Avaya Solution & Interoperability Test Lab

Application Notes for configuring Fonolo Voice Call-Backs with Avaya IP Office Server Edition using SIP Trunks – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Fonolo Voice Call-Backs application to interoperate with Avaya IP Office Server Edition using SIP trunks.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any 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 Voice Call-Backs (VCB) to interoperate with Avaya IP Office Server Edition (IP Office) using SIP trunks. VCB provides functionality to replace hold-time with a call-back. The solution combines hosted services with optional hardware (to keep voice data on-premise). The solution communicates via SIP/RTP. The VCB 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 VCB application.

2. General Test Approach and Test Results

The interoperability compliance testing focused on verifying inbound and outbound calls flows between IP Office and VCB.

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 these DevConnect Application Notes 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 these Application Notes, the interface between Avaya systems and the Fonolo Voice Call Back did not include use of any specific encryption features as requested by Fonolo.

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

- Establishment of SIP trunks between Fonolo and IP Office.
- Incoming calls to a skill set queue on IP Office using Avaya Aura Contact Center Select can be directed to the VCB application via the SIP trunks.
- The VCB application can call into an agent in a skill set queue and also make an outbound call and connect them together when an agent is available.
- DTMF transmission.

2.2. Test Results

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

- VCB only supports G.711u codec.
- VCB only supports RFC2833 for DTMF transmission.

2.3. Support

Technical support on Fonolo VCB 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, Avaya Contact Center Select (Contact Center Select) and Avaya phones were used during compliance testing. As shown in **Figure 1**, SIP trunks were used to connect Fonolo VCB with IP Office directly. IP Office is connected to SIP Service provider using SIP Trunk. A skill set queue is configured on Contact Center Select with a few agents belonging to a queue. The configuration allowed the enterprise site to use SIP trunking for calls to and from VCB 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.
A CDN of 4345 was configured on Contact Center Select and agents from IP Office were log in to skillset Skill1.

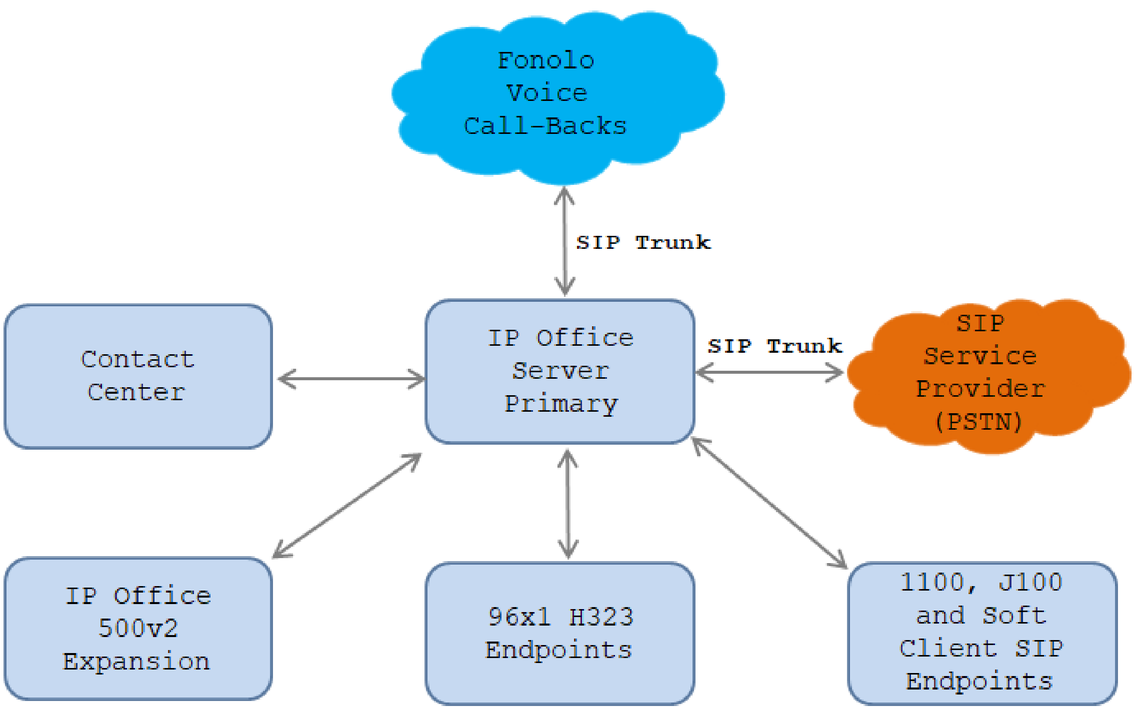


Figure 1: Avaya IP Office Network with Fonolo Voice Call-Backs

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office Server Edition (Primary) running on Virtual Environment	11.0.4.1.0 build 11
Avaya IP Office IP500 V2 (Expansion)	11.0.4.1.0 build 11
Avaya IP Office Manager running on Windows 10	11.0.4.1.0 build 11
Avaya Contact Center Select running on Virtual Environment	7.0.1.1
Avaya Telephones: • 1140 IP (SIP) Deskphone • 9641 IP (H323) Deskphone • J129 SIP Deskphone	 04.04.23.00 6.68 4.0.5.0
Fonolo Voice Call-Backs	3.5

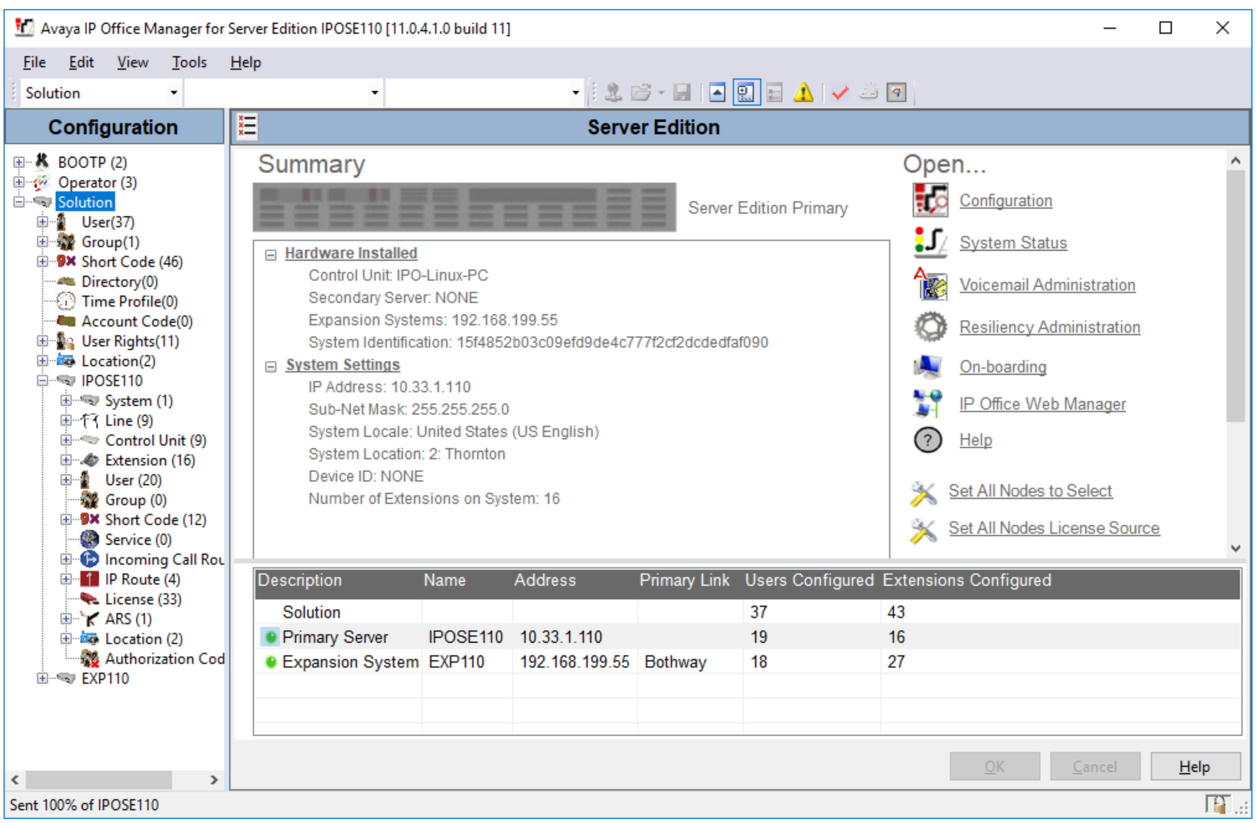
Note*:* Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office IP500 V2 and also when deployed with IP Office Server Edition in all configurations.

5. Configure Avaya IP Office

Configuration and verification operations on the Avaya 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 and the same configuration applies to the Expansion (IP500 V2) system too. 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** Error! Reference source not found.. The configuration operations described in this section can be summarized as follows:

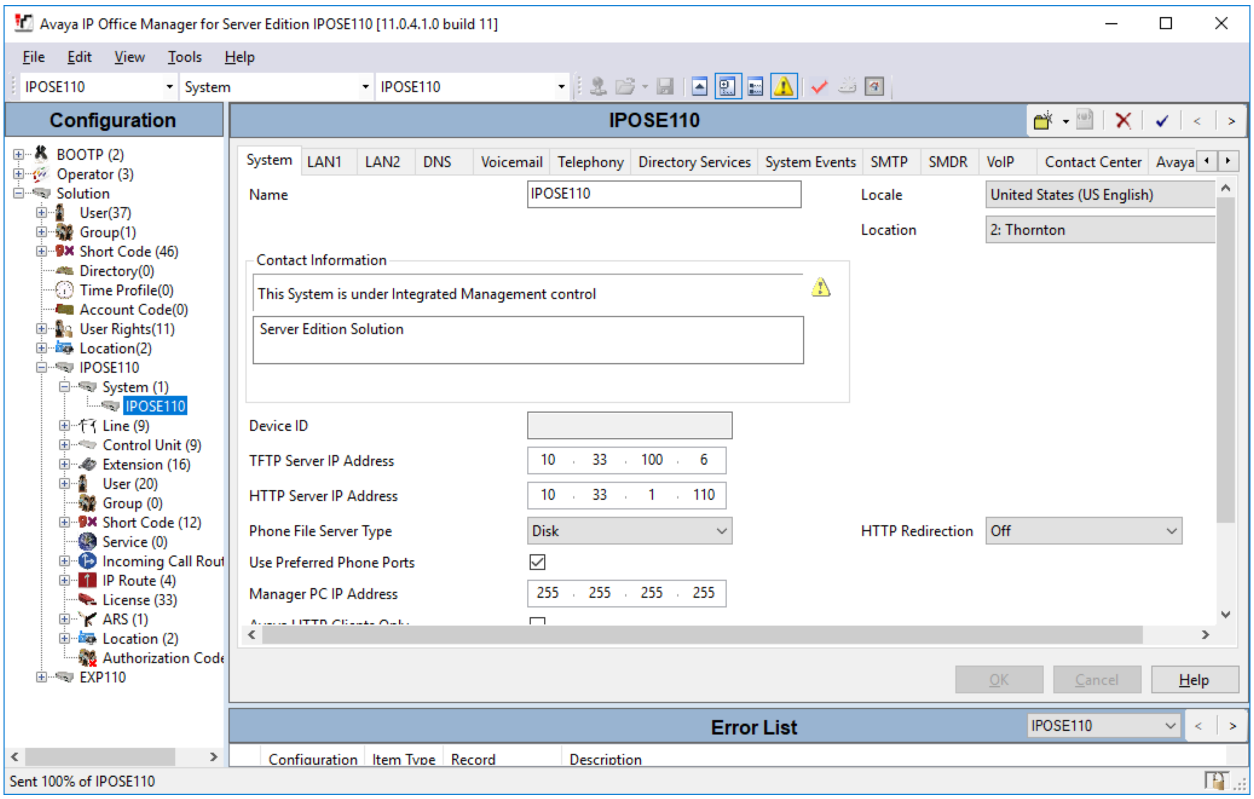
- Configure System Locale.
- Configure System.
- Create SIP Line.
- Configure Incoming Call Route.
- Create Short Code..
- Save Configuration.

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, 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

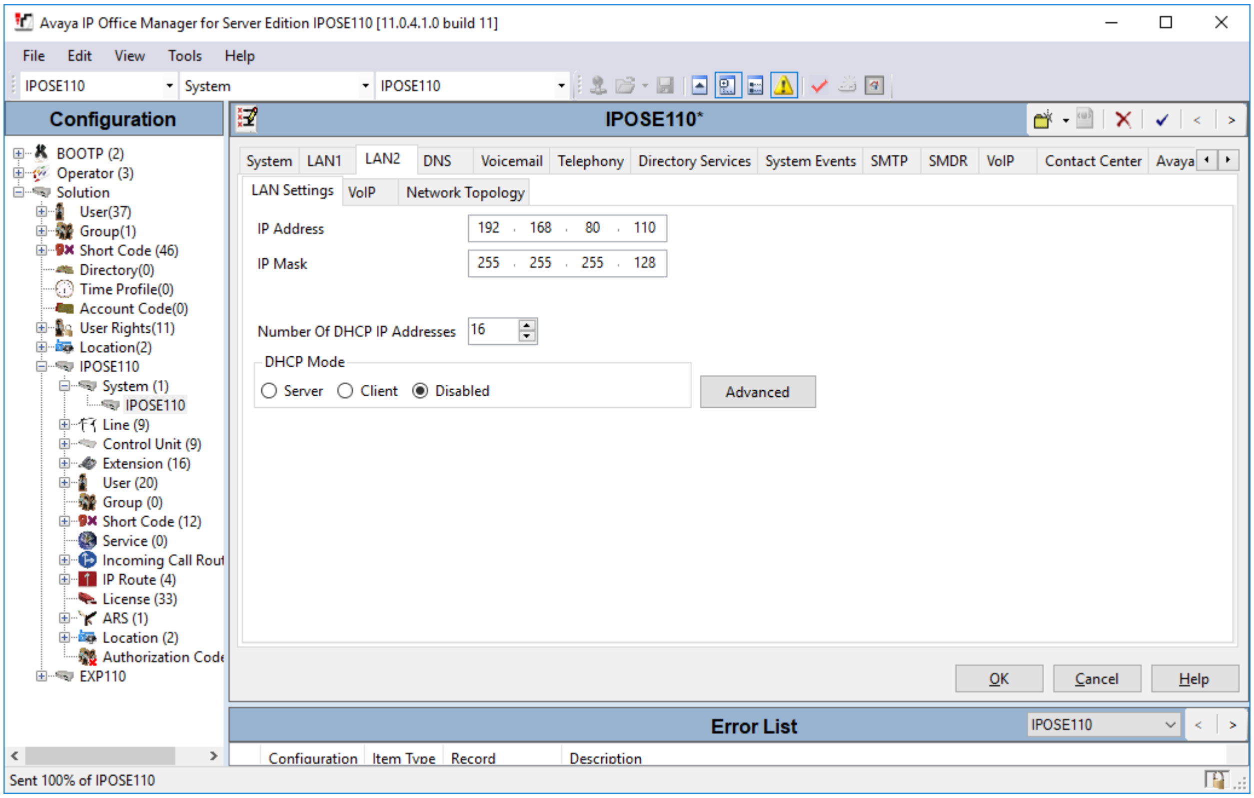
The locale is usually the country where the IP Office is installed. By selecting the correct country a number of system defaults for that country will be used by the IP Office. To configure the locale, select **IPOSE110 → System →IPOSE110** from the IP Office Configuration Tree. During compliance testing the System was called **IPOSE110** for the Primary Server and **EXP110** 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 **IPOSE110 → System → IPOSE110** to display the screen in the right pane, where **IPOSE110** is the name of the IP Office system.

