**Avaya Solution & Interoperability Test Lab**

# Application Notes for Med-Pat XLIP V2 with Avaya Aura® Session Manager R8.1 and Avaya Aura® Communication Manager R8.1 – Issue 1.0

## Abstract

These Application Notes describe the configuration steps required for Med-Pat XLIP V2 to interoperate with Avaya Aura® Session Manager R8.1 and Avaya Aura® Communication Manager R8.1. Med-Pat XLIP V2 is a one-piece SIP hospital phone that can register with Avaya Aura® Session Manager as a SIP endpoint in support of voice communications.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any 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 for Med-Pat XLIP V2 to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Med-Pat XLIP V2 is a one-piece SIP telephone that can register with Avaya Aura® Session Manager as a SIP endpoint. Med-Pat XLIP V2 (XLIP) is typically deployed in hospital patient rooms. Med-Pat has noted that the handset does not yet support TLS and media encryption so the tests are limited to the TCP protocol.

# 2. General Test Approach and Test Results

The general test approach was to place calls to and from XLIP and exercise basic telephone operations. The main objectives were to verify the following:

- Registration
- Codecs (G.711, G.722, and G.729)
- Inbound calls
- Outbound calls
- Call termination (origination/destination)
- Avaya Features using FAC (Call Pickup, Call Forward, and Find Me)
- Message Waiting Indicator (MWI)
- Voicemail
- Serviceability

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, XLIP did not include any encryption features as requested by Med-Pat.

## 2.1. Interoperability Compliance Testing

The interoperability compliance test included features and serviceability. The focus of interoperability compliance testing was primarily on verifying call establishment on XLIP. XLIP operations such as inbound calls, outbound calls, hold/resume, transfer, conference, Facility Access Codes, and its interactions with Session Manager, Communication Manager, and other Avaya SIP, and H.323 phones were verified. The serviceability testing introduced failure scenarios to see if XLIP can recover from loss of ethernet connectivity by cable disconnection while a call is active.

## 2.2. Test Results

All test cases were executed. The following observations were noted during compliance testing:

- The following features are not supported by XLIP:
  - Call Mute, Hold, and Resume
  - Call Transfer
  - Conference Call
  - Long Hold Recall Timer
  - Call Park
  - MWI
- When the call is terminated at the far-end, a busy tone is heard on XLIP handset for three to five seconds. As per the signaling, the call terminates properly. Med-Pat indicates this is expected and acceptable.
- XLIP does not indicate a message is waiting via stutter tone. Med-Pat indicates this is not currently enabled and will be added in a future release.
- Using '#' as a leading digit for Feature access codes does not work. The workaround is to use another digit such as '*'.

## 2.3. Support

For Technical Support:

- Med-Pat XLIP V2 please visit http://www.med-pat.com
- Telephone: 800 626-0410

# 3. Reference Configuration

Once XLIP registers as a SIP endpoint with Session Manager, it can place and receive voice calls with various supported features as listed above in **Section 2.1**. The reference configuration used for the compliance test is shown in **Figure 1** below.

