Avaya Solution & Interoperability Test Lab

Application Notes for Aiphone IX Series PC Video Master Stations (IX-SOFT) with Avaya Aura® Communication Manager and Avaya Aura® Session Manager – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Aiphone IX Series PC Video Master Station (IX-SOFT) 1.6 which was compliance tested with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1.

The overall objective of the interoperability compliance testing was to verify Aiphone IX Series PC Video Master Station (IX-SOFT) functionality in an environment comprised of Avaya Aura® and various Avaya endpoints. Aiphone IX Series PC Video Master Station is a SIP based PC softphone.

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

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

1. Introduction

These Application Notes describe the configuration steps required for Aiphone IX-SOFT PC Video Master Station to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

The Aiphone IX Series PC Master Station (IX-SOFT) is part of Aiphone IX Series Master Stations. The Aiphone IX Series PC Master Station (IX-SOFT) acts as a SIP phone when connected to Avaya Aura®. It supports the PC's built-in camera allowing for two-way video. Additionally, the Aiphone IX Series PC Master Station (IX-SOFT) has intercom features that include paging, line supervision, device check, picture in picture when using 3rd party ONVIF Profile S cameras, and an intuitive map (not tested). The map can be used to answer and place calls, release doors, and pull up cameras from devices in the system.

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

2. General Test Approach and Test Results

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

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

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

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

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

2.1. Interoperability Compliance Testing

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

- SIP Registration
- Calls to Avaya SIP Audio & Video endpoints
- Calls to Avaya H.323 Audio endpoints
- Calls to Avaya Digital & Analog endpoints
- Attended transfers
- Calls to PSTN via ISDN-PRI Trunk
- Serviceability testing focusing on recovery from Ethernet disconnect/reconnect

2.2. Test Results

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

2.3. Support

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

Japan

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

USA, Canada

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

France

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

Australia, New Zealand

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

Singapore

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

United Kingdom

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

3. Reference Configuration

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

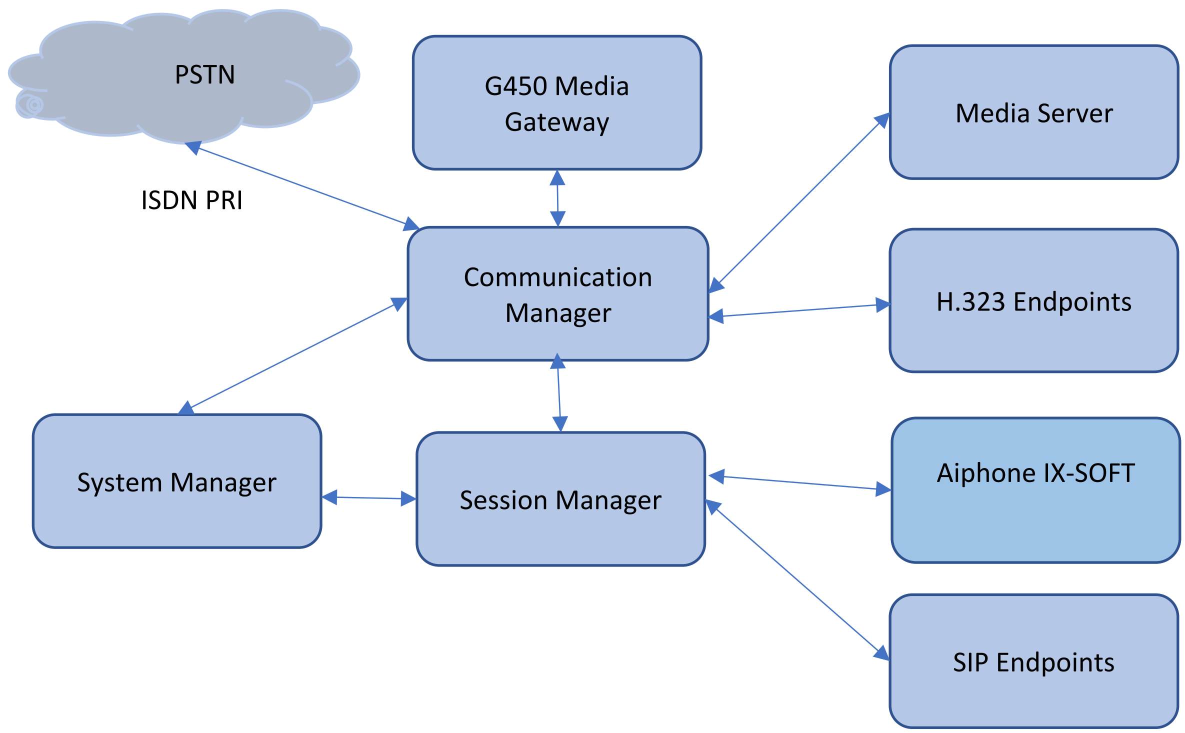


Figure 1: Test Configuration of Aiphone IX-SOFT with Avaya Aura®

4. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya Aura® Communication Manager	8.1.1.0.0.890.25763 (FP1)
Avaya Aura® Session Manager	8.1.1.0.811021
Avaya Aura® System Manager	8.1.1.0.0310782 (FP1)
Avaya 9600 Series H.323 IP Deskphones	6.8304
Avaya J179 SIP Phone	4.0.7.0.70
Avaya IX Workspace	3.14.0.53.10
Avaya H175 Collaboration Station	1.0.2.3
Avaya Vantage K175 Phone	2.0.1
Aiphone IX Series PC Video Master Station IX-SOFT	1.6

5. Configure Avaya Aura® Communication Manager

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

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

These steps were performed using an SSH Terminal session.

5.1. Verify System Capacity (License)

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

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

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

                                                              USED
                            Platform Maximum Ports:  48000     73
                                  Maximum Stations:  36000     48
                          Maximum XMOBILE Stations:  36000      0
                Maximum Off-PBX Telephones - EC500:  41000      0
                Maximum Off-PBX Telephones -   OPS:  41000     30
                Maximum Off-PBX Telephones - PBFMC:  41000      0
                Maximum Off-PBX Telephones - PVFMC:  41000      0
                Maximum Off-PBX Telephones - SCCAN:      0      0
                     Maximum Survivable Processors:  313        0

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

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

```
display system-parameters customer-options                       Page   2 of 12
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                 Maximum Administered H.323 Trunks: 12000      0
       Maximum Concurrently Registered IP Stations:  2400      2
         Maximum Administered Remote Office Trunks: 12000      0
Max Concurrently Registered Remote Office Stations:  2400      0
          Maximum Concurrently Registered IP eCons:  128       0
     Max Concur Reg Unauthenticated H.323 Stations:   100      0
                    Maximum Video Capable Stations:  36000     2
               Maximum Video Capable IP Softphones:   2400    19
                   Maximum Administered SIP Trunks: 12000     10
  Max Administered Ad-hoc Video Conferencing Ports: 12000      0
   Max Number of DS1 Boards with Echo Cancellation: 688    0
```

5.2. Define the Dial Plan

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

```
change dialplan analysis                                         Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all            Percent Full: 1

    Dialed   Total  Call      Dialed   Total  Call      Dialed   Total  Call
    String   Length Type      String   Length Type      String   Length Type
 1             3  dac
 2             5  ext
 3             5  ext
 4             5  aar
 7             5  ext
 8             1  fac
 9             1  fac
 *             3  fac
 #             3  fac
```

