



Avaya Solution & Interoperability Test Lab

Application Notes for Aiphone IX Series Video Door Stations (IX-DV) 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-DV) 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-DV) 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-DV was used.

The Aiphone IX Series Video Door Stations (IX-DV) are part of Aiphone IX Series Door Stations. The Video Door Stations, IX-DV, 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-DV 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-DV 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-DV, 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-DV, 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-DV.

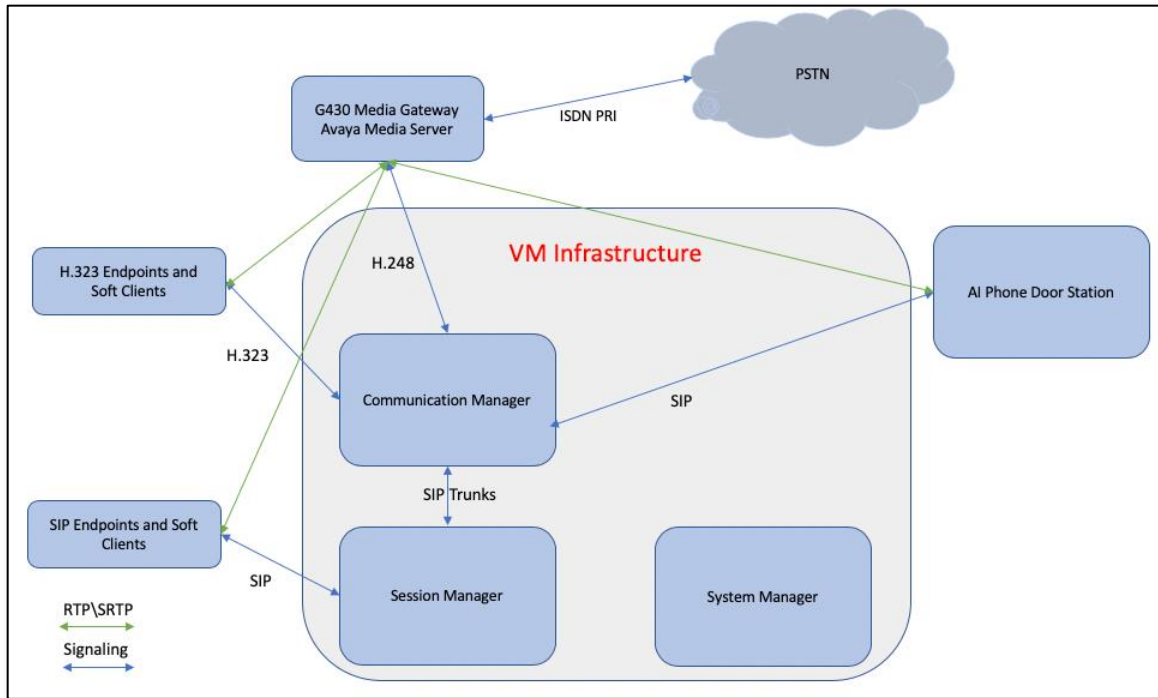


Figure 1: Test Configuration of Aiphone IX-DV with Avaya Aura®

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-DV	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		
<u>1</u>			<u>3</u> <u>dac</u>								
<u>2</u>			<u>5</u> <u>ext</u>								
<u>3</u>			<u>5</u> <u>ext</u>								
<u>4</u>			<u>5</u> <u>aar</u>								
<u>7</u>			<u>5</u> <u>ext</u>								
<u>8</u>			<u>1</u> <u>fac</u>								
<u>9</u>			<u>1</u> <u>fac</u>								
<u>*</u>			<u>3</u> <u>fac</u>								
<u>#</u>			<u>3</u> <u>fac</u>								

