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.

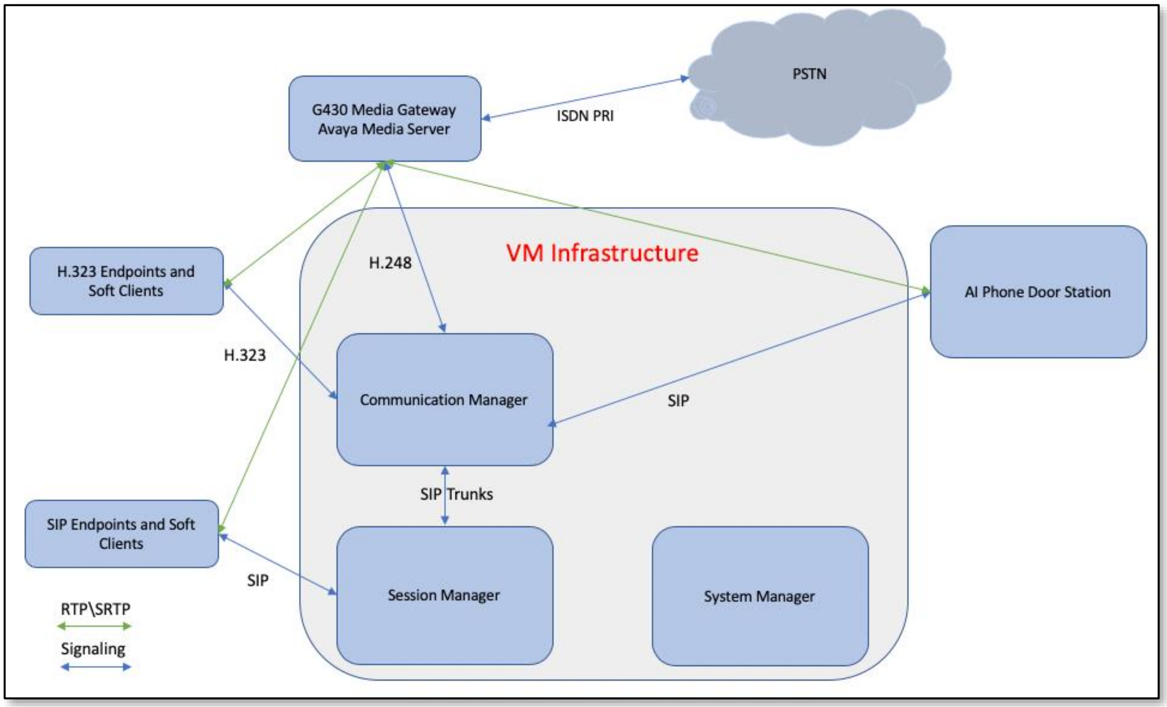


Figure 1: Test Configuration of Aiphone IX-SPMIC with Avaya Aura®

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**.

