



DevConnect Program

Application Notes for Aiphone IX Series 2 Audio Door Stations (IX-SS-2GT) with Avaya IP Office Server Edition - 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 IP Office Server Edition 11.1 and Avaya IP Office 500V2 Expansion System 11.1. 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 IP Office 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 Avaya DevConnect Program.

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 IP Office Server Edition 11.1 and Avaya IP Office 500V2 Expansion System 11.1. 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 IP Office 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 IP Office.
- 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 to either IP Office Server Edition or IP Office 500 V2 Expansion System (not simultaneously).

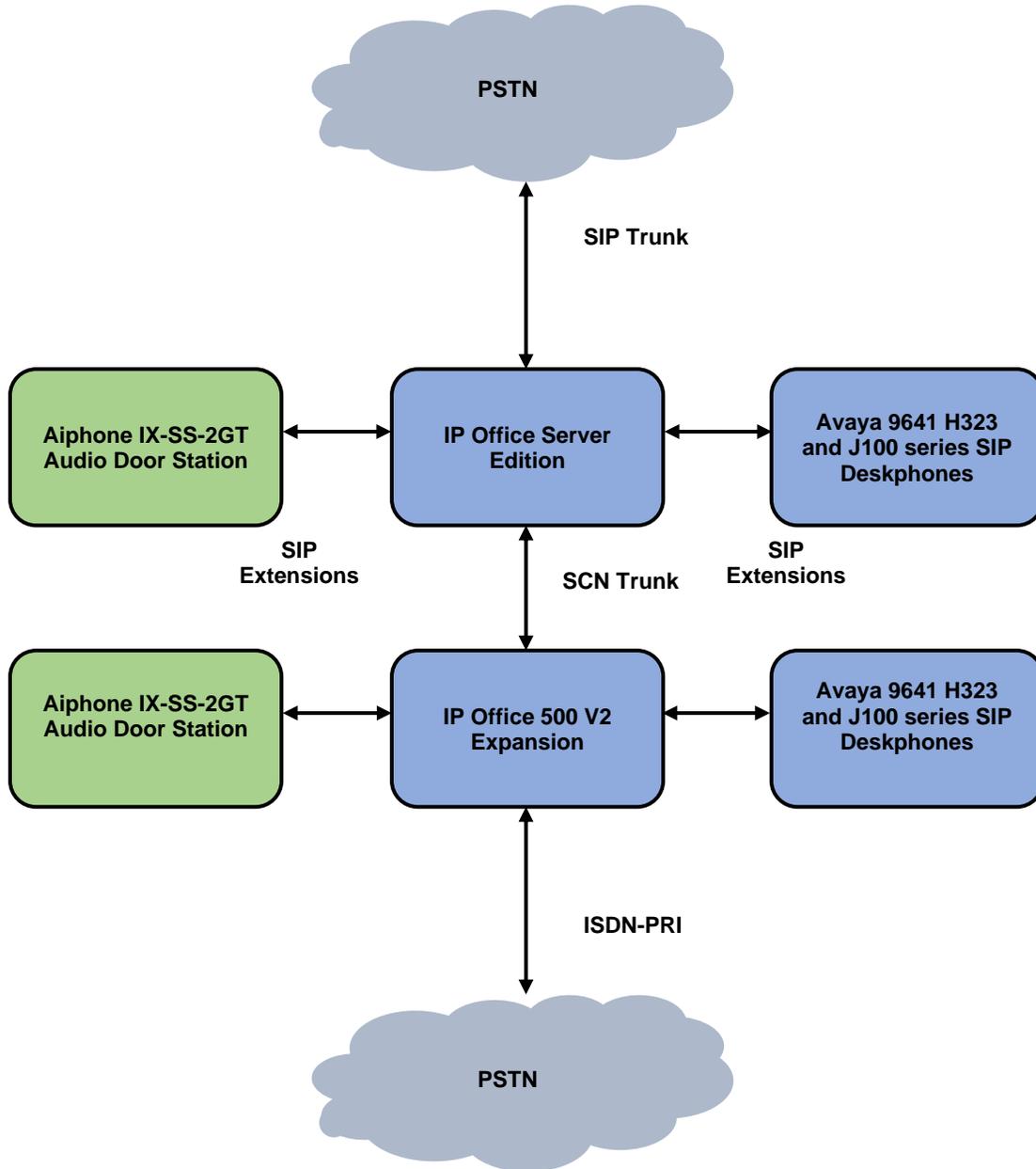


Figure 1: Avaya SIP Telephony Network with Aiphone IX-SS-2GT Audio Door Station

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office Server Edition	11.1.2.4.0 build 18 (FP2 SP4)
Avaya IP Office 500V2 Expansion System	11.1.2.4.0 build 18 (FP2 SP4)
Avaya 96x1 Series IP Deskphones	6.8.5.2.3 (H.323)
Avaya J100 Series IP Phones	4.0.10.3.2 (SIP)
Aiphone IX-SS-2GT Audio Door Station	7.00

Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and when deployed with IP Office Server Edition in all configurations.

