



DevConnect Program

Application Notes for Configuring Enghouse Trio Version 9 with Avaya IP Office Server Edition Release 11.1 – 1.0

Abstract

These Application Notes describe the configuration steps required for Enghouse Trio to interoperate with Avaya IP Office Server Edition using SIP trunk.

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 Avaya DevConnect Program.

1. Introduction

These Application Notes outline the steps necessary to configure Trio from Enghouse Interactive AB (Trio) to interoperate with Avaya IP Office Server Edition (IP Office). Trio is a client/server-based application running on Microsoft Windows Server operating systems. Trio provides users with attendant answering position for IP Office, as well as a call forward function that provides spoken information about the absent status of a regular user (extension) being called. The Trio Attendant client provides a view of regular users' contacts and schedules and was installed on the same server as the Trio Server but can be installed on a separate platform if required. Trio connects to the IP Office Server Edition using a SIP trunk.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise voice network using IP Office. The Trio server uses a SIP trunk to connect to the IP Office. See **Figure 1** for a network diagram. An Incoming Call Route and Short Code were configured on the IP Office Primary and Expansion to route calls to Trio. Calls placed to the Trio server automatically places a call to the telephone the Attendant is using for answering purposes. When the attendant answers the call, the Trio server bridges the two calls. When the attendant extends the call to another telephone, Trio server performs a SIP path replacement, and the caller and the called user are now directly connected. It is possible to have multiple Trio attendant positions on an IP Office system.

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 Trio did not include use of any specific encryption features as requested by Enghouse Interactive AB.

2.1. Interoperability Compliance Testing

The interoperability compliance testing included feature and serviceability testing. The serviceability testing introduced failure scenarios to see if Trio could resume after a link failure with IP Office. The testing included:

- Incoming internal and external calls
- Outgoing internal and external calls
- Supervised and unsupervised transfer with answer
- Directing calls from busy extensions and extensions that do not answer
- Call queuing and retrieval

2.2. Test Results

The tests were all functional in nature and performance testing was not included. All test cases passed successfully.

2.3. Support

For technical support for Enghouse Interactive AB products, please use the following web link.
<http://www.trio.com/web/Support.aspx>

Enghouse Interactive AB can also be contacted as follows.

- **Phone:** +46 (0)8 457 30 00
- **Fax:** +46 (0)8 31 87 00
- **E-mail:** triosupport@enghouse.com

3. Reference Configuration

Figure 1 illustrates the network topology used during compliance testing. The Avaya solution consists of an IP Office Server Edition which consist of Primary and Expansion System, has a SIP Trunk connection from Primary server to the Trio server.

SIP and H323 phones were configured on the IP Office to generate outbound/inbound calls to/from the PSTN. A SIP trunk was configured to connect to the simulated PSTN. An Avaya H.323 station on Primary was used as the Trio Attendant telephone during compliance testing.

Note: The Trio Attendant (client) was installed on the same server as the Trio Server but can be installed on a separate platform if required.

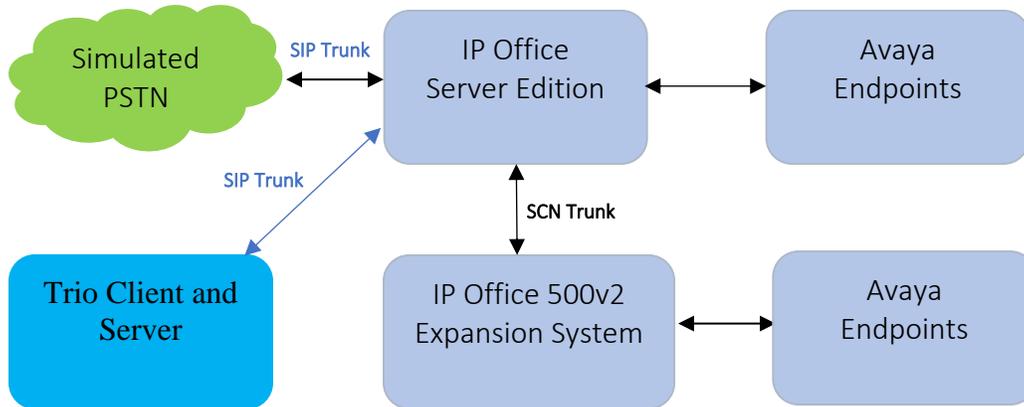


Figure 1: Avaya and Trio Reference Configuration

4. Equipment and Software Validated

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

Equipment	Release/Version
Avaya IP Office Server	11.1.2.4.0 Build 18
Avaya IP Office 500V2	11.1.2.4.0 Build 18
Avaya Endpoints: 9641GS (H.323)	6.854
J189 (SIP)	4.1.0.0.23
Enghouse Interactive AB Trio running on Microsoft Windows 2016 Server	9.2

Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office IP500V2 and also when deployed with IP Office Server Edition in all configurations

5. Configure Avaya IP Office Server Edition

The document assumes that IP Office Server Edition has been installed and configured.

Configuration and verification operations on the IP Office illustrated in this section were all performed using Avaya IP Office Manager. 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:

- Launch Avaya IP Office Manager
- Verify IP Office license
- Obtain LAN IP address
- Enable SIP trunks
- Administer SIP line
- Administer incoming call route
- Administer short code
- Save Configuration

5.1. Launch Avaya IP Office Manager

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 (not shown). The **Avaya IP Office Manager for Server Edition** screen is displayed as shown in the screen below. Click on **Configuration** that is highlighted on the right side of the screen below.

The screenshot shows the Avaya IP Office Manager for Server Edition application. The main window displays the 'Server Edition' configuration for 'IPOSE110'. The 'Open...' menu on the right is expanded, and the 'Configuration' option is highlighted with a red rectangular box. Below the summary information, there is a table with the following data:

Description	Name	Address	Primary Link	Users Configured	Extensions Configured
Solution				24	25
Primary Server	IPOSE110	10.33.1.110		19	13
Expansion System	IPO500	192.168.11.55	Bothway	5	12

