



Application Notes for Valcom VE8090R SIP Intercom Controller with Avaya IP Office Server Edition using SIP Endpoint - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Valcom VE8090R SIP Intercom Controller with Avaya IP Office Server Edition. Valcom VE8090R SIP Intercom Controller provides access to Valcom VoIP audio endpoints, such as Valcom VIP-430A IP Wall Speakers, from telephone servers. For this compliance test, Valcom VE8090R SIP Intercom Controller interfaced with Avaya IP Office Server Edition as a SIP endpoint. The Valcom VE8090R SIP Intercom Controller supports two-way audio intercom 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 VE8090R SIP Intercom Controller with Avaya IP Office Server Edition. Valcom VE8090R SIP Intercom Controller provides access to Valcom VoIP audio endpoints, such as Valcom VIP-430A IP Wall Speakers, from telephone servers. For this compliance test, Valcom VE8090R SIP Intercom Controller interfaced with Avaya IP Office Server Edition as a SIP endpoint. The Valcom VE8090R SIP Intercom Controller supports two-way audio intercom calls and one-way audio group paging calls.

When the Valcom VE8090R SIP Intercom Controller is configured in SIP station mode and a call is placed to it using its direct dial SIP extension, the VE8090R 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 call (one-way audio) to one or more Valcom speakers.

In addition, the VIP-430A IP Wall Speaker established intercom calls by pressing the call button. Pressing the call button would place a call to the specified destination in the VE8090R configuration. Pressing the call button during an active call, terminates the call.

In SIP station mode with a single SIP line/extension, VE8090R can support one call at a time. However, VE8090R can support up to four SIP lines/extensions.

Note: Valcom has indicated that other products in the SIP Intercom Controller family share the same hardware circuitry, software, SIP stack and firmware version 3.20.14, which was compliance tested; therefore, this testing also applies to those products. The differences between the products are detailed in **Attachment 1**. 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 VE8090R SIP Intercom Controller using the Valcom VIP-430A IP Wall Speakers, Avaya SIP / H.323 IP Deskphones, and the PSTN. Two-way audio intercom calls and one-way audio group paging calls were exercised.

The serviceability testing focused on verifying that the Valcom VE8090R SIP Intercom Controller came back into service after 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 VE8090R SIP Intercom Controller did not include use of any specific encryption features as requested by Valcom.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of VE8090R with IP Office Server Edition.
- Establishing two-way audio intercom calls between VIP-430A IP Wall Speaker, via VE8090R, Avaya H.323 / SIP Deskphones, and PSTN in both directions.
- Establishing one-way group paging calls from Avaya H.323 / SIP Deskphones to VIP-430A IP Wall Speakers via VE8090R.
- Verifying that higher priority group calls have precedence over active, lower priority group calls.
- Terminating active calls by pressing the call button on the VIP-430A IP Wall Speaker.
- Establishing calls between VE8090R and Avaya H.323 / SIP Deskphones registered to IP Office Server Edition and IP Office 500 V2 Expansion.
- Support of G.711 mu-law codec and UDP transport protocol.
- Support of direct IP-to-IP media, which allows IP endpoints to send audio RTP packets directly to each other without using media resources on IP Office.
- Proper system recovery after a restart of VE8090R.

2.2. Test Results

All test cases passed.

2.3. Support

For technical support and information on Valcom VE8090R SIP Intercom Controller, 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, Avaya J129 SIP Deskphones, and Avaya 1100/1200 Series SIP Deskphones registered to Avaya IP Office Server Edition and Avaya IP Office 500 V2 Expansion.
- Valcom VE8090R SIP Intercom Controller provided connectivity to Avaya IP Office Server Edition as a SIP endpoint. Valcom VIP-430A IP Wall Speakers serving as the Valcom audio endpoint, and Valcom VIP-102B IP Solutions Setup Tool to configure Valcom products.

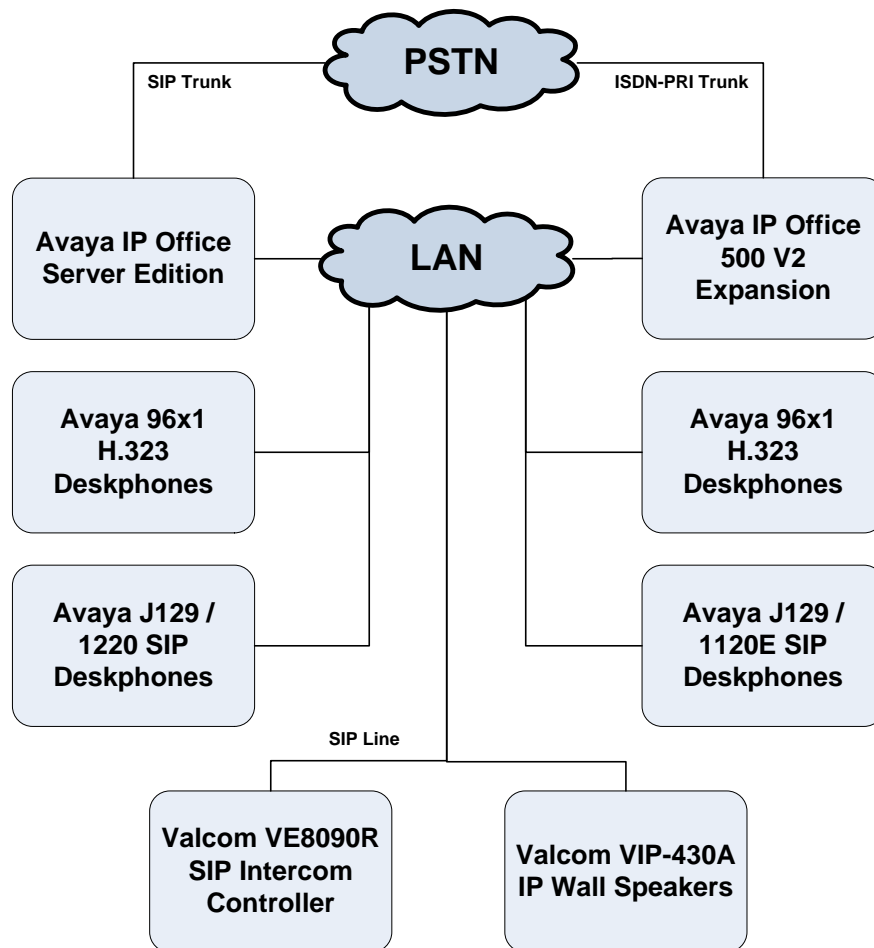


Figure 1: Avaya SIP Network with Valcom VE8090R SIP Intercom Controller and Valcom VIP-430A IP Wall Speakers

