



# **Application Notes for QSC Q-Sys SIP Softphone with Avaya Aura® Session Manager R7.0 and Avaya Aura® Communication Manager R7.0 – 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 R7.0 and Avaya Aura® Communication Manager R7.0. 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 R7.0 and Avaya Aura® Communication Manager R7.0. The QSC Q-Sys platform SIP Softphone (Q-Sys 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 is a 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 range of Linux-based platforms, 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 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.

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.

## 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 Avaya Session Manager. The following feature tests worked during compliance test:

- Registration
- Codecs G.711
- Inbound calls
- Outbound calls
- Call termination (origination/destination)
- Avaya Feature Access Codes
  - Call Pickup
  - Call Forward (Unconditional, Busy/no answer)
  - Find Me
- Voicemail
- Serviceability

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)
- G729/G722 Codec

The following observations were made during the testing:

- During compliance testing it was noted that when an inbound call to Q-Sys Softphone was terminated at Q-Sys Softphone end, the BYE message challenge from Communication Manager was not being authenticated. Q-Sys delivered the fix for this issue in **R5.1.53** of the Q-Sys platform software.
- G722 codec is supported by Q-Sys Softphone but during compliance testing, it was determined that the current implementation does not interoperate properly with Communication Manager. QSC is looking into this issue and will address this issue in subsequent release, and until then G722 codec should not be used by Q-Sys Softphone with Communication Manager.
- An issue was found with Q-Sys Softphone using RFC2833 for DTMF. QSC implemented a workaround to use SIP-INFO message for DTMF in **R5.1.53** of the software platform. Using SIP-INFO message for DTMF causes the calls not to shuffle. See **Section 7** to set this field.

## 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) 957-7150

### *Q-SYS 24/7 Emergency Support\**

Tel: +1-888-252-4836 (U.S./Canada)

Tel: +1-949-791-7722 (non-U.S.)

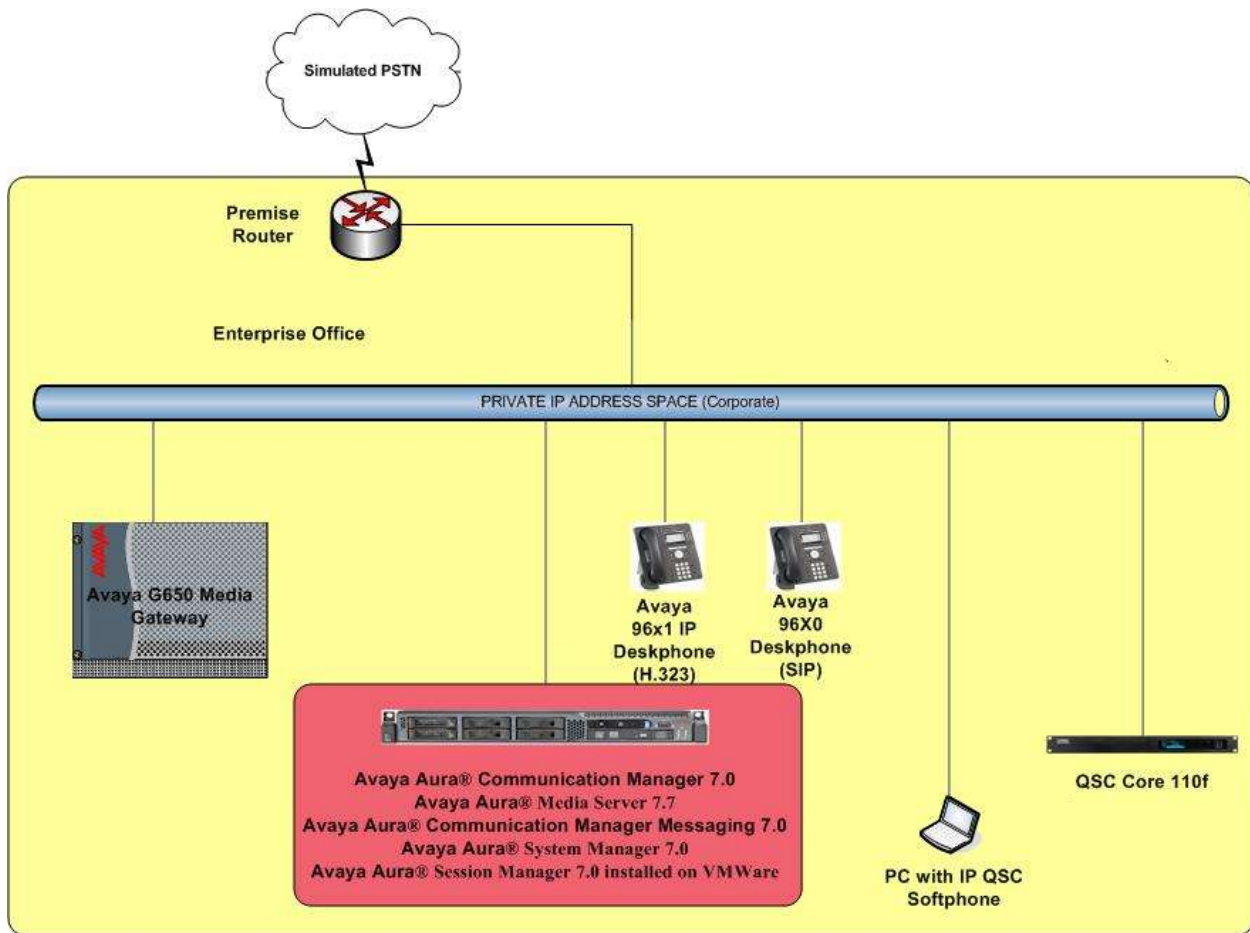
### *Q-SYS Support Email*

[qsyssupport@qsc.com](mailto: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	R7.0 (7.0.0.0.16266)
Avaya Aura® Session Manager installed on VMWare	R7.0 (7.0.0.0.700007)
Avaya Aura® Communication Manager installed on VMWare	R7.0 (vcm-07.00.0.441.0-22903)
Avaya Aura® Media Server installed on VMWare	R7.7 (v.7.7.0.226)
Avaya Aura® Communication Manager Messaging installed on VMWare	R7.0 (vcmm-07.00.0.441.0)
Avaya 96x1 IP Deskphone (H323)	R6.2.2313
Avaya 96x0 IP Deskphone (SIP)	R2.6.9.1
QSC Q-Sys Designer	5.1.53
QSC Q-Sys Core 110f	5.1.53
QSC Q-Sys Softphone	5.1.53

## 5. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Locations that can be occupied by SIP Entities
- SIP Entities and corresponding Entity Links between Session Manager and Communication Manager/Communication Manager Messaging
- Define Communication Manager as Administrable Entity (i.e., Managed Element).
- Application Sequence
- Add 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.

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

### 5.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. To add a location, navigate to **Home→Elements→Routing→Domains** and click the **New** (not shown) button on the right.

The following screen will then be shown. Fill in the following:

- **Name:** The authoritative domain name (e.g., *avaya.com*)
- **Type:** Set to *sip* (default)
- **Notes:** Descriptive text (optional)

Click **Commit**.

