td>17</td><td></td></tr> </tbody> </table> | Line Group ID | Incomin | 0 | | 0 | | 17 | | 17 | | <table border="1"> <thead> <tr> <th>Standard</th> <th>Voice Recording</th> <th>Destinations</th> </tr> </thead> <tbody> <tr> <td colspan="2">Bearer Capability</td> <td>Any Voice</td> </tr> <tr> <td colspan="2">Line Group ID</td> <td>17</td> </tr> <tr> <td colspan="2">Incoming Number</td> <td></td> </tr> <tr> <td colspan="2">Incoming Sub Address</td> <td></td> </tr> <tr> <td colspan="2">Incoming CLI</td> <td></td> </tr> <tr> <td colspan="2">Locale</td> <td></td> </tr> <tr> <td colspan="2">Priority</td> <td>1 - Low</td> </tr> <tr> <td colspan="2">Tag</td> <td></td> </tr> <tr> <td colspan="2">Hold Music Source</td> <td>System Source</td> </tr> </tbody> </table> | Standard | Voice Recording | Destinations | Bearer Capability | | Any Voice | Line Group ID | | 17 | Incoming Number | | | Incoming Sub Address | | | Incoming CLI | | | Locale | | | Priority | | 1 - Low | Tag | | | Hold Music Source | | System Source |
| Line Group ID | Incomin | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 0 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 0 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 17 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| 17 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Standard | Voice Recording | Destinations | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Bearer Capability | | Any Voice | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Line Group ID | | 17 | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Incoming Number | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Incoming Sub Address | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Incoming CLI | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Locale | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Priority | | 1 - Low | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Tag | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |
| Hold Music Source | | System Source | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | | |

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

| 17 | | | |
|----------|-----------------|--------------|--------------------|
| Standard | Voice Recording | Destinations | |
| | TimeProfile | Destination | Fallback Extension |
| ▶ | Default Value | . | |

5.11. Short Code /Automatic Route Selection

Define a short code to route outbound traffic to the SIP line. To create a short code, right-click on **Short Code** in the Navigation Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **8N;**. This short code will be invoked when the user dials 8 followed by any number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N"@aslab.centixvoip.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 name used by the AT&T SIP proxy server during the compliance test follows the @ sign in the above expression.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.8**. This short code will use this line group when placing outbound calls.
- Default values may be used for all other parameters.

The screenshot displays the Avaya configuration interface. On the left, the 'IP Offices' navigation pane shows a tree structure with 'Short Code (64)' selected. The main area is divided into two panes. The left pane, titled 'Short Code', contains a table with columns 'Code' and 'Telephone'. The right pane, titled '8N;; Dial', shows the configuration for a specific short code.

| Code | Telephone |
|--------|-----------|
| *12*N# | N |
| *13*N# | N |
| *15 | |
| *16 | |
| *17 | ?U |
| *18 | |
| *19 | |
| *20*N# | N |
| *21*N# | N |
| *29 | |
| *30 | |
| *31 | |

The configuration for '8N;; Dial' is as follows:

- Short Code: 8N;;
- Feature: Dial
- Telephone Number: N"@aslab.centixvoip.net"
- Line Group ID: 17
- Locale: United States (US English)
- Force Account Code: ☐

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

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

9N: Dial

Short Code

Code: 9N

Feature: Dial

Telephone Number: N

Line Group ID: 50: Main

Locale: United States (US English)

Force Account Code: ☐

The following screen shows the example ARS configuration for the route **Main**. Note the sequence of **Xs** used in the **Code** column of several entries to specify the exact numbers of digits to be expected, following the access code and the first digit on the string. For example, for local calls in Puerto Rico, the user dialed 9, then 10 digit numbers starting with a 7. This type of setting results in a much quicker response in the delivery of the call by the IP Office. Also note that the AT&T domain name does not need to be specified in the **Telephone Number** field of the entries. It is automatically inserted by IP Office in the outbound SIP-URI, using the value from the **ITSP Domain Name** field, entered in **Section 5.8**.

