



Avaya Solution & Interoperability Test Lab

Application Notes for Mobile Heartbeat MH-CURE SIP Clients with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.1

Abstract

These Application Notes describe the configuration steps required to integrate the Mobile Heartbeat MH-CURE SIP clients with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. MH-CURE SIP clients registered with Avaya Aura® Session Manager as SIP endpoints.

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 steps required to integrate the Mobile Heartbeat MH-CURE SIP clients with Avaya Aura® Session Manager (Session Manager) and Avaya Aura® Communication Manager (Communication Manager). MH-CURE SIP clients registered with Avaya Aura® Session Manager as SIP endpoints.

MH-CURE SIP clients are software applications that run on mobile iOS devices. The integration of MH-CURE SIP clients makes it possible to perform all telephony management via Avaya Aura® environment and then simply assigning extensions to MH-CURE SIP clients. For hospital environments utilizing Avaya Aura® and MH-CURE, the integration makes system administration more efficient. It also provides a seamless user experience when calls involve both mobile iOS devices running the MH-CURE SIP clients and Avaya Endpoints. During the compliance test MH-CURE SIP clients registered with Session Manager via SIP/TCP with the use of RTP/UDP for audio. Configuration of MH-CURE SIP clients is performed via MH-CURE Application Server.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between MH-CURE SIP clients, Avaya SIP/H.323 endpoints and the PSTN, and exercising basic telephony features, such as hold, mute, call transfer and conference. Additional telephony features, such as call forward, follow me, call park/unpark, and call pickup were also verified using Communication Manager Features Access Codes (FACs).

The serviceability testing focused on verifying that MH-CURE SIP clients returned to service after re-connecting the network connection or rebooting the MH-CURE SIP clients.

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 this DevConnect Application Note 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 SIP clients did not utilize encryption capabilities.

This test was conducted in a lab environment simulating a basic customer enterprise network environment. The testing focused on the standards-based interface between the Avaya solution and the third party solution. The results of testing are therefore considered to be applicable to either a premise-based deployment or to a hosted or cloud deployment where some elements of the third party solution may reside beyond the boundaries of the enterprise network, or at a different physical location from the Avaya components.

Readers should be aware that network behaviors (e.g. jitter, packet loss, delay, speed, etc.) can vary significantly from one location to another, and may affect the reliability or performance of the overall solution. Different network elements (e.g. session border controllers, soft switches, firewalls, NAT appliances, etc.) can also affect how the solution performs.

If a customer is considering implementation of this solution in a cloud environment, the customer should evaluate and discuss the network characteristics with their cloud service provider and network organizations, and evaluate if the solution is viable to be deployed in the cloud.

The network characteristics required to support this solution are outside the scope of these Application Notes. Readers should consult the appropriate Avaya and third party documentation for the product network requirements. Avaya makes no guarantee that this solution will work in all potential deployment configurations.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of MH-CURE SIP clients with Session Manager.
- Calls between MH-CURE SIP clients and Avaya SIP/H.323 Endpoints with Direct IP Media (Shuffling) enabled and disabled.
- Calls between MH-CURE SIP clients and the PSTN.
- Support of G.711Mu codec.
- Proper recognition of DTMF tones.
- Basic telephony features, including hold, mute, redial, multiple calls, blind/attended transfer, attended conference, and long duration calls.
- Extended telephony features using Communication Manager FACs for Call Forward, Follow Me, 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 SIP clients and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observations noted:

- In a scenario where called number from MH-CURE SIP client is invalid or busy, related tones are not played; call simply disappears on the client.

2.3. Support

For technical related to MH-CURE SIP clients, contact Mobile Heartbeat Support via the Mobile Heartbeat website.

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya Aura® SIP-based network that includes the following products:

- Avaya Aura® Communication Manager running in a virtual environment with an Avaya G450 Gateway. Avaya G450 Gateway was connected to the PSTN via an ISDN-PRI trunk.
- Media resources in the Avaya G450 Media Gateway and Avaya Aura® Media Server.
- Avaya Aura® Session Manager connected to Avaya Aura® Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP Endpoints.
- Avaya Aura® System Manager used to configure Avaya Aura® Session Manager.
- Avaya 96x1 and J Series H.323 and SIP Deskphones.
- MH-CURE Application Server running on a Windows Server 2016 and MH-CURE SIP clients running on iOS devices.

