



Avaya Solution & Interoperability Test Lab

Application Notes for Aiphone IX Series 2 Audio Door Stations (IX-SS-2GT) with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate Aiphone IX Series 2 Audio Door Station (IX-SS-2GT) Version 7.00 with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1. The Aiphone IX-SS-2GT Audio Door Station, which is part of the Aiphone IX Series 2 Audio Door Stations, was used for the compliance test. Aiphone IX-SS-2GT Audio Door Station is a flush mount, weather resistant audio door station. It has one dry contact that can be used to release doors when activated by a phone. Aiphone IX-SS-2GT Audio Door Station registers with Avaya Aura® Session Manager as a SIP endpoint.

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 to integrate Aiphone IX Series 2 Audio Door Station (IX-SS-2GT) Version 7.00 with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1. The Aiphone IX-SS-2GT Audio Door Station, which is part of the Aiphone IX Series 2 Audio Door Stations, was used for the compliance test. Aiphone IX-SS-2GT Audio Door Station is a flush mount, weather resistant audio door station. It has one dry contact that can be used to release doors when activated by a phone. Aiphone IX-SS-2GT Audio Door Station (IX-SS-2GT) registers with Avaya Aura® Session Manager as a SIP endpoint.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing audio calls between Aiphone IX-SS-2GT Audio Door Station, Avaya SIP and H.323 telephones, and the PSTN, and exercising basic telephony features, such as hold/resume, mute/unmute, transfer, conference, call forwarding, and call coverage from an Avaya IP endpoint. Additional telephony features, such as call forward and call coverage, were also verified.

The serviceability testing focused on verifying that the Aiphone IX-SS-2GT Audio Door Station comes back into service after re-connecting the Ethernet cable.

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 Aiphone IX-SS-2GT Audio Door Station did not include use of any specific encryption features as requested by Aiphone.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of IX-SS-2GT with Session Manager.
- Audio calls between IX-SS-2GT and Avaya SIP and H.323 deskphones with Direct IP Media (Shuffling) enabled and disabled.
- Audio calls between IX-SS-2GT and the PSTN.
- G.711 codec support.
- UDP transport protocol.
- IX-SS-2GT placing, answering, and terminating calls.
- DTMF tones recognition via input of Door Release Authorization Authentication Key.
- Basic telephony features, including hold/resume, mute/unmute, transfer, and 3-way conference, initiated from an Avaya IP endpoint.
- Proper system recovery after re-establishing IP connectivity to IX-SS-2GT.

2.2. Test Results

All test cases executed passed successfully.

2.3. Support

For technical support of Aiphone IX Series 2 Audio Door Stations, contact Aiphone Technical Support via phone or website.

- Phone: +1 (800) 692-0200
- Web: <https://www.aiphone.com/support/technical-support>

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network. Aiphone IX-SS-2GT Audio Door Station registered with Session Manager and was configured as an Off-PBX Station (OPS) on Communication Manager.

