Avaya Solution & Interoperability Test Lab

Application Notes for Spectralink 84-Series Wireless Telephones 7.5 with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Spectralink 84-Series Wireless Telephones with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Spectralink 84-Series Wireless Telephones registered with Avaya Aura® Session Manager via SIP. The Spectralink wireless telephones communicate with Avaya Aura® Session Manager over a converged wireless network.

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 Spectralink 84-Series Wireless Telephones with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Spectralink 84-Series Wireless Telephones registered with Avaya Aura® Session Manager via SIP. The Spectralink 8440 Wireless Telephones were used for the compliance test. The Spectralink wireless telephones communicate with Avaya Aura® Session Manager over a converged wireless network.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between Spectralink 84-Series Wireless Telephones, Avaya SIP / H.323 deskphones, and the PSTN, and exercising basic telephony features, such as hold, mute, transfer and conference. Additional telephony features, such as call forward, call coverage, call park/unpark, and call pickup were also verified using Communication Manager Feature Access Codes (FACs).

The serviceability testing focused on verifying that the Spectralink 84-Series Wireless Telephones come back into service after re-connecting the access point or rebooting the phone.

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 Spectralink 84-Series Wireless Telephones did not include use of any specific encryption features as requested by Spectralink.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of Spectralink 84-Series Wireless Telephones with Session Manager.
- Calls between Spectralink 84-Series Wireless Telephones and Avaya SIP / H.323 deskphones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between the Spectralink 84-Series Wireless Telephones and the PSTN.
- G.711, G.729 and G.722 codec support.
- Proper recognition of DTMF tones.
- Basic telephony features, including hold, mute, redial, multiple calls, and 3-party 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.
- Use of programmable buttons on the Spectralink 84-Series Wireless Telephones.
- Proper system recovery after a restart of the Spectralink 84-Series Wireless Telephones and loss of IP connectivity.

2.2. Test Results

All test cases passed with the exception that blind conference is not supported.

2.3. Support

For technical support and information on Spectralink 84-Series Wireless Telephones, contact Spectralink technical support at:

- Phone: 1-800-775-5330
- Website: http://support.spectralink.com/
- Email: technicalsupport@spectralink.com

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 with Avaya G430/G450 Media Gateway.
- Media resources in the Avaya G430/G450 Media Gateway and Avaya Aura® Media Server (not shown in figure).
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya Messaging serving as the voicemail system.
- Avaya 96x1 Series H.323 and SIP Deskphones.
- Spectralink 8440 Wireless Telephones.
- FTP and DHCP Servers that provide configuration data and IP network information to Spectralink 84-Series Wireless Telephones (not shown).
- A Spectralink-approved wireless access point was used to provide Spectralink wireless telephones access to the converged wireless network (not shown).

Spectralink 84-Series Wireless Telephones registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.

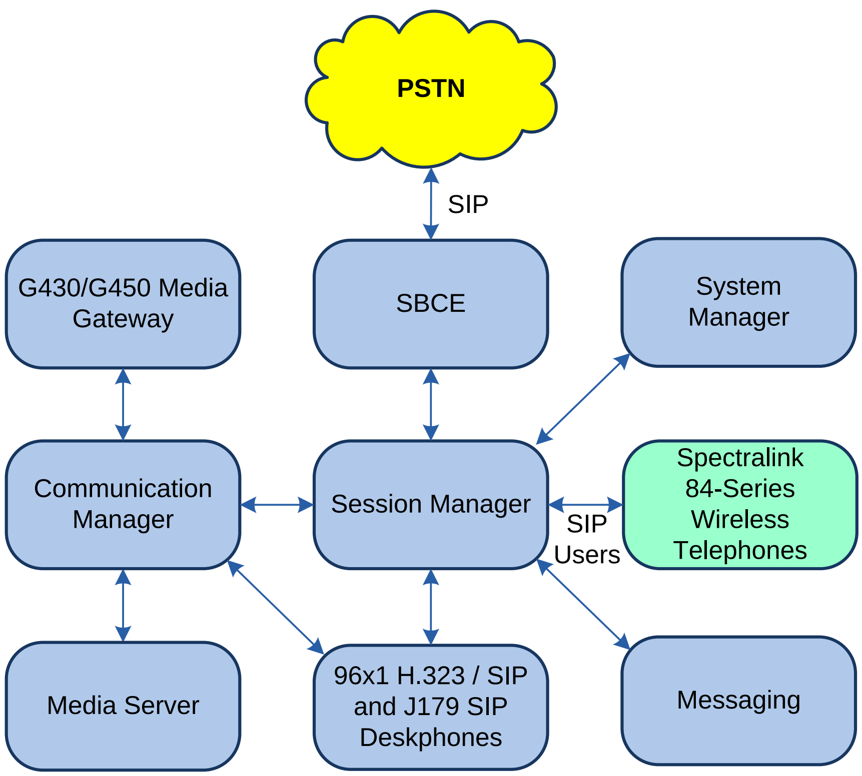


Figure 1: Avaya SIP Network with Spectralink 84-Series Wireless Telephones

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	10.1.0.1.0-SP1
Avaya G430 Media Gateway	FW 42.8.0
Avaya G450 Media Gateway	FW 42.7.0
Avaya Aura® Media Server	v.10.1.0.77
Avaya Aura® System Manager	10.1.0.1 Build No. – 10.1.0.0.537353 Software Update Revision No: 10.1.0.1.0614394 Service Pack 1
Avaya Aura® Session Manager	10.1.0.1.1010105
Avaya Session Border Controller for Enterprise	10.1.1.0-35-21872
Avaya Aura® Messaging	11.0.0.3204
Avaya 96x1 Series IP Deskphones	6.8.5.3.2 (H.323) 7.1.15.0.14 (SIP)
Avaya J100 Series Phones	4.0.13.0.6
Spectralink 84-Series Wireless Telephones	7.5.1.2274

