Avaya Solution & Interoperability Test Lab

Application Notes for Algo 8188 SIP Ceiling Speaker Version 3.3.3 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager Release 8.1 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Algo 8188 SIP Ceiling Speaker to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Algo 8188 SIP Ceiling Speaker is a SIP-based device that can register with Avaya Aura® Session Manager as two separate SIP endpoints, one for loud ringing and one for voice paging.

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 for Algo 8188 SIP Ceiling Speaker to interoperate with Avaya Aura® Session Manager 8.1. Algo 8188 SIP Ceiling Speaker is a SIP-based device that can register with Avaya Aura® Session Manager as two separate SIP endpoints, one for loud ringing and one for voice paging.

The 8188 supports two SIP extensions which behave differently – RING and PAGE. One or both may be used depending on the application. If the RING extension is called the 8188 will not answer. Instead, it will flash a light pattern until the inbound call stops ringing. Typically the RING extension is programmed as part of a hunt group so that it receives a ring signal simultaneously with one or more devices. The simultaneous ringing at the desk phone and the 8188 Ceiling Speaker is accomplished via the bridge feature.

If the PAGE extension is called, the 8188 will auto answer and allow paging over its internal speaker. The 8188 can represent 8188, 8189, 8198, 8186, 8196 since it shares the same SIP stack.

2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were manually placed to the loud ringing and voice paging extensions, with call controls such as hold/resume, unattended, attended transfer and conference performed from the caller.

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 the Algo 8188 did not include use of any specific encryption features, as requested by Algo.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. The loud ringing feature testing included registration, internal and external caller, interactions with the voice paging extension, and interactions with desk phone features such as coverage, call forwarding, and do not disturb. The voice paging feature testing included registration, media shuffling, internal and external caller, interactions with the loud ringing extension, and interactions with caller actions such as drop, hold/reconnect, blind/attended transfer, and blind/attended conference.

The serviceability testing focused on verifying the ability of Algo 8188 SIP Ceiling Speaker to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet cable to the device.

2.2. Test Results

The objectives outlined in **Section 2.1** were verified. All test cases passed.

2.3. Support

Technical support on Algo 8188 SIP Ceiling Speaker can be obtained through the following:

- Phone: + 1 604 454 3792
- Web: http://www.algosolutions.com/support
- Email: support@algosolutions.com

3. Reference Configuration

Figure 1 illustrates the test configuration used during the compliance testing between the Avaya Aura® Communication Manager and Avaya Aura® Session Manager and Algo 8188 SIP Ceiling Speaker. The Algo 8188 communicated with Avaya systems through an Avaya switch with Power over Ethernet (PoE) and registered with Session Manager as two separate SIP endpoints, and the extensions used for the testing: one for Voice Paging and one for Loud Ringer. The SIP trunk was also configured to connect from Avaya Session Border Controller for Enterprise to Service Provider for test cases off-net via SIP trunk.

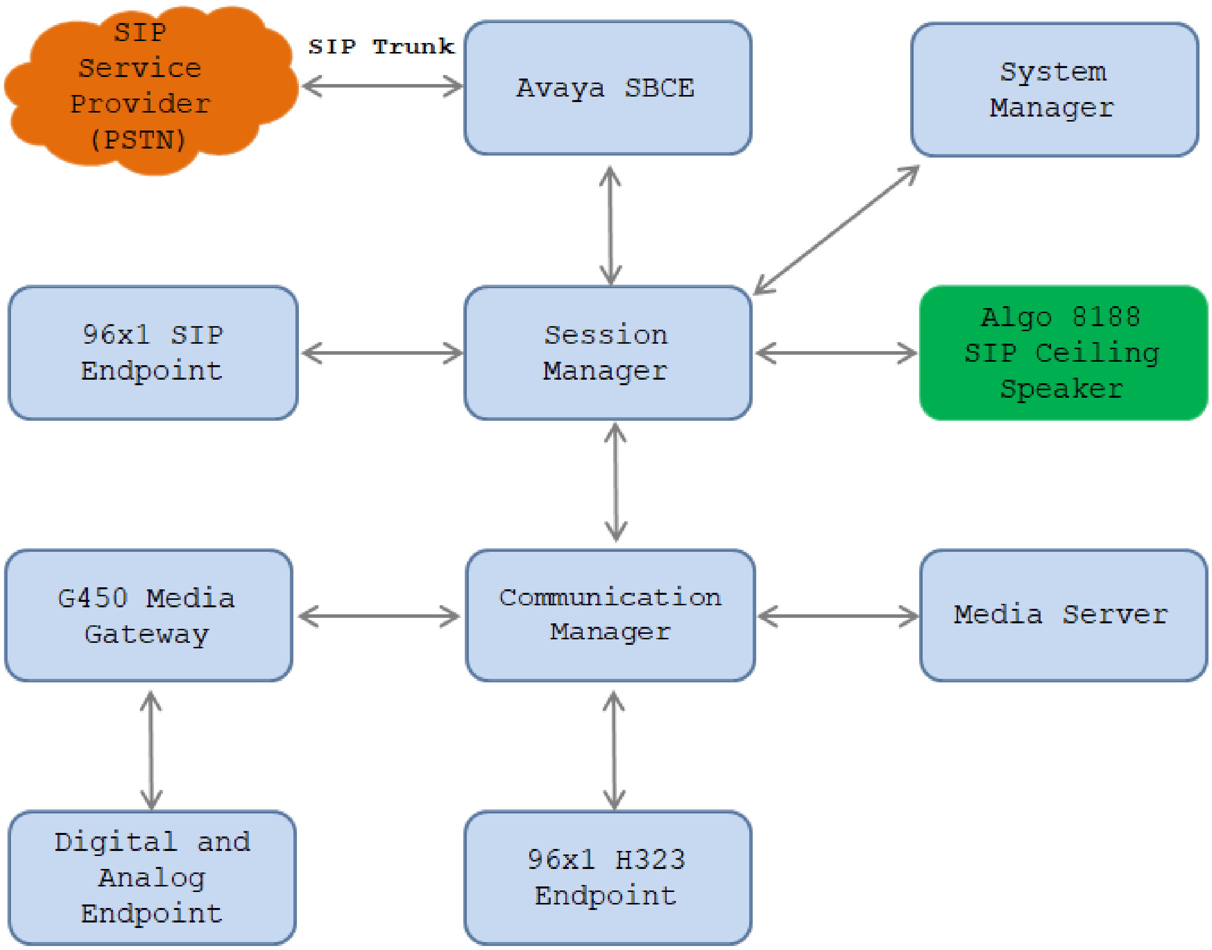


Figure 1: Test Configuration Diagram

The following table indicates the IP addresses that were assigned to the systems in the test configuration diagram:

Description	IP Address
System Manager	10.33.1.10
Session Manager Signaling	10.33.1.12
Communication Manager	10.33.1.6
Session Border Controller	10.33.1.51
Media Server	10.33.1.30
G450 Media Gateway	10.33.1.40
96x1 Endpoints	10.33.5.40-10.33.5.46
Algo 8188 SIP Ceiling Speaker	192.168.12.139

4. Equipment and Software Validated

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

Equipment/Software	Version/Release
Avaya Aura® System Manager running on Virtualized Environment	8.1.3 Build 8.1.3.0.1011784
Avaya Aura® Session Manager running on Virtualized Environment	8.1.3 Build 8.1.3.0.813014
Avaya Aura® Communication Manager running on Virtualized Environment	8.1.3 Build 8.1.3.2.0.890.26989
Avaya Aura® Server Media running on Virtualized Environment	8.0 Build 8.0.2.163
Avaya G450 Media Gateway	41 .34 .0
Avaya 96x1 IP Deskphones	7.1.9.0.8 (SIP) 6.85.11 (H323)
Algo 8188 SIP Ceiling Speaker Firmware	3.3.3

5. Configure Avaya Aura® Communication Manager

Configuration and verification operations on Communication Manager illustrated in this section were all performed using Avaya Site Administrator Emulation Mode. The information provided in this section describes the configuration of Communication Manager for this solution. It is implied a working system is already in place, including SIP trunks to a Session Manager. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**. The configuration described in this section can be summarized as follows:

- Verify System Capacity
- Define the Dial Plan

Note: Any settings not in **Bold** in the following screen shots may be left as Default.

5.1. Verify System Capacity

The license file installed on the system controls these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative. Use the **display system-parameters customer-options** command to determine these values. On **Page 1**, verify that the **Maximum Off-PBX Telephones** allowed in the system is sufficient. One OPS station is required per SIP device.

```
display system-parameters customer-options                      Page   1 of  10
                                OPTIONAL FEATURES

     G3 Version: V16                                 Software Package: Enterprise
       Location: 2                                    System ID (SID): 1
       Platform: 28                                   Module ID (MID): 1

                                                                USED
                                Platform Maximum Ports: 65000 290
                                      Maximum Stations: 41000 44
                              Maximum XMOBILE Stations: 41000 0
                    Maximum Off-PBX Telephones - EC500: 41000 0
                    Maximum Off-PBX Telephones -   OPS: 41000 14
                    Maximum Off-PBX Telephones - PBFMC: 41000 0
                    Maximum Off-PBX Telephones - PVFMC: 41000 0
                    Maximum Off-PBX Telephones - SCCAN: 41000 0
                         Maximum Survivable Processors: 313   0

        (NOTE: You must logoff & login to effect the permission changes.)
```

On **Page 2** of the **system-parameters customer-options form**, verify that the number of **Maximum Administered SIP Trunks** supported by the system is sufficient.

```
display system-parameters customer-options                        Page   2 of  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                                  USED
                     Maximum Administered H.323 Trunks: 12000 16
           Maximum Concurrently Registered IP Stations: 18000 2
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 41000 1
                   Maximum Video Capable IP Softphones: 18000 4
                       Maximum Administered SIP Trunks: 24000 180
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

        (NOTE: You must logoff & login to effect the permission changes.)
```

5.2. Define the Dial Plan

Use the **change dialplan analysis** command to define the dial plan used in the system. This includes all telephone extensions. In the sample configuration, telephone extensions are 4 digits long and begin with **33** and **34**.

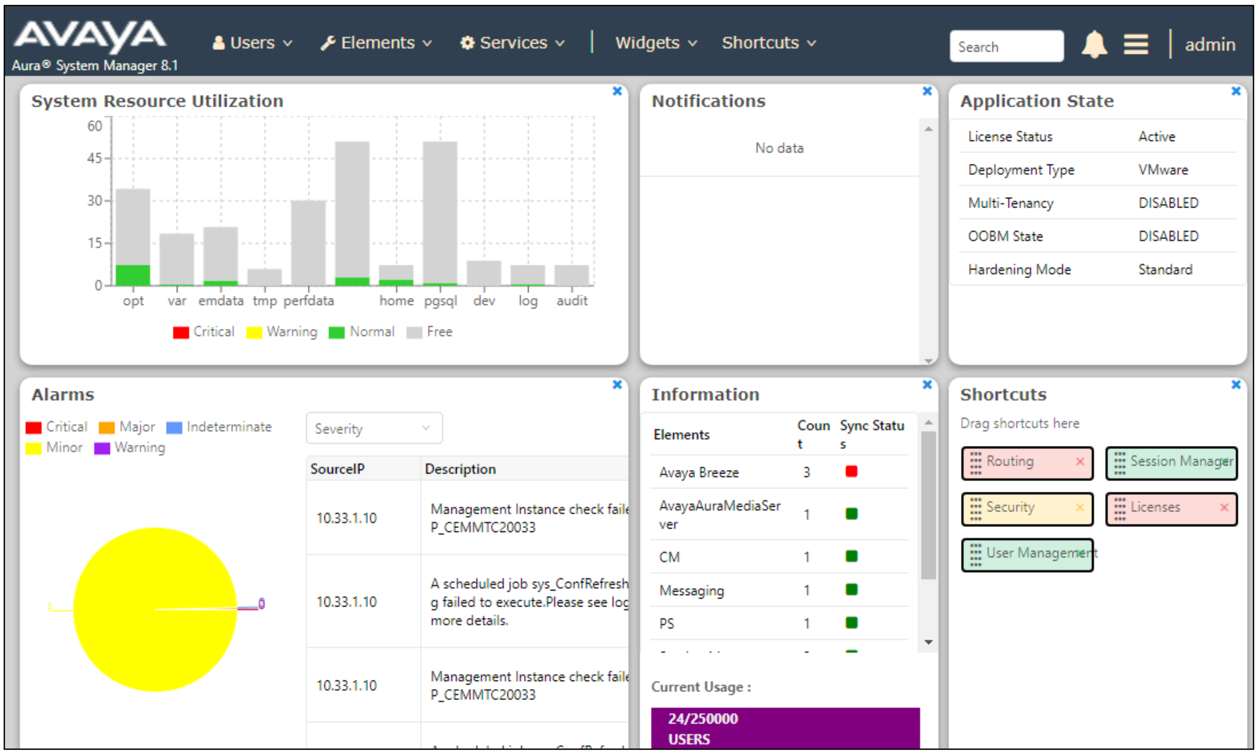
```
change dialplan analysis                                           Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all          Percent Full: 1

    Dialed   Total  Call      Dialed   Total  Call      Dialed   Total  Call
    String   Length Type      String   Length Type      String   Length Type
   33          4   ext
   34          4   ext
   *           3   fac
   #           3   fac
```

