



## Avaya Solution & Interoperability Test Lab

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# Application Notes for Aiphone IX Series Video Master Stations (IX-MV7) R5.4 and Avaya Aura<sup>®</sup> Communication Manager and Avaya Aura<sup>®</sup> Session Manager R8.1 – Issue 1.0

## Abstract

These Application Notes describe the procedures for configuring Aiphone IX Series Video Master Stations (IX-MV7) which was compliance tested with Avaya Aura<sup>®</sup> Communication Manager and Avaya Aura<sup>®</sup> Session Manager.

The overall objective of the interoperability compliance testing was to verify Aiphone IX Series Video Master Stations (IX-MV7) functionalities in an environment comprised of Avaya Aura<sup>®</sup> and various Avaya endpoints. Aiphone IX Series Video Master 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 Master Stations to interoperate with Avaya Aura<sup>®</sup> Communication Manager and Avaya Aura<sup>®</sup> Session Manager. During the compliance testing, Aiphone IX-MV7 was used.

The Aiphone IX Series Video Master Stations (IX-MV7) are part of Aiphone IX Series Master Stations. The Video Master Stations, IX-MV7, act as SIP phones when connected to Avaya Aura<sup>®</sup>. They have a built-in camera allowing for H.264 based two-way video, and a 7-inch screen. Additionally, the Master Stations have intercom features that include paging, line supervision, device check, and picture in picture when using 3rd party ONVIF Profile S cameras (not tested).

During the compliance test, Aiphone IX-MV7 registered as a 3<sup>rd</sup> party SIP phone using UDP to Avaya Aura<sup>®</sup> Session Manager.

## 2. General Test Approach and Test Results

The focus of this interoperability compliance testing was to verify that the Aiphone IX-MV7 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<sup>®</sup> 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-MV7, 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-MV7, 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](mailto: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](mailto: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-MV7.

