Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 9.1 with Clearcom SIP Trunk Services – 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 Clearcom SIP Trunk Services.

The SIP Trunking service offered by Clearcom 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 procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider Clearcom 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 Clearcom SIP Trunk Services referenced within these Application Notes is designed for business customers in Mexico. Customers using this service with this Avaya enterprise solution are able to place and receive PSTN calls via a broadband WAN connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as analog and/or ISDN-PRI.

The Avaya enterprise solution can be configured to authenticate with the SIP service provider using either SIP Trunk Registration or Static IP Authentication. These Application Notes cover the configuration of the Avaya IP Office using SIP Trunk Registration with Clearcom.

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 Clearcom SIP Trunk Services 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 and local directory assistant.
- Codecs G.729A, G.711A and G.711MU.
- Fax.
- Caller ID presentation and Caller ID restriction.
- DTMF transmission using RFC 2833.
- Voicemail navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, and conference.
- Off-net call transfer, call forwarding and twinning.

The following functionality is not supported by the service provider and it was not tested:

- Operator and operator assisted calls.

Inbound toll-free and emergency calls are supported, but were not tested as part of the compliance test

2.2. Test Results

Interoperability testing of Clearcom SIP Trunk Services was completed with successful results for all test cases with the observations and limitations described below:

- **Caller ID on outbound calls**: On calls originating from Avaya IP Office extensions to PSTN telephones, the caller ID number shown on the PSTN endpoint was always the main DID number assigned by Clearcom to the SIP trunk, not the specific DID assigned to that extension. This includes calls to "twinned" mobile phones, and calls that were forwarded or transferred back on the SIP trunk to the PSTN, where the number displayed on the PSTN endpoint was the main DID number on the trunk, not the originator's caller's ID.
  This may be a requirement of the Clearcom service for all outbound calls, it is listed here simply as an observation.
- **Caller ID on inbound calls**: On inbound calls made from the test lab in the U.S., the caller IDs shown on the enterprise extensions occasionally showed "Unavailable", while in other cases showed numbers corresponding to local PSTN numbers in Mexico, not the CLI of the original caller. Calls made from a test number in Mexico showed the correct caller ID.
- **Outbound Calling Party Number (CPN) Block**: Clearcom did not allow outbound calls with privacy enabled. When an Avaya IP Office user activated "Withhold Number" to enable user privacy on an outbound call, Avaya IP Office sent "anonymous" in the "From" header and the "Privacy:id" header, while the caller information was still being sent in the
  "P-Asserted-Identity" header. Clearcom responded with a "403 PSTN calls are forbidden" message and the call was rejected.
- **Outbound call from an enterprise extension to a busy PSTN number**: Clearcom did not send a "486 Busy Here" message on an outbound call to a PSTN number that was busy, as it was expected on this condition. There was no direct impact to the user, who heard busy tone.
- **Call transfer to the PSTN using REFER**: PSTN calls that were transferred back to the network using REFER messages did not work properly. Calls that were blind transferred dropped. On attended transfers, the REFER message was accepted by Clearcom with a 202 message, but the trunks were not released. For the reasons above, REFER was left disabled in the Avaya IP Office for the tests. With REFER disabled, blind and attended call transfers to the PSTN were allowed to complete, with the caveat that the IP Office was not released from the call path, and two trunks remained busy for the complete duration of the call.
- **Fax support**: Fax calls using the T.38 protocol failed during the test. G.711 fax was also tested, but it behaved unreliably. Fax should not be used in this solution.
- **Incoming Call , SIP Trunk Signaling Failure**: When the SIP trunk was forced to an "Out of Service" condition, and an incoming call was attempted to one of the DID numbers, it took from 15 to 30 seconds, depending on the source of the call, for the caller to receive an error recording from the network. This amount of time seems excessive in this condition.

2.3. Support

For technical support on the Clearcom SIP Trunk Services offer, visit http://www.clearcom.mx/

3. Reference Configuration

Figure 1 illustrates the sample Avaya SIP-enabled enterprise solution, connected to Clearcom SIP Trunk Services through a public Internet WAN connection.

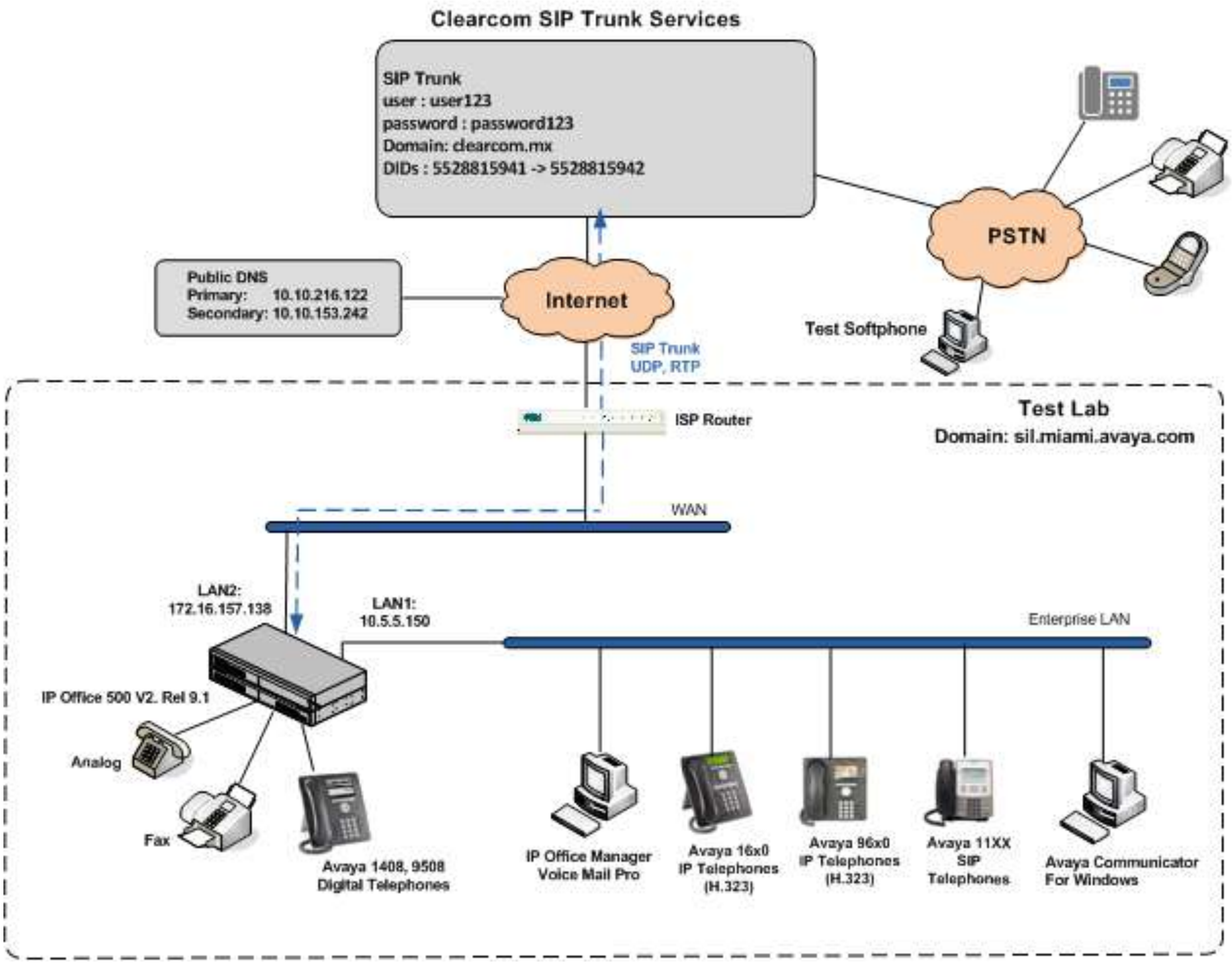


Figure 1: Test Configuration

Note that for security purposes, all public IP addresses of the network elements and public PSTN numbers 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 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 a router, 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.

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

4. Equipment and Software Validated

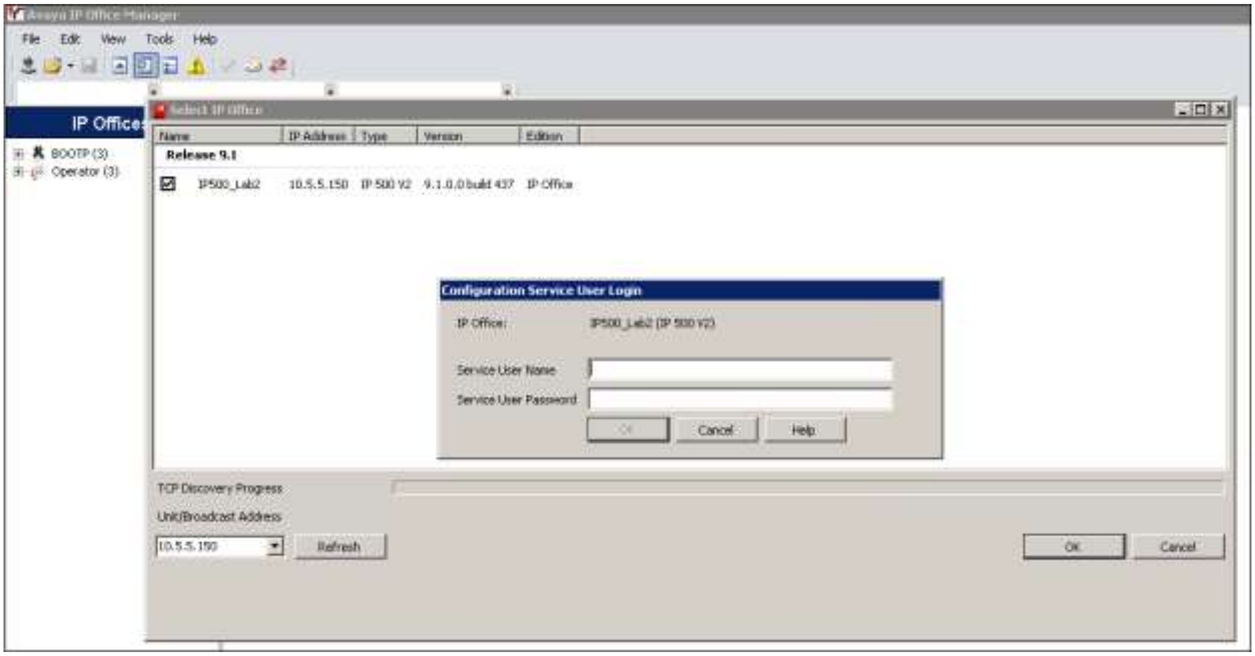
The following equipment and software were used for the sample configuration provided:

Component	Version
Avaya	
Avaya IP Office 500v2	9.1.0.437
Avaya IP Office Digital Expansion Module DCPx16	9.1.0.437
Avaya IP Office Manager	9.1.0.0.Build 437
Avaya IP Office Voicemail Pro	9.1.0.166
Avaya 1608 IP Telephone (H.323)	1.3.5
Avaya 9640 IP Telephone (H.323)	Avaya one-X Deskphone Edition S3.230A
Avaya 1140E IP Telephone (SIP)	04.04.18.00
Avaya Digital Telephone 1408	40.0
Avaya Digital Phone 9508	0.55
Avaya Communicator for Windows	2.0.3.30
Clearcom	
OpenSIPS Softswitch	1.9
OpenSIPS Session Border Controller	1.9

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.