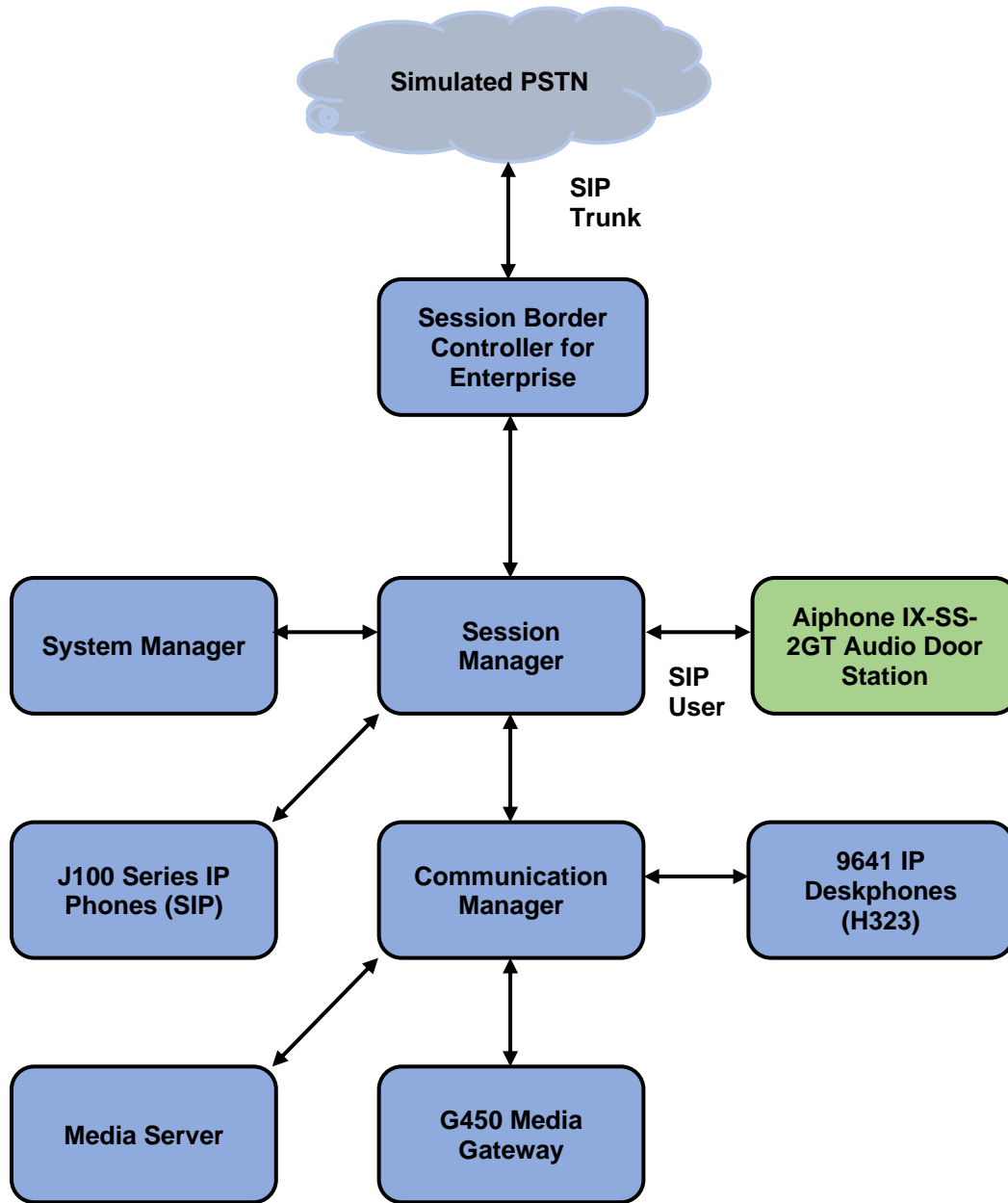


Figure 1: Avaya SIP Telephony Network with Aiphone IX-SS-2GT Audio Door Station

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	10.1.2.0 FP2 01.0.974.0-27783
Avaya G450 Media Gateway	FW 42.18.0
Avaya Aura® Media Server	10.1.0.125
Avaya Aura® System Manager	10.1.2.0 Feature Pack 2 10.1.2.0.0715476
Avaya Aura® Session Manager	10.1.2.0 Feature Pack 2 10.1.0.02.1012016
Avaya Session Border Controller for Enterprise	10.1.0.0-32-21432
Avaya 96x1 Series IP Deskphones	6.8.5.4 (H.323)
Avaya J100 Series IP Phones	4.1.0.0.9 (SIP)
Aiphone IX-SS-2GT Audio Door Station	7.00

5. Configure Avaya Aura® Communication Manager

This section describes the configuration of a SIP trunk to Session Manager and routing calls to IX-SS-2GT. Administration of Communication Manager was performed using the System Access Terminal (SAT). The following configuration is covered:

- Verify Communication Manager license.
- Administer IP Node Names.
- Administer IP Codec Set.
- Administer IP Network Region.
- Administer SIP Trunk to Session Manager.
- Configure Private Numbering.

5.1. Verify Communication Manager License

Using the SAT, verify that the Off-PBX Telephones (OPS) option is enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

On **Page 1**, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints, including IX-SS-2GT Audio Door Stations, that will be deployed.

```
display system-parameters customer-options                               Page 1 of 12
                                OPTIONAL FEATURES

G3 Version: V20                                                         Software Package: Enterprise
Location: 2                                                             System ID (SID): 1
Platform: 28                                                            Module ID (MID): 1

                                USED
Platform Maximum Ports: 48000    108
Maximum Stations: 150            73
Maximum XMOBILE Stations: 36000  0
Maximum Off-PBX Telephones - EC500: 150  0
Maximum Off-PBX Telephones - OPS: 150  42
Maximum Off-PBX Telephones - PBFMC: 150  0
Maximum Off-PBX Telephones - PVFMC: 150  0
Maximum Off-PBX Telephones - SCCAN: 0    0
Maximum Off-PBX Telephones - EMX: 150   0
Maximum Survivable Processors: 313    0

(NOTE: You must logoff & login to effect the permission changes.)
```

5.2. Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name for Communication Manager (*procr*) and Session Manager (*sm10*). The host names will be used in other configuration screens of Communication Manager.

```
change node-names ip                                     Page 1 of 2
                                                    IP NODE NAMES
      Name                IP Address
aes10                    10.64.110.247
ams10                    10.64.110.214
default                  0.0.0.0
procr                   10.64.110.213
procr6                   ::
sm10                   10.64.110.212

( 6 of 6 administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

5.3. Administer IP Codec Set

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to IX-SS-2GT. The form is accessed via the **change ip-codec-set 1** command. Note the codec set number since it will be used in the IP Network Region covered in the next section. For the compliance test, the G.711MU codec was verified. The following IP codec set is configured with G.711MU.

Media encryption was enabled for Avaya IP endpoints. IX-SS-2GT wasn't configured to support SRTP, so the *none* option was also included under **Media Encryption**.

```
change ip-codec-set 1                                     Page 1 of 2
                                                    IP MEDIA PARAMETERS
      Codec Set: 1
      Audio          Silence      Frames   Packet
      Codec          Suppression  Per Pkt  Size(ms)
1: G.711MU         n           2       20
2:
3:
4:
5:
6:
7:

      Media Encryption                               Encrypted SRTP: best-effort
1: 1-srtp-aescm128-hmac80
2: 10-srtp-aescm256-hmac80
3: none
4:
5:
```

5.4. Administer IP Network Region

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IX-SS-2GT and IP endpoints without using media resources in the Avaya G450 Media Gateway or Avaya Media Server. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region 1) is specified in the SIP signaling group.