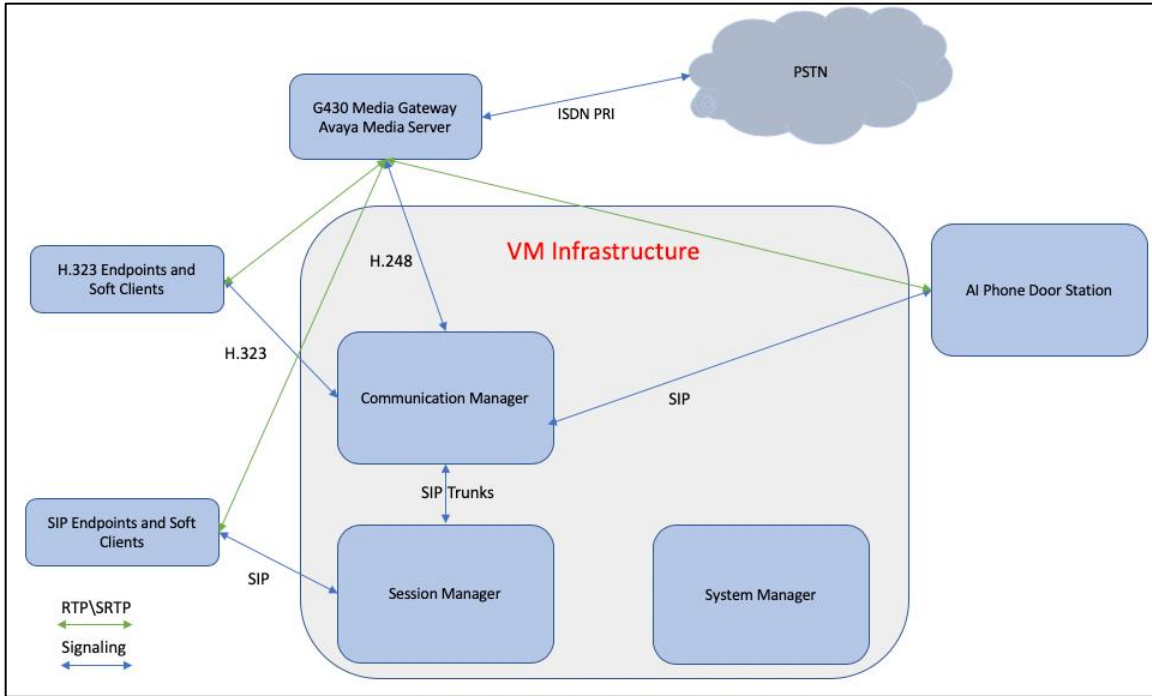


Figure 1: Test Configuration of Aiphone IX-MV7 with Avaya Aura®

## 4. Equipment and Software Validated

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

<b>Equipment</b>	<b>Software/Firmware</b>
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 Master Station IX-MV7	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   Total   Call   Dialed   Total   Call   Dialed   Total   Call
    String   Length  Type   String   Length  Type   String   Length  Type
1         3   dac
2         5   ext
3         5   ext
4         5   aar
7       5 ext
8         1   fac
9         1   fac
*         3   fac
#         3   fac
  
```

### 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

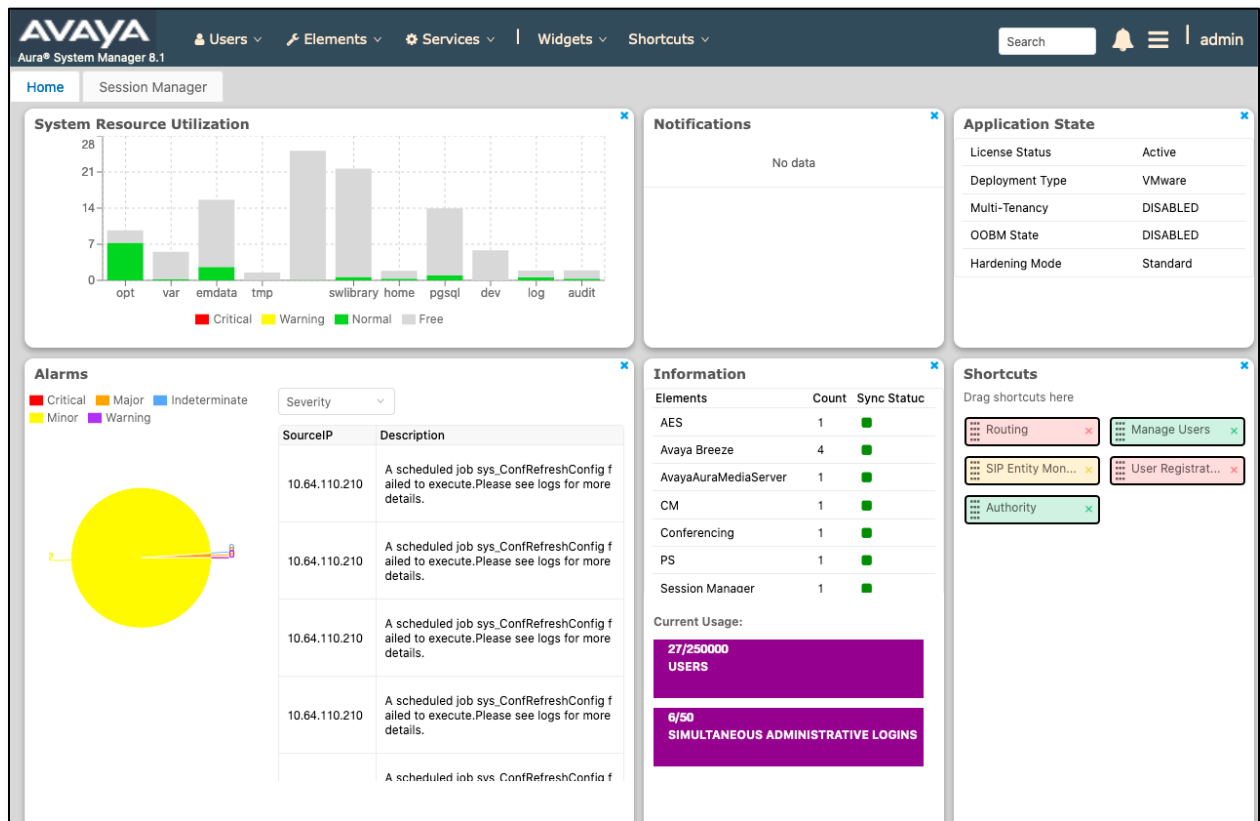
Incoming Dialog Loopbacks: eliminate           Bypass If IP Threshold Exceeded? n
  DTMF over IP: rtp-payload           RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 65           Direct IP-IP Audio Connections? n
  Enable Layer 3 Test? y           IP Audio Hairpinning? y
                                     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-MV7. 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.

<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-MV7. Click **User Management** → **Manage Users** → **New** (not shown) and configure the following in the **Identity** tab.

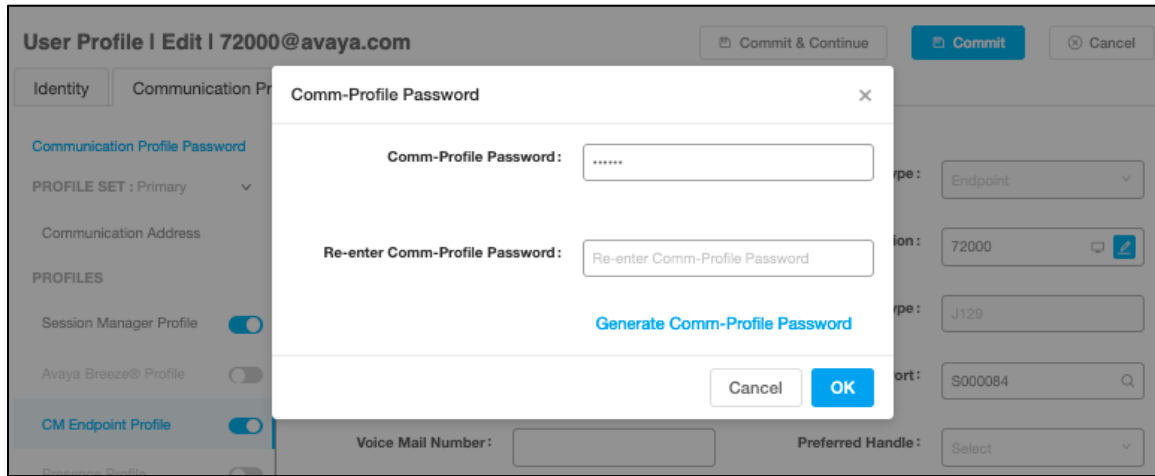
