



Avaya Solution & Interoperability Test Lab

Application Notes for Enghouse Interactive Communications Center 11.1 with Avaya IP Office Server Edition 11.1 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Enghouse Interactive Communications Center 11.1 to interoperate with Avaya IP Office Server Edition 11.1.

Enghouse Interactive Communications Center is a multi-channel and multi-contact solution that can handle voice, fax, web, and email contacts. The compliance testing focused on the voice integration with Avaya IP Office Server Edition using the MTCTI-3 and SIP trunk interfaces.

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

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

1. Introduction

These Application Notes describe the configuration steps required for Enghouse Interactive Communications Center (EICC) 11.1 to interoperate with Avaya IP Office Server Edition 11.1.

EICC is a multi-channel and multi-contact solution that can handle voice, fax, web, and email contacts. The compliance testing focused on the voice integration with IP Office Server Edition using the MTCTI-3 and SIP trunk interfaces.

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

The MTCTI-3 interface is used by EICC to monitor agent users on IP Office and to support agent call control and call forwarding settings. Agents are configured as users on the two IP Office systems with ACD functionality provided by EICC. Agents have desktops running the Enghouse Interactive TouchPoint client application that are used to log into EICC, set proper work modes and call forwarding, and control calls.

Incoming ACD calls are routed by IP Office over the SIP trunk to EICC. EICC determines an available agent for the call, establishes a separate audio connection over the SIP trunk with the available agent user on IP Office, and bridges the audio path of the agent connection with the customer connection at EICC. In the case that there is no available agent, EICC can provide queue treatment and configured progress announcements to the caller.

2. General Test Approach and Test Results

The feature test cases were performed both automatically and manually. Upon start of the EICC application, the application establishes MTCTI-3 connection with IP Office to obtain a list of users on both IP Office systems and to initiate users monitoring.

For the manual part of testing, incoming calls were made from the PSTN to EICC. Manual call controls from the TouchPoint client application were exercised to verify features such as answering and transferring of calls.

The serviceability test cases were performed manually by disconnecting and reconnecting the Ethernet connection to the EICC server and to the TouchPoint client.

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

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

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

For the testing associated with these Application Notes, the interfaces between IP Office and EICC included the encrypted MTCTI-3 and non-encrypted SIP trunk connections as requested by Enghouse Interactive.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing focused on verifying the following on EICC:

- Proper SIP exchanges including OPTIONS, G.711, G.722, codec negotiation, and session refresh.
- Use of MTCTI-3 capabilities to monitor users, support call control via TouchPoint client application, and set call forwarding.
- Proper handling of call scenarios including incoming ACD calls to different called numbers associated with configured queues on EICC, screen pop, hold/resume, blind/attended transfer, attended conference, announcement, call forwarding, non-ACD call, queuing, outgoing call, multiple calls, multiple agents, and long duration.

The feature testing call flows included calls within the primary IP Office system, calls within the expansion IP Office system, as well as calls between the two IP Office systems.

The serviceability testing focused on verifying the ability of EICC to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connection to the EICC server and to the TouchPoint client.

2.2. Test Results

All test cases were executed and verified. The following were observations on EICC from the compliance testing.

- In general, mixed use of agent desktop and phone to perform call control actions are supported. For the conference feature, however, all actions need to start and complete from the same source.
- By default, the FreeSWITCH SIP stack on EICC can support session refresh from 120 seconds and up.
- Special characters as part of a dial string is not supported by TouchPoint, and the workaround is to use the agent phone for such dialing.

2.3. Support

Technical support on EICC can be obtained through the following:

- **Phone:** (800) 513-2810
- **Web:** www.enghouseinteractive.com
- **Email:** usa.support@enghouse.com

3. Reference Configuration

The configuration used for the compliance testing is shown in **Figure 1**. Each IP Office system has connectivity to the PSTN for testing of cross systems PSTN scenarios.

The detailed administration of IP Office resources is not the focus of these Application Notes and will not be described. As shown in **Figure 1** below, MTCTI-3 and SIP trunk were used between the primary IP Office system and EICC.

A five-digits dial plan was used to facilitate routing with EICC. Unique extension ranges were assigned to users on the primary IP Office system (210xx), to users on the expansion IP Office system (220xx), and to EICC (2177x).

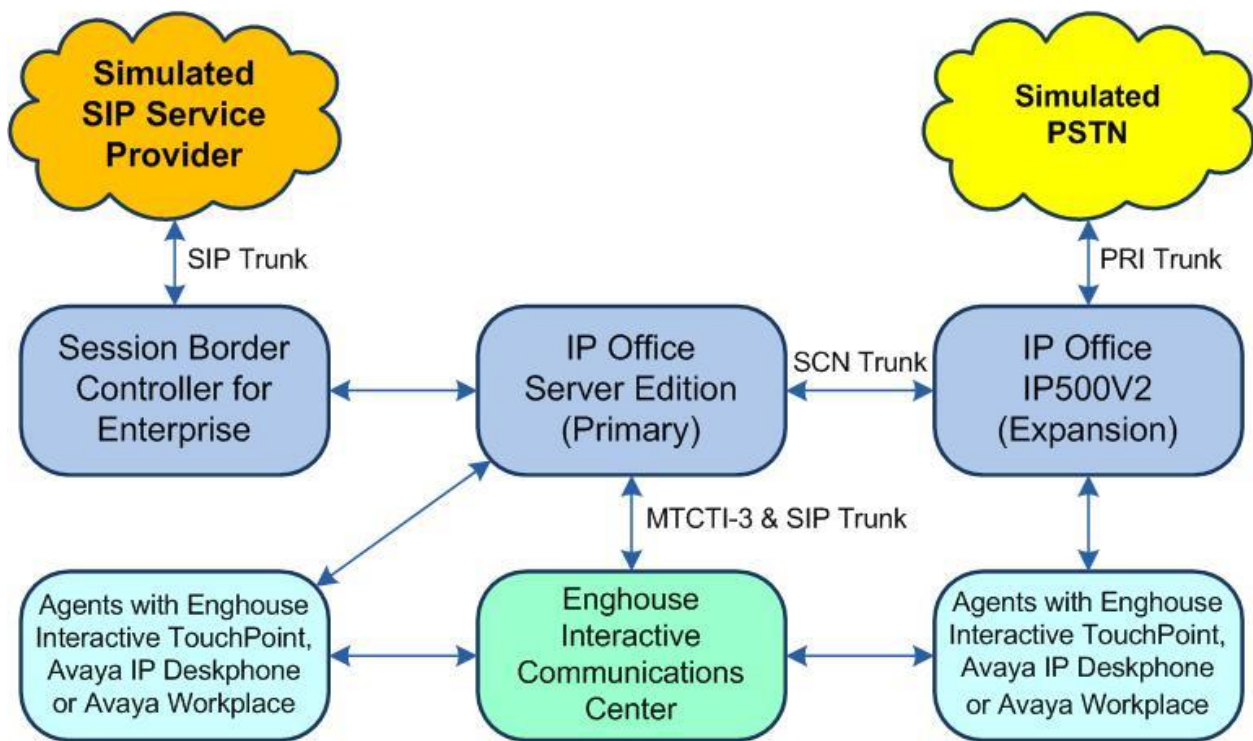


Figure 1: Compliance Testing Configuration

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office Server Edition (Primary) in Virtual Environment	11.1.0.1.0
Avaya IP Office on IP500 V2 (Expansion)	11.1.0.1.0
Avaya Workplace (SIP)	3.12.0.65.6
Avaya 1608-I IP Deskphone (H.323)	1.3120
Avaya 9611G IP Deskphone (H.323)	6.8202
Enghouse Interactive Communications Center on Windows Server 2019 <ul style="list-style-type: none">Avaya MTCTI-3 API (ipo_mtcti3.proto)	11.1.0.16117 Standard 1.0
Enghouse Interactive TouchPoint on Windows 10 Pro	11.1.0.16117

Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with IP Office Server Edition in all configurations.

5. Configure Avaya IP Office

This section provides the procedures for configuring the IP Office systems. The procedures include the following area:

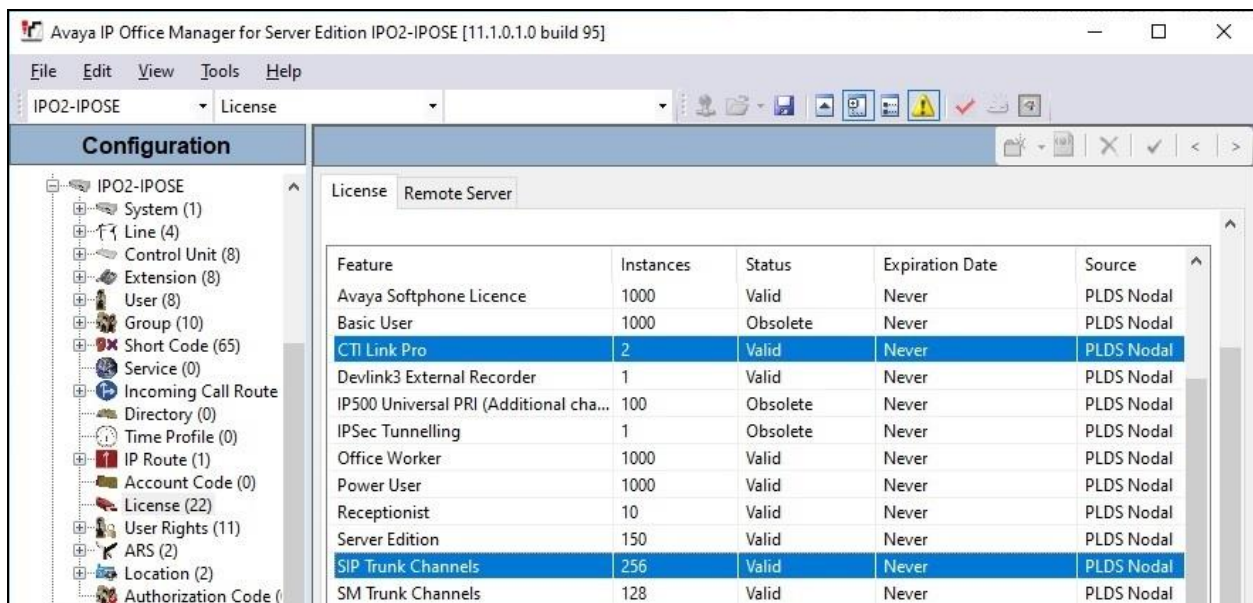
- Verify license
- Administer system
- Administer line
- Administer incoming call route
- Administer short codes
- Administer users
- Administer rights group
- Administer service user

5.1. Verify License

From a PC running the IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Select the proper primary IP Office system, and log in using the appropriate credentials.

The **Avaya IP Office Manager for Server Edition IPO2-IPOSE** screen is displayed, where **IPO2-IPOSE** is the name of the primary IP Office system.

From the configuration tree in the left pane, select **License** under the primary IP Office system, in this case “IPO2-IPOSE”, and a list of licenses is displayed in the right pane. Verify that there are sufficient licenses for **CTI Link Pro**, in this case two licenses are required with two IP Office systems. Also verify that there are sufficient licenses for **SIP Trunk Channels** as shown below.

