Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 9.0 to support Swisscom SME SIP Trunking – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking on an enterprise solution consisting of Avaya IP Office 9.0, to interoperate with Swisscom SME SIP Trunking.

Swisscom is a member of the Avaya DevConnect Service Provider program. The Swisscom SME SIP Trunking service provides customers with PSTN access via a SIP trunk between the enterprise and the Swisscom 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 procedures for configuring Session Initiation Protocol (SIP) Trunking on an enterprise solution consisting of Avaya IP Office 9.0, to interoperate with Swisscom SME SIP Trunking.

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 Swisscom SME SIP Trunking service referenced within these Application Notes is designed for small and medium size enterprises. 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.

As a required component of the Swisscom SME SIP Trunking service offer, a SIP Enterprise Session Border Router (ESBR) server is deployed by the service provider at the enterprise site, as Customer Premises Equipment (CPE).

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 Swisscom SME SIP Trunking 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.

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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:

- 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.
- Codec G.711A.
- 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.
- Fax is not supported in the Swisscom SME SIP Trunking service offer.
- The REFER method of network call redirection is not supported on the Swisscom network.

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2.2. Test Results

Interoperability testing of the Swisscom SME SIP Trunking service with and Avaya IP Office 9.0 was completed with successful results for all test cases with the exception of the observations and limitations described below:

- **Response to OPTIONS**: During the compliance test, Swisscom was not configured to send SIP OPTIONS messages on the SIP trunk, but responded to the OPTIONS sent by the IP Office with a "401 Unauthorized" message. Since the OPTIONS messages were used to check the status of the network connectivity to the service provider, any response received from Swisscom was sufficient to achieve that purpose.
- **RFC2833 Payload Type 120** – Soft client Avaya Flare Experience for Windows uses payload type 120, which is not configurable. When using this soft client as one of the extensions at the enterprise to make outbound calls on the SIP trunk, Swisscom seems to dynamically negotiate and to accept the payload type 120 sent on the SDP offer of the outbound INVITE from the IP Office, but DTMF transmissions failed when using this specific soft client at the enterprise. This issue was sent to Swisscom for investigation. Additionally, there is a ticket currently open with Avaya IP Office development requesting the change of the payload type when using the Flare to the payload type of the SIP trunk (JIRA IPOFFICE-41776). A fix is expected in a future software release of the IP Office.

2.3. Support

For technical support and contact information on the Swisscom SME SIP Trunking service offer, visit http://www.swisscom.ch/en/business/sme/help/contact-page-sme.html

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3. Reference Configuration

Figure 1 illustrates the sample Avaya SIP-enabled enterprise solution, connected to the Swisscom SME SIP Trunking service through a public Internet WAN connection.

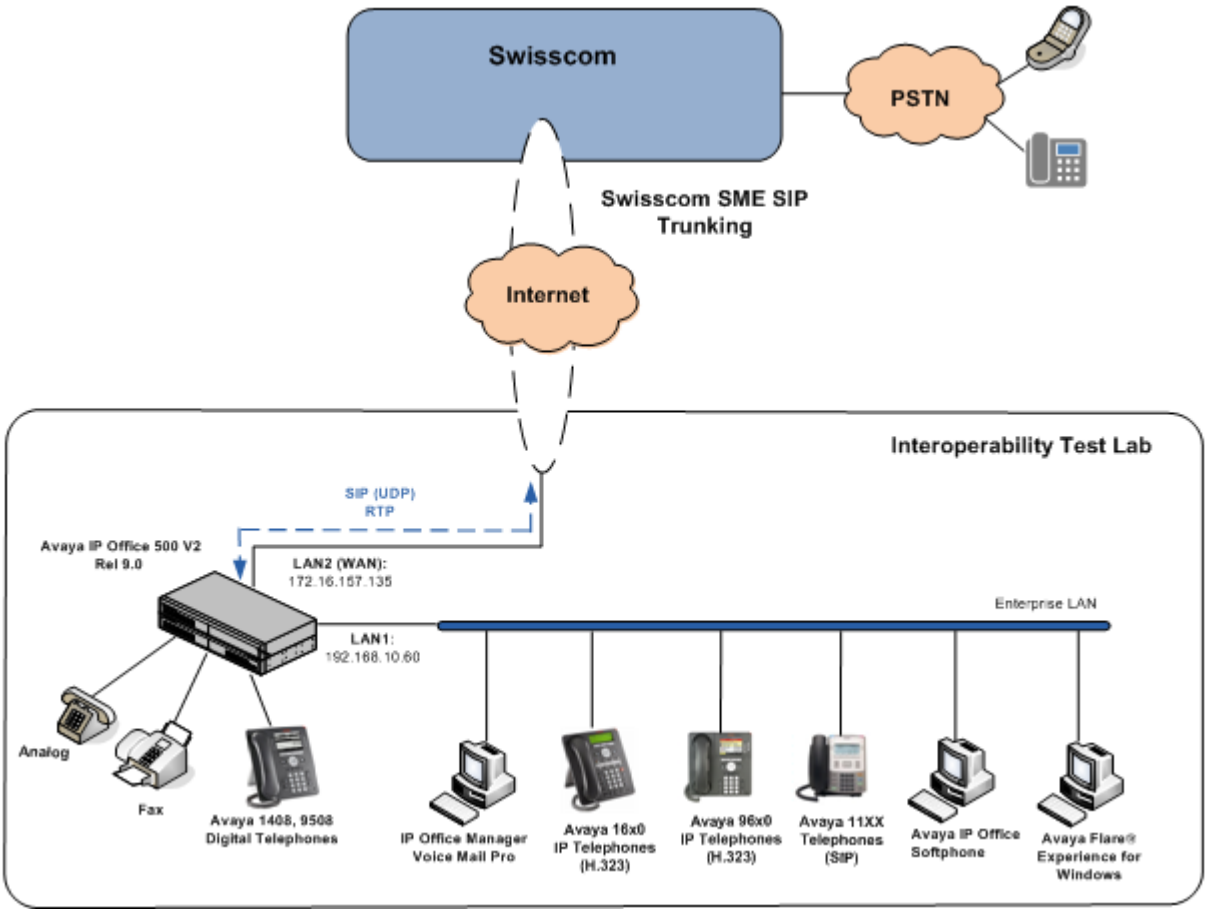

Figure 1: Test Configuration

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.

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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 the reference configuration, the Swisscom ESBR server, also known as "Connect Box", was required and installed at the enterprise site, acting as a SIP interface between the Avaya IP Office and the Swisscom network.

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 router or data firewall. A complete discussion of the configuration of these devices is beyond the scope of these Application Notes. However, it should be noted that all SIP and RTP traffic between the service provider and the Avaya IP Office system must be allowed to pass through these devices.

4. Equipment and Software Validated

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

Component	Version
Avaya	
Avaya IP Office 500v2	9.0.300.941
Avaya IP Office Digital Expansion Module DCPx16	9.0.300.941
Avaya IP Office Manager	9.0.3.0.Build 860
Avaya IP Office Voicemail Pro	9.0.300.212
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)	04.04.14.00
Avaya Digital Telephone 1408	38.0
Avaya Digital Phone 9508	0.55
Avaya IP Office Softphone	3.2.3.49.68975
Avaya IP Office Flare Experience for Windows	1.1.4.23
Swisscom	
Broadsoft Softswitch	R 16
Acme Packet SBC	Not Available
Connect Box, Patton SmartNode SN5300	Trinity 3.5.2_14085

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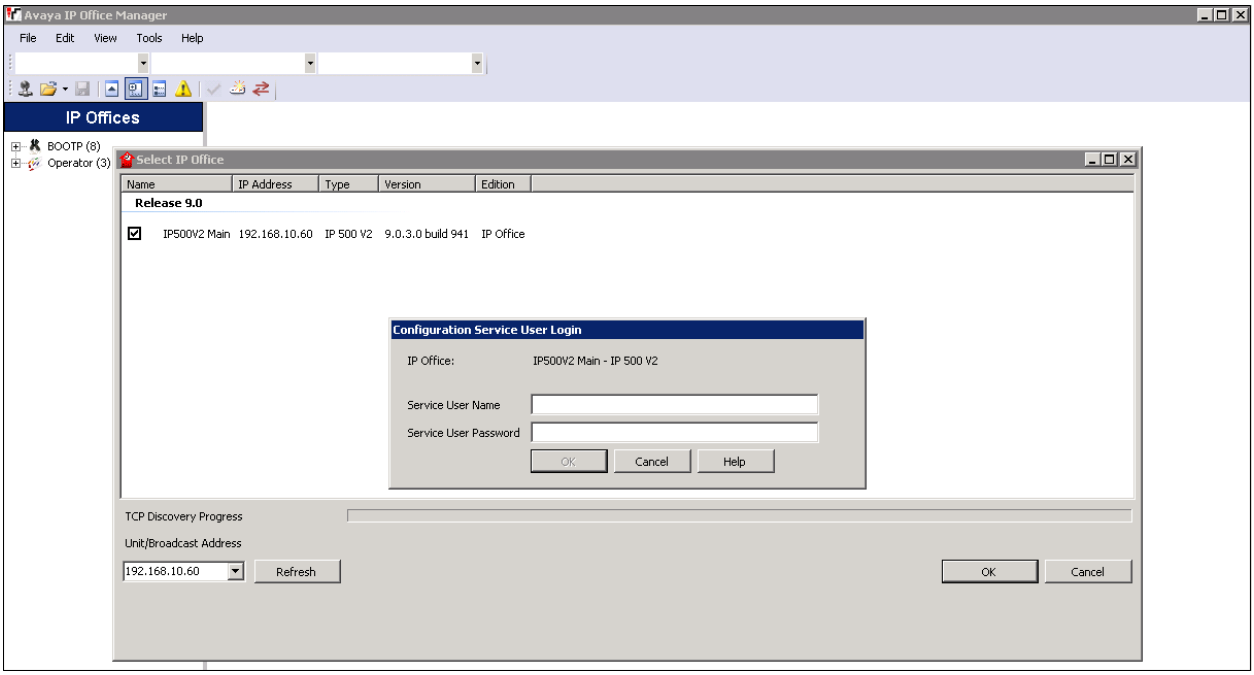
5. Configure IP Office

This section describes the Avaya IP Office configuration necessary to support connectivity to the Swisscom SME SIP Trunking 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.

