



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

### **Abstract**

These Application Notes describe the configuration steps required to integrate the Spectralink 84-Series Wireless Telephones with Avaya Aura® Session Manager and Avaya Aura® Communication 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 802.11 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® Session Manager and Avaya Aura® Communication Manager. The Spectralink 84-Series Wireless Telephones registered with Avaya Aura® Session Manager via SIP. The Spectralink 8440 Wireless Telephone was used for the compliance test. The Spectralink wireless telephones communicate with Avaya Aura® Session Manager over a converged 802.11 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) and Feature Name Extensions (FNEs).

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.

### 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 and FNEs 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:

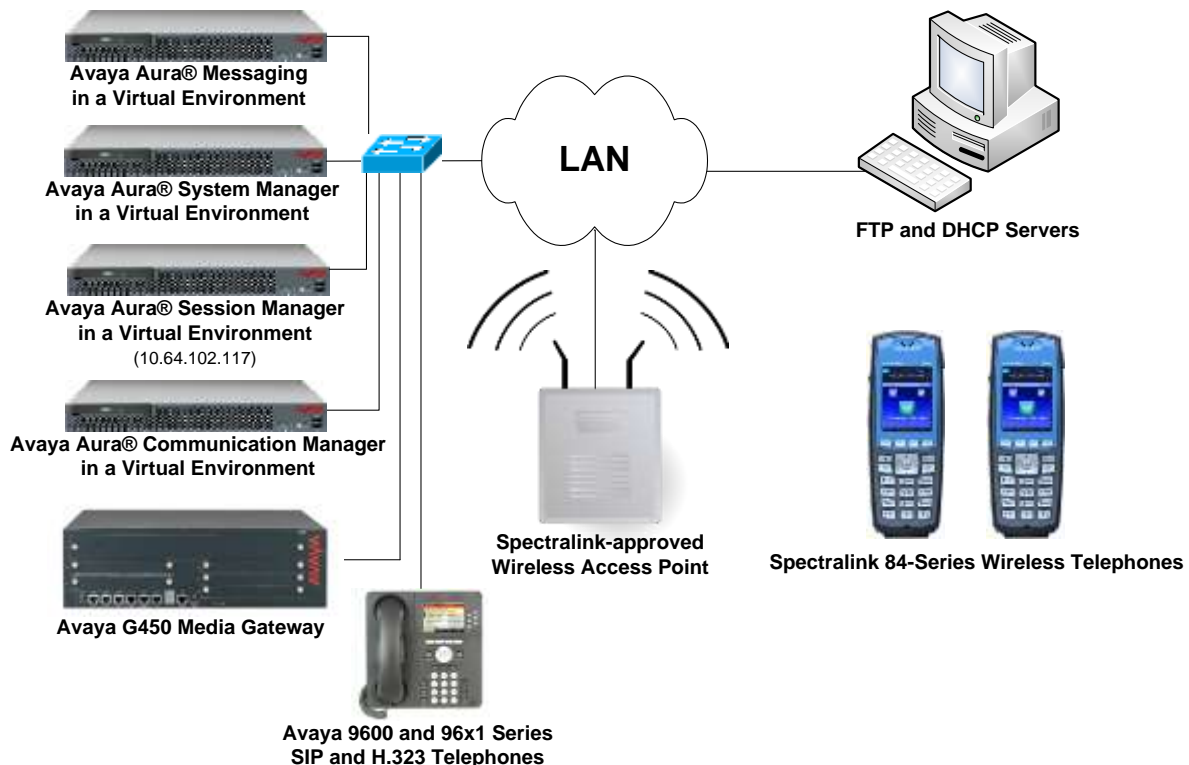
- Phone: 1-800-775-5330
- Website: <http://support.spectralink.com/>
- Email: [technicalsupport@spectralink.com](mailto:technicalsupport@spectralink.com)

### 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 an Avaya G450 Media Gateway.
- Media resources in the Avaya 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 Aura® Messaging serving as the voicemail system.
- Avaya 9600 and 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 8440.
- A Spectralink-approved wireless access point was used to provide Spectralink handsets access to the converged 802.11 wireless network.

