Avaya Solution & Interoperability Test Lab

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

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.

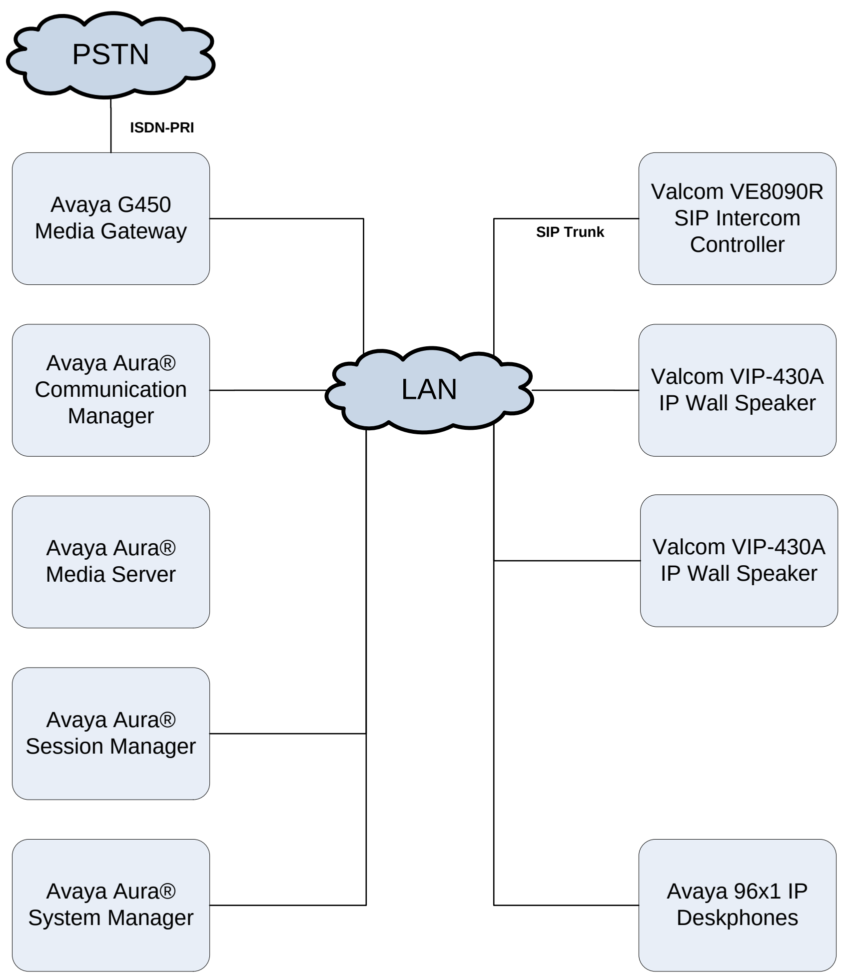


Figure 1: Avaya SIP Network with Valcom VE8090R SIP Intercom Controller and Valcom VIP-430A IP Wall Speakers

4. Equipment and Software Validated

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

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 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        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  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
                                           Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                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           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    784                   5    5     40       lev0         n
```

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

6. Configure Avaya Aura® Session Manager

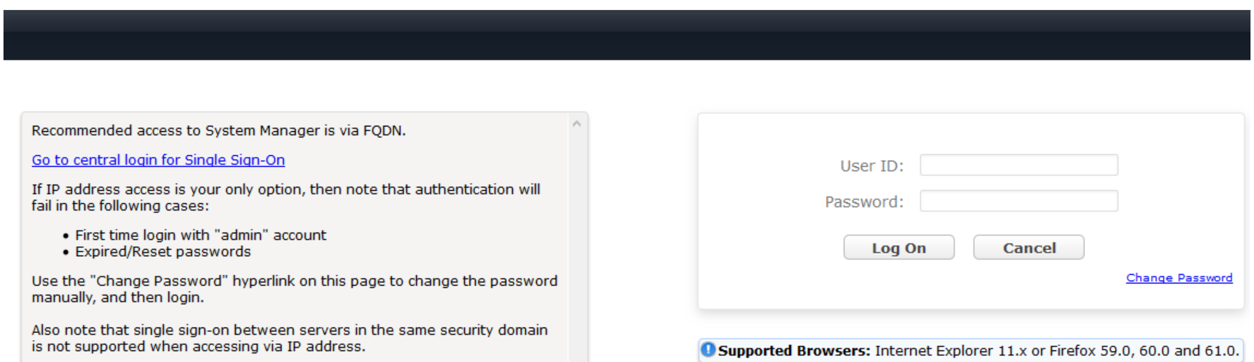
This section provides the procedure for configuring Session Manager. The procedures include adding the following items:

- 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.



6.2. Administer SIP Entities

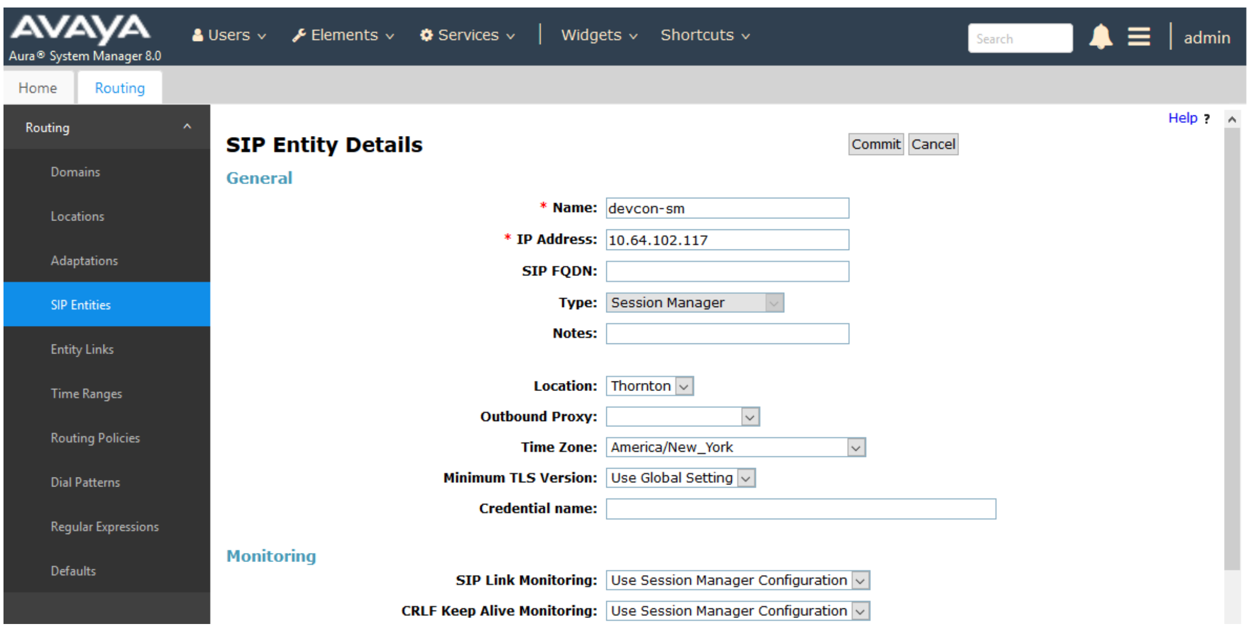
This section covers the configuration of SIP Entities for Session Manager and VE8090R.