6. Configure Avaya Aura® Session Manager

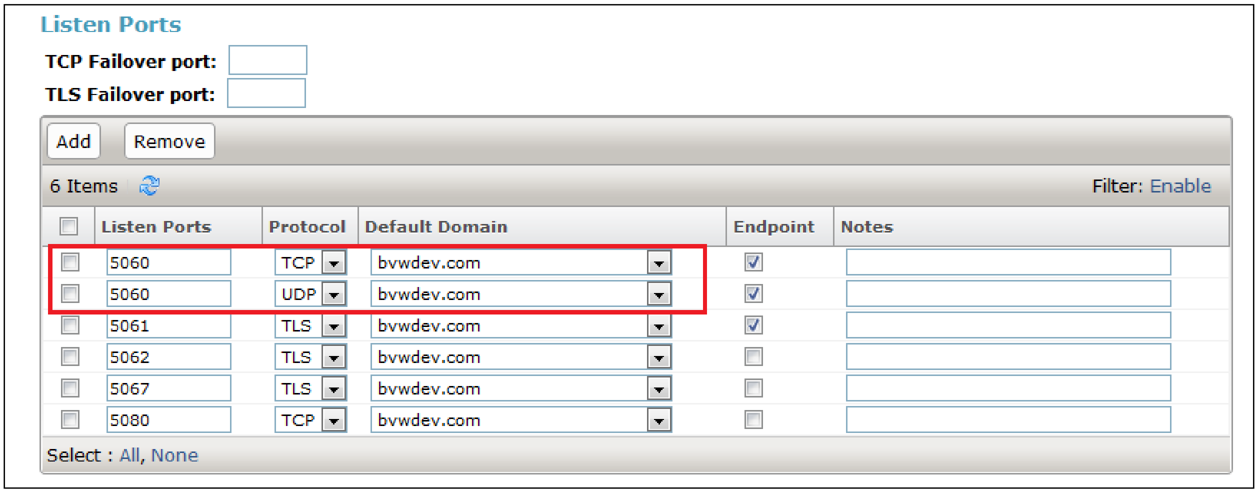
This section describes aspects of the Session Manager configuration required for interoperating with Algo 8188. It is assumed that the Domains, Locations, SIP entities, Entity Links, Routing Policies, Dial Patterns and Application Sequences have been configured where appropriate for Communication Manager and Session Manager.

Session Manager is managed via System Manager. Using a web browser, access **https://<ip-addr of System Manager>/SMGR**. In the **Log On** screen, enter appropriate **User ID** and **Password** and click the **Log On** button.



6.1. Verify Session Manager Listen Port for SIP Endpoint Registration

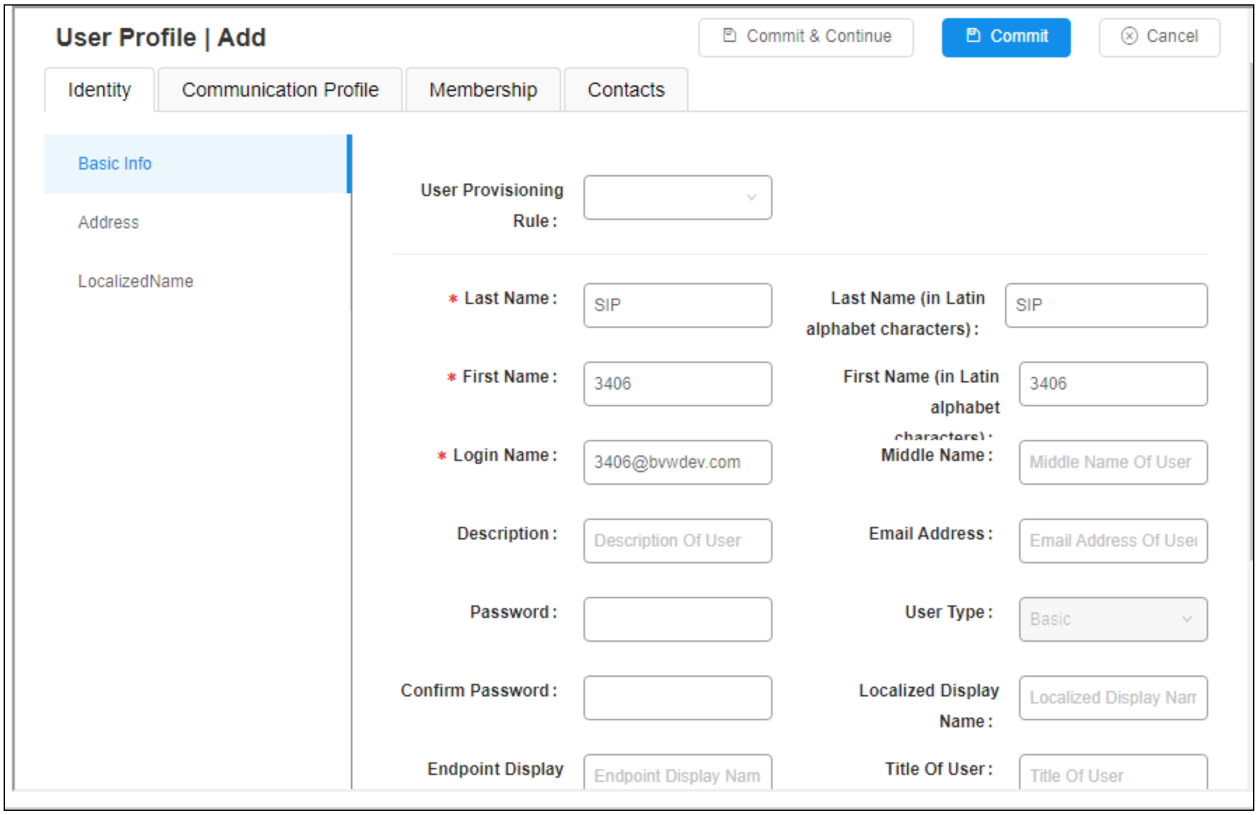
Each Session Manager Entity must be configured so that SIP endpoint can register to it using UDP, TCP, or TLS. From the web interface click **Routing → SIP Entities** (not shown) and select the Session Manager entity used for registration. In the compliance test, **TCP** and **UDP** listen ports were used. The TCP and UDP entries are highlighted below.