5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration necessary to support connectivity to Clearcom SIP Trunk Services. 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.

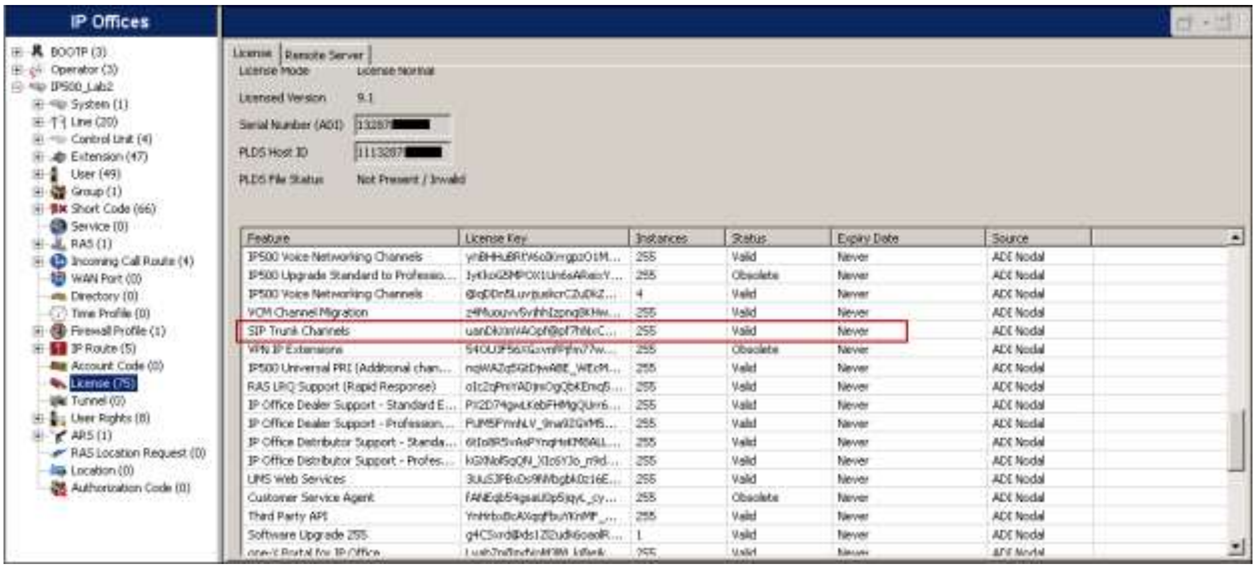
The appearance of the IP Office Manager can be customized using the View menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side and the Details pane on the right side. These panes will be referenced throughout the Avaya IP Office configuration.

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

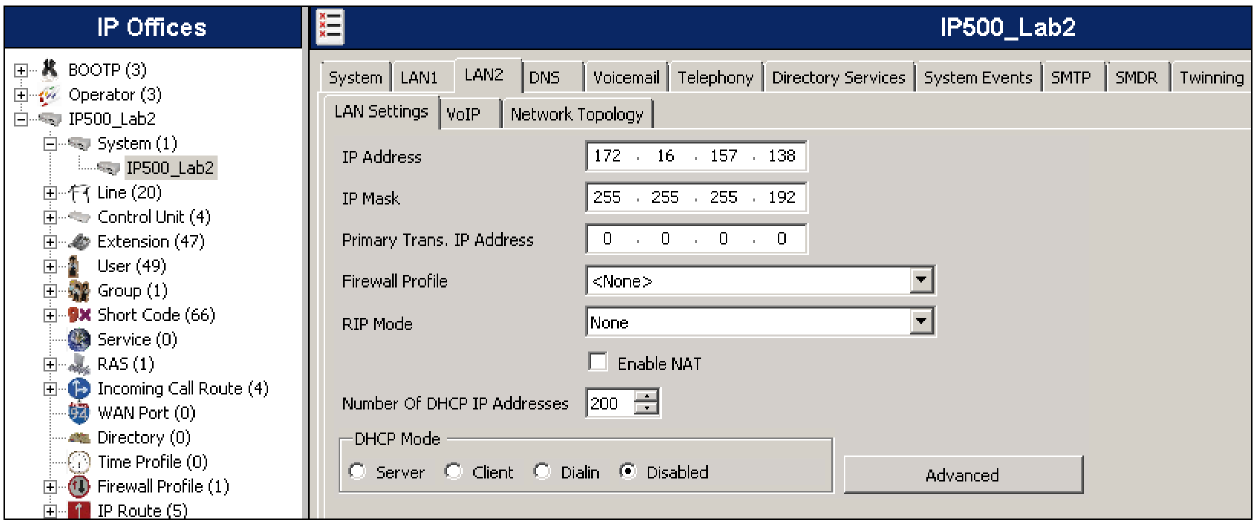
The configuration and features described in these Application Notes require the Avaya IP Office system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

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.

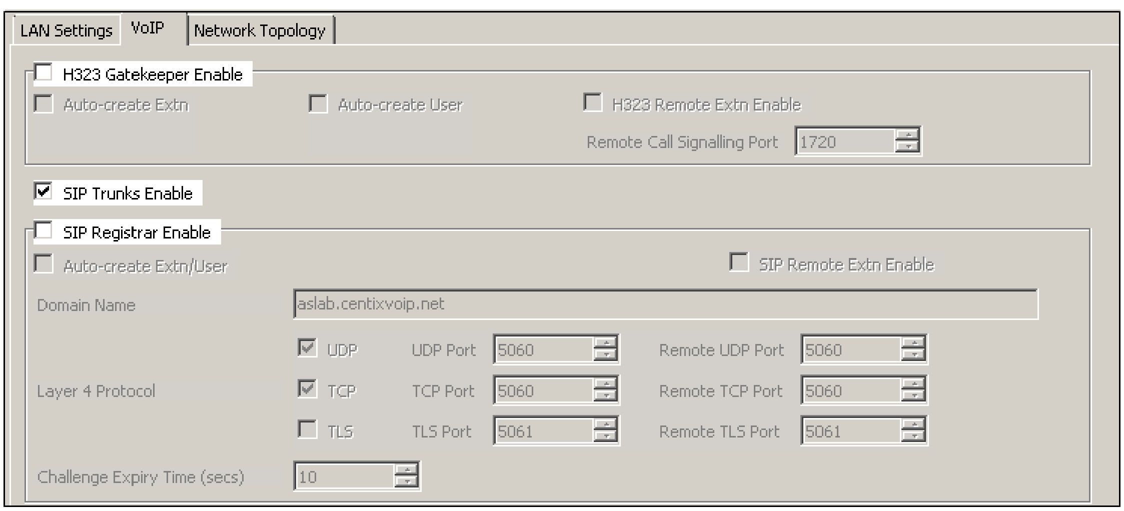


5.2. LAN2 Settings

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.

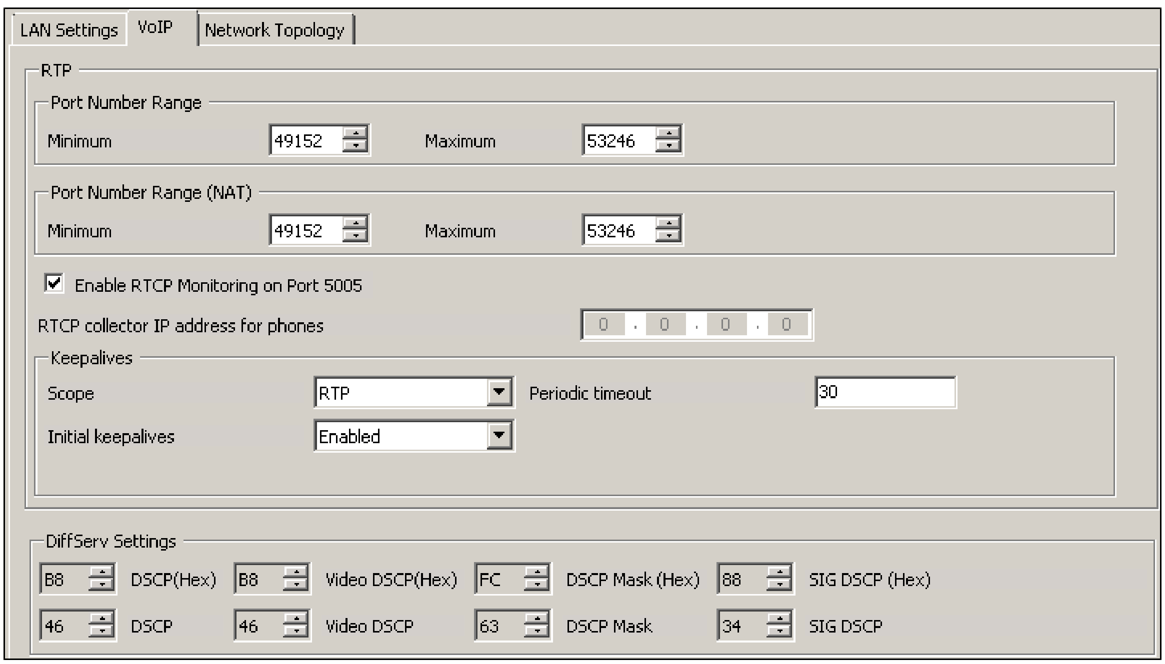


The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2.

