



Avaya Solution & Interoperability Test Lab

Application Notes for Aiphone IX Series 2 Video Door Stations (IX-DVM) with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate Aiphone IX Series 2 Video Door Stations (IX-DVM) Version 5.75 with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. The Aiphone IX-DVM Video Door Station, which is part of the Aiphone IX Series 2 Video Door Stations, was used for the compliance test. The Aiphone IX-DVM Video Door Station is a surface mount, weather resistant video door station. It has one dry contact that can be used to release doors when activated by a phone and can provide one-way video to a video capable SIP phone. The Aiphone IX-DVM Video Door Station registers with Avaya Aura® Session Manager as a SIP endpoint.

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

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

1. Introduction

These Application Notes describe the configuration steps required to integrate Aiphone IX Series 2 Video Door Stations (IX-DVM) Version 5.75 with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. The Aiphone IX-DVM Video Door Station, which is part of the Aiphone IX Series 2 Video Door Stations, was used for the compliance test. The Aiphone IX-DVM Video Door Station is a surface mount, weather resistant video door station. It has one dry contact that can be used to release doors when activated by a phone and can provide one-way video to a video capable SIP phone. The Aiphone IX-DVM Video Door Station (IX-DVM) registers with Avaya Aura® Session Manager as a SIP endpoint.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing audio and video calls between Aiphone IX-DVM Video Door Station, Avaya SIP and H.323 telephones, Avaya Workplace Client for Windows, Avaya Vantage™ K155, 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-DVM Video Door Station come 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-DVM Video 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-DVM with Session Manager.
- Audio calls between IX-DVM and Avaya SIP and H.323 deskphones with Direct IP Media (Shuffling) enabled and disabled.
- Audio and video calls between IX-DVM, Workplace, and Vantage K155 with Direct IP Media (Shuffling) enabled and disabled. One-way video from IX-DVM to Workplace and Vantage K155 was verified.
- Audio calls between IX-DVM and the PSTN.
- G.711 codec support.
- UDP transport protocol.
- Door release by Avaya IP endpoint using DTMF.
- IX-DVM placing, answering, and terminating calls using contactless call sensor.
- Basic telephony features, including hold/resume, mute/unmute, redial, transfer, and 3-way conference, initiated from an Avaya IP endpoint.
- Proper system recovery after re-establishing IP connectivity to IX-DVM.

2.2. Test Results

All test cases passed with the following observation:

- When a video call is originated from Workplace or Vantage, Communication Manager sends *video* media description with *inactive* media attribute in the SIP INVITE SDP to IX-DVM when Direct IP Media (Shuffling) is disabled. As a result, IX-DVM doesn't send video. If Shuffling is enabled, this issue doesn't occur.

2.3. Support

For technical support of Aiphone IX Series 2 Video Door Stations, contact Aiphone Technical Support via phone or website.

USA, Canada

- Phone: +1 (800) 692-0200
- Web: <https://www.aiphone.com/home/support>
- Email: tech@aiphone.com

Australia, New Zealand

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

France

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

Japan

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

Singapore

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

United Kingdom

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

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network that includes the following products:

- Communication Manager with G450 Media Gateway and Avaya Aura® Media Server for media resources.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP endpoints.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya 96x1 Series SIP and H.323 Deskphones.
- Avaya J100 Series SIP Deskphones.
- Avaya Workplace Client and Avaya Vantage™ video capable SIP endpoints.
- IX-DVM Video Door Station.

Aiphone IX-DVM Video Door Station registered with Session Manager and was configured as Off-PBX Stations (OPS) on Communication Manager.

