



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.

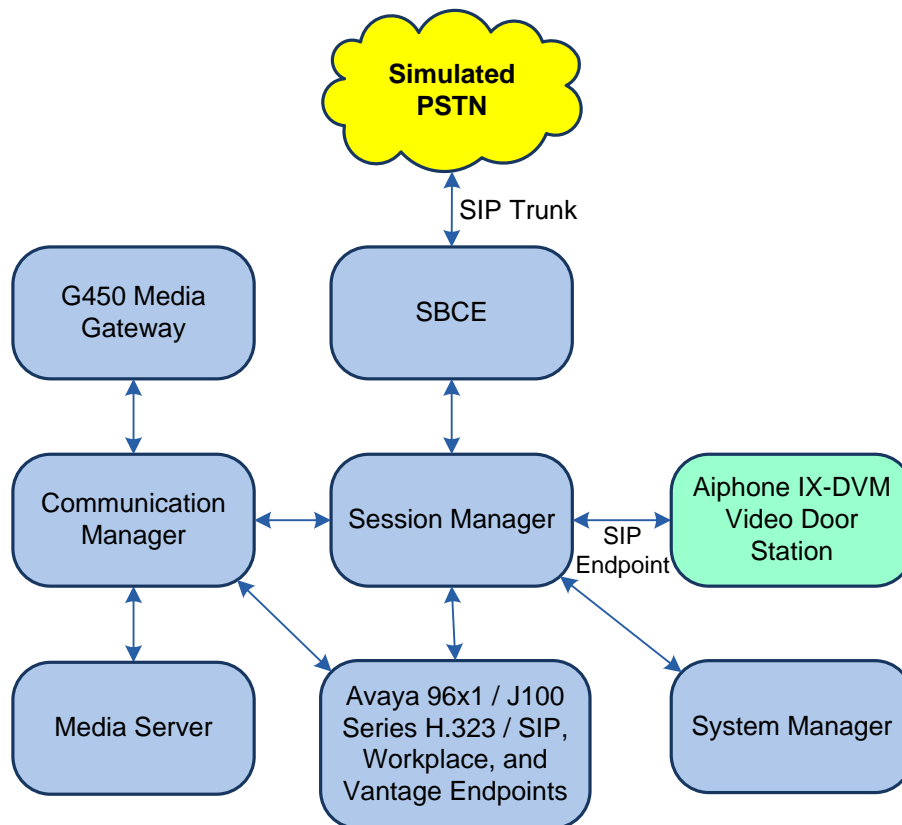


Figure 1: Avaya SIP Telephony Network with Aiphone IX-DVM Video Door Station

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 Codec | Silence Suppression | Frames Per Pkt |
| 1: G.711MU | n | 2 |
| 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)
                                off                                  0      ECM: y
                                US                                   3
                                H.323 Clear-channel                 0
                                SIP 64K Data                         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 | | |
| Codec Set: 1 | Intra-region IP-IP Direct Audio: yes | |
| | Inter-region IP-IP Direct Audio: yes | |
| UDP Port Min: 2048 | IP Audio Hairpinning? n | |
| 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 | | |
| Incoming Dialog Loopbacks: eliminate | | Bypass If IP Threshold Exceeded? n |
| DTMF over IP: rtp-payload | | RFC 3389 Comfort Noise? n |
| Session Establishment Timer(min): 3 | Direct IP-IP Audio Connections? y | |
| Enable Layer 3 Test? y | IP Audio Hairpinning? n | |
| H.323 Station Outgoing Direct Media? n | Initial IP-IP Direct Media? y | |
| | | 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 |
| <p>PROTOCOL VARIATIONS</p> <p>Mark Users as Phone? n</p> <p>Prepend '+' to Calling/Alerting/Diverting/Connected Number? n</p> <p>Send Transferring Party Information? n</p> <p>Network Call Redirection? n</p> <p>Send Diversion Header? n</p> <p>Support Request History? y</p> <p>Telephone Event Payload Type:</p> <p>Convert 180 to 183 for Early Media? n</p> <p>Always Use re-INVITE for Display Updates? n</p> <p>Identity for Calling Party Display: P-Asserted-Identity</p> <p>Block Sending Calling Party Location in INVITE? n</p> <p>Accept Redirect to Blank User Destination? n</p> <p>Enable Q-SIP? n</p> <p>Interworking of ISDN Clearing with In-Band Tones: keep-channel-active</p> <p>Request URI Contents: may-have-extra-digits</p> | |

