Avaya Solution & Interoperability Test Lab

Application Notes for SIP Softphone by QSC Q-Sys platform with Avaya Aura® Session Manager R8.1 and Avaya Aura® Communication Manager R8.1 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for QSC Q-Sys platform SIP Softphone to interoperate with Avaya Aura® Session Manager R8.1 and Avaya Aura® Communication Manager R8.1. The QSC Q-Sys platform SIP Softphone is a conferencing phone that can register with Avaya Aura® Session Manager as a SIP endpoint in support of voice communications and conferencing requirements.

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 QSC Q-Sys platform SIP Softphone to interoperate with Avaya Aura® Session Manager R8.1 and Avaya Aura® Communication Manager R8.1. The QSC Q-Sys platform SIP Softphone is a conferencing phone that can register with Avaya Aura® Session Manager as a SIP endpoint in support of voice communications and conferencing requirements.

The Q-Sys Core is an Intel-based embedded Linux PC digital audio processor used for commercial installations. The Q-Sys Softphone feature of Q-Sys Core is completely virtual, requiring no additional hardware to function. The typical application of the Q-Sys Softphone is to provide a telephony endpoint to both conference rooms and wide-area paging. The Q-Sys Core product line runs on a Linux-based platform, which handle audio Digital Signal Processing tasks and support very scalable input/output channel counts from small to large, either locally or across a layer 3 infrastructure. The Q-Sys Softphone is a highly configurable SIP-based telephone endpoint which, because of its component nature within the Q-Sys environment, is accessible from any Windows-based desktop, iOS device or using the TCP-based External Control Protocol.

2. General Test Approach and Test Results

The general test approach was to place calls to and from the Q-Sys Softphone and exercise basic telephone operations. The main objectives were to verify the following:

- Registration
- Codecs (G.711, G.722, and G.729)
- Inbound calls
- Outbound calls
- Hold/Resume
- Call Transfer and Conferencing (Blind and Attended)
- Call termination (origination/destination)
    - Avaya Features using Feature Access Code (FAC) Call Park/Unpark
    - Call Pickup
    - Call Forward (Unconditional, Busy/no answer)
    - Find Me
- Voicemail using Communication manager Messaging (CMM)
- Message Waiting Indicator (MWI)
- Serviceability

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 the QSC S-Sys Softphone utilized enabled capabilities of TLS and SRTP.

2.1. Interoperability Compliance Testing

The interoperability compliance test included features and serviceability. The focus of interoperability compliance testing was primarily on verifying call establishment on Q-Sys Softphone. The Q-Sys Softphone operations such as inbound calls, outbound calls, hold/resume, transfer, conference, Feature Access Codes, and its interactions with Session Manager, Communication Manager, and other Avaya SIP, and H.323 phones were verified. The serviceability testing introduced failure scenarios to see if Q-Sys Softphone can recover from failures.

2.2. Test Results

The test objectives were verified. For serviceability testing, Q-Sys Softphone operated properly after recovering from failures such as network disconnects, and resets of Q-Sys Softphone and Session Manager.

The following features are not supported by Q-Sys at this time:

- Call Hold/Resume
- Call Transfer
- Three party conference
- Call Park/Unpark
- MWI (Message Waiting Indicator)

SRTP support for inbound calls to the softphone worked, but outbound calls using SRTP failed with an SDP Fault error on Communication Manager due to incompatible cyphers contained in the offer. QSC engineers provided a patch resolving this which is anticipated to be included in a future release of the product.

2.3. Support

Technical support on QSC Q-Sys can be obtained through the following:

Application Engineering and Technical Services
Monday - Friday 7 AM to 5 PM PST (Excludes Holidays)
Tel. 800-772-2834 (U.S. only)
Tel. +1 (714) 754-6175

Q-SYS 24/7 Emergency Support*
Tel: +1-888-252-4836 (U.S./Canada)
Tel: +1-949-791-7722 (non-U.S.)

Q-SYS Support Email
qsyssupport@qsc.com
(Immediate email response times not guaranteed)

3. Reference Configuration

Once Q-Sys Softphone registers as a SIP endpoint with Session Manager, it can place and receive voice calls with various supported features as listed above in **Section 2.1**. The reference configuration used for the compliance test is shown in **Figure 1** below.

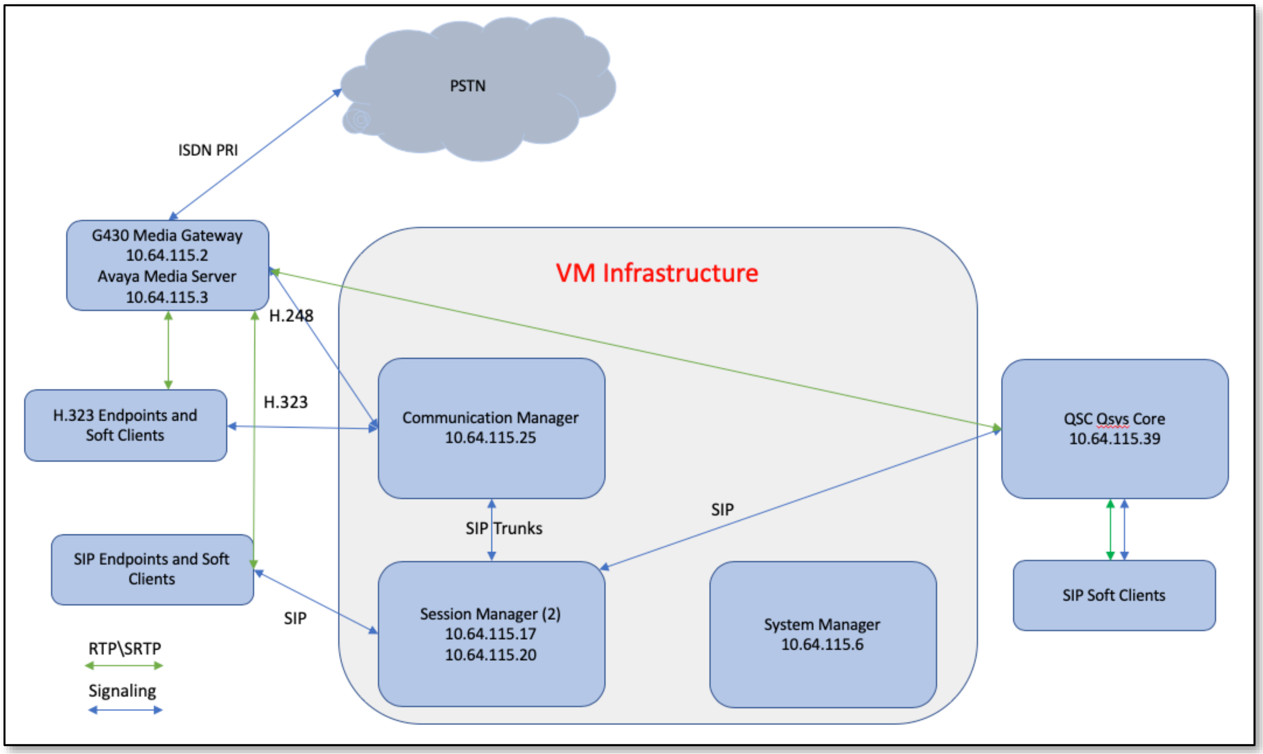


Figure 1: Q-Sys SIP Softphone with Avaya Aura® Session Manager and Avaya Aura® Communication Manager

