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.

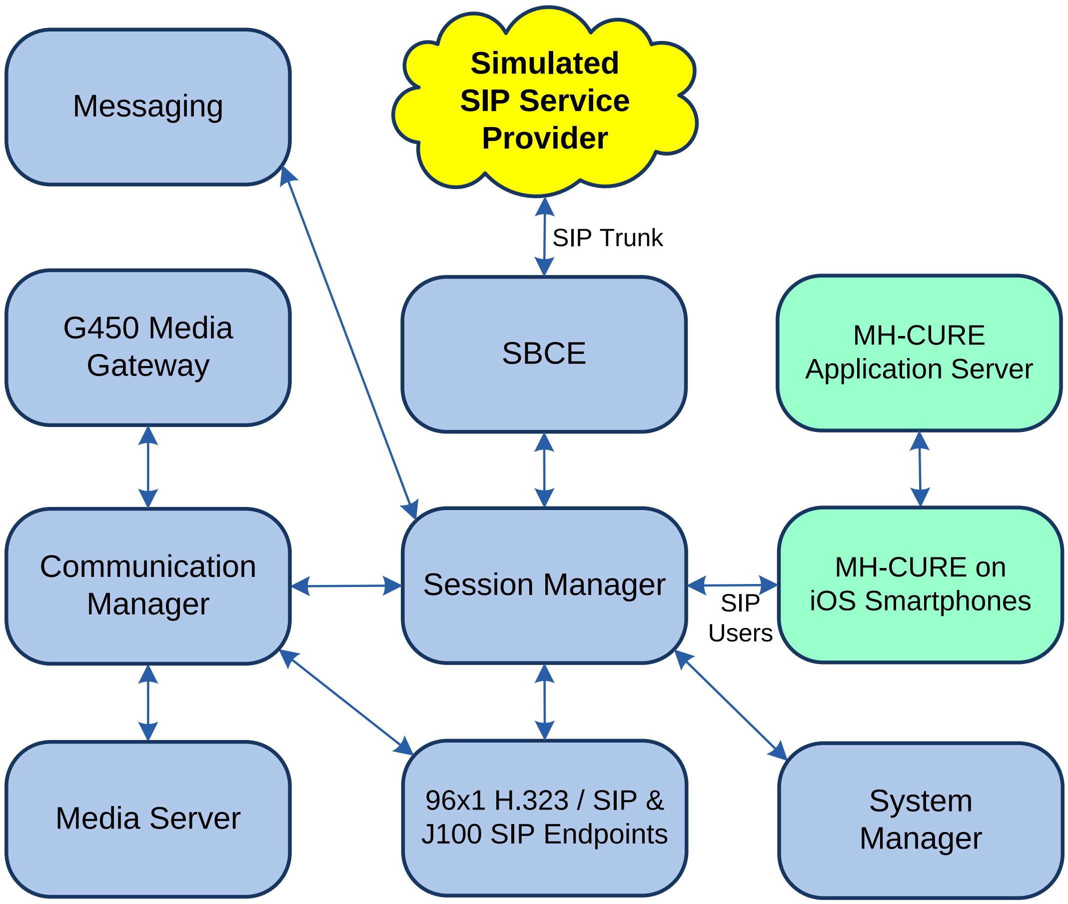


Figure 1: Avaya SIP Telephony Network with MH-CURE

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        Silence      Frames   Packet
    Codec        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: 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                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: 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
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for 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
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              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
                                                UUI Treatment: shared
                                              Maximum Size of UUI 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
                             Dgts                                      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

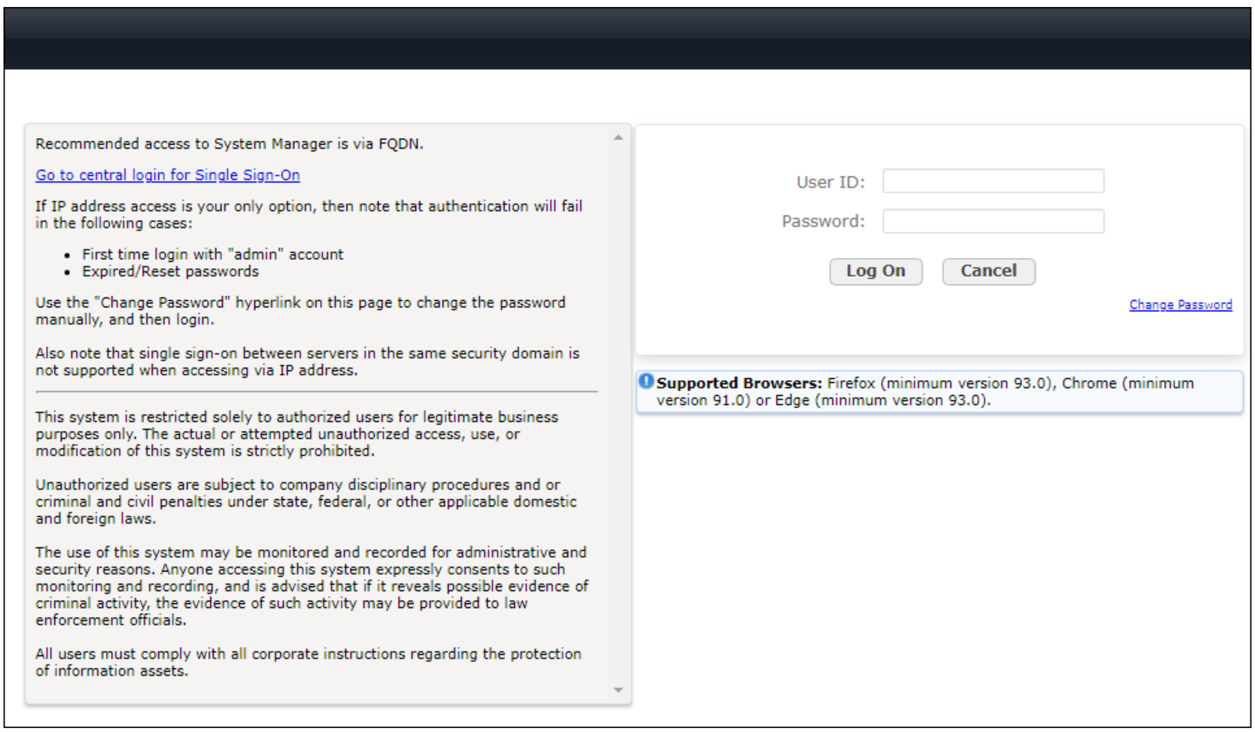
This section covers the procedure for adding a SIP user in Session Manager. The configuration covers:

- 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.