5.2. Verify IP Office License

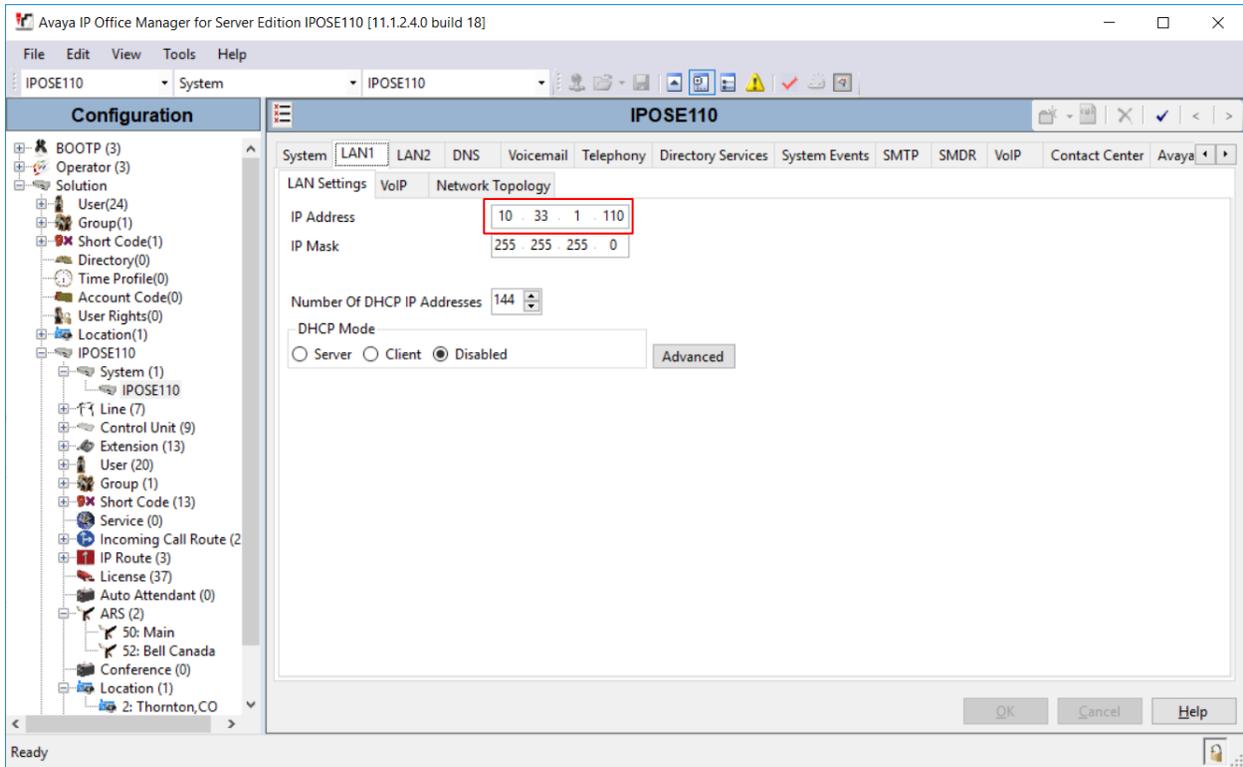
Once the **Avaya IP Office Manager for Server Edition** screen is displayed, from the configuration tree in the left pane, select the Primary System, which in this case is **IPOSE110** and click on **License** to display the **License** screen in the right pane. Verify that the **Feature** for **SIP Trunk Channels Status** is “Valid”, and that the **Instances** value is sufficient for the desired maximum number of simultaneous calls. If there is insufficient capacity of SIP Trunks, contact an Avaya representative to make the appropriate changes.

The screenshot shows the Avaya IP Office Manager for Server Edition interface. The left pane displays a configuration tree with 'IPOSE110' selected. The right pane shows the 'License' configuration screen with a table of features and their instances.

Feature	Instances	Status	Expiration Date	Source
Avaya Softphone Licence	1000	Obsolete	Never	PLDS Nodal
SM Trunk Channels	128	Obsolete	Never	PLDS Nodal
Web Collaboration	64	Obsolete	Never	PLDS Nodal
Avaya Contact Center Select	5	Obsolete	Never	PLDS Nodal
Allow Virtualization	5	Valid	Never	PLDS Nodal
Devlink3 External Recorder	5	Obsolete	Never	PLDS Nodal
Basic User	1000	Obsolete	Never	PLDS Nodal
Call Analytics	5	Obsolete	Never	PLDS Nodal
SP Soft Branding	5	Obsolete	Never	PLDS Nodal
VM Media Manager	5	Obsolete	Never	PLDS Nodal
Additional Voicemail Pro Ports	2	Valid	Never	WebLM
VMPro TTS Professional	1	Valid	Never	WebLM
Power User	4	Valid	Never	WebLM
Avaya IP endpoints	12	Valid	Never	WebLM
SIP Trunk Channels	100	Valid	Never	WebLM
CTI Link Pro	1	Valid	Never	WebLM
3rd Party IP Endpoints	1	Valid	Never	WebLM
Server Edition	1	Valid	Never	WebLM
SM Trunk Channels	128	Valid	Never	WebLM
Avaya Contact Center Select	1	Valid	Never	WebLM
VM Media Manager	1	Valid	Never	WebLM

5.3. Obtain LAN IP Address

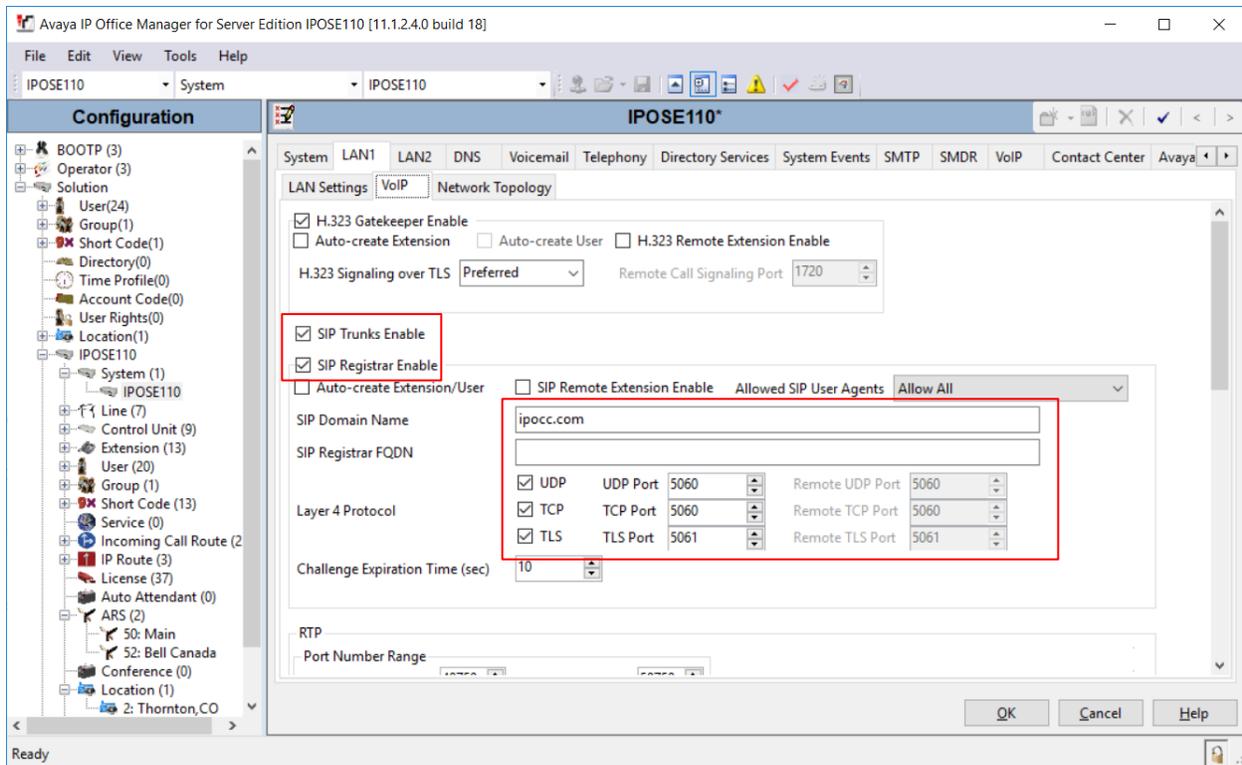
From the configuration tree in the left pane, navigate to **System (1)** to display the IP Office screen in the right pane, where **IPOSE110** is the name of the IP Office Primary system. Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the **IP Address**, which will be used later while configuring Trio in **Section Error! Reference source not found.** Note that IP Office can support SIP trunks on the LAN1 and/or LAN2 interfaces, and the compliance testing used the LAN1 interface.



5.4. Enable SIP Trunks

Select the **VoIP** sub-tab and ensure the configuration is as shown below:

- Check **SIP Trunks Enable** box.
- Check **SIP Registrar Enable** box.
- **Domain Name:** During compliance testing “ipocc.com” was used.
- Check **UDP** and **TCP** protocol with the correct port numbers. However, during testing only UDP was used.

