

Avaya Solution & Interoperability Test Lab

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 18.0 with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1. Spectralink Versity Enterprise Wi-Fi Smartphones registered with Avaya Aura® Session Manager via SIP. The following Spectralink Versity series handsets were covered in the compliance test: Versity 9240, Versity 9540, and Versity 9640. Spectralink Versity Enterprise Wi-Fi Smartphones communicated with the Avaya SIP network over a wireless network access point.

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 and Avaya Aura® Session Manager. Spectralink Versity Enterprise Wi-Fi Smartphones registered with Avaya Aura® Session Manager via SIP. The following Spectralink Versity series handsets were covered in the compliance test: Versity 9240, Versity 9540, and Versity 9640. Spectralink Versity Enterprise Wi-Fi Smartphones communicated with the Avaya SIP network over a wireless network access point.

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 these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and Spectralink 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.
- TCP 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 on Spectralink Versity Enterprise Wi-Fi Smartphones, contact Spectralink Technical Support at:

- Phone: 1-800-775-5330
- Website: <u>https://support.spectralink.com/</u>
- Email: <u>technicalsupport@spectralink.com</u>

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 G430/G450 Media Gateway.
- Media resources in Avaya G430/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

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4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	10.1.0.1.0-SP1
Avaya G430 Media Gateway	FW 42.8.0
Avaya G450 Media Gateway	FW 42.7.0
Avaya Aura® Media Server	v.10.1.0.77
Avaya Aura® System Manager	10.1.0.1
	Build No. – 10.1.0.0.537353
	Software Update Revision No:
	10.1.0.1.0614394
	Service Pack 1
Avaya Aura® Session Manager	10.1.0.1.1010105
Avaya Session Border Controller for Enterprise	10.1.1.0-35-21872
Avaya Messaging	11.0.0.3204
Avaya 96x1 Series IP Deskphones	6.8.5.3.2 (H.323)
	7.1.13.0.4 (SIP)
Avaya J100 Series IP Phones	4.0.13.0.6
Spectrolink Versity Entermise Wi Ei	18.0.44091 (BizPhone Application)
Spectralink versity Enterprise w1-F1	2.5.2.1733 (Versity 96 Series – Android 10)
Smarphones	1.5.2.1385 (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: V20
                                              Software Package: Enterprise
      Location: 2
                                               System ID (SID): 1
      Platform: 28
                                               Module ID (MID): 1
                                                          USED
                              Platform Maximum Ports: 48000 106
                                  Maximum Stations: 36000
                                                             36
                           Maximum XMOBILE Stations: 36000
                                                              0
                  Maximum Off-PBX Telephones - EC500: 41000
                                                              0
                  Maximum Off-PBX Telephones - OPS: 41000
                                                             22
                  Maximum Off-PBX Telephones - PBFMC: 41000
                                                               0
                  Maximum Off-PBX Telephones - PVFMC: 41000
                                                               0
                  Maximum Off-PBX Telephones - SCCAN: 0
                                                               0
                       Maximum Survivable Processors: 313
                                                               0
       (NOTE: You must logoff & login to effect the permission changes.)
```

5.2. Administer IP Node Names

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

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

Spectralink Versity was not configured to support SRTP, so *none* was also included under **Media Encryption**.

