Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 9.1 to support Alestra SIP Trunk Service on the Broadsoft Platform – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking in Avaya IP Office 9.1, to interoperate with the Alestra SIP Trunk Service on the Broadsoft platform.

The SIP Trunking service offered by Alestra provides customers with PSTN access via a SIP trunk between the enterprise and the service provider's network, as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as the observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

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.

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1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the Alestra SIP Trunk Service, operating on the Broadsoft platform, and an Avaya IP Office solution.

In the sample configuration, the Avaya solution consists of an Avaya IP Office 500v2 Release 9.1, Avaya Voicemail Pro and Avaya IP Office soft clients and deskphones, including SIP, H.323, digital, and analog endpoints.

The Alestra SIP Trunk Service referenced within these Application Notes is designed for business customers in Mexico. Customers using this service with this Avaya 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 all the Avaya equipment for the SIP-enabled solution was installed at the Avaya Solution and Interoperability Lab. The enterprise site was configured to connect to the Alestra SIP Trunk Service via a broadband connection.

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.

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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 with the service provider.
- 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 Communicator for Windows softphones.
- Various call types including: local, outbound toll free, long distance national and long distance international.
- Codecs G.729A, G.711A and G.711MU.
- Fax.
- 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:

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

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2.2. Test Results

Interoperability testing of the Alestra SIP Trunk Service was completed with successful results for all test cases with 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 DIDs 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 carrier 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 as expected on this condition.
- **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.
- **Voice quality when using codec G.729A**: During the compliance test it was observed poor voice quality on calls when codec G729A was used. The voice quality was normal when using codecs G.711A and G.711U. This condition may be limited to the testing environment and not be present in a real customer scenario, it is listed here as an observation.
- **Fax support**: Inbound and outbound fax calls using the T.38 protocol or the G.711 pass-through mode failed during the test. Fax should not be used with this solution.
- **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.

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2.3. Support

For technical support on the Alestra 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 SIP Trunk service through a public Internet WAN connection.

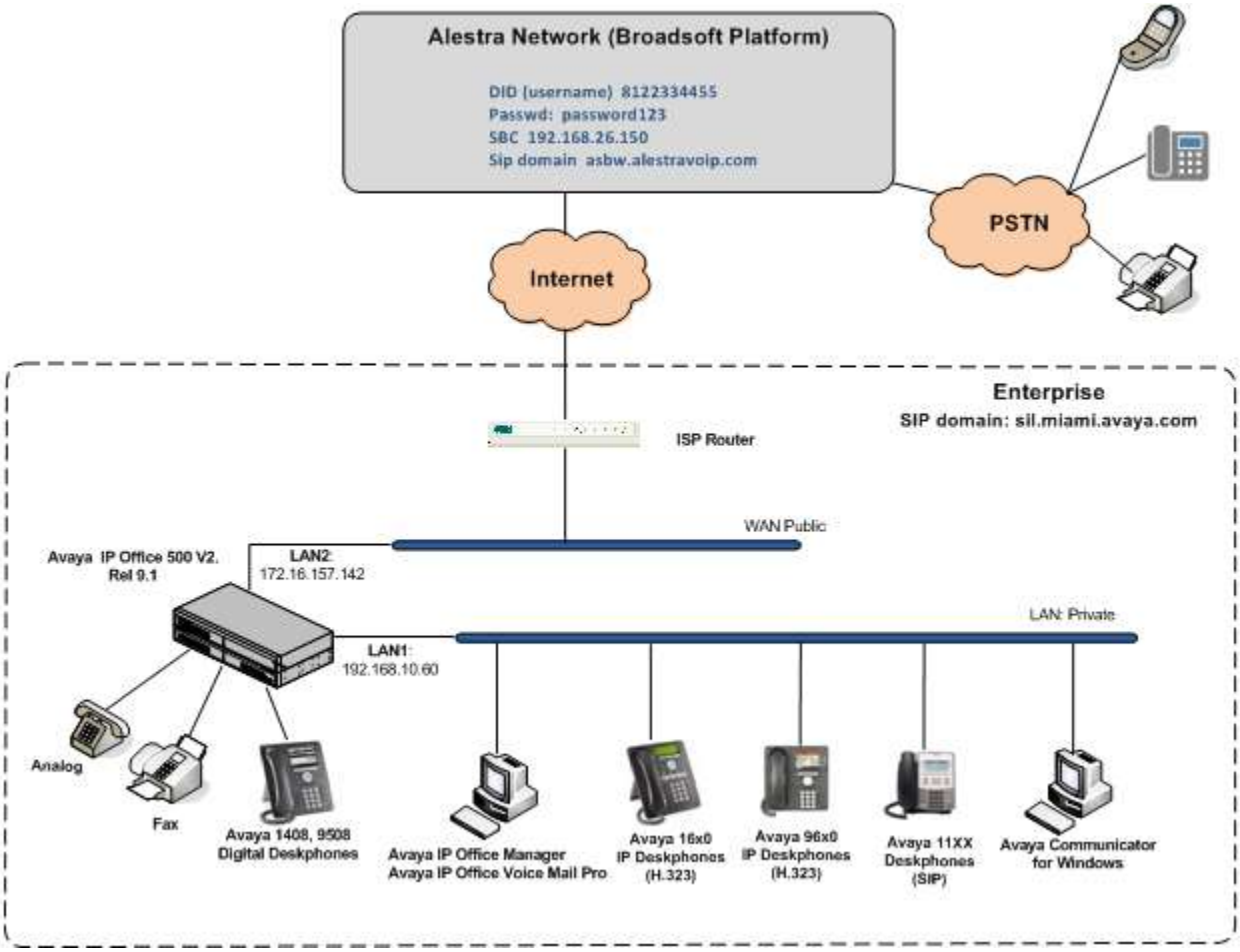

Figure 1: Test Configuration

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

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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 Deskphones (with H.323 firmware), Avaya 1140E IP Deskphones (with SIP firmware), Avaya 1408 and 9508D Digital Deskphones, analog telephones and PCs running Avaya Communicator 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 telephones.

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 routers, data firewall, etc. 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.

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4. Equipment and Software Validated

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

Component	Version
Avaya	
Avaya IP Office 500v2	9.1.100.10
Avaya IP Office Digital Expansion Module DCPx16	9.1.100.10
Avaya IP Office Manager	9.1.1.0.Build 10
Avaya IP Office Voicemail Pro	9.1.100.3
Avaya 1608 IP Deskphone (H.323)	1.3.5
Avaya 9640 IP Deskphone (H.323)	Avaya one-X Deskphone Edition S3.230A
Avaya 1140E IP Deskphone (SIP)	04.04.18.00
Avaya Digital Deskphone 1408	40.0
Avaya Digital Deskhone 9508	0.55
Avaya Communicator for Windows	2.0.3.30
Alestra	
Broadsoft Softswitch	Release 17 SP 4
Acme Packet SBC	V6.2
Lucent 5ESS	V16.1

Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2, and also when deployed with all configurations of IP Office Server Edition without T.38 Fax Service. (T.38 fax is not supported on IP Office Server Edition). Note that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks.

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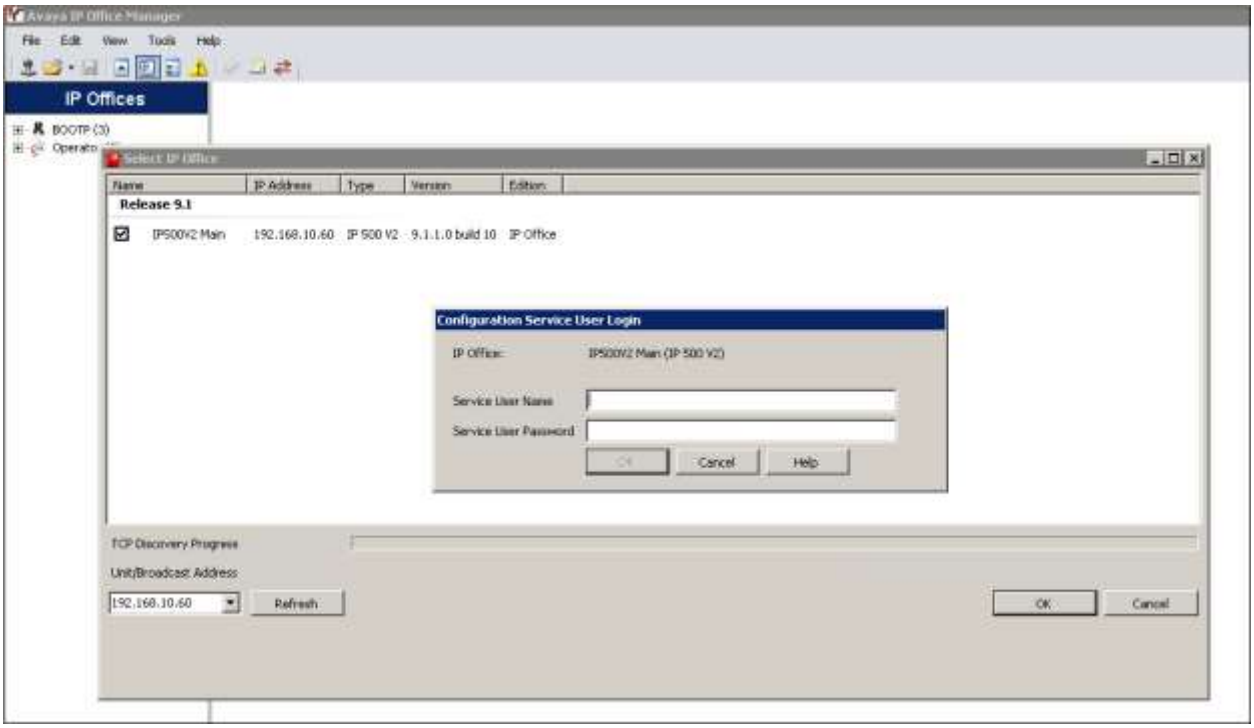
5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration necessary to support connectivity to the Alestra SIP Trunk Service. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the PC running 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 using 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 are assumed to be already in place, and they are not part of these Application Notes.