5. Configure Avaya IP Office Server Edition

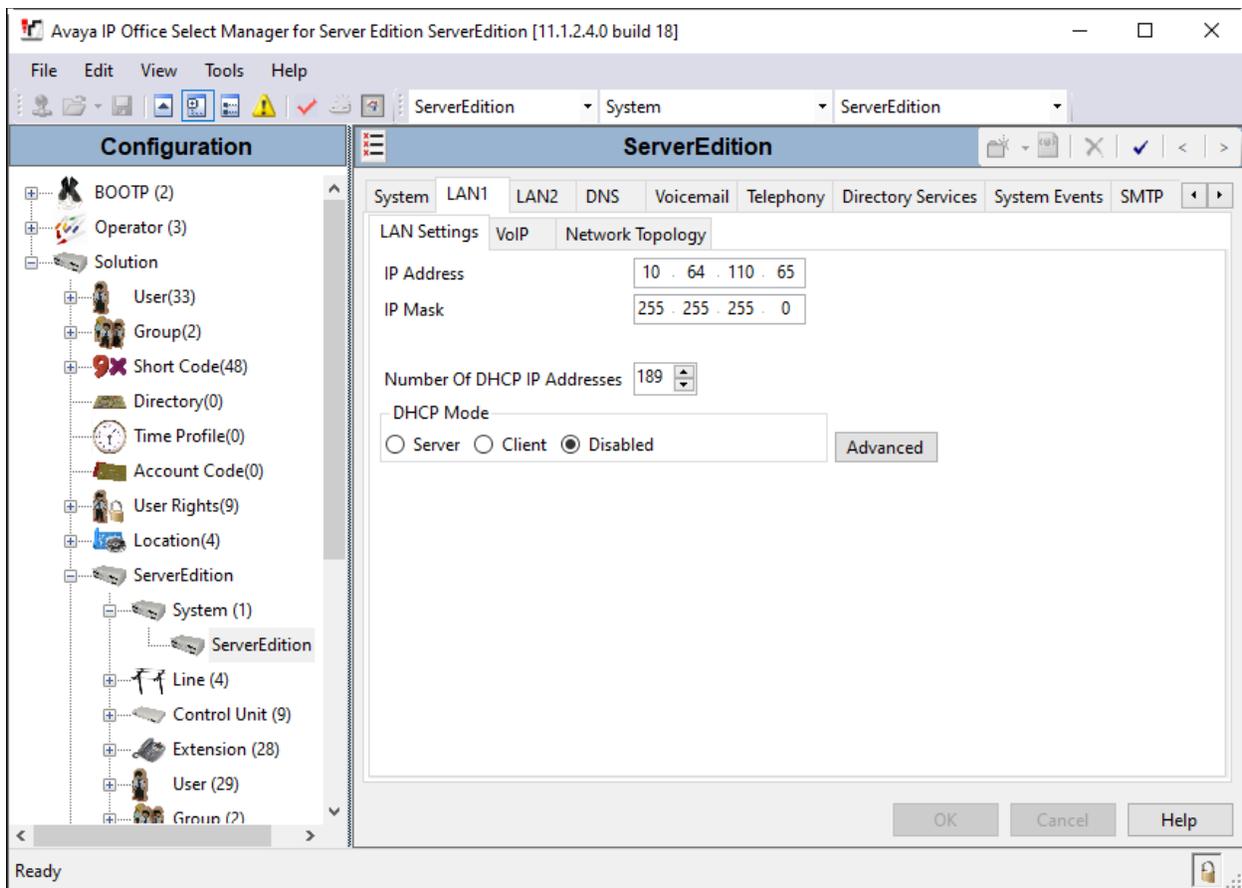
This section provides the procedures for configuring Avaya IP Office Server Edition. The procedures include the following areas:

- Obtain LAN IP Address
- Administer SIP Registrar
- Administer SIP Extension for IX-SS-2GT
- Administer SIP User for IX-SS-2GT

Note: This section covers the configuration of Avaya IP Office Server Edition, but the configuration is the same for Avaya IP Office 500 V2 Expansion System.