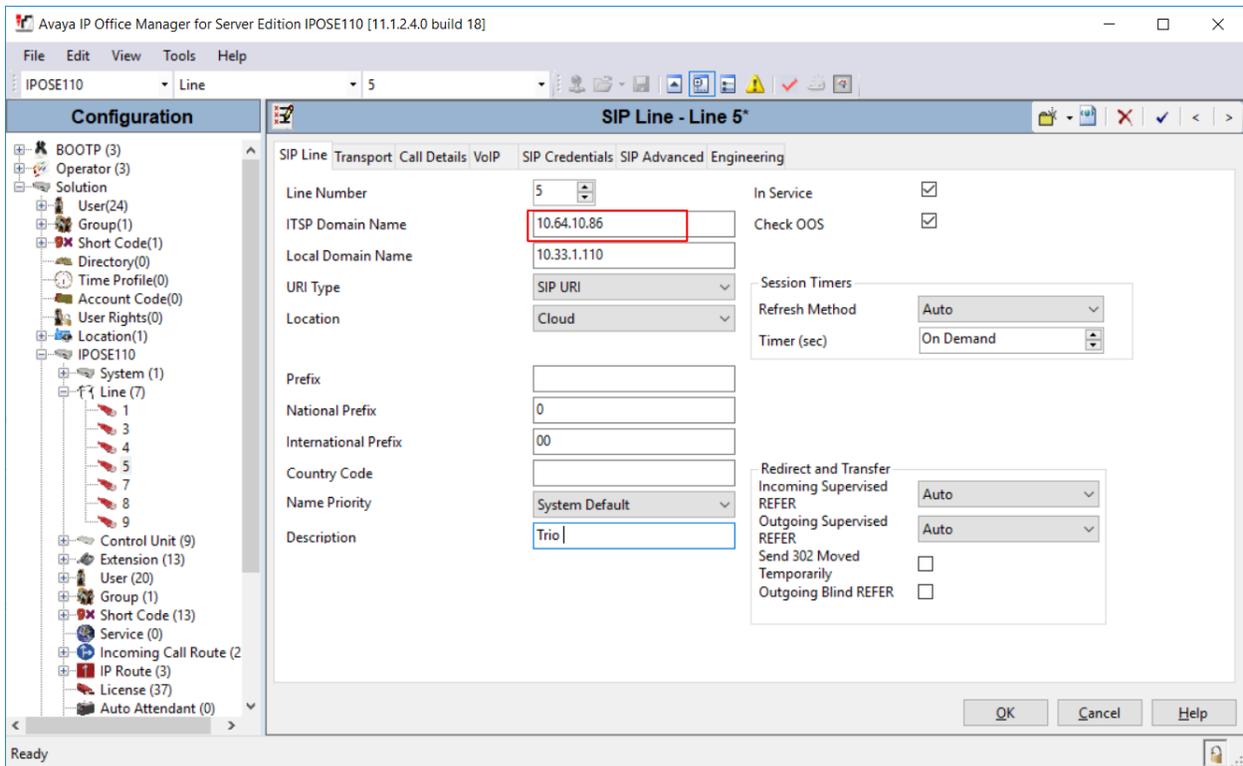


5.5. Administer SIP Line

From the configuration tree in the left pane, right-click on **Line**, now select **New** → **SIP Line** from the pop-up list to add a new SIP line (not shown). During compliance testing **Line 5** was added. Select the **SIP Line** tab in the right pane and configure the following:

- **ITSP Domain Name:** IP address of Trio Server.
- Enter an appropriate description in the **Description** field (optional).

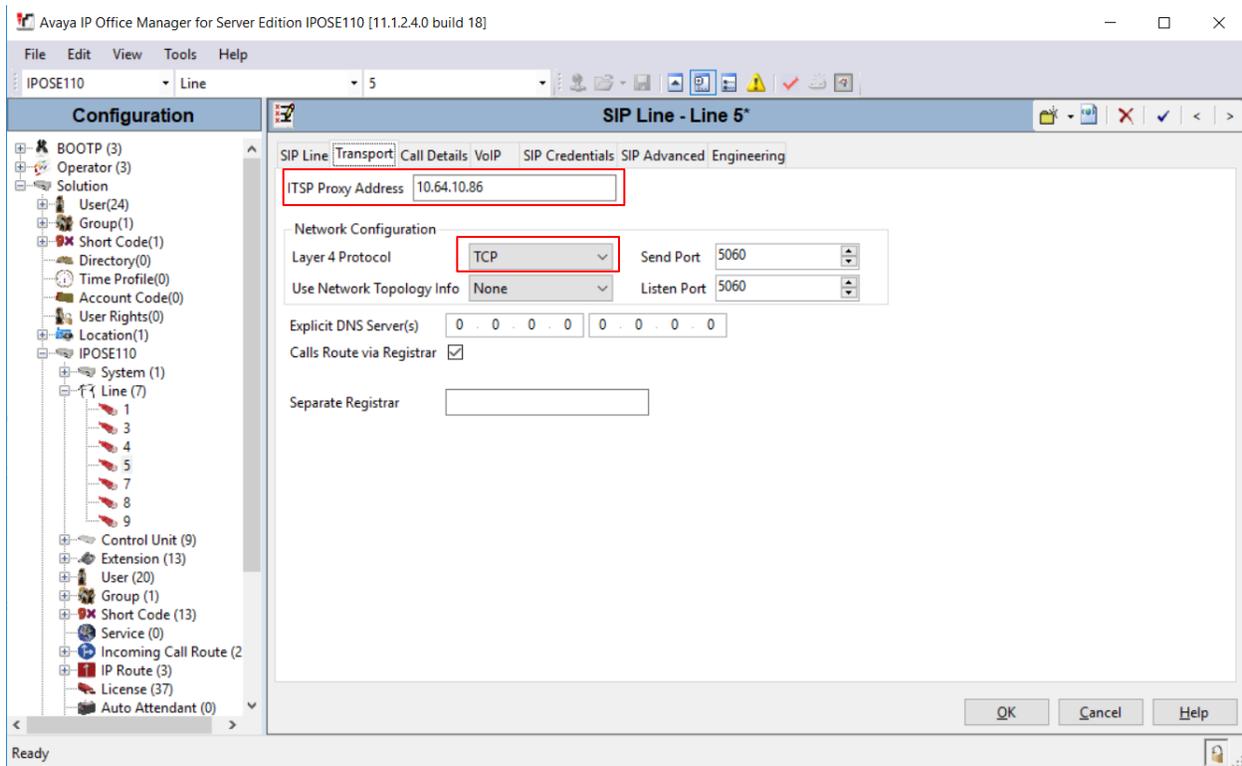
Retain default values for all other fields.



