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 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.

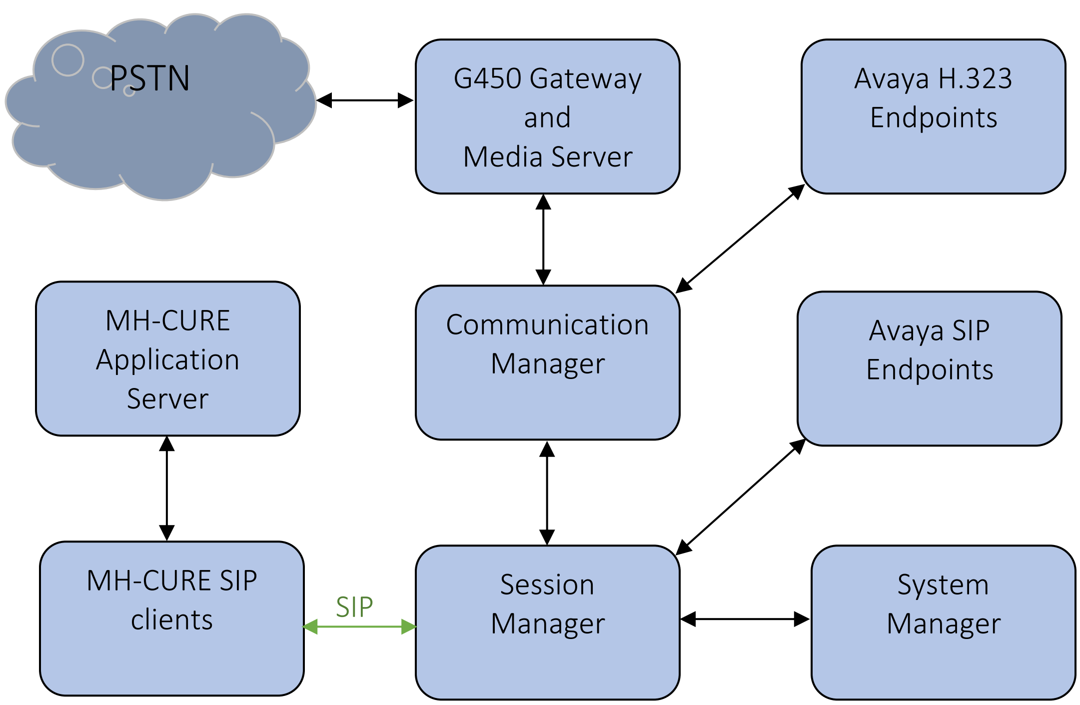


Figure 1: Avaya Aura® with MH-CURE

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
ipo15050            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:
                                             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 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

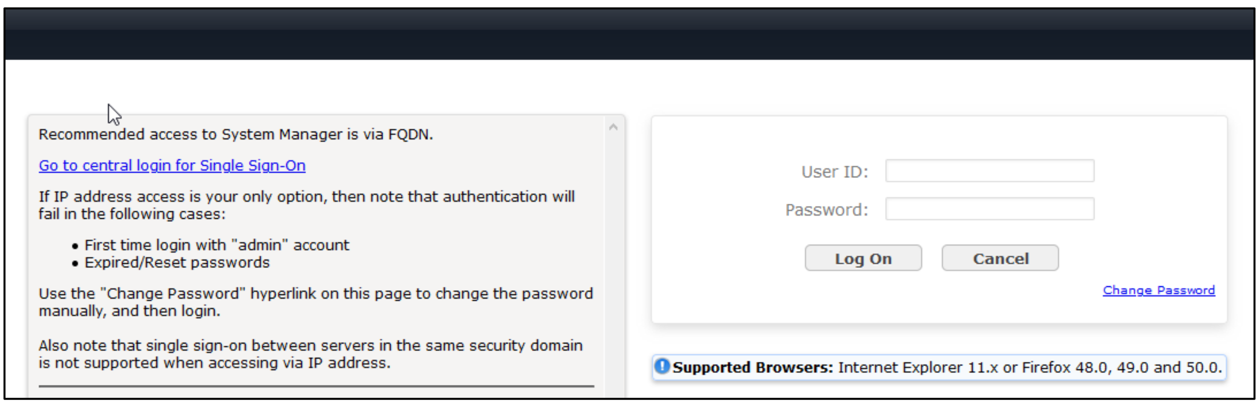
This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- 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.

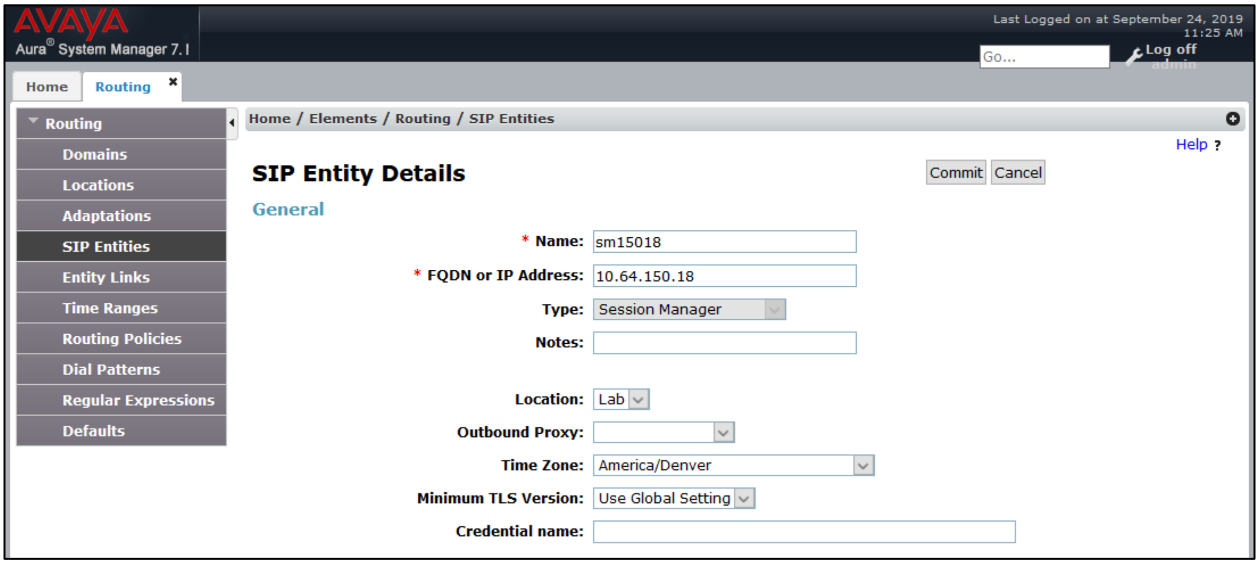
6.1. Launch System Manager

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.

