



Avaya Solution & Interoperability Test Lab

Application Notes for Aiphone IX Series Video Door Stations (IX-EA) R5.4 and Avaya Aura[®] Communication Manager and Avaya Aura[®] Session Manager R8.1 – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Aiphone IX Series Video Door Stations (IX-EA) which was compliance tested with Avaya Aura[®] Communication Manager and Avaya Aura[®] Session Manager.

The overall objective of the interoperability compliance testing was to verify Aiphone IX Series Video Door Stations (IX-EA) functionalities in an environment comprised of Avaya Aura[®] and various Avaya endpoints. Aiphone IX Series Video Door Stations are SIP based door phones.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required for Aiphone IX Series Video Door Stations to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. During the compliance testing, Aiphone IX-EA was used.

The Aiphone IX Series Video Door Stations (IX-EA) are part of Aiphone IX Series Door Stations. The Video Door Stations, IX-EA, act as SIP phones when connected to Avaya Aura®. The Video Door Stations come in both surface mount and flush mount varieties. All door stations have dry contacts that can be used to release doors when activated by another intercom or phone. The dry contacts can also be used to trigger external signaling devices, such as strobes.

During the compliance test, Aiphone IX-EA registered as a 3rd party SIP phone using UDP to Avaya Aura® Session Manager.

2. General Test Approach and Test Results

The focus of this interoperability compliance testing was to verify that the Aiphone IX-EA can register as a SIP endpoint on Session Manager, and is able to originate and receive audio and video calls to and from the Avaya Aura® system.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and Aiphone did not utilize secure capabilities.

2.1. Interoperability Compliance Testing

The general test approach was to place calls to and from, Aiphone IX-EA, and exercise basic telephone operations. The main objectives were to verify the following:

- Registration
- Calls to Avaya SIP Audio & Video endpoints
- Calls to Avaya H.323 Audio endpoints
- Calls to Avaya Digital & Analog endpoints
- Calls to PSTN via SIP Trunks
- Call termination (origination/destination)
- Serviceability

2.2. Test Results

The test objectives were verified, and the features tested worked as expected.

2.3. Support

For technical support on Aiphone IX-EA, please contact Aiphone via the following:

Japan

- Web: <https://www.aiphone.co.jp/>
- Phone: 052-228-9961

USA, Canada

- Web: <https://www.aiphone.com/home>
- Email: tech@aiphone.com
- Phone: 800-692-0200

France

- Web: <https://www.aiphone.fr/>
- Phone: 01 69 11 46 00

Australia, New Zealand

- Web: <https://www.aiphone.com.au/>
- Phone: (02)80364507

Singapore

- Web: <http://www.aiphone.com.sg/>
- Email: admin@aiphone.com.sg
- Phone: 6534-1135

United Kingdom

- Web: <https://www.aiphone.co.uk/>
- Phone: 020-7507-6250

3. Reference Configuration

Figure 1 illustrates a sample configuration consisting of Avaya Aura® components and Aiphone IX-EA.

