



Avaya Solution & Interoperability Test Lab

Application Notes for SIP Softphone by QSC Q-Sys platform with SIP Trunks to 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 use a SIP Trunk connection with Avaya Aura® Session Manager 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 use a SIP Trunk with Avaya Aura® Session Manager 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
- 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, 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.

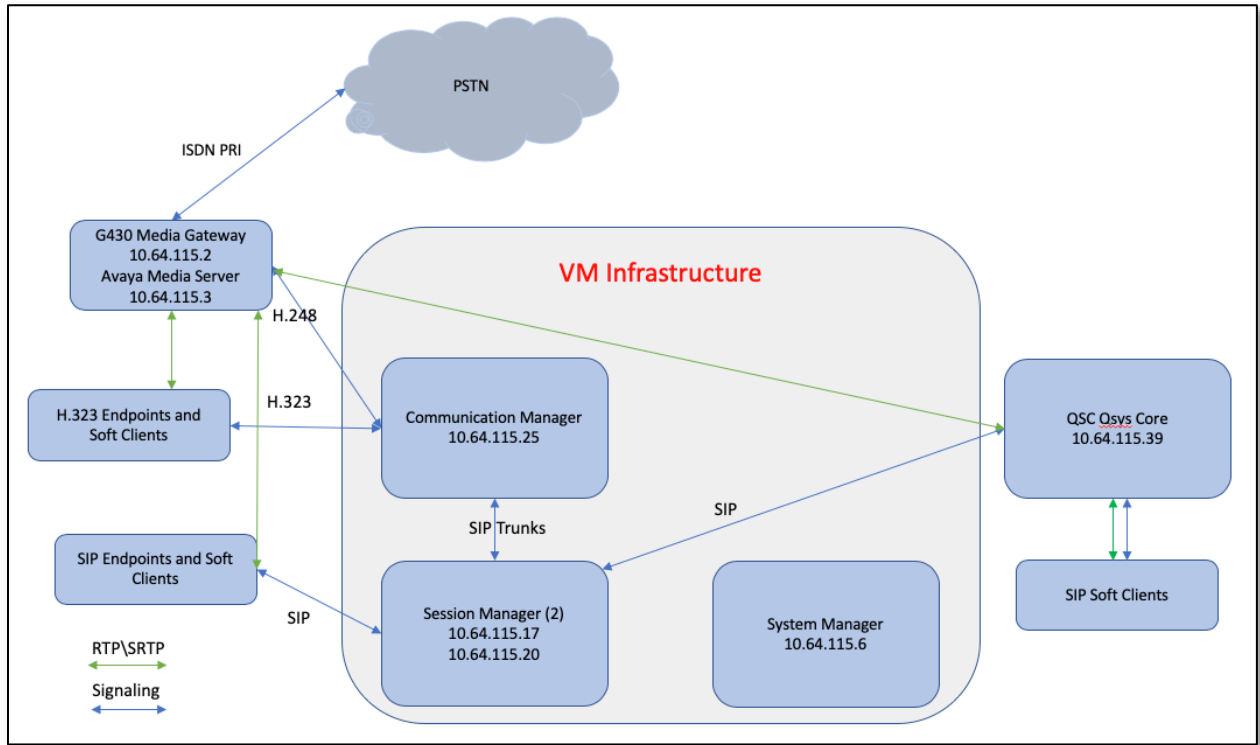


Figure 1: Q-Sys SIP Softphone with Avaya Aura[®] Session Manager and Avaya Aura[®] Communication Manager

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 a SIP Trunk to the QSC Q-Sys Core server.

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 Trunk

The QSC Q-Sys Core server was entered as a SIP Entity on Session Manager using the following steps.

5.1.1. Add SIP Entity

Navigate to **Home → Elements → Routing → SIP Entities** and configure as follows. This configuration is automatically synchronized with Session Manager.

Enter values for the following required attributes for SIP Entity:

- **Name:** Enter a name for the Entity
- **FQDN or IP Address:** Enter the Address of the QSC Q-Sys Core server
- **Type:** Select *SIP Trunk*
- **Adaptation:** Select the QSC Adaptation (created below)
- **Location:** Select a valid location (optional)
- **SIP Link Monitoring:** Select *Link Monitoring Disabled*

Click **Add** to create an Entity Link to Session Manager. Note that this configuration was tested with both UDP and TLS and the corresponding ports. The UDP configuration is shown below.

