Avaya Solution & Interoperability Test Lab

Application Notes for AEi Communications SDB-8000 SIP DECT Telephone with Avaya Aura® Session Manager and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate AEi Communications SDB-8000 SIP DECT Telephones with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. SDB-8000 SIP DECT Telephones serve the hospitality industry and provide the following features: speakerphone, hold, message waiting indicator (MWI), and programmable buttons. In the compliance test, SDB-8000 SIP DECT Telephones successfully registered with Session Manager, established calls with the PSTN and other Avaya SIP and H.323 telephones, and executed telephony and hospitality features using Avaya Aura® Communication Manager Feature Access Codes (FACs) and Feature Name Extensions (FNEs).

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 steps required to integrate AEi Communications SDB-8000 SIP DECT Telephones with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. SDB-8000 SIP DECT Telephones serve the hospitality industry and provide the following features: speakerphone, hold, message waiting indicator (MWI), and programmable buttons. In the compliance test, SDB-8000 SIP DECT Telephones successfully registered with Avaya Aura® Session Manager, established calls with the PSTN and other Avaya SIP and H.323 telephones, and executed telephony and hospitality features using Avaya Aura® Communication Manager Feature Access Codes (FACs) and Feature Name Extensions (FNEs).

2. General Test Approach and Test Results

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.

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between AEi Communications SDB-8000 SIP DECT Telephones and Avaya SIP and H.323 telephones. Basic telephony features, such as hold, speaker, and hospitality features including wake up calls and updating housekeeping status for a guest's room were also exercised. In addition, other extended telephony features, such as call forwarding and call pickup were also exercised using FNEs.

The serviceability testing focused on verifying that the AEi Communications SDB-8000 SIP DECT Telephone comes back into service after re-connecting the Ethernet connection or rebooting the SIP phone.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of SDB-8000 with Session Manager.
- Calls between SDB-8000 and Avaya SIP and H.323 telephones with Direct IP-IP Media (Shuffling) enabled and disabled.
- Support of multiple incoming and outgoing calls, using L1 and L2.
- G.711 codec support.
- Proper recognition of DTMF tones.
- Long call duration and long hold duration.
- Extended telephony features using Communication Manager FNEs and FACs, such as Hospitality Wakeup calls, Housekeeping Status Access Codes, Call Forwarding and Call Pickup.

- Voicemail coverage, MWI support, and logging into voicemail system to retrieve voicemail messages.
- Use of programmable buttons on SDB-8000.
- Proper system recovery after a restart of the SDB-8000 and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observations noted:

- The SDB-8000 SIP DECT Telephone does not support conference.
- The SDB-8000 SIP DECT Telephone does not support transfer.

2.3. Support

For technical support on the AEi Communications SDB-8000 SIP DECT Telephone, contact AEi Communications Support via phone, email, or website.

- **Phone:** (650) 552-9416
- **Email:** sales@aeicommunications.com
- **Web:** http://www.aeicommunications.com/contact.html

3. Reference Configuration

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

- Avaya Aura® Communication Manager running in a virtual environment with an Avaya G450 Media Gateway and/or Avaya Aura® Media Server.
- 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® Messaging served as the voicemail system.
- Avaya 9600 Series SIP and H.323 Telephones.
- AEi Communications SDB-8000 SIP DECT Telephones.

AEi Communications SDB-8000 SIP DECT Telephones registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.

