



Application Notes for Configuring Avaya IP Office 8.1 with Servicio Troncal SIP de Axtel – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between the service provider Axtel and Avaya IP Office 8.1.

The Servicio Troncal SIP de Axtel provides PSTN access via a SIP trunk between the enterprise and the Axtel 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 Axtel and an Avaya IP Office solution.

In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 8.1, 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 for the authentication with Axtel.

The Servicio Troncal SIP de Axtel referenced within these Application Notes is designed for business customers in Mexico. 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.

During the next pages in these Application Notes, the name in Spanish of the service provider's offer, "Servicio Troncal SIP de Axtel", will be also mentioned using the English translation "Axtel SIP Trunk Service".

2. General Test Approach and Test Results

A simulated enterprise site was configured in the test lab using Avaya IP Office, connected to the Axtel SIP Trunk Service by means of a SIP trunk over the public Internet. 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.
- 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, etc.
- Codecs G729B, G.711A 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.

2.2. Test Results

Interoperability testing with Axtel was completed with successful results with the exception of the observations/limitations described below:

- **SIP Network REFER with Blind Transfer:** During the compliance tests, REFER support needed to be disabled in the IP Office SIP Line, to allow the completion of calls that are blind transferred back to Axtel. Otherwise the call transfer failed. The implication is that for this type of call, the IP Office is not released after the call is transferred, and two channels in the SIP trunks remain busy for the entire duration of the call.
- **Fax:** T.38 fax is not currently supported in the Axtel network. Fax using G.711 pass-through was tested instead, but it behaved unreliably, with results varying depending on the specific carrier where the remote fax machine was subscribed. It is recommended not to use fax with this solution.
- **Calls from twinned mobile phones to idle-appearance FNE:** This feature can be used to provide dial tone from the IP Office to provisioned twinned mobile devices (e.g. cell phone). For this feature to work, it requires the number entered on the **Twinned Mobile** field on the host extension to exactly match the CLI of the device making the incoming call to the IP Office.

The dial plan used for local calls in Monterrey, Mexico, requires 8 digit dialing for outbound calls, without the area code (81), but incoming calls to the IP Office show the complete 10 digit number, including the area code, in the From header of the arriving INVITE. This disparity in the outbound and inbound numbering plans conflicts with the IP Office FNE functionality. Further digit manipulation would be required on the Axtel network if the full implementation of this feature is required.

2.3. Support

For technical support on the Axtel SIP Trunk Service offer, visit www.axtel.mx.

3. Reference Configuration

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

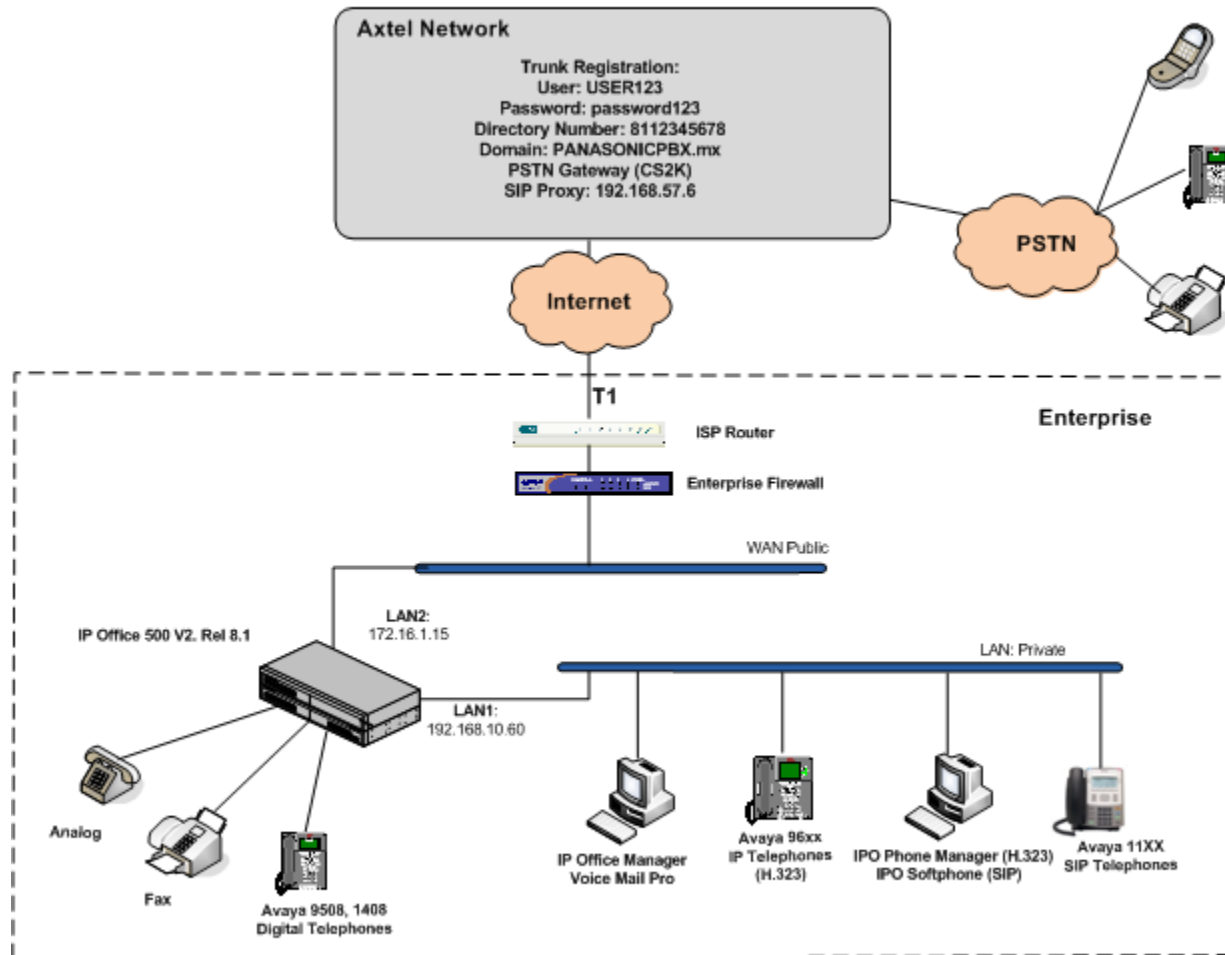


Figure 1: Test Configuration

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. Also the credentials used for the trunk registration are fictitious, and not the ones used during the compliance tests.

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.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to make calls across the SIP trunk to Axtel. The short code 9 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to Axtel. For outbound local calls the user dialed 9 followed by eight digits. Other calls, like mobile phones, Toll Free, long distance, international, etc., use different number lengths, and should be provisioned accordingly in the IP Office, with their respective entries on the Short Code or ARS forms.