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 | | | | | | | | | | | | | | | | | | |
|--|-------------|------------|----------------|------------|-----------------------|-----------|--|---|---|--|--|---|-----------------------|--|--|--|--|--|----------------------|
| <p>NUMBERING - PRIVATE FORMAT</p> <table border="1"> <thead> <tr> <th>Ext Len</th> <th>Ext Code</th> <th>Trk Grp(s)</th> <th>Private Prefix</th> <th>Total Len</th> <th></th> </tr> </thead> <tbody> <tr> <td>5</td> <td>7</td> <td></td> <td></td> <td>5</td> <td>Total Administered: 1</td> </tr> <tr> <td colspan="5"></td> <td>Maximum Entries: 540</td> </tr> </tbody> </table> | | Ext Len | Ext Code | Trk Grp(s) | Private Prefix | Total Len | | 5 | 7 | | | 5 | Total Administered: 1 | | | | | | Maximum Entries: 540 |
| Ext Len | Ext Code | Trk Grp(s) | Private Prefix | Total 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 | | | | | | | | | | | | |
|--|-------------|------------------|---------------|----------|---------------|----------|----------|----|---|---|--|-----|---|
| <p>UNIFORM DIAL PLAN TABLE</p> <p>Percent Full: 0</p> <table border="1"> <thead> <tr> <th>Matching Pattern</th> <th>Len</th> <th>Del</th> <th>Insert Digits</th> <th>Net Conv</th> <th>Node Num</th> </tr> </thead> <tbody> <tr> <td>78</td> <td>5</td> <td>0</td> <td></td> <td>aar</td> <td>n</td> </tr> </tbody> </table> | | Matching Pattern | Len | Del | Insert Digits | Net Conv | Node Num | 78 | 5 | 0 | | aar | n |
| Matching Pattern | Len | Del | Insert Digits | Net Conv | Node Num | | | | | | | | |
| 78 | 5 | 0 | | aar | n | | | | | | | | |

| AAR DIGIT ANALYSIS TABLE | | | | | | Page 1 of 2 |
|--------------------------|-----------|-----------|---------------|-------------|-----------------|-------------|
| Location: all | | | | | Percent Full: 2 | |
| Dialed String | Total Min | Total Max | Route Pattern | Call Type | Node Num | ANI Req'd |
| 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 |

| 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 | FRL | NPA | Pfx | Hop | Toll | No. | Inserted | | | | | DCS/ | IXC |
| No | | | Mrk | Lmt | List | Del | Digits | | | | | QSIG | |
| | | | | | | | Dgts | | | | | Intw | |
| 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 | | | 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 top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The left sidebar shows a tree view with 'Routing' expanded, and 'SIP Entities' selected. The main content area is titled 'SIP Entity Details' and contains two sections: 'General' and 'Monitoring'. 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 in the top right.

| Field | Value |
|----------------------------|-----------------------------------|
| 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 |
| Credential name | |
| SIP Link Monitoring | Use Session Manager Configuration |
| CRLF Keep Alive Monitoring | Use Session Manager Configuration |

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.

Avaya Aura System Manager 8.1

Home / User Management

Manage Users

Search

+ New

| | First Name | Surname | Display Name | Login Name | SIP Handle |
|--------------------------|------------|---------|--------------|-----------------|------------|
| <input type="checkbox"/> | SIP | 78000 | 78000, SIP | 78000@avaya.com | 78000 |
| <input type="checkbox"/> | SIP | 78001 | 78001, SIP | 78001@avaya.com | 78001 |
| <input type="checkbox"/> | SIP | 78002 | 78002, SIP | 78002@avaya.com | 78002 |

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.