4. Equipment and Software Validated

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

Equipment	Software
Avaya Aura® System Manager installed on VMWare	R8.1 (8.1.0.0.079880)
Avaya Aura® Session Manager installed on VMWare	R8.1 (8.1.0.0.810007)
Avaya Aura® Communication Manager installed on VMWare	R8.1 (vcm-018-01.0.890.0)
Avaya Aura® Media Server installed on VMWare	R8.0 (v.8.0.0.169)
Avaya Aura® Communication Manager Messaging installed on VMWare	R7.0 (vcmm-07.00.0.441.0)
Avaya 96x1 IP Deskphone (H323)	R6.6506
Avaya 96x0 IP Deskphone (H.323)	R3.280
Avaya JXX SIP Deskphone	R3.0.0.0.16
QSC Q-Sys Designer	8.1.1
QSC Q-Sys Core 110f	8.1.1
QSC Q-Sys Softphone	8.1.1

5. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding SIP Users.

Configuration is accomplished by accessing the browser-based GUI of System Manager using the URL "**https://<*ip-address*>/SMGR**", where <*ip-address*> is the IP address of System Manager. Log in with the appropriate credentials.

A typical existing environment of Avaya Aura® components was used, this section covers only the modifications required for testing the QSC solution.

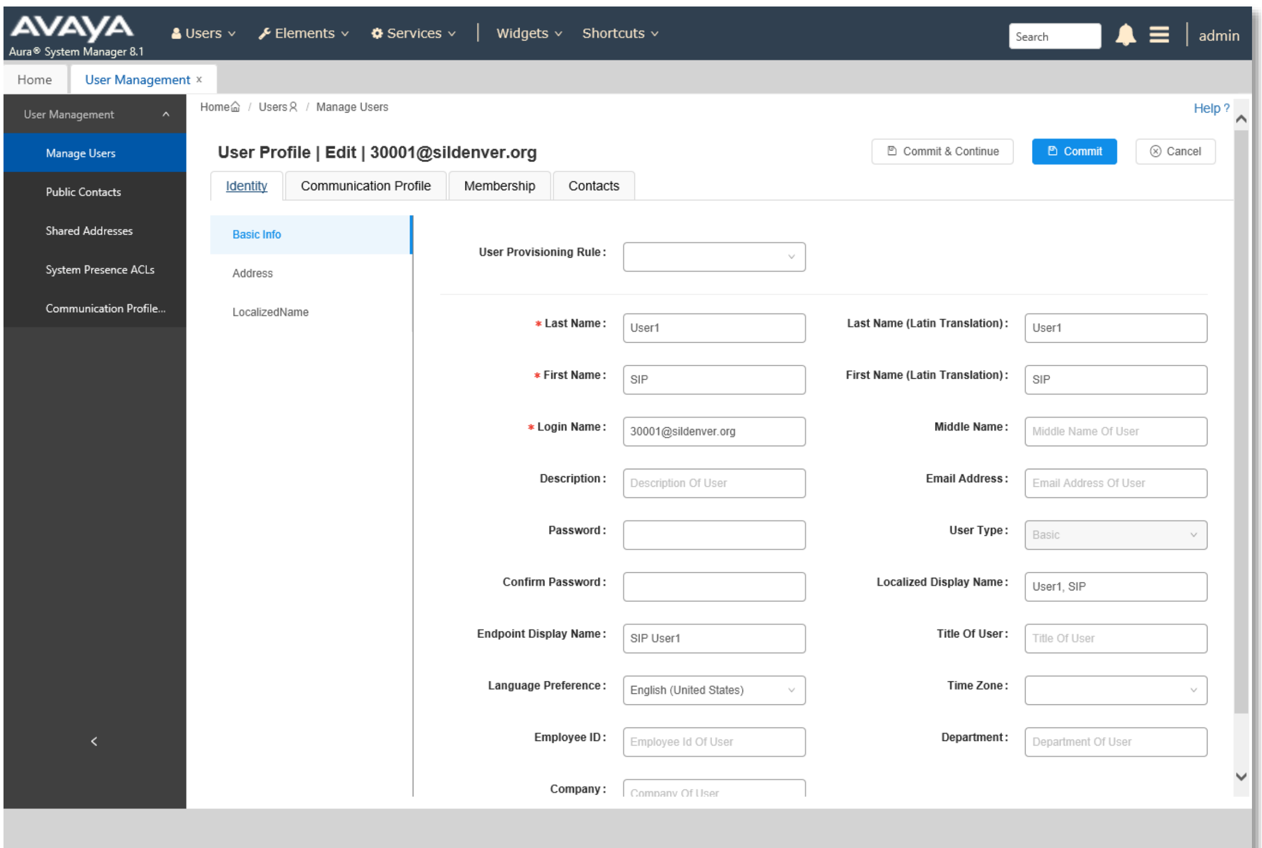
Note that the fields modified in this section are for this reference configuration only; defaults are used for all other fields.

5.1. Add SIP Users

Q-Sys Softphone was entered as a SIP user on Session Manager using the following steps. Navigate to **Home→Users→User Management→Manage Users** and configure as follows. This configuration is automatically synchronized with Communication Manger.

Enter values for the following required attributes for a SIP user in the **New User Profile** form:

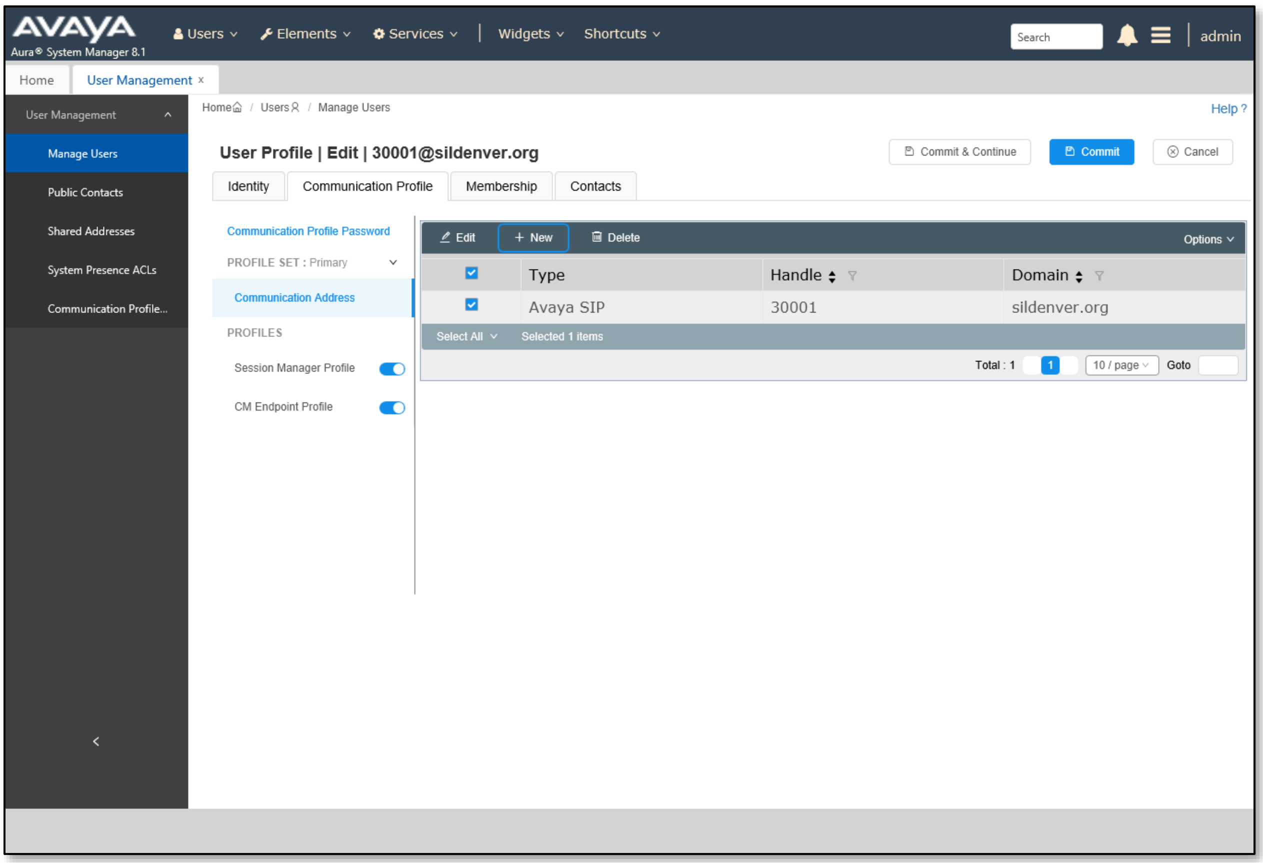
- **Last Name**: Enter the last name of the user
- **First Name**: Enter the first name of the user
- **Login Name**: Enter *<extension>@<sip domain>* of the user (e.g., ***30001@sildenver.org***)
- **Password**: Enter the password used to register with System Manager
- **Confirm Password**: Re-enter the password from above



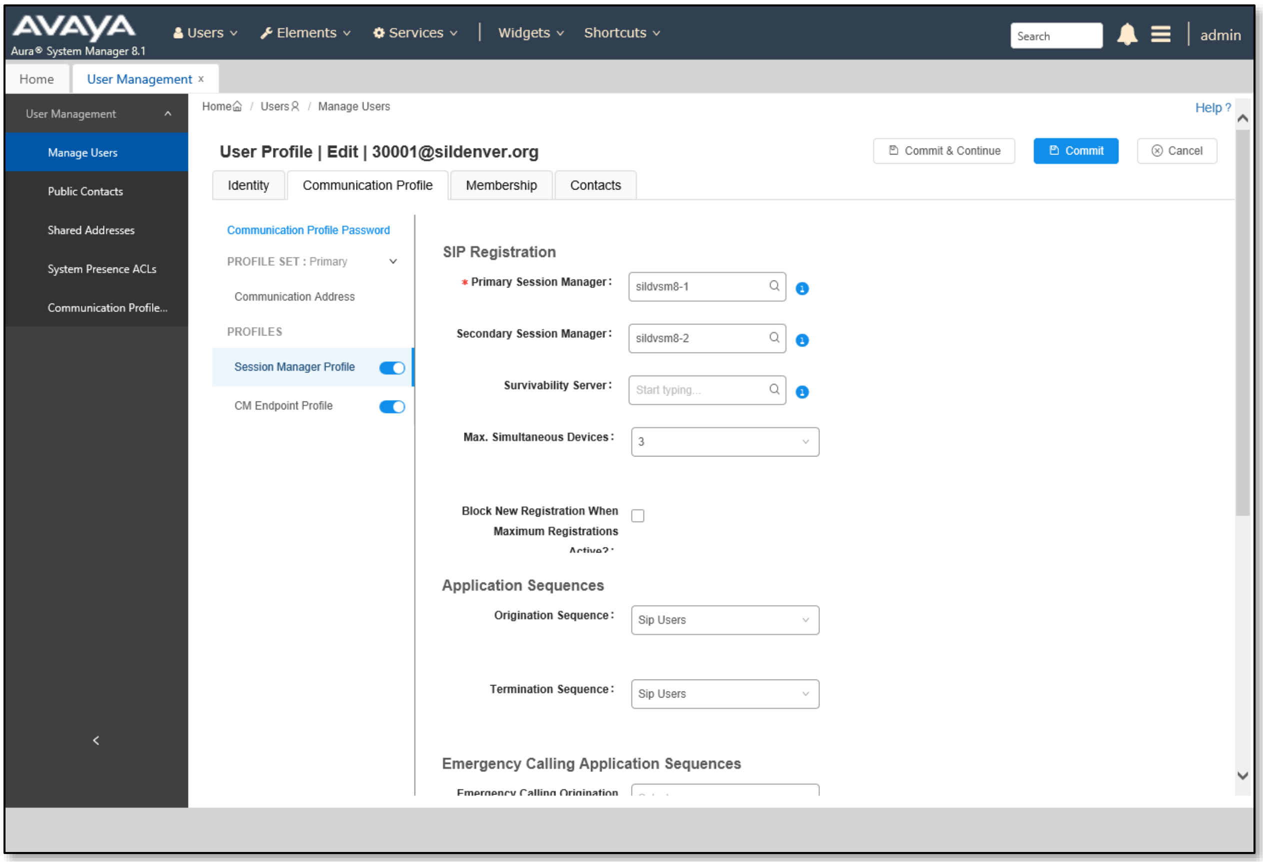
Click the **Communication Profile** tab, then click **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

- **Type**: Select ***Avaya SIP*** (default)
- **Fully Qualified Address**: Enter extension number and SIP domain

The screen below shows the information when adding a new SIP user to the sample configuration. Click **OK**.