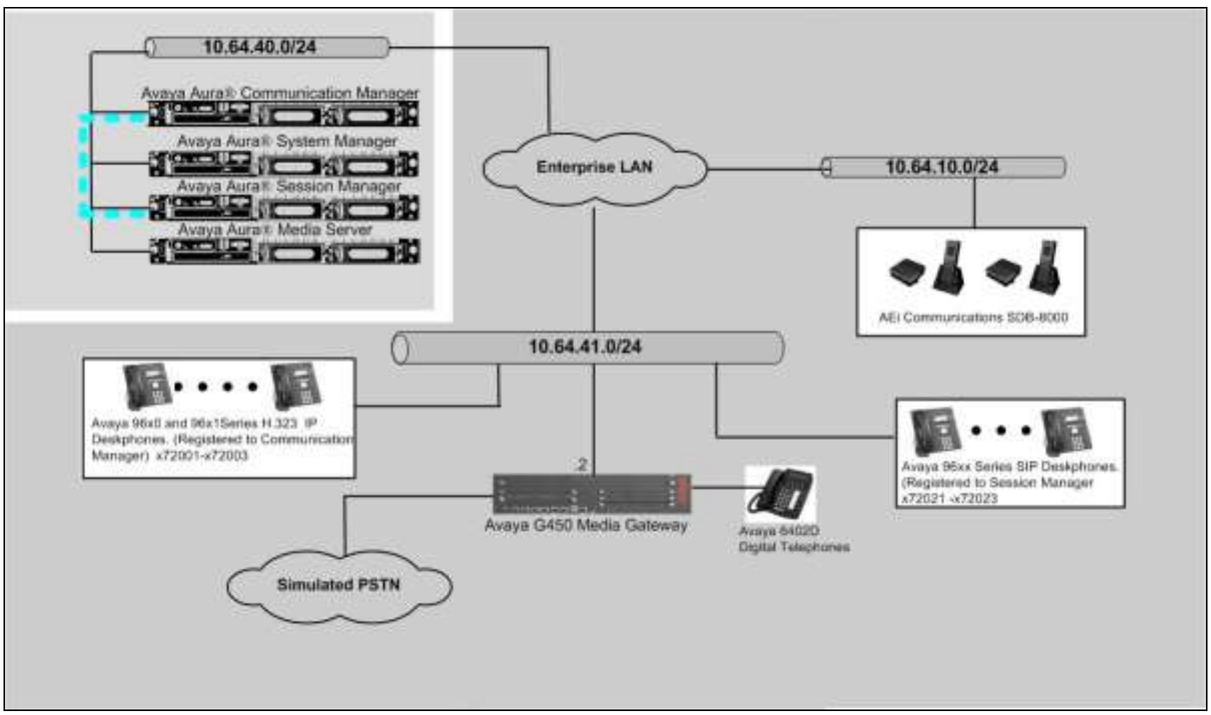


Figure 1: Avaya SIP Network with AEi Communications SDB-8000 SIP DECT Telephones

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 in a Virtual Environment with an Avaya G450 Media Gateway	7.0.1.0 FP1 (R017x.00.0.441.0 with Patch 23012)
Avaya Aura® Media-Server in a Virtual Environment	7.7.0.226
Avaya Aura® System Manager in a Virtual Environment	7.0.1
Avaya Aura® Session Manager in a Virtual Environment	7.0.0.0.700007
Avaya Aura® Messaging	6.3.3 SP 5
Avaya 9650C	3.25 (H.323)
Avaya 96x1 Series IP Deskphones	6.6.115 (H.323)
Avaya 96x1 Series IP Deskphones	7.0.0.39 (SIP)
AEi Communications SDB-8000 SIP DECT Telephone	IPS8XXX_A02

5. Configure Avaya Aura® Communication Manager

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

- Verify Communication Manager license
- Administer IP Network Region and IP Codec Set
- Administer Hospitality Features

Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials.

Note: It is assumed that basic configuration of the Communication Manager has already been completed, such as the SIP trunk to Session Manager. The SIP station configuration for AEi Communications SDB-8000 is configured through Avaya Aura® System Manager in **Section 6.2**.