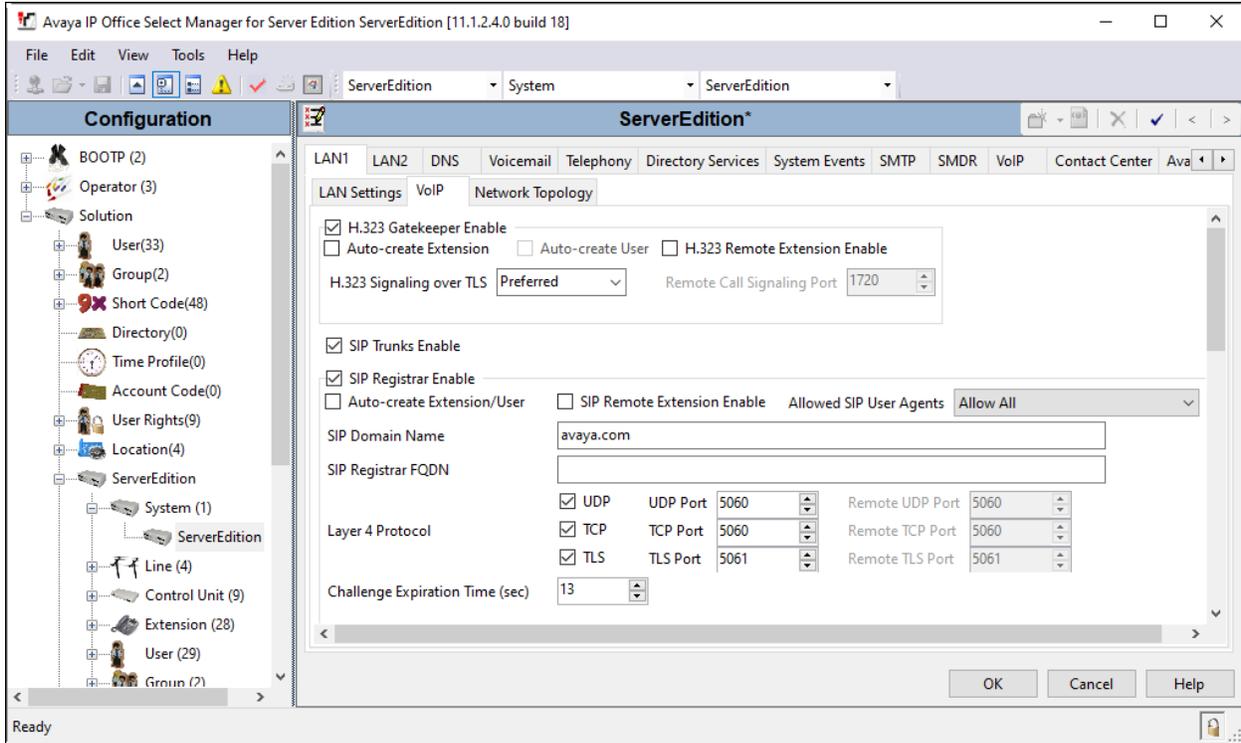
5.1. Obtain LAN IP Address

From a PC running the IP Office Manager application, on the configuration tree in the left pane, select **System** to display the **System** screen for the IP Office Server Edition in the right pane. Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the **IP Address**, which will be used later to configure IX-SS-2GT.



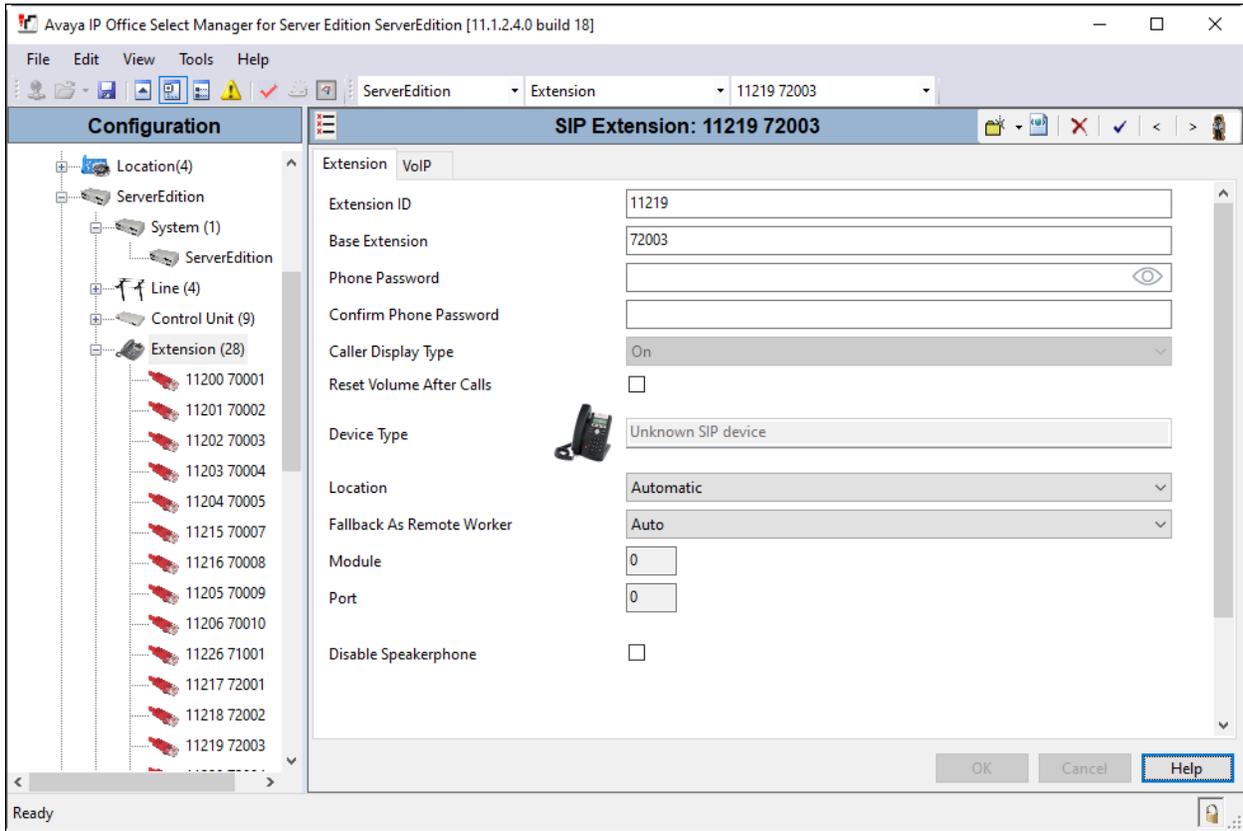
5.2. Administer SIP Registrar

Select the **VoIP** sub-tab. Ensure that **SIP Registrar Enable** is checked and enter a valid **SIP Domain Name**. In the compliance testing, the **SIP Domain Name** field was set to *avaya.com*. UDP transport protocol was enabled for the **Layer 4 Protocol**, which was used by IX-SS-2GT.