Select the **LAN2** tab, IP Office can support SIP on the LAN1 and/or LAN2 interfaces, however during compliance testing the LAN2 interface was used. From the **LAN Settings** sub-tab, note the **IP Address** configured, which is ***192.168.80.110***. This IP Address is required during VCB configuration.

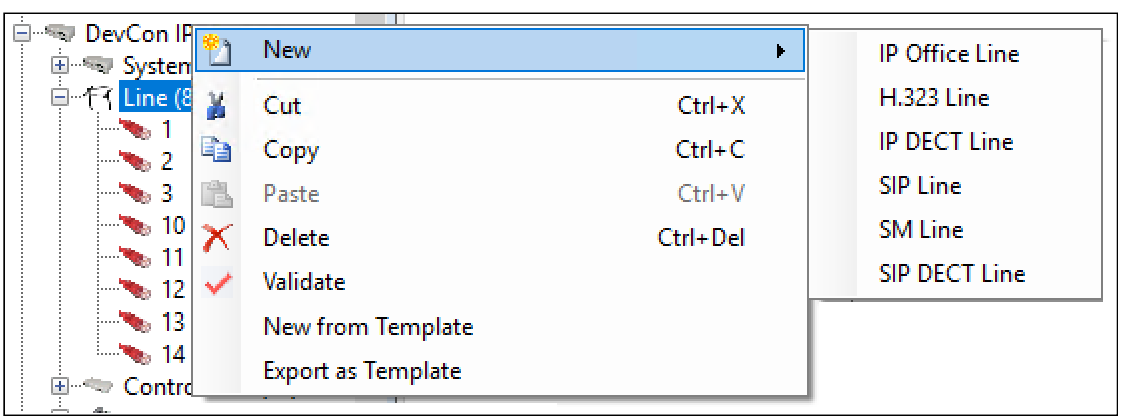


5.3. Create SIP Lines

During compliance testing two SIP lines were created. One line was for outgoing calls from IP Office to VCB and the other line was for incoming calls from VCB to IP Office.

5.3.1. Configure Outgoing SIP Line

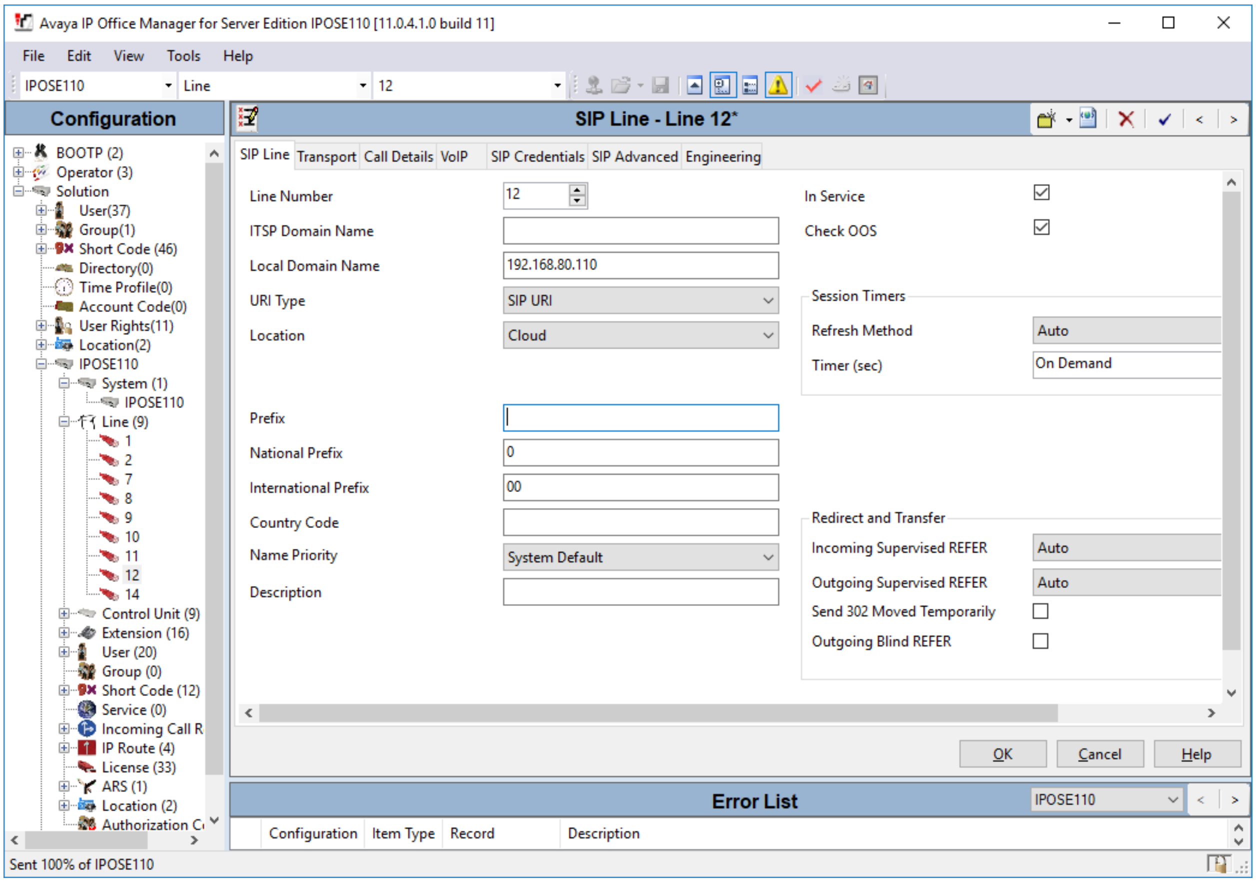
To create the SIP line from the IP Office to VCB for outgoing calls, navigate to **System → Line** and right click on **Line** followed by **New →SIP Line** as shown in the screen below. In this example, line **12** was created to connect to VCB.



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 ***12*** was configured.
- **ITSP Domain Name:** Configure Domain name of VCB or leave this field blank.
- **Local Domain Name:** Enter the LAN2 IP address.
- **Description:** Provide a description for the SIP Line. This field is optional.

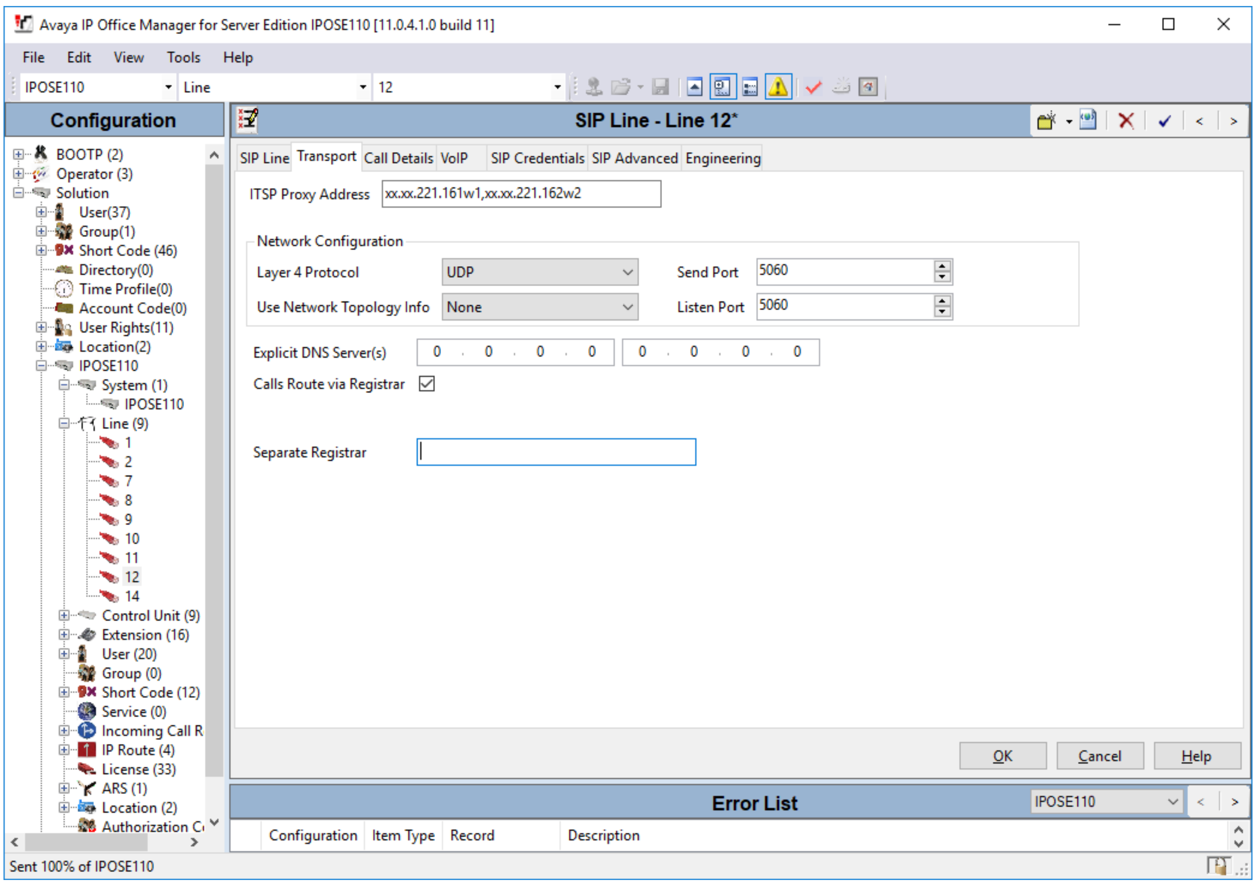
Retain default values for all remaining fields.



In the **Transport** tab enter IP address of VCB in the **ITSP Proxy Address** field. During compliance testing ***xx.xx.221.161w1, xx.xx.221.162w2*** was configured. VCB 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.221.161w1*, *xx.xx.221.162w1*, the weighting values assign1 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 ***UDP*** from the drop down menu.

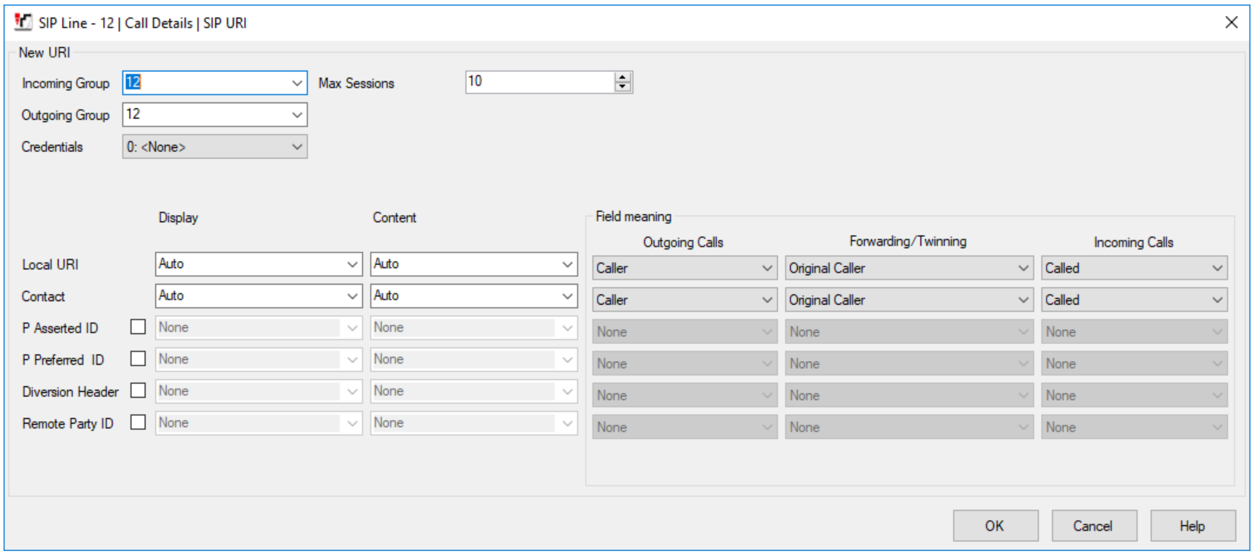
Retain default values for remaining fields. For compliance testing **UDP** protocol was tested.



