

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 20.2 with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. 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 20.2 with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. 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 and G.729 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; call simply disappears on the client.

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

- Communication Manager with a G450 Media Gateway and Avaya Aura® Media Server providing media resources.
- Session Manager connected to Communication Manager via a SIP trunk.
- Session Manager connected to the PSTN via Avaya Session Border Controller for Enterprise (SBCE).
- Avaya Aura® System Manager used to configure Session Manager and SIP stations on Communication Manager.
- Avaya Messaging serving as the voicemail system.
- Avaya H.323 and SIP Deskphones.
- MH-CURE Clients running on iOS smartphones with wireless LAN device providing network access (not shown).
- MH-CURE Application Server providing SIP configuration to MH-CURE clients.

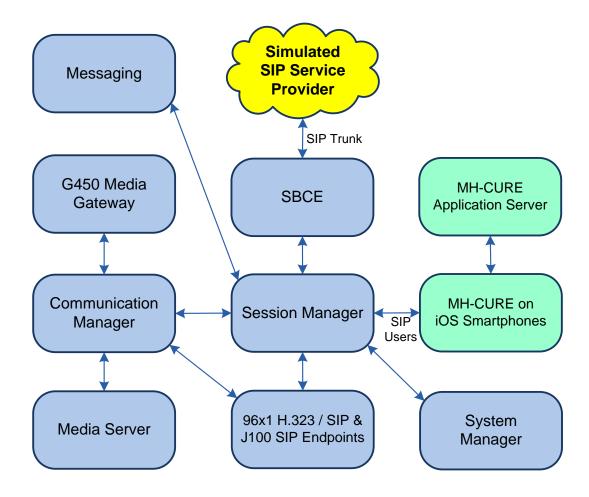


Figure 1: Avaya SIP Telephony Network with MH-CURE

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	8.1.3.0.1-FP3P1
Avaya G450 Media Gateway	FW 41.24.0
Avaya Aura® Media Server	v.8.0.2.138
Avaya Messaging	10.8.2 SP1
Avaya Aura® System Manager	8.1.3.0 Build No. – 8.1.0.0.733078 Software Update Revision No: 8.1.3.0.1012091 Feature Pack 3
Avaya Aura® Session Manager	8.1.3.0.813014
Avaya 96x1 Series IP Deskphone	6.8502 (H.323) 7.1.11.0.8 (SIP)
Avaya J100 Series SIP Deskphone	4.0.7.1.5
MH-CURE Client running on iOS 14.3 Smartphone	20.2.3.5
MH-CURE Application Server running on Windows Server 2016	20.2.3.109

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.

```
Page 1 of 12
display system-parameters customer-options
                             OPTIONAL FEATURES
    G3 Version: V18
                                              Software Package: Enterprise
      Location: 2
                                               System ID (SID): 1
      Platform: 28
                                               Module ID (MID): 1
                                                           USED
                              Platform Maximum Ports: 48000
                                  Maximum Stations: 36000
                           Maximum XMOBILE Stations: 36000
                  Maximum Off-PBX Telephones - EC500: 41000
                  Maximum Off-PBX Telephones - OPS: 41000 16
                  Maximum Off-PBX Telephones - PBFMC: 41000
                                                             0
                  Maximum Off-PBX Telephones - PVFMC: 41000
                                                             0
                  Maximum Off-PBX Telephones - SCCAN: 0
                                                              0
                       Maximum Survivable Processors: 313
       (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 (*devcon-sm*). 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
default
                 0.0.0.0
devcon-aes
                  10.64.102.119
devcon-ams
                   10.64.102.118
                   10.64.102.117
devcon-sm
procr
                   10.64.102.115
procr6
                   ::
( 6 of 6 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 and G.729 codecs were verified. The following IP codec set is configured with G.711MU.

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
                                                                 1 of
                                                           Page
                       IP MEDIA PARAMETERS
   Codec Set: 1
   Audio
Codec
             Silence Frames
             Suppression Per Pkt Size(ms)
1: G.711MU
               n 2
                                   20
2:
3:
4:
5:
6:
7:
    Media Encryption
                                    Encrypted SRTCP: best-effort
1: 1-srtp-aescm128-hmac80
2: none
3:
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 the 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
                                                                        1 of 20
                                                                 Page
                               IP NETWORK REGION
Region: 1 NR Group: 1 Location: 1 Authoritative
                Authoritative Domain: avaya.com
   Name:
                                Stub Network Region: n
MEDIA PARAMETERS
                                Intra-region IP-IP Direct Audio: yes
     Codec Set: 1
                                Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048
                                           IP Audio Hairpinning? n
  UDP Port Max: 50999
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 5.1.
- 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 10
                                                            Page 1 of
                               SIGNALING GROUP
Group Number: 10

IMS Enabled? n
                            Group Type: sip
                       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: devcon-sm
Near-end Listen Port: 5061
                                         Far-end Listen Port: 5061
                                       Far-end Network Region: 1
Far-end Domain: avaya.com
DTMF over IP: rtp-payload

Session Establishment Timer(min): 3

Enable Laver 3 Tost?
                                          Bypass If IP Threshold Exceeded? 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 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 10

TRUNK GROUP

Group Number: 10

Group Type: sip

Group Name: To devcon-sm

Direction: two-way

Dial Access? n

Queue Length: 0

Service Type: tie

Auth Code? n

Member Assignment Method: auto

Signaling Group: 10

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 10
TRUNK FEATURES

ACA Assignment? n

Measured: none

Maintenance Tests? y

Suppress # Outpulsing? n

Numbering Format: private

UUI Treatment: service-provider
Maximum Size of UUI Contents: 128
Replace Restricted Numbers? n
Replace Unavailable Numbers? n

Hold/Unhold Notifications? y

Modify Tandem Calling Number: no

Show ANSWERED BY on Display? y
```

