Avaya Solution & Interoperability Test Lab

Application Notes for Zebra Workforce Connect Voice Client with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.0

Abstract

These Application Notes describe the configuration of Zebra Workforce Connect Voice Client 9.0 with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. Zebra Workforce Connect Voice Client runs on Zebra TC51, TC56, and TC57 Android-based Touch Computers and registers with Avaya Aura® Session Manager as a SIP endpoint through an 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 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 of Zebra Workforce Connect Voice Client 9.0 with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. Zebra Workforce Connect Voice Client runs on Zebra TC51, TC56, and TC57 Android-based Touch Computers and registers with Avaya Aura® Session Manager as a SIP endpoint through an enterprise wireless LAN.

Zebra Workforce Connect Voice Client provides the capability to customize its user interface by adding telephony feature buttons. Some features are supported locally by the client or through Communication Manager via Feature Access Codes (FACs) or feature buttons. As an Avaya AST device, the Zebra Workforce Connect Voice Client retrieves feature buttons configured in the SIP user via Avaya Personal Profile Manager (PPM) during the SIP registration process. Zebra Workforce Connect Voice Client supports the following feature buttons:

Automatic Callback	Call Unpark	Extended Call Pickup
Call Forward All	Call Pickup	Priority
Call Forward Busy/DA	Directed Pickup	
Call Park	Exclusion	

The following features are also supported by Zebra Workforce Connect Voice Client without being configured as feature buttons in the SIP user on Communication Manager.

Attended Conference	Call Hold	Redial
Attended Transfer	Message Waiting Indicator	Programmable Buttons
Blind Transfer	Mute	Voicemail Button

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between the Zebra Workforce Connect Voice Client and Avaya H.323 / SIP Deskphones and the PSTN, and exercising telephony features listed in **Section** 1 using Communication Manager FACs and feature buttons.

The serviceability testing focused on verifying that the Zebra Workforce Connect Voice Client came back into service after rebooting the Android handset 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 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 Zebra Workforce Connect Voice Client did not include use of any specific encryption features as requested by Zebra.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of the Workforce Connect Voice Client with Session Manager as an Avaya AST device.
- Multi Device Access (MDA), where multiple Workforce Connect Voice Clients and an Avaya SIP Deskphone registered to the same SIP extension. Inbound/outbound calls and telephony features were verified on the multiple devices.
- Calls between the Workforce Connect Voice Client and Avaya H.323 / SIP Deskphones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between the Workforce Connect Voice Client and the PSTN.
- G.711, G.729, and G.722 codec support.
- Proper recognition of DTMF tones.

- Basic telephony features, including Hold, Mute, Redial, Blind and Supervised Transfer, and Attended Conference.
- Additional telephony features via feature buttons configured on Communication Manager, including Automatic Callback, Call Forward, Call Park/Unpark, Call Pickup, Directed Pickup, Extended Call Pickup, Exclusion, and Priority. As an Avaya AST device, Workforce Connect Voice Client retrieves feature button info from Session Manager using PPM.
- Voicemail coverage, MWI support, and logging into voicemail system to retrieve messages.
- Use of programmable buttons on the Workforce Connect Voice Client, including the use of Communication Manager FACs.
- Proper system recovery after a restart of the Workforce Connect Voice Client and loss of wireless network connectivity.

2.2. Test Results

All test cases passed with the following observation:

- If an outgoing call from the Workforce Connect Voice Client doesn't complete for any reason (e.g., invalid number, busy, or call blocked), the client displays "Ended" and terminates the call.

2.3. Support

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

- **Phone:** +1 (800) 653-5350
- **Web:** https://www.zebra.com/us/en/about-zebra/contact-zebra/contact-tech-support.html

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network that includes the following products:

- Communication Manager with a G450 Media Gateway and Avaya Aura® Media Server providing media resources.
- Session Manager connected to Communication Manager via a SIP trunk.
- Session Manager connected to the PSTN via Avaya Session Border Controller for Enterprise (SBCE).
- Avaya Aura® System Manager used to configure Session Manager and SIP stations on Communication Manager.
- Aura® Messaging serving as the voicemail system.
- Avaya H.323 and SIP Deskphones.
- Zebra Workforce Connect Voice Clients running on TC51, TC56, and TC57 Touch Computers with wireless LAN device providing network access (not shown).

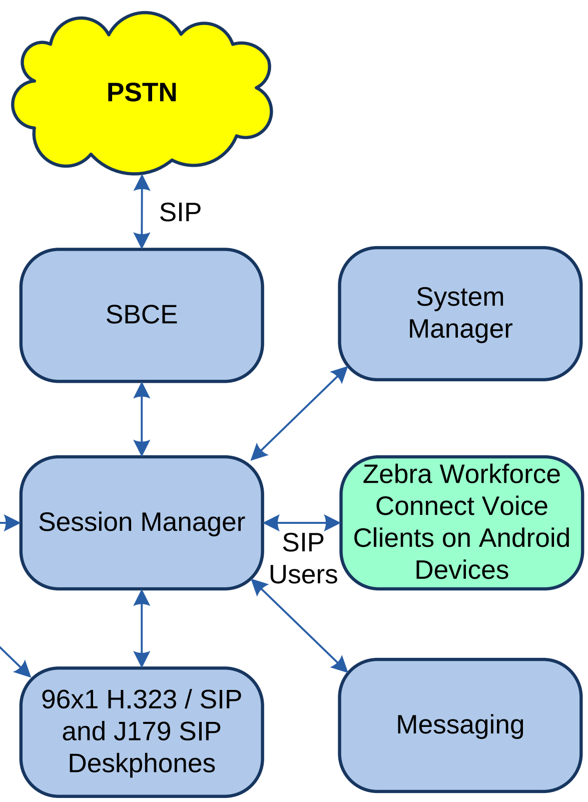


Figure 1: Avaya SIP Telephony Network with Zebra Workforce Connect Voice Clients

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	8.1.2.0-FP2 (R018x.00.0.890.0 with Patch 26095)
Avaya G450 Media Gateway	FW 41.24.0
Avaya Aura® Media Server	v.8.0.2.93
Avaya Aura® System Manager	8.1.2.0 Build No. – 8.1.0.0.733078 Software Update Revision No: 8.1.2.0.0611167 Feature Pack 2
Avaya Aura® Session Manager	8.1.2.0.812039
Avaya Aura® Messaging	7.1.3.1.0-FP3SP1
Avaya 96x1 Series IP Deskphone	6.8502 (H.323) 7.1.11.0.8 (SIP)
Avaya J100 Series SIP Deskphone	4.0.7.1.5 (SIP)
Zebra Workforce Connect Voice Client running on: ▪ TC51 Touch Computer ▪ TC56 Touch Computer ▪ TC57 Touch Computer	9.0.20306 Android 7.1.2 Android 8.1.0 Android 9

5. Configure Avaya IP Office Server Edition

This section describes the configuration of a SIP trunk to Session Manager and routing calls to Workforce Connect Voice Clients.   Administration of Communication Manager was performed using the System Access Terminal (SAT). The following configuration is covered:

- **IP Node Names** to associate names with IP addresses.
- **IP Codec Set** to specify the codec type used for calls to Workforce Connect Voice Clients.
- **IP Network Region** to specify the SIP domain name, the IP codec set, and enable IP-IP direct audio (i.e., Shuffling).
- **SIP trunk** for calls towards Session Manager and Workforce Connect Voice Clients.
- **Private Numbering** to allow the caller’s extension to be sent over the SIP trunk.
- **Call Routing** to route calls to Workforce Connect Voice Clients using AAR.

5.1. 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
```

5.2. Administer IP Codec Set

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to Workforce Connect Voice Clients. The form is accessed via the **change ip-codec-set 1** command. Note the codec set number since it will be used in the IP Network Region covered in the next section. For the compliance test, G.711MU, G.729, and G.722-64K codecs were verified. The following IP codec set is configured with G.711MU.

Media encryption was enabled for Avaya IP Deskphones. Workforce Connect Voice Clients 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 SRTCP: best-effort
 1: 1-srtp-aescm128-hmac80
 2: none
 3:
 4:
 5:
```

5.3. Administer IP Network Region

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 the Workforce Connect Voice Clients and IP endpoints without using media resources in the Avaya G450 Media Gateway or Avaya 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. This codec set is used when its corresponding network region (i.e., IP Network Region 1) is specified in the SIP signaling group.

```
change ip-network-region 1                                        Page   1 of  20
                               IP NETWORK REGION
  Region: 1       NR Group: 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
```

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*.
- Specify the Ethernet processor (*procr*) of Communication Manager and 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 in **Section 5.1**.
- 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
                                          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? 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 the Workforce Connect Voice Clients. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, 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. Accept the default values for the remaining fields.

```
add trunk-group 10                                          Page   1 of  5
                                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
                                                     Signaling Group: 10
                                                   Number of Members: 10
```

On **Page 3** of the trunk group form, set the **Numbering Format** field to *private*. This field specifies the format of the calling party number sent to the far-end.

```
add trunk-group 10                                          Page   3 of  5
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                         Maintenance Tests? y

   Suppress # Outpulsing? n  Numbering Format: private
                                              UUI Treatment: service-provider
                                            Maximum Size of UUI Contents: 128
                                               Replace Restricted Numbers? n
                                              Replace Unavailable Numbers? n

                                                Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? y
```