5.3. Enable IP Video

Use the **change signaling-group** command to enable IP video in the system. This signaling group is used for calls routed over the SIP trunk between Connection Manager and Session Manager.

```
change signaling-group 1                                        Page   1 of   3
                                SIGNALING GROUP

 Group Number: 1             Group Type: sip
  IMS Enabled? n       Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? n        Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: SM                       Clustered? n
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                 Far-end Node Name: sm81
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1

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

5.4. Configure IP Codec Set

Use the **change ip-codec-set** command to set audio codec types in the **Audio Codec** fields as necessary. As an example, the codec is configured as **G7.711MU**.

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

                          IP CODEC SET
    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU       n          2        20
 2:
Media Encryption                   Encrypted SRTCP: best-effort
 1:none
 2:
 3:
 4:
 5:
```

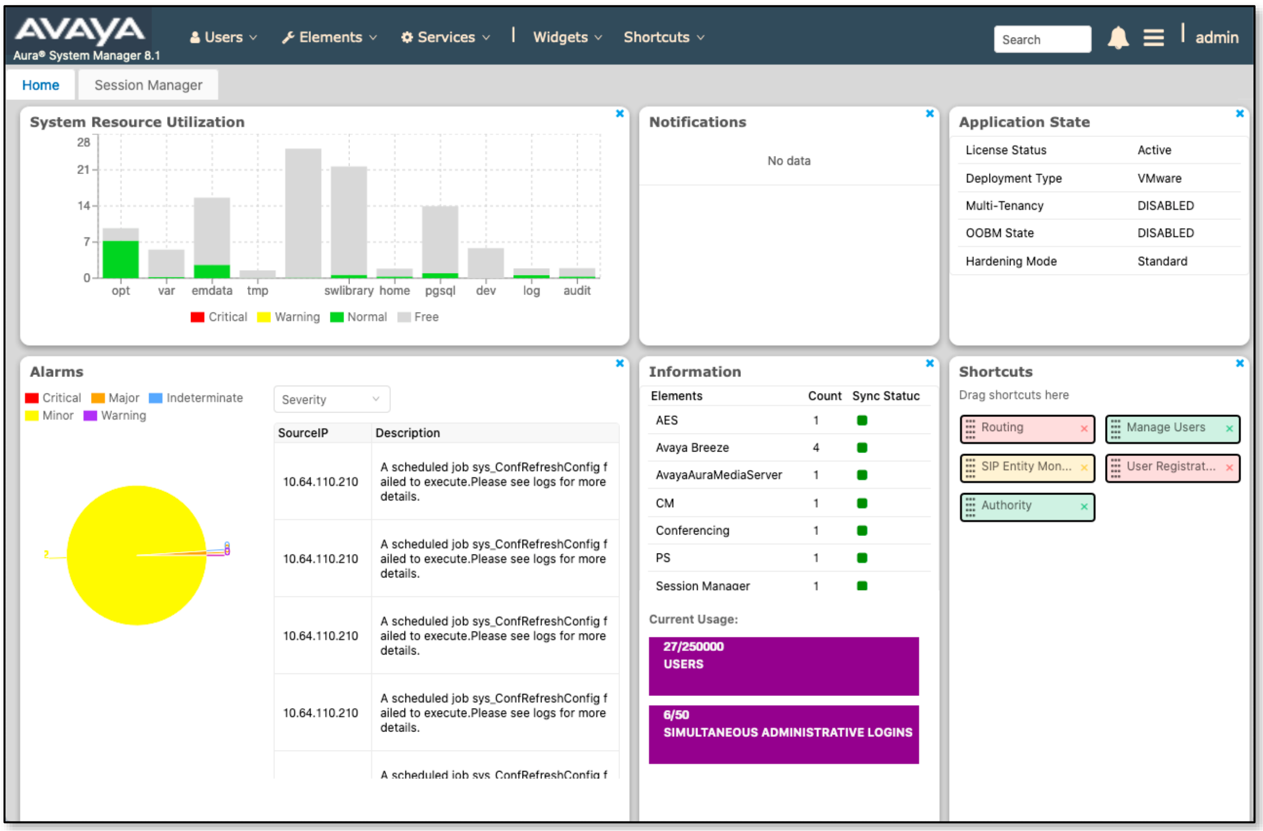
Configure Page 2 of the IP Codec Set, enable **Allow Direct-IP Multimedia?**

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

                          IP MEDIA PARAMETERS

                              Allow Direct-IP Multimedia? y
              Maximum Call Rate for Direct-IP Multimedia: 10240:Kbits
     Maximum Call Rate for Priority Direct-IP Multimedia: 10240:Kbits


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


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

6. Configure Avaya Aura® Session Manager

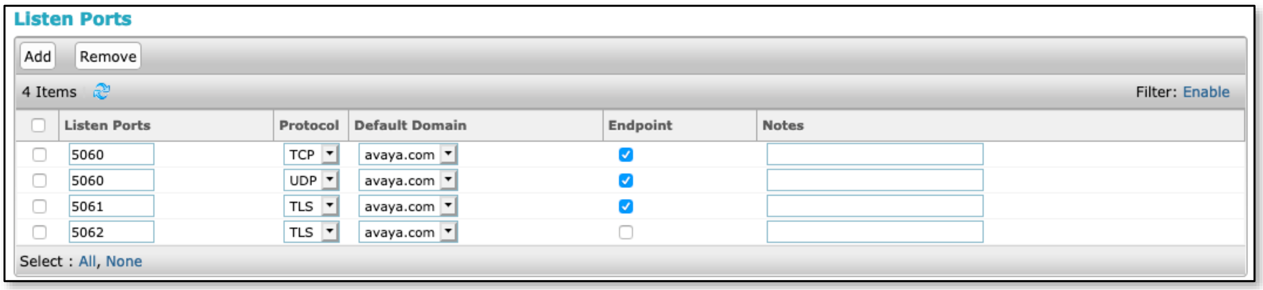
This section describes aspects of the Session Manager configuration required for interoperating with Aiphone IX-SOFT. It is assumed that the Domains, Locations, SIP entities, Entity Links, Routing Policies, Dial Patterns and Application Sequences have been configured where appropriate for Communication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

