



Application Notes for Configuring Alestra Enlace IP SIP Trunk Service (Broadsoft platform) with Avaya IP Office 9.0 using SIP Trunk Registration– Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between the Alestra Enlace IP SIP Trunk service on the Broadsoft platform and Avaya IP Office 9.0, using SIP Trunk Registration.

The Enlace IP SIP Trunk service offered by Alestra provides customers with PSTN access via a SIP trunk between the enterprise and the Alestra 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 Alestra Enlace IP SIP Trunk service on the Broadsoft platform and Avaya IP Office 9.0, using SIP Trunk Registration.

In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 9.0, Avaya Voicemail Pro, Avaya IP Office soft clients and deskphones, including SIP, H.323, digital, and analog endpoints.

The Alestra Enlace IP SIP Trunk service referenced within these Application Notes is designed for enterprise business customers in Mexico. Customers using this service with the Avaya SIP-enabled enterprise solution are able to place and receive PSTN calls via a broadband WAN connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as analog and/or ISDN-PRI.

2. General Test Approach and Test Results

A simulated enterprise site containing the Avaya IP Office equipment was installed at the Avaya Solution and Interoperability Lab. The enterprise site was configured to connect to the Alestra Enlace IP SIP Trunk service via a broadband connection to 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.

Testing was performed with IP Office 500 v2 R9.0, but it also applies to IP Office Server Edition R9.0. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R9.0 to support analog or digital endpoints or trunks.

2.1. Interoperability Compliance Testing

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

- Response to SIP OPTIONS queries.
- Incoming PSTN calls to 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 Avaya IP Office Softphone.
- Inbound and outbound PSTN calls to/from Avaya Flare® Experience for Windows softphone.
- Various call types including: local, long distance, international, outbound toll-free, etc.
- Codecs G729A and G.711A.
- Fax using T.38 and G711 pass-through methods.
- 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 transfer, call forwarding and twinning.

Items not supported or not tested included the following:

- Inbound toll-free and emergency calls are supported but were not tested as part of the compliance test
- Operator services such as dialing 0 or 0 + 10 digits are not supported.

2.2. Test Results

Interoperability testing of the Alestra Enlace IP SIP trunk service was completed with successful results for all test cases with the exception of the observations and limitations described below:

- **Caller ID on incoming calls from the U.S.:** Calls originating from PSTN telephones in the U.S. to DID numbers in Mexico assigned to the IP Office SIP trunk displayed a caller ID “anonymous” on the enterprise extensions. The “From” header on the INVITE of these incoming calls was “anonymous@anonymous.invalid”. This seems to be a PSTN restriction for all international inbound calls to Mexico from the U.S., not limited just to Alestra. This behavior is not necessarily indicative of a limitation of the combined Alestra/Avaya solution, and it is listed here simply as an observation.
- **No matching codec on outbound call:** On an outbound call containing only one codec in its SDP offer that was not supported by the service provider, Alestra responded sending back a “480 Temporarily Unavailable” error code, instead of the expected “488 Not Acceptable Here”. There is no impact to the user, who hears an error tone.
- **Caller ID on call forward to the PSTN:** On inbound calls that were forwarded back out to another user on the PSTN, the caller ID number shown on the receiving end was the DID number assigned to the SIP trunk, not the caller ID of the call originator.
- **Outbound Calling Party Number (CPN) Block:** When an enterprise user activated “Withhold Number” on an outbound call, IP Office sent “anonymous” in the From header, included the “Privacy:id” header, and the complete DID number in the P-Asserted-Identity (PAI) header of the outbound INVITE, as expected. In this scenario, Alestra returned a “404 Not Found” and calls failed, suggesting that the PAI header is not inspected for authentication purposes in the Alestra network.
- **T.38 Fax:** Testing of T38 fax while interworking with the G.729A codec for the voice setup was successful. During testing of the T.38 fax using codec G.711 for the voice setup, Alestra returned a “488 Not Acceptable Here” error code to the T.38 re-INVITE from IP Office using SDP with T.38 parameters and the fax calls dropped. Since G729A voice setup is the normal mode of operation in this solution, this limitation has no real impact to the user.
Additionally, inbound and outbound fax calls using the G.711 pass-through Fax Transport Support mode were tested successfully.
- **Response to OPTIONS:** During the compliance test, Alestra responded to OPTIONS messages sent from the IP Office with a “405 Method Not Allowed” message. Since the OPTIONS messages were used to check the status of the network connectivity to the service provider, any response received from Alestra was sufficient to achieve that purpose.

2.3. Support

For technical support on the Alestra Enlace IP SIP Trunk service offer, visit <http://www.alestra.com.mx/>

3. Reference Configuration

Figure 1 illustrates the sample Avaya SIP-enabled enterprise solution, connected to the Alestra Enlace IP SIP Trunk service through a public Internet WAN connection.