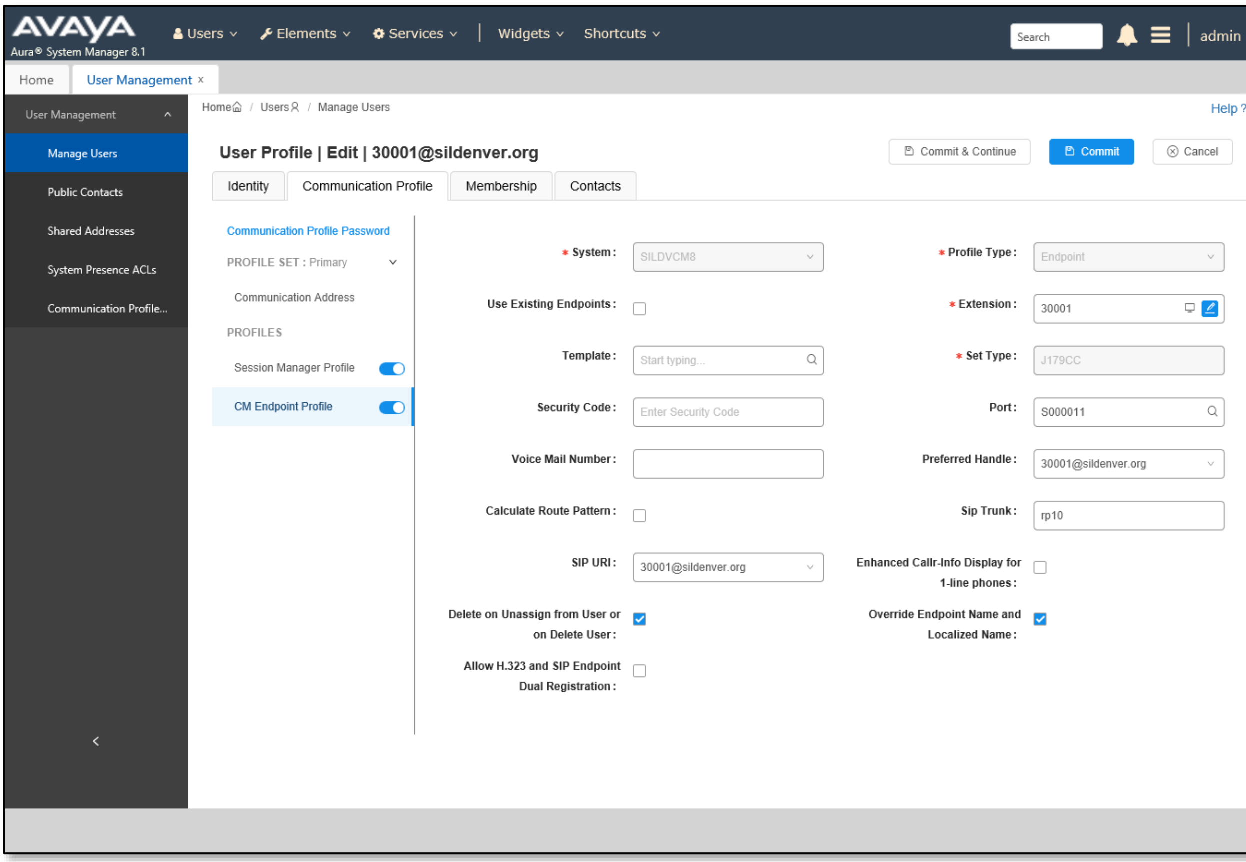
In the **Session Manager Profile** section, specify the Session Manager(s) and assign the **Application Sequence** to both the **Originating Sequence** and **Termination Sequence** fields.



In the **CM Endpoint Profile** section, fill in the following fields:

- **System**: Select the managed element corresponding to Communication Manager
- **Profile Type**: Select ***Endpoint***
- **Use Existing Stations**: If field is not selected, the station will automatically be added in Communication Manager
- **Extension**: Enter extension number of the SIP user
- **Template**: Select a template for type of SIP phone, this endpoint was previously configured using the J179CC template.

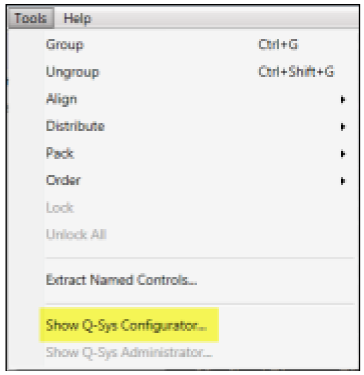
Click **Commit**.



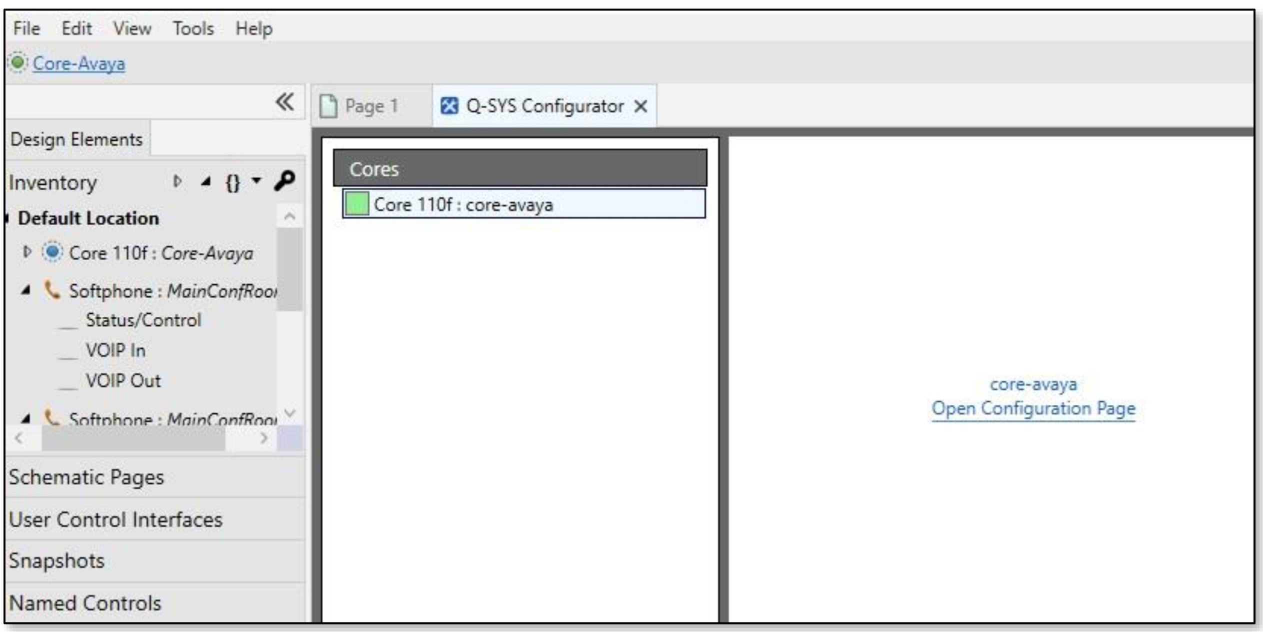
If additional endpoint features are required, click on the blue icon in the extension field to display the endpoint editor. Defaults were used in this testing.

6. Configure QSC Q-Sys Core and Q-Sys Softphone

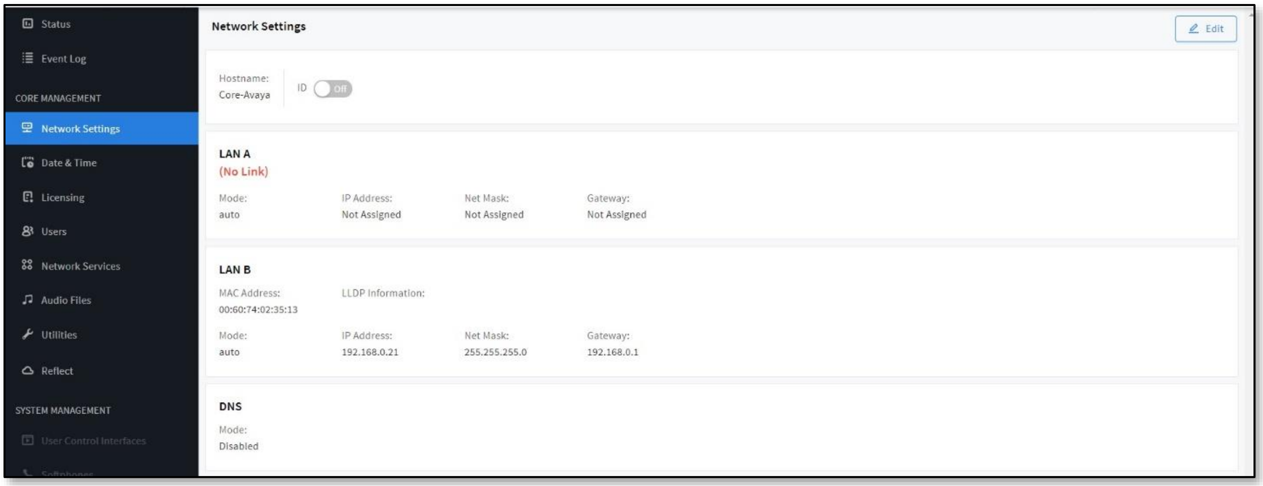
Run the Q-Sys Designer software on a Windows desktop. Select **Tools → Show Q-Sys Configurator**.



