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.

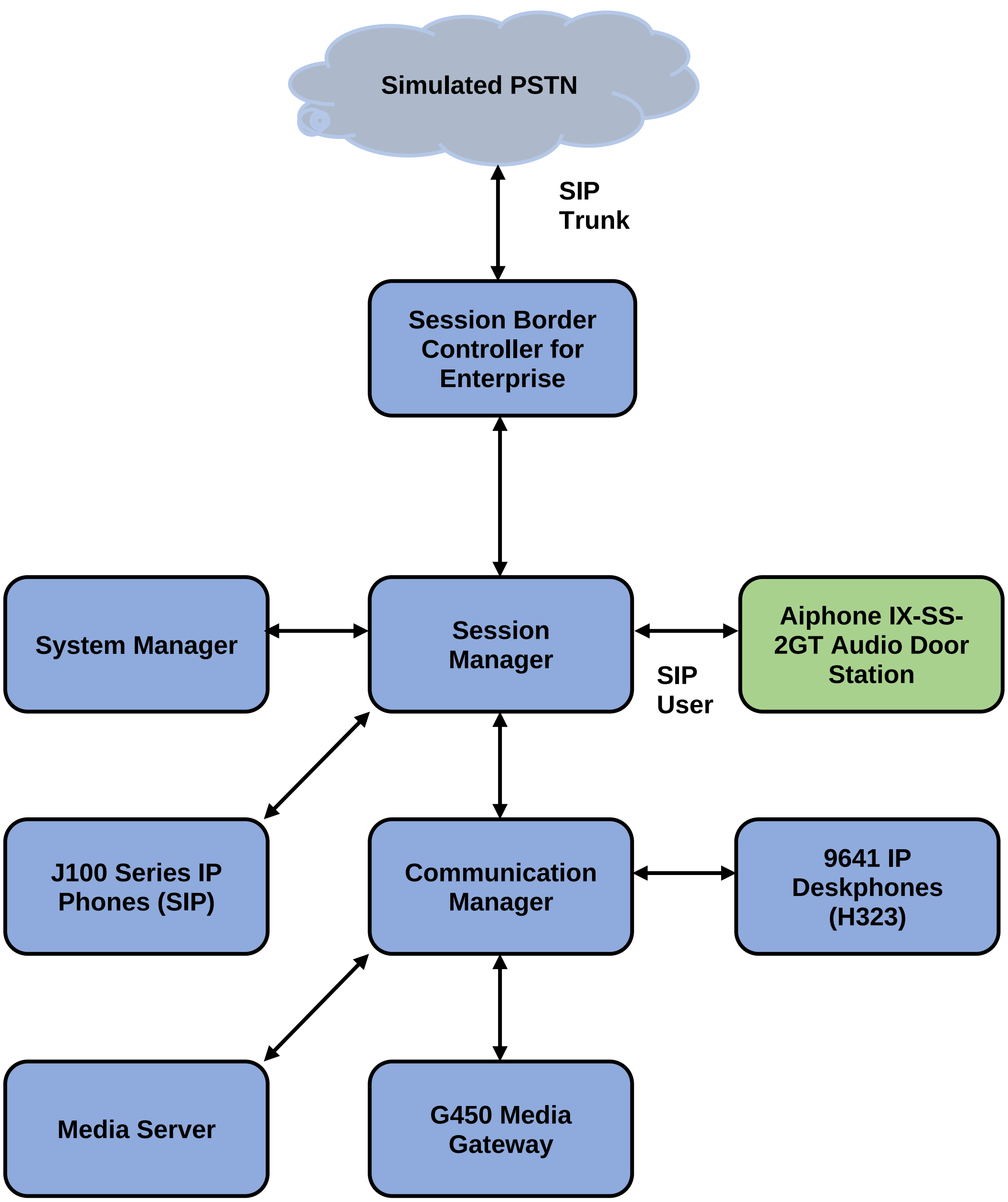


Figure 1: Avaya SIP Telephony Network with Aiphone IX-SS-2GT Audio Door Station

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 SRTCP: 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      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                      RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 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
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  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

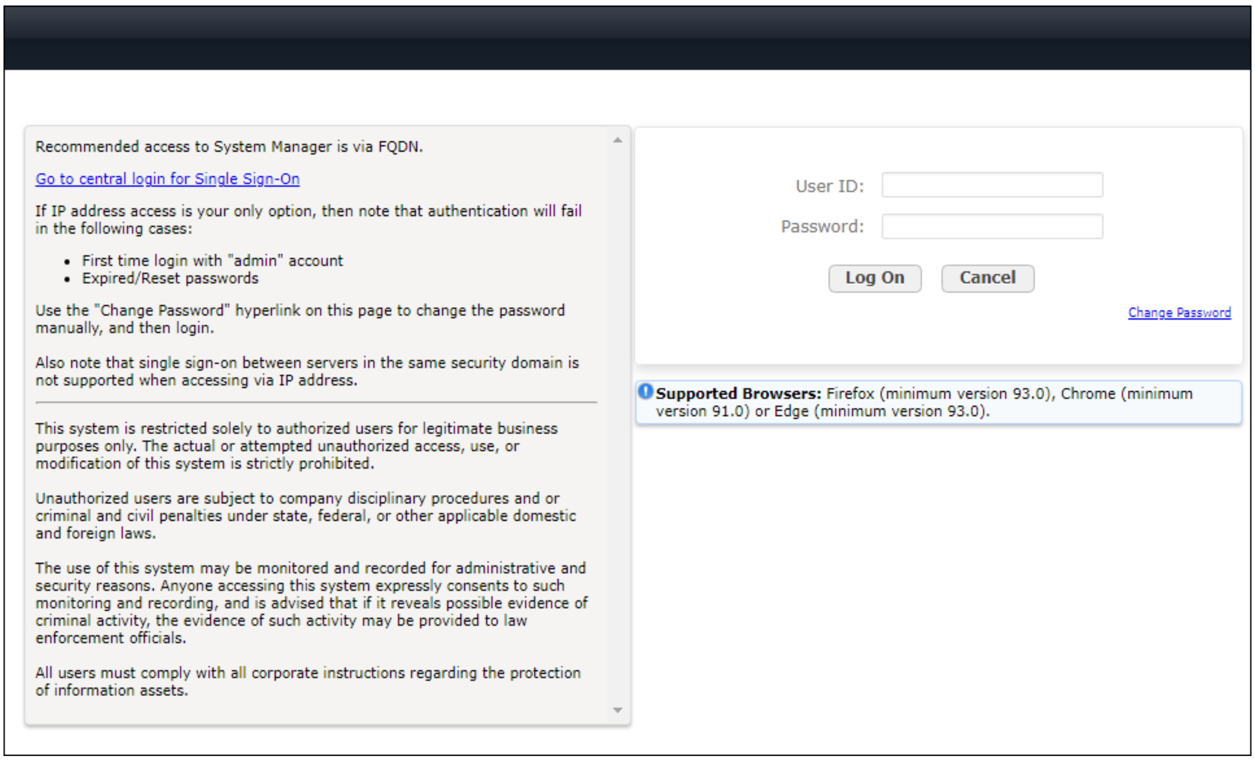
This section covers the procedure for adding a SIP user in Session Manager. The configuration covers:

- 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.

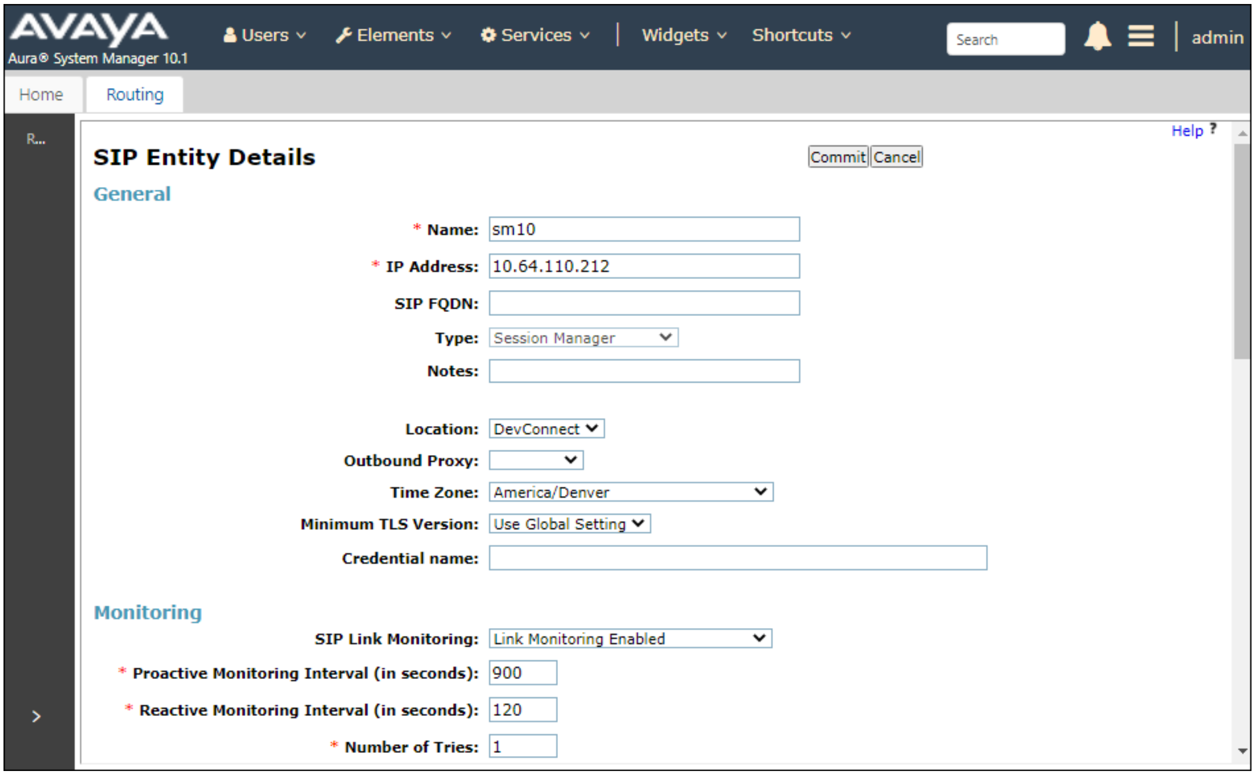
6.1. Launch System Manager

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.