The screenshot shows the Avaya Aura System Manager 7.0 web interface. The top navigation bar includes 'Home' and 'Routing'. The left sidebar shows a tree view with 'Routing' expanded, containing 'Domains', 'Locations', 'Adaptations', 'SIP Entities', 'Entity Links', and 'Time Ranges'. The main content area is titled 'Domain Management' and shows a breadcrumb path 'Home / Elements / Routing / Domains'. There are 'Commit' and 'Cancel' buttons in the top right. Below the title, it says '1 Item' with a refresh icon. A table displays the domain configuration:

Name	Type	Notes
*avaya.com	sip	Used for Devconnect Testing

## 5.2. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, navigate to **Home→Elements→Routing→Locations** and click on the **New** (not shown) button on the right. The following screen will then be shown. Fill in the following:

Under **General**:

- **Name:** A descriptive name
- **Notes:** Descriptive text (optional)

Under **Location Pattern**:

- **IP Address Pattern:** A pattern used to logically identify the location
- **Notes:** Descriptive text (optional)

The screen below shows addition of the **Location\_102** used for Communication Manager and other entities. Similarly a location was defined for Session Manager. Click **Commit** to save the Location definition.

The screenshot displays the 'Location Details' configuration page in the Avaya Aura System Manager 7.0 interface. The page is divided into several sections:

- General:** Includes fields for **Name** (set to 'Location\_102') and **Notes** (set to 'Entities in Subnet 102').
- Dial Plan Transparency in Survivable Mode:** Includes an **Enabled** checkbox (unchecked), **Listed Directory Number**, and **Associated CM SIP Entity** fields.
- Overall Managed Bandwidth:** Includes **Managed Bandwidth Units** (set to 'Kbit/sec'), **Total Bandwidth**, **Multimedia Bandwidth**, and an **Audio Calls Can Take Multimedia Bandwidth** checkbox (checked).
- Per-Call Bandwidth Parameters:** Includes **Maximum Multimedia Bandwidth (Intra-Location)** (2000 Kbit/Sec), **Maximum Multimedia Bandwidth (Inter-Location)** (2000 Kbit/Sec), **Minimum Multimedia Bandwidth** (64 Kbit/Sec), and **Default Audio Bandwidth** (80 Kbit/Sec).
- Alarm Threshold:** Includes **Overall Alarm Threshold** (80 %), **Multimedia Alarm Threshold** (80 %), **Latency before Overall Alarm Trigger** (5 Minutes), and **Latency before Multimedia Alarm Trigger** (5 Minutes).
- Location Pattern:** A table with one item: **IP Address Pattern** with the value **\*10.64.102.\***.

The page also features a **Commit** button and a **Cancel** button in the top right corner.



## 5.3. Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager and Communication Manager. The screens below also show the corresponding Entity Links.

### 5.3.1. Session Manager Entity

To add a SIP Entity, navigate to **Home→Elements→Routing→SIP Entities**, and click on **New** (not shown) and configure as follows:

Under **General**:

- **Name:** Any descriptive name
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager
- **Type:** Select *Session Manager*
- **Location:** Select one of the locations defined previously
- **Time Zone:** Time zone for this location

Under **Listen Ports**, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Listen Ports:** Port number on which the system listens for SIP requests
- **Protocol:** Transport protocol to be used to send SIP requests
- **Default Domain:** The domain used for the enterprise (e.g. *avaya.com*)

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.

**Avaya System Manager 7.1.1**

**SIP Entity Details**

**General**

Name:

FQDN or IP Address:

Type:

Location:

Outbound Proxy:

Time Zone:

Credential name:

SIP Link Monitoring:

**Entity Links**

Name	SIP Entity 1	Protocol	SIP Entity 2	Port	Connection Policy	Ready New Service
+SMN, ON Messaging	SMN	TCP	ON Messaging	*5060	Trusted	<input type="checkbox"/>
+SMN, ON Voice Mail	SMN	TCP	ON Voice Mail	*5060	Trusted	<input type="checkbox"/>

Select: all, none

**Listen Ports**

TCP Follower port:

TLS Follower port:

Name	Protocol	Default Domain	Notes
5060	TCP	avaya.com	Default TCP port

### 5.3.2. Communication Manager Entity

The following screen displays the Communication Manager entity configured for this reference configuration.

**AVAYA**  
Aura System Manager 7.0

Home / Elements / Routing / SIP Entities

#### SIP Entity Details

Commit Cancel

**General**

\* Name: CM70Proct

\* FQDN or IP Address: 10.64.102.150

Type: CH

Notes: CM 7.0 Proc, Ethernet

Adaptation: [v]

Location: location\_102

Time Zone: America/Denver

\* SIP Timer B/F (in seconds): 4

Credential name: [v]

Securable: [v]

Call Detail Recording: none

**Loop Detection**

Loop Detection Mode: On

Loop Count Threshold: 5

Loop Detection Interval (in msec): 200

**SIP Link Monitoring**

SIP Link Monitoring: Use Session Manager Configuration

Supports Call Admission Control: [v]

Shared Bandwidth Manager: [v]

Primary Session Manager Bandwidth Association: [v]

Backup Session Manager Bandwidth Association: [v]

**Entity Links**

Override Port & Transport with DNS SRV: [v]

Add Remove

1 Item

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
* SM70_CH70Proct_50	SM70	TCP	* 5050	CM70Proct	* 5060	trusted	[v]

### 5.3.3. Communication Manager Messaging Entity

The following screen displays the Communication Manager Messaging entity configured for this reference configuration.

The screenshot displays the Avaya System Manager 7.2 interface. The left sidebar shows a navigation menu with options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Mail Patterns, Regular Expressions, and Defaults. The main content area is titled "SIP Entity Details" and includes a "General" tab. The configuration fields are as follows:

- Name:** CM Messaging
- FQDN or IP Address:** 10.64.102.151
- Type:** Protocol Messaging
- Notes:** CM Messaging
- Adaptation:** (dropdown menu)
- Location:** Location\_002
- Time Zone:** America/Denver
- SIP Timer R/T (in seconds):** 5
- Credential name:** (text field)
- Securable:** ☐
- Call Detail Recording:** agree
- Loop Detection Mode:** On
- Loop Count Threshold:** 5
- Loop Detection Interval (in msec):** 200
- SIP Link Monitoring:** Use Session Manager Configuration
- Supports Call Admission Control:** ☐
- Shared Bandwidth Manager:** ☐
- Primary Session Manager Bandwidth Association:** (dropdown menu)
- Backup Session Manager Bandwidth Association:** (dropdown menu)
- Override Port & Transport with DNS SRV:** ☐