```
2
change ip-codec-set 1
                                                                    1 of
                                                              Page
                        IP MEDIA PARAMETERS
   Codec Set: 1
Audio
Codec
1: G.711MU
              Silence Frames
                                     Packet
              Suppression Per Pkt Size(ms)
                n 2
                                       20
2:
3:
4:
5:
6:
7:
    Media Encryption
                                      Encrypted SRTCP: best-effort
1: 1-srtp-aescm128-hmac80
2: 2-srtp-aescm128-hmac32
3: none
4:
5:
```

5.4. Administer SIP Trunk to Session Manager

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tls*.
- Set the **Enforce SIPS URI for SRTP** field to *n*.
- Specify Communication Manager (*procr*) and the Session Manager as the two ends of the signaling group in the Near-end Node Name field and the Far-end Node Name field, respectively. These field values are taken from the IP Node Names form.
- Ensure that the TLS port value of 5061 is configured in the Near-end Listen Port and the Far-end Listen Port fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.
- Initial IP-IP Direct Media was enabled.

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

add signaling-group 10	Page 1 of 2
SIGNALING	GROUP
Group Number: 10 Group Type:	sip
IMS Enabled? n Transport Method:	tls
Q-SIP? n	
IP Video? y Priority Video?	n Enforce SIPS URI for SRTP? n
Peer Detection Enabled? y Peer Server:	SM Clustered? n
Prepend '+' to Outgoing Calling/Alerting,	Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Al	lerting/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
Fa	ar-end Network Region: 1
Far-end Domain: avaya.com	
	Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate	RFC 3389 Comfort Noise? n
DTMF over IP: rtp-payload	Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3	IP Audio Hairpinning? n
Enable Layer 3 Test? y	Initial IP-IP Direct Media? y

Configure the **Trunk Group** form as shown below. This trunk group is used for SIP calls to Spectralink Versity, Avaya SIP deskphones, and Avaya Messaging. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie* or *public-ntwrk*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure default values for the remaining fields.

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

      TRUNK GROUP
      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
      Night Service:

      Queue Length: 0
      Auth Code? n
      Member Assignment Method: auto

      Signaling Group: 10
      Number of Members: 10
```

5.5. Administer AAR Call Routing

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

change aar analysis 78						Page 1 of 2	
	AA	R DI	GIT ANALYS	SIS TABI	ΞE		
			Location:	all		Percent Full: 1	
Dialed	Tota	1	Route	Call	Node	ANI	
String	Min	Max	Pattern	Туре	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.

chai	nge route-pa	ttern 10		Page	1 of 3
		Pattern Nu	umber: 10 Pattern Name: To	devcon-sm	
	SCCAN? n	Secure SIP? n	Used for SIP stations? n		
	Grp FRL NPA	Pfx Hop Toll 1	No. Inserted		DCS/ IXC
	No	Mrk Lmt List I	Del Digits		QSIG
		I	Ogts		Intw
1:	10 0				n user
2:					n user
3:					n user
4:					n user
5:					n user
6:					n user
	BCC VALUE	TSC CA-TSC	ITC BCIE Service/Feature PARM	I Sub Numbe	ring LAR
	012M4W	Request		Dgts Forma	t
1:	yyyyyn	n	rest	- unk-u	nk none
2:	yyyyyn	n	rest		none

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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	
or to central login for bingre bigh on	User ID:
If IP address access is your only option, then note that authentication will fail in the following cases:	Password:
 First time login with "admin" account Expired/Reset passwords 	Log On Cancel
Use the "Change Password" hyperlink on this page to change the password manually, and then login.	Change Passw
Also note that single sign-on between servers in the same security domain is	
not supported when accessing via IP address.	• Supported Browsers: Firefox (minimum version 93.0), Chrome (minimum version 91.0) or Edge (minimum version 93.0).

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

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

Avra® System	Manager 10.1	Users 🗸 🎤 Elements 🗸 🔅 Services	✓ │ Widgets ✓ Shortcuts ✓	Search	admin
Home	Routing				
Routing	^	SIP Entity Details		Commit Cancel	Help ? 🔺
Domain	ıs	General			
Locatio	ns	* Name:	devcon-sm]	
		* IP Address:	10.64.102.117]	
Conditi	ons	SIP FQDN:]	
Adapta	tions ×	Type:	Session Manager 🗸 🗸		
SIP Enti	ities	Notes:]	- 1
Entity Li	inks	Location:	Thornton 🗸		
		Outbound Proxy:	~		
Time Ra	anges	Time Zone:	America/New_York 🗸		
Routing	9 Policies	Minimum TLS Version:	Use Global Setting 🗸		
Dial Dat	Home V	Credential name:			
	uems .	Monitoring			
Regular	Expressions	SIP Link Monitoring:	Use Session Manager Configuration \checkmark		
Default	s	CRLF Keep Alive Monitoring:	Use Session Manager Configuration \checkmark		

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 TCP network transport.

