



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Fonolo In-Call Rescue with Avaya IP Office Server Edition using SIP Trunks – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Fonolo In-Call Rescue application to interoperate with Avaya IP Office Server Edition using SIP trunks. In-Call Rescue provides functionality to replace hold-time with a call-back.

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

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

1. Introduction

These Application Notes describe the configuration steps required for Fonolo In-Call Rescue (ICR) to interoperate with Avaya IP Office Server Edition (IP Office) using SIP trunks. ICR provides functionality to replace hold-time with a call-back and during this compliance testing was hosted on the cloud by Fonolo. The solution communicates via SIP/RTP. The ICR functionality was compliance tested utilizing SIP trunks to IP Office. The configuration allowed IP Office to use SIP trunking for calls to and from the ICR application. The ICR is a contact center solution where instead of a caller staying in the queue when agents are all busy, can request to get a call back when an agent becomes available.

When a caller encounters a scenario where no agents are available in a call center environment and IP Office is part of that environment, the caller is presented with options by the call center to either continue waiting in the queue or receive a call back from the call center. If the caller chose the latter, then the call center directs the caller to ICR via IP Office SIP trunks where ICR then provides a message to the caller to leave a call back number, so that ICR can call back the caller when an agent becomes available. Once ICR receives the confirmed call back number from the caller, ICR uses SIP trunk with IP Office to call back into the call center) and wait in the queue until an agent becomes available. When an agent becomes available, ICR informs the agent that there is a call waiting and if the agent would like to get connected to the caller. If the agent accepts to connect to the caller, ICR then calls the caller via SIP trunks to IP Office and connects the caller with the available agent. When ICR makes an outbound call to the caller and agent via IP Office, it makes two SIP INVITE requests. One to the available agent and one to the caller and then mixes the audio within the ICR server.

The IP Office Server Edition configuration consisted of two IP Office systems, a primary Linux server and an expansion IP500V2 that were connected via Small Community Network (SCN) trunks.

The SIP trunks connection from ICR can be with either the primary Linux server or the expansion IP500V2 IP Office system. The configuration shown in these Application Notes used the primary Linux server IP Office system for SIP trunks connectivity.

For security purposes public and Lab IP addresses have been altered in this document.

2. General Test Approach and Test Results

The interoperability compliance testing focused on verifying inbound and outbound calls flows between IP Office and ICR. The feature test cases were performed manually. Calls were placed manually from users on the PSTN to a call center Control Directory Number (CDN). During compliance testing Avaya Contact Center Select (Contact Center Select) was used to emulate a call center. Assumption was made during compliance testing in the Contact Center Select script to direct callers to ICR when no agents are available. When caller connected with ICR, ICR read the call back number of the caller or asked caller to input a new call back number. ICR recognized the Dual Tone Multi Frequency (DTMF) input provided by the caller confirming the call back number. For compliance testing purposes, agents were made available after the above

call between caller and ICR is completed. ICR then called into the call center CDN and connected with an available agent. ICR provided a recording informing the agent of a call in waiting and if the agent wants to get connected to the PSTN caller. Agent accepted the call by using DTMF input. ICR then made the second outbound call to the PSTN caller via IP Office and if the PSTN caller answered the call he/she then get connected with the agent. During compliance testing agents were present on both primary and expansion IP Office systems.

The serviceability test cases were performed manually by disconnecting and reconnecting the SIP trunk connection to ICR.

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and Fonolo utilized enabled capabilities of TLS and SRTP.

This test was conducted in a lab environment simulating a basic customer enterprise network environment. The testing focused on the standards-based interface between the Avaya solution and the third party solution. The results of testing are therefore considered to be applicable to either a premise-based deployment or to a hosted or cloud deployment where some elements of the third party solution may reside beyond the boundaries of the enterprise network, or at a different physical location from the Avaya components.

Readers should be aware that network behaviors (e.g. jitter, packet loss, delay, speed, etc.) can vary significantly from one location to another, and may affect the reliability or performance of the overall solution. Different network elements (e.g. session border controllers, soft switches, firewalls, NAT appliances, etc.) can also affect how the solution performs.

If a customer is considering implementation of this solution in a cloud environment, the customer should evaluate and discuss the network characteristics with their cloud service provider and network organizations, and evaluate if the solution is viable to be deployed in the cloud.

The network characteristics required to support this solution are outside the scope of these Application Notes. Readers should consult the appropriate Avaya and third party documentation for the product network requirements. Avaya makes no guarantee that this solution will work in all potential deployment configurations.

2.1. Interoperability Compliance Testing

The ICR application is hosted in a cloud environment by Fonolo. SIP trunks were used to connect the ICR application with IP Office. The following features and functionality were covered during compliance testing:

- Establishment of SIP trunks connectivity between ICR and IP Office including session refresh.
- Testing of G.711MU codec.
- Incoming calls to a CDN of Contact Center Select via IP Office can be redirected to the ICR application via the SIP trunks based on Contact Center Select scripting. Outgoing calls from ICR to the CDN via IP Office when callers decide on Call back. During this compliance testing Contact Center Select was used to simulate a call center environment and is not the scope of these Application Notes.
- The ICR application can make outbound call to the caller via IP Office who had selected the call back option and merge the call between the caller and available agents (on both primary and expansion) . The outbound call is made via IP Office using SIP INVITE.
- DTMF transmission to ensure that options selected by the caller and agent is accepted correctly by ICR.

Serviceability testing focused on verifying the ability of ICR to recover from adverse conditions, such as the SIP trunks going down and reboot of IP Office.

2.2. Test Results

All test cases were executed and passed with the following exceptions/observations:

- ICR only supports G.711 codec variants.
- ICR only supports RFC2833 for DTMF transmission.

2.3. Support

Technical support on Fonolo ICR can be obtained through the following:

- **Phone:** 1-855-366-2500 (Toll-free)
- **Web:** <https://fonolo.com/contact/>
- **Email:** support@fonolo.com

3. Reference Configuration

A simulated enterprise site consisting of IP Office, Contact Center Select and Avaya telephones were used during compliance testing. As shown in **Figure 1**, SIP trunks were used to connect ICR with IP Office directly. IP Office is connected to an emulated PSTN using T1/PRI. A CDN and skillset is configured on Contact Center Select with a few agents belonging to the configured skillset. The configuration allowed the enterprise site to use SIP trunking for calls to and from ICR and IP Office.

During compliance testing inbound calls to Fonolo were sent to two of Fonolo's specific servers and outbound calls from Fonolo came from four of Fonolo's other servers. This architecture was implemented by Fonolo due to some PBX vendors cannot support inbound and outbound calls on the same SIP trunk. Due to this design intent of Fonolo, inbound and outbound calls to and from Fonolo were handled by different servers. All these servers were hosted on the cloud by Fonolo.

A CDN of 33000 was configured on Contact Center Select and agents from IP Office on both primary and expansion were logged into the pertinent skillset.

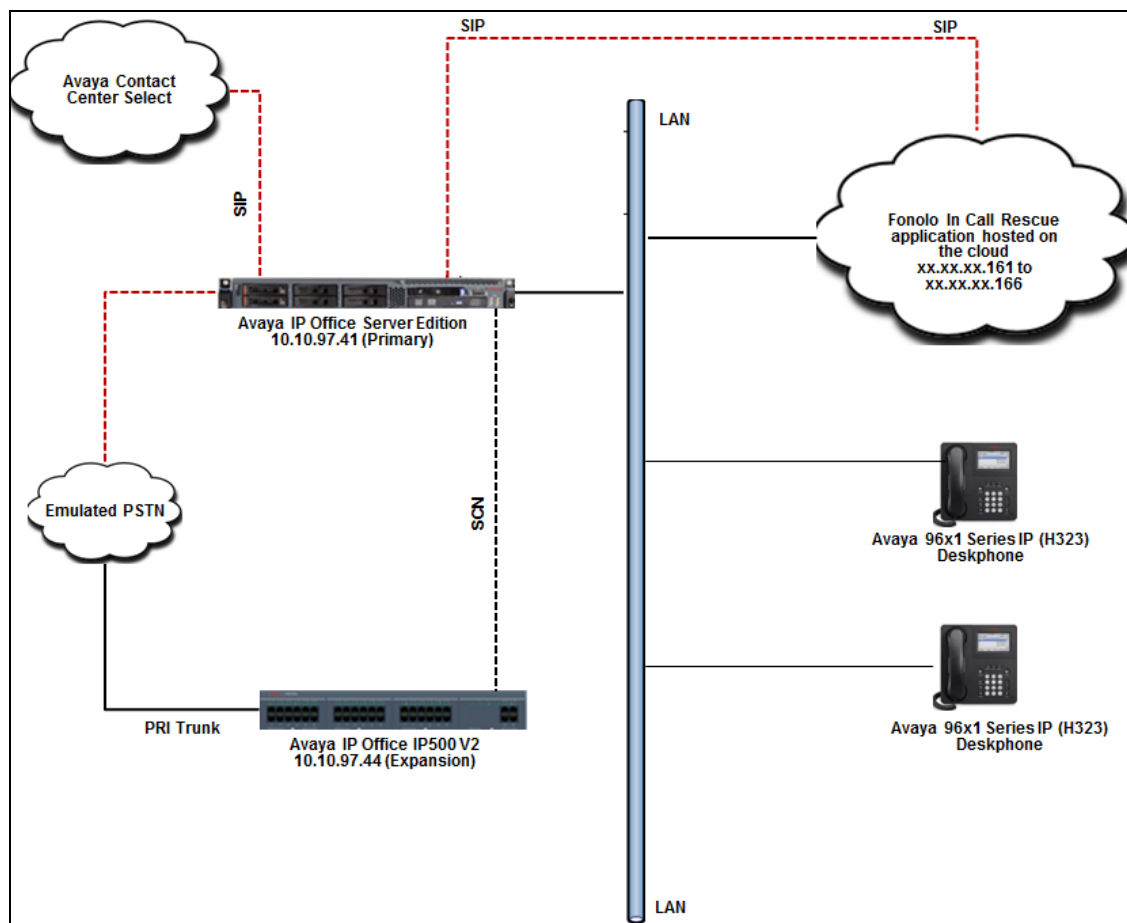


Figure 1: Avaya IP Office Network with Fonolo In-Call Rescue

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office Linux (Primary)	10.1.0.0.0 build 237
Avaya IP Office IP500 V2 (Expansion)	10.1.0.0.0 build 237
Avaya Contact Center Select installed on VMware	7.0.1.1
Avaya Telephones: <ul style="list-style-type: none">• 9650 IP (H323) Deskphone• 1140 IP (SIP) Deskphone• 9641 IP (H323) Deskphone	3.270B 04.04.23.00 6.6401
Fonolo In-Call Rescue	3.0