Main

ARS

ARS Route Id: 50

Route Name: Main

Dial Delay Time: System Default (4)

Secondary Dial tone: ☒ SystemTone

Check User Call Barring: ☒

In Service: ☒ Out of Service Route: <None>

Time Profile: <None> Out of Hours Route: <None>

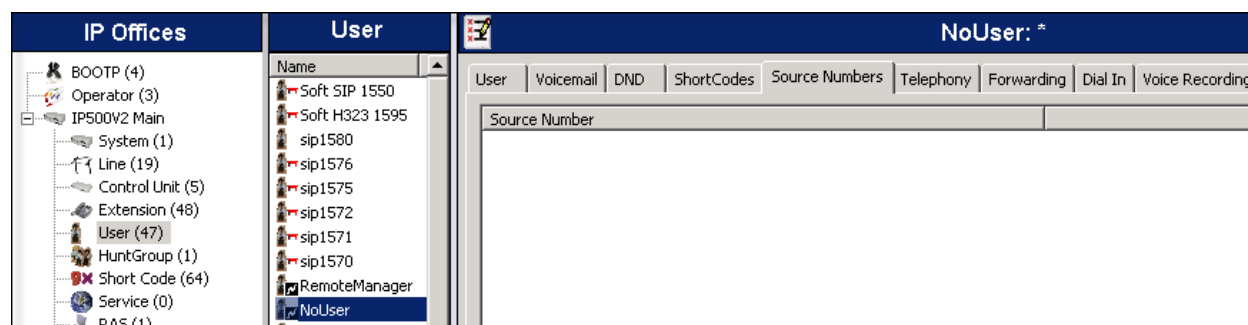
| Code | Telephone Number | Feature | Line Group ID |
|-------------|------------------|----------------|---------------|
| 11 | 911 | Dial Emergency | 17 |
| 911 | 911 | Dial Emergency | 17 |
| 7XXXXXXXXXX | 7N | Dial | 17 |
| 1XXXXXXXXXX | 1N | Dial | 17 |
| 0XXXXXXXXXX | 0N | Dial | 17 |
| 411 | 411 | Dial | 17 |
| 611 | 611 | Dial | 17 |

Add... Remove Edit...

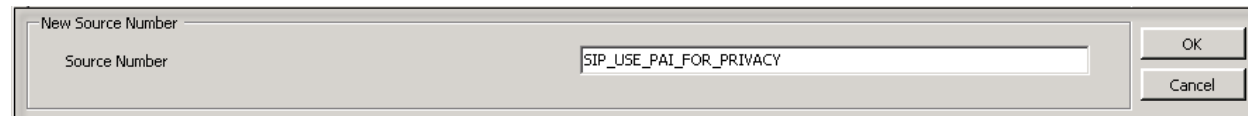
5.12. Privacy/Anonymous Calls

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

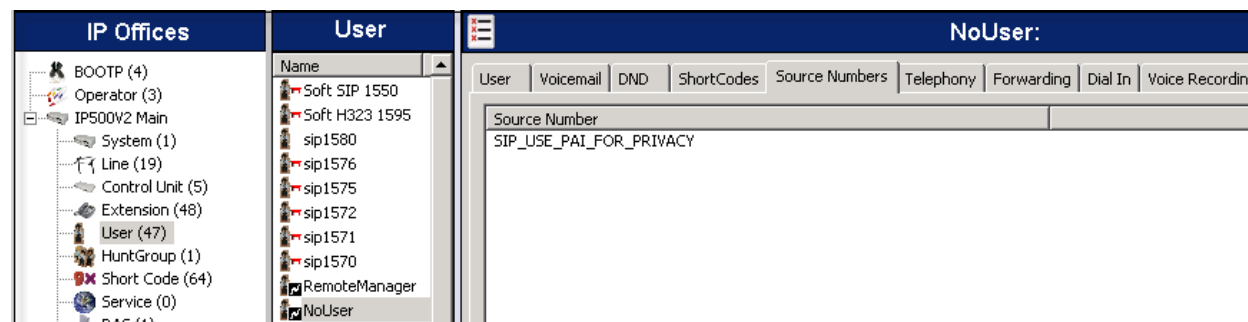
To configure Avaya IP Office to use PAI for privacy calls, navigate to **User → NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



At the bottom of the Details Pane, the **Source Number** field will appear. Enter **SIP_USE_PA1_FOR_PRIVACY**. Click **OK**.