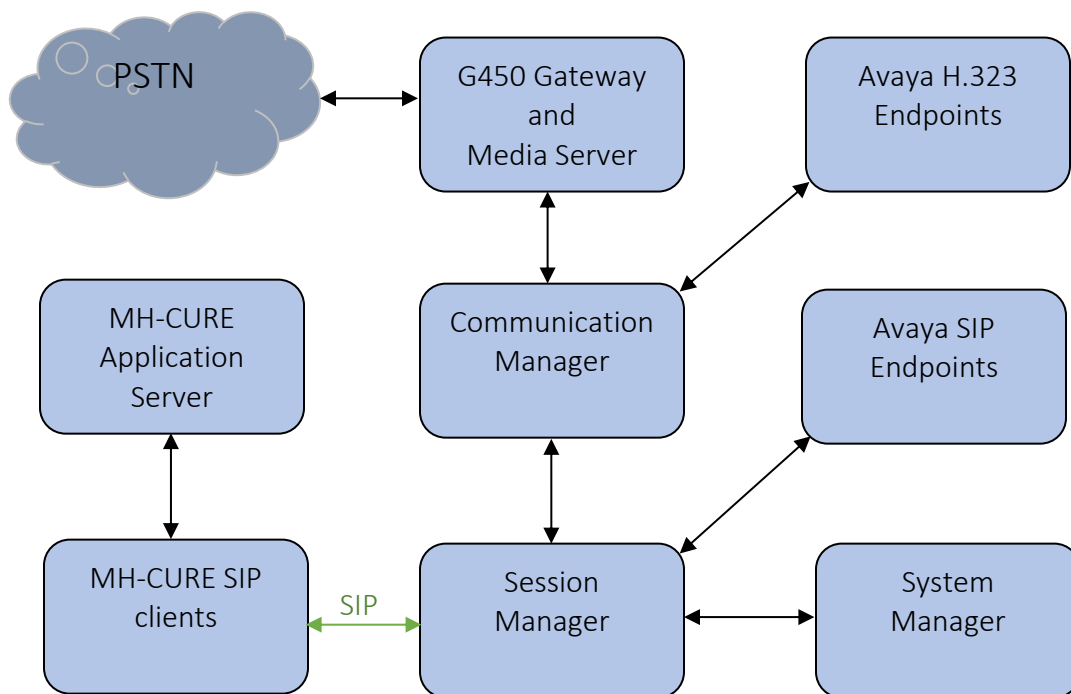


Figure 1: Avaya Aura® with MH-CURE

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	7.1.3.3.0-FP3SP3
Avaya G450 Media Gateway	FW 40.19.1
Avaya Aura® Media Server	8.0.0.205
Avaya Aura® Session Manager	7.1.3.3.713307
Avaya Aura® System Manager	7.1.3.3
Avaya Aura® Communication Manager Messaging	7.1.3.1.0-FP3SP1
Avaya 9600 Series IP Deskphones	6.8.2 (H.323) 7.1.6.1 (SIP)
Avaya J100 Series IP Phones	6.8.2 (H.323) 4.0.2.1 (SIP)
Mobile Heartbeat MH-CURE Application Server running on Windows server 2016 Mobile Heartbeat MH-CURE SIP client running on iOS mobile devices v12.1.3	R19.2.5

5. Configure Avaya Aura® Communication Manager

This section provides the procedure for configuring Communication Manager. The procedure includes the following areas:

- Verify license
- Administer IP Node Names
- Administer IP Network Region and IP Codec Set
- Administer SIP Trunk to Session Manager
- Administer AAR Call Routing

Use the System Access Terminal (SAT) to configure Communication Manager and log in with appropriate credentials.

Note: It is assumed that basic configuration, such as voicemail coverage, has already been configured. The SIP station configuration for MH-CURE SIP clients is configured through Avaya Aura® System Manager in **Section 6.2**.

5.1. Verify 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 that will be deployed.

display system-parameters customer-options		Page 1 of 12
OPTIONAL FEATURES		
G3 Version: V17	Software Package: Enterprise	
Location: 2	System ID (SID): 1	
Platform: 28	Module ID (MID): 1	
		USED
Platform Maximum Ports:	6400	180
Maximum Stations:	2400	29
Maximum XMOBILE Stations:	2400	0
Maximum Off-PBX Telephones - EC500:	9600	2
Maximum Off-PBX Telephones - OPS:	9600	14
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

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 (*sm15018*). 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	
ams1509	10.64.150.9	
cm102201	10.64.102.201	
cms15012	10.64.150.12	
default	0.0.0.0	
ipol15050	10.64.150.50	
msgserver	10.64.150.15	
procr	10.64.150.14	
procr6	::	
sm102204	10.64.102.204	
sm15018	10.64.150.18	

5.3. Administer IP Network Region and IP Codec Set

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 Avaya G450 Media Gateway or Avaya Aura® 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.

```
change ip-network-region 1                                     Page 1 of 20

                                IP NETWORK REGION

Region: 1
Location: 1      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
```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to MH-CURE SIP clients. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set '1' was specified in IP Network Region '1' shown above. MH-CURE SIP clients were tested using G.711Mu codec.

```
change ip-codec-set 1                                     Page 1 of 2

                                IP CODEC SET

Codec Set: 1

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

5.4. 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 Communication Manager (*procr*) and the 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.
- 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.
- 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*.

add signaling-group 2		Page 1 of 2
SIGNALING GROUP		
Group Number: 2	Group Type: sip	
IMS Enabled? n	Transport Method: tls	
Q-SIP? n		
IP Video? y	Priority Video? n	Enforce SIPS URI for SRTP? y
Peer Detection Enabled? y	Peer Server: SM	
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: sm15018	
Near-end Listen Port: 5061	Far-end Listen Port: 5061	
	Far-end Network Region: 1	
Far-end Domain:		
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 SIP clients, Avaya Endpoints, and Communication Manager Messaging. 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 22
                                     TRUNK GROUP

Group Number: 2                                     Group Type: sip          CDR Reports: y
  Group Name: sm15018                               COR: 1          TN: 1          TAC: 102
  Direction: two-way                                Outgoing Display? n
  Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: tie                                   Auth Code? n
                                                Member Assignment Method: auto
                                                Signaling Group: 2
                                                Number of Members: 20
  
