

Avaya Solution & Interoperability Test Lab

Application Notes for Valcom One-Way and Talkback IP Speakers with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate Valcom One-Way and Talkback IP Speakers running SW Rev 3.24.5 and SIP Rev sw1.60.38 with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. For this compliance test, Valcom VIP-130AL-GY IP Secure One-Way Paging Horn and VIP-160A IP Talkback 8" Ceiling Speaker were used. These Valcom IP speakers register with Avaya Aura® Session Manager as SIP endpoints.

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 Valcom One-Way and Talkback IP Speakers running SW Rev 3.24.5 and SIP Rev sw1.60.38 with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. For this compliance test, Valcom VIP-130AL-GY IP Secure One-Way Paging Horn and VIP-160A IP Talkback 8" Ceiling Speaker were used. These Valcom IP speakers register with Avaya Aura® Session Manager as SIP endpoints.

The Valcom VIP-130AL-GY IP Secure One-Way Paging Horn is a self-contained paging system which enables paging over an IP network. When a call is placed to the Valcom One-Way IP Speaker, the device automatically answers the call and provides one-way communication to the device.

The Valcom VIP-160A IP Talkback 8" Ceiling Speaker supports both incoming and outgoing pages and hands-free two-way communication. When the call button is pressed on a Valcom Talkback IP Speaker, the device initiates a call to a preconfigured destination that resides on Avaya Aura® Communication Manager.

Valcom offers IP Ceiling Speakers, IP Wall Speakers, and IP Horns 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 One-Way and Talkback IP Speakers, Avaya SIP / H.323 IP Deskphones, and the PSTN. Two-way audio intercom calls with the Talkback IP Speaker and one-way audio paging calls with the One-Way IP Speaker were exercised. The serviceability testing focused on verifying that the Valcom IP speakers 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 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 interface between Avaya systems and Valcom One-Way and Talkback IP Speakers 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 IP speakers with Session Manager.
- Establishing one-way audio paging calls from Avaya H.323 / SIP Deskphones and PSTN to the VIP-130AL Paging Horn.
- Establishing two-way audio intercom calls between VIP-160A Ceiling Speaker, Avaya H.323 / SIP Deskphones, and PSTN in both directions.
- Originating and terminating calls through Avaya SIP telephony network.
- Originating calls from the VIP-160A Ceiling Speaker to a predefined number using the call button.
- Terminating active calls by pressing the call button on the VIP-160A Ceiling Speaker.
- Support of G.711 mu-law and G.722 codecs.
- Support of TCP and UDP transport protocols.
- Support of direct IP-to-IP media (also known as "Shuffling" which allows IP endpoints to send audio RTP packets directly to each other without using media resources on the Avaya Media Gateway or Avaya Aura® Media Server).
- Proper recovery after a restart of IP speakers.

2.2. Test Results

All test cases passed with the following observations:

- Valcom One-Way and Talkback IP Speakers always advertise G.722 and G.711 (in that order) in the SIP SDP. The codec selection on the IP speaker only enforces the codec when the IP speaker initiates a call to a Valcom gateway (not covered by this compliance test).
- When an outgoing call from an IP speaker fails for whatever reason, such as invalid number, phone busy, or trunk calls blocked, or if the IP speaker doesn't register successfully via SIP, the IP speaker plays, "All circuits are busy at the present time."
- Standalone Valcom IP speakers do not support group paging calls using multicast audio without an optional Valcom controller.

2.3. Support

For technical support and information on Valcom One-Way and Talkback IP Speakers, 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>

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3. Reference Configuration

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

- Avaya Aura® Communication Manager with an Avaya G450 Media Gateway.
- Media resources in the Avaya G450 Media Gateway and Avaya Aura® Media Server.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP deskphones and Valcom One-Way and Talkback IP Speakers.
- Avaya Aura® Session Manager connected to Avaya Session Border Controller for Enterprise (SBCE) via a SIP trunk for access to the simulated PSTN.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya Session Border Controller for Enterprise
- Avaya 96x1 Series H.323 and SIP Deskphones.
- Avaya J100 Series SIP Deskphones.
- Valcom IP speakers, including the Valcom VIP-130AL-GY IP Secure One-Way Paging Horn and the Valcom VIP-160A IP Talkback 8" Ceiling Speaker configured with the ValcomVIP-102B IP Solutions Setup Tool.