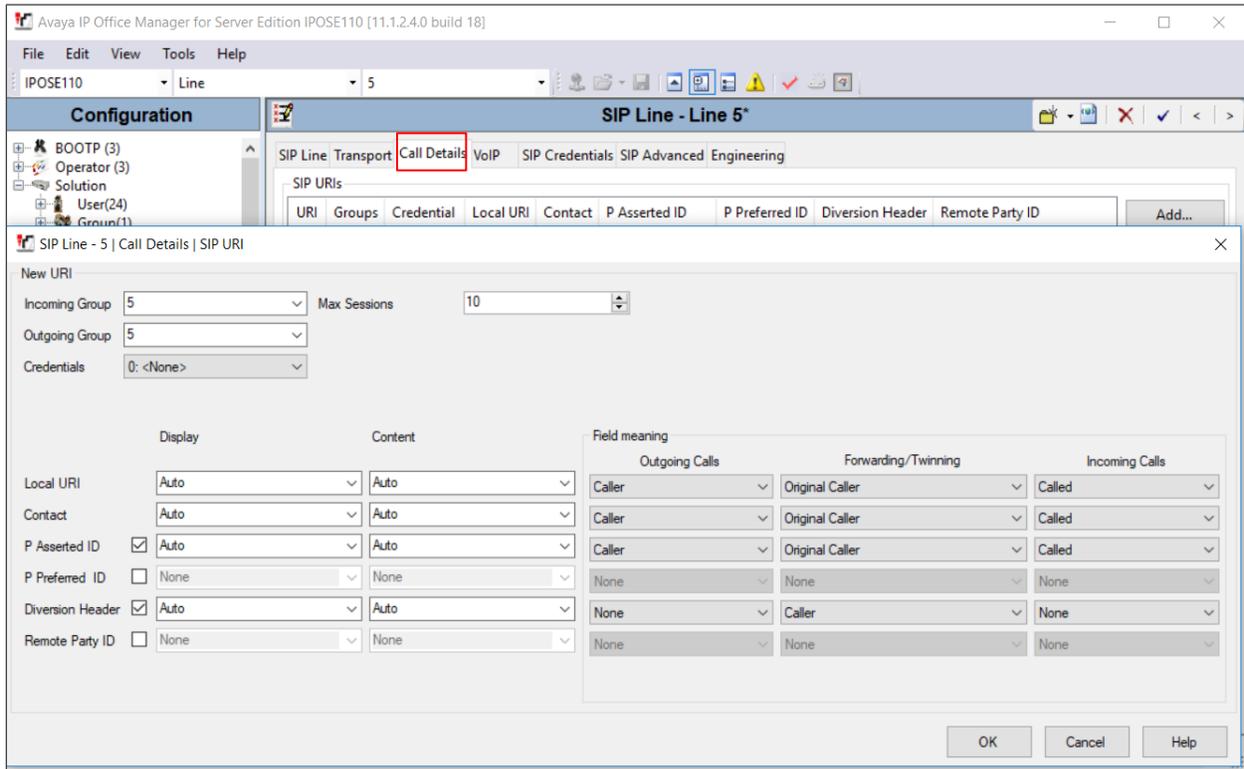
Select the **Transport** tab in the right pane and configure the following:

- **ITSP Proxy Address:** IP address of Trio Server.
- Under **Network Configuration** → **Layer 4 protocol**, select “TCP” and its **Send Port** as “5060”.

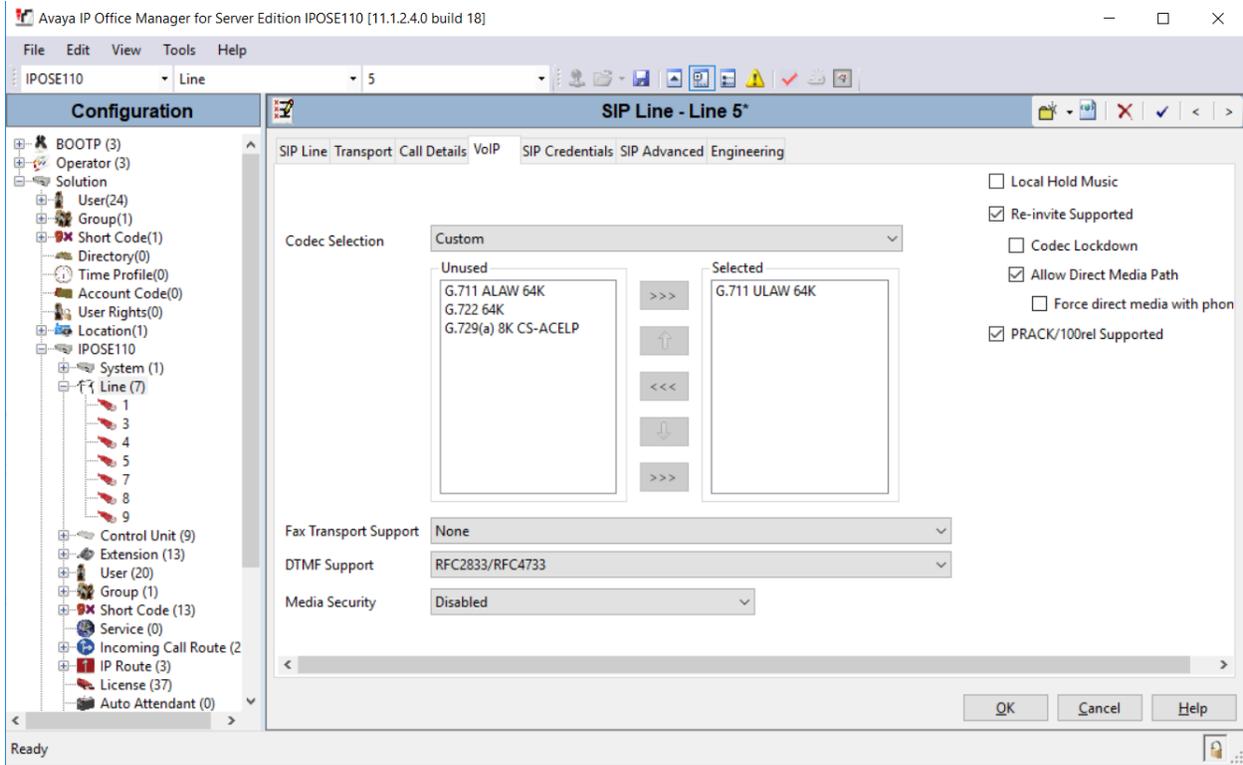
Retain default values for all other fields.



Select the **Call Details** tab and under **SIP URIs** click on **Add** to display the **New URI** section as shown below. Enter an unused group number such as “5” for **Incoming Group** and **Outgoing Group**. Set **Max Sessions** to the maximum number of simultaneous calls allowed, during compliance testing “10” was configured. Enable the **P Asserted ID** and **Diversion Header** boxes and retain the default values in the remaining fields. Click **OK**.

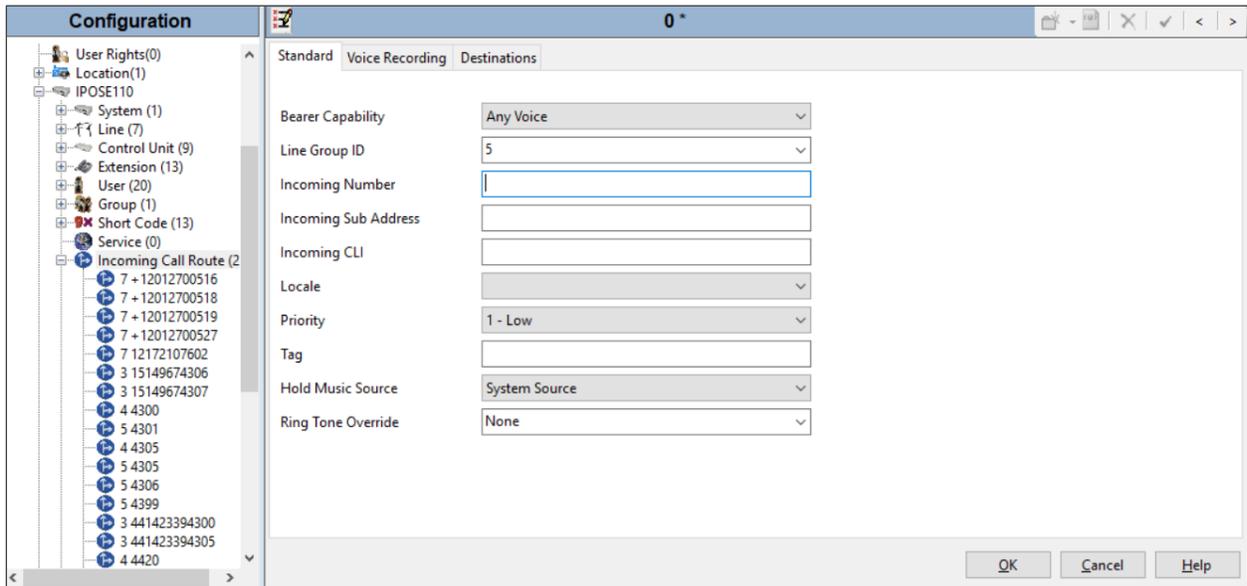


Select the **VoIP** tab. The default **Codec Selection** in the system is shown below and the same was used for compliance testing. In the **VoIP** tab ensure that for **DTMF Support**, **RFC2833/RFC4733** is selected from the drop-down menu. Retain default values for all remaining fields. During compliance testing only, the **G.711ULAW** codec was tested.