For inbound calls, Axtel sent to the IP Office only the last 4 digits of the 10 digit Directory Number in the Request URI and the To headers of inbound INVITE messages, and the complete 10 digit number of the calling party, including the area code, in the From header.

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 all SIP and RTP traffic between the service provider and the Avaya IP Office system must be allowed to pass through these devices.

4. Equipment and Software Validated

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

Component	Version
Avaya	
Avaya IP Office 500v2	8.1 (43)
Avaya IP Office Digital Expansion Module DCPx16	10.1 (43)
Avaya IP Office Manager	10.1 (43)
Avaya IP Office Voicemail Pro	8.1.810.0
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)	04.03.12.00
Avaya Digital Telephone 1408	16
Avaya Digital Phone 9508	0.39
Avaya IP Office Softphone	3.2.3.20.64770
Avaya IP Office Phone Manager	4.2.42
Servicio Troncal SIP de Axtel	
Nortel CS2K	12.0.0.6

5. Configure IP Office

This section describes the Avaya IP Office configuration necessary to support connectivity to the Axtel 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 these 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.

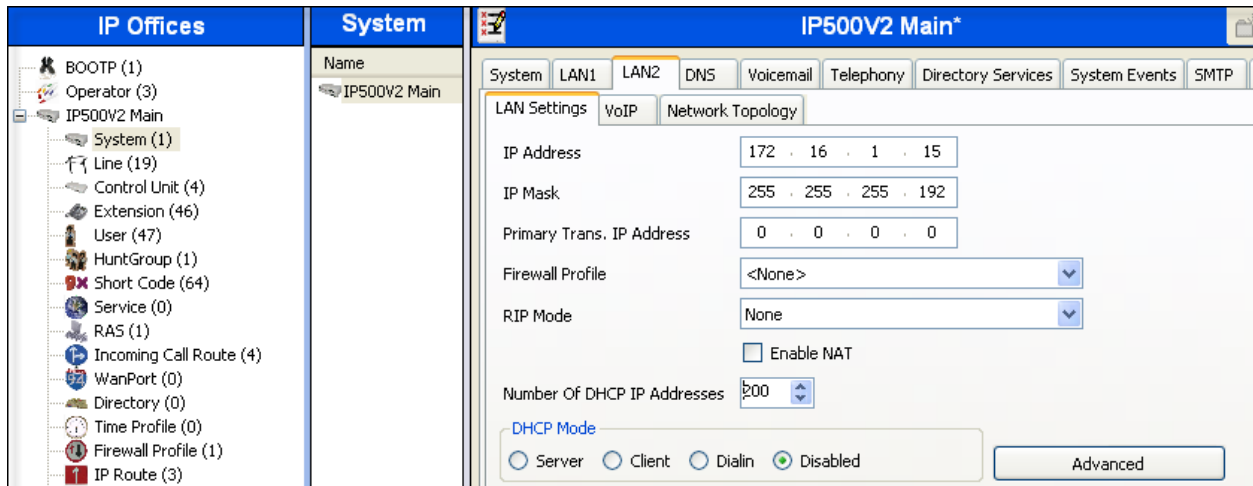
To verify that there is a SIP Trunk Channels License with sufficient capacity, click **License** in the Navigation pane and **SIP Trunk Channels** in the Group pane. Confirm that there is a valid license with sufficient “Instances” (trunk channels) in the Details pane. Note that the actual License Key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

The screenshot displays the Avaya IP Office configuration interface. It is divided into three main panes: IP Offices, License, and SIP Trunk Channels.

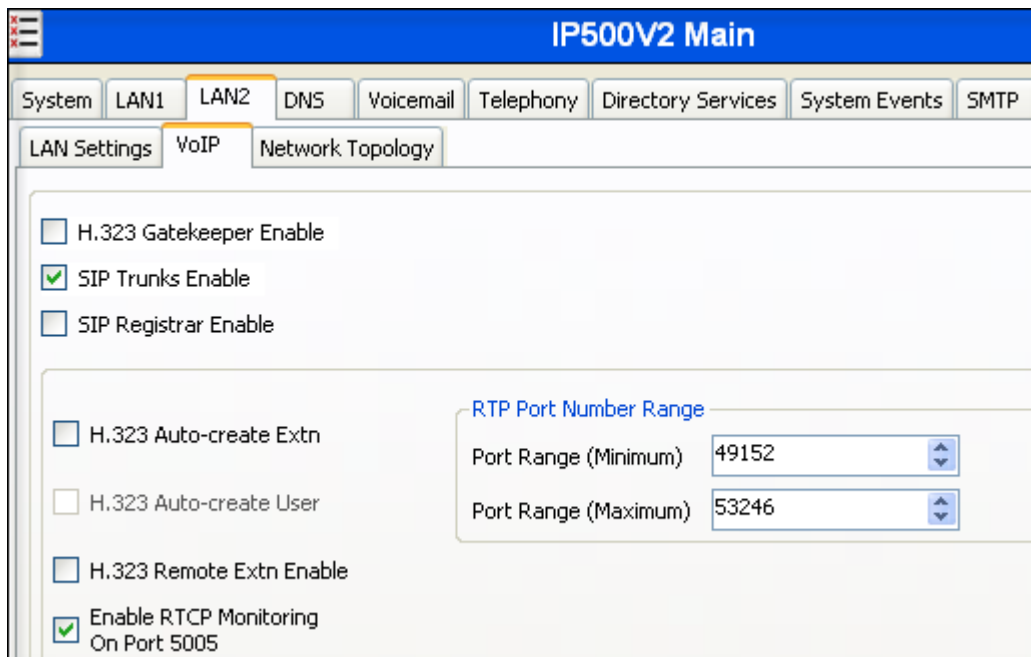
- IP Offices:** A tree view showing various system components such as BOOTP (4), Operator (3), IP500V2 Main, System (1), Line (19), Control Unit (5), Extension (45), User (47), HuntGroup (1), Short Code (63), Service (0), RAS (1), Incoming Call Route (4), WanPort (0), Directory (0), Time Profile (0), Firewall Profile (1), IP Route (3), Account Code (0), License (74), Tunnel (0), User Rights (8), and ARS (1).
- License:** A list of license types including IP500 Voice Networkin, IP5ec Tunnelling, Microsoft CRM Integra, Mobile Worker, Mobility Features, Office Worker, one-X Portal for IP Off, Phone Manager Pro, Phone Manager Pro (p), Phone Manager Pro IP, Power User, Preferred Edition (Voic), Preferred Edition Addit, Preferred/Advanced to, Proactive Reporting, RAS LRQ Support (Rap), Receptionist, Report Viewer, SIP Trunk Channels, Small Office Edition VC, and Small Office Edition Wil.
- SIP Trunk Channels:** A details pane for the selected license type. It shows the following information:
 - License Key: XnMtHe6b9DXL42M [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.