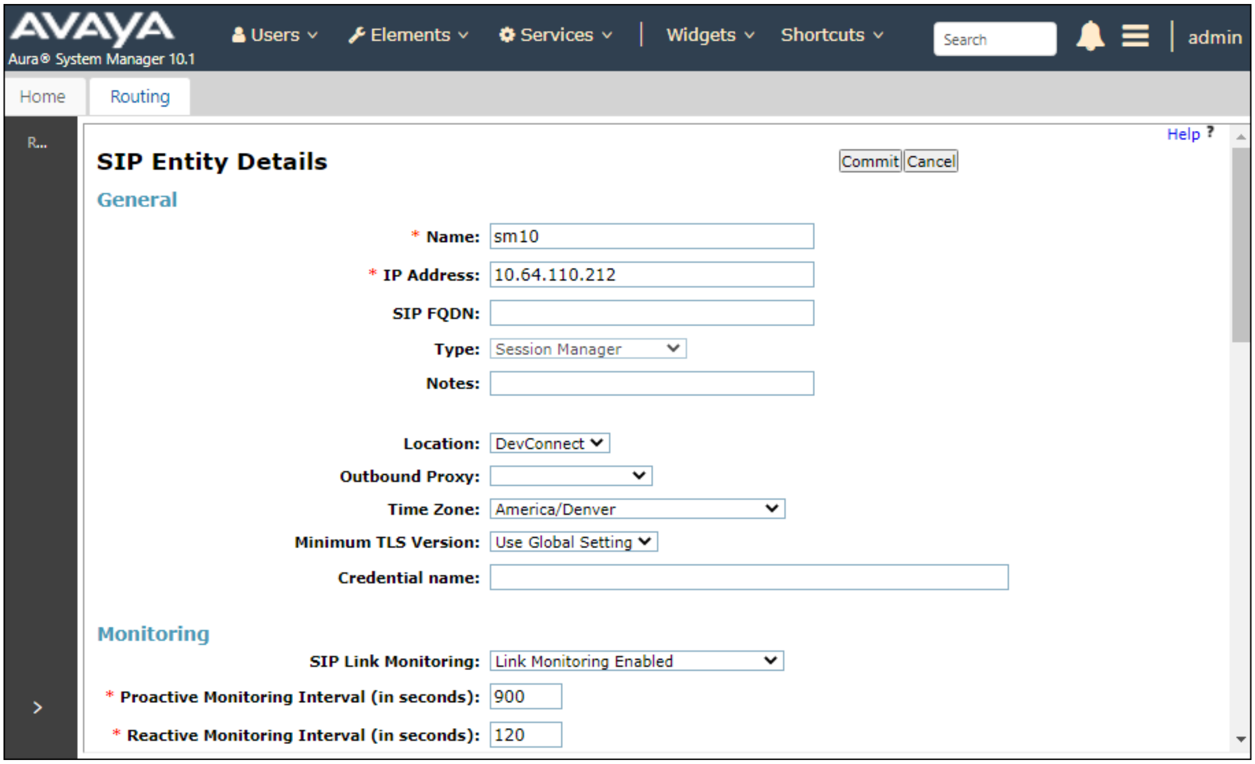
6.1. Launch System Manager

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.

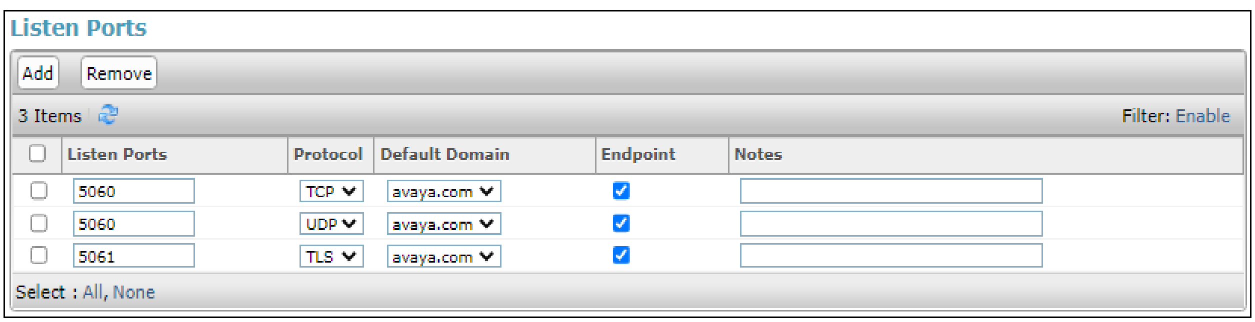


6.2. Set Network Transport Protocol for MH-CURE Clients

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

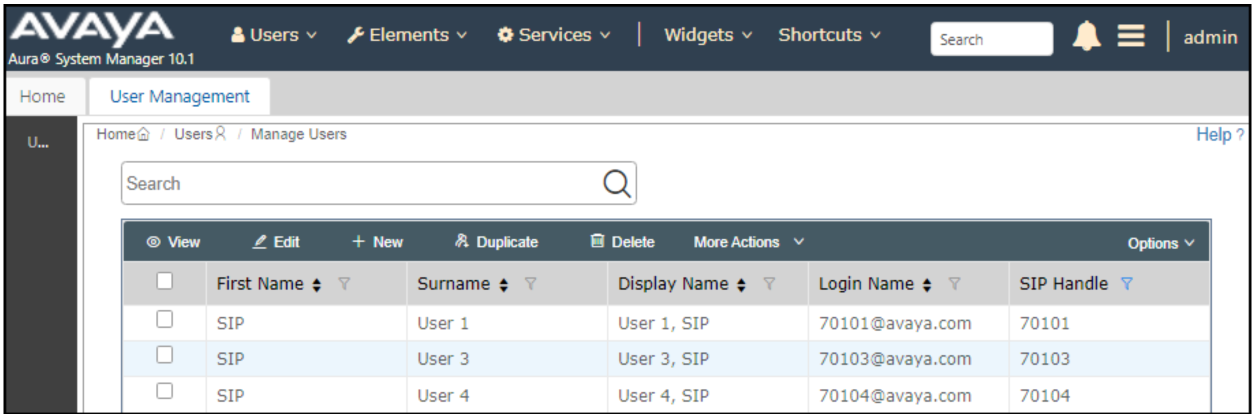


Scroll down to the **Listen Ports** section and verify that the transport network protocol used by MH-CURE clients is specified in the list below. For the compliance test, the solution used TCP network transport.



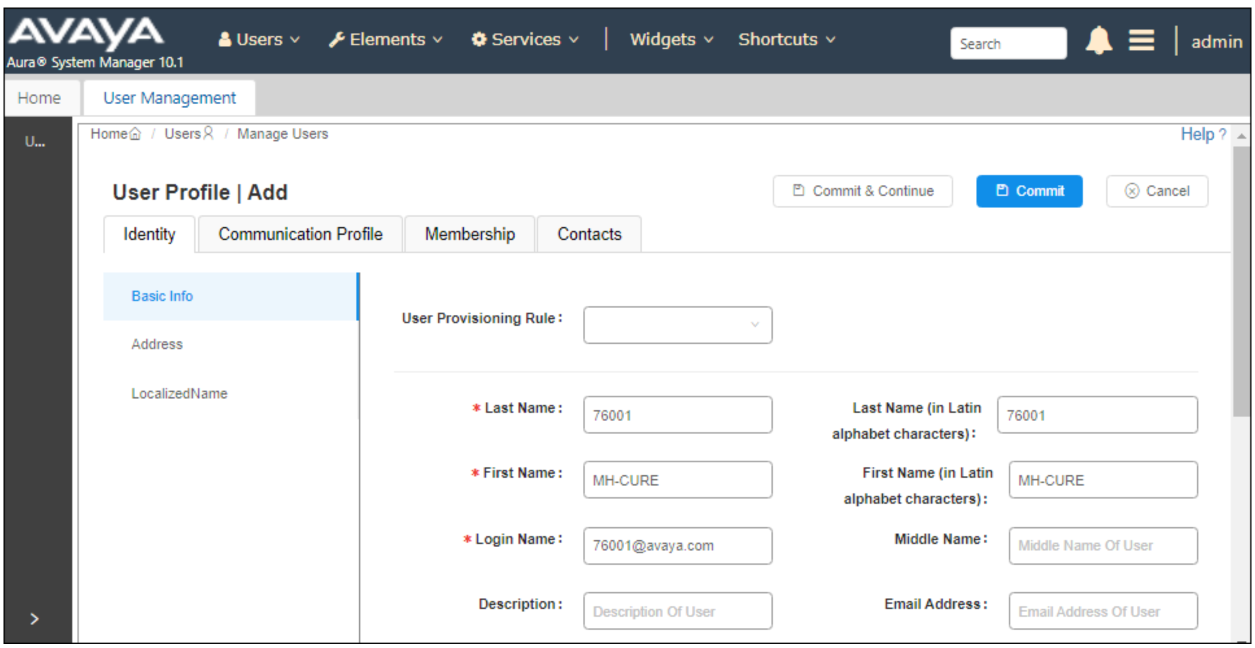
6.3. Administer SIP User

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.