```
change ip-network-region 1                                     Page 1 of 20
                                                           IP NETWORK REGION
  Region: 1          NR Group: 1
Location: 1         Authoritative Domain: avaya.com
  Name: Main        Stub Network Region: n
MEDIA PARAMETERS   Intra-region IP-IP Direct Audio: yes
                   Codec Set: 1           Inter-region IP-IP Direct Audio: yes
  UDP Port Min: 2048                               IP Audio Hairpinning? n
  UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
  Call Control PHB Value: 46
    Audio PHB Value: 46
    Video PHB Value: 26
802.1P/Q PARAMETERS
  Call Control 802.1p Priority: 6
    Audio 802.1p Priority: 6
    Video 802.1p Priority: 5
H.323 IP ENDPOINTS
  H.323 Link Bounce Recovery? y
  Idle Traffic Interval (sec): 20
  Keep-Alive Interval (sec): 5
  Keep-Alive Count: 5
                                                           AUDIO RESOURCE RESERVATION PARAMETERS
                                                           RSVP Enabled? n
```


Administer SIP Trunk to Session Manager

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tls*.
- Specify the Ethernet processor (*procr*) of Communication Manager and Session Manager (*sm10*) as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form in **Section 5.2**.
- Ensure that the TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.
- Set **Initial IP-IP Direct Media** field to *y*.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
display signaling-group 1                                     Page 1 of 3
                                SIGNALING GROUP

Group Number: 1                      Group Type: sip
  IMS Enabled? n                    Transport Method: tls
    Q-SIP? n
    IP Video? y                      Priority Video? n          Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y Peer Server: SM                      Clustered? n
  Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
  Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
  Alert Incoming SIP Crisis Calls? n
  Near-end Node Name: procr                Far-end Node Name: sm10
  Near-end Listen Port: 5061              Far-end Listen Port: 5061
                                          Far-end Network Region: 1

Far-end Domain: avaya.com

Incoming Dialog Loopbacks: eliminate          Bypass If IP Threshold Exceeded? n
                                          RFC 3389 Comfort Noise? n
  DTMF over IP: rtp-payload                Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3          IP Audio Hairpinning? n
  Enable Layer 3 Test? y                    Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? n      Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for SIP calls to IX-SS-2GT. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Accept the default values for the remaining fields.

```

display trunk-group 1                                     Page 1 of 5
                                     TRUNK GROUP
Group Number: 1                Group Type: sip                CDR Reports: y
  Group Name: SM Trunk 1                COR: 1                TN: 1                TAC: 101
  Direction: two-way                Outgoing Display? n
  Dial Access? n                Night Service:
Queue Length: 0
Service Type: tie                Auth Code? n
                                     Member Assignment Method: auto
                                     Signaling Group: 1
                                     Number of Members: 10

```

On **Page 3** of the trunk group form, set the **Numbering Format** field to *private*. This field specifies the format of the calling party number sent to the far-end.

```

display trunk-group 1                                     Page 3 of 5
TRUNK FEATURES
  ACA Assignment? n                Measured: both
                                     Maintenance Tests? y

  Suppress # Outpulsing? n Numbering Format: private
                                     UUI Treatment: shared
                                     Maximum Size of UUI Contents: 128
                                     Replace Restricted Numbers? n
                                     Replace Unavailable Numbers? n

                                     Modify Tandem Calling Number: no
  Send UCID? y

  Show ANSWERED BY on Display? y

  DSN Term? n

```

On **Page 4** of the trunk group form, the default settings were used as shown below.

```

display trunk-group 1                                     Page 5 of 5
                PROTOCOL VARIATIONS

                Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                Send Transferring Party Information? n
                Network Call Redirection? n

                Send Diversion Header? n
                Support Request History? y
                Telephone Event Payload Type: 101

                Convert 180 to 183 for Early Media? n
                Always Use re-INVITE for Display Updates? n
Resend Display UPDATE Once on Receipt of 481 Response? n
                Identity for Calling Party Display: P-Asserted-Identity
                Block Sending Calling Party Location in INVITE? n
                Accept Redirect to Blank User Destination? n
Enable Q-SIP? n
Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                Request URI Contents: may-have-extra-digits
    
```

5.5. Configure Private Numbering

Configure the **Numbering – Private Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with ‘7’ whose calls are routed over any trunk group, including SIP trunk group 1, have their extension sent.

```

change private-numbering 0                               Page 1 of 2
                NUMBERING - PRIVATE FORMAT

Ext Ext          Trk      Private      Total
Len Code        Grp(s)    Prefix      Len
5 7
5                    Total Administered: 1
                    Maximum Entries: 540
    