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



6.1. Verify Session Manager Listen Port for SIP Endpoint Registration

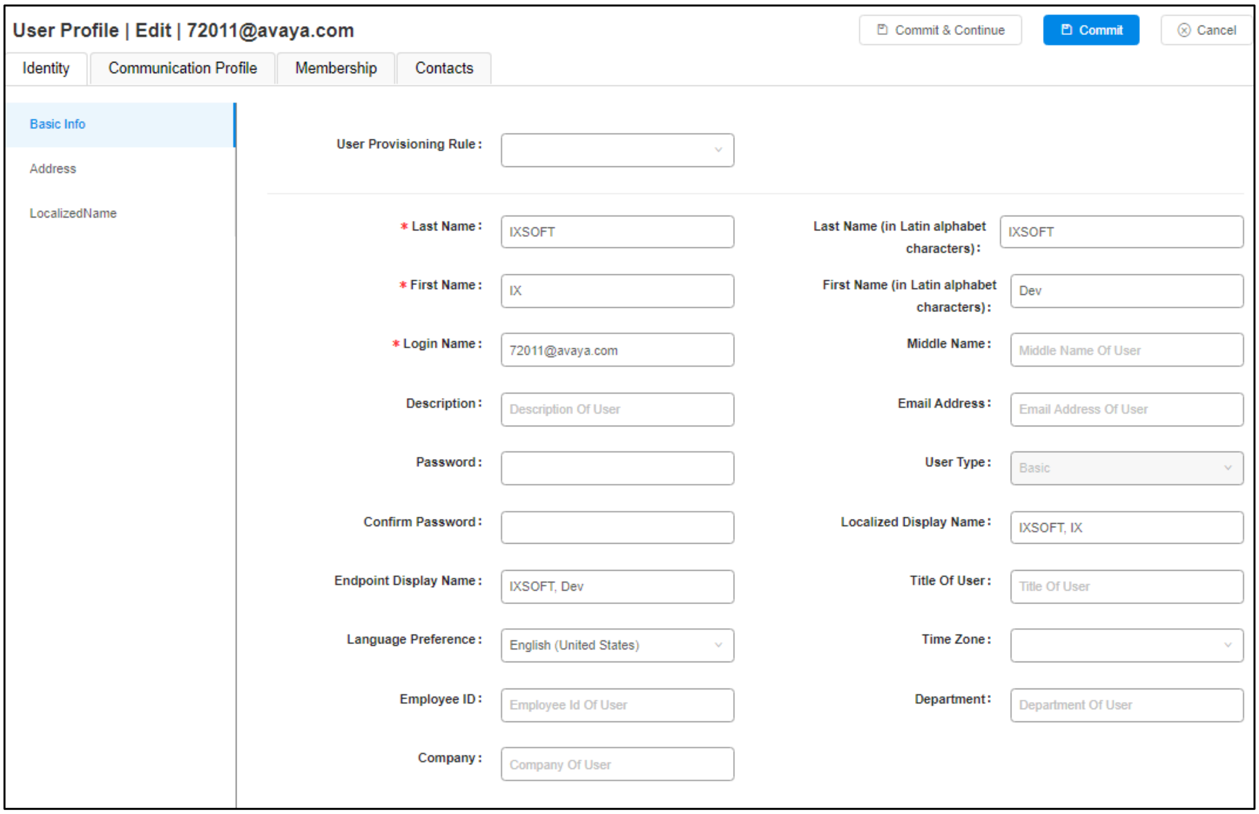
Each Session Manager Entity must be configured so that SIP endpoint can register to it using UDP, TCP, or TLS. From the web interface click **Routing → SIP Entities** (not shown) and select the Session Manager entity used for registration. In the compliance test, **TCP** and **UDP** listen ports were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



6.2. Add a SIP User

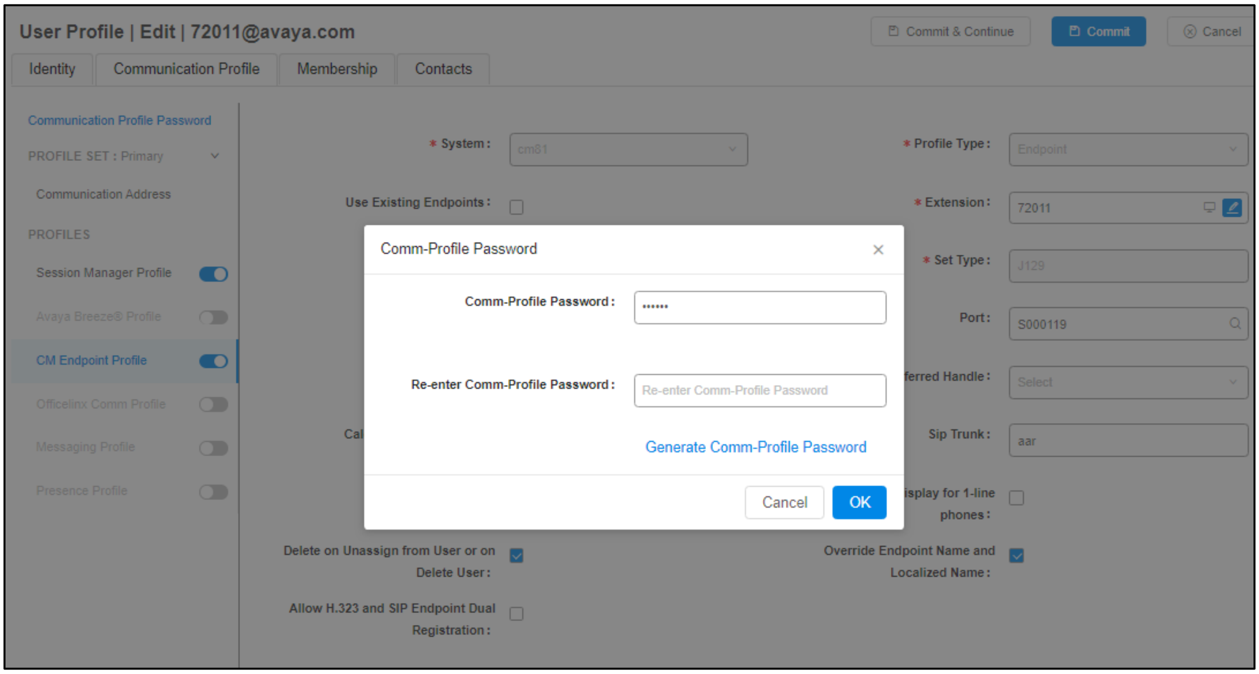
A SIP user must be added for Aiphone IX-SOFT. Click **User Management** → **Manage Users** → **New** (not shown) and configure the following in the **Identity** tab.

- **First Name** and **Last Name -** Enter an identifying name
- **Login Name** Enter the extension number followed by the domain, in this case **72011@avaya.com**

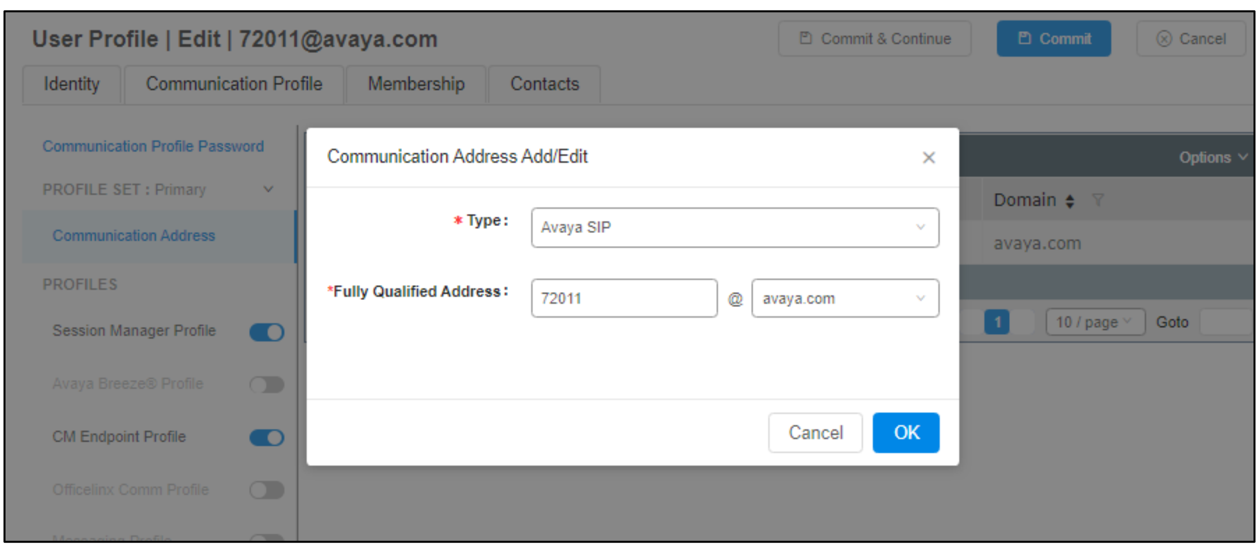


Note in this and subsequent steps, press **Commit & Continue** after making entries or selections.

