



Avaya Solution & Interoperability Test Lab

Application Notes for Mobile Heartbeat MH-CURE with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.0

Abstract

These Application Notes describe the configuration of Mobile Heartbeat MH-CURE 20.5 with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1. MH-CURE connects and mobilizes clinicians and operational staff across the healthcare enterprise. MH-CURE client runs on a smartphone and connects to an MH-CURE application server for its SIP configuration. The MH-CURE client application then registers directly with Avaya Aura® Session Manager as a SIP endpoint through an enterprise wireless LAN.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the 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 of Mobile Heartbeat MH-CURE 20.5.2 with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1. MH-CURE connects and mobilizes clinicians and operational staff across the healthcare enterprise. The MH-CURE client runs on a smartphone and connects to an MH-CURE application server for its SIP configuration. The MH-CURE client application then registers directly with Avaya Aura® Session Manager as a SIP endpoint through an enterprise wireless LAN.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between the MH-CURE clients and Avaya H.323 / SIP Deskphones and the PSTN, and exercising telephony features, such as Hold/Resume, Mute/Unmute, Call Coverage, Call Transfer, and Conference. Additional telephony features, such as Call Forward, Call Pickup, and Call Park/Unpark, were also verified using Communication Manager Feature Access Codes (FACs).

The serviceability testing focused on verifying that MH-CURE clients came back into service after re-establishing IP network connectivity and rebooting the MH-CURE clients and application server.

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 MH-CURE did not include use of any specific encryption features as requested by Mobile Heartbeat.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of MH-CURE clients with Session Manager.
- Calls between MH-CURE clients and Avaya H.323 / SIP Deskphones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between MH-CURE clients and the PSTN.
- G.711 codec support.
- Proper recognition of DTMF tones.
- Basic telephony features, including Hold, Mute, Redial, multiple calls, Blind and Supervised Transfer, and Attended Conference.
- Extended telephony features using Communication Manager FACs for Call Forward, Call Park/Unpark, and Call Pickup.
- Voicemail coverage, MWI support, and logging into voicemail system to retrieve voice messages.
- Proper system recovery after a restart of MH-CURE clients and loss of wireless network connectivity.

2.2. Test Results

All test cases passed with the following observation:

- If an outgoing call from an MH-CURE client doesn't complete for any reason (e.g., invalid number, busy, or call blocked), related tones are not played; the call simply disappears on the client.
- MH-CURE does not support the Communication Manager Long Hold Recall Timer feature. No recall notification is made.

2.3. Support

For MH-CURE technical support, contact Mobile Heartbeat technical support via phone or website.

- **Phone:** +1 (781) 238-0000
- **Web:** <https://www.mobileheartbeat.com/contact-us/>

3. Reference Configuration

Figure 1: Avaya SIP Telephony Network with MH-CURE

illustrates a sample configuration with an Avaya SIP-based network that includes the following products.

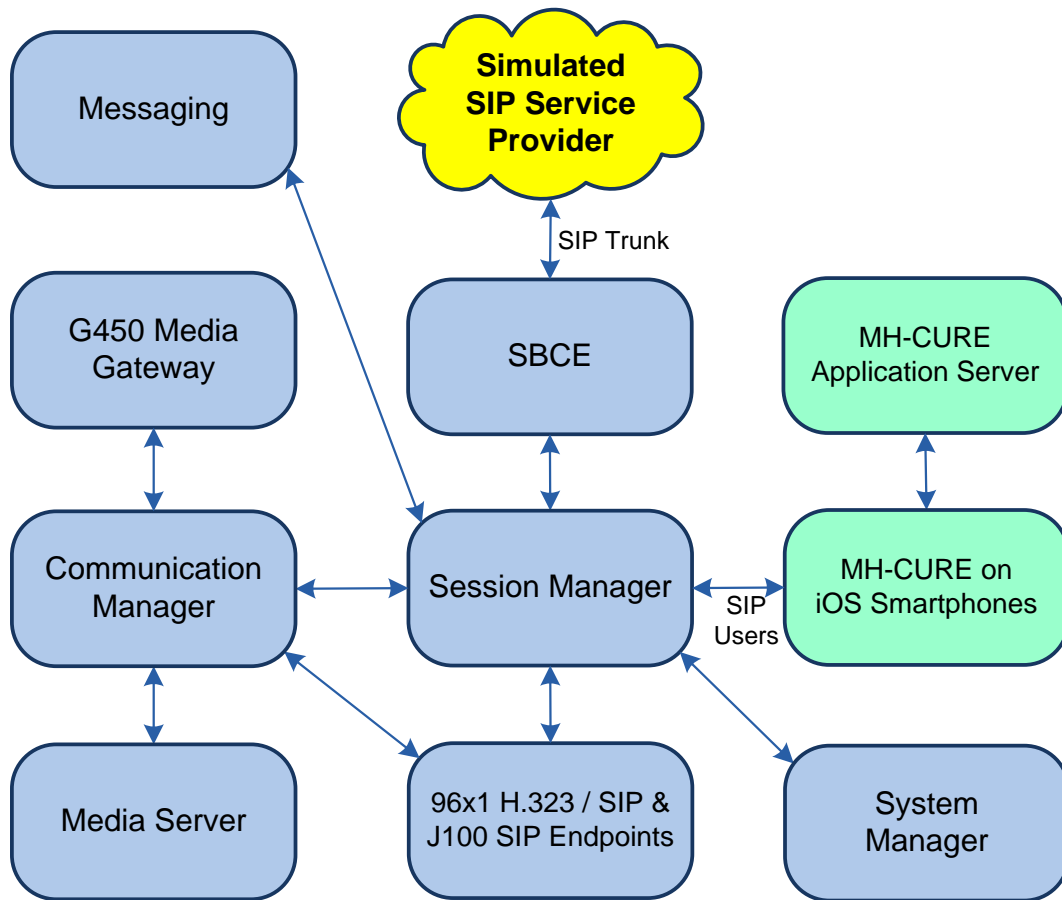


Figure 1: Avaya SIP Telephony Network with MH-CURE

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on Virtualized Environment	10.1.0.2-SP2 01.0.974.0-27607
Avaya G450 Media Gateway	42.7.0
Avaya Aura® Media Server running on Virtualized Environment	10.1.0.101
Avaya Aura® Messaging running on Virtualized Environment	7.2-5
Avaya Aura® System Manager running on Virtualized Environment	10.1.0.2 Service Pack 2 10.1.0.2.0715160
Avaya Aura® Session Manager running on Virtualized Environment	10.1.0.2 Service Pack 2 10.1.0.02.1010215
Avaya Session Border Controller for Enterprise running on Virtualized Environment	10.1.0.0-32-21432
Avaya 9641G IP Deskphone	6.8.5.3.2 (H.323)
Avaya J179 IP Phone	4.0.13.0.6 (SIP)
MH-CURE Client running on iOS 15.7.1 Smartphone	20.5.2.12
MH-CURE Application Server running on Windows Server 2016	20.5.1.3

5. Configure Avaya Aura® Communication Manager

This section describes the configuration of a SIP trunk to Session Manager and routing calls to MH-CURE clients. Administration of Communication Manager was performed using the System Access Terminal (SAT). The following configuration is covered:

- **Optional Features** to verify Communication Manager license.
- **IP Node Names** to associate names with IP addresses.
- **IP Codec Set** to specify the codec type used for calls to MH-CURE clients.
- **IP Network Region** to specify the SIP domain name, the IP codec set, and enable IP-IP direct audio (i.e., Shuffling).
- **SIP trunk** for calls towards Session Manager and MH-CURE clients.
- **Private Numbering** to allow the caller's extension to be sent over the SIP trunk.
- **Call Routing** to route calls to MH-CURE clients using AAR.

