



Application Notes for Valcom VE8090R SIP Intercom Controller with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Endpoint - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Valcom VE8090R SIP Intercom Controller with Avaya 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 as a SIP endpoint. The Valcom VE8090R SIP Intercom Controller supports two-way audio intercom calls and one-way audio group paging calls.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required to integrate the Valcom VE8090R SIP Intercom Controller with Avaya 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 as a SIP endpoint. The Valcom VE8090R SIP Intercom Controller supports two-way audio intercom calls and one-way audio group paging calls.

When the Valcom VE8090R SIP Intercom Controller is configured in SIP station mode and a call is placed to it using its direct dial SIP extension, the VE8090R plays dial tone back to the caller. The caller can then dial a Valcom speaker Dial Code or Group Code to establish an intercom call (two-way audio) with a single Valcom speaker or a group call (one-way audio) to one or more Valcom speakers.

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

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

Note: Valcom has indicated that other products in the SIP Intercom Controller family share the same hardware circuitry, software, SIP stack and firmware version 3.20.14, which was compliance tested; therefore, this testing also applies to those products. The differences between the products are detailed in **Attachment 1**. For additional details contact Valcom Support, as noted in **Section 2.3**.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between the Valcom VE8090R SIP Intercom Controller using the Valcom VIP-430A IP Wall Speakers, Avaya SIP / H.323 IP Deskphones, and the PSTN. Two-way audio intercom calls and one-way audio group paging calls were exercised.

The serviceability testing focused on verifying that the Valcom VE8090R SIP Intercom Controller came back into service after a reboot.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and Valcom VE8090R SIP Intercom Controller did not include use of any specific encryption features as requested by Valcom.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of VE8090R with Session Manager.
- 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.
- 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

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.

VE8090R SIP Intercom Controller registered with Session Manager as a SIP endpoint and was configured as Off-PBX Stations (OPS) on Communication Manager.

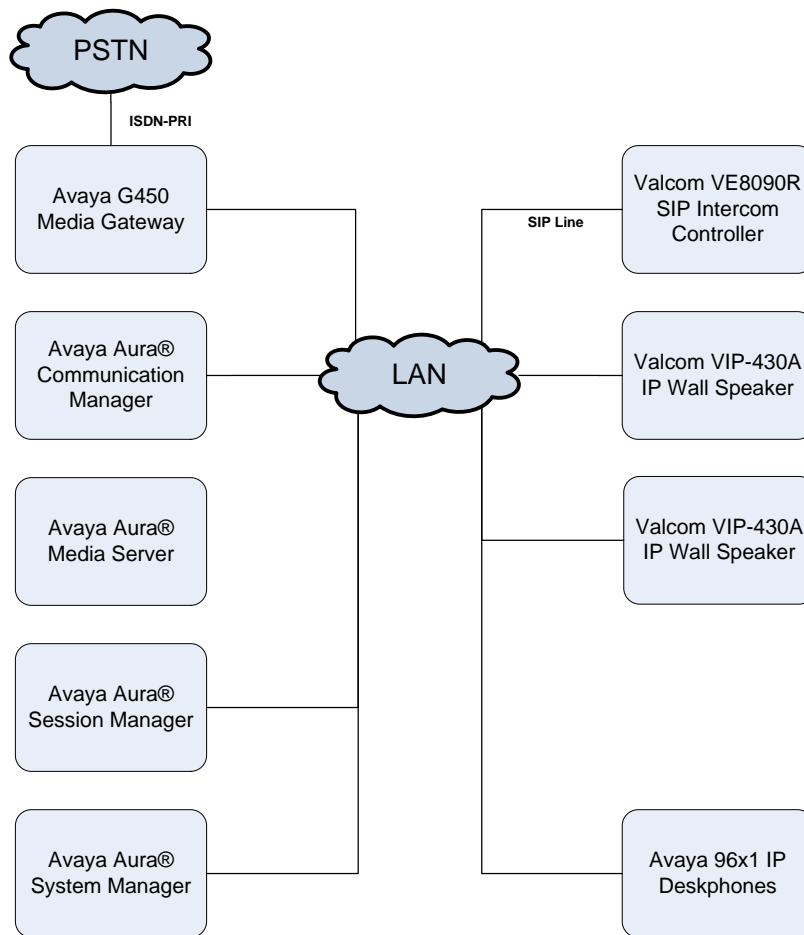


Figure 1: Avaya SIP Network with Valcom VE8090R SIP Intercom Controller and Valcom VIP-430A IP Wall Speakers

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
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	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:

- 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 Valcom VE8090R SIP Intercom Controller is configured 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.

```
display system-parameters customer-options                               Page 1 of 12
                                OPTIONAL FEATURES

G3 Version: V18                                     Software Package: Enterprise
Location: 2                                           System ID (SID): 1
Platform: 28                                         Module ID (MID): 1

                                                USED
Platform Maximum Ports: 48000      87
Maximum Stations: 36000 26
Maximum XMOBILE Stations: 36000 0
Maximum Off-PBX Telephones - EC500: 41000 0
Maximum Off-PBX Telephones - OPS: 41000 17
Maximum Off-PBX Telephones - PBFMC: 41000 0
Maximum Off-PBX Telephones - PVFMC: 41000 0
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
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
```

5.3. 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
                                                           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 was tested using 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:          |                     |                |                  |


```


5.4. 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*.
- 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
DTMF over IP: rtp-payload                    RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3           Direct IP-IP Audio Connections? y
  Enable Layer 3 Test? y                     IP Audio Hairpinning? n
H.323 Station Outgoing Direct Media? n       Initial IP-IP Direct Media? y
                                             Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for SIP calls to/from VE8090R, 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

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
  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      Type      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.