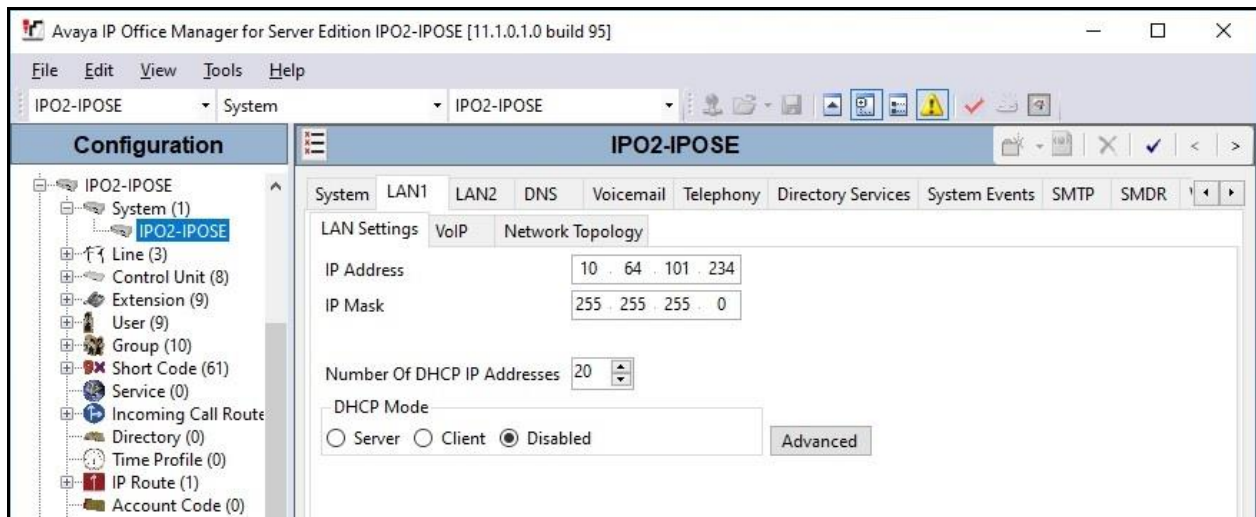


Feature	Instances	Status	Expiration Date	Source
Avaya Softphone Licence	1000	Valid	Never	PLDS Nodal
Basic User	1000	Obsolete	Never	PLDS Nodal
CTI Link Pro	2	Valid	Never	PLDS Nodal
Devlink3 External Recorder	1	Valid	Never	PLDS Nodal
IP500 Universal PRI (Additional cha...	100	Obsolete	Never	PLDS Nodal
IPSec Tunnelling	1	Obsolete	Never	PLDS Nodal
Office Worker	1000	Valid	Never	PLDS Nodal
Power User	1000	Valid	Never	PLDS Nodal
Receptionist	10	Valid	Never	PLDS Nodal
Server Edition	150	Valid	Never	PLDS Nodal
SIP Trunk Channels	256	Valid	Never	PLDS Nodal
SM Trunk Channels	128	Valid	Never	PLDS Nodal

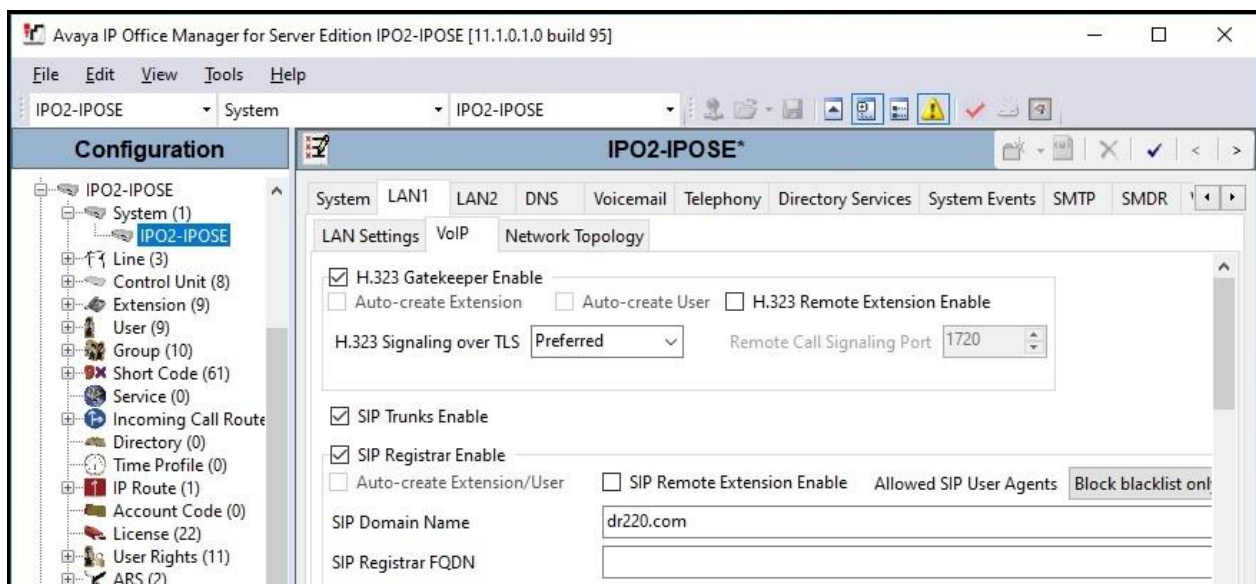
5.2. Administer System

From the configuration tree in the left pane, select **System** under the IP Office system used for SIP trunk connection with EICC, to display the system screen in the right pane.

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 to configure EICC.

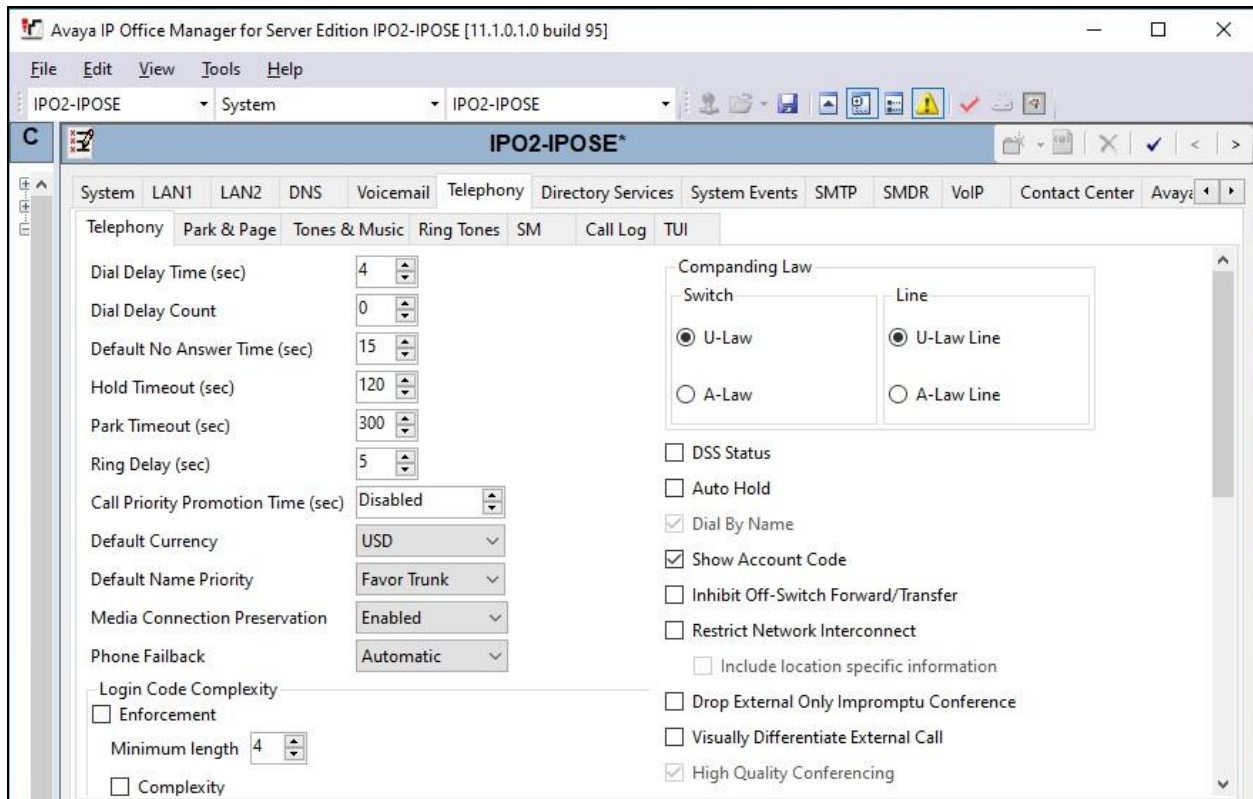


Select the **VoIP** sub-tab. Make certain that **SIP Trunks Enable** is checked, as shown below. Retain the default values in the remaining fields.



Select the **Telephony** tab, followed by the **Telephony** sub-tab in the right pane. Uncheck **Inhibit Off-Switch Forward/Transfer** if transfer from EICC to PSTN destinations is desired. In the compliance testing, this parameter was disabled.

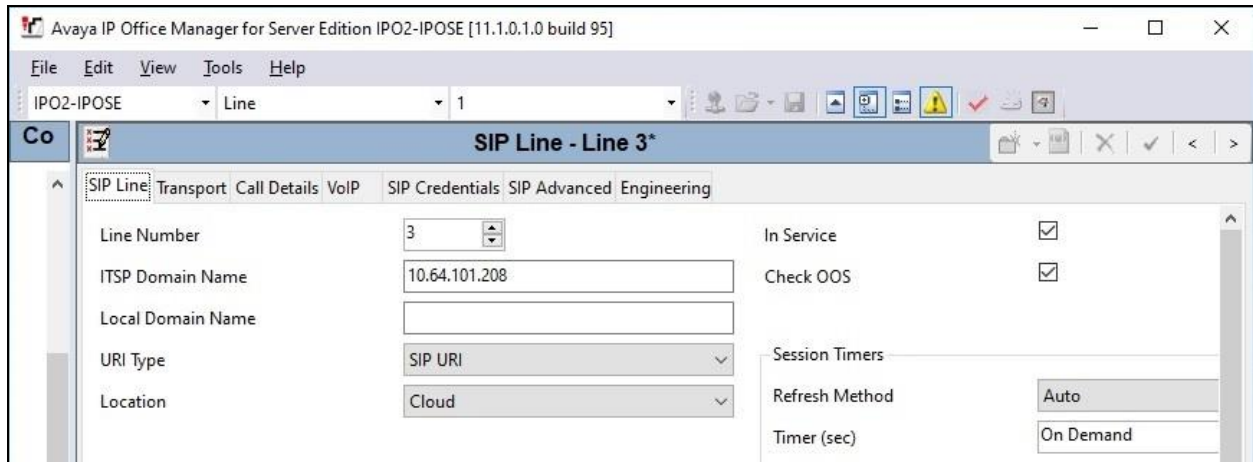
Uncheck **Drop External Only Impromptu Conference**, this setting allows IP Office to not end a conference when there are external parties on the call after the last internal party drops.



5.3. Administer Line

From the configuration tree in the left pane, right-click **Line** under the IP Office system used for SIP trunk connection with EICC, and select **New** → **SIP Line** from the pop-up list to add a new SIP line (not shown).

Select the **SIP Line** tab. For **ITSP Domain Name**, enter the IP address of the EICC server. Retain the defaults in the remaining fields.



