Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office Release 9.1 to support Claro SIP Trunking Service – Issue 1.0

Abstract

These Application Notes describe the steps necessary for configuring Session Initiation Protocol (SIP) Trunk Service for an enterprise solution consisting of Avaya IP Office Release 9.1 to interoperate with Claro SIP Trunking Service.

The test was performed to verify SIP trunk features including basic calls, call forward (all calls, busy, no answer), call transfer (blind and consult), conference, and voice mail. The calls were placed to and from the PSTN with various Avaya endpoints.

Claro SIP Trunking Service provides PSTN access via a SIP trunk between the enterprise and Claro'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.

Claro is a member of the Avaya DevConnect Service Provider Program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the steps necessary for configuring Session Initiation Protocol (SIP) Trunk Service for an enterprise solution consisting of Avaya IP Office Release 9.1 to interoperate with Claro SIP Trunking Service.

In the sample configuration, the Avaya IP Office solution consists of Avaya IP Office (hereafter referred to as IP Office) 500v2 Release 9.1 and various Avaya endpoints, including Avaya Communicator for Windows, Avaya IP Office Video Softphone and Avaya deskphones, including SIP, H.323, digital, and analog.

During the interoperability testing, feature test cases were executed to ensure interoperability between Claro and IP Office.

During the interoperability testing, a VPN connection was used to connect the simulated Avaya enterprise network to Claro's network via the public Internet. The connection could also be done without the use of VPN.

The Claro SIP Trunking Service referenced within these Application Notes is designed for business customers. Customers using this service with the IP Office solution are able to place and receive PSTN calls via a broadband WAN connection using the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as analog and/or ISDN-PRI trunks. This approach generally results in lower cost for the enterprise.

The terms "Service Provider" and "Claro" will be used interchangeable throughout these Application Notes.

2. General Test Approach and Test Results

The general test approach was to simulate an enterprise site in the Avaya Solution & Interoperability Test Lab by connecting IP Office to Claro SIP Trunking Service via the public internet, as depicted in **Figure 1**.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

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2.1. Interoperability Compliance Testing

To verify SIP trunk interoperability, the following areas were tested for compliance:

- SIP OPTIONS queries and responses.
- Incoming calls from the PSTN were routed to the DID numbers assigned by Claro. Incoming PSTN calls were terminated to the following endpoints: Avaya 96x0 Series IP Deskphones (H.323), Avaya 96x1 Series IP Deskphones (H.323), Avaya 1100 Series IP Deskphones (SIP), Avaya Communicator for Windows, Avaya IP Office Video Softphone, Avaya 1400 Series Digital Deskphones, Avaya 9500 Series Digital Deskphones, and analog Deskphones.
- Outgoing calls to the PSTN were routed via Claro's network to various PSTN destinations.
- Caller ID presentation.
- Proper disconnect when the caller abandons the call before the call is answered.
- Proper disconnect via normal call termination by the caller or the called parties.
- Proper disconnect by the network for calls that are not answered (with voicemail off).
- Proper response to busy endpoints.
- Proper response/error treatment when dialing invalid PSTN numbers.
- Proper Codec negotiation and two way speech-path. (Testing was performed with codecs: G.711MU, G.711A and G.729A, Claro's preferred codec order).
- No matching codecs.
- Proper early media transmissions.
- Voicemail and DTMF tone support (leaving and retrieving voice mail messages from PSTN phones).
- Outbound Toll-Free calls, interacting with IVR (Interactive Voice Response systems).
- Calling number blocking (Privacy).
- Call Hold/Resume (long and short duration).
- Call Forward (unconditional, busy, no answer).
- Blind Call Transfers.
- Consultative Call Transfers.
- Station Conference.
- Mobility twinning of incoming calls to mobile phones.
- Simultaneous active calls.
- Long duration calls (over one hour).
- Proper response/error treatment to all trunks busy.
- Proper response/error treatment when disabling SIP connection.
- T.38 fax.
- G.711 Pass-through fax.

Note: Remote worker was not tested with this solution since the Avaya Session Border Controller for Enterprise (Avaya SBCE) was not included in the configuration used during the compliance testing. The Avaya SBCE is a required component for enterprises planning to deploy Remote Workers.

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Items not supported or not tested included the following:

- The SIP REFER method for call redirection was not tested for reasons noted under **Section 2.2**.
- Inbound toll-free calls, Outbound Toll-Free calls, 911 calls (emergency), "0" calls (Operator), 0+10 digits calls (Operator Assisted), and 411 calls (Local Directory Assistance) were not tested.

2.2. Test Results

Interoperability testing of Claro SIP Trunking Service was completed successfully with the following observations/limitations.

- **SIP REFER**: Calls from the PSTN to IP Office that are re-directed to another PSTN endpoint by the IP Office user, with REFER enabled in IP Office, did not work properly. Testing was done with REFER disabled in IP Office, refer to **Section 5.4.3**.
- **Caller ID display on Call Forwards to the PSTN**: For Calls from the PSTN to IP Office, that were Forwarded back out to the PSTN, the caller ID number displayed at the PSTN was always of the first DID number assigned to the SIP Trunk , regardless of the PSTN number being used to originate the call.
- **Caller ID display on Mobile Twinning**: For Mobile Twinning calls, the Caller ID display at the Mobile/cellular station was always of the first DID number assigned to the SIP Trunk, regardless of the PSTN number being used to originate the call.
- **No matching codec on outbound calls**: If an unsupported audio codec is received by Claro on the SIP Trunk (e.g., 722-64K), Claro will respond with "500 Server Internal Error" instead of "488 Not Acceptable Here", and the user will hear re-order. This issue does not have any user impact, it is listed here simply as an observation.

2.3. Support

For support on Claro systems visit the corporate Web page at:
http://www.claro.com.do/wps/portal/do/sc/empresas.

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. Alternatively, in the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus.

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3. Reference Configuration

Figure 1 illustrates the test configuration used for the DevConnect compliance testing. The test configuration simulates an enterprise site with an Avaya SIP-enabled enterprise solution connected to the Claro SIP Trunking Service through the public Internet.

The Avaya components used to create the simulated enterprise customer site includes:

- Avaya IP Office 500v2.
- Avaya Voicemail Pro for IP Office.
- Avaya 96x0 Series H.323 IP Deskphones.
- Avaya 96x1 Series H.323 IP Deskphones.
- Avaya 1100 Series SIP IP Deskphones.
- Avaya Communicator for Windows Softphone.
- Avaya IP Office Video Softphone.
- Avaya 1408 Digital Deskphones.
- Avaya 9508 Digital Deskphones.
- Analog Deskphone.
- Fax Machines.

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 IP Office is connected to the enterprise LAN (private IP network) while the LAN2 port is connected to Claro's network via a VPN connection. Endpoints include Avaya 96x0 and 96x1 Series IP Deskphones (with H.323 firmware), Avaya 1100 Series IP Deskphones (with SIP firmware), Avaya 1408 and 9508 Digital Deskphones, Analog Deskphones, Fax Machines, and a PC running Avaya Communicator for Windows and Avaya IP Office Video Softphones. The site also has a Windows PC running Avaya IP Office Manager to configure and administer the IP Office system, and Avaya Voicemail Pro providing voice messaging service to the IP Office users. Mobile Twinning is configured for some of the IP Office users so that calls to these user's extensions will also ring and can be answered at the configured mobile phones.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to make calls across the SIP trunk to the PSTN via Claro's network. The short code 9 was stripped off by IP Office but the remaining "N" digits were sent unaltered to the network. Refer to **Section 5.6**.

In an actual customer configuration, the enterprise site may include additional network components between the service provider and the IP Office system, such as firewalls. 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 IP Office system must be allowed to pass through these devices.

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For confidentiality and privacy purposes, public IP addresses, user names, passwords and DID numbers used during the compliance testing have been masked.

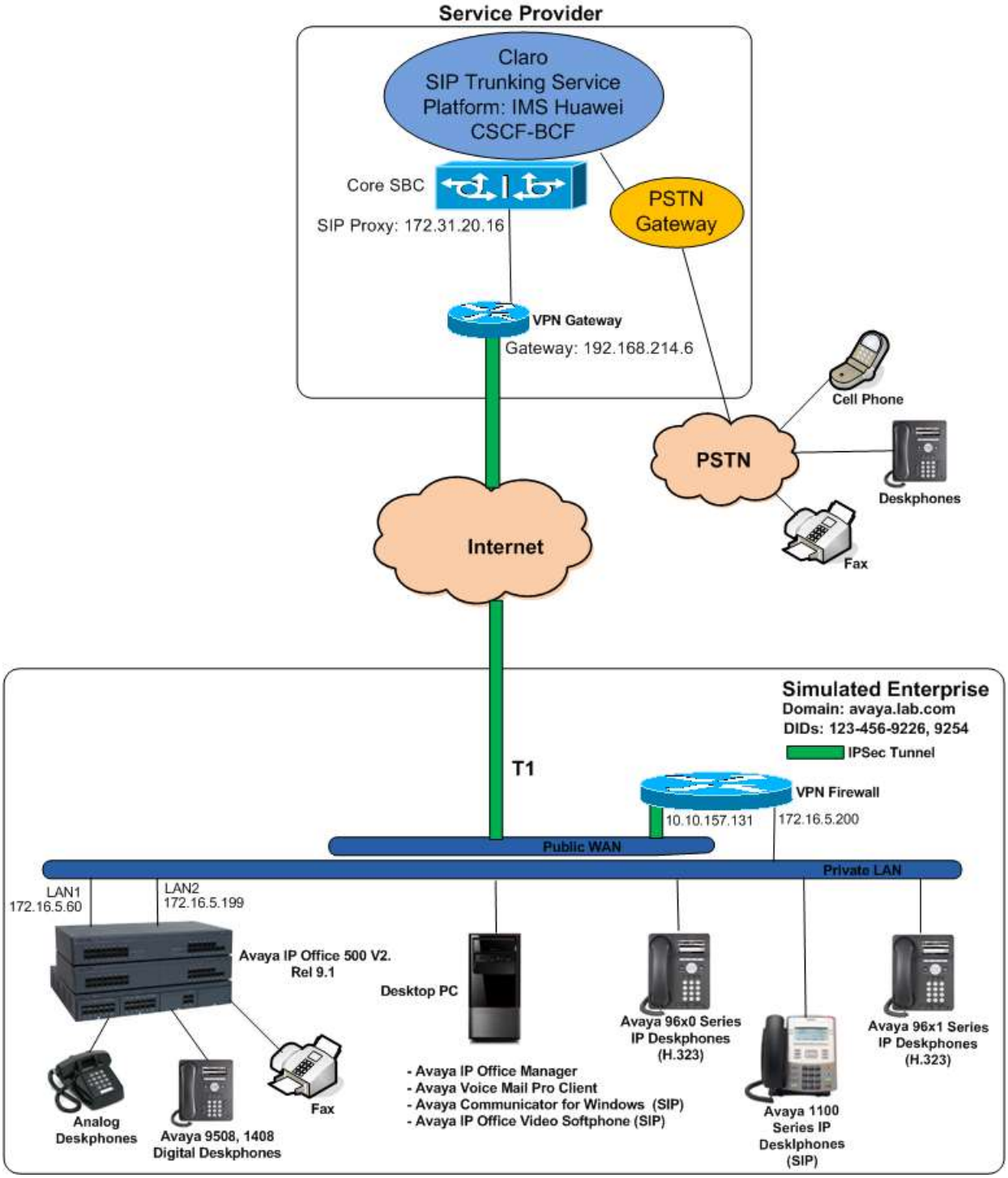

Figure 1: Avaya Interoperability Test Lab Configuration

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4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Avaya IP Office 500v2	9.1.1.0 Build 10
Avaya IP Office DIG DCPx16 V2	9.1.1.0 Build 10
Avaya IP Office Manager	9.1.1.0 Build 10
Avaya Voicemail Pro Client	9.1.1.0 Build 3
Avaya 96x0 Series IP Deskphones (H.323)	Avaya one-X® Deskphone Edition S3.230A
Avaya 96x1 Series IP Deskphones (H.323)	Avaya one-X® Deskphone H.323 Version 6.4014
Avaya 1140E IP Deskphones (SIP)	SIP1140e Ver. 04.04.18.00
Avaya Communicator for Windows	2.0.3.30
Avaya IP Office Video Softphone	3.2.3.49 68975
Avaya Digital Deskphones 1408	40.0
Avaya Digital Deskphones 9508	0.55
Lucent Analog Phone	--
Claro	
IMS Huawei CSCF-BCF	V100R010C00SPC100
SBC Huawei SessionEngine2600	V200R009ENGC30SPC100

Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500v2 and also when deployed with all configurations of IP Office Server Edition without T.38 Fax Service.

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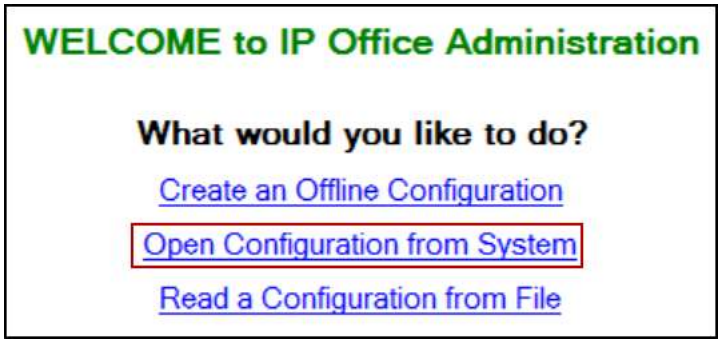
5. Configure Avaya IP Office

IP Office is configured through the Avaya IP Office Manager application. From the PC running Avaya IP Office Manager, select **Start → Programs → IP Office → Manager** to launch the application. A screen that includes the following may be displayed.



Select **Open Configuration from System**. If the above screen does not appear, the configuration may be alternatively opened by navigating to **File → Open Configuration** at the top of the Avaya IP Office Manager window. Select the proper IP Office system from the pop-up window and log in with the appropriate credentials.

The appearance of the Avaya IP Office Manager can be customized using the **View** menu. In the screens presented in this document, the **View** menu was configured to show the **Navigation** pane on the left side, omit the **Group** pane in the center, and show the **Details** pane on the right side. Since the **Group** pane has been omitted, its content is shown as submenus in the **Navigation** pane. These panes (**Navigation**, **Group** and **Details**) will be referenced throughout the IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the service provider is assumed to already be in place.

In the sample configuration, the MAC address **00E00706530F** was used as the system name. All navigation described in the following sections (e.g., **License → SIP Trunk Channels**) appears as submenus underneath the system name **00E00706530F** in the **Navigation** Pane.

