Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 9.1 to support MTS SIP Trunking Service – 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 MTS SIP Trunking Service.

The SIP Trunking service offered by MTS 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.

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the MTS SIP Trunking Service 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 MTS SIP Trunking Service referenced within these Application Notes is designed for business customers. 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 MTS SIP Trunking 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.

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, long distance national, long distance international, outbound toll free, operator (0), operator assisted calls (0+10) and local directory assistant.
- Codec 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:

- Inbound toll-free and emergency calls are supported but were not tested as part of the compliance test.
- Codec G.729 is not currently supported.

2.2. Test Results

Interoperability testing of the MTS SIP Trunking Service was completed with successful results for all test cases with the observations and limitations described below:

- **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, MTS responded sending back a "487 Request Terminated" 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.
- **Outbound Calling Party Number (CPN) Block**: When an enterprise user activated "Withhold Number" on an outbound call for privacy purposes, IP Office sent "anonymous" in the From header, included the "Privacy:id" header, and the complete DID number in the P-Asserted-Identity (PAI) and Contact headers of the outbound INVITE. But calls with "anonymous" in the From header are not allowed by MTS, which rejected the call with a "403" error message.
- **Call transfer to the PSTN using REFER**: Blind call transfers of incoming calls that are transferred back to the PSTN using the REFER message did not work properly. The REFER message sent by the enterprise was accepted by MTS with a 202 response, but the call between the two final endpoints dropped.
  REFER was left disabled in the IP Office for the tests. Blind and attended call transfers to the PSTN completed successfully with REFER disabled. The caveat is that the IP Office was not released from the call path after the call was transferred, and two trunks remained occupied for the complete duration of the call.
- **Outbound Fax**: On outbound fax calls using the T.38 protocol, it was observed that during the negotiation process for the transition from a G.711 voice setup to a T.38 fax call, the MTS Genband softswitch sends in the SDP of its re-INVITE to the IP Office both the m lines for audio and image (T38), with the audio m line assigned a value of "0" and the image line assigned a valid port. The IP Office sends a 200OK with SDP containing an m audio line only. A subsequent re-INVITE from the IP Office with m image line is rejected by MTS with a "500 Internal Server Error" message.
  Ticket **JIRA IPOFFICE-98597** was opened with the IP Office development team to investigate this issue, and it is pending a resolution.
  As a workaround to this situation, the extensions used for faxing in the IP Office can be set as "Fax Machine" in the Extension/Analogue tab. By doing this, T.38 is offered on the SDP of the initial INVITE from the IP Office, avoiding the transition from voice to fax. Outbound and inbound T.38 fax calls were tested successfully by using this setting.
  The issue described above also affected the testing of outbound fax in G.711 pass-through mode. MTS uses T.38 fax with fallback to G.711 on its network. If a user doesn't support T.38 fax, MTS expects a 488 or similar error code from the user to their T.38 offer (with the two m lines on its SDP), to then send a new INVITE with a G.711 offer. IP Office sends instead a 200 OK with an audio m line only to MTS and the fax call fails. The workaround described above does not address this behavior.
  G.711 fax should not be used with this solution at this time.

3. Reference Configuration

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

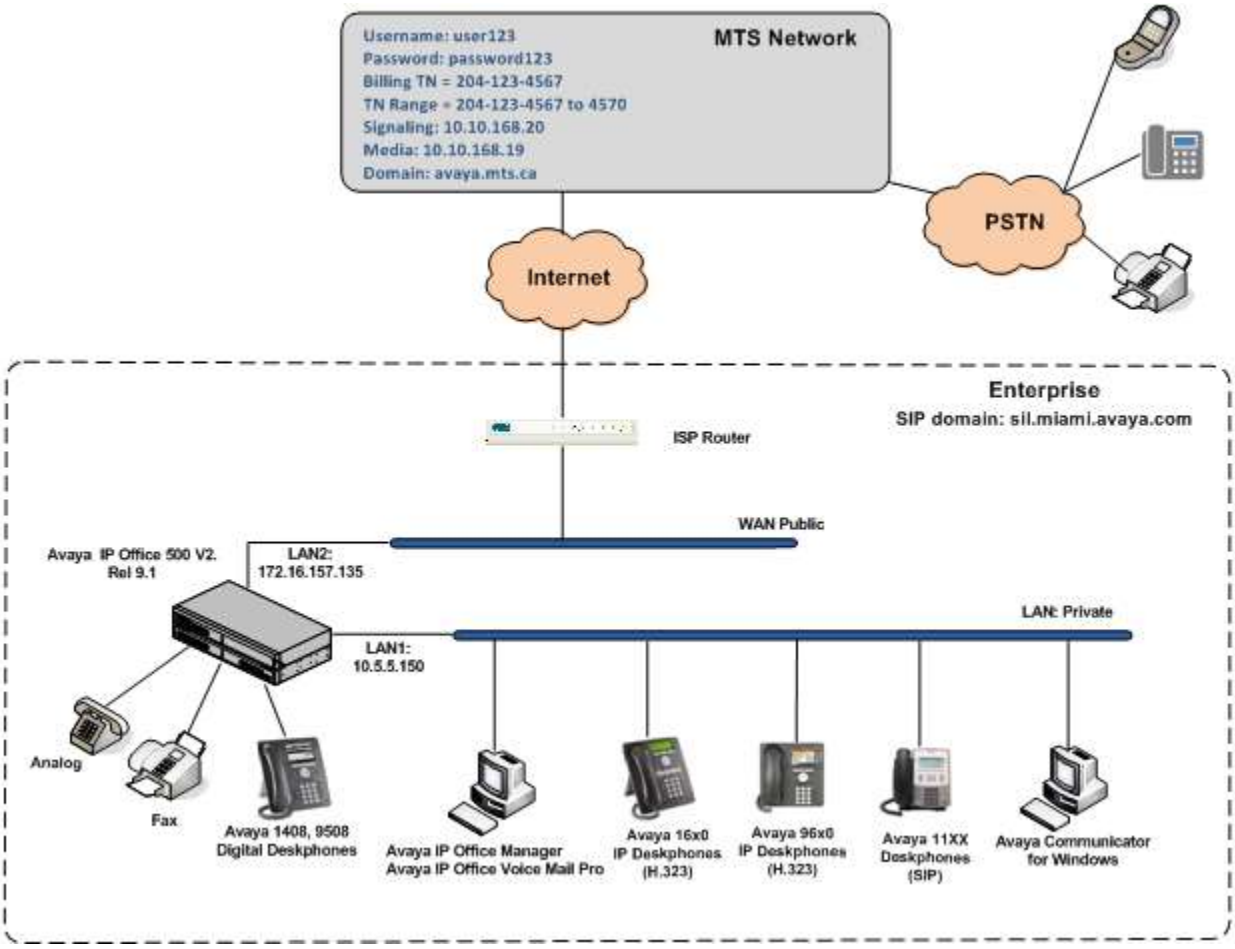


Figure 1: Test Configuration

Note that for security purposes, all public IP addresses, SIP trunk credentials and DID numbers shown throughout these Application Notes have been edited so the actual values used 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 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.

4. Equipment and Software Validated

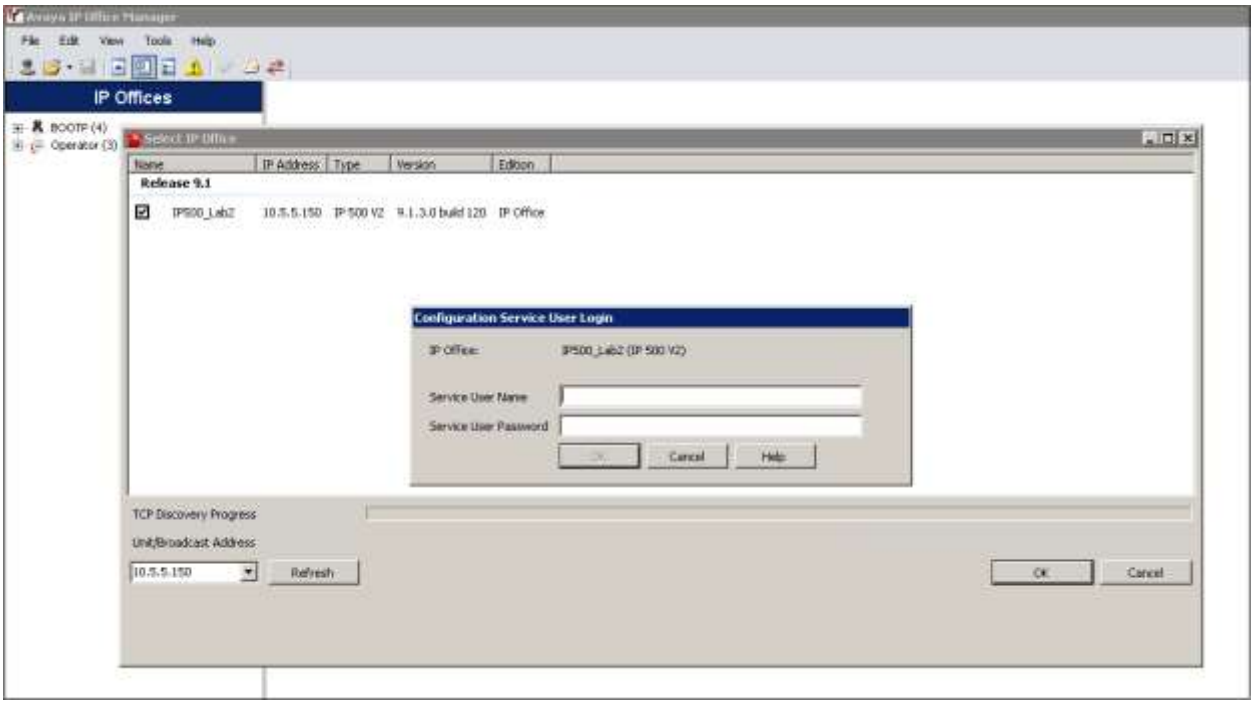
The following equipment and software were used for the sample configuration provided:

Component	Version
Avaya	
Avaya IP Office 500v2	9.1.300.120
Avaya IP Office Digital Expansion Module DCPx16	9.1.300.120
Avaya IP Office Manager	9.1.3.0.Build 120
Avaya IP Office Voicemail Pro	9.1.300.2
Avaya 1608 IP Deskphone (H.323)	1.360A
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 Deskphone 9508	0.55
Avaya Communicator for Windows	2.0.3.30
MTS Inc.	
Genband A2/Experius	10.3
Genband S3/Q20	v8.3.8.4

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.