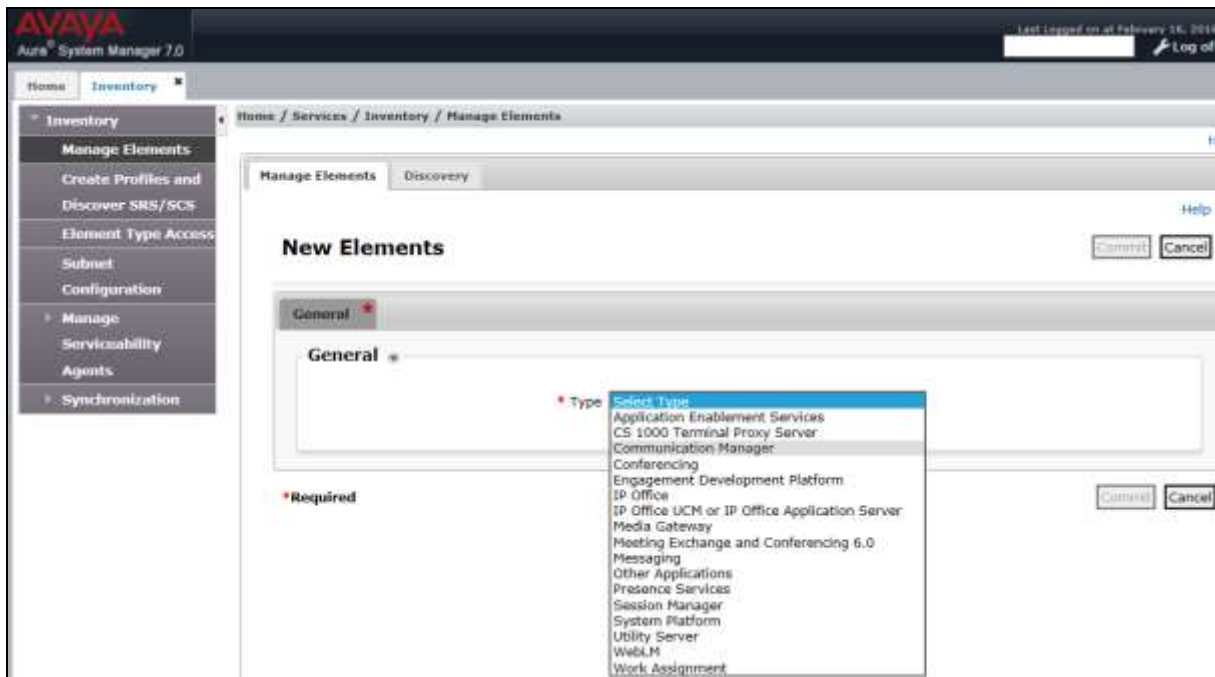
At the bottom, there is a table for "Entity Links" with columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Connection Policy, and Display Name. The table contains one entry:

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Display Name
* SIP CM Messaging	SIP CM	SIP	* 5060	CM Messaging	* 5060	Trusted	

## 5.4. Define Communication Manager as a Managed Element

Before adding SIP users, Communication Manager must be added to System Manager as a managed element. This action allows System Manager to access Communication Manager over its administration interface. Using this administration interface, System Manager will notify Communication Manager when new SIP users are added.

To define Communication Manager as a managed element, navigate to **Home→Services→Inventory→Manage Elements** on the left and click on the **New** (not shown) button on the right. In the **Type** field that is displayed, select *Communication Manager*.



In the **Add Communication Manager** screen, fill in the following fields as follows:

Under *General Attributes*:

- **Name:** Enter an identifier for Communication Manager
- **Hostname or IP Address:** Enter the IP address of the administration interface for Communication Manager
- **Login:** Enter login used for administration access  
Communication Manager instance
- **Authentication Type:** Select the **Password** button
- **Password:** Enter a valid password
- **Confirm Password:** This should match the password entered in the **Password** field above

Click **Commit** to save.

The screenshot displays the 'Add Communication Manager' configuration page in the Avaya Aura System Manager 7.0. The left sidebar contains a navigation tree with 'Inventory' expanded, showing 'Manage Elements' and 'Discovery' sub-items. The main panel has a breadcrumb trail 'Home / Services / Inventory / Manage Elements' and a 'Help' link. The 'Add Communication Manager' form is divided into two tabs: 'General Attributes (0)' and 'SNMP Attributes (5)'. The 'General Attributes' tab is active, showing fields for Name (CM70), Hostname or IP Address (10.64.102.150), Login, Authentication Type (Password selected), Password (masked), Confirm Password (masked), SSH Connection (checked), RSA SSH Fingerprint (Primary IP), and RSA SSH Fingerprint (Alternate IP). On the right side of the form, there are fields for Description (Communication Manager), Alternate IP Address, Enable Notifications (unchecked), Port (5022), Location, and Add to Communication Manager (checked). At the bottom right, there are 'Commit', 'Clear', and 'Cancel' buttons. The top of the interface shows the Avaya logo and 'Aura System Manager 7.0'.

## 5.5. Add Application Sequence

Navigate to **Home**→**Elements**→**Session Manager**→**Application Configuration**→**Applications** and configure as follows:

- **Name:** Enter any descriptive name
- **SIP Entity:** Select the Communication Manager SIP Entity configured in **Section 5.3.2**
- **CM System for SIP Entity:** Select the system configured in **Section 5.4**

Click **Commit** to save the application configuration.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The left sidebar contains a navigation menu with options like Session Manager, Dashboard, Session Manager Administration, Communication Profile Editor, Network Configuration, and Device and Location Configuration. The main content area is titled 'Application Editor' and contains a form for creating a new application. The form fields are: Name (CM70), SIP Entity (CM70Procr), CM System for SIP Entity (CM70), and Description (CM 7.0). There are 'Commit' and 'Cancel' buttons at the top right of the form area.

Next, define the **Application Sequence** for Communication Manager as shown below.

The screenshot shows the Avaya Aura System Manager 7.0 interface for the 'Application Sequence Editor'. The left sidebar is the same as the previous screenshot. The main content area is titled 'Application Sequence Editor' and contains a form for creating a new application sequence. The form fields are: Name (CM70Sequencing) and Description (App. Sequencing with CM 7.0). Below the form is a table titled 'Applications in this Sequence' which contains one item: CM70, CM70Procr, Mandatory, and CM 7.0. There are 'Commit' and 'Cancel' buttons at the top right of the form area.

Sequence Order (first to last)	Name	SIP Entity	Mandatory	Description
<input type="checkbox"/>	CM70	CM70Procr	<input checked="" type="checkbox"/>	CM 7.0

## 5.6. 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, as verified in **Section 6.3**.

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., **50071@avaya.com**)
- **Password:** Enter the password used to register with System Manager
- **Confirm Password:** Re-enter the password from above

The screenshot shows the Avaya Aura System Manager 7.0 interface. The breadcrumb navigation is Home / Users / User Management / Manage Users. The page title is "User Profile Edit: 50071@avaya.com". The left sidebar contains a menu with "User Management" expanded, showing "Manage Users", "Public Contacts", "Shared Addresses", "System Presence", "ACLs", "Communication", "Profile Password", and "Policy". The main content area has tabs for "Identity", "Communication Profile", "Membership", and "Contacts". The "Identity" tab is active, showing the "User Provisioning Rule" dropdown set to "User Provisioning Rule". Below this, the "Identity" section contains the following fields: "Last Name" (SIP), "Last Name (Latin Translation)" (SIP), "First Name" (50071), "First Name (Latin Translation)" (50071), "Middle Name" (empty), "Description" (QSC SIP Softphone), "Update Time" (April 7, 2016 11:29:13 A), "Login Name" (50071@avaya.com), "Authentication Type" (Basic), "Change Password" link, "Source" (local), "Localized Display Name" (SIP\_50071), "Endpoint Display Name" (SIP\_50071), "Title" (empty), "Language Preference" (English (United States)), and "Time Zone" (-6:0 Mountain Time (US & Can...)).