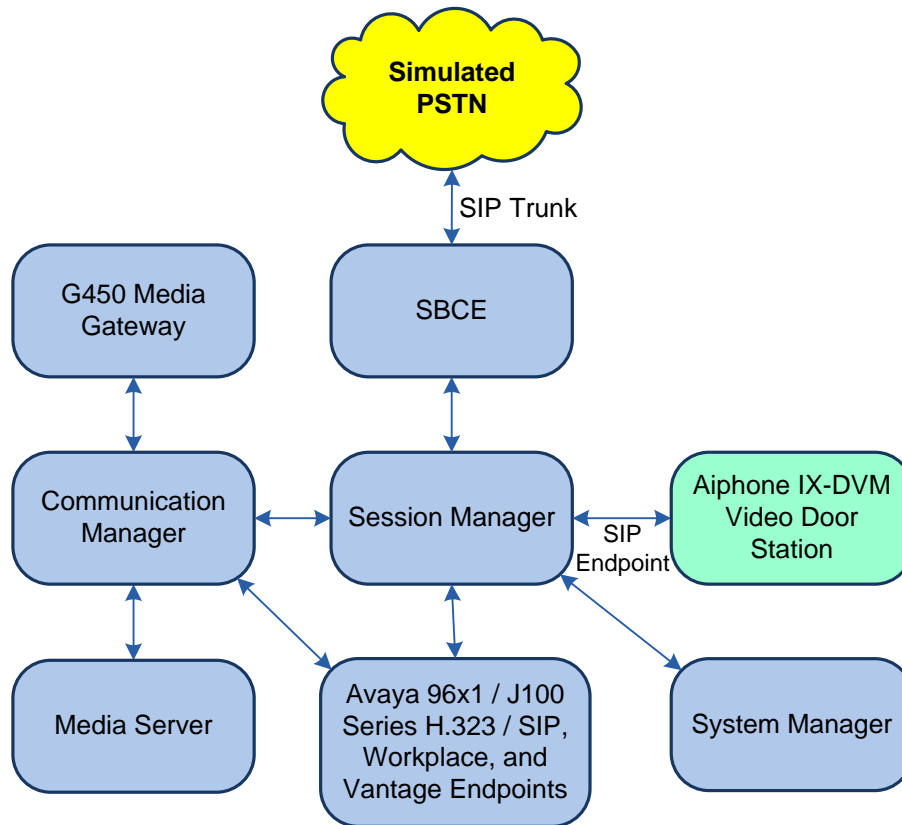


Figure 1: Avaya SIP Telephony Network with Aiphone IX-DVM Video Door Station

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	8.1.3.2.0-FP3SP2
Avaya G450 Media Gateway	FW 41.24.0
Avaya Aura® Media Server	v.8.0.2.138
Avaya Aura® System Manager	8.1.3.1 Build No. – 8.1.0.0.733078 Software Update Revision No: 8.1.3.1.1012493 Service Pack 1
Avaya Aura® Session Manager	8.1.3.1.813113
Avaya Workplace Client for Windows	3.19.0.72.19
Avaya Vantage™ K155	3.0.0.1.0006
Avaya 96x1 Series IP Deskphones	6.8502 (H.323) 7.1.13.0.4 (SIP)
Avaya J100 Series SIP Deskphones	4.0.9.0.4
Aiphone IX-DVM Video Door Station	5.75

5. Configure Avaya Aura® Communication Manager

This section describes the configuration of a SIP trunk to Session Manager and routing calls to IX-DVM. Administration of Communication Manager was performed using the System Access Terminal (SAT). The following configuration is covered:

- **Optional Features** to verify Communication Manager license.
- **IP Node Names** to associate names with IP addresses.
- **IP Codec Set** to specify the codec type used for calls to IX-DVM.
- **IP Network Region** to specify the SIP domain name, the IP codec set, and enable IP-IP direct audio (i.e., Shuffling).
- **SIP trunk** for calls towards Session Manager and IX-DVM.
- **Private Numbering** to allow the caller's extension to be sent over the SIP trunk.
- **Call Routing** to route calls to IX-DVM using AAR.

5.1. Verify Communication Manager License

Using the SAT, verify that the Off-PBX Telephones (OPS) option is enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

On **Page 1**, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints, including IX-DVM Video Door Stations, that will be deployed.

```
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    107
Maximum Stations: 36000         36
Maximum XMOBILE Stations: 36000  0
Maximum Off-PBX Telephones - EC500: 41000 0
Maximum Off-PBX Telephones - OPS: 41000 22
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.)
```

5.2. Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name for Communication Manager (*procr*) and Session Manager (*devcon-sm*). The host names will be used in other configuration screens of Communication Manager.

```
change node-names ip                                     Page 1 of 2
                                                    IP NODE NAMES
      Name                IP Address
default                 0.0.0.0
devcon-aes              10.64.102.119
devcon-ams              10.64.102.118
devcon-sm              10.64.102.117
procr                  10.64.102.115
procr6                  ::
( 6 of 6 administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

5.3. Administer IP Codec Set

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to IX-DVM. The form is accessed via the **change ip-codec-set 1** command. Note the codec set number since it will be used in the IP Network Region covered in the next section. For the compliance test, G.711MU codecs was verified. The following IP codec set is configured with G.711MU.

Media encryption was enabled for Avaya IP endpoints. IX-DVM wasn't configured to support SRTP, so the *none* option was also included under **Media Encryption**.

```
change ip-codec-set 1                                   Page 1 of 2
                                                    IP MEDIA PARAMETERS
Codec Set: 1
      Audio              Silence      Frames      Packet
      Codec             Suppression Per Pkt   Size (ms)
1: G.711MU              n           2          20
2:
3:
4:
5:
6:
7:
Media Encryption                               Encrypted SRTP: best-effort
1: 1-srtp-aescm128-hmac80
2: none
3:
4:
5:
```


On **Page 2**, enable **Allow Direct-IP Multimedia** and set **Maximum Call rate for Direct-IP Multimedia** and **Maximum Call Rate for Priority Direct-IP Multimedia** to *4096 Kbits* as shown below.

```
change ip-codec-set 1                                     Page 2 of 2

                IP MEDIA PARAMETERS

                Allow Direct-IP Multimedia? y
                Maximum Call Rate for Direct-IP Multimedia: 4096:Kbits
                Maximum Call Rate for Priority Direct-IP Multimedia: 4096:Kbits

                Mode                Redun-                Packet
                t.38-standard        dancy                Size (ms)
FAX                0      ECM: y
Modem                off                0
TDD/TTY                US                3
H.323 Clear-channel  n                0
SIP 64K Data         n                0                20

Media Connection IP Address Type Preferences
1: IPv4
2:
```

5.4. Administer IP Network Region

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IX-DVM and IP endpoints without using media resources in the Avaya G450 Media Gateway or Avaya Media Server. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region 1) is specified in the SIP signaling group.

