



## **Application Notes for Workforce Connect Voice Client running on MC40 Mobile Computer from Zebra Technologies with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.0**

### **Abstract**

These Application Notes describe the integration of the Workforce Connect Voice Client running on MC40 mobile computer with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Workforce Connect Voice Client runs on Android based Voice enabled mobile computers. Workforce Connect Voice Client on MC40 registers with Avaya Aura® Session Manager as a SIP endpoint through the enterprise wireless LAN.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any 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 integration of the Workforce Connect (WFC) Voice Client running on an MC40 mobile computer with Avaya Aura® Communication Manager (Communication Manager) and Avaya Aura® Session Manager (Session Manager). The WFC Voice Client runs on Android based Voice enabled mobile computers. The WFC Voice Client registers with Avaya Aura® Session Manager as a SIP endpoint through the enterprise wireless LAN.

The WFC Voice Client provides the capability to customize its user interface by adding telephony feature buttons.

Feature buttons that are associated with telephony features are supported locally by WFC Voice Client such as Do Not Disturb, Hold, and Call Transfer etc. Refer to WFC Voice Client documentation for a full list of feature buttons supported, and button configuration and operation.

The features tested by this solution are listed below.

Automatic Redial	Call Forward	Message Waiting Indicator
Call Hold/Resume	Call Park/Unpark	Speed Dial Buttons
Consultation Hold	Do Not Disturb	
Transfer	Call Pickup (only using	
Conference	feature access code)	

## 2. General Test Approach and Test Results

This section details the general approach to the testing, what was covered, and results of the testing. If the testing was successfully concluded but it was necessary to implement workarounds or certain non-critical features did not work, it should be noted in **Section 2.2**.

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between the WFC Voice Client and Avaya SIP, H.323, and digital telephones and exercising basic telephony features, such as hold, mute, transfer, and conference. Additional telephony features, such as call forward, EC500, call park/unpark, and call pickup were also verified.

The serviceability testing focused on verifying that the WFC Voice Client comes back into service after rebooting it or the wireless LAN device.

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's formal testing and Declaration of Conformity is provided only on the headsets/Smartphones that carry the Avaya brand or logo. Avaya may conduct testing of non-Avaya headset/handset to determine interoperability with Avaya phones. However, Avaya does not conduct the testing of non-Avaya headsets/Smartphones for: Acoustic Pressure, Safety, Hearing Aid Compliance, EMC regulations, or any other tests to ensure conformity with safety, audio quality, long-term reliability or any regulation requirements. As a result, Avaya makes no representations whether a particular non-Avaya headset will work with Avaya's telephones or with a different generation of the same Avaya telephone.

Since there is no industry standard for handset interfaces, different manufacturers utilize different handset/headset interfaces with their telephones. Therefore, any claim made by a headset vendor that its product is compatible with Avaya telephones does not equate to a guarantee that the headset will provide adequate safety protection or audio quality.

## 2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of WFC Voice Client with Avaya Aura® Session Manager.
- Calls between WFC Voice Client and Avaya SIP, H.323, and digital telephones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between WFC Voice Client and PSTN.
- G.711MU/A and G.729 codec support.
- Proper recognition of DTMF tones.
- Basic telephony features, including hold, mute, redial, multiple calls, call display, blind and supervised transfer, and attended conference.
- Extended telephony features using MC40 feature buttons for Do Not Disturb, Call Forward, Call Park/Unpark and Call Pickup (only using feature access code).
- Voicemail coverage, MWI support, and logging into voicemail system to retrieve messages.
- Proper system recovery after a restart of the WFC Voice Client and loss of wireless connectivity.

## 2.2. Test Results

All test cases passed with the following observations noted:

- WFC Voice Client does not support Initial IP-IP Direct Media. During integration, ensure that the Initial IP-IP Direct Media value is configured as “n” (see **Section 5.5**). Due to this reason, not all calls use direct media and certain calls use the media resources from Avaya Aura® Communication Manager, hence ensure that Avaya Aura® Communication Manager Media cards have sufficient DSPs while implementing this solution.
- Due to the above reason, certain calls that do not use direct media do not support codec negotiation at G722 and hence this codec should not be used while implementing this solution.
- Call Pickup key is not a programmable feature button in MC40 and hence this feature was tested using only feature access code. All other features mentioned in these Application Notes were tested using feature buttons programmed in MC40.

