



## Avaya Solution & Interoperability Test Lab

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# 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.

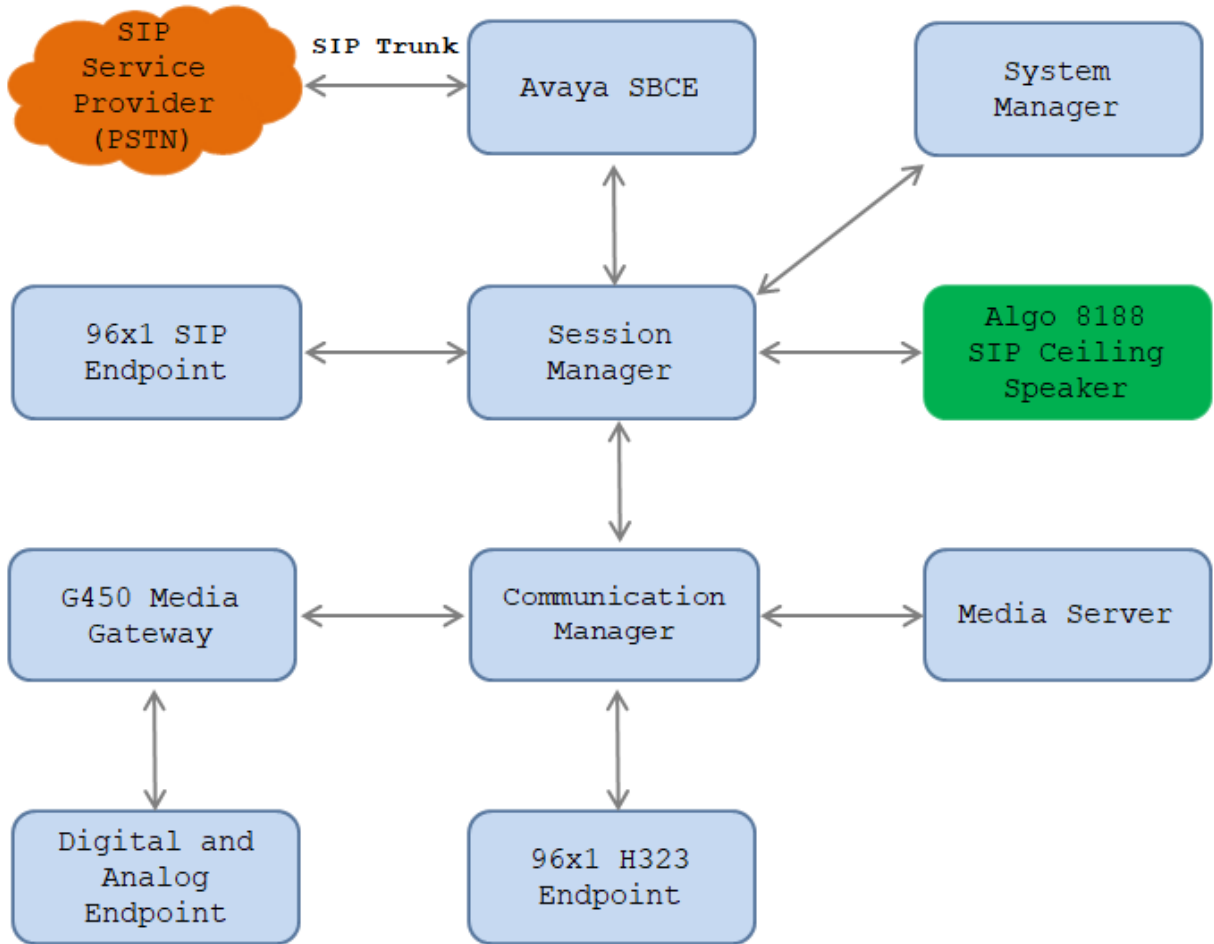
## 2.3. Support

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:

<b>Description</b>	<b>IP Address</b>
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:

<b>Equipment/Software</b>	<b>Version/Release</b>
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

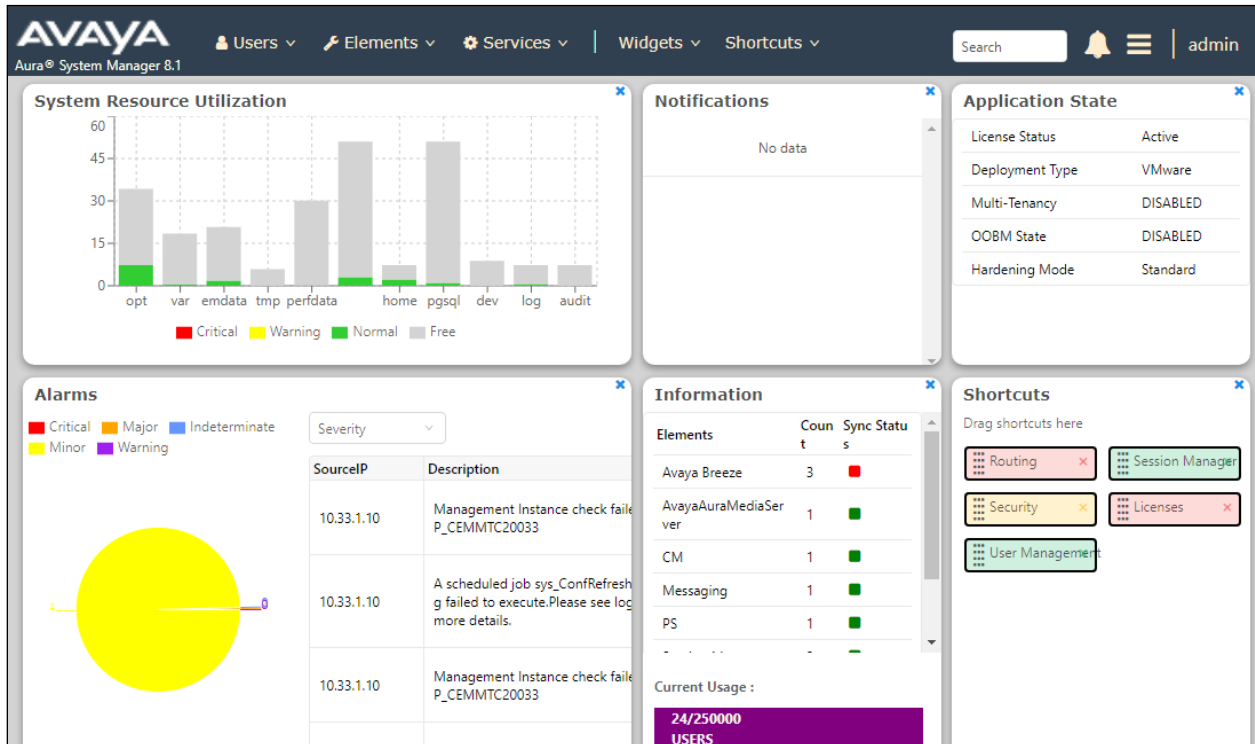
    Dialed   Total Call   Dialed   Total Call   Dialed   Total Call
    String   Length Type    String   Length Type    String   Length Type
33         4  ext
34         4  ext
    *         3   fac
    #         3   fac
  
```