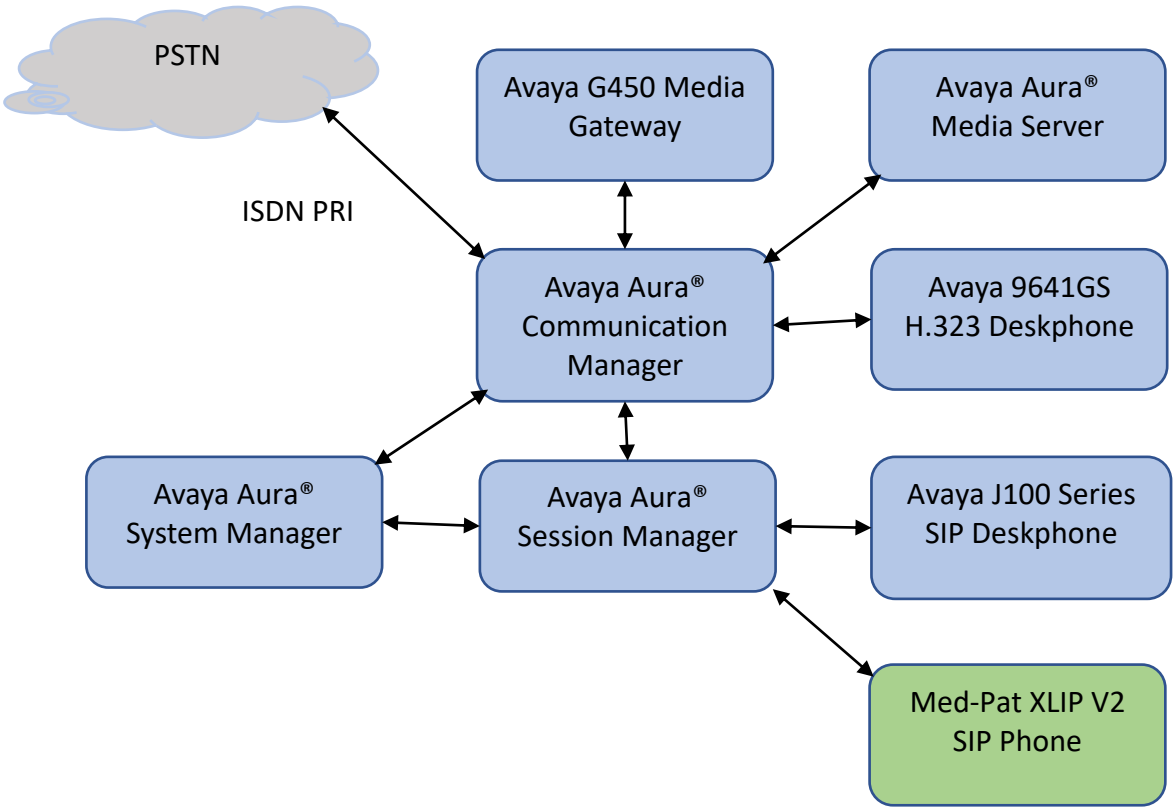



**Figure 1: Med-Pat XLIP V2 with Avaya Aura® Session Manager and Avaya Aura® Communication Manager**

# 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 a virtual server | 8.1.3.0.1.890.26685 |
| Avaya Aura® System Manager running on a virtual Machine | 8.1.3.0.813014 |
| Avaya Aura® Session Manager running on a virtual server | 8.1.3.0.813014 |
| Avaya Aura® Media Server running on a virtual server | 8.0.0 169 |
| Avaya G450 Media Gateway | 41.34.1 |
| Avaya 9641GS Deskphone (H.323) | R6_8_5_11-050321 |
| Avaya J100 Series Deskphone (SIP) | R4_0_9_0-040821 |
| Med-Pat XLIP V2 | V0.0.23 |

# 5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring Communication Manager. The procedures include the following areas:

- Verify the Communication Manager OPS (Off-PBX station) Licensed Capacity
- Administer IP Network Region
- Administer IP Codec Set

Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials. The configuration steps illustrate field values changed for this reference configuration. Default values were used for all other fields.

**Note:** It is assumed that basic configuration of the Communication Manager has already been completed, such as the SIP trunk to Session Manager. The SIP station configuration for XLIP is configured through System Manager in **Section 6.2.**

## 5.1. Verify Communication Manager OPS Licensed Capacity

Using the SAT, verify that the Off-PBX Stations (OPS) and SIP Trunks features are 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: V18                              Software Package: Enterprise
       Location: 2                                 System ID (SID): 1
       Platform: 28                                Module ID (MID): 1

                                                          USED
                                 Platform Maximum Ports: 48000    111
                                       Maximum Stations: 36000     86
                                Maximum XMOBILE Stations: 36000      0
                    Maximum Off-PBX Telephones - EC500: 41000        0
                    Maximum Off-PBX Telephones -   OPS: 41000       51
                    Maximum Off-PBX Telephones - PBFMC: 41000        0
                    Maximum Off-PBX Telephones - PVFMC: 41000        0
                    Maximum Off-PBX Telephones - SCCAN:     0        0
                         Maximum Survivable Processors: 313   0

        (NOTE: You must logoff & login to effect the permission changes.)
```

## 5.2. Administer IP Network Region

This IP network region is for the signaling group associated with the SIP trunk between Session Manager and Communication Manager. This form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. Verify the following values:

- **Authoritative Domain**: The applicable domain (e.g., **avaya.com**)
- **Codec Set**: The codec set number from **Section 5.3**

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 Avaya G450 Media Gateway or Avaya Media Server.

```
change ip-network-region 1                                    Page   1 of  20
                             IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name: Main
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
```

## 5.3. Administer IP Codec Set

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to XLIP. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set **1** is specified in **IP Network Region 1** shown above. The form shows the list of codecs tested. Enter values for the following:

- **Audio Codec**: The audio codec tested
- **Media Encryption**: **none** as an option to assure TCP and no encryption is available.