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 Avaya 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 Avaya IP Office back to the network, over the same SIP trunk.

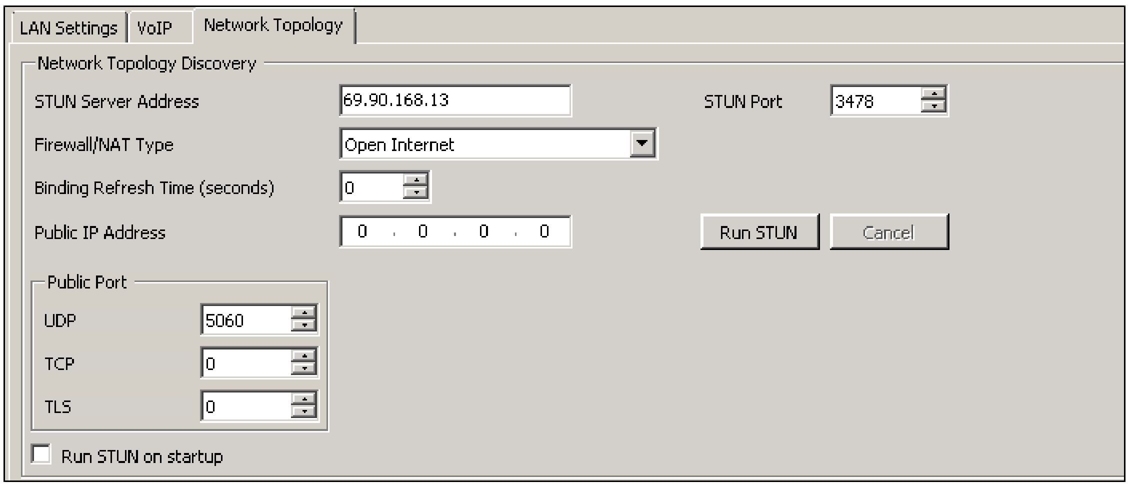
Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling.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

All other parameters should be set according to customer requirements.



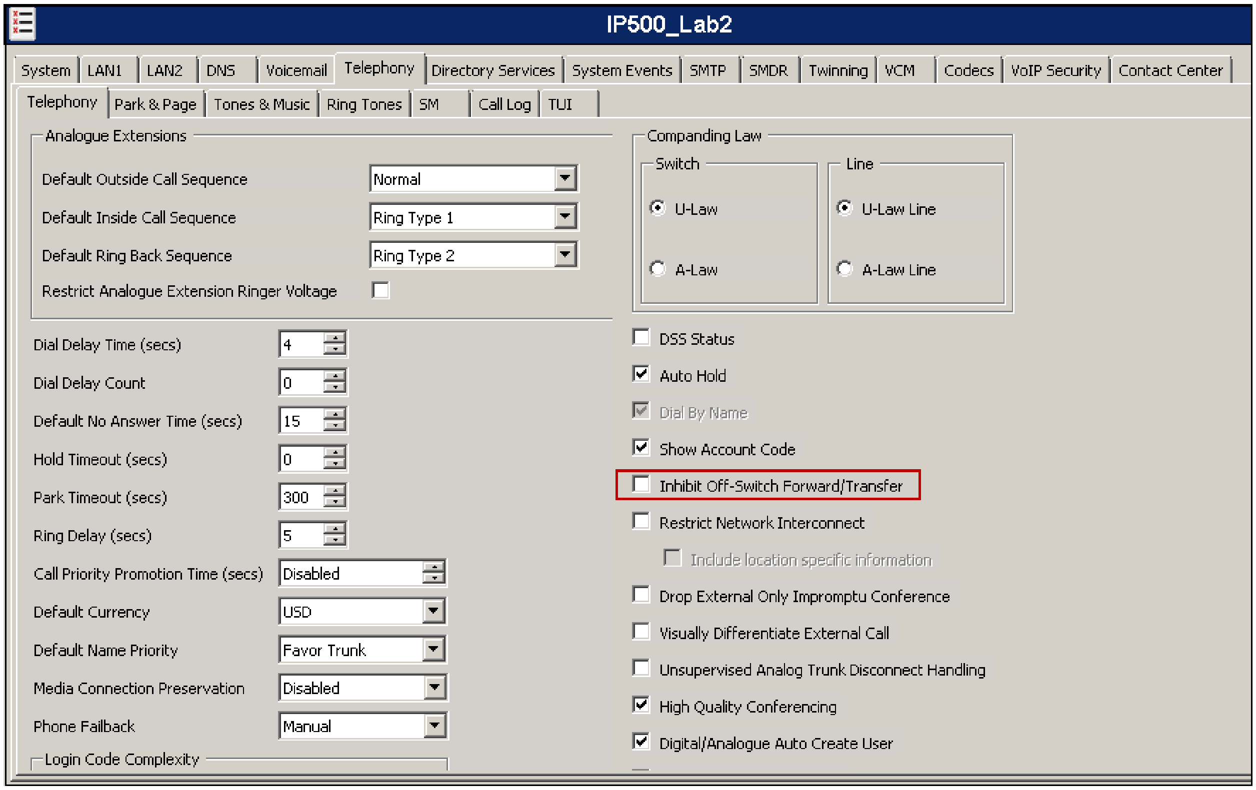
On the **Network Topology** tab in the Details pane, configure the following parameters:

- 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. This parameter was left at the default value ***0***. This means that the Avaya IP Office will send OPTIONS messages using its default interval of 300 seconds.
- 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.



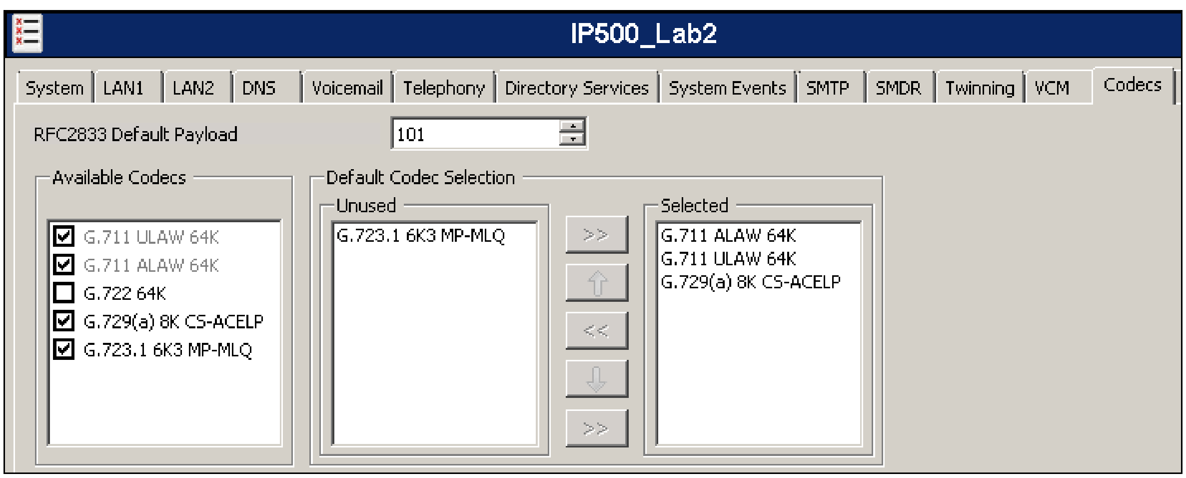
5.4. Twinning Calling Party Settings

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 Avaya 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.

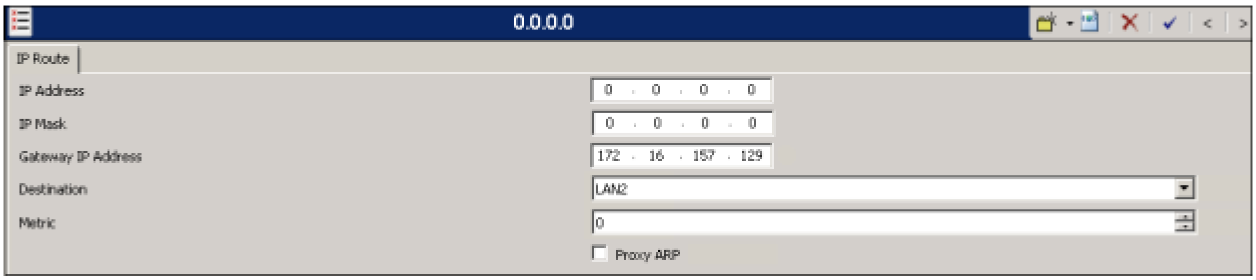


5.6. IP Route

Create an IP route to specify the IP address of the gateway or router where the Avaya IP Office needs to send the packets in order to route calls to the Clearcom network.

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

- Set the **IP Address** and **IP Mask** to **0.0.0.0** to make this the default route.
- Set **Gateway IP Address** to the IP Address of the router on the IP Office subnet 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.
- Click **OK** (not shown) to save any changes.



5.7. Administer SIP Line

A SIP line is created to establish the SIP connection between the Avaya IP Office and Clearcom SIP Trunk Services. 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**.

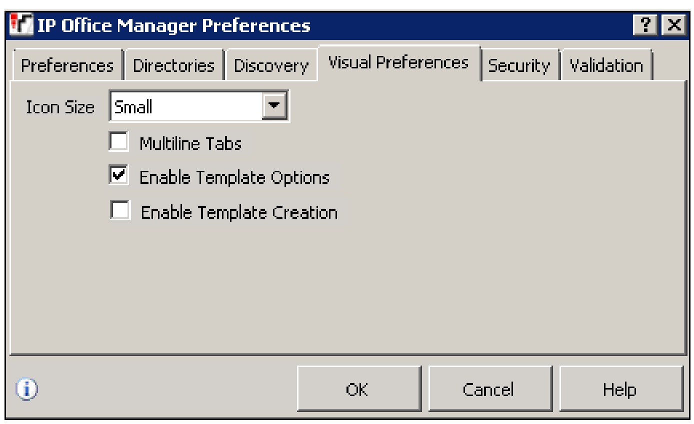
5.7.1. Importing a SIP Line Template

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.