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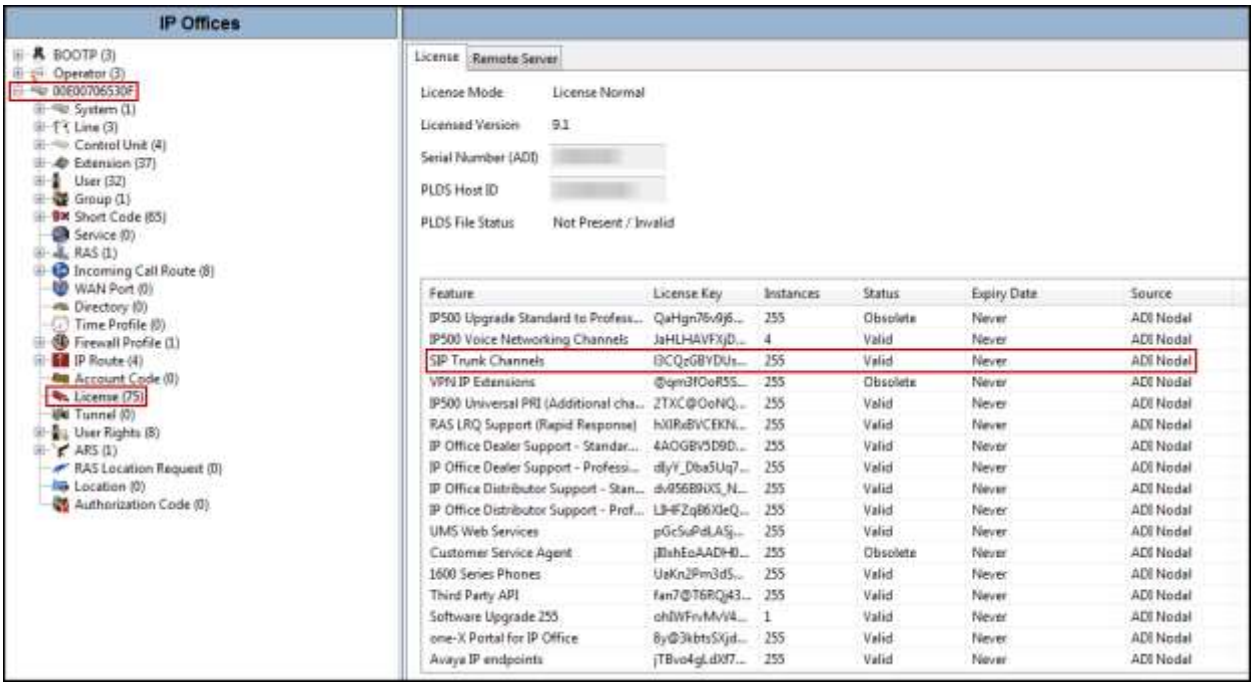
5.1. Licensing and Physical Hardware

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

To verify that there is a SIP trunk channels license with sufficient capacity; click **License** in the **Navigation** pane. Confirm a valid license with sufficient **Instances** (trunk channels) in the Details pane. Note that the full **License Key** in the screen below is not shown for security purposes.



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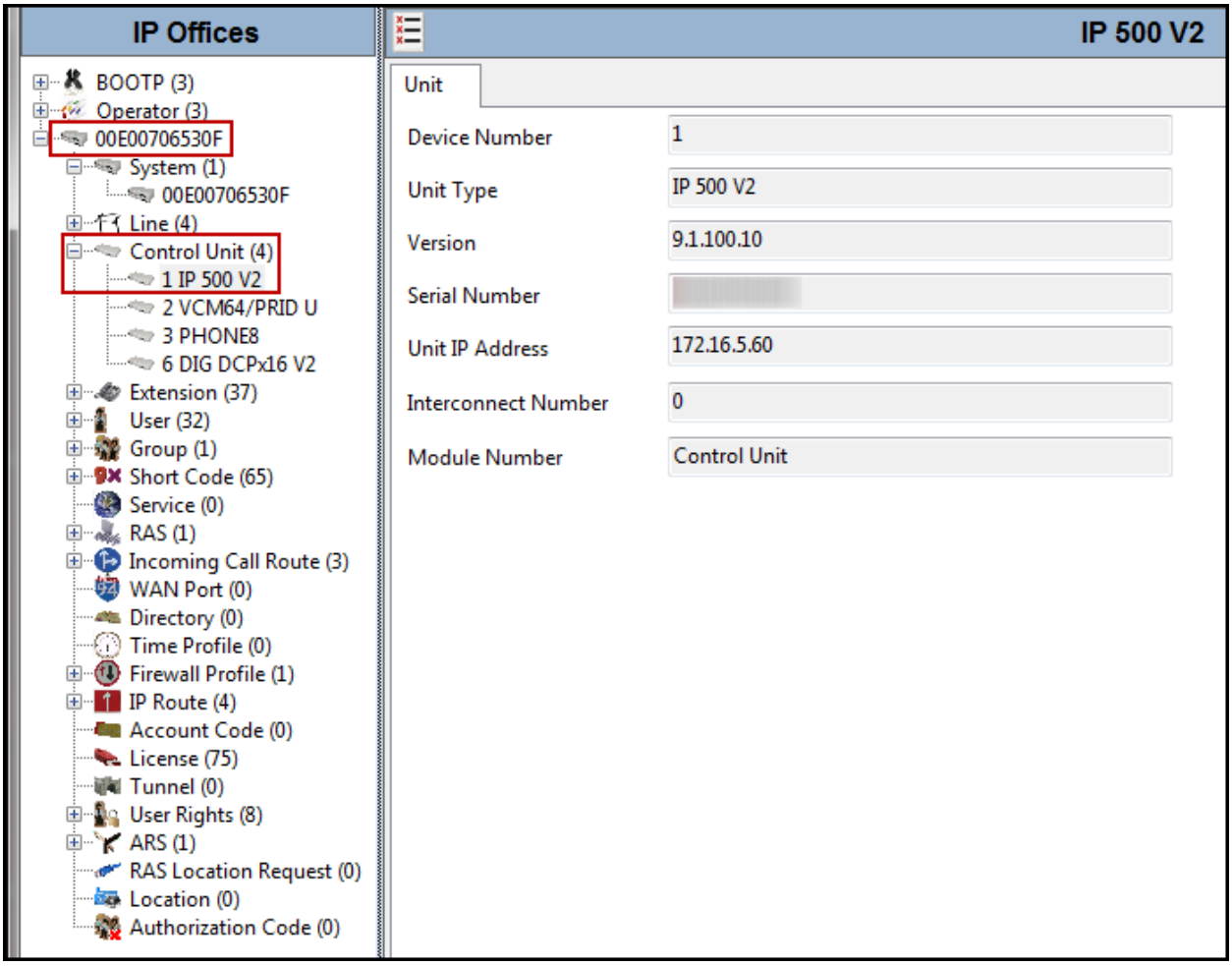
To view the physical hardware comprising IP Office, expand the components under the **Control Unit** in the **Navigation** pane. In the sample configuration, the Avaya IP Office 500v2 is equipped with analog and digital extension expansion modules, as well as a VCM64 (Voice Compression Module) for supporting VoIP codecs. An IP Office hardware configuration with a VCM component is necessary to support SIP Trunking Services.

To view the details of the component, select the component in the **Navigation** pane. The following screen shows the details of the **Avaya IP 500 V2**.



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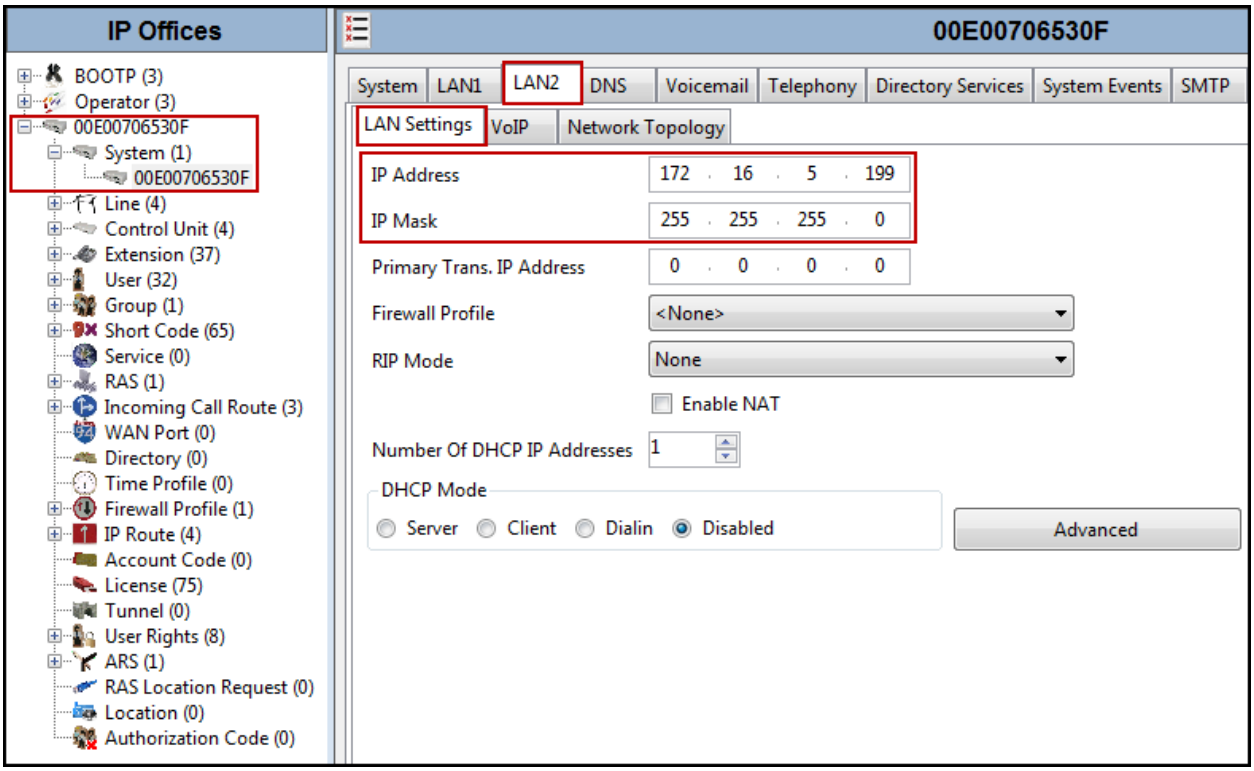
5.2. System

Configure the necessary system settings. In an IP Office system, the LAN2 tab settings correspond to the IP Office WAN port (public network side) and the LAN1 tab settings correspond to the LAN port (private network side).

5.2.1. System – LAN2 Tab

In the sample configuration, the IP Office WAN port was used to connect to Claro. The LAN2 settings correspond to the WAN port on the Avaya IP Office 500v2. To access the LAN2 settings, first navigate to **System** → *<Name>*, where *<Name>* is the system name assigned to IP Office. In the case of the compliance test, the system name is the MAC address **00E00706530F**. Next, navigate to the **LAN2** → **LAN Settings** tab in the **Details** pane. Set the **IP Address** field to a private IP address on the enterprise LAN side (private IP network). Set the **IP Mask** field to the mask used on the enterprise LAN side (private IP network). All other parameters should be set to default or according to customer requirements. Click **OK** to commit (not shown).

Note: A private IP address on the enterprise LAN side (private IP network) was used instead of a public IP address, since a VPN connection was used to connect to Claro's network. When a VPN connection is not used, a public IP address, instead of a private IP address, is assigned to **LAN2**.



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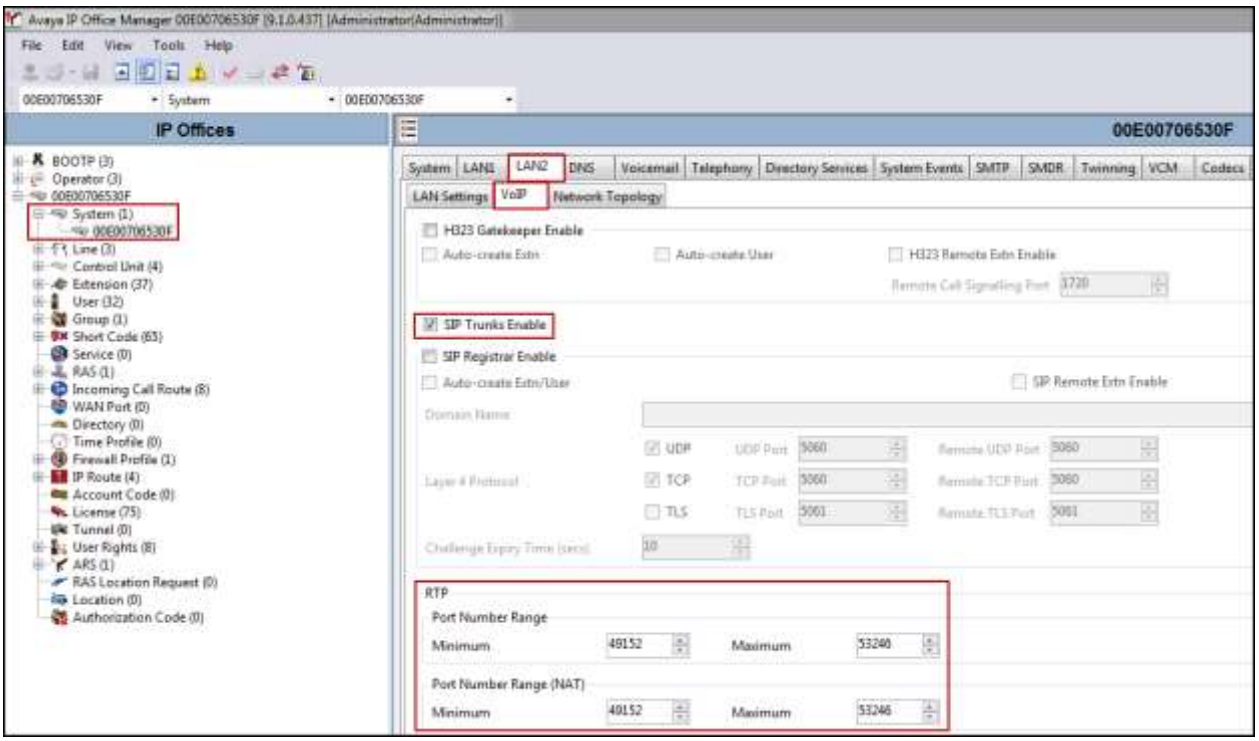
On the **VoIP** tab in the **Details** pane, configure the following parameters:

- Check the **SIP Trunks Enable** box to enable the configuration of SIP trunks.
- The **RTP Port Number Range** can be customized to a specific range of receive ports for RTP media. Based on this setting, IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2.



