



Avaya Solution & Interoperability Test Lab

Application Notes for Valcom V-9972 Universal Paging Interface with Avaya IP Office Server Edition using SIP Trunk - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Valcom V-9972 Universal Paging Interface with Avaya IP Office Server Edition 11.1 and Avaya IP Office 500 V2 Expansion System 11.1. Valcom V-9972 Universal Paging Interface provides access to paging systems, such as Valcom VIP-430A IP Wall Speakers, which was used in the compliance test. For this compliance test, Valcom V-9972 Universal Paging Interface connected to Avaya IP Office Server Edition via a SIP trunk. The Valcom V-9972 Universal Paging Interface supports two-way audio intercom (talkback) calls and one-way audio group paging calls.

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 describe the configuration steps required to integrate the Valcom V-9972 Universal Paging Interface with Avaya IP Office Server Edition 11.1 and Avaya IP Office 500 V2 Expansion System 11.1. Valcom V-9972 Universal Paging Interface provides access to paging systems, such as Valcom VIP-430A IP Wall Speakers, which was used in the compliance test. For this compliance test, Valcom V-9972 Universal Paging Interface connected to Avaya IP Office Server Edition via a SIP trunk. The Valcom V-9972 Universal Paging Interface supports two-way audio intercom (talkback) calls and one-way audio group paging calls.

When a call is routed to the Valcom V-9972 Universal Paging Interface, the V-9972 plays dial tone back to the caller. The caller can then dial a Valcom speaker Dial Code or Group Code to establish an intercom call (two-way audio) with a single Valcom speaker or a group paging call (one-way audio) to one or more Valcom speakers.

Alternatively, the Valcom VIP-430A IP Wall Speaker can establish intercom calls by pressing its call button. Pressing the call button would place a call to the specified destination in the V-9972 configuration. Pressing the call button during an active call, terminates the call.

All calls to/from the VIP-430A IP Wall Speaker go through the V-9972. Communication between V-9972 and VIP-430A IP Wall Speaker uses unicast for intercom (talkback) calls and multicast for paging calls.

Valcom offers Universal Paging Adapters as different products/models to accommodate different environments. They share the same SIP stack and firmware version, therefore, this testing also applies to those products, as detailed in **Attachment 1. Section 4** of this document shows the actual products/models and SIP Stack and software versions that were tested. For additional details, contact Valcom Support, as noted in **Section 2.3**.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between the Valcom V-9972 Universal Paging Interface with the Valcom VIP-430A IP Wall Speaker, Avaya SIP / H.323 IP Deskphones, and the PSTN. Two-way audio intercom calls and one-way audio group paging calls were exercised. In addition, basic telephony features were exercised from Avaya SIP / H.323 IP Deskphones, such as hold/resume, call transfer, and conference.

The serviceability testing focused on verifying that the Valcom V-9972 Universal Paging Interface came back into service after reconnecting the network connection or a reboot.

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 Valcom V-9972 Universal Paging Interface used TLS/SRTP encryption features.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Establishing a SIP trunk between V-9972 and IP Office Server Edition and IP Office 500V2 Expansion and verifying the exchange of SIP Options messages.
- Calls between V-9972 and Avaya H.323/SIP endpoints with Direct IP Media (Shuffling) enabled and disabled. Shuffling allows IP endpoints to send audio RTP packets directly to each other without using media resources in Avaya IP Office.
- Establishing two-way audio intercom calls between VIP-430A IP Wall Speaker, via V-9972, Avaya H.323 / SIP Deskphones, and PSTN in both directions.
- Establishing one-way paging calls from Avaya H.323 / SIP Deskphones to VIP-430A IP Wall Speaker via V-9972.
- Verifying that higher priority paging calls take precedence over existing lower priority intercom calls.
- Terminating active calls by pressing the call button on the VIP-430A IP Wall Speaker.
- Support of G.711 mu-law codec.
- Support of TLS/SRTP using mutual TLS authentication.
- Since the VIP-430A IP Wall Speaker does not provide a keypad or feature buttons, basic telephony features, such as hold/resume, call transfer, and conference were performed from Avaya H.323/SIP Deskphones.
- Long duration calls and outbound calls from V-9972 that were rejected due to dialing an invalid number or a busy station.
- Proper system recovery after re-establishing network connectivity to the V-9972 or restarting the V-9972.

2.2. Test Results

All test cases passed.

2.3. Support

For technical support and information on Valcom V-9972 Universal Paging Interface, contact Valcom Technical Support at:

- Phone: +1 (800) 825-2661 or +1 (540) 563-2000
- Website: <https://www.valcom.com/Support/techsupport.html>
- Email: support@valcom.com

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network that includes the following products:

- Avaya IP Office Server Edition and Avaya IP Office 500 V2 Expansion connected via a SCN trunk and configured via Avaya IP Office Manager.
- PSTN connectivity provided by a SIP trunk on Avaya IP Office Server Edition and an ISDN-PRI trunk on Avaya IP Office 500 V2 Expansion System.
- Avaya 96x1 Series H.323 Deskphones and Avaya J100 Series SIP Deskphones registered to Avaya IP Office Server Edition and Avaya IP Office 500 V2 Expansion.
- Valcom V-9972 Universal Paging Interface connected to Avaya IP Office Server Edition via a SIP trunk. Valcom VIP-430A IP Wall Speaker serving as the Valcom audio endpoint, and using Valcom VIP-102B IP Solutions Setup Tool to configure Valcom products.

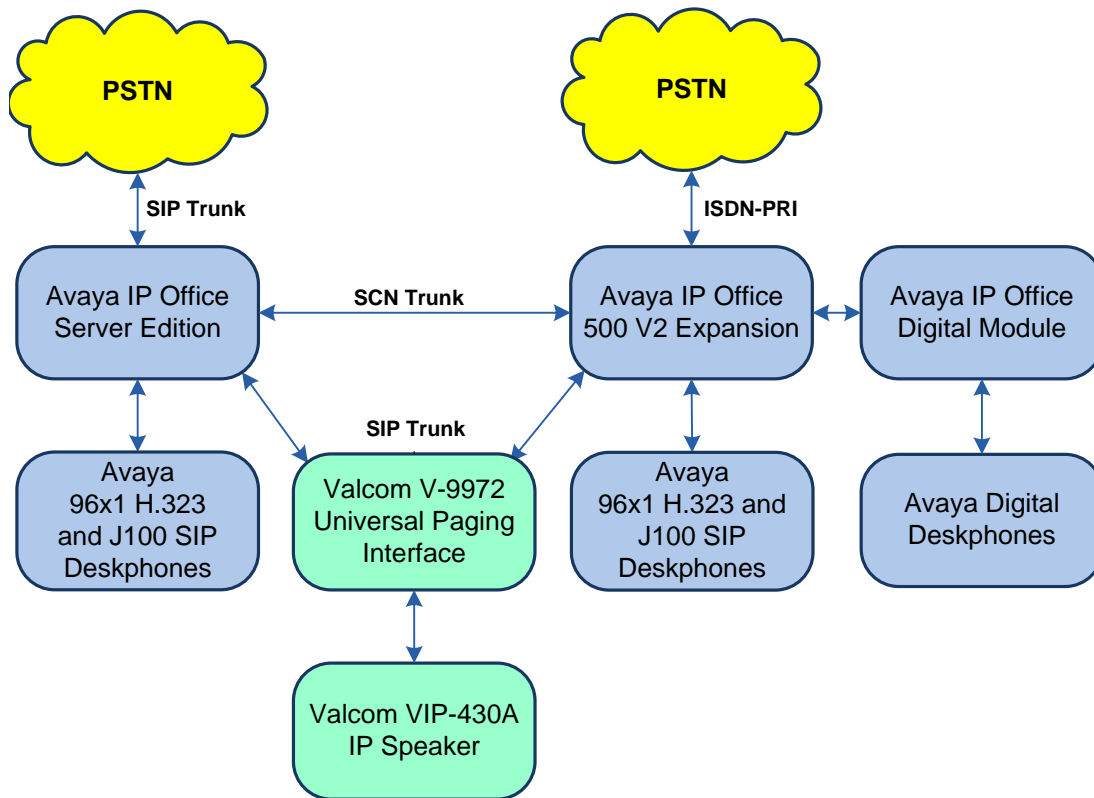


Figure 1: Avaya SIP Network with Valcom V-9972 Universal Paging Interface and Valcom VIP-430A IP Wall Speakers

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	11.1.2.1.0 build 3
Avaya IP Office 500 V2 Expansion	11.1.2.1.0 build 3
Avaya 96x1 Series IP Deskphones	6.8304 (H.323)
Avaya J100 Series IP Deskphones	4.0.7.0.7 (SIP)
Valcom V-9972 Universal Paging Interface, including optional L9972-2 feature license	3.00.14
Valcom VIP-430A IP Wall Speaker	3.23.7
Valcom VIP-102B IP Solutions Setup Tool	8.4.0.0

Note: 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 Server Edition

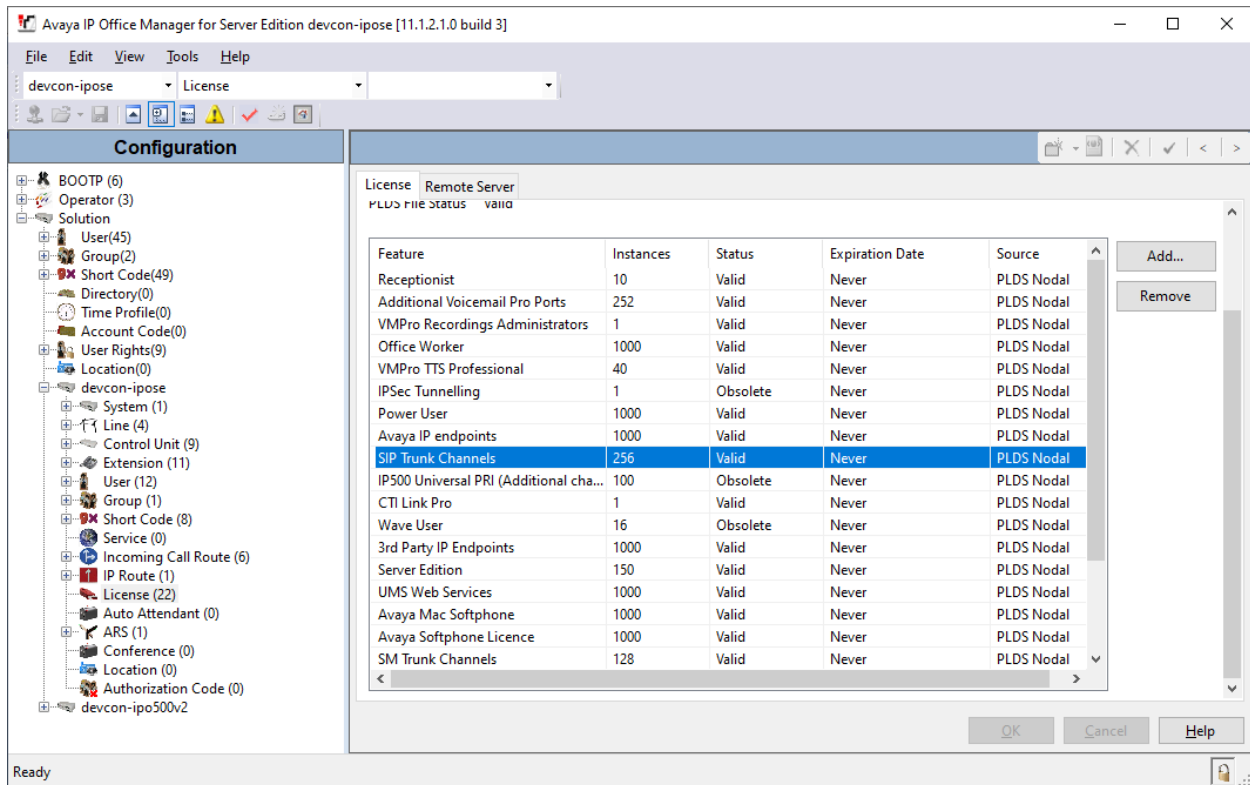
This section provides the procedures for configuring Avaya IP Office. The procedures include the following areas:

- Verify IP Office license
- Obtain LAN IP address
- Enable SIP trunks
- Administer SIP line
- Administer incoming call route
- Administer short code
- Install Valcom V-9972 Universal Paging Interface TLS Certificate

5.1. Verify IP Office License

From a PC with Avaya IP Office Manager installed, select **Start** → **Programs** → **IP Office** → **Manager** to launch the Manager application. Select the required IP Office system and log in with the appropriate credentials.