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	11.0.0.2.0 Build 23
Avaya IP Office 500 V2 Expansion	11.0.0.2.0 Build 23
Avaya 96x1 Series IP Deskphones	6.6604 (H.323)
Avaya 1100/1200 Series IP Deskphones	04.04.26.00 (SIP)
Avaya J129 SIP Deskphones	3.0.0.1.6 (6)
Valcom VE8090R SIP Intercom Controller	3.20.14
Valcom VIP-430A IP Wall Speaker	3.20.15
Valcom VIP-102B IP Solutions Setup Tool on Windows 10	7.5.0.0

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

Configure Avaya IP Office Server Edition

This section provides the procedure for configuring Avaya IP Office Server Edition. The procedure includes the following areas:

- Verify IP Office License
- Obtain LAN IP Address
- Administer SIP Registrar
- Administer SIP Extension
- Administer SIP User

Note: Integration of IP Office 500 V2 Expansion and call routing to the PSTN are outside the scope of these Application Notes.

4.1. 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.

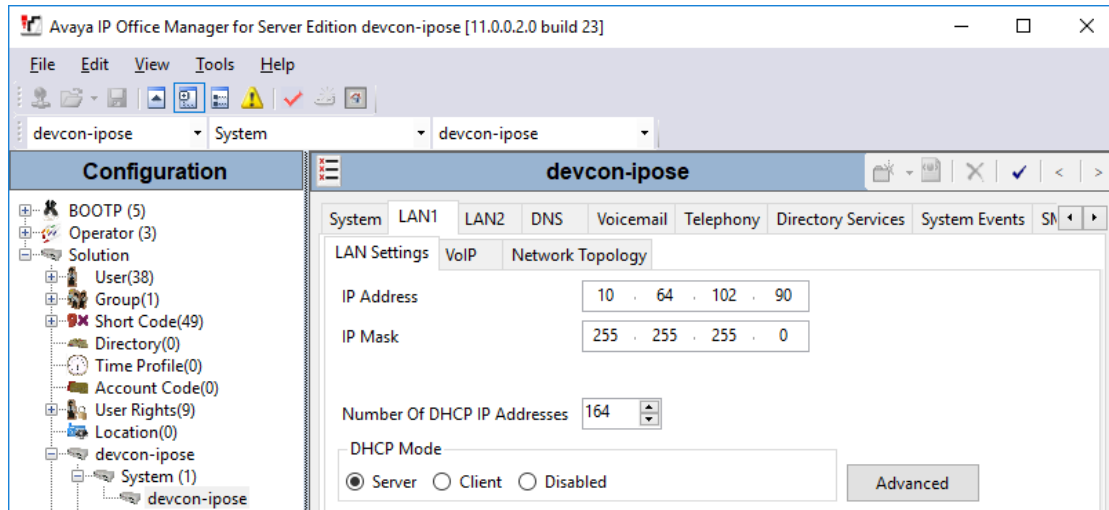
The **Avaya IP Office Manager for Server Edition** screen is displayed. From the configuration tree in the left pane, select **License** to display the license screen in the right pane. Verify that the **License Status** is “Valid” for **3rd Party IP Endpoints**.

The screenshot shows the Avaya IP Office Manager for Server Edition application window. The title bar reads "Avaya IP Office Manager for Server Edition devcon-ipose [11.0.0.2.0 build 23]". The interface includes a menu bar (File, Edit, View, Tools, Help) and a toolbar. The left pane shows a "Configuration" tree with various system components. The right pane displays the "License" configuration screen, which includes a table of license features and their status.

Feature	Instances	Status	Expiration Date	Source
Receptionist	10	Valid	Never	PLDS Nodal
Additional Voicemail Pro Ports	252	Valid	Never	PLDS Nodal
VMPro Recordings Administrators	1	Valid	Never	PLDS Nodal
Office Worker	1000	Valid	Never	PLDS Nodal
VMPro 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
Web Collaboration	64	Valid	Never	PLDS Nodal
Avaya Contact Center Select	1	Valid	Never	PLDS Nodal

