Avaya Solution & Interoperability Test Lab

Application Notes for Revolabs FLX UC 1000 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 Revolabs FLX UC 1000 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Revolabs FLX UC 1000 is a SIP conference phone that registers with Avaya Aura® Session Manager as a SIP endpoint in support of voice communications and enterprise conferencing.

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 Revolabs FLX UC 1000 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Revolabs FLX UC 1000 is a SIP conference phone that registers with Avaya Aura® Session Manager as a SIP endpoint in support of voice communications and enterprise conferencing. In the compliance test, Revolabs FLX UC 1000 successfully registered with Avaya Aura® Session Manager, established calls with other Avaya SIP and H.323 telephones, and exercised telephony features, such as hold, call transfer, 3-way conference, and call forwarding.

2 General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between Revolabs FLX UC 1000 and Avaya SIP and H.323 telephones and exercising basic telephony features, such as hold, mute, attended transfer and attended conference. Additional telephony features, such as call forward, call coverage, call park/unpark, and call pickup were also verified.

The serviceability testing focused on verifying that Revolabs FLX UC 1000 came back into service after re-connecting the Ethernet connect or rebooting the phone.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1 Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of FLX UC 1000 with Session Manager.
- Calls between FLX UC 1000 and Avaya SIP and H.323 telephones with Direct IP-IP Media (Shuffling) enabled and disabled.
- Support of G.711, G.722 and G.729 codecs.
- Proper recognition of DTMF tones.
- Basic telephony features including hold, mute, and redial. In addition, attended transfer and attended conference.
- Long call duration and long hold duration.
- Extended telephony features using Communication Manager Feature Access Codes (FACs) and Feature Name Extensions (FNEs) such as Call Forwarding, Call Park/Unpark and Call Pickup.
- Voicemail coverage, MWI support, and logging into voicemail system to retrieve voicemail messages.

2.2 Test Results

All test cases passed with the following observations noted:

- Revolabs FLX UC 1000 does not support blind transfer or blind conference. It supports attended transfer and attended conference.
- The G.729 codec was not always displayed as an option under the audio codecs supported by Revolabs FLX UC 1000.

2.3 Support

For technical support on the Revolabs FLX UC 1000, contact Revolabs Customer Support via phone, email, or website.

- **Phone:** (800) 326-1088
- **Web:** http://www.revolabs.com/support/product-line/uc-1000
- **Email:** support@revolabs.com

3 Reference Configuration

The network diagram shown in **Figure 1** illustrates a sample configuration with an Avaya SIP-based network that includes the following Avaya products:

- Avaya Aura® Communication Manager running on an Avaya S8300 Server with a G450 Media Gateway.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya Aura® Communication Manager Messaging serving as the voicemail system.
- Avaya 9600 Series SIP and H.323 Telephones.
- Revolabs FLX UC 1000 Conference Phones.

Revolabs FLX UC 1000 registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.

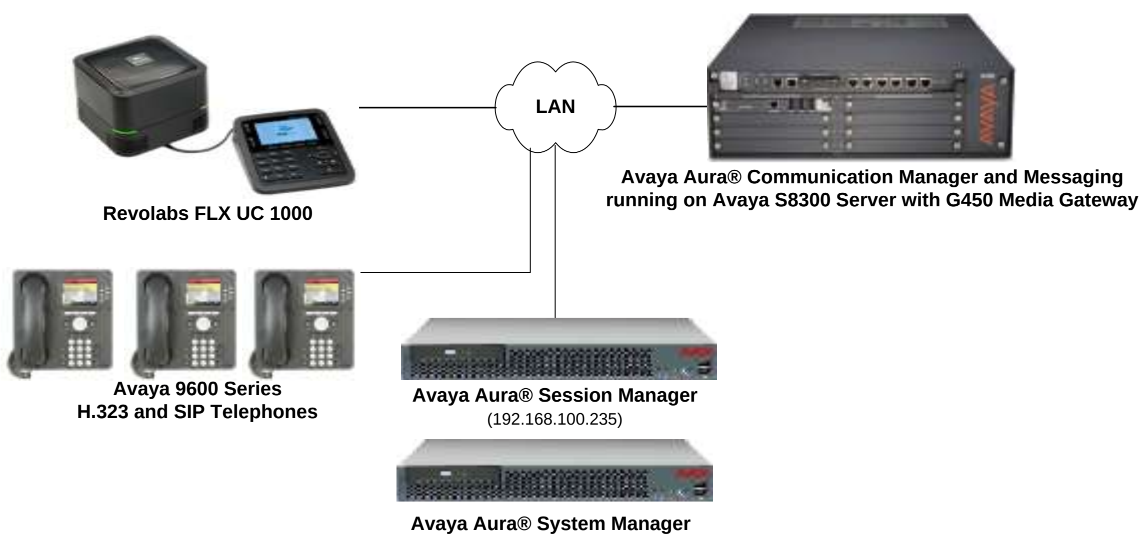


Figure 1: Revolabs FLX UC 1000 with Avaya SIP Telephony Network

4 Equipment and Software Validated

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

Equipment	Software
Avaya Aura® Communication Manager running on S8300 Server	6.3.9 SP 9.1 (R016x.03.0.124.0 with Patch 22098)
Avaya G450 Media Gateway	Gateway FW 36.12.0
Avaya Aura® Session Manager running on an S8800 Server	6.3.11.0.631103
Avaya Aura® System Manager	6.3.11 Build No. 6.3.0.8.5682-6.3.84751 Software Update Revision No: 6.3.11.8.2933
Revolabs FLX UC 1000	1.1.0.163

5 Configure Avaya Aura® Communication Manager

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

- Administer IP Network Region
- Administer IP Codec Set
- Administer SIP Stations

Communication Manager is configured through the System Access Terminal (SAT).

Note: It is assumed that basic configuration of Communication Manager has already been completed, including the SIP trunk to Session Manager. The SIP station configuration for revolabs FLX UC 1000 is configured through Avaya Aura® System Manager in **Section 6.3**.

5.1 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 *devcon.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G450 Media Gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: devcon.com
    Name:
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: 65535
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
```

5.2 Administer IP Codec Set

In the **IP Codec Set** form, specify the audio codec(s) supported for calls routed over the SIP trunk to FLX UC 1000. The form is accessed via the **change ip-codec-set 1** command. For the compliance test, G.711MU, G.729, and G.722 codecs were used. In the IP codec set form, specify the appropriate codec being used by FLX UC 1000. Below is the IP codec set configured for G.711 mu-law.

```
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:
 3:
```

5.3 Administer SIP Stations

The SIP station for FLX UC 1000 was configured through System Manager as described in **Section 6.3**. This section describes how the configuration would appear on Communication Manager. In the station form, the **Type** field should be set to *9630SIP*, an available IP **Port** will be allocated automatically, and a descriptive **Name** should be specified as shown below. For the compliance test, the SIP station was configured with extension 46201.

```
display station 46201                                           Page   1 of   6
                                     STATION

Extension: 46201                         Lock Messages? n             BCC: 0
     Type: 9630SIP                       Security Code:                TN: 1
     Port: S00058                      Coverage Path 1: 1             COR: 1
     Name: Revolabs, FLX UC 1000       Coverage Path 2:               COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19
                                                 Message Lamp Ext: 46201

        Display Language: english

          Survivable COR: internal
   Survivable Trunk Dest? y                           IP SoftPhone? n

                                                          IP Video? n
```

Configure the **Stations with Off-PBX Telephone Integration** form so that calls destined for a FLX UC 1000 conference phone are routed to Session Manager. On this form, specify the extension of the SIP endpoint and set the **Application** field to *OPS*. The **Phone Number** field is set to the digits to be sent over the SIP trunk. In this case, the SIP telephone extensions configured on Session Manager also match the extensions of the corresponding stations on Communication Manager. However, this is not a requirement. Finally, the **Trunk Selection** field is set to *aar*. This field specifies Auto Alternate Routing (AAR) routing. In this case, the **Trunk Selection** field would be set to *aar* to trigger AAR routing. Configuration of the **AAR Analysis** and **Route Pattern** forms would also be required.