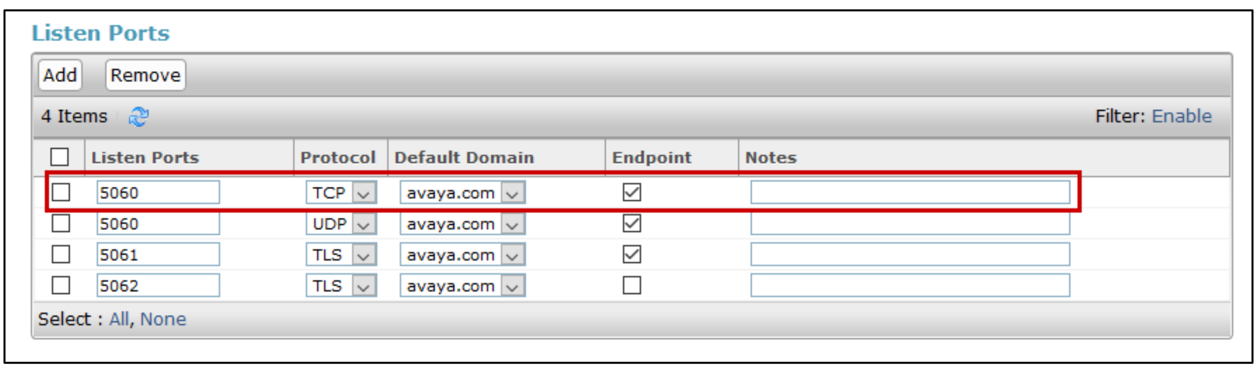


6.2. Set Network Transport Protocol for MH-CURE SIP clients

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

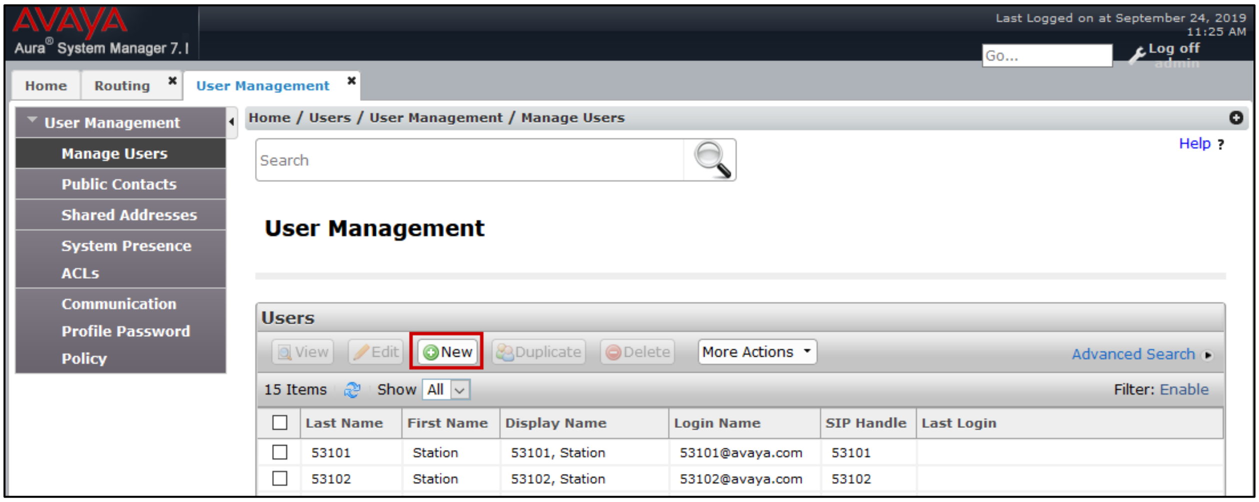


Scroll down to the **Listen Ports** section and verify that the transport network protocol used by MH-CURE SIP clients is specified in the list below. For the compliance test, the solution used TCP network transport.