```

5.5. AAR Call Routing

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 “5” to route pattern 2 as shown below.

```

change aar analysis 5                               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
  5              5   5      2         lev0          n
  
```

Configure a preference in **Route Pattern 2** to route calls over SIP trunk group 2 as shown below.

```

change route-pattern 2                               Page 1 of 3
      Pattern Number: 2          Pattern Name: sm15018
  SCCAN? n      Secure SIP? n      Used for SIP stations? n

  Grp FRL NPA Pfx Hop Toll No.  Inserted          DCS/ IXC
  No          Mrk Lmt List Del  Digits          QSIG
                                     Dgts          Intw
  1: 2        0                  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      lev0-pvt  none
  2: y y y y y n      n      rest      none
  
```

6. Configure Avaya Aura® Session Manager

This section provides the procedure for configuring Session Manager. The procedures include the following areas:

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

Note: It is assumed that basic configuration of Session Manager has already been performed. This section will focus on the configuration of a SIP user for MH-CURE SIP clients.

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 SIP clients

From the System Manager **Home** screen, select **Elements** → **Routing** → **SIP Entities** (not shown) and edit the SIP Entity for Session Manager shown below.

The screenshot shows the Avaya Aura System Manager 7.1 interface. The top navigation bar includes 'Home' and 'Routing'. The left sidebar lists various configuration options, with 'SIP Entities' selected. The main content area is titled 'SIP Entity Details' and shows the 'General' tab for the entity 'sm15018'. The settings include:

- Name: sm15018
- FQDN or IP Address: 10.64.150.18
- Type: Session Manager
- Notes: (empty)
- Location: Lab
- Outbound Proxy: (empty)
- Time Zone: America/Denver
- Minimum TLS Version: Use Global Setting
- Credential name: (empty)

Buttons for 'Commit', 'Cancel', and 'Help' are visible at the top right of the form.

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

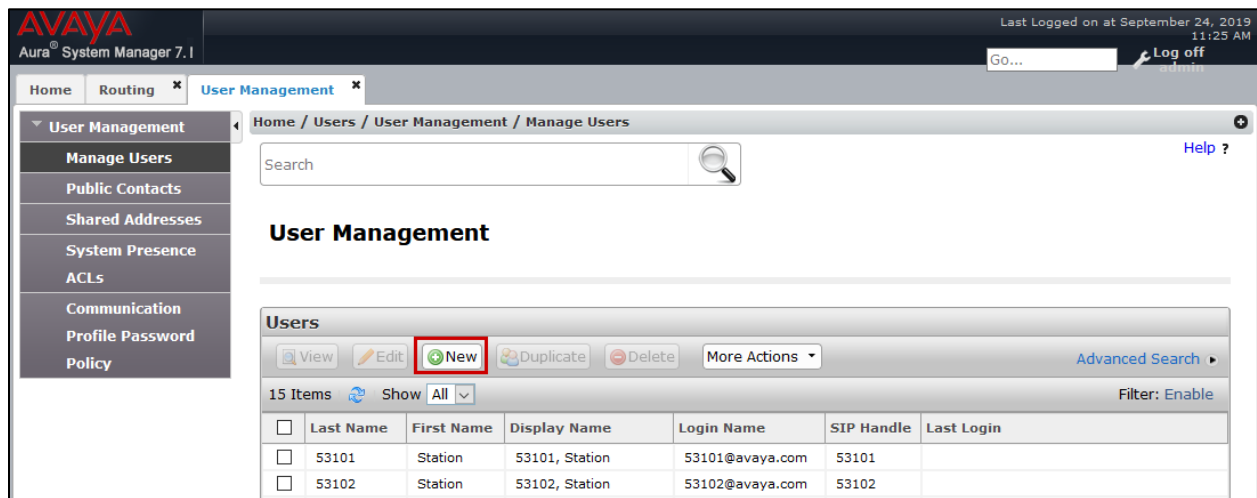
The screenshot shows the 'Listen Ports' configuration section. It contains a table with 4 items. The first item, 5060 TCP, is highlighted with a red box. The table columns are Listen Ports, Protocol, Default Domain, Endpoint, and Notes.

Listen Ports	Protocol	Default Domain	Endpoint	Notes
5060	TCP	avaya.com	<input checked="" type="checkbox"/>	
5060	UDP	avaya.com	<input checked="" type="checkbox"/>	
5061	TLS	avaya.com	<input checked="" type="checkbox"/>	
5062	TLS	avaya.com	<input type="checkbox"/>	

Buttons for 'Add' and 'Remove' are at the top left. A 'Filter: Enable' link is at the top right. A 'Select: All, None' link is at the bottom left.

6.3. Administer SIP User

In the **Home** 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 **User Profile | Add** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter “<ext>@<domain>”, where “<ext>” is the desired MH-CURE SIP client extension and “<domain>” is the applicable SIP domain name from **Section 5.3**. Retain the default values in the remaining fields.