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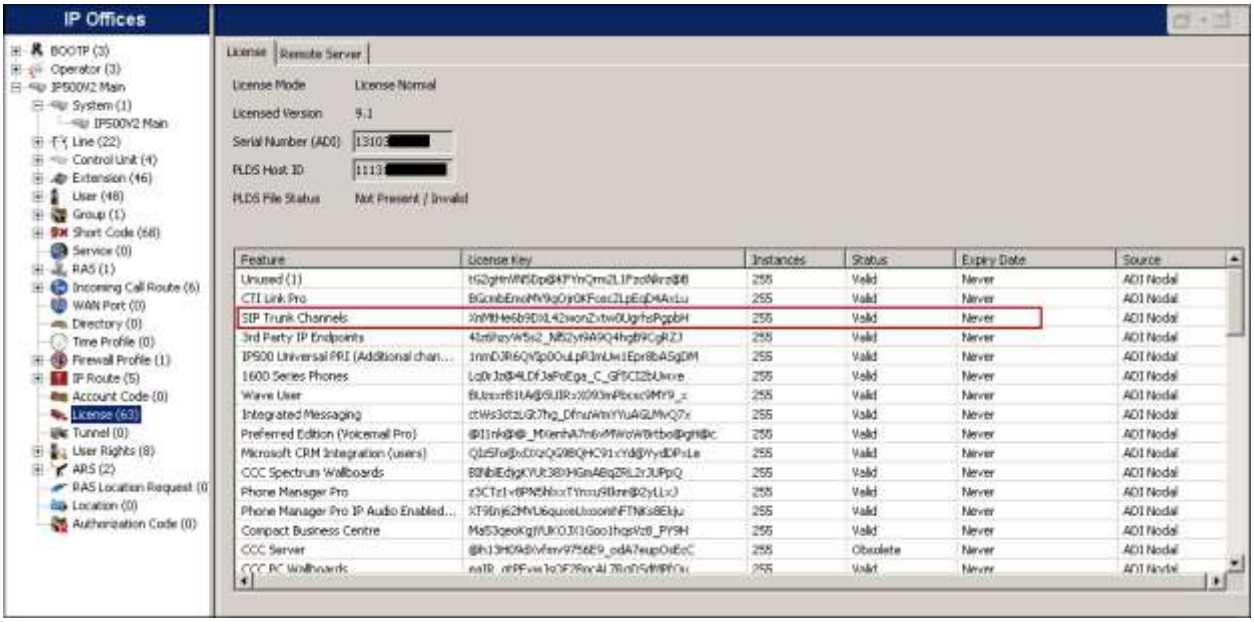
5.1. Licensing

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.

In the reference configuration, ***IP500_Main*** was used as the system name. Under the system name on the Navigation pane, select **License**. Confirm that there is a valid **SIP Trunk Channels** license with sufficient "Instances" in the Details pane, enough to support the number of channels to be deployed on the SIP trunk to the service provider.



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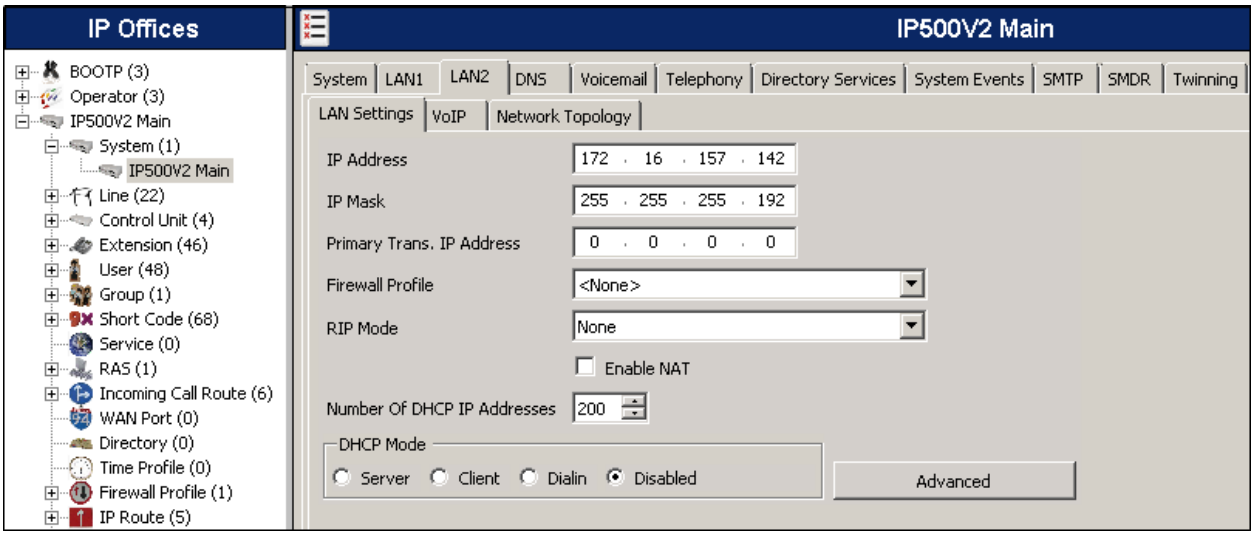
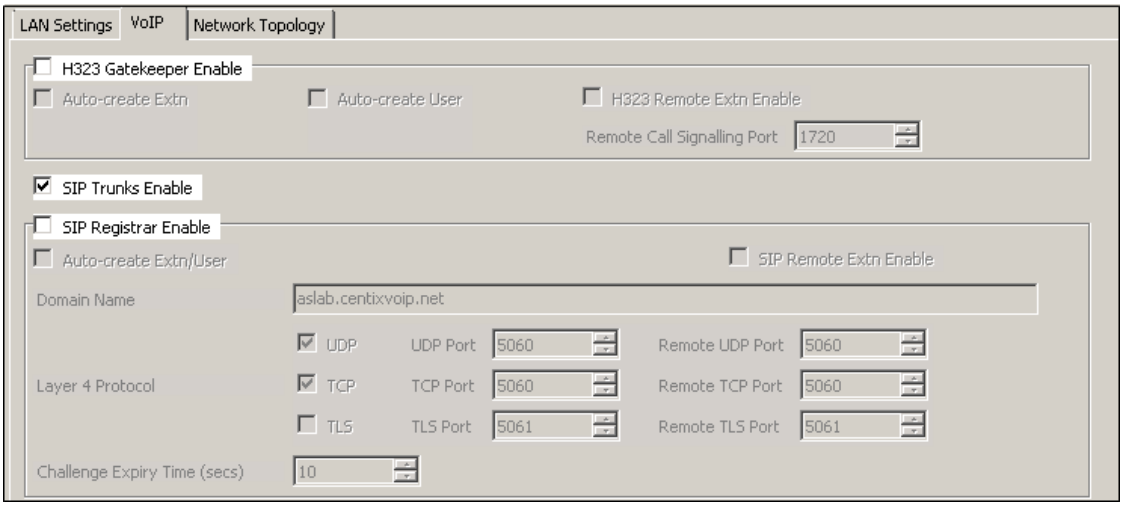
5.2. LAN2 Settings

In the sample configuration, 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)** under the system name in the Navigation pane and select the **LAN2 → LAN Settings** tab in the Details pane. Set the **IP Address** and **IP Mask** fields to the IP address and subnet mask assigned to the Avaya IP Office LAN2 port. 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 on this interface.



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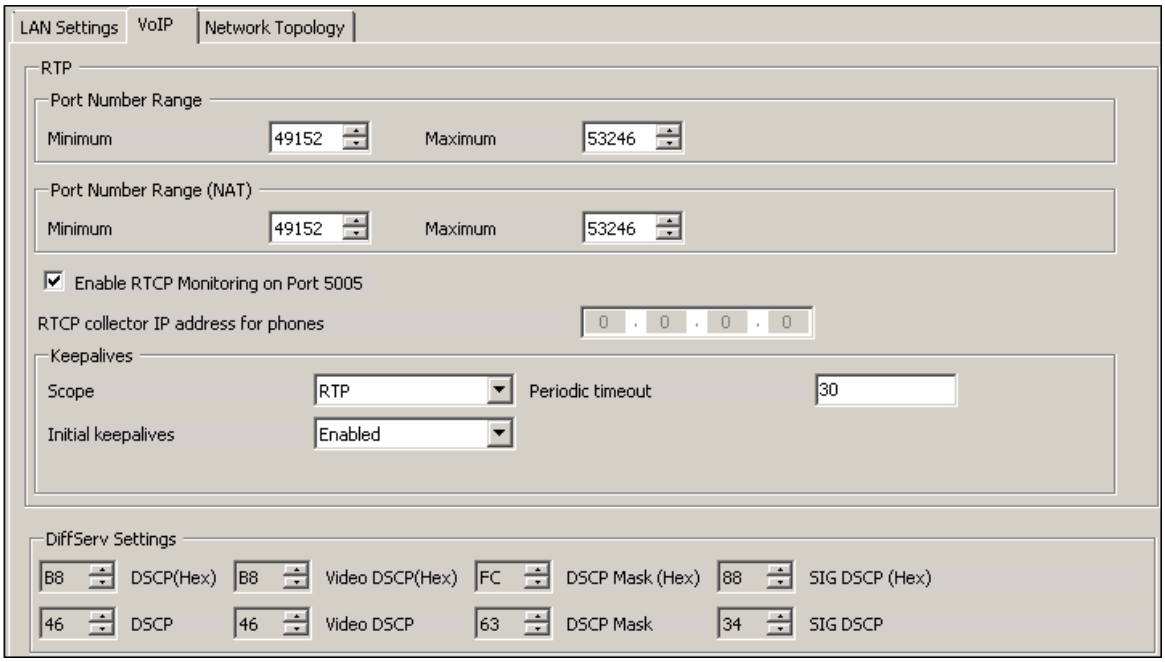
The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2.