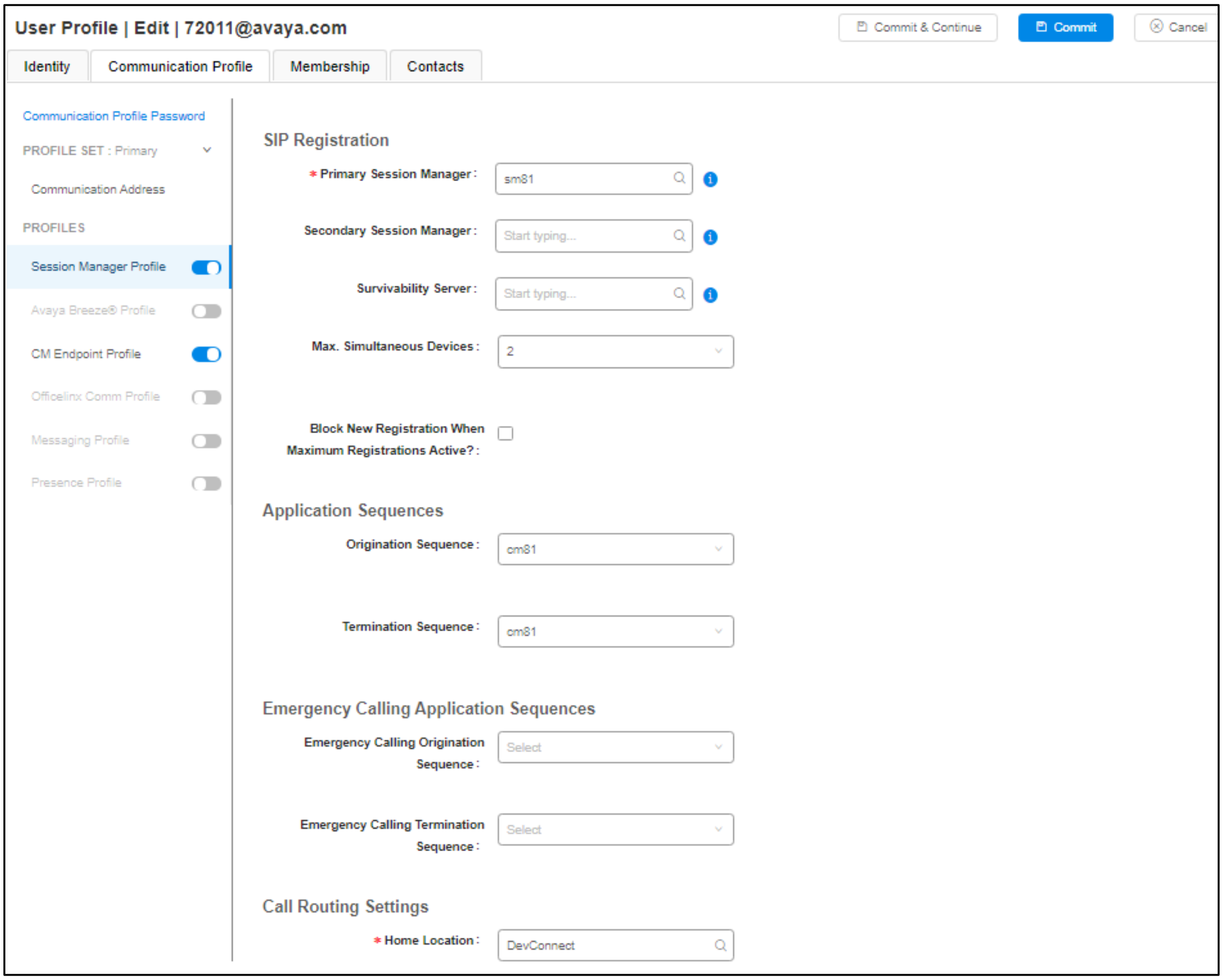
Click the **Communication Profile** tab and in the **Communication Profile Password** and **Confirm Password** fields, enter a numeric password. This will be used to register the device during login.



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



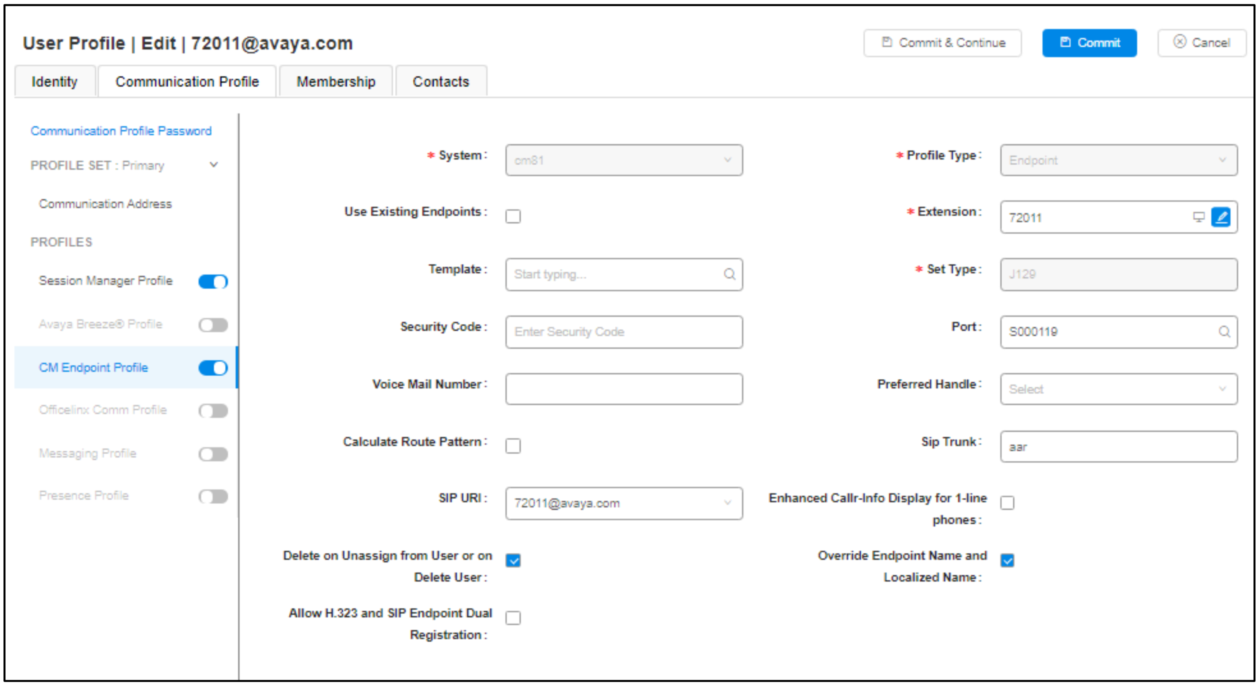
Click on the **Session Manager Profile link** and configure the **Primary Session Manager, Max Simultaneous Devices, Origination Application Sequence, Termination Application Sequence** and **Home Location,** from the respective drop-down lists.