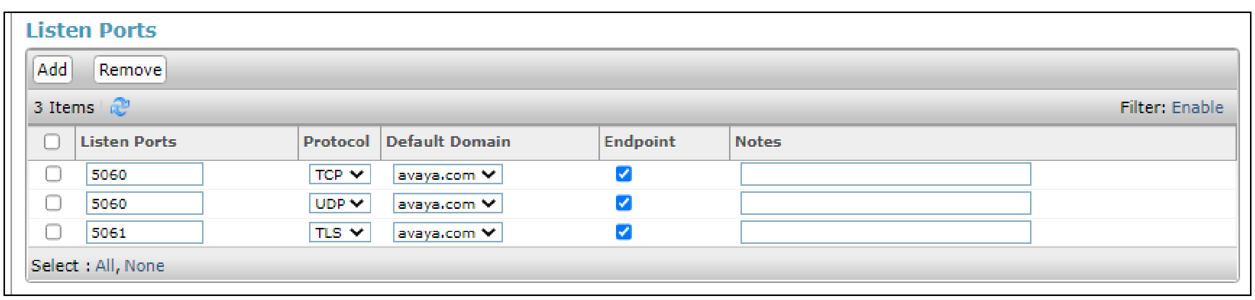


6.2. Set Network Transport Protocol for IX-SS-2GT

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

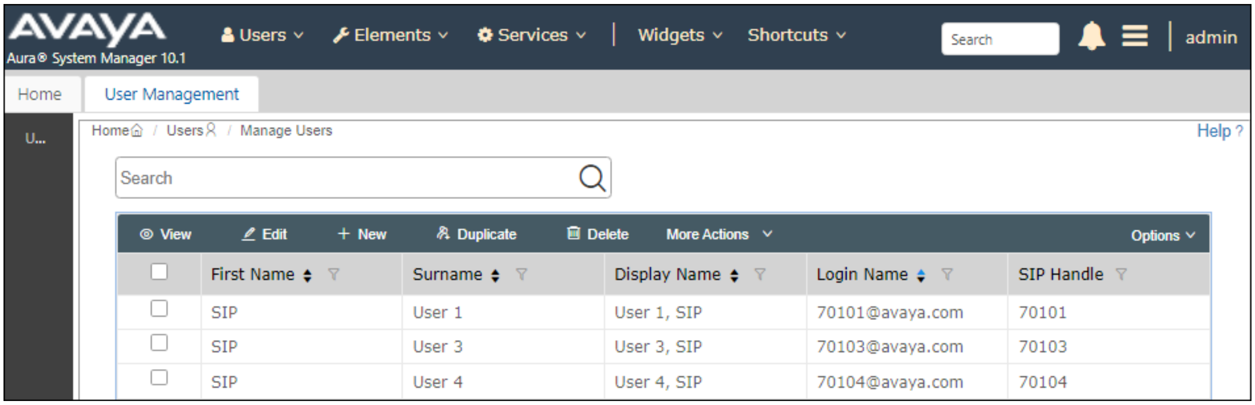


Scroll down to the **Listen Ports** section and verify that the transport network protocol used by IX-SS-2GT is specified in the list below. For the compliance test, the solution used UDP network transport.



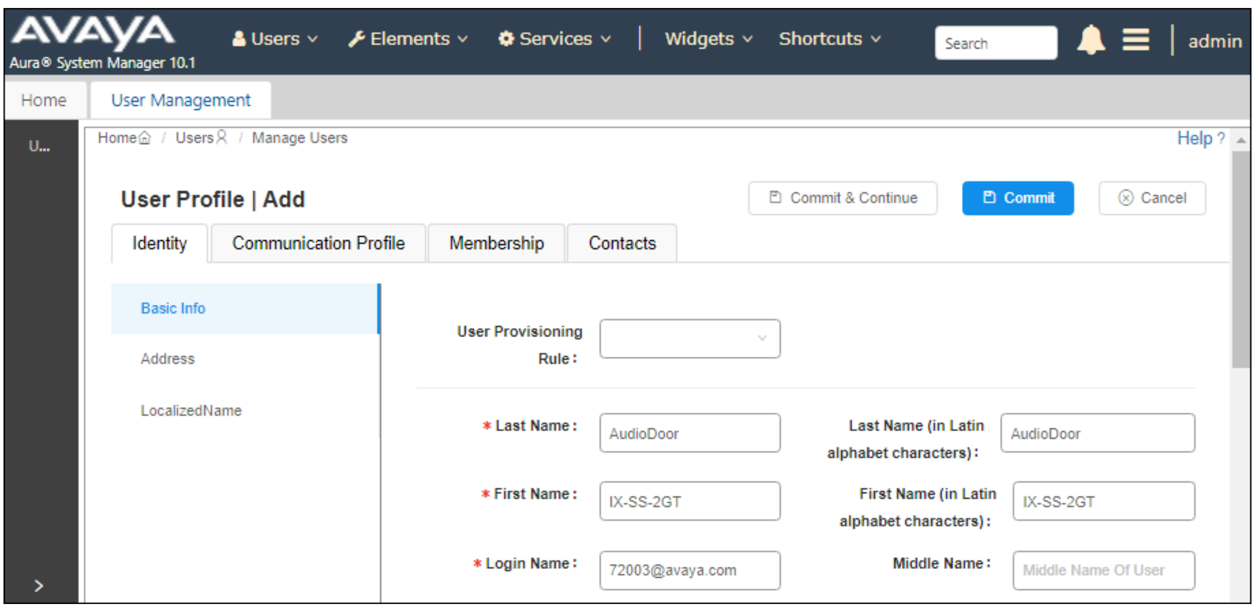
6.3. Administer SIP User

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.



