



Application Notes for Spectralink Versity Enterprise Wi-Fi Smartphones with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Spectralink Versity Enterprise Wi-Fi Smartphones with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. The Spectralink Versity Enterprise Wi-Fi Smartphones registered with Avaya Aura® Session Manager via SIP, including the following Spectralink Versity series handsets: Versity 9240, Versity 9540, and Versity 9640. The Spectralink Versity Enterprise Wi-Fi Smartphones communicate with Avaya SIP network over an 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 Versity Enterprise Wi-Fi Smartphones with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. The Spectralink Versity Enterprise Wi-Fi Smartphones registered with Avaya Aura® Session Manager via SIP using UDP transport, including the following Spectralink Versity series handsets: Versity 9240, Versity 9540, and Versity 9640. The Spectralink Versity Enterprise Wi-Fi Smartphones communicate with Avaya SIP network over an 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 Versity, 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 Versity came back into service after re-connecting the access point and rebooting the phones.

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 this DevConnect Application Note 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 this Application Note, the interface between Avaya systems and Spectralink Versity 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 Versity with Session Manager
- Calls between Spectralink Versity and Avaya SIP/H.323 deskphones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between Spectralink Versity and the PSTN.
- UDP transport protocol.
- Support of G.711, G.729, and G.722 codecs.
- Proper recognition of DTMF tones.
- Basic telephony features, including hold, mute, redial, multiple calls, blind/attended transfer, attended conference, and long duration calls.
- Extended telephony features using Communication Manager FACs for Call Forward, Follow Me, Call Park/Unpark, and Call Pickup.
- Voicemail coverage, MWI support, and logging into voicemail system to retrieve voice messages.
- Proper system recovery after a restart of the Spectralink Versity and loss of IP network connectivity.

2.2. Test Results

All test cases passed.

2.3. Support

For technical support and information on Spectralink Versity Enterprise Wi-Fi Smartphones, contact Spectralink technical support at:

- Phone: 1-800-775-5330
- Website: <https://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 with an Avaya G450 Media Gateway.
- Media resources in Avaya G450 Media Gateway and Avaya Aura® Media Server.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP deskphones.
- Session Manager connected to the PSTN via Avaya Session Border Controller for Enterprise (SBCE).
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya 96x1 Series H.323 / SIP Deskphones and Avaya J100 Series SIP Deskphones.
- Spectralink Versity Enterprise Wi-Fi Smartphones, included the Versity 9240, Versity 9540, and Versity 9640.

Spectralink Versity Enterprise Wi-Fi Smartphones registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.

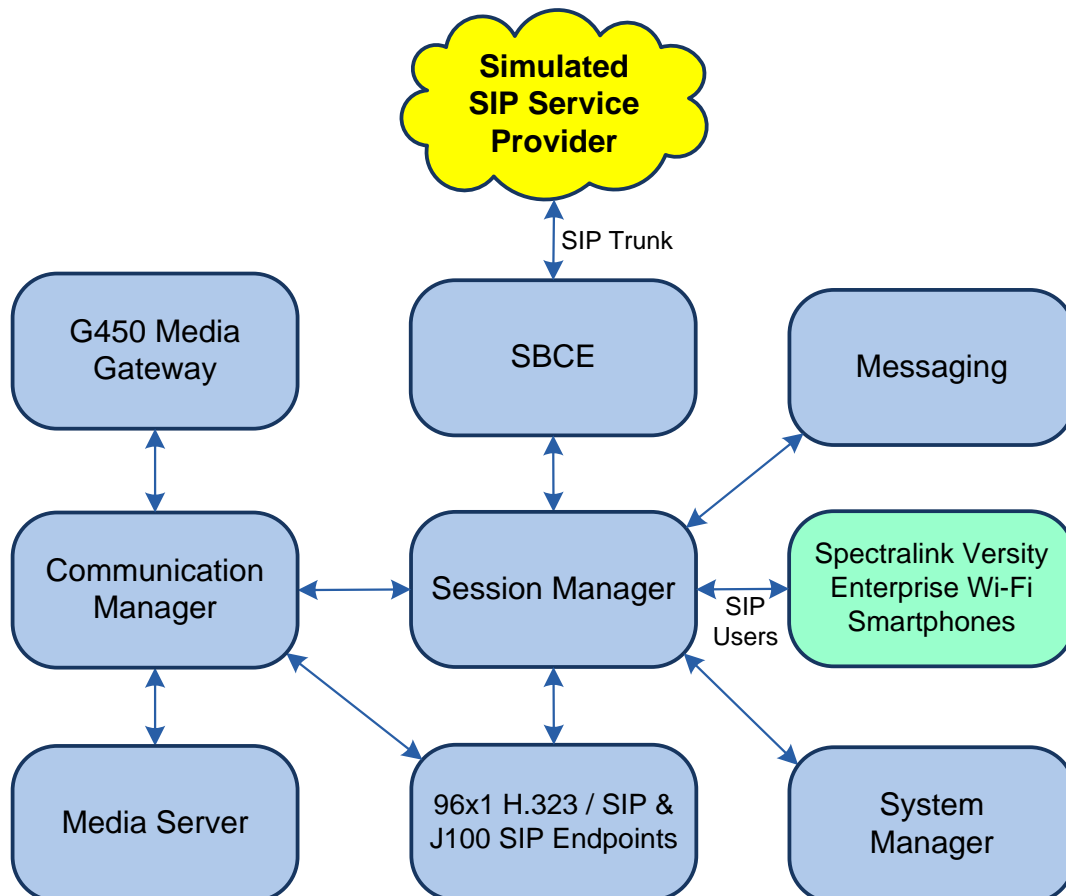


Figure 1: Avaya SIP Network with Spectralink Versity Enterprise Wi-Fi Smartphones