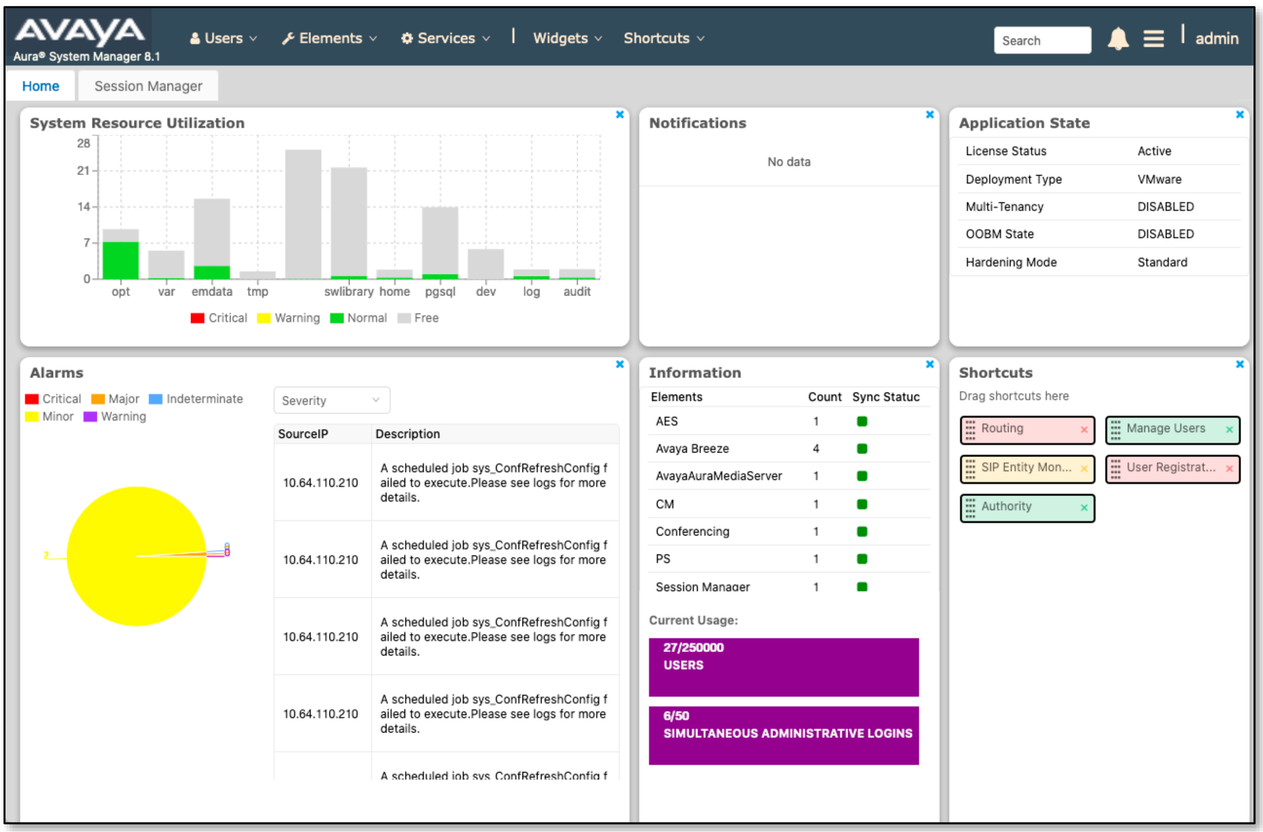
```
change dialplan analysis                                      Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all            Percent Full: 1

    Dialed   Total  Call      Dialed   Total  Call      Dialed   Total  Call
    String   Length Type      String   Length Type      String   Length Type
 1             3  dac
 2             5  ext
 3             5  ext
 4             5  aar
 7             5  ext
 8             1  fac
 9             1  fac
 *             3  fac
 #             3  fac
```

6. Configure Avaya Aura® Session Manager

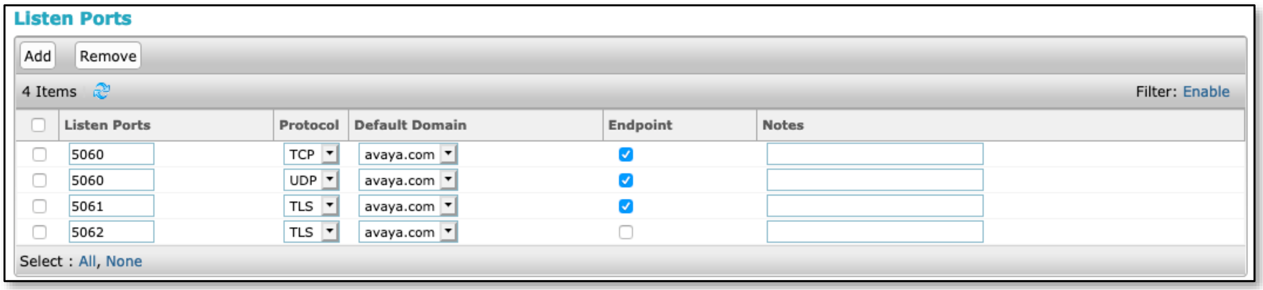
This section describes aspects of the Session Manager configuration required for interoperating with Aiphone IX-SPMIC. It is assumed that the Domains, Locations, SIP entities, Entity Links, Routing Policies, Dial Patterns and Application Sequences have been configured where appropriate for Communication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