SIP Entity Details [Commit] [Cancel] [Help ?]

General

- Name: QSC
- FQDN or IP Address: 10.64.115.139
- Type: SIP Trunk
- Notes:
- Adaptation: QSC
- Location: Data Center
- Time Zone: America/Denver
- SIP Timer B/F (in seconds): 4
- Minimum TLS Version: Use Global Setting
- Credential name:
- Securable:
- Call Detail Recording: egress

Loop Detection

- Loop Detection Mode: On
- Loop Count Threshold: 5
- Loop Detection Interval (in msec): 200

Monitoring

- SIP Link Monitoring: Link Monitoring Disabled
- CRLF Keep Alive Monitoring: CRLF Monitoring Disabled
- Supports Call Admission Control:
- Shared Bandwidth Manager:
- Primary Session Manager Bandwidth Association:
- Backup Session Manager Bandwidth Association:

Entity Links

Override Port & Transport with DNS SRV:

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
QSC	sildvsm8-1	UDP	5060	QSC	5060	trusted	<input type="checkbox"/>

Select : All, None

SIP Responses to an OPTIONS Request

5.1.2. Add Adaptation

Navigate to **Home → Elements → Routing → Adaptations** and click New (not shown) to create a header manipulation rule for the SIP Trunk. The purpose is to set the destination domain to the IP Address of the Q-Sys Core server on messages sent to the server, and the lab environment domain for messages sent to Session Manager. Provide the following:

- **Adaptation Name:** Enter a unique name
- **Module Name:** Select *DigitConversionAdapter*
- **Module Parameter Type:** Select Name-Value-Parameter
- **Click Add to specify the following parameters:**
 - **Name:** *fromto* **Value:** *true*
 - **Name:** *iodstd* **Value:** *sildenver.org*
 - **Name:** *iosrcd* **Value:** *sildenver.org*
 - **Name:** *odstd* **Value:** *10.64.115.39*
- Click **Add** to create an Incoming Digit Conversion rule to remove the + on incoming messages from QSC
- Click **Add** to create an Outgoing Digit Conversion rule to add the + on outgoing messages to QSC

Adaptation Details Commit Cancel Help ?

General

• **Adaptation Name:** QSC

• **Module Name:** DigitConversionAdapter

Module Parameter Type: Name-Value Parameter

<input type="checkbox"/>	Name	Value
<input type="checkbox"/>	fromto	true
<input type="checkbox"/>	iodstd	sildenver.org
<input type="checkbox"/>	iosrcd	sildenver.org

Select : All, None Page 1 of 2

Egress URI Parameters:

Notes:

Digit Conversion for Incoming Calls to SM

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
<input type="checkbox"/>	*+	6	6		1		both		

Select : All, None Filter: Enable

Digit Conversion for Outgoing Calls from SM

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
<input type="checkbox"/>	*+	6	6		1		both		

