**Avaya Solution & Interoperability Test Lab**

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

## Abstract


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

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

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

# 1. Introduction

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

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

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

Valcom offers IP Ceiling Speakers, IP Wall Speakers, and IP Horns as different products/models to accommodate different environments. They share the same SIP stack and firmware version, therefore, this testing also applies to those products, as detailed in **Attachment 1**. **Section 4** of this document shows the actual products/models and SIP Stack and software versions that were tested. For additional details, contact Valcom Support, as noted in **Section 2.3**.

# 2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between the Valcom One-Way and Talkback IP Speakers, Avaya SIP / H.323 IP Deskphones, and the PSTN. Two-way audio intercom calls with the Talkback IP Speaker and one-way audio paging calls with the One-Way IP Speaker were exercised. The serviceability testing focused on verifying that the Valcom IP speakers came back into service after a reboot.

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

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

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

For the testing associated with these Application Notes, the interface between Avaya systems and Valcom One-Way and Talkback IP Speakers did not include use of any specific encryption features as requested by Valcom.

## 2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

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

## 2.2. Test Results

All test cases passed with the following observations:

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

## 2.3. Support

For technical support and information on Valcom One-Way and Talkback IP Speakers, contact Valcom Technical Support at:

- Phone: +1 (800) 825-2661 or +1 (540) 563-2000
- Website: 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 with an Avaya G450 Media Gateway.
- Media resources in the Avaya G450 Media Gateway and Avaya Aura® Media Server.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP deskphones and Valcom One-Way and Talkback IP Speakers.
- Avaya Aura® Session Manager connected to Avaya Session Border Controller for Enterprise (SBCE) via a SIP trunk for access to the simulated PSTN.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya Session Border Controller for Enterprise
- Avaya 96x1 Series H.323 and SIP Deskphones.
- Avaya J100 Series SIP Deskphones.
- Valcom IP speakers, including the Valcom VIP-130AL-GY IP Secure One-Way Paging Horn and the Valcom VIP-160A IP Talkback 8” Ceiling Speaker configured with the ValcomVIP-102B IP Solutions Setup Tool.

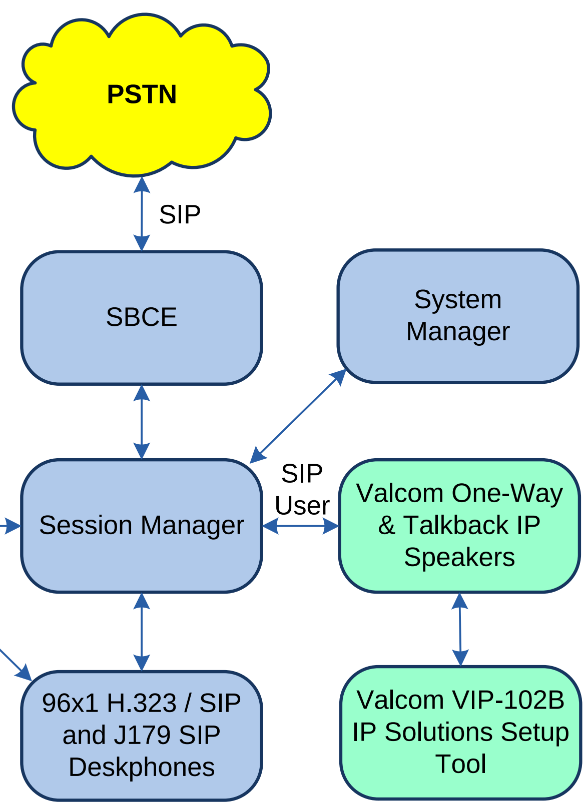



**Figure 1: Avaya SIP Network with Valcom One-Way and Talkback IP Speakers**

# 4. Equipment and Software Validated

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

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

# 5. Configure Avaya Aura® Communication Manager

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

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

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

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

## 5.1. Verify Communication Manager License

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

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

```
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    91
                                      Maximum Stations: 36000    30
                              Maximum XMOBILE Stations: 36000     0
                    Maximum Off-PBX Telephones - EC500: 41000     0
                    Maximum Off-PBX Telephones -   OPS: 41000    16
                    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 IP speaker. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set '1' was specified in IP Network Region '1' shown above. The default settings of the **IP Codec Set** form are shown below. Valcom IP speakers were tested using G.711 and G.722 codecs.

```
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:
```

## 5.4. Administer SIP Trunk to Session Manager

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

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

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

```
add signaling-group 10                                          Page   1 of   2
                                SIGNALING GROUP

 Group Number: 10             Group Type: sip
  IMS Enabled? n       Transport Method: tls
        Q-SIP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: SM                      Clustered? n
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                 Far-end Node Name: devcon-sm
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1

Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
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? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

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

```
add trunk-group 10                                              Page   1 of  22
                                TRUNK GROUP

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
```

## 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                                                                 n   user
 2:                                                                        n   user
 3:                                                                        n   user
 4:                                                                        n   user
 5:                                                                        n   user
 6:                                                                        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

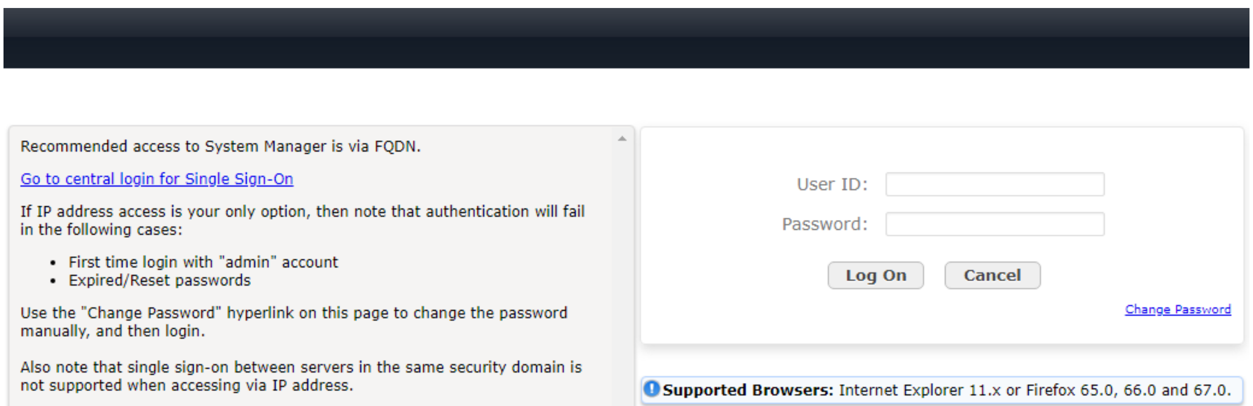
This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- 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 One-Way and Talkback IP Speakers.