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.



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.

The screenshot displays the configuration interface for Avaya IP Office, divided into three sections:

- DiffServ Settings:** This section contains six spinners for configuring DSCP values. The top row shows: DSCP(Hex) set to 88, DSCP Mask (Hex) set to FC, and SIG DSCP (Hex) set to 88. The bottom row shows: DSCP set to 46, DSCP Mask set to 63, and SIG DSCP set to 34.
- DHCP Settings:** This section includes: Primary Site Specific Option Number (SSON) set to 176; Secondary Site Specific Option Number (SSON) set to 242; VLAN set to Not Present; 1100 Voice VLAN Site Specific Option Number (SSON) set to 232; and 1100 Voice VLAN IDs set to an empty text field.
- IPO-SNet Firewall:** A dropdown menu is set to <None>.
- RTP Keepalives:** This section includes: Scope set to Disabled with a Periodic timeout of 0; and Initial keepalives set to Enabled.

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. Since no network address translation (NAT) was used in the compliance test, 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 with the 'Network Topology' tab selected. The 'Network Topology Discovery' section contains the following fields and values:

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		
Public Port	5060		

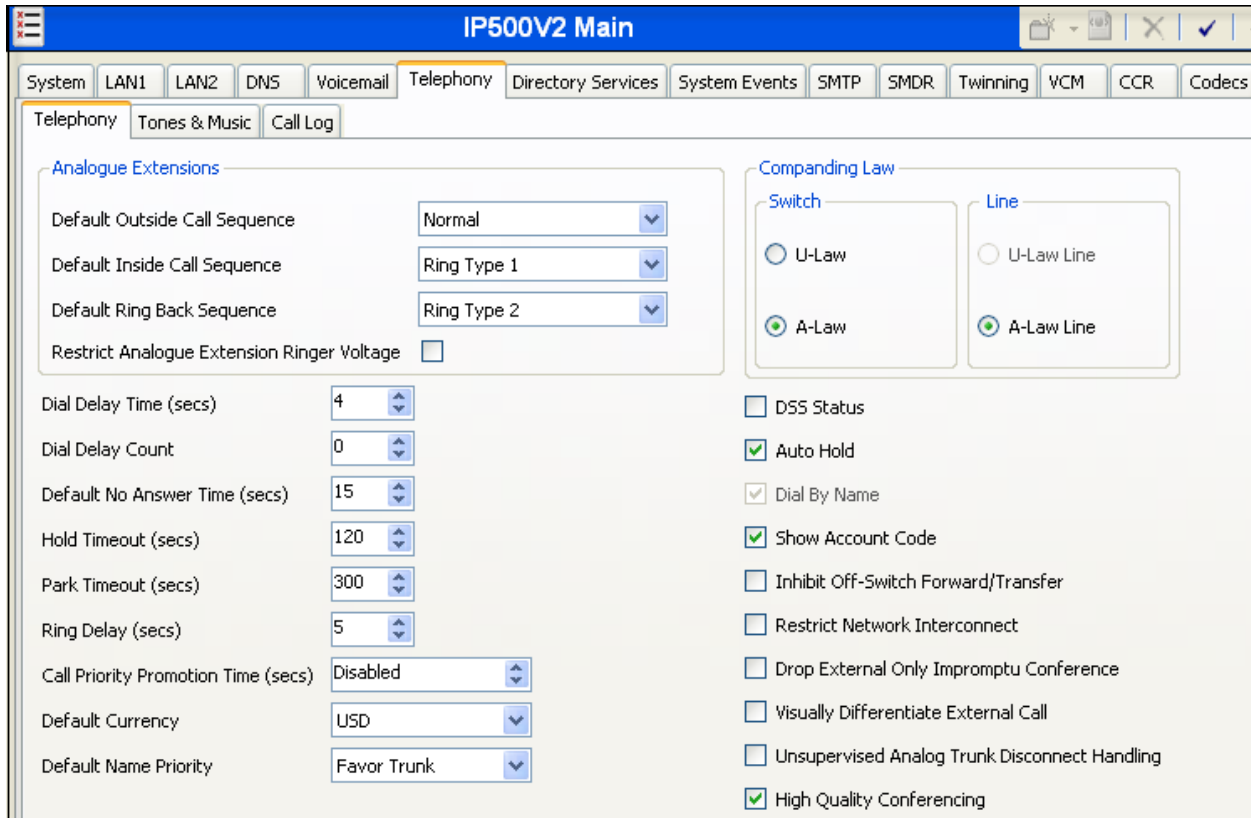
Buttons: Run STUN, Cancel

Run STUN on startup

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

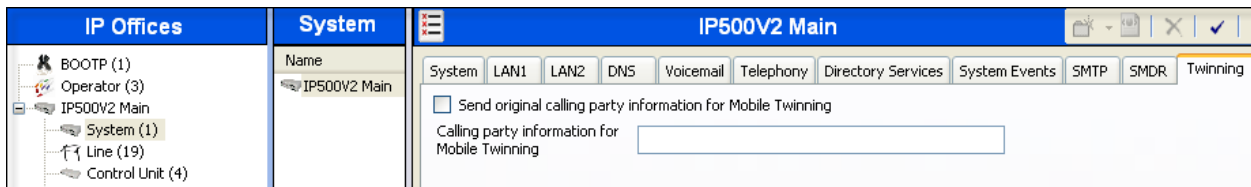
5.3. System Telephony Settings

Navigate to the **Telephony** → **Telephony** Tab in the Details Pane. Choose the **Companding Law** typical for the enterprise location. In Mexico, **A-Law** 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.



5.4. Twinning Calling Party Settings

Navigate to the **Twining** tab on the Details Pane. Uncheck the **Send original calling party information for Mobile Twining** box. This will allow the Caller ID for Twining to be controlled by the setting on the SIP Line (**Section 5.6**). This setting also impacts the Caller ID for call forwarding.



5.5. 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 subnet where the SIP Proxy is located on the Axtel network. On the left navigation pane, right-click on **IP Route** and select **New**.

- Set the **IP Address** and **IP Mask** of the remote Axtel 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.