```
display 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: G.711A            n         2        20
 3: G.722             n         2        20
 4: G.729             n         2        20
 5:
 6
 7:
Media Encryption                  Encrypted SRTCP: best-effort
 1:1-srtp-aescm128-hmac80
 2:2-srtp-aescm128-hmac32
 3:none
 4:
 5:
```

# 6. Configure Avaya Aura® Session Manager

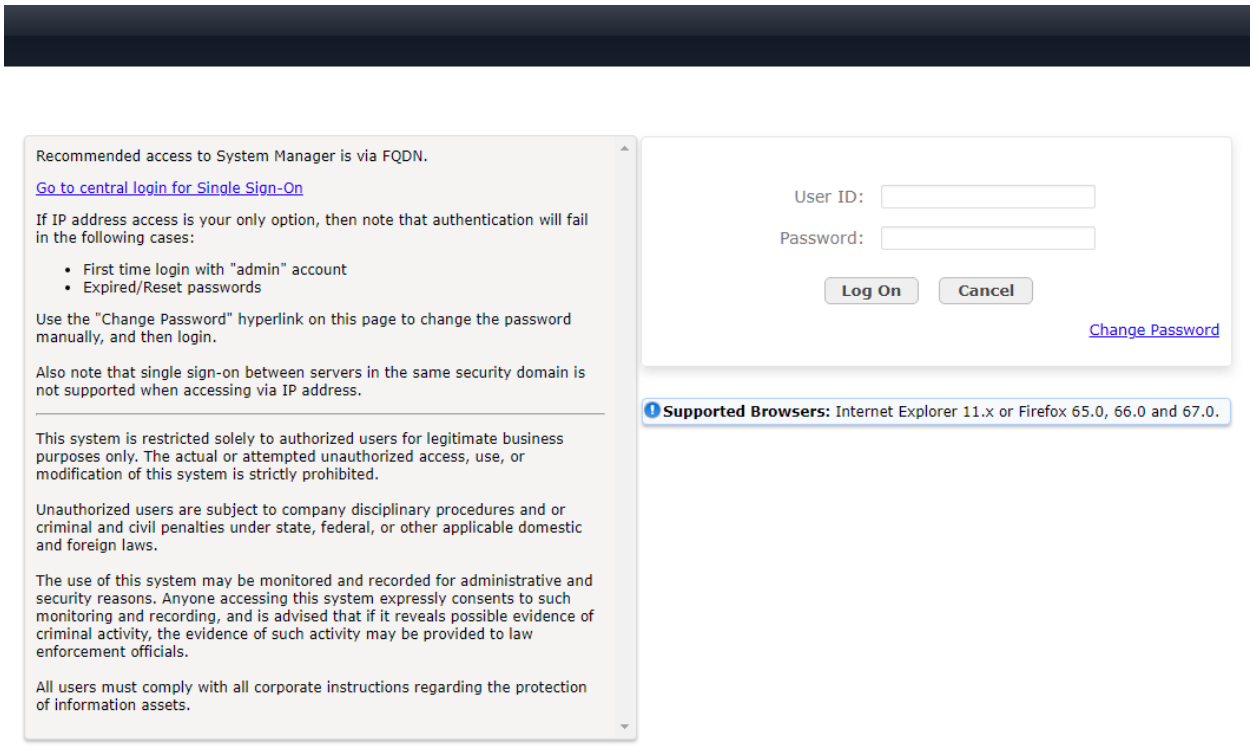
This section provides the steps for configuring Session Manager. The steps include adding the following areas:

- Launch System Manager
- Administer SIP Users

**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 XLIP.

## 6.1. Launch System Manager

Access Session Manager Administration web interface by entering **http://<ip-address>/SMGR** in a web browser, where <ip-address> is the IP address of System Manager. Log in using the appropriate credentials.



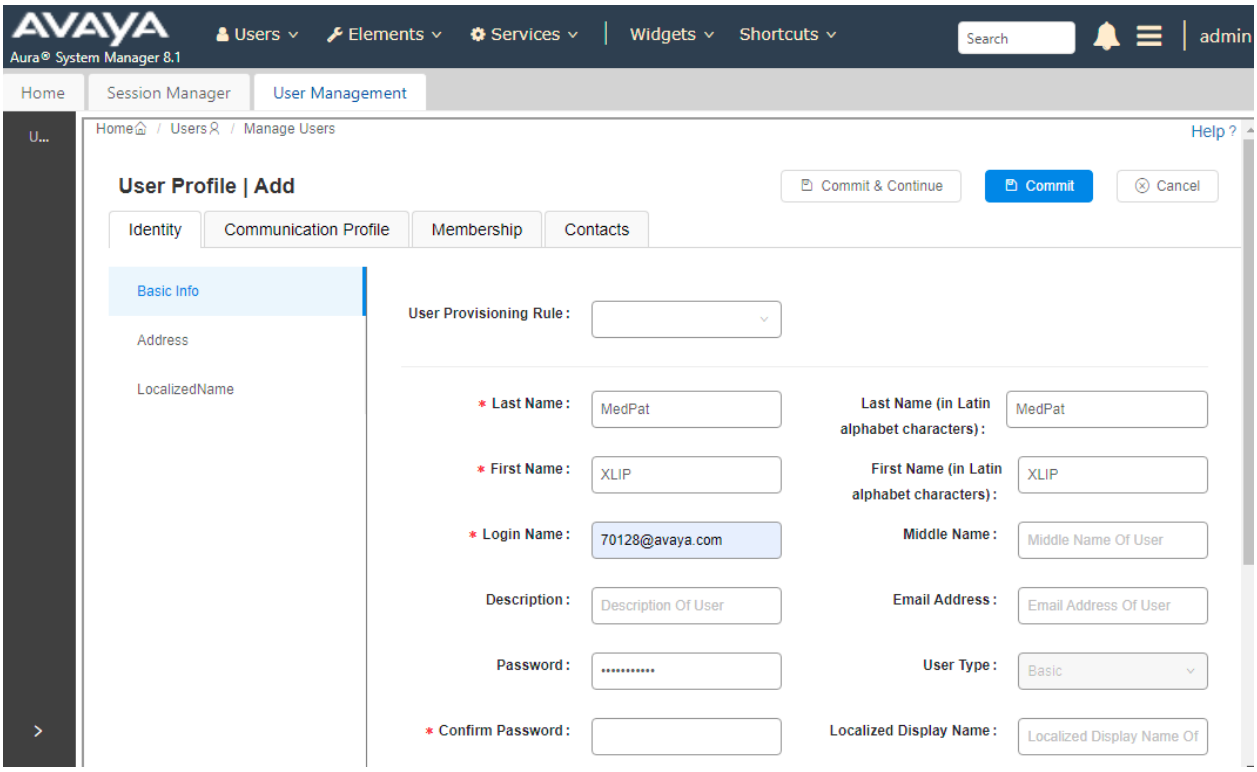
## 6.2. Administer SIP Users

XLIP is administered as a SIP user on Session Manager by the following steps. This configuration is automatically synchronized with Communication Manager. In Session Manager, select **Users →User Management →Manage Users** to display the **User Management** screen (not shown). Click **+ New** to add a user.

