Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Algo 8028 SIP Doorphone Version 2.7.4 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager Release 8.0 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Algo 8028 SIP Doorphone to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Algo 8028 SIP Doorphone is a SIP-based device that can register with Avaya Aura® Session Manager as a SIP endpoint and enables conversations and remote entry using door release features.

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 8028 SIP Doorphone to interoperate with Avaya Aura ® Communication Manager and Avaya Aura® Session Manager. Algo 8028 SIP Doorphone is a SIP-based device that can register with Avaya Aura® Session Manager as a SIP endpoint using UDP protocol and enables conversations and remote entry using door release features.

Algo 8028 SIP Doorphone (hereafter referred as 8028) is an outdoor rated IP intercom compatible with premise based and hosted SIP communication servers. By connecting to the VoIP telephone system, arriving visitors and guests can be greeted from any telephone or client and allowed entry by a simple key press.

2. General Test Approach and Test Results

The feature test cases were performed manually. The focus of this interoperability compliance testing was to verify if the 8028 can register as a SIP endpoint to Session Manager and able to make a call to and from a telephone on the Session Manager and able to open the door when the key is pressed on the phone.

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 the Algo 8028 SIP Doorphone did not include use of any specific encryption features as requested by Algo.

2.1. Interoperability Compliance Testing

The Compliance testing verified that the 8028 was able to interoperate with the telephones residing on the Aura systems. The following interoperability areas were covered:

- The 8028 can register to the Session Manager as a SIP endpoint.
- The 8028 can make a call to an endpoint on Communication Manager and establish a clear speech path.
- Endpoints on Communication Manager can call the extension assigned to the 8028 and establish speech path between the endpoint and the 8028.
- Endpoints on Communication Manager can send required DTMF tones and therefore ensure the remote door release features work successfully.

The serviceability testing focused on verifying the ability of the 8028 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 8028 SIP Doorphone 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 Communication Manager and Session Manager and the 8028. The 8028 communicated with Aura systems through Avaya switch with Power over Ethernet (PoE) and registered with Session Manager as SIP endpoint. The PRI T1 trunk was also configured to connect from Avaya G450 Media Gateway to PSTN for test cases off-net via PRI T1 trunk.

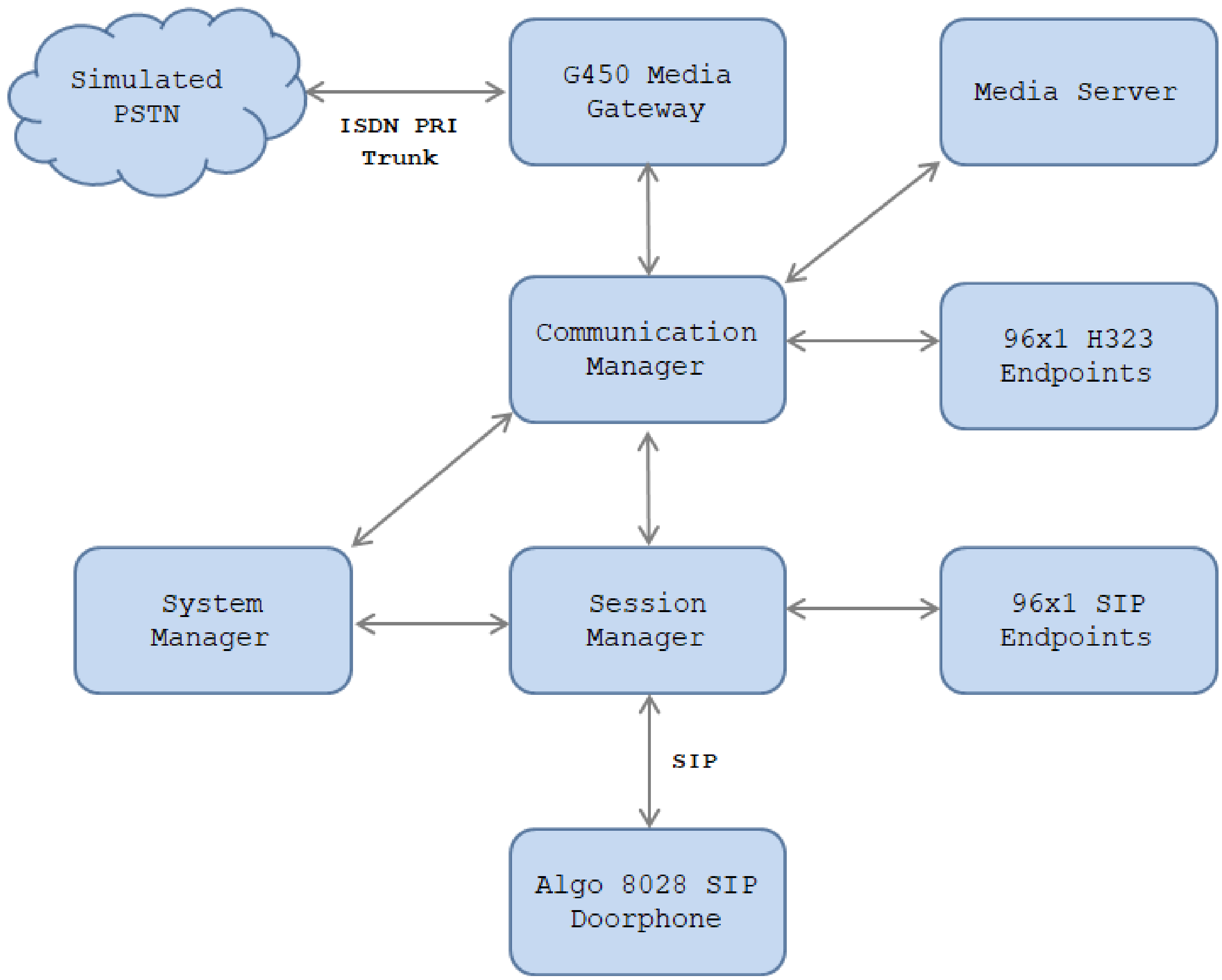


Figure 1: Test Configuration Diagram

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
Breeze Signaling	10.33.1.16
Communication Manager	10.33.1.6
Media Server	10.33.1.30
G450 Media Gateway	10.33.1.40
96x1 Endpoints	10.33.5.40-10.33.5.46
Algo 8028 SIP Doorphone	10.33.5.50