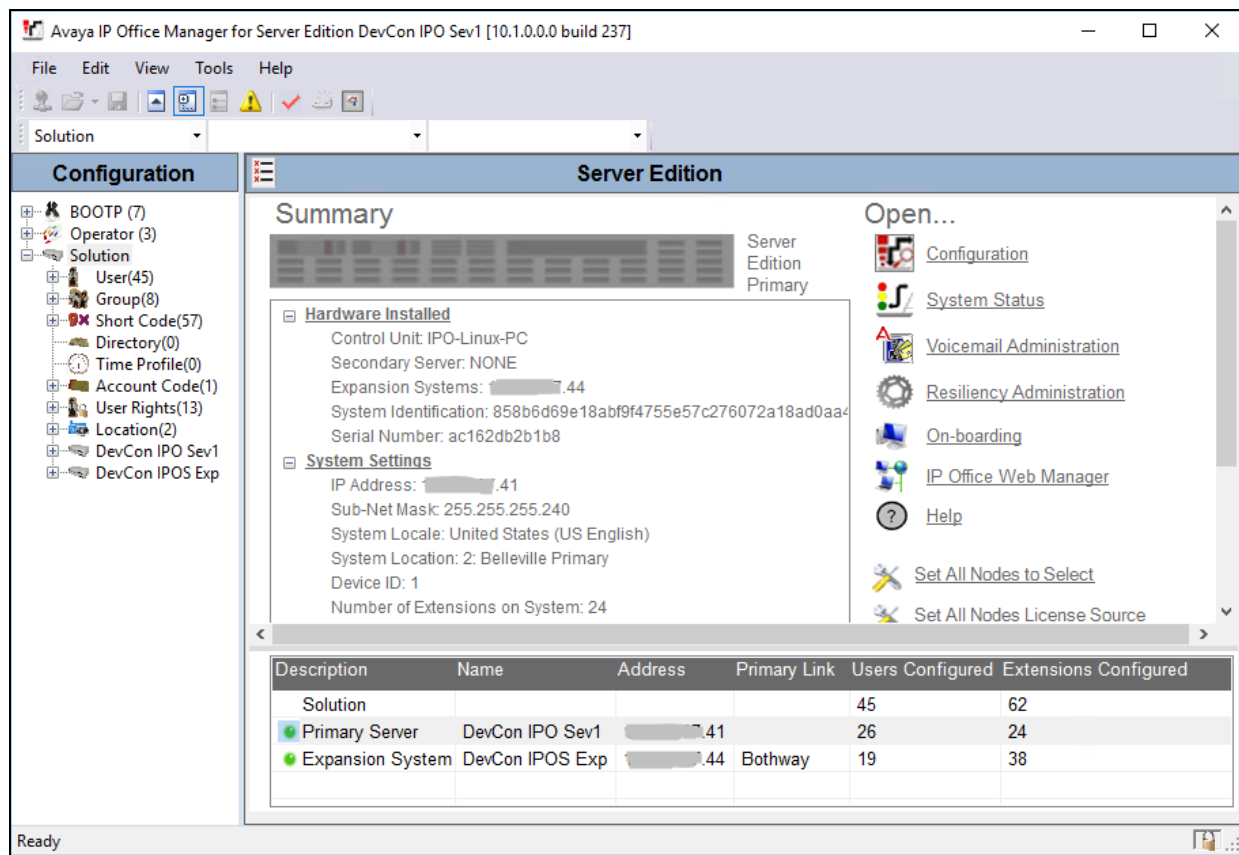
Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office IP500 V2 and also when deployed with IP Office Server Edition in all configurations.

5. Configure Avaya IP Office

Configuration and verification operations on the IP Office illustrated in this section were all performed using Avaya IP Office Manager. The information provided in this section describes the configuration done on the Primary (Linux server) system. It is implied a working system is already in place with the necessary licensing. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**. The configuration operations described in this section can be summarized as follows:

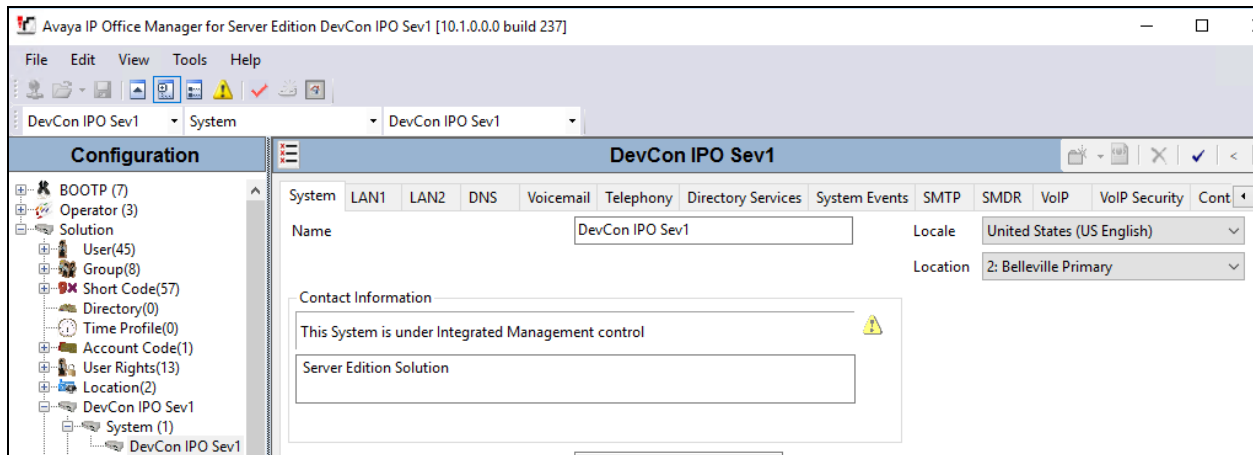
- Configure System Locale
- Configure System
- Create SIP Lines
- Configure Incoming Call Route
- Create Short Code (Route Calls)
- Save Configuration

From a PC running the IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the Manager application. Select the proper IP Office system. In this case the primary server and log in using the appropriate credentials. The **Avaya IP Office Manager for Server Edition** screen is displayed as shown below.



5.1. Configure System Locale

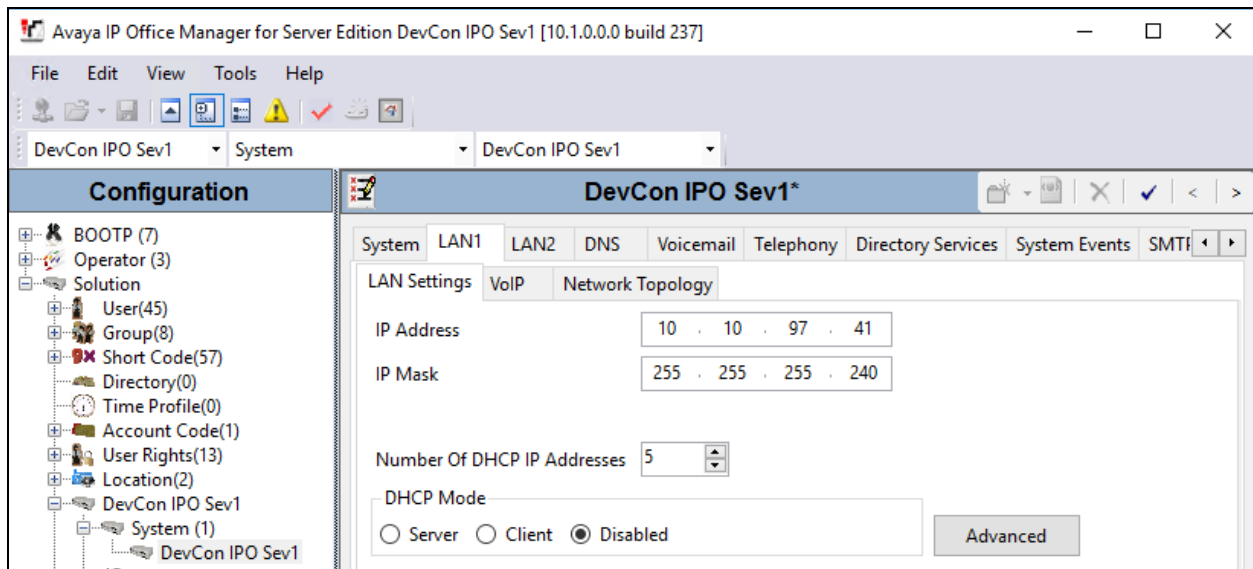
The Locale is usually the country where the IP Office is installed. By selecting the correct country a number of system defaults for that country will be used by the IP Office. To configure the Locale, select **DevCon IPO Sev1** → **System** → **DevCon IPO Sev1** from the IP Office Configuration Tree. During compliance testing the System was called **DevCon IPO Sev1** for the Primary Server and **DevConIPOS Exp** for the Expansion IP500V2. In the right hand pane select the **System** tab, and from the **Locale** dropdown menu select the appropriate country (i.e. **United States (US English)**) as shown in the screen below. Click the **OK** button to save.



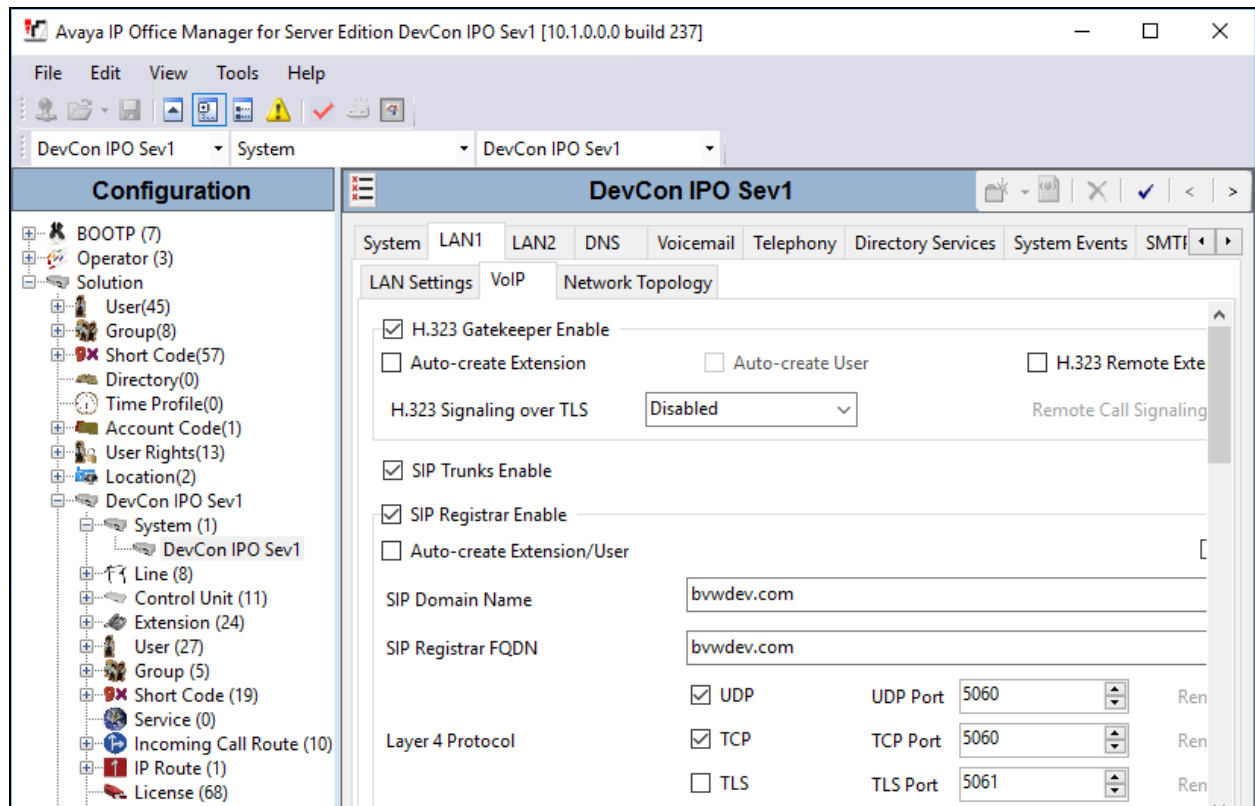
5.2. Configure System

From the configuration tree in the left pane, select **DevCon IPO Sev1** → **System** → **DevCon IPO Sev1** to display the screen in the right pane, where **DevCon IPO Sev1** is the name of the IP Office system.

Select the **LAN1** tab, IP Office can support SIP on the LAN1 and/or LAN2 interfaces, however during compliance testing the LAN1 interface was used. From the **LAN Settings** sub-tab, note the **IP Address** configured, which is **10.10.97.41**. This IP address is required during ICR configuration.



Select the **VoIP** sub-tab. Ensure that **SIP Trunks Enable** and **SIP Registrar Enable** boxes are checked, as shown below.