```
change off-pbx-telephone station-mapping 46201                 Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number     Trunk       Config  Dual
 Extension                  Prefix                      Selection   Set     Mode
 46201             OPS           -     46201             aar        1
```

6 Configure Avaya Aura® Session Manager

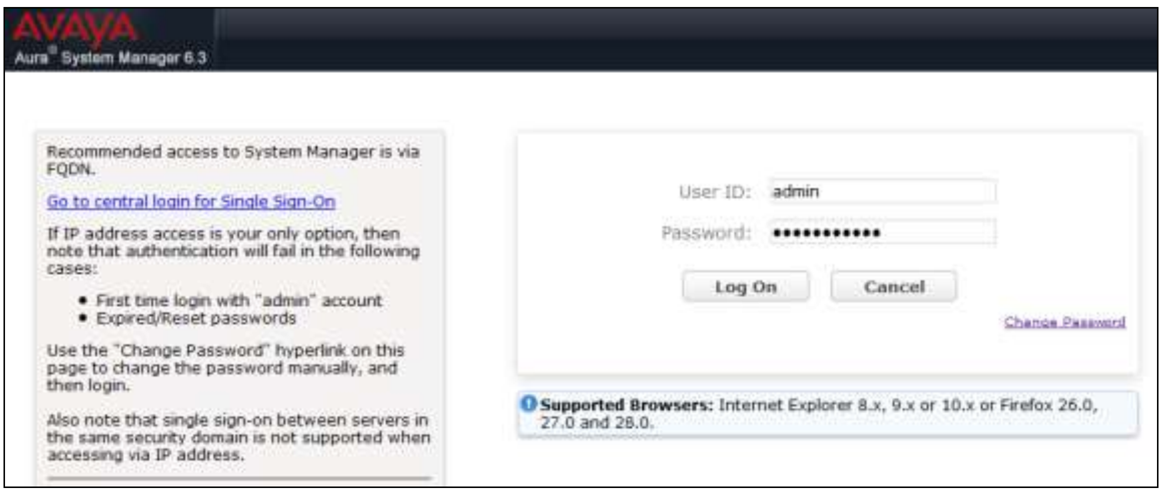
This section provides the procedures for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Specify the SIP port and transport protocol to be used by Revolabs FLX UC 1000
- 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 use for Revolabs FLX UC 1000 conference phone.

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.



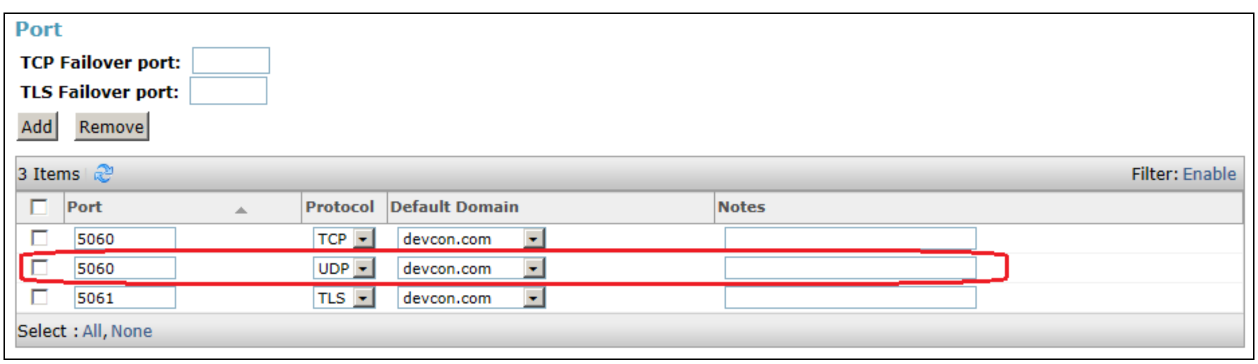
6.2 Specify SIP Port and Transport Protocol

In the Session Manager SIP Entity displayed below, scroll down to the **Port** section to configure the SIP port and transport protocol to be used by FLX UC 1000.

Under *Port*, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Port:** Port number on which the system listens for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain** The domain used for the enterprise (e.g., *devcon.com*).

Defaults can be used for the remaining fields. Click **Commit** (not shown) to save the SIP Entity definition.



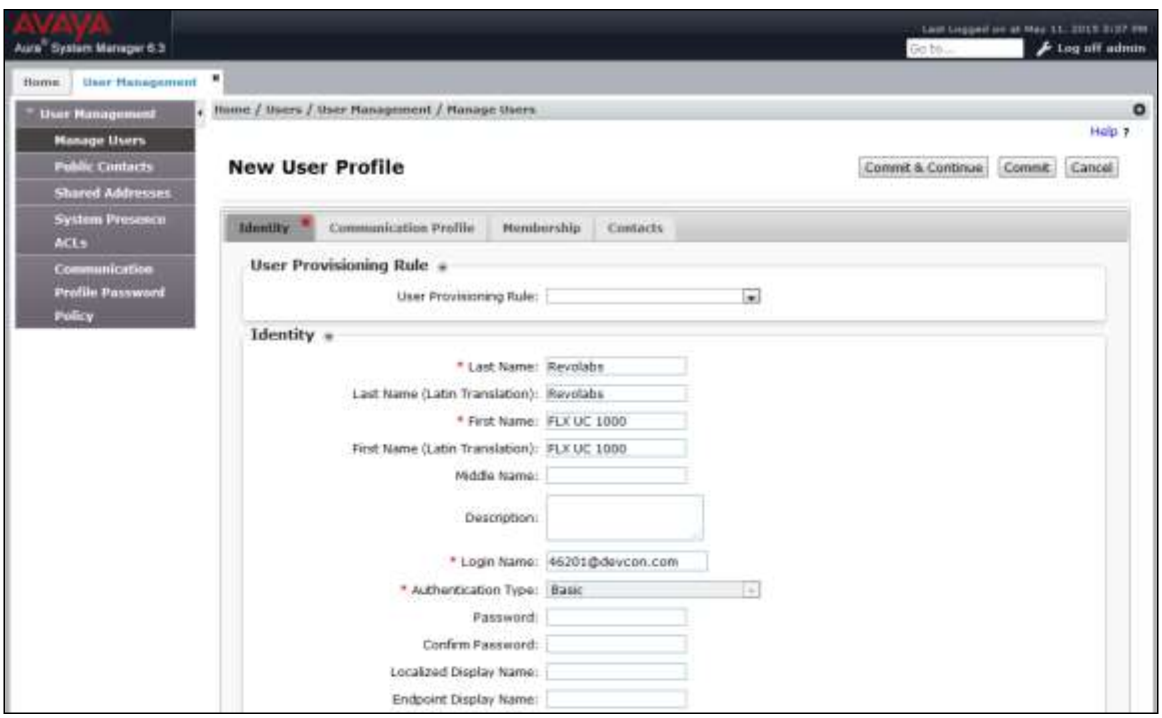