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	8.0 Build 8.0.0.0.931077
Avaya Aura® Session Manager running on Virtualized Environment	8.0 Build 8.0.0.0.800035
Avaya Aura® Communication Manager running on Virtualized Environment	8.0 Build 8.0.0.1.2.822 Patch 24826
Avaya Aura® Server Media running on Virtualized Environment	8.0 Build 8.0.0.117
Avaya G450 Media Gateway	40 .10 .0
Avaya 96x1 IP Deskphones	7.1.3.0.8 (SIP) 6.6604 (H323)
Avaya 1416 Digital Deskphone	Fw 1
Algo 8028 SIP Doorphone Firmware Kernel Version	 2.7.4 R1.5

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**.

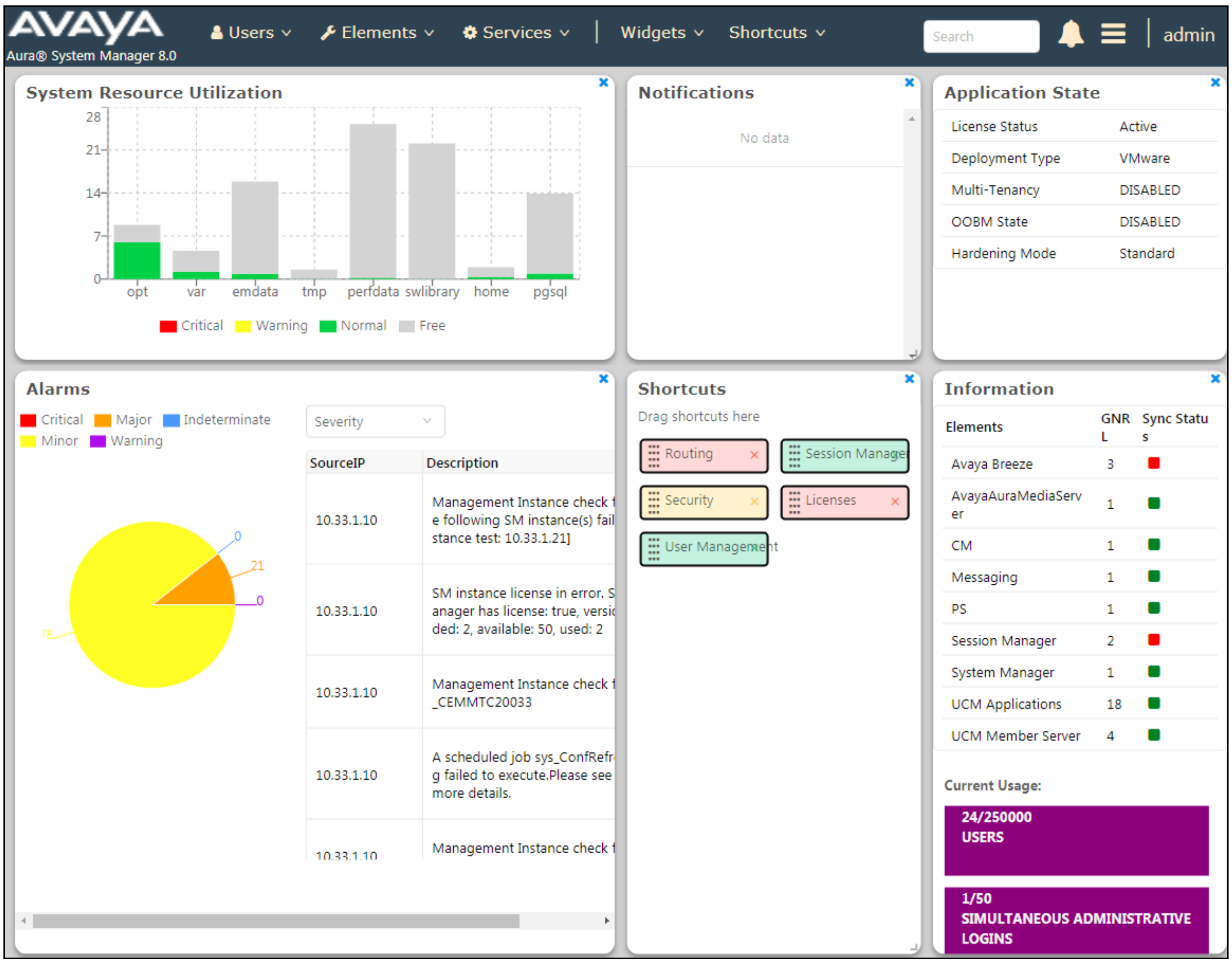
```
change dialplan analysis                                      Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all          Percent Full: 1

    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

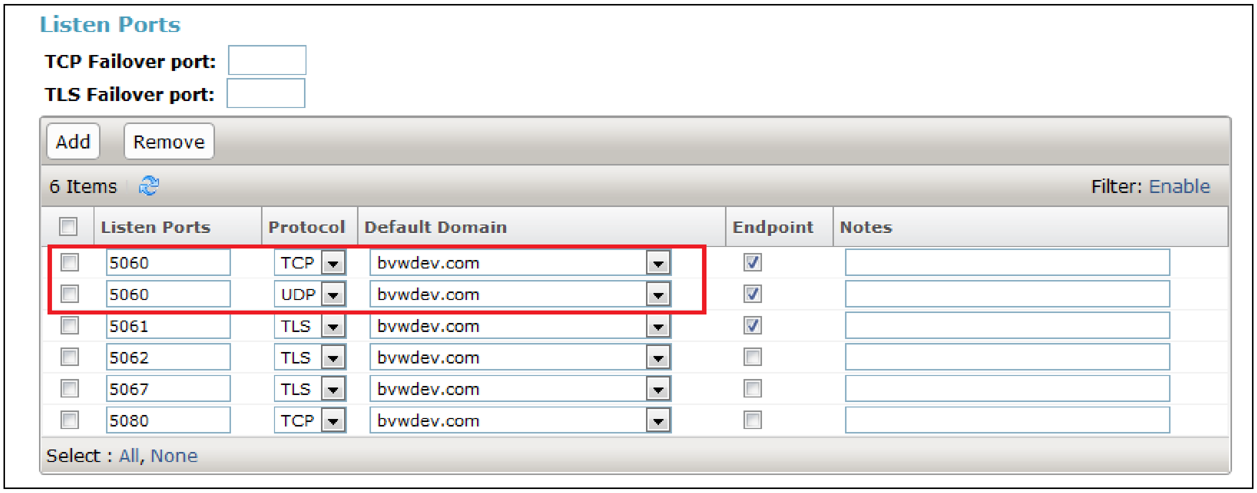