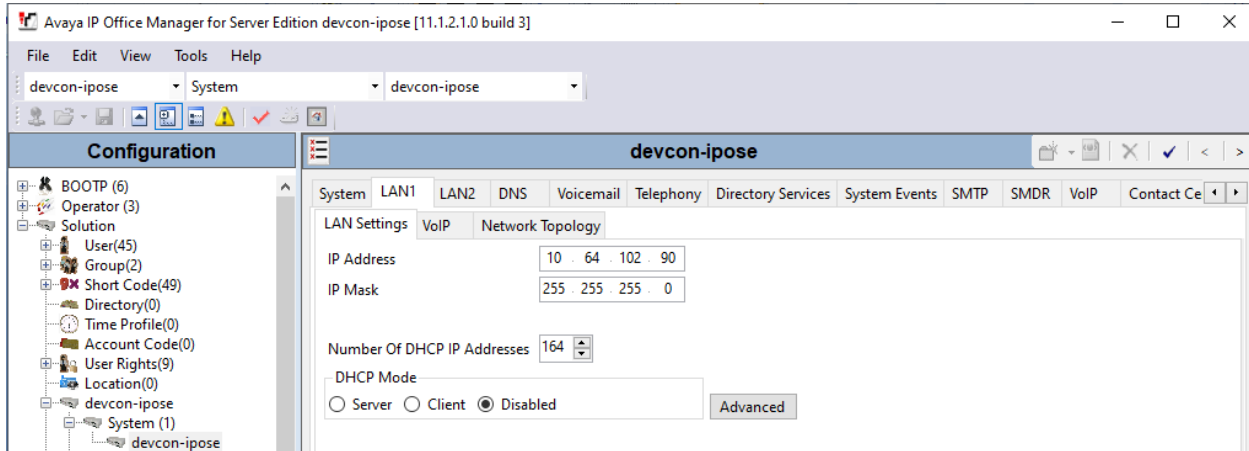
The **Avaya IP Office Manager for Server Edition** screen is displayed. From the configuration tree in the left pane, select **License** to display the license screen in the right pane. Verify that the **License Status** is “Valid” for **SIP Trunk Channels**, and that the **Instances** value is sufficient.



Feature	Instances	Status	Expiration Date	Source
Receptionist	10	Valid	Never	PLDS Nodal
Additional Voicemail Pro Ports	252	Valid	Never	PLDS Nodal
VMPPro Recordings Administrators	1	Valid	Never	PLDS Nodal
Office Worker	1000	Valid	Never	PLDS Nodal
VMPPro TTS Professional	40	Valid	Never	PLDS Nodal
IPSec Tunnelling	1	Obsolete	Never	PLDS Nodal
Power User	1000	Valid	Never	PLDS Nodal
Avaya IP endpoints	1000	Valid	Never	PLDS Nodal
SIP Trunk Channels	256	Valid	Never	PLDS Nodal
IP500 Universal PRI (Additional cha...	100	Obsolete	Never	PLDS Nodal
CTI Link Pro	1	Valid	Never	PLDS Nodal
Wave User	16	Obsolete	Never	PLDS Nodal
3rd Party IP Endpoints	1000	Valid	Never	PLDS Nodal
Server Edition	150	Valid	Never	PLDS Nodal
UMS Web Services	1000	Valid	Never	PLDS Nodal
Avaya Mac Softphone	1000	Valid	Never	PLDS Nodal
Avaya Softphone Licence	1000	Valid	Never	PLDS Nodal
SM Trunk Channels	128	Valid	Never	PLDS Nodal

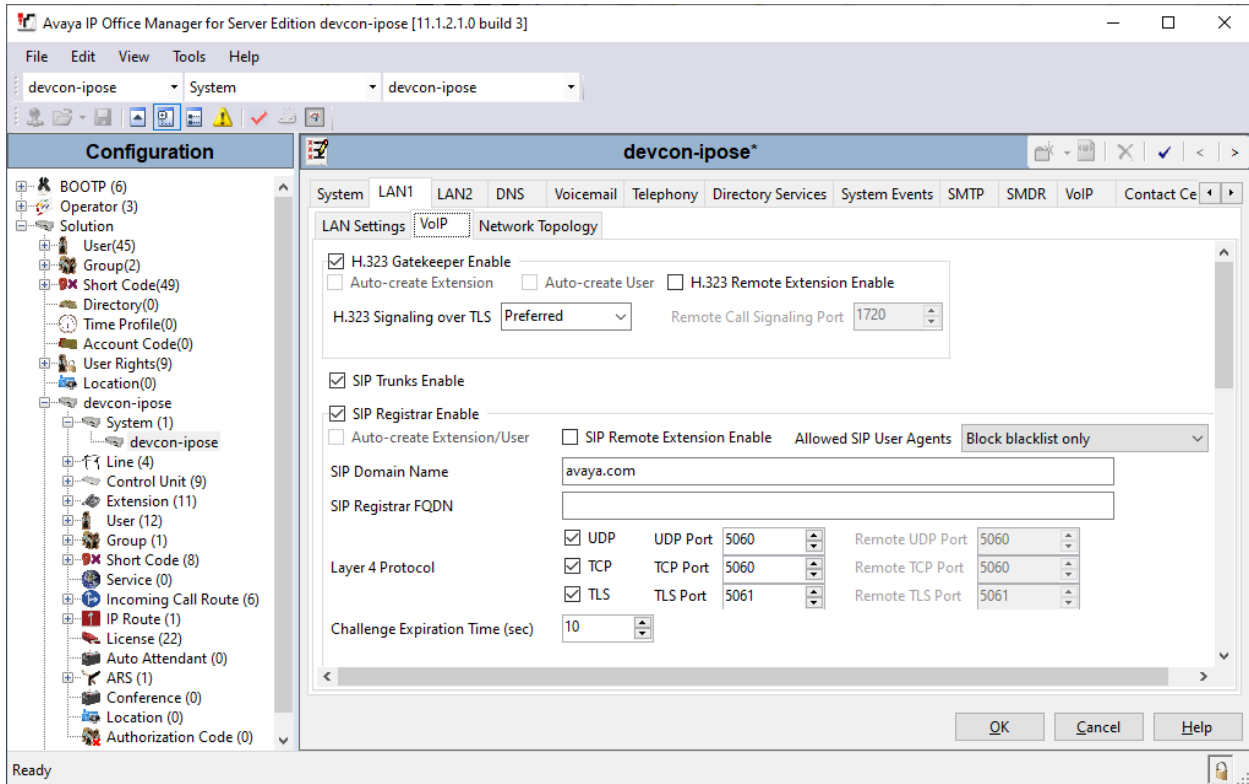
5.2. Obtain LAN IP Address

From the configuration tree in the left pane, select **System** to display the **System** screen for the IP Office Server Edition 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 V-9972.



5.3. Enable SIP Trunks

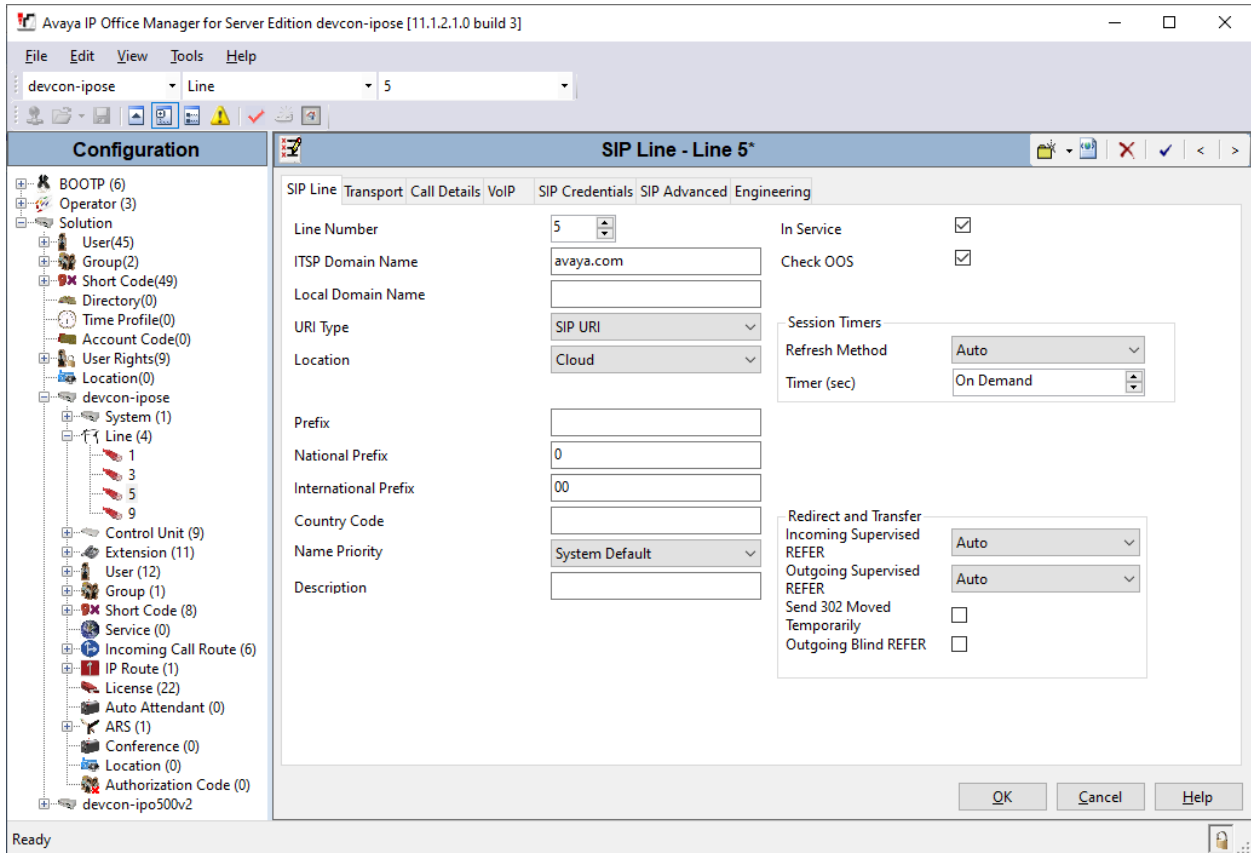
Select the **VoIP** sub-tab. Ensure that **SIP Trunks Enable** is checked as shown below.



