



Avaya Solution & Interoperability Test Lab

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.

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 an attendant answering position for IP Office, as well as a call referral function that provides spoken information about the status of the extension called. The Trio Enterprise Attendant client provides a view of contacts, schedules, and communication tasks 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.

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 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.

It is possible to have multiple Trio attendant positions on an IP Office system. A variety of Avaya telephones were installed and configured on the IP Office.

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

2.1. Interoperability Compliance Testing

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 to busy extensions
- Call queuing and retrieval
- Loop detection for busy and unanswered extensions
- Absence detection

2.2. Test Results

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

- Trio Enterprise only supports RFC2833/RFC4733 for DTMF tone and this has to be taken into account when configuring the SIP Line in **Section 5.2**.
- Trio Enterprise uses TAPI client to set diversion which is used for activating and deactivating absence. In an IP Office environment which includes a server and expansion, TAPI client needs to be installed on every Trio Enterprise server that wants to set diversion on the phones that are configured on the server 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

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, has a SIP Trunk connection from Primary server to the Trio Enterprise server.

TAPI clients are configured on each Trio Enterprise server (1 and 2) connected to each IP Office as shown in test configuration below. This enables the Trio Enterprise to control a telephone via the IP Office to set diversion to activate or deactivate absence.

SIP and H323 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 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.

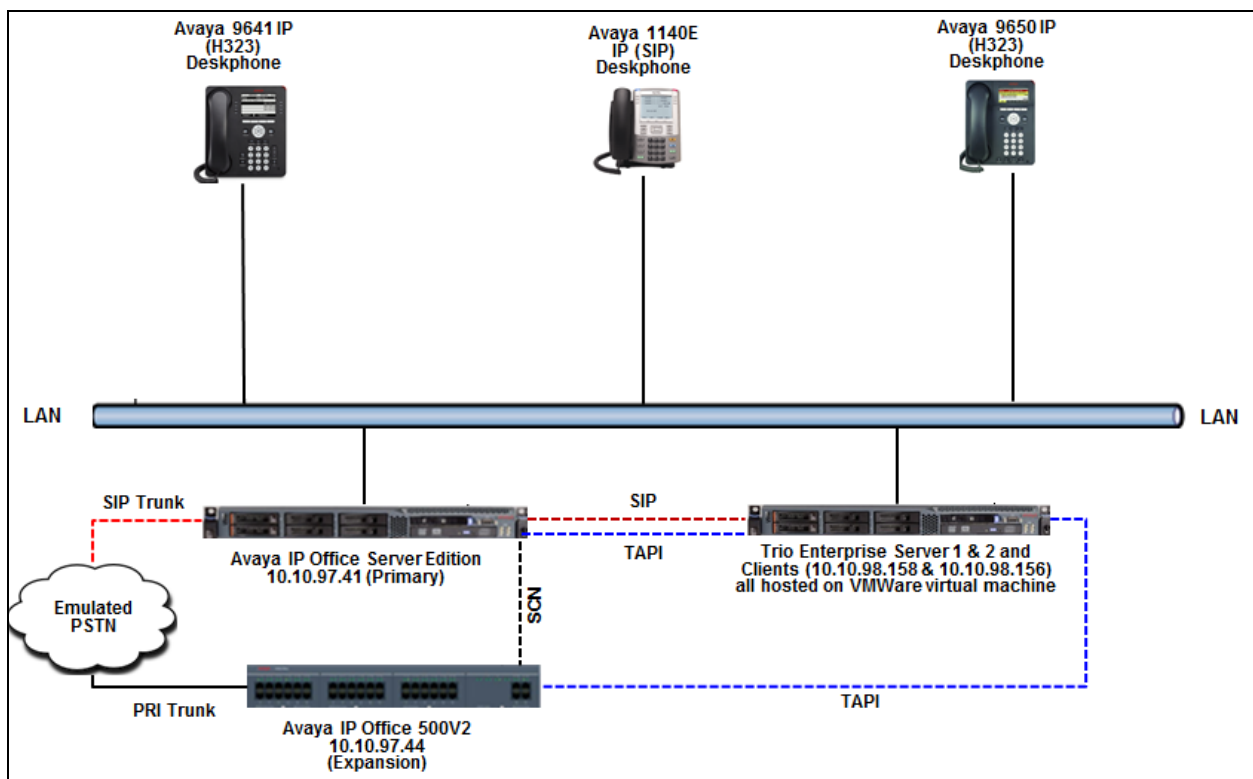


Figure 1: Avaya and Trio Enterprise Reference Configuration

4. Equipment and Software Validated

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

Equipment	Release/Version
Avaya IP Office Primary Linux	10.0 SP2
Avaya IP Office IP500V2	10.0 SP2
Avaya Telephones: <ul style="list-style-type: none">• 9650 IP (H323) Deskphone• 1140 IP (SIP) Deskphone• 9641 IP (H323) Deskphone	3.270B 04.04.26.00 6.6302
Trio Enterprise running on Microsoft Windows 2008 R2 Server TAPI3	Version 6.2.35 1.0.7

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.

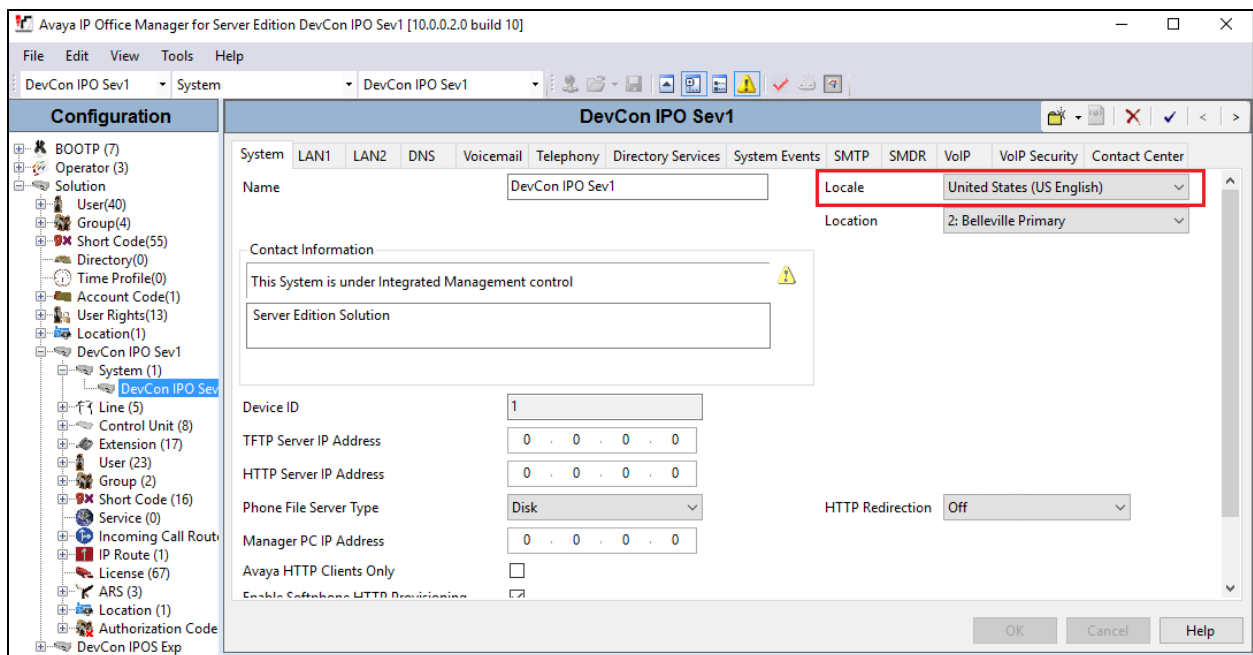
5. Avaya IP Office Configuration

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 of the Avaya IP Office for this solution. 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 9**. The configuration operations described in this section can be summarized as follows:

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

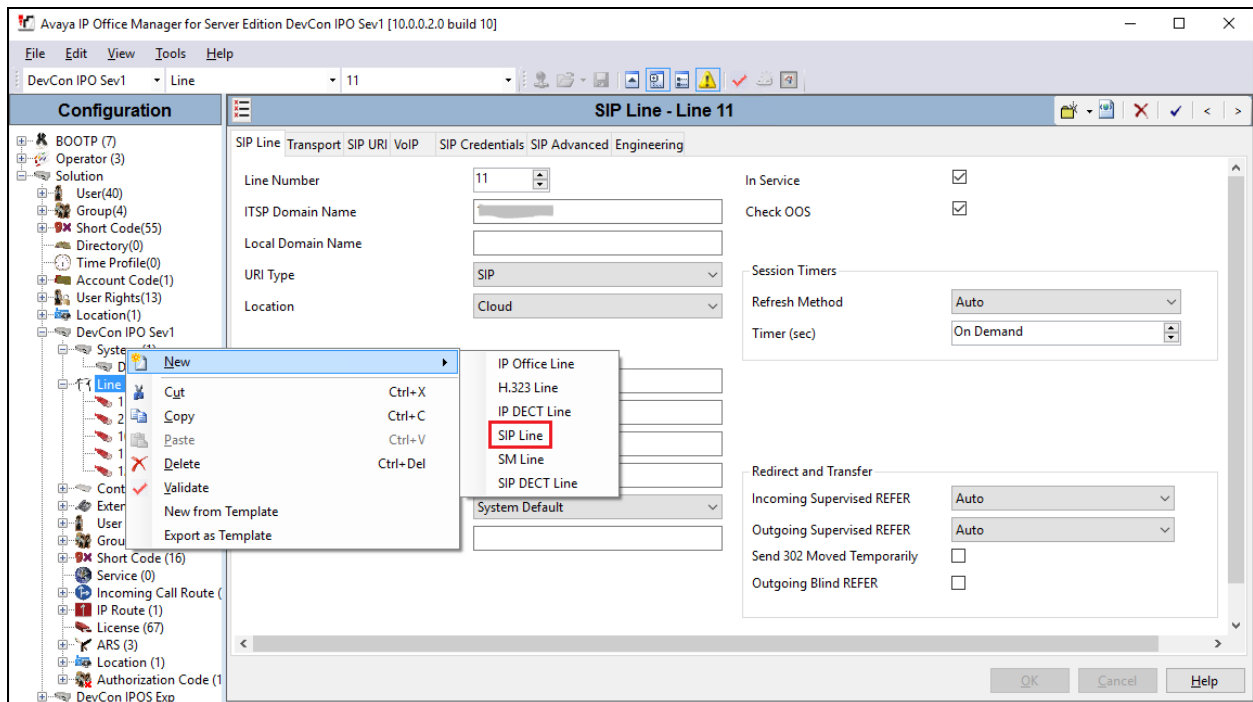
5.1. Configure System Locale

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



5.2. Create SIP Trunk

To create the SIP trunk from the IP Office to Trio Enterprise, navigate to **System → Line** and right click on **Line** followed by **New → SIP Line** as shown in the screen below. In this example, line **11** was created to connect to Trio Enterprise.