The **SIP_USE_PA1_FOR_PRIVACY** parameter will appear in the list of Source Numbers as shown below.



5.13. Save Configuration

Navigate to **File → Save Configuration** in the menu bar at the top left of the screen to save the configuration performed in the preceding sections (not shown).

6. AT&T Mobility SIP Trunking Configuration

AT&T Mobility in Puerto Rico is responsible for the configuration of the AT&T SIP Trunk service. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise site. AT&T will provide the customer the necessary information to configure the Avaya IP Office SIP connection to AT&T including:

- IP address of the AT&T SIP Proxy server.
- AT&T SIP domain name.
- Supported codecs.
- DID numbers.
- Username and password to be used for SIP Credentials.
- All IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices.

7. Verification Steps

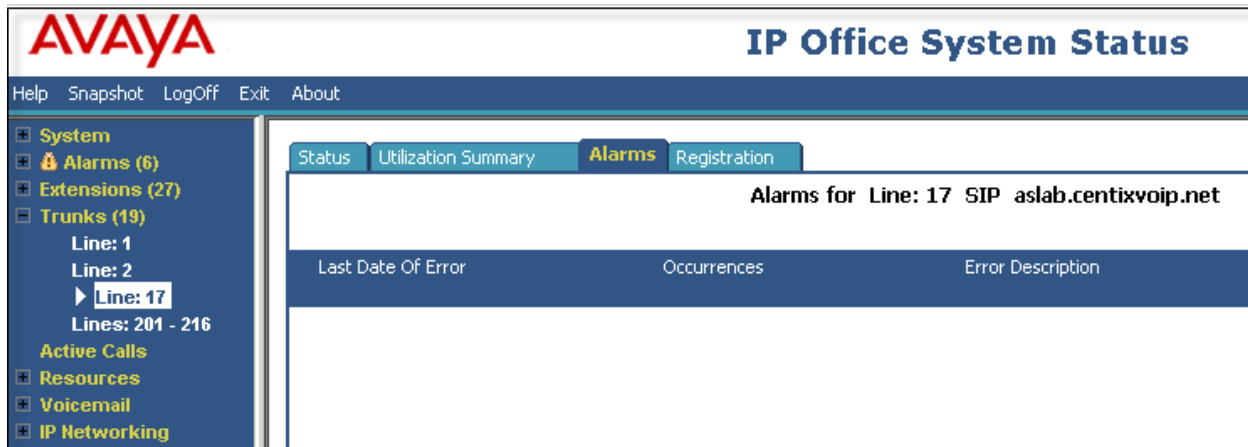
The following steps may be used to verify the configuration:

- Use the Avaya IP Office System Status application to verify the state of the SIP connection. Launch the application from **Start → Programs → IP Office → System Status** on the PC where Avaya IP Office Manager was installed. Log in using the appropriate credentials and select the SIP line of interest from the left pane. On the **Status** tab in the right pane, verify that the **Current State** is *Idle* for each channel (assuming no active calls at present time).

