



## **Application Notes for Valcom VE8090R SIP Intercom Controller with Avaya Aura® Communication Manager and Avaya Aura® Session Manager 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 Aura® Communication Manager and Avaya Aura® Session Manager. 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 Aura® Session Manager via a SIP trunk. 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 Aura® Communication Manager and Avaya Aura® Session Manager. 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 Aura® Session Manager via a SIP trunk. 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 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 784xx 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.

Dialed Digits	Digits Received by VE8090R	Call Type
78403	403	Intercom Call to Speaker 1
78404	404	Intercom Call to Speaker 2
78410	410	Group Call to Speaker 1 (Speaker 1 is the only group member)
78420	420	Group Call to Speaker 2 (Speaker 2 is the only group member)
78499	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 Session Manager 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.
- Originating and terminating calls through Avaya SIP telephony network.
- Terminating active calls by pressing the call button on the VIP-430A IP Wall Speaker.
- Support of G.711 mu-law codec and UDP transport protocol.
- 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 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:

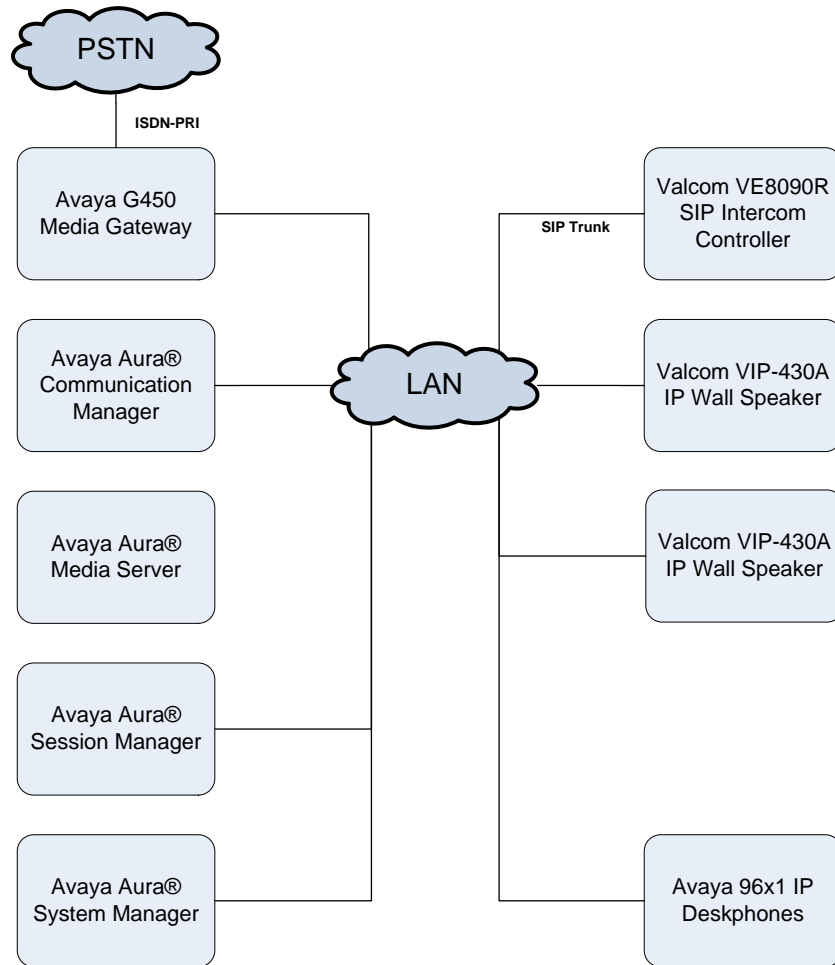
- Phone: +1 (800) 825-2661 or +1 (540) 563-2000
- Website: <https://www.valcom.com/Support/techsupport.html>
- Email: [support@valcom.com](mailto:support@valcom.com)

### 3. Reference Configuration

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

- Avaya Aura® Communication Manager running in a virtual environment with an Avaya G450 Media Gateway. Avaya G450 Media Gateway was connected to the PSTN via an ISDN-PRI trunk.
- 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.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya 96x1 Series H.323 and SIP Deskphones.
- Valcom VE8090R SIP Intercom Controller and Valcom VIP-430A IP Wall Speakers.

A SIP trunk was established between VE8090R SIP Intercom Controller and Session Manager.



**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:

<b>Equipment/Software</b>	<b>Release/Version</b>
Avaya Aura® Communication Manager	8.0.1.0.0-FP1 (R018x.00.0.822.0 with Patch 25031)
Avaya G450 Media Gateway	FW 38.21.1
Avaya Aura® Media Server	v.8.0.0.173
Avaya Aura® Session Manager	8.0.1.0801007
Avaya Aura® System Manager	8.0.1.0 Build No. – 8.0.0.0.931077 Software Update Revision No: 8.0.1.0.038826 Feature Pack 1
Avaya 96x1 Series IP Deskphones	6.7104 (H.323) 7.1.4.0.11 (SIP)
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

## 5. Configure Avaya Aura® Communication Manager

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

- 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.

### 5.1. 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
devcon-sm              10.64.102.117
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
```

## 5.2. 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
                                                           IP 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 VE8090R. 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. VE8090R supports the G.711 codec.