5.1. Verify Communication Manager License

Using the SAT, verify that the Off-PBX Telephones (OPS) option is 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, including MH-CURE users, that will be deployed. Only basic SIP endpoint licensing is required for this solution.

display system-parameters customer-options				Page 1 of 12	
OPTIONAL FEATURES					
G3 Version: V20		Software Package: Enterprise			
Location: 2		System ID (SID): 1			
Platform: 28		Module ID (MID): 1			
USED					
Platform Maximum Ports:		48000	108		
Maximum Stations:		150	73		
Maximum XMOBILE Stations:		36000	0		
Maximum Off-PBX Telephones - EC500:		150	0		
Maximum Off-PBX Telephones - OPS:		150	42		
Maximum Off-PBX Telephones - PBFMC:		150	0		
Maximum Off-PBX Telephones - PVFMC:		150	0		
Maximum Off-PBX Telephones - SCCAN:		0	0		
Maximum Survivable Processors:		313	0		
(NOTE: You must logoff & login to effect the permission changes.)					

5.2. Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name for Communication Manager (*procr*) and Session Manager (*sm10*). The host names will be used in other configuration screens of Communication Manager.

change node-names ip		Page 1 of 2
IP NODE NAMES		
Name	IP Address	
aes10	10.64.110.247	
aes811	10.64.110.209	
ams10	10.64.110.214	
default	0.0.0.0	
procr	10.64.110.213	
procr6	::	
sm10	10.64.110.212	
(7 of 7 administered node-names were displayed)		
Use 'list node-names' command to see all the administered node-names		
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name		

5.3. Administer IP Codec Set

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to MH-CURE clients. The form is accessed via the **change ip-codec-set 1** command. Note the codec set number since it will be used in the IP Network Region covered in the next section. For the compliance test, G.711MU was verified. The following IP codec set is configured as such.

Media encryption was enabled for Avaya IP Deskphones. MH-CURE clients weren't configured to support SRTP, so the *none* option was also included under **Media Encryption**.

change ip-codec-set 1		Page 1 of 2
IP MEDIA PARAMETERS		
Codec Set: 1		
Audio Codec	Silence Suppression	Frames Per Pkt Packet Size (ms)
1: G.711MU	n	2 20
2:		
3:		
4:		
5:		
6:		
7:		
Media Encryption		Encrypted SRTP: best-effort
1: 1-srtp-aescm128-hmac80		
2: 10-srtp-aescm256-hmac80		
3: none		
4:		
5:		

5.4. Administer IP Network Region

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 MH-CURE clients and IP endpoints without using media resources in Avaya G450 Media Gateway or Avaya Media Server. The **IP Network Region** form also specifies the **IP 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 1) is specified in the SIP signaling group.

change ip-network-region 1		Page 1 of 20
IP NETWORK REGION		
Region: 1	NR Group: 1	
Location: 1	Authoritative Domain: avaya.com	
Name: Main	Stub Network Region: n	
MEDIA PARAMETERS		
Codec Set: 1	Intra-region IP-IP Direct Audio: yes	
	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		

5.5. Administer SIP Trunk to Session Manager

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:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tls*.
- Specify the Ethernet processor (*procr*) of Communication Manager and Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form in **Section 5.2**.
- Ensure that the TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

add signaling-group 1		Page 1 of 2
SIGNALING GROUP		
Group Number: 10	Group Type: sip	
IMS Enabled? n	Transport Method: tls	
Q-SIP? n		
IP Video? n	Enforce SIPS URI for SRTP? n	
Peer Detection Enabled? y	Peer Server: SM	Clustered? n
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y		
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n		
Alert Incoming SIP Crisis Calls? n		
Near-end Node Name: procr	Far-end Node Name: sm10	
Near-end Listen Port: 5061	Far-end Listen Port: 5061	
	Far-end Network Region: 1	
Far-end Domain: avaya.com		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? y	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Initial IP-IP Direct Media? n	
	Alternate Route Timer(sec): 6	

Configure the **Trunk Group** form as shown below. This trunk group is used for SIP calls to MH-CURE clients. 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. Accept the default values for the remaining fields.

add trunk-group 1		Page 1 of 5	
TRUNK GROUP			
Group Number: 1	Group Type: sip	CDR Reports: y	
Group Name: SM Trunk 1	COR: 1	TN: 1	TAC: 101
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n			
Queue Length: 0	Auth Code? n		
Service Type: tie	Member Assignment Method: auto		
	Signaling Group: 1		
	Number of Members: 10		

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 1		Page 3 of 5	
TRUNK FEATURES			
ACA Assignment? n	Measured: both	Maintenance Tests? y	
Suppress # Outpulsing? n	Numbering Format: private	UII Treatment: shared	
		Maximum Size of UII Contents: 128	
		Replace Restricted Numbers? n	
		Replace Unavailable Numbers? n	
	Modify Tandem Calling Number: no		
Send UCID? y			
Show ANSWERED BY on Display? y			
DSN Term? n			

5.6. Configure Private Numbering

Configure the **Numbering – Private 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 ‘7’ whose calls are routed over any trunk group, including SIP trunk group 10, have their extension sent.

change private-numbering 0				Page 1 of 2	
NUMBERING - PRIVATE FORMAT					
Ext	Ext	Trk	Private	Total	
Len	Code	Grp (s)	Prefix	Len	
5	7			5	Total Administered: 1
					Maximum Entries: 540

5.7. AAR Call Routing

Configure the uniform dial plan table to route calls using AAR for dialed digits that are 5-digits long and begin with ‘76’. This would cover call routing to MH-CURE client extensions (i.e., 76001 – 76003).

change uniform-dialplan 76					Page 1 of 2	
UNIFORM DIAL PLAN TABLE						
					Percent Full: 0	
Matching			Insert		Node	
Pattern	Len	Del	Digits	Net	Conv	Num
76	5	0		aar	n	

SIP calls to Session Manager are routed over a SIP trunk via AAR call routing. Configure the AAR analysis form and add an entry that routes digits beginning with “76” to route pattern 10 as shown below. Note that the **Call Type** was set to *lev0*. This route calls to SIP stations and to MH-CURE clients.

change aar analysis 76						Page 1 of 2	
AAR DIGIT ANALYSIS TABLE							
Location: all						Percent Full: 0	
	Dialed	Total		Route	Call	Node	ANI
	String	Min	Max	Pattern	Type	Num	Reqd
76		5	5	1	lev0		n
77		5	5	1	aar		n
78		5	5	1	aar		n
8		5	5	1	aar		n
							n
							n

Configure a preference in **Route Pattern 10** to route calls over SIP trunk group 10 as shown below.

change route-pattern 1										Page	1	of	4							
Pattern Number: 1										Pattern Name: main										
SCCAN? n		Secure SIP? y		Used for SIP stations? n																
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted	DCS/ IXC												
No			Mrk	Lmt	List	Del	Digits	QSIG												
								Intw												
1:	1	0							n	user										
2:									n	user										
3:									n	user										
4:									n	user										
5:									n	user										
6:									n	user										
BCC VALUE										TSC	CA-TSC		ITC	BCIE	Service/Feature	PARM	Sub	Numbering	LAR	
0 1 2 M 4 W										Request						Dgts		Format		
1:	y	y	y	y	y	n	n			rest				pub-unk		none				
2:	y	y	y	y	y	n	n			rest						none				
3:	y	y	y	y	y	n	n			rest						none				
4:	y	y	y	y	y	n	n			rest						none				
5:	y	y	y	y	y	n	n			rest						none				
6:	y	y	y	y	y	n	n			rest						none				

6. Configure Avaya Aura® Session Manager

This section covers the procedure for adding a SIP user in Session Manager. The configuration covers:

- Launch System Manager
- Set Network Transport Protocol for MH-CURE Clients
- Administer SIP User

Note: It is assumed that basic configuration of Session Manager has already been performed.