Avaya IP Office Manager for Server Edition IPO2-IPOSE [11.1.0.1.0 build 95]

File Edit View Tools Help

IPO2-IPOSE Line 1

SIP Line - Line 3*

SIP Line Transport Call Details VoIP SIP Credentials SIP Advanced Engineering

Line Number: 3 In Service: ☒

ITSP Domain Name: 10.64.101.208 Check OOS: ☒

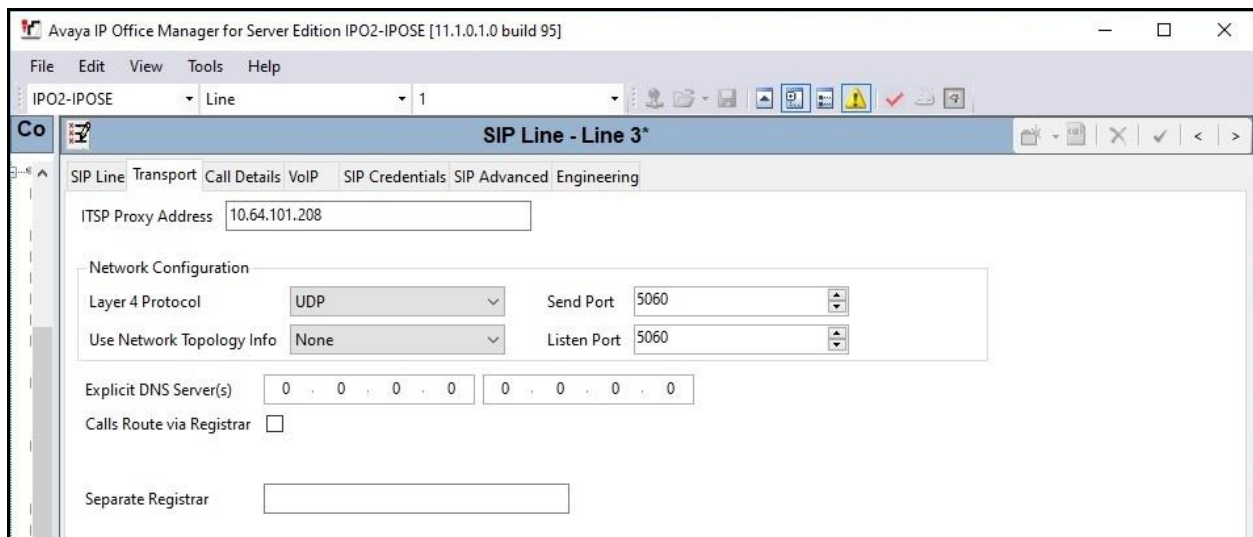
Local Domain Name:

URI Type: SIP URI Session Timers:

Location: Cloud Refresh Method: Auto

Timer (sec): On Demand

Select the **Transport** tab. For **ITSP Proxy Address**, enter the IP address of the EICC server. Retain the default values in the remaining fields. Note that EICC can support TLS, UDP and TCP, and the compliance testing used the UDP protocol.



Avaya IP Office Manager for Server Edition IPO2-IPOSE [11.1.0.1.0 build 95]

File Edit View Tools Help

IPO2-IPOSE Line 1

SIP Line - Line 3*

SIP Line Transport Call Details VoIP SIP Credentials SIP Advanced Engineering

ITSP Proxy Address: 10.64.101.208

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

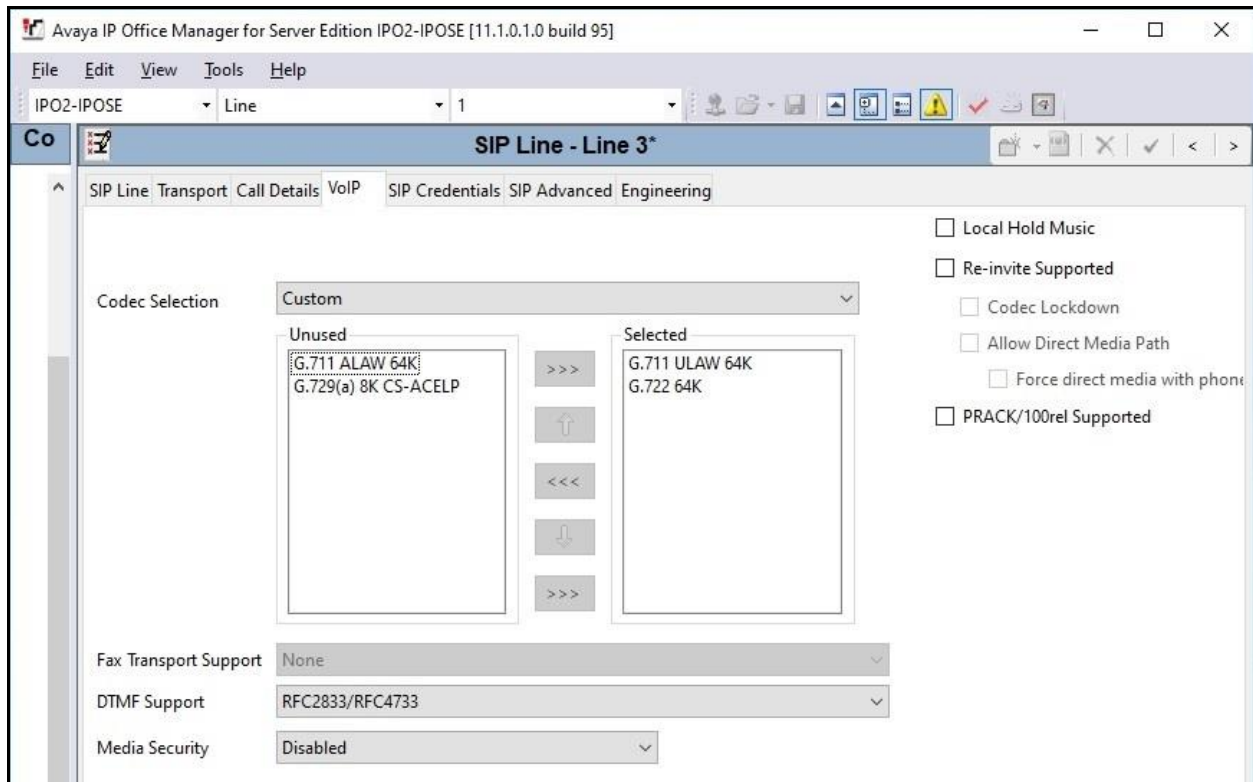
Calls Route via Registrar: ☐

Separate Registrar:

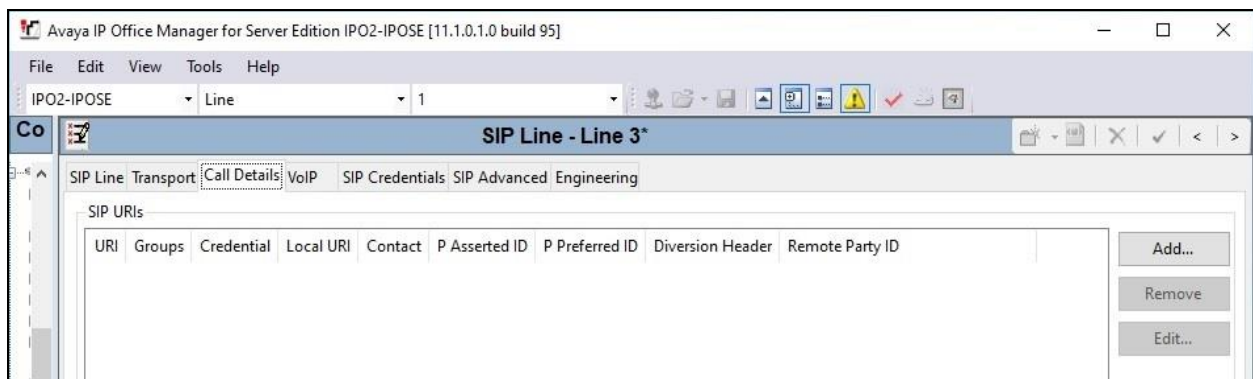
Select the **VoIP** tab. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Codec Selection:** “Custom”
- **Selected:** Retain the relevant G.711 codec variant and “G.722 64K”.
- **Media Security:** “Disabled”

Note that EICC only supports G.711 and G.722 codecs, and does not support media shuffling in the current release.



Select the **Call Details** tab, followed by **Add** in the **SIP URIs** sub-section. Create two SIP URIs, one to match incoming calls from EICC and another to match outgoing calls to EICC.



5.3.1. SIP URI for Incoming Calls from EICC

Create a SIP URI for matching of incoming calls from EICC. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Incoming Group:** An available incoming group number that will be used to match.
- **Outgoing Group:** An available outgoing group number that will not be used.
- **Max Sessions:** The desired number of channels.

Note that the **Outgoing Group** number associated with this SIP URI will not be used but is required to be configured.

SIP Line - 3 | Call Details | SIP URI

New URI

Incoming Group: 3 Max Sessions: 5

Outgoing Group: 93

Credentials: 0: <None>

	Display	Content
Local URI	Auto	Auto
Contact	Auto	Auto
P Asserted ID	<input type="checkbox"/> None	None
P Preferred ID	<input type="checkbox"/> None	None
Diversion Header	<input type="checkbox"/> None	None
Remote Party ID	<input type="checkbox"/> None	None

Field meaning

Outgoing Calls	Forwarding/Twinning	Incoming Calls
Caller	Original Caller	Called
Caller	Original Caller	Called
None	None	None
None	None	None
None	None	None
None	None	None

OK Cancel Help

5.3.2. SIP URI for Outgoing Calls to EICC

Create a SIP URI for matching of outgoing calls to EICC. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Incoming Group:** An available incoming group number that will not be used.
- **Outgoing Group:** An available outgoing group number that will be used to match.
- **Max Sessions:** The desired number of channels.
- **Local URI Display:** “Use Internal Data”
- **Local URI Content:** “Use Internal Data”
- **Contact Display:** “Use Internal Data”
- **Contact Content:** “Use Internal Data”
- **P Asserted ID:** Check this parameter.

Note that the **Incoming Group** number associated with this SIP URI will not be used but is required to be configured.

Display		Content		Field meaning		
				Outgoing Calls	Forwarding/Twinning	Incoming Calls
Local URI	Use Internal Data	Use Internal Data		Caller	Original Caller	Called
Contact	Use Internal Data	Use Internal Data		Caller	Original Caller	Called
P Asserted ID	<input checked="" type="checkbox"/> Auto	Auto		Caller	Original Caller	Called
P Preferred ID	<input type="checkbox"/> None	None		None	None	None
Diversion Header	<input type="checkbox"/> None	None		None	None	None
Remote Party ID	<input type="checkbox"/> None	None		None	None	None

The screen below shows the two SIP URIs that were created in the compliance testing.

URI	Groups	Credential	Local URI	Contact	P Asserted ID	P Preferred ID	Diversion Header	Remote Party ID
1	3 93	0: <None>	Auto	Auto				
2	93 3	0: <None>	Use Internal Data	Use Internal Data	Auto			

5.4. Administer Incoming Call Route

From the configuration tree in the left pane, right-click on **Incoming Call Route** under the IP Office system used for SIP trunk connection with EICC and select **New** from the pop-up list to add a new route for incoming calls from EICC.

For **Incoming Number**, enter “*”. For **Line Group ID**, select the incoming group number from **Section 5.3.1**, in this case “3”. Retain the default value in the remaining fields.

The screenshot shows the Avaya IP Office Manager for Server Edition interface. The left pane displays the configuration tree with 'Incoming Call Route' selected. The right pane shows the configuration fields for the selected route. The 'Standard' tab is active, and the 'Line Group ID' is set to '3'. The 'Incoming Number' is set to '*'. The 'Incoming Sub Address' and 'Incoming CLI' fields are empty. The 'Locale' is set to '1 - Low'. The 'Priority' is set to '1 - Low'. The 'Tag' field is empty. The 'Hold Music Source' is set to 'System Source'. The 'Ring Tone Override' is set to 'None'.

Field	Value
Bearer Capability	Any Voice
Line Group ID	3
Incoming Number	*
Incoming Sub Address	
Incoming CLI	
Locale	1 - Low
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

Select the **Destinations** tab. For **Destination**, enter “.” to match any dialed number from EICC.