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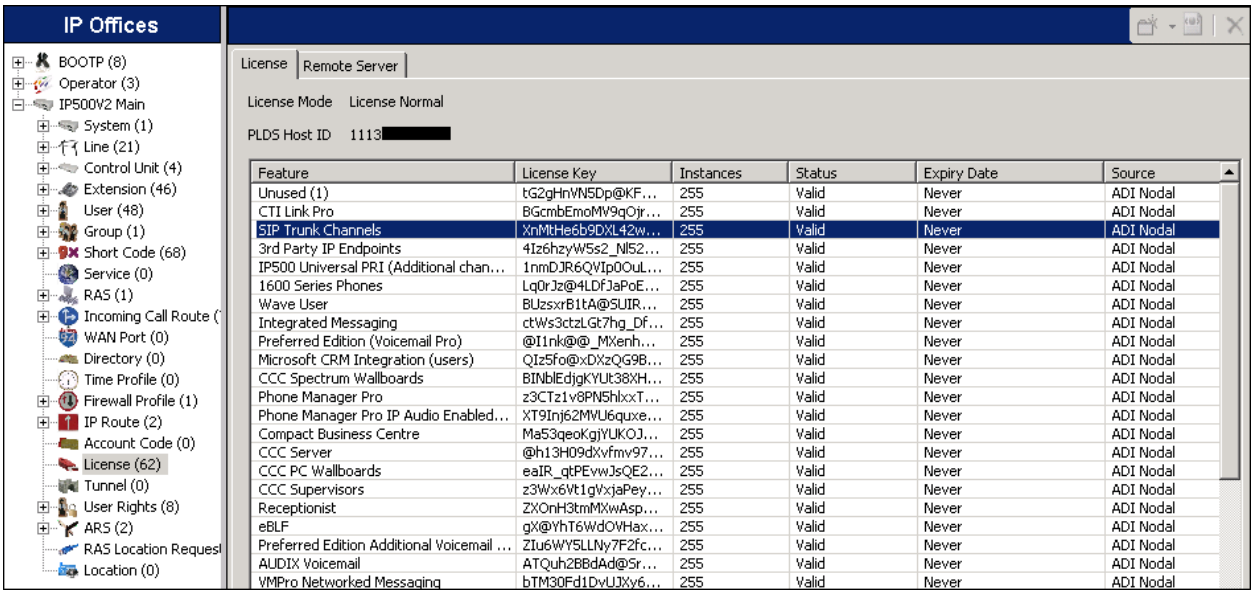
5.1. Licensing

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, ***IP500V2 Main*** was used as the system name. To verify that there is a SIP Trunk Channels License with sufficient capacity, navigate to **IP500V2 Main** in the Navigation pane and select **License**. Confirm that there is a valid license with sufficient "Instances" (trunk channels) in the Details pane.



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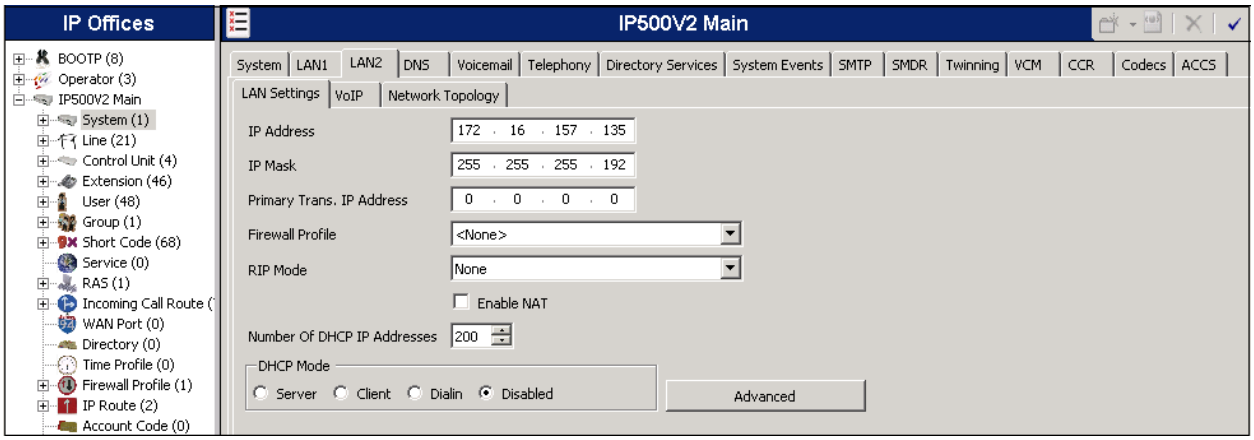
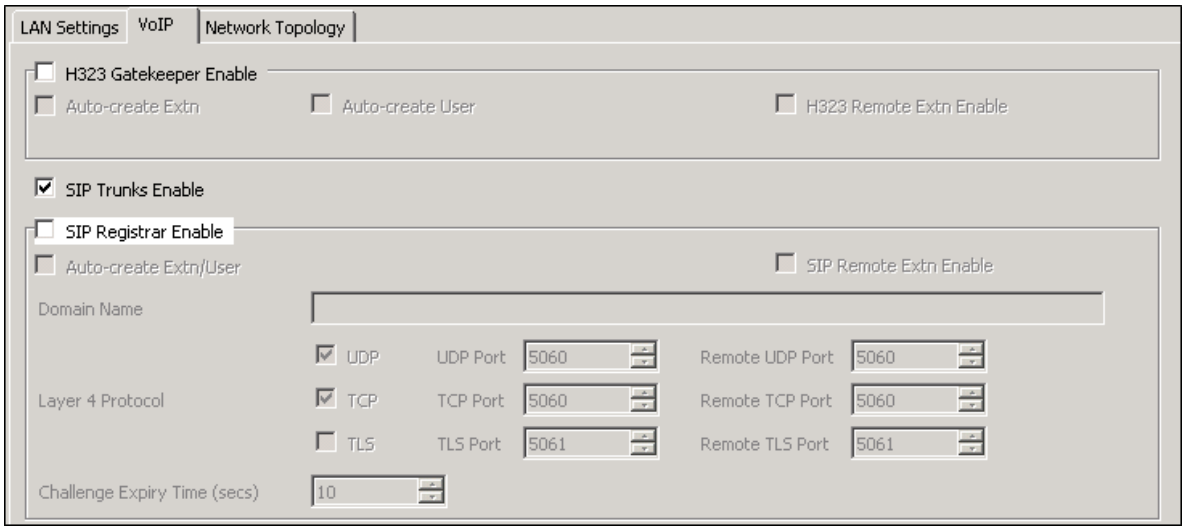
5.2. LAN2 Settings

In the reference 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 **IP500V2 Main → System (1)** 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 on this interface. Leave the **SIP Registrar Enable** box unchecked, since no SIP users are expected to register to the Avaya IP Office using this interface.



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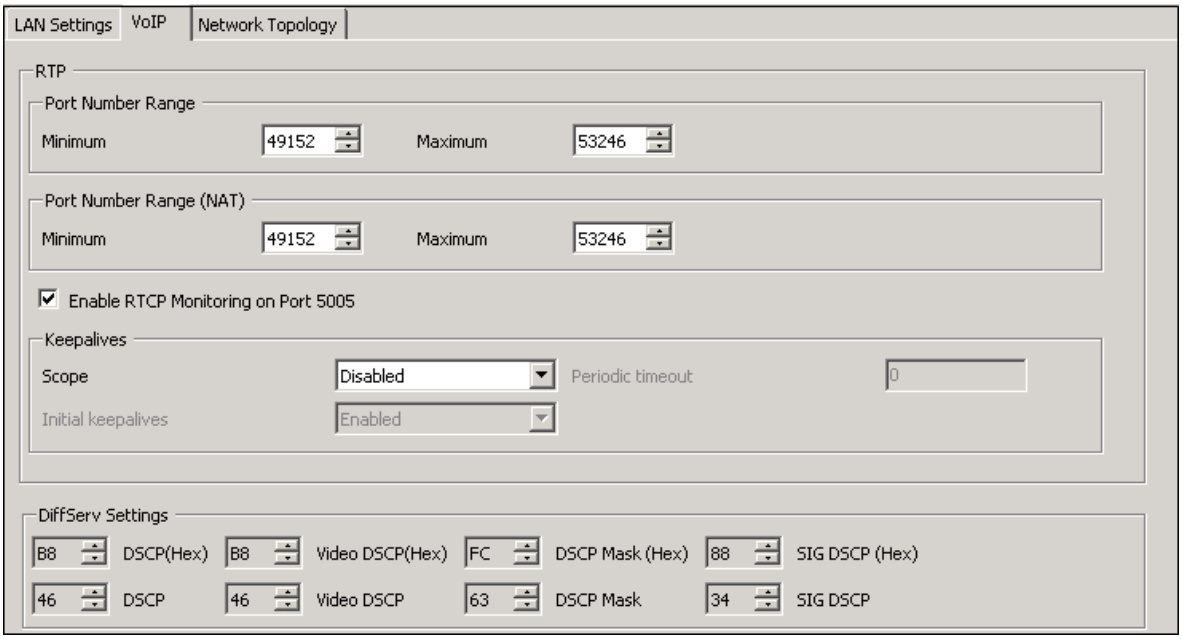
The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2.