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

- **System -** Select the relevant Communication Manager SIP Entity from the drop-down list
- **Profile Type -** Select **Endpoint** from the drop-down list
- **Extension -** Enter the required extension number, in this case **72011**
- **Port -** The "IP" is auto filled out by the system

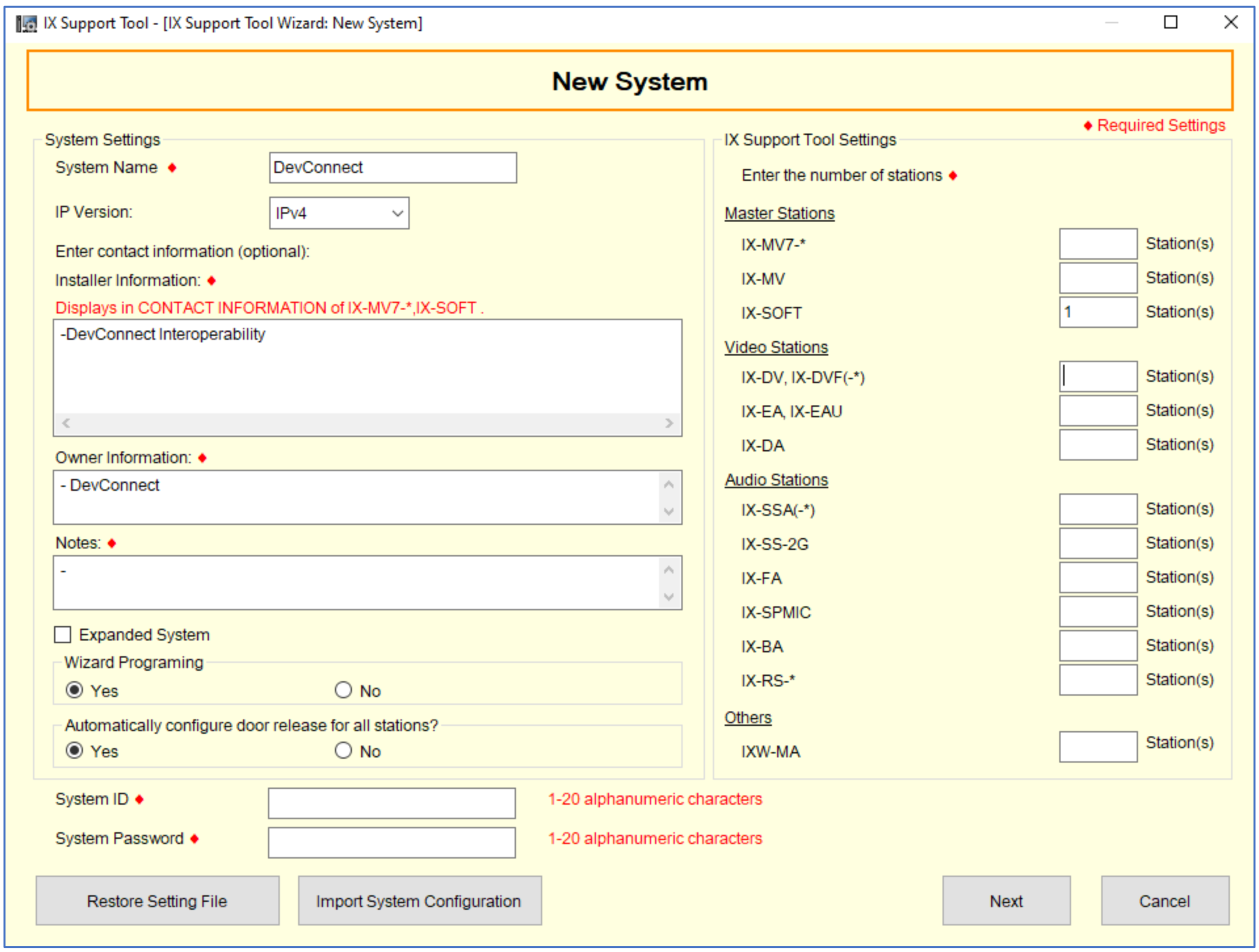
  Click on **Endpoint Editor** in the Extension field to edit Communication Manager settings if desired.



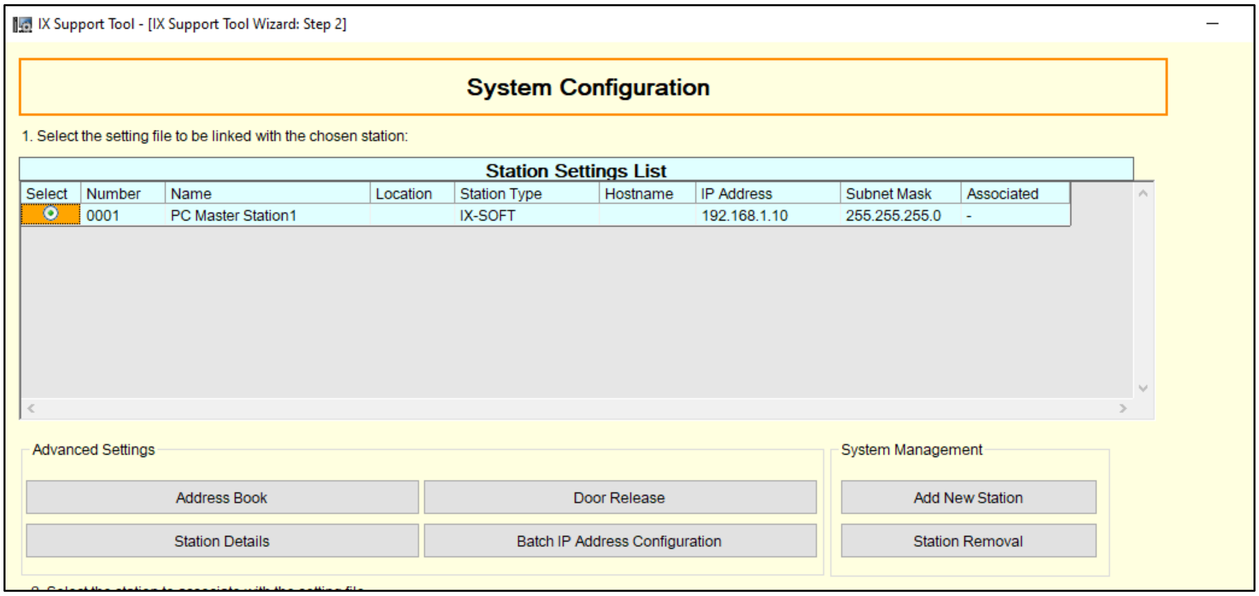
6.3. Configure Aiphone IX Series Video Master Station

This section provides steps to configure Aiphone IX-SOFT.