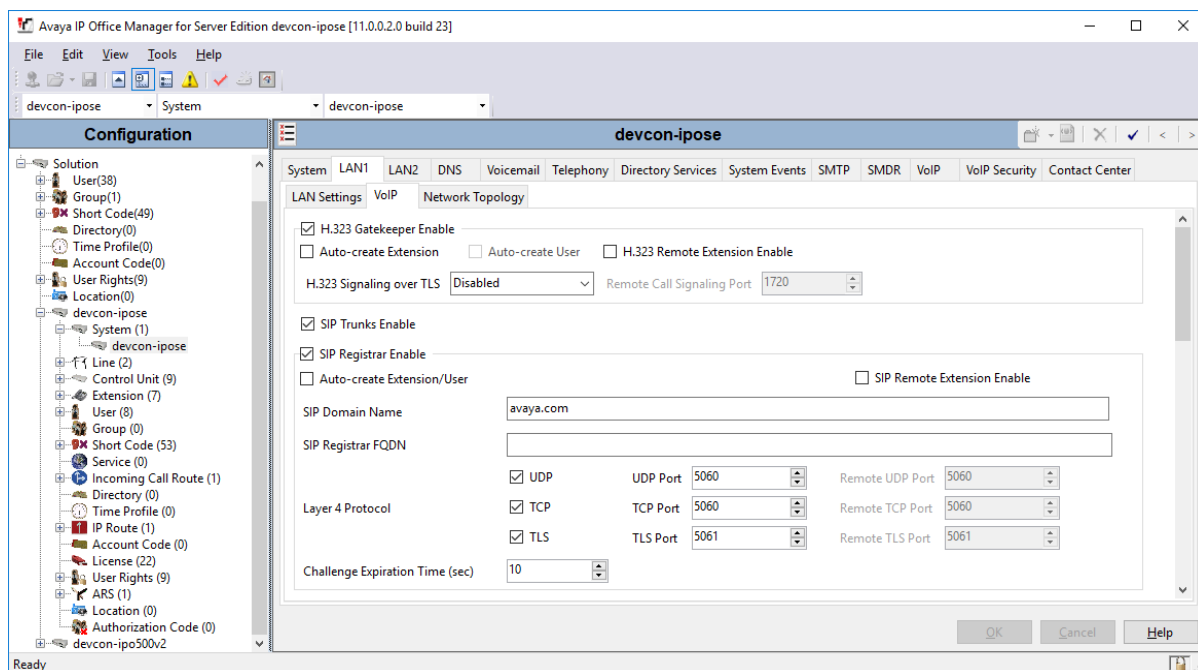
4.2. Obtain LAN IP Address

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 VE8090R.



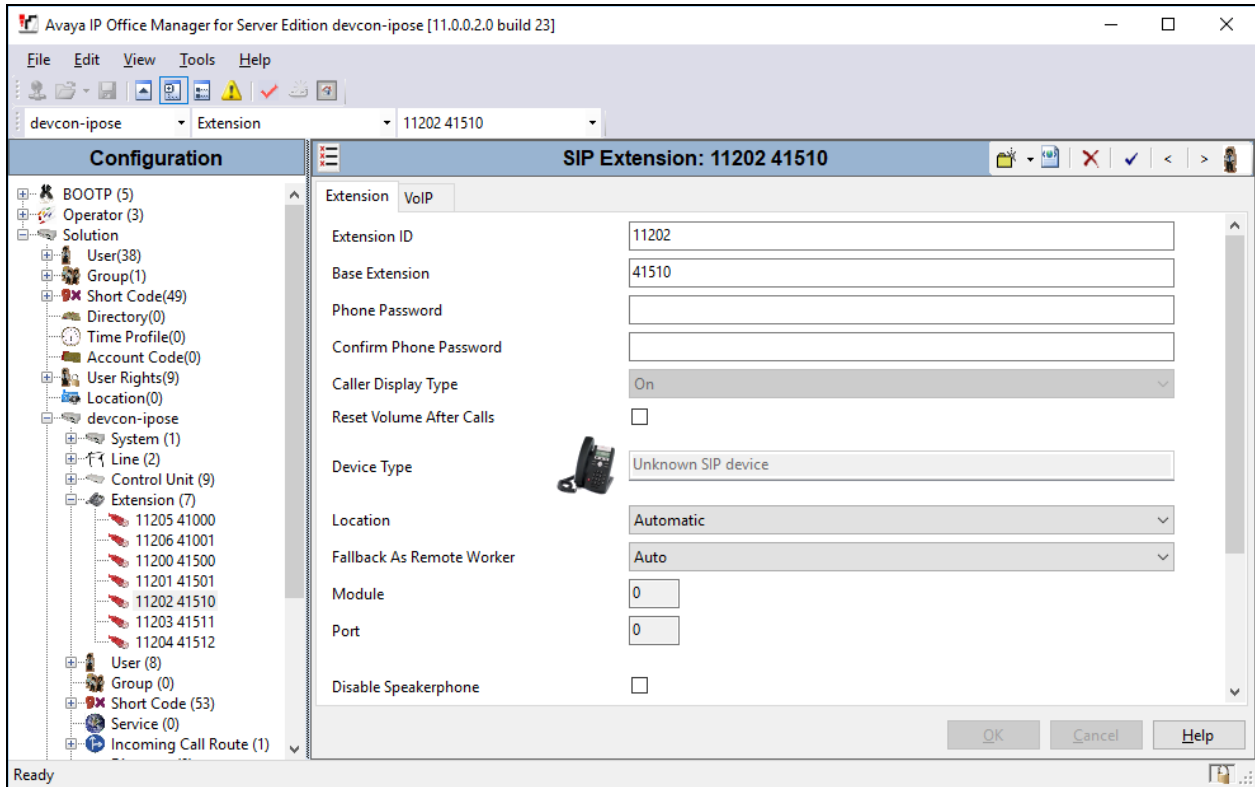
4.3. Administer SIP Registrar

Select the **VoIP** sub-tab. Ensure that **SIP Registrar Enable** and that UDP transport is selected, which will be used by VE8090R. Also, enter a valid **Domain Name**.