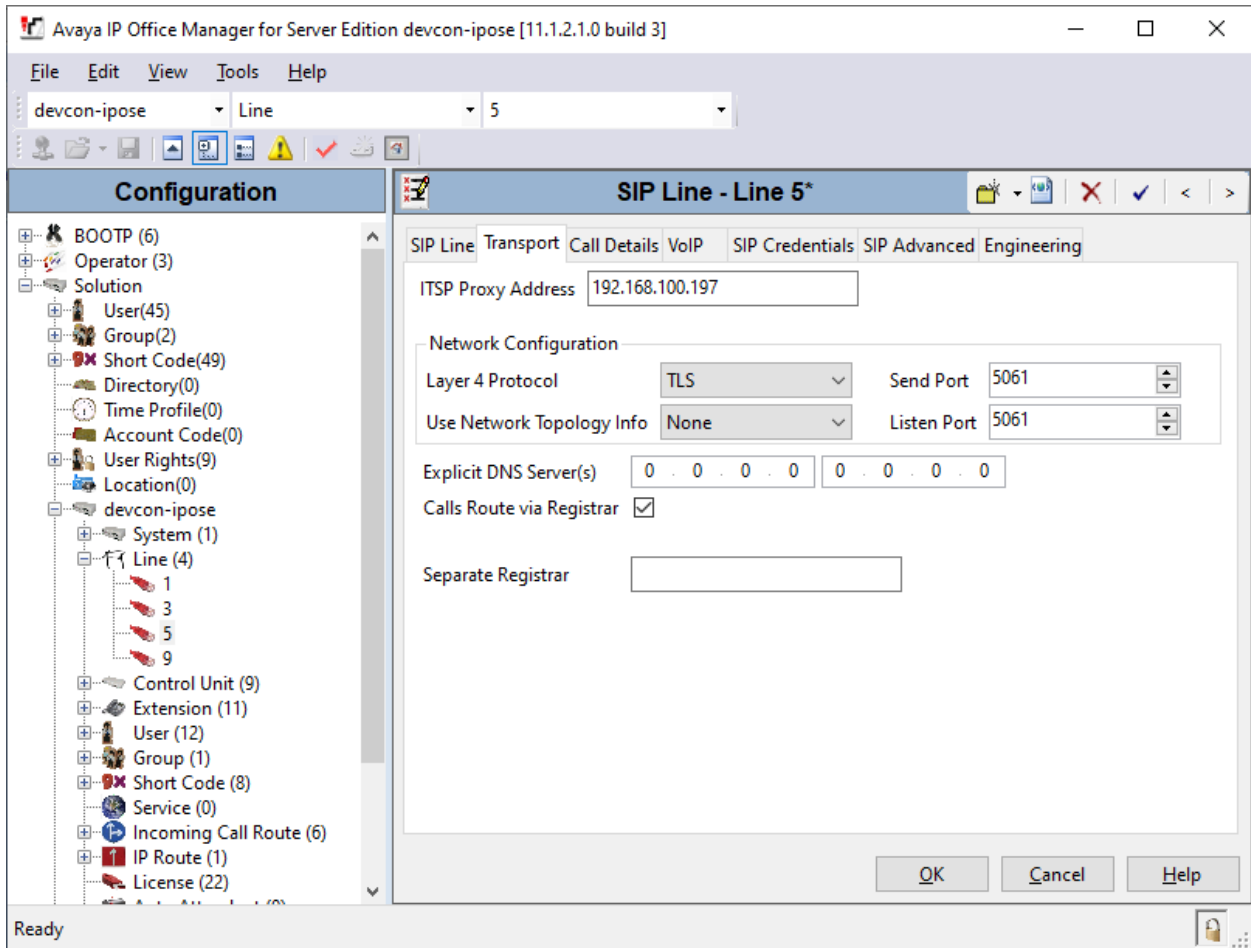
5.4. Administer SIP Line

From the configuration tree in the left pane, right-click on **Line**, and select **New** → **SIP Line** from the pop-up list to add a new SIP line (not shown).

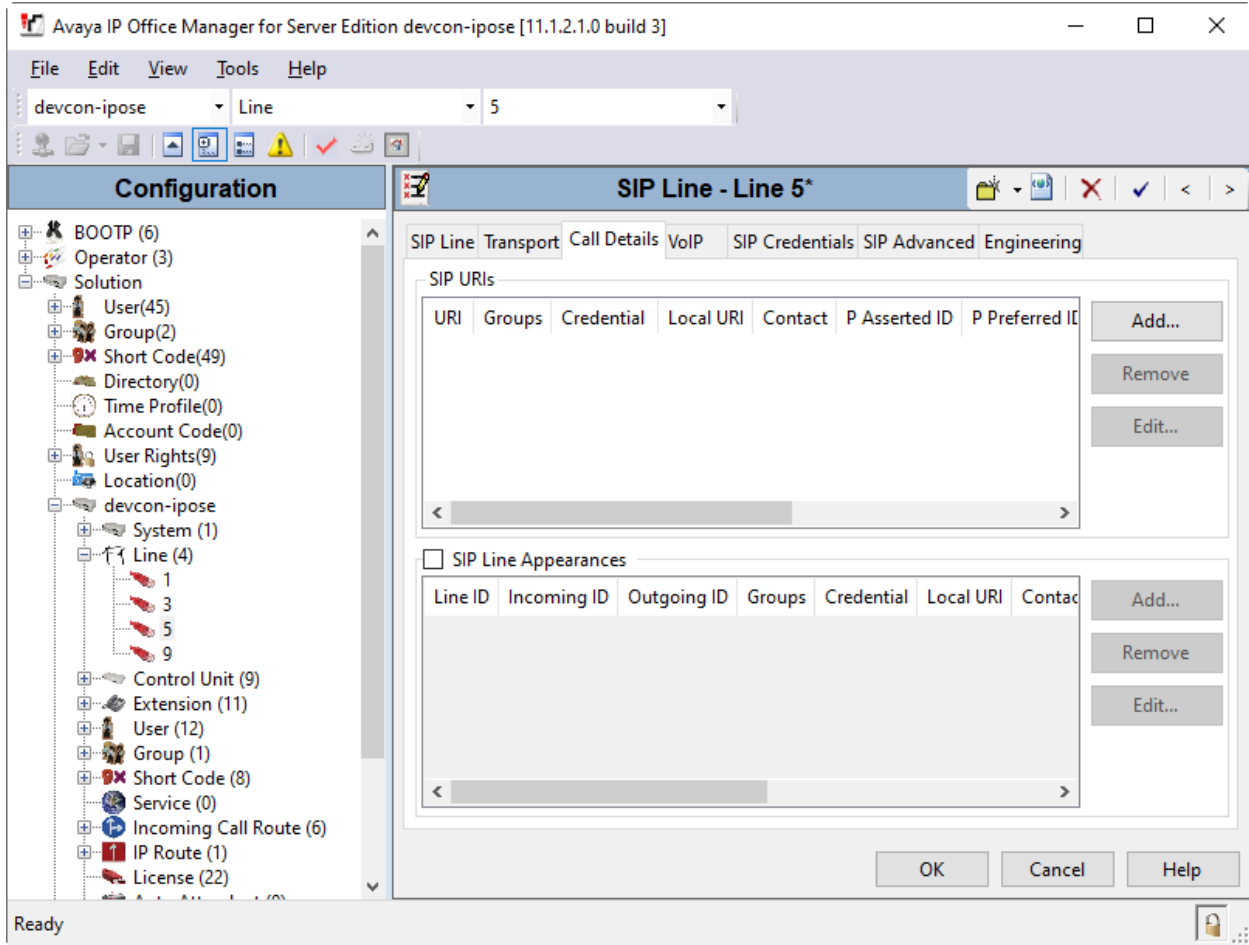
Select the **SIP Line** tab. Select an available **Line Number** (e.g., 5) and set **ITSP Domain Name** to the appropriate SIP domain (e.g., *avaya.com*).



Select the **Transport** tab. For **ITSP Proxy Address**, enter the IP address of V-9972, set the **Layer 4 Protocol** to *TLS* and the **Send Port** and **Listen Port** to the TLS port (e.g., *5061*). Retain the default values for the remaining fields.



Select the **Call Details** tab, and click **Add** to display the **Call Details | SIP URI** section.



In the **Call Details | SIP URI** window, set the **Incoming Group** and **Outgoing Group** to a valid number (e.g., 5) and set the **Max Sessions**.