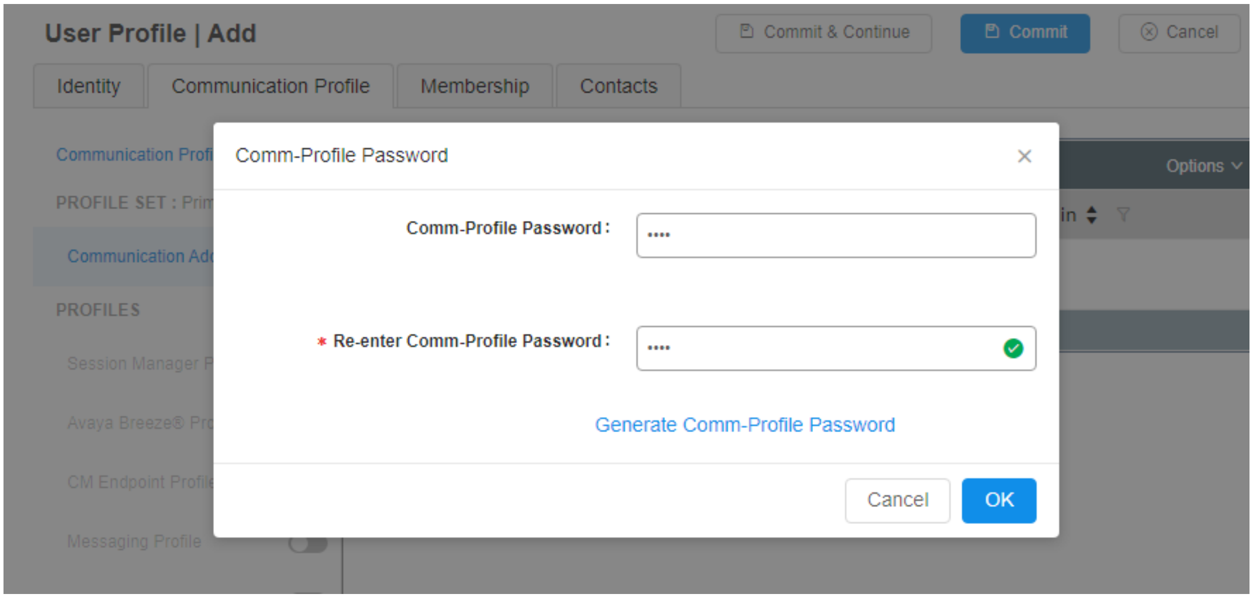
6.2. Add a SIP User

Two SIP users will be created for Algo 8188 Ring and Page. Click **User Management → Manage Users → New** (not shown) and configure the following in the **Identity** tab.

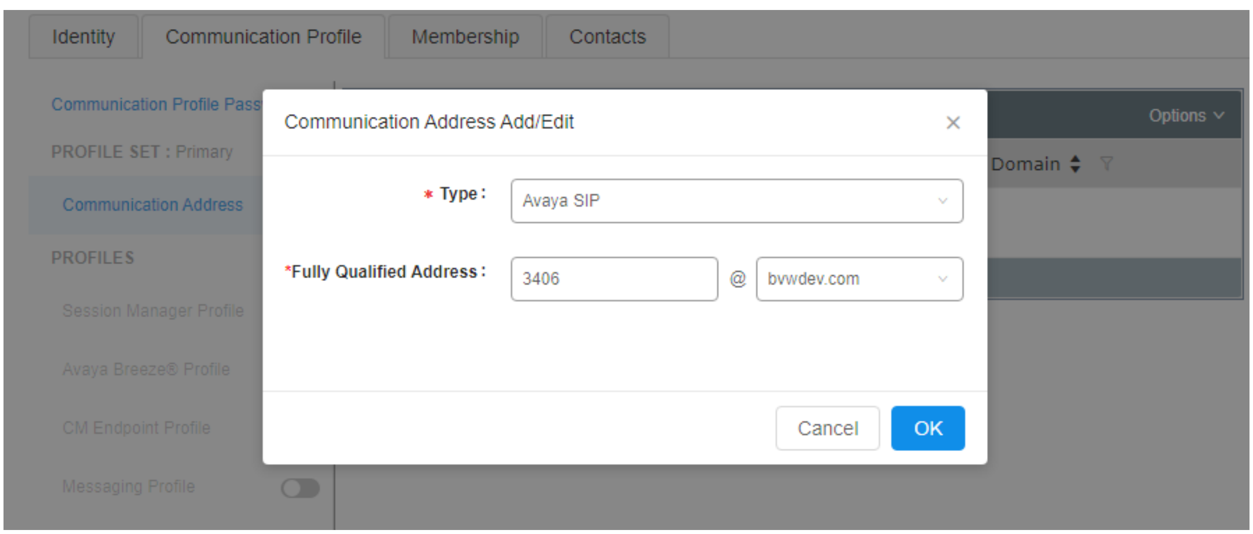
- **First Name** and **Last Name**   Enter an identifying name
- **Login Name**   Enter the extension number followed by the domain, in this case **3406@bvwdev.com**