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	8.1.3.0.1-FP3P1
Avaya G450 Media Gateway	FW 41.24.0
Avaya Aura® Media Server	v.8.0.2.138
Avaya Aura® Session Manager	8.1.3.0 Build No. – 8.1.0.0.733078 Software Update Revision No: 8.1.3.0.1012091 Feature Pack 3
Avaya Aura® System Manager	8.1.3.0.813014
Avaya Messaging	10.8.2 SP1
Avaya 96x1 Series IP Deskphone	6.8502 (H.323) 7.1.1.0.9 (SIP)
Avaya J100 Series IP Deskphone	4.0.7.1.5 (SIP)
Spectralink Versity Enterprise Wi-Fi Smartphones	13.1.27141 (BizPhone Application) 2.1.0.894 (Versity 95, 96 Series – Android 10) 1.2.0.611 (Versity 92 Series – Android 10)

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 appropriate credentials.

Note: It is assumed that basic configuration, such as voicemail coverage, has already been configured. The SIP station configuration for Spectralink Versity is configured through System Manager in **Section 6.2**.

5.1. Verify Communication Manager 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: V18                                     Software Package: Enterprise
Location: 2                                         System ID (SID): 1
Platform: 28                                       Module ID (MID): 1

                                                USED
Platform Maximum Ports: 48000                      106
Maximum Stations: 36000                            35
Maximum XMOBILE Stations: 36000                   0
Maximum Off-PBX Telephones - EC500: 41000         0
Maximum Off-PBX Telephones - OPS: 41000          21
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 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: 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 Spectralink Versity. 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. Spectralink Versity was tested using G.711, G.722 and G.729 codecs. Specify the desired codecs in the **IP Codec Set** form as per customer requirements.

Media encryption was enabled for Avaya IP Deskphones. Spectralink Versity wasn't configured to support SRTP, so the *none* option 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 SRTP: best-effort
1: 1-srtp-aescm128-hmac80
2: none
3:
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*.

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? 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
Incoming Dialog Loopbacks: eliminate                Bypass If IP Threshold Exceeded? n
                                     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? n
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 22
                                     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: tie                                     Auth Code? n
Member Assignment Method: auto
Signalng 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
Intw
1: 10      0
2:
3:
4:
5:
6:
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

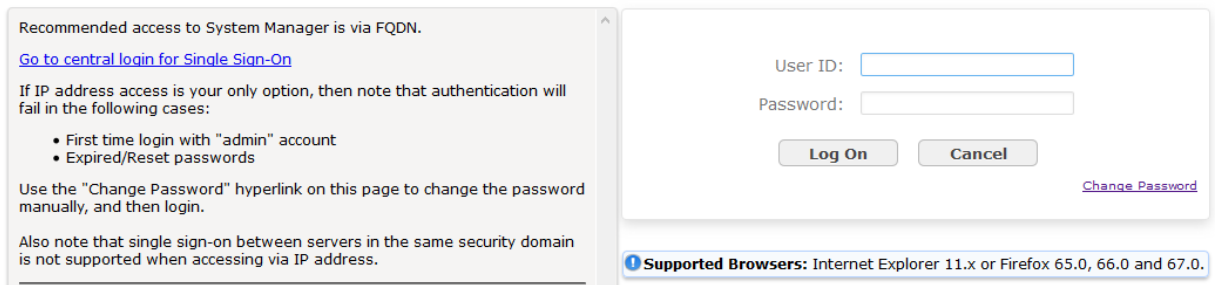
This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Set Network Transport Protocol for Spectralink Versity Enterprise Wi-Fi Smartphones
- 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 Spectralink Versity Enterprise Wi-Fi Smartphones.

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.



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 11.x or Firefox 65.0, 66.0 and 67.0.

6.2. Set Network Transport Protocol for Spectralink Versity Enterprise Wi-Fi Smartphones

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

AVAYA Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ Widgets ▾ Shortcuts ▾ Search 🔍 admin

Home Routing

Routing

- Domains
- Locations
- Conditions
- Adaptations ▾
- SIP Entities**
- Entity Links
- Time Ranges
- Routing Policies
- Dial Patterns ▾
- Regular Expressions
- Defaults

SIP Entity Details

Commit Cancel Help ?

General

* Name: devcon-sm

* IP Address: 10.64.102.117

SIP FQDN:

Type: Session Manager ▾

Notes:

Location: Thornton ▾

Outbound Proxy: ▾

Time Zone: America/New_York ▾

Minimum TLS Version: Use Global Setting ▾

Credential name:

Monitoring

SIP Link Monitoring: Use Session Manager Configuration ▾

CRLF Keep Alive Monitoring: Use Session Manager Configuration ▾

Scroll down to the **Listen Ports** section and verify that the transport network protocol used by Spectralink Versity is specified in the list below. For the compliance test, the solution used UDP network transport.