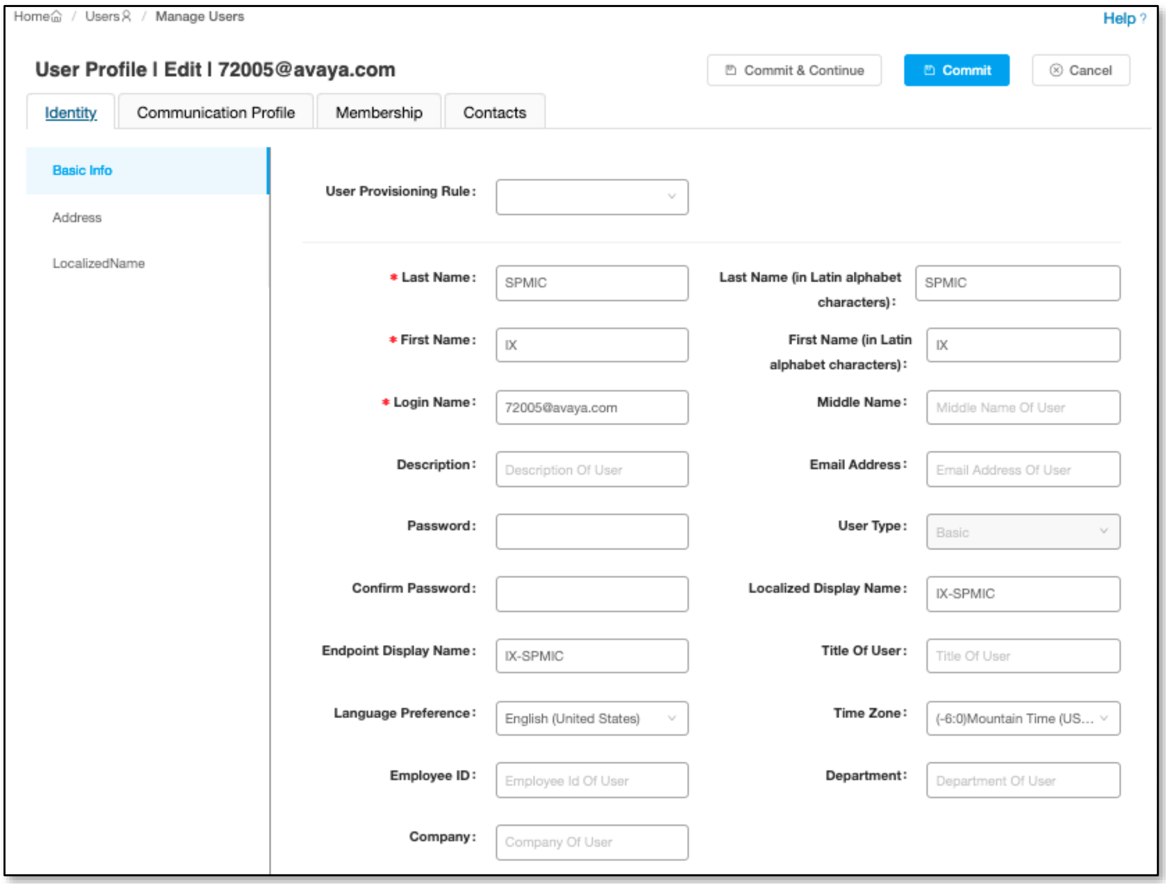
Each Session Manager Entity must be configured so that SIP endpoint can register to it using UDP, TCP, or TLS. From the web interface click **Routing → SIP Entities** (not shown) and select the Session Manager entity used for registration. In the compliance test, **TCP** and **UDP** listen ports were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



6.2. Add a SIP User

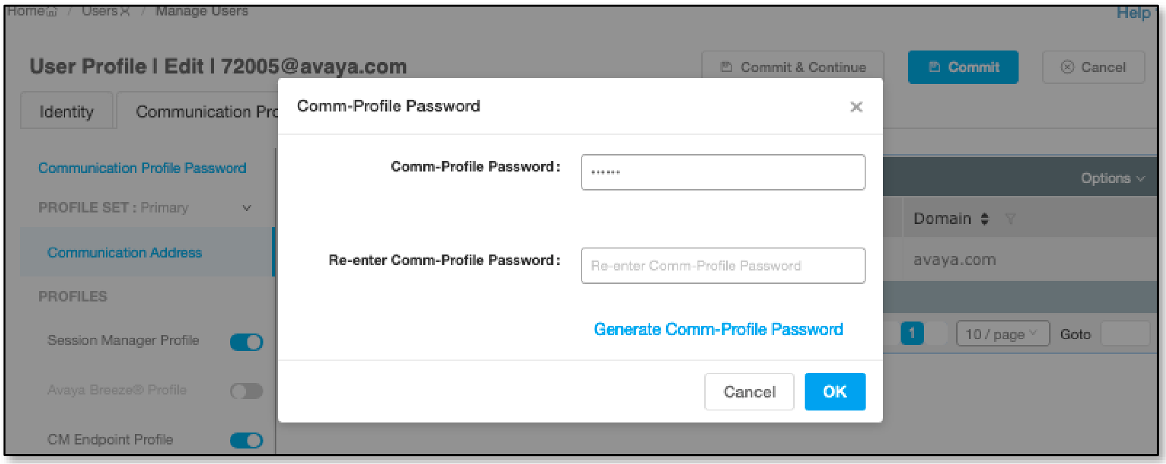
A SIP user must be added for Aiphone IX-SPMIC. Click **User Management** → **Manage Users** → **New** (not shown) and configure the following in the **Identity** tab.