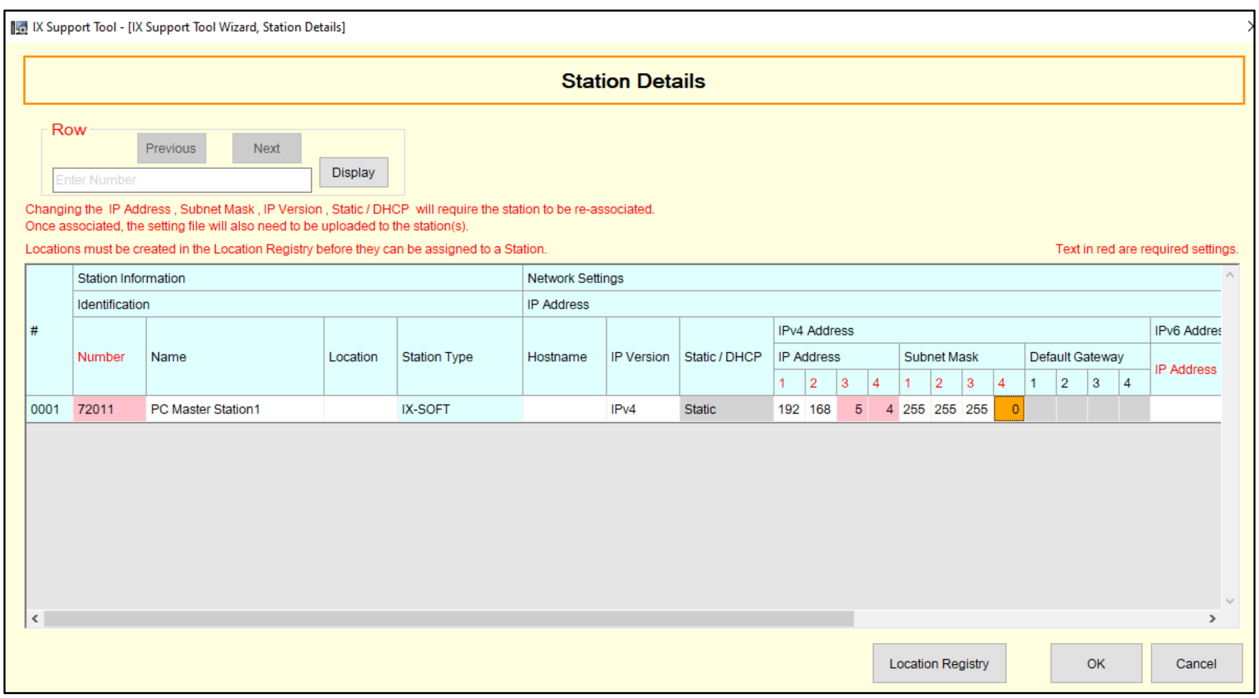
To configure Aiphone IX-SOFT, the Aiphone IX Support Tool must be used. Install and open. Log into the Support Tool using appropriate credentials. Once logged in, the new system dialog opens. Enter appropriate **System Name, Installer Information, Owner Information** and **Notes** as applicable. Input a number for the **IX-SOFT Master Stations** box. Enter **System ID** and **System Password**. Select **Next**.



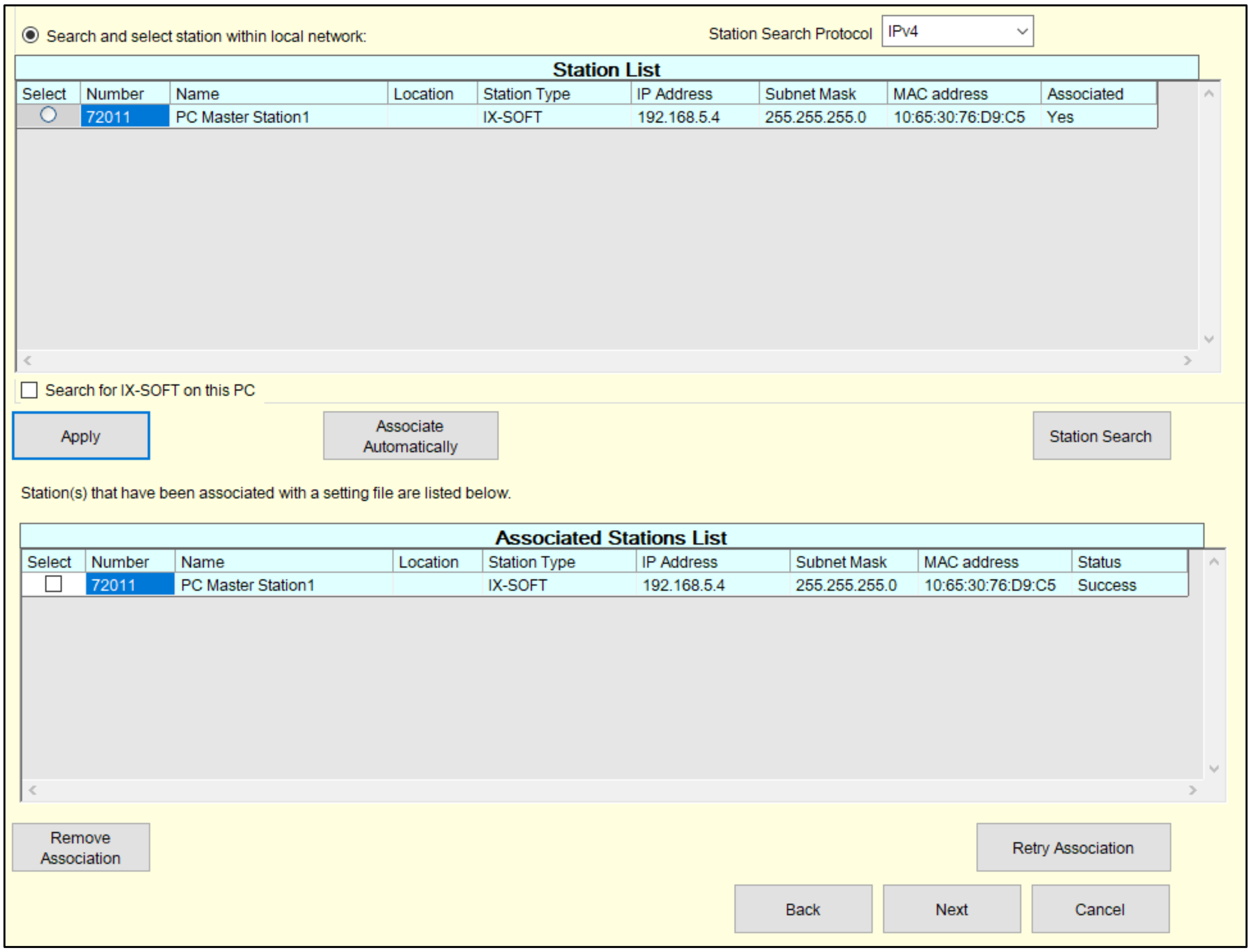
The **System Configuration** dialog opens. Select the IX-SOFT station found and click on **Station Details.**



Update the **Number**, and **IP Address**, and **Subnet Mask** fields. Click **OK**.