```

change route-pattern 10                             Page 1 of 3
      Pattern Number: 10      Pattern Name: To devcon-sm
      SCCAN? n      Secure SIP? n      Used for SIP stations? n

      Grp FRL NPA Pfx Hop Toll No.      Inserted      DCS/ IXC
      No      Mrk Lmt List Del      Digits      QSIG
      Dgts      Intw

1: 10      0
2:
3:
4:
5:
6:
                                     n      user
                                     n      user
                                     n      user
                                     n      user
                                     n      user

      BCC VALUE      TSC CA-TSC      ITC BCIE Service/Feature PARM Sub      Numbering LAR
      0 1 2 M 4 W      Request      Dgts      Format

1: y y y y y n      n      rest      unk-unk      none
2: y y y y y n      n      rest      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 VE8090R SIP Intercom Controller.

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. Set Network Transport Protocol

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

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:** (empty)
- Type:** Session Manager
- Notes:** (empty)
- Location:** Thornton
- Outbound Proxy:** (empty)
- Time Zone:** America/New_York
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty)
- SIP Link Monitoring:** Use Session Manager Configuration
- CRLF Keep Alive Monitoring:** 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.

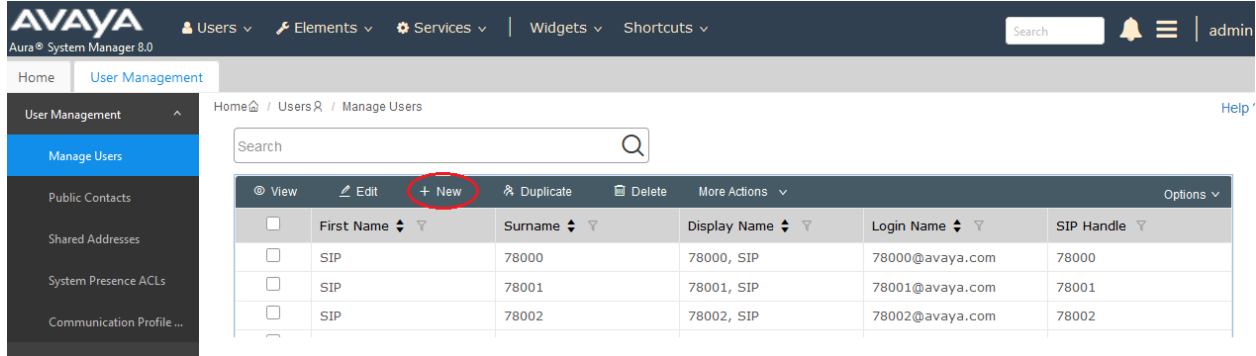
Listen Ports

<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

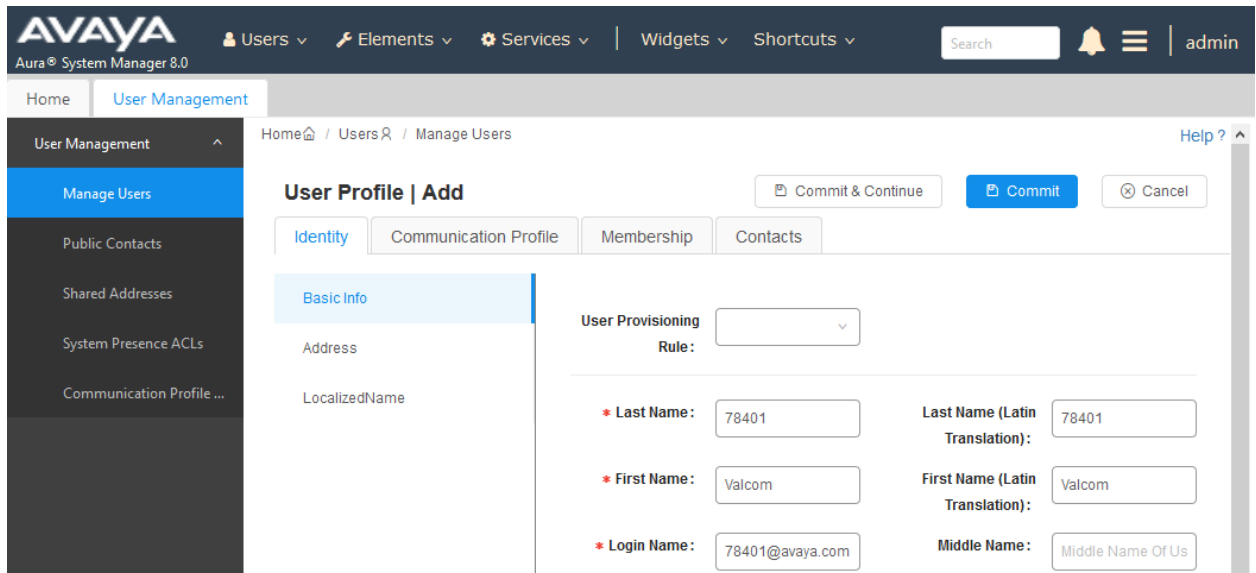
6.3. Administer SIP User

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



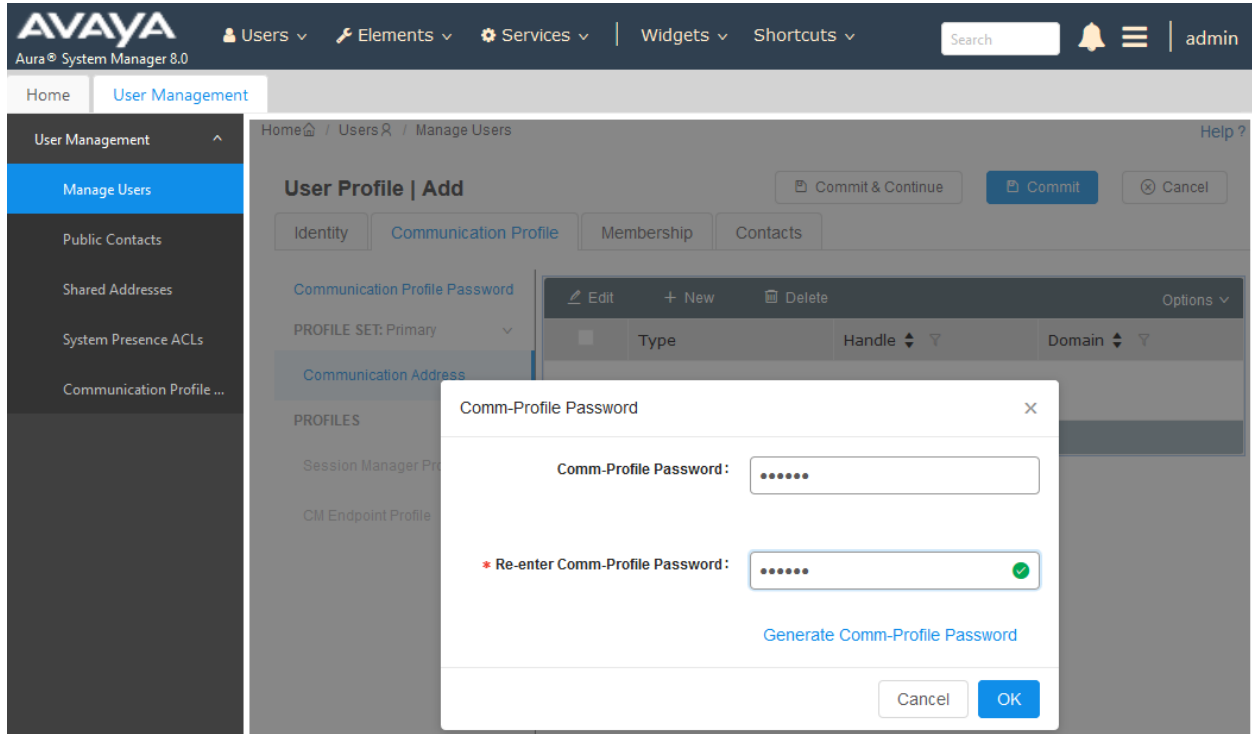
6.3.1. Identity

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 VE8090R SIP extension and “<domain>” is the applicable SIP domain name from **Section 5.3**. Retain the default values in the remaining fields.