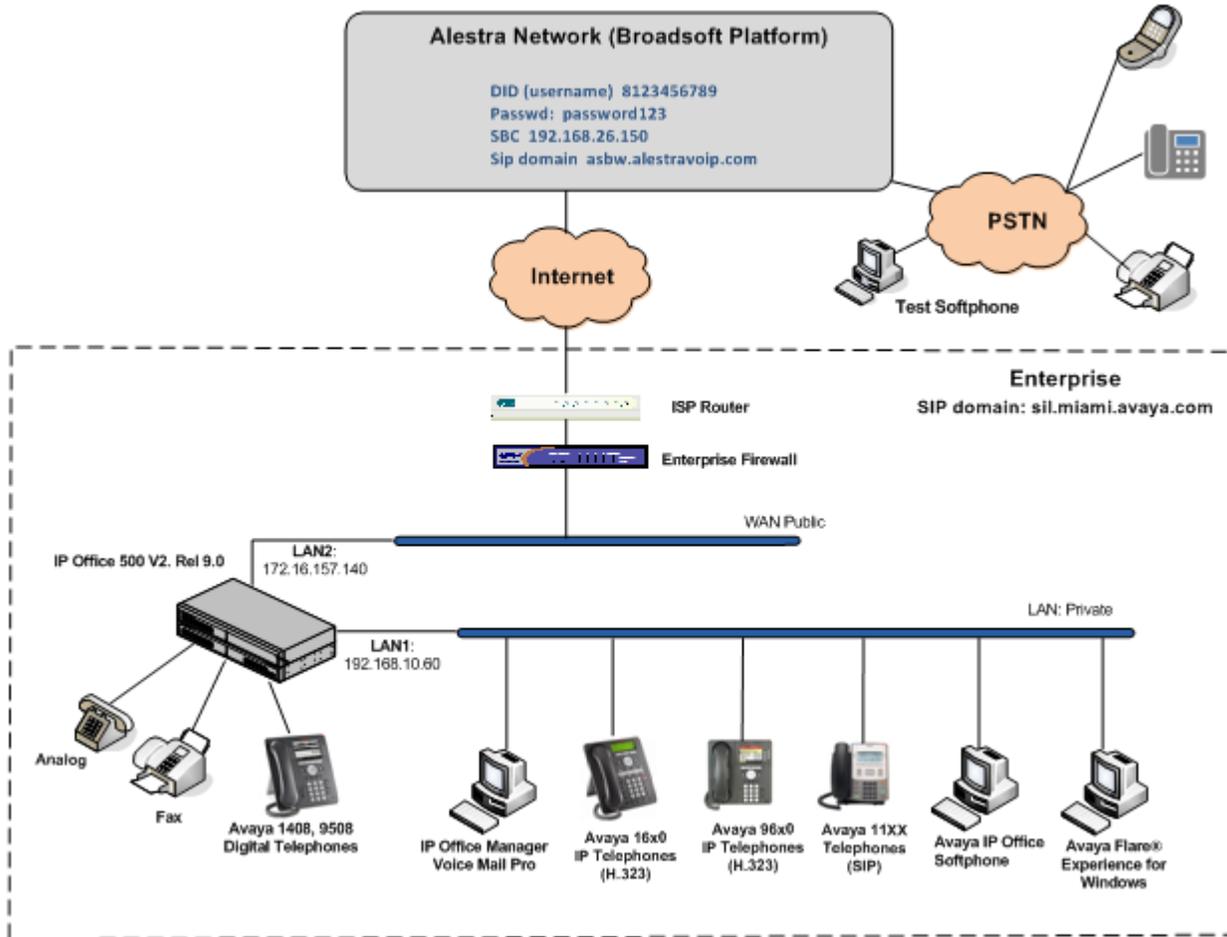


Figure 1: Test Configuration

Note that for security purposes all public IP addresses of the network elements, public PSTN numbers and SIP trunk credentials shown throughout these Application Notes have been edited so the actual values 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 1600 and 9600 Series IP Telephones (with H.323 firmware), Avaya 1140E IP Telephones (with SIP firmware), Avaya 1408 and 9508D Digital Telephones, analog telephones and PCs running Avaya IP Office Softphone and Avaya Flare® Experience for Windows. The site also has a Windows 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 all SIP and RTP traffic between the service provider and the Avaya IP Office system must be allowed to pass through these devices.

During the compliance test, in addition to the DID number assigned to the SIP trunk, Alestra provided a local test number in Monterrey, Mexico. A SIP-based softphone was registered to this local PSTN number and was used to originate and terminate local calls to and from the PSTN to the enterprise

4. Equipment and Software Validated

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

Component	Version
Avaya	
Avaya IP Office 500v2	9.0.200.860
Avaya IP Office Digital Expansion Module DCPx16	9.0.200.860
Avaya IP Office Manager	9.0.2.0.Build 860
Avaya IP Office Voicemail Pro	9.0.2.0.41
Avaya 1608 IP Telephone (H.323)	1.3 SP3
Avaya 9640 IP Telephone (H.323)	Avaya one-X Deskphone Edition 3.2
Avaya 1140E IP Telephone (SIP)	04.03.18.00
Avaya Digital Telephone 1408	32.0
Avaya Digital Phone 9508	0.45
Avaya IP Office Softphone	3.2.3.49.68975
Avaya IP Office Flare Experience for Windows	1.1.4.23
Alestra	
Broadsoft Softswitch	Release 17 SP 4
Acme Packet SBC	V6.2
Lucent 5ESS	V16.1

5. Configure IP Office

This section describes the Avaya IP Office configuration necessary to support connectivity to the Alestra Enlace IP SIP Trunk service. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the PC running Avaya IP Office Manager, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration** (not shown), 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 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.

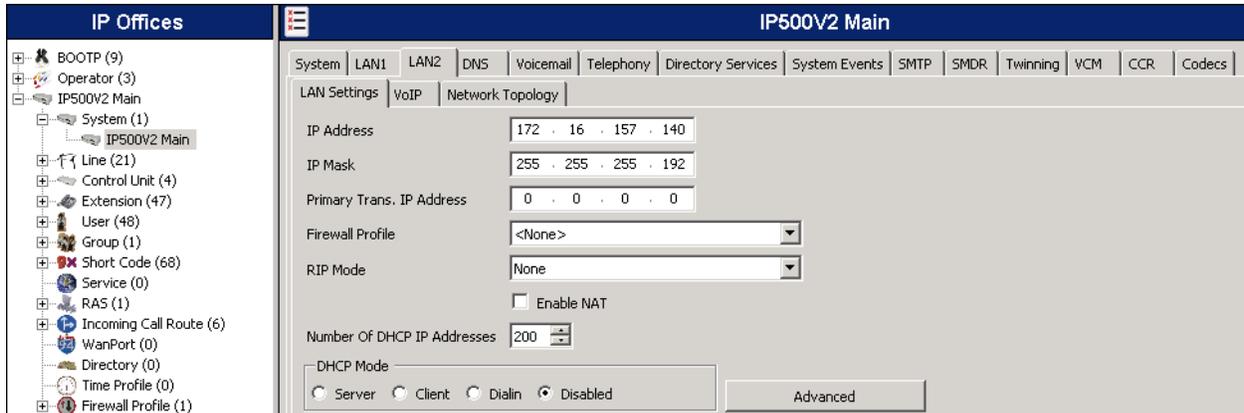
To verify that there is a SIP Trunk Channels License with sufficient capacity; click **License** in the Navigation pane. Confirm that there is a valid license with sufficient “Instances” (trunk channels) in the Details pane.

The screenshot displays the IP Office configuration interface. On the left, the 'IP Offices' navigation pane shows a tree structure with 'License (62)' selected. The main area shows the 'License' tab for a 'Remote Server' with PLDS Host ID 11131. A table lists various licenses with columns for Feature, License Key, Instances, Status, and Expiry Date. The 'SIP Trunk Channels' license is highlighted with a red border, showing 255 instances, a status of 'Valid', and an expiry date of 'Never'.

