

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

<u>qsyssupport@qsc.com</u> (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	R3.0.0.16
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

Aura® System Manager 8.1	Users 🗸 🌶 Elements 🗸 🌣 Ser	vices ~ Widgets ~ Shori	tcuts ~	s	earch 🔶 🔔 🗧 admin		
Home User Managemen							
User Management 🔹 ^	User Management A Home A / Users R / Manage Users Help?						
Manage Users	Manage Users User Profile Edit 30001@sildenver.org						
Public Contacts	Identity Communication Pro	ofile Membership Contact	s				
Shared Addresses	Basic Info						
System Presence ACLs	System Presence ACLs Address						
Communication Profile	LocalizedName	* Last Name :	User1	Last Name (Latin Translation) :	User1		
		* First Name :	SIP	First Name (Latin Translation):	SIP		
		* Login Name :	30001@sildenver.org	Middle Name :	Middle Name Of User		
		Description :	Description Of User	Email Address :	Email Address Of User		
		Password :		User Type :	Basic v		
		Confirm Password :		Localized Display Name :	User1, SIP		
		Endpoint Display Name :	SIP User1	Title Of User:	Title Of User		
		Language Preference :	English (United States)	Time Zone :			
<		Employee ID :	Employee Id Of User	Department :	Department Of User		
		Company :	Company Of User]	~		

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**.



Communication Address Add/Edit			
* Type :	Avaya SIP	~	
*Fully Qualified Address:	30001 @ sildenver.org	~	
	Cancel	ОК	

RAB; Reviewed: SPOC 2/4/2020 Solution & Interoperability Test Lab Application Notes ©2020 Avaya Inc. All Rights Reserved. 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.

Aura® System Manager 8.1	Jsers 🗸 🎤 Elements 🗸 🌣 Servic	ces ~ Widgets ~ Shortc	uts v	Search 💄	☰ admin
Home User Management	x				
User Management 🔹 ^	Home🏠 / Users 🎗 / Manage Users				Help?
Manage Users	User Profile Edit 30001@	@sildenver.org		🖻 Commit & Continue 🖻 Commit	⊗ Cancel
Public Contacts	Identity Communication Profile	e Membership Contacts			_
Shared Addresses	Communication Profile Password				
System Presence ACLs	PROFILE SET : Primary V	SIP Registration			
Communication Profile	Communication Address	* Primary Session Manager:	sildvsm8-1 Q		
	PROFILES	Secondary Session Manager:	sildvsm8-2 Q		
	Session Manager Profile	Survivability Server:	Start typing Q		
	CM Endpoint Profile	Survivability Surver	Start typing Q		
		Max. Simultaneous Devices:	3 ~ _		
		Block New Registration When Maximum Registrations			
		Application Sequences			
		Origination Sequence:	Sip Users v		
<		Termination Sequence:	Sip Users v		
		Emergency Calling Applica	ation Sequences		~
		Emergency Calling Origination			*

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 Select a template for type of SIP phone, this endpoint was **Template**:
 - previously configured using the J179CC template.

Click Commit.

Aura® System Manager 8.1						
Home User Managemen	t x					
User Management A Home A / Users A / Manage Users H						
Manage Users	User Profile Edit 30001@	gsildenver.org		🖻 Commit & Continue	Commit © Cancel	
Public Contacts	Identity Communication Profile	e Membership Contacts				
Shared Addresses	Communication Profile Password					
System Presence ACLs	PROFILE SET : Primary V	* System :	SILDVCM8 ~	* Profile Type :	Endpoint ~	
Communication Profile	Communication Address	Use Existing Endpoints :		* Extension :	30001 🖵 💆	
	PROFILES	Template :	Start typing Q	* Set Type :	J179CC	
	Session Manager Profile 💽		Start typing Q		11/400	
	CM Endpoint Profile	Security Code:	Enter Security Code	Port :	S000011 Q	
		Voice Mail Number:		Preferred Handle :	30001@sildenver.org v	
		Columbra Danta Dattarra		Cia Taunha		
		Calculate Route Pattern :		Sip Trunk :	rp10	
		SIP URI :	30001@sildenver.org ~	Enhanced Callr-Info Display for 1-line phones :		
		Delete on Unassign from User or				
		on Delete User:	-	Localized Name :	-	
		Allow H.323 and SIP Endpoint Dual Registration :				
<	1					

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** \rightarrow **Show Q-Sys Configurator**.

Took	Help	
	Group	Ctrl+G
	Ungroup	Ctrl+Shift+G
	Align	•
	Distribute	•
	Pack	•
	Order	•
	Lock	
	Unlock All	
	Extract Named Controls	
	Show Q-Sys Configurator	
	Show Q-Sys Administrator	

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.

Status	Network Settings					🖉 Edit
🗮 Event Log	Hostname:					
CORE MANAGEMENT	Core-Avaya	ОП				
Network Settings						
Co Date & Time	LAN A (No Link)					
E Licensing	Mode: auto	IP Address: Not Assigned	Net Mask: Not Assigned	Gateway: Not Assigned		
8 Users	uneres.	0.00				
88 Network Services	LAN B					
♫ Audio Files	MAC Address: 00:60:74:02:35:13	LLDP Information:				
🖌 Utilities	Mode: auto	IP Address: 192.168.0.21	Net Mask: 255.255.255.0	Gateway: 192.168.0.1		
C Reflect						
SYSTEM MANAGEMENT	DNS					
User Control Interfaces	Mode: Disabled					
C. Sakabaner						

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

Core Properties		
Name	Core-Avaya	
Location	Default Location	-
Model	Core 110f	-
Is Redundant	No	
External USB Audio	Disabled	-
USB Bridging		
USB Video Bridge	Disabled	
USB Audio Bridge	Disabled	-
Telephony		
Telephone Country	USA	-
Telephone Tone Output	No	-

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 $I/O \rightarrow Softphone$ 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:

^ Graphic Tools	
T H· C	
^ Properties	
Name	MainConfRoom
Location	Default Location 🔻

From the Inventory list, drag the individual components of the Softphone into the work area by clicking on the sicon. 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).