In the **RTP Keepalives** section, set the **Scope** field to ***RTP***. Set the **Periodic timeout** to ***30*** and **Initial keepalives** to ***Enabled***. This will cause the IP Office to send RTP keepalive packets at the beginning of the calls and periodically thereafter, to avoid problems of media deadlock resulting in no audio situations that can occur with certain types of forwarded calls that are routed from the IP Office back to the network, over the same SIP trunk.

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.



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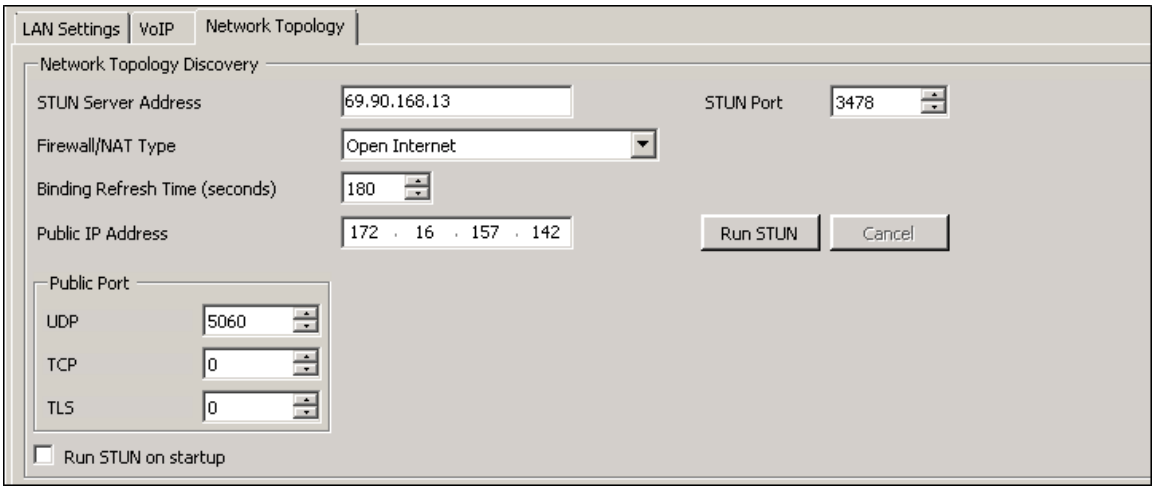
On the **Network Topology** tab in the Details pane, configure the following parameters:

- 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, settings obtained by STUN lookups are ignored. The IP address used is the one assigned to the interface.
- **Binding Refresh Time (seconds)** is used to determine the frequency at which Avaya IP Office will send SIP OPTION messages to the SIP trunk using this interface. If this parameter is left at the default value 0, the IP Office will send OPTIONS messages using its default interval of 300 seconds. A different value can be used if an interval lower than 300 seconds is needed. In the reference configuration, ***180*** seconds was used.
- Set **Public IP Address** to the IP address that was set for LAN2.
- Set **Public Port** to ***5060*** for **UDP**.
- Defaults were used for all other fields.



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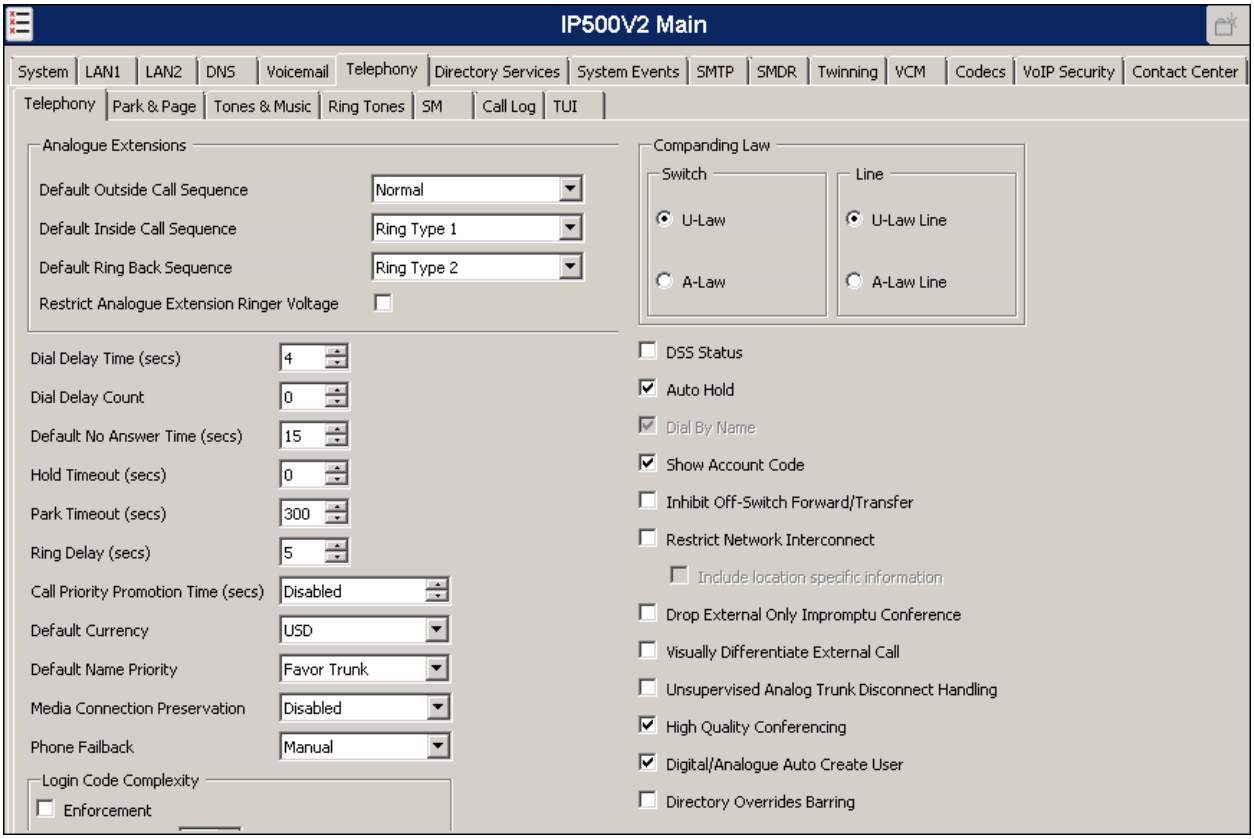
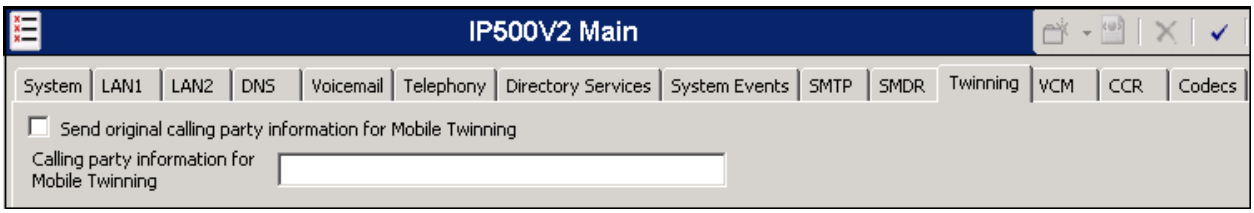
5.3. System Telephony Settings

Navigate to the **Telephony → Telephony** Tab in the Details Pane. 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 **Twinning** tab on the Details Pane. Uncheck the **Send original calling party information for Mobile Twinning** box. This will allow the Caller ID for Twinning to be controlled by the setting on the SIP Line (**Section 5.7**). This setting also impacts the Caller ID for call forwarding.



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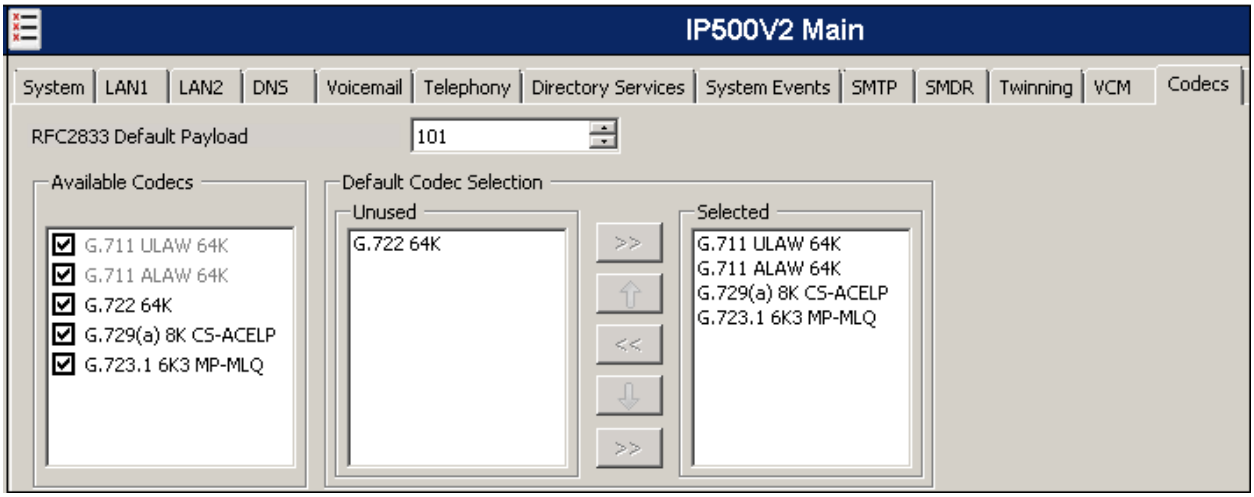
5.5. System Codecs Settings

Navigate to the **Codecs** tab in the Details Pane. The **RFC2833 Default Payload** field 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.