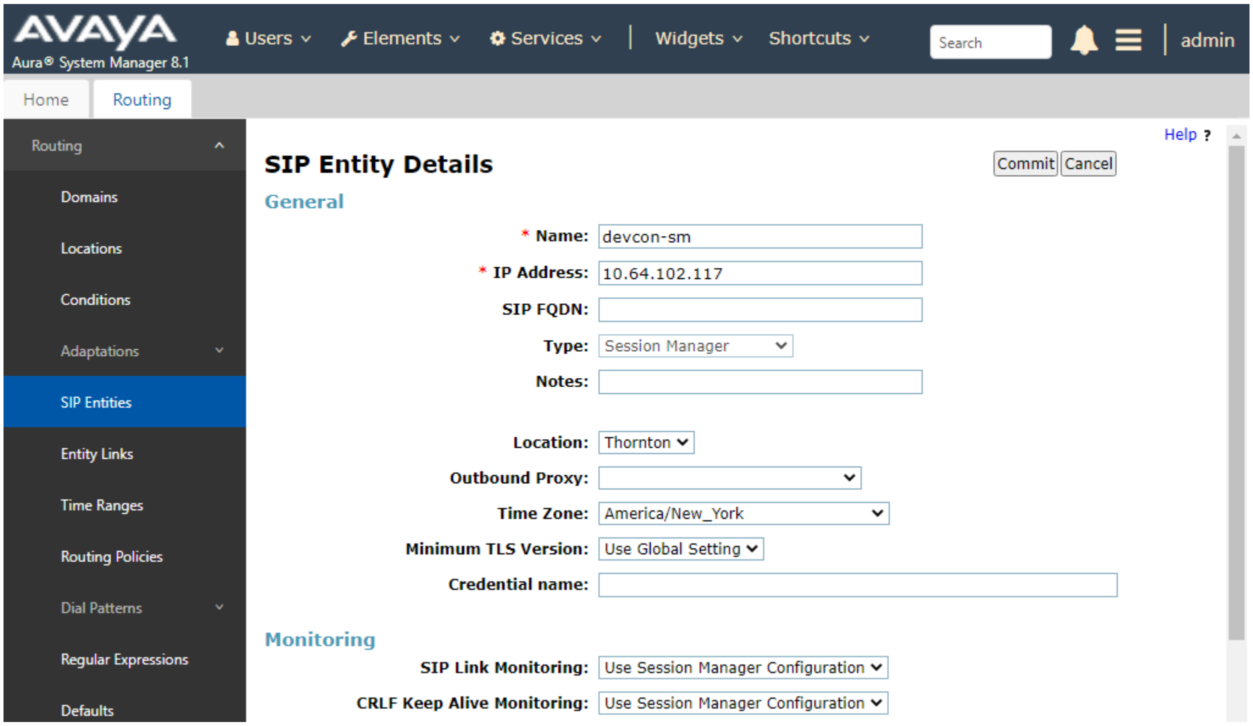
## 6.1. Launch System Manager

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

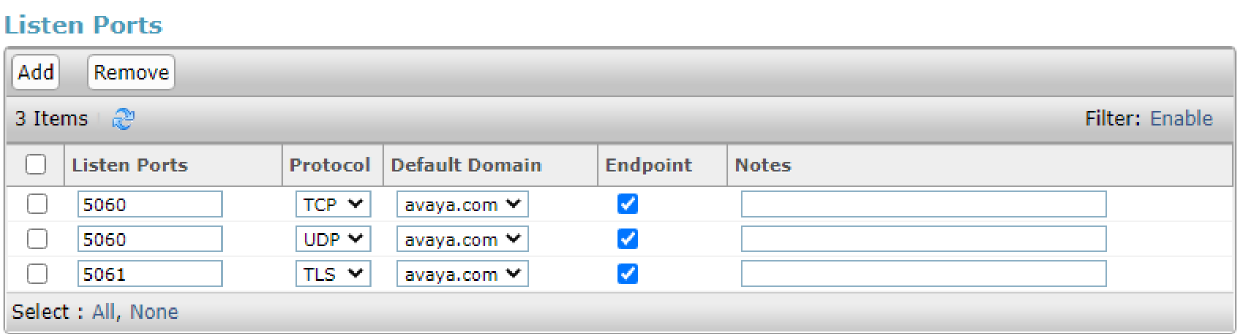


## 6.2. Set Network Transport Protocol

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

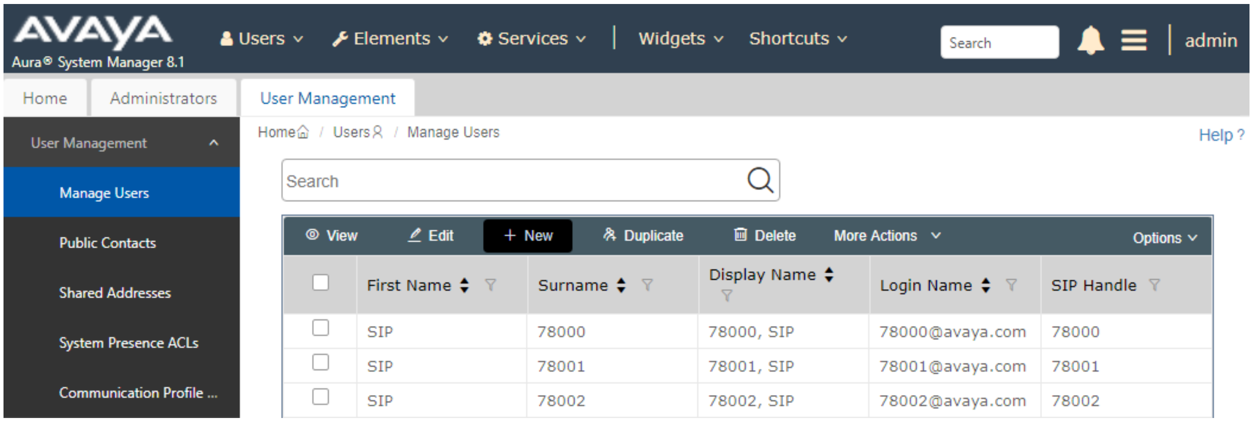



Scroll down to the **Listen Ports** section and verify that the transport network protocol used by the IP speaker is specified in the list below. For the compliance test, the solution was verified with UDP and TCP network transport.



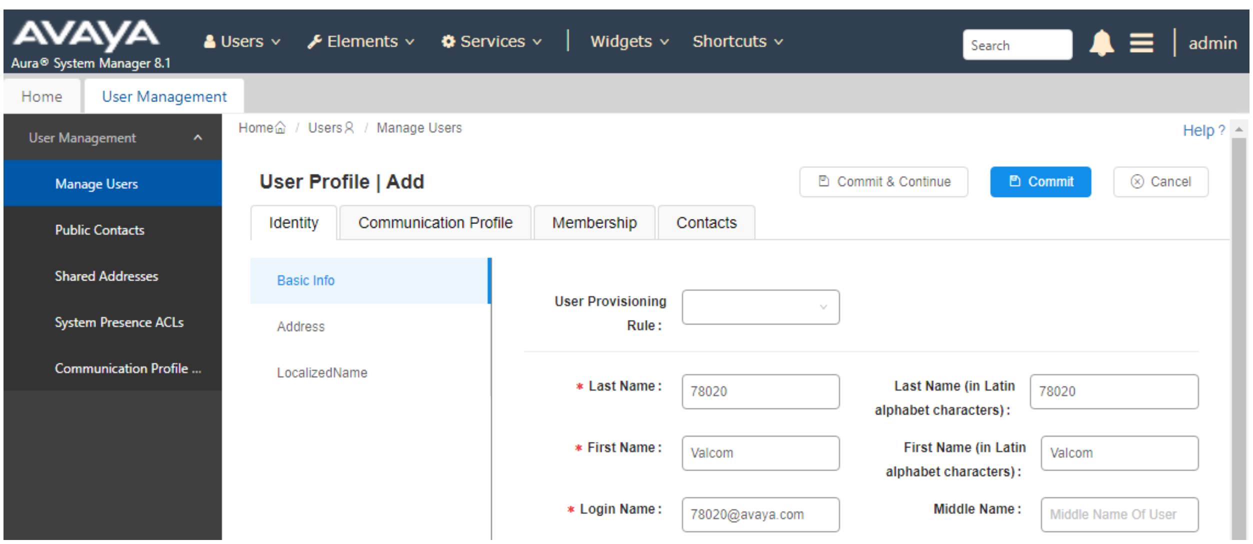
## 6.3. Administer SIP User

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.