Listen Ports

Add Remove

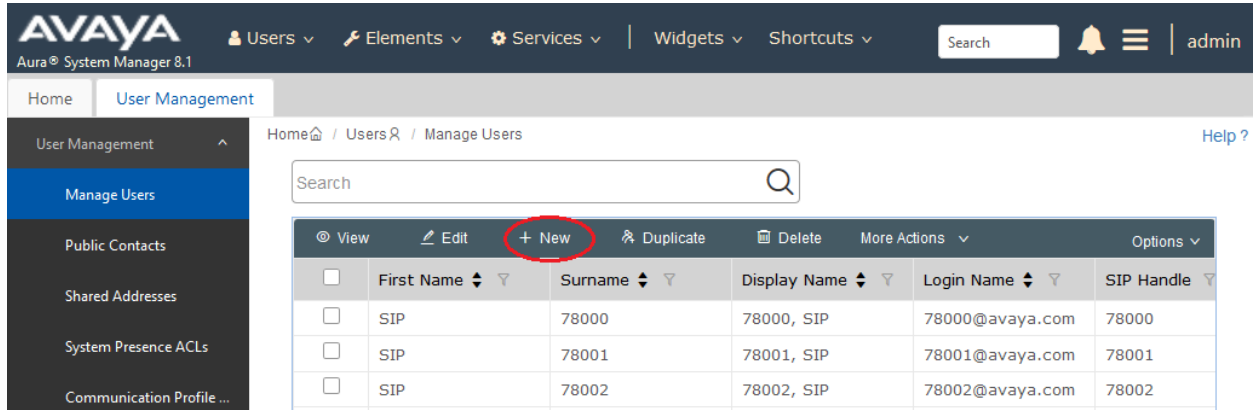
3 Items Filter: Enable

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

Select : All, None

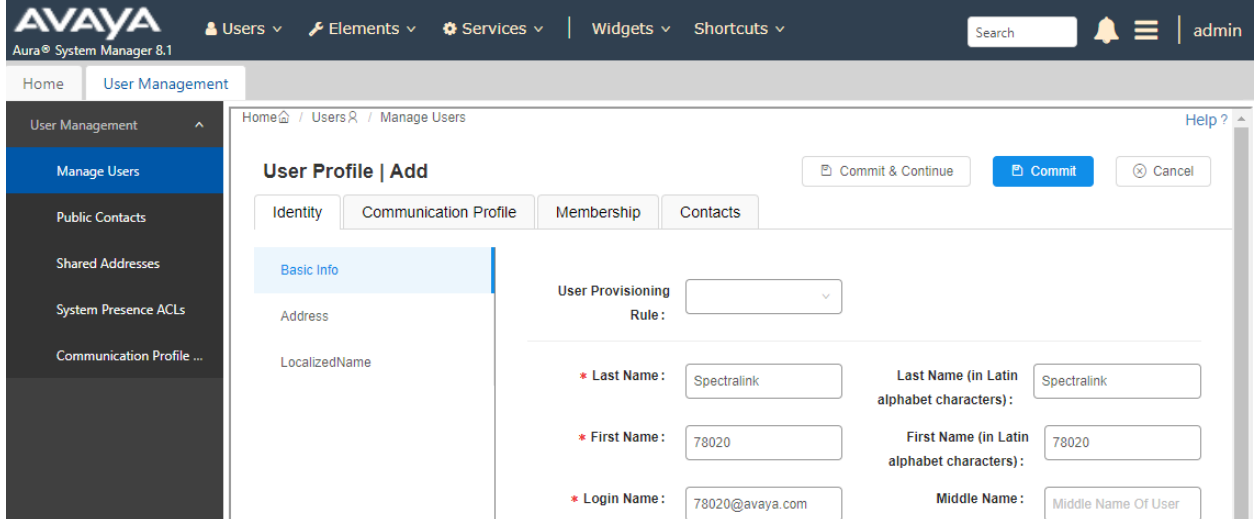
6.3. Administer SIP User

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.



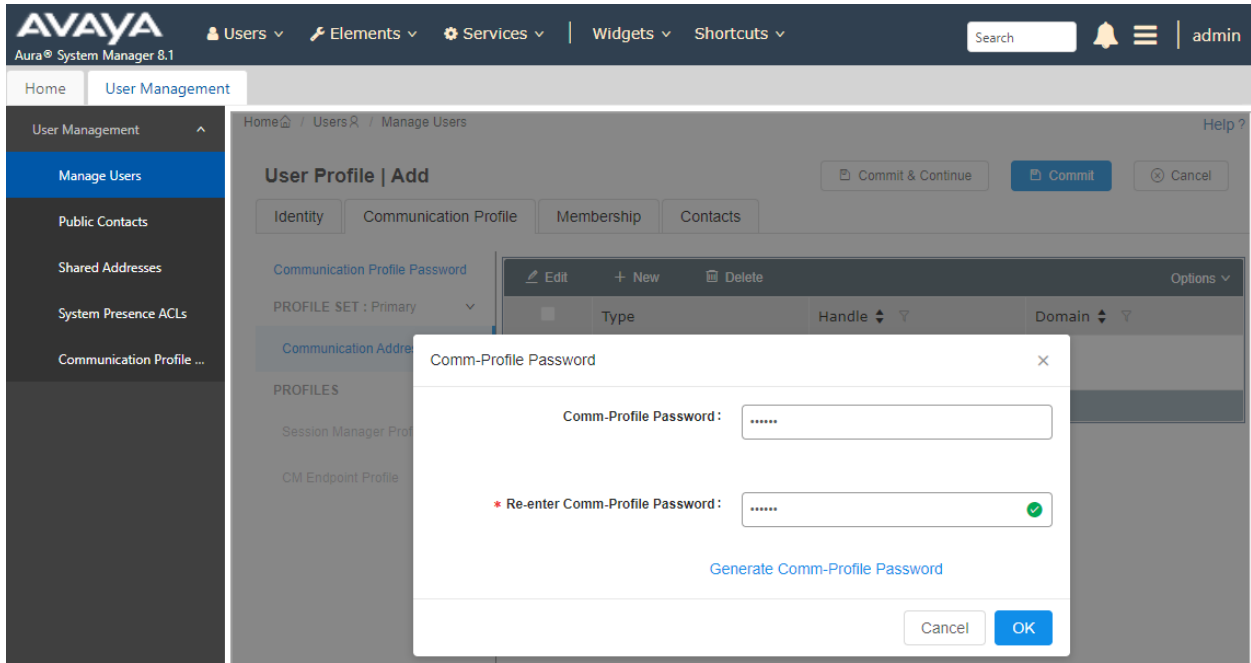
6.3.1. Identity

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 Versity SIP extension and “<domain>” is the applicable SIP domain name from **Section 5.3**. Retain the default values in the remaining fields.