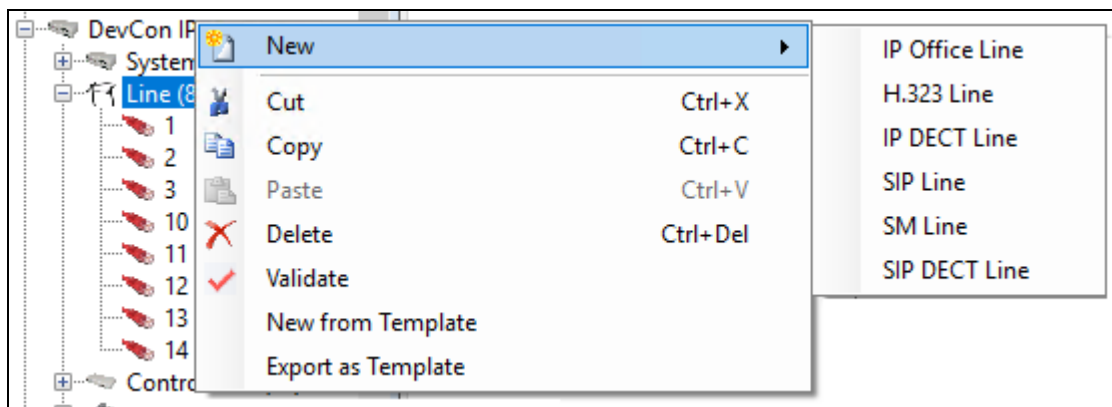


5.3. Create SIP Lines

During compliance testing two SIP lines were created since ICR used different servers for inbound and outbound calls. One line was for outgoing calls from IP Office to ICR, which is an inbound call for ICR and the other line was for incoming calls to IP Office from ICR, which is an outbound call for ICR.

5.3.1. Configure Outgoing SIP Line

To create the SIP line from the IP Office for outgoing calls to ICR, navigate to **System → Line** and right click on **Line** followed by **New → SIP Line** as shown in the screen below. In this example, line **13** was created to connect to ICR. In terms of ICR, this is an inbound call coming to ICR from IP Office.



Screen below shows the configuration of the SIP Line in the subsequent **SIP Line** window; enter the following in the **SIP Line** tab.

- **Line Number:** Line number 13 was configured.
- **ITSP Domain Name:** Configure domain name of ICR or leave this field blank.
- **Description:** Provide a description for the SIP Line. This field is optional.

Retain default values for all remaining fields.

The screenshot displays the Avaya IP Office Manager for Server Edition DevCon IPO Sev1 [10.1.0.0.0 build 237] interface. The main window is titled "SIP Line - Line 13*" and contains several tabs: SIP Line, Transport, SIP URI, VoIP, SIP Credentials, SIP Advanced, and Engineering. The "SIP Line" tab is active, showing the configuration for Line 13. The left sidebar shows a tree view of the system configuration, including BOOTP (7), Operator (3), Solution, User(45), Group(8), Short Code(57), Directory(0), Time Profile(0), Account Code(1), User Rights(13), Location(2), DevCon IPO Sev1, System (1), Line (8), Control Unit (1), Extension (24), User (27), Group (5), Short Code (19), Service (0), and Incoming Call (1). The main configuration area for Line 13 includes the following fields and options:

- Line Number: 13
- ITSP Domain Name: (blank)
- Local Domain Name: (blank)
- URI Type: SIP
- Location: Cloud
- Prefix: (blank)
- National Prefix: 0
- International Prefix: 00
- Country Code: (blank)
- Name Priority: System Default
- Description: Outgoing to Fonolo
- In Service: ☒
- Check OOS: ☒
- Session Timers: Refresh Method: Auto, Timer (sec): On Demand
- Redirect and Transfer: Incoming Supervised REFER: Auto, Outgoing Supervised REFER: Auto, Send 302 Moved Temporarily: ☐, Outgoing Blind REFER: ☐

At the bottom of the window, there are buttons for OK, Cancel, and Help. The status bar at the bottom left shows "Ready".

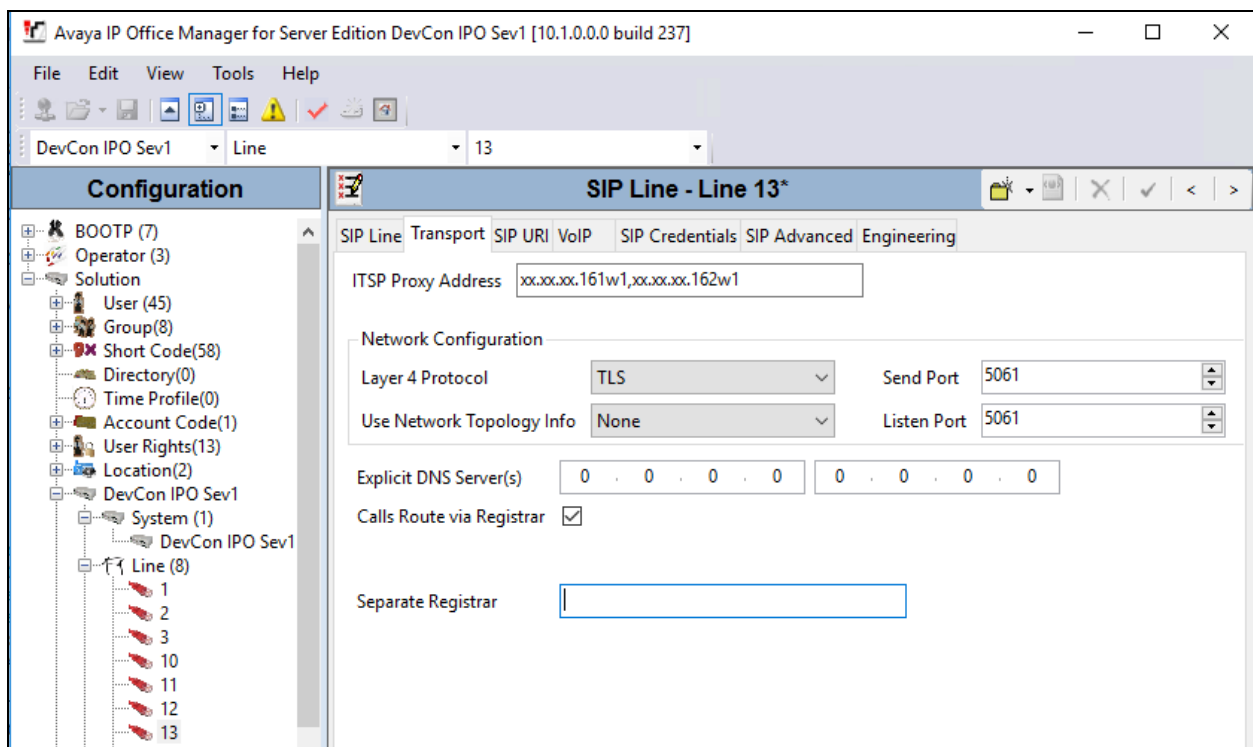
In the **Transport** tab enter IP address of ICR in the **ITSP Proxy Address** field. During compliance testing *xx.xx.xx.161w1,xx.xx.xx.162w1* was configured. ICR requested the use of two servers for outgoing calls.

In the **ITSP Proxy Address** field, a list of up to four IP addresses, with each address separated by a comma or space can be configured as per IP Office design. The addresses can include an indication of the relative call weighting of each address compared to the others. This is done by adding a *wN* suffix to the address where *N* is the weighting value.

For example during compliance testing for IP addresses *xx.xx.xx.161w1, xx.xx.xx.162w1*, the weighting values assign 1 times the weight of calls to both addresses. The default weight if not specified is 1. A weight of 0 can be used to disable an address.

Under **Network Configuration**, in the **Layer 4 Protocol** field, select *TLS* from the drop down menu.

Retain default values for remaining fields. For compliance testing TLS protocol was tested.



In the **SIP URI** tab click on the **Add** button. In the subsequent window, enter the following:

- **Local URI:** Select *Auto*.
- **Contact:** Select *Auto*.
- **Display Name:** Select *Auto*.
- **Incoming Group:** Select an available group number. During compliance testing *13* was selected.
- **Outgoing Group:** Select an available group number. During compliance testing *13* was selected.
- **Max Sessions:** During compliance testing *10* were configured.

Retain default values for all other remaining fields and click the **OK** button.

Avaya IP Office Manager for Server Edition DevCon IPO Sev1 [10.1.0.0.0 build 237]

File Edit View Tools Help

DevCon IPO Sev1 Line 13

Configuration

SIP Line - Line 13

SIP Line Transport SIP URI VoIP SIP Credentials SIP Advanced Engineering

URI	Groups	Local URI	Contact	Display Name	Identity	Header	Originator Number	Send Caller ID
-----	--------	-----------	---------	--------------	----------	--------	-------------------	----------------

Add... Remove Edit...