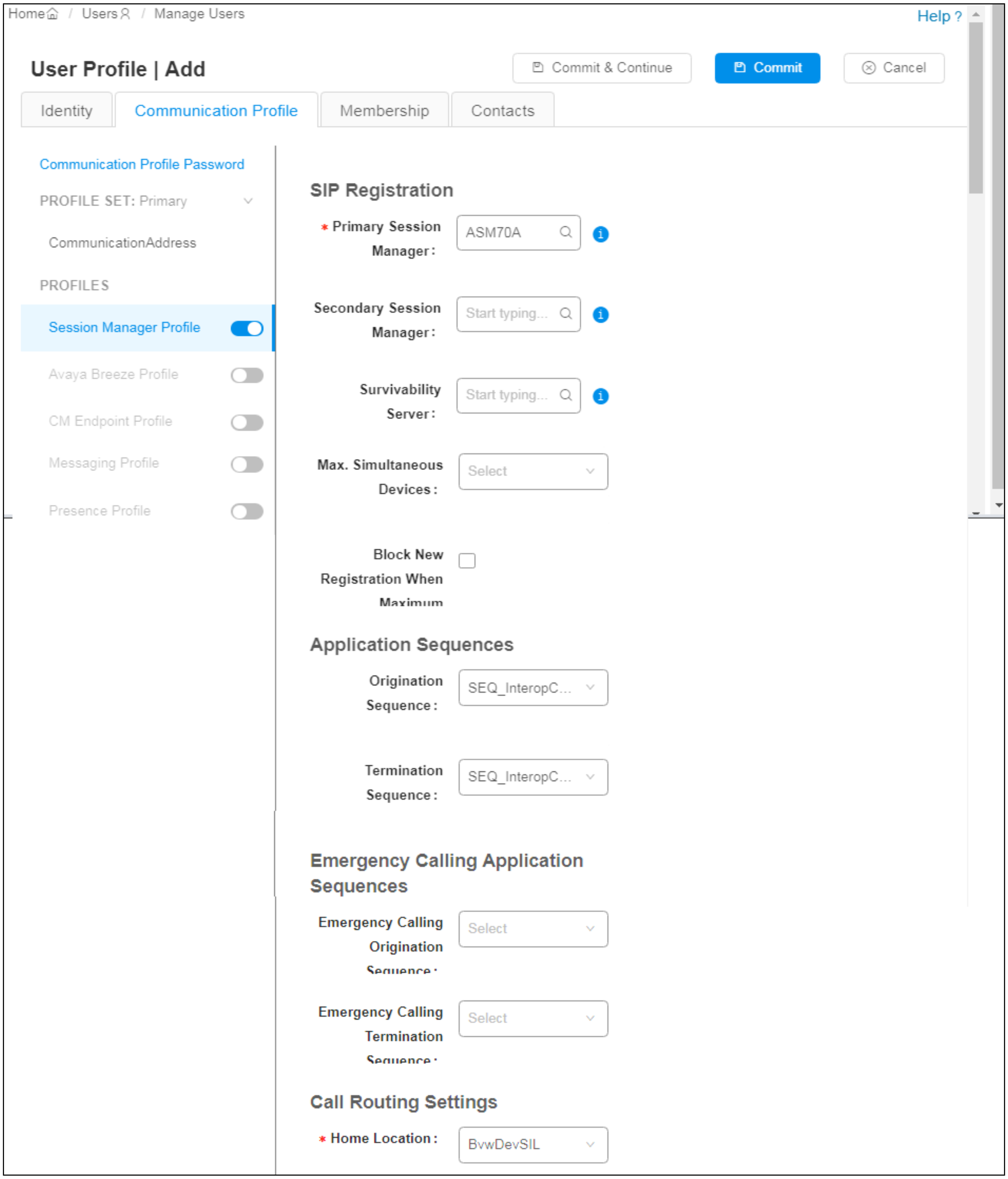
Click the **Communication Profile** tab and in the **Communication Profile Password** and **Confirm Password** fields, enter a numeric password. This will be used to register the Network Ceiling Speaker during login.



In the **Communication Address** section, for **Type** select **Avaya SIP** from the drop down list. In the **Fully Qualified Address** field enter the extension number as required and select the appropriate **Domain** from the drop down list. Click **OK** when done.



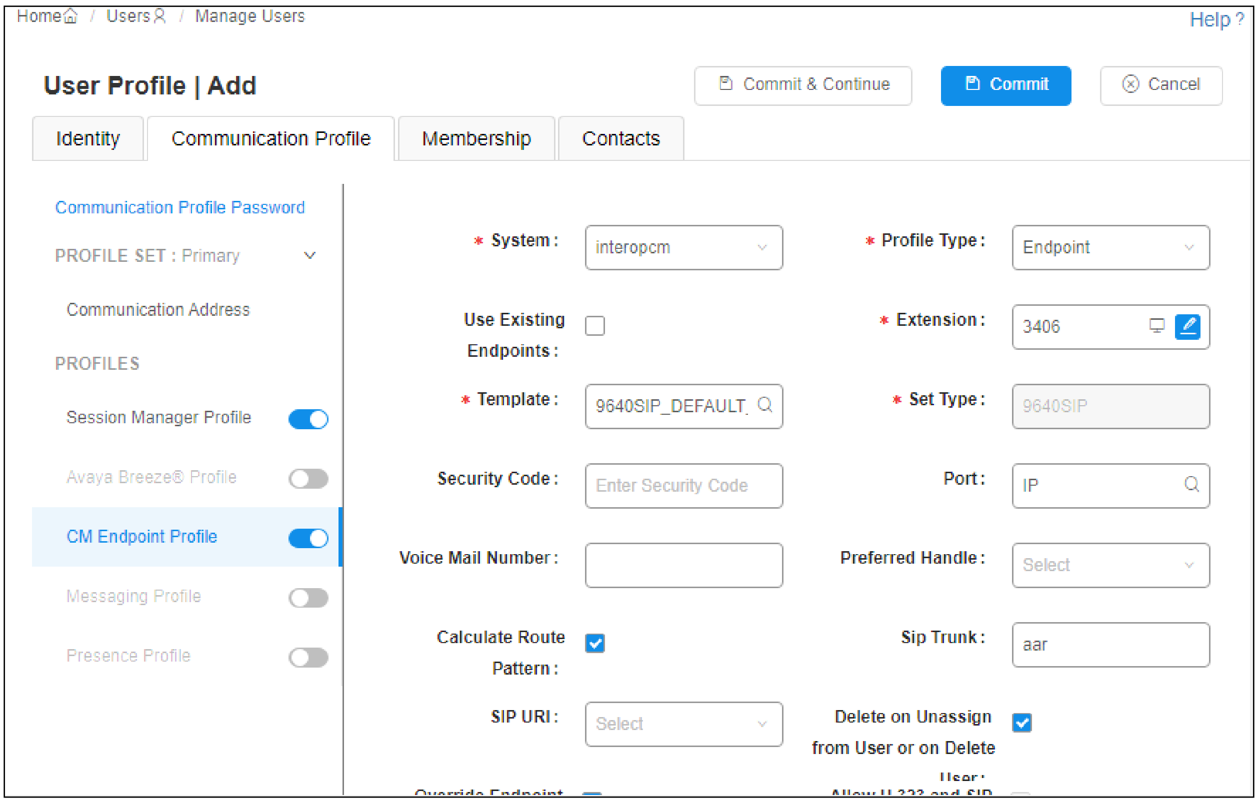
Place a tick in the **Session Manager Profile** check box and configure the **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence** and **Home Location**, from the respective drop-down lists. The Primary Session Manager used was **ASM70A**.



Place a tick in the **CM Endpoint Profile** check box and configure as follows:

- **System** Select the relevant Communication Manager SIP Entity from the drop down list
- **Profile Type** Select **Endpoint** from the drop down list
- **Extension** Enter the required extension number, in this case **3406**
- **Template** Select **9640SIP_DEFAULT_CM_8_1** from the drop down list
- **Port** The "IP" is auto filled out by the system

Click on **Commit** to save.



Repeat the procedure above to add another SIP user 3407 for Algo 8188 Ring.