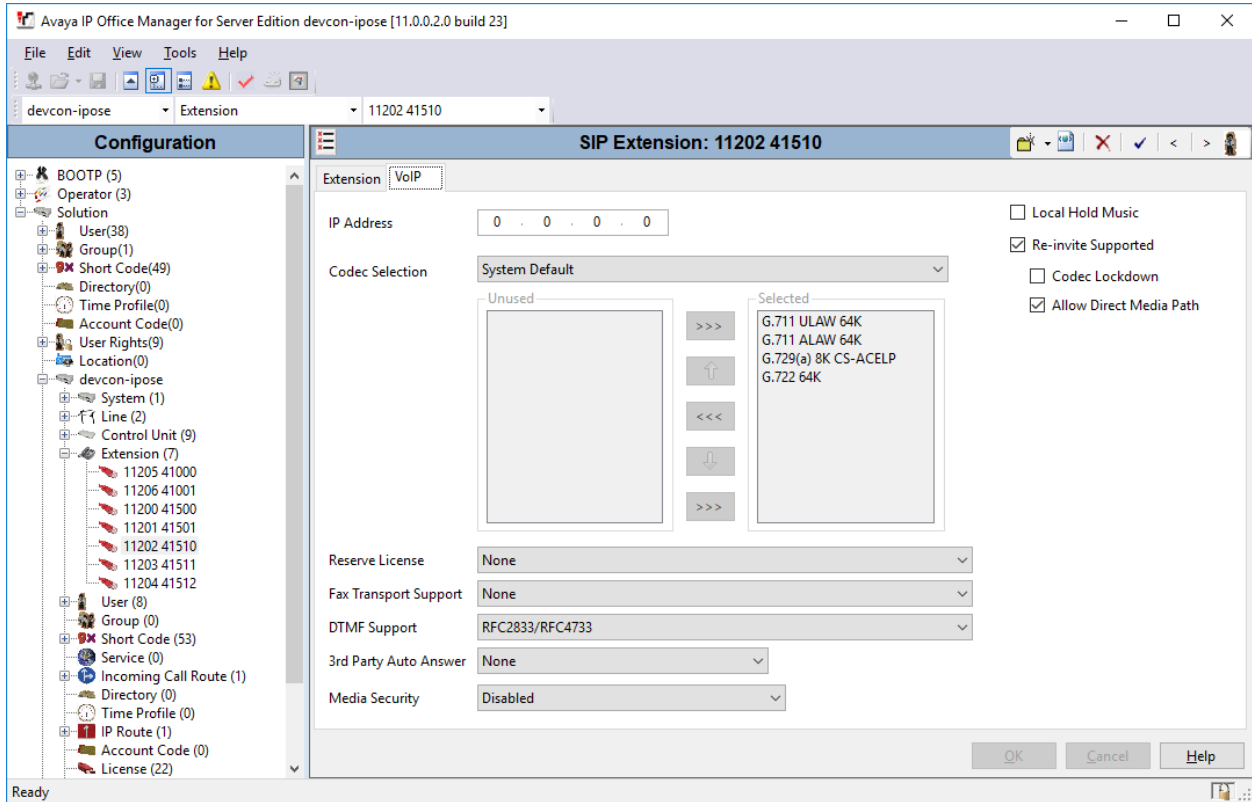


4.4. Administer SIP Extension

From the configuration tree in the left pane, right-click on **Extension** and select **New** → **SIP** from the pop-up list to add a new SIP extension (not shown). Enter the desired extension for the **Base Extension** field as shown below. In this example, VE8090R was assigned extension 41510.



Select the **VoIP** tab and retain the default values in the all fields. VE8090R supports G.711 codec only.



4.5. Administer SIP User

From the configuration tree in the left pane, right-click on **User** and select **New** from the pop-up list. Enter desired values for the **Name** and **Full Name** fields. For the **Extension** field, enter the SIP extension created in **Section 4.4**. The **Extension** field specifies the username that will be used by VE8090R to register with IP Office Server Edition. Note that the SIP authentication password was configured in the **Telephony** tab below instead of in the **Password** field shown below.

The screenshot displays the Avaya IP Office Manager for Server Edition interface. The left pane shows a configuration tree with the 'User' folder expanded, and the '41510 Valcom41510' user selected. The right pane shows the configuration details for this user, with the 'Telephony' tab active. The configuration fields are as follows:

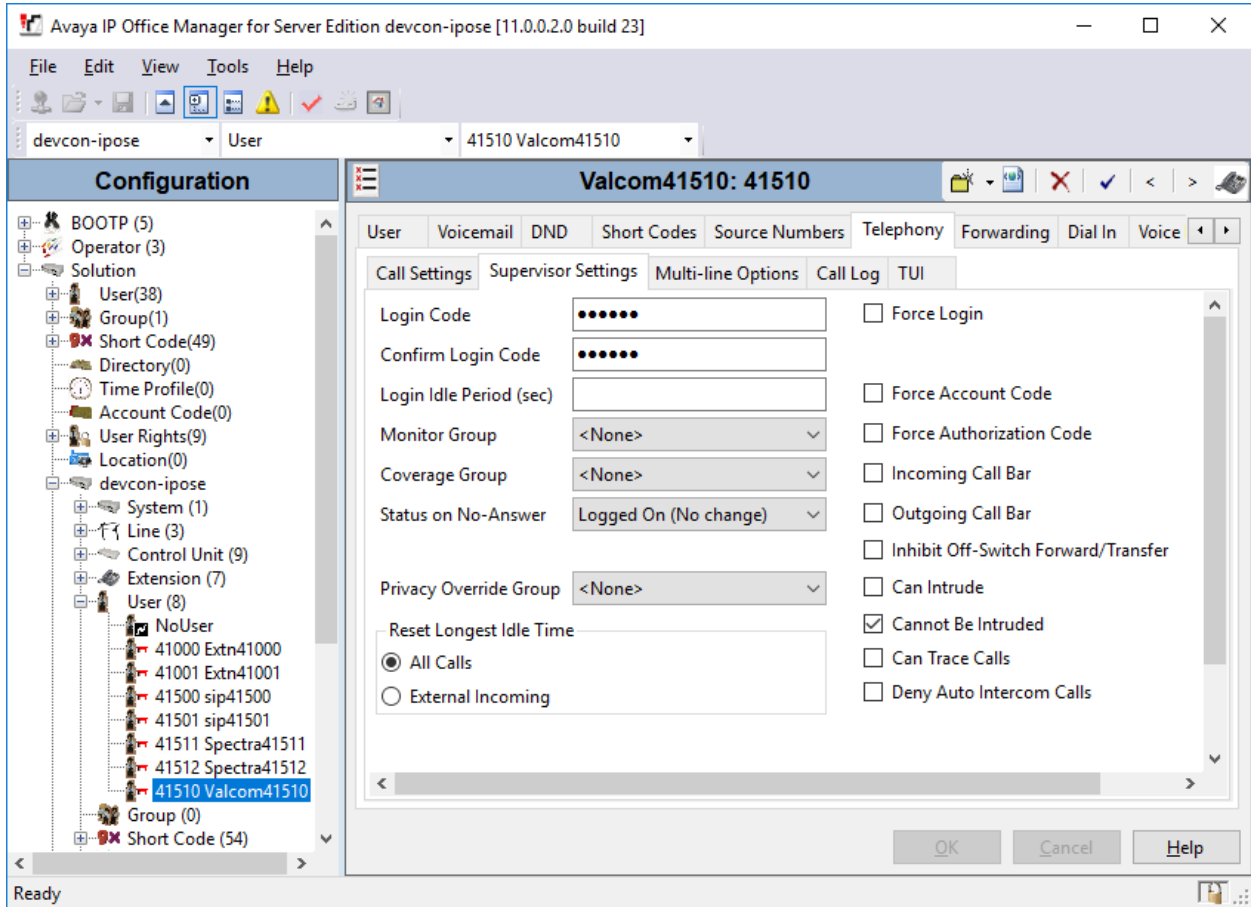
Field	Value
Name	Valcom41510
Password	
Confirm Password	
Unique Identity	
Conference PIN	
Confirm Audio Conference PIN	
Account Status	Enabled
Full Name	Valcom
Extension	41510
Email Address	
Locale	
Priority	5
System Phone Rights	None
Profile	Basic User

