



Avaya Solution & Interoperability Test Lab

Application Notes for Aiphone IX Series Audio Door Stations (IX-SPMIC) R5.4 and Avaya Aura[®] Communication Manager and Avaya Aura[®] Session Manager R8.1 – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Aiphone IX Series Audio Door Stations (IX-SPMIC) which was compliance tested with Avaya Aura[®] Communication Manager and Avaya Aura[®] Session Manager.

The overall objective of the interoperability compliance testing was to verify Aiphone IX Series Audio Door Stations (IX-SPMIC) functionalities in an environment comprised of Avaya Aura[®] and various Avaya endpoints. Aiphone IX Series Audio Door Stations are SIP based door phones.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any 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 Aiphone IX Series Audio Door Stations to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. During the compliance testing, Aiphone IX-SPMIC was used.

The Aiphone IX Series Audio Door Stations (IX-SPMIC) are part of Aiphone IX Series Door Stations. The Audio Door Stations, IX-SPMIC, act as SIP phones when connected to Avaya Aura®. The Audio Door Stations come in both surface mount and flush mount varieties. All door stations have dry contacts that can be used to release doors when activated by another intercom or phone. The dry contacts can also be used to trigger external signaling devices, such as strobes.

Calls from Aiphone IX-SPMIC are originated and terminated via a URL. Aiphone IX-SPMIC cannot receive calls.

During the compliance test, Aiphone IX-SPMIC registered as a 3rd party SIP phone using UDP to Avaya Aura® Session Manager.

2. General Test Approach and Test Results

The focus of this interoperability compliance testing was to verify that the Aiphone IX-SPMIC can register as a SIP endpoint on Session Manager, and is able to originate and receive audio calls to and from the Avaya Aura® system.

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 Aiphone did not utilize secure capabilities.

2.1. Interoperability Compliance Testing

The general test approach was to place calls from Aiphone IX-SPMIC, and exercise basic telephone operations. The main objectives were to verify the following:

- Registration
- Calls to Avaya SIP Audio endpoints
- Calls to Avaya H.323 Audio endpoints
- Calls to Avaya Digital & Analog endpoints
- Calls to PSTN via SIP Trunks
- Call termination (origination/destination)
- Serviceability

2.2. Test Results

The test objectives were verified, and the features tested worked as expected.

2.3. Support

For technical support on Aiphone IX-SPMIC, please contact Aiphone via the following:

- Web: <https://www.aiphone.co.jp/>
- Phone: 052-228-9961

3. Reference Configuration

Figure 1 illustrates a sample configuration consisting of Avaya Aura® components and Aiphone IX-SPMIC.