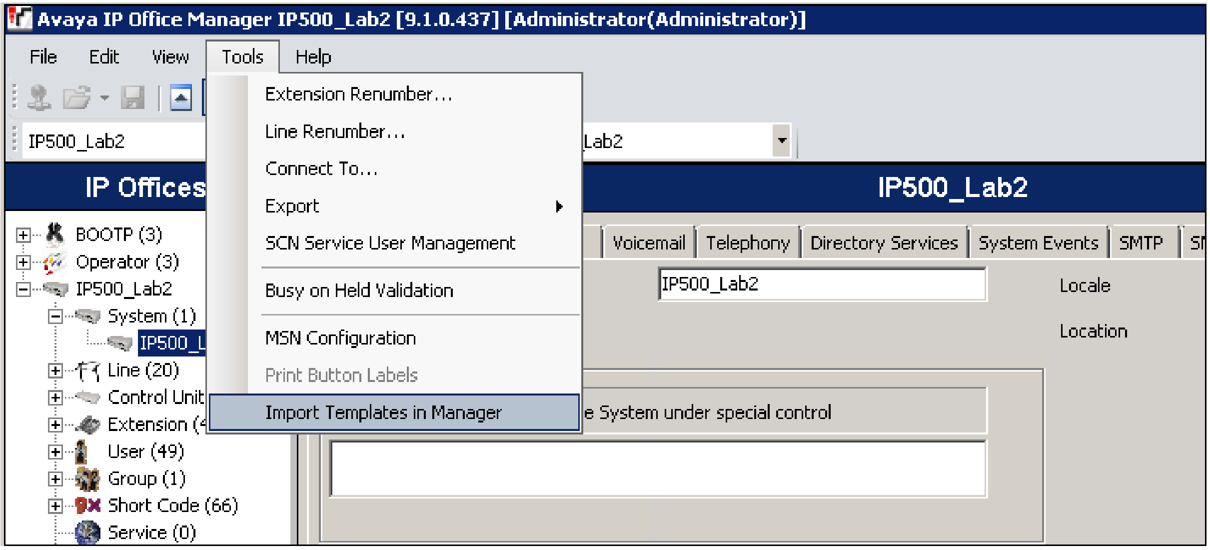
1. Copy a previously created template file to a location (e.g., *\Temp*) on the same computer where IP Office Manager is installed. By default, the template file name will have the format **AF_<*user supplied text*>_SIPTrunk.xml**, where the **<*user supplied text*>** portion is entered during template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

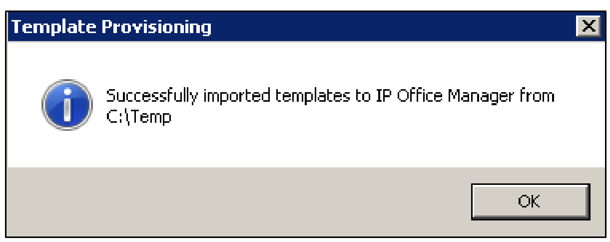


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**.

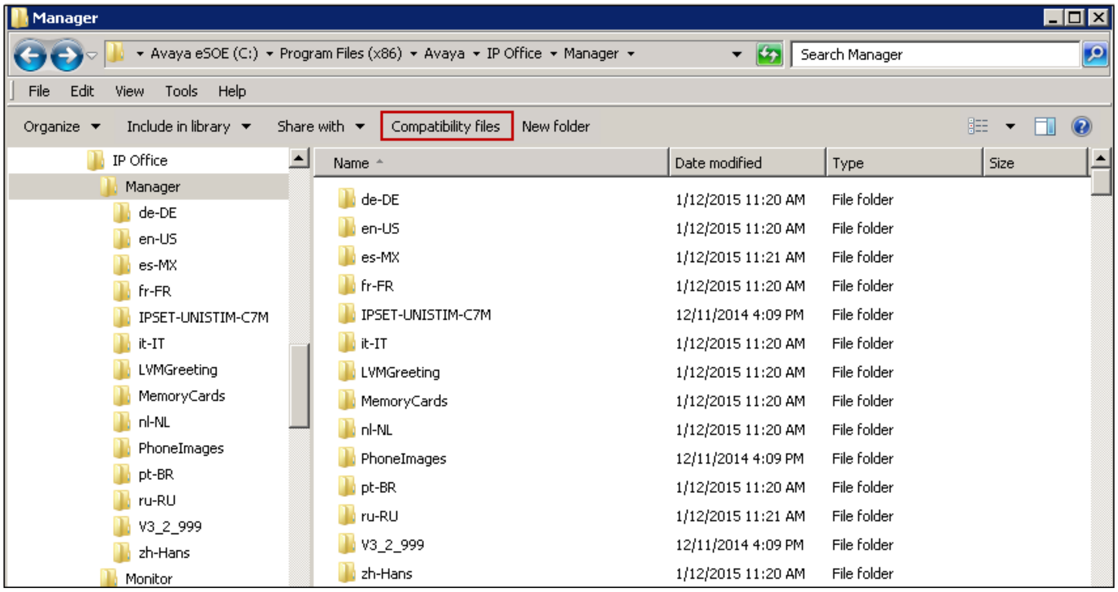


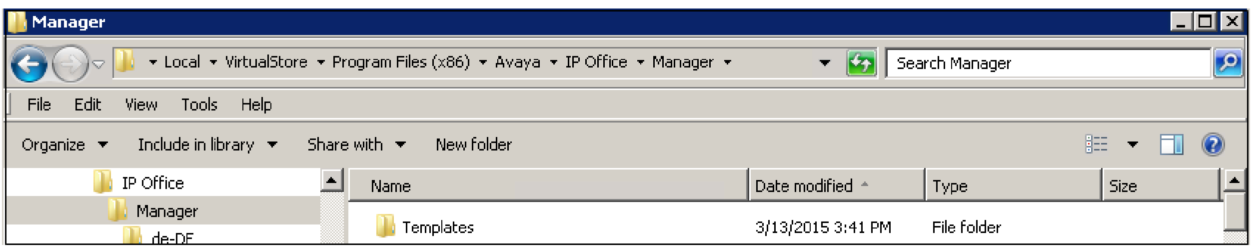
4. A folder browser will open (not shown). Select the directory used in **step 1** to store the template (e.g., *Temp*). In the reference configuration, template file **AF_*Clearcom*_ SIPTrunk.xml** was imported. The template file is automatically copied into the default template location, **C:\Program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

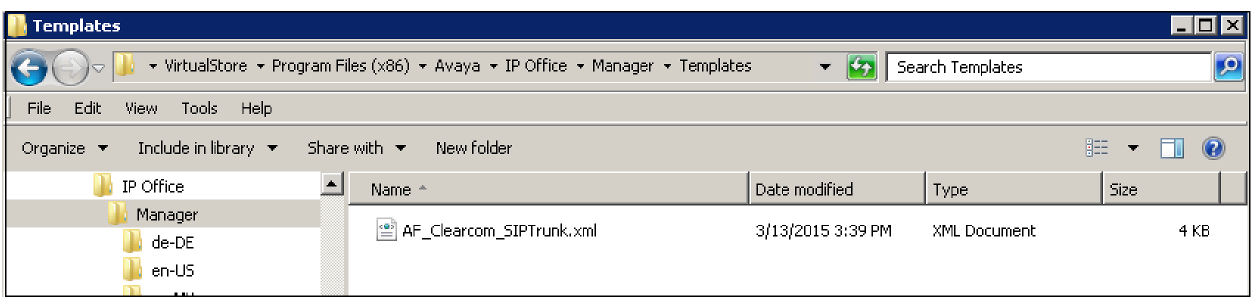
5. After the import is complete, a final import status pop-up window will open stating success or failure. Click **OK**.



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.

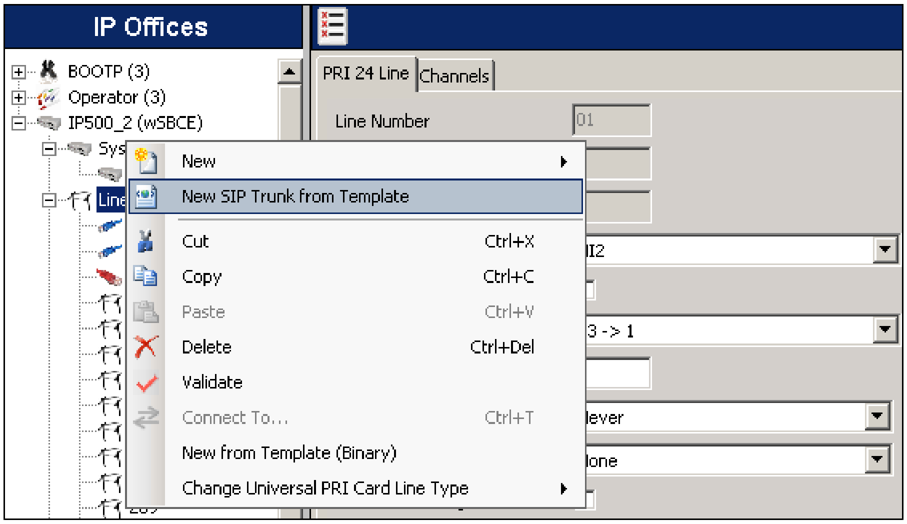






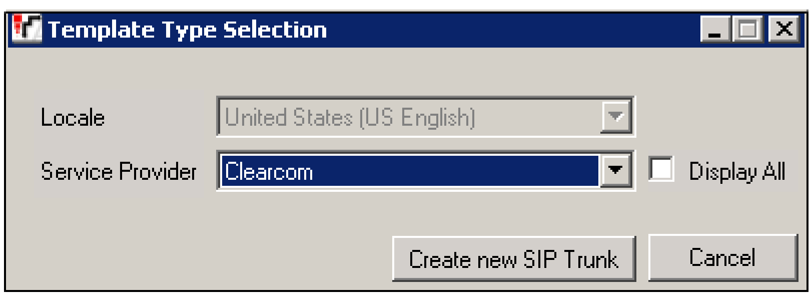
5.7.2. Creating a SIP Trunk from an XML Template

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 you check the **Display All** box, then the full template file name is displayed.