```
change ip-network-region 1                                     Page 1 of 20
                                                           IP NETWORK REGION
  Region: 1          NR Group: 1
Location: 1        Authoritative Domain: avaya.com
  Name:              Stub Network Region: n
MEDIA PARAMETERS   Intra-region IP-IP Direct Audio: yes
                   Inter-region IP-IP Direct Audio: yes
                   IP Audio Hairpinning? n
  Codec Set: 1
  UDP Port Min: 2048
  UDP Port Max: 50999
DIFFSERV/TOS PARAMETERS
  Call Control PHB Value: 46
  Audio PHB Value: 46
  Video PHB Value: 26
802.1P/Q PARAMETERS
  Call Control 802.1p Priority: 6
  Audio 802.1p Priority: 6
  Video 802.1p Priority: 5
AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS   RSVP Enabled? n
  H.323 Link Bounce Recovery? y
  Idle Traffic Interval (sec): 20
  Keep-Alive Interval (sec): 5
  Keep-Alive Count: 5
```

5.5. Administer SIP Trunk to Session Manager

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tls*.
- Specify the Ethernet processor (*procr*) of Communication Manager and Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form in **Section 5.2**.
- Set **IP Video** field to *y*.
- Ensure that the TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.
- Set **Initial IP-IP Direct Media** field to *y*.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
add signaling-group 10                                     Page 1 of 2
                                     SIGNALING GROUP
Group Number: 10                                         Group Type: sip
  IMS Enabled? n                                         Transport Method: tls
    Q-SIP? n
  IP Video? y                                           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: devcon-sm
  Near-end Listen Port: 5061                             Far-end Listen Port: 5061
                                                         Far-end Network Region: 1
Far-end Domain: avaya.com
                                                         Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                    RFC 3389 Comfort Noise? n
  DTMF over IP: rtp-payload                             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                      IP Audio Hairpinning? n
  Enable Layer 3 Test? y                                Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? n                  Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for SIP calls to IX-DVM. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Accept the default values for the remaining fields.

```

add trunk-group 10                                     Page 1 of 5
                                     TRUNK GROUP
Group Number: 10                                     Group Type: sip                                     CDR Reports: y
  Group Name: To devcon-sm                           COR: 1                                     TN: 1       TAC: 1010
  Direction: two-way                                 Outgoing Display? n
  Dial Access? n                                     Night Service:
  Queue Length: 0
Service Type: tie                                  Auth Code? n
                                                    Member Assignment Method: auto
                                                    Signaling Group: 10
                                                    Number of Members: 10

```

On **Page 3** of the trunk group form, set the **Numbering Format** field to *private*. This field specifies the format of the calling party number sent to the far-end.

```

add trunk-group 10                                     Page 3 of 5
TRUNK FEATURES
  ACA Assignment? n                                   Measured: none
                                                    Maintenance Tests? y

  Suppress # Outpulsing? n Numbering Format: private
                                                    UUI Treatment: service-provider
                                                    Maximum Size of UUI Contents: 128
                                                    Replace Restricted Numbers? n
                                                    Replace Unavailable Numbers? n

                                                    Hold/Unhold Notifications? y
  Modify Tandem Calling Number: no

  Show ANSWERED BY on Display? y

```

On **Page 4** of the trunk group form, the default settings were used as shown below.

```

add trunk-group 10                                     Page 5 of 5
                PROTOCOL VARIATIONS

                Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                Send Transferring Party Information? n
                Network Call Redirection? n

                Send Diversion Header? n
                Support Request History? y
                Telephone Event Payload Type:

                Convert 180 to 183 for Early Media? n
                Always Use re-INVITE for Display Updates? n
                Identity for Calling Party Display: P-Asserted-Identity
Block Sending Calling Party Location in INVITE? n
                Accept Redirect to Blank User Destination? n
                Enable Q-SIP? n
Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                Request URI Contents: may-have-extra-digits
    
```

5.6. Configure Private Numbering

Configure the **Numbering – Private Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with ‘7’ whose calls are routed over any trunk group, including SIP trunk group 10, have their extension sent.

```

change private-numbering 0                             Page 1 of 2
                NUMBERING - PRIVATE FORMAT

Ext Ext          Trk      Private      Total
Len Code        Grp(s)    Prefix      Len
  5  7
  5          Total Administered: 1
                Maximum Entries: 540
    
```

5.7. AAR Call Routing

Configure the uniform dial plan table to route calls using AAR for dialed digits that are 5-digits long and begin with ‘78’. This would cover call routing to IX-DVM (e.g., 78010).

```

change uniform-dialplan 7                             Page 1 of 2
                UNIFORM DIAL PLAN TABLE

                Percent Full: 0

Matching          Insert          Node
Pattern          Len Del        Digits      Net Conv Num
  78                5  0          aar          n
    
```

SIP calls to Session Manager are routed over a SIP trunk via AAR call routing. Configure the AAR analysis form and add an entry that routes digits beginning with “78” to route pattern 10 as shown below. Note that the **Call Type** was set to *lev0*. This entry routes calls to SIP stations and to IX-DVM.

```
change aar analysis 7
```

Page 1 of 2

AAR DIGIT ANALYSIS TABLE
Location: all Percent Full: 2

	Dialed String	Total		Route Pattern	Call Type	Node Num	ANI Reqd
		Min	Max				
7		7	7	254	aar		n
78		5	5	10	lev0		n
8		7	7	254	aar		n
9		7	7	254	aar		n
							n
							n

Configure a preference in **Route Pattern 10** to route calls over SIP trunk group 10 as shown below.