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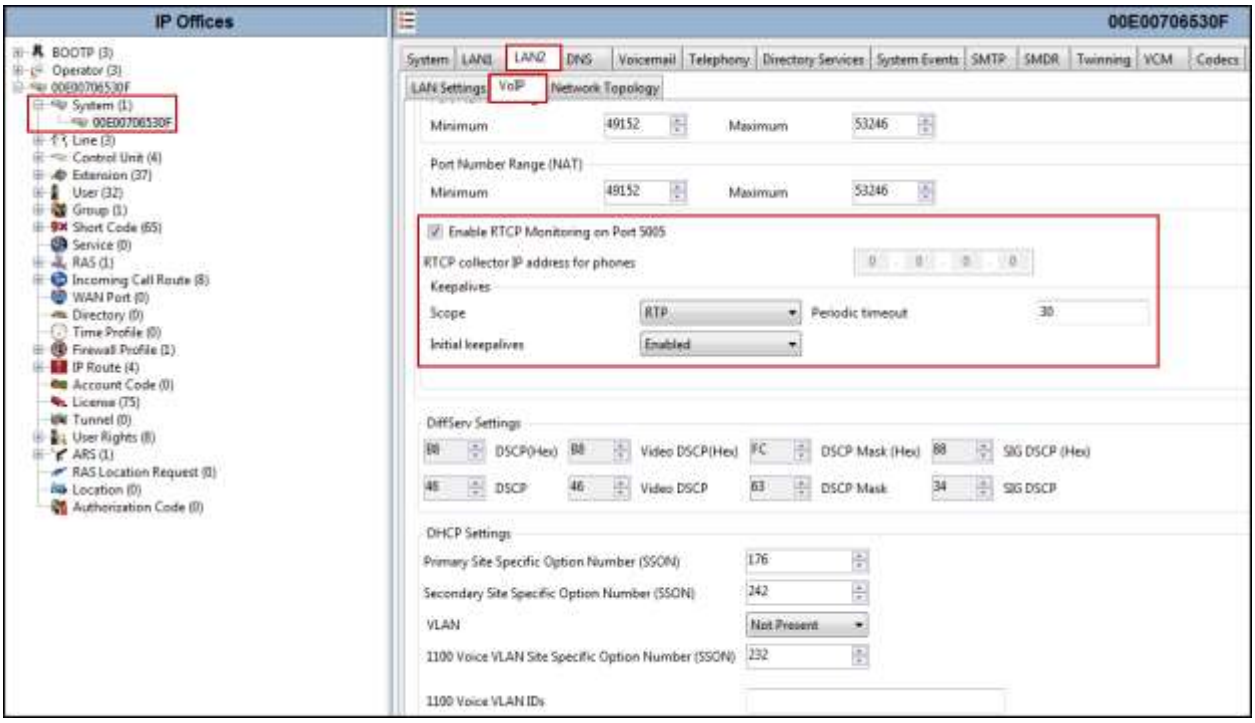
Scroll down the page.

- In the **RTP Keepalives** section, set the **Scope** to **RTP**. Set the **Periodic timeout** to **30** and the **Initial keepalives** parameter to **Enabled**. These settings will cause IP Office to send a RTP keepalive packet starting at the time of initial connection and every 30 seconds thereafter if no other RTP traffic is present. This facilitates the flow of media in cases where each end of the connection is waiting to see media from the other, as well as helping to keep firewall ports open for the duration of the call.
- In the **DiffServ Settings** section, 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 and are also the default values. For a customer installation, if the default values are not sufficient, appropriate values will be provided by the customer.
- All other parameters should be set to default or according to customer requirements.
- Click **OK** to commit (not shown).



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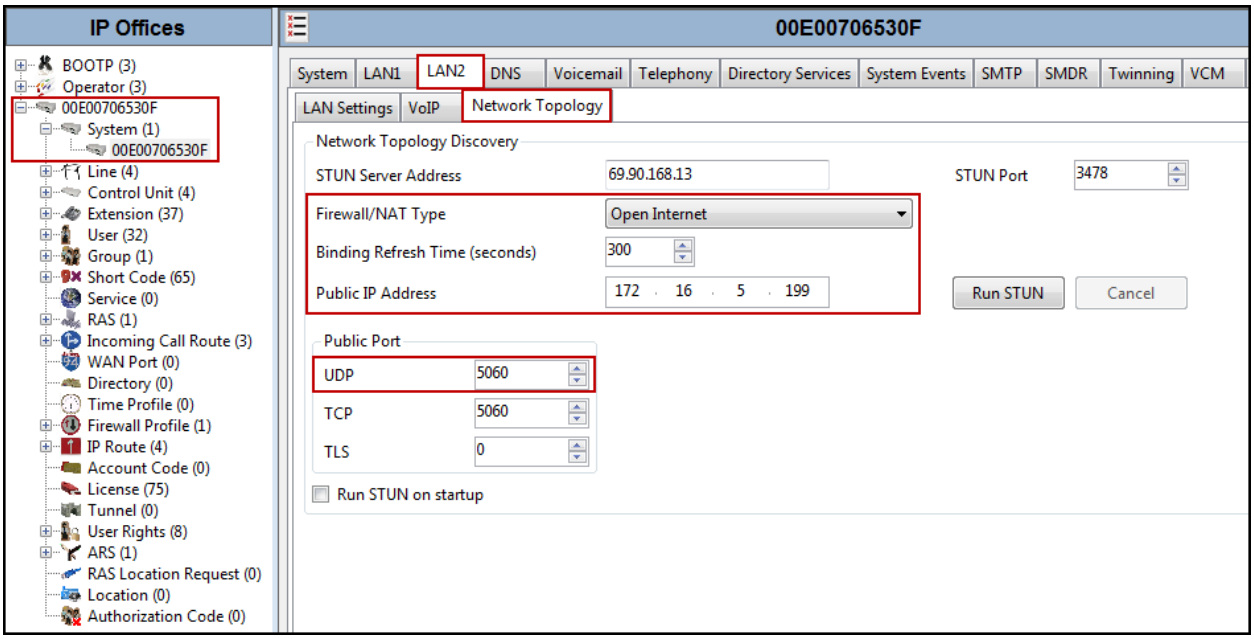
On the **Network Topology** tab in the **Details** pane, configure the following parameters:

- Select the **Firewall/NAT Type** from the pull-down menu that matches the network configuration. Since no firewall or network address translation (NAT) device was used between IP Office and Claro, the parameter was set to **Open Internet**.
- Set the **Binding Refresh Time (seconds)** to a desired value, the value of **300 (or every 5 minutes)** was used during the compliance testing. This value is used to determine the frequency that IP Office will send OPTIONS heartbeat to the service provider.
- Set **Public IP Address** to the IP address of the IP Office WAN port (**LAN2**) assigned above.
- In the **Public Port** section, next to the transport protocol **UDP**, select the UDP port on which IP Office will listen.
- All other parameters should be set to default or according to customer requirements.
- Click **OK** to commit (not shown).



Note: In the compliance test, the LAN1 interface was used to connect IP Office to the enterprise site IP network (private network). The LAN1 interface configuration is not directly relevant to the interface with the Claro SIP Trunking Service, and therefore is not described in these Application Notes.

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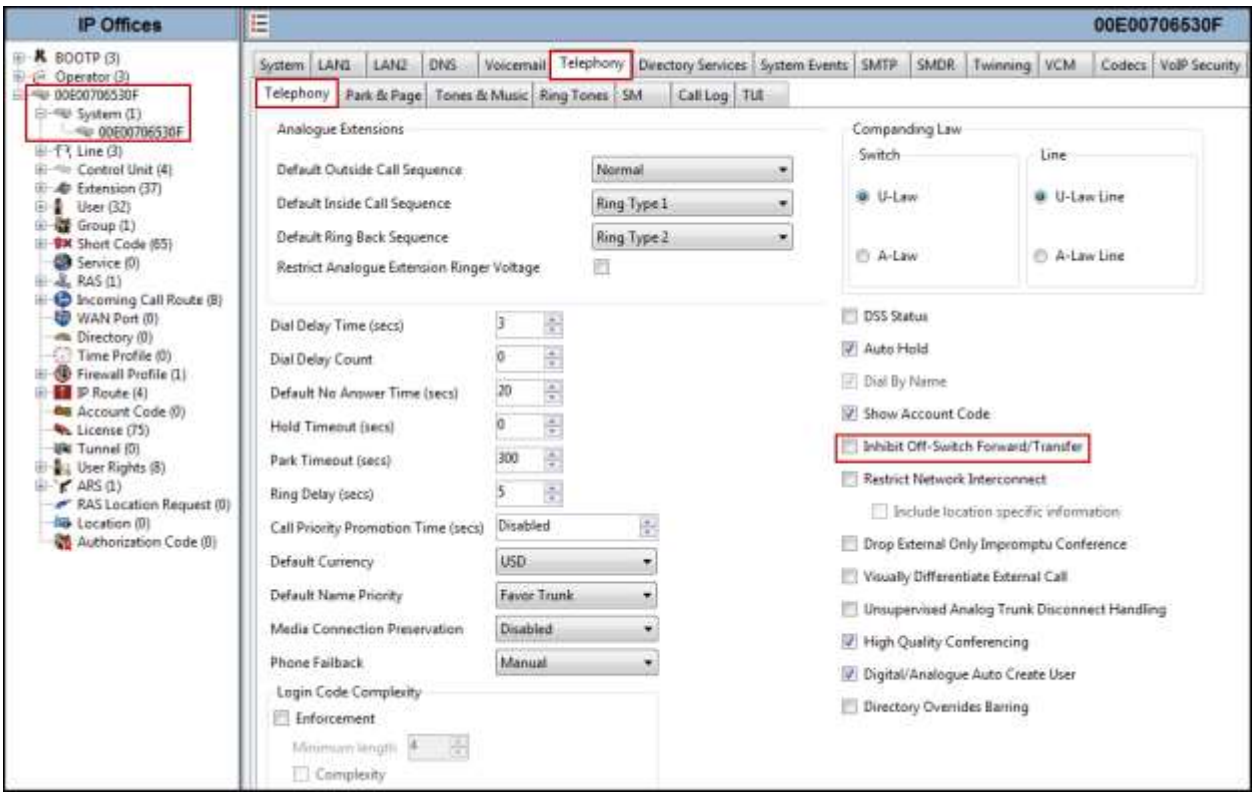
5.2.2. System - Telephony Tab

To access the 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 transfer to the PSTN. If for security reasons, incoming calls should not be allowed to transfer back to the PSTN, then leave this setting checked. All other parameters should be set to default or according to customer requirements. Click **OK** to commit (not shown).



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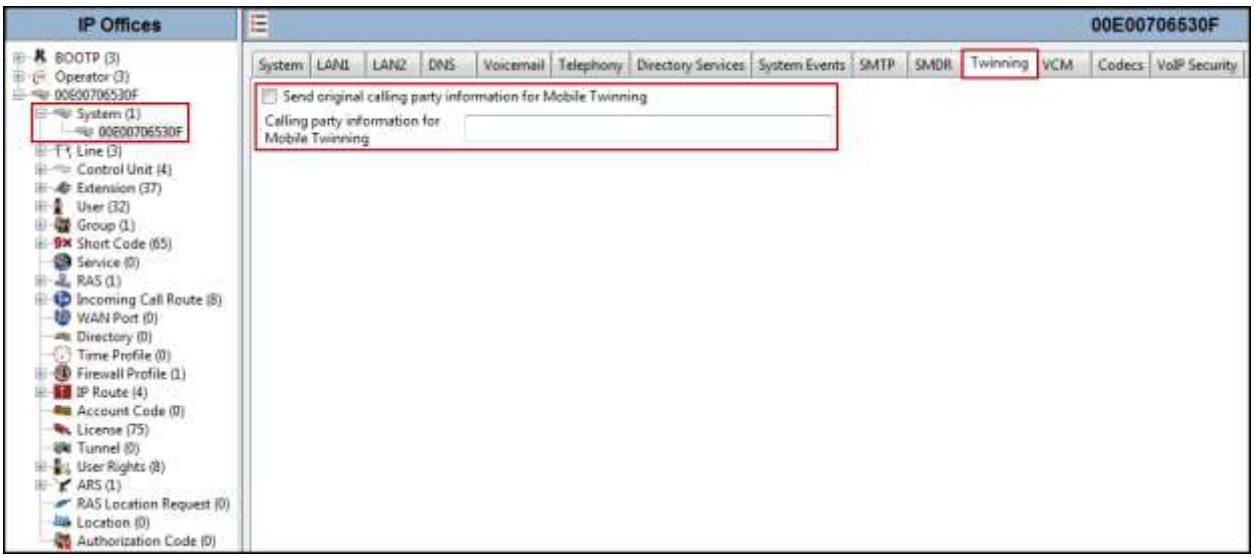
5.2.3. System - Twinning Tab

To view or change the System Twinning settings, navigate to the **Twinning** tab in the **Details** pane as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning** is left blank. Click **OK** to commit (not shown).



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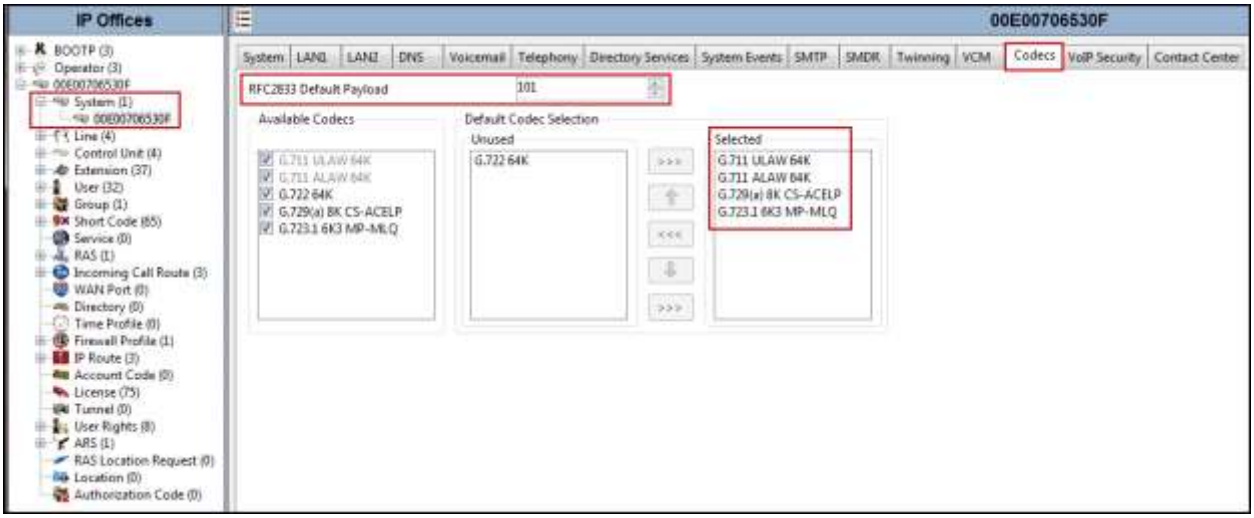
5.2.4. System – Codecs Tab

For **Codecs** settings, navigate to the **System (1) → 00E00706530F** in the **Navigation** pane, select the **Codecs** tab and configure the following parameters:

- In the **Codecs** tab of the **Details** pane, select or enter **101** for **RFC2833 Default Payload**. This setting was recommended by Claro for use with out-band DTMF tone transmissions.
- For codec selection, select the codecs and codec order of preference on the right, under the **Selected** column. 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 phones (SIP and H.323) will use the system default codec selection shown here, unless configured otherwise for a specific extension. The example below shows the codecs used for IP phones (SIP and H.323), the system's default codecs and order was used.