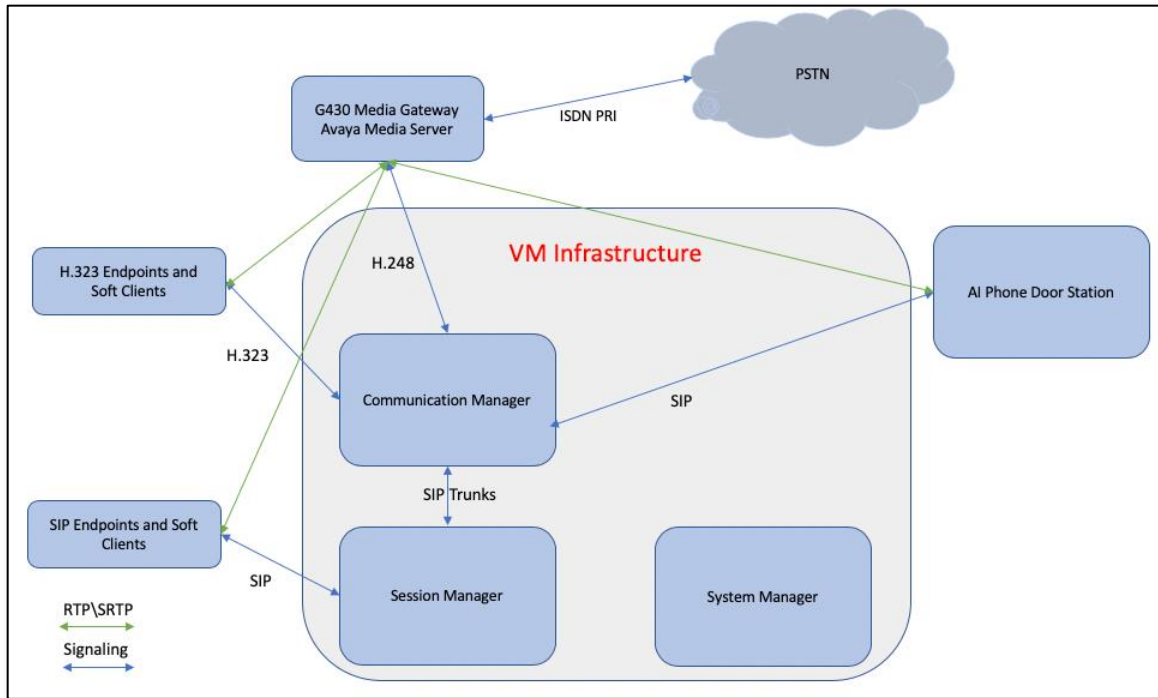


Figure 1: Test Configuration of Aiphone IX-SPMIC with Avaya Aura®

4. Equipment and Software Validated

The following equipment and software were used for the test configuration.

Equipment	Software/Firmware
Avaya Aura® Communication Manager	8.1.1.0.0.890.25763 (FP1)
Avaya Aura® Session Manager	8.1.1.0.811021
Avaya Aura® System Manager	8.1.1.0.0310782 (FP1)
Avaya 9600 Series H.323 IP Deskphones	6.8304
Avaya J129 SIP Phone	4.0.4.0.10
Avaya IX Workspace	3.7.0.102.3
Avaya H175 Collaboration Station	1.0.2.3
Avaya Vantage K175 Phone	3.5.0
Avaya 9504 Digital Phone	0.55
Avaya 6210 Analogue Telephone	-
Aiphone IX Series Audio Door Station IX-SPMIC	5.40

5. Configure Avaya Aura® Communication Manager

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

- Verify System Capacity (License)
- Define Dial Plan

These steps were performed using an SSH Terminal session.

5.1. Verify System Capacity (License)

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 12
OPTIONAL FEATURES				
G3 Version: V18		Software Package: Enterprise		
Location: 2		System ID (SID): 1		
Platform: 28		Module ID (MID): 1		
			USED	
Platform Maximum Ports:			48000	73
Maximum Stations:			36000	48
Maximum XMOBILE Stations:			36000	0
Maximum Off-PBX Telephones - EC500:			41000	0
Maximum Off-PBX Telephones - OPS:			41000	27
Maximum Off-PBX Telephones - PBFMC:			41000	0
Maximum Off-PBX Telephones - PVFMC:			41000	0
Maximum Off-PBX Telephones - SCCAN:			0	0
Maximum Survivable Processors:			313	0
(NOTE: You must logoff & login to effect the permission changes.)				

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 12
OPTIONAL FEATURES			
IP PORT CAPACITIES		USED	
Maximum Administered H.323 Trunks:		12000	0
Maximum Concurrently Registered IP Stations:		2400	3
Maximum Administered Remote Office Trunks:		12000	0
Max Concurrently Registered Remote Office Stations:		2400	0
Maximum Concurrently Registered IP eCons:		128	0
Max Concur Reg Unauthenticated H.323 Stations:		100	0
Maximum Video Capable Stations:		36000	0
Maximum Video Capable IP Softphones:		2400	16
Maximum Administered SIP Trunks:		12000	10
Max Administered Ad-hoc Video Conferencing Ports:		12000	0
Max Number of DS1 Boards with Echo Cancellation:		688	0

