Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Clearfly Communications SIP Trunking Service with Avaya IP Office 9.1 with Registration - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Avaya IP Office Release 9.1 to interoperate with the Clearfly Communications SIP Trunking Service with registration. Clearfly Communications is a member of the Avaya DevConnect Service Provider program.

The Clearfly Communications SIP Trunking Service provides PSTN access via a SIP trunk between an enterprise site and the Clearfly Communications network as an alternative to legacy analog or digital trunks. This approach generally results in easier maintenance and lower cost for the business customer.

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

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

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1. Introduction

These Application Notes describe the procedures for configuring an enterprise solution using Avaya IP Office Release 9.1 to interoperate with the Clearfly Communications SIP Trunking Service with registration.

In the sample configuration, the enterprise solution consists of an Avaya IP Office Server Edition which also includes the Avaya Preferred Edition (a.k.a. Voicemail Pro) messaging application, Avaya one-X® Portal and Avaya WebRTC Gateway. Endpoints used in the test environment included Avaya H.323 and SIP desk phones, Avaya Communicator for Windows and Avaya Communicator for Web. Customers using this Avaya IP Office enterprise solution with the Clearfly Communications SIP Trunking Service are able to place and receive PSTN calls via a broadband WAN connection using SIP. This converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI.

For brevity, Clearfly Communications could be referred to as “Clearfly” in the remainder of this document. The term “service provider” was also used to refer to Clearfly Communications.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Avaya IP Office to connect to Clearfly Communications via the public Internet. The configuration shown in **Figure 1** was used to exercise the features and functionality tests listed in **Section 2.1**.

The Clearfly Communications SIP Trunking Service passed compliance testing with any observations or limitations described in **Section 2.2**.

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 compliance test.

- SIP OPTIONS queries and responses.
- Incoming calls from the PSTN to H.323 and SIP telephones at the enterprise. All inbound calls from the PSTN were routed from the service provider across the SIP trunk to the enterprise.
- Outgoing calls to the PSTN from H.323 and SIP telephones at the enterprise. All outbound calls to the PSTN were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound calls to the Avaya Communicator for Windows (SIP soft client).

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- Inbound and outbound calls to the Avaya Communicator for Web (WebRTC).
- Various call types including: local, long distance, toll-free, international, local directory assistance, and emergency 911 calls.
- G.711u and G.729a codecs.
- DTMF transmission using RFC 2833.
- Caller ID presentation and Caller ID restriction.
- User features such as hold and resume, internal call forwarding, transfer, and conference.
- Off-net call transfer, conference, forwarding and mobility (mobile twinning).
- Use of the SIP REFER method for call transfers to the PSTN.
- Voicemail navigation for inbound and outbound calls, and voicemail Message Waiting Indicator (MWI).
- Inbound and outbound long duration and long hold time call stability.
- Response to incomplete call attempts and trunk errors.

Fax was not tested as part of this compliance test. However, since fax was tested previously with Avaya IP Office 9.1 without registration (**Section 9**) and the use of registration is not believed to impact fax operation, then the results of the previous fax testing applies to this test configuration using registration. Any fax related observation made during the previous testing is included in **Section 2.2**.

2.2. Test Results

Interoperability compliance testing of the Clearfly SIP Trunking Service was completed with successful results for all test cases with the exception of the observations/limitations described below.

- **Caller ID with Blind Transfer from SIP endpoints** – If an outbound PSTN call from an Avaya IP Office SIP endpoint is blind transferred (transferred before answered) by the endpoint to another PSTN destination, then the caller ID shows the trunk pilot number as the calling party. It was expected that the caller ID would show the transferring party (or the connected PSTN party) as the caller. Upon investigation, it was learned that the Clearfly softswitch is overwriting the value sent from Avaya IP Office with the trunk pilot number. Possible changes to this behavior are under consideration.
- **REFER Signaling** – When using the SIP REFER method for off-net call transfer, post-REFER signaling between Clearfly and the enterprise was sometimes not completely clean. Clearfly would issue a BYE, immediately after accepting the REFER message from the enterprise, to terminate the call with the original enterprise caller. Due to timing, the enterprise could respond to the subsequent messages from Clearfly (to the enterprise caller) with "405 Method Not Allowed" or "481 Call Leg/Transaction does not exist" since the call was already terminated by the previous BYE. This problem did not negatively affect the call transfer itself – off-net call transfer to the PSTN was successfully verified in the compliance test.
- **T.38 Fax** – During previous compliance testing, inbound T.38 fax worked properly with Avaya IP Office issuing the reINVITE message towards Clearfly to re-negotiate to the T.38 codec after call connect. However, outbound T.38 faxes almost always fell back to G.711u pass-through (treating fax as regular voice calls with best effort). Clearfly passes fax call

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signaling straight through to/from its terminating carriers, and therefore does not guarantee codec re-negotiation to T.38 on outbound faxes from the service side.

- **Direct Media** – The Direct Media capability on Avaya IP Office allows IP endpoints to send RTP media directly to each other rather than having all the media flow through the Avaya IP Office, using up VoIP and relay resources. This capability is not supported by Avaya IP Office on the SIP trunk connection which allows T.38 fax in addition to voice calls. Consequently, Direct Media was disabled for the test circuit configured for the compliance test.
- **Remote Worker** – Remote Worker (phones connected directly to the public Internet function as enterprise local extensions) is not supported by the combined Avaya/Clearfly solution as documented in these Application Notes since its setup requires the use of an Avaya Session Border Controller for Enterprise (Avaya SBCE) whereas in the tested solution the Avaya SBCE was not used.

Items not supported by the Clearfly SIP Trunking Service included the following:

- **Operator Calls** – Clearfly does not support Operator (0) and Operator-Assisted (0 + 10-digits) calls.
- **Session Timer –** Session timer based on RFC 4028 is not supported by Clearfly. Instead, Clearfly uses a similar approach via reINVITE polling: Clearfly would re-INVITE an active SIP dialog every 15-minutes. During compliance testing, the enterprise sent session refresh UPDATE messages towards Clearfly with the configured session timer on Avaya IP Office.

2.3. Support

For support on the Clearfly SIP Trunking Service, please contact Clearfly Communications via the following:

- Web: https://www.clearfly.net/support/
- Phone: (866) 652-7520

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com.

3. Reference Configuration

Figure 1 illustrates the test configuration showing an enterprise site connected to the Clearfly SIP Trunking Service via the public Internet.

Within the enterprise site is an Avaya IP Office Server Edition running in a VMware virtual environment. Avaya IP Office Server Edition includes Avaya Preferred Edition (a.k.a. Voicemail Pro) for voicemail and the Avaya one-X® Portal and Avaya WebRTC Gateway which together provides support for WebRTC clients. Endpoints include various Avaya IP Telephones (with H.323 and SIP firmware) and the Avaya Communicator for Windows (SIP) and Avaya Communicator for Web (WebRTC) soft clients. The site also has a Windows PC running Avaya IP Office Manager for administering the Avaya IP Office.

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Mobility Twinning is configured for some of the Avaya IP Office users so that calls to these user phones will also ring and can be answered at the configured mobile phones.

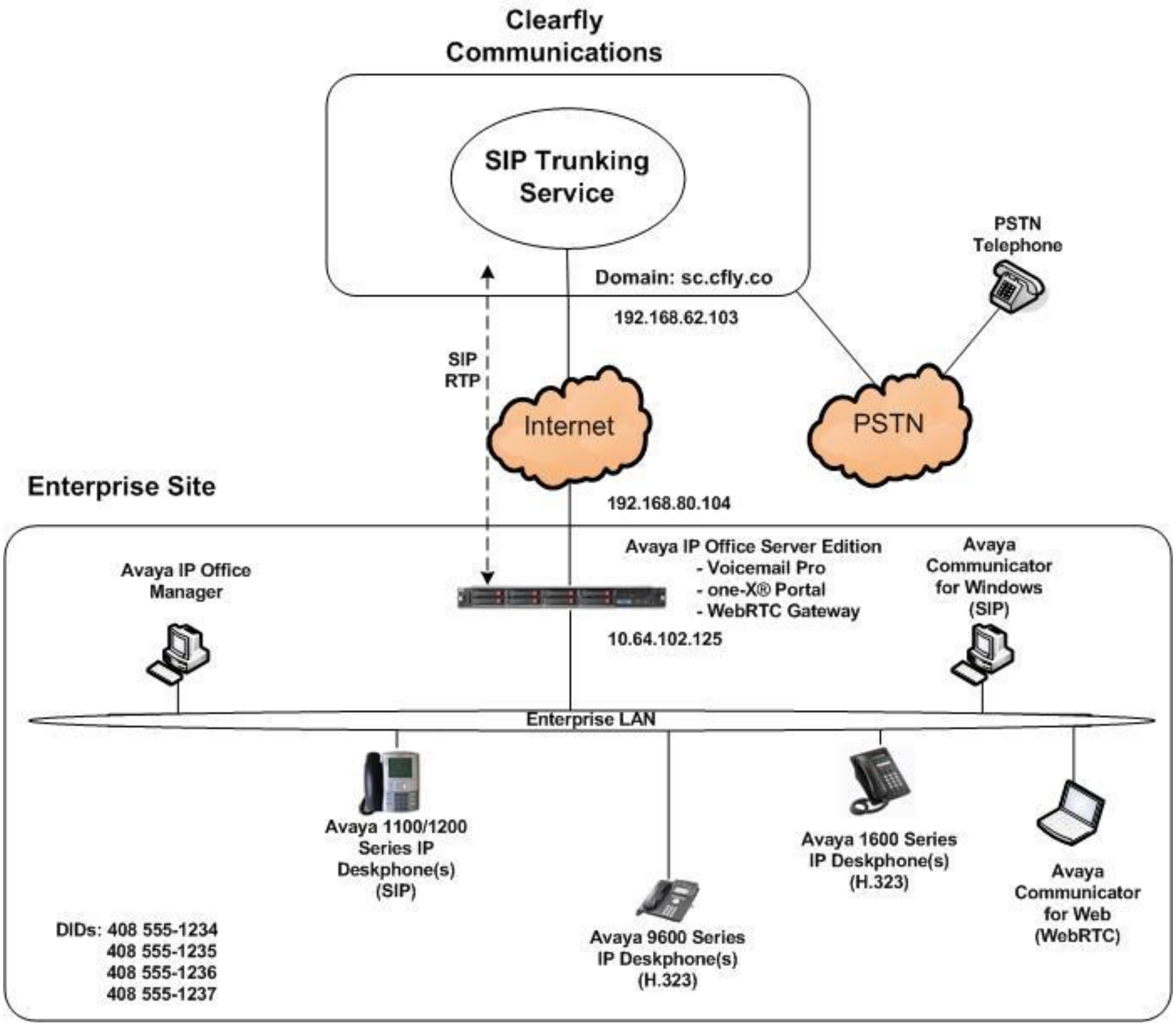

Figure 1: Avaya IP Office with Clearfly Communications SIP Trunking Service

For security reasons, any actual public IP addresses used in the configuration have been replaced with private IP addresses in these Application Notes.

During compliance testing, enterprise users dialed a prefix digit 9 plus N digits to send an outbound call to the number N across the SIP trunk to Clearfly. The short code of 9 was stripped off by Avaya IP Office but the remaining N digits were sent to the service provider network. For calls within the North American Numbering Plan (NANP), the user dialed 11 (1 + 10) digits for long distance and local calls. Thus, for these NANP calls, Avaya IP Office sent 11 digits in the Request URI and the To header of an outbound SIP INVITE message. Clearfly sent 10 digits in the Request URI and the To header of inbound SIP INVITE messages.

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In an actual customer configuration, the enterprise site may also include additional network components between the service provider and the enterprise network such as a data firewall. A complete discussion of the configuration of these devices is beyond the scope of these Application Notes. However, it should be noted that SIP and RTP traffic between the service provider and Avaya IP Office must be allowed to pass through these devices.

The administration of the Avaya Preferred Edition (Voicemail Pro) messaging service, Avaya one-X® Portal, Avaya WebRTC Gateway and endpoints on Avaya IP Office is standard. Since these configuration tasks are not directly related to the interoperation with the Clearfly SIP Trunking Service, they are not included in these Application Notes.

4. Equipment and Software Validated

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

Avaya Telephony Components	
Equipment / Software	**Release / Version**
Avaya IP Office Server Edition running in a VMware virtual environment • Avaya Preferred Edition (Voicemail Pro) • Avaya one-X® Portal • Avaya WebRTC Gateway	9.1.8.0 build 172 9.1.8.0 build 9 9.1.8.0 build 13 9.1.8.0 build 1013
Avaya IP Office Manager	9.1.8.0 build 172
Avaya 1616 IP Deskphone (H.323) running Avaya one-X® Deskphone Value Edition	1.3 SP5
Avaya 9641G IP Deskphone (H.323) running Avaya one-X® Deskphone Edition	6.6.1 (6.6115)
Avaya 1140E IP Deskphone (SIP)	4.04.23.00
Avaya Communicator for Windows	2.0.3 (2.0.3.55)
Avaya Communicator for Web	1.0.16.1718
Clearfly Communications Components	
Equipment / Software	**Release / Version**
Metaswitch Softswitch	v9.3
Metaswitch Perimeta Session Border Controller	v3.9

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 (T.38 fax is not supported on IP Office Server Edition). Note that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks.

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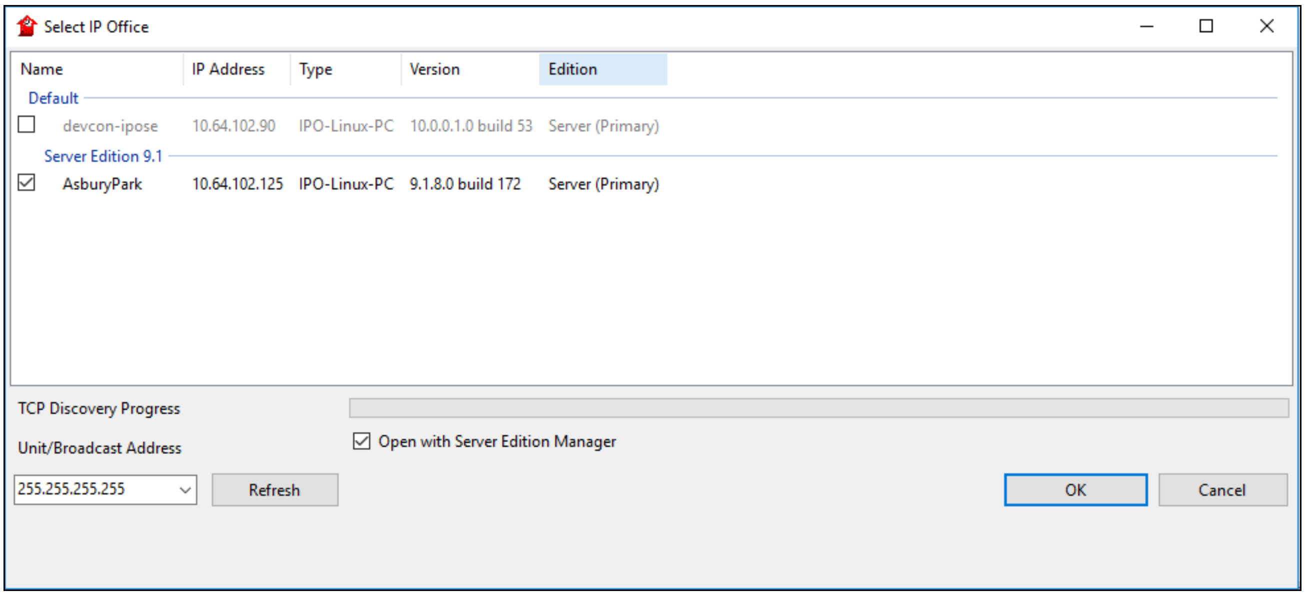
5. Configure Avaya IP Office

The configuration screens shown in this document were taken from an Avaya IP Office Server Edition system and as such may vary slightly from the same screen on an Avaya IP Office 500V2 system. However, as previously stated the configuration applies to both.