```

change ip-codec-set 1                                     Page 1 of 2
                                                           IP CODEC SET
  Codec Set: 1
  

| Audio Codec | Silence Suppression | Frames Per Pkt | Packet Size (ms) |
|-------------|---------------------|----------------|------------------|
| 1: G.711MU  | n                   | 2              | 20               |
| 2:          |                     |                |                  |
| 3:          |                     |                |                  |


```



### 5.3. 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*.
- 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*.
- Enable **Initial IP-IP Direct Media**.

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
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

Incoming Dialog Loopbacks: eliminate          Bypass If IP Threshold Exceeded? n
                                     RFC 3389 Comfort Noise? n
DTMF over IP: rtp-payload                Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3          IP Audio Hairpinning? n
  Enable Layer 3 Test? y                  Initial IP-IP Direct Media? 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 VE8090R and Avaya SIP deskphones. 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. Accept the default values for the remaining fields.

```

add trunk-group 10                                     Page 1 of 22
                                                    TRUNK GROUP
Group Number: 10                                     Group Type: sip                                     CDR Reports: y
  Group Name: To devcon-sm                           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

```

On **Page 5**, set the Identity for **Calling Party Display** to *From* so that the VIP-430A IP Wall Speaker's name and number is displayed on the Avaya H.323 / SIP Deskphones for calls originated from the speaker. If the field is left at the default value of *P-Asserted-Identity*, only the speaker's number is displayed (i.e., no name).

```

change trunk-group 10                               Page 5 of 5
                                                    PROTOCOL VARIATIONS
                                                    Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
  Send Transferring Party Information? n
  Network Call Redirection? n
  Send Diversion Header? n
  Support Request History? y
  Telephone Event Payload Type:
  Convert 180 to 183 for Early Media? n
  Always Use re-INVITE for Display Updates? n
  Identity for Calling Party Display: From
  Block Sending Calling Party Location in INVITE? n
  Accept Redirect to Blank User Destination? n
  Enable Q-SIP? n
Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
Request URI Contents: may-have-extra-digits

```

## 5.4. 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 add an entry that routes digits beginning with “784” to route pattern 40 as shown below.

```
change aar analysis 78
```

Page 1 of 2

AAR DIGIT ANALYSIS TABLE  
Location: all                      Percent Full: 1

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
<b>784</b>	<b>5</b>	<b>5</b>	<b>40</b>	<b>lev0</b>		<b>n</b>

Configure a preference in **Route Pattern 40** to route calls over SIP trunk group 10 as shown below. The leading dialed digits are deleted so that only the last 3 digits are sent to VE8090R. For example, if 78404 is dialed, only 404 is sent as the digits in the SIP Invite message.