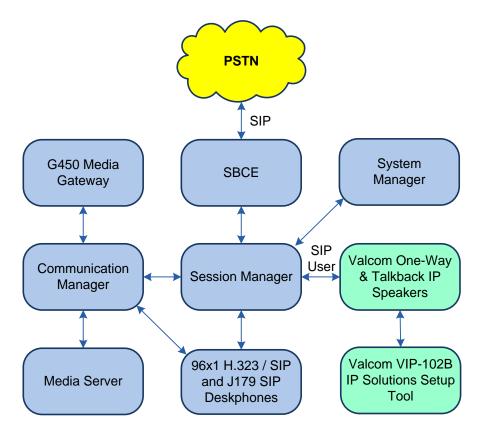


Figure 1: Avaya SIP Network with Valcom One-Way and Talkback IP Speakers

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	8.1.2.0.0-FP2
Avaya G450 Media Gateway	FW 41.24.0
Avaya Aura® Media Server	v.8.0.2.93
Avaya Aura® System Manager	8.1.2.0 Build No. – 8.1.0.0.733078 Software Update Revision No: 8.1.2.0.0611167 Feature Pack 2
Avaya Aura® Session Manager	8.1.2.0.812039
Avaya 96x1 Series IP Deskphones	6.8304 (H.323) 7.1.9.0.8 (SIP)
Avaya J100 Series IP Deskphones	4.0.5.0.10 (SIP)
Valcom VIP-130AL-GY IP Secure One-Way Paging Horn	Software Rev: 3.24.5 SIP Rev: sw1.60.38
Valcom VIP-160A IP Talkback 8" Ceiling Speaker	Software Rev: 3.24.5 SIP Rev: sw1.60.38
Valcom VIP-102B IP Solutions Setup Tool	8.1.0.0

5. Configure Avaya Aura® Communication Manager

This section provides the procedure for configuring Communication Manager. The procedure includes the following areas:

- Verify Communication Manager license
- Administer IP Node Names
- Administer IP Network Region and IP Codec Set
- Administer SIP Trunk Group to Session Manager
- Administer AAR Call Routing

Use the System Access Terminal (SAT) to configure Communication Manager and log in with appropriate credentials.

Note: The SIP station configuration for the Valcom IP speakers is performed through System Manager in **Section 6.3**.

5.1. Verify Communication Manager License

Using the SAT, verify that the Off-PBX Telephones (OPS) option is enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

On **Page 1**, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

```
Page 1 of 12
display system-parameters customer-options
                             OPTIONAL FEATURES
    G3 Version: V18
                                              Software Package: Enterprise
                                               System ID (SID): 1
      Location: 2
      Platform: 28
                                               Module ID (MID): 1
                                                          USED
                              Platform Maximum Ports: 48000 91
                              Maximum Stations: 36000
                                                              30
                                                             0
                            Maximum XMOBILE Stations: 36000
                  Maximum Off-PBX Telephones - EC500: 41000
                                                              0
                  Maximum Off-PBX Telephones - OPS: 41000
                                                              16
                  Maximum Off-PBX Telephones - PBFMC: 41000
                                                              0
                                                              0
                   Maximum Off-PBX Telephones - PVFMC: 41000
                  Maximum Off-PBX Telephones - SCCAN: 0
                                                              0
                       Maximum Survivable Processors: 313
                                                               0
       (NOTE: You must logoff & login to effect the permission changes.)
```

5.2. Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name for Communication Manager (*procr*) and Session Manager (*devcon-sm*). The host names will be used in other configuration screens of Communication Manager.

```
change node-names ip
                                                              Page
                                                                    1 of
                                                                           2
                                IP NODE NAMES
   Name
                   IP Address
default
                 0.0.0.0
devcon-aes
                  10.64.102.119
devcon-ams
                   10.64.102.118
                   10.64.102.117
devcon-sm
procr
                   10.64.102.115
procr6
                   • •
( 6 of 6 administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

5.3. Administer IP Network Region and IP Codec Set

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G450 Media Gateway or Media Server. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager.

```
change ip-network-region 1
                                                               Page 1 of 20
                              TP NETWORK REGION
  Region: 1
Location: 1
               Authoritative Domain: avaya.com
   Name:
                               Stub Network Region: n
MEDIA PARAMETERS
                               Intra-region IP-IP Direct Audio: yes
     Codec Set: 1
                              Inter-region IP-IP Direct Audio: yes
  UDP Port Min: 2048
                                          IP Audio Hairpinning? n
  UDP Port Max: 50999
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 46
       Audio PHB Value: 46
       Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
       Audio 802.1p Priority: 6
       Video 802.1p Priority: 5
                                     AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS
                                                        RSVP Enabled? n
 H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
           Keep-Alive Count: 5
```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to IP speaker. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set '1' was specified in IP Network Region '1' shown above. The default settings of the **IP Codec Set** form are shown below. Valcom IP speakers were tested using G.711 and G.722 codecs.

Page

1 of

2

```
change ip-codec-set 1

IP CODEC SET

Codec Set: 1

Audio Silence Frames Packet

Codec Suppression Per Pkt Size(ms)

1: G.711MU n 2 20

2:

3:
```

5.4. Administer SIP Trunk to Session Manager

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tls*.
- Set the **Enforce SIPS URI for SRTP** field to *n*.
- Specify Communication Manager (*procr*) and the Session Manager as the two ends of the signaling group in the Near-end Node Name field and the Far-end Node Name field, respectively. These field values are taken from the IP Node Names form.
- Ensure that the TLS port value of 5061 is configured in the Near-end Listen Port and the Far-end Listen Port fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

add signaling-group 10 Page 1 of 2 SIGNALING GROUP Group Number: 10 Group Type: sip IMS Enabled? n Transport Method: tls Q-SIP? n IP Video? n Enforce SIPS URI for SRTP? n Peer Detection Enabled? y Peer Server: SM Clustered? n Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n Alert Incoming SIP Crisis Calls? n Near-end Node Name: procr Far-end Node Name: devcon-sm Near-end Listen Port: 5061 Far-end Listen Port: 5061 Far-end Network Region: 1 Far-end Domain: avaya.com Bypass If IP Threshold Exceeded? n RFC 3389 Comfort Noise? n Incoming Dialog Loopbacks: eliminate DTMF over IP: rtp-payload Direct IP-IP Audio Connections? y Session Establishment Timer(min): 3 IP Audio Hairpinning? n Initial IP-IP Direct Media? n Enable Layer 3 Test? y H.323 Station Outgoing Direct Media? n Alternate Route Timer(sec): 6

Configure the **Trunk Group** form as shown below. This trunk group is used for SIP calls to/from IP speakers, Avaya SIP Deskphones, and Avaya Aura® Messaging. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

add trunk-group 10 Page 1 of 22 TRUNK GROUP coup Number: 10 Group Name: To devcon-sm Group Type: sip CDR Reports: y Group Number: 10 COR: 1 TN: 1 TAC: 1010 Direction: two-way Outgoing Display? n Dial Access? n Night Service: Queue Length: 0 Service Type: tie Auth Code? n Member Assignment Method: auto Signaling Group: 10 Number of Members: 10

5.5. Administer AAR Call Routing

SIP calls to Session Manager are routed over a SIP trunk via AAR call routing. Configure the AAR analysis form and enter add an entry that routes digits beginning with "78" to route pattern "10" as shown below.

change aar analysis 78					Page 1 of	2	
	AAR DIGIT ANALYSIS TABLE						
	Location: all				Percent Full: 1		
Dialed	Total	Route	Call	Node	ANI		
String	Min Max	Pattern	Туре	Num	Reqd		
78	5 5	10	lev0		n		

Configure a preference in **Route Pattern** 10 to route calls over SIP trunk group 10 as shown below.

char	nge 1	coute	-pat	terr	n 10]	Page	1 of	3
					Pat	tern 1	Number	c: 10		Patt	tern 1	Name:	то	devco	on-sm		
	SCCA	AN? n		Seci	ire S	SIP? 1	n	Used	for	SIP	stat	ions?	n				
	_							_									
	Grp	FRL	NPA	Pfx	Нор	TOII	No.									DCS/	IXC
	No			Mrk	Lmt	List	Del	Digit	s							QSIG	
							Dgts									Intw	
1:	10	0														n	user
2:																n	user
3:																n	user
4:																n	user
5:																n	user
6:																n	user
	BCC	VAL	UE	TSC	CA-	rsc	ITC	BCIE	Serv	/ice	/Feat	ure PA	ARM	Sub	Numbe	ring	LAR
	0 1	2 M	4 W		Req	uest								Dgts	Forma	t	
1:	уу	уу	y n	n			rest	-							unk-u	nk	none
2:	УУ	УУ	y n	n			rest	5									none

6. Configure Avaya Aura® Session Manager

This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Set Network Transport Protocol
- Administer SIP User

Note: It is assumed that basic configuration of Session Manager has already been performed. This section will focus on the configuration of a SIP user for Valcom One-Way and Talkback IP Speakers.

6.1. Launch System Manager

Access the System Manager Web interface by using the URL "https://ip-address" in an Internet browser window, where "ip-address" is the IP address of the System Manager server. Log in using the appropriate credentials.

Recommended access to System Manager is via FQDN.	•
Go to central login for Single Sign-On	User ID:
If IP address access is your only option, then note that authentication will fail in the following cases:	Password:
 First time login with "admin" account Expired/Reset passwords 	Log On Cancel
Use the "Change Password" hyperlink on this page to change the password manually, and then login.	Change Password
Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.	• Supported Browsers: Internet Explorer 11.x or Firefox 65.0, 66.0 and 67.0.

6.2. Set Network Transport Protocol

From the System Manager Home screen, select **Elements** \rightarrow **Routing** \rightarrow **SIP Entities** and edit the SIP Entity for Session Manager shown below.

AVAYA Aura® System Manager 8.1	Users 🗸 🥜 Elements 🗸 🎄 Services 🛇	 Widgets <> Shortcuts 	Search	admin
Home Routing				
Routing ^	SIP Entity Details		Commit Cancel	Help ? 🔺
Domains	General			
Locations	* Name:	devcon-sm		
	* IP Address:	10.64.102.117		
Conditions	SIP FQDN:			
Adaptations 🗸 🗸	Туре:	Session Manager		
SIP Entities	Notes:			
Entity Links	Location:	Thornton 🗸		
	Outbound Proxy:	~		
Time Ranges	Time Zone:	America/New_York 🗸		
Routing Policies	Minimum TLS Version:	Use Global Setting 🗸		
Dial Patterns 🗸 🗸	Credential name:			
	Monitoring			
Regular Expressions		Use Session Manager Configuration $ullet$		
Defaults	CRLF Keep Alive Monitoring:	Use Session Manager Configuration \checkmark		

Scroll down to the **Listen Ports** section and verify that the transport network protocol used by the IP speaker is specified in the list below. For the compliance test, the solution was verified with UDP and TCP network transport.

Listen Ports

Add	Add Remove										
3 Ite	ms I 🍣					Filter: Enable					
	Listen Ports	Protocol	Default Domain	Endpoint	Notes						
	5060	TCP 🗸	avaya.com 🗙	Image: A start and a start							
	5060	UDP 🗸	avaya.com 💙	~							
	5061	TLS 💙	avaya.com 💙	~							
Selec	t : All, None										

6.3. Administer SIP User

In the Home screen (not shown), select Users \rightarrow User Management \rightarrow Manage Users to display the User Management screen below. Click New to add a user.

Aura® System Manager 8.1	Users 🗸 🎤	Elements 🗸 🔅 S	Services ~ Widge	ts v Shortcuts v	Search	🗌 🐥 🚍 adr	
Home Administrators	User Manage	ement					
User Management A Home 🏠 / Users R / Manage Users Help ?							
Manage Users Q							
Public Contacts	© View	v <u>∕</u> Edit	+ New 🎄 Duplicate	🔟 Delete Mor	e Actions 🗸	Options V	
Shared Addresses		First Name 🖨 🍸	Surname 🖨 🍸	Display Name 🖨	Login Name 🖨 🍸	SIP Handle \forall	
System Presence ACLs		SIP	78000	78000, SIP	78000@avaya.com	78000	
		SIP	78001	78001, SIP	78001@avaya.com	78001	
Communication Profile		SIP	78002	78002, SIP	78002@avaya.com	78002	

6.3.1. Identity

The New User Profile screen is displayed. Enter desired Last Name and First Name. For Login Name, enter "< ext > @ < domain >", where "< ext >" is the desired IP speaker extension and "< domain >" is the applicable SIP domain name from Section 5.3. Retain the default values in the remaining fields.

Avra® System	m Manager 8.1	Jsers 🗸 🎤 El	ements 🗸 🔅 Services	· √ Widgets √	Shortcuts v	Search	🗼 🗮 admin
Home	User Management						
User Man	agement ^	Home 🛆 / Users	R / Manage Users				Help? 🔺
Mana	age Users	User Pro	file Add			Commit & Continue	🗈 Commit 🛛 🛞 Cancel
Publi	c Contacts	Identity	Communication Profile	Membership	Contacts		
Share	ed Addresses	Basic Info		User Provisioning		~	
Syste	m Presence ACLs	Address		Rule :			
Comr	munication Profile	LocalizedN	ame	* Last Name :	78020	Last Name (in Latii alphabet characters) :	70020
				* First Name :	Valcom	First Name (in La alphabet characters	Velicom
				* Login Name :	78020@avaya.com	Middle Name	e: Middle Name Of User

6.3.2. Communication Profile

Select the **Communication Profile** tab. Next, click on **Communication Profile Password**. For **Comm-Profile Password** and **Re-enter Comm-Profile Password**, enter the desired password for the SIP user to use for registration. Click **OK**.

Aura® Syste	m Manager 8.1	Users 🗸 🎤 Elen	nents 🗸 🔅 Services 🕯	V Widgets	 Shortcuts 	S 🗸 Sear	ch 🔶	≡ admin
Home	Administrators	User Managemen	nt					
User Mar	nagement ^	Home☆ / Users &	/ Manage Users					Help?
Man	age Users	User Profil	e Add			Commit & Continue	🗈 Commit	S Cancel
Publ	ic Contacts	Identity	Communication Profile	Membership	Contacts			
Shar	ed Addresses	Communication	Profile Password	Edit + New	🗊 Delete		_	Options 🗸