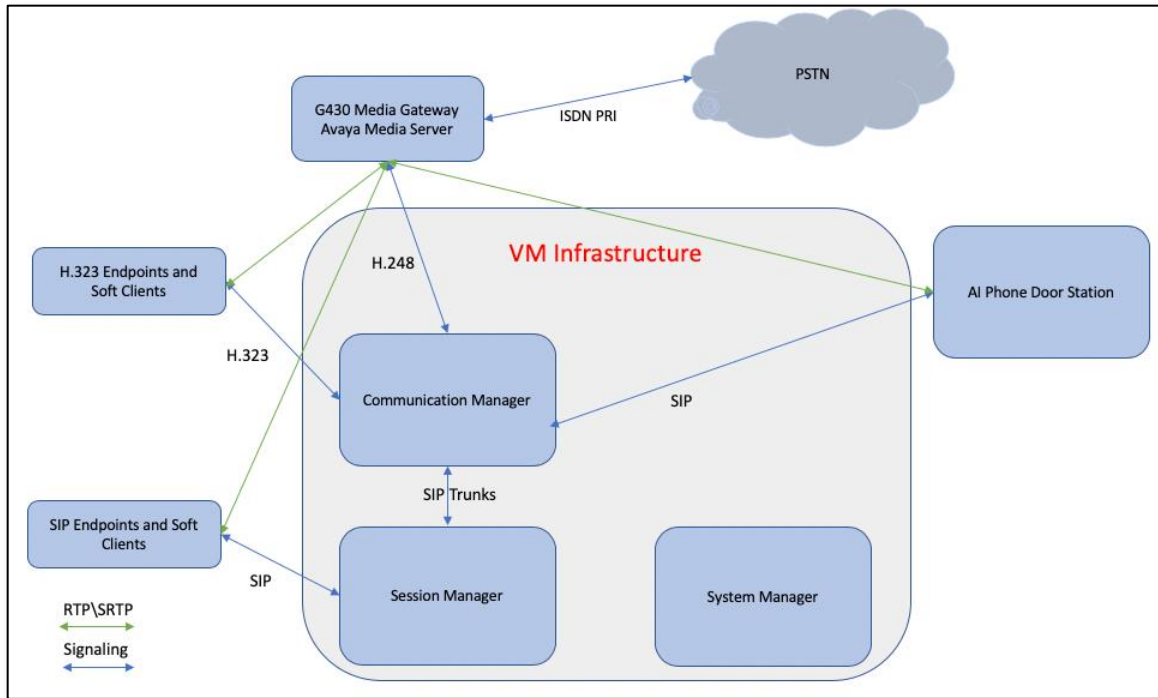


Figure 1: Test Configuration of Aiphone IX-EA with Avaya Aura®

4. Equipment and Software Validated

The following equipment and software were used for the test configuration.

Equipment	Software/Firmware
Avaya Aura® Communication Manager	8.1.1.0.0.890.25763 (FP1)
Avaya Aura® Session Manager	8.1.1.0.811021
Avaya Aura® System Manager	8.1.1.0.0310782 (FP1)
Avaya 9600 Series H.323 IP Deskphones	6.8304
Avaya J129 SIP Phone	4.0.4.0.10
Avaya IX Workspace	3.7.0.102.3
Avaya H175 Collaboration Station	1.0.2.3
Avaya Vantage K175 Phone	3.5.0
Avaya 9504 Digital Phone	0.55
Avaya 6210 Analogue Telephone	-
Aiphone IX Series Video Door Station IX-EA	5.40

5. Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Communication Manager. The procedures include the following areas:

- Verify System Capacity (License)
- Define Dial Plan
- Enable IP Video

These steps were performed using an SSH Terminal session.

5.1. Verify System Capacity (License)

The license file installed on the system controls these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative. Use the **display system-parameters customer-options** command to determine these values. On **Page 1**, verify that the **Maximum Off-PBX Telephones** allowed in the system is sufficient. One OPS station is required per SIP device.

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

G3 Version: V18                                     Software Package: Enterprise
Location: 2                                           System ID (SID): 1
Platform: 28                                         Module ID (MID): 1

                                USED
Platform Maximum Ports: 48000      73
Maximum Stations: 36000      48
Maximum XMOBILE Stations: 36000      0
Maximum Off-PBX Telephones - EC500: 41000      0
Maximum Off-PBX Telephones - OPS: 41000      27
Maximum Off-PBX Telephones - PBFMC: 41000      0
Maximum Off-PBX Telephones - PVFMC: 41000      0
Maximum Off-PBX Telephones - SCCAN: 0      0
Maximum Survivable Processors: 313      0

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

On **Page 2** of the **system-parameters customer-options form**, verify that the number of **Maximum Administered SIP Trunks** supported by the system is sufficient.