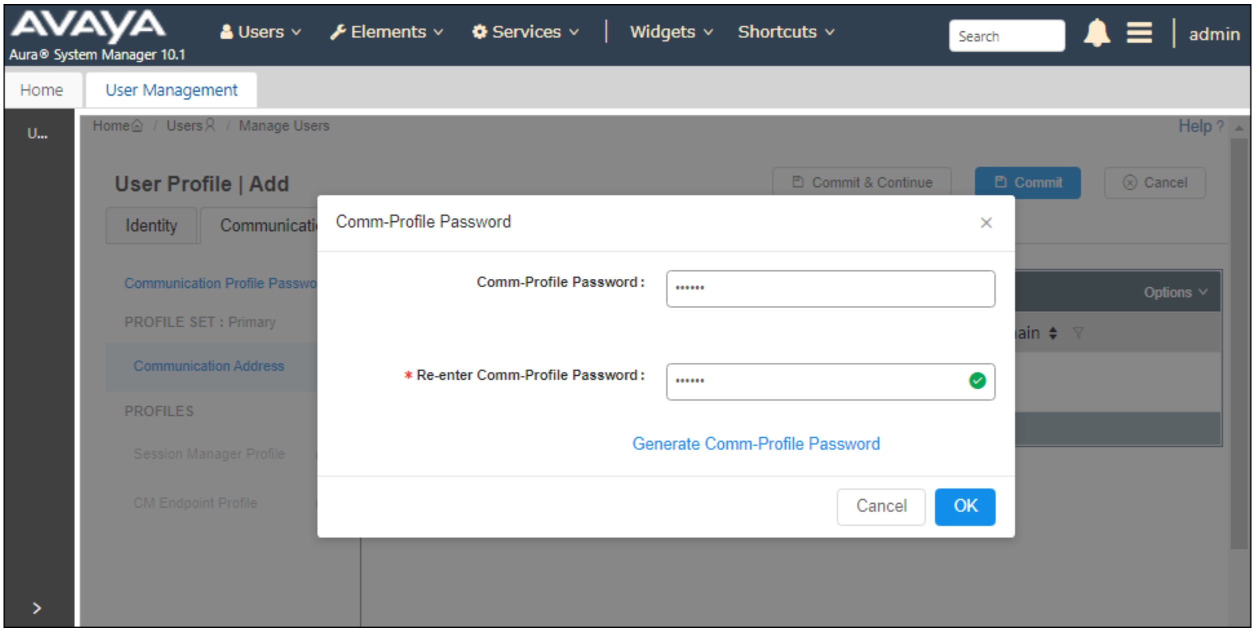
6.3.1. Identity

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.



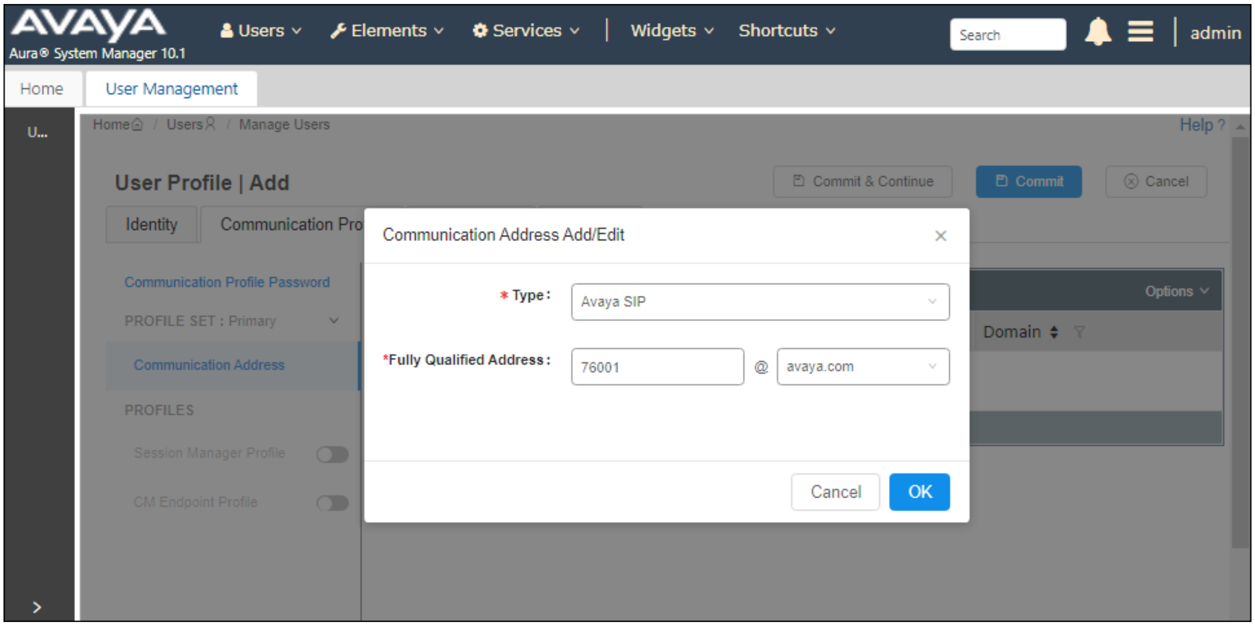
6.3.2. Communication Profile

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**.



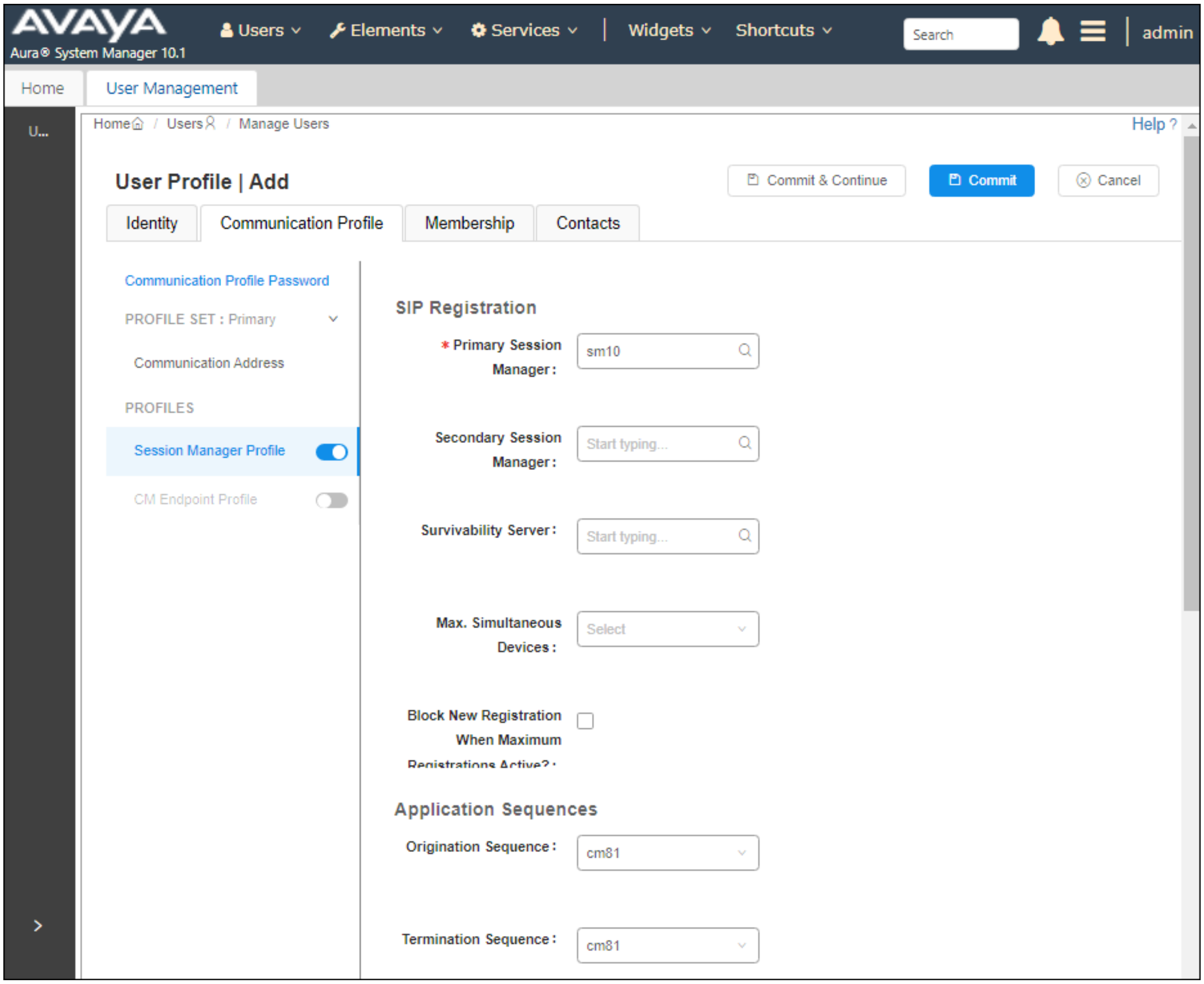
6.3.3. Communication Address

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**.

