



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Cox SIP Trunking with Avaya IP Office 9.0 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider Cox Communications and Avaya IP Office 9.0.

Cox SIP Trunking provides PSTN access via a SIP trunk between the enterprise and the Cox network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

Cox 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 procedures for configuring Session Initiation Protocol (SIP) Trunking between service provider Cox Communications (here on referred as Cox) and an Avaya IP Office solution. In the sample configuration, the Avaya IP Office solution consists of an Avaya IP Office 500v2 Release 9.0, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, digital, and analog endpoints.

The Cox SIP Trunking service referenced within these Application Notes is designed for business customers. The service enables local and long distance PSTN calling via standards-based SIP trunks as an alternative to legacy analog or digital trunks, without the need for additional TDM enterprise gateways and the associated maintenance costs.

The Cox SIP Trunking service uses Digest Authentication for outbound calls from the enterprise, using challenge-response authentication for each call to the Cox network based on a configured user name and password (provided by Cox and configured in IP Office). This call authentication scheme as specified in SIP RFC 3261 provides security and integrity protection for SIP signaling.

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 Cox SIP Trunking service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality tests listed in **Section 2.1**.

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

2.1. Interoperability Compliance Testing

A simulated enterprise site with Avaya IP Office was connected to Cox SIP Trunking service. To verify SIP Trunking interoperability, the following features and functionality were exercised during the interoperability compliance test:

- Response to SIP OPTIONS queries.
- Incoming PSTN calls to various phone types. Phone types included H.323, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types. Phone types included H.323, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from the Avaya IP Office Softphone.
- Inbound and outbound long holding time call stability.

- Various call types including: local, long distance, international, outbound toll-free, operator service and directory assistance.
- Codec G.711MU.
- Caller number/ID presentation.
- Privacy requests (i.e., caller anonymity) and Caller ID restriction for inbound and outbound calls.
- DTMF transmission using RFC 2833.
- Voicemail navigation for inbound and outbound calls.
- Telephony features such as hold and resume, transfer, and conference.
- Use of SIP Re-INVITE and REFER for call transfer to PSTN
- FAX G.711 Pass Through
- Off-net call forwarding.
- Twinning to mobile phones on inbound calls.

2.2. Test Results

Cox SIP Trunking passed compliance testing.

Items not supported or not tested included the following:

- G.729 codec is not supported.
- T.38 Fax is not supported.
- Cox system does not support REFER for call transfer at the moment.
- Cox does not use either PPI or PAI for the purposes of privacy.

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

- **Call Display on PSTN Phone** – Call display was not properly updated on PSTN phone involved in a call transfer. After the call transfer was completed, the PSTN phone did not display the actual connected party but instead showed the party that initiated the transfer. SIP signaling trace showed that the enterprise IP Office did not send an UPDATE message to the network to update the call display of the PSTN phone in call transfer. However, it does not affect the end user.
- **Call Display on PSTN Phone on PSTN Hold and Resume** – Call display was not properly updated on PSTN phone involved in PSTN Hold and Resume operation. After the inbound call from PSTN to Avaya IP Office was hold on PSTN phone, when the call was resumed on the PSTN phone, the PSTN phone displayed trunk number instead of the calling party ID.
- **Call Redirection (Transfers) Using REFER method** – Cox system does not properly response to REFER request with 202 Accepted but with 403 Forbidden or 603 Decline response codes. The calls are being transferred successfully with 2 ways voice path.

2.3. Support

For technical support on the Avaya products described in these Application Notes visit <http://support.avaya.com>.

For technical support on Cox SIP Trunking, contact Cox at <http://www.cox.com>.

3. Reference Configuration

Figure 1 below illustrates the test configuration. The test configuration shows an enterprise site connected to Cox SIP Trunking service through the public IP network. For confidentiality and privacy purposes, actual public IP addresses used in this testing have been masked out and replaced with fictitious IP addresses throughout the document.

Located at the enterprise site are Edgewater EdgeMarc 4550 series (EdgeMarc SIP in B2BUA mode, referred to as EdgeMarc throughout this document) and Avaya IP Office 500v2 with the MOD DGTL STA16 (Module Digital Station 16) expansion which provides connections for 16 digital stations to the PSTN, the extension PHONE 8 card which provides connections for 8 analog stations to the PSTN as well as 64-channel VCM (Voice Compression Module) for supporting VoIP codecs. The LAN port of Avaya IP Office is connected to LAN port of the EdgeMarc while WAN port of EdgeMarc is connected to public IP network. Endpoints include an Avaya 9600 Series IP Telephone (with H.323 firmware), an Avaya 9508 Digital Telephones, an Avaya Symphony 2000 Analog Telephone and an Avaya IP Office Softphone. A separate Windows XP PC runs Avaya IP Office Manager to configure and administer the Avaya IP Office.

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.

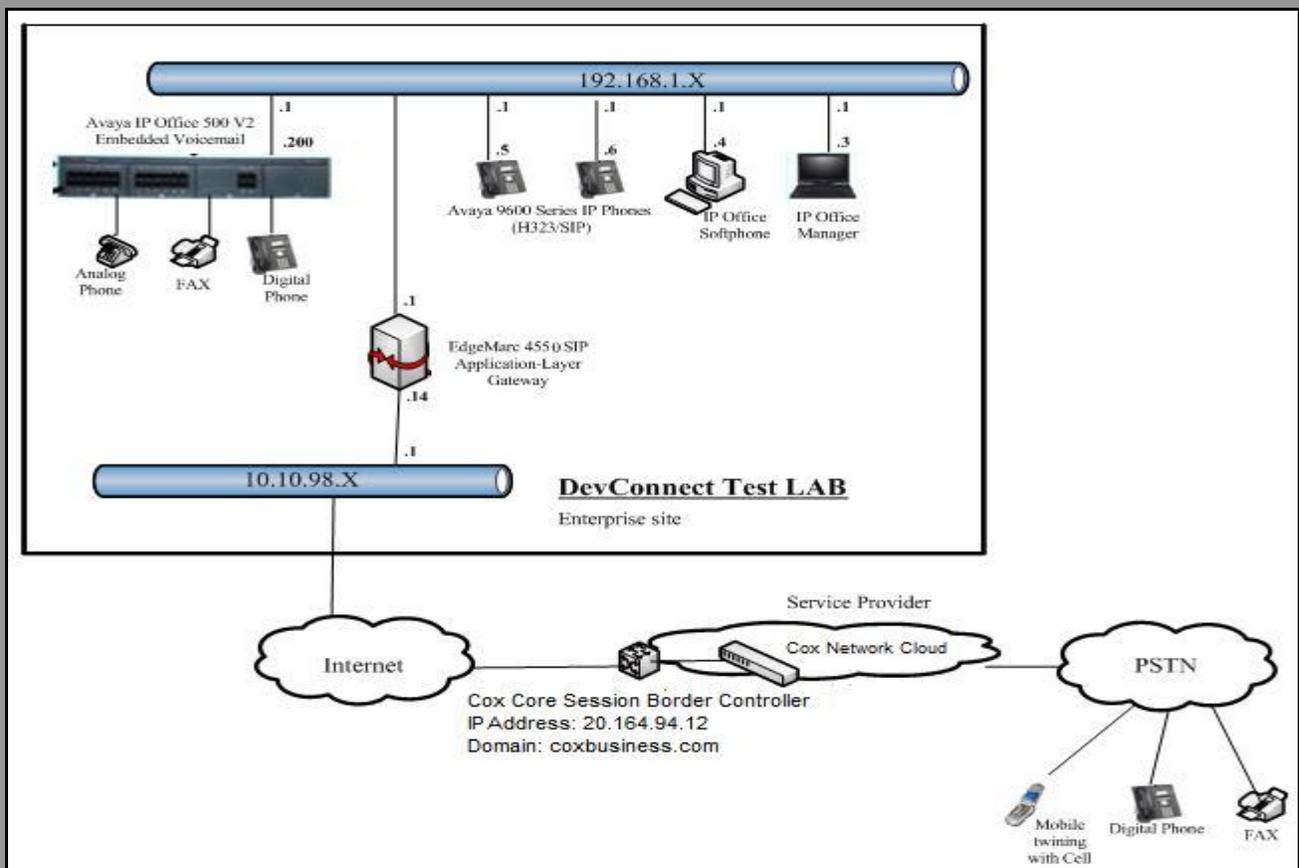


Figure 1: Test Configuration for Avaya IP Office with Cox SIP Trunking Service

Note: Cox system required that EdgeMarc to be resided on the same subnet of Avaya IP Office as shown in the Figure 1. EdgeMarc is part of Cox SIP trunking service offering.