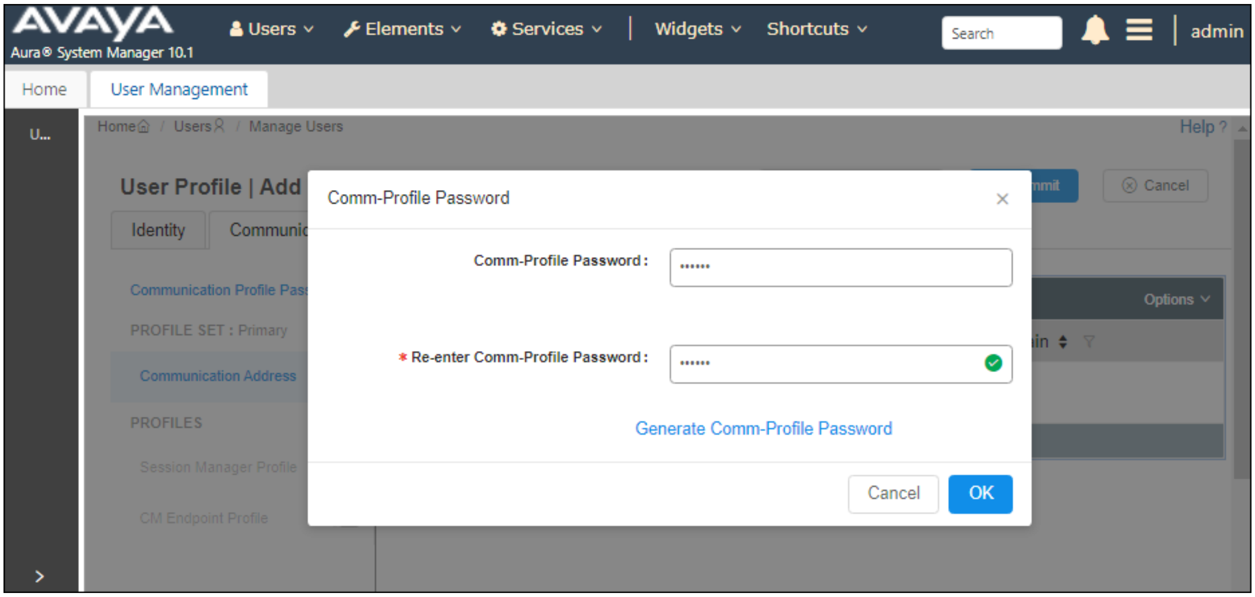
6.3.1. Identity

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



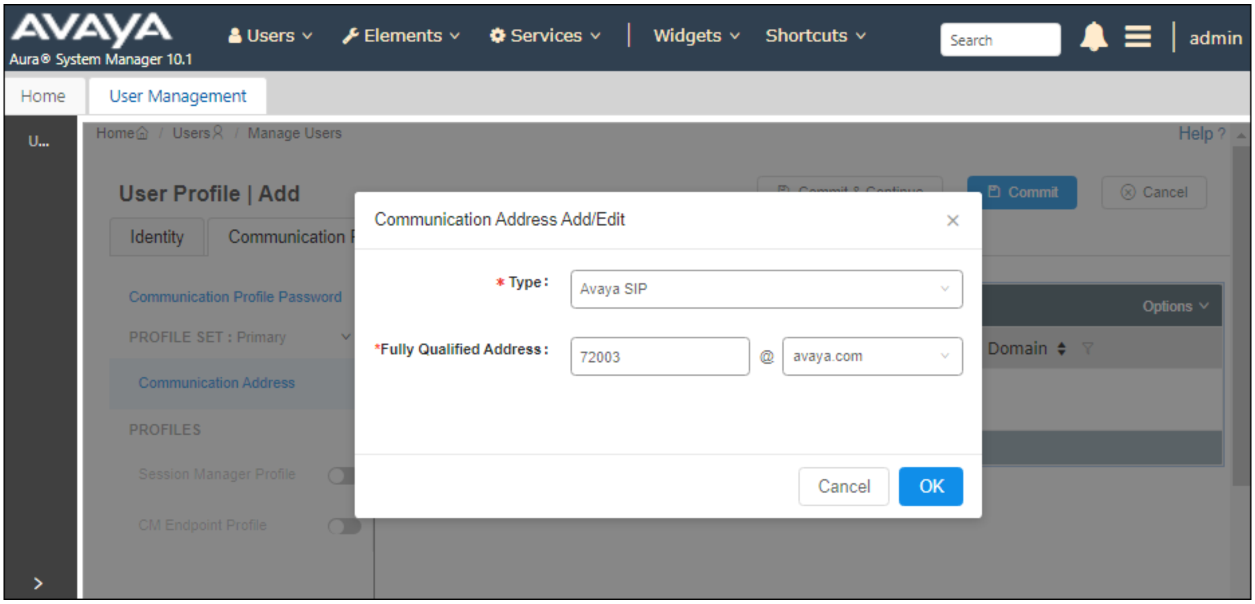
6.3.2. Communication Profile Password

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**.