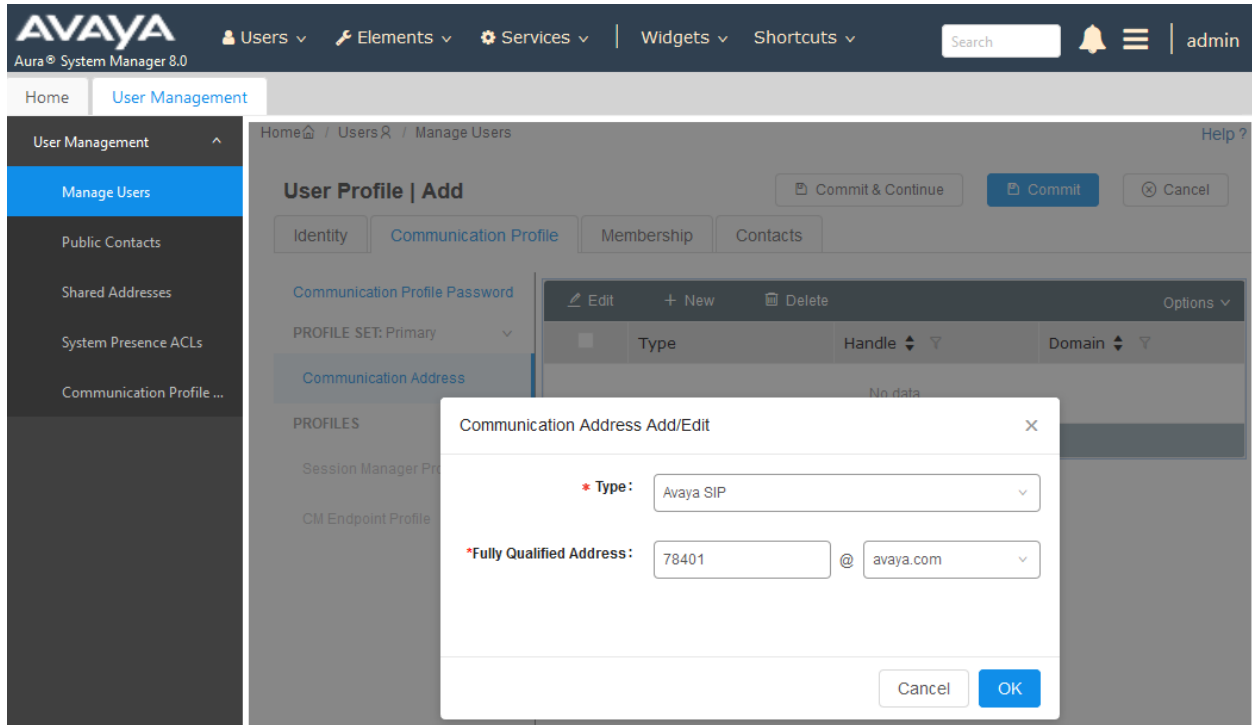
6.3.2. Communication Profile

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**.



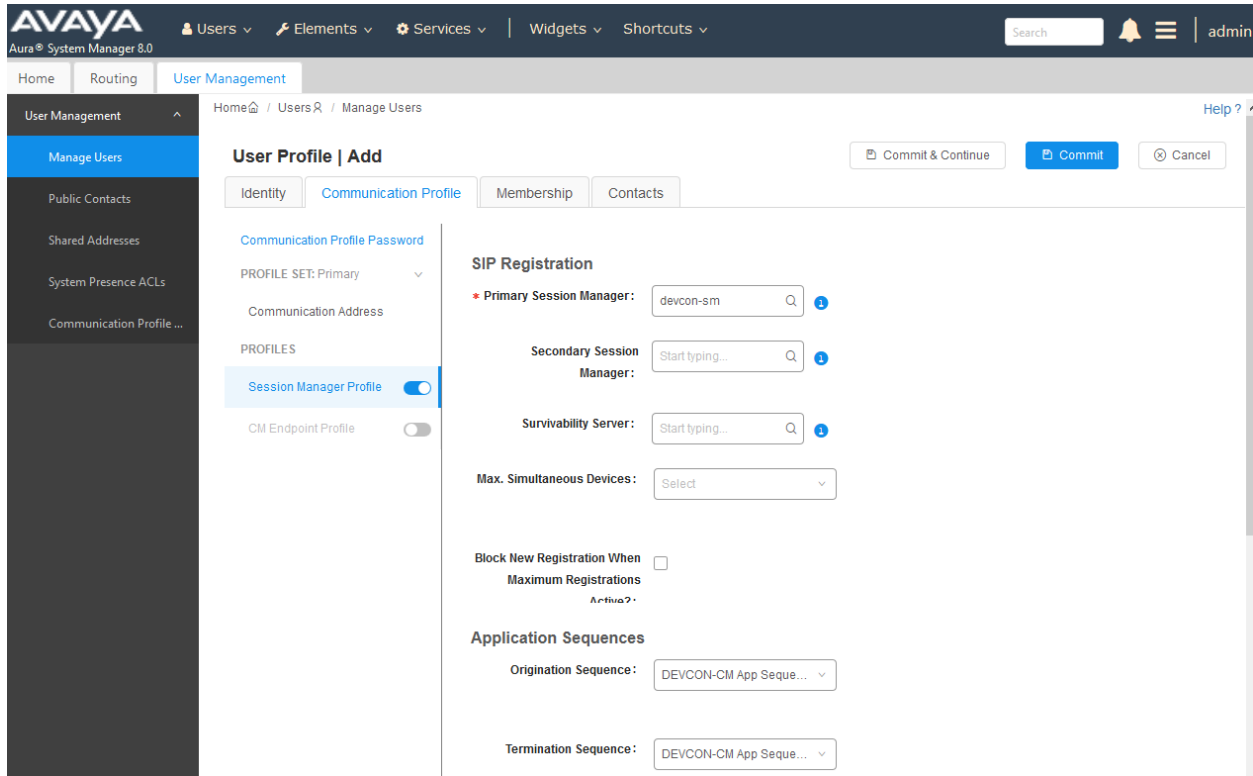
6.3.3. Communication Address

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**.

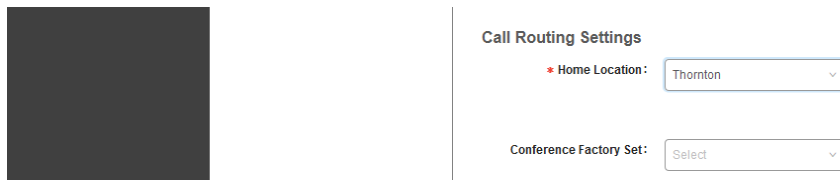


6.3.4. Session Manager Profile

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.



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 `9600SIP_DEFAULT_CM_8_0`. For **Port**, click and select `IP`. Retain the default values in the remaining fields.