5.2. Define the Dial Plan

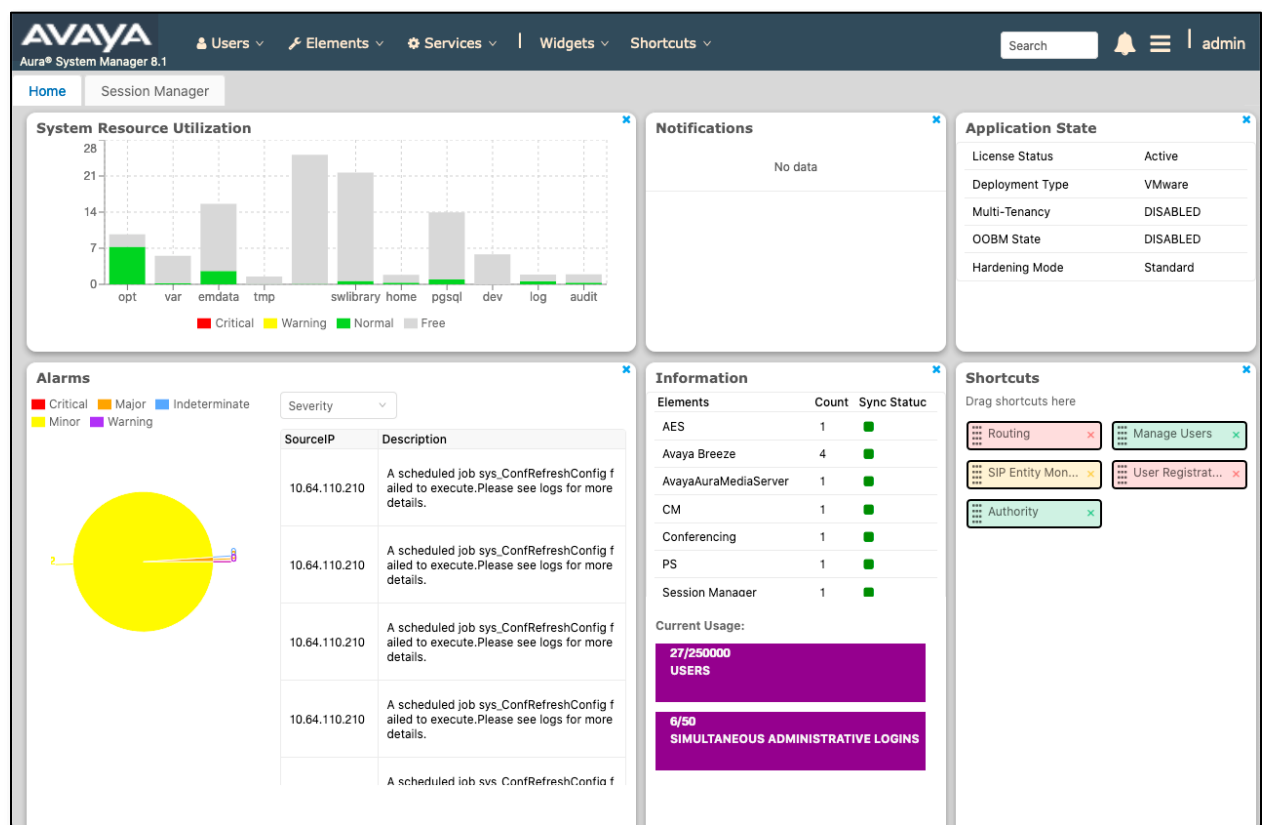
Use the **change dialplan analysis** command to define the dial plan used in the system. This includes all telephone extensions. In the sample configuration, telephone extensions are 5 digits long and begin with 7.

change dialplan analysis						Page 1 of 12					
DIAL PLAN ANALYSIS TABLE											
Location: all						Percent Full: 1					
Dialed String			Total Call Length Type			Dialed String			Total Call Length Type		
1			3 <u>dac</u>								
2			5 <u>ext</u>								
3			5 <u>ext</u>								
4			5 <u>aar</u>								
7			5 <u>ext</u>								
8			1 <u>fac</u>								
9			1 <u>fac</u>								
*			3 <u>fac</u>								
#			3 <u>fac</u>								

6. Configure Avaya Aura® Session Manager

This section describes aspects of the Session Manager configuration required for interoperating with Aiphone IX-SPMIC. 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.