On **Page 4** of the trunk group form, the default settings were used as shown below.

```
add trunk-group 10                                          Page   5 of  5
                              PROTOCOL VARIATIONS

                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? n

                                     Send Diversion Header? n
                                   Support Request History? y
                              Telephone Event Payload Type:

                        Convert 180 to 183 for Early Media? n
                  Always Use re-INVITE for Display Updates? n
                        Identity for Calling Party Display: P-Asserted-Identity
            Block Sending Calling Party Location in INVITE? n
                 Accept Redirect to Blank User Destination? n
                                              Enable Q-SIP? n
          Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                                Request URI Contents: may-have-extra-digits
```

5.5. Configure Private Numbering

Configure the **Numbering – Private Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with '7' whose calls are routed over any trunk group, including SIP trunk group 10, have their extension sent.

```
change private-numbering 0                                          Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk        Private           Total
Len Code             Grp(s)     Prefix            Len
 5  7                                             5     Total Administered: 1
                                                           Maximum Entries: 540
```

5.6. AAR Call Routing

Configure the uniform dial plan table to route calls using AAR for dialed digits that are 5-digits long and begin with '78'. This would cover call routing to the Workforce Connect Voice Client extensions (i.e., 78050 – 78052).

```
change uniform-dialplan 7                                           Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                                  Percent Full: 0

  Matching                   Insert                  Node
  Pattern          Len Del       Digits        Net Conv Num
 78                 5   0                      aar  n
```