New URI

Local URI Auto

Contact Auto

Display Name Auto

Identity

Identity None

Header P Asserted ID

Forwarding And Twinning

Originator Number

Send Caller ID None

Diversion Header None

Registration 0: <None>

Incoming Group 13

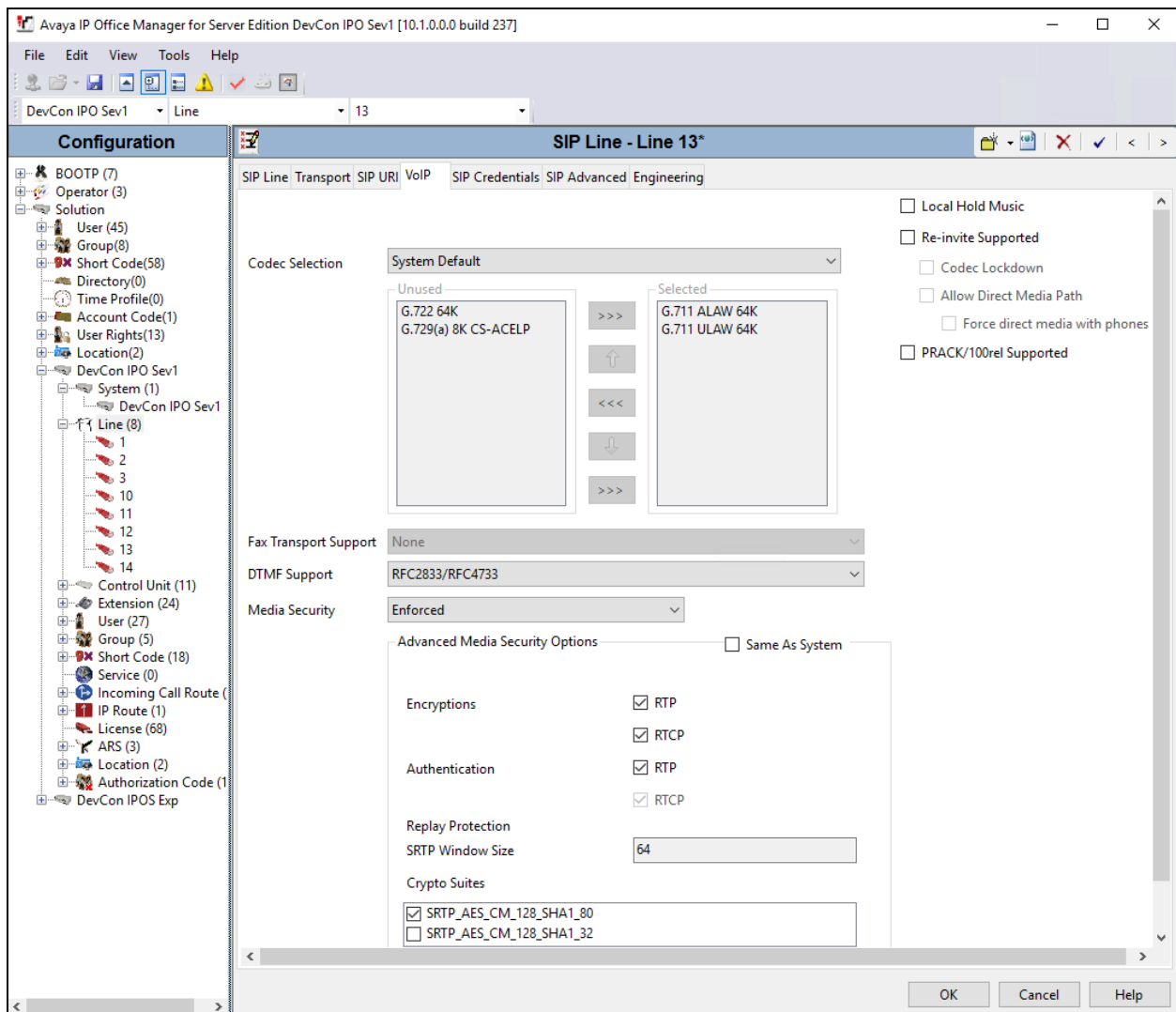
Outgoing Group 13

Max Sessions 10

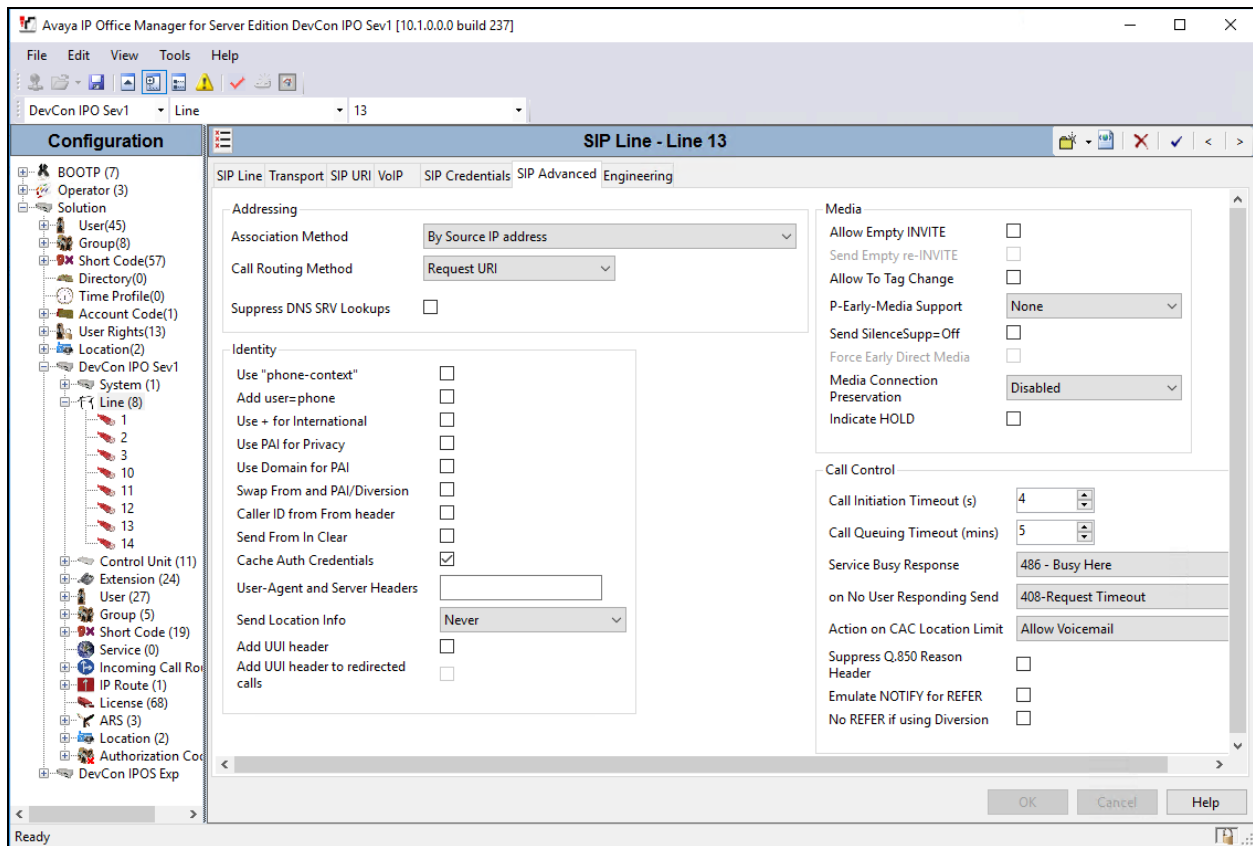
OK Cancel

In the **VoIP** tab ensure that for **DTMF Support**, *RFC2833/RFC4733* is selected from the drop down menu. Select *Enforced* from the drop down menu for **Media Security**. Check all boxes for **Encryptions** and **Authentication** fields and under **Crypto Suites** check the box for *SRTP_AES_CM_128_SHA1_80*.

Retain default values for all remaining fields. During compliance testing only the **G.711 ULAW** codec was tested as shown in the screen below.

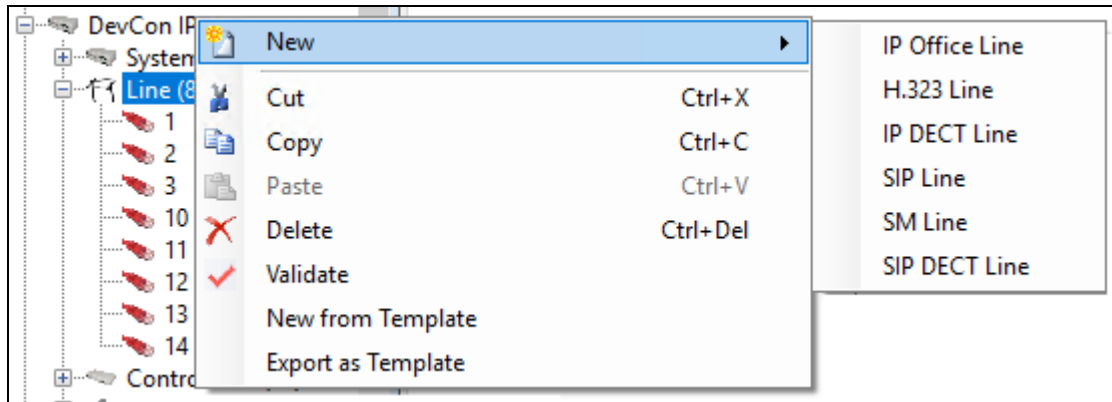


Default values were used for the remaining fields in the **SIP Credentials**, **SIP Advanced** and **Engineering** tabs. Screen below shows an example of the **SIP Advanced** tab with the default values. Click on the **OK** button to complete the configuration of the new SIP Line.



5.3.2. Configure Incoming SIP Line

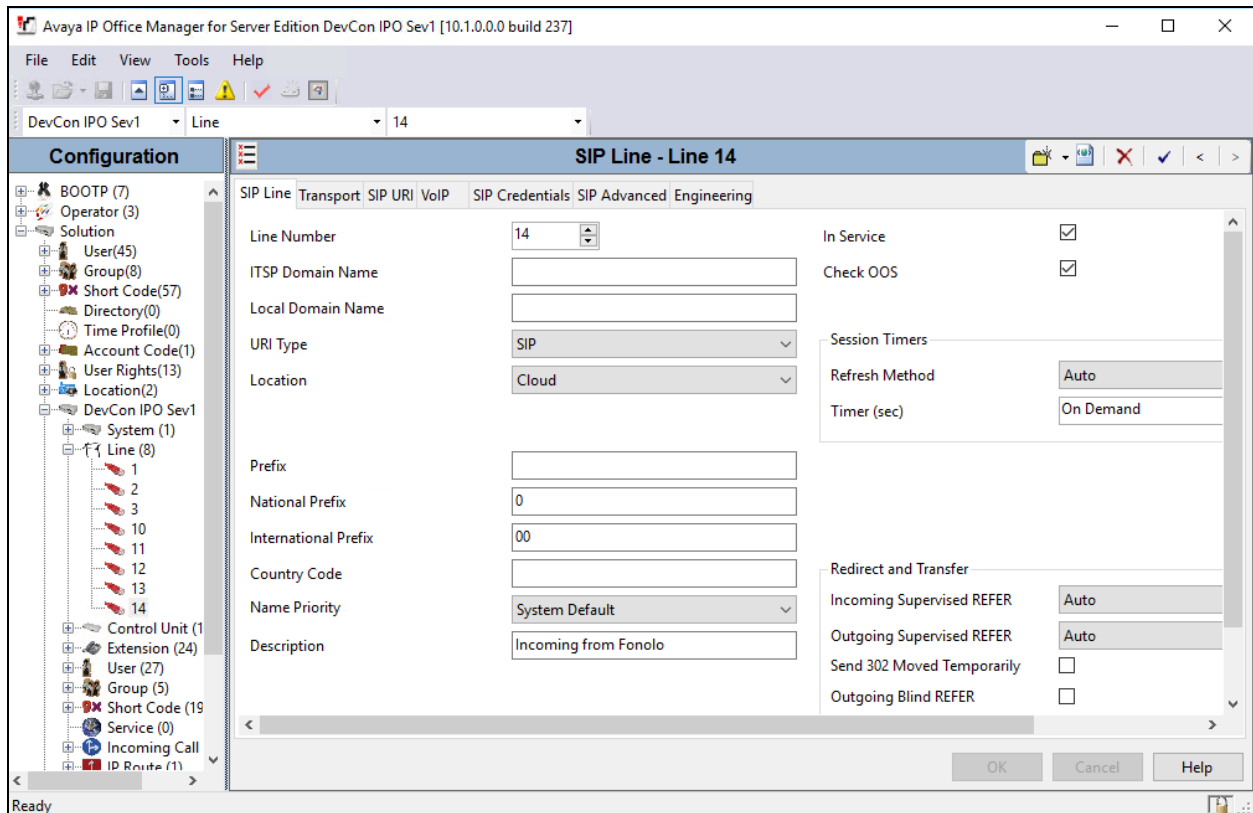
To create the SIP line from the IP Office for incoming calls from ICR, navigate to **System** → **Line** and right click on **Line** followed by **New** → **SIP Line** as shown in the screen below. In this example, line **14** was created to connect to ICR. In terms of ICR, this is an outbound that ICR makes to IP Office.



Screen below shows the configuration of the SIP Line in the subsequent **SIP Line** window; enter the following in the **SIP Line** tab.

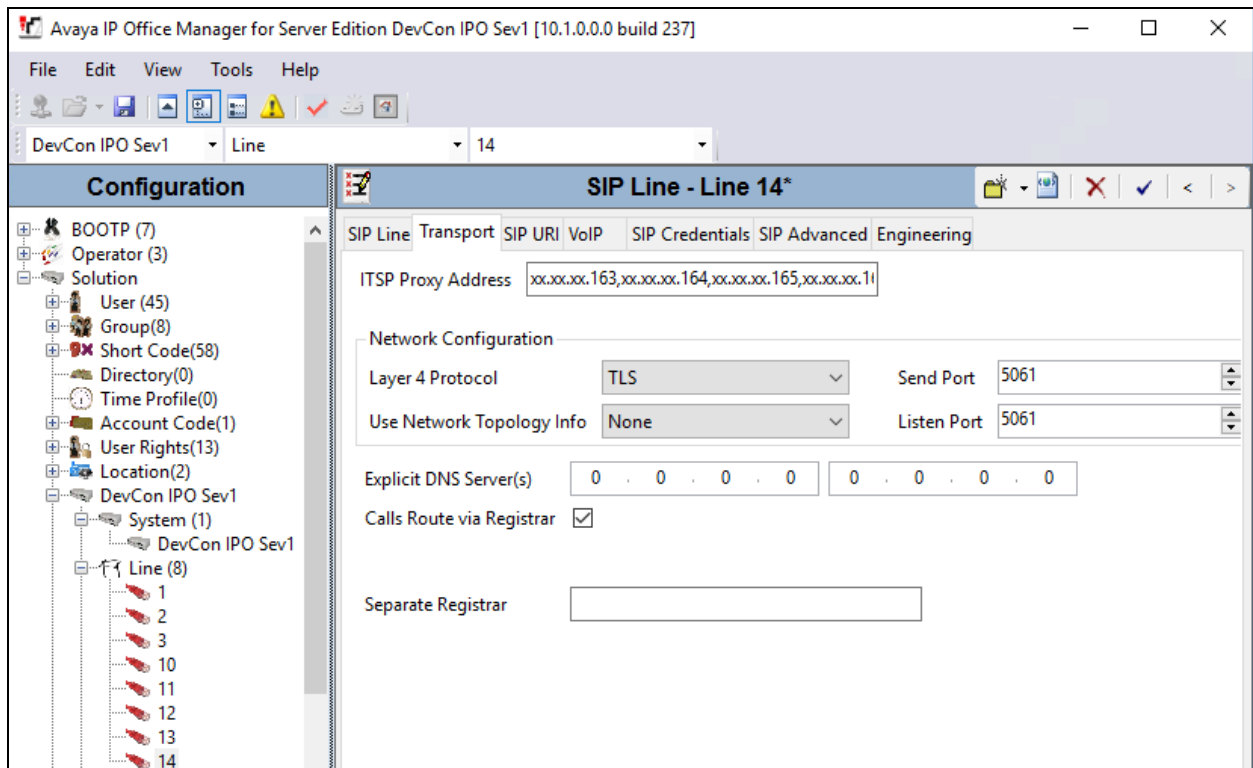
- **Line Number:** Line number *14* was configured.
- **ITSP Domain Name:** Configure domain name of ICR or leave this field blank.
- **Description:** Provide a description for the SIP Line. This field is optional.