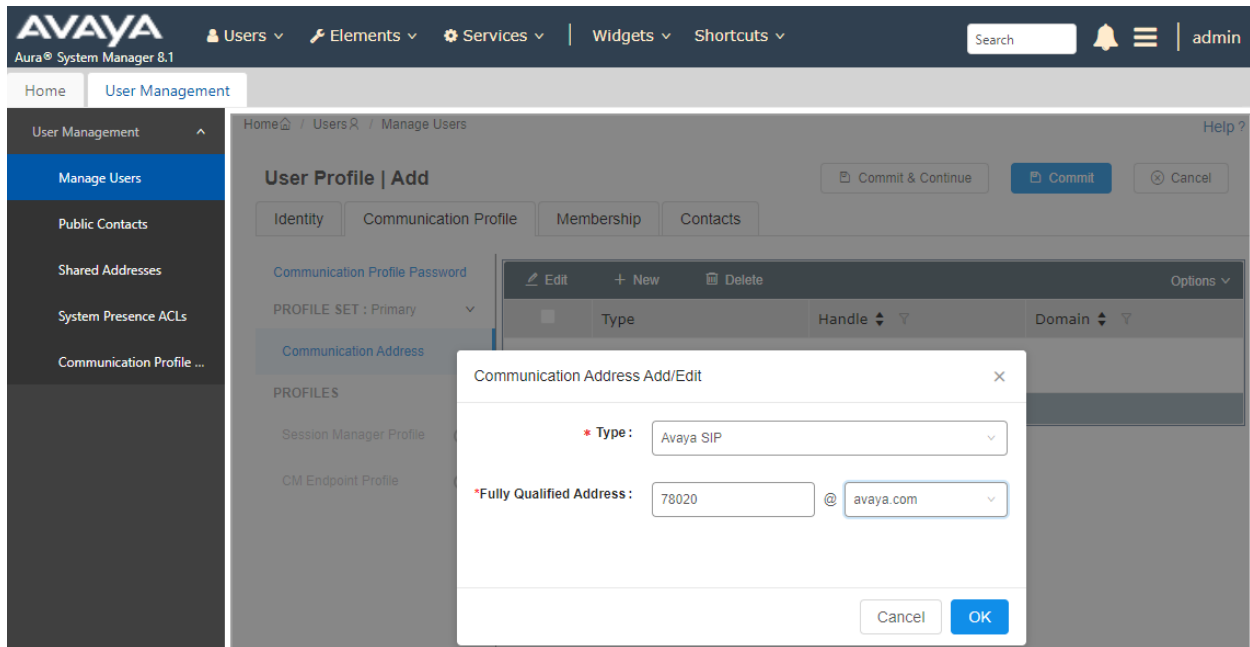
6.3.2. Communication Profile

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.

The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes the Avaya logo, 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts' menus, along with a search bar and a user profile icon labeled 'admin'. The main content area is titled 'User Profile | Add' and features tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' tab is active, showing a 'Communication Profile Password' field and a 'PROFILE SET' dropdown set to 'Primary'. Below this, there are 'PROFILES' with a toggle for 'Session Manager Profile' (which is turned on) and 'CM Endpoint Profile' (which is turned off). The 'SIP Registration' section contains several fields: 'Primary Session Manager' (set to 'devcon-sm'), 'Secondary Session Manager' (placeholder 'Start typing...'), 'Survivability Server' (placeholder 'Start typing...'), and 'Max. Simultaneous Devices' (set to 'Select'). There is also a 'Block New Registration When Maximum Registrations Active?' checkbox. The 'Application Sequences' section includes 'Origination Sequence' and 'Termination Sequence', both set to 'DEVCON-CM App S...'. At the top right of the form, there are buttons for 'Commit & Continue', 'Commit', and 'Cancel'.

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

The screenshot shows the 'Call Routing Settings' section of the Avaya Aura System Manager 8.1 interface. It includes a 'Home Location' field set to 'Thornton' and a 'Conference Factory Set' dropdown menu set to 'Select'.

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**.

The screenshot displays the Avaya Aura System Manager 8.1 interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 8.1', and menu items for Users, Elements, Services, Widgets, and Shortcuts. A search bar and user profile (admin) are also visible. The main content area is titled 'User Profile | Add' and features tabs for Identity, Communication Profile, Membership, and Contacts. The 'Communication Profile' tab is active, showing various configuration fields. On the left, a sidebar lists navigation options, with 'CM Endpoint Profile' selected and its toggle switch turned on. The configuration fields include:

- System:** devcon-cm
- Profile Type:** Endpoint
- Extension:** 78020
- Set Type:** 9641SIP
- Port:** IP
- Template:** 9641SIP_DEFAULT_CM_8_1
- Security Code:** Enter Security Code
- Preferred Handle:** Select
- Voice Mail Number:** (empty field)
- Sip Trunk:** aar
- Use Existing Endpoints:**
- Calculate Route Pattern:**
- SIP URI:** Select
- Delete on Unassign from User or on Delete User:**
- Override Endpoint Name and Localized Name:**
- Allow H.323 and SIP Endpoint Dual Registration:**

Buttons for 'Commit & Continue', 'Commit', and 'Cancel' are located at the top right of the form.