5. Configure Avaya Aura® Communication Manager

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

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

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

Note: It is assumed that basic configuration of the Communication Manager has already been completed. The SIP station configuration for the Spectralink 84-Series Wireless Telephones 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: V20                              Software Package: Enterprise
       Location: 2                                 System ID (SID): 1
       Platform: 28                                Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 48000    106
                                      Maximum Stations: 36000     36
                              Maximum XMOBILE Stations: 36000      0
                    Maximum Off-PBX Telephones - EC500: 41000      0
                    Maximum Off-PBX Telephones -   OPS: 41000     22
                    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 Node Names

In the **IP Node Names** form, assign an IP address and host name for Communication Manager (*procr*) and Session Manager (*devcon-sm*). The host names will be used in other configuration screens of Communication Manager.

```
change node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
default             0.0.0.0
devcon-aes          10.64.102.119
devcon-ams          10.64.102.118
devcon-sm           10.64.102.117
procr               10.64.102.115
procr6              ::

( 6  of 6    administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

5.3. Administer IP 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 Media Gateway or Avaya Aura® 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.

```
change ip-network-region 1                                        Page   1 of  20
                               IP NETWORK REGION
  Region: 1       NR Group: 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 the Spectralink 84-Series Wireless Telephones. 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. The default settings of the **IP Codec Set** form are shown below. The Spectralink 84-Series Wireless Telephones were tested using G.711, G.729, and G.722 codecs. Specify the desired codecs in the **IP Codec Set** form as per customer requirements.

Spectralink Versity was not configured to support SRTP, so *none* 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: 2-srtp-aescm128-hmac32
 3: none
 4:
 5:
```

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*.
- Set the **Enforce SIPS URI for SRTP** field to *n*.
- 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.
- 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*.
- **Initial IP-IP Direct Media** was enabled.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
add signaling-group 10                                      Page   1 of   2
                                SIGNALING GROUP

 Group Number: 10             Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? n        Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: SM                       Clustered? n
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                  Far-end Node Name: devcon-sm
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1

Far-end Domain: avaya.com
                                             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? y
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 Spectralink Versity, Avaya SIP deskphones, and Avaya Messaging. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie* or *public-ntwrk*, 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 default values for the remaining fields.

```
add trunk-group 10                                           Page   1 of  5
                                TRUNK GROUP

Group Number: 10                   Group Type: sip           CDR Reports: y
  Group Name: To devcon-sm                COR: 1        TN: 1        TAC: 1010
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 10
                                                     Number of Members: 10
```

5.5. Administer AAR Call Routing

SIP calls to Session Manager are routed over a SIP trunk via AAR call routing. Configure the AAR analysis form and enter add an entry that routes digits beginning with "78" to route pattern 10 as shown below.

```
change aar analysis 78                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 1

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    78                    5    5     10       lev0         n
```

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