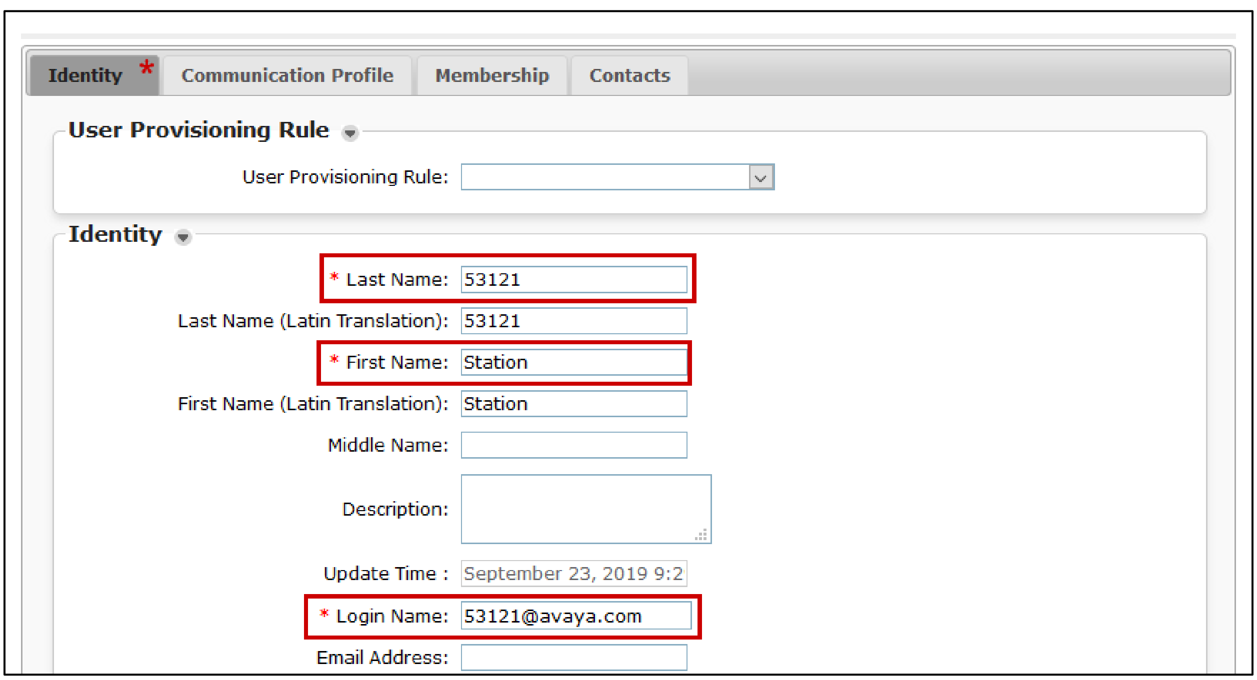
6.3. Administer SIP User

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.



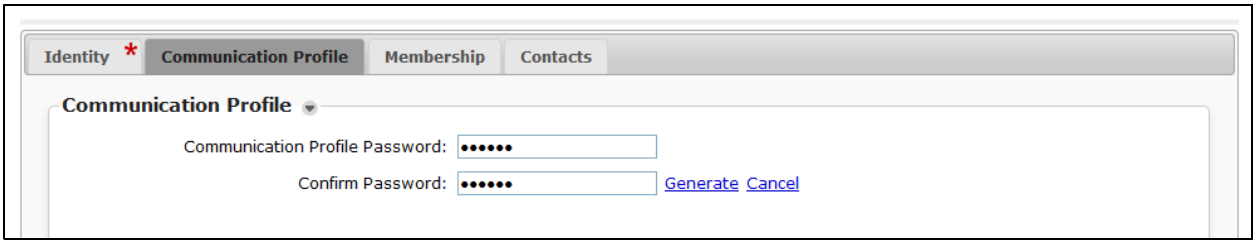
6.3.1. Identity

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.



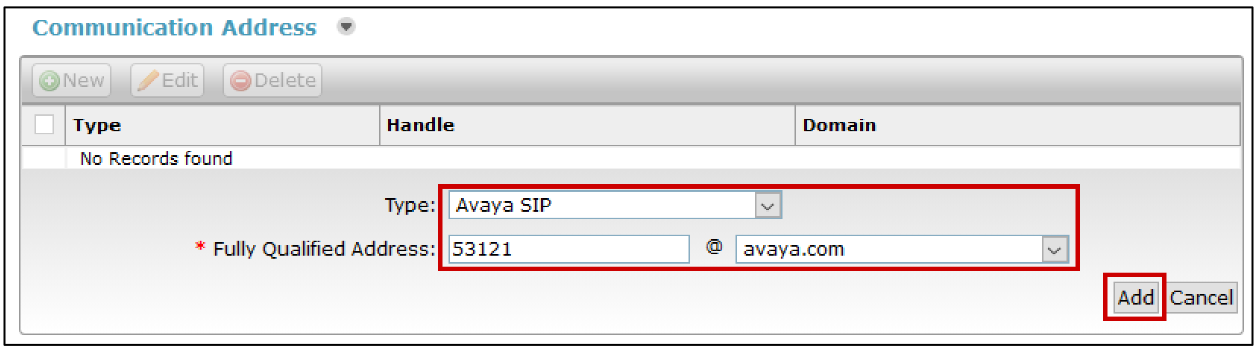
6.3.2. Communication Profile

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**.



