

Avaya Solution & Interoperability Test Lab

### Application Notes for Valcom VE8090R SIP Intercom Controller 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 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 via a SIP trunk. The Valcom VE8090R SIP Intercom Controller supports two-way audio intercom calls or 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 via a SIP trunk. The Valcom VE8090R SIP Intercom Controller supports two-way audio intercom calls or one-way audio group paging calls.

When the Valcom VE8090R SIP Intercom Controller is configured in SIP trunk mode, the digits in the SIP Invite are interpreted by VE8090R to be Valcom speaker Dial Code or Group Code. For this compliance test, 5-digit numbers in the format of 414xx were dialed, where the last 3 digits (i.e., 4xx) mapped to the Dial Code or Group Code. Refer to the table below for dialing examples.

<b>Dialed Digits</b>	Digits Received by VE8090R	Call Type
41403	403	Intercom Call to Speaker 1
41404	404	Intercom Call to Speaker 2
41410	410	Group Call to Speaker 1
		(Speaker 1 is the only group member)
41420	420	Group Call to Speaker 2
		(Speaker 2 is the only group member)
41499	499	Group Call to Speaker 1 & 2

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 trunk mode, multiple incoming and outgoing calls can be placed to/from the VE8090R.

**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:

- Establishing a SIP trunk between VE8090R and IP Office Server Edition and verifying the exchange of SIP Options messages.
- 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.
- Multiple simultaneous calls with VE8090R.
- 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.

JAO; Reviewed:	Solution & Interoperability Test Lab Application Notes	3 of 23
SPOC 4/9/2019	©2019 Avaya Inc. All Rights Reserved.	VE8090R-TRK-IPO

#### 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: <u>https://www.valcom.com/Support/techsupport.html</u>
- Email: <u>support@valcom.com</u>

### 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 via a SIP trunk. Valcom VIP-430A IP Wall Speakers served as the Valcom audio endpoint, and Valcom VIP-102B IP Solutions Setup Tool to configure Valcom products.





JAO; Reviewed:
SPOC 4/9/2019

Solution & Interoperability Test Lab Application Notes ©2019 Avaya Inc. All Rights Reserved.

## 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
- Enable SIP trunks
- Administer SIP line
- Administer incoming call route
- Administer short code

#### 4.1. Verify IP Office License

From a PC with Avaya IP Office Manager installed, select Start  $\rightarrow$  Programs  $\rightarrow$  IP Office  $\rightarrow$  Manager to launch the Manager application. Select the proper 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**. Verify that the **SIP Trunk Channels** license is "Valid", and that the **Instances** value is sufficient for the desired maximum number of calls.

Eile Edit View Iools Help	4	· ·							
Configuration						C <sup>1</sup>	+ <sup>(0)</sup>	$\times  $ $\checkmark$	<   >
	^	License Remote Server							
PX Short Code(49)     Directory(0)     Time Profile(0)     Get Rights(9)     Get Rights(9)     Get Rights(9)     Get Control Unit (9)     Get		Feature Receptionist Additional Voicemail Pro Ports VMPro Recordings Administrators Office Worker VMPro TTS Professional IPSec Tunnelling Power User Avaya IP endpoints SIP Trunk Channels IP500 Universal PRI (Additional cha CTI Link Pro Wave User 3rd Party IP Endpoints Server Edition UMS Web Services Avaya Mac Softphone Avaya Softphone Licence SM Trunk Channels	Instances 10 252 1 1000 40 1 1000 256 1000 1 1000 150 1000 150 1000 1000 1000 128	Status Valid Valid Valid Valid Valid Valid Valid Obsolete Valid Obsolete Valid Valid Valid Valid Valid Valid Valid Valid	Expiration Date Never	Source PLDS Nodal PLDS Nodal		Add Remove	
Location (0)		Web Collaboration	64	Valid	Never	PI DS Nodal			~
devcon-ipo500v2	~					<u>O</u> K	<u>C</u> ano	cel <u>H</u>	elp