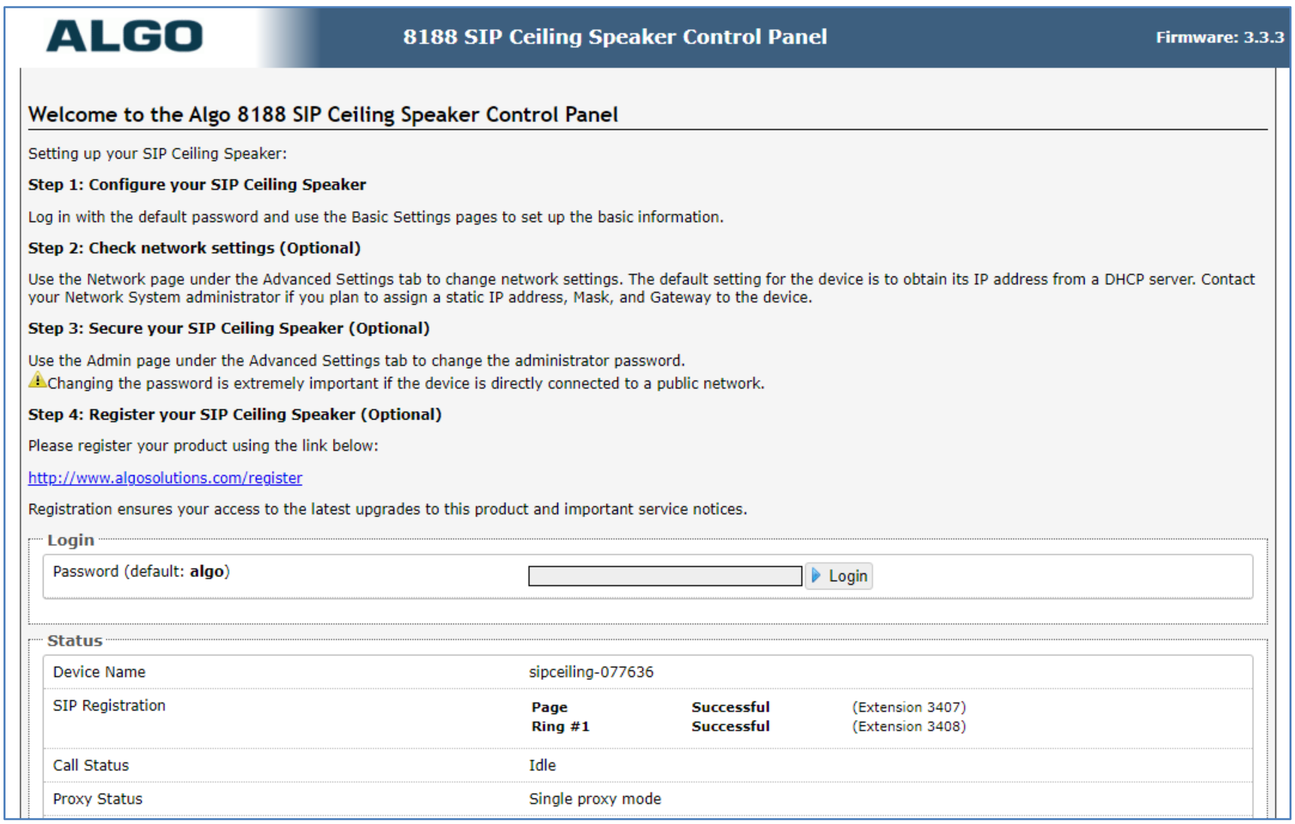
7. Configure 8188 SIP Ceiling Speaker

This section provides the procedures for configuring Algo 8188 SIP Ceiling Speaker. The procedures include the following areas:

- Launch web interface.
- Administer configuration.

7.1. Launch Web Interface

Access the 8188 SIP Ceiling Speaker web-based interface by using the URL "http://ip-address" in an Internet browser window, where "ip-address" is the IP address of the 8188. The **Welcome to the Algo 8188 SIP Ceiling Speaker Control Panel** screen is displayed, as shown below. Log in using the appropriate credentials.



7.2. Administer Algo 8188

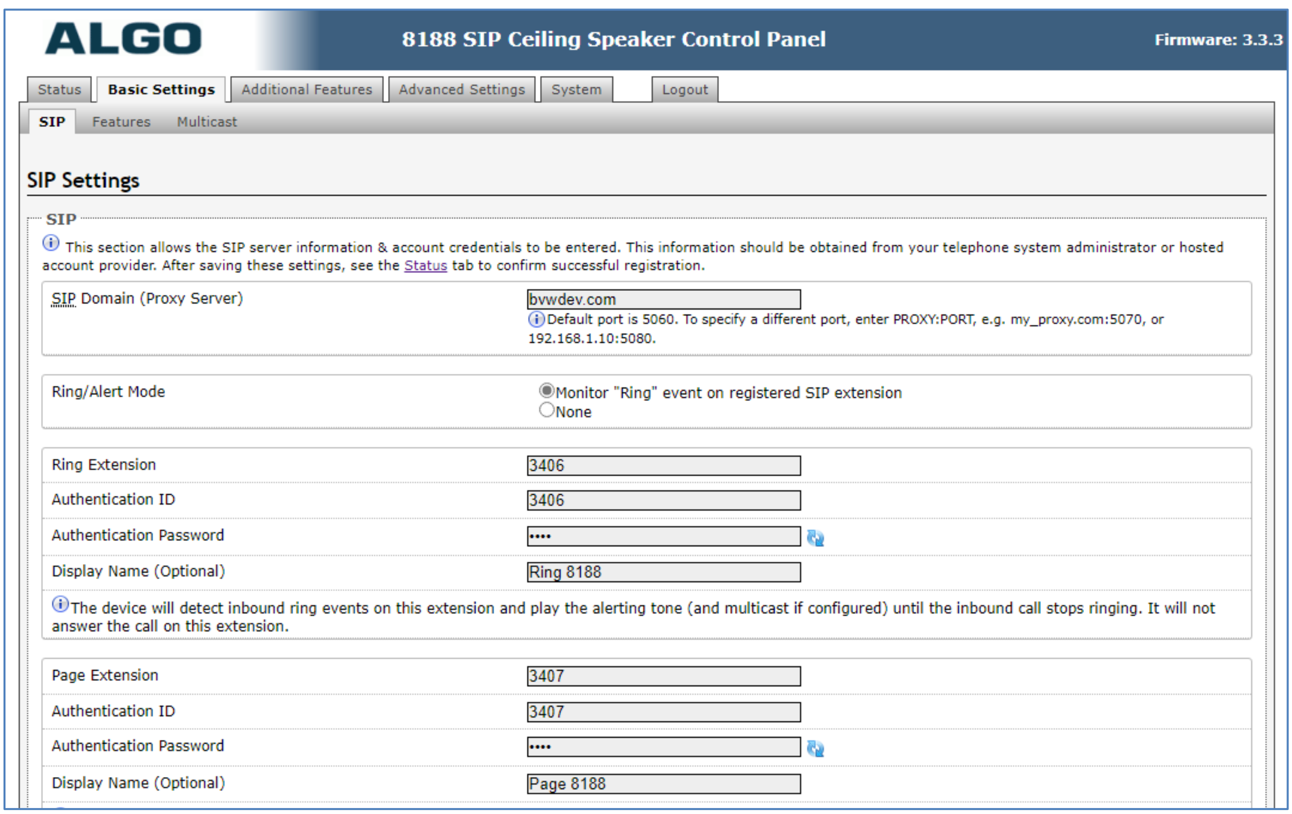
Select **Basic Settings → SIP** from the top menu, to display the screen below. Configure the **SIP Settings** section toward the bottom of the screen as desired to match the configuration. Enter the following values for the specified fields, and retain the default values in the remaining fields.