display system-parameters customer-options		Page	2 of 12
OPTIONAL FEATURES			
IP PORT CAPACITIES		USED	
Maximum Administered H.323 Trunks:		12000	0
Maximum Concurrently Registered IP Stations:		2400	3
Maximum Administered Remote Office Trunks:		12000	0
Max Concurrently Registered Remote Office Stations:		2400	0
Maximum Concurrently Registered IP eCons:		128	0
Max Concur Reg Unauthenticated H.323 Stations:		100	0
Maximum Video Capable Stations:		36000	0
Maximum Video Capable IP Softphones:		2400	16
Maximum Administered SIP Trunks:		12000	10
Max Administered Ad-hoc Video Conferencing Ports:		12000	0
Max Number of DS1 Boards with Echo Cancellation:		688	0

5.2. Define the Dial Plan

Use the **change dialplan analysis** command to define the dial plan used in the system. This includes all telephone extensions. In the sample configuration, telephone extensions are 5 digits long and begin with 7.

change dialplan analysis						Page 1 of 12					
DIAL PLAN ANALYSIS TABLE											
Location: all						Percent Full: 1					
Dialed String			Total Call Length Type			Dialed String			Total Call Length Type		
1			3 <u>dac</u>								
2			5 <u>ext</u>								
3			5 <u>ext</u>								
4			5 <u>aar</u>								
7			5 <u>ext</u>								
8			1 <u>fac</u>								
9			1 <u>fac</u>								
*			3 <u>fac</u>								
#			3 <u>fac</u>								

5.3. Enable IP Video

Use the **change signaling-group** command to enable IP video in the system.

```
change 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: sm81
Near-end Listen Port: 5061                      Far-end Listen Port: 5061
                                                Far-end Network Region: 1

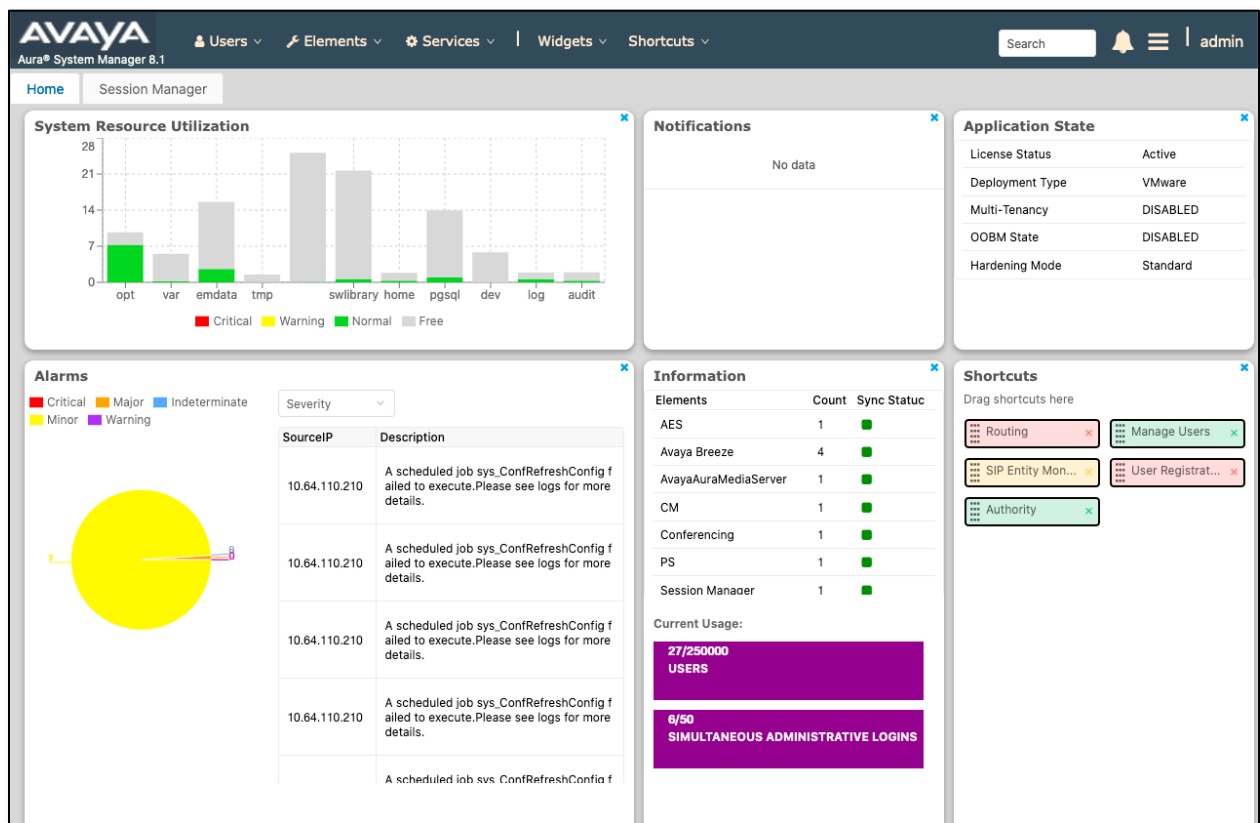
Far-end Domain: avaya.com

Incoming Dialog Loopbacks: eliminate                      Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload                      RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 65                      Direct IP-IP Audio Connections? n
Enable Layer 3 Test? y                      IP Audio Hairpinning? y
                                                Alternate Route Timer(sec): 6
```