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

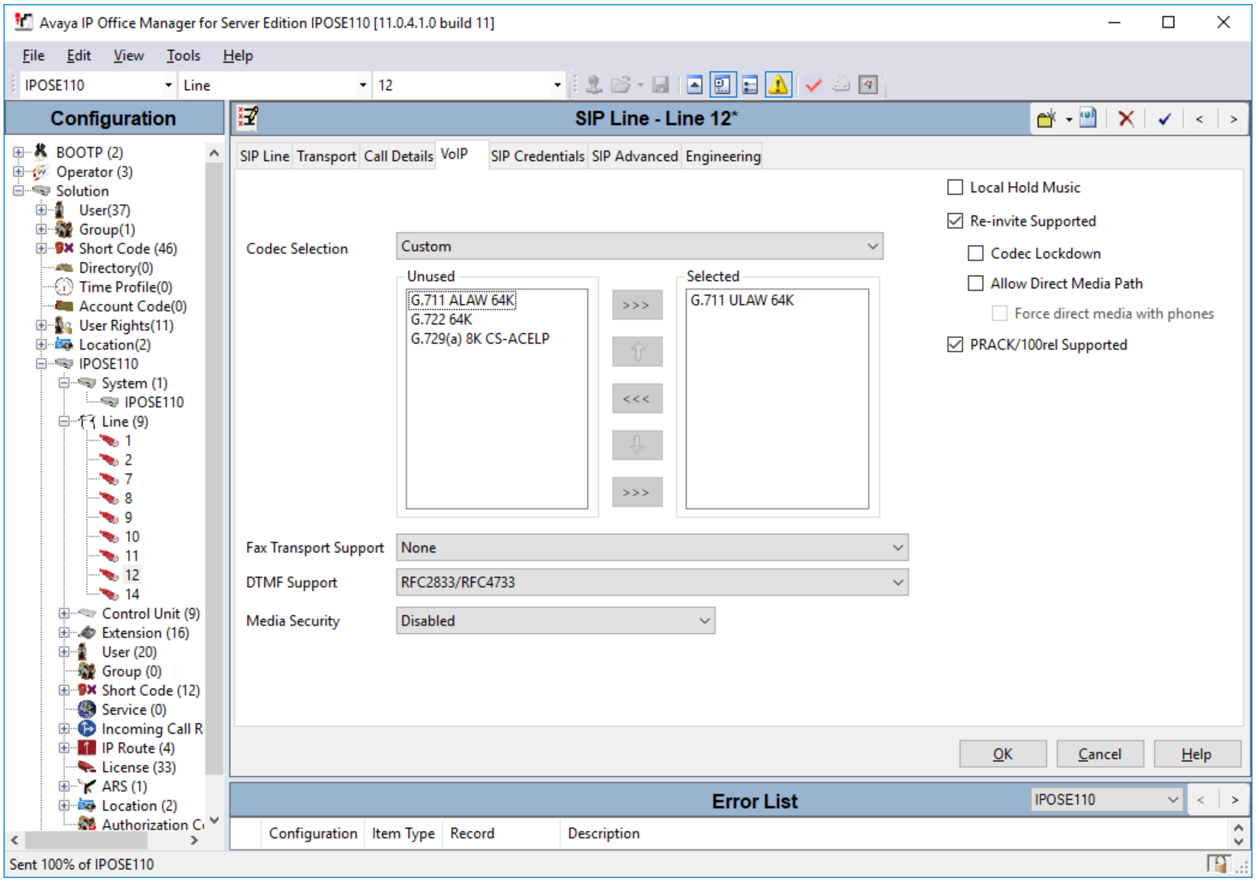
- **Incoming Group**: Select the SIP Line number which is ***12***.
- **Outcoming Group**: Select the SIP Line number which is ***12***.
- **Local URI:** Select ***Auto***.
- **Contact:** Select ***Auto***.

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

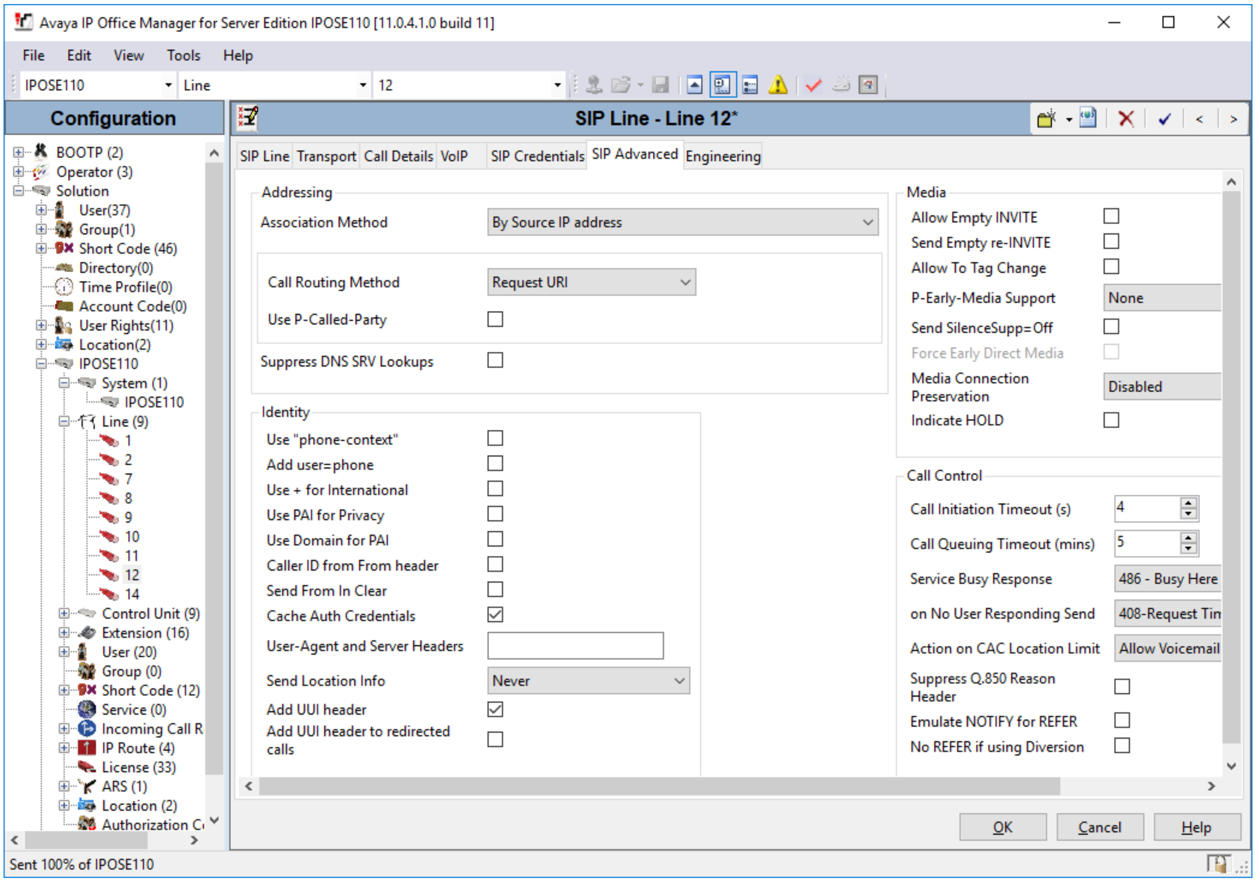


In the **VoIP** tab ensure that for **DTMF Support**, *RFC2833/RFC4733* is selected from the drop down menu. Select ***Disable*** from the drop down menu for **Media Security**. Check **Re-invite Supported** and **PRACT/100rel Supported** checkbox.

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

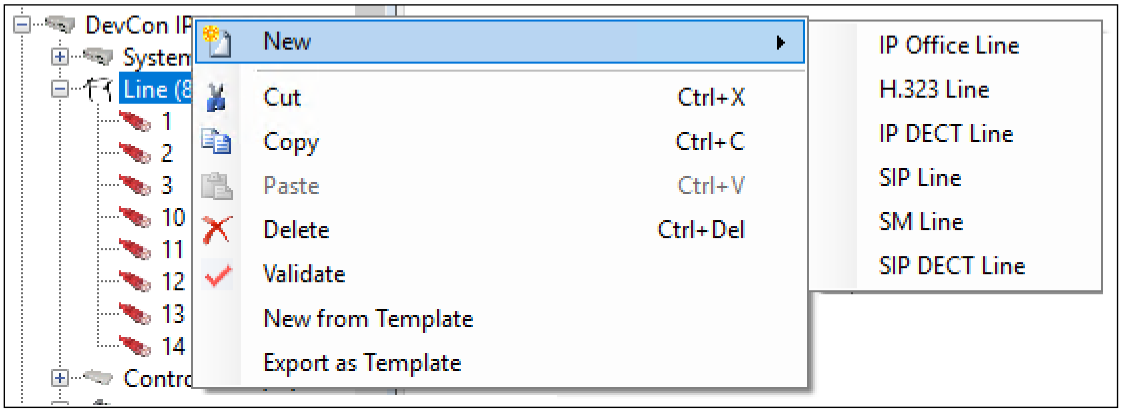


Default values were used for the remaining fields in the **SIP Credentials**, **SIP Advanced** and **Engineering** tabs. Screen below shows an example of the **SIP Advanced** tab with the default values. Click on the **OK** button to complete the configuration of the new SIP Line.



5.3.2. Configure Incoming SIP Line

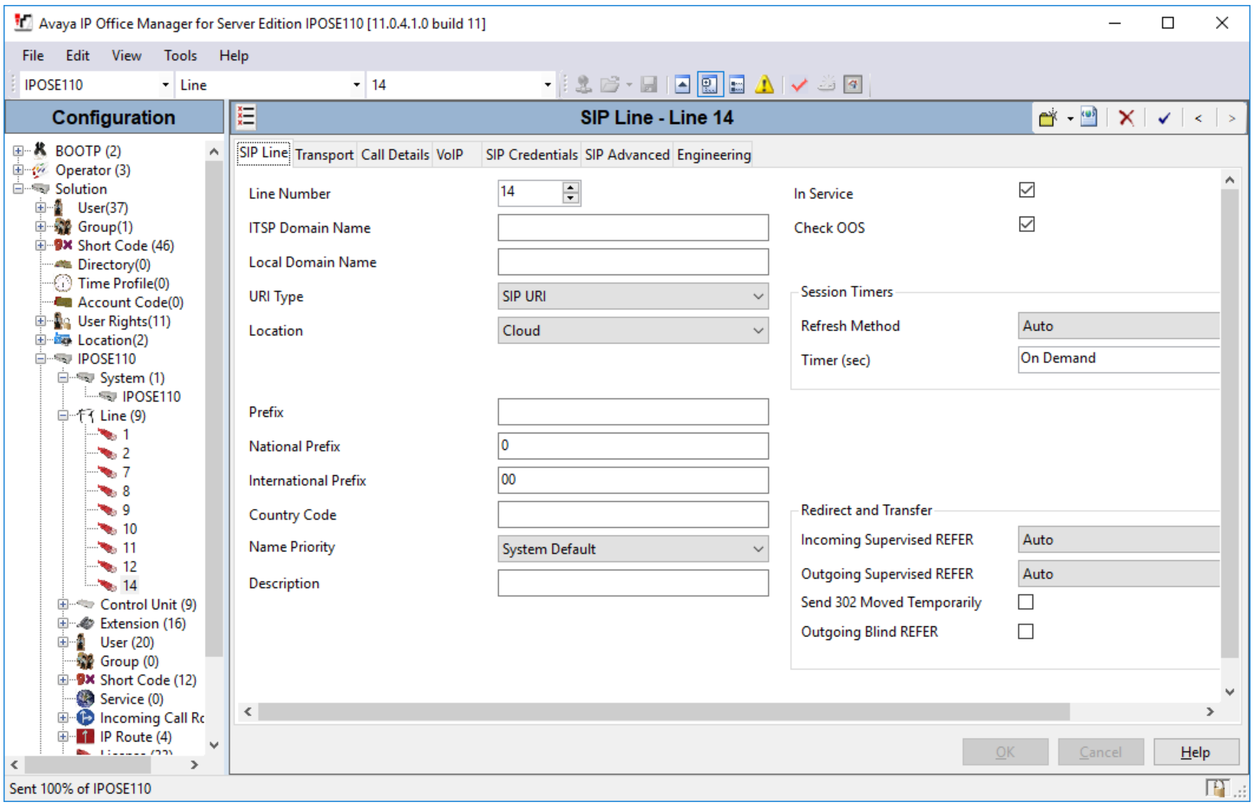
To create the SIP line from the IP Office to VCB for incoming calls, navigate to **System → Line** and right click on **Line** followed by **New →SIP Line** as shown in the screen below. In this example, line **14** was created to connect to VCB.



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

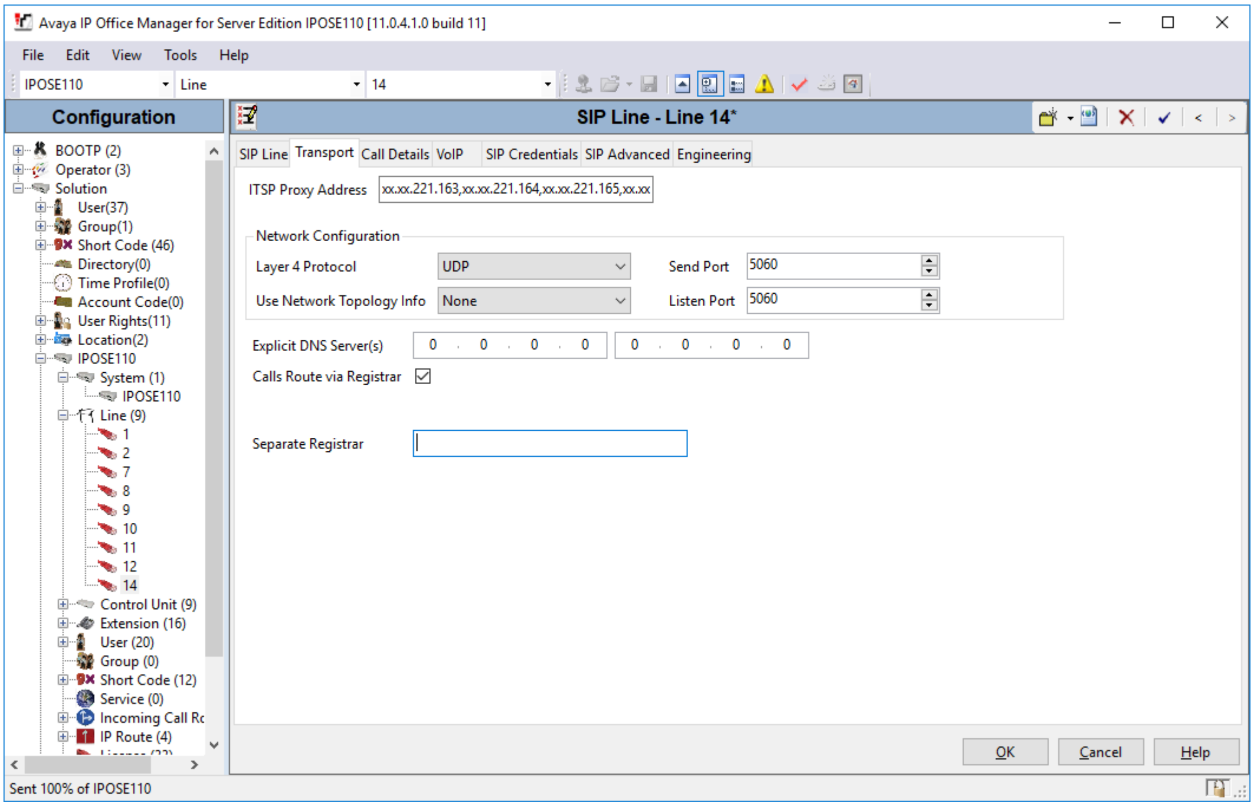
- **Line Number:** Line number ***14*** was configured.
- **ITSP Domain Name:** Configure Domain name of VCB or leave this field blank.
- **Description:** Provide a description for the SIP Line. This field is optional.