5. Configure Avaya IP Office

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

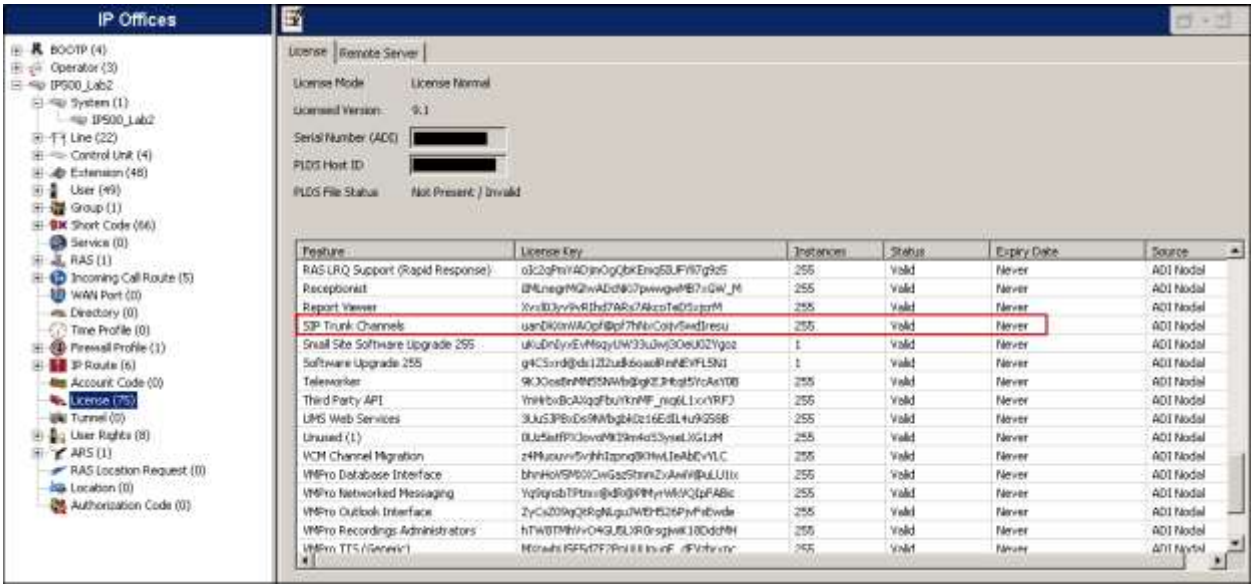
The appearance of the IP Office Manager can be customized using the View menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side and the Details pane on the right side. These panes will be referenced throughout the Avaya IP Office configuration.

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.

5.1. Licensing

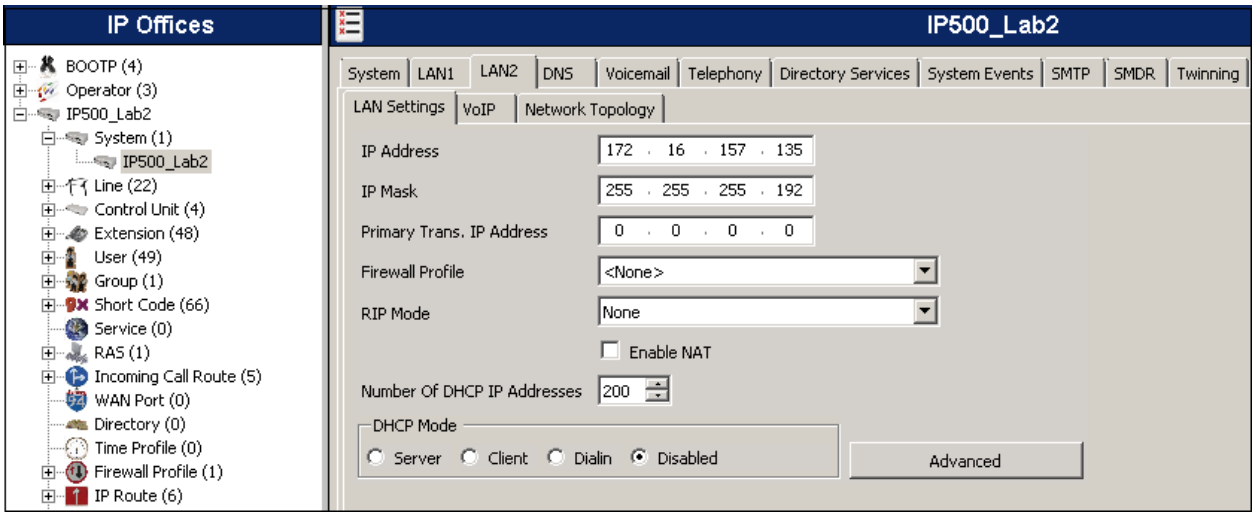
The configuration and features described in these Application Notes require the IP Office system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

In the reference configuration, ***IP500_Lab2*** 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.

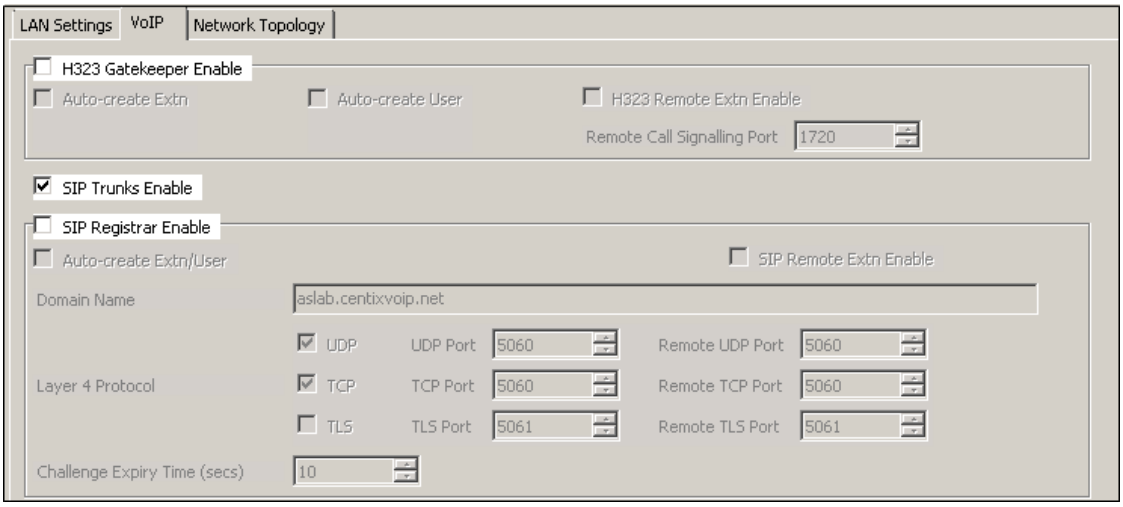


5.2. LAN2 Settings

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.

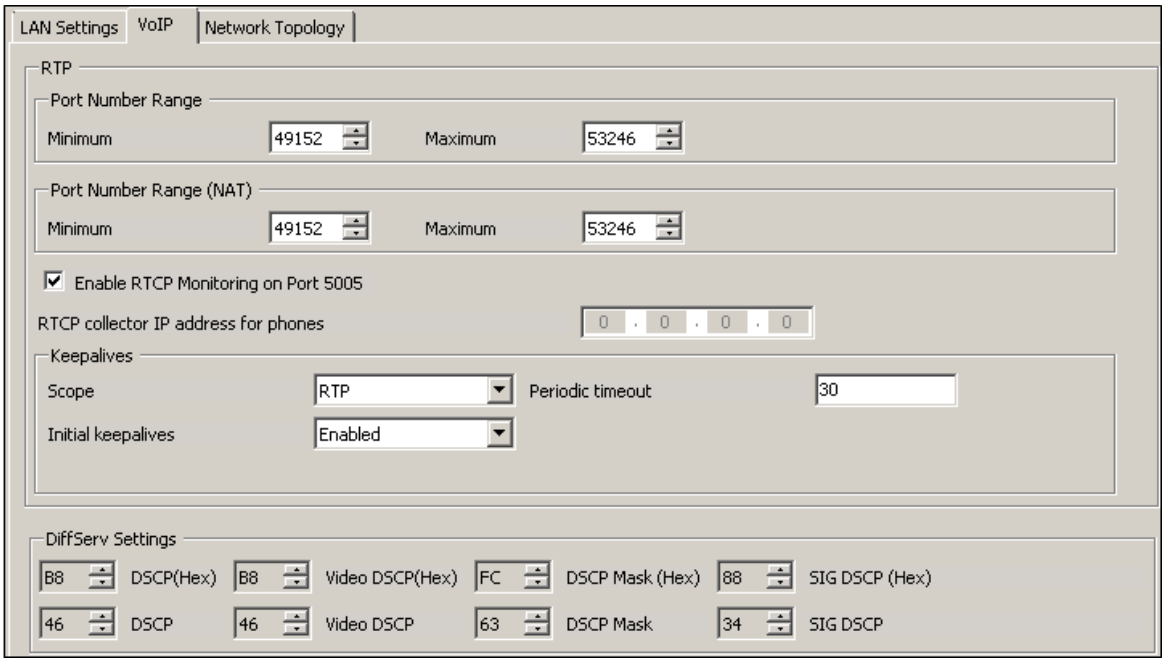


The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2.

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.

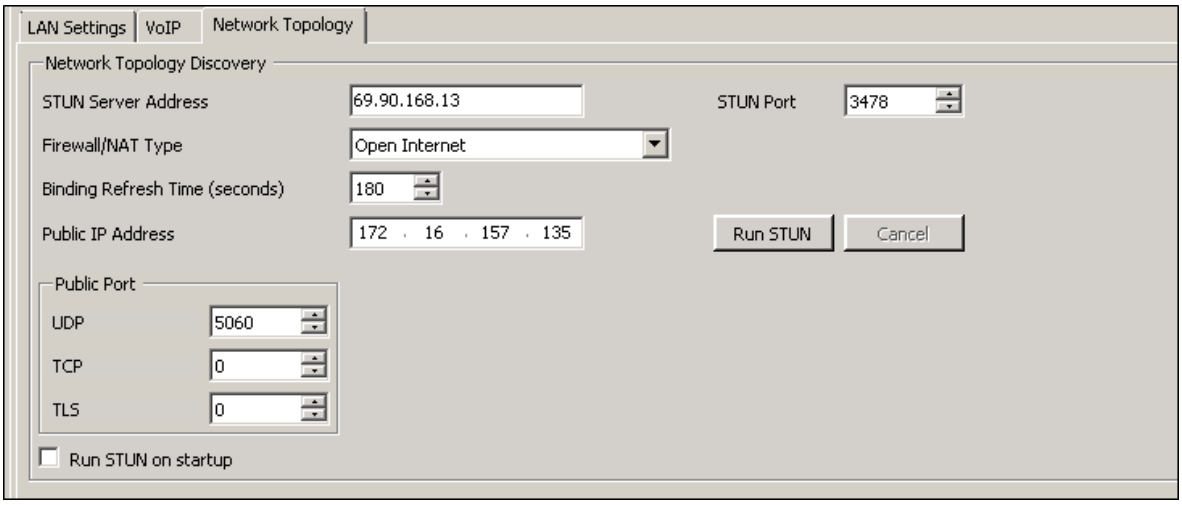
Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling.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

All other parameters should be set according to customer requirements.