Avaya IP Office is configured through the Avaya IP Office Manager PC application. From the PC running Avaya IP Office Manager, select **Start → All Programs → IP Office → Manager** to launch the application. A **Select IP Office** pop-up window is displayed as shown below. Select the proper Avaya IP Office system from the pop-up window and click **OK** to log in with the appropriate credentials (not shown).



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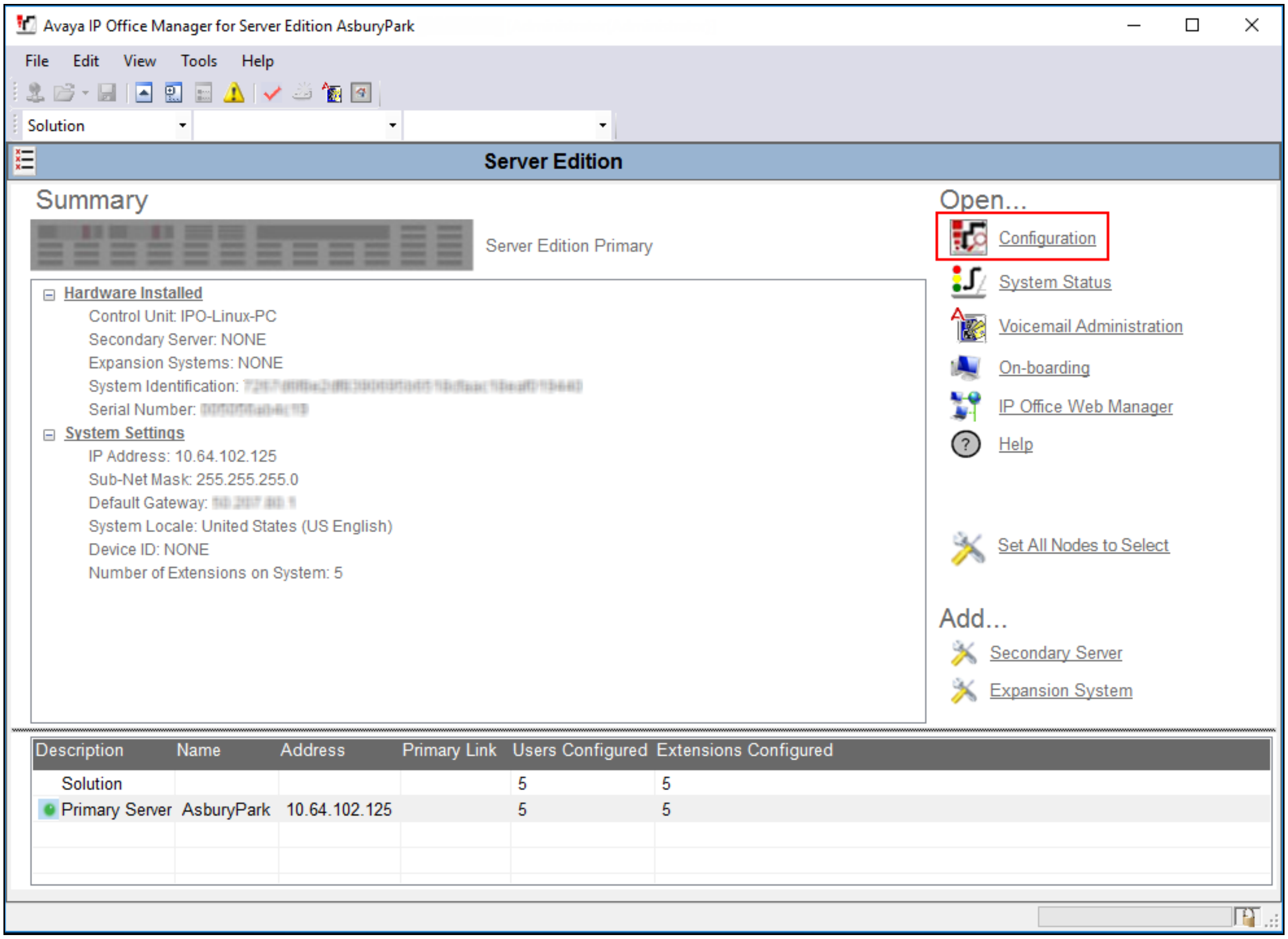
After logging in an Avaya IP Office 500V2 system, the configuration will be displayed. On an Avaya IP Office Server Edition system, the following screen will appear. Click **Configuration** to display the configuration.



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 and Details) will be referenced throughout the Avaya IP Office configuration.

All licensing and feature configuration that is not directly related to the interface with the service provider (such as administering IP endpoints) is assumed to already be in place.

In the sample configuration, **AsburyPark** was used as the system name. All navigation described in the following sections (e.g., **IP Route → 0.0.0.0)** appears as submenus underneath the system name **AsburyPark** (or **Solution → AsburyPark**) in the Navigation Pane. The configuration screens highlight values/settings configured for the compliance test. Defaults were used for other values and may be customized based upon requirements in the field.

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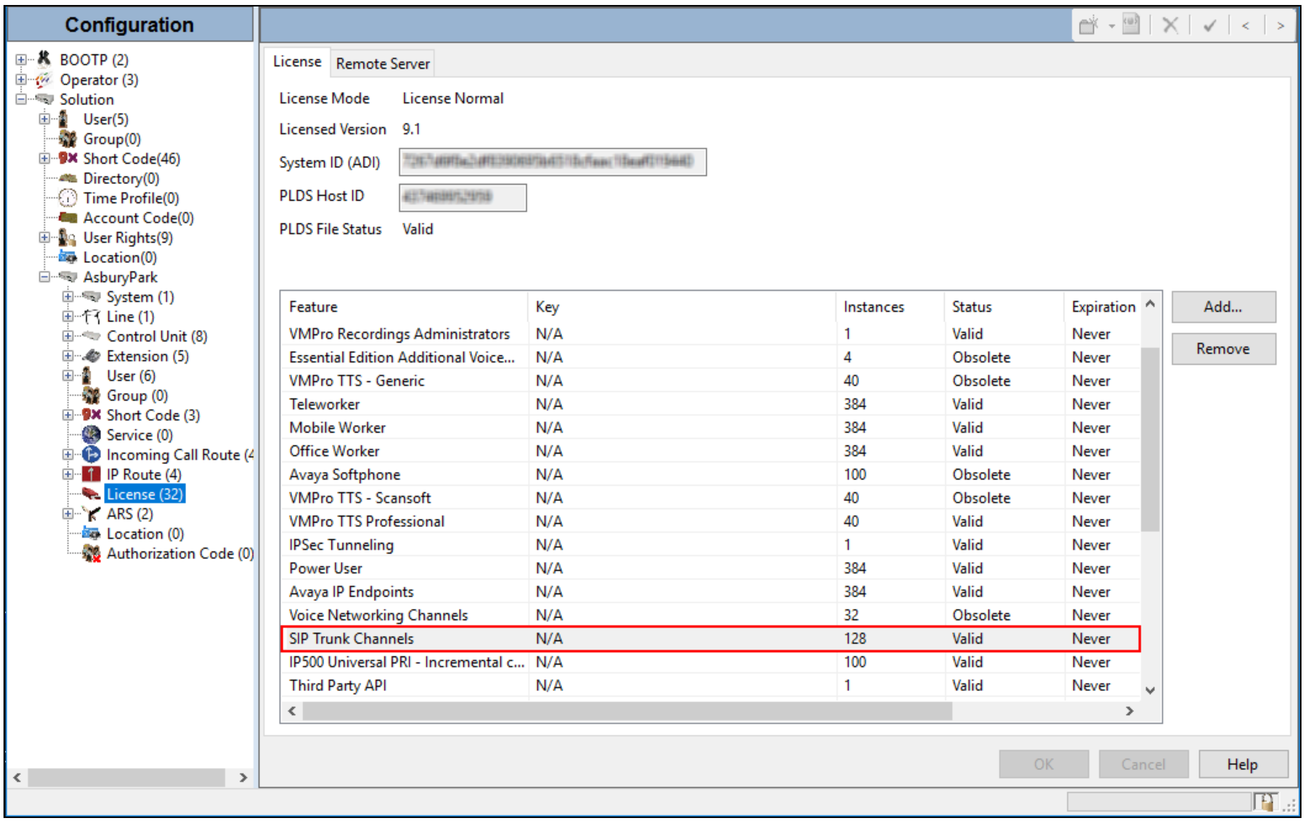
5.1. Licensing

The configuration and features described in these Application Notes require Avaya IP Office 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 exists with sufficient **Instances** (trunk channels) in the Details Pane.



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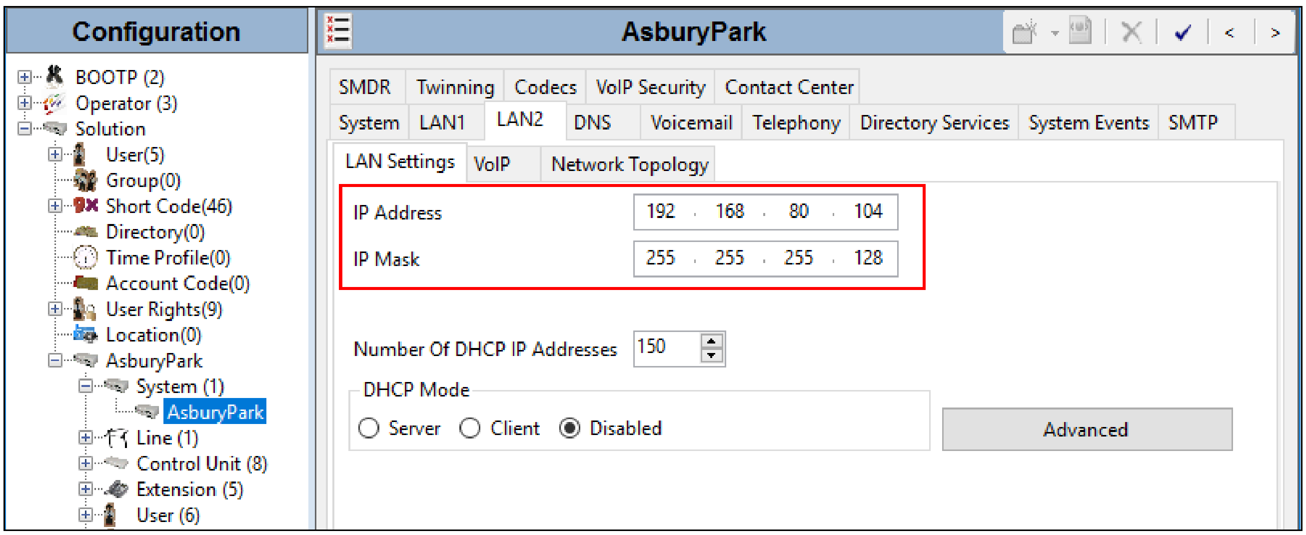
5.2. System

This section configures the necessary system settings.

5.2.1. System – LAN2 Tab

In the sample configuration, **AsburyPark** was used as the system name and the WAN port (LAN2) was used to connect the Avaya IP Office to the public Internet (see **Figure 1**). The LAN2 settings correspond to the WAN interface on Avaya IP Office. To access the LAN2 settings, first navigate to **System → AsburyPark** in the Navigation Pane and then navigate to the **LAN2→ LAN Settings** tab in the Details Pane. Set the **IP Address** field to the IP address assigned to the Avaya IP Office WAN port. Set the **IP Mask** field to the proper mask value.



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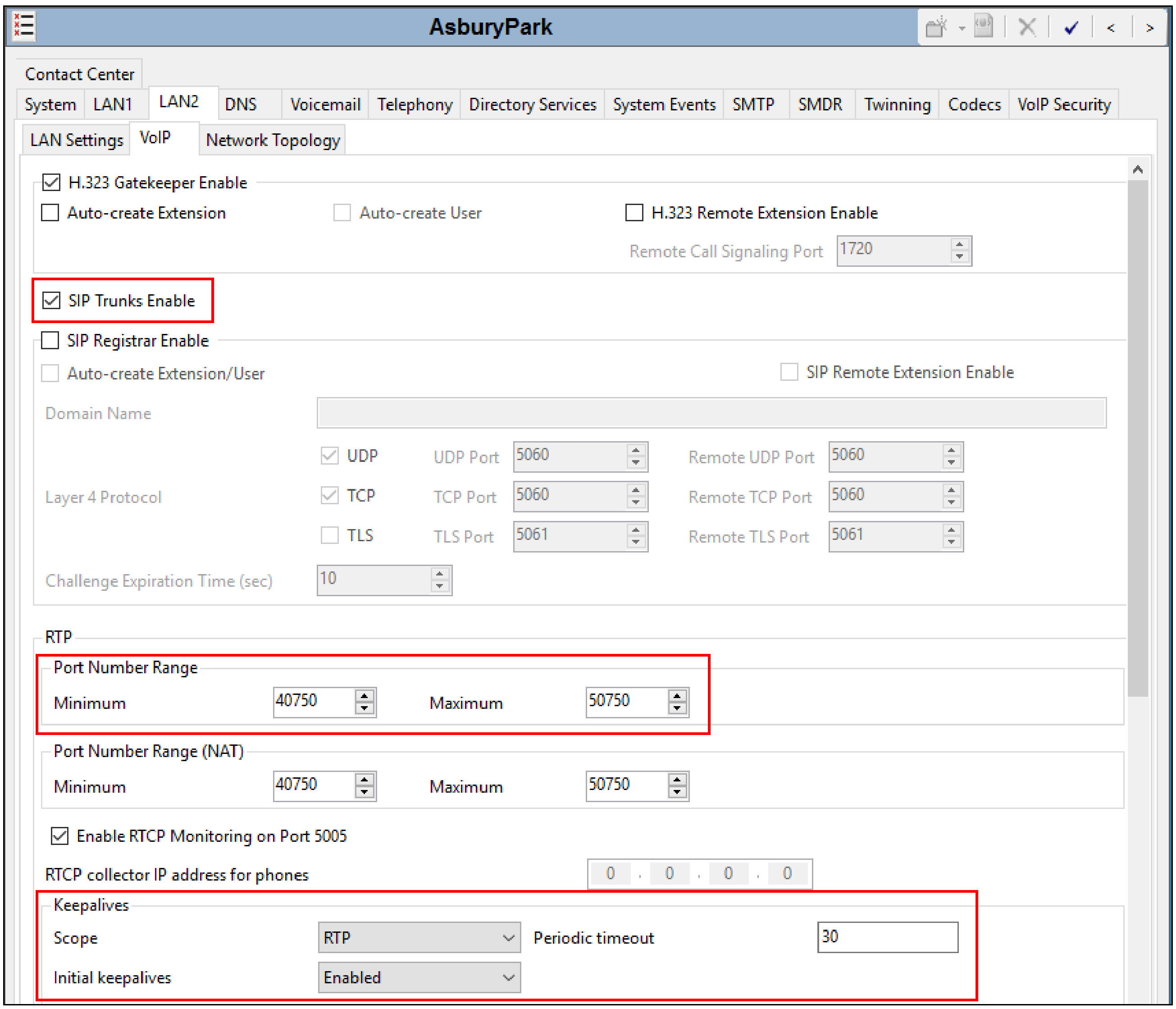
On the **VoIP** tab of LAN2 in the Details Pane, configure the following parameters:

- Check the **SIP Trunks Enable** box to enable the configuration of SIP trunks.
- In the **RTP** section, the **RTP Port Number Range** can be customized to a specific range of receiving ports for the RTP media, as agreed with the service provider. Based on this setting, Avaya IP Office would request RTP media be sent to a port in the configurable range for calls using LAN2.
- In the **Keepalives** section, select ***RTP*** for **Scope**; select ***Enabled*** for **Initial keepalives**; enter ***30*** for **Periodic timeout**. These settings direct Avaya 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 for media from the other, as well as helping to keep firewall (if used) ports open for the duration of the call.