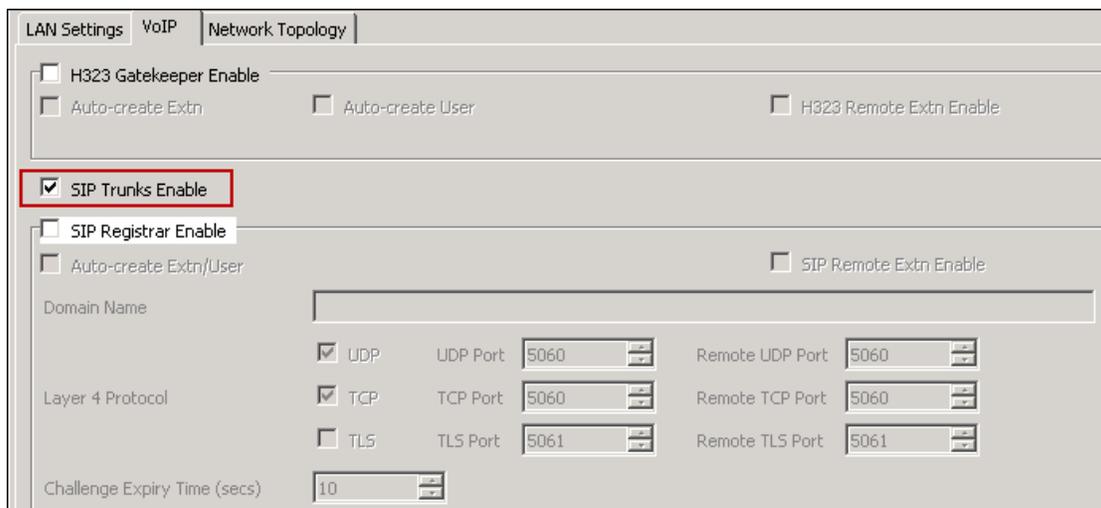
Feature	License Key	Instances	Status	Expiry Date
Unused (1)	tG2gHnVn5Dp@KFYnQ...	255	Valid	Never
CTI Link Pro	BGcmbEmoMw9qOir0KF...	255	Valid	Never
SIP Trunk Channels	XnMtHe6b9DXL42wonZ...	255	Valid	Never
3rd Party IP Endpoints	4Iz6hzyW5s2_N52y19...	255	Valid	Never
IP500 Universal PRI (Additional chan...)	1nmDJR6QVIp0OuLpRI...	255	Valid	Never
1600 Series Phones	Lq0rJz@4LdfJaPoEga_...	255	Valid	Never
Wave User	BUzsrB1tA@5UIRxx0...	255	Valid	Never
Integrated Messaging	ctWs3ctzLgt7hg_Dfnu...	255	Valid	Never
Preferred Edition (Voicemail Pro)	@I1nk@_MXenhA7n...	255	Valid	Never
Microsoft CRM Integration (users)	QIz5fo@xDxzQG9BQH...	255	Valid	Never
CCC Spectrum Wallboards	BINbEdjgKYU38XHGm...	255	Valid	Never
Phone Manager Pro	z3CTz1v8PN5hxxTYnx...	255	Valid	Never
Phone Manager Pro IP Audio Enabled...	XT9Inj62MVU6guxeUx...	255	Valid	Never
Compact Business Centre	Ma53qeoKgjYUKOJX1G...	255	Valid	Never
CCC Server	@h13H09dXvfmv9756...	255	Valid	Never
CCC PC Wallboards	eaIR_qtPEvwJsQE2Boc...	255	Valid	Never
CCC Supervisors	z3Wx6Vt1gVxjaPeyWv...	255	Valid	Never
Receptionist	ZXOnH3tmMXwAspMM...	255	Valid	Never
eBLF	gX@YhT6WdOVHaxL2K...	255	Valid	Never
Preferred Edition Additional Voicemail ...	ZIu6WY5LLNy7F2fUw...	255	Valid	Never
AUDIX Voicemail	ATQuh2BBdAd@5r8ev...	255	Valid	Never
VMPPro Networked Messaging	bTM30Fd1DvUJxy6IA2...	255	Valid	Never
VMPPro Database Interface	ja9y6Nv8QGdZUIm0Qi...	255	Valid	Never
VMPPro VB Script	dqVgn5ynLV7BF7rC4eu...	255	Valid	Never
VMPPro Recordings Administrators	A3csyyBvdO02dZ9yRk...	255	Valid	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 pane and select 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.

All other parameters should be set according to customer requirements.

The screenshot displays the configuration interface for Avaya IP Office, specifically the VoIP settings under the Network Topology tab. The interface is divided into several sections:

- RTP Section:**
 - Port Number Range:** Minimum is set to 49152 and Maximum is set to 53246.
 - Port Number Range (NAT):** Minimum is set to 49152 and Maximum is set to 53246.
 - Enable RTCP Monitoring on Port 5005**
 - Keepalives:**
 - Scope: Disabled
 - Periodic timeout: 0
 - Initial keepalives: Enabled
- DiffServ Settings Section:**
 - DSCP (Hex): B8
 - Video DSCP (Hex): B8
 - DSCP Mask (Hex): FC
 - SIG DSCP (Hex): 88
 - DSCP: 46
 - Video DSCP: 46
 - DSCP Mask: 63
 - SIG DSCP: 34

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 **300**. 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.
- Under **Public Port**, set **UDP** to **5060**.
- All other parameters should be set according to customer requirements.

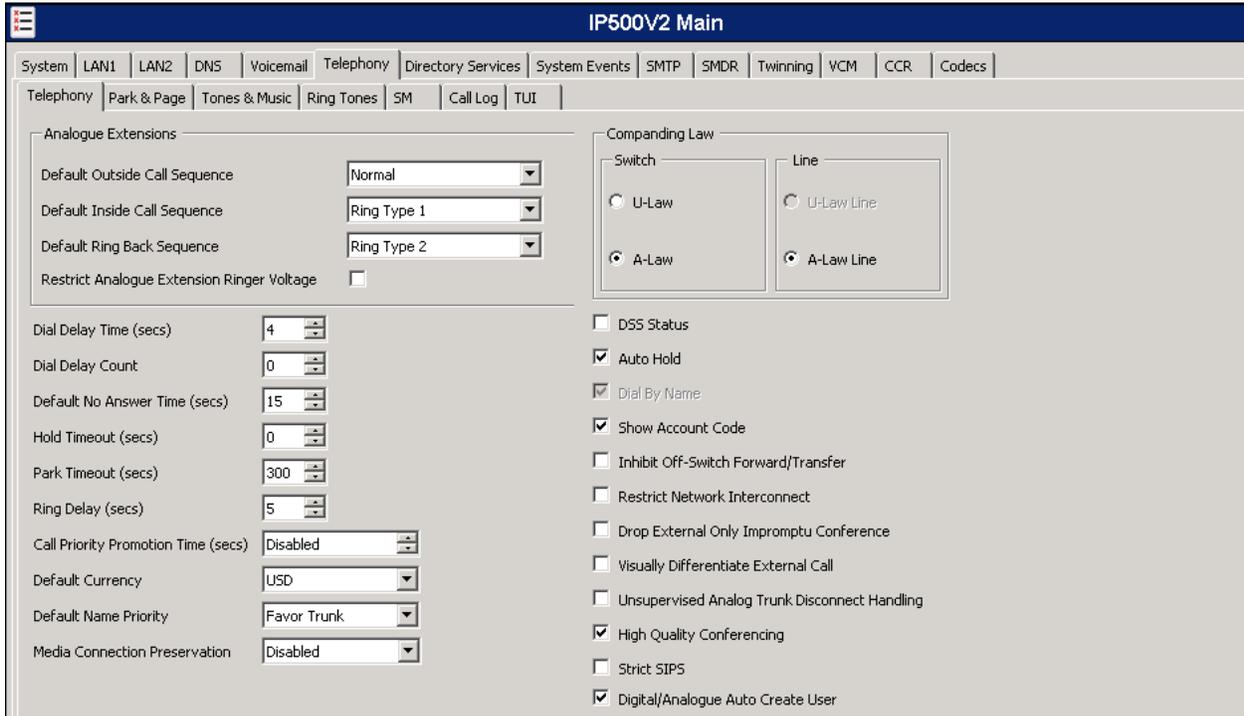
The screenshot shows the 'Network Topology' configuration window. It includes the following fields and values:

- STUN Server Address: 69.90.168.13
- STUN Port: 3478
- Firewall/NAT Type: Open Internet
- Binding Refresh Time (seconds): 300
- Public IP Address: 172 . 16 . 157 . 140
- Public Port: UDP (5060), TCP (0), TLS (0)
- 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 Alestra SIP trunk, and therefore is not described in these Application Notes.