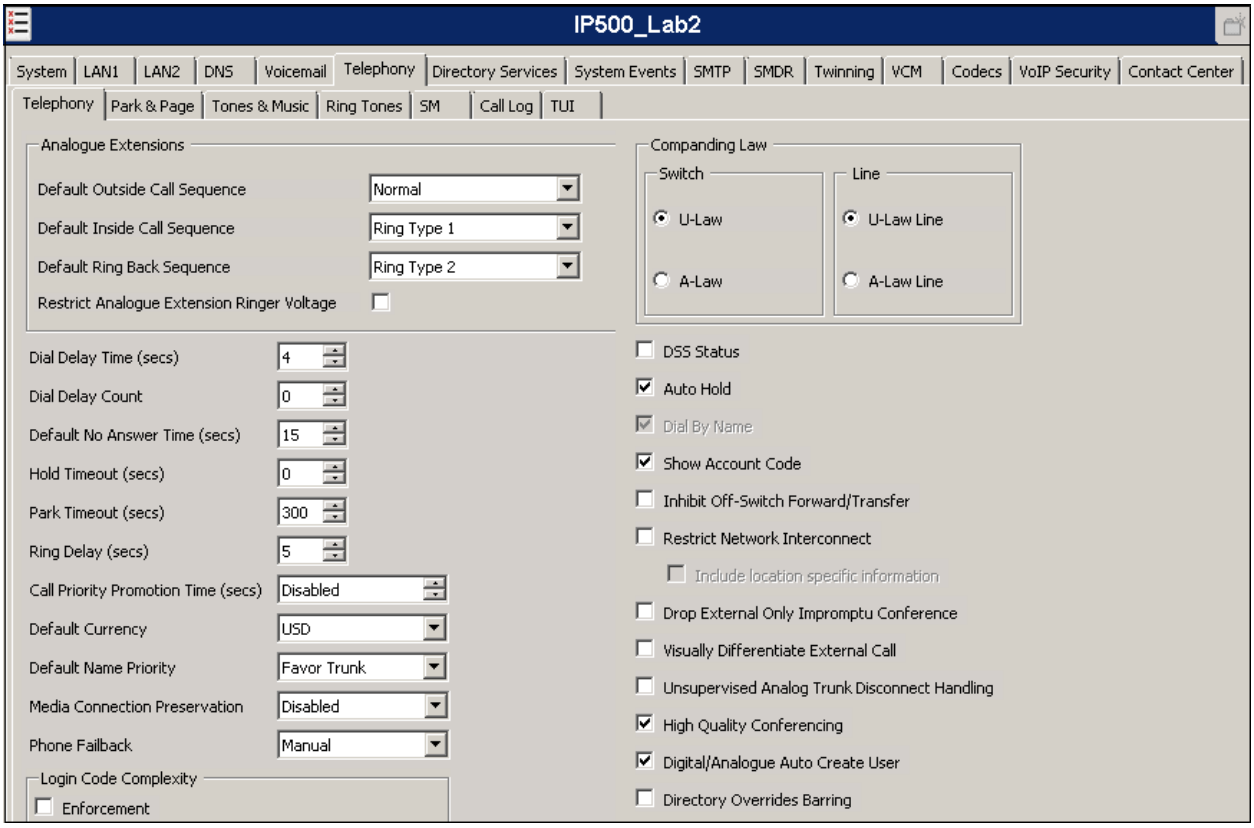
On the **Network Topology** tab in the Details pane, configure the following parameters:

- 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.



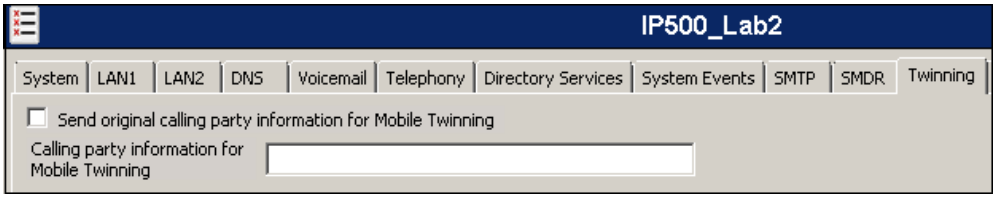
5.3. System Telephony Settings

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

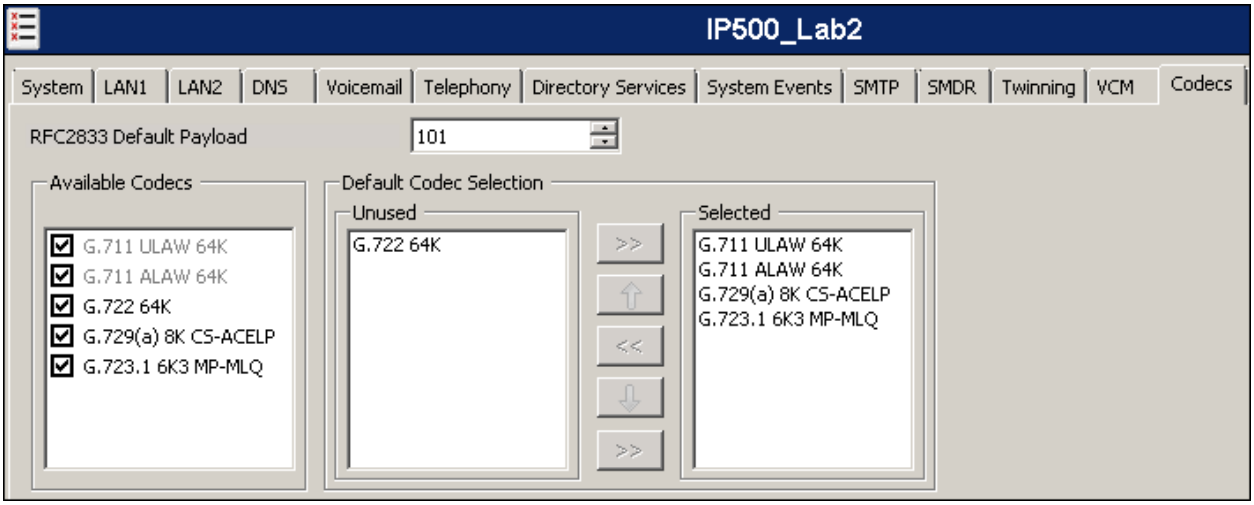
Navigate to the **Twinning** tab on the Details Pane. Uncheck the **Send original calling party information for Mobile Twinning** box. This will allow the Caller ID for Twinning to be controlled by the setting on the SIP Line (**Section 5.7**). This setting also impacts the Caller ID for call forwarding.



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.

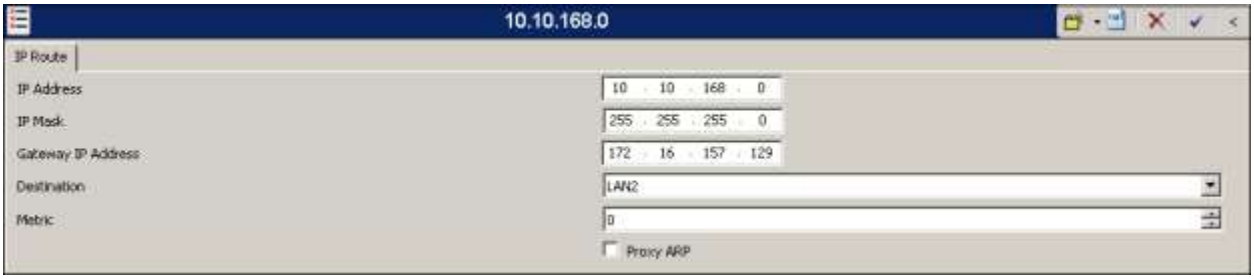
Click **OK** (not shown) to save any changes made to any of the various **System** tabs.



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 MTS 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 MTS 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 created to establish the SIP connection between the Avaya IP Office and the MTS 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** 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**.

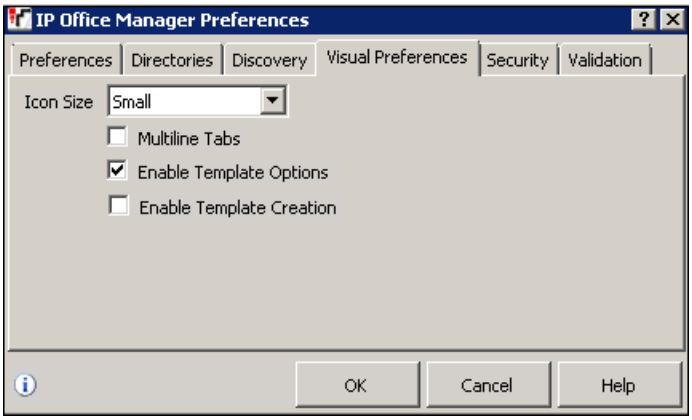
5.7.1. Importing a SIP Line Template

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.

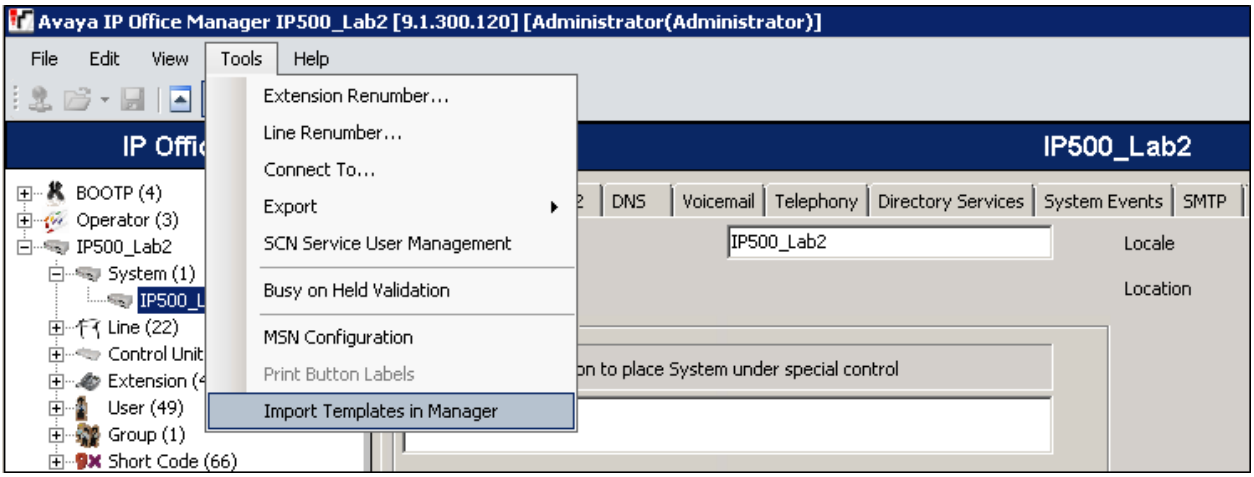
1. Copy a previously created template file to a location (e.g., *\Temp*) on the same computer where IP Office Manager is installed. By default, the template file name will have the format **AF_<*user supplied text*>_SIPTrunk.xml**, where the **<*user supplied text*>** portion is entered during template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

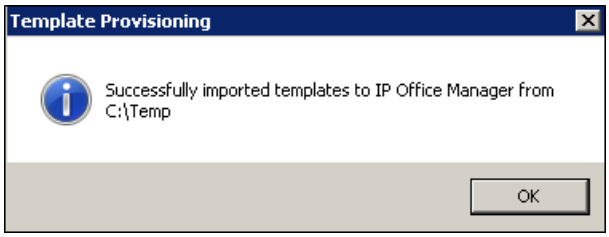


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**.