SIP calls to Session Manager are routed over a SIP trunk via AAR call routing. Configure the AAR analysis form and add an entry that routes digits beginning with "78" to route pattern 10 as shown below. Note that the **Call Type** was set to *lev0*. This routes calls to SIP stations and to the Workforce Connect Voice Clients.

```
change aar analysis 7                                               Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all            Percent Full: 2

          Dialed            Total     Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    7                     7    7     254       aar          n
    78                    5    5     10        lev0         n
    8                     7    7     254       aar          n
    9                     7    7     254       aar          n
                                                            n
                                                            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
                             Dgts                                     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 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
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

6. Configure Avaya Aura® Session Manager

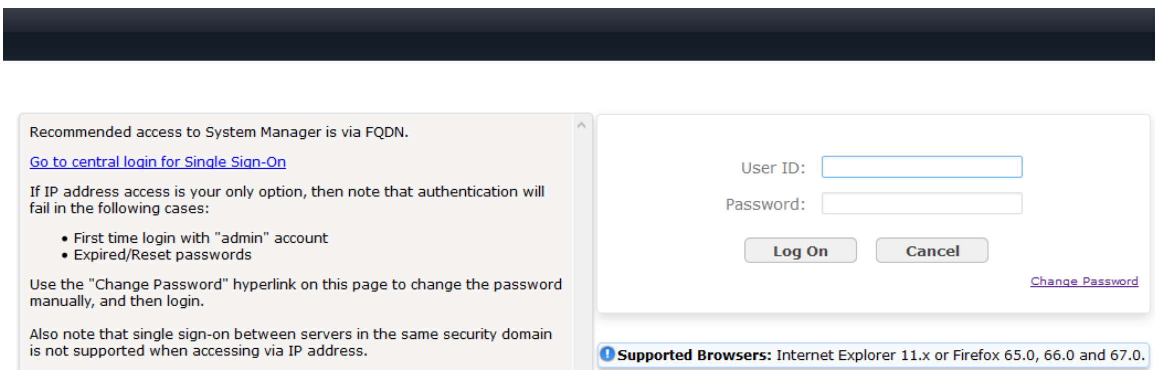
This section covers the procedure for adding a SIP user in Session Manager. The configuration covers:

- Launch System Manager
- Set Network Transport Protocol for Workforce Connect Voice Clients
- Administer SIP User

Note: It is assumed that basic configuration of Session Manager has already been performed.

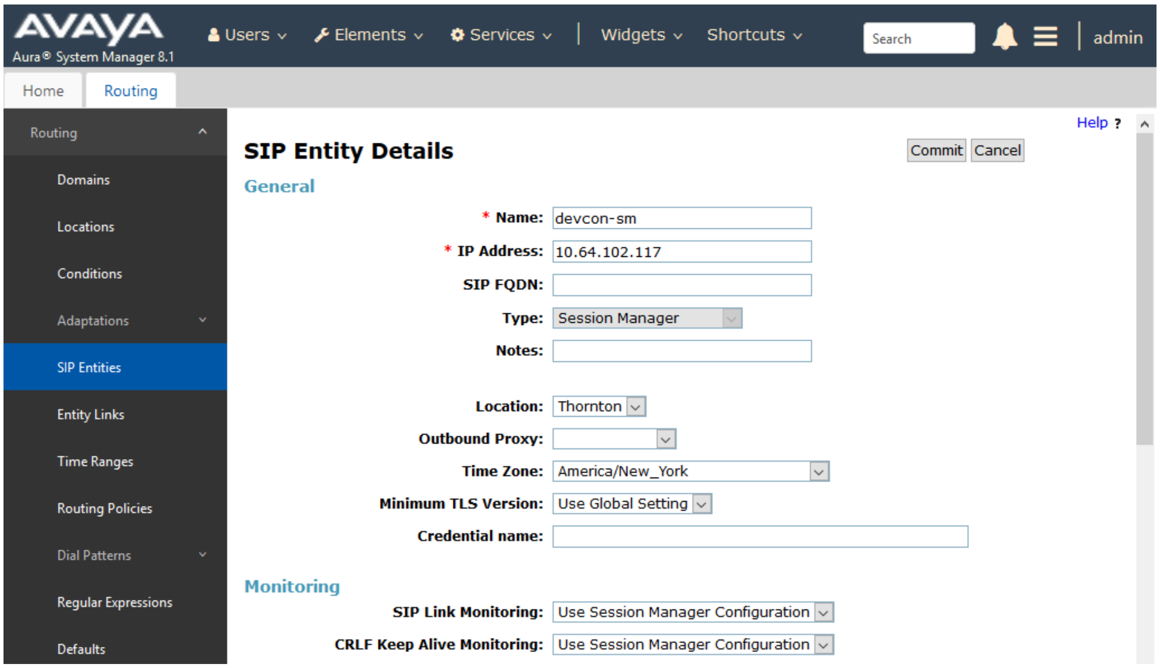