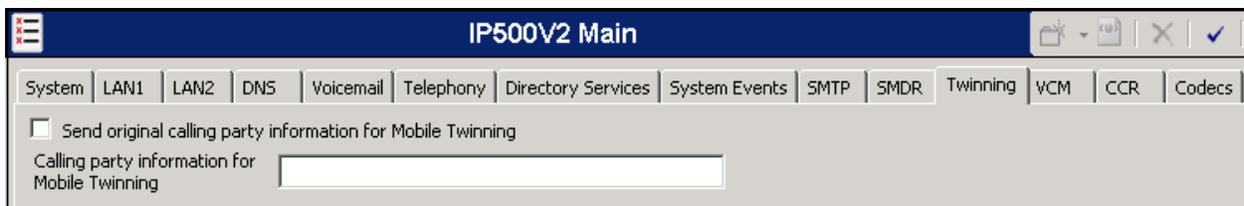
5.3. System Telephony Settings

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.



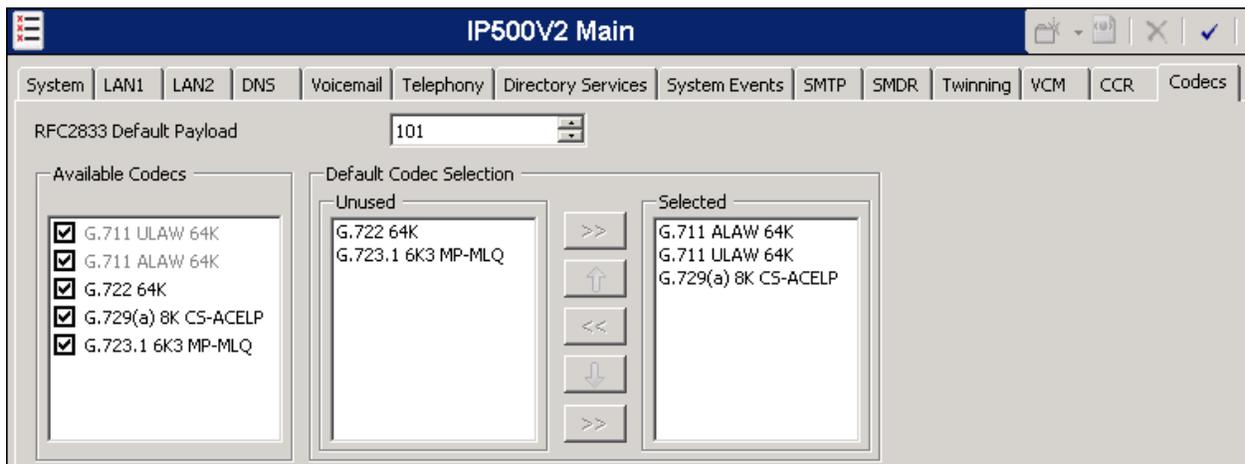
5.4. Twinning Calling Party Settings

Navigate to the **Twining** tab on the Details Pane. Uncheck the **Send original calling party information for Mobile Twinning** box. This will allow the Caller ID for Twinning 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. System Codecs Settings

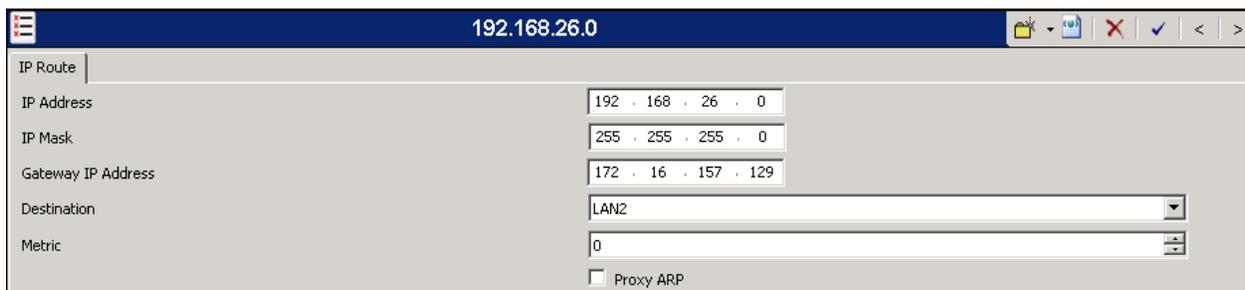
Navigate to the **Codecs** tab in the Details Pane. The **RFC2833 Default Payload** field is new in IP Office release 9.0. It allows the manual configuration of the payload type used on SIP calls that are initiated by the IP Office. The default value **101** was used. The list of **Available Codecs** 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 **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.



5.6. 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 Alestra network. On the left navigation pane, right-click on **IP Route**. Select **New** (not shown).

- Set the **IP Address** and **IP Mask** of the remote subnet of the Alestra SIP Proxy.
- 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.



5.7. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the Alestra SIP Trunking service. This line will carry outbound and inbound traffic between to and from the service provider. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.7.1** to create the SIP Line from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:

- IP addresses
- SIP Credentials (if applicable)
- SIP URI entries
- Setting of the **Use Network Topology Info** field on the Transport tab.

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.7.2 – 5.7.5**.

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

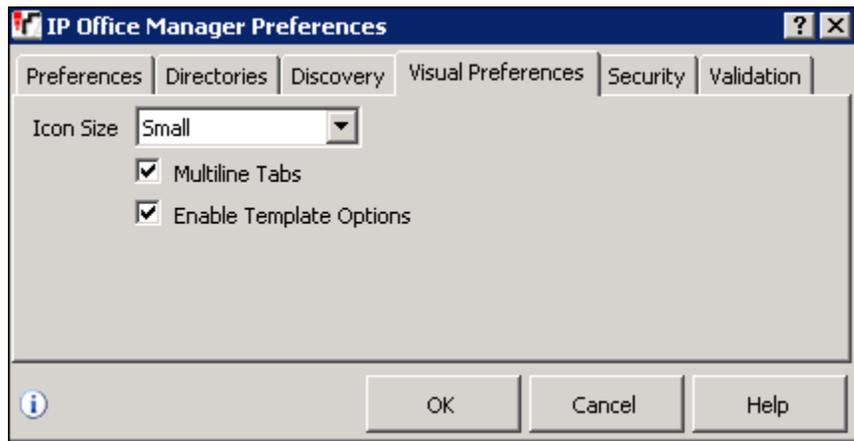
- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required

Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.7.2 – 5.7.6**.

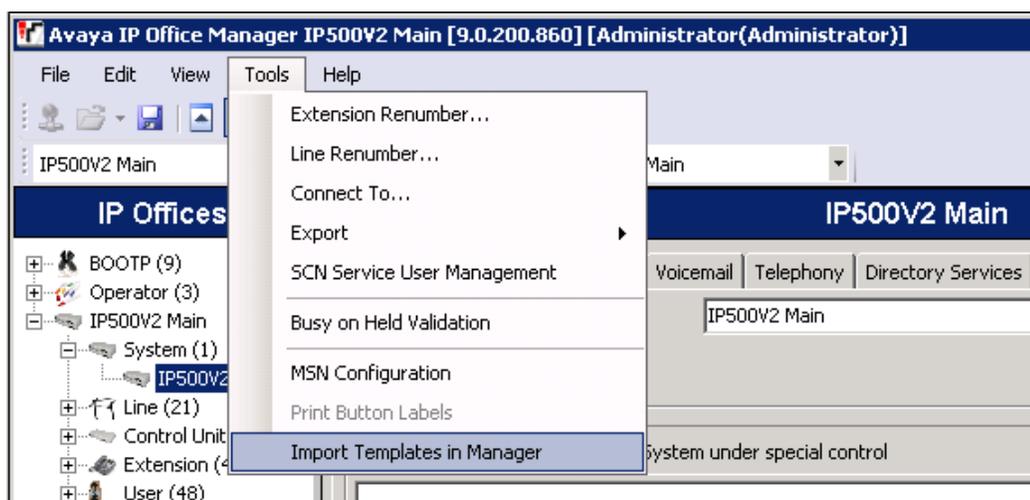
5.7.1. SIP Line From Template

Complete the following steps to create a SIP Line from the template associated with these Application Notes:

1. Copy the template file to the computer where IP Office Manager is installed. Rename the template file to **MX_Alestra_SIPTrunk.xml**. The file name is important in locating the proper template file in **Step 5**.
2. Verify that template options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the IP Office Manager Preferences window that appears, select the **Visual Preferences** tab. Verify that the box is checked next to **Enable Template Options**. Click **OK**.