```
change route-pattern 10
```

Page 1 of 3

Pattern Number: 10 **Pattern Name: To devcon-sm**

SCCAN? n Secure SIP? n Used for SIP stations? n

Grp No	FRL	NPA	Pfx Mrk	Hop Lmt	Toll List	No. Del	Inserted Digits	DCS/ QSIG	IXC
1:	10	0						n	user
2:								n	user
3:								n	user
4:								n	user
5:								n	user
6:								n	user

	BCC	VALUE	TSC	CA-TSC	ITC	BCIE	Service/Feature	PARM	Sub	Numbering	LAR
	0	1	2	M	4	W	Request	Dgts	Format		
1:	y	y	y	y	y	n	n	rest		unk-unk	none
2:	y	y	y	y	y	n	n	rest			none
3:	y	y	y	y	y	n	n	rest			none
4:	y	y	y	y	y	n	n	rest			none
5:	y	y	y	y	y	n	n	rest			none
6:	y	y	y	y	y	n	n	rest			none

6. Configure Avaya Aura® Session Manager

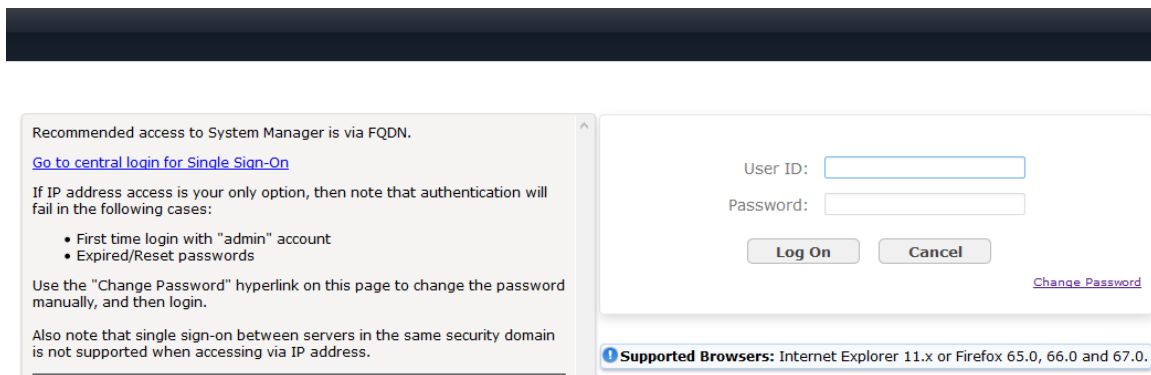
This section covers the procedure for adding a SIP user in Session Manager. The configuration covers:

- Launch System Manager
- Set Network Transport Protocol for IX-DVM
- Administer SIP User

Note: It is assumed that basic configuration of Session Manager has already been performed.

6.1. Launch System Manager

Access the System Manager Web interface by using the URL <https://<ip-address>> in an Internet browser window, where *<ip-address>* is the System Manager IP address. Log in using the appropriate credentials.



Recommended access to System Manager is via FQDN.
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

User ID:

Password:

[Change Password](#)

Supported Browsers: Internet Explorer 11.x or Firefox 65.0, 66.0 and 67.0.

6.2. Set Network Transport Protocol for IX-DVM

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

The screenshot shows the Avaya Aura System Manager 8.1 interface. The left sidebar is expanded to 'Routing' > 'SIP Entities'. The main content area displays 'SIP Entity Details' for the entity 'devcon-sm'. The 'General' section includes fields for Name (devcon-sm), IP Address (10.64.102.117), SIP FQDN, Type (Session Manager), Notes, Location (Thornton), Outbound Proxy, Time Zone (America/New_York), Minimum TLS Version (Use Global Setting), and Credential name. The 'Monitoring' section includes SIP Link Monitoring and CRLF Keep Alive Monitoring, both set to 'Use Session Manager Configuration'. Buttons for 'Commit' and 'Cancel' are visible at the top right.

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

Listen Ports

Add Remove

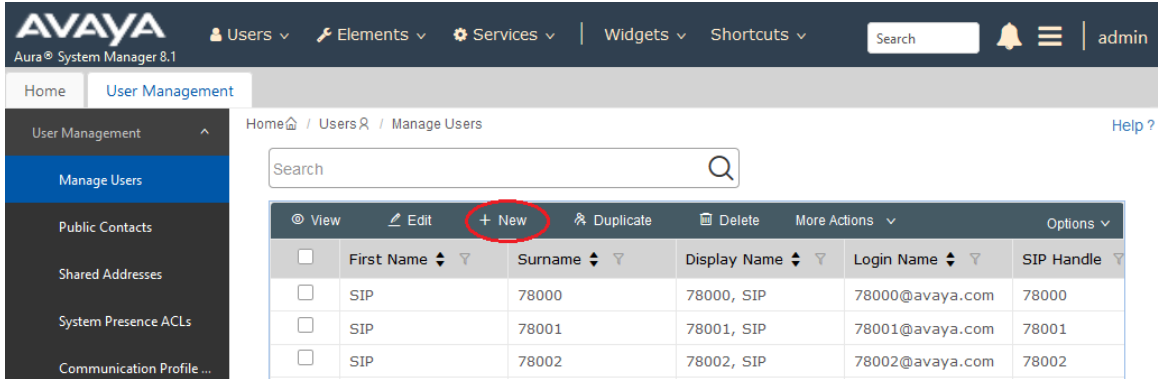
3 Items Filter: Enable

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

Select : All, None

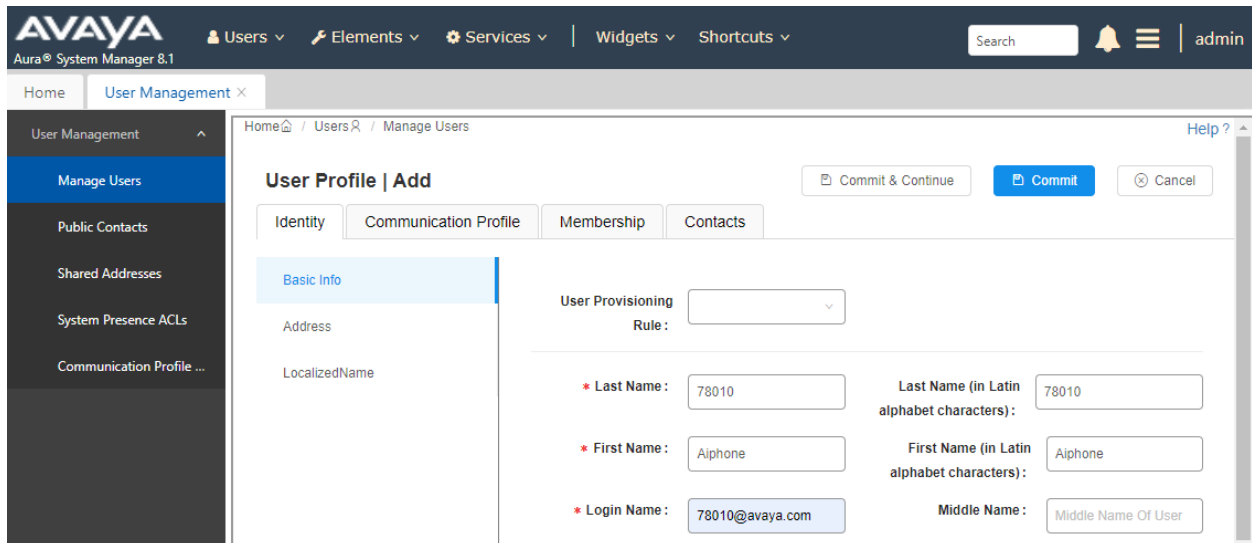
6.3. Administer SIP User

In the subsequent screen (not shown), select **Users** → **User Management** → **Manage Users** to display the **User Management** screen below. Click **New** to add a user.



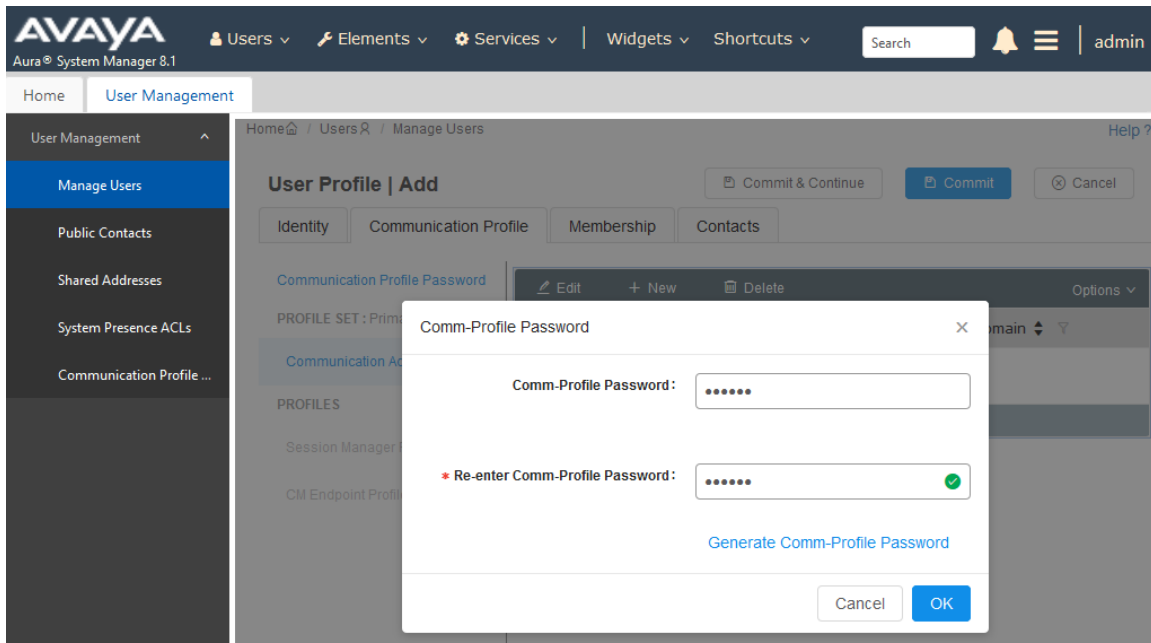
6.3.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter `<extension>@<domain>`, where `<extension>` is the desired IX-DVM SIP extension and `<domain>` is the applicable SIP domain name from **Section 5.4**. Retain the default values in the remaining fields.



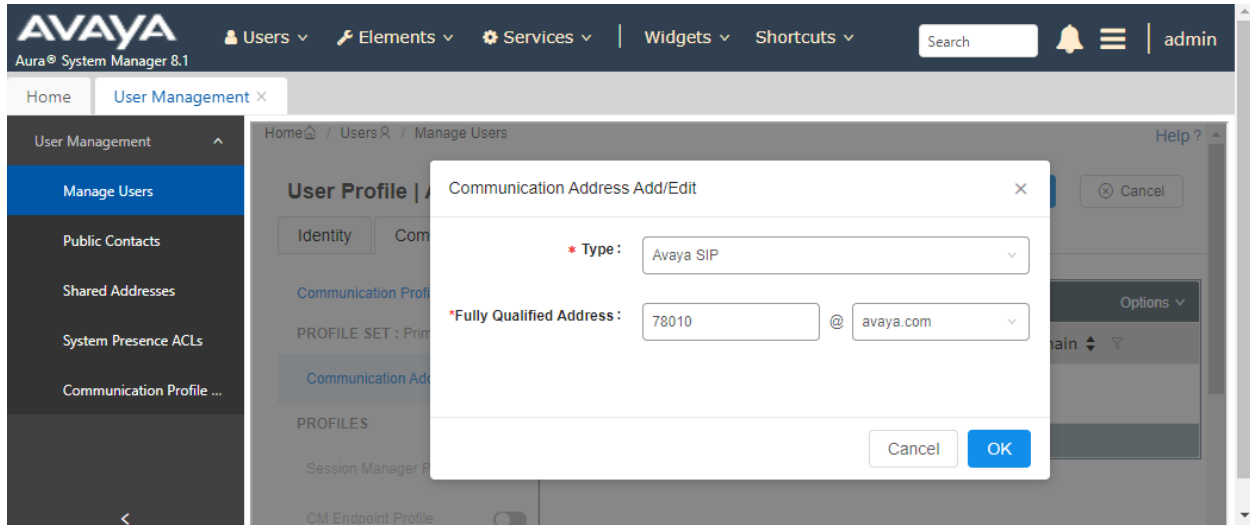
6.3.2. Communication Profile

Select the **Communication Profile** tab. Next, click on **Communication Profile Password**. For **Comm-Profile Password** and **Re-enter Comm-Profile Password**, enter the desired password for the SIP user to use for registration. Click **OK**.



6.3.3. Communication Address

Click on **Communication Address** and then click **New** to add a new entry. The **Communication Address Add/Edit** dialog box is displayed as shown below. For **Type**, retain *Avaya SIP*. For **Fully Qualified Address**, enter the SIP user extension and select the domain name to match the login name from **Section 6.3.1**. Click **OK**.



6.3.4. Session Manager Profile

Click on toggle button by **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.