## 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.

**Listen Ports**

TCP Failover port:

TLS Failover port:

6 Items  Filter: [Enable](#)

<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Endpoint	Notes
<input type="checkbox"/>	5060	TCP	bvwdev.com	<input checked="" type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	5060	UDP	bvwdev.com	<input checked="" type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	5061	TLS	bvwdev.com	<input checked="" type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	5062	TLS	bvwdev.com	<input type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	5067	TLS	bvwdev.com	<input type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	5080	TCP	bvwdev.com	<input type="checkbox"/>	<input type="text"/>

Select : [All](#), [None](#)

## 6.2. Add a SIP User

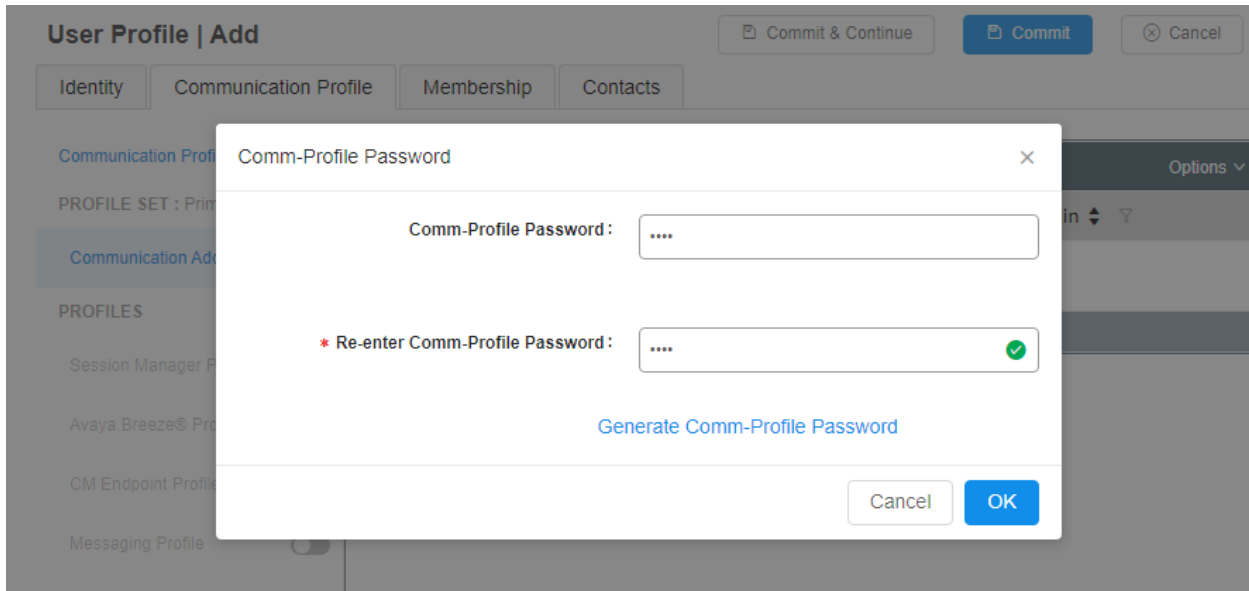
Two SIP users will be created for Algo 8188 Ring and Page. Click **User Management** → **Manage Users** → **New** (not shown) and configure the following in the **Identity** tab.

- **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**

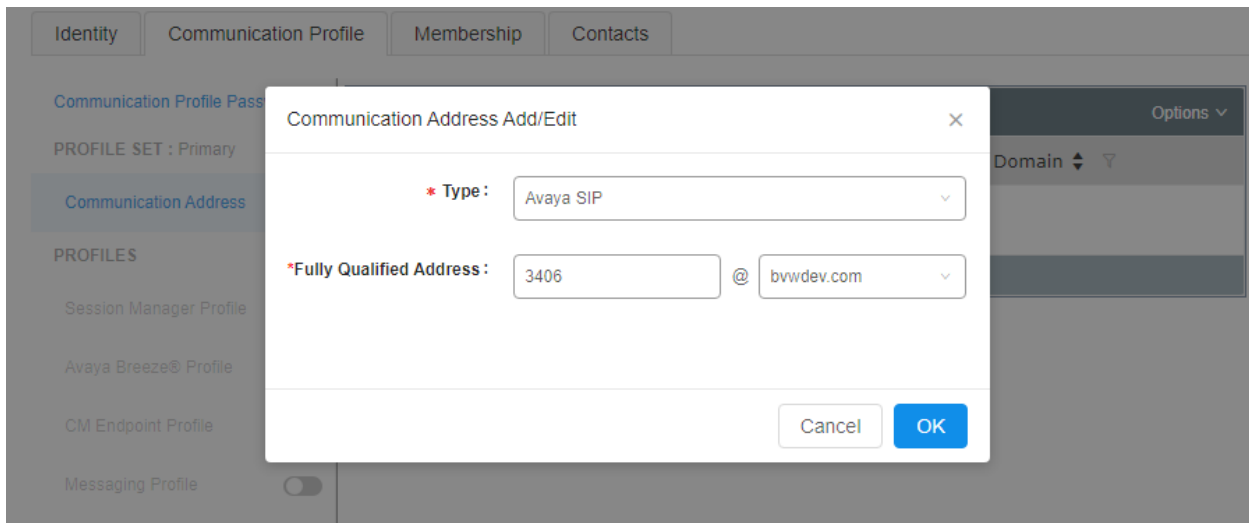
The screenshot shows the 'User Profile | Add' form with the 'Identity' tab selected. The form is divided into several sections:

- User Provisioning Rule:** A dropdown menu.
