



Avaya Solution & Interoperability Test Lab

Application Notes for Aiphone IX Series 2 Audio Door Stations (IX-RS-BT) 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-RS-BT) Version 7.00 with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1. The Aiphone IX-RS-BT Audio Door Station, which is part of the Aiphone IX Series 2 Audio Door Station was used for the compliance test. Aiphone IX-RS-BT Audio Door Station is a surface 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-RS-BT 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-RS-BT) Version 7.00 with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1. The Aiphone IX-RS-BT Audio Door Station, which is part of the Aiphone IX Series 2 Audio Door Stations, was used for the compliance test. Aiphone IX-RS-BT Audio Door Station is a surface 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-RS-BT Audio Door Station (IX-RS-BT) 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-RS-BT 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-RS-BT 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-RS-BT 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-RS-BT with Session Manager.
- Audio calls between IX-RS-BT and Avaya SIP and H.323 deskphones with Direct IP Media (Shuffling) enabled and disabled.
- Audio calls between IX-RS-BT and the PSTN.
- G.711 codec support.
- UDP transport protocol.
- IX-RS-BT 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-RS-BT.

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-RS-BT Audio Door Station registered with Session Manager and was configured as an Off-PBX Station (OPS) on Communication Manager.

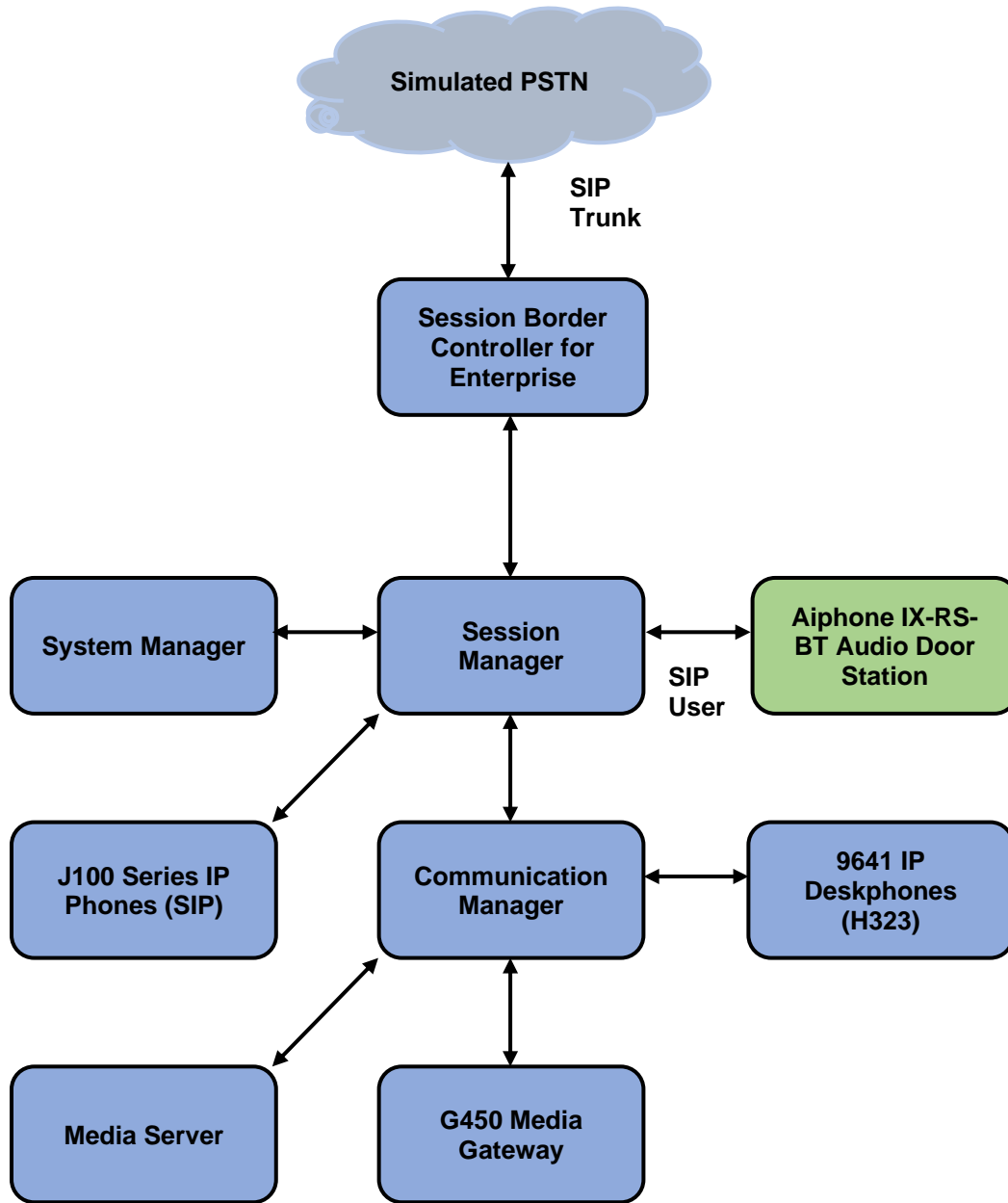


Figure 1: Avaya SIP Telephony Network with Aiphone IX-RS-BT Audio Door Station