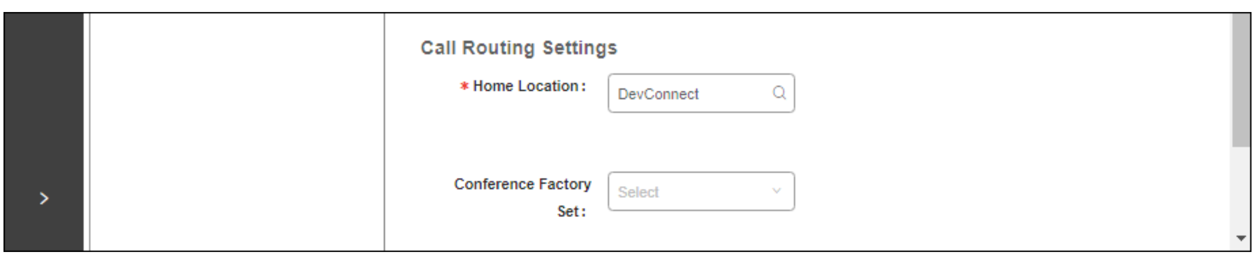


6.3.4. Session Manager Profile

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.

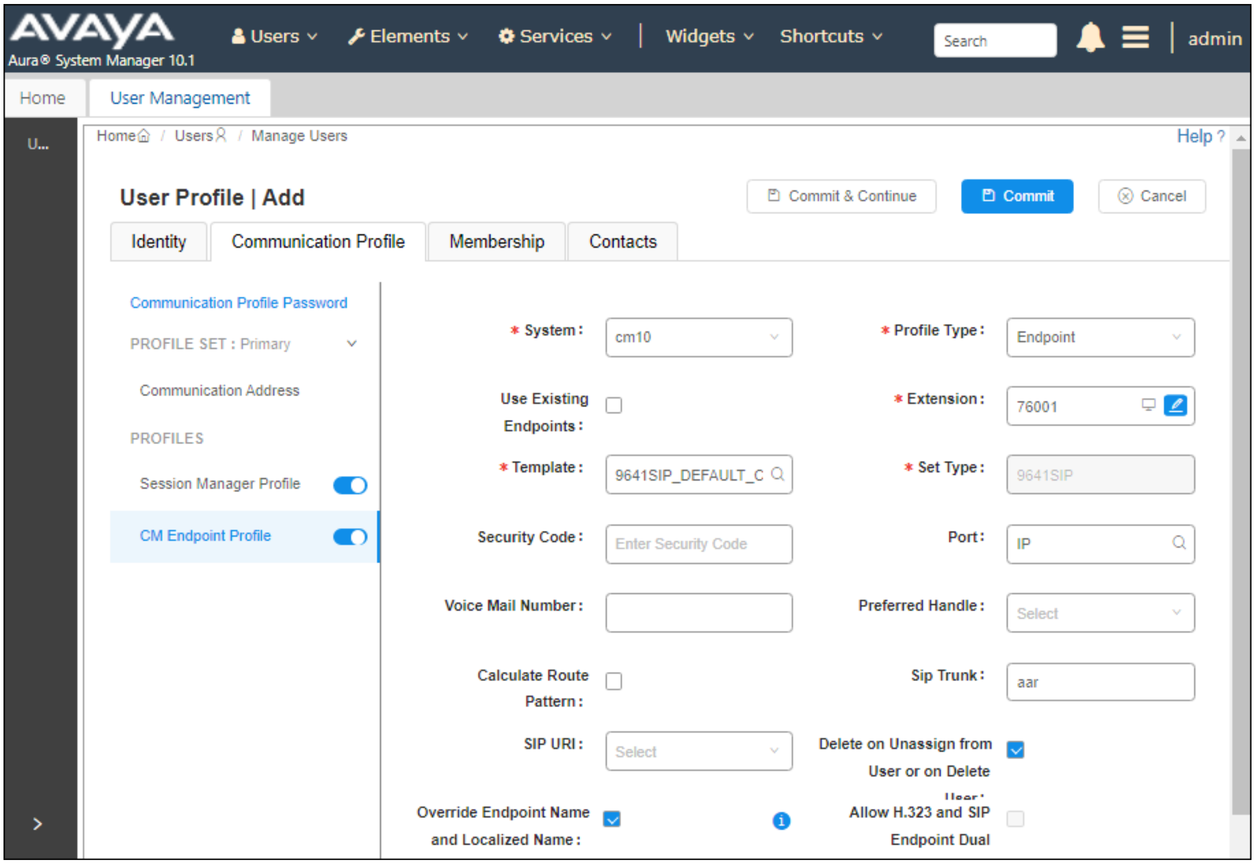


Scroll down to the **Call Routing Settings** section to configure the **Home Location**.