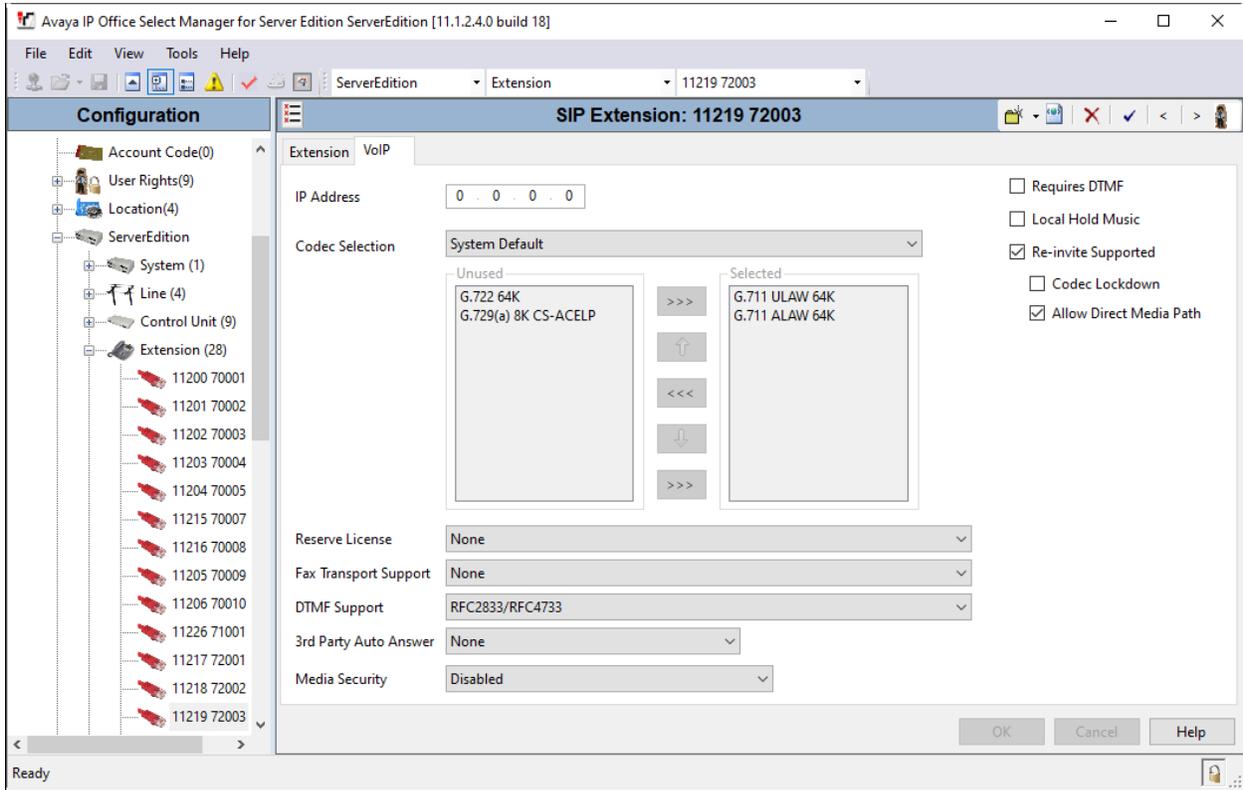


5.3. Administer SIP Extension for IX-SS-2GT

From the configuration tree in the left pane, right-click on **Extension** and select **New** → **SIP** from the pop-up list to add a new SIP extension. Enter the desired extension for the **Base Extension** field as shown below. In this example, IX-SS-2GT was assigned extension 72003. This is the extension that IX-SS-2GT will use to register with IP Office Server Edition.