Listen Ports

Add	Remove					
3 Ite	ms I 🍣					Filter: Enable
	Listen Ports	Protocol	Default Domain		Endpoint	Notes
	5060	TCP 🗸	avaya.com	~		
	5060	UDP 🗸	avaya.com	~		
	5061	TLS 💙	avaya.com	~	✓	
Selec	t : All, None					

6.3. Administer SIP User

In the Home screen (not shown), select Users \rightarrow User Management \rightarrow Manage Users to display the User Management screen below. Click New to add a user.

Avra® System Manager 10.1	Ser:	s v 🌶	é Elements 🗸 🔅 S	ervices	✓ Widgets	s v Shortcuts v	Search	🕽 🗮 admin
Home User Manageme	nt							
User Management 🔹 🔨	Hom	ne☆ / Us	ers 8 / Manage Users					Help?
Manage Users		Search				Q		
Public Contacts		Ø Viev	v <u>/</u> Edit +	New	冬 Duplicate	🔟 Delete 🛛 More Ac	tions V	Options V
Shared Addresses			First Name 🖨 🍸	Surn	ame 🛊 🛛	Display Name 🖨 🍸	Login Name 🖨 🍸	SIP Handle 🛛
			SIP	7800	0	78000, SIP	78000@avaya.com	78000
System Presence ACLs			SIP	7800	1	78001, SIP	78001@avaya.com	78001
Communication Profile			SIP	7800	2	78002, SIP	78002@avaya.com	78002
			SIP	7800	3	78003, SIP	78003@avaya.com	78003

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.

Aura® System Manager 10.1	Users 🗸 🌾 Elements 🗸 🔅 Se	rvices ~ Widgets ~	Shortcuts v	Search	📄 🙏 🗮 admin
Home User Management					
User Management 🔷	Home☆ / UsersՋ / Manage Users				Help?
Manage Users	User Profile Add		E	D Commit & Continue	Commit 🛞 Cancel
Public Contacts	Identity Communication Pr	ofile Membership	Contacts		
Shared Addresses	Basic Info	Heer Dravisioning			
System Presence ACLs	Address	Rule:		~	
Communication Profile	LocalizedName	* Last Name :	On a startint	Last Name (in Latin	
			Spectralink	alphabet characters):	Spectrainik
		* First Name :	78020	First Name (in Latin	78020
		* Login Name :	78020@avaya.com	Middle Name :	Middle Name Of User

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**.

AVAYA Aura® System Manager 10.1	Users 🗸 🎤 Elements 🗸 🔅 Se	ervices ~ Widgets ~ Shortcuts	s 🗸 Sea	arch 🔷 🙏 🚍 🛛 admin
Home User Managemen	E			
User Management 🔷	Home合 / Users / Manage Users			Help ?
Manage Users	User Profile Add		Commit & Continue	🖻 Commit 🛞 Cancel
Public Contacts	Identity Communication Pr	rofile Membership Contacts		
Shared Addresses	Communication Profile Password	🖉 Edit 🕂 New 🖻 Delete		Options 🗸
System Presence ACLs	PROFILE SET : Primary Y	Туре	Handle 🗢 🛛	Domain 🕈 🍸
Communication Profile	Communication Address		No dolo	_
	PROFILES Comm-P	rofile Password		×
	Session Manager Pro	Comm-Profile Password :	,	
	CM Endpoint Profile			
		* Re-enter Comm-Profile Password :	,	 Ø
		Generate	Comm-Profile Password	
		Generate		
			Cancel	ок

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.

