**Avaya Solution & Interoperability Test Lab**

# Application Notes for Configuring Avaya IP Office Release 9.1 to support Cincinnati Bell Business SIP Trunking Service - Issue 1.0

## Abstract


These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking on an enterprise solution consisting of Avaya IP Office 9.1, to interoperate with Cincinnati Bell Business 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.

Cincinnati Bell Business SIP Trunking Service provides PSTN access via a SIP Trunk between the enterprise and Cincinnati Bell's network as an alternative to legacy analog or ISDN-PRI 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.

Cincinnati Bell 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.

# 1. Introduction

These Application Notes describe the steps necessary for configuring Session Initiation Protocol (SIP) Trunking service between Cincinnati Bell and an Avaya SIP-enabled enterprise solution.

In the configuration used during the testing, the Avaya SIP-enabled enterprise solution consists of Avaya IP Office 500v2 Release 9.1 (hereafter referred to as IP Office), Avaya Communicator for Windows and Avaya Deskphones, including SIP, H.323, digital, and analog.

The Cincinnati Bell Business SIP Trunking Service referenced within these Application Notes is designed for business customers. Customers using this service with the Avaya 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" or "Cincinnati Bell" 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 Solution & Interoperability Test Lab by connecting IP Office to the Cincinnati Bell Business 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.

## 2.1 Interoperability Compliance Testing

To verify the Cincinnati Bell Business SIP Trunking service offering with Avaya IP Office, the following features and functionalities were exercised during the compliance testing:

- Response to SIP OPTIONS queries.
- Incoming PSTN calls to various Avaya endpoints, including SIP, H.323, digital and analog at the enterprise. All incoming calls from the PSTN were routed to the enterprise across the SIP Trunk from the service provider networks.
- Outgoing PSTN calls from Avaya endpoints including SIP, H.323, digital and analog telephone at the enterprise. All outgoing calls to the PSTN were routed from the enterprise across the SIP trunk to the service provider networks.
- Incoming and outgoing PSTN calls to/from Avaya Communicator for Windows.
- Dialing plans including long distance, outbound toll-free, etc.
- Caller ID presentation and Caller ID restriction.
- 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 coverage to voicemail off).
- Proper response to busy endpoints.
- Proper response/error treatment when dialing invalid PSTN numbers.
- Codec G.711MU and G.729(a) (Cincinnati Bell supported audio codec).
- Proper response to no matching codecs.
- T.38 fax.
- G.711 fax Pass-through.
- Proper early media transmissions.
- Voicemail and DTMF tone support using RFC 2833 (leaving and retrieving voice mail messages, etc.).
- Outbound Toll-Free calls, interacting with IVR (Interactive Voice Response systems).
- 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.

Items not supported or not tested included the following:

- Inbound toll-free calls and 911 emergency calls are supported but were not tested as part of the compliance test.
- Notification of intermediate call states (via NOTIFY messages) for a call that is redirected with a REFER message.

## 2.2 Test Results

Interoperability testing with Cincinnati Bell Business SIP Trunking service was successfully completed with the exception of observations/limitations described below:

- **No ring back tone on PSTN stations after Blind Transfers to the PSTN**: Ring back tone was not heard (only silence) on PSTN stations after calls from the PSTN to IP Office H.323 stations are Blind Transferred back out to another PSTN station (external transfer) and while the PSTN station the call was transferred to was ringing (Scenario: PSTN_Station_1 →IPO_H.323_Station →Blind Transfer → PSTN_ Station _2). This issue is only seen on IP Office H.323 endpoints, this issue is not seen on IP Office SIP endpoints. This issue is under investigation.
- **No matching codec on outbound calls**: If an unsupported audio codec is received by Cincinnati Bell on the SIP Trunk (e.g., 711A), Cincinnati Bell will respond with “503 Service Unavailable” instead of “488 Not Acceptable Here”, the user will hear a series of tones. This issue does not have any user impact, and should not be seen since the codecs will be matched during the installation, it is listed here simply as an observation.
- **Operator –assisted calls**: Operator-assisted calls (0 + 10 digits) are routed the same as direct dialed calls (1 + 10 digits).

## 2.3 Support

For support on Cincinnati Bell Business SIP Trunking service visit the corporate Web page at: https://www.cincinnatibell.com/customer_support/

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.

# 3. Reference Configuration

**Figure 1** illustrates the test configuration used. The test configuration simulates an enterprise site with an Avaya SIP-enabled enterprise solution connected to the Cincinnati Bell Business SIP Trunking service through the public Internet.

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

- Avaya IP Office 500v2.
- Avaya IP Office Voicemail Pro.
- Avaya 96x0 Series H.323 IP Deskphones.
- Avaya 96x1 Series H.323 IP Deskphones.
- Avaya 1100 Series SIP IP Deskphones.
- Avaya Communicator for Windows.
- Avaya 1408 Digital Telephones.
- Avaya 9508 Digital Telephones.

Located at the simulated enterprise site is Avaya IP Office 500v2 with analog and digital extension expansion modules, as well as a VCM64 (Voice Compression Module) for supporting VoIP codecs. The **LAN1** port of Avaya IP Office is connected to the enterprise LAN (private IP network) while the LAN2 port is connected to the public IP network.

The transport protocol between IP Office and Cincinnati Bell's network, across the public Internet, is SIP over UDP.

For inbound calls, the calls flowed from Cincinnati Bell's network, across the public Internet, to IP Office.

Outbound calls to the PSTN were first processed by IP Office. Once IP Office selected the proper SIP trunk; the call was routed, across the public Internet, to Cincinnati Bell's network.

For the purposes of the compliance test, users dialed a short code of 9 + N digits to make calls across the SIP trunk to Cincinnati Bell (refer to **Section 5.8**). The short code 9 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to the network. Since Cincinnati Bell is a U.S. based company, a country member of the North American Numbering Plan (NANP), the users dialed 7 or 10 digits for local calls, and 11 (1 + 10) digits for other calls between the NANP.

In an actual customer configuration, the enterprise site may also include additional network components between Cincinnati Bell and the enterprise. 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 the enterprise must be allowed to pass through these devices.

For confidentiality and privacy purposes, actual public IP addresses and DID numbers used during the compliance test have been replaced with fictitious IP addresses and DID numbers throughout the Application Notes.

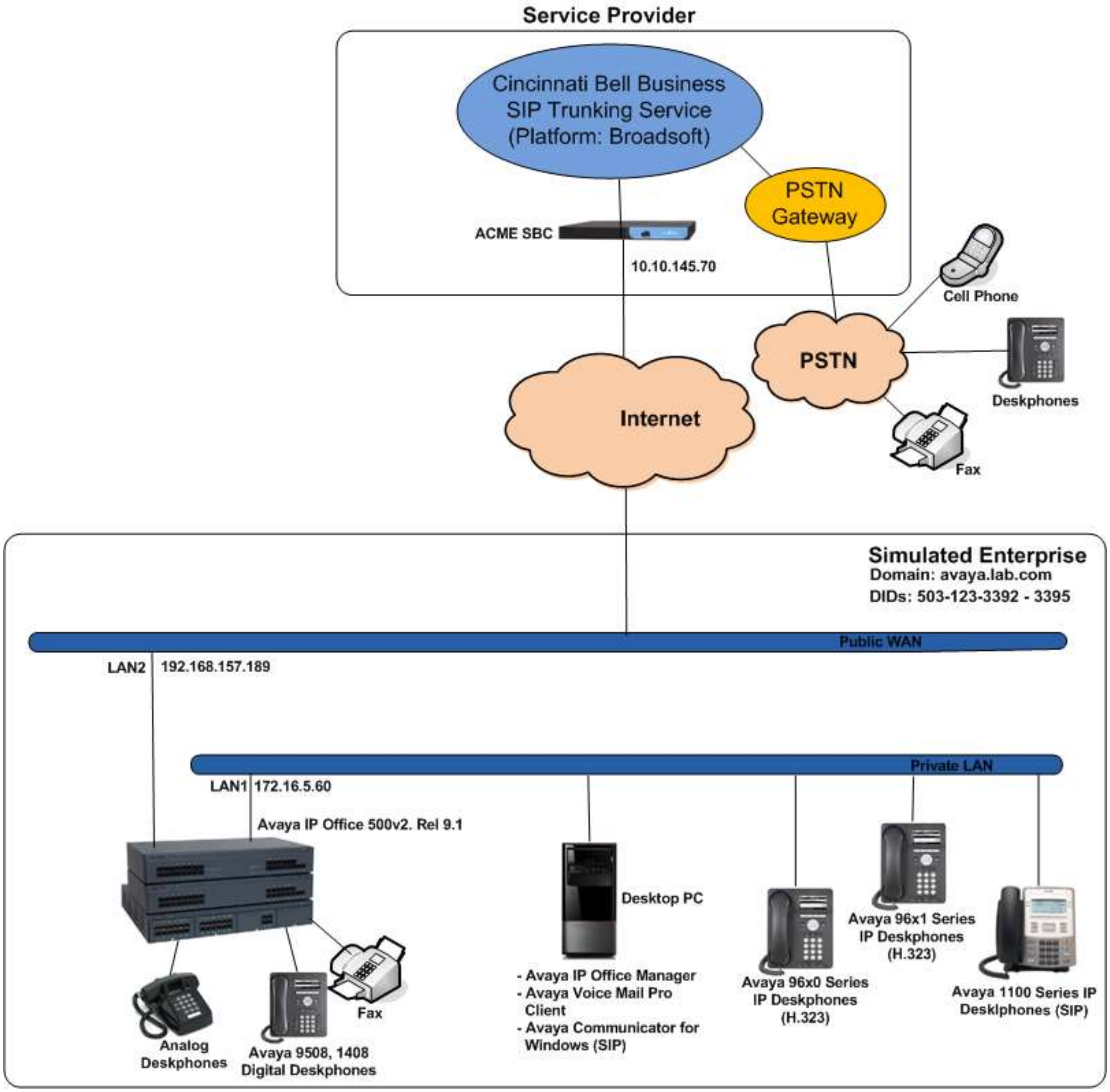



**Figure 1: Avaya Interoperability Test Lab Configuration**.

# 4. Equipment and Software Validated

The following equipment and software/firmware were used for the compliance testing.

| Equipment/Software | Release/Version |
|---|---|
| **Avaya** | |
| Avaya IP Office 500v2 | 9.1.6.0 Build 153 |
| Avaya IP Office DIG DCPx16 V2 | 9.1.6.0 Build 153 |
| Avaya IP Office Manager | 9.1.6.0 Build 153 |
| Avaya Voicemail Pro Client | 9.1.6.0 Build 2 |
| Avaya 96x0 IP Deskphones (H.323) | Avaya one-X® Deskphone Edition S3.230A |
| Avaya 96x1 Series IP Deskphones (H.323) | 6.6029 |
| Avaya 1120E IP Deskphones (SIP) | SIP1120e Ver. 04.04.18.00 |
| Avaya Communicator for Windows | 2.0.3.40 |
| Avaya Digital Deskphones 1408 | 40.0 |
| Avaya Digital Deskphones 9508 | 0.55 |
| Lucent Analog Phone | -- |
| **Cincinnati Bell** | |
| Broadworks | R20 |
| Acme Packet 6300 Series SBC | ScZ7.2.0 |

**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.

# 5. Configure IP Office

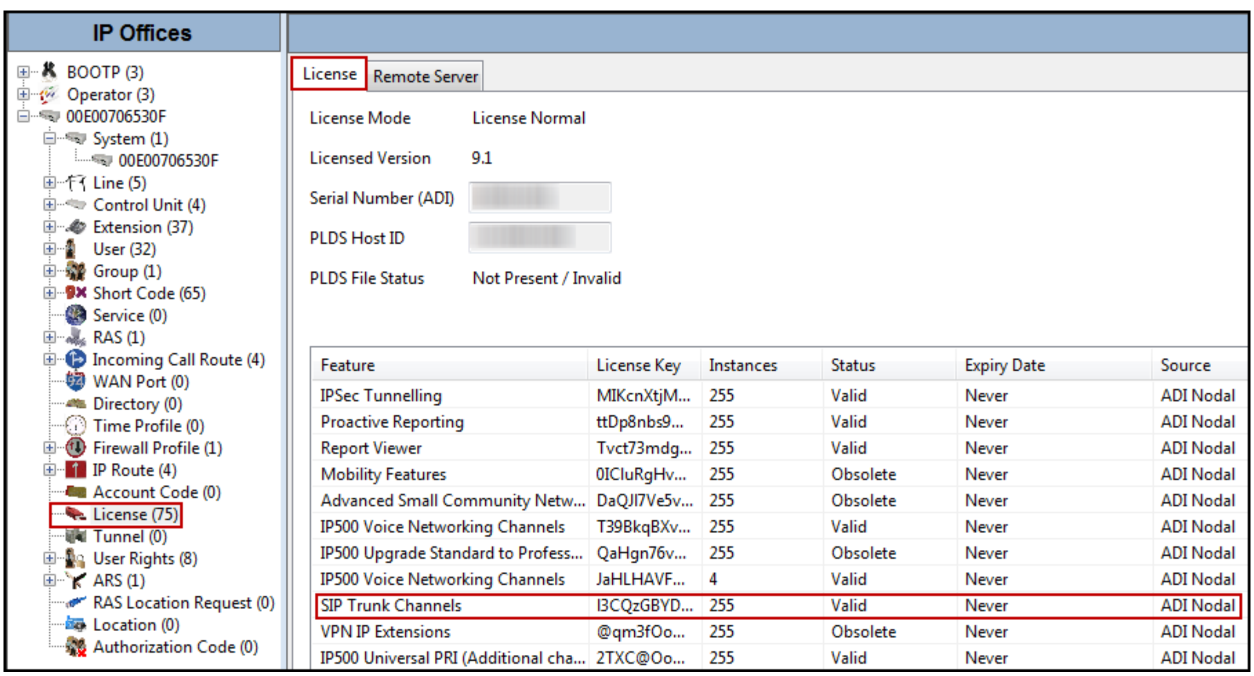
This section describes the IP Office configuration required to interwork with Cincinnati Bell Business SIP Trunking service. IP Office is configured through Avaya IP Office Manager (IP Office Manager) which is a PC application. On the PC, select **Start → Programs → IP Office → Manager** to launch IP Office Manager. Navigate to **File → Open Configuration**, select the proper IP Office from the pop-up window, and log in with the appropriate credentials. A management window will appear as shown in the next sections. The appearance of IP Office Manager can be customized using the **View** menu (not shown). In the screenshots presented in this section, the **View** menu was configured to show the **Navigation Pane** on the left side and the **Details Pane** on the right side. These panes will be referenced throughout these Application Notes.

These Application Notes assume the basic installation and configuration of IP Office have already been completed and are not discussed here. For further information on IP Office, please consult References in **Section 9**.

## 5.1 Licensing

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

To verify that there is a SIP Trunk Channels License with sufficient capacity; click **License**, then from the license tab, locate **SIP Trunk Channels**. Confirm that there is a valid license with sufficient "Instances" (trunk channels) in the Details pane. Note that the full License Keys in the screen below is not shown for security purposes.