This section describes aspects of the Session Manager configuration required for interoperating with Aura Alliance Client. It is assumed that the Domains, Locations, SIP entities, Entity Links, Routing Policies, Dial Patterns and Application Sequences have been configured where appropriate for Communication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

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. Check Session Manager Ports

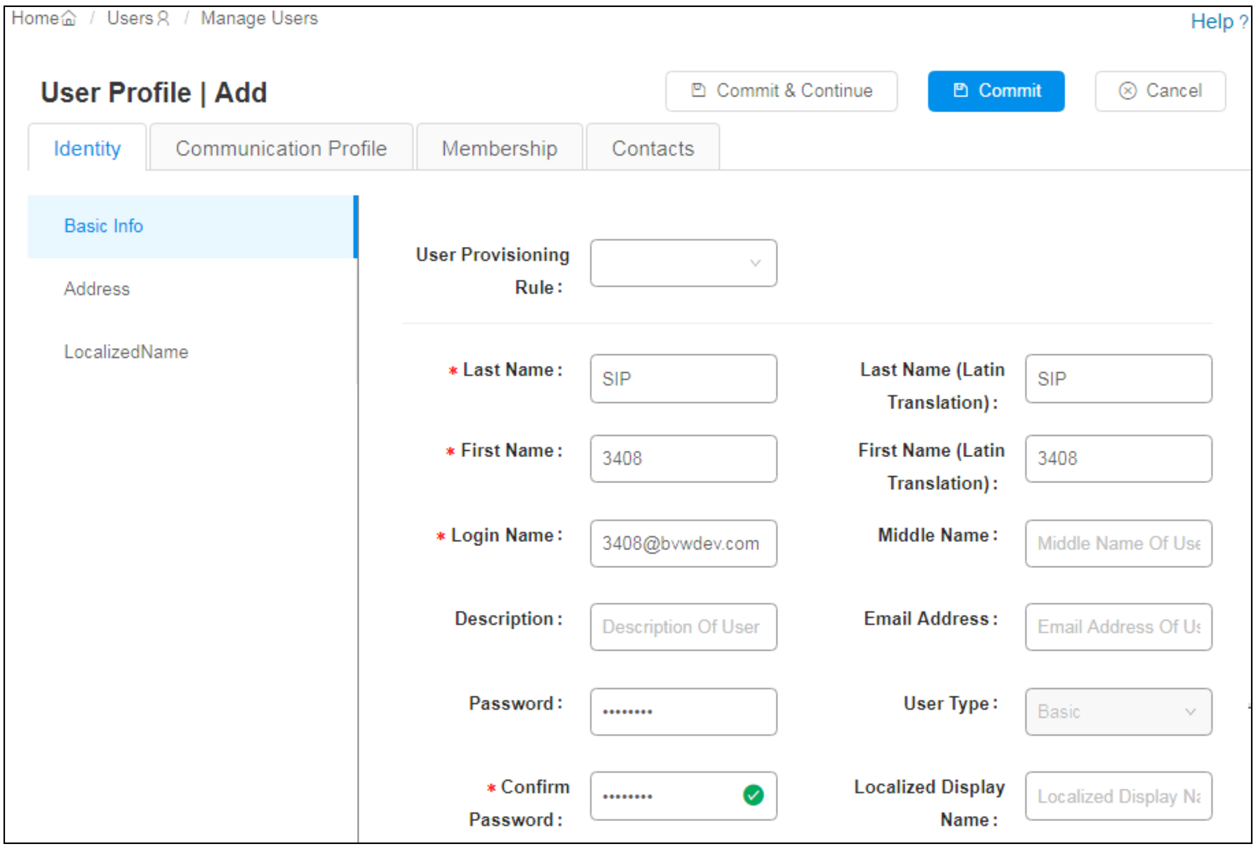
Each Session Manager Entity must be configured so that the 8028 can register to it using UDP/TCP. From the web interface click **Routing → SIP Entities** (not shown) and select the Session Manager entity used for registration. Make sure that **TCP** and **UDP** entries are present. The TCP and UDP entries are highlighted below.



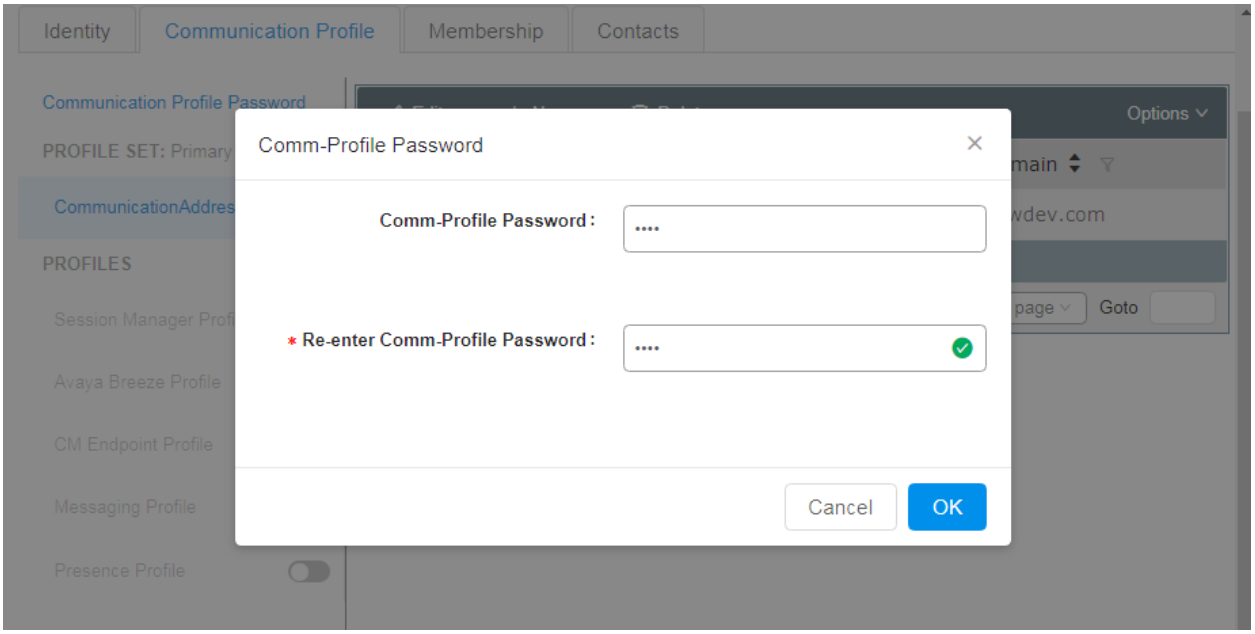
6.2. Add a SIP User

The 8028 SIP user must be added as a user. A user must be added for the 8028. Click **User Management → Manage Users → New** (not shown) and configure the following in the **Identity** tab.