6.3 Add SIP User

Add the SIP user for FLX UC 1000 mentioned in **Section 5.3.** To add a new SIP user, navigate to **Users → User Management** from the main web page. From **User Management**, select **Manage Users** from the left pane from the left pane and then click **New** button (not shown) on the right.

Enter values for the following required attributes for a new SIP user in the **Identity** section of the new user form.

- **Last Name:** Enter the last name of the user.
- **First Name:** Enter the first name of the user.
- **Login Name:** Enter *<extension>@<sip domain>* of the user (e.g., *46201@devcon.com*).
- **Authentication Type:** Select *Basic*.

The screen below shows the information when adding a new SIP user to the sample configuration.



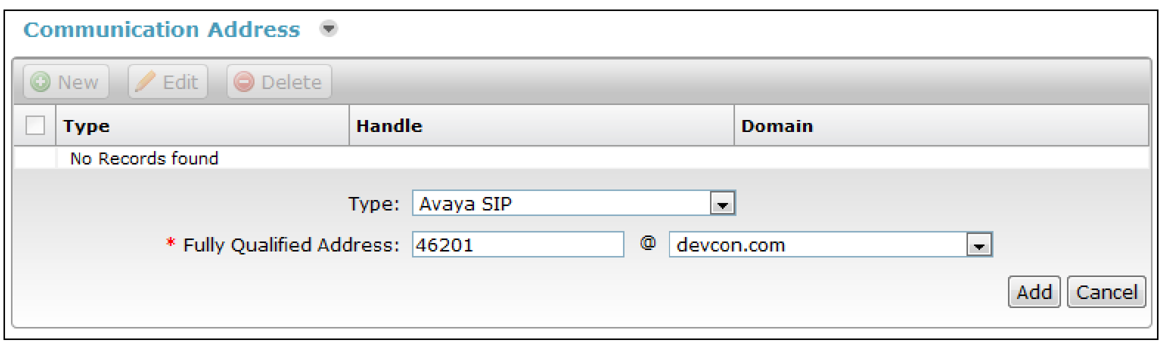
Select the **Communication Profile** tab and configure the following fields:

- **Communication Profile Password:** Enter the password which will used by FLX UC 1000 to log into Session Manager.
- **Confirm Password:** Re-enter the password from above.

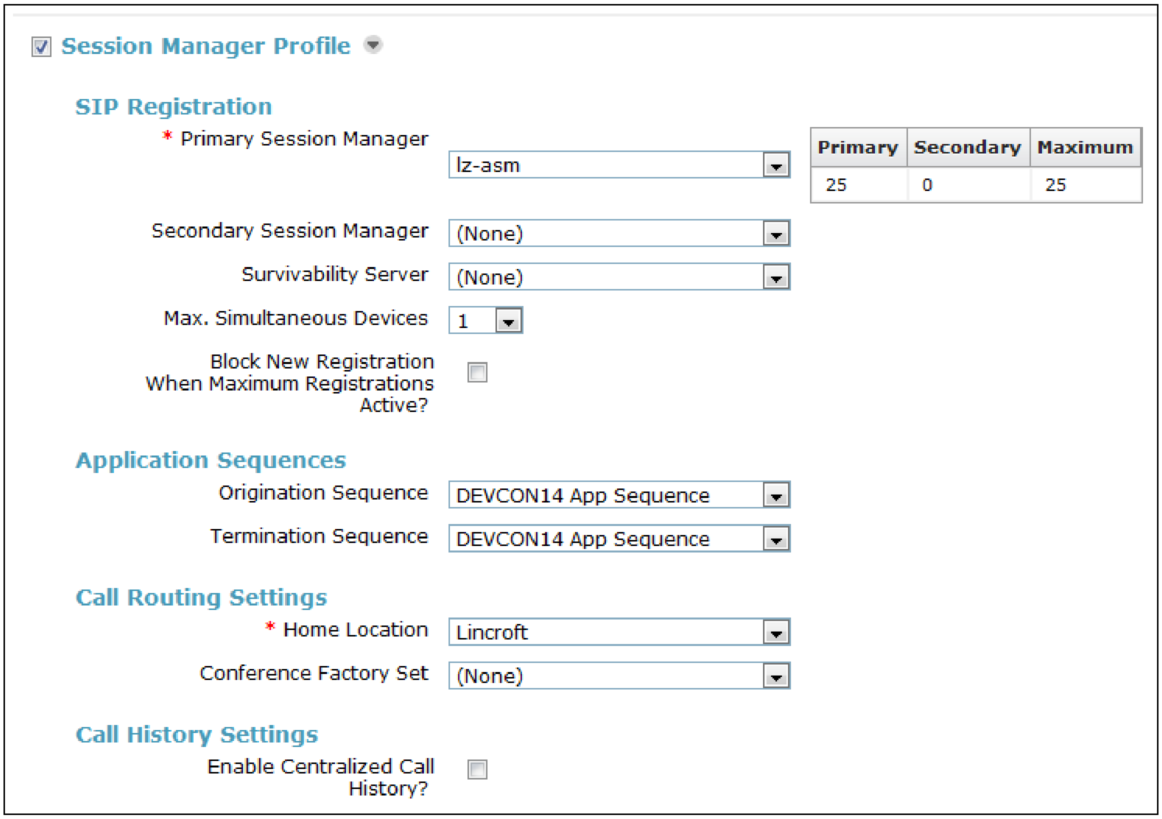
Click **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

- **Type:** Select *Avaya SIP*.
- **Fully Qualified Address:** Enter extension number and select SIP domain.

The screen below shows the information when adding a new SIP user to the sample configuration. Click **Add**.