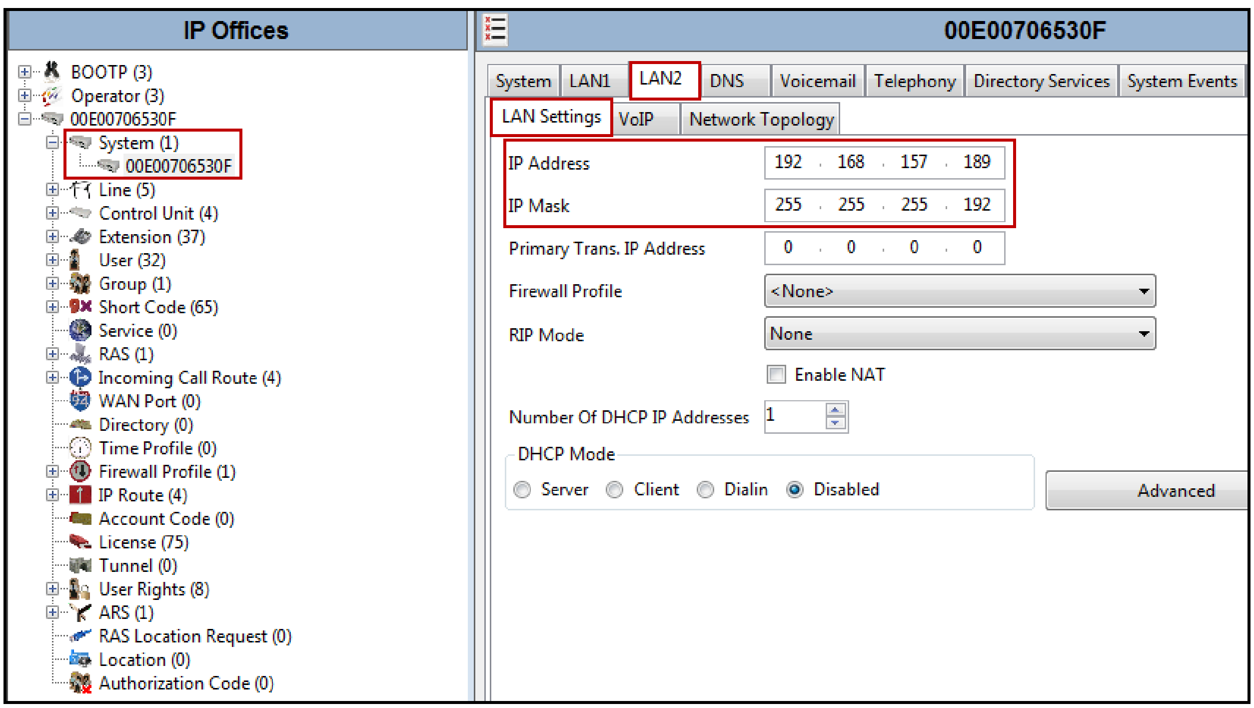
## 5.2 System

Configure the necessary system settings. In an Avaya IP Office the LAN2 tab settings correspond to the Avaya IP Office WAN port (public network side) and the LAN1 tab settings correspond to the LAN port (private network side). For the compliance test, the **LAN1** interface was used to connect Avaya IP Office to the enterprise private network (LAN); **LAN2** was used to connect to the public network.

### 5.2.1 System – LAN2 Tab

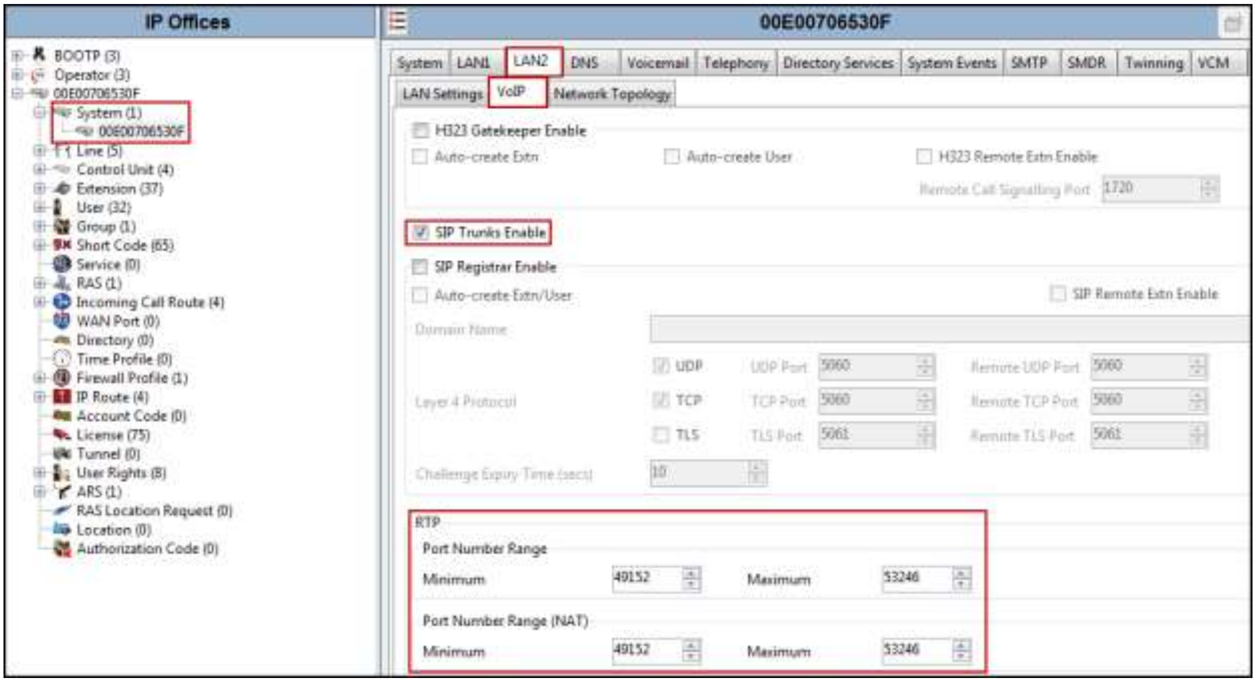
In the sample configuration, the IP Office WAN port was used to connect to Cincinnati Bell's network across the public Internet. The LAN2 settings correspond to the WAN port on the Avaya IP Office 500v2. To access the LAN2 settings, first navigate to **System** → <*Name*>, where <*Name*> is the system 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. The **LAN2** settings for the compliance testing were configured with following parameters:

- Set the **IP Address** field to the public IP address assigned to the IP Office WAN port, e.g. **192.168.157.189**.
- Set the **IP Mask** field to the subnet mask of the public network, e.g. **255.255.255.192**.
- All other parameters should be set according to customer requirements.
- Click **OK** to commit (not shown).



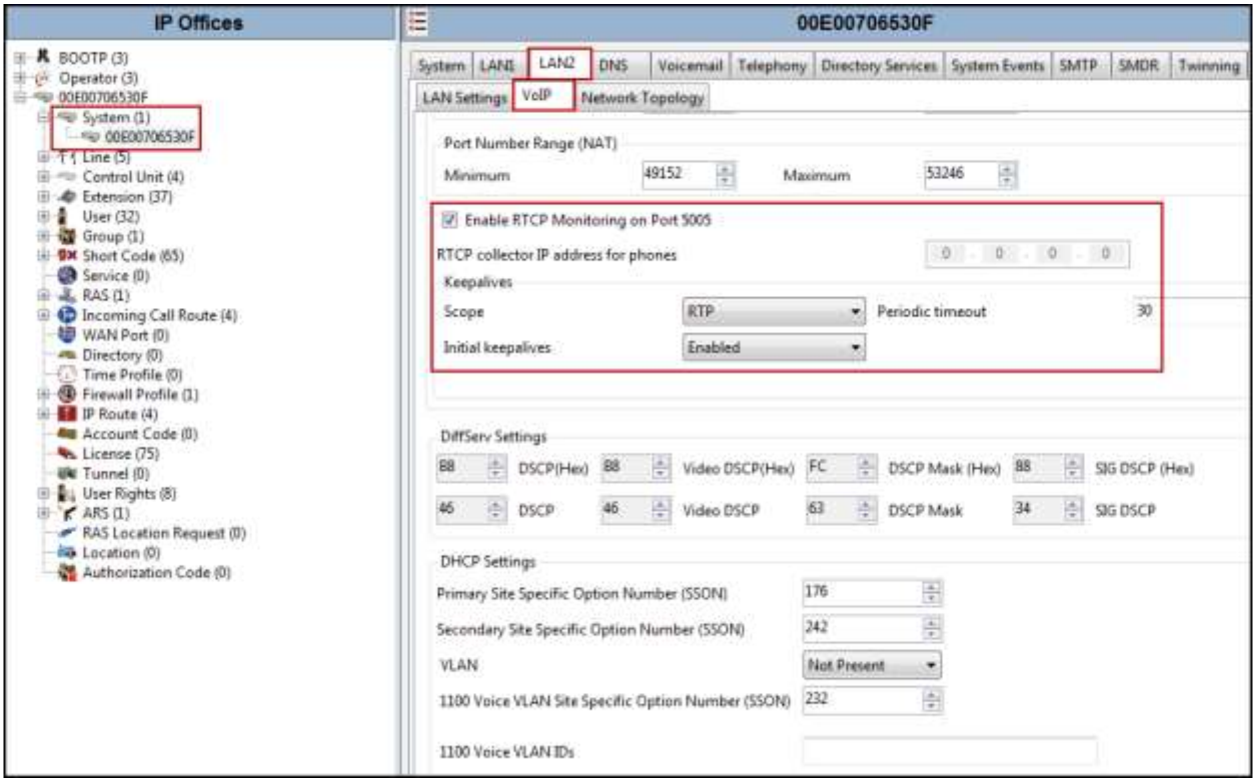
On the **VoIP** tab in the Details Pane, configure the following parameters:

- 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.



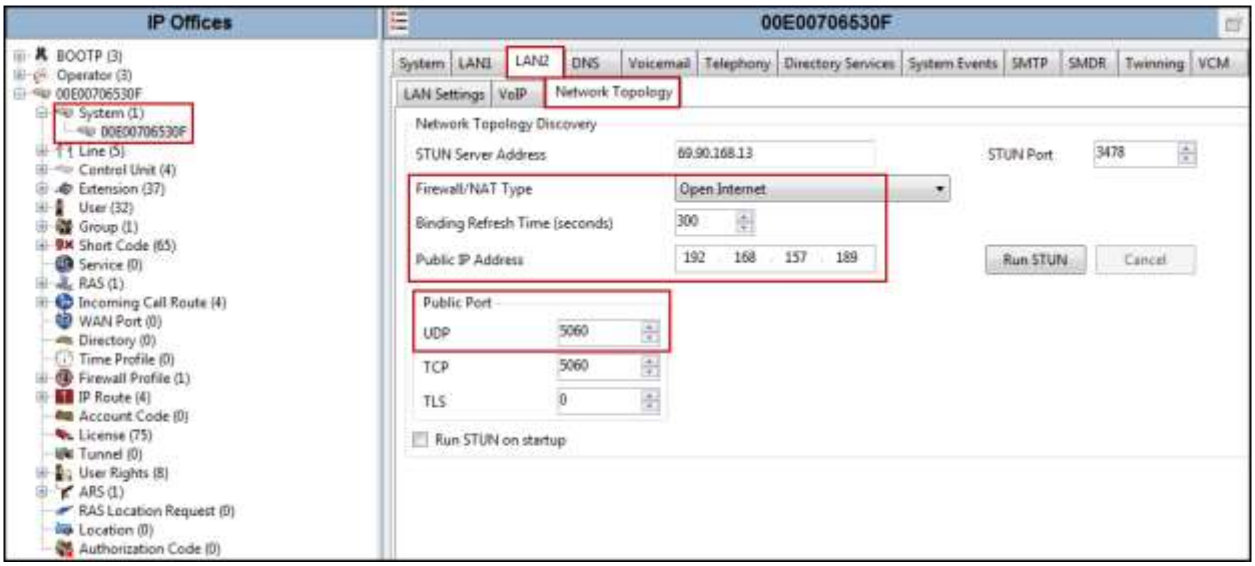
Scroll down the page.

- 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).



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 Cincinnati Bell, 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.
- 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).

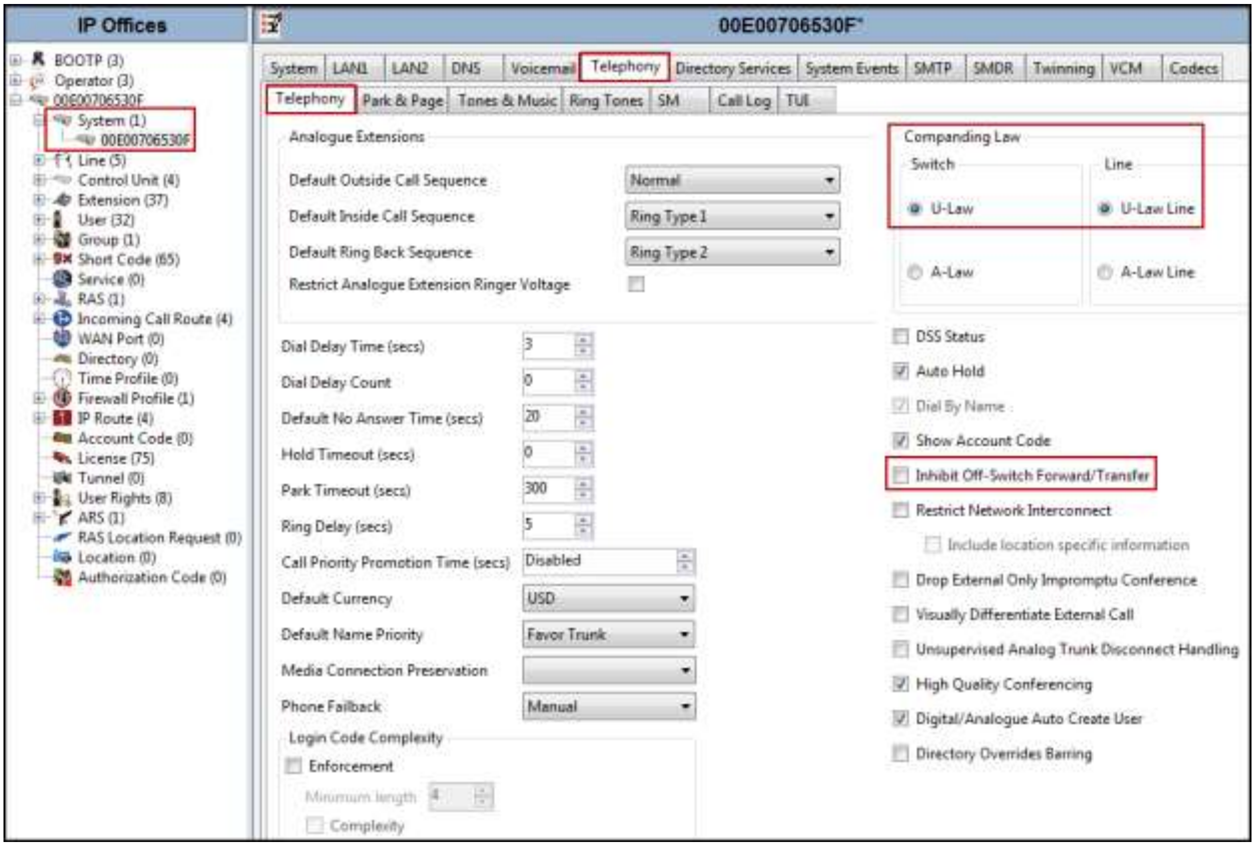



**Note**: In the compliance test, the LAN1 interface was used to connect IP Office to the enterprise site IP network (private network). The LAN1 interface configuration is not directly relevant to the interface with the Cincinnati Bell Business SIP Trunking Service, and therefore is not described in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

## 5.2.2 System - Telephony Tab

Navigate to the **Telephony → Telephony** Tab in the Details Pane, configure the following parameters:

- Choose the **Companding Law** typical for the enterprise location, **U-Law** was used.
- Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfers to the PSTN via the SIP trunk to the service provider.
- All other parameters should be set according to customer requirements.
- Click **OK** to commit (not shown).