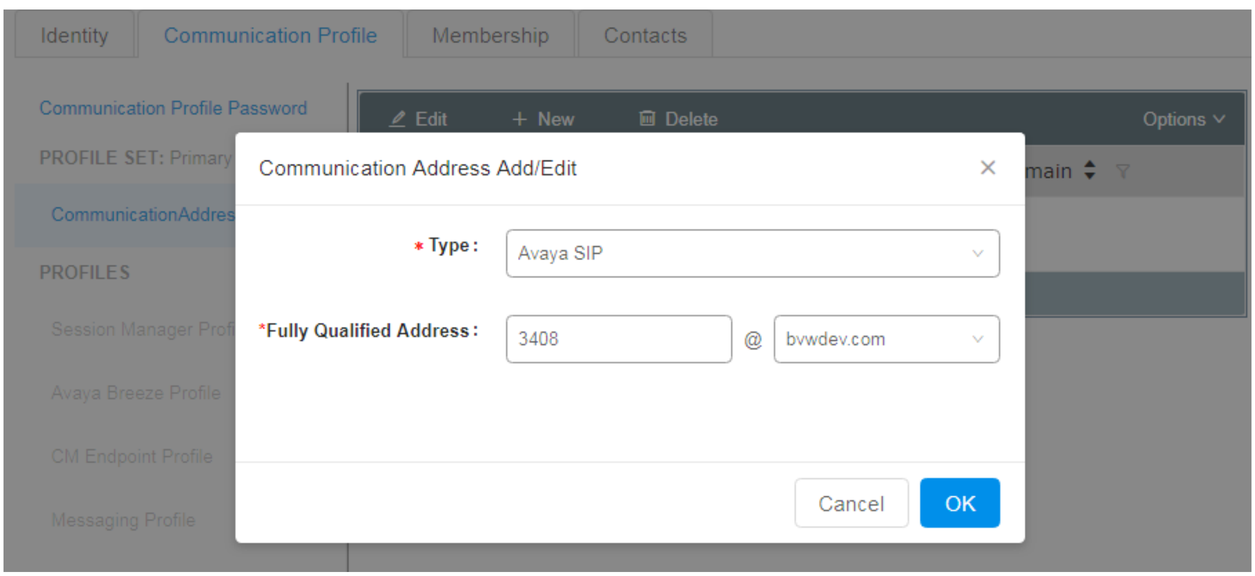
- **First Name** and **Last Name** Enter an identifying name
- **Login Name** Enter the extension number followed by the domain, in this case **3408@bvwdev.com**
- **User Type** Select **Basic** from the drop down list
- **Password** and **Confirm Password** Enter and confirm a password



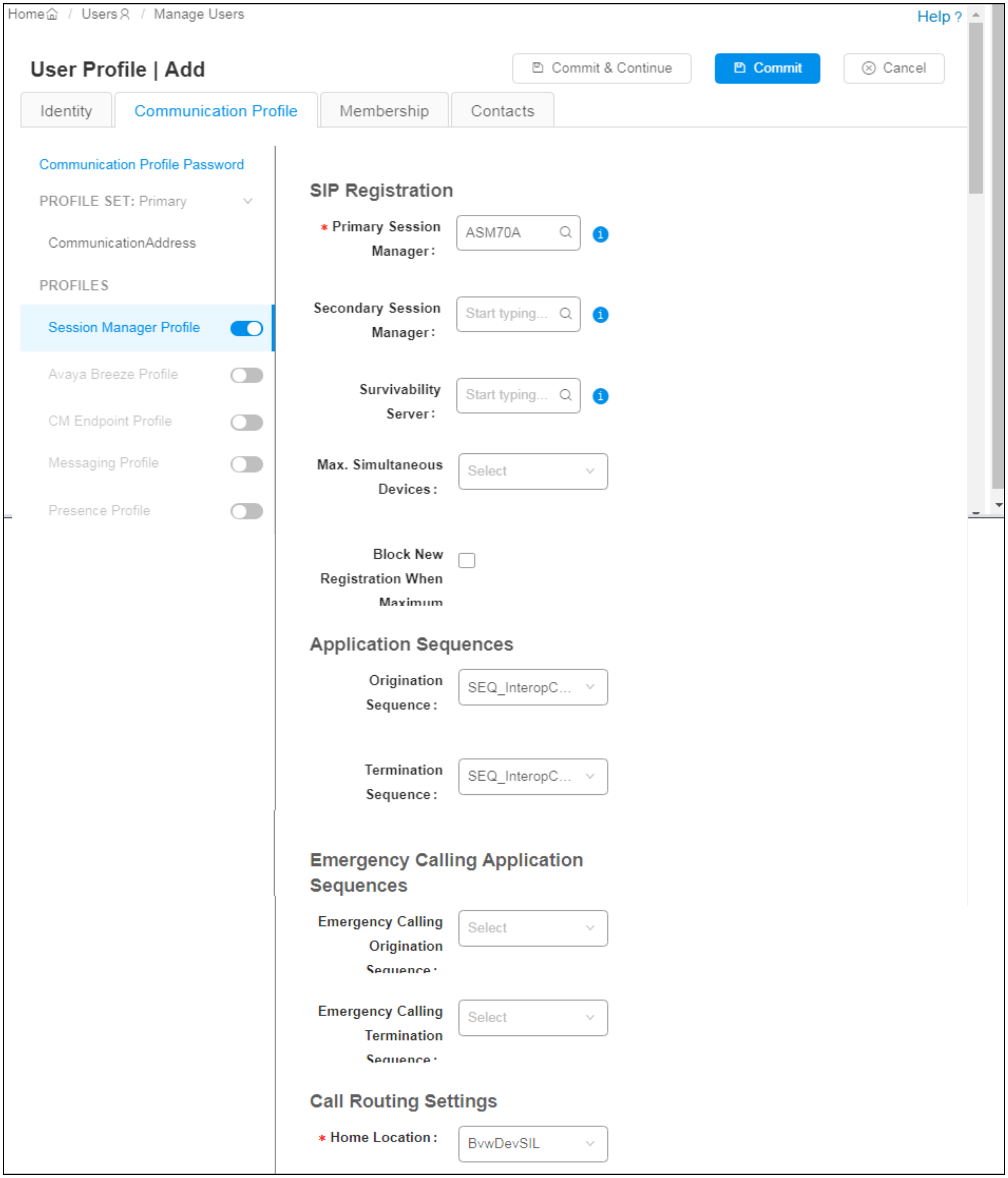
Click the **Communication Profile** tab and in the **Communication Profile Password** and **Confirm Password** fields, enter a numeric password. This will be used to register the 8028 during login.



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.

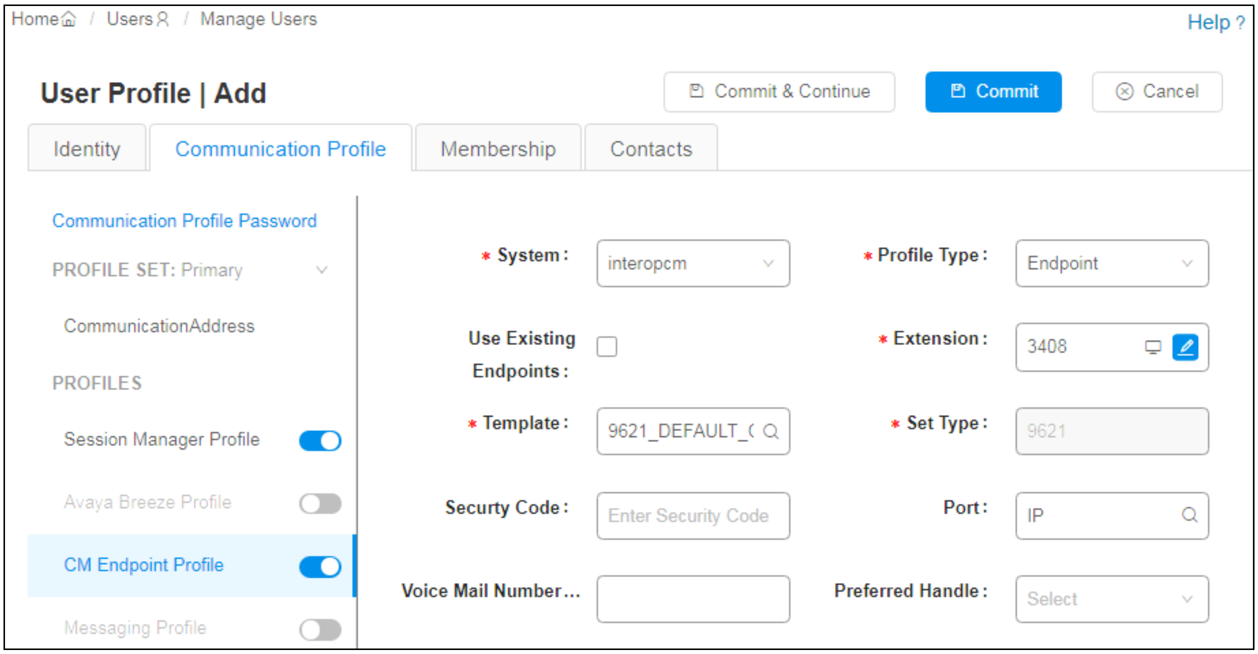


Enable the **Session Manager Profile** and configure the **Primary Session Manager, Origination Application Sequence, Termination Application Sequence** and **Home Location,** from the respective drop down lists. The Primary Session Manager used was **ASM70A**.



Enable **CM Endpoint Profile** 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 **3408**