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).

New Endpoint
Help ?

Done

[Save As Template]

* **System**

* **Template**

* **Port Name**

Display Extension Ranges

* **Extension**

Set Type

Security Code

General Options (G) *
Feature Options (F)
Site Data (S)
Abbreviated Call Dialing (A)
Enhanced Call Fwd (E)

Button Assignment (B)
Profile Settings (P)
Group Membership (M)

<p>* Class of Restriction (COR) <input type="text" value="1"/></p> <p>* Emergency Location Ext <input type="text" value="78020"/></p> <p>* Tenant Number <input type="text" value="1"/></p> <p>* SIP Trunk <input type="text" value="Qaar"/></p> <p>Coverage Path 1 <input type="text" value="15"/></p> <p>Lock Message <input type="checkbox"/></p> <p>Multibyte Language <input type="text" value="Not Applicable"/></p> <p>SIP URI <input type="text"/></p> <p>Attendant <input type="checkbox"/></p> <p style="color: green; font-weight: bold; margin-top: 5px;">Primary Session Manager</p> <p>IPv4: <input type="text"/></p>	<p>* Class Of Service (COS) <input type="text" value="1"/></p> <p>* Message Lamp Ext. <input type="text" value="78020"/></p> <p>Type of 3PCC Enabled <input type="text" value="None"/></p> <p>Coverage Path 2 <input type="text"/></p> <p>Localized Display Name <input type="text"/></p> <p>Enable Reachability for Station Domain Control <input type="text" value="system"/></p> <p>IPv6: <input type="text"/></p>
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7. Configure Spectralink Versity Enterprise Wi-Fi Smartphones

This section covers the SIP configuration of the Spectralink Versity Enterprise Wi-Fi Smartphones. Refer to [4] for more information on configuring Spectralink Versity. The configuration was performed via the **Biz Phone Settings** menu on the smartphone. The procedure covers the following areas:

- Configure DHCP Server
- Configure SIP Phone Settings

7.1. Configure DHCP Server

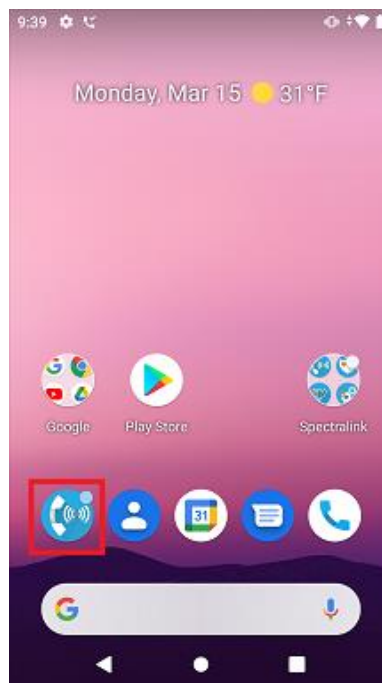
Spectralink Versity must first acquire several IP network settings before proceeding with provisioning. These settings were automatically obtained from a DHCP server. Alternatively, Spectralink Versity could be configured with static IP addresses, but for the compliance test, a DHCP server was used.

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

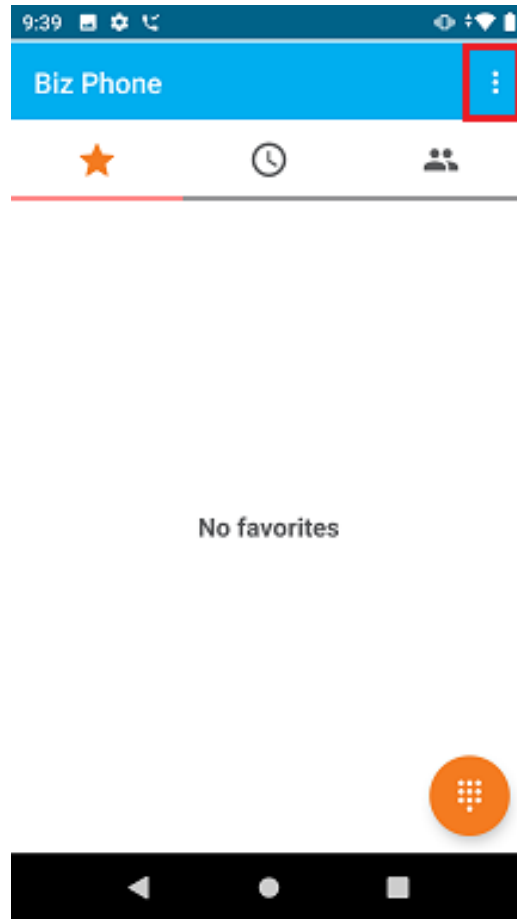
- Option 3: Default Gateway
- Option 6: DNS Server (optional)