For the purposes of the compliance test, Avaya IP Office users dialed a short code of 5 + N digits to send digits across the SIP trunk to Cox system. The short code of 5 was stripped off by Avaya IP Office but the remaining N digits were sent unaltered to Cox. For calls within the North American Numbering Plan (NANP), the user would dial 11 (1 + 10) digits. Thus for these NANP calls, Avaya IP Office would send 11 digits in the Request URI and the To field of an outbound SIP INVITE message. It was configured to send 10 digits in the From field. For inbound calls, Cox SIP Trunking sent 10 digits in the Request URI and the To field of inbound SIP INVITE messages.

Cox uses the phone number in the From header of a SIP INVITE message to authenticate the calling party. Thus, a call will be rejected by the network unless the From header contains a number known to Cox. This is especially important for calls inbound from the PSTN which are redirected back to the PSTN by call forwarding or twinning. For call forwarding, Avaya IP Office sends the number of the forwarding phone in the From header. This is a number known to Cox. As a result, the call display on the destination phone shows the forwarding party not the original caller. For twinning, this behavior can be slightly altered through configuration. See **Sections 5.3** and **5.4** for details.

Note that the calling party authentication using the phone number by Cox, as mentioned above, is in addition to the Digest Authentication by Cox. During call setup, SIP signaling exchanges are using user name and password as configured in Avaya IP Office for all calls from the enterprise to Cox.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and the Avaya IP Office such as a session border controller or 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 the Avaya IP Office must be allowed to pass through these devices.

4. Equipment and Software Validated

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

Avaya Telephony Components	
Equipment	Release
Avaya IP Office 500v2	9.0.0.829
Avaya IP Office Manager	9.0.0.829
Avaya Voicemail Pro for IP Office	9.0.0.829
Avaya 11x0 IP Telephone (SIP)	SIP11x0e04.03.12.00
Avaya 9630G IP Telephone (H.323)	Avaya one-X® Deskphone Edition S3.2
Avaya IP Office Softphone	3.2.3.20 64770
Avaya Digital Telephone (9508)	N/A
Avaya Symphony 2000 Analog Telephone	N/A
Cox SIP Trunking Service Components	
Component	Release
Edgewater EdgeMarc 4550	Version 11.6.14
ACME/Oracle 9200	nnSD710m3p4 and nnSD710m4p2
Broadsoft	AS, NS = R19

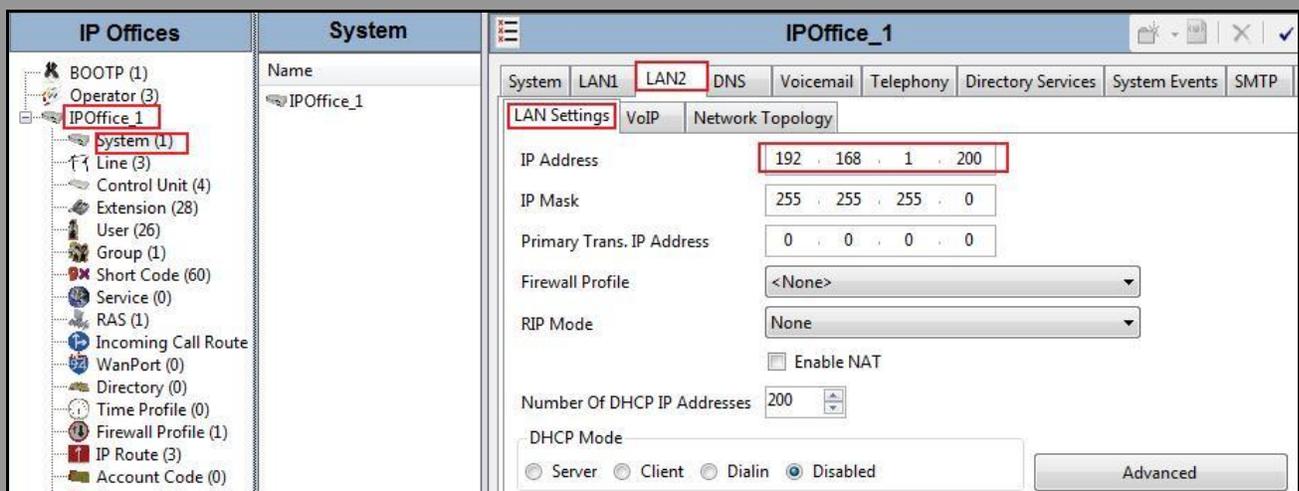
5. Configure IP Office

This section describes the Avaya IP Office configuration to support connectivity to Cox SIP Trunking service. Avaya IP Office is configured through the Avaya IP Office Manager PC application.

From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application (not shown). Navigate to **File → Open Configuration**, select the proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials. A management window will appear similar to the one shown in the next section. The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center, and the Details pane on the right side. These panes will be referenced throughout the Avaya IP Office configuration. Proper licensing as well as standard feature configurations that are not directly related to the interface with the service provider (such as LAN interface to the enterprise site and IP Office Softphone support) is assumed to be already in place.

5.1. LAN Settings

In the sample configuration, the **IPOffice_1** was used as the system name and LAN2 port of Avaya IP Office was used to connect to LAN port of EdgeMarc. To access the LAN settings, first navigate to **System (1) → IPOffice_1** in the Navigation and Group Panes 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 LAN port. Set the **IP Mask** field to the mask used on the public network. All other parameters should be set according to customer requirements. Click **OK** (not shown).



Select the **VoIP** tab as shown in the following screen. The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such the 9600-Series IP Telephones used in the sample configuration. The **SIP Trunks Enable** box must be checked to enable the configuration of SIP trunks to Cox. The **SIP Registrar Enable** box is checked to allow Avaya IP Office Softphone usage. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2. The specific values used for the compliance test are shown in the example below. All other parameters should be set according to customer requirements. Click **OK** (not shown).

