



## **Avaya Solution & Interoperability Test Lab**

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### **Application Notes for Enghouse Interactive AB Trio Enterprise with Avaya IP Office Server Edition - Issue 1.0**

#### **Abstract**

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

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 outline the steps necessary to configure Trio Enterprise from Enghouse Interactive AB (Trio Enterprise) to interoperate with Avaya IP Office Server Edition (IP Office). Trio Enterprise is a client/server-based application running on Microsoft Windows Server operating systems. Trio Enterprise 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 Enterprise 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 Enterprise connects to the IP Office using a SIP trunk. Trio Enterprise is supplied with all prerequisite software including the relevant version of Avaya TAPI. Trio Enterprise uses TAPI client to set forwarding (diversion) which is used for activating and deactivating absence. This is accomplished by Trio sending TAPI requests to IP Office to forward calls of an absent user to Trio. When a user is absent, direct calls to the extension is forwarded to Trio Enterprise via SIP trunk and handled by Trio based on the absence reason set by the user.

## 2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise voice network using IP Office. The Trio Enterprise 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 Enterprise. Calls placed to the Trio Enterprise server automatically places a call to the telephone the Attendant is using for answering purposes. When the attendant answers the call, the Trio Enterprise server bridges the two calls. When the attendant extends the call to another telephone, Trio Enterprise server performs a SIP path replacement, and the caller and the called user are now directly connected. Trio Enterprise uses TAPI client to send request to IP Office to forward a user's extension. This request can be done from the user or from the Attendant. Calls to the Trio Enterprise system may be placed in a queue, until an attendant is available.

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 Enterprise 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 Enterprise 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
- Loop detection for busy and unanswered extensions
- Call Forwarding (Absence) detection

## **2.2. Test Results**

Tests were performed to ensure interoperability between Trio Enterprise and IP Office. The tests were completed with the following observation:

- Trio Enterprise uses TAPI client to set diversion which is used for activating and deactivating call forwarding (absence) feature of Trio Enterprise. In an IP Office environment, which includes the Server Edition and Expansion, TAPI client needs to be installed on every Trio Enterprise server that wants to set call forwarding on the phones that are configured on the Server Edition and Expansion.

## **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](mailto:triosupport@enghouse.com)

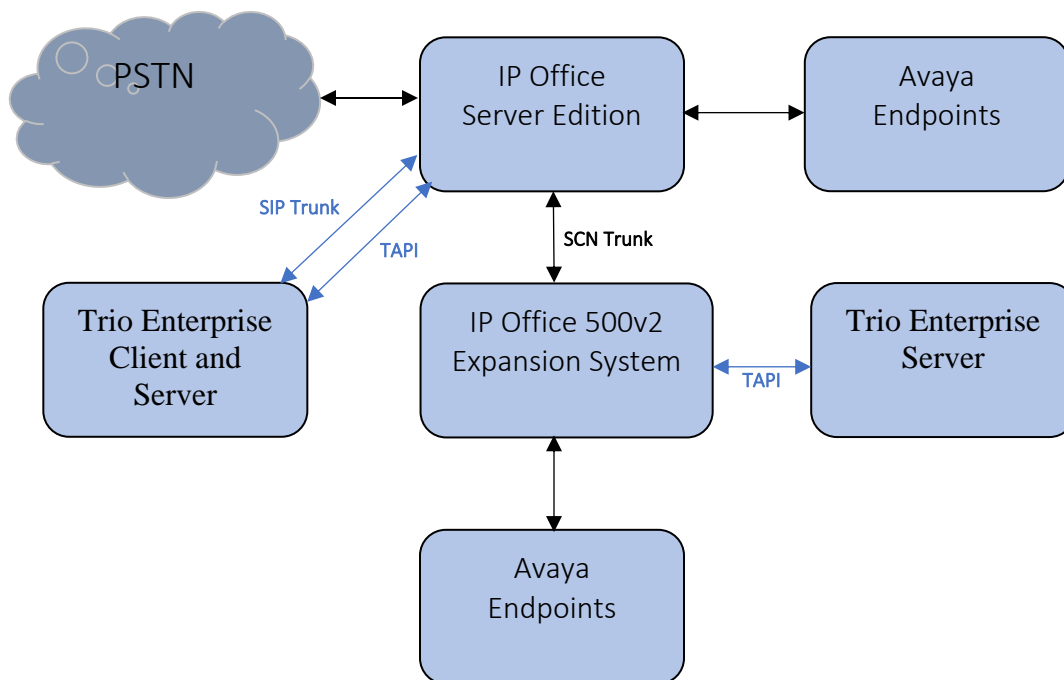
### 3. Reference Configuration

**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 IP500V2 Expansion, has a SIP trunk connection from Primary server to the Trio Enterprise server.

TAPI clients are configured on each Trio Enterprise server connected to each IP Office as shown in test configuration below. This enables the Trio Enterprise to send TAPI requests via the IP Office to set call forwarding to activate or deactivate absence on Primary and Expansion.

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

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



**Figure 1: Avaya and Trio Enterprise 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 Linux Server	11.0.4.0.0 Build 74
Avaya IP Office 500V2 Server	11.0.4.0.0 Build 74
Avaya Endpoints: 9641GS (H.323) J179 (SIP) K175 Vantage Phone (SIP) H175	6.8002 4.0.0.0.21 2.0.1.0 1.0.2.3
Trio Enterprise running on Microsoft Windows 2008 R2 Server TAPI3 Installer	8.0  1.0.0.44

***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

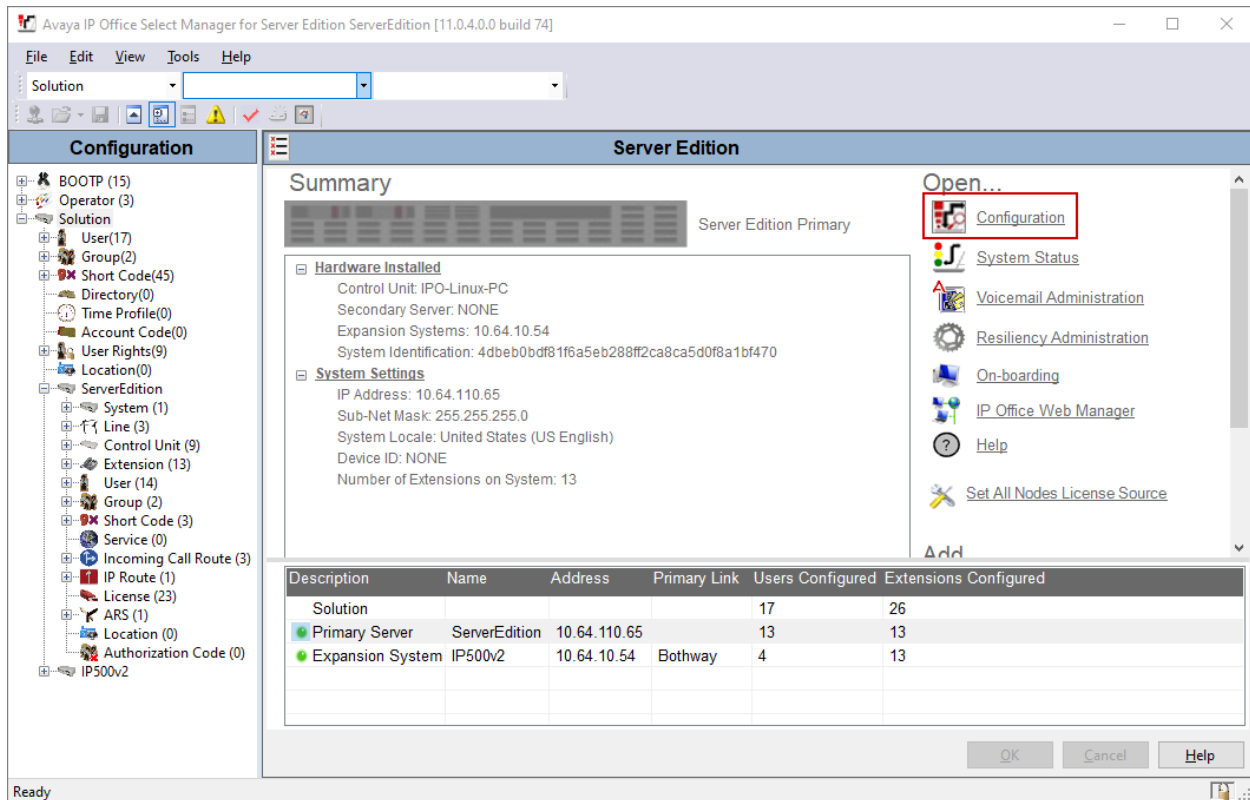
The document assumes that IP Office Server Edition has been installed and configured.

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 9**. 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
- Enable TAPI

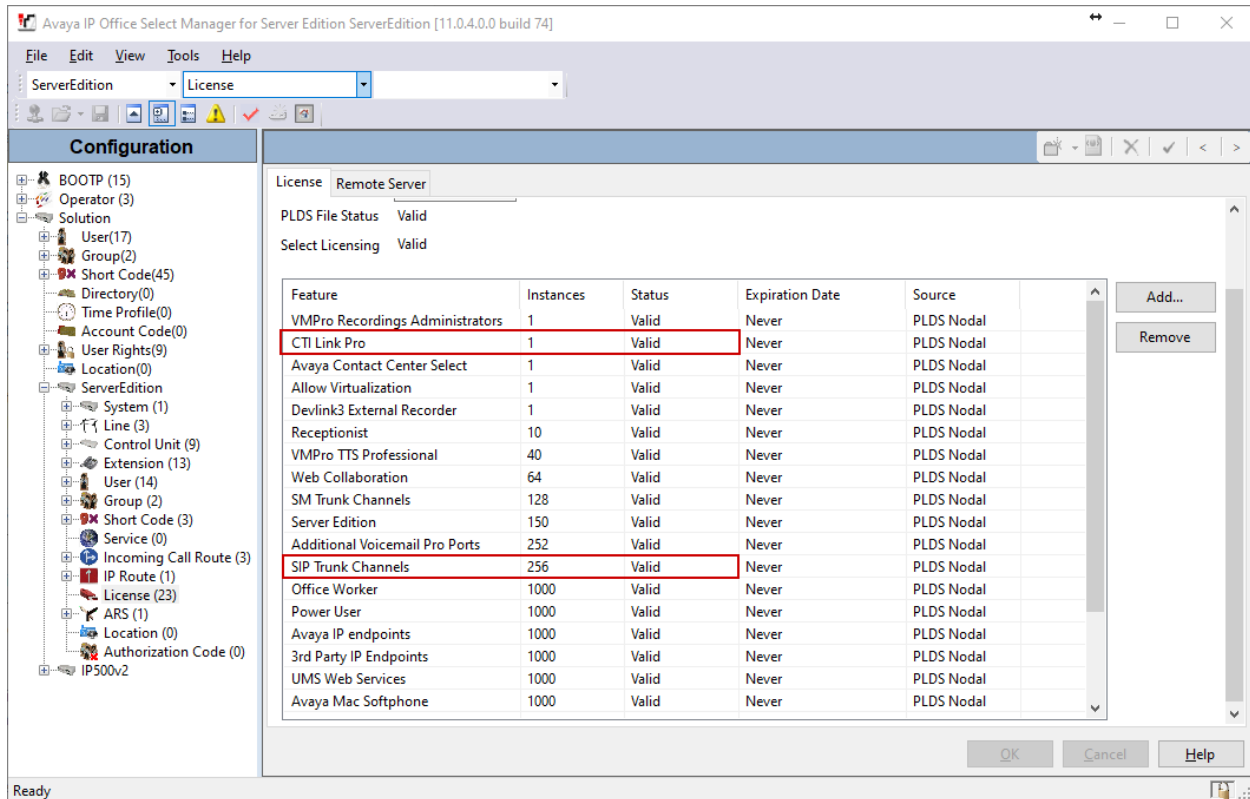
## 5.1. Launch Avaya IP Office Manager

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.



## 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 **DevCon IPO Sev1**, 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. Ensure the same for **CTI Link Pro** also.



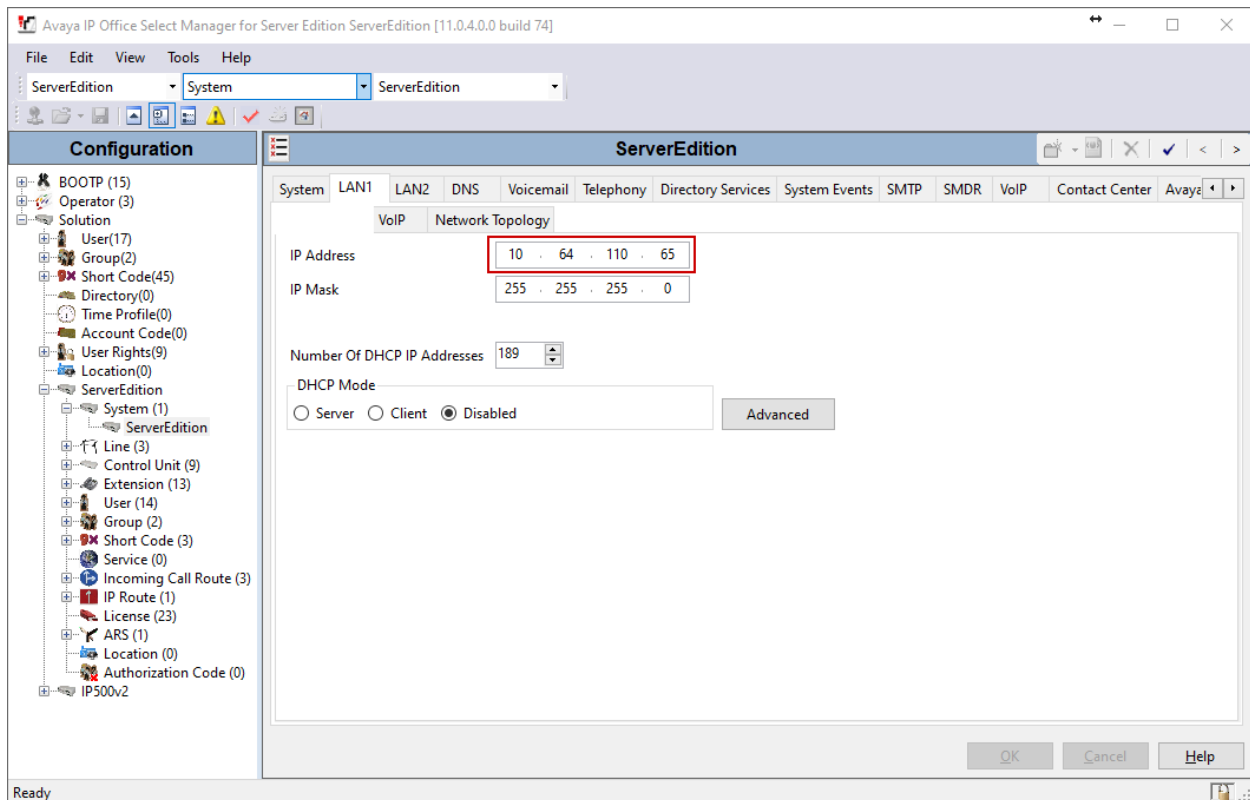
The screenshot shows the Avaya IP Office Select Manager for Server Edition ServerEdition [11.0.4.0.0 build 74] window. The 'License' tab is selected, and the 'Remote Server' sub-tab is active. The 'PLDS File Status' is 'Valid' and 'Select Licensing' is 'Valid'. A table lists various features and their instances. The 'CTI Link Pro' and 'SIP Trunk Channels' rows are highlighted with red boxes.