Field	Value
IP Address	192 . 168 . 57 . 0
IP Mask	255 . 255 . 255 . 0
Gateway IP Address	172 . 16 . 1 . 254
Destination	LAN2
Metric	0
Proxy ARP	<input type="checkbox"/>

5.6. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the Axtel SIP Trunk Service. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New**→ **SIP Line**.

5.6.1. SIP Line Tab

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*.
- Check **Caller ID from From header**. This field is new in IP Office 8.1. It allows to choose the header used as the source for caller ID information on incoming calls.
- Set **Send Caller ID** to *None*.
- Uncheck the **REFER support** box. IP Office will not send REFER headers for calls that are transferred back to the PSTN. This was necessary as a workaround to the problem found with Blind Call Transfer described in **Section 2.2**.
- Set **UPDATE Supported** to *Auto*.
- Default values may be used for all other parameters.

IP Offices	Line	SIP Line - Line 17			
<ul style="list-style-type: none"> BOOTP (1) Operator (3) IP500V2 Main System (1) Line (19) Control Unit (4) Extension (46) User (47) HuntGroup (1) Short Code (64) Service (0) RAS (1) Incoming Call Rout 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) RAS Location Requ E911 System (1) 	<ul style="list-style-type: none"> Line Number 1 2 17 201 202 203 204 205 206 207 208 209 210 211 212 213 214 215 216 	<ul style="list-style-type: none"> SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials 	<ul style="list-style-type: none"> Line Number: 17 ITSP Domain Name: PANASONICPBX.mx Prefix: National Prefix: 0 Country Code: International Prefix: 00 Send Caller ID: None Association Method: By Source IP address REFER Support: <input type="checkbox"/> Incoming: Always Outgoing: Always UPDATE Supported: Auto 	<ul style="list-style-type: none"> 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 Caller ID from From header: <input checked="" type="checkbox"/> Send From In Clear: <input type="checkbox"/> User-Agent and Server Headers: 	

5.6.2. Transport Tab

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

- Set the **ITSP Proxy Address** to the IP address of the Axtel proxy server.
- Set the **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to **LAN2** as configured in **Section 5.2**.
- Set the **Send Port** to **5060**.
- Default values may be used for all other parameters.

The screenshot shows a configuration window with several tabs: SIP Line, Transport (selected), SIP URI, VoIP, T38 Fax, and SIP Credentials. The Transport tab is active, displaying the following settings:

- ITSP Proxy Address: 192.168.57.6
- Network Configuration section:
 - Layer 4 Protocol: UDP (dropdown menu)
 - Send Port: 5060 (spin box)
 - Use Network Topology Info: LAN 2 (dropdown menu)
 - Listen Port: 5060 (spin box)
- Explicit DNS Server(s): 0 . 0 . 0 . 0 (two input fields)
- Calls Route via Registrar:
- Separate Registrar: (empty text input field)

5.6.3. SIP URI Tab

SIP URIs entries need to be created to match each incoming number that Avaya IP Office and Axtel will accept on this line. For the compliance test, and since Axtel used different numbering formats for inbound and outbound calls, two different SIP URI entries were needed, to be used for calls travelling in each direction. 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 incoming calls, a single SIP URI entry was created with the wildcard * in the **Local URI**, **Contact** and **Display** fields. With this setting the IP Office will accept any incoming call arriving on the SIP trunk. The routing of the call will still be performed by the system Incoming Call Route that matched the 4 digit number sent to the IP Office by Axtel. This number corresponds to the last 4 digits of the Directory Number assigned to the IP Office. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact** and **Display Name** to the wildcard *.
- 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. For the compliance test, a new incoming 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.

The screenshot shows the configuration interface for a SIP URI. The 'SIP URI' tab is active. Below the tab, there is a table with columns: Channel, Groups, Via, Local URI, Contact, Display Name, PAI, Credential, and Max Calls. To the right of the table are buttons for 'Add...', 'Remove', and 'Edit...'. Below the table is the 'Edit Channel' dialog box. The fields in the dialog are: Via (172.16.1.15), Local URI (*), Contact (*), Display Name (*), PAI (None), Registration (0: <None>), Incoming Group (17), Outgoing Group (0), and Max Calls per Channel (5). To the right of the dialog are 'OK' and 'Cancel' buttons.

Axtel required every outbound call made from the enterprise to have the complete 10 digit Directory Number as the user part of the SIP URI of the From and Contact headers in all messages entering their network. A second entry was created with the following parameters:

- Set **Local URI**, **Contact** and **Display Name** to the 10 digit Directory Number assigned to the IP Office.
- Associate this line with an outgoing line group by entering a line group number in 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 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.

The screenshot shows a configuration window with tabs for SIP Line, Transport, SIP URI, VoIP, T38 Fax, and SIP Credentials. The SIP URI tab is active. At the top, there is a table with columns: Channel, Groups, Via, Local URI, Contact, Display Name, PAI, Credential, and Max Calls. The table contains one row with values: 1, 17 0, 6..., *, *, *, N..., 0: <Non..., 5. To the right of the table are buttons for Add..., Remove, and Edit... Below the table is a 'New Channel' section with the following fields:

Via	172.16.1.15
Local URI	8112345678
Contact	8112345678
Display Name	8112345678
PAI	None
Registration	0: <None>
Incoming Group	0
Outgoing Group	17
Max Calls per Channel	5

At the bottom right of the 'New Channel' section are buttons for OK and Cancel.

5.6.4. VoIP Tab

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. The buttons allow setting an explicit list of codecs to be used on the line, in that specific order of preference.
- Since fax was not reliable during testing, it is recommended to disable fax by selecting **None** under **Fax Transport Support**. However, if fax is to be used, select **G711**.
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Check the **VoIP Silence Suppression** box, to enable the use in the IP Office of codec G729B, which is the codec preferred by Axtel.
- 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 Axtel.
- Default values may be used for all other parameters.

The screenshot shows the configuration interface for the VoIP tab. At the top, there are tabs for 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'VoIP' tab is active. The 'Codec Selection' dropdown is set to 'Custom'. Below this, there are two lists: 'Unused' and 'Selected'. The 'Unused' list contains 'G.723.1 6K3 MP-MLQ'. The 'Selected' list contains 'G.729(a) 8K CS-ACELP', 'G.711 ALAW 64K', and 'G.711 ULAW 64K'. Between the lists are buttons for moving items: '>>', '<<', '<', '>', and '>>'. Below the lists, there are several configuration fields: 'Fax Transport Support' is set to 'None', 'Call Initiation Timeout (s)' is set to 4, and 'DTMF Support' is set to 'RFC2833'. On the right side, there are five checkboxes: 'VoIP Silence Suppression' (checked), 'Re-invite Supported' (checked), 'Use Offerer's Preferred Codec' (unchecked), 'Codec Lockdown' (unchecked), and 'PRACK/100rel Supported' (checked).