6.2.1. Avaya Aura® Session Manager

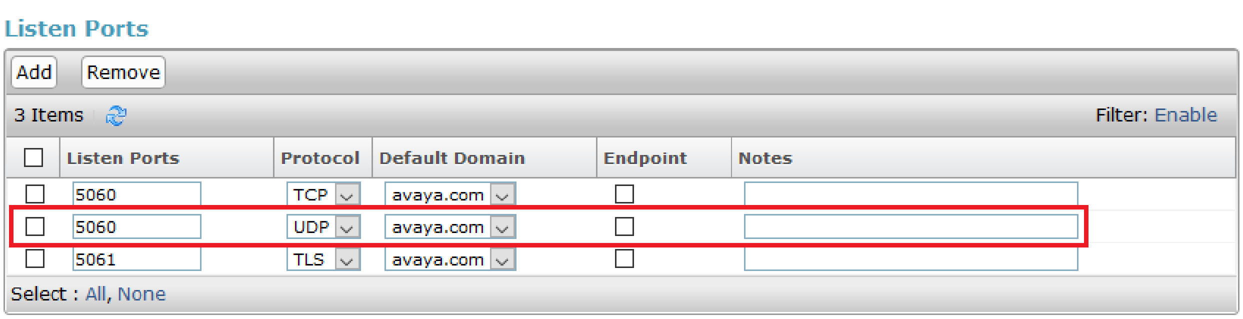
From the System Manager **Home** screen, navigate to **Elements → Routing → SIP Entities** and click on the **New** button (not shown). The following screen is displayed. Fill in the following:

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.



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.



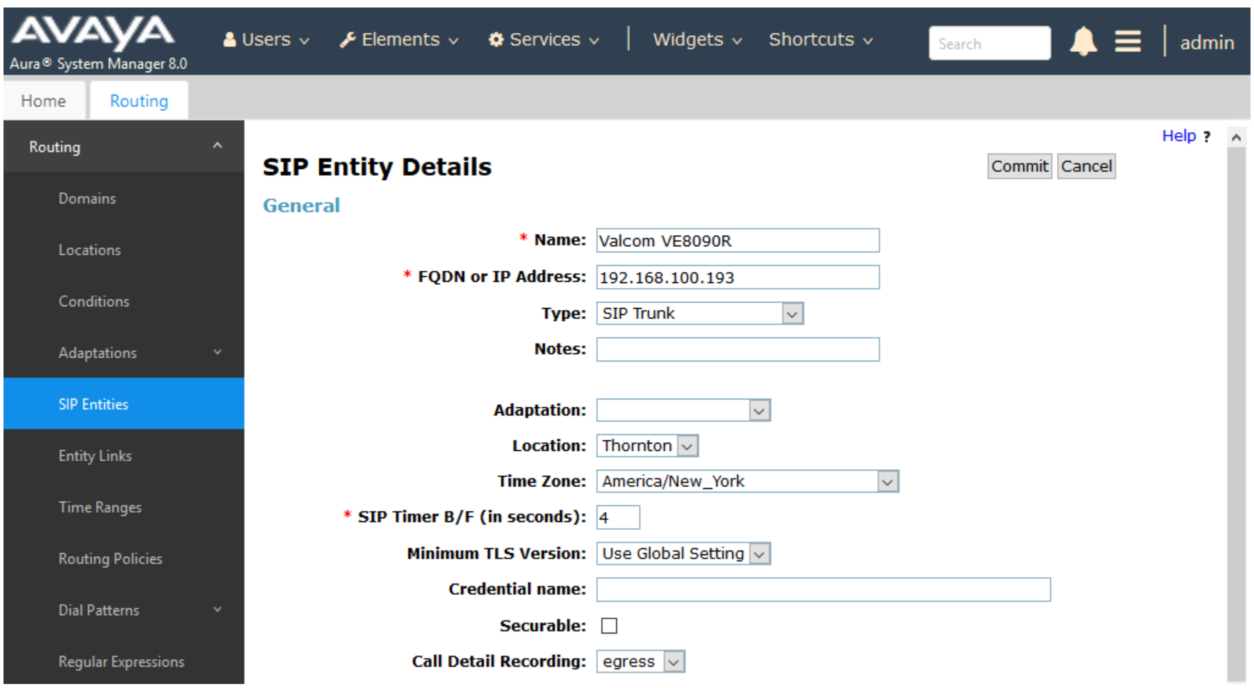
6.2.2. Valcom VE8090R SIP Intercom Controller

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.