- Basic Info:** A sidebar menu with 'Basic Info' selected.
- Fields:**
  - \* Last Name:** SIP
  - Last Name (in Latin alphabet characters):** SIP
  - \* First Name:** 3406
  - First Name (in Latin alphabet characters):** 3406
  - \* Login Name:** 3406@bvwdev.com
  - Middle Name:** Middle Name Of User
  - Description:** Description Of User
  - Email Address:** Email Address Of User
  - Password:** (empty)
  - User Type:** Basic
  - Confirm Password:** (empty)
  - Localized Display Name:** Localized Display Name
  - Endpoint Display:** Endpoint Display Name
  - Title Of User:** Title Of User

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**.

Home / Users / Manage Users Help ?

### User Profile | Add

Commit & Continue Commit Cancel

Identity **Communication Profile** Membership Contacts

**Communication Profile Password**

PROFILE SET: Primary

CommunicationAddress

PROFILES

- Session Manager Profile**
- Avaya Breeze Profile
- CM Endpoint Profile
- Messaging Profile
- Presence Profile

#### SIP Registration

\* Primary Session Manager:  ⓘ

Secondary Session Manager:  ⓘ

Survivability Server:  ⓘ

Max. Simultaneous Devices:

Block New Registration When Maximum

#### Application Sequences

Origination Sequence:

Termination Sequence:

#### Emergency Calling Application Sequences

Emergency Calling Origination Sequence:

Emergency Calling Termination Sequence:

#### Call Routing Settings

\* Home Location:

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.

The screenshot shows the 'User Profile | Add' configuration page. The 'Communication Profile' tab is active. On the left, there is a sidebar with 'Communication Profile Password' (PROFILE SET: Primary) and 'PROFILES' (Session Manager Profile, Avaya Breeze® Profile, **CM Endpoint Profile**, Messaging Profile, Presence Profile). The main area contains the following fields:

- \* System: interopcm
- \* Profile Type: Endpoint
- Use Existing Endpoints:
- \* Extension: 3406
- \* Template: 9640SIP\_DEFAULT
- \* Set Type: 9640SIP
- Security Code: Enter Security Code
- Port: IP
- Voice Mail Number: (empty)
- Preferred Handle: Select
- Calculate Route Pattern:
- SIP URI: Select
- Sip Trunk: aar
- Delete on Unassign from User or on Delete:

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

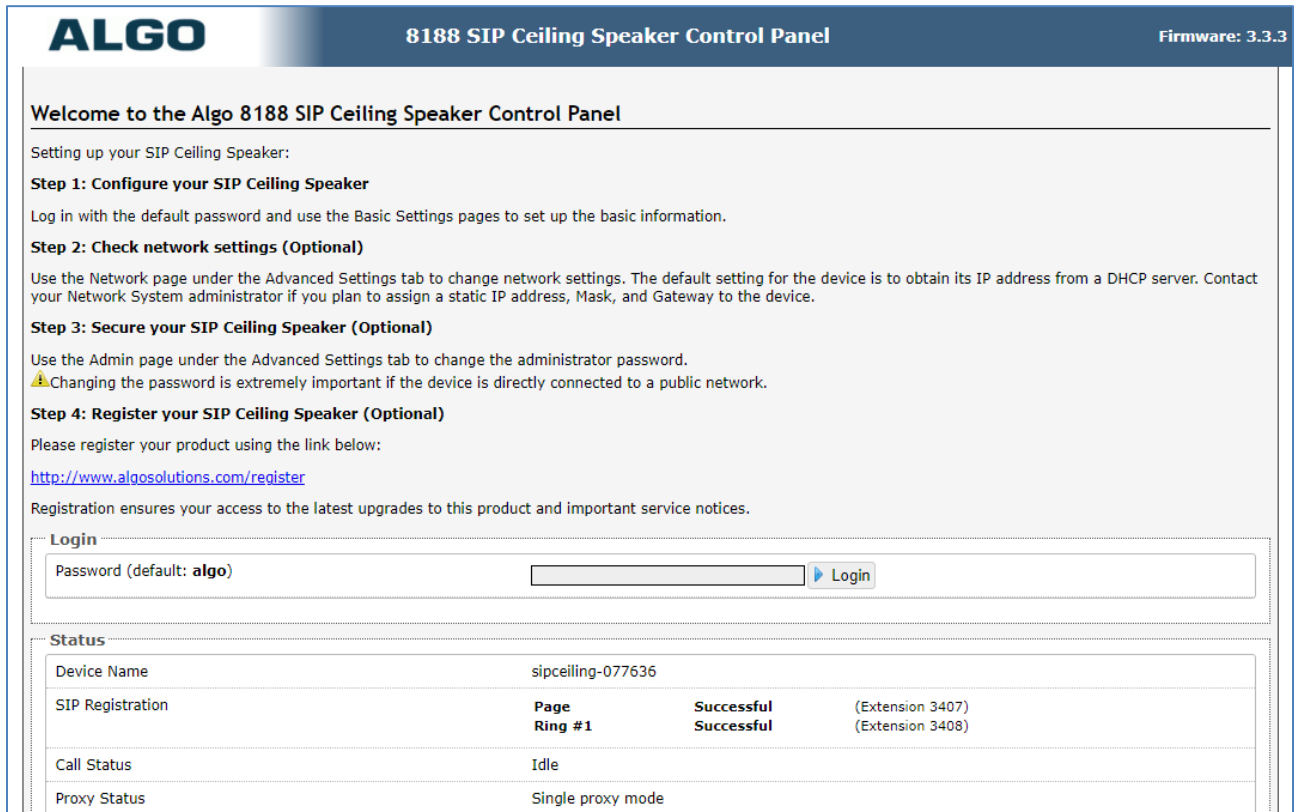
## 7. Configure 8188 SIP Ceiling Speaker

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.