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:

<b>Equipment/Software</b>	<b>Release/Version</b>
Avaya Aura® Communication Manager with an Avaya G450 Media Gateway	7.0.1.0 SP 1 (R017x.00.0.441.0 with Patch 23012)
Avaya Aura® Media Server	7.7.0.226
Avaya Aura® Session Manager	7.0.1 (7.0.1.0.701007)
Avaya Aura® System Manager	7.0.1 (Build No. 7.0.0.016266 Software Update Revision No: 7.0.1.0.064859 Feature Pack 1)
Avaya Aura® Messaging	6.3.2 SP 2 Patch 3
Avaya 9600 Series IP Deskphones	3.260A (H.323) 2.6.16.1 (SIP)
Avaya 96x1 Series IP Deskphones	7.0.1.0.46 (SIP)
Spectralink 84-Series Wireless Telephones	4.14.0.2071

## 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 Network Region and IP Codec Set

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, such as the SIP trunk to Session Manager. However, implementers should ensure sufficient Maximum Administered SIP Trunks licenses are available to accommodate the traffic between Communication Manager and the Session Manager. The SIP station configuration for the Spectralink 84-Series Wireless Telephones are 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 56
Maximum Stations: 2400 18
Maximum XMOBILE Stations: 2400 0
Maximum Off-PBX Telephones - EC500: 9600 0
Maximum Off-PBX Telephones - OPS: 9600 10
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

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

## 5.2. 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 **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
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: 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
  
```

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.

```

change ip-codec-set 1                                       Page 1 of 2
                                                           IP CODEC SET
  Codec Set: 1
  