Since T.38 Fax will not be used with this solution, the **T38 Fax** tab will not be visited.

5.6.5. SIP Credentials Tab

SIP credentials must be created for the registration of the SIP trunk to the Axtel network.

To create a SIP Credentials entry, first select the **SIP Credentials** tab in the Detail pane. 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 value provided by the service provider.
- Leave the **Registration required** box checked.
- Leave the **Expiry** field with the default value of **60** minutes. The actual registration expiration time is negotiated and agreed upon as part of the registration exchange.
- Click **OK**.

The screenshot shows a dialog box titled "New SIP Credentials". It contains the following fields and controls:

- User name:** Text input field containing "USER123".
- Authentication Name:** Text input field containing "USER123".
- Contact:** Empty text input field.
- Password:** Text input field containing "*****".
- Expiry (mins):** Spin box containing the value "60".
- Registration required:** A checked checkbox.
- Buttons:** "OK" and "Cancel" buttons are located on the right side of the dialog.

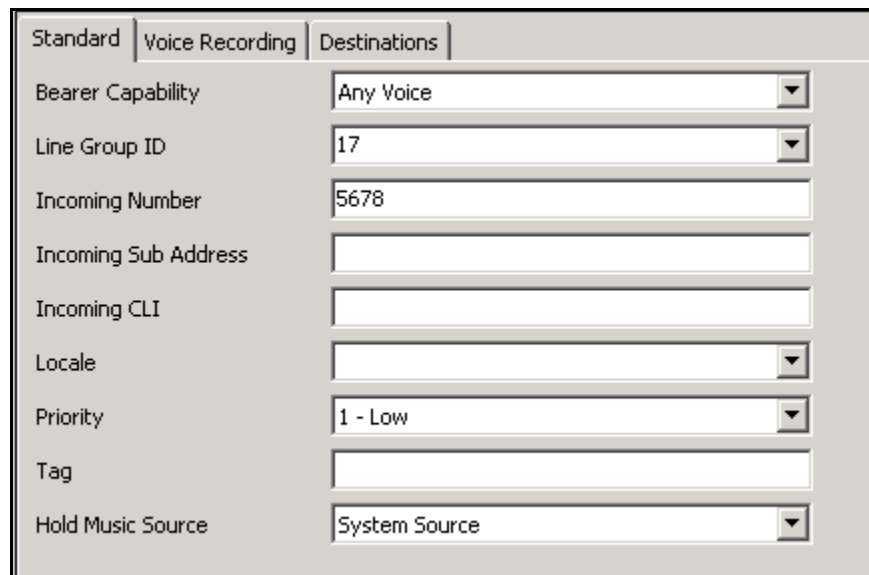
5.7. Incoming Call Route

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, where only one Directory Number was assigned to the IP office by the service provider, only one incoming route was needed.

From the left Navigation Pane, right-click on **Incoming Call Route**. Select **New**. On the Details Pane, under the **Standard** tab, set the parameters as shown below:

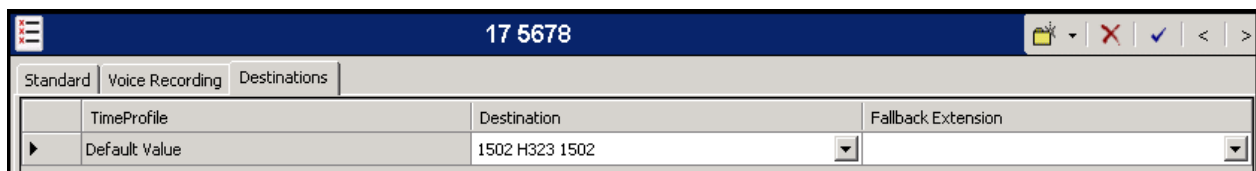
- Set **Bearer Capacity** to *Any Voice*.
- Set the **Line Group ID** to the incoming line group of the SIP line defined in **Section 5.6.3**.
- Set the **Incoming Number** to the last 4 digits of the Directory Number.
- Default values may be used for all other parameters.



The screenshot shows the configuration dialog for an Incoming Call Route, with the 'Standard' tab selected. The fields are as follows:

Field	Value
Bearer Capacity	Any Voice
Line Group ID	17
Incoming Number	5678
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. Click the **OK** button (not shown). In this example, incoming calls to 5678 on line 17 are routed to extension 1502.



The screenshot shows the 'Destinations' tab of the configuration dialog. The title bar indicates '17 5678'. The table below shows the destination configuration:

TimeProfile	Destination	Fallback Extension
Default Value	1502 H323 1502	

5.8. Short Code

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"@PANASONICPBX.mx"**. 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 Axtel 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 5.6.3**. This short code will use this line group when placing outbound calls.
- Default values may be used for all other parameters.

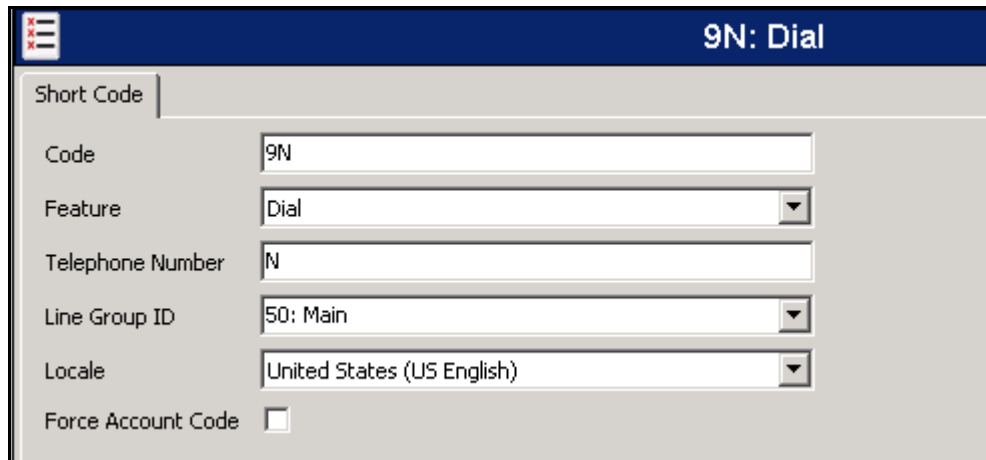
The screenshot displays the Avaya SIP Line configuration interface. On the left, the 'IP Offices' navigation pane shows a tree structure with 'Short Code (64)' selected. The main area is divided into two panes: 'Short Code' and '8N;; Dial'. The 'Short Code' pane shows a list of short codes from *00 to *09. The '8N;; Dial' pane shows the configuration for a short code with the following fields:

Field	Value
Code	8N;
Feature	Dial
Telephone Number	N"@PANASONICPBX.mx"
Line Group ID	17
Locale	United States (US English)
Force Account Code	<input type="checkbox"/>

5.9. Automatic Route Selection

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. Other features like time-based routing criteria 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.