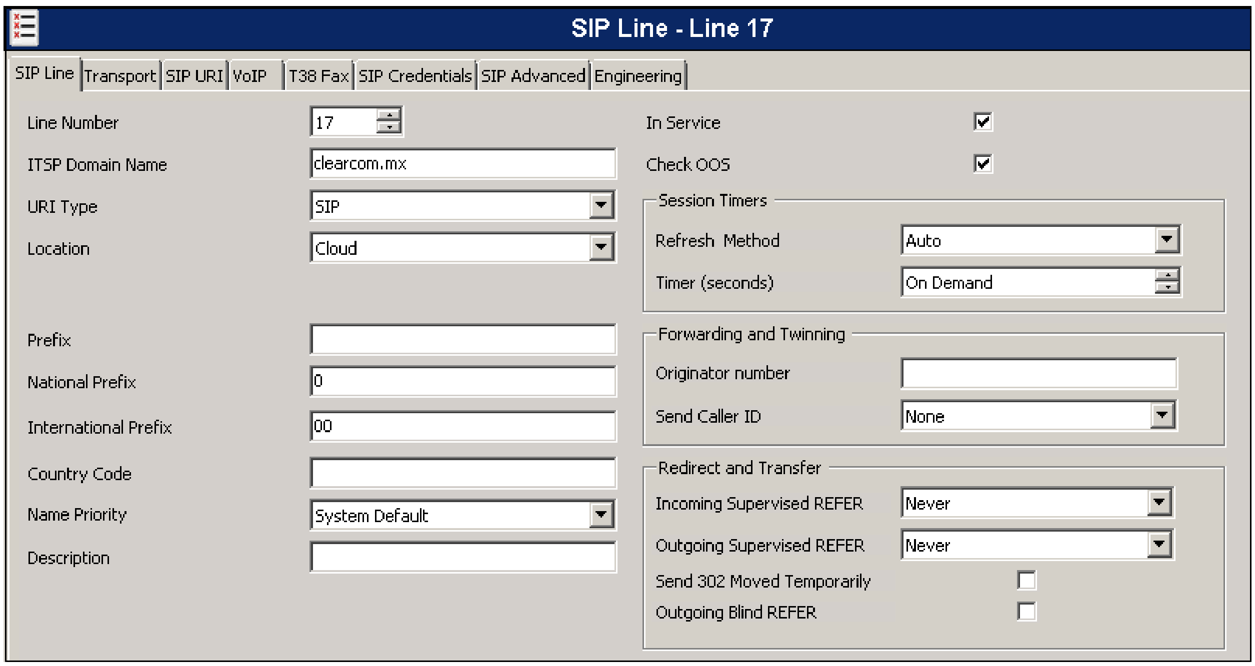
Click **Create new SIP Trunk** to finish creating the trunk.

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:

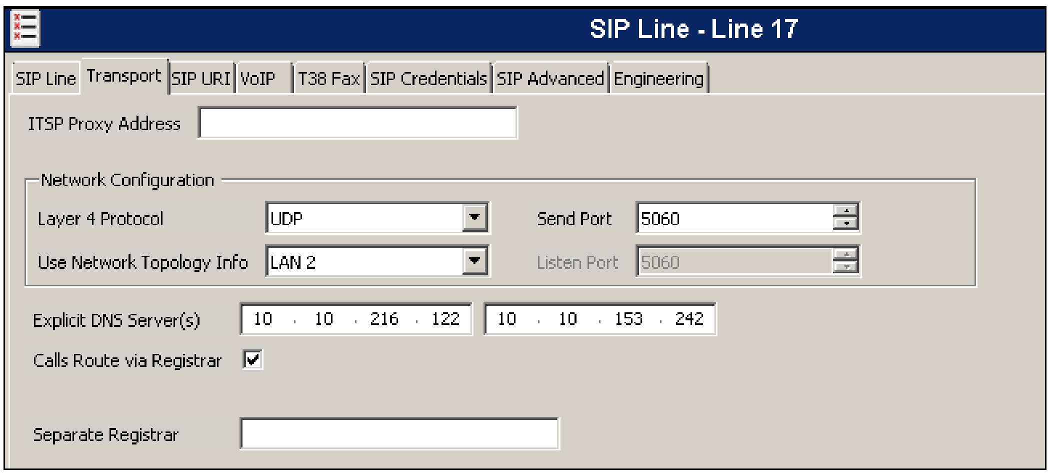
- Set the **ITSP Domain Name** to ***clearcom.mx***, the domain known and expected by Clearcom on the SIP trunk. Avaya IP Office will use this domain as the host portion of the SIP URI of SIP headers in messages sent to the network.
- Check the **In Service** box.
- Check the **Check OOS** box.
- On the **Forwarding and Twinning** section, set **Send Caller ID** to ***None***. This field is not used in this configuration. On outbound calls, the caller ID number shown on the PSTN end was always the main number assigned by Clearcom to the enterprise, regardless of the actual number sent in any of the origination headers from the Avaya IP Office.
- On the **Redirect and Transfer section**, set **Incoming Supervised REFER** and **Outgoing Supervised REFER** to ***Never***, to disable the use of REFER in call transfers to the PSTN. See **Section 2.2**.
- Default values may be used for all other parameters.



5.7.4. Transport Tab

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

- Leave the **ITSP Proxy Address** blank. Avaya IP Office will use the ITSP Domain Name entered in the **SIP Line** tab to resolve the public IP address of the Clearcom SIP proxy server, using DNS resolution.
- 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***.
- Set **Explicit DNS Server(s)** to the IP addresses of the primary and secondary public DNS Servers used by the enterprise. This information should be provided by the local ISP.
- Default values may be used for all other parameters.

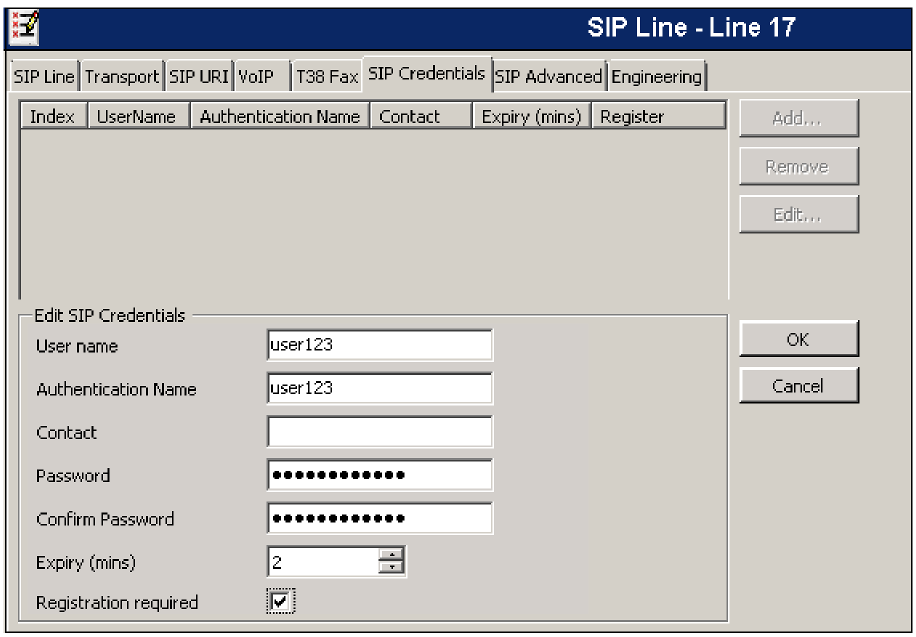


5.7.5. SIP Credentials

Clearcom 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**, **Authentication Name** and **Password** to the values provided by the service provider.
- Set **Expiry (mins)** to a value acceptable to the service provider. This setting defines how often Avaya IP Office needs to send REGISTERs to Clearcom in order to renew the SIP trunk registration. The Expiry value is negotiated and agreed as part of the registration process. For the compliance test, a value of ***2*** minutes was used.
- Check the **Registration required** box.
- Click **OK**.



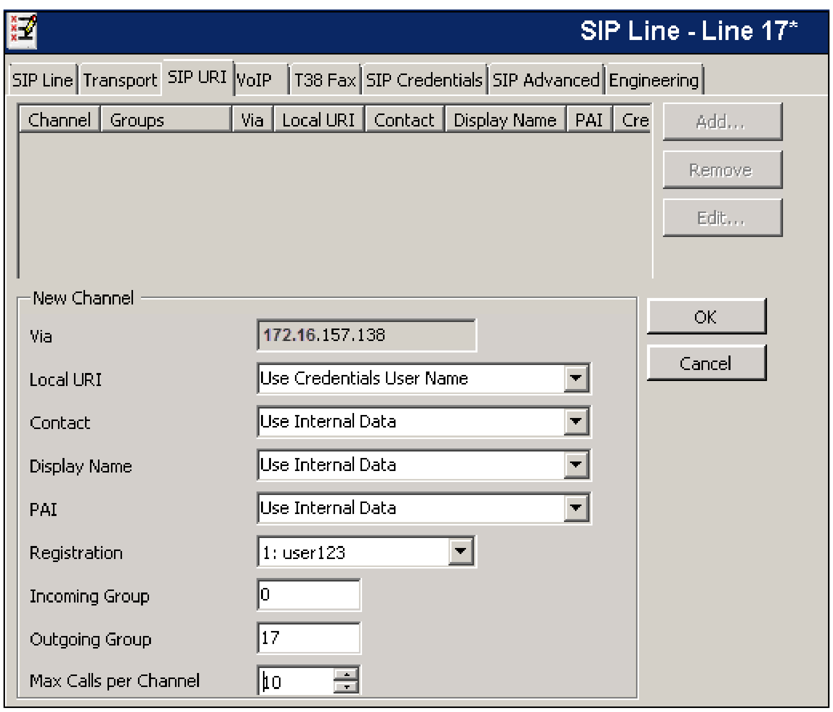
5.7.6. SIP URI Tab

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:

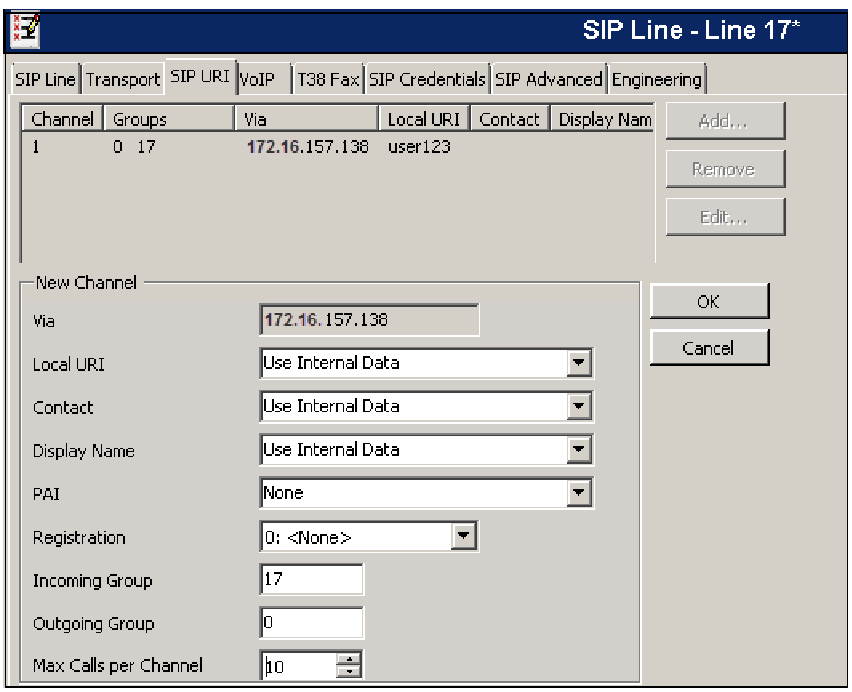
- Set **Local URI** to ***Use Credentials User Name***. Clearcom required the user name configured in **Section 5.7.5** to be sent in the "From" header of all outbound requests sent to the network.
- Set **Contact**, **Display Name** and **PAI** to ***Use Internal Data***.
- Under **Registration**, select ***1: <user123>*** from the pull-down menu. Clearcom uses Digest Authentication to challenge all calls made from the enterprise to the PSTN.

  Avaya 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**.

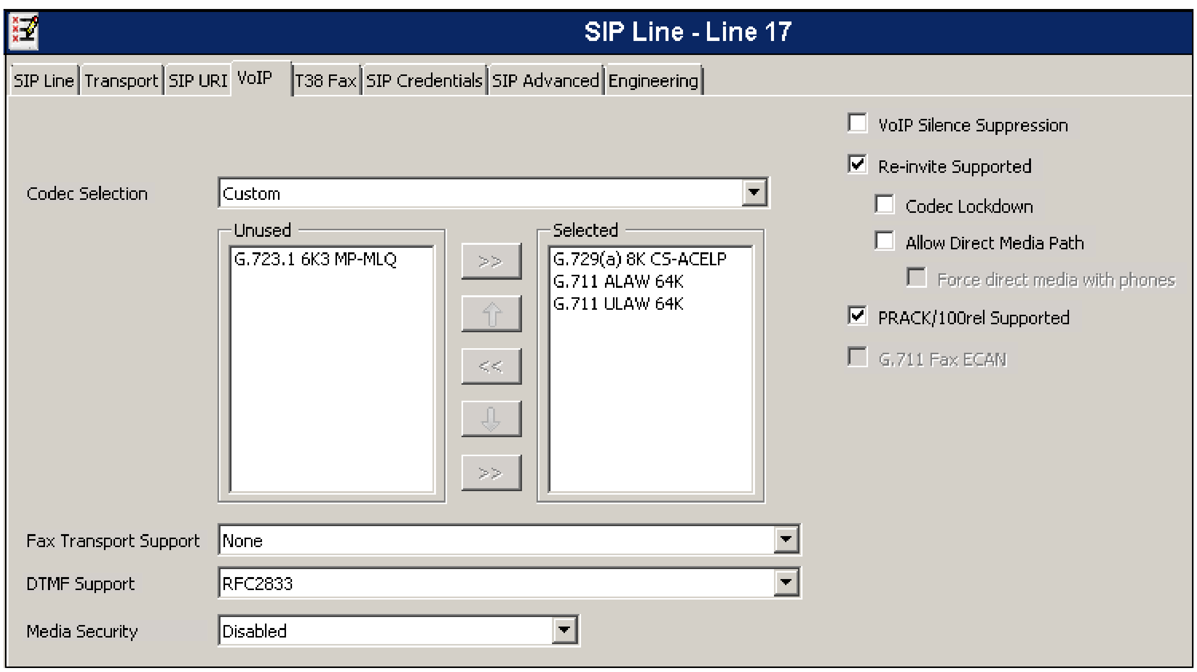


Additional SIP URIs may be required to allow inbound calls to numbers not associated with a user, such as a short code. These URIs are created in the same manner as shown previously, with the exception that the incoming DID number is entered directly in the **Local URI**, **Contact**, and **Display Name** 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. During the compliance test, **G729A**, **G711A and G711U**, in this order of preference, were the codecs supported by Clearcom.
- Set **Fax Transport Support** to *None*. See **Section 2.2**.
- Set the **DTMF Support** field to ***RFC2833.*** This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Check the **Re-invite Supported** box to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk.
- Check the **PRACK/100rel Supported** box, to advertise the support for provisional responses and Early Media to the service provider.
- Default values may be used for all other parameters.

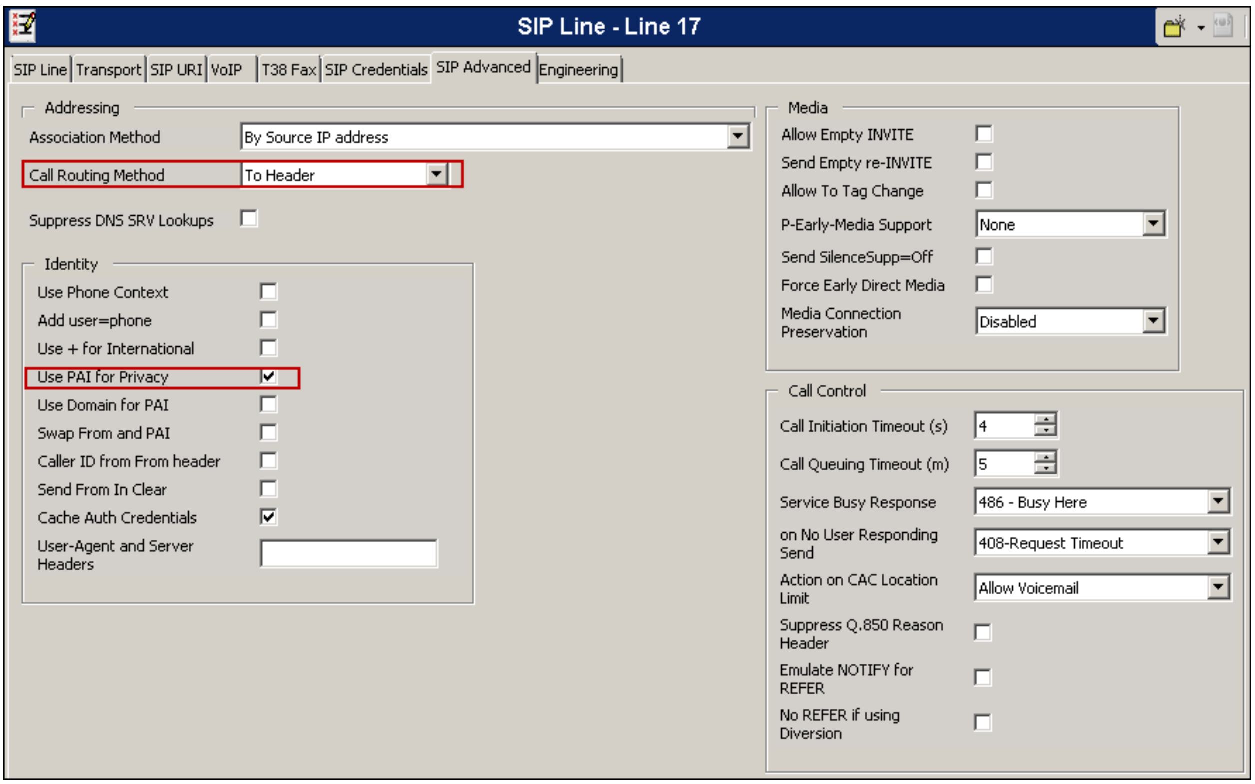


5.7.8. SIP Advanced Tab

On the **SIP Advanced** tab, select ***To Header*** from the **Call Routing Method** pull down menu. On inbound calls, Clearcom will include the user name associated with the SIP trunk credentials in the Request URI header of every incoming INVITEs, while the actual DID number of the party being called is sent in the "To" header. Avaya IP Office will use the "To" header on the incoming INVITE to route the call to the intended destination, ignoring the Request URI header in this decision.

For outbound calls with privacy enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with "anonymous". Avaya IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing purposes. By default, Avaya IP Office will use the PPI header for privacy. To configure Avaya IP Office to use the PAI header for privacy calls, check the box for **Use PAI for Privacy**.

All other fields retained their default values.



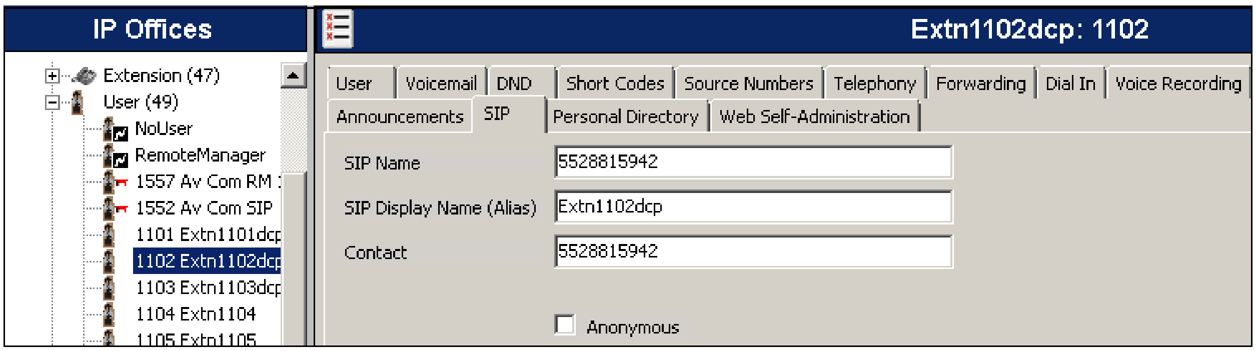
Click **OK** (not shown) to save any changes made to any of the various "SIP Line" tabs.