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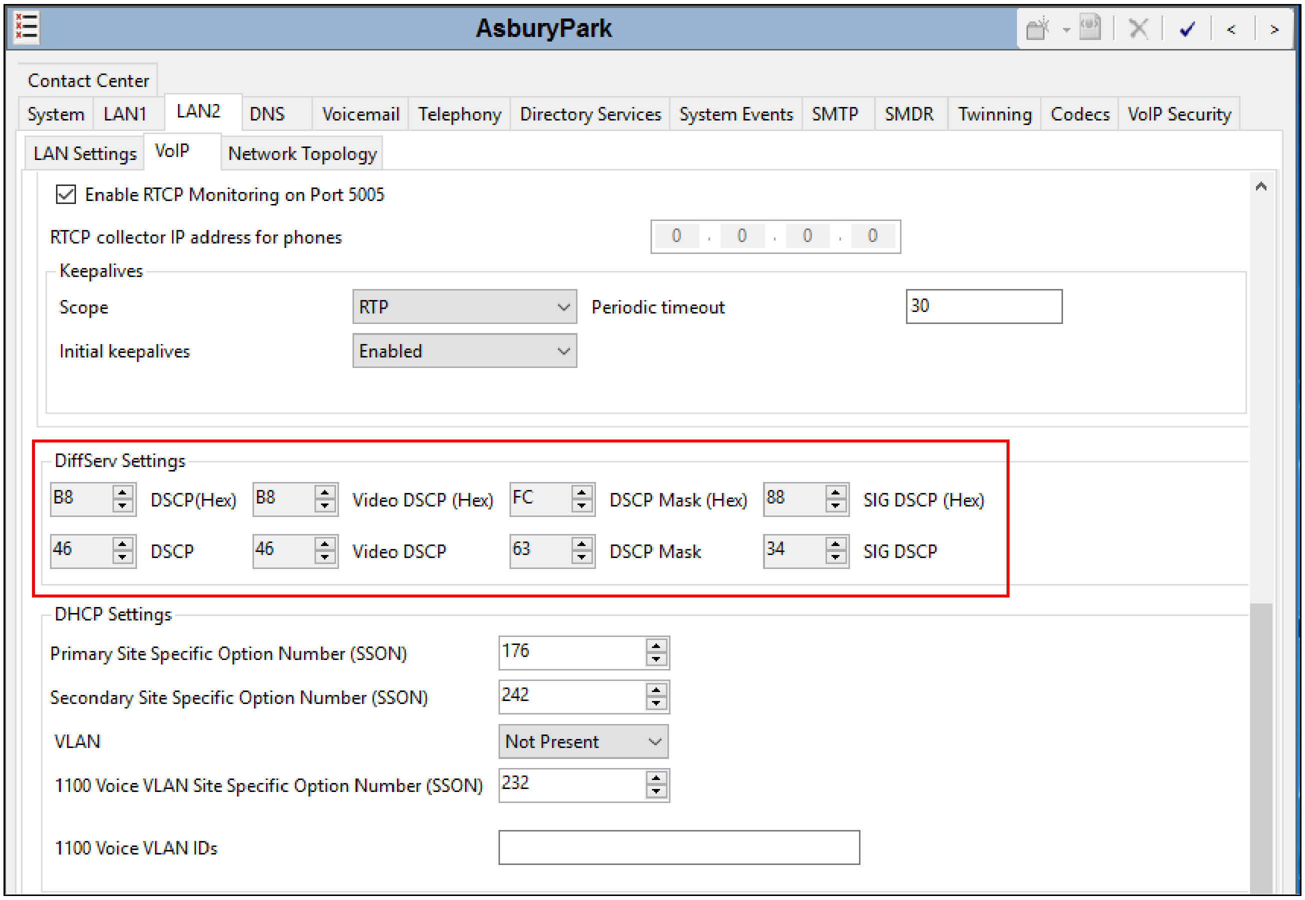
Scroll down to the **DiffServ Settings** section. Avaya IP Office can be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling. The specific values used for the compliance test are shown in the screen below and are also the default values. For a customer installation, if the default values are not sufficient, appropriate values should be provided by the customer.



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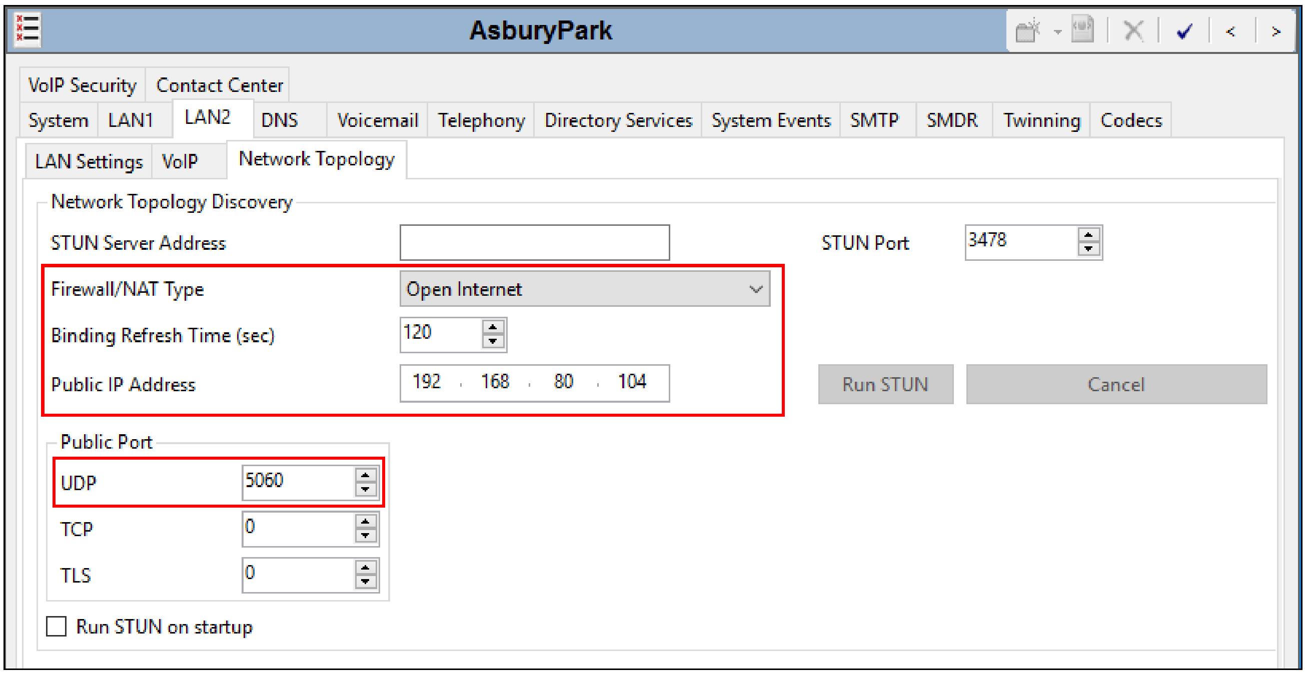
On the **Network Topology** tab of LAN2 in the Details Pane, configure the following parameters:

- Select **Firewall/NAT Type** from the pull-down menu that matches the network configuration. No firewall or network address translation (NAT) device was used in the compliance test as shown in **Figure 1**, so the parameter was set to ***Open Internet***. With the ***Open Internet*** setting, the **STUN Server Address** is not used.
- Set **Binding Refresh Time (seconds)** to a desired value. This value is used as one input to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider. See **Section 5.9** for complete details.
- Set **Public IP Address** to the IP address of the Avaya IP Office WAN port.
- Set **Public Port** to ***5060*** for **UDP**.



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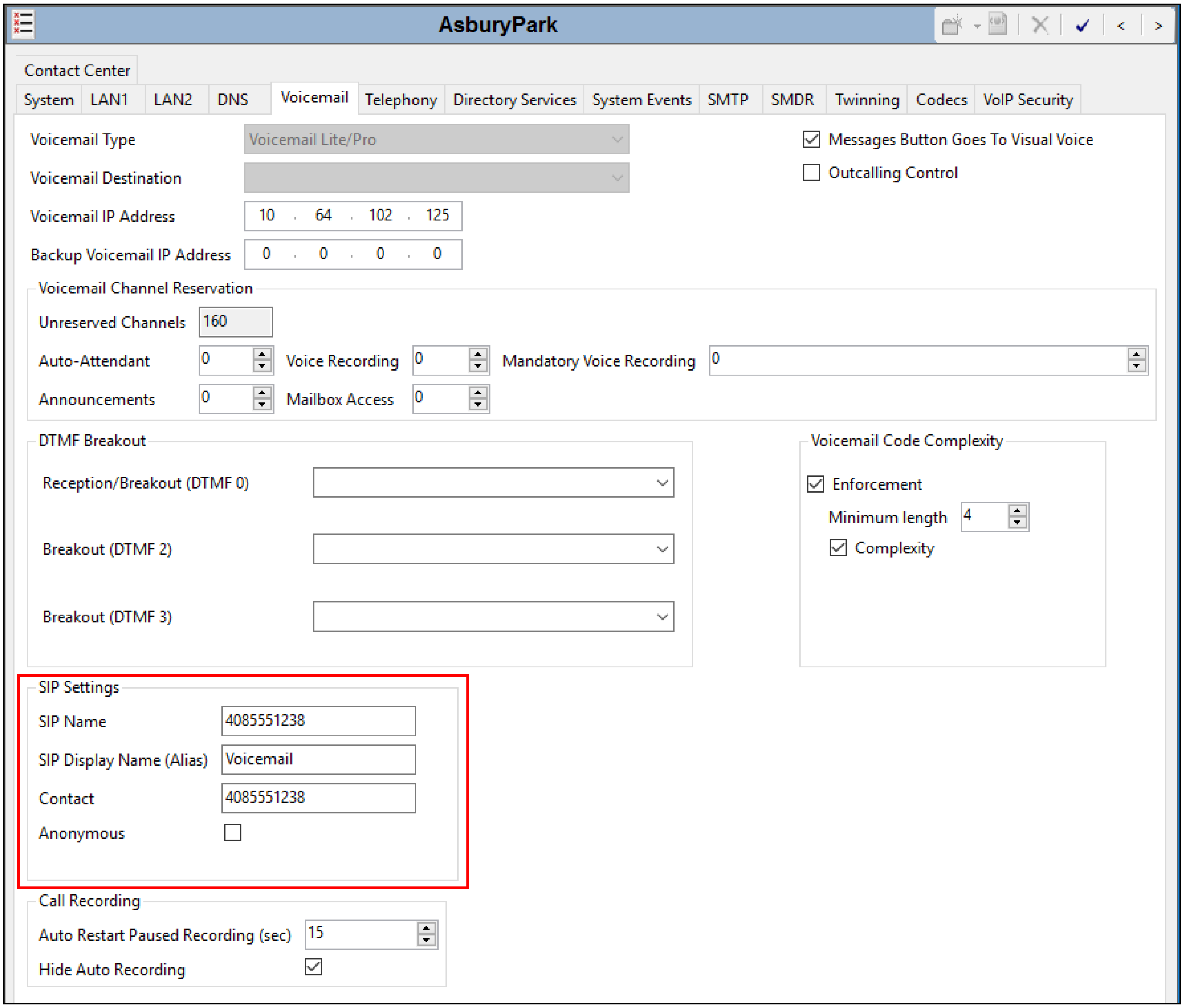
5.2.2. System - Voicemail Tab

In the **Voicemail** tab of the Details Pane, configure the **SIP Settings** section. The **SIP Name** and **Contact** are set to one of the DID numbers provided by Clearfly. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. Uncheck the **Anonymous** box to allow the Voicemail Caller ID information to be sent to the network.



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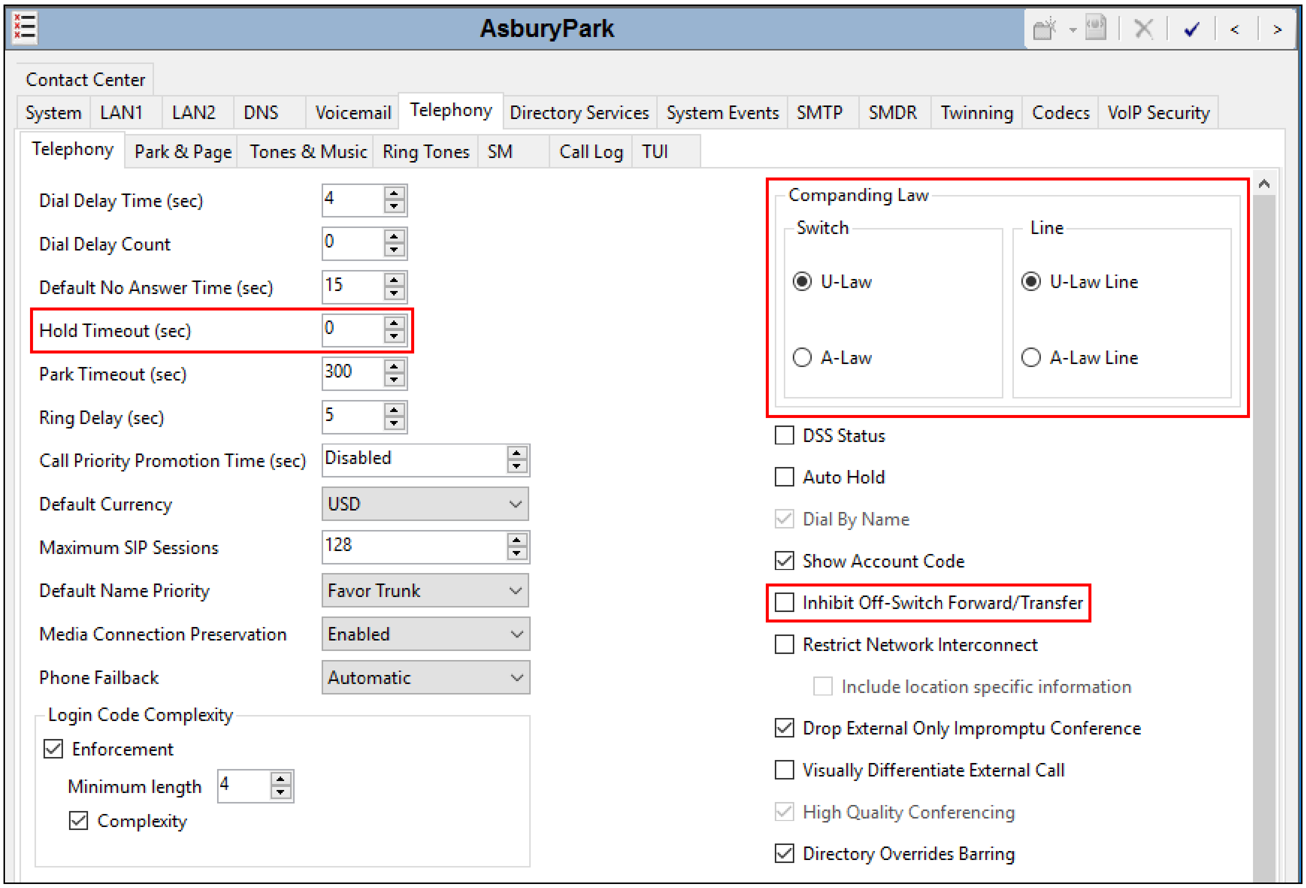
5.2.3. System - Telephony Tab

Navigate to the **Telephony → Telephony** tab in the Details Pane. Enter or select ***0*** for **Hold Timeout (sec)** so that calls on hold will not time out. Choose the **Companding Law** typical for the enterprise site. For the compliance test, ***U-Law*** was used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the service provider per customer business policies. Note that this configuration might pose a security issue (Toll Fraud). Customers should exercise caution with this configuration.