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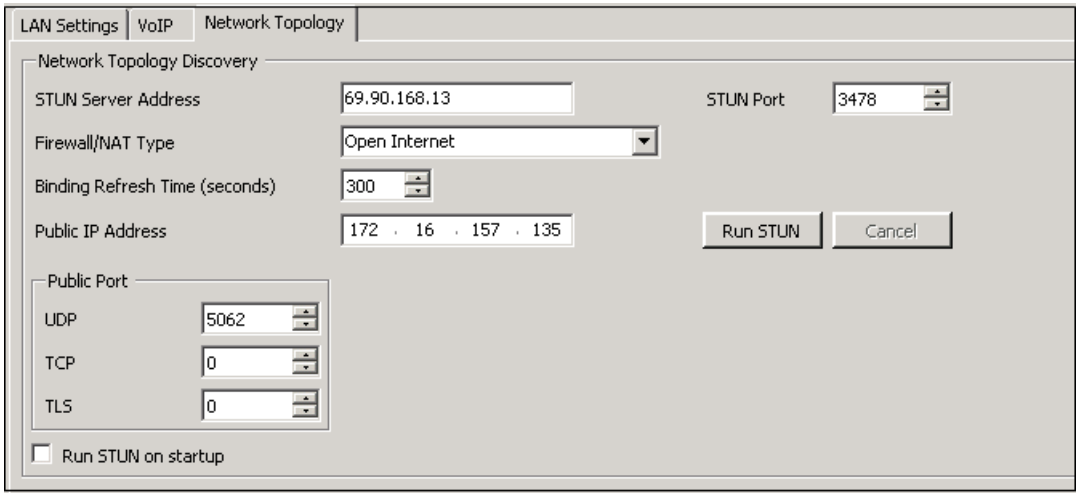
On the **Network Topology** tab in the Details pane, configure the following parameters:

- 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, 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 ***5062***, as specified by Swisscom.
- All other parameters should be set according to customer requirements.



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 to the SIP trunk to Swisscom, and therefore is not described in these Application Notes.

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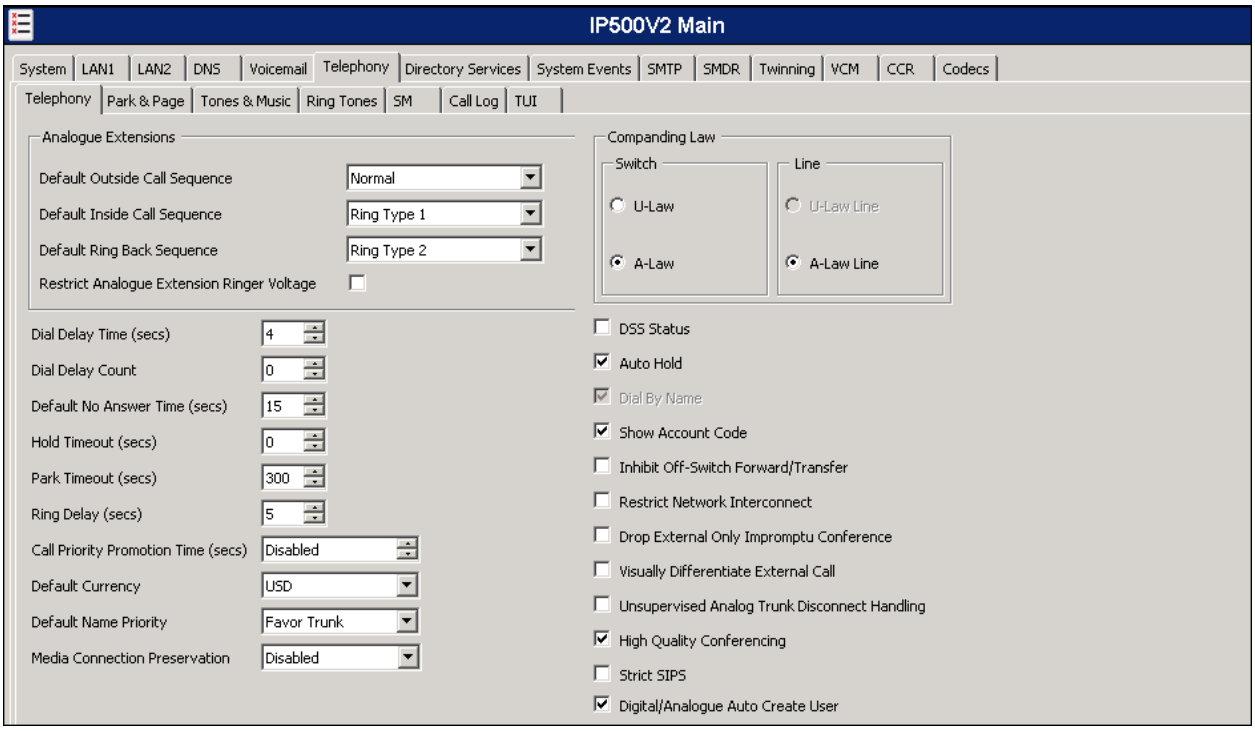
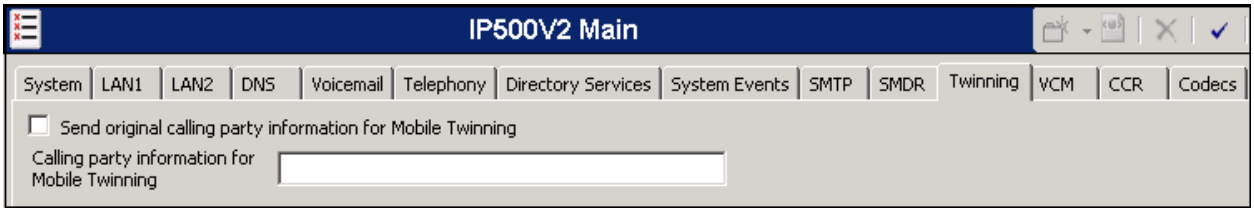
5.3. System Telephony Settings

Navigate to the **Telephony → Telephony** Tab in the Details Pane. Choose the **Companding Law** typical for the enterprise location. ***A-Law*** was 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 **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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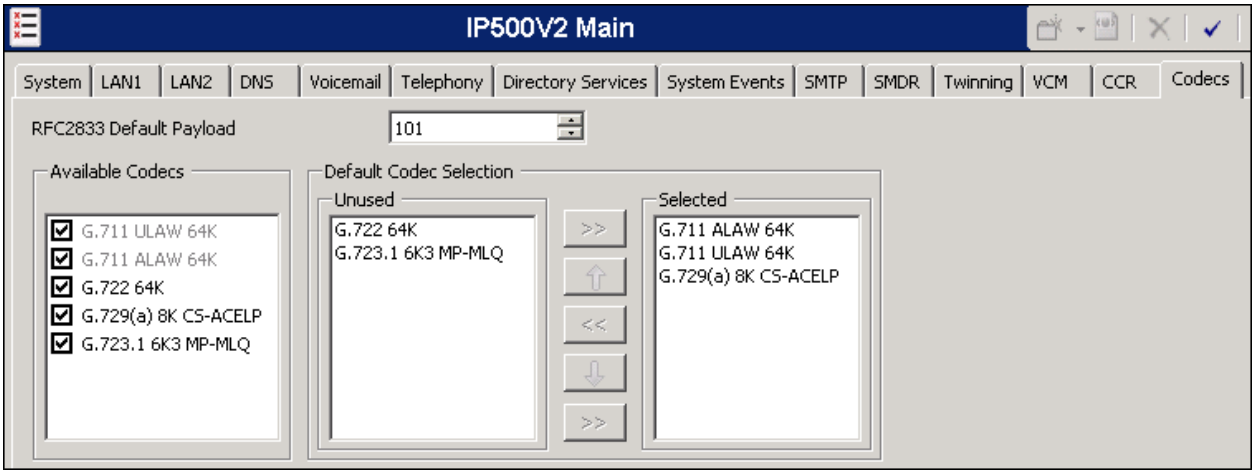
5.5. System Codecs Settings

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. Since Swisscom supports dynamic payload negotiation, 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.



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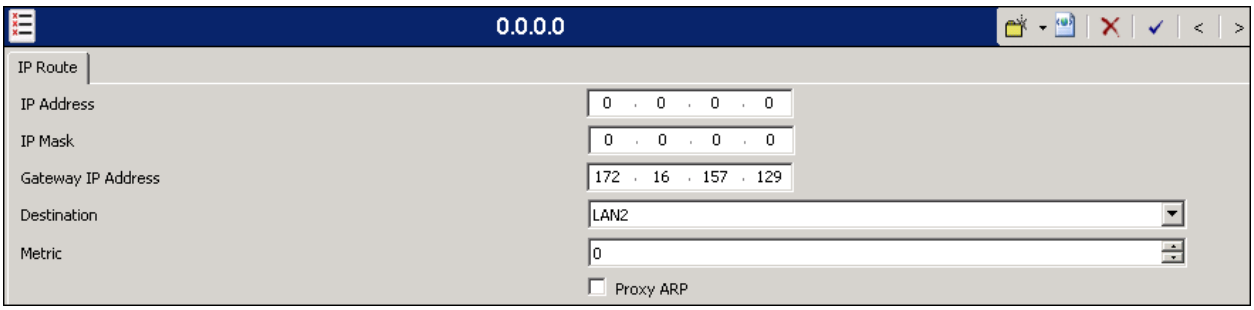
5.6. IP Route

In the reference configuration, the IP Office LAN2 interface and the Swisscom ESBR server resided on the same subnet, so an IP route was not necessary. In an actual customer configuration, these two interfaces may be in different subnets, and in that case an IP route would have to be created 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 Swisscom ESBR server is located.