Avaya Aura System Manager 8.1

Home / User Management

User Profile | Add

Commit & Continue Commit Cancel

Identity Communication Profile Membership Contacts

Basic Info

Address

LocalizedName

User Provisioning Rule: Rule

* Last Name: 78010 Last Name (in Latin alphabet characters): 78010

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

* Login Name: 78010@avaya.com Middle Name: Middle Name Of User

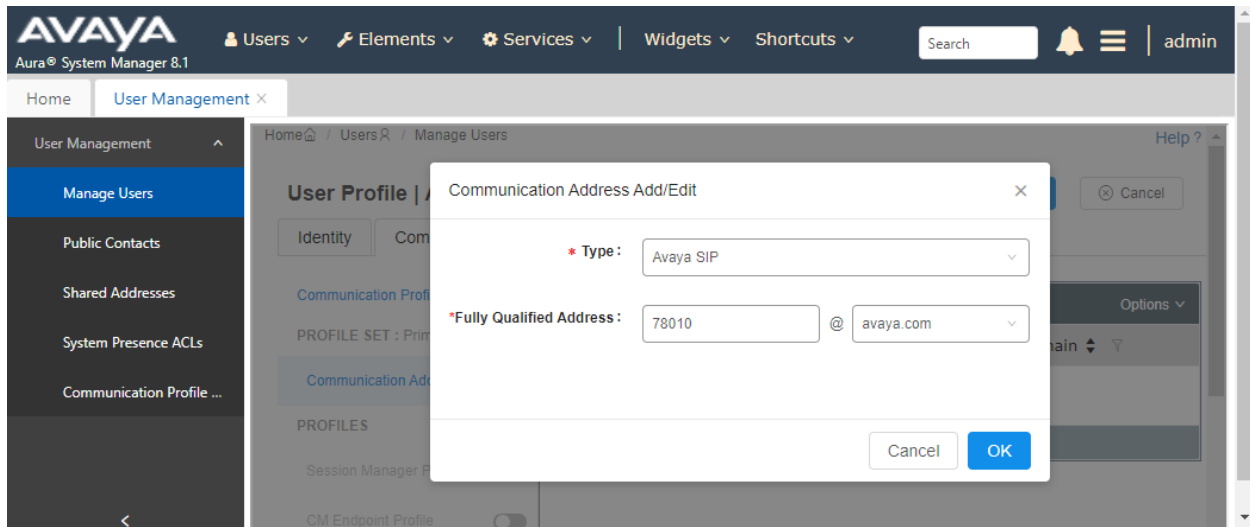
6.3.2. Communication Profile

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**.

The screenshot displays the Avaya Aura System Manager 8.1 web interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 8.1', and tabs for 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. A search bar and a user profile 'admin' are also visible. The left sidebar shows a 'User Management' menu with options like 'Manage Users', 'Public Contacts', 'Shared Addresses', 'System Presence ACLs', and 'Communication Profile ...'. The main content area is titled 'User Profile | Add' and has tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' tab is active, and the 'Communication Profile Password' sub-tab is selected. A modal dialog box titled 'Comm-Profile Password' is open in the foreground. It contains two password input fields: 'Comm-Profile Password' and 'Re-enter Comm-Profile Password'. The second field has a red asterisk and a green checkmark, indicating a match. Below the fields is a link 'Generate Comm-Profile Password'. At the bottom of the dialog are 'Cancel' and 'OK' buttons. The background interface shows a list of profiles and a 'Commit & Continue' button.

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 shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 8.1', and tabs for Users, Elements, Services, Widgets, and Shortcuts. A search bar and user profile (admin) are on the right. The left sidebar shows 'User Management' with a sub-menu 'Manage Users'. The main content area is titled 'User Profile | Add' and has 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 'Session Manager Profile' and 'CM Endpoint Profile' both toggled on. The 'SIP Registration' section includes fields for 'Primary Session Manager' (devcon-sm), 'Secondary Session Manager' (Start typing...), 'Survivability Server' (Start typing...), 'Max. Simultaneous Devices' (1), and a checkbox for 'Block New Registration When Maximum Registrations Active?'. The 'Application Sequences' section has dropdowns for 'Origination Sequence' and 'Termination Sequence', both set to 'DEVCON-CM App S...'. Action buttons 'Commit & Continue', 'Commit', and 'Cancel' are at the top right.

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

The screenshot shows the 'Call Routing Settings' section of the Avaya Aura System Manager 8.1 interface. It includes a field for 'Home Location' set to 'Thornton' and a dropdown for 'Conference Factory Set' set to 'Select'. The section is titled 'Call Routing Settings'.