6. Configure Avaya Aura® Session Manager

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

Session Manager is managed via System Manager. Using a web browser, access **https://<ip-addr of System Manager>/SMGR**. In the **Log On** screen, enter appropriate **User ID** and **Password** and click the **Log On** button.



6.1. Verify Session Manager Listen Port for SIP Endpoint Registration

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

Listen Ports

AddRemove

4 ItemsFilter: Enable

<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Endpoint	Notes
<input type="checkbox"/>	5060	TCP	avaya.com	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	5060	UDP	avaya.com	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	5061	TLS	avaya.com	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	5062	TLS	avaya.com	<input type="checkbox"/>	

Select : All, None

6.2. Add a SIP User

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

- **First Name** and **Last Name** - Enter an identifying name
- **Login Name** Enter the extension number followed by the domain, in this case **72003@avaya.com**

Home / Users / Manage Users Help ?

User Profile | Edit | 72003@avaya.com Commit & Continue Commit Cancel

Identity | Communication Profile | Membership | Contacts

Basic Info

Address

LocalizedName

User Provisioning Rule:

* Last Name: Last Name (in Latin alphabet characters):

* First Name: First Name (in Latin alphabet characters):

* Login Name: Middle Name:

Description: Email Address:

Password: User Type:

Confirm Password: Localized Display Name:

Endpoint Display Name: Title Of User:

Language Preference: Time Zone:

Employee ID: Department:

Company:

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

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

The screenshot shows a web interface for editing a user profile. A modal dialog titled "Comm-Profile Password" is open. It contains two input fields: "Comm-Profile Password:" with a masked value "*****" and "Re-enter Comm-Profile Password:" with the placeholder text "Re-enter Comm-Profile Password". Below these fields is a blue link "Generate Comm-Profile Password". At the bottom of the dialog are "Cancel" and "OK" buttons. The background page shows the "Communication" tab selected, with sections for "Communication Profile Password", "Communication Address", and "PROFILES".

In the **Communication Address** section, for **Type** select **Avaya SIP** from the drop-down list. In the **Fully Qualified Address** field enter the extension number as required and select the appropriate **Domain** from the drop-down list. Click **OK** when done.

The screenshot shows the same web interface, but the modal dialog is now titled "Communication Address Add/Edit". It contains a "Type:" dropdown menu with "Avaya SIP" selected. Below it is a "Fully Qualified Address:" field with "72003" entered, followed by an "@" symbol and a domain dropdown menu with "avaya.com" selected. At the bottom of the dialog are "Cancel" and "OK" buttons. The background page shows the "Communication Address" section selected.

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

User Profile | Edit | 72003@avaya.com

Commit & ContinueCommitCancel

IdentityCommunication ProfileMembershipContacts

Communication Profile Password

PROFILE SET : Primary

Communication Address

PROFILES

Session Manager Profile

Avaya Breeze® Profile

CM Endpoint Profile

Presence Profile

SIP Registration

Primary Session Manager: sm81

Secondary Session Manager: Start typing...

Survivability Server: Start typing...

Max. Simultaneous Devices: 2

Block New Registration When Maximum Registrations Active?