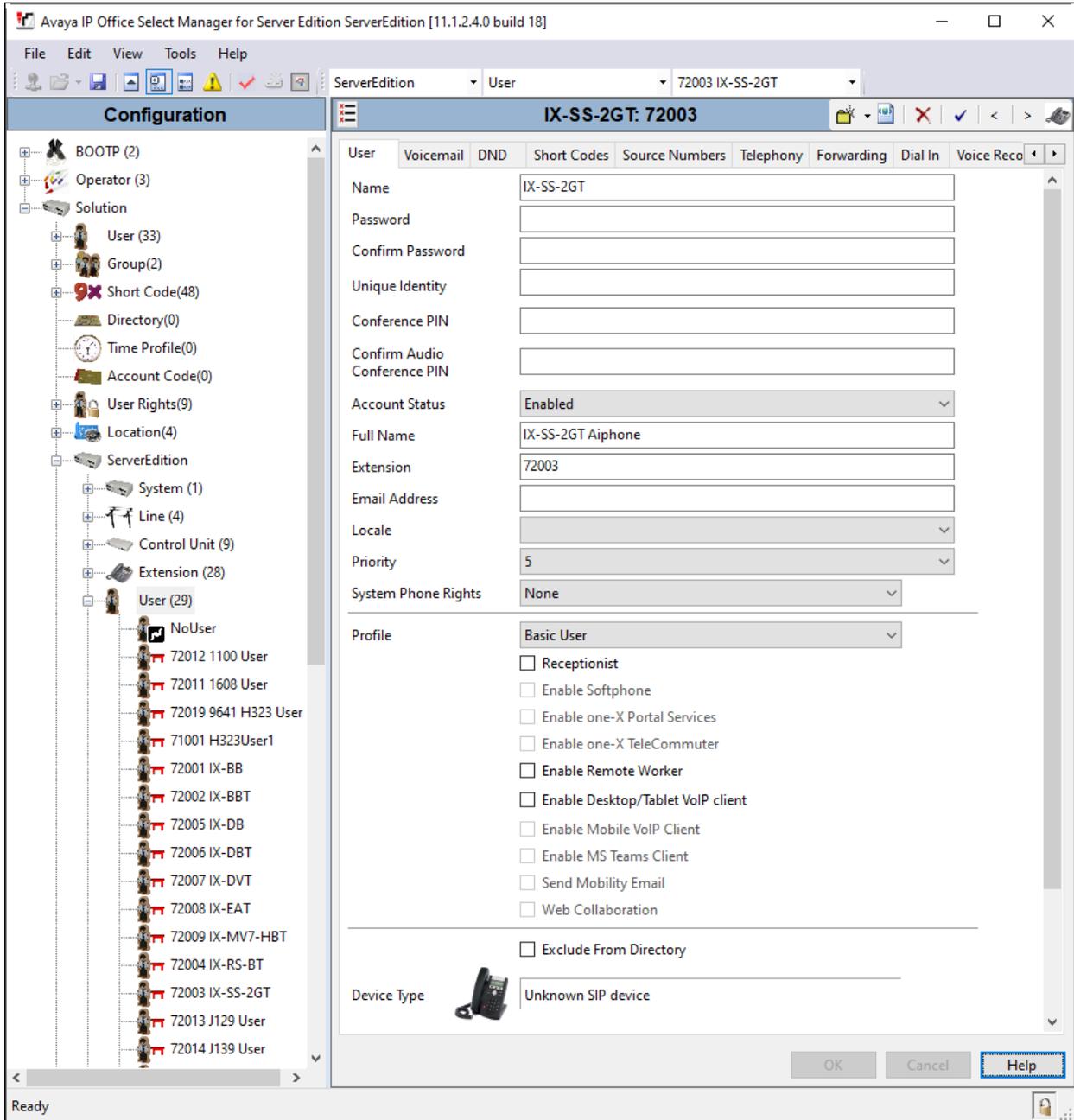


Select the **VoIP** tab and retain the default values. During the compliance test, IX SS-2GT was tested with *G.711 ULaw* codec. Enable **Allow Direct Media Path** so that audio/RTP flows directly between two SIP endpoints without using media resources in Avaya IP Office Server Edition. **Media Security** was *disabled* for IX-SS-2GT.