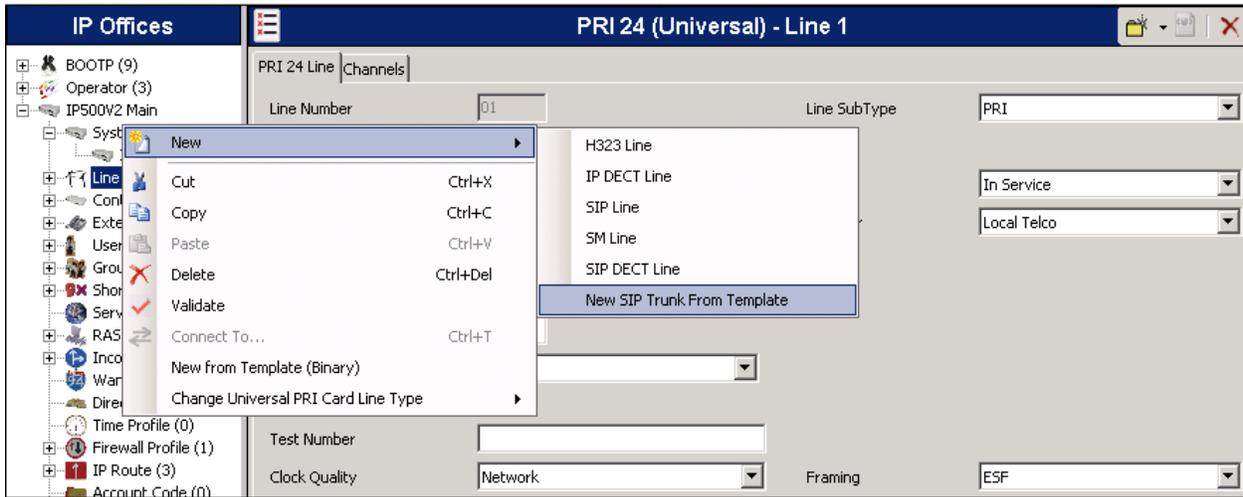


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default template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



In the pop-up window (not shown) that appears, select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window (not shown) will appear stating success or failure. Click **OK** (not shown) to continue. If preferred, this step may be skipped if the template file is copied directly to the IP Office template directory.

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



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



6. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.7.2 – 5.7.5**.

5.7.2. SIP Line Tab

On the **SIP Line** tab in the Details Pane, configure (or verify) the parameters as shown below:

- Set the **ITSP Domain Name** to the domain known and expected by Alestra on the SIP trunk. IP Office will use this domain as the host portion of the SIP URI of SIP headers in messages sent to the network.
- Check the **In Service** box.
- Check the **Check OOS** box. With this option selected, IP Office will check the responses to SIP OPTIONS messages sent to the service provider to determine the operational status of the SIP Line.
- Set **Call Routing Method** to **Request URI**.
- Set **Send Caller ID** to **None**. This field determines the method to be used to send the original calling party ID in scenarios of call forward to the PSTN and twinning. Selecting any other option (Diversion Header, Remote party ID, etc.) made these types of calls to be rejected by Alestra.
- Check the **REFER support** box. The REFER method was supported during the network call redirection of call transfer and call forward to the PSTN.
- Default values may be used for all other parameters.

The screenshot displays the configuration interface for a SIP Line in IP Office. The left pane shows a tree view of the system hierarchy, including BOOTP (9), Operator (3), IP500V2 Main, System (1), IP500V2 Main, and Line (21) with sub-entries 1 through 216, Control Unit (4), and Extension (46). The main pane is titled 'SIP Line - Line 17' and contains the following configuration fields:

Field	Value	Field	Value
Line Number	17	In Service	<input checked="" type="checkbox"/>
ITSP Domain Name	asbw.alestravoip.com	URI Type	SIP
Prefix		Check OOS	<input checked="" type="checkbox"/>
National Prefix	0	Call Routing Method	Request URI
Country Code		Originator number for forwarded and twinning calls	
International Prefix	00	Name Priority	System Default
Send Caller ID	None	Caller ID from From header	<input type="checkbox"/>
Association Method	By Source IP address	Send From In Clear	<input type="checkbox"/>
<input checked="" type="checkbox"/> REFER Support		User-Agent and Server Headers	
Incoming	Auto	Service Busy Response	486 - Busy Here
Outgoing	Auto	Action on CAC Location Limit	Allow Voicemail

5.7.3. Transport Tab

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

- Set the **ITSP Proxy Address** to the IP address of the Alestra SIP 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 the configuration interface for 'SIP Line - Line 17'. The 'Transport' tab is selected. The 'ITSP Proxy Address' is set to '192.168.26.150'. The 'Network Configuration' section includes 'Layer 4 Protocol' set to 'UDP', 'Send Port' set to '5060', and 'Use Network Topology Info' set to 'LAN 2'. The 'Listen Port' is also set to '5060'. The 'Explicit DNS Server(s)' are set to '0 . 0 . 0 . 0'. The 'Calls Route via Registrar' checkbox is checked. The 'Separate Registrar' field is empty.

5.7.4. SIP Credentials

SIP Credentials must be created when using SIP trunk registration with the service provider

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

- Set **User name** and **Authentication Name** to the value provided by the service provider. For the compliance test, the User name matched the DID number assigned by Alestra to the SIP trunk.
- Set **Password** to the value provided by the service provider.
- Set the **Expiry (mins)** field to **60**. Alestra required the registration to be renewed every 60 minutes. The registration expiration time is negotiated and agreed as part of the registration exchange.
- Check the **Registration required** box.
- Click **OK**.

The screenshot shows a software interface titled "SIP Line - Line 17". It features a tabbed menu with "SIP Line", "Transport", "SIP URI", "VoIP", "T38 Fax", and "SIP Credentials" selected. Below the tabs is a table with columns: "Index", "UserName", "Authentication Name", "Contact", "Expiry (mins)", and "Register". To the right of the table are buttons for "Add...", "Remove", and "Edit...". Below the table is a section titled "Edit SIP Credentials" containing the following fields:

User name	8123456789
Authentication Name	8123456789
Contact	8123456789
Password	*****
Expiry (mins)	60
Registration required	<input checked="" type="checkbox"/>

At the bottom right of the "Edit SIP Credentials" section are "OK" and "Cancel" buttons.