- **Template** Select **9621SIP_DEFAULT_CM_8_0** from the drop down list
- **Port** The "IP" is auto filled out by the system



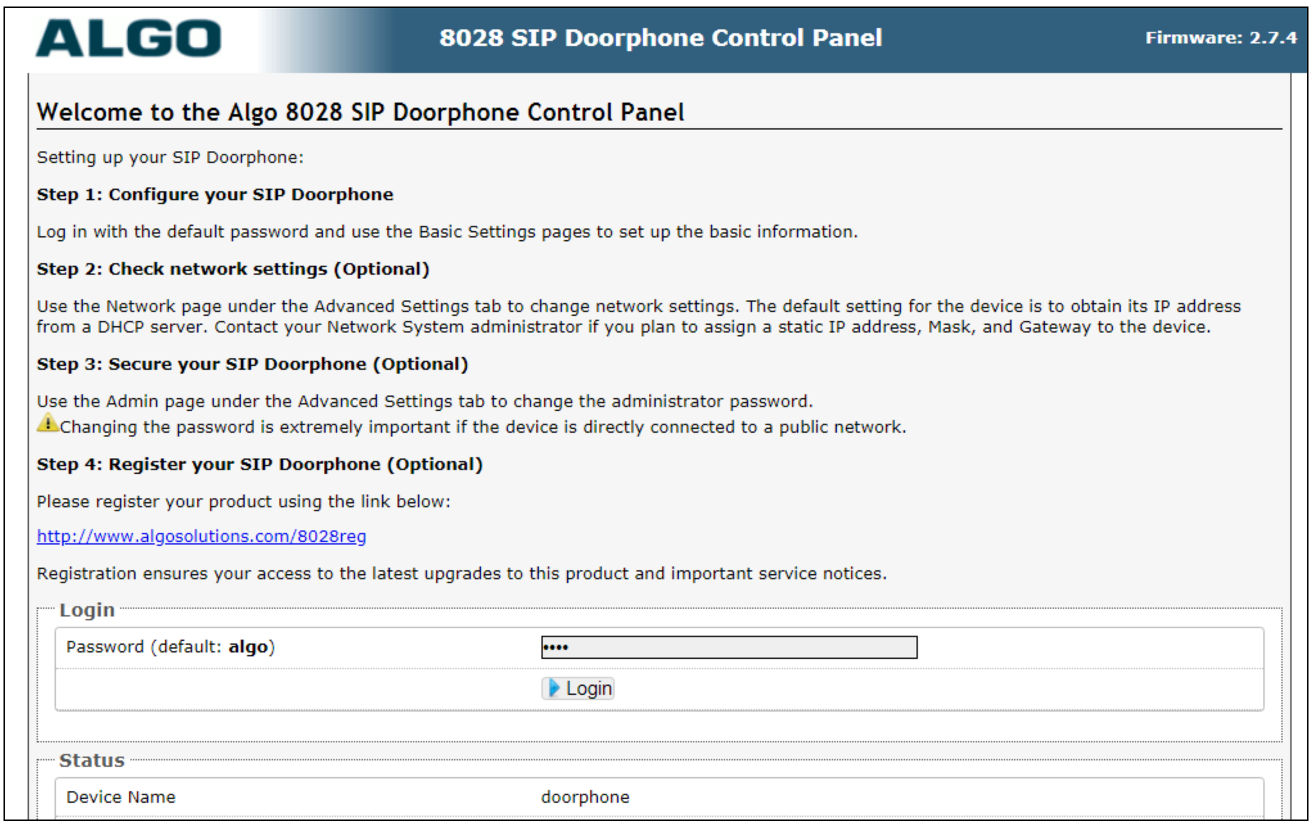
7. Configure 8028 SIP Doorphone

This section provides the procedures for configuring the 8028. The procedures include the following areas:

- Launch web interface.
- Administer configuration.

7.1. Launch Web Interface

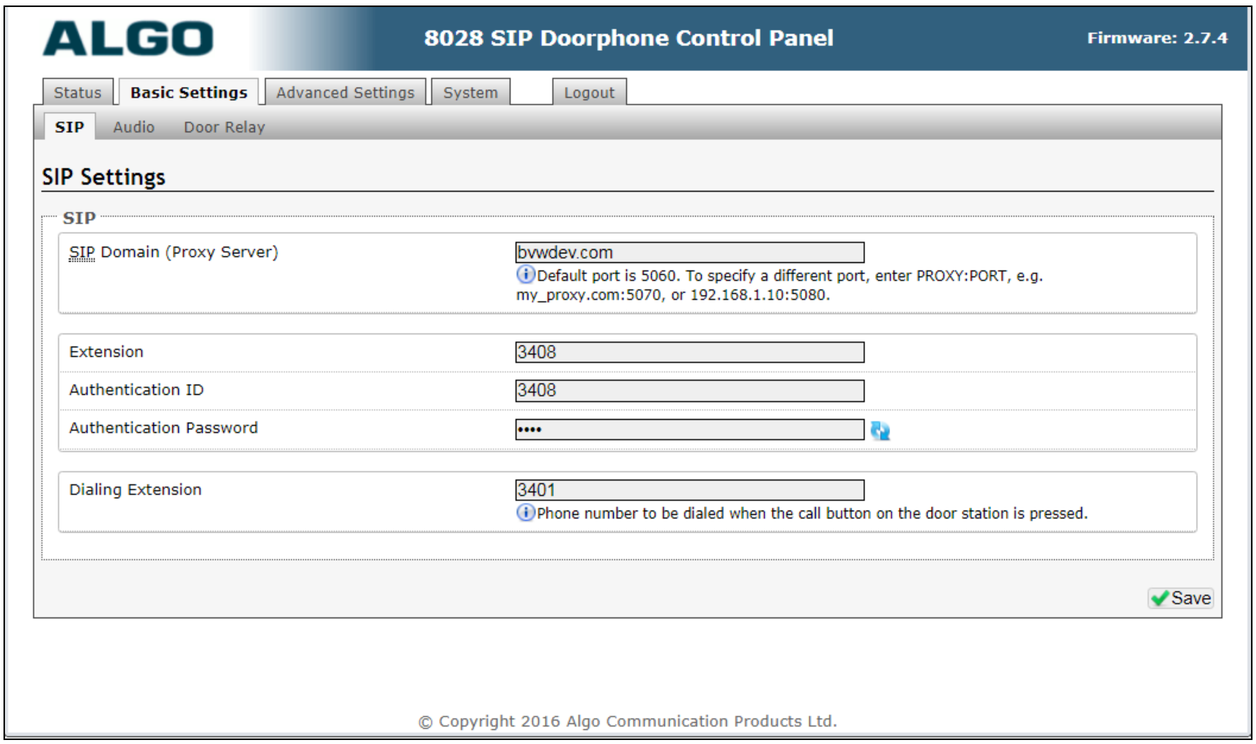
Access the 8028 SIP Doorphone web-based interface by using the URL “http://ip-address” in an Internet browser window, where “ip-address” is the IP address of the 8028 SIP Doorphone. The IP address of the 8028 can be spoken by using the call buttons on the door station of the 8028. The **Welcome to the Algo 8028 SIP Doorphone Control Panel** screen is displayed, as shown below. Log in using the appropriate credentials.



7.2. Administer Algo 8028 SIP Doorphone

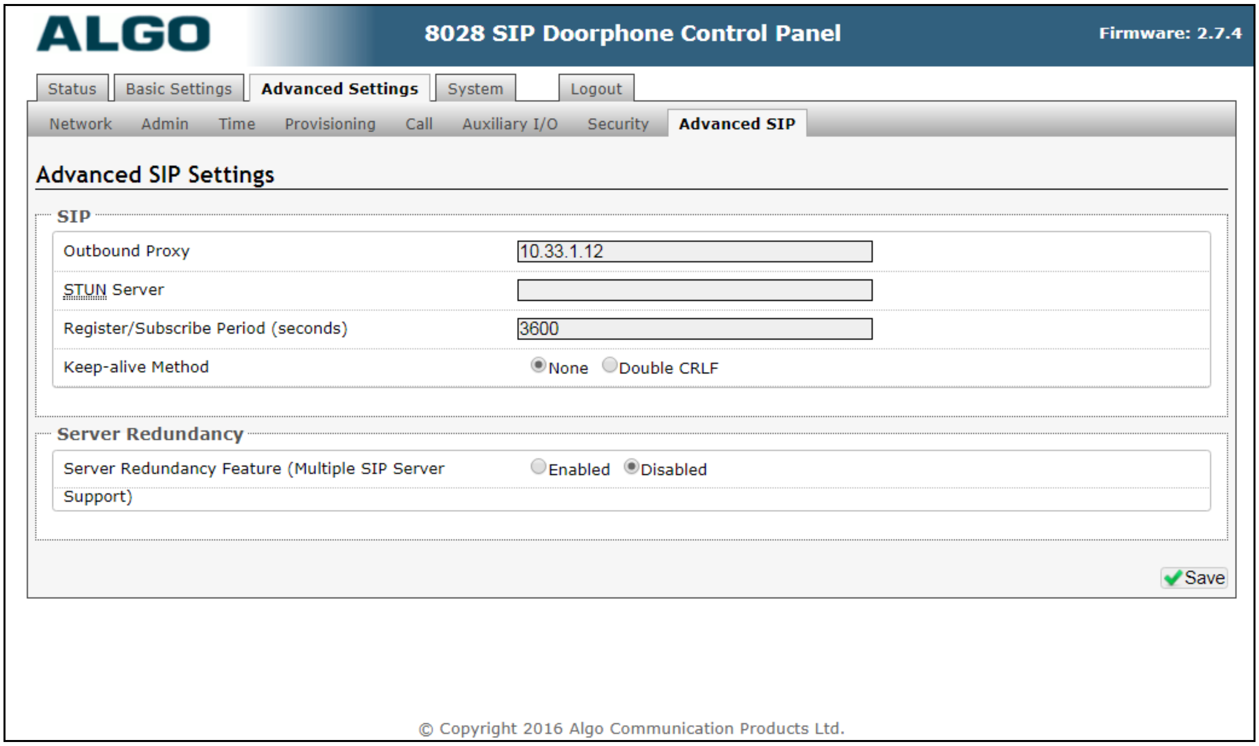