**ALGO** 8188 SIP Ceiling Speaker Control Panel Firmware: 3.3.3

**Welcome to the Algo 8188 SIP Ceiling Speaker Control Panel**

Setting up your SIP Ceiling Speaker:

**Step 1: Configure your SIP Ceiling Speaker**  
Log in with the default password and use the Basic Settings pages to set up the basic information.

**Step 2: Check network settings (Optional)**  
Use the Network page under the Advanced Settings tab to change network settings. The default setting for the device is to obtain its IP address from a DHCP server. Contact your Network System administrator if you plan to assign a static IP address, Mask, and Gateway to the device.

**Step 3: Secure your SIP Ceiling Speaker (Optional)**  
Use the Admin page under the Advanced Settings tab to change the administrator password.  
⚠️ Changing the password is extremely important if the device is directly connected to a public network.

**Step 4: Register your SIP Ceiling Speaker (Optional)**  
Please register your product using the link below:  
<http://www.algosolutions.com/register>  
Registration ensures your access to the latest upgrades to this product and important service notices.

**Login**

Password (default: algo)

**Status**

Device Name	sipceiling-077636		
SIP Registration	<b>Page Ring #1</b>	<b>Successful Successful</b>	(Extension 3407) (Extension 3408)
Call Status	Idle		
Proxy Status	Single proxy mode		

## 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.

The screenshot displays the ALGO 8188 SIP Ceiling Speaker Control Panel interface. The top navigation bar includes tabs for Status, Basic Settings (selected), Additional Features, Advanced Settings, System, and Logout. The main content area is titled 'SIP Settings' and contains two sections: 'SIP' and 'Page Extension'. The 'SIP' section includes a descriptive note, a text field for 'SIP Domain (Proxy Server)' with the value 'bwvdev.com', a radio button for 'Ring/Alert Mode' set to 'Monitor "Ring" event on registered SIP extension', and text fields for 'Ring Extension' (3406), 'Authentication ID' (3406), 'Authentication Password' (masked), and 'Display Name (Optional)' (Ring 8188). The 'Page Extension' section includes text fields for 'Page Extension' (3407), 'Authentication ID' (3407), 'Authentication Password' (masked), and 'Display Name (Optional)' (Page 8188).