Listen Ports

AddRemove

4 ItemsFilter: Enable

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

Select : All, None

6.2. Add a SIP User

A SIP user must be added for Aiphone IX-SPMIC. 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 **72005@avaya.com**

Home / Users / Manage Users Help ?

User Profile | Edit | 72005@avaya.com Commit & Continue Commit Cancel

Identity | Communication Profile | Membership | Contacts

Basic Info

Address

LocalizedName

User Provisioning Rule:

* Last Name: Last Name (in Latin alphabet characters):

* First Name: First Name (in Latin alphabet characters):

* Login Name: Middle Name:

Description: Email Address:

Password: User Type:

Confirm Password: Localized Display Name:

Endpoint Display Name: Title Of User:

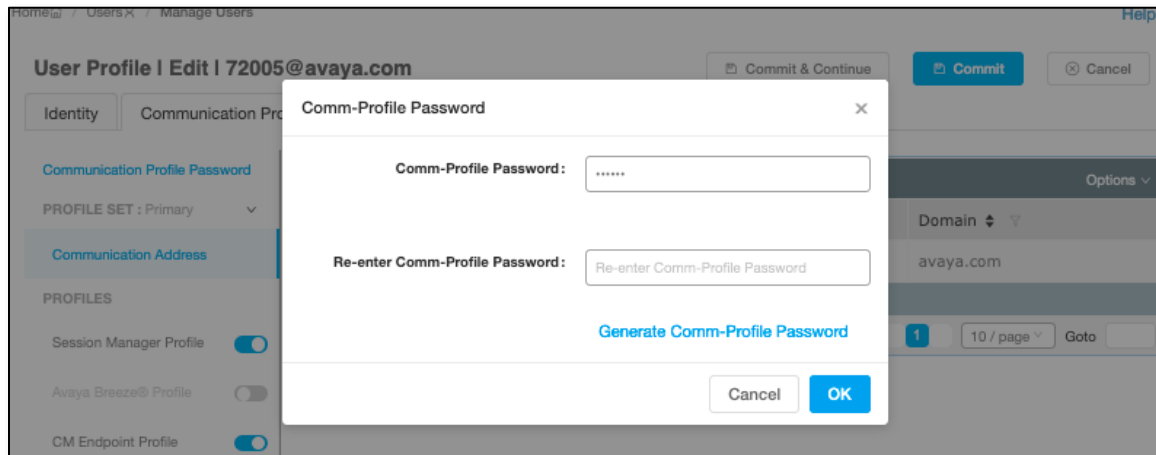
Language Preference: Time Zone:

Employee ID: Department:

Company:

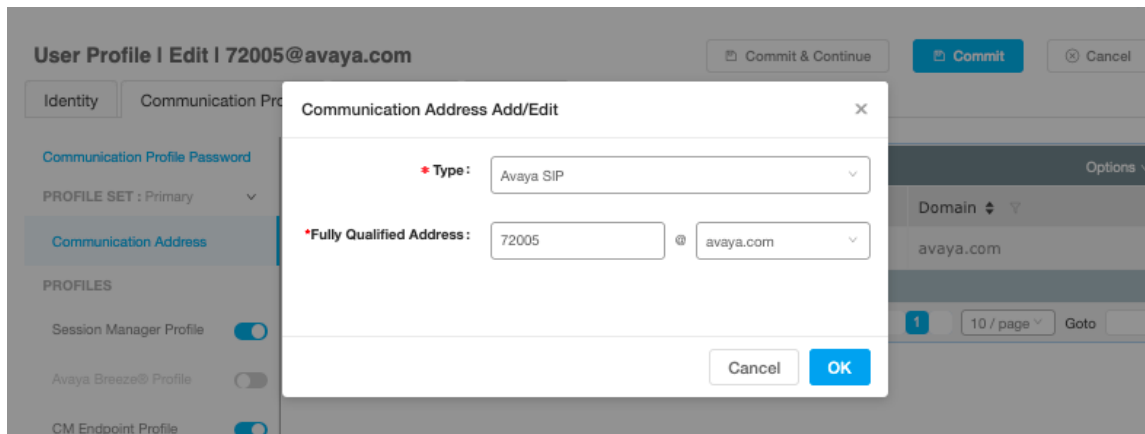
Note in this and subsequent steps, press **Commit & Continue** after making entries or selections.