Status/Control	MainConfRoom		×
Dialer Tone C Dialing Dial String Progress	ontrol - ×	1 2 3	Recent Calls
Off Hook Ringing		4 5 6 7 8 9 * 0 #	
Status		Continue with DTMF	Simulate incoming call

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 \rightarrow Save to Core & Run

New Design	Ctrl+N
Open	Ctrl+O
Save	Ctrl+S
Save As	
Check Design	Shift+F6
Save to Core & Run	F5
Load from Core & Connect	•
Emulate	F6
Disconnect	F7
Recently Opened Designs	•
Preferences	
Close	

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** \rightarrow **Show Q-Sys Core Manager** or use the B 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.

Q-SYS Core Manager	r							-	o ×
CORE MANAGER	Core Name: Core-Avaya	System Status: • Running: I fault						Access Control I	s not enabled Go to Users
Status		Softphones						Cancel	Save
i≣ Event Log		Shared Settings				Audio Codecs			
CORE MANAGEMENT		Core Interface:		SIP Port:		G.711 alaw			+
Network Settin		LAN B	~	5061	0	G.711 ulaw			÷.
🖨 Date & Time		Logging:		SRTP: Enabled		G.726 32k (AAL2)			+
E Licensing		DTMF INFO:		RFC2833 DTMF Type:		✓ G.729			4
8 Users	8 Users Enabled			101	0	G.726 32k		++-	
88 Network Services Stun:					G.722		++++		
Audio Files		Enabled							
🖌 Utilities		Softphones List							_
🛆 Reflect						MainConfRoom			
SYSTEM MANAGEMENT		• OK	MainConfRoom			Username:	CID Name (optional):		
User Control In		• OK	MainConfRoom-2			MainConfRoom	MainConfRoom		
Softphones		• Fault ①	Softphone-2			Transport: UDP TCP TLS			
8 Contacts						Proxy:	Backup Proxy (optional):		
D4 Cameras						10.64.115.17	Enter backup proxy ad	dress	

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.

Username:	CID Name (optional):
30001	MainConfRoom
Transport:	
● UDP ○ TCP ○ TLS	
Proxy:	Backup Proxy (optional):
10.64.115.17	Enter backup proxy address
Register with Proxy: Yes	
Authentication ID:	Password:
30001	·····
Domain (optional):	Registration Timeout (optional):

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

	Cancel Save
Audio Codecs	
✔ G.711 alaw	4
🔽 G.711 ulaw	+1+
G.726 32k (AAL2)	\pm

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

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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:

ome	Routing Rou	ting	Session I	Manager Replicat	ion										
	<u> </u>	ung	000010111	indiagon replicat											Help :
ission N	Manager ^	Use	r Reai	strations											
Dash	board	Select r	ows to send	notifications to devices. Cl	ick on Details col	lumn for comp	lete								
Sessi	ion Manager	registra	ition status.											Custo	mize 🖲
		Vie	w T De	fault Export For	ce Unregister	AST Dev		Reload *	Failback As of	2:06 PM			Advanc		
Globa	al Settings				ee onregister	Notificat	tions:	Kelobu							
Comr	munication Pr	4 Iter	ms 🤕 Si	how All -										ilter: E	nable
			Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Register		
Netw	ork Config V		Show	30003@sildenver.org	SIP	User2	Data Center	10.64.115.33			1/1		Prim	Sec	Surv
Devic	ce and Loca Y		Show	30003@sildenver.org	SIP	User2 User3	Data Center	10.64.115.33			2/5		€ (AC)		
			→Show	30006@sildenver.org	SIP	User3	Data Center	10.64.115.40			2/5	2	(AC)		
Appli	ication Conf ~		↓Hide	30001@sildenver.org	SIP	User1	Data Center	10.64.115.39			1/3		Ø		
Syste	em Status 🔷														
		User	Registra	tion Device Simultan	eous History										
s	SIP Entity Mon						0001@sildenver.org								
N	Managed Ban						0.64.115.39:5060								
	g	Actual Location Active Controller													
s	Security Modu														
						bscriptions	-								
s	SIP Firewall St					instance Id									
	Registration S				Primary Registration Interru		ue Dec 03 13:26:28	MST 2019							
	tegistration 5				ry Registration I										
	Jser Registrat				condary Registr										
				Secondary Reg	gistration Interru	upted Time	-								
s	Session Counts				ry Registration I										
	<				irvivable Registr										
				Survivable Rec	istration Interru	inted Time									

• 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
                                                                Page 7 of 8
status station 30002
                     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
```

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```
status trunk 10
                             TRUNK GROUP STATUS
Member
       Port Service State
                                    Mtce Connected Ports
                                    Busv
0010/0001 T000001 in-service/active no T000005
0010/0002 T000002 in-service/idle
                                    no
                                  no
0010/0003 T000003 in-service/idle
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
0010/0009 T000009 in-service/idle
                                    no
                                    no
0010/0010 T000010 in-service/idle
                                    no
                                                                Page 3 of 3
status trunk 10/0001
                     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/q729/20ms
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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