### 5.2.3 System - Twinning Tab

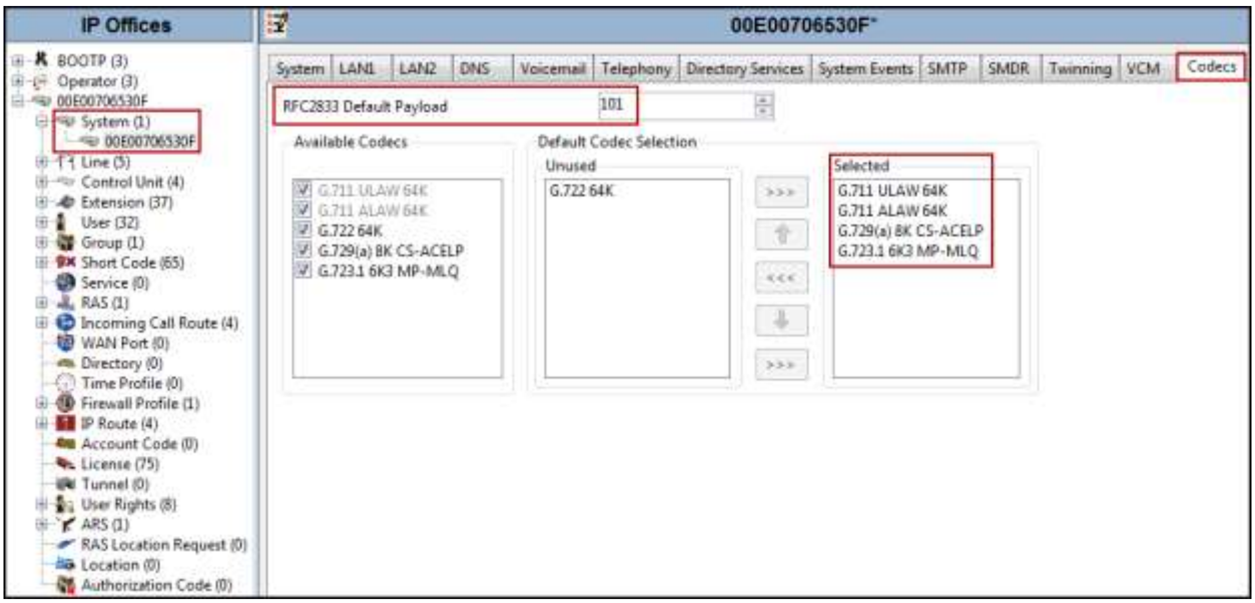
Navigate to the **Twinning** tab on the Details Pane, configure the following parameters:

- Uncheck the **Send original calling party information for Mobile Twinning** box. This will allow the Caller ID for Twinning to be controlled by the setting on the SIP Line (**Section 5.4**). This setting also impacts the Caller ID for call forwarding.
- Click **OK** to commit (not shown).

## 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:

- The **RFC2833 Default Payload** field allows for the manual configuration of the payload type used on SIP calls that are initiated by the IP Office. The default value **101** was used.
- 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 lines and phones (SIP and H.323) will use the system default codec selection shown here, unless configured otherwise for a specific line or extension. The example below shows the codecs used for IP phones (SIP and H.323), the system's default codecs and order was used.
- Click **OK** to commit (not shown).



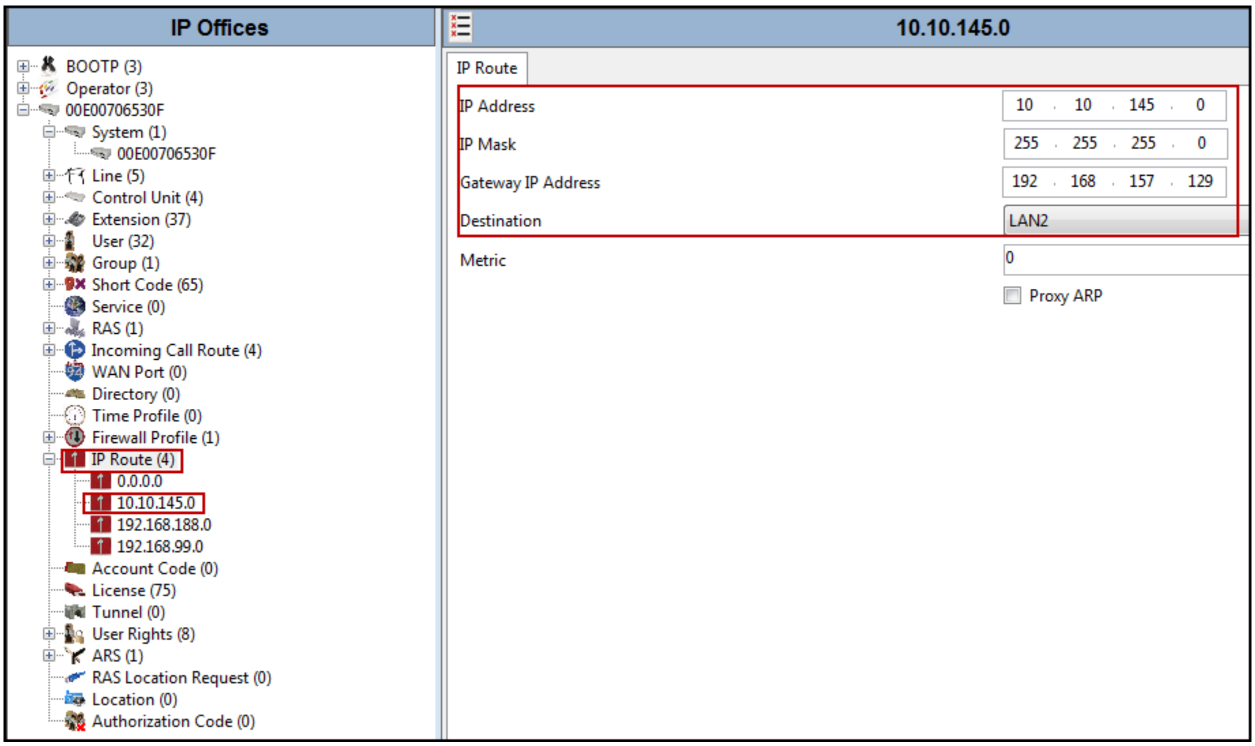

**Note**: The codec selections defined under this section (System – Codecs Tab) are the codecs selected for the IP phones/extensions. The codec selections defined under **Section 5.4.6** (SIP Line – VoIP tab) are the codecs selected for the SIP Line (Trunk).

## 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 Cincinnati Bell'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 **10.10.145.0**.
- Set the **IP Mask** to **255.255.255.0**.
- Set **Gateway IP Address** to the IP address of the default router for the public network where IP Office is connected.
- Set **Destination** to **LAN2** from the drop-down list.
- Click the **OK** to commit (not shown).



## 5.4 SIP Line

A SIP Line is needed to establish the SIP connection between IP Office and Cincinnati Bell Business 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 (if used).
- 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 **5.4.8**.

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 **5.4.8**.

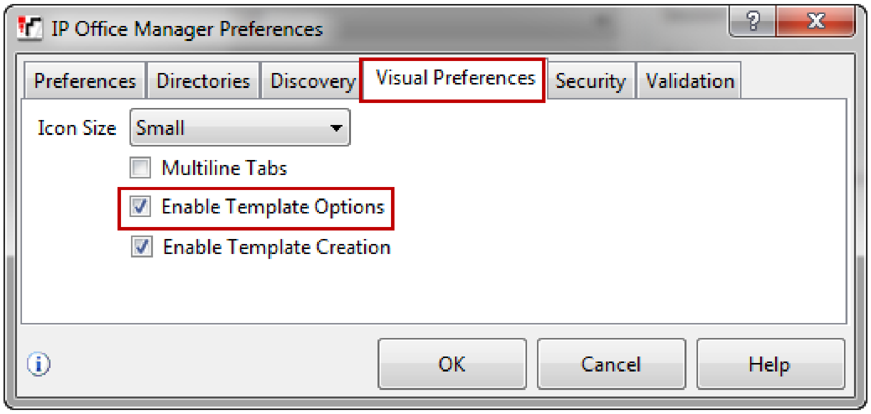
### 5.4.1 Importing a SIP Line Template

**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.

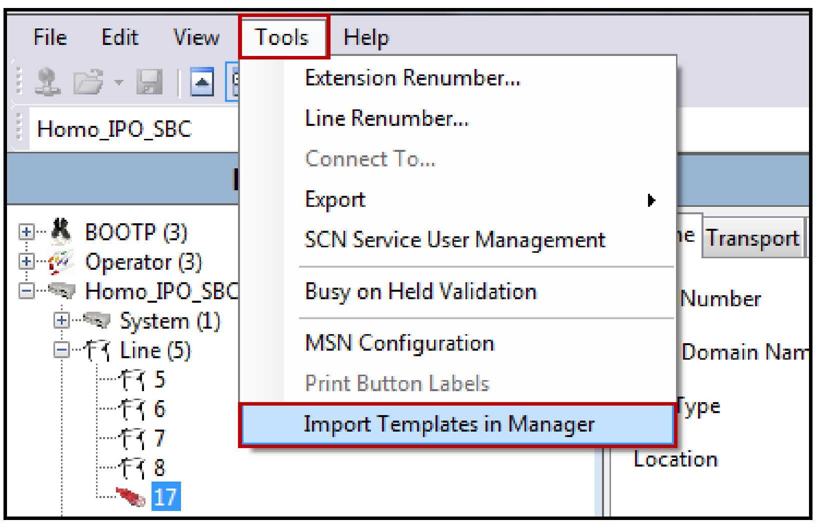
1. Copy a previously created template file to a location (e.g., C:*Temp*) on the same computer where IP Office Manager is installed. By default, the template file name will have the format **AF_<*user supplied text*>_SIPTrunk.xml**, where the **<*user supplied text*>** portion is entered during template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

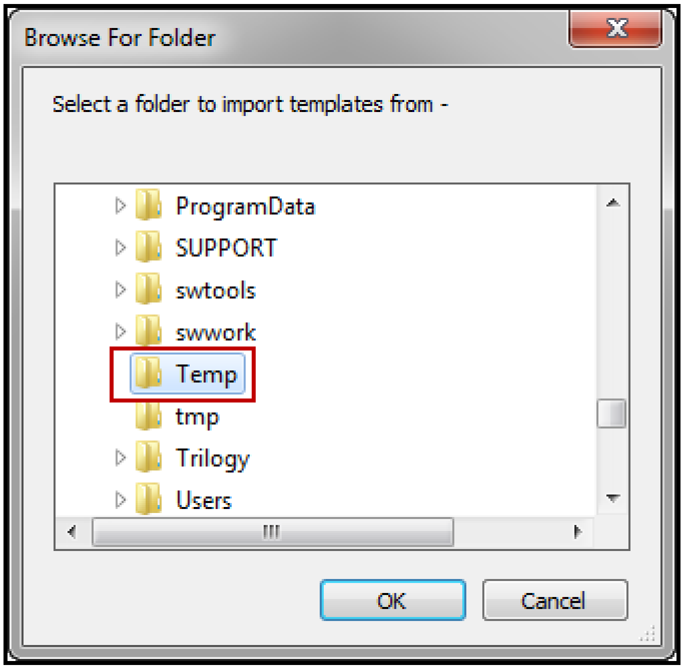
2. Verify that Template Options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the IP Office Manager Preferences window that appears, select the **Visual Preferences** tab. Check the box next to **Enable Template Options**. Click **OK**.




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

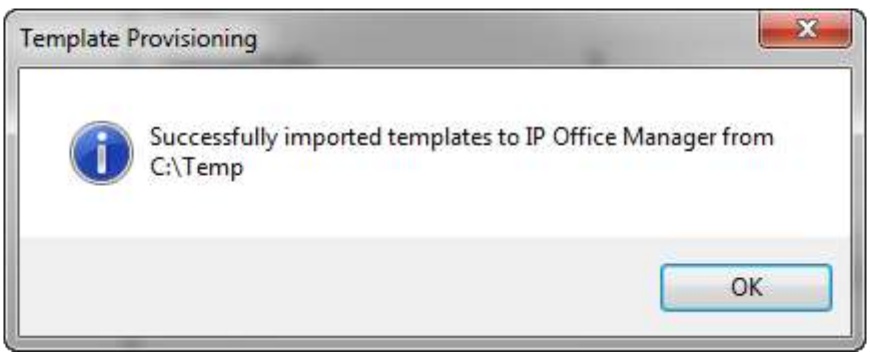


4. A folder browser will open. Select the directory used in **step 1** to store the template(s) (e.g., C:*Temp*).

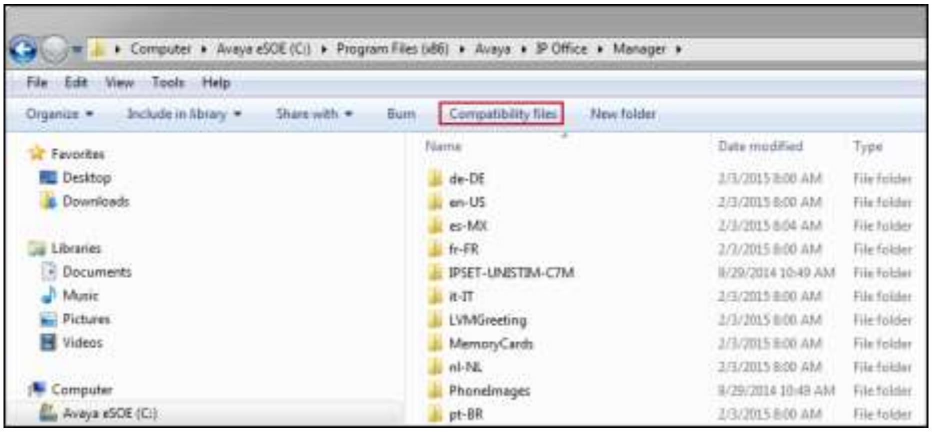


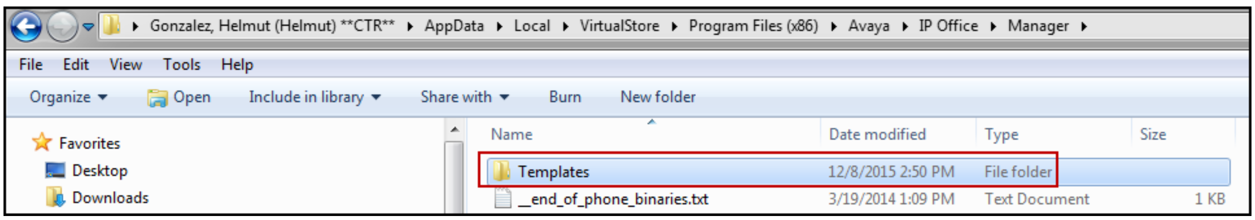

In the reference configuration, template files **AF_Cincinnati Bell IPO 9.1_SIPTrunk.xml** was imported. The template files are automatically copied into the IP Office default template location, **C:\Program Files\Avaya\IP Office\Manager\Templates**.

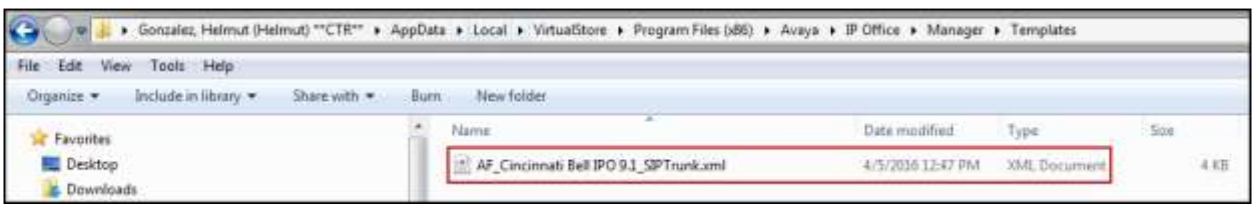
5. After the import is complete, a final import status pop-up window will open stating success or failure. Click **OK**.