```

6. Configure Avaya Aura® Session Manager

This section covers the procedure for adding a SIP user in Session Manager. The configuration covers:

- Launch System Manager
- Set Network Transport Protocol for IX-SS-2GT
- Administer SIP User

Note: It is assumed that basic configuration of Session Manager has already been performed.

6.1. Launch System Manager

Access the System Manager Web interface by using the URL <https://<ip-address>> in an Internet browser window, where <ip-address> is the System Manager IP address. Log in using the appropriate credentials.

Recommended access to System Manager is via FQDN.

[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

All users must comply with all corporate instructions regarding the protection of information assets.

User ID:

Password:

[Change Password](#)

Supported Browsers: Firefox (minimum version 93.0), Chrome (minimum version 91.0) or Edge (minimum version 93.0).

6.2. Set Network Transport Protocol for IX-SS-2GT

From the System Manager **Home** screen, select **Elements** → **Routing** → **SIP Entities** and edit the SIP Entity for Session Manager shown below.

The screenshot shows the Avaya Aura System Manager 10.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The main content area is titled 'SIP Entity Details' and is divided into 'General' and 'Monitoring' sections. The 'General' section contains the following fields:

- Name:** sm10
- IP Address:** 10.64.110.212
- SIP FQDN:** (empty)
- Type:** Session Manager
- Notes:** (empty)
- Location:** DevConnect
- Outbound Proxy:** (empty)
- Time Zone:** America/Denver
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty)

The 'Monitoring' section contains the following fields:

- SIP Link Monitoring:** Link Monitoring Enabled
- Proactive Monitoring Interval (in seconds):** 900
- Reactive Monitoring Interval (in seconds):** 120
- Number of Tries:** 1

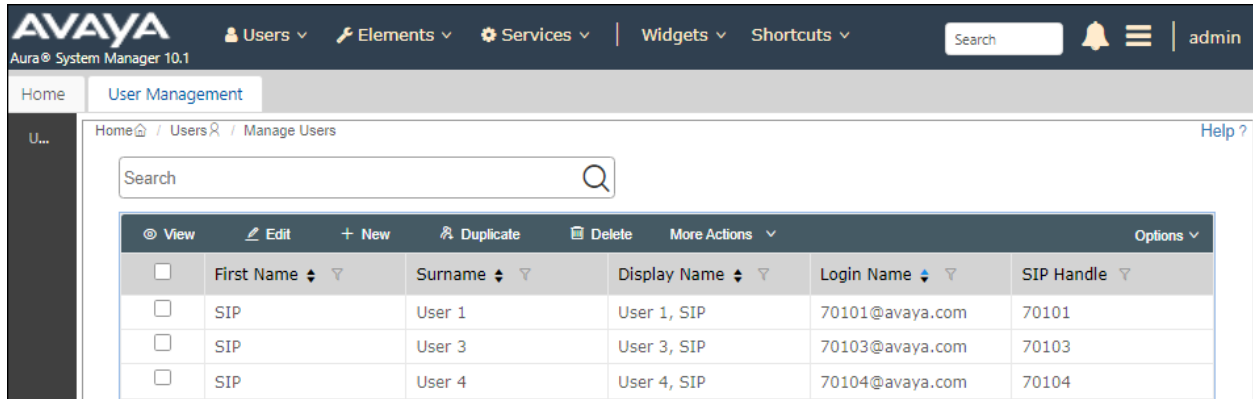
Scroll down to the **Listen Ports** section and verify that the transport network protocol used by IX-SS-2GT is specified in the list below. For the compliance test, the solution used UDP network transport.

The screenshot shows the 'Listen Ports' section of the Avaya Aura System Manager 10.1 interface. It displays a table with 3 items. The table has the following columns: Listen Ports, Protocol, Default Domain, Endpoint, and Notes. The data is as follows:

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

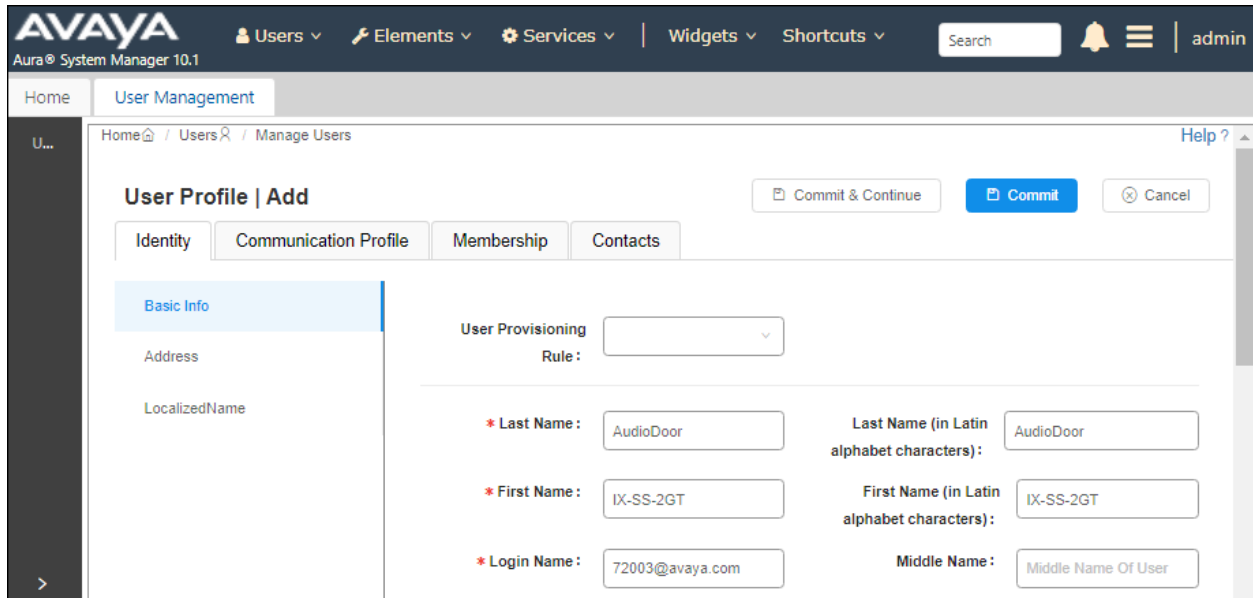
6.3. Administer SIP User

In the subsequent screen (not shown), select **Users** → **User Management** → **Manage Users** to display the **User Management** screen below. Click **New** to add a user.



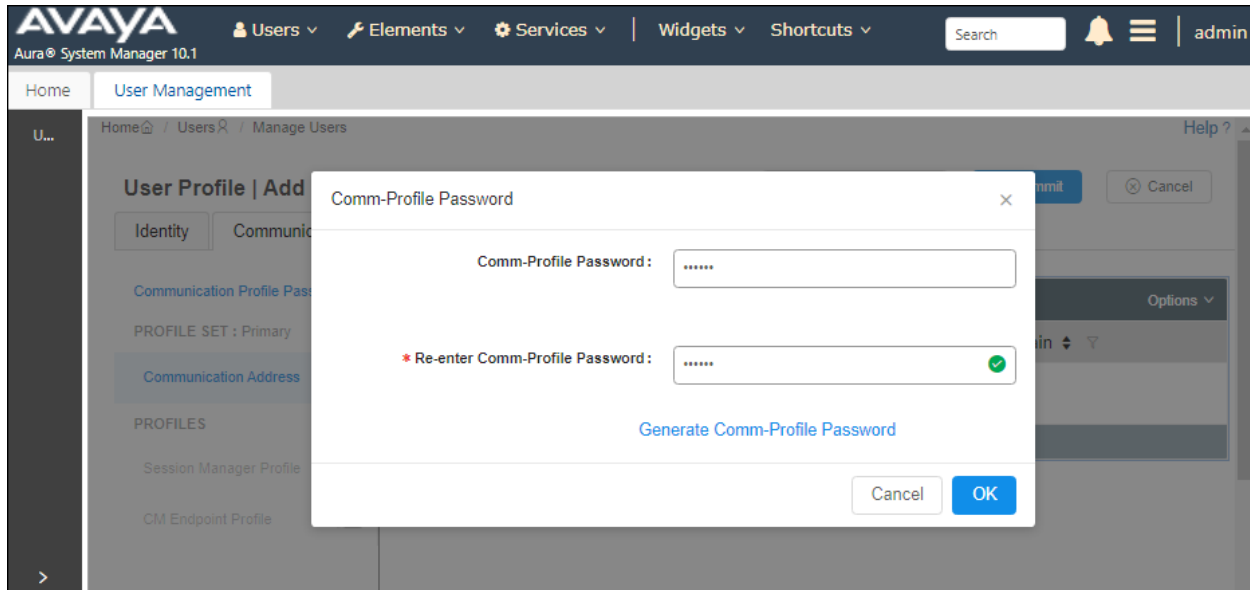
6.3.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter $\langle extension \rangle @ \langle domain \rangle$, where $\langle extension \rangle$ is the desired IX-SS-2GT SIP extension and $\langle domain \rangle$ is the applicable SIP domain name from **Section 0**. Retain the default values in the remaining fields.



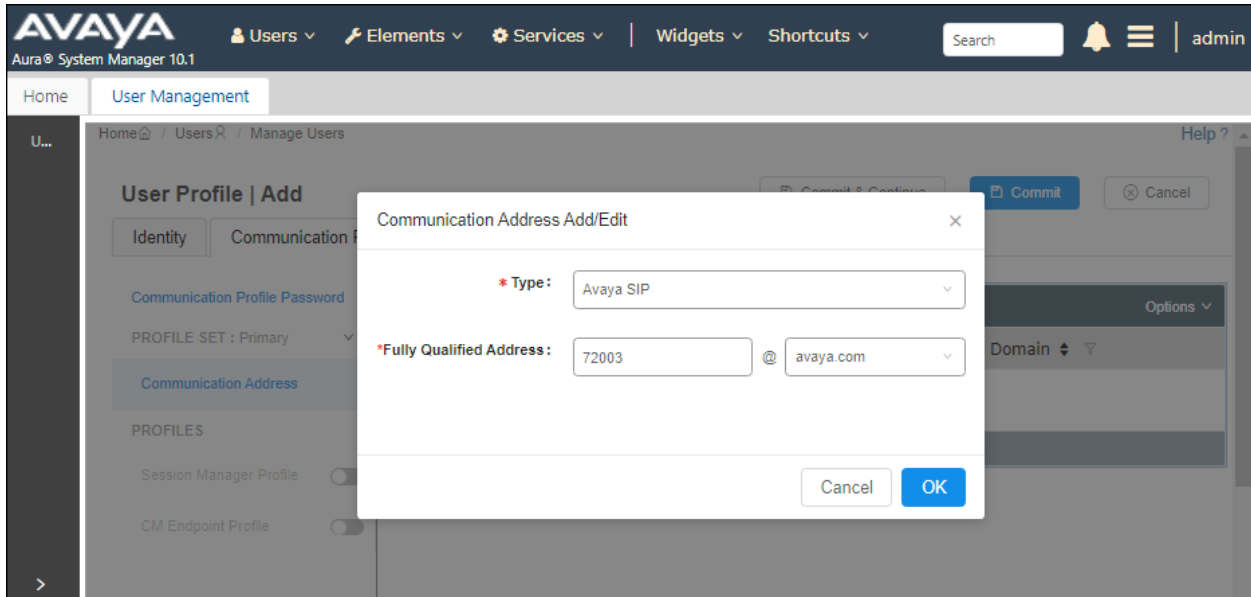
6.3.2. Communication Profile Password