```
change route-pattern 10                                         Page   1 of   3
                    Pattern Number: 10     Pattern Name: To devcon-sm
    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: 10   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                               unk-unk   none
 2: y y y y y n  n            rest                                         none
```

6. Configure Avaya Aura® Session Manager

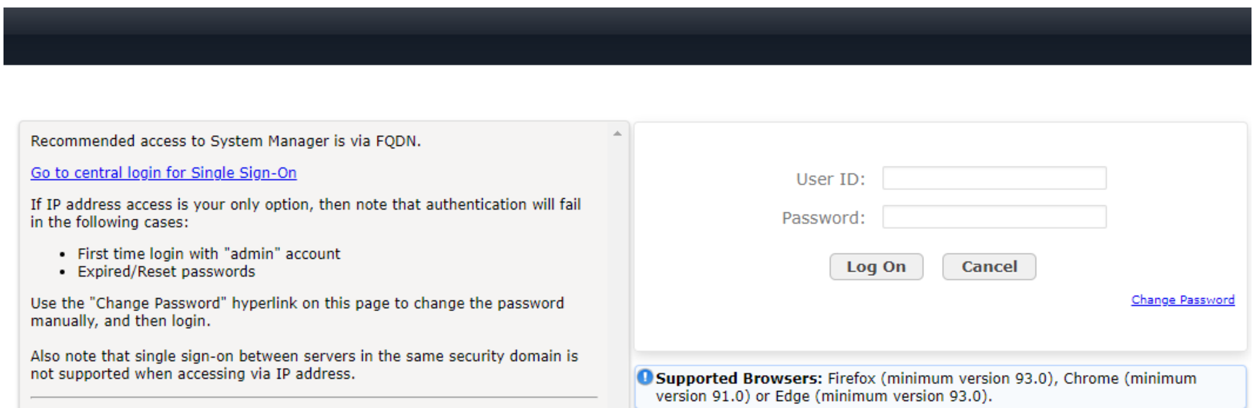
This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Set Network Transport Protocol for Spectralink 84-Series Wireless Telephones
- 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 the Spectralink 84-Series Wireless Telephones.

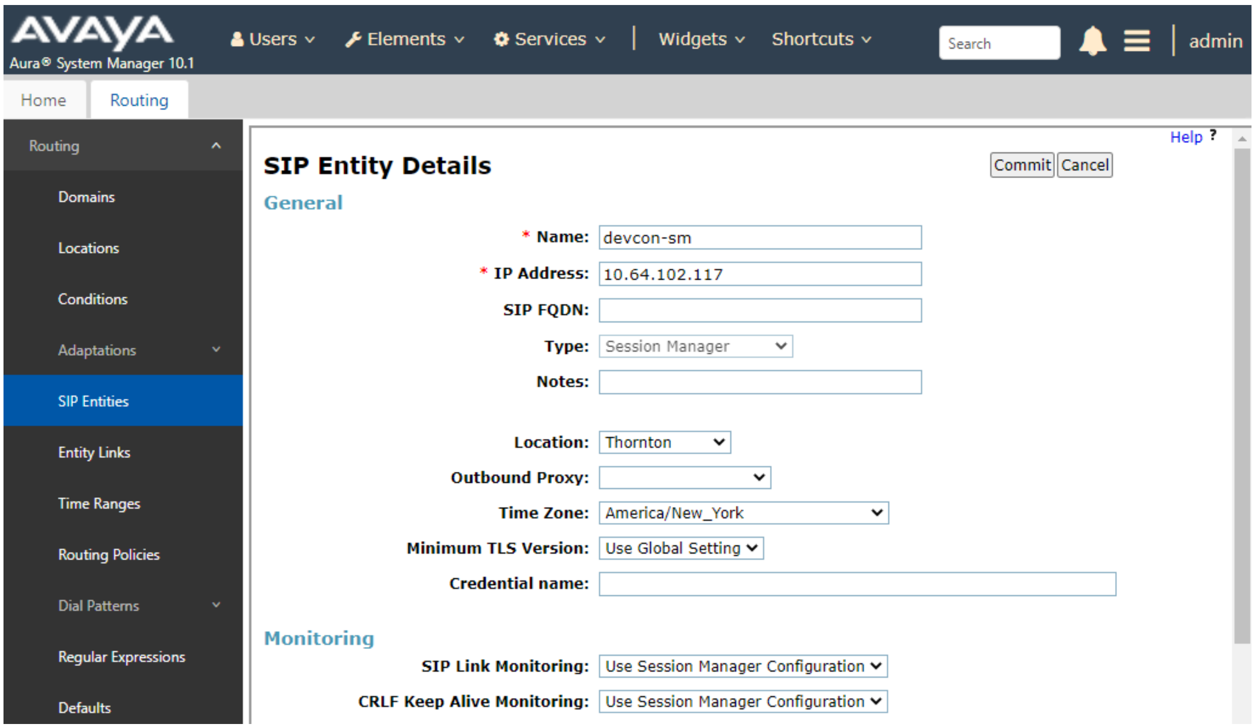
6.1. Launch System Manager

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.

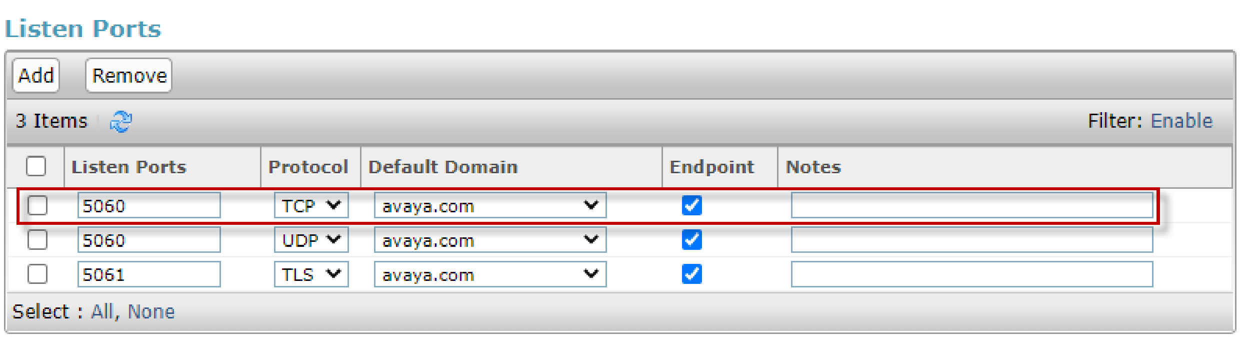


6.2. Set Network Transport Protocol for Spectralink 84-Series Wireless Telephones

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

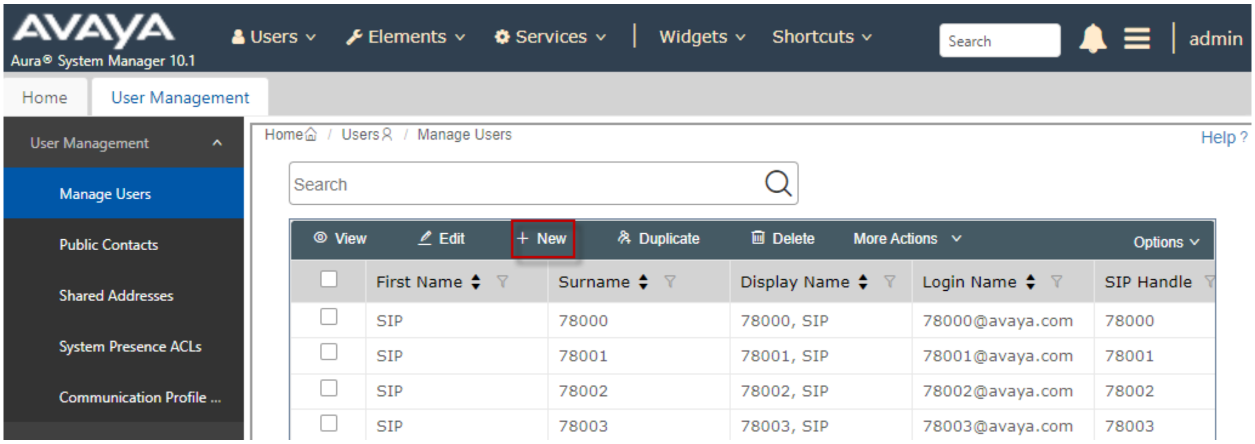


Scroll down to the **Listen Ports** section and verify that the transport network protocol used by Spectralink telephones is specified in the list below. For the compliance test, the Spectralink telephones used TCP network transport as specified in the `site.cfg` file configured in **Section 7.3**.



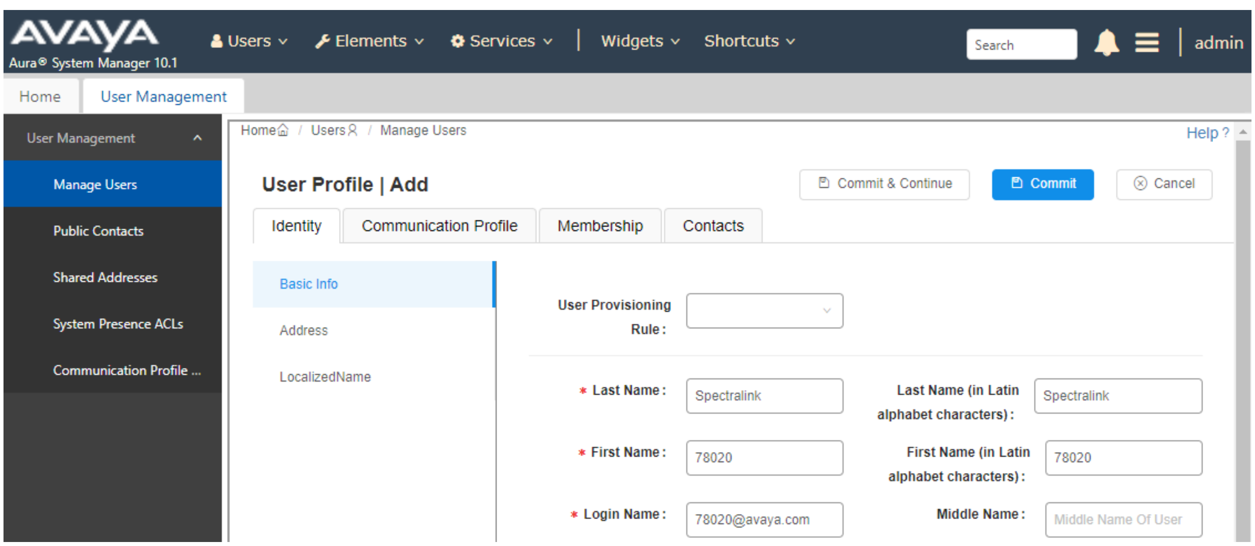
6.3. Administer SIP User

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.



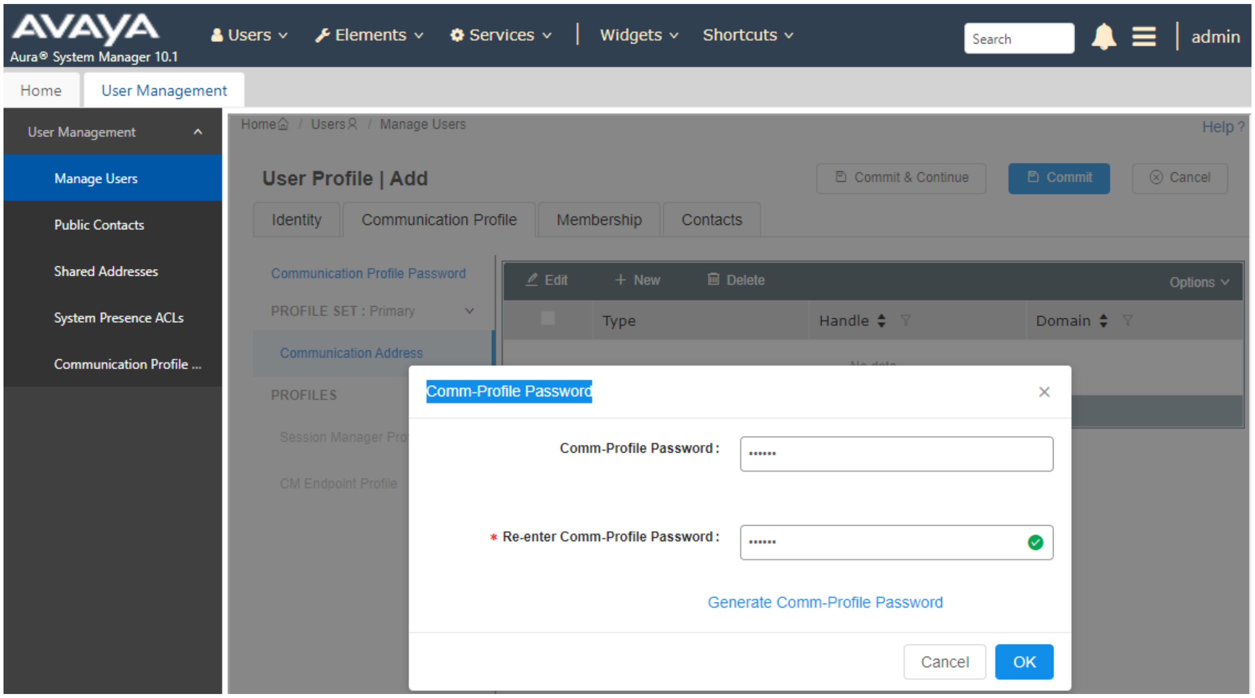
6.3.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter "*<ext>@<domain>*", where "*<ext>*" is the desired Spectralink 84-Series Wireless Telephone SIP extension and "*<domain>*" is the applicable SIP domain name from **Section 5.2**. Retain the default values in the remaining fields.