SIP Line - 5 | Call Details | SIP URI

New URI

Incoming Group Max Sessions

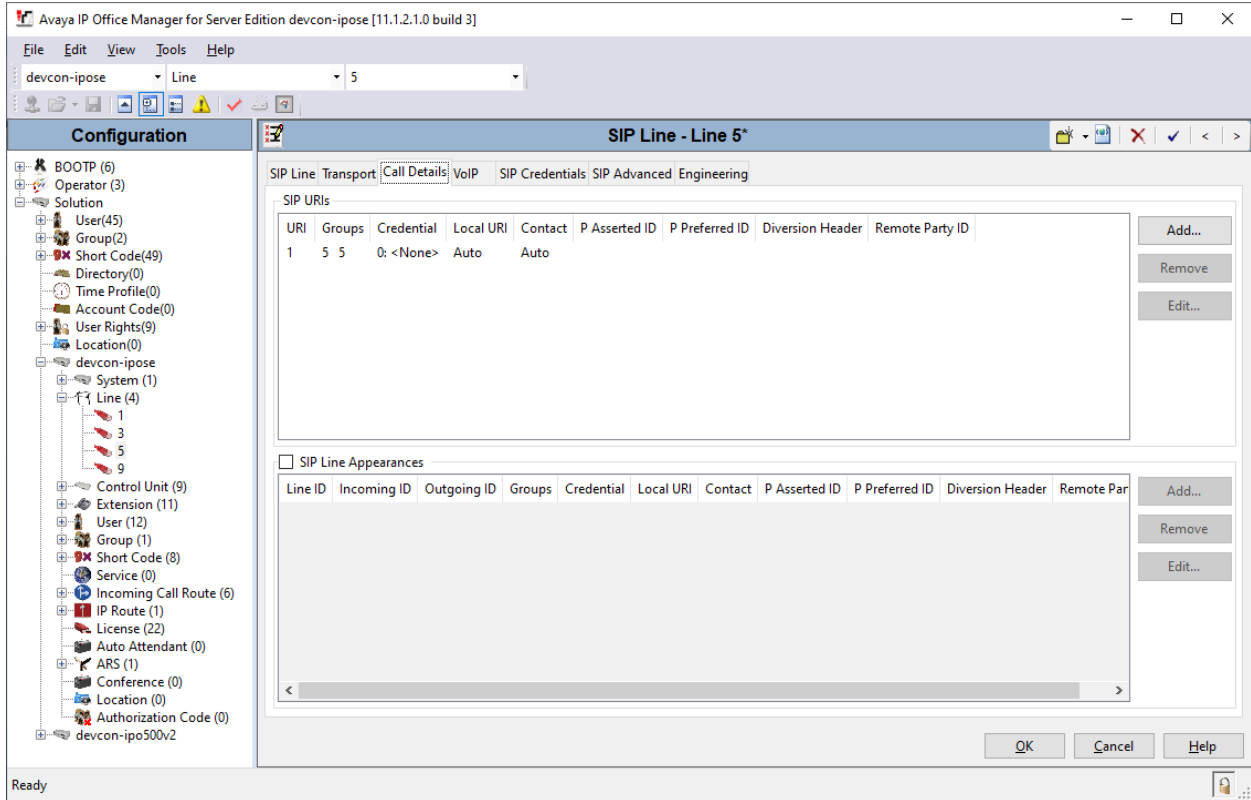
Outgoing Group

Credentials

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

OK Cancel Help

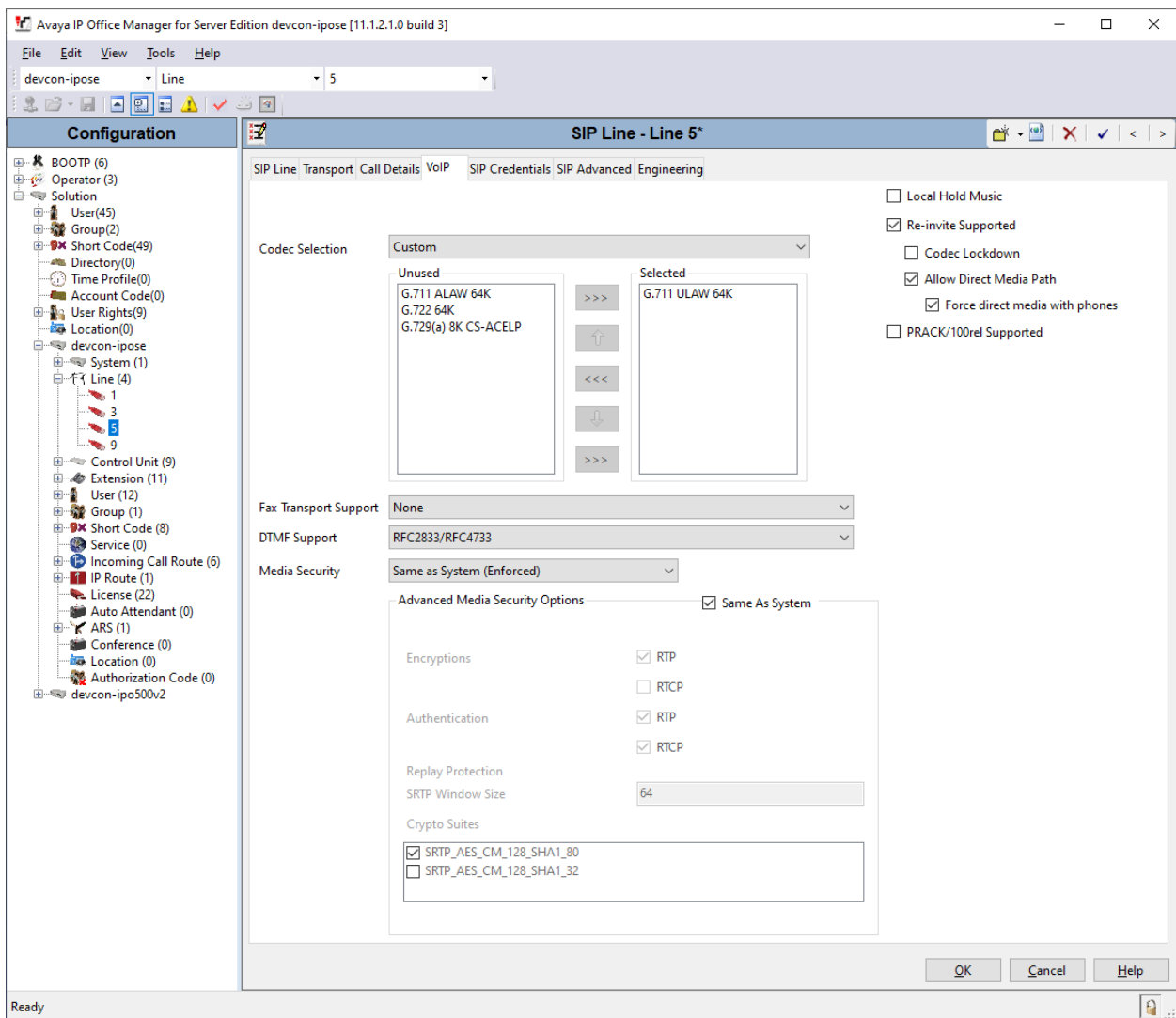
The **Call Details** tab is updated as shown below.



Select the **VoIP** tab. For **Codec Selection**, select *Custom* and select G.711 ULAW 64K, which is the codec supported by V-9972 with IP speakers. Enable **Allow Direct Media Path** and **Force direct media with phones** so that audio/RTP flows directly between V-9972 and H.323/SIP endpoints without using media resources in Avaya IP Office Server Edition.

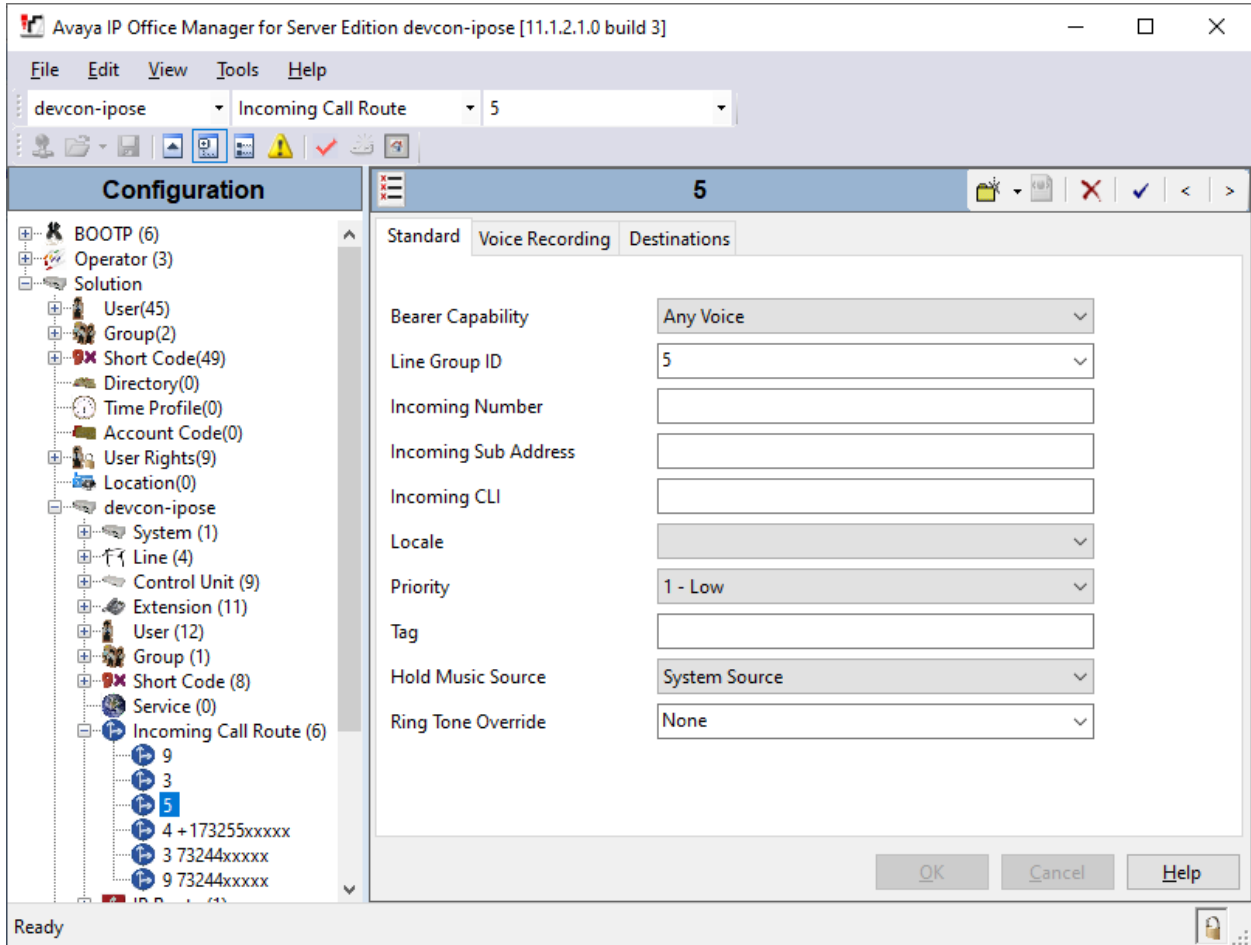
Media Security was enabled for V-9972. Since the V-9972 does not support SDP Capability Negotiation (RFC5939), IP Office Server Edition should only offer SRTP in the SIP SDP. Therefore, the **Media Security** field must be set to *Enforced*. In addition, when SRTP is enabled for V-9972, unencrypted RTCP is required.

Note: The Media Security section shown below was used for Avaya H.323 / SIP Deskphones and the Web Socket SCN trunk between IP Office Server Edition and IP Office 500 V2 Expansion.

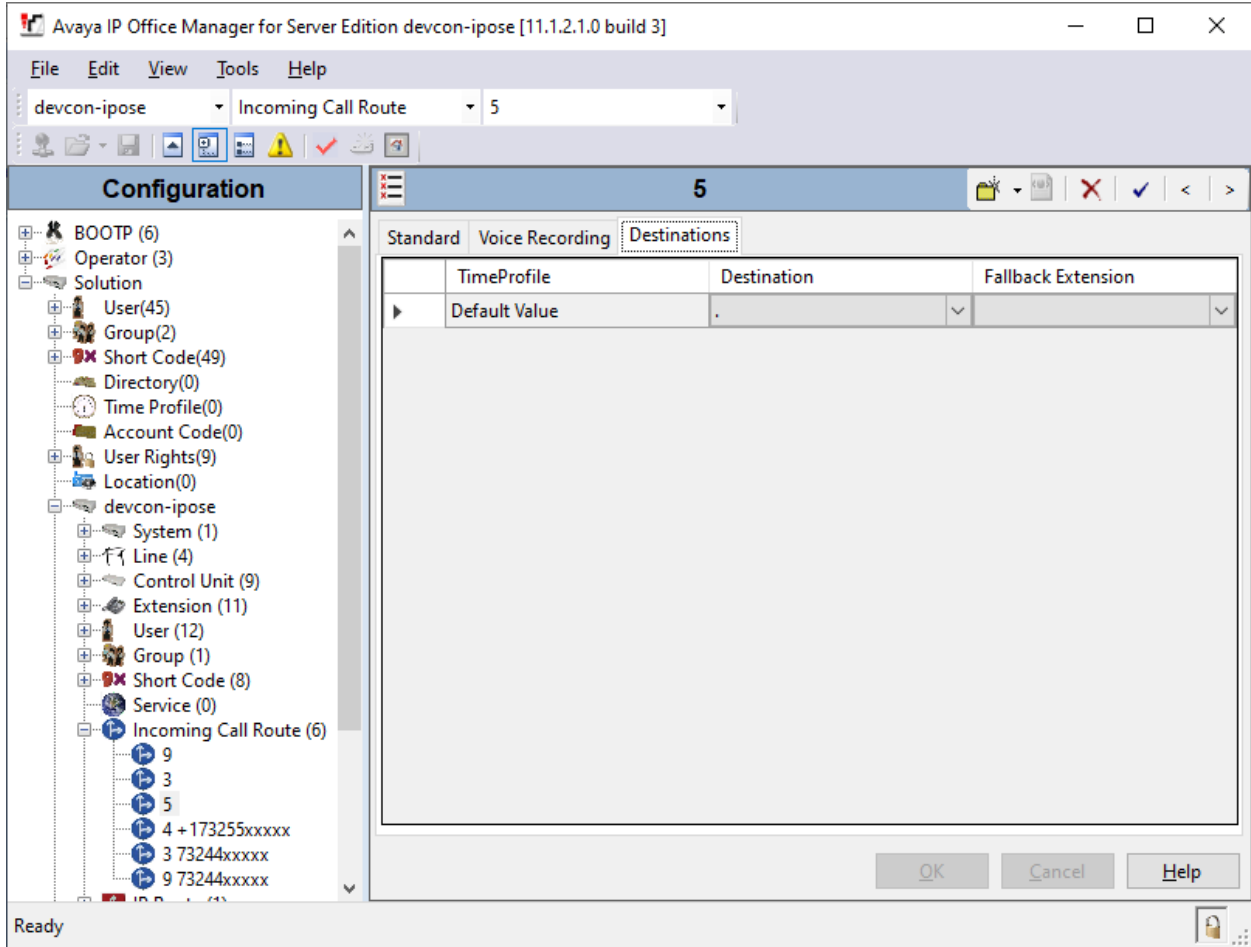


5.5. Administer Incoming Call Route

From the configuration tree in the left pane, right-click on **Incoming Call Route**, and select **New** from the pop-up list to add a new route. For **Line Group Id**, select the incoming group number from **Section 5.4**, which corresponds to the SIP line, in this case “5”.