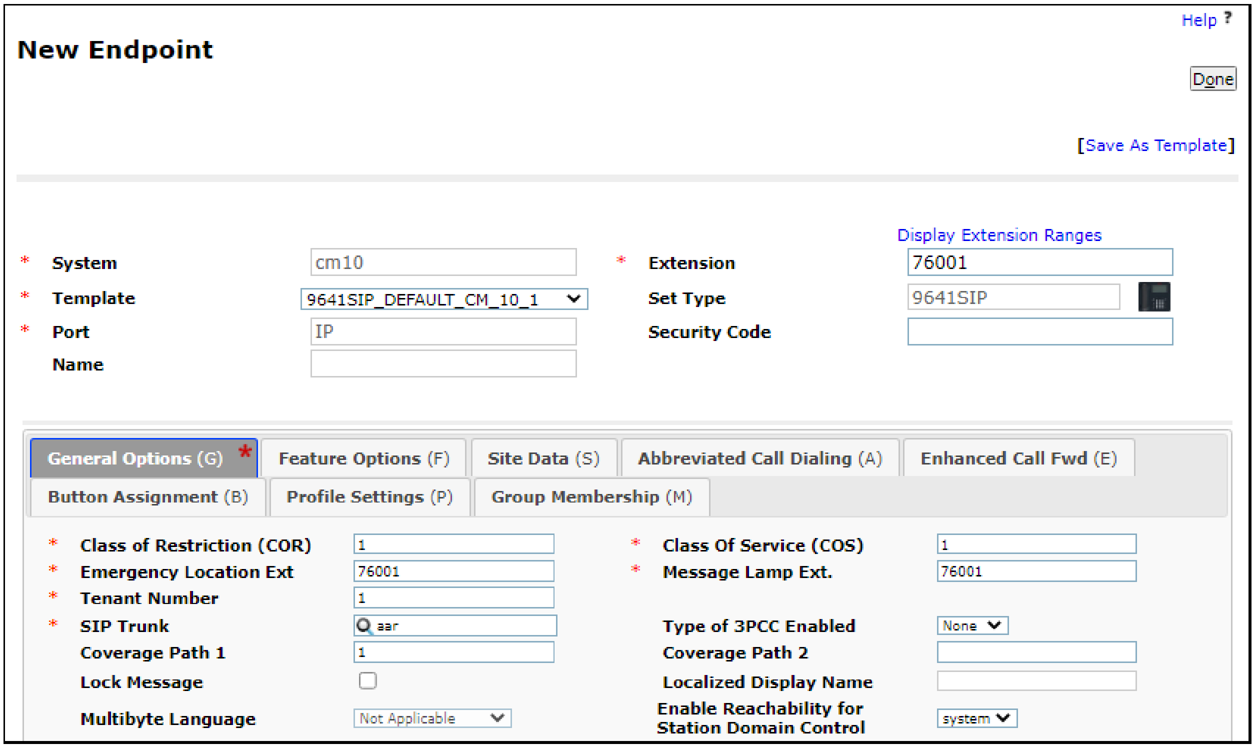


6.3.5. CM Endpoint Profile

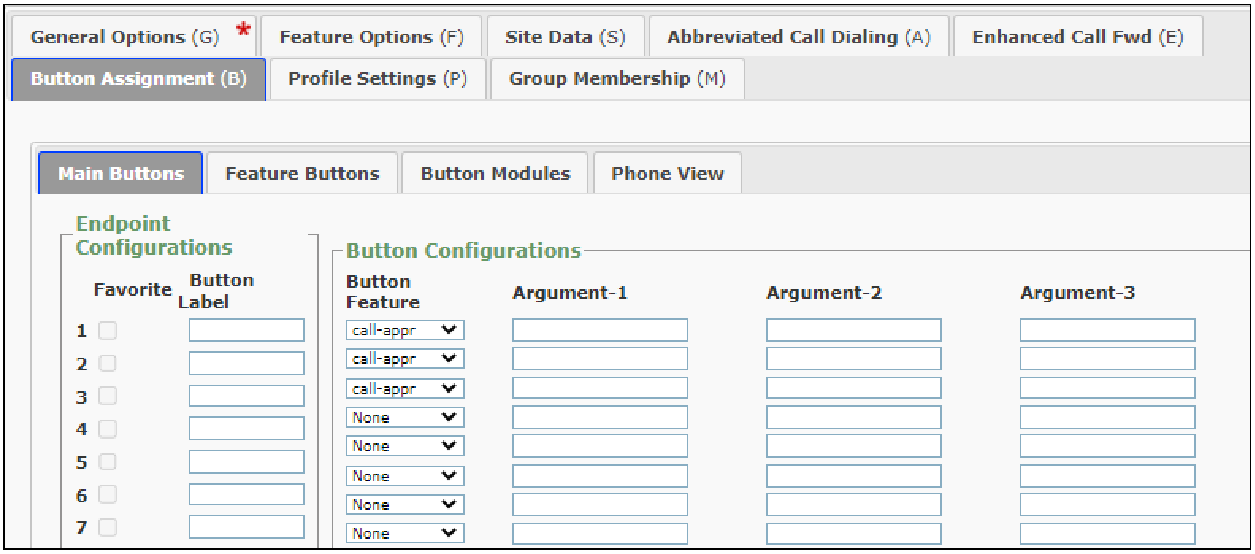
Click on the toggle button by **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.1**. For **Template**, select *9641SIP_DEFAULT_CM_10_1*. For **Port**, click and select *IP*.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



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.