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-RS-BT 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-RS-BT. 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-RS-BT 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-RS-BT. 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-RS-BT 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-RS-BT 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   AUDIO RESOURCE RESERVATION PARAMETERS
    H.323 Link Bounce Recovery? y      RSVP Enabled? n
    Idle Traffic Interval (sec): 20
    Keep-Alive Interval (sec): 5
    Keep-Alive Count: 5
```


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-RS-BT. 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
                                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-RS-BT
- 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-RS-BT

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 'SIP Entity Details' configuration page in the Avaya Aura System Manager 10.1 interface. The page is titled 'SIP Entity Details' and has 'Commit' and 'Cancel' buttons. The 'General' section includes 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 includes 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-RS-BT is specified in the list below. For the compliance test, the solution used UDP network transport.

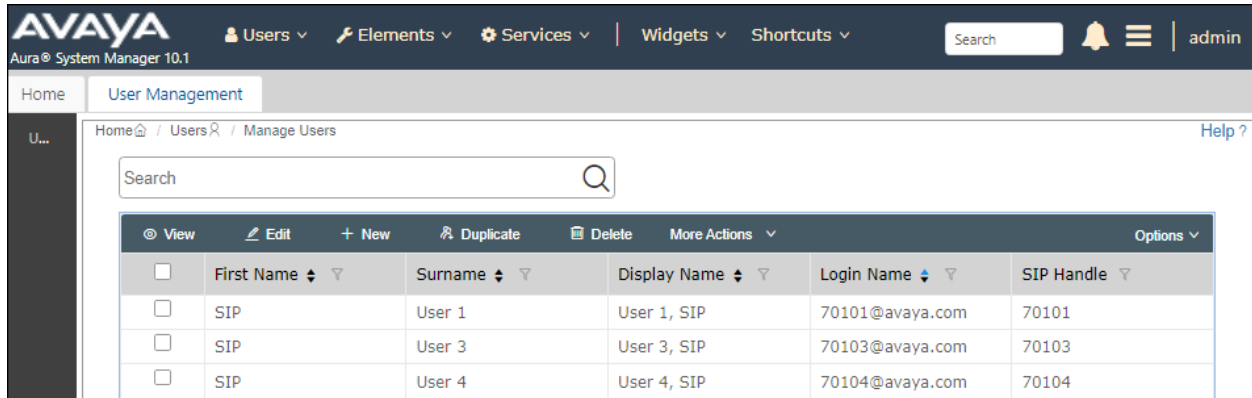
The screenshot shows the 'Listen Ports' configuration table in the Avaya Aura System Manager 10.1 interface. The table has 3 items and a 'Filter: Enable' button. The table columns are Listen Ports, Protocol, Default Domain, Endpoint, and Notes.

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"/>	

Select : All, None

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.