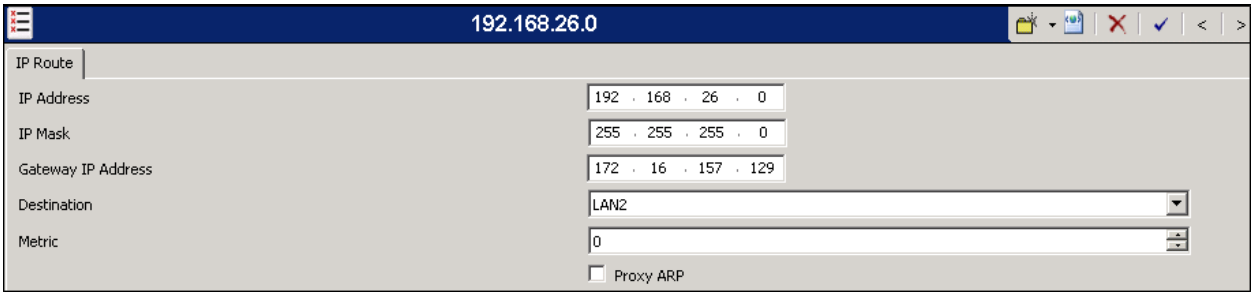
Click **OK** (not shown) to save any changes made to any of the various **System** tabs.



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.



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5.7. Administer SIP Line

A SIP line is created to establish the SIP connection between the Avaya IP Office and the Alestra SIP Trunk 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** and **Section 5.7.2** 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 5.7.3 – 5.7.8**.

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 5.7.3 – 5.7.8**.

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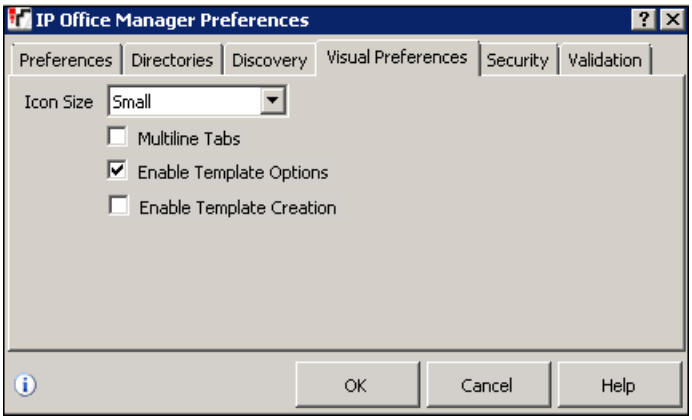
5.7.1. Importing a SIP Line Template

Note – DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500v2) and IP Office Server Edition systems. Alternatively, binary templates may be generated. However, binary templates include all the configuration parameters of the Trunk, including sensitive customer specific information. Therefore, binary templates should only be used for cloning trunks within a specific customer's environment.

1. Copy a previously created template file to a location (e.g., *\Temp*) on the same computer where IP Office Manager is installed. By default, the template file name will have the format **AF_<*user supplied text*>_SIPTrunk.xml**, where the **<*user supplied text*>** portion is entered during template file creation.

   Note – If necessary, the **<*user supplied text*>** portion of the template file name may be modified, however the **AF_<*user supplied text*>_SIPTrunk.xml** format of the file name must be maintained. For example, an original template file **AF_*TEST* _SIPTrunk.xml** could be changed to **AF_*Test1*_SIPTrunk.xml**. The template file name is selected in **Section 5.7.2** to create a new SIP Line.

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. Check the box next to **Enable Template Options**. Click **OK**.



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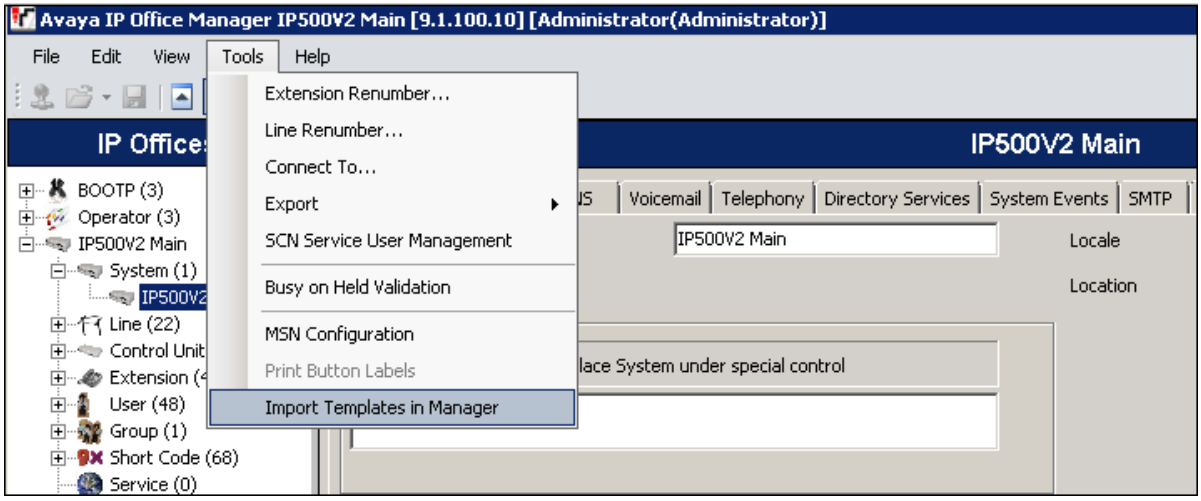
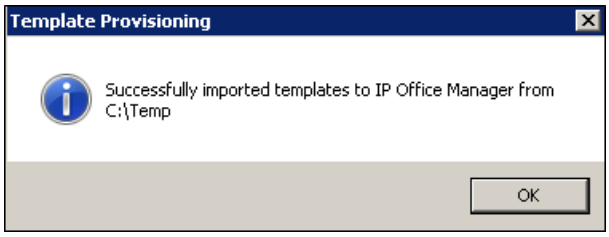
3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**.



4. A folder browser will open (not shown). Select the directory used in **step 1** to store the template (e.g., *\Temp*). In the reference configuration, template file **AF_*Alestra*_ SIPTrunk.xml** was imported. The template file is automatically copied into the default template location, **C:\Program Files\Avaya\IP Office\Manager\Templates**.

5. After the import is complete, a final import status pop-up window will open stating success or failure. Click **OK**.



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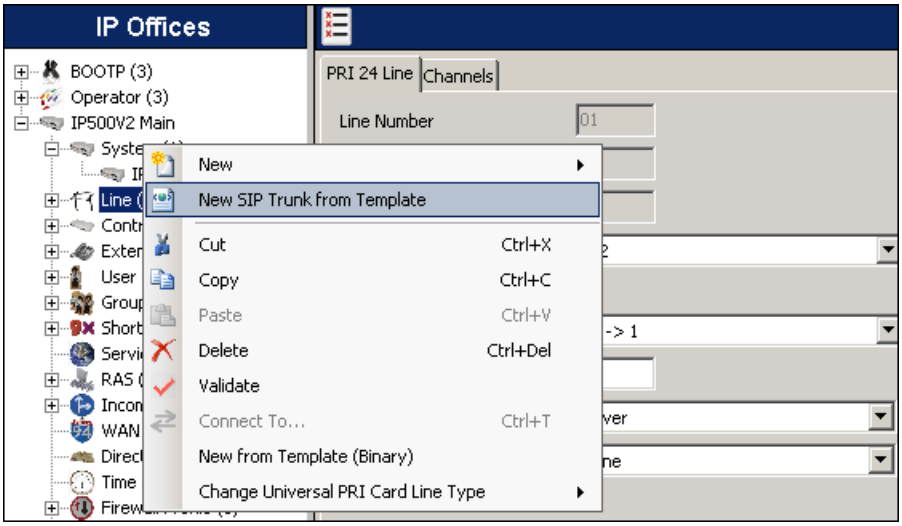
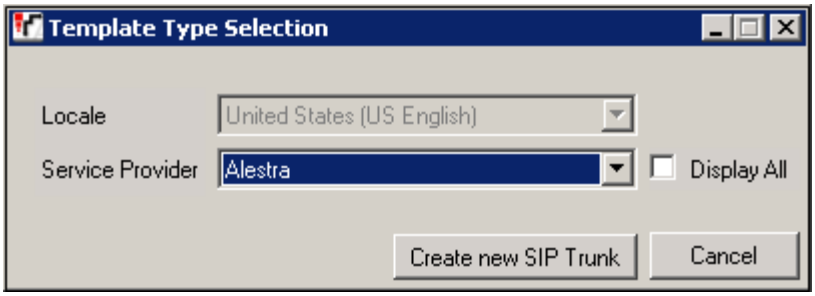
5.7.2. Creating a SIP Trunk from an XML Template

1. To create the SIP Trunk from a template, right-click on **Line** in the Navigation Pane, and select **New SIP Trunk from Template**.