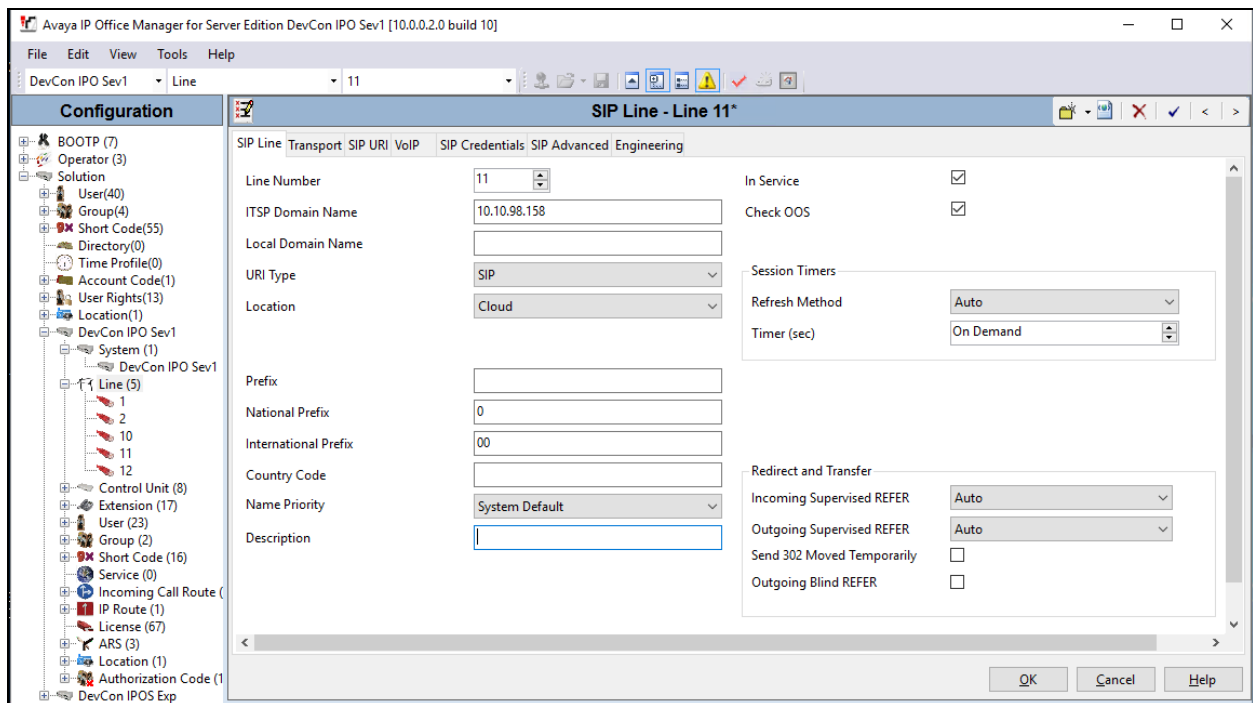


Screen below shows the configuration of the SIP Line in the subsequent **SIP Line** window; enter the following in the **SIP Line** tab.

Note: The **Line number** is filled in automatically.

- **ITSP Domain Name:** Enter the IP address of the Trio Server, **10.10.98.158**.

Retain default values for all remaining fields.



In the **Transport** tab enter the IP address of the Trio Enterprise Server in the **ITSP Proxy Address** field. Retain default values for remaining fields. For compliance testing only UDP protocol was tested.

Avaya IP Office Manager for Server Edition DevCon IPO Sev1 [10.0.0.2.0 build 10]

File Edit View Tools Help

DevCon IPO Sev1 Line 11

Configuration

SIP Line - Line 11*

SIP Line Transport SIP URI VoIP SIP Credentials SIP Advanced Engineering

ITSP Proxy Address 10.10.98.158

Network Configuration

Layer 4 Protocol UDP Send Port 5060

Use Network Topology Info None Listen Port 5060

Explicit DNS Server(s) 0 . 0 . 0 . 0 0 . 0 . 0 . 0

Calls Route via Registrar ☒

Separate Registrar

OK Cancel Help

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

- **Local URI:** Enter *
- **Contact:** Enter *
- **Display Name:** Enter *
- **Send Caller ID:** Select **Diversion Header** from the dropdown menu
- **Incoming Group:** Select **11** as configured earlier in this section
- **Outgoing Group:** Select **11** as configured earlier in this section

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

Avaya IP Office Manager for Server Edition DevCon IPO Sev1 [10.0.0.2.0 build 10]

File Edit View Tools Help

DevCon IPO Sev1 Line 11

Configuration

SIP Line - Line 11*

SIP Line Transport SIP URI VoIP SIP Credentials SIP Advanced Engineering

URI	Groups	Local URI	Contact	Display Name	Identity	Header	Originator Number	Send Caller ID	Diversion Header	Credential	Max
1	11	11	Auto	Auto	Auto	None	PAI	Diversion	None	0: <Non...	10

Add... Remove Edit...

OK Cancel

Edit URI

Local URI * Contact * Display Name *

Identity: None Header: P Asserted ID

Forwarding And Twinning

Originator Number

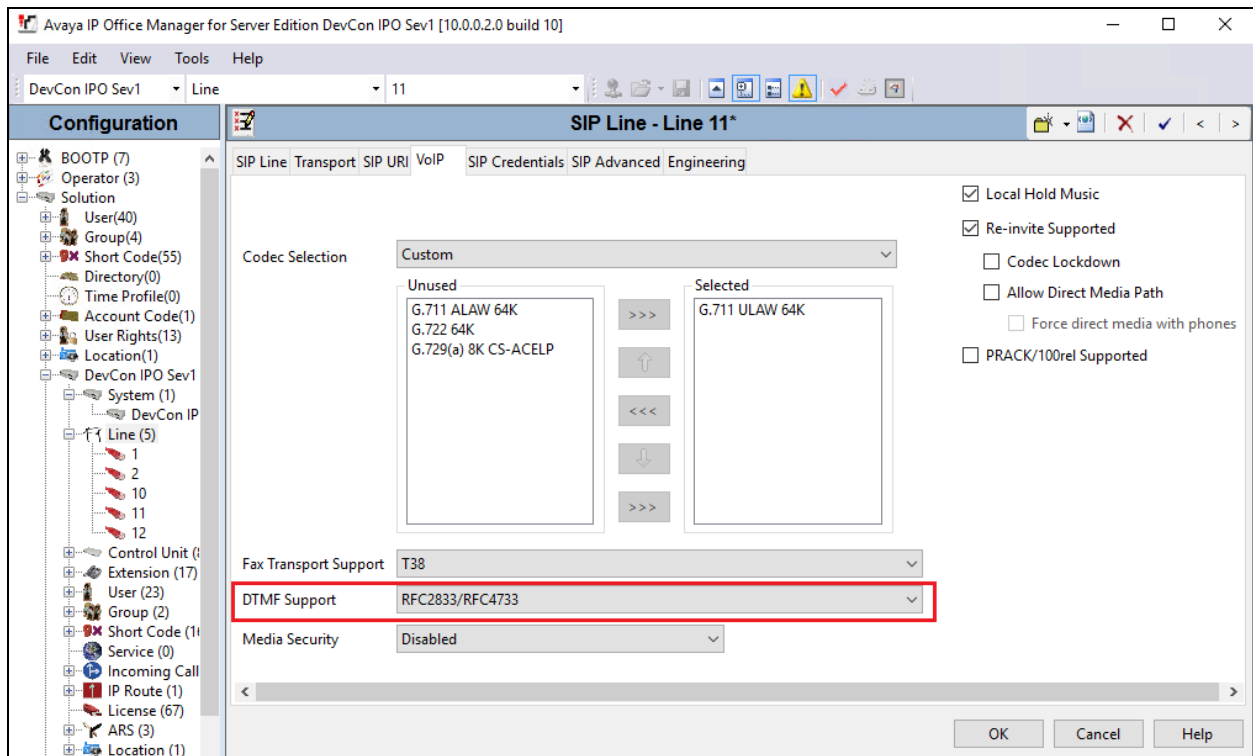
Send Caller ID: Diversion Header

Diversion Header: None Registration: 0: <None>

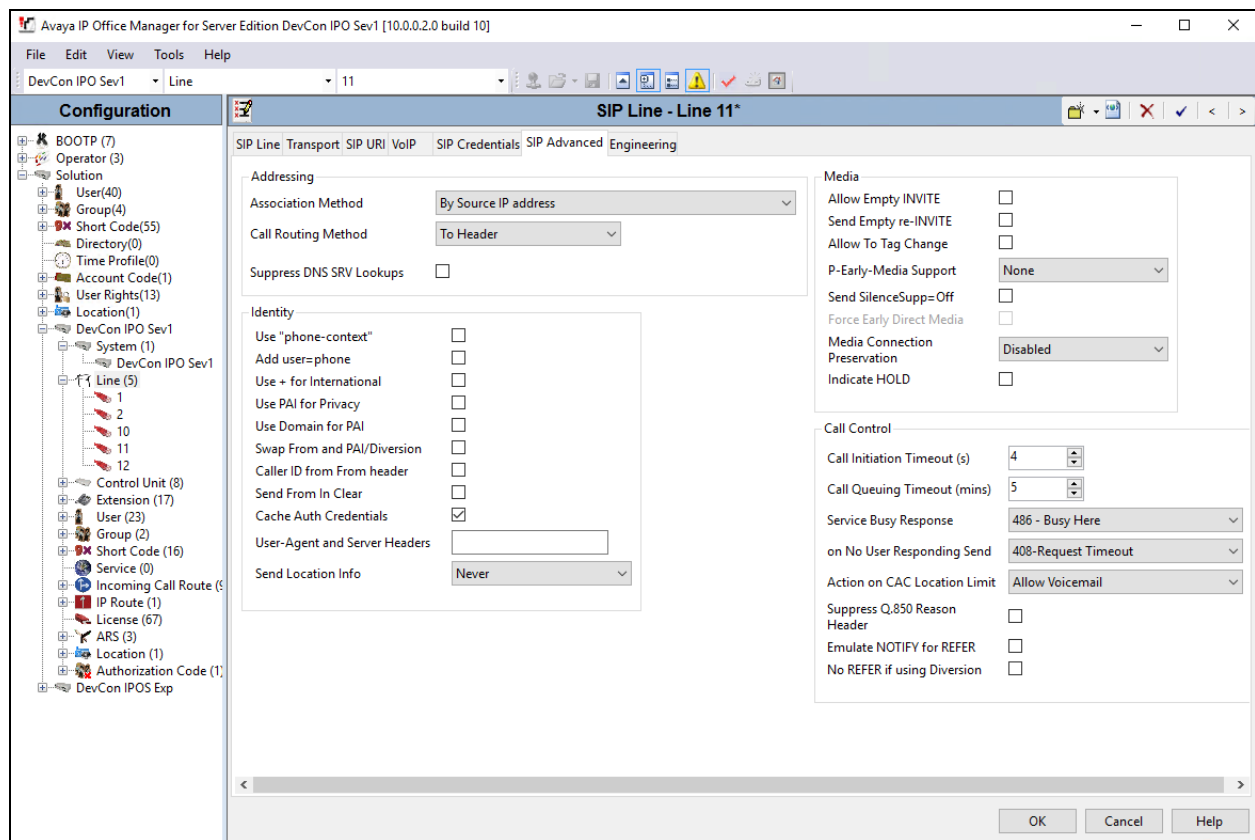
Incoming Group: 11 Outgoing Group: 11

OK Cancel Help

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.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. Configure Incoming Call Route

To configure the Incoming Call Route, navigate to **DevCon IPO Sev1 → Incoming Call Route** and right click on **Incoming Call Route** followed by **New** (not shown). In the subsequent window, enter the following in the **Standard** tab.

- **Bearer Capability:** Select **AnyVoice** from the drop down menu
- **Line Group ID:** Select **11**, the SIP Line as configured in **Section 5.2**

Retain default values for all remaining values.