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 device during login.



The screenshot shows the 'User Profile | Edit | 72005@avaya.com' page with the 'Communication Profile' tab selected. A modal dialog titled 'Comm-Profile Password' is open. It contains two text input fields: 'Comm-Profile Password' and 'Re-enter Comm-Profile Password'. Below these fields is a blue link labeled 'Generate Comm-Profile Password'. At the bottom of the dialog are 'Cancel' and 'OK' buttons. The background interface shows the 'Communication Address' section with a dropdown for 'Domain' set to 'avaya.com'.

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.



The screenshot shows the 'User Profile | Edit | 72005@avaya.com' page with the 'Communication Address' section selected. A modal dialog titled 'Communication Address Add/Edit' is open. It contains a dropdown menu for 'Type' with 'Avaya SIP' selected. Below it is a 'Fully Qualified Address' field with '72005' entered, followed by an '@' symbol and a dropdown menu for the domain with 'avaya.com' selected. At the bottom of the dialog are 'Cancel' and 'OK' buttons. The background interface shows the 'Communication Address' section with a dropdown for 'Domain' set to 'avaya.com'.

Click on the **Session Manager Profile** link and configure the **Primary Session Manager**, **Max Simultaneous Devices**, **Origination Application Sequence**, **Termination Application Sequence** and **Home Location**, from the respective drop-down lists.

User Profile | Edit | 72005@avaya.com

Commit & Continue

Commit

Cancel

Identity

Communication Profile

Membership

Contacts

Communication Profile Password

PROFILE SET : Primary

Communication Address

PROFILES

Session Manager Profile

Avaya Breeze® Profile

CM Endpoint Profile

Presence Profile

SIP Registration

Primary Session Manager

sm81

Secondary Session Manager

Start typing...

Survivability Server

Start typing...

Max. Simultaneous Devices

2

Block New Registration When Maximum Registrations Active?

Application Sequences

Origination Sequence

cm81

Termination Sequence

cm81

Emergency Calling Application Sequences

Emergency Calling Origination Sequence

Select

Emergency Calling Termination Sequence

Select

Call Routing Settings

Home Location

DevConnect

Click the **CM Endpoint Profile** link 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 **72005**
- **Template** - Select **J129_DEFAULT_CM_8_1** from the drop-down list
- **Port** - The “IP” is auto filled out by the system

Click on **Endpoint Editor** in the Extension field to edit Communication Manager settings if desired.

The screenshot displays the 'User Profile | Edit | 72005@avaya.com' interface. The 'Communication Profile' tab is active. On the left, a sidebar lists profile types: 'Communication Profile Password', 'PROFILE SET : Primary', 'Communication Address', 'PROFILES', 'Session Manager Profile' (checked), 'Avaya Breeze® Profile' (unchecked), 'CM Endpoint Profile' (checked), and 'Presence Profile' (unchecked). The main area contains various configuration fields: 'System' (cm81), 'Profile Type' (Endpoint), 'Extension' (72005), 'Set Type' (J129), 'Port' (S000092), 'Template' (Start typing...), 'Security Code' (Enter Security Code), 'Voice Mail Number', 'Preferred Handle' (72005@avaya.com), 'Sip Trunk' (aar), 'SIP URI' (72005@avaya.com), 'Enhanced Callr-Info Display for 1-line phones' (unchecked), 'Delete on Unassign from User or on Delete User' (checked), 'Override Endpoint Name and Localized Name' (checked), and 'Allow H.323 and SIP Endpoint Dual Registration' (unchecked). At the top right, there are buttons for 'Commit & Continue', 'Commit', and 'Cancel'.