The screenshot displays the Avaya IP Office configuration interface for 'IPOffice_1'. The left-hand navigation pane shows a tree structure with 'IPOffice_1' selected, and 'System (1)' highlighted. The main configuration area is titled 'IPOffice_1' and has several tabs: System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, Twinning, VCM, and CC. The 'LAN2' tab is active, and the 'VoIP' sub-tab is selected. The configuration is divided into several sections:

- LAN Settings:** Contains checkboxes for 'H323 Gatekeeper Enable' (checked), 'Auto-create Extn' (unchecked), 'Auto-create User' (unchecked), and 'H323 Remote Extn Enable' (unchecked).
- SIP Settings:** Contains checkboxes for 'SIP Trunks Enable' (checked), 'SIP Registrar Enable' (checked), 'Auto-create Extn/User' (checked), and 'SIP Remote Extn Enable' (unchecked).
- Domain Name:** A text input field.
- Layer 4 Protocol:** A table of settings:

<input checked="" type="checkbox"/>	UDP	UDP Port	5060	Remote UDP Port	5060
<input checked="" type="checkbox"/>	TCP	TCP Port	5060	Remote TCP Port	5060
<input type="checkbox"/>	TLS	TLS Port	5061	Remote TLS Port	5061
- Challenge Expiry Time (secs):** A numeric input field set to 10.
- RTP Section:**
 - Port Number Range:** Minimum: 49152, Maximum: 53246.
 - Port Number Range (NAT):** Minimum: 49152, Maximum: 53246.
 - Enable RTCP Monitoring on Port 5005

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. 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 this configuration, STUN will not be used.
- Set **Binding Refresh Time (seconds)** to **60**. 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. **Public Port** is set to **5060**
- All other parameters should be set according to customer requirements.
- Click **OK** (not shown).

The screenshot displays the configuration window for 'IPOffice_1' in the 'System' tab, specifically the 'Network Topology' sub-tab for the 'LAN2' interface. The 'Network Topology Discovery' section contains the following settings:

STUN Server Address	192.168.10.13	STUN Port	3478
Firewall/NAT Type	Open Internet		
Binding Refresh Time (seconds)	60		
Public IP Address	192 . 168 . 1 . 200		

Below this, the 'Public Port' section is configured as follows:

UDP	5060
TCP	0
TLS	0

The 'Run STUN on startup' checkbox is currently unchecked. The interface also shows a tree view on the left with 'IP Office 1' selected, and various tabs at the top including 'System', 'LAN1', 'LAN2', 'DNS', 'Voicemail', 'Telephony', 'Directory Services', 'System Events', 'SMTP', 'SMDR', and 'Twinnir'.

In the compliance test, the LAN1 interface was used to connect the Avaya IP Office to the enterprise site IP network. The LAN1 interface configuration is not directly relevant to the interface with Cox SIP Trunking service, and therefore is not described in these Application Notes.

5.2. System Telephony Settings

Navigate to the **Telephony** → **Telephony** Tab in the Details Pane. Choose the **Companding Law** typical for the enterprise location. For North America, **ULAW** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the service provider across the SIP trunk. Set **Hold Timeout (secs)** to **600**.

The screenshot shows the configuration window for IOffice_1. The 'Telephony' tab is active. In the 'Analogue Extensions' section, 'Default Outside Call Sequence' is 'Normal', 'Default Inside Call Sequence' is 'Ring Type 1', and 'Default Ring Back Sequence' is 'Ring Type 2'. The 'Hold Timeout (secs)' is set to 600. In the 'Companding Law' section, 'U-Law' is selected for both 'Switch' and 'Line'. The 'Inhibit Off-Switch Forward/Transfer' checkbox is unchecked.

5.3. Twinning Calling Party Settings

When using twinning, the calling party number displayed on the twinned phone is controlled by two parameters. These parameters only affect twinning and do not impact the messaging or operation of other redirected calls such as forwarded calls. The first parameter is the **Send original calling party information for Mobile Twinning** box on the **System** → **Twinning** tab. The second parameter is the **Send Caller ID** parameter on the **SIP Line** form (shown in **Section 5.4**).

For the compliance testing, the **Send original calling party information for Mobile Twinning** as shown below was unchecked. This setting allows **Send Caller ID** parameter that was set to **Diversion Header** in **Section 5.4.2** to be used. IP Office will send the following in the “From” header:

- On calls from an internal extension to a twinned phone, IP Office sends Calling Party Number of the originating extension.
- On calls from the PSTN to a twinned phone, IP Office sends Calling Party Number of the originating PSTN party.
- Click **OK** to commit (not shown).

The screenshot shows the configuration window for IOffice_1, specifically the 'Twinning' tab. The 'Send original calling party information for Mobile Twinning' checkbox is unchecked. Below this checkbox, there is a text box for 'Calling party information for Mobile Twinning'.

5.4. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and Cox 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 IP Office Manager to create a SIP Line. Follow the steps in **Section 5.4.1** to create the SIP Line from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:

- IP addresses.
- SIP Credentials (if applicable).
- SIP URI entries.
- Setting of the **Use Network Topology Info** field on the Transport tab.

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

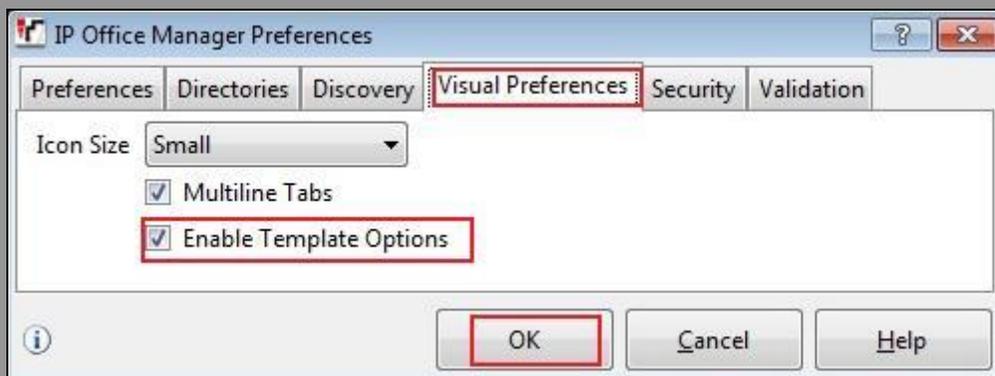
Also, the following SIP Line settings are not supported on Basic Edition:

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

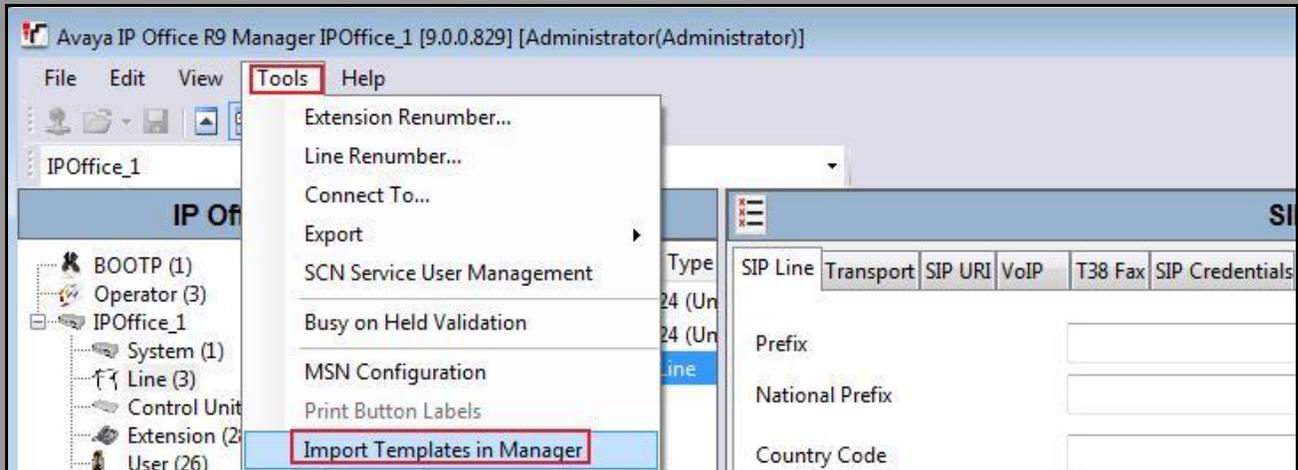
Alternatively, a SIP Line can be created manually. To do so right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.4.2**.