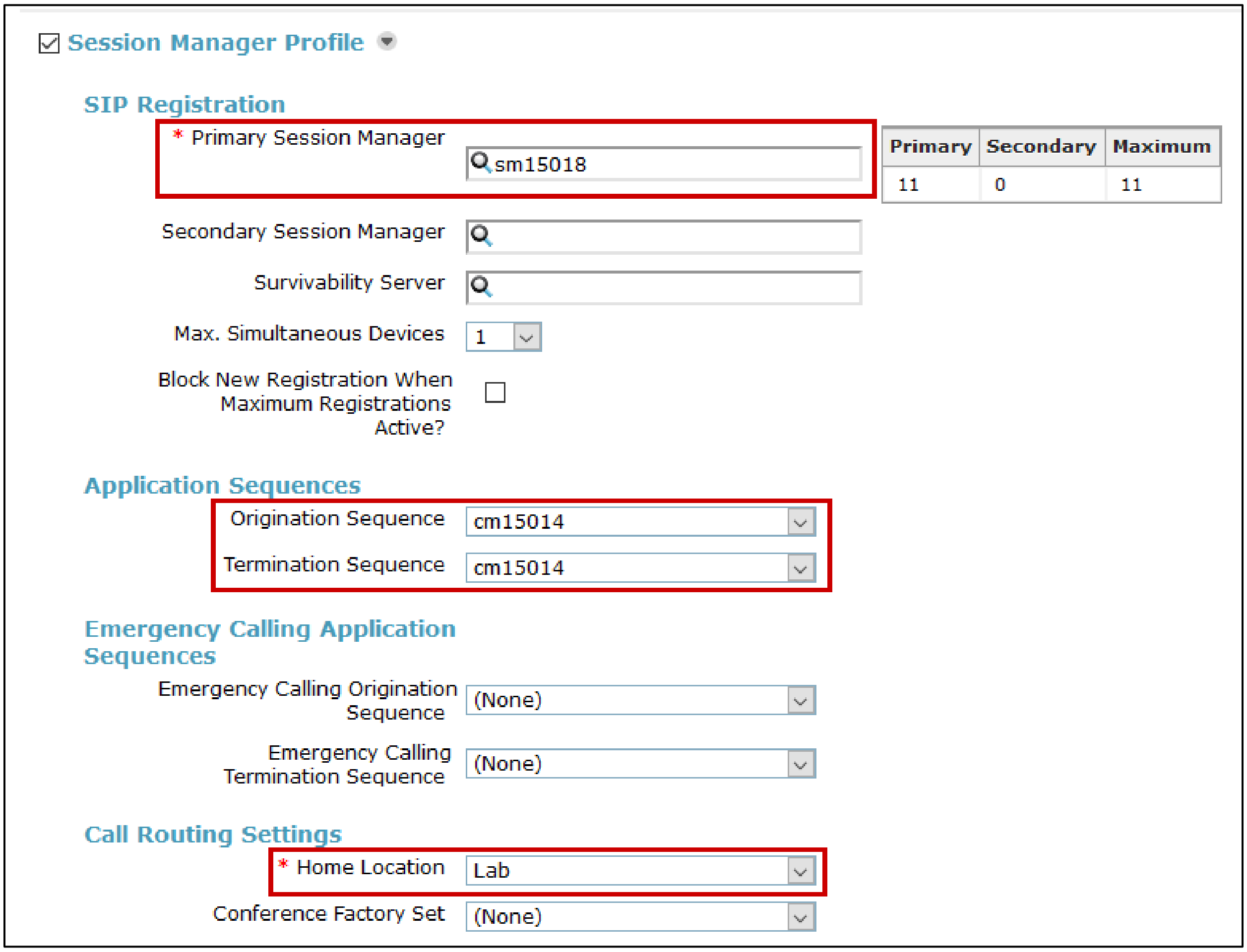
6.3.3. Communication Address

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**.



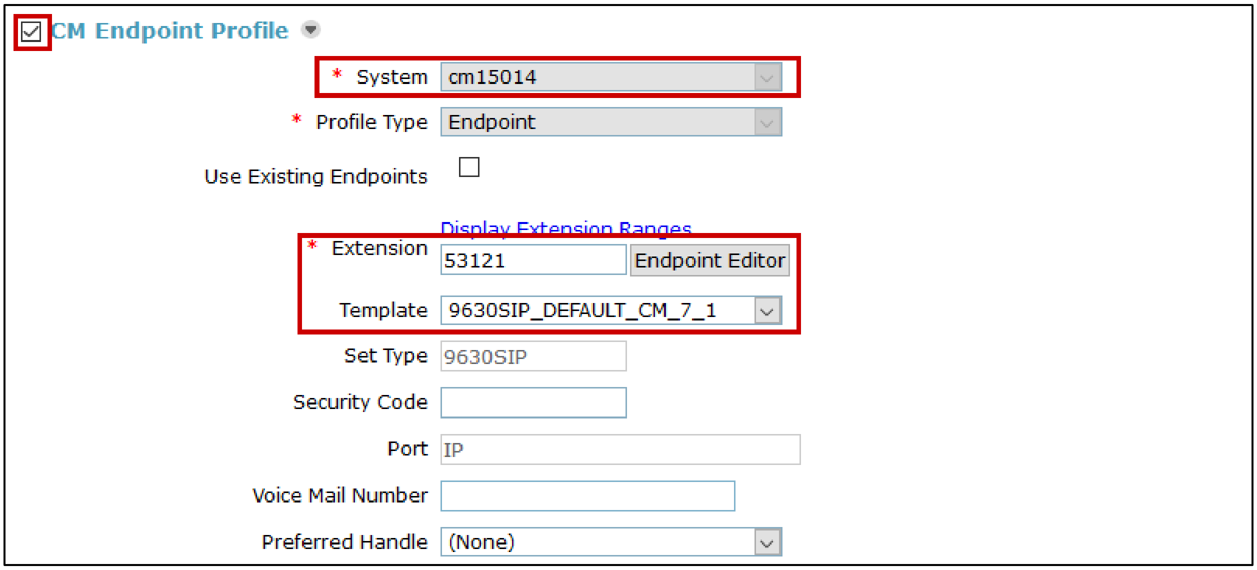
6.3.4. Session Manager Profile

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**.