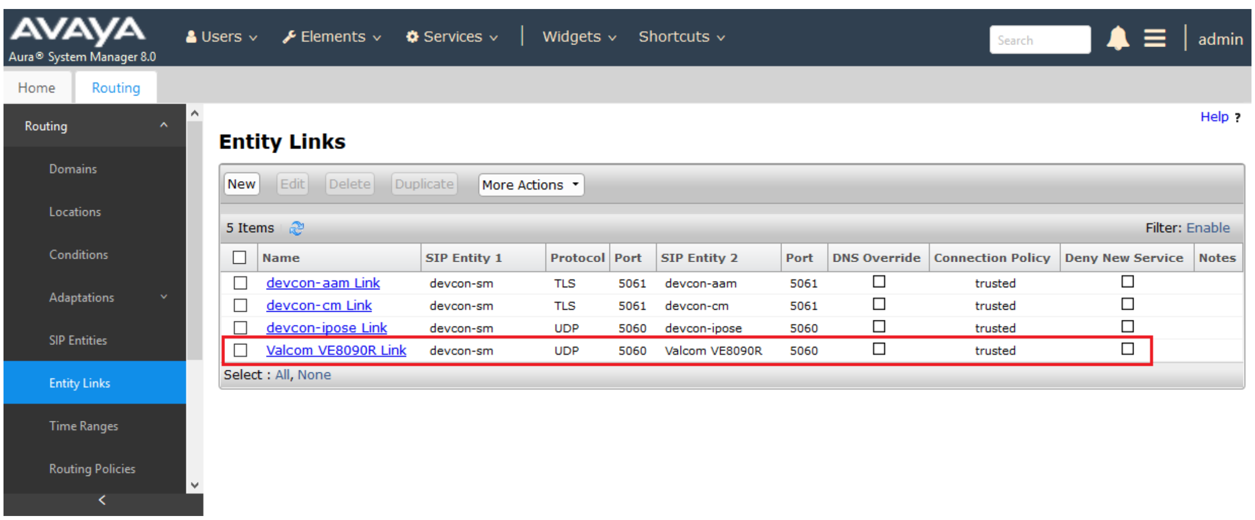


6.3. Administer Entity Link between Session Manager and VE8090R

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.



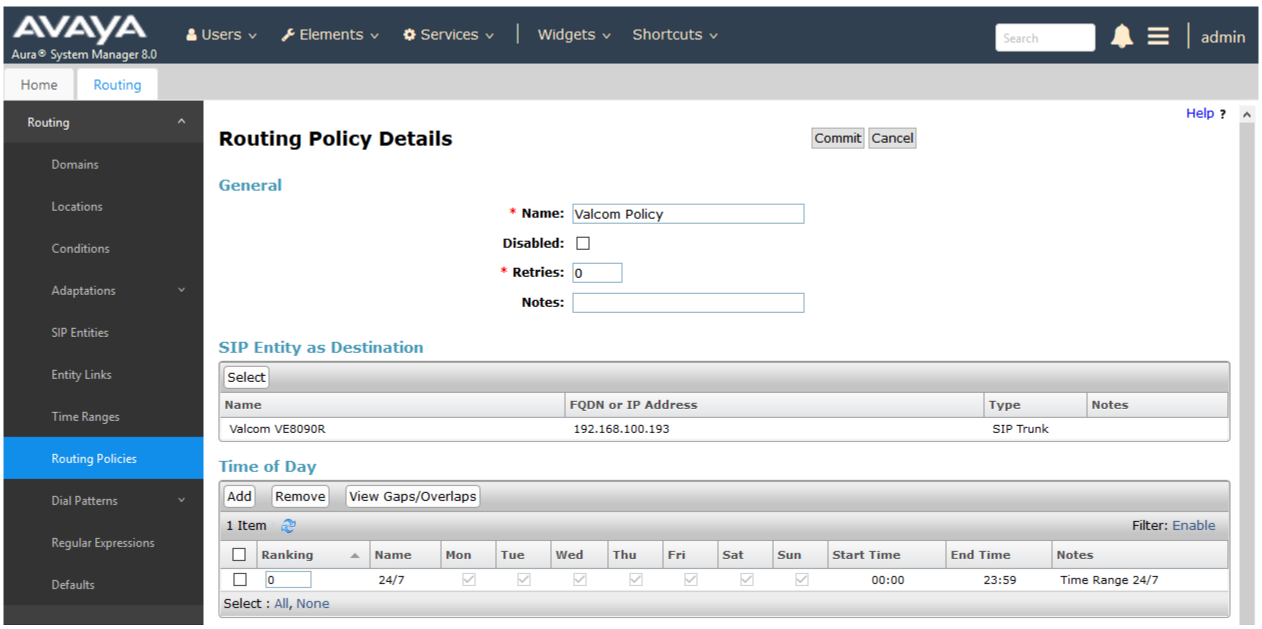
6.4. Add Routing Policy

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.



6.5. Add Dial Pattern

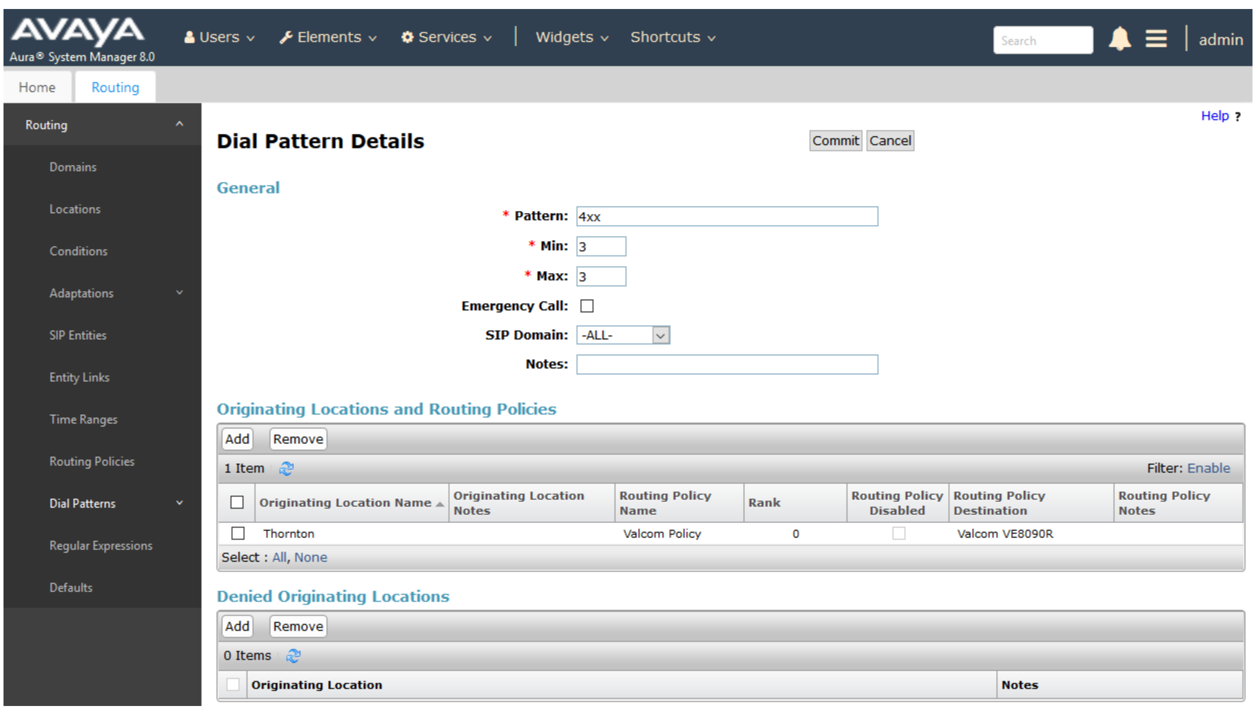
Dial patterns must be defined to direct calls to the appropriate SIP Entity. In the sample configuration, a 3-digit number beginning with '4' will be routed to VE8090R. To add a dial pattern, navigate to **Elements → Routing → Dial Patterns** and click on the **New** button (not shown). Fill in the following:

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.