Syste	em Presence ACLs	PROFILE SET	: Primary 🗸 🗸	Туре		Handle 🔷 🍸	Domain	\$ 7
Com	munication Profile	Communicatio				No doto		
		PROFILES	Comm-Profile Passwor	d			×	
		Session Mana	Con	nm-Profile Password	:			
		CM Endpoint I						
			* Re-enter Con	nm-Profile Password	:		Ø	
				(Generate Com	m-Profile Password		
						Cano	cel OK	

6.3.3. Communication Address

Click on **Communication Address** and then click **New** to add a new entry. The **Communication Address Add/Edit** dialog box is displayed as shown below. For **Type**, select *Avaya SIP*. For **Fully Qualified Address**, enter the SIP user extension and select the domain name to match the login name from **Section 6.3.1**. Click **OK**.

Avra® System Mar		Users 🗸 🛛 🎤 Elements	s 🗸 🔅 Services 🗸	Widgets	∽ Shortcu	its v Se	earch	♣ ≡	admin
Home Ad	ministrators	User Management							
User Manageme	ent ^	Home 🗟 / Users 🎗 / Ma	anage Users						Help?
Manage Us	ers	User Profile A	Add		E	Commit & Continue	🖻 Com	nmit 🤅	Cancel
Public Cont	tacts	Identity Com	munication Profile	Membership	Contacts				
Shared Add	dresses	Communication Profi	le Password	Edit + New	Delei	te		_	Options ~
System Pre	sence ACLs	PROFILE SET : Prim	iary 🗸	Туре		Handle 🖨 🕅	D	omain 🖨 🦷	,
Communica	ation Profile	Communication Add	dress			No data			
		PROFILES	Communication Ac	ddress Add/Edit			×		
		Session Manager P CM Endpoint Profile	*	Type: Avaya S	IP		~		
			*Fully Qualified Add	78020		@ avaya.com	~		
						Cancel	ОК		

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6.3.4. Session Manager Profile

Click on toggle button by **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.

Avra® System Manager 8.1	Users 🗸 🎤 Elements 🗸 🔅 Serv	vices v Widgets v Sh	ortcuts v	(Search \blacksquare admin
Home Administrators	User Management				
User Management ^	Home☆ / Users옷 / Manage Users				Help ? 🔺
Manage Users	User Profile Add			Commit & Continue	Commit 🛞 Cancel
Public Contacts	Identity Communication Pro	file Membership Conta	licts		
Shared Addresses	Communication Profile Password				
System Presence ACLs	PROFILE SET : Primary V	SIP Registration			
Communication Profile	Communication Address	* Primary Session Manager :	devcon-sm Q		
	PROFILES	Secondary Session			
	Session Manager Profile	Manager:	Start typing Q		
	CM Endpoint Profile	Survivability Server :	Start typing Q		
		Max. Simultaneous Devices :	Select v]	
		Block New Registration			
		When Maximum Registrations Active?			
		Application Sequences	i		
		Origination Sequence :	DEVCON-CM App Seque v]	
				,	
		Termination Sequence :	DEVCON-CM App Seque v]	

Scroll down to the **Call Routing Settings** section to configure the **Home Location**.



6.3.5. CM Endpoint Profile

Click on the toggle button by **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.1**. For **Template**, select *9641SIP_DEFAULT_CM_8_1*. For **Port**, click and select *IP*. Retain the default values in the remaining fields.

AVAYA Aura® System Manager 8.1	Users 🗸 🎤 Elements 🗸 🎄 Service	es ~ Widgets ~ She	ortcuts v	Search	📕 🔔 🗮 admin
Home Administrators	User Management				
User Management 🔷	Home🏠 / Users ႙ / Manage Users				Help ?
Manage Users	User Profile Add			🖻 Commit & Continue	🗈 Commit 🛞 Cancel
Public Contacts	Identity Communication Profile	Membership Conta	ucts		
Shared Addresses	Communication Profile Password				
System Presence ACLs	PROFILE SET : Primary V	* System :	devcon-cm v	* Profile Type :	Endpoint v
Communication Profile	Communication Address	Use Existing Endpoints :		* Extension:	78020 🖵 💆
	PROFILES				
	Session Manager Profile	* Template :	9641SIP_DEFAULT_CM_8_ Q	* Set Type :	9641SIP
	CM Endpoint Profile	Security Code :	Enter Security Code	Port:	Q Q
		Voice Mail Number :		Preferred Handle :	
		voice mail number:		Preterrea Handle :	Select v
		Calculate Route Pattern :		Sip Trunk :	aar
		SIP URI :	Select	Delete on Unassign from User	
				or on Delete User :	_
		Override Endpoint Name and Localized Name :		Allow H.323 and SIP Endpoint Dual Registration :	

7. Configure Valcom One-Way and Talkback IP Speakers