On **Page 4** of the trunk group form, the default settings were used as shown below.

```
add trunk-group 10
                                                                   5 of 5
                                                             Page
                              PROTOCOL VARIATIONS
                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? n
                                     Send Diversion Header? n
                                   Support Request History? y
                              Telephone Event Payload Type:
                        Convert 180 to 183 for Early Media? n
                  Always Use re-INVITE for Display Updates? n
                        Identity for Calling Party Display: P-Asserted-Identity
            Block Sending Calling Party Location in INVITE? n
                 Accept Redirect to Blank User Destination? n
                                              Enable Q-SIP? n
          Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                                Request URI Contents: may-have-extra-digits
```

5.6. Configure Private Numbering

Configure the **Numbering – Private Format** form to send the calling party number to the farend. 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-n	_	JUMBERING - PRIV	ATE FORMA	Page 1	of	2
Ext Ext Len Code 5 7	Trk Grp(s)	Private Prefix	Total Len 5	Total Administered: Maximum Entries:		

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 '78'. This would cover call routing to MH-CURE client extensions (i.e., 78300 – 78302).

change unifor	m-dialplan 7	Page 1 of 2		
	UNI	Percent Full: 0		
				reicent ruii. 0
Matching		Insert	Node	
Pattern	Len Del	Digits	Net Conv Num	
78	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 "78" 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 7	Page 1 of 2							
			GIT ANALYS Location:		Percent Full: 2			
Dialed	Tot	al	Route	Call	Node	ANI		
String	Min	Max	Pattern	Type	Reqd			
7	7	7	254	n				
78	5	5	10	lev0	n			
8	7	7	254	n				
9	7	7	254	aar		n		
					n			
						n		

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

char	nge i	cout	e-pa	tter	n 10]	Page	1 of	3
					Pat	tern 1	Numbei	r: 10		Patter	n Name	: To	devc	on-sm		
	SCC	N?	n	Sec	are :	SIP?	n	Used	for	SIP st	tations	? n				
	${\tt Grp}$	FRI	NP	Pfx	Hop	Toll	No.	Inser	rted						DCS/	IXC
	No			Mrk	Lmt	List	Del	Digit	s						QSIG	}
							Dgts								Intw	•
1:	10	0													n	user
2:															n	user
3:															n	user
4:															n	user
5:															n	user
6:															n	user
	BCC	C VA	LUE	TSC	CA-	rsc	ITC	BCIE	Serv	ice/Fe	eature	PARM	Sub	Numbe:	ring	LAR
	0 1	2 M	14 V	7	Req	uest							Dgts	Forma		
1:	у у	УУ	ууг	ı n			rest	t						unk-u	nk	none
2:	У У	УУ	y r	n n			rest	t								none
3:	У У	УУ	y r	n n			rest	t								none
4:	У У	УУ	y r	n n			rest	t								none
5:	У У	УУ	y r	n n			rest	t								none
6:	У У	УУ	y r	n n			rest	t								none

6. Configure Avaya Aura® Session Manager

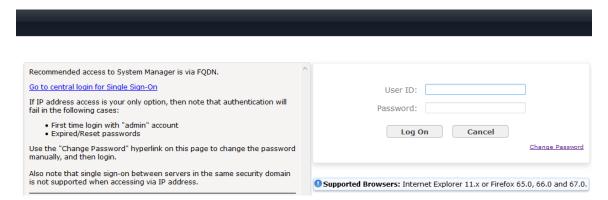
This section covers the procedure for adding a SIP user in Session Manager. The configuration covers:

- 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.