6.1. Launch System Manager

Access the System Manager Web interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of the System Manager server. Log in using the appropriate credentials.

Recommended access to System Manager is via FQDN.

[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

All users must comply with all corporate instructions regarding the protection of information assets.

User ID:

Password:

[Change Password](#)

Supported Browsers: Firefox (minimum version 93.0), Chrome (minimum version 91.0) or Edge (minimum version 93.0).

6.2. Set Network Transport Protocol for MH-CURE Clients

From the System Manager **Home** screen, select **Elements** → **Routing** → **SIP Entities** and edit the SIP Entity for Session Manager shown below.

The screenshot shows the Avaya Aura System Manager 10.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The 'Routing' tab is selected. The 'SIP Entity Details' form for 'sm10' is displayed. The 'General' section includes fields for Name (sm10), IP Address (10.64.110.212), SIP FQDN, Type (Session Manager), Notes, Location (DevConnect), Outbound Proxy, Time Zone (America/Denver), Minimum TLS Version (Use Global Setting), and Credential name. The 'Monitoring' section includes SIP Link Monitoring (Link Monitoring Enabled), Proactive Monitoring Interval (900), and Reactive Monitoring Interval (120). Buttons for 'Commit' and 'Cancel' are at the top right.

Field	Value
Name	sm10
IP Address	10.64.110.212
SIP FQDN	
Type	Session Manager
Notes	
Location	DevConnect
Outbound Proxy	
Time Zone	America/Denver
Minimum TLS Version	Use Global Setting
Credential name	
SIP Link Monitoring	Link Monitoring Enabled
Proactive Monitoring Interval (in seconds)	900
Reactive Monitoring Interval (in seconds)	120

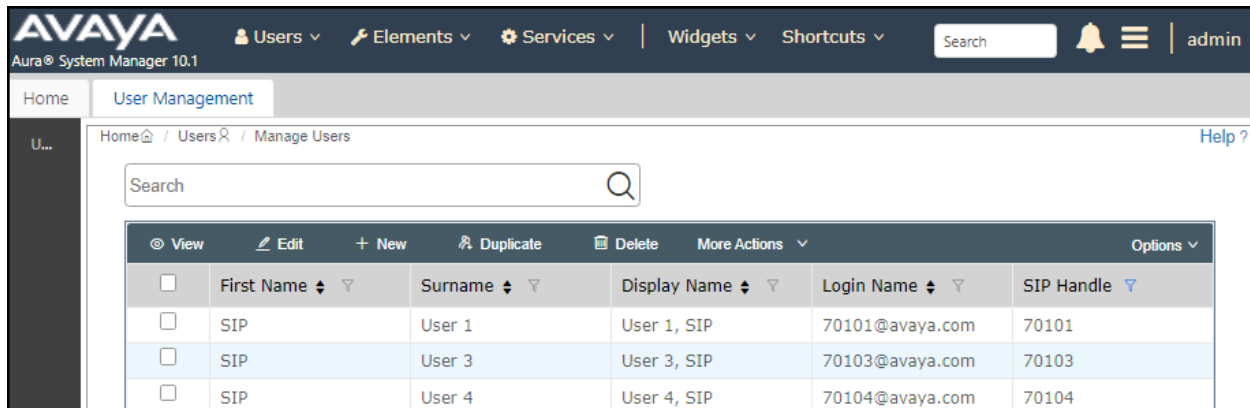
Scroll down to the **Listen Ports** section and verify that the transport network protocol used by MH-CURE clients is specified in the list below. For the compliance test, the solution used TCP network transport.

The screenshot shows the 'Listen Ports' section. It includes 'Add' and 'Remove' buttons, a '3 Items' indicator, and a 'Filter: Enable' button. A table lists three listen ports: 5060 (TCP), 5060 (UDP), and 5061 (TLS). Each row has a checkbox, a text input for the port number, a dropdown for the protocol, a dropdown for the default domain (all set to 'avaya.com'), a checkbox for the endpoint (all checked), and a text input for notes. At the bottom, there is a 'Select : All, None' option.

Listen Ports	Protocol	Default Domain	Endpoint	Notes
<input type="checkbox"/> 5060	TCP	avaya.com	<input checked="" type="checkbox"/>	
<input type="checkbox"/> 5060	UDP	avaya.com	<input checked="" type="checkbox"/>	
<input type="checkbox"/> 5061	TLS	avaya.com	<input checked="" type="checkbox"/>	

6.3. Administer SIP User

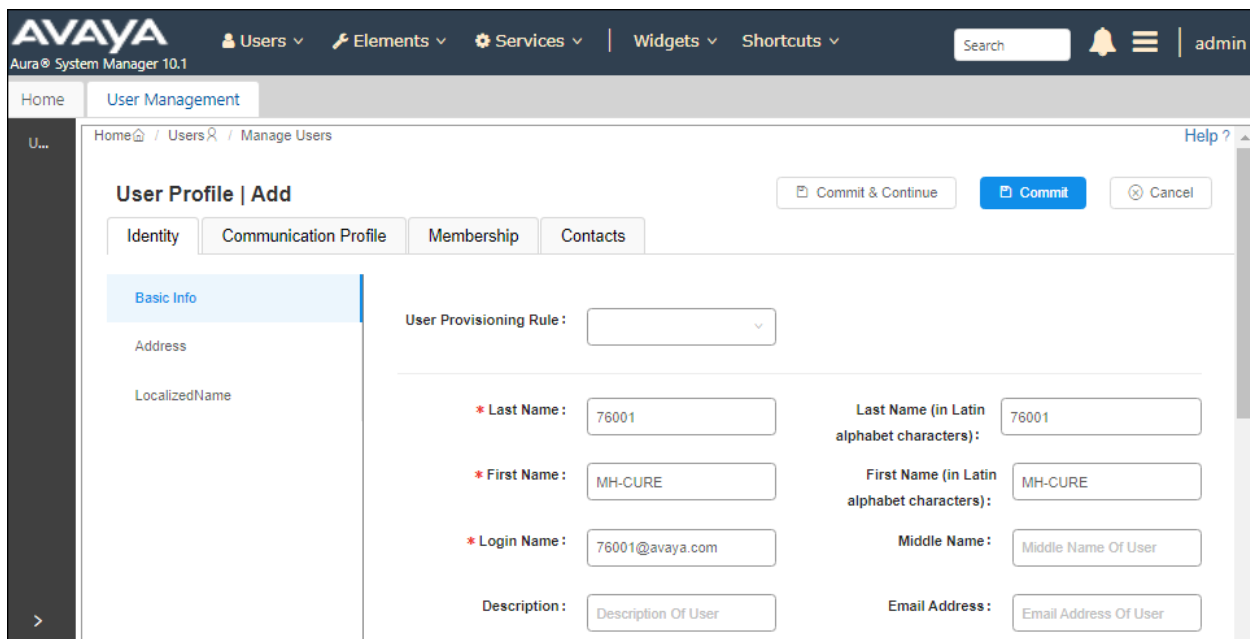
In the subsequent screen (not shown), select **Users** → **User Management** → **Manage Users** to display the **User Management** screen below. Click **+New** to add a user.



	First Name	Surname	Display Name	Login Name	SIP Handle
<input type="checkbox"/>	SIP	User 1	User 1, SIP	70101@avaya.com	70101
<input type="checkbox"/>	SIP	User 3	User 3, SIP	70103@avaya.com	70103
<input type="checkbox"/>	SIP	User 4	User 4, SIP	70104@avaya.com	70104

6.3.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter `<extension>@<domain>`, where `<extension>` is the desired MH-CURE client SIP extension and `<domain>` is the applicable SIP domain name from **Section 5.4**. Retain the default values in the remaining fields.