Below the profile dropdown, several checkboxes are visible, all of which are currently unchecked:

- Receptionist
- Enable Softphone
- Enable one-X Portal Services
- Enable one-X TeleCommuter
- Enable Remote Worker
- Enable Desktop/Tablet VoIP client
- Enable Mobile VoIP Client
- Send Mobility Email
- Web Collaboration

The interface includes a menu bar (File, Edit, View, Tools, Help), a toolbar, and a status bar at the bottom showing 'Ready'.

Select the **Supervisor Settings** sub-tab and enter a desired **Login Code**. The **Login Code** is the password that will be used by VE8090R to register with IP Office Server Edition.



5. Configure Valcom VE8090R SIP Intercom Controller

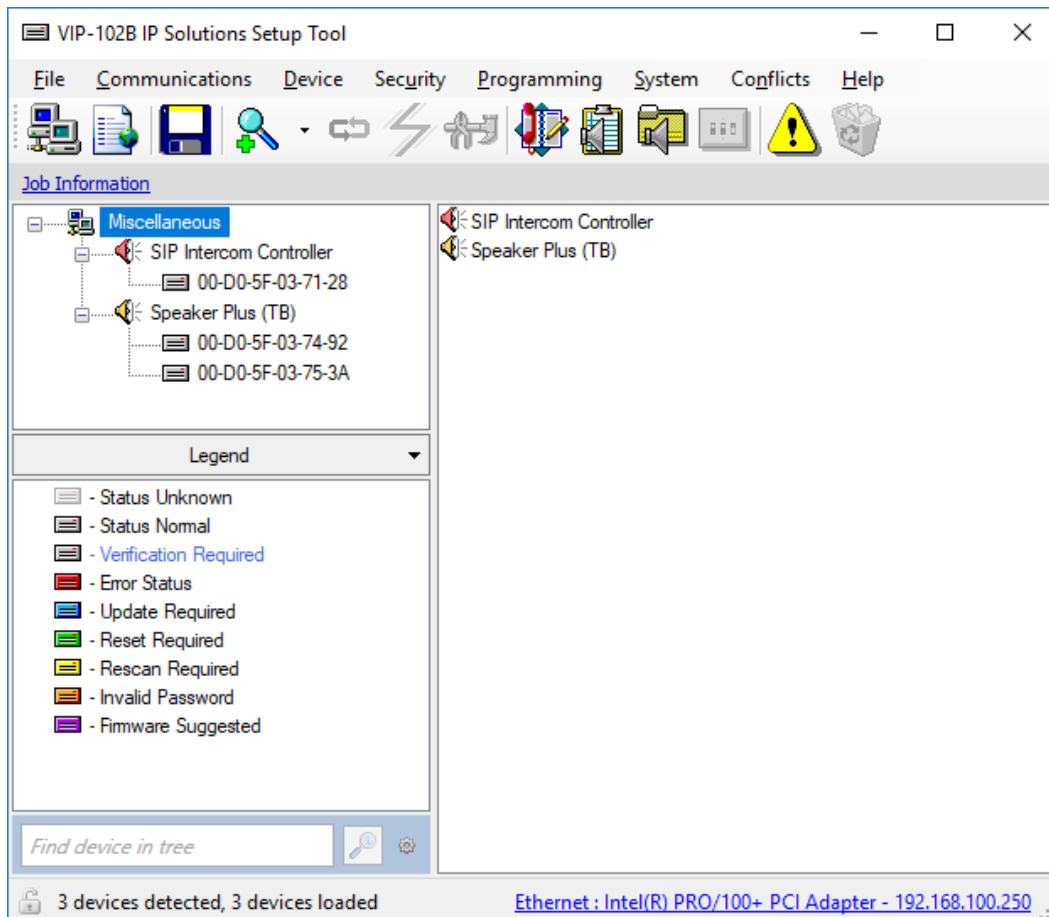
This section covers the configuration of VE8090R using the Valcom VIP-102B IP Solutions Setup Tool. The configuration covers the following areas:

- Launch Valcom VIP-102B IP Solutions Setup Tool
- Configure the Network Settings of Valcom VE8090R SIP Intercom Controller
- Configure SIP Parameters of Valcom VE8090R SIP Intercom Controller
- Verify Codec Settings
- Update SIP Intercom Controller 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] for details.

5.1. 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.



5.2. Configure the Network Settings of Valcom VE8090R SIP Intercom Controller

Click the MAC/hardware address under SIP Intercom Controller in the left pane and select the **Network** tab. VE8090R must first acquire IP network settings before proceeding with provisioning. These network settings were automatically obtained from a DHCP server as shown below. Alternatively, VE8090R could be configured with static IP addresses, but for the compliance test, a DHCP server was used.