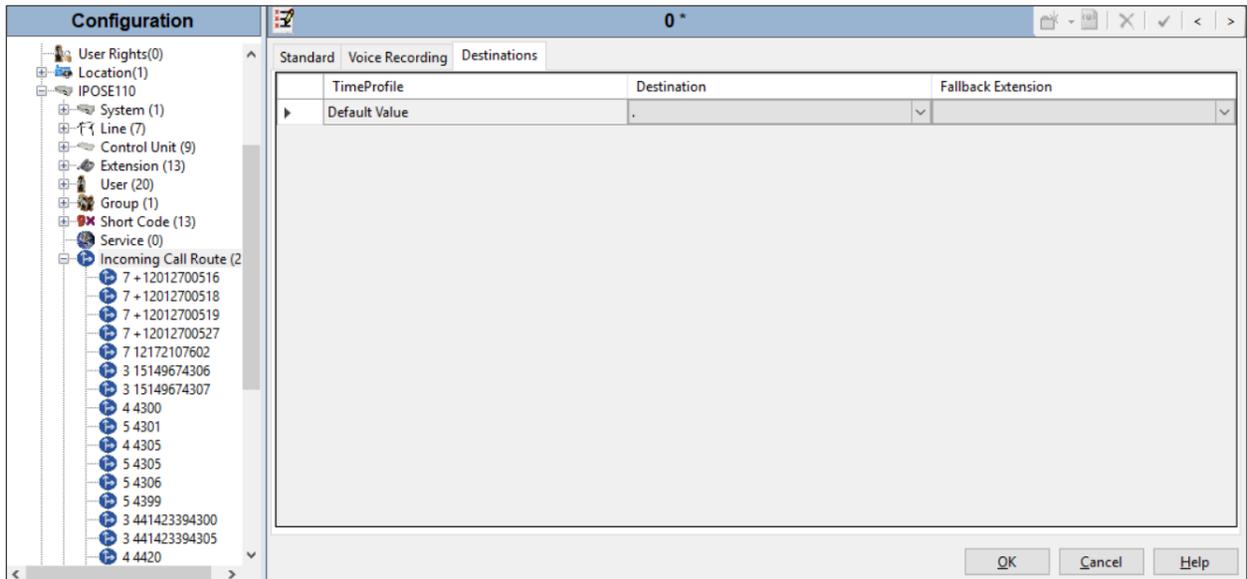


5.6. Administer Incoming Call Route

From the configuration tree seen in the left pane, right-click on the **Incoming Call Route**. Select **New** from the pop-up list (not shown) to add a new route. For **Line Group ID**, select the incoming group number from **Section 5.5**, in this case “5”. Retain default values for all other fields.



Select the **Destinations** tab. For **Destination**, enter “.” to match any dialed number from Trio and click on the **OK** button to complete the configuration.



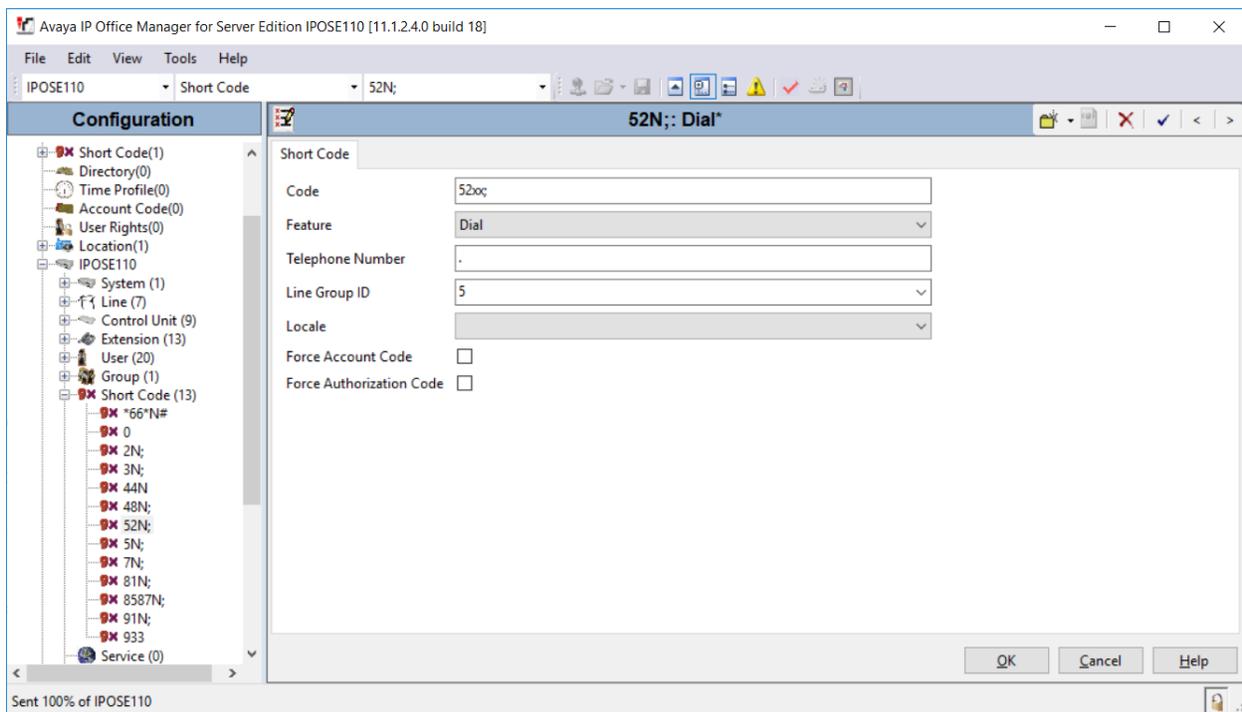
5.7. Administer Short Code

From the configuration tree in the left pane, right-click on **Short Code** and select **New** from the pop-up list (not shown) to add a new short code to route calls to Trio. In the compliance testing, 52XX dialing plan was used for calls to be routed over the SIP trunks to Trio.

Configure the following values:

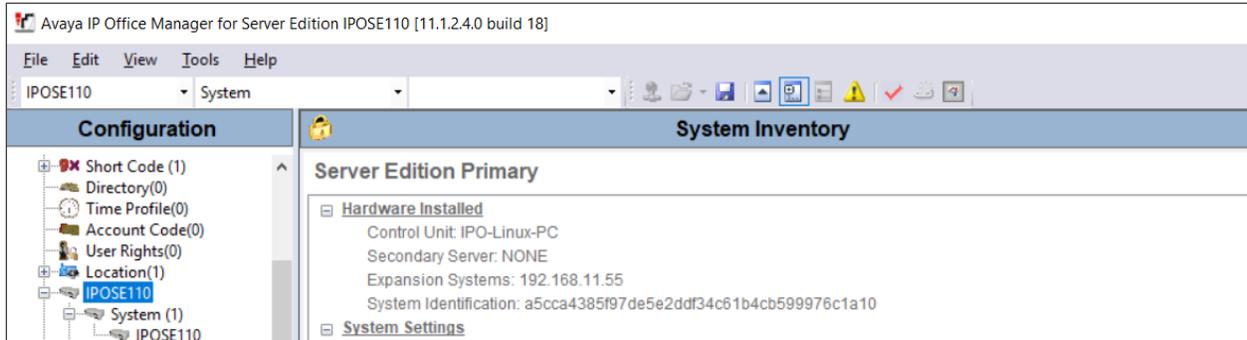
- **Code:** Enter “52XX”.
- **Feature:** Keep the default value of “Dial”.
- **Telephone Number:** Enter “.”.
- **Line Group ID:** Select “5 which is the outgoing group number configured in **Section 5.5**.