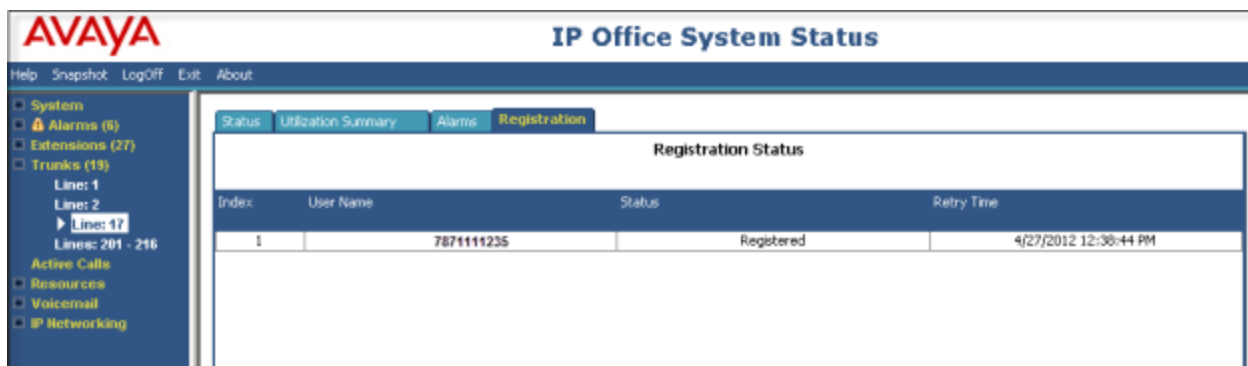
The screenshot displays the Avaya IP Office System Status application. The left sidebar shows a tree view with categories like System, Alarms (6), Extensions (27), Trunks (19), Active Calls, Resources, Voicemail, and IP Networking. The 'Trunks' category is expanded, showing 'Line: 1', 'Line: 2', and 'Line: 17' (selected). The main pane shows the 'Status' tab for the selected line. The 'SIP Trunk Summary' section displays various parameters: Peer Domain Name (aslab.centrixvoip.net), Resolved Address (10.1.1.25), Line Number (17), Number of Administered Channels (10), Number of Channels in Use (0), Administered Compression (G729 A, G711 Mu), Silence Suppression (Off), SIP Trunk Channel Licenses (Unlimited), SIP Trunk Channel Licenses in Use (0), and SIP Device Features (REFER (Incoming and Outgoing)). A green circle indicates 0% utilization. Below this is a table with columns for Channel Number, URI, Call G..., Ref, Current State, Time in State, Remote Media Ad..., Codec, Connec..., Caller ID or Diale..., Other Party on Call, Direction of Call, Round Trip De..., Receive Jitter, Receive Packet ..., Transmit Jitter, and Transmit Packet ... The table lists 10 channels, all of which are in the 'Idle' state.

| Channel Number | URI | Call G... | Ref | Current State | Time in State | Remote Media Ad... | Codec | Connec... | Caller ID or Diale... | Other Party on Call | Direction of Call | Round Trip De... | Receive Jitter | Receive Packet ... | Transmit Jitter | Transmit Packet ... |
|----------------|-----|-----------|-----|---------------|---------------|--------------------|-------|-----------|-----------------------|---------------------|-------------------|------------------|----------------|--------------------|-----------------|---------------------|
| 1 | | | | Idle | 01:17:17 | | | | | | | | | | | |
| 2 | | | | Idle | 1 day 2... | | | | | | | | | | | |
| 3 | | | | Idle | 1 day 2... | | | | | | | | | | | |
| 4 | | | | Idle | 1 day 2... | | | | | | | | | | | |
| 5 | | | | Idle | 1 day 2... | | | | | | | | | | | |
| 6 | | | | Idle | 1 day 2... | | | | | | | | | | | |
| 7 | | | | Idle | 1 day 2... | | | | | | | | | | | |
| 8 | | | | Idle | 1 day 2... | | | | | | | | | | | |
| 9 | | | | Idle | 1 day 2... | | | | | | | | | | | |
| 10 | | | | Idle | 1 day 2... | | | | | | | | | | | |

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



- Select the **Registration** tab and verify that the user name used in the SIP credentials for the SIP line has a **Status** of *Registered*.



- 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 system with two-way audio.

8. Conclusion

These Application Notes describe the procedures required to configure SIP trunk connectivity between Avaya IP Office 8.0 and the AT&T Mobility in Puerto Rico SIP Trunk Service, as shown in **Figure 1**.

AT&T Mobility in Puerto Rico SIP Trunk Service passed compliance testing. Interoperability testing of the sample configuration was completed with successful results for all test cases with the exception of the observations/limitations described in **Section 2.2**.