Retain default values for all remaining fields.



In the **Transport** tab enter IP address of ICR in the **ITSP Proxy Address** field. During compliance testing *xx.xx.xx.163,xx.xx.xx.164,xx.xx.xx.165,xx.xx.xx.166* was configured. ICR requested the use of four servers for incoming calls. In this field, a list of up to four IP addresses, with each address separated by a comma or space can be configured as per IP Office design. Under **Network Configuration**, in the **Layer 4 Protocol** field, select *TLS* from the drop down menu.

Retain default values for remaining fields. For compliance testing TLS protocol was tested.



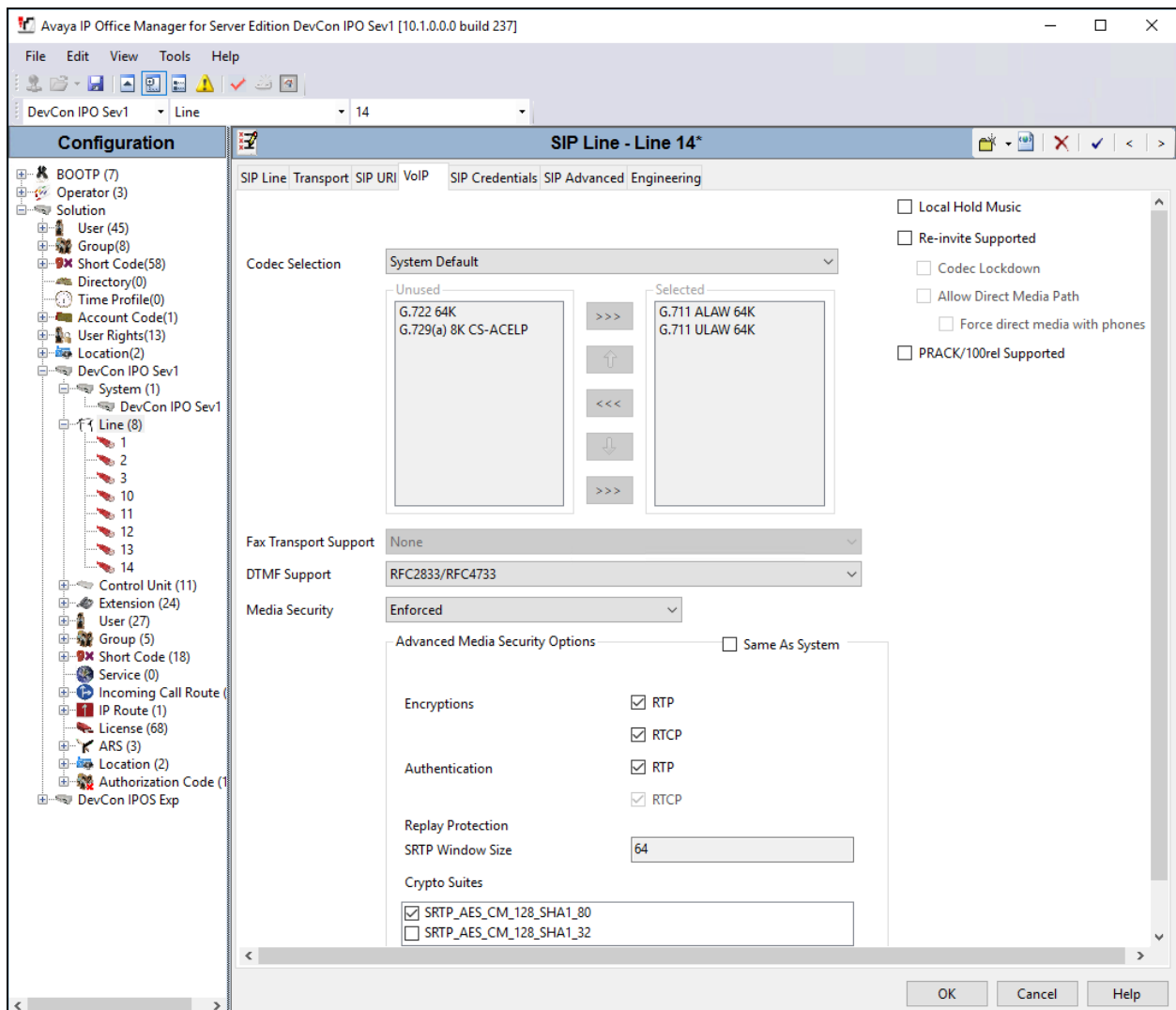
In the **SIP URI** tab click on the **Add** button. In the subsequent window, enter the following and retain default values for all other remaining fields and click the **OK** button.

- **Local URI:** Select *Auto*.
- **Contact:** Select *Auto*.
- **Display Name:** Select *Auto*.
- **Incoming Group:** Select an available group number. During compliance testing *14* was selected.
- **Outgoing Group:** Select an available group number. During compliance testing *14* was selected.
- **Maximum Sessions:** During compliance testing *10* were configured.

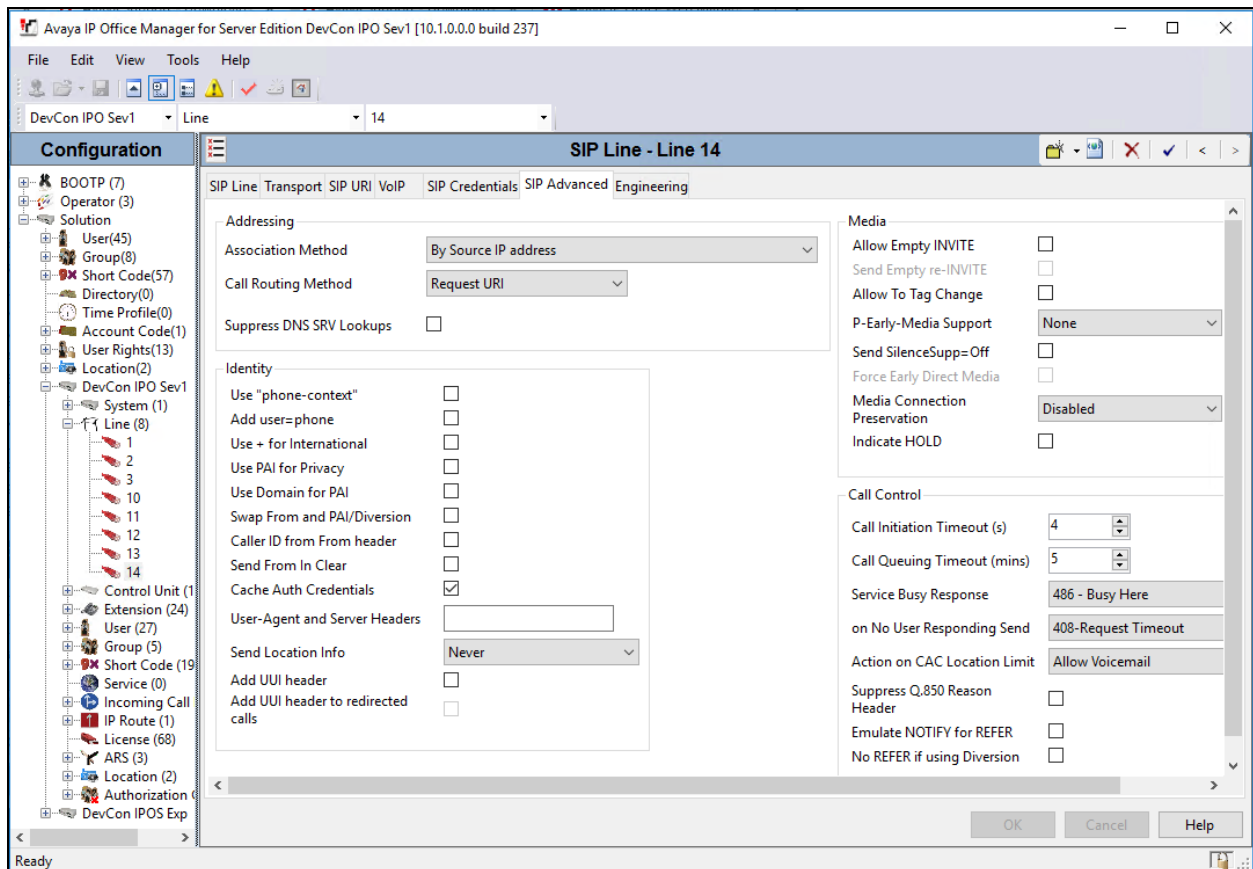
The screenshot shows the Avaya IP Office Manager for Server Edition DevCon IPO Sev1 [10.1.0.0.0 build 237] interface. The 'SIP Line - Line 14' configuration window is open, showing the 'SIP URI' tab. The 'New URI' dialog box is displayed, with fields for Local URI (Auto), Contact (Auto), Display Name (Auto), Identity (None), Header (P Asserted ID), Forwarding And Twinning (Originator Number, Send Caller ID), Diversion Header (None), Registration (0: <None>), Incoming Group (14), Outgoing Group (14), and Max Sessions (10).

In the **VoIP** tab ensure that for **DTMF Support**, *RFC2833/RFC4733* is selected from the drop down menu. Select *Enforced* from the drop down menu for **Media Security**. Check all boxes for **Encryptions** and **Authentication** fields and under **Crypto Suites** check the box for *SRTP_AES_CM_128_SHA1_80*.

Retain default values for all remaining fields. During compliance testing only the **G.711 ULAW** codec was tested as shown in the screen below.



Default values were used for the remaining fields in the **SIP Credentials**, **SIP Advanced** and **Engineering** tabs. Screen below shows an example of the **SIP Advanced** tab with the default values. Click on the **OK** button to complete the configuration of the new SIP Line.