Retain default values for all remaining fields.



In the **Transport** tab enter IP address of VCB 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. VCB 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 ***UDP*** from the drop down menu.

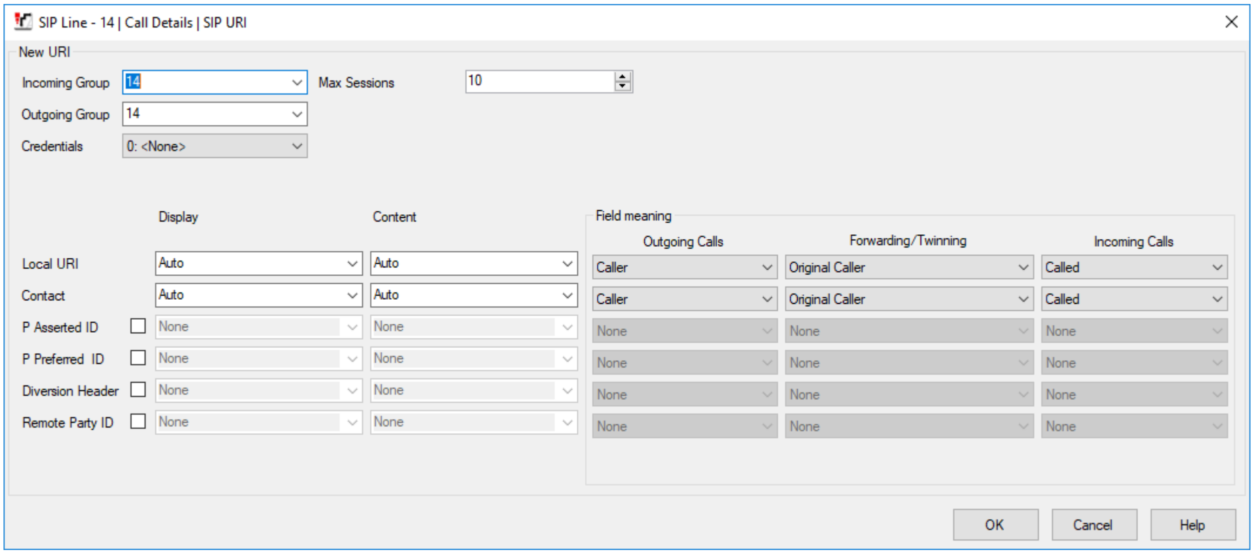
Retain default values for remaining fields. For compliance testing ***UDP*** protocol was tested.



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

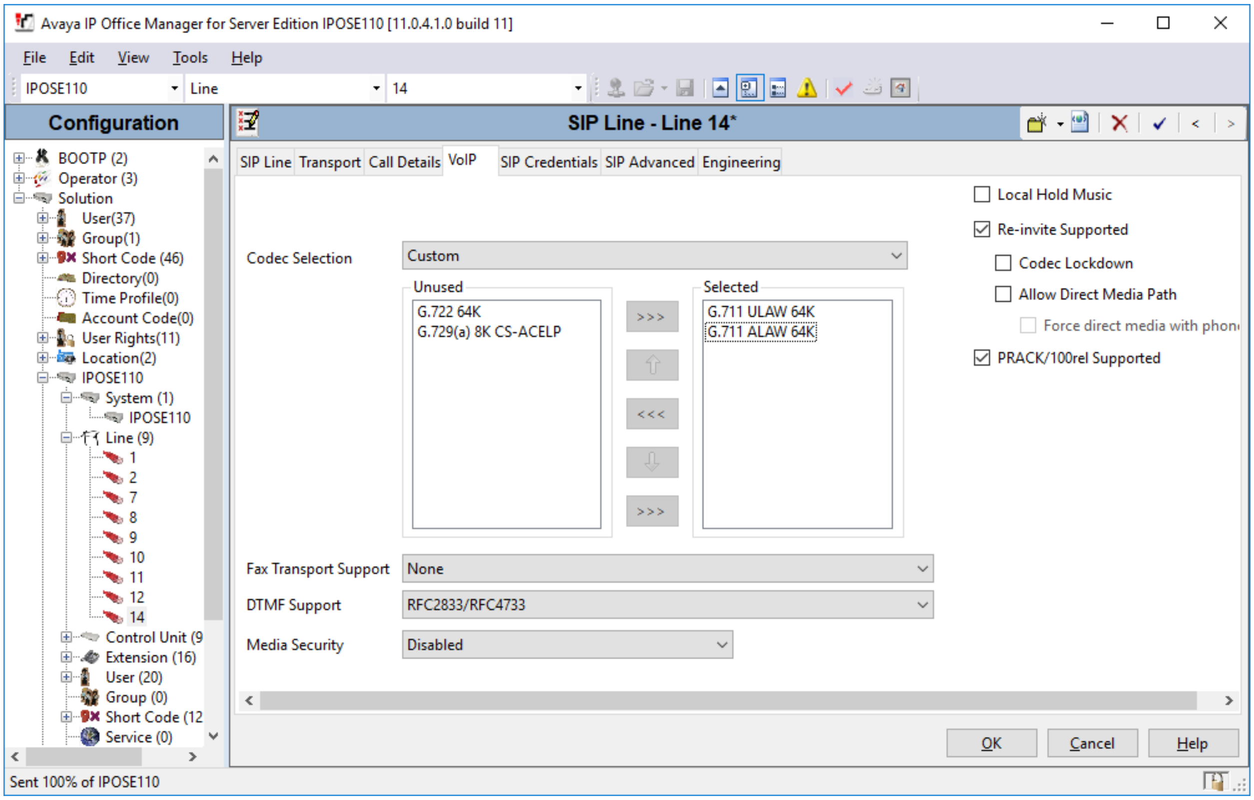
- **Incoming Group:** Select ***14*** as configured earlier in this section.
- **Outgoing Group** Select ***14*** as configured earlier in this section.
- **Local URI:** Select *Auto*.
- **Contact:** Select *Auto*.

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

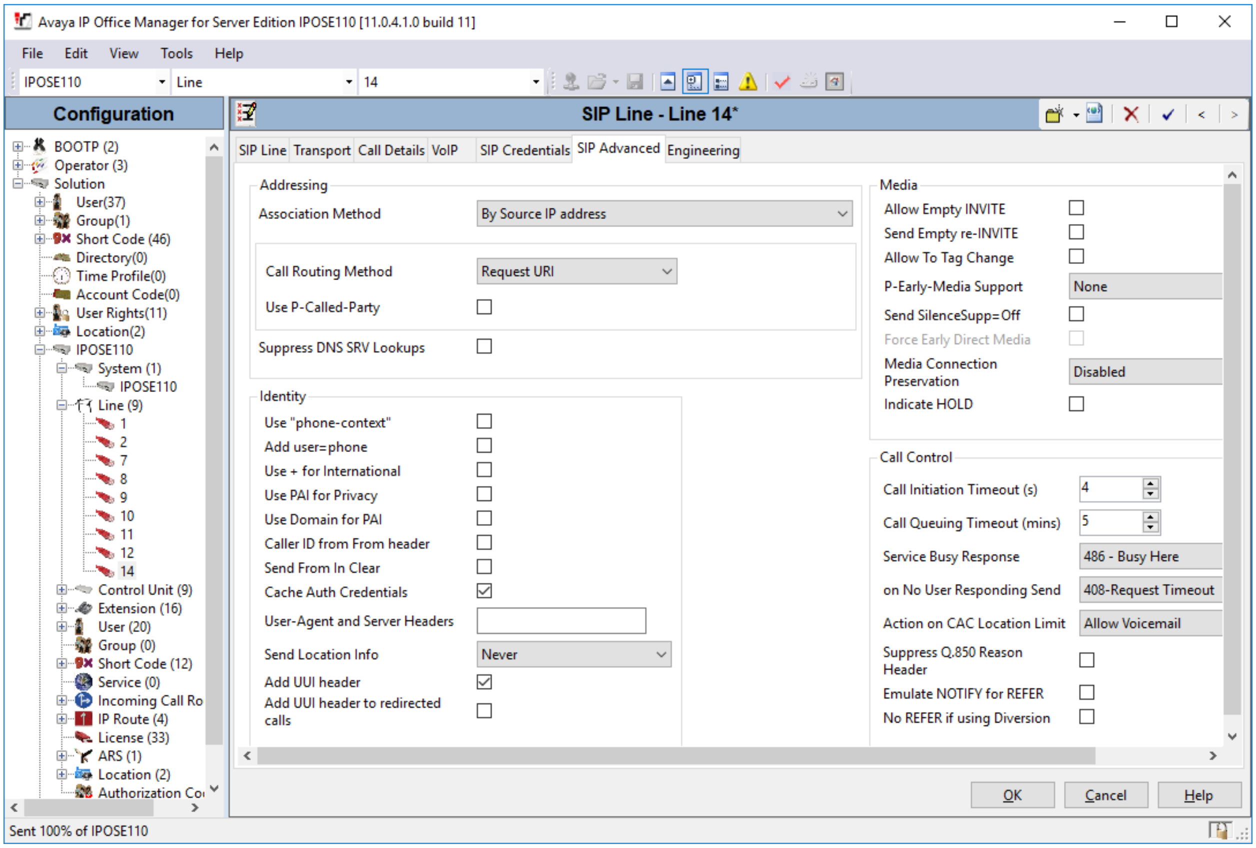


In the **VoIP** tab, ensure that for **DTMF Support**, *RFC2833/RFC4733* is selected from the drop down menu. Select ***Disable*** from the drop down menu for **Media Security**. Check **Re-invite Supported** and **PRACT/100rel Supported** checkbox.

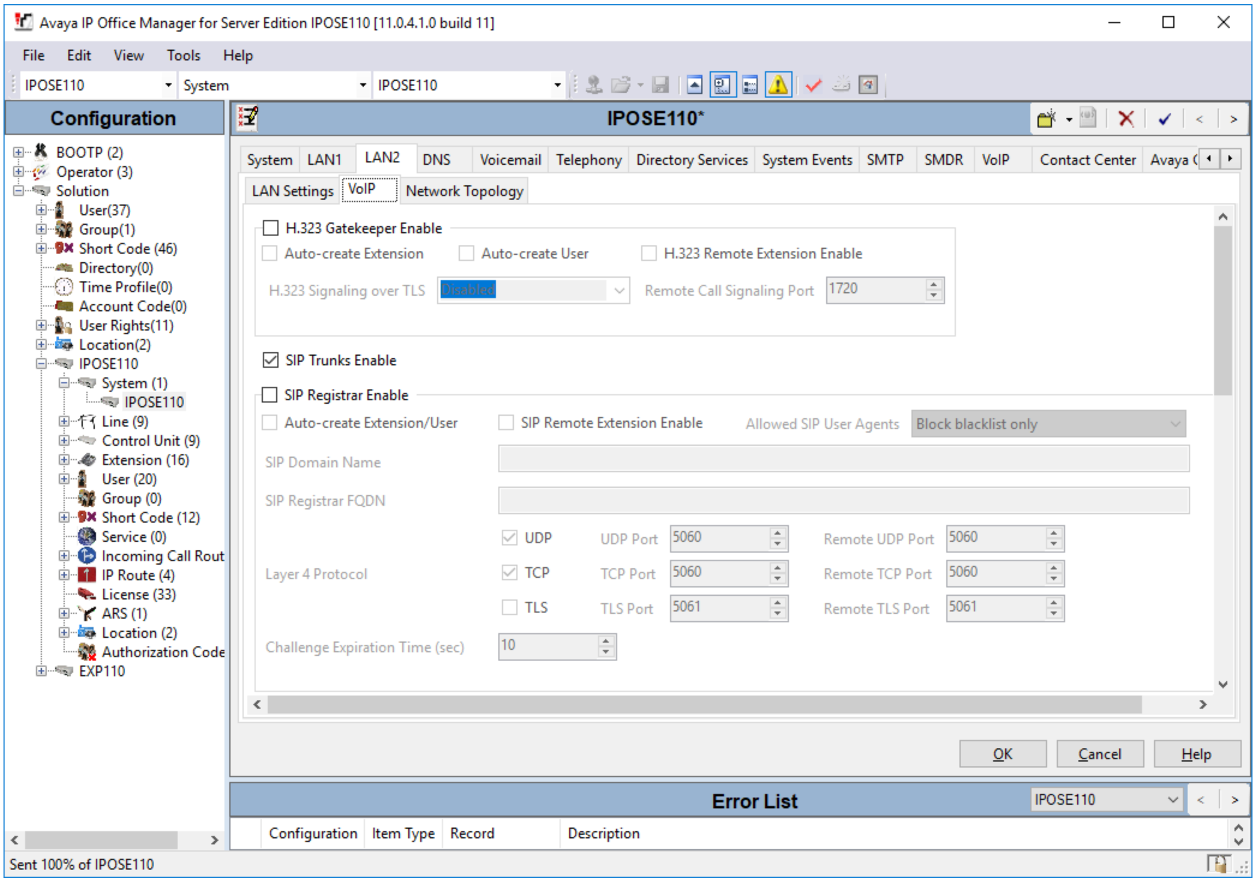
Retain default values for all remaining fields. During compliance testing the **G.711 ULAW** codec was tested as shown in the screen below.