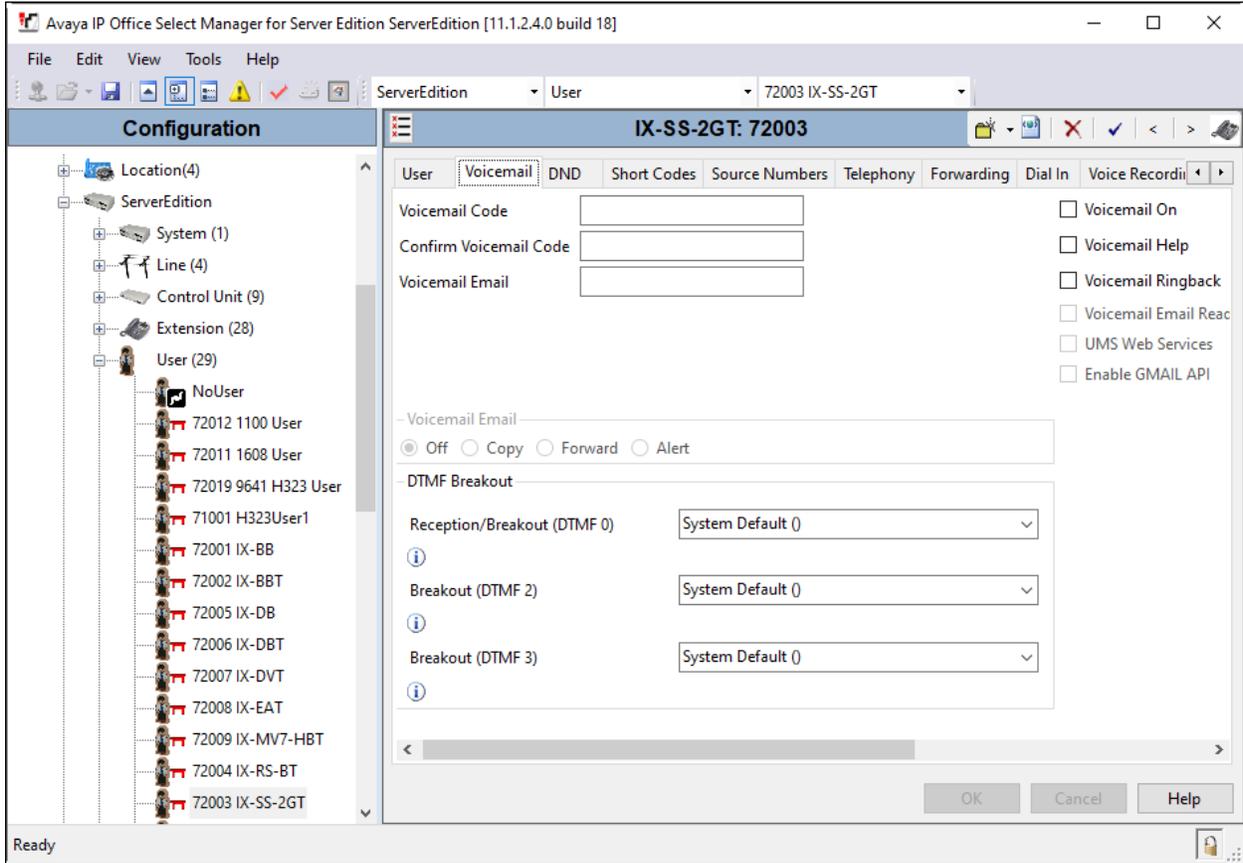


5.4. Administer SIP User for IX-SS-2GT

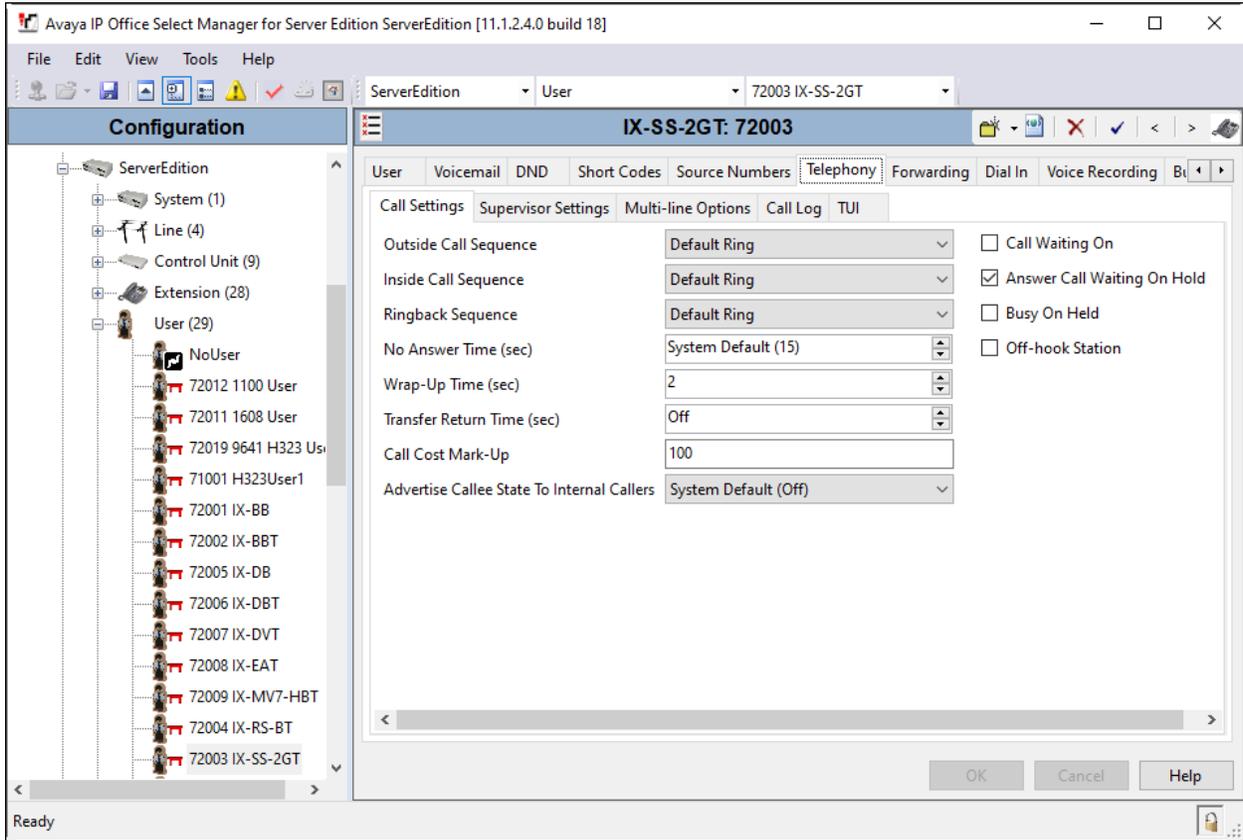
From the configuration tree in the left pane, right-click on **User** and select **New** from the pop-up list. Enter desired values for the **Name** and **Full Name** fields. For the **Extension** field, enter the SIP extension from **Section 5.3** (e.g., *72003*).