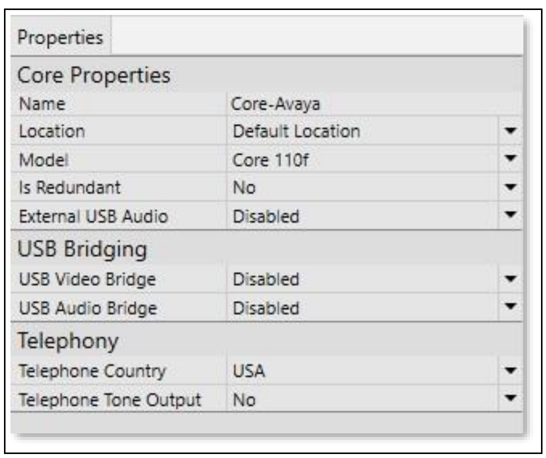
Select the Core model being configured for Softphone usage from the list of discovered Cores in the left column of the Q-Sys Configurator. Click on the link in the middle of the window that says **Open Configuration Page**.



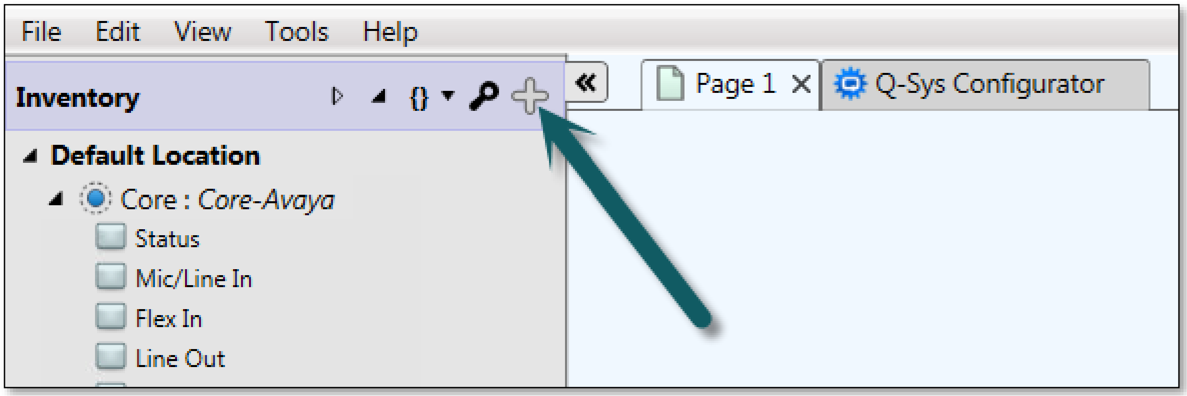
Set the appropriate network settings on the network interface used to connect to Session Manager. **LAN A** is often used for connection to Q-Sys peripherals, and **LAN B** is often chosen to isolate VoIP traffic from other Q-Sys audio and control traffic. End user can configure any network interface for use with Communication Manager. For this testing, **LAN A** was chosen and configured as shown below. Click **Edit** to change the network settings and then **Save**.



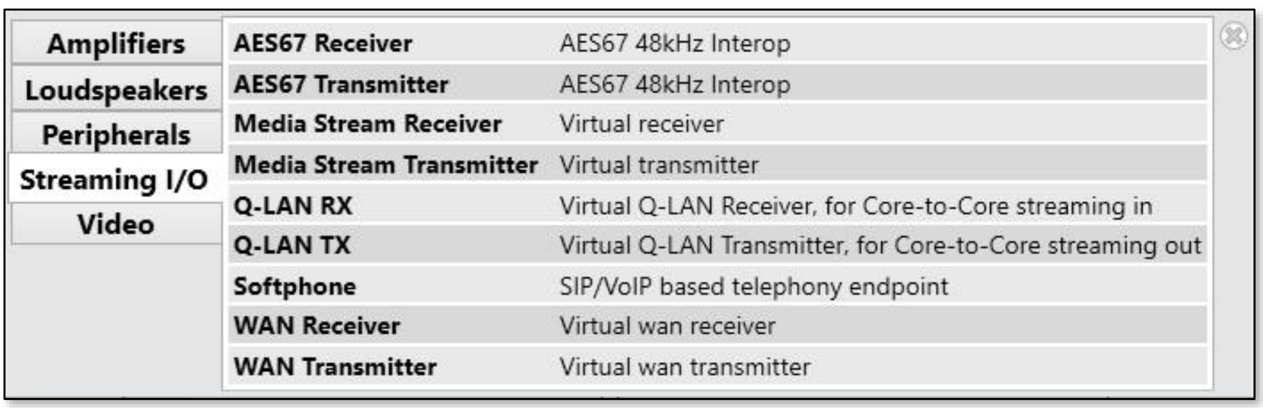
Copy the Core name from the Configurator to the Core Properties section on your design and choose the proper model.



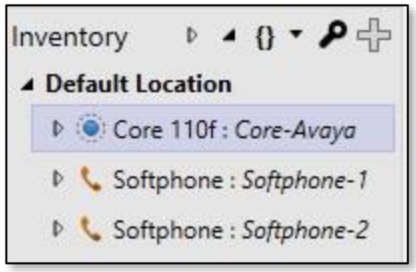
In the far-left column of Q-Sys Designer, click on plus sign to add a Softphone instance to the current design file.



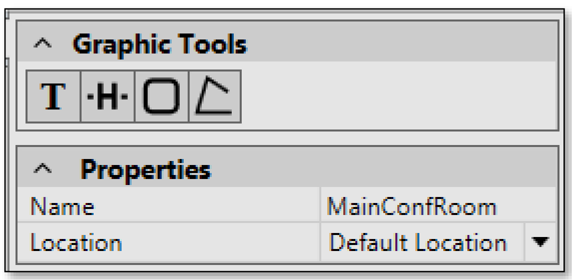
The **Add Inventory Item** menu will appear to the right of the plus button showing all of the items available to add to the design. Navigate to and click **Streaming I/O → Softphone** to insert a Softphone instance to the Inventory list. The **Add Inventory Menu** disappears. Follow the same steps to add additional items to the design as needed.