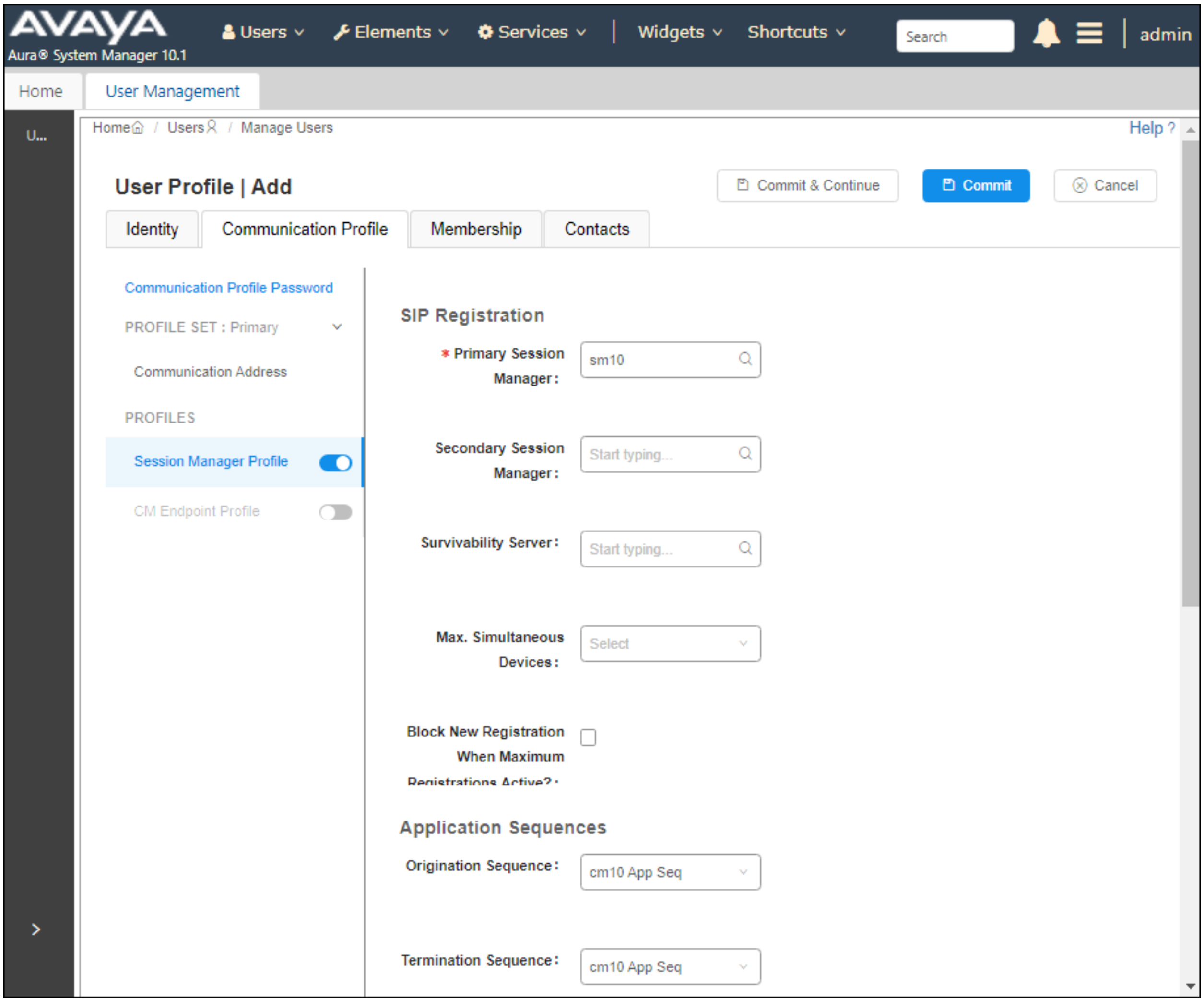
6.3.3. Communication Address

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**.

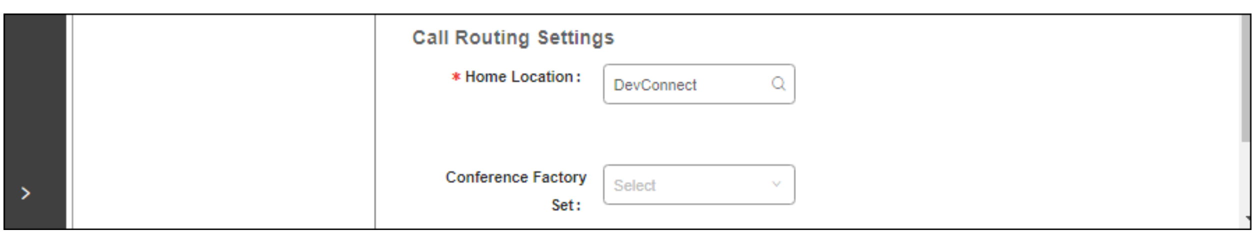


6.3.4. Session Manager Profile

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.