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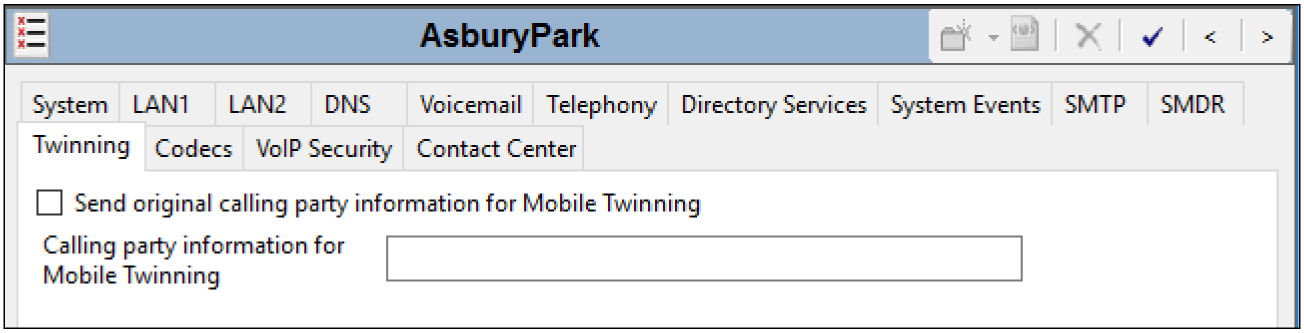
5.2.4. System - Twinning Tab

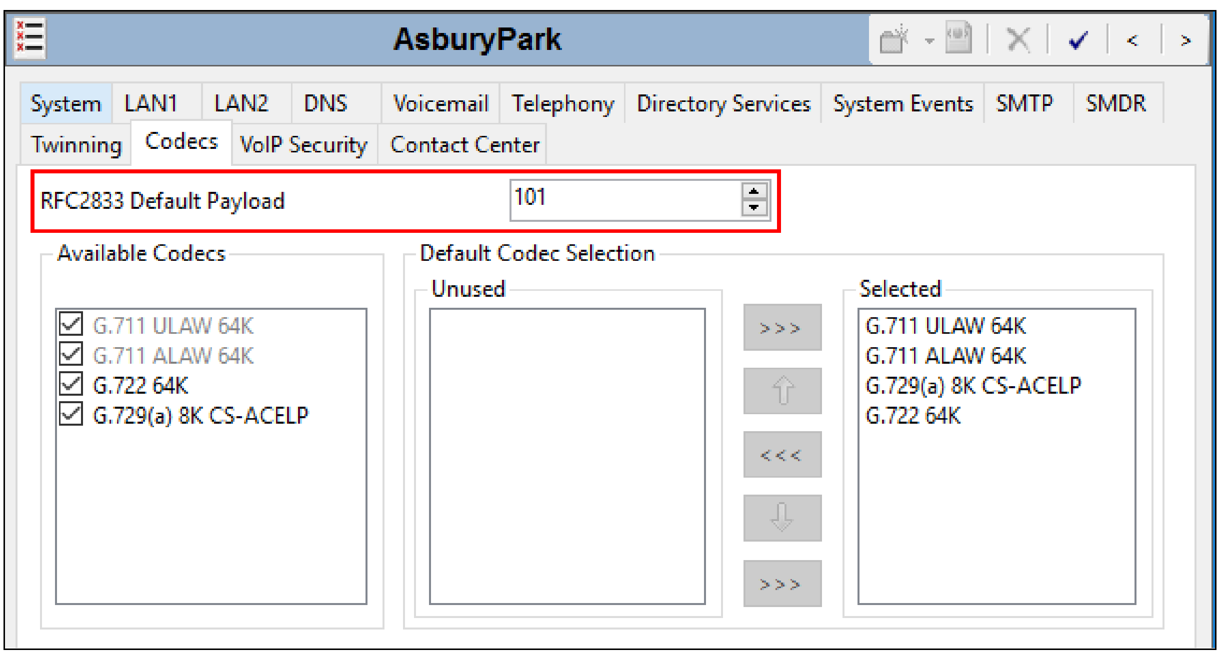
To view or change the System Twinning settings, navigate to the **Twinning** tab in the Details Pane as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked in the sample configuration, and the **Calling party information for Mobile Twinning**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



5.2.5. System – Codecs Tab

In the **Codecs** tab of the Details Pane, select or enter ***101*** for **RFC2833 Default Payload**. This setting was preferred by Clearfly for use with out-of-band DTMF tone transmissions.

On the left, observe the list of **Available Codecs**. In the screen below, which is not intended to be prescriptive, the box next to each codec is checked, making all the codecs available in other screens where codec configuration may be performed. The **Default Codec Selection** area enables the codec preference order to be configured on a system-wide basis. By default, all IP (SIP and H.323) lines and extensions will assume the system default codec selection, unless configured otherwise for the specific line or extension.



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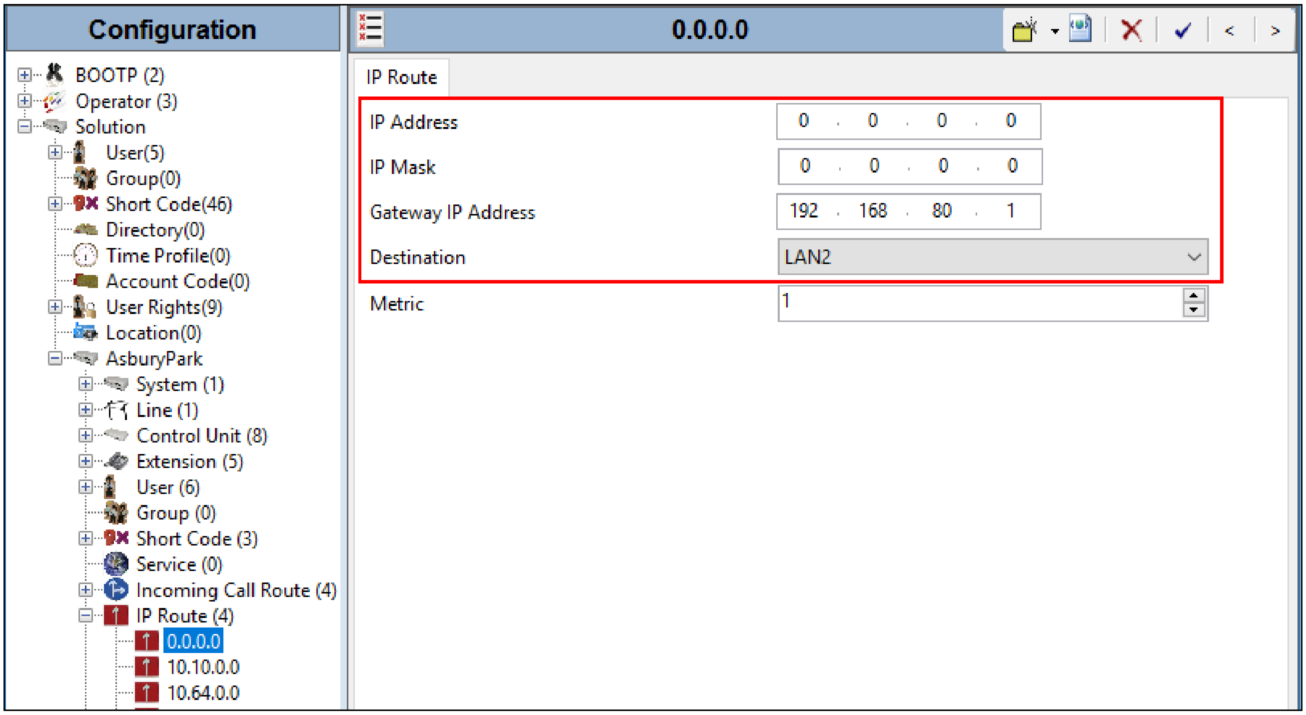
5.3. IP Route

Navigate to **IP Route → 0.0.0.0** in the left Navigation Pane if a default route already exists. Otherwise, to create the default route, right-click on **IP Route** and select **New** (not shown). Create and verify a default route with the following parameters:

- Set **IP Address** and **IP Mask** to ***0.0.0.0***.
- Set **Gateway IP Address** to the IP address of the gateway for the public Internet to which the Avaya IP Office WAN port is connected.
- Set **Destination** to ***LAN2*** from the drop-down list.



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5.4. Administer SIP Line

A SIP Line is needed to establish the SIP connection between Avaya IP Office and the Clearfly network. 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 **Section 5.4.1** to create the SIP Line from the 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 Avaya IP Office Standard Edition (500 V2) and Avaya IP Office Server Edition systems.

Some items relevant to a specific customer environment are not included in the template associated with these Application Notes, 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 SIP Line **Transport** tab

Therefore, it is important that the SIP Line configuration be reviewed and updated after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.4.2** through **5.4.8**.

Also, the following SIP Line settings are not supported on Avaya IP Office Basic Edition:

- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required

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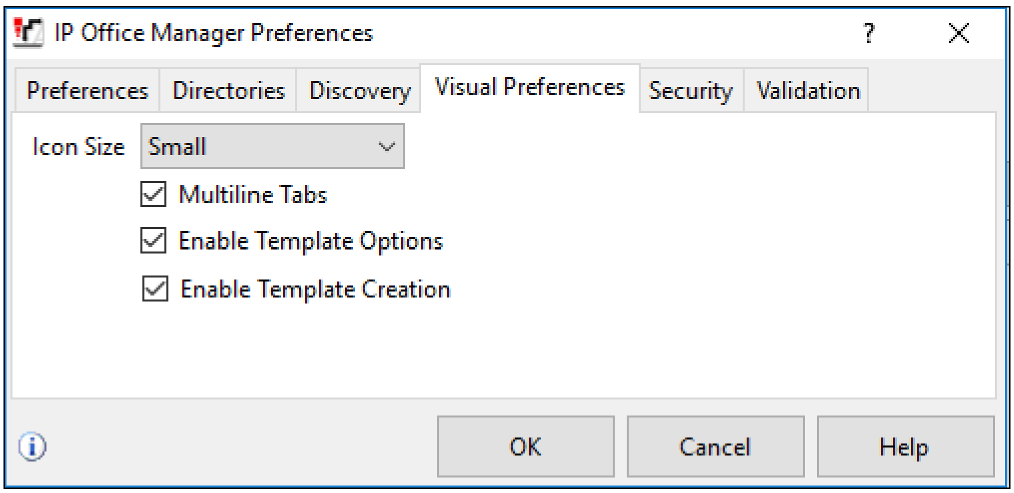
5.4.1. Create SIP Line From Template

1. Copy the template file to a location (e.g., C:\Temp) on the computer where Avaya IP Office Manager is installed. Rename the template file to

    AF_ClearflyReg_SIPTrunk.xml.

    The file name is important in locating the proper template file in **Step 4**.

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



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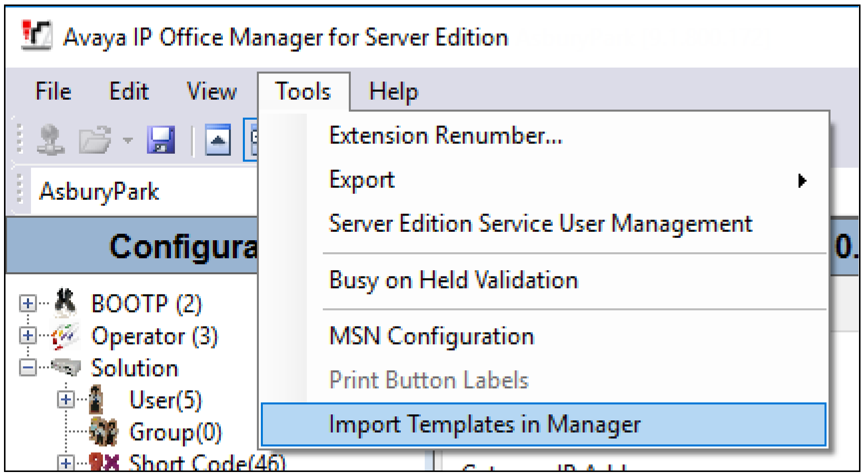
3. Import the template into Avaya IP Office Manager. From Avaya IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the Avaya IP Office template directory and make the template available in the Avaya IP Office Manager pull-down menus in **Step 4**. The default template location is **C:\Users**<*UserName*>**\AppData\Local\VirtualStore\Program Files\Avaya\IP Office\Manager\Templates**.



In the pop-up window that appears (not shown), select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window (not shown) will appear stating success or failure. Click **OK** (not shown) to continue.

If preferred, this step may be skipped if the template file is copied directly to the Avaya IP Office template directory.