2. In the subsequent **Template Type Selection** pop-up window, from the **Service Provider** pull-down menu, select the XML template name from **Section 5.7.1**.

> **Note** – The drop down menu will display the ***<user supplied text>*** part of the template file name (see **Section 5.7.1**). If the **Display All** box is checked, then the full template file name is displayed.



Click **Create new SIP Trunk** to finish creating the trunk.

3. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.7.3 – 5.7.8.**

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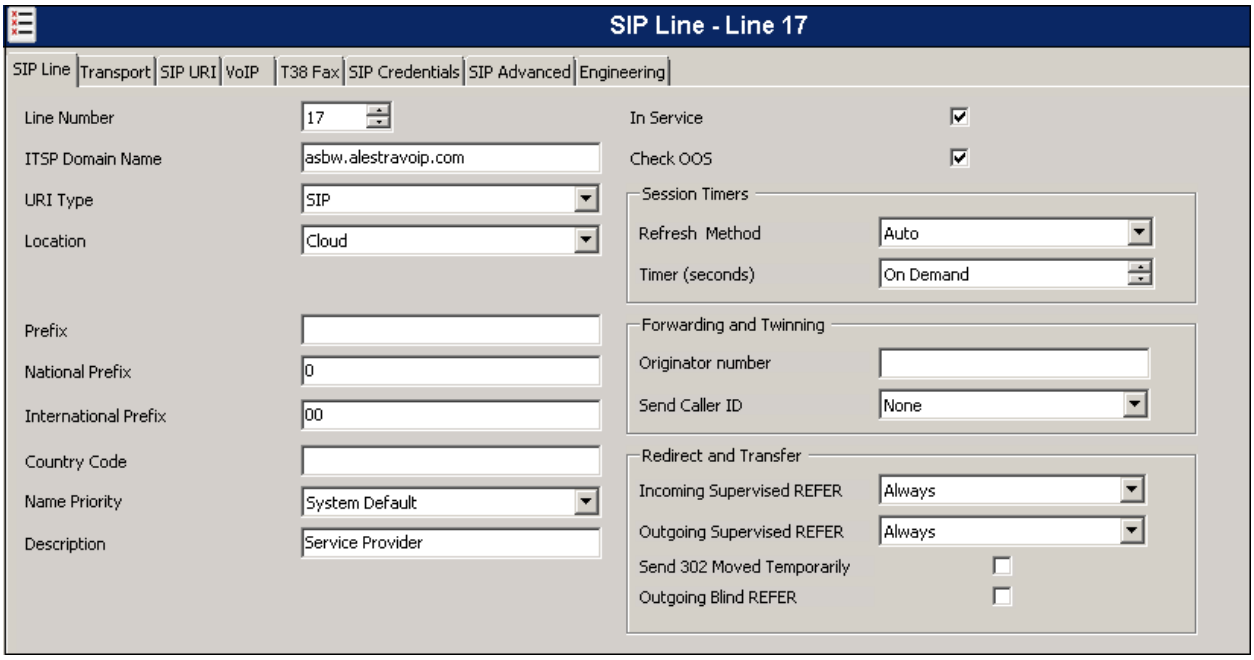
5.7.3. SIP Line Tab

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. In the reference configuration, this was ***asbw.alestravoip.com***. 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.
- On the **Forwarding and Twinning** section, 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.) caused these types of calls to be rejected by Alestra.
- On the **Redirect and Transfer section**, set **Incoming Supervised REFER** and **Outbound Supervised REFER** to ***Always***.
- Default values may be used for all other parameters.



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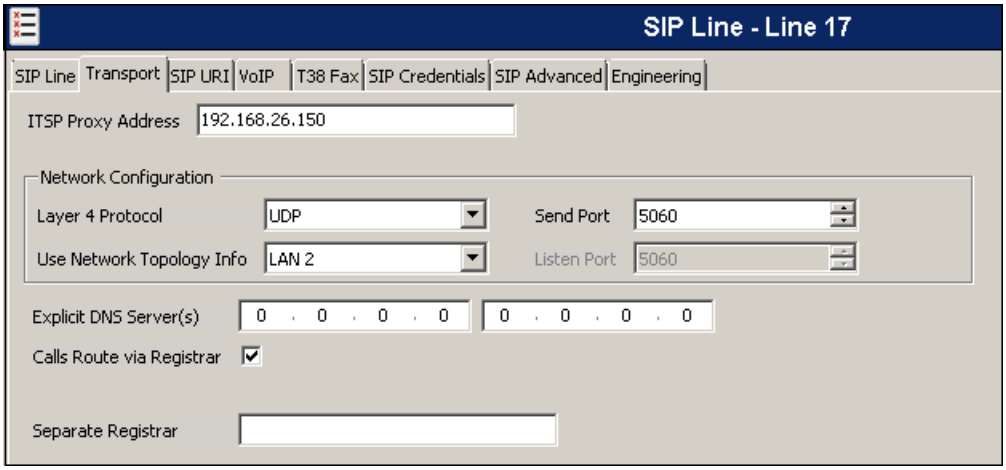
5.7.4. Transport Tab

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.



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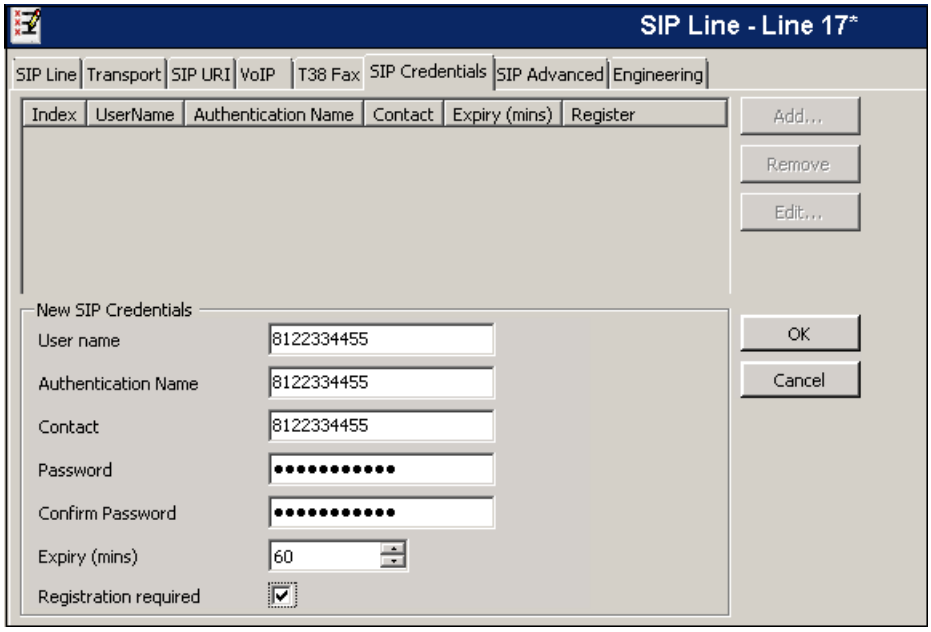
5.7.5. SIP Credentials

Alestra required the use of SIP Credentials for the registration of the SIP trunk 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**.



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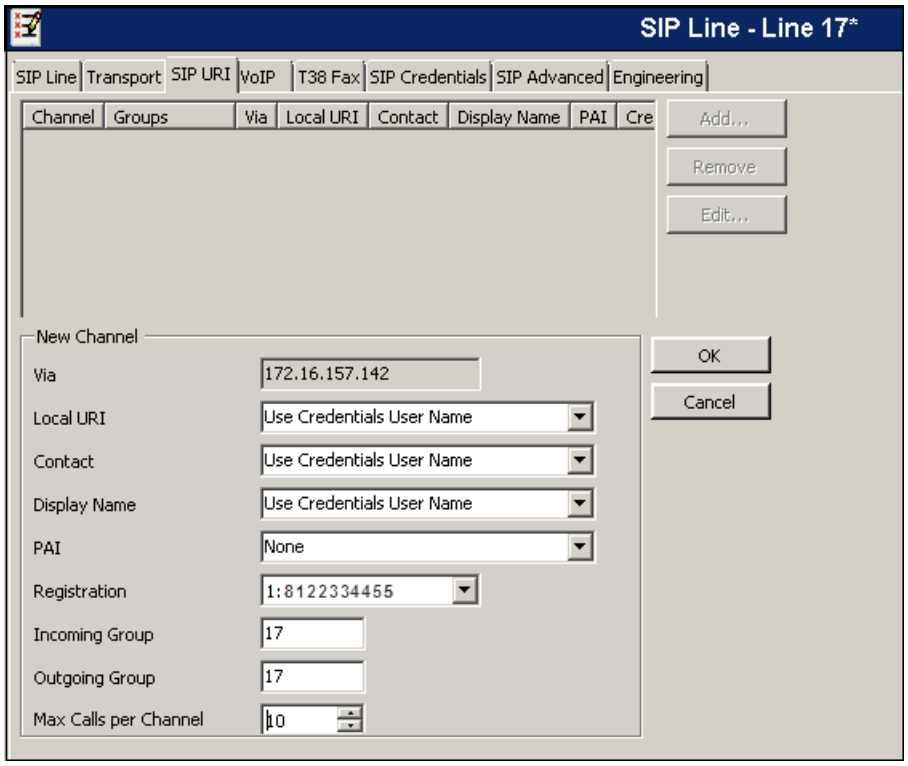
5.7.6. SIP URI Tab

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: <8122334455>*** 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.