Retain default values for all other fields and click on **OK** (not shown) to complete the configuration. Configure the same short code on the Expansion system. However, note that the Line Group in the Expansion will depend on the Line number that is configured for SCN to connect to the Primary



5.8. Save Configuration

Navigate to **File** → **Save Configuration** (not shown) in the menu bar at the top of the screen or click on the **Save Icon** as shown below to save the configuration performed in the preceding sections.

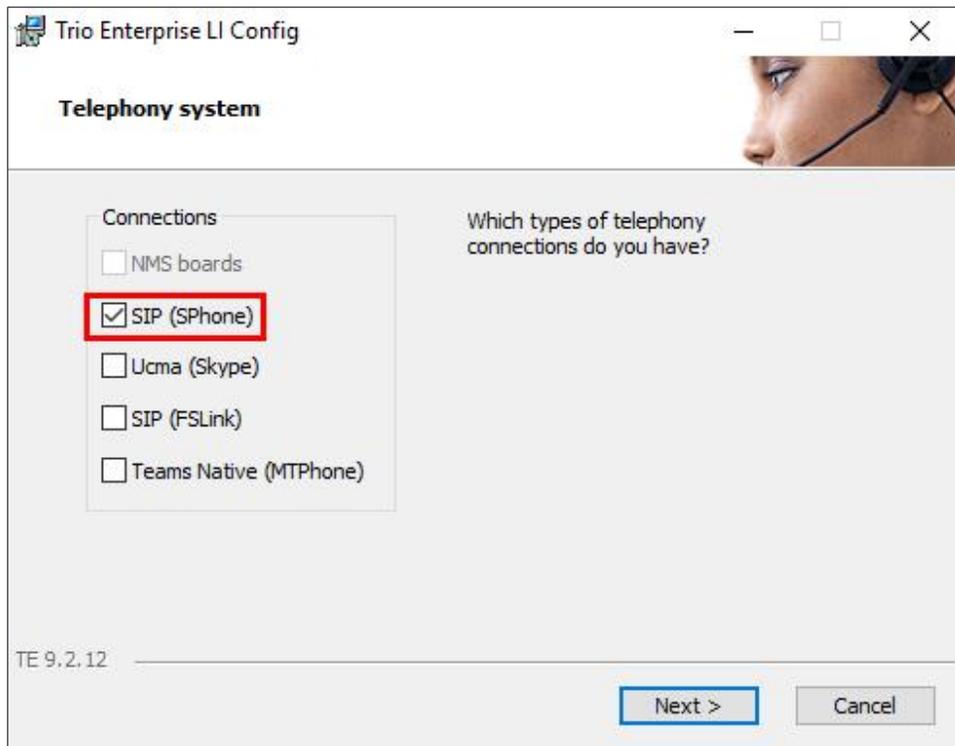


6. Configure Enghouse Trio

Trio Enterprise connects as a SIP Line Trunk to IP Office. This section shows how to configure Trio Enterprise to successfully connect to IP Office using SIP trunk. The installation of the Trio Enterprise software is assumed to be completed and the Trio services are up and running. The steps to configure a SIP Trunk are as follows.

Launch the 'TeleVoice Config' shortcut,  TeleVoice Config

When the configuration window opens, check the **SIP** check box followed by the **Next** button.



In the subsequent window, enter the **License site number** and **Line licence** as supplied directly by Enghouse Interactive AB or the Trio Enterprise reseller. Click on the **Next** button to continue.

Line license

License site number:

Line license:

No Line license key results in demo mode where four channels can be used.

Text-to-Speech license

TTS channel license:

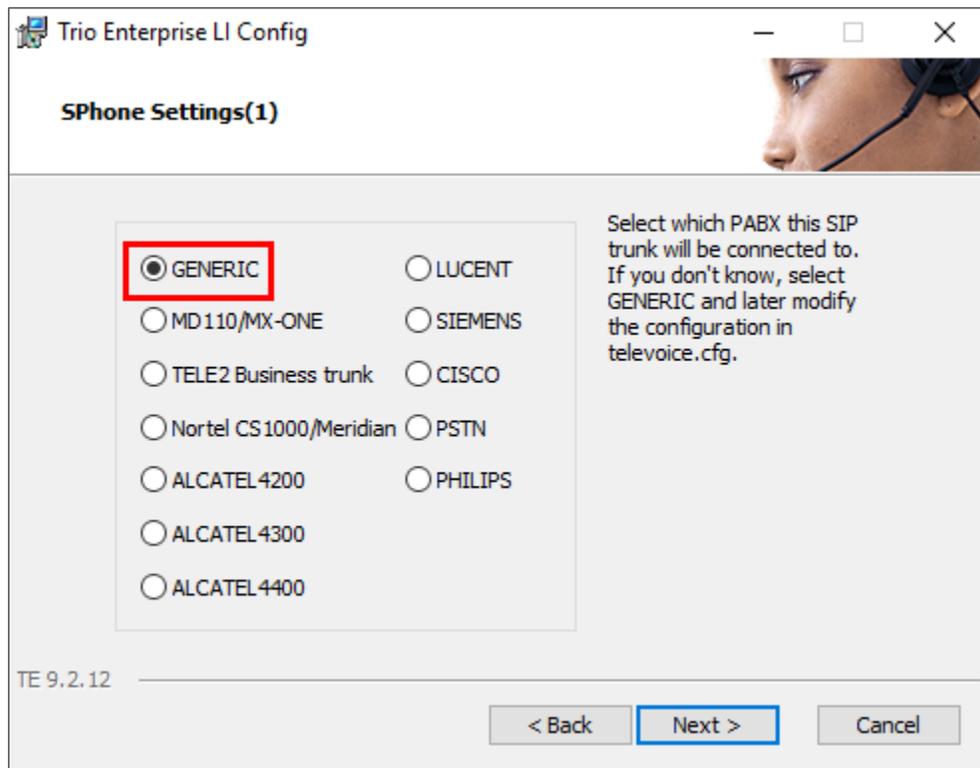
TTS voice license:

No Text-to-Speech licenses results in demo mode where a single channel and a single voice can be used.