Avaya IP Office Manager for Server Edition DevCon IPO Sev1 [10.0.0.2.0 build 10]

File Edit View Tools Help

DevCon IPO Sev1 Incoming Call Route 11

Configuration

Standard Voice Recording Destinations

11

Bearer Capability Any Voice

Line Group ID 11

Incoming Number

Incoming Sub Address

Incoming CLI

Locale

Priority 1 - Low

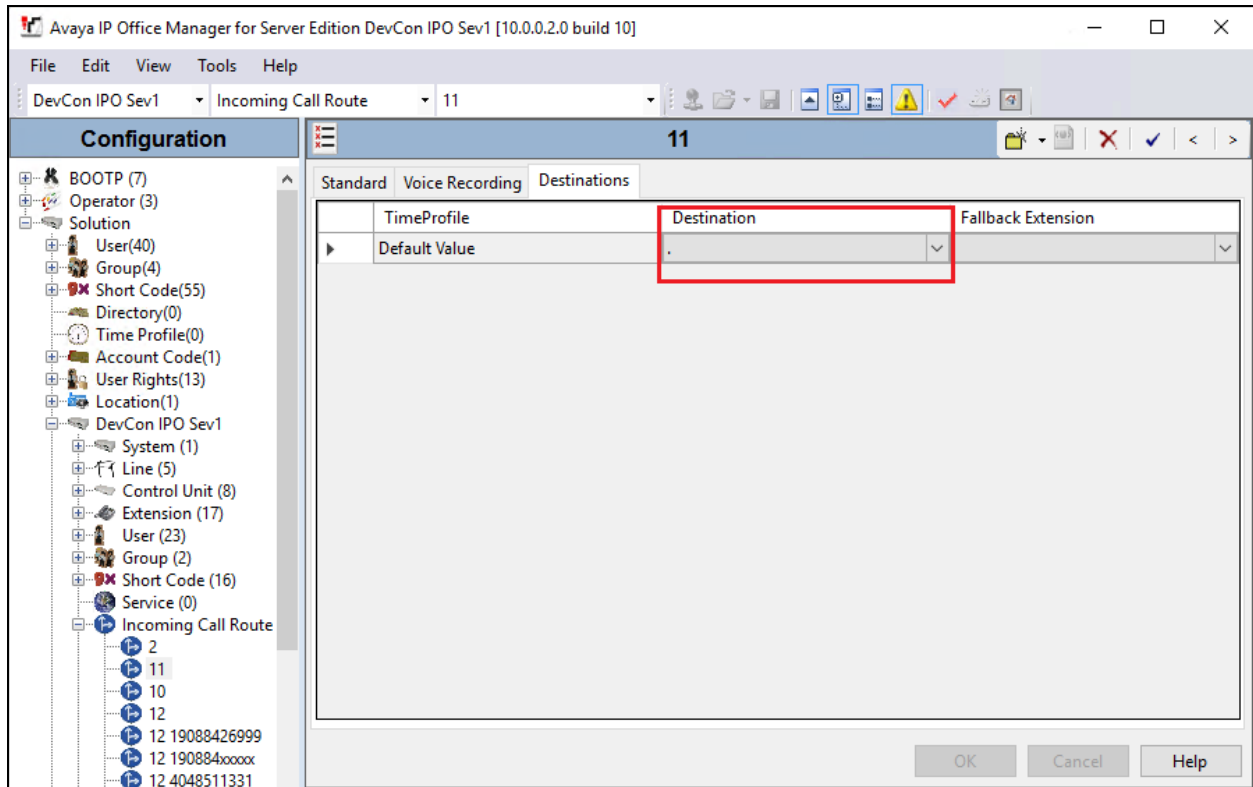
Tag

Hold Music Source holdmusic

Ring Tone Override None

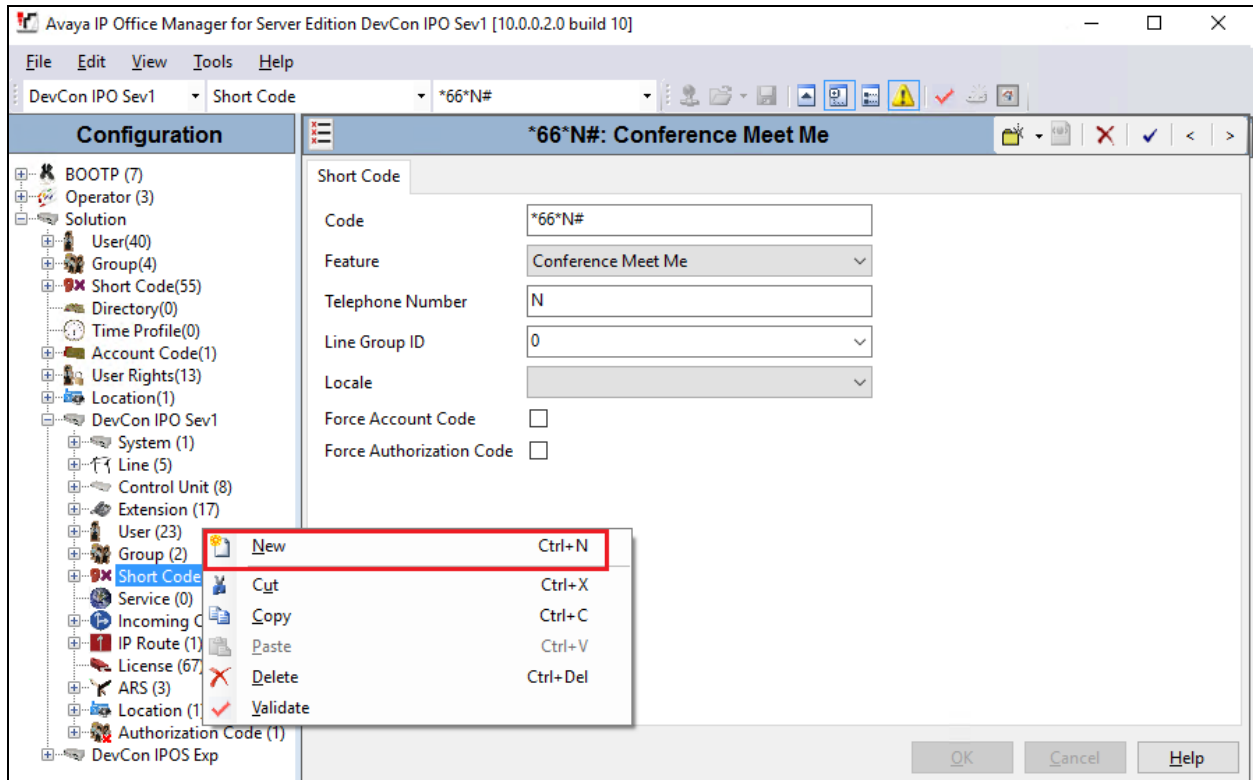
OK Cancel Help

In the **Destinations** tab, under the **Destination** column enter.. Retain default values for all remaining fields and click the **OK** button.



5.4. Create Short Code (Route Calls)

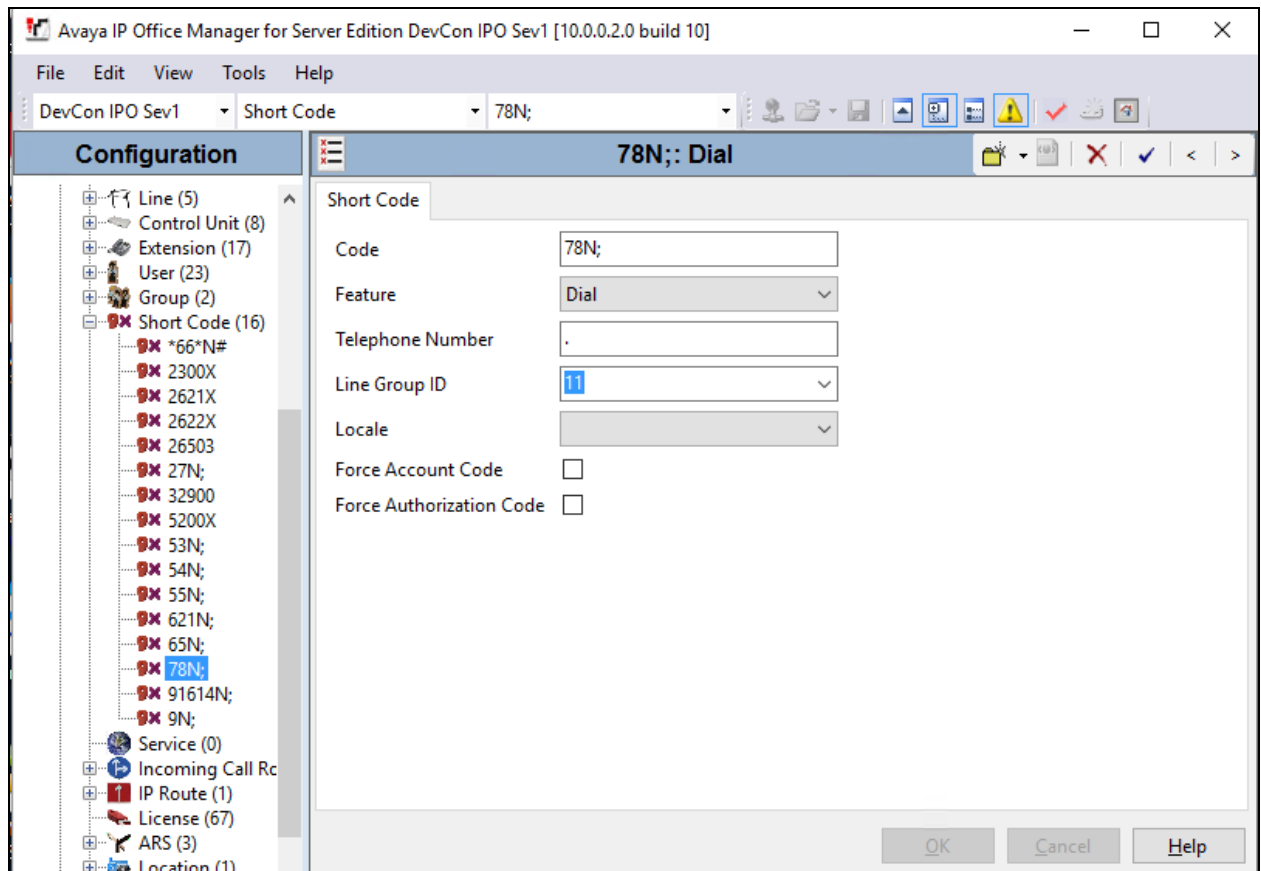
A Short Code needs to be configured on the IP Office to route calls to Trio Enterprise. Navigate to **DevCon IPO Sev1 → Short Codes**, and then right click and select **New** as shown in the screen below.



In the subsequent window, enter the following:

- **Code:** Enter the number range that will be routed to Trio Enterprise (during compliance testing, all numbers beginning with 78 were sent to Trio Enterprise, therefore **78N;** was entered)
- **Feature:** Select **Dial** from the dropdown menu
- **Telephone Number:** Enter .
- **Group Line ID:** Enter **11**, the SIP Line configured in **Section 5.2**

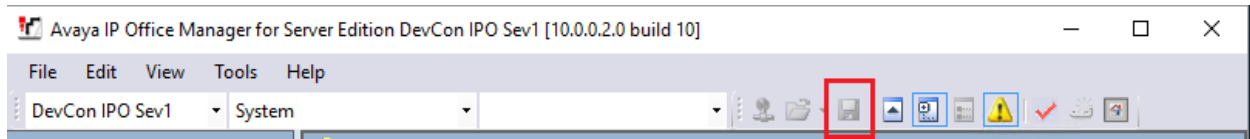
Retain default values for all remaining fields and click the **OK** button.