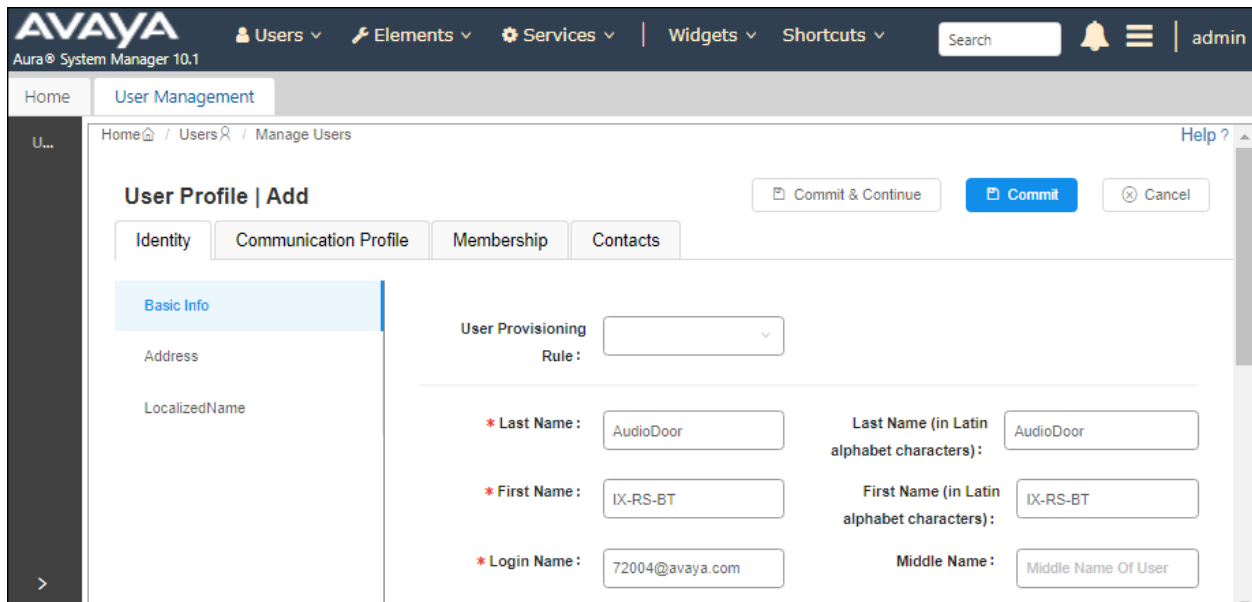


The screenshot shows the Avaya Aura System Manager 10.1 User Management interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. A search bar and a user profile icon labeled 'admin' are also present. The main content area displays a table of users with the following columns: First Name, Surname, Display Name, Login Name, and SIP Handle. The table contains three rows of user data.

<input type="checkbox"/>	First Name	Surname	Display Name	Login Name	SIP Handle
<input type="checkbox"/>	SIP	User 1	User 1, SIP	70101@avaya.com	70101
<input type="checkbox"/>	SIP	User 3	User 3, SIP	70103@avaya.com	70103
<input type="checkbox"/>	SIP	User 4	User 4, SIP	70104@avaya.com	70104

6.3.1. Identity

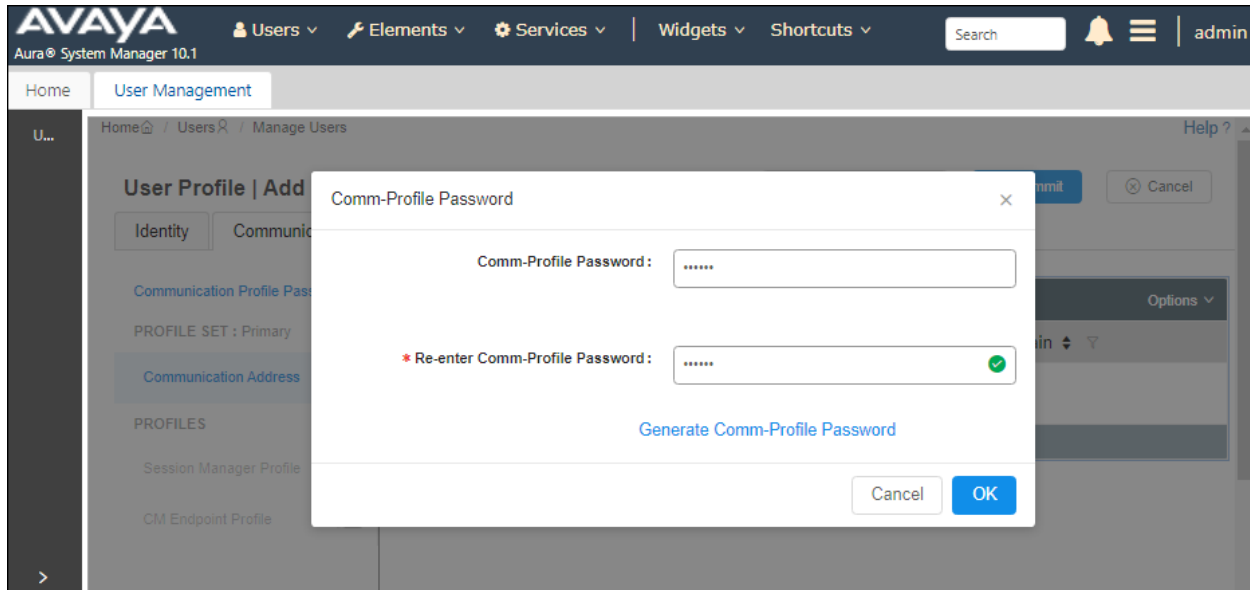
The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter $\langle extension \rangle @ \langle domain \rangle$, where $\langle extension \rangle$ is the desired IX-RS-BT SIP extension and $\langle domain \rangle$ is the applicable SIP domain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



The screenshot shows the 'User Profile | Add' screen in the Avaya Aura System Manager 10.1. The interface includes a navigation bar with 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The main content area is titled 'User Profile | Add' and features a sidebar with tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Identity' tab is selected, showing a 'Basic Info' section with the following fields: 'User Provisioning