7. Configure Aiphone IX Series Audio Door Station

This section provides steps to configure Aiphone IX-SPMIC.

To configure Aiphone IX-SPMIC, using a web browser, navigate to <https://<IP Address of IX-SPMIC>/webset.cgi?login> and log in using appropriate credentials.

AIPHONE IX System

ID:

Enter ID and password

Password:

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Once logged in, for the **Number** field, type in the SIP extension that is being configured (from **Section 6.2**), and a desired **Name**. Select **Update** to save changes.

The screenshot shows the 'Station Information' page for an 'Audio Sub Station' of type 'IX-SPMIC'. The left sidebar contains links for 'Station Information' (Identification, ID and Password, Language, Time, Expanded System) and 'Network Settings' (IP Address, DNS, SIP, Audio, Packet Priority). The main content area is titled 'Station Information' and includes a 'Required Settings' section. Under the 'Identification' sub-section, there are three fields: 'Number' (value: 72005, hint: 3-5 digits), 'Name' (value: IX-SPMIC, hint: 1-24 alphanumeric characters(*1)), and 'Location' (value: DevConnect, hint: 1-24 alphanumeric characters(*1)). A note at the bottom states: '(*1)Certain characters may not be displayed correctly on IX-MV and IX-MV7.*' An 'Update' button is in the top right corner.

From the left, select **Network Settings** → **SIP** and configure as follows:

- **SIP Signaling Port:** Set to **5060**.
- **User Agent:** Type in a desired value.
- **ID:** SIP Extension number from **Section 6.2**.
- **Password:** SIP Extension password from **Section 6.2**.
- **IPv4 Address:** LAN IP Address of Session Manager
- **Port:** Set to **5060**.

Once done, select **Update** to save changes.

The screenshot shows the 'Network Settings' page for an 'Audio Sub Station' of type 'IX-SPMIC'. The left sidebar is the same as the previous screenshot, but the 'SIP' option under 'Network Settings' is selected. The main content area is titled 'Network Settings' and includes a 'SIP' sub-section. Under 'SIP Connections', there are two fields: 'SIP Signaling Port' (value: 5060, hint: 1-65535) and 'User Agent' (value: IX-SPMIC, hint: 1-36 alphanumeric characters). Under 'SIP Server', there are five fields: 'Primary Server ID' (value: 72005, hint: 1-24 alphanumeric characters), 'Password' (value: *****, hint: 1-24 alphanumeric characters), 'IPv4 Address' (value: 10.64.110.212, hint: 1.0.0.1-223.255.255.254 or host), 'IPv6 Address' (value: ::FF:0-FFFF:FFFF:FFFF:FFFF, hint: ::FF:0-FFFF:FFFF:FFFF:FFFF), and 'Port' (value: 5060, hint: 1-65535). An 'Update' button is in the top right corner.

From the left, select **Call Settings** → **Called Stations** and configure as follows:
The numbers configured here will be dialed when the button on the IX-SPMIC is pressed.

- **Station Number:** Type in an extension number that will be called for a given line.
- **IPv4:** Type in the LAN IP Address for Session Manager.

Select **Update** to save changes.

AIPHONE IX System Setting
Category: Audio Sub Station Station Type: IX-SPMIC

Call Settings

•Called Stations (for Door)

Option Input #:

Station Number must be 3-5 digits. (3-32 digits for VoIP Phone)
IPv4 must be 1.0.0.1-223.255.255.254 or hostname(1-64 alphanumeric characters).
IPv6 must be ::FF:0:FEFF:FFFF:FFFF:FFFF:FFFF:FFFF or hostname(1-64 alphanumeric characters).
Enter SIP Primary Server IP address for VoIP Phone, set only one VoIP Phone per call group.
Station Type must be "VoIP Phone" when calling via SIP server.
U = Unicast, M = Multicast

#	Station Number	IPv4 Address	IPv6 Address
1	70103	10.64.110.212	
2			

From the left, select **Function Settings** → **CGI** and select the **Enable** radio button for **CGI Functionality**. This enables Aiphone IX-SPMIC to place calls via a CGI URL.