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

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 ***Extn1102dcp***. 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 Clearcom. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. Click **OK** (not shown) to save any changes.



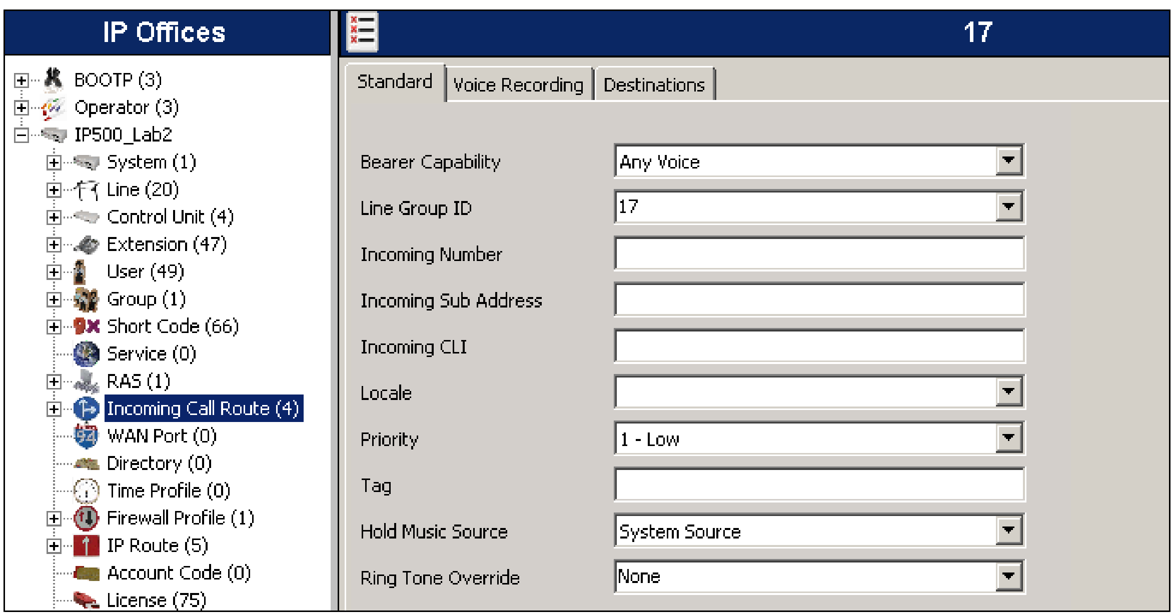
5.9. Incoming Call Route

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

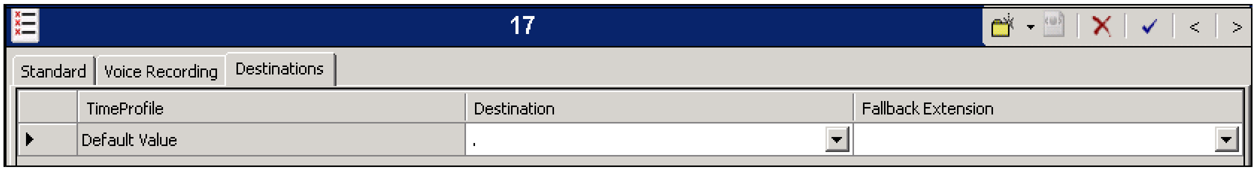
In a scenario like the one used for the compliance test, only one incoming route was needed, which allowed any incoming number arriving on the SIP trunk to reach any predefined extension in the Avaya IP Office. The routing decision for the call is based on the parameters previously configured for the **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.



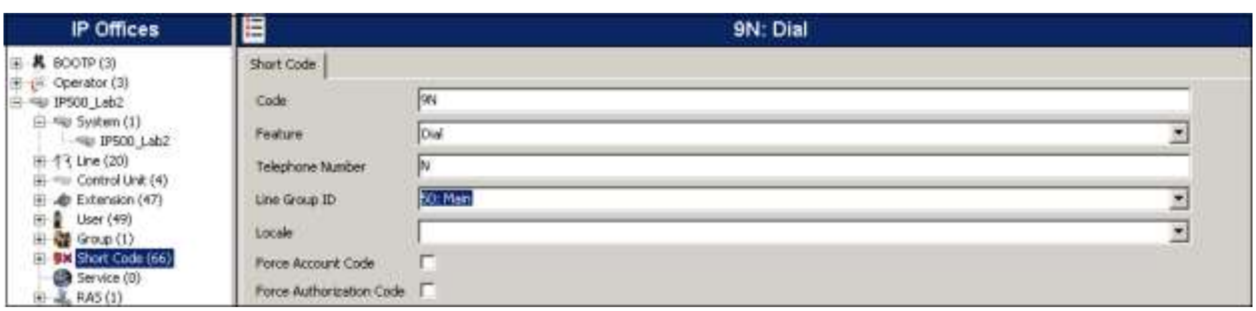
Additional incoming call routes may be required to allow inbound calls to numbers not associated with a user, such as a short code. These routes are created in the same manner as shown, with the exception that the incoming DID number is entered directly in the **Incoming Number** field on the **Standard**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

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 5.11**

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

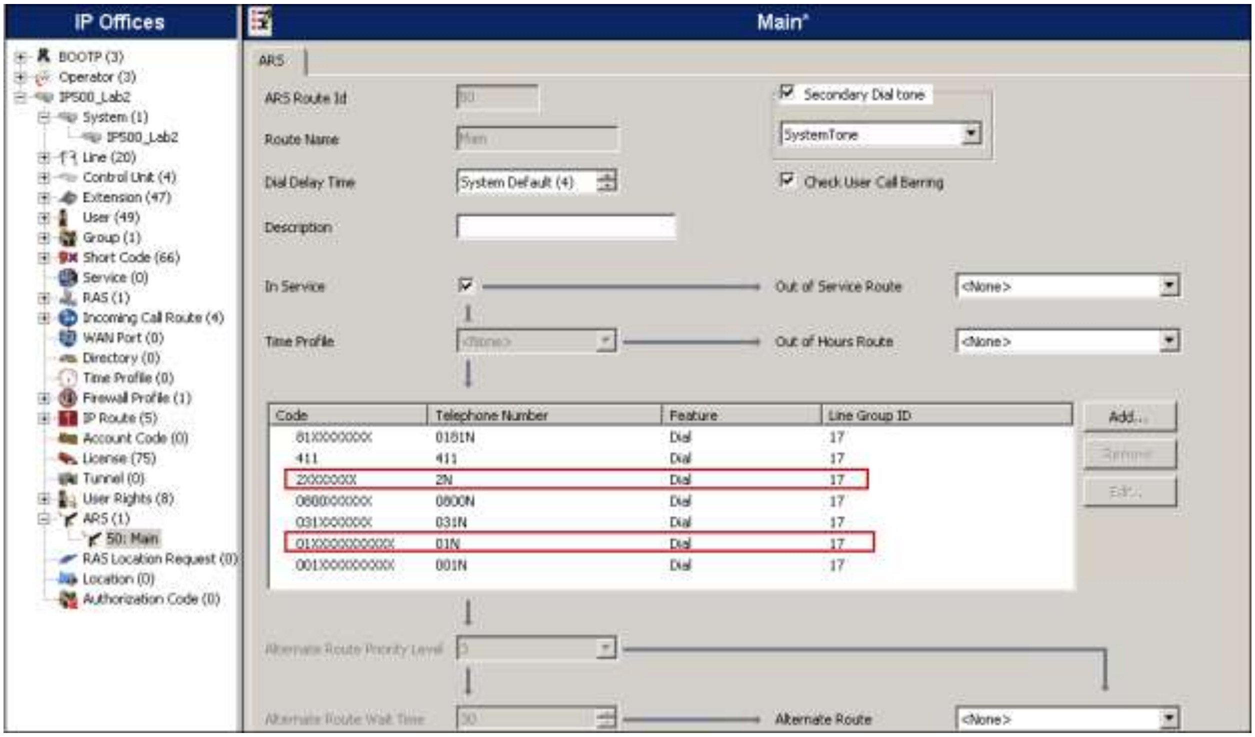
- In the **Code** field, enter the dial string which will trigger this short code, in this case ***9N***. This short code will be invoked when the user dials 9 followed by any number.
- Set **Feature** to ***Dial***. This is the action that the short code will perform.
- Set **Telephone Number** to **N**.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



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.

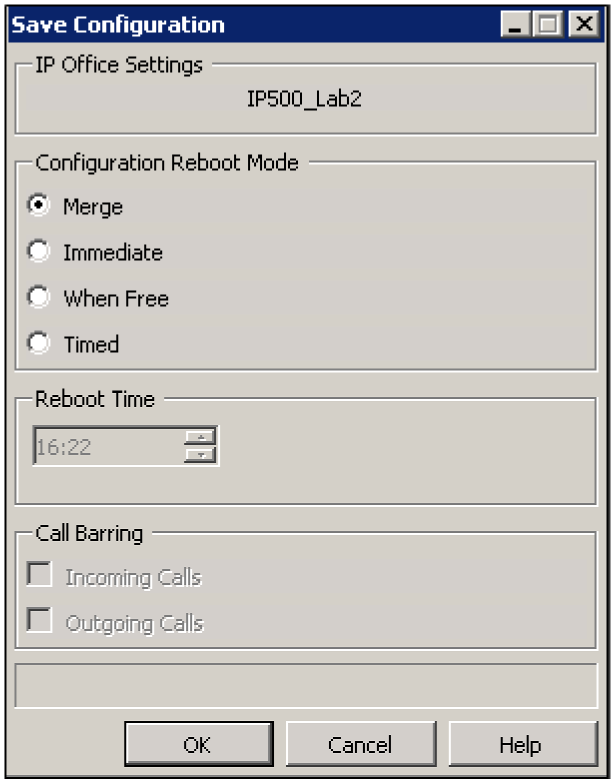
The following screen shows the ARS configuration for the route ***50: Main***. The example shows a subset of the dialed strings tested as part of the compliance test. See **Section 2.1** for the complete list of call types tested. Note the sequence of ***X***s used in the **Code** column of some entries, 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 Avaya IP Office. The highlighted entries show that for example, for calls in the local area code, the user dialed 9 plus the 8 digit local number, starting with a 2, which was the range of local numbers used during the compliance test. For national long distance calls in Mexico, the user dialed 9, then 01, followed by 10 digit numbers.