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, version information, and various menu items like Users, Elements, Services, Widgets, and Shortcuts. A search bar and user profile (admin) are also present. The left sidebar shows the 'User Management' section with options like Manage Users, Public Contacts, Shared Addresses, System Presence ACLs, and Communication Profile ... 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 a dropdown for 'PROFILE SET : Primary' and a 'Communication Address' section. Below these are 'PROFILES' with toggle switches for 'Session Manager Profile' and 'CM Endpoint Profile' (which is currently turned on). The main configuration area contains several 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), 'Calculate Route Pattern' (checked), 'SIP URI' (Select), 'Enhanced Callr-Info Display for 1-line phones' (unchecked), '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' (unchecked). Action buttons at the top right include 'Commit & Continue', 'Commit', and 'Cancel'.

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 ▼ MWI Served User Type None ▼ Per Station CPN - Send Calling Number None ▼ IP Phone Group ID <input type="text"/> Remote Soft Phone Emergency Calls as-on-local ▼ LWC Reception spe ▼ AUDIX Name None ▼ EC500 State enabled ▼ Short/Prefixed Registration Allowed default ▼ Music Source <input type="text"/> | Auto Answer none ▼ Coverage After Forwarding system ▼ Display Language english ▼ Hunt-to Station <input type="text"/> Loss Group 19 Survivable COR internal ▼ Time of Day Lock Table None ▼ Voice Mail Number <input type="text"/> Bridging Tone for This Extension None ▼ | | | | | | | | | | | | | | | | | | | | | | | | | |
| Features <table border="1"> <tbody> <tr> <td><input type="checkbox"/> Always Use</td> <td><input type="checkbox"/> Idle Appearance Preference</td> </tr> <tr> <td><input type="checkbox"/> IP Audio Hairpinning</td> <td><input type="checkbox"/> IP SoftPhone</td> </tr> <tr> <td><input type="checkbox"/> Bridged Call Alerting</td> <td><input checked="" type="checkbox"/> LWC Activation</td> </tr> <tr> <td><input type="checkbox"/> Bridged Idle Line Preference</td> <td><input type="checkbox"/> CDR Privacy</td> </tr> <tr> <td><input checked="" type="checkbox"/> Coverage Message Retrieval</td> <td><input checked="" type="checkbox"/> Direct IP-IP Audio Connections</td> </tr> <tr> <td><input type="checkbox"/> Data Restriction</td> <td><input type="checkbox"/> H.320 Conversion</td> </tr> <tr> <td><input checked="" type="checkbox"/> Survivable Trunk Dest</td> <td><input checked="" type="checkbox"/> IP Video</td> </tr> <tr> <td><input type="checkbox"/> Bridged Appearance Origination Restriction</td> <td><input type="checkbox"/> Per Button Ring Control</td> </tr> <tr> <td><input checked="" type="checkbox"/> Restrict Last Appearance</td> <td></td> </tr> <tr> <td><input type="checkbox"/> Turn on mute for remote off-hook attempt</td> <td></td> </tr> <tr> <td><input type="checkbox"/> IP Hoteling</td> <td></td> </tr> </tbody> </table> | | | | | <input type="checkbox"/> Always Use | <input type="checkbox"/> Idle Appearance Preference | <input type="checkbox"/> IP Audio Hairpinning | <input type="checkbox"/> IP SoftPhone | <input type="checkbox"/> Bridged Call Alerting | <input checked="" type="checkbox"/> LWC Activation | <input type="checkbox"/> Bridged Idle Line Preference | <input type="checkbox"/> CDR Privacy | <input checked="" type="checkbox"/> Coverage Message Retrieval | <input checked="" type="checkbox"/> Direct IP-IP Audio Connections | <input type="checkbox"/> Data Restriction | <input type="checkbox"/> H.320 Conversion | <input checked="" type="checkbox"/> Survivable Trunk Dest | <input checked="" type="checkbox"/> IP Video | <input type="checkbox"/> Bridged Appearance Origination Restriction | <input type="checkbox"/> Per Button Ring Control | <input checked="" type="checkbox"/> Restrict Last Appearance | | <input type="checkbox"/> Turn on mute for remote off-hook attempt | | <input type="checkbox"/> IP Hoteling | |
| <input type="checkbox"/> Always Use | <input type="checkbox"/> Idle Appearance Preference | | | | | | | | | | | | | | | | | | | | | | | | | |
| <input type="checkbox"/> IP Audio Hairpinning | <input type="checkbox"/> IP SoftPhone | | | | | | | | | | | | | | | | | | | | | | | | | |
| <input type="checkbox"/> Bridged Call Alerting | <input checked="" type="checkbox"/> LWC Activation | | | | | | | | | | | | | | | | | | | | | | | | | |
| <input type="checkbox"/> Bridged Idle Line Preference | <input type="checkbox"/> CDR Privacy | | | | | | | | | | | | | | | | | | | | | | | | | |
| <input checked="" type="checkbox"/> Coverage Message Retrieval | <input checked="" type="checkbox"/> Direct IP-IP Audio Connections | | | | | | | | | | | | | | | | | | | | | | | | | |
| <input type="checkbox"/> Data Restriction | <input type="checkbox"/> H.320 Conversion | | | | | | | | | | | | | | | | | | | | | | | | | |
| <input checked="" type="checkbox"/> Survivable Trunk Dest | <input checked="" type="checkbox"/> IP Video | | | | | | | | | | | | | | | | | | | | | | | | | |
| <input type="checkbox"/> Bridged Appearance Origination Restriction | <input type="checkbox"/> Per Button Ring Control | | | | | | | | | | | | | | | | | | | | | | | | | |
| <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