Click the **Communication Profile** tab and enter values for the following required fields:

- **Communication Profile Password:** Enter a valid password.
- **Confirm Password:** Make sure that it matches the password entered above

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 **Add**.

The screenshot displays the 'New User Profile' configuration window in Avaya System Manager 7.0. The interface includes a left-hand navigation pane with options like 'Manage Users', 'Public Conversations', and 'System Parameters'. The main area is titled 'New User Profile' and contains three tabs: 'Identity', 'Communication Profile', and 'Address'. The 'Communication Profile' tab is currently selected, showing input fields for 'Communication Profile Password' and 'Confirm Password'. Below these, there is a 'Name' field with a dropdown menu set to 'Primary'. The 'Address' tab is also visible, showing a 'Type' dropdown set to 'Avaya SIP' and a 'Fully Qualified Address' field with the value '5001@10.10.10.10'. The form includes 'Add' and 'Cancel' buttons at the bottom right.



In the **Session Manager Profile** section, specify the Session Manager entity configured in **Section 5.3.1** and assign the **Application Sequence** defined in **Section 5.5** to both the **Originating Sequence** and **Termination Sequence** fields. Additionally, set **Home Location** field to *Session Manager* configured in **Section 5.2**.

☒ **Session Manager Profile**

### SIP Registration

\* Primary Session Manager

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices

Block New Registration When Maximum Registrations Active?
☒

Primary	Secondary	Maximum
12	0	12

### Application Sequences

Origination Sequence

Termination Sequence

### Call Routing Settings

\* Home Location

Conference Factory Set

### Call History Settings

Enable Centralized Call History?
☐

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

- **System:** Select the managed element corresponding to Communication Manager in **Section 5.4**
- **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 template for type of SIP phone which is set to **9621SIP\_DEFAULT\_CM\_7\_0** for Q-Sys Softphone

Click **Commit** (not shown).

☒ **CM Endpoint Profile**

\* System

CM70

\* Profile Type

Endpoint

Use Existing Endpoints

☐

\* Extension

50071

Endpoint Editor

\* Template

9621SIP\_DEFAULT\_CM\_7\_0

Set Type

9621SIP

Security Code

Port

IP

Voice Mail Number

Preferred Handle

50071@avaya.com

Calculate Route Pattern

☐

Sip Trunk

aar

Enhanced Callr-Info display for 1-line phones

☐

Delete Endpoint on Unassign of Endpoint from User or on Delete User

☒

Override Endpoint Name and Localized Name

☒

Allow H.323 and SIP Endpoint Dual Registration

☐

## 6. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring Q-Sys Softphone as an Off-PBX Station (OPS) and configuring a SIP trunk between Communication Manager and Session Manager. Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials. Note that the fields modified in this section are for this reference configuration only; defaults are used for all other fields.

### 6.1. Verify OPS and SIP Trunk Capacity

Using the SAT, verify that the Off-PBX Telephones (OPS) and SIP Trunks features are enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative. On **Page 1**, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

display system-parameters customer-options		Page	1 of 11
OPTIONAL FEATURES			
G3 Version: V16	Software Package: Enterprise		
Location: 2	System ID (SID): 1		
Platform: 28	Module ID (MID): 1		
		USED	
	Platform Maximum Ports: 6400	25	
	Maximum Stations: 2400	10	
	Maximum XMOBILE Stations: 2400	0	
	Maximum Off-PBX Telephones - EC500: 9600	0	
	<b>Maximum Off-PBX Telephones - OPS: 9600</b>	<b>5</b>	
	Maximum Off-PBX Telephones - PBFMC: 9600	0	
	Maximum Off-PBX Telephones - PVFMC: 9600	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 SIP trunks supported by the system is sufficient.

display system-parameters customer-options	Page	2 of 11
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	4000	0
Maximum Concurrently Registered IP Stations:	2400	2
Maximum Administered Remote Office Trunks:	4000	0
Maximum Concurrently Registered Remote Office Stations:	2400	0
Maximum Concurrently Registered IP eCons:	68	0
Max Concur Registered Unauthenticated H.323 Stations:	100	0
Maximum Video Capable Stations:	2400	0
Maximum Video Capable IP Softphones:	2400	0
<b>Maximum Administered SIP Trunks:</b>	<b>4000</b>	<b>160</b>
Maximum Administered Ad-hoc Video Conferencing Ports:	4000	0
Maximum Number of DS1 Boards with Echo Cancellation:	80	0
Maximum TN2501 VAL Boards:	10	0
Maximum Media Gateway VAL Sources:	50	0
Maximum TN2602 Boards with 80 VoIP Channels:	128	0
Maximum TN2602 Boards with 320 VoIP Channels:	128	0
Maximum Number of Expanded Meet-me Conference Ports:	300	0
(NOTE: You must logoff & login to effect the permission changes.)		

## 6.2. Configure SIP Trunk

In the **IP Node Names** form, assign an IP address and host name for Session Manager (**ASM70**), Communication Manager Messaging (**CMM70**) and Media Server (**AMS70**). The host names will be used throughout the other configuration screens of Communication Manager.

change node-names ip	Page	1 of 2
IP NODE NAMES		
Name	IP Address	
default	0.0.0.0	
<b>ASM70</b>	<b>10.64.102.157</b>	
<b>CMM70</b>	<b>10.64.102.151</b>	
<b>AMS70</b>	<b>10.64.102.158</b>	
<b>procr</b>	<b>10.64.102.150</b>	
procr6	::	

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is **avaya.com**. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Media Gateway or Media Server. The **IP Network Region** form also specifies the **Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region 2) is specified in the SIP signaling group.

```

change ip-network-region 2                                     Page 1 of 20

                                IP NETWORK REGION

Region: 2
Location: 1           Authoritative Domain: avaya.com
Name: Main Network Region
MEDIA PARAMETERS                      Intra-region IP-IP Direct Audio: yes
    Codec Set: 2                      Inter-region IP-IP Direct Audio: yes
    UDP Port Min: 2048                  IP Audio Hairpinning? n
    UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
    Call Control PHB Value: 46
    Audio PHB Value: 46
    Video PHB Value: 26
802.1P/Q PARAMETERS
    Call Control 802.1p Priority: 6
    Audio 802.1p Priority: 6
    Video 802.1p Priority: 5          AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                      RSVP Enabled? n
H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5

```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to Q-Sys Softphone. The form is accessed via the **change ip-codec-set 2** command. Note that IP codec set 2 was specified in IP Network Region 2 shown above. The following form shows the list of codecs tested. The order of these codecs was changed to support the some of the codecs for reasons listed in **Section 2.2**.

```

change ip-codec-set 2                                     Page 1 of 2

                                IP Codec Set

Codec Set: 2

Audio      Silence      Frames      Packet
Codec      Suppression  Per Pkt    Size (ms)
1: G.711MU      n           2          20
2: G.711A      n           2          20
3: G.722-64K    2          20
4:
5:
6:
7:

```

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the Signaling Group form as follows:

- **Group Type:** Set to *sip*
- **Transport Method:** Set to *tcp*
- **Near-end Node Name:** Set to *procr* node configured in this section
- **Far-end Node Name:** Set to *ASM70* node configured in this section
- **Far-end Network Region:** Set to network region configured in this section
- **Far-end Domain:** Set to *avaya.com* to match the Session Manager domain configured in **Section 5.1**
- Verify **Direct IP-IP Audio Connections** field is set to *y* for shuffling
- Verify **DTMF over IP** field is set to the default value of *rtp-payload* indicating DTMF transmission using RFC 2833

add signaling-group 2		Page 1 of 1
SIGNALING GROUP		
Group Number: 2	Group Type: sip	
IMS Enabled? n	Transport Method: tcp	
Q-SIP? n		
IP Video? n	Enforce SIPS URI for SRTP? y	
Peer Detection Enabled? y	Peer Server: SM	
Near-end Node Name: procr	Far-end Node Name: ASM70	
Near-end Listen Port: 5060	Far-end Listen Port: 5060	
	Far-end Network Region: 2	
	Far-end Secondary Node Name:	
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? y	
Session Establishment Timer(min): 3	IP Audio Hairpinning? n	
Enable Layer 3 Test? y	Initial IP-IP Direct Media? n	
H.323 Station Outgoing Direct Media? n	Alternate Route Timer(sec): 6	

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to the SIP Phones. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

add trunk-group 2		Page 1 of 21	
TRUNK GROUP			
Group Number: 2	<b>Group Type: sip</b>	CDR Reports: y	
<b>Group Name: SIP Endpoints/CM Messaging</b>	COR: 1	TN: 1	TAC: 102
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n			
Queue Length: 0			
<b>Service Type: tie</b>	Auth Code? n		
	Member Assignment Method: auto		
	<b>Signaling Group: 2</b>		
	<b>Number of Members: 15</b>		

On **Page 3** of the **Trunk Group** form, set the **Numbering Format** field to *private*. This field specifies the format of the calling party number sent to the far-end.

add trunk-group 2		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
<b>Numbering Format: private</b>			
UI Treatment: service-provider			
Replace Restricted Numbers? n			
Replace Unavailable Numbers? n			
Modify Tandem Calling Number: no			
Show ANSWERED BY on Display? y			
DSN Term? n			

Configure the **Private Numbering Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with **5** and whose calls are routed over any trunk group, including SIP trunk group **2**, have the number sent to the far-end for display purposes.

change private-numbering 0		Page 1 of 2	
NUMBERING - PRIVATE FORMAT			
Ext Len	Ext Code	Trk Grp(s)	Private Prefix
5	33	10	5
5	58	10	5
<b>5</b>	<b>5</b>	<b>2</b>	<b>5</b>
5	600	10	5
			Total Administered: 4
			Maximum Entries: 540

### 6.3. Configure Signaling Group for Avaya Aura® Media Server

Another signaling group was created between Communication Manager and Media Server to provide media resources for IP telephony in parallel with Media Gateway G650 resource.

Following signaling group was created for this reference configuration:

- **Group Type:** Set to *sip*
- **Transport Method:** Set to *tcp*
- **Peer Detection Enable:** Set to *n*
- **Peer Server:** Set to *AMS*
- **Near-end Node Name:** Set to *procr* node shown in **Section 6.2**
- **Far-end Node Name:** Set to *AMS70* node configured in **Section 6.2**
- **Far-end Network Region:** Set to network region configured in **Section 6.2**

```
add signaling-group 3                                     Page 1 of 1
                                     SIGNALING GROUP
Group Number: 3                Group Type: sip
                               Transport Method: tcp
Peer Detection Enabled? n Peer Server: AMS
Near-end Node Name: procr      Far-end Node Name: AMS70
Near-end Listen Port: 5060     Far-end Listen Port: 5060
                               Far-end Network Region: 2
Far-end Domain: 10.64.102.158
```

### 6.4. Verify SIP Stations

Use the **display station** command to view each Q-Sys Softphone SIP station configured in **Section 5.6**.

```
display station 50071                                     Page 1 of 6
                                     STATION
Extension: 50071                Lock Messages? n          BCC: 0
Type: 9621SIP                  Security Code:             TN: 1
Port: S00003                   Coverage Path 1: 1          COR: 1
Name: 50071 SIP                Coverage Path 2:          COS: 1
                               Hunt-to Station:
STATION OPTIONS
                               Time of Day Lock Table:
Loss Group: 19                 Message Lamp Ext: 40012
Display Language: english
Survivable COR: internal
Survivable Trunk Dest? y       IP SoftPhone? n
                               IP Video? n
```



Use the **display off-pbx-telephone station-mapping** to verify proper entry of Q-Sys Softphone SIP station in Communication Manager.

display off-pbx-telephone station-mapping 50071							Page	1	of	3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION										
Station	Application	Dial	CC	Phone	Trunk	Config	Dual			
Extension		Prefix		Number	Selection	Set	Mode			
50071	OPS	-		50071	aar	1				

On **Page 2**, verify that the **Call Limit** matches the number of *call-appr* entries in the station form.

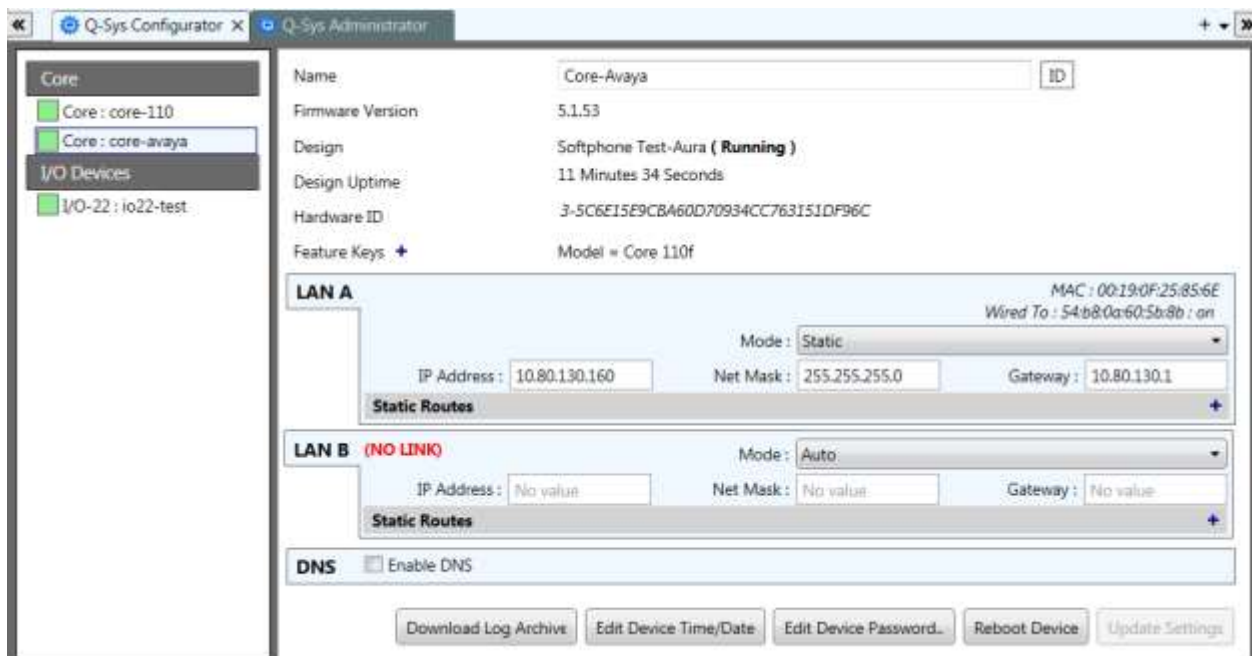
display off-pbx-telephone station-mapping 50071							Page	2	of	3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION										
Station	Appl	Call	Mapping	Calls	Bridged	Location				
Extension	Name	Limit	Mode	Allowed	Calls					
50071	OPS	3	both	all	none					