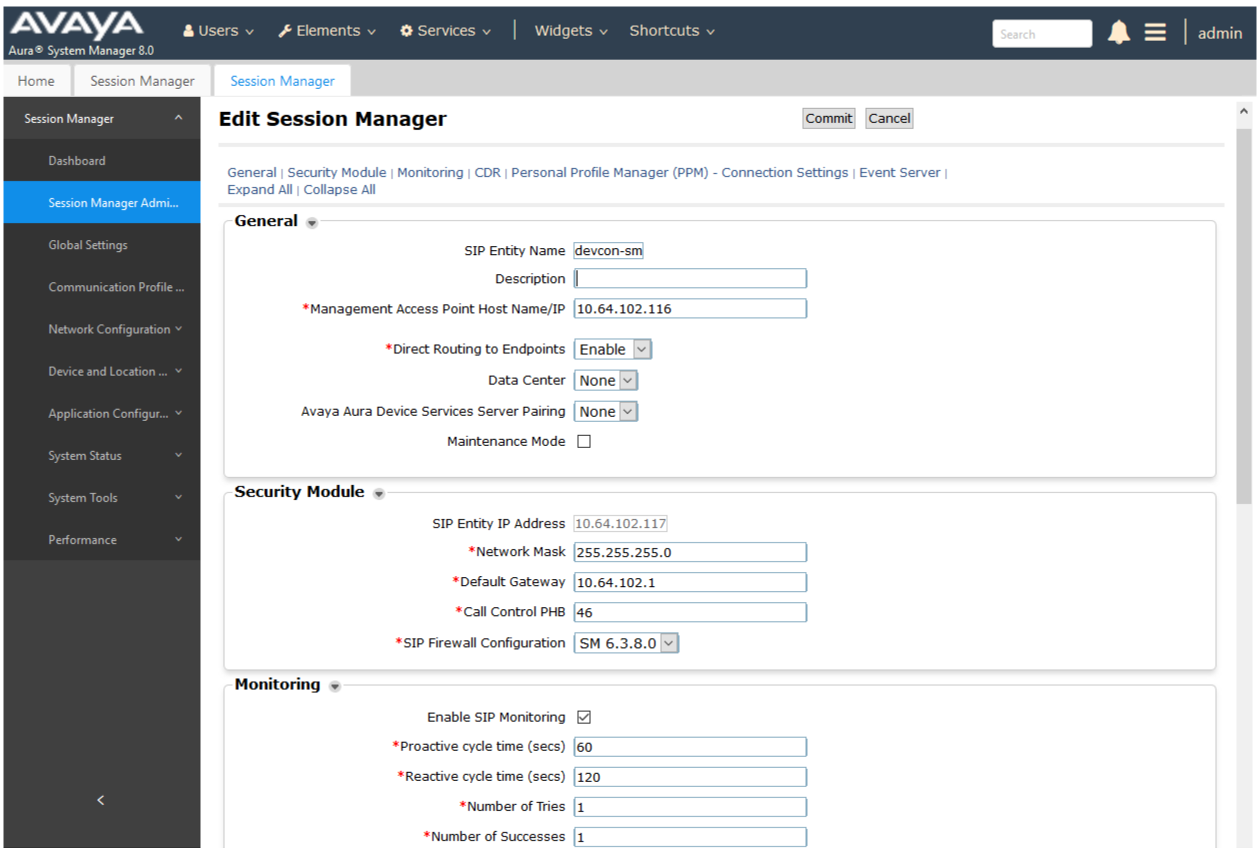
Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows the dial pattern definition for VE8090R.



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.