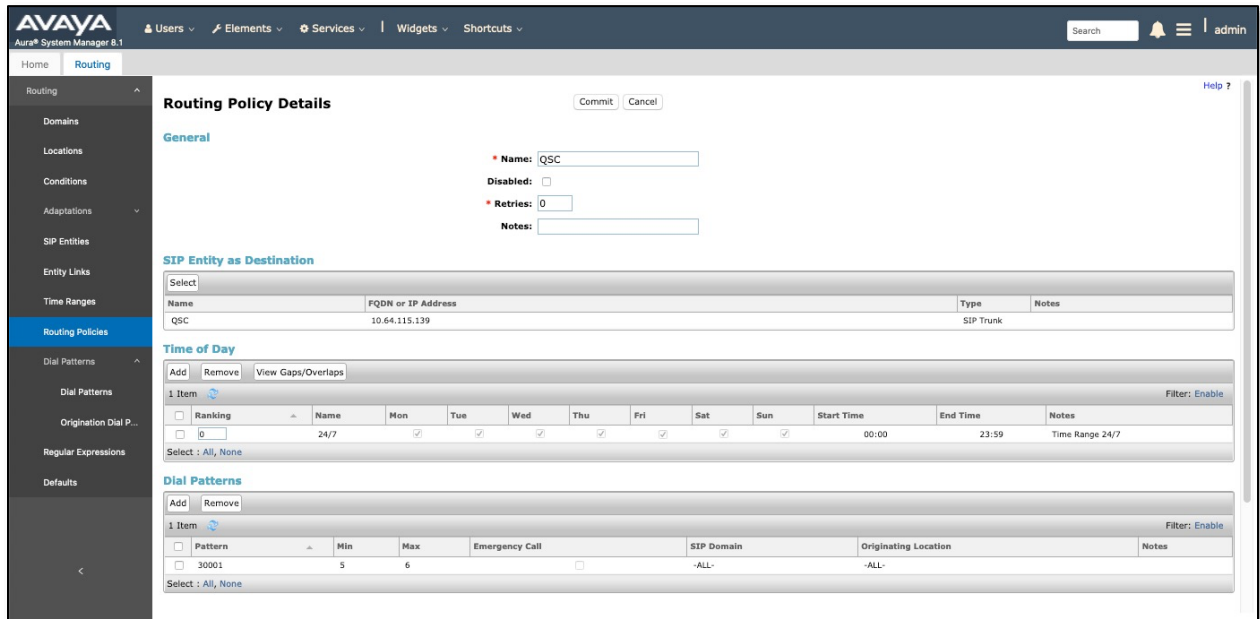
Select : All, None Filter: Enable

Commit Cancel

5.1.3. Add Routing Policy

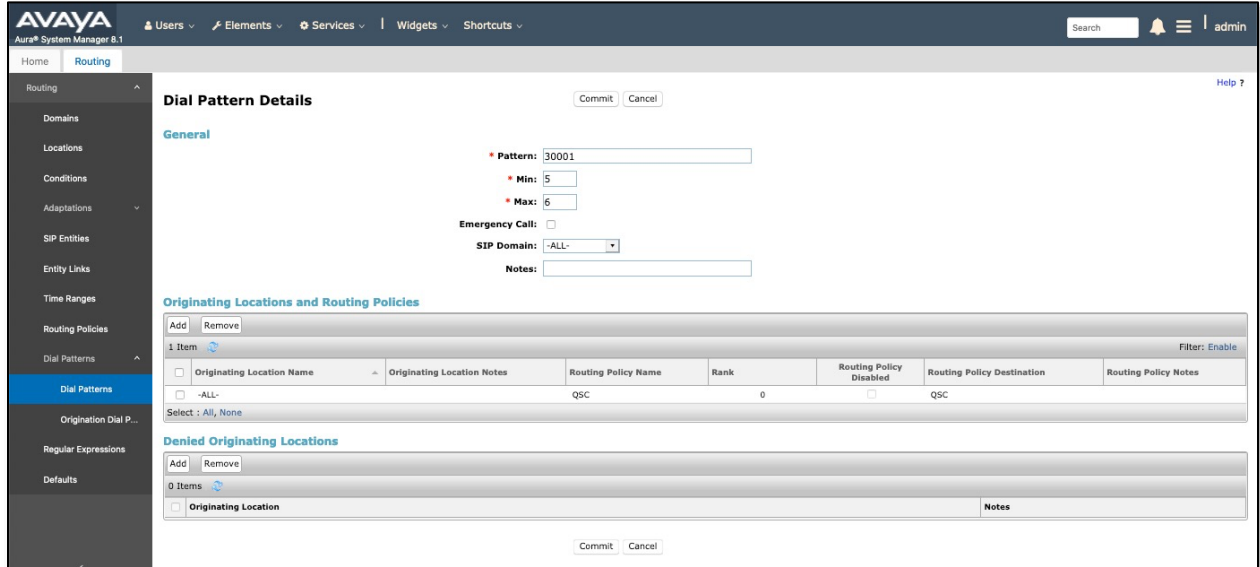
Navigate to Home → Elements → Routing → Routing Policies and click New (not shown) to create a routing policy rule for the SIP Trunk. The purpose is to set the destination for calls to the softphone extensions to the QSC solution. Provide the following:

- **Name:** A generally descriptive name for the Routing Policy.
- **SIP Entity as Destination:** Click **Select** and choose the SIP Entity created in the previous step (not shown).
- **Dial Patterns:** Click **Add** to display the Dial Pattern Details page, more details on this below.