A similar short code can be configured for user to dial to set absence; however during compliance testing, user dialled **78003** to reach the diversion number on Trio Enterprise and entered the required codes configured on Trio Enterprise to activate and deactivate absence.

5.5. Save Configuration

Once all the configurations are complete, the changes need to be saved on the IP Office System. To achieve this, click on the Save Icon as shown in the screen below. 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 Trio Enterprise

Trio Enterprise connects as a SIP Line Trunk to Avaya IP Office. This section shows how to configure Trio Enterprise to successfully connect to Avaya 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.

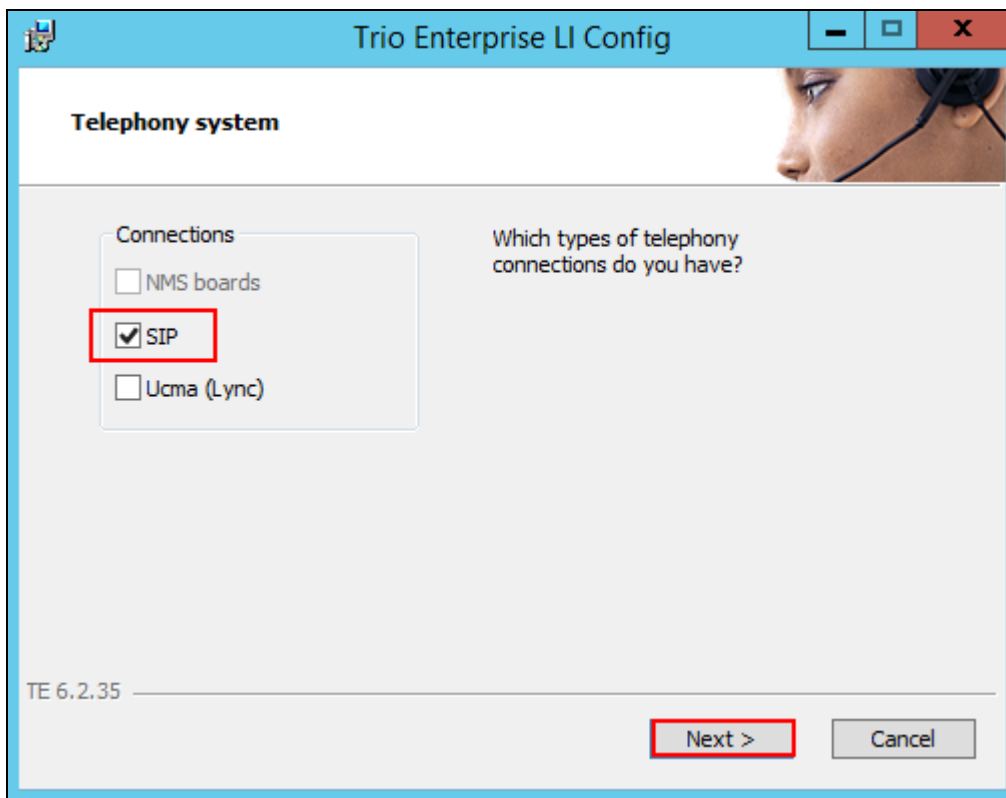
6.1. Configure Trio Enterprise to use SIP Trunks

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).

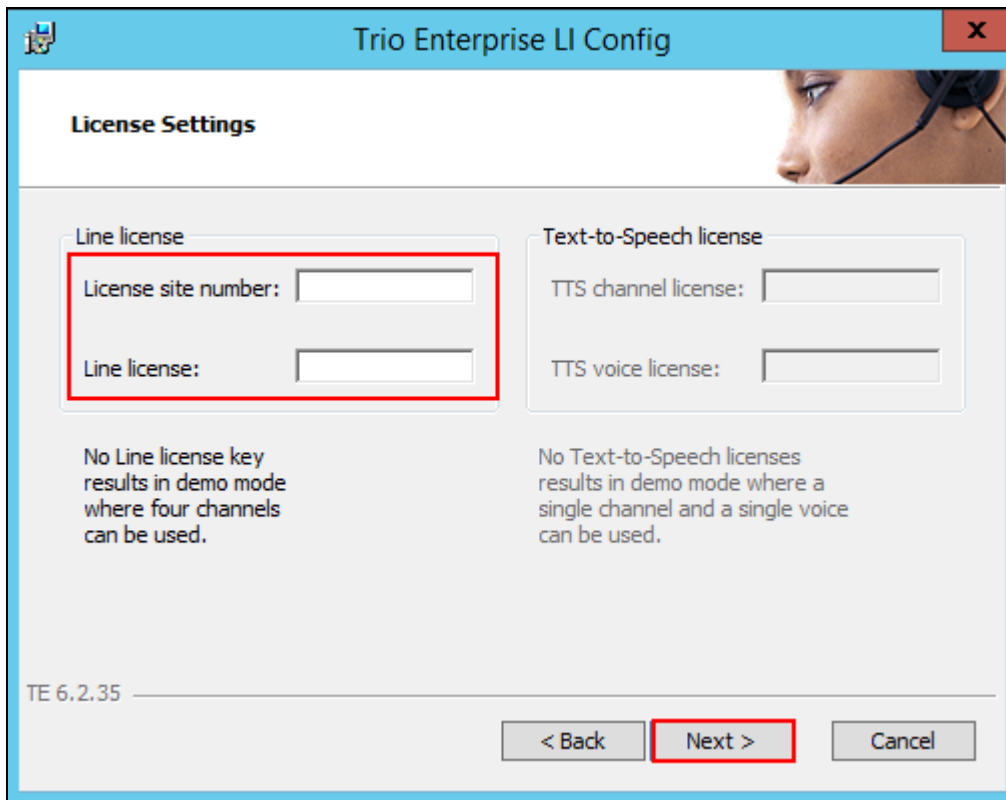


Launch the 'TeleVoice Config' shortcut

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 screenshot shows a software configuration window titled "Trio Enterprise LI Config". The window has a blue header bar with a close button (X) in the top right corner. Below the header, the title "License Settings" is displayed. 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:". These two fields are enclosed in a red rectangular box. Below these fields, a note states: "No Line license key results in demo mode where four channels can be used." The right column is titled "Text-to-Speech license" and contains two text input fields: "TTS channel license:" and "TTS voice license:". Below these fields, a note 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 version "TE 6.2.35" is displayed. At the bottom right, there are three buttons: "< Back", "Next >", and "Cancel". The "Next >" button is highlighted with a red rectangular box.

Trio Enterprise LI Config

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 6.2.35

< Back **Next >** Cancel

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

Trio Enterprise LI Config

SIP 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
☐ MD110/MX-ONE ☐ SIEMENS
☐ PHILIPS ☐ CISCO
☐ Nortel CS1000/Meridian ☐ PSTN
☐ ALCATEL4200
☐ ALCATEL4300
☐ ALCATEL4400

TE 6.2.35

< 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
- **Codecs:** Check the box **Enable the for G711 mu-law codec**

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

Trio Enterprise LI Config

SIP Settings(2)

SIP settings

Local IP: 10.10.98.158

Port: 5060

Target IP: 10.10.97.41

Port: 5060

Number of channels: 30

Codecs

☒ Enable G711 mu-law codec

TE 6.2.35

< 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

SIP Settings(3)

Address Space (AS)

- ☒ Use LI Address Space
- ☐ AS Name:
- ☐ No Address Space

Sip Options

- ☒ UPDATE support

Routing

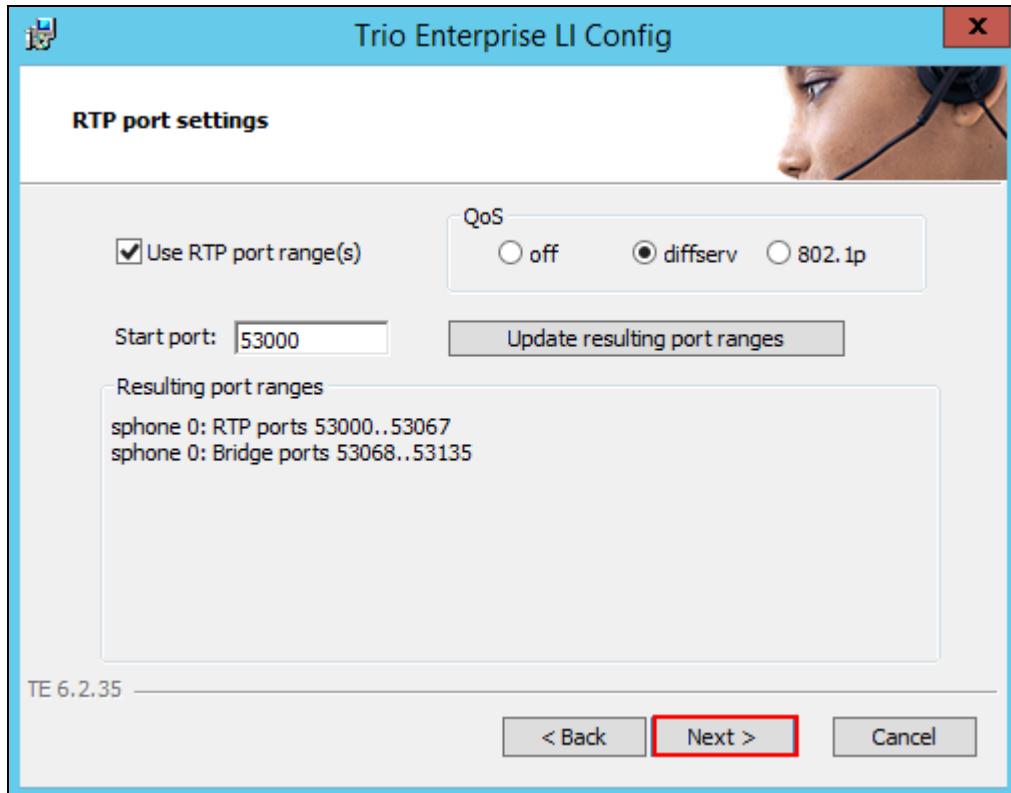
- ☒ Enable IP routing

TE 6.2.35

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.