Aura® System Manager 10.1	Users 🗸 🎤 Elements 🗸 🔅 Serv	vices ~ Widgets ~	Shortcuts v	Search	📄 🜲 🗮 admin
Home User Management	E Constanting of the second				
User Management ^	Home슯 / Users 였 / Manage Users				Help ? 🔺
Manage Users	User Profile Add		D	Commit & Continue	ommit 🛞 Cancel
Public Contacts	Identity Communication Prot	file Membership C	ontacts		
Shared Addresses	Communication Profile Password	SIP Registration			
System Presence ACLs	PROFILE SET : Primary V	* Primary Session			
Communication Profile	Communication Address	Manager:	devcon-sm Q		
	PROFILES			_	
	Session Manager Profile	Secondary Session Manager:	Start typing Q		
	CM Endpoint Profile			_	
		Survivability Server:	Start typing Q		
		Max. Simultaneous	Select ~		
		Devices :			
		Block New Registration			
		When Maximum			
		Application Sequen	ces		
		Origination Sequence:	DEVCON-CM App S v		
<		Termination Sequence:	DEVCON-CM App S v		
	1			-	

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



* Home Location :	Thornton	Q

6.3.5. CM Endpoint Profile

Click on the toggle button by **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.1**. For **Template**, select *9641SIP_DEFAULT_CM_8_1*. For **Port**, click and select *IP*. Retain the default values in the remaining fields. Click on the Endpoint Editor (i.e., Edit icon in Extension field) to configure the **Coverage Path**.



Navigate to the **General Options** tab and set the **Coverage Path 1** field to the voicemail coverage path. Click **Done** (not shown) to return to the previous web page and then **Commit** to save the configuration (not shown).

New Endpoint					÷	lelp ?
·····						D <u>o</u> ne
					[Save As Tem	plate]
					Display Extension Ranges	
* System	System devcon-cm * Extension				78020	
* Template g	641SIP_DEFAULT_	CM_8_1 🗸		Set Type	9641SIP	
* Port	[P			Security Code		
Name						
						_
General Options (G) * Feat	ure Options (F)	Site Data (S) /	Abbreviated Call Dialing (A)	Enhanced Call Fwd (E)	
Button Assignment (B) Pro	file Settings (P)	Group Mem	bersh	ip (M)		
* Class of Restriction (COR)	1		*	Class Of Service (COS)	1	
* Emergency Location Ext	78020		*	Message Lamp Ext.	78020	
* Tenant Number	1					
* SIP Trunk	Qaar			Type of 3PCC Enabled	None 🗸	
Coverage Path 1	15			Coverage Path 2		
Lock Message				Localized Display Name		
Multibyte Language	Not Applicable	~		Station Domain Control	system 🗸	

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] in Section 10 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.



JAO; Reviewed: SPOC 11/16/2022 Solution & Interoperability Test Lab Application Notes ©2022 Avaya Inc. All Rights Reserved. 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.

9:18 🖪 🏟 🕊	@ ‡♀ ₿				
Biz Phone	Clear call log				
*	Speed dial				
	Call forwarding				
	Settings				
	Biz Status				
	About				
No favorites					



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9:18 🖬 🏟 🕊	_⊕ ‡♀ 🗎
← Biz Phone settings	
Admin settings	
Enable SIP ON	
Registration 1 SIP settings for registration 1	
Registration 2 SIP settings for registration 2	
Common settings Settings common to both registrations	
LDAP settings Corporate directory settings	
Emergency contact Emergency contact numbers	

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

- **SIP** server: •
 - Set to the Session Manager IP address (e.g.,
- SIP server port:
- **Transport:**
- SRTP enable:
- Extension number:
- **Username:**
- 10.64.102.117). Set to appropriate SIP port (e.g., 5060).
- Set to *TCP* transport protocol.
 - Disable this option.
- Set to the SIP extension (e.g., 78020).
 - 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.



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

- Voicemail retrieval address:
- Force subscription to message Waiting notifications:
- Disable call waiting:

Set to the voicemail pilot number (e.g., 78600).

Enable this option. Set to *OFF*.

Accept the default values for the remaining parameters.

