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/](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.
4. Equipment and Software Validated

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

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

G3 Version: V17
Location: 2
Platform: 28
Software Package: Enterprise
System ID (SID): 1
Module ID (MID): 1
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 - PBFC: 9600  0
Maximum Off-PBX Telephones - PFVC: 9600  0
Maximum Off-PBX Telephones - SCCAN: 0  0
Maximum Survivable Processors: 313  0

(Optional: 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.

<table>
<thead>
<tr>
<th>Region: 1</th>
<th>Location: 1</th>
<th>Authoritative Domain: avaya.com</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name:</td>
<td>Stub Network Region: n</td>
<td></td>
</tr>
<tr>
<td>MEDIA PARAMETERS</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Codec Set: 1</td>
<td>Intra-region IP-IP Direct Audio: yes</td>
<td></td>
</tr>
<tr>
<td>UDP Port Min: 2048</td>
<td>Inter-region IP-IP Direct Audio: yes</td>
<td></td>
</tr>
<tr>
<td>UDP Port Max: 3329</td>
<td>IP Audio Hairpinning? n</td>
<td></td>
</tr>
<tr>
<td>DIFFSERV/TOS PARAMETERS</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Call Control PHB Value: 46</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Audio PHB Value: 46</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Video PHB Value: 26</td>
<td></td>
<td></td>
</tr>
<tr>
<td>802.1P/Q PARAMETERS</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Call Control 802.1p Priority: 6</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Audio 802.1p Priority: 6</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Video 802.1p Priority: 5</td>
<td>AUDIO RESOURCE RESERVATION PARAMETERS</td>
<td></td>
</tr>
<tr>
<td>H.323 IP ENDPOINTS</td>
<td>RSVP Enabled? n</td>
<td></td>
</tr>
<tr>
<td>H.323 Link Bounce Recovery? y</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Idle Traffic Interval (sec): 20</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Keep-Alive Interval (sec): 5</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Keep-Alive Count: 5</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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.

<table>
<thead>
<tr>
<th>Audio Codec</th>
<th>Silence Suppression n</th>
<th>Frames Per Pkt 2</th>
<th>Packet Size (ms) 20</th>
</tr>
</thead>
<tbody>
<tr>
<td>1: G.711MU</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>2:</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>3:</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>4:</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>5:</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>6:</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>7:</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
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.

![Communication Profile Tab](image)

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.

![Communication Address Section](image)
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).

<table>
<thead>
<tr>
<th>Selected Item</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>System</td>
<td>devcon-cm</td>
</tr>
<tr>
<td>Profile Type</td>
<td>Endpoint</td>
</tr>
<tr>
<td>Use Existing Endpoints</td>
<td></td>
</tr>
<tr>
<td>Extension</td>
<td>78005</td>
</tr>
<tr>
<td>Template</td>
<td>9630SIP_DEFAULT_CM_7_0</td>
</tr>
<tr>
<td>Set Type</td>
<td>9630SIP</td>
</tr>
<tr>
<td>Security Code</td>
<td></td>
</tr>
<tr>
<td>Port</td>
<td>IP</td>
</tr>
<tr>
<td>Voice Mail Number</td>
<td></td>
</tr>
<tr>
<td>Preferred Handle</td>
<td>(None)</td>
</tr>
<tr>
<td>Calculate Route Pattern</td>
<td></td>
</tr>
<tr>
<td>Sip Trunk</td>
<td>aar</td>
</tr>
<tr>
<td>Enhanced Caller-Info display for 1-line phones</td>
<td></td>
</tr>
<tr>
<td>Delete Endpoint on Unassign of Endpoint from User or on Delete User</td>
<td>Yes</td>
</tr>
<tr>
<td>Override Endpoint Name and Localized Name</td>
<td></td>
</tr>
<tr>
<td>Allow H.323 and SIP Endpoint Dual Registration</td>
<td></td>
</tr>
</tbody>
</table>
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.

![Edit Endpoint](image1.png)

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.

![Edit Endpoint](image2.png)
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"
  </SIPserver>
  <dialplan>
    dialplan.impossibleMatchHandling="2"
    dialplan.digitmap=""
  </dialplan>
  <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"
      />
    </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.


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