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 menu items for Users, Elements, Services, Widgets, and Shortcuts. A search bar and user profile (admin) are also visible. The main content area is titled 'User Profile | Add' and has tabs for Identity, Communication Profile, Membership, and Contacts. The 'Communication Profile' tab is active. On the left, a sidebar shows 'User Management' with 'Manage Users' selected, and a 'CM Endpoint Profile' toggle is turned on. The main form contains the following fields and options:

- System:** devcon-cm
- Profile Type:** Endpoint
- Extension:** 78401
- Set Type:** 9600SIP
- Template:** 9600SIP_DEFAULT_CM
- Sub Type:** Select
- Terminal Number:** [Four empty input boxes]
- System ID:** Enter System Id
- Security Code:** Enter Security Code
- Port:** IP
- Voice Mail Number:** [Empty input box]
- Preferred Handle:** Select
- Calculate Route Pattern:**
- Sip Trunk:** aar
- SIP URI:** Select
- Enhanced Callr-Info display for 1-line phones:**
- 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 VE8090R SIP Intercom Controller

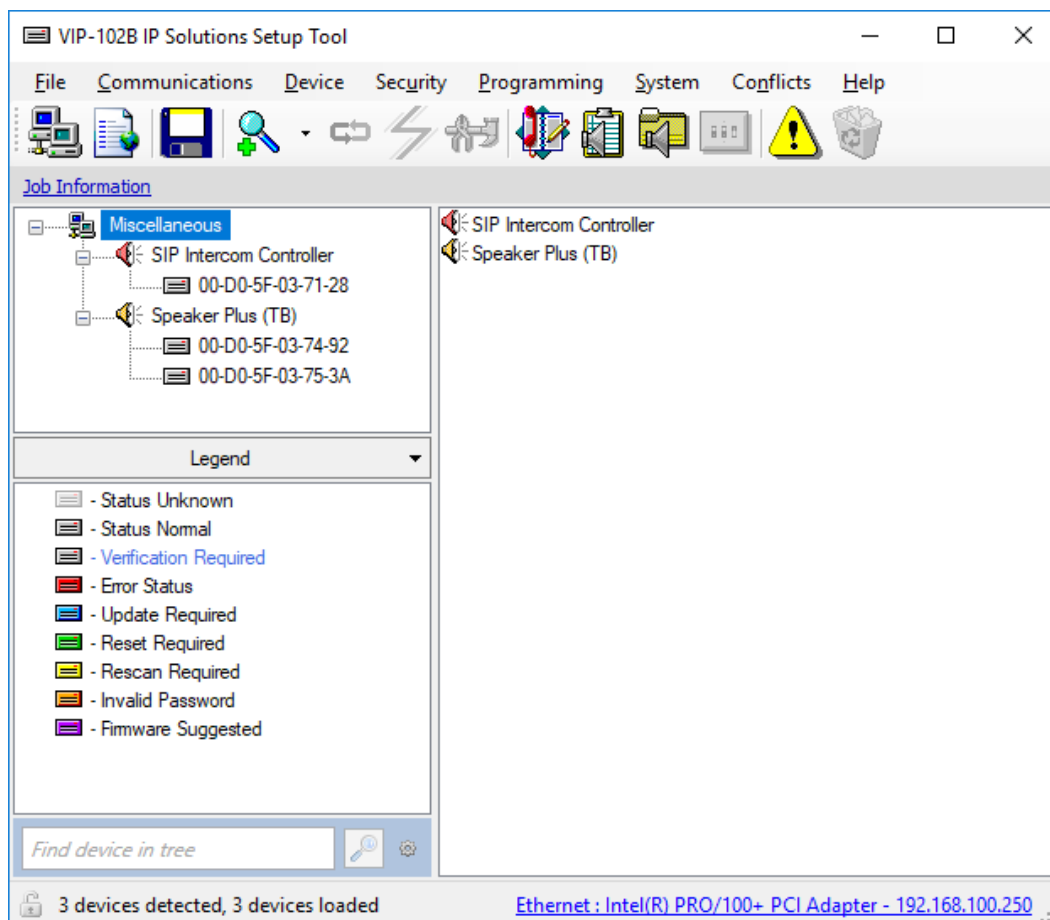
This section covers the configuration of VE8090R using the Valcom VIP-102B IP Solutions Setup Tool. The configuration covers the following areas:

- 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.