User Profile | Add

Commit & Continue Commit Cancel

Identity Communication Profile Membership Contacts

Basic Info

Address

LocalizedName

User Provisioning Rule:

* Last Name: 76001 Last Name (in Latin alphabet characters): 76001

* First Name: MH-CURE First Name (in Latin alphabet characters): MH-CURE

* Login Name: 76001@avaya.com Middle Name: Middle Name Of User

Description: Description Of User Email Address: Email Address Of User

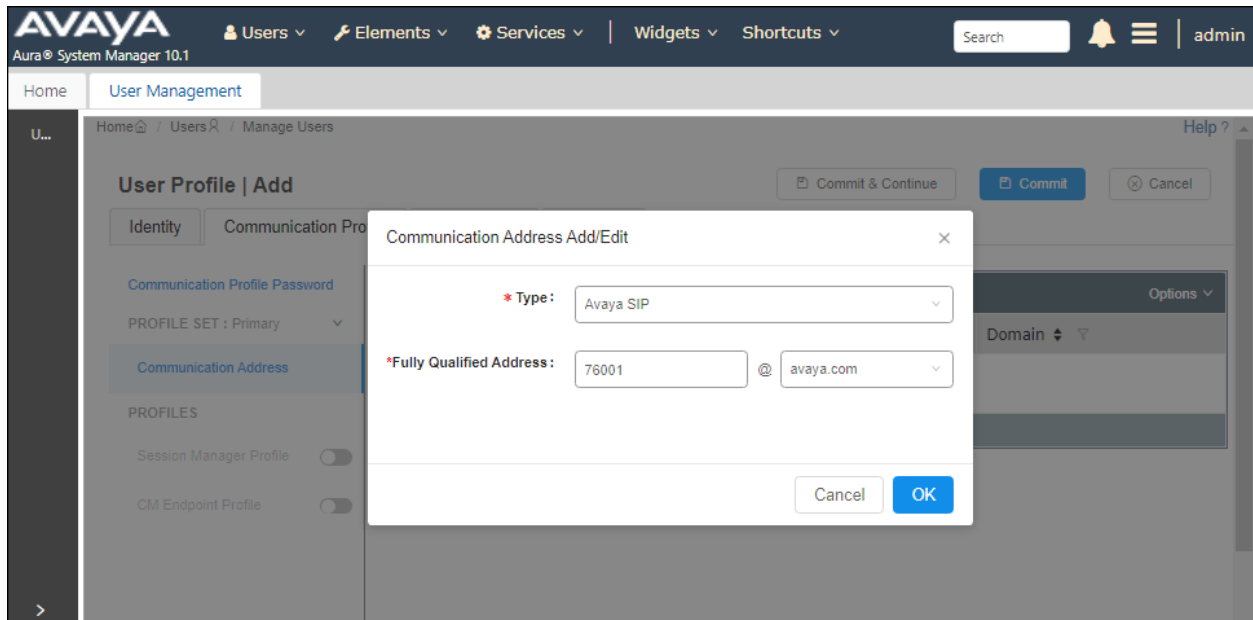
6.3.2. Communication Profile

Select the **Communication Profile** tab. Next, click on **Communication Profile Password**. For **Comm-Profile Password** and **Re-enter Comm-Profile Password**, enter the desired password for the SIP user to use for registration. Click **OK**.

The screenshot shows the Avaya Aura System Manager 10.1 interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 10.1', and various menu items like 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. A search bar and a user profile 'admin' are also present. The main content area is titled 'User Management' and shows a 'User Profile | Add' dialog box. The 'Communication Profile Password' tab is selected, displaying two password input fields. The first field is labeled 'Comm-Profile Password' and the second is labeled 'Re-enter Comm-Profile Password'. The second field has a green checkmark, indicating the passwords match. Below the fields is a link labeled 'Generate Comm-Profile Password'. At the bottom of the dialog box are 'Cancel' and 'OK' buttons. The background shows the 'User Profile | Add' form with tabs for 'Identity' and 'Communication', and a list of profiles including 'Session Manager Profile' and 'CM Endpoint Profile'.

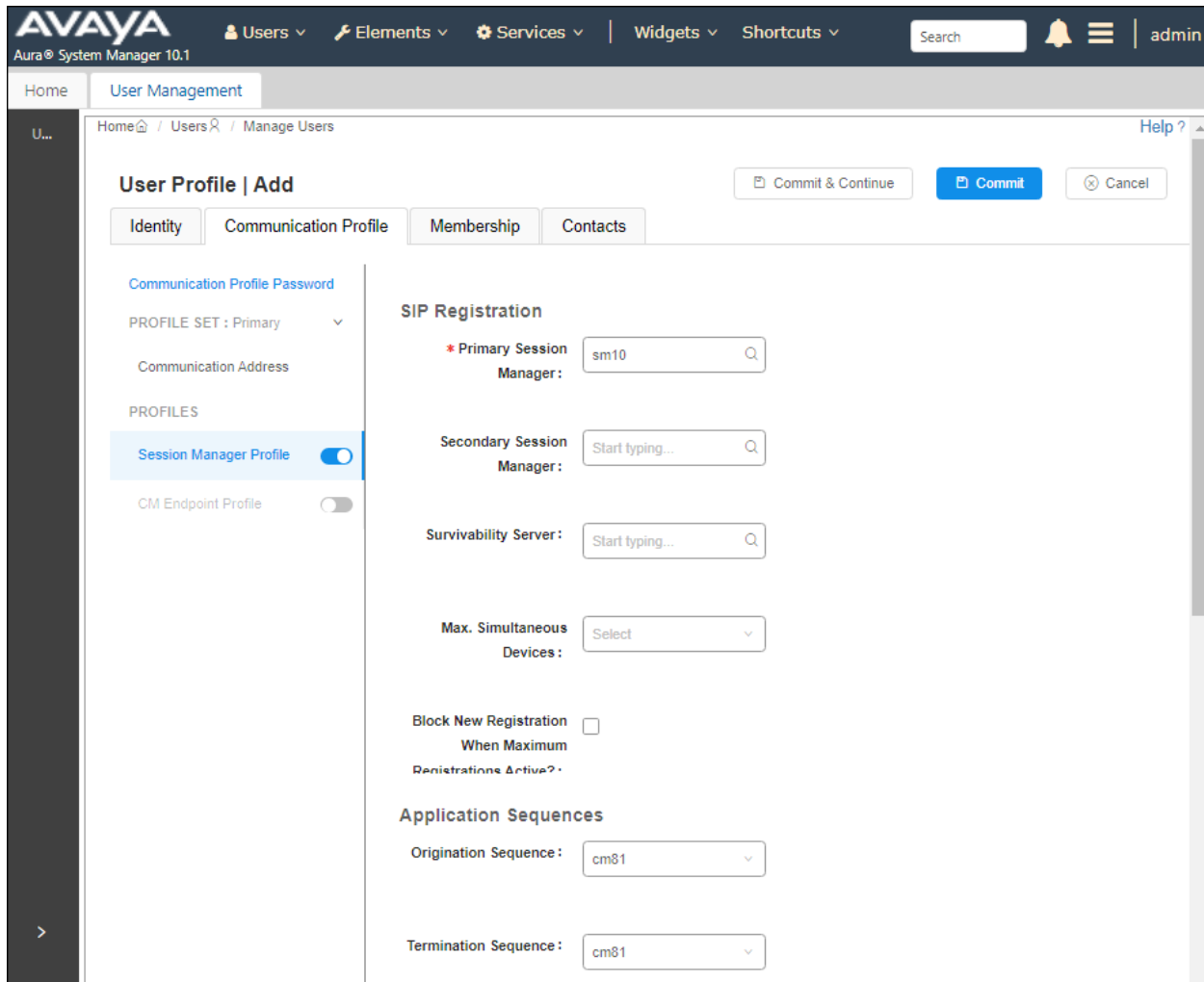
6.3.3. Communication Address

Click on **Communication Address** and then click **New** to add a new entry. The **Communication Address Add/Edit** dialog box is displayed as shown below. For **Type**, retain *Avaya SIP*. For **Fully Qualified Address**, enter the SIP user extension and select the domain name to match the login name from **Section 6.3.1**. Click **OK**.