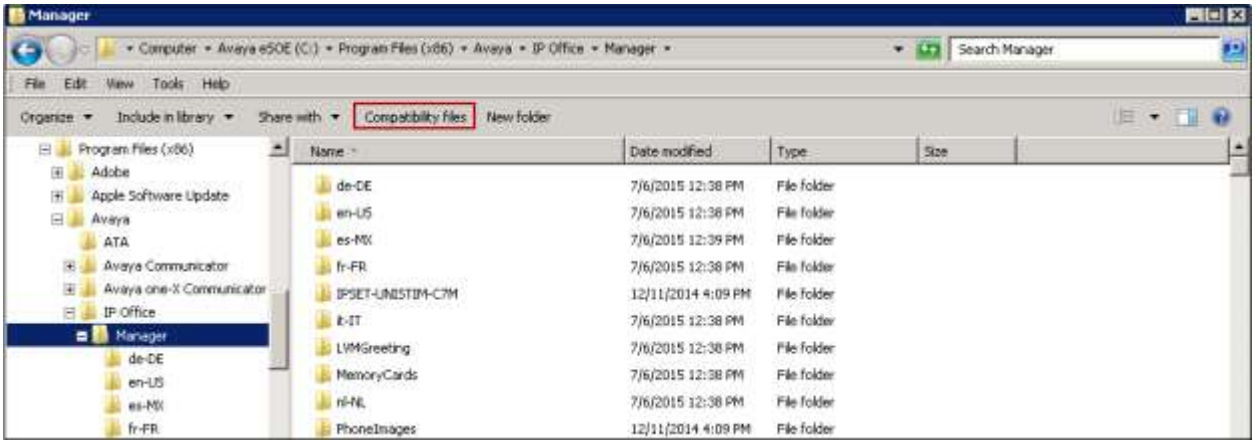


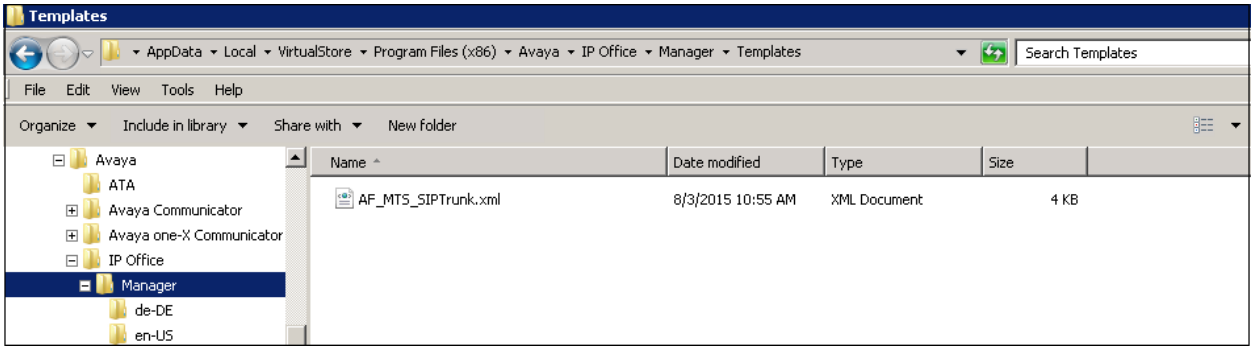
4. A folder browser will open (not shown). Select the directory used in **step 1** to store the template (e.g., *\Temp*). In the reference configuration, template file **AF_*MTS*_ SIPTrunk.xml** was imported. The template file is automatically copied into the default template location, **C:\Program Files\Avaya\IP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

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



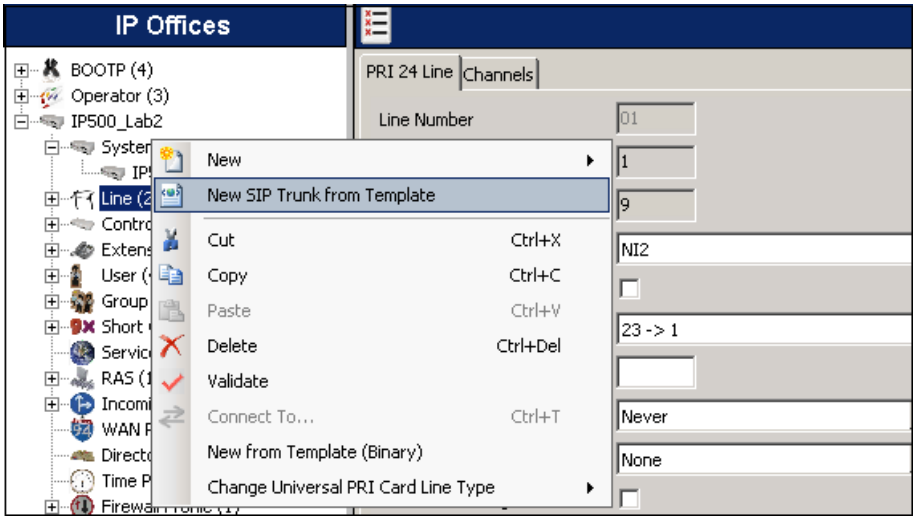
Note –Windows 7 (and later) locks the Avaya IP Office 9.1 **Templates** directory, and it cannot be viewed. To enable browsing of the **Templates** directory, open Windows Explorer, navigate to **C:\Program Files\Avaya\IP Office\Manager** (or C:\Program Files (x86)\Avaya\IP Office\Manager), and then click on the **Compatibility files** option shown below. The **Templates** directory and its contents can then be viewed.





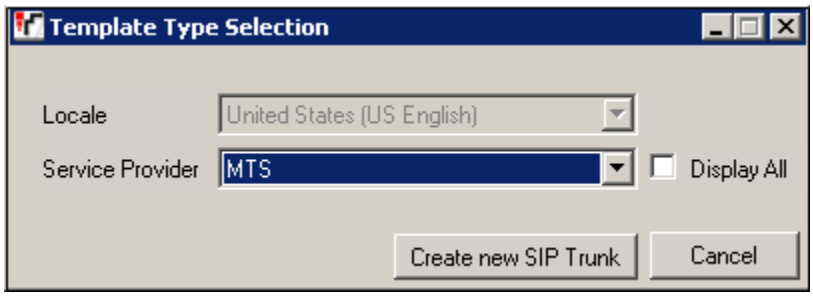
5.7.2. Creating a SIP Trunk from an XML Template

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.



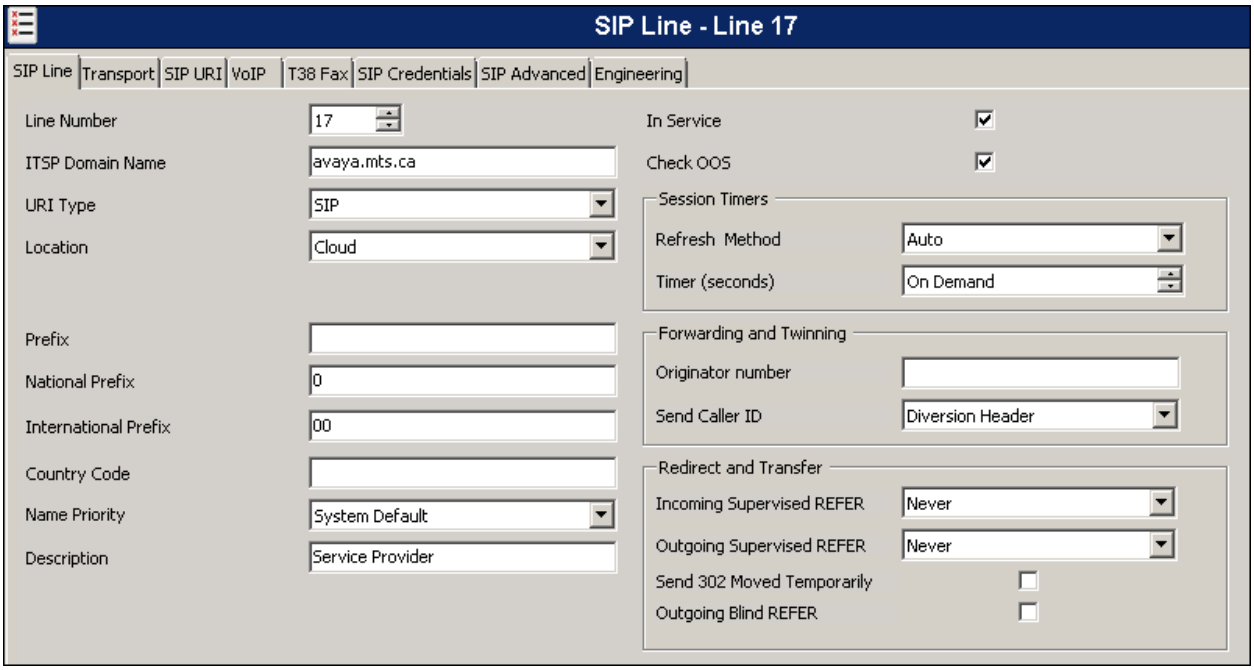
Click **Create new SIP Trunk** to finish creating the trunk.