Note: The codec selections defined under this section (System – Codecs Tab) are the codecs selected for the IP phones/extensions. The codec selections defined under **Section 5.4.6** (SIP Line – VoIP tab) are the codecs selected for the SIP Line (Trunk).

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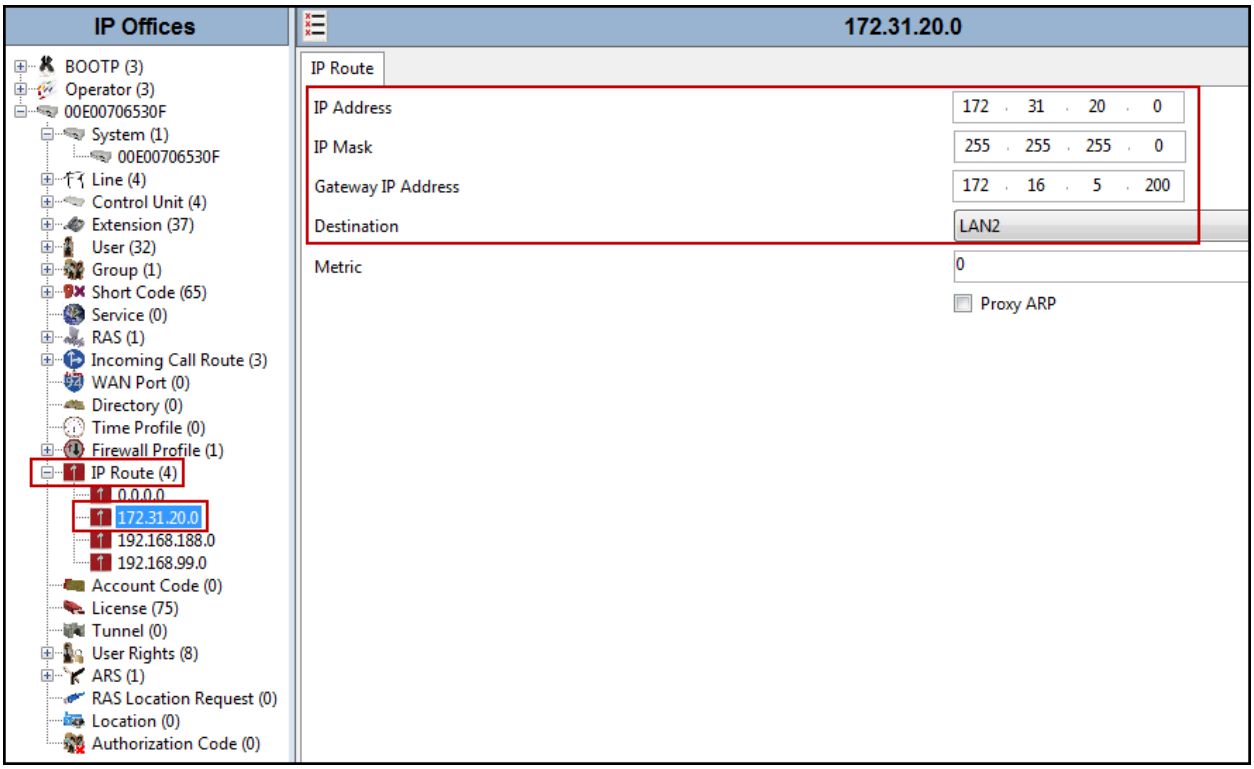
5.3. IP Route

Create an IP route to specify the IP address of the gateway or router where IP Office needs to send the packets in order to route calls to Claro's network (if located in a different subnet).

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

- Set **IP Address** to **172.31.20.0** (IP subnet assigned to Claro's SIP Proxy).
- Set the **IP Mask** to **255.255.255.0** (IP Mask assigned to Claro's SIP Proxy).
- Set **Gateway IP Address** to the IP address of the default router for the public network where IP Office is connected. Gateway IP Address **172.16.5.200** was used in the sample configuration, which correspond to the enterprise LAN side (private IP network) IP address assigned to the VPN firewall. Refer to **Figure 1**.
- Set **Destination** to **LAN2** from the drop-down list.
- Click the **OK** to commit (not shown).



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5.4. SIP Line

A SIP Line is needed to establish the SIP connection between IP Office and Claro SIP Trunking Service. 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 Avaya IP Office Manager to create a SIP Line. Follow the steps in **Sections 5.4.1** and **5.4.2** 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 trunk Registration Credentials.
- 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 5.4.3** to **5.4.7.**

Alternatively, a SIP Line can be created manually. To do so, right-click on **Line** in the **Navigation** pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.4.3** to **5.4.7**

5.4.1. 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., C:*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 5.4.2**, **Step 2**, to create a new SIP Line.

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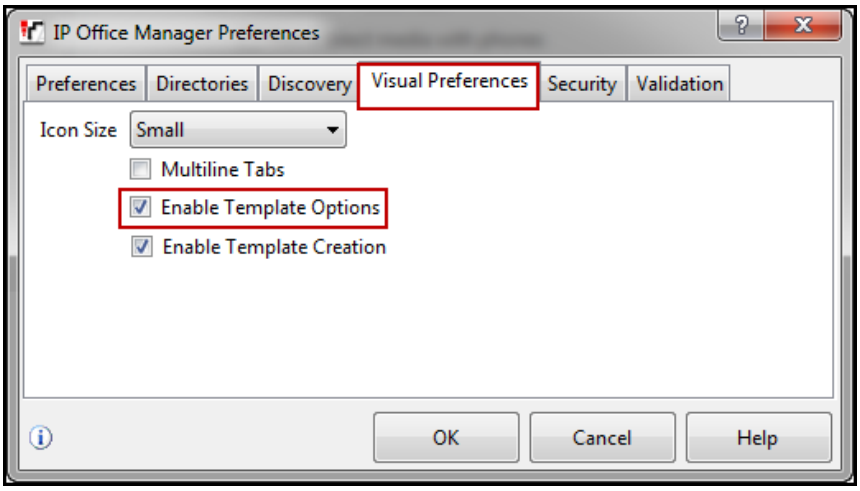
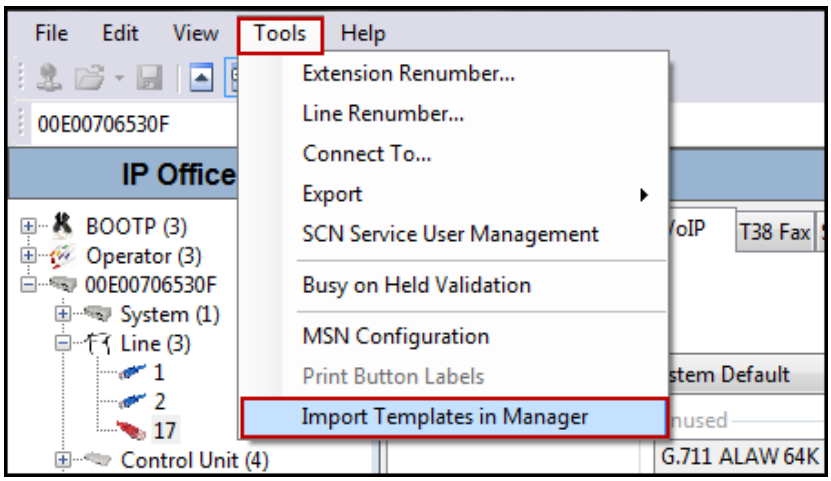
2. Verify that Template Options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the **IP Office Manager Preferences** window that appears, select the **Visual Preferences** tab. Check the box next to **Enable Template Options**. Click **OK**.



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



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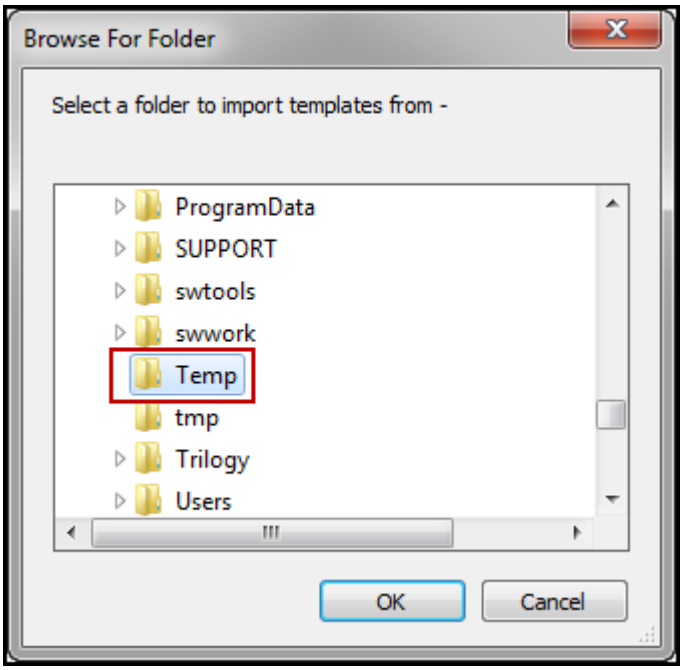
4. A folder browser will open. Select the directory used in **Step 1** to store the template(s) (e.g., C:*Temp*).



In the reference configuration, template files **AF_Claro Dominican Republic_ SIPTrunk.xml** was imported. The template files are automatically copied into the IP Office default template location, **C:\Program Files\Avaya\IP Office\Manager\Templates**.

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

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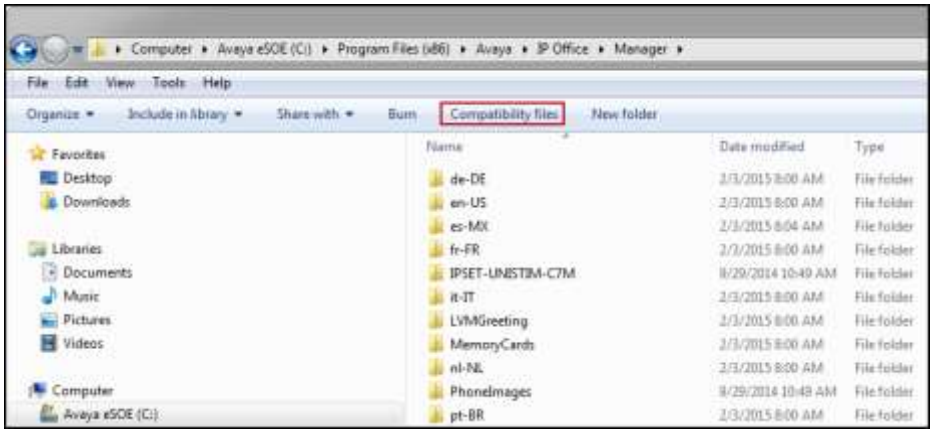
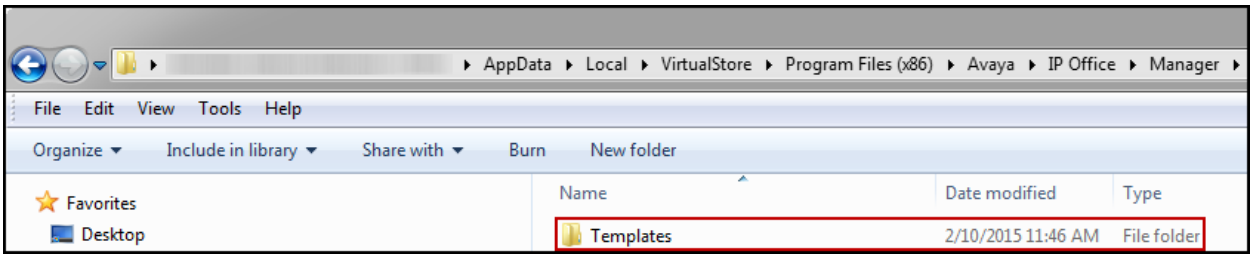
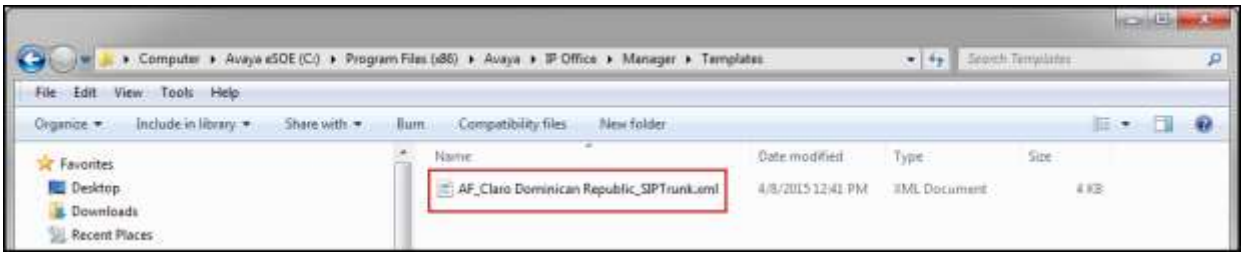
Note –Windows 7 (and later) locks the Avaya IP Office 9.1 **Templates** directory, and it cannot be viewed. To enable browsing of the **Templates** directory, open Windows Explorer, navigate to **C:\Program Files\Avaya\IP Office\Manager\Templates** (or C:*Program Files (x86)*\Avaya\IP Office\Manager\Templates),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







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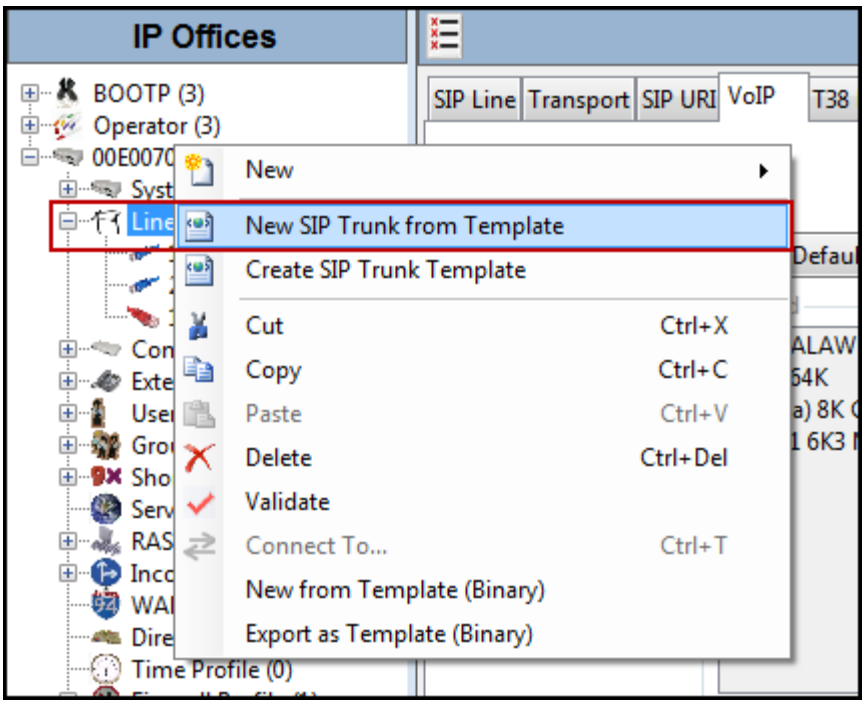
5.4.2. Creating a SIP Trunk from an XML Template

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 5.4.1**. Click **Create new SIP Trunk**.

