



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.

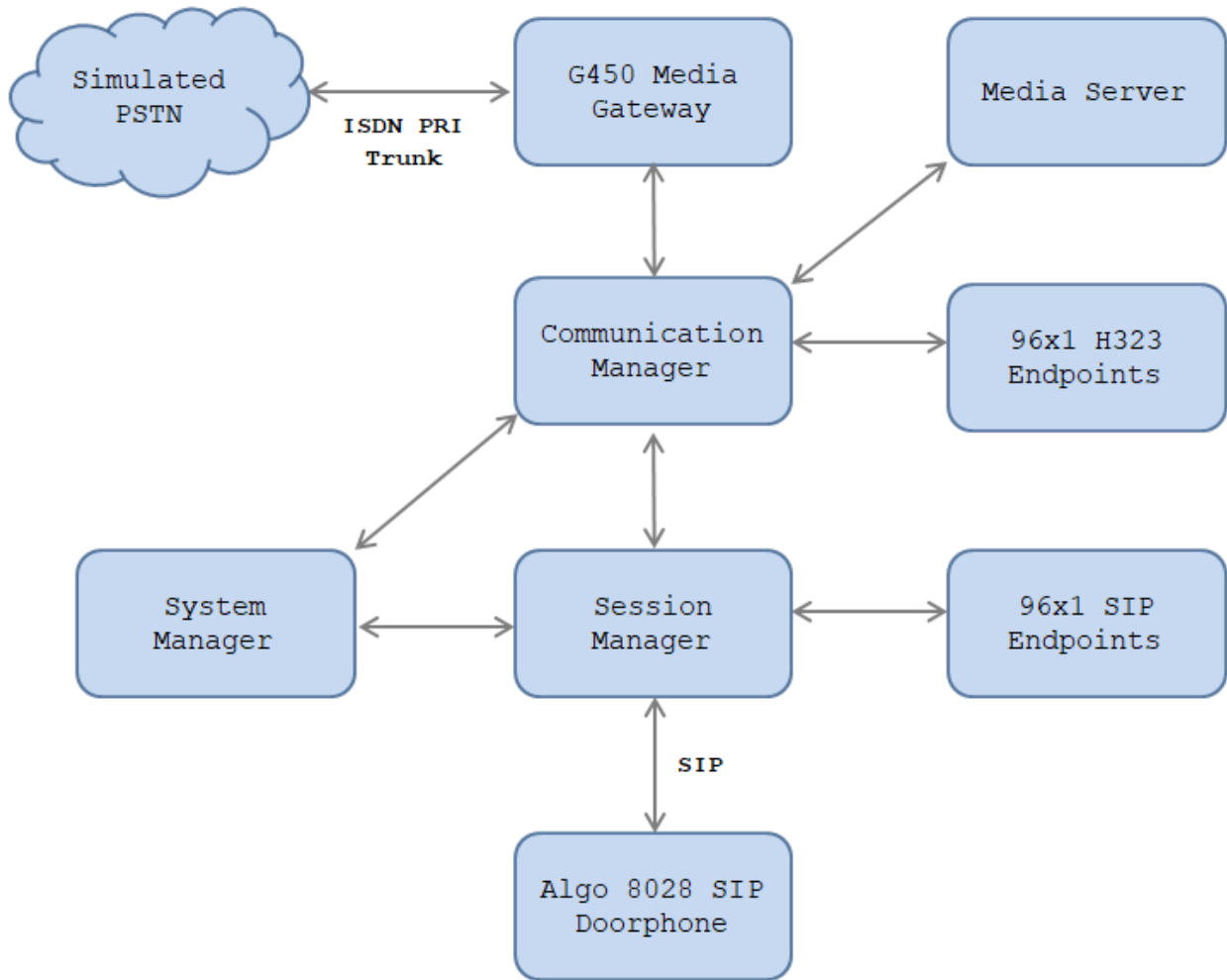


Figure 1: Test Configuration Diagram

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	2.7.4
Kernel Version	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
10
                                                    Page 2 of
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

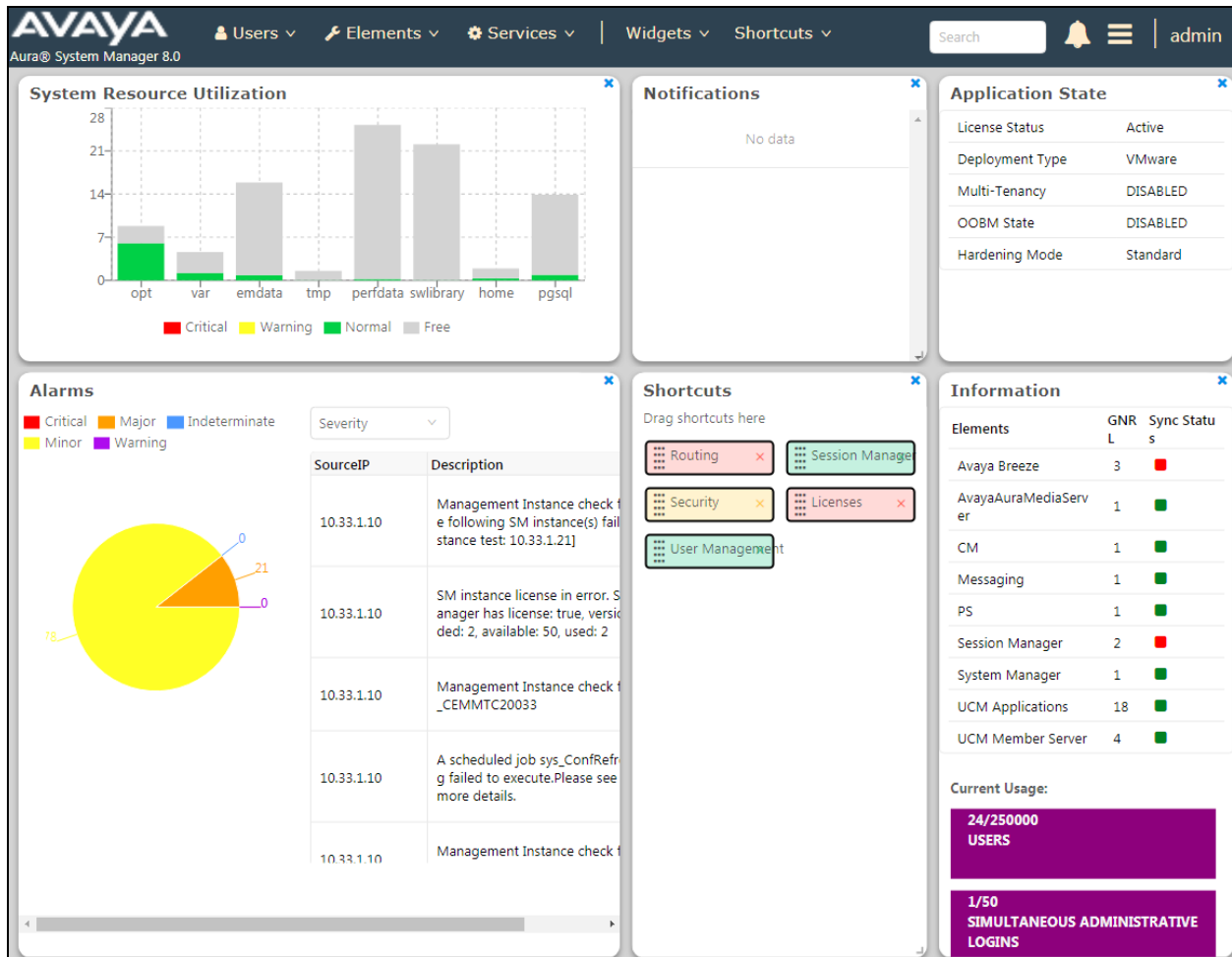
Dialed   Total Call   Dialed   Total Call   Dialed   Total Call
String   Length Type      String   Length Type      String   Length Type
33      4  ext
34      4  ext
*         3   fac
#         3   fac

```

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.

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

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

Home / Users / Manage Users Help ?