Feature	Instances	Status	Expiration Date	Source
VMPRO Recordings Administrators	1	Valid	Never	PLDS Nodal
CTI Link Pro	1	Valid	Never	PLDS Nodal
Avaya Contact Center Select	1	Valid	Never	PLDS Nodal
Allow Virtualization	1	Valid	Never	PLDS Nodal
Devlink3 External Recorder	1	Valid	Never	PLDS Nodal
Receptionist	10	Valid	Never	PLDS Nodal
VMPRO TTS Professional	40	Valid	Never	PLDS Nodal
Web Collaboration	64	Valid	Never	PLDS Nodal
SM Trunk Channels	128	Valid	Never	PLDS Nodal
Server Edition	150	Valid	Never	PLDS Nodal
Additional Voicemail Pro Ports	252	Valid	Never	PLDS Nodal
SIP Trunk Channels	256	Valid	Never	PLDS Nodal
Office Worker	1000	Valid	Never	PLDS Nodal
Power User	1000	Valid	Never	PLDS Nodal
Avaya IP endpoints	1000	Valid	Never	PLDS Nodal
3rd Party IP Endpoints	1000	Valid	Never	PLDS Nodal
UMS Web Services	1000	Valid	Never	PLDS Nodal
Avaya Mac Softphone	1000	Valid	Never	PLDS Nodal



### 5.3. Obtain LAN IP Address

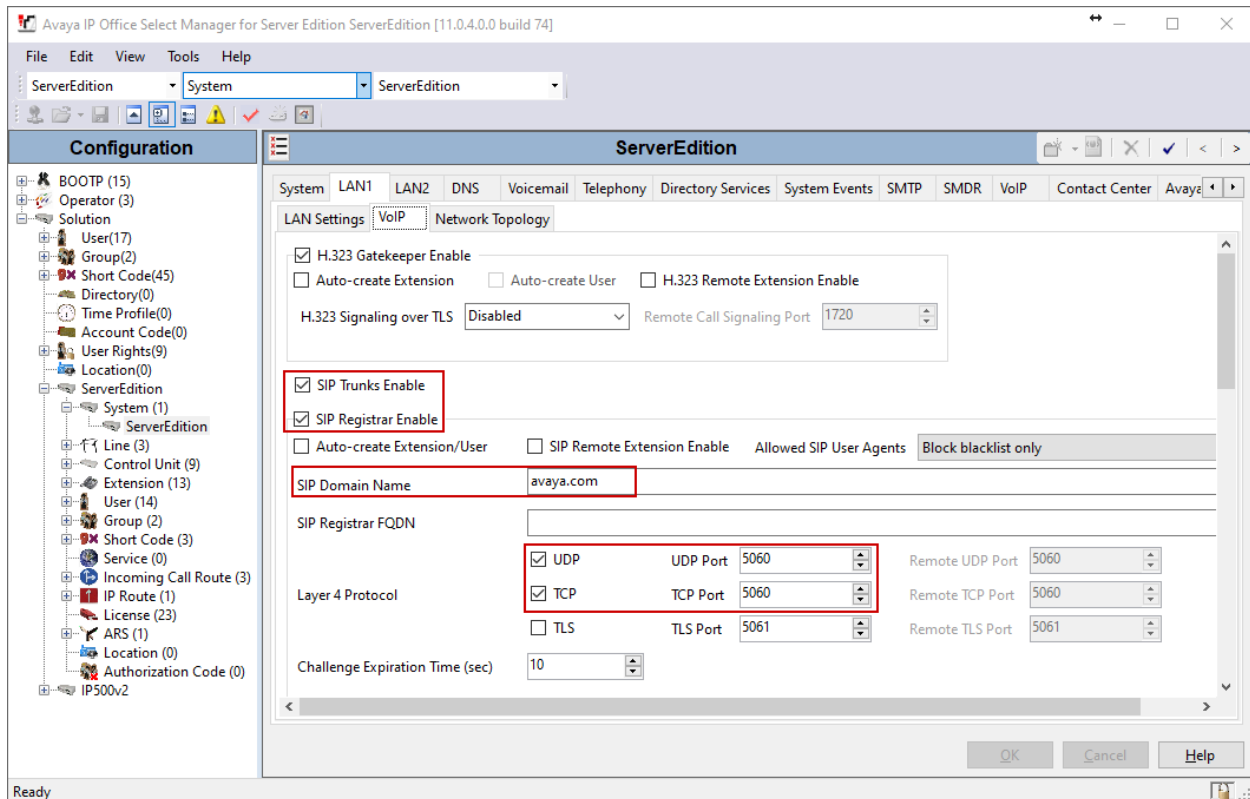
From the configuration tree in the left pane, navigate to **System (1)** to display the IP Office screen in the right pane, where **ServerEdition** is the name of the IP Office Primary system. Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the **IP Address**, which will be used later while configuring Trio Enterprise in **Section 6**. 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 “bvwddev.com” was used.
- Check **UDP** and **TCP** protocol with the correct port numbers. However, during testing only UDP was used.

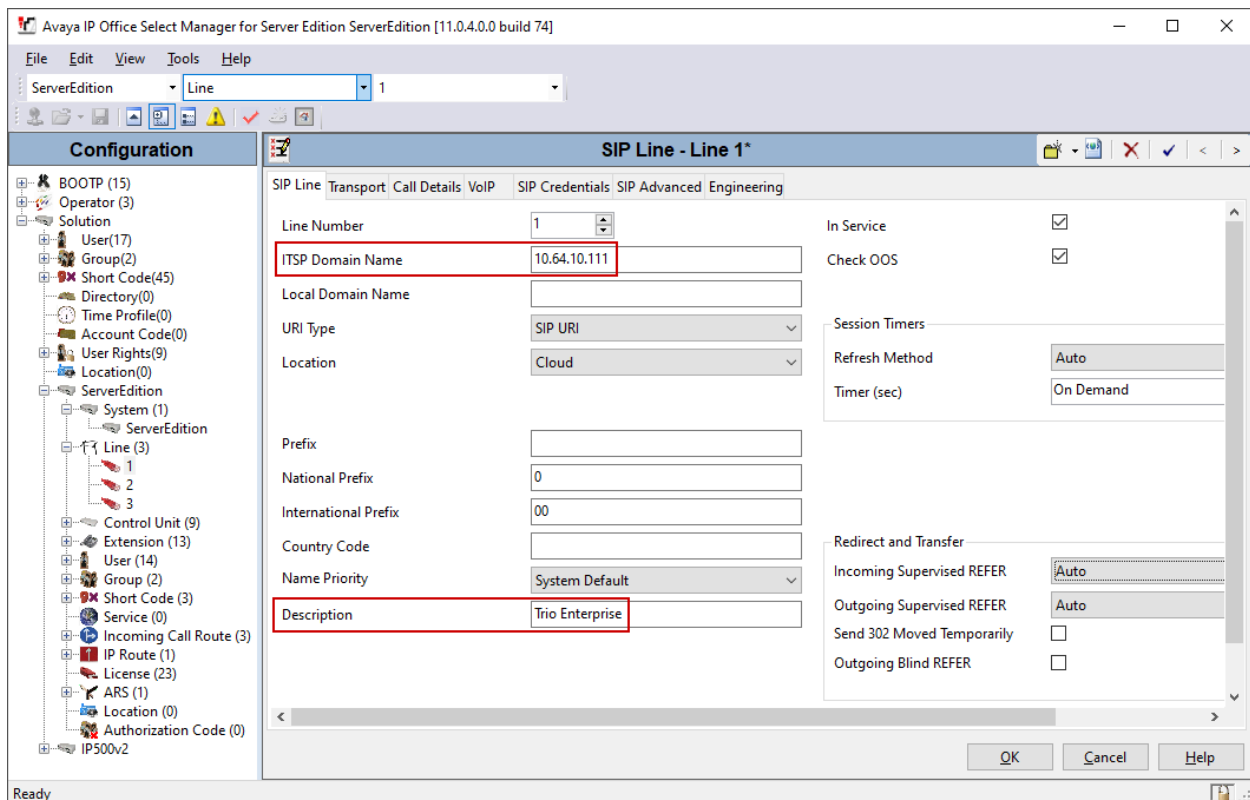


## 5.5. Administer SIP Line

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 1** was added. Select the **SIP Line** tab in the right pane and configure the following:

- **ITSP Domain Name:** IP address of Trio Enterprise Server tevsip01.
- Enter a valid description in the **Description** field (optional).

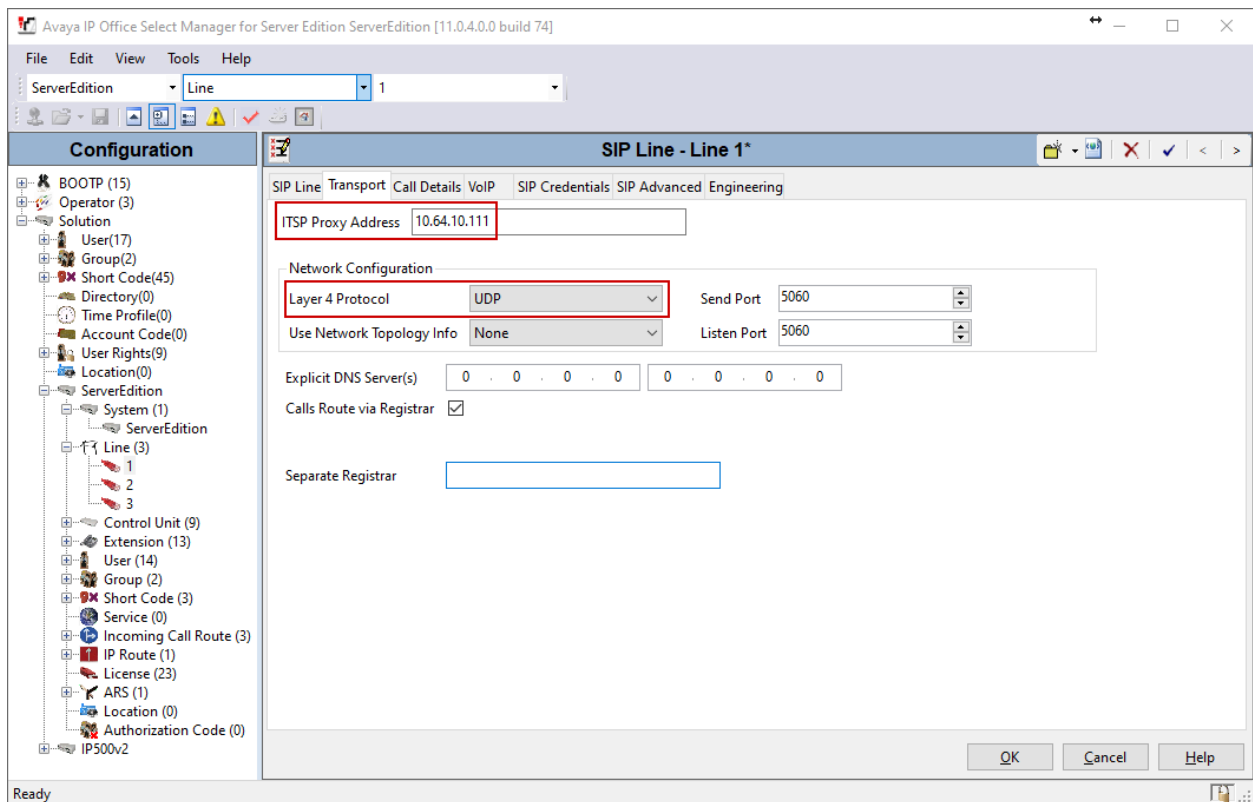
Retain default values for all other fields.



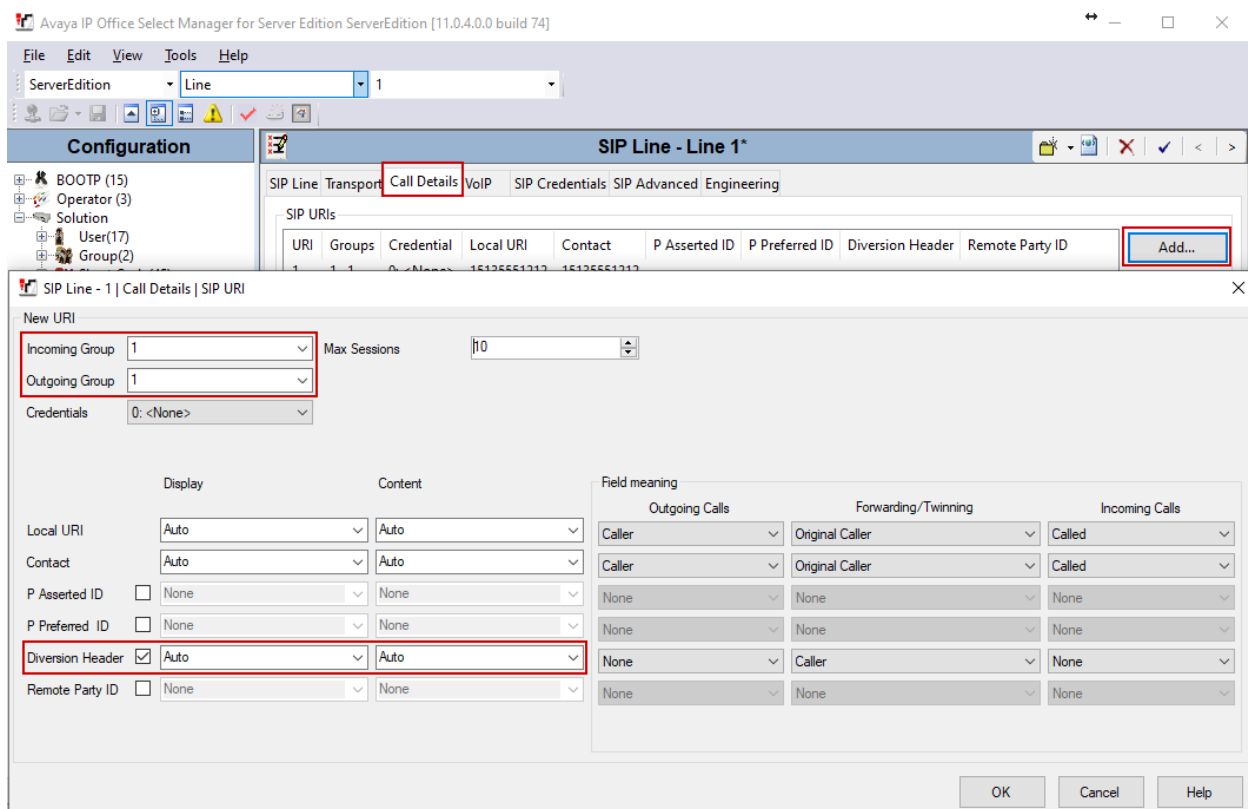
Select the **Transport** tab in the right pane and configure the following:

- **ITSP Proxy Address:** IP address of Trio Enterprise Server tevsipo1.
- Under **Network Configuration** → **Layer 4 protocol**, select “UDP” and its **Send Port** as “5060”.