### 6.2.1. Identity

Enter values for the following required attributes for a new SIP user in the **New User Profile** screen:

- **Last Name**: Enter the last name of the user, e.g., **MedPat**
- **First Name**: Enter the first name of the user, e.g., **XLIP**
- **Login Name**: Enter <extension>@<sip domain> of the user (e.g., **70128@avaya.com**)
- **Password**: Enter an appropriate password
- **Confirm Password**: Re-enter the password from above

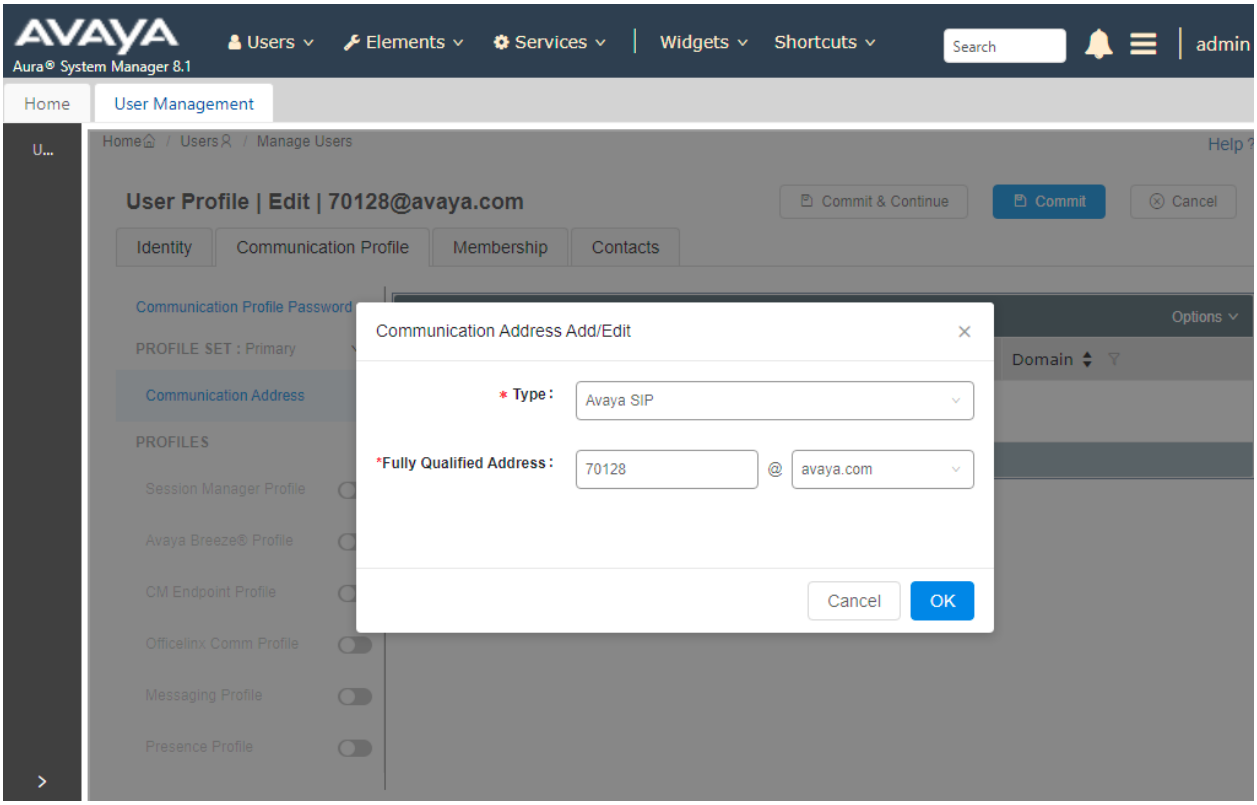



Press **Commit & Continue** after making entries or selections.

### 6.2.2. Communication Address

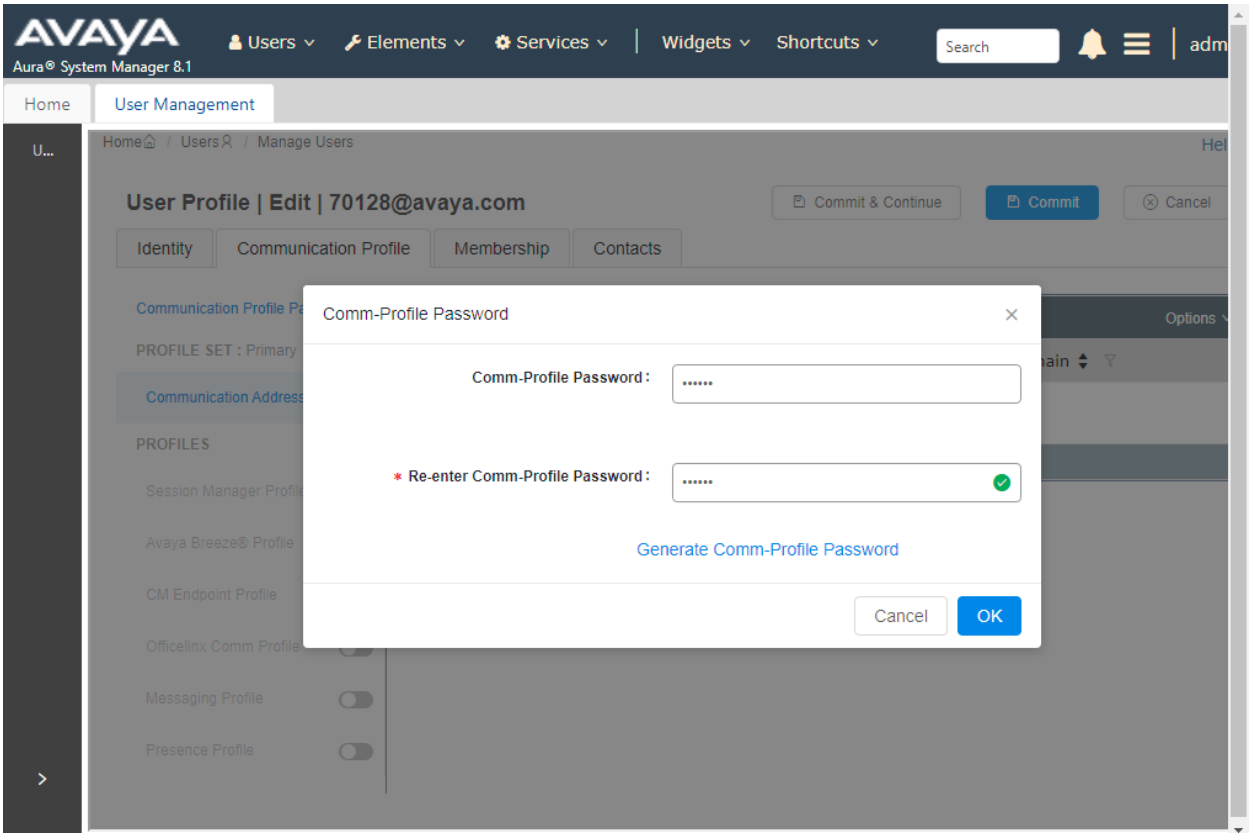
Select **Communication Address** in the left list and click **+ New** (not shown).
Enter the following attributes for the **Communication Address**:

- **Type:** Select **Avaya SIP** from the drop-down list
- **Fully Qualified Address:** Enter the extension number (e.g.,**70128**)
- **Domain:** Enter the domain (e.g., **avaya.com**)



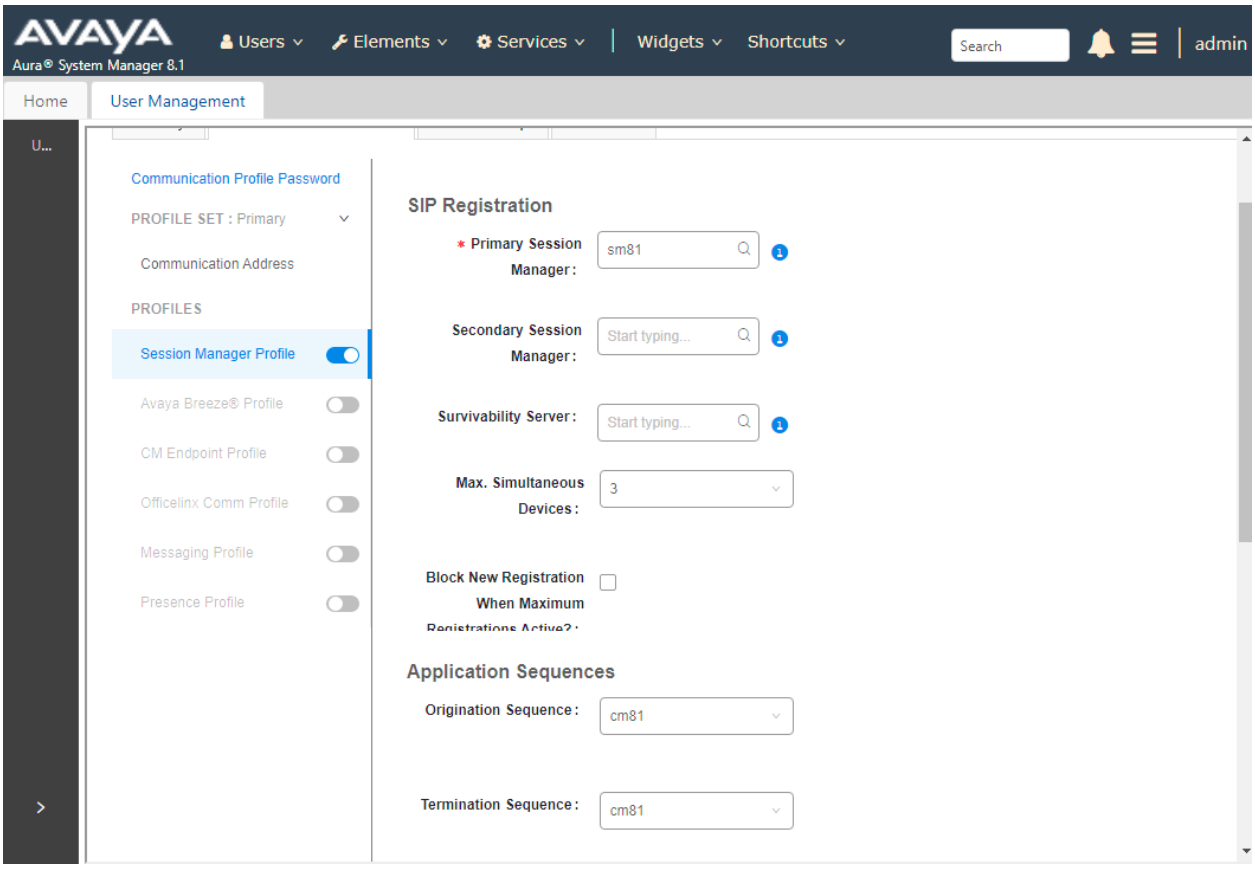