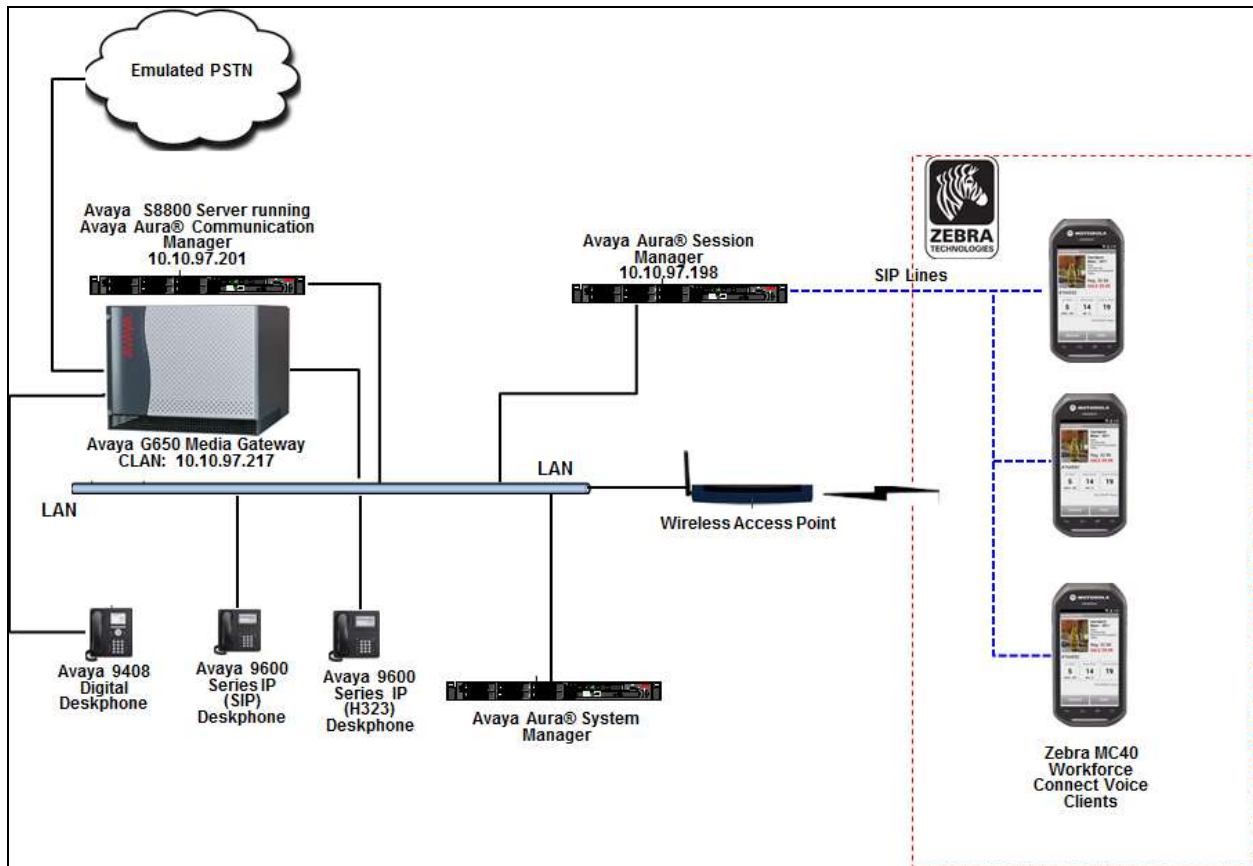
## 2.3. Support

For technical support on the Workforce Connect Voice Client, contact Zebra Technologies support via phone or website.

- **Phone:** 1.800.653.5350
- **Web:** <http://www.zebra.com>

### 3. Reference Configuration

**Figure 1** illustrates a sample configuration consisting of MC40 Workforce Connect Voice Clients with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. In this configuration, the Workforce Connect Voice Client runs on the MC40 mobile computer and connects to the enterprise wireless network. The WFC Voice Client registers with Session Manager via SIP.



**Figure 1: MC40 Workforce Connect Voice Client with Avaya Aura® Communication Manager and Avaya Aura® Session Manager**

## 4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura <sup>®</sup> System Manager running on an Avaya S8800 Server	6.3.14.11.3595
Avaya Aura <sup>®</sup> Communication Manager running on an Avaya S8800 Server	6.3-03.0.124.0
Avaya Aura <sup>®</sup> Session Manager running on an Avaya S8800 Server	6.3.14.0.631402
Avaya 96xx Series Deskphone	9621 H.323 Release 6.6029
Avaya 96xx Series Deskphone	9640 SIP Release 2.6.11.4
Avaya 94xx Series Digital Deskphone	9408 FW Version 12
Workforce Connect Voice Client running on MC40 Wireless handset	6.0.542

## 5. Configure Avaya Aura® Communication Manager

It is assumed that a fully functioning Communication Manager is in place with the necessary licensing with a SIP Trunk in place to Session Manager. For further information on the configuration of Communication Manager please see **Section 10** of these Application Notes. The following sections go through the following.