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



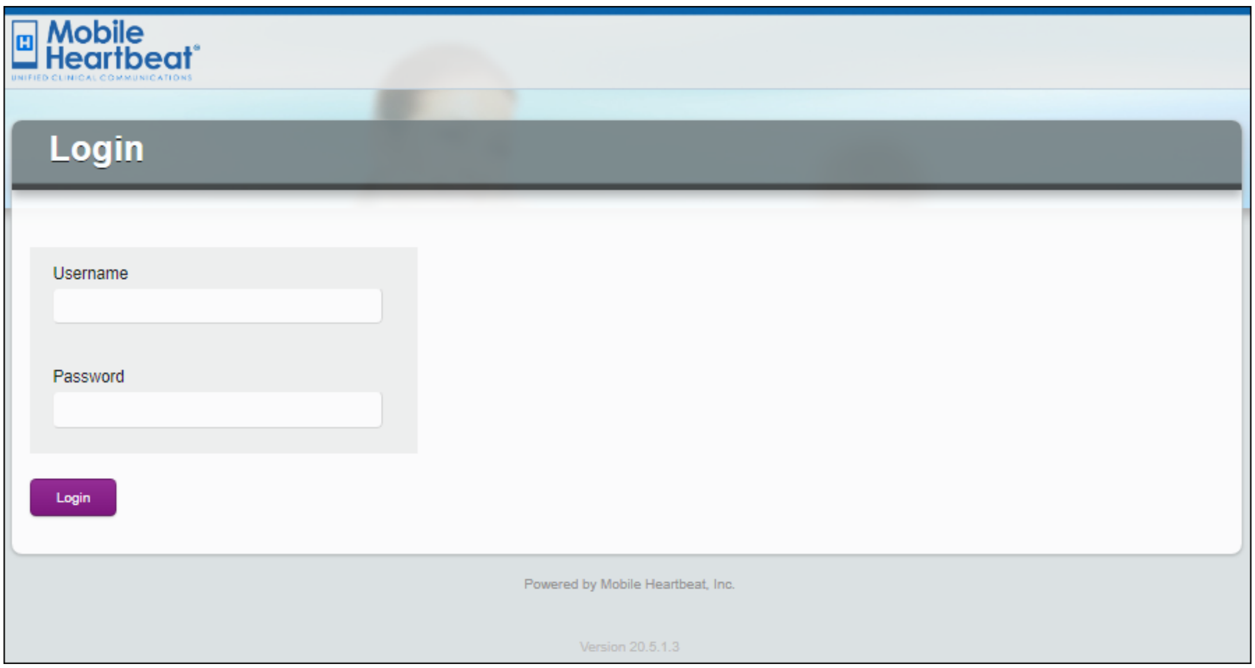
7. Configure MH-CURE

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

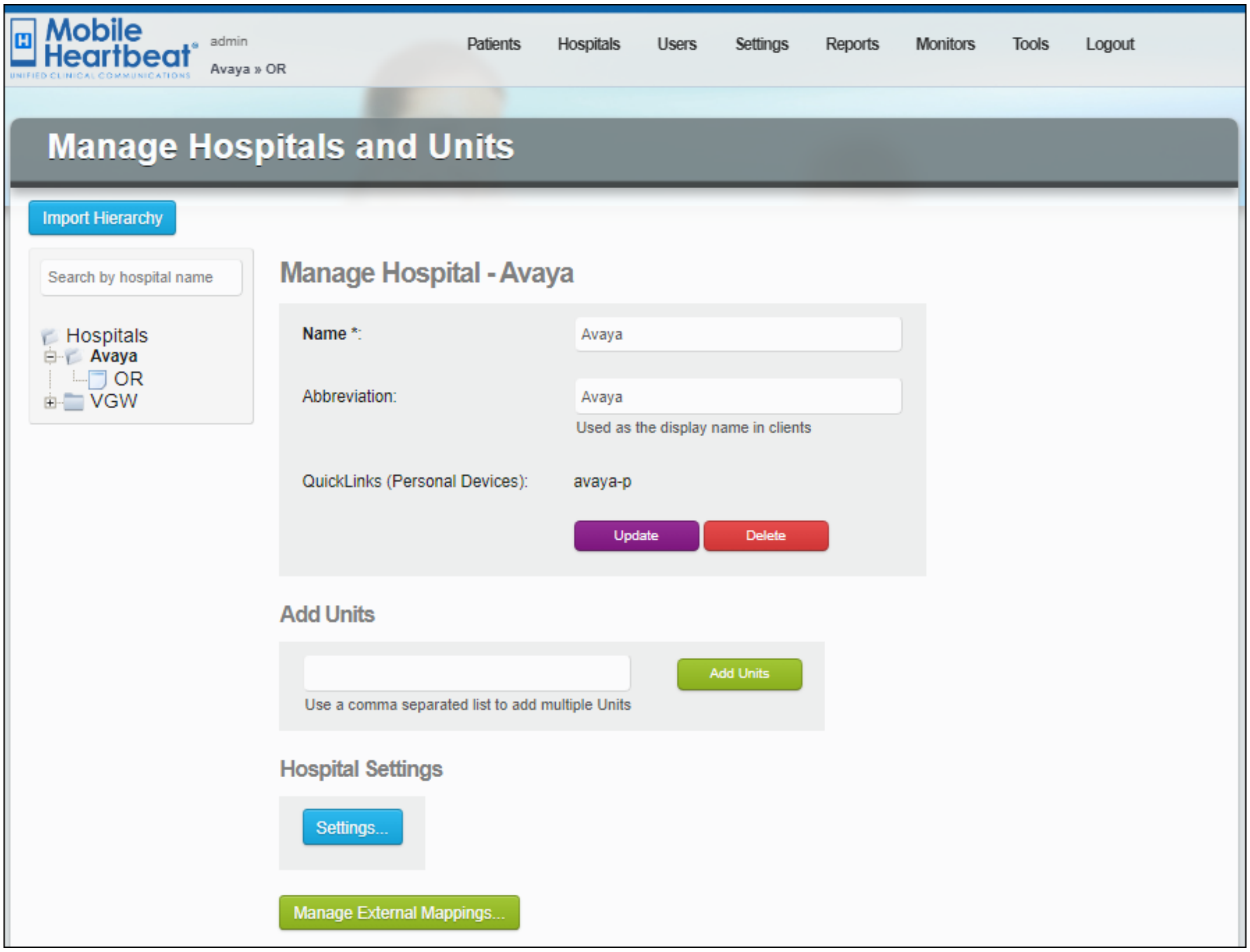
7.1. Log into MH-CURE Web Admin Tool

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

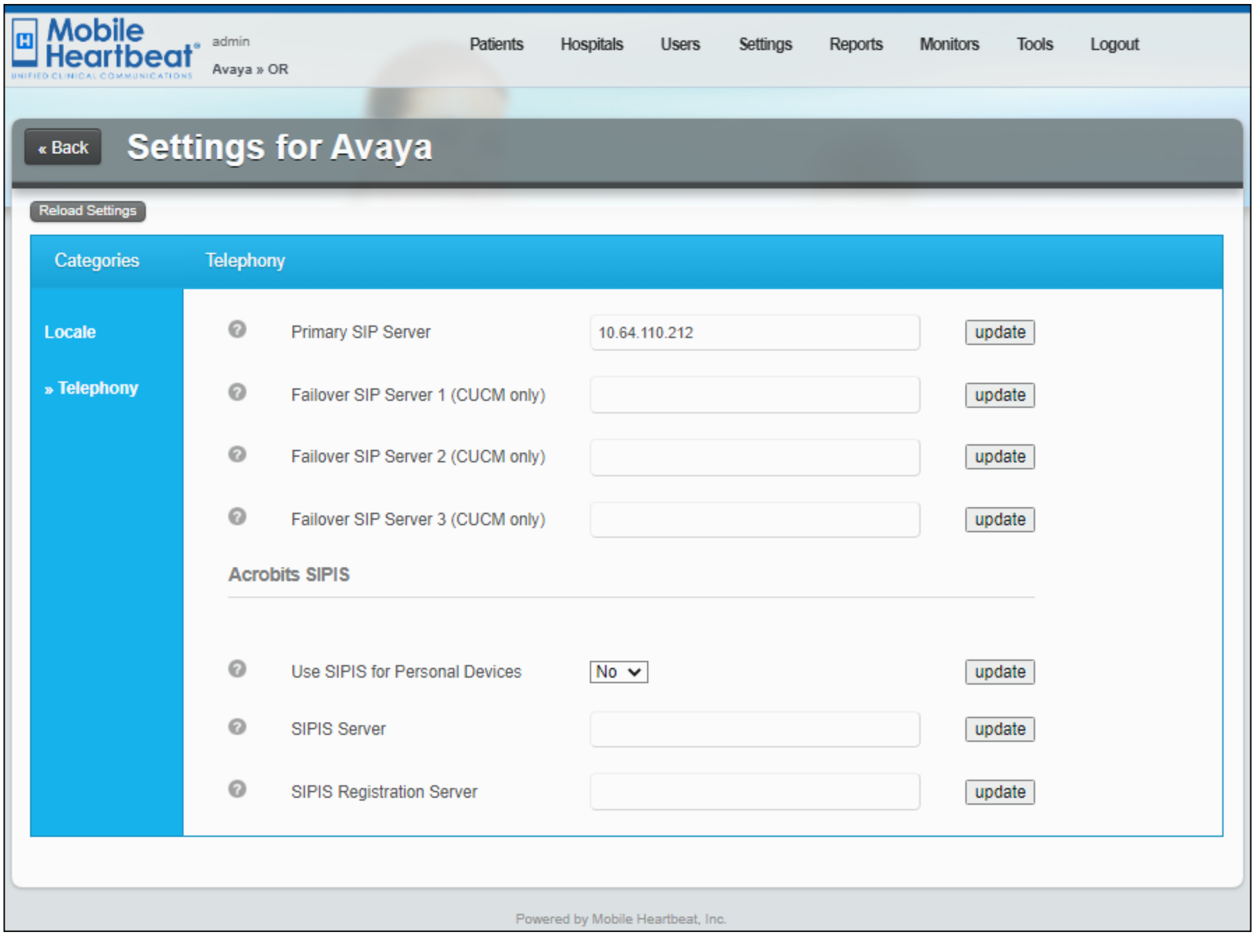


7.2. Configure MH-CURE for SIP Endpoint Service

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

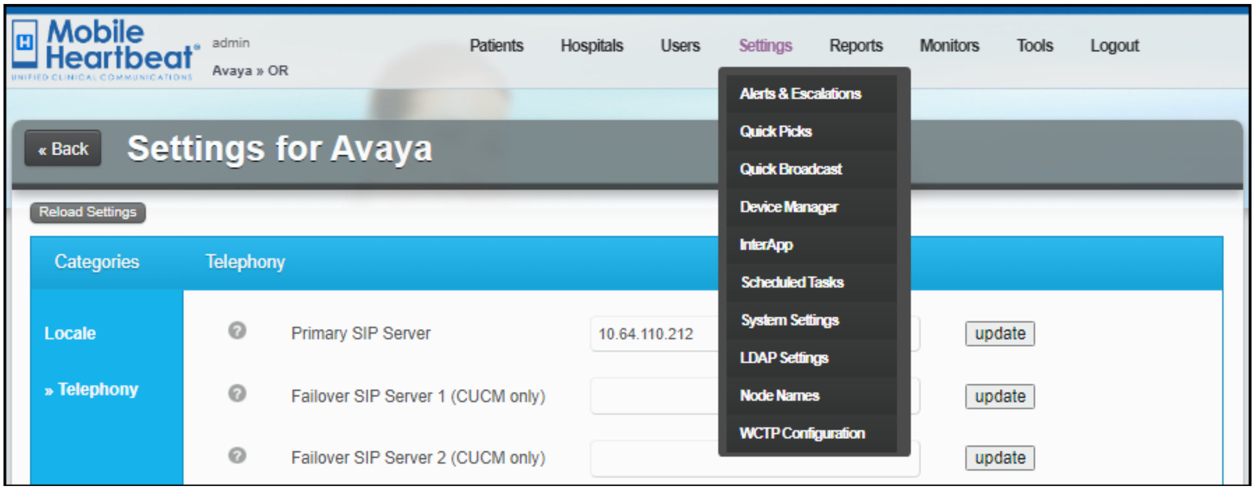


Click **Telephony** in the left pane. Set **Primary SIP Server** to the Session Manager IP address (e.g., *10.64.110.212*) from **Section 5.2**. Click the **Update** button to the right of the field.