To create an IP route, on the left navigation pane, right-click on **IP Route**. Select **New** (not shown).

- Set the **IP Address** and **IP Mask** of the subnet of the Swisscom ESBR server, or enter **0.0.0.0** to make this the default route.
- Set **Gateway IP Address** to the IP Address of the default router in the IP Office subnet.
- Set **Destination** to ***LAN2*** from the pull-down menu.
- Click **OK** (not shown) to save any changes.



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5.7. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and Swisscom SME SIP Trunking. 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**.

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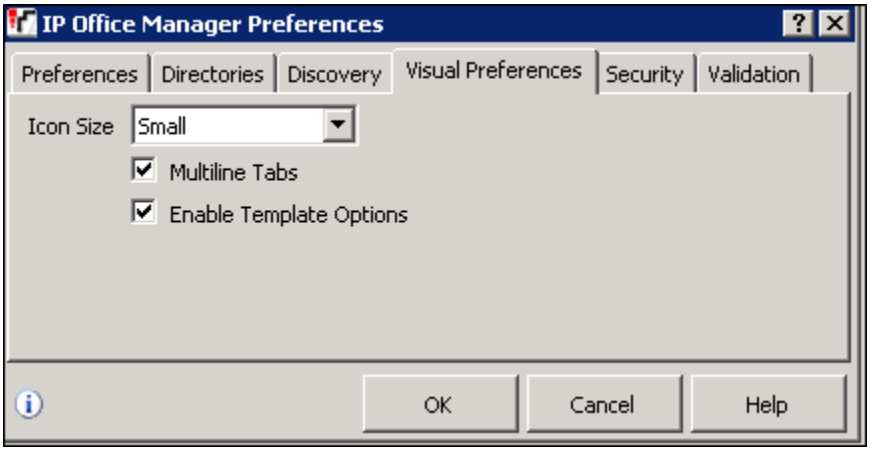
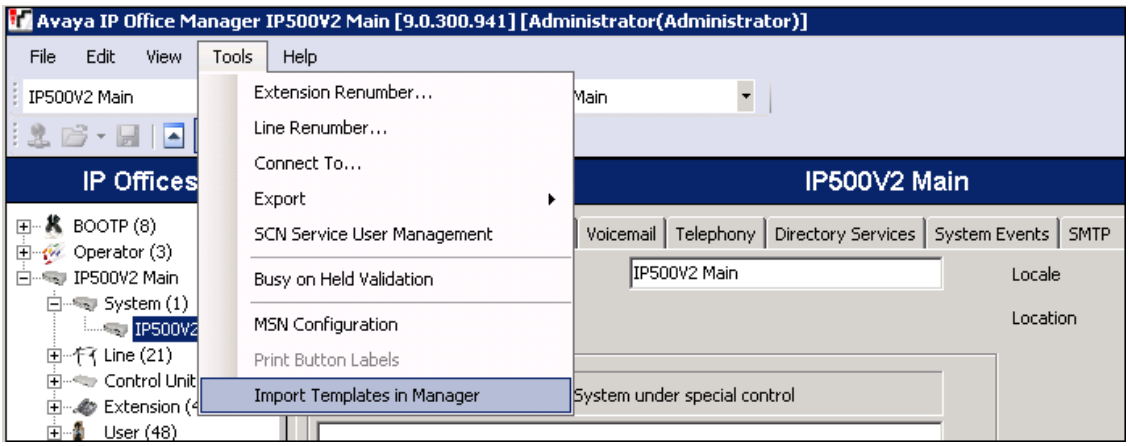
5.7.1. SIP Line From Template

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 **CH_Swisscom_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**.



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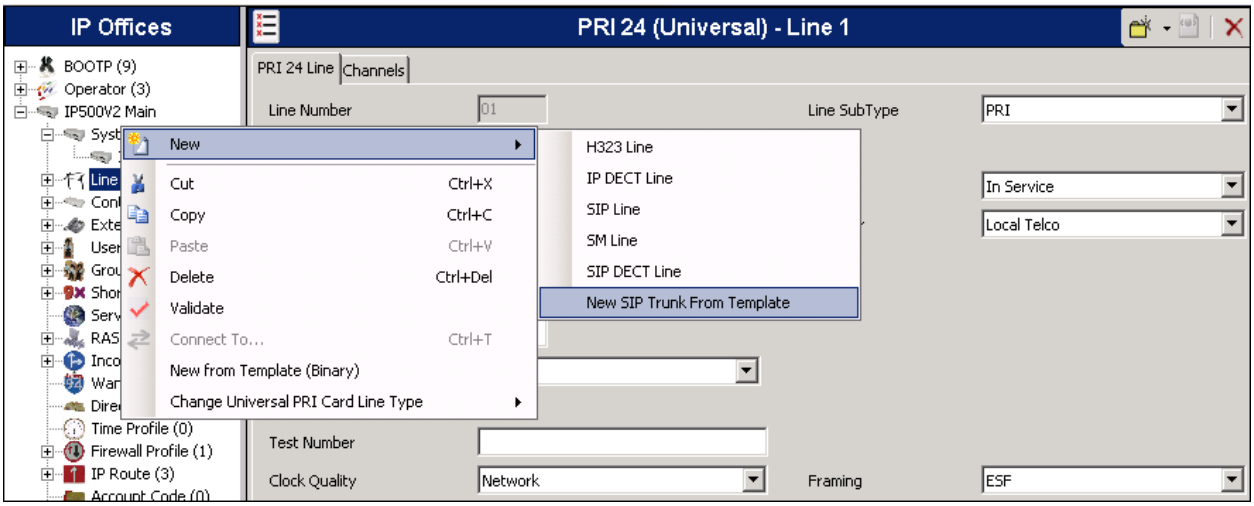
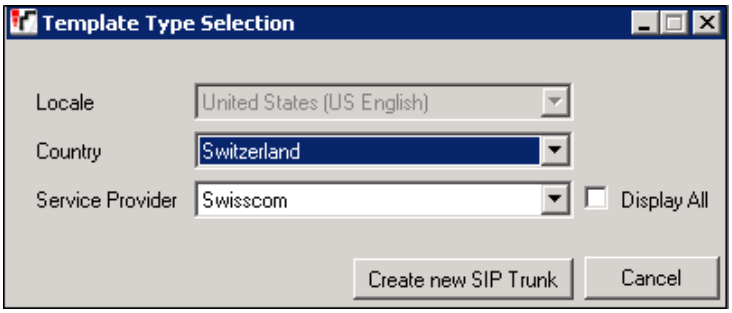
In the pop-up window (not shown) that appears, select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window (not shown) will appear stating success or failure. Click **OK** (not shown) to continue. If preferred, this step may 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 **Switzerland** from the **Country** pull-down menu and select **Swisscom** from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name created in **Step 1** (**CH_Swisscom_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.**

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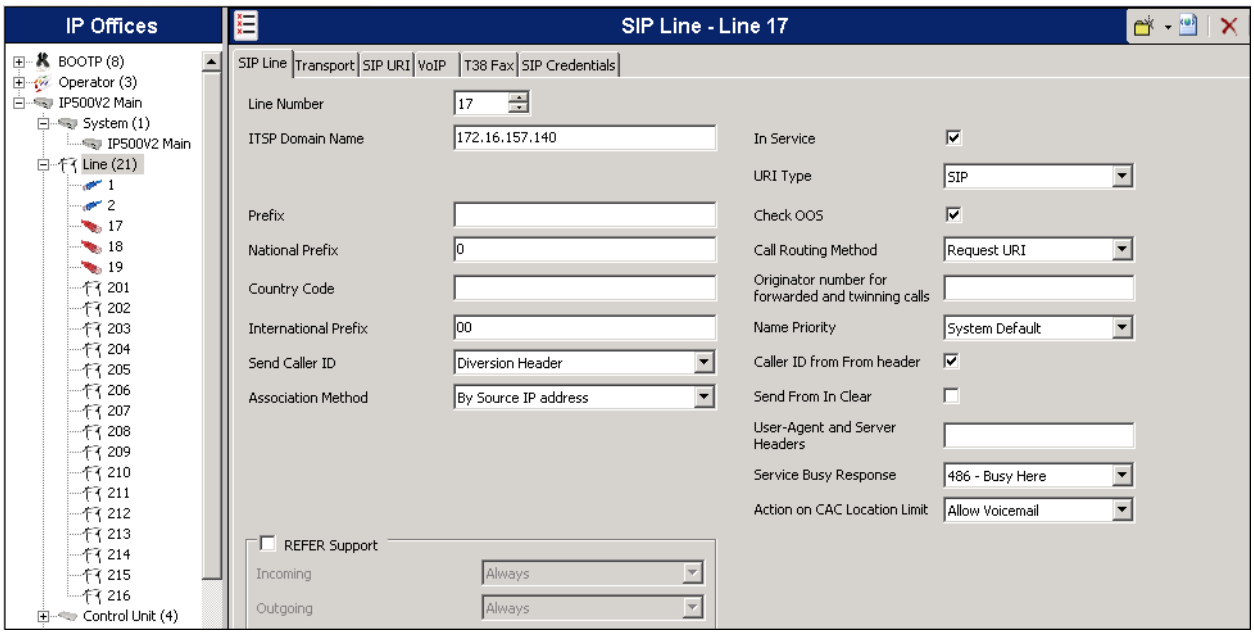
5.7.2. SIP Line Tab

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

- Set the **ITSP Domain Name** to the IP address of the Swisscom ESBR server. IP Office will use this IP address as the host portion of the SIP URI of SIP headers in messages sent to Swisscom.
- Check the **In Service** box.
- Check the **Check OOS** box. With this option selected, the 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 **Send Caller ID** to ***Diversion Header***. With this setting, Avaya IP Office will include the Diversion Header for calls that are redirected via Call Forward or Mobile Twinning out the SIP Line to Swisscom.
- Since REFER is not supported, uncheck the **REFER support** box. .
- Default values may be used for all other parameters.