- Dial Plan Analysis
- IP Interfaces
- Network Region
- IP Codec
- Signalling Group

### 5.1. Configure Dial Plan Analysis

Use the **change dialplan analysis** command to configure the dial plan using the parameters shown below. Extension numbers (**ext**) are those beginning with **53**.

change dialplan analysis						Page 1 of 12			
DIAL PLAN ANALYSIS TABLE									
Location: all						Percent Full: 7			
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	
53	5	ext							

### 5.2. Configure IP Interfaces

Shown below is an example of the nodes names used in the compliance testing. Note that MC40 Workforce Voice Connect does not feature in this setup and only the name and IP address of Session Manager and Communication Manager processor is added. Use the **change node-names ip** command to configure the IP address of Session Manager and Communication Manager processor. **SM61** is the **Name** used for Session Manager and **10.10.97.198** is the **IP Address**. **procr** is the **Name** used for Communication Manager processor and **10.10.97.201** is the **IP Address**.

change node-names ip		IP NODE NAMES	
Name	IP Address		
SM61	10.10.97.198		
procr	10.10.97.201		
CLAN1	10.10.97.217		

### 5.3. Configure Network Region

Use the **change ip-network-region x** (where x is the network region to be configured) command to assign an appropriate domain name to be used by Communication Manager. In the example below *bvwdev.com* is used. Note this domain is also configured in **Section 6.1** of these Application Notes.

```
change ip-network-region 1                                     Page 1 of 20

                                IP NETWORK REGION

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

### 5.4. Configure IP-Codec-Set

Use the **change ip-codec-set x** (where x is the ip-codec set used) command to designate a codec set compatible with MC40 Workforce Voice Connect clients, which supports *G.711MU*, *G.729*, and *G.711A*.

```
change ip-codec-set 1                                         Page 1 of 2

                                IP CODEC SET

Codec Set: 1

Audio      Silence      Frames      Packet
Codec      Suppression   Per Pkt    Size(ms)
1: G.711MU      n           2         20
2: G.729        n           2         20
3: G.711A       n           2         20
```



## 5.5. Configure SIP Signaling Group

For the compliance test, a signaling group and the associated SIP trunk group and route was used for connecting Avaya Aura® Communication Manager with the Avaya Aura® Session Manager. The configuration of SIP trunk group and route is outside the scope of these Application Notes. Only the configuration of the SIP Signaling Group is shown below. For further details on other fields refer to **Section 10**.

Use the **change signaling-group x** (where x is the signaling-group used) command to configure the SIP Signaling Group

- The **Group Type** was set to *sip*.
- The **Transport Method** was set to *tcp*. As a result, the **Near-end Listen Port** and **Far-end Listen Port** are automatically set to *5060*.
- The **Near-end Node Name** was set to *procr*, the node name that maps to the IP address of the circuit pack used to connect to Session Manager. Node names are defined using the **change node-names ip** command (see **Section 5.2**).
- The **Far-end Node Name** was set to *SM61*. This node name maps to the IP address of the Session Manager server as defined using the **change node-names ip** command (see **Section 5.2**).
- The **Far-end Network Region** was set to *1*. This is the IP network region which contains Session Manager (see **Section 5.3**).
- The **Far-end Domain** was set to *bvwdev.com*. This domain is sent in the headers of SIP INVITE messages for calls originating from and terminating to Session Manager using this signaling group.
- **Direct IP-IP Audio Connections** was set to *y*. This field must be set to *y* to enable Media Shuffling on the trunk level.
- **Initial IP-IP Direct Media** was set to *n*.

Retain default values for all other fields.

```

display signaling-group 1
                                     Page 1 of 2
                                     SIGNALING GROUP

Group Number: 1                     Group Type: sip
IMS Enabled? n                     Transport Method: tcp
Q-SIP? n
IP Video? y                         Priority Video? n           Enforce SIPS URI for SRTP? y
Peer Detection Enabled? y Peer Server: SM
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: SM61
Near-end Listen Port: 5060          Far-end Listen Port: 5060
                                     Far-end Network Region: 1

Far-end Domain: bvwddev.com

Incoming Dialog Loopbacks: eliminate Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload           RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3 Direct IP-IP Audio Connections? y
Enable Layer 3 Test? y               IP Audio Hairpinning? n
H.323 Station Outgoing Direct Media? n Initial IP-IP Direct Media? n
                                     Alternate Route Timer(sec): 30