5.3. Enable IP Video

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

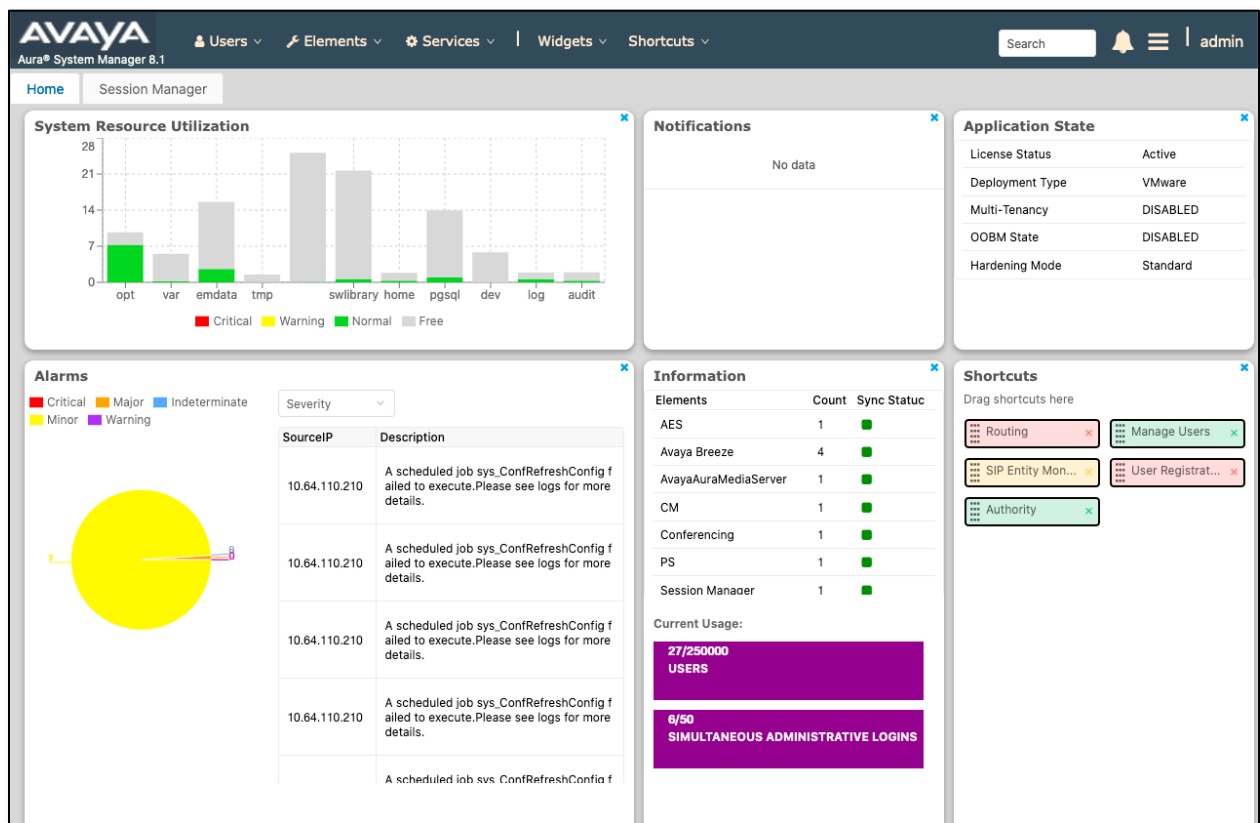
```
change signaling-group 1                                     Page 1 of 3
                                SIGNALING GROUP
Group Number: 1                      Group Type: sip
IMS Enabled? n                      Transport Method: tls
Q-SIP? n
IP Video? y                      Priority Video? n          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
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? n
Session Establishment Timer(min): 65                      IP Audio Hairpinning? y
Enable Layer 3 Test? 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-DV. 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 Items 

Filter: [Enable](#)

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

Select : [All](#), [None](#)

6.2. Add a SIP User

A SIP user must be added for Aiphone IX-DV. 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 **72002@avaya.com**

The screenshot displays the Avaya Aura System Manager 8.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The left sidebar shows 'User Management' with 'Manage Users' selected. The main content area is titled 'User Profile | Edit | 72002@avaya.com' and features tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Identity' tab is active, showing a 'Basic Info' section with fields for 'Last Name' (DV), 'First Name' (IX), 'Login Name' (72002@avaya.com), 'Description', 'Password', 'Confirm Password', 'Endpoint Display Name' (DV, IX), 'Language Preference' (English (United States)), 'Employee ID', and 'Company'. There are also fields for 'Last Name (in Latin alphabet characters)', 'First Name (in Latin alphabet characters)', 'Middle Name', 'Email Address', 'User Type' (Basic), 'Localized Display Name' (DV, IX), 'Title Of User', 'Time Zone', and 'Department'. Buttons for 'Commit & Continue', 'Commit', and 'Cancel' are visible at the top right of the form.

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 the 'User Profile | Edit | 72002@avaya.com' interface. The 'Communication Profile' tab is selected. A modal dialog titled 'Comm-Profile Password' is open, featuring two password input fields: 'Comm-Profile Password' and 'Re-enter Comm-Profile Password'. A 'Generate Comm-Profile Password' link is positioned below the second field. The dialog includes 'Cancel' and 'OK' buttons at the bottom right. In the background, the 'Communication Address' section is visible, showing a domain dropdown set to 'avaya.com'.

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 'User Profile | Edit | 72002@avaya.com' interface. The 'Communication Address' section is active. A modal dialog titled 'Communication Address Add/Edit' is open. It contains a 'Type' dropdown menu with 'Avaya SIP' selected, and a 'Fully Qualified Address' field with '72002' entered. A domain dropdown menu to the right of the address field shows 'avaya.com' selected. The dialog has 'Cancel' and 'OK' buttons at the bottom right. The background shows the 'Communication Profile' tab and various profile settings.