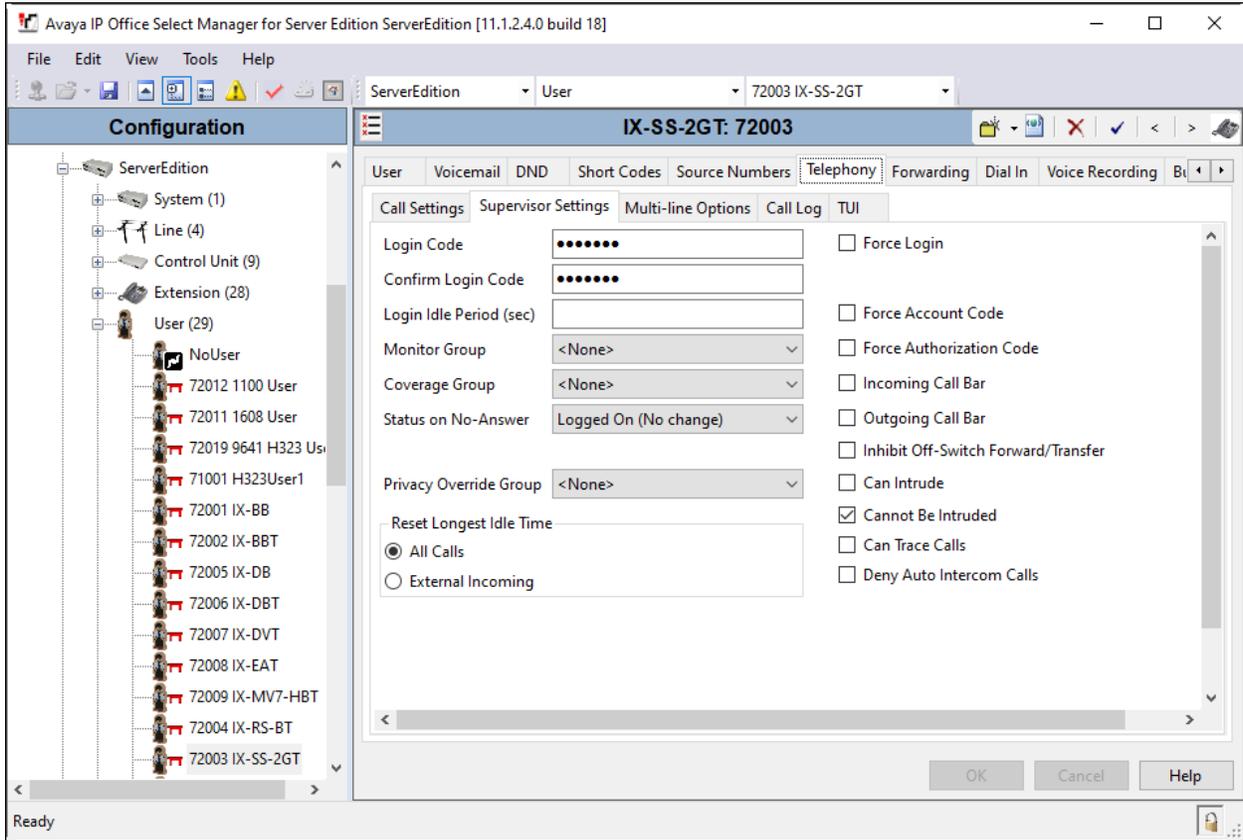
Select the **Voicemail** tab and disable voicemail for IX-SS-2GT



Select the **Telephony** tab followed by the **Call Settings** sub-tab. Note the settings below for the user.



Select the **Supervisor Settings** sub-tab and enter a desired **Login Code**. The **Login Code** is the password that will be used by IX-SS-2GT to register with IP Office Server Edition.



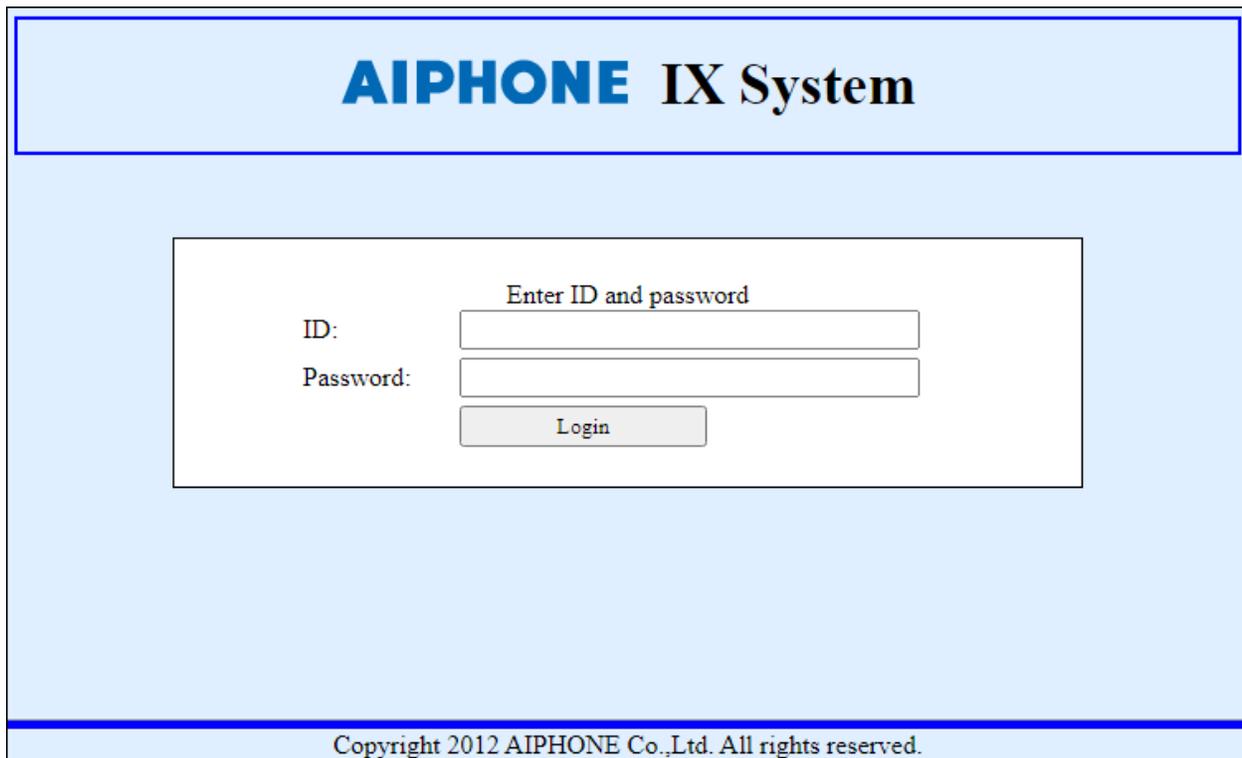
6. Configure Aiphone IX-SS-2GT Audio Door Station

This section provides the procedure for configuring IX-SS-2GT to provide SIP connectivity to IP Office. 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

6.1. 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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6.2. 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 displays the AIPHONE IX System Setting web interface. The page title is "Station Information" and the category is "Audio Stations". The station type is "IX-SS-2GT". The left sidebar contains a navigation menu with "Station Information" selected, showing sub-items: Identification, ID and Password, Language, Time, Expanded System, Network Settings, IP Address, DNS, SIP, Audio, Packet Priority, and NTP. The main content area is titled "Station Information" and contains a "Required Settings" section with 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.