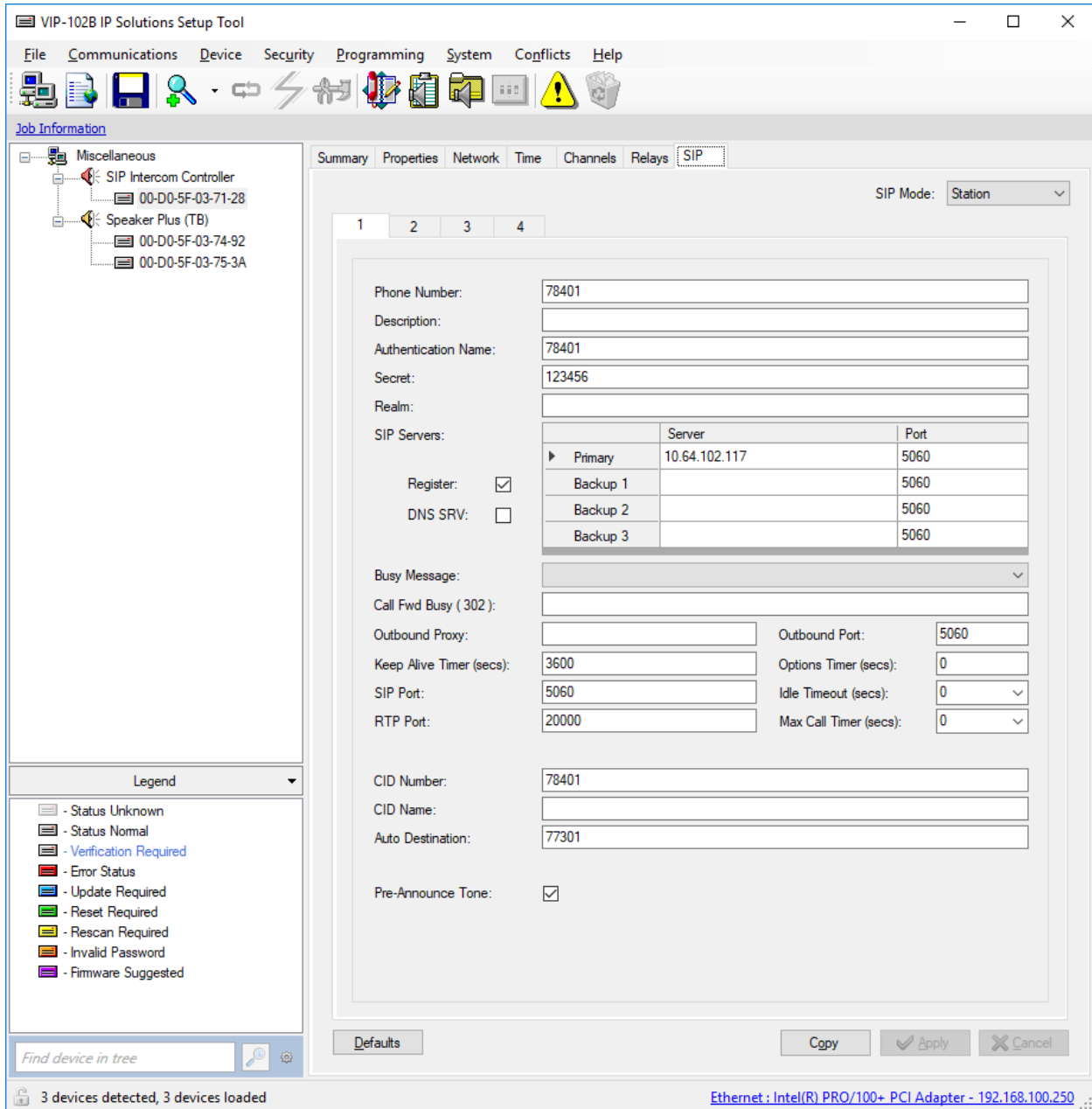
The screenshot displays the 'VIP-102B IP Solutions Setup Tool' interface. The left pane shows a tree view under 'Miscellaneous' with 'SIP Intercom Controller' selected, showing its MAC address '00-D0-5F-03-71-28'. Below the tree is a 'Legend' section with various status icons. The main pane shows the 'Network' tab with the following configuration:

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

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

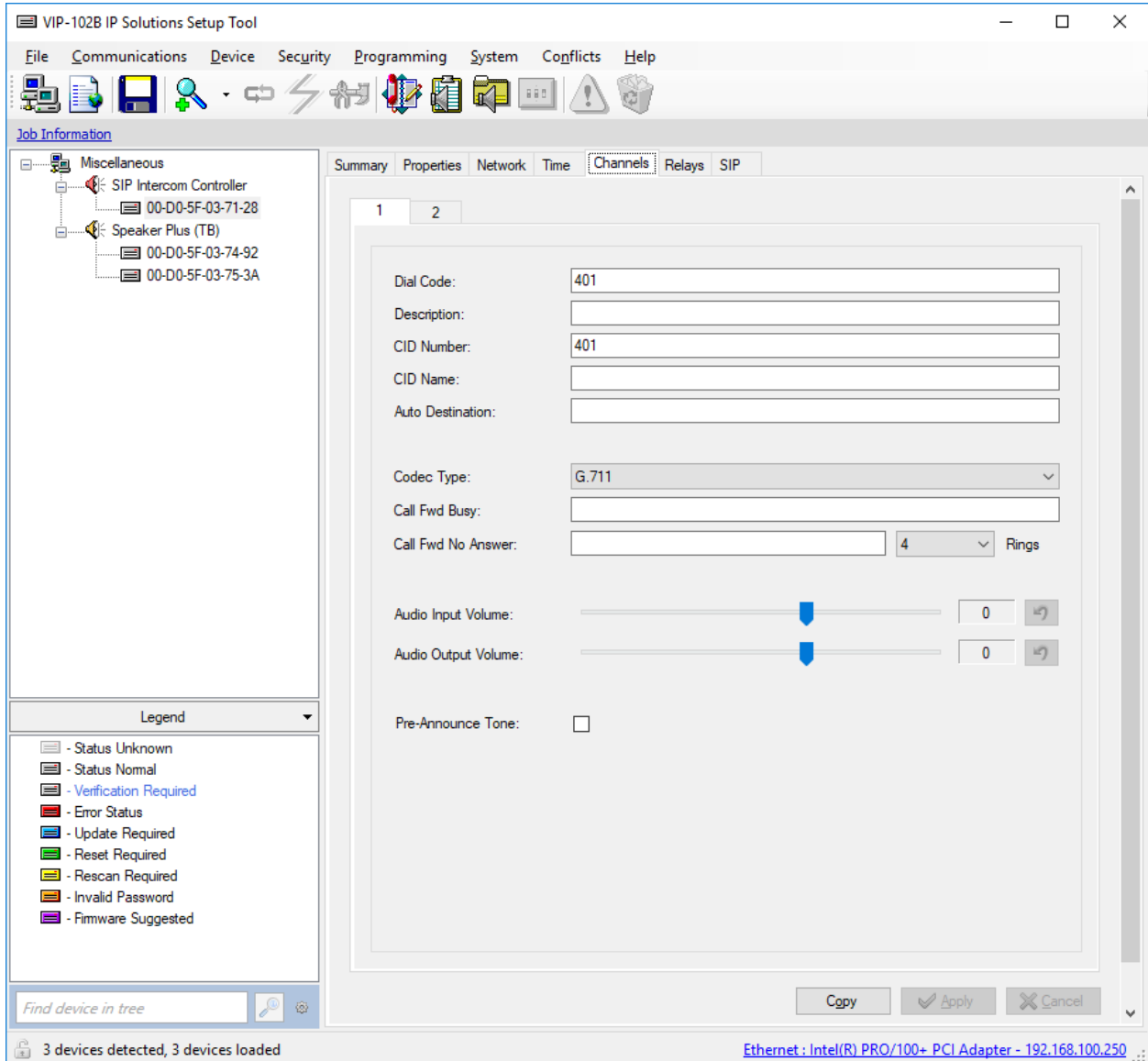
7.3. Configure SIP Parameters of Valcom VE8090R SIP Intercom Controller