## 7. Configure QSC Q-Sys Core and Q-Sys Softphone

On an installed PC start, Q-Sys Designer software (not shown). Select **Tools** → **Show Q-Sys Configurator**.



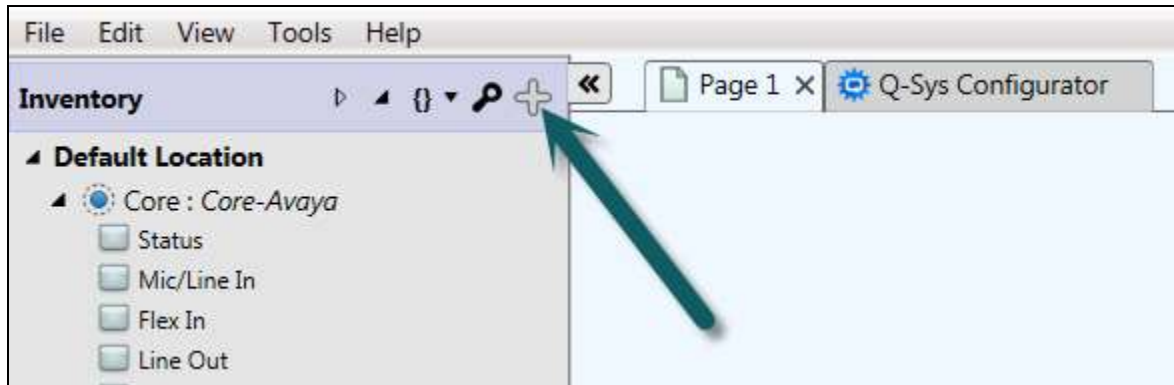
Select the Core model being configured for Softphone usage from the list of discovered Cores in the left column of the Q-Sys Configurator. Set the appropriate network settings on the network interface used to connect to Session Manager. **LAN A** is often used for connection to Q-Sys peripherals, so the AUX port 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 Session Manager. For this testing, **LAN A** was chosen and configured as shown below. After configuring the network settings, click the **Update Settings** button.



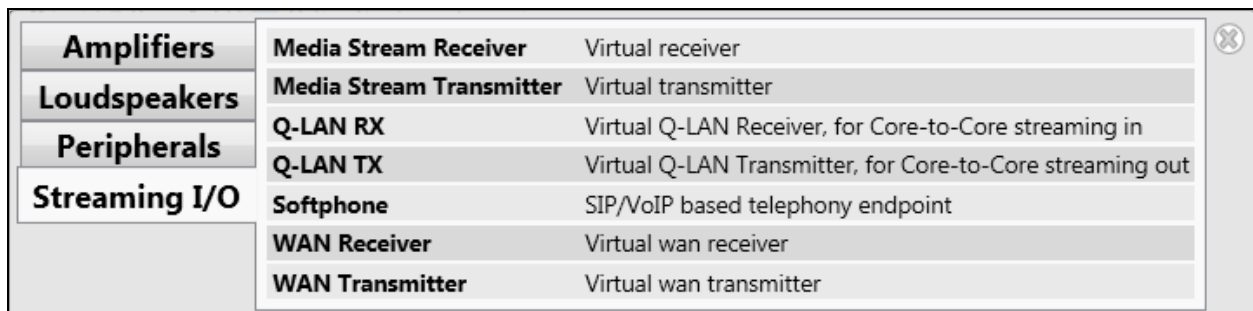
Copy the Core name from the Configurator to the Core Property name and choose the proper model.



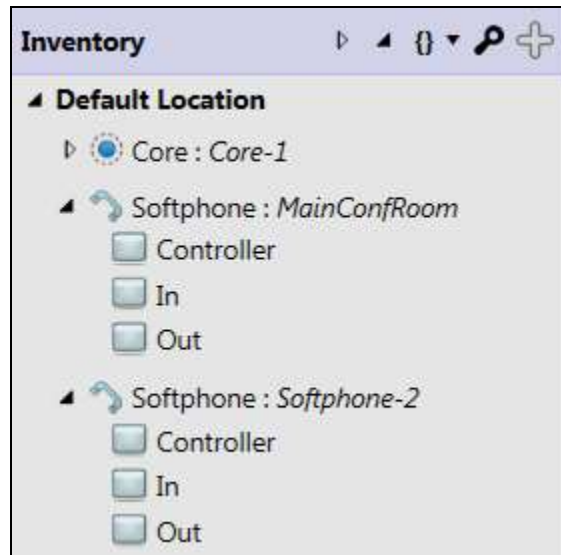
In the far left column of Q-Sys Designer, click on plus sign to add a Softphone instance to the current design file.



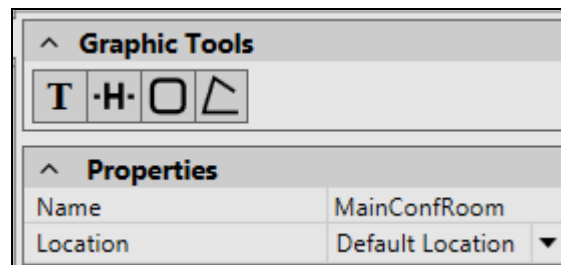
The **Add Inventory Item** menu will appear to the right of the plus button showing all of the items available to add to the design. Navigate to and click **Streaming I/O → Softphone** to insert a Softphone instance to the Inventory list. The **Add Inventory Menu** disappears. Follow the same steps to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