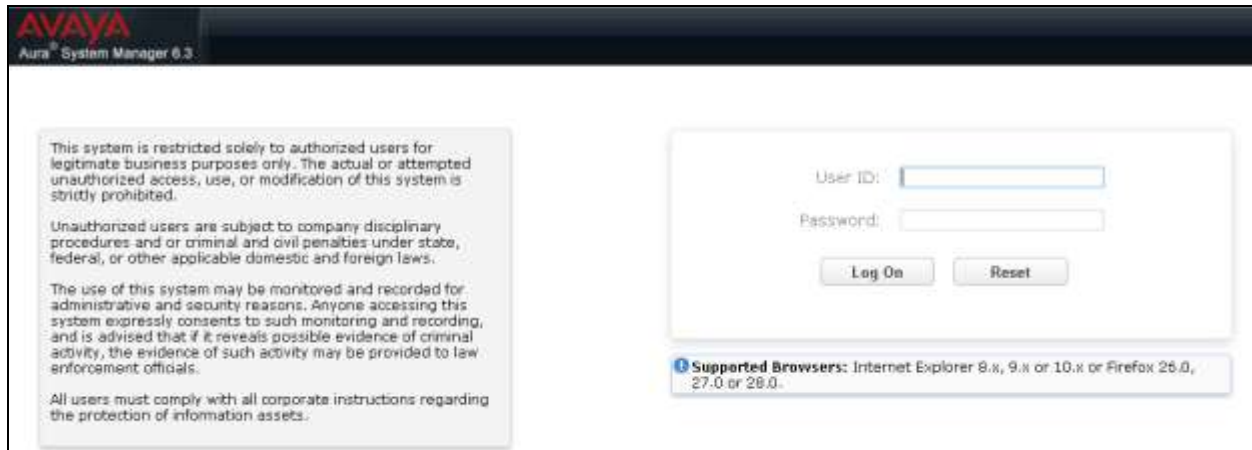
```

## 6. Configure Avaya Aura® Session Manager

The WFC Voice Clients are added to Session Manager as SIP Users. In order to make changes in Session Manager, a web session to System Manager is opened.

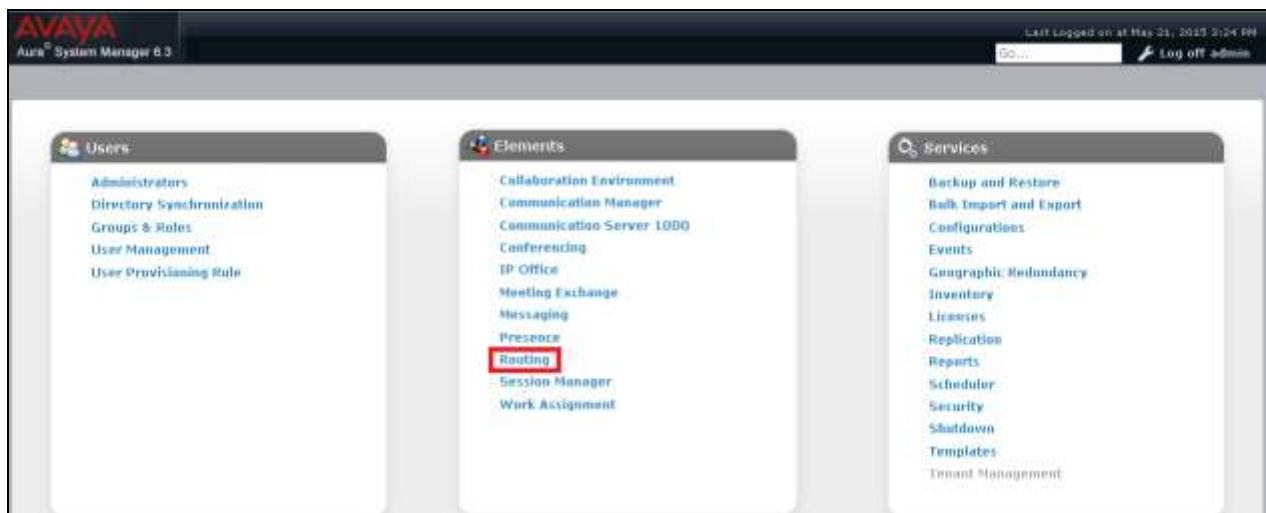
### 6.1. Configuration of a Domain

Navigate to <http://<System Manager IP Address>/>, enter the appropriate credentials and click on **Log On** as shown below.



The image shows the login page of Avaya Aura System Manager 6.3. The page has a dark header with the Avaya logo and 'Aura System Manager 6.3'. The main content area is white and contains a login form on the right and a disclaimer on the left. The login form has fields for 'User ID:' and 'Password:', followed by 'Log On' and 'Reset' buttons. Below the form, a blue box lists supported browsers: Internet Explorer 8.x, 9.x or 10.x or Firefox 26.0, 27.0 or 28.0. The disclaimer on the left states that the system is restricted to authorized users and that unauthorized use is prohibited.

Once logged in click on **Routing** as highlighted below.



Click on **Domains** in the left window. If there is not a domain already configured, click on **New** highlighted below.

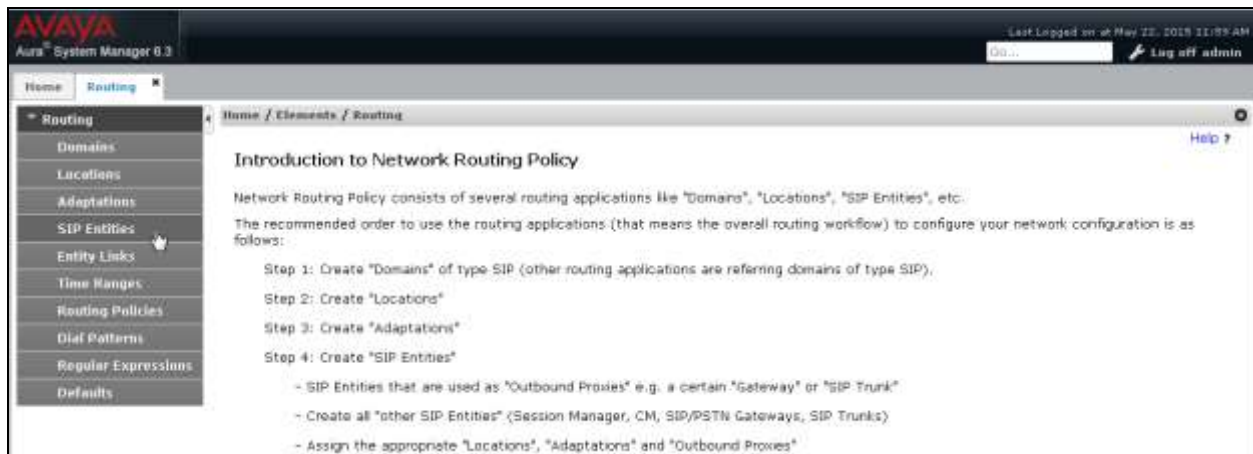


Note the domain **Name** used in the compliance testing was **bwvdev.com**. Note this domain is also referenced in **Section 5.3**. Once the domain name is entered, click on **Commit** (not shown) to save this value.



## 6.2. Configuration of SIP Entities

Log into System Manager as described in **Section 6.1** above, click on **SIP Entities** shown below.



Clicking on **SIP Entities** shows what SIP Entities have been added to the system and allows the addition of any new SIP Entity that may be required. Please note the SIP Entities present for this Compliance Testing.

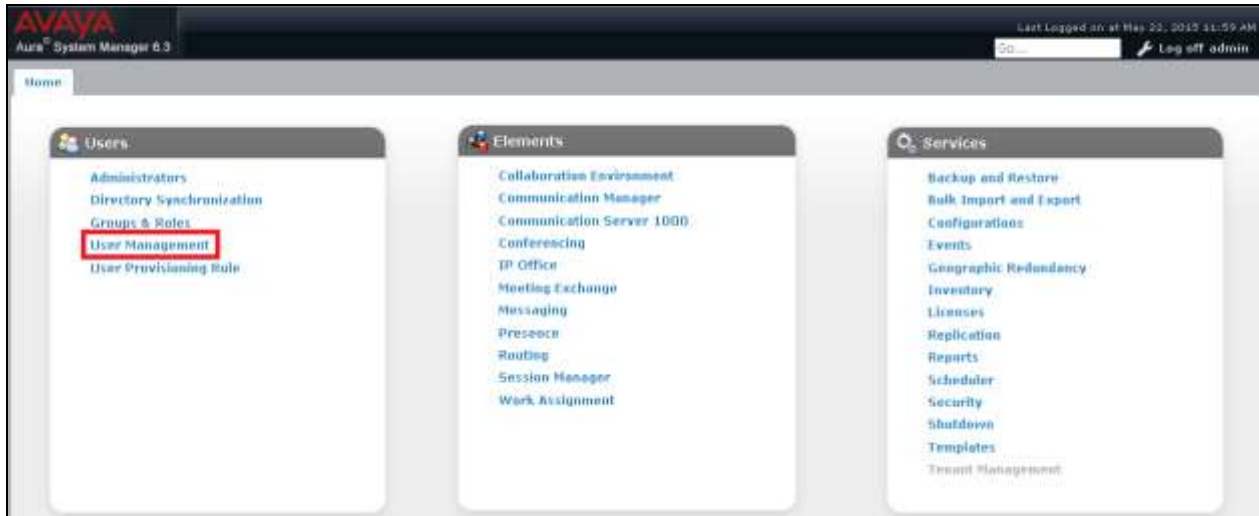
- Session Manager SIP Entity
- Communication Manager SIP Entity

**Note:** There is no SIP Entity present or required for WFC Voice Clients.



### 6.3. Adding Workforce Connect Voice Client as SIP User

From the **Home** page, click on **User Management** as highlighted below.



Click on **Manage Users**. Click on **New** as highlighted below to add a new SIP user.