5.4.1. Create SIP line from Template

1. Copy the template file to the computer where IP Office Manager is installed. Rename the template file to **CoxSIPTrkIPO9.xml**. The file name is important in locating the proper template file in **Step 5**.
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. Verify that the box is checked next to **Enable Template Options**. Click **OK**.

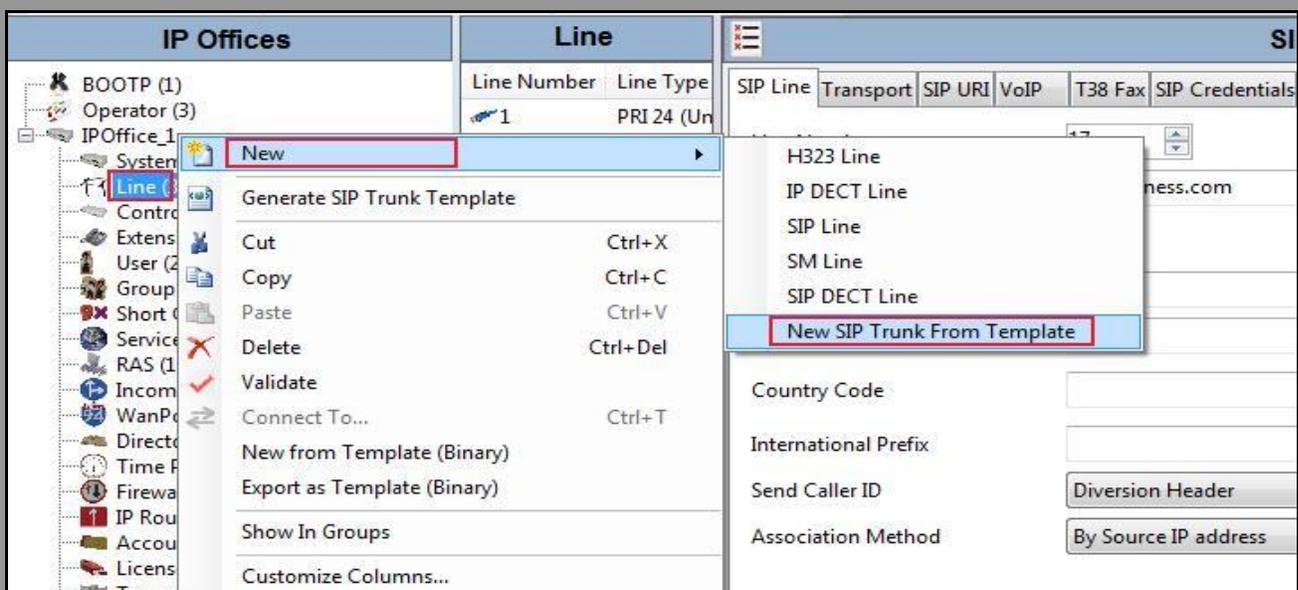


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools** → **Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default template location is **C:\Program Files\Avaya\IP Office\Manager\Templates**.



In the pop-up window (not shown) that appears, 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 IP Office template directory.

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



5. In the subsequent Template Type Selection pop-up window, select **United States** from the **Country** pull-down menu and select **Cox Communications** from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name (**CoxSIPTrkIPO9.xml**) created in **Step 1**. Click **Create new SIP Trunk** to finish creating the trunk.



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

5.4.2. Create SIP Line Manually

To create a SIP line, begin by navigating to **Line** in the left Navigation Pane, then right-click in the Group Pane and select **New** → **SIP Line**. On the **SIP Line** tab in the Details Pane, configure the parameters as shown below:

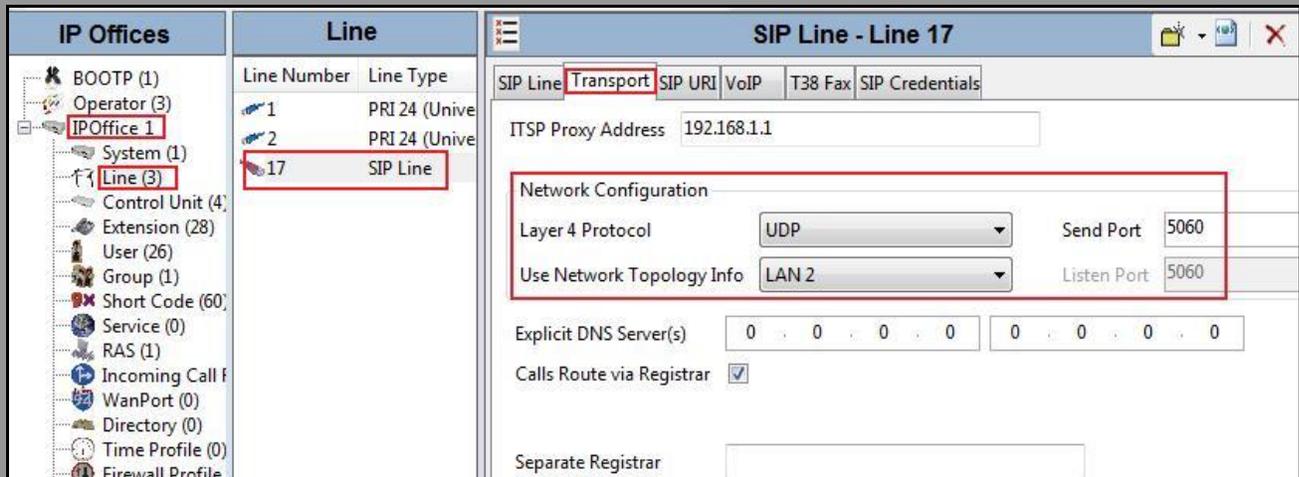
- Set **ITSP Domain Name** to the enterprise domain so that IP Office uses this domain as the host portion of SIP URI in SIP headers such as the From header.
- Set **Send Caller ID** to *Diversion Header*.
- Check the **In Service** box.
- Check the **Check OOS** box. With this option selected, IP Office will use the SIP OPTIONS method to periodically check the SIP Line.
- Default values may be used for all other parameters.

The area of the screen entitled **REFER Support** is used to enable/disable SIP REFER for call transfers. The default values of “Auto” for **Incoming** and **Outgoing** effectively disable use of SIP REFER. To enable SIP REFER, select “Always” from the drop-down menu for **Incoming** and **Outgoing**. In the compliance test, The REFER method was not tested since Cox system does not support REFER at this point. Click OK to commit (not shown).