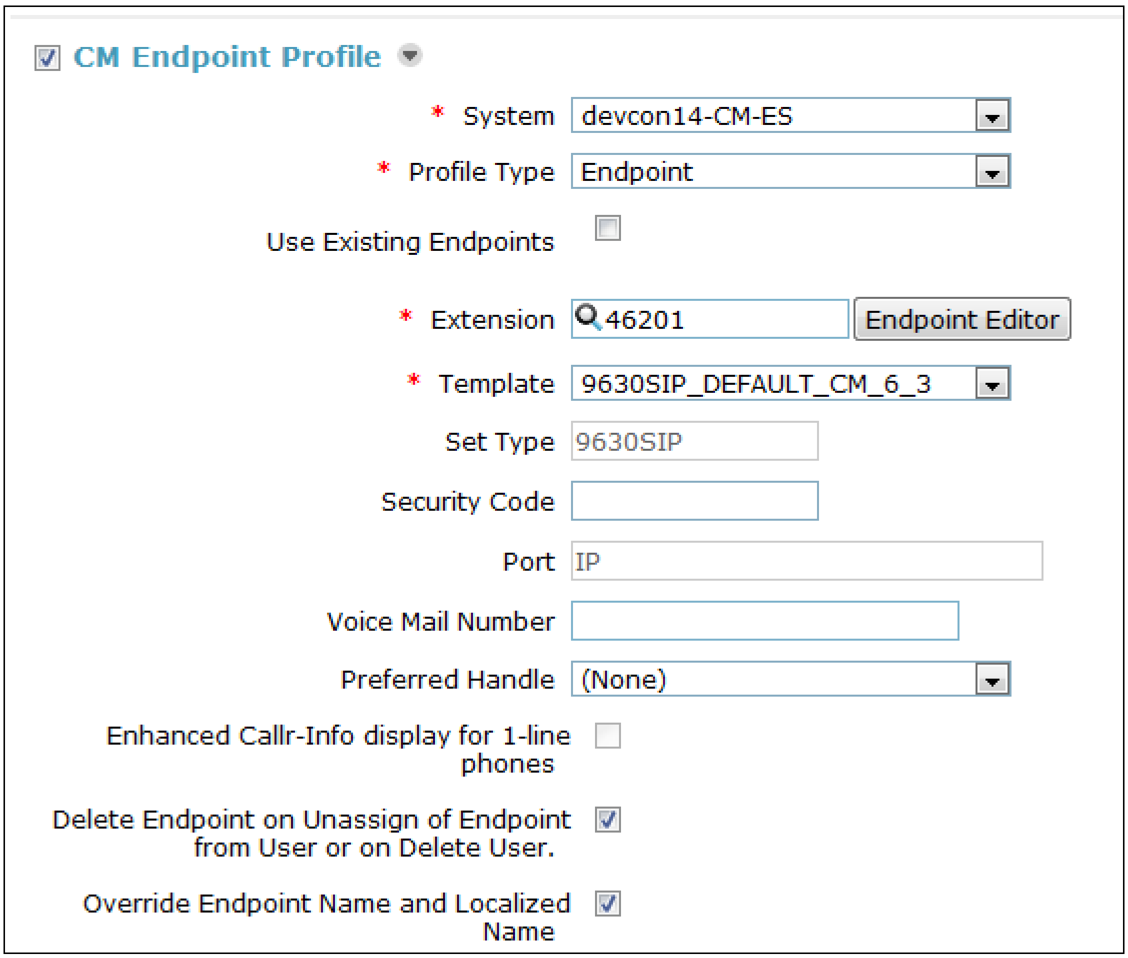
In the *Session Manager Profile* section, specify the Session Manager entity from **Section** Error! Reference source not found. for **Primary Session Manager** and assign the **Application Sequence** defined in **Section** Error! Reference source not found. to both the originating and terminating sequence fields. Set the **Home Location** field to the **Location** configured in **Section** Error! Reference source not found..



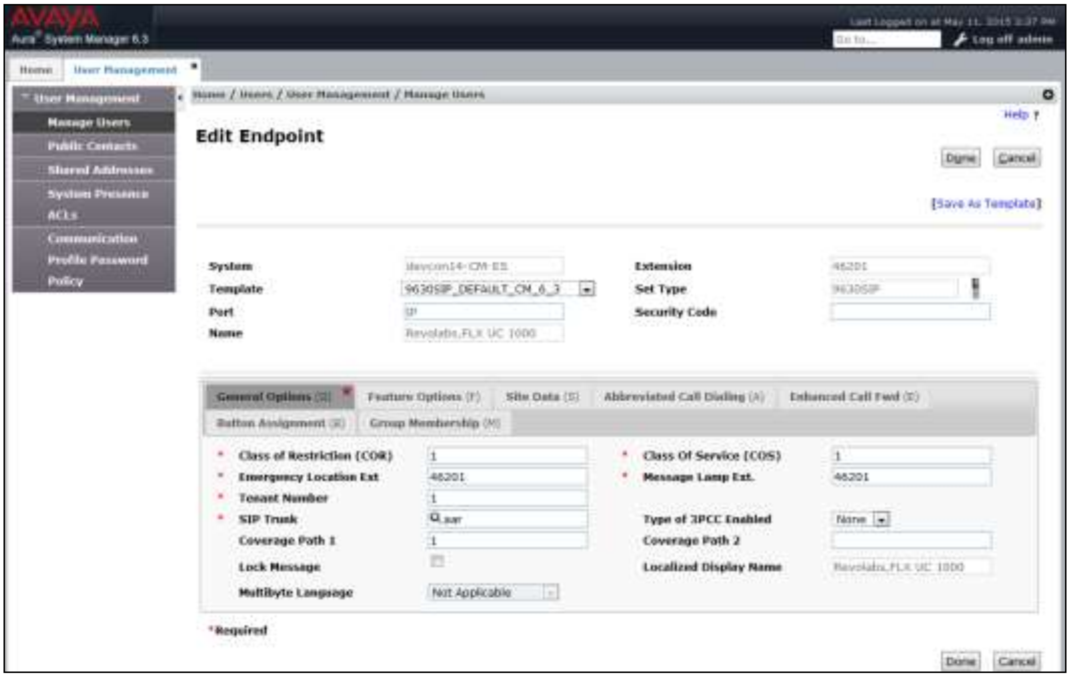
In the **CM Endpoint Profile** section, fill in the following fields:

- **System:** Select the managed element corresponding to Communication Manager.
- **Profile Type:** Select *Endpoint*.
- **Use Existing Stations:** If field is not selected, the station will automatically be added in Communication Manager.
- **Extension:** Enter extension number of SIP user.
- **Template:** Select template for 9630 SIP phone.
- **Port:** Enter *IP*.

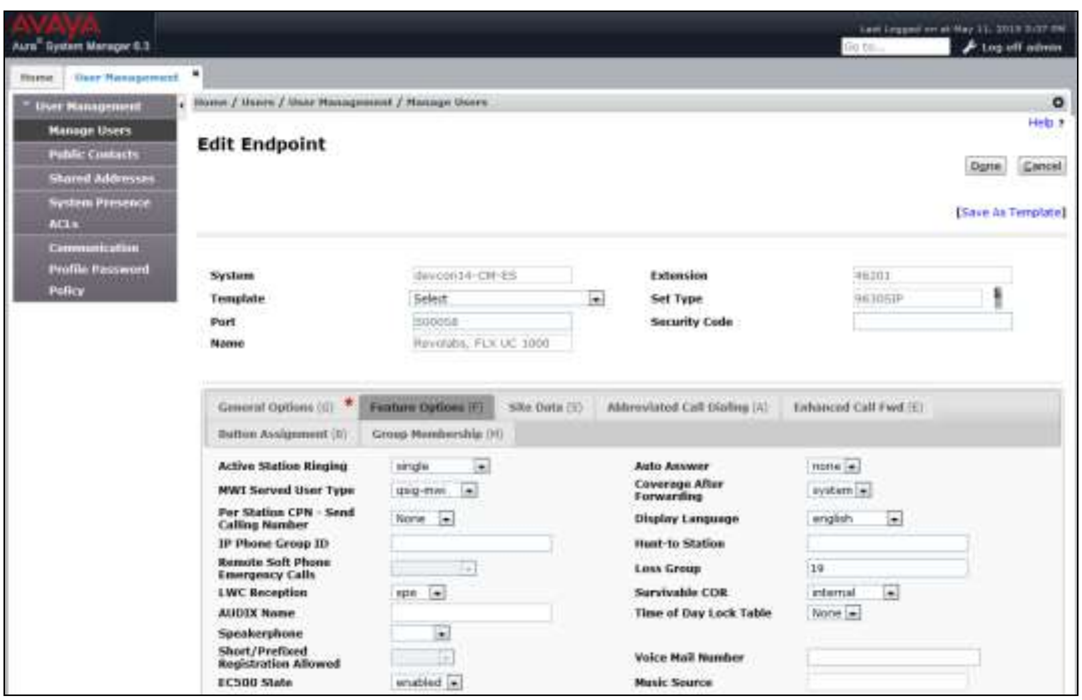
The screen below shows the information when adding a new SIP user to the sample configuration. Click **Commit** (not shown) to add the SIP user.



Under **CM Endpoint Profile**, click on the **Endpoint Editor** button to display the **Edit Endpoint** web page shown below. In the **General Options** tab, set **Coverage Path 1** to the voicemail coverage path.