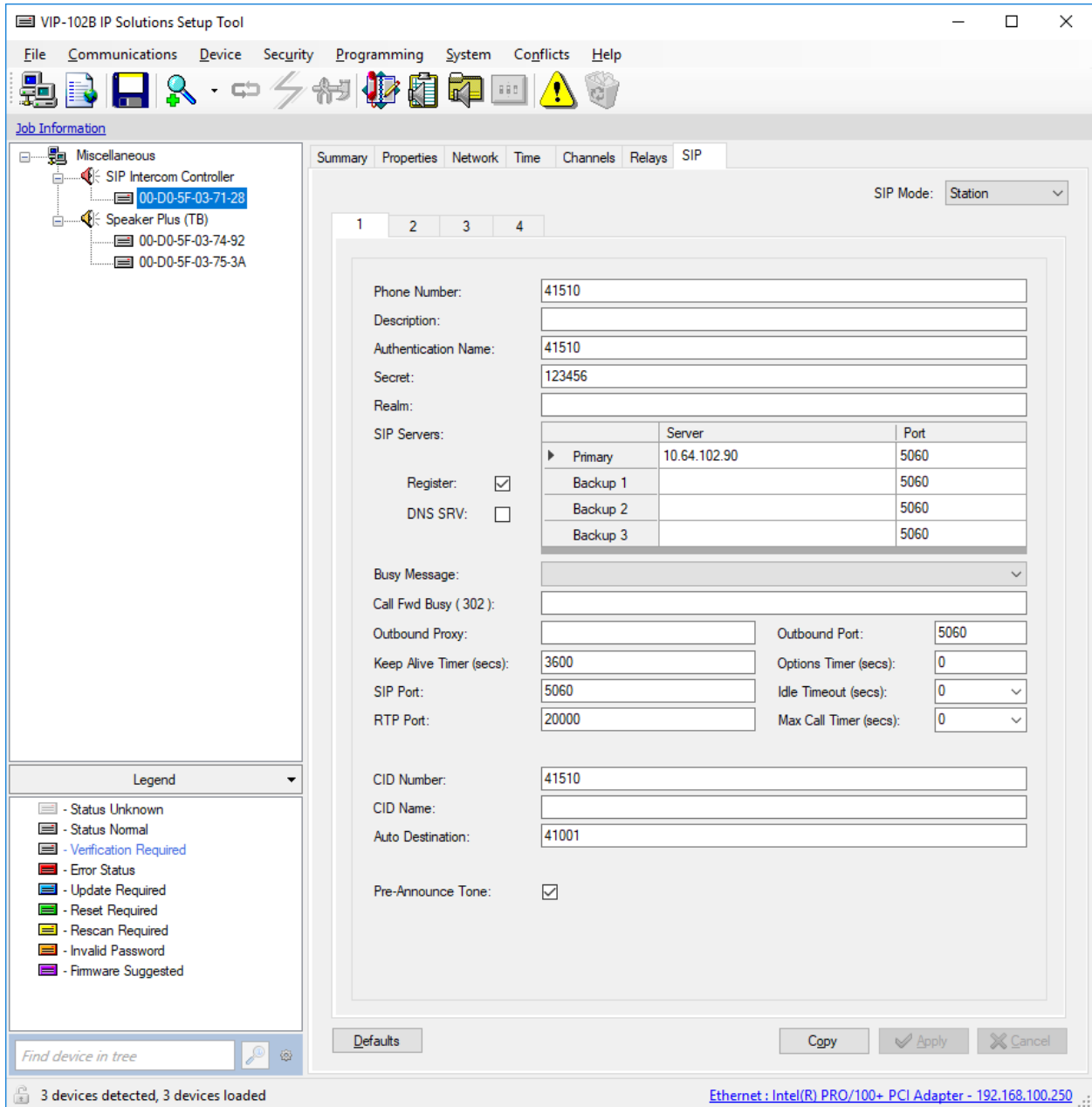
The screenshot displays the 'VIP-102B IP Solutions Setup Tool' interface. The left pane shows a tree view under 'Miscellaneous' with three devices: 'SIP Intercom Controller' (MAC: 00-D0-5F-03-71-28), 'Speaker Plus (TB)' (MAC: 00-D0-5F-03-74-92), and another 'Speaker Plus (TB)' (MAC: 00-D0-5F-03-75-3A). The 'SIP Intercom Controller' is selected, and the 'Network' tab is active. The network configuration fields are as follows:

Host Name:	<input type="text"/>
Domain Name:	<input type="text"/>
Use DHCP:	<input checked="" type="checkbox"/>
IP Address:	192.168.100.193
Subnet Mask:	255.255.255.0
Gateway IP Address:	192.168.100.1
Preferred DNS Server:	192.168.1.1
Alternate DNS Server:	<input type="text"/>
SIP SDP NAT:	<input type="text"/>
Continuous Beacon:	<input checked="" type="checkbox"/>
Use Syslog Daemon:	<input type="checkbox"/>
Syslog Daemon:	<input type="text"/>

At the bottom, there are buttons for 'Local Network', 'Copy', 'Apply', and 'Cancel'. The status bar at the bottom indicates '3 devices detected, 3 devices loaded' and the active network interface: 'Ethernet : Intel(R) PRO/100+ PCI Adapter - 192.168.100.250'.