**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.

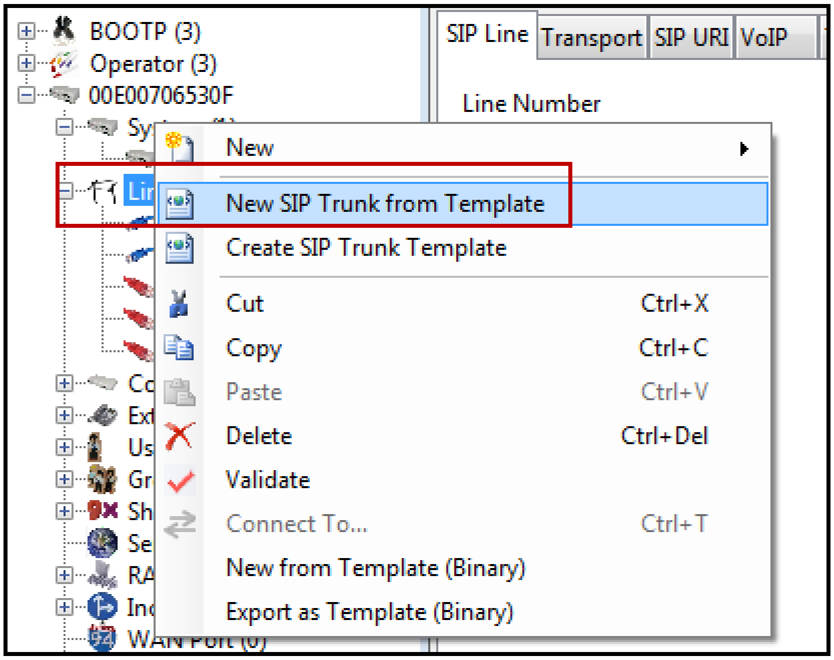


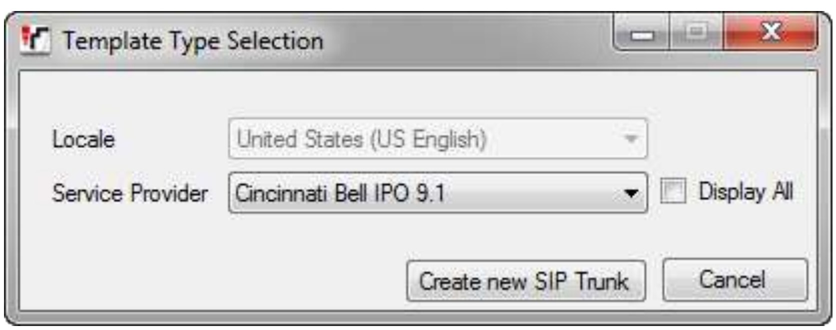
### 5.4.2 Creating a SIP Trunk from an XML Template

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

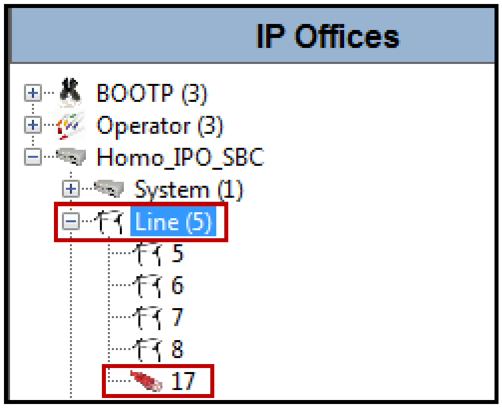



2. In the subsequent **Template Type Selection** pop-up window, from the **Service Provider** pull-down menu, select the XML template name from **Section 5.4.1**. Click **Create new SIP Trunk**.

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



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

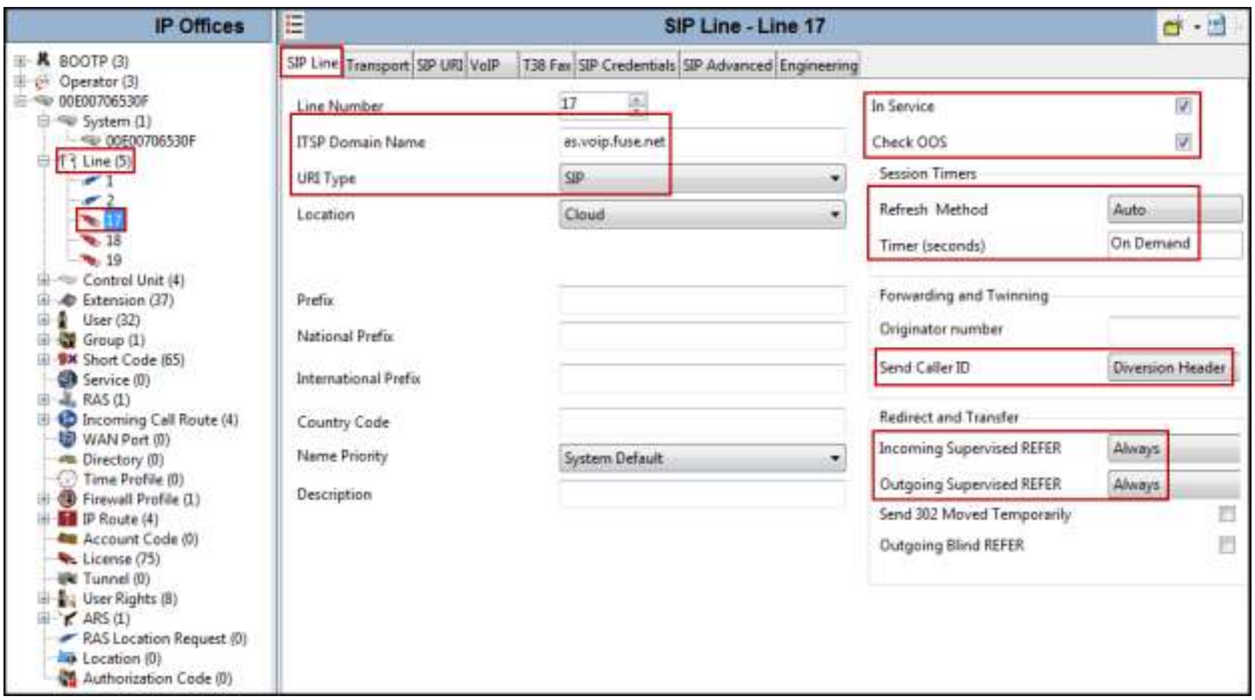



It is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.4.3** to **5.4.8**.

### 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:

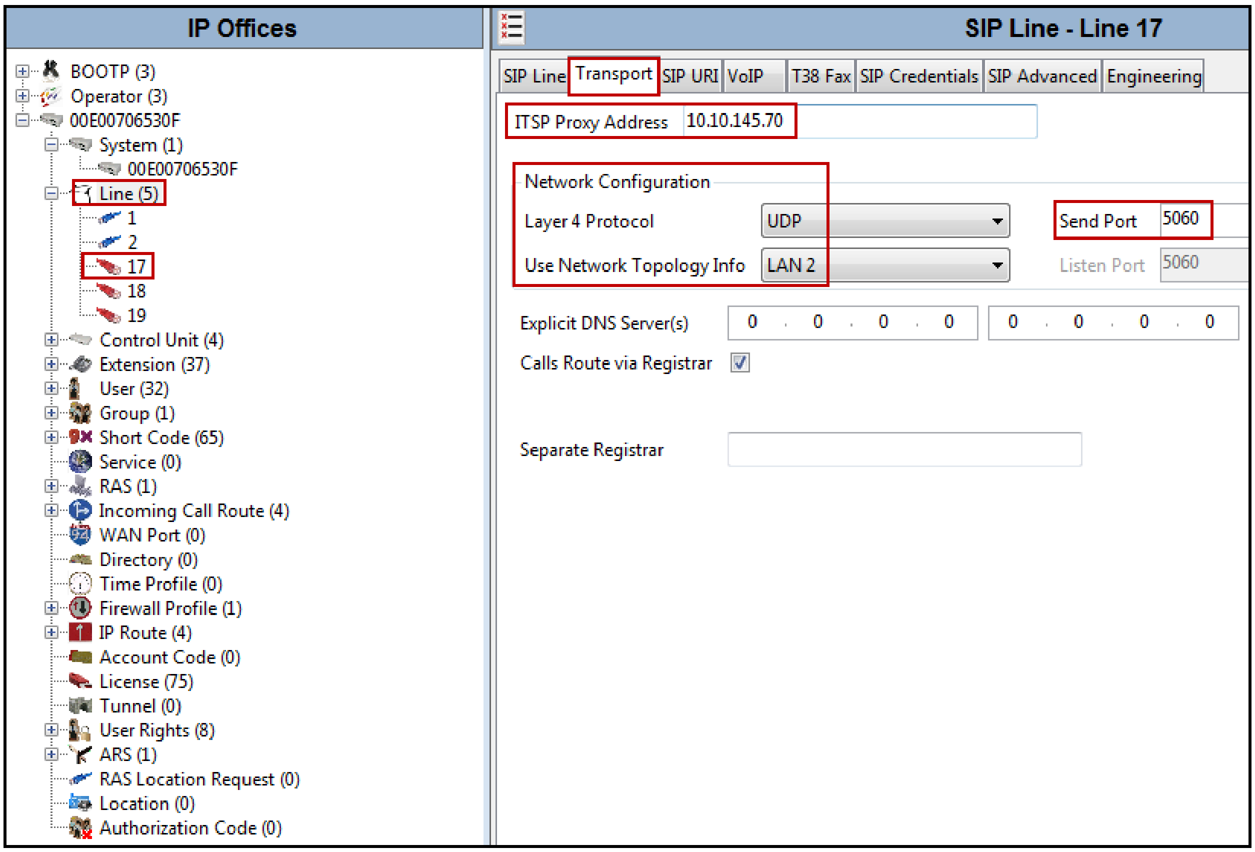
- Set the **ITSP Domain Name** to the domain name of the Service Provider, e.g. **as.voip.fuse.net**.
- Verify that **URI Type** is set to **SIP**.
- Verify that **In Service** box is checked, which is the default value. This makes the trunk available to incoming and outgoing calls.
- Verify that **Check 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** Support and **Outgoing Supervised REFER** to **Always**.
- All other parameters should be set to default or according to customer requirements.
- Click **OK** to commit (not shown).



### 5.4.4 SIP Line - Transport Tab

Select the **Transport** tab; configure the parameters as shown below:

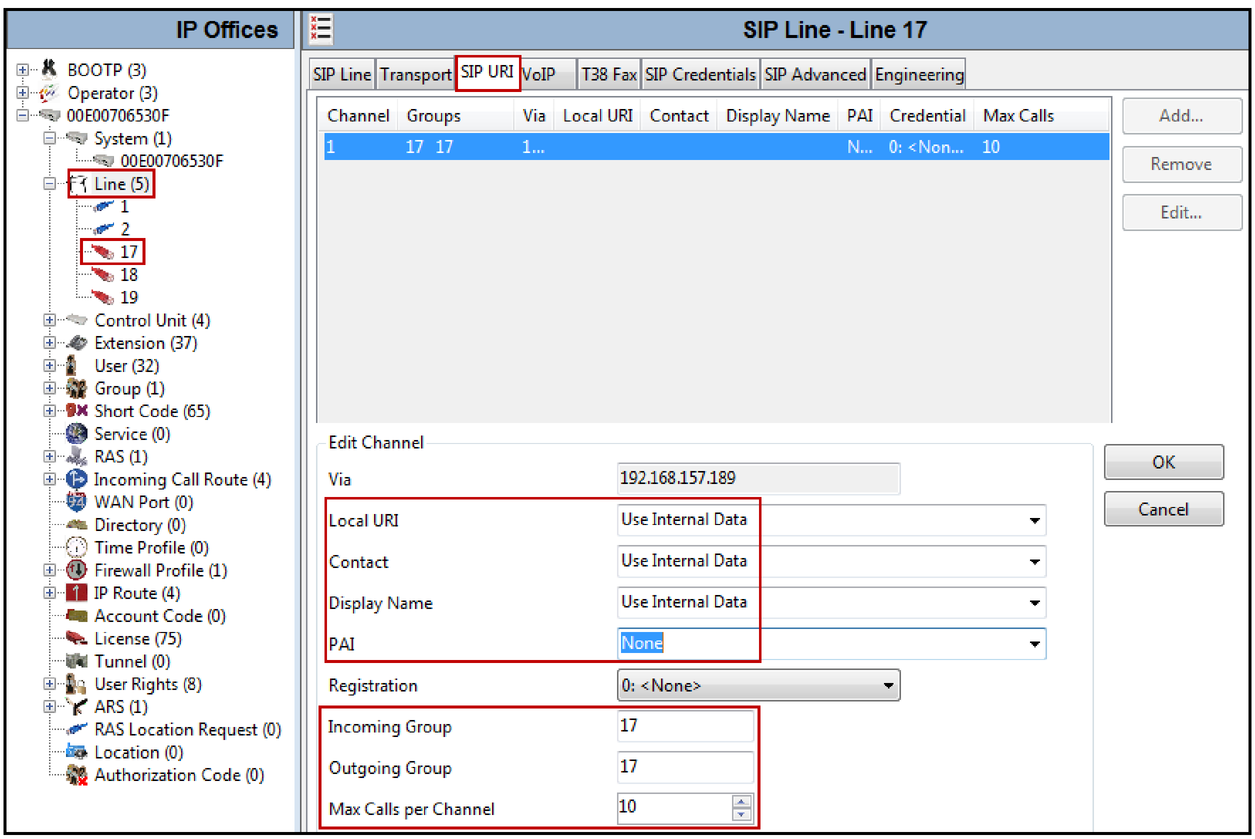
- Set the **ITSP Proxy Address** to the IP address of the Service Provider's SIP Proxy, as shown on **Figure 1**.
- Set the **Layer 4 Protocol** to **UDP**.
- Set **Use Network Topology Info** to **LAN2**. The LAN2 settings correspond to the WAN port on the Avaya IP Office 500v2, used by the SIP Line 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).



### 5.4.5 SIP Line - SIP URI Tab

A SIP URI entry needs to be created to match each incoming number that Avaya 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 Avaya IP Office user. The entry was created with the parameters shown below:

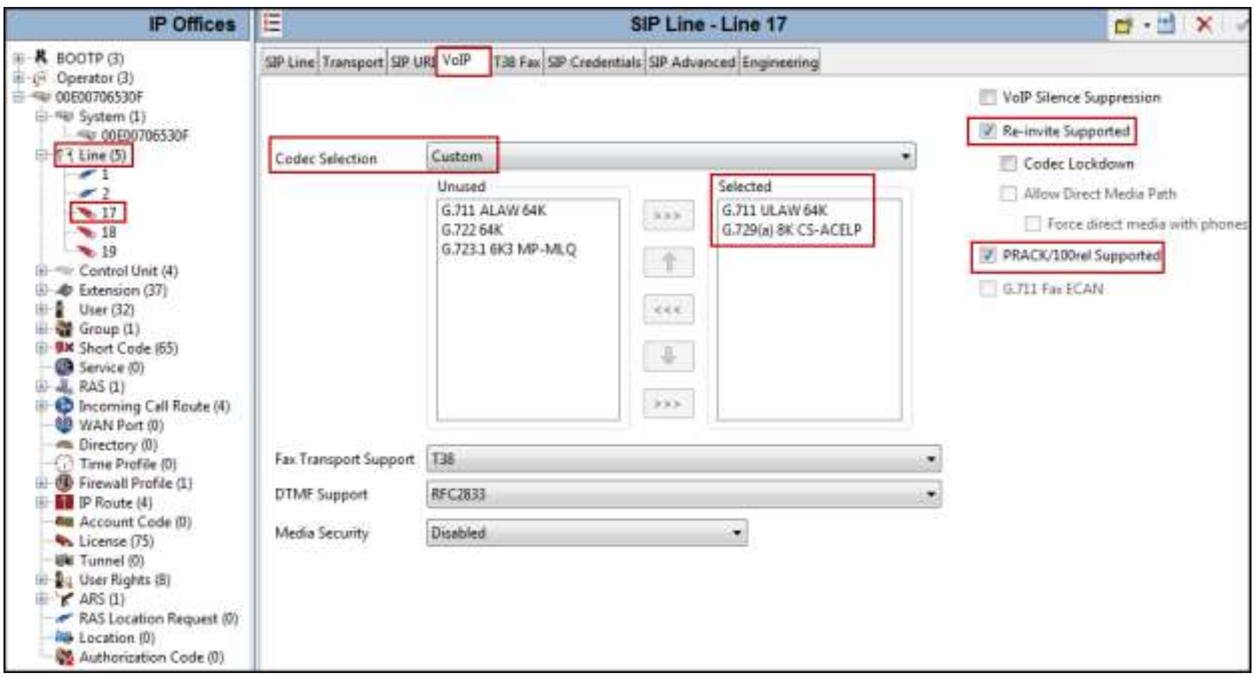
- Set **Local URI**, **Contact, Display Name** to **Use Internal Data**.
- Set **PAI** to **None**.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, 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.
- Click **OK**.
- Click **OK** again to commit (not shown).