5.1. Verify 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: V17                              Software Package: Enterprise
       Location: 2                                 System ID (SID): 1
       Platform: 28                                Module ID (MID): 1

                                                            USED
                                Platform Maximum Ports: 6400  200
                                      Maximum Stations: 2400  47
                              Maximum XMOBILE Stations: 2400  0
                    Maximum Off-PBX Telephones - EC500: 9600  1
                    Maximum Off-PBX Telephones -   OPS: 9600  22
                    Maximum Off-PBX Telephones - PBFMC: 9600  0
                    Maximum Off-PBX Telephones - PVFMC: 9600  0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 313   1




        (NOTE: You must logoff & login to effect the permission changes.)
```

5.2. 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 the Avaya G450 Media Gateway or Avaya Aura® Media Server. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name:                       Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 16390                     IP Audio Hairpinning? n
   UDP Port Max: 16999
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 the SDB-8000 SIP DECT Telephone. 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. The SDB-8000 SIP DECT Telephone supports G.711Mu.

```
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:
 4:
 5:

     Media Encryption                   Encrypted SRTCP: enforce-unenc-srtcp
 1: none
 2:
```

5.3. Administer Hospitality Features

This section covers the configuration of two hospitality features: wakeup calls and housekeeping status. A hotel guest may enter the wake up feature access code (FAC) followed by the time for the wakeup call in *hhmm* format, where *hh* is the hour and *mm* is the minute. The housekeeping status of a hotel room may be changed by dialing the housekeeping status access code from the hotel room phone.

5.3.1. Administer Feature Name Extensions (FNEs)

Prior to dialing the wakeup call, the SIP user must first receive dial tone from Communication Manager. This is achieved by first dialing the **Idle Appearance Select FNE** configured as shown below. Afterwards, the wakeup call access code may be dialed. The housekeeping status access codes may be dialed directly (FAC) without dialing the **Idle Appearance Select FNE**.

```
change off-pbx-telephone feature-name-extensions set 1          Page   2 of   2
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME

       Exclusion (Toggle On/Off): 78018
      Extended Group Call Pickup: 78019
          Held Appearance Select: 78020
          Idle Appearance Select: 78021
              Last Number Dialed: 78022
            Malicious Call Trace:
     Malicious Call Trace Cancel:
             Off-Pbx Call Enable: 78025
            Off-Pbx Call Disable: 78026
                   Priority Call: 78027
                          Recall: 78028
                  Send All Calls: 78029
           Send All Calls Cancel: 78030
               Transfer Complete: 78031
             Transfer On Hang-Up: 78032
          Transfer to Voice Mail: 78033
         Whisper Page Activation: 78034
```

5.3.2. Administer Features Access Codes (FACs)

In the **Feature Access Code (FAC)** form configure the **Automatic Wakeup Call Access Code** and the **Housekeeping Status (Client Room) Access Codes**, as needed, as shown below. The FACs should comply with the dial plan.

```
change feature-access-codes                                    Page   8 of  10
                              FEATURE ACCESS CODE (FAC)
                                Hospitality Features

               Automatic Wakeup Call Access Code: 150
   Housekeeping Status (Client Room) Access Code: 151
   Housekeeping Status (Client Room) Access Code: 152
   Housekeeping Status (Client Room) Access Code: 153
   Housekeeping Status (Client Room) Access Code: 154
   Housekeeping Status (Client Room) Access Code: 155
   Housekeeping Status (Client Room) Access Code: 156
       Housekeeping Status (Station) Access Code: 157
       Housekeeping Status (Station) Access Code: 158
       Housekeeping Status (Station) Access Code: 159
       Housekeeping Status (Station) Access Code: 160
          Verify Wakeup Announcement Access Code: 161
                Voice Do Not Disturb Access Code: 162
```

5.3.3. Allow Wake-up Calls

In the **Hospitality** form, enable **Room Activated Wakeup With Tones**. Communication Manager will prompt the user with tones when enabling a wakeup call. For example, a 3-burst confirmation tone will be played to prompt the user to enter the wakeup time.