User Profile | Add

[Commit & Continue](#) [Commit](#) [Cancel](#)

[Identity](#) [Communication Profile](#) [Membership](#) [Contacts](#)

Basic Info

Address

LocalizedName

User Provisioning Rule:

* Last Name: Last Name (Latin Translation):

* First Name: First Name (Latin Translation):

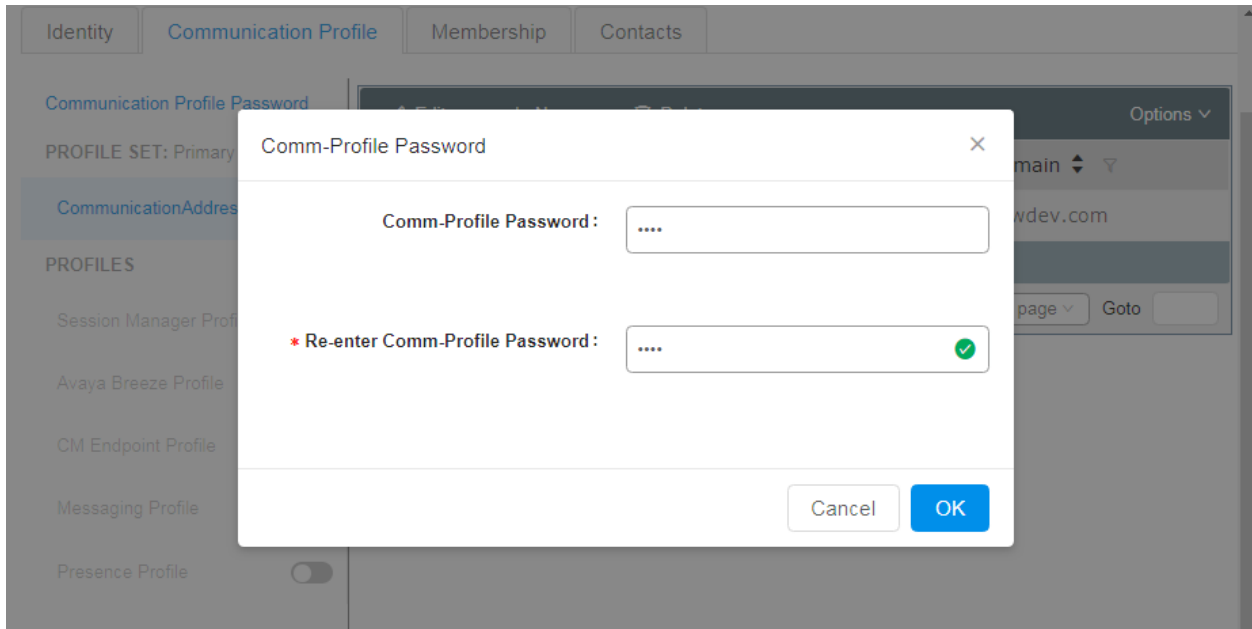
* Login Name: Middle Name:

Description: Email Address:

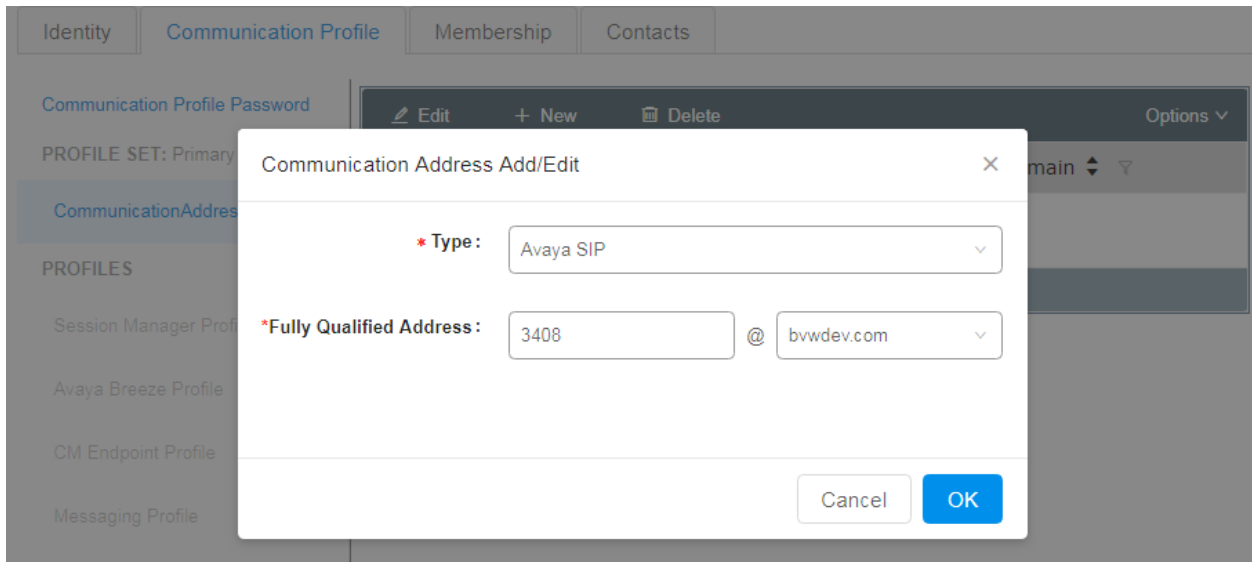
Password: User Type:

* Confirm Password: Localized Display Name:

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**.

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: ASM70A

Secondary Session Manager: Start typing...

Survivability Server: Start typing...

Max. Simultaneous Devices: Select

Block New Registration When Maximum

Application Sequences

Origination Sequence: SEQ_InteropC...