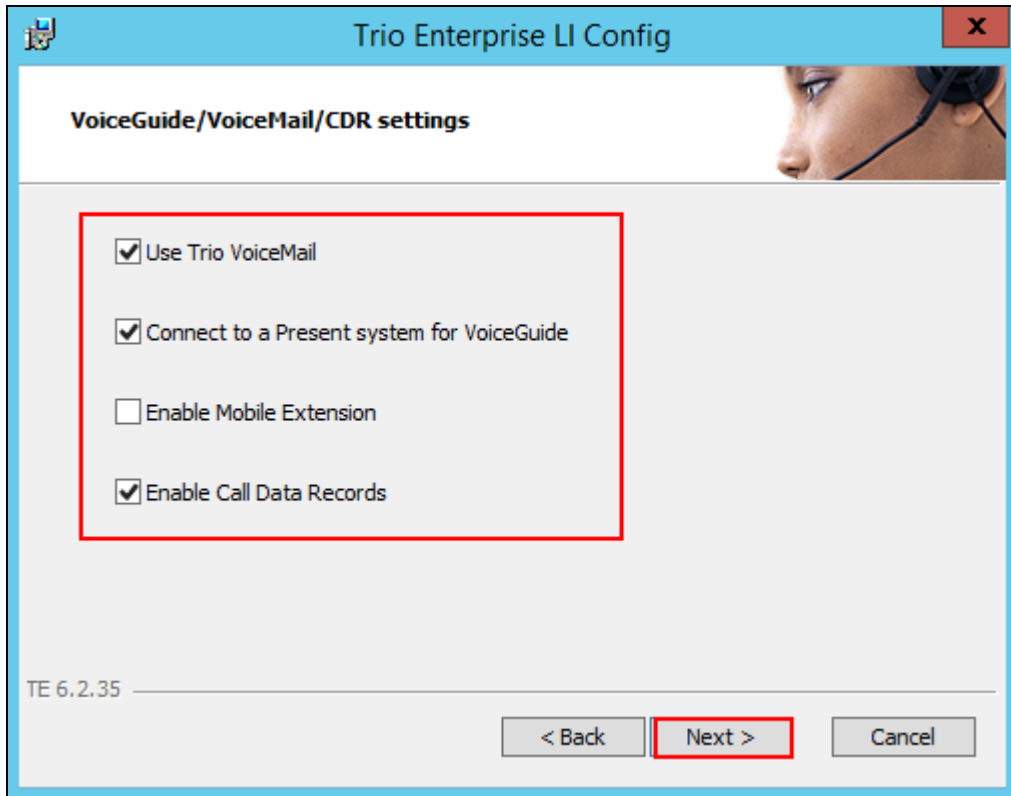


The screenshot shows the 'Trio Enterprise LI Config' window with the 'RTP port settings' tab selected. The 'Use RTP port range(s)' checkbox is checked. The 'QoS' section has three radio buttons: 'off', 'diffserv' (which is selected), and '802.1p'. The 'Start port' field contains the value '53000'. An 'Update resulting port ranges' button is located to the right of the 'Start port' field. Below this, a text box displays the 'Resulting port ranges' as follows:
sphone 0: RTP ports 53000..53067
sphone 0: Bridge ports 53068..53135
At the bottom of the window, there are three buttons: '< Back', 'Next >' (which is highlighted with a red border), and 'Cancel'. The version 'TE 6.2.35' is displayed in the bottom left corner.

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.

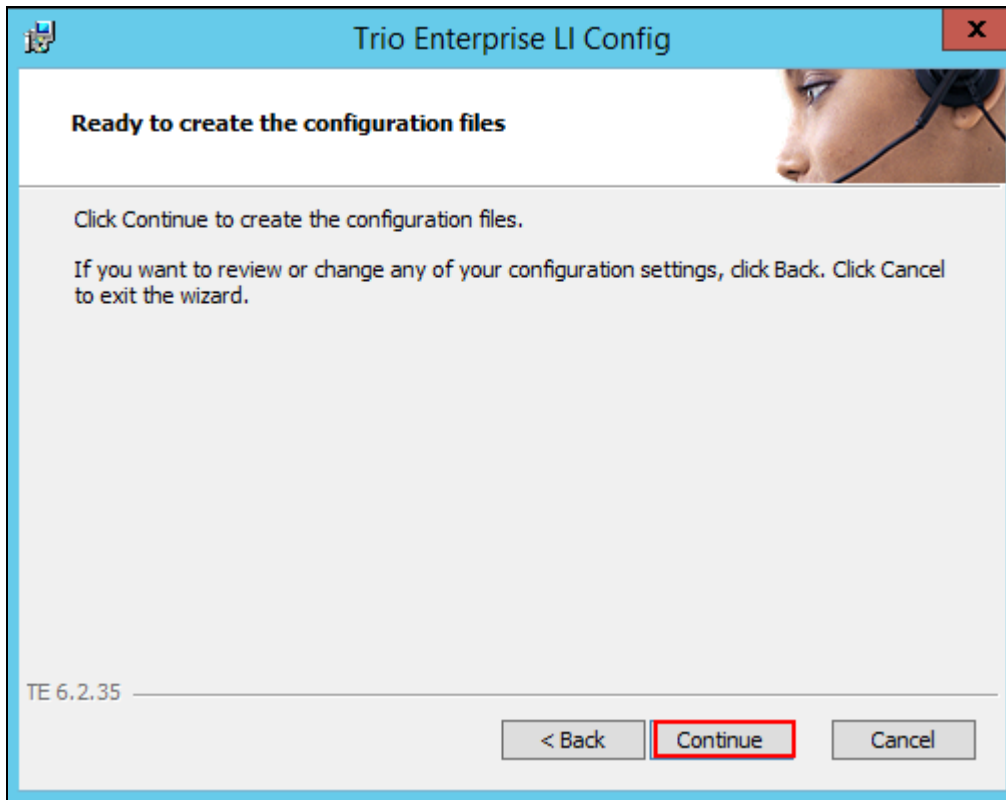


The screenshot shows a window titled "Trio Enterprise LI Config" with a close button (X) in the top right corner. Below the title bar, there is a header "VoiceGuide/VoiceMail/CDR settings" and a small image of a person wearing a headset. The main content area contains four settings, each with a checkbox:

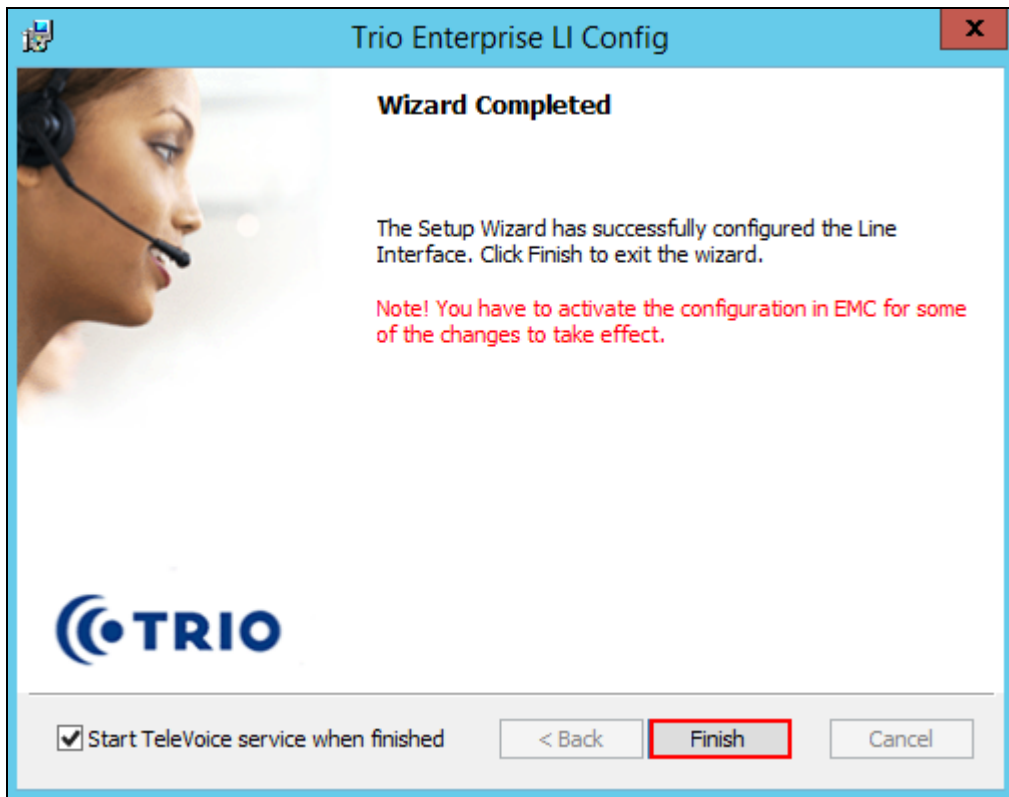
- ☒ Use Trio VoiceMail
- ☒ Connect to a Present system for VoiceGuide
- ☐ Enable Mobile Extension
- ☒ Enable Call Data Records

At the bottom left, the text "TE 6.2.35" is displayed. At the bottom right, there are three buttons: "< Back", "Next >" (highlighted with a red box), 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.



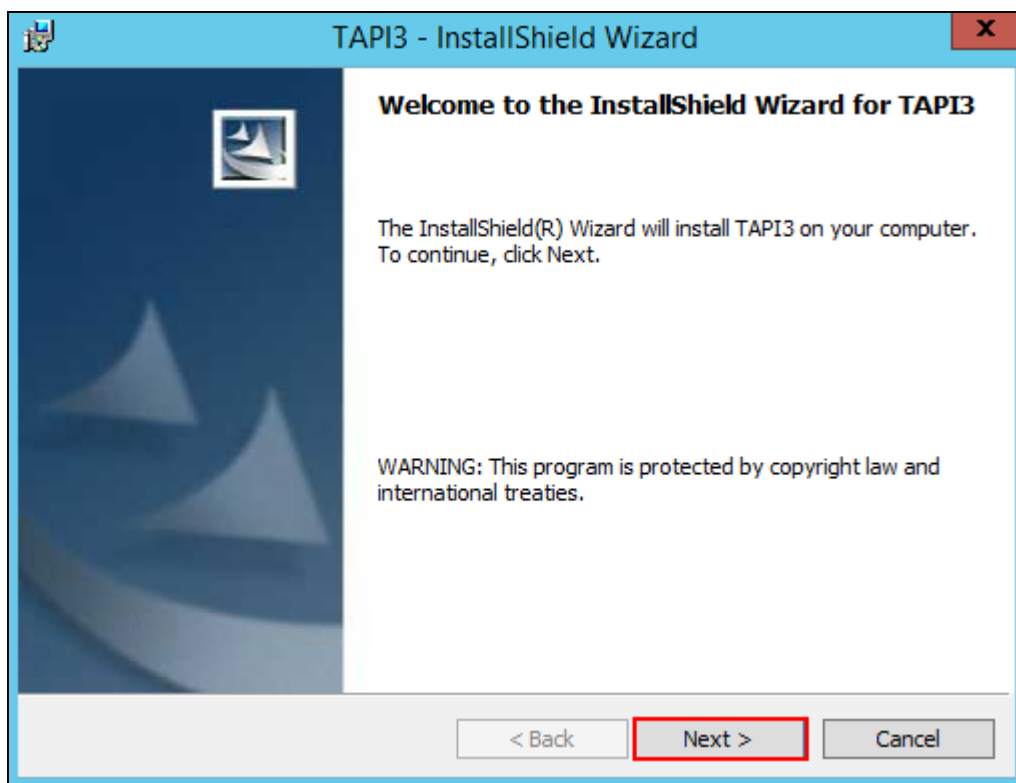
6.2. Configure Absence Integration

Absence functionality for Avaya IP Office is utilized using the TAPI interface. The Avaya IP Office TAPI is required on each Trio server so as to allow Trio Enterprise to interoperate with IP Office Primary Server and IP500V2. 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

Select a User Name

Please select or enter a User Name. This is the name that the workstation software installed on this PC will use. The User Name may already be configured on your telephone system.