6.1. Launch System Manager

Access the System Manager Web interface by using the URL "https://ip-address" in an Internet browser window, where "ip-address" is the IP address of the System Manager server. Log in using the appropriate credentials.

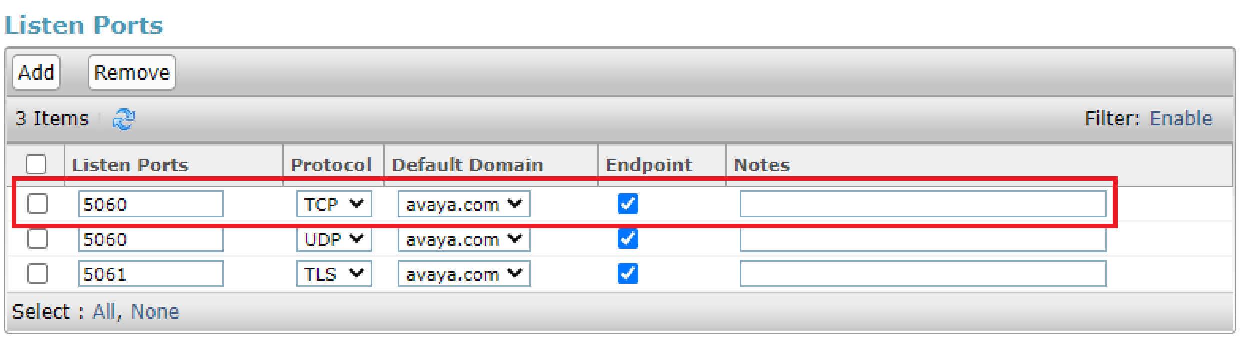


6.2. Set Network Transport Protocol for Zebra Workforce Connect Voice Clients

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

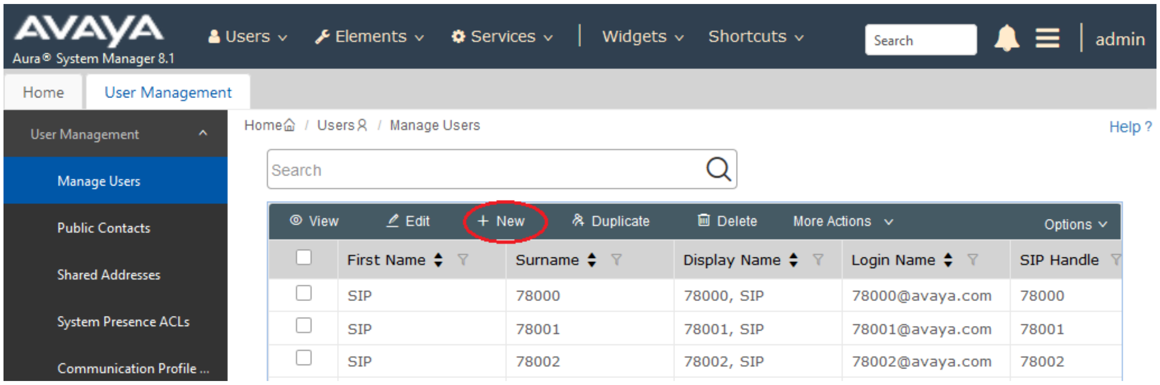


Scroll down to the **Listen Ports** section and verify that the transport network protocol used by Workforce Connect Voice Clients is specified in the list below. For the compliance test, the solution used TCP network transport.



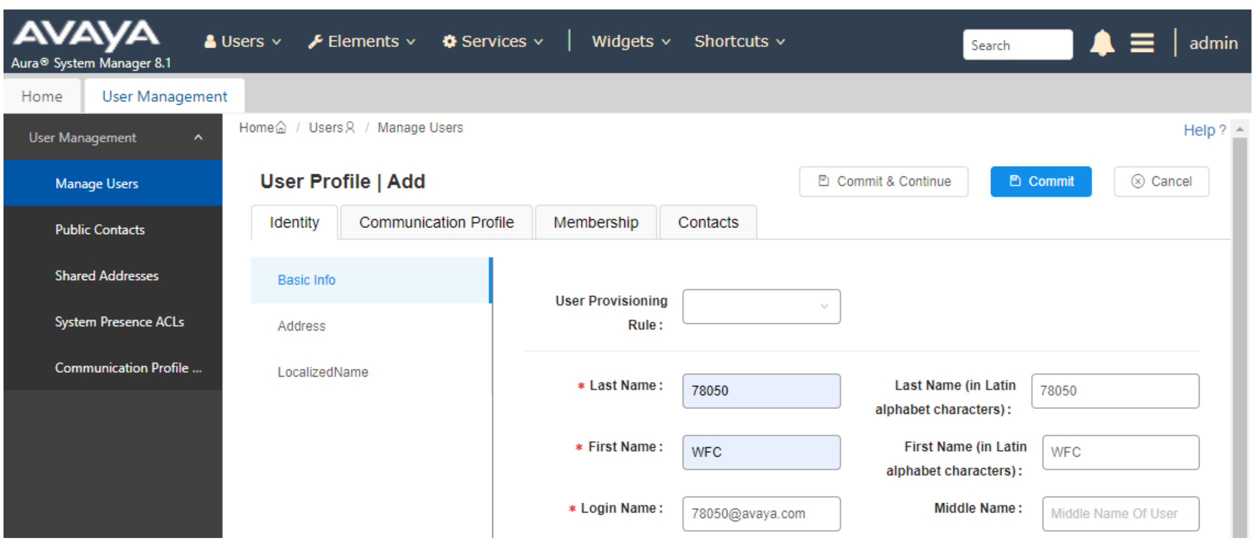
6.3. Administer SIP User

In the subsequent screen (not shown), select **Users → User Management → Manage Users** to display the **User Management** screen below. Click **New** to add a user.



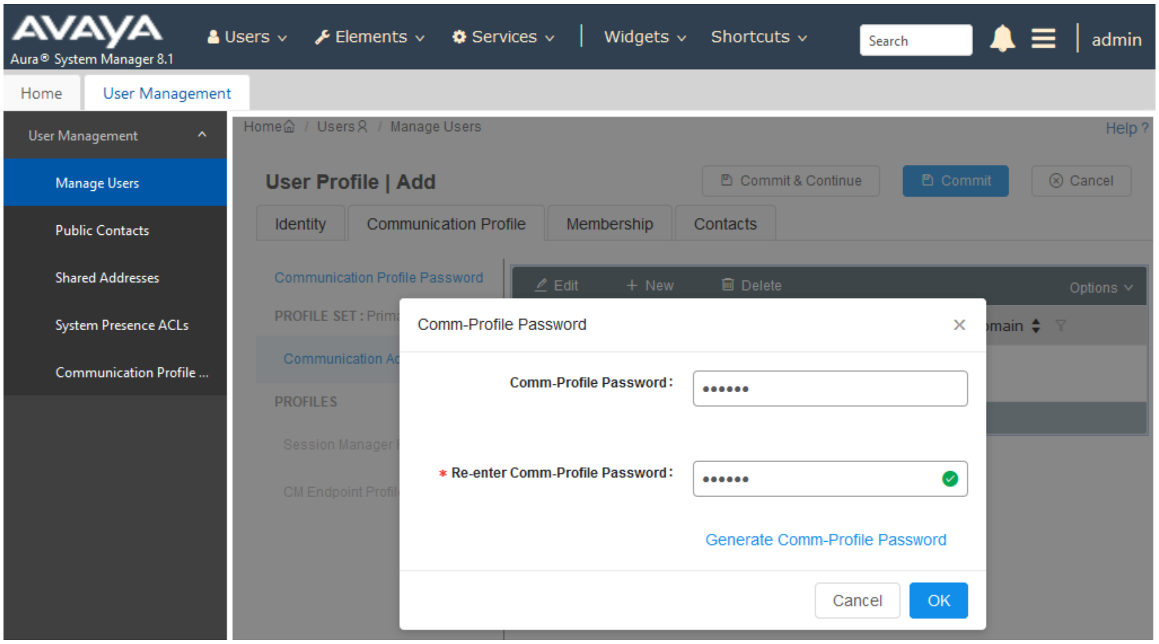
6.3.1. Identity

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 Workforce Connect Voice Client SIP extension and "*<domain>*" is the applicable SIP domain name from **Section 5.3**. Retain the default values in the remaining fields.



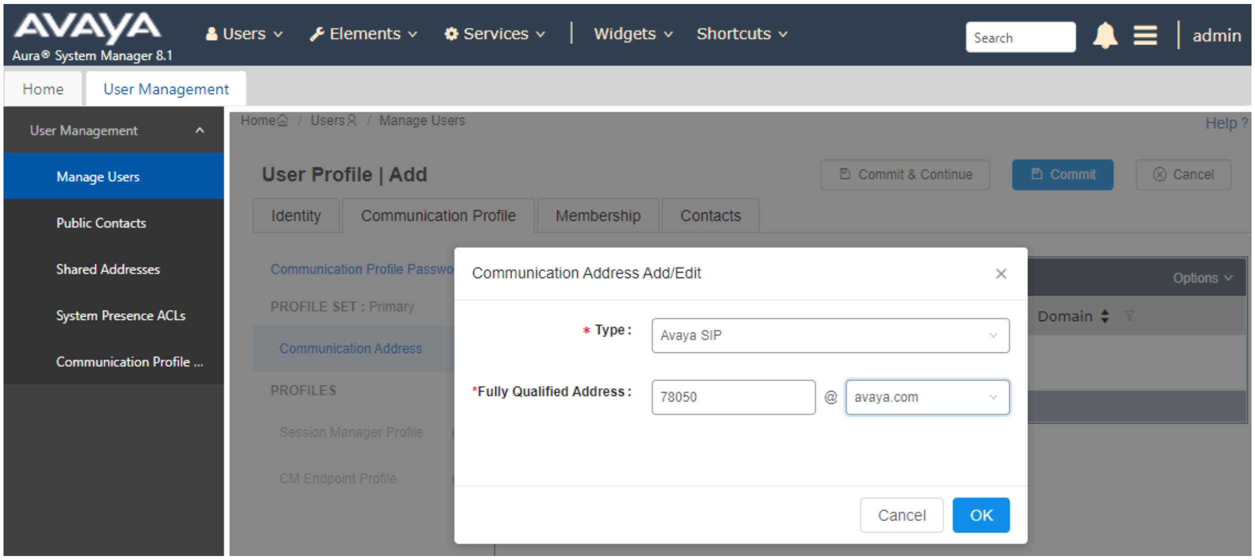
6.3.2. Communication Profile

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**.



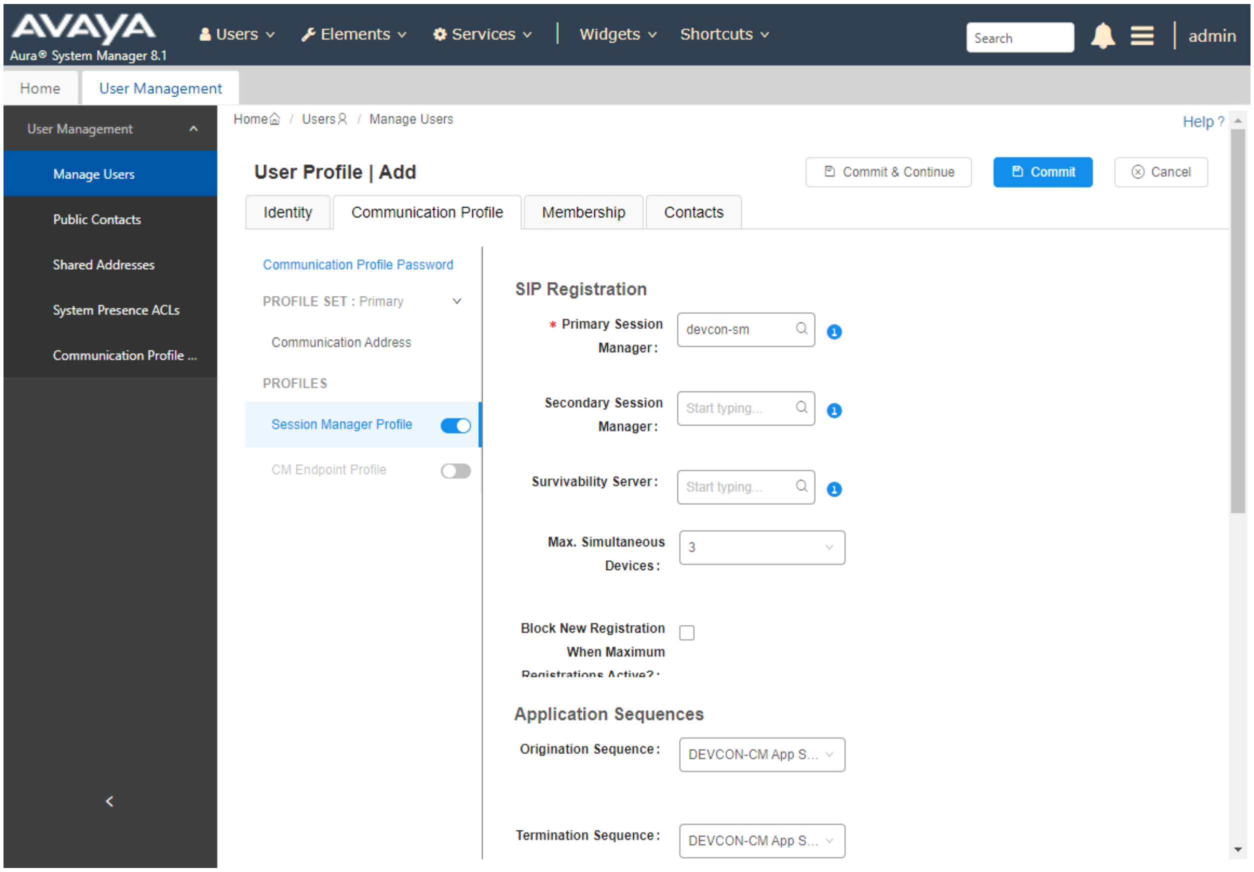
6.3.3. Communication Address

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**, retain *Avaya SIP*. For **Fully Qualified Address**, enter and select the SIP user extension and domain name to match the login name from **Section 6.3.1** Click **OK**.

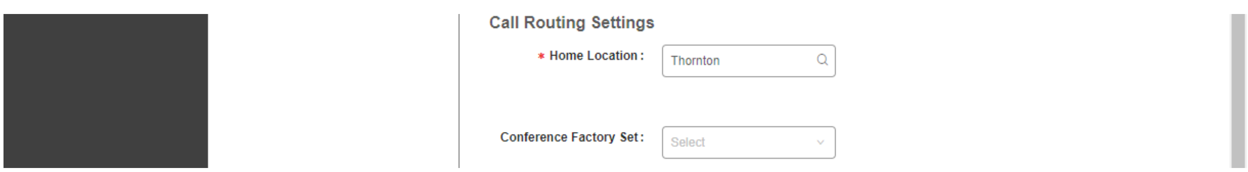