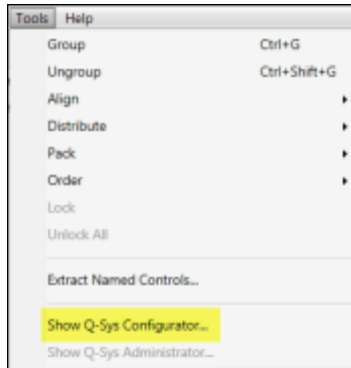
On the Dial Pattern Details page, enter the digit pattern, min and max digits, and SIP Domain to be used for this rule.

Click on **Add** to assign the Originating Location and Destination.

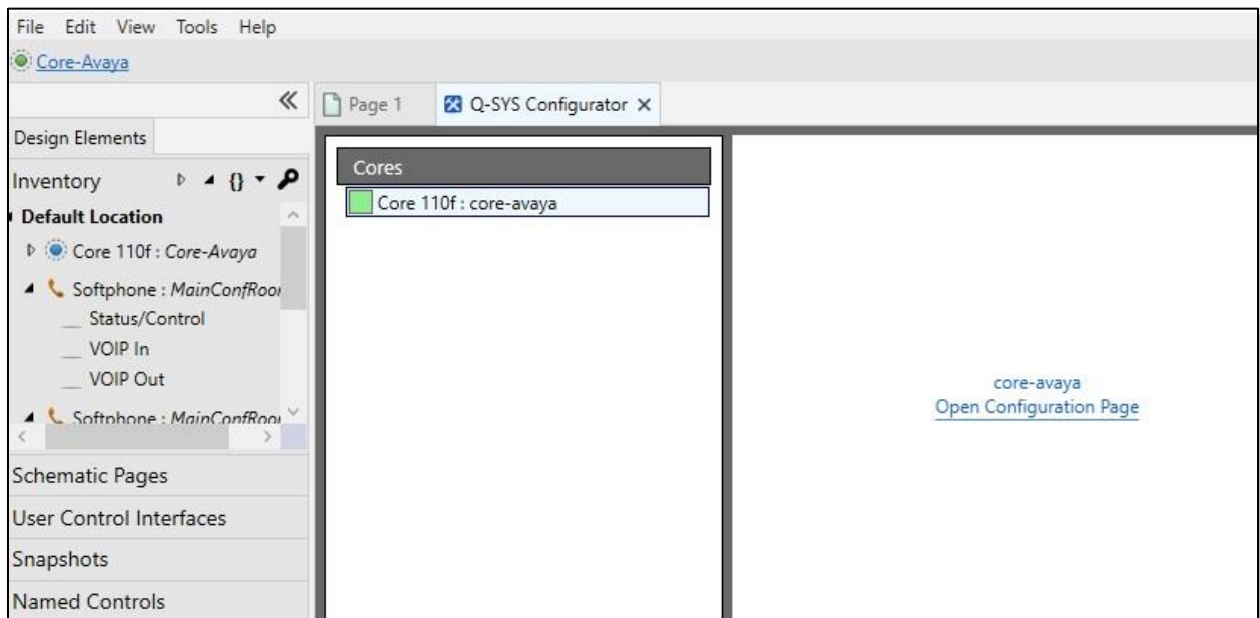


6. Configure QSC Q-Sys Core and Q-Sys Softphone

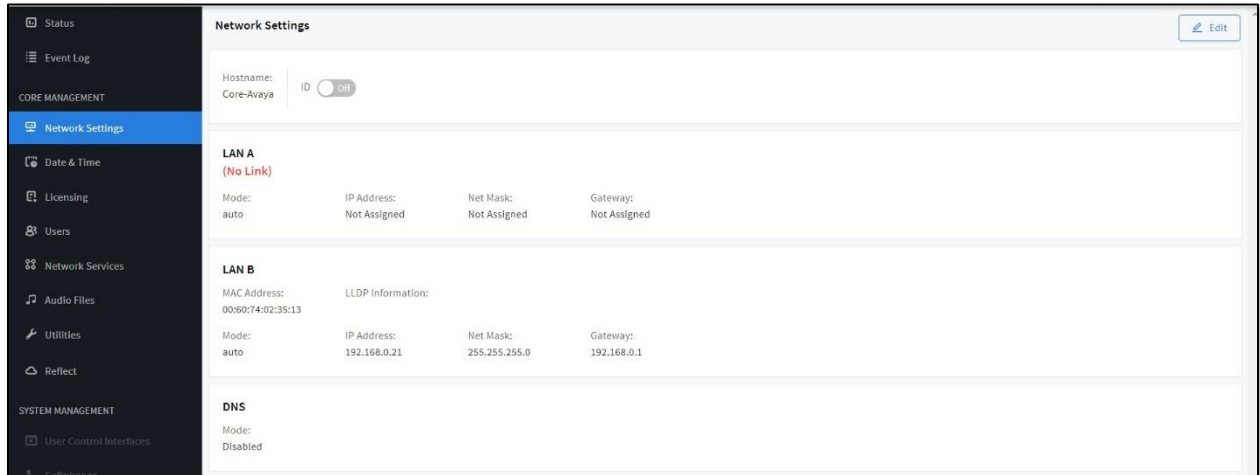
Run the Q-Sys Designer software on a Windows desktop. Select **Tools** → **Show Q-Sys Configurator**.