Rule' (dropdown), '* Last Name' (text input with value 'AudioDoor'), '* First Name' (text input with value 'IX-RS-BT'), and '* Login Name' (text input with value '72004@avaya.com'). There are also fields for 'Last Name (in Latin alphabet characters)', 'First Name (in Latin alphabet characters)', and 'Middle Name'.

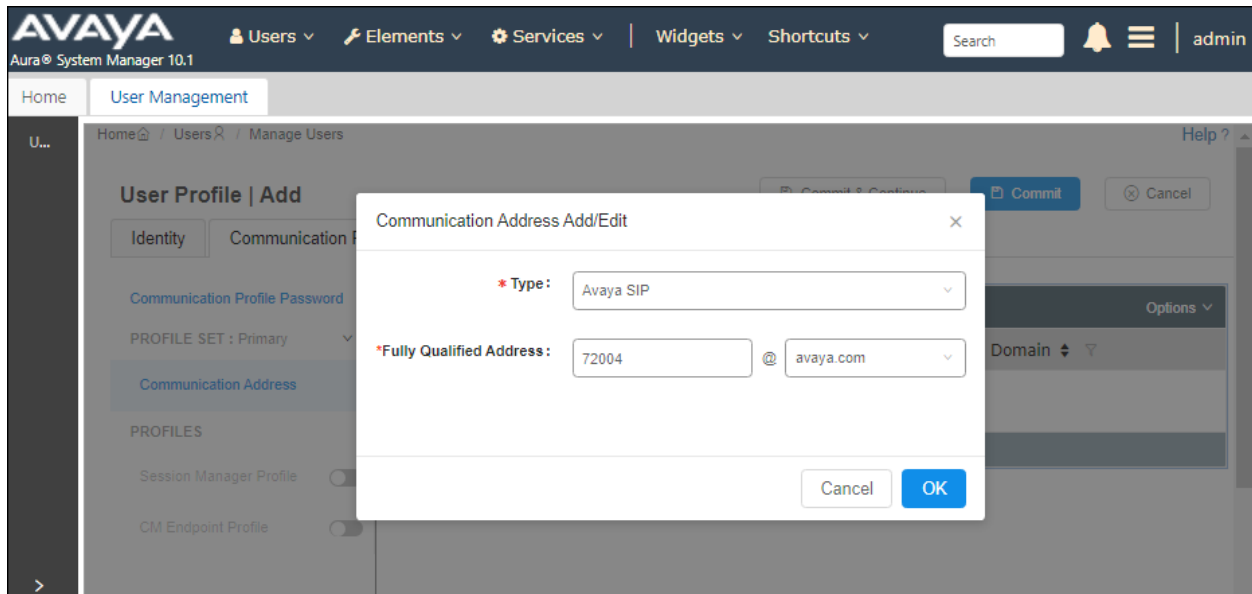
6.3.2. Communication Profile Password

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 'Session Manager Profile' toggle is turned on. The 'SIP Registration' section includes fields for Primary Session Manager (sm10), Secondary Session Manager (Start typing...), Survivability Server (Start typing...), Max. Simultaneous Devices (Select), and Block New Registration When Maximum Registrations Active? (checkbox). The 'Application Sequences' section includes Origination Sequence (cm10 App Seq) and Termination Sequence (cm10 App Seq).

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

The screenshot shows the 'Call Routing Settings' section of the form. The 'Home Location' field is set to DevConnect. The 'Conference Factory Set' dropdown is set to Select.

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, 'Aura System Manager 10.1', and menu items for 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. A search bar and user profile 'admin' are also visible. 