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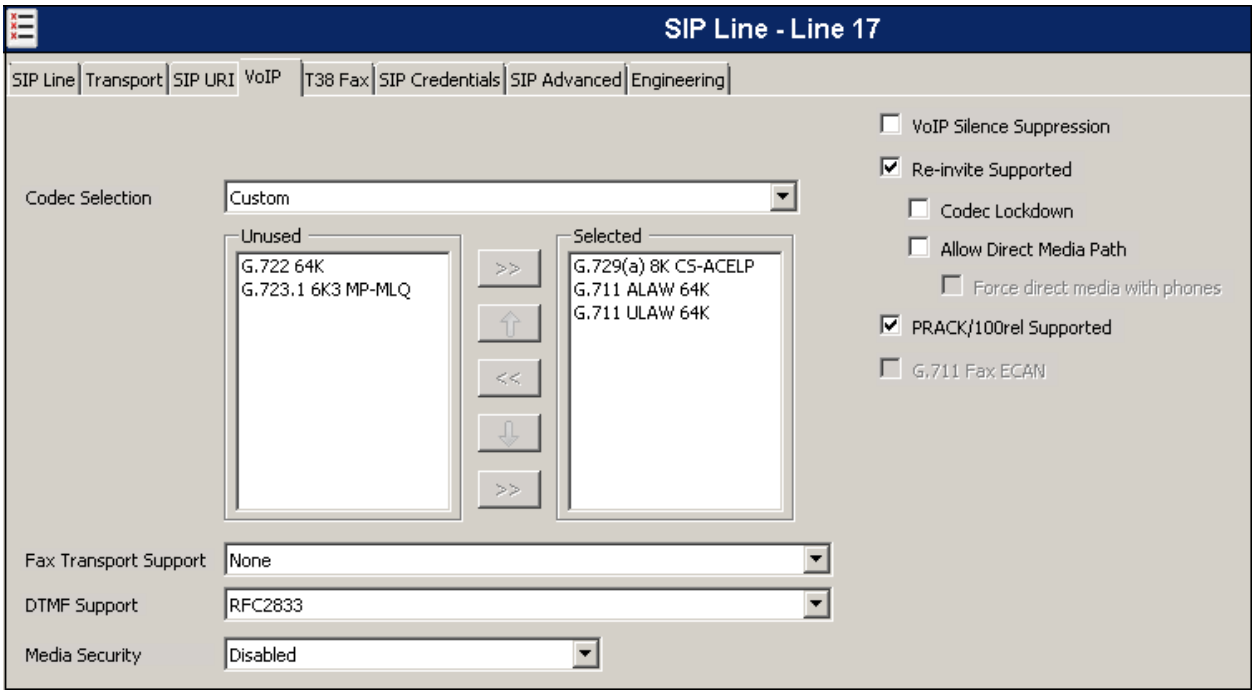
5.7.7. VoIP Tab

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. During the compliance test, **G729A**, **G711A and G711U**, in this order of preference, were the codecs supported by Alestra.
- Set **Fax Transport Support** to ***None***. See **Section 2.2**.
- 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 the service provider.
- Default values may be used for all other parameters.



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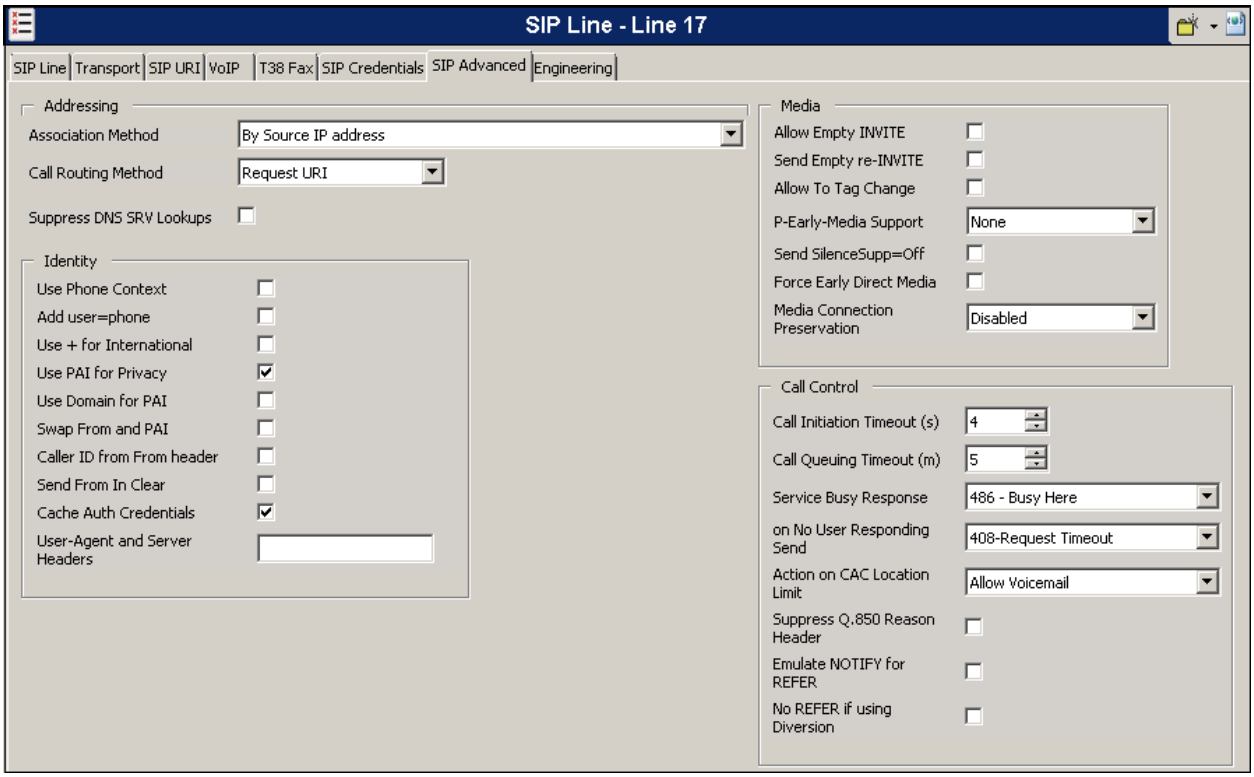
5.7.8. SIP Advanced Tab

For outbound calls with privacy enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with "anonymous". 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 purposes. By default, IP Office will use the PPI header for privacy. To configure IP Office to use the PAI header for privacy calls, check the box for **Use PAI for Privacy**.

All other fields retained their default values.



Click **OK** (not shown) to save any changes made to any of the various "SIP Line" tabs.

No changes were made to the **T.38 Fax** and the **Engineering** tabs, so they will not be visited**.**

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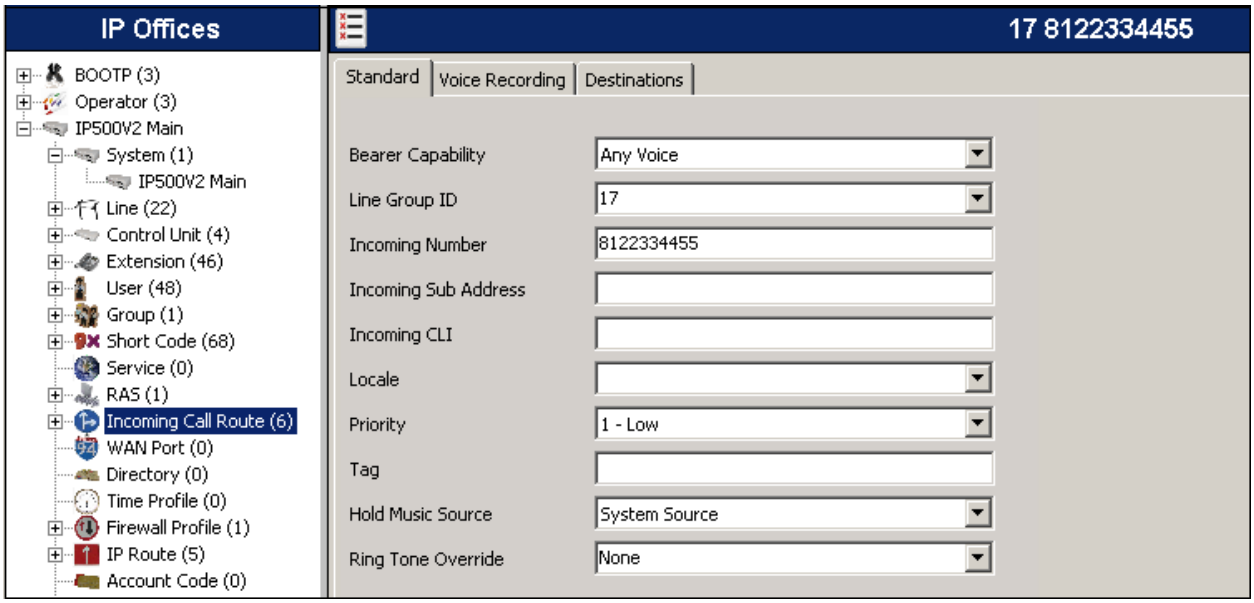
5.8. Incoming Call Route

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.