Select the **Communication Profile** tab. Next, click on **Communication Profile Password**. For **Comm-Profile Password** and **Re-enter Comm-Profile Password**, enter the desired password for the SIP user to use for registration. Click **OK**.



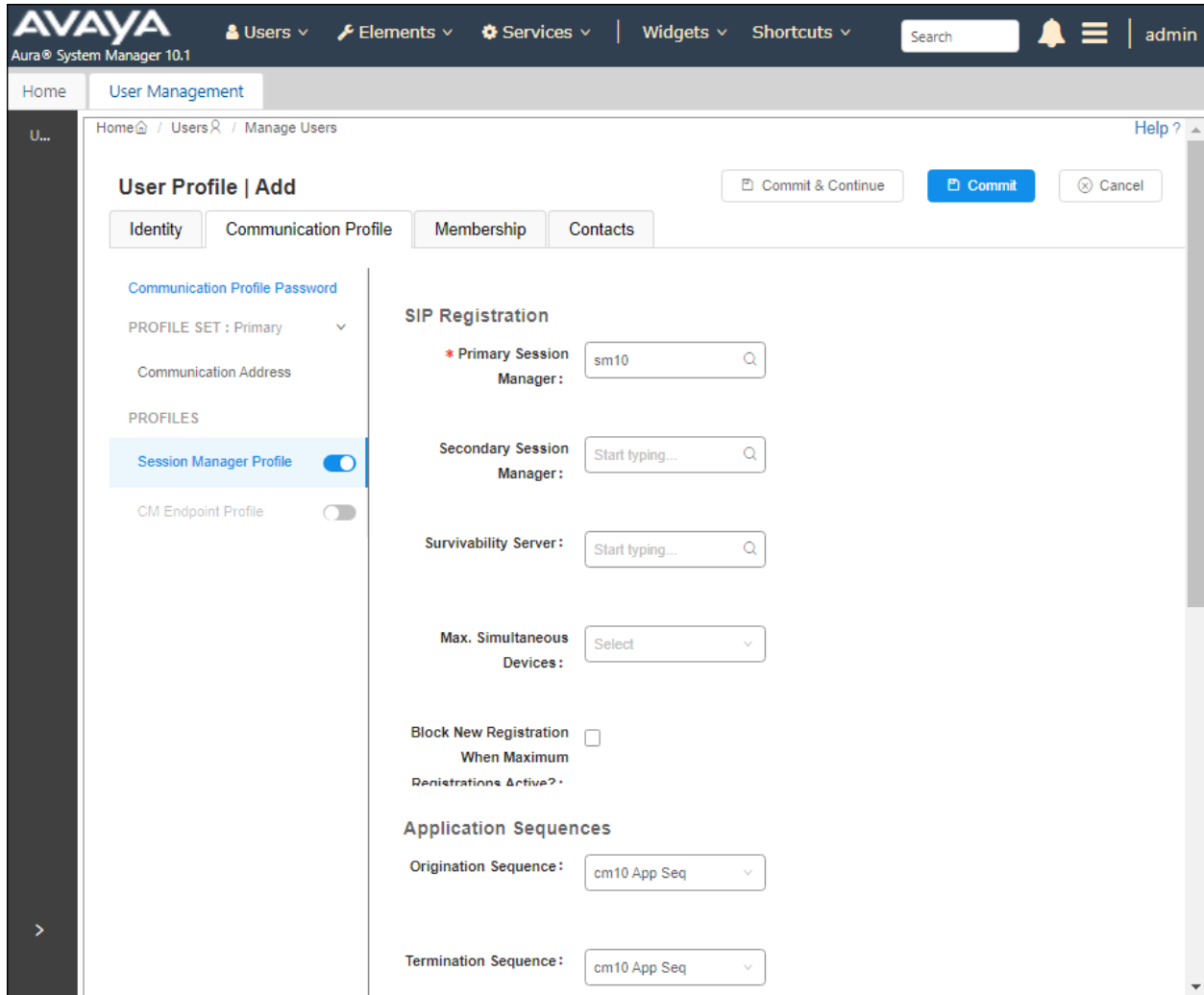
6.3.3. Communication Address

Click on **Communication Address** and then click **New** to add a new entry. The **Communication Address Add/Edit** dialog box is displayed as shown below. For **Type**, retain *Avaya SIP*. For **Fully Qualified Address**, enter the SIP user extension and select the domain name to match the login name from **Section 6.3.1**. Click **OK**.



6.3.4. Session Manager Profile

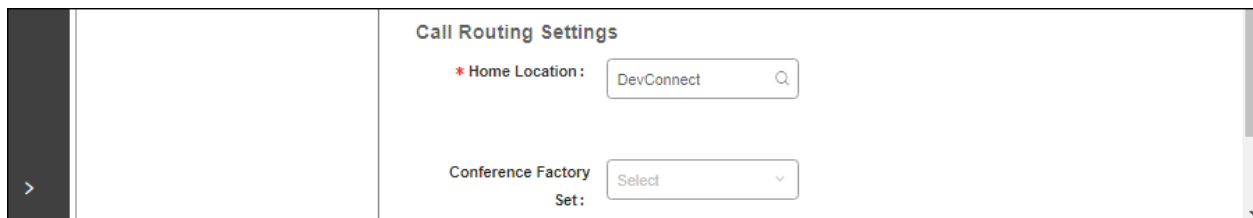
Click on toggle button by **Session Manager Profile**. For **Primary Session Manager**, **Origination Sequence** and **Termination Sequence** (Application Sequences), and **Home Location**, select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.



The screenshot shows the 'User Profile | Add' form in the Avaya Aura System Manager 10.1 interface. The 'Communication Profile' tab is active, and the 'Session Manager Profile' toggle is turned on. The form includes the following fields and options:

- Communication Profile Password:** PROFILE SET : Primary (dropdown), Communication Address (text input).
- PROFILES:** Session Manager Profile (toggle, ON), CM Endpoint Profile (toggle, OFF).
- SIP Registration:**
 - * Primary Session Manager: sm10 (text input)
 - Secondary Session Manager: Start typing... (text input)
 - Survivability Server: Start typing... (text input)
 - Max. Simultaneous Devices: Select (dropdown)
 - Block New Registration When Maximum Registrations Active?:
- Application Sequences:**
 - Origination Sequence: cm10 App Seq (dropdown)
 - Termination Sequence: cm10 App Seq (dropdown)

Scroll down to the **Call Routing Settings** section to configure the **Home Location**.



The screenshot shows the 'Call Routing Settings' section of the form. It includes the following fields:

- * Home Location: DevConnect (text input)
- Conference Factory Set: Select (dropdown)

6.3.5. CM Endpoint Profile

Click on the toggle button by **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.1**. For **Template**, select *9641SIP_DEFAULT_CM_10_1*. For **Port**, click and select *IP*. Retain the default values in the remaining fields.

The screenshot displays the Avaya Aura System Manager 10.1 interface. The top navigation bar includes the Avaya logo, 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts' menus, a search bar, a notification bell, and the user 'admin'. The breadcrumb trail shows 'Home > Users > Manage Users'. The main content area is titled 'User Profile | Add' and features tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' tab is active. On the left, a sidebar shows 'Communication Profile Password' (PROFILE SET: Primary), 'Communication Address', and 'PROFILES' with 'Session Manager Profile' and 'CM Endpoint Profile' (checked) toggle switches. The main form contains the following fields: '* System:' (cm10), '* Profile Type:' (Endpoint), 'Use Existing Endpoints:' (checkbox), '* Extension:' (72003), '* Template:' (9641SIP_DEFAULT_CM_10_1), '* Set Type:' (9641SIP), 'Security Code:' (Enter Security Code), 'Port:' (IP), 'Voice Mail Number:', 'Preferred Handle:' (Select), 'Calculate Route Pattern:' (checkbox), 'Sip Trunk:' (aar), 'SIP URI:' (Select), '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' (checkbox). Action buttons 'Commit & Continue', 'Commit', and 'Cancel' are at the top right.