6.3.4. Session Manager Profile

Click on toggle button by **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.



The screenshot shows the 'User Profile | Add' form in the Avaya Aura System Manager 10.1 interface. The 'Session Manager Profile' toggle is turned on. The form includes the following fields:

- Communication Profile Password**: PROFILE SET : Primary
- Communication Address**
- PROFILES**:
 - Session Manager Profile**: ☒
 - CM Endpoint Profile**: ☐
- SIP Registration**:
 - * Primary Session Manager**: sm10
 - Secondary Session Manager**: Start typing...
 - Survivability Server**: Start typing...
 - Max. Simultaneous Devices**: Select
 - Block New Registration When Maximum Registrations Active?**: ☐
- Application Sequences**:
 - Origination Sequence**: cm81
 - Termination Sequence**: cm81

Scroll down to the **Call Routing Settings** section to configure the **Home Location**.



The screenshot shows the 'Call Routing Settings' section of the form. It includes the following fields:

- * Home Location**: DevConnect
- Conference Factory Set**: Select

6.3.5. CM Endpoint Profile

Click on the toggle button by **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.1**. For **Template**, select *9641SIP_DEFAULT_CM_10_1*. For **Port**, click and select *IP*. Retain the default values in the remaining fields. Click on **Endpoint Editor** (i.e., Edit icon in **Extension** field) to configure the **Coverage Path**.

The screenshot displays the Avaya Aura System Manager 10.1 User Management interface. The top navigation bar includes the Avaya logo, navigation links for Users, Elements, Services, Widgets, and Shortcuts, a search bar, and a user profile icon labeled 'admin'. The main content area is titled 'User Profile | Add' and features four tabs: Identity, Communication Profile, Membership, and Contacts. The 'Communication Profile' tab is active, showing a form for adding a new profile. On the left, a sidebar lists 'Communication Profile Password', 'PROFILE SET : Primary', 'Communication Address', 'PROFILES', 'Session Manager Profile' (disabled), and 'CM Endpoint Profile' (enabled). The main form fields include: * System (cm10), * Profile Type (Endpoint), * Extension (76001) with an edit icon, * Template (9641SIP_DEFAULT_C), * Set Type (9641SIP), Security Code (Enter Security Code), Port (IP), Voice Mail Number, Preferred Handle (Select), Calculate Route Pattern (disabled), SIP URI (Select), Sip Trunk (aar), Delete on Unassign from User or on Delete (checked), and Override Endpoint Name and Localized Name (checked). A blue information icon is visible next to the 'Override Endpoint Name' checkbox.

Navigate to the **General Options** tab and set the **Coverage Path 1** field to the voicemail coverage path. This provides voicemail coverage for the SIP user. In this example, coverage path 1 was used.

New Endpoint Help ?

[Done](#)

[\[Save As Template\]](#)

* **System** * **Extension** Display Extension Ranges

* **Template** * **Set Type**

* **Port** * **Security Code**

Name

General Options (G) *	Feature Options (F)	Site Data (S)	Abbreviated Call Dialing (A)
Button Assignment (B)	Profile Settings (P)	Group Membership (M)	Enhanced Call Fwd (E)

<p>* Class of Restriction (COR) <input type="text" value="1"/></p> <p>* Emergency Location Ext <input type="text" value="76001"/></p> <p>* Tenant Number <input type="text" value="1"/></p> <p>* SIP Trunk <input type="text" value="aar"/></p> <p>Coverage Path 1 <input type="text" value="1"/></p> <p>Lock Message <input type="checkbox"/></p> <p>Multibyte Language <input type="text" value="Not Applicable"/></p>	<p>* Class Of Service (COS) <input type="text" value="1"/></p> <p>* Message Lamp Ext. <input type="text" value="76001"/></p> <p>Type of 3PCC Enabled <input type="text" value="None"/></p> <p>Coverage Path 2 <input type="text"/></p> <p>Localized Display Name <input type="text"/></p> <p>Enable Reachability for Station Domain Control <input type="text" value="system"/></p>
---	---

In the **Button Assignment** tab, select the **Main Buttons** sub-tab and accept the default settings with three call appearances.

General Options (G) *	Feature Options (F)	Site Data (S)	Abbreviated Call Dialing (A)
Button Assignment (B)	Profile Settings (P)	Group Membership (M)	Enhanced Call Fwd (E)

Main Buttons	Feature Buttons	Button Modules	Phone View
---------------------	------------------------	-----------------------	-------------------

Endpoint Configurations

Favorite	Button Label
<input type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text"/>

Button Configurations

Button Feature	Argument-1	Argument-2	Argument-3
call-appr ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>
call-appr ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>
call-appr ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>
None ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>
None ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>
None ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>
None ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>
None ▼	<input type="text"/>	<input type="text"/>	<input type="text"/>

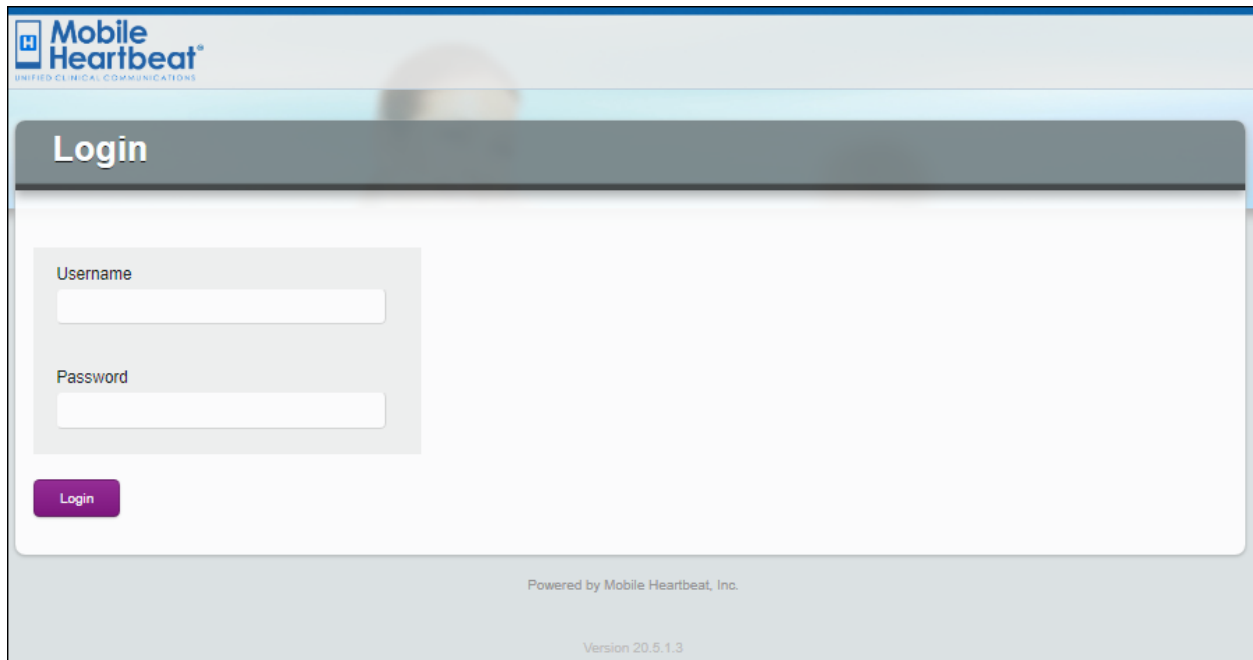
7. Configure MH-CURE

This section provides the procedure for configuring MH-CURE to provide the clients with SIP connectivity to Session Manager. Configuration for MH-CURE is performed via MH-CURE Administrative web User Interface.