```
change system-parameters hospitality                           Page   2 of   3
                             HOSPITALITY

            Dual Wakeups? n    Daily Wakeup? n    VIP Wakeup? n

                            Room Activated Wakeup With Tones? y
                    Time of Scheduled Wakeup Activity Report:
                     Time of Scheduled Wakeup Summary Report:
           Time of Scheduled Emergency Access Summary Report:
                                          Announcement Type: silence


          Length of Time to Remain Connected to Announcement: 30
             Extension to Receive Failed Wakeup LWC Messages:
            Routing Extension on Unavailable Voice Synthesis:
                    Display Room Information in Call Display? n
                          Automatic Selection of DID Numbers? n
                         Custom Selection of VIP DID Numbers? n
                                  Number of Digits from PMS:
                                           PMS Sends Prefix? n
                      Number of Digits in PMS Coverage Path: 3
                                     Digit to Insert/Delete:
```

5.3.4. Allow Housekeeping Status Updates

To allow housekeeping to change the housekeeping status of a guests room by dialing the appropriate access code, **Client Room** must be enabled on the COS assigned to the SIP phone. In this example, **Client Room** was enabled for COS 1, which was assigned to the AEi SDB-8000 phone.

```
change cos-group 1                                                Page   1 of   2
CLASS OF SERVICE         COS Group: 1   COS Name:

                                 0  1  2  3  4  5  6  7  8  9 10 11 12 13 14 15
 Auto Callback                   n  y  y  n  y  n  y  n  y  n  y  n  y  n  y  n
 Call Fwd-All Calls              n  y  n  y  y  n  n  y  y  n  n  y  y  n  n  y
 Data Privacy                    n  y  n  n  n  y  y  y  y  n  n  n  n  y  y  y
 Priority Calling                n  y  n  n  n  n  n  n  n  y  y  y  y  y  y  y
 Console Permissions             n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Off-hook Alert                  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Client Room                     n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Restrict Call Fwd-Off Net       y  n  y  y  y  y  y  y  y  y  y  y  y  y  y  y
 Call Forwarding Busy/DA         n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Personal Station Access (PSA)   n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding All         n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding B/DA        n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Trk-to-Trk Transfer Override    n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 QSIG Call Offer Originations    n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Contact Closure Activation      n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
```

6. Configure Avaya Aura® Session Manager

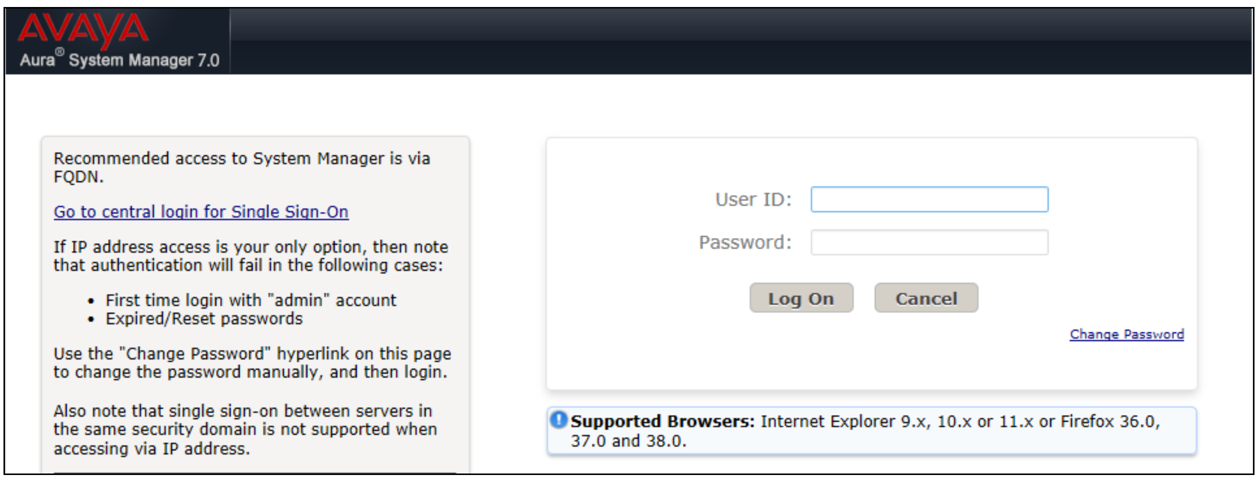
This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- 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 the SDB-8000 SIP DECT Telephone.