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.**

5.7.3. SIP Line Tab

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

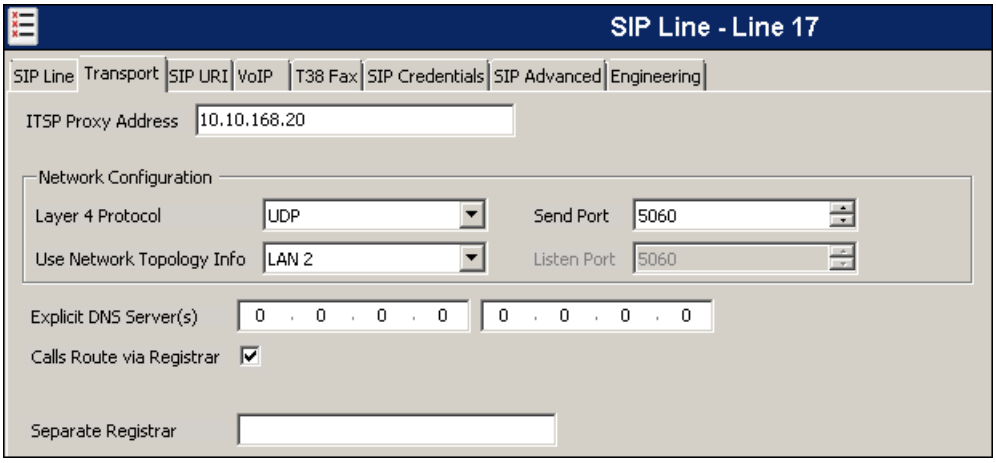
- Set the **ITSP Domain Name** to the domain known and expected by MTS on the SIP trunk. In the reference configuration, this was ***avaya.mts.ca***. IP Office will use this domain as the host portion of the SIP URI of SIP headers in messages sent to the network.
- Check the **In Service** box.
- Check the **Check OOS** box.
- On the **Forwarding and Twinning** section, set **Send Caller ID** to ***Diversion Header***. Avaya IP Office will include the Diversion header to be able to send the original calling party ID, in scenarios of call forward to the PSTN and twinning.
- On the **Redirect and Transfer section**, set **Incoming Supervised REFER** and **Outbound Supervised REFER** to ***Never***. REFER was disabled during the compliance test. See **Section 2.2** for details.
- Default values may be used for all other parameters.



5.7.4. Transport Tab

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

- Set the **ITSP Proxy Address** to the IP address of the MTS 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.

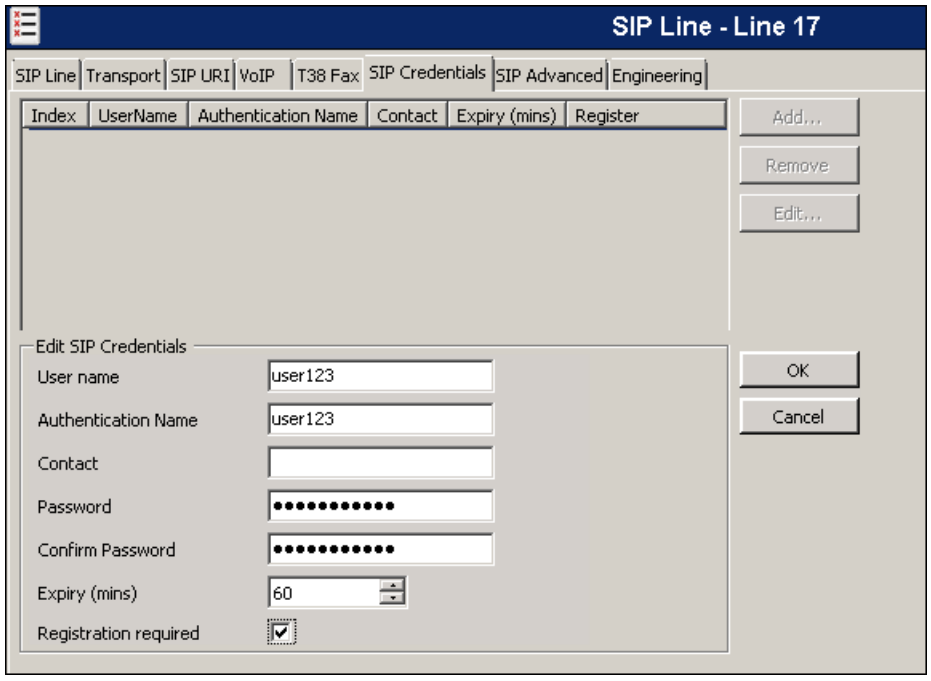


5.7.5. SIP Credentials

MTS required the use of SIP Credentials for the registration of the SIP trunk. These SIP Credentials are also used to authenticate outbound 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** and **Authentication Name** to the value provided by the service provider.
- Set **Password** to the value provided by the service provider.
- Leave the **Expiry (mins)** field to the default value **60**. This field defines how often the service provider requires the SIP trunk registration to be renewed. The actual registration expiration time is negotiated and agreed as part of the registration exchange.
- Check the **Registration required** box.
- Click **OK**.



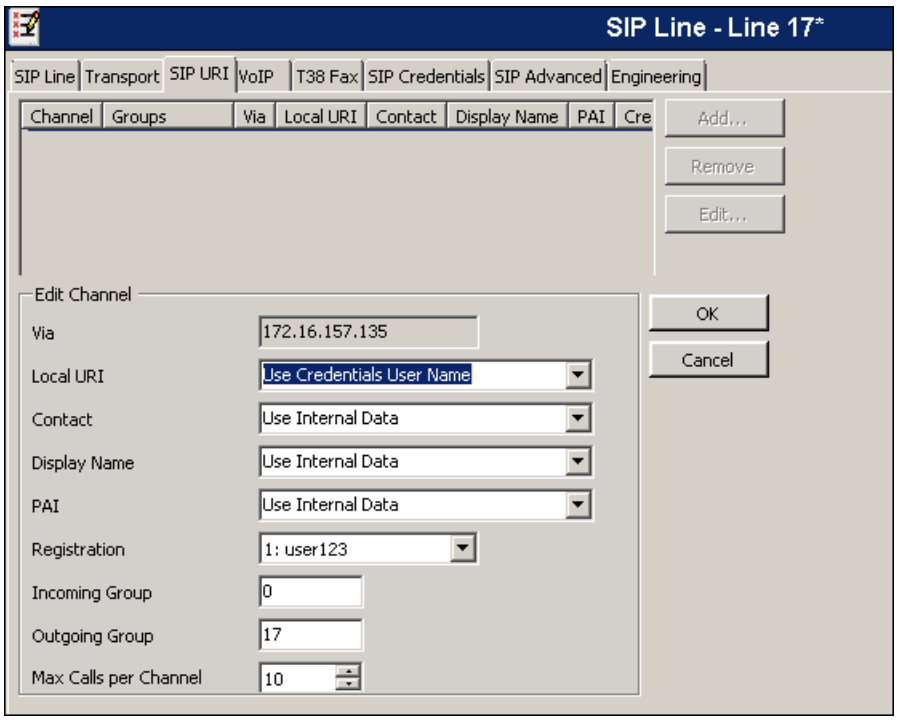
5.7.6. SIP URI Tab

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. In the reference configuration, two SIP URI entries were defined, one for outbound calls to the SIP trunk and another one for inbound calls to the Avaya IP Office.

To set the SIP URI for outbound calls, select the **SIP URI** tab and click the **Add** button. The **New Channel** area will appear at the bottom of the pane. Set the parameters as shown below:

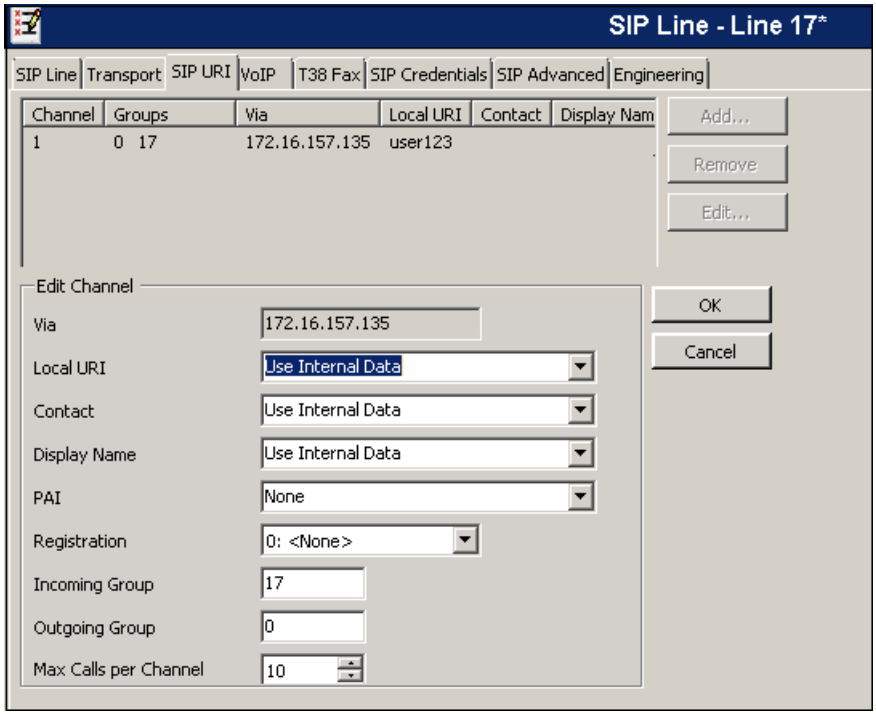
- Set **Local URI** to ***Use Credentials User Name***. MTS required the user name configured in **Section 5.7.5** to be sent in the "From" header of all outbound messages sent to the network.
- Set **Contact**, **Display Name** and **PAI** to ***Use Internal Data***.
- Under **Registration**, select ***1: <user123>*** from the pull-down menu. MTS 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.
- Leave the **Incoming Group** field with the default value ***0***.
- Set the **Outgoing Group** field to ***17***. 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.
- Click **OK**.



To set the SIP URI for inbound calls, select the **SIP URI** tab and click the **Add** button. The **New Channel** area will appear at the bottom of the pane. Set the parameters as 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 ***None***.
- Under **Registration**, select ***0: <None>*** from the pull-down menu.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. For the compliance test, a new incoming and outgoing group 17 was defined that only contains this line (line 17).
- Leave the **Outgoing Group** field with the default value ***0***.
- Set **Max Calls per Channel** to the number of simultaneous calls to be allowed on the SIP trunk using this SIP URI pattern.
- Click **OK**.