The screenshot shows the Avaya IP Office Manager for Server Edition interface. The left pane displays the configuration tree with 'Incoming Call Route' selected. The right pane shows the configuration fields for the selected route. The 'Destinations' tab is active, and the 'Destination' is set to '.'. The 'Fallback Extension' is set to 'Default Value'.

TimeProfile	Destination	Fallback Extension
Default Value	.	Default Value

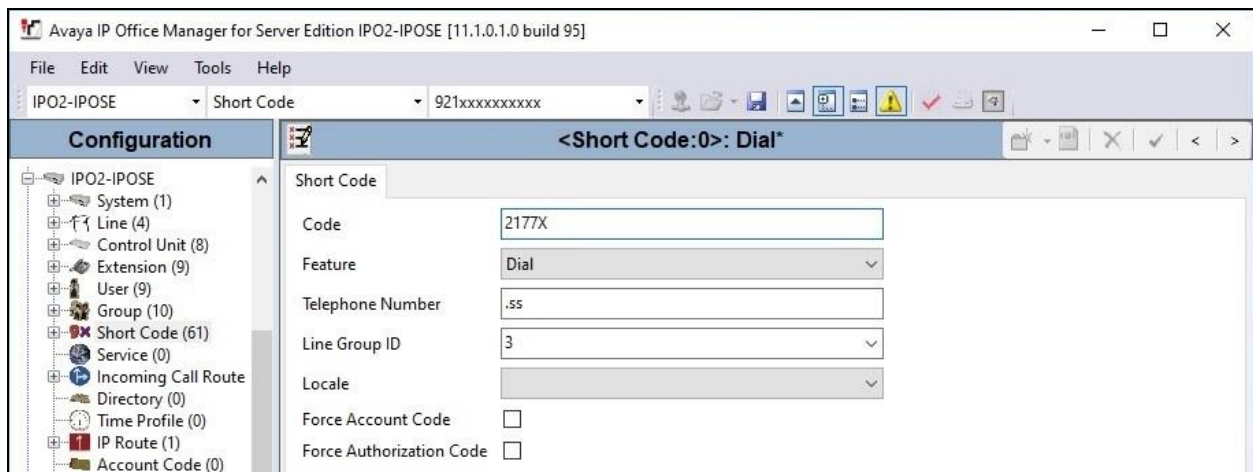
5.5. Administer Short Codes

Configure a set of short codes for routing of outgoing calls to EICC and for routing of incoming calls from EICC to the PSTN.

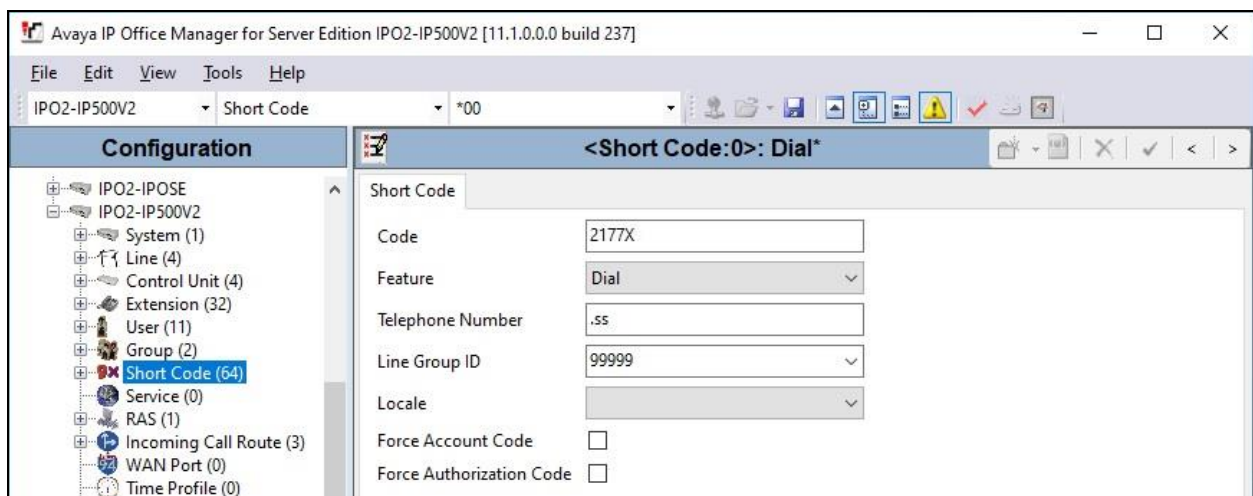
5.5.1. Short Codes for Outgoing Calls to EICC

From the configuration tree in the left pane, right-click on **Short Code** under the primary IP Office system and select **New** from the pop-up list to add a new short code for outgoing calls to EICC. In the compliance testing, all calls to 2177X are routed over the SIP trunk to EICC.

For **Code**, enter the appropriate value, in this case “2177X”. For **Telephone Number**, enter “.ss” to pass the calling number as is to EICC. For **Line Group ID**, select the outgoing group number from **Section 5.3.2**. Retain the default values in the remaining fields.



Repeat this section to add similar short code for the expansion IP Office system, which is named **IPO2-IP500V2** in this case. For **Line Group ID**, select the applicable outgoing group ID for the SCN trunk that connects to the primary IP Office system, in this case “99999” as shown below.



5.5.2. Short Codes for Incoming Calls from EICC to PSTN

From the configuration tree in the left pane, right-click on **Short Code** under the primary IP Office system and select **New** from the pop-up list to add a new short code for incoming calls from EICC to the PSTN. In the compliance testing, EICC will use the E.164 format to add prefix “+1” followed by the ten-digits number for outbound domestic calls to the PSTN, and therefore a new short code is created for routing of such calls.

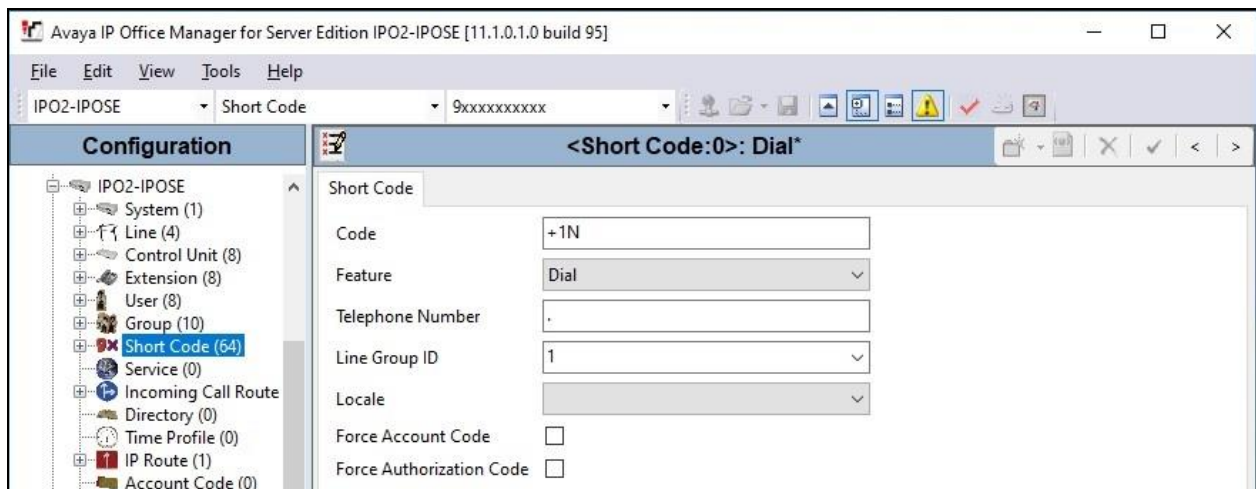
For **Code**, enter the appropriate value, in this case “+1N”. Note that this is only a sample and the actual code value may differ depending on the customer dial plan.

For **Telephone Number**, enter a pertinent value for the customer configuration. In the compliance testing, IP Office was already using E.164 format, therefore “.” was used to retain the same format.

For **Line Group ID**, enter the outgoing group number for the PSTN line, in this case “1”.

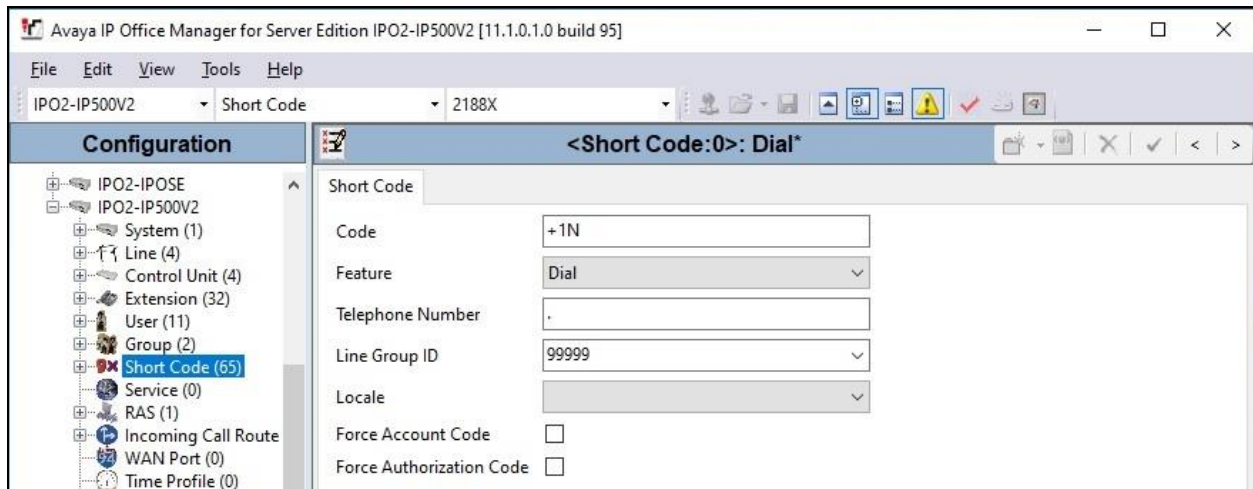
Retain the default values in the remaining fields.

Create additional short code as necessary for routing of international calls from EICC to the PSTN if applicable.



Repeat this section to add similar short code(s) for the expansion IP Office system, which is named **IPO2-IP500V2** in this case. For **Line Group ID**, select the applicable outgoing group ID for the SCN trunk that connects to the primary IP Office system, in this case “99999”.

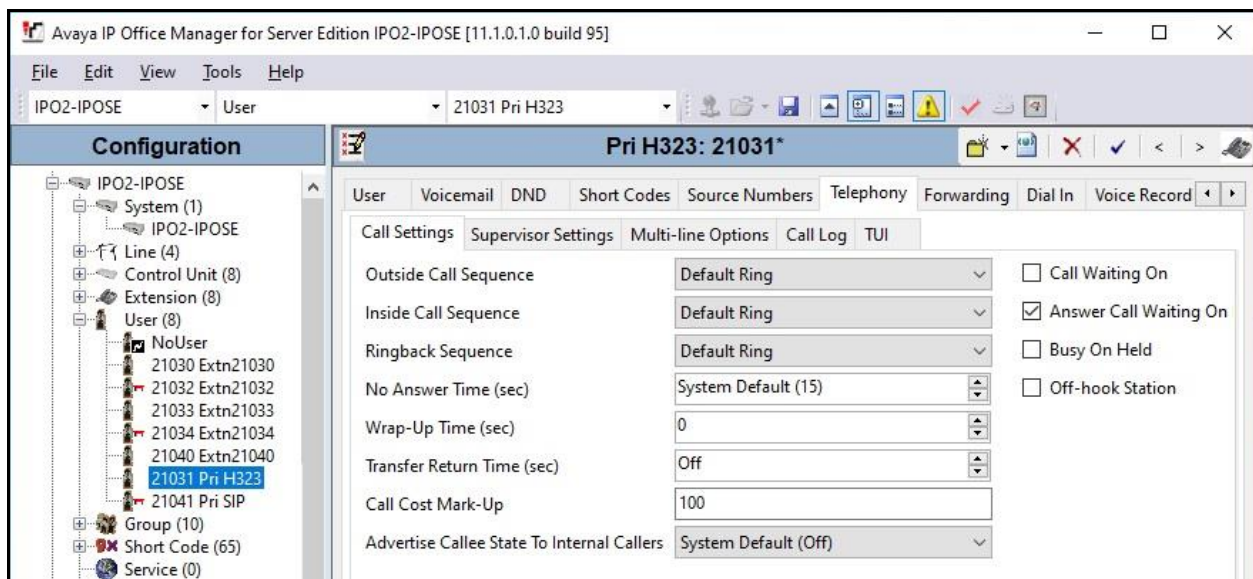
In the compliance testing, one short code was created on the expansion IP Office system as shown below.