6.3.4. Session Manager Profile

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.

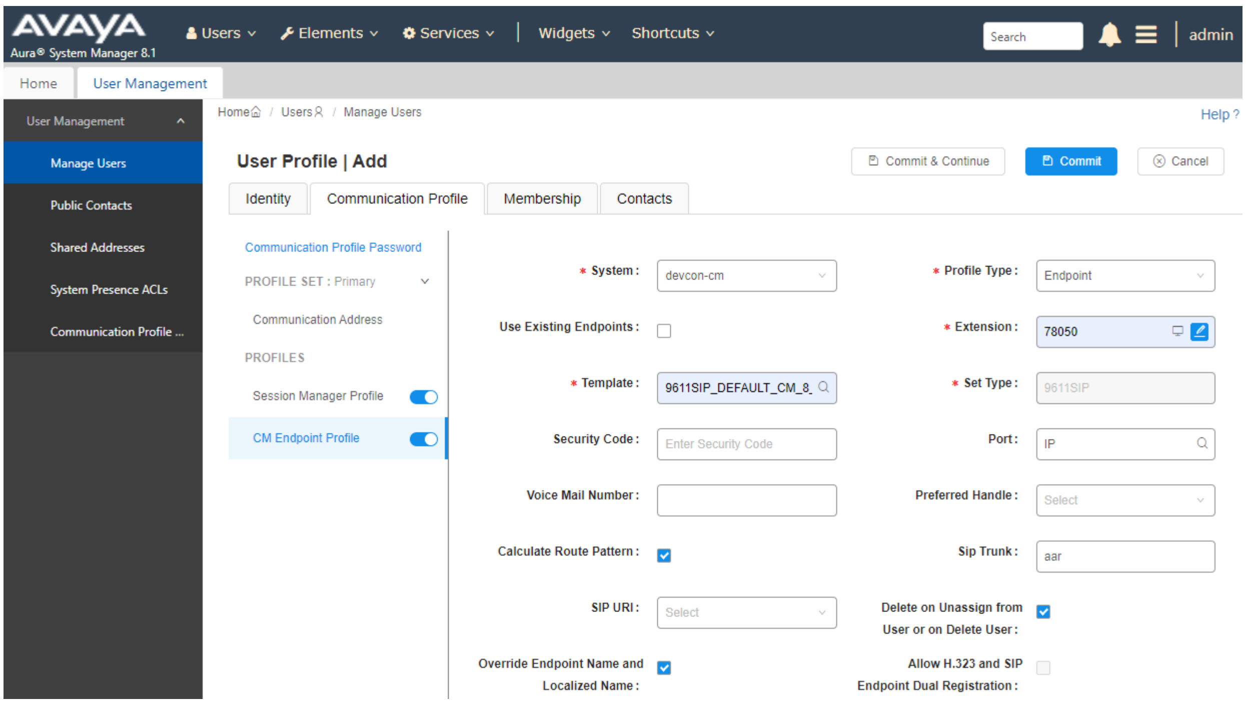


Scroll down to the **Call Routing Settings** section to configure the **Home Location**.

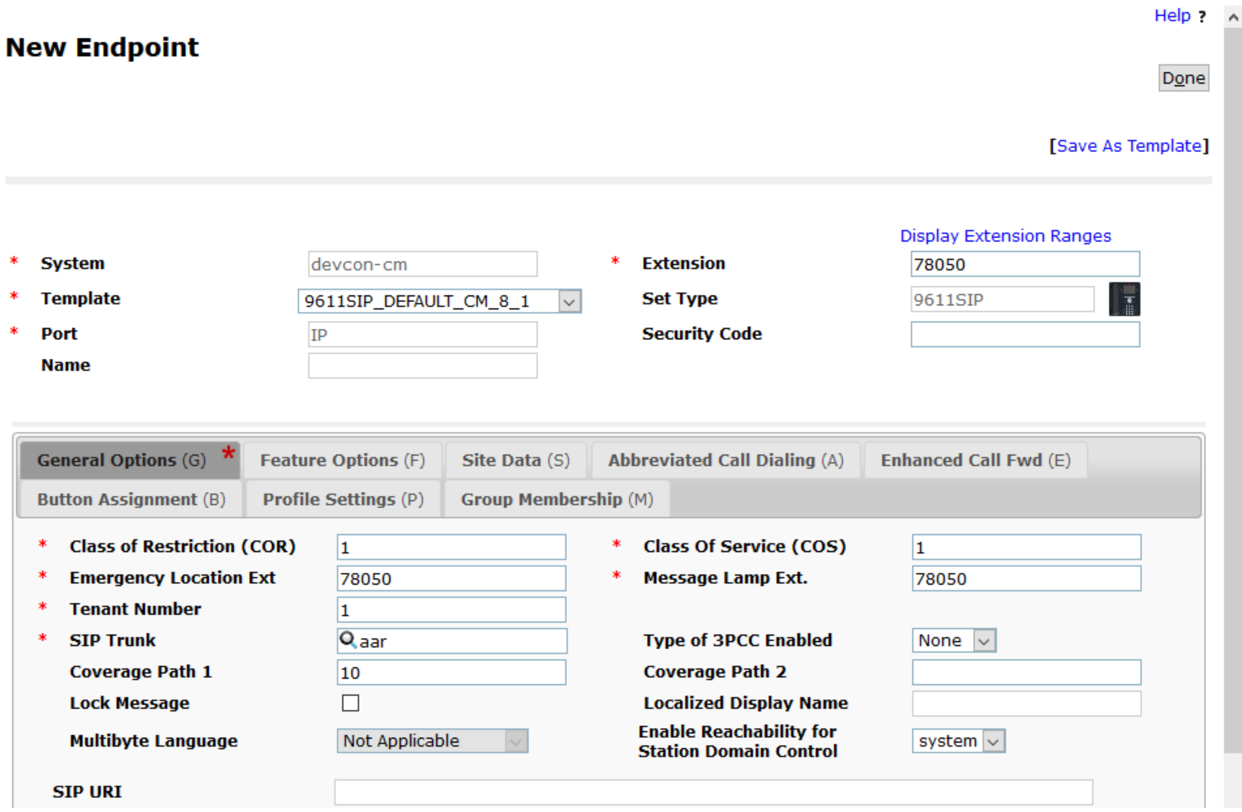


6.3.5. CM Endpoint Profile

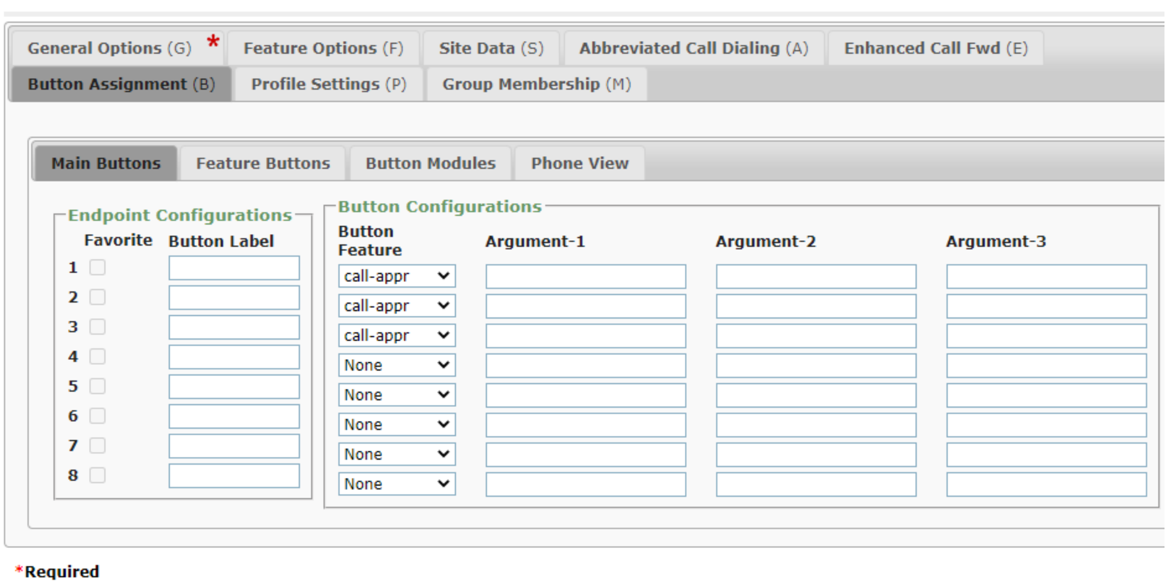
Click on the toggle button by **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.1**. For **Template**, select *9611SIP_DEFAULT_CM_8_1*. For **Port**, click and select *IP*. 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** and enabled bridged calls.



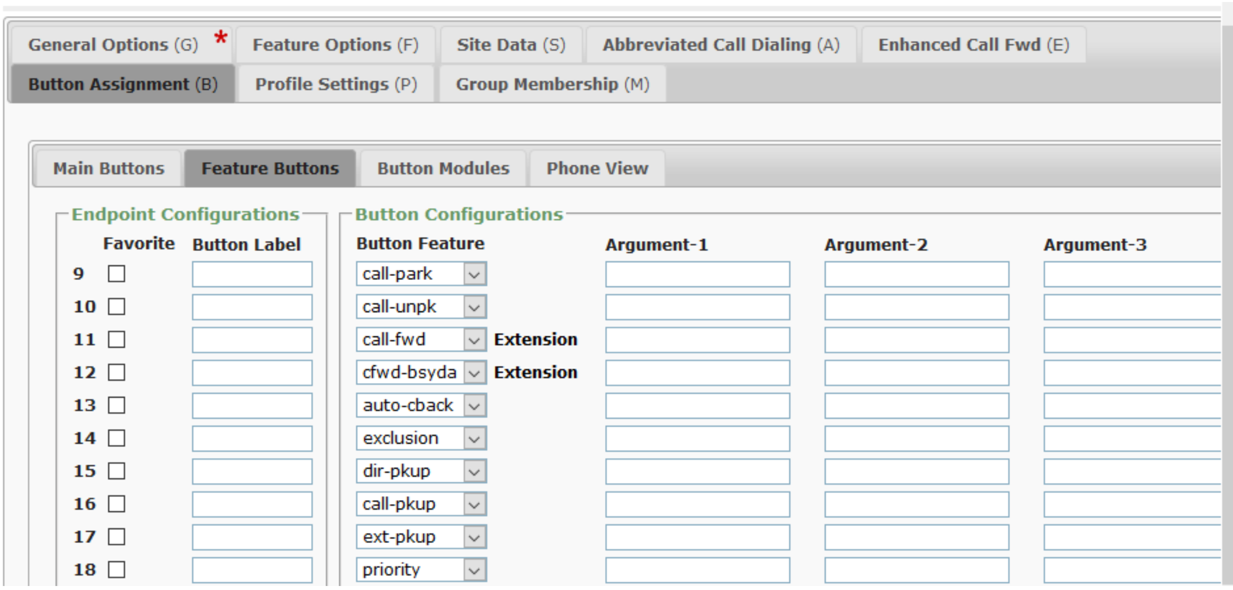
Navigate to the **General Options** tab and set the **Coverage Path 1** field to the voicemail coverage path. This provides voicemail coverage for the SIP user. In this example, coverage path *10* was used.



In the **Button Assignment** tab, select the **Main Buttons** sub-tab and accept the default settings with three call appearances.



In the **Button Assignment** tab, select the **Feature Buttons** sub-tab and configure the feature buttons below and listed in **Section 1**. As Avaya AST devices, Workforce Connect Voice Clients will be automatically configured with these feature buttons. Click **Done** (not shown) when complete, followed by **Commit** on the previous page.



7. Configure 46xxsettings.txt File

The Workforce Connect Voice Client retrieves the voicemail pilot number from the **MSGNUM** parameter in the **46xxsettings.txt** file that resides on a HTTP server. In the compliance test, the HTTP server IP address was 192.168.100.250, which was configured in the Workforce Connect Voice Clients in **Section 8**.