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.

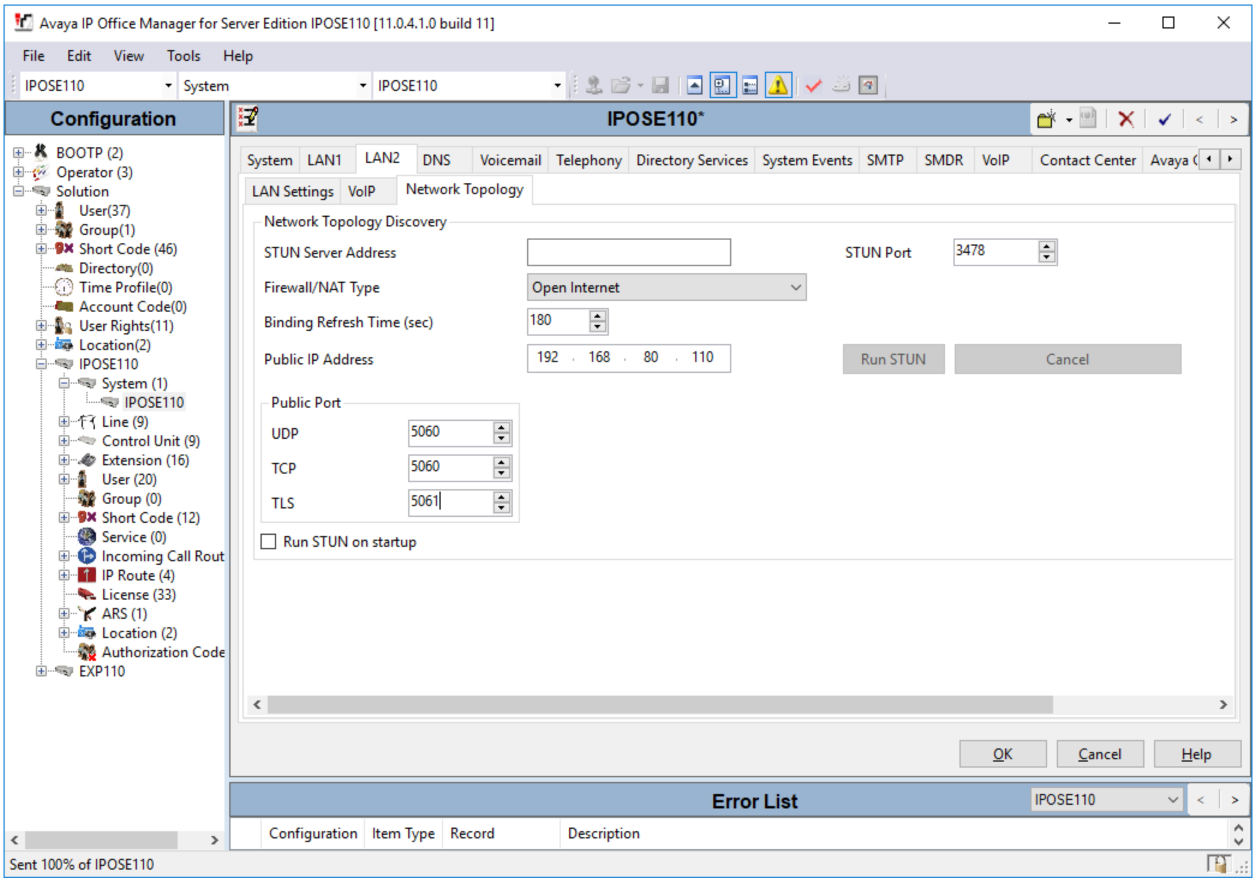


Select the **VoIP** sub-tab. Ensure that **SIP Trunks Enable** is checked, as shown below. Leave other fields as default.



Select the **Network Topology** sub-tab. In the **Network Topology Discovery** section, select ***Open Internet*** in the **Firewall/NAT Type** field, time in seconds in the **Binding Refresh Time (sec)** field and enter the LAN2 IP address in the **Public IP Address** field.

In the **Public Port** section, enter the **UDP** ***5060***, **TCP** ***5060*** and **TLS** ***5061***.

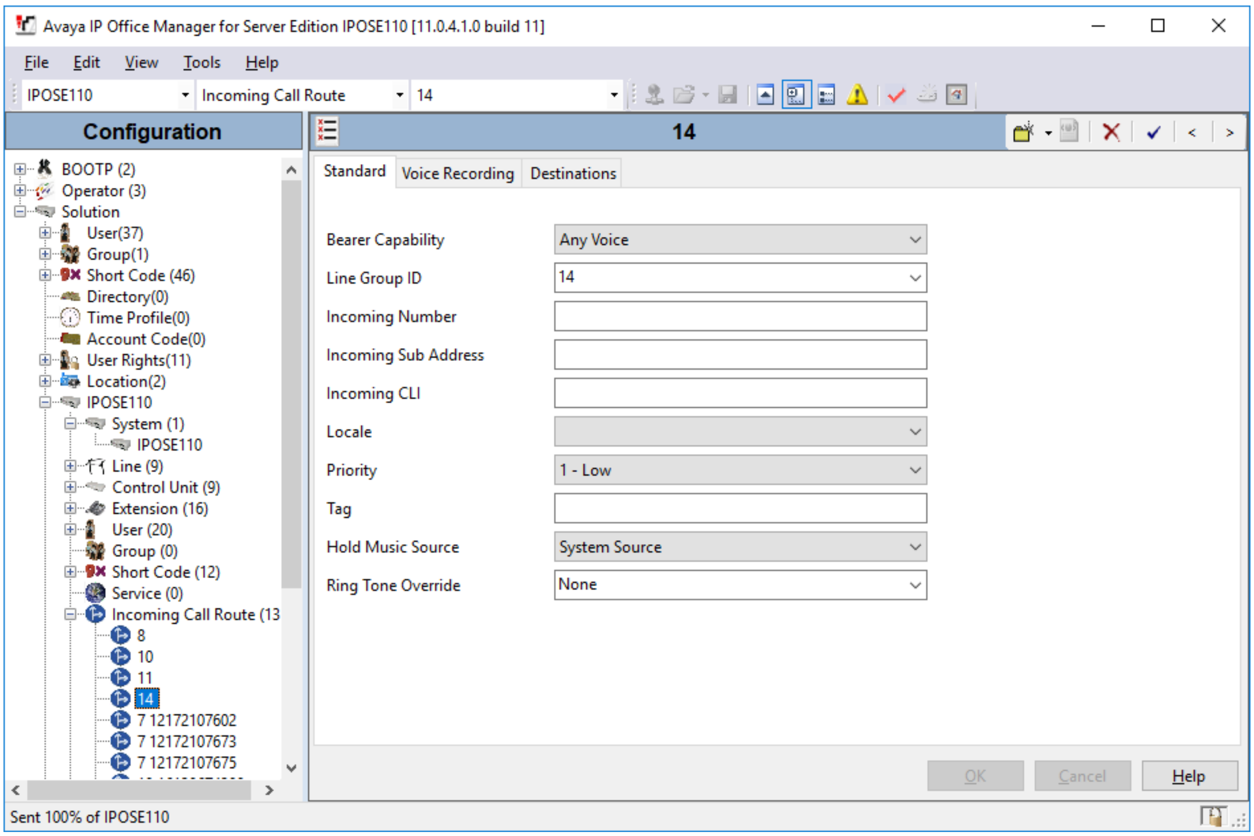


5.4. Configure Incoming Call Route

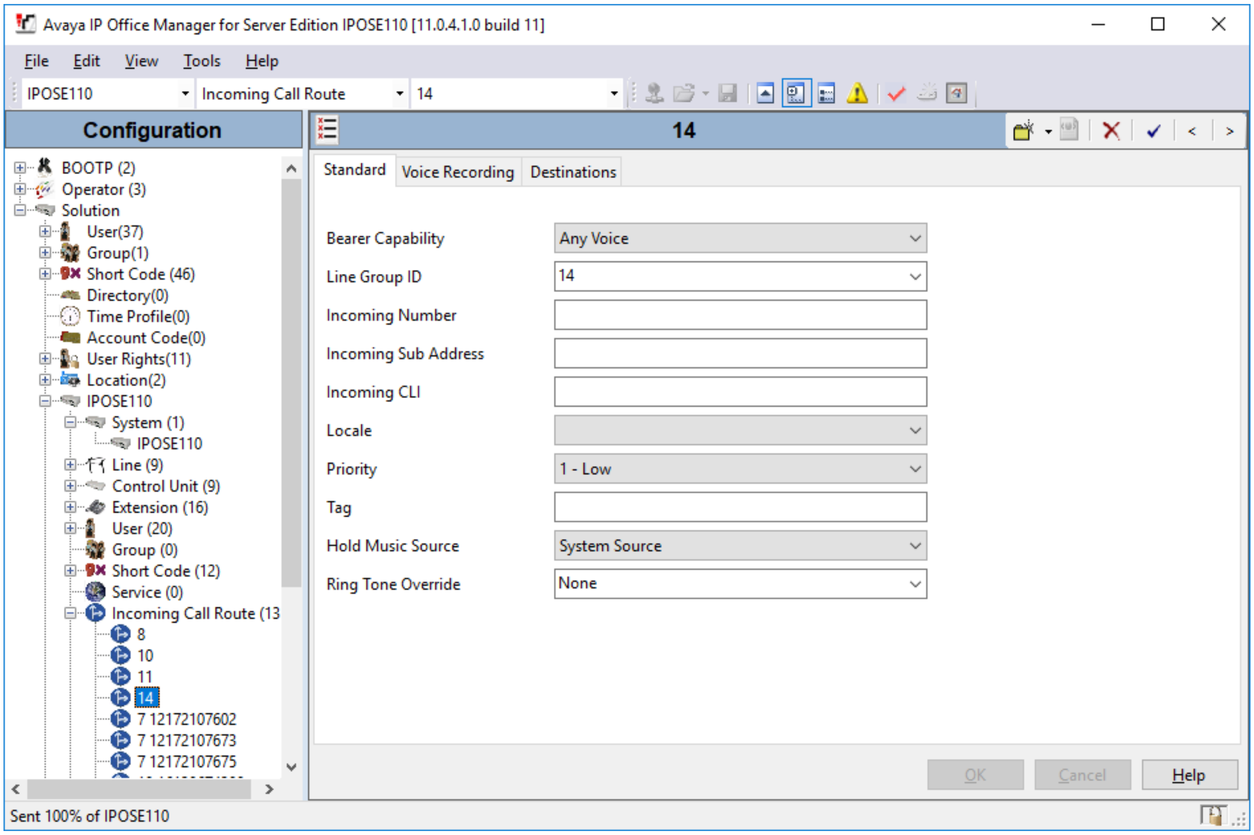
To configure the Incoming Call Route, navigate to **IPOSE110 → Incoming Call Route** and right click on **Incoming Call Route** followed by **New** (not shown). In the subsequent window, enter the following in the **Standard** tab.

- **Bearer Capability:** Select ***AnyVoice*** 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.

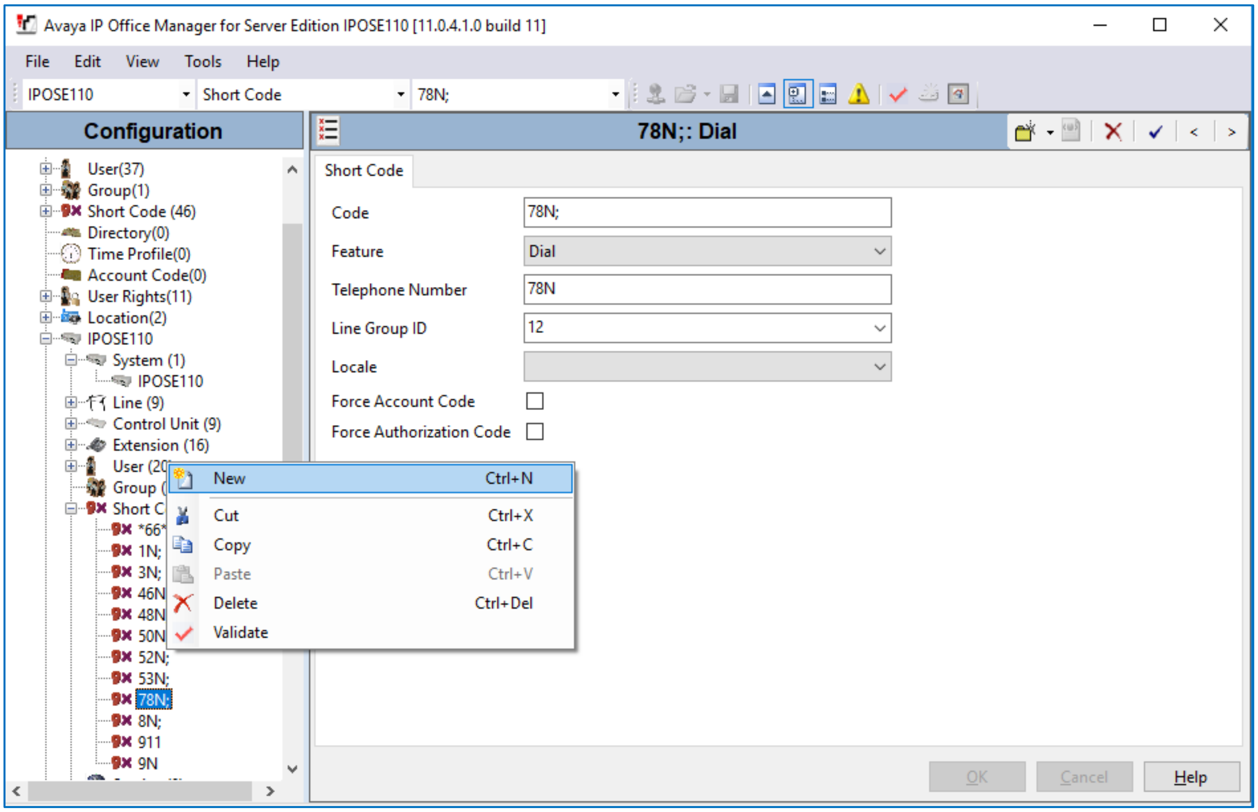


In the **Destinations** tab, under the **Destination** column enter **“.”.** Retain default values for all remaining fields and click the **OK** (not shown) button.