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 | 72002@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

RAB; Reviewed:
SPOC 9/30/2020

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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 **72002**
- **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 | 72002@avaya.com' interface. The 'Communication Profile' tab is active. On the left, a sidebar shows 'PROFILES' with 'CM Endpoint Profile' selected. The main area contains various configuration fields:

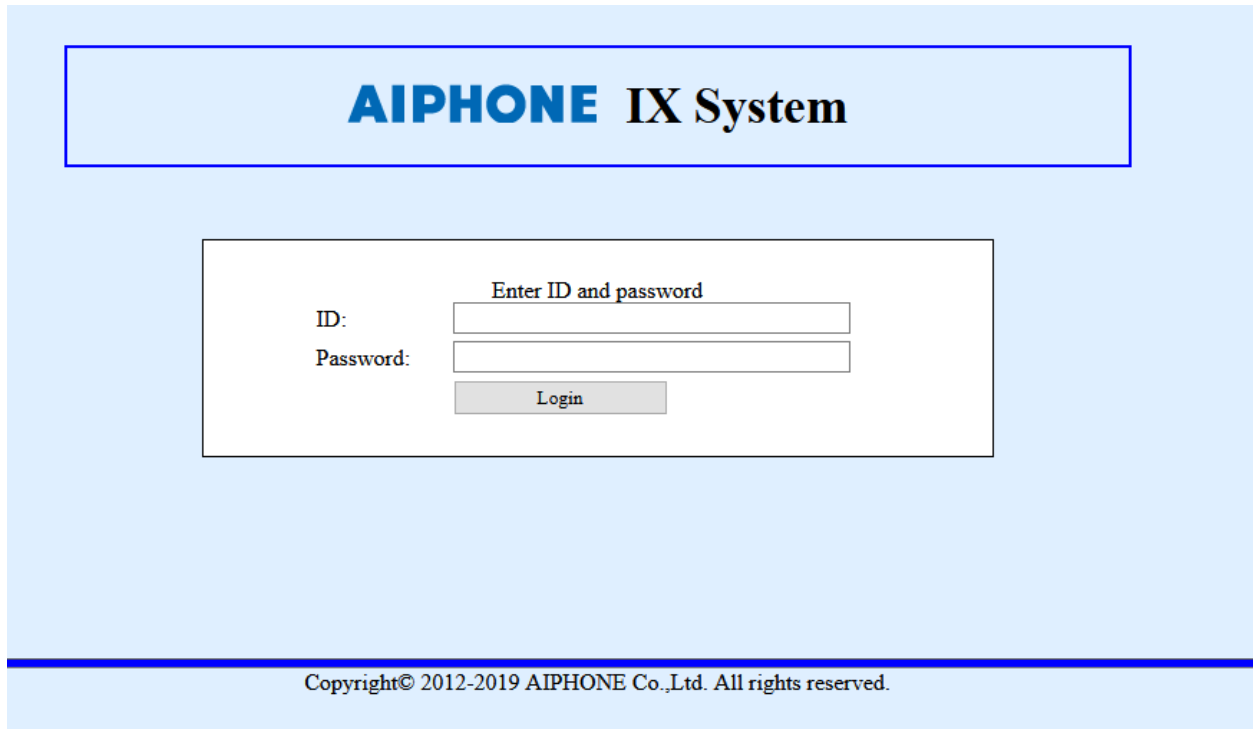
- System:** cm81
- Profile Type:** Endpoint
- Extension:** 72002 (with an 'Endpoint Editor' icon)
- Set Type:** J129
- Port:** S000085
- Preferred Handle:** 72002@avaya.com
- Sip Trunk:** aar
- Template:** Start typing...
- Security Code:** Enter Security Code
- Voice Mail Number:**
- Calculate Route Pattern:**
- SIP URI:** Select
- Enhanced Callr-Info Display for 1-line phones:** (unchecked)
- 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)

Buttons at the top right include 'Commit & Continue', 'Commit', and 'Cancel'.

7. Configure Aiphone IX Series Video Door Station

This section provides steps to configure Aiphone IX-DV.

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



AIPHONE IX System

Enter ID and password

ID:

Password:

Login

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Once logged in, for the **Number** field, type in the SIP extension that is being configured (from **Section 6.2**), and a desired **Name**. Select **Update** to save changes.

AIPHONE IX System Setting

Category: Video Stations Station Type: IX-DV, IX-DVF(-*)

Station Information

Identification

Number 72002 3-5 digits

Name IX-DV 1-24 alphanumeric characters(*)

Location Dev Connect 1-24 alphanumeric characters(*)

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

Update

From the left, select **Network Settings** → **SIP** and configure as follows:

- **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**.

Once done, select **Update** to save changes.

AIPHONE IX System Setting

Category: Video Stations Station Type: IX-DV, IX-DVF(-*)

Network Settings

SIP

SIP Connections

SIP Signaling Port 5060 1-65535

User Agent IX-DV 1-36 alphanumeric characters

SIP Server

Primary Server

ID 72002 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(1-64 alphanumeric characters)

IPv6 Address ::FF:0:FEFF:FFFF:FFFF:FFFF:FFFF:FFFF or hostname(1-64 alphanumeric characters)

Port 5060 1-65535

Update