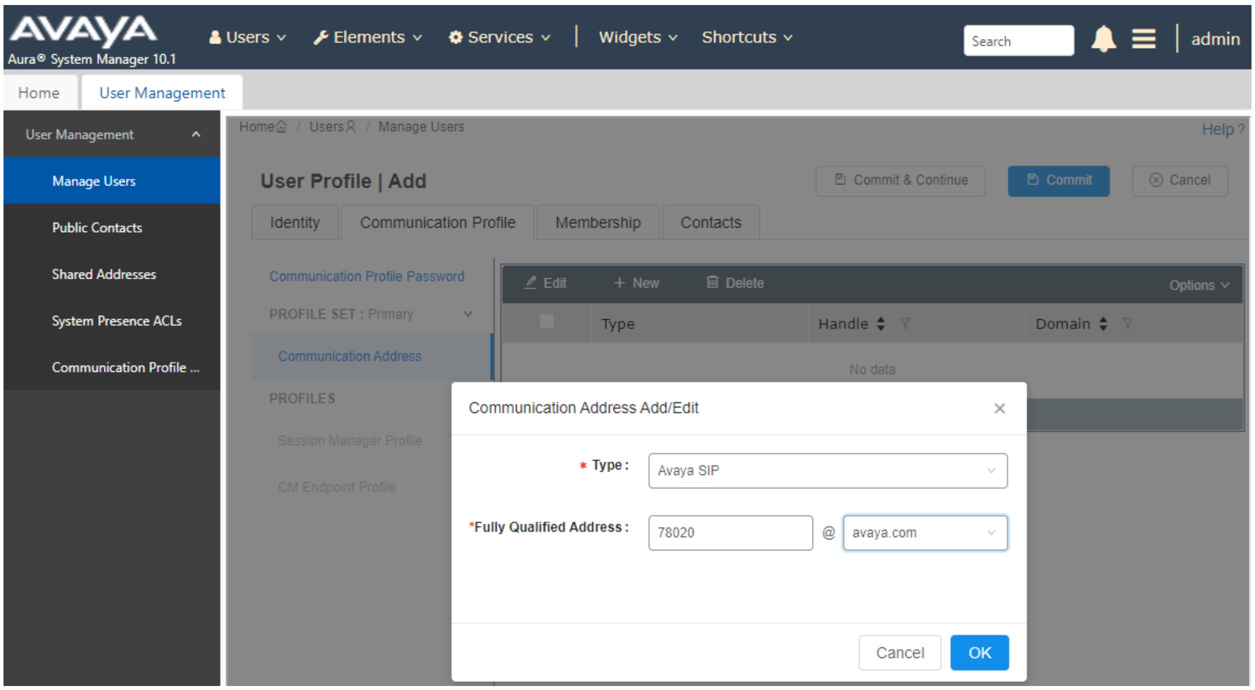
6.3.2. Communication Profile

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**.



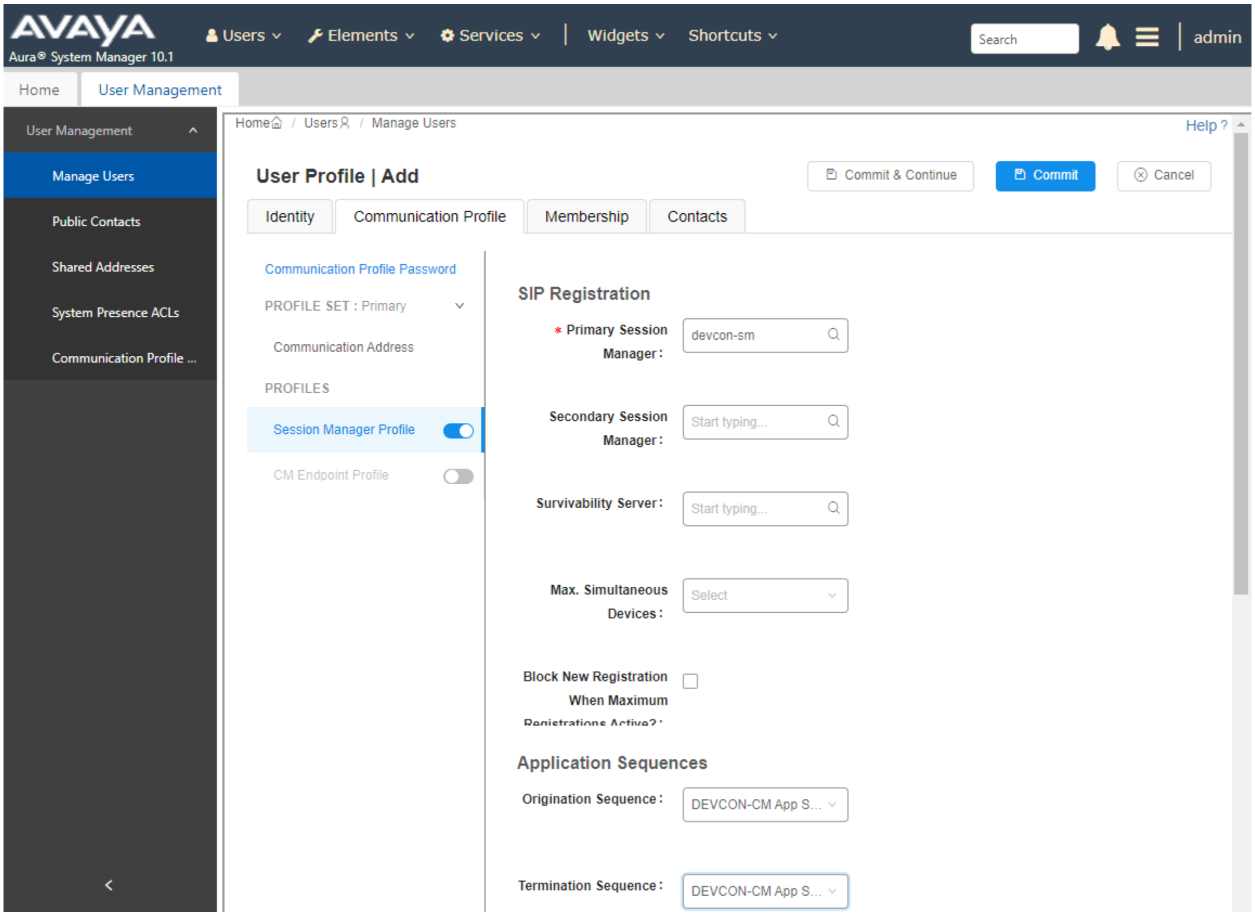
6.3.3. Communication Address

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**, 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 **OK**.

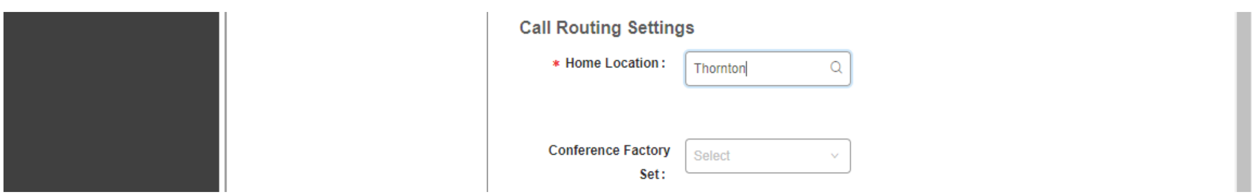


6.3.4. Session Manager Profile

Click on toggle button by **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.

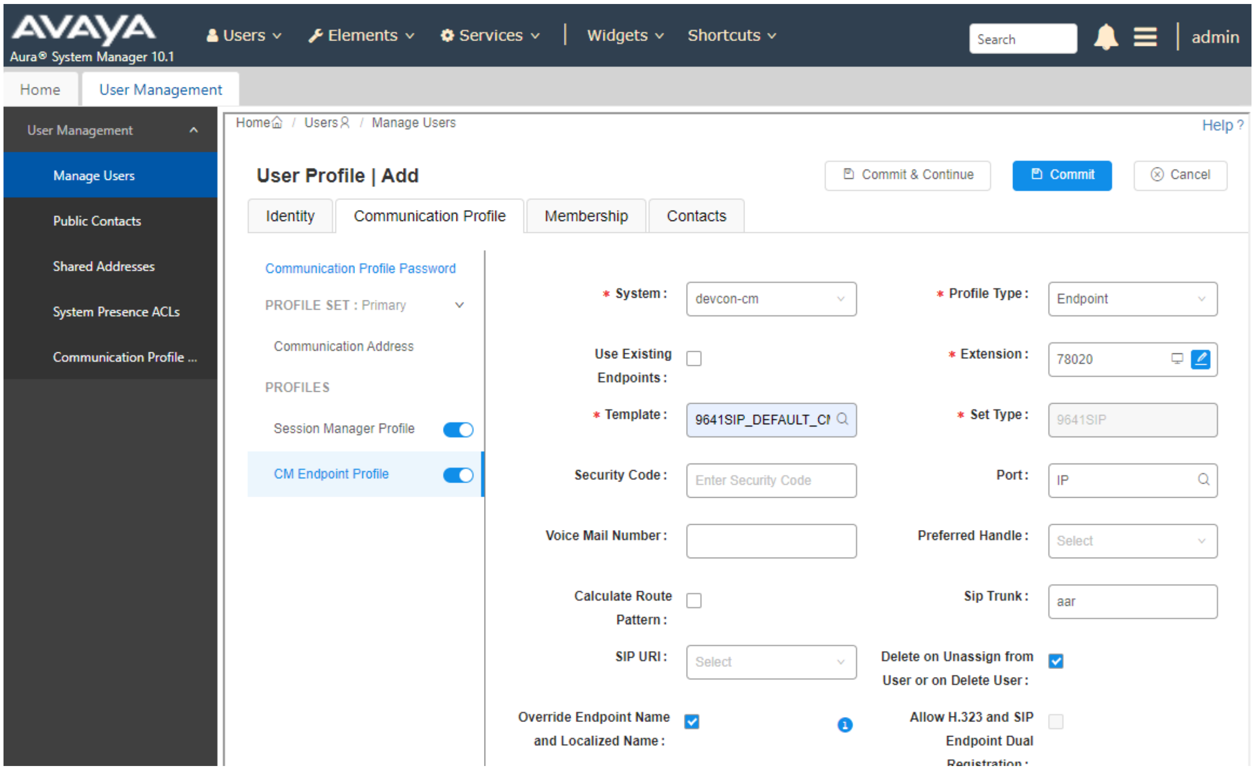


Scroll down to the **Call Routing Settings** section to configure the **Home Location**.

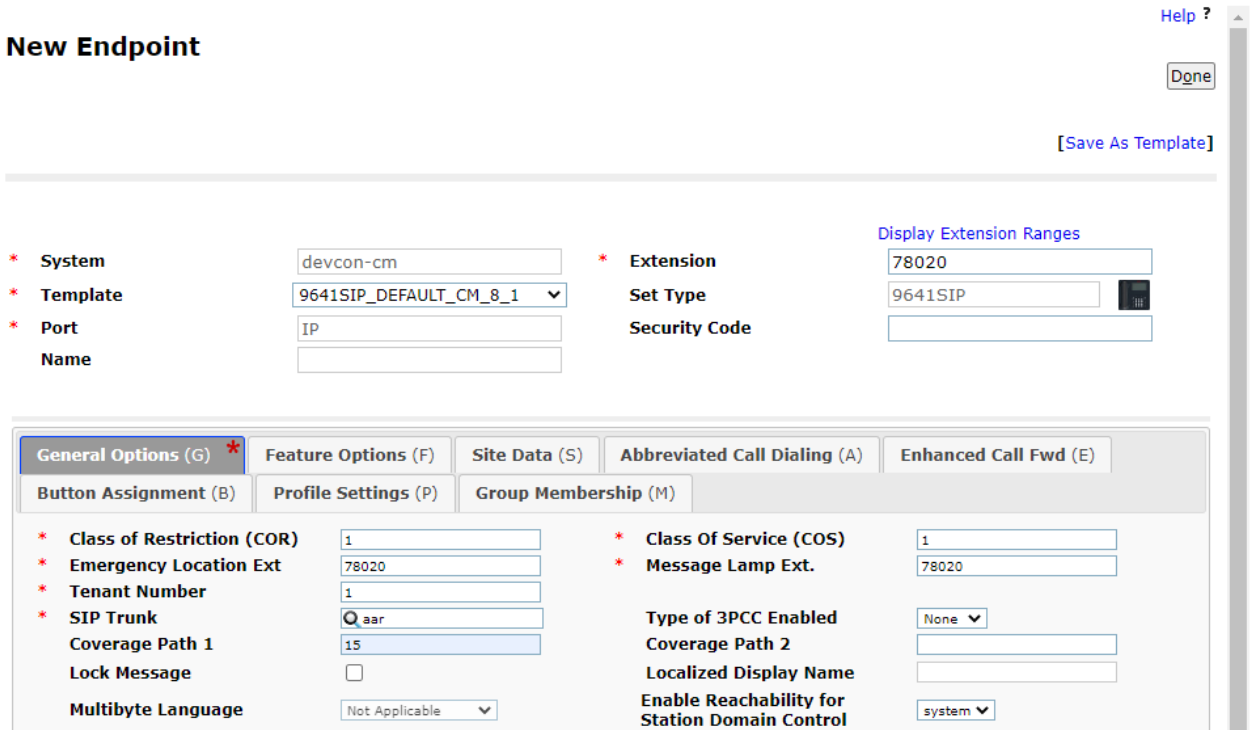


6.3.5. CM Endpoint Profile

Click on the toggle button by **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.1**. For **Template**, select *9641SIP_DEFAULT_CM_8_1*. For **Port**, click and select *IP*. Retain the default values in the remaining fields. Click on the 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. Click **Done** (not shown) to return to the previous web page and then **Commit** to save the configuration (not shown).



7. Configure Spectralink 84-Series Wireless Telephones

This section covers the SIP configuration of the Spectralink 84-Series Wireless Telephones. Refer to **[5] in Section 9** for more information on configuring the Spectralink 84-Series Wireless Telephones. The procedure covers the following areas:

- Configure DHCP Server
- Configure FTP Server
- Edit `site.cfg`
- Edit `<mac-address>-ext.cfg`

7.1. Configure DHCP Server