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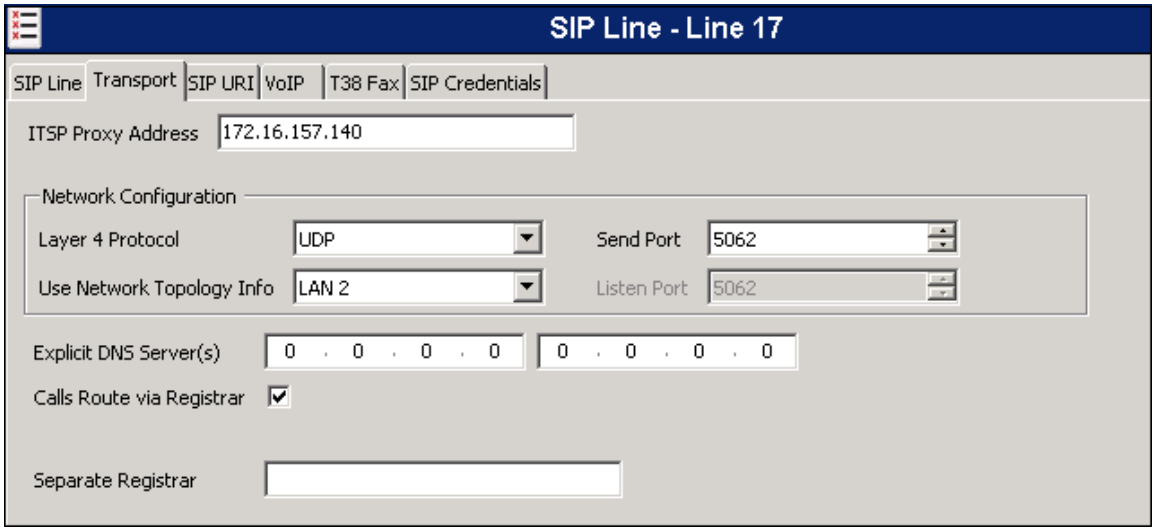
5.7.3. Transport Tab

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

- Set the **ITSP Proxy Address** to the IP address of the Swisscom ESBR 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 ***5062***, as specified by Swisscom.
- Default values may be used for all other parameters.



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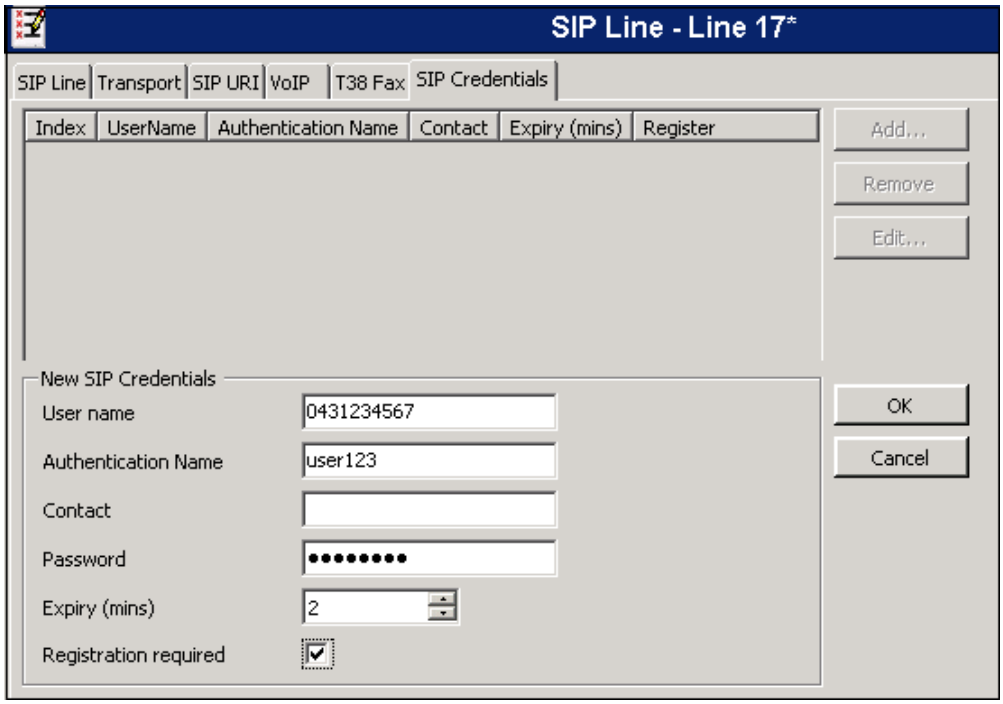
5.7.4. SIP Credentials

Swisscom required the use of SIP Credentials for the registration of the SIP trunk with the service provider. These SIP Credentials are also used to authenticate calls made from the enterprise on the SIP trunk to the PSTN.

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**, **Authentication Name** and **Password** to the values provided by the service provider. For the compliance test the User name was the trunk network ID, the pilot DID number assigned by Swisscom to the SIP trunk.
- Set the **Expiry (mins)** field to **2**. Swisscom required the registration to be renewed every 120 seconds.
- Check the **Registration required** box.
- Click **OK**.



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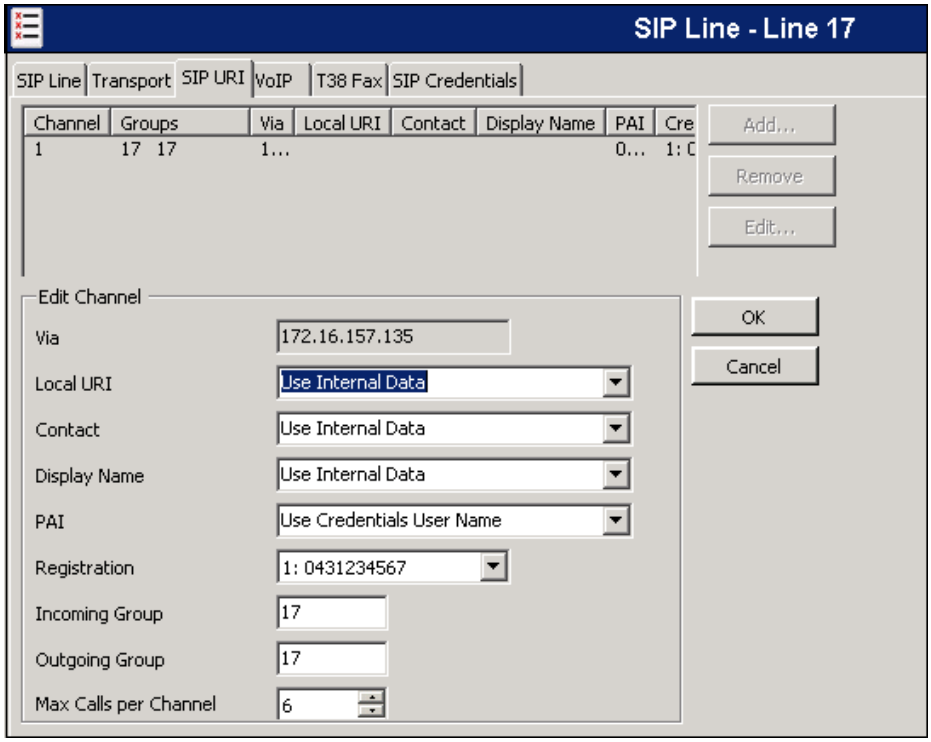
5.7.5. SIP URI Tab

A SIP URI entry needs to be created to match each number that Avaya IP Office and the service provider 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. In the example screen below, a previously configured entry was edited to use the parameters shown below:

- Set **Local URI**, **Contact** and **Display Name** to ***Use Internal Data***. This setting allows calls on this line that have a SIP URI that matches the number set in the **SIP** tab of any user as shown later in **Section 5.8**.
- Set **PAI** to ***Use Credentials User Name***. Swisscom required the trunk network ID to be sent in the PAI header of all requests sent to the network.
- Under **Registration**, select ***1: <0431234567>*** from the pull-down menu. Swisscom uses Digest Authentication to challenge all calls made from the enterprise to the PSTN.