5.6. Administer Users

From the configuration tree in the left pane, expand **User** and select the first user that will be used by EICC as an agent, in this case “20031 Pri H323”. Select the **Telephony** tab, followed by the **Call Settings** sub-tab. Set **Wrap-Up Time (sec)** to “0” as shown below.

Repeat this section for all users on both IP Office systems that will be used by EICC as agents. In the compliance testing, two users on each IP Office system were configured.



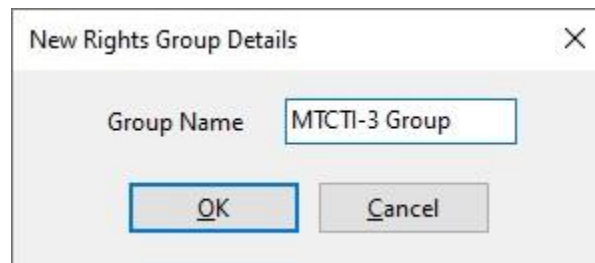
5.7. Administer Rights Group

From the configuration tree in the left pane, select **File → Advanced → Security Settings** from the top menu. The **Avaya IP Office Manager for Server Edition – Security Administration** screen is displayed.

Select and right-click on **Rights Groups** in the left pane, followed by **New** from the pop-up list to add a new rights group for MTCTI-3.

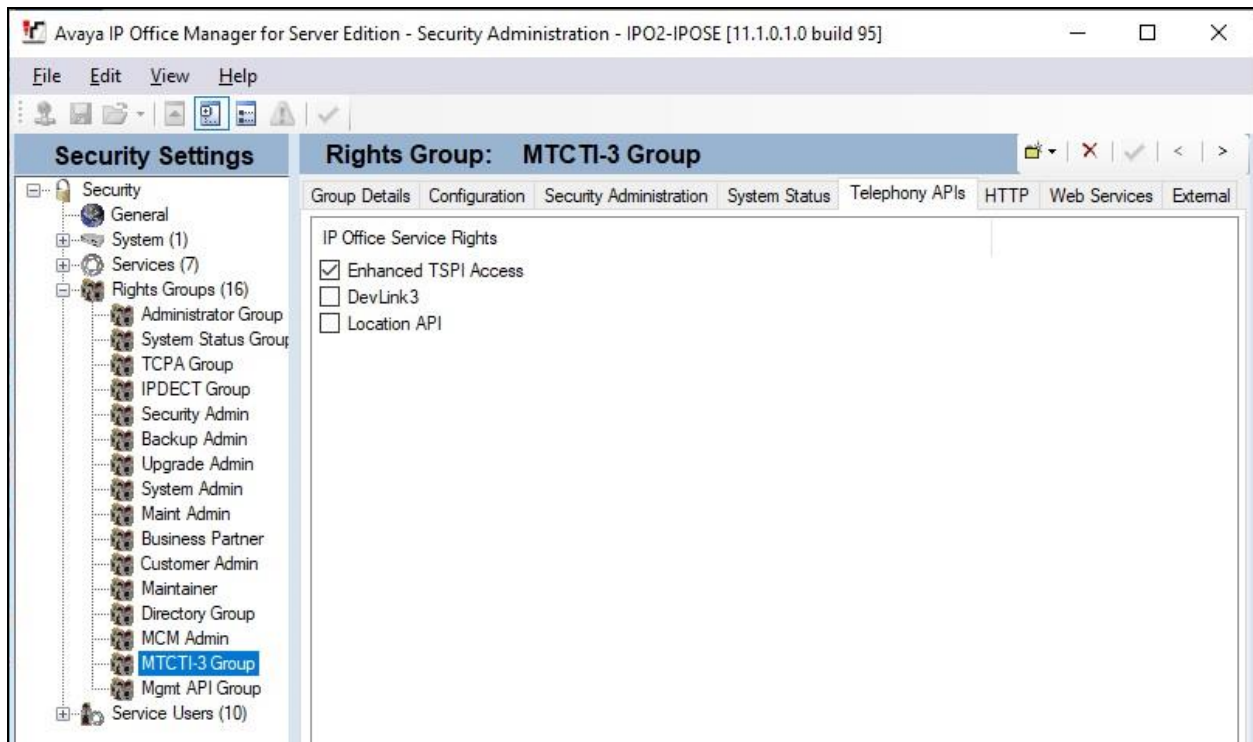


The **New Rights Group Details** dialog box is displayed. For **Group Name**, enter a descriptive name, as shown below.



The screen is updated with **Rights Group: MTCTI-3 Group** shown in the right pane, where **MTCTI-3 Group** is the name of the newly created rights group.

Select the **Telephony APIs** tab in the right pane, and check **Enhanced TSPI Access**, as shown below.



5.8. Administer Service User

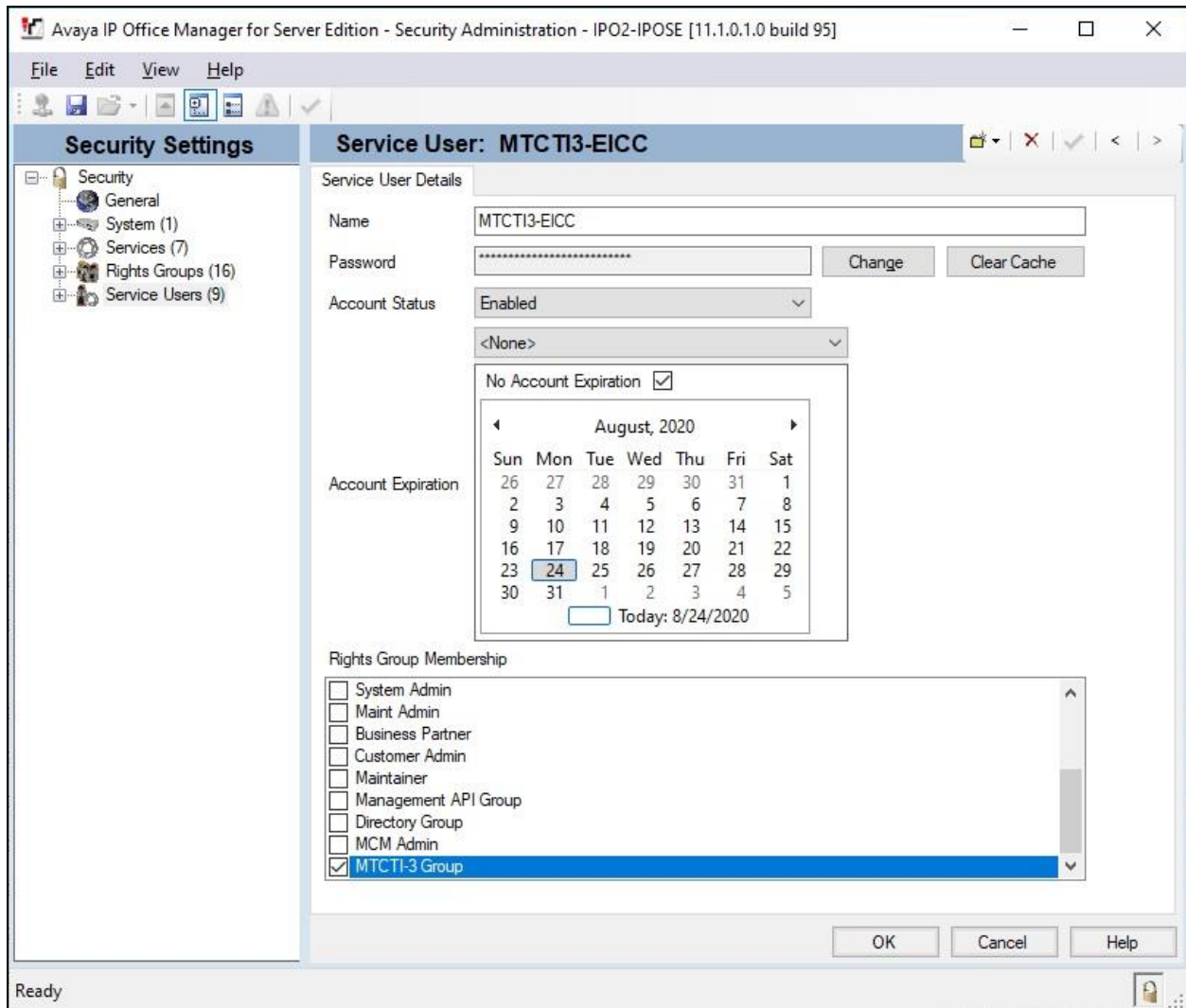
From the **Avaya IP Office Manager for Server Edition – Security Administration** screen shown in **Section 5.7**, select and right-click on **Service Users** in the left pane, followed by **New** from the pop-up list to add a new service user for MTCTI-3.

The **New Service User Details** dialog box is displayed. Enter desired name and password, as shown below.

The 'New Service User Details' dialog box is shown. It has a title bar with the text 'New Service User Details'. Inside, there are three text input fields. The first is labeled 'New User Name' and contains the text 'MTCTI3-EICC'. The second is labeled 'New User Password' and contains a series of dots. The third is labeled 'Re-enter New User Password' and also contains a series of dots. At the bottom of the dialog, there are two buttons: 'OK' and 'Cancel'.

The screen is updated with **Service User: MTCTI3-EICC** shown in the right pane, where **MTCTI3-EICC** is the name of the newly created service user from above.

Scroll the **Rights Group Membership** in the bottom right pane as necessary and check the newly added rights groups from **Section 5.7**, as shown below.



6. Configure Enghouse Interactive Communications Center

This section provides the procedures for configuring EICC. The procedures include the following areas:

- Administer phone system type
- Administer SIP PBX
- Administer services
- Verify licenses
- Administer queues
- Administer agent login class
- Administer agents

The configuration of EICC is typically performed by Enghouse Interactive installation technicians or third-party resellers. The procedural steps are presented in these Application Notes for informational purposes.

6.1. Administer Phone System Type

As part of the installation wizard, the **Phone System Type** screen below will be displayed. Set **PBX Type** to “SIP PBX” as shown and continue to complete the installation.



Phone System Type

Define Phone System Type

Select the type of Phone PBX to be installed. The PBX will be added to the database and Application Manager will launch afterwards for further configuration steps.

This step must be completed before beginning the Configure Phone System Data step.

PBX Type:

6.2. Administer SIP PBX

From the EICC server, launch the web interface by using the URL **`http://localhost:9090/ccadmin/system/sippbx/configuration`** in an Internet browser window.

The screen below is displayed. Log in using the administrator credentials.