The screenshot shows a configuration window titled "9N: Dial". The window has a tab labeled "Short Code". The configuration fields are as follows:

Field	Value
Code	9N
Feature	Dial
Telephone Number	N
Line Group ID	50: Main
Locale	United States (US English)
Force Account Code	<input type="checkbox"/>

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. This type of setting results in a much quicker response in the delivery of the call by the IP Office. For example, during the compliance tests, the user dialed 9, then 8 digit numbers starting with a 1 for local calls. Also note that the Axtel test domain name does not need to be specified in the **Telephone Number** column 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 5.6.1**.

The screenshot displays the ARS configuration for the 'Main' route. The left sidebar shows a tree view of IP Office components, with 'ARS (1)' selected. The main configuration area includes the following settings:

- ARS Route Id: 50
- Route Name: Main
- Dial Delay Time: System Default (4)
- In Service: (Out of Service Route: <None>)
- Time Profile: <None> (Out of Hours Route: <None>)
- Secondary Dial tone: (SystemTone)
- Check User Call Barring:

Below the settings is a table of ARS entries:

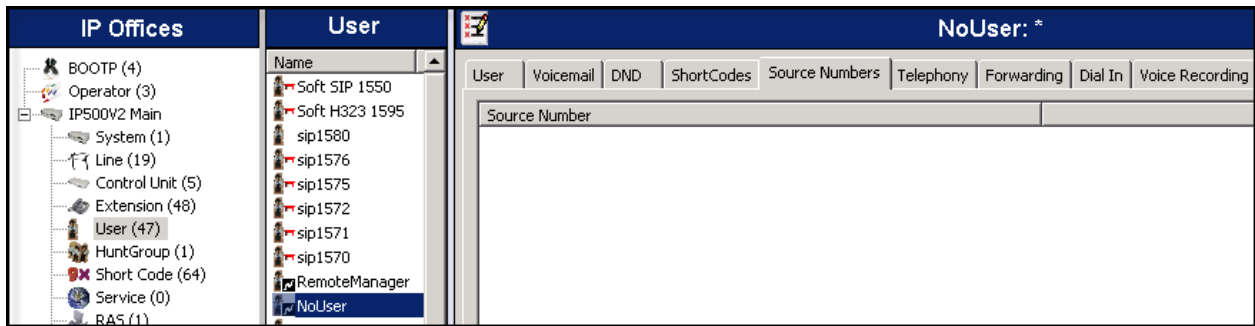
Code	Telephone Number	Feature	Line Group ID
11	911	Dial Emergency	17
911	911	Dial Emergency	17
8XXXXXXXX	8N	Dial	17
1XXXXXXXX	1N	Dial	17
0XXXXXXXXXXXX	0N	Dial	17
411	411	Dial	17
611	611	Dial	17

Buttons for 'Add...', 'Remove', and 'Edit...' are located to the right of the table.

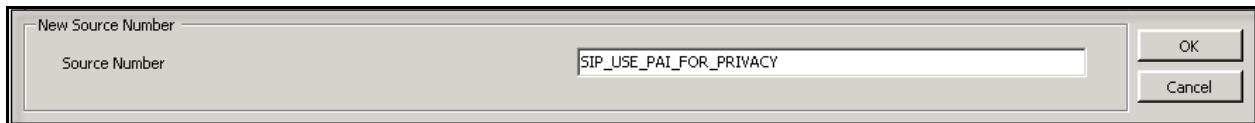
5.10. Privacy/Anonymous Calls

For outbound calls with privacy (anonymous) enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with “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.

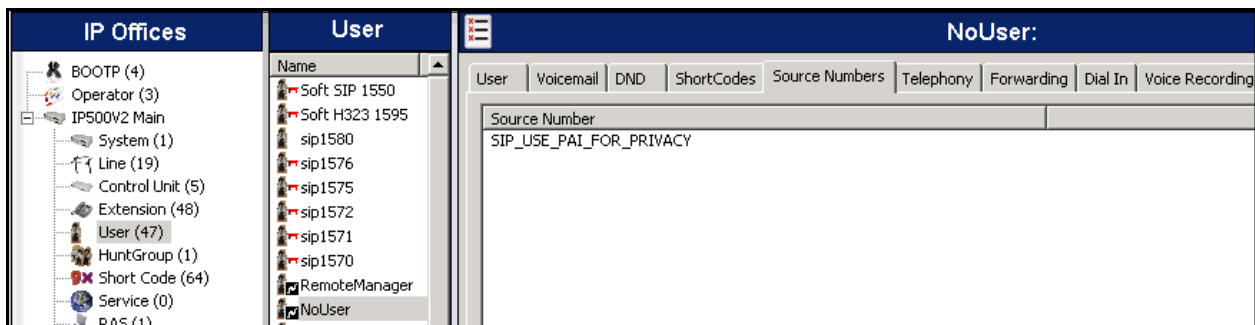
To configure Avaya IP Office to use PAI for privacy calls, navigate to **User** → **NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



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.



5.11. 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. Axtel SIP Trunking Configuration

Axtel is responsible for the configuration of the Axtel SIP Trunk Service. Axtel will provide the customer the necessary information to configure the Avaya IP Office SIP trunk connection to Axtel, including:

- IP address of the Axtel SIP Proxy server.
- Axtel SIP domain name.
- Supported codecs and order of preference.
- Directory Number.
- 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).

IP Office R8.1 System Status - IP500V2 Main (192.168.10.60) - IP500 V2 8.1 (43)

AVAYA IP Office System Status

Help Snapshot LogOff Exit About

- System
- Alarms (0)
- Extensions (0)
- Trunks (3)
 - Line: 1
 - Line: 2
 - Line: 3
- Active Calls (0)
- Resources (0)
- Voicemail (0)
- IP Network (0)

SIP Trunk Summary

Peer Domain Name: PANASONICPBX.mx
 Resolved Address: 192.168.57.6
 Line Number: 17
 Number of Administered Channels: 10
 Number of Channels in Use: 0
 Administered Compression: G729 A, G711 A, G711 Mu
 Silence Suppression: On
 SIP Trunk Channel Licenses: Unlimited
 SIP Trunk Channel Licenses in Use: 0 0%
 SIP Device Features:

Channel Number	U...	Call Ref	Current State	Time in State	Remote Media ...	Co...	Conn...	Caller ID or ...	Other Party on Call	Directi...	Round Trip D...	Receive Jitter	Receive Packe...	Transmi t Jitter	Trans...
1			Idle	08:46...											
2			Idle	08:46...											
3			Idle	08:46...											
4			Idle	08:46...											
5			Idle	08:46...											
6			Idle	08:46...											
7			Idle	08:46...											