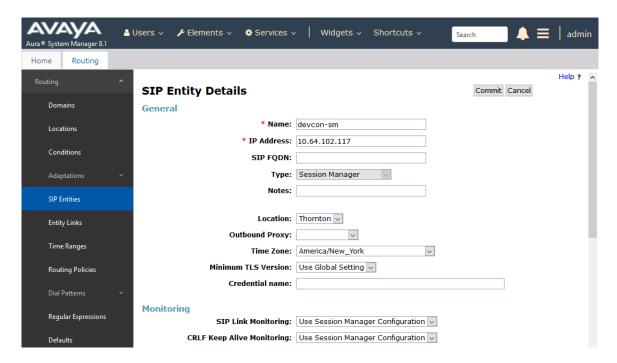
6.1. Launch System Manager

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.



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.



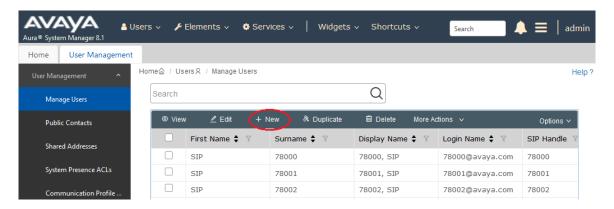
Scroll down to the **Listen Ports** section and verify that the transport network protocol used by MH-CURE clients is specified in the list below. For the compliance test, the solution used TCP network transport.

Listen Ports



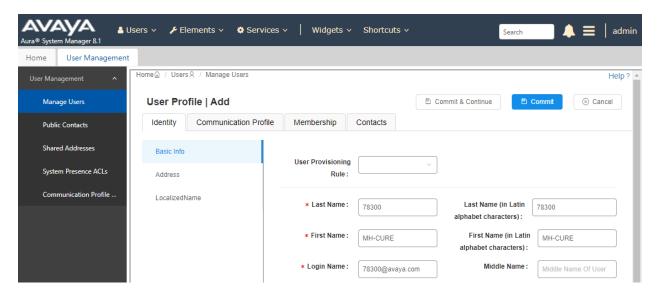
6.3. Administer SIP User

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.



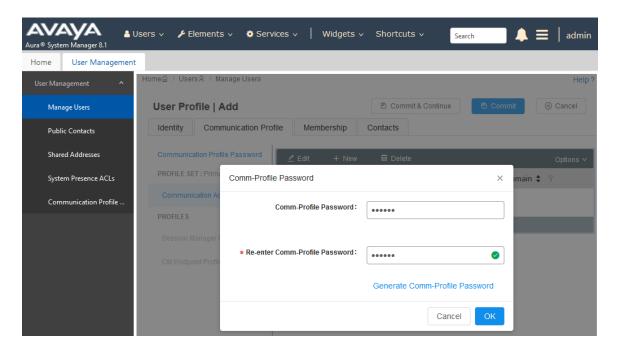
6.3.1. Identity

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.



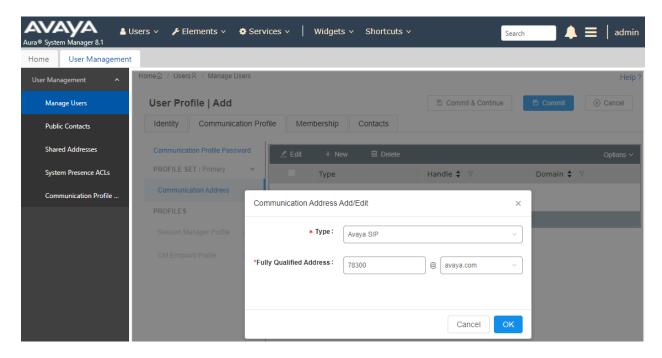
6.3.2. Communication Profile

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**.