In the **Feature Options** tab, set **MWI Served User Type** to the appropriate value, such as *qsig-mwi*. Click **Done**.



7 Configure Revolabs FLX UC 1000

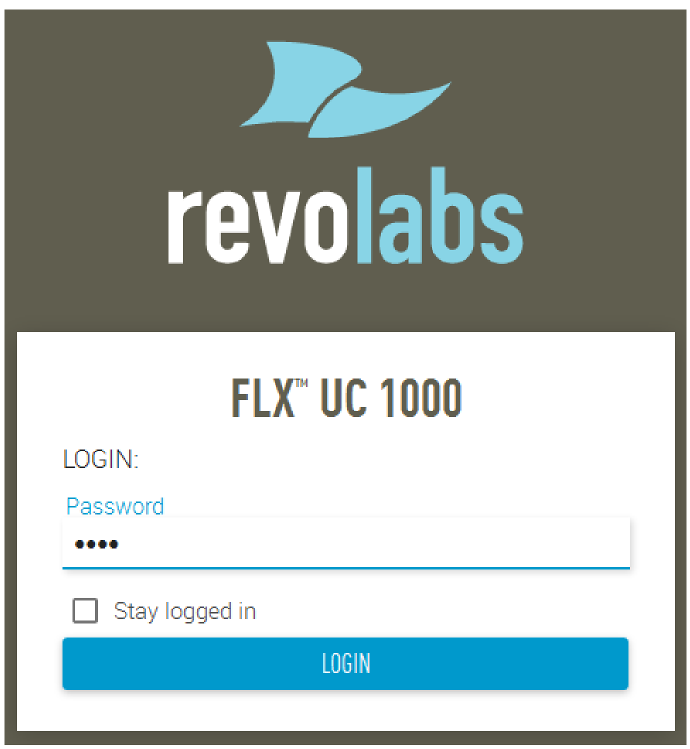
This section provides the procedures for configuring Revolabs FLX UC 1000. The procedures fall into the following areas:

- Launch web interface.
- Administer network settings.
- Administer SIP settings.
- Configure SIP Port and Transport Protocol.
- Enable Message Waiting Indicator (MWI).
- Configure Audio Codecs.

7.1 Launch Web Interface

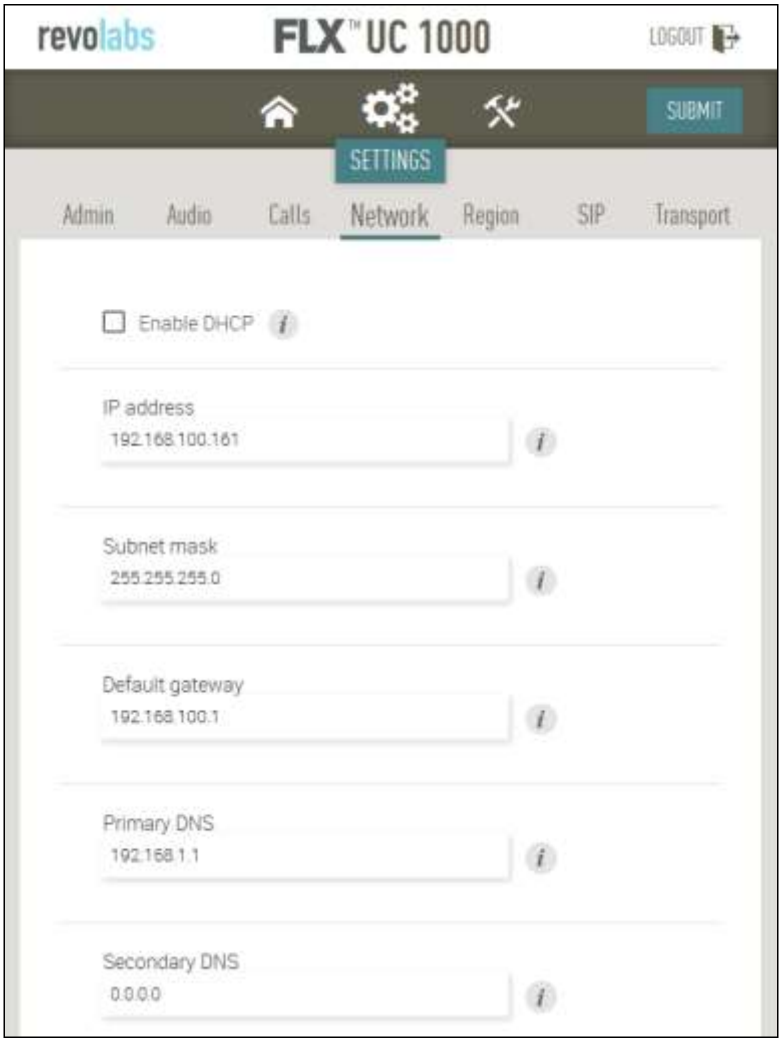
The Revolabs FLX UC 1000 may be configured directly from its keypad and display or using its web interface. For the compliance test, the configuration was performed through the web interface as described in this section. Access the Revolabs FLX UC 1000 web interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of the SIP conference phone. Log in using the appropriate credentials and then click **Enter**.

Note: Initially, the network parameters can be configured directly from the Revolabs FLX UC 1000 keypad and display as described in **[3].** After the IP address is changed to match the customer’s network, configuration can proceed through the web interface, if desired.



7.2 Administer Network Settings