- **Sip Domain (Proxy Server):** Enter the SIP domain name as shown in **Section 6.1**.
- **Ring/Alert Mode:** Select **Monitor "Ring" event on registered SIP extension**.
- **Ring Extension:** Enter the extension of SIP user as configured in **Section 6.2**.
- **Authentication ID:** Enter the SIP user name as configured in **Section 6.2**.
- **Authentication Password:** Enter the SIP user login code as configured in **Section 6.2**.

In the Page section:

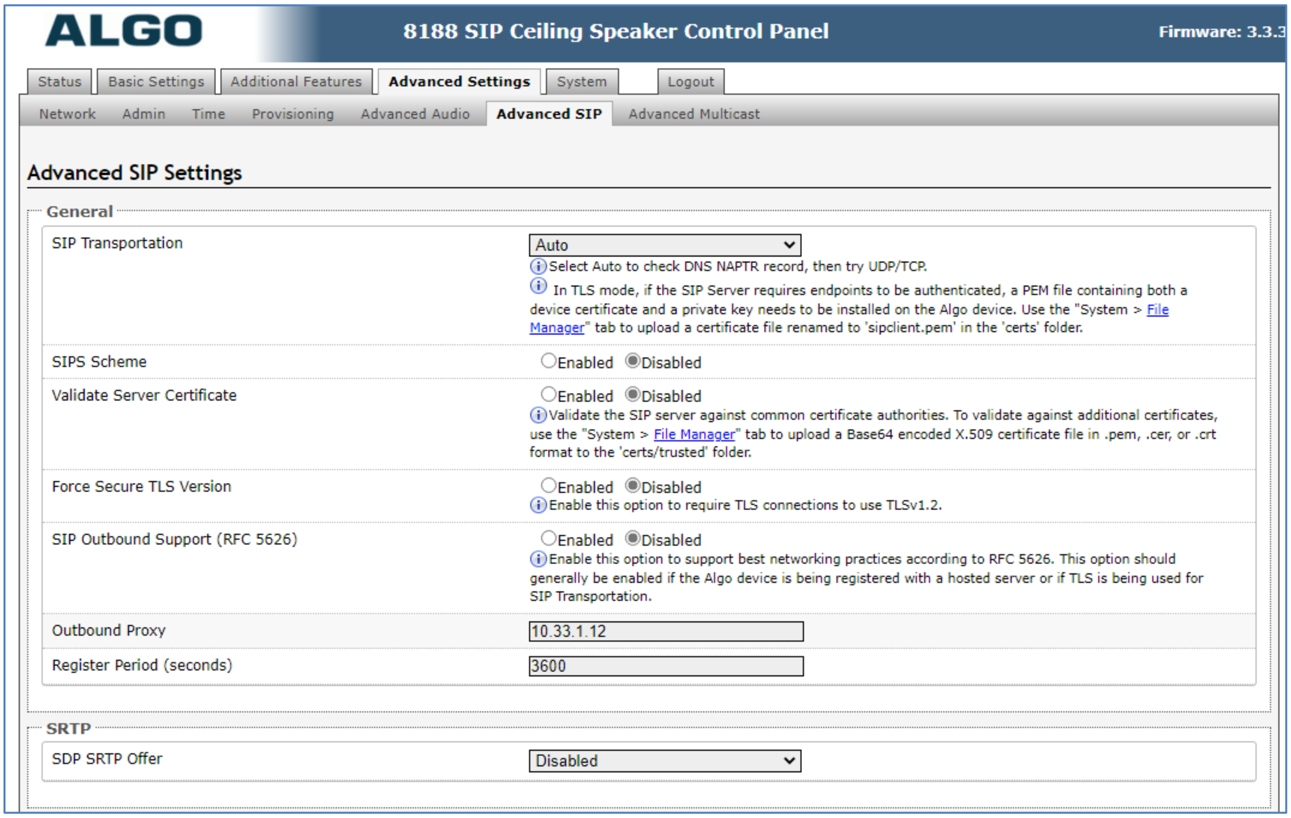
- **Page Extension:** Enter the voice paging SIP user extension from **Section 6.2**.
- **Authentication ID:** Enter the voice paging SIP user name from **Section 6.2**.
- **Authentication Password:** Enter the voice paging SIP user login code from **Section 6.2**.

Click on **Save** button to save the configuration.



Navigate to **Advanced Settings → Advanced SIP**. The **Advanced SIP** page is displayed, enter the signaling IP address of Session Manager in the **Outbound Proxy** and keep other values at default.

Click on **Save** button (not shown) to save the configuration.