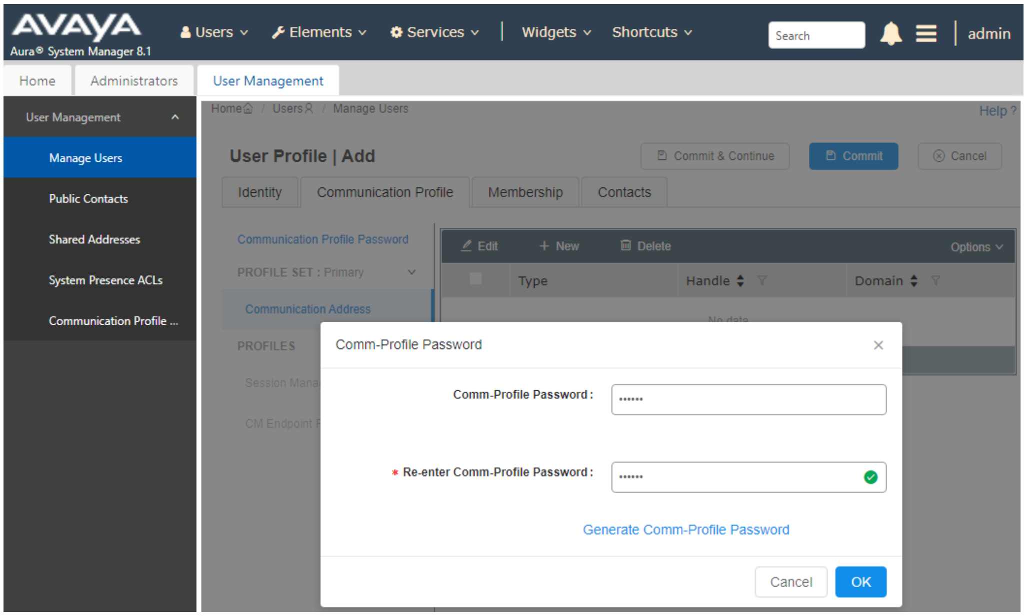

### 6.3.1. Identity

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 IP speaker extension and "*<domain>*" is the applicable SIP domain name from **Section 5.3**. Retain the default values in the remaining fields.



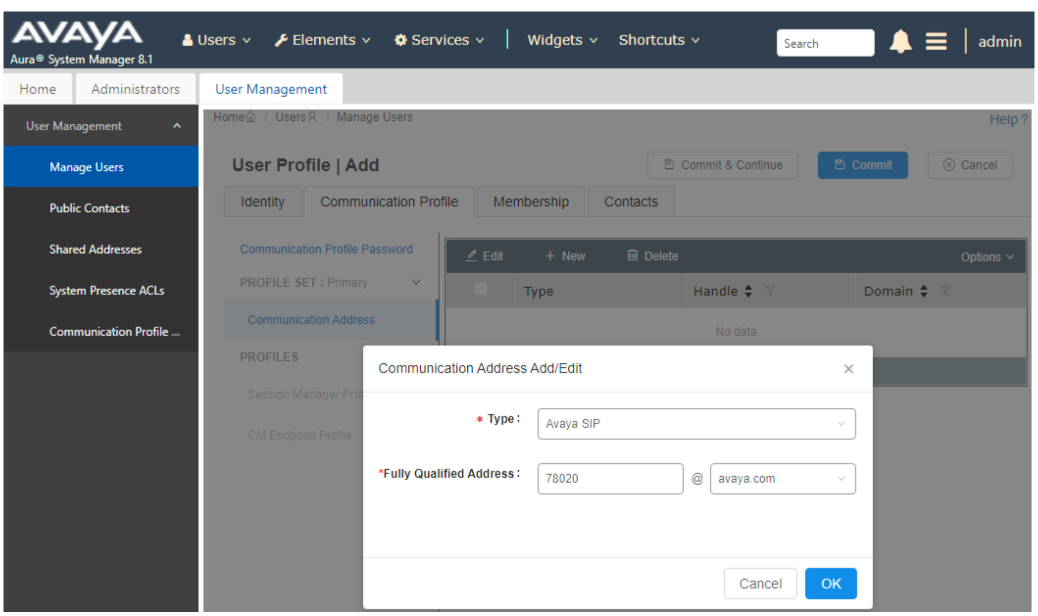
## 6.3.2. Communication Profile

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**.