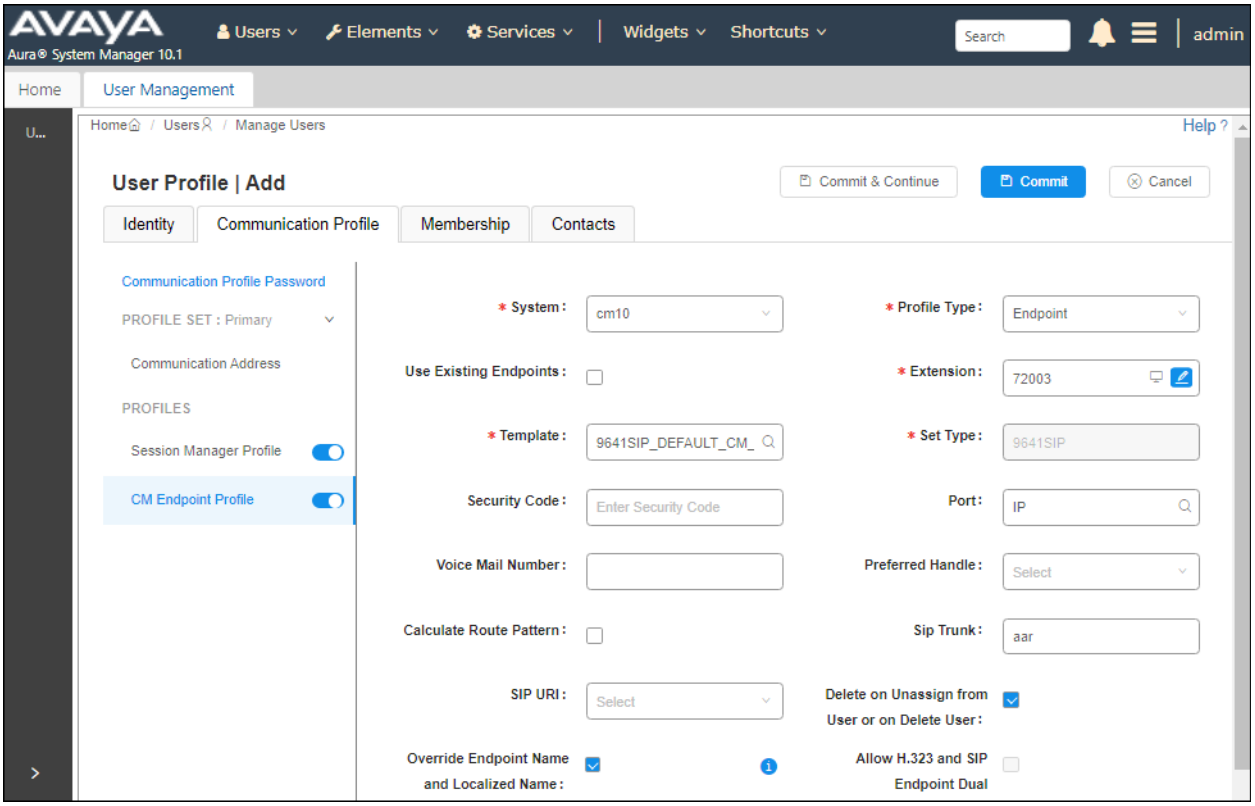


Scroll down to the **Call Routing Settings** section to configure the **Home Location**.



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.



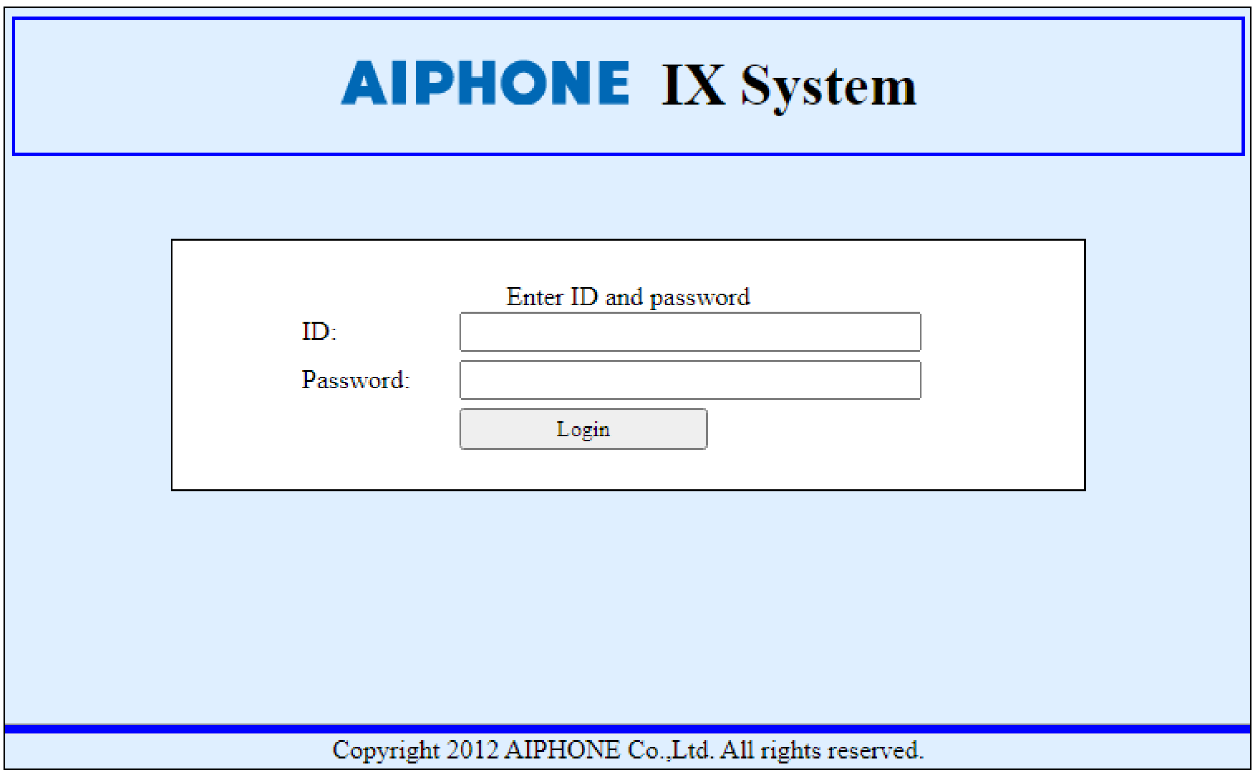
7. Configure Aiphone IX-SS-2GT Audio Door Station

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

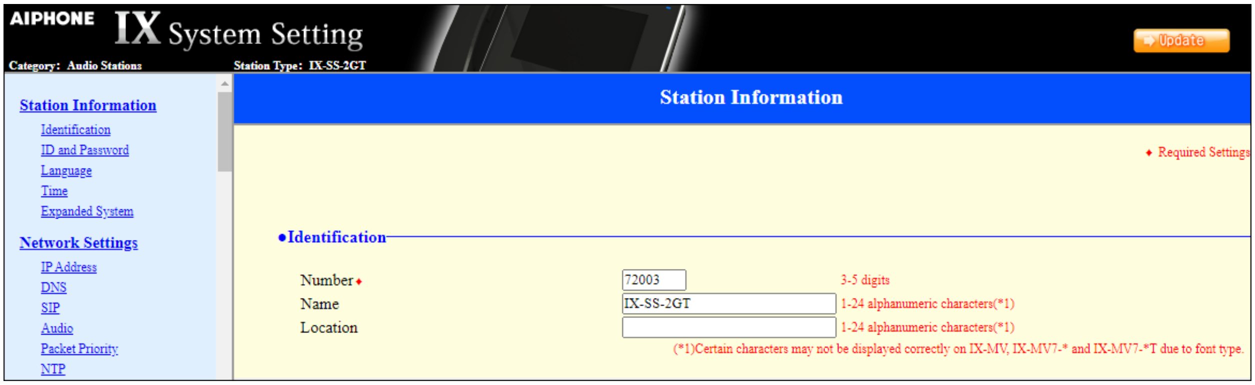
7.1. Log into Aiphone IX System Web Interface

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.