5.7.5. SIP URI Tab

A SIP URI entry needs to be created to match each number that Avaya IP Office and Alestra will accept on this line. Select the **SIP URI** tab, click the **Add** button and the **New Channel** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit...** button. In the example screen below, a previously configured entry is edited.

For the compliance test, a single SIP URI entry was created that matched the DID number assigned to the SIP trunk by Alestra. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact** and **Display Name** to *Use Credentials User Name*. Set **PAI** to *None*.
- Under **Registration**, select *1: <8123456789>* from the pull-down menu, which corresponds to the set of SIP credentials defined in the previous section.
- 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 calls to be allowed on the SIP trunk using this SIP URI pattern.

The screenshot shows the 'SIP Line - Line 17' configuration window. The 'SIP URI' tab is selected. A table lists the SIP URI entries:

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Cre
1	17 17	1...	81234...	8123...	8123456789	N...	1: E

Buttons for 'Add...', 'Remove', and 'Edit...' are visible. The 'Edit Channel' dialog is open, showing the following fields:

- Via: 172.16.157.140
- Local URI: Use Credentials User Name
- Contact: Use Credentials User Name
- Display Name: Use Credentials User Name
- PAI: None
- Registration: 1: 8123456789
- Incoming Group: 17
- Outgoing Group: 17
- Max Calls per Channel: 10

Buttons for 'OK' and 'Cancel' are also present.

5.7.6. 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 the specific order of preference for the codecs to be used on the line, as shown.
- Set **Fax Transport Support** to **T38**.
- 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 **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 Alestra.
- Default values may be used for all other parameters.

The screenshot shows the configuration interface for a SIP Line (Line 17) in the VoIP tab. The interface includes several sections:

- Codec Selection:** A dropdown menu is set to "Custom". Below it are two lists: "Unused" (containing G.722 64K and G.723.1 6K3 MP-MLQ) and "Selected" (containing G.729(a) 8K CS-ACELP, G.711 ALAW 64K, and G.711 ULAW 64K). Navigation buttons (right arrow, up arrow, left arrow, down arrow, and right arrow) are positioned between the lists.
- Fax Transport Support:** A dropdown menu set to "T38".
- Location:** A dropdown menu set to "Cloud".
- Call Initiation Timeout (s):** A numeric input field set to "4".
- DTMF Support:** A dropdown menu set to "RFC2833".
- Checkboxes:** On the right side, there are several checkboxes: "VoIP Silence Suppression" (unchecked), "Allow Direct Media Path" (unchecked), "Re-invite Supported" (checked), "Codec Lockdown" (unchecked), "PRACK/100rel Supported" (checked), "Force direct media with phones" (unchecked), and "G.711 Fax ECAN" (unchecked).

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

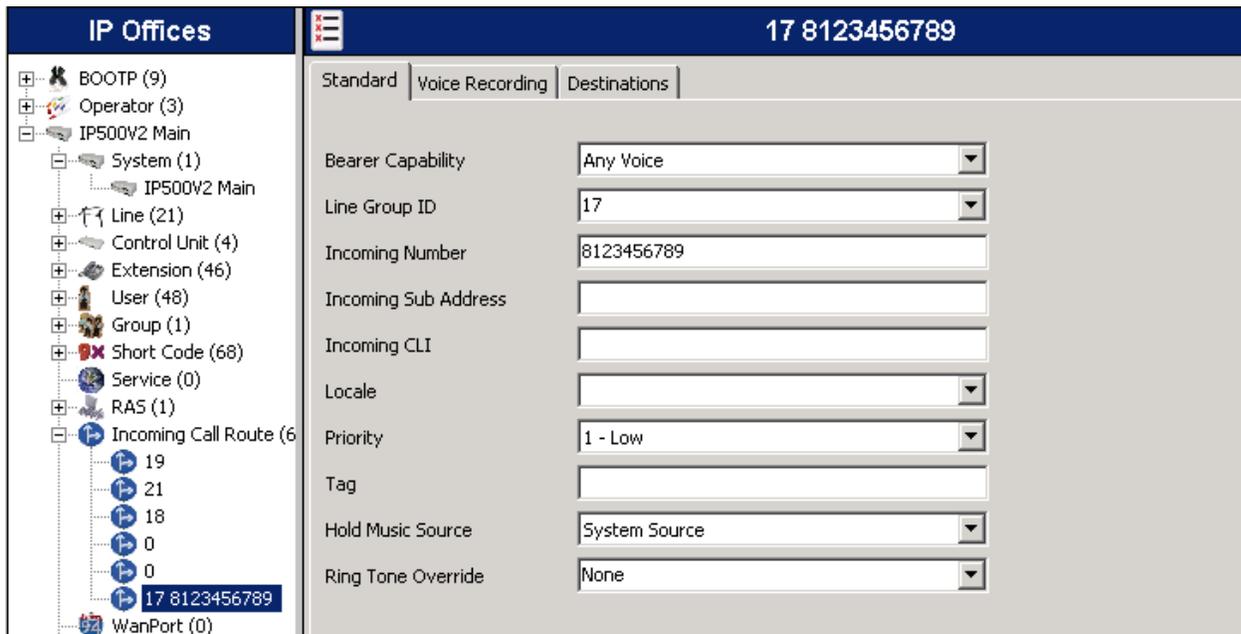
5.8. Incoming Call Route

Incoming call routes map inbound DID numbers on a specific line to internal extensions, hunt groups, short codes, etc, within the IP Office system. Incoming call routes are defined for each DID number assigned by the service provider.

To add a new incoming call route, from the left Navigation Pane, right-click on **Incoming Call Route** and select **New** (not shown). The screen below shows the route for the DID number assigned to the enterprise by Alestra, **8123456789** in this example.

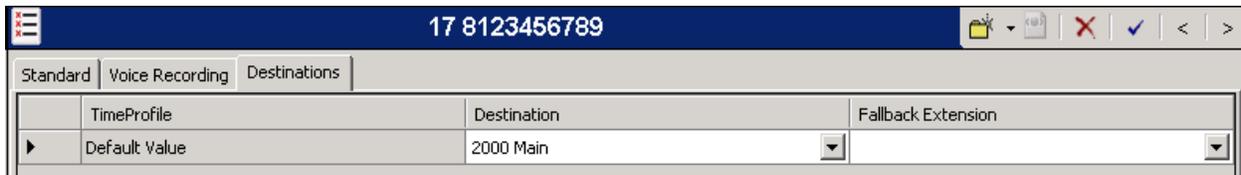
On the Details Pane, under the **Standard** tab, set the parameters as show below:

- Set **Bearer Capacity** to *Any Voice*.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.7**.
- Set the **Incoming Number** to the 10 digit DID number assigned by Alestra.
- Default values may be used for all other parameters.



Field	Value
Bearer Capability	Any Voice
Line Group ID	17
Incoming Number	8123456789
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

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 DID 8123456789 on line 17 are routed to hunt group **2000 Main**.