User Name : TAPI

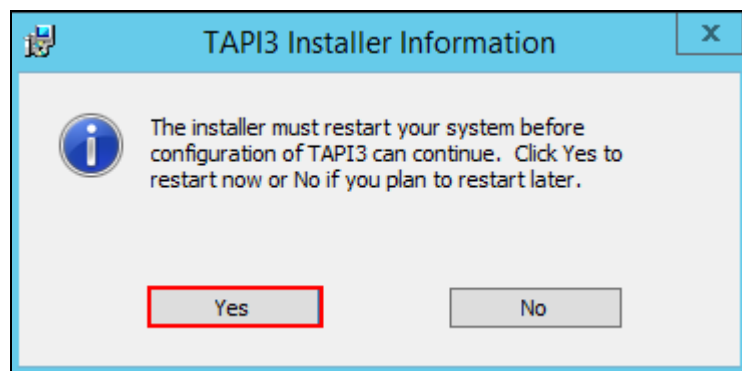
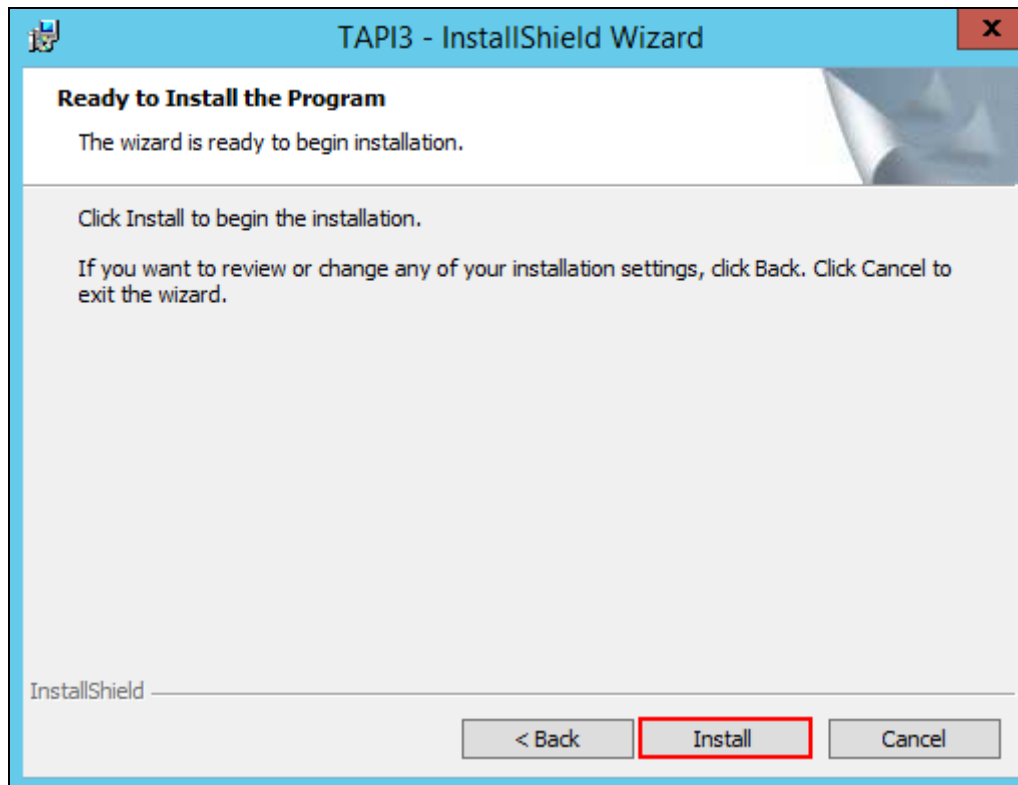
User Password :

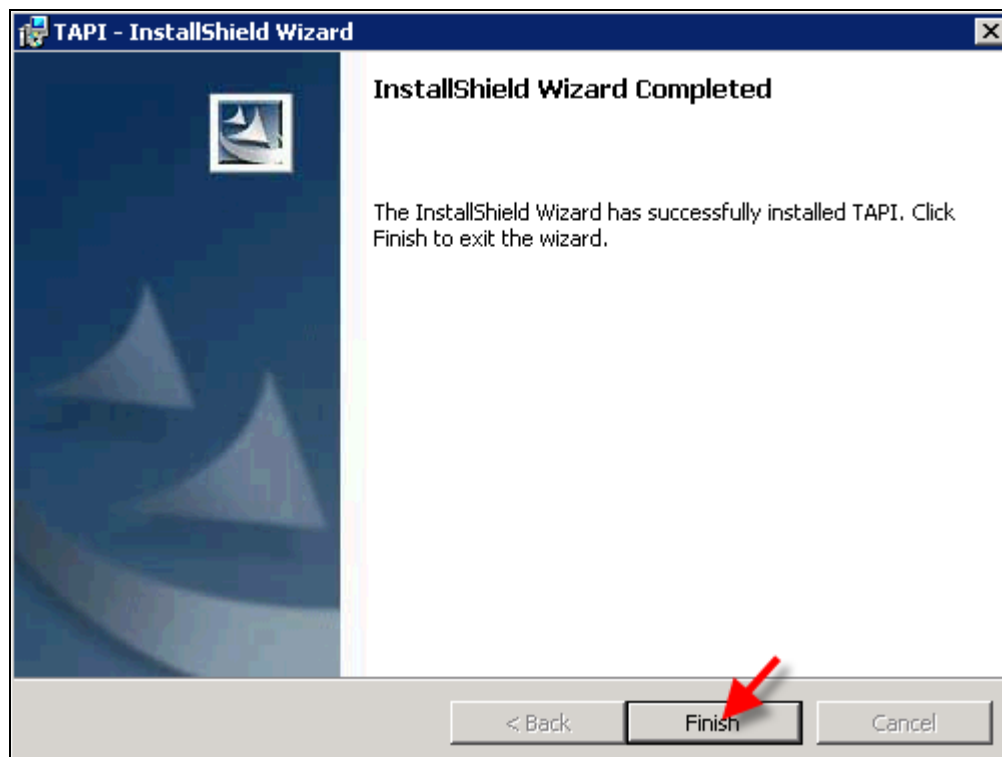
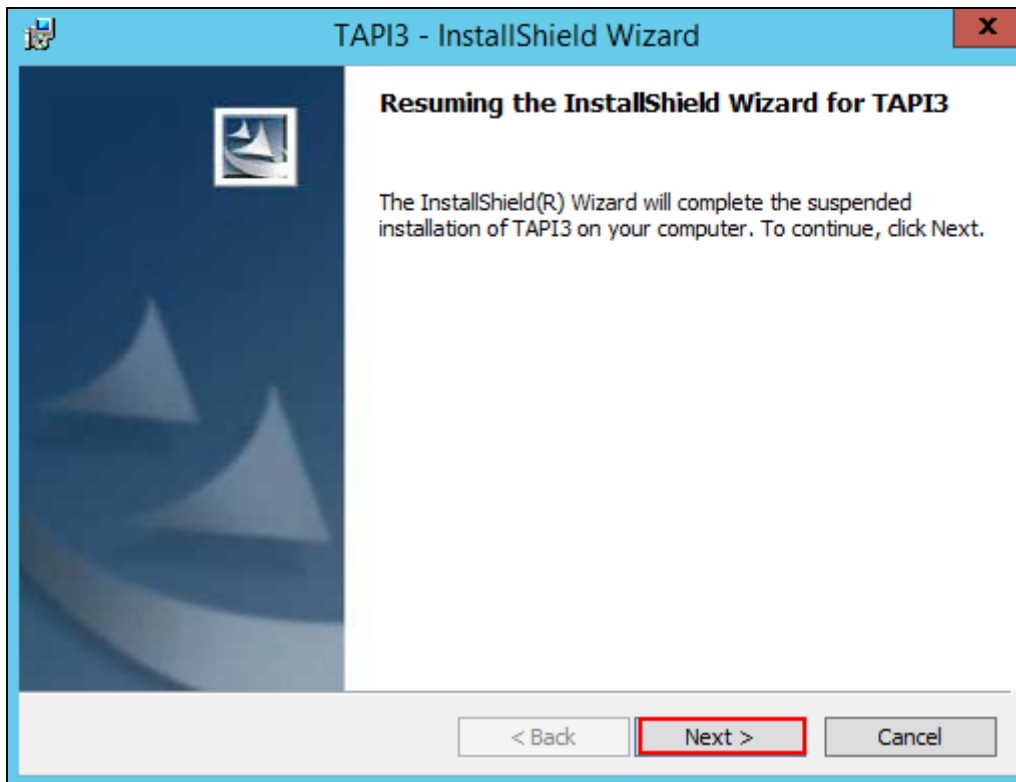
IP Address : 10.10.97.41

Browse..