The screenshot displays the configuration window for a SIP Line (Line 17). The interface is divided into several sections:

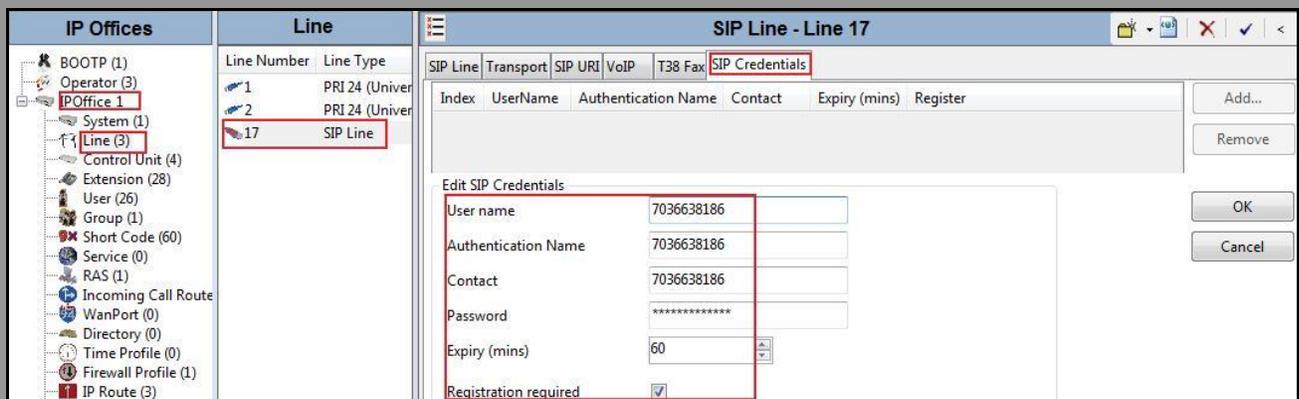
- Line List:** A table showing the configuration for Line 17, which is a SIP Line.
- Configuration Fields:** Fields for Line Number (17), ITSP Domain Name (coxbusiness.com), Prefix, National Prefix, Country Code, International Prefix, Send Caller ID (Diversion Header), and Association Method (By Source IP address).
- Service Options:** Checkboxes for In Service (checked) and Check OOS (checked). A dropdown for URI Type is set to SIP.
- Call Routing:** Call Routing Method (Request URI), Originator number for forwarded and twinning calls, and Name Priority (System Default).
- Caller ID:** Caller ID from From header (checked), Send From In Clear (unchecked), and User-Agent and Server Headers.
- Service Response:** Service Busy Response (503 - Service Unavailable) and Action on CAC Location Limit (Allow Voicemail).
- REFER Support:** A section with a checkbox for REFER Support (unchecked) and dropdowns for Incoming and Outgoing (both set to Auto).
- Session Refresh:** Method for Session Refresh (Reinvite).

Select the **Transport** tab. The **ITSP Proxy Address** is set to the EdgeMarc LAN IP Address. As shown in **Figure 1**, this IP Address is **192.168.1.1**. In the **Network Configuration** area, **UDP** is selected as the **Layer 4 Protocol**, and the **Send Port** is set to the port number provided by Cox. The **Use Network Topology Info** parameter is set to **LAN 2**. This associates the SIP Line with the parameters in the **System → LAN2 → Network Topology** tab. Other parameters retain default values in the screen below. Click OK to commit (not shown).



A SIP Credentials entry must be created for Digest Authentication used by Cox SIP trunking service to authenticate calls from the enterprise to the PSTN. To create a SIP Credentials entry, first select the **SIP Credentials** tab. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane (not shown). To edit an existing entry, click an entry in the list at the top, and click the **Edit...** button (not shown). At the bottom of the screen, the Edit Channel area will be opened. In the example screen below, a previously configured entry is edited. The entry was created with the parameters shown below:

- Set **User name** and **Authentication Name** to the value provided by the service provider.
- Set **Password** to the value provided by the service provider. **Expiry (mins)** is set to **60**
- Check the **Registration required** option. Cox does require registration for Digest Authentication.
- Click **OK** to commit (not shown).



A **SIP URI** entry must be created to match each incoming number that Avaya IP Office will accept on this line. Select the **SIP URI** tab, click the **Add** button and then **New Channel** area will appear at the bottom of the pane (not shown). To edit an existing entry, click an entry in the list at the top, and click the **Edit...** button (not shown). In the example screen below, a previously configured entry is 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** and **Display Name** to *Internal Data*. This setting allows calls on this line which SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.6**.
- Set **PAI** to *Internal Data*. With this setting IP Office will populate the SIP P-Asserted-Identity header on outgoing calls with the data set in the **SIP** tab of the call initiating **User** as shown in **Section 5.6**.
- For **Registration**, select the account credentials previously configured on the line's **SIP Credentials** tab.
- Associate this line with an incoming line group 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, 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** to commit.

The screenshot below shows the SIP URI entry for **Channel 1**

The screenshot displays the Avaya IP Office configuration interface for 'SIP Line - Line 17'. The 'SIP URI' tab is selected, and the 'Edit Channel' dialog is open for Channel 1. The dialog shows the following configuration:

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	17 17	1...					1:70366...	10
2	17 17	1...	703663...	70366...		N...	0: <Non...	10

The 'Edit Channel' dialog for Channel 1 shows the following fields:

- Via: 192.168.1.200
- Local URI: Use Internal Data
- Contact: Use Internal Data
- Display Name: Use Internal Data
- PAI: Use Internal Data
- Registration: 1:7036638186
- Incoming Group: 17
- Outgoing Group: 17
- Max Calls per Channel: 10

SIP URI entry **Channel 2** was similarly created for incoming calls appropriately to pre-define DID numbers **7036638122** to access to Feature Name Extension 00 (FNE00). The Short Codes for FNE00 was defined in **Section 5.5** to provide Dial Tone and Mobile Callback for mobility extension.

The **Channel 2**, as shown in the screenshot below, was configured with following parameters.

- Set the **Local URI** and **Contact** fields to pre-define DID number **7036638122** appropriately for **Channel 2**.
- Associate **Incoming Group** and **Outgoing Group** to SIP Line **17**.
- Set the **Max Calls per Channel** field to **10**.
- Other parameters retain default values.
- Click **OK** to commit.

The screenshot below shows the SIP URI entry for **Channel 2**

The screenshot displays the configuration for SIP Line - Line 17. The 'Edit Channel' dialog is open for Channel 2. The configuration parameters are as follows:

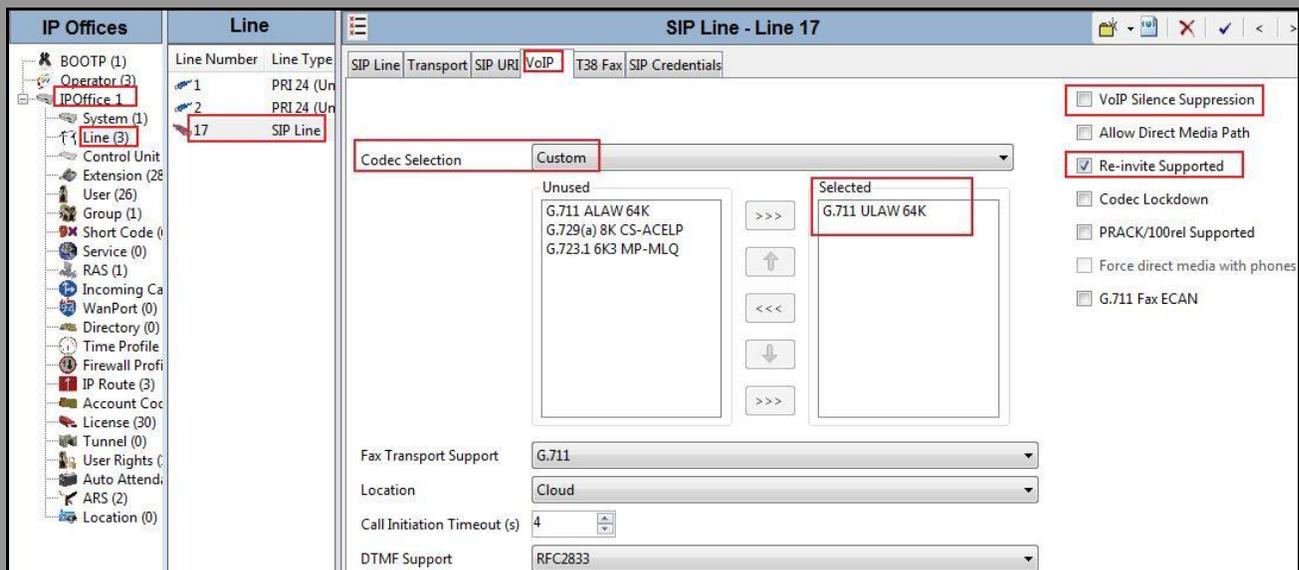
Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	17 17	1...					1: 70366...	10
2	17 17	1...	703663...	70366...	N...	0: <Non...		10