This section covers the configuration of the VIP-160A IP Talkback 8" Ceiling Speaker using the Valcom VIP-102B IP Solutions Setup Tool. The configuration of the VIP-130AL-GY IP Secure One-Way Paging Horn is similar, unless otherwise specified. The configuration covers the following areas:

- Launch the Valcom VIP-102B IP Solutions Setup Tool
- Configure the Network Settings of Valcom One-Way and Talkback IP Speakers
- Configure SIP Parameters of Valcom One-Way and Talkback IP Speakers
- Verify Codec Settings
- Specify Call Destination
- Update SIP Intercom Controller with the New Configuration

7.1. 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.

VIP-102B IP Solutions Setup Tool	_		×
File Communications Device Security Programming System Conflicts	<u>H</u> elp		
🎭 📄 🖳 🗞 · 🗢 🗲 🚧 🕼 🕼 💷 🔥	(a)		
Job Information			
■ Miscellaneous ● Miscellaneous ● Speaker Plus (FOW) ● ● ● ● 00-D0-5F-04-A7-F8 ● ● ● ● ● ● ● 00-D0-5F-04-A7-F8 ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● <			
Legend -			
 Status Unknown Status Nomal 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 : Intel(R) PRO/100+ PCI Ad	dapter - 1	92.168.100).250:

7.2. Configure the Network Settings of Valcom One-Way and Talkback IP Speakers

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

VIP-102B IP Solutions Setup Tool		- 🗆 X
<u>File</u> <u>Communications</u> <u>Device</u> Sec <u>u</u> rity	y <u>P</u> rogramming <u>S</u> ystem Co <u>n</u> fl	licts <u>H</u> elp
🔩 🔒 🔚 🕵 · 🗢 🎸	1 💭 💭 🦉 🗱	<u>1</u>
Job Information		
E Miscellaneous E Speaker Plus (FOW) E O-D0-5F-04-A7-F8	Summary Options Properties Network	rk Time Talkback Channels Inputs Relays Group Membership SIP
Speaker Plus (TB)	Host Name:	00-D0-5F-04-98-BB
🗐 00-D0-5F-04-98-BB	Domain Name:	
	Use DHCP:	Eallback
	IP Address:	192.168.100.191
	Subnet Mask:	255.255.255.0 - (/24) 🗸
	Gateway IP Address:	192.168.100.1
	Preferred DNS Server:	192.168.1.1
	Alternate DNS Server:	
	SIP SDP NAT:	
	Continuous Beacon:	
	SNMP Enabled:	SNMP Options
	Use Syslog Daemon:	Transport: UDP V
	Syslog Daemon:	Port: 514
Legend 👻		
- Status Unknown		
 Status Normal Verification Required 		
 Error Status 		
 Update Required Reset Required 		
🖃 - Rescan Required		
Invalid Password Firmware Suggested	Local Network	Copy 🖉 Apply 💥 Cancel
Find device in tree		
2 devices detected, 2 devices loaded		Ethernet : Intel(R) PRO/100+ PCI Adapter - 192.168.100.250

7.3. Configure SIP Parameters of Valcom One-Way and Talkback IP Speakers

From the **VIP-102B IP Solutions Setup Tool**, navigate to the **SIP** tab of the IP speaker. For **Transport**, select UDP or TCP transport. Set the **Phone Number** and **Authentication** to the SIP extension (e.g., 78020) and **Secret** to the SIP password used to register with Session Manager. Select the **Register** checkbox and and set the **Primary Server** to the Session Manager IP address (e.g., 10.64.102.117). Leave all other fields at their default values. Click **Apply**.

VIP-102B IP Solutions Setup Tool					_		×
File Communications Device Security	<u>P</u> rogramming <u>S</u> ystem Co	nflicts <u>H</u> elp					
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) 2 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 						
Job Information					(
■ ■ Miscellaneous Speaker Plus (FOW)	ummary Options Properties Net	work Time Talkb	back Channels In	nputs Relays Group	Membership SIP		
00-D0-5F-04-A7-F8	Transport: Accept: TCP + U	IDP, Originate: UDP		\sim			
Speaker Plus (TB)	1						
🗐 00-D0-5F-04-98-BB							
	Phone Number:	78020					
	Description:	VIP-160A				=	
	Authentication Name:	78020				=	
	Secret:	123456				=	
	Realm:	avaya.com				=	
	SIP Servers:		Server		Port		
		Primary	10.64.102.117		5060		
	Register:	Backup 1			5060	_	
	DNS SRV:	Backup 2			5060	_	
		Backup 3			5060	_	
	Busy Message:					\sim	
	Call Fwd Busy (302):						
	Outbound Proxy:			Outbound Port:	5060		
	Keep Alive Timer (secs):	3600		Options Timer (secs)	: 0		
	SIP Port:	5060		Idle Timeout (secs):	0	~	
	RTP Port:	20000		Max Call Timer (secs	i): 0	~	
Legend 🗸	CID Number:	78020					
- Status Unknown	CID Name:	VIP-160A					
Status on a low line							
 Verification Required Error Status 							
 Error Status Update Required 							
🖃 - Reset Required							