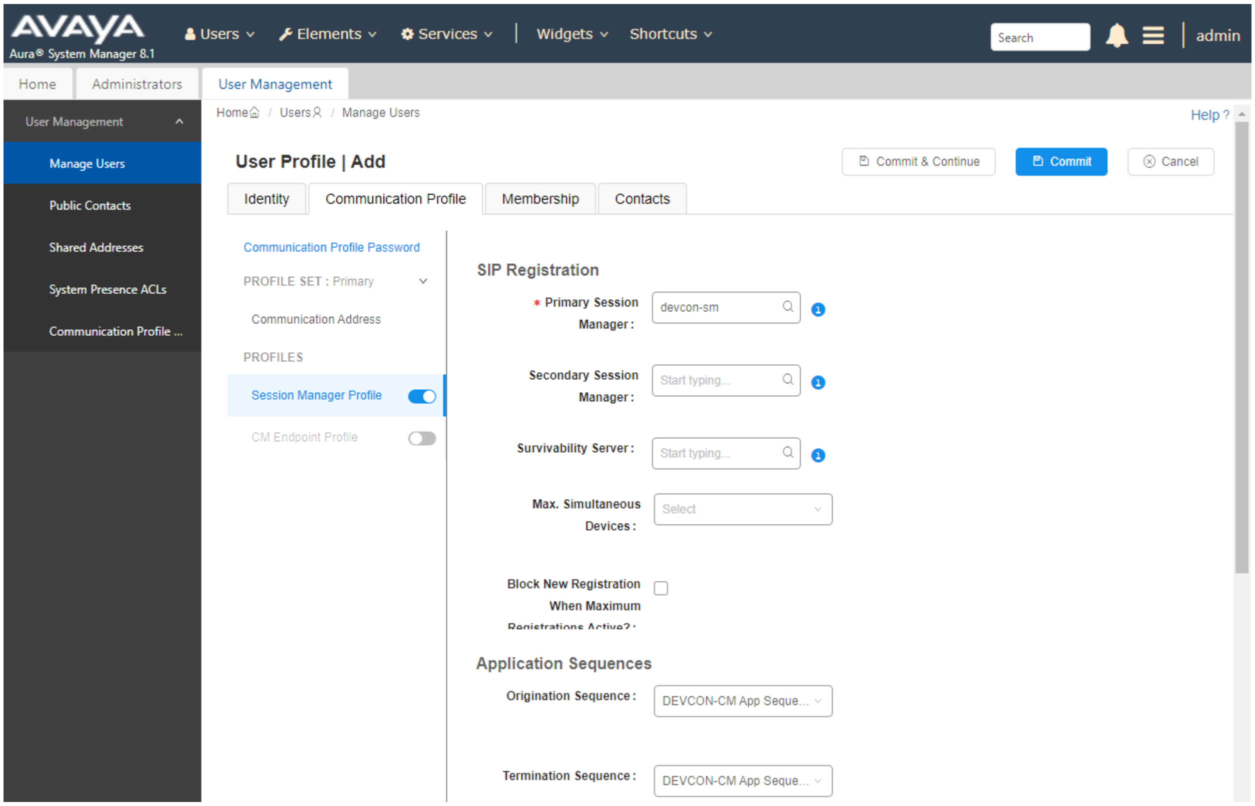

## 6.3.3. Communication Address

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**.

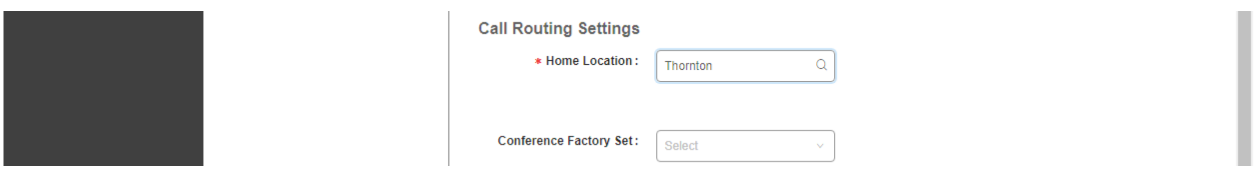


### 6.3.4. Session Manager Profile

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.

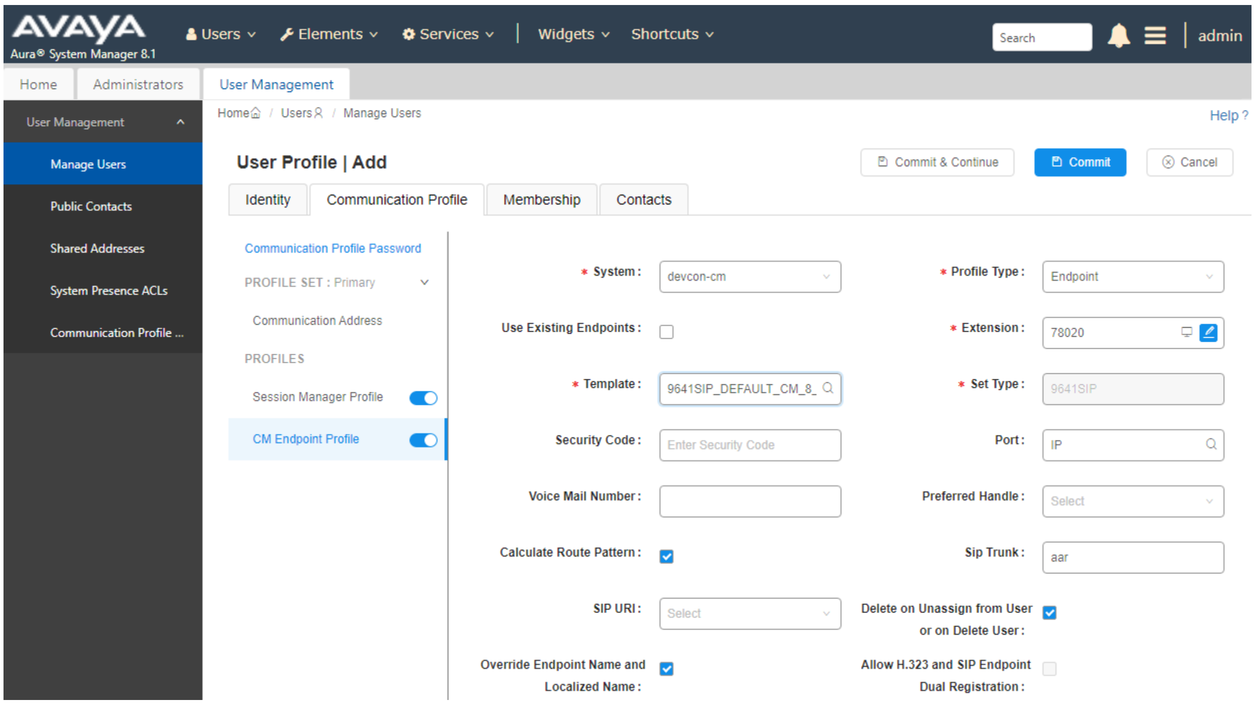



Scroll down to the **Call Routing Settings** section to configure the **Home Location**.



### 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 *9641SIP_DEFAULT_CM_8_1*. For **Port**, click and select *IP*. Retain the default values in the remaining fields.



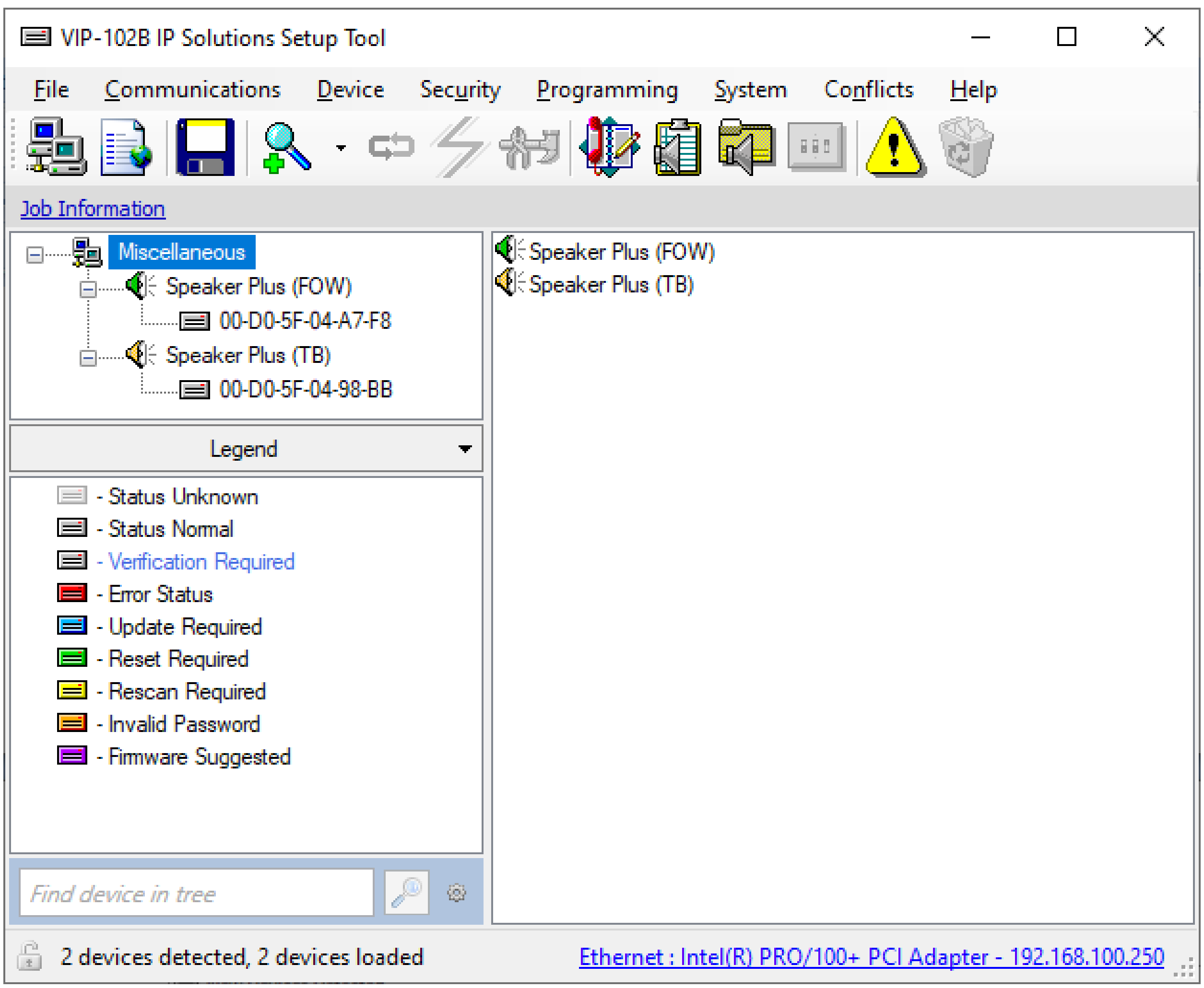
# 7. Configure Valcom One-Way and Talkback IP Speakers