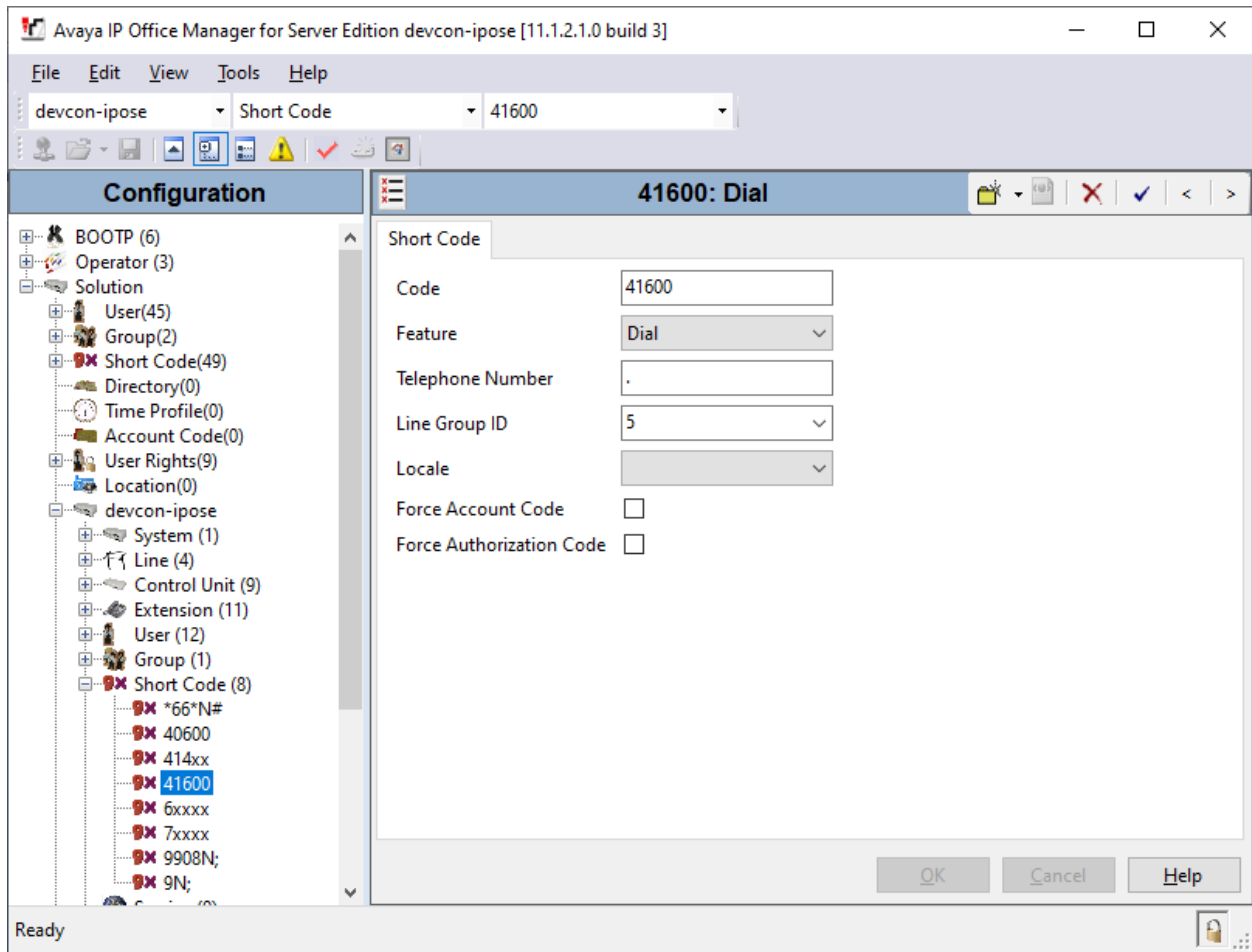


Select the **Destinations** tab. For **Destination**, enter “.” to match any dialed number by V-9972.



5.6. Administer Short Code

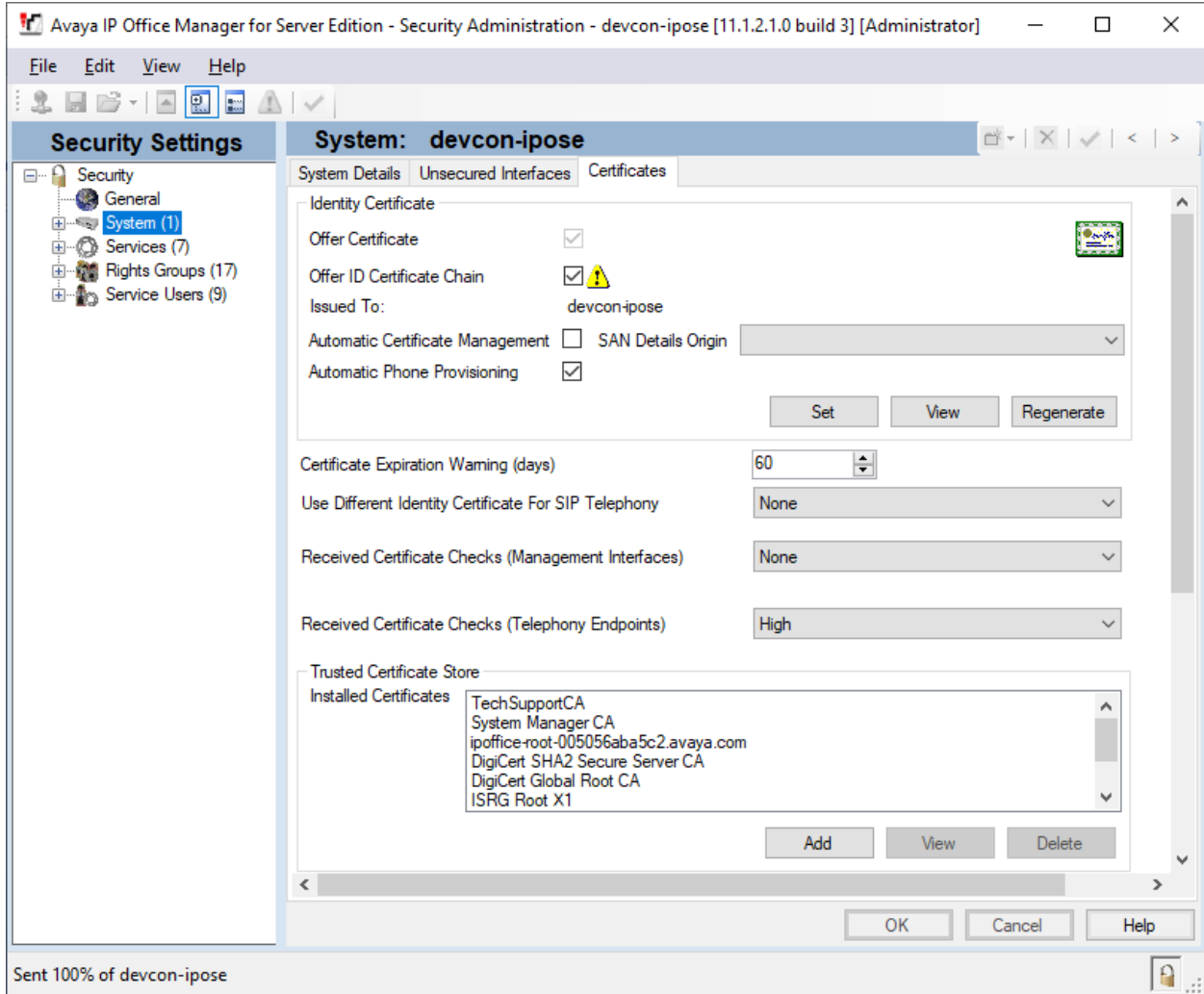
From the configuration tree in the left pane, right-click on **Short Code** and select **New** from the pop-up list to add a new short code for calls to V-9972. For the compliance test, 41600 was used to route calls to V-9972. Set **Telephone Number** to . so that 41600 would be sent to V-9972 and set **Line Group Id** to the outgoing group number from **Section 5.4**, which corresponds to the SIP line.



5.7. Install Valcom V-9972 Universal Paging Interface TLS Certificate

From **Avaya IP Office Manager for Server Edition**, navigate to **File → Advanced → Security Settings**. The **Security Settings** window is displayed. In the left pane, navigate to **Security → System**, and then select the **Certificates** tab to display the window below.

To enable mutual TLS authentication, install the V-9972 TLS certificate shown below as *TechSupportCA* and set **Received Certificate Checks (Telephony Endpoints)** to *High*.



6. Configure Valcom V-9972 Universal Paging Interface

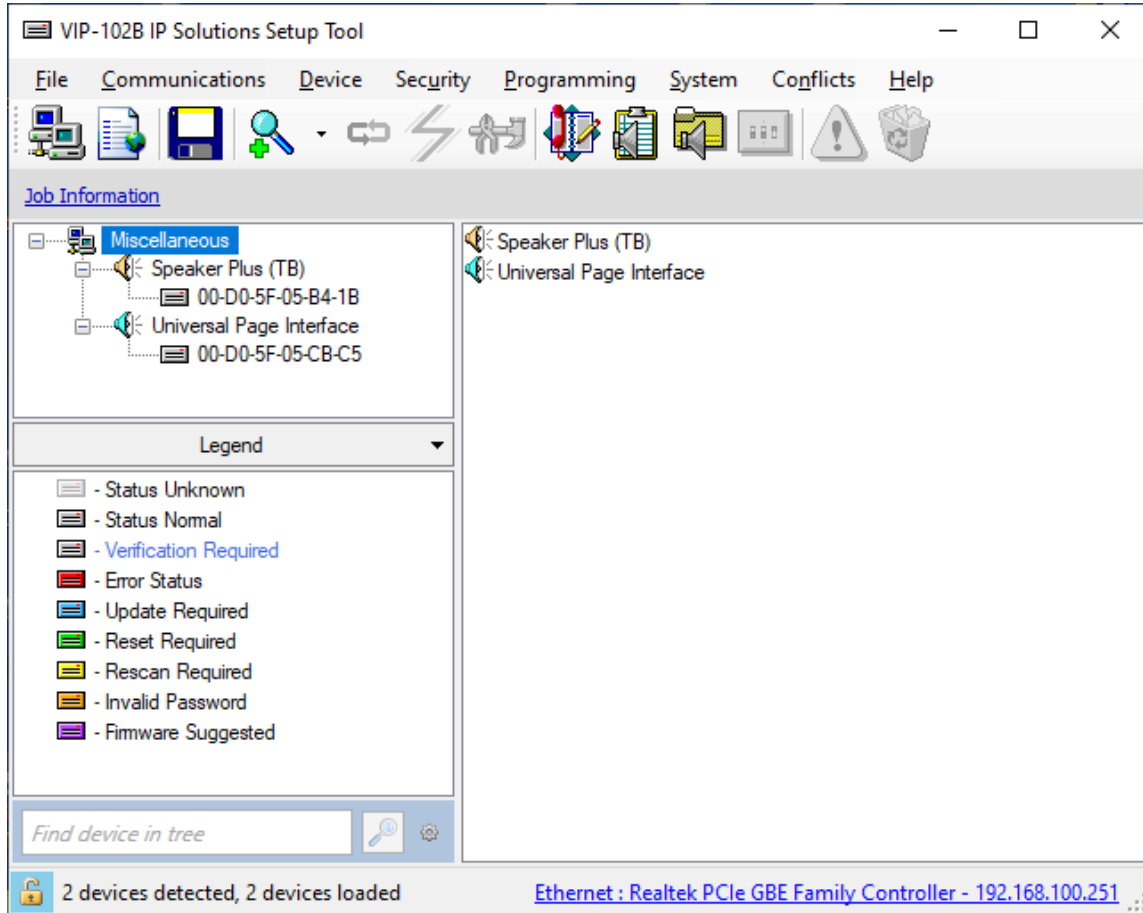
This section covers the configuration of Valcom V-9972 Universal Paging Interface using the Valcom VIP-102B IP Solutions Setup Tool. The configuration covers the following areas:

- Launch the Valcom VIP-102B IP Solutions Setup Tool
- Configure the Network Settings
- Configure Time
- Install System Manager CA TLS Certificate
- Configure SIP Parameters
- Verify Codec Settings
- Update Universal Paging Interface with the New Configuration

Note: These Application Notes do not cover the configuration of the Valcom VIP-430A IP Wall Speakers, Audio Groups, or the assignment of Dial Codes to Valcom speakers. Refer to [2] and [3] for details.