#### 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. Enable SIP Trunks

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

🐮 Avaya IP Office Manager for Server Edition devo	ipose [11.0.0.2.0 build 23]	- 🗆 ×
File Edit View Iools Help		
devcon-ipose • System	• devcon-ipose •	
Configuration	E devcon-ipose	ini - 1   ×   ✓   <   >
Gerator (3)     Solution     Solution     Gerator (3)     Solution     Gerator (3)     Solution     Gerator (3)     Gerator (3)     Gerator (4)     Gera	System       LAN1       LAN2       DNS       Voicemail       Telephony       Directory Services       System Events       SMDP         LAN Settings       VoiP       Network Topology         H.323       Gatekeeper Enable       Auto-create Extension       Auto-create User       H.323       Remote Call Signaling Port       1720       Image: Call Signaling Port       Image: Call Signaling Port<	R VolP VolP Security • • Remote Extension Enable DP Port 5060 • • LS Port 5061 • •
User Rights (9)	<pre></pre>	×
← Location (0)	Ōĸ	<u>C</u> ancel <u>H</u> elp
Ready		III .::

JAO; Reviewed: SPOC 4/9/2019

Solution & Interoperability Test Lab Application Notes ©2019 Avaya Inc. All Rights Reserved. 8 of 23 VE8090R-TRK-IPO

#### 4.4. Administer SIP Line

From the configuration tree in the left pane, right-click on Line and select New  $\rightarrow$  SIP Line from the pop-up list (not shown) to add a new SIP line. Select the **Transport** tab in the right pane. For **ITSP Proxy Address**, enter the VE8090R IP address. Retain the default values for the remaining fields.

🐮 Avaya IP Office Manager for Server Edit	ion devcon-ipose [11.0.0.2.0 build 23]	-	- 🗆 X
<u>F</u> ile <u>E</u> dit <u>V</u> iew <u>T</u> ools <u>H</u> elp			
🗄 2. 🗁 - 🖃 🖪 💽 🖬 🛕 🗸 🖉			
devcon-ipose 👻 Line	• 2 •		
Configuration	E SIP Line - Line 2	📥 🗕 🔤	X   ✓   <   >
	SIP Line       Transport       Call Details       VolP       SIP Credentials       SIP Advanced       Engineering         ITSP Proxy Address       192.168.100.193		
			Hala
Ready Account Code (0)	2		

🕅 Avava IP Office Manager for Server Edi	tion devcon-ipose [11.0.0.2.0 build 23]		
Eile Edit View Tools Hele			- ~
devcon-ipose • Line	• 2 •		
Configuration	🗳 SIP Line - Line 2*	🛉 - 🔛   🗙	✓   <   >
BOOTP (5)	SIP Line Transport Call Details VoIP SIP Credentials SIP Advanced Engineering		
Solution	SIP URIs		
⊞∎ User(38) ⊕∰ Group(1)	URI Groups Credential Local URI Contact P Asserted ID P Preferred ID Diversion Header Rem	iote Party II	Add
Short Code(49)     Directory(0)			Remove
Time Profile(0)			Edit
B- S User Rights(9)			
evcon-ipose	٢	>	
□ - 行 T Line (3)			
9	Line ID Incoming ID Outgoing ID Groups Credential Local URI Contact PAsserted ID PPrefer	red ID Div	Add
🖅 🤝 Control Unit (9)			Remove
Group (0)			Edit
🕀 🥬 Short Code (54)			
Service (0)			
Directory (0)		,	
······································			
Account Code (0)	<u>O</u> K	<u>C</u> ancel	<u>H</u> elp
Ready			IT .::

Select the **Call Details** tab to display the **SIP URIs** section as shown below.

Click **Add** to display the **New URI** section shown below. Enter the SIP line number for **Incoming Group** and **Outgoing Group**. Set **Max Sessions** to the desired maximum number of simultaneous calls allowed. Retain the default values in the remaining fields.