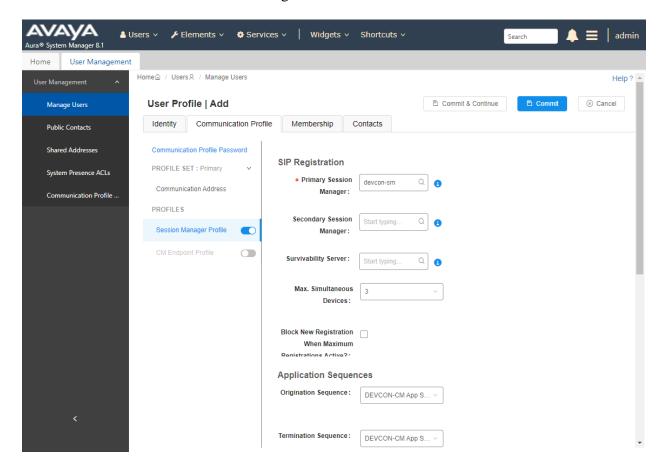
6.3.3. Communication Address

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**.

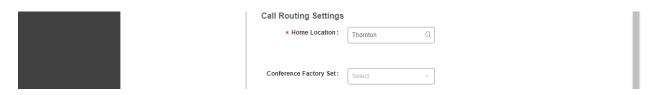


6.3.4. Session Manager Profile

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.

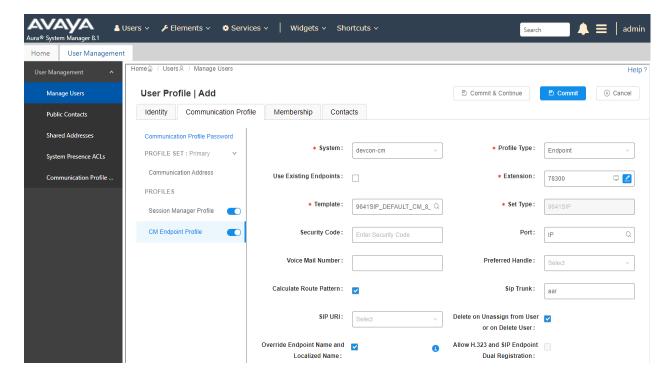


Scroll down to the **Call Routing Settings** section to configure the **Home Location**.

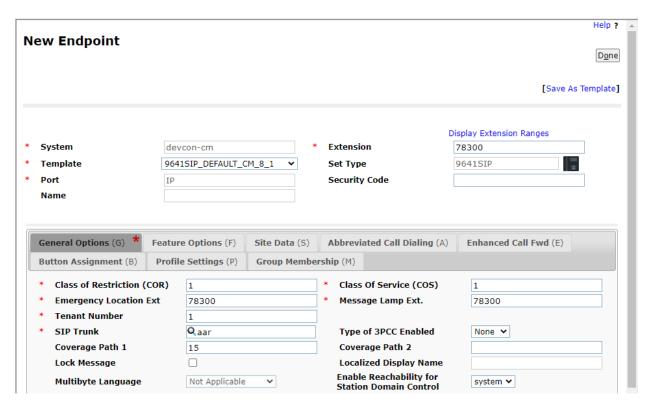


6.3.5. CM Endpoint Profile

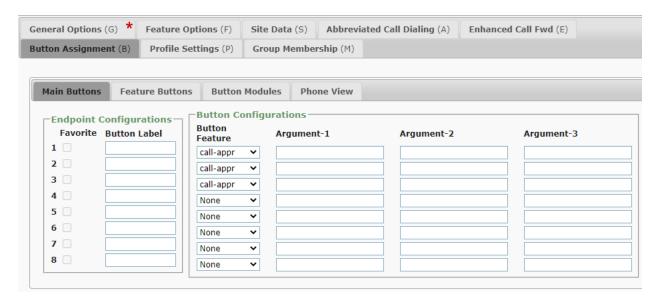
Click on the toggle button by **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.1**. For **Template**, select *9641SIP_DEFAULT_CM_8_1*. For **Port**, click and select *IP*. Retain the default values in the remaining fields. Click on **Endpoint Editor** (i.e., Edit icon in **Extension** field) to configure the **Coverage Path**.



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 15 was used.