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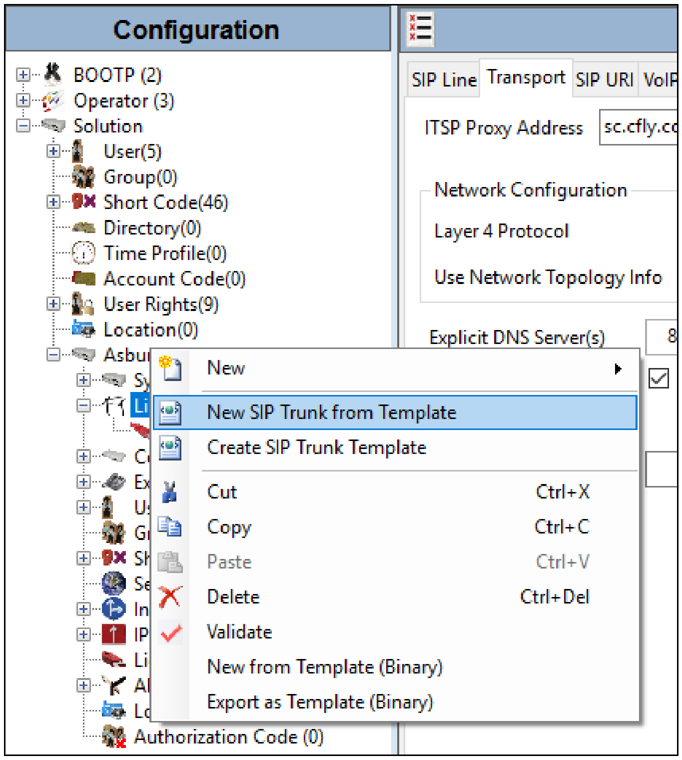
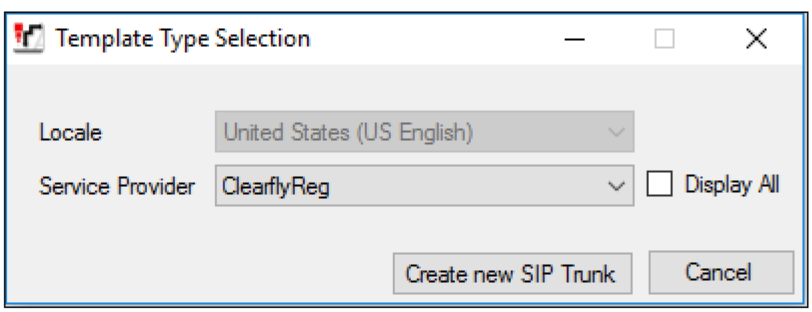
4. To create the SIP Trunk from the template, right-click on **Line** in the Navigation Pane, then select **New SIP Trunk from Template**.



In the subsequent **Template Type Selection** pop-up window, select ***ClearflyReg*** from the **Service Provider** drop-down list as shown below. This selection corresponds to parts of the template file name as specified in **Step 1**. Click **Create new SIP Trunk** to finish creating the trunk.



Note that the newly created SIP Line may not immediately appear in the Navigation pane until the configuration was saved, closed and reopened in Avaya IP Office Manager.

5. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.4.2** through **5.4.8**.

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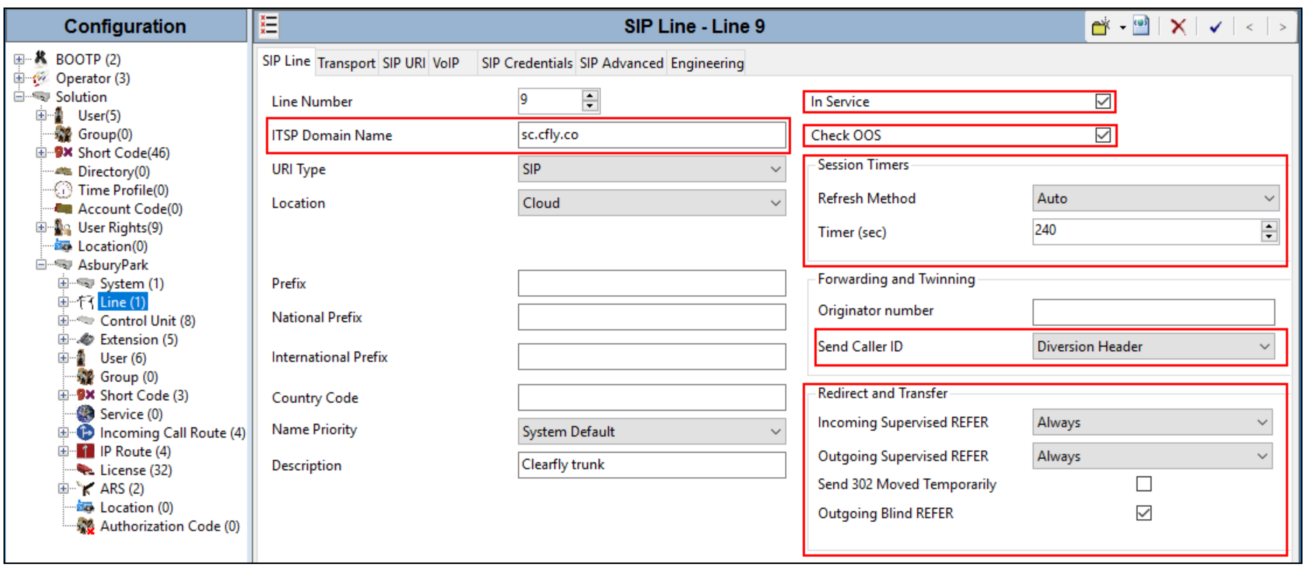
5.4.2. SIP Line – SIP Line Tab

In the **SIP Line** tab of the Details Pane, configure the parameters as shown below:

- Set **ITSP Domain Name** to the service provider SIP domain provided by Clearfly.
- Check the **In Service** box.
- Check the **Check OOS** box. Avaya IP Office will check the SIP OPTIONS response from the far end to determine whether to take the SIP Line out of service.
- In the **Session Timers** section, set **Refresh Method** to ***Auto***. With this setting Avaya IP Office will send UPDATE messages for session refresh if the other party supports UPDATE. If UPDATE is not supported, re-INVITE messages are sent. Set **Timer (sec)** to a desired value. Avaya IP Office will send out session refresh UPDATE or re-INVITE at half of the specified timer value.
- Set **Send Caller ID** under **Forwarding and Twinning** to ***Diversion Header***. With this setting and the related configuration in **Section 5.2.4**, Avaya IP Office will include the Diversion Header for calls that are redirected via Mobile Twinning out the SIP Line to the PSTN. It will also include the Diversion Header for calls that are forwarded out the SIP Line.
- Under **Redirect and Transfer**, select ***Always*** for **Incoming Supervised REFER** and **Outgoing Supervised REFER**. Clearfly supports use of the REFER method for supervised off-net call transfer. **Outgoing Blind REFER** is checked to enable use of REFER for blind transfers as well.



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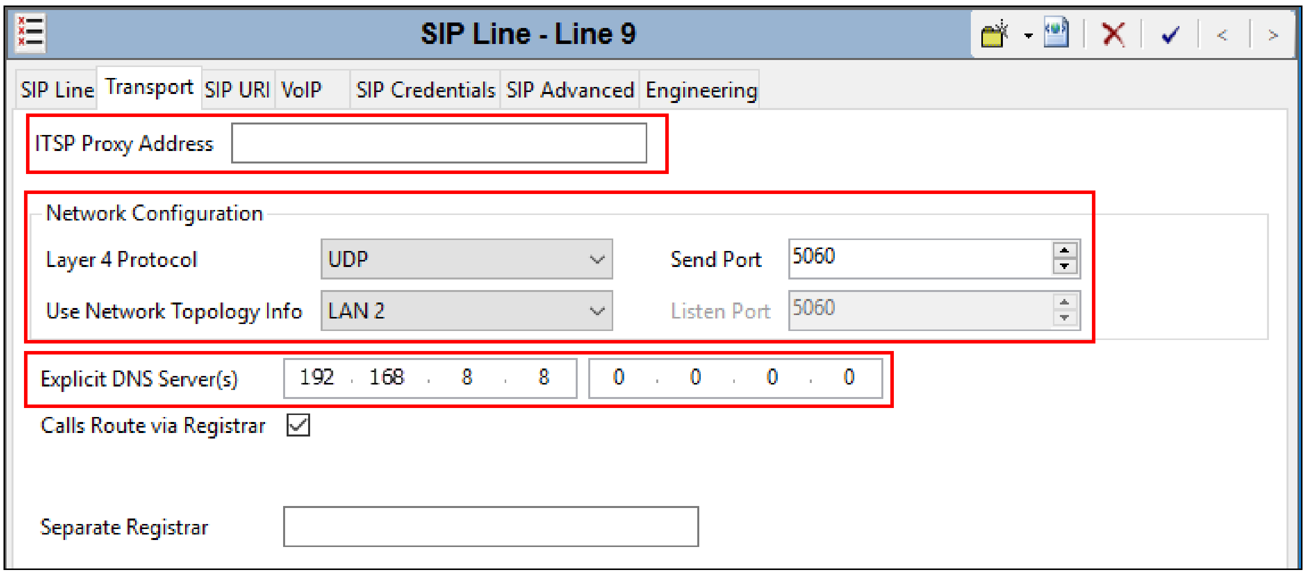
5.4.3. SIP Line – Transport Tab

Navigate to the **Transport** tab and set the following:

- Leave the **ITSP Proxy Address** blank or set it to the **ITSP Domain Name** from **Section 5.4.2**. In either case, the address will be determined from a DNS lookup of the **ITSP Domain Name**.
- Set the **Layer 4 Protocol** to ***UDP***.
- Set **Use Network Topology Info** to the network port used by the SIP line to access the far-end as configured in **Section 5.2.1**.
- Set the **Send Port** to ***5060***.
- Set the **Explicit DNS Server(s)** field to the IP address of the DNS server to use to resolve the **ITSP Domain Name**. This DNS server will only be used for this SIP Line. Alternatively, a system wide DNS server can be configured in a similar manner on the **System → DNS** tab (not shown).



5.4.4. SIP Line – SIP URI Tab

Select the **SIP URI** tab to create or edit a SIP URI entry. A SIP URI entry matches each incoming number that Avaya IP Office will accept on this line. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane. For the compliance test, a single SIP URI entry was created to match any DID number assigned to Avaya IP Office users.

- Set **Local URI** to ***Use Internal Data***. This setting allows calls on this line whose Request-Line SIP URI matches the **SIP Name** set on the **SIP** tab of any **User** as shown in **Section 5.6**. For outbound calls, the From header will be set from the **SIP Name** data.
- Set **Contact** and **Display Name** to ***Use Internal Data***. This setting will cause the Contact and Display Name data for outbound messages to be set from the corresponding fields on the **SIP** tab of the individual **User** as shown in **Section 5.66**.

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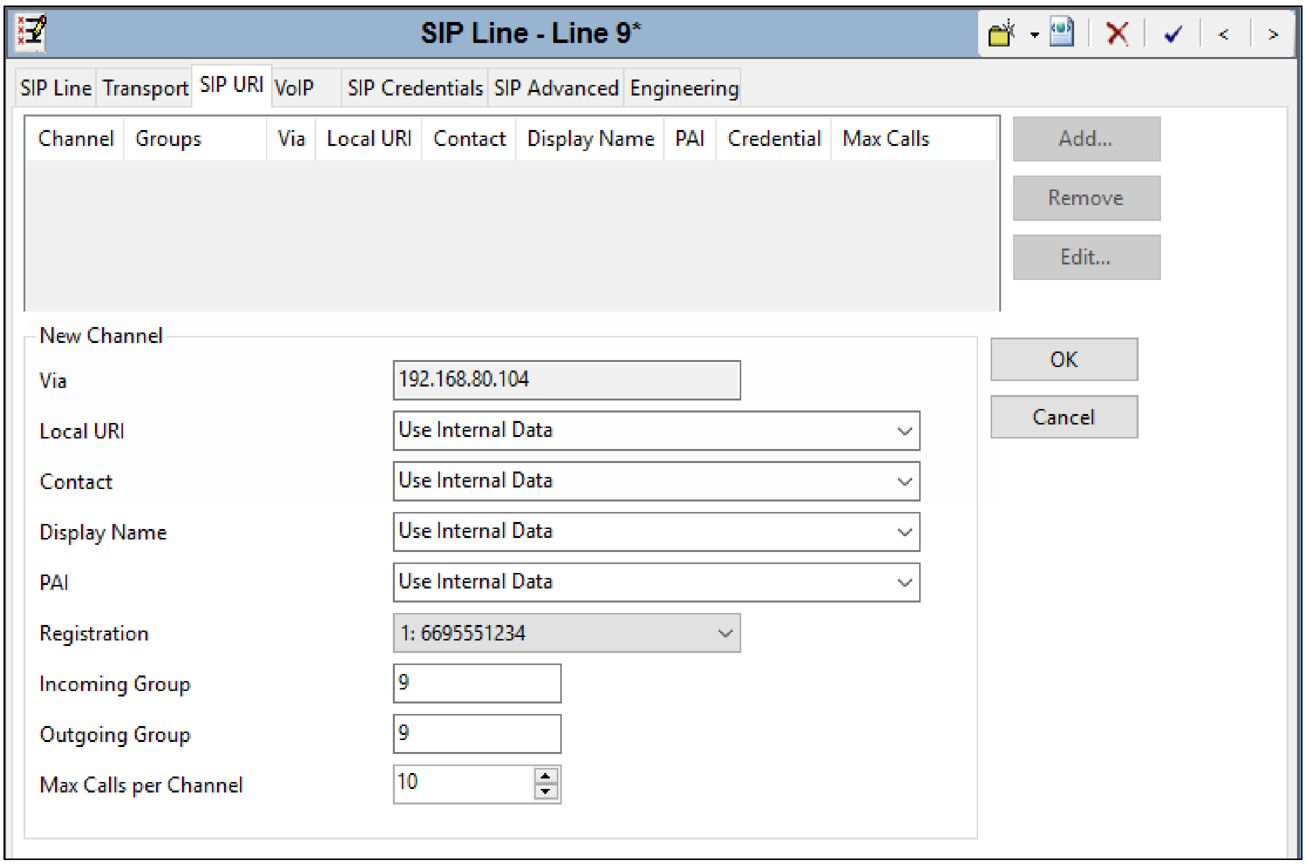
- Set **PAI** to ***Use Internal Data***. This setting directs Avaya IP Office to send the PAI (P-Asserted-Identity) header when appropriate. The PAI header will be populated from the **SIP Name** in the **SIP** tab of the call initiating **User** as shown in **Section 5.66**.
- For the **Registration** field, select the SIP credentials configured in **Section 5.4.7** 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. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. For the compliance test, the incoming and outgoing group ***9*** was specified. Note that this group number can be different than the SIP Line number.
- Set **Max Calls per Channel** to the number of simultaneous SIP calls allowed using this SIP URI pattern.



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

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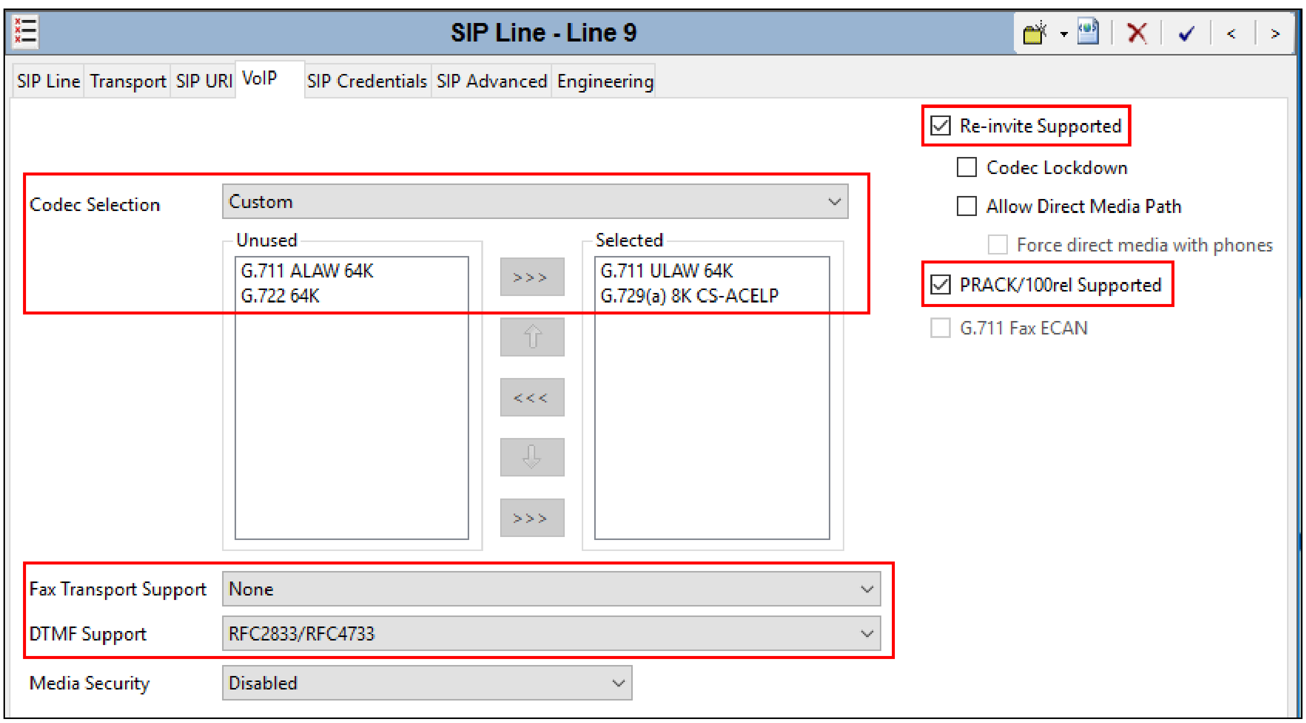
5.4.5. SIP Line – VoIP Tab

Select the **VoIP** tab. Set the parameters as shown below.