The screenshot displays the 'User Profile | Add' configuration page in the Avaya Aura System Manager 8.1 interface. The left sidebar shows the 'User Management' menu with 'Manage Users' selected. The main content area is divided into tabs: 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' tab is active, showing a 'Communication Profile Password' field and a 'PROFILE SET' dropdown set to 'Primary'. Below this, there are 'PROFILES' with 'Session Manager Profile' and 'CM Endpoint Profile' both toggled on. The 'SIP Registration' section contains several fields: 'Primary Session Manager' (devcon-sm), 'Secondary Session Manager' (Start typing...), 'Survivability Server' (Start typing...), and 'Max. Simultaneous Devices' (1). There is also a 'Block New Registration When Maximum Registrations Active?' checkbox. The 'Application Sequences' section shows 'Origination Sequence' and 'Termination Sequence' both set to 'DEVCON-CM App S...'. At the bottom, the 'Call Routing Settings' section is partially visible, showing 'Home Location' (Thornton) and 'Conference Factory Set' (Select).

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

This close-up screenshot shows the 'Call Routing Settings' section. It includes a 'Home Location' field with the value 'Thornton' and a 'Conference Factory Set' dropdown menu currently set to 'Select'.

6.3.5. CM Endpoint Profile

Click on the toggle button by **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.1**. For **Template**, select *9641SIP_DEFAULT_CM_8_1*. For **Port**, click and select *IP*. Retain the default values in the remaining fields. Click on **Endpoint Editor** (i.e, Edit icon in **Extension** field) to enable **IP Video**.

The screenshot displays the Avaya Aura System Manager 8.1 interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 8.1', and navigation menus for Users, Elements, Services, Widgets, and Shortcuts. A search bar and user profile (admin) are also visible. The main content area is titled 'User Profile | Add' and features tabs for Identity, Communication Profile, Membership, and Contacts. The 'Communication Profile' tab is active, showing a 'Communication Profile Password' section with 'PROFILE SET : Primary' and 'Communication Address'. Below this is a 'PROFILES' section with two toggle switches: 'Session Manager Profile' (off) and 'CM Endpoint Profile' (on). The main form area contains various fields: 'System' (devcon-cm), 'Profile Type' (Endpoint), 'Extension' (78010), 'Template' (9641SIP_DEFAULT_CM_8_1), 'Set Type' (9641SIP), 'Security Code' (Enter Security Code), 'Port' (IP), 'Voice Mail Number', 'Preferred Handle' (Select), 'Sip Trunk' (aar), 'SIP URI' (Select), 'Enhanced Callr-Info Display for 1-line phones' (off), 'Delete on Unassign from User or on Delete User' (checked), 'Override Endpoint Name and Localized Name' (checked), and 'Allow H.323 and SIP Endpoint Dual Registration' (off). Action buttons 'Commit & Continue', 'Commit', and 'Cancel' are located at the top right of the form.

In the **Endpoint Editor**, navigate to the **Feature Options** tab and enable **IP Video** under **Features** as shown below.

General Options (G) *	Feature Options (F)	Site Data (S)	Abbreviated Call Dialing (A)	Enhanced Call Fwd (E)
Button Assignment (B)	Profile Settings (P)	Group Membership (M)		
Active Station Ringing	single ▾	Auto Answer	none ▾	
MWI Served User Type	None ▾	Coverage After Forwarding	system ▾	
Per Station CPN - Send Calling Number	None ▾	Display Language	english ▾	
IP Phone Group ID	<input type="text"/>	Hunt-to Station	<input type="text"/>	
Remote Soft Phone Emergency Calls	as-on-local ▾	Loss Group	19	
LWC Reception	spe ▾	Survivable COR	internal ▾	
AUDIX Name	None ▾	Time of Day Lock Table	None ▾	
EC500 State	enabled ▾	Voice Mail Number	<input type="text"/>	
Short/Prefixed Registration Allowed	default ▾	Bridging Tone for This Extension	None ▾	
Music Source	<input type="text"/>			
Features				
<input type="checkbox"/> Always Use	<input type="checkbox"/> Idle Appearance Preference	<input type="checkbox"/> IP SoftPhone	<input checked="" type="checkbox"/> LWC Activation	<input type="checkbox"/> CDR Privacy