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


```

## 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 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.

AVAYA  
Aura® System Manager 7.0

Recommended access to System Manager is via FQDN.  
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

User ID:

Password:

[Change Password](#)

**Supported Browsers:** Internet Explorer 9.x, 10.x or 11.x or Firefox 36.0, 37.0 and 38.0.



## 6.2. Set Network Transport Protocol for Spectralink

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

The screenshot shows the Avaya System Manager 7.0 interface. The breadcrumb path is Home / Elements / Routing / SIP Entities. The page title is 'SIP Entity Details' with a 'General' tab selected. The form contains the following fields:

- Name: devcon-sm
- FQDN or IP Address: 10.64.102.117
- Type: Session Manager
- Notes: (empty)
- Location: Thornton
- Outbound Proxy: (empty)
- Time Zone: America/New\_York
- Credential name: (empty)

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**.

### Listen Ports

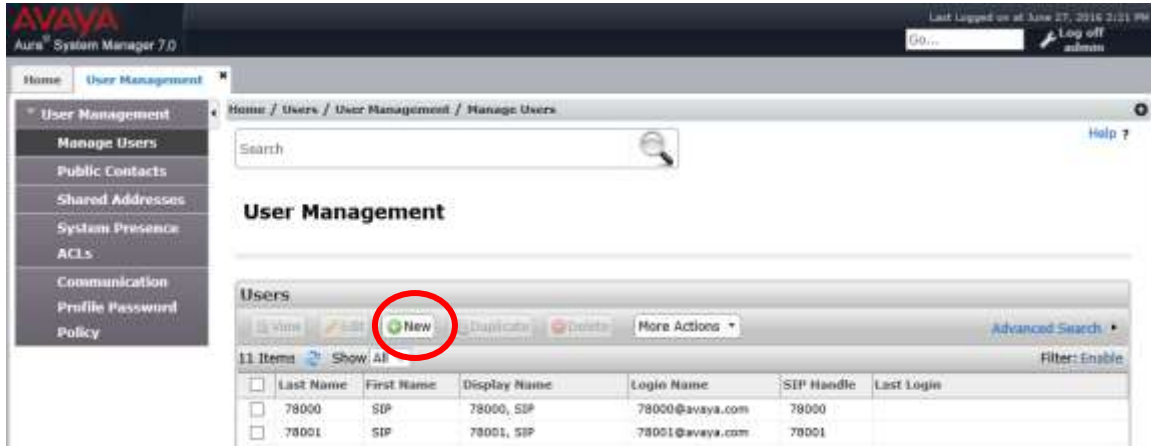
TCP Failover port:   
TLS Failover port:

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

Select : All, None

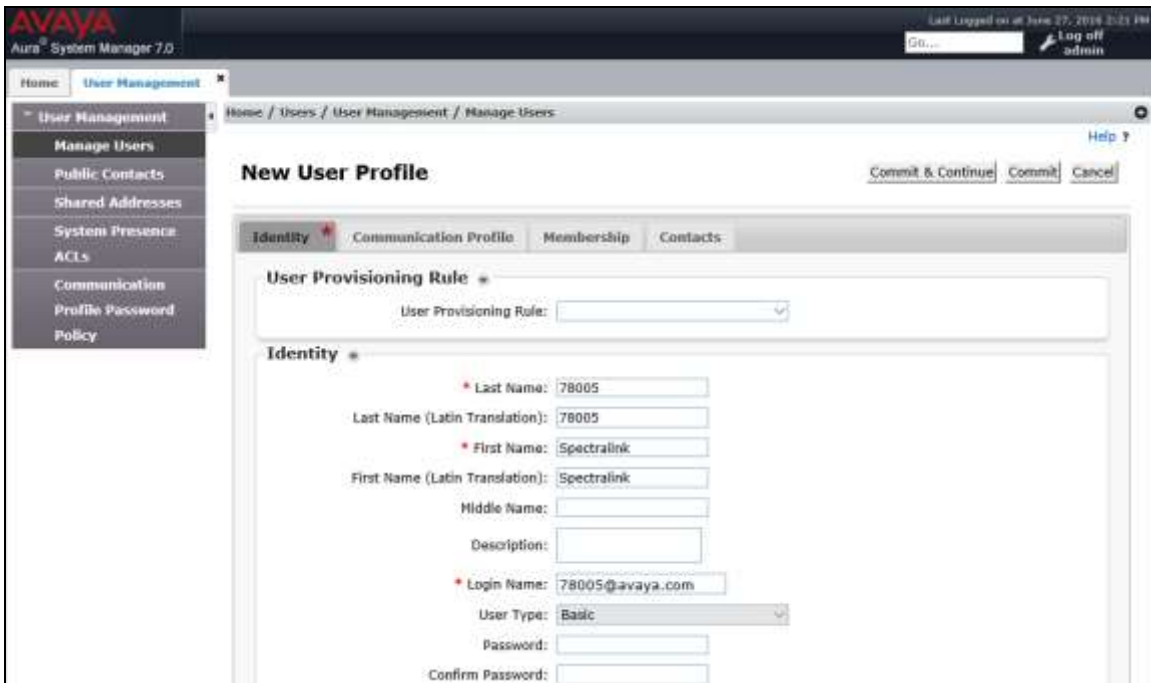
### 6.3. Administer SIP User

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.



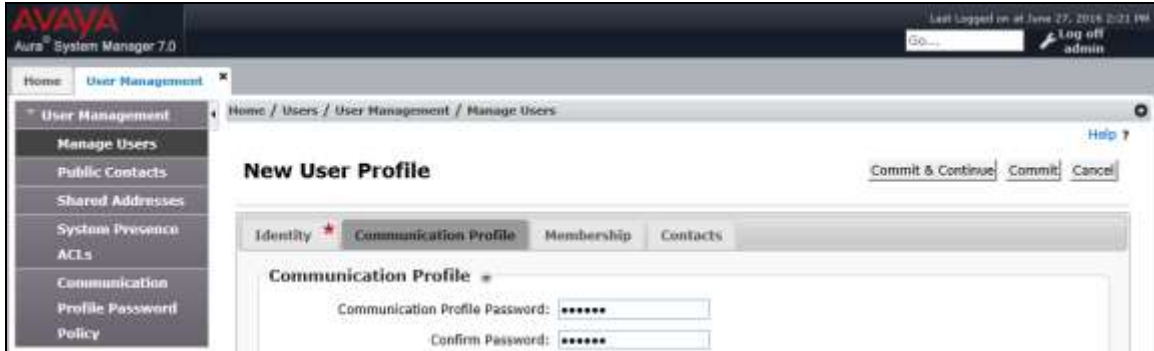
#### 6.3.1. Identity

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

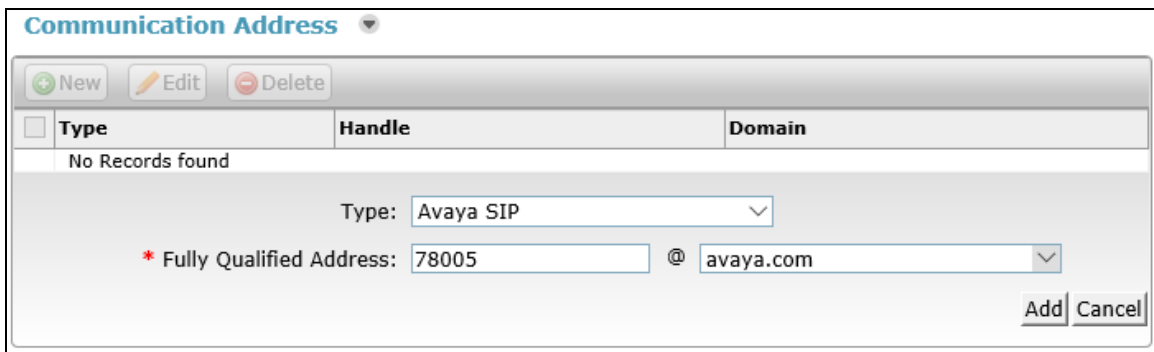
Select the **Communication Profile** tab. For **Communication Profile Password** and **Confirm Password**, enter the desired password for the SIP user to use for registration.



The screenshot shows the 'New User Profile' page in the Avaya Aura System Manager 7.0 interface. The 'Communication Profile' tab is active, and the 'Communication Profile Password' and 'Confirm Password' fields are filled with masked text. The 'Commit & Continue', 'Commit', and 'Cancel' buttons are visible at the top right of the form.