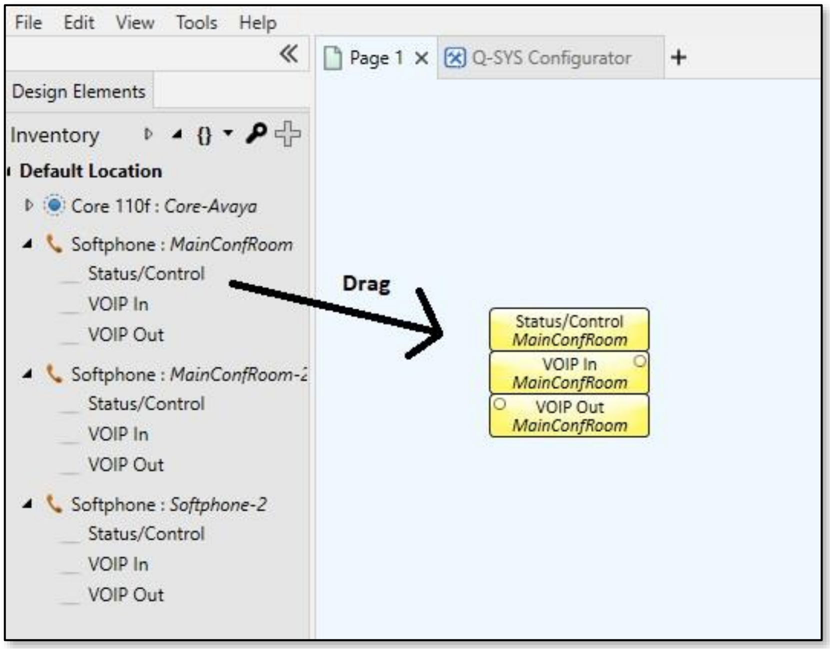
This screenshot shows two softphones were added to the Inventory list of the design file.



The name of each Softphone instance will be in the form Softphone-*n*, where *n* is an ascending integer value. The name may be changed to describe the room from which it will be used or the intended use of the instance. To change the name representing each Softphone instance, click on a Softphone instance in the Inventory Item list and edit the Name property field in the **Properties** section to the upper right side of the Q-Sys Designer user interface. The following screenshot shows that the Name property has been changed to describe the room where the Softphone instance will be used:

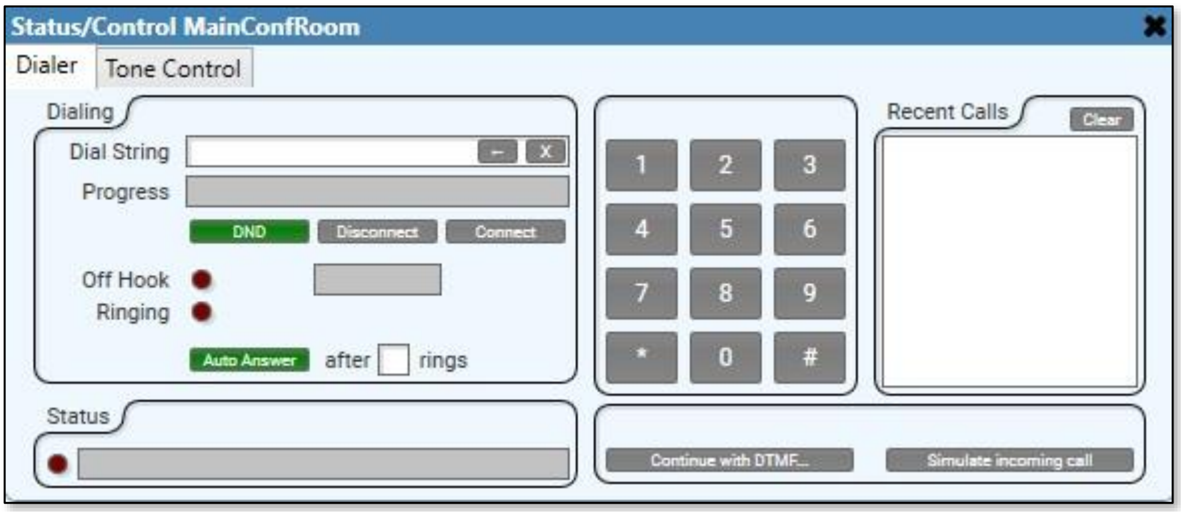


From the Inventory list, drag the individual components of the Softphone into the work area by clicking on the  icon. Alternately, click on the Softphone name and drag into the work area to place all three sub-components into the design at one time (as shown below).



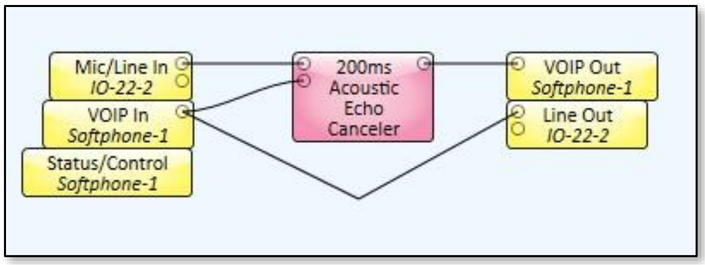
The three sub-components each Softphone instance provides are the following:

- **Status/Control** – this block contains the user interface for the dialpad, off/on hook, redial, auto-answer, local do-not-disturb (not the Communication Manager Do Not Disturb) and a Flash Hook button. Overall Status and call progress blocks, Off Hook and Ringing LEDs are provided for creating user interfaces for monitoring. There is also a call timer.
- **VoIP In** – brings the incoming telephony audio into the Q-Sys design for routing as an audio source (Audio from Session Manager)
- **VoIP Out** – receives the audio from the Q-Sys design which is intended to be sent as telephony audio (Audio to Session Manager).

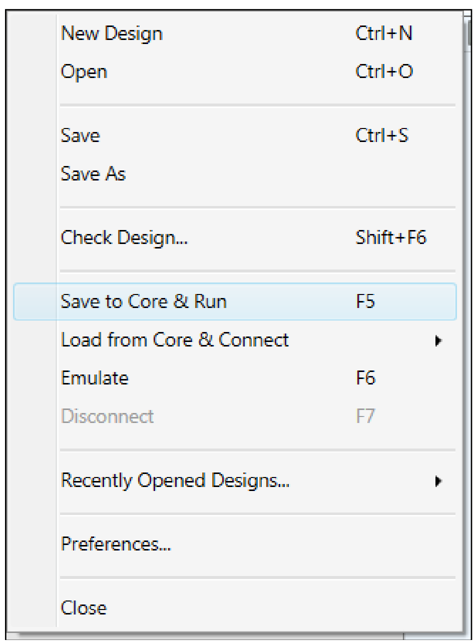


Add components to the design by dragging, then "wire" them similarly to the softphone as shown below. To draw a wire between components, click within an audio pin circle and then drag the mouse and release the mouse button over another appropriate pin (outputs to inputs).

An **Acoustic Echo Canceller** audio component should be included in the telephony signal path. This component is necessary in order to cancel any far-end audio which may be received by room microphones from being returned to the far end. The lower pin on the left side of the AEC component is the "Reference" pin. Any audio received on this pin will be automatically removed from the audio signal received from the room microphone(s) before being sent into the Q-Sys **Softphone Out** block for transmission.