9. Additional References

- [1] *IP Office 8.0 Installation Manual*, Document Number 15-601042, March 2012
- [2] *IP Office Manager Manual 10.0*, Document Number 15-601011, April 2012
- [3] *IP Office Release 8.0 Implementing Voicemail Pro*, Document Number 15-601064, December, 2011
- [4] *IP Office System Status Application*, Document Number 15-601758, November 2011
- [5] *Avaya IP Office Knowledgebase*, <http://marketingtools.avaya.com/knowledgebase/>

Product documentation for Avaya products may be found at <http://support.avaya.com>. Product documentation for AT&T Mobility SIP Trunk Service is available from AT&T Mobility.

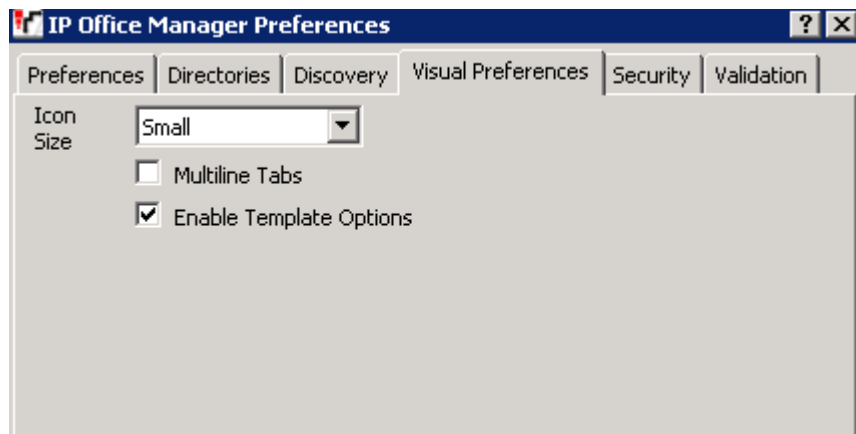
Appendix: SIP Line Template

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

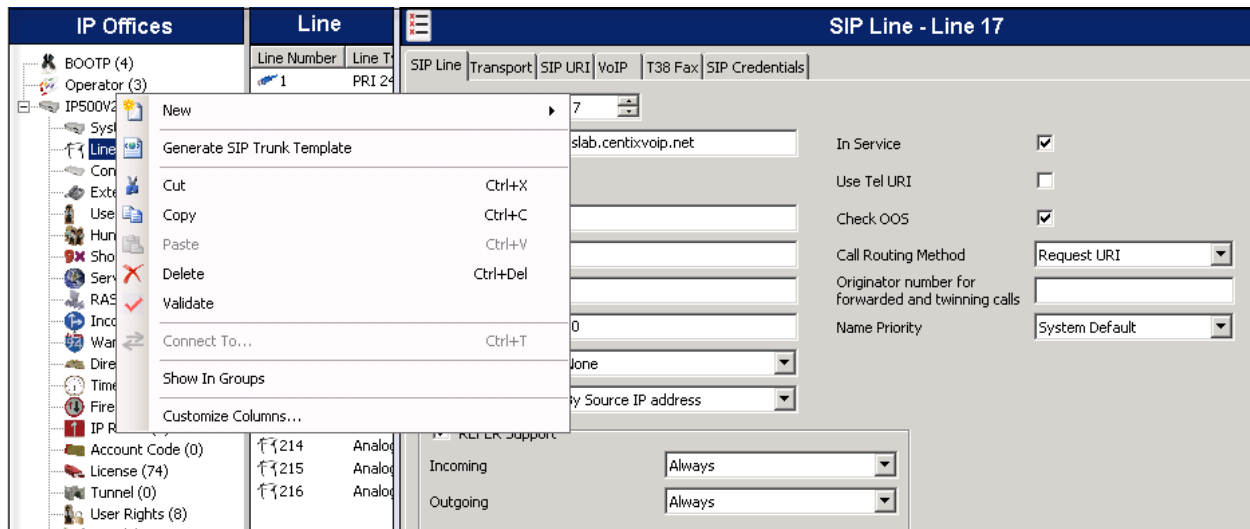
Not all of the configuration information is included in the SIP Line Template, therefore, it is critical that the SIP Line configuration be verified/updated after a template has been imported, and additional configuration be supplemented using **Section 5.8** in these Application Notes as a reference.