- **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**

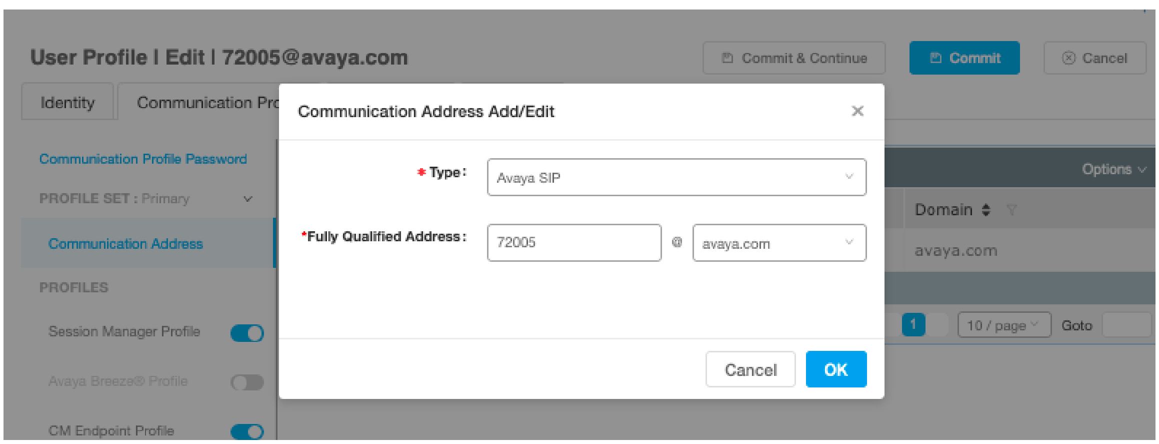


Note in this and subsequent steps, press **Commit & Continue** after making entries or selections.