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**.

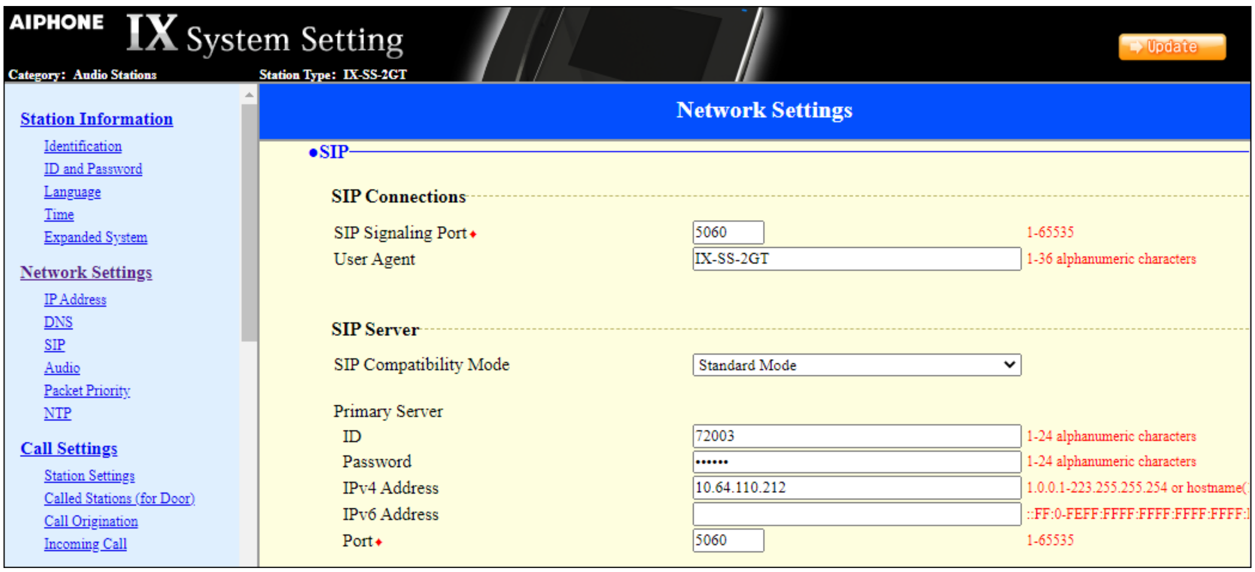


7.3. Administer SIP Parameters

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

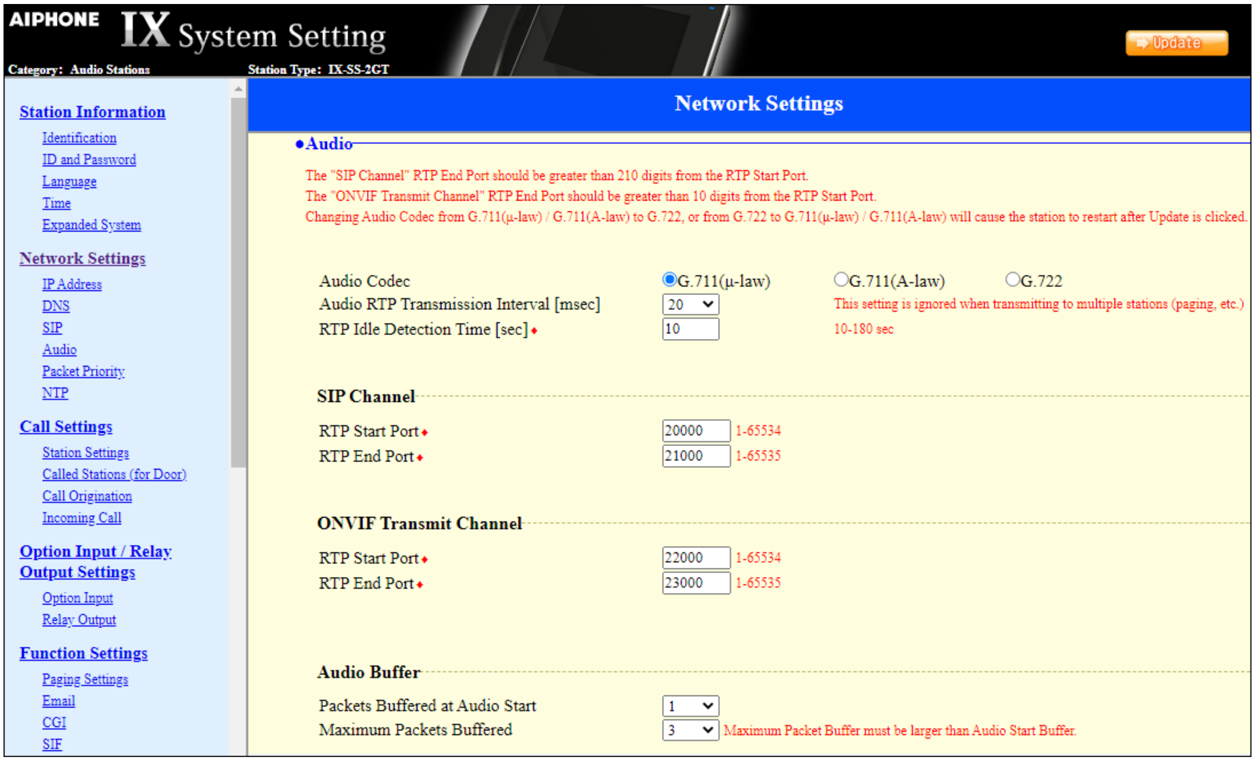
- **SIP Signaling Port:** Set to *5060*.
- **User Agent:** Enter desired value (e.g., *IX-SS-2GT*).
- **ID:** Set to SIP extension (e.g., *72003*) from **Section 6.3.1.**
- **Password:** Enter SIP password from **Section 6.3.2**.
- **IPv4 Address:** Set to signaling IP address of Session Manager (e.g.,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

Click **Update** to save changes.