- Rescan Required							
Invalid Password							
🔲 - Firmware Suggested							
Find device in tree	<u>D</u> efaults			Сору	V Apply 🛛 💥	<u>Cancel</u>	
2 devices detected, 2 devices loaded			Ether	net : Intel(R) PRO/100	0+ PCI Adapter - 1	92.168.10	0.250 .::

7.4. Verify Codec Settings

Navigate to the **Channels** tab shown below. For this solution, the IP speaker will always advertise G.722 and G.711 in the SDP section of the SIP INVITE message. The setting of the **Codec Type** will not impact the codecs supported by the IP speaker. The **Codec Type** may be left at the default value and both codecs will still be supported.

VIP-102B IP Solutions Setup Tool				_		×
<u>File Communications Device Security P</u>	rogramming <u>S</u> ystem C	Co <u>n</u> flicts <u>H</u> elp				
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			"			
	mary Options Properties Ne	etwork Time Talkback Channels	Inputs Relays Group Membersh	ip SIP		
🔲 00-D0-5F-04-A7-F8	1					
	Channel Mode:	Talkback One-Way				
	Dial Code:	801				
	Description:					
	CID Number:	801				
	CID Name:					
	Codec Type:	G.711		\sim		
	Call Fwd Busy:					
	Audio Input Volume:		0	2		
	Audio Output Volume:		-20	5		
	Pre-Announce Tone:	Ringback Alert Tone:				
	Privacy Tone:	Incomplete Call Message:				
Legend -						
- Status Unknown						
I - Status Normal II - Verification Required						
🚍 - Error Status						
I - Update Required ■ - Reset Required						
I - Rescan Required I - Invalid Password	Group Membership	ample Page	Copy 🖉 Apply	💥 <u>C</u> ance	el.	
- Firmware Suggested						
Find device in tree						
2 devices detected, 2 devices loaded		Et	thernet : Intel(R) PRO/100+ PCI Ad	apter - 192.	168.100	.250

7.5. Specify Call Destination

For Talkback IP speakers with a call switch button only, a **SIP Auto Destination** may be configured to specify the number that should be dialed when the call switch button is pressed. In the following example, when the call switch button is pressed, the IP speaker will dial 77301. After the call is established, the IP speaker can terminate the call by pressing the call switch button again.

VIP-102B IP Solutions Setup Tool		_		\times
<u>File</u> <u>Communications</u> <u>Device</u> Sec <u>u</u> ri	ty <u>P</u> rogramming <u>S</u> ystem Co <u>n</u> flicts <u>H</u> elp			
🏭 🗟 🔲 🕵 · 🛥 🍊	+= 🏰 🗿 💷 <u> ()</u> 🎯			
Job Information				
Miscellaneous	Summary Options Properties Network Time Talkback Channels Inputs Relays Group Membership	CID		
Speaker Plus (FOW)	Summary Options Properties Network mile Tarkback Channels "Option Relays Group Membership	SIF		
	Configuration: Normal Input		\sim	
Speaker Plus (TB) Image: Control of the second s	1			
E 00-D0-DF-04-38-BB				
	Input Function: Call Switch	\sim		
	Auto Destination:			
	Sec Auto Destination:			
	SIP Auto Destination: 77301			
	SIP Sec Auto Destination:			
Legend -				
📃 - Status Unknown				
 Status Normal Verification Required 				
- Error Status				
Update Required				
 Reset Required Rescan Required 				
Invalid Password				
🚍 - Firmware Suggested				
	Copy 🖌 Apply	💢 <u>C</u> anc	.el	
Find device in tree				
2 devices detected, 2 devices loaded	Ethernet : Intel(R) PRO/100+ PCI Adapt	ter - 192	.168.10	0.250 .::

7.6. 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	×
<u>File</u> <u>Communications</u> <u>D</u> evice	Sec <u>u</u> rity <u>P</u> rogramming <u>S</u> ystem	Co <u>n</u> flicts <u>H</u> elp	
🛃 📑 🔚 😵 - 🕶	- 6 🖓 🕸 🗿 🖓	··· 🚹 🍘	
Job Information			
	Summary Options Properties	Network Time Talkback Channels Inputs Relays Group Membership	SIP
€ 00-D0-5F-04-A7-F8 	Name:		
	<u>S</u> can Device	00-D0-5F-04-98-BB	
	<u>R</u> eset Device	192.168.100.191	
	<u>U</u> pdate Device	Speaker Plus (TB) : Dual-Mode	
	Verify Device	1	
Legend	Re <u>c</u> over Device	Device - (data is retrieved from the device) $\qquad \checkmark$	
🖃 - Status Unknown	<u>A</u> ssign IP Address	3.24.5	
Status Normal	<u>P</u> rogram Firmware	5.33	
 Verification Required Error Status 	Re <u>m</u> ove Device	G3 - 1	
Update Required	Remove And Ignore	1.13.00	
 Reset Required Rescan Required 	View Cached <u>F</u> iles	Product Name = VIP-160A	