The Spectralink 84-Series Wireless Telephones must first acquire several IP network settings before proceeding with provisioning. These settings were automatically obtained from a DHCP server. Alternatively, the Spectralink telephones could be configured with static IP addresses, but for the compliance test, a DHCP server was used.

In addition to obtaining IPv4 addresses from the DHCP server for each Spectralink telephone, the DHCP server also provided the following settings:

- Option 3: Default Gateway
- Option 6: DNS Server (optional)
- Option 66: FTP Server (or Provisioning Server)

7.2. Configure FTP Server

By default, Spectralink sets FTP as the provisioning protocol on Spectralink wireless telephones. For the compliance test, a free and popular server, FileZilla Server, available for Windows was used. Refer to **[5]** for instructions on setting up the FTP server, such as specifying the FTP username and password. The Spectralink telephones will receive configuration parameters from XML files placed on the FTP or Provisioning server and will also upload log files detailing their operation. The two required XML files are `site.cfg` and `<mac-address>-ext.cfg` described in the following sections. The uploaded log files will appear as `<mac-address>-app.log` files, where `<mac-address>` is the MAC address of the Spectralink handset. These XML files are located in the folder specified in the FTP server configuration.

7.3. Edit `site.cfg` File

The `site.cfg` file will be used by all of the Spectralink handsets and should provide parameters that are common to all phones. The following parameters were set in this file:

- **reg.1.server.1.address** Set to the SIP signaling IP address of Session Manager.
- **reg.1.server.1.transport** Set to TCP transport.
- **msg.mwi.1.callback** Set to the voicemail pilot number.