- Log into MH-CURE Web Admin Tool
- Configure MH-CURE for SIP Endpoint Service
- Configure Voicemail Number
- Configure MH-CURE Users

7.1. Log into MH-CURE Web Admin Tool

Access the MH-CURE Web Admin Tool. Navigate to the Mobile Heartbeat site-specific link on an internet browser. Log in using appropriate credentials.

The screenshot shows the Mobile Heartbeat login interface. At the top left is the Mobile Heartbeat logo with the tagline 'UNIFIED CLINICAL COMMUNICATIONS'. Below the logo is a large grey header bar with the word 'Login' in white. Underneath the header is a white login box containing two input fields: 'Username' and 'Password'. Below these fields is a purple 'Login' button. At the bottom of the page, there is a footer area with the text 'Powered by Mobile Heartbeat, Inc.' and 'Version 20.5.1.3'.

7.2. Configure MH-CURE for SIP Endpoint Service

Click **Hospitals** in top menu. On the left pane, expand the Hospital tree, and click on the hospital name (e.g., Avaya). Click on the **Settings...** button.

The screenshot displays the 'Mobile Heartbeat' web application interface. The top navigation bar includes the logo, user 'admin', and links for Patients, Hospitals, Users, Settings, Reports, Monitors, Tools, and Logout. The main header is 'Manage Hospitals and Units'. On the left, there is an 'Import Hierarchy' button and a search box 'Search by hospital name'. Below the search box is a tree view showing 'Hospitals' expanded, with 'Avaya' selected, and 'OR' and 'VGW' as sub-items. The main content area is titled 'Manage Hospital - Avaya'. It contains a form with the following fields: 'Name *:' with value 'Avaya', 'Abbreviation:' with value 'Avaya' (with a note 'Used as the display name in clients'), and 'QuickLinks (Personal Devices):' with value 'avaya-p'. There are 'Update' and 'Delete' buttons. Below this is an 'Add Units' section with a text input field and an 'Add Units' button, with a note 'Use a comma separated list to add multiple Units'. At the bottom, there is a 'Hospital Settings' section with a 'Settings...' button and a 'Manage External Mappings...' button.

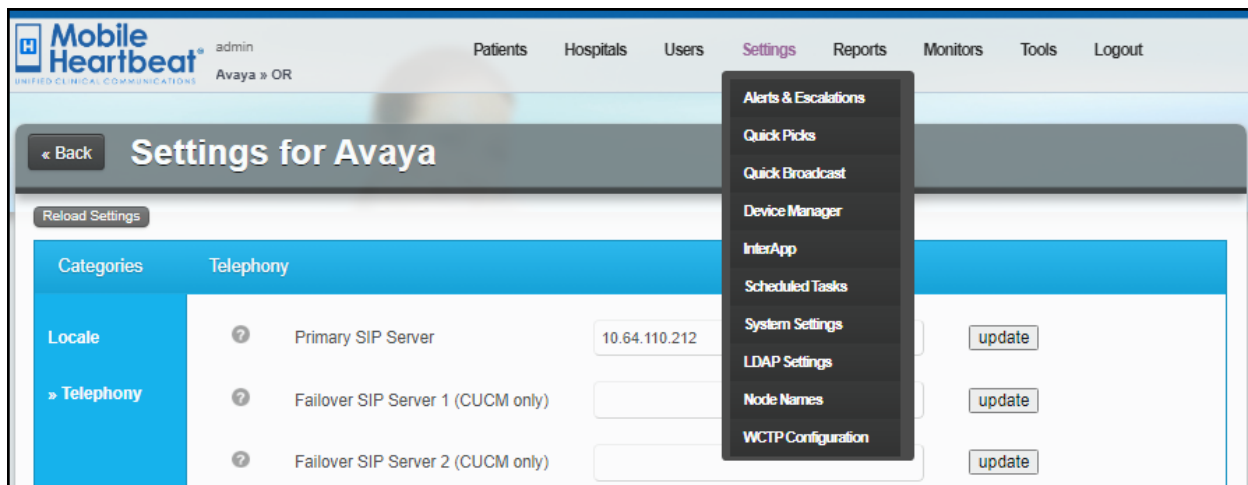
Click **Telephony** in the left pane. Set **Primary SIP Server** to the Session Manager IP address (e.g., *10.64.110.212*) from **Section 5.2**. Click the **Update** button to the right of the field.

The screenshot shows the 'Settings for Avaya' interface. At the top, there's a navigation bar with 'Mobile Heartbeat' logo, 'admin', 'Avaya » OR', and links for Patients, Hospitals, Users, Settings, Reports, Monitors, Tools, and Logout. Below this is a 'Settings for Avaya' header with a 'Back' button. A 'Reload Settings' button is also present. The main content area is divided into 'Categories' and 'Telephony'. Under 'Telephony', there are several settings: 'Primary SIP Server' (set to 10.64.110.212), 'Failover SIP Server 1 (CUCM only)', 'Failover SIP Server 2 (CUCM only)', and 'Failover SIP Server 3 (CUCM only)', each with an 'update' button. Below these is a section for 'Acrobix SIPIS' with settings: 'Use SIPIS for Personal Devices' (set to No), 'SIPIS Server', and 'SIPIS Registration Server', each with an 'update' button. The footer indicates 'Powered by Mobile Heartbeat, Inc.'

Categories	Telephony	
Locale » Telephony	Primary SIP Server 10.64.110.212 <input type="button" value="update"/>	
	Failover SIP Server 1 (CUCM only) <input type="text"/> <input type="button" value="update"/>	
	Failover SIP Server 2 (CUCM only) <input type="text"/> <input type="button" value="update"/>	
	Failover SIP Server 3 (CUCM only) <input type="text"/> <input type="button" value="update"/>	
	Acrobix SIPIS	
	Use SIPIS for Personal Devices No <input type="button" value="update"/>	
	SIPIS Server <input type="text"/> <input type="button" value="update"/>	
SIPIS Registration Server <input type="text"/> <input type="button" value="update"/>		