Note – The drop down menu will display the ***<user supplied text>*** part of the template file name (see **Section 5.4.1**). If the **Display All** box is checked, then the full template file name is displayed.

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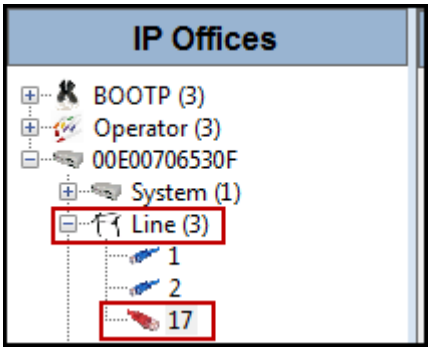
The newly created SIP Line will appear in the **Navigation** pane (e.g., SIP Line **17**).



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 5.4.3** to **5.4.7**.

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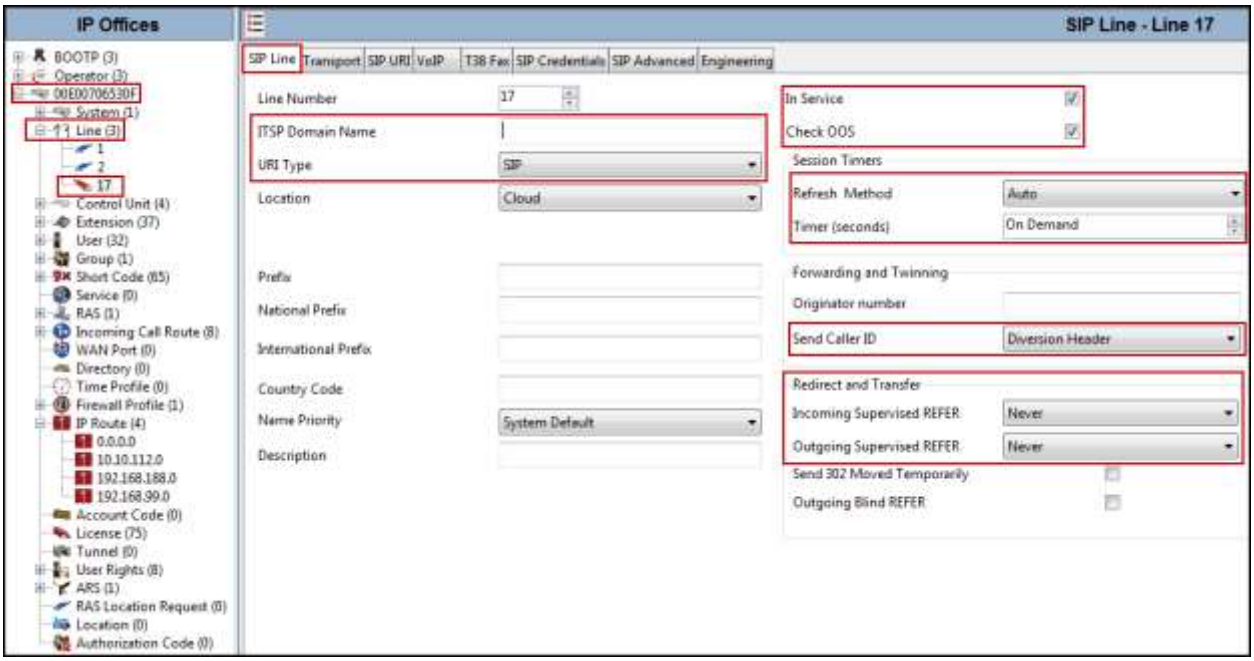
5.4.3. SIP Line – SIP Line Tab

On the **SIP Line** tab in the **Details** pane, configure or verify the parameters as shown below.

- Leave the **ITSP Domain Name** blank. Note that if this field is left blank, IP Office inserts the far end's ITSP Proxy Address from the Transport tab as the ITSP Domain in the SIP messaging.
- Verify that **URI Type** is set to **SIP**.
- Verify that **In Service** box is checked, which is the default value. This makes the trunk available to incoming and outgoing calls.
- Verify that **Check OOS** box is checked, the default value. IP Office will use the SIP OPTIONS method to periodically check the SIP Line. The time between SIP OPTIONS sent by IP Office will use the Binding Refresh Time for LAN2, as shown in **Section 5.2.1**.
- Verify that **Refresh Method** is set to **Auto**.
- Verify that **Timer (seconds)** is set to **On Demand**.
- Set **Send Caller ID** to **Diversion Header**.
- Under **Redirect and Transfer**, set **Incoming Supervised REFER** and **Outgoing Supervised REFER** to **Never** (Refer to **Sections 2.1** and **2.2**).
- All other parameters should be set to default or according to customer requirements.
- Click **OK** to commit (not shown).



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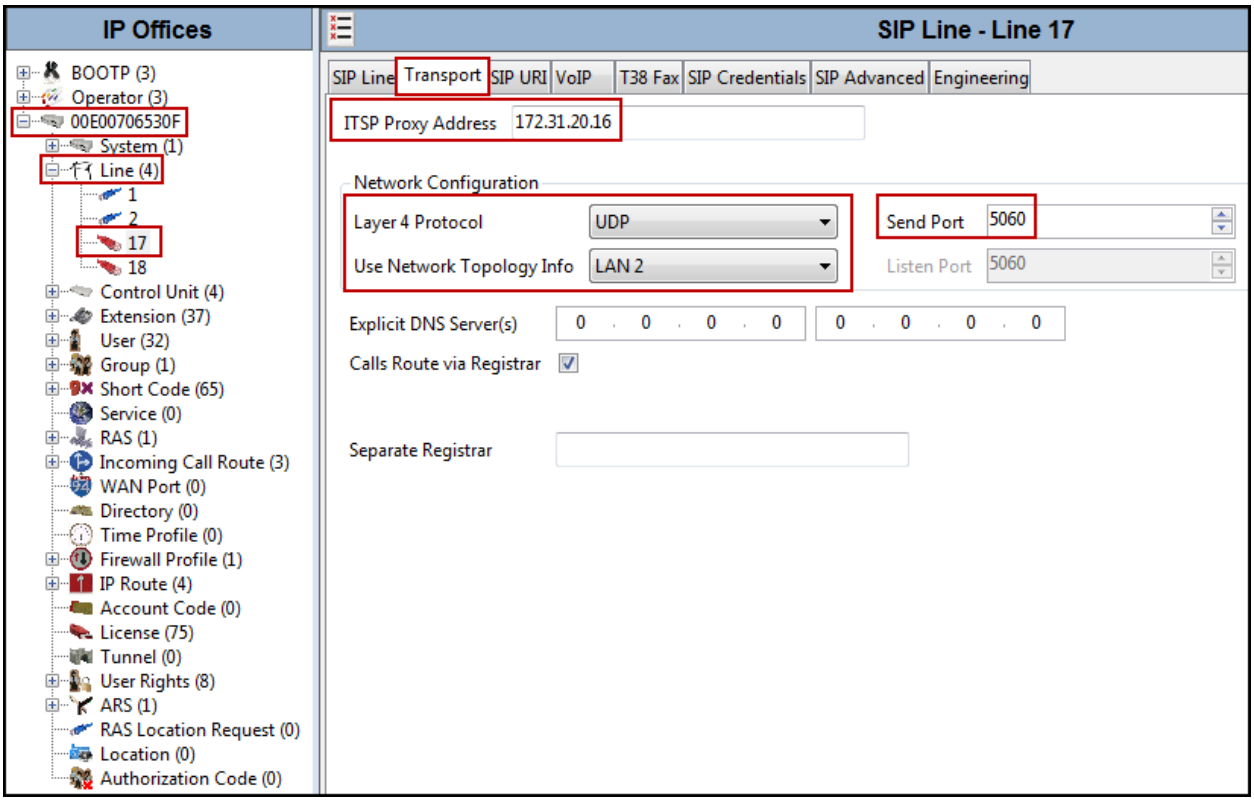
5.4.4. SIP Line - Transport Tab

Select the **Transport** tab. Set or verify the parameters as shown below.

- Set **ITSP Proxy Address** to **172.31.20.16**, the IP address of Claro's SIP Proxy.
- Set **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to **LAN2**. The LAN2 settings correspond to the WAN port on the Avaya IP Office 500v2, used by the SIP Line to access the far-end, configured in **Section 5.2.1**.
- Set the **Send Port** to **5060**.
- Default values may be used for all other parameters.
- Click **OK** to commit (not shown).



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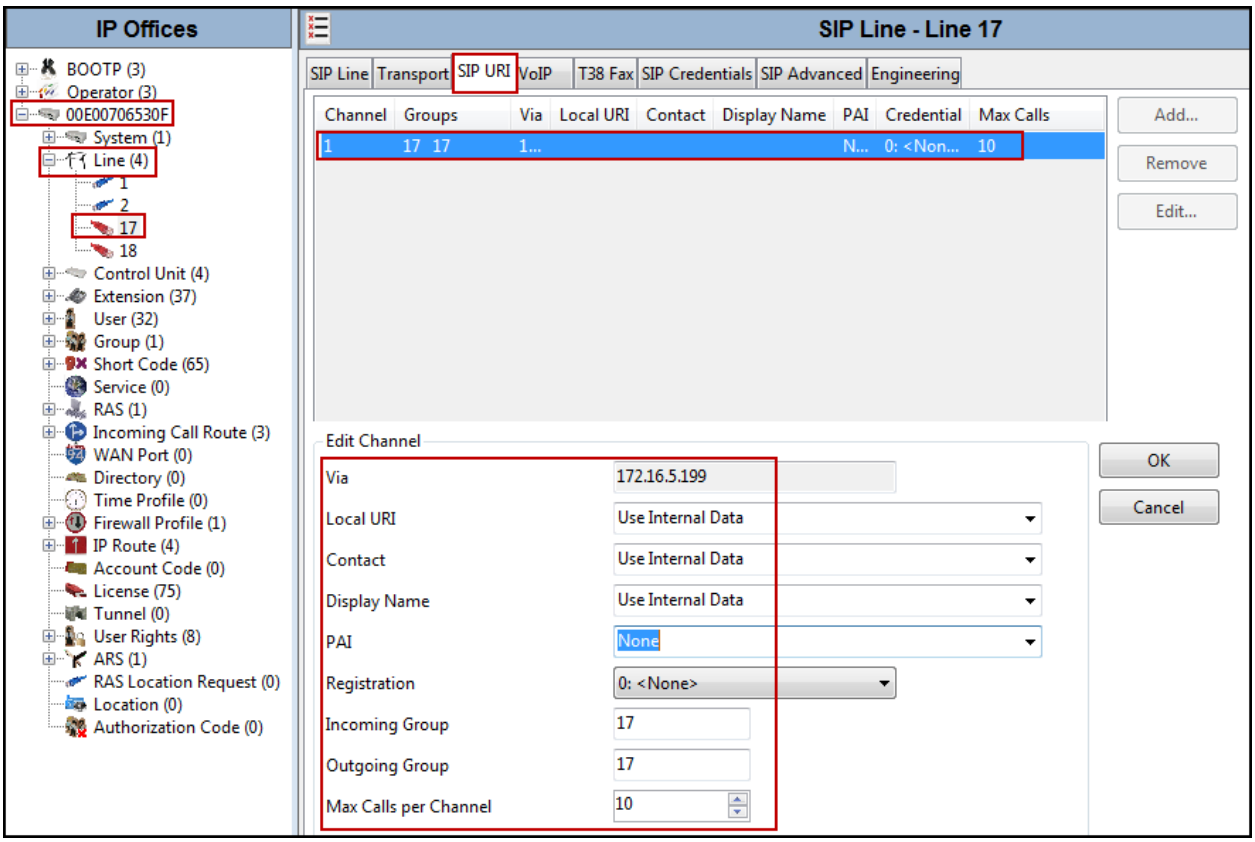
5.4.5. SIP Line - SIP URI Tab

A SIP URI entry needs to be created to match each incoming number that IP Office will accept on this line. Select the **SIP URI** tab, and then click the **Add…** button and the **New Channel** area will appear at the bottom of the pane. To edit an existing entry, click an entry in the list at the top, and click the **Edit…** button. In the example screen below, a previously configured entry was edited. For the compliance test, a single SIP URI entry was created that matched any DID number assigned to an IP Office user. The entry was created with the parameters shown below:

- Set **Local URI**, **Contact**, **Display Name** to **Use Internal Data**.
- Set **PAI** to **None**.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group **17** was defined that only contains this line (line 17).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern. This field should be set according to customer requirements; in the sample configuration the value of **10** was used.
- Click **OK** to commit.
- Click **OK** to commit again (not shown).