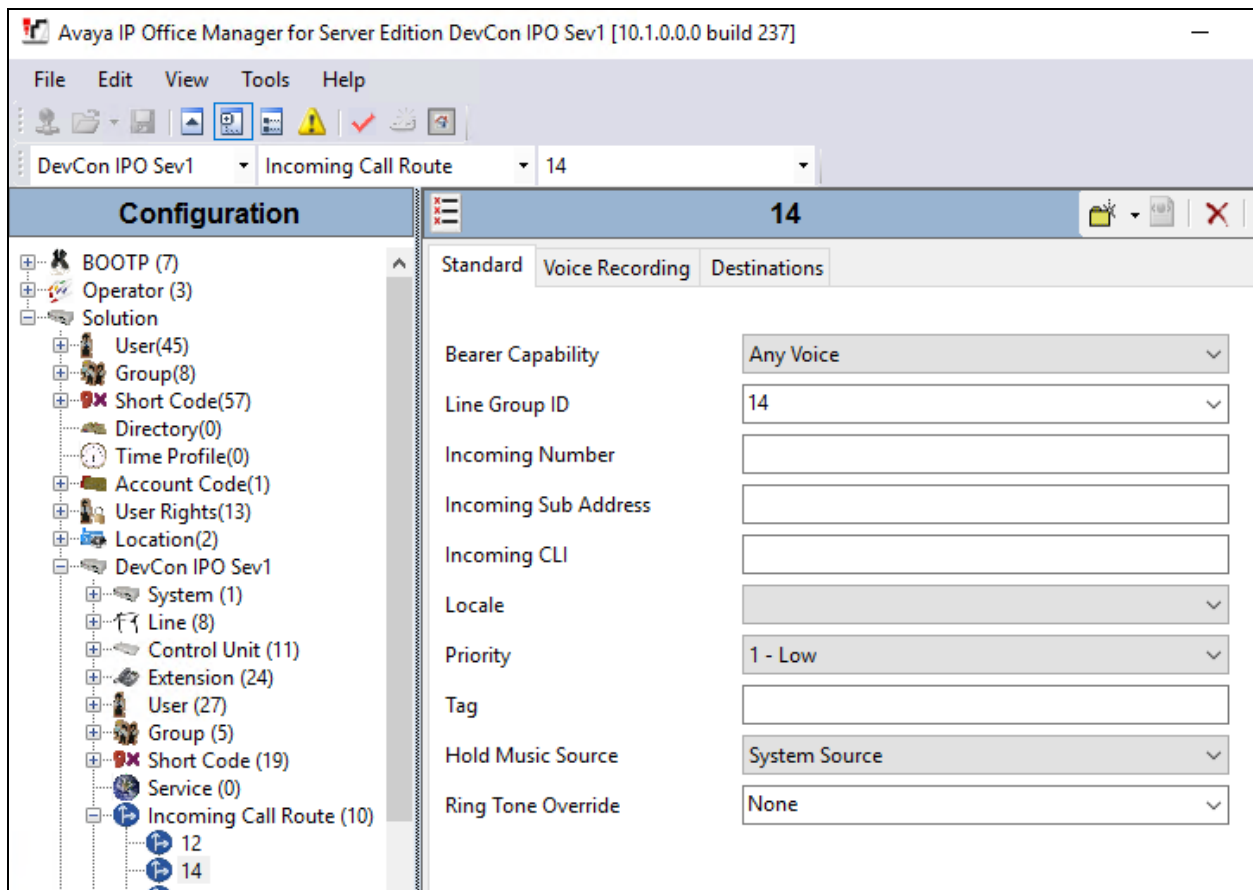


5.4. Configure Incoming Call Route

To configure an incoming call route for routing of incoming calls from ICR, navigate to **DevCon IPO Sev1 → Incoming Call Route** and right click on **Incoming Call Route** followed by **New** (not shown). In the subsequent window, enter the following in the **Standard** tab. This route is used by ICR to make an outbound call to IP Office.

- **Bearer Capability:** Select *Any Voice* from the drop down menu.
- **Line Group ID:** Select *14*, the incoming SIP Line as configured in **Section 5.3.2**.

Retain default values for all remaining values.



Avaya IP Office Manager for Server Edition DevCon IPO Sev1 [10.1.0.0.0 build 237]

File Edit View Tools Help

DevCon IPO Sev1 Incoming Call Route 14

Configuration 14

Standard Voice Recording Destinations

Bearer Capability Any Voice

Line Group ID 14

Incoming Number

Incoming Sub Address

Incoming CLI

Locale

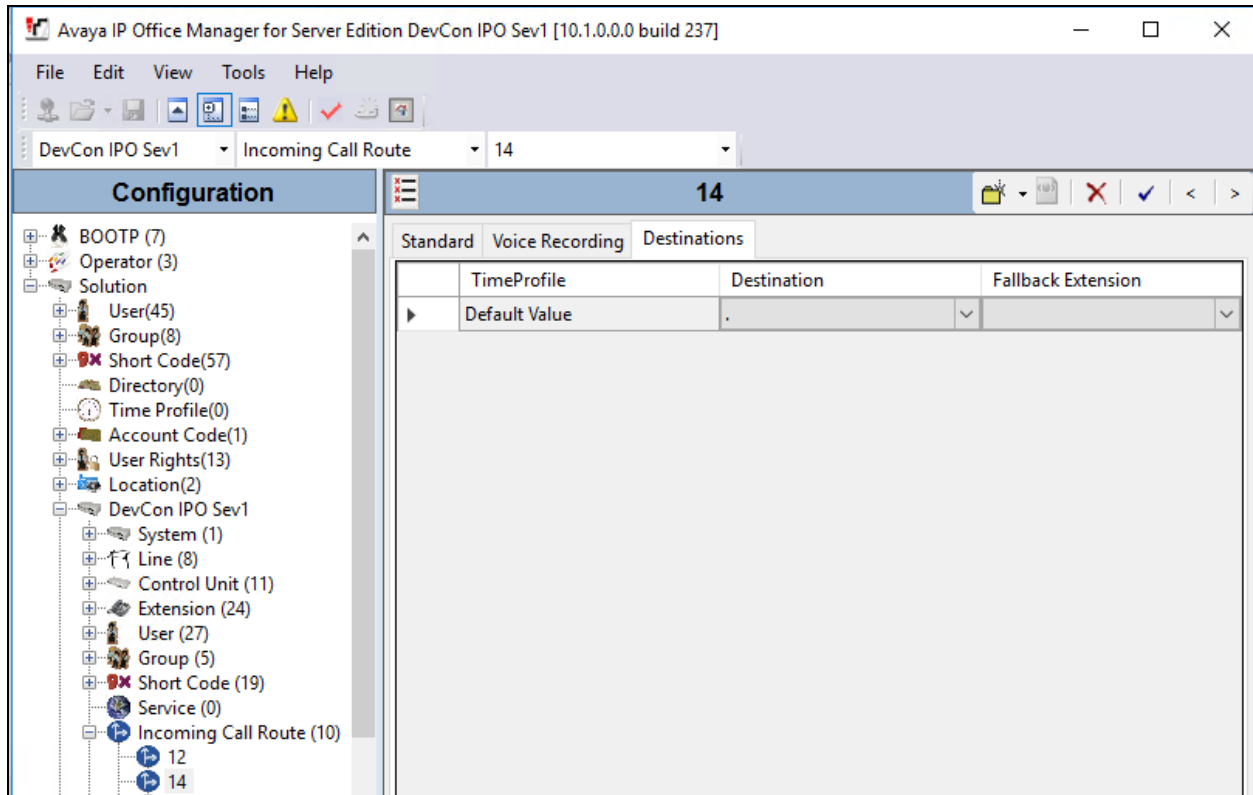
Priority 1 - Low

Tag

Hold Music Source System Source

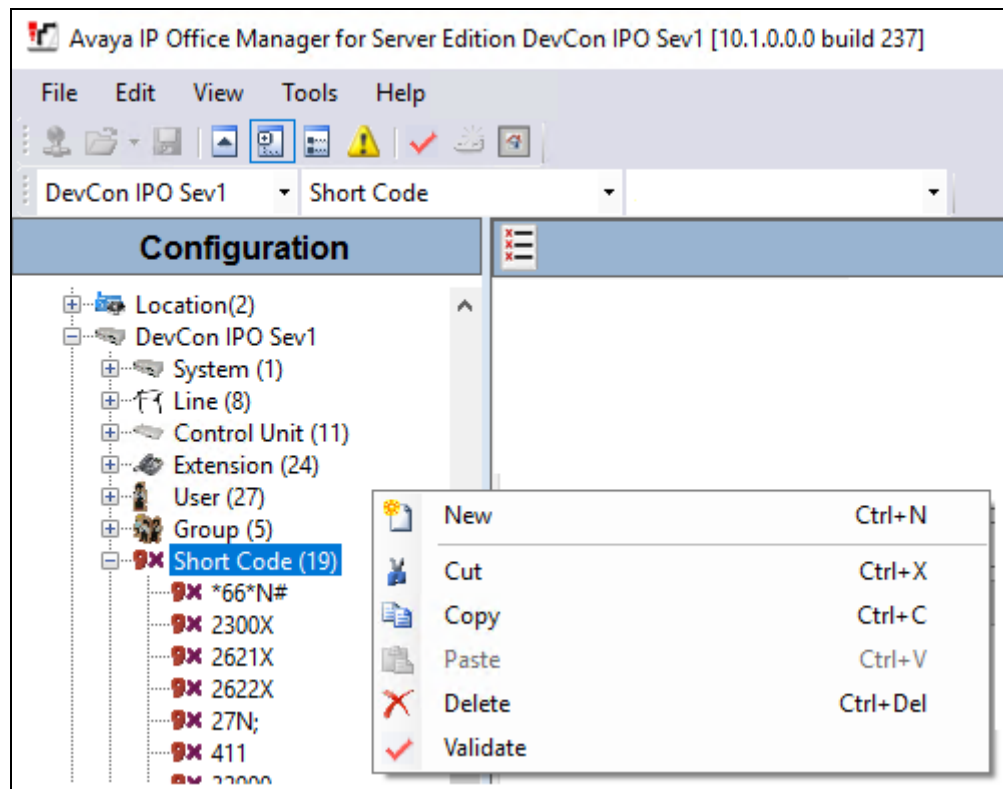
Ring Tone Override None

In the **Destinations** tab, under the **Destination** column enter “.”. Retain default values for all remaining fields and click the **OK** (not shown) button.



5.5. Create Short Code (Route Calls)

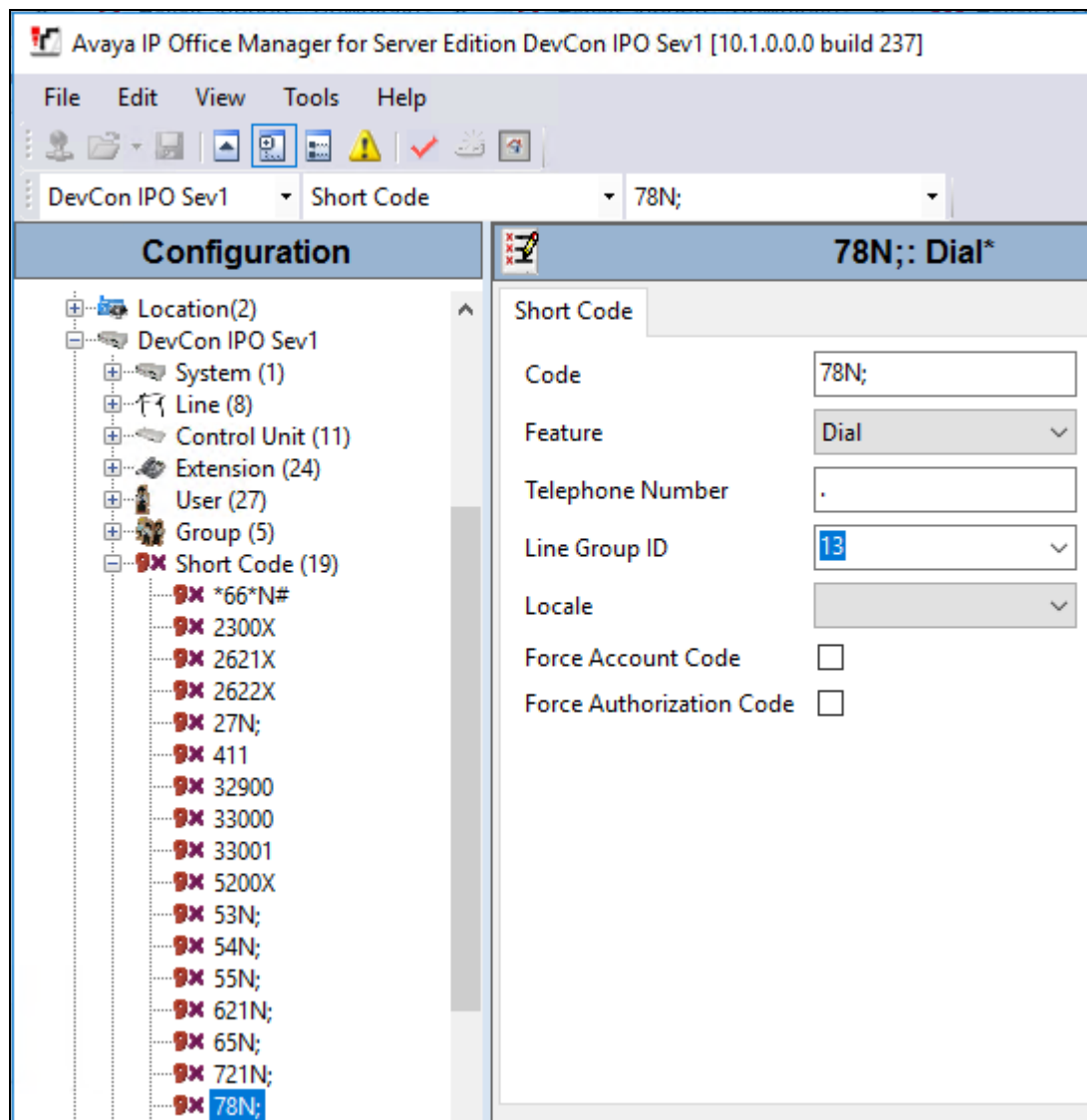
A Short Code needs to be configured on the IP Office to route calls to ICR. Navigate to **DevCon IPO Sev1** → **Short Codes**, and then right click and select **New** as shown in the screen below.