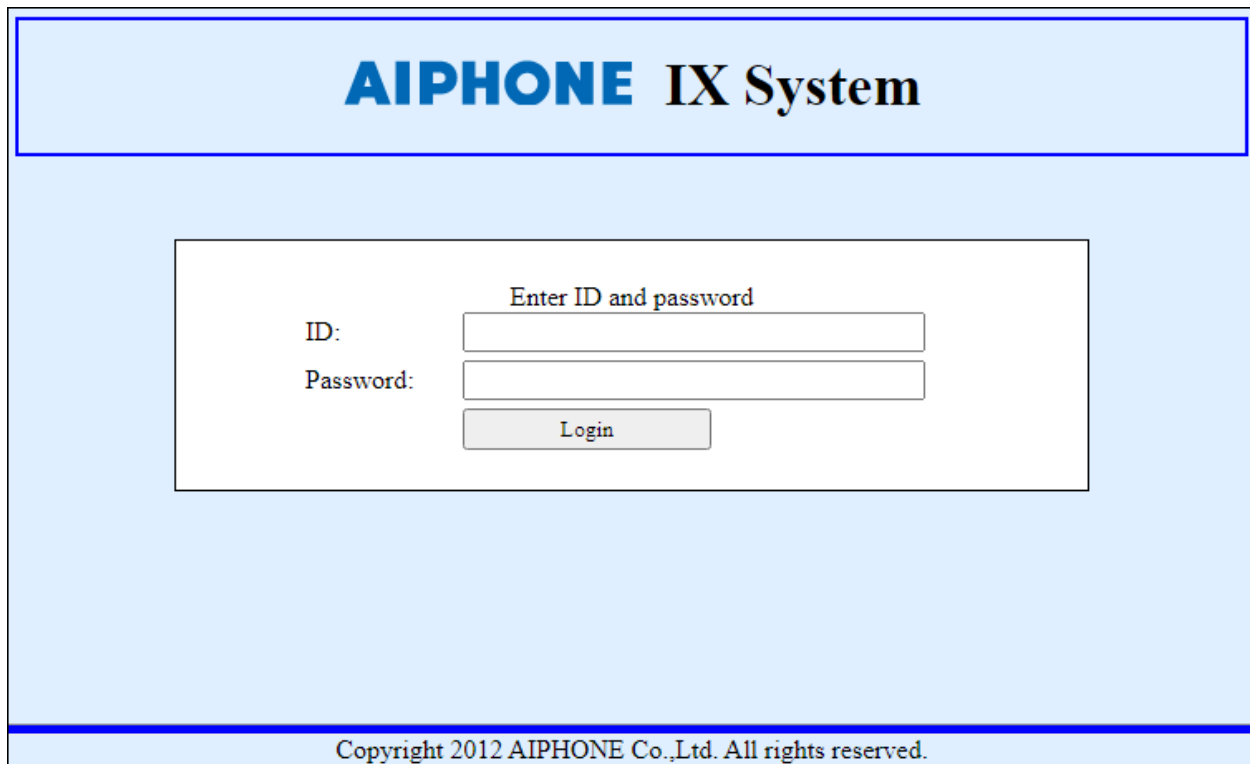
7. Configure Aiphone IX-SS-2GT Audio Door Station

This section provides the procedure for configuring IX-SS-2GT to provide SIP connectivity to Session Manager. Configuration of IX-SS-2GT is performed via Aiphone IX System web interface.

- Log into Aiphone IX System Web Interface
- Administer Station Information
- Administer SIP Parameters
- Administer Audio Settings
- Administer Call Settings

7.1. Log into Aiphone IX System Web Interface

Access the Aiphone IX System Web Interface by using the URL <https://<ip-address>/webset.cgi?login> in an Internet browser, where <ip-address> is the IX-SS-2GT IP address. Select language (not shown) and log in using the appropriate credentials.



AIPHONE IX System

Enter ID and password

ID:

Password:

Login

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7.2. Administer Station Information

Navigate to **Station Information** → **Identification** and set the **Number** to the IX-SS-2GT SIP extension (e.g., 72003). Input an appropriate **Name**.

The screenshot shows the AIPHONE IX System Setting web interface. The top navigation bar includes the AIPHONE logo, the text "IX System Setting", and an "Update" button. Below the navigation bar, the page is titled "Station Information" and shows "Category: Audio Stations" and "Station Type: IX-SS-2GT".

The left sidebar contains a navigation menu with the following items:

- Station Information
 - Identification
 - ID and Password
 - Language
 - Time
 - Expanded System
- Network Settings
 - IP Address
 - DNS
 - SIP
 - Audio
 - Packet Priority
 - NTP

The main content area is titled "Station Information" and contains a section for "Identification". The "Identification" section has the following fields:

Field	Value	Validation
Number	72003	3-5 digits
Name	IX-SS-2GT	1-24 alphanumeric characters(*1)
Location		1-24 alphanumeric characters(*1)

1) Certain characters may not be displayed correctly on IX-MV, IX-MV7- and IX-MV7-*T due to font type.

7.3. Administer SIP Parameters

Navigate to **Network Settings** → **SIP** from the left pane and configure the following parameters:

- **SIP Signaling Port:** Set to *5060*.
- **User Agent:** Enter desired value (e.g., *IX-SS-2GT*).
- **ID:** Set to SIP extension (e.g., *72003*) from **Section 6.3.1**.
- **Password:** Enter SIP password from **Section 6.3.2**.
- **IPv4 Address:** Set to signaling IP address of Session Manager (e.g., *10.64.110.212*).
- **Port:** Set to *5060*.

Click **Update** to save changes.