Under the **Identity** tab, fill in the user's **Last Name** and **First Name** as shown below. Enter the **Login Name** and ensure **Authentication Type** is set to **Basic**.

The screenshot shows the 'Identity' tab of a user provisioning interface. The 'User Provisioning Rule' dropdown is set to 'User Provisioning Rule:'. Below this, the 'Identity' section contains several input fields: 'Last Name' (SetA), 'Last Name (Latin Translation)' (SetA), 'First Name' (WFC), 'First Name (Latin Translation)' (WFC), 'Middle Name' (empty), 'Description' (empty), 'Update Time' (August 6, 2015 11:30), 'Login Name' (53130@bvwdev.com), and 'Authentication Type' (Basic). A 'Change Password' link is visible at the bottom.

Under the **Communication Profile** tab, enter a suitable **Communication Profile Password** and click on **Done** when added. Note that this password is required when configuring the WFC Voice Client in **Section 7**. Click on **New** to add a new **Communication Address**.

The screenshot shows the 'Communication Profile' tab of the same user provisioning interface. The 'Communication Profile Password' field is filled with a masked password and has an 'Edit' button next to it. Below this, there are buttons for 'New', 'Delete', 'Done', and 'Cancel'. The 'Name' field is set to 'Primary' and is marked as the 'Default'. At the bottom, there is a 'Communication Address' section with a 'New' button highlighted.

Select **Type** as *Avaya SIP* and enter the extension number and the domain for the **Fully Qualified Address** and click on **Add** once finished.

Communication Address ▼

New Edit Delete

Type	Handle	Domain
Avaya SIP	53130	bvwdev.com

Select : All, None

Type: Avaya SIP ▼

\* Fully Qualified Address: 53130 @ bvwdev.com ▼

Add Cancel

Ensure **Session Manager Profile** is checked and enter the **Primary Session Manager** details, enter the **Origination Sequence**, the **Termination Sequence** and the **Home Location** as highlighted below. Note that **DevCM-SEQ** is an application sequence that corresponds to the Communication Manager in the test configuration and has been configured in the system previously.

☒ Session Manager Profile ▼

**SIP Registration**

\* Primary Session Manager DevSM ▼

Primary	Secondary	Maximum
44	0	44

Secondary Session Manager (None) ▼

Survivability Server (None) ▼

Max. Simultaneous Devices 1 ▼

Block New Registration When Maximum Registrations Active? ☐