6.3. 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 5.3**.
- **Password:** Enter SIP password from **Section 5.4**.
- **IPv4 Address:** Set to signaling IP address of IP Office (e.g., *10.64.110.65*).
- **Port:** Set to *5060*.

Click **Update** to save changes.

The screenshot displays the 'AIPHONE IX System Setting' web interface. The left sidebar contains a navigation menu with categories: Station Information, Network Settings, and Call Settings. The 'Network Settings' category is selected, and the 'SIP' sub-section is active. The main content area shows the following configuration fields:

Parameter	Value	Validation/Range
SIP Signaling Port	5060	1-65535
User Agent	IX-SS-2GT	1-36 alphanumeric character
SIP Compatibility Mode	Standard Mode	Dropdown menu
Primary Server ID	72003	1-24 alphanumeric character
Password	*****	1-24 alphanumeric character
IPv4 Address	10.64.110.65	1.0.0.1-223.255.255.254 or h
IPv6 Address		::FF:0-FE:FF:FFFF:FFFF:FF
Port	5060	1-65535

An 'Update' button is visible in the top right corner of the interface.

6.4. 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 main title is "AIPHONE IX System Setting" with an "Update" button in the top right. The left sidebar contains a navigation menu with categories: Station Information, Network Settings, Call Settings, Option Input / Relay, Output Settings, and Function Settings. The "Network Settings" category is selected, and the "Audio" sub-section is active. The main content area is titled "Network Settings" and contains the following configuration options:

- Audio**
 - Warning: The "SIP Channel" RTP End Port should be greater than 210 digits from the RTP Start Port.
 - Warning: The "ONVIF Transmit Channel" RTP End Port should be greater than 10 digits from the RTP Start Port.
 - Warning: 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
 - Note: This setting is ignored when transmitting to multiple stations (paging, etc.) 10-180 sec
- SIP Channel**
 - RTP Start Port: 20000 (range 1-65534)
 - RTP End Port: 21000 (range 1-65535)
- ONVIF Transmit Channel**
 - RTP Start Port: 22000 (range 1-65534)
 - RTP End Port: 23000 (range 1-65535)
- Audio Buffer**
 - Packets Buffered at Audio Start: 1
 - Maximum Packets Buffered: 3 (Note: Maximum Packet Buffer must be larger than Audio Start Buffer.)

6.5. 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., 72015), set the **IPv4 Address** to the signaling IP address of IP Office (e.g., 10.64.110.65), 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	72015	10.64.110.65		VoIP Phone
2				
3				

7. Verification Steps

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

1. Verify that IX-SS-2GT has successfully registered with with IP Office. Launch **IP Office System Status** and navigate to **Extensions** → **<SIP Extension>**, where **<SIP Extension>** is the IX-SS-2GT extension. Verify that the **Current State** is *Idle* as shown below

The screenshot shows the Avaya IP Office System Status application window. The title bar reads "Avaya IP Office System Status - ServerEdition (10.64.110.65) - IP Office Linux PC 11.1.2.4.0 build 18". The main window has a blue header with the Avaya logo and the title "IP Office System Status". Below the header is a menu bar with "Help", "Snapshot", "LogOff", "Exit", and "About". A left-hand navigation pane contains a tree view with categories: System, Alarms (2), Extensions (7), Trunks (4), Active Calls, Resources, Voicemail, IP Networking, and Locations. The "Extensions (7)" category is expanded, and extension 72003 is selected. The main content area displays the "Extension Status" for extension 72003. The status is "Idle". Below the status information is a table with columns: Call Ref, Current State, Time in State, Calling Number or Called Number, Direction, and Other Party on Call. The table shows one row with "Idle" state and "00:01:42" time in state. At the bottom of the window, there are buttons for "Trace", "Trace All", "Pause", "Ping", "Call Details", "Print...", and "Save As...". The system tray at the bottom right shows the time "2:30:34 PM" and the status "Online".

Call Ref	Current State	Time in State	Calling Number or Called Number	Direction	Other Party on Call
	Idle	00:01:42			

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

8. Conclusion

These Application Notes describe the administration steps required to integrate Aiphone IX Series 2 Audio Door Stations (IX-SS-2GT) with Avaya IP Office. The Aiphone IX-SS-2GT Audio Door Station successfully registered with IP Office as a SIP endpoint and audio calls were verified. All test cases executed passed with no observations noted.

9. References

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

Avaya product documentation is available at <https://support.avaya.com>.

[1] *Administering Avaya IP Office using Manager*, Release 11.1, available at <http://support.avaya.com> as an HTML document.

Aiphone product documentation is available at <https://www.aiphone.com>.

[2] *Aiphone IX Door Stations Web Setting Manual*, Software version 6.00 or later, available from Aiphone.

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