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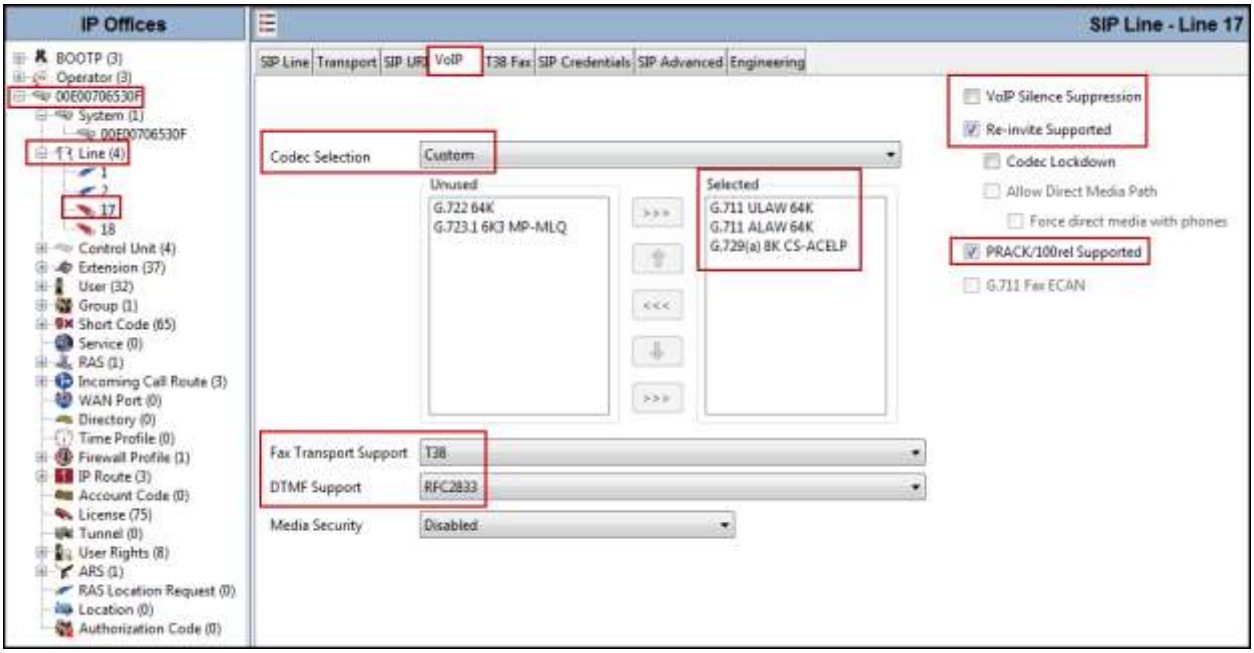
5.4.6. SIP Line - VoIP Tab

Select the **VoIP** tab, to set the Voice over Internet Protocol parameters of the SIP Line. Set or verify the parameters as shown below.

- In the sample configuration, the **Codec Selection** was configured using the **Custom** option, allowing an explicit order of codecs to be specified for the SIP Line. The buttons allow setting the specific order of preference for the codecs to be used on the SIP Line, as shown. Codec's G.711ULAW, G.711ALAW and G.729(a) were selected for audio.
- Select **T.38** for **Fax Transport Support** (select **G.711** instead of **T.38** if G.711 fax pass-through is to be used, instead of T.38 fax).
- Set the **DTMF Support** field to **RFC2833**. This directs 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 reliable provisional responses and Early Media to Claro.
- Default values may be used for all other parameters.
- Click OK to commit (not shown).



Note: The codec selections defined under this section (SIP Line – VoIP tab) are the codecs selected for the SIP Line (Trunk). The codec selections defined under **Section 5.2.4** (System – Codec tab) are the codecs selected for the IP phones/extension (H.323 and SIP).

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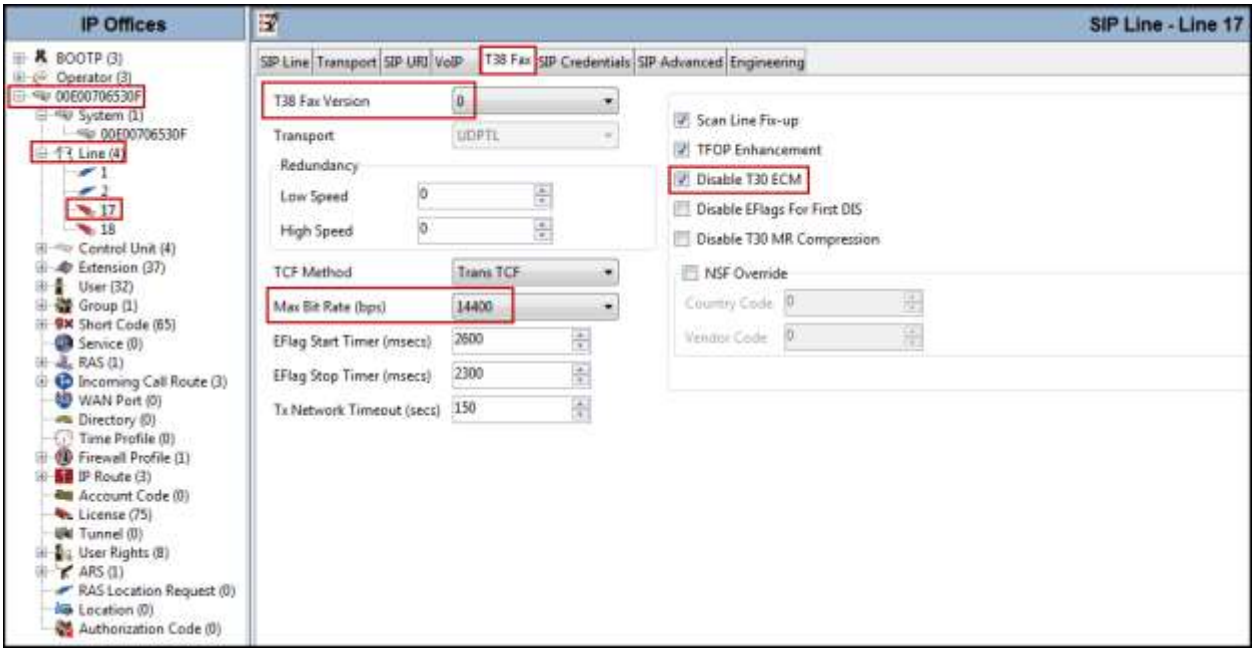
Select the **T38 Fax** tab to set the Fax over Internet Protocol parameters of the SIP line. Set the parameters as shown below (note that this tab will be disabled if **G.711 Fax Transport Support** is selected above).

- Uncheck **Use Default Values** at the bottom of the screen (not shown).
- Set **T38 Fax Version** to **0**. Claro SIP Trunking supports T.38 fax version 0.
- Set **Max Bit Rate (bps)** to **14400**, the highest fax bit rate that Avaya IP Office supports for T.38 faxing.
- Check the **Disable T30 ECM** option.
- Default values may be used for all other parameters.
- Click **OK** to commit (not shown).



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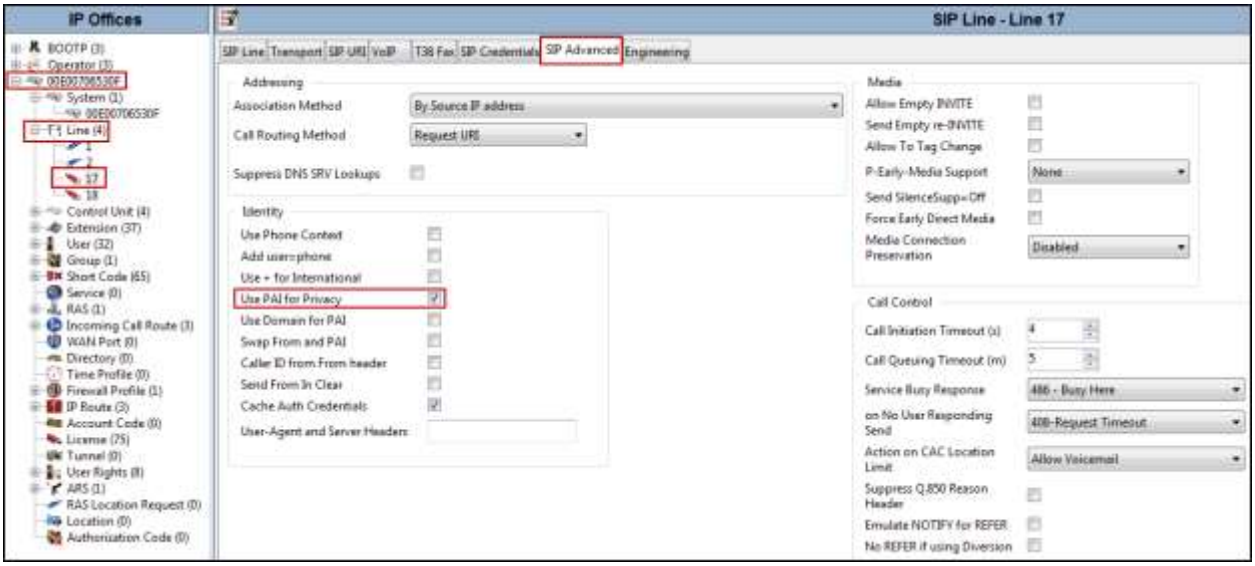
5.4.7. SIP Line – SIP Advanced Tab

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

- Check the box for **Use PAI for Privacy.**
- Default values may be used for all other parameters.
- Click **OK** to commit (not shown).



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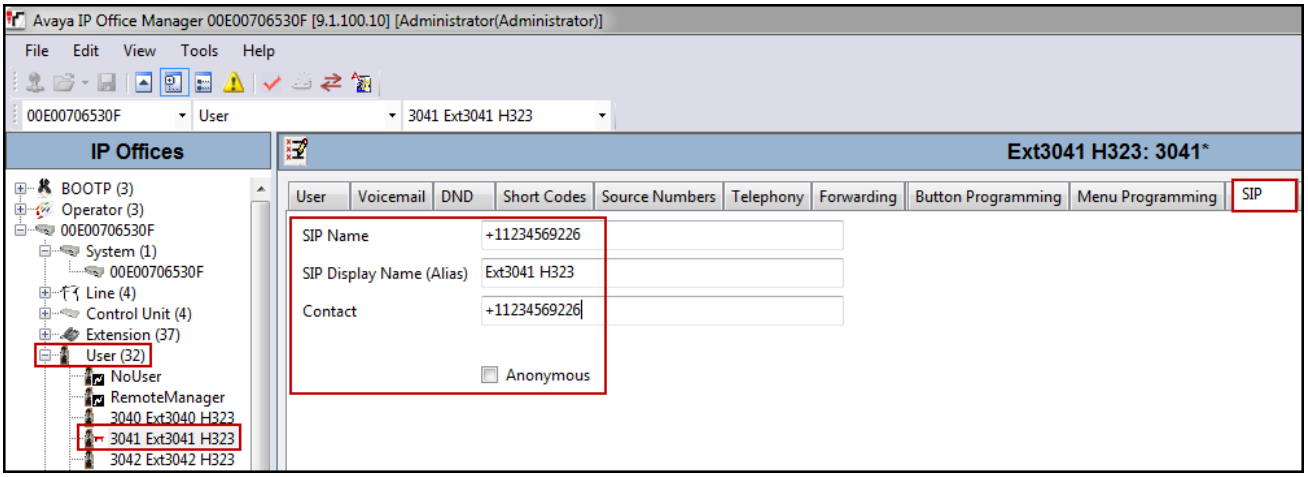
5.5. Users

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP Line defined in **Section 5.4**. To configure these settings, first navigate to **User** → *Name* in the **Navigation** pane where *Name* is the name of the user to be modified. In the example below, the name of the user is **Ext3041 H323**. Select the **SIP** tab in the **Details** pane. The values entered for the **SIP Name** allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP Line (see **Section 5.4.5**). The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise by Claro. Note that a "+" sign was added at the beginning of the DID number for each user under **SIP Name** and **Contact**. Claro supports the E.164 numbering format, Claro's network adds a "+" sign to the numbers it includes in SIP messages it sends to the enterprise, IP Office also adds the "+" sign to the numbers in SIP messages it sends to Claro. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network. This can also be accomplished by activating Withhold Number on H.323 Deskphones (not shown). Click the **OK** to commit (not shown).



5.6. Outbound Call Routing

For outbound call routing, a combination of system short codes and Automatic Route Selection (ARS) entries are used. With ARS, features like time-based routing criteria and alternate routing can be specified so that a call can re-route automatically if the primary route or outgoing line group is not available. While detailed coverage of ARS is beyond the scope of these Application Notes, and alternate routing was not used in the reference configuration, this section includes some basic screen illustrations of the ARS settings used during the compliance testing.

5.6.1. Short Codes and Automatic Route Selection

To create a short code to be used for ARS, right-click on **Short Code** on the **Navigation** pane and select **New**. The screen below shows the short code **9N** created (note that the semi-colon is not used here). In this case, when the IP Office user dials 9 plus any number **N**, instead of being directed to a specific Line Group ID, the call is directed to **Line Group 50: Main**, which is configurable via ARS.

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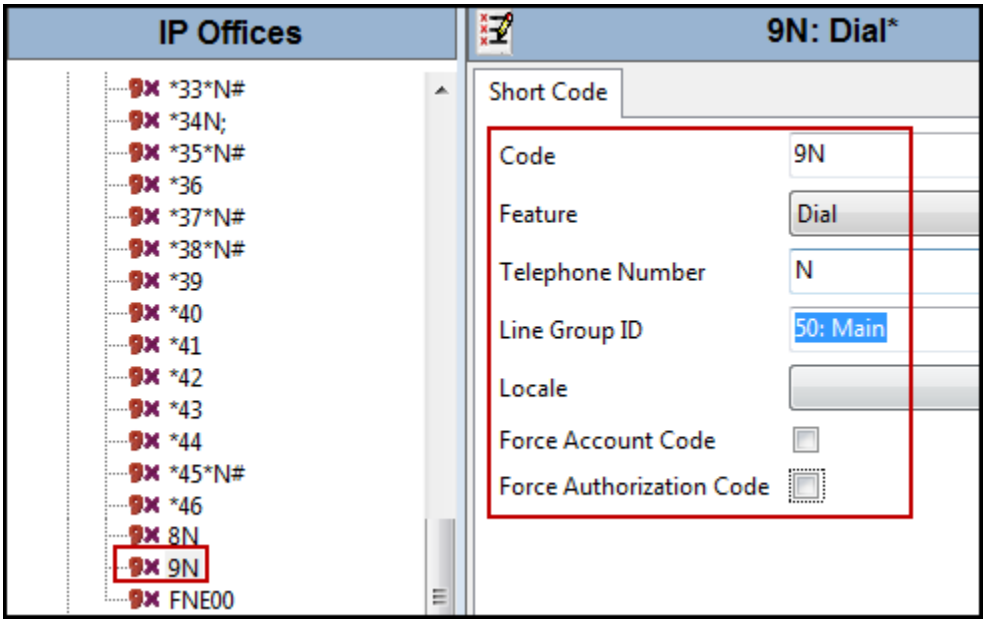
- In the **Code** field, enter the dial string which will trigger this short code. In this case, **9N** was used (note that the semi-colon is not used here).
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N**. The value **N** represents the number dialed by the user after removing the **9** prefix. This value is passed to ARS.
- Set the **Line Group ID** to **50: Main** to be directed to **Line Group 50: Main**, which is configurable via ARS.
- Click the **OK** to commit (not shown).