<input type="checkbox"/> IP Audio Hairpinning	<input type="checkbox"/> Direct IP-IP Audio Connections	<input type="checkbox"/> H.320 Conversion	<input checked="" type="checkbox"/> IP Video	<input type="checkbox"/> Per Button Ring Control
<input type="checkbox"/> Bridged Call Alerting	<input type="checkbox"/> H.320 Conversion	<input checked="" type="checkbox"/> IP Video	<input type="checkbox"/> Per Button Ring Control	
<input type="checkbox"/> Bridged Idle Line Preference				
<input checked="" type="checkbox"/> Coverage Message Retrieval				
<input type="checkbox"/> Data Restriction				
<input checked="" type="checkbox"/> Survivable Trunk Dest				
<input type="checkbox"/> Bridged Appearance Origination Restriction				
<input checked="" type="checkbox"/> Restrict Last Appearance				
<input type="checkbox"/> Turn on mute for remote off-hook attempt				
<input type="checkbox"/> IP Hoteling				

7. Configure Aiphone IX-DVM Video Door Station

This section provides the procedure for configuring IX-DVM to provide SIP connectivity to Session Manager. Configuration of IX-DVM is performed via Aiphone IX System web interface.

- Log into Aiphone IX System Web Interface
- Administer Station Information
- Administer SIP Parameters
- Administer Video SIP Channel
- Administer Audio Settings
- Administer Call Settings

7.1. Log into Aiphone IX System Web Interface

Access the Aiphone IX System Web Interface by using the URL <https://<ip-address>/webset.cgi?login> in an Internet browser, where <ip-address> is the IX-DVM IP address. Select language (not shown) and log in using the appropriate credentials.

AIPHONE IX System

Enter ID and password

ID:

Password:

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7.2. Administer Station Information

Navigate to **Station Information** → **Identification** and set the **Number** to the IX-DVM SIP extension (e.g., 78010).

The screenshot shows the AIPHONE IX System Setting web interface. The top navigation bar includes the AIPHONE logo, the text 'IX System Setting', and an 'Update' button. Below the navigation bar, the page is titled 'Station Information' and shows 'Category: Video Stations' and 'Station Type: IX-DVM'. A left sidebar contains a menu with 'Station Information' (selected) and 'Network Settings'. Under 'Station Information', there are links for 'Identification', 'ID and Password', 'Language', 'Time', and 'Expanded System'. Under 'Network Settings', there are links for 'IP Address', 'DNS', 'SIP', 'Multicast Address', 'Video', 'Audio', 'Packet Priority', and 'NTP'. The main content area is titled 'Station Information' and contains a section for 'Identification' with a red asterisk indicating required settings. The 'Number' field is set to '78010' with a red note '3-5 digits'. The 'Name' field is set to 'IX-DVM' with a red note '1-24 alphanumeric characters(*1)'. The 'Location' field is empty with a red note '1-24 alphanumeric characters(*1)'. A red note at the bottom states: '(*1)Certain characters may not be displayed correctly on IX-MV and IX-MV7-* due to font type.'

7.3. Administer SIP Parameters

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

- **SIP Signaling Port:** Set to *5060*.
- **User Agent:** Enter desired value (e.g., *IX-DVM*).
- **ID:** Set to SIP extension (e.g., *78010*) from **Section 6.3.1**.
- **Password:** Enter SIP password from **Section 6.3.2**.
- **IPv4 Address:** Set to signaling IP address of Session Manager (e.g., *10.64.102.117*).
- **Port:** Set to *5060*.

Click **Update** to save changes.