📶 SIP Line - 2	Call D	etails   SIP URI					>	×	
New URI									
Incoming Group	Incoming Group 2 Max Sessions 10								
Outgoing Group 2									
Credentials	0: <1	lone> ~							
		Display		Content	Field meaning				
					Outgoing Calls	Forwarding/Twinning	Incoming Calls		
Local URI		Auto	~	Auto ~	Caller	<ul> <li>Original Caller</li> </ul>	Called $\checkmark$		
Contact		Auto	~	Auto ~	Caller	<ul> <li>Original Caller</li> </ul>	Called $\checkmark$		
P Asserted ID		None	$\sim$	None 🗸	None	None	None 🗸		
P Preferred ID		None	$\sim$	None 🗸	None	None	None ~		
Diversion Header		None	$\sim$	None	None	None	None ~		
Remote Party ID		None	$\sim$	None 🗸	None	None 🗸	None ~		
						ОК	Cancel Help	1	

Select the **VoIP** tab to view the codecs allowed. VE8090R supports G.711 codec only. Check **Re-invite Supported**, **Allow Direct Media Path**, and retain the default values in the remaining fields.

🖞 Avaya IP Office Manager for Server Edition devcon-ipose [11.0.0.2.0 build 23] – 🗆 X								
<u>F</u> ile <u>E</u> dit <u>V</u> iew <u>T</u> ools <u>H</u> elp	<u>Eile Edit View Iools H</u> elp							
🕴 🖄 - 🔛 🖪 🔝 🔝 人 🛹 🐸 🕑								
devcon-ipose 🝷 Line	✓ 2	•						
Configuration	<b>17</b>	SIP Line - Line 2*	📸 - 🔛   🗙   🖌   <   >					
BOOTP (5)     Operator (3)     Solution     User(38)     User(38)     Off(0)     Directory(0)     Oirectory(0)     Oire	SIP Line Transport Call Codec Selection Fax Transport Support DTMF Support Media Security	Details       VoIP       SIP Credentials       SIP Advanced       Engineering         System Default          Unused       Selected         G.711 ULAW 64K       G.711 ALAW 64K         G.722 64K       G.722 64K         None          RFC2833/RFC4733	<ul> <li>□ Local Hold Music</li> <li>☑ Re-invite Supported</li> <li>□ Codec Lockdown</li> <li>☑ Allow Direct Media Path</li> <li>□ Force direct media with phones</li> <li>□ PRACK/100rel Supported</li> </ul>					
Account Code (0)			<u>O</u> K <u>Cancel H</u> elp					
Ready	1		F¶ .::					

### 4.5. 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 (not shown) to add a new route. For **Line Group Id**, select the incoming group number from **Section 4.4**, which corresponds to the SIP line, in this case "2".



Select the **Destinations** tab. For **Destination**, enter "." to match any dialed number from VE8090R.

🖞 Avaya IP Office Manager for Server Edition devcon-ipose [11.0.0.2.0 build 23] - 🛛 🗙						
<u>F</u> ile <u>E</u> dit <u>V</u> iew <u>T</u> ools <u>H</u> elp	9					
devcon-ipose 👻 Incoming Call Rou	te • 2 •					
Configuration	2		📸 - 🔤   🗙   🗸   <   >			
BOOTP (5)	Standard Voice Recording Destination	S				
Solution	TimeProfile D	estination	Fallback Extension			
∎ User(38)	Default Value .	\ \	/			
⊞∰ Group(1)						
Directory(0)						
······································						
Account Code(0)						
Location(0)						
devcon-ipose						
⊕ ~ < Control Unit (9)						
⊞						
⊞¶ User (8)						
Broup (0)						
Service (0)						
🖃 🕩 Incoming Call Route (2)						
Directory (0)						
- ① Time Profile (0)						
i IP Route (1)						
License (22)		<u>O</u> K	Cancel <u>H</u> elp			
Ready						

### 4.6. Administer Short Code

From the configuration tree in the left pane, right-click on **Short Code** and select **New** from the pop-up list (not shown) to add a new short code for calls to VE8090R. In the compliance test, 3-digit numbers beginning with '4' were sent to VE8090R in the SIP Invite that matched a speaker dial code or group code.

For **Code**, enter "414xx". For **Feature**, select *Dial* from the drop-down list. For **Telephone Number**, enter the value shown below where "4N" is the 3-digit dial code and "192.168.100.193" is the VE8090R IP address. For **Line Group Id**, enter the outgoing group number from **Section 4.4**, which corresponds to the VE8090R SIP trunk.