- **First Name** and **Last Name** - Enter an identifying name
- **Login Name** Enter the extension number followed by the domain, in this case **72000@avaya.com**

The screenshot shows the 'User Profile | Edit | 72000@avaya.com' interface. It features a navigation bar with 'Commit & Continue', 'Commit', and 'Cancel' buttons. Below the navigation bar are tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Identity' tab is active, and the 'Basic Info' section is highlighted in the left sidebar. The main content area contains the following fields:

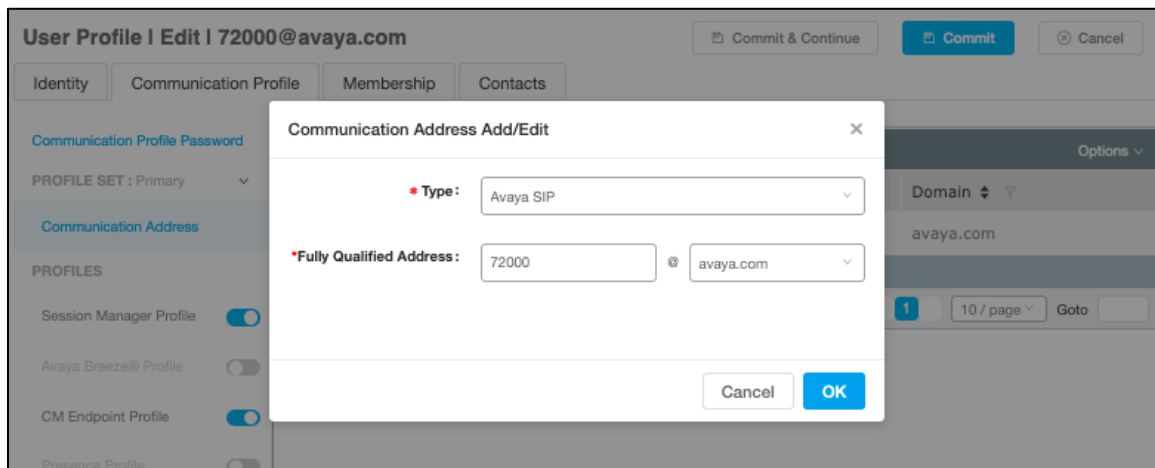
- User Provisioning Rule: [Dropdown]
- \* Last Name: [Text: MV7]
- Last Name (in Latin alphabet characters): [Text: MV7]
- \* First Name: [Text: IX]
- First Name (in Latin alphabet characters): [Text: Dev]
- \* Login Name: [Text: 72000@avaya.com]
- Middle Name: [Text: Middle Name Of User]
- Description: [Text: Description Of User]
- Email Address: [Text: Email Address Of User]
- Password: [Text: ]
- User Type: [Dropdown: Basic]
- Confirm Password: [Text: ]
- Localized Display Name: [Text: MV7, IX]
- Endpoint Display Name: [Text: MV7, Dev]
- Title Of User: [Text: Title Of User]
- Language Preference: [Dropdown: English (United States)]
- Time Zone: [Dropdown: ]
- Employee ID: [Text: Employee Id Of User]
- Department: [Text: Department Of User]
- Company: [Text: Company Of User]

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.