```
<openSIP>
      <SIPserver
             reg.1.server.1.address="10.64.102.117"
             reg.1.server.1.expires="120"
             reg.1.server.1.transport="TCPpreferred"
           />
       <dialplan
             dialplan.impossibleMatchHandling="2"
             dialplan.digitmap="" />
      <DND_CallForwarding
             voIpProt.SIP.serverFeatureControl.dnd="0"
             voIpProt.SIP.serverFeatureControl.cf="0"
             voIpProt.SIP.use486forReject="1"
           >
      </DND_CallForwarding>
      <voicemail
             up.oneTouchVoicemail="1"
             up.mwiVisible="1"
             msg.mwi.1.callBackMode="contact"
             msg.mwi.1.callBack="78500"
           np.normal.alert.messageWaiting.tonePattern="silent">
      </voicemail>
</openSIP>
```

7.4. Edit `<mac-address>-ext.cfg` Files

There will be one of these XML files per handset. This file should contain parameters that are handset-specific and that aren't specified in the `site.cfg` file because they are unique to a particular phone. Edit the following parameters in each `<mac-address>-ext.cfg` file:

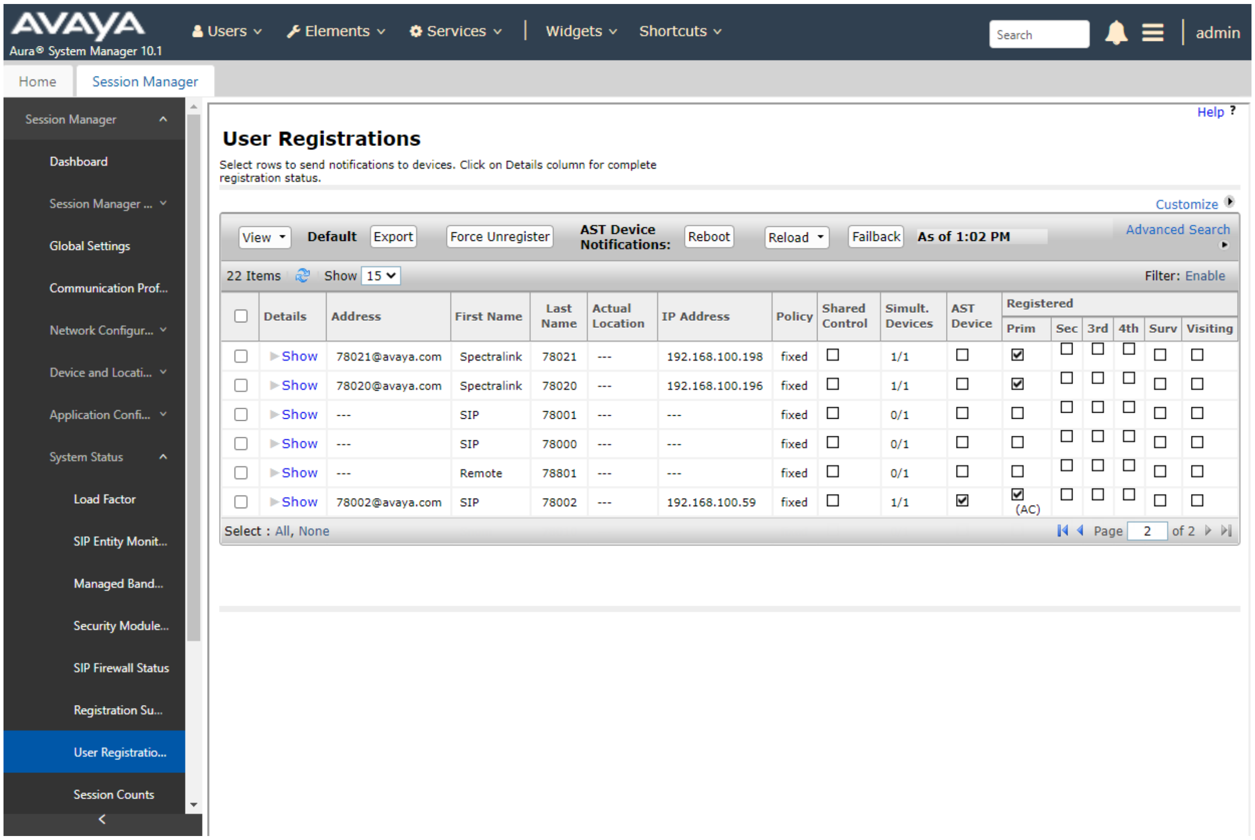
- **reg.1.address** Set to the SIP extension of the handset (e.g., *78020*).
- **reg.1.label** Set to the SIP extension of the handset.
- **reg.1.displayName** Set to the SIP extension of the handset.
- **reg.1.auth.userId** Set to the SIP extension of the handset, which is the authentication user ID for registering with Session Manager.
- **reg.1.auth.password** Set to the SIP password used for SIP registration with Session Manager.
- **msg.mwi.1.subscribe** Set to the SIP extension of the handset to subscribe to MWI.