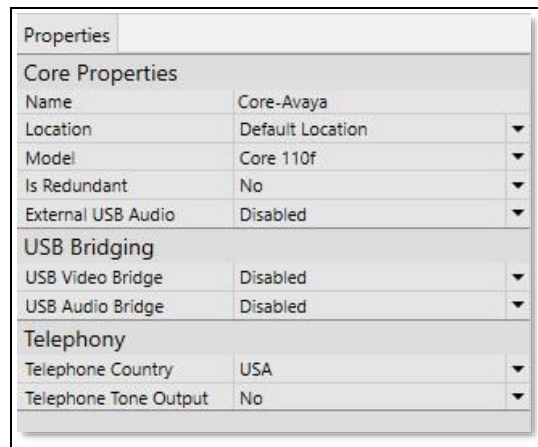
Select the Core model being configured for Softphone usage from the list of discovered Cores in the left column of the Q-Sys Configurator. Click on the link in the middle of the window that says **Open Configuration Page**.



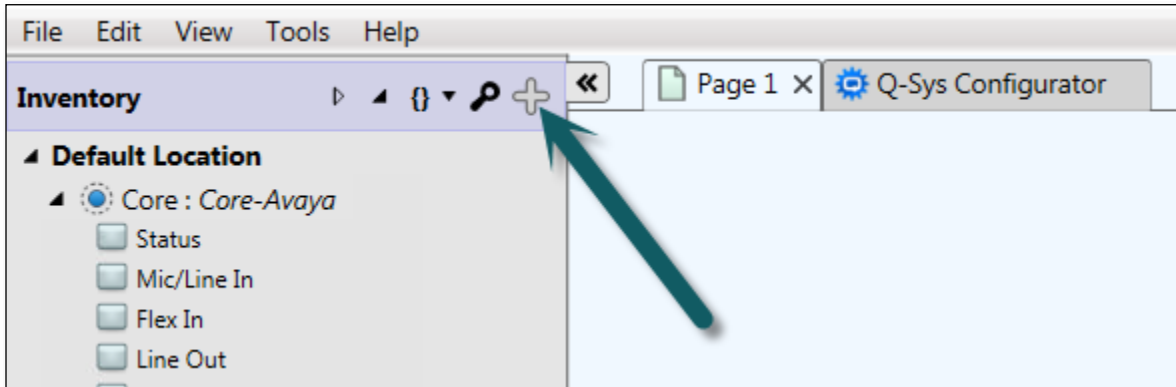
Set the appropriate network settings on the network interface used to connect to Session Manager. **LAN A** is often used for connection to Q-Sys peripherals, and **LAN B** is often chosen to isolate VoIP traffic from other Q-Sys audio and control traffic. End user can configure any network interface 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.



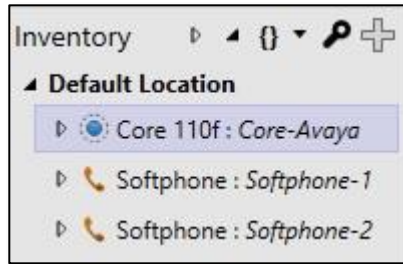
In the far-left column of Q-Sys Designer, click on plus sign to add a Softphone instance to the current design file.