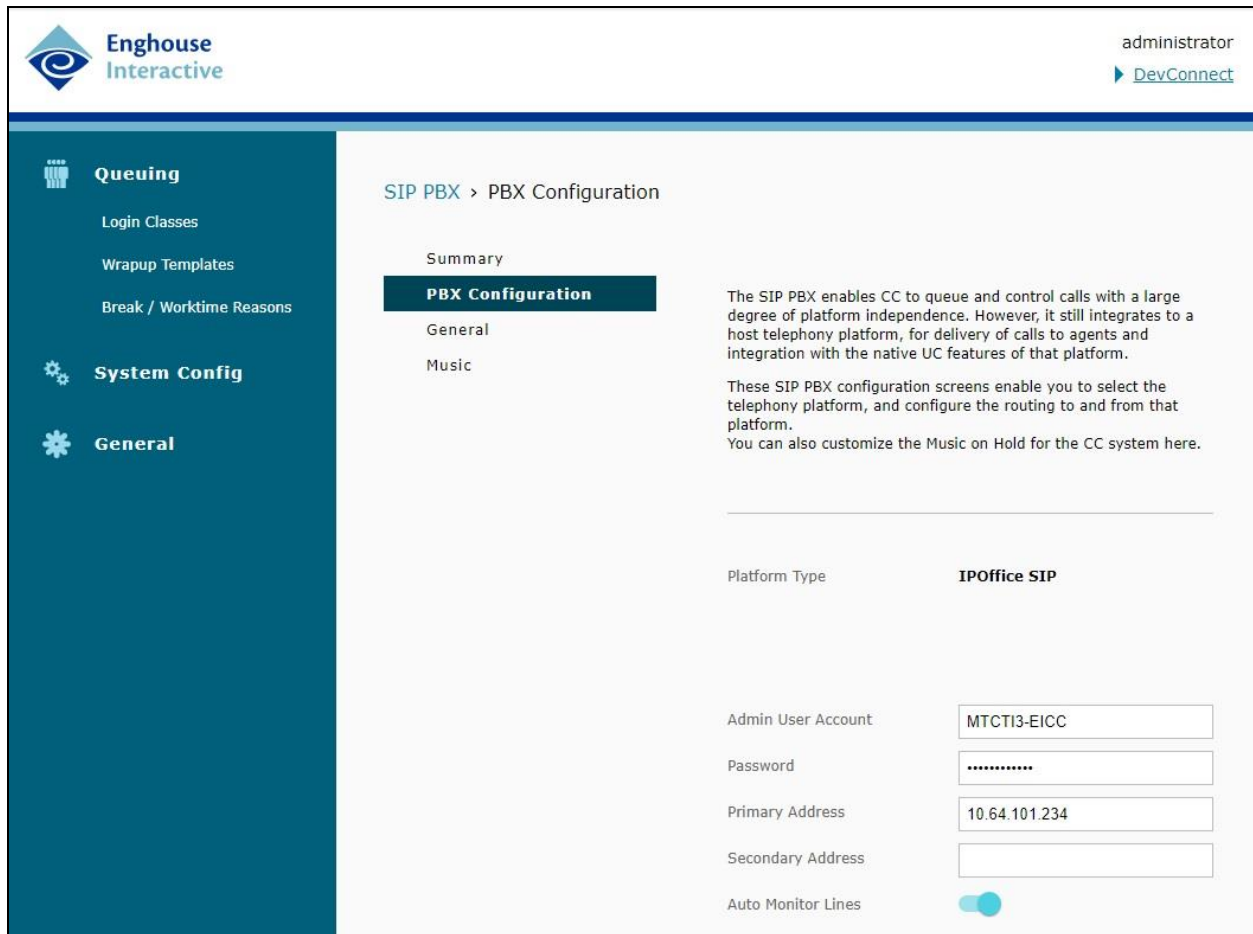
v11.1.0.14805 CU; API v11.1.0.14805; WS v11.1.0.14805

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The screen below is displayed. Select **PBX Configuration** from the right pane.

Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Admin User Account:** The service user credentials from **Section 5.8**.
- **Password:** The service user credentials from **Section 5.8**.
- **Primary Address:** IP address of the primary IP Office system from **Section 5.2**.



Enghouse Interactive administrator
DevConnect

SIP PBX > PBX Configuration

Summary

PBX Configuration

General

Music

The SIP PBX enables CC to queue and control calls with a large degree of platform independence. However, it still integrates to a host telephony platform, for delivery of calls to agents and integration with the native UC features of that platform.

These SIP PBX configuration screens enable you to select the telephony platform, and configure the routing to and from that platform.

You can also customize the Music on Hold for the CC system here.

Platform Type **IPOffice SIP**

Admin User Account

Password

Primary Address

Secondary Address

Auto Monitor Lines ☒

Select **General** from the right pane.

For **External Trunk One**, enter the IP address of the primary IP Office system from **Section 5.2**.

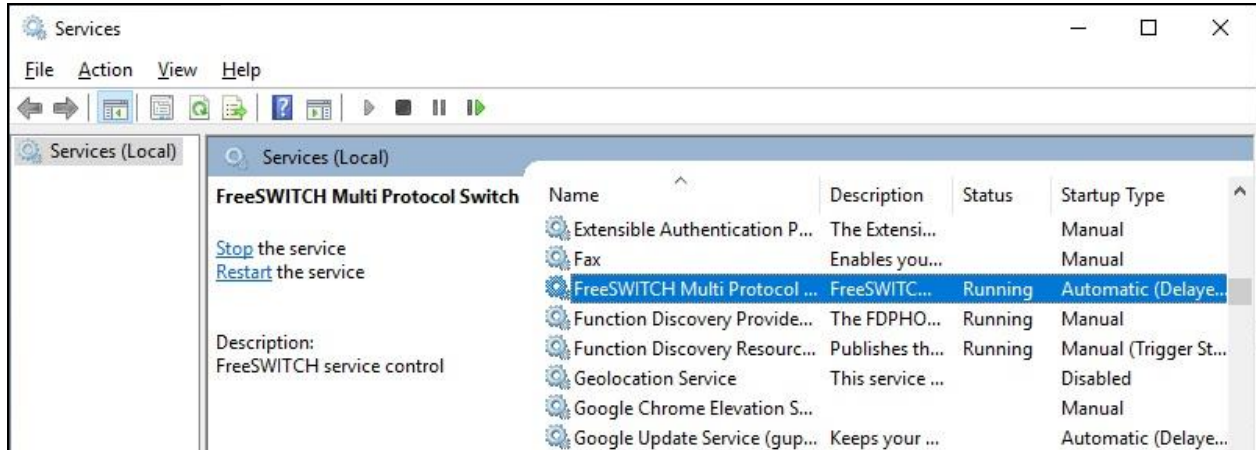
Retain the default values in the remaining fields.

The screenshot shows the Enghouse Interactive web interface. The top header includes the Enghouse Interactive logo and the user 'administrator' with a 'DevConnect' link. The left sidebar contains navigation options: 'Queuing' (with sub-items 'Login Classes', 'Wrapup Templates', and 'Break / Worktime Reasons'), 'System Config' (with a gear icon), and 'General' (with a flower icon). The main content area is titled 'SIP PBX > General'. It features a 'Summary' section with a 'PBX Configuration' sub-section containing 'General' (highlighted) and 'Music'. A descriptive text block states: 'An incoming call has characteristics that are recognized by the SIP Queuing Controller and managed accordingly. Use this screen to configure system responses to exceptions that may prevent a call from proceeding to a queue.' Below this is a configuration table with the following fields and values:

Queue Timeout	4000
Failover	<input checked="" type="checkbox"/> Return 480 - Temporarily una...
Failover Destination	
External Trunk One	10.64.101.234
External Trunk Two	
External Trunk Three	
External Trunk Four	
Tone Pattern	UNITED STATES
SIP Transport Protocol	UDP
Allow external voice connectivity	<input type="checkbox"/>

6.3. Administer Services

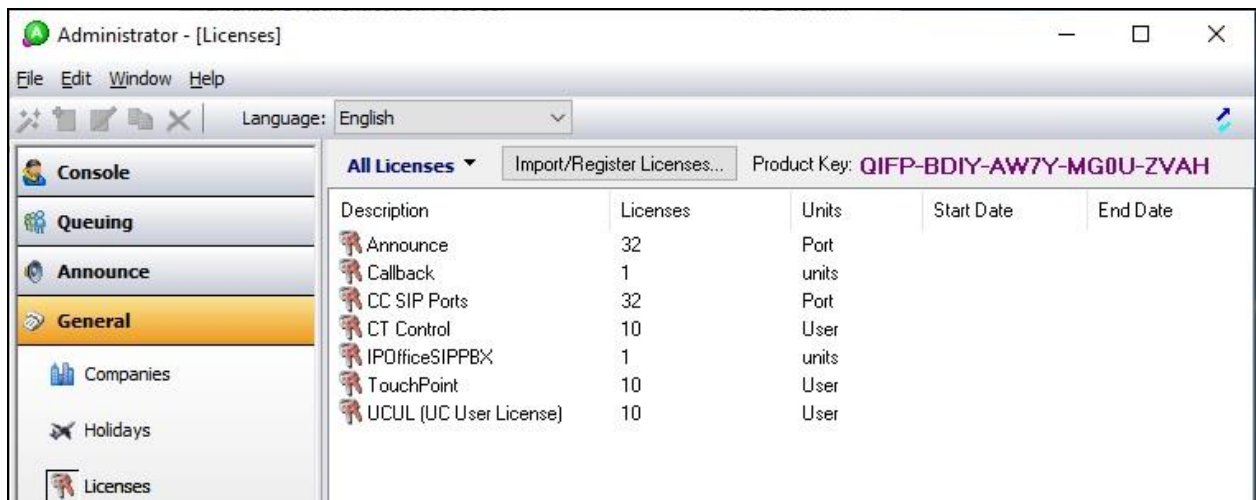
From the EICC server, select **Start → Windows Administrative Tools → Services** to display the **Services** screen. Start the **FreeSWITCH Multi Protocol Switch** service, as shown below.



6.4. Verify Licenses

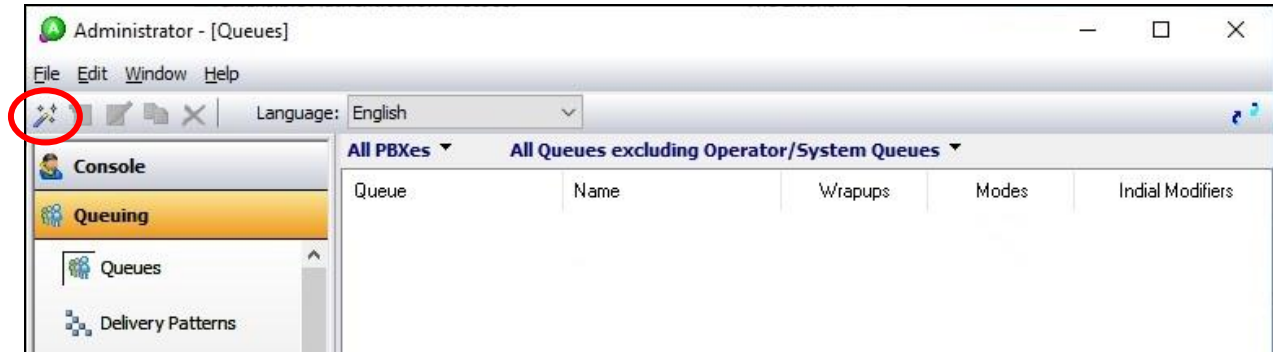
At the conclusion of installation, the **Administrator** screen is displayed. Select **General → Licenses** from the left pane, to display **All Licenses** in the right pane.

Verify that the following licenses are in place: **Announce**, **CC SIP Ports**, **CT Control**, **IPOfficeSIPPBX**, **TouchPoint**, and **UCUL (UC User License)**.