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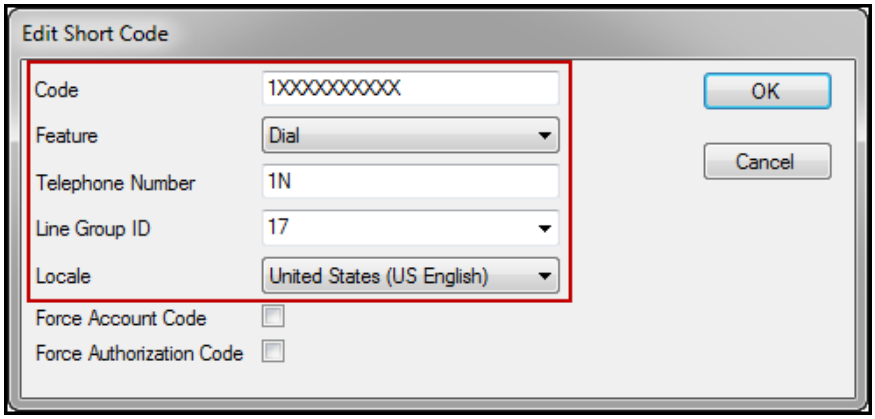
The following screen shows a sample ARS configuration for the route **Main**. Note the sequence of **X**'s used in the **Code** column of the entries to specify the exact number of digits to be expected, following the access code and the first set of digits on the string. This type of setting results in a much quicker response in the delivery of the call by IP Office.

To create a short code to be used for ARS, select **ARS → 50: Main** on the **Navigation** pane and click **Add**.

- In the **Code** field, enter the dial string which will trigger this short code. In this case, **1** followed by **10 X**'s to represent the exact number of digits.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **1N**. The value **N** represents the additional number of digits dialed by the user after dialing **1** (The **9** will be stripped off).
- Set the **Line Group ID** to the Line Group number being used for the SIP Line, in this case Line **Group ID 17** was used.
- Set **Locale** to **United States (US English)**.
- Click **OK** to commit.



Repeat the above procedure for additional dial patterns to be used by the enterprise to dial out from IP Office.

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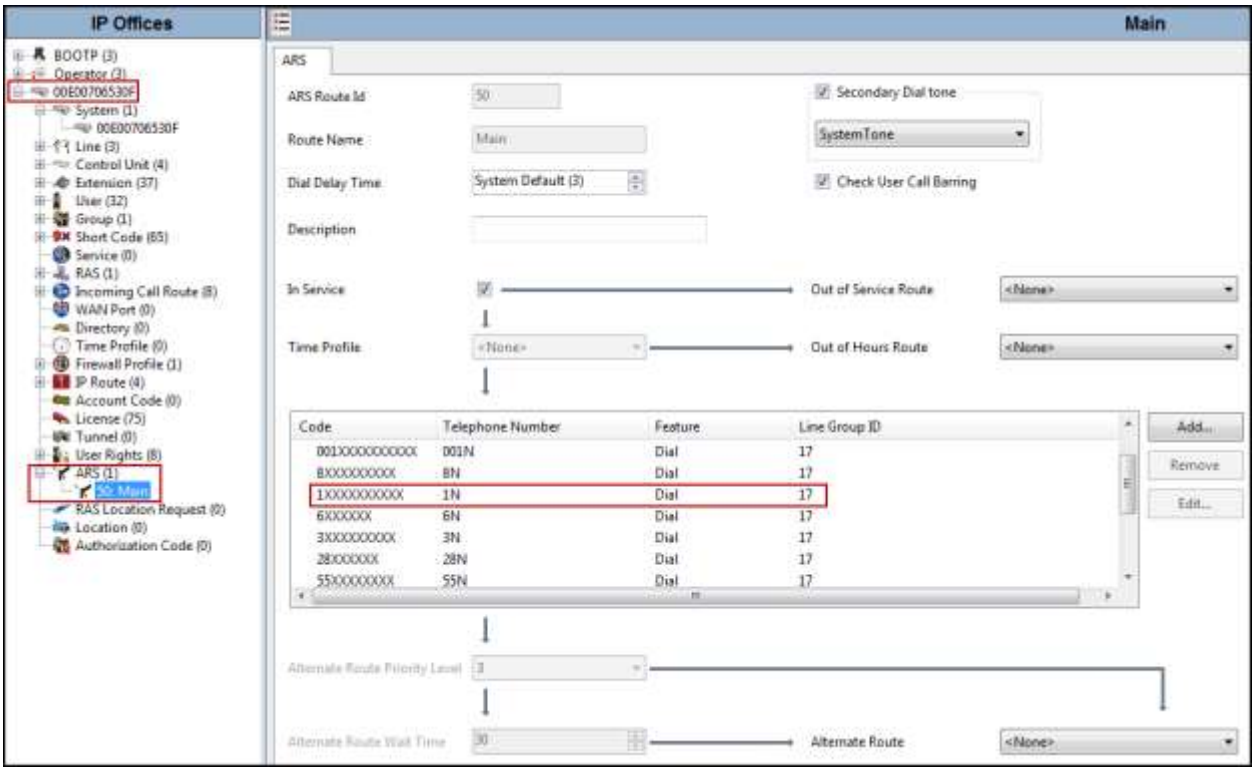
The example highlighted below shows that for calls in the North American numbering plan, the user dialed **9**, followed by **1** and **10** digits (represented by **10 X**'s). The **9** is stripped off, the remaining digits, including the **1**, are included in the SIP INVITE message IP Office sends to Claro.



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5.7. Incoming Call Route

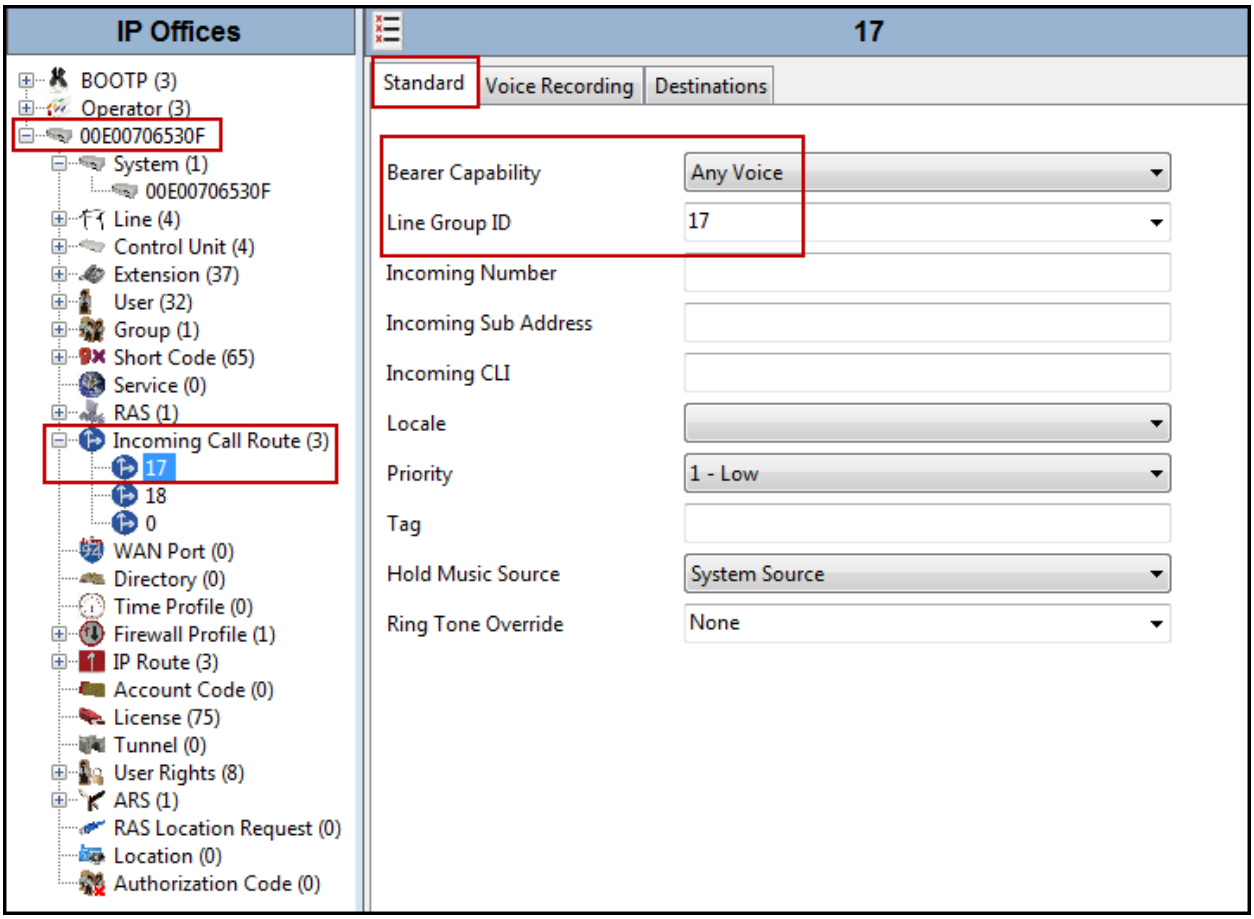
An incoming call route maps inbound DID numbers on a specific line to internal extensions, hunt groups, short codes, etc., within the IP Office system. In a scenario like the one used for the compliance test, only one incoming route is needed, which allows any incoming number arriving on the SIP trunk to reach any predefined extension in IP Office. The routing decision for the call is based on the parameters previously configured for **Call Routing Method** and **SIP URI (Section 5.4.5)** and the users **SIP Name** and **Contact**, already populated with the assigned Claro DID numbers **(Section 5.5)**.

5.7.1. Incoming Call Route – Standard Tab

To create an incoming call route, right-click **Incoming Call Routes** in the **Navigation** pane and select **New**.

On the **Standard** tab of the **Details** pane, enter the parameters as shown below.

- Set the **Bearer Capacity** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.4**.
- Default values can be used for all other fields.



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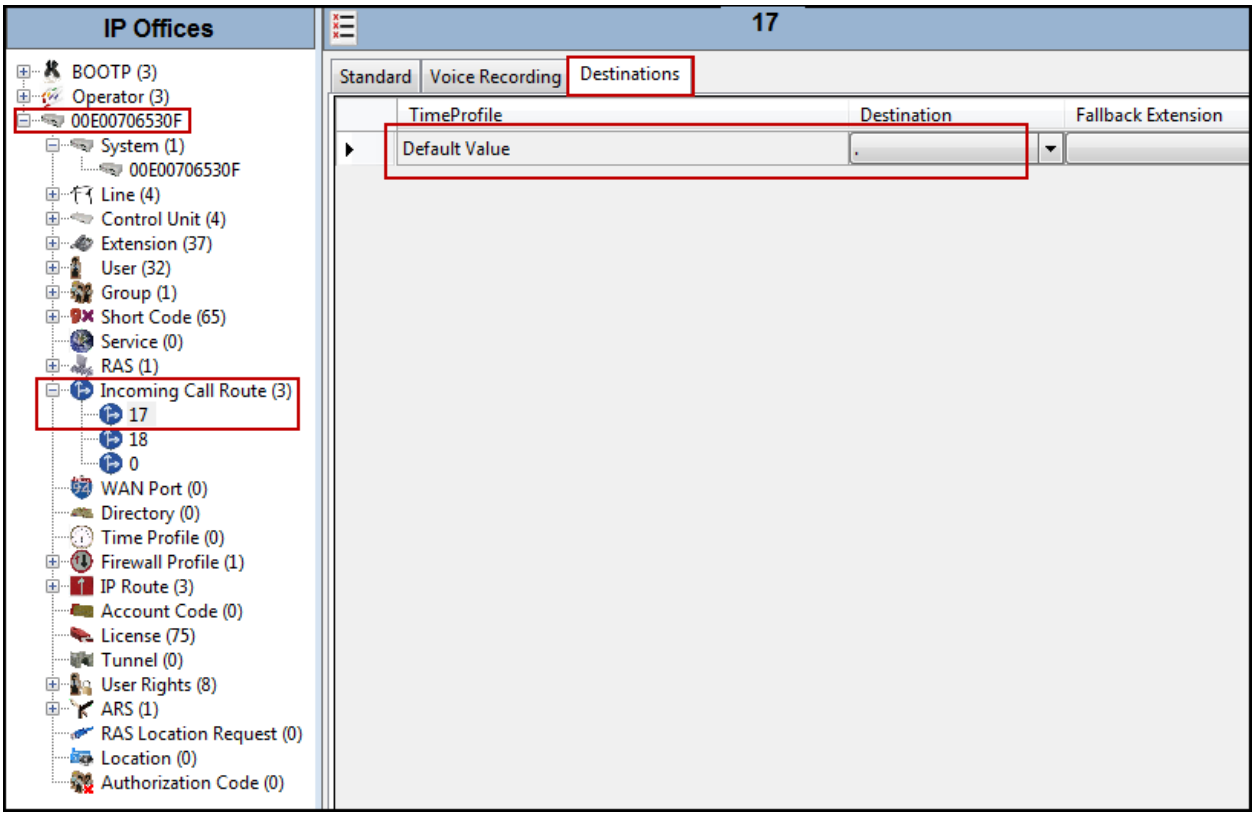
5.7.2. Incoming Call Route – Destinations Tab

- Under the **Destinations** tab, enter "**.**" for the **Default Value**. This setting will allow the call to be routed to any destination with a value on its **SIP Name** field, entered on the **SIP** tab of that **User**, which matches the number present on the user part of the incoming Request URI.
- Click **OK** to commit (not shown)



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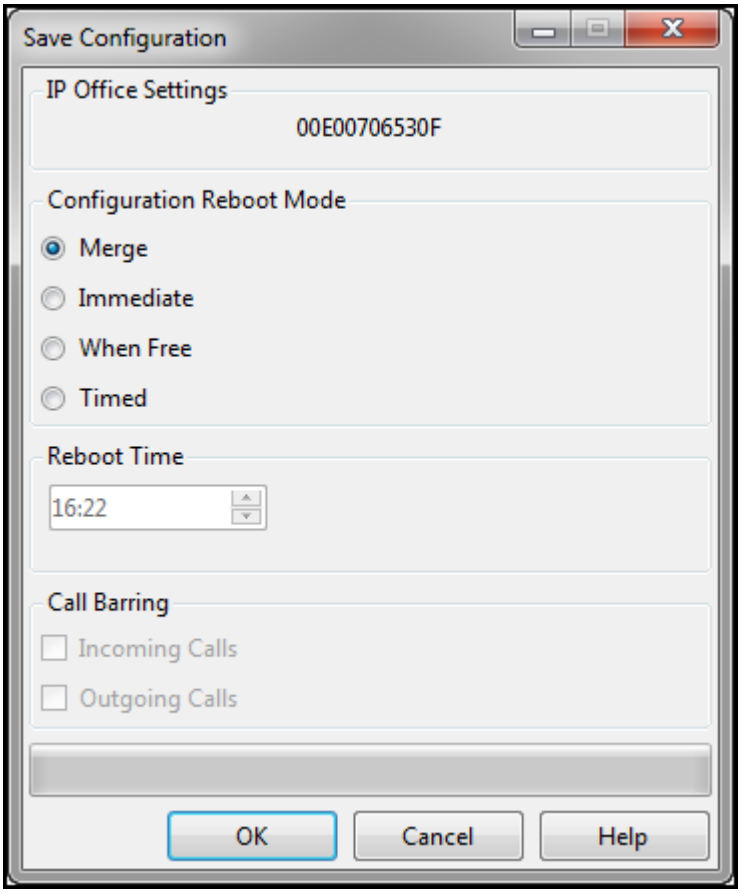
5.8. Save Configuration

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

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



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6. Claro SIP Trunking Service Configuration

To use Claro's SIP Trunking Services, a customer must request the service from Claro using the established sales processes. The process can be started by contacting Claro via the corporate web site at: http://www.claro.com.do/wps/portal/do/sc/empresas and requesting information.