7. Configure Valcom VE8090R SIP Intercom Controller

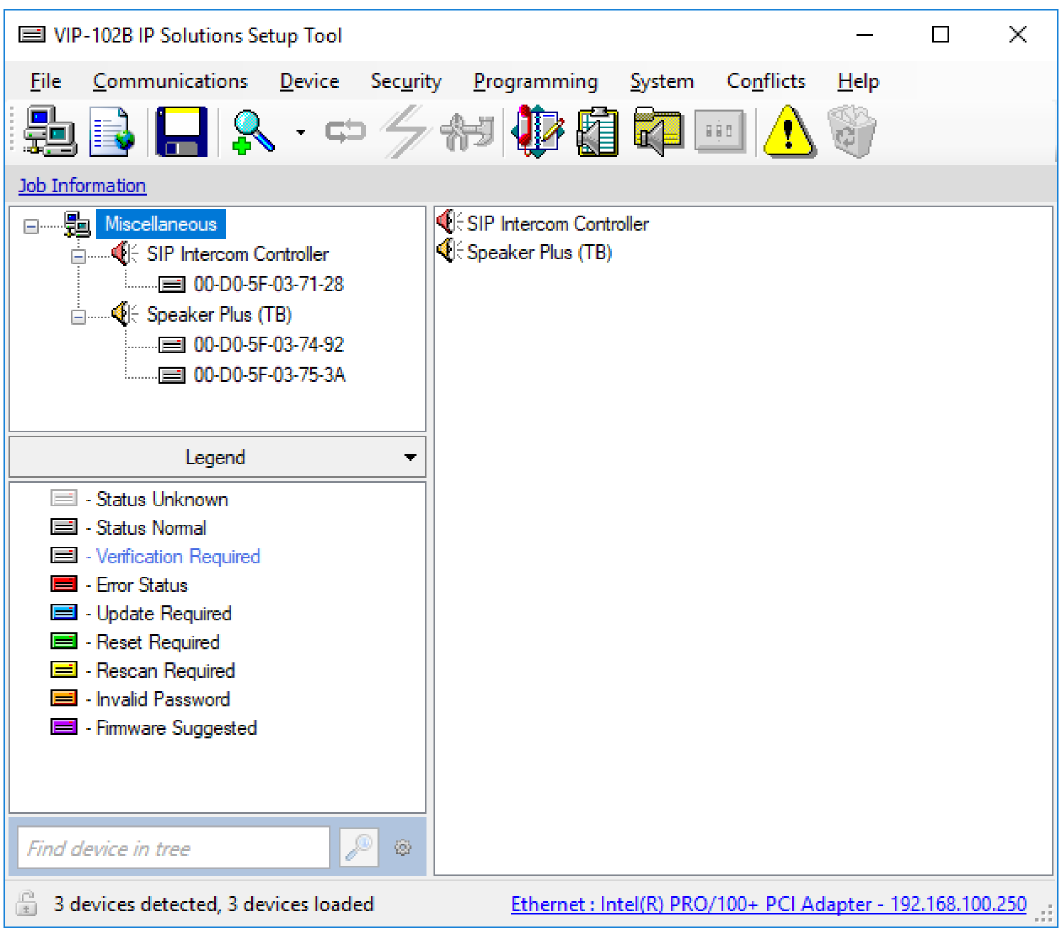
This section covers the configuration of VE8090R using the Valcom VIP-102B IP Solutions Setup Tool. The configuration covers the following areas:

- 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.

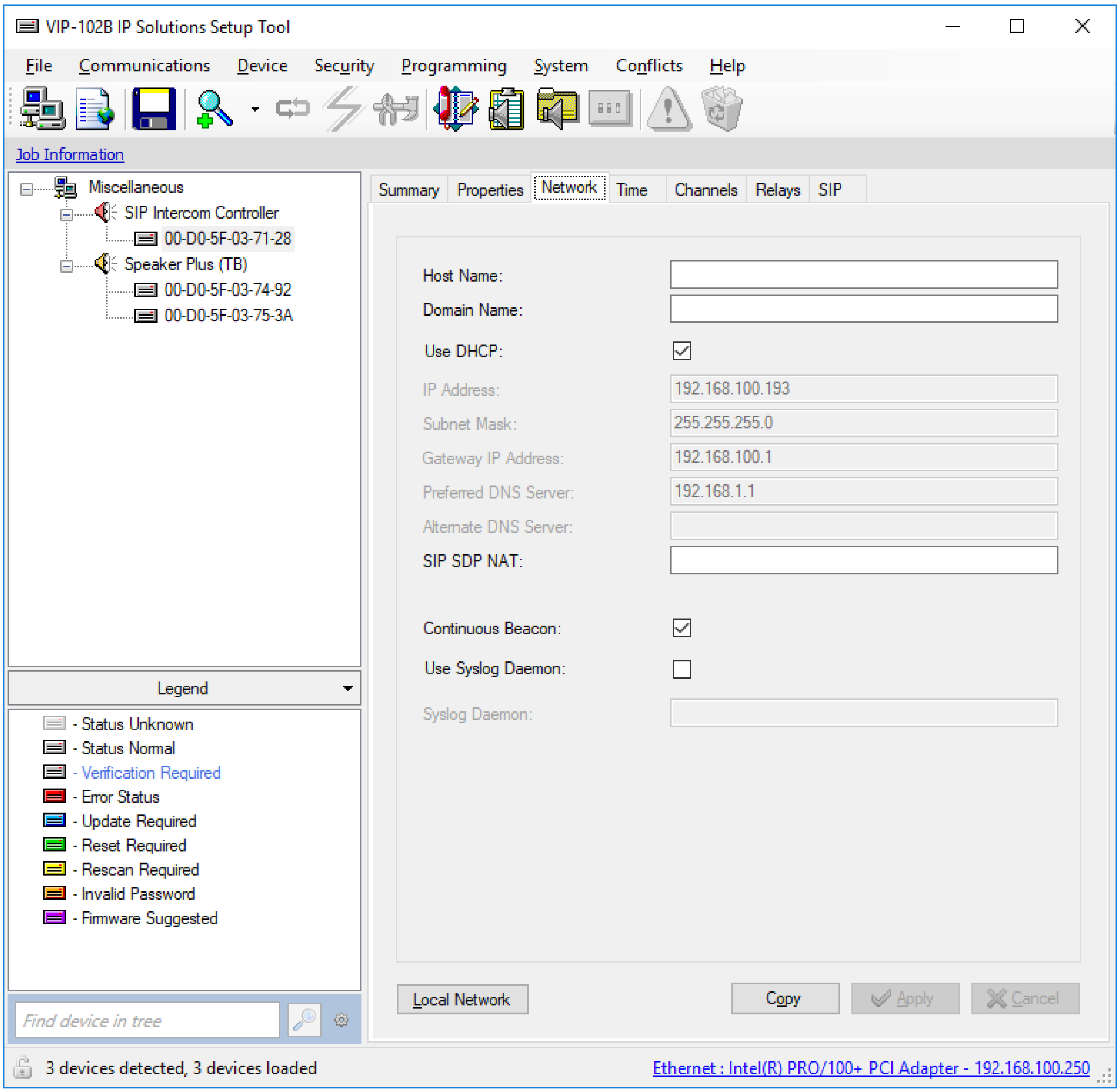
7.1. Launch Valcom VIP-102B IP Solutions Setup Tool