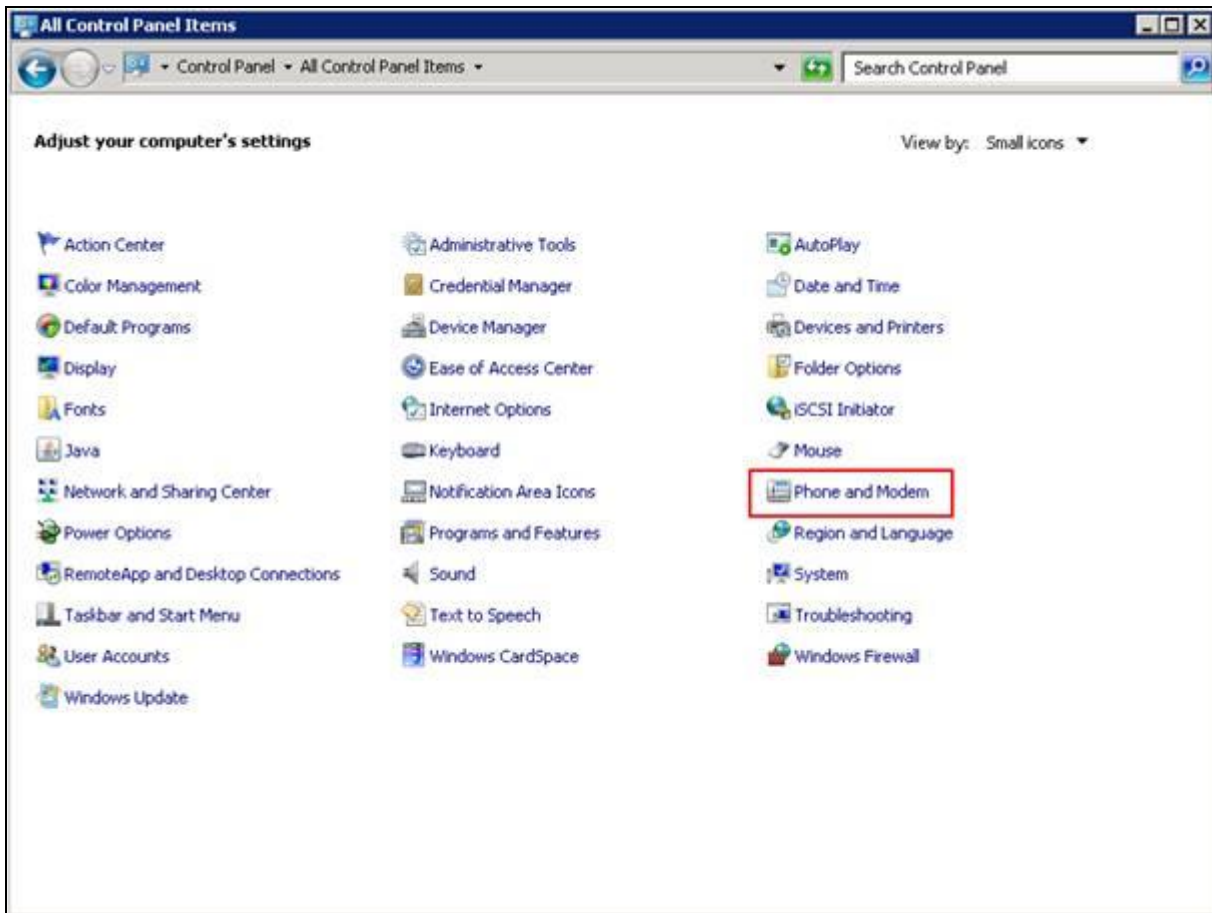
< Back Next > Cancel

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

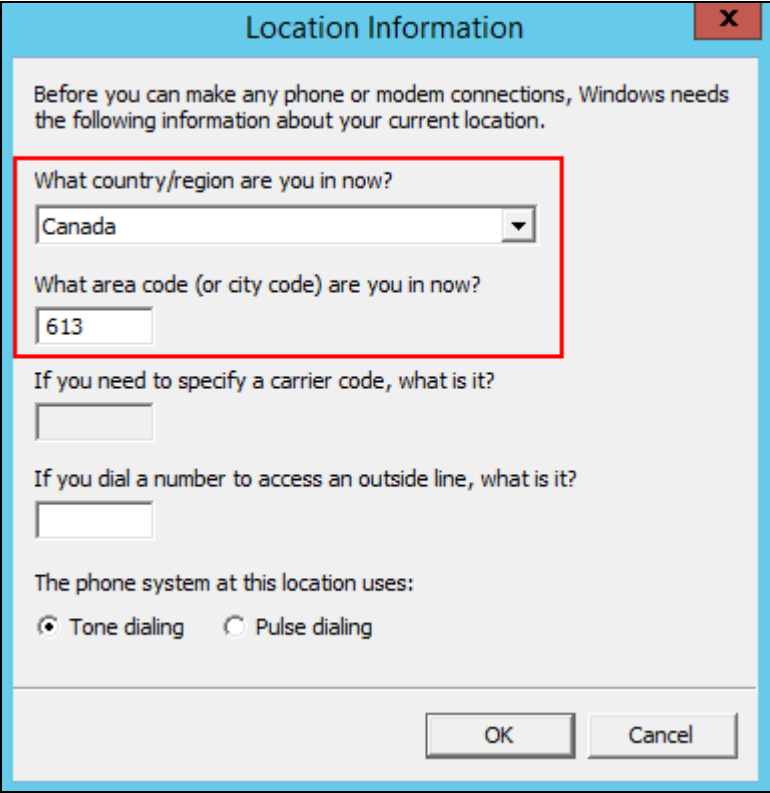




From the Windows OS Control Panel, select **Phone and Modem** as shown in the screen below.



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



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?

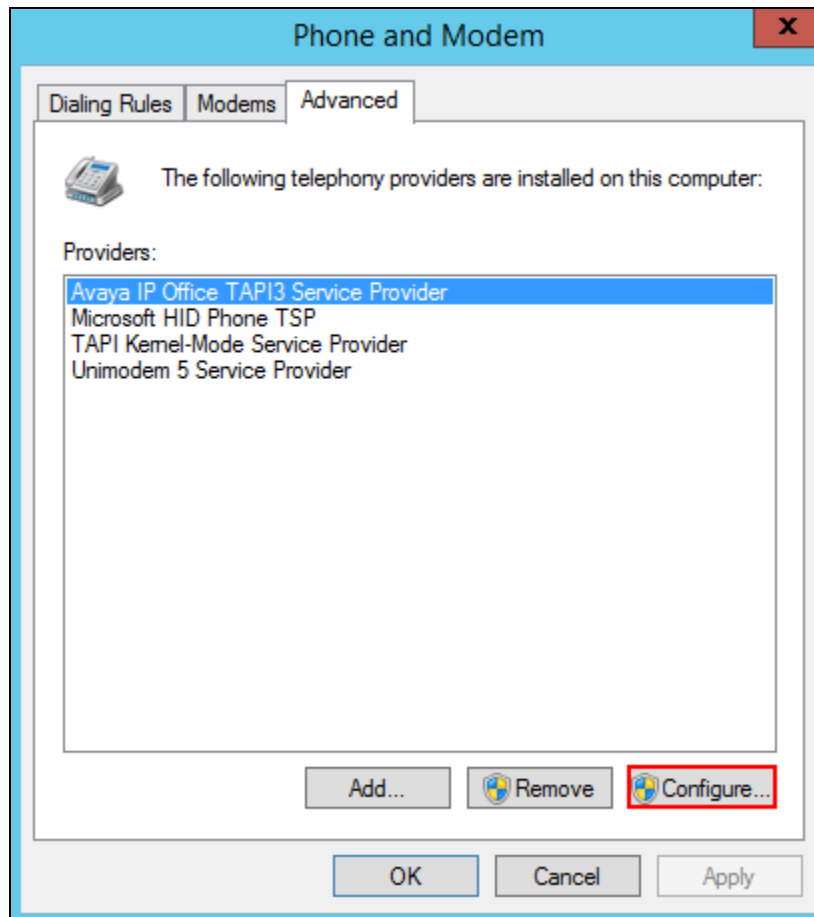
If you dial a number to access an outside line, what is it?

The phone system at this location uses:

☒ Tone dialing ☐ Pulse dialing

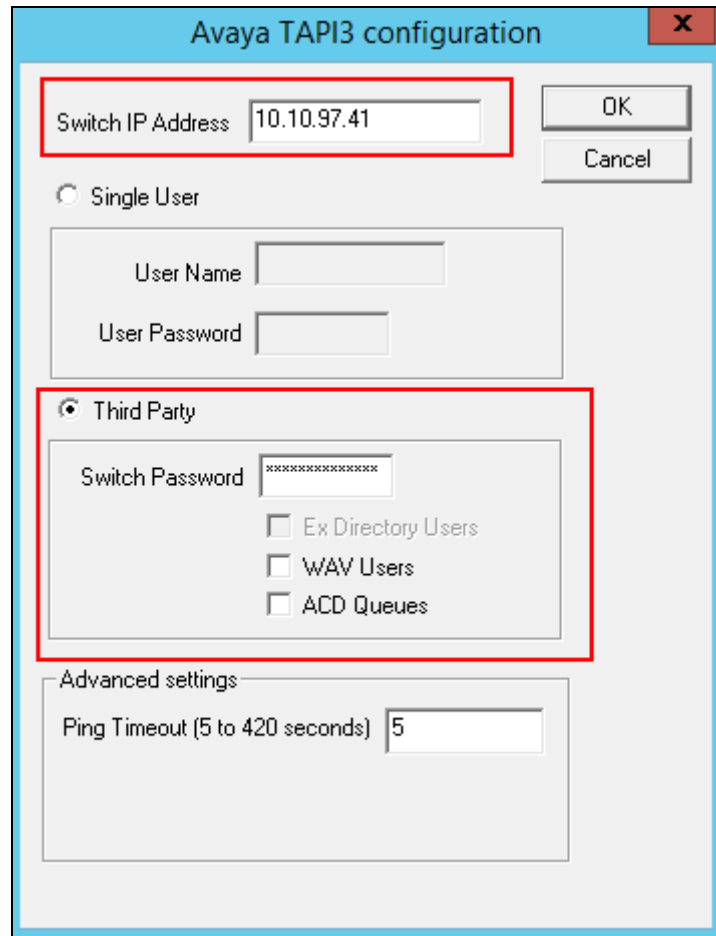
OK Cancel

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 '10.10.97.41' entered, and 'OK' and 'Cancel' buttons to its right. 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 sections. The first section contains 'User Name' and 'User Password' fields. The second section, which is highlighted with a red border, contains a 'Switch Password' field with a masked password (represented by asterisks), and three checkboxes: 'Ex Directory Users', 'WAV Users', and 'ACD Queues'. At the bottom, there is an 'Advanced settings' section with a 'Ping Timeout (5 to 420 seconds)' field set to '5'.