  IP Office will use this set of SIP credentials, defined in the previous section, to authenticate outbound calls to the service provider.
- 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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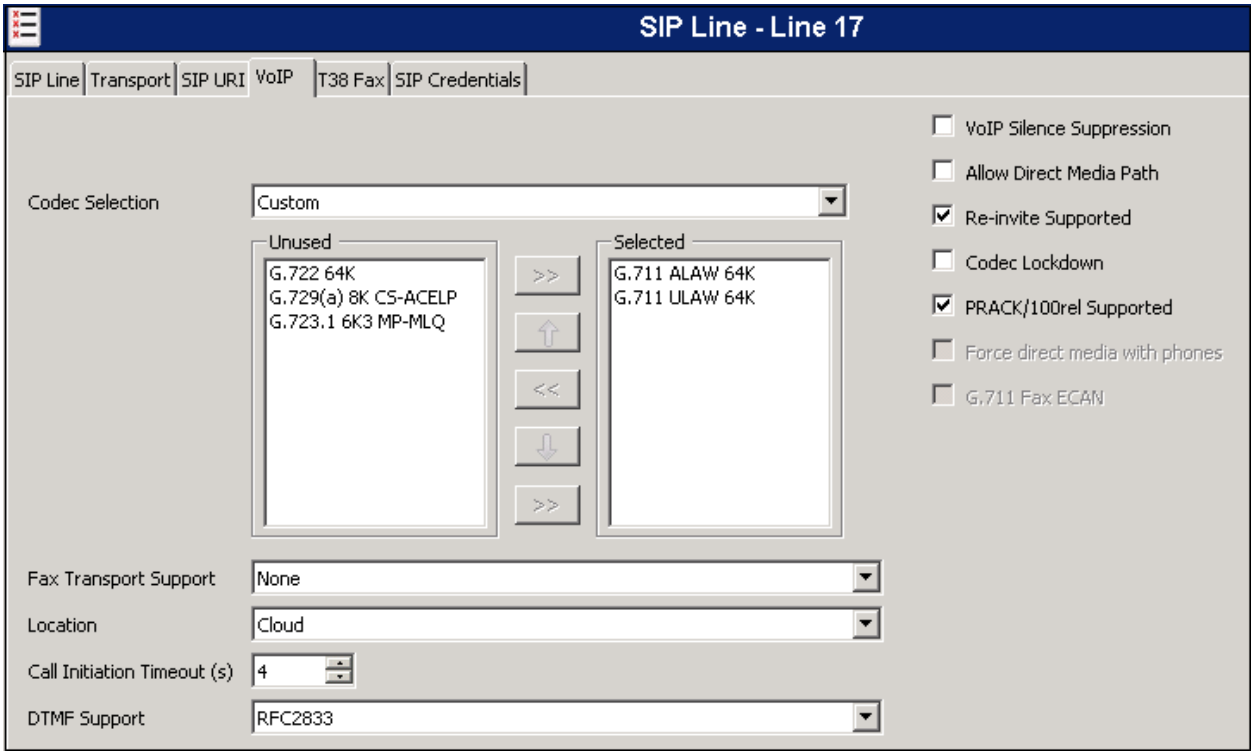
Additional SIP URIs may be required to allow inbound calls to numbers not associated with a user, such as a short code. These URIs are created in the same manner as shown previously, with the exception that the incoming DID number is entered directly in the **Local URI**, **Contact**, and **Display Name** fields, and only the **Incoming Group** needs to be associated to the SIP line.

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.
- Since fax is not supported by Swisscom, set **Fax Transport Support** to ***None***.
- Set the **DTMF Support f**ield 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.
- Click **OK** (not shown) to save any changes made to any of the various "SIP Line" tabs.



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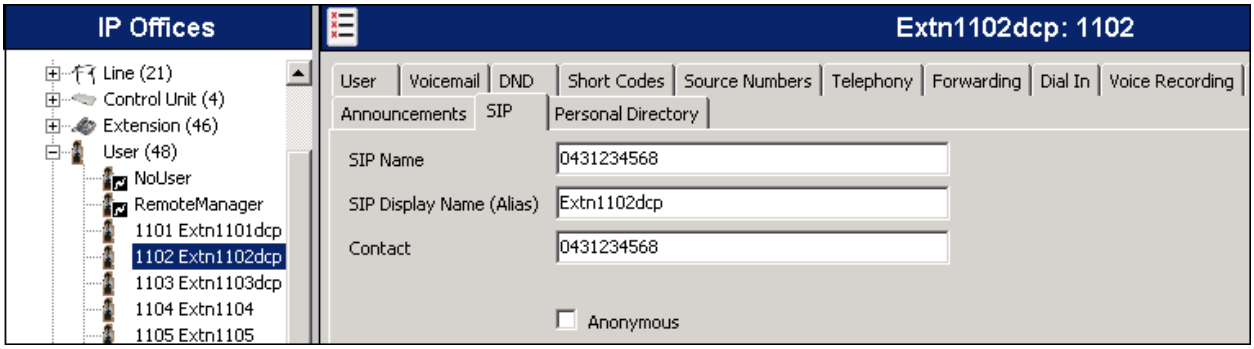
5.8. Users

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.7**. To configure these settings, navigate to **User** in the left Navigation Pane and select the name of the user to be modified. In the example below, the name of the user is ***Extn 1102dcp***. Select the **SIP** tab in the Details Pane.

The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From and Contact headers for outgoing SIP trunk calls. In addition, these settings are used to match against the SIP URI of incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.7.5**). The example below shows the settings for user "Extn1102dcp". The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise by Swisscom. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. Click **OK** (not shown) to save any changes.



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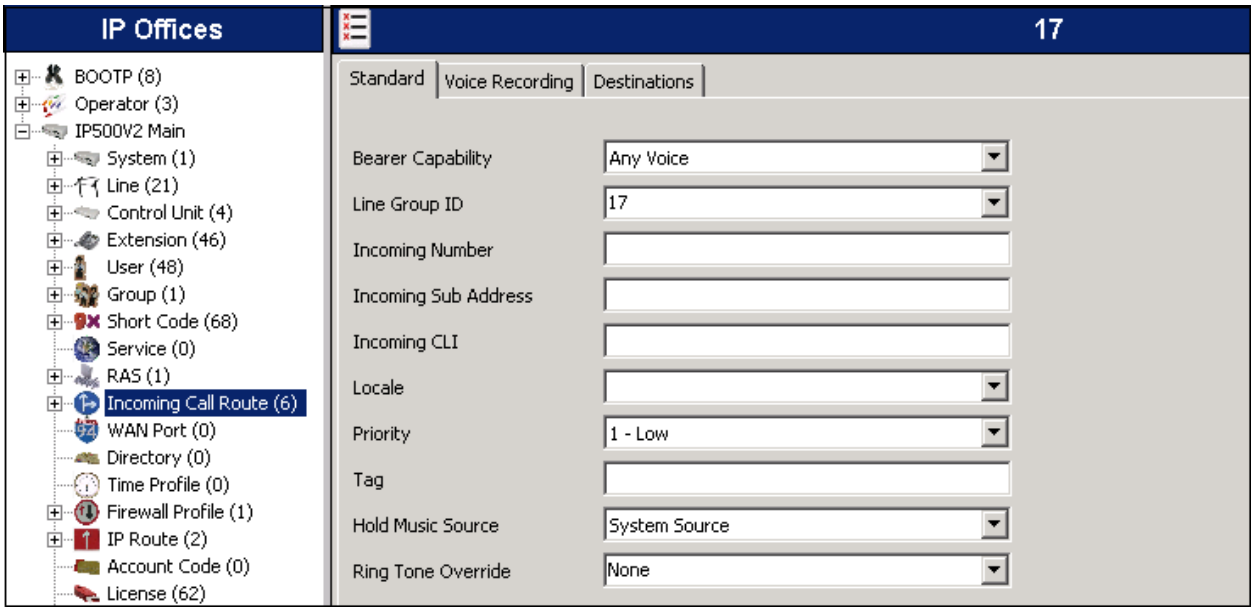
5.9. Incoming Call Route

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.

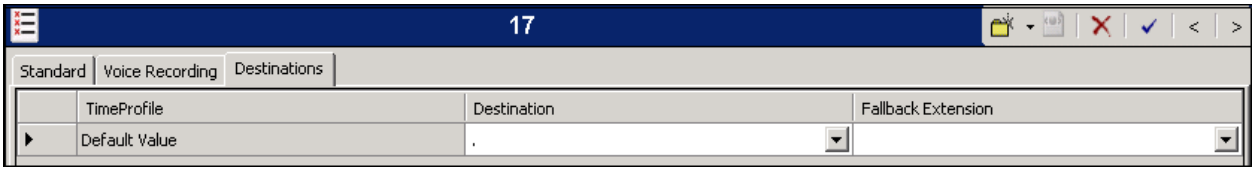
In a scenario like the one used for the compliance test, only one incoming route was needed, which allowed any incoming number arriving on the SIP trunk to reach any predefined extension in the IP Office. The routing decision for the call is based on the parameters previously configured for the **SIP URI (Section 5.7.5)** and the users **SIP Name** and **Contact**, already populated with the assigned Swisscom DID numbers **(Section 5.8)**

To add a new incoming call route, from the left Navigation Pane, right-click on **Incoming Call Route** and select **New** (not shown). 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**.
- Default values may be used for all other parameters.