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

Alarms for Line: 17 SIP PANASONICPBX.mx

Last Date Of Error	Occurrences	Error Description

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

Registration Status

Index	User Name	Status	Retry Time
1	USER123	Registered	6/11/2012 11:14:03 AM

- 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 8.1 and the Axtel SIP Trunk Service, as shown in **Figure 1**.

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 8.1 Installation Manual*, Document Number 15-601042, August 2012
- [2] *IP Office Manager Manual 10.0*, Document Number 15-601011, August 2012
- [3] *IP Office Release 8.1 Implementing Voicemail Pro*, Document Number 15-601064, June, 2012
- [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 the Axtel SIP Trunk Service is available from Axtel.

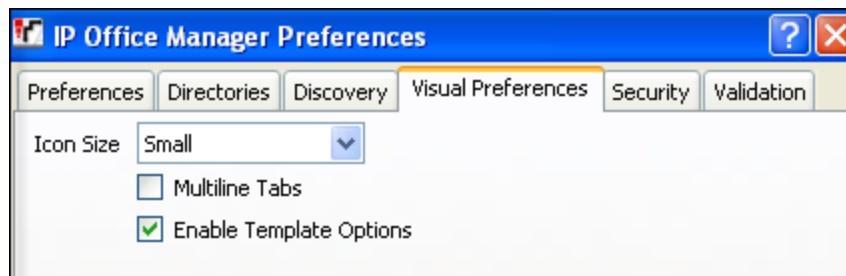
Appendix: SIP Line Template

Avaya IP Office Release 8.1 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.

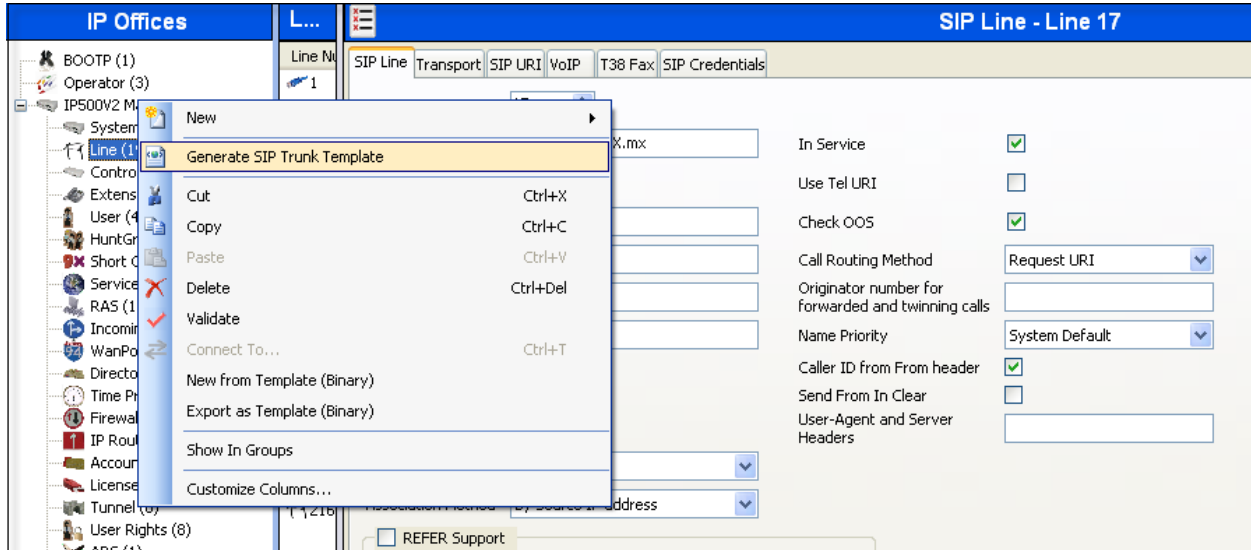
Not all of the configuration information is included in the SIP Line Template, therefore, it is critical that the SIP Line configuration be verified/updated after a template has been imported, and additional configuration be supplemented using **Section 5.6** in these Application Notes as a reference.

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

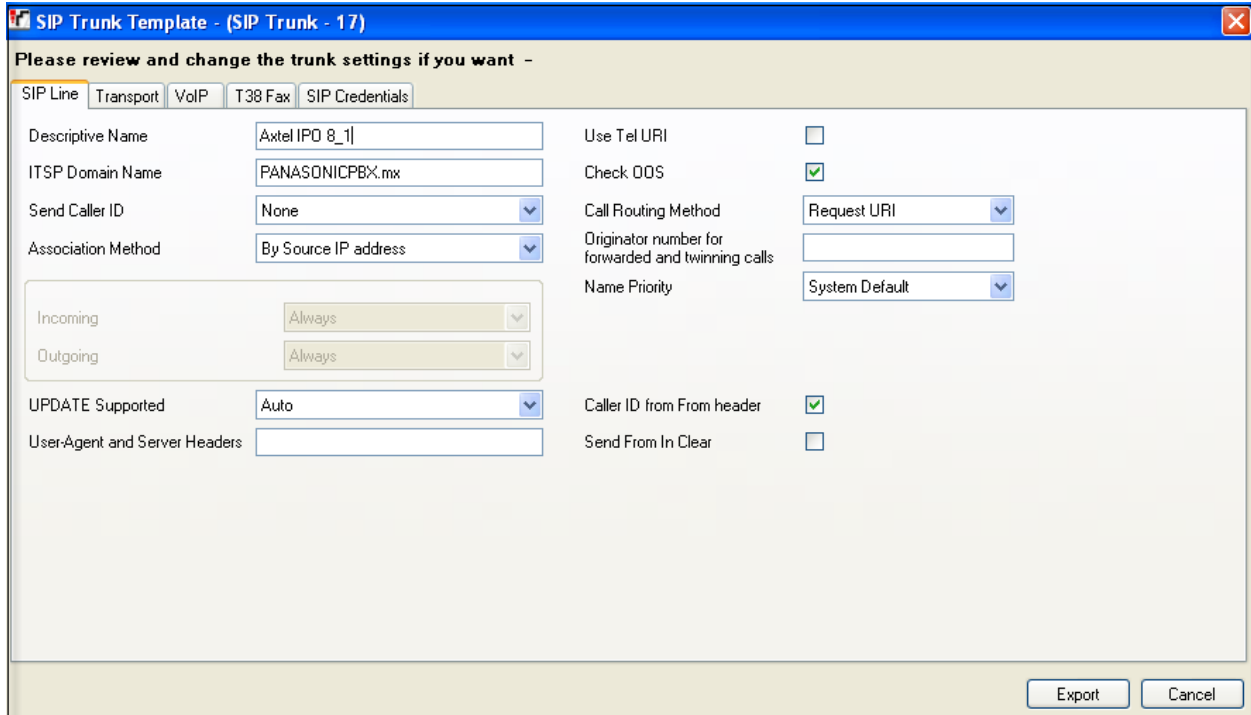
To enable template support, select **File**, then **Preferences**. On the **Visual Preferences** tab, check the **Enable Template Options** box.