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.



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.

The screenshot shows the 'User Profile | Edit | 72000@avaya.com' interface. The 'Communication Profile' tab is active. On the left, the 'Session Manager Profile' is toggled on. The main configuration area includes:

- 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:  (with a question mark icon)
- 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

Buttons for 'Commit & Continue', 'Commit', and 'Cancel' are visible at the top right.

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 **72000**
- **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 | 72000@avaya.com' interface. The 'Communication Profile' tab is active. On the left, a sidebar shows profile settings: '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 the following fields:

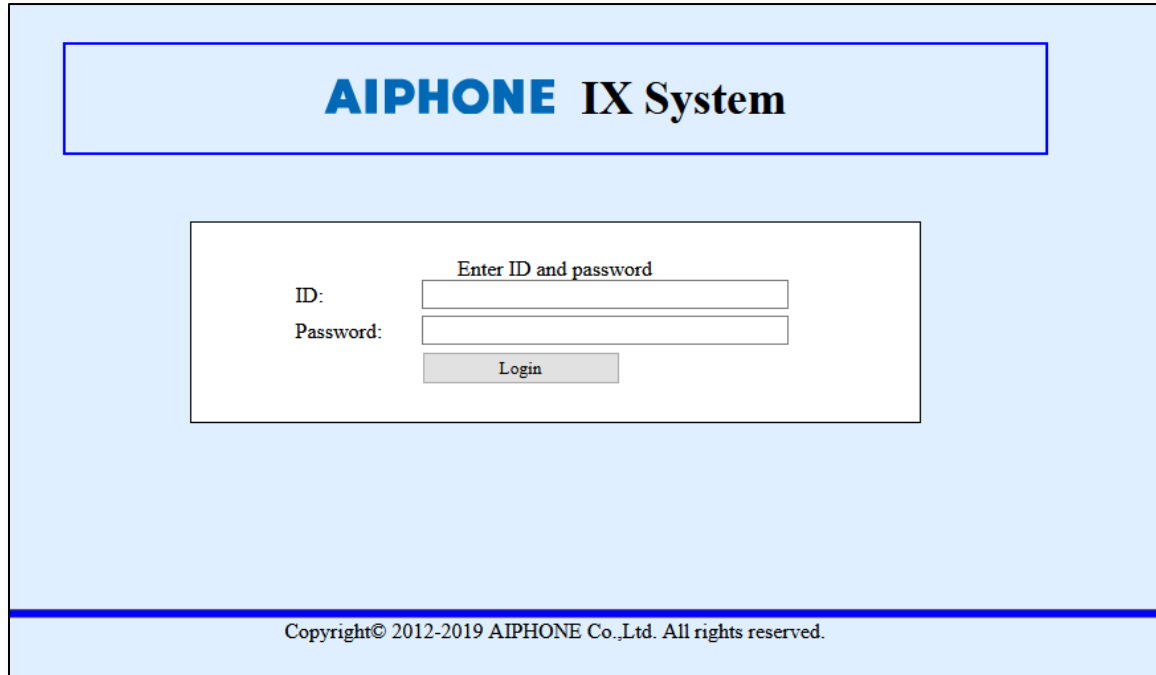
- System:** cm81
- Profile Type:** Endpoint
- Extension:** 72000
- Set Type:** J129
- Port:** S000084
- Template:** Start typing...
- Security Code:** Enter Security Code
- Preferred Handle:** Select
- Voice Mail Number:** (empty)
- Sip Trunk:** aar
- SIP URI:** 72000@avaya.com
- Enhanced Call-Info Display for 1-line phones:** (unchecked)
- Override Endpoint Name and Localized Name:** (checked)
- Use Existing Endpoints:** (unchecked)
- Calculate Route Pattern:** (unchecked)
- 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 Master Station