The screenshot shows the Avaya Aura System Manager 7.1 User Profile | Add screen. The 'Identity' tab is selected. The 'Last Name' field is set to '53121', the 'First Name' field is set to 'Station', and the 'Login Name' field is set to '53121@avaya.com'. These fields are highlighted with red boxes.

User Provisioning Rule:

Identity

* Last Name:

Last Name (Latin Translation):

* First Name:

First Name (Latin Translation):

Middle Name:

Description:

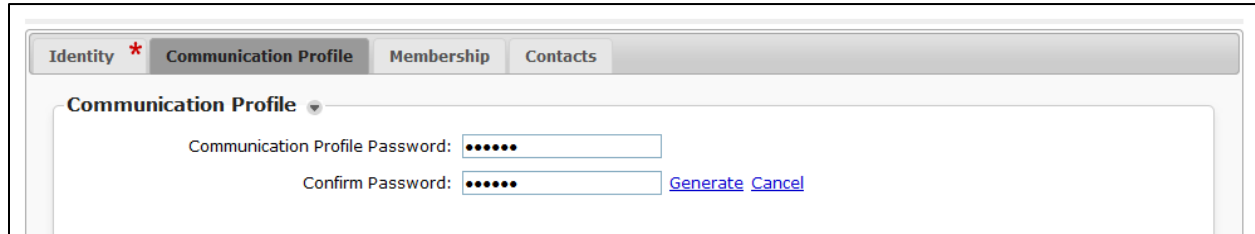
Update Time:

* Login Name:

Email Address:

6.3.2. Communication Profile

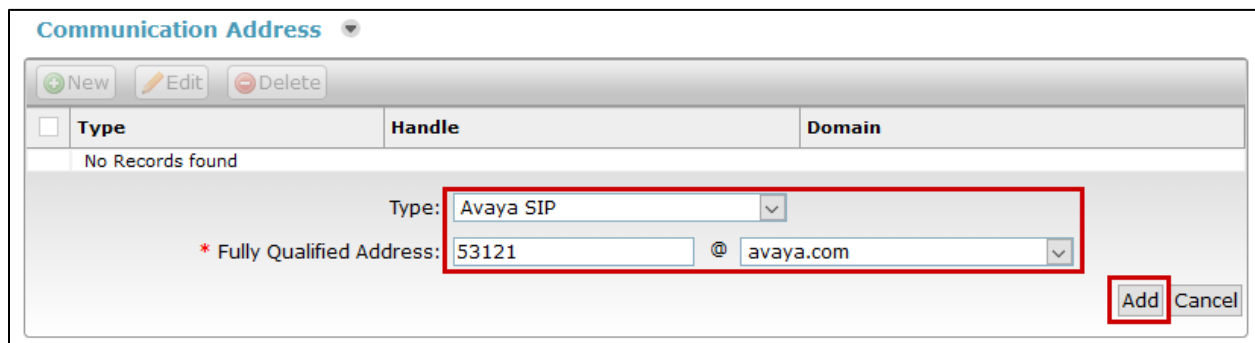
Select the **Communication Profile** tab. Next, click on **Communication Profile Password**. For **Comm-Profile Password** and **Re-enter Comm-Profile Password**, enter the desired password for the MH-CURE SIP user to use for registration. Click **OK**.



The screenshot shows a web interface with four tabs: Identity, Communication Profile, Membership, and Contacts. The 'Communication Profile' tab is selected. Below the tabs, there is a section titled 'Communication Profile'. Inside this section, there are two password input fields: 'Communication Profile Password' and 'Confirm Password'. Both fields contain six dots. To the right of the 'Confirm Password' field are two links: 'Generate' and 'Cancel'.

6.3.3. Communication Address

Under **Communication Address** click **New** to add a new entry. The **Communication Address Add/Edit** dialog box is displayed as shown below. For **Type**, select *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 **Add**.



The screenshot shows a 'Communication Address' dialog box. At the top, there are three buttons: 'New', 'Edit', and 'Delete'. Below these buttons is a table with three columns: 'Type', 'Handle', and 'Domain'. The table is empty, and a message 'No Records found' is displayed. Below the table, there is a 'Type' dropdown menu with 'Avaya SIP' selected. To the right of the 'Type' dropdown is a 'Fully Qualified Address' field. This field contains the text '53121' followed by an '@' symbol and a dropdown menu with 'avaya.com' selected. At the bottom right of the dialog box are two buttons: 'Add' and 'Cancel'. The 'Add' button is highlighted with a red box.

6.3.4. Session Manager Profile

Check box for **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, and **Termination Application Sequence**, select the values corresponding to the applicable Session Manager and Communication Manager. Select Retain the default values in the remaining fields. For **Call Routing Settings** section to configure the **Home Location**.

☒ **Session Manager Profile**

SIP Registration

* Primary Session Manager

sm15018

Primary	Secondary	Maximum
11	0	11

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices

1

Block New Registration When Maximum Registrations Active?