7.2. Configure SIP Phone Settings

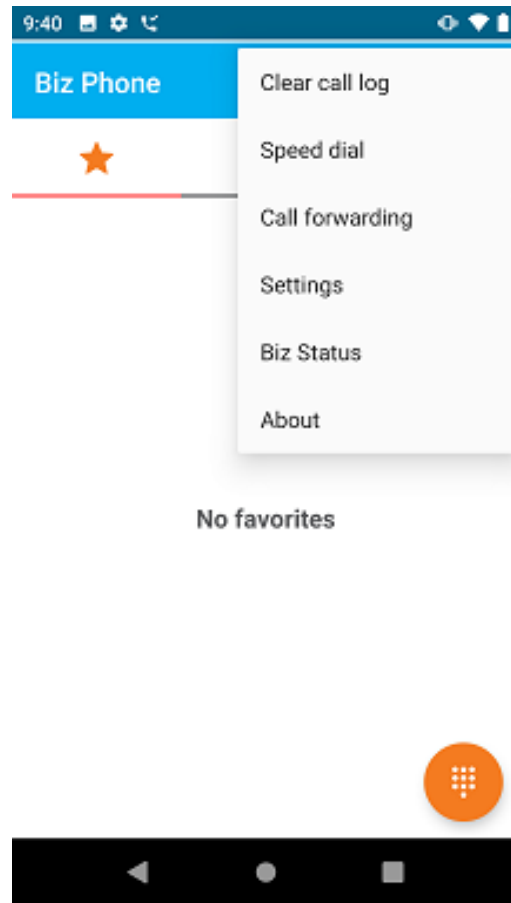
Click on the **Biz Phone** app icon on the smartphone as shown below.



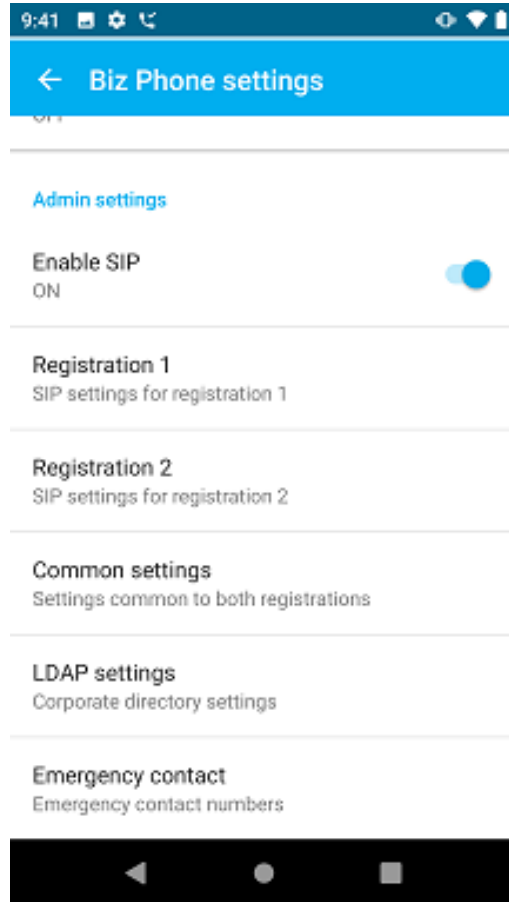
In the **Biz Phone** screen shown below, click on the overflow menu (i.e., 3 dots in upper right-hand corner).



From the menu, select **Settings** to access the Biz Phone settings.

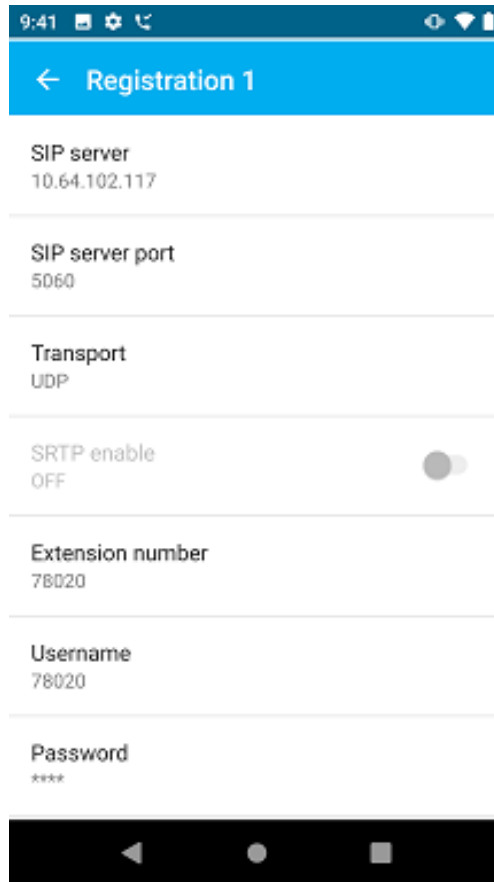


Under the **Admin settings** section, turn on the **Enable SIP** option as shown below and select the **Registration 1** option to display the SIP settings.



In the **Registration 1** screen, configure the following parameters:

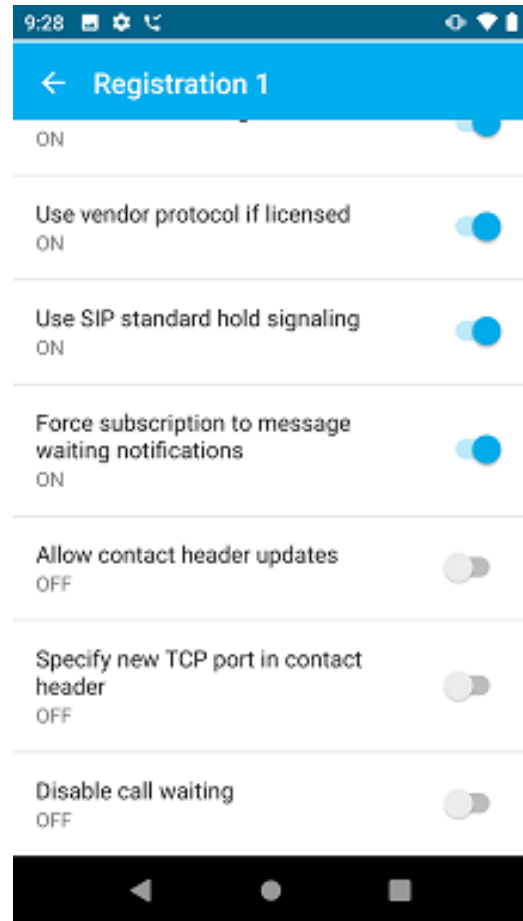
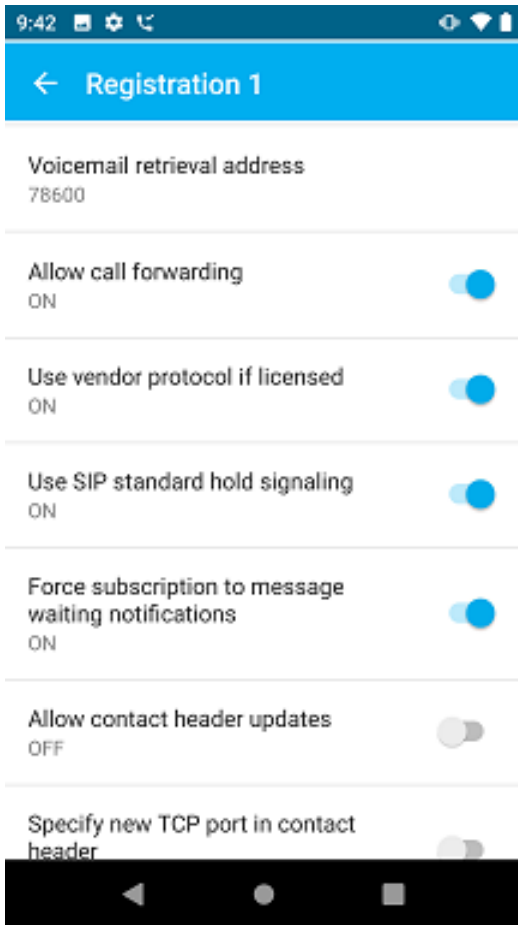
- **SIP server:** Set to the Session Manager IP address (e.g., *10.64.102.117*).
- **SIP server port:** Set to appropriate SIP port (e.g., *5060*).
- **Transport:** Set to *UDP* transport protocol.
- **SRTP enable:** Disable this option.
- **Extension number:** Set to the SIP extension (e.g., *78020*).
- **Username:** Set to the SIP extension (e.g., *78020*).
- **Password:** Set to the SIP password specified as the **Comm-Profile Password** in **Section 6.3.2**.



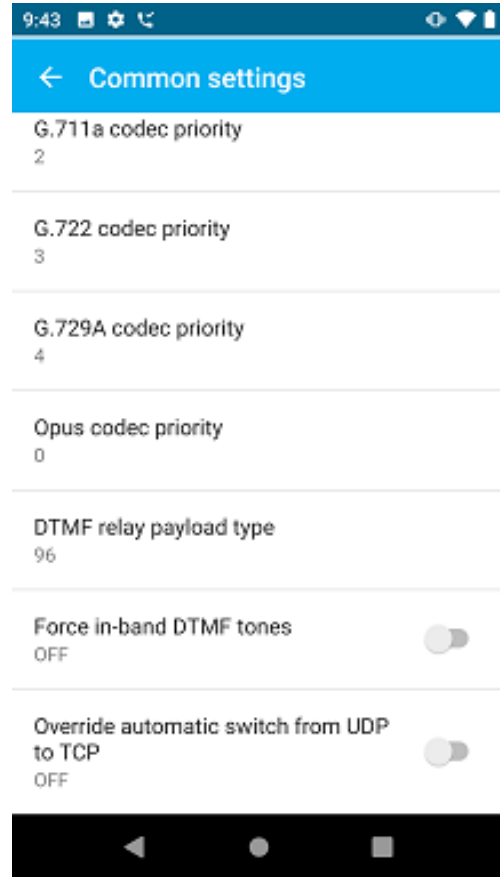
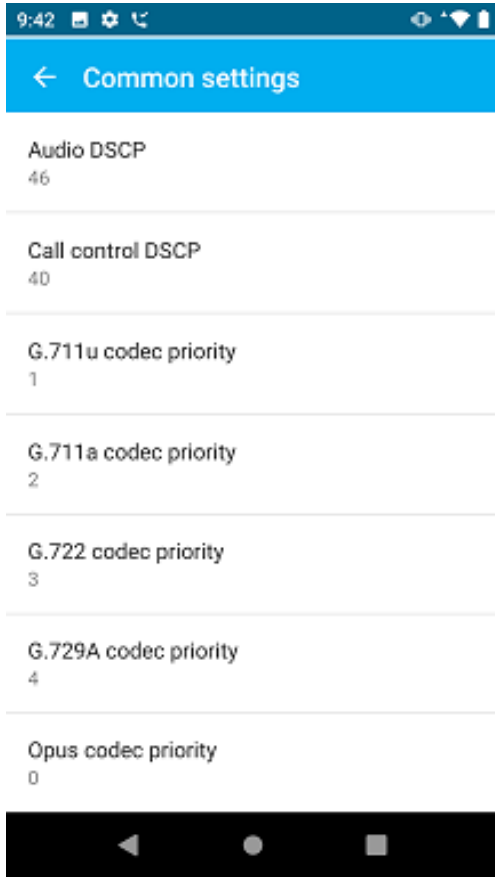
Scroll down to the bottom half of the **Registration 1** screen and configure the following parameters:

- **Voicemail retrieval address:** Set to the voicemail pilot number (e.g., 78500).
- **Force subscription to message Waiting notifications:** Enable this option.
- **Disable call waiting:** Set to *OFF*.

Accept the default values for the remaining parameters.



In the main screen of the **Biz Phone** application, select **Common** settings (not shown) to prioritize the codecs as needed.