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



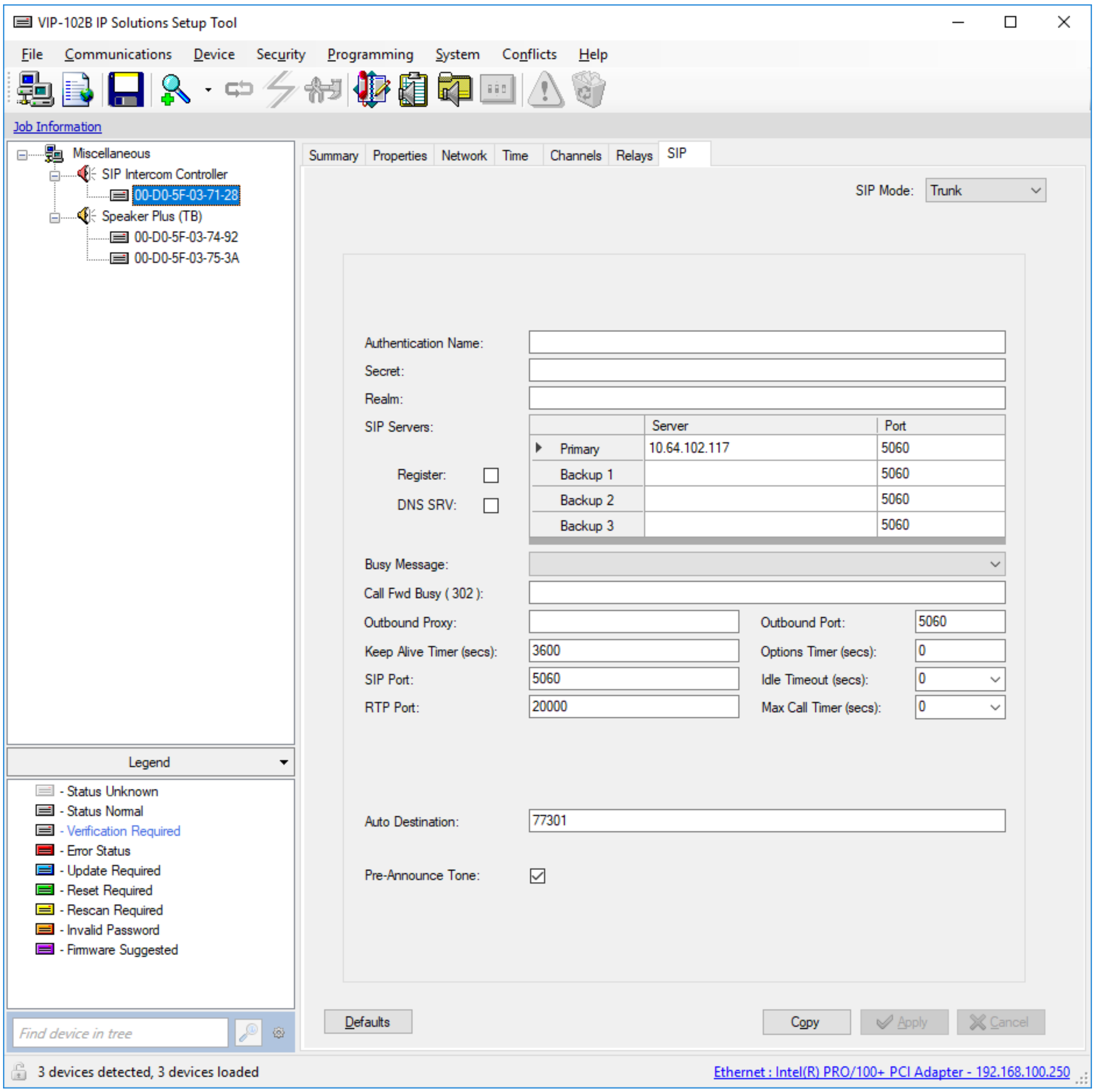
7.2. Configure the Network Settings of Valcom VE8090R SIP Intercom Controller

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.



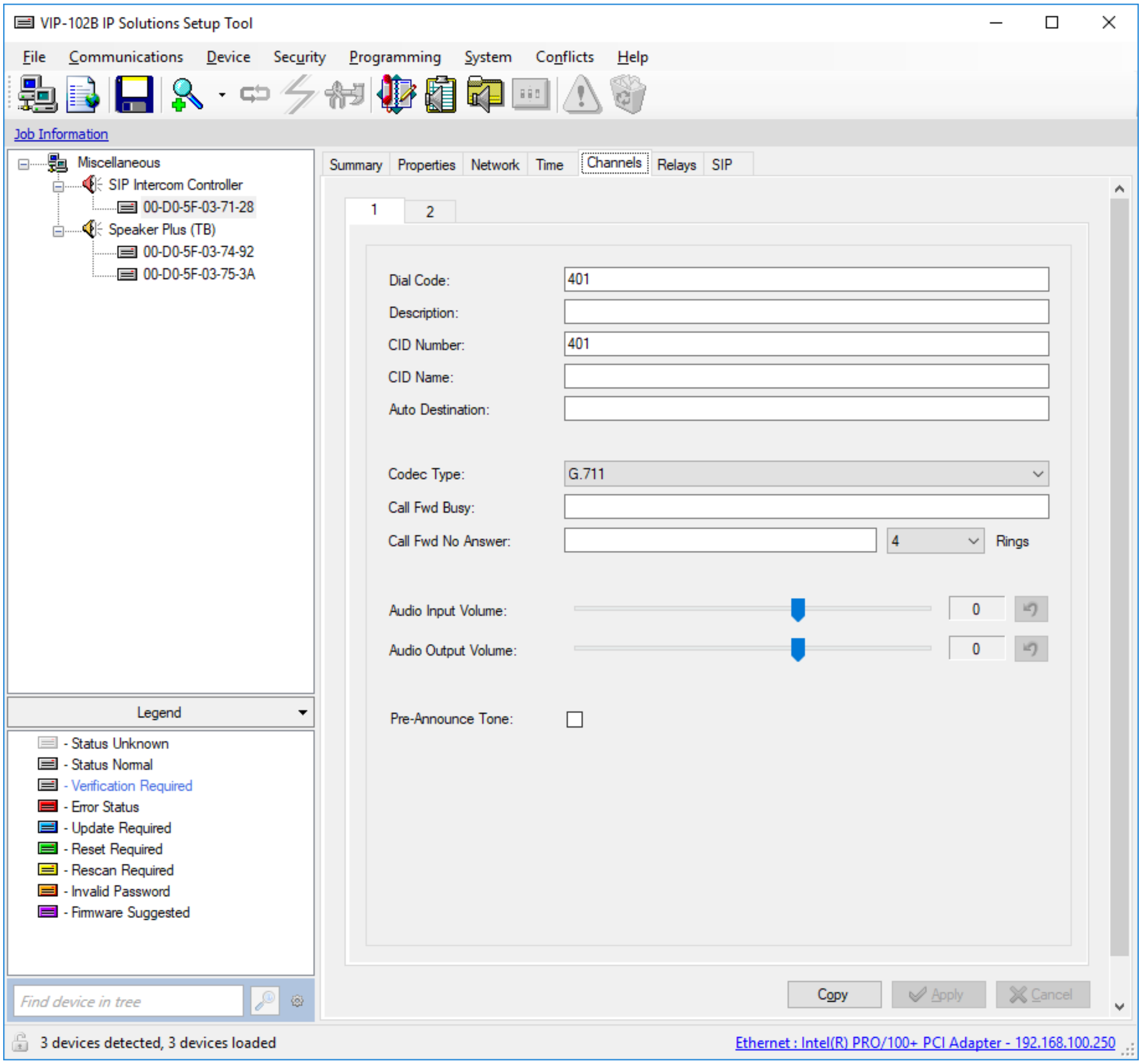
7.3. Configure SIP Parameters of Valcom VE8090R SIP Intercom Controller

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**.



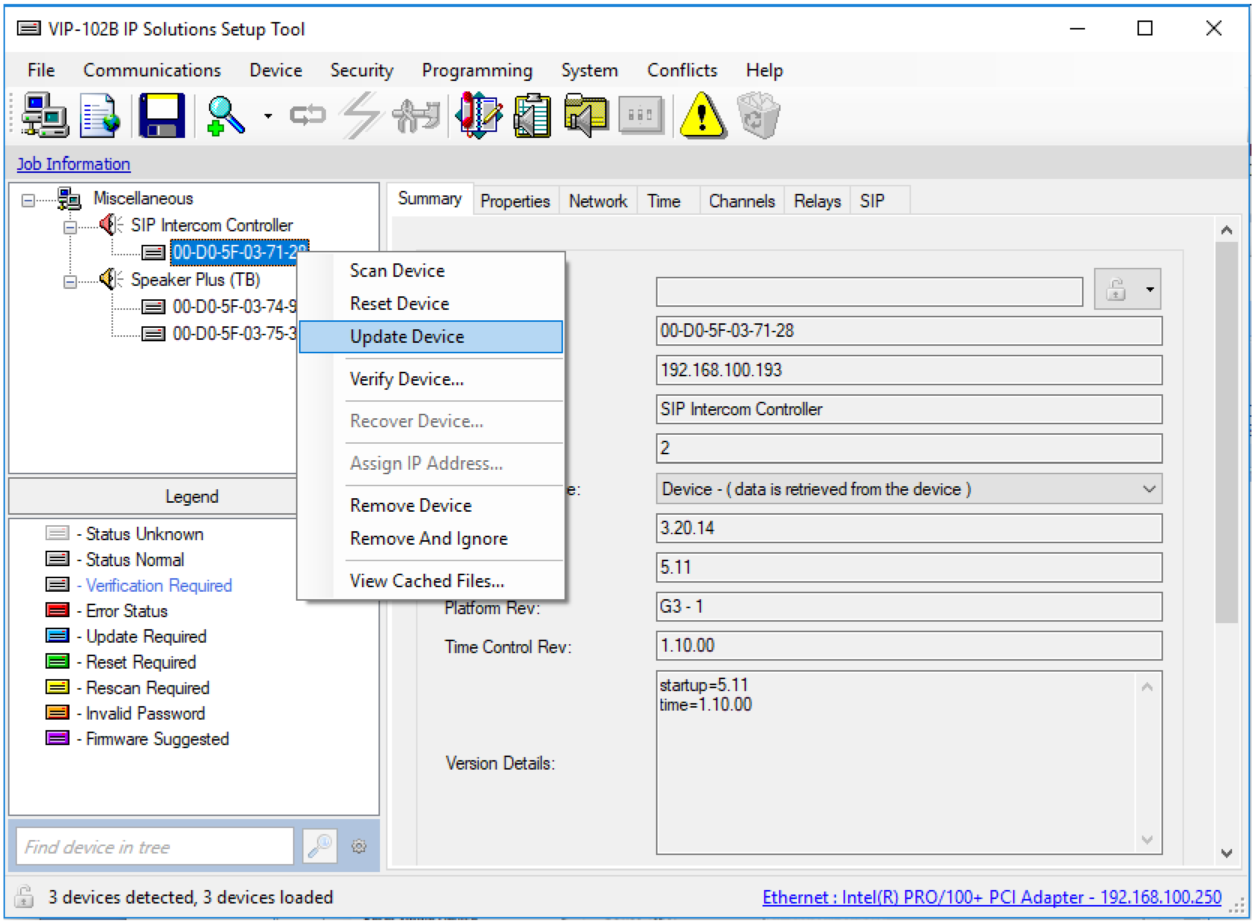