### 6.2.3. Communication Profile

Click the **Communication Profile** tab and in the **Comm-Profile Password** and **Re-enter Comm-Profile Password** fields, enter a numeric password. This will be used to register the device during login. Click **OK**.



### 6.2.4. Session Manager Profile

Click on the **Session Manager Profile** slide button. For **Primary Session Manager**, **Origination Sequence**, **Termination Sequence,** and **Home Location** (not shown), select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.

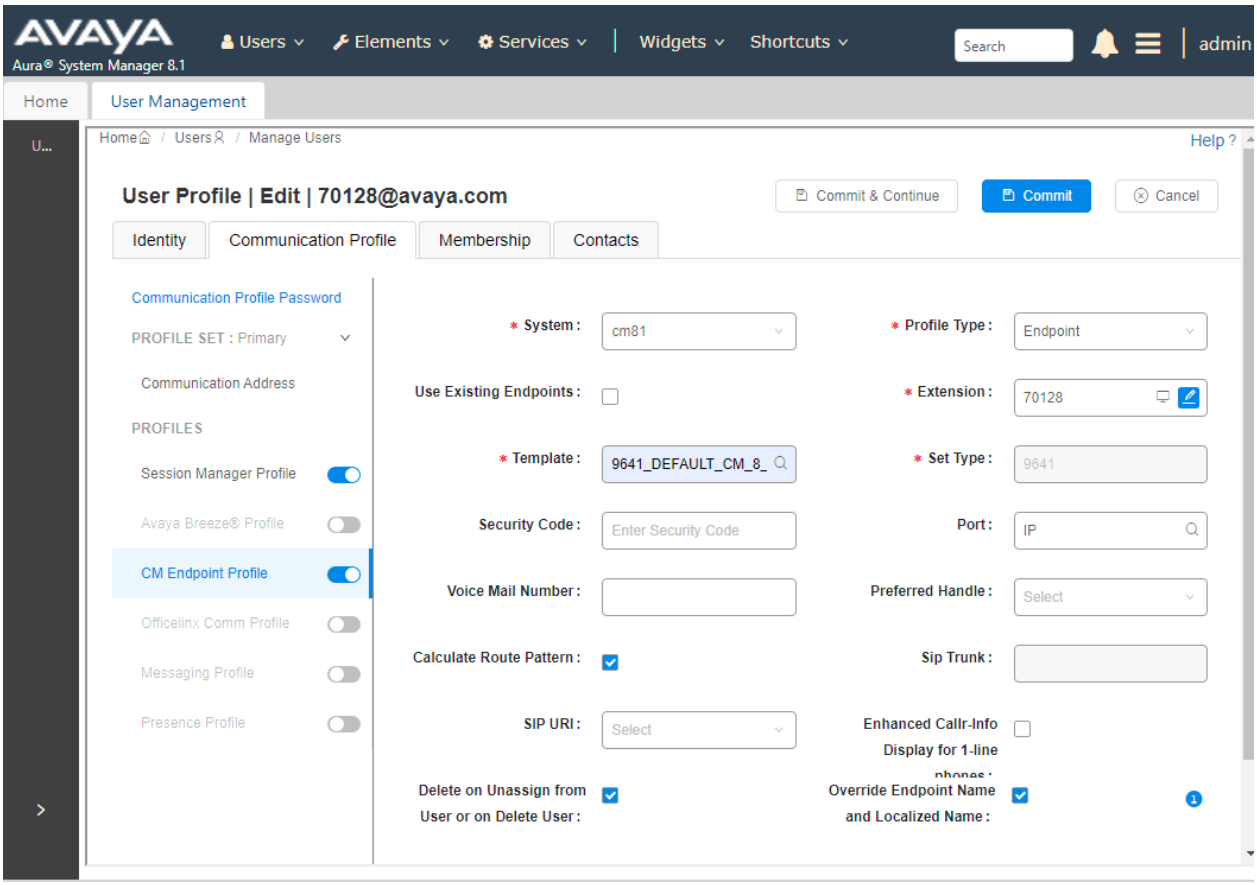


### 6.2.5. CM Endpoint Profile

Click on the **CM Endpoint Profile** slide button. Fill in the following fields:

- **System**: Select the relevant Communication Manager SIP Entity (e.g., **cm81**)
- **Profile Type**: Select **Endpoint**
- **Template**: Select **9641_DEFAULT_CM_8_1** for **XLIP**
- **Extension**: Enter the extension number (e.g., **70128**)

Click on **Endpoint Editor** in the Extension field to edit Communication Manager settings if desired. Click **Commit**.



# 7. Configure Med-Pat XLIP

This section provides the steps for configuring XLIP. The steps to configure XLIP to integrate with Communication manager are as follows:

- Configure IP Address of XLIP