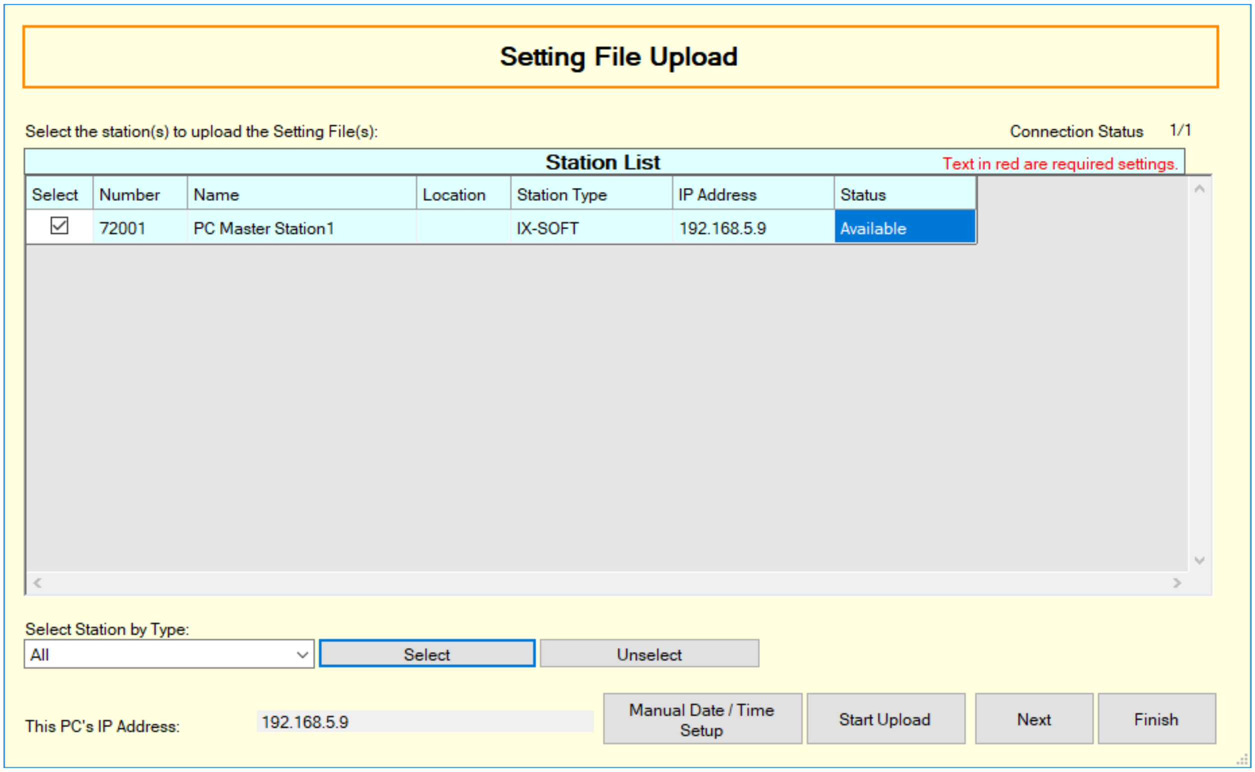


In the **System Configuration** dialog, select **Associate** to link the found softphone to the configuration. The **Status** column in the **Associated Stations** list should show 'Success'.



Click **Next**.

The Setting File Upload dialog opens. Select the PC master station in the Station list. Select the checkbox for the station and click the **Start Upload** button.

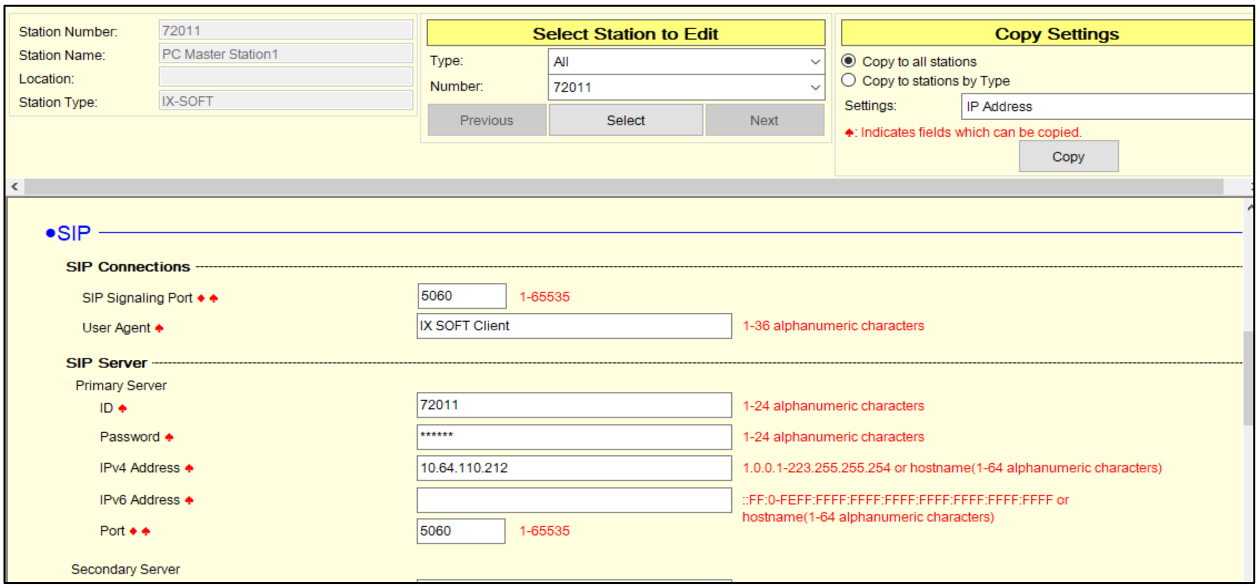


The station status should show successful upon upload completion.

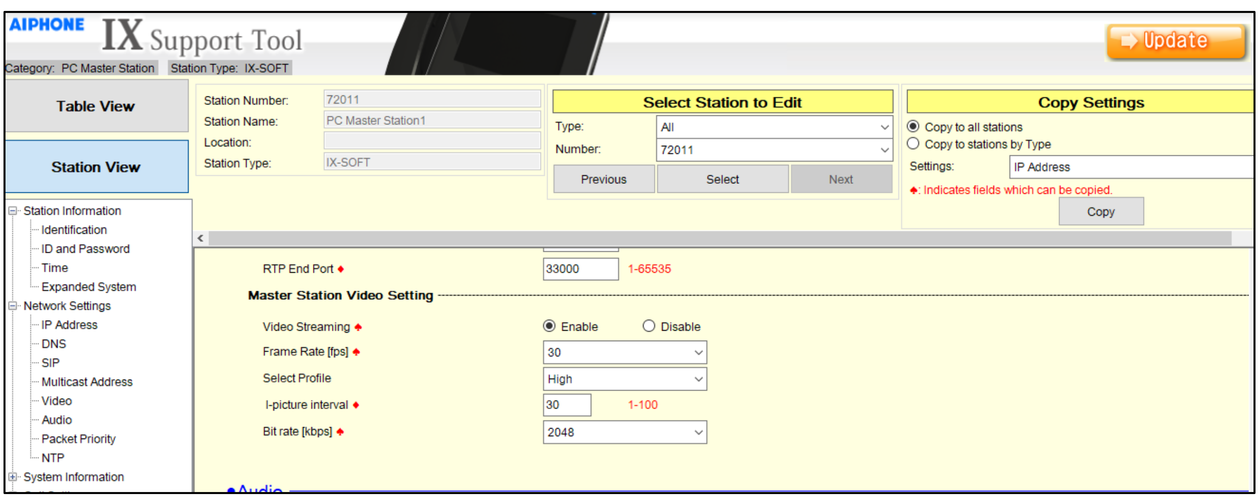
In the Station View, under Station Information, select an ID for the **user agent** (not shown). Under SIP, configure with the following values.