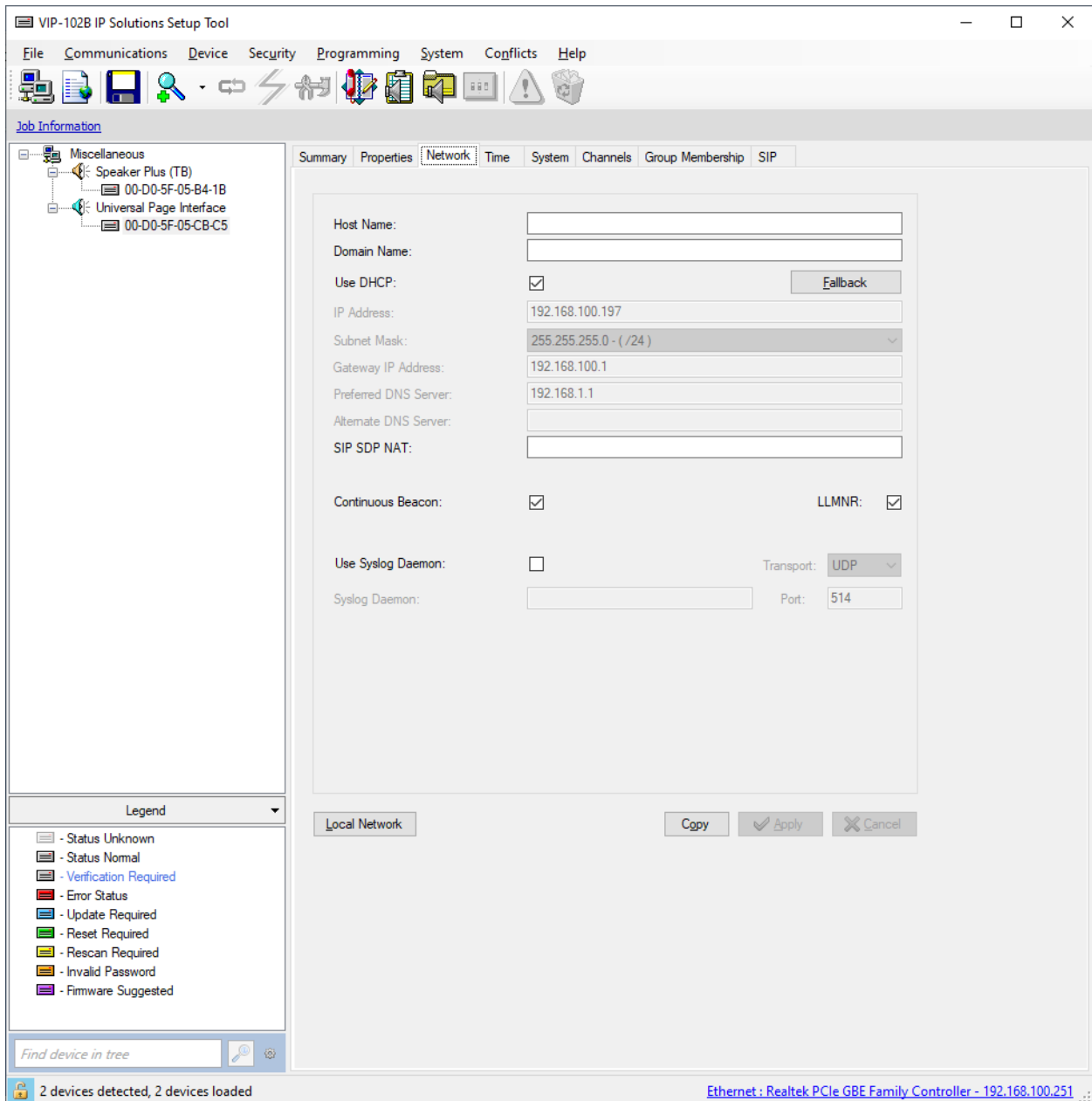
6.1. Launch Valcom VIP-102B IP Solutions Setup Tool

Launch the **VIP-102B IP Solutions Setup Tool** and follow the prompts. The main window is displayed as shown below.



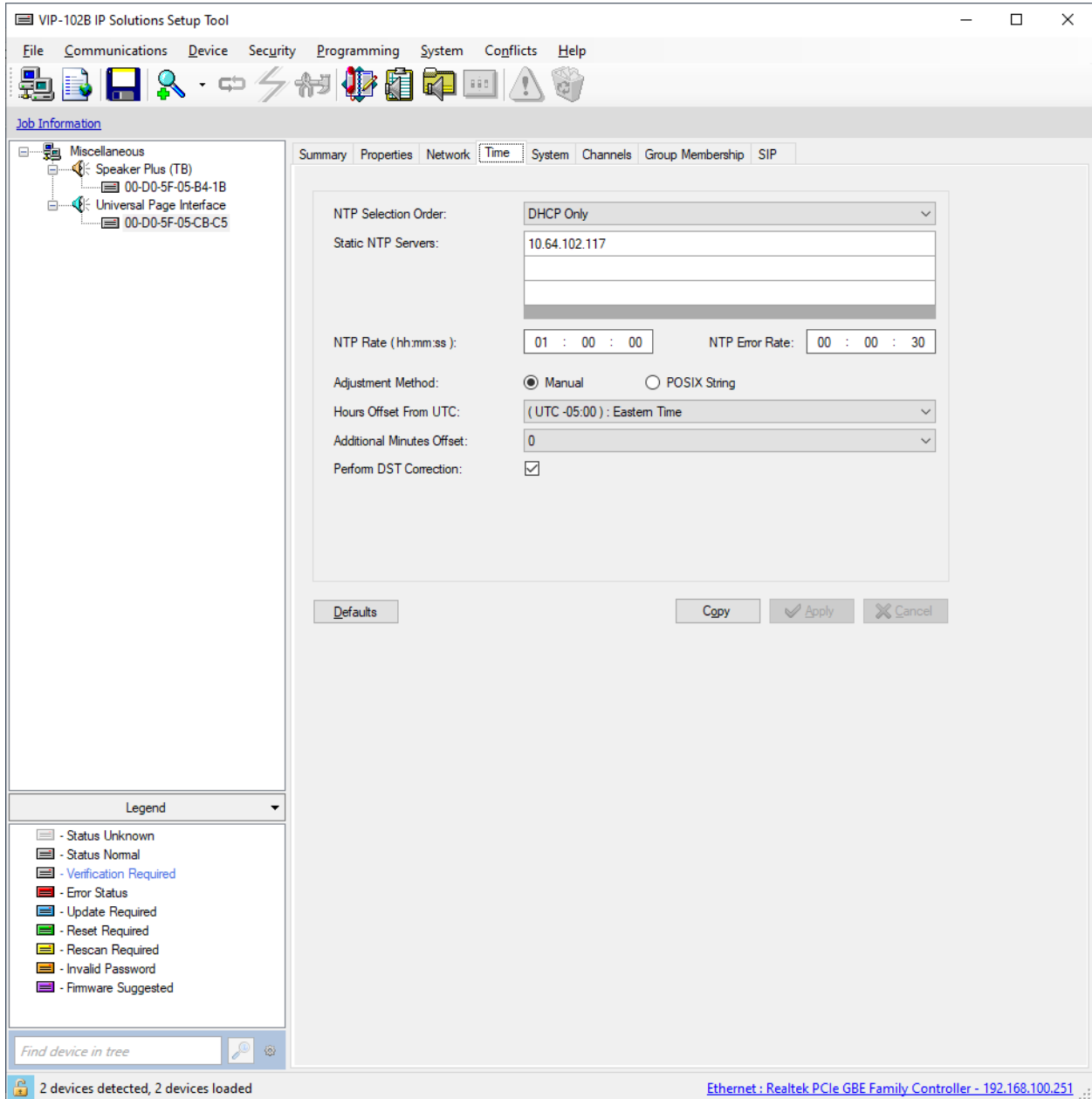
6.2. Configure the Network Settings

Click the MAC/hardware address under Universal Page Interface in the left pane and select the **Network** tab. V-9972 must first acquire IP network settings before proceeding with provisioning. These network settings were automatically obtained from a DHCP server as shown below. Alternatively, V-9972 could be configured with static IP addresses, but for the compliance test, DHCP was used.