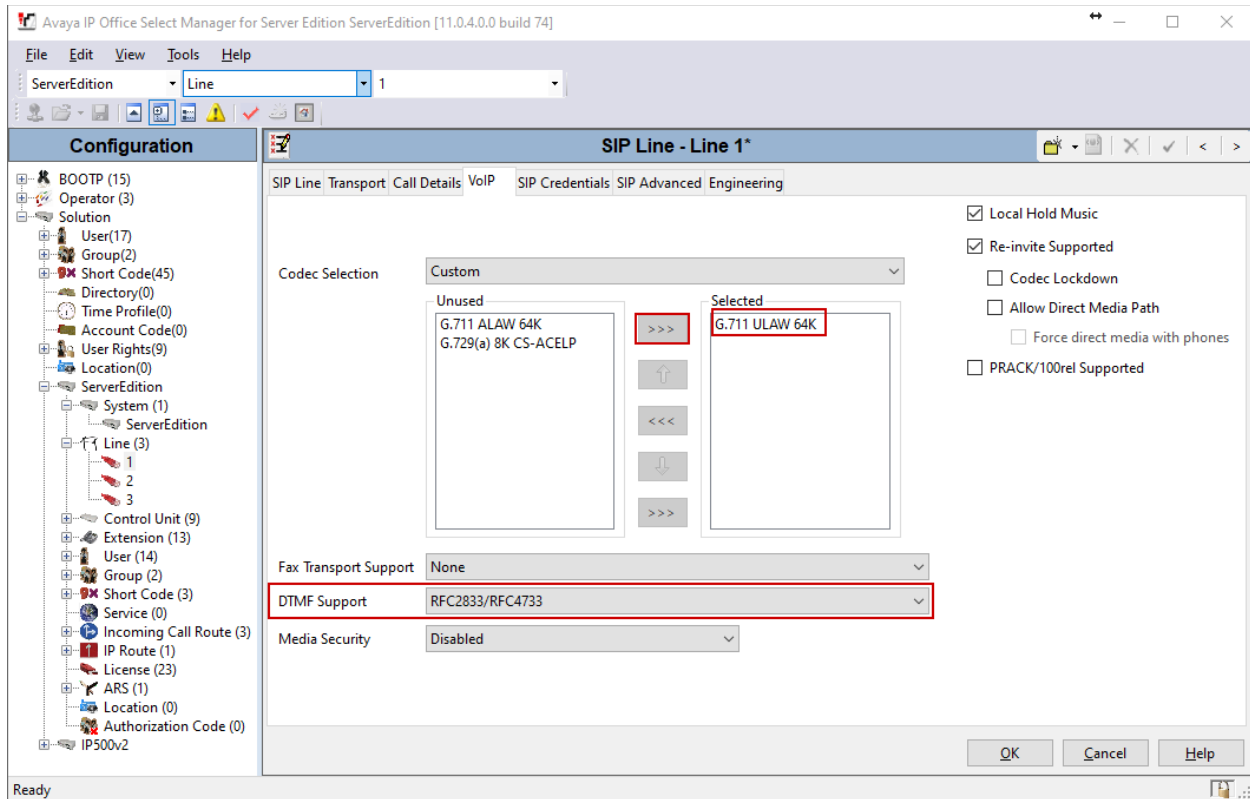
Retain default values for all other fields.



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 “1” 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 **Diversion Header** box and retain the default values in the remaining fields. Click **OK**.

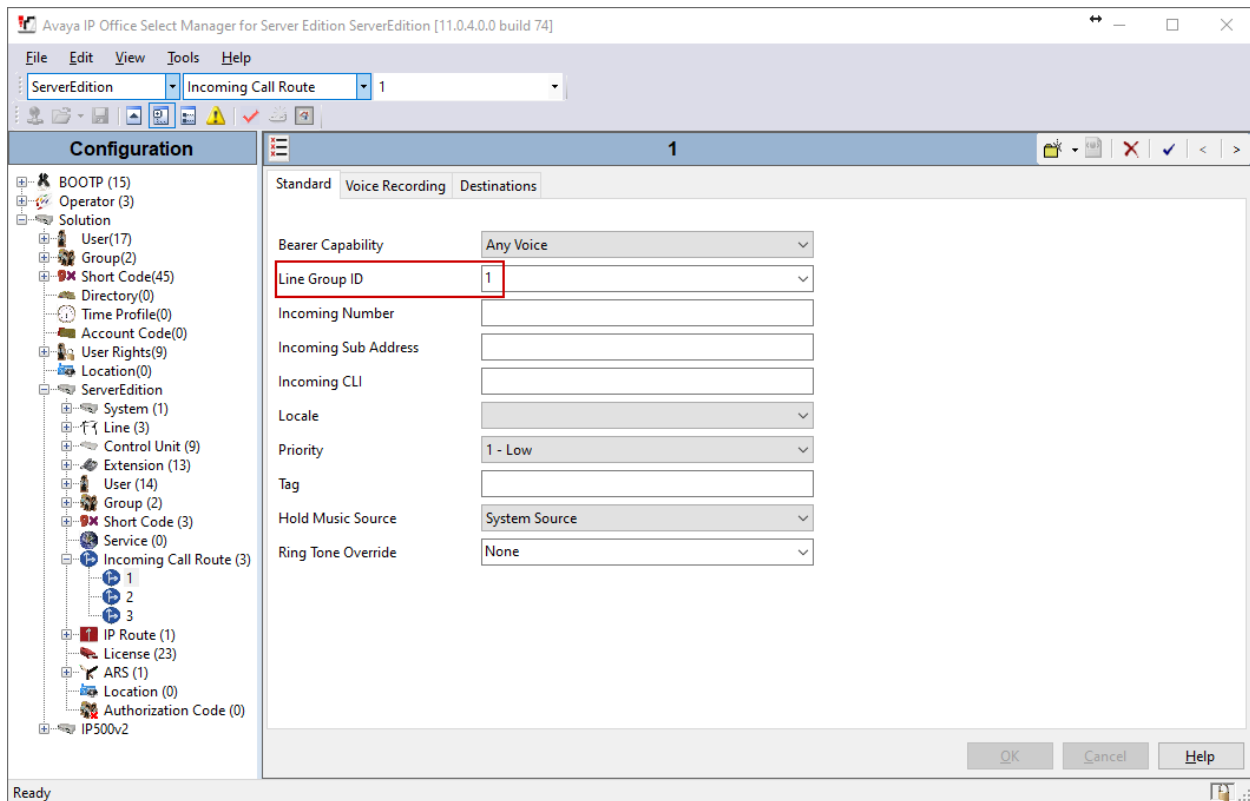


Select the **VoIP** tab. The default **Codec Selection** in the system is shown below and the same was used for compliance testing. In the **VoIP** tab ensure that for **DTMF Support**, **RFC2833/RFC4733** is selected from the drop-down menu. Retain default values for all remaining fields. During compliance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

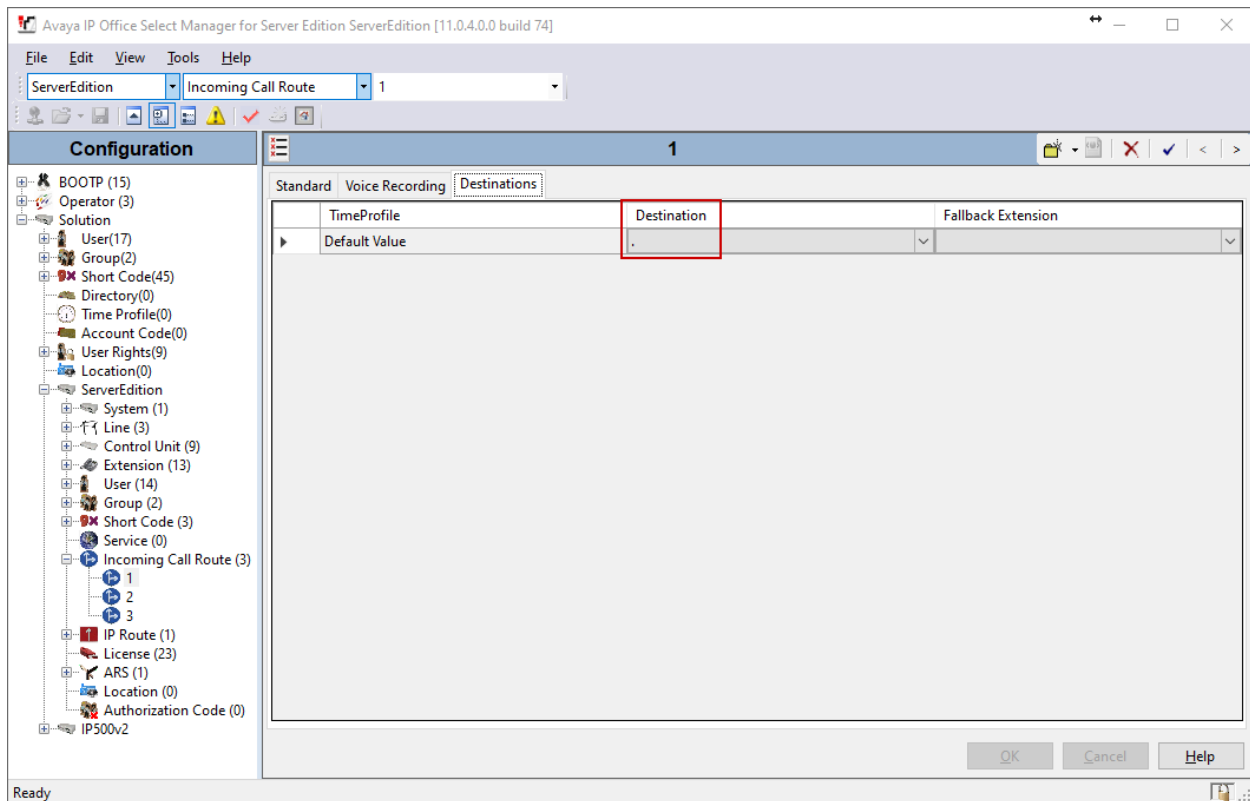


## 5.6. Administer Incoming Call Route

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 “1”. Retain default values for all other fields.



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





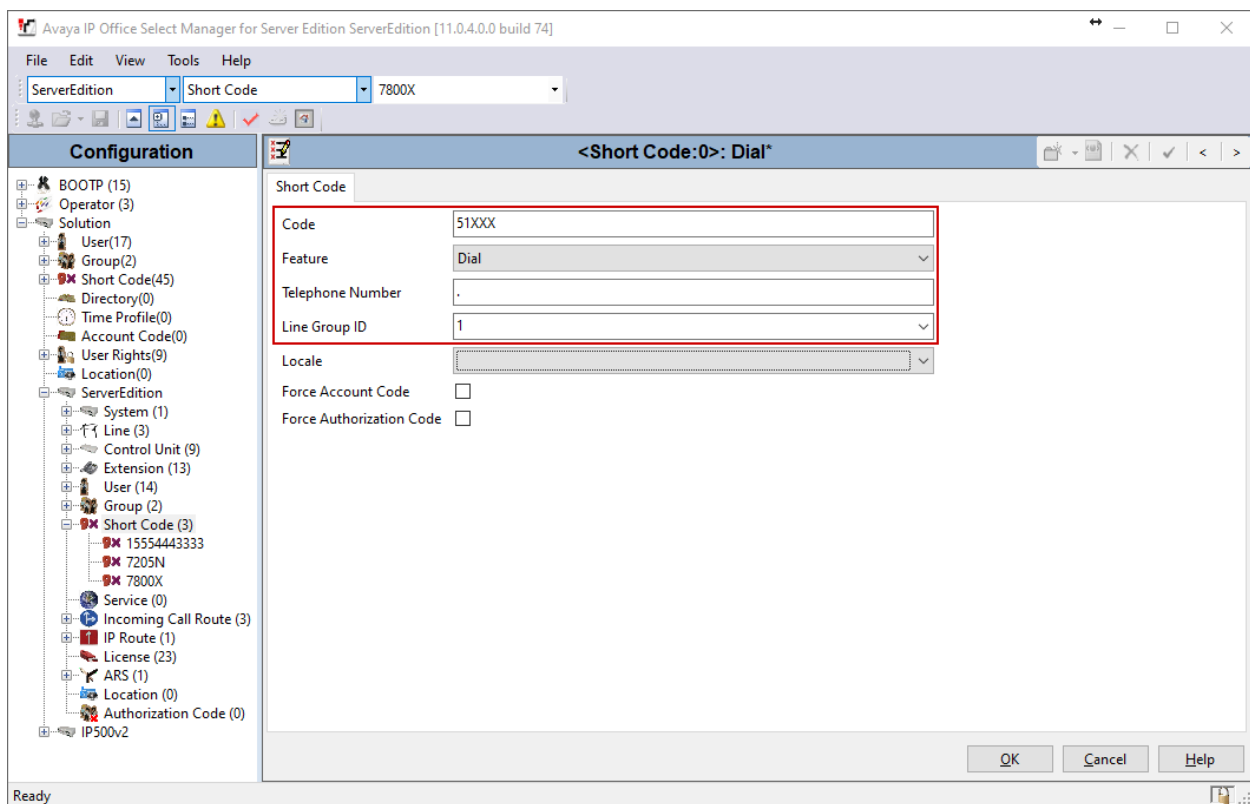
## 5.7. Administer Short Code

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 Enterprise. In the compliance testing, 51XXX dialling plan was used for calls to be routed over the SIP trunks to Trio Enterprise.