5.5. Create Short Codes

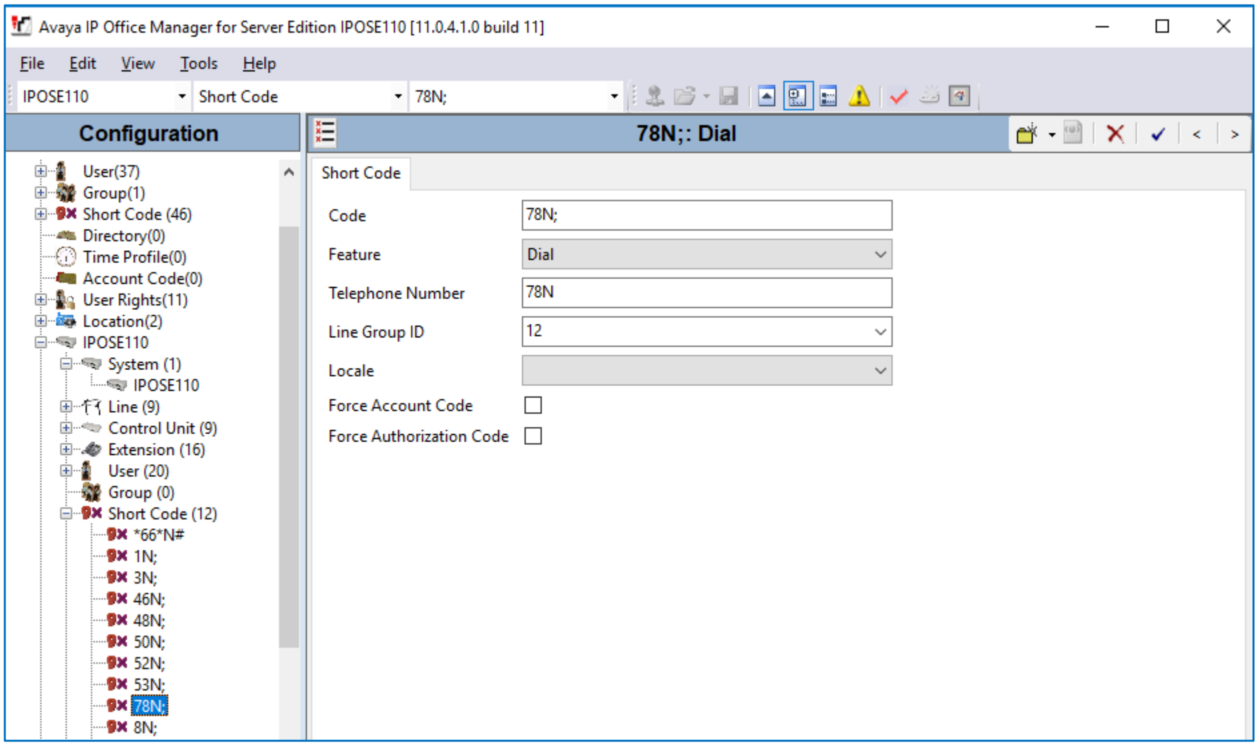
A Short Code needs to be configured on the IP Office to route calls to VCB. Navigate to **IPOSE110 → Short Codes,** and then right click and select **New** as shown in the screen below.



In the subsequent window, enter the following:

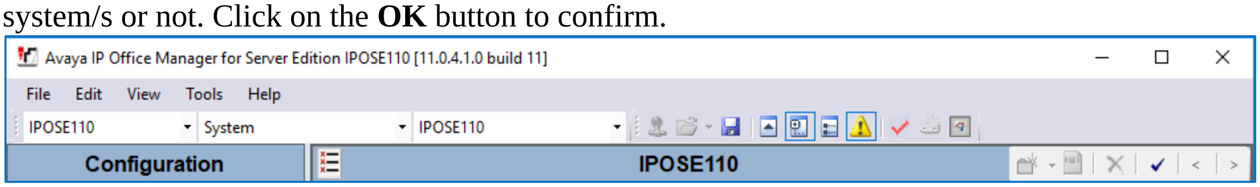
- **Code:** Enter the number range that will be routed to VCB (during compliance testing, all numbers beginning with 78xxx were sent to VCB, therefore *78N;* was entered)
- **Feature:** Select *Dial* from the dropdown menu
- **Telephone Number:** Enter "78N"
- **Group Line ID:** Enter ***12***, the outgoing 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 on 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.

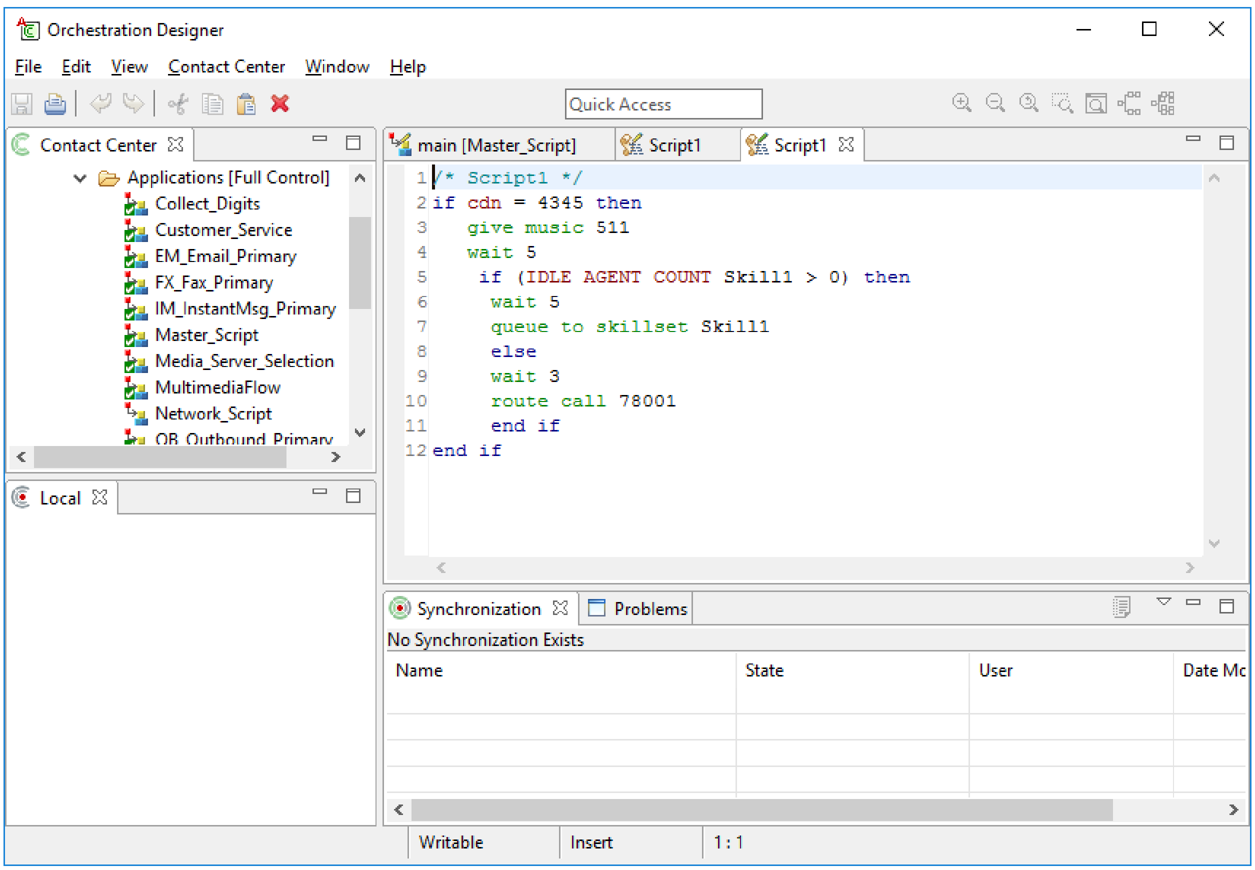


6. Configure Avaya Contact Center Select

The administration of the routing and basic connectivity between Avaya IP Office and Avaya Contact Center Select or the setting up of Skill set, CDN, Scripts, Agents for a contact center type environment are not the focus of these Application Notes; however, some details are provided only for informational purposes and completeness.

This section provides a sample script that was used during the compliance testing. When a call is directed to this script, caller is connected to VCB if there are no agents available in the Skill set queue.

From Avaya Contact Center Select Launchpad, navigate to **Scripting → Orchestration Designer → Launch Orchestration Designer** (not shown) to open the Orchestration Designer window as shown below. A basic script is configured in the example below where when a CDN is reached and if there is no agent available, call is to be routed to Fonolo.

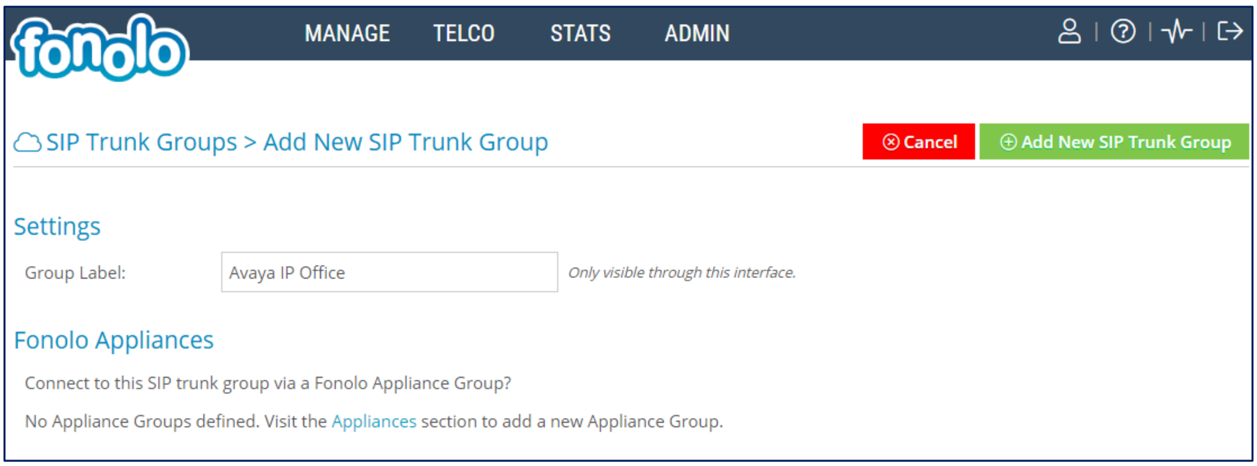


7. Configure Fonolo Voice Call Backs

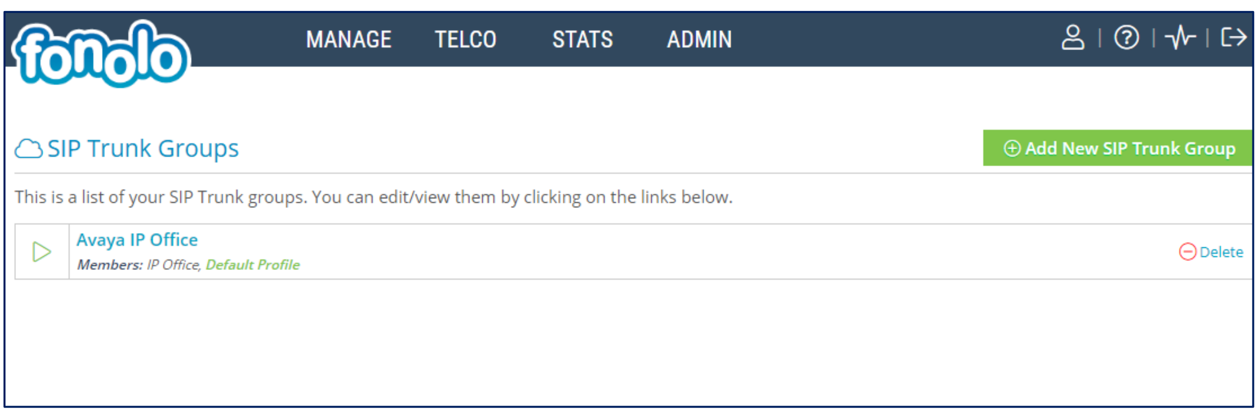
This section provides a "snapshot" of VCB configuration used during compliance testing. VCB is typically configured for customers by Fonolo. The screen shots and partial configuration shown below, supplied by Fonolo, are provided only for reference. They represent only an example of the configuration GUI of VCB, available through the Fonolo Customer Portal at https://portal.fonolo.com/. Other configurations are possible. Contact Fonolo for details on how to configure VCB.

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 right of the page. Define a new label to identify this SIP trunk group. During compliance testing ***Avaya IP Office*** was used as the label.