📶 Avaya IP Office Manager for Server Edition	n devcon-ipose [11.0.0.2.0 bui	ld 23]		_	· 🗆	×
Eile Edit View Tools Help	9					
devcon-ipose		<u> </u>				
Configuration		414xx: Dial		📥 🗕 🔤	× 🗸	< >
→ -9×       Short Code (54)       ^        9×       *00      9×       *01        9×       *02      9×       *03        9×       *03      9×       *04         -9×       *05      9×       *06         -9×       *06      9×       *08         -9×       *08      9×       *09         -9×       *10*N#      9×       *12*N#         -9×       *12*N#      9×       *12*N#         -9×       *12*N#      9×       *16         -9×       *16      9×       *18         -9×       *19      9×       *22*N#         -9×       *21*N#      9×       *30	Short Code Code Feature Telephone Number Line Group ID Locale Force Account Code Force Authorization Code	414xx Dial ~ 4N"@192.168.100.193" 2 ~				
•••••••••••••••••••••••••••••••••••••			<u>О</u> К	<u>C</u> ance		<u>H</u> elp
Ready						- Fi .::

# 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 the 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.

VIP-102B IP Solutions Setup Tool		– 🗆 X
File Communications Device Securi	ity <u>P</u> rogramming <u>S</u> ystem Co <u>n</u> f	ilicts <u>H</u> elp
🎭 🔒 🔒 🐥 · 🗢 🗲	ka 🕸 🚺 🗱 💷	
Job Information		
Miscellaneous  Miscellaneous  SIP Intercom Controller  00-D0-5F-03-71-28	Summary Properties Network Time	Channels Relays SIP
	Host Name: Domain Name:	
	Use DHCP:	
	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:	
	SIP SDP NAT:	
	Continuous Beacon:	
Lagand	Use Syslog Daemon:	
- Status Unknown	Syslog Daemon:	
<ul> <li>Status Vormal</li> </ul>		
<ul> <li>Verification Required</li> <li>From Status</li> </ul>		
<ul> <li>Update Required</li> </ul>		
Reset Required		
<ul> <li>Invalid Password</li> </ul>		
🖃 - Firmware Suggested		
Find device in tree	Local Network	Copy & Apply & Cancel
3 devices detected, 3 devices loaded		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 *Trunk*, 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. The **Register** checkbox should be unchecked and all other fields should be left at their default values. Click **Apply**.

IVIP-102B IP Solutions Setup Tool					-		×
<u>File Communications Device Security P</u>	rogramming <u>S</u> ystem Co	nflicts <u>H</u> elp					
🔚 🗈 🗖 🍳 - 🗂 4	1 🎝 🖻 🛜 📖						
Job Information							
Image: Second controller         Summ           Image: Second controller         Image:	nary Properties Network Tin	ne Channels Re	lays SIP	SIF	P Mode: Tru	ık	~
	Authentication Name:						
	Secret:						
	Realm:						
	SIP Servers:	Dimensi	Server		Port		
	Register:	Primary Backup 1	10.04.102.30		5060		
		Backup 1 Backup 2			5060		
	DINS SRV:	Backup 3			5060		
					-		
	Busy Message:					~	
	Call Fwd Busy ( 302 ):				5000		
	Outbound Proxy:			Outbound Port:	5060		
	Keep Alive Timer (secs):	3600		Options Timer (secs)	): 0		
	SIP Port:	20000		Idle Timeout (secs):	、 (0		
	RTP Port:	20000		Max Call Timer (secs	s): U	~	
Legend							
Status Normal     Verification Required	Auto Destination:	41001					
- Error Status							
Update Required	Pre-Announce Tone:	$\checkmark$					
- Reset Required     Rescan Required     - Invalid Password     - Firmware Suggested							
Find device in tree	<u>D</u> efaults			Сору	✓ <u>A</u> pply	X <u>C</u> an	cel
3 devices detected, 3 devices loaded			Etherne	t : Intel(R) PRO/100+ I	PCI Adapter -	192.168.1	00.250

Solution & Interoperability Test Lab Application Notes ©2019 Avaya Inc. All Rights Reserved.