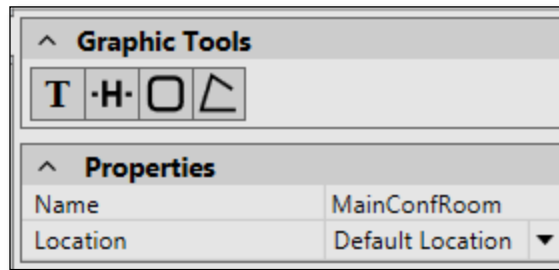
The **Add Inventory Item** menu will appear to the right of the plus button showing all of the items available to add to the design. Navigate to and click **Streaming I/O → Softphone** to insert a Softphone instance to the Inventory list. The **Add Inventory Menu** disappears. Follow the same steps to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


Amplifiers	AES67 Receiver	AES67 48kHz Interop
Loudspeakers	AES67 Transmitter	AES67 48kHz Interop
Peripherals	Media Stream Receiver	Virtual receiver
Streaming I/O	Media Stream Transmitter	Virtual transmitter
Video	Q-LAN RX	Virtual Q-LAN Receiver, for Core-to-Core streaming in
	Q-LAN TX	Virtual Q-LAN Transmitter, for Core-to-Core streaming out
	Softphone	SIP/VoIP based telephony endpoint
	WAN Receiver	Virtual wan receiver
	WAN Transmitter	Virtual wan transmitter

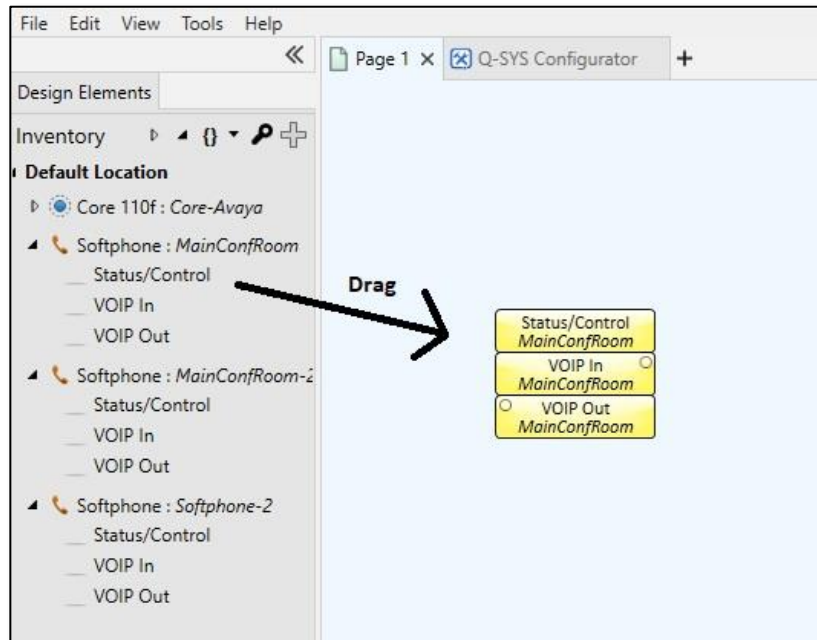
This screenshot shows two softphones were added to the Inventory list of the design file.



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:

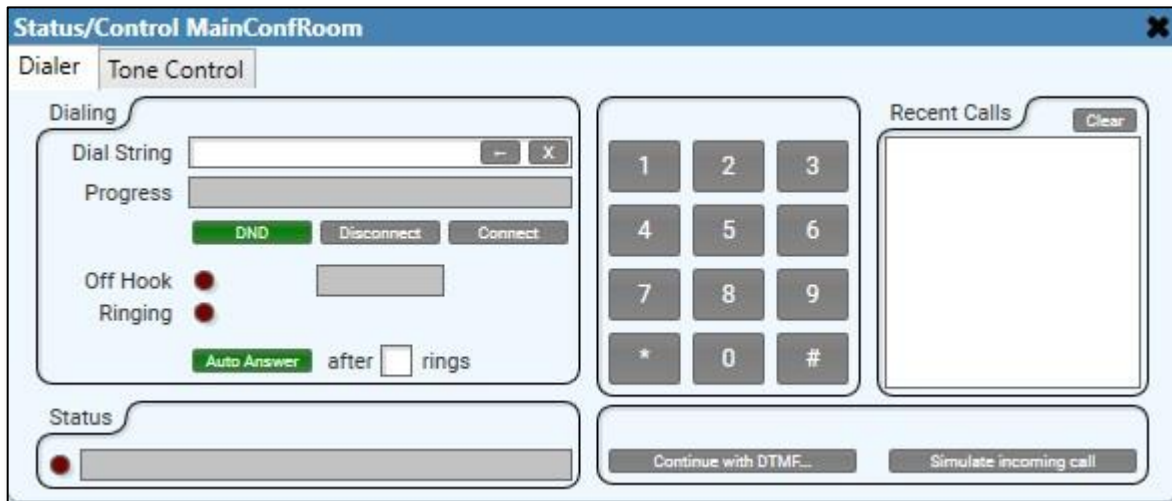


From the Inventory list, drag the individual components of the Softphone into the work area by clicking on the  icon. Alternately, click on the Softphone name and drag into the work area to place all three sub-components into the design at one time (as shown below).



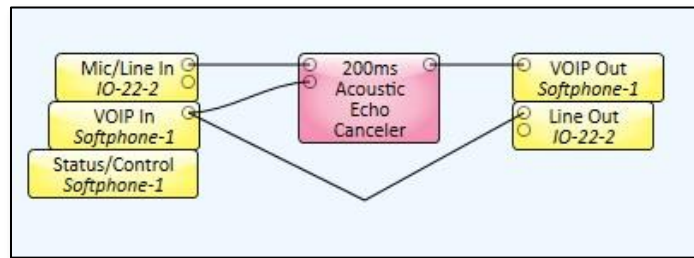
The three sub-components each Softphone instance provides are the following:

- **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).

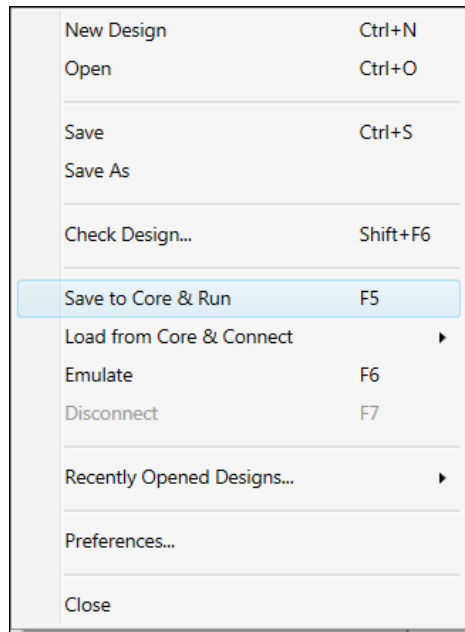


Add components to the design by dragging, then “wire” them similarly to the softphone as shown below. To draw a wire between components, click within an audio pin circle and then drag the mouse and release the mouse button over another appropriate pin (outputs to inputs).


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.



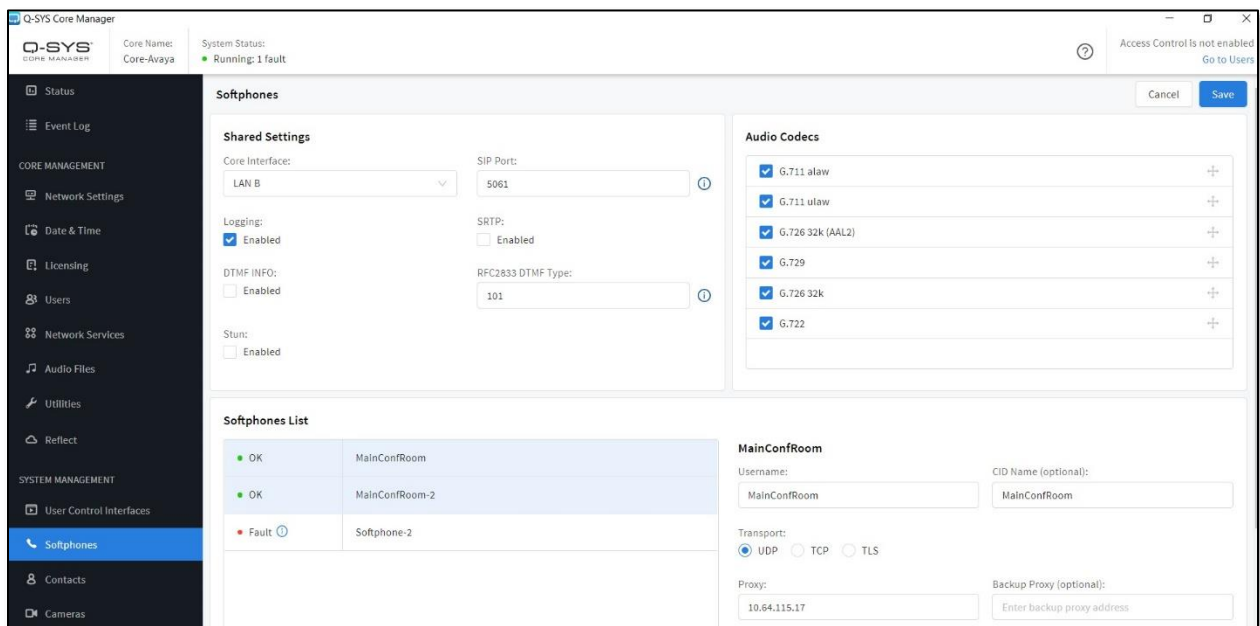
To “Deploy” the design to the Q-Sys Core press **F5** or **File → Save to Core & Run**