Application Sequences

Origination Sequence: cm81

Termination Sequence: cm81

Emergency Calling Application Sequences

Emergency Calling Origination Sequence: Select

Emergency Calling Termination Sequence: Select

Call Routing Settings

Home Location: DevConnect

Click the **CM Endpoint Profile** link and configure as follows:

- **System** - Select the relevant Communication Manager SIP Entity from the drop-down list
- **Profile Type** - Select **Endpoint** from the drop-down list
- **Extension** - Enter the required extension number, in this case **72003**
- **Template** - Select **J129_DEFAULT_CM_8_1** from the drop-down list
- **Port** - The “IP” is auto filled out by the system

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

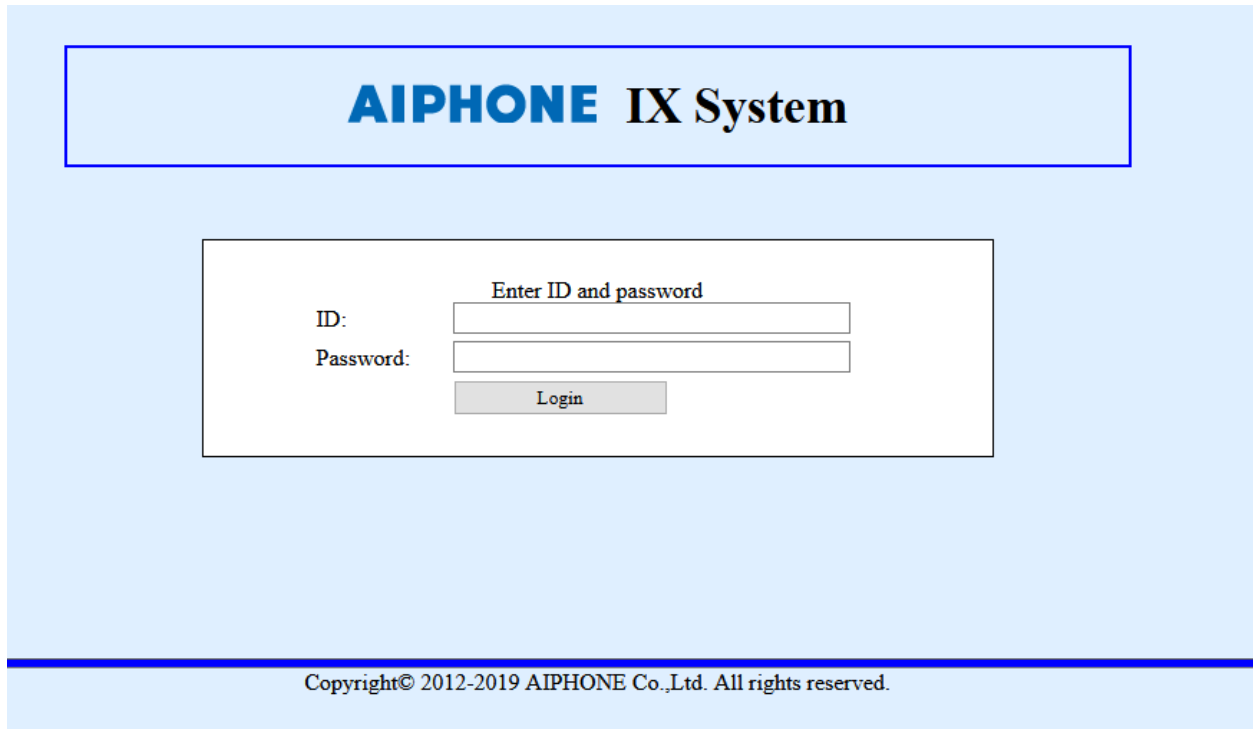
The screenshot displays the 'User Profile | Edit | 72003@avaya.com' interface. The 'Communication Profile' tab is active, showing various configuration fields. On the left, a sidebar lists profile types: 'Communication Profile Password', 'PROFILE SET : Primary', 'Communication Address', 'PROFILES', 'Session Manager Profile' (checked), 'Avaya Breeze® Profile' (unchecked), 'CM Endpoint Profile' (checked), and 'Presence Profile' (unchecked). The main area contains fields for 'System' (cm81), 'Profile Type' (Endpoint), 'Extension' (72003), 'Set Type' (J129), 'Port' (S000086), 'Template' (Start typing...), 'Security Code' (Enter Security Code), 'Voice Mail Number', 'Preferred Handle' (Select), 'Sip Trunk' (aar), 'SIP URI' (Select), 'Enhanced Callr-Info Display for 1-line phones - Override Endpoint Name and Localized Name' (checked), 'Delete on Unassign from User or on Delete User' (checked), and 'Allow H.323 and SIP Endpoint Dual Registration' (unchecked). Buttons for 'Commit & Continue', 'Commit', and 'Cancel' are at the top right.