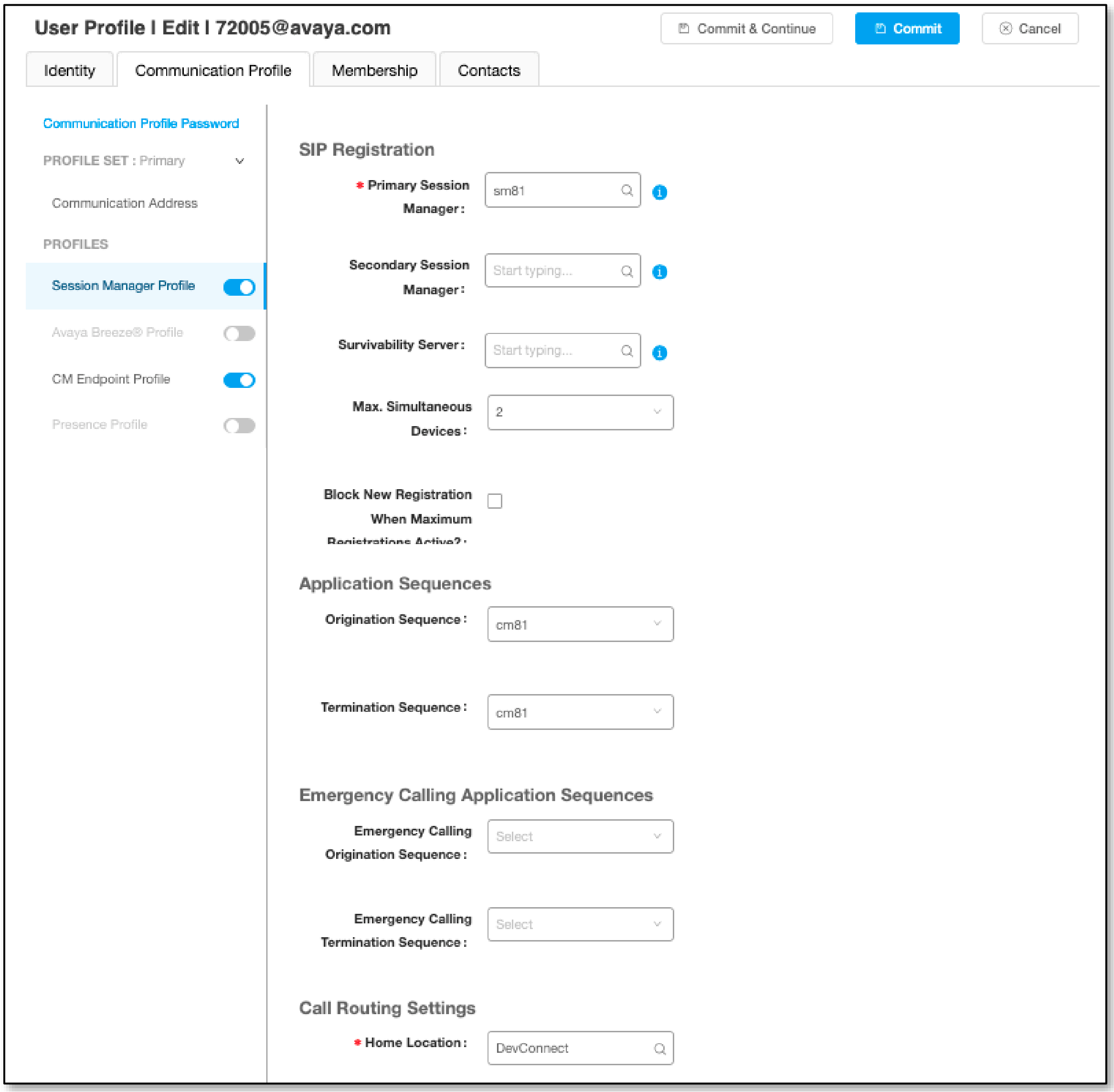
Click the **Communication Profile** tab and in the **Communication Profile Password** and **Confirm Password** fields, enter a numeric password. This will be used to register the device during login.



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.



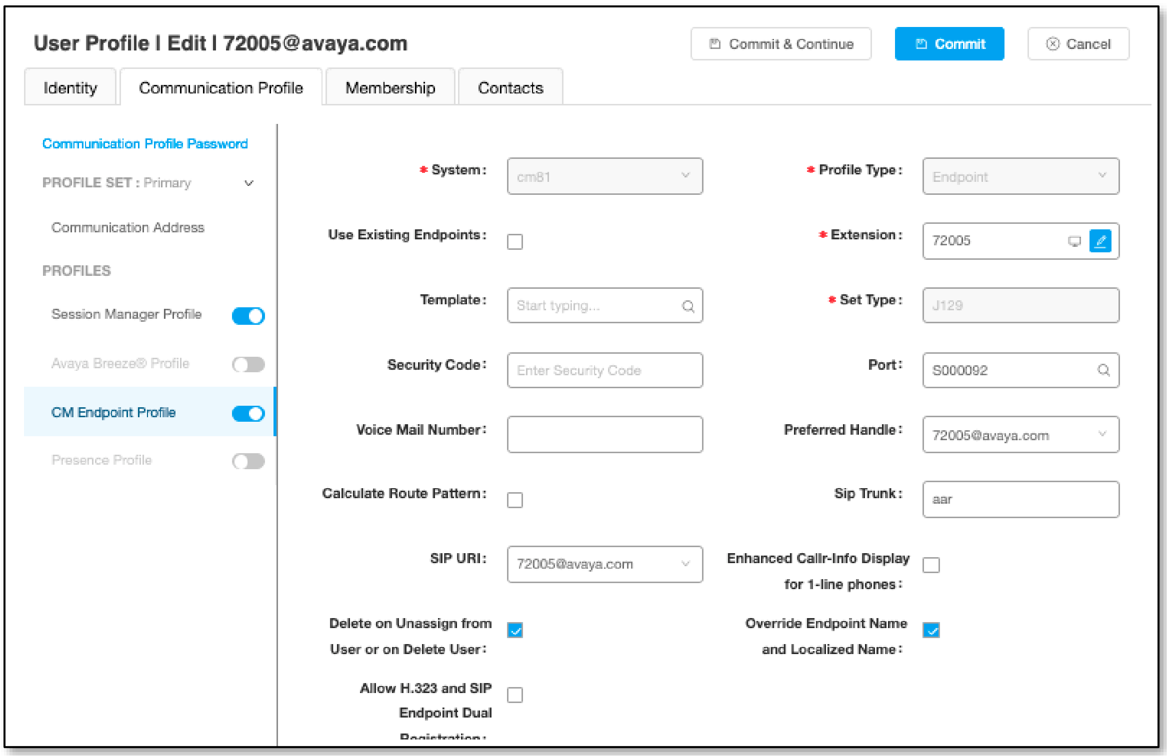
Click on the **Session Manager Profile link** and configure the **Primary Session Manager, Max Simultaneous Devices, Origination Application Sequence, Termination Application Sequence** and **Home Location,** from the respective drop-down lists.