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.



6. Clearcom SIP Trunking Configuration

Clearcom 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. Clearcom will provide the customer the necessary information to configure the SIP trunk connection from the enterprise site to the network, including:

- SIP Trunk registration credentials (User name, password).
- Clearcom's SIP Domain Name.
- DID numbers.
- Supported codecs and order of preference
- Any IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices.

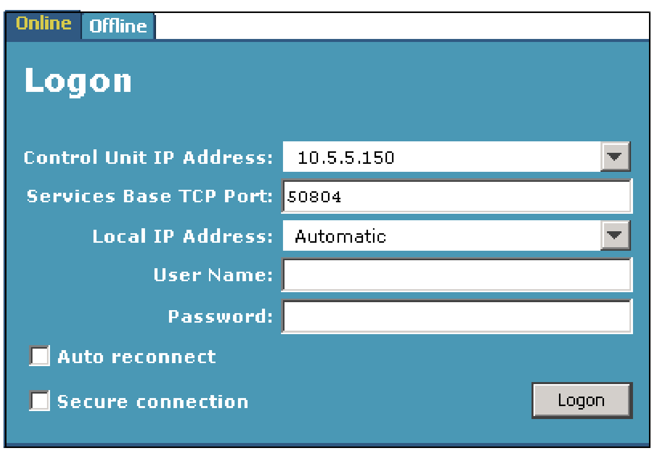
This information is used to complete the configuration of the Avaya IP Office discussed in the previous sections.

7. Verification Steps

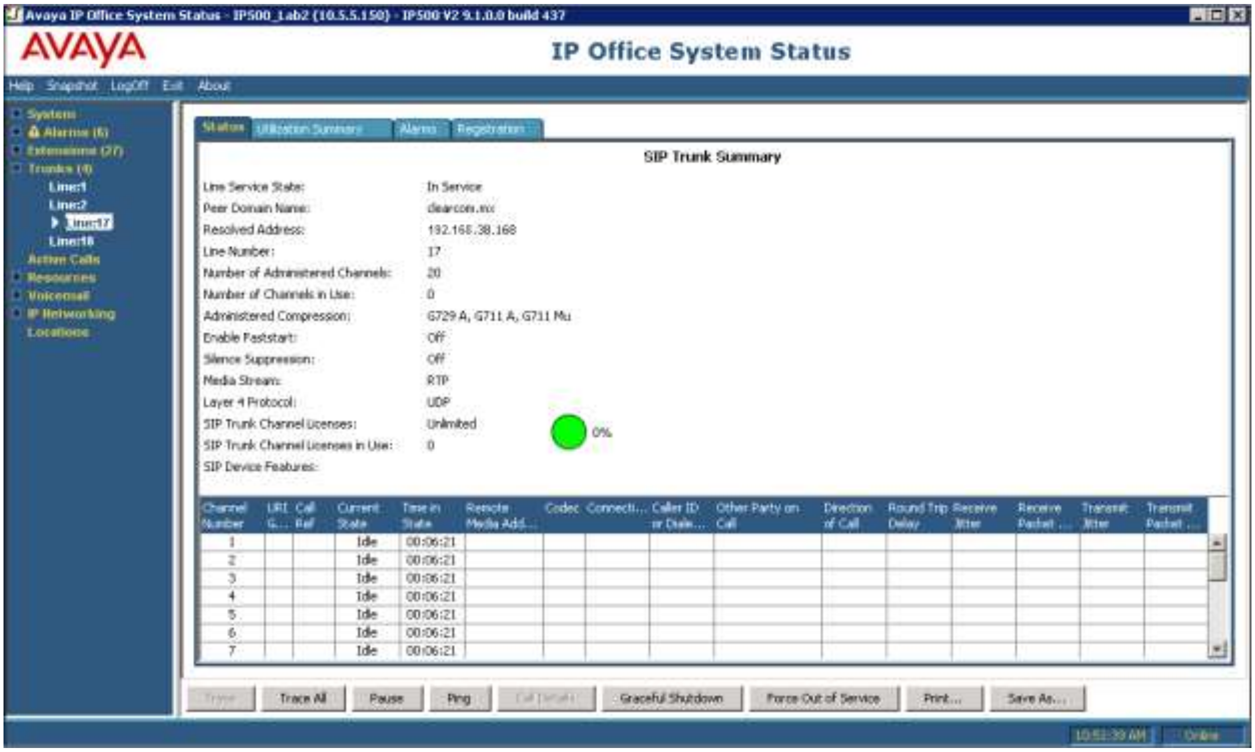
The Avaya IP Office System Status and Monitor applications are useful tools used for the verification and troubleshooting of the SIP connection to the service provider.

7.1. System Status

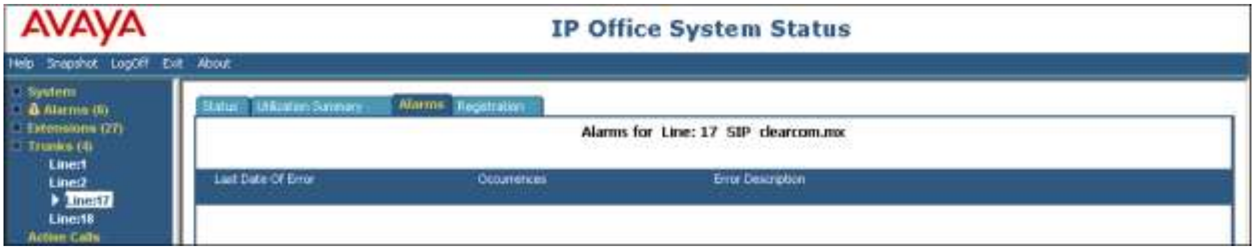
The Avaya IP Office System Status application can be used to verify the service state of the SIP line. Launch the application from **Start → Programs → IP Office → System Status** on the PC where Avaya IP Office Manager was installed. Under **Control Unit IP Address** select the IP address of the Avaya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



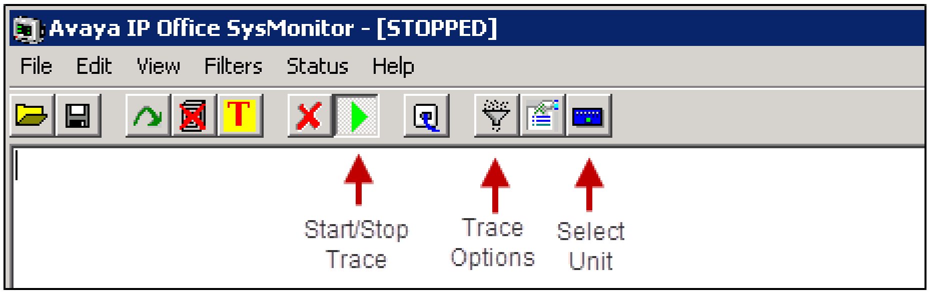
Select the **Alarms** tab and verify that no alarms are active on the SIP line.



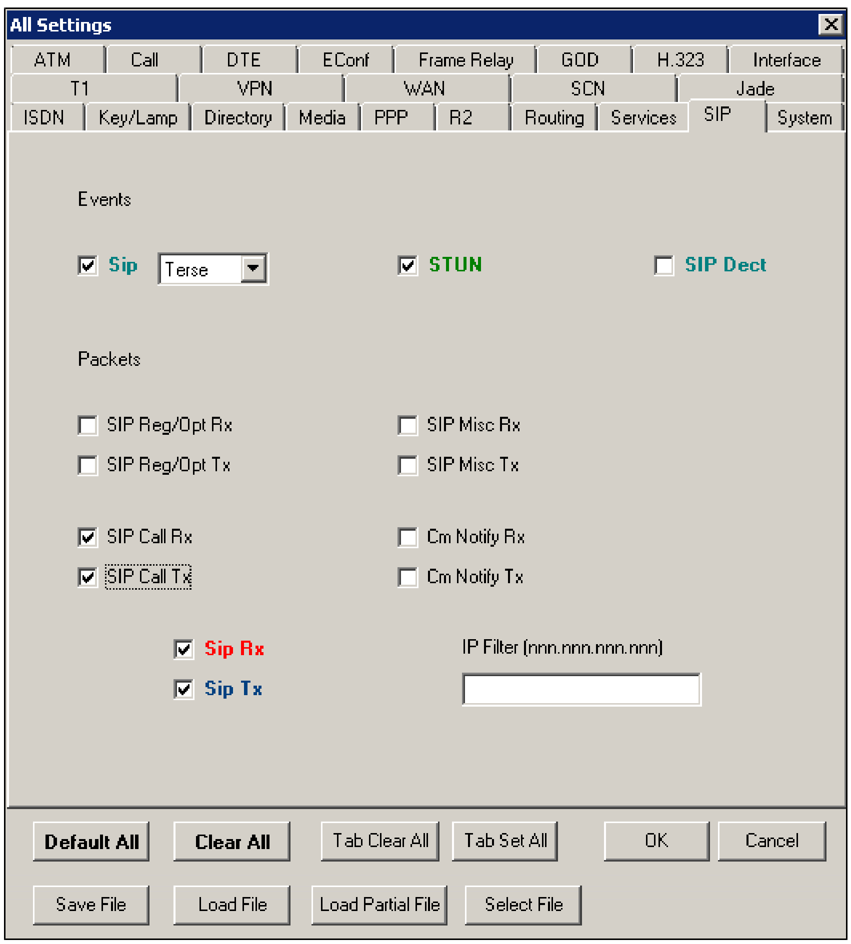
On the **Registration** tab, verify that the trunk is successfully registered with the service provider.

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 Avaya 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 in the Avaya IP Office Release 9.1, to connect to Clearcom SIP Trunk Services, 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, January 2015
https://downloads.avaya.com/css/P8/documents/101005082

[2] *Administering Avaya IP Office Platform with Manager*, *Release 9.1.0*, January 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, October 2014
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 Clearcom SIP Trunk Service is available from Clearcom.

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.