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



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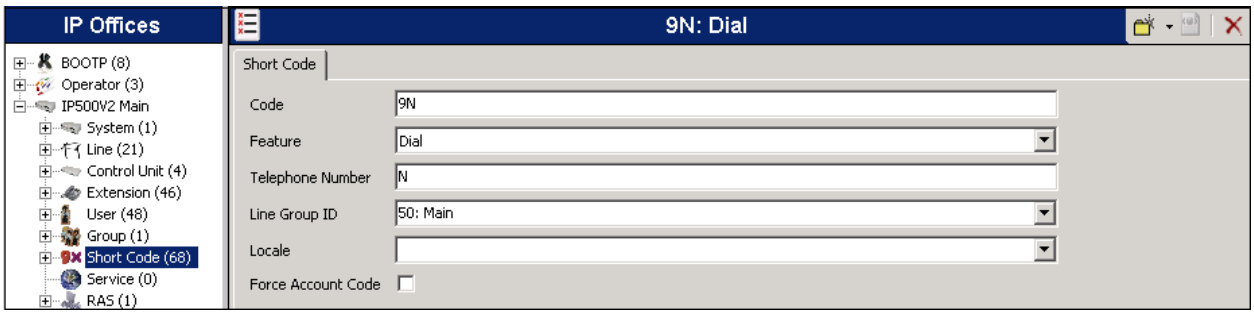
Additional incoming call routes may be required to allow inbound calls to numbers not associated with a user, such as a short code. These routes are created in the same manner as shown, with the exception that the incoming DID number is entered directly in the **Incoming Number** field on the **Standard** tap, and the specific destination (short code, etc.) needs to be entered on the **Default Value** field of the **Destinations** tab. Click **OK** (not shown) to save any changes.

5.10. Short Code

Avaya IP Office uses 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 5.11**

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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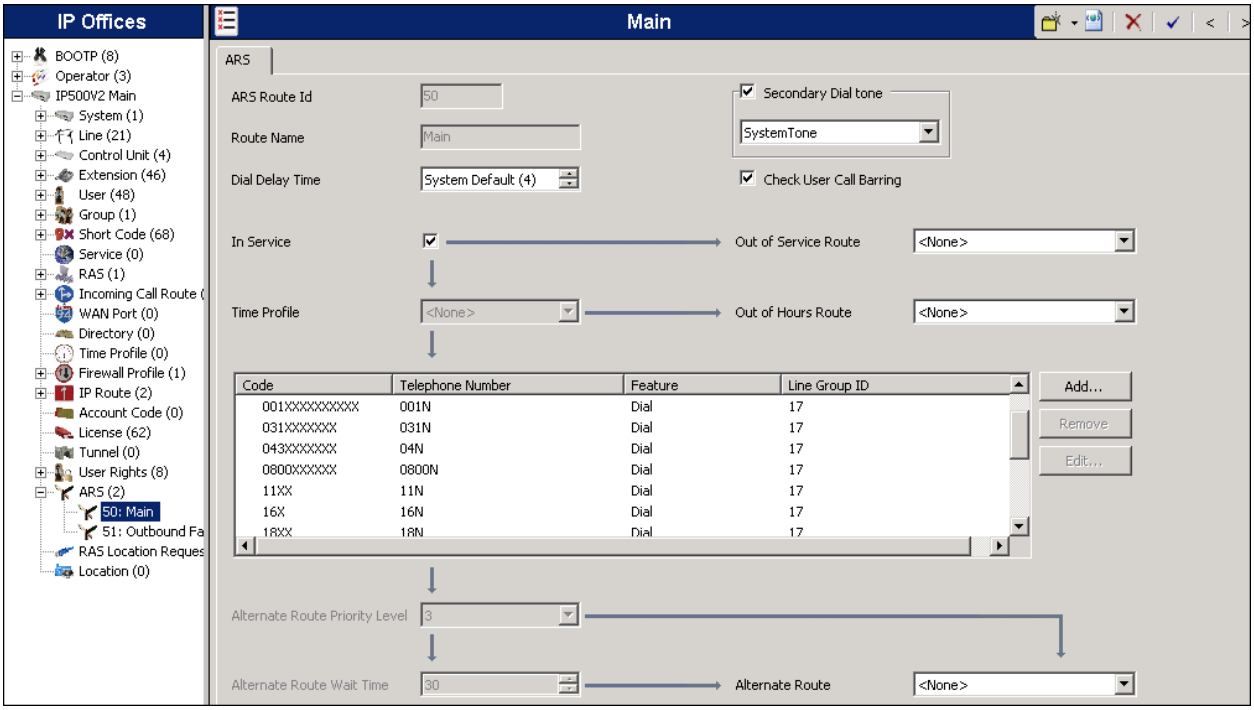
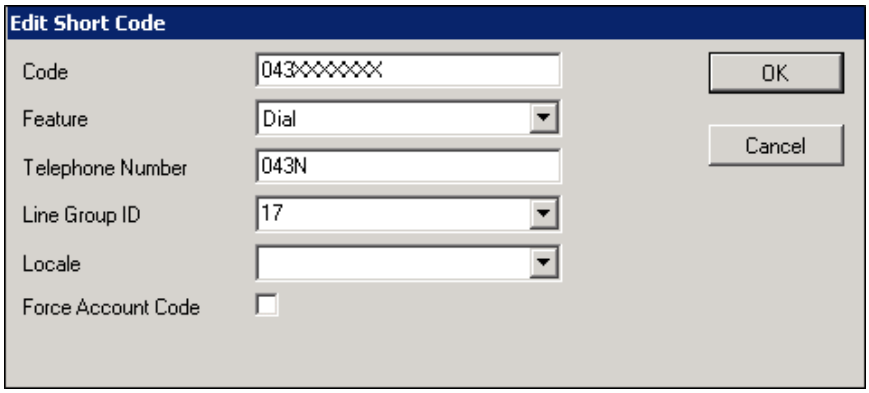
5.11. Automatic Route Selection

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***. Note the sequence of ***X***s 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 digits on the string. This type of setting results in a much quicker response in the delivery of the calls by the IP Office.



For example, for local calls, the user dialed 9 plus the 10 digit local number, starting with the area code 043 and then the remaining 7 digits. Other entries were created for different are codes, international calls, toll free, information, etc.



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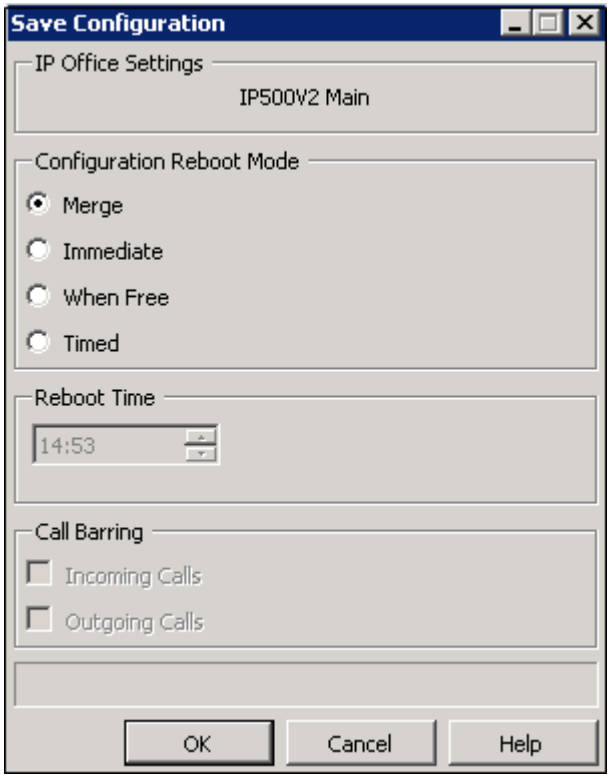
5.12. Save Configuration

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. Swisscom SME SIP Trunking Configuration

The service provider is responsible for the configuration of the Swisscom SME SIP Trunking service, including the provisioning of the Swisscom ESBR server (Connect Box) in the customer's network. Swisscom will require from the customer the information needed to configure the service, which includes:

- IP address used to reach the Avaya IP Office at the enterprise. In the reference configuration, this is the IP Office LAN2 IP address.
- IP address of the default gateway/router and DNS server used to reach the public network

Swisscom will provide the customer the necessary information to configure the Avaya IP Office SIP trunk connection, including:

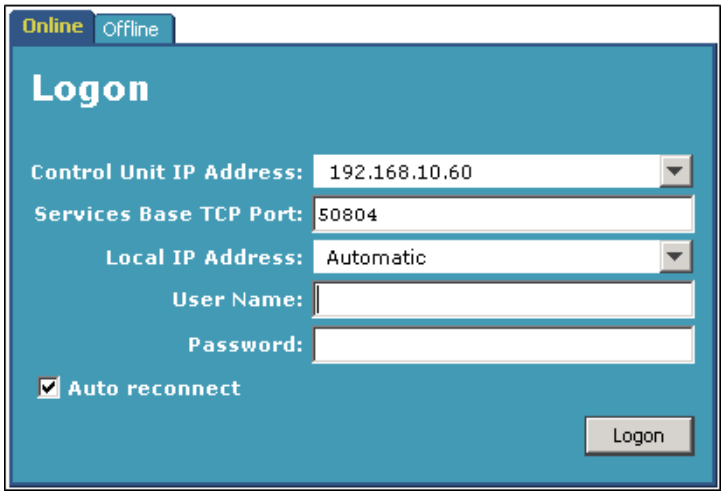
- IP address, protocol and port used to reach the Swisscom ESBR server
- Credentials for the SIP trunk registration.
- Supported codecs and order of preference.
- DID numbers.
- All IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security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