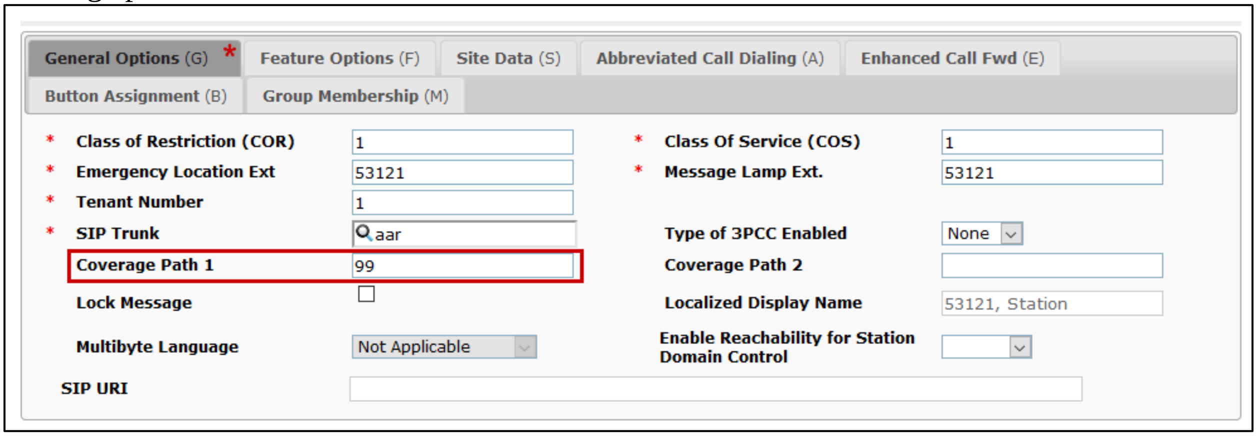


6.3.5. CM Endpoint Profile

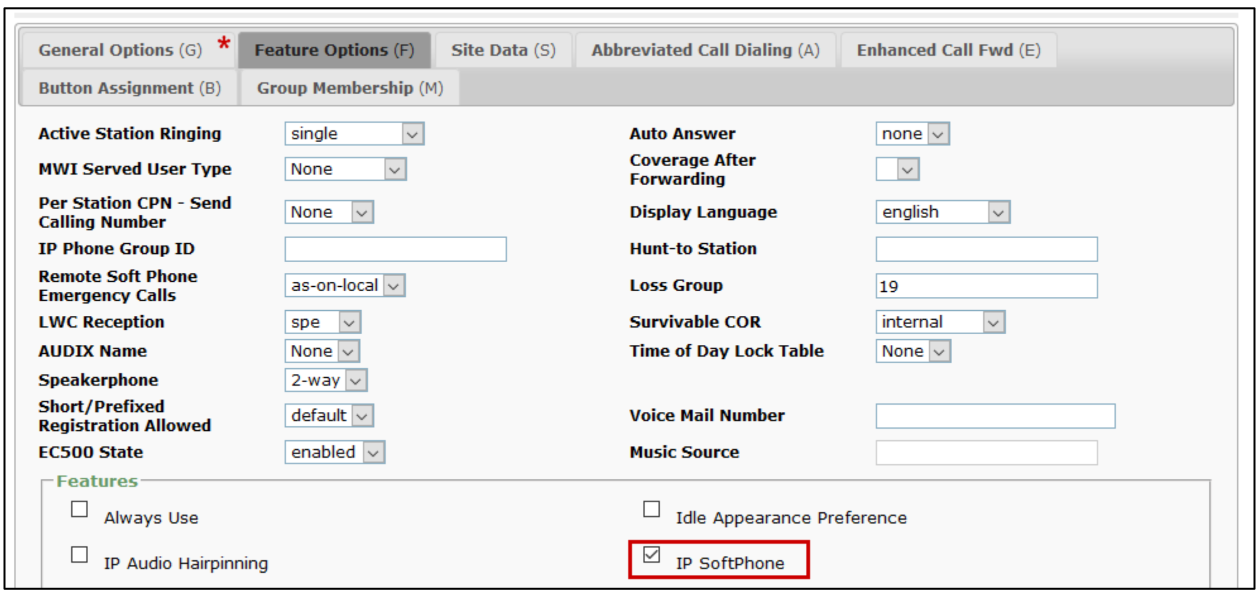
Check box for **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.1**. For **Template**, select *9630SIP_DEFAULT_CM_7_1*. Retain the default values in the remaining fields. 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*.



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



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).