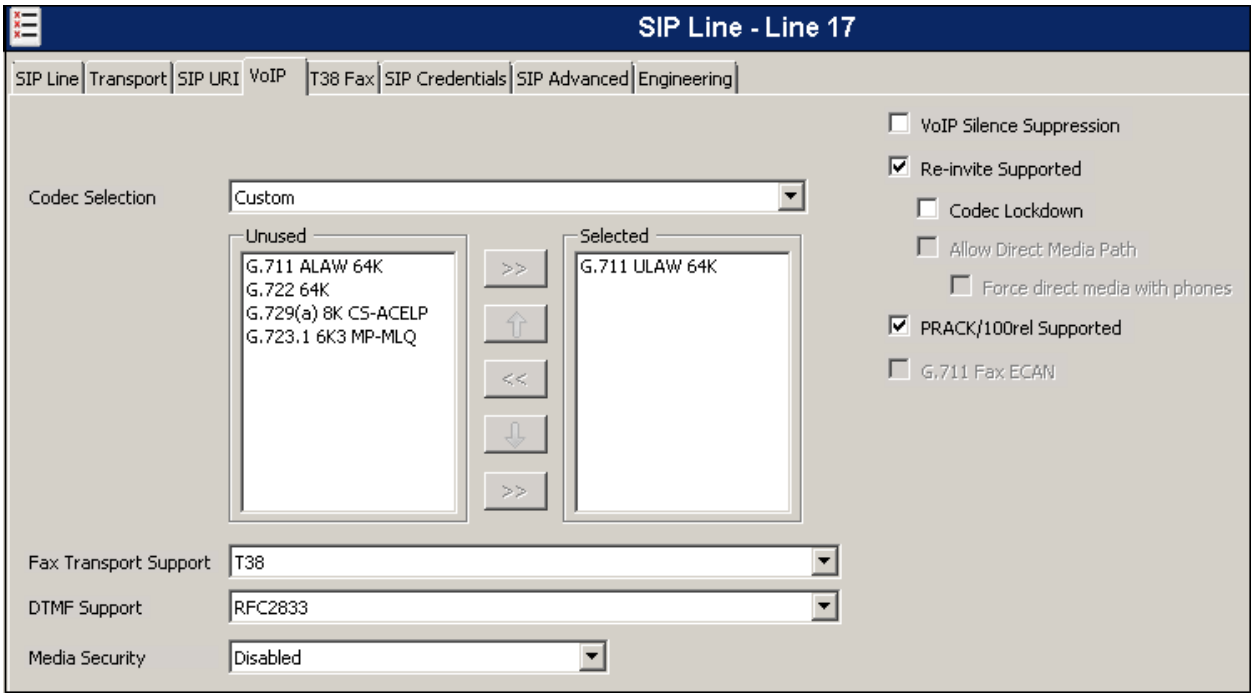


Additional SIP URIs may be required to allow inbound calls to numbers not associated with a user, such as a short code. These URIs are created in the same manner as shown previously, with the exception that the incoming DID number is entered directly in the **Local URI**, **Contact**, and **Display Name** fields.

5.7.7. VoIP Tab

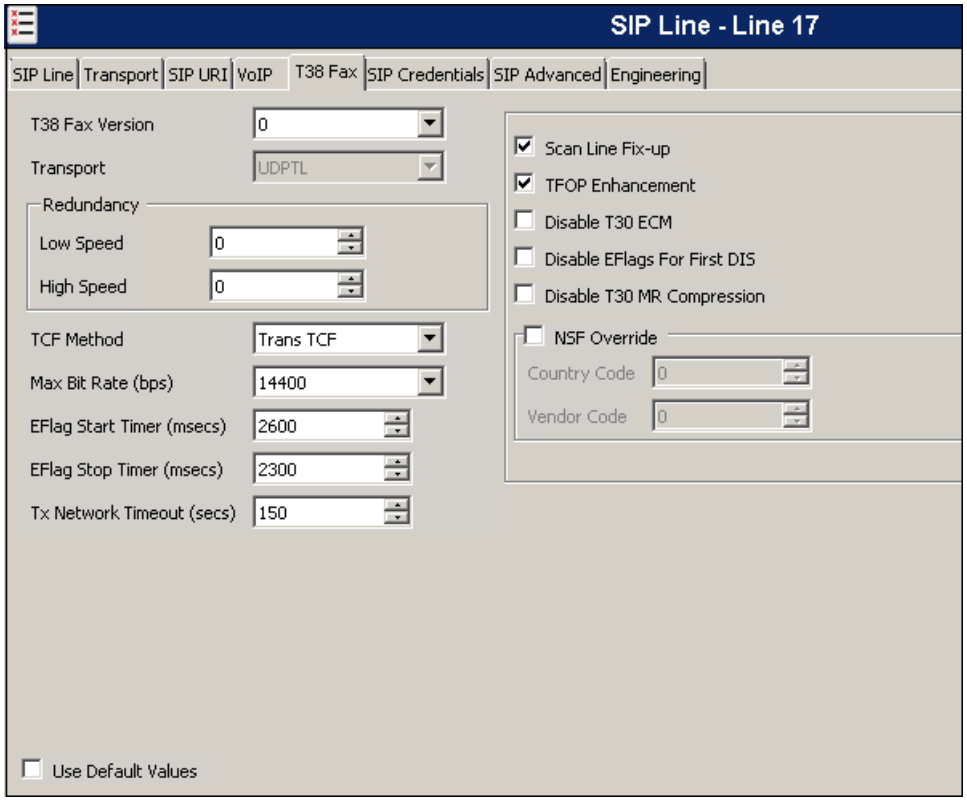
Select the **VoIP** tab to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- 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. In the example codec G.711ULAW, the only codec supported by MTS on the SIP trunk, is placed under the **Selected** column.
- 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 the service provider.
- Default values may be used for all other parameters.



5.7.8. T38 Fax Tab.

On the **T38 Fax** tab, uncheck the **Use Default Values** box at the bottom of the tab. Set the **T38 Fax Version** to version ***0***.



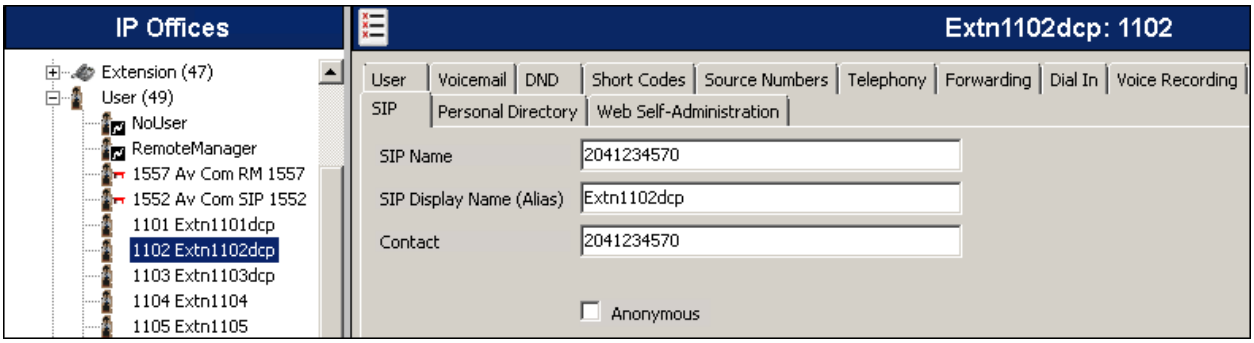
Click **OK** (not shown) to save all the changes made to any of the various "SIP Line" tabs.

No changes were made to the **SIP Advanced** and the **Engineering** tabs, so they will not be visited**.**

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.

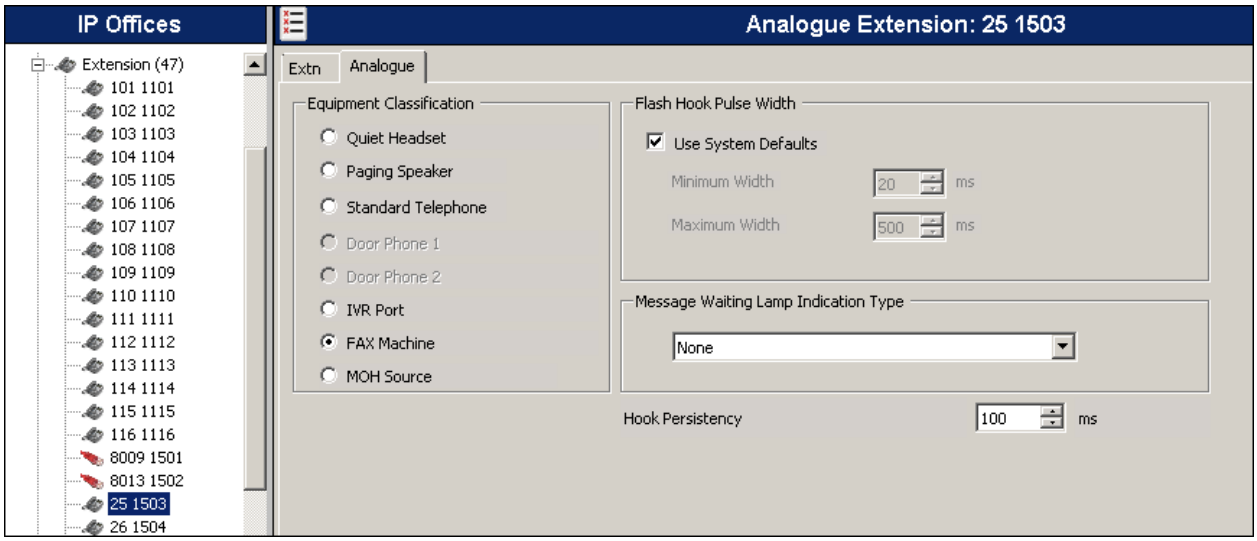
The values entered for the **SIP Name** and **Contact** fields are used to match against the SIP URI of incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.7.6**). The example below shows the settings for user "Extn1102dcp". The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise by MTS. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. Click **OK** (not shown) to save any changes.



5.9. Fax Extensions

As specified in **Section 2.2**, during the compliance test it was determined that in order for both outbound and inbound T.38 fax calls to complete successfully, analog extensions used as fax machines in the IP Office needed to specifically be classified as such. By doing this, IP Office offered T.38 in the SDP of the initial INVITE of outbound calls made from these fax extensions. These calls connected directly at T.38, avoiding the G.711MU voice setup and the problems that occurred with the subsequent re-INVITE from MTS. On incoming calls to these extensions, IP Office still sent a T.38 re-INVITE, which was accepted by MTS, after the initial voice connection using codec G.711MU.

Under **Extension** on the left Navigation pane, select the extension to be used as a fax machine. On the Details pane, select the **Analogue** tab. On the **Equipment Classification** section, check the **FAX Machine** option. The screen below shows the configuration for extension 1503, used as a fax machine in the sample configuration.