This section provides steps to configure Aiphone IX-MV7.

To configure Aiphone IX-MV7, using a web browser, navigate to <https://<IP Address of IX-MV7>/webset.cgi?login> and log in using appropriate credentials.



**AIPHONE IX System**

Enter ID and password

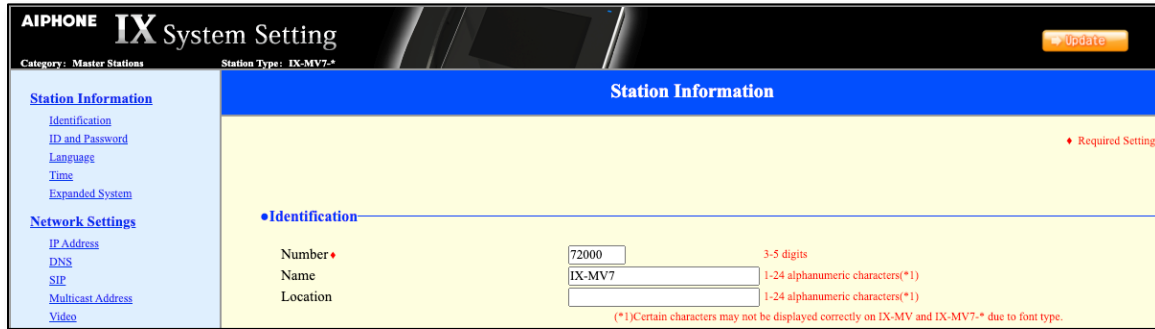
ID:

Password:

Login

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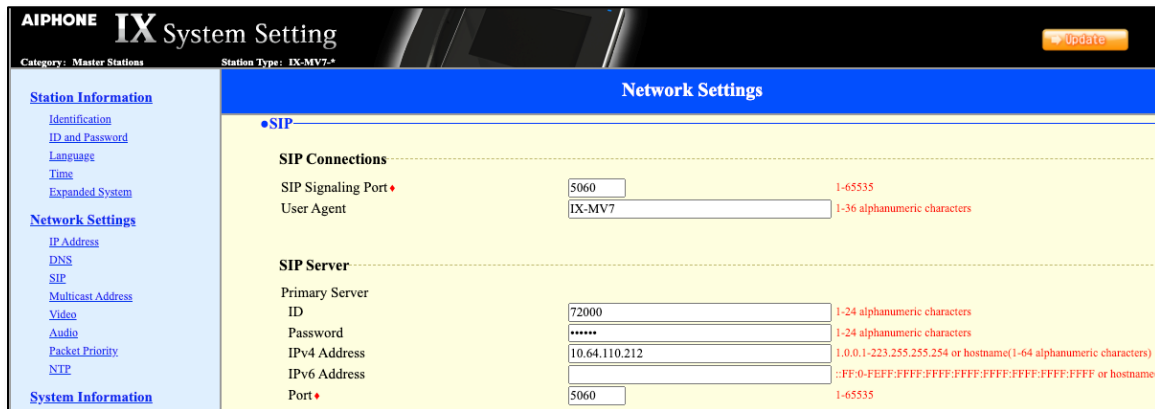
Once logged in, for the **Number** field, type in the SIP extension that is being configured (from **Section 6.2**), and a desired **Name**. Select **Update** to save changes.



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.

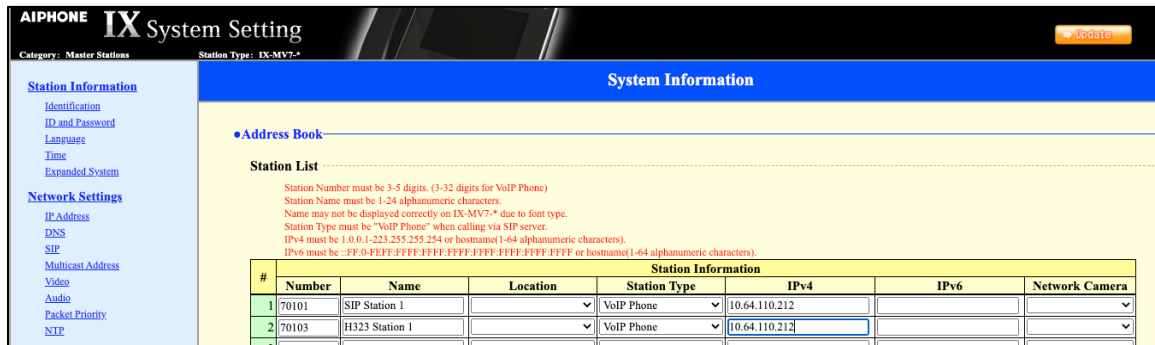




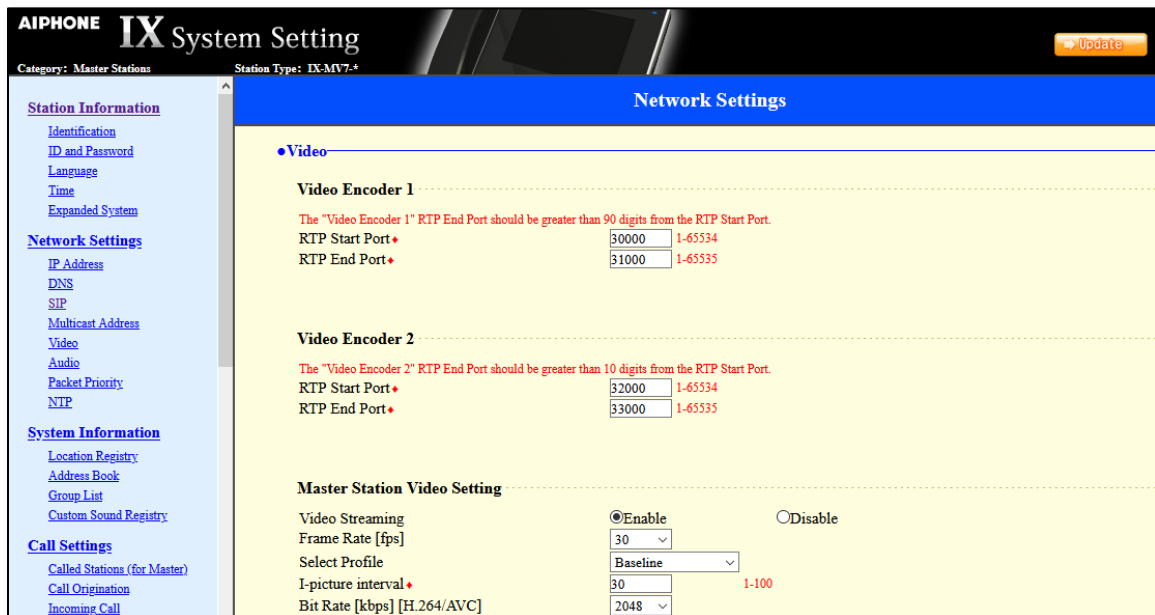
From the left, select **System Information** → **Address Book** and configure as follows:  
 The numbers configured here are added to the Address Book of IX-MV7, which makes it easier to call the number by tapping the screen instead of typing the extension.

- **Number:** Type in an extension number on IP Office that will be called for a given line
- **Name:** Desired name for the extension
- **Station Type:** Set to **VoIP Phone**
- **IPv4:** Type in the LAN IP Address for Session Manager.

Select **Update** to save changes.



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



## 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 main content area is titled "User Registrations" and includes a table with the following data:

Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered
<input checked="" type="checkbox"/>	72000@avaya.com	IX	MV7	---	192.168.4.138	<input type="checkbox"/>	<input type="checkbox"/>	1/2	<input type="checkbox"/>	<input checked="" type="checkbox"/>
<input type="checkbox"/>	---	IX	DA	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/2	<input type="checkbox"/>	<input type="checkbox"/>

Place a call from Aiphone IX-MV7 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:192.168.4.138:20000/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 Tunneler 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: 192.168.4.138 : 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-MV7 was compliance tested with Avaya Aura<sup>®</sup>. Aiphone IX-MV7 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-MV7 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/](https://www.aiphone.co.uk/featured_item/ix2/)

## Appendix A

Following devices are based on the same firmware as IX-MV7:

- IX-MV7-B
- IX-MV7-W
- IX-MV7-HB
- IX-MV7-HW
- IXMV7HBLA
- IXMV7HWLAIX-DVF-AC

The difference in each IX-MV7 devices is their mounting method:

- IX-MV7-B
  - Black
- IX-MV7-W
  - White
- IX-MV7-HB
  - Black
  - Handset
- IX-MV7-HW
  - White
  - Handset
- IXMV7HBLA
  - Black
  - Handset
  - Hearing aid
- IXMV7HWLA
  - White
  - Handset
  - Hearing aid

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