To configure network settings, click on **Settings → Network** at the top of the web page. Disable DHCP and configure the **IP address**, **Subnet mask**, **Default gateway**, and **Primay DNS**. Click **Submit** when done. The **Primary DNS** is required so that FLX UC 1000 can reach the NTP servers.

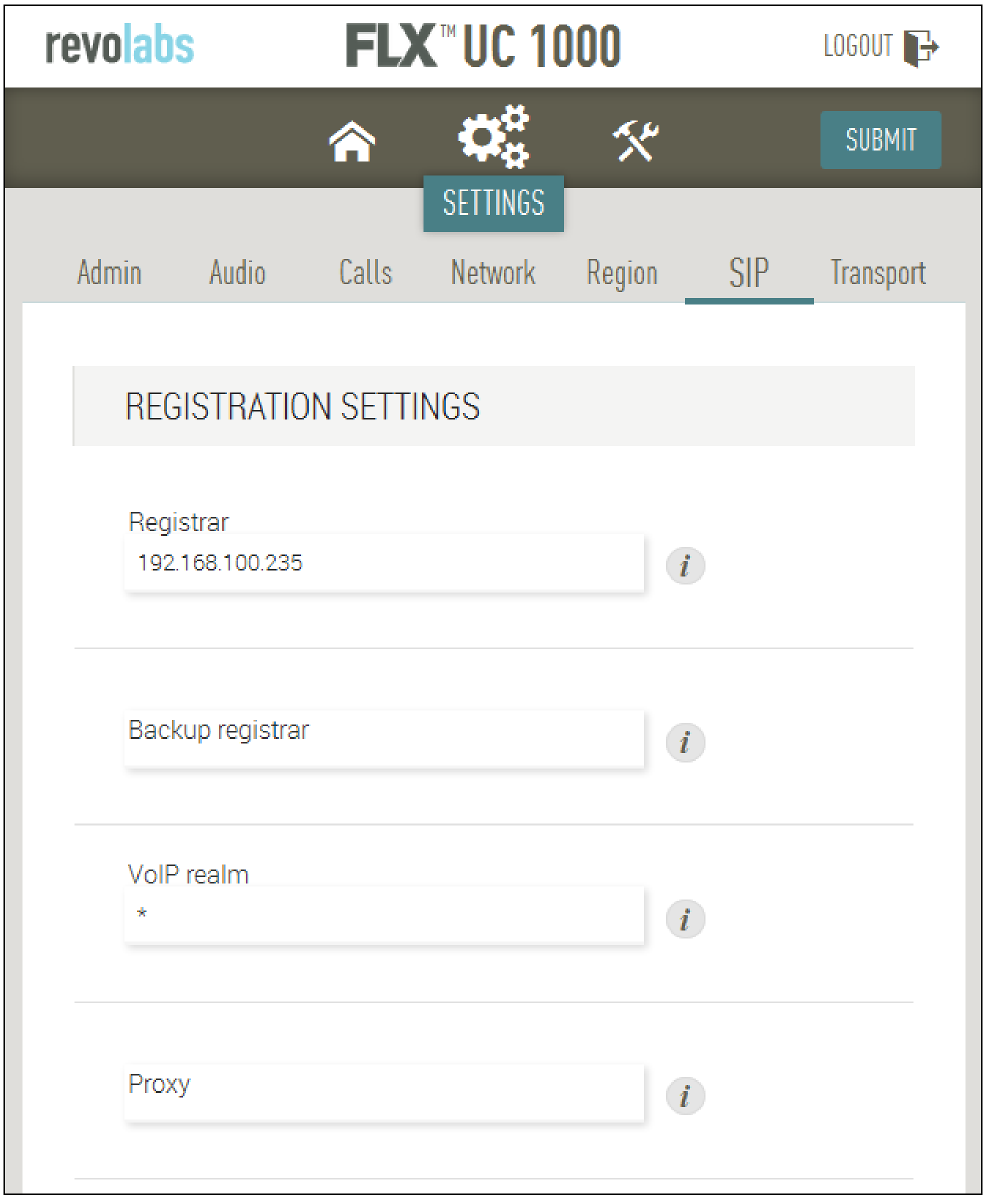


7.3 Administer SIP Settings

To configure SIP settings, click on **Settings** → **SIP** at the top of the web page. Configure the following fields:

- **Registrar** Enter the IP address of Session Manager signaling interface.
- **VoIP realm** Set to * so that FLX UC 1000 responds to calls from all domains.

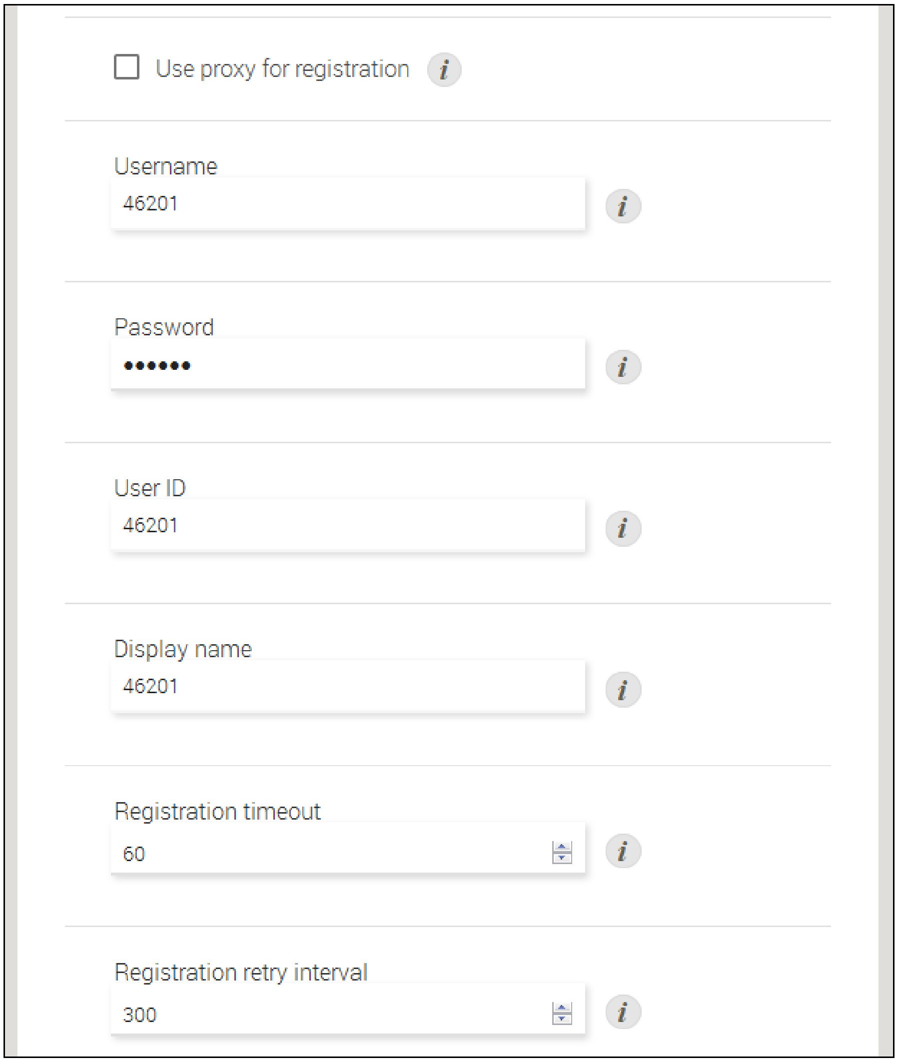
Retain the default values for the other fields.



Scroll down and set the following fields:

- **Username** Set to the SIP extension configured in **Section 6.3**.
- **Password** Enter the password configured in the **Communication Profile Password** field in **Section 6.3**.
- **User ID** Enter the SIP extension.
- **Display name** Enter the SIP extension.

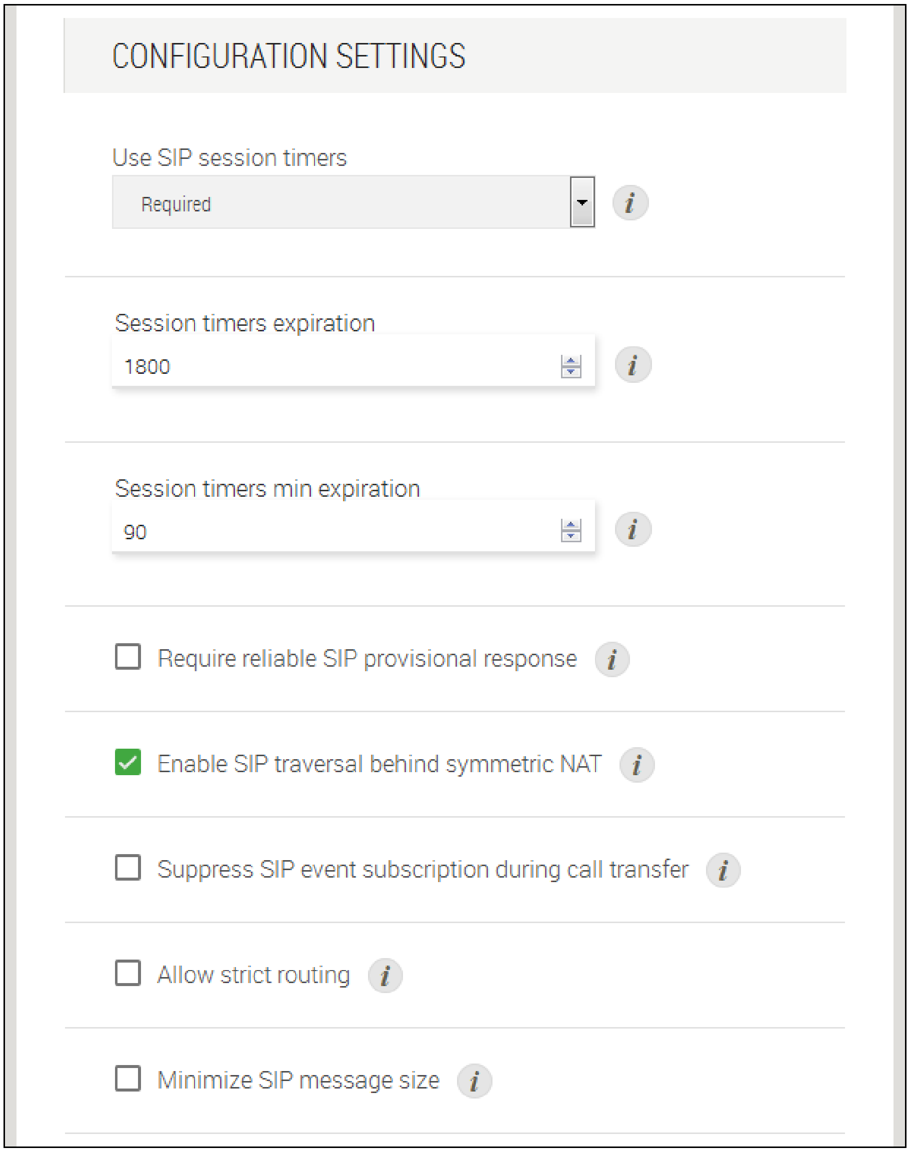
Retain the default values for the other fields.



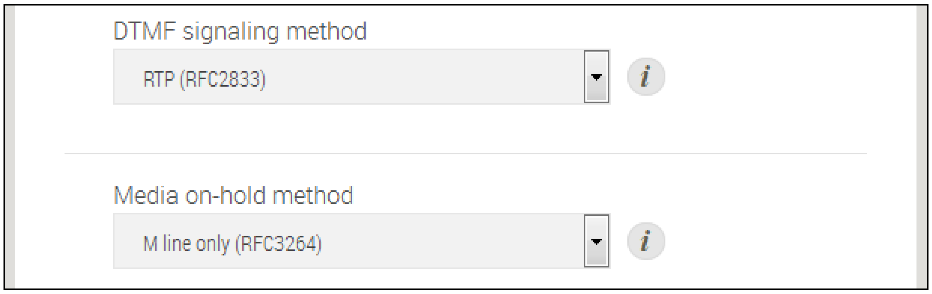
Scroll down to the Configuration Settings section and configure the following fields:

- **Use SIP session timers** Set this field to **Required**.
- **Session timers expiration** Enter the desired value for the session refresh interval.