**Application Sequences**

Origination Sequence DevCM-SEQ ▼

Termination Sequence DevCM-SEQ ▼

**Call Routing Settings**

\* Home Location Belleville ▼

Conference Factory Set (None) ▼



Ensure that **CM Endpoint Profile** is selected. Select *DevCM* as **System** and *Endpoint* for **Profile Type**. Enter *53130* for **Extension** and choose the *9611SIP\_DEFAULT\_CM\_6\_3* as the **Template**. Click **Endpoint Editor** to configure the buttons and features for that handset on Communication Manager.

**CM Endpoint Profile**

\* System: DevCM

\* Profile Type: Endpoint

Use Existing Endpoints: ☐

\* Extension: 53130 Endpoint Editor

Template: 9611SIP\_DEFAULT\_CM\_6\_3

Set Type: 9611SIP

Security Code:

Port: IP

Voice Mail Number:

Preferred Handle: (None)

Enhanced Callr-Info display for 1-line phones: ☐

Delete Endpoint on Unassign of Endpoint from User or on Delete User: ☒

Override Endpoint Name and Localized Name: ☒

Screen below shows the window when **Endpoint Editor** is clicked. During compliance testing all default values were retained.

**Endpoint Editor**

General Options (G) | Feature Options (F) | Site Data (S) | Abbreviated Call Dialing (A) | Enhanced Call Forward (E) | Button Assignment (B) | Profile Settings (P) | Group Membership (M)

\* Class of Restriction (COR): 1

\* Emergency Location Ext.: 53130

\* Tenant Number: 1

\* SIP Trunk: aar

Coverage Path 1: 1

Lock Message: ☐

Multibyte Language: Not Applicable

\* Class of Service (COS): 1

\* Message Lamp Ext.: 53130

Type of 3PCC Enabled: None

Coverage Path 2: 1

Localized Display Name: SWA, WFE


\*Required

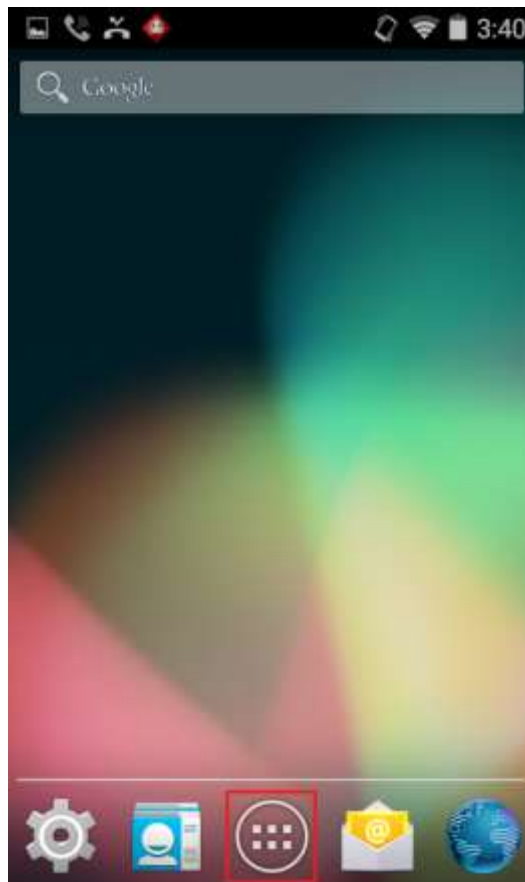
Done Cancel

## 7. Configure Workforce Connect Voice Client

This section provides the procedures for configuring the WFC Voice Client for SIP connectivity to Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

**Note:** Connecting the MC40 Android-based voice-enabled mobile computer to the wireless network and configuring feature buttons on the WFC Voice Client are outside the scope of these Application Notes.

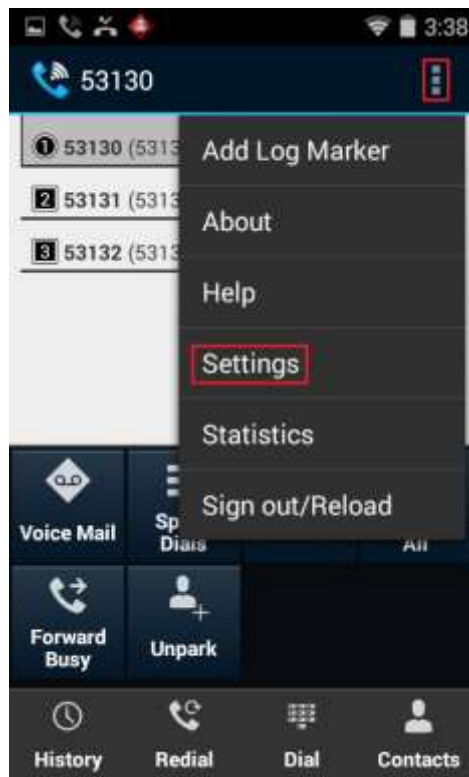
Power on the WFC Voice Client and unlock the MC40 Android-based voice-enabled mobile computer. The following screen is displayed. Tap on the  button, 3<sup>rd</sup> button highlighted below.