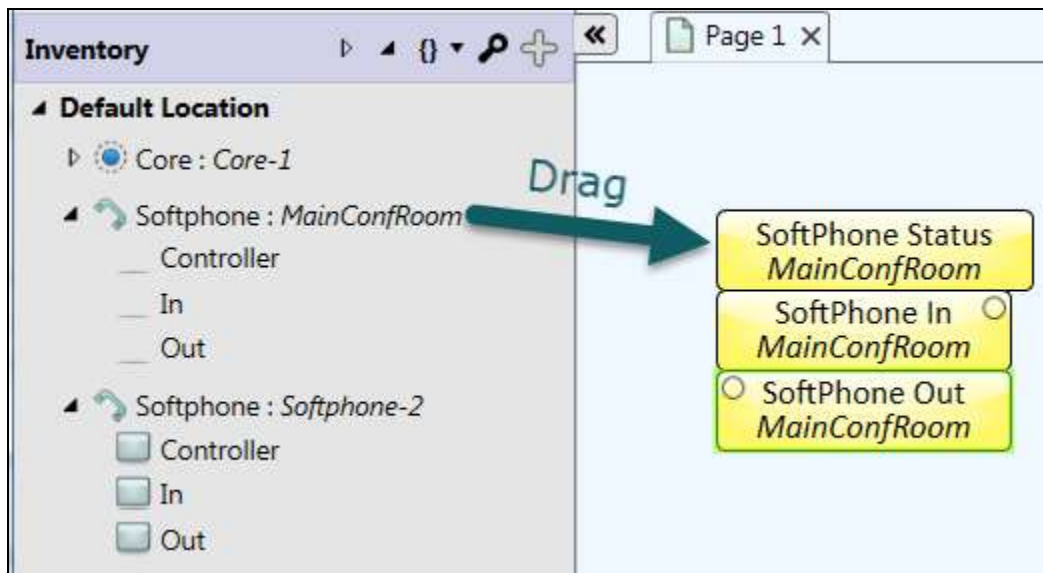
This screenshot shows two softphones were added to the Inventory list of the design file.



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:

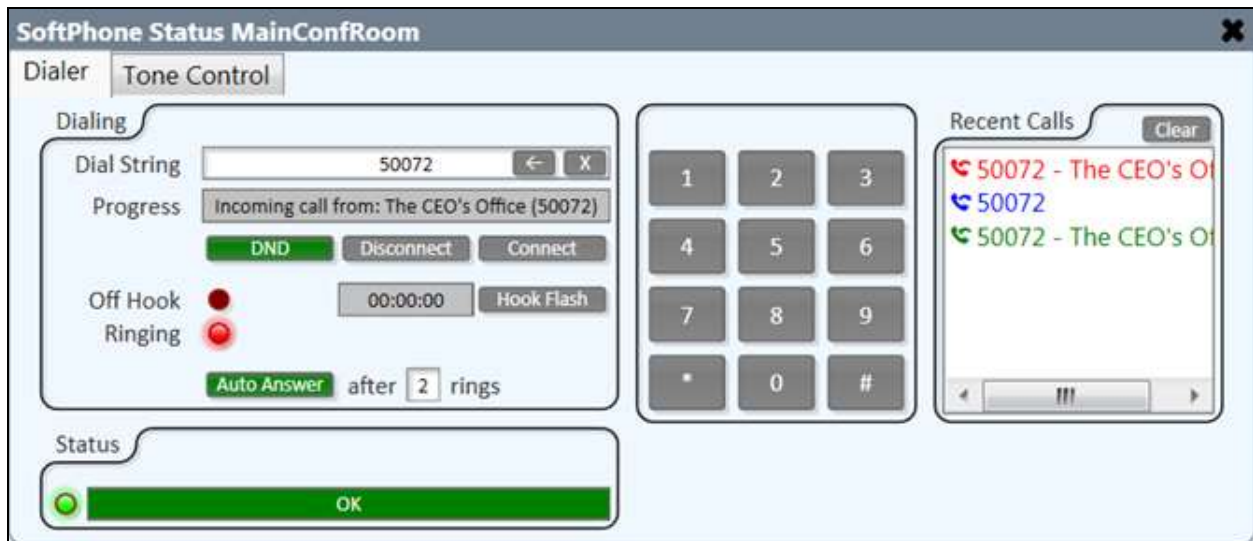


From the Inventory list, drag the individual components of the Softphone into the work area by clicking on the  icon. Alternately, click on the Softphone name and drag into the work area to place all three sub-components into the design at one time (as shown below).



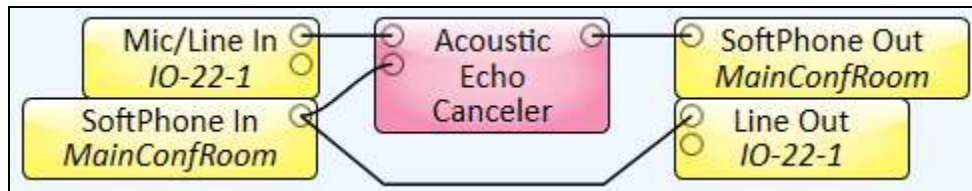
The three sub-components each Softphone instance provides are the following:

- **Softphone Status** – 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.
- **In** – brings the incoming telephony audio into the Q-Sys design for routing as an audio source (Audio from Session Manager)
- **Out** – receives the audio from the Q-Sys design which is intended to be sent as telephony audio (Audio to Session Manager).

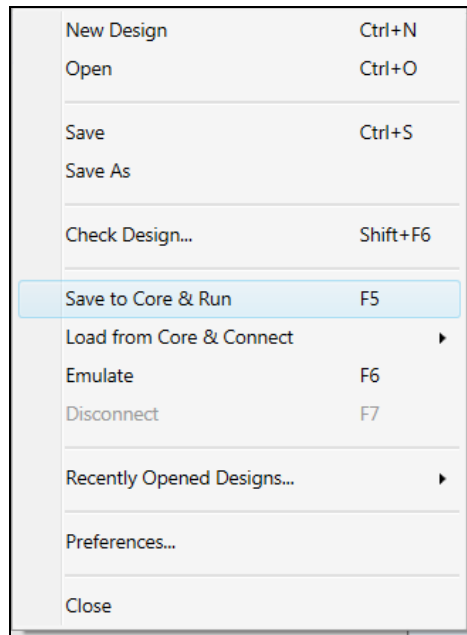


Add components to the design by dragging, then “wire” them similarly to the softphone as shown below. To draw a wire between components, click within an audio pin circle and then drag the mouse and release the mouse button over another appropriate pin (outputs to inputs).


An **Acoustic Echo Cancellor** 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.