ID:

Enter ID and password

Enter ID

Password:

Enter Password

Login

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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 header includes the AIPHONE logo, the text 'IX System Setting', and an 'Update' button. Below the header, the 'Category' is 'Video Stations' and the 'Station Type' is 'IX-DVM'. The left sidebar contains a navigation menu with 'Station Information' (expanded) and 'Network Settings'. Under 'Station Information', the sub-items are 'Identification', 'ID and Password', 'Language', 'Time', and 'Expanded System'. Under 'Network Settings', the sub-items are 'IP Address', 'DNS', 'SIP', 'Multicast Address', 'Video', 'Audio', 'Packet Priority', and 'NTP'. The main content area is titled 'Station Information' and features a red 'Required Settings' indicator. The 'Identification' section is active, showing three fields: 'Number' (with a red required indicator), 'Name', and 'Location'. The 'Number' field contains '78010' with a '3-5 digits' hint. The 'Name' field contains 'IX-DVM' with a '1-24 alphanumeric characters(*1)' hint. The 'Location' field is empty with a '1-24 alphanumeric characters(*1)' hint. A footnote at the bottom states: '(*1) Certain characters may not be displayed correctly on IX-MV and IX-MV7-* due to font type.'

AIPHONE IX System Setting Update

Category: Video Stations Station Type: IX-DVM

Station Information

Identification
ID and Password
Language
Time
Expanded System

Network Settings
IP Address
DNS
SIP
Multicast Address
Video
Audio
Packet Priority
NTP

Station Information

Required Settings

Identification

Number • 78010 3-5 digits
Name IX-DVM 1-24 alphanumeric characters(*1)
Location 1-24 alphanumeric characters(*1)

(*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 displays the 'AIPHONE IX System Setting' web interface. The top navigation bar includes 'Category: Video Stations' and 'Station Type: IX-DVM'. A left sidebar contains a tree view with categories: 'Station Information' (sub-items: Identification, ID and Password, Language, Time, Expanded System), 'Network Settings' (sub-items: IP Address, DNS, SIP, Multicast Address, Video, Audio, Packet Priority, NTP), 'System Information' (sub-item: Custom Sound Registry), and 'Call Settings'. The main content area is titled 'Network Settings' and features a 'SIP' section. This section is divided into two sub-sections: 'SIP Connections' and 'SIP Server'. In 'SIP Connections', 'SIP Signaling Port' is set to 5060 (with a red '1-65535' constraint) and 'User Agent' is set to IX-DVM (with a red '1-36 alphanumeric characters' constraint). In 'SIP Server', 'SIP Compatibility Mode' is set to 'Standard Mode'. Under 'Primary Server', 'ID' is 78010 (1-24 alphanumeric), 'Password' is masked (1-24 alphanumeric), 'IPv4 Address' is 10.64.102.117 (1.0.0.1-223.255.255.254 or host), 'IPv6 Address' is empty (::FF:0-FEFF:FFFF:FFFF:FFFF:), and 'Port' is 5060 (1-65535). An 'Update' button is located in the top right corner.

| Field | 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-FEFF: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 left sidebar contains a navigation menu with the following sections: 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), and Function Settings. The main content area is titled 'Network Settings' and is divided into three sections: Miscellaneous, Multicast Address, and Video. The Miscellaneous section includes fields for Register Transmission Interval [sec] (3600), DTMF digit interval timeout [sec] (5), and Call health check timer (80-3600 sec). The Multicast Address section includes fields for IPv4 (224.0.0.0-239.255.255.255) and IPv6 (FF10::0-FFFF:FFFF:FFFF:FFFF:FFFF:FFFF). The Video section includes a warning about the RTP End Port and fields for Resolution (640x480(VGA)), Wide View (Disable), Frame Rate [fps] (30), Select Profile (High), I-picture interval (30), Bit rate [kbps] (1024), RTP Start Port (30000), and RTP End Port (31000).