The screenshot shows the 'AIPHONE IX System Setting' web interface. The top navigation bar includes 'Category: Video Stations' and 'Station Type: IX-DVM'. A blue header for 'Network Settings' is visible. On the left, a sidebar lists various settings categories: Station Information, Network Settings, System Information, and Call Settings. The main content area is titled 'SIP' and contains two sections: 'SIP Connections' and 'SIP Server'. The 'SIP Connections' section includes fields for 'SIP Signaling Port' (set to 5060) and 'User Agent' (set to IX-DVM). The 'SIP Server' section includes a dropdown for 'SIP Compatibility Mode' (Standard Mode) and several input fields for 'Primary Server' details: ID (78010), Password (masked with dots), IPv4 Address (10.64.102.117), IPv6 Address (empty), and Port (5060). Each field has a red text label indicating its format or constraints. An 'Update' button is located in the top right corner.

Parameter	Value	Constraint
SIP Signaling Port	5060	1-65535
User Agent	IX-DVM	1-36 alphanumeric characters
SIP Compatibility Mode	Standard Mode	
Primary Server ID	78010	1-24 alphanumeric characters
Primary Server Password	*****	1-24 alphanumeric characters
Primary Server IPv4 Address	10.64.102.117	1.0.0.1-223.255.255.254 or host
Primary Server IPv6 Address		::FF:0-FE:FF:FFFF:FFFF:FFFF
Primary Server Port	5060	1-65535

7.4. Administer Video SIP Channel

Navigate to **Network Settings** → **Video** in the left pane and configure the video settings as shown below.

The screenshot displays the AIPHONE IX System Setting web interface. The top navigation bar includes the product name and a category of 'Video Stations'. The left sidebar contains a tree view of settings categories, with 'Network Settings' selected and 'Video' highlighted. The main content area is titled 'Network Settings' and is divided into several sections: 'Miscellaneous', 'Multicast Address', and 'Video'. The 'Video' section is further divided into 'SIP Channel' settings. The 'SIP Channel' section includes a warning message and several configuration options for resolution, wide view, frame rate, select profile, I-picture interval, bit rate, RTP start port, and RTP end port.

Setting	Value	Default/Range
Register Transmission Interval [sec]	3600	10-14400
DTMF digit interval timeout [sec]	5	1-10
Call health check timer	80-3600 sec (dropdown) 90 sec (input)	Do not transmit re-INVITE, 80-3600 sec
For Call IPv4	[] . [] . [] . []	224.0.0.0-239.255.255.255
For Call IPv6	[]	FF10::0-FF1F:FFFF:FFFF:FFFF:FFFF:FFFF:F
Resolution	<input checked="" type="radio"/> 640x480(VGA)	<input type="radio"/> 320x240(QVGA)
Wide View	<input checked="" type="radio"/> Disable	<input type="radio"/> Enable
Frame Rate [fps]	30 (dropdown)	
Select Profile	High (dropdown)	
I-picture interval	30	1-100
Bit rate [kbps]	1024 (dropdown)	
RTP Start Port	30000	1-65534
RTP End Port	31000	1-65535

7.5. 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 web interface. The top navigation bar includes the product name and an 'Update' button. The left sidebar contains a tree view with categories: Station Information, Network Settings, System Information, Call Settings, and Option Input / Relay. The main content area is titled 'Network Settings' and features a sub-section for 'Audio'. This section contains several configuration fields: 'Audio Codec' (radio buttons for G.711(μ-law), G.711(A-law), and G.722), 'Audio RTP Transmission Interval [msec]' (dropdown menu set to 20), and 'RTP Idle Detection Time [sec]' (input field set to 10). Below these are sections for 'SIP Channel' and 'ONVIF Transmit Channel', each with 'RTP Start Port' and 'RTP End Port' fields. The 'Audio Buffer' section includes 'Packets Buffered at Audio Start' (dropdown menu set to 1) and 'Maximum Packets Buffered' (dropdown menu set to 3). Red text provides warnings about RTP port ranges and codec changes.

Station Information

- Identification
- ID and Password
- Language
- Time
- Expanded System

Network Settings

- IP Address
- DNS
- SIP
- Multicast Address
- Video
- Audio
- Packet Priority
- NTP

System Information

- Custom Sound Registry

Call Settings

- Station Settings
- Called Stations (for Door)
- Call Origination
- Incoming Call
- Contactless Call

Option Input / Relay

Output Settings

- Option Input
- Relay Output

Network Settings

Audio