Scroll through the applications until the **WFCVoice** application is found. Select **WFCVoice**



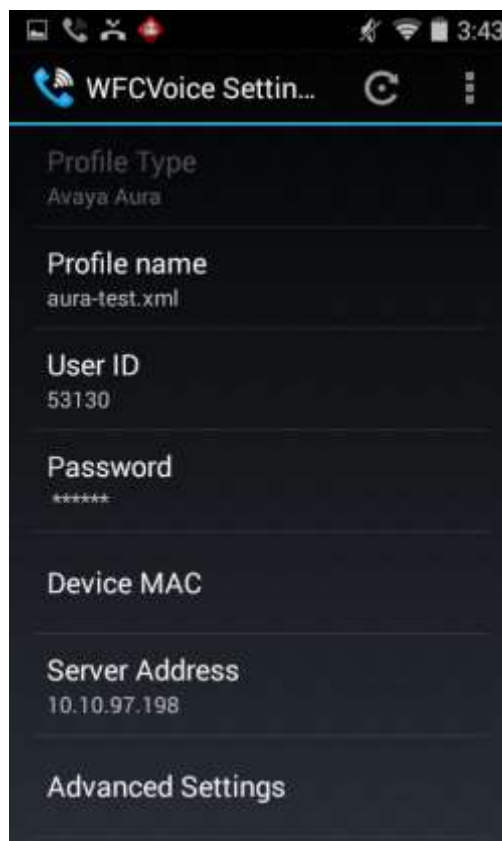
From the **WFCVoice** screen shown below, touch  on the MC40 Android-based voice-enabled mobile computer to display the menu below. From the menu, select **Settings**.



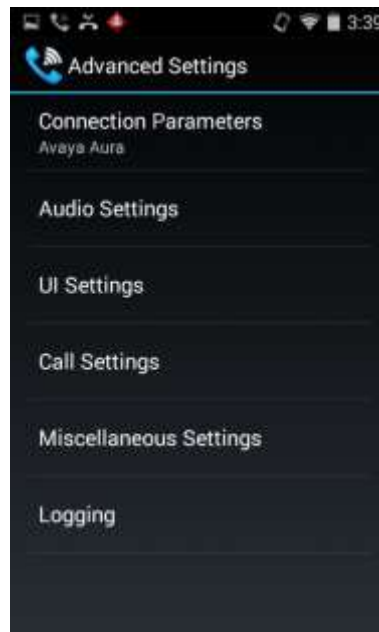
The **WFCVoice Settings** screen is displayed. Tap on the following fields and configure them as follows.

Field Name	Description
<b>Profile Type</b>	Provide a descriptive name for profile type (e.g., <i>Avaya Aura</i> ).
<b>User ID</b>	Specify the SIP extension configured in <b>Section 6.3</b> . In this example, the SIP extension was <i>53130</i> . This is the SIP extension that WFC Voice client will use to register with Session Manager.
<b>Password</b>	Specify the SIP password configured on Session Manager in <b>Section 6.3</b> . WFC Voice client will use this password to register with Session Manager.
<b>Server Address</b>	This is the IP address of Session Manager. In this example, the address is <i>10.10.97.198</i> .

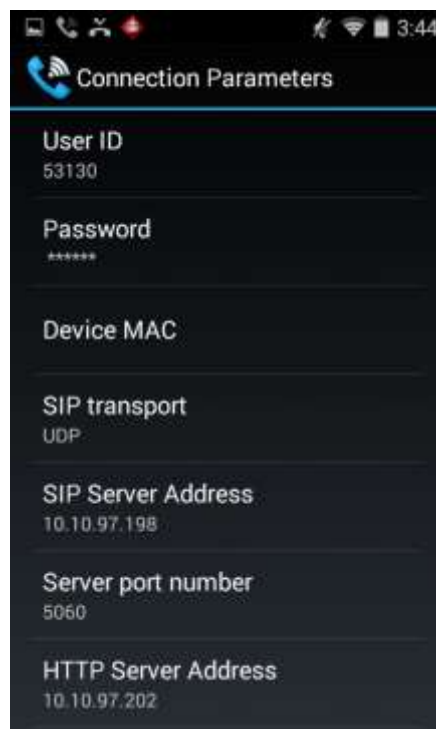
The following WF Connect screen shows these fields configured appropriately for the test configuration. These field values should be modified to correspond to the customer's network. Next, select **Advanced Settings** to review other SIP parameters.



From the **Advanced Settings** screen, select **Connection Parameters Avaya Aura** to display the **Connection Parameters**.



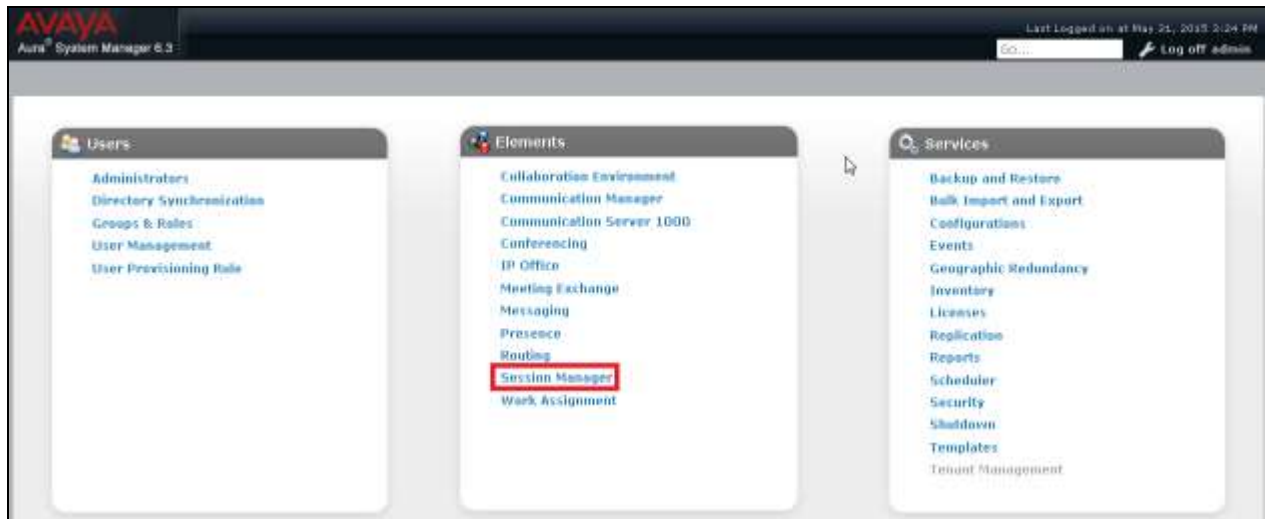
The **Connect Parameters** screen shows that *UDP* was used as the **SIP transport** and *5060* was used as the **Server port number**. Also note that an **HTTP Server Address** was configured which in most cases is the IP address of a Utility Server where the 46xxsettings file reside.



## 8. Verification Steps

The following steps can be taken to ensure that connections between WFC Voice Client and Session Manager and Communication Manager are up.

Log into System Manager as done previously in **Section 6.1**, select **Session Manager** as highlighted below.



Select **System Status** and **User Registrations** in the left column (not shown). This screen displays the users that are currently registered with Session Manager. The WFC Voice clients should show as being registered as highlighted below for extension 53130.