- Select ***Custom*** for **Codec Selection**.
- Choose ***G.711 ULAW 64K*** and ***G.729(a) 8K CS-ACELP*** for the **Selected** codecs by using the left and right arrows to move codecs between the **Selected** and **Unused** boxes. Use the up and down arrows in the middle to order these two codecs. The G.711u and G.729a codecs are supported by Clearfly. G.711u was configured as the preferred codec for the compliance test as shown below.
- Select ***None*** for **Fax Transport Support**. This was necessary because testing was performed using Avaya IP Office Server Edition which does not support T.38 and no Expansion System was present to support G.711u fax. T.38 and G.711u fax is supported by Clearfly and may be enabled when the Avaya IP Office platform supports it. To enable fax on the Clearfly SIP trunk, see the fax configuration in the corresponding Application Notes for Clearfly SIP Trunking without registration **[9]** (**Section 9**).
- Select ***RFC2833/RFC4733*** for **DTMF Support**. This directs Avaya IP Office to send DTMF tones as out-of-band RTP events as per RFC2833/RFC4733.
- Check the **Re-invite Supported** option box. When enabled, reINVITE can be used during a call session to change the characteristics of the session including codec renegotiation.
- Check the **PRACK/100rel Supported** option box. This setting enables support by Avaya IP Office for the PRACK (Provisional Reliable Acknowledgement) message on SIP trunks.



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5.4.6. SIP Line – T38 Fax

This tab is only present if the Avaya IP Office platform supports it. The compliance test used an Avaya IP Office Server Edition system which does not support T.38 fax so this tab is not available. To configure fax on an Avaya IP Office platform that supports it, see the fax configuration in the corresponding Application Notes for Clearfly SIP Trunking without registration **[9]** (**Section 9**).

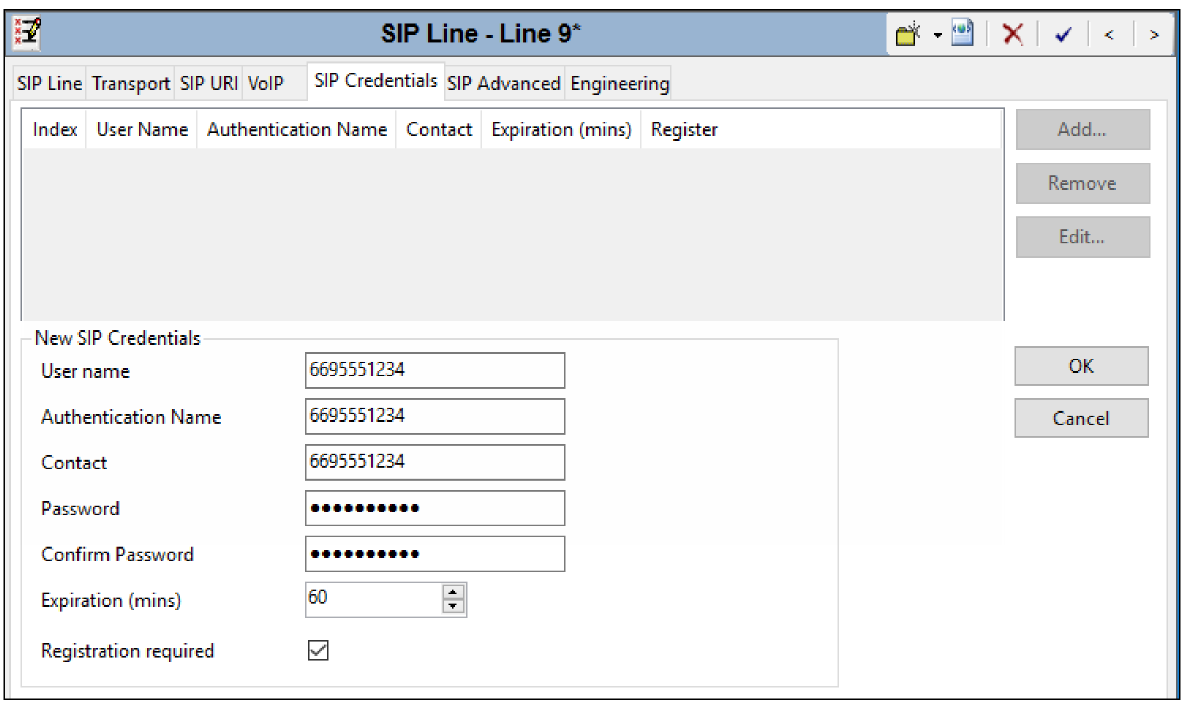
5.4.7. SIP Line – SIP Credentials Tab

SIP credentials are used to register the SIP trunk with a service provider that requires SIP registration. SIP credentials are also used to provide the required information for digest authentication of outbound calls. SIP credentials are unique per customer and therefore customers must contact the service provider to obtain the proper registration and/or digest authentication credentials for their deployment.

To enter the SIP credentials, select the **SIP Credentials** tab and click **Add**. In the **New SIP Credentials** area that appears, enter the information as shown below.

- Set the **User name**, **Authentication Name**, and **Contact** to the user name provided by Clearfly.
- Set the **Password** and **Confirm Password** to the password provided by Clearfly.
- Set **Expiry (mins)** to **60**.
- Check the **Registration required** box.

Click **OK**.



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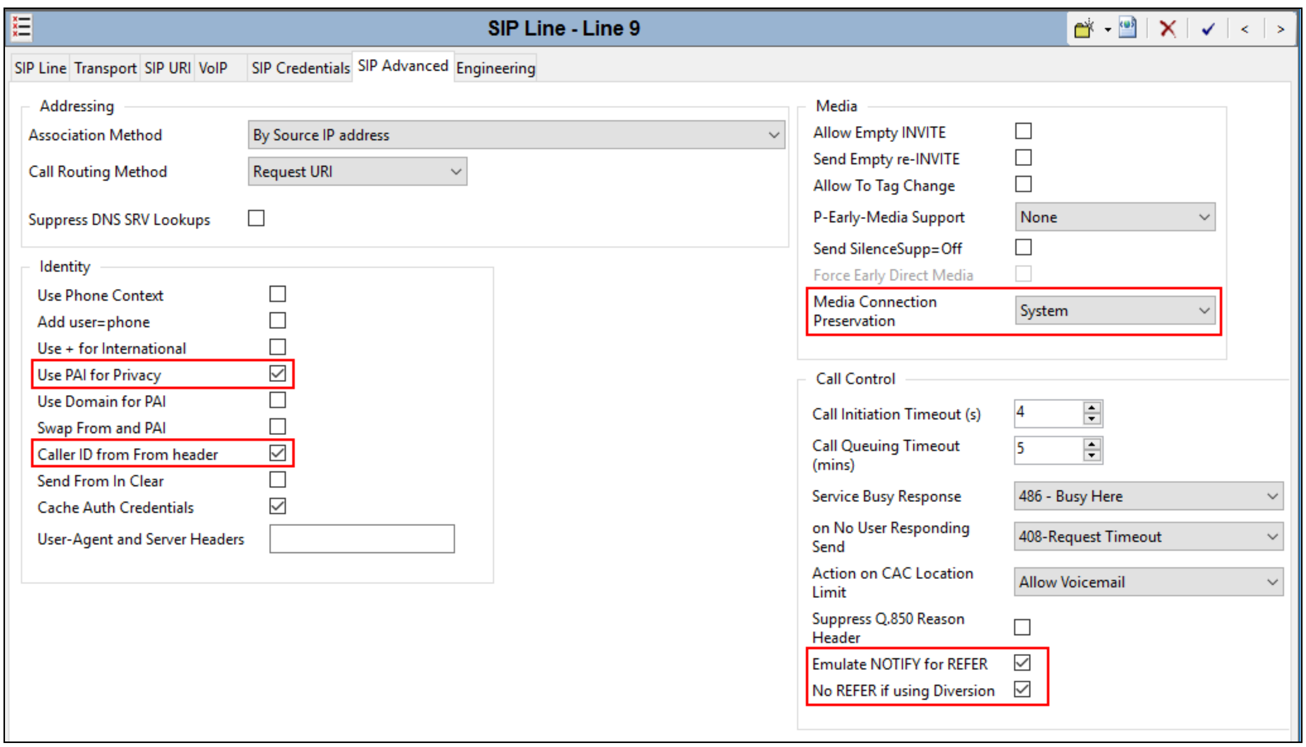
5.4.8. SIP Line – SIP Advanced Tab

Select the **SIP Advanced** tab to configure advanced SIP Line parameters.

In the **Identity** area, the **Use PAI for Privacy** box is checked for Avaya IP Office to use the P-Asserted-Identity (PAI) SIP header for privacy-requested outbound calls. With this configuration, Avaya IP Office will populate the From and Contact headers of the anonymous outbound call INVITE with "anonymous" as the URI user part, but include the normal calling user information in the PAI header. The **Caller ID from From header** box is checked for Avaya IP Office to use the Caller ID information in the From SIP header rather than the PAI or Contact SIP header for inbound calls.

In the **Media** area, select ***System*** for **Media Connection Preservation** to allow established calls to continue despite brief network failures.

In the **Call Control** area, **Emulate NOTIFY for REFER** is checked. This is required for SIP endpoints that perform REFER-based transfers across the SIP line. **No REFER if using Diversion** is checked to prevent Avaya IP Office from using the SIP REFER method on call forward scenarios that use the Diversion SIP header.



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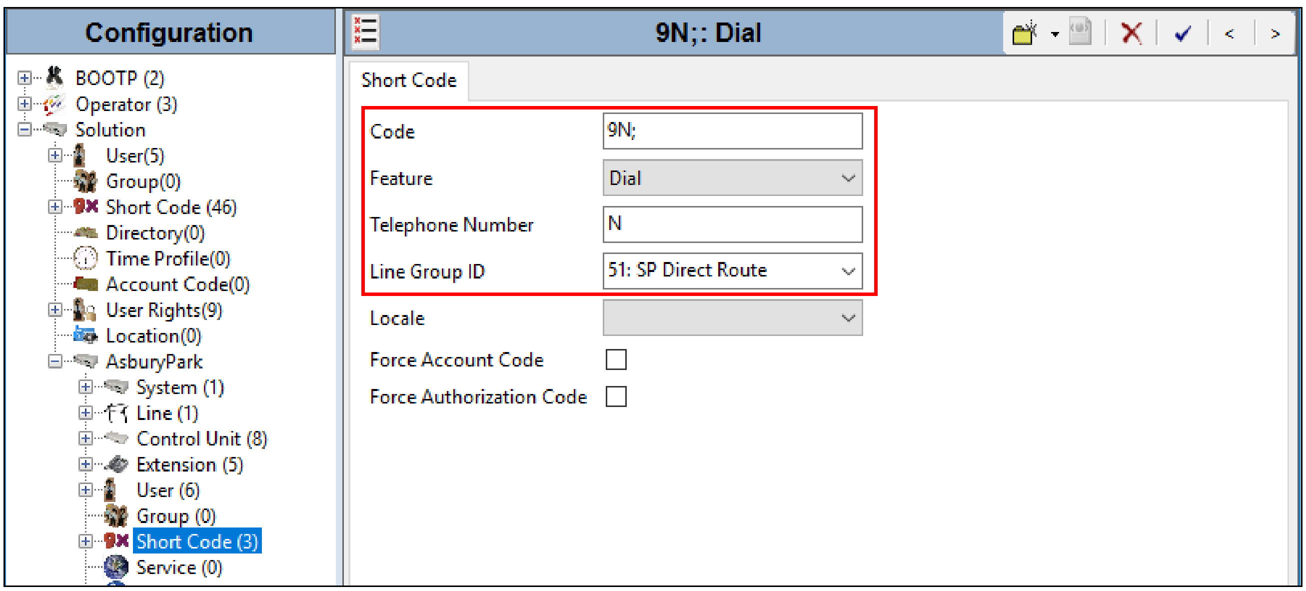
5.5. Short Code

Define a short code to route outbound calls to the SIP Line. To create a short code, right-click on **Short Code** in the Navigation Pane and select **New** (not shown). 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, followed by a semi-colon. The ***9N;*** short code, used for the compliance test, 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***. This field is used to construct the Request URI and the To header in the outgoing SIP INVITE message. The value ***N*** represents the number dialed by the user after removing the **9** prefix. This number is passed to the ARS route specified in the **Line Group Id** field.
- Set the **Line Group Id** to the ARS route to be used (**Section 5.8)**. This short code will use this route when placing outbound calls.



Optionally, add or edit a short code used to access the SIP Line anonymously. In the screen shown below, the short code ****67N;*** is illustrated. This short code is similar to the **9N;** short code except that the **Telephone Number** field begins with the letter ***W***, which means "withhold the outgoing calling line identification". In the case of the compliance test, when a user dialed *67 plus the destination number, Avaya IP Office would include the user's telephone number (DID number assigned to the user) in the **P-Asserted-Identity** (PAI) header, populate the URI user part with "anonymous" in the From and Contact headers, and include the **Privacy: id** header in the outbound INVITE message. Consequently, Clearfly would prevent presentation of the caller ID to the called PSTN destination.