```
change route-pattern 40
```

Page 1 of 3

Pattern Number: 40                      **Pattern Name: Valcom VE8090R**

SCCAN? n      Secure SIP? n      Used for SIP stations? n

Grp No	FRL	NPA	Pfx Mrk	Hop Lmt	Toll List	No. Del	Inserted Digits	DCS/ QSIG Intw	IXC user
1:	10	0				2		n	user
2:								n	user

## 6. Configure Avaya Aura® Session Manager

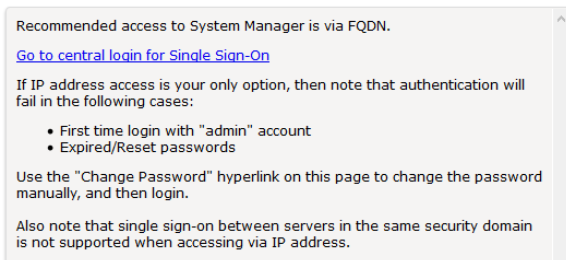
This section provides the procedure for configuring Session Manager. The procedures include adding the following items:

- Launch System Manager
- Set Network Transport Protocol
- Administer SIP Entities for Session Manager and VE8090R
- Administer Entity Link between Session Manager and VE8090R
- Add Routing Policy
- Add Dial Pattern
- Enable Monitoring on Session Manager

**Note:** It is assumed that basic configuration of Session Manager has already been completed. This section will focus on the configuration of the SIP trunk to Valcom VE8090R SIP Intercom Controller and routing calls to it.

### 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.



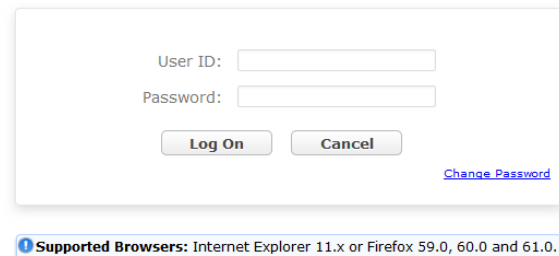
Recommended access to System Manager is via FQDN.  
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.



User ID:

Password:

[Change Password](#)

**Supported Browsers:** Internet Explorer 11.x or Firefox 59.0, 60.0 and 61.0.

## 6.2. Administer SIP Entities

This section covers the configuration of SIP Entities for Session Manager and VE8090R.

### 6.2.1. Avaya Aura® Session Manager

From the System Manager **Home** screen, navigate to **Elements** → **Routing** → **SIP Entities** and click on the **New** button (not shown). The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Select *Session Manager*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

The screenshot shows the Avaya Aura System Manager 8.0 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The main content area is titled 'SIP Entity Details' and is divided into 'General' and 'Monitoring' sections. The 'General' section contains the following fields: Name (devcon-sm), IP Address (10.64.102.117), SIP FQDN, Type (Session Manager), Notes, Location (Thornton), Outbound Proxy, Time Zone (America/New\_York), Minimum TLS Version (Use Global Setting), and Credential name. The 'Monitoring' section contains SIP Link Monitoring and CRLF Keep Alive Monitoring, both set to Use Session Manager Configuration.

Scroll down to the **Listen Ports** section and verify that the transport network protocol used by VE8090R is specified in the list below. For the compliance test, the solution used UDP network transport.

#### Listen Ports

<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Endpoint	Notes
<input type="checkbox"/>	5060	TCP	avaya.com	<input type="checkbox"/>	