User Registrations												
Select rows to send notifications to devices. Click on Details column for complete registration status.												
View ▾ Default Force Unregister AST Device Notifications: Reboot Reload ▾ Failback As of 4:40 PM Customize ▾ Advanced Search ▾												
39 Items Show 15 ▾ Filter: Enable												
<input type="checkbox"/>	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered	
											Prim	Sec
<input type="checkbox"/>	Show	---	Ba	Bay	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	53102@bvwdev.com	53102	Avaya SIP	---	10.10.5.73	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>
<input type="checkbox"/>	Show	53130@bvwdev.com	WFC	SetA	---	10.10.5.221	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>

Launch **WFCVoice** and verify that the SIP extension has been registered. When **WFCVoice** is registered with Session Manager, it would display the SIP extension as shown below without any other status information, such as *Initializing*.



Verify basic telephony features by establishing calls with the WFC Voice Client.

## 9. Conclusion

These Application Notes describe the integration of the Workforce Connect Voice Client running on an MC40 mobile computer with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Workforce Connect Voice Client registered successfully with Avaya Aura® Session Manager as a SIP endpoint through the enterprise wireless LAN.

Incoming and outgoing calls were placed to/from the WFC Voice Clients and telephony features were exercised. All test cases passed with observations noted in **Section 2.2**.

## 10. References

This section references documentation relevant to these Application Notes. The Avaya product documentation is available at <http://support.avaya.com> where the following documents can be obtained.

- [1] *Administering Avaya Aura® Communication Manager*, Document ID 03-300509
- [2] *Avaya Aura® Communication Manager Feature Description and Implementation*, Document ID 555-245-205
- [3] *Implementing Avaya Aura® Session Manager* Document ID 03-603473
- [4] *Administering Avaya Aura® Session Manager*, Doc ID 03-603324

Product Documentation for Zebra Technologies can be obtained from a product supplier or may be accessed at <https://www.zebra.com/> (login required).



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