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

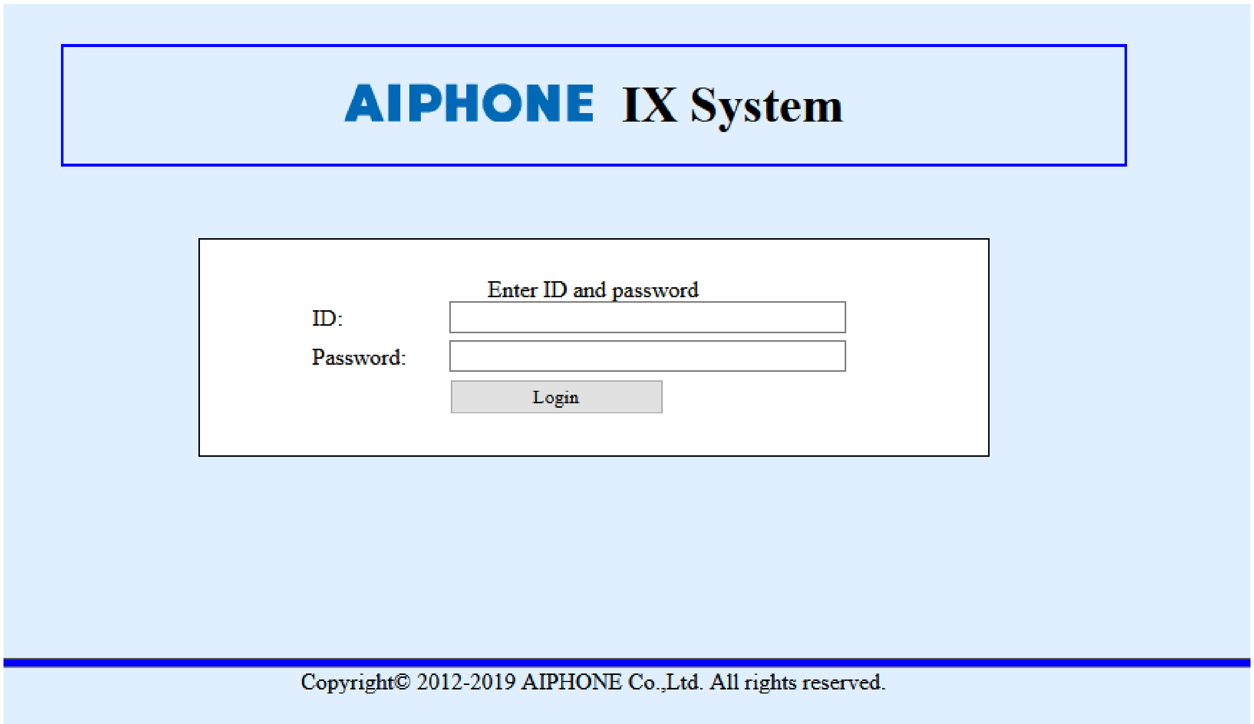
Click on **Endpoint Editor** in the Extension field to edit Communication Manager settings if desired.