Configure the following values:

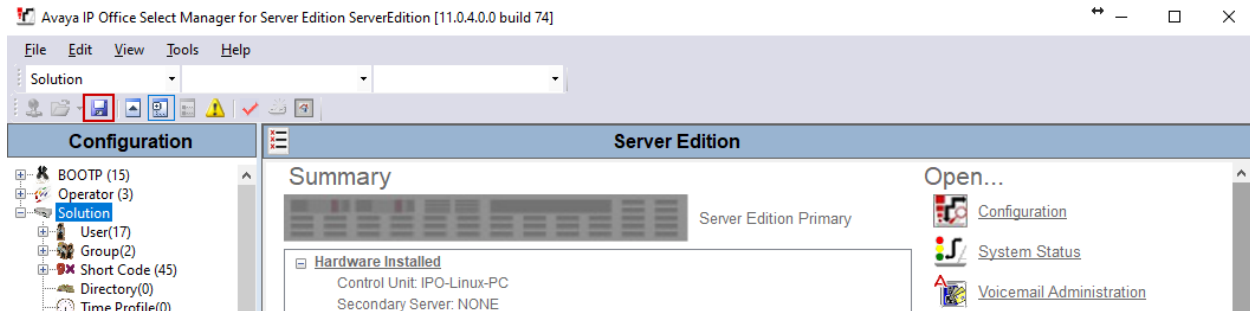
- **Code:** Enter “51XXX”.
- **Feature:** Keep the default value of “Dial”.
- **Telephone Number:** Enter “.”.
- **Line Group ID:** Select “1” which is the outgoing group number configured in **Section 5.5**.

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

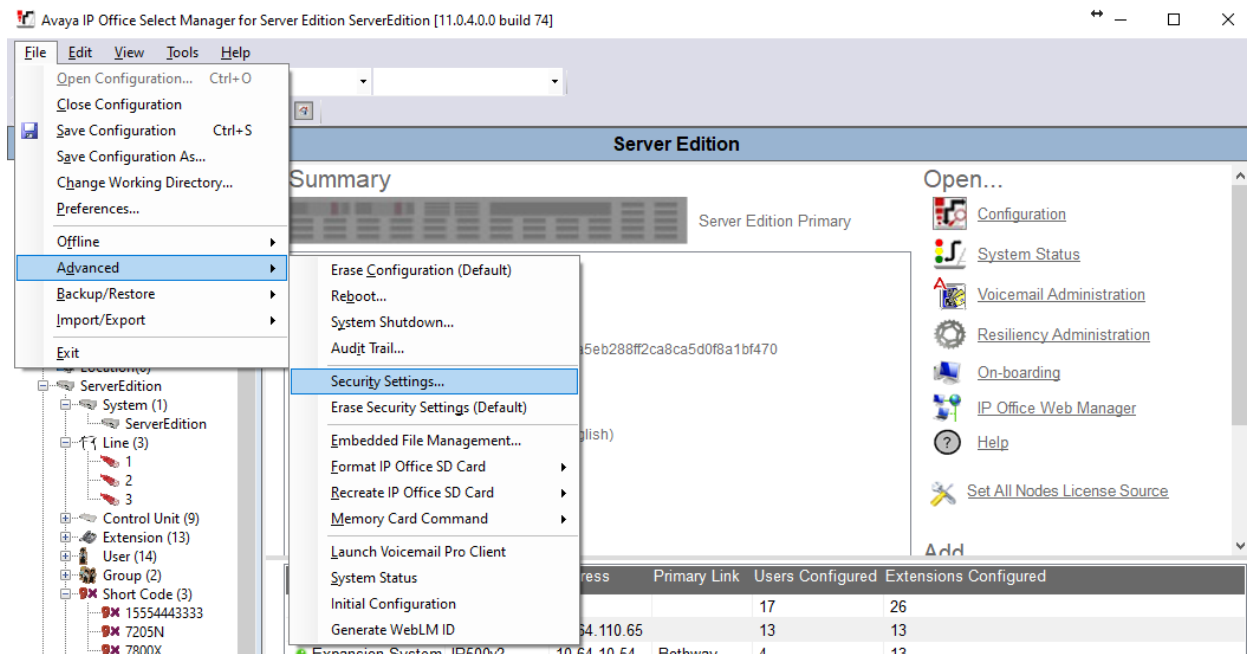
Navigate to **File → Save Configuration** (not shown) in the menu bar at the top of the screen or click on the **Save** Icon as shown below to save the configuration performed in the preceding sections.



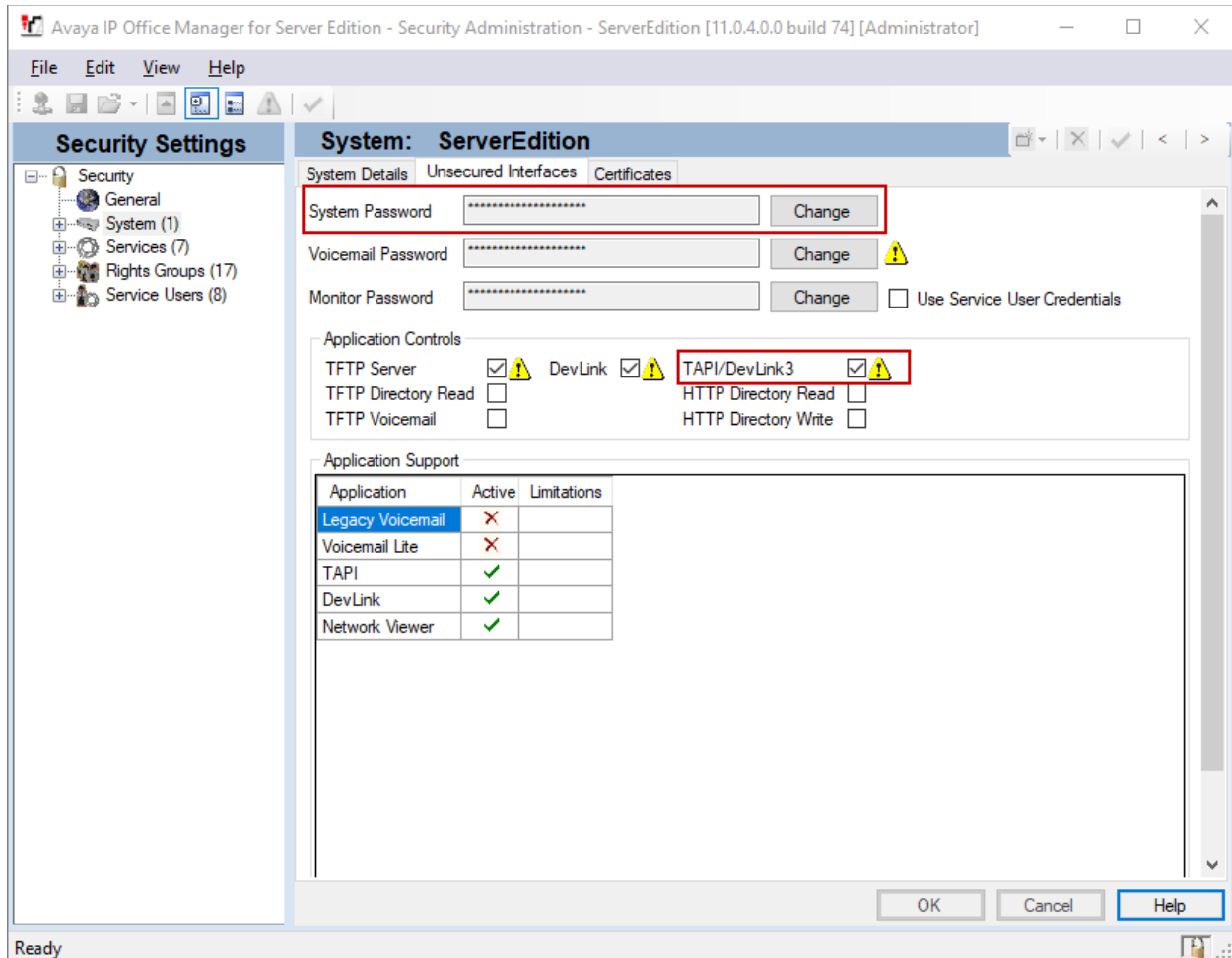
## 5.9. Enable TAPI

For Trio Enterprise to send TAPI request to IP Office, ensure that TAPI feature is enabled on both Primary and Expansion IP Office.

From **Avaya IP Office Manager for Server Edition**, navigate to **File → Advanced → Security Settings** as shown in the screen below.



Select **System** and under the **Unsecured Interfaces** tab ensure that **TAPI/DevLink3** box is selected as shown below. Then click on **OK** (not shown) and save the settings. Repeat the same on IP Office Expansion system too. Also, note that the password configured for **System** is used for TAPI connectivity.



## 6. Configure Trio Enterprise from Enghouse Interactive AB

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.

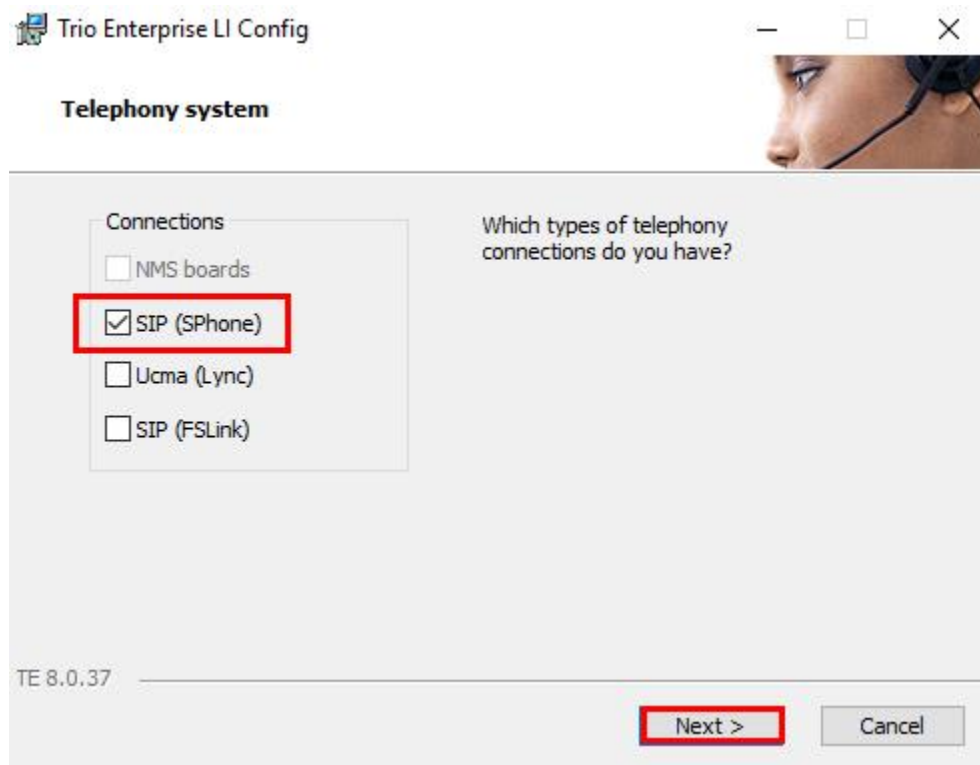
### 6.1. Configure Trio Enterprise to use SIP Trunks

Access Windows services. Select **Start → Run**, then type **services.msc** into the command line and press return (not shown). When the services window opens, locate the **Trio Televoice service**, right click and select **stop** to stop the service (not shown).

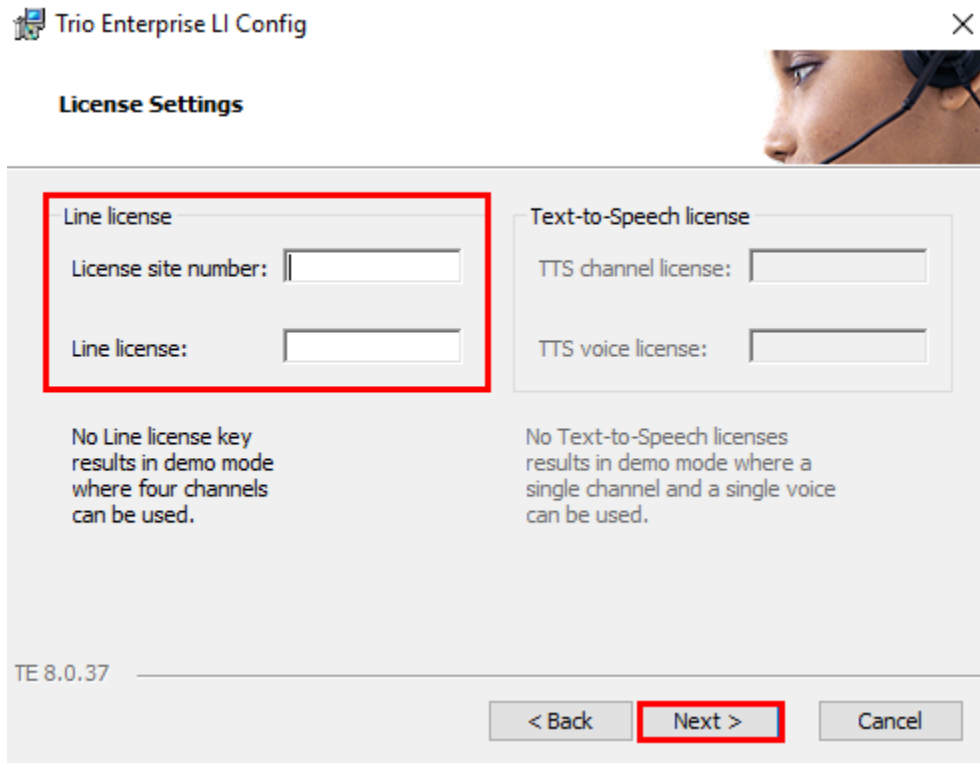
Launch the ‘TeleVoice Config’ shortcut,



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.



The image shows a screenshot of the 'Trio Enterprise LI Config' window. The title bar includes a small icon, the text 'Trio Enterprise LI Config', and a close button (X). Below the title bar is a header area with the text 'License Settings' on the left and a profile picture of a person wearing a headset on the right. The main content area is divided into two columns. The left column is titled 'Line license' and contains two text input fields: 'License site number:' and 'Line license:'. This entire column is enclosed in a red rectangular box. The right column is titled 'Text-to-Speech license' and contains two text input fields: 'TTS channel license:' and 'TTS voice license:'. Below these input fields, there are two paragraphs of text. The left paragraph states: 'No Line license key results in demo mode where four channels can be used.' The right paragraph states: 'No Text-to-Speech licenses results in demo mode where a single channel and a single voice can be used.' At the bottom left of the window, the text 'TE 8.0.37' is displayed. At the bottom right, there are three buttons: '< Back', 'Next >', and 'Cancel'. The 'Next >' button is highlighted with a red rectangular box.