Select **Basic Settings → SIP** from the top menu, to display the screen below. Configure the **SIP Account** section toward the bottom of the screen as desired to match the configuration. Enter the following values for the specified fields, and retain the default values in the remaining fields.

- **SIP Domain (Proxy Server):** Enter the SIP domain as shown in **Section 6.1**.
- **Extension:** The SIP user extension as configured in **Section 6.2**.
- **Authentication ID:** The SIP user name from **Section 6.2**.
- **Authentication password:** The SIP user communication profile password as configured in **Section 6.2**
- **Dialing Extension:** Enter an extension on the Communication Manager system for dialing out from the call button in the 8028 Door station.



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 to save the configuration.

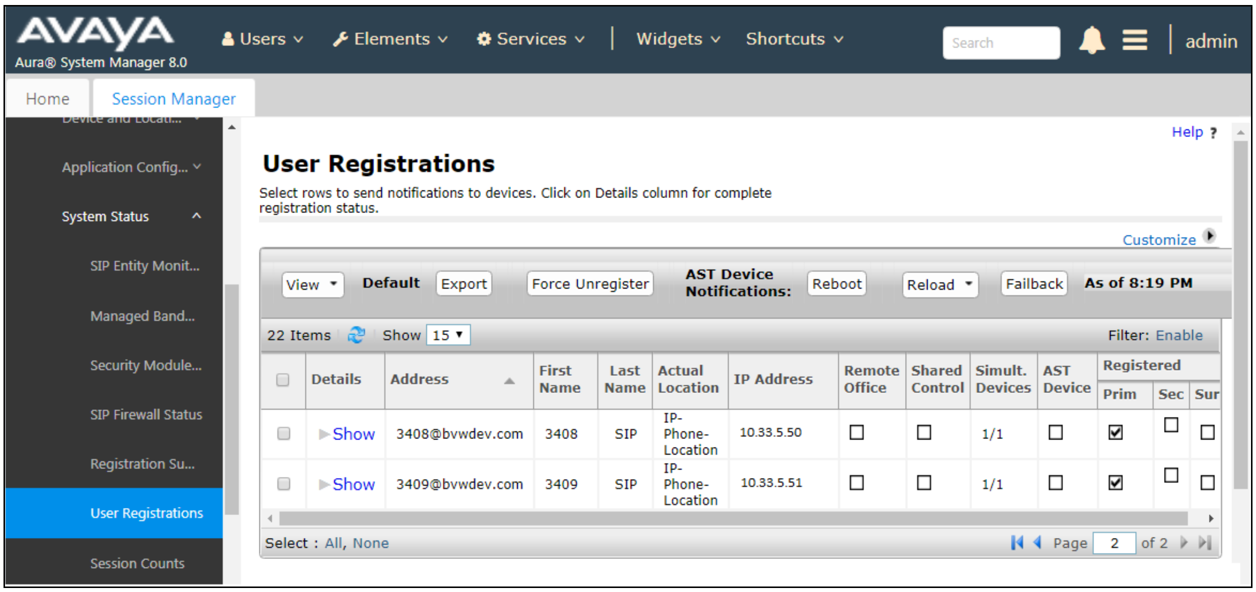


8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Aura® Session Manager and Algo 8028 SIP Doorphone.