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

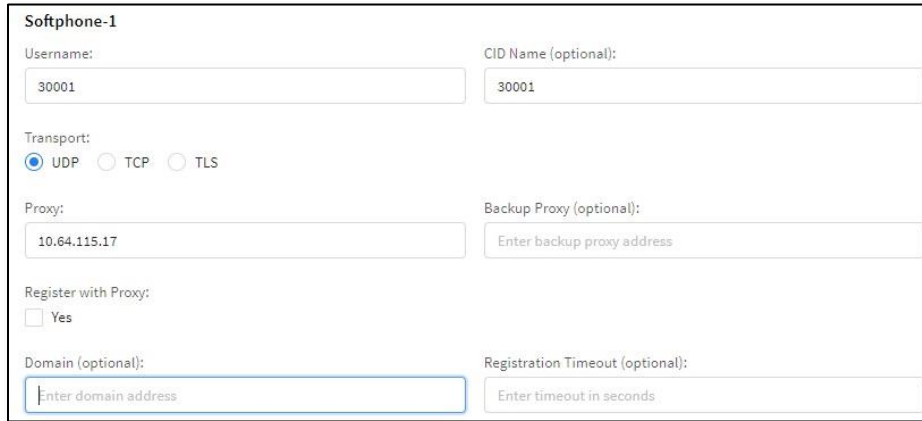
Manager. Navigate to **Tools** → **Show Q-Sys Core Manager** or use the  button which appears in the upper right corner of the user interface. Once the Core Manager appears, select the **Softphones** tab.

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.

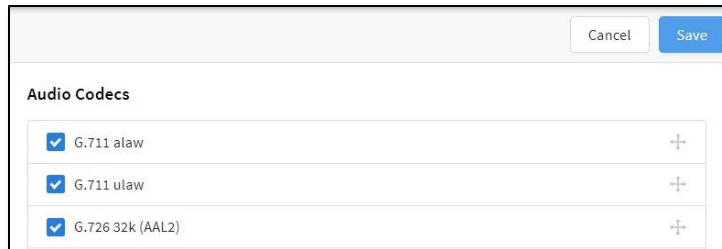
Register with proxy should be set to **No** when SIP Trunking is being used.



The screenshot shows a configuration form for 'Softphone-1'. It includes the following fields and options:

- Username:** 30001
- CID Name (optional):** 30001
- Transport:** UDP, TCP, TLS
- Proxy:** 10.64.115.17
- Backup Proxy (optional):** Enter backup proxy address
- Register with Proxy:** Yes
- Domain (optional):** Enter domain address
- Registration Timeout (optional):** Enter timeout in seconds

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



The screenshot shows a dialog box titled 'Audio Codecs' with 'Cancel' and 'Save' buttons. It contains a list of audio codecs with checkboxes and plus signs:

Audio Codec	Selected	Action
G.711 alaw	<input checked="" type="checkbox"/>	+
G.711 ulaw	<input checked="" type="checkbox"/>	+
G.726 32k (AAL2)	<input checked="" type="checkbox"/>	+

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 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

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