Termination Sequence: SEQ_InteropC...

Emergency Calling Application Sequences

Emergency Calling Origination Sequence: Select

Emergency Calling Termination Sequence: Select

Call Routing Settings

* Home Location: BvwDevSIL

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

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

* **System**: interopcm

* **Profile Type**: Endpoint

Use Existing Endpoints:

* **Extension**: 3408

* **Template**: 9621_DEFAULT_CM_8_0

* **Set Type**: 9621

Security Code: Enter Security Code

Port: IP

Voice Mail Number...

Preferred Handle: Select

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.

ALGO 8028 SIP Doorphone Control Panel Firmware: 2.7.4

Welcome to the Algo 8028 SIP Doorphone Control Panel

Setting up your SIP Doorphone:

Step 1: Configure your SIP Doorphone

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 Doorphone (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 Doorphone (Optional)

Please register your product using the link below:
<http://www.algosolutions.com/8028reg>

Registration ensures your access to the latest upgrades to this product and important service notices.

Login

Password (default: algo)

Status

Device Name	doorphone
-------------	-----------

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.

The screenshot displays the '8028 SIP Doorphone Control Panel' interface. At the top, the 'ALGO' logo is on the left, '8028 SIP Doorphone Control Panel' is in the center, and 'Firmware: 2.7.4' is on the right. Below the header is a navigation bar with tabs for 'Status', 'Basic Settings', 'Advanced Settings', 'System', and 'Logout'. The 'Basic Settings' tab is active, and within it, the 'SIP' sub-tab is selected. The main content area is titled 'SIP Settings' and contains several input fields:

- SIP Domain (Proxy Server):** The input field contains 'bwdev.com'. Below it is a help icon and text: 'Default port is 5060. To specify a different port, enter PROXY:PORT, e.g. my_proxy.com:5070, or 192.168.1.10:5080.'
- Extension:** The input field contains '3408'.
- Authentication ID:** The input field contains '3408'.
- Authentication Password:** The input field contains four dots '....' and a help icon.
- Dialing Extension:** The input field contains '3401'. Below it is a help icon and text: 'Phone number to be dialed when the call button on the door station is pressed.'

At the bottom right of the settings area is a green 'Save' button with a checkmark icon. At the bottom center of the page is the copyright notice: '© Copyright 2016 Algo Communication Products Ltd.'

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.

The screenshot displays the ALGO 8028 SIP Doorphone Control Panel interface. The top navigation bar includes 'Status', 'Basic Settings', 'Advanced Settings' (selected), 'System', and 'Logout'. Below this, a secondary menu shows 'Network', 'Admin', 'Time', 'Provisioning', 'Call', 'Auxiliary I/O', 'Security', and 'Advanced SIP' (selected). The main content area is titled 'Advanced SIP Settings' and is divided into two sections: 'SIP' and 'Server Redundancy'. In the 'SIP' section, the 'Outbound Proxy' field is set to '10.33.1.12', 'STUN Server' is empty, 'Register/Subscribe Period (seconds)' is '3600', and 'Keep-alive Method' is set to 'None'. The 'Server Redundancy' section shows 'Server Redundancy Feature (Multiple SIP Server Support)' set to 'Disabled'. A green 'Save' button is located at the bottom right of the settings area. The footer contains the copyright notice: '© Copyright 2016 Algo Communication Products Ltd.'