**License Settings**

**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 8.0.37

< Back   **Next >**   Cancel

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

Trio Enterprise LI Config

**SPhone Settings(1)**

Select which PABX this SIP trunk will be connected to. If you don't know, select GENERIC and later modify the configuration in televoice.cfg.

☒ **GENERIC** ☐ LUCENT

☐ MD 110/MX-ONE ☐ SIEMENS

☐ TELE2 Business trunk ☐ CISCO

☐ Nortel CS1000/Meridian ☐ PSTN

☐ ALCATEL 4200 ☐ PHILIPS

☐ ALCATEL 4300

☐ ALCATEL 4400

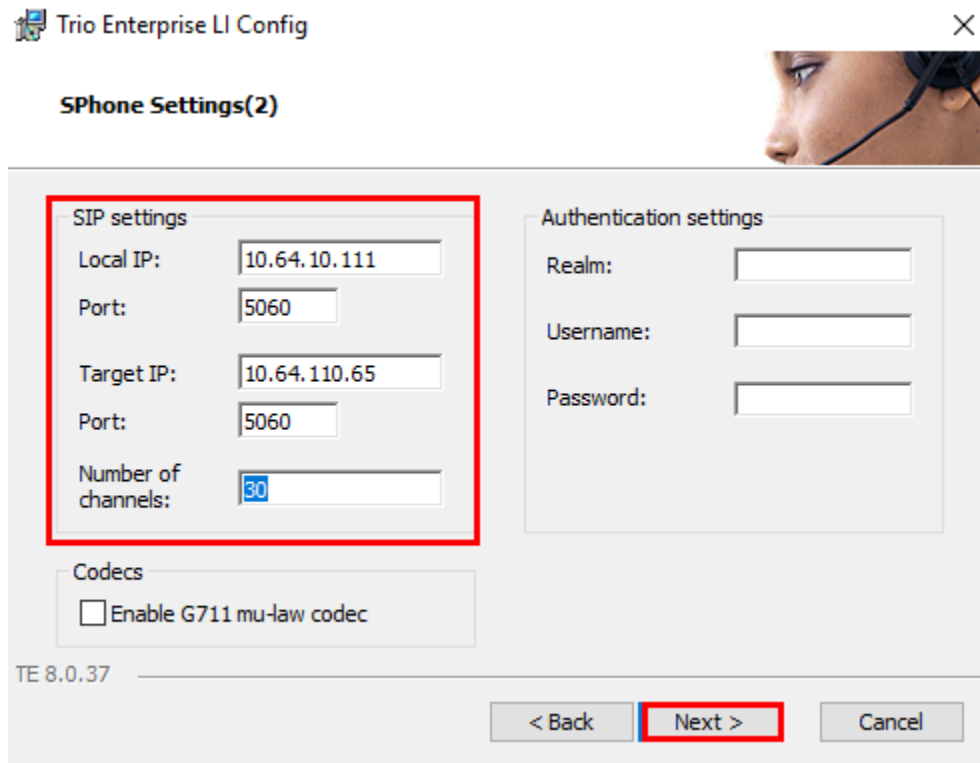
TE 8.0.37

< Back **Next >** Cancel

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 the 'Trio Enterprise LI Config' window with the 'SPhone Settings(2)' tab selected. The 'SIP settings' section is highlighted with a red box. The 'Authentication settings' section is also visible. The 'Codecs' section has an unchecked checkbox for 'Enable G711 mu-law codec'. The 'Next >' button is highlighted with a red box.

SIP settings	
Local IP:	10.64.10.111
Port:	5060
Target IP:	10.64.110.65
Port:	5060
Number of channels:	30

Authentication settings	
Realm:	
Username:	
Password:	

Codecs

☐ Enable G711 mu-law codec

TE 8.0.37

< Back   **Next >**   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.

Trio Enterprise LI Config

**SPhone Settings(3)**

Address Space (AS)

☒ Use LI Address Space

☐ AS Name:

☐ No Address Space

Sip Options

☒ UPDATE support

Routing

☒ Enable IP routing

TE 8.0.37

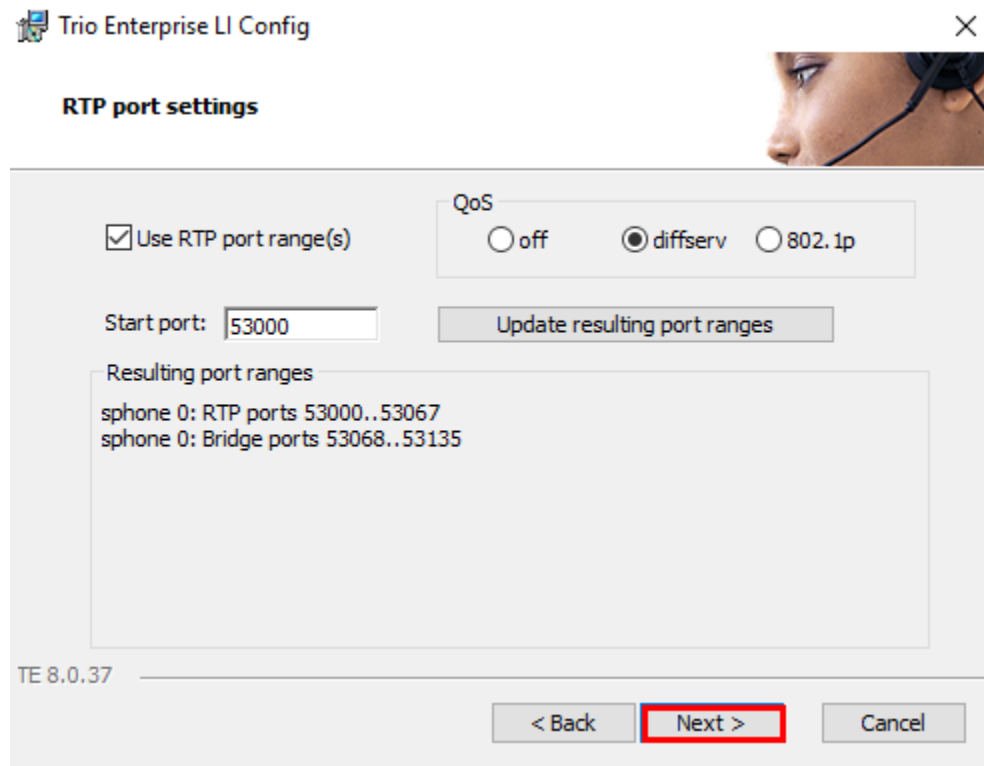
Additional SIP Trunk

< Back **Next >** Cancel

In the subsequent window enter the following settings:

- **Use RPT port range(s):** Check the box.
- **QoS:** Select “diffserv”.
- **Start port:** Enter “53000”.

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



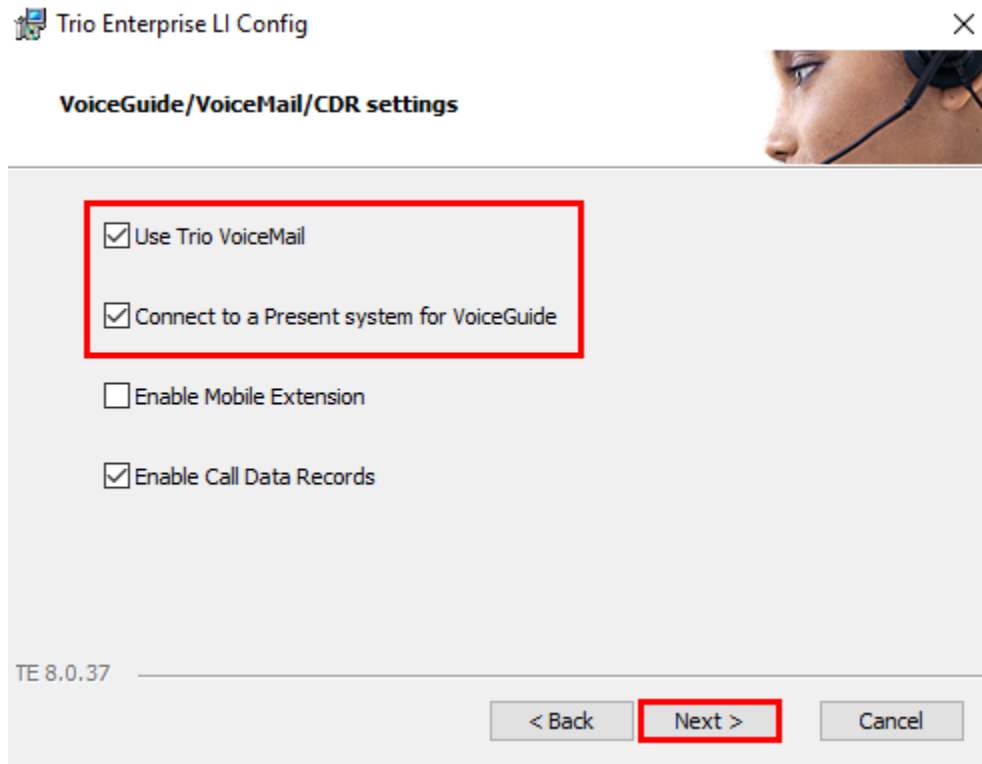
The screenshot shows the 'RTP port settings' window in the Trio Enterprise LI Config application. The window has a title bar with the application name and a close button. The main content area is titled 'RTP port settings' and contains the following elements:

- A checkbox labeled 'Use RTP port range(s)' which is checked.
- A 'QoS' section with three radio buttons: 'off', 'diffserv' (which is selected), and '802.1p'.
- A 'Start port' text box containing the value '53000'.
- An 'Update resulting port ranges' button.
- A 'Resulting port ranges' section with a text area displaying:  
sphone 0: RTP ports 53000..53067  
sphone 0: Bridge ports 53068..53135
- A version indicator 'TE 8.0.37' at the bottom left.
- Navigation buttons at the bottom: '< Back', 'Next >' (highlighted with a red border), and 'Cancel'.

In the subsequent window enter the following settings:

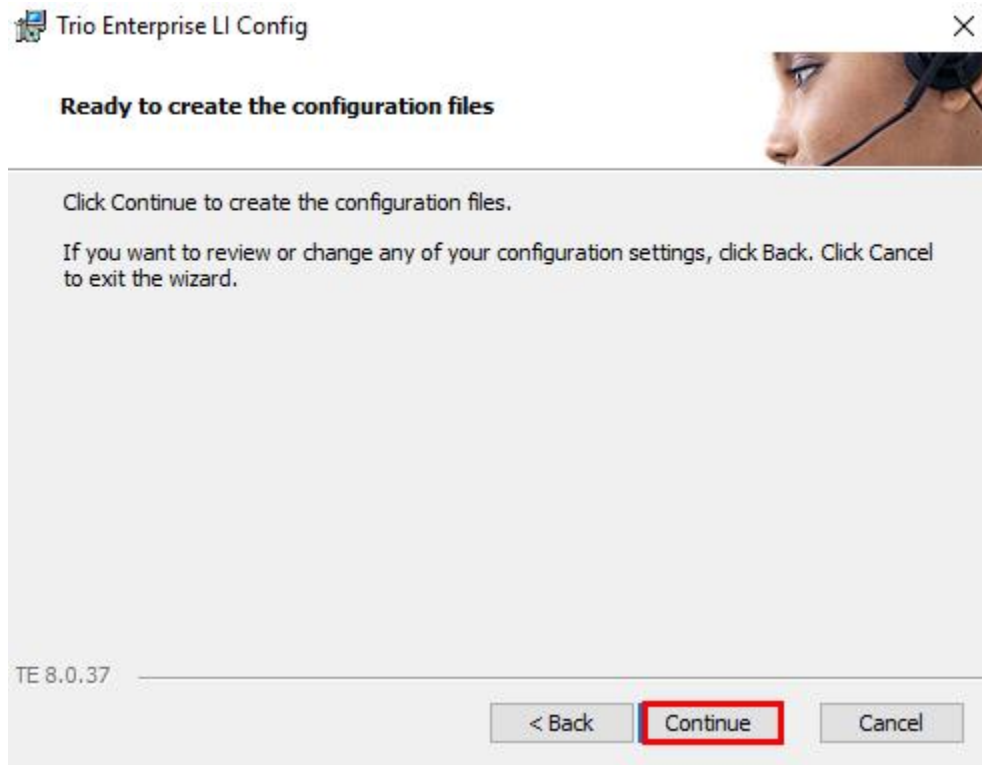
- **Use Trio VoiceMail:** Check the box.
- **Connect to a Present system for VoiceGuide:** Check the box.
- **Enable Call Data Records:** Check the box.

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

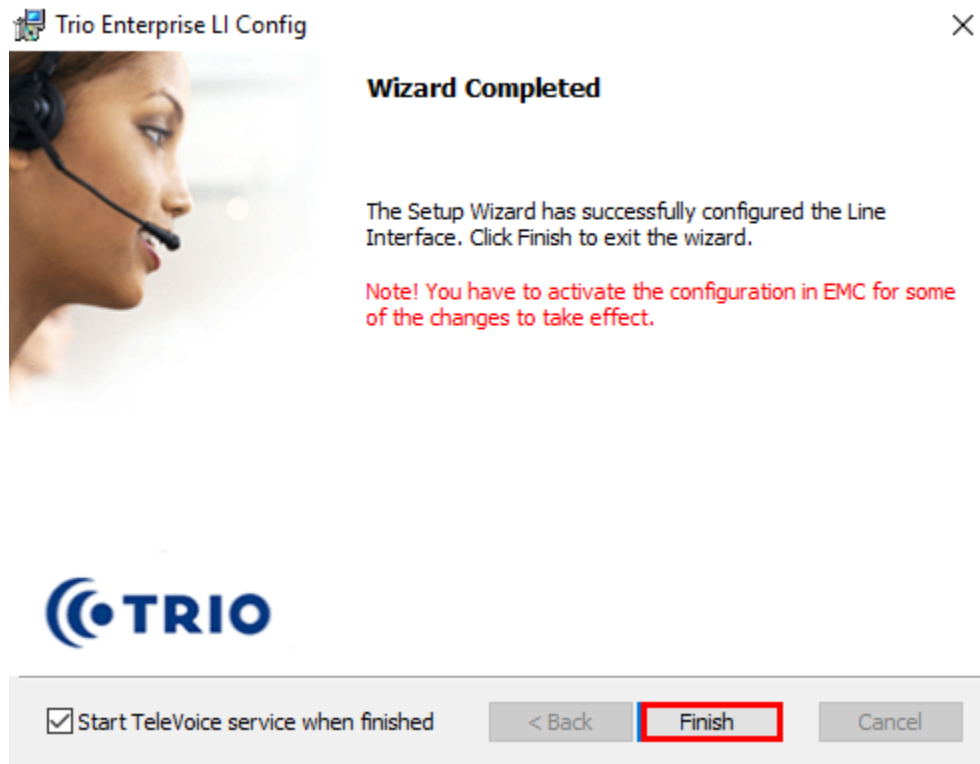


The screenshot shows a window titled "Trio Enterprise LI Config" with a close button (X) in the top right corner. Below the title bar, the text "VoiceGuide/VoiceMail/CDR settings" is displayed. The main area contains four checkboxes: "Use Trio VoiceMail" (checked), "Connect to a Present system for VoiceGuide" (checked), "Enable Mobile Extension" (unchecked), and "Enable Call Data Records" (checked). A red rectangle highlights the first two checked options. At the bottom left, the text "TE 8.0.37" is visible. At the bottom right, there are three buttons: "< Back", "Next >" (highlighted with a red rectangle), and "Cancel".