The screenshot shows the 'AIPHONE IX System Setting' web interface. The top navigation bar includes 'Station Type: IX-SS-2GT' and an 'Update' button. The left sidebar lists various settings categories: Station Information, Network Settings, and Call Settings. The main content area is titled 'Network Settings' and contains a 'SIP' section. This section is divided into three sub-sections: 'SIP Connections', 'SIP Server', and 'SIP Client'. The 'SIP Connections' section includes 'SIP Signaling Port' (set to 5060) and 'User Agent' (set to IX-SS-2GT). The 'SIP Server' section includes 'SIP Compatibility Mode' (set to Standard Mode), 'Primary Server ID' (set to 72003), 'Password' (masked with asterisks), 'IPv4 Address' (set to 10.64.110.212), 'IPv6 Address' (set to ::FF:0:FEFF:FFFF:FFFF:FFFF:), and 'Port' (set to 5060). Each input field has a corresponding validation message on the right.

Parameter	Value	Validation
SIP Signaling Port	5060	1-65535
User Agent	IX-SS-2GT	1-36 alphanumeric characters
SIP Compatibility Mode	Standard Mode	
Primary Server ID	72003	1-24 alphanumeric characters
Password	*****	1-24 alphanumeric characters
IPv4 Address	10.64.110.212	1.0.0.1-223.255.255.254 or hostname
IPv6 Address	::FF:0:FEFF:FFFF:FFFF:FFFF:	::FF:0:FEFF:FFFF:FFFF:FFFF:
Port	5060	1-65535

7.4. Administer Audio Settings

Navigate to **Network Settings** → **Audio** in the left pane and set **Audio Codec** to select *G.711 (u-law)*.

The screenshot displays the 'AIPHONE IX System Setting' web interface. The top navigation bar includes 'Category: Audio Stations' and 'Station Type: IX-SS-2GT'. The main content area is titled 'Network Settings' and is divided into several sections:

- Audio**: This section contains several settings:
 - Audio Codec**: Three radio buttons are present: G.711(u-law), G.711(A-law), and G.722.
 - Audio RTP Transmission Interval [msec]**: A dropdown menu set to '20'.
 - RTP Idle Detection Time [sec]**: A text input field set to '10'. A red note below states: 'This setting is ignored when transmitting to multiple stations (paging, etc.) 10-180 sec'.
- SIP Channel**: A section separated by a dashed line, containing:
 - RTP Start Port**: A text input field set to '20000' with a range indicator '1-65534'.
 - RTP End Port**: A text input field set to '21000' with a range indicator '1-65535'.
- ONVIF Transmit Channel**: Another section separated by a dashed line, containing:
 - RTP Start Port**: A text input field set to '22000' with a range indicator '1-65534'.
 - RTP End Port**: A text input field set to '23000' with a range indicator '1-65535'.
- Audio Buffer**: A section separated by a dashed line, containing:
 - Packets Buffered at Audio Start**: A dropdown menu set to '1'.
 - Maximum Packets Buffered**: A dropdown menu set to '3'. A red note below states: 'Maximum Packet Buffer must be larger than Audio Start Buffer.'

The left sidebar contains a navigation menu with categories: Station Information, Network Settings, Call Settings, Option Input / Relay, Output Settings, and Function Settings. The 'Audio' link under Network Settings is highlighted.

7.5. Administer Call Settings

Navigate to **Call Settings** in the left pane and set the **Call Button Function** to *Call, Answer Call, End Communication* in the **Station Information** section.