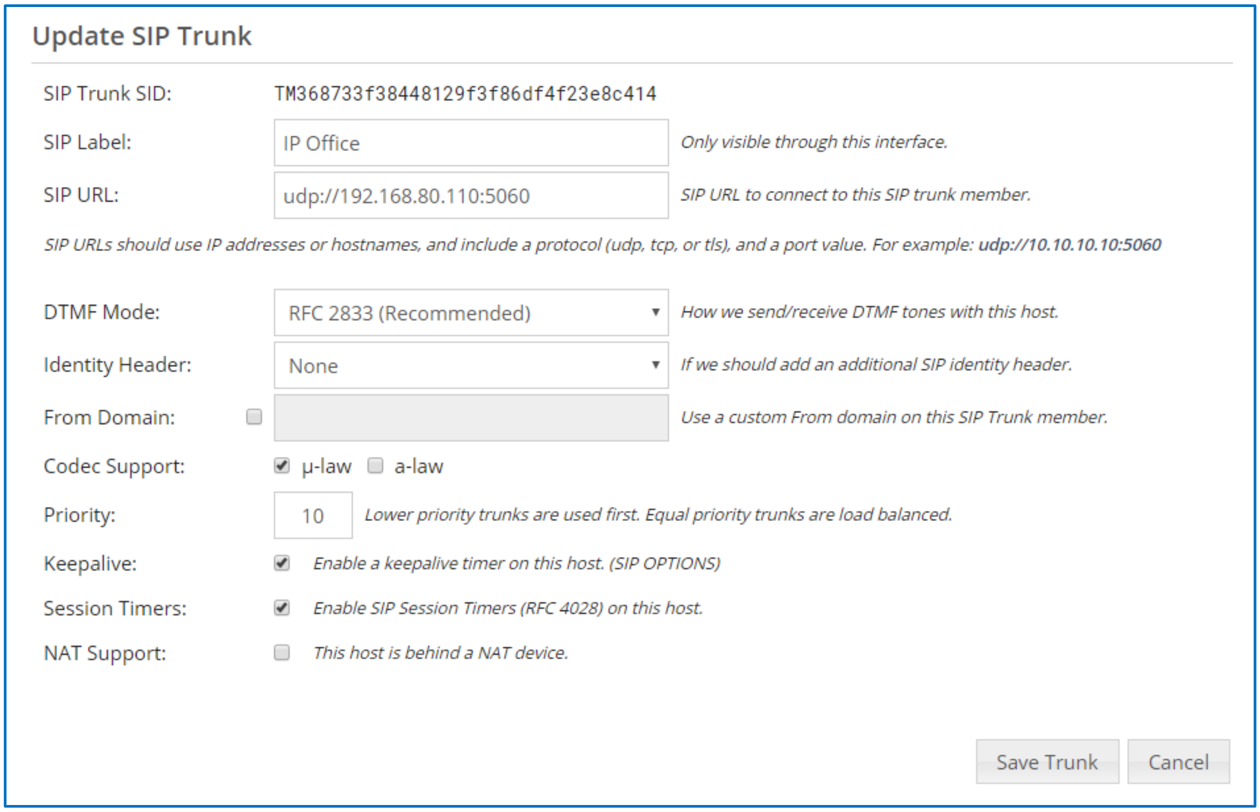
Under the **Members** tab in this new SIP trunk group, click the **Add New Member** button, and the **Add New SIP Trunk** dialog will appear as shown below.



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**: 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 an 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 enabled.
- **Session Timers**: Enable this field to keep session refreshed for long duration call. Default is enabled.
- **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. Default is disabled.

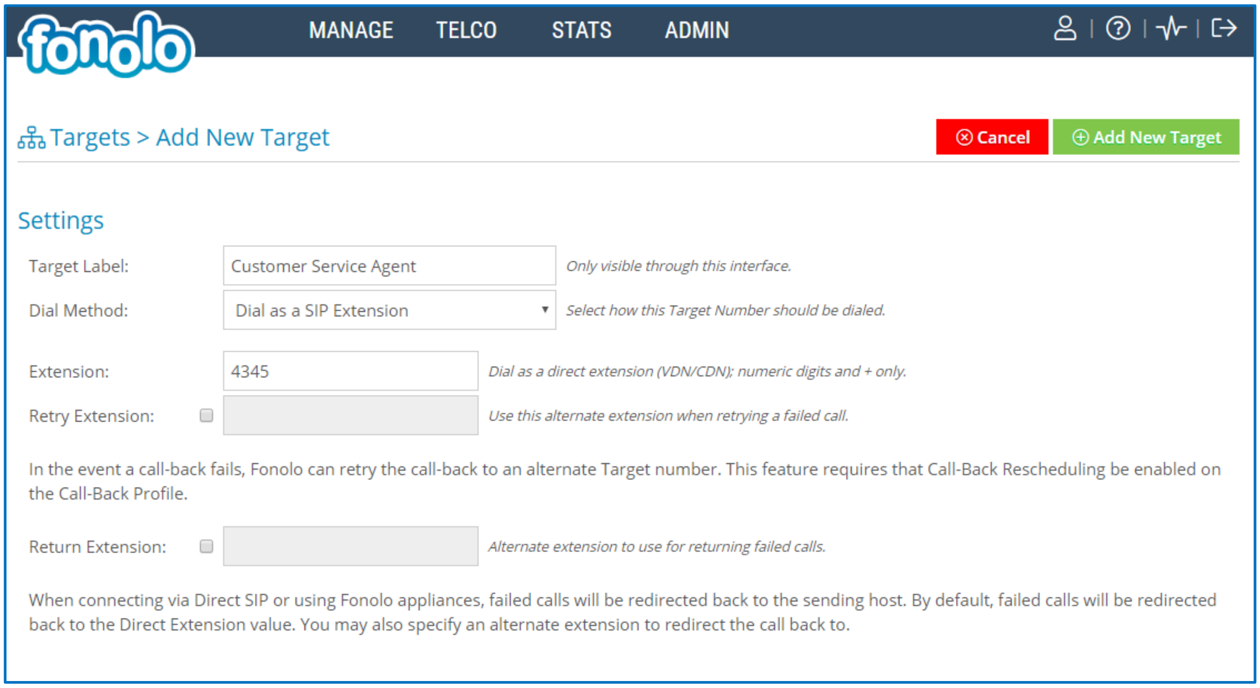
Add the IP address of IP Office, formatted as a fully qualified URL, defining the protocol and SIP port, then click the **Save Trunk** button. During compliance testing, the protocol **UDP** and port **5060** is used for the SIP service to IP Office, and the default values for the remaining SIP trunk group member settings.



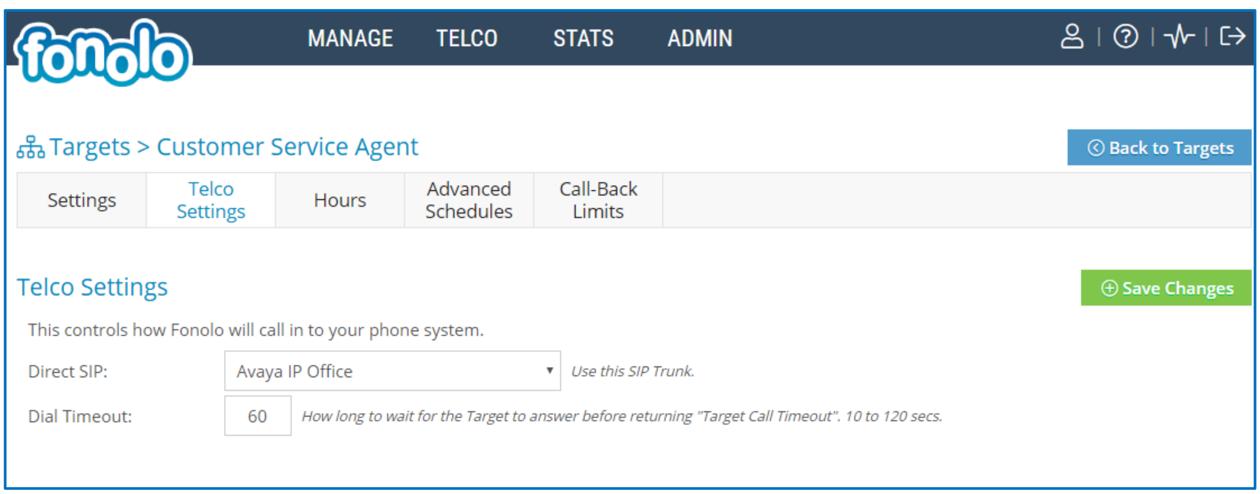
7.2. Add Agent Call Back Endpoint

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 Agent** was used as the label. Select the **Extension** option (shown below), and enter the CDN to reach the skill set queue on IP Office.

During compliance testing, CDN ***4345*** was configured on Contact Center Select which was accessible via IP Office. Then click on the **Add New Target** button to save this Target.



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 ***Avaya IP Office*** SIP trunk, added in **Section** Error! Reference source not found., and then click the **Save Changes** button.

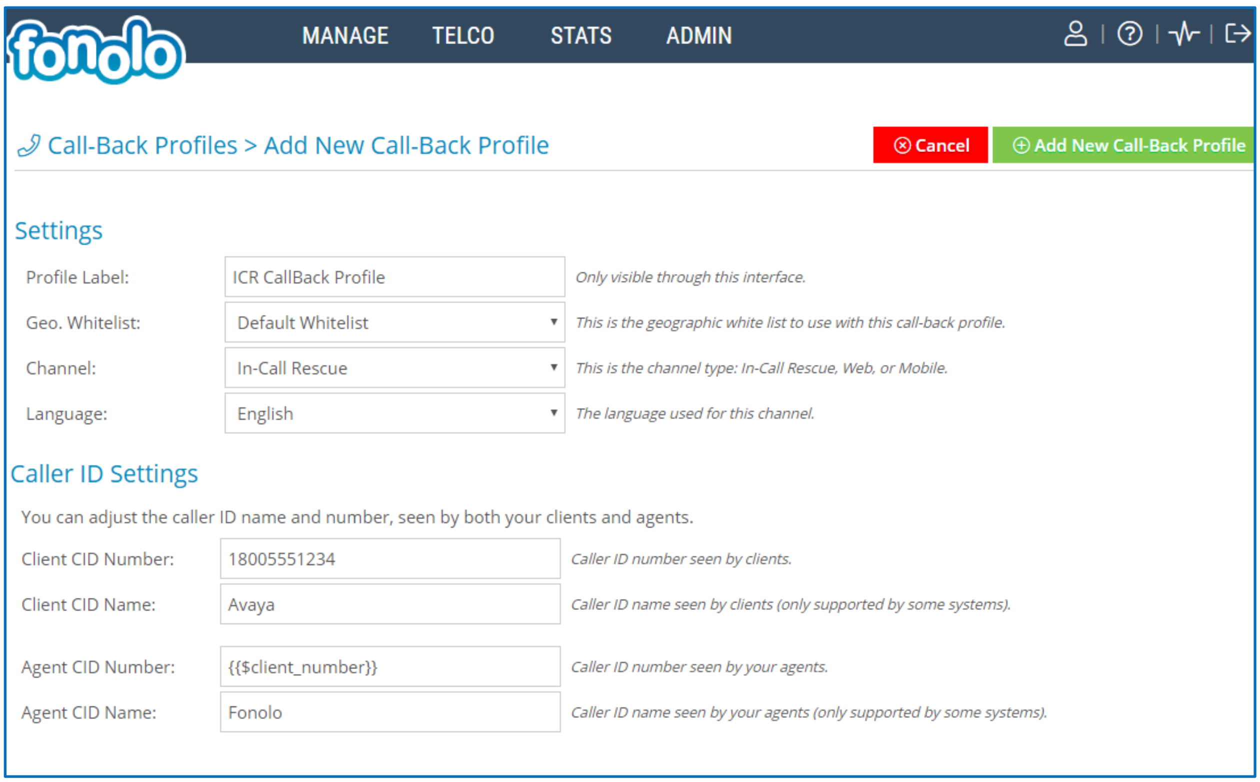


7.1. Add a New Call-Back Profile

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

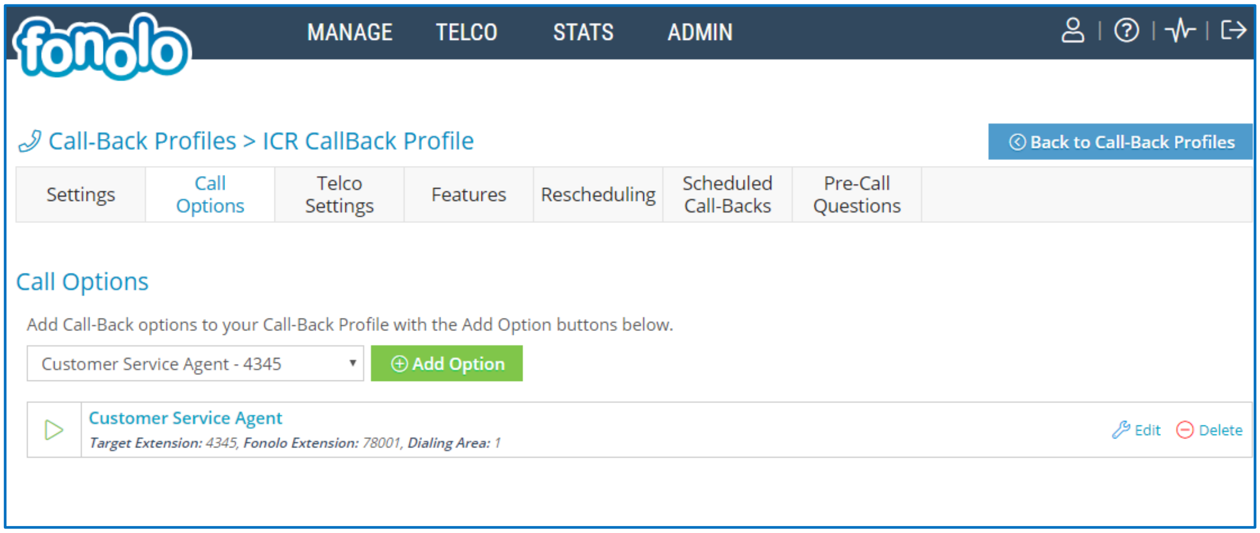
- **Profile Label:** A label to identify this new profile.
- **Geo Whitelist:** A geographic whitelist to use for this new profile.
- **Channel:** Select **Voice Call-Backs**.
- **Language:** Select the appropriate language for this skill set queue.
- **Customer CID Number:** The Caller-ID number the 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.