## 5.4.6 SIP Line - VoIP Tab

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

- In the sample configuration, the **Codec Selection** was configured using the **Custom** option, allowing an explicit 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. Cincinnati Bell supports codec G.711ULAW and G.729(a) for audio, with G.711ULAW being the preferred codec, which is shown at the top.
- Select **T.38** for **Fax Transport Support** (Refer to **Section 2.1**).
- Set the **DTMF Support** field to **RFC2833.** This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Check the **Re-invite Supported** box to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk.
- Check the **PRACK/100rel Supported** box, to advertise the support for reliable provisional responses and Early Media to Cincinnati Bell.
- Default values may be used for all other parameters.
- Click **OK** to commit (not shown).

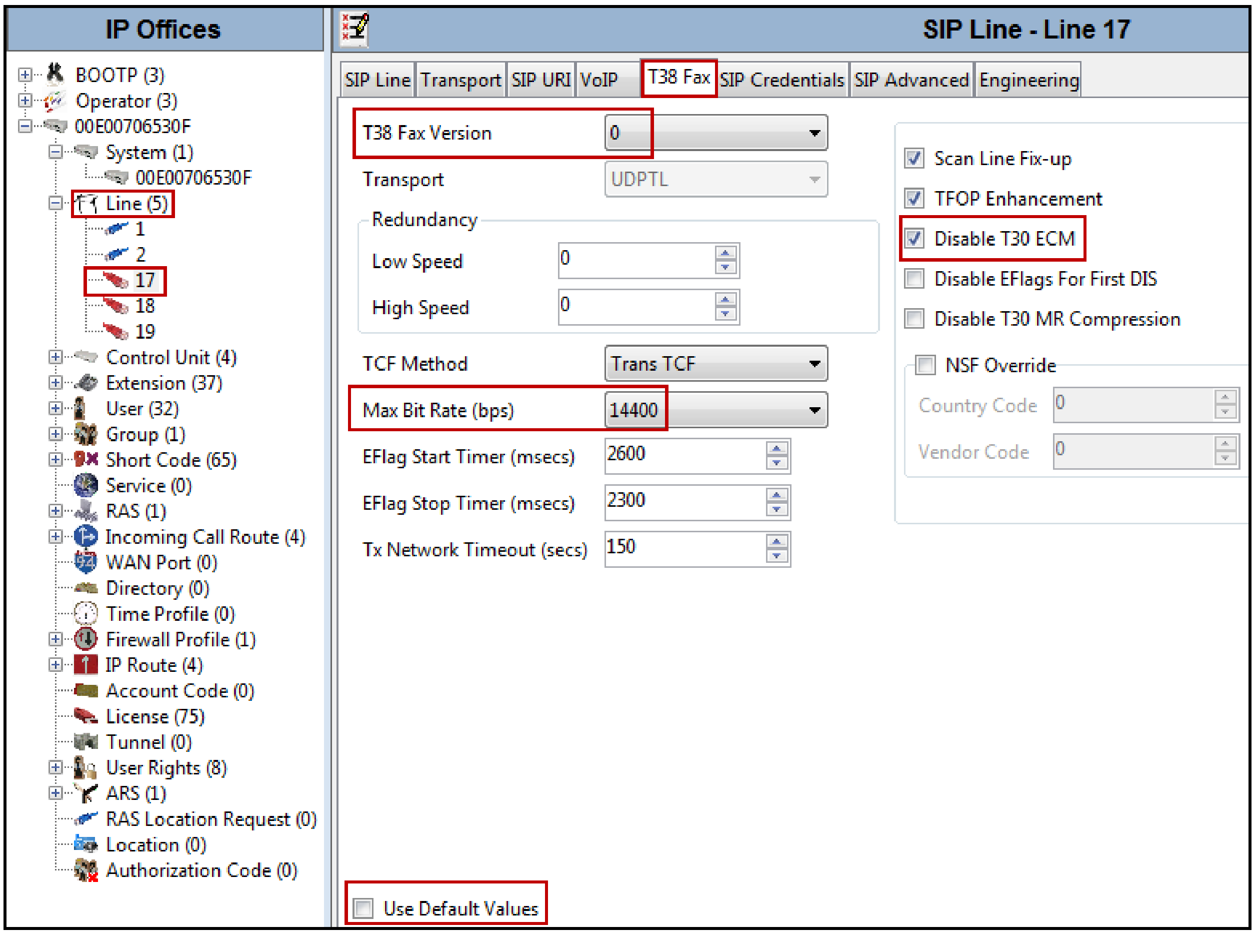



**Note**: The codec selections defined under this section (SIP Line – VoIP tab) are the codecs selected for the SIP Line (Trunk). The codec selections defined under **Section 5.2.4** (System – Codecs tab) are the codecs selected for the IP phones/extension (H.323 and SIP).

### 5.4.7 SIP Line – T.38 Fax Tab

Select the **T38 Fax** tab to set the Fax over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

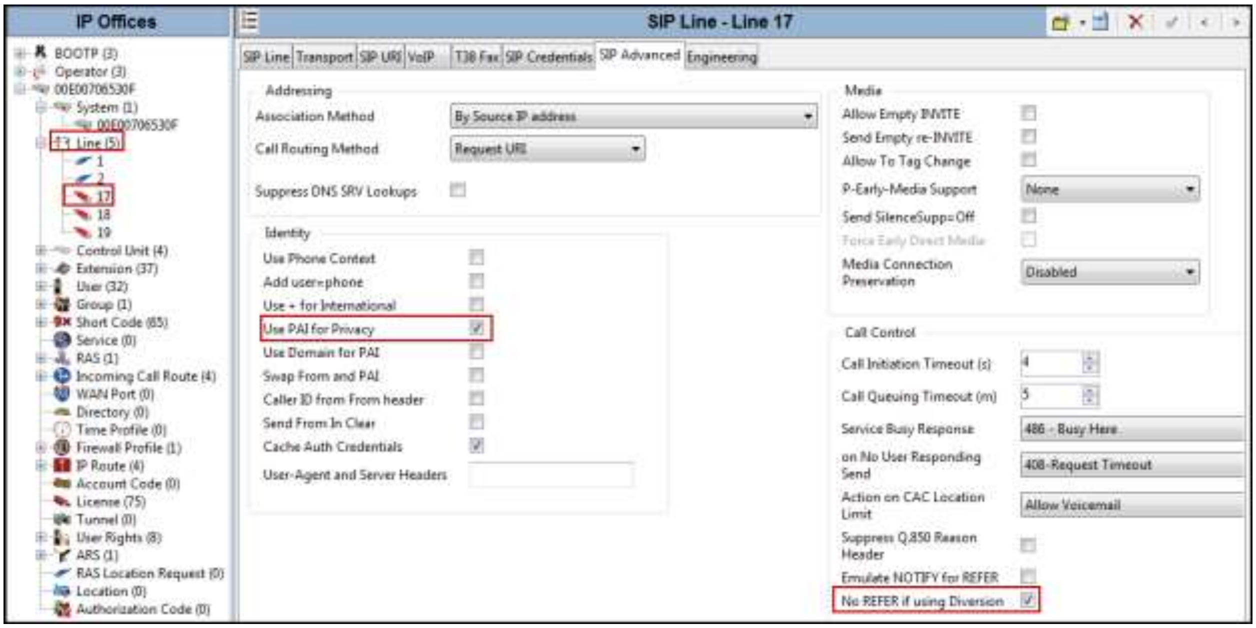
- Uncheck **Use Default Values** at the bottom of the screen.
- Set **T38 Fax Version** to **0**. Cincinnati Bell Business SIP Trunking supports T.38 fax version 0.
- Set **Max Bit Rate (bps)** to **14400**, the highest fax bit rate that Avaya IP Office supports for T.38 faxing.
- Check the **Disable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



### 5.4.8 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:

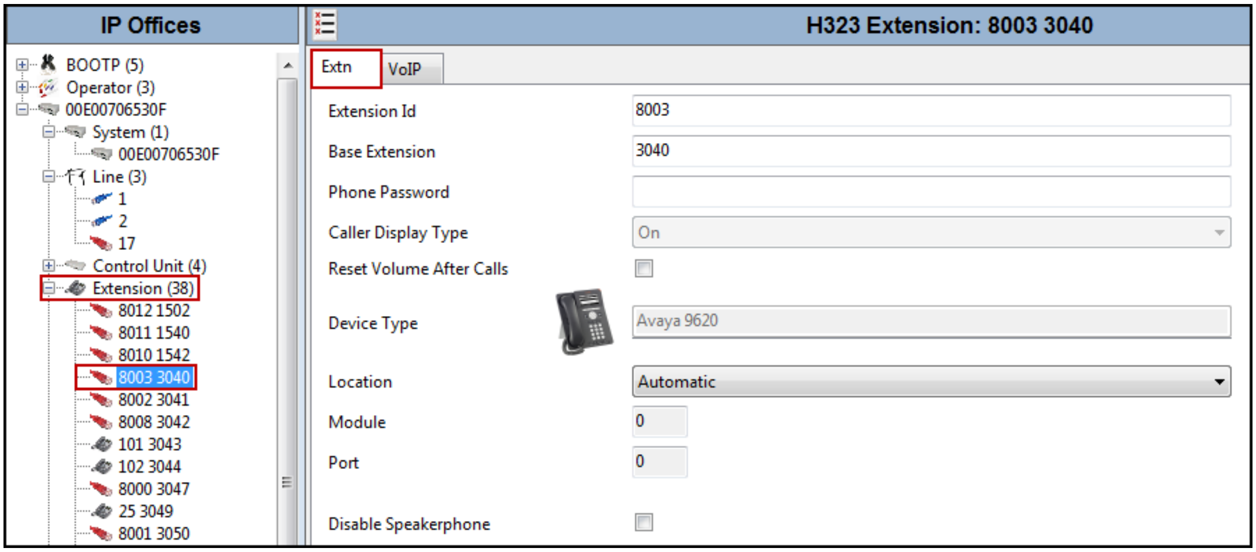
- Check the box for **Use PAI for Privacy.**
- Check **No REFER if using Diversion**. This directs IP Office not to send REFER if the SIP message contains the Diversion header.
- Default values may be used for all other parameters.
- Click OK to commit (not shown).



## 5.5 Extension

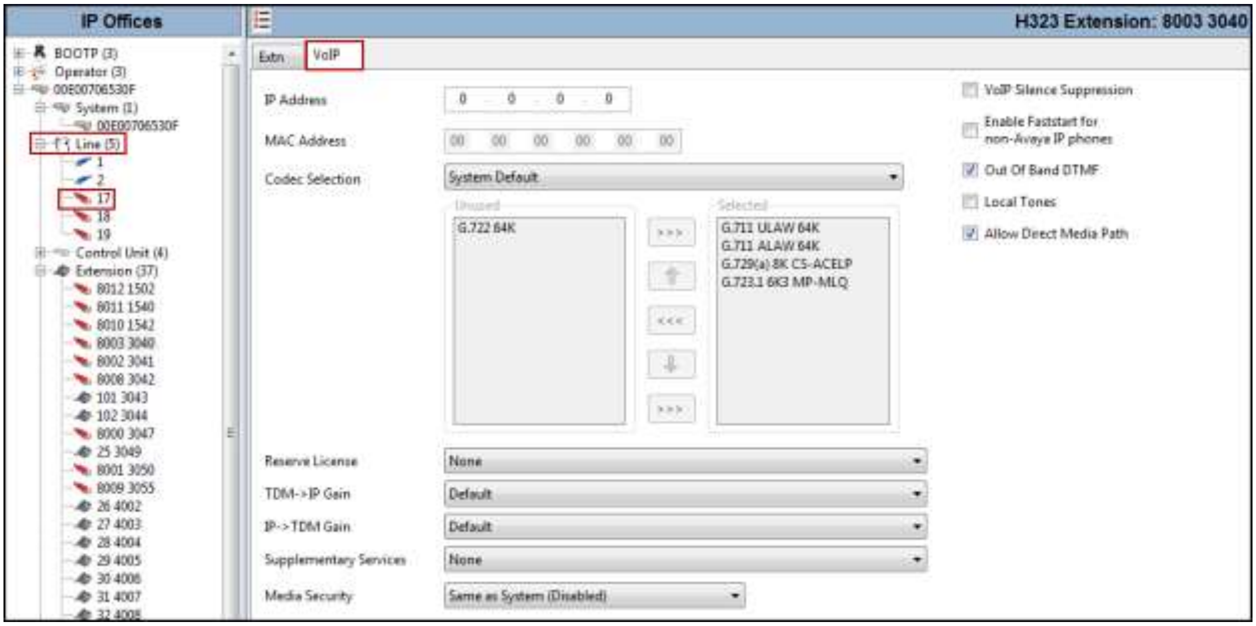
In this section, an example of an Avaya IP Office Extension will be illustrated. In the interests of brevity, not all users and extensions will be presented, since the configuration can be easily extrapolated to other users and extensions. To add an Extension, right click on **Extension** then select **New → Select H323 or SIP**.

Select the **Extn** tab. Following is an example of extension 3040; this extension corresponds to an H.323 extension.



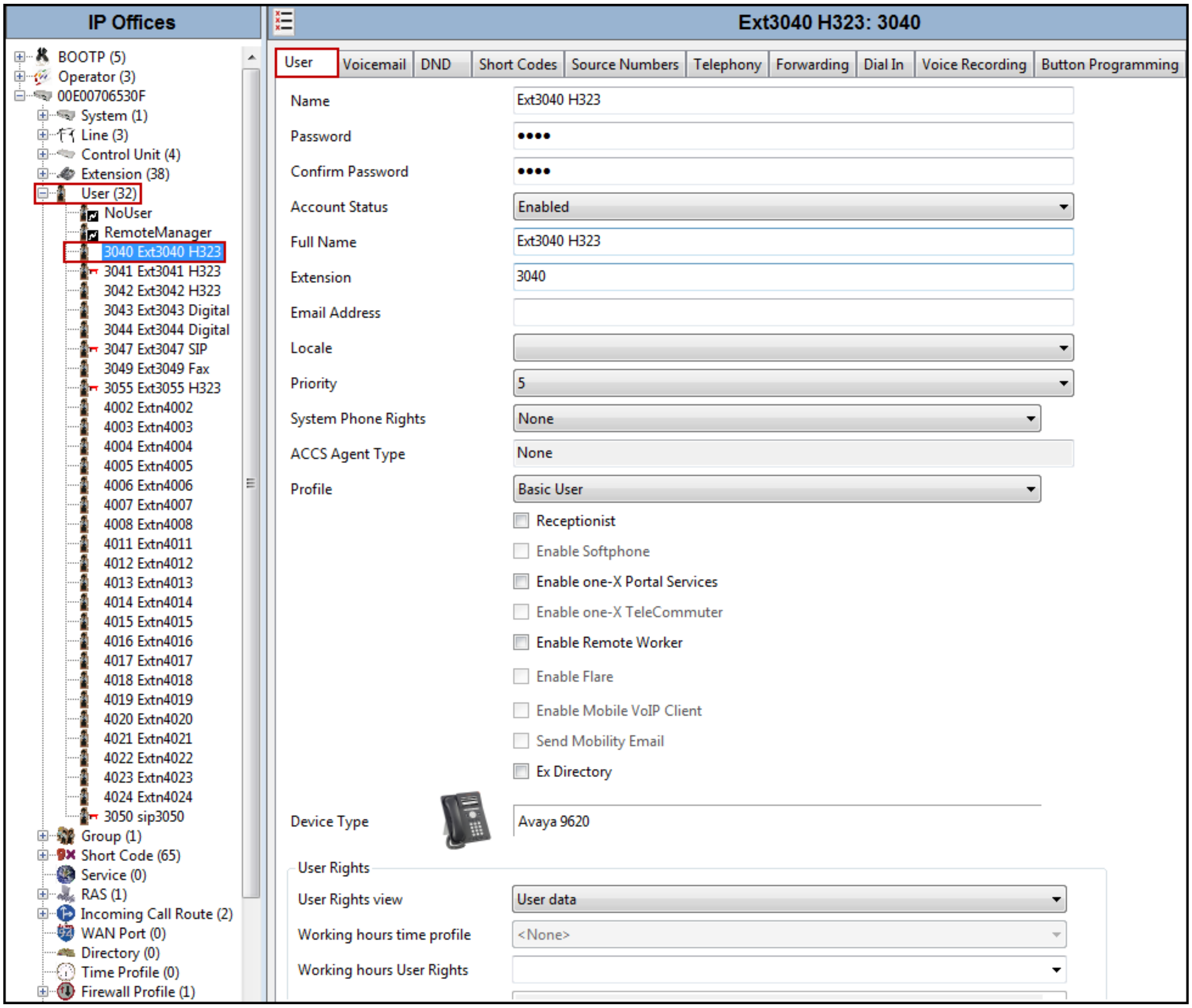
Select the **VOIP** tab. Use default values on VoIP tab. Following is an example for Extension 3040; this extension corresponds to an H.323 extension.