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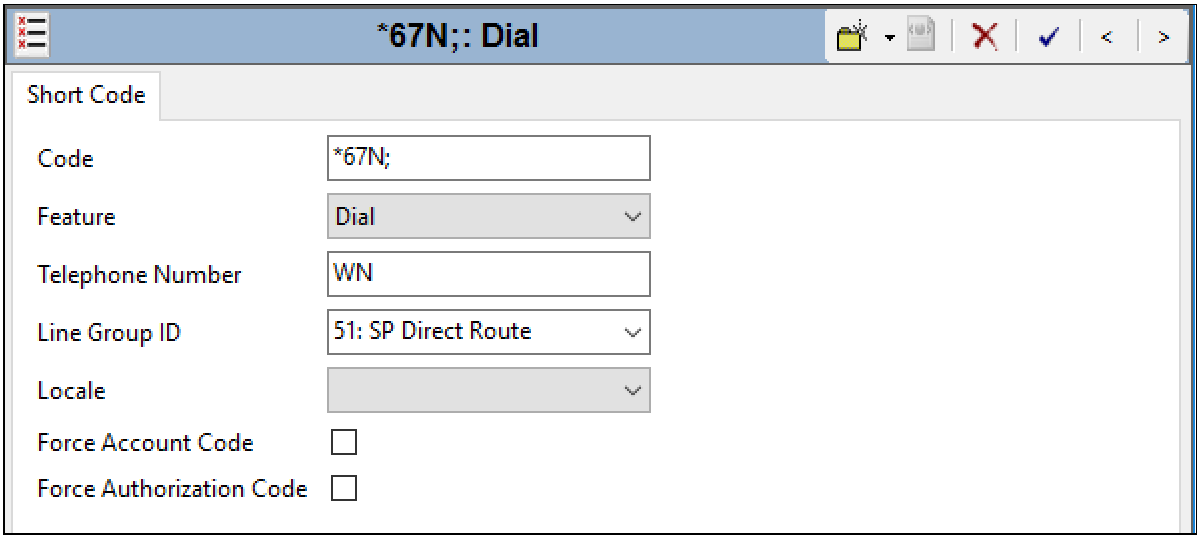

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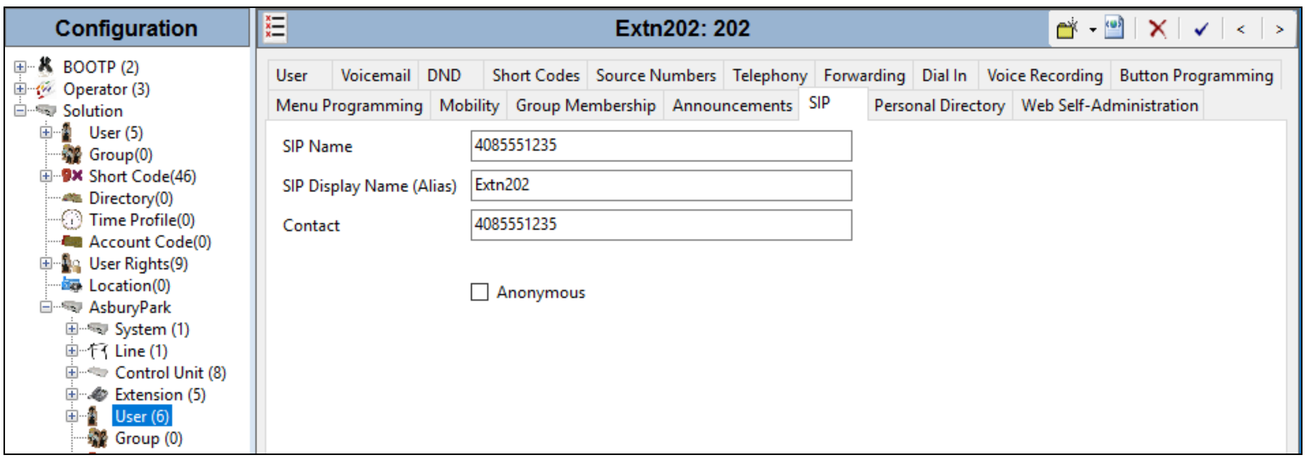
5.6. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP Line. 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 "Extn202" at extension 202. Select the **SIP** tab in the Details Pane. The **SIP Name** and **Contact** are set to one of the DID numbers provided by Clearfly. The **SIP Display Name (Alias)** can optionally be configured with a descriptive text string. The value entered for the **Contact** field will be used in the Contact header for outgoing SIP INVITEs to the service provider. The value entered for the **SIP Name** is used as the user part of the SIP URI in the From header for outgoing SIP INVITEs.

If outbound calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user information from the network (or alternatively use the ***67N;** short code as defined in **Section 5.5**).



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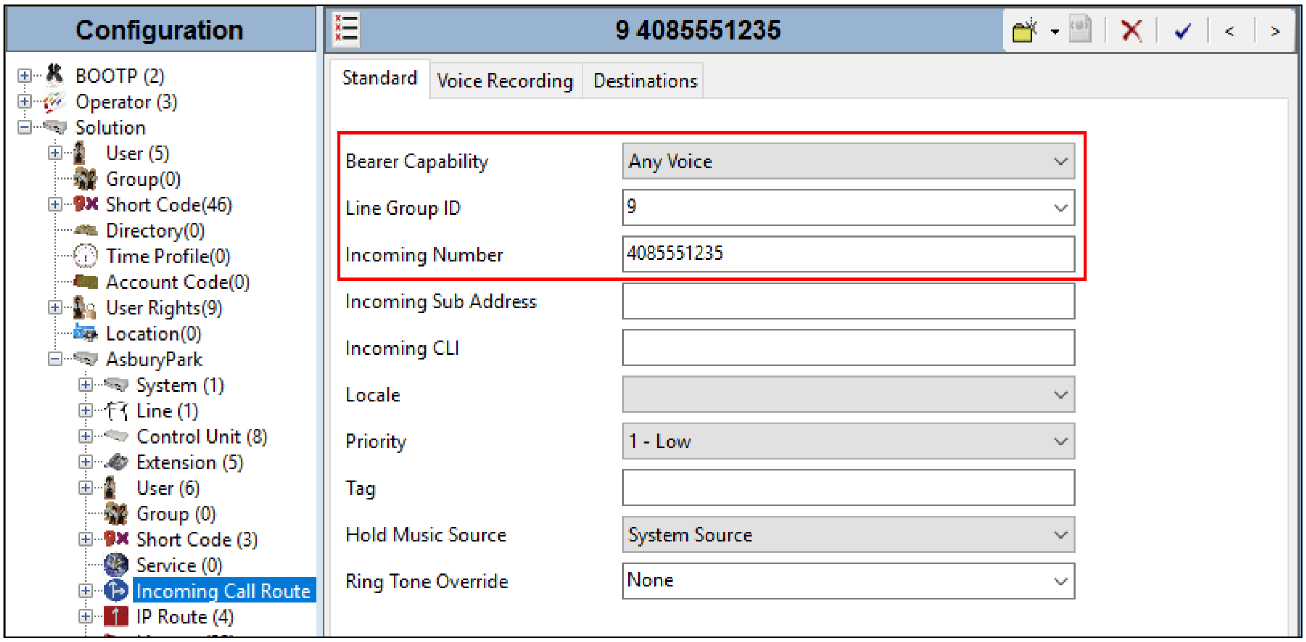
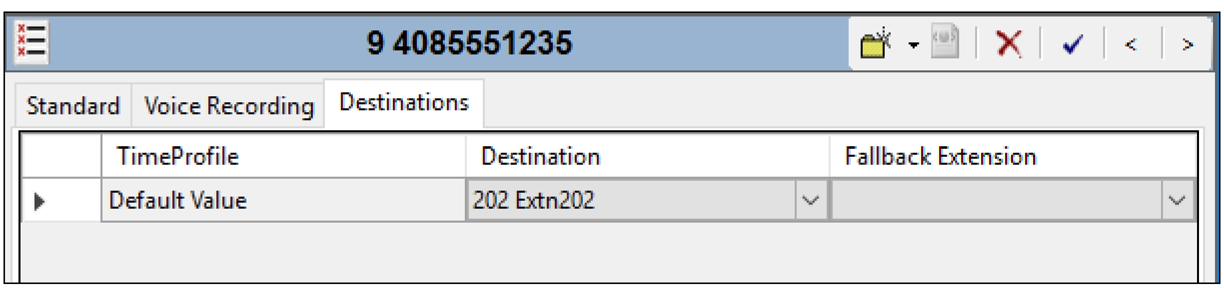
5.7. Incoming Call Route

An incoming call route maps an inbound DID number on a specific line to an internal destination. This procedure should be repeated for each DID number provided by the service provider. To create an incoming call route, right-click **Incoming Call Route** in the Navigation Pane and select **New** (not shown). On the **Standard** tab in the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capacity** to ***Any Voice***.
- Set the **Line Group Id** to the **Incoming Group** of the SIP Line defined in **Section 5.4.4**.
- Set the **Incoming Number** to the incoming number on which this route should match.



On the **Destinations** tab, select the destination from the pull-down list of the **Destination** field. In this example, incoming calls to 4085551235 on Incoming Group 9 are to be routed to the user "Extn202" at extension 202.



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5.8. ARS and Alternate Routing

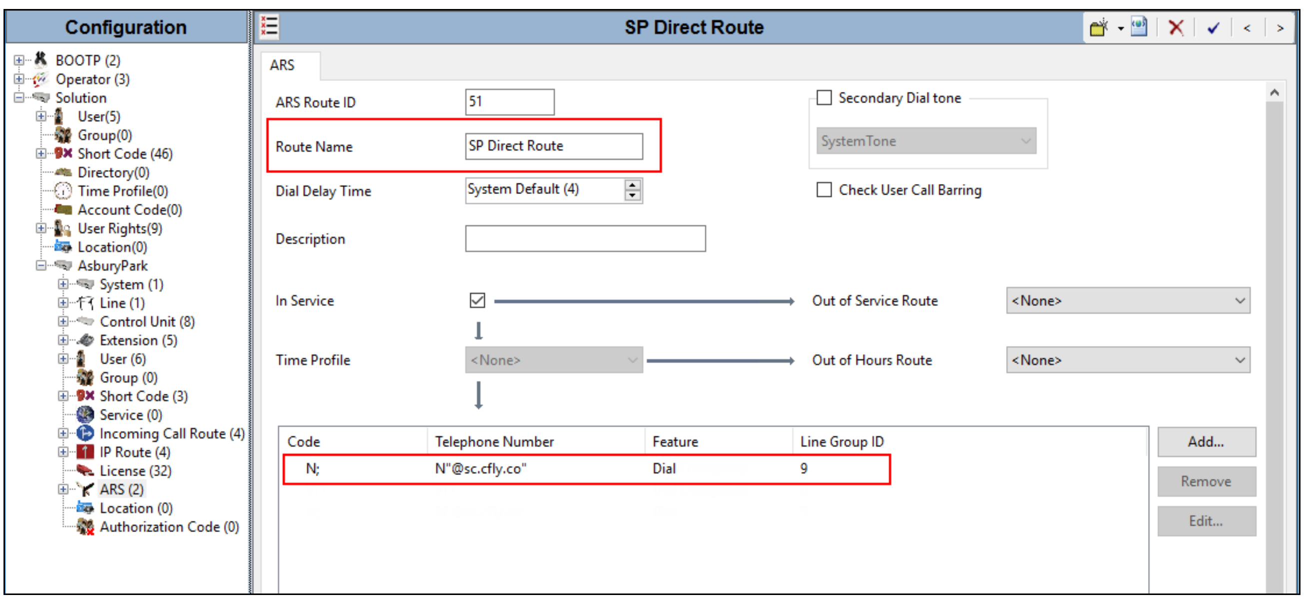
Alternate Route Selection (ARS) is used to route outbound traffic to the SIP line. To define a new ARS route, right-click **ARS** in the Navigation pane and select **New**. In the Details pane that appears, enter a name for the route in the **Route Name** field and a collection of matching patterns (similar to short codes) can be entered to route calls as shown below.

For the compliance test, one entry was created. The entry matches on any number **N**.

To create an entry, click the **Add** button and enter the following in the pop-up window (not shown).

- In the **Code** field, enter the pattern to match the number passed to ARS from the short code in **Section 5.5**. The value **N;** will match any number.
- For **Code N;**, set **Telephone Number** to **N"@sc.cfly.co"**. This field is used to construct the Request-URI and To headers in the outgoing SIP INVITE message. The value **N** represents the complete number passed to ARS. The domain **sc.cfly.co** is the SIP domain provided by Clearfly.
- Set **Feature** to **Dial**. This is the action that the entry will perform.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.4.4**. This entry will use this line group when placing the outbound call.

Click the **OK** button (not shown).



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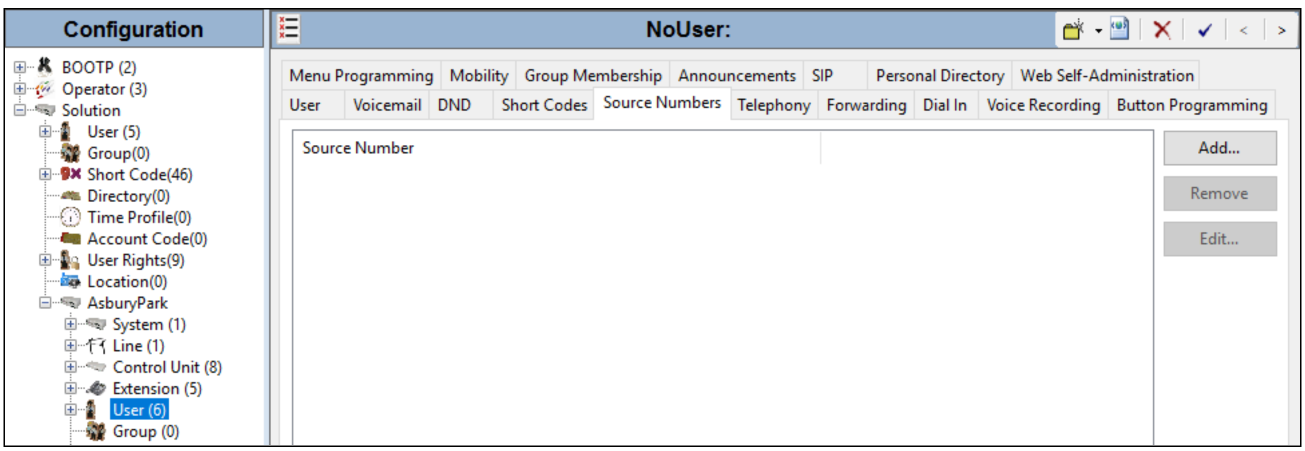
5.9. SIP Options

Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. By default, Avaya IP Office Release 9.1 sends out OPTIONS every 300 seconds. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.2.1** and the **SIP_OPTIONS_PERIOD** parameter (in minutes) that can be set on the **Source Number** tab of the **noUser** user. The OPTIONS period is determined in the following manner:

- To use the default value, set **Binding Refresh Time** to 300. OPTIONS will be sent at the 300 second frequency.
- To establish a period of less than 300 seconds, do not define the **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 300 seconds. The OPTIONS message period will be equal to the **Binding Refresh Time** setting.
- To establish a period greater than 300 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 300 seconds. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD** settings.

To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User → NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



At the bottom of the Details Pane, the **Source Number** field will appear. Enter ***SIP_OPTIONS_PERIOD=X***, where ***X*** is the desired value in minutes. Click **OK**.

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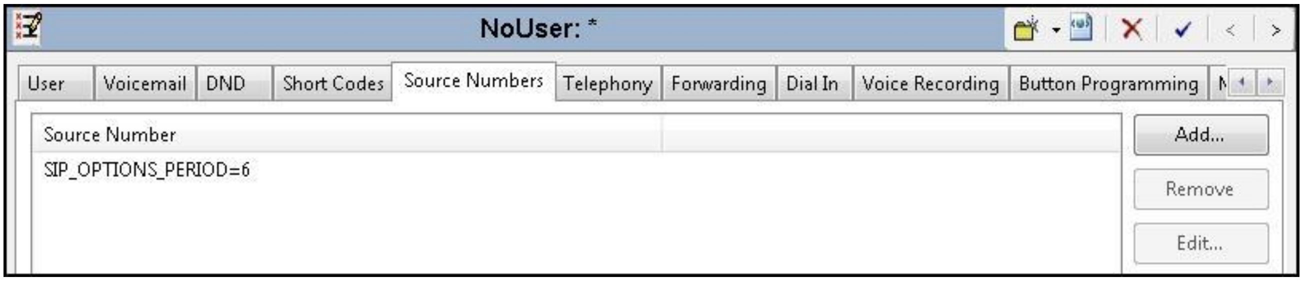
The **SIP_OPTIONS_PERIOD** parameter will appear in the list of Source Numbers as shown below. Click **OK** at the bottom of the screen (not shown).