<input type="checkbox"/>	5060	UDP	avaya.com	<input type="checkbox"/>	
<input type="checkbox"/>	5061	TLS	avaya.com	<input type="checkbox"/>	

Select : All, None

## 6.2.2. Valcom VE8090R SIP Intercom Controller

A SIP Entity must be added for VE8090R. To add a SIP Entity, navigate to **Elements** → **Routing** → **SIP Entities** and click on the **New** button (not shown). The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** VE8090R IP address.
- **Type:** Select *SIP Trunk*.
- **Location:** Select one of the locations previously defined.
- **Time Zone:** Time zone for this location.

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.

The screenshot shows the Avaya Aura System Manager 8.0 interface. The top navigation bar includes the Avaya logo, 'Aura © System Manager 8.0', and several menu items: 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. A search bar and a user profile 'admin' are also visible. The main content area is titled 'SIP Entity Details' and has a 'General' tab selected. The configuration fields are as follows:

* Name:	Valcom VE8090R
* FQDN or IP Address:	192.168.100.193
Type:	SIP Trunk
Notes:	
Adaptation:	
Location:	Thornton
Time Zone:	America/New_York
* SIP Timer B/F (in seconds):	4
Minimum TLS Version:	Use Global Setting
Credential name:	
Securable:	<input type="checkbox"/>
Call Detail Recording:	egress

### 6.3. Administer Entity Link between Session Manager and VE8090R

The SIP trunk between Session Manager and VE8090R is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *Valcom VE8090R Link*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select UDP transport protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the *Valcom VE8090R* SIP entity.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** Selected *Trusted*. *Note: If the link is not trusted, calls from the associated SIP Entity specified in Section 6.2.2 will be denied.*

Click **Commit** to save the Entity Link definition.

The screenshot shows the Avaya Aura System Manager 8.0 interface. The top navigation bar includes the Avaya logo, user information (Users), and various menu items (Elements, Services, Widgets, Shortcuts). A search bar and a notification bell are also present. The main content area is titled 'Entity Links' and features a table with 5 items. The table columns are: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, DNS Override, Connection Policy, Deny New Service, and Notes. The 'Valcom VE8090R Link' is highlighted with a red border. Below the table, there is a 'Select' dropdown menu with options 'All, None'.

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	DNS Override	Connection Policy	Deny New Service	Notes