TimeProfile	Destination	Fallback Extension
Default Value	2000 Main	

5.9. 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** (not shown). On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

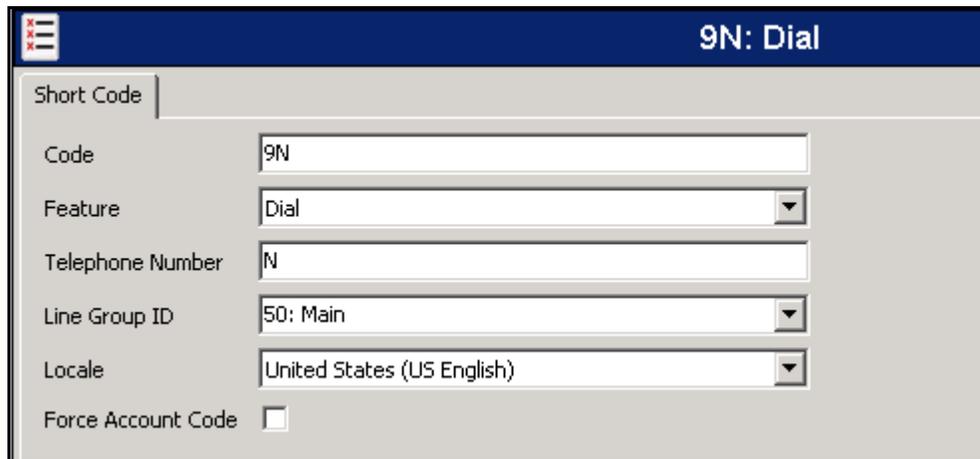
- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, **8N;**. This short code will be invoked when the user dials 8 followed by any number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N"@asbw.alestravoip.com"**. 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.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.7**. This short code will use this line group when placing outbound calls.
- Default values may be used for all other parameters.

IP Offices		8N;: Dial	
Short Code		Code	8N;
		Feature	Dial
		Telephone Number	N"@asbw.alestravoip.com"
		Line Group ID	50: Main
		Locale	United States (US English)
		Force Account Code	<input type="checkbox"/>

5.10. 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** (not shown). 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 the entries to specify the exact number 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, for local calls, the user dialed 9 plus the 8 digit local number, starting with a 1 or a 2, which was the range of local numbers used during the compliance test. For national long distance calls in Mexico, the user dialed 9, then 01, followed by 10 digit numbers.

The screenshot displays the configuration for the 'Main' route in the IP Office system. The left-hand pane shows a tree view of the system configuration, with 'ARS (2)' expanded to show '50: Main'. 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:

The central table lists the following entries:

Code	Telephone Number	Feature	Line Group ID
411	411	Dial	17
2XXXXXXX	2N	Dial	17
040	040	Dial	17
01XXXXXXXXXX	01N	Dial	17
01800XXXXXXX	01800N	Dial	17
00XXXXXXXXXX	00N	Dial	17
001XXXXXXXXXX	001N	Dial	17

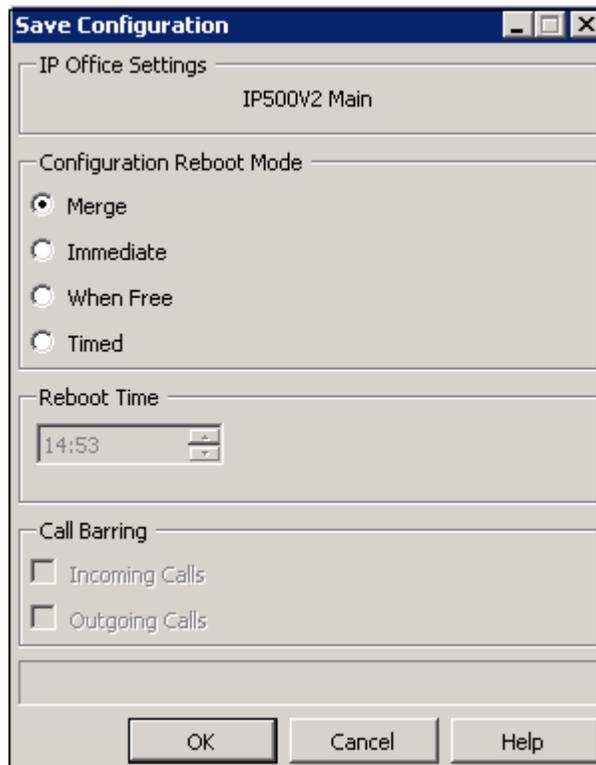
Additional settings at the bottom include:

- Alternate Route Priority Level: 3
- Alternate Route Wait Time: 30
- Alternate Route: <None>

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.

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



The image shows a 'Save Configuration' dialog box with the following sections and controls:

- IP Office Settings:** A text field containing 'IP500V2 Main'.
- Configuration Reboot Mode:** A group box containing four radio buttons: 'Merge' (selected), 'Immediate', 'When Free', and 'Timed'.
- Reboot Time:** A time selection control showing '14:53'.
- Call Barring:** A group box containing two checkboxes: 'Incoming Calls' and 'Outgoing Calls', both of which are currently unchecked.
- Buttons:** Three buttons at the bottom: 'OK', 'Cancel', and 'Help'.

6. Alestra Enlace IP SIP Trunking Configuration

Alestra is responsible for the configuration of the Alestra Enlace IP SIP Trunk service. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise. Alestra will provide the customer the necessary information to configure the Avaya IP Office SIP trunk connection, including:

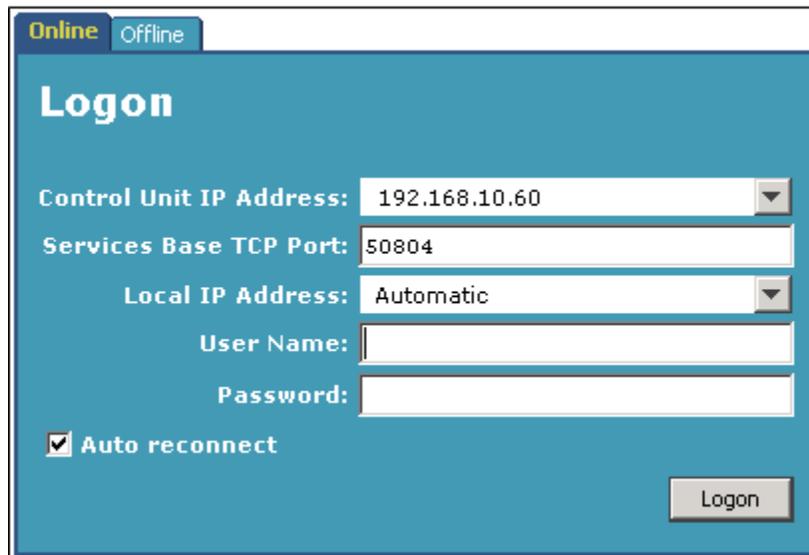
- IP address of the Alestra SIP Proxy server.
- Credentials for the SIP trunk registration.
- Supported codecs and order of preference.
- DID numbers.
- All IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices.

7. Verification Steps

The Avaya IP Office System Status and Monitor applications are useful tools used for the verification of the IP Office configuration and for troubleshooting the SIP connection to the service provider.