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

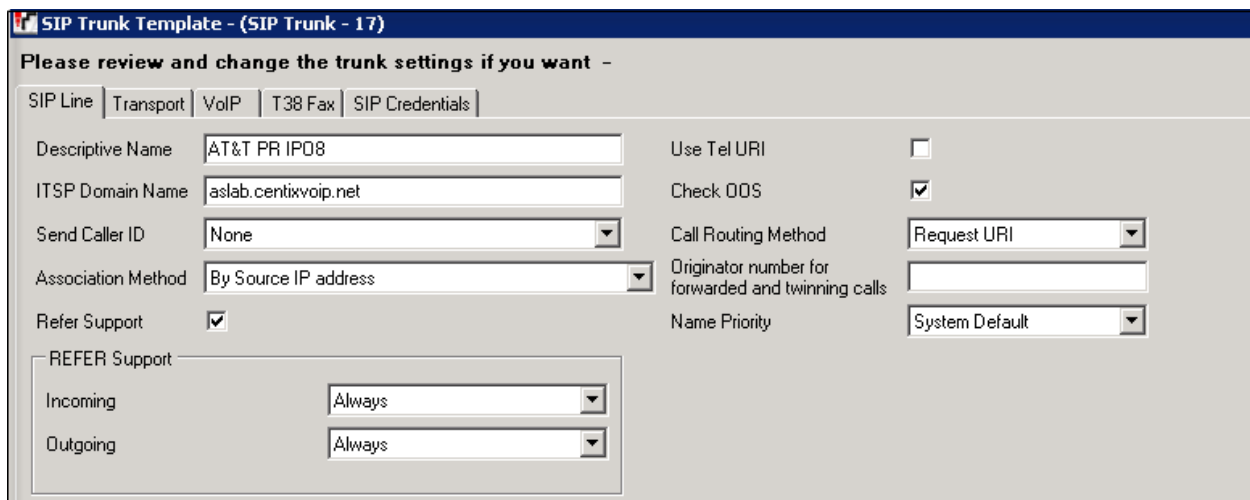
To enable template support, select **File**, then **Preferences**. On the **Visual Preferences** tab, check the **Enable Template Options** box.



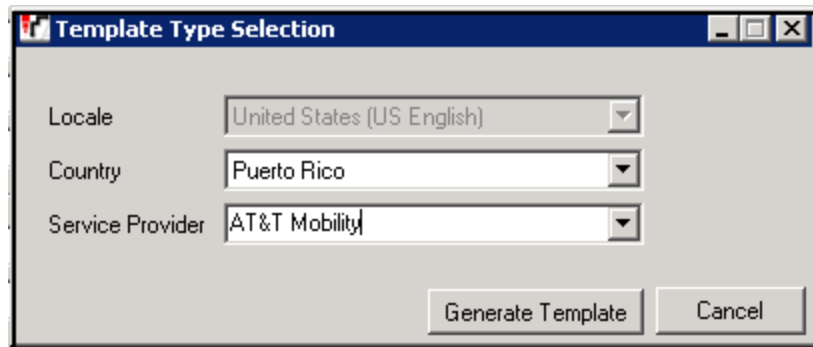
To create a SIP Line Template from the configuration, on the left Navigation Pane, right click the Sip Line (17), and select **Generate SIP Trunk Template**.



The trunk's settings are displayed as configured in **Section 5.8**. Enter a descriptive name for the template, adjust the settings if required, and then click on **Export**.



On the next screen, **Template Type Selection**, select the **Country**, enter the name for the **Service Provider**, and click **Generate Template**.