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) as toggle options. The main form fields include: '* System' (cm10), '* Profile Type' (Endpoint), 'Use Existing Endpoints' (unchecked), '* Extension' (72004), '* 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' (unchecked), 'SIP Trunk' (aar), 'SIP URI' (Select), 'Delete on Unassign from User or on Delete User' (checked), and 'Override Endpoint Name and Localized Name' (checked). A 'Commit & Continue' button, a blue 'Commit' button, and a 'Cancel' button are located at the top right of the form area.

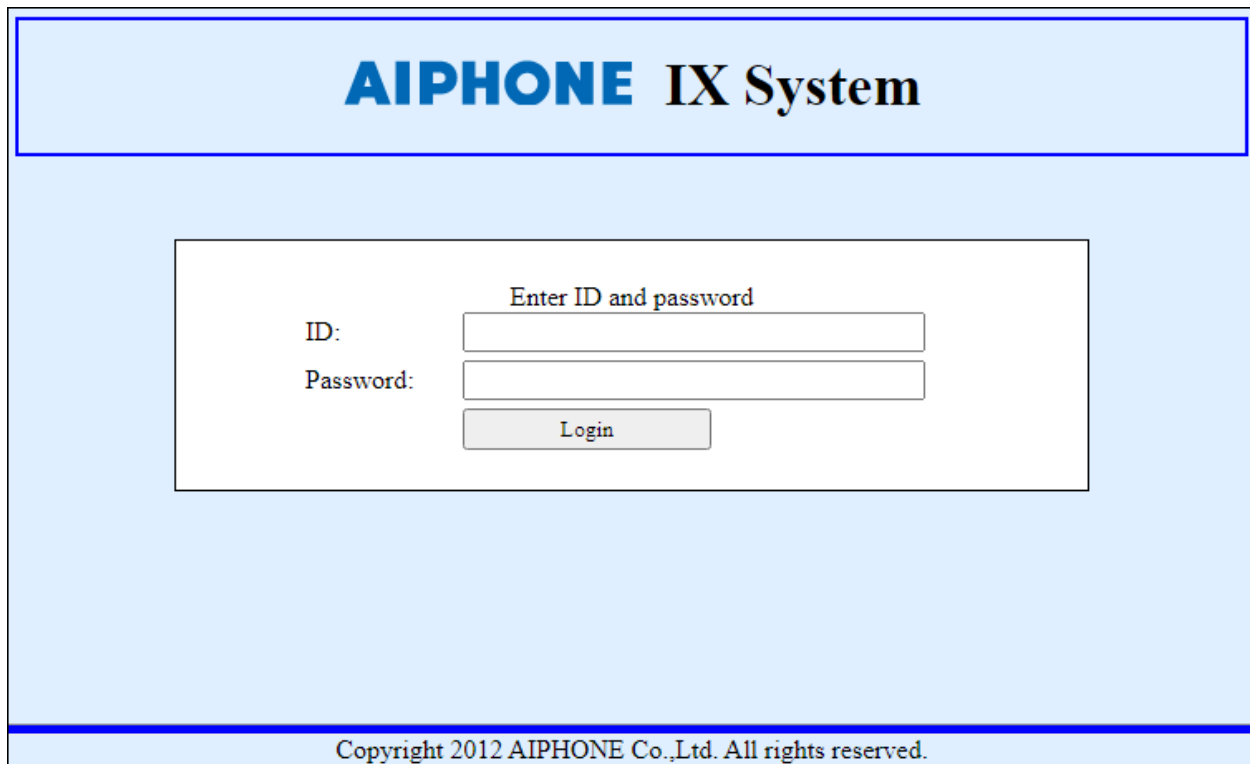
7. Configure Aiphone IX-RS-BT Audio Door Station

This section provides the procedure for configuring IX-RS-BT to provide SIP connectivity to Session Manager. Configuration of IX-RS-BT 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-RS-BT 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-RS-BT SIP extension (e.g., 72004). Input an appropriate **Name**.

The screenshot shows the AIPHONE IX System Setting web interface. The page title is "Station Information" and the category is "Handset Sub Station". The station type is "IX-RS-T". The interface is divided into two main sections: "Station Information" and "Network Settings". The "Station Information" section is currently selected and contains the following fields:

Field	Value	Validation
Number	72004	3-5 digits
Name	IX-RS-BT	1-24 alphanumeric characters(*1)
Location		1-24 alphanumeric characters(*1)

Below the fields, there is a note: (*1)Certain characters may not be displayed correctly on IX-MV, IX-MV7-* and IX-MV7-*T due to font type.

The "Network Settings" section is currently collapsed and contains the following fields:

Field	Value
IP Address	
DNS	
SIP	
Audio	
Packet Priority	
NTP	

The interface also includes a navigation menu on the left with links for "Station Information" (sub-menu: Identification, ID and Password, Language, Time, Expanded System) and "Network Settings" (sub-menu: IP Address, DNS, SIP, Audio, Packet Priority, NTP). An "Update" button is located in the top right corner.

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-RS-BT*).
- **ID:** Set to SIP extension (e.g., *72004*) 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 displays the 'AIPHONE IX System Setting' web interface. The top navigation bar includes 'Category: Handset Sub Station' and 'Station Type: IX-RS-T'. A blue header bar reads 'Network Settings'. On the left, a sidebar lists various settings categories: Station Information, Network Settings, and Call Settings. The main content area is titled 'SIP' and contains two sections: 'SIP Connections' and 'SIP Server'. The 'SIP Connections' section includes 'SIP Signaling Port' (5060) and 'User Agent' (IX-RS-BT). The 'SIP Server' section includes 'SIP Compatibility Mode' (Standard Mode), 'Primary Server' (72004), 'ID' (72004), 'Password' (*****), 'IPv4 Address' (10.64.110.212), 'IPv6 Address' (empty), and 'Port' (5060). Each input field has a red error message below it, such as '1-65535' for the signaling port and '1-24 alphanumeric characters' for the ID and password. An orange 'Update' button is located in the top right corner.