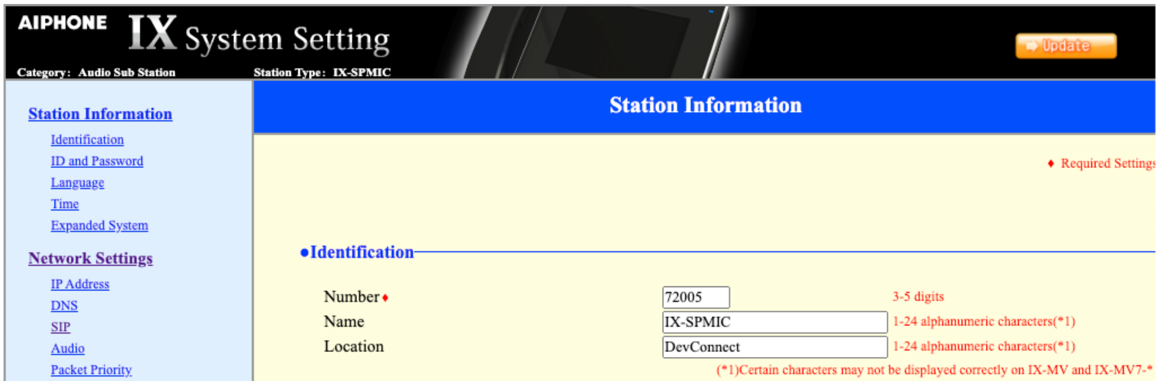
7. Configure Aiphone IX Series Audio Door Station

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.



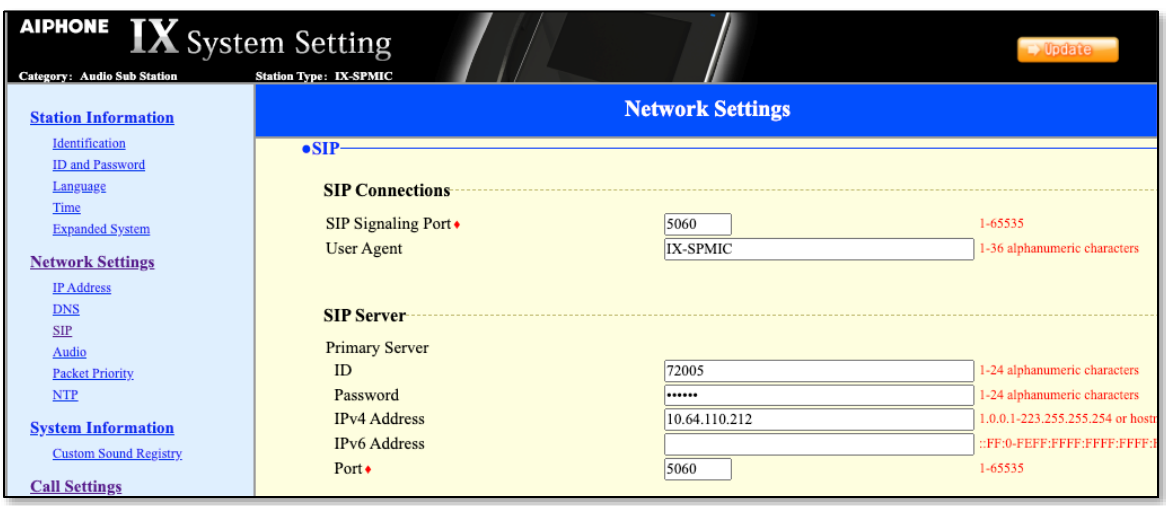
Once logged in, for the **Number** field, type in the SIP extension that is being configured (from **Section 6.2**), and a desired **Name.** Select **Update** to save changes.



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**.

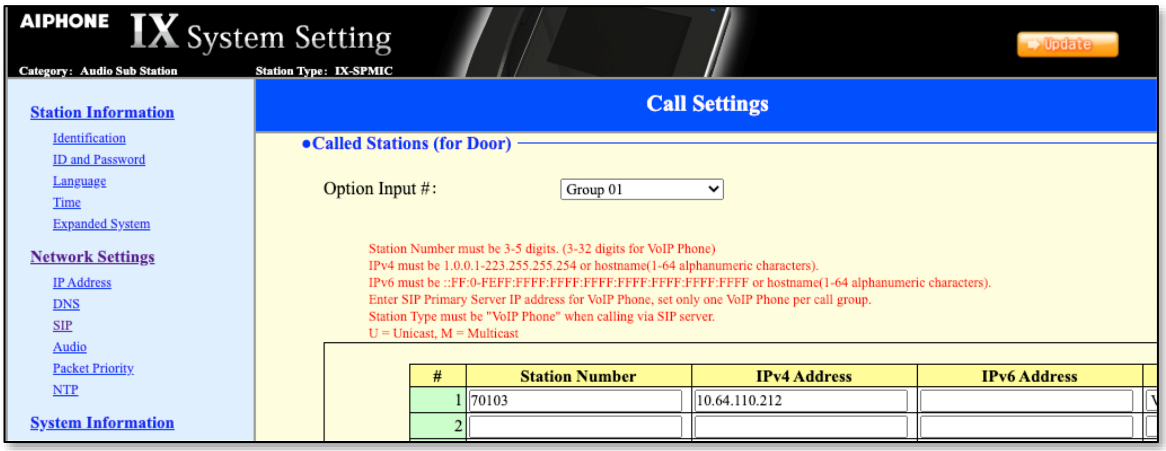
Once done, select **Update** to save changes.