Application Sequences

Origination Sequence

cm15014

Termination Sequence

cm15014

Emergency Calling Application Sequences

Emergency Calling Origination Sequence

(None)

Emergency Calling Termination Sequence

(None)

Call Routing Settings

* Home Location

Lab

Conference Factory Set

(None)

KJA; Draft:
SPOC 1/17/2020

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6.3.5. CM Endpoint Profile

Check box for **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 *9630SIP_DEFAULT_CM_7_1*. Retain the default values in the remaining fields. Click on the Endpoint Editor (i.e, Edit icon in Extension field) to configure the other options. Three MH-CURE SIP endpoints were used during compliance test. Each of the MH-CURE SIP endpoint extension were configured with different **Template**: *9630SIP_DEFAULT_CM_7_1*, *9608SIP_DEFAULT_CM_7_1* and *9641SIP_DEFAULT_CM_7_1*.

☒ **CM Endpoint Profile**

* Systemcm15014

* Profile TypeEndpoint

Use Existing Endpoints☐

Display Extension Ranges

* Extension53121Endpoint Editor

Template9630SIP_DEFAULT_CM_7_1

Set Type9630SIP

Security Code

PortIP

Voice Mail Number

Preferred Handle(None)

Navigate to the **General Options** tab and set the **Coverage Path 99** field to the voicemail coverage path.

The screenshot shows the 'General Options (G)' tab selected. The 'Coverage Path 1' field is highlighted with a red box and contains the value '99'. Other fields include 'Class of Restriction (COR)' set to '1', 'Emergency Location Ext' set to '53121', 'Tenant Number' set to '1', 'SIP Trunk' set to 'aar', 'Class Of Service (COS)' set to '1', 'Message Lamp Ext.' set to '53121', 'Type of 3PCC Enabled' set to 'None', 'Coverage Path 2' is empty, 'Localized Display Name' set to '53121, Station', 'Enable Reachability for Station Domain Control' is set to a dropdown, 'Lock Message' is unchecked, 'Multibyte Language' is set to 'Not Applicable', and 'SIP URI' is empty.

Select the **Feature Options** tab and check box for **IP SoftPhone**. Click **Done** (not shown) to return to the previous web page and then **Commit** to save the configuration (not shown).

The screenshot shows the 'Feature Options (F)' tab selected. The 'IP SoftPhone' checkbox is checked and highlighted with a red box. Other fields include 'Active Station Ringing' set to 'single', 'MWI Served User Type' set to 'None', 'Per Station CPN - Send Calling Number' set to 'None', 'IP Phone Group ID' is empty, 'Remote Soft Phone Emergency Calls' set to 'as-on-local', 'LWC Reception' set to 'spe', 'AUDIX Name' set to 'None', 'Speakerphone' set to '2-way', 'Short/Prefixed Registration Allowed' set to 'default', 'EC500 State' set to 'enabled', 'Auto Answer' set to 'none', 'Coverage After Forwarding' is unchecked, 'Display Language' set to 'english', 'Hunt-to Station' is empty, 'Loss Group' set to '19', 'Survivable COR' set to 'internal', 'Time of Day Lock Table' set to 'None', 'Voice Mail Number' is empty, 'Music Source' is empty, 'Always Use' is unchecked, 'Idle Appearance Preference' is unchecked, and 'IP Audio Hairpinning' is unchecked.