Parameter	Value	Validation
SIP Signaling Port	5060	1-65535
User Agent	IX-RS-BT	1-36 alphanumeric characters
SIP Compatibility Mode	Standard Mode	
Primary Server ID	72004	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-FE:FF:FFFF:FFFF:FFFF: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 interface. The top header shows 'AIPHONE IX System Setting' and 'Station Type: IX-RS-4T'. The left sidebar contains a navigation menu with categories: Station Information, Network Settings, Call Settings, Option Input / Relay, Output Settings, and Function Settings. The main content area is titled 'Network Settings' and is currently on the 'Audio' sub-page. The 'Audio' section includes several configuration fields: 'Audio Codec' (radio buttons for G.711(u-law), G.711(A-law), and G.722), 'Audio RTP Transmission Interval [msec]' (dropdown menu set to 20), and 'RTP Idle Detection Time [sec]' (input field set to 10). Below these are sections for 'SIP Channel' and 'ONVIF Transmit Channel', each with 'RTP Start Port' and 'RTP End Port' fields. The 'Audio Buffer' section at the bottom has 'Packets Buffered at Audio Start' (dropdown set to 1) and 'Maximum Packets Buffered' (dropdown set to 3). Red text provides warnings about RTP End Port values and a note that the buffer must be larger than the audio start buffer. An 'Update' button is located in the top right corner.

Station Information

- Identification
- ID and Password
- Language
- Time
- Expanded System

Network Settings

- IP Address
- DNS
- SIP
- Audio
- Packet Priority
- NTP

Call Settings

- Station Settings
- Called Stations (for Door)
- Call Origination
- Incoming Call

Option Input / Relay

Output Settings

- Option Input
- Relay Output

Function Settings

- Paging Settings
- Email
- CGI
- SIF

Network Settings

•Audio

The "SIP Channel" RTP End Port should be greater than 210 digits from the RTP Start Port.
The "ONVIF Transmit Channel" RTP End Port should be greater than 10 digits from the RTP Start Port.
Changing Audio Codec from G.711(u-law) / G.711(A-law) to G.722, or from G.722 to G.711(u-law) / G.711(A-law) will cause the station to restart after Update is clicked.

Audio Codec G.711(u-law) G.711(A-law) G.722
Audio RTP Transmission Interval [msec] 20
RTP Idle Detection Time [sec] 10
This setting is ignored when transmitting to multiple stations (paging, etc.)
10-180 sec

SIP Channel

RTP Start Port 20000 1-65534
RTP End Port 21000 1-65535

ONVIF Transmit Channel

RTP Start Port 22000 1-65534
RTP End Port 23000 1-65535

Audio Buffer

Packets Buffered at Audio Start 1
Maximum Packets Buffered 3 Maximum Packet Buffer must be larger than Audio Start Buffer.

Update

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.

The screenshot shows the 'Call Settings' page in the AIPHONE IX System Setting web interface. The left sidebar contains a navigation menu with categories like Station Information, Network Settings, Call Settings, Option Input / Relay, and Function Settings. The main content area is titled 'Call Settings' and includes sections for 'Station Information' and 'Called Stations (for Door)'. The 'Station Information' section has a 'Call Button Function' dropdown menu set to 'Call, Answer Call, End Communication'. The 'Called Stations (for Door)' section has an 'Option Input #' dropdown set to 'Group 01'. Below this, there is a table with columns for '#', 'Station Number', 'IPv4 Address', 'IPv6 Address', and 'Station Type'. The table contains three rows, with the first row having values 1, 70103, 10.64.110.212, and VoIP Phone. A red asterisk indicates required settings.

Station Information

Call Button Function:

Cancel Call, End Communication disabled when using Option Input call.

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-FE:FF:FFFF: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
 IX-DA, IX-DB, IX-DBT, IX-BA, IX-BB or IX-BBT must be registered under Called Stations (Door/Sub Stations) to use Door Release while in communication with IX-DA, IX-DB or IX-DBT.
 IX-RS-* and IX-RS-*T cannot place a call to IX-DA, IX-DB, IX-DBT, IX-BA, IX-BB or IX-BBT even IX-DA, IX-DB, IX-DBT, IX-BA, IX-BB or IX-BBT is under Called Stations.

#	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-RS-BT Audio Door Station.

1. Verify that IX-RS-BT 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-RS-BT 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-RS-BT) with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Aiphone IX-RS-BT 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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