- Launch Web Interface
- Configure SIP account
- Modify Codec Settings

## 7.1. Configure IP Address of XLIP

XLIP is configured for DHCP as a factory default. The following steps provide network connectivity and determine the phone IP address for use in launching administration detailed in **Section 7.2**:

- Connect the WAN port of XLIP to a Power over Ethernet (PoE) switch
- Determine the assigned IP address using an IP scanner or a keycode that is input to XLIP. In this case, enter the code **47*#** (e.g., '**IP*#**') to get the IP address read out

**Note:** Additional keycodes that may be useful during setup are:

- **47*#** IP address
- **48*#** Firmware Version
- **49*#** Factory Reset
- **50*#** VLAN Tag On/Off (Only on or off)
- **51*#** DHCP/Static IP Mode
- **52*#** Last 4 Octets of MAC Readback

## 7.2. Launch Web Interface

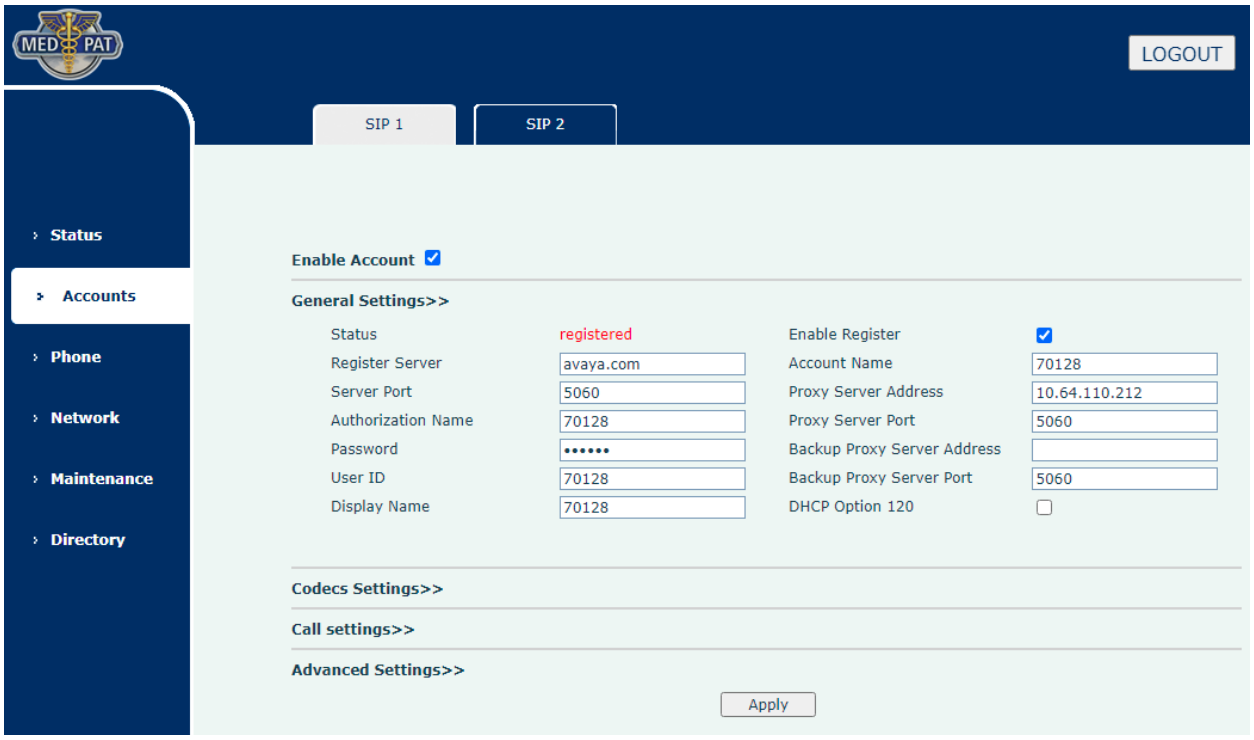
- Invoke the web interface using the IP address from **Section 7.1** using the URL **http://<IP address>.**
- In the login dialog, enter an appropriate password