In the subsequent window shown below, click on **Continue** button.



On the **Wizard Completed** page check the **Start TeleVoice service when finished** check box, followed by the **Finish** button.



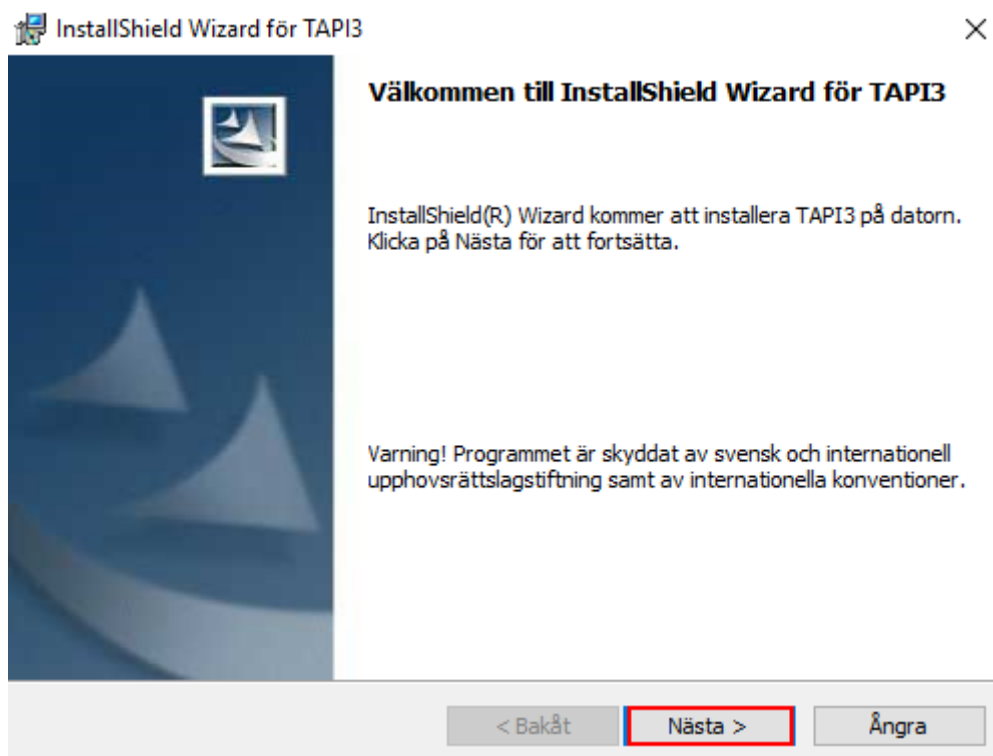
## 6.2. Configure Absence Integration

Absence functionality for IP Office is utilized using the TAPI interface. Avaya IP Office TAPI is required on each Trio Enterprise server to allow Trio Enterprise to interoperate with IP Office. In the example below, only the installation of TAPI software on Primary is explained. It is implied that the TAPI software and Enterprise company directory is already installed. (It is important that the TAPI software installation was run as administrator to ensure that the application receives the correct rights to run).

### 6.2.1. Installing and Configuring TAPI

After downloading the TAPI software, launch **tapiSetup.exe** (not shown).

The installation wizard screen is shown as below. Click on **Next** button.



In the screen shown below configure the following values.

**User Name:** Enter “TAPI”.

**User Password:** Keep it blank.

**IP Address:** Enter the IP address of the IP Office Primary Server.

Välj användarnamn

Välj eller ange ett användarnamn. Det är detta namn som arbetsstationsmjukvaran på denna PC kommer att använda. Användarnamnet kanske redan är konfigurerat på ert telefonsystem.

Användarnamn : TAPI

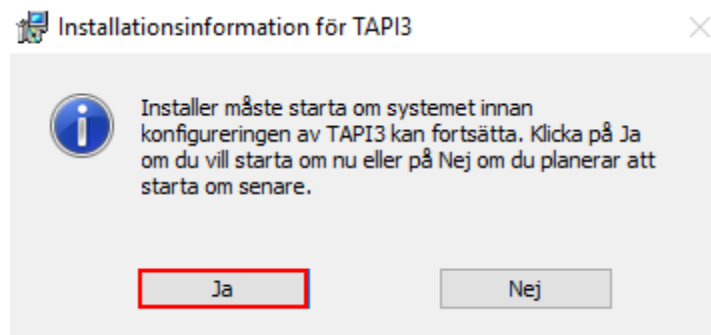
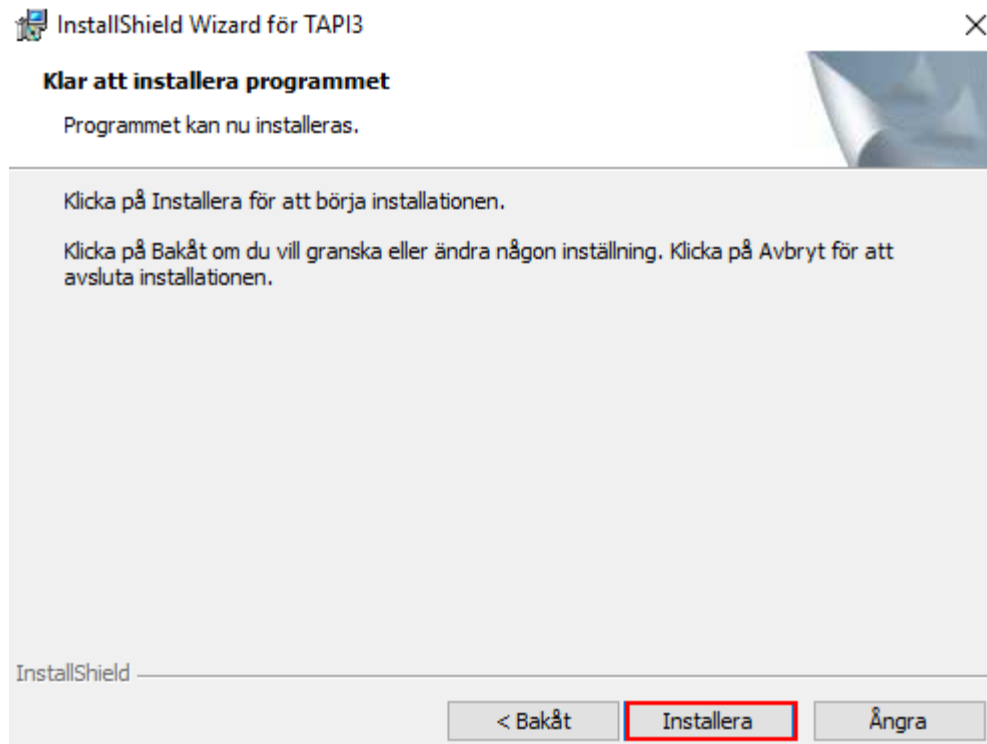
Användarlösenord :

IP-adress : 10.64.110.65

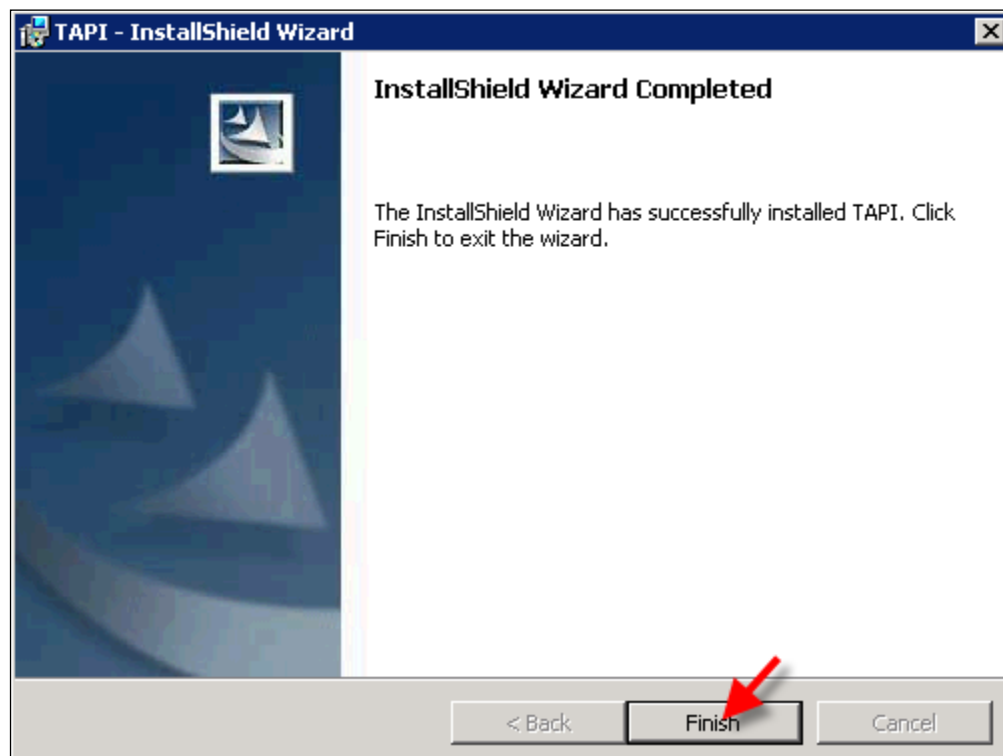
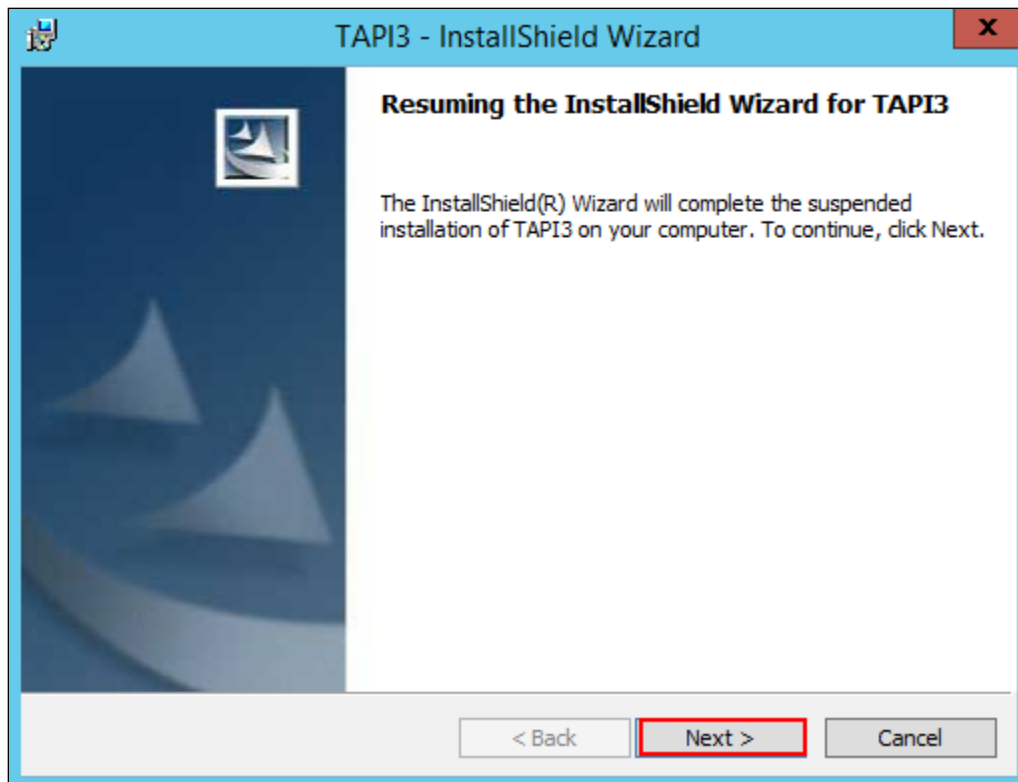
Bläddra..