The 'Edit Channel' dialog fields are:

- Via: 192.168.1.200
- Local URI: 7036638122
- Contact: 7036638122
- Display Name: Use Internal Data
- PAI: None
- Registration: 0: <None>
- Incoming Group: 17
- Outgoing Group: 17
- Max Calls per Channel: 10

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

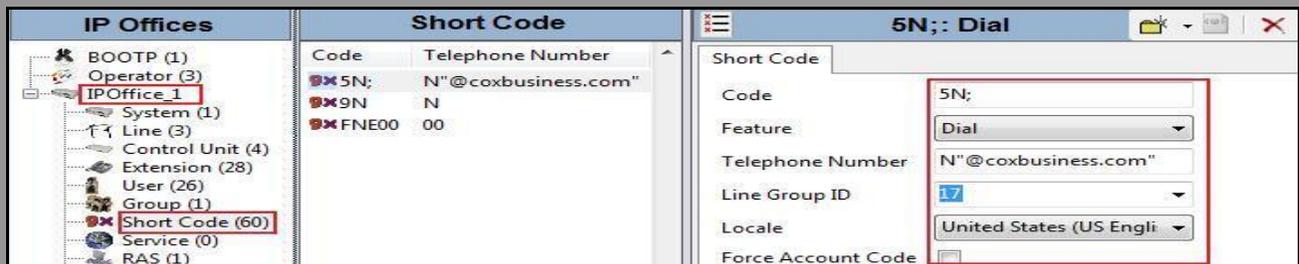
- The **Codec Selection** can be selected by choosing **Custom** from the pull-down menu, allowing an explicit ordered list of codecs to be specified. Select **G.711 ULAW 64K**, codecs cause Avaya IP Office to include this codec, supported by the Cox SIP Trunking service, in the Session Description Protocol (SDP) offer.
- Set **Fax Transport Support** to **G711** from the pull-down menu (T.38 faxing is not currently supported by Cox).
- Set the **DTMF Support** field to **RFC2833** from the pull-down menu. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Uncheck the **VoIP Silence Suppression** box. By unchecking the **VoIP Silence Suppression** box, calls can be established with the G.729 codec but without silence suppression. Since Cox does not support G.729 at this point, this parameter is optional.
- Check the **Re-invite Supported** box.
- Default values may be used for all other parameters.
- Click **OK** to commit (not shown).



5.5. Short Code

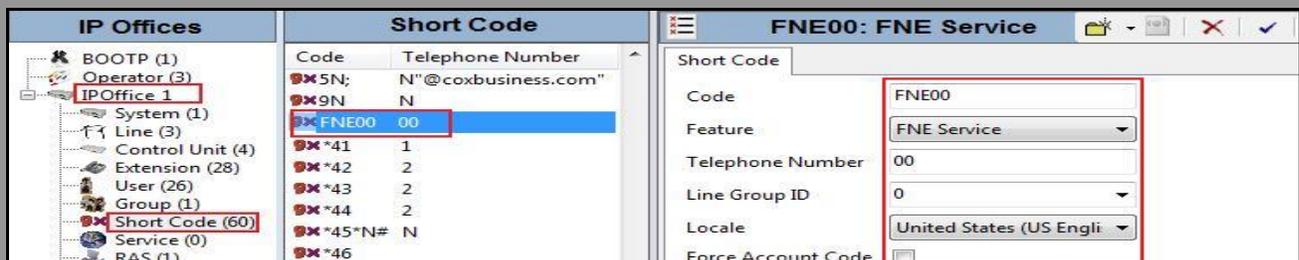
Define a short code to route outbound traffic to the SIP line. To create a short code, select **Short Code** in the left Navigation Pane, then right-click in the Group Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters for the new short code to be created. The screen below shows the details of the previously administered “5N;” short code used in the test configuration.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. In this case, *5N;*, this short code will be invoked when the user dials 5 followed by any number.
- Set **Feature** to *Dial*. This is the action that the short code will perform.
- Set **Telephone Number** to *N"@coxbusiness.com"*. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. The value *N* represents the number dialed by the user. The host part following the “@” is the domain of the service provider network.
- 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**. This short code will use this line group when placing the outbound call.
- Set **Locale** to *United State (US English)*.
- Click OK to commit (not shown).



For incoming calls from mobility extension to FNE features hosted by IP Office to provide **Dial Tone** functionality, Short Code **FNE00** was created. The **FNE00** was configured with the following parameters.

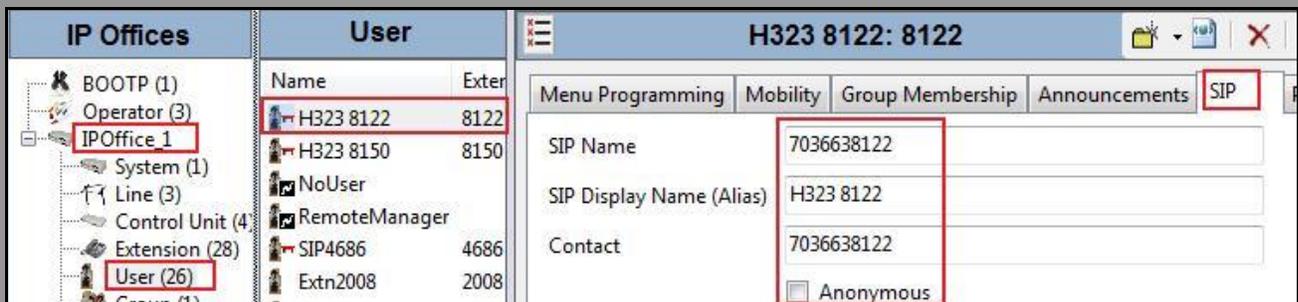
- In the **Code** field, enter the FNE feature code as *FNE00* for **Dial Tone**.
- Set the **Feature** field to *FNE Service*.
- Set the **Telephone Number** field to *00* for **FNE00**.
- Set the **Line Group ID** field to *0*.
- Retain default values for other fields.
- Click OK to commit (not shown).