From the **VIP-102B IP Solutions Setup Tool**, navigate to the **SIP** tab of the SIP Intercom Controller. Set the **SIP Mode** to *Station*, 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. Select the **Register** checkbox and leave all other fields at their default values. Click **Apply**.



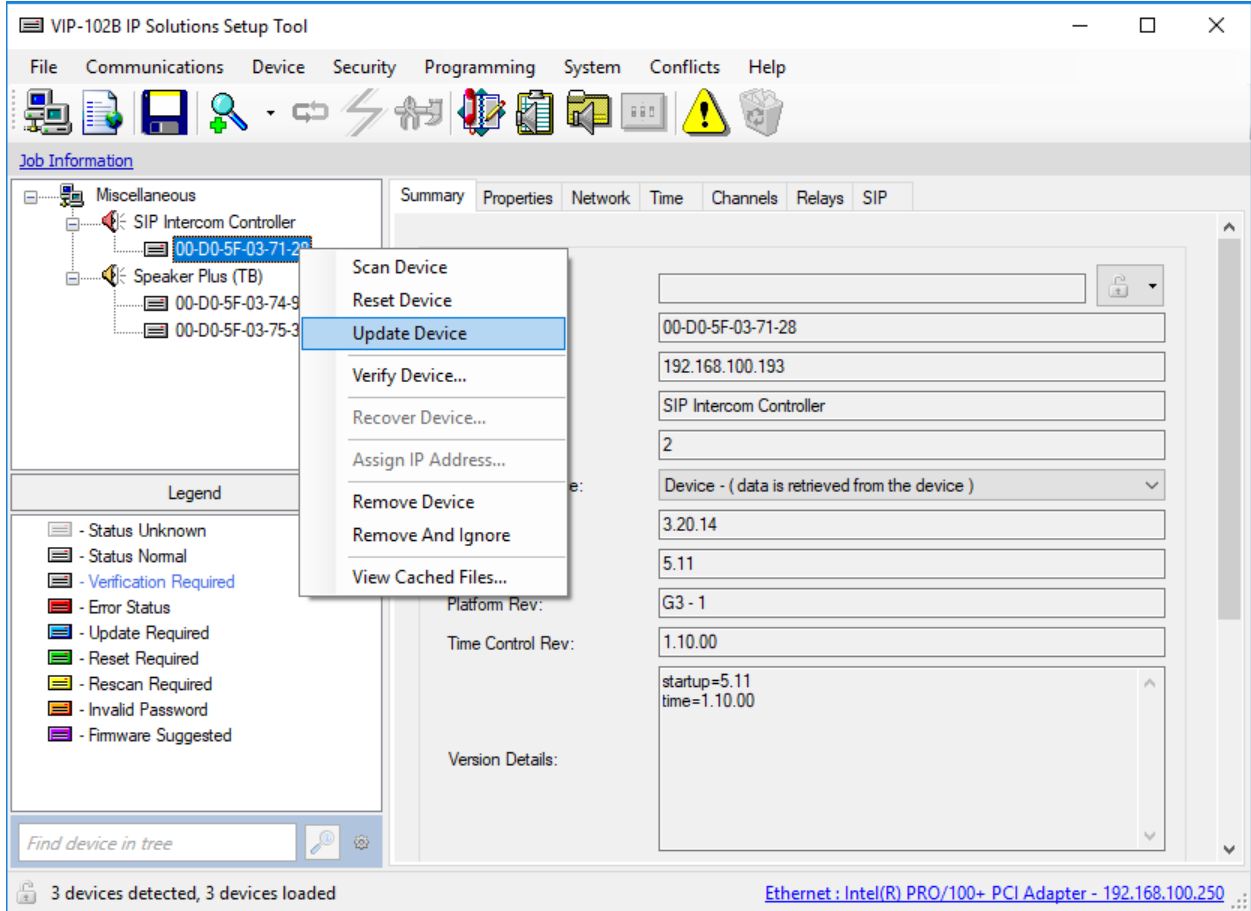
7.4. Verify Codec Settings

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

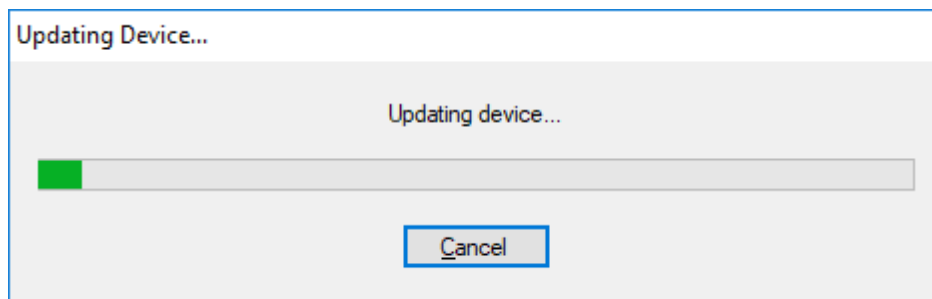


7.5. Update SIP Intercom Controller with the New Configuration

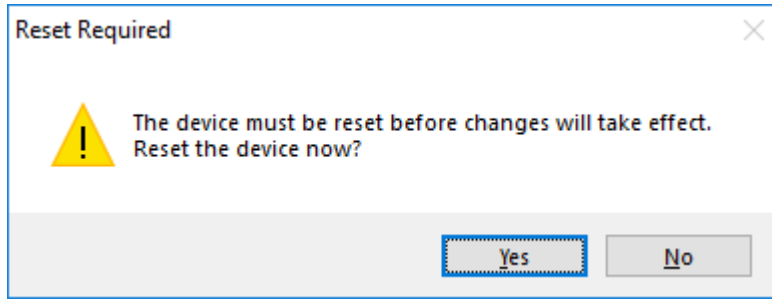
From the **VIP-102B IP Solutions Setup Tool**, right-mouse click on the MAC/hardware address of the SIP Intercom Controller and select **Update Device** from the pop-up menu as shown below.



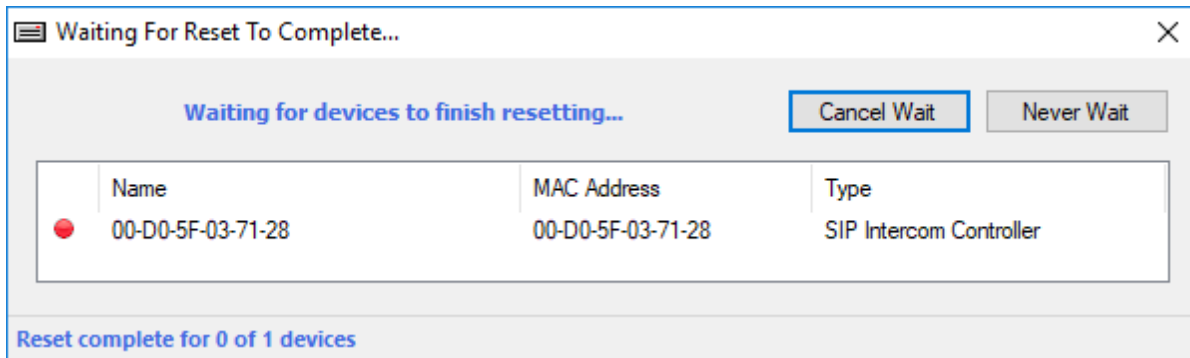
The following window is displayed indicating that the device is being updated.



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 VE8090R has successfully registered with Session Manager. In System Manager, navigate to **Elements** → **Session Manager** → **System Status** → **User Registrations** to check the registration status.

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 'User Registrations' and contains a table of 17 items. The table has the following columns: Details, Address, First Name, Last Name, Actual Location, IP Address, Remote Office, Shared Control, Simult. Devices, AST Device, and Registered (Prim, Sec, Surv). The row for '78401@avaya.com' is highlighted with a red box. The 'Registered' column for this row shows 'Prim' checked and 'Sec' and 'Surv' unchecked. The page also includes a search bar, a 'Filter: Enable' button, and a 'Page 1 of 2' indicator.

	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered		
											Prim	Sec	Surv
<input type="checkbox"/>	Show	78000@avaya.com	SIP	78000	---	192.168.100.54	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	Equinox	78040	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	78030@avaya.com	Agent	SIP	---	192.168.100.49	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	78002@avaya.com	SIP	78002	---	192.168.100.53	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	78001@avaya.com	SIP	78001	---	192.168.100.58	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	SIP	78400	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	78401@avaya.com	Valcom	78401	---	192.168.100.193	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

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.



Declaration of Conformance

March 4, 2019

Jeff Gartner
Senior Manager
DevConnect Program
Avaya

Dear Jeff Gartner:

We, Valcom Inc, declare under sole responsibility that product series named SIP Intercom Controller all share the same hardware circuitry, software, SIP stack and firmware version. Therefore, the products are expected to behave in the same manner. The differences between the different models in the series are detailed in the table below.

Sincerely,

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

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