Field	Value
System	cm81
Profile Type	Endpoint
Extension	72003
Set Type	J129
Port	S000086
Template	Start typing...
Security Code	Enter Security Code
Voice Mail Number	
Preferred Handle	Select
Sip Trunk	aar
SIP URI	Select
Enhanced Callr-Info Display for 1-line phones - Override Endpoint Name and Localized Name	Checked
Delete on Unassign from User or on Delete User	Checked
Allow H.323 and SIP Endpoint Dual Registration	Unchecked

7. Configure Aiphone IX Series Video Door Station

This section provides steps to configure Aiphone IX-EA.

To configure Aiphone IX-EA, using a web browser, navigate to <https://<IP Address of IX-EA>/webset.cgi?login> and log in using appropriate credentials.



AIPHONE IX System

Enter ID and password

ID:

Password:

Login

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AIPHONE

IX System Setting

Category: Video Stations

Station Type: IX-EA, IX-EAU

Update

Station Information

Identification

ID and Password

Language

Time

Expanded System

Network Settings

IP Address

DNS

SIP

Multicast Address

Video

Station Information

◆ Required Settings

◆ Identification

Number◆720033-5 digits

NameIX-EA1-24 alphanumeric characters(*1)

Location1-24 alphanumeric characters(*1)

(*1)Certain characters may not be displayed correctly on IX-MV and IX-MV7.*

- **SIP Signaling Port:** Set to **5060**.
- **User Agent:** Type in a desired value.
- **ID:** SIP Extension number from **Section 6.2**.
- **Password:** SIP Extension password from **Section 6.2**.
- **IPv4 Address:** LAN IP Address of Session Manager
- **Port:** Set to **5060**.

AIPHONE IX System Setting		Update																											
Category: Video Stations	Station Type: IX-EA, IX-EAU																												
Station Information Identification ID and Password Language Time Expanded System	Network Settings <ul style="list-style-type: none"> SIP <table border="1"> <thead> <tr> <th colspan="3">SIP Connections</th> </tr> </thead> <tbody> <tr> <td>SIP Signaling Port ♦</td> <td><input type="text" value="5060"/></td> <td>1-65535</td> </tr> <tr> <td>User Agent</td> <td><input type="text" value="IX-EA"/></td> <td>1-36 alphanumeric characters</td> </tr> </tbody> </table> <table border="1"> <thead> <tr> <th colspan="3">SIP Server</th> </tr> </thead> <tbody> <tr> <td>Primary Server ID</td> <td><input type="text" value="72003"/></td> <td>1-24 alphanumeric characters</td> </tr> <tr> <td>Password</td> <td><input type="password" value="*****"/></td> <td>1-24 alphanumeric characters</td> </tr> <tr> <td>IPv4 Address</td> <td><input type="text" value="10.64.110.212"/></td> <td>1.0.0.1-223.255.255.254 or host</td> </tr> <tr> <td>IPv6 Address</td> <td><input type="text"/></td> <td>::FF:0-FE:FF:FFFF:FFFF:FFFF::</td> </tr> <tr> <td>Port ♦</td> <td><input type="text" value="5060"/></td> <td>1-65535</td> </tr> </tbody> </table> 		SIP Connections			SIP Signaling Port ♦	<input type="text" value="5060"/>	1-65535	User Agent	<input type="text" value="IX-EA"/>	1-36 alphanumeric characters	SIP Server			Primary Server ID	<input type="text" value="72003"/>	1-24 alphanumeric characters	Password	<input type="password" value="*****"/>	1-24 alphanumeric characters	IPv4 Address	<input type="text" value="10.64.110.212"/>	1.0.0.1-223.255.255.254 or host	IPv6 Address	<input type="text"/>	::FF:0-FE:FF:FFFF:FFFF:FFFF::	Port ♦	<input type="text" value="5060"/>	1-65535
SIP Connections																													
SIP Signaling Port ♦	<input type="text" value="5060"/>	1-65535																											
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Password	<input type="password" value="*****"/>	1-24 alphanumeric characters																											
IPv4 Address	<input type="text" value="10.64.110.212"/>	1.0.0.1-223.255.255.254 or host																											
IPv6 Address	<input type="text"/>	::FF:0-FE:FF:FFFF:FFFF:FFFF::																											
Port ♦	<input type="text" value="5060"/>	1-65535																											
Network Settings IP Address DNS SIP Multicast Address Video Audio Packet Priority NTP																													
System Information																													