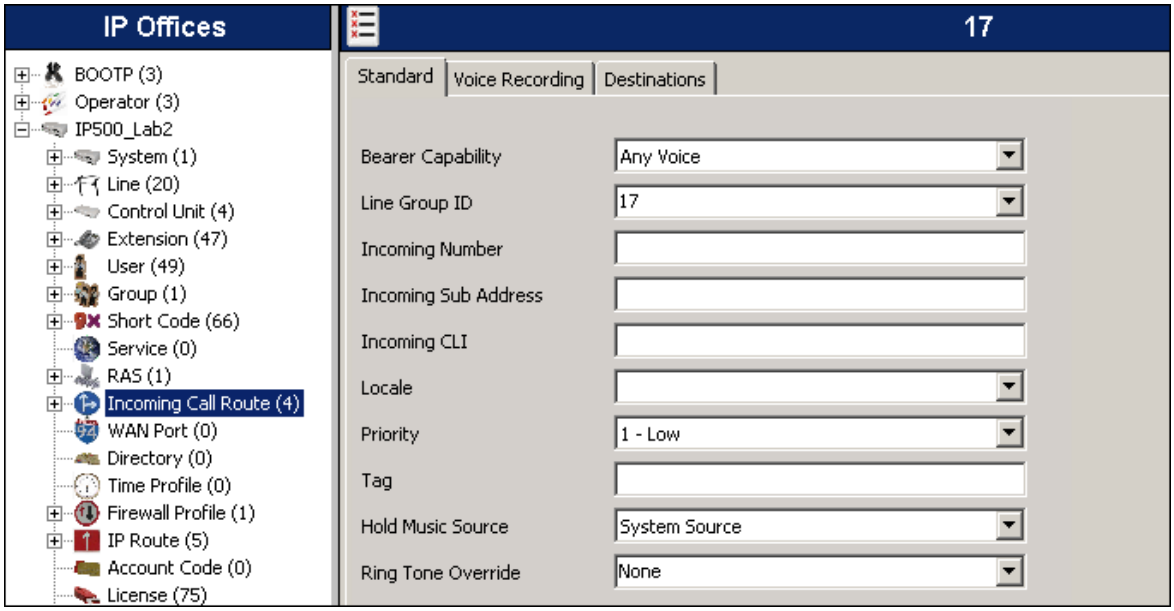
5.10. Incoming Call Route

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.

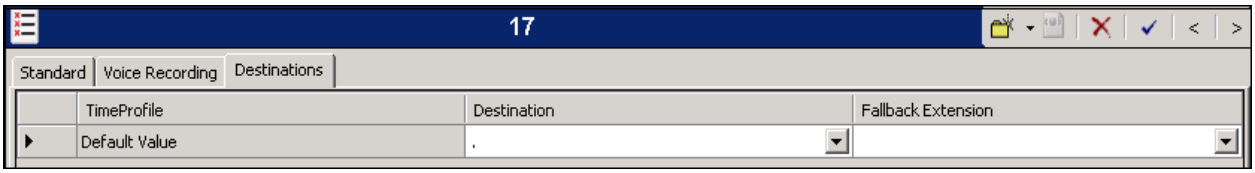
In a scenario like the one used for the compliance test, only one incoming route was needed, which allowed any incoming number arriving on the SIP trunk to reach any predefined extension in the IP Office. The routing decision for the call is based on the parameters previously configured for the **SIP URI (Section 5.7.6)** and the users **SIP Name** and **Contact**, already populated with the assigned 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 Capability** 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 "To" header of the incoming INVITE.



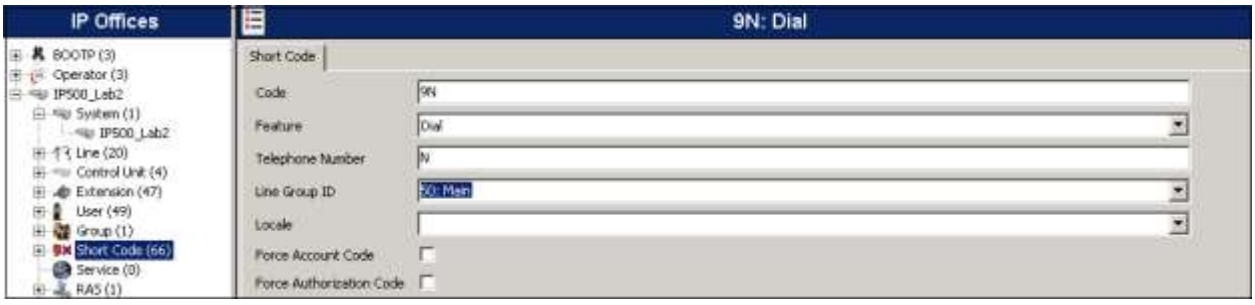
Additional incoming call routes may be required to allow inbound calls to numbers not associated with a user, such as a short code. These routes are created in the same manner as shown, with the exception that the incoming DID number is entered directly in the **Incoming Number** field on the **Standard** tab, 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.

5.11. 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 5.12**.

On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

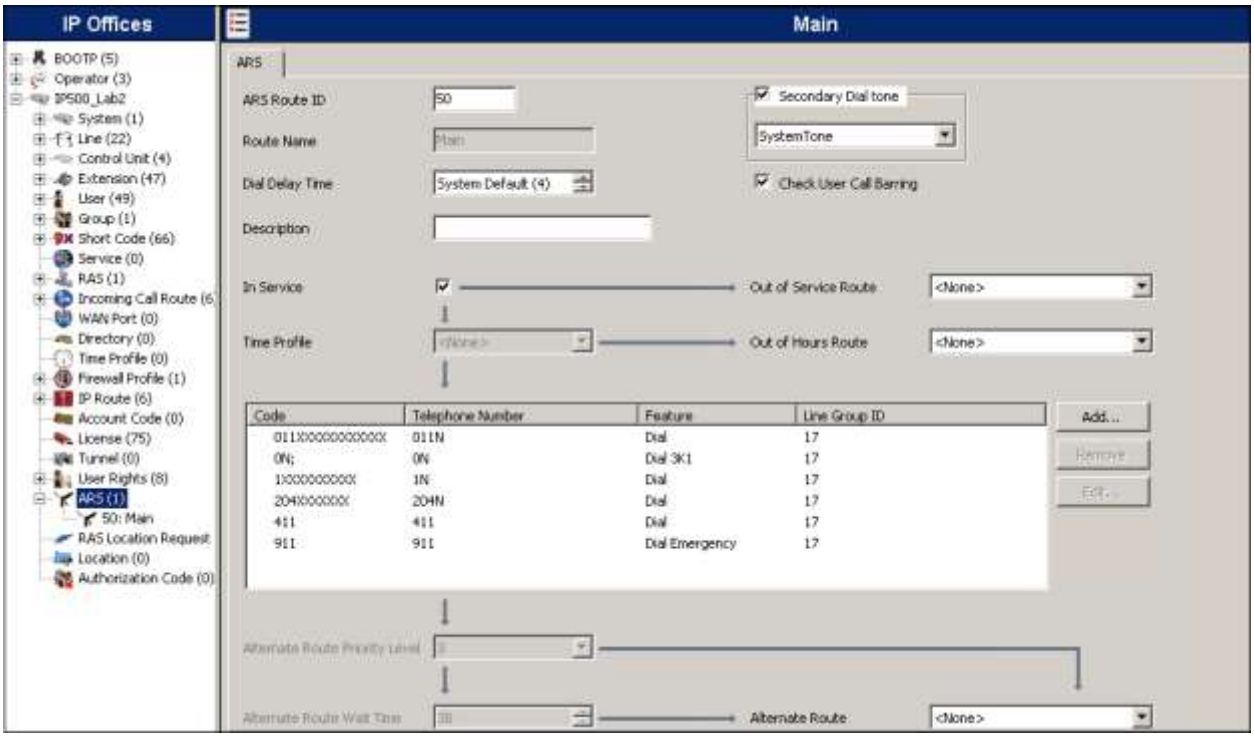
- In the **Code** field, enter the dial string which will trigger this short code, in this case ***9N***. This short code will be invoked when the user dials 9 followed by any number.
- Set **Feature** to ***Dial***. This is the action that the short code will perform.
- Set **Telephone Number** to **N**. 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).



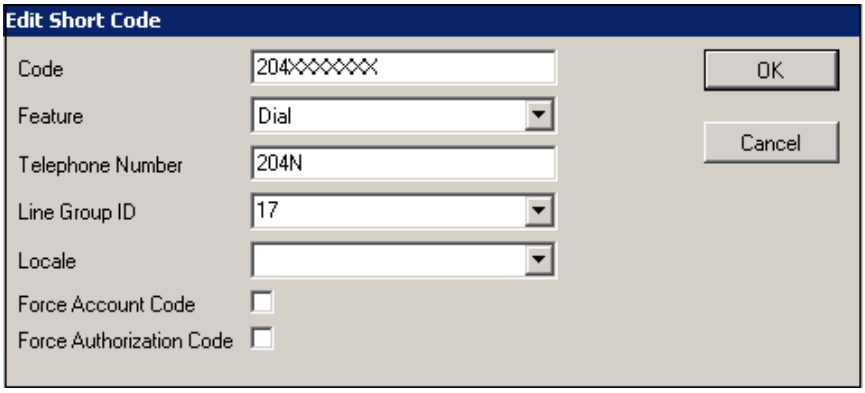
5.12. Automatic Route Selection

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.



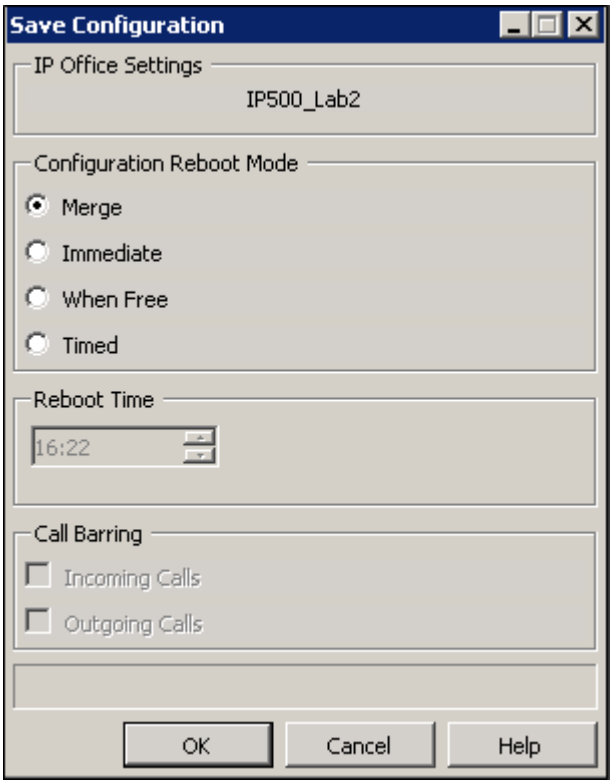
For example, during the compliance test, to dial local PSTN calls the user dialed 9 plus the 10 digit local number, starting with the area code 204 and then the remaining 7 digits.



