

Avaya Solution & Interoperability Test Lab

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/

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.

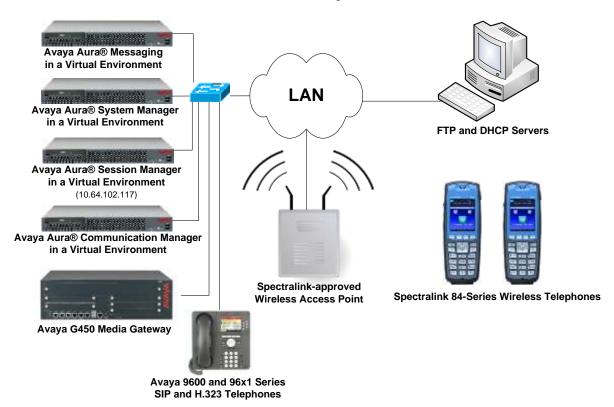


Figure 1: Avaya SIP Network with Spectralink 84-Series Wireless Telephones

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
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
                               Platform Maximum Ports: 6400
                                    Maximum Stations: 2400
                             Maximum XMOBILE Stations: 2400
                   Maximum Off-PBX Telephones - EC500: 9600
                   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
                        Maximum Survivable Processors: 313
        (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
                                                                     1 of 20
                              TP 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