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

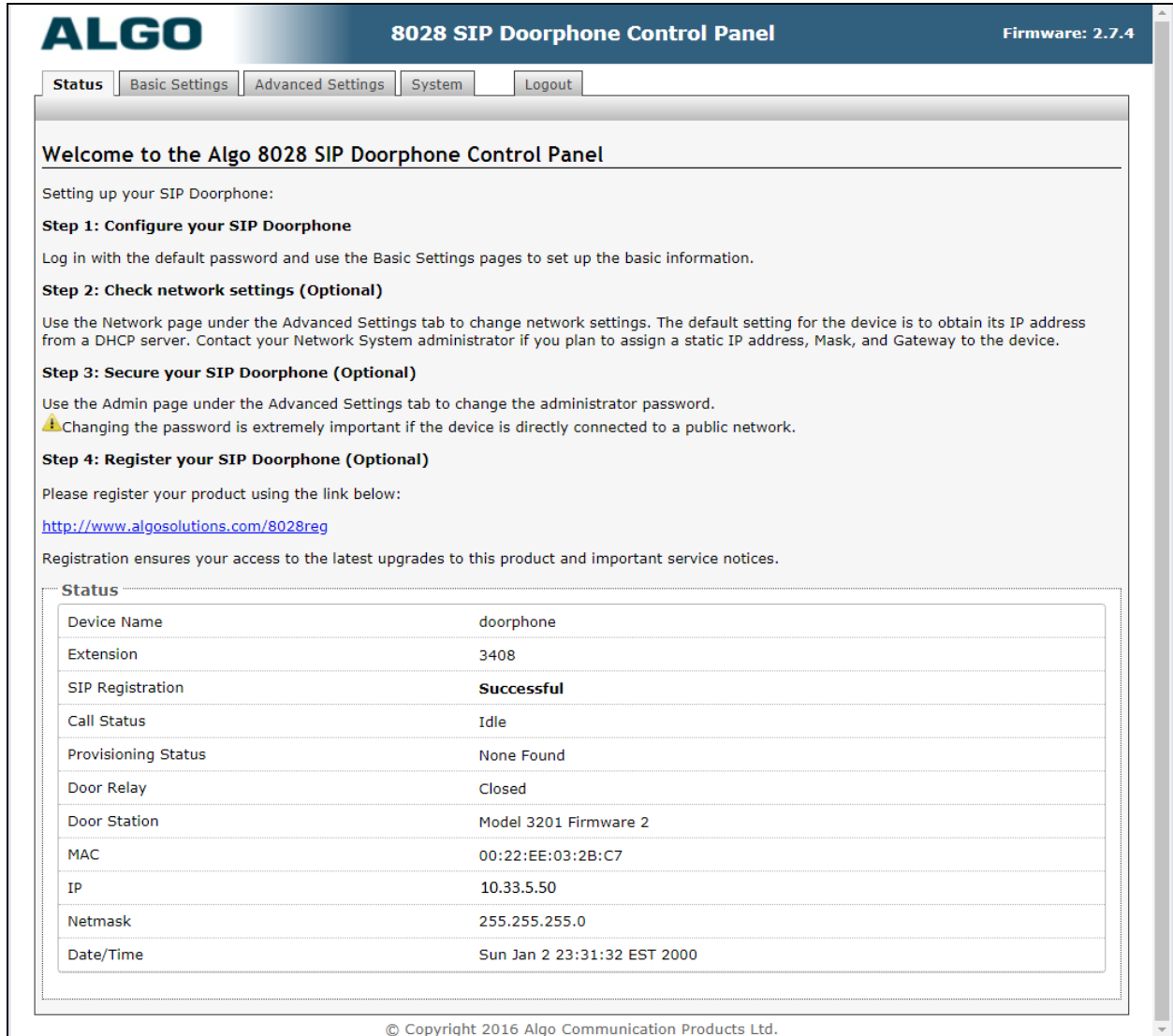
From the System Manager dashboard select **Session Manager** from the **Elements** section (not shown). From the left hand menu select **System Status**→**User Registrations** (not shown). The 8028 extension is listed and a tick under **Registered** for the Session Manager as it is registered to.

The screenshot displays the Avaya Aura System Manager 8.0 interface. The main content area is titled "User Registrations" and includes a table of user registration data. The table has the following columns: Details, Address, First Name, Last Name, Actual Location, IP Address, Remote Office, Shared Control, Simult. Devices, AST Device, and Registered (Prim, Sec, Sur). Two rows are visible in the table, both with the "Registered" checkbox checked.

	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered		
											Prim	Sec	Sur
<input type="checkbox"/>	Show	3408@bvwddev.com	3408	SIP	IP-Phone-Location	10.33.5.50	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	3409@bvwddev.com	3409	SIP	IP-Phone-Location	10.33.5.51	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

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.



ALGO 8028 SIP Doorphone Control Panel Firmware: 2.7.4

Status Basic Settings Advanced Settings System Logout

Welcome to the Algo 8028 SIP Doorphone Control Panel

Setting up your SIP Doorphone:

Step 1: Configure your SIP Doorphone
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 Doorphone (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 Doorphone (Optional)
Please register your product using the link below:
<http://www.algosolutions.com/8028reg>
Registration ensures your access to the latest upgrades to this product and important service notices.

Status

Device Name	doorphone
Extension	3408
SIP Registration	Successful
Call Status	Idle
Provisioning Status	None Found
Door Relay	Closed
Door Station	Model 3201 Firmware 2
MAC	00:22:EE:03:2B:C7
IP	10.33.5.50
Netmask	255.255.255.0
Date/Time	Sun Jan 2 23:31:32 EST 2000

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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>

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