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-DV 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. It features a table with columns for Station Number, IPv4 Address, IPv6 Address, Station Type, and Protocol. The table has two rows, both with Station Number 70101. The first row has IPv4 Address 10.64.110.212 and Station Type VoIP Phone. The second row has empty fields. A red warning message is displayed above the table, stating: '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. If designating "M", multicast IP addresses must be configured for the station(s).' An 'Update' button is located in the top right corner.

#	Station Number	IPv4 Address	IPv6 Address	Station Type	Protocol
1	70101	10.64.110.212		VoIP Phone	U
2					

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. It features two sub-sections: 'Video Encoder 1' and 'Video Encoder 2'. The 'Video Encoder 1' section includes settings for Resolution (320x240(QVGA) or 640x480(VGA)), Frame Rate [fps] (15), Select Profile (High), I-picture interval (15), Bit rate [kbps] (1024), RTP Start Port (30000), and RTP End Port (31000). The 'Video Encoder 2' section includes settings for Second Video Encoder (Enable or Disable), Video Codec (H.264/AVC or Motion-JPEG), Resolution (1280x720(HD)), Frame Rate [fps] (10), Select Profile [H.264 / AVC] (High), I-picture interval [H.264 / AVC] (10), Bit rate [kbps] [H.264 / AVC] (2048), Select Quality [Motion-JPEG] (6), RTP Start Port (32000), and RTP End Port (33000). Red warning messages are displayed above each encoder section, stating: 'The "Video Encoder 1" RTP End Port should be greater than 90 digits from the RTP Start Port.' and 'The "Video Encoder 2" RTP End Port should be greater than 10 digits from the RTP Start Port.' An 'Update' button is located 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.

The screenshot shows the Avaya Aura System Manager 8.1 web interface. The top navigation bar includes links for Users, Elements, Services, Widgets, and Shortcuts. The left sidebar shows the Session Manager menu. The main content area is titled 'User Registrations' and displays a table of user registrations. The table has columns for Details, Address, First Name, Last Name, Actual Location, IP Address, Remote Office, Shared Control, Simult. Devices, AST Device, and Registered status. The table shows three entries, with the second entry (72002@avaya.com) selected.

Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered
<input type="checkbox"/> -- Show	72005@avaya.com	IX	SPMIC	---	192.168.4.135	<input type="checkbox"/>	<input type="checkbox"/>	1/2	<input type="checkbox"/>	<input checked="" type="checkbox"/>
<input checked="" type="checkbox"/> -- Show	72002@avaya.com	IX	DV	---	192.168.4.137	<input type="checkbox"/>	<input type="checkbox"/>	1/2	<input type="checkbox"/>	<input checked="" type="checkbox"/>
<input type="checkbox"/> -- Show	72003@avaya.com	IX	EA	---	192.168.4.136	<input type="checkbox"/>	<input type="checkbox"/>	1/2	<input type="checkbox"/>	<input checked="" type="checkbox"/>

Place a call from Aiphone IX-DV 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                                     Page 4 of 4

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:192.168.4.137:20000/g711u/20ms
```