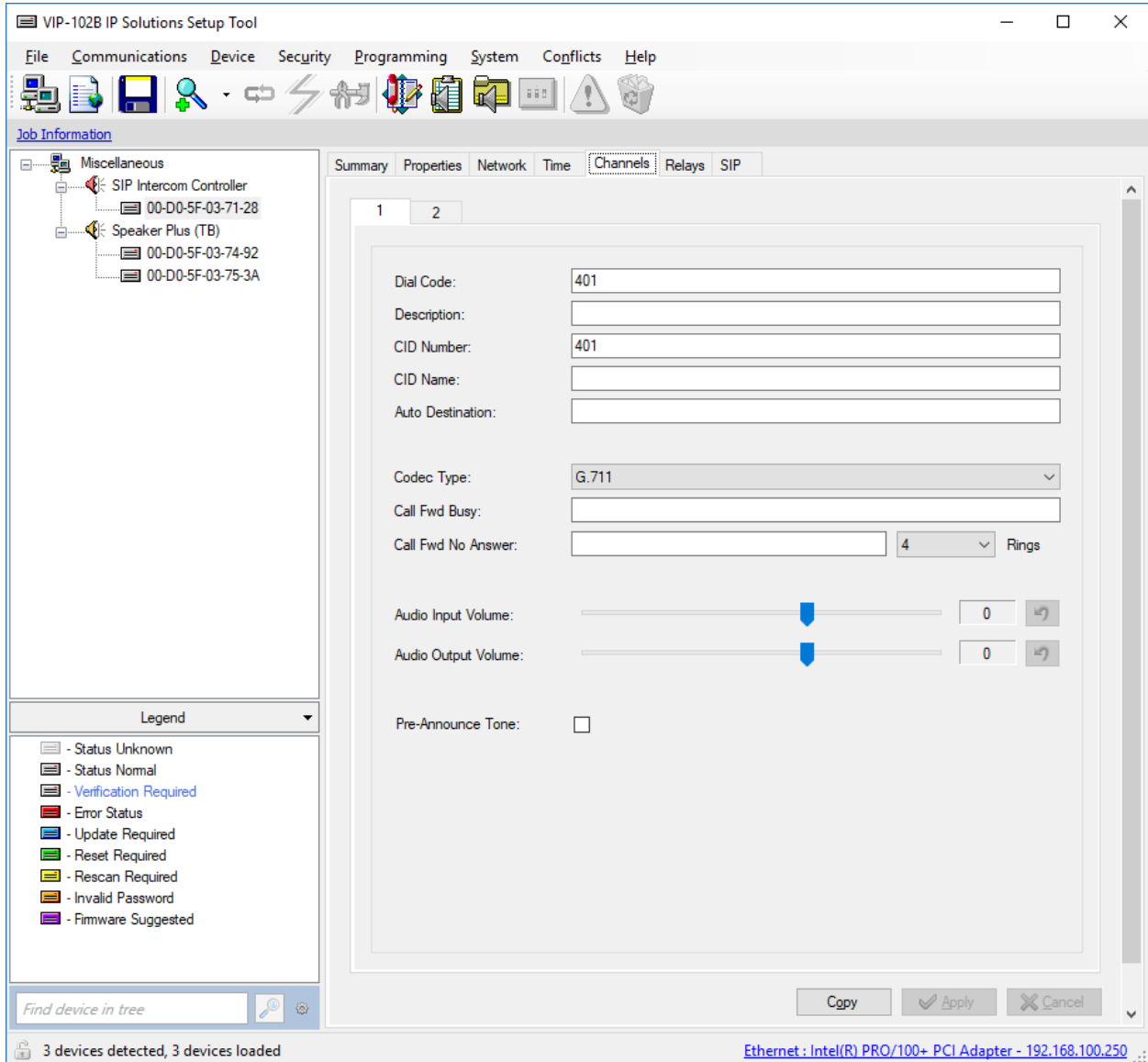
5.3. Configure SIP Parameters of Valcom VE8090R SIP Intercom Controller

From the **VIP-102B IP Solutions Setup Tool**, navigate to the **SIP** tab of the SIP Intercom Controller. Set the **SIP Mode** to *Station*, the **Primary Server** to the IP Office Server Edition IP address (i.e., *10.64.102.90*), and the **Auto Destination** to the number that should be dialed when the call button on the VIP-430A IP Wall Speaker is pressed as shown below. Select the **Register** checkbox and leave all other fields at their default values. Click **Apply**.



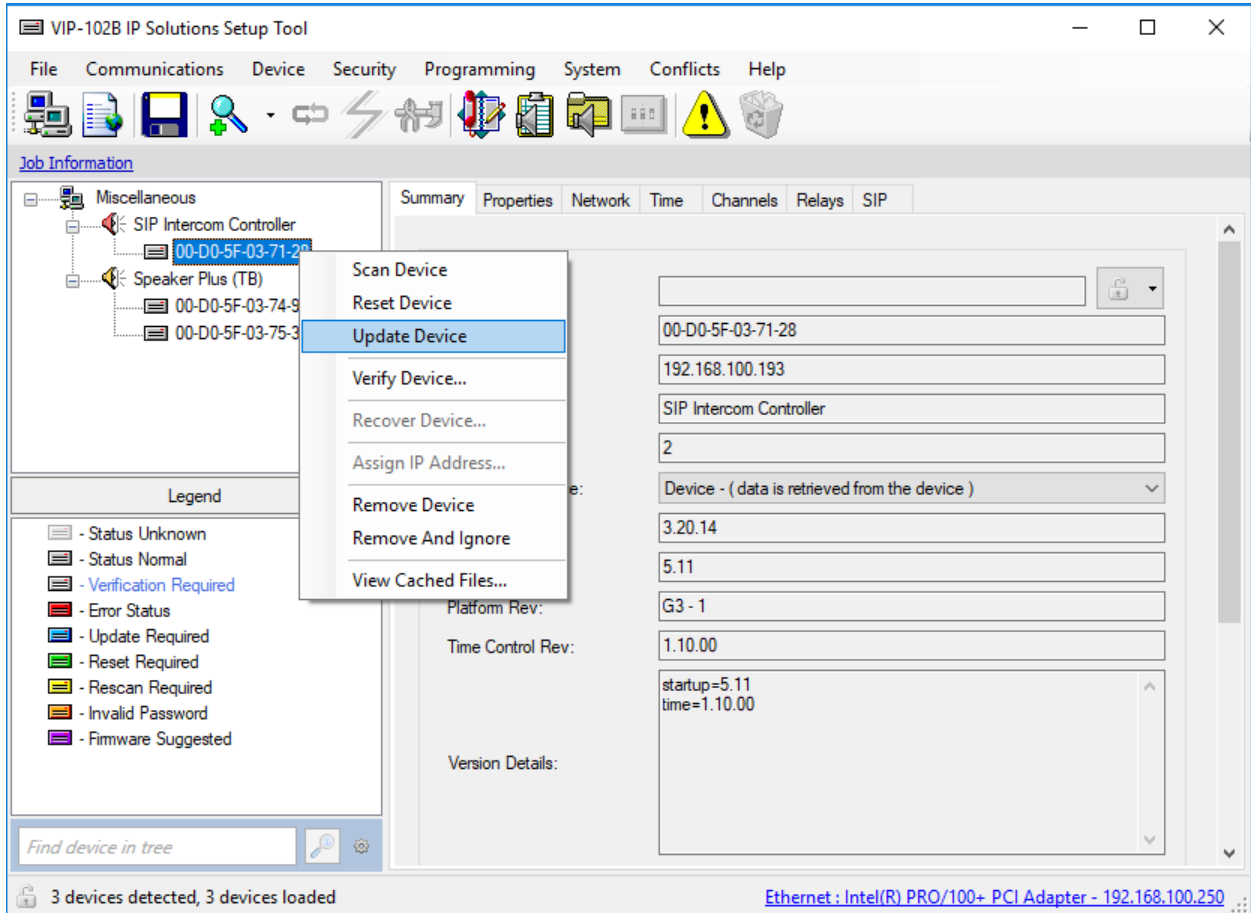
5.4. Verify Codec Settings

Navigate to the **Channels** tab shown below. The Codec Type should be set G.711, currently the only option.