```
##
## Voice Mail Telephone Number
##   Specifies the telephone number to be dialed
##   automatically when the telephone user presses the
##   Messaging button.  The specified number is used to
##   connect to the user's Voice Mail system.
##
## Note 1: This parameter setting is ignored for extensions
##        configured as 96xx station types on the call server.
## Note 2: PSTN_VM_NUM shall be used with IP Office and 3PCC SIP environments instead
of MSGNUM.
##
SET MSGNUM 78500
```

8. Configure Zebra Workforce Connect Voice Client

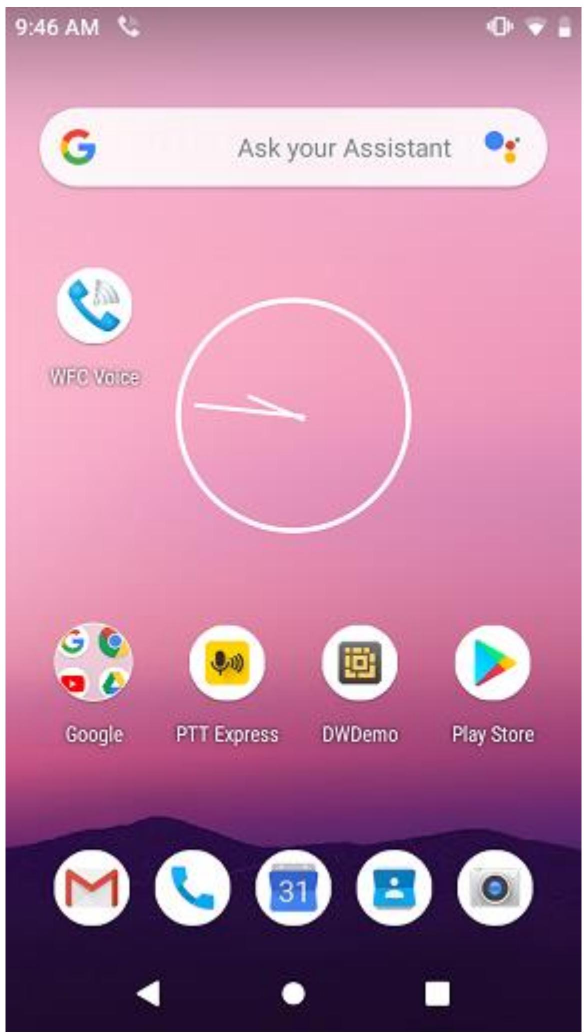
This section provides the procedures for configuring the Workforce Connect Voice Client for SIP connectivity to Session Manager.

Note: Connecting the Zebra TC51, TC56, and TC57 Touch Computers to the wireless network and configuring feature buttons on the Workforce Connect Voice Client are outside the scope of these Application Notes.

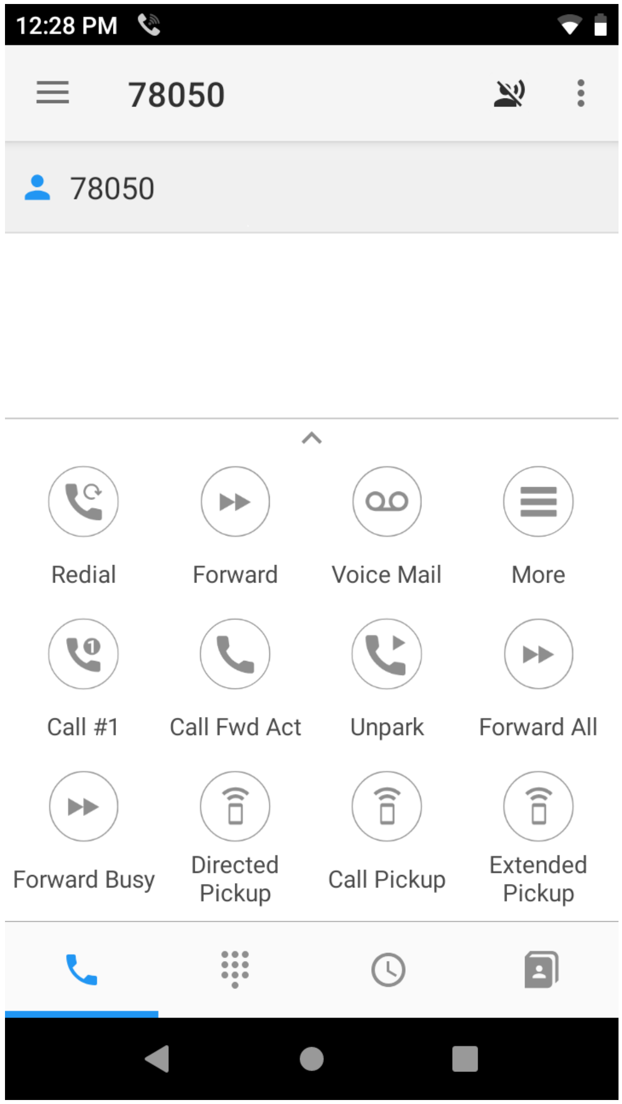
Power on the Workforce Connect Voice Client and unlock the Zebra touch computer.

A similar screen to the one below is displayed. Tap on the 

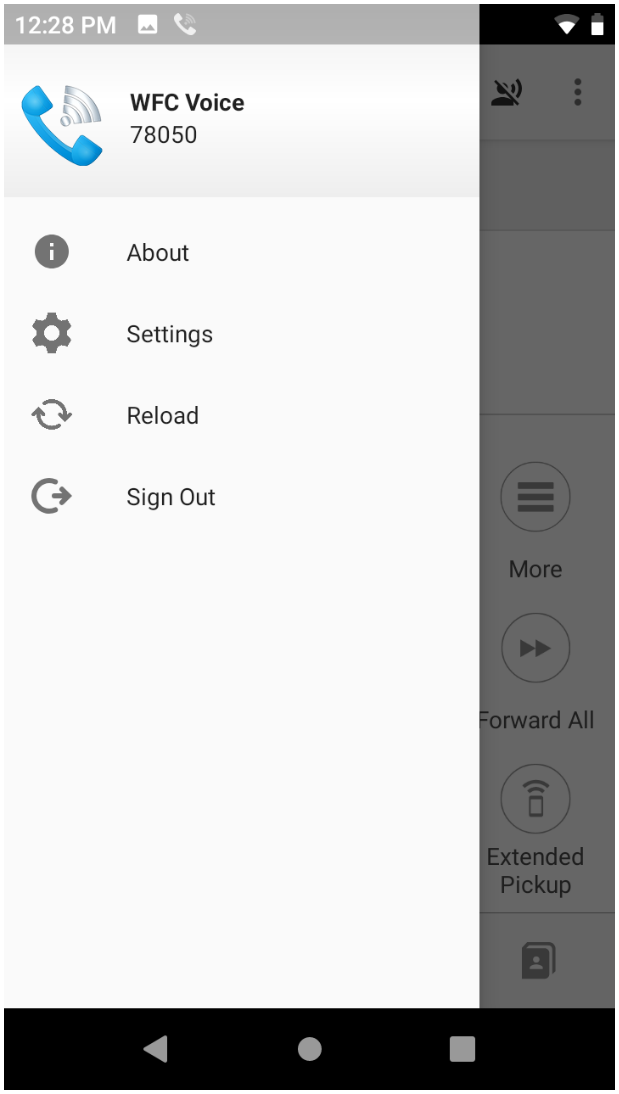
icon.



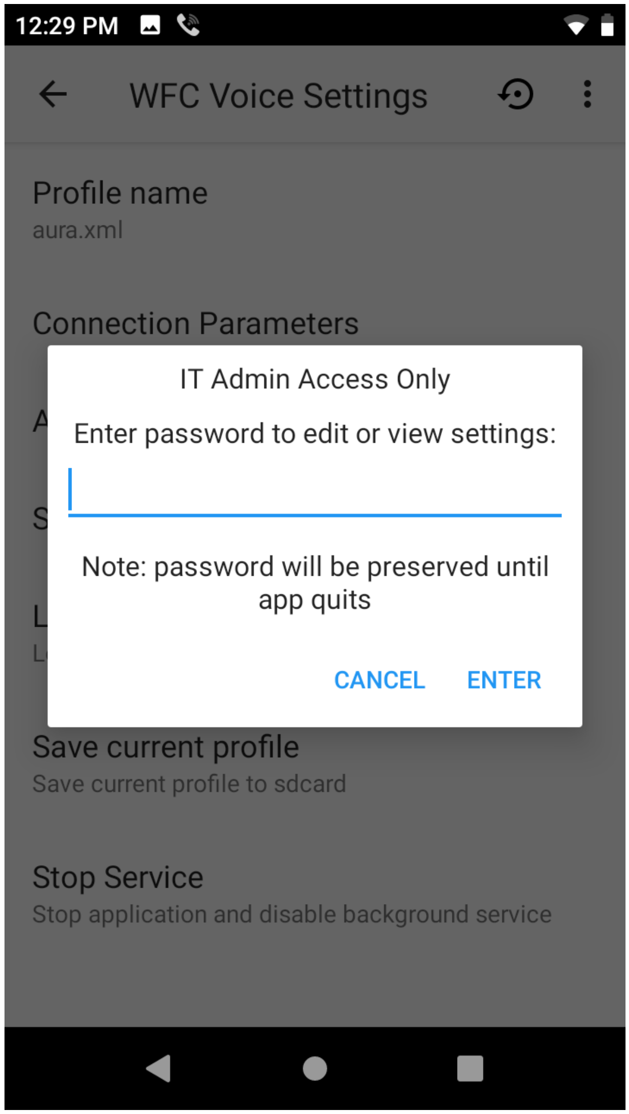
Tap on the **Menu** in the following screen.



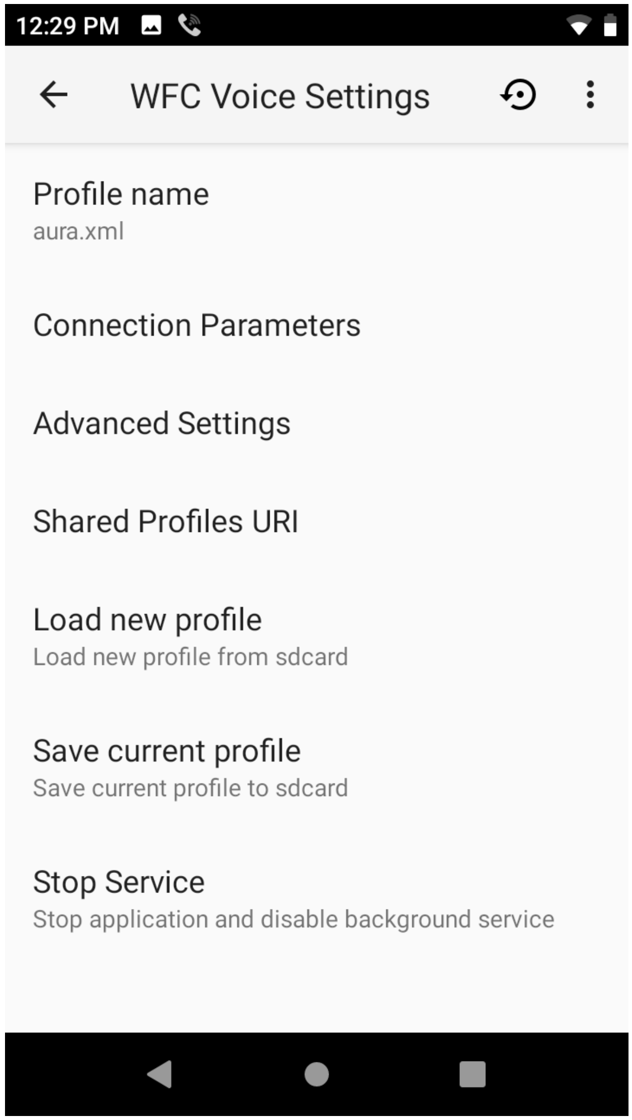
Select the **Settings** option from the pop-up menu shown below.



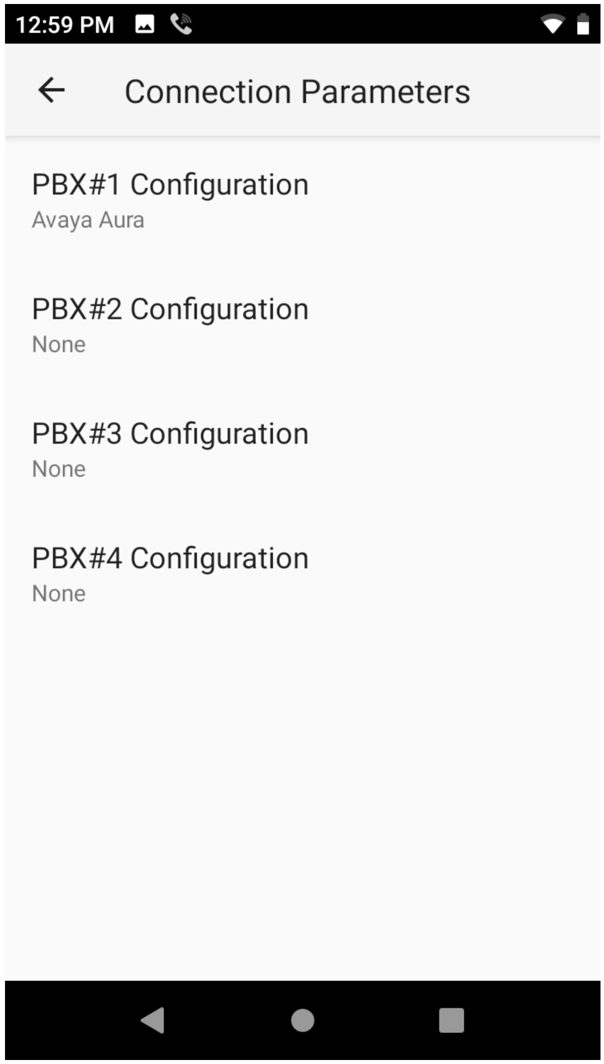
Log in with the appropriate credentials when prompted to access the **WFC Voice Settings**.