The "SIP Channel" RTP End Port should be greater than 210 digits from the RTP Start Port.
The "ONVIF Transmit Channel" RTP End Port should be greater than 10 digits from the RTP Start Port.
Changing Audio Codec from G.711(μ-law) / G.711(A-law) to G.722, or from G.722 to G.711(μ-law) / G.711(A-law) will cause the station to restart after Update is clicked

Audio Codec G.711(μ-law) G.711(A-law) G.722
Audio RTP Transmission Interval [msec] 20
RTP Idle Detection Time [sec] 10
This setting is ignored when transmitting to multiple stations (paging, etc.)
10-180 sec

SIP Channel

RTP Start Port 20000 1-65534
RTP End Port 21000 1-65535

ONVIF Transmit Channel

RTP Start Port 22000 1-65534
RTP End Port 23000 1-65535

Audio Buffer

Packets Buffered at Audio Start 1
Maximum Packets Buffered 3 Maximum Packet Buffer must be larger than Audio Start Buffer.

7.6. 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., 78040), set the **IPv4 Address** to the signaling IP address of Session Manager (e.g., 10.64.102.117), 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
 If designating "M", multicast IP addresses must be configured for the station(s).

#	Station Number	IPv4 Address	IPv6 Address	Station Type	Protocol
1	78040	10.64.102.117		VoIP Phone	U
2					
3					

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager, Session Manager, and Aiphone IX-DVM Video Door Station.

1. Verify that IX-DVM has successfully registered with Session Manager. In System Manager, navigate to **Elements** → **Session Manager** → **System Status** → **User Registrations** to check the registration status as shown below.

The screenshot shows the Avaya Aura System Manager 8.1 interface. The main content area displays the 'User Registrations' page. The page includes a navigation sidebar on the left, a top header with the Avaya logo and user information, and a main content area with 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' (with sub-columns for Prim, Sec, Surv, and Visiting). The row for '78010@avaya.com' is highlighted in red. The table also shows a 'Show 15' dropdown and a 'Filter: Enable' option.

	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered			
											Prim	Sec	Surv	Visiting
<input type="checkbox"/>	Show	---	Agent	78004	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	78030@avaya.com	Agent	78030	---	192.168.100.49	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	78003@avaya.com	SIP	78003	---	192.168.100.64	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	Equinox	78040	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	78010@avaya.com	Aiphone	78010	---	192.168.100.180	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

2. Establish inbound and outbound video calls to IX-DVM with Workplace and/or Vantage and verify two-way audio and one-way video.

9. Conclusion

These Application Notes describe the administration steps required to integrate Aiphone IX Series 2 Video Door Stations (IX-DVM) with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Aiphone IX-DVM Video Door Station successfully registered with Avaya Aura® Session Manager as a SIP endpoint and audio and video calls were verified. All test cases passed with observations noted in **Section 2.2**.

10. References

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

- [1] *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 10, March 2021, available at <http://support.avaya.com>.
- [2] *Administering Avaya Aura® System Manager for Release 8.1.x*, Release 8.1.x, Issue 11, April 2021, available at <http://support.avaya.com>.
- [3] *Administering Avaya Aura® Session Manager*, Release 8.1.x, Issue 8, February 2021, available at <http://support.avaya.com>.
- [4] *Aiphone IX-DVM Video Door Station Installation Manual*, Issue Date: Oct.2021, available from Aiphone.
- [5] *Aiphone IX Series Operation Manual*, Software version 5.75 or later, available from Aiphone.

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