 Invalid Password Firmware Suggested 	Version Details:	startup=5.33 time=1.13.00 rescue=2.02 softwarerev=03.24.05 siprev=sw1.60.38 SwFilterRev=2	
Find device in tree		×	¥
2 devices detected, 2 devices loade	d	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.

	🛛 Wai	iting For Reset To Complete		>	×
		Waiting for devices t	to finish resetting	Cancel Wait Never Wait]
		Name	MAC Address	Туре]
	•	00-D0-5F-04-98-BB	00-D0-5F-04-98-BB	Speaker Plus (TB)	
R	eset co	omplete for 0 of 1 devices			

8. Verification Steps

This section provides the tests that may be performed to verify proper configuration of Valcom One-Way and Talkback IP Speakers with Avaya Aura® Session Manager and Avaya Aura® Communication Manager.

Verify that the IP speaker has successfully registered with Session Manager. In System Manager, navigate to Elements → Session Manager → System Status → User Registrations to check the registration status.

me Session Manager														He
ession Manager 🔷	Use	er Reai	strations											пе
Dashboard	Select		I notifications to device	s. Click on Detai	ls column f	or complete								
Session Manager Ad	_											С	uston	niz
Global Settings	Vi	ew 🔹 De	fault Export	Force Unregis		ST Device otifications:	Reboot Reloa	ad 🔹 🛛 Fai	back As of	10:47 AM		Advan	ced S	iea
Communication Prof	16 It	ems I 🍣 I	Show 15 🗸									Filt	er: Er	nab
		Details	Address	First Name	Last Name	Actual Location	IP Address	Remote	Shared Control	Simult. Devices	AST		istered	
Network Configur Y		► Show	78051@avaya.com	WFC	78051		192.168.100.194			1/1		Prim	Sec	S
Device and Locati Y		► Show	78000@avaya.com	SIP	78000		192.168.100.194			1/1		(AC)		۔ ۲
Application Confi Y		► Show	78021@avaya.com	78021	Valcom		192.168.100.193			1/1		(AC)		
		►Show	78052@avaya.com	WFC	78052		192.168.100.197			1/1		(AC)		0
System Status 🔷		► Show		WFC	78050					0/3				0
SIP Entity Monit		►Show	78003@avaya.com	SIP	78003		192.168.100.64			1/1	V	(AC)		C
Managed Band		►Show	78020@avaya.com	78020	Valcom		192.168.100.191			1/1				0
		►Show		Remote	78801					0/1				0
Security Module		►Show	78002@avaya.com	SIP	78002		192.168.100.59			1/1	V	(AC)		E
SIP Firewall Status	Selec	t: All, Non	e								4 4 Pa	ge 1	of 2	

- 2. Place a call to a Valcom IP speaker. Verify two-way audio for Talkback IP speakers and one-way audio for One-Way IP speakers. Terminate the call from the Avaya IP Deskphone or by pressing the call button on the speaker.
- 3. Place an intercom call by pressing the call button on a Talkback IP speaker. Verify two-way audio to the call destination. Terminate the call from the IP speaker by pressing the call button.

9. Conclusion

These Application Notes described the configuration steps required to integrate Valcom One-Way and Talkback IP Speakers with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Intercom and paging calls were established with Valcom VIP-130AL-GY IP Secure One-Way Paging Horn, VIP-160A IP Talkback 8" Ceiling Speaker, Avaya H.323 / SIP Deskphones, and the PSTN. All feature and serviceability test cases were completed successfully with observations noted in **Section 2.2**.

10. References

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

- [1] *Administering Avaya Aura*® *Communication Manager*, Release 8.1.x, Issue 8, November 2020, available at <u>http://support.avaya.com</u>.
- [2] *Administering Avaya Aura*® *System Manager for Release* 8.1.x, Release 8.1.x, Issue 8, November 2020, available at <u>http://support.avaya.com</u>.
- [3] Administering Avaya Aura® Session Manager, Release 8.1.x, Issue 7, October 2020, available at <u>http://support.avaya.com</u>.
- [4] Valcom VIP-102B IP Solutions Setup Tool Version 8.1.0.0 Reference Manual, Revision 12 7/10/20, available at <u>http://www.valcom.com/vipsetuptool</u>.

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

ATTACHMENT 1



Declaration of Conformance

December 1, 2020

Jeff Gartner Senior Manager DevConnect Program Avaya

Dear Jeff Gartner:

We, Valcom Inc, declare under sole responsibility that product series named IP Ceiling Speakers, IP Wall Speakers and IP Horns 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