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
                                Page 2 of 4
                                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:    192.168.4.130       : 40750

Video Near:   192.168.4.137       : 30000
Video Far:    192.168.4.130       : 45752
Video Port:   T000007

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

```

9. Conclusion

Aiphone IX-DV was compliance tested with Avaya Aura®. Aiphone IX-DV 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-DV 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-DV:

- IX-DV
- IX-DVF
- IX-DVF-P
- IX-DVF-RA
- IX-DVF-2RA
- IX-DVF-RA-FR
- IX-DVF-2RA-FR
- IX-DVF-4
- IXDVFL
- IXDVFLAC
- IXDVF2L
- IXDVF2LAC
- IXDVF4L
- IXDVF4LAC
- IXDVF6L
- IXDVF6LAC
- IXDVFA
- IX-4DVF
- IX-2DVF
- IX-DVF Slim
- IX-DVF-AC

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

- IX-DV
 - Surface mounting
- IX-DVF
 - Flush mounting
- IX-DVF-P
 - Flush mounting
 - Card reader
- IX-DVF-RA
 - Flush mounting
 - Emergency call button
- IX-DVF-2RA
 - Flush mounting
 - Normal call button and emergency call button
- IX-DVF-RA-FR
 - Flush mounting
 - Emergency call button
 - French notation
- IX-DVF-2RA-FR

- Flush mounting
 - Normal call button and emergency call button
 - French notation
- IX-DVF-4
 - Flush mounting
 - 4 call buttons
- IXDVFL
 - Flush mounting
 - Hearing aid
- IXDVFLAC
 - Flush mounting
 - Hearing aid
 - 10-key pad
- IXDVF2L
 - Flush mounting
 - Hearing aid
 - 2 call buttons
- IXDVF2LAC
 - Flush mounting
 - Hearing aid
 - 2 call buttons
 - 0-key pad
- IXDVF4L
 - Flush mounting
 - 4 call buttons
- IXDVF4LAC
 - Flush mounting
 - 4 call buttons
 - 10-key pad
- IXDVF6L
 - Flush mounting
 - 6 call buttons
- IXDVF6LAC
 - Flush mounting
 - 6 call buttons
 - 10-key pad
- IXDVFA
 - Flush mounting
- IX-4DVF
 - Flush mounting
 - 4 piezo buttons
- IX-2DVF
 - Flush mounting
 - 2 piezo buttons
- IX-DVF Slim

- Flush mounting
 - Miniaturized panel
- IX-DVF-AC
 - Flush mounting
 - 10-key pad

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