In the subsequent window, enter the following:

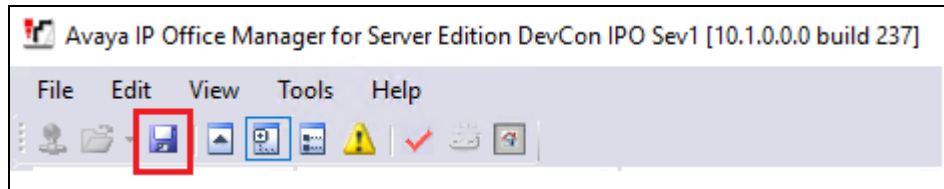
- **Code:** Enter the number range that will be routed to ICR (during compliance testing, all 78xxx numbers were sent to ICR, therefore 78N; was entered).
- **Feature:** Select *Dial* from the drop down menu.
- **Telephone Number:** Enter “.”.
- **Group Line ID:** Enter 13, the outgoing SIP Line configured in **Section 5.3.1**.

Retain default values for all remaining fields and click the **OK** (not shown) button.



5.6. Save Configuration

Once all the configurations are complete, the changes need to be saved on the IP Office system. Click the Save icon as shown in the screen below to save the changes. A subsequent window will appear (not shown) asking the user to proceed with the changes made to the IP Office system/s or not. Click on the **OK** button to confirm.



7. Configure Fonolo In-Call Rescue

This section provides a “snapshot” of ICR configuration used during compliance testing. ICR is typically configured for customers by Fonolo. The screen shots and partial configuration shown below, supplied by Fonolo, are provided only for reference. These represent only an example of the configuration GUI of ICR, available through the Fonolo Customer Portal at <https://portal.fonolo.com/>. Other configurations are possible. Contact Fonolo for details on how to configure ICR. The configuration operations described in this section can be summarized as follows:

- Add a New SIP Trunk Group
- Adding the Agent Call-Back Endpoint
- Adding a New Call-Back Profile

7.1. Add a New SIP Trunk Group

Navigate to **Telco → SIP Trunks** and click the **Add New SIP Trunk Group** button (not shown) at the top of the page. Define a new label to identify this SIP trunk group. During compliance testing **IP Office 10.1** was used as the label. Then select **Save Changes** (not shown).

Under the **Members** tab in this new SIP trunk group, click the **Add New Member** button (not shown), and the **Add New SIP Trunk** dialog will appear as shown below.

Under **Add New SIP Trunk**:

- **SIP URL:** The IP address of the IP Office formatted as a fully qualified URL, defining the protocol and SIP port.
- **DTMF Mode:** The mode to use for sending DTMF tones. Default is RFC 2833.
- **Identity Header:** Whether to include an identity header (either Remote-Party-ID or P-Asserted-Identity). Default is none.
- **Codec Support:** The list of audio codecs to use. Default is μ -law.
- **Priority:** A numeric value that can be used to determine failover or load balance groups when more than one SIP trunk group member is defined. Members with lower priority values are used first; members with a equal priority values are load balanced
- **Keepalive:** This instructs the Fonolo platform to perform regular keep-alive using SIP OPTIONS requests, based on the number of seconds defined. Default is disabled.
- **Session Timers:** If Fonolo should enable SIP Session Timers (RFC 4028). Default is disabled.
- **NAT Support:** If the SIP trunk group member specified is located behind a NAT (Network Address Translation) device. Fonolo can compensate for the un-reachable RTP data specified in the SDP body of the INVITE request, using symmetric RTP.

Add the IP address of IP Office, formatted as a fully qualified URL, defining the protocol and SIP port, then click the **Save Trunk** button. During compliance testing, the protocol **TLS** and port **5061** is used for the SIP service with IP Office, and the default values for the remaining SIP trunk group member settings.

Add New SIP Trunk

SIP URL:

*SIP URLs should use IP addresses or hostnames, and include a protocol (udp, tcp, or tls), and a port value. For example: **udp://10.10.10.10:5060***

DTMF Mode:

Identity Header:

Codec Support: ☒ µ-law ☐ a-law

Priority:

Keepalive: ☒ Enable a keepalive timer on this host. (SIP OPTIONS)

Session Timers: ☐ Enable SIP Session Timers (RFC 4028) on this host.

NAT Support: ☐ This host is behind a NAT device.

During compliance testing, the short code to dial outbound (to PSTN) from IP Office was “9N”. Therefore, all outbound calls to customer call-back phone numbers via the IP Office needed to be prefixed with a “9”. Under the **CALL ROUTING** tab in this new SIP trunk group, enter the digit “9” in the **Default Prefix** input field (shown below), and then click **Save Changes**.

The screenshot shows the Fonolo web interface for configuring SIP Trunks. The top navigation bar includes 'MANAGE', 'TELCO', 'STATS', and 'ADMIN'. The main header shows 'SIP Trunks > IP Office 10.1' with a 'Back to SIP Trunk Groups' button. Below this is a tabbed interface with 'SETTINGS', 'MEMBERS', 'CALL ROUTING' (selected), and 'ACLS'. The 'CALL ROUTING' section contains three input fields: 'Default Country' (set to '(+1) United States, Canada, & Island N'), 'Default Format' (set to 'International Format (w/ country code)', and 'Default Prefix' (set to '9'). The 'Default Prefix' field is highlighted with a red rectangle. To the right of these fields are buttons for 'Test Phone Number' and 'Save Changes'. Below the 'CALL ROUTING' section is the 'INTERNATIONAL DIALING' section, which includes 'International Format' (set to 'International Format (w/ country code)' and 'International Prefix' (set to '011').

SETTINGS	MEMBERS	CALL ROUTING	ACLS
<h3>CALL ROUTING</h3> <p>Default Country: (+1) United States, Canada, & Island N <small>Default local country code.</small></p> <p>Default Format: International Format (w/ country code) <small>Default dialing format for numbers in this country.</small></p> <p>Default Prefix: 9 <small>Default prefix to add to all numbers dialed in this country.</small></p> <p>Test Phone Number Save Changes</p> <h3>INTERNATIONAL DIALING</h3> <p>International Format: International Format (w/ country code) <small>Dialing format for international numbers.</small></p> <p>International Prefix: 011 <small>Dialing code for international numbers. (e.g. 011)</small></p>			

7.2. Adding the Agent Call-Back Endpoint

Navigate to **Manage** → **Targets** and click the **Add New Target** button (not shown). Define a new label to identify this new Target. During compliance testing **Customer Service Agents** was used as the label. Select the **Extension** option (shown below), and enter the CDN to reach the pertinent skillset via IP Office.

During compliance testing, CDN 33000 was pre-configured on Contact Center Select which was accessible via IP Office. Then click on the **Add New Target** button to save this Target.

The screenshot shows the 'Add New Target' form in the Fonolo interface. The top navigation bar includes 'MANAGE', 'TELCO', 'STATS', and 'ADMIN'. The breadcrumb trail is 'Targets > Add New Target'. There are 'Cancel' and 'Add New Target' buttons. The 'SETTINGS' section contains the following fields:

- Target Label:** A text input field with the value 'Customer Service Agents'. A note next to it says 'Only visible through this interface.'
- Phone Number:** A radio button option that is currently unselected. A note next to it says 'Dial as a complete phone number, including the country code.'
- Extension:** A radio button option that is currently selected. A text input field next to it contains the value '33000'. A note next to it says 'Dial as a direct extension (VDN/CDN); numeric digits only.'

Below these fields, there is a paragraph of text: 'When connecting via Direct SIP or using Fonolo appliances, failed calls transfers can be redirected back to the sending host. By default, failed calls will be redirected back to the Direct Extension value. You may also specify an alternate extension to redirect the call back to.'

At the bottom, there is an **Alternate Extension:** checkbox which is currently unchecked, followed by an empty text input field and a note: 'Alternate extension to use for returning failed calls.'

From the **TELCO SETTINGS** section of the newly added Target, select the SIP trunk to use for this Target, from the **Direct SIP** drop down menu shown below. Select the **IP Office 10.1** SIP trunk, added in **Section 7.1**, and then click the **Save Changes** button.

The screenshot shows the 'TELCO SETTINGS' section for the 'Customer Service Agents' target in the Fonolo interface. The top navigation bar is the same. The breadcrumb trail is 'Targets > Customer Service Agents'. There is a 'Back to Targets' button. Below the breadcrumb, there is a tabbed interface with tabs for 'SETTINGS', 'TELCO SETTINGS' (which is active), 'HOURS', 'ADVANCED SCHEDULES', and 'CALL-BACK LIMITS'. A 'Save Changes' button is located at the top right of the 'TELCO SETTINGS' section.

The 'TELCO SETTINGS' section contains the following information:

- A descriptive text: 'This controls how Fonolo will call in to your phone system.'
- Direct SIP:** A radio button option that is currently selected. A dropdown menu next to it shows 'IP Office 10.1'. A note next to it says 'Use this SIP Trunk.'
- Dial Timeout:** A text input field with the value '60'. A note next to it says 'How long to wait for the Target to answer before returning "Target Call Timeout". 10 to 120 secs.'

7.3. Adding a New Call-Back Profile

Navigate to **Manage → Call-Back Profiles** and click on the **Add New Profile** button, and configure the new profile:

- **Profile Label:** A label to identify this new profile.
- **Geo Whitelist:** A geographic whitelist to use for this new profile.
- **Channel:** Select **In-Call Rescue**.
- **Language:** Select the appropriate language for this skill set queue.
- **Customer CID Number:** The Caller-ID number the customer will see.
- **Customer CID Name:** The Caller-ID name the customer will see.
- **Agent CID Number:** The Caller-ID number the agent will see.
- **Agent CID Name:** The Caller-ID name the agent will see.

Click the **Add New Call-Back Profile** button to add this new profile.

fonolo MANAGE TELCO STATS ADMIN

Call-Back Profiles > Add New Call-Back Profile

Cancel Add New Call-Back Profile

SETTINGS

Profile Label:	ICR Profile	Only visible through this interface.
Geo. Whitelist:	Default Whitelist	This is the geographic white list to use with this call-back profile.
Channel:	In-Call Rescue	This is the channel type: In-Call Rescue, Web, or Mobile.
Language:	English	The language used for this channel.

CALLER ID SETTINGS

You can adjust the caller ID name and number, seen by both your clients and agents.

Client CID Number:	18005551234	Caller ID number seen by clients.
Client CID Name:	Acme Company	Caller ID name seen by clients (only supported by some systems).
Agent CID Number:	{{ \$client_number }}	Caller ID number seen by your agents.
Agent CID Name:	Fonolo	Caller ID name seen by your agents (only supported by some systems).