### 5.4. Verify Codec Settings

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

WP-1028 IP Solutions Setup Tool     File Communications Levice Security Programming System Conflicts Help     Sub Information					
bit Communications Device Security Programming System Cogficts Help   Bit Bit Bit Bit Bit Bit Bit Bit Bit   Bit Bit Bit Bit Bit Bit Bit Bit   Bit Bit Bit Bit Bit Bit Bit   Bit Bit Bit Bit Bit Bit Bit   Bit Bit Bit Bit Bit Bit Bit   Bit Bit Bit Bit Bit Bit Bit   Bit Bit Bit Bit Bit Bit Bit   Bit Bit Bit Bit Bit Bit Bit   Bit Bit Bit Bit Bit Bit Bit   Bit Bit Bit Bit Bit Bit Bit   Bit Bit Bit Bit Bit Bit Bit   Bit Bit Bit Bit Bit Bit Bit   Bit Bit Bit Bit Bit Bit Bit   Bit Bit Bit Bit Bit Bit Bit   Bit Bit Bit Bit Bit Bit Bit   Bit Bit Bit Bit Bit Bit Bit   Bit Bit Bit Bit Bit Bit Bit   Bit Bit Bit Bit Bit Bit   Bit Bit	VIP-102B IP Solutions Setup Tool			- 🗆	×
Image: Image	<u>File</u> <u>Communications</u> <u>Device</u> Sec <u>u</u> rity	<u>P</u> rogramming <u>S</u> ystem C	o <u>n</u> flicts <u>H</u> elp		
but formation     Summary Properties Network Time Channels Relays SIP     I 2     Bal Code:     401     Description:     ClD Number:     Code:     Ado Destination:     Code:     Code:     Code:     Ado Destination:     Code:     Code:     Code:     Code:     Ado Destination:     Code:	💂 🗟 🔚 💲 · 🗢 🖉 ·	19 🐌 🚺 👫 🔤			
Macellaneous   Macellaneous   Summary Properties Network Time Diameds Relays SIP     00005F037128   00005F03753A     1   2     Dal Code:   001005F03753A     Dal Code:   011   Description:   CID Number:   401   Code:   0   Code:   0   Cal Find Bary:   Can Find Bary:   Cal Find Bary:   Can Find Bary:  <	Job Information				
Audio Input Volume:   0   0   Audio Output Volume:   0  0	Image: Second controller           Image: Second control controller           Image: Second control control controller           Image: Second control	Summary Properties Network T 1 2 Dial Code: Description: CID Number: CID Name: Auto Destination: Codec Type: Call Fwd Busy: Call Fwd No Answer:	Imme         Channels         Relays         SIP           401	Rings	
Legend       ▼            • Status Unknown         • Status Nomal         • Verfication Required         • Enror Status         • Update Required         • Reset Required         • Reset Required         • Rescan Required         • Invalid Password         • Invalid Password         • Firmware Suggested         • Firmd device in tree         Ø         • Output         • Ou		Audio Input Volume: Audio Output Volume:		) ) )	
Image: A rescan required       Image: A rescan req       Image: A rescan req<	Legend ▼ Status Unknown □ - Status Nomal □ - Verification Required □ - Error Status □ - Update Required □ - Reset Required □ - Reset Required	Pre-Announce Tone:			
2 devices detected 2 devices loaded Ethernet, lots (0) 000 (000 000 4 L + 100 100 000 000 000 000 000 000 000 0				∑ Cance	