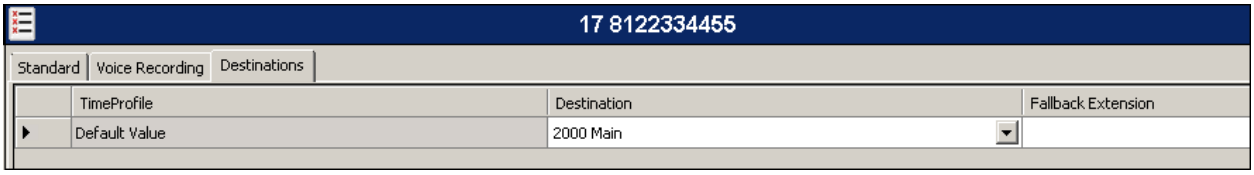
To add a new incoming call route, from the left Navigation Pane, right-click on Incoming Call Route and select New (not shown). The screen below shows the route for the DID number assigned to the enterprise by Alestra, ***8122334455*** in this example.

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.



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 8122334455 on line 17 are routed to hunt group ***2000 Main***.



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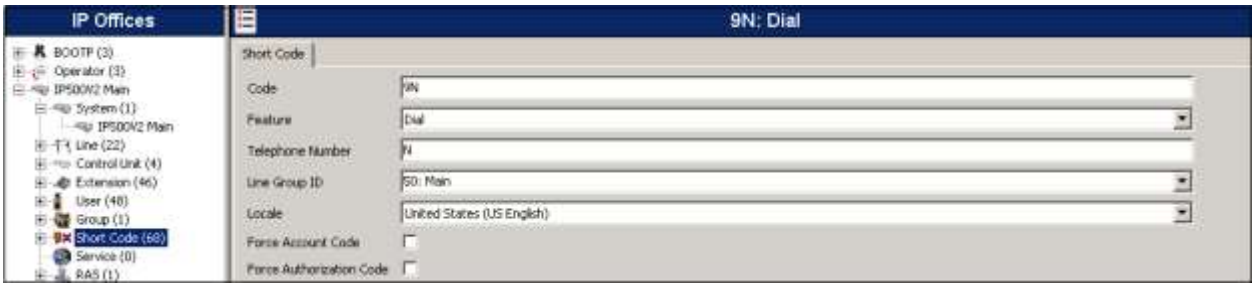
5.9. Short Code

In the reference configuration, Avaya IP Office used Automatic Route Selection (ARS) to route outbound traffic to the SIP line. A short code is needed to send the outbound traffic to the ARS route. To create the short code used for ARS, right-click on **Short Code** in the Navigation Pane and select **New** (not shown). The screen below shows the creation of the short code ***9N*** used in the reference configuration. When the Avaya IP Office users dialed 9 plus any number N, calls were directed to **Line Group *50: Main***, configurable via ARS and defined next in **Section 5.10**.

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, in this case ***9N***. This short code will be invoked when the user dials 9 followed by any number.
- Set **Feature** to ***Dial***. This is the action that the short code will perform.
- Set **Telephone Number** to **N**. The value **N** represents the number dialed by the user after removing the **9** prefix.
- Set the **Line Group ID** to the ARS route to be used. In the example shown, the call is directed to **Line Group *50: Main***.
- Click **OK** (not shown).



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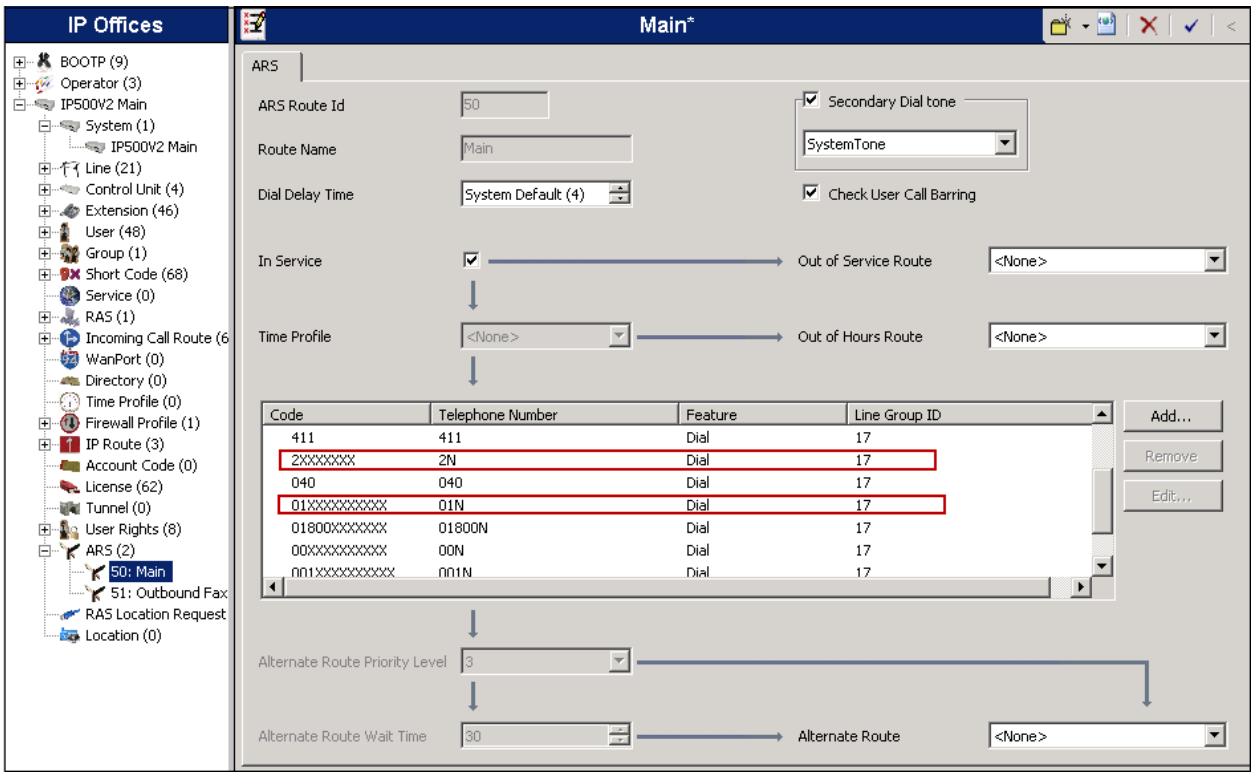
5.10. Automatic Route Selection

While detailed coverage of ARS is beyond the scope of these Application Notes, this section includes some basic screen illustrations of the ARS settings used during the compliance test.

The following screen shows the ARS configuration for the route ***50: Main***. The example shows a subset of the dialed strings tested as part of the compliance test. See **Section 2.1** for the complete list of call types tested. Note the sequence of ***X***s used in the **Code** column of some entries, to specify the exact number of digits to be expected following the access code and the first digits on the string. This type of setting results in a much quicker response in the delivery of the calls by the IP Office. The highlighted entries show that for example, for calls in the local area code, the user dialed 9 plus the 8 digit local number, starting with 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.



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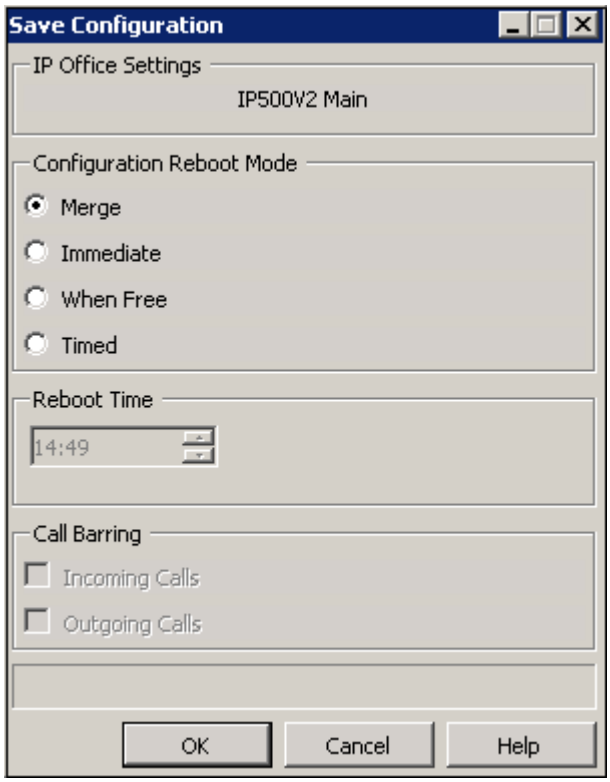
5.11. Save Configuration

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.



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6. Alestra SIP Trunking Configuration

Alestra is responsible for the configuration of the SIP Trunk service in its network. The customer will need to provide the IP address and port used to reach the Avaya IP Office at the enterprise. Alestra will provide the customer the necessary information to configure the SIP trunk connection from the enterprise site to the network, 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.

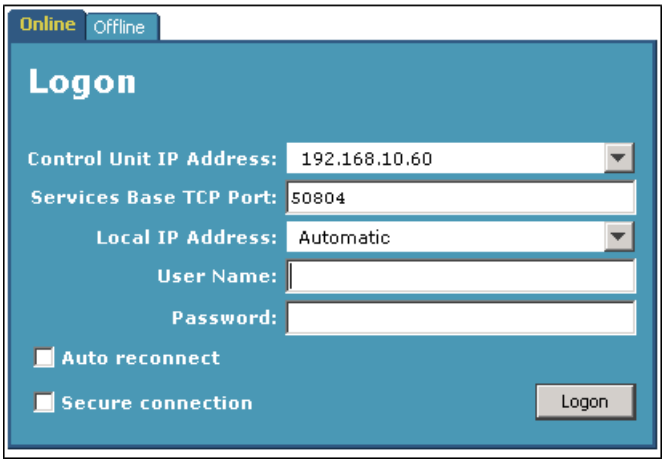
This information is used to complete the configuration of the Avaya IP Office discussed in the previous sections.