By default, all IP phones (SIP and H.323) will use the system default codec selection configured under the System Codecs tab (**Section 5.2.4**), unless configured otherwise for a specific extension by selecting **Custom** under **Codec Selection** on the screenshot shown below. The example below shows the codecs used for IP phones (SIP and H.323).

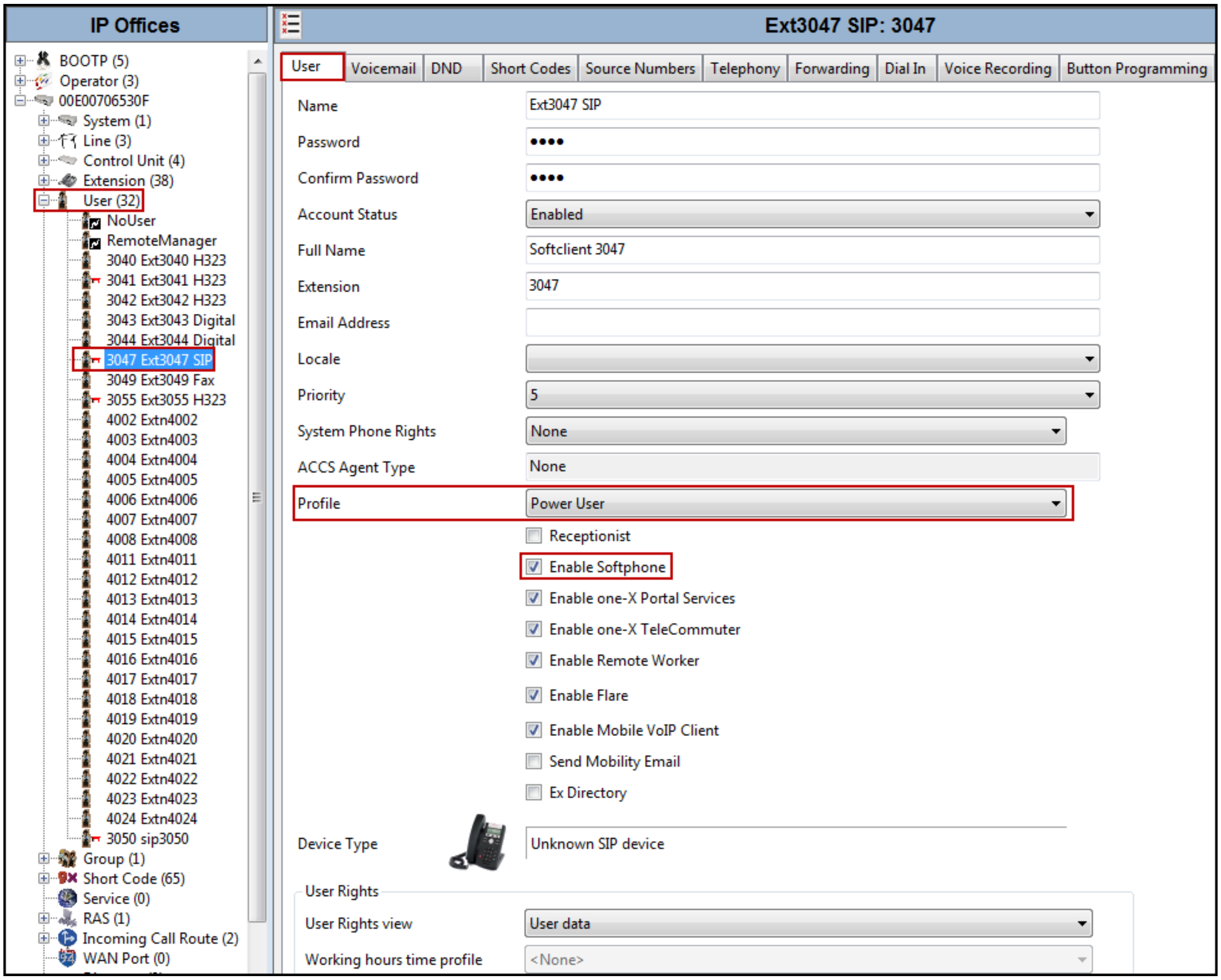


## 5.6 Users

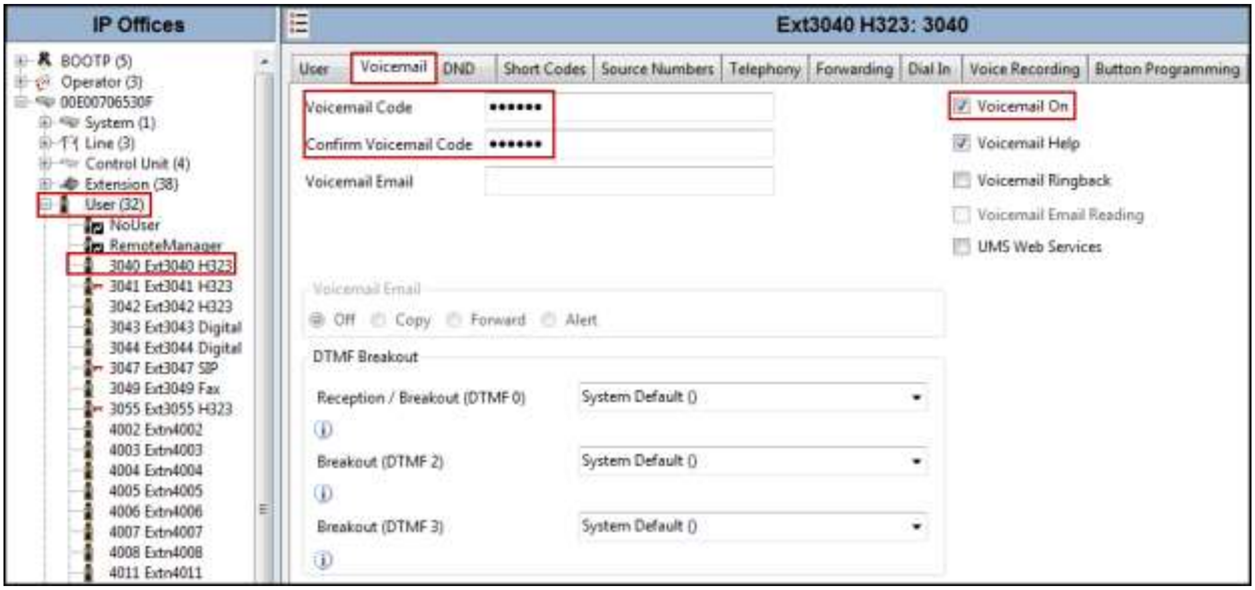
Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.4**. To configure these settings, first navigate to **User** in the left Navigation Pane, and then select the name of the user to be modified in the center Group Pane. In the example below, the name of the user is **Ext3040 H323**.



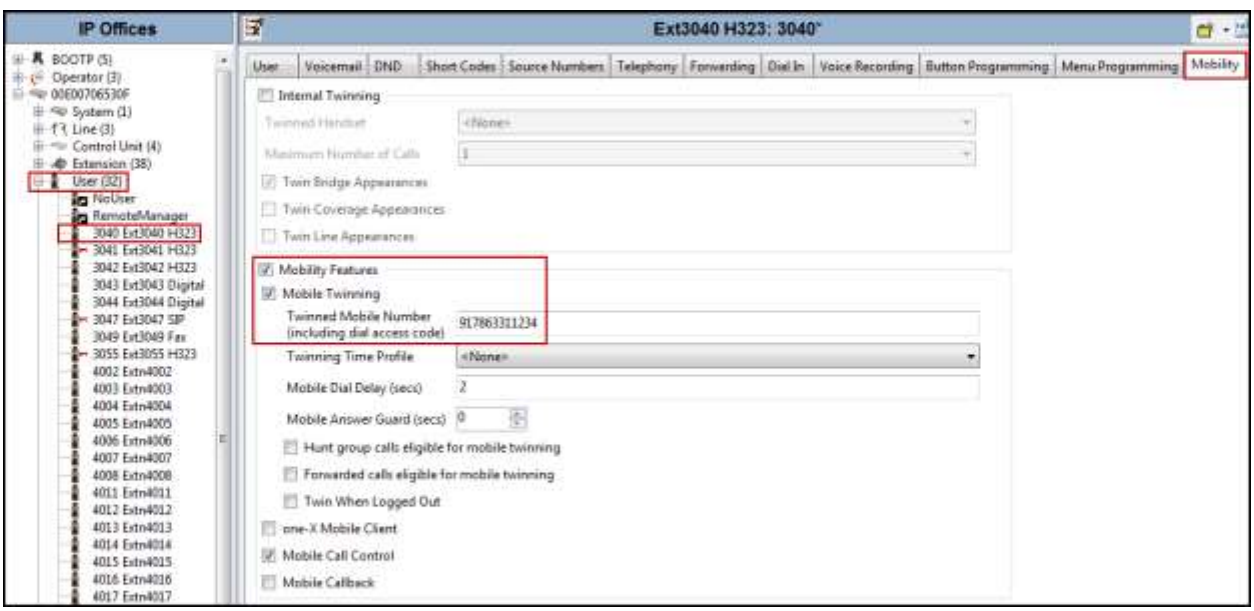
In the example below, the name of the user is “Ext3047 SIP”. This is an Avaya IP Office Softphone user, set the Profile to **Power User** and check **Enable Softphone**.



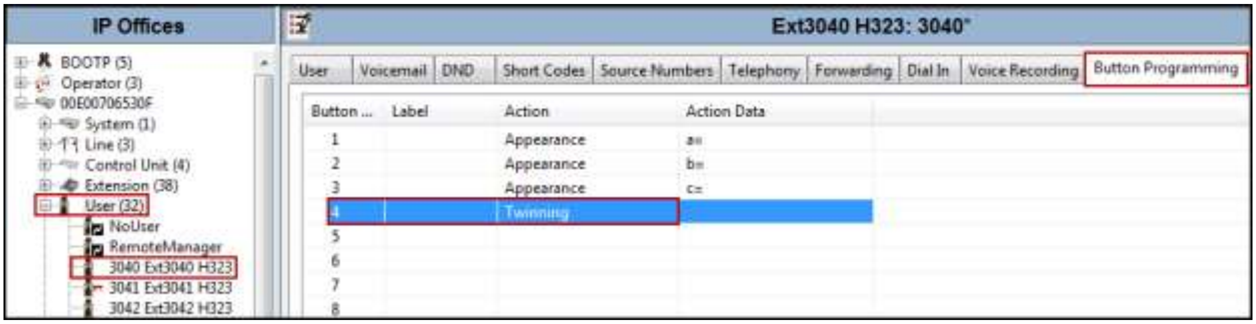
Select the **Voicemail** tab. The following screen shows the **Voicemail** tab for the user with extension 3040. The **Voicemail On** box is checked. Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters. In the verification of these Application Notes, incoming calls from Cincinnati Bell to this user were redirected to Voicemail Pro after no answer. Voicemail messages were recorded and retrieved successfully. Voice mail navigation and retrieval were performed locally and from PSTN telephones to test DTMF using RFC 2833.




Select the **Mobility** tab. In the sample configuration user 3040 was one of the users configured to test the Mobile Twinning feature. The following screen shows the **Mobility** tab for User 3040. The **Mobility Features** and **Mobile Twinning** boxes are checked**.** The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned telephone, including the dial access code "9", in this case **917863311234**. Other options can be set according to customer requirements.

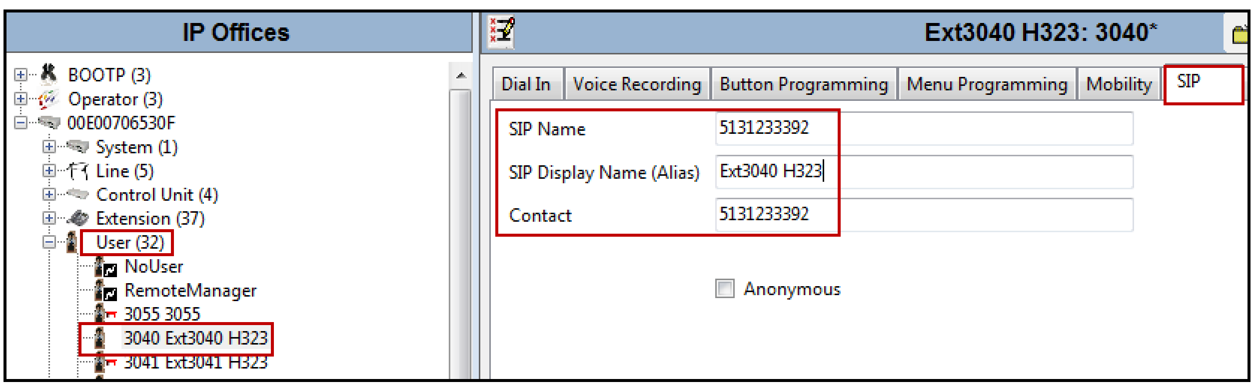


To program a key on the telephone to turn Mobil Twinning on and off, select the **Button Programming** tab on the user, then select the button to program to turn Mobil Twinning on and off, click on **Edit → Emulation → Twinning** (not shown). In the sample below, button **4** was programmed to turn Mobil Twinning on and off on user 3040.




Select the **SIP** tab, the values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the "From" and "Contact" headers for outgoing SIP trunk calls. In addition, these settings are used to match against the SIP URI of incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.4**). The example below shows the settings for user "Ext3040 H323". The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise by Cincinnati Bell. In the example, DID number **5131233392** was used. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name.

If all calls involving this user should be considered private, then the **Anonymous** box may be checked to withhold the Caller ID information from the network.