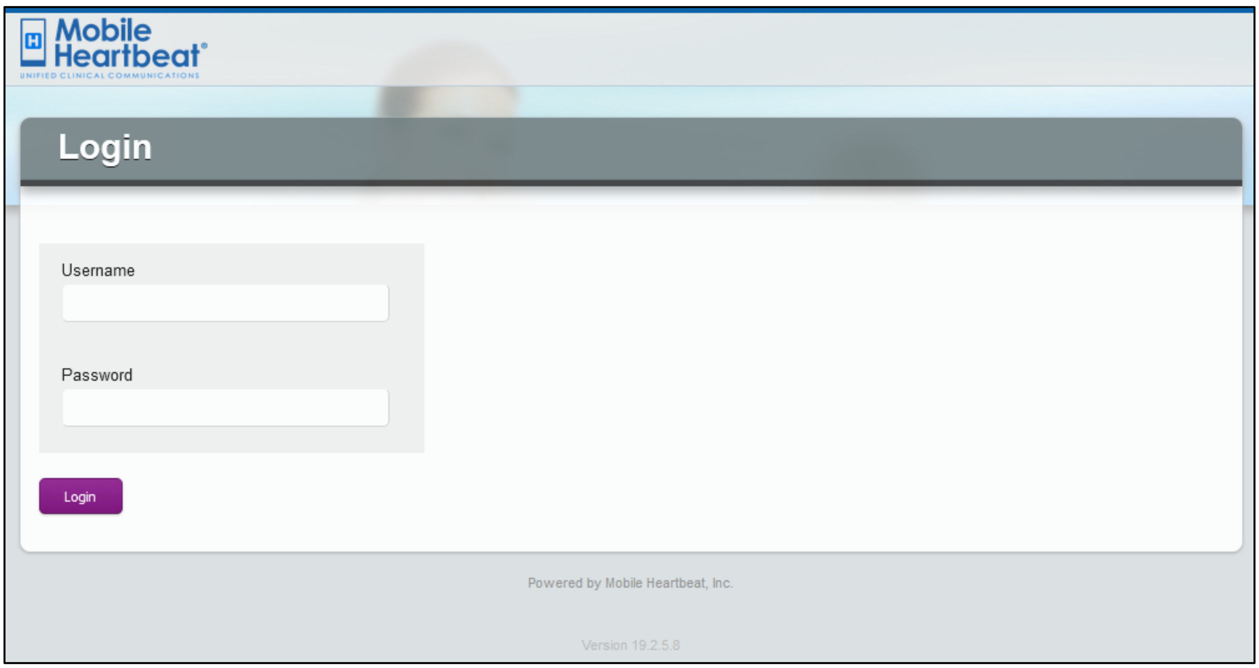
7. Configure Mobile Heartbeat MH-CURE

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

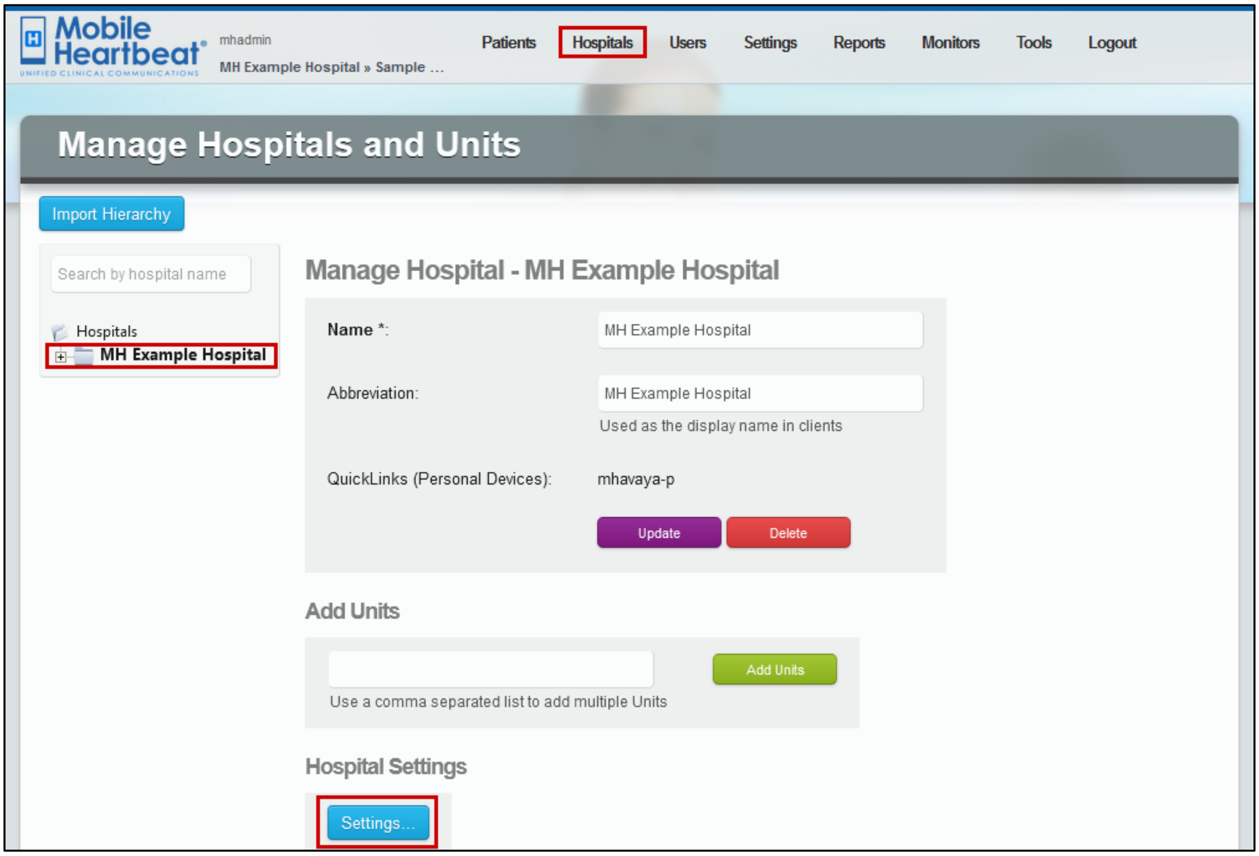
7.1. Log into MH-CURE web UI

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.

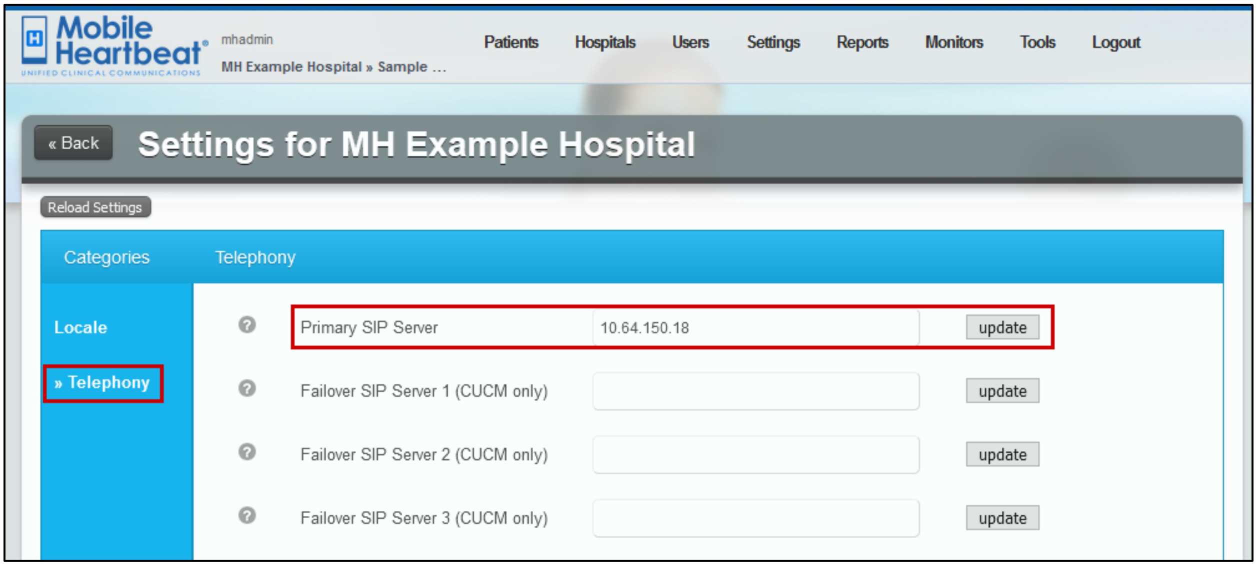


7.2. Configure MH-CURE for SIP Endpoint Service

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.