During the signup process, Claro and the customer will discuss details about the preferred method to be used to connect the customer's enterprise network to Claro's network. Claro will provide IP addresses, Direct Inward Dialed (DID) numbers to be assigned to the enterprise, etc. This information is used to complete the Avaya IP Office configuration discussed in the previous sections.

Claro is responsible for the configuration of Claro SIP Trunking Service. The customer will need to provide the public IP address used to reach IP Office at the enterprise.

During the interoperability testing, a VPN connection was used to connect the simulated enterprise site to Claro's network via the public Internet. The connection could also be done without the use of a VPN connection.

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7. Verification Steps

This section provides verification steps that may be performed to verify that the solution is configured properly.

The following steps may be used to verify the configuration:

- Verify that endpoints at the enterprise site can place calls to the PSTN.
- Verify that endpoints at the enterprise site can receive calls from the PSTN.
- Verify that users at the PSTN can end active calls to endpoints at the enterprise by hanging up.
- Verify that endpoints at the enterprise can end active calls to PSTN users by hanging up.

7.1. Avaya IP Office System Status

The following steps can also be used to verify the configuration.

Use the **Avaya IP Office System Status** application to verify the state of SIP connections. Launch the application from **Start → Programs → IP Office → System Status** on the PC where Avaya IP Office System Status is installed, log in with the proper credentials.

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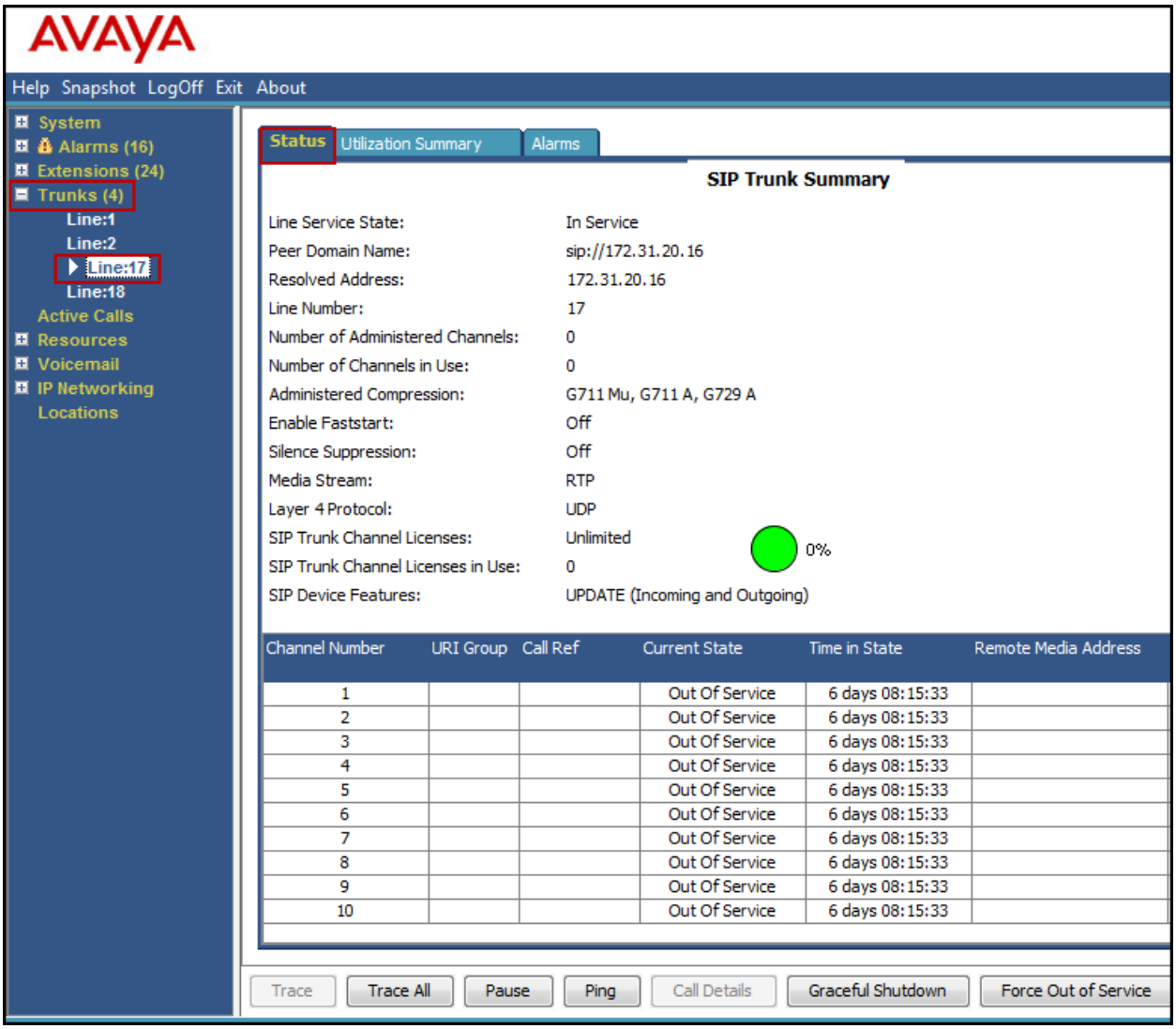
Select the SIP Line under **Trunks** from the left pane. On the **Status** tab in the right pane, verify the **Current State** is **Idle** for each channel. Note that the **Current State** of the trunk channels shown on the screenshot is **Out Of Service** since the link in between Claro and the enterprise was down at the time the screenshot was taken.



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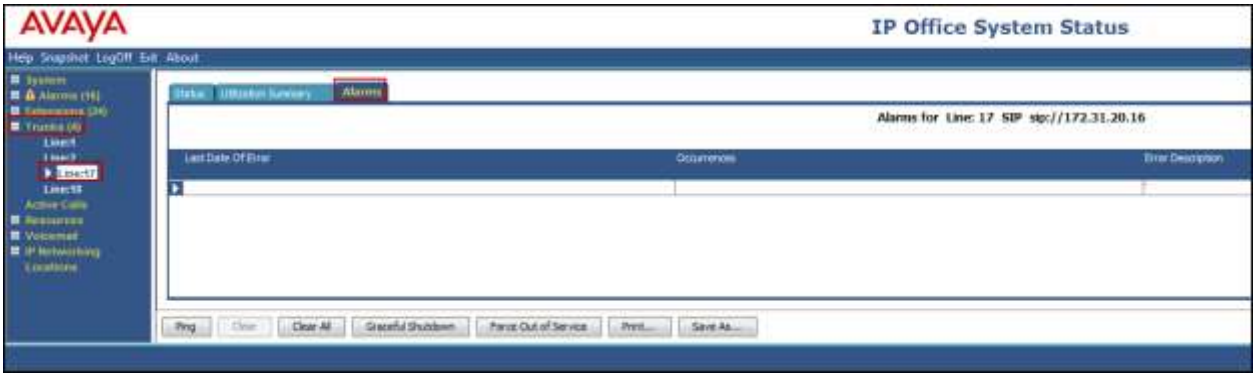
Select the **Alarms** tab and verify that no alarms are active on the SIP Line.



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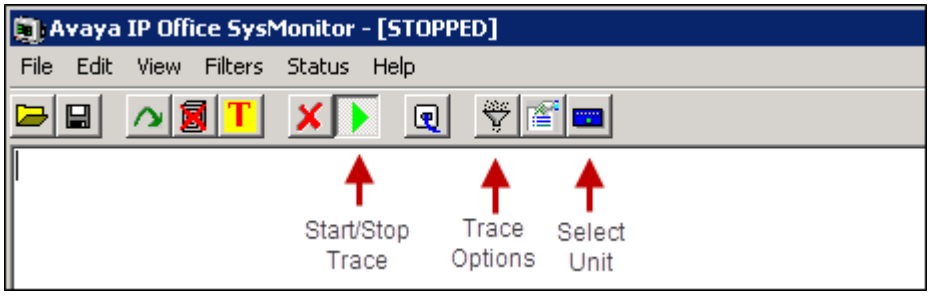
7.2. Avaya IP Office Sys Monitor

The **Avaya IP Office Sys 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 Sys Monitor was installed. Click the **Select Unit** icon on the taskbar and select the IP address of the IP Office system under verification.



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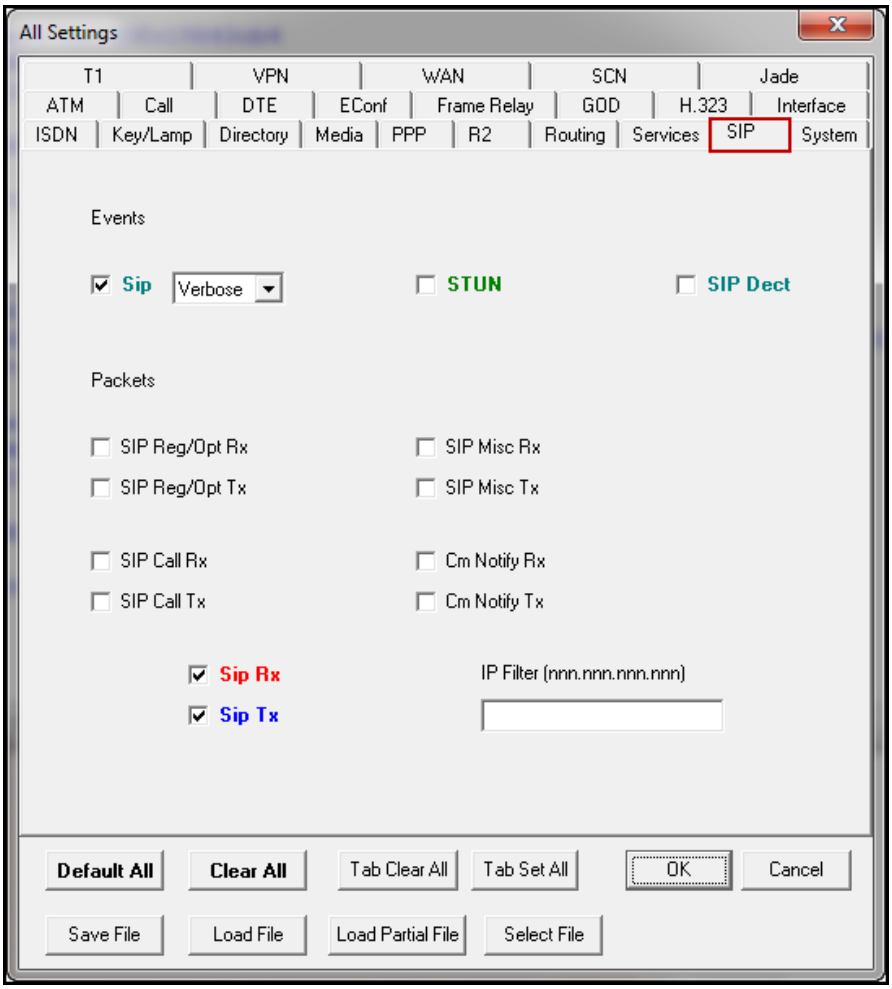
Clicking the **Trace Options** icon on the taskbar, selecting the **SIP** tab allows modifying the threshold used for capturing events, types of packets to be captured, filters, etc. Additionally, the color used to represent the packets in the trace can be customized by right clicking on the type of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



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8. Conclusion

These Application Notes describe the configuration steps necessary for configuring Session Initiation Protocol (SIP) Trunk Service for an enterprise solution consisting of Avaya IP Office Release 9.1 to interoperate with Claro SIP Trunking Service. Claro SIP Trunking Service is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. It provides a flexible, cost-saving alternative to traditional hardwired telephony trunks.

Claro SIP Trunking Service passed compliance testing with the observations/limitations outlined in the scope of testing in **Section 2.1** as well as under test results in **Section 2.2**.

9. References

This section references the documentation relevant to these Application Notes. Product documentation for Avaya IP Office, including the following, is available at:
http://support.avaya.com/

[1] *Avaya IP Office Platform Solution Description, Release 9.1*, Issue 1, December 2014.
[2] *Avaya IP Office Platform Feature Description, Release 9.1*, Issue 1, December 2014
[3] *IP Office Platform 9.1 Deploying Avaya IP Office Platform IP500 V2*, Document Number 15-601042, Issue 30g, January 2015.
[4] *Administering Avaya IP Office Platform with Manager*, Release 9.1.0, Issue 10.02, January 2015.
[5] *IP Office Platform 9.1 Using Avaya IP Office Platform System Status*, Document 15-601758, Issue 10b, October 30, 2014.
[6] *IP Office Platform 9.1 Using IP Office System Monitor*, Document 15-601019, Issue 06b, November 13, 2014.
[7] *Using Avaya Communicator for Windows on IP Office*, Release 9.1, December 2014.
[8] *Administering Avaya Communicator on IP Office, Release 9.1*, December 2014.

Additional Avaya IP Office documentation can be found at:
http://marketingtools.avaya.com/knowledgebase/

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