AIPHONE IX System Setting
Category: Video Stations Station Type: IX-DVM

Network Settings

Miscellaneous

Register Transmission Interval [sec] 3600 10-14400
DTMF digit interval timeout [sec] 5 1-10
Call health check timer 80-3600 sec 90 sec Do not transmit re-INVITE, 80-3600 sec

•Multicast Address

For Call IPv4 224.0.0.0-239.255.255.255
IPv6 FF10::0-FFFF:FFFF:FFFF:FFFF:FFFF:FFFF

•Video

SIP Channel

The "SIP Channel" RTP End Port should be greater than 90 digits from the RTP Start Port.

Resolution ☐ 320x240(QVGA) ☒ 640x480(VGA)
Wide View ☐ Enable ☒ Disable
Frame Rate [fps] 30
Select Profile High
I-picture interval 30 1-100
Bit rate [kbps] 1024
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 header shows 'AIPHONE IX System Setting' with a category of 'Video Stations' and a station type of 'IX-DVM'. An 'Update' button is in the top right. The left sidebar contains a navigation menu with sections: 'Station Information' (Identification, ID and Password, Language, Time, Expanded System), 'Network Settings' (IP Address, DNS, SIP, Multicast Address, Video, Audio, Packet Priority, RTP), 'System Information' (Custom Sound Registry), 'Call Settings' (Station Settings, Called Stations (for Door), Call Origination, Incoming Call, Contactless Call), and 'Option Input / Relay' (Output Settings, Option Input, Relay Output). The main content area is titled 'Network Settings' and features a blue header. Below this, the 'Audio' section is highlighted. It includes several red warning messages: '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.', and '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'. The configuration fields are as follows: 'Audio Codec' with radio buttons for G.711(μ-law) (selected), G.711(A-law), and G.722; 'Audio RTP Transmission Interval [msec]' set to 20; 'RTP Idle Detection Time [sec]' set to 10; 'SIP Channel' with RTP Start Port 20000 and RTP End Port 21000; 'ONVIF Transmit Channel' with RTP Start Port 22000 and RTP End Port 23000; and 'Audio Buffer' with 'Packets Buffered at Audio Start' set to 1 and 'Maximum Packets Buffered' set to 3. A red note states 'Maximum Packet Buffer must be larger than Audio Start Buffer.'

| Section | Parameter | Value | Notes |
|------------------------|--|--|---|
| Audio | Audio Codec | <input checked="" type="radio"/> G.711(μ-law) <input type="radio"/> G.711(A-law) <input type="radio"/> G.722 | |
| | Audio RTP Transmission Interval [msec] | 20 | |
| | RTP Idle Detection Time [sec] | 10 | |
| 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.

AIPHONE IX System Setting
Category: Video Stations Station Type: IX-DVM Update

Call Settings

• **Station Information**

Call Button Function: Call, Answer Call, End Communication
"Cancel Call, End Communication" disabled when using Option Input call.

• **Called Stations (for Door)**

Option Input #: Group 01

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 displays the 'User Registrations' page in the Avaya Aura System Manager 8.1 interface. The page title is 'User Registrations' and it includes a sub-header: 'Select rows to send notifications to devices. Click on Details column for complete registration status.' The interface features a navigation sidebar on the left with options like 'Communication Prof...', 'Network Configur...', 'Device and Locati...', 'Application Confi...', 'System Status', 'SIP Entity Monit...', 'Managed Band...', 'Security Module...', 'SIP Firewall Status', 'Registration Su...', and 'User Registratio...'. The main content area shows a table of 22 items. 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: Prim, Sec, Surv, Visiting). The last row, representing an Aiphone device, is highlighted with a red border. The table data is as follows:

| | 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"/> |
| <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"/> |
| <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"/> |

At the bottom of the table, there is a 'Select : All, None' option and a pagination control showing 'Page 1 of 2'.

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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