AIPHONE IX System Setting
Category: Audio Sub Station Station Type: IX-SPMIC

Function Settings

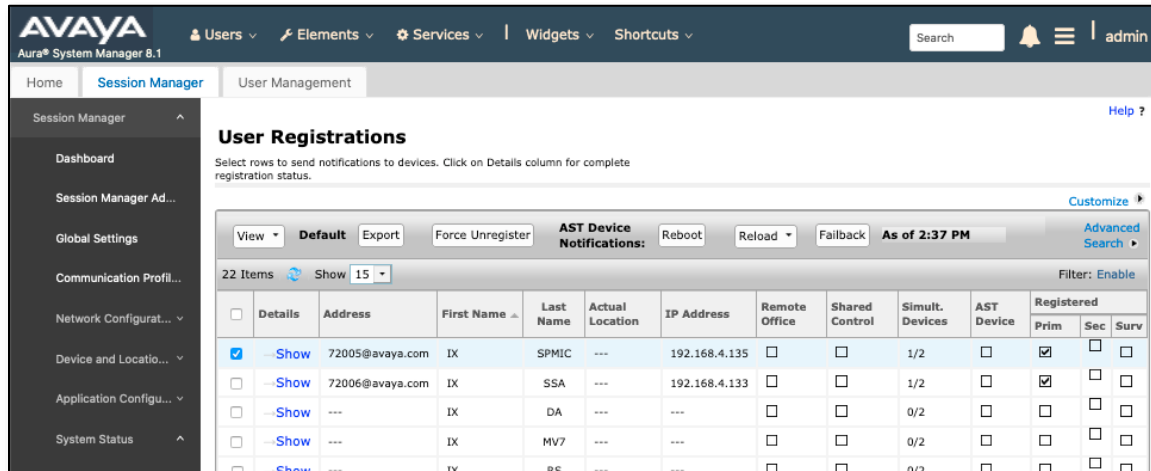
•CGI

CGI Functionality ☒ Enable ☐ Disable

8. Verification Steps

The following steps may be used to verify the configuration:

- In the System Manager web interface, navigate to Elements → Session Manager → System Status → User Registrations to confirm successful registration.



Place a call from Aiphone IX-SPMIC to an Avaya endpoint. The state of the call be viewed on Communication Manager using the **status trunk** command in a SAT Terminal session:

```
status trunk 1

TRUNK GROUP STATUS

Member      Port      Service State      Mtce Connected Ports
              Busy

0001/0001 T000001 in-service/active no T000007
0001/0002 T000002 in-service/idle no
0001/0003 T000003 in-service/idle no
0001/0004 T000004 in-service/idle no
0001/0005 T000005 in-service/idle no
0001/0006 T000006 in-service/idle no
0001/0007 T000007 in-service/active no T000001
```

To view the status of the endpoints connected to the SIP Trunk, and codecs in use, use **status trunk 1/0001** where /0001 is a trunk port connected to the call.

```
status trunk 1/0001 Page 4 of 4

SRC PORT TO DEST PORT TALKPATH

src port: T000001
T000007:TX:192.168.4.130:40750/g711u/20ms
001V062:RX:10.64.50.54:2054/g711u/20ms:TX:ctxID:542
001V061:RX:ctxID:542:TX:10.64.50.54:2056/g711u/20ms
T000001:RX:192.168.4.135:20000/g711u/20ms
```

Calls be placed by using the following URL:

<https://<IP Address of IX-SPMIC>/CallUP.cgi?ID=admin123&PW=admin123>

Once calls are connected, calls can be terminated by using the following URL:

<https://<IP Address of IX-SPMIC>/CallTalkEnd.cgi?ID=admin123&PW=admin123>

9. Conclusion

Aiphone IX-SPMIC was compliance tested with Avaya Aura[®]. Aiphone IX-SPMIC functioned properly for feature and serviceability.

10. Additional References

Avaya product documentation can be found at: <http://support.avaya.com>

Documentation related to Aiphone IX-SPMIC can be found at:

<https://www.aiphone.co.jp/products/business/ix/>

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