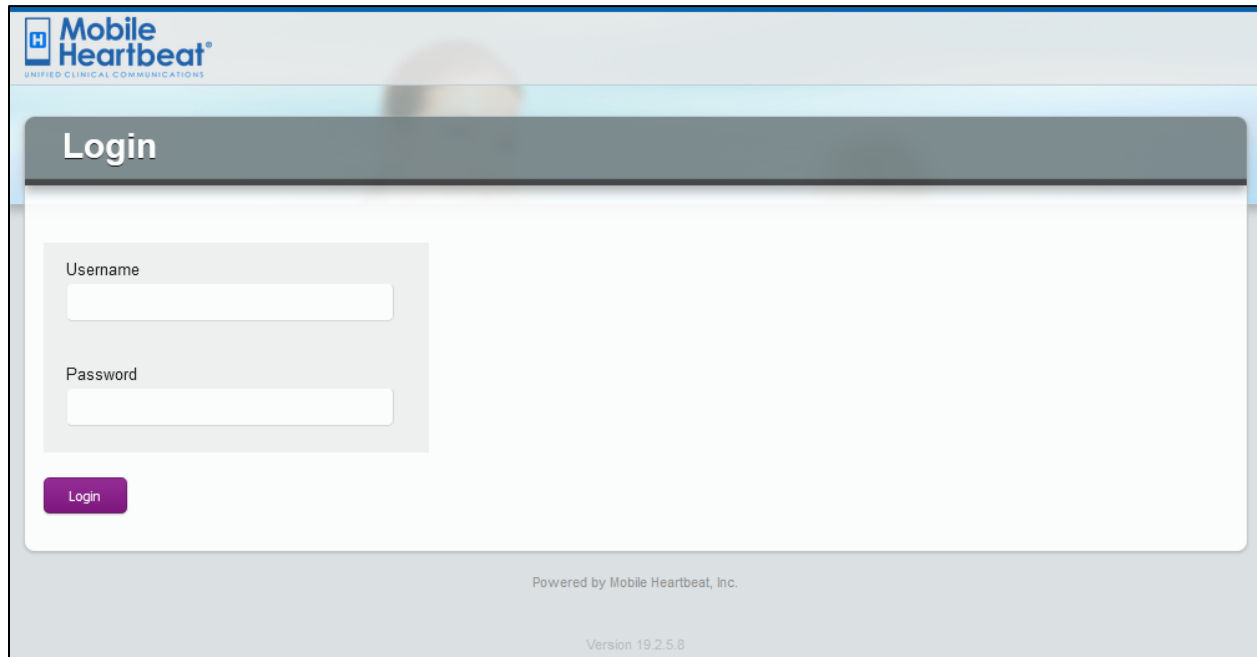
7. Configure Mobile Heartbeat MH-CURE

Configuration for MH-CURE SIP clients is performed via MH-CURE Administrative web User Interface.

- Log into MH-CURE web UI
- Configure MH-CURE for SIP Endpoint Service
- Configure Voicemail Number
- Configure User Extensions

7.1. Log into MH-CURE web UI

Via a browser, navigate to <https://<MH-CURE>/heartbeat/> where MH-CURE is the IP-Address/FQDN and port of the MH-CURE Administrative web UI. Log on 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 dark grey header bar with the word 'Login' in white. The main content area is white and contains a login form with two input fields: 'Username' and 'Password'. Below these fields is a purple 'Login' button. At the bottom of the page, there is a footer with the text 'Powered by Mobile Heartbeat, Inc.' and 'Version 19.2.5.8'.

7.2. Configure MH-CURE for SIP Endpoint Service

Click **Hospitals** in top menu. On the left side expand tree, and click on the hospital name, “MH Example Hospital” in this case. Click on the **Settings..** button.

The screenshot displays the 'Mobile Heartbeat' web interface. The top navigation bar includes links for Patients, Hospitals (highlighted), Users, Settings, Reports, Monitors, Tools, and Logout. The left sidebar shows a tree view with 'Hospitals' expanded and 'MH Example Hospital' selected. The main content area is titled 'Manage Hospitals and Units' and contains a form for 'Manage Hospital - MH Example Hospital'. This form includes input fields for 'Name *' (MH Example Hospital), 'Abbreviation' (MH Example Hospital), and 'QuickLinks (Personal Devices)' (mhavaya-p), along with 'Update' and 'Delete' buttons. Below the form is an 'Add Units' section with a text input and an 'Add Units' button. At the bottom, the 'Hospital Settings' section features a 'Settings...' button, which is highlighted with a red box.

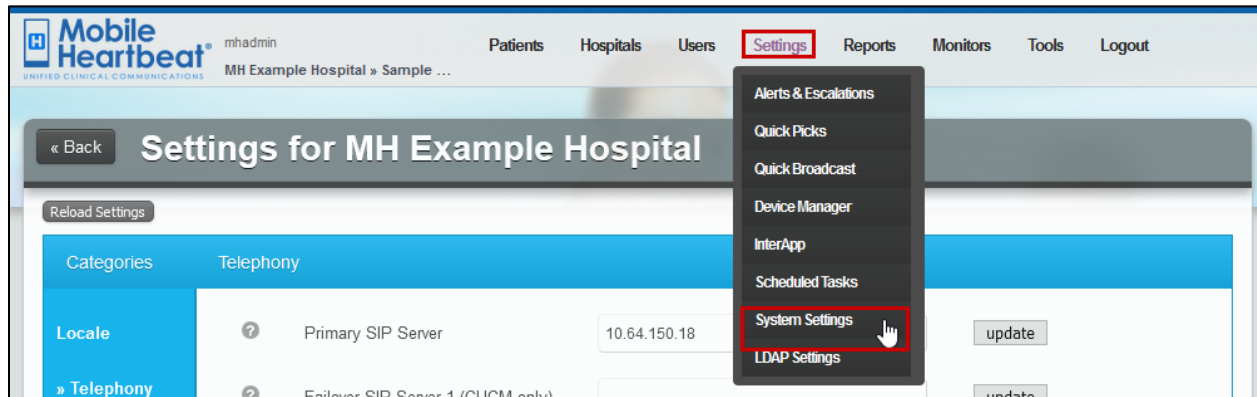
Click **Telephony** on the left side. Change **Primary SIP Server** to IP Address of Session Manager from **Section 5.2**, and click the Update button on right.

The screenshot shows the 'Settings for MH Example Hospital' interface. The top navigation bar includes 'Patients', 'Hospitals', 'Users', 'Settings', 'Reports', 'Monitors', 'Tools', and 'Logout'. The 'Settings' section is active, and the 'Telephony' category is selected in the left sidebar. The 'Primary SIP Server' field is highlighted with a red box, showing the IP address '10.64.150.18' and an 'update' button. Below it are three 'Failover SIP Server' fields, each with an 'update' button.