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

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

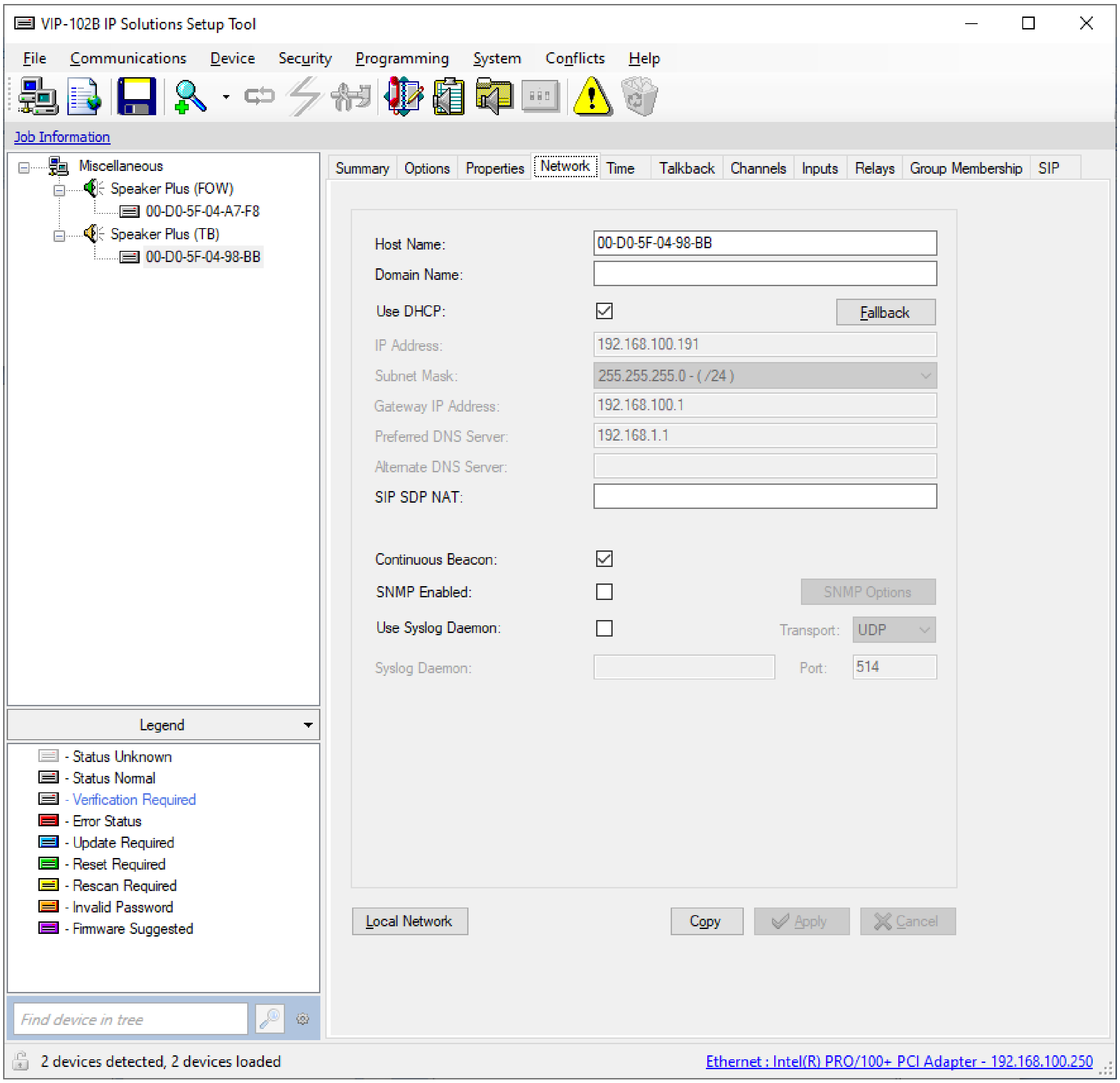
## 7.1. Launch Valcom VIP-102B IP Solutions Setup Tool