<input type="checkbox"/>	<a href="#">devcon-aam Link</a>	devcon-sm	TLS	5061	devcon-aam	5061	<input type="checkbox"/>	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	<a href="#">devcon-cm Link</a>	devcon-sm	TLS	5061	devcon-cm	5061	<input type="checkbox"/>	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	<a href="#">devcon-ipose Link</a>	devcon-sm	UDP	5060	devcon-ipose	5060	<input type="checkbox"/>	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	<a href="#">Valcom VE8090R Link</a>	devcon-sm	UDP	5060	Valcom VE8090R	5060	<input type="checkbox"/>	trusted	<input type="checkbox"/>	

## 6.4. Add Routing Policy

A routing policy describes the conditions under which calls will be routed to the VE8090R SIP entity. To add a routing policy, navigate to **Elements** → **Routing** → **Routing Policies** and click on the **New** button (not shown). The following screen is displayed. Fill in the following:

Under *General*:

Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:

Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition. The following screen shows the Routing Policy for VE8090R.

The screenshot displays the Avaya Aura System Manager 8.0 interface. The top navigation bar includes the Avaya logo, user information (Users), and various menu options (Elements, Services, Widgets, Shortcuts). A search bar and a user profile icon (admin) are also present. The main content area is titled "Routing Policy Details" and includes "Commit" and "Cancel" buttons. The "General" section contains fields for "Name" (Valcom Policy), "Disabled" (checkbox), "Retries" (0), and "Notes". The "SIP Entity as Destination" section features a "Select" button and a table with columns for Name, FQDN or IP Address, Type, and Notes. The "Time of Day" section includes "Add", "Remove", and "View Gaps/Overlaps" buttons, a "Filter: Enable" option, and a table with columns for Ranking, Name, days of the week (Mon-Sun), Start Time, End Time, and Notes. The table shows one item with a ranking of 0, a name of 24/7, and a time range of 00:00 to 23:59.

Name	FQDN or IP Address	Type	Notes
Valcom VE8090R	192.168.100.193	SIP Trunk	

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	☑	☑	☑	☑	☑	☑	☑	00:00	23:59	Time Range 24/7



## 6.5. Add Dial Pattern

Dial patterns must be defined to direct calls to the appropriate SIP Entity. In the sample configuration, a 3-digit number beginning with '4' will be routed to VE8090R. To add a dial pattern, navigate to **Elements** → **Routing** → **Dial Patterns** and click on the **New** button (not shown). Fill in the following:

Under *General*:

- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **SIP Domain** SIP domain of dial pattern.
- **Notes** Comment on purpose of dial pattern (optional).

Under *Originating Locations and Routing Policies*:

Click **Add**, and then select the appropriate location and routing policy from the list.

Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows the dial pattern definition for VE8090R.

**Dial Pattern Details** Commit Cancel

**General**

\* **Pattern:**

\* **Min:**

\* **Max:**

**Emergency Call:**

**SIP Domain:**

**Notes:**

**Originating Locations and Routing Policies**

1 Item Filter: Enable

<input type="checkbox"/>	Originating Location Name <sup>▲</sup>	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Thornton		Valcom Policy	0	<input type="checkbox"/>	Valcom VE8090R	

Select : All, None

**Denied Originating Locations**

0 Items Filter: Enable

<input type="checkbox"/>	Originating Location	Notes
--------------------------	----------------------	-------

## 6.6. Enable Monitoring on Avaya Aura® Session Manager

Verify that monitoring is enabled for Session Manager. Navigate to Elements → Session Manager → Session Manager Administration, select the appropriate Session Manager and click Edit (not shown). This assumes that Session Manager has already been configured System Manager.

Next, scroll down to the **Monitoring** section, which determines how frequently Session Manager sends SIP Options messages to VE8090R. Ensure that monitoring is enabled and use default values for the remaining fields. Click **Commit** to add this Session Manager. In the following configuration, Session Manager sends a SIP Options message every 60 secs. If there is no response, Session Manager will send a SIP Options message every 120 secs.

**AVAYA** Aura® System Manager 8.0

Users v Elements v Services v Widgets v Shortcuts v Search admin

Home Session Manager Session Manager

**Edit Session Manager** Commit Cancel

General | Security Module | Monitoring | CDR | Personal Profile Manager (PPM) - Connection Settings | Event Server | Expand All | Collapse All

**General**

SIP Entity Name devcon-sm

Description

\*Management Access Point Host Name/IP 10.64.102.116

\*Direct Routing to Endpoints Enable

Data Center None

Avaya Aura Device Services Server Pairing None

Maintenance Mode

**Security Module**

SIP Entity IP Address 10.64.102.117

\*Network Mask 255.255.255.0

\*Default Gateway 10.64.102.1

\*Call Control PHB 46

\*SIP Firewall Configuration SM 6.3.8.0

**Monitoring**

Enable SIP Monitoring

\*Proactive cycle time (secs) 60

\*Reactive cycle time (secs) 120

\*Number of Tries 1

\*Number of Successes 1

## 7. Configure Valcom VE8090R SIP Intercom Controller

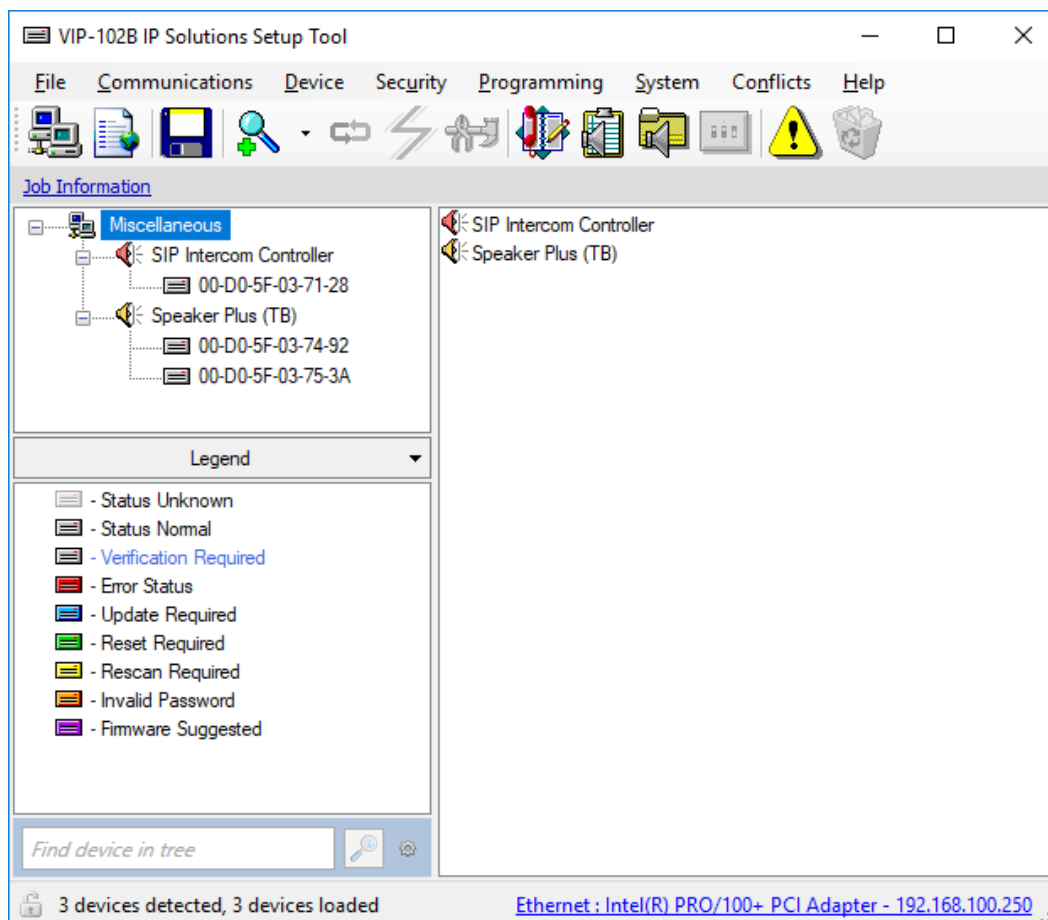
This section covers the configuration of VE8090R using the Valcom VIP-102B IP Solutions Setup Tool. The configuration covers the following areas:

- 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 [4] for details.