The Agent CID values support replaceable MACROS:

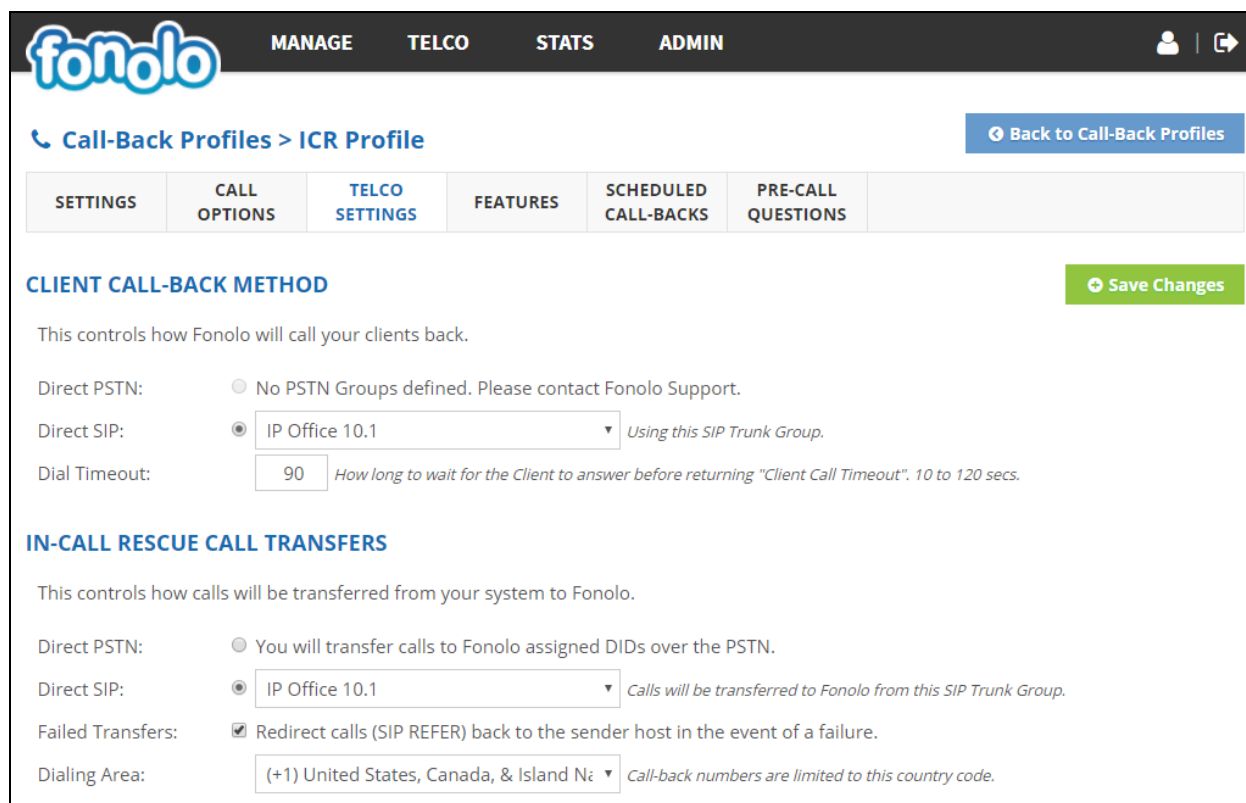
- The **{{ \$client_number }}** MACRO will be replaced with the clients call-back number.

From the **CALL OPTIONS** section of the new **Call-Back Profile**, select the Target added in **Section 7.2** (from the drop-down menu highlighted below), and click the **Add Option** link to add the CDN value to the section on the right, as shown below, then click the **Save Changes** button.

This associates the Target CDN with this new **Call-Back Profile**. Multiple call options can be associated with a single **Call-Back Profile**, one for each skill call-backs are being offered on.

The screenshot displays the Fonolo user interface. At the top, there is a navigation bar with the 'fonolo' logo and tabs for 'MANAGE', 'TELCO', 'STATS', and 'ADMIN'. Below this, the breadcrumb 'Call-Back Profiles > ICR Profile' is shown next to a 'Back to Call-Back Profiles' button. A horizontal menu contains tabs for 'SETTINGS', 'CALL OPTIONS' (which is active), 'TELCO SETTINGS', 'FEATURES', 'SCHEDULED CALL-BACKS', and 'PRE-CALL QUESTIONS'. The 'CALL OPTIONS' section is titled and includes a 'Save Changes' button. A text instruction reads: 'Add Call-Back options to your Call-Back Profile with the Add Option buttons below.' Below this instruction, a dropdown menu is highlighted with a red rectangle; it currently shows 'Customer Service Agents - 33000'. Underneath the dropdown, there is a card for 'Customer Service Agents #33000'. This card features a blue 'Add Option' button, which is also highlighted with a red rectangle. To the right of this card, another card for 'Customer Service Agents #33000' is visible, showing 'Edit' and 'Delete' options.

From the **TELCO SETTINGS** section of the new **Call-Back Profile**, select the **IP Office 10.1** SIP trunk group created in **Section 7.1** as the **Direct SIP** value under both the **CLIENT CALL-BACK METHOD**, and the **IN-CALL RESCUE CALL TRANSFERS** section, as shown below, then click the **Save Changes** button.



fonolo MANAGE TELCO STATS ADMIN

Call-Back Profiles > ICR Profile [Back to Call-Back Profiles](#)

SETTINGS CALL OPTIONS **TELCO SETTINGS** FEATURES SCHEDULED CALL-BACKS PRE-CALL QUESTIONS

CLIENT CALL-BACK METHOD [Save Changes](#)

This controls how Fonolo will call your clients back.

Direct PSTN: ☐ No PSTN Groups defined. Please contact Fonolo Support.

Direct SIP: ☒ IP Office 10.1 Using this SIP Trunk Group.

Dial Timeout: How long to wait for the Client to answer before returning "Client Call Timeout". 10 to 120 secs.

IN-CALL RESCUE CALL TRANSFERS

This controls how calls will be transferred from your system to Fonolo.

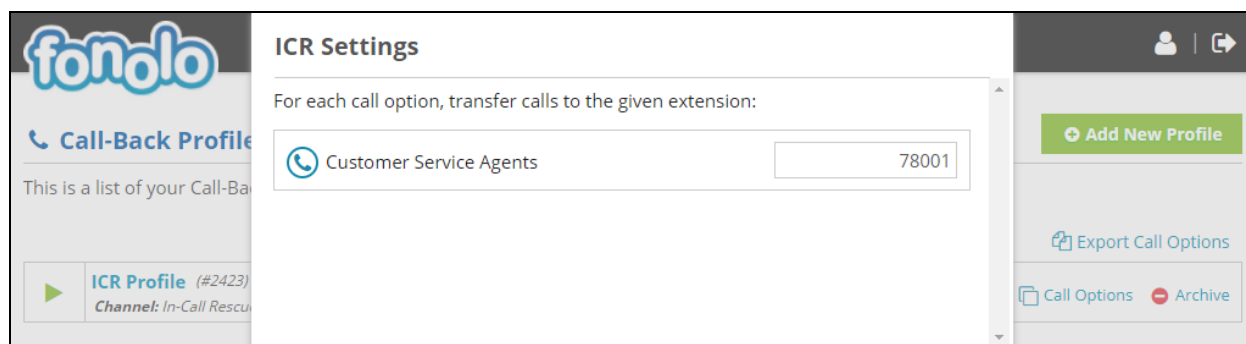
Direct PSTN: ☐ You will transfer calls to Fonolo assigned DIDs over the PSTN.

Direct SIP: ☒ IP Office 10.1 Calls will be transferred to Fonolo from this SIP Trunk Group.

Failed Transfers: ☒ Redirect calls (SIP REFER) back to the sender host in the event of a failure.

Dialing Area: Call-back numbers are limited to this country code.

Navigate to **Manage → Call-Back Profiles** and click on the **In-Call Rescue** link on the newly created **Call-Back Profile** (not shown). The **ICR Settings** dialog will appear (shown below), and include the inbound extensions to use for each pertinent Contact Center Select CDN. These are the extensions to transfer calls to, on the ICR system, when a call opts-in for a call-back. During compliance testing, the extension **78001** is configured on the Fonolo system.



fonolo

Call-Back Profile

This is a list of your Call-Back Profiles

ICR Profile (#2423)
Channel: In-Call Rescue

ICR Settings

For each call option, transfer calls to the given extension:

Customer Service Agents	78001
-------------------------	-------

[Add New Profile](#)

[Export Call Options](#)

[Call Options](#) [Archive](#)

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of IP Office, and ICR.

From the **Avaya IP Office System Status** window for the Primary server, verify that the **Trunks** configured are in service. Screen below shows the SIP Line trunks *13* and *14* have *In Service* under the **Line Service State** field.

The screenshot shows the Avaya IP Office System Status window for DevCon IPO Sev1. The left sidebar contains a tree view with categories: System, Alarms (9), Extensions (6), Trunks (8), Active Calls, Resources, Voicemail, IP Networking, and Locations. Under Trunks (8), Line 13 is selected. The main pane displays the SIP Trunk Summary for Line 13, which is in the 'In Service' state. The summary includes fields for Peer Domain Name, Resolved Address, Line Number, Number of Administered Channels, Number of Channels in Use, Administered Compression, Enable Faststart, Silence Suppression, Media Stream, Layer 4 Protocol, SIP Trunk Channel Licenses, and SIP Trunk Channel Licenses in Use. A green progress indicator shows 0% usage. Below the summary is a table with 13 columns: C..., U..., C..., C..., T..., Re..., C..., C..., C..., Other, Di..., R..., R..., R... Par... The table has 4 rows, all showing 'Idle' status. At the bottom, there are buttons for Trace, Trace All, Pause, Ping, Call Details, Graceful Shutdown, Force Out of Service, Print..., and Save As... The status bar at the bottom right shows the time 3:58:02 PM and the status Online.

C...	U...	C...	C...	T...	Re...	C...	C...	C...	Other	Di...	R...	R...	R...	Par...
1				Idle	0...									
2				Idle	0...									
3				Idle	0...									
4				Idle	0...									

The following tests were also performed to verify proper configuration of ICR with IP Office.

- PSTN caller is able to select the call back option and get redirected to ICR via IP Office.
- PSTN caller is able to hear the ICR menu and make the required choices.
- ICR is able to recognize the choices made by the PSTN user.
- ICR is able to call the CDN and wait for an available agent.
- ICR is able to call out to the PSTN caller and connect them to an available agent.

8.1. Verify Fonolo In-Call Rescue

In the Fonolo customer portal, verify the link status of the SIP trunk group to IP Office, by navigating to **Telco → SIP Trunks**. Each SIP trunk group member will have a response time value, indicating the network latency (in milliseconds) between the Fonolo network, and IP Office. A positive **Response Time** value indicates a positive link status.

The screenshot shows the Fonolo customer portal interface. At the top, there is a navigation bar with the Fonolo logo and tabs for MANAGE, TELCO, STATS, and ADMIN. Below the navigation bar, the breadcrumb trail reads "SIP Trunks > IP Office 10.1". A "Back to SIP Trunk Groups" button is visible in the top right. The main content area has tabs for SETTINGS, MEMBERS, CALL ROUTING, and ACLS. The "MEMBERS" tab is selected, showing the "TRUNK GROUP MEMBERS" section. A green "Add New Member" button is present. Below this, a message states: "Fonolo will select a SIP Trunk from this group for each Call-Back placed." A table lists the trunk group members. The first member is "tls://10.10.97.41:5061" with a "DTMF Mode: RFC 2833, Using NAT: No" and a "Response Time: 31ms" (highlighted with a red box) and "Priority: 10". A "Delete" button is located to the right of the member entry.

SETTINGS	MEMBERS	CALL ROUTING	ACLs			
TRUNK GROUP MEMBERS Add New Member						
Fonolo will select a SIP Trunk from this group for each Call-Back placed.						
<table border="1"><tr><td></td><td>tls://10.10.97.41:5061 DTMF Mode: RFC 2833, Using NAT: No Response Time: 31ms Priority: 10</td><td> Delete</td></tr></table>					tls://10.10.97.41:5061 DTMF Mode: RFC 2833, Using NAT: No Response Time: 31ms Priority: 10	Delete
	tls://10.10.97.41:5061 DTMF Mode: RFC 2833, Using NAT: No Response Time: 31ms Priority: 10	Delete				

Additional information is available through the **Stats → Graphs** section of the Fonolo customer portal (not shown).

9. Conclusion

These Application Notes describe the configuration steps required for Fonolo In-Call Rescue to successfully interoperate with Avaya IP Office Server Edition. All feature and serviceability test cases were completed and passed with the exceptions/observations if any noted in **Section 2.2**.

10. Additional References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at <http://support.avaya.com>.

1. *Deploying IP Office™ Platform Server Edition Solution*, Release 10.1, June 2017.
2. *Administering Avaya IP Office™ Platform with Manager*, Release 10.1, June 2017.
3. *Deploying Avaya IP Office™ Platform IP500 V2*, 15-601042 Issue 32f - (20 July 2017).

Fonolo provides their documentation upon delivery of their products/services.

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