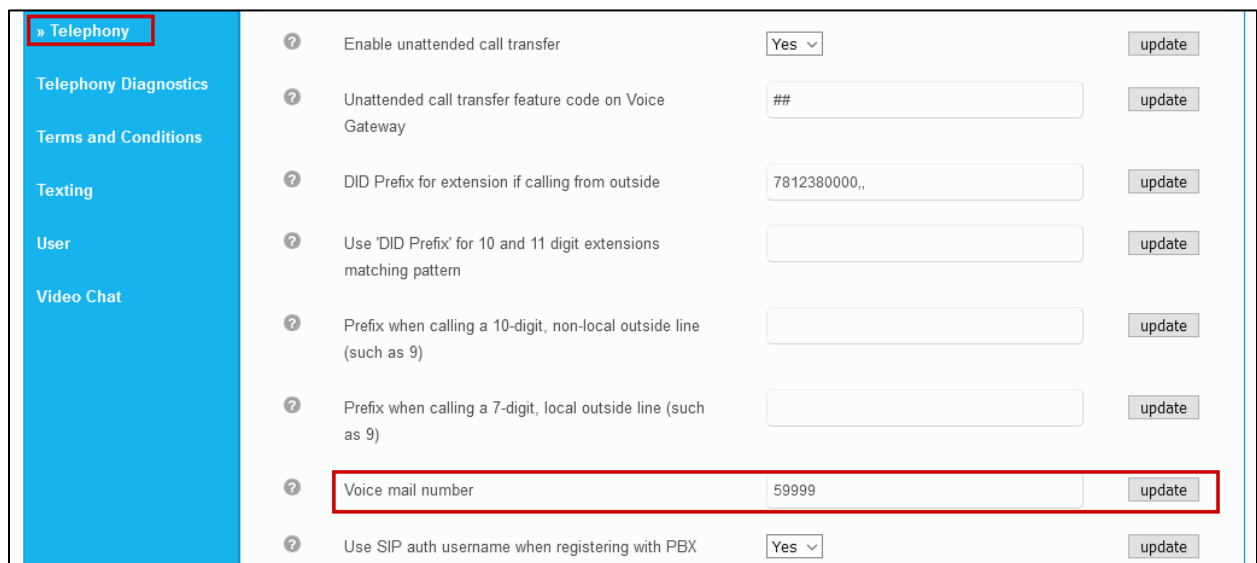
Categories	Telephony
Locale	<div>Primary SIP Server 10.64.150.18 update</div>
» Telephony	<div>Failover SIP Server 1 (CUCM only) update</div>
	<div>Failover SIP Server 2 (CUCM only) update</div>
	<div>Failover SIP Server 3 (CUCM only) update</div>

7.3. Configure Voicemail Number

From the top, navigate to **Settings** → **System Settings**.

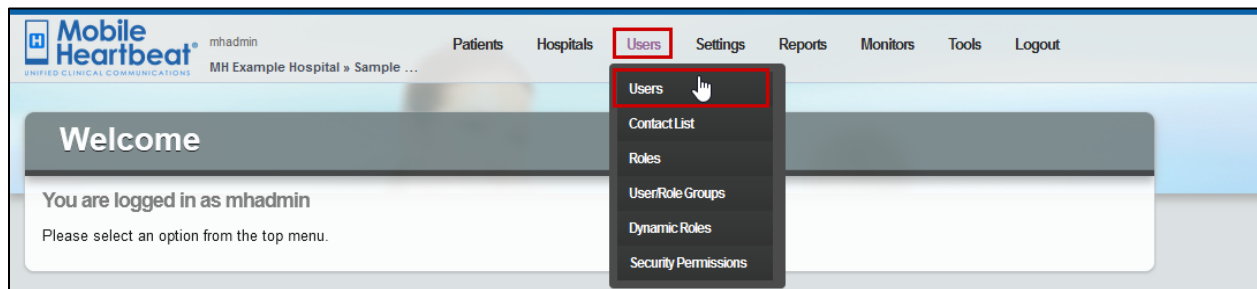


Select **Telephony** on left side. Scroll down and locate **Voice mail Number** field. Type in the pilot number used in the environment. During the compliance test, 59999 was configured as the Voicemail pilot number for Communication Manager Messaging.