### 6.3.3. Communication Address

In the **Communication Address** sub-section, click **New** to add a new entry. The **Communication Address** sub-section is updated with additional fields as shown below. For **Type**, retain “Avaya SIP”. For **Fully Qualified Address**, enter and select the SIP user extension and domain name to match the login name from **Section 6.3.1**. Click **Add**.



The screenshot shows the 'Communication Address' sub-section. It features a table with columns 'Type', 'Handle', and 'Domain'. Below the table, the 'Type' is set to 'Avaya SIP' and the 'Fully Qualified Address' is set to '78005@avaya.com'. The 'Add' and 'Cancel' buttons are visible at the bottom right.

Type	Handle	Domain
No Records found		

Type: Avaya SIP  
\* Fully Qualified Address: 78005 @ avaya.com

### 6.3.4. Session Manager Profile

Scroll down to check and expand **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.

**Session Manager Profile** ▾

**SIP Registration**

\* Primary Session Manager 

Primary	Secondary	Maximum
11	0	11

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices  ▾

Block New Registration When Maximum Registrations Active?

**Application Sequences**

Origination Sequence  ▾

Termination Sequence  ▾

**Call Routing Settings**

\* Home Location  ▾

Conference Factory Set  ▾

**Call History Settings**

Enable Centralized Call History?

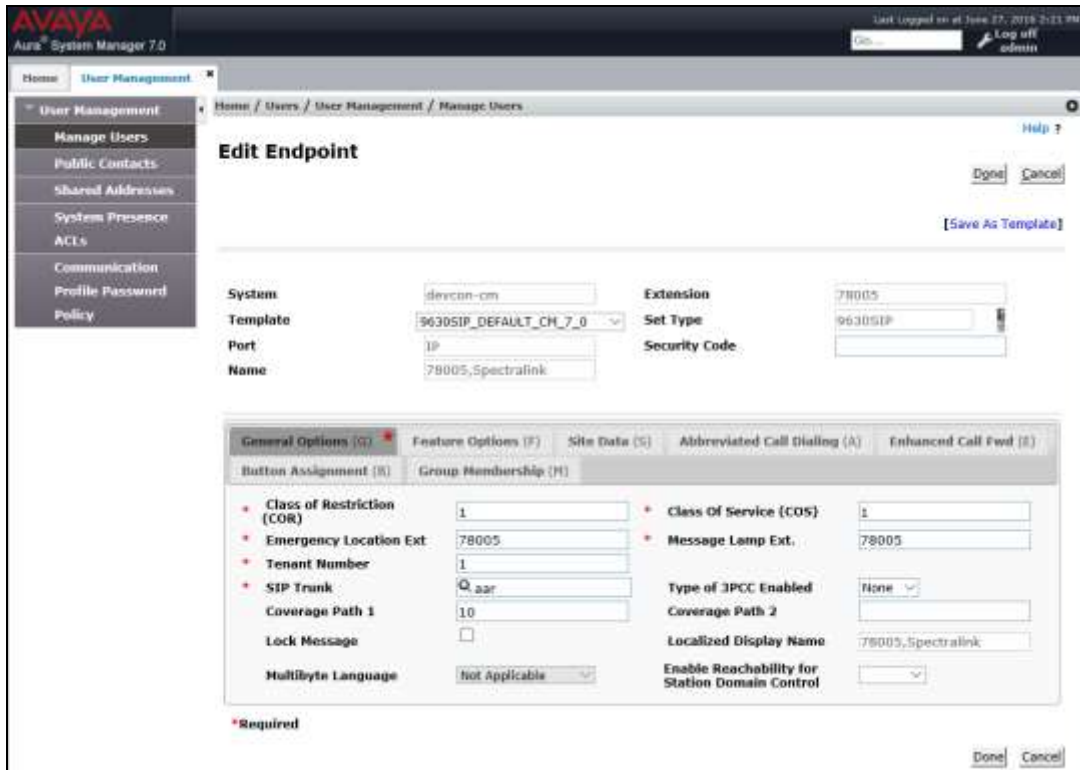
### 6.3.5. CM Endpoint Profile

Scroll down to check and expand **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\_0*. For **Port**, click and select *IP*. Retain the default values in the remaining fields. Click **Commit** to save the configuration (not shown).

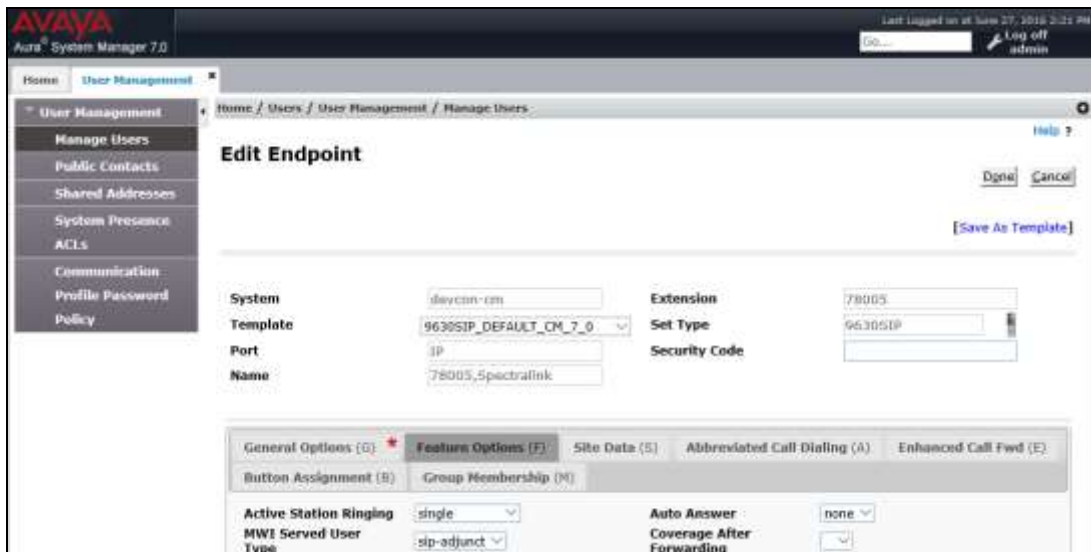
**CM Endpoint Profile** ▾

- \* System  ▾
- \* Profile Type  ▾
- Use Existing Endpoints
- \* Extension
- \* Template  ▾
- Set Type
- Security Code
- Port
- Voice Mail Number
- Preferred Handle  ▾
- Calculate Route Pattern
- Sip Trunk
- Enhanced Callr-Info display for 1-line phones
- Delete Endpoint on Unassign of Endpoint from User or on Delete User
- Override Endpoint Name and Localized Name
- Allow H.323 and SIP Endpoint Dual Registration