5.5. Update SIP Intercom Controller with the New Configuration

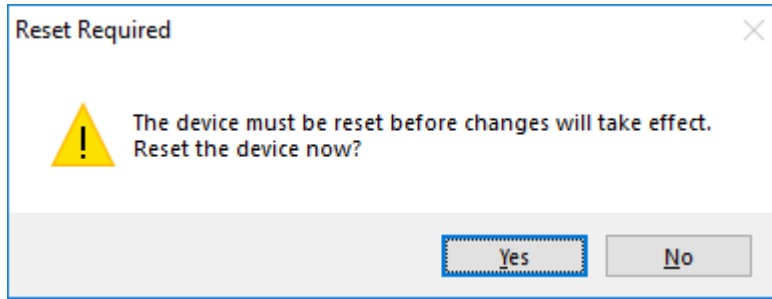
From the **VIP-102B IP Solutions Setup Tool**, right-mouse click on the MAC/hardware address of the SIP Intercom Controller and select **Update Device** from the pop-up menu as shown below.



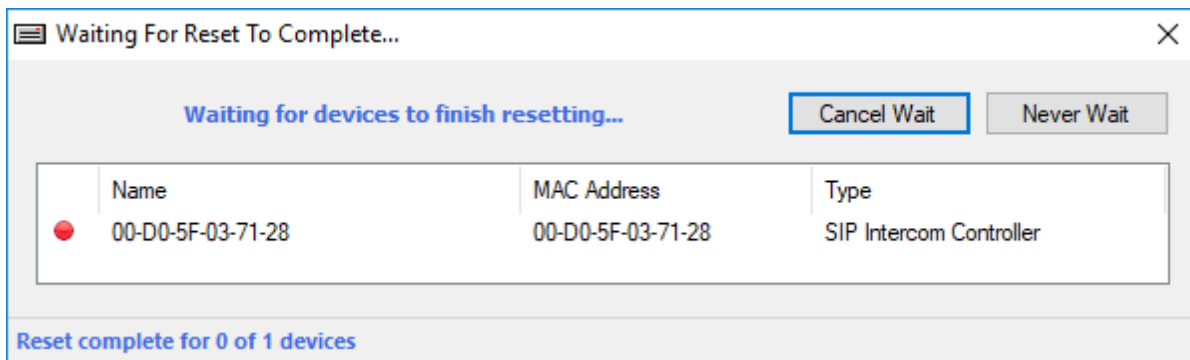
The following window is displayed indicating that the device is being updated.



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 complete, the window will disappear.



6. Verification Steps

This section provides the tests that may be performed to verify proper configuration of Valcom VE8090R SIP Intercom Controller with Avaya IP Office Server Edition.

1. Verify that VE8090R has successfully registered with IP Office Server Edition. Launch **IP Office System Status** and navigate to **Extensions** → **<SIP Extension>**, where **<SIP Extension>** is the VE8090R extension. Verify that the **Current State** is *Idle* as shown below.

The screenshot shows the Avaya IP Office System Status web interface. The title bar indicates the system is running on a Linux PC (10.64.102.90) with build 23. The main content area displays the 'Extension Status' for extension 41510. The status is 'Idle'. Below the status information is a table with columns: Call Ref, Current State, Time in State, Calling Number or Called Number, Direction, and Other Party on Call. The table shows one entry with 'Idle' state and '00:01:40' time in state. At the bottom of the interface, there are buttons for 'Trace', 'Trace All', 'Pause', 'Ping', 'Call Details', 'Print...', and 'Save As...'. The system time is 12:57:43 PM and the system is 'Online'.

Call Ref	Current State	Time in State	Calling Number or Called Number	Direction	Other Party on Call
	Idle	00:01:40			

1. Place a call to VE8090R, and at the dial tone, enter a speaker dial code to place 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 speaker.
2. Place a call to VE8090R, and at the dial tone, enter a group code to place a group call from an Avaya IP Deskphone to a group of Valcom speakers. Verify one-way audio. Terminate the call from the Avaya IP Deskphone.

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

7. Conclusion

These Application Notes described the configuration steps required to integrate Valcom VE8090R SIP Intercom Controller with Avaya IP Office Server Edition. Intercom and group calls were established with Valcom VE8090R SIP Intercom Controller, Valcom VIP-430A IP Wall Speakers, Avaya H.323 / SIP Deskphones, and the PSTN. All feature and serviceability test cases were completed successfully.

8. References

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

- [1] *Administering Avaya IP Office Platform with Manager*, Release 11.0 FP4, February 2019, available at <http://support.avaya.com>.
- [2] *Valcom VIP-102B IP Solutions Setup Tool Version 7.5.0.0 Reference Manual*, Revision 7 – 10/4/18, available at <http://www.valcom.com/vipsetuptool>.

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Declaration of Conformance

March 4, 2019

Jeff Gartner
Senior Manager
DevConnect Program
Avaya

Dear Jeff Gartner:

We, Valcom Inc, declare under sole responsibility that product series named SIP Intercom Controller all 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 the series are detailed in the table below.

Sincerely,

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

Model	Software Rev.	Description
VE8090	3.20.14	SIP Intercom Controller, wall mount, sold direct
VE8090R	3.20.14	SIP Intercom Controller, rack mount, sold direct
VIP-890	3.20.14	SIP Intercom Controller, wall mount, sold through distributors
VIP-890R	3.20.14	SIP Intercom Controller, rack mount, sold through distributors