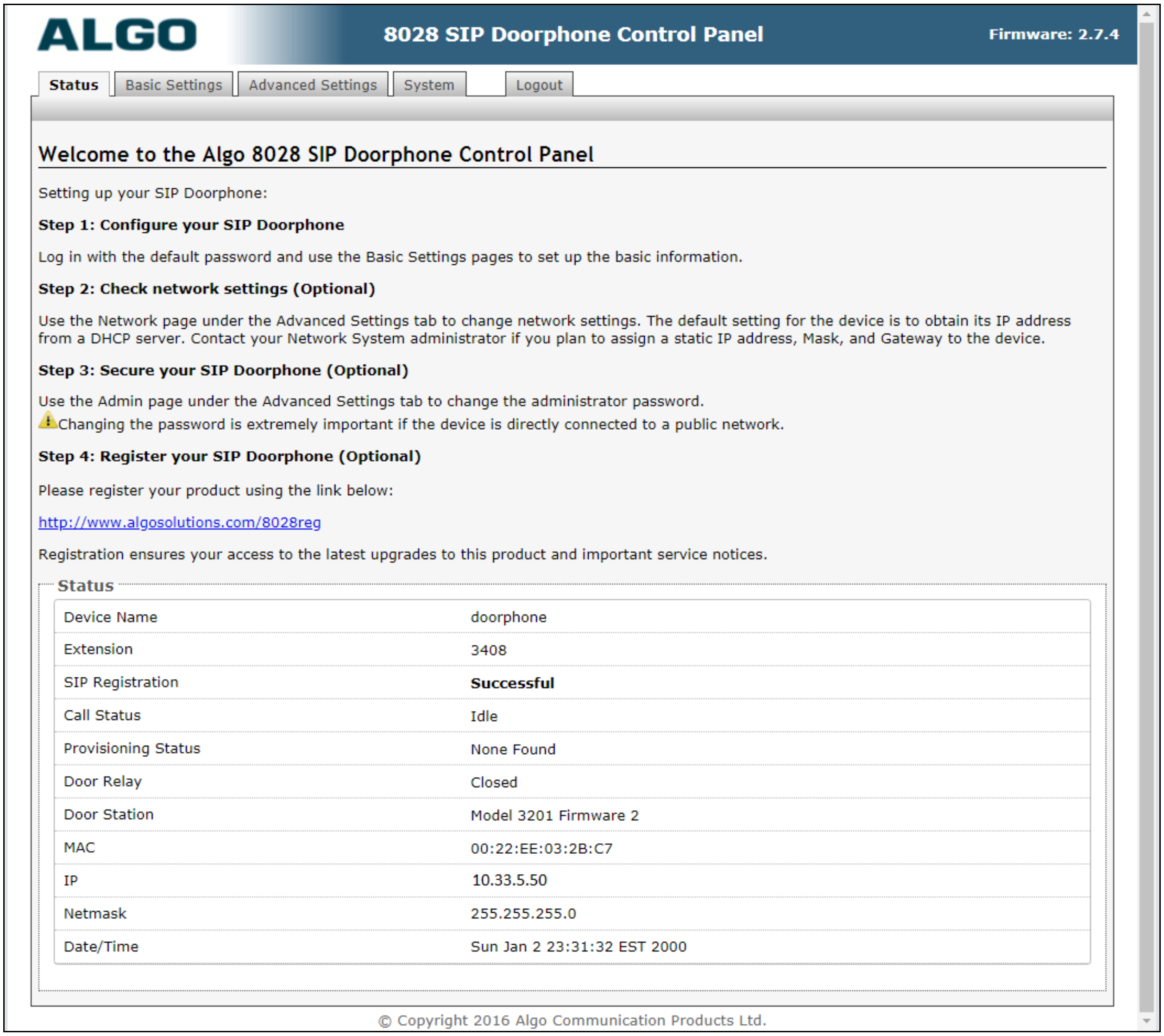
8.1. Verify Registration to Avaya Aura® Session Manager

From the System Manager dashboard select **Session Manager** from the **Elements** section (not shown). From the left hand menu select **System Status→User Registrations** (not shown). The 8028 extension is listed and a tick under **Registered** for the Session Manager as it is registered to.



8.2. Verify Algo 8028

From the 8028 web-based interface, select **Status** from the top menu. Verify that **SIP Registration** displays “Successful” as shown below.



The following tests were conducted to verify the solution between the 8028 and Session Manager.

- Verify that when the call button on the 8028 Door Station is pressed the endpoint on the Communication Manager rings and a clear speech path is established
- Verify that the solution works with different Avaya endpoints (e.g. digital, analog, IP etc.) and that DTMF tones generated from these different endpoints are able to unlock the door release
- Verify that the 8028 goes into an idle state when the call is completed
- Verify that the 8028 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 8028 SIP Doorphone with Avaya Aura® Session Manager and Avaya Aura® Communication Manger. All of the executed test cases have passed and met the objectives outlined in **Section 2.1**, with some exceptions outlined in **Section 2.2**.

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 8.0, August 2018, Document Number 03-300509, Issue 1.
[2] Avaya Aura® Communication Manager Feature Description and Implementation, Release 8.0, August 2018, Document Number 555-245-205, Issue 1.
[3] Administering Avaya Aura® Session Manager, Release 8.0, Issue 1 August 2018
[4] Administering Avaya Aura® System Manager, Release 8.0, Issue 1, August, 2018

Product documentation for the Algo 8028 SIP Doorphone products may be found at:
http://www.algosolutions.com/8028

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.