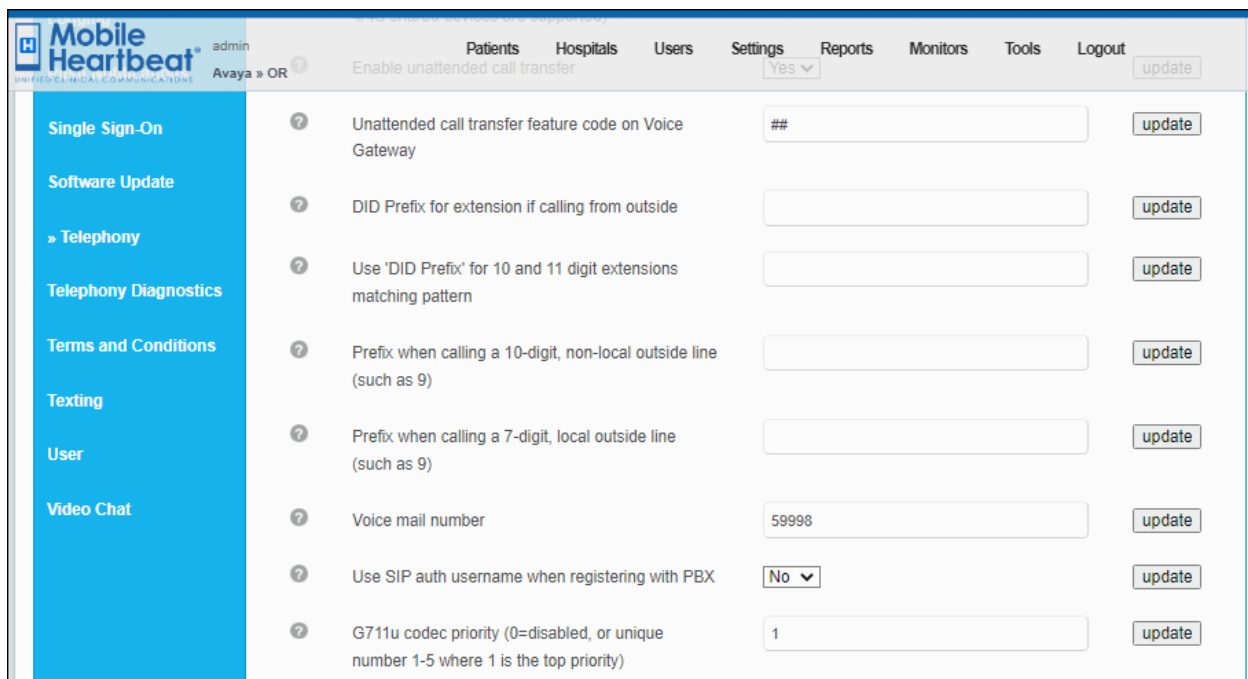
7.3. Configure Voicemail Number

To configure the voicemail number, first navigate to **Settings** → **System Settings**.



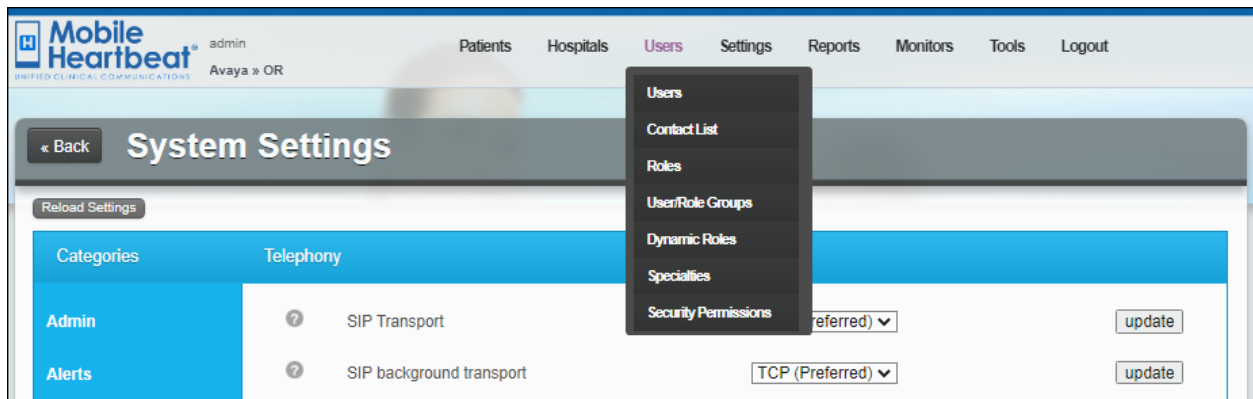
Next, select **Telephony** in left pane. Scroll down and locate the **Voice mail number** field. Type in the voicemail pilot number. For the compliance test, 59998 was the voicemail pilot number for Avaya Messaging.

Note: The **Telephony** page also provides the configuration of transport protocol (e.g., TCP) and the codec settings (not shown).

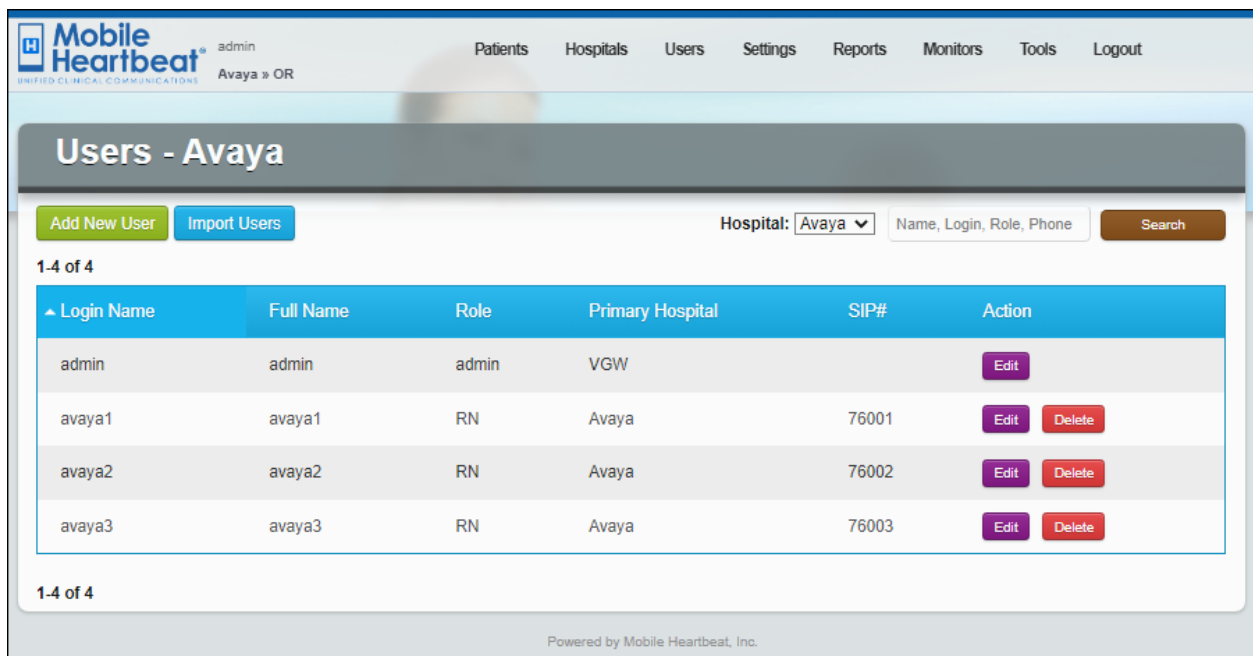


7.4. Configure MH-CURE Users

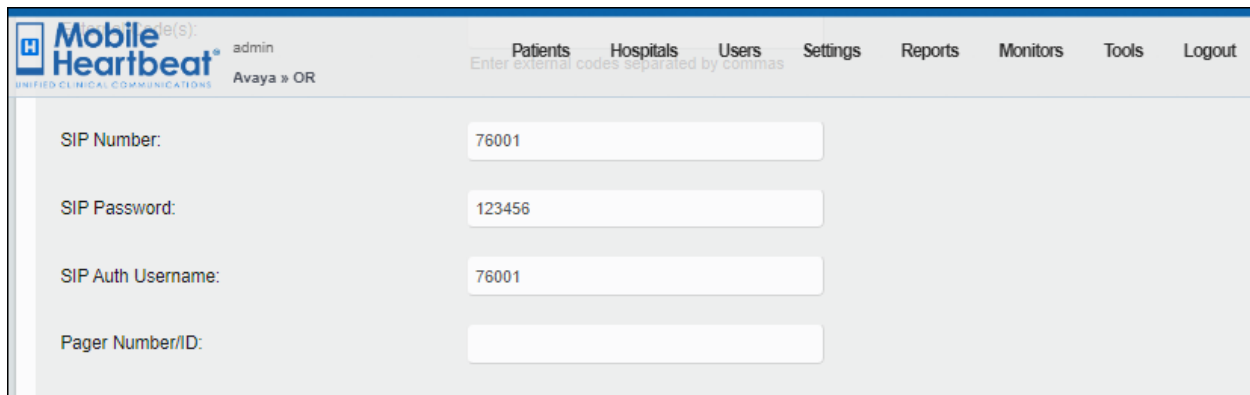
To add or modify an existing user, first navigate to **Users** → **Users** from the top menu.