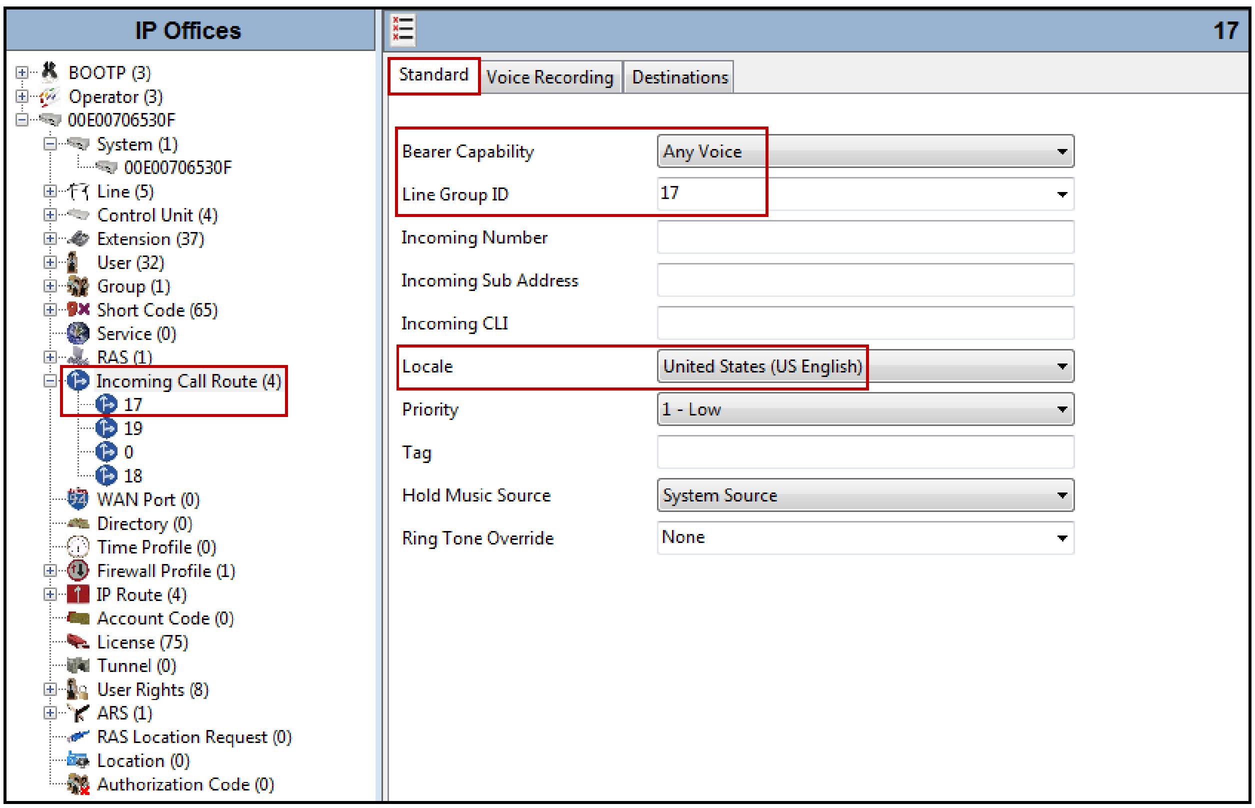
## 5.7 Incoming Call Route

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.

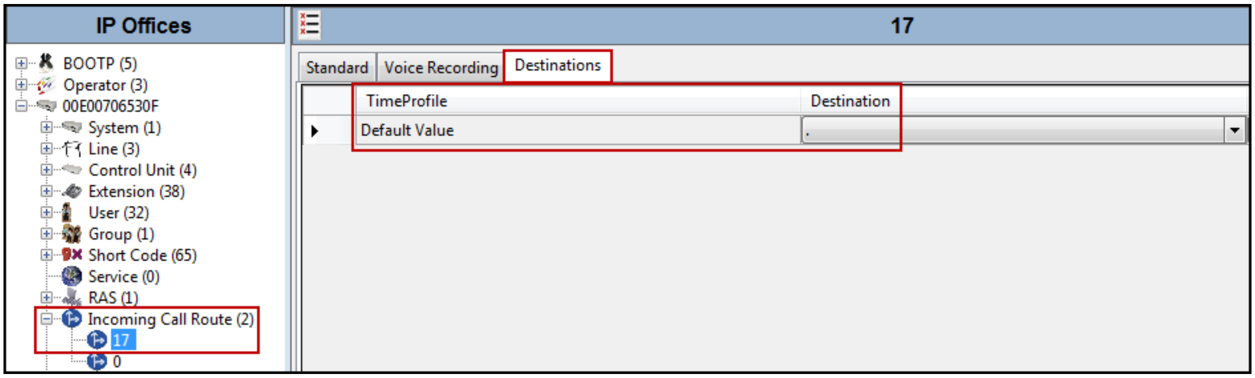
In a scenario like the one used for the compliance test, only one incoming route is needed, which allows any incoming number arriving on the SIP trunk to reach any predefined extension in IP Office. The routing decision for the call is based on the parameters previously configured for **Call Routing Method** and **SIP URI (Section 5.4.5)** and the users **SIP Name** and **Contact**, already populated with the assigned Cincinnati Bell DID numbers **(Section 5.6)**.

From the left Navigation Pane, right-click on **Incoming Call Route** and select **New**.
On the Details Pane (not shown), under the **Standard** tab, set the parameters as show bellow:

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



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

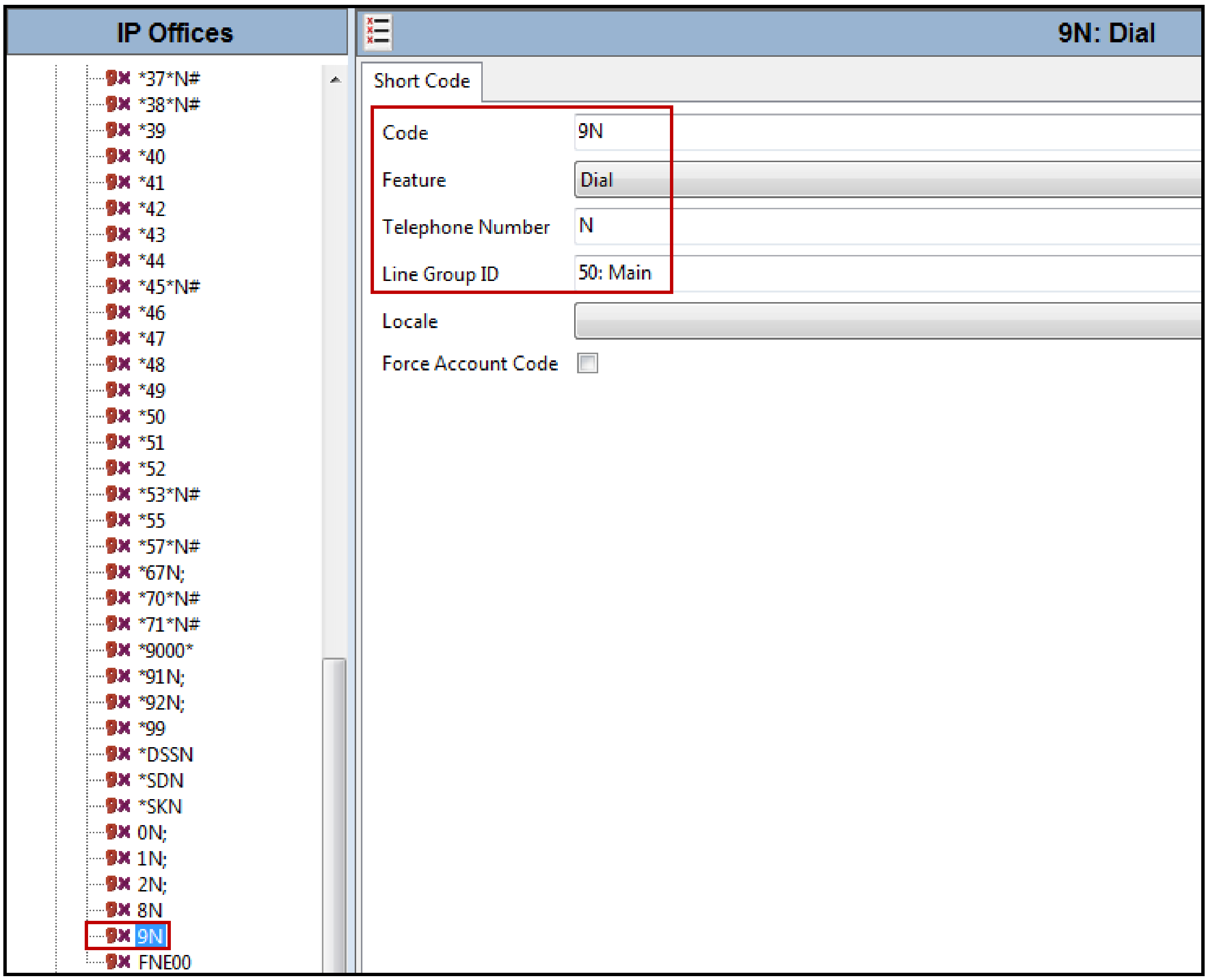


## 5.8 Outbound Call Routing

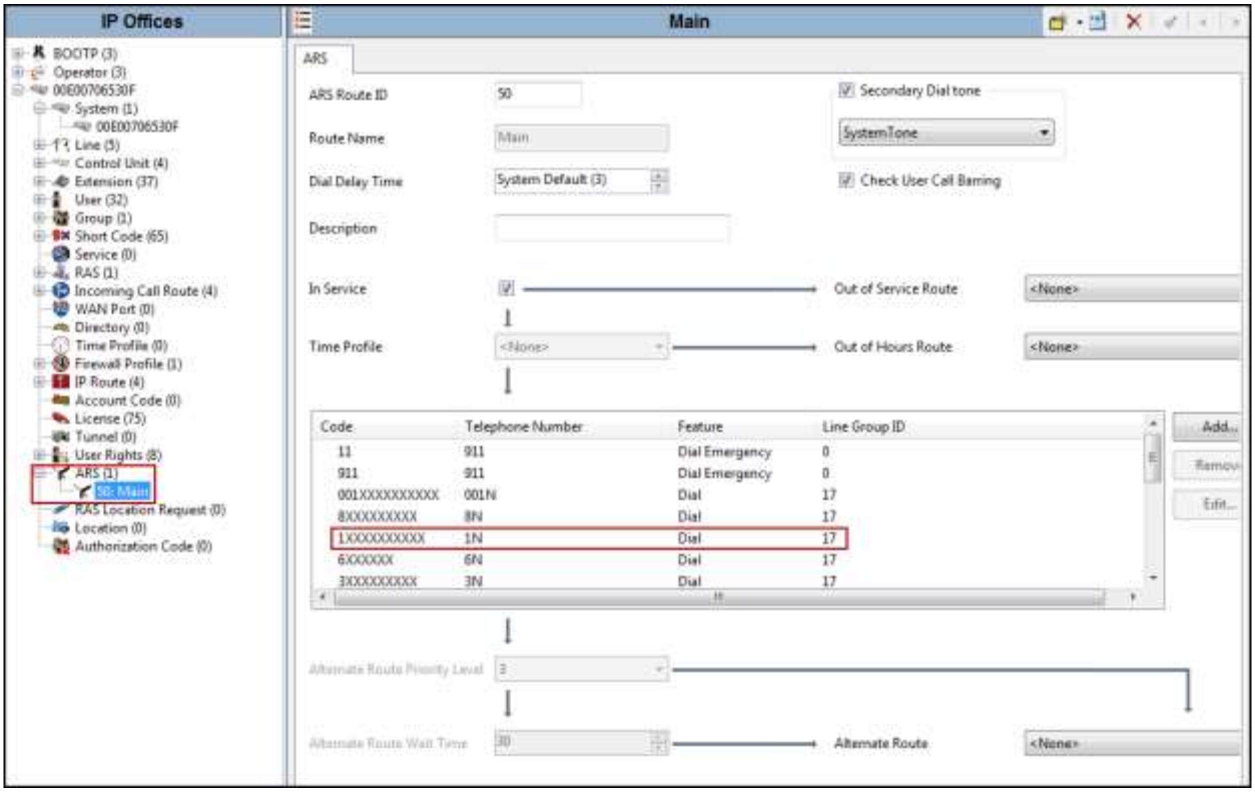
For outbound call routing, a combination of system short codes and Automatic Route Selection (ARS) entries are used. With ARS, features like time-based routing criteria and alternate routing can be specified so that a call can re-route automatically if the primary route or outgoing line group is not 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.

### 5.8.1 Short Codes and Automatic Route Selection

To create the short code used for ARS, right-click on **Short Code** in the Navigation Pane and select **New** (not shown). The screen below shows the creation of the short code ***9N*** used in the reference configuration. When the Avaya IP Office users dialed 9 plus any number N, calls were directed to **Line Group *50: Main***, configurable via ARS and defined next in this section.



The following screen shows the example 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 digit on the string. This type of setting results in a much quicker response in the delivery of the call by IP Office. The first example highlighted below shows that for calls to area codes in the North American Numbering Plan, the user dialed 9, followed by 11 digits, starting with a 1.

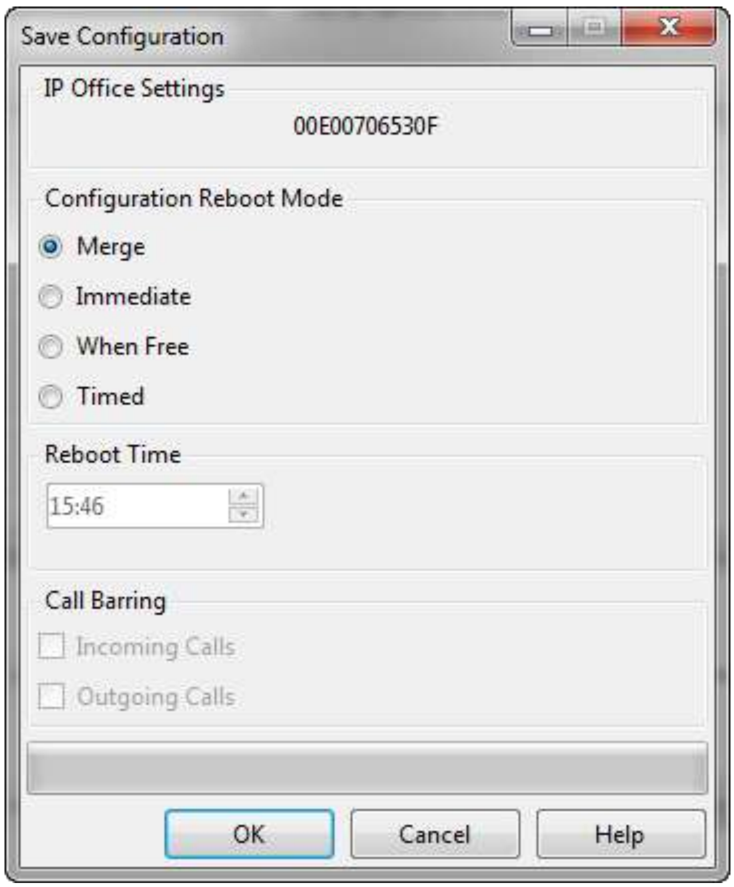


## 5.9 Save Configuration

When desired, send the configuration changes made in Avaya IP Office Manager to the Avaya IP Office server in order for the changes to take effect.

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

Once the configuration is validated, a screen similar to the following will appear, with either the **Merge** or the **Immediate** radio button chosen based on the nature of the configuration changes made since the last save. Note that clicking OK may cause a service disruption due to system reboot. Click **OK** if desired.



# 6. Cincinnati Bell Business SIP Trunk Service Configuration

To use the Cincinnati Bell Business SIP Trunking service offering, a customer must request the service from Cincinnati Bell using the established sales processes. The process can be started by contacting Cincinnati Bell via the corporate web site at:
https://www.cincinnatibell.com/customer_support/

During the signup process, Cincinnati Bell and the customer will discuss details about the preferred method to be used to connect the customer's enterprise network to Cincinnati Bell's network. Cincinnati Bell 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.

# 7. Verification and Troubleshooting

This section provides verification steps that may be performed in the field to verify that the solution is configured properly. This section also provides a list of useful troubleshooting tips that can be used to troubleshoot the solution.

## 7.1 Verification Steps

The following steps may be used to verify the configuration:

- Verify that endpoints at the enterprise site can place calls to PSTN and that calls remain active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.
- Verify that endpoints at the enterprise site can receive calls from PSTN and that calls can remain active for more than 35 seconds.
- Verify that the user on the PSTN side can end an active call by hanging up.
- Verify that an Avaya endpoint at the enterprise site can end an active call by hanging up.

## 7.2 Protocol Traces

The following SIP message headers are inspected using sniffer trace analysis tool:

- Request-URI: Verify the request number and SIP domain.
- From: Verify the display name and display number.
- To: Verify the display name and display number.
- P-Asserted-Identity: Verify the display name and display number.
- Privacy: Verify privacy masking with “user, id”.
- Diversion: Verify the display name and display number.