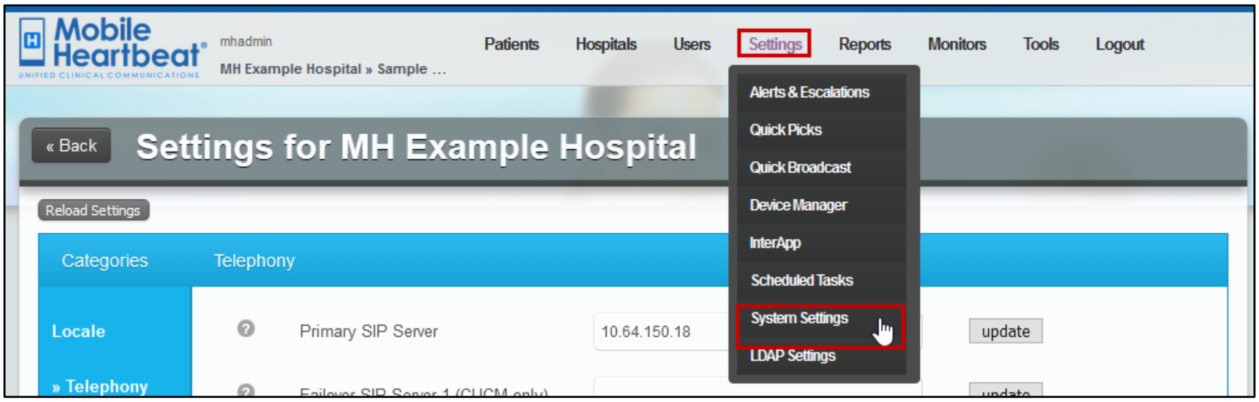


Click **Telephony** on the left side. Change **Primary SIP Server** to IP Address of Session Manager from **Section 5.2**, and click the Update button on right.

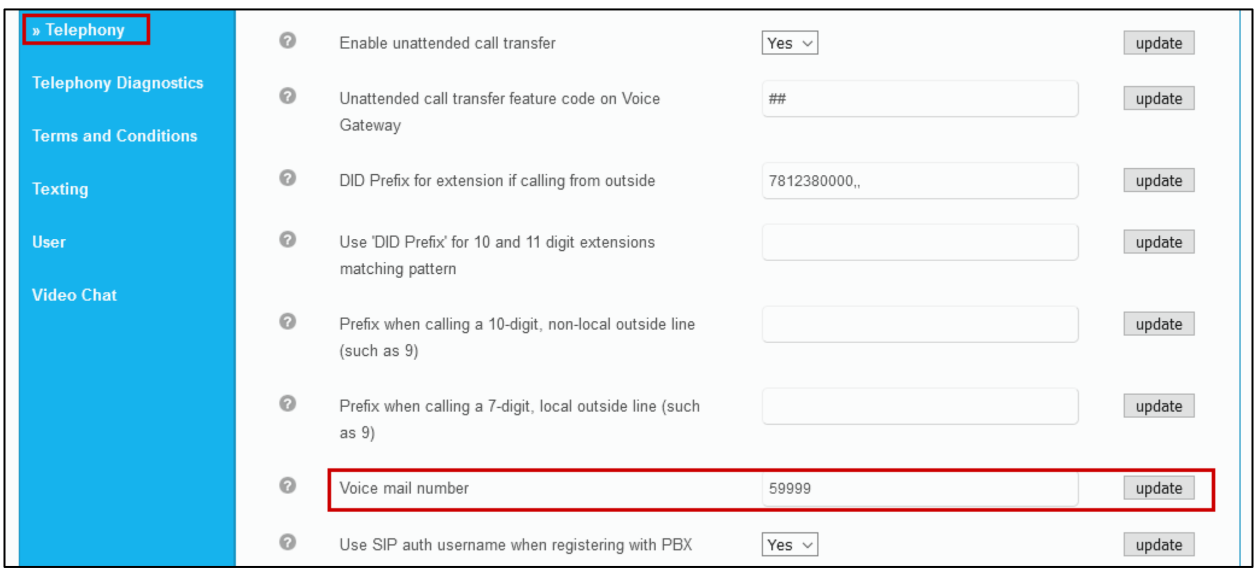


7.3. Configure Voicemail Number

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

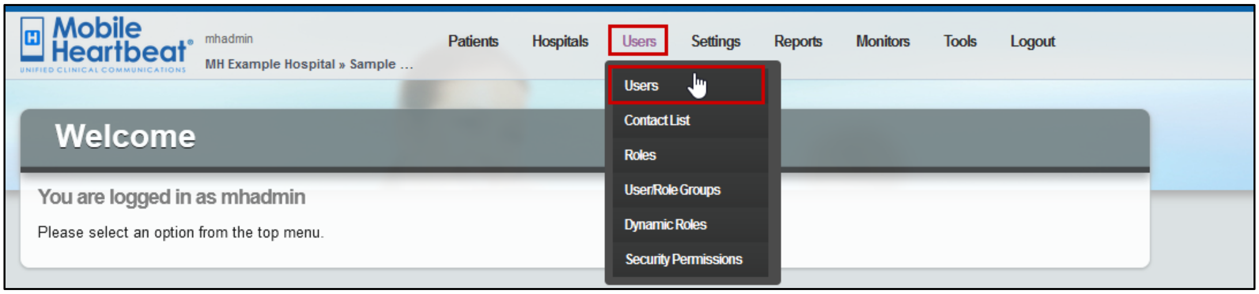


Select **Telephony** on left side. Scroll down and locate **Voice mail Number** field. Type in the pilot number used in the environment. During the compliance test, 59999 was configured as the Voicemail pilot number for Communication Manager Messaging.