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.

**ALGO** 8188 SIP Ceiling Speaker Control Panel Firmware: 3.3.3

Status Basic Settings Additional Features **Advanced Settings** System Logout

Network Admin Time Provisioning Advanced Audio **Advanced SIP** Advanced Multicast

### Advanced SIP Settings

**General**

SIP Transportation	Auto	<p>ⓘ Select Auto to check DNS NAPTR record, then try UDP/TCP.</p> <p>ⓘ In TLS mode, if the SIP Server requires endpoints to be authenticated, a PEM file containing both a device certificate and a private key needs to be installed on the Algo device. Use the "System &gt; <a href="#">File Manager</a>" tab to upload a certificate file renamed to 'sipclient.pem' in the 'certs' folder.</p>
SIPS Scheme	<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled	
Validate Server Certificate	<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled	<p>ⓘ Validate the SIP server against common certificate authorities. To validate against additional certificates, use the "System &gt; <a href="#">File Manager</a>" tab to upload a Base64 encoded X.509 certificate file in .pem, .cer, or .crt format to the 'certs/trusted' folder.</p>
Force Secure TLS Version	<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled	<p>ⓘ Enable this option to require TLS connections to use TLSv1.2.</p>
SIP Outbound Support (RFC 5626)	<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled	<p>ⓘ Enable this option to support best networking practices according to RFC 5626. This option should generally be enabled if the Algo device is being registered with a hosted server or if TLS is being used for SIP Transportation.</p>
Outbound Proxy	10.33.1.12	
Register Period (seconds)	3600	

**SRTP**

SDP SRTP Offer	Disabled	
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## 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

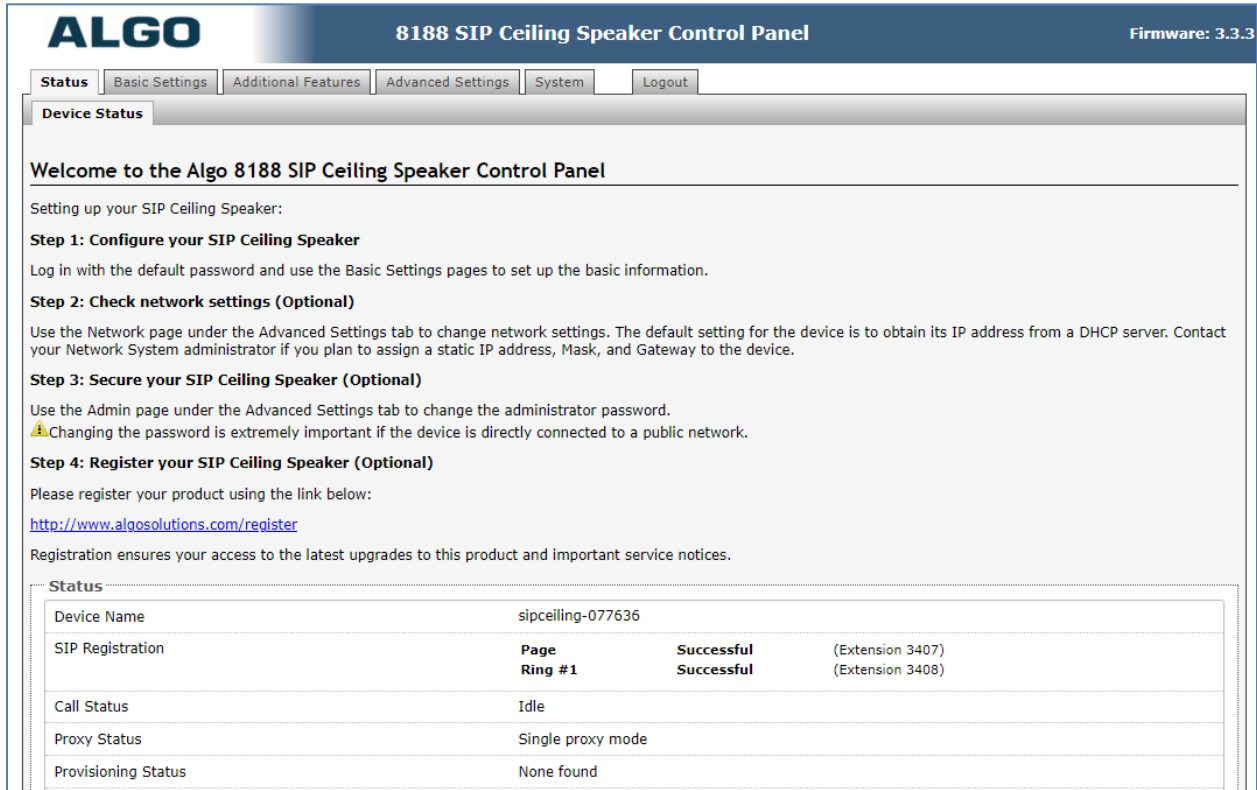
From the System Manager dashboard select **Session Manager** from the **Elements** section (not shown). From the left hand menu select **System Status**→**User Registrations**. The Algo 8188 Ring and Page extensions are listed and a tick under **Registered** for the Session Manager as it is registered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