In the **WFC Voice Settings** shown below, **select Connection Parameters**. Note that **Profile name** specifies an .xml file that contains additional settings related to audio settings and feature buttons that can be customized further than described in these Application Notes. This file was in the **WFConnect** folder on the Android device.

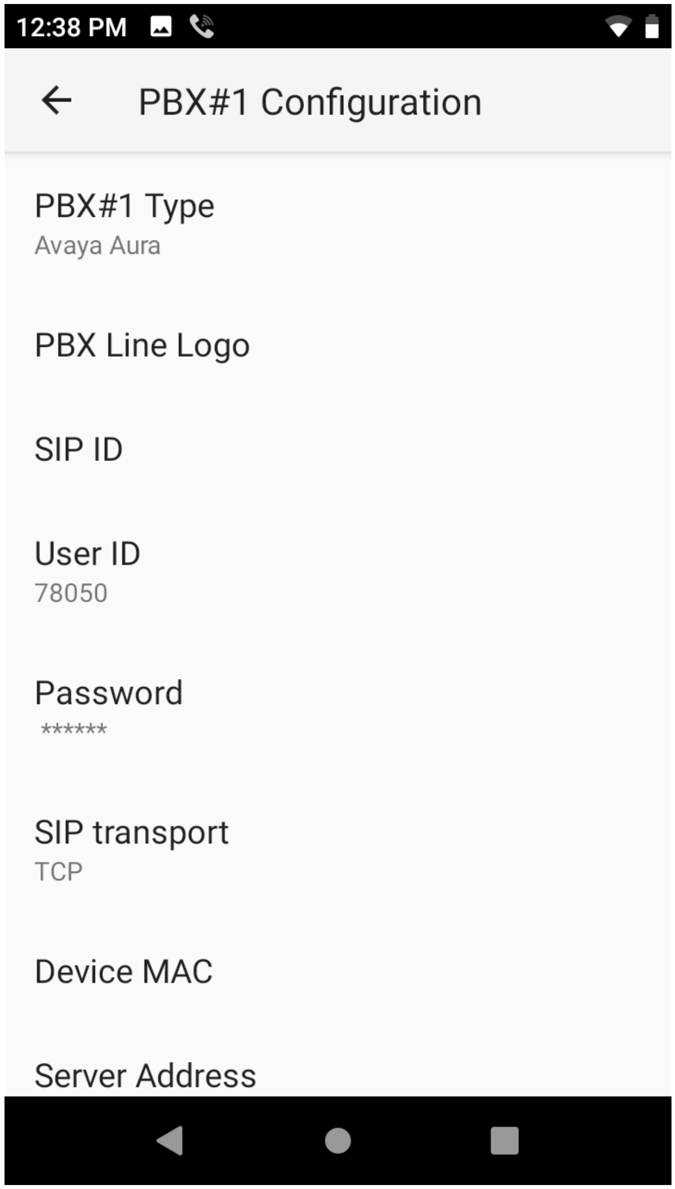


In the **Connection Parameters** shown below, select **PBX#1 Configuration**.



Under **PBX#1 Configuration**, configure the following parameters:

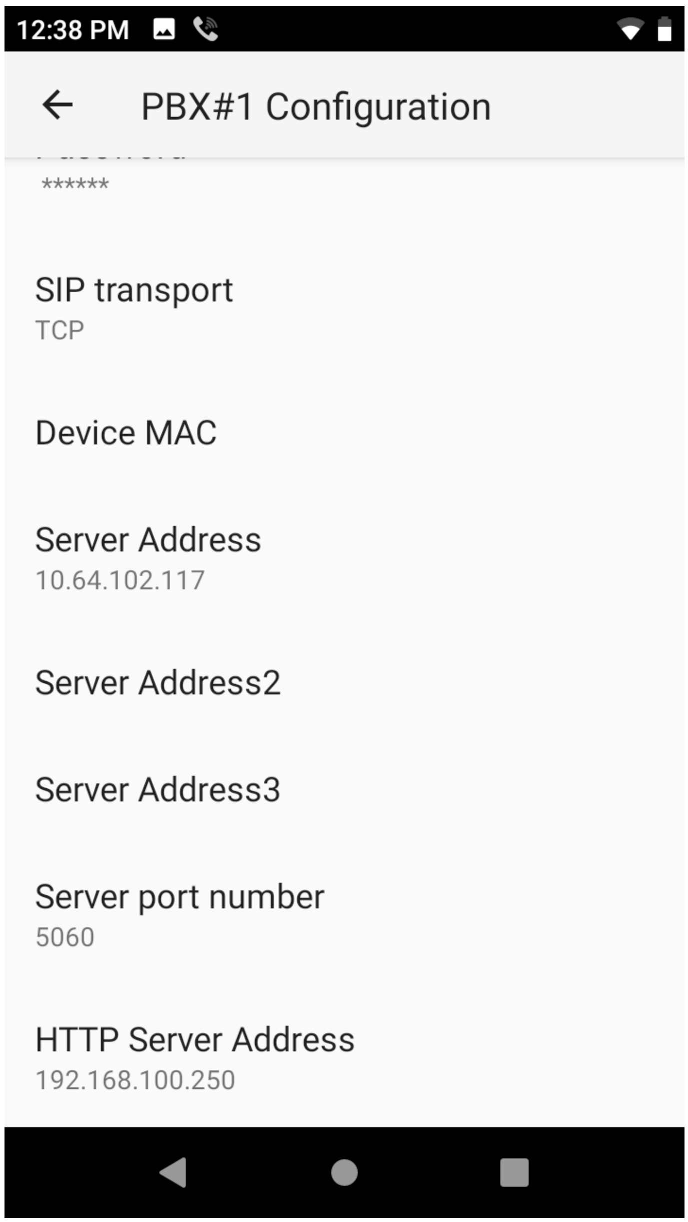
- **PBX#1 Type:** Set to *Avaya Aura*.
- **User ID:** Set to SIP extension (e.g., *78050*).
- **Password:** Set to SIP password configured in **Section 6.3.2**.
- **SIP Transport:** Set to *TCP* transport.



Scroll down and set the following parameters:

- **Server Address:** Set to IP address of Session Manager signaling interface (e.g., *10.64.102.117*).
- **Server port number:** Set to SIP port *5060*.
- **HTTP Server Address:** Set to the HTTP server IP address with the 46xxsettings.txt file to populate the voicemail pilot number. Refer to **Section 7**.

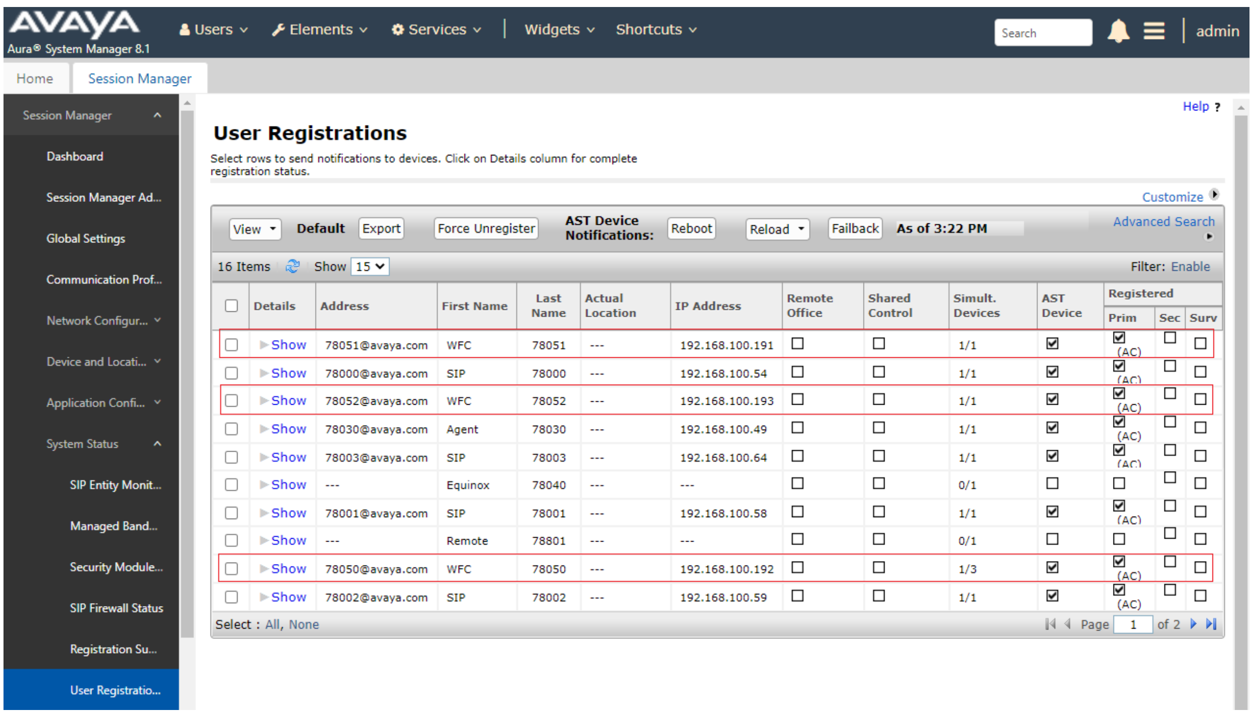
Return to the Workforce Connect Voice Client main screen to register client with Session Manager.



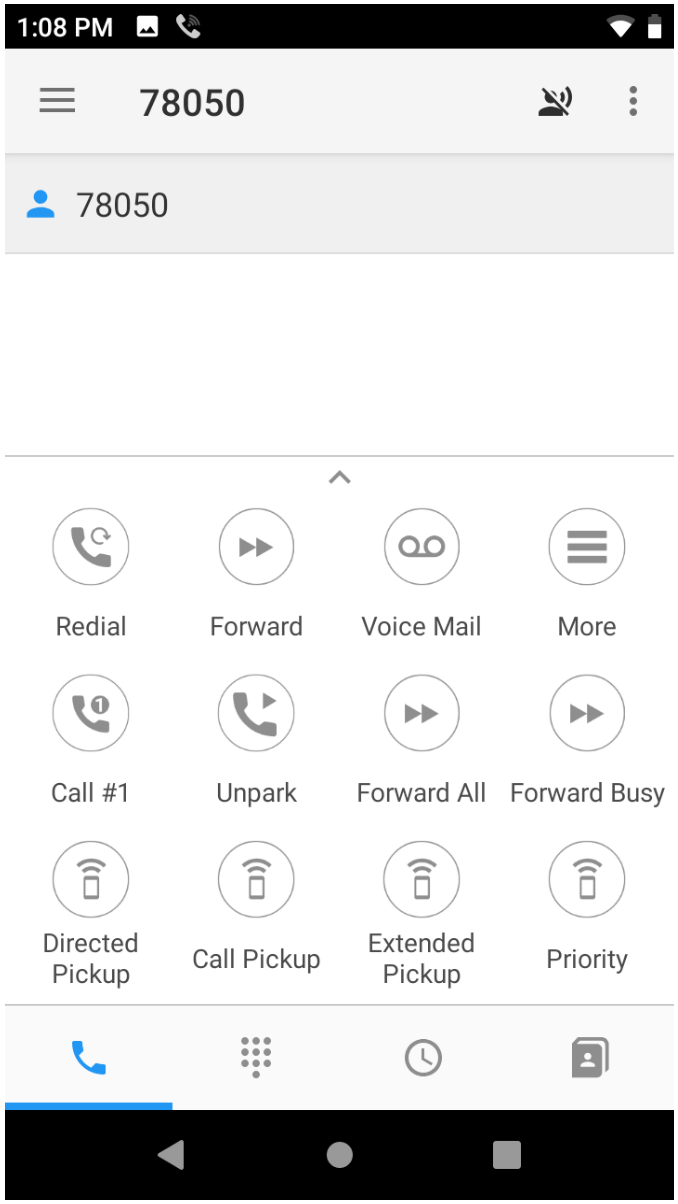
9. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and the Zebra Workforce Connect Voice Clients.

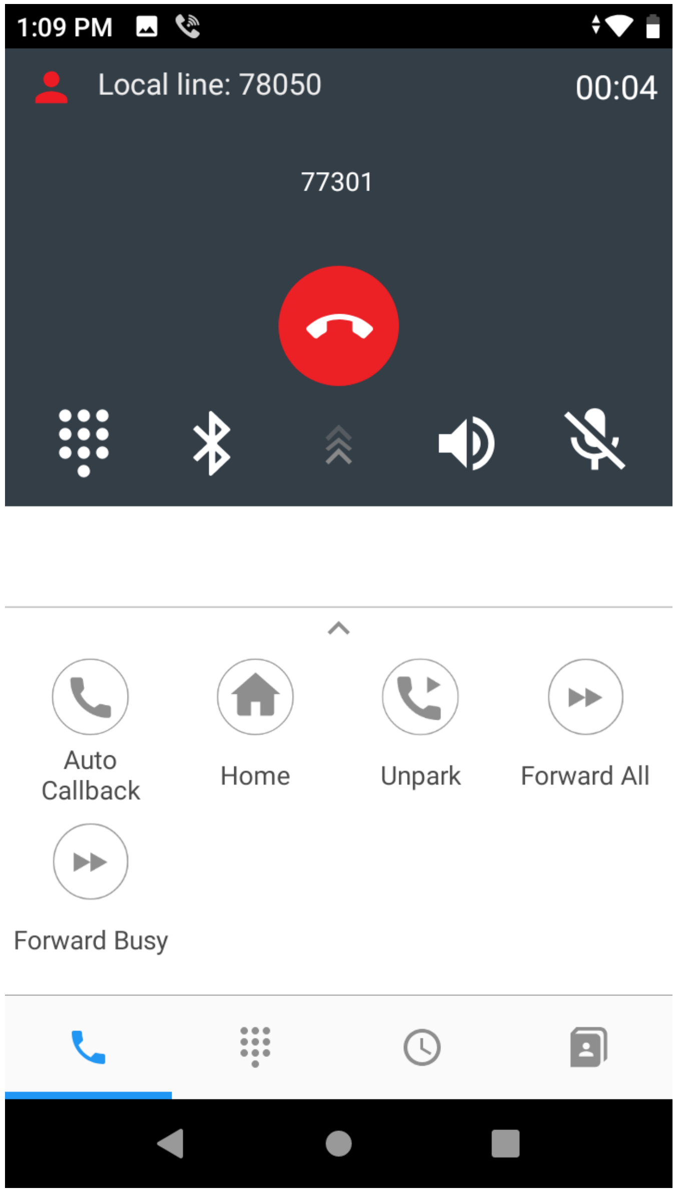