```
<?xml version="1.0" encoding="utf-8" standalone="yes"?>
<handsetConfig xmlns:xsi="http://www.w3.org/2001/XMLScheme-instance"
xsi:noNamespaceSchemaLocation="handsetConfig.xsd">
  <LineRegistration>
       <openSIPTelephony>
          <TelephonyLine1
             reg.1.address="78020"
             reg.1.label="78020"
             reg.1.displayName="78020"
           reg.1.auth.userId="78020"
           reg.1.auth.password="123456"
           msg.mwi.1.subscribe="78020"
         >
          </TelephonyLine1>
          <!-- Additional lines: -->
          <!-- * -->
          <!-- Additional telephony lines can be addded (reg.3, etc...) by copying the
TelephonyLine1 group above and -->
          <!-- editing appropriately-->
    </openSIPTelephony>
  </LineRegistration>
</handsetConfig>
```

7.5. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the Spectralink 84-Series Wireless Telephones with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

1. Verify that the Spectralink 84-Series Wireless Telephone has successfully registered with Session Manager. In System Manager, navigate to **Elements → Session Manager → System Status → User Registrations** to check the registration status.



2. Establish a call between 84-Series wireless telephone and a local Avaya deskphone. 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.

```
status trunk 10/1                                             Page   2 of   3
                                CALL CONTROL SIGNALING