< Åter    Nästa>    Avbryt

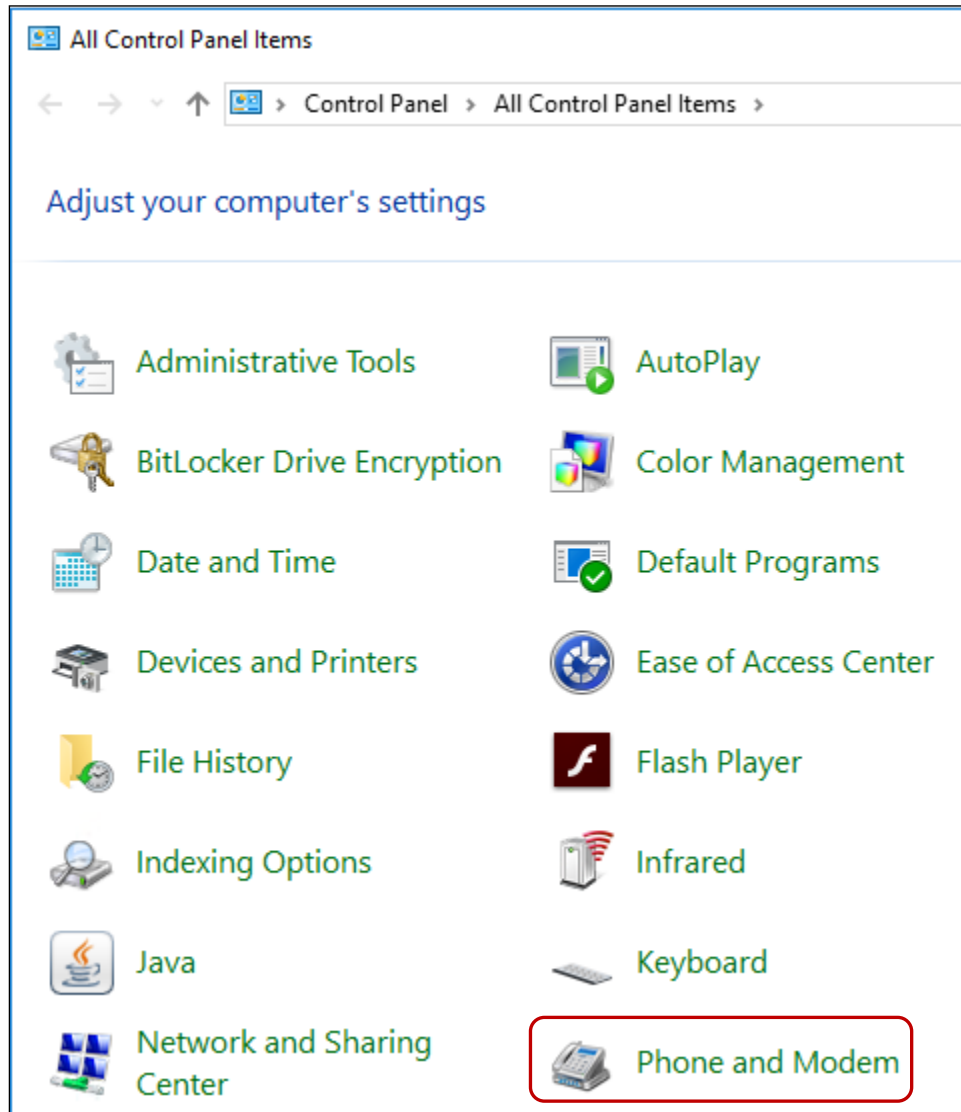
Follow the next instructions as shown in the screens below until the installation wizard process is completed. Click on **Install** to proceed



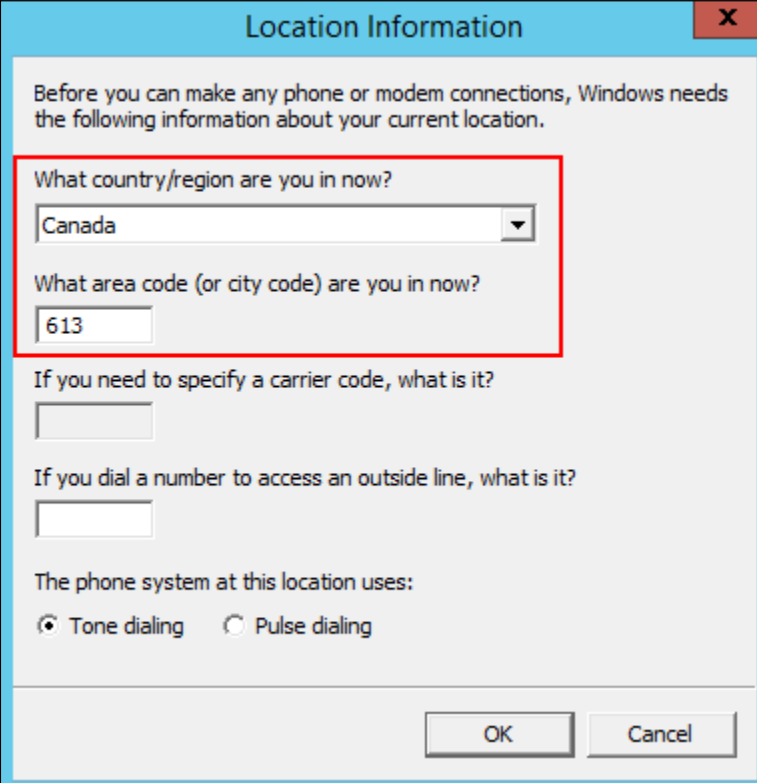




From the Windows OS Control Panel, select **Phone and Modem** as shown in the screen below.



In the **Location Information** screen shown below, select the required country and enter the required area code. Click on the **OK** button.

A screenshot of a Windows-style dialog box titled "Location Information". The dialog has a blue header bar with the title and a red close button. The main area is light gray and contains several prompts. The first prompt, "What country/region are you in now?", is followed by a dropdown menu showing "Canada". The second prompt, "What area code (or city code) are you in now?", is followed by a text box containing "613". These two prompts are enclosed in a red rectangular box. Below these are two more prompts: "If you need to specify a carrier code, what is it?" and "If you dial a number to access an outside line, what is it?", each followed by an empty text box. At the bottom, there is a section "The phone system at this location uses:" with two radio buttons: "Tone dialing" (which is selected) and "Pulse dialing". At the very bottom right are "OK" and "Cancel" buttons.

Location Information

Before you can make any phone or modem connections, Windows needs the following information about your current location.

What country/region are you in now?

Canada

What area code (or city code) are you in now?

613

If you need to specify a carrier code, what is it?

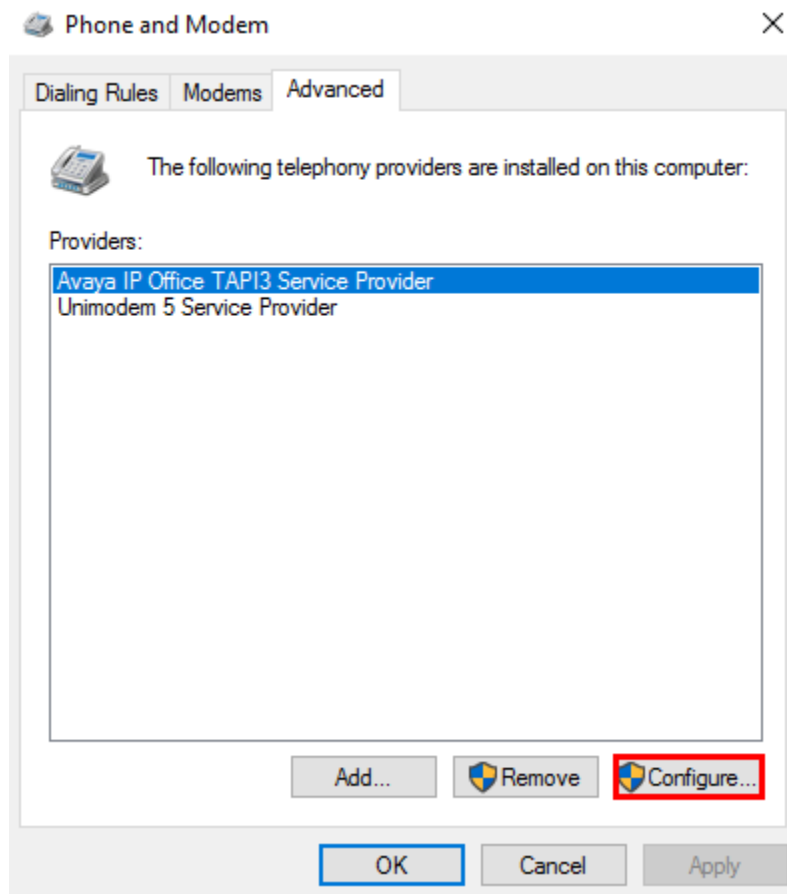
If you dial a number to access an outside line, what is it?

The phone system at this location uses:

☒ Tone dialing ☐ Pulse dialing

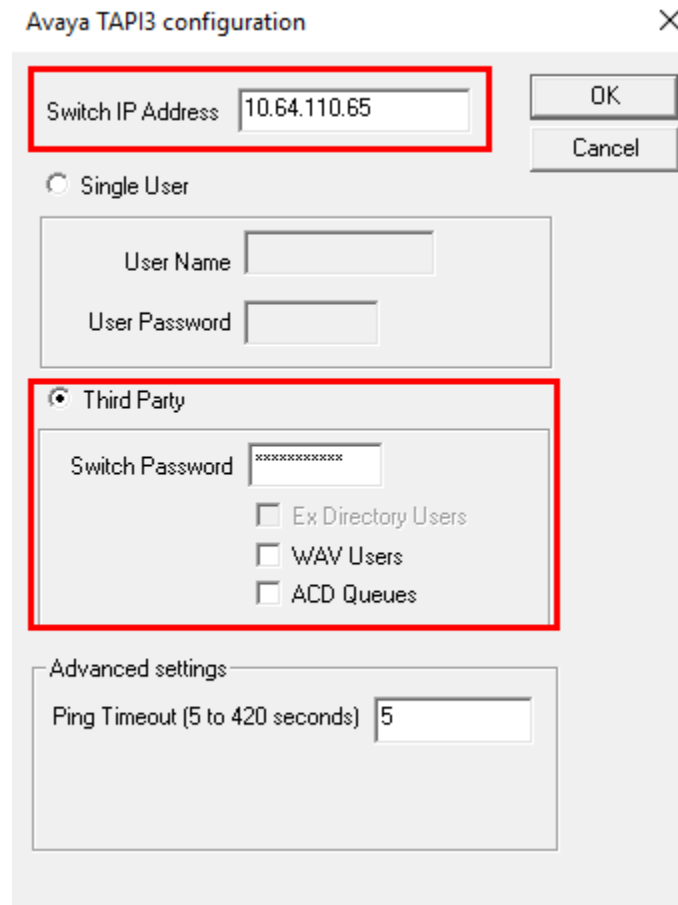
OK Cancel

In the next screen shown below, from the **Advanced tab**, select **Avaya IP Office TAPI3 Service Provider** and click on the **Configure** button.



In the **Avaya TAPI3 configuration** screen shown below, the **Switch IP Address** field is populated automatically by the IP Address of the Primary Server, which was configured earlier in this section during the installation wizard process. Click on the **Third Party** radio button and enter the **Switch Password**, which is the password of IP Office Primary Server.

Retain default settings for all other fields and click on the **OK** button.



The image shows a screenshot of the 'Avaya TAPI3 configuration' dialog box. The dialog has a title bar with the text 'Avaya TAPI3 configuration' and a close button (X). Inside the dialog, there are several sections. At the top, there is a 'Switch IP Address' field with the value '10.64.110.65' entered. To the right of this field are 'OK' and 'Cancel' buttons. Below this, there are two radio buttons: 'Single User' and 'Third Party'. The 'Third Party' radio button is selected. Below the radio buttons, there are two main sections. The first section, under the 'Single User' radio button, contains 'User Name' and 'User Password' fields. The second section, under the 'Third Party' radio button, contains a 'Switch Password' field with a masked password 'xxxxxxxxxx'. Below the 'Switch Password' field are three checkboxes: 'Ex Directory Users', 'WAV Users', and 'ACD Queues'. At the bottom of the dialog, there is an 'Advanced settings' section with a 'Ping Timeout (5 to 420 seconds)' field set to '5'.

Avaya TAPI3 configuration

Switch IP Address 10.64.110.65

OK

Cancel

☐ Single User

User Name

User Password

☒ Third Party

Switch Password xxxxxxxxxxxx

☐ Ex Directory Users

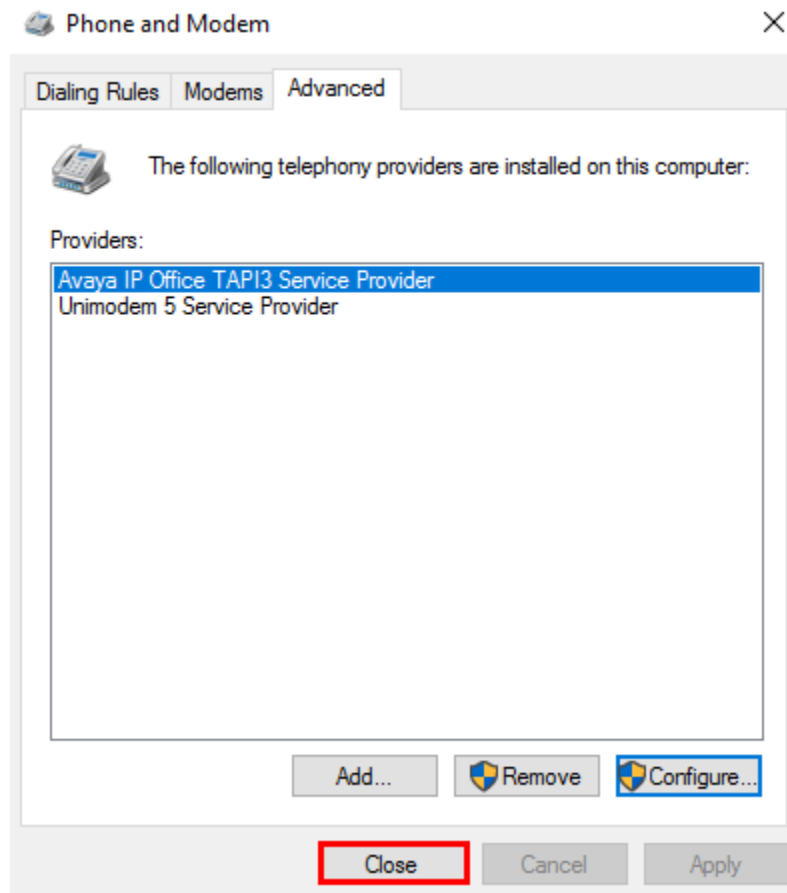
☐ WAV Users

☐ ACD Queues

Advanced settings

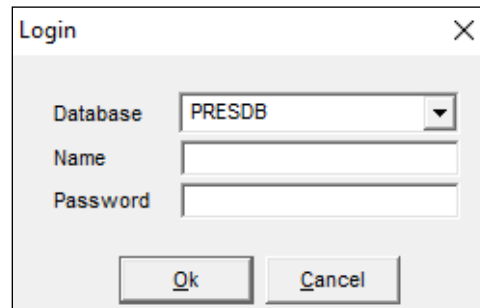
Ping Timeout (5 to 420 seconds) 5

Click on the **Close** button as shown in the screen below to complete the configuration.