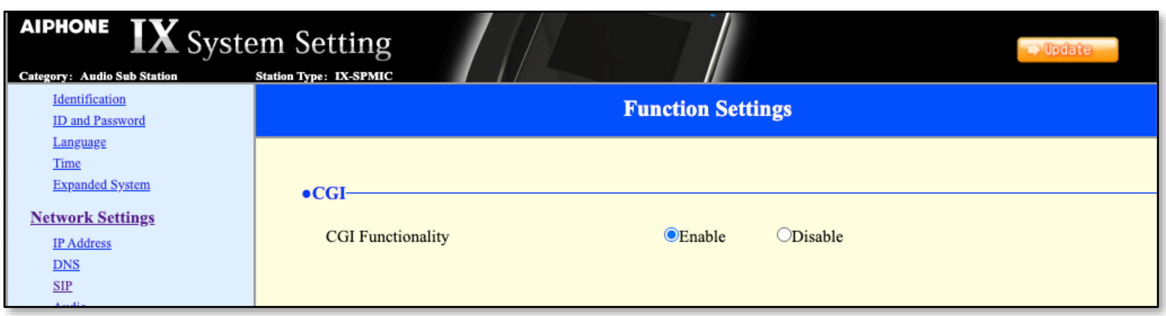
From the left, select **Call Settings → Called Stations** and configure as follows:
The numbers configured here will be dialed when the button on the IX-SPMIC is pressed.

- **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.



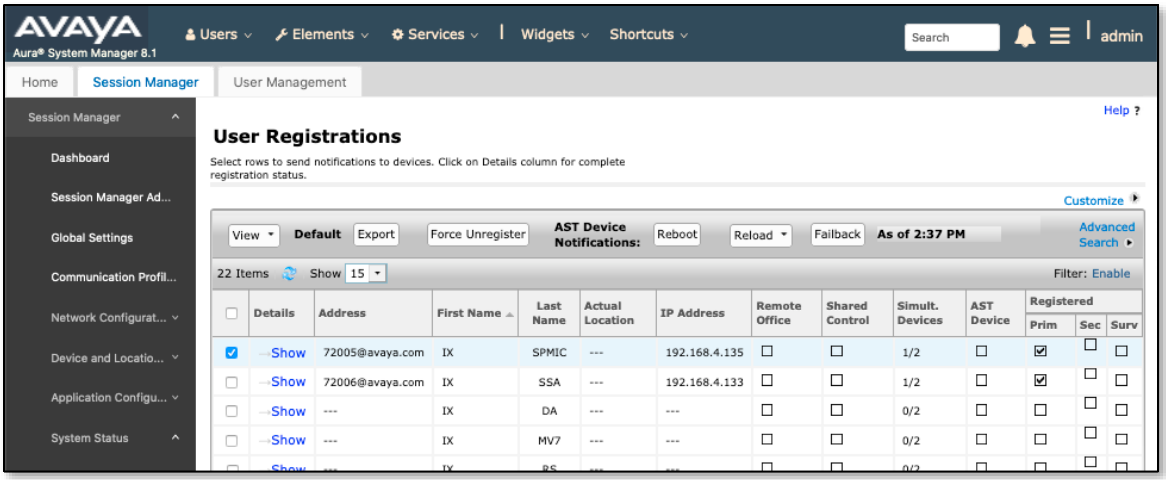
From the left, select **Function Settings → CGI** and select the **Enable** radio button for **CGI Functionality.** This enables Aiphone IX-SPMIC to place calls via a CGI URL.



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/

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.