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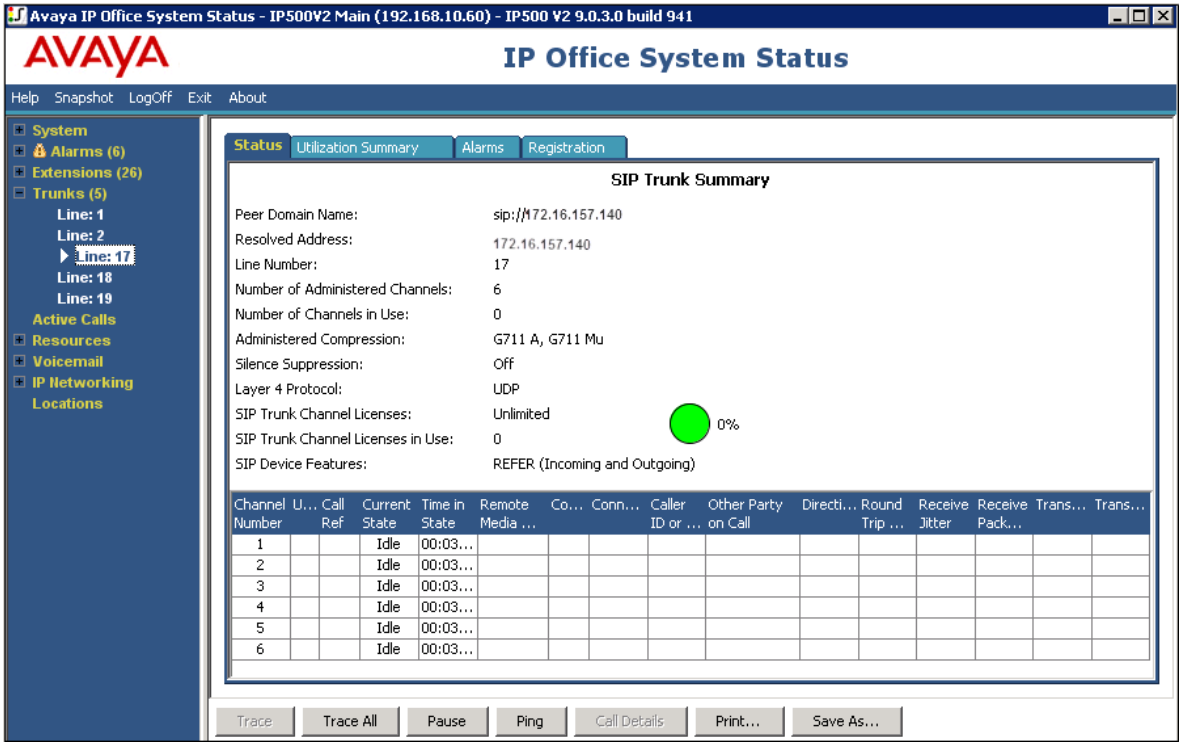
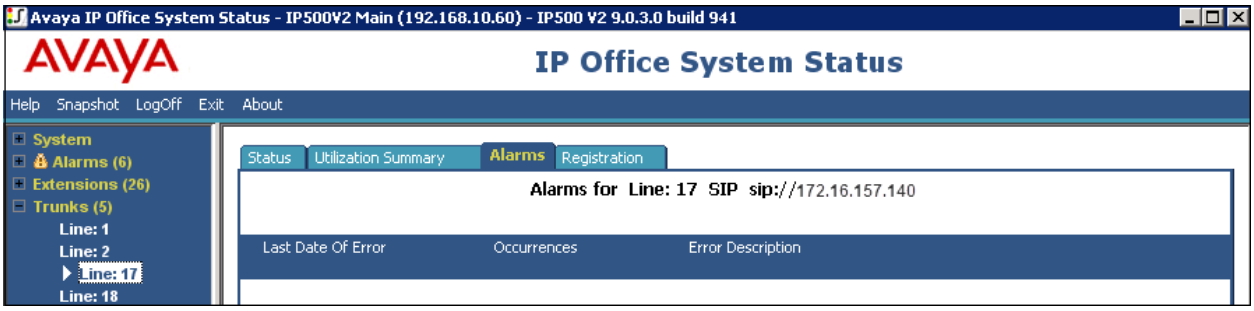
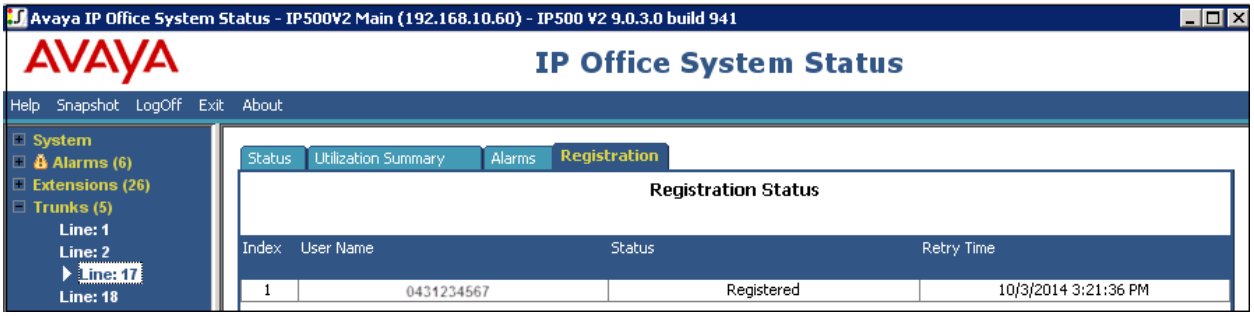
Select the SIP line of interest from the left pane (**Line 17** in the reference configuration). On the **Status** tab in the right pane, verify that the **Current State** is ***Idle*** for each channel (assuming no active calls at present time).



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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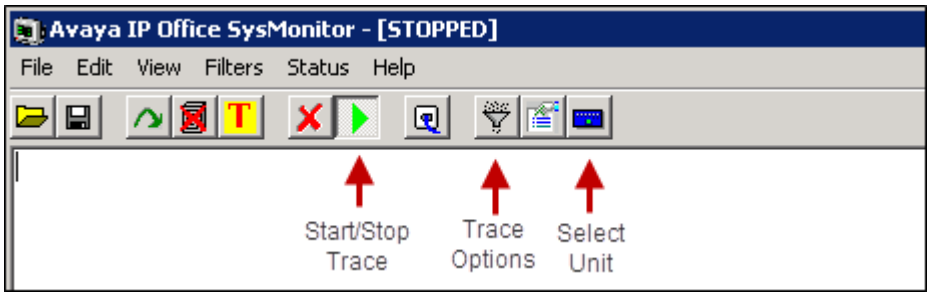
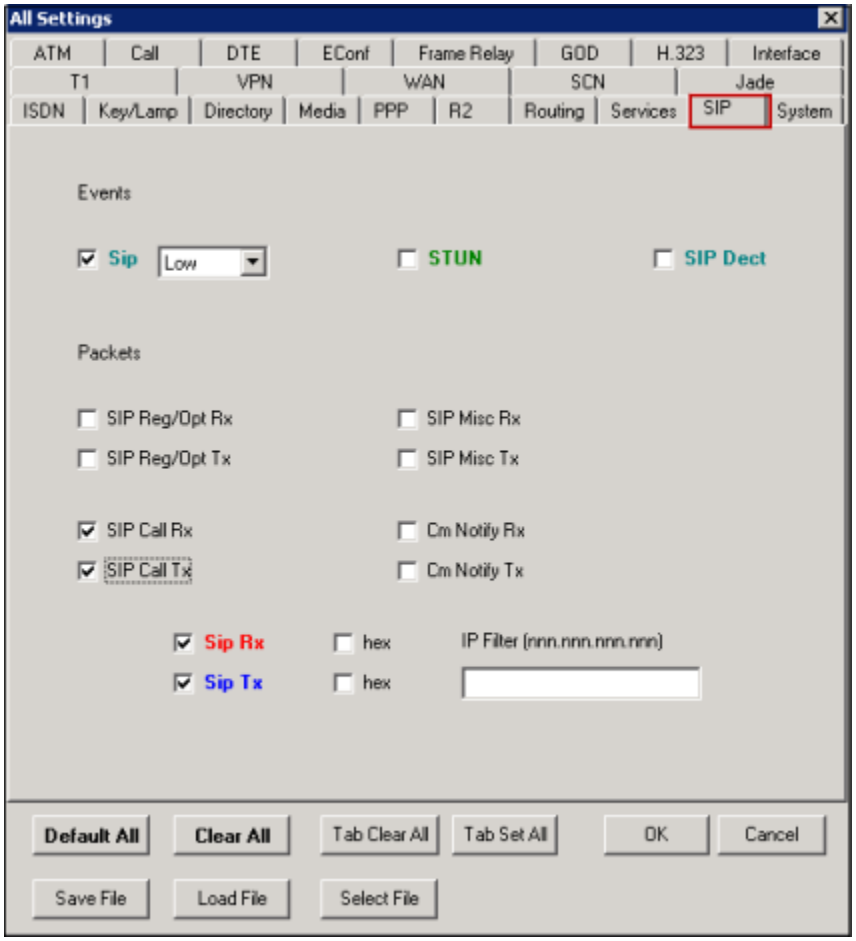
7.2. Monitor

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 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.



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8. Conclusion

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between Swisscom SME SIP Trunking and Avaya IP Office 9.0, 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 9.0.3 Installing IP500/IP500* V2. Document 15-601042, June 2014
https://downloads.avaya.com/css/P8/documents/100174004

[2] *Avaya IP Office Manager Release 9.0.3*, Document 15-601011, May 2014
https://downloads.avaya.com/css/P8/documents/100174478

[3] *Administering Avaya Flare® Experience for IPad devices and Windows, Release 9.0*, May 2014
https://downloads.avaya.com/css/P8/documents/100175132

[4] *IP Office 9.0 Using System Status*, Document Number 15-601758, August 2013
https://downloads.avaya.com/css/P8/documents/100173994

[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 Swisscom SME SIP Trunking service is available from Swisscom.

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