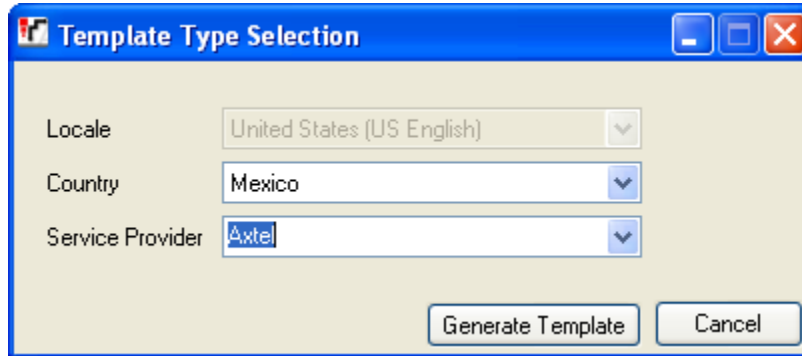
To create a SIP Line Template from the configuration, on the left Navigation Pane, right click the Sip Line (17), and select **Generate SIP Trunk Template**.



The trunk's settings are displayed as configured in **Section 5.6**. 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 **MX_Axtel_SIPTrunk.xml**:

```

<?xml version="1.0" encoding="utf-8" ?>
<Template xmlns="urn:SIPTrunk-schema">
<TemplateType>SIPTrunk</TemplateType>
<Version>20120825</Version>
<SystemLocale>enu</SystemLocale>
<DescriptiveName>Axtel IPO 8_1</DescriptiveName>
<ITSPDomainName>PANASONICPBX.mx</ITSPDomainName>
<SendCallerID>CallerIDNone</SendCallerID>
<ReferSupport>false</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>true</CallerIDfromFromheader>
<PerformUserLevelPrivacy>false</PerformUserLevelPrivacy>
<ITSPProxy>192.168.57.6</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>

```

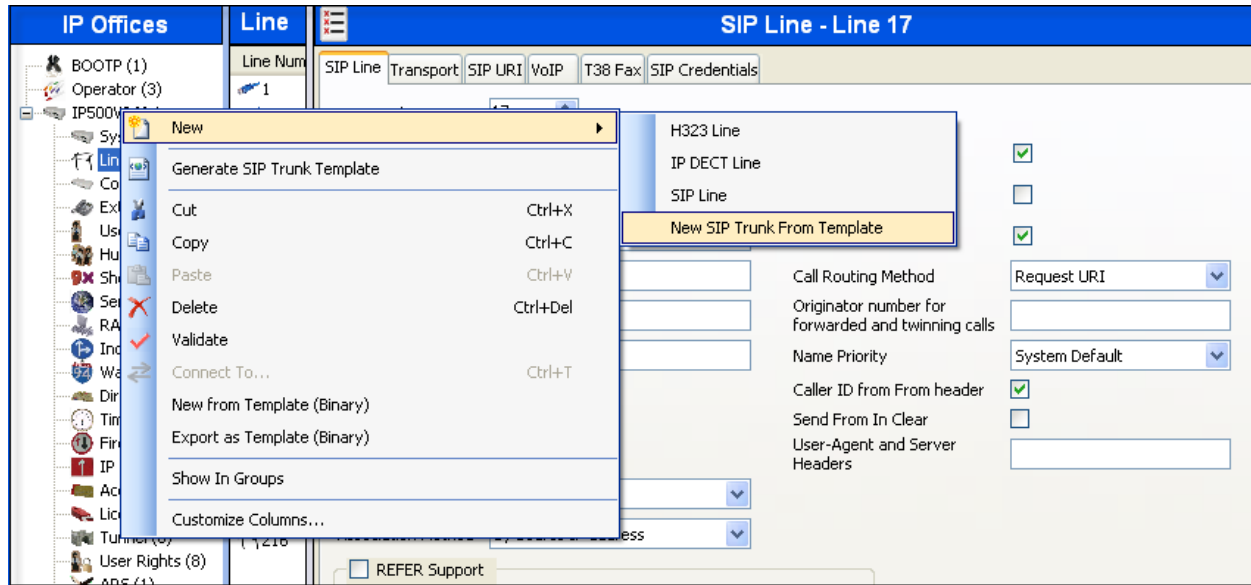
```

<UseAdvVoiceCodecPrefs>true</UseAdvVoiceCodecPrefs>
<AdvCodecPref>G.729(a) 8K CS-ACELP,G.711 ALAW 64K,G.711 ULAW
  64K</AdvCodecPref>
<CallInitiationTimeout>4</CallInitiationTimeout>
<DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
<VoipSilenceSupression>true</VoipSilenceSupression>
<ReinviteSupported>true</ReinviteSupported>
<FaxTransportSupport>FOIP_NONE</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>

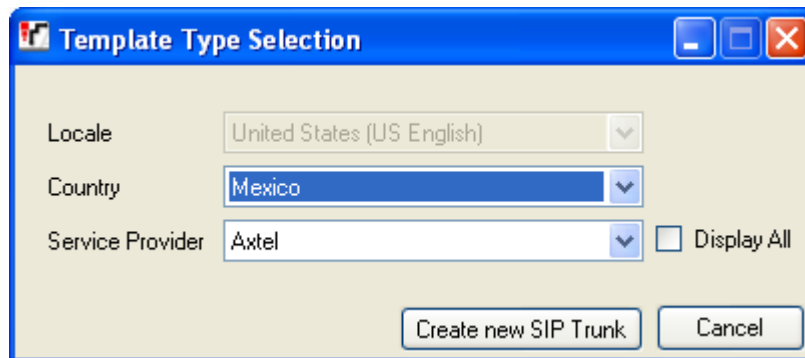
```

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.



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