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



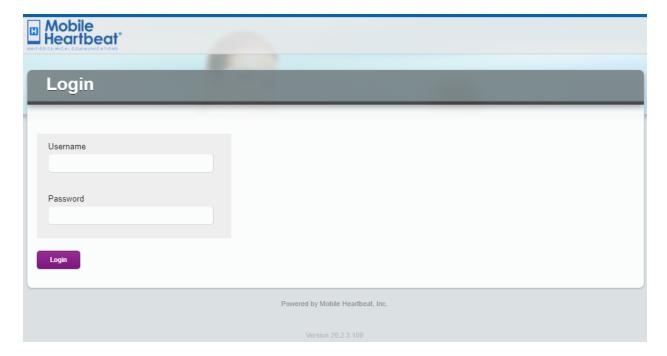
7. Configure MH-CURE

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

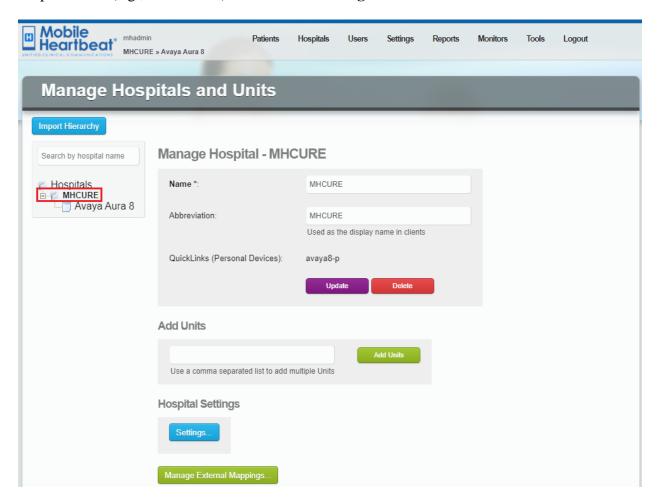
7.1. Log into MH-CURE Web Admin Tool

Access the MH-CURE Web Admin Tool by using the URL "Error! Hyperlink reference not valid." in an Internet browser, where $\langle FQDN \rangle$ is the FQDN of the MH-CURE Application Server. Log in using appropriate credentials.

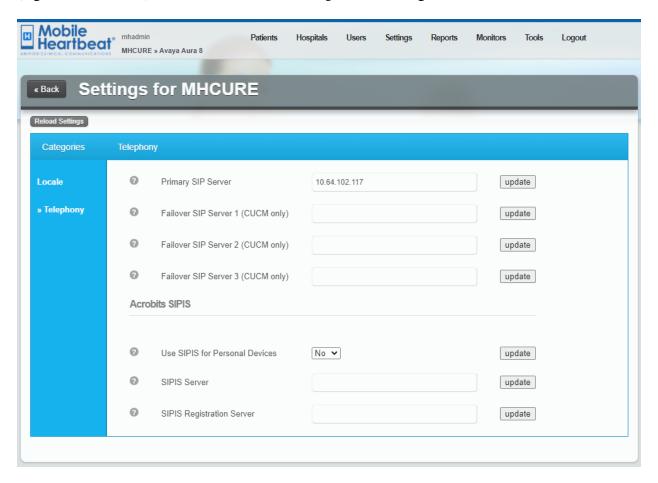


7.2. Configure MH-CURE for SIP Endpoint Service

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

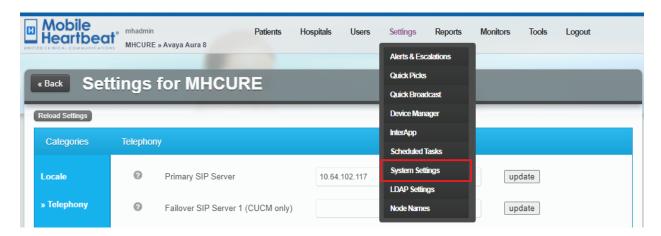


Click **Telephony** in the left pane. Set **Primary SIP Server** to the Session Manager IP address (e.g., 10.64.102.117) from **Section 5.1**. Click **Update** to the right of the field.