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



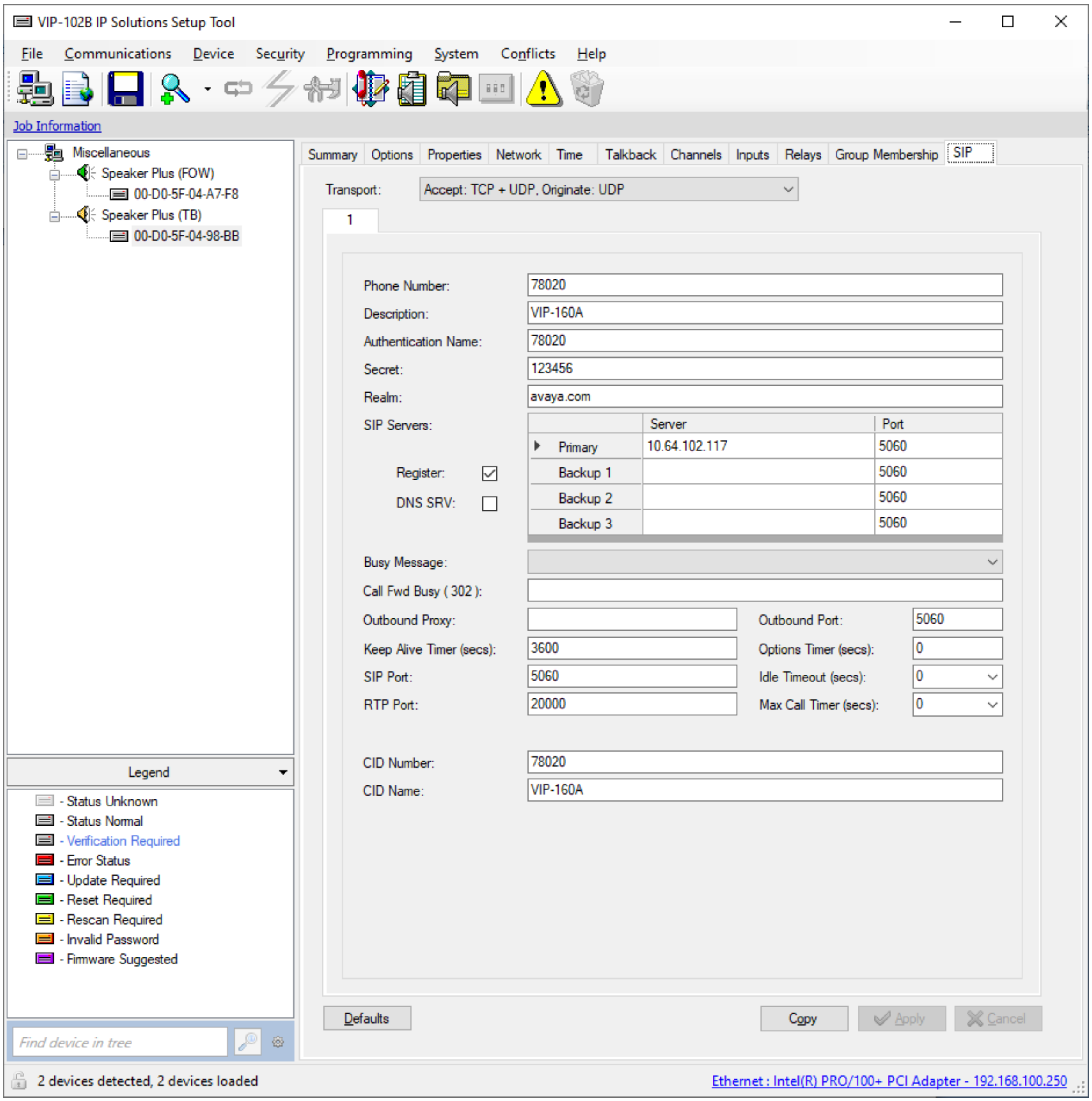
## 7.2. Configure the Network Settings of Valcom One-Way and Talkback IP Speakers

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



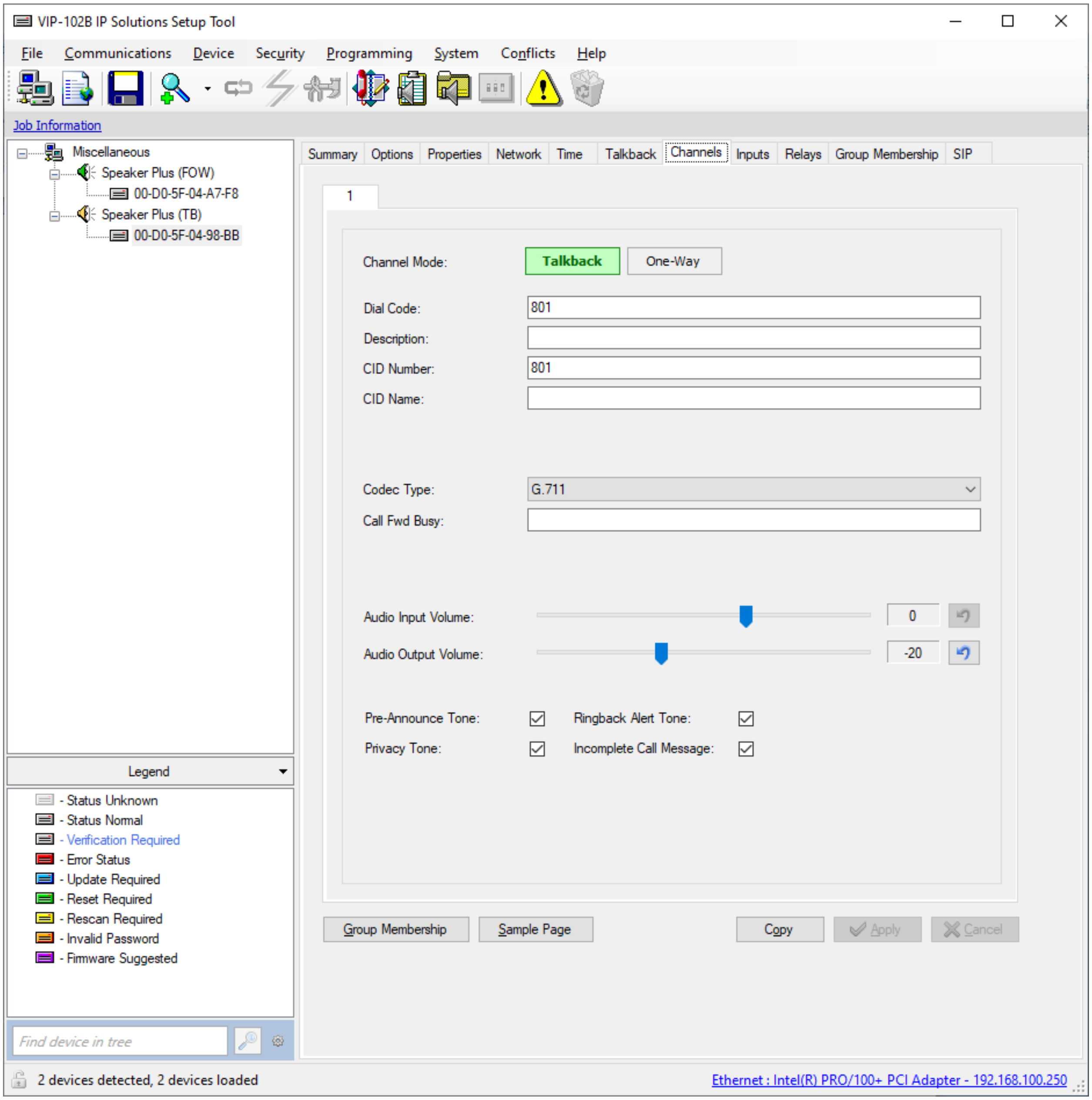
## 7.3. Configure SIP Parameters of Valcom One-Way and Talkback IP Speakers