The following is the exported SIP Line Template file **PR_AT&T Mobility_SIPTrunk.xml**:

```
<?xml version="1.0" encoding="utf-8" ?>
<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20120430</Version>
  <SystemLocale>enu</SystemLocale>
  <DescriptiveName>AT&T PR IPO8</DescriptiveName>
  <ITSPDomainName>aslab.centixvoip.net</ITSPDomainName>
  <SendCallerID>CallerIDNone</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>
  <ITSPProxy>10.1.1.25</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>
```

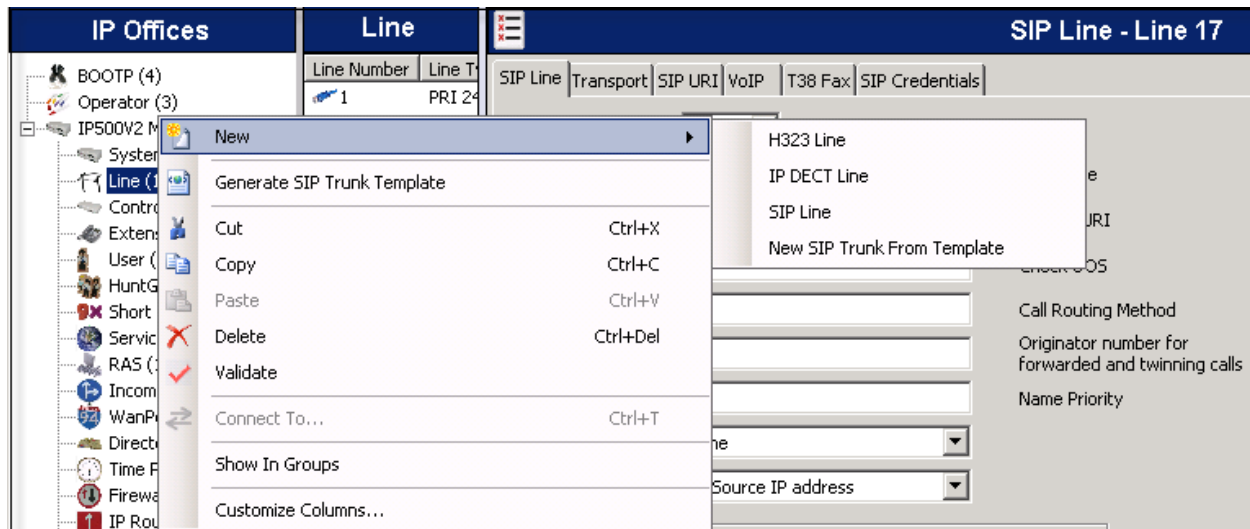
```

<ReinviteSupported>true</ReinviteSupported>
<FaxTransportSupport>FOIP_T38FB</FaxTransportSupport>
<UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
<CodecLockdown>false</CodecLockdown>
<Rel100Supported>true</Rel100Supported>
<T38FaxVersion>3</T38FaxVersion>
<Transport>UDPTL</Transport>
<LowSpeed>0</LowSpeed>
<HighSpeed>0</HighSpeed>
<TCFMethod>Trans_TCF</TCFMethod>
<MaxBitRate>FaxRate_14400</MaxBitRate>
<EflagStartTimer>2600</EflagStartTimer>
<EflagStopTimer>2300</EflagStopTimer>
<UseDefaultValues>true</UseDefaultValues>
<ScanLineFixup>true</ScanLineFixup>
<TFOPEnhancement>true</TFOPEnhancement>
<DisableT30ECM>false</DisableT30ECM>
<DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
<DisableT30MRCompression>false</DisableT30MRCompression>
<NSFOVERRIDE>false</NSFOVERRIDE>
<SIPCredentials>
  <Expiry>60</Expiry>
  <RegistrationRequired>true</RegistrationRequired>
</SIPCredentials>
</Template>

```

To import the template into a new IP Office system, copy and paste the exported xml template file to the Templates directory (C:\Program Files\Avaya\IP Office\Manager\Templates) on the PC where IP Office Manager for the new system is running.

Next, import the template into the new IP Office system by creating a new SIP Line as shown in the screenshot below. In the Navigation Pane on the left, right-click on **Line** then navigate to **New, New SIP Trunk From Template**:



On the next screen, **Template Type Selection**, verify that the information in the **Country** and **Service Provider** fields is correct. If more than one template is present, use the drop-down menus to select the required template. Click **Create new SIP Trunk** to finish the process.



©2012 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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.