8. Verification Steps

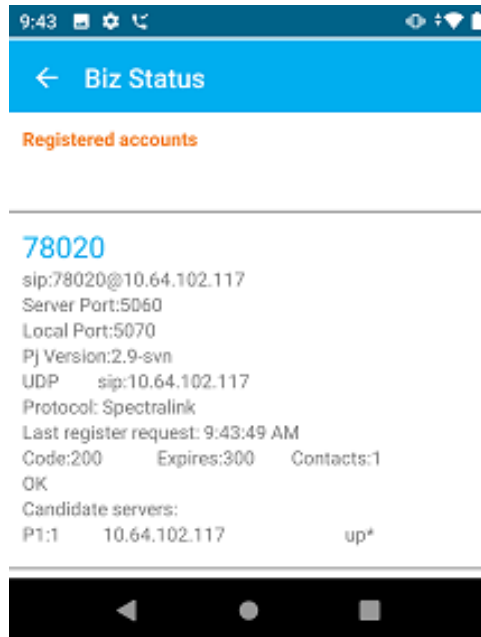
This section provides the tests that can be performed to verify proper configuration of Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and Spectralink Versity Enterprise Wi-Fi Smartphones.

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

The screenshot shows the Avaya System Manager 8.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The main content area is titled 'User Registrations' and contains a table of 21 items. The table has the following columns: Details, Address, First Name, Last Name, Actual Location, IP Address, Remote Office, Shared Control, Simult. Devices, AST Device, and Registered (Prim, Sec, Surv, Visiting). The table shows several entries, including SIP and Spectralink devices, with their respective IP addresses and registration status.

Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered			
										Prim	Sec	Surv	Visiting
▶ Show	---	SIP	78003	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
▶ Show	78000@avaya.com	SIP	78000	---	192.168.100.54	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
▶ Show	78021@avaya.com	Spectralink	78021	---	192.168.100.192	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
▶ Show	78030@avaya.com	Agent	78030	---	192.168.100.49	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
▶ Show	78022@avaya.com	Spectralink	78022	---	192.168.100.193	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
▶ Show	78020@avaya.com	Spectralink	78020	---	192.168.100.191	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

- Alternatively, the registration status can also be checked on Spectralink Versity by opening the **Biz Status** application. Note that the server status on the last line indicates an *up** status.



- Establish a call between Spectralink Versity 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

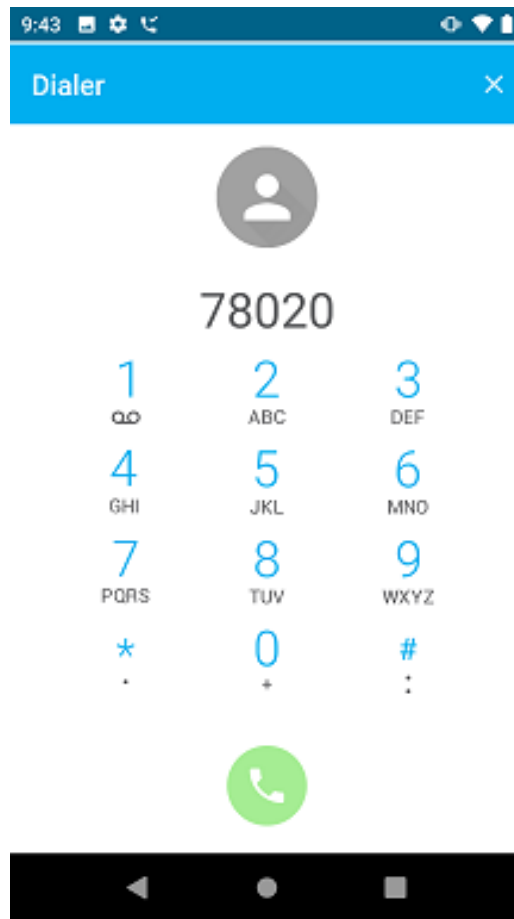
Near-end Signaling Loc: PROCR
  Signaling IP Address Port
  Near-end: 10.64.102.115 : 5061
  Far-end: 10.64.102.117 : 5061
H.245 Near:
H.245 Far:
  H.245 Signaling Loc: H.245 Tunneled in Q.931? no

Audio Connection Type: ip-direct Authentication Type: None
  Near-end Audio Loc: Codec Type: G.711MU
  Audio IP Address Port
  Near-end: 192.168.100.55 : 15964
  Far-end: 192.168.100.191 : 43205

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

```

4. From the Spectralink Versity Dialer shown below, place a call to an Avaya IP Deskphone or another Spectralink Versity smartphone. While the call is active, basic telephony features can be exercised to verify proper operation.



9. Conclusion

These Application Notes described the configuration steps required to integrate Spectralink Versity Enterprise Wi-Fi Smartphones with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. Spectralink Versity Enterprise Wi-Fi Smartphones were able to establish calls with H.323 / SIP deskphones and the PSTN. In addition, basic telephony features were verified. All feature and serviceability test cases were completed successfully with observations noted in **Section 2.2**.

10. References

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

- [1] *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 8, November 2020, available at <http://support.avaya.com>.
- [2] *Administering Avaya Aura® System Manager for Release 8.1.x*, Release 8.1.x, Issue 8, November 2020, available at <http://support.avaya.com>.
- [3] *Administering Avaya Aura® Session Manager*, Release 8.1.x, Issue 7, October 2020, available at <http://support.avaya.com>.
- [4] *Spectralink Versity Smartphone User Guide*, Release 1.8, 720-0060-000 version H, May 2020, available at <https://support.spectralink.com/versity>.

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