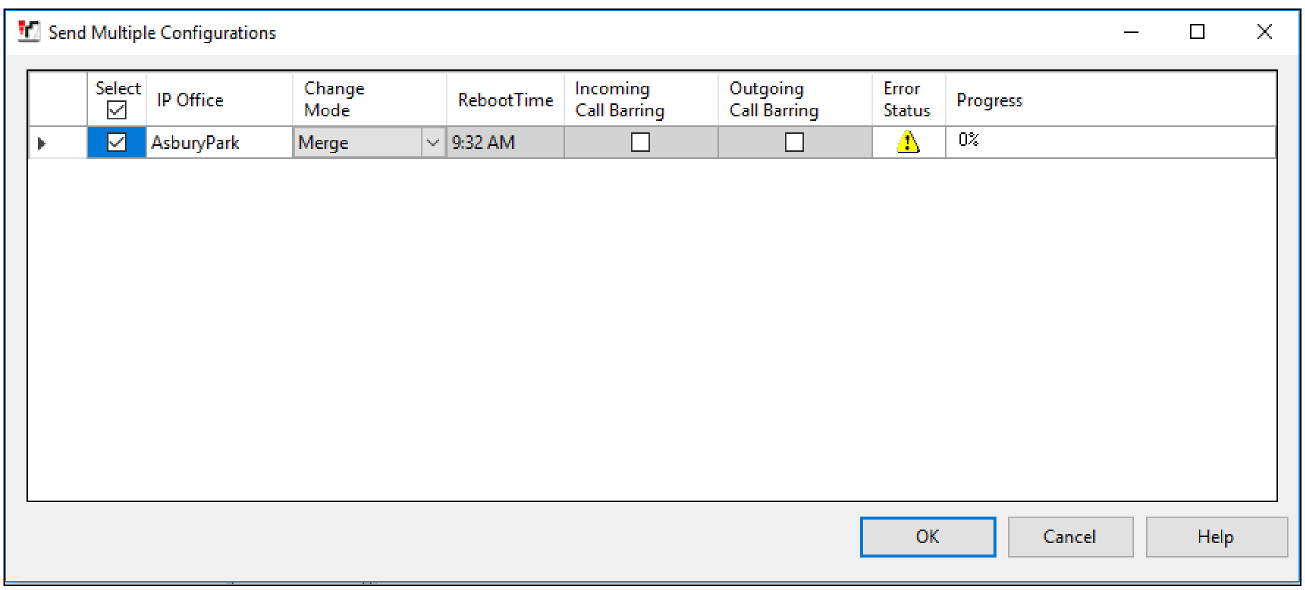


For the compliance test, an OPTIONS period of 2 minutes was desired. The **Binding Refresh Time** was set to ***120*** seconds in **Section 5.2.1**. Thus, there was no need to define **SIP_OPTIONS_PERIOD**.

5.10. 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 screen will appear, with either **Merge** or **Immediate** automatically selected under the **Change Mode**, based on the nature of the configuration changes made since the last save. Note that clicking **OK** may cause a system reboot or a service disruption. Click **OK** to proceed.



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6. Clearfly Communications SIP Trunking Configuration

Clearfly is responsible for the configuration of its SIP Trunking Service. Clearfly will provide the customer the necessary information to configure the Avaya IP Office including:

- SIP Domain of the Clearfly SIP Trunking Service
- Transport and port for the Clearfly SIP connection to the Avaya IP Office at the enterprise
- SIP Credentials
- DID numbers to assign to users at the enterprise
- Supported codecs and their preference order

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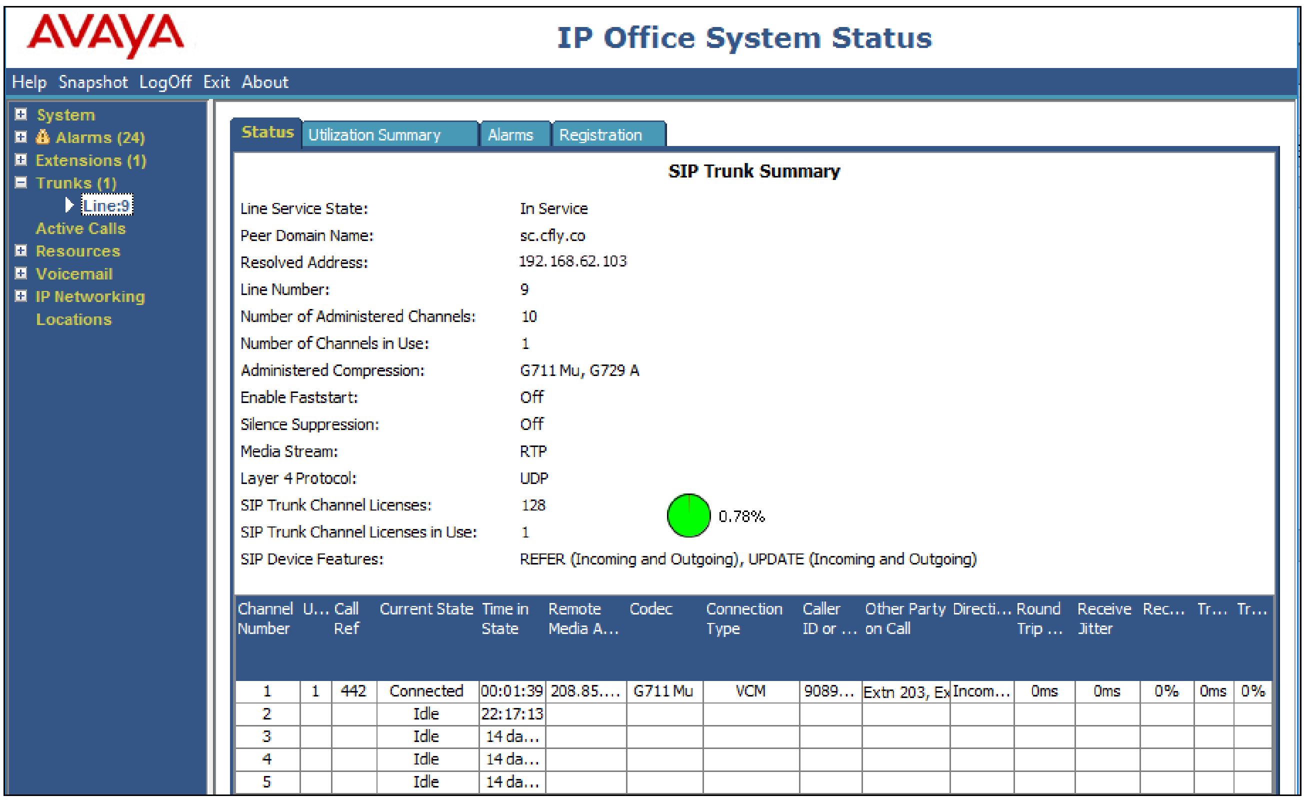
7. Verification Steps

This section provides verification steps that may be performed to verify the solution configuration.

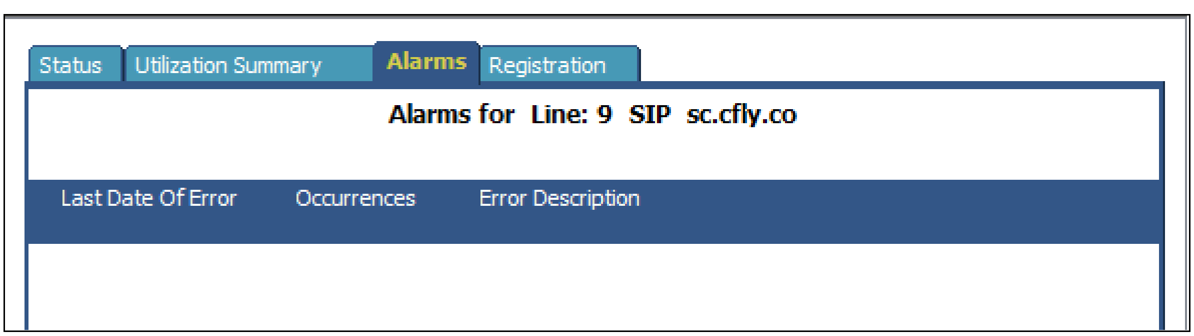
7.1. Avaya IP Office System Status

Use the Avaya IP Office System Status application to check the SIP Line channels state and alarms:

- Launch the application from **Start → Programs → IP Office → System Status** on the Avaya IP Office Manager PC. Select the SIP Line under **Trunks** from the left pane. In the **Status** tab in the right pane, verify the **Current State** is ***Idle*** for channels not taken by active calls; the state should be ***Connected*** for the channels engaged in active calls with the PSTN.



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



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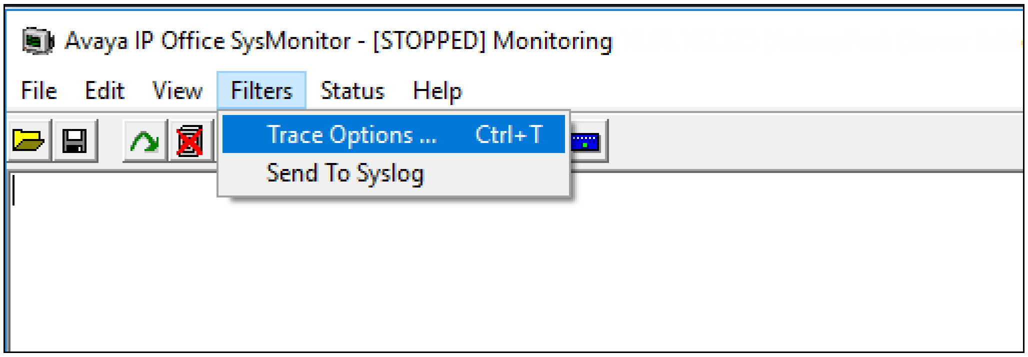
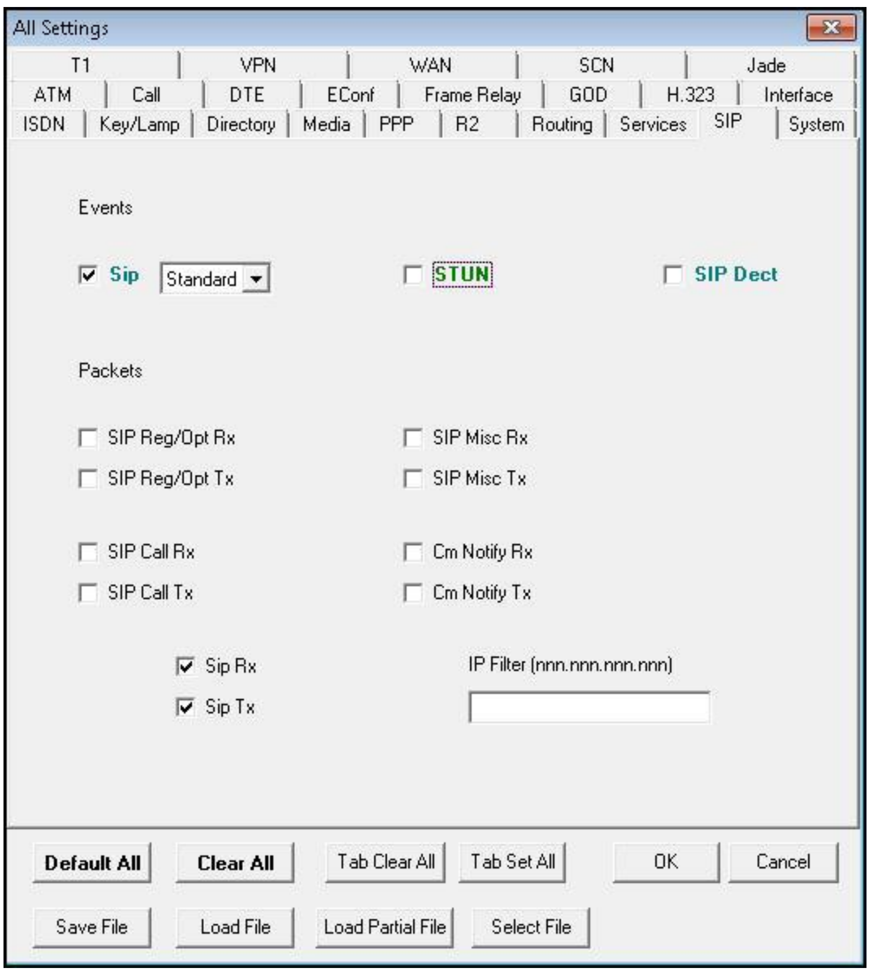
7.2. Avaya IP Office Monitor

The Avaya IP Office Monitor application can be used to monitor and troubleshoot Avaya IP Office. Avaya IP Office Monitor can be accessed from **Start → Programs → IP Office → Monitor** on the Avaya IP Office Manager PC. The application allows monitored information to be customized. To customize, select **Filters → Trace Options…** as shown below:



The following screen shows the **SIP** tab of trace options. In this example, **Standard Sip Events** and the **Sip Rx** and **Sip Tx** boxes are checked.



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8. Conclusion

The Clearfly Communications SIP Trunking Service with registration passed compliance testing with Avaya IP Office 9.1. These Application Notes describe the configuration necessary to connect Avaya IP Office 9.1 to Clearfly as shown in **Figure 1**. Test results and observations are noted in **Section 2.2**.

9. Additional References

[1] *IP Office™ Platform 9.1, Deploying Avaya IP Office™ Platform IP500 V2,* Document Number 15-601042, Issue 30g, January 2015.

[2] *Administering Avaya IP Office™ Platform with Manager*, Release 9.1, Issue 10.04, February 2015.

[3] *IP Office ™ Platform 9.1, Administering Avaya IP Office™ Platform Voicemail Pro*, Document Number 15-601063, Issue 10c, December 2014.

[4] *IP Office™ Platform 9.1, Using IP Office System Monitor*, Document Number 15-601019, Issue 06b, November 2014.

[5] *IP Office™ Platform 9.1, Using Avaya IP Office™ Platform System Status*, Document Number 15-601758, October 2014.

[6] *Using Avaya Communicator on IP Office*, Release 9.1, December 2014.

[7] RFC 3261 *SIP: Session Initiation Protocol*, http://www.ietf.org/.

[8] RFC 2833 *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, http://www.ietf.org/.

[9] *Application Notes for Configuring Clearfly Communications SIP Trunking Service with Avaya IP Office 9.1*, Issue 1.0, June 15, 2015.

Product documentation for Avaya products may be found at http://support.avaya.com or http://marketingtools.avaya.com/knowledgebase/ipoffice/general/rss2html.php?XMLFILE=manuals.xml&TEMPLATE=pdf_feed_template.html.

Product documentation for Clearfly SIP Trunking Service is available from Clearfly Communications.

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