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

- **Station Number:** Type in an extension number that will be called for a given line.
- **IPv4:** Type in the LAN IP Address for Session Manager.

Select **Update** to save changes.

The screenshot shows the 'Call Settings' section of the AIPHONE IX System Setting web interface. The left sidebar contains navigation links for Station Information, Network Settings, System Information, Call Settings, Option Input / Relay, Output Settings, and Function Settings. The main content area is titled 'Call Settings' and includes a 'Called Stations (for Door)' section. Below this, there is a table with columns for #, Station Number, IPv4 Address, IPv6 Address, Station Type, and Protocol. The table contains two rows of data. A red warning message is displayed above the table, stating that the Station Number must be 3-5 digits, IPv4 must be a valid IP address, and IPv6 must be a valid IPv6 address or hostname. The 'Update' button is visible in the top right corner.

#	Station Number	IPv4 Address	IPv6 Address	Station Type	Protocol
1	70101	10.64.110.212		VoIP Phone	U
2	72000	10.64.110.212		IX-MV7-*	U

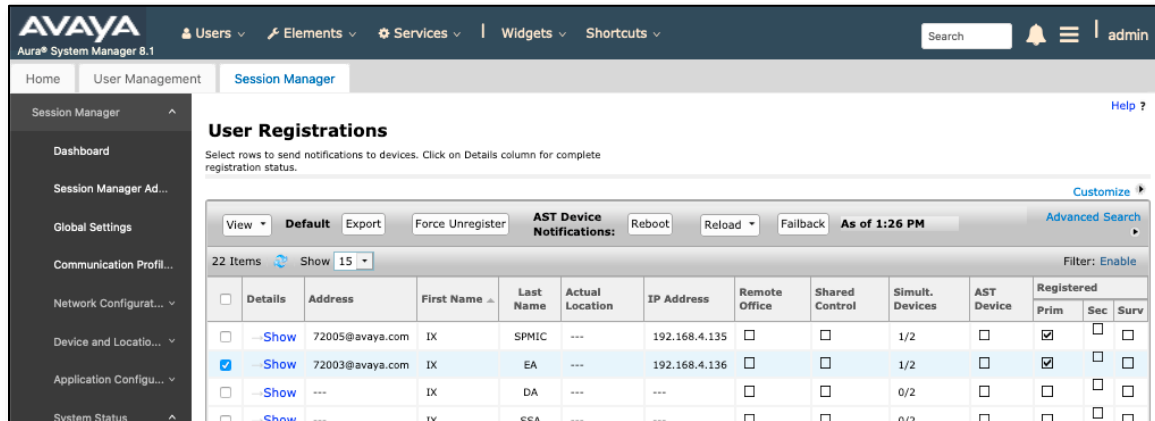
Continuing from above, scroll down to the **Video** sub section and verify the Video Encoder settings are as shown below.