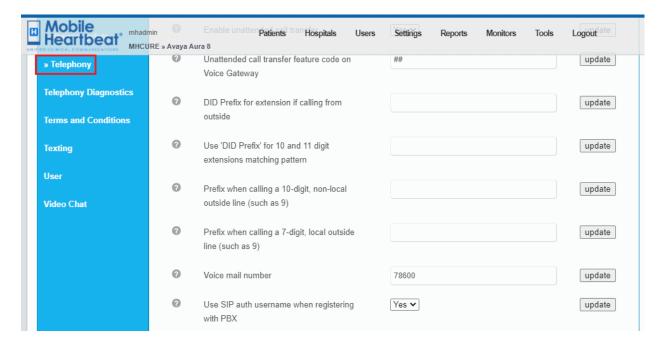
7.3. Configure Voicemail Number

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



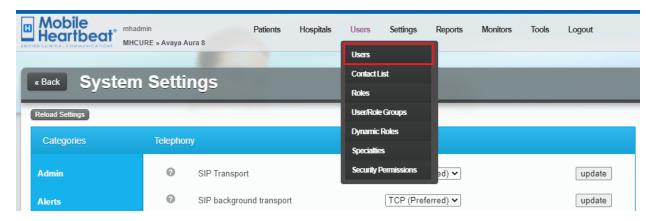
Next, select **Telephony** in left pane. Scroll down and locate the **Voice mail number** field. Type in the voicemail pilot number. For the compliance test, 78600 was the voicemail pilot number for Avaya Messaging.

Note: The **Telephony** page also provides the configuration of transport protocol (e.g., TCP) and the codec settings. The default settings were used, but to allow G.729 codec, the codec must be enabled.

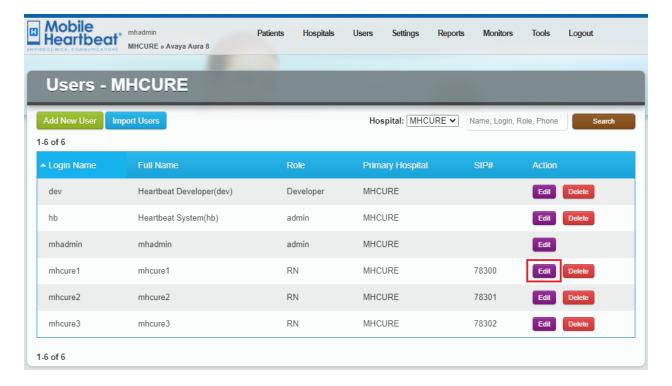


7.4. Configure MH-CURE Users

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 associated with the **mhcure1** user.



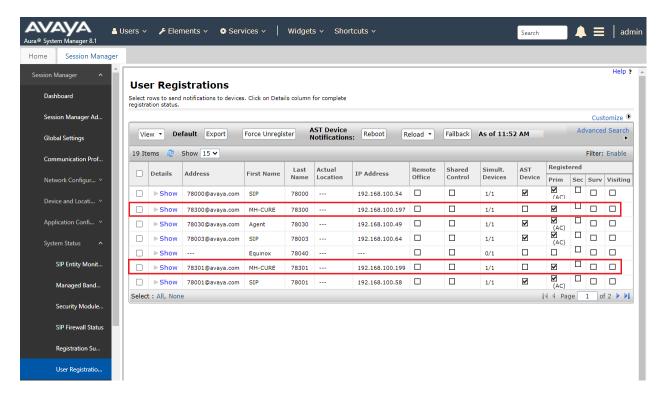
Configure the user extension as configured in **Section 6.3.1** for **SIP Number** and **SIP Auth Username**. Configure the **SIP Password** as configured in **Section 6.3.2**. Once done, click **Update** at the bottom of the screen (not shown).

SIP Number:	78300
SIP Password:	123456
SIP Auth Username:	78300

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.

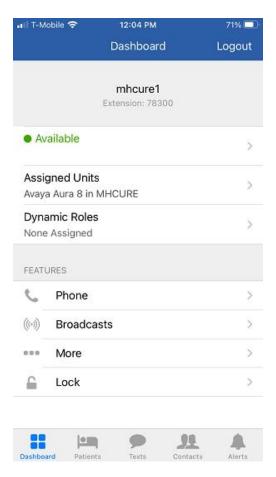
1. Verify that MH-CURE clients have successfully registered with Session Manager. In System Manager, navigate to **Elements → Session Manager → System Status → User Registrations** to check the registration status as shown below.



2. From an iOS mobile device, launch the MH-CURE app and login 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 the Avaya and Mobile Heartbeat documentation relevant to these Application Notes.

- [1] *Administering Avaya Aura*® *Communication Manager*, Release 8.1.x, Issue 8, November 2020, available at http://support.avaya.com.
- [2] *Administering Avaya Aura*® *System Manager for Release 8.1.x*, Release 8.1.x, Issue 8, November 2020, available at http://support.avaya.com.
- [3] *Administering Avaya Aura*® *Session Manager*, Release 8.1.x, Issue 7, October 2020, available at http://support.avaya.com.
- [4] MH-CURE 20.2 Web Admin Guide, MH00333, Revision 1.0.
- [5] MH-CURE 20.2 iOS Shared User Guide, MH00324, Revision 1.0.

©2021 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and TM are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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