7.4. Administer Audio Settings

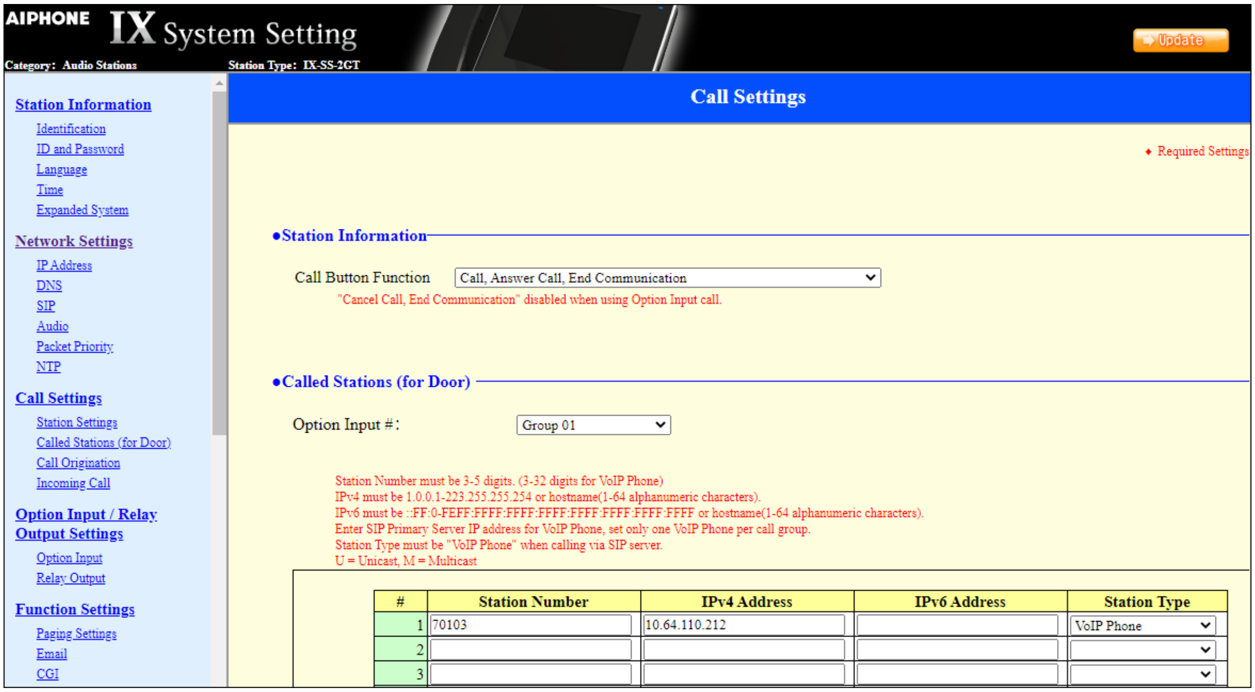
Navigate to **Network Settings → Audio** in the left pane and set **Audio Codec** to select *G.711 (u-law).*



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.

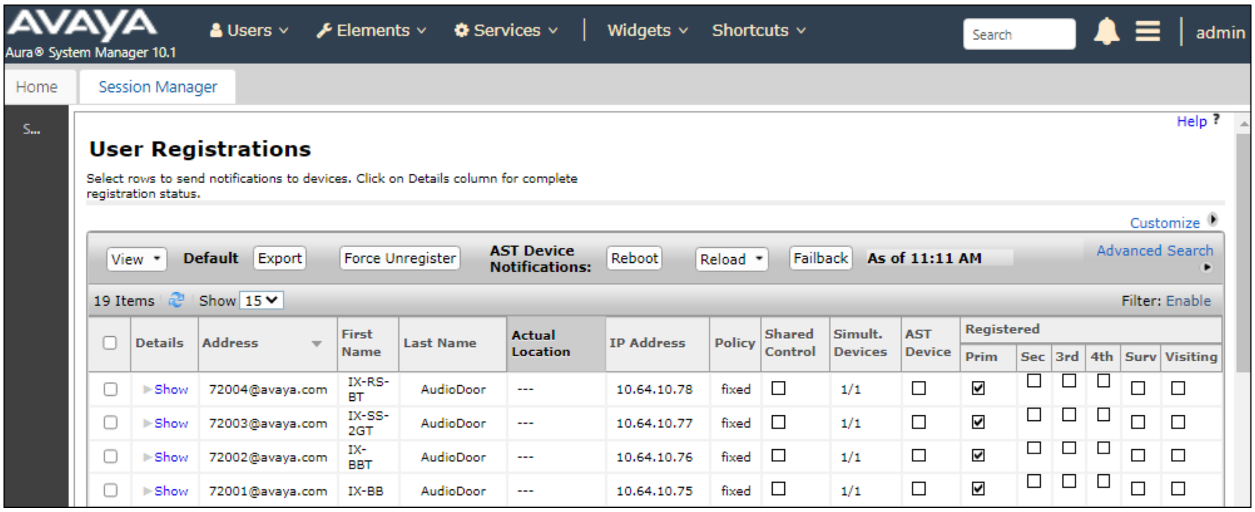
In the **Called Stations (for Door)** section, add an entry that specifies the number that should be dialed when the call button is pressed. Set the **Station Number** to the called number (e.g., *70103*), set the **IPv4 Address** to the signaling IP address of Session Manager (e.g., *10.64.110.212*),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



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.



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.