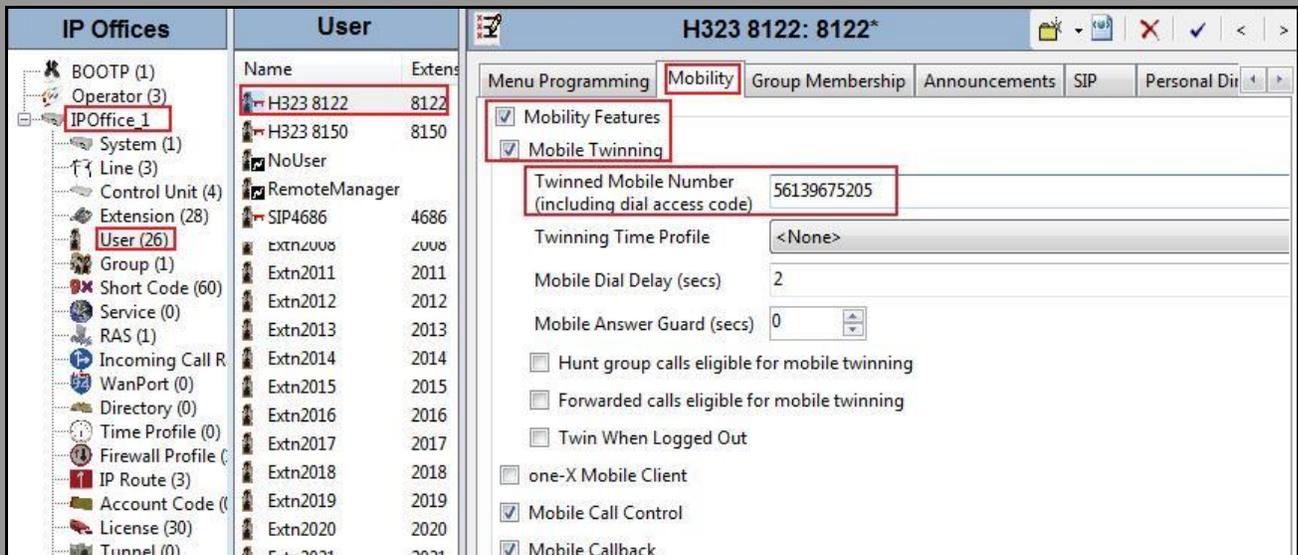
5.6. User

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 select **User** in the left Navigation Pane, 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 “H323-8122”. Select the **SIP** tab in the Details Pane.

The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From header for outgoing SIP trunk calls. They also allow matching of the SIP URI for 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 H323-8122. The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise from Cox. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered as private, then the **Anonymous** box may be checked to withhold the user’s information from the network. Click OK to commit (not shown).



One of the H.323 IP Phones at the enterprise site uses the Mobile Twinning feature. The following screen shows the **Mobility** tab for User H323-8122. 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 mobile telephone, in this case **56139675205**. Other options can be set according to customer requirements. Click OK to commit (not shown).



5.7. Incoming Call Route

An incoming call route maps an inbound DID number on a specific line to an internal extension. This procedure should be repeated for each DID number provided by the service provider. To create an incoming call route, select **Incoming Call Route** in the left Navigation Pane, then right-click in the center Group Pane and select **New**. On the **Standard** tab of the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capacity** to *Any Voice*.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.4**.
- Set the **Incoming Number** to the incoming number on which this route should match.
- Set **Locale** to *United State (US English)*.
- Default values can be used for all other fields.
- Click OK to commit (not shown).

IP Offices	Incoming Call Route	17 7036638122																																																
BOO TP (1) Operator (3) IP Office 1 System (1) Line (3) Control Unit (4) Extension (28) User (26) Group (1) Short Code (60) Service (0) RAS (1) Incoming Call Route WanPort (0) Directory (0) Time Profile (0) Firewall Profile (1) IP Route (3) Account Code (0) License (30) Tunnel (0)	<table border="1"><thead><tr><th>Line Group ID</th><th>Incoming Number</th><th>Desti</th></tr></thead><tbody><tr><td>17</td><td>7036638122</td><td>8122</td></tr><tr><td>17</td><td>7036638150</td><td>8150</td></tr><tr><td>17</td><td>7036638133</td><td>Voice</td></tr><tr><td>17</td><td>7036638126</td><td>4686</td></tr></tbody></table>	Line Group ID	Incoming Number	Desti	17	7036638122	8122	17	7036638150	8150	17	7036638133	Voice	17	7036638126	4686	<table border="1"><thead><tr><th>Standard</th><th>Voice Recording</th><th>Destinations</th></tr></thead><tbody><tr><td>Bearer Capability</td><td>Any Voice</td><td></td></tr><tr><td>Line Group ID</td><td>17</td><td></td></tr><tr><td>Incoming Number</td><td>7036638122</td><td></td></tr><tr><td>Incoming Sub Address</td><td></td><td></td></tr><tr><td>Incoming CLI</td><td></td><td></td></tr><tr><td>Locale</td><td>United States (US English)</td><td></td></tr><tr><td>Priority</td><td>1 - Low</td><td></td></tr><tr><td>Tag</td><td></td><td></td></tr><tr><td>Hold Music Source</td><td>System Source</td><td></td></tr><tr><td>Ring Tone Override</td><td>None</td><td></td></tr></tbody></table>	Standard	Voice Recording	Destinations	Bearer Capability	Any Voice		Line Group ID	17		Incoming Number	7036638122		Incoming Sub Address			Incoming CLI			Locale	United States (US English)		Priority	1 - Low		Tag			Hold Music Source	System Source		Ring Tone Override	None	
Line Group ID	Incoming Number	Desti																																																
17	7036638122	8122																																																
17	7036638150	8150																																																
17	7036638133	Voice																																																
17	7036638126	4686																																																
Standard	Voice Recording	Destinations																																																
Bearer Capability	Any Voice																																																	
Line Group ID	17																																																	
Incoming Number	7036638122																																																	
Incoming Sub Address																																																		
Incoming CLI																																																		
Locale	United States (US English)																																																	
Priority	1 - Low																																																	
Tag																																																		
Hold Music Source	System Source																																																	
Ring Tone Override	None																																																	

On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. In this example, incoming calls to 7036638122 on line 17 are routed to extension **8122**. Click OK to commit (not shown).

IP Offices	Incoming Call Route	17 7036638122																					
BOO TP (1) Operator (3) IP Office 1 System (1) Line (3)	<table border="1"><thead><tr><th>Line Group ID</th><th>Incoming Number</th><th>Desti</th></tr></thead><tbody><tr><td>17</td><td>7036638122</td><td>8122</td></tr><tr><td>17</td><td>7036638150</td><td>8150</td></tr><tr><td>17</td><td>7036638133</td><td>Voice</td></tr></tbody></table>	Line Group ID	Incoming Number	Desti	17	7036638122	8122	17	7036638150	8150	17	7036638133	Voice	<table border="1"><thead><tr><th>Standard</th><th>Voice Recording</th><th>Destinations</th></tr></thead><tbody><tr><td>TimeProfile</td><td>Destination</td><td>Fallback Extension</td></tr><tr><td>Default Value</td><td>8122 H323 8122</td><td></td></tr></tbody></table>	Standard	Voice Recording	Destinations	TimeProfile	Destination	Fallback Extension	Default Value	8122 H323 8122	
Line Group ID	Incoming Number	Desti																					
17	7036638122	8122																					
17	7036638150	8150																					
17	7036638133	Voice																					
Standard	Voice Recording	Destinations																					
TimeProfile	Destination	Fallback Extension																					
Default Value	8122 H323 8122																						

5.8. Privacy/Anonymous Calls

For outbound calls with privacy (anonymous) enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with “restricted” and “anonymous” respectively. Avaya IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing. For the compliance test, Cox does not use either PPI or PAI for the purposes of privacy.

5.9. SIP Options

Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. 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 5.1** 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:

- If no **SIP_OPTIONS_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 44 seconds is used.
- To establish a period less than 42 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 42 secs. The OPTIONS message period will be equal to the **Binding Refresh Time**.
- To establish a period greater than 42 seconds, a **SIP_OPTIONS_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 42 secs. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD**.

To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User** → **noUser** in the Navigation / Group Panes. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button (not shown). 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**.

The screenshot shows the Avaya IP Office configuration interface. On the left, the 'IP Offices' tree is expanded to 'IPOffice 1', and the 'User' list shows 'NoUser' selected. The main pane is titled 'NoUser:' and has the 'Source Numbers' tab selected. The 'Source Number' field is empty, and the 'New Source Number' field contains 'SIP_OPTIONS_PERIOD=2'. The 'OK' button is highlighted with a red box.