### 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.

VIP-102B IP Solutions Setup Tool			- 0	×
File Communications Device	Security Programming	System Conflicts Help		
🛃 🗟 🔚 왔 - 🖙	/~ 👫 🐌 🗿	🗊 💷 <u>1</u> 🎯		
Job Information				
□ Miscellaneous □ ≪ SIP Intercom Controller	Summary Properties	Network Time Channels Relays SIP		^
□ = 00-D0-5F-03-71-2 <sup>a1</sup>	Scan Device			
■	Reset Device		* *	
🗐 00-D0-5F-03-75-3	Update Device	00-D0-5F-03-71-28		
	Verify Device	192.168.100.193		
		SIP Intercom Controller		
	Kecover Device	2		
	Assign IP Address			
Legend	Remove Device	e: Device - ( data is retrieved from the device )	~	
🖃 - Status Unknown	Remove And Ignore	3.20.14		
<ul> <li>Status Normal</li> <li>Verification Required</li> </ul>	View Cached Files	5.11		
- Error Status	Platform Rev:	G3 - 1		
Update Required	Time Control Rev	r: 1.10.00		
🖃 - Reset Required		startup=5.11		
Invalid Password		time=1.10.00		
🔳 - Firmware Suggested	V · D · 1			
	Version Details:			
Find device in tree				
🔒 3 devices detected, 3 devices loaded Ethernet : Intel(R) PRO/100+ PCI Adapter - 192.168.100.250				

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

Updating Device		
	Updating device	
	<u>C</u> ancel	

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.

Waiting for devices	to finish resetting	Cancel Wait Never Wait
Name	MAC Address	Туре
0-D0-5F-03-71-28	00-D0-5F-03-71-28	SIP Intercom Controller
	Waiting for devices t Name 10-D0-5F-03-71-28	Waiting for devices to finish resetting         Name       MAC Address         00-D0-5F-03-71-28       00-D0-5F-03-71-28

### 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.

1. From Avaya IP Office for Server Edition, select File → Advanced → System Status to launch the System Status application, and log in using the appropriate credentials.

The **IP Office System Status** screen is displayed. Expand **Trunks** in the left pane and select the SIP line from **Section 4.4**, in this case "2".

Verify that the **SIP Trunk Summary** screen shows that the **Line Service State** is *In Service*.

🗾 Avaya IP Office System S	Status - devcon-ipose (10.64.102.90) - IP Office Linux PC 11.0.0.2.0 build 23	-	×
AVAYA	<b>IP Office System Status</b>		
Help Snapshot LogOff Exit	t About		
<ul> <li>System</li> <li>Å Alarms (7)</li> <li>Extensions (4)</li> <li>Trunks (3) <ol> <li>Line: 1</li> <li>Line: 2</li> <li>Line: 9</li> <li>Active Calls</li> </ol> </li> <li>Resources</li> <li>Voicemail</li> <li>IP Networking <ol> <li>Locations</li> </ol> </li> </ul>	Status       Utilization Summary       Alarms         SIP Trunk Summary         Line Service State:       In Service         Peer Domain Name:       sip://192.168.100.193         Resolved Address:       192.168.100.193         Line Number:       2         Number of Administered Channels:       10         Number of Channels in Use:       0         Administered Compression:       G711 Mu, G711 A, G729 A, G722         Enable Faststart:       Off         Media Stream:       RTP         Layer 4 Protocol:       UDP         SIP Trunk Channel Licenses:       255         SIP Trunk Channel Licenses:       0         SIP Device Features:       0         Chan U Call Current Time in Remote Co Conn Caller Other Party Direc Round Receive Recei         1       Idle       00:1         2       Idle       00:1       Idle       Idle         3       Idle       00:1       Idle       Idle       Idle         4       Idle       00:1       Idle       Idle       Idle       Idle       Idle         5       Idle       00:1       Idle       Idle       Idle       Idle       Idle	Trans Tr	ans
	1:22:45 PM	On	ine 🔒

- 2. Dial 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.
- 3. Dial 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.

JAO; Reviewed:	Solution & Interoperability Test Lab Application Notes	21 of 23
SPOC 4/9/2019	©2019 Avaya Inc. All Rights Reserved.	VE8090R-TRK-IPO

4. 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 <u>http://support.avaya.com</u>.
- [2] *Valcom VIP-102B IP Solutions Setup Tool Version 7.5.0.0 Reference Manual*, Revision 7 10/4/18, available at <u>http://www.valcom.com/vipsetuptool</u>.

#### ©2019 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and <sup>TM</sup> are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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 <u>devconnect@avaya.com</u>.

#### ATTACHMENT 1



#### **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 Ellivon

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