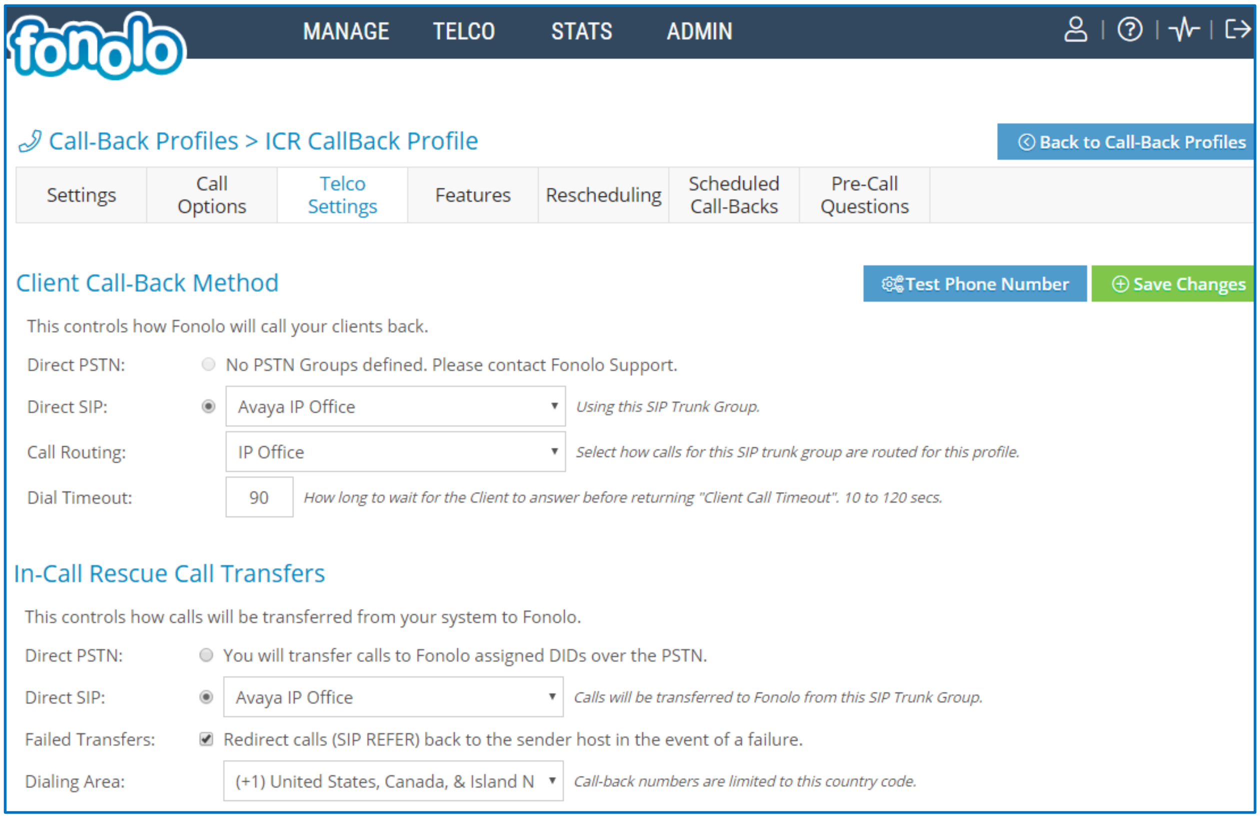


From the **CALL OPTIONS** section of the new **Call-Back Profile**, select the Target added in **Section** Error! Reference source not found. (from the drop-down menu highlighted below), and click the **Add Option** link to add the CDN value to the section on the right, as shown below, then click the **Save Changes** button.

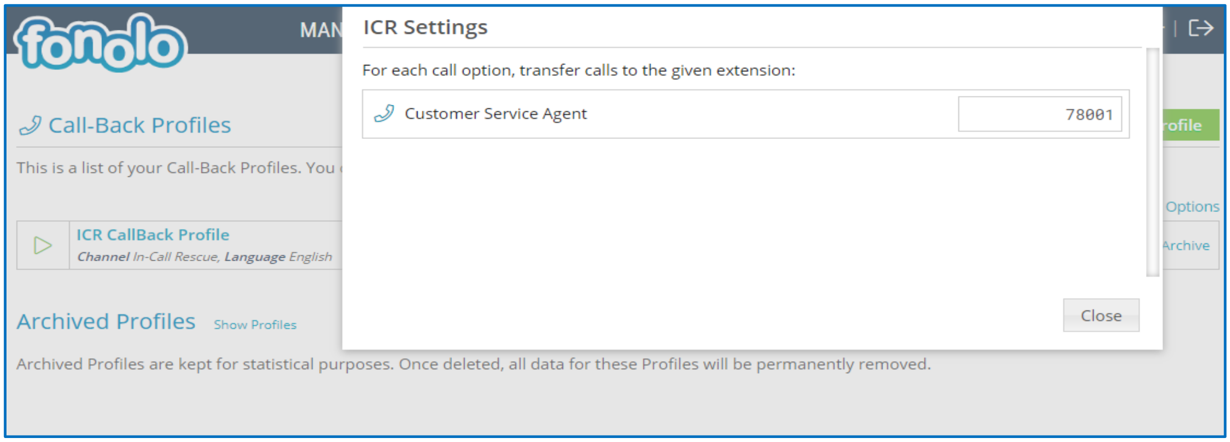
This associates the Target CDN with this new **Call-Back Profile**. Multiple call options can be associated with a single **Call-Back Profile**, one for each skill call-backs are being offered on.



From the **TELCO SETTINGS** section of the new **Call-Back Profile**, select the ***Avaya IP Office*** SIP trunk group created in **Section** Error! Reference source not found. 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.



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

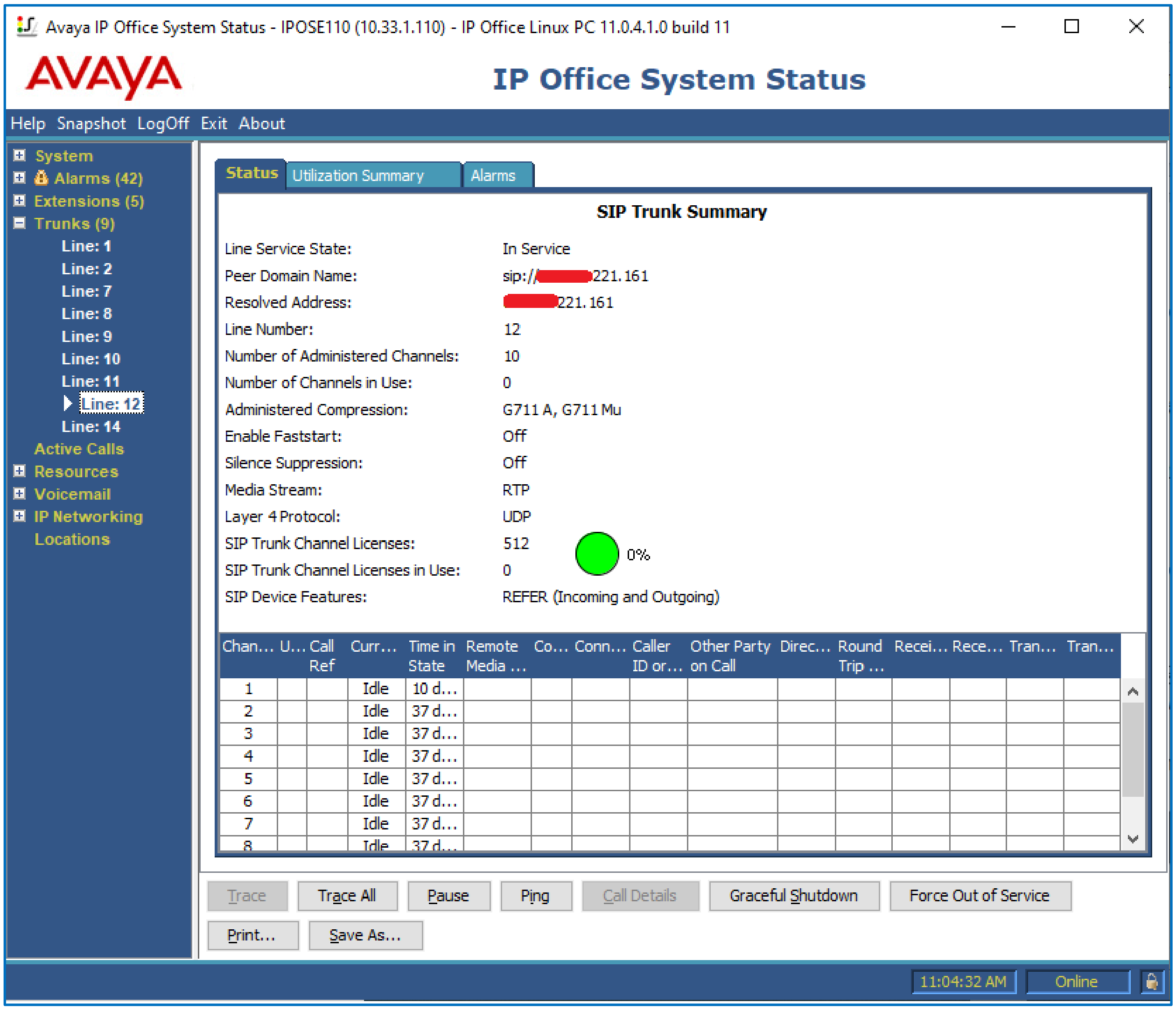


8. Verification Steps

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

8.1. Verify IP Office System Status

From the Avaya IP Office System Status window for the Primary server, verify that the Trunks configured are In Service. Screen bellows shows one of the SIP Line trunk ***12*** has *In Service* under the **Line Service State** field.

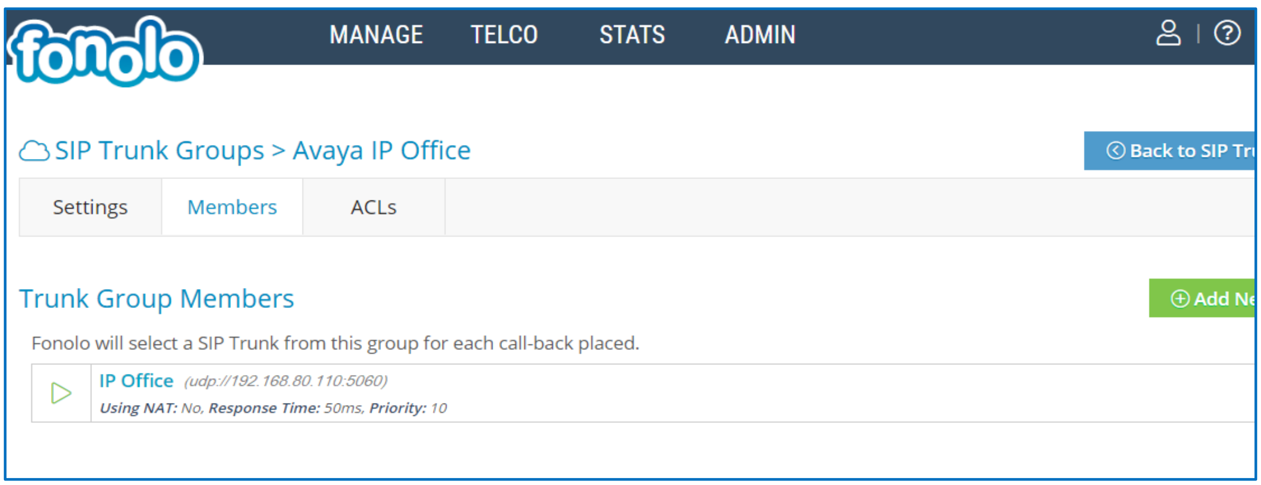


The following tests were also performed to verify proper configuration of VCB with Avaya IP Office.

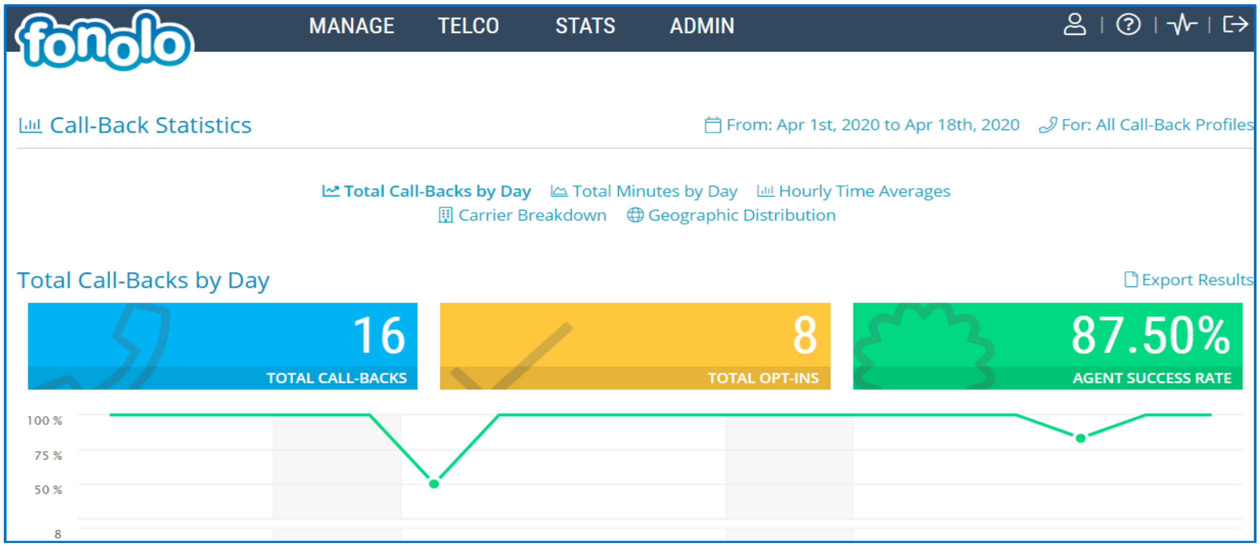
- PSTN caller is able to select the call back option and IP Office is able to direct this call to VCB.
- PSTN caller is able to hear the VCB menu and make the required choices.
- VCB is able to recognize the choices made by the PSTN user.
- VCB is able to call the queue and wait for an available agent.
- VCB is able to call out to the PSTN caller and connect them to an available agent.

8.2. Verify Fonolo Voice Call-Backs

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.



Additional information is available through the **Stats → Graphs** section of the Fonolo customer portal.



9. Conclusion

These Application Notes describe the configuration steps required for Fonolo Voice Call-Backs to successfully interoperate with Avaya IP Office Server Edition. All feature and serviceability test cases were completed and passed with the exceptions/observations noted in **Section** Error! Reference source not found..

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 11, June 2019.
2. *Administering Avaya IP Office™ Platform with Manager,* Release 11, June 2019.
3. *Deploying Avaya IP Office™ Platform IP500 V2,* 15-601042 Issue 32f - (20 July 2019).

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

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.