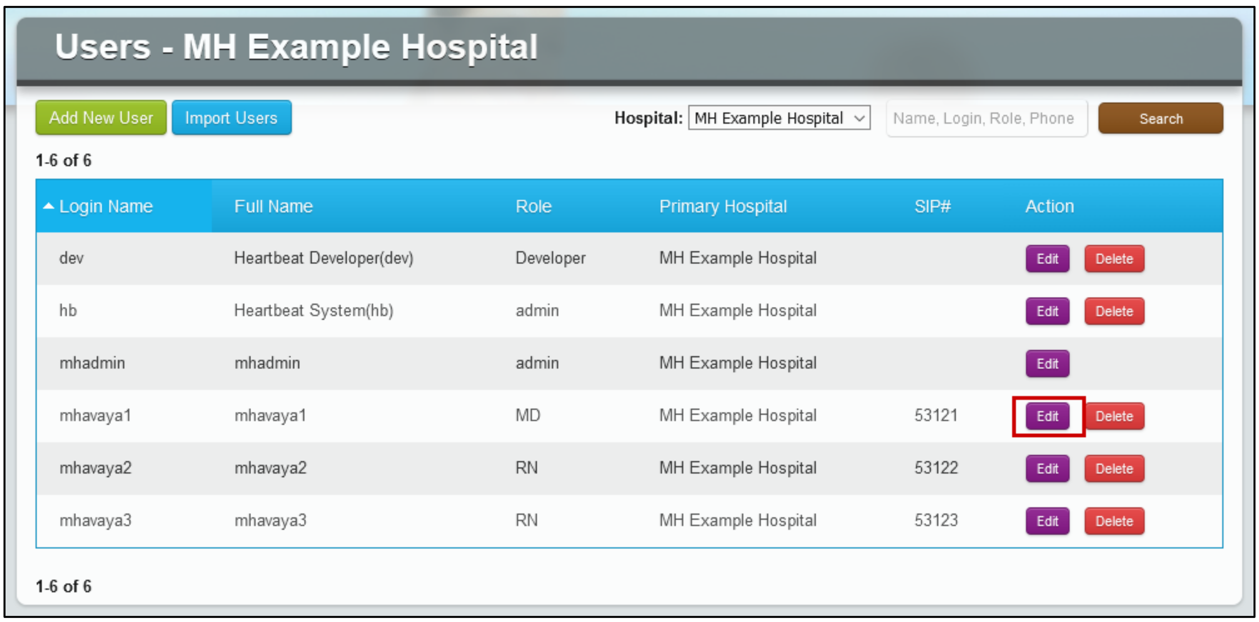


7.4. Configure User Extensions

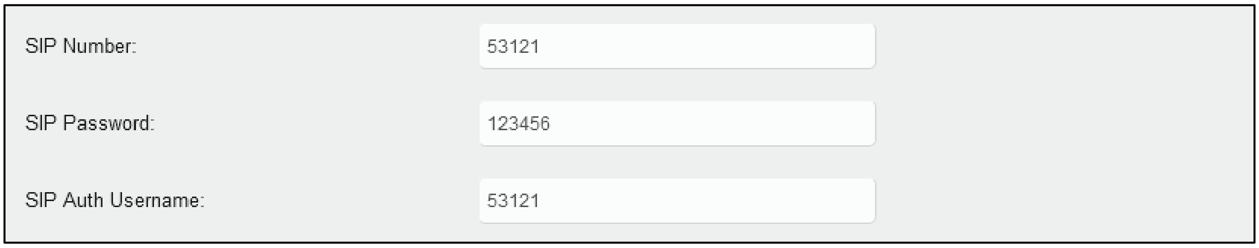
From the top, navigate to **Users → Users.**



Edit an existing user to configure the user.



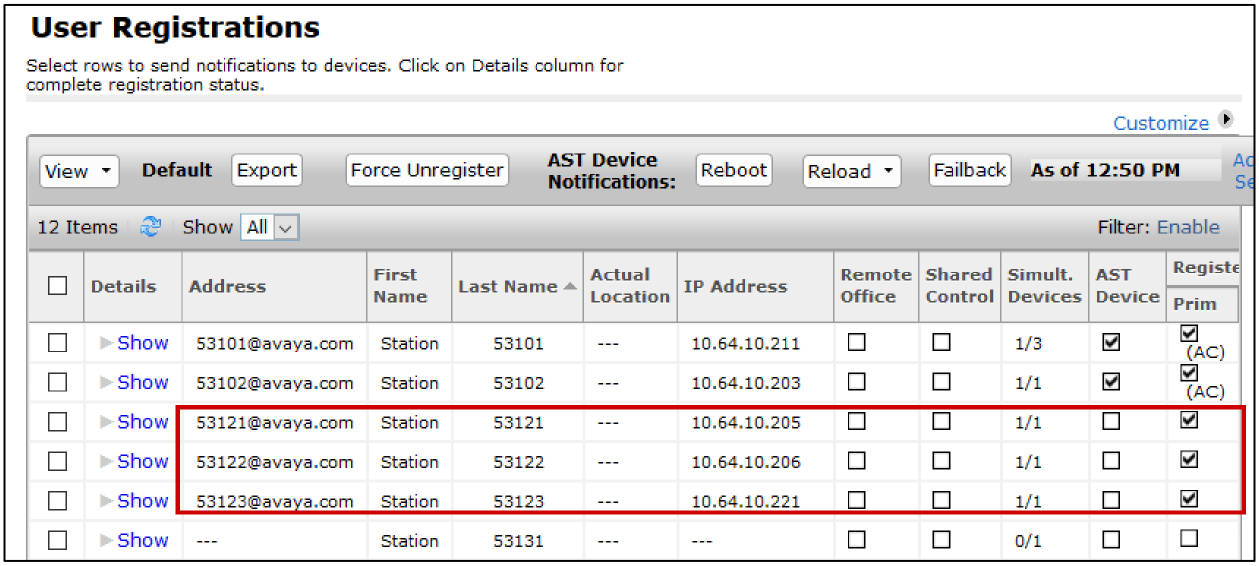
Configure the user extension as configured in **Section 6.3.1** for **SIP Number** and **SIP Auth Username.** Configure the **SIP Password** as configured in **6.3.2.** Once done click **Update** at the bottom of the screen (not shown).



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

☐	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registe Prim
☐	Show	53101@avaya.com	Station	53101	---	10.64.10.211	☐	☐	1/3	☑	☑ (AC)
☐	Show	53102@avaya.com	Station	53102	---	10.64.10.203	☐	☐	1/1	☑	☑ (AC)
☐	Show	53121@avaya.com	Station	53121	---	10.64.10.205	☐	☐	1/1	☐	☑
☐	Show	53122@avaya.com	Station	53122	---	10.64.10.206	☐	☐	1/1	☐	☑
☐	Show	53123@avaya.com	Station	53123	---	10.64.10.221	☐	☐	1/1	☐	☑
☐	Show	---	Station	53131	---	---	☐	☐	0/1	☐	☐

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.

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.