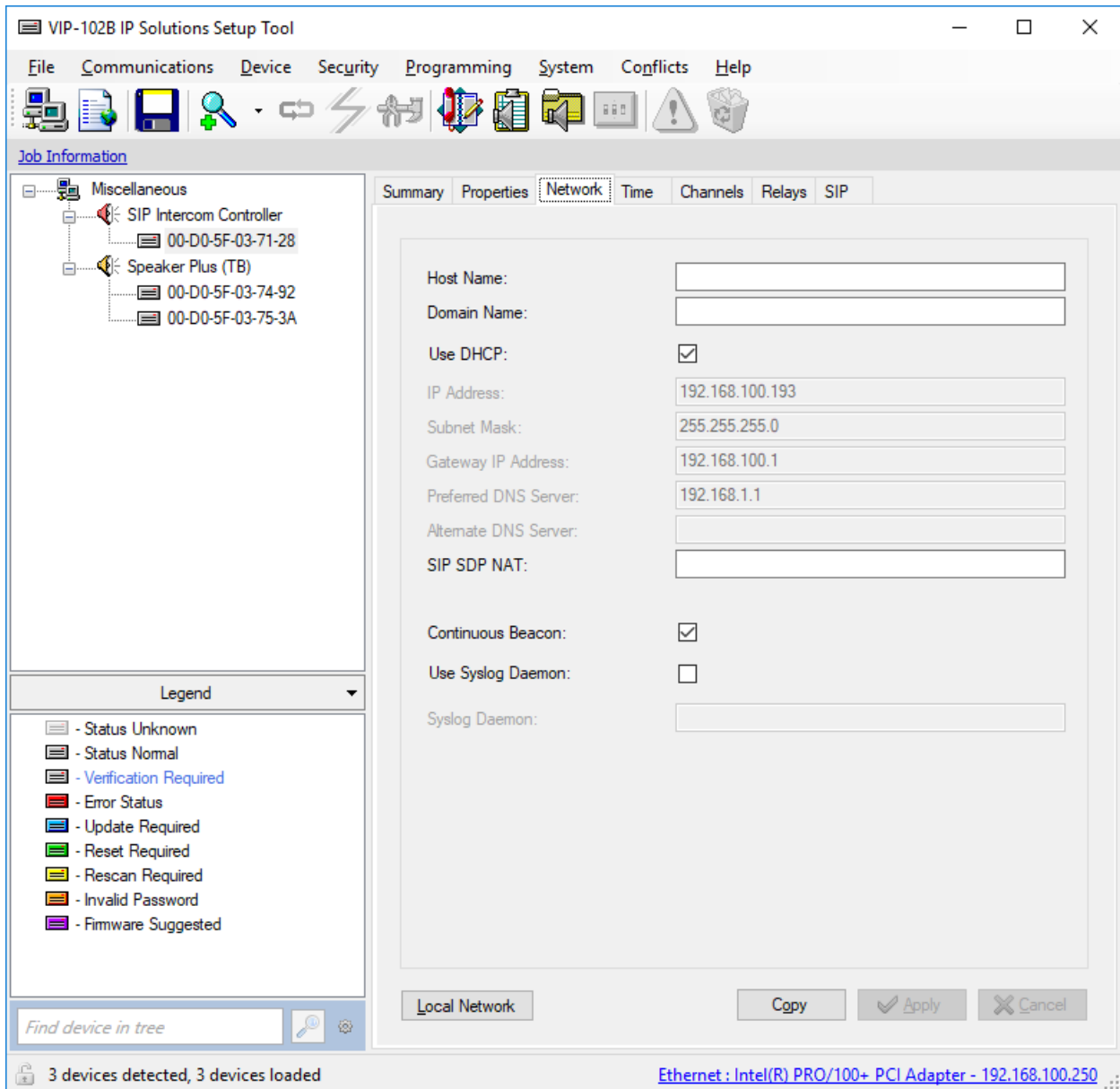
### 7.1. Launch Valcom VIP-102B IP Solutions Setup Tool

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



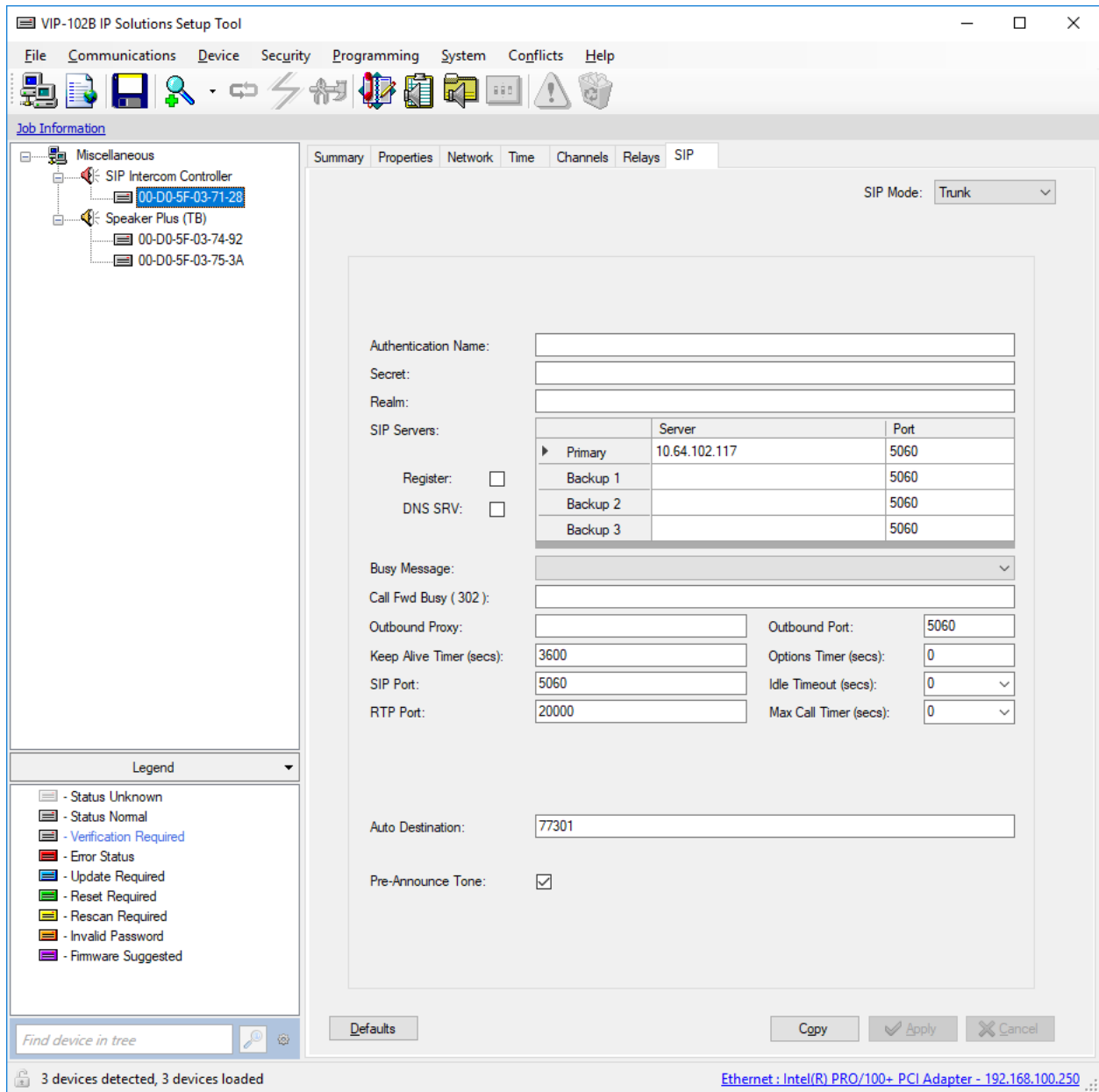
## 7.2. Configure the Network Settings of Valcom VE8090R SIP Intercom Controller

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.



### 7.3. 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 Session Manager IP address (i.e., *10.64.102.117*), 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**.



## 7.4. Verify Codec Settings

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

The screenshot displays the 'VIP-102B IP Solutions Setup Tool' interface. The 'Channels' tab is active, showing configuration for channel 1. The 'Codec Type' is set to 'G.711'. Other fields include Dial Code (401), CID Number (401), and Call Fwd No Answer (4 Rings). Audio input and output volumes are set to 0. A legend at the bottom left lists various status icons. The status bar at the bottom indicates '3 devices detected, 3 devices loaded' and the network connection is 'Ethernet : Intel(R) PRO/100+ PCI Adapter - 192.168.100.250'.

**Job Information**

- Miscellaneous
  - SIP Intercom Controller
    - 00-D0-5F-03-71-28
  - Speaker Plus (TB)
    - 00-D0-5F-03-74-92
    - 00-D0-5F-03-75-3A

**Legend**

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

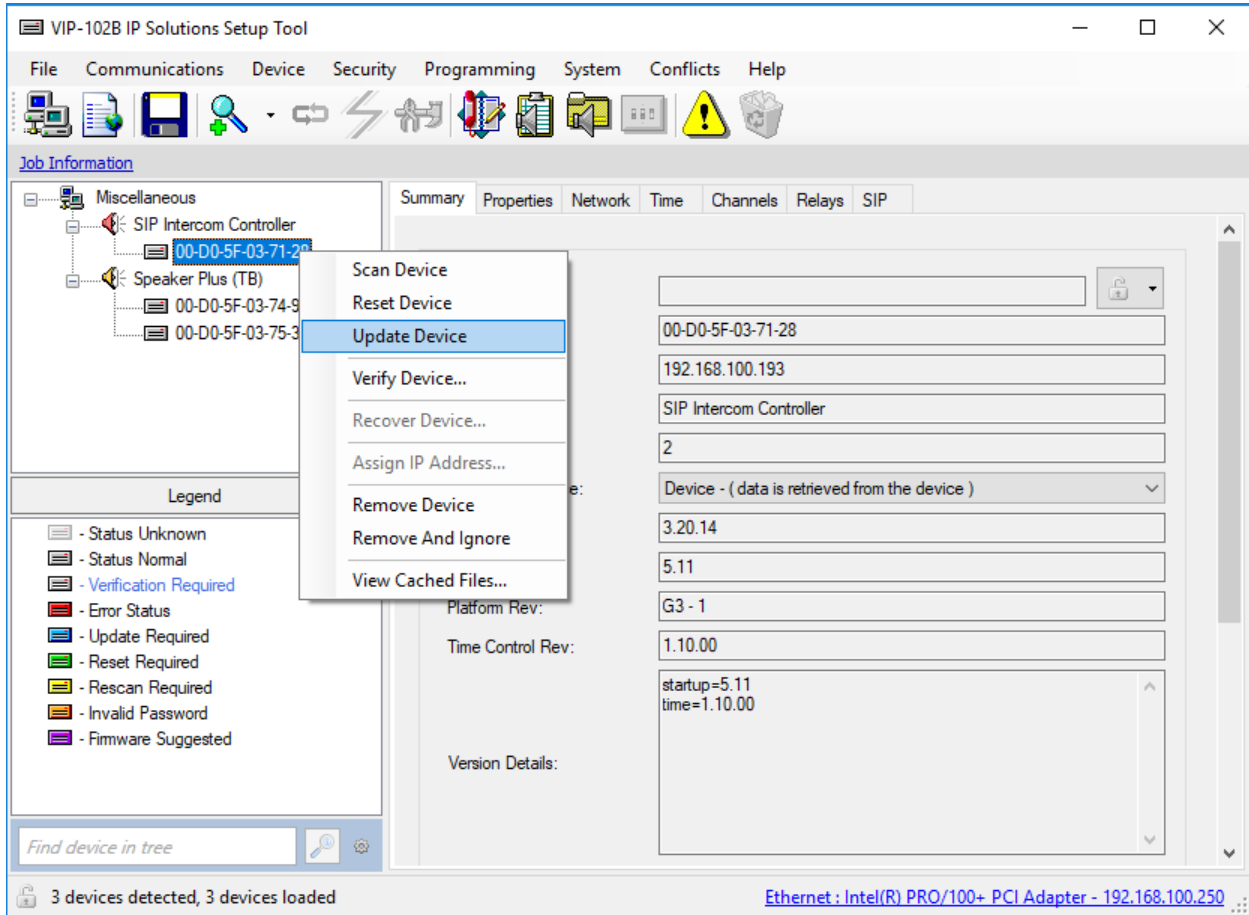
**Channels Configuration (Channel 1)**

Dial Code: 401  
Description:   
CID Number: 401  
CID Name:   
Auto Destination:   
Codec Type: G.711  
Call Fwd Busy:   
Call Fwd No Answer: 4 Rings  
Audio Input Volume: 0  
Audio Output Volume: 0  
Pre-Announce Tone:

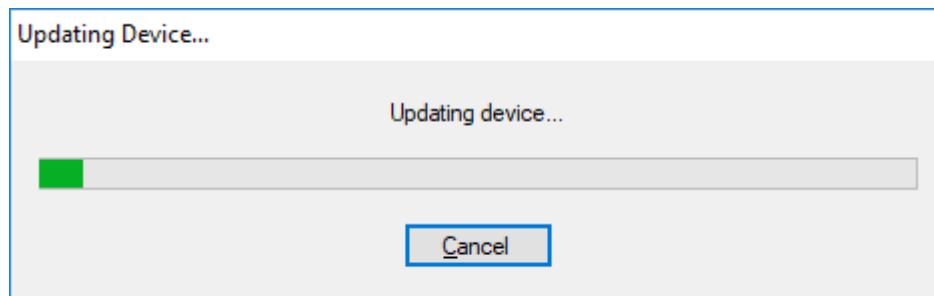
3 devices detected, 3 devices loaded  
Ethernet : Intel(R) PRO/100+ PCI Adapter - 192.168.100.250

## 7.5. Update SIP Intercom Controller with the New Configuration

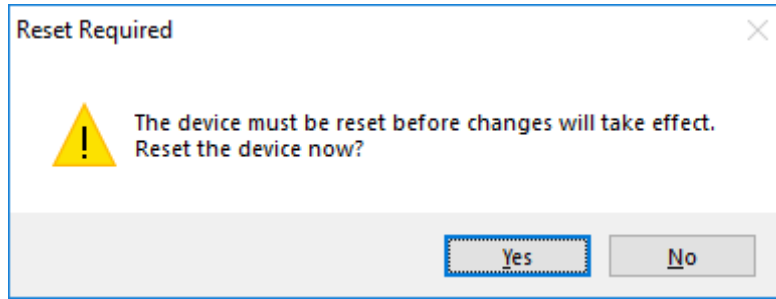
From the **VIP-102B IP Solutions Setup Tool**, right-mouse click on the MAC/hardware address of the SIP Intercom Controller and select **Update Device** from the pop-up menu as shown below.



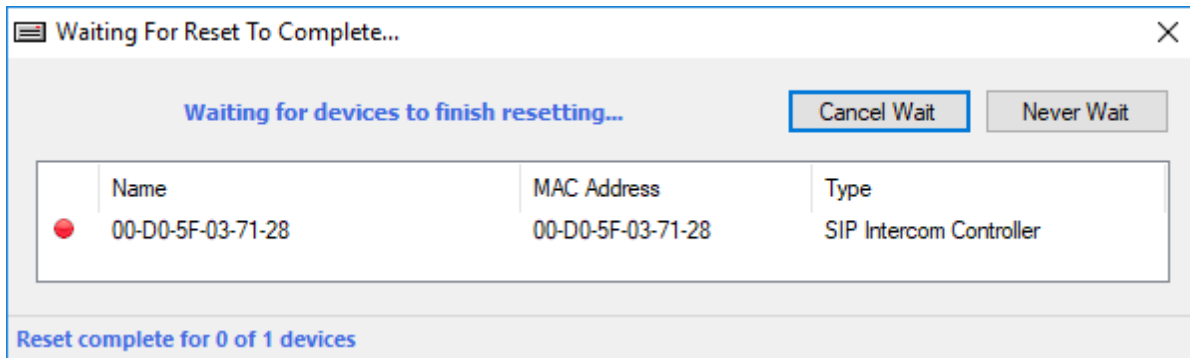
The following window is displayed indicating that the device is being updated.



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.

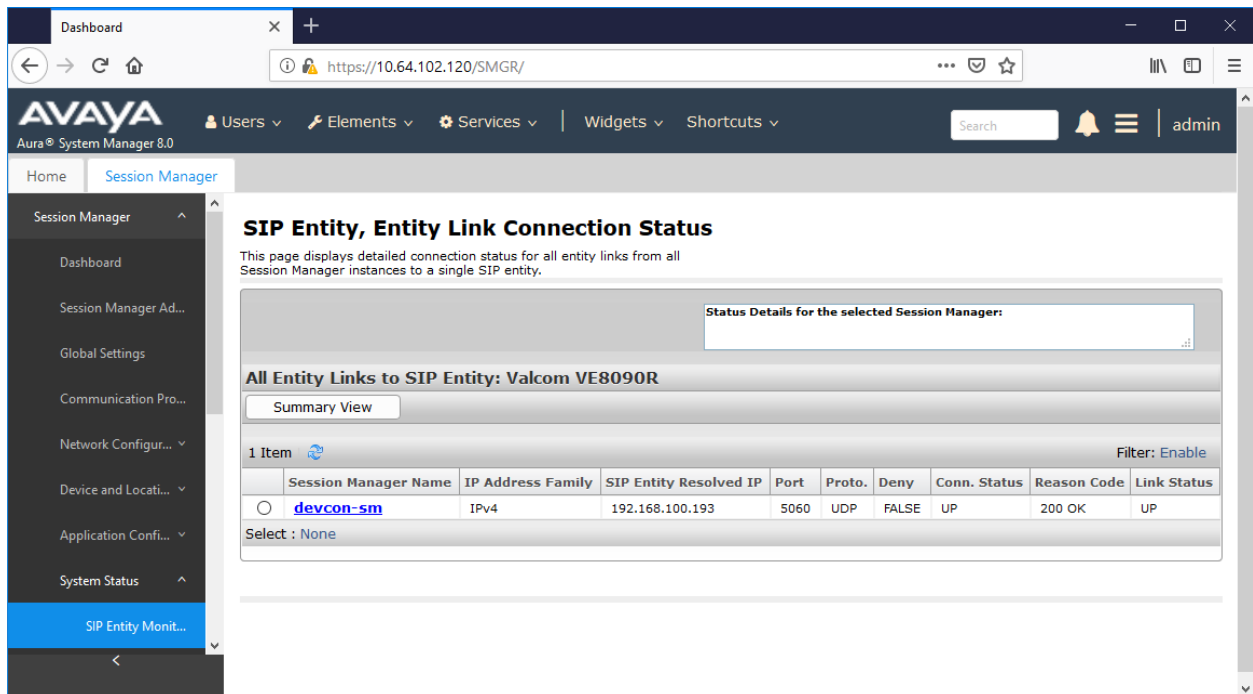




## 8. Verification Steps

This section provides the tests that may be performed to verify proper configuration of Valcom VE8090R SIP Intercom Controller with Avaya Aura® Session Manager and Avaya Aura® Communication Manager.