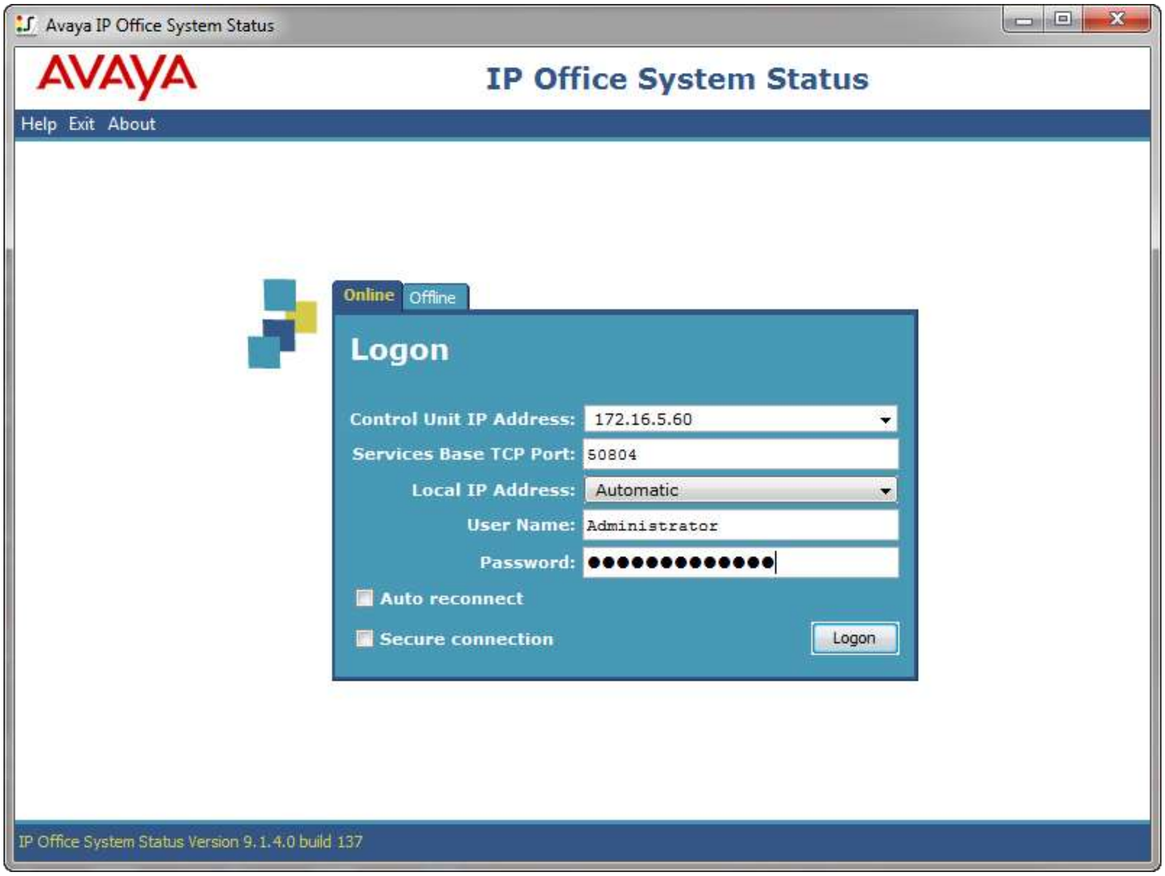
The following attributes in SIP message body are inspected using sniffer trace analysis tool:

- Connection Information (c line): Verify IP addresses of near end and far end endpoints.
- Time Description (t line): Verify session timeout value of near end and far end endpoints.
- Media Description (m line): Verify audio port, codec, DTMF event description.
- Media Attribute (a line): Verify specific audio port, codec, ptime, send/ receive ability, DTMF events.

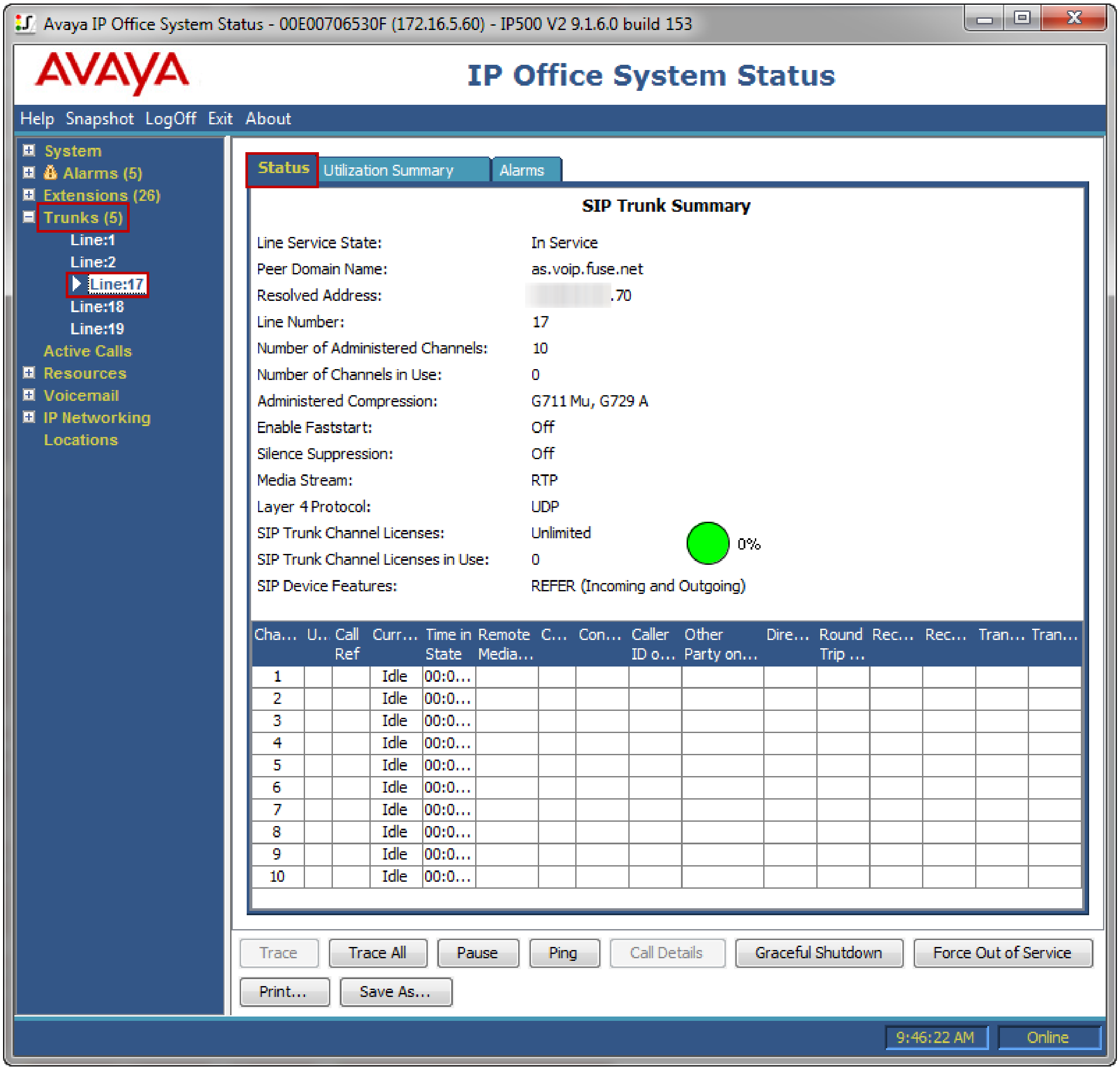
## 7.3 IP Office System Status

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

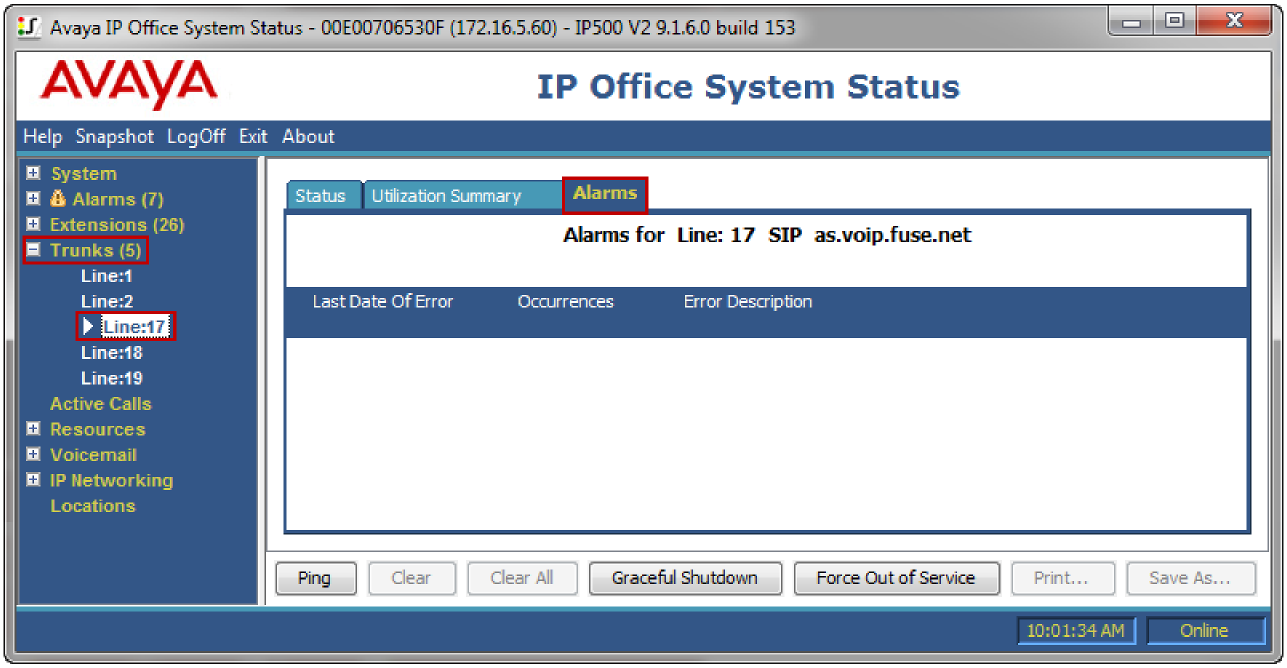
Use the Avaya IP Office **System Status** application to verify the state of SIP connections. Launch the application from **Start → Programs → IP Office → System Status** on the PC where IP Office Manager is installed, log in with the proper credentials.



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

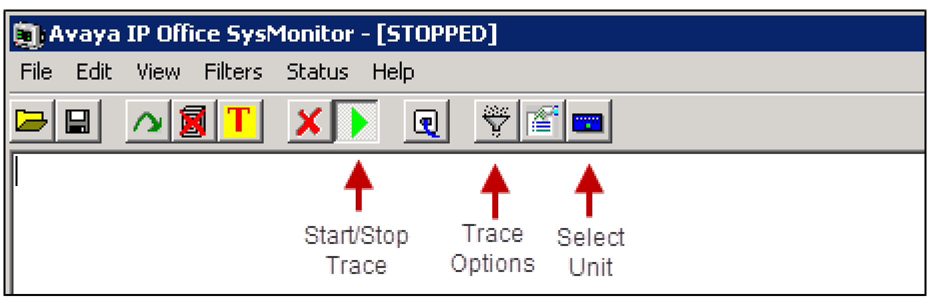


- Select the **Alarms** tab and verify that no alarms are active on the SIP Line.

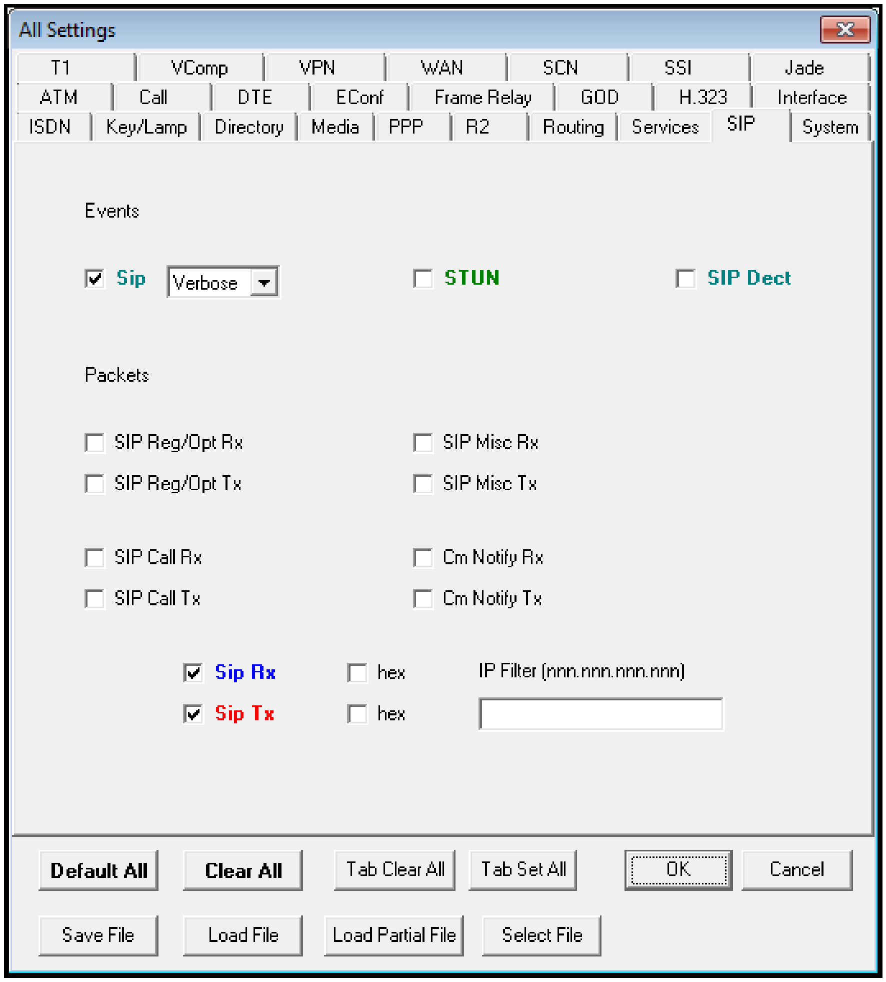



## 7.4 IP Office Monitor

The Avaya IP Office Monitor application can be used to monitor and troubleshoot signaling messaging on the SIP trunk. Launch the application from **Start → Programs → IP Office → Monitor** on the PC where Avaya IP Office Manager was installed. Click the **Select Unit** icon on the taskbar and Select the IP address of the IP Office system under verification.




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


All Settings
T1
VComp
VPN
WAN
SCN
SSI
Jade
ATM
Call
DTE
EConf
Frame Relay
GOD
H.323
Interface
ISDN
Key/Lamp
Directory
Media
PPP
R2
Routing
Services
SIP
System
Events
Sip
Verbose
STUN
SIP Dect
Packets
SIP Reg/Opt Rx
SIP Misc Rx
SIP Reg/Opt Tx
SIP Misc Tx
SIP Call Rx
Cm Notify Rx
SIP Call Tx
Cm Notify Tx
Sip Rx
hex
IP Filter (nnn.nnn.nnn.nnn)
Sip Tx
hex
Default All
Clear All
Tab Clear All
Tab Set All
OK
Cancel
Save File
Load File
Load Partial File
Select File

# 8. Conclusion

These Application Notes describe the procedures required to configure SIP trunk connectivity between Avaya IP Office 9.1 and Cincinnati Bell Business SIP Trunking Service, as shown in **Figure 1**.

Interoperability testing was completed successfully 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] *Deploying Avaya IP Office Platform IP500 V2*, Document Number 15-601042, Issue 30zc, March 21, 2016.
[2] *Using Avaya IP Office Platform System Status*, Document Number 15-601758, Issue 10f, August 2015.
[3] *Administering Avaya IP Office Platform Voicemail Pro*, Document Number 15-601063, Issue 10m, February 05, 2016.
[4] *Using IP Office System Monitor*, Document Number 15-601019, Issue 06g, February 08, 2016.

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

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.