## 7.3. Configure SIP Account

The SIP account used to register with Session Manager is configured here. From the Web Interface, Select **Accounts** from left pane. Click on the **Enable Account** checkbox. The account settings become visible. Input the following values:

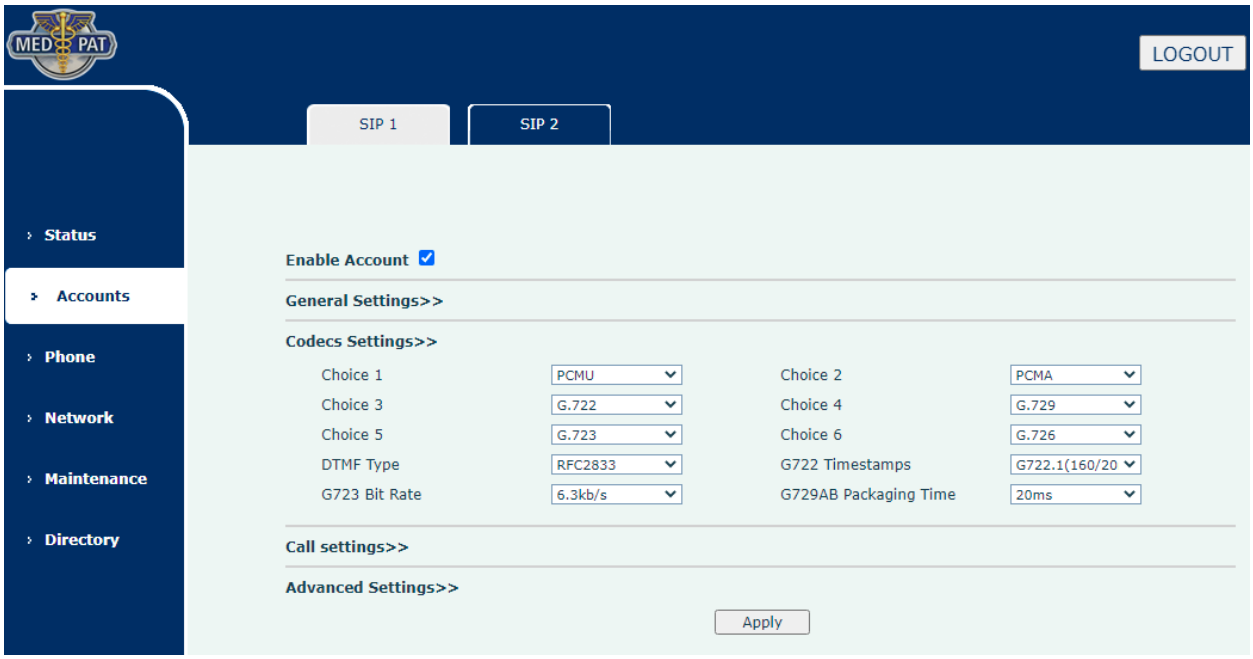
- **Register Server:** The Session Manager domain (e.g., **avaya.com**)
- **Account Name:** Any appropriate descriptive string (e.g., **70128)**
- **Proxy Server Address:** Session Manager IP address
- **Authorization Name:** SIP user ID as created in **Section 6.2.1** (e.g., **70128**)
- **Password**: SIP user password used in **Section 6.2.3**
- **User ID:** User extension used for SIP user created in **Section 6.2** (e.g., **70128**)
- **Display Name:** Any appropriate descriptive string (e.g., **70128**)

Click **Apply.**



## 7.4. Modify Codec Settings

Continuing from the **Accounts** pane, select **Codecs Settings**>>. Modify the priority order and codecs as desired and click **Apply**.



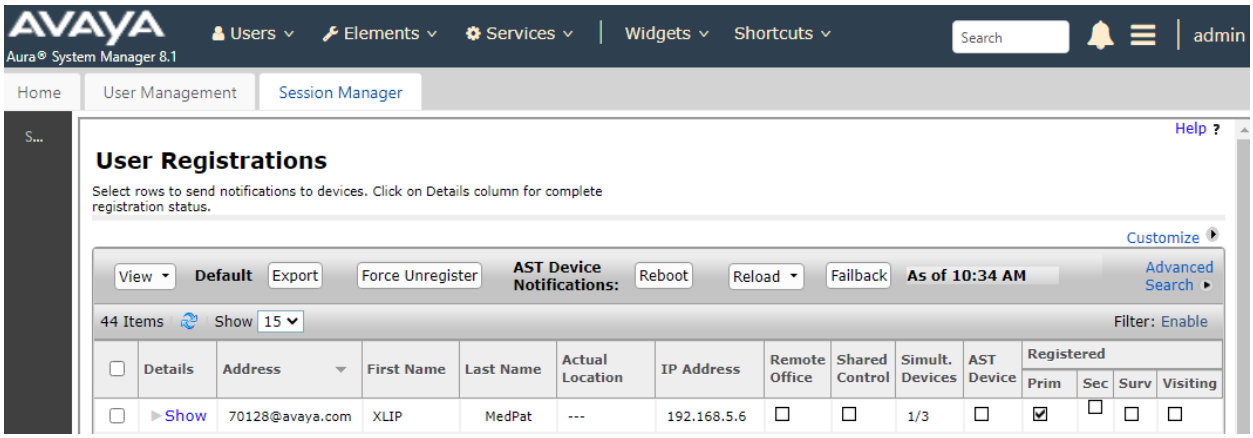
# 8. Verification Steps

The proper configuration of XLIP with Avaya Session Manager and Avaya Communication Manager is verified by the following.

## 8.1. View Session Manager Status

Verify XLIP has successfully registered with Session Manager. In System Manager, navigate to **Elements →Session Manager →System Status →User Registrations** to check the registration status.

- Verify that XLIP (here **70128**) is registered with Session Manager by noting the registered users include **70128**.




## 8.2. Call Verification

Verify that basic calls can be made from and to XLIP and another telephone registered to Session Manager.

# 9. Conclusion

These Application Notes describe the configuration steps required for Med-Pat XLIP V2 SIP one-piece telephone to successfully interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

# 10. Additional References

This section references the product documentation relevant to these Application Notes.

Product documentation for Avaya products may be found at http://support.avaya.com.

1. *Administering Avaya Aura® Communication Manager*, Issue 10, Release 8.1.x, March 2021.
2. *Administering Avaya Aura® Session Manager*, Issue 8, Release 8.1.x, February 2021.
3. *Administering Avaya Aura® System Manager*, Issue 11, Release 8.1.x, April 2021.

Product documentation for Med-Pat products may be found at http://www.med-pat.com.

4. *VOIP Phone Model XL-IP User Manual*.