Next, add a new user or edit an existing user. In this example, an existing user will be modified. Click the **Edit** button to the right of the avaya1 user.



Configure the user extension as configured in **Section 6.3.1** for **SIP Number** and **SIP Auth Username**. Configure the **SIP Password** as configured in **Section 6.3.2**. Once done, click **Update** at the bottom of the screen (not shown).



The screenshot shows the 'Mobile Heartbeat' web interface. The top navigation bar includes the logo, a user profile 'admin', and tabs for 'Patients', 'Hospitals', 'Users', 'Settings', 'Reports', 'Monitors', 'Tools', and 'Logout'. Below the navigation bar, there is a section for SIP configuration with four input fields: 'SIP Number' (containing '76001'), 'SIP Password' (containing '123456'), 'SIP Auth Username' (containing '76001'), and 'Pager Number/ID' (empty). The interface is light gray with blue accents.

SIP Number:	76001
SIP Password:	123456
SIP Auth Username:	76001
Pager Number/ID:	

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and MH-CURE.

1. Verify that MH-CURE clients have successfully registered with Session Manager. In System Manager, navigate to **Elements** → **Session Manager** → **System Status** → **User Registrations** to check the registration status as shown below.

AVAYA

[Users](#)
[Elements](#)
[Services](#)
[Widgets](#)
[Shortcuts](#)

[admin](#)

[Home](#)
[Session Manager](#)

User Registrations

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

View

Default

Export

Force Unregister

AST Device Notifications:

Reboot

Reload

Failback

As of 8:48 AM

Advanced Search

3 Items Found

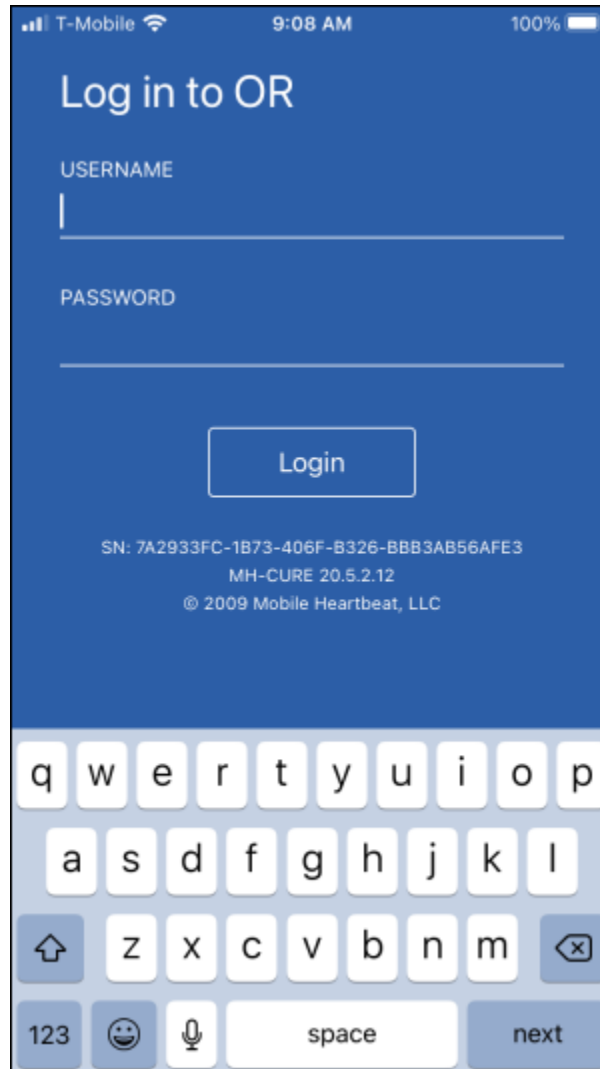
Show All

Filter: Enable, Clear

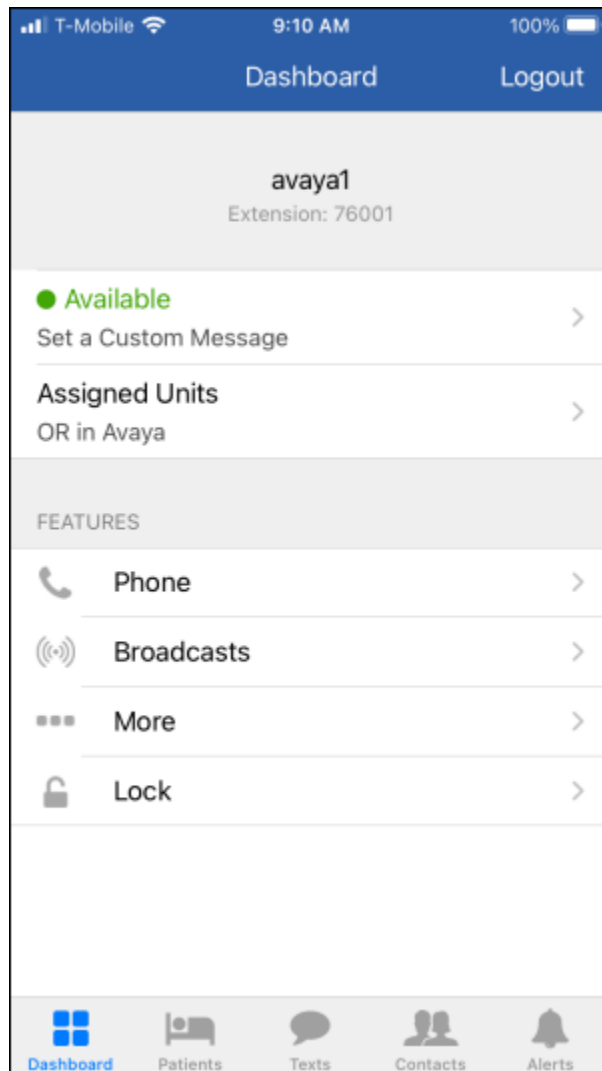
	Details	Address	First Name	Last Name	Actual Location	IP Address	Policy	Shared Control	Simult. Devices	AST Device	Registered					
											Prim	Sec	3rd	4th	Surv	Visiting
<input type="checkbox"/>	▶ Show	76003@avaya.com	MH-CURE	76003	---	192.168.4.24	fixed	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	▶ Show	76002@avaya.com	MH-CURE	76002	---	192.168.4.22	fixed	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	▶ Show	76001@avaya.com	MH-CURE	76001	---	192.168.4.21	fixed	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

Select : All, None

2. From an iOS mobile device, launch the MH-CURE app and log in with the appropriate credentials.



3. Once logged in, verify the MH-CURE client has registered with Session Manager. The MH-CURE should specify its SIP extension and a status of *Available*.



4. Establish a call from the MH-CURE client to an Avaya IP deskphone. While the call is active, verify two-way audio and exercise basic telephony features.

9. Conclusion

These Application Notes describe the integration of MH-CURE with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. MH-CURE clients successfully registered with Avaya Aura® Session Manager as a SIP endpoint through an enterprise wireless LAN. Incoming and outgoing calls were placed to/from MH-CURE clients and telephony features were exercised. All test cases passed with observations noted in **Section 2.2**.

10. References

This section references Avaya and Mobile Heartbeat documentation relevant to these Application Notes.

- [1] *Administering Avaya Aura® Communication Manager*, Release 10.1.x, Issue 2, September 2022, available at <http://support.avaya.com>.
- [2] *Administering Avaya Aura® System Manager*, Release 10.1.x, Issue 7, September 2022, available at <http://support.avaya.com>.
- [3] *Administering Avaya Aura® Session Manager*, Release 10.1, Issue 4, September 2022, available at <http://support.avaya.com>.
- [4] *MH-CURE 20.4 Web Admin User Guide*, MH00398, Revision 1.2.
- [5] *MH-CURE 20.4 iOS Shared User Guide*, MH00392, Revision 2.0.

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