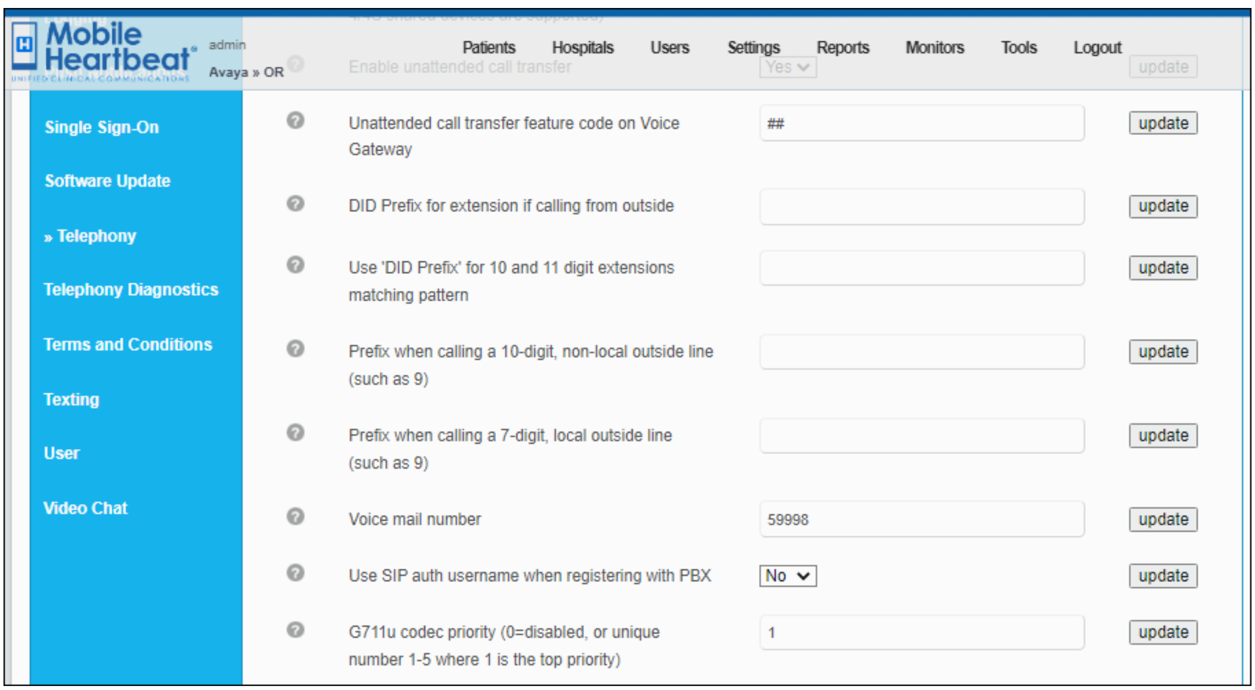
7.3. Configure Voicemail Number

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



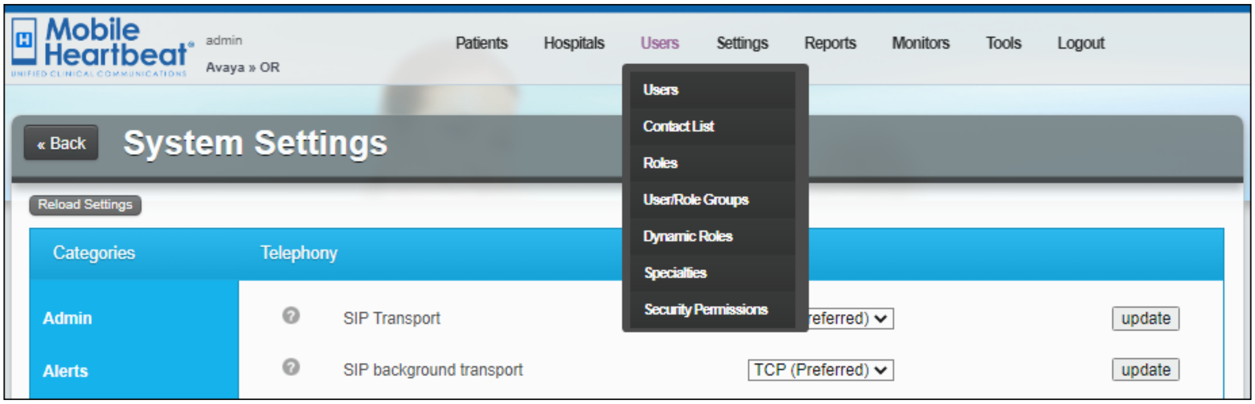
Next, select **Telephony** in left pane. Scroll down and locate the **Voice mail number** field. Type in the voicemail pilot number. For the compliance test, 59998 was the voicemail pilot number for Avaya Messaging.

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

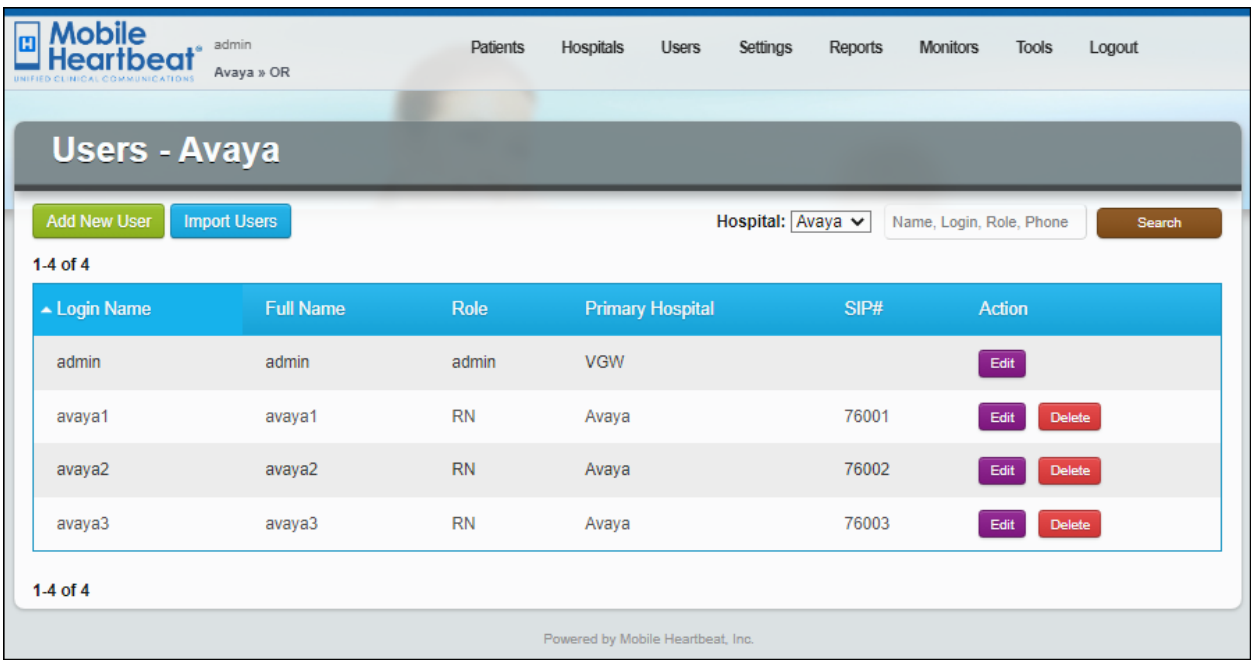


7.4. Configure MH-CURE Users

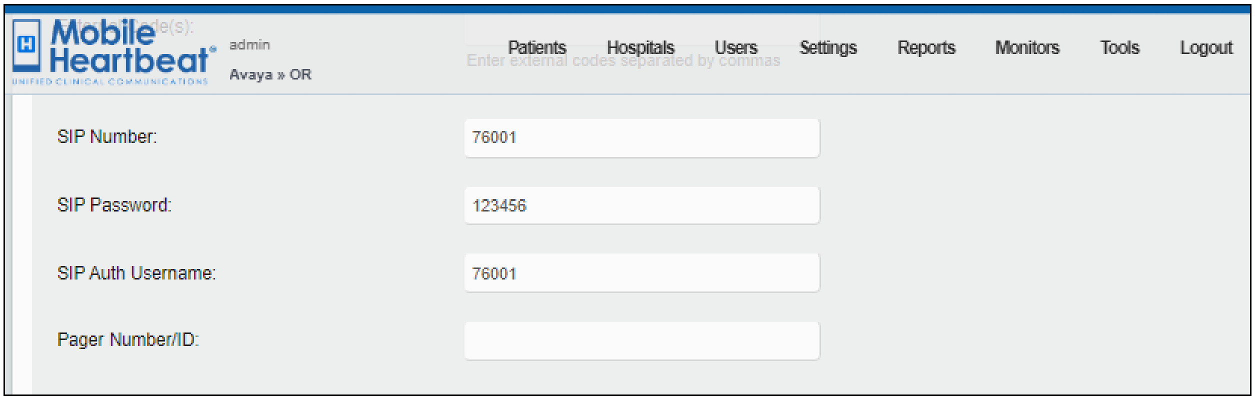
To add or modify an existing user, first navigate to **Users → Users** from the top menu.