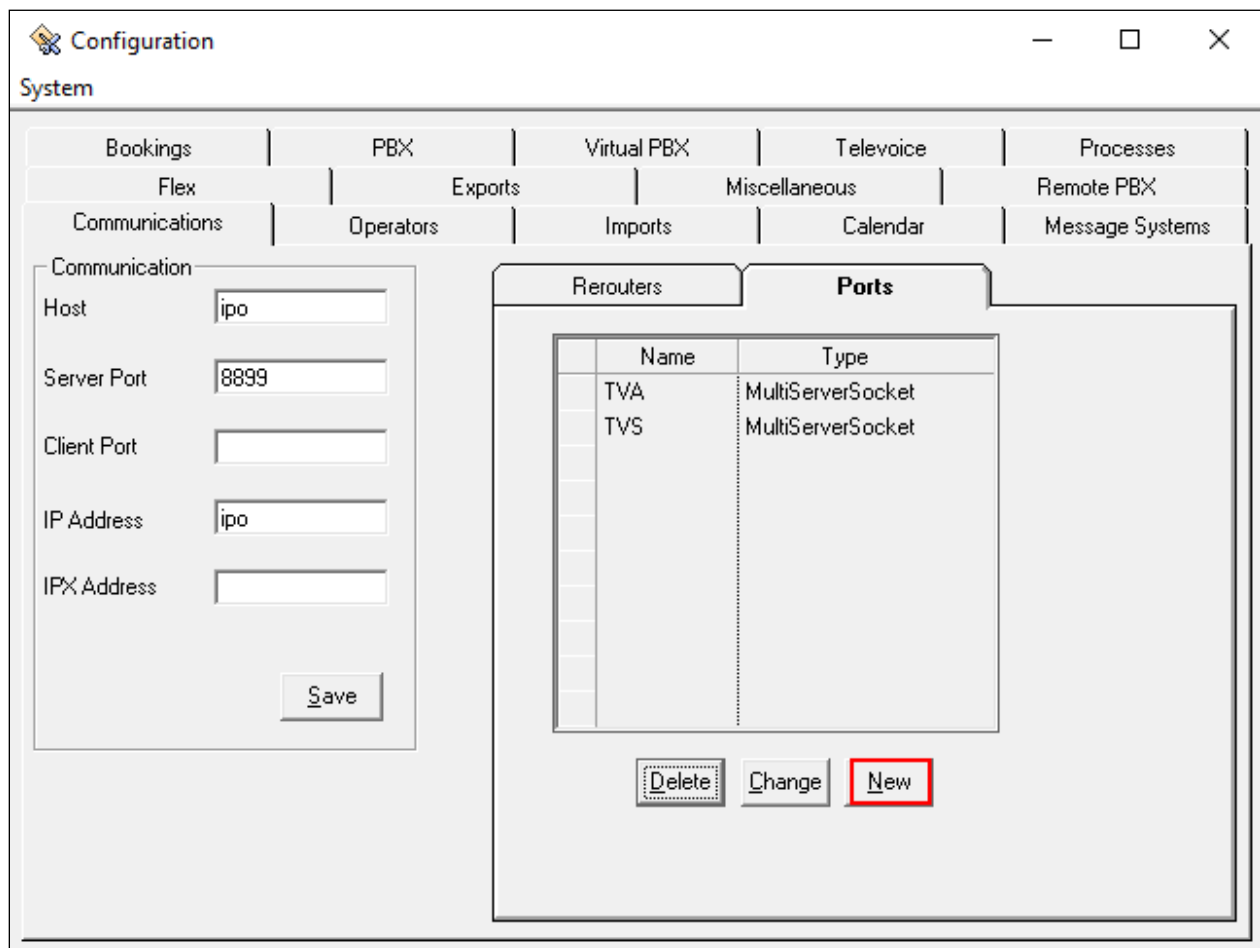
### 6.2.2. Configure Absence in Trio Enterprise

To configure the Absence connector; navigate to **Start → Programs → Trio Enterprise → Trio Present Setup** (not shown). Use the correct credentials to login as shown below.



A login dialog box titled "Login" with a close button (X) in the top right corner. It contains three input fields: "Database" with a dropdown menu showing "PRESDB", "Name" with an empty text box, and "Password" with an empty text box. At the bottom are "Ok" and "Cancel" buttons.

From the screen shown below, select **Communication → Ports** and then click on **New** as shown in the screen below.



A configuration window titled "Configuration" with a "System" sub-header. It features a tabbed interface with various categories: Bookings, PBX, Virtual PBX, Televoice, Processes, Flex, Exports, Miscellaneous, Remote PBX, Communications, Operators, Imports, Calendar, and Message Systems. The "Communication" tab is selected, showing fields for Host (ipo), Server Port (8899), Client Port, IP Address (ipo), and IPX Address, with a "Save" button. The "Ports" sub-tab is also selected, displaying a table with columns "Name" and "Type".

Name	Type
TVA	MultiServerSocket
TVS	MultiServerSocket

Below the table are "Delete", "Change", and "New" buttons. The "New" button is highlighted with a red border.

In the **Port** screen shown below, configure the following and click on the **OK** button.

- Select the **Client Socket** radio button.
- **Port Name:** Enter “TAPI (PLD)”.
- **Port No:** Enter “5299”.
- **IP Address:** Enter “127.0.0.1”.
- **Address Family:** Select “Internet (TCP/UDP)” from the drop-down menu.

The screenshot shows a dialog box titled "Port" with a close button (X) in the top right corner. On the left, under the "Type" section, there are four radio buttons: "Serial", "Server Socket", "Multi Socket Server", and "Client Socket". The "Client Socket" option is selected and highlighted with a red rectangle. To the right of the radio buttons, there is a red-bordered box containing four input fields: "Port Name" with the text "TAPI (PLD)", "Port No" with the text "5299", "IP Address" with the text "127.0.0.1", and "Address Family" with a dropdown menu showing "Internet (TCP/UDP)". Below this red box, there is a checkbox labeled "MD110 AAU" which is unchecked. Underneath the checkbox are two more input fields: "User Id:" and "Password:". At the bottom of the dialog box, there are two buttons: "OK" and "Cancel", both of which are highlighted with red rectangles.



From the screen shown below, select **PBX** and then click on **New**.

The screenshot shows a window titled "Configuration" with a "System" tab. The window contains a navigation pane on the left with a tree structure. The tree has a root node "System" with several sub-nodes: "Flex", "Exports", "Miscellaneous", and "Remote PBX". The "Flex" node is expanded, showing sub-nodes: "Communications", "Bookings", "Operators", "Virtual PBX", "Calendar", "Televoice", "Message Systems", and "Processes". The "Operators" node is selected, and the "PBX" sub-node is highlighted. The main area of the window displays a table with the following columns: "Id", "Type", "Name", "Port", "Prefix", "Net Grou", "Msg Wait", "Signal", "Ext. L", "Term L", and "Rea". The table is currently empty. Below the table, there are three buttons: "Delete", "Change", and "New". The "New" button is highlighted with a red border.

Id	Type	Name	Port	Prefix	Net Grou	Msg Wait	Signal	Ext. L	Term L	Rea
----	------	------	------	--------	----------	----------	--------	--------	--------	-----

Configure the **PBX** window as shown below.

- **Type:** Click on the “Tapi Generic” radio button.
- **Port:** Select “TAPI (PLD)” from the drop-down menu.
- **PbxName:** Enter an informative name.
- **Referral destination:** Enter the number that the calls should be forwarded to when a call forwarding (absence) is activated. This number is configured on the Trio Enterprise server for absence treatment. During compliance testing “51001” was configured.

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

The screenshot shows the 'PBX' configuration window. On the left, under the 'Type' section, the 'Tapi Generic' radio button is selected and highlighted with a red box. To the right, a group of fields is enclosed in a red rectangular box: 'Port' is set to 'TAPI (PLD)', 'PbxName' is 'IPO', 'Prefix' is empty, 'Domain' is empty, 'Extension Length' is '4', 'Net Group' is checked, and 'Referral Destination' is '51001'. Below these, 'Express No. to Voicemail' is empty. At the bottom left, the 'OK' button is highlighted with a red box. The 'Virtual' section on the left lists 'MCX', 'Microsoft Lync', 'Telenor MB', 'Generic Linestate', and 'MiCloud', all of which are unselected. The 'Address Space' field is set to '1'.

### 6.3. Configure Trio Enterprise Attendant

Trio Enterprise Attendant is a separate application to Trio Enterprise server and can run concurrently on the same platform. The attendant uses a regular Communication Manager telephone to make and receive calls, which are directed to the telephone by Trio Enterprise server. Start agent login by browsing to <https://<te server>/cc1/triowebagent>

- **User ID:** Enter a valid user ID
- **Password:** Enter a valid Password

Note this user ID and password is created during the user creation

Go to the setup tab (not shown)

- **Phone number:** Select the Communication Manager telephone number that will be used as the agent's audio device (number 70101 in this example).
- **Phone type:** Select "Standard phone" from the dropdown menu

Click on the **SAVE** button to save settings.

Trio Web Agent

tecmm/CC1/TrioWebAgent/settings

Trio Web Agent Search... ? LOG OUT

Agent 70101

LOGGED OFF

NORMAL

Search Queue Agents Services

Cases (0)

TE 8.0.37

### Settings

#### Login

Phone type \* Standard phone Phone number \* 70101

Location \* Default

Use Contact Center license

SAVE

#### Language

Language Browser language

#### Notifications

Use notifications

Play sound on new notification

## 7. Verification Steps

To verify that IP Office is connected to Trio Enterprise 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 1**. This is the SIP Line configured in **Section 5.5** for IP Office to connect to Trio Enterprise. 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 - 005056ABB116 (10.64.110.65) - IP Office Linux PC 11.0.4.0.0 build 74

### IP Office System Status

Help Snapshot LogOff Exit About

- System
- Alarms (3)
- Extensions (11)
- Trunks (3)
  - Line: 1
  - Line: 2
  - Line: 3
- Active Calls
- Resources
- Voicemail
- IP Networking
- Locations

#### SIP Trunk Summary

Line Service State: In Service  
Peer Domain Name: 10.64.10.111  
Resolved Address: 10.64.10.111  
Line Number: 1  
Number of Administered Channels: 10  
Number of Channels in Use: 0  
Administered Compression: G711 A  
Enable Faststart: Off  
Silence Suppression: Off  
Media Stream: RTP  
Layer 4 Protocol: UDP  
SIP Trunk Channel Licenses: 256  
SIP Trunk Channel Licenses in Use: 0  
SIP Device Features: REFER (Incoming and Outgoing), UPDATE (Incoming and Outgoing)

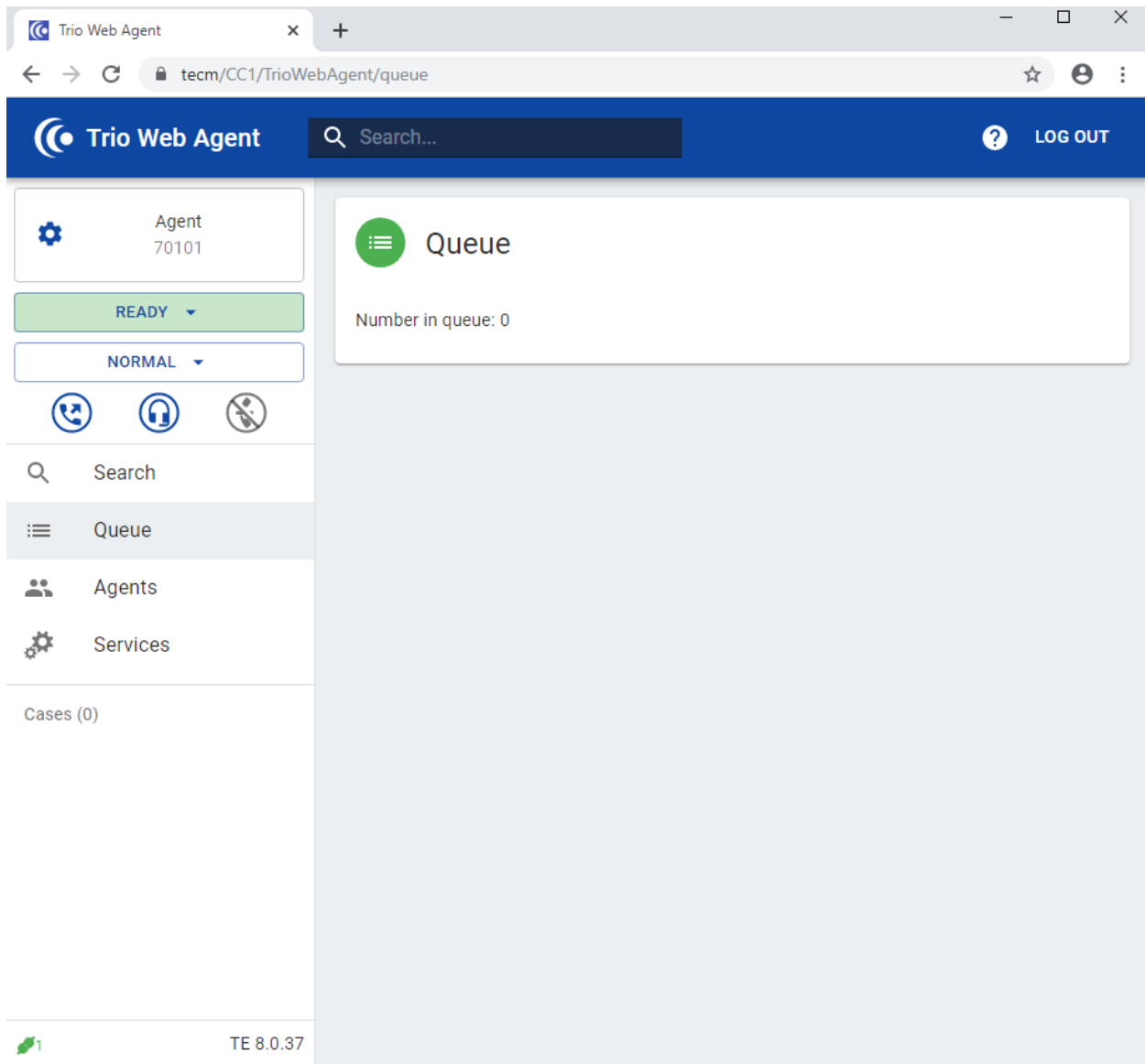
Cha...	U..	Call Ref	Curr...	Time in Remote	C...	Con...	Caller ID o...	Other Party on...	Dire...	Round Trip ...	Rec...	Rec...	Tran...	Tran...
1			Idle	00:0...										
2			Idle	00:0...										
3			Idle	00:3...										
4			Idle	01:2...										
5			Idle	01:2...										
6			Idle	01:2...										
7			Idle	01:2...										

Trace Trace All Pause Ping Call Details Graceful Shutdown Force Out of Service

Print... Save As...

11:41:31 PM Online

To verify that Trio Enterprise is connected to Communication Manager via Session Manager, log in to the Trio Enterprise Attendant at <https://<te server>/cc1/triowebagent>. Complete log in with the appropriate credentials as shown in **Section Error! Reference source not found..** The Trio Enterprise Attendant window appears as shown below. Select **Ready** from the drop-down box.



## 8. Conclusion

These Application Notes describe the configuration steps required for Enghouse Interactive Trio Enterprise to successfully interoperate with Avaya IP Office Server Edition. All feature and serviceability test cases were completed with any observations noted in **Section 2.2.**

## 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 11.0, May 2018.
- [2] *Deploying IP Office Essential Edition (IP500 V2)*, Release 11.0, 15-601042 Issue 33k - (Tuesday, October 9, 2018).
- [3] *Administering Avaya IP Office™ Platform with Manager*, Release 11.0, Issue 17a, August 2018.

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

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