The screenshot displays the Avaya Aura System Manager 8.1 interface. The top navigation bar includes the Avaya logo, user information, and search. The left sidebar shows the navigation menu with 'User Registrations' selected. The main content area is titled 'User Registrations' and contains a table of user registration details. The table has columns for 'Details', 'Address', 'First Name', 'Last Name', 'Actual Location', 'IP Address', 'Remote Office', 'Shared Control', 'Simult. Devices', 'AST Device', and 'Registered' (with sub-columns for Prim, Sec, Surv, and Visi). The row for '3408@bvwdev.com' is highlighted with a red box, indicating it is registered.

	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered			
											Prim	Sec	Surv	Visi
<input type="checkbox"/>	<a href="#">Show</a>	---	3404	SIP	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	<a href="#">Show</a>	---	Ascom	3420	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	<a href="#">Show</a>	3401@bvwdev.com	3401	SIP	---	192.168.11.11	<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"/>	<a href="#">Show</a>	3400@bvwdev.com	3400	SIP	---	192.168.11.5	<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"/>	<a href="#">Show</a>	3408@bvwdev.com	3408	SIP	---	192.168.12.137	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>		<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	<a href="#">Show</a>	3407@bvwdev.com	3407	SIP	---	192.168.12.137	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>		<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	<a href="#">Show</a>	3406@bvwdev.com	3406	SIP	---	192.168.12.139	<input type="checkbox"/>	<input type="checkbox"/>	1/2	<input type="checkbox"/>	<input checked="" type="checkbox"/>		<input type="checkbox"/>	<input type="checkbox"/>

## 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 screenshot shows the web interface for the Algo 8188 SIP Ceiling Speaker. The top navigation bar includes the ALGO logo, the title "8188 SIP Ceiling Speaker Control Panel", and the firmware version "Firmware: 3.3.3". Below the navigation bar are tabs for "Status", "Basic Settings", "Additional Features", "Advanced Settings", "System", and "Logout". The "Status" tab is selected, and the "Device Status" sub-tab is active. The main content area displays a welcome message and four setup steps: Step 1 (Configure your SIP Ceiling Speaker), Step 2 (Check network settings), Step 3 (Secure your SIP Ceiling Speaker), and Step 4 (Register your SIP Ceiling Speaker). Below the steps is a "Status" section with a table showing the current device status.

Status			
Device Name	sipceiling-077636		
SIP Registration	Page Ring #1	Successful Successful	(Extension 3407) (Extension 3408)
Call Status	Idle		
Proxy Status	Single proxy mode		
Provisioning Status	None found		

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>

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