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.



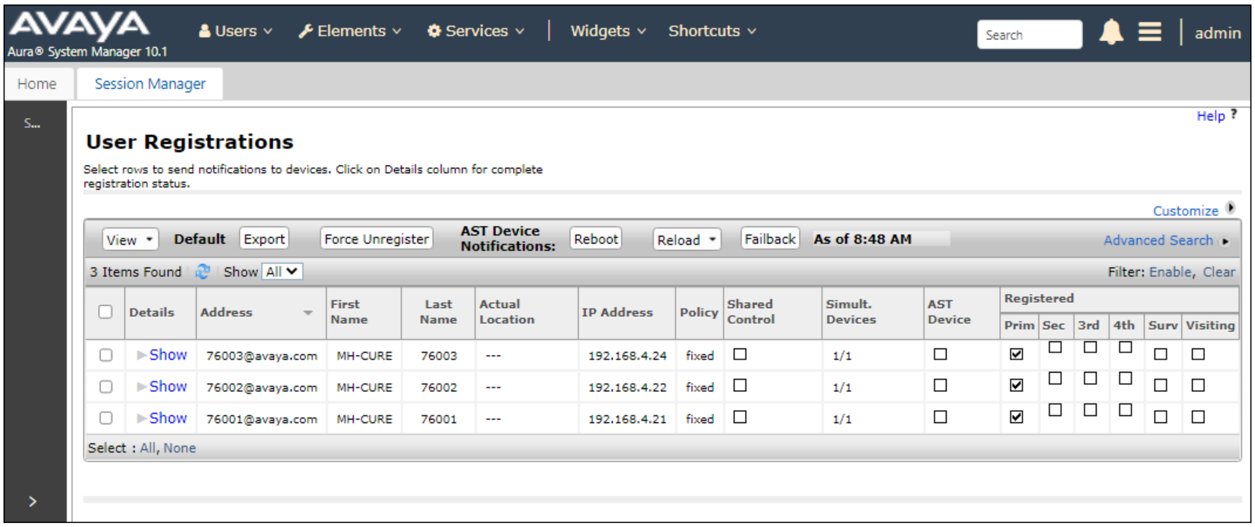
Configure the user extension as configured in **Section 6.3.1** for **SIP Number** and **SIP Auth Username**. Configure the **SIP Password** as configured in **Section 6.3.2**. Once done, click **Update** at the bottom of the screen (not shown).



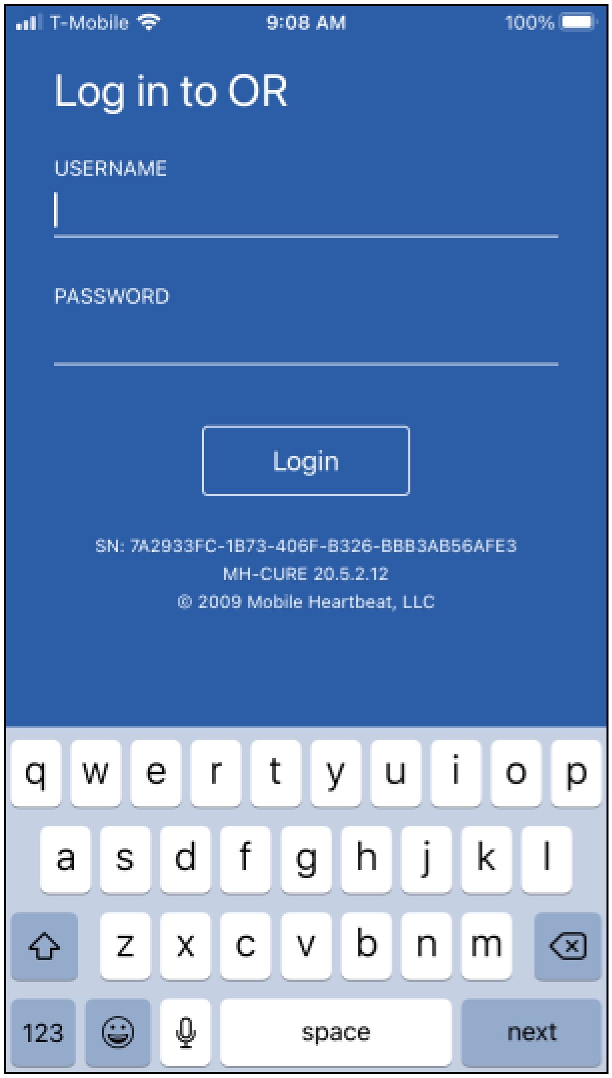
8. Verification Steps

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.

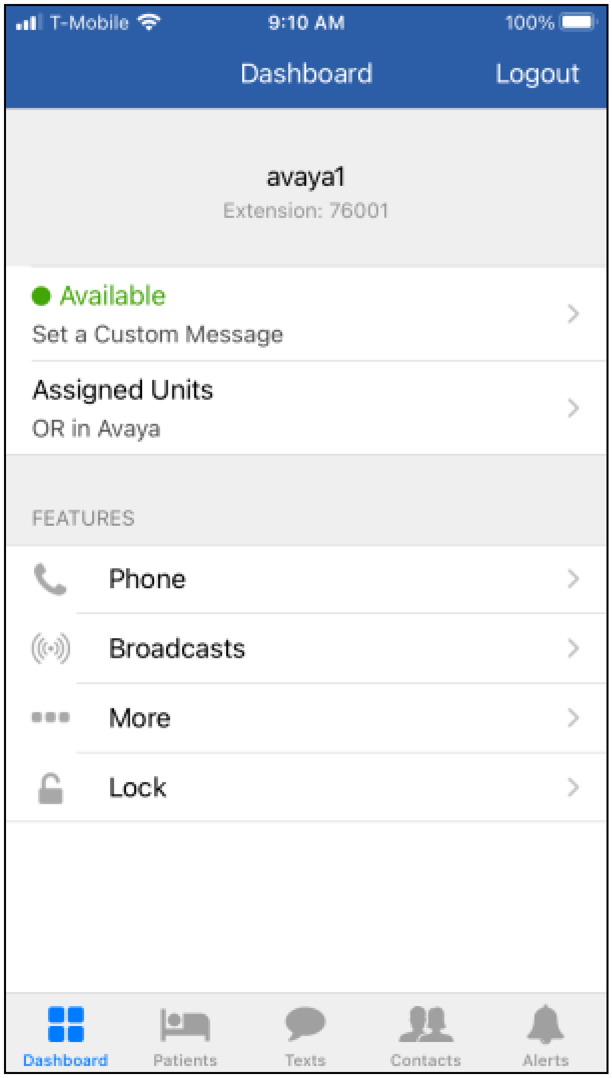
1. Verify that MH-CURE clients have successfully registered with Session Manager. In System Manager, navigate to **Elements → Session Manager → System Status → User Registrations** to check the registration status as shown below.



2. From an iOS mobile device, launch the MH-CURE app and 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.

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.