7.1. System Status

The Avaya IP Office System Status application can be used to verify the service state of the SIP line. Launch the application from **Start → Programs → IP Office → System Status** on the PC where IP Office Manager was installed. Under **Control Unit IP Address** select the IP address of the IP Office system under verification. Log in using the appropriate credentials



The screenshot shows the 'Logon' window of the Avaya IP Office System Status application. At the top, there are two tabs: 'Online' (selected) and 'Offline'. The window has a blue header with the title 'Logon'. Below the header, there are several input fields and a checkbox:

- Control Unit IP Address:** A dropdown menu with '192.168.10.60' selected.
- Services Base TCP Port:** A text input field containing '50804'.
- Local IP Address:** A dropdown menu with 'Automatic' selected.
- User Name:** An empty text input field.
- Password:** An empty text input field.
- Auto reconnect**

A 'Logon' button is located at the bottom right of the window.

Select the SIP line of interest from the left pane (**Line 17** in the reference configuration). 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 shows the Avaya IP Office System Status interface. The left pane shows a tree view with 'Line: 17' selected. The main pane is on the 'Status' tab, displaying the 'SIP Trunk Summary' for Line 17. The summary includes details such as Peer Domain Name (asbw.alestravoip.com), Resolved Address (192.168.26.150), Line Number (17), and Number of Administered Channels (10). A green progress indicator shows 0% utilization. Below the summary is a table of channel states.

Channel Number	URI G...	Call Ref	Current State	Time in State	Remote Media Ad...	Codec	Connec...	Caller ID or Dial...	Other Party on Call	Direction of Call	Round Trip De...	Receive Jitter	Receive Packet ...	Transmit Jitter	Transmit Packet ...
1			Idle	00:00:26											
2			Idle	00:00:40											
3			Idle	5 days ...											
4			Idle	6 days ...											
5			Idle	6 days ...											
6			Idle	6 days ...											
7			Idle	6 days ...											
8			Idle	6 days ...											
9			Idle	6 days ...											
10			Idle	6 days ...											

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

The screenshot shows the Avaya IP Office System Status interface with the 'Alarms' tab selected. The main pane displays 'Alarms for Line: 17 SIP asbw.alestravoip.com'. Below this, there is a table with columns for 'Last Date Of Error', 'Occurrences', and 'Error Description', which is currently empty.

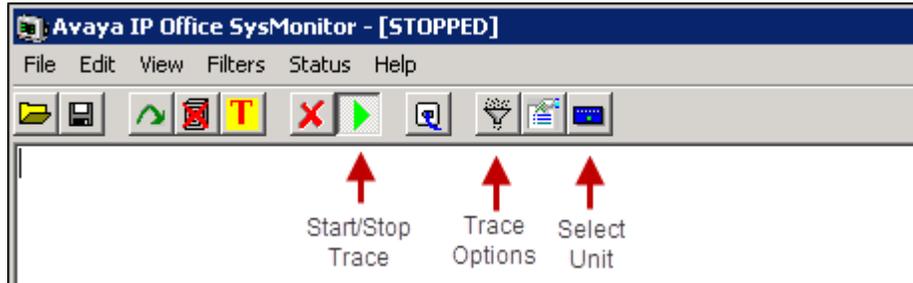
On the **Registration** tab, verify that the trunk is successfully registered with the service provider.

The screenshot shows the Avaya IP Office System Status interface with the 'Registration' tab selected. The main pane displays 'Registration Status' for Line 17. Below this, there is a table with columns for 'Index', 'User Name', 'Status', and 'Retry Time'.

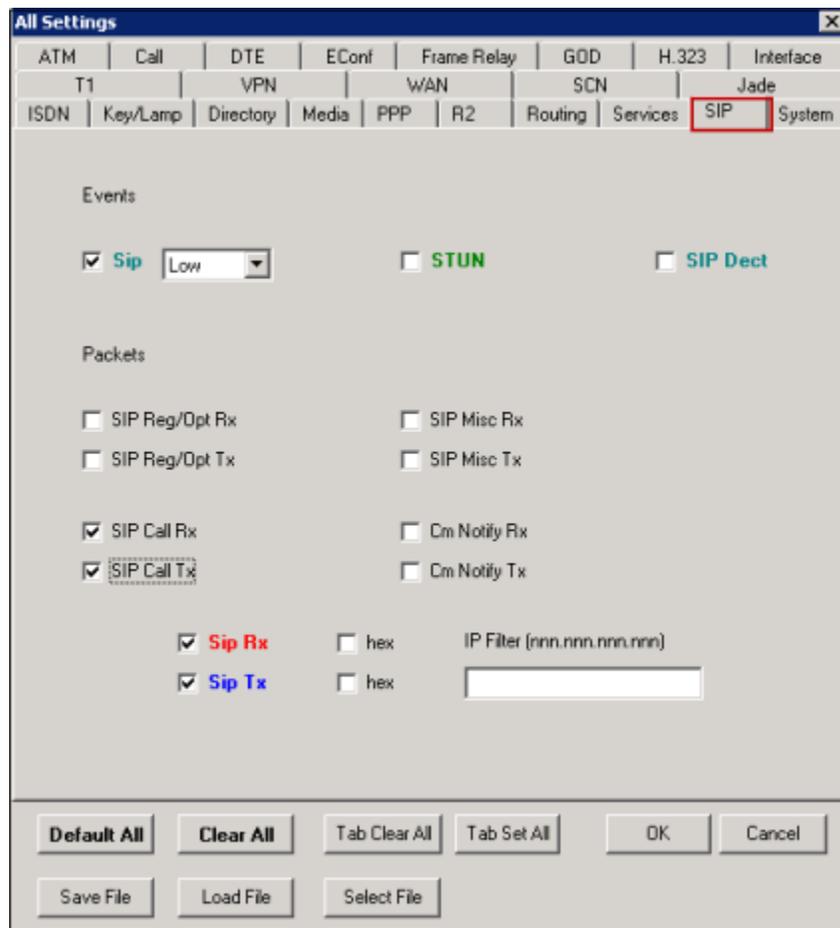
Index	User Name	Status	Retry Time
1	8123456789	Registered	6/23/2014 11:10:32 AM

7.2. Monitor

The Avaya IP Office SysMonitor application can be used to monitor and troubleshoot signaling messaging on the SIP trunk. Launch the application from **Start → Programs → IP Office → Monitor** on the PC where IP Office Manager was installed. Click the **Select Unit** icon on the taskbar and Select the IP address of the IP Office system under verification.



Clicking the **Trace Options** icon on the taskbar and selecting the **SIP** tab allows modifying the threshold used for capturing events, types of packets to be captured, filters, etc. Additionally, the color used to represent the packets in the trace can be customized by right clicking on the type of packet and selecting to the desired color.



8. Conclusion

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between the Alestra Enlace IP SIP Trunk service on the Broadsoft platform and Avaya IP Office 9.0, using SIP Trunk Registration, 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] *Avaya IP Office 9.0, Installing IP500/IP500 V2*. Document 15-601042, February 2014
<https://downloads.avaya.com/css/P8/documents/100174004>
- [2] *Avaya IP Office Manager Release 9.0*, Document 15-601011, January 2014
<https://downloads.avaya.com/css/P8/documents/100174478>
- [3] *Administering Avaya Flare® Experience for iPad devices and Windows, Release 9.0*, September 2013
<https://downloads.avaya.com/css/P8/documents/100175132>
- [4] *IP Office System Status Application*, Document Number 15-601758, May 2013
<https://downloads.avaya.com/css/P8/documents/100150298>
- [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 Alestra Enlace IP SIP Trunk Service is available from Alestra.

©2014 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.