1. Verify that the Workforce Connect Voice Clients have successfully registered with Session Manager. In System Manager, navigate to **Elements → Session Manager → System Status → User Registrations** to check the registration status as shown below. Note that the Workforce Connect Voice Clients use Avaya AST to retrieve feature button information.



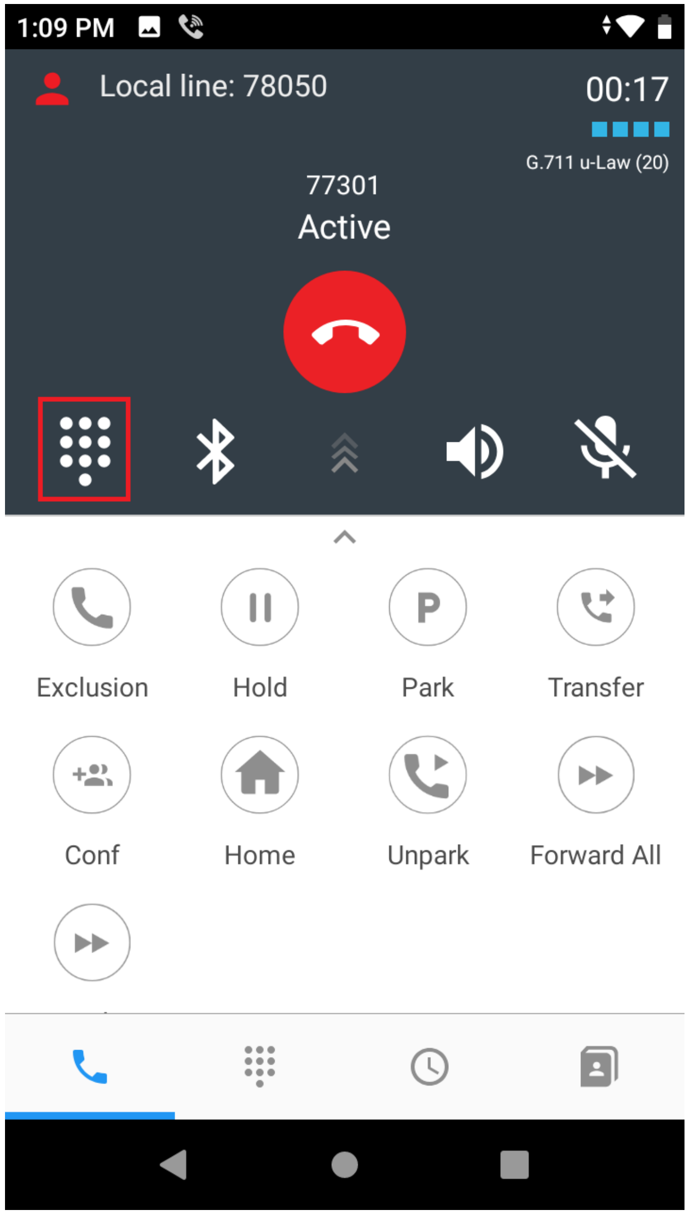
2. On an Android device, launch the Workforce Connect Voice Client and verify that the SIP extension has registered. When registered, it should display the SIP extension and feature buttons as shown below.



3. Select the keypad on the home screen above and place a call to another phone. While the call is ringing, the following feature buttons should be available.



4. Once the call is answered, the following feature buttons should appear. Use the dialer in the gray area in the red box to send DTMF. Verify two-way audio.



5. During the active call, verify basic telephony features with the Zebra Workforce Connect Voice Client.

10. Conclusion

These Application Notes describe the integration of the Zebra Workforce Connect Voice Clients with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Zebra Workforce Connect Voice Clients registered successfully with Avaya Aura® Session Manager as a SIP endpoint through an enterprise wireless LAN. Incoming and outgoing calls were placed to/from the Zebra Workforce Connect Voice Clients and telephony features were exercised using feature buttons and FACs. All test cases passed with observations noted in **Section 2.2**.

11. 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.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.