9:19 🖪 🏟 🕊	0 🖓 🗎	9:20 🖬 🏟 🕊	● ‡♀ 🗎
← Registration 1		← Registration 1	
Voicemail retrieval address		ON	
Allow coll forwarding		Use vendor protocol if licensed	
ON		Use SIP standard hold signaling	
Use vendor protocol if licensed		ON	
Use SIP standard hold signaling		Force subscription to message waiting notifications	
ON		Allow contact header undates	
Force subscription to message waiting notifications		ON	
		Specify new TCP port in contact header	
OFF		OFF	
Specify new TCP port in contact header		Disable call waiting OFF	
< ● ■		• • •	

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9:20 🖪 💠 🕊	● \$\$ ₿	9:20 🖪 🌩 🕊	● \$♀ ∎
\leftarrow Common settings		\leftarrow Common settings	
Audio DSCP 46		G.711a codec priority 2	
Call control DSCP 40		G.722 codec priority 3	
G.711u codec priority		G.729A codec priority 4	
G.711a codec priority 2		Opus codec priority 0	
G.722 codec priority 3		DTMF relay payload type	
G.729A codec priority		Force in-band DTMF tones	
Opus codec priority 0		Override automatic switch from UDP to TCP OFF	
< ● ■		< ● ■	

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.

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.

AV Aura® Sy	rstem Manager 10.1	占 Users 🕚	🗸 🎤 Ele	ements 🗸 🔅 Ser	vices v	Widg	ets v Sł	nortcuts v					Search				≡	admin
Home	Session Manage	er																
Sessio	n Manager 🔷 🔨	A																Help ?
D	ashboard	Select registr	er Regi rows to send ation status.	strations I notifications to device	. Click on Deta	ils columr	n for complete	9										
Se	ession Manager 🗡																Cust	:omize 🕨
G	lobal Settings	View Default Export Force Unregister AST Device Reboot Reload Failback As of 1:02 PM Advanced Search																
C	ommunication Prof	22 It	ems 🛛 💝 👘	Show 15 ¥								_					Filter:	Enable
N	abuark Canfigura V		Details	Address	First Name	Last Name	Actual Location	IP Address	Policy	Shared Control	Simult. Devices	AST Device	Regist	ered	3rd	4th	Surv	Visiting
IN IN	etwork Conligur +		▶ Show	78021@avava.com	Spectralink	78021		192.168.100.198	fixed		1/1							
D	evice and Locati 💙		▶ Show	78020@avaya.com	Spectralink	78020		192.168.100.196	fixed		1/1		~					
A	pplication Confi		▶ Show		SIP	78001			fixed		0/1							
			► Show		SIP	78000			fixed		0/1							
Sj	/stem Status 🔹 🔨		►Show		Remote	78801			fixed		0/1							
	Load Factor		► Show	78002@avaya.com	SIP	78002		192.168.100.59	fixed		1/1	✓	(AC)					
	SIP Entity Monit	Selec	t : All, Non	e										14 4	Pag	je 🔤	2 of	2 ▶ ▶
	Managed Band																	
	Security Module																	
	SIP Firewall Status																	
	Registration Su																	
	User Registratio																	
	Session Counts	-																
	<																	

2. 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.





3. 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 Sign	aling 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 Sign	aling Loc: H.	245 Tunneled in Q.931? no						
Audio Connec	tion Type: ip-direct	Authentication Type: None						
Near-end	Audio Loc:	Codec Type: G.711MU						
Audio	IP Address	Port						
Near-end:	192.168.100.59	: 2048						
Far-end:	192.168.100.196	: 46239						
Video Near:								
Video Far:								
Video Port:								
Video Near-	end Codec:	Video Far-end Codec:						

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 and Avaya Aura® Session Manager. 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 10.1, Issue 1, December 2021, available at <u>http://support.avaya.com</u>.
- [2] Administering Avaya Aura® System Manager, Release 10.1.x, Issue 6, June 2022, available at http://support.avaya.com.
- [3] Administering Avaya Aura® Session Manager, Release 10.1.x, Issue 3, April 2022, available at http://support.avaya.com.
- [4] Spectralink Versity Smartphone Family User Guide, Release 1.4 for Versity 92 Series, Release 2.3 for Versity 95/96 Series, 720-0075-000 Rev E, February 2022, available at <u>https://support.spectralink.com/versity</u>.

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