6.3. Configure the Time

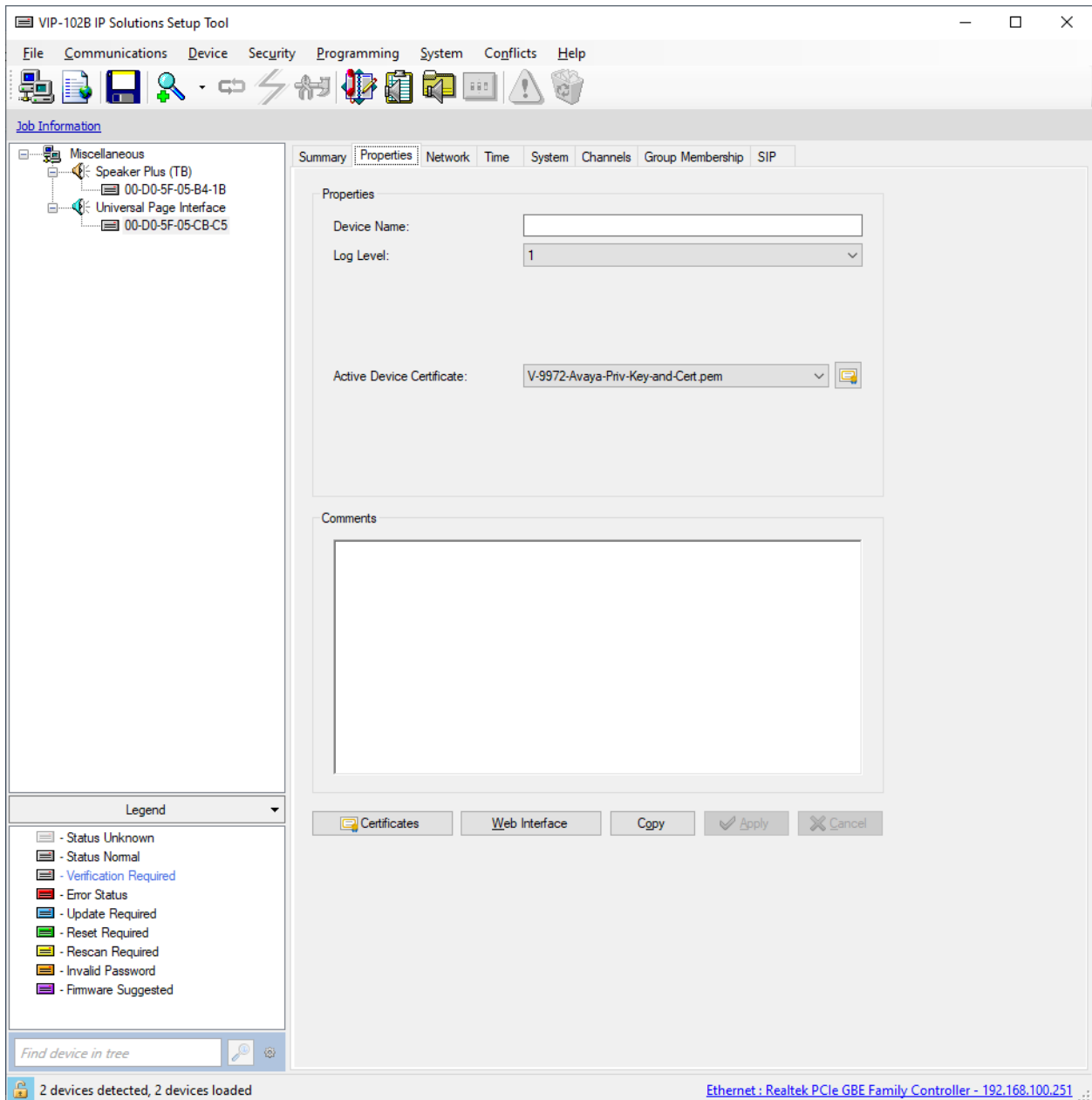
Navigate to the **Time** tab and set the Static NTP Servers to ensure the proper date/time on the device.



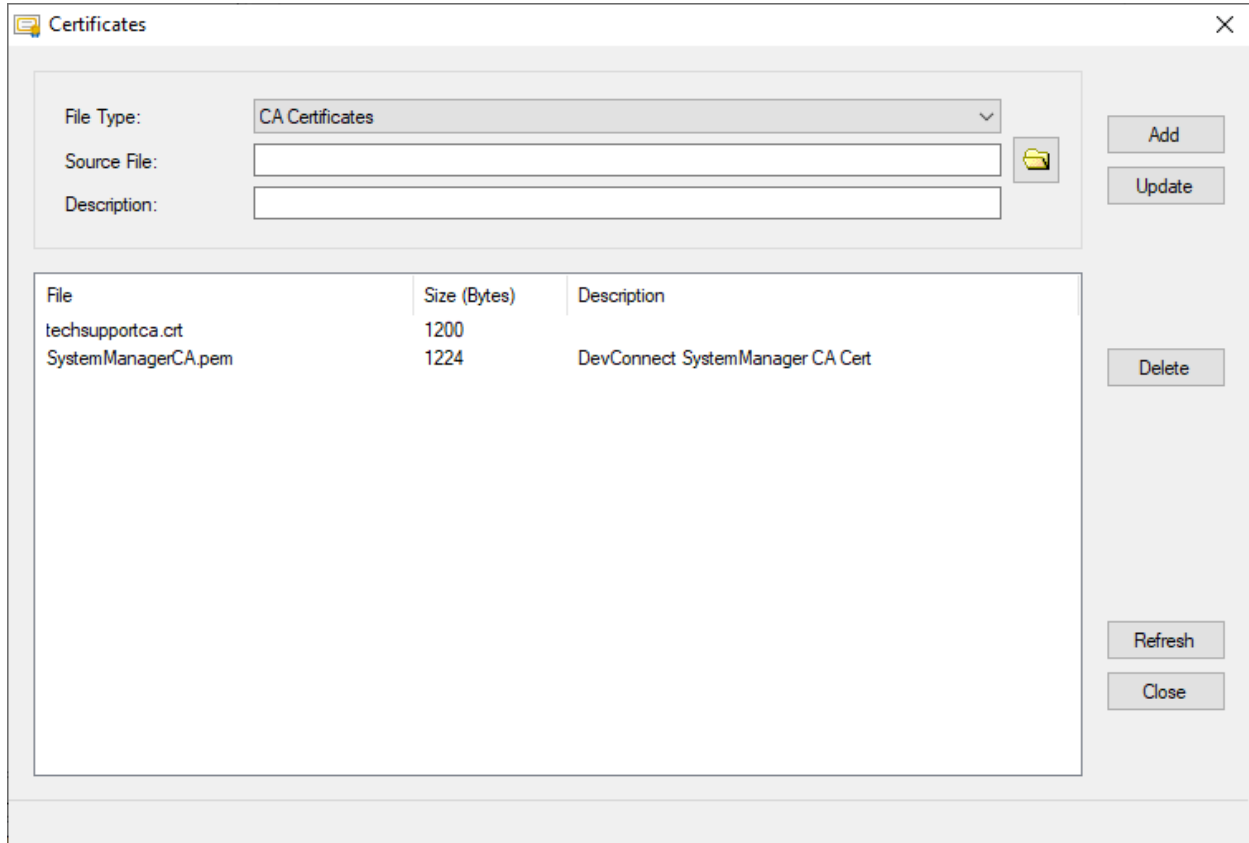
6.4. Install the System ManagerCA TLS Certificate

Navigate to the **Properties** tab to install the System Manager CA certificate. Note that the V-9972 has a device certificate (*V-9972-Avaya-Priv-Key-and-Cert.pem*) signed by a different CA other than the System Manager. Click on **Certificates**.

Note: Avaya Aura® System Manager was used as the Certificate Authority.



In the **Certificate** dialog box, add the System Manager CA TLS certificate. Note that the certificate has already been imported as shown below. In addition, the V-9972 root certificate (*techsupportca.crt*) is also installed. This certificate must be installed on IP Office to support mutual TLS authentication as shown in **Section 5.7**.



6.5. Configure SIP Parameters

From the **VIP-102B IP Solutions Setup Tool**, navigate to the **SIP** tab of the Universal Page Interface and configure the parameters as follows.

- **Transport:** Set to *Accept: TLS, Originate: TLS*.
- **Phone Number:** Set to number that will be routed to V-9972 (e.g., *41600*).
- **Description:** Provide optional description.
- **Authentication Name:** Leave blank.
- **Secret:** Leave blank.
- **Realm:** Set to SIP domain (e.g., *avaya.com*).
- **Validate Remote Certificate:** Enable this option so that V-9972 validates the remote TLS certificate installed in **Section 6.4**.
- **Primary Server:** Set to IP Office IP address (i.e., *10.64.102.90*).
- **Port:** Set to TLS port (e.g., *5061*).
- **Register:** Disable this option.
- **Max Calls:** Specify maximum number of calls (e.g., *4*). For example, V-9972 could establish an intercom call to the IP speaker and then a higher priority paging call to the same IP speaker. In addition, V-9972 could establish up to four calls to four different IP speakers (not tested).
- **SRTP:** Enable SRTP and then select *Media Encryption Mandatory*.
- **Auto Destination:** Set to the number that should be dialed when the call button on the VIP-430A IP Wall Speaker is pressed.

Accept the values in the remaining fields and click **Apply**.

VIP-102B IP Solutions Setup Tool

File Communications Device Security Programming System Conflicts Help

Job Information

Miscellaneous

- Speaker Plus (TB)
 - 00-D0-5F-05-B4-1B
 - Universal Page Interface
 - 00-D0-5F-05-CB-C5

Summary Properties Network Time System Channels Group Membership SIP

Transport: Accept: TLS, Originate: TLS

1 2 3 4

Phone Number: 41600

Description: VIP-430A

Authentication Name:

Secret:

Realm: avaya.com Validate Remote Certificate:

SIP Servers:

	Server	Port
Primary	10.64.102.90	5061
Backup 1		5061
Backup 2		5061
Backup 3		5061

Register:

DNS SRV:

Max Calls: 4 SRTP: Media Encryption Mandatory

Busy Message:

Call Fwd Busy (302): Ring Timeout (secs): None

Outbound Proxy: Outbound Port: 5061

Keep Alive Timer (secs): 600 Options Timer (secs): 40

SIP Port: 5061 Idle Timeout (secs): 0

RTP Port: 20000 Max Call Timer (secs): 0

Night Ring: Night Ring Group:

CID Number: 41600

CID Name: VIP-430A

Auto Destination: 41501

Channel Priority: Medium

Legend

- Status Unknown
- Status Normal
- Verification Required
- Error Status
- Update Required
- Reset Required
- Rescan Required
- Invalid Password
- Firmware Suggested

Find device in tree

2 devices detected, 2 devices loaded

Ethernet: Realtek PCIe GBE Family Controller - 192.168.100.251

Defaults Status Copy Apply Cancel

6.6. Verify Codec Settings

Navigate to the **Channels** tab shown below. The Codec Type should be set G.711, currently the only option supported with VIP-430A IP Wall Speaker.

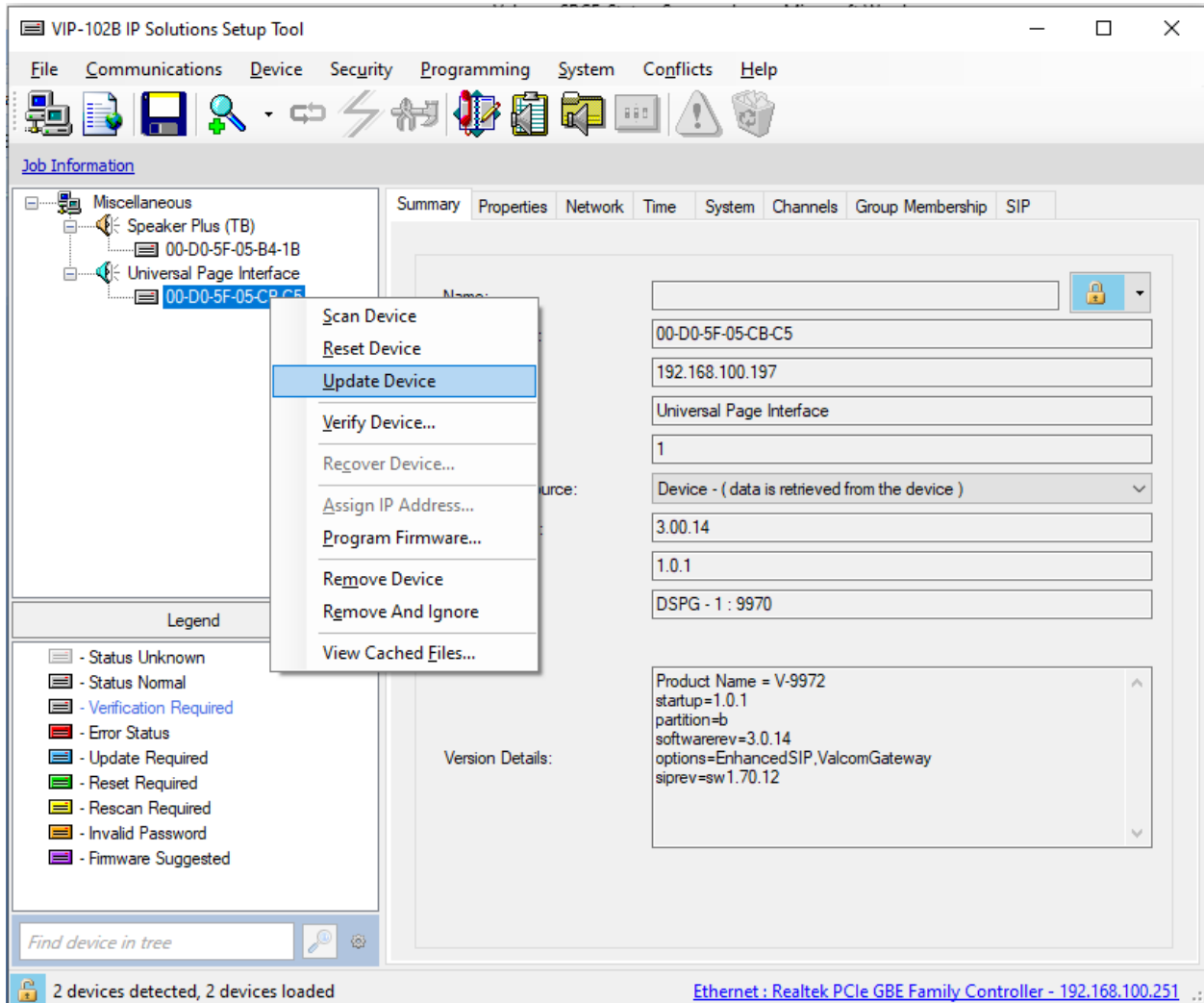
The screenshot displays the 'VIP-102B IP Solutions Setup Tool' interface. The 'Channels' tab is active, showing configuration fields for a channel. The fields are as follows:

- Dial Code: 802
- Description: (empty)
- CID Number: 802
- CID Name: (empty)
- Auto Destination: (empty)
- Codec Type: G.711 (selected in a dropdown menu)
- Call Fwd Busy: (empty)
- Call Fwd No Answer: (empty) with a '4 Rings' dropdown menu

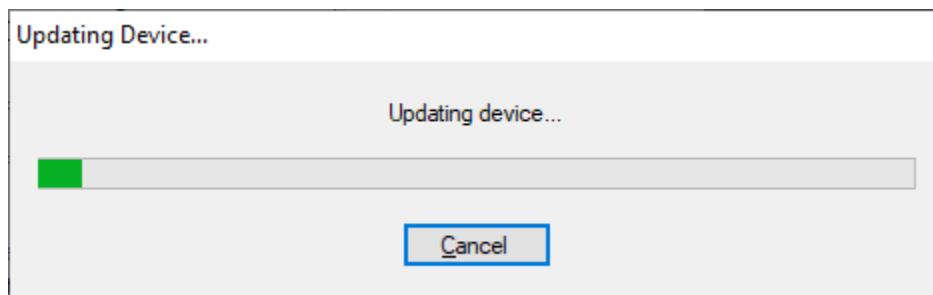
At the bottom of the configuration area, there are buttons for 'Group Membership', 'Sample Page', 'Copy', 'Apply', and 'Cancel'. The status bar at the bottom indicates '2 devices detected, 2 devices loaded' and the network address 'Ethernet : Realtek PCIe GBE Family Controller - 192.168.100.251'.

6.7. Update Universal Page Interface with the New Configuration

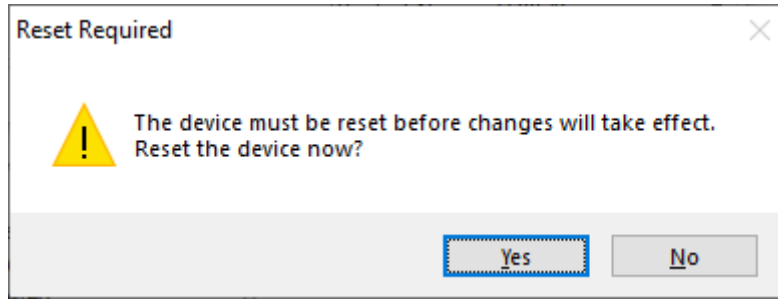
From the **VIP-102B IP Solutions Setup Tool**, right-mouse click on the MAC/hardware address of the Universal Page Interface and select **Update Device** from the pop-up menu as shown below.



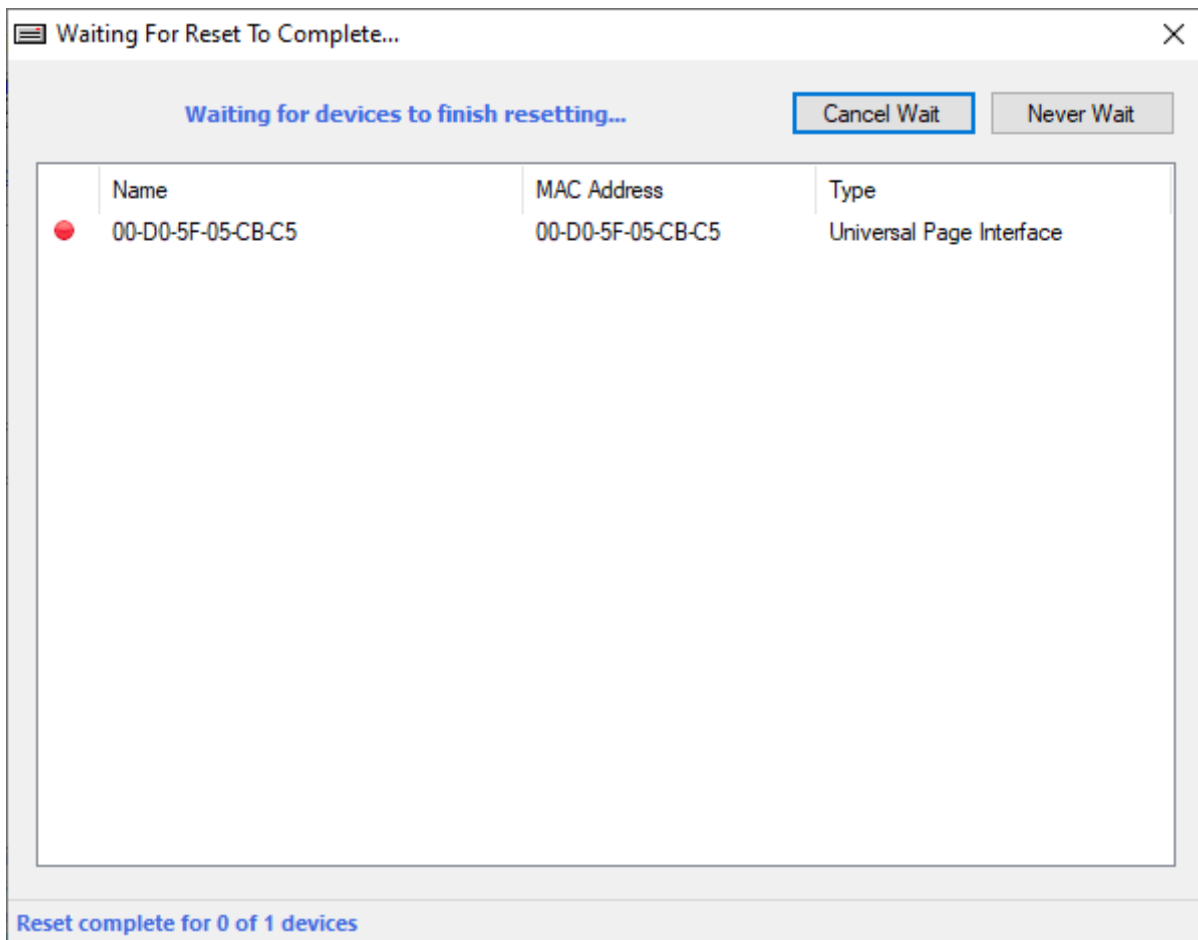
The following window is displayed indicating that the device is being updated.



A device reset is required so respond with **Yes** when prompted.



The following window will be displayed while the device is being reset. When the reset is completed, the window will disappear.



7. Verification Steps

This section provides the tests that may be performed to verify proper configuration of Valcom V-9972 Universal Paging Interface with Avaya IP Office Server Edition.

1. From **IP Office System Status**, expand **Trunks** in the left pane and select the SIP line from **Section 5.4**, in this case “5”.

Verify that the **SIP Trunk Summary** screen shows **Media Stream** is set to *SRTP*, **Layer 4 Protocol** is set to *TLS*, and **Current State** of channels is set to *Idle*.

The screenshot displays the Avaya IP Office System Status application. The left sidebar shows a tree view with 'Trunks (4)' expanded to 'Line: 5'. The main content area shows the 'SIP Trunk Summary' for Line 5. The summary includes the following details:

- Line Service State: In Service
- Peer Domain Name: avaya.com
- Resolved Address: 192.168.100.197
- Line Number: 5
- Number of Administered Channels: 10
- Number of Channels in Use: 0
- Administered Compression: G711 Mu
- Silence Suppression: Off
- Media Stream: SRTP
- Layer 4 Protocol: TLS
- SIP Trunk Channel Licenses: 256
- SIP Trunk Channel Licenses in Use: 0 (0%)
- SIP Device Features: REFER (Incoming and Outgoing)

Below the summary is a table with the following columns: Channel Number, URI G..., Ref, Call 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, and Transmit Packe... The table contains 10 rows, all with 'Idle' in the Call State column and '00:03:19' in the Time in State column.

1. Place a call to the V-9972 and at the dial tone, enter the dial code for the IP speaker to establish an intercom call from an Avaya IP deskphone to a Valcom speaker. Verify two-way audio. Terminate the call from the Avaya IP deskphone or by pressing the call button on the IP speaker.
2. Place a call to the V-9972 and at the dial tone, enter the dial code a group page code to establish a one-way paging call from an Avaya IP deskphone to IP speaker(s). Verify one-way audio. Terminate the call from the Avaya IP deskphone.

3. Place an intercom call by pressing the call button on the IP speaker. Verify two-way audio to the call destination. Terminate the call.

8. Conclusion

These Application Notes described the configuration steps required to integrate Valcom V-9972 Universal Paging Interface with Avaya IP Office Server Edition using a SIP trunk. Intercom and group paging calls were established with Valcom V-9972 Universal Paging Interface, Valcom VIP-430A IP Wall Speakers, Avaya H.323 / SIP Deskphones, and the PSTN. All feature and serviceability test cases were completed successfully.

9. References

This section references the Avaya and Valcom documentation relevant to these Application Notes.

- [1] *Administering Avaya IP Office Platform using Manager*, Release 11.1, August 16, 2021.
- [2] *Valcom VIP-102B IP Solutions Setup Tool Version 8.4.0.0 Reference Manual*, Revision 17 – 3/16/22, available at <https://www.valcom.com/resources/documents-manuals>.
- [3] *Valcom V-9972 Universal Page Interface Configuration Guide*, Rev. 3.1, available at <https://www.valcom.com/resources/documents-manuals>.

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Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.



Declaration of Conformance

May 20, 2022

Jeff Gartner
Senior Manager
DevConnect Program
Avaya

Dear Jeff Gartner:

We, Valcom Inc, declare under sole responsibility that product series named Universal Paging Adapter, including product models V-9972, V-9972-2 or VRCPA share the same hardware circuitry, software, SIP stack and firmware version. Therefore, the products are expected to behave in the same manner. The differences between the different models in each series are generally cosmetic in nature, such as enclosure shape or color, mounting arrangement, etc.

Sincerely,

/s/ David Ellison

David Ellison
Technical Support Manager
Valcom Inc
dellison@valcom.com