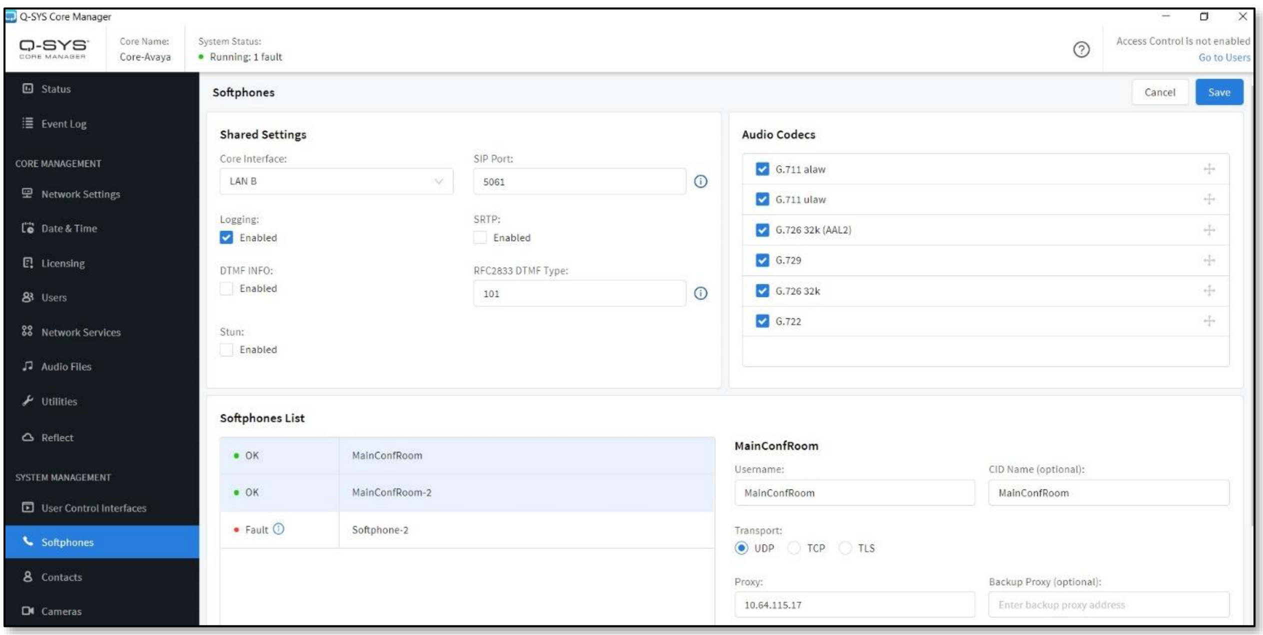


To "Deploy" the design to the Q-Sys Core press **F5** or **File → Save to Core & Run**

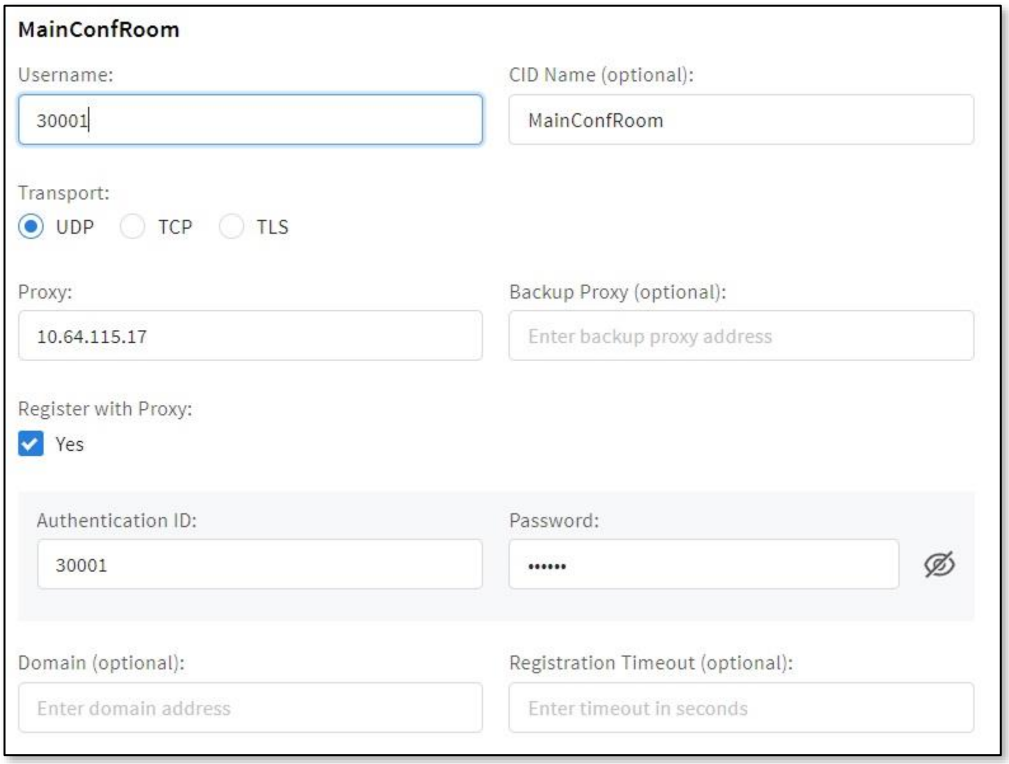


Once the design is running and a green Core Status LED is visible in the upper left of the user interface, further configuration of the Softphone is required to allow registration to the Session Manager. Navigate to **Tools → Show Q-Sys Core Manager** or use the  button which appears in the upper right corner of the user interface. Once the Core Manager appears, select the **Softphones** tab.

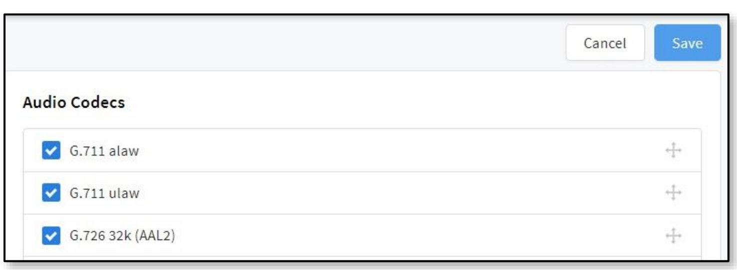
Select the codecs you wish to use with your configuration or click them all if you wish to have Session Manager control which codecs are used. Configure the port that matches your Session Manager configuration. For normal configurations this will usually be 5060. If encryption is being used it will normally be 5061. By default the Softphone uses RFC2833 and DTMF Type 101. If needed change these values. The Softphone also supports the DTMF INFO method as well. When any changes are made within the Core Manager then you will need to click the **Save** button on the upper right. Changes are effective after the configuration is saved.



Click an entry in the Softphones list to be configured. A box in the lower right of the screen will show with the parameters for that softphone. Enter the softphone registration details similarly to those shown below, which correspond to the details configured within Session Manager . The Username field will be the extension number chosen in **Section 5.1**. The **CID name** is what the Softphone will report as the display name. Register with proxy should be set to **Yes** unless SIP Trunking is being used (see document describing SIP Trunking Configuration). The **Proxy** corresponds to the **IP Address** in **Section 5.1** and **Authentication ID** corresponds to the **Name** configured in **Section 5.1**. Password is same as configured in **Section 5.1**. The **Domain (Optional)** can be configured to report the organization's domain as part of the URI.