In the **CM Endpoint Profile** sub-section, click the **Endpoint Editor** button to display the page below. In the **General Options** tab, specify that coverage path that points to the voicemail system in the **Coverage Path 1** field. This provides voicemail coverage for the SIP user. In this example, coverage path 10 was used.



In the **Feature Options** tab, set the **MWI Served User Type** field to *sip-adjunct*. This allows MWI to be enabled for the SIP user. The voicemail system was connected via SIP to Session Manager.



## 7. Configure Spectralink 84-Series Wireless Telephones

This section covers the SIP configuration of the Spectralink 84-Series Wireless Telephones. Refer to [4] 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 handsets. For the compliance test, a free and popular server, FileZilla Server, available for Windows was used. Refer to [4] 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., 78005).
- **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/XMLSchema-instance"
xsi:noNamespaceSchemaLocation="handsetConfig.xsd">
  <LineRegistration>
    <openSIPTelephony>
      <TelephonyLine1
        reg.1.address="78005"
        reg.1.label="78005"
        reg.1.displayName="78005"
        reg.1.auth.userId="78005"
        reg.1.auth.password="123456"
        msg.mwi.1.subscribe="78005"
      >
    </TelephonyLine1>
    <!-- Additional lines: -->
    <!-- * -->
    <!-- Additional telephony lines can be added (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 Telephone 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. 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. The following Avaya product documentation is available at [support.avaya.com](http://support.avaya.com).

- [1] *Administering Avaya Aura® Communication Manager*, Release 7.0.1, Issue 2, May 2016, Document Number 03-300509.
- [2] *Administering Avaya Aura® Session Manager*, Release 7.0.1, Issue 2, May 2016.

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

- [3] *Spectralink 84-Series Wireless Telephone User Guide*, 1725-86720-000 Rev: N, May 2016.
- [4] *Spectralink 84-Series Wireless Telephone Administration Guide*, 1725-86984-000 Rev: N, June 2016.
- [5] *Spectralink 84-Series Wireless Telephone Deployment Guide*, 1725-86724-000 Rev: T, March 2016.

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