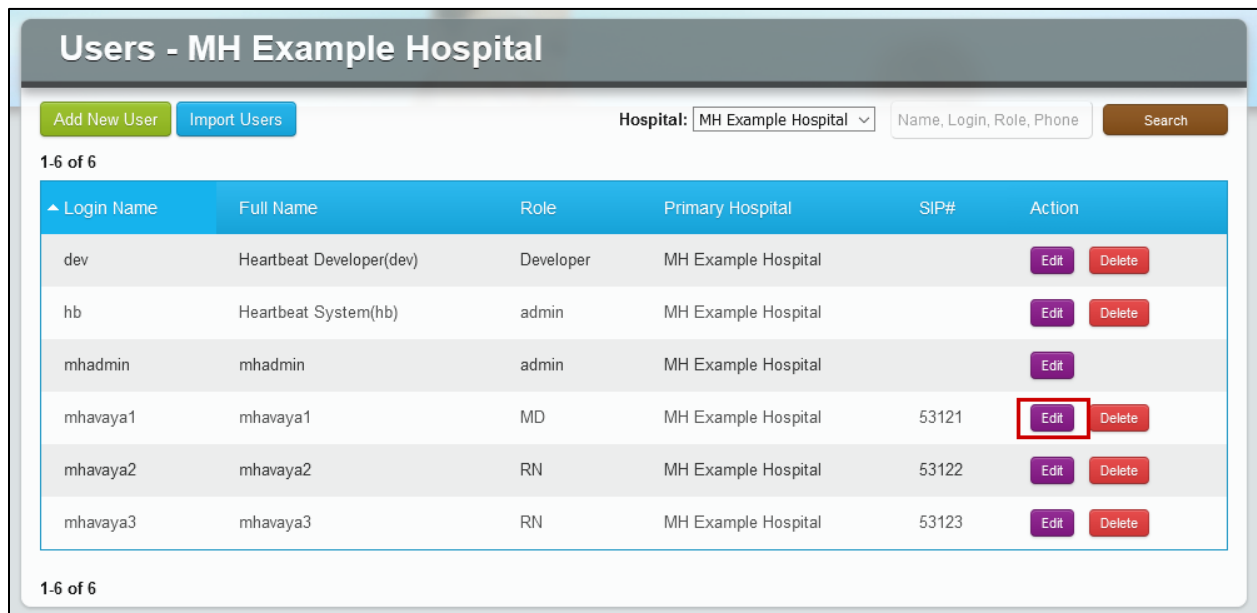


7.4. Configure User Extensions

From the top, navigate to **Users** → **Users**.



Edit an existing user to configure the 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 **6.3.2**. Once done click **Update** at the bottom of the screen (not shown).

SIP Number:	<input type="text" value="53121"/>
SIP Password:	<input type="text" value="123456"/>
SIP Auth Username:	<input type="text" value="53121"/>

8. Verification Steps

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

1. Verify that MH-CURE SIP clients have successfully registered with Session Manager. In System Manager, navigate to **Elements → Session Manager → System Status → User Registrations** to check the registration status. During compliance test, extensions 53121, 53122 and 53123 were used by MH-CURE SIP clients to register to Session Manager.

User Registrations											
Select rows to send notifications to devices. Click on Details column for complete registration status.											
Customize											
View Default Export Force Unregister AST Device Notifications: Reboot Reload Failback As of 12:50 PM											
12 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 Prim
<input type="checkbox"/>	Show	53101@avaya.com	Station	53101	---	10.64.10.211	<input type="checkbox"/>	<input type="checkbox"/>	1/3	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)
<input type="checkbox"/>	Show	53102@avaya.com	Station	53102	---	10.64.10.203	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)
<input type="checkbox"/>	Show	53121@avaya.com	Station	53121	---	10.64.10.205	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>
<input type="checkbox"/>	Show	53122@avaya.com	Station	53122	---	10.64.10.206	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>
<input type="checkbox"/>	Show	53123@avaya.com	Station	53123	---	10.64.10.221	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>
<input type="checkbox"/>	Show	---	Station	53131	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>

2. Establish a call between MH-CURE SIP client and a local Avaya Endpoint. The **status trunk** command may be used to view the active call status. The trunk that is being monitored here is the trunk to Session Manager. This command should specify the trunk group and trunk member used for the call. On **Page 2**, **Audio Connection Type** will set to *ip-direct* if the call is shuffled. The **Codec Type** is also displayed.

```
status trunk 2/1                                     Page 2 of 3
CALL CONTROL SIGNALING
Near-end Signaling Loc: PROCR
  Signaling   IP Address      Port
  Near-end:   10.64.150.14     : 5061
  Far-end:    10.64.150.18     : 5061
H.245 Near:
H.245 Far:
H.245 Signaling Loc:      H.245 Tunneled in Q.931? no
Audio Connection Type: ip-direct      Authentication Type: None
  Near-end Audio Loc:      Codec Type: G.711MU
  Audio      IP Address      Port
  Near-end:   10.64.10.202    : 3098
  Far-end:    10.64.10.206    : 19858
Video Near:
Video Far:
Video Port:
Video Near-end Codec:      Video Far-end Codec:
```

3. While the call is active, basic telephony features can be exercised to verify proper operation.

9. Conclusion

These Application Notes described the configuration steps required to integrate Mobile Heartbeat MH-CURE SIP clients with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. MH-CURE SIP clients were able to establish calls with Avaya H.323 / SIP Endpoints and the PSTN with SIP/TCP and RTP/UDP. In addition, basic telephony features were verified. All feature and serviceability test cases were completed successfully with observations noted in **Section 2.2**.

10. References

This section references the Avaya and Mobile Heartbeat documentation relevant to these Application Notes. The Avaya product documentation is available at <http://support.avaya.com>

- [1] *Administering Avaya Aura® Communication Manager*, Release 7.1.3, Issue 1, July 2018.
- [2] *Administering Avaya Aura® System Manager for Release*, Release 7.1.3, Issue 4, September 2017.
- [3] *Administering Avaya Aura® Session Manager*, Release 7.1.3, Issue 4, July 2018.

Documentation related to MH-CURE SIP clients can be directly obtained from Mobile Heartbeat.

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