Retain the default value for the other fields.

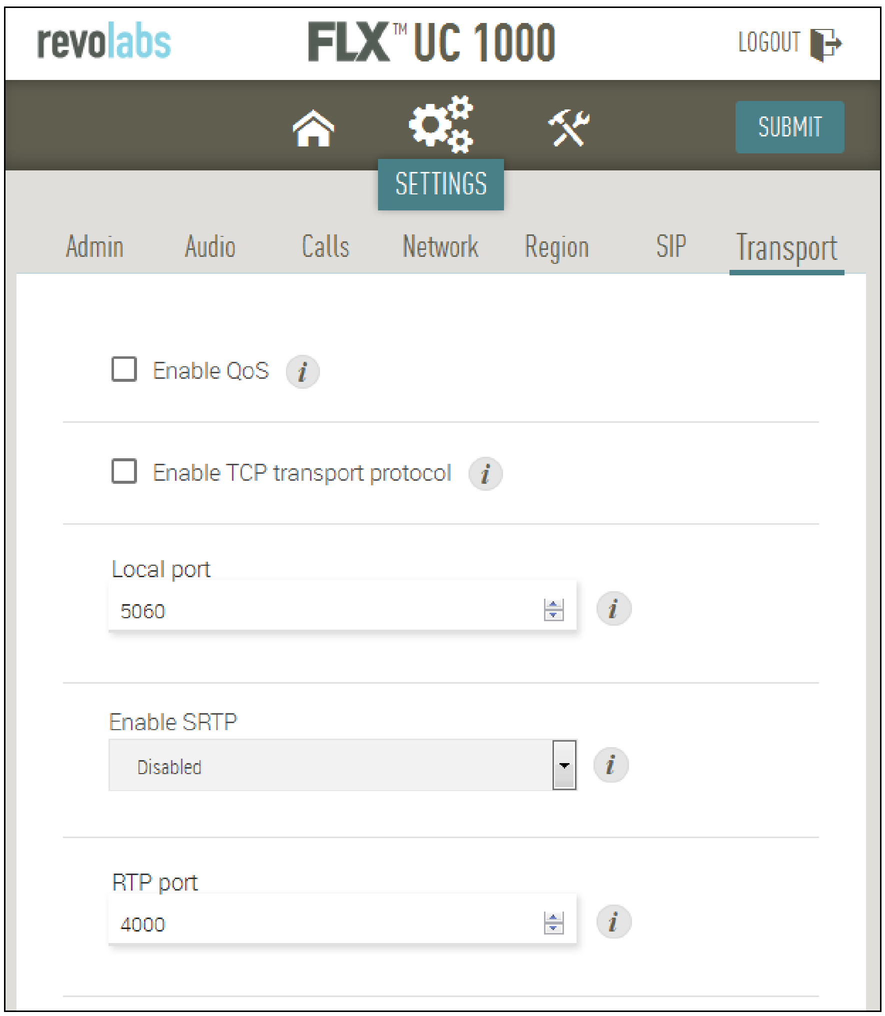


Set **DTMF signaling method** to *RTP (RFC2833)* as shown below. Click **Submit** (not shown).



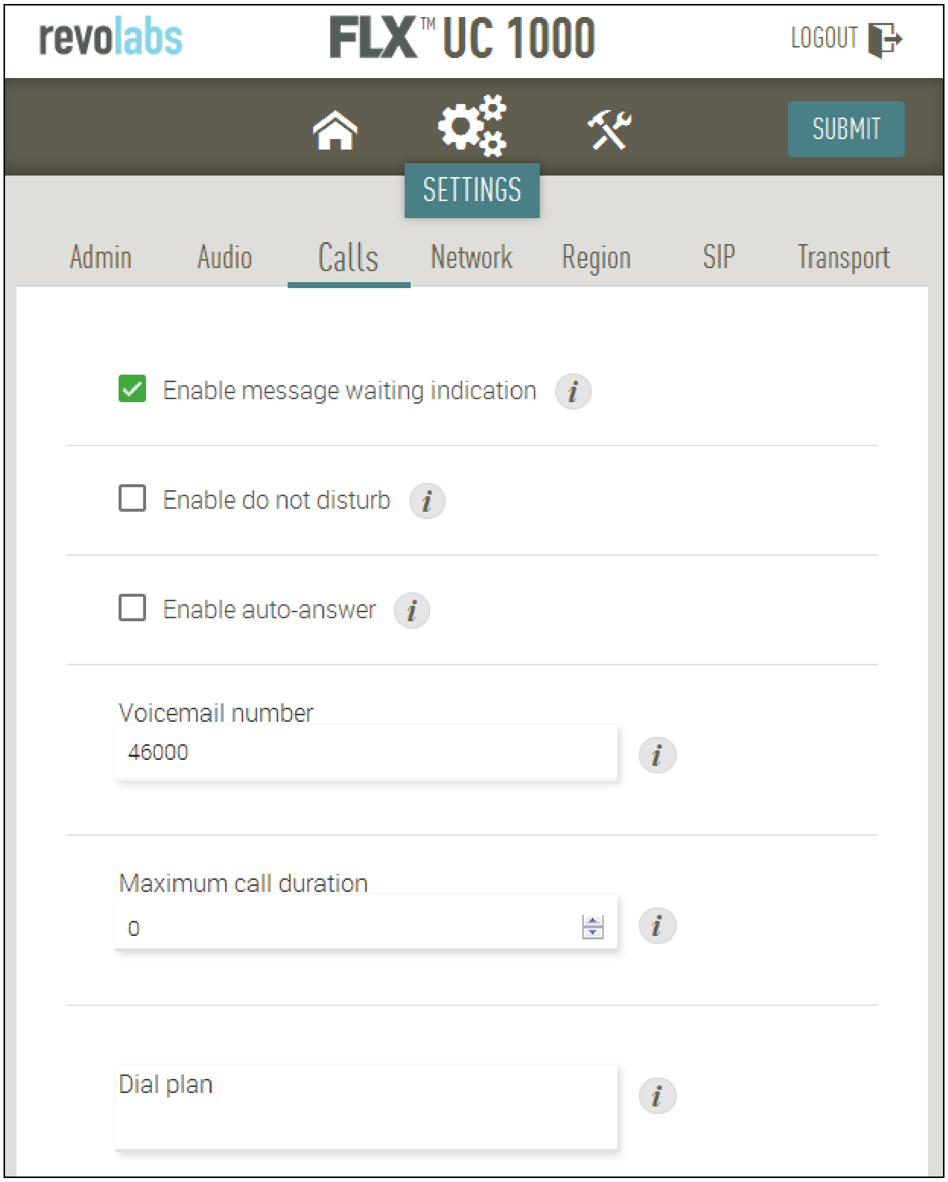
7.3.1 Configure SIP Port and Transport Protocol

Under **Settings → Transport**, configure the SIP port and transport protocol. The **Local port** is set to *5060*, and by default, the transport protocol is set to UDP, unless TCP is enabled below. These are the default values.



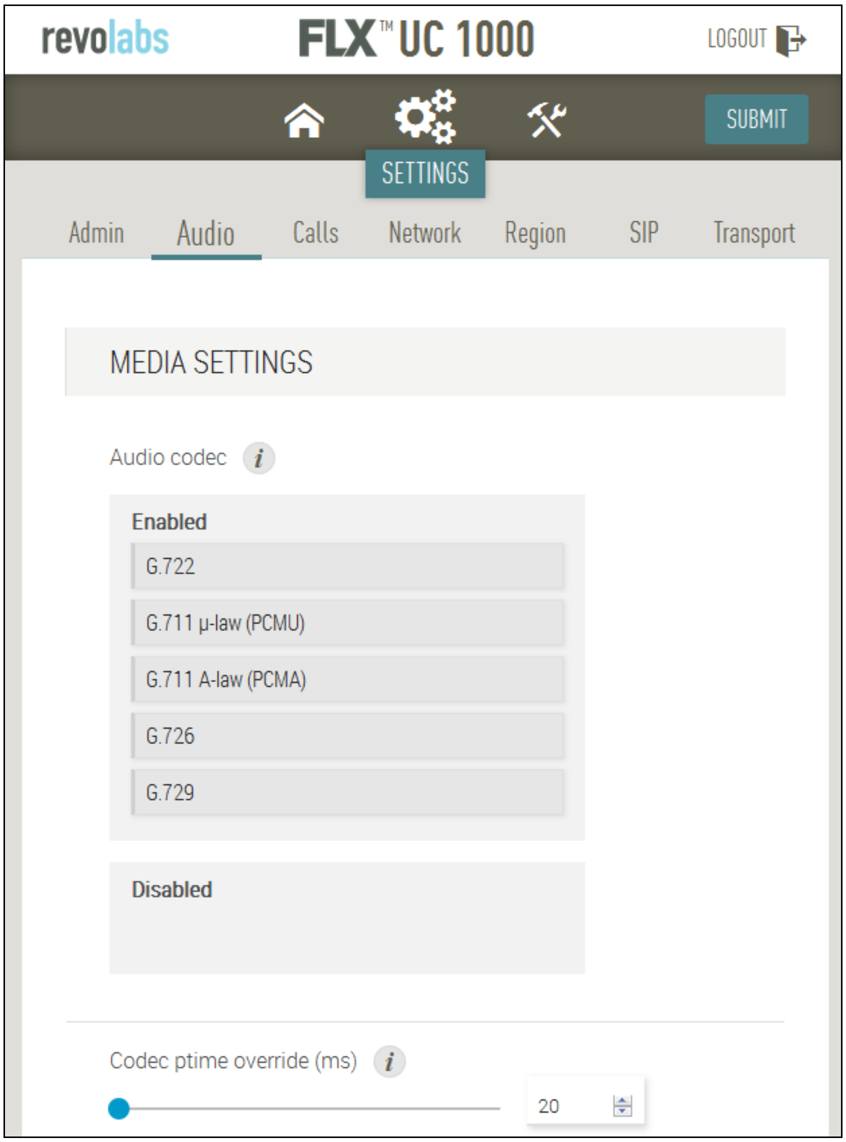
7.4 Enable Message Waiting Indicator (MWI)

Enable MWI under **Settings → Calls** and set the **Voicemail number** as shown below.



7.5 Administer Media Settings

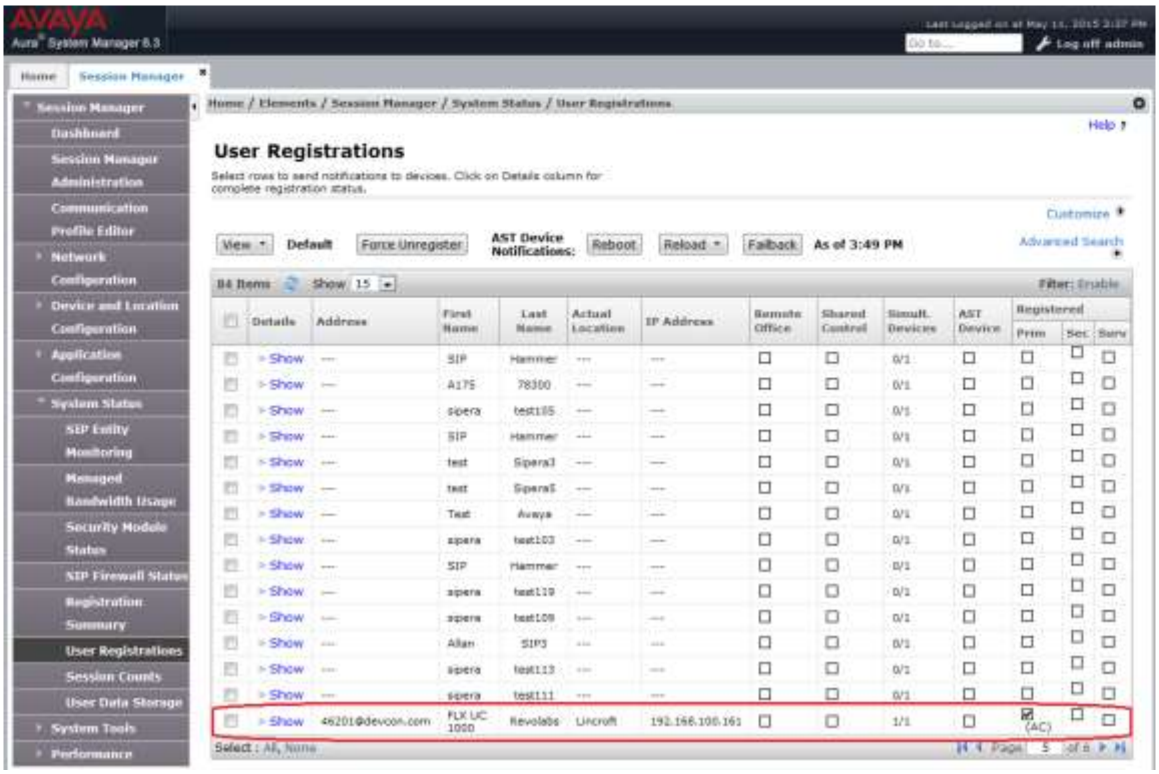
Under **Settings → Audio**, the supported audio codecs are displayed along with their priority order. The web page below shows the default settings. This web page allows audio codecs to be disabled or the codec precedence to be modified.



8 Verification Steps

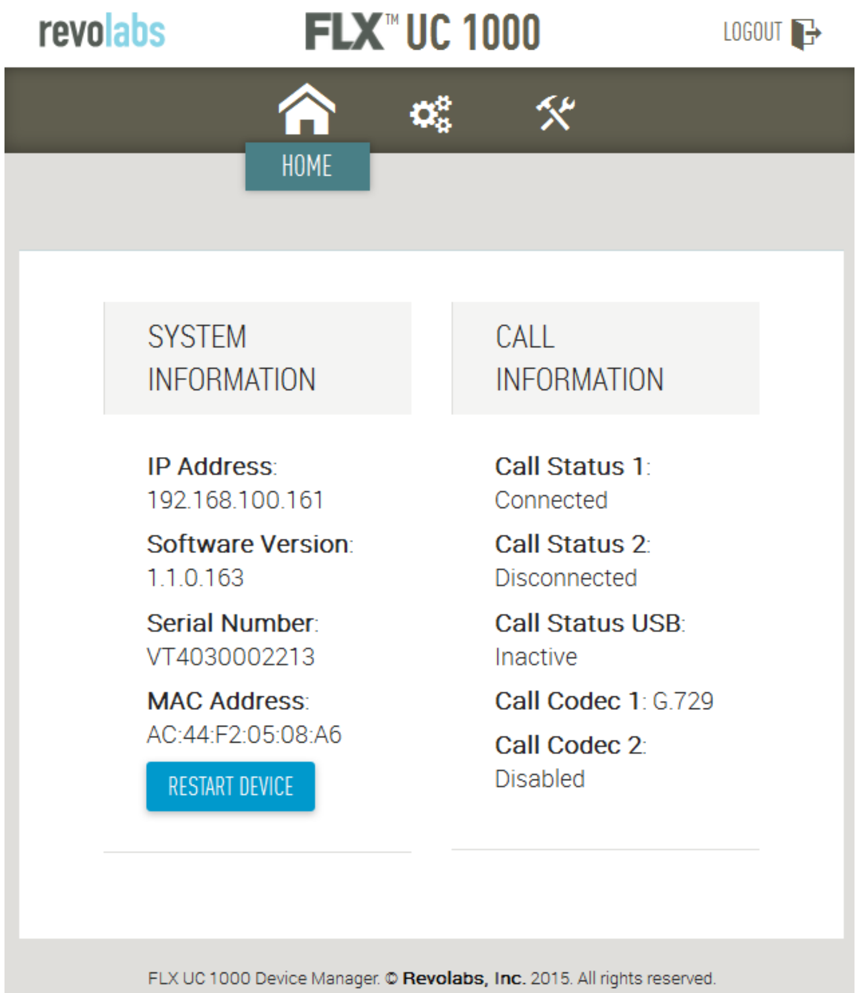
This section provides the tests that can be performed to verify proper configuration of Avaya Aura® Communication Manager, Avaya Aura® Session Manager and Revolabs FLX UC 1000.

1. Verify that Revolabs FLX UC 1000 has successfully registered with Session Manager. In System Manager, navigate to **Elements → Session Manager → System Status → User Registrations** to check the registration status.



2. Verify basic telephony features by establishing calls between Revolabs FLX UC 1000 and another phone.

3. In the **Home** webpage of Revolabs FLX UC 1000, the **Call Status** will be set to *Connected* when it is active on a call and the **Call Codec** used for the call will also be displayed as shown below.



9 Conclusion

These Application Notes describe the configuration steps required to integrate Revolabs FLX UC 1000 with an Avaya SIP telephony network. Revolabs FLX UC 1000 was able to register with Avaya Aura® Session Manager, successfully establish calls through Avaya Aura® Communication Manager to H.323 and SIP stations, and exercise telephony features. All feature and serviceability test cases were completed successfully with observations noted in **Section 2.2**.

10 References

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

[1] *Administering Avaya Aura® Communication Manager*, Release 6.3, Issue 10.0, June 2014, Document Number 03-300509.

[2] *Administering Avaya Aura® Session Manager*, Release 6.3, Issue 7, September 2014.

[3] *Revolabs FLX UC 1000 IP & USB Conference Phone Installation and Operation Guide, March 2015 (Rev 2.1.0).*

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.