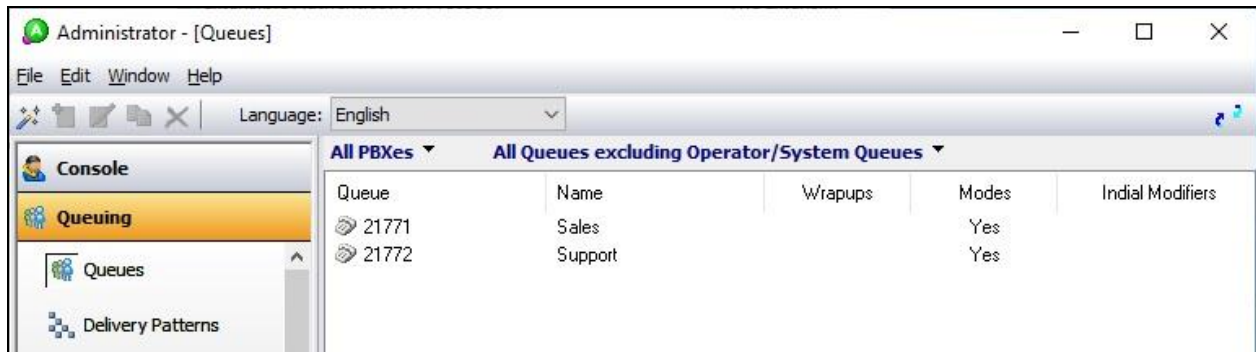
6.5. Administer Queues

Select **Queuing** → **Queues** from the left pane, followed by the **Add Wizard** icon located at the upper left corner of the screen.



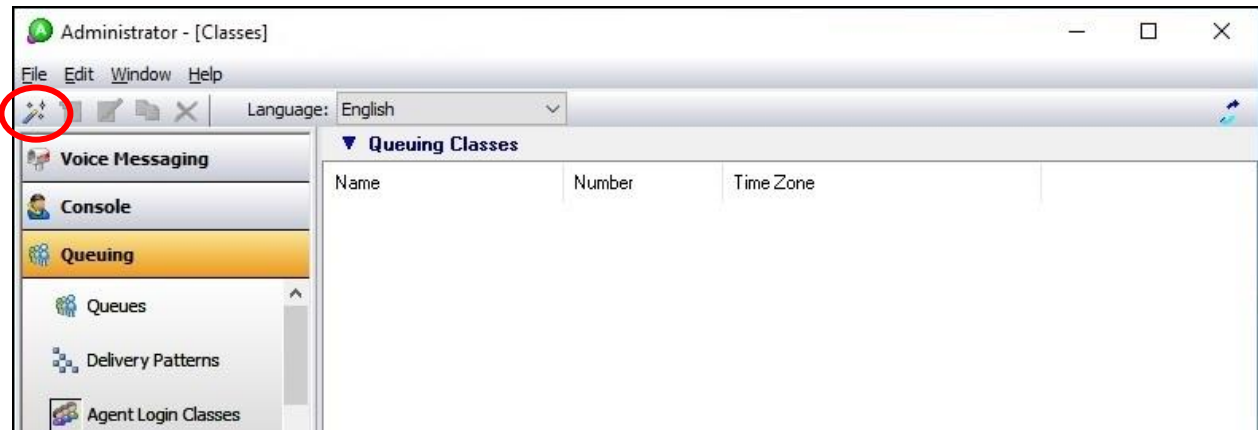
Follow the **Adding a New Queue Wizard** in the subsequent screens (not shown) to configure a new queue with desired name and queue extension.

Repeat this section to create the desired number of queues. In the compliance testing, two queues were created as shown below.

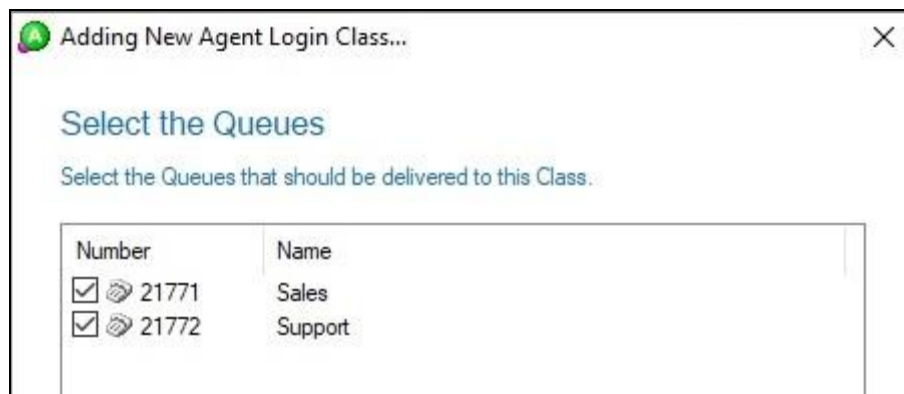


6.6. Administer Agent Login Class

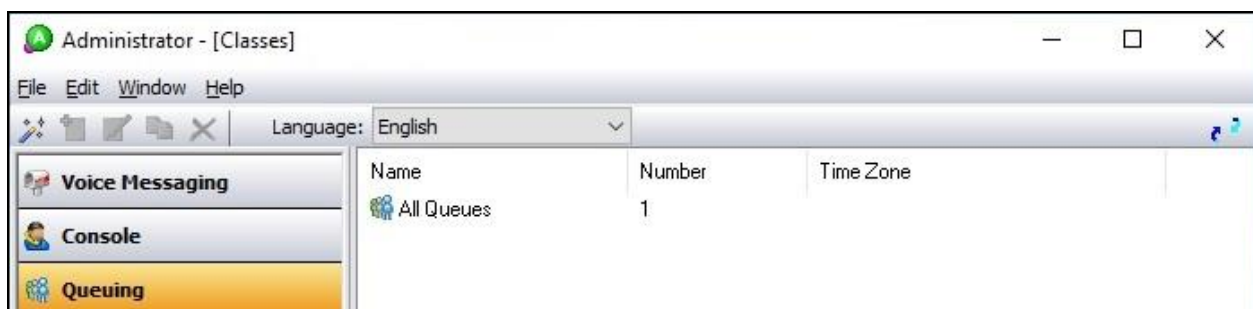
Select **Queuing** → **Agent login Classes** from the left pane, followed by the **Add Wizard** icon located at the upper left corner of the screen.



Follow the **Adding New Agent Login Class Wizard** in the subsequent screens to configure a new agent login class. In the **Select the Queues** screen, select the Sales and Support queues created from **Section 6.5**, as shown below.

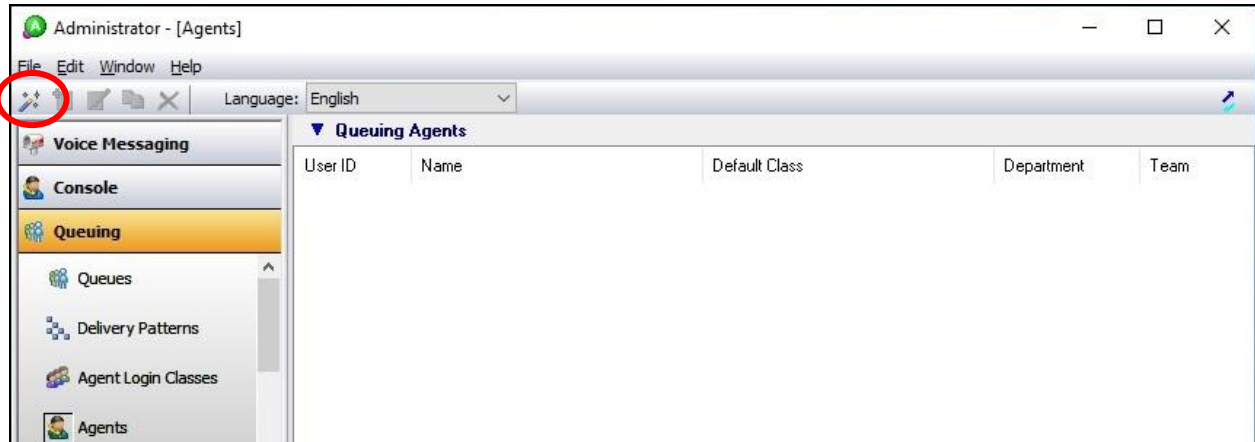


In the compliance testing, one agent login class was created, as shown below.



6.7. Administer Agents

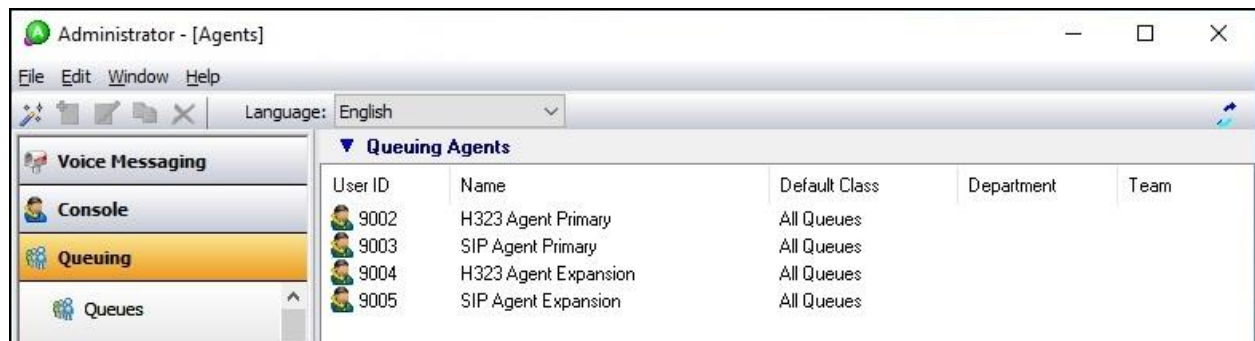
Select **Queuing** → **Agents** from the left pane, followed by the **Add Wizard** icon located at the upper left corner of the screen.



Follow the **Add Agent Wizard** in the subsequent screens to configure a corresponding entry for each agent. In the **Select Agent Login Class** screen, select the agent login class created from **Section 6.6**, as shown below.



In the compliance testing, four agents were created as shown below.



7. Verification Steps

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

7.1. Verify Avaya IP Office

From the **Avaya IP Office Manager for Server Edition IPO2-IPOSE** screen shown in **Section 5.1**, select **File → Advanced → System Status** to launch the System Status application, and log in using the appropriate credentials.

The **Avaya IP Office System Status – IPO2-IPOSE** screen is displayed. Expand **Trunks** in the left pane and select the SIP line from **Section 5.3**, in this case “3”.

Verify that the **SIP Trunk Summary** screen shows all channels with **Current State** of “Idle”, as shown below.