In the **Called Stations (for Door)** section, add an entry that specifies the number that should be dialed when the call button is pressed. Set the **Station Number** to the called number (e.g., 70103), set the **IPv4 Address** to the signaling IP address of Session Manager (e.g., 10.64.110.212), and set **Station Type** to *VoIP Phone*. Only one VoIP phone may be specified.

Station Information

Call Button Function:

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	Station Type
1	70103	10.64.110.212		VoIP Phone
2				
3				

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager, Session Manager, and Aiphone IX-SS-2GT Audio Door Station.

1. Verify that IX-SS-2GT has successfully registered with Session Manager. In System Manager, navigate to **Elements** → **Session Manager** → **System Status** → **User Registrations** to check the registration status as shown below.

The screenshot shows the Avaya Aura System Manager 10.1 interface. The main content area is titled "User Registrations" and includes a table with 19 items. The table columns are: Details, Address, First Name, Last Name, Actual Location, IP Address, Policy, Shared Control, Simult. Devices, AST Device, and Registered (Prim, Sec, 3rd, 4th, Surv, Visiting). The table shows four rows of registered devices, all with a 'Prim' status checked.

	Details	Address	First Name	Last Name	Actual Location	IP Address	Policy	Shared Control	Simult. Devices	AST Device	Registered					
											Prim	Sec	3rd	4th	Surv	Visiting
<input type="checkbox"/>	Show	72004@avaya.com	IX-RS-BT	AudioDoor	---	10.64.10.78	fixed	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	72003@avaya.com	IX-SS-2GT	AudioDoor	---	10.64.10.77	fixed	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	72002@avaya.com	IX-BBT	AudioDoor	---	10.64.10.76	fixed	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	72001@avaya.com	IX-BB	AudioDoor	---	10.64.10.75	fixed	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

2. Establish inbound and outbound calls to IX-SS-2GT with Avaya SIP and/or Avaya H.323 endpoints and verify two-way audio.

9. Conclusion

These Application Notes describe the administration steps required to integrate Aiphone IX Series 2 Audio Door Stations (IX-SS-2GT) with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Aiphone IX-SS-2GT Audio Door Station successfully registered with Avaya Aura® Session Manager as a SIP endpoint and audio calls were verified. All test cases executed passed.

10. References

This section references the Avaya and Aiphone documentation relevant to these Application Notes.

- [1] *Administering Avaya Aura® Communication Manager*, Release 10.1.x, Issue 5, March 2023, available at <http://support.avaya.com>.
- [2] *Administering Avaya Aura® System Manager*, Release 10.1.x, Issue 8, February 2023, available at <http://support.avaya.com>.
- [3] *Administering Avaya Aura® Session Manager*, Release 10.1.x, Issue 5, February 2023, available at <http://support.avaya.com>.
- [4] *Aiphone IX Door Stations Web Setting Manual*, Software version 6.00 or later, available from Aiphone.

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