5.13. Save Configuration

Navigate to **File → Save Configuration** in the menu bar at the top left of the screen to save the configuration performed in the preceding sections.

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.



6. MTS SIP Trunking Service Configuration

MTS is responsible for the configuration of the SIP Trunking 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. MTS will provide the customer the necessary information to configure the SIP trunk connection from the enterprise site to the network, including:

- IP address and port of the MTS SIP Proxy server.
- Credentials for the SIP trunk registration (user name, password).
- MTS SIP Domain.
- 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.

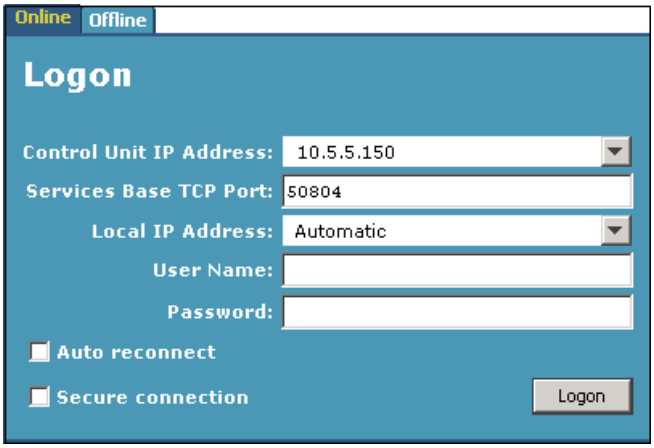
This information is used to complete the configuration of the Avaya IP Office discussed in the previous sections.

7. Verification Steps

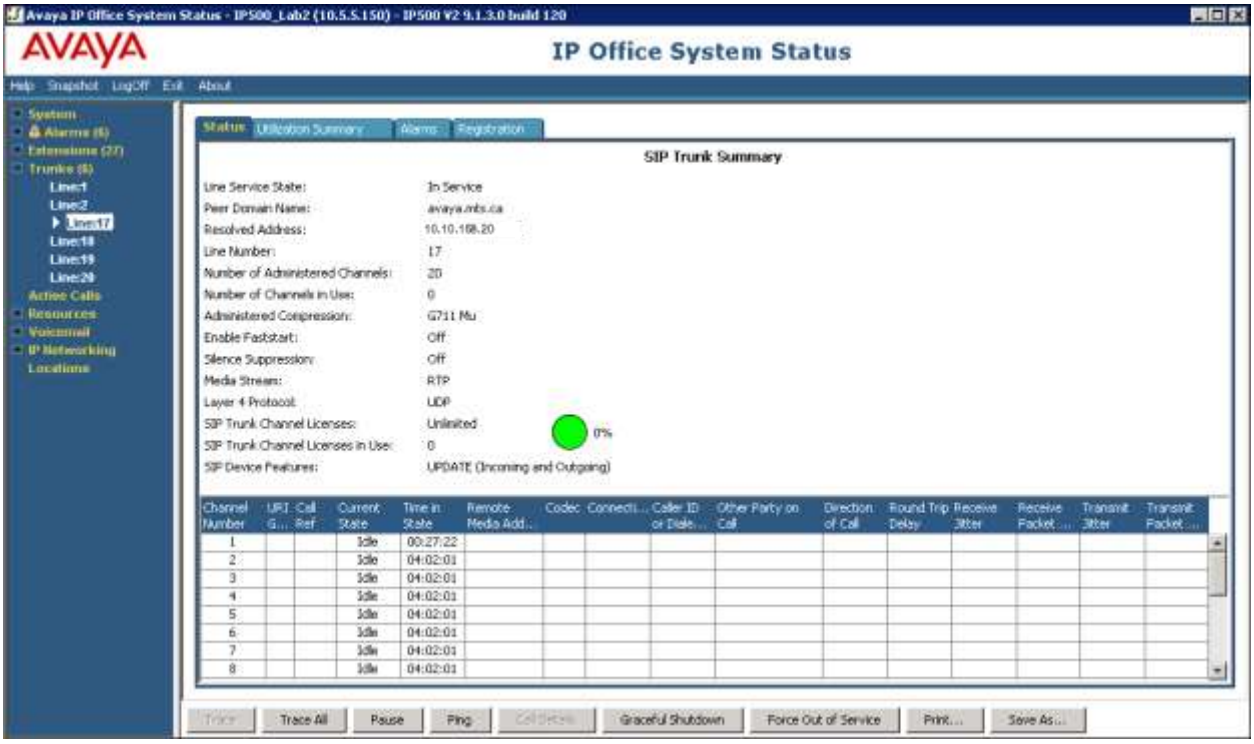
The Avaya IP Office System Status and Monitor applications are useful tools used for the verification and troubleshooting of the SIP connection to the service provider.

7.1. System Status

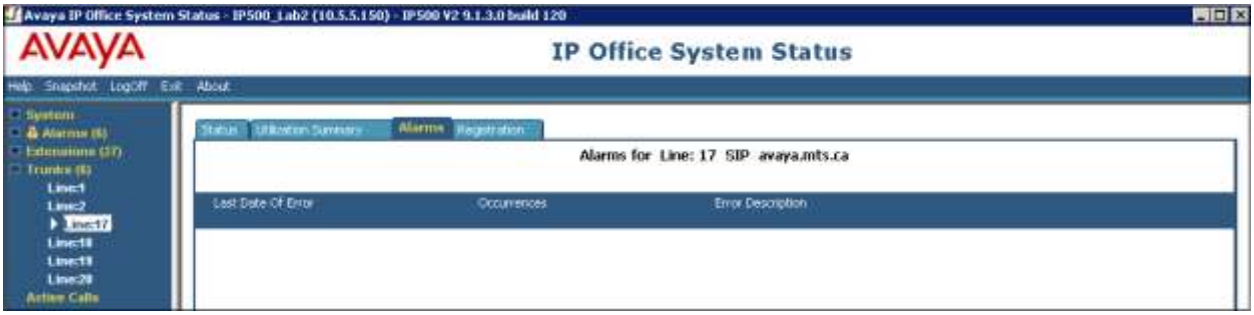
The Avaya IP Office System Status application can be used to verify the service state of the SIP line. Launch the application from **Start → Programs → IP Office → System Status** on the PC where Avaya IP Office Manager was installed. Under **Control Unit IP Address** select the IP address of the IP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



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).



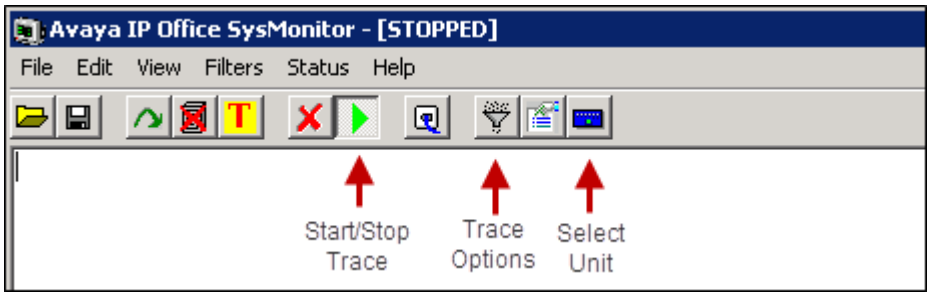
S**elec**t the **Alarms** tab and verify that no alarms are active on the SIP line.



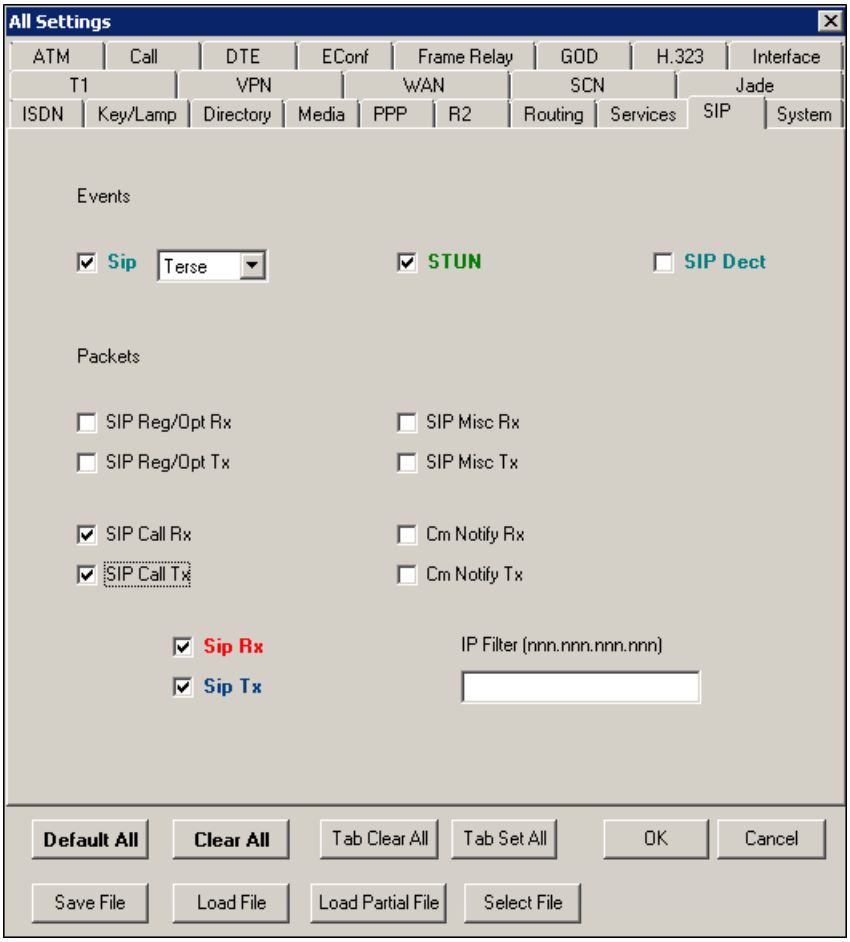
On the **Registration** tab, verify that the trunk is successfully registered with the service provider.

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.



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 MTS SIP Trunking Service, as shown in **Figure 1**.

Interoperability testing of the sample configuration was completed with successful results for all test cases with the exception of the observations/limitations described in **Section 2.2**.

9. Additional References

[1] *IP Office Platform 9.1, Deploying Avaya IP Office Platform IP500V2*, Document 15-601042, July 2015
https://downloads.avaya.com/css/P8/documents/101005082

[2] *Administering Avaya IP Office Platform with Manager, Release 9.1*, July 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 the MTS SIP Trunking Service is available from MTS.

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.