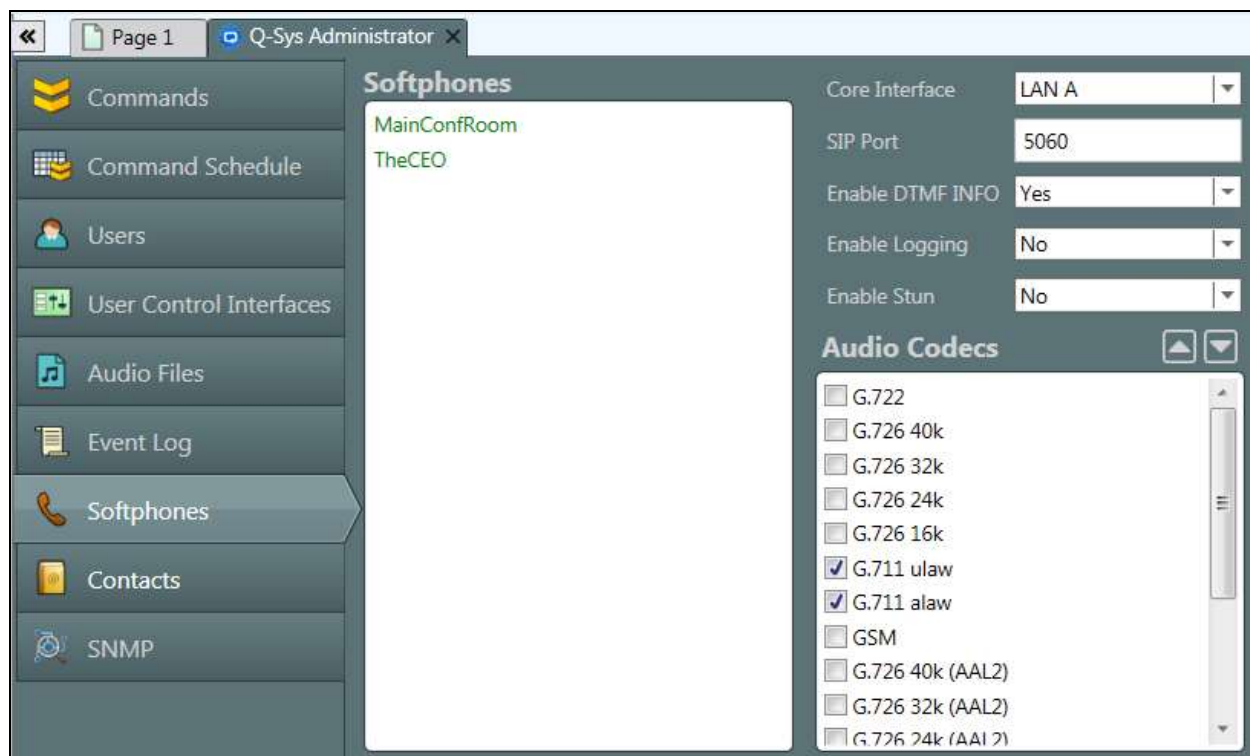


To “Deploy” the design to the Q-Sys Core press **F5** or **File → Save to Core & Run**



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

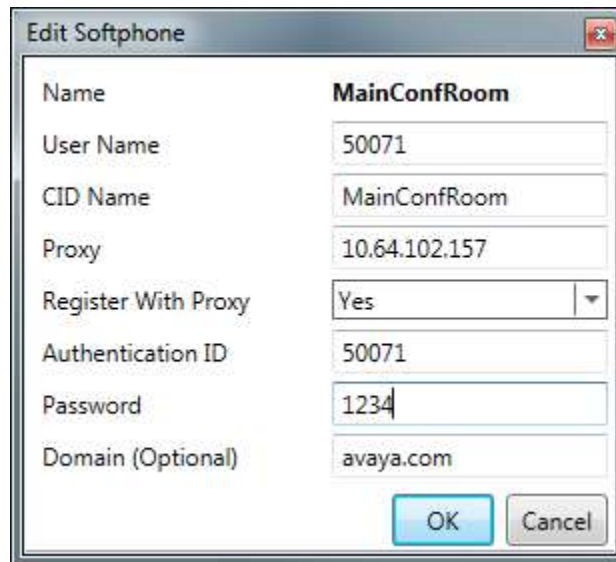
Manager. Navigate to **Tools → Q-Sys Administrator** or use the  button which appears in the upper right corner of the user interface. Once the Q-Sys Administrator appears, select the **Softphones** tab. **Enable DTMF INFO** field should be set to **Yes** (See **Section 2.2** for further discussion on this issue)



Deselect all Audio Codecs except for the two G.711 codecs. When any changes are made within the Administrator, a red bar will appear to indicate that the settings will need to be confirmed to become active. Wait until all Administrator steps are completed prior to clicking the **Update** button



Double-click an entry in the Softphones list to be configured. When the **Edit Softphone** dialog box appears, enter the softphone registration details similarly to those shown below, which correspond to the details configured within Session Manager. The User Name field will be the extension number chosen in **Section 5.4**. The **CID name** is what the Softphone will report as the display name. The **Proxy** corresponds to the **IP Address** in **Section 5.2** and **Authentication ID** corresponds to the **Name** configured in **Section 5.5**. Password is same as configured in **Section 5.5**. The **Domain (Optional)** can be configured to report the organization's domain as part of the URI.

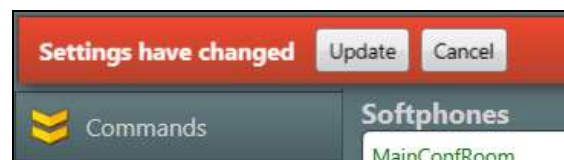


The 'Edit Softphone' dialog box contains the following fields and values:

Field	Value
Name	MainConfRoom
User Name	50071
CID Name	MainConfRoom
Proxy	10.64.102.157
Register With Proxy	Yes
Authentication ID	50071
Password	1234
Domain (Optional)	avaya.com

Buttons: OK, Cancel

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



The notification bar shows the following elements:

- Red bar with text: **Settings have changed**
- Buttons: **Update**, **Cancel**
- Left sidebar: **Commands** (with a yellow icon)
- Right sidebar: **Softphones** (with a green icon), **MainConfRoom**

## 8. 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:

Home

Session Manager

Session Manager

Dashboard

Session Manager Administration

Communication Profile Editor

Network Configuration

Device and Location Configuration

Application Configuration

System Status

SIP Entity Monitoring

Managed

Bandwidth Usage

Security Module Status

Home / Elements / Session Manager / System Status / User Registrations

Help

User Registrations

Select rows to send notifications to devices. Click on Details column for complete registration status.

Customize

View ▾ Default Force Unregister AST Device Notifications Reboot Reload ▾ Fallback As of 10:22 AM

Advanced Search

10 Items Show All ▾

Filter: enable

<input type="checkbox"/>	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered From	Stat
<input type="checkbox"/>	Show	90073@avaya.com	50056	SIP	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	90073@avaya.com	50073	SIP	---	10.80.120.140	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	50057	SIP	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	90050@avaya.com	50056	SIP	---	10.80.120.48	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>
<input type="checkbox"/>	Show	90055@avaya.com	50055	SIP	---	10.80.120.44	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	50073	SIP	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	50074	SIP	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	50053	SIP	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	50054	SIP	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	50071@avaya.com	50071	SIP	---	10.80.120.140	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	(AC)	<input type="checkbox"/>

- Verify that basic calls can be made from and to Q-Sys Softphone.

## 9. 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**.

## 10. Additional References

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

- [1] [Deploying Avaya Aura® System Manager, Release 7.0, January 2015](#)
- [2] [Administering Avaya Aura® System Manager, Release 7.0, January 2016](#)
- [3] [Deploying Avaya Aura® Session Manager on VMWare, Release 7.0, August 2015](#)
- [4] [Administering Avaya Aura® Session Manager, Release 7.0, August 2015](#)
- [5] [Deploying Avaya Aura® Communication Manager in Virtualized Environment, Release 7.0, August 2015](#)
- [6] [Deploying and Updating Avaya Aura® Media Server Appliance, Release 7.7, October 2015](#)
- [7] [Implementing and Administering Avaya Aura® Media Server, Release 7.7, January 2016](#)
- [8] [Deploying Avaya Aura® Communication Manager Messaging, Release 7.0, September 2015](#)

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