Near-end Signaling Loc: PROCR
  Signaling   IP Address                             Port
   Near-end:  10.64.102.115                        : 5061
    Far-end:  10.64.102.117                        : 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:  192.168.100.59                       : 2048
    Far-end:  192.168.100.196                      : 46239

 Video Near:
  Video Far:
 Video Port:
  Video Near-end Codec:               Video Far-end Codec:
```

3. Verify basic telephony features by establishing calls between a Spectralink 84-Series Wireless Telephone and another phone.

8. Conclusion

These Application Notes have described the administration steps required to integrate the Spectralink 84-Series Wireless Telephones with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Spectralink 84-Series Wireless Telephones successfully registered with Session Manager and basic and supplementary telephony features were verified. All test cases passed with observations noted in **Section 2.2**.

9. Additional References

This section references the Avaya and Spectralink documentation relevant to these Application Notes.

[1] *Administering Avaya Aura® Communication Manager*, Release 10.1, Issue 1, December 2021, available at http://support.avaya.com.
[2] *Administering Avaya Aura® System Manager*, Release 10.1.x, Issue 6, June 2022, available at http://support.avaya.com.
[3] *Administering Avaya Aura® Session Manager*, Release 10.1.x, Issue 3, April 2022, available at http://support.avaya.com.

The following Spectralink documentation may be found at http://support.spectralink.com/products/wi-fi/spectralink-84-series-wireless-telephone.

[4] *Spectralink 84-Series Wireless Telephone User Guide*, 1725-86720-000 Rev: W, January 2022.
[5] Spectralink 84-Series Wireless Telephone Administration Guide, 1725-86984-000 Rev: AC, January 2022.
[6] Spectralink 84-Series Wireless Telephone Deployment Guide, Spectralink Software Versions 4.3.x to 6.2.x, 1725-86724-000 Rev: Y.

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.