1. Verify that the SIP trunk between VE8090R and Session Manager has been established successfully. In System Manager, navigate to **Elements** → **Session Manager** → **System Status** → **SIP Entity Monitoring**, and then click on the Valcom VE8090R SIP entity (not shown) to check the Entity Link connection status is *UP*.



The screenshot shows the Avaya Aura System Manager 8.0 interface. The browser address bar displays <https://10.64.102.120/SMGR/>. The navigation menu on the left includes 'Session Manager', 'Dashboard', 'Session Manager Ad...', 'Global Settings', 'Communication Pro...', 'Network Configur...', 'Device and Locati...', 'Application Conf...', and 'System Status'. The 'SIP Entity Monit...' option is selected. The main content area is titled 'SIP Entity, Entity Link Connection Status' and includes a sub-header 'All Entity Links to SIP Entity: Valcom VE8090R'. Below this, there is a 'Summary View' button and a table with one item. The table columns are: Session Manager Name, IP Address Family, SIP Entity Resolved IP, Port, Proto., Deny, Conn. Status, Reason Code, and Link Status. The single row shows 'devcon-sm' with IP Address Family 'IPv4', SIP Entity Resolved IP '192.168.100.193', Port '5060', Proto. 'UDP', Deny 'FALSE', Conn. Status 'UP', Reason Code '200 OK', and Link Status 'UP'. A 'Filter: Enable' button is also visible.

Session Manager Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
devcon-sm	IPv4	192.168.100.193	5060	UDP	FALSE	UP	200 OK	UP

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.
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.

## 9. Conclusion

These Application Notes described the configuration steps required to integrate Valcom VE8090R SIP Intercom Controller with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. 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.

## 10. References

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

- [1] *Administering Avaya Aura® Communication Manager*, Release 8.0.1, Issue 3, December 2018, available at <http://support.avaya.com>.
- [2] *Administering Avaya Aura® System Manager for Release 8.0.1*, Release 8.0.x, Issue 7, January 2019, available at <http://support.avaya.com>.
- [3] *Administering Avaya Aura® Session Manager*, Release 8.0.1, Issue 3, December 2018, available at <http://support.avaya.com>.
- [4] *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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Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at [devconnect@avaya.com](mailto:devconnect@avaya.com).



## 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,

A handwritten signature in black ink that reads "David Ellison". The signature is written in a cursive style.

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

<b>Model</b>	<b>Software Rev.</b>	<b>Description</b>
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