Avaya TAPI3 configuration

Switch IP Address 10.10.97.41

OK

Cancel

☐ Single User

User Name

User Password

☒ Third Party

Switch Password

☐ Ex Directory Users

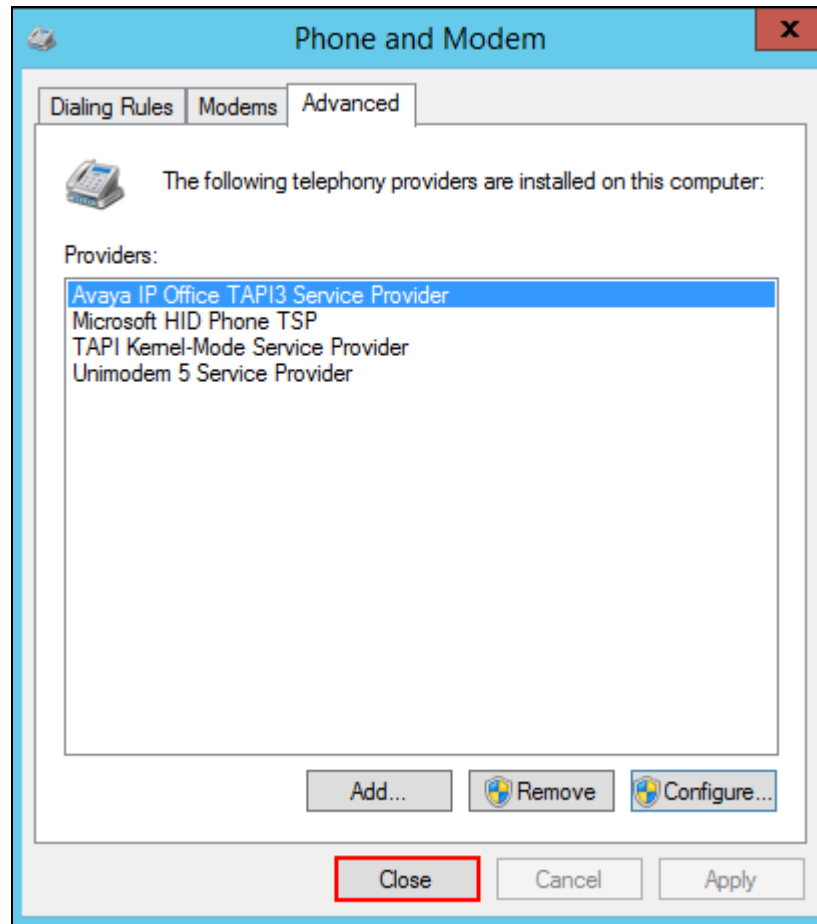
☐ WAV Users

☐ ACD Queues

Advanced settings

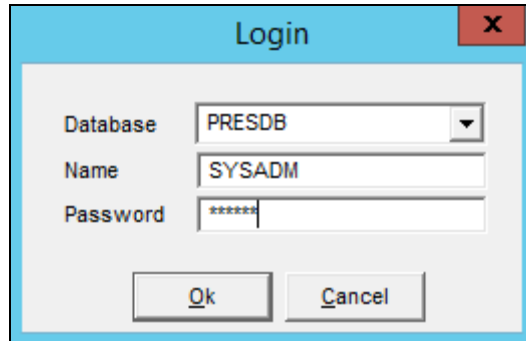
Ping Timeout (5 to 420 seconds) 5

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



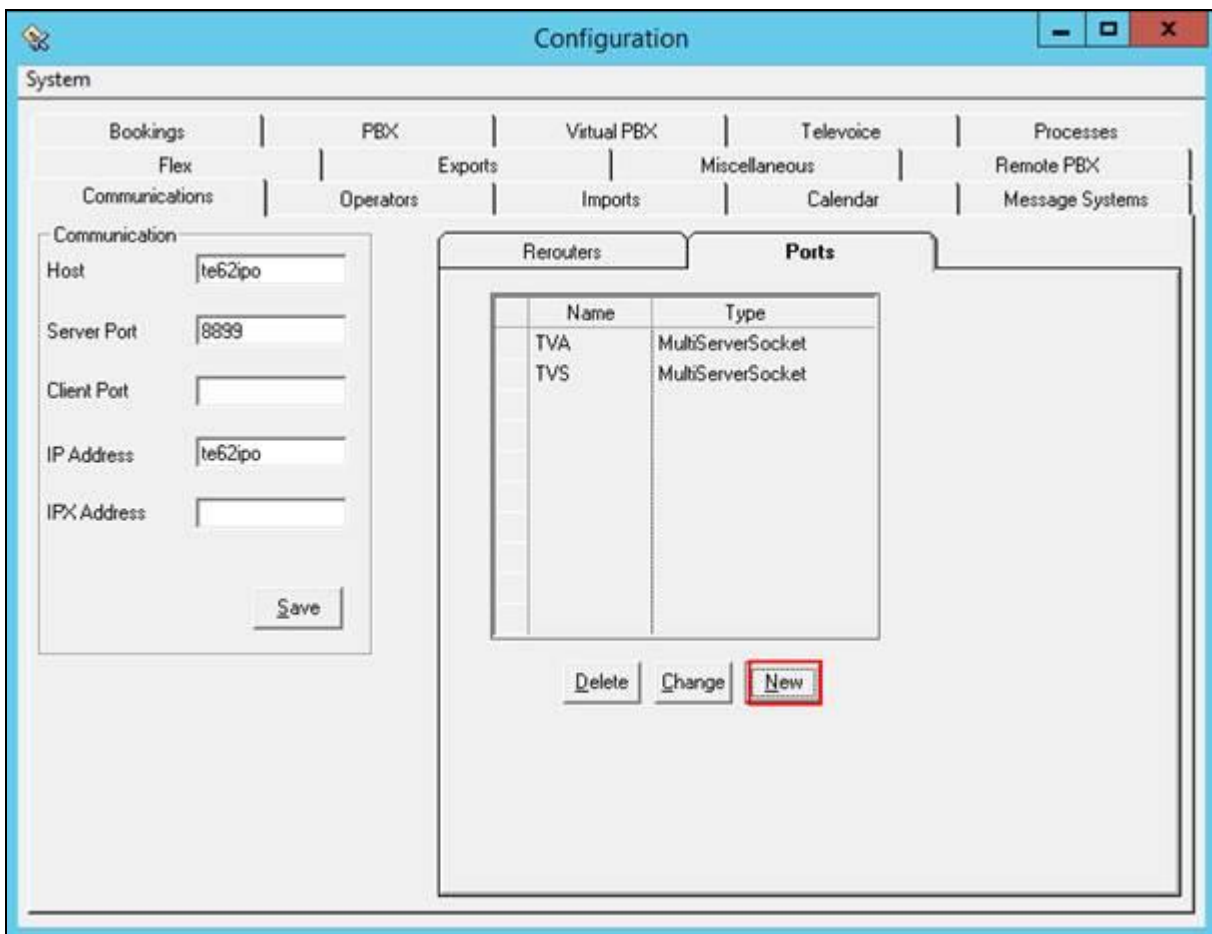
6.2.2. Configure Absence in Trio Enterprise

To configure the Absence connect; 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 the text "SYSADM", and "Password" with masked characters "*****". 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 close button (X) in the top right corner. It features a tabbed interface with the following tabs: Bookings, PBX, Virtual PBX, Televoice, Processes, Flex, Exports, Miscellaneous, Remote PBX, Communications, Operators, Imports, Calendar, and Message Systems. The "Communications" tab is selected, showing a "Communication" section with fields for Host (te62ipo), Server Port (8899), Client Port, IP Address (te62ipo), and IPX Address, along with a "Save" button. To the right, the "Ports" section is active, displaying a table with two 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 box.

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 'Port' configuration window. On the left, under 'Type', the 'Client Socket' radio button is selected. To the right, the 'Port Name' is 'Tapi (PLD)', 'Port No' is '5299', and 'IP Address' is '127.0.0.1'. The 'Address Family' dropdown is set to 'Internet (TCP/UDP)'. There is an unchecked checkbox for 'MD110 AAU' and empty fields for 'User Id' and 'Password'. The 'OK' button at the bottom is highlighted with a red rectangle.

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

The screenshot shows a window titled "Configuration" with a standard Windows-style title bar (minimize, maximize, close buttons). Below the title bar is a tab labeled "System". Under the "System" tab, there are several sub-tabs: Flex, Exports, Miscellaneous, Remote PBX, Communications, Operators, Imports, Calendar, Message Systems, Bookings, PBX, Virtual PBX, Televoice, and Processes. The "PBX" sub-tab is selected and highlighted. Below the sub-tabs is 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 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
----	------	------	------	--------	----------	----------	--------	--------	--------	-----

Buttons: Delete, Change, New

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 extensions should be forwarded to when a referral is activated. This number is configured on the Trio Enterprise server for absence treatment.

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

The screenshot shows the PBX configuration window. The 'Type' section on the left has the 'Tapi Generic' radio button selected. The configuration fields on the right include: Port (Tapi (PLD)), PbxName (Avaya IPO), Prefix (empty), Domain (empty), Extension Length (4), Net Group (checked), and Referral Destination (78003). The OK button is highlighted at the bottom left.

6.3. Configure Trio Enterprise Attendant

Trio Attendant is a separate application to Trio Enterprise server and can run concurrently on the same platform. The attendant uses a regular IP Office telephone to make and receive calls, which are directed to the phone by Trio Enterprise server. The steps to configure Trio Attendant are to launch the 'Agent Client' shortcut as shown below.



The window below opens. Enter a valid **User ID** and **Password**. Note this user ID and password is created during the installation of TRIO Enterprise Server. For **Extension**, select the IP Office telephone number that will be used as the agent's audio device (number **26112** in this example). Ensure the correct Trio Enterprise server is selected if there is more than one on the network (default is the current Trio server). Confirm **Phone type** is set to **Standard phone**. Click on the **OK** button when finished.

A screenshot of the 'Trio Agent - Login' window. The window has a title bar with 'Trio Agent - Login' and a close button. The main area features the 'Trio Enterprise®' logo at the top left, a background image of a smiling woman wearing a headset, and a key icon. Below the logo, there are several input fields and dropdown menus: 'User ID' (containing 'alt'), 'Password' (masked with dots), a 'Windows login' checkbox, 'Phone number' (containing '26112'), 'Phone type' (dropdown menu showing 'Standard phone'), 'Location' (dropdown menu showing 'Location 1'), 'Work mode' (dropdown menu showing 'Switchboard operator'), and 'Server' (dropdown menu showing 'win-gvs7gtbd3bs'). At the bottom, there are two checkboxes: 'Log in with Contact Center license (e-mail, fax, voice mail and tasks)' (unchecked) and 'Log in with Enterprise Attendant license (extended switchboard features)' (checked). At the very bottom are three buttons: 'OK', 'Guest', and 'Cancel'.

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 11**. This is the SIP Line configured in **Section 5.2** 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**.

Avaya IP Office System Status - DevCon IPO Sev1 () - IP Office Linux PC 10.0.0.2.0 build 10

AVAYA IP Office System Status

Help Snapshot LogOff Exit About

System
Alarms (12)
Extensions (6)
Trunks (5)
Line: 1
Line: 2
Line: 10
Line: 11
Line: 12
Active Calls
Resources
Voicemail
IP Networking
Locations

Status Utilization Summary Alarms

SIP Trunk Summary

Line Service State: In Service

Peer Domain Name: 10.10.98.158
Resolved Address: 10.10.98.158
Line Number: 11
Number of Administered Channels: 10
Number of Channels in Use: 0
Administered Compression: G711 Mu
Enable Faststart: Off
Silence Suppression: Off
Media Stream: RTP
Layer 4 Protocol: UDP
SIP Trunk Channel Licenses: 128
SIP Trunk Channel Licenses in Use: 0
SIP Device Features: REFER (Incoming and Outgoing)

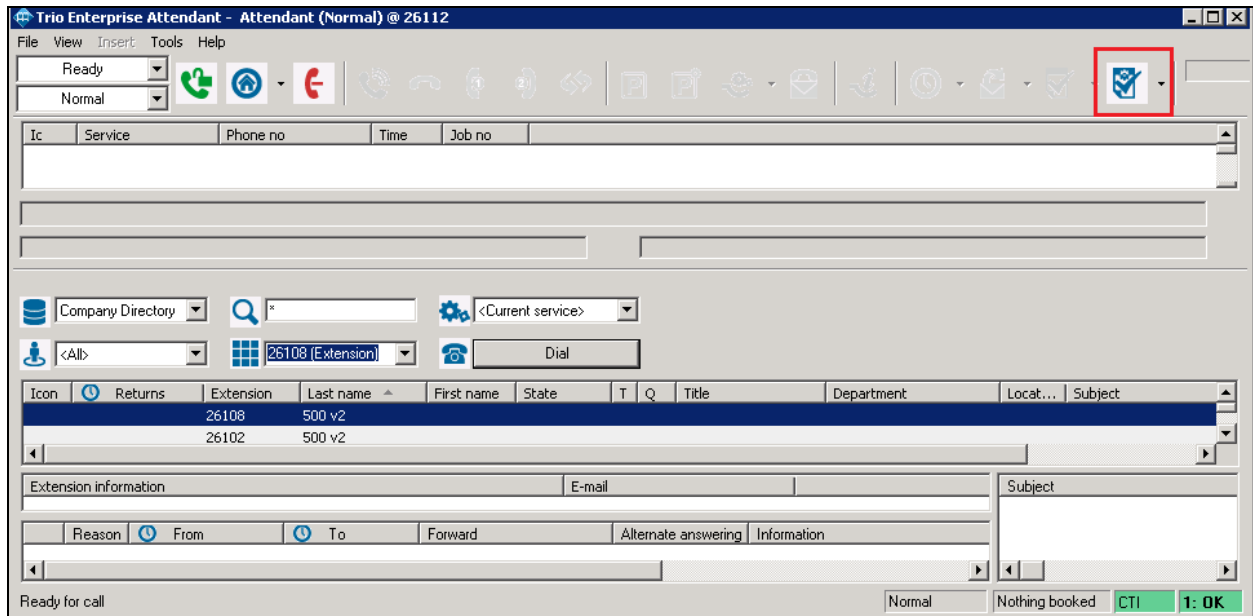
0%

Channel Number	URI	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	Round Trip Delay	Receive Jitter	Receive Packet Loss	Transmit Jitter	Transmit Packet Loss
1			Idle	4 days 15...											
2			Idle	4 days 15...											
3			Idle	4 days 15...											

Trace Trace All Pause Ping Call Details Graceful Shutdown Force Out of Service Print... Save As...

10:55:46 AM Online

To verify that Trio Enterprise is connected to IP Office, log in to the Trio Enterprise Attendant at **Start → Programs → Trio Enterprise → Contact Centre → Agent Client** (not shown) or launch the shortcut mentioned in **Section 6.3**. Complete log in with the appropriate credentials as shown in **Section 6.3**. The Trio Agent window appears as shown below. Select **Ready** from the drop down box (confirm the traffic light goes green in the small icon to the right of 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 Avaya documentation relevant to these Application Notes. The Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Deploying IP Office™ Platform Server Edition Solution*, Release 10.0.
- [2] *Administering Avaya IP Office™ Platform with Manager*, Release 10.0.
- [3] *Deploying Avaya IP Office™ Platform IP500 V2*, 15-601042 Issue 31I.

All information on the product installation and configuration TRIO Enterprise Server can be found at <http://www.trio.com>

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