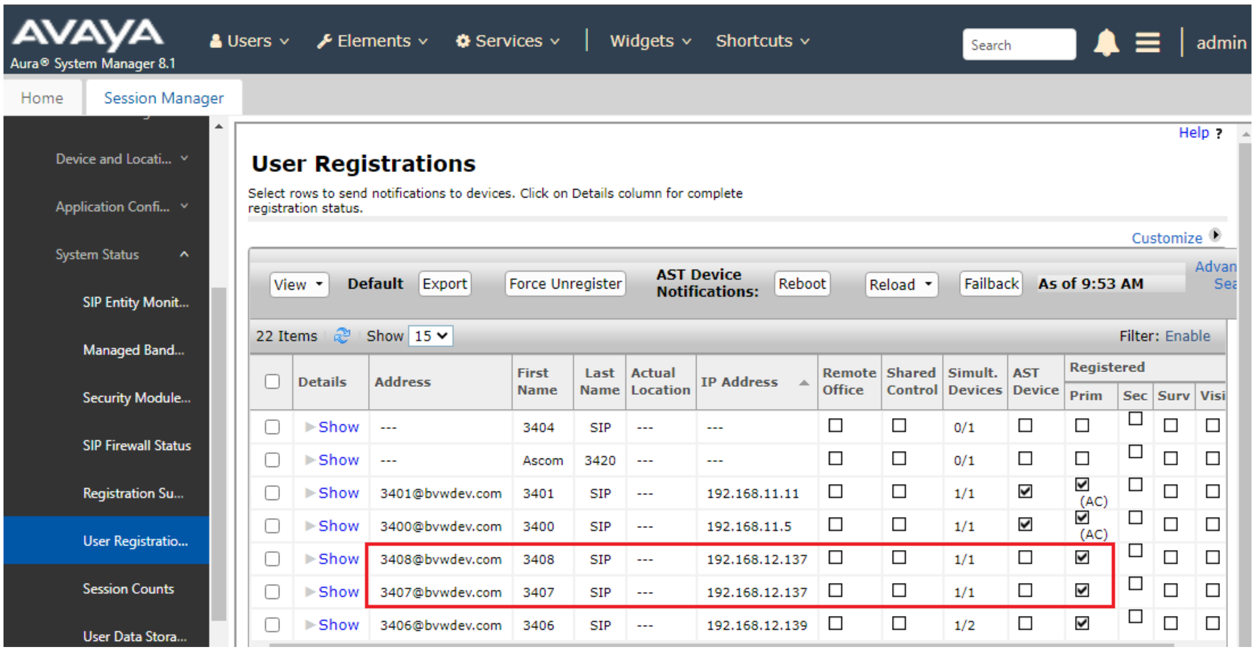


8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Aura® Session Manager and Algo 8188 SIP Ceiling Speaker.

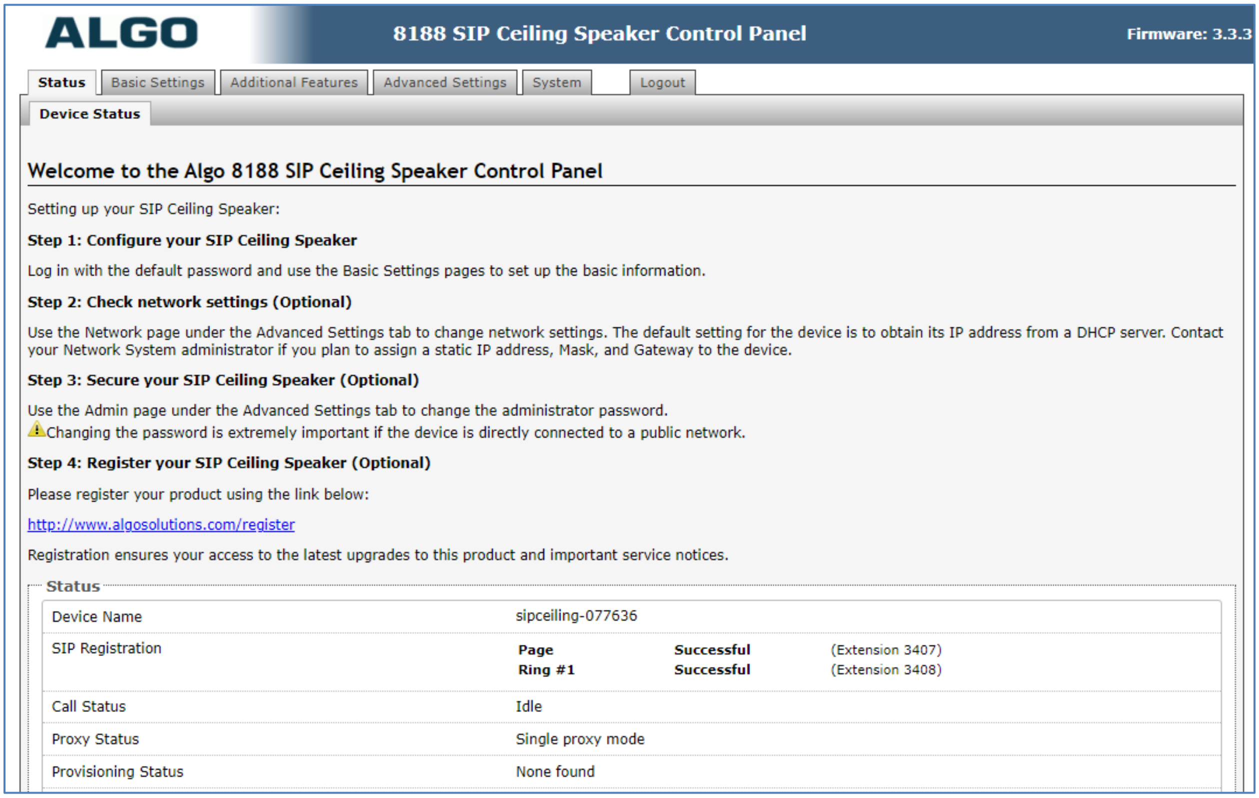
8.1. Verify Registration to Avaya Aura® Session Manager

From the System Manager dashboard select **Session Manager** from the **Elements** section (not shown). From the left hand menu select **System Status→User Registrations**. The Algo 8188 Ring and Page extensions are listed and a tick under **Registered** for the Session Manager as it is registered to.



8.2. Verify Algo 8188

From the Algo 8188 web-based interface, select **Status** from the top menu. Verify that **SIP Registration** displays “Ring – Successful” and “Page – Successful”, as shown below.



The following tests were conducted to verify the solution between the Algo 8188 and Communication Manager and Session Manager.

- Verify that the incoming call to the bridged extension on the Communication Manager that rings the 8188 Ring and the 8188 Ring stops ringing if the bridge extension answers the call.
- Verify that the incoming call to the 8188 Page is automatically answered with clear audio path.
- Verify that the telephone that places the incoming call to the 8188 Page can do conference, transfer, mute, un-mute and provide busy tone if it is on another call.
- Verify that the solution works with different Avaya clients (e.g., digital, analog, IP etc.).
- Verify that 8188 goes into an idle state when the call is completed.
- Verify that the 8188 re-registers without issues if the Ethernet cable is unplugged and plugged back in.

9. Conclusion

These Application Notes describe the configuration steps required to integrate the Algo 8188 SIP Ceiling Speaker with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. All of the executed test cases have passed and met the objectives outlined in **Section 2.1**.

10. Additional References

Product documentation for the Avaya Aura may be found at:
https://support.avaya.com/css/Products/

Avaya Aura Documents:

[1] Administering Avaya Aura® Communication Manager, Release 8.1.x, Issue 12, July 2021.
[2] Administering Avaya Aura® Session Manager, Release 8.1.x, Issue 10, September 2021.

Product documentation for the Algo 8188 SIP Ceiling Speaker products may be found at:
http://www.algosolutions.com/8188

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.