TE 9.2.12

< Back Next > Cancel

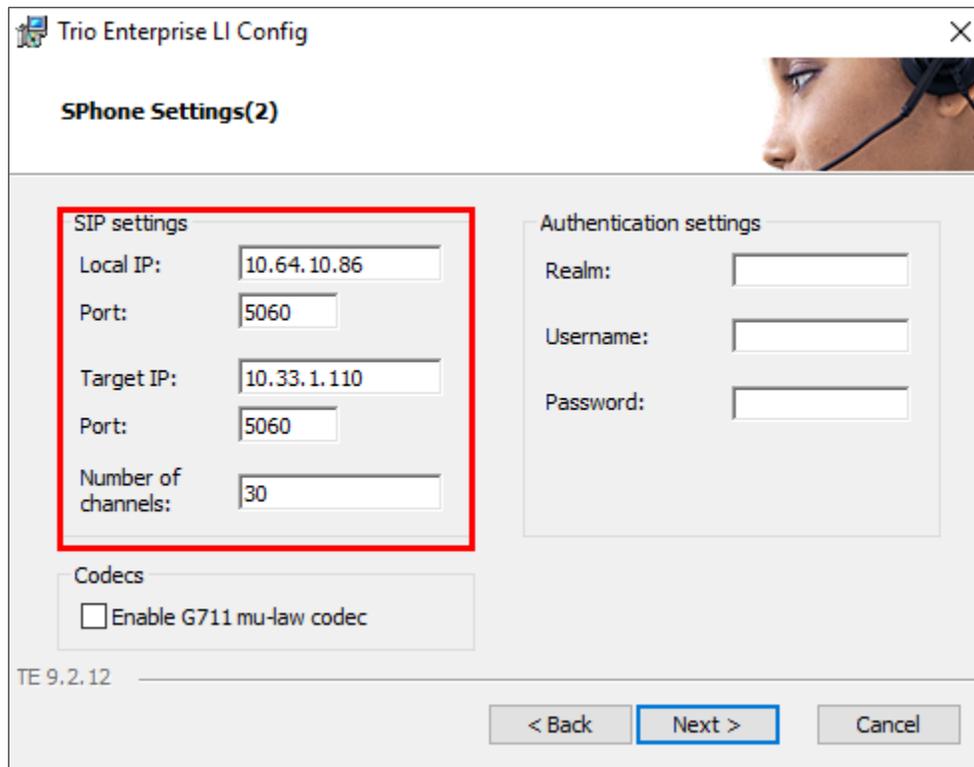
In the subsequent window, click on the **GENERIC** radio button followed by the **Next** button to continue.



In the subsequent window enter the following settings:

- **Local IP:** Enter the local IP address of the Trio Enterprise server.
- **Port:** Enter the SIP Port “5060”.
- **Target IP:** Enter the IP address of the IP Office Primary Server.
- **Port:** Enter the SIP Port “5060”.
- **Number of channels:** Enter “30” as the number of channels.

Click on the **Next** button to continue.



The screenshot shows a window titled "Trio Enterprise LI Config" with a close button (X) in the top right corner. Below the title bar, there is a header "SPhone Settings(2)" and a small image of a person wearing a headset. The main content area is divided into two sections: "SIP settings" and "Authentication settings". The "SIP settings" section is highlighted with a red border and contains the following fields: "Local IP:" with the value "10.64.10.86", "Port:" with the value "5060", "Target IP:" with the value "10.33.1.110", "Port:" with the value "5060", and "Number of channels:" with the value "30". The "Authentication settings" section contains three fields: "Realm:", "Username:", and "Password:", all of which are empty. Below these sections is a "Codecs" section with a checkbox labeled "Enable G711 mu-law codec" which is currently unchecked. At the bottom left of the window, the text "TE 9.2.12" is visible. At the bottom right, there are three buttons: "< Back", "Next >" (which is highlighted with a blue border), and "Cancel".

In the subsequent window enter the following settings:

- **Use LI Address Space:** Click on the radio button.
- **Enable IP routing:** Check the box.
- **UPDATE support:** Check the box.

Click on the **Next** button to continue.

The screenshot shows a configuration window titled "Trio Enterprise LI Config" with a close button in the top right corner. The main heading is "SPhone Settings(3)".

Under the heading "Address Space (AS)", there are three radio button options: "Use LI Address Space" (which is selected), "AS Name:" (with an empty text input field), and "No Address Space".

Under the heading "Sip Options", there is a checked checkbox for "UPDATE support".

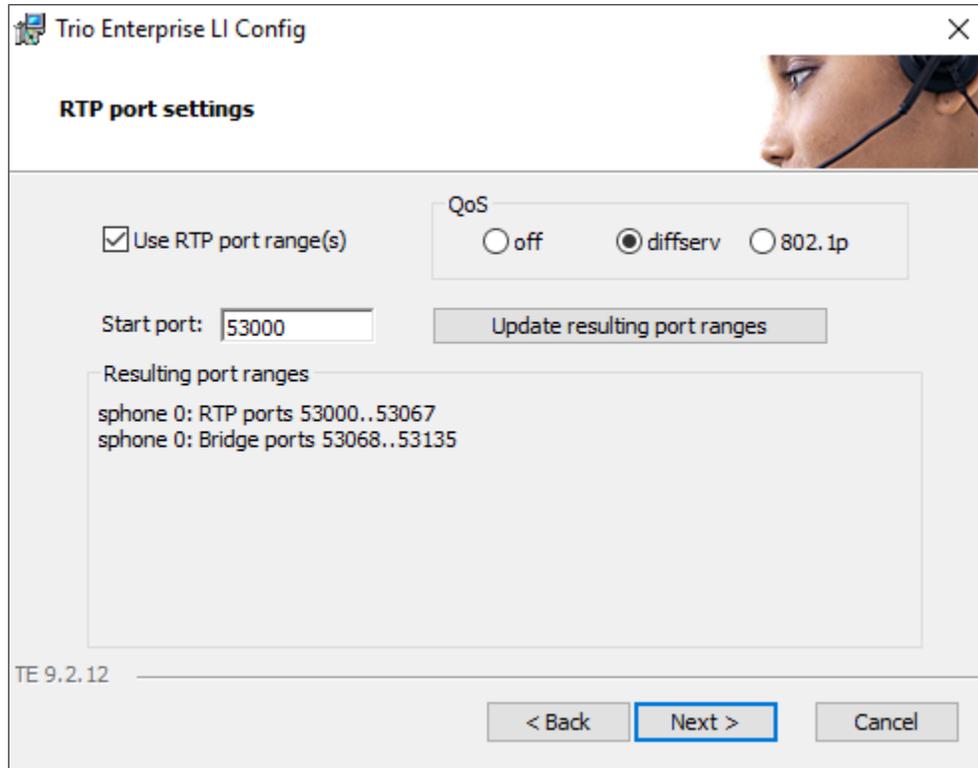
Under the heading "Routing", there is a checked checkbox for "Enable IP routing".

At the bottom left, the version number "TE 9.2.12" is displayed. At the bottom right, there are three buttons: "Additional SIP Trunk", "< Back", and "Next >" (which is highlighted with a blue border), and "Cancel".

In the subsequent window enter the following settings:

- **Use RPT port range(s):** Check the box.
- **diffserv:** Click on the radio button.
- **Start port:** Enter “53000”.

Click on the **Next** button to continue.



The screenshot shows a window titled "Trio Enterprise LI Config" with a close button in the top right corner. The window content is titled "RTP port settings" and features a background image of a woman wearing a headset. The settings are as follows:

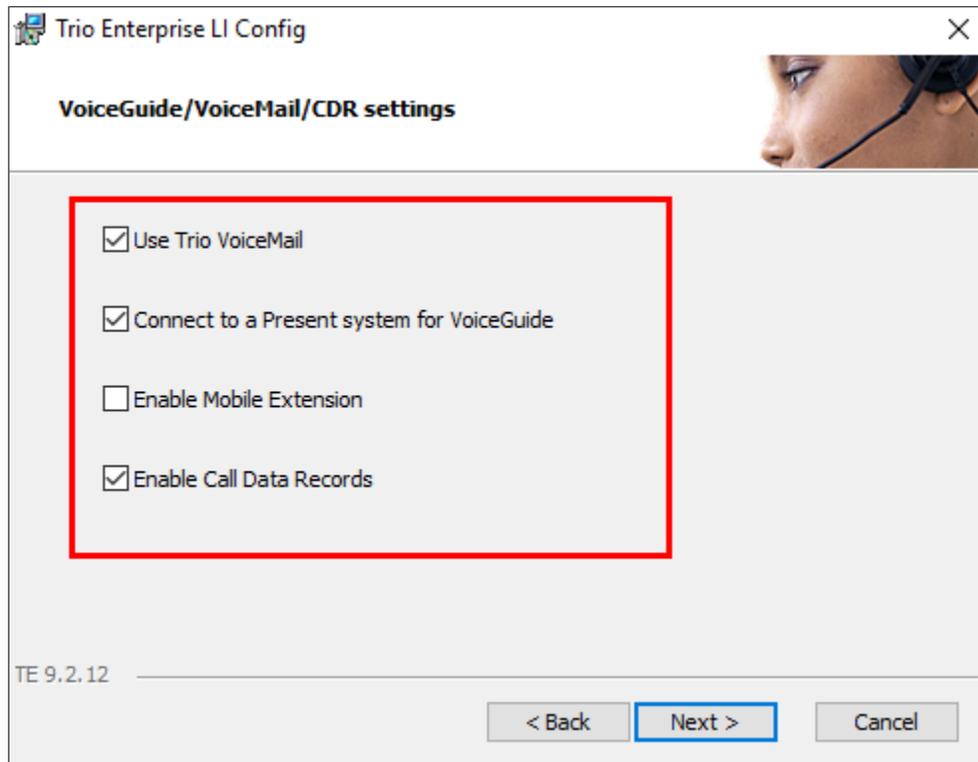
- Use RTP port range(s)
- QoS: off, diffserv, 802.1p
- Start port:
- Resulting port ranges:
 - sphone 0: RTP ports 53000..53067
 - sphone 0: Bridge ports 53068..53135

At the bottom left, the version "TE 9.2.12" is displayed. At the bottom right, there are three buttons: "< Back", "Next >" (highlighted with a blue border), and "Cancel".

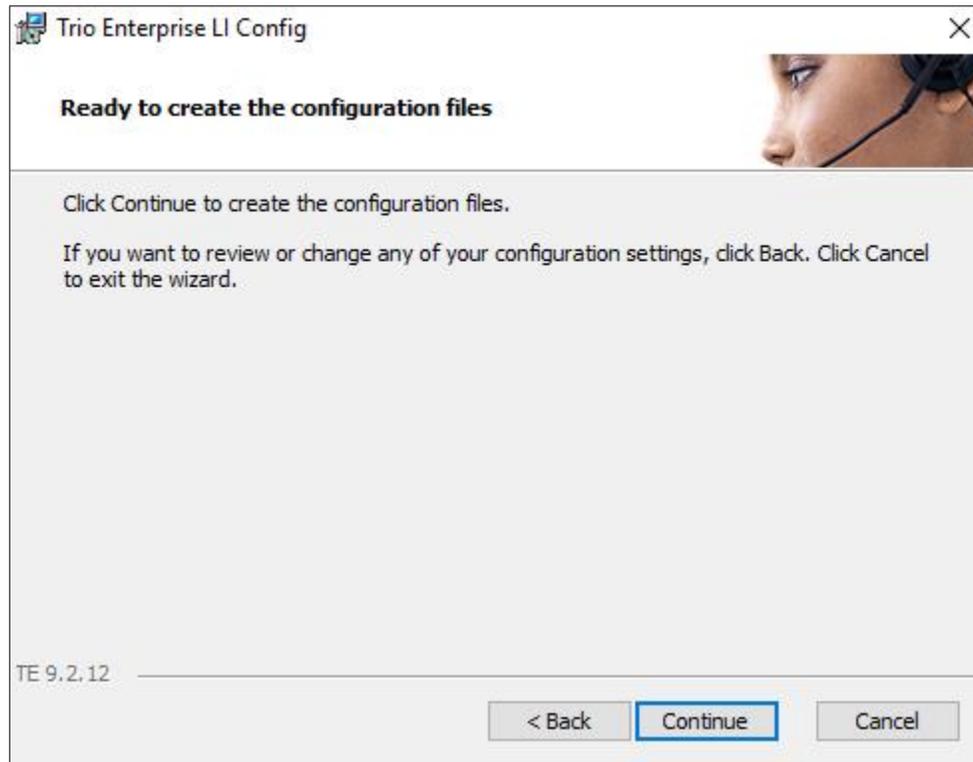
In the subsequent window enter the following settings:

- **Use Trio VoiceMail:** Check the box.
- **Connect to a Present system for VoiceGuide:** Check the box.
- **Enable Call Data Records:** Check the box.

Click on the **Next** button to continue.



In the subsequent window shown below, click on **Continue** button to complete the configuration for Trio.



7. Verification Steps

To verify that IP Office is connected to Trio via SIP trunk, in the PC hosting the IP Office Manager application, navigate to **Start → All apps → IP Office → System Status** (not shown). The **System Status** screen of the Primary Server is seen as shown below.

From the left menu, navigate to **Trunks → Line 5**. This is the SIP Line configured in **Section 5.5** for IP Office to connect to Trio. Ensure that the **Line Service State** is **In Service** and the **Current State** column for the channels read as **Idle** or **Connected** for an active call.

Avaya IP Office System Status - IPOSE110 (10.33.1.110) - IP Office Linux PC 11.1.2.4.0 build 18

AVAYA IP Office System Status

Help Snapshot LogOff Exit About

- System
- Alarms (10)
- Extensions (1)
- Trunks (7)
 - Line: 1
 - Line: 3
 - Line: 4
 - ▶ Line: 5
 - Line: 7
 - Line: 8
 - Line: 9
- Active Calls
- Resources
- Voicemail
- IP Networking
- Locations

SIP Trunk Summary

Line Service State: In Service
 Peer Domain Name: 10.64.10.86
 Resolved Address: 10.64.10.86
 Line Number: 5
 Number of Administered Channels: 0
 Number of Channels in Use: 0
 Administered Compression: G711 Mu
 Silence Suppression: Off
 Media Stream: RTP
 Layer 4 Protocol: TCP
 SIP Trunk Channel Licenses: 100
 SIP Trunk Channel Licenses in Use: 0 0%
 SIP Device Features:

Channel Number	URI G...	Call Ref	Current State	Time in State	Remote Media Address	Codec	Connection Type	Caller ID or Dialed Digits	Other Party on Call	Direction of Call
1			Idle	18 days 2...						
2			Idle	18 days 2...						
3			Idle	18 days 2...						
4			Idle	18 days 2...						
5			Idle	18 days 2...						
6			Idle	18 days 2...						
7			Idle	18 days 2...						
8			Idle	18 days 2...						
9			Idle	18 days 2...						
10			Idle	18 days 2...						

Trio calls the Avaya IP Office station to make it as the attendant and ready to receive incoming call. PSTN user places a call to Trio. Trio bridges the call to the attendant and verify the call is established between the Trio attendant and PSTN user.

8. Conclusion

These Application Notes describe the configuration steps required for Enghouse Interactive AB Trio Version 9 to successfully interoperate with Avaya IP Office Server Edition Release 11.1. All feature and serviceability test cases were completed with observations noted in **Section** Error! Reference source not found..

9. Additional References

This section references the product documentation relevant to these Application Notes.

Product documentation for Avaya products may be found at <http://support.avaya.com>.

- [1] Deploying IP Office™ Platform Server Edition Solution, Release 11.1.2, May 2022.
- [2] Deploying IP Office Essential Edition (IP500 V2), Release 11.1.2, April 2022.
- [3] Administering Avaya IP Office™ Platform with Manager, Release 11.1.2, April 2022

Product Documentation for Enghouse Interactive AB can be obtained in the installed software or at: <http://enghouseinteractive.com>.

©2023 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.