7.4. Verify Codec Settings

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

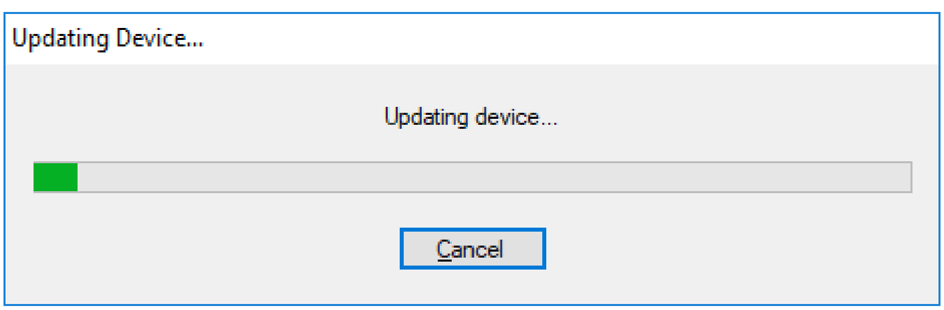


7.5. Update SIP Intercom Controller with the New Configuration

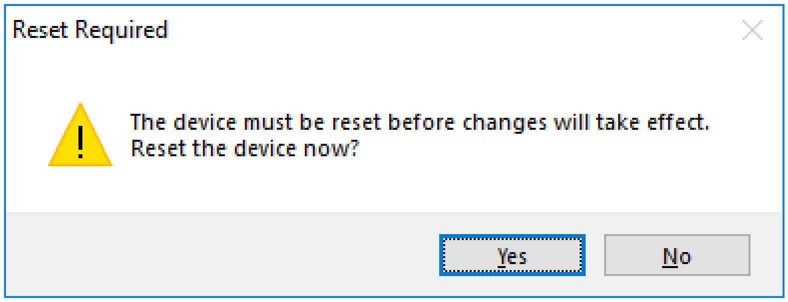
From the **VIP-102B IP Solutions Setup Tool**, right-mouse click on the MAC/hardware address of the SIP Intercom Controller and select Update Device from the pop-up menu as shown below.



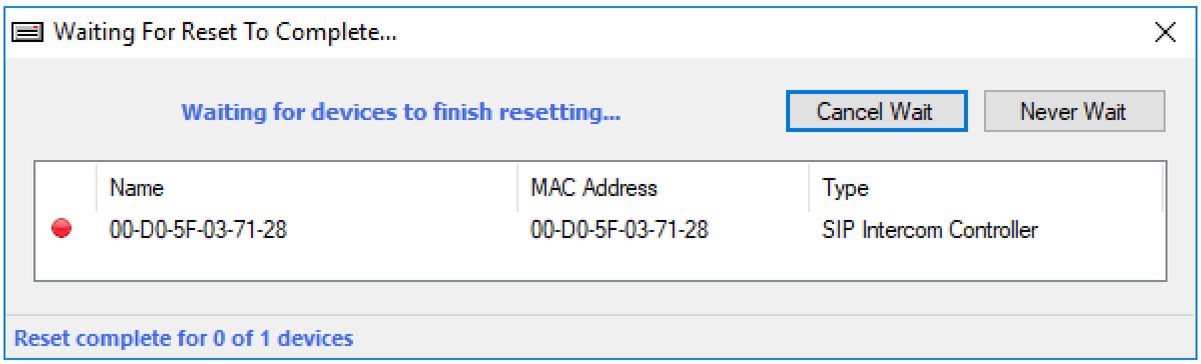
The following window is displayed indicating that the device is being updated.



A device reset is required so respond with **Yes** when prompted.



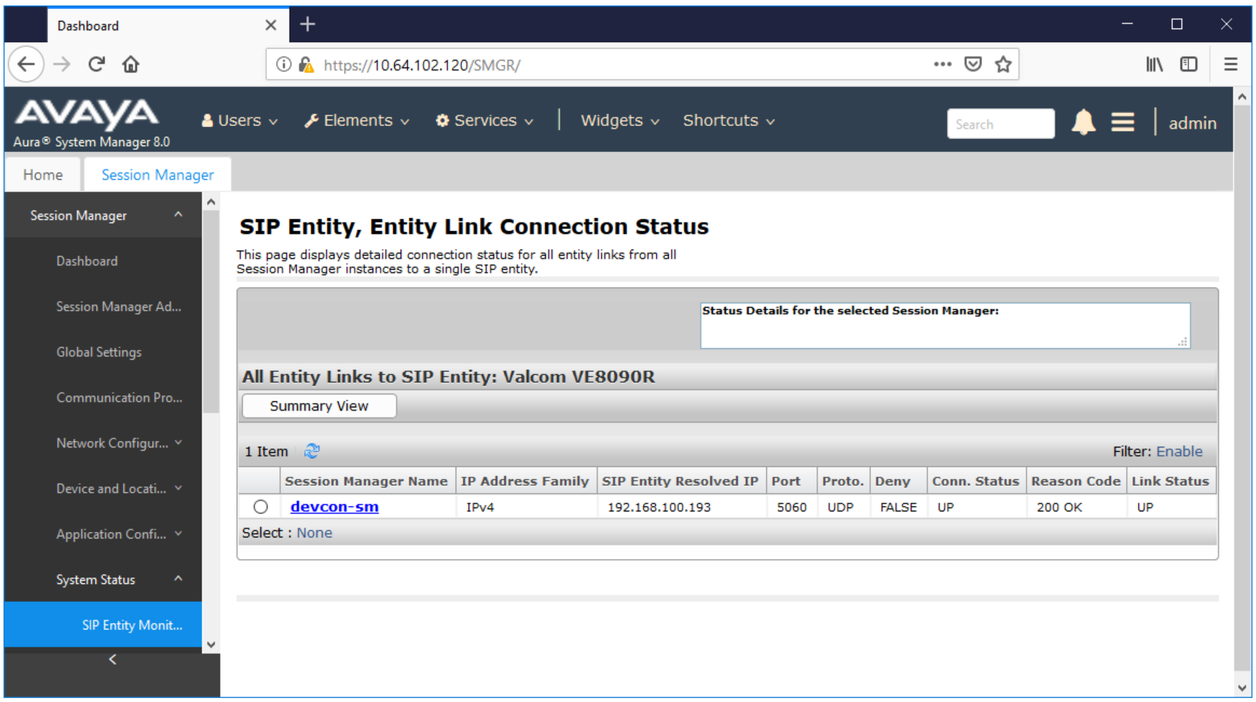
The following window will be displayed while the device is being reset. When the reset is complete, the window will disappear.



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*.



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.

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