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:
```

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.

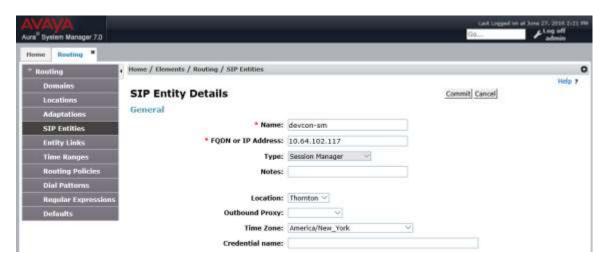
6.1. Launch System Manager

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.



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.

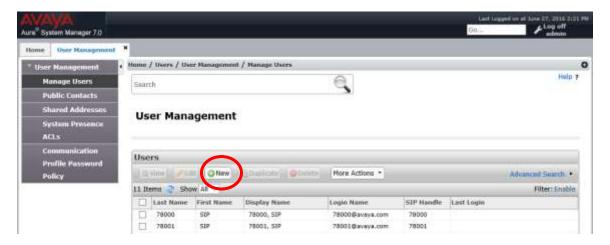


Scroll down to the **Listen Ports** section and verify that the transport network protocol used by Spectralink telephones is specified in the list below. For the compliance test, the Spectralink telephones used TCP network transport as specified in the site.cfg file configured in **Section 7.3**.



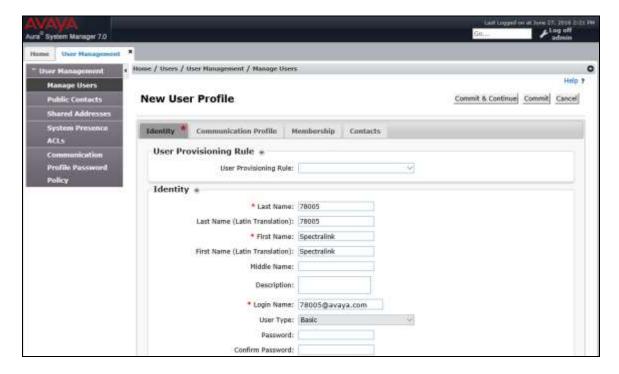
6.3. Administer SIP User

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.



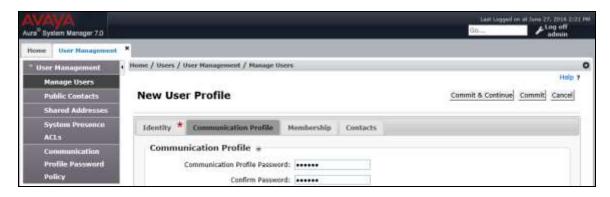
6.3.1. Identity

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.



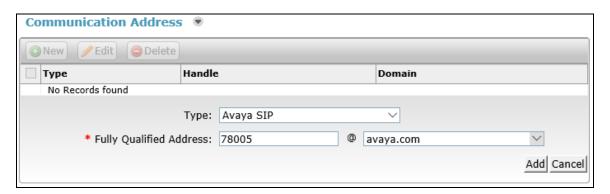
6.3.2. Communication Profile

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



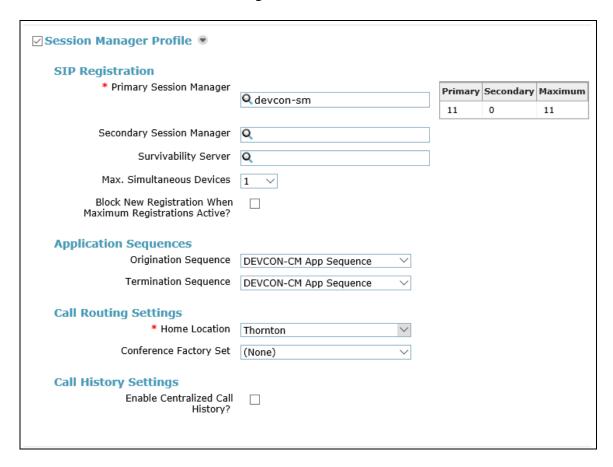
6.3.3. Communication Address

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**.



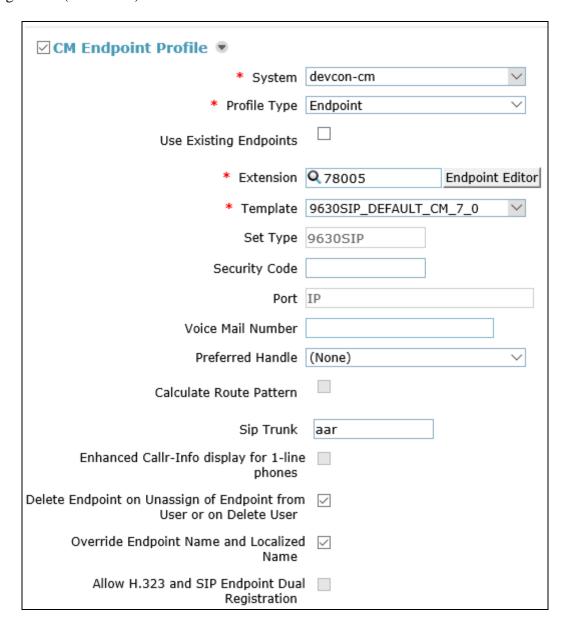
6.3.4. Session Manager Profile

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.

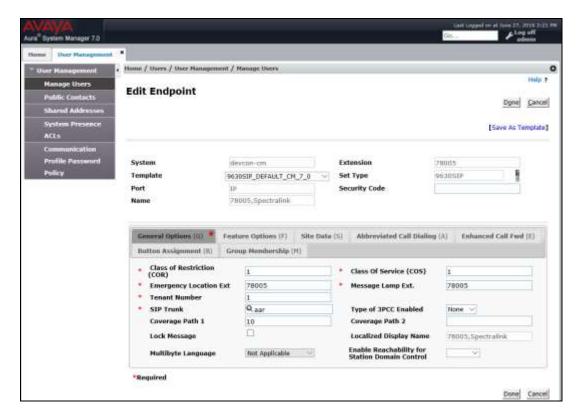


6.3.5. CM Endpoint Profile

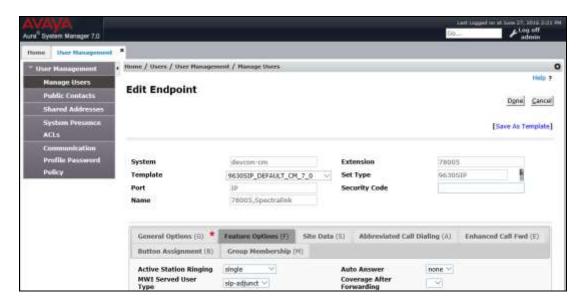
Scroll down to check and expand **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.1**. For **Template**, select *9630SIP_DEFAULT_CM_7_0*. For **Port**, click and select *IP*. Retain the default values in the remaining fields. Click **Commit** to save the configuration (not shown).



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
 reg.1.displayName
 Set to the SIP extension of the handset.
 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"</pre>
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 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 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.

- [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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