The **SIP_OPTIONS_PERIOD** parameter will appear in the list of Source Numbers as shown below. For the compliance test, an OPTIONS period of 1 minute was desired. The **Binding Refresh Time** was set to **60** seconds (1 minute) in **Section 5.1**. The **SIP_OPTIONS_PERIOD** was set to 2 minutes. Avaya IP Office chose the OPTIONS period as the smaller of these two values (1 minute). Click the **OK** button (not shown).

The screenshot shows the Avaya IP Office configuration interface. On the left, the 'IP Offices' tree is expanded to 'IPOffice 1', and the 'User' list shows 'NoUser' selected. The main pane is titled 'NoUser:' and has the 'Source Numbers' tab selected. The 'Source Number' field contains 'SIP_OPTIONS_PERIOD=2'. The 'OK' button is highlighted with a red box.

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.

6. Cox SIP Trunking Configuration

Cox is responsible for the configuration of Cox SIP Trunking service. It is also worth to mention here that Edgewater EdgeMarc 4550 device running in B2BUA mode is part of Cox solution and Cox is responsible for configuration and support it. The customer will need to provide the IP address used to reach the Avaya IP Office at the enterprise. Cox will provide the customer the necessary information to configure the Avaya IP Office SIP connection to Cox. The provided information from Cox includes:

- IP address of the Cox SIP proxy.
- Supported codecs.
- DID numbers.
- IP addresses and port numbers used for signaling or media through any security devices.

7. Verification Steps

The following steps may be used to verify the configuration:

- Verify that a phone connected to PSTN can successfully place a call to the Avaya IP Office with two-way audio.
- Verify that a phone connected to Avaya IP Office can successfully place a call to the PSTN with two-way audio.
- Using a network sniffing tool e.g. Wireshark to monitor the SIP signalling between the enterprise and Cox. The sniffer traces are captured at the public interface of the Avaya SBCE.

Following screenshots show an example incoming call from Cox to the enterprise.

- Incoming INVITE request from Cox.

```
INVITE sip:7036638122@10.10.98.14:5060 SIP/2.0
Via: SIP/2.0/UDP 20.164.94.12:5060;branch=z9hG4bKp3g6b0593cii51qgtqp301nqp2
From: "BELLEVILLE ON"<sip:6139675206@20.164.94.12;user=phone>;tag=SD36v5e01-196394093-1389037203792-
To: "Test - SIP Trunk Atlanta - FCPS"<sip:7036638122@coxbusiness.com;user=phone;access-vasb02=ACCESS-VASB02-e7s8pg0p3s6pe>
Call-ID: SD36v5e01-8c7205fec4db7cb135a0498acdd7ee5c-vrvvfv3
CSeq: 881383593 INVITE
Contact: <sip:20.164.94.12:5060;transport=udp>
Supported: 100rel
Allow: ACK, BYE, CANCEL, INFO, INVITE, OPTIONS, PRACK, REFER, NOTIFY, UPDATE
Accept: multipart/mixed, application/media_control+xml, application/sdp
Max-Forwards: 69
Content-Type: application/sdp
Content-Length: 181

v=0
o=BroadWorks 818053829 1 IN IP4 20.164.94.12
s=-
c=IN IP4 20.164.94.12
t=0 0
m=audio 59946 RTP/AVP 0 101
a=rtpmap:101 telephone-event/8000/1
a=fmtp:101 0-15
a=ptime:20
```

- Outgoing 200OK response from the enterprise.

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 20.164.94.12:5060;branch=z9hG4bKp3g6b0593cii51qgtqp301nqp2
From: "BELLEVILLE ON" <sip:6139675206@20.164.94.12;user=phone>;tag=SD36v5e01-196394093-1389037203792-
To: "Test - SIP Trunk Atlanta - FCPS"
<sip:7036638122@coxbusiness.com;user=phone;access-vasb02=ACCESS-VASB02-e7s8pg0p3s6pe>;tag=X57Ed!HXLqEk!uhzT62E5116459c9f7a
Call-ID: SD36v5e01-8c7205fec4db7cb135a0498acdd7ee5c-vrvvfv3
CSeq: 881383593 INVITE
Contact: <sip:7036638122@10.10.98.14:5060>
Content-Type: application/sdp
Content-Length: 203

v=0
o=UserA 376989473 3344333447 IN IP4 10.10.98.14
s=Session SDP
c=IN IP4 10.10.98.14
t=0 0
m=audio 16428 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

Following screenshots show an example outgoing call from the enterprise to Cox.

- Outgoing INVITE request from the enterprise.

```
INVITE sip:6139675206@coxbusiness.com:5060 SIP/2.0
Via: SIP/2.0/UDP 10.10.98.14:5060;branch=z9hG4bK109445413
From: "H323 8122"
<sip:7036638122@coxbusiness.com>;tag=X57Ed!HXLqEk!uhzT65995B94495decd
To: <sip:6139675206@coxbusiness.com>
Call-ID: 2024392884@10.10.98.14
CSeq: 807 INVITE
Contact: <sip:7036638122;tgrp=tg4770489482131;trunk-context=coxbusiness.com@10.10.98.14:5060;transport=udp;user=phone>
Authorization: Digest username="7036638186", realm="BroadWorks",
nonce="BroadWorksXhq453divT6558a0BW", uri="sip:coxbusiness.com",
response="8fa2df750dbd195405899f0c40fcab75", algorithm=MD5, cnonce="52ca972cf3683488",
qop=auth, nc=00000001
Max-Forwards: 70
User-Agent: ewb2bua/11.6.14
Content-Type: application/sdp
Content-Length: 251

v=0
o=UserA 2893914044 3686528984 IN IP4 10.10.98.14
s=Session SDP
c=IN IP4 10.10.98.14
t=0 0
m=audio 16448 RTP/AVP 18 0 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

- Incoming 200OK response from Cox.

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.10.98.14:5060;branch=z9hG4bK109445413
From: "H323 8122"
<sip:7036638122@coxbusiness.com>;tag=X57Ed!HXLqEk!uhzT65995B94495decd
To: <sip:6139675206@coxbusiness.com>;tag=SDvabl999-698133581-1389037553575
Call-ID: 2024392884@10.10.98.14
CSeq: 807 INVITE
Contact: <sip:20.164.94.12:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Accept: multipart/mixed,application/media_control+xml,application/sdp
Content-Type: application/sdp
Content-Length: 189

v=0
o=BroadWorks 818138391 1 IN IP4 20.164.94.12
s=-
c=IN IP4 20.164.94.12
t=0 0
m=audio 59732 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

8. Conclusion

Cox SIP Trunking passed compliance testing. These Application Notes describe the procedures required to configure the SIP connection between Avaya IP Office and Cox SIP Trunking service as shown in **Figure 1**.

9. Additional References

- [1] IP Office 9.0 Installation, Document number 15-601042 Issue 28, 11 October 2013
- [2] IP Office 9.0 Manager 9.0, Document number 15-601011 Issue 9.01, 09 September 2013
- [3] IP Office 9.0 Administering Voicemail Pro, Document number 15-601063 Issue 9.0 Release 1.0, September 2013
- [4] IP Office Embedded Voicemail User Guide (IP Office Mode), Document number 15-604067 Issue 9.0, 10 September 2013

Product documentation for Avaya products may be found at <http://support.avaya.com>. Additional IP Office documentation can be found at: <http://marketingtools.avaya.com/knowledgebase/>

Product documentation for Cox SIP Trunking is available from Cox.

10. Change History

Issue	Date	Reason
0.1	1/10/2014	Initial issue

©2014 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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.