7. Verification Steps

The Avaya IP Office System Status and Monitor applications are useful tools used for the verification and troubleshooting of 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 Avaya 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



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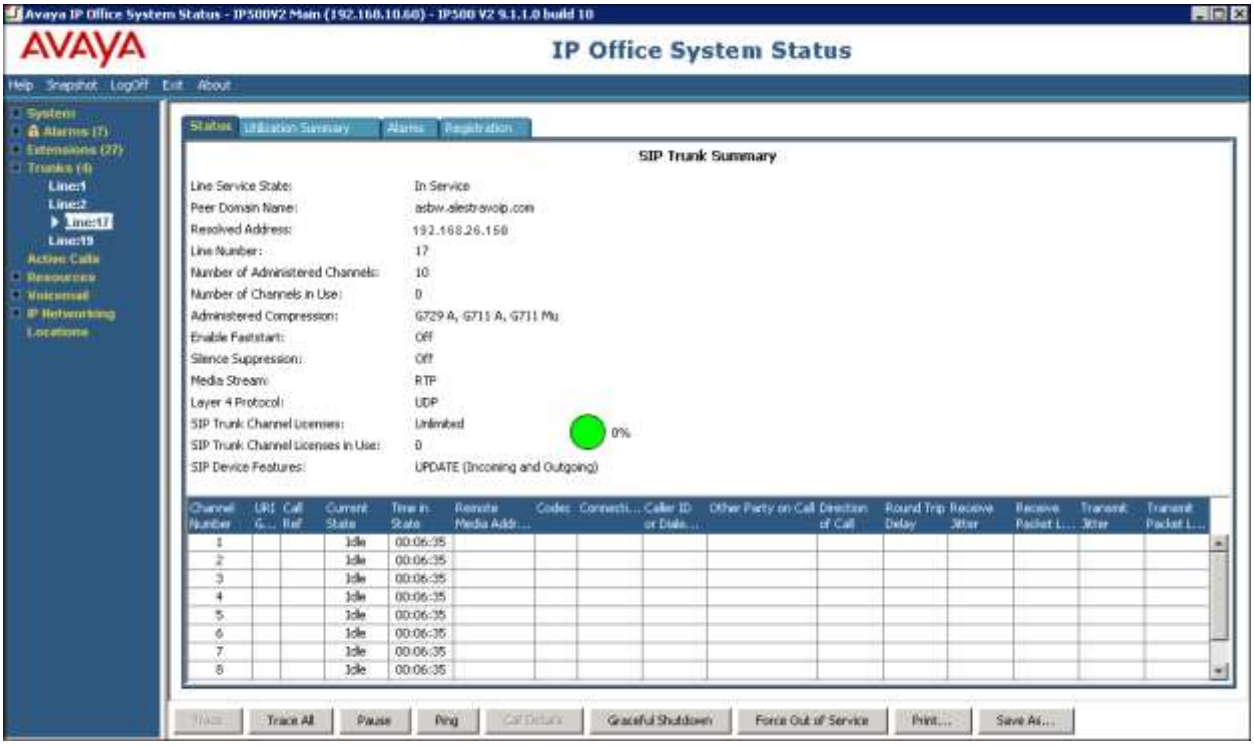
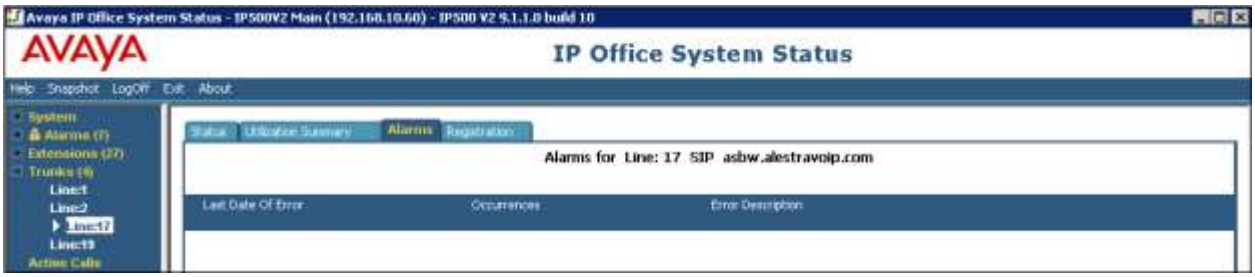
Select the SIP line of interest from the left pane (**Line 17** in the reference configuration). On the **Status** tab in the right pane, verify that the **Current State** is ***Idle*** for each channel (assuming no active calls at present time).



S**elec**t the **Alarms** tab and verify that no alarms are active on the SIP line.



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

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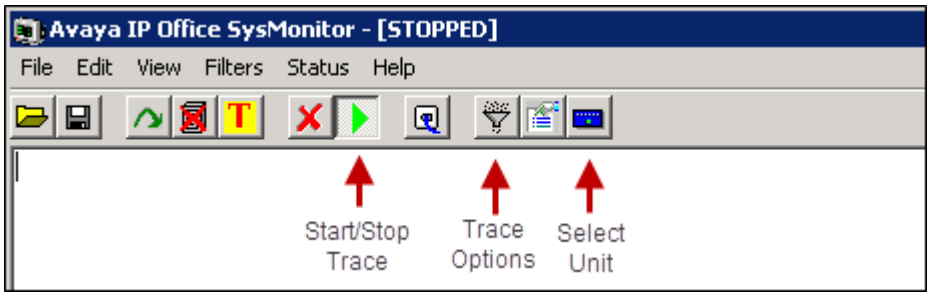
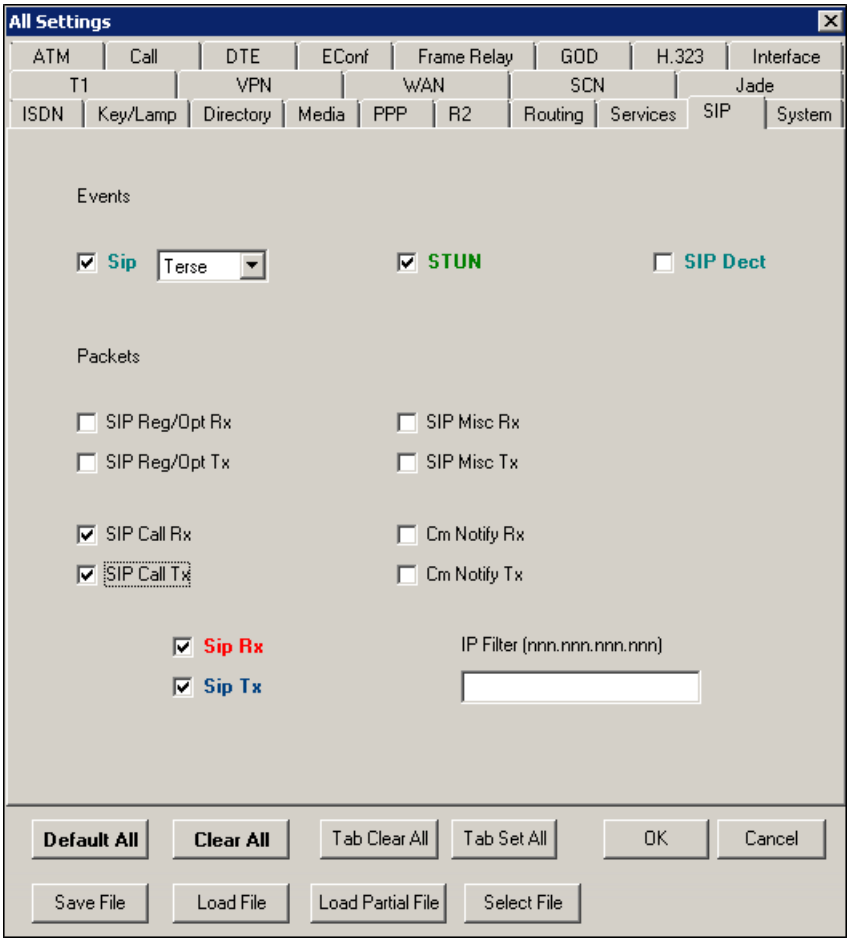
7.2. Monitor

The Avaya IP Office Monitor 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 Avaya 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.



Click the **Trace Options** icon on the taskbar and select the **SIP** tab to modify 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 the desired color.



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8. Conclusion

These Application Notes describe the procedures required to configure SIP trunk connectivity on the Avaya IP Office Release 9.1, to support the Alestra SIP Trunk Service on the Broadsoft platform, 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 Platform 9.1, Deploying Avaya IP Office Platform IP500V2*, Document 15-601042, April 2015
https://downloads.avaya.com/css/P8/documents/101005082

[2] *Administering Avaya IP Office Platform with Manager, Release 9.1*, March 2015
https://downloads.avaya.com/css/P8/documents/101005673

[3] *Administering Avaya Communicator on IP Office, Release 9.1*, December 2014
https://downloads.avaya.com/css/P8/documents/101005862

[4] *IP Office Platform 9.1, Using Avaya IP Office Platform System Status*, Document 15-601758, April 2015
https://downloads.avaya.com/css/P8/documents/101005061

[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 Alestra SIP Trunk Service is available from Alestra.

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