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

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

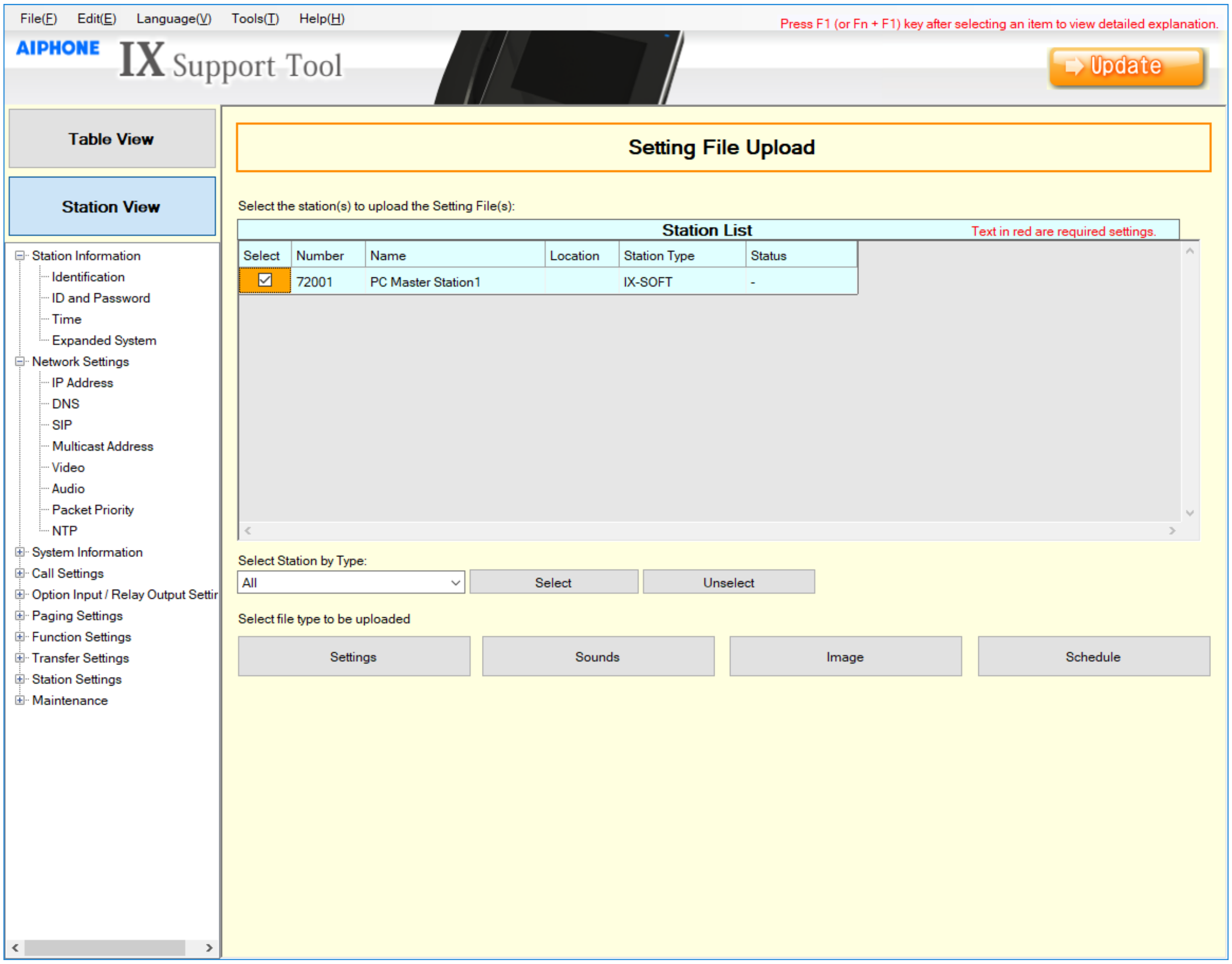


Continuing from above, scroll down to the **Master Station Video Setting** sub section and verify the Video settings are as shown below.



Click the update button.

Upload the configuration changes to the station. Select **File →Upload Settings to Station**. Select the PC Master Station 1 and click the **Settings** button to upload the configuration.

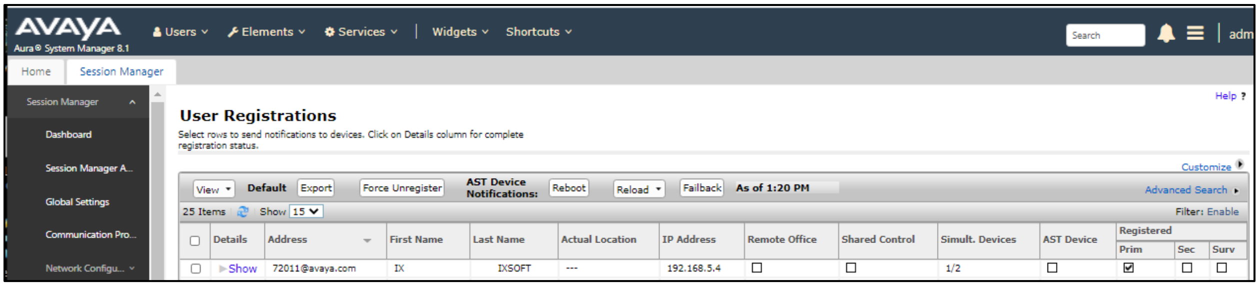


The Station List status should show successful upon completion.

7. Verification Steps

The following steps may be used to verify the configuration:

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



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

```
status trunk 1

                              TRUNK GROUP STATUS

Member     Port     Service State       Mtce Connected Ports
                                        Busy

0001/0001 T000001 in-service/active   no   T000002
0001/0002 T000002 in-service/active   no   T000001
0001/0003 T000003 in-service/idle     no
0001/0004 T000004 in-service/idle     no
0001/0005 T000005 in-service/idle     no
0001/0006 T000006 in-service/idle     no
0001/0007 T000007 in-service/idle     no
```

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

```
status trunk 1/0001                                         Page   3 of   3
                      SRC PORT TO DEST PORT TALKPATH
src port: T000001
T000001:TX:192.168.5.4:20002/g711u/20ms
T000002:RX:192.168.5.3:5010/g711u/20ms
```

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

```
status trunk 1/0007                                            Page   2 of   3
                                CALL CONTROL SIGNALING

Near-end Signaling Loc: PROCR
  Signaling   IP Address                                        Port
   Near-end:  10.64.110.213                                   : 5061
    Far-end:  10.64.110.212                                   : 5061
 H.245 Near:
  H.245 Far:
   H.245 Signaling Loc:           H.245 Tunneled in Q.931? no

 Audio Connection Type: ip-direct     Authentication Type: None
    Near-end Audio Loc:                        Codec Type: G.711MU
   Audio      IP Address                                        Port
   Near-end:  192.168.5.3                                     : 5010
    Far-end:  192.168.5.4                                     : 20002

 Video Near:  192.168.5.3                                     : 5010
  Video Far:  192.168.5.4                                     : 30000
 Video Port:  T000001
  Video Near-end Codec:     H.264     Video Far-end Codec: H.264
```

8. Conclusion

Aiphone IX-SOFT was compliance tested with Avaya Aura®. Aiphone IX-SOFT functioned properly in feature and serviceability tests.

9. Additional References

The following Avaya product documentation can be found at http://support.avaya.com:

[1] *Administering Avaya Aura® Communication Manager*, Release 8.1.x, 10 December 2020.
[2] *Administering Avaya Aura® Session Manager*, Release 8.1.x , 12 October 2020.

Documentation related to Aiphone IX-SOFT can be found at:
Japan: https://www.aiphone.co.jp/products/business/ix/
USA, Canada: https://www.aiphone.com/home/products/ix-series
France: https://www.aiphone.fr/catalogue/interphonie-ip-protocole-sip-ix/
Australia, New Zealand: https://www.aiphone.com.au/product/ix/
Singapore: http://www.aiphone.com.sg/
United Kingdom: https://www.aiphone.co.uk/featured_item/ix2/

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