Once the necessary changes have been made to the Softphone tab of the Administrator, click the **Save** button for the changes to take effect.

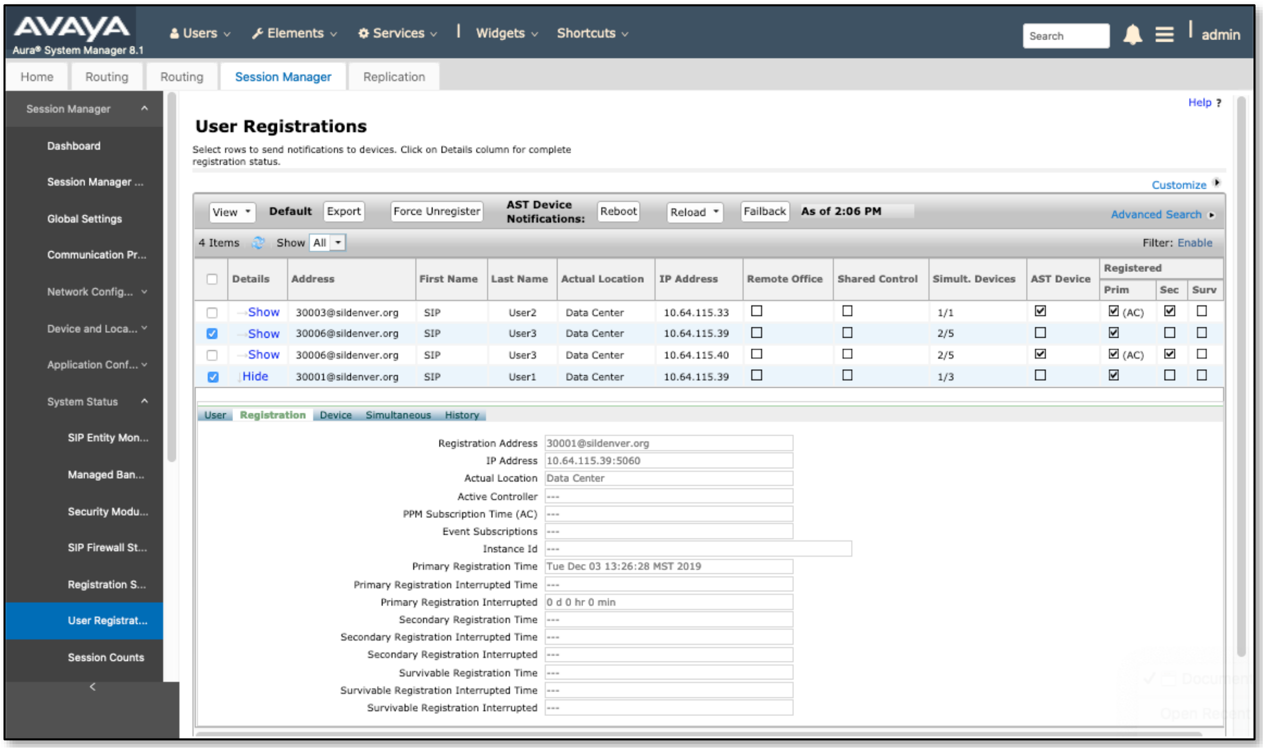


If the Softphone has registered successfully with Session Manager, it will show green in the Softphones List.

7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Session Manager and Communication Manager with Q-Sys Softphone.

- Verify that Q-Sys Softphone is registered with Session Manager. The following screen shows the registered SIP users with Session Manager:



- Verify codecs and encryption using status station (for calls connected to H.323 and DCP stations) or status trunk (for calls connected to SIP stations) commands in Communication Manager:

```
status station 30002                                            Page    8 of    9
                      SRC PORT TO DEST PORT TALKPATH
src port: S000005
S000005:TX:10.64.115.36:2062/g729a/20ms/1-srtp-aescm128-hmac80
001V011:RX:10.64.115.2:2056/g729/20ms/1-srtp-aescm128-hmac80:TX:ctxID:106
001V012:RX:ctxID:106:TX:10.64.115.2:2052/g729/20ms/1-srtp-aescm128-hmac80
T000001:RX:10.64.115.39:16486/g729/20ms/1-srtp-aescm128-hmac80

status station 30002                                            Page    7 of    8
                      SRC PORT TO DEST PORT TALKPATH
src port: S000005
S000005:TX:10.64.115.36:2062/g722-64/20ms/1-srtp-aescm128-hmac80
AMS1:RX:10.64.115.3:6008/g722-64/20ms/1-srtp-aescm128-hmac80:TX:cnfID:0
AMS1:RX:cnfID:0:TX:10.64.115.3:6006/g722-64/20ms/1-srtp-aescm128-hmac80
T000001:RX:10.64.115.39:16404/g722-64/20ms/1-srtp-aescm128-hmac80
```

```
status trunk 10

                             TRUNK GROUP STATUS

Member    Port    Service State      Mtce Connected Ports
                                     Busy

0010/0001 T000001 in-service/active  no   T000005
0010/0002 T000002 in-service/idle    no
0010/0003 T000003 in-service/idle    no
0010/0004 T000004 in-service/idle    no
0010/0005 T000005 in-service/active  no   T000001
0010/0006 T000006 in-service/idle    no
0010/0007 T000007 in-service/idle    no
0010/0008 T000008 in-service/idle    no
0010/0009 T000009 in-service/idle    no
0010/0010 T000010 in-service/idle    no


status trunk 10/0001                                          Page   3 of   3
                      SRC PORT TO DEST PORT TALKPATH
src port: T000001
T000001:TX:10.64.115.39:16446/g729/20ms
AMS1:RX:10.64.115.3:6174/g729/20ms:TX:cnfID:0
AMS1:RX:cnfID:0:TX:10.64.115.3:6176/g729/20ms/1-srtp-aescm128-hmac80
T000005:RX:10.64.115.40:5004/g729/20ms/1-srtp-aescm128-hmac80


dst port: T000005


status trunk 10/0005                                          Page   3 of   3
                      SRC PORT TO DEST PORT TALKPATH
src port: T000005
T000005:TX:10.64.115.40:5004/g729/20ms/1-srtp-aescm128-hmac80
AMS1:RX:10.64.115.3:6176/g729/20ms/1-srtp-aescm128-hmac80:TX:cnfID:0
AMS1:RX:cnfID:0:TX:10.64.115.3:6174/g729/20ms
T000001:RX:10.64.115.39:16446/g729/20ms

dst port: T000001
```

8. Conclusion

These Application Notes describe the configuration steps required for QSC Q-Sys Softphone to successfully interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. All feature and serviceability test cases were completed with the exceptions noted in **Section 2.2**.

9. Additional References

This section references the product documentation available at support.avaya.com relevant to these Application Notes.

[1] Administering Avaya Aura® Communication Manager, Release 8.1.x, November 2019
[2] Administering Avaya Aura® Session Manager, Release 8.1.1, October 2019
[3] Implementing and Administering Avaya Aura® Media Server, Release 8.0.x, December 2019

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.