Avaya IP Office System Status - IPO2-IPOSE (10.64.101.234) - IP Office Linux PC 11.1.0.1.0 build 95

IP Office System Status

Help Snapshot LogOff Exit About

- System
- Alarms (13)
- Extensions (2)
- Trunks (4)
 - Line: 1
 - Line: 2
 - Line: 3**
 - Line: 8
- Active Calls
- Resources
- Voicemail
- IP Networking
- Locations

SIP Trunk Summary

Line Service State: In Service
Peer Domain Name: 10.64.101.208
Resolved Address: 10.64.101.208
Line Number: 3
Number of Administered Channels: 10
Number of Channels in Use: 0
Administered Compression: G711 Mu, G722
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)

0%

Channel Number	U...	Call Ref	Current State	Time in State	Remote Media A...	Co...	Conne...	Caller ID or Dial...	Other Party on Call	Direction of Call	Round Trip D...	Receive Jitter	Receive Packe...	Transmit Jitter	Transmit Packe...
1			Idle	00:00:23											
2			Idle	00:00:23											
3			Idle	00:00:23											
4			Idle	00:00:23											
5			Idle	00:00:23											
6			Idle	00:00:23											
7			Idle	00:00:23											
8			Idle	00:00:23											
9			Idle	00:00:23											
10			Idle	00:00:23											

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

10:25:01 AM Online

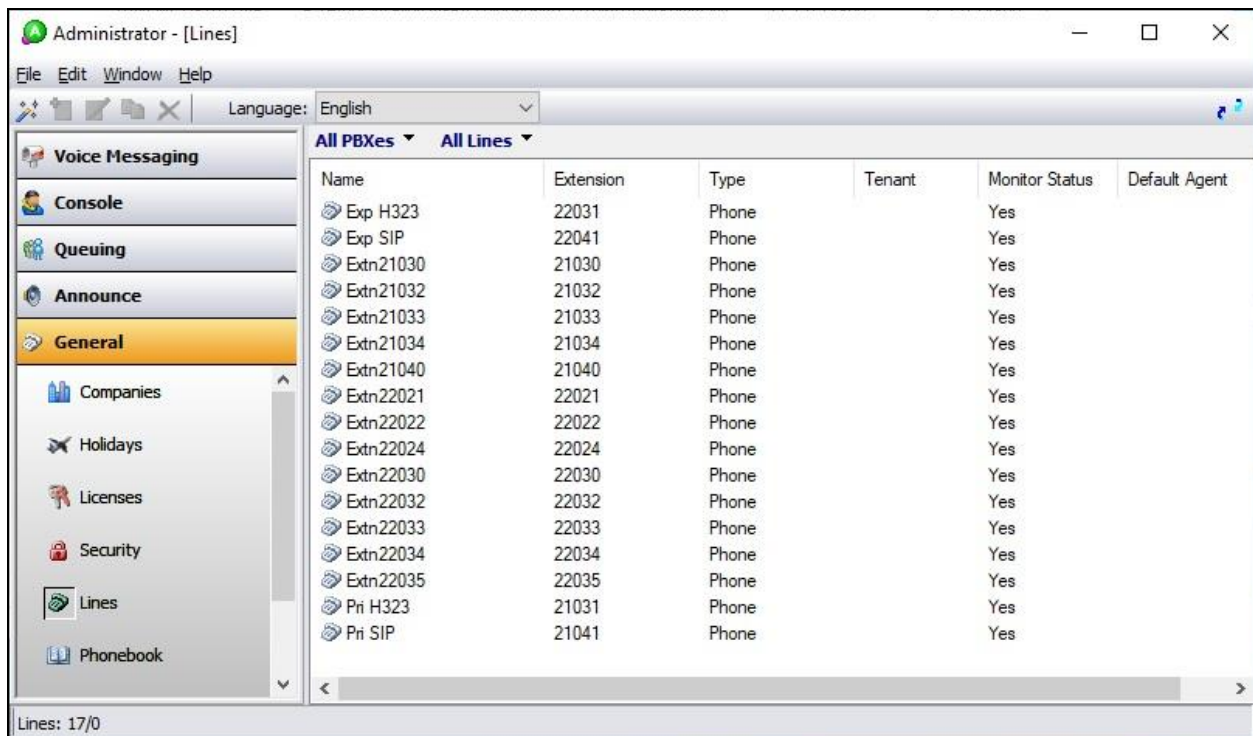
7.2. Verify Enghouse Interactive Communications Center

This section provides tests that can be performed to verify the EICC server and the TouchPoint client.

7.2.1. EICC Server

From the **Administrator** screen on the EICC server, select **General** → **Lines**. Verify that a listed of users from both IP Office systems are listed as shown below, which were picked up via the MTCTI-3 interface.

Note that by default, EICC monitors all received users from IP Office.



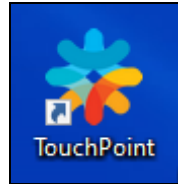
The screenshot shows the 'Administrator - [Lines]' window. The left sidebar has a menu with 'General' selected. The main area displays a table of lines under the 'All Lines' tab. The table has columns for Name, Extension, Type, Tenant, Monitor Status, and Default Agent. The data is as follows:

Name	Extension	Type	Tenant	Monitor Status	Default Agent
Exp H323	22031	Phone		Yes	
Exp SIP	22041	Phone		Yes	
Extn21030	21030	Phone		Yes	
Extn21032	21032	Phone		Yes	
Extn21033	21033	Phone		Yes	
Extn21034	21034	Phone		Yes	
Extn21040	21040	Phone		Yes	
Extn22021	22021	Phone		Yes	
Extn22022	22022	Phone		Yes	
Extn22024	22024	Phone		Yes	
Extn22030	22030	Phone		Yes	
Extn22032	22032	Phone		Yes	
Extn22033	22033	Phone		Yes	
Extn22034	22034	Phone		Yes	
Extn22035	22035	Phone		Yes	
Pri H323	21031	Phone		Yes	
Pri SIP	21041	Phone		Yes	


At the bottom left of the window, it says 'Lines: 17/0'.

7.2.2. TouchPoint Client

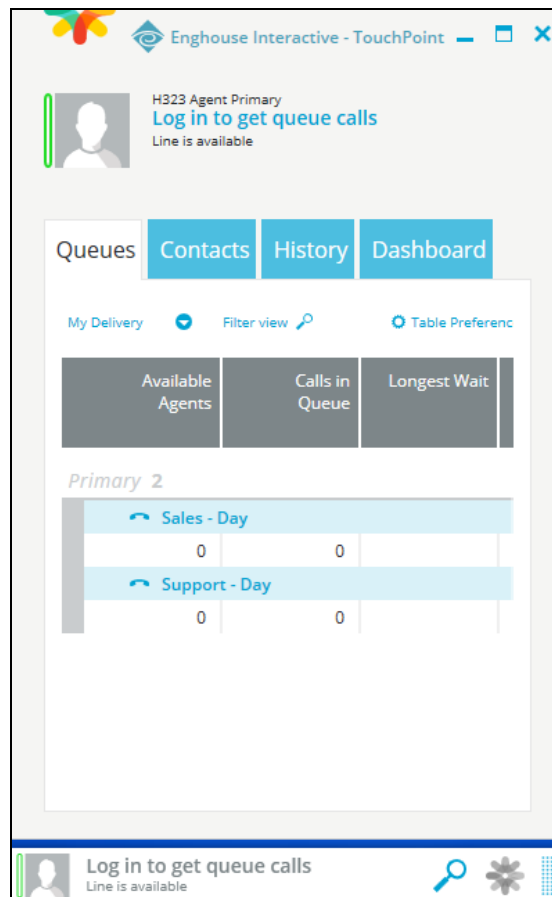
From an agent desktop running the TouchPoint client application, double-click on the **TouchPoint** shortcut icon shown below, which was created as part of installation.



The **Enghouse Interactive TouchPoint** login screen below is displayed. Enter the login name associated with an agent from **Section 6.7**, and use the generic default PIN value from EICC. Retain the default value in the remaining fields.

A screenshot of the Enghouse Interactive TouchPoint login screen. The screen has a white background with a blue border. At the top left is the Enghouse logo, a colorful flower-like shape. Below it, the text "Enghouse Interactive TouchPoint" is displayed in a black, sans-serif font. There are two input fields: the first contains the text "H323 Agent Primary", and the second contains four black dots, indicating a password field. Below the password field is a checkbox with a blue checkmark and the text "Remember me". At the bottom right is a blue button with the text "Open TouchPoint" in white. A small blue "x" icon is in the top right corner of the window.

The **TouchPoint** screen is displayed, along with a Call Bar above the system tray, as shown below. Click on **Log in to get queue calls** toward the top of the screen.

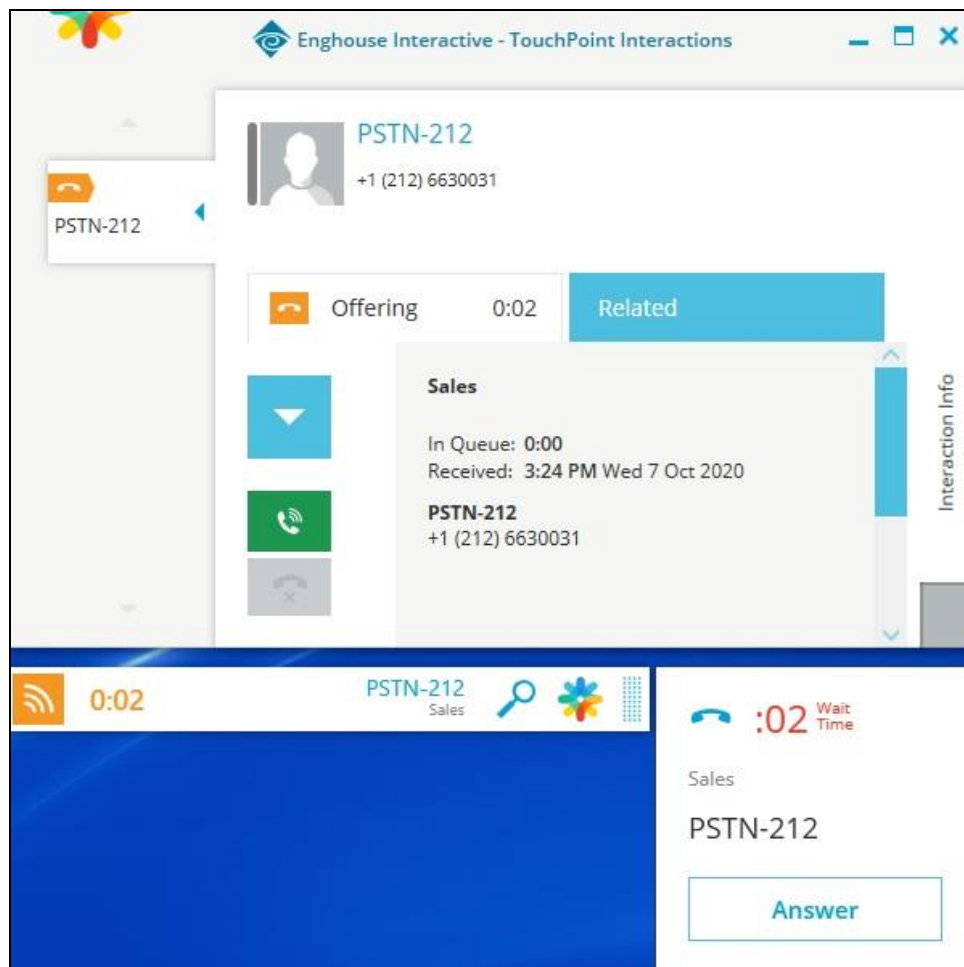


In the drop-down window, select **Log in to Queues** as shown below.



Make an incoming ACD call from PSTN to the Sales queue on EICC, with available agent “21031”. Verify that the agent desktop is populated with a **TouchPoint Interactions** screen with an **Offering** tab, along with a Pop-up Notification box, and that the existing Call Bar is updated to reflect the active call.

Click **Answer** from the Pop-up Notification box and verify that the agent is connected to the PSTN caller with two-way talk paths.



8. Conclusion

These Application Notes describe the configuration steps required for Enghouse Interactive Communications Center 11.1 to successfully interoperate with Avaya IP Office Server Edition 11.1 using the MTCTI-3 and SIP trunk interfaces. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

9. Additional References

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

1. *Administering Avaya IP Office™ Platform with Manager*, Release 11.1, Issue 1, April 2020, available at <http://support.avaya.com>.
2. *First-time Installation and Server Setup – SIP IPO PBX Installation Manual*, Communications Center version CC 11.1, August 2020, available at <https://partnerportal.enghouseinteractive.com/user/login>.

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