The screenshot shows the 'Network Settings' section of the AIPHONE IX System Setting web interface. The left sidebar contains navigation links for Station Information, Network Settings, System Information, Call Settings, Option Input / Relay, Output Settings, and Function Settings. The main content area is titled 'Network Settings' and includes a 'Video' section. Below this, there are two sub-sections: 'Video Encoder 1' and 'Video Encoder 2'. The 'Video Encoder 1' section includes settings for Resolution, Frame Rate, Select Profile, I-picture interval, Bit rate, RTP Start Port, and RTP End Port. The 'Video Encoder 2' section includes settings for Second Video Encoder, Video Codec, Resolution, Frame Rate, Select Profile, I-picture interval, Bit rate, Select Quality, RTP Start Port, and RTP End Port. The 'Update' button is visible in the top right corner.

8. Verification Steps

The following steps may be used to verify the configuration:

- In the System Manager web interface, navigate to Elements → Session Manager → System Status → User Registrations to confirm successful registration.



Place a call from Aiphone IX-EA to an Avaya endpoint. The state of the call be viewed on Communication Manager using the **status trunk** command in a SAT Terminal session:

```
status trunk 1
```

TRUNK GROUP STATUS				
Member	Port	Service State	Mtce	Connected Ports Busy
0001/0001	T000001	in-service/active	no	T000007
0001/0002	T000002	in-service/idle	no	
0001/0003	T000003	in-service/idle	no	
0001/0004	T000004	in-service/idle	no	
0001/0005	T000005	in-service/idle	no	
0001/0006	T000006	in-service/idle	no	
0001/0007	T000007	in-service/active	no	T000001

To view the status of the endpoints connected to the SIP Trunk, and codecs in use, use **status trunk 1/0001** where /0001 is a trunk port connected to the call.

```
status trunk 1/0001
```

SRC PORT TO DEST PORT TALKPATH	
src port:	T000001
T000007:TX:	192.168.4.130:40750/g711u/20ms
001V062:RX:	10.64.50.54:2054/g711u/20ms:TX:ctxID:542
001V061:RX:ctxID:	542:TX:10.64.50.54:2056/g711u/20ms
T000001:RX:	192.168.4.136:20000/g711u/20ms

Page 4 of 4

To verify video codecs used, scroll to page 2 and note the Video Near-end Codec and Video Far-end Codec and highlighted below.

```

status trunk 1/0007
CALL CONTROL SIGNALING
Near-end Signaling Loc: PROCR
  Signaling      IP Address      Port
  Near-end:      10.64.110.213    : 5061
  Far-end:       10.64.110.212    : 5061
H.245 Near:
H.245 Far:
  H.245 Signaling Loc:          H.245 Tunneled in Q.931? no

Audio Connection Type: ip-tdm      Authentication Type: None
  Near-end Audio Loc: MG1          Codec Type: G.711MU
  Audio      IP Address      Port
  Near-end:   10.64.50.54     : 2054
  Far-end:    192.168.4.136   : 40750

Video Near: 192.168.4.130      : 30000
Video Far:  192.168.4.136     : 45752
Video Port: T000007

Video Near-end Codec:      H.264      Video Far-end Codec: H.264

```

9. Conclusion

Aiphone IX-EA was compliance tested with Avaya Aura®. Aiphone IX-EA functioned properly for feature and serviceability.

10. Additional References

Avaya product documentation can be found at: <http://support.avaya.com>

Documentation related to Aiphone IX-EA can be found at:

Japan: <https://www.aiphone.co.jp/products/business/ix/>

USA, Canada: <https://www.aiphone.com/home/products/ix-series>

France: <https://www.aiphone.fr/catalogue/interphonie-ip-protocole-sip-ix/>

Australia, New Zealand: <https://www.aiphone.com.au/product/ix/>

Singapore: <http://www.aiphone.com.sg/>

United Kingdom: https://www.aiphone.co.uk/featured_item/ix2/

Appendix A

Following devices are based on the same firmware as IX-EA:

- IX-EA
- IX-EAU

The difference in each IX-EA devices is their mounting method:

- IX-EA
 - Surface mounting
- IX-EAU
 - Flush mounting

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