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



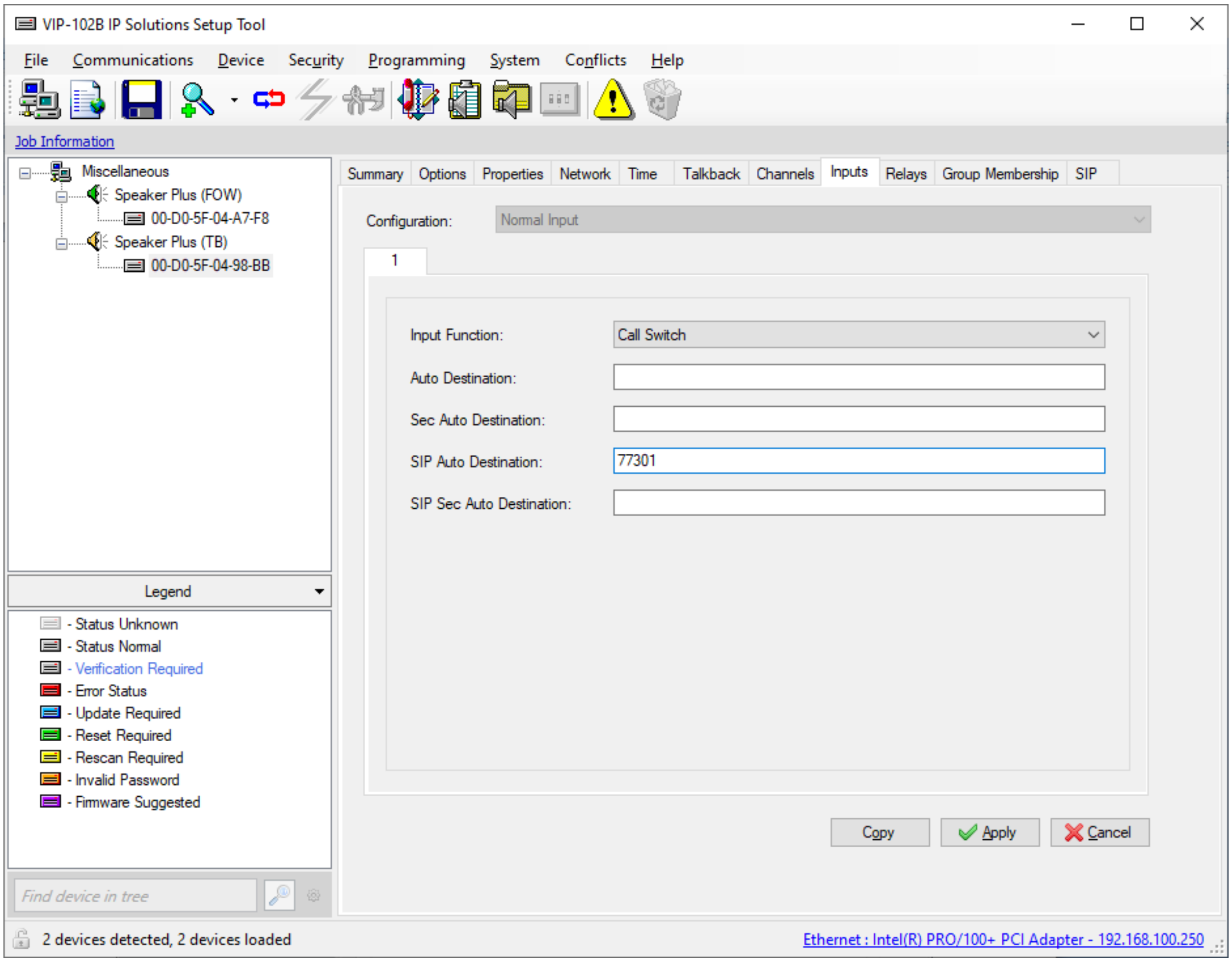
## 7.4. Verify Codec Settings

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



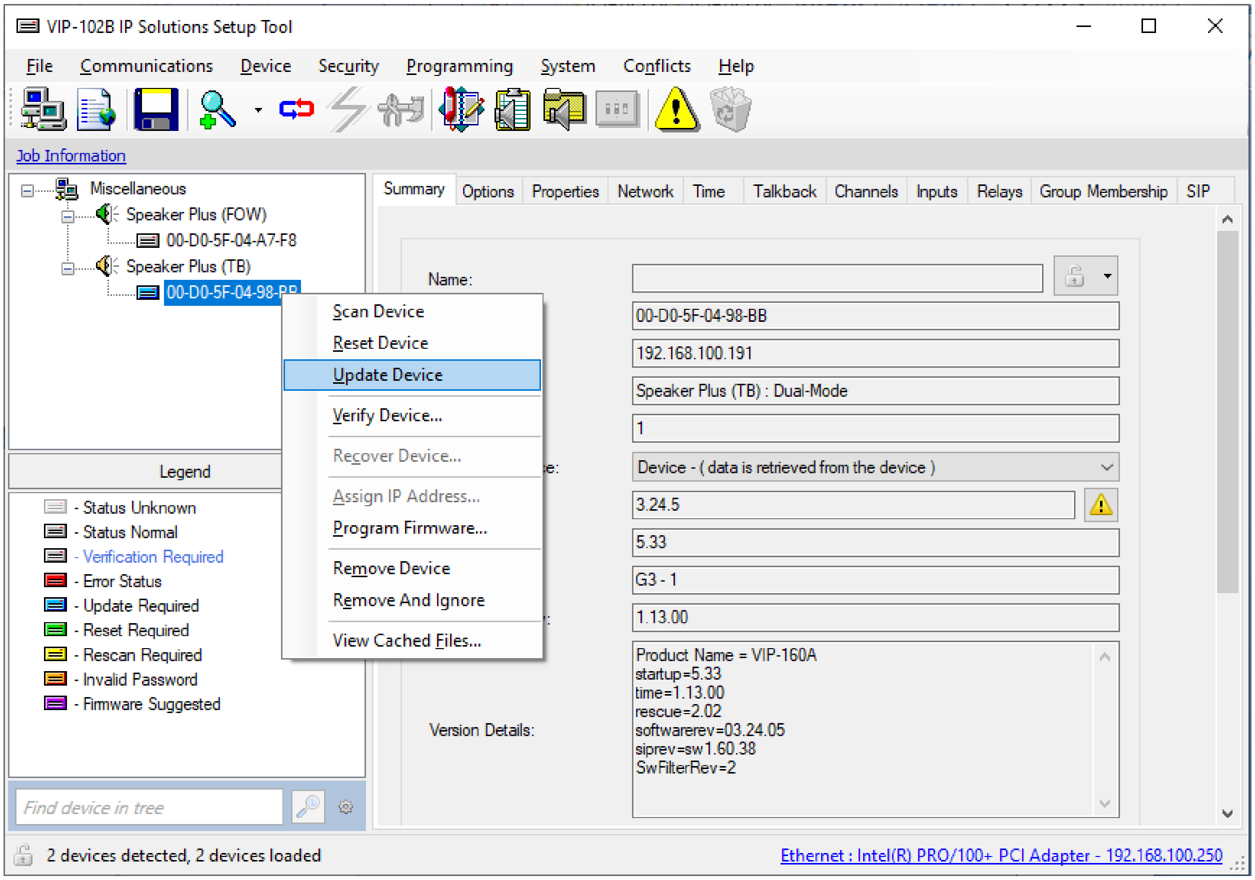
## 7.5. Specify Call Destination

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

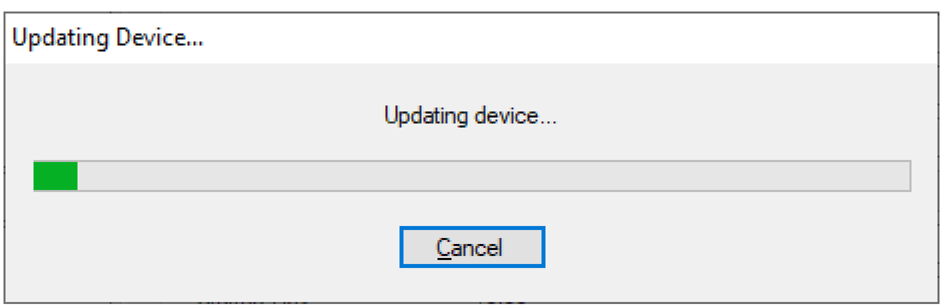


## 7.6. Update SIP Intercom Controller with the New Configuration

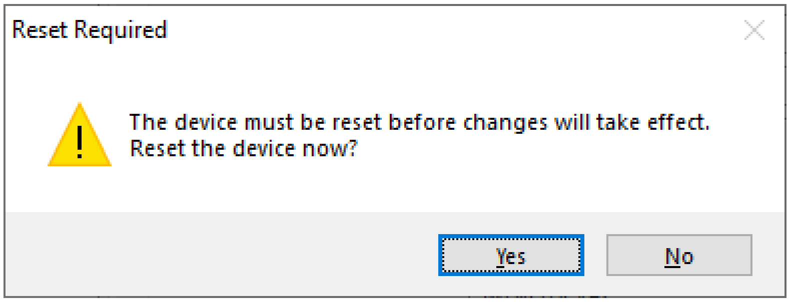
From the **VIP-102B IP Solutions Setup Tool**, right-mouse click on the MAC/hardware address of the SIP Intercom Controller and select **Update Device** from the pop-up menu as shown below.



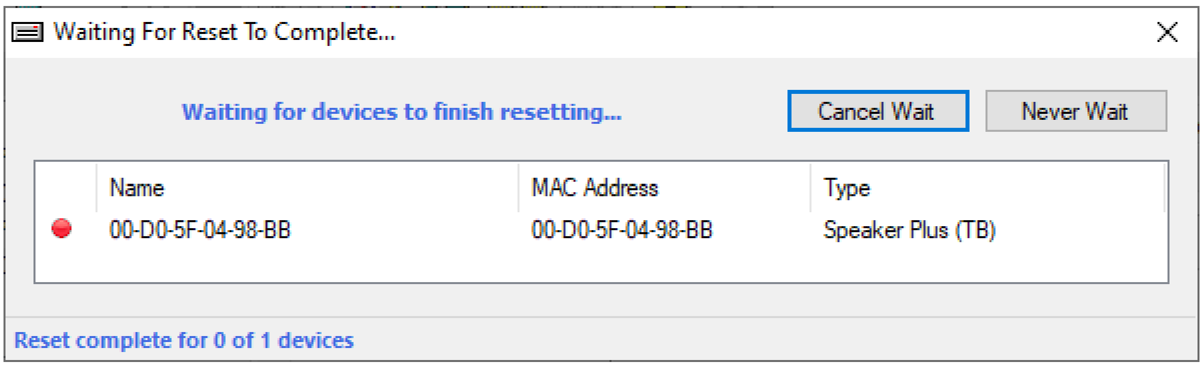

The following window is displayed indicating that the device is being updated.



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



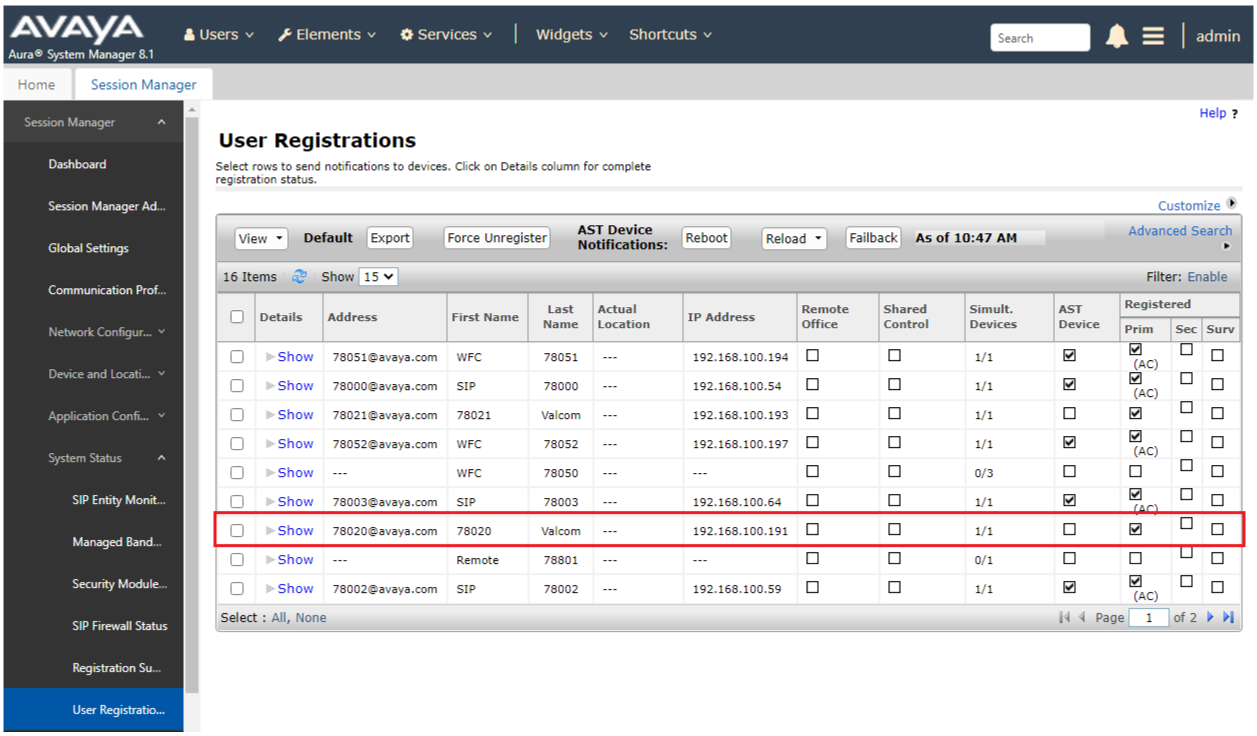

The following window will be displayed while the device is being reset. When the reset is complete, the window will disappear.



## 8. Verification Steps

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

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




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

3. Place an intercom call by pressing the call button on a Talkback IP speaker. Verify two-way audio to the call destination. Terminate the call from the IP speaker by pressing the call button.

# 9. Conclusion

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

# 10. References

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


[1] *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 8, November 2020, available at http://support.avaya.com.

[2] *Administering Avaya Aura® System Manager for Release 8.1.x*, Release 8.1.x, Issue 8, November 2020, available at http://support.avaya.com.

[3] *Administering Avaya Aura® Session Manager*, Release 8.1.x, Issue 7, October 2020, available at http://support.avaya.com.

[4] *Valcom VIP-102B IP Solutions Setup Tool Version 8.1.0.0 Reference Manual*, Revision 12 – 7/10/20, 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.

VALCOM®

## Declaration of Conformance

**December 1, 2020**

Jeff Gartner
Senior Manager
DevConnect Program
Avaya

**Dear Jeff Gartner:**

We, Valcom Inc, declare under sole responsibility that product series named IP Ceiling Speakers, IP Wall Speakers and IP Horns share the same hardware circuitry, software, SIP stack and firmware version. Therefore, the products are expected to behave in the same manner. The differences between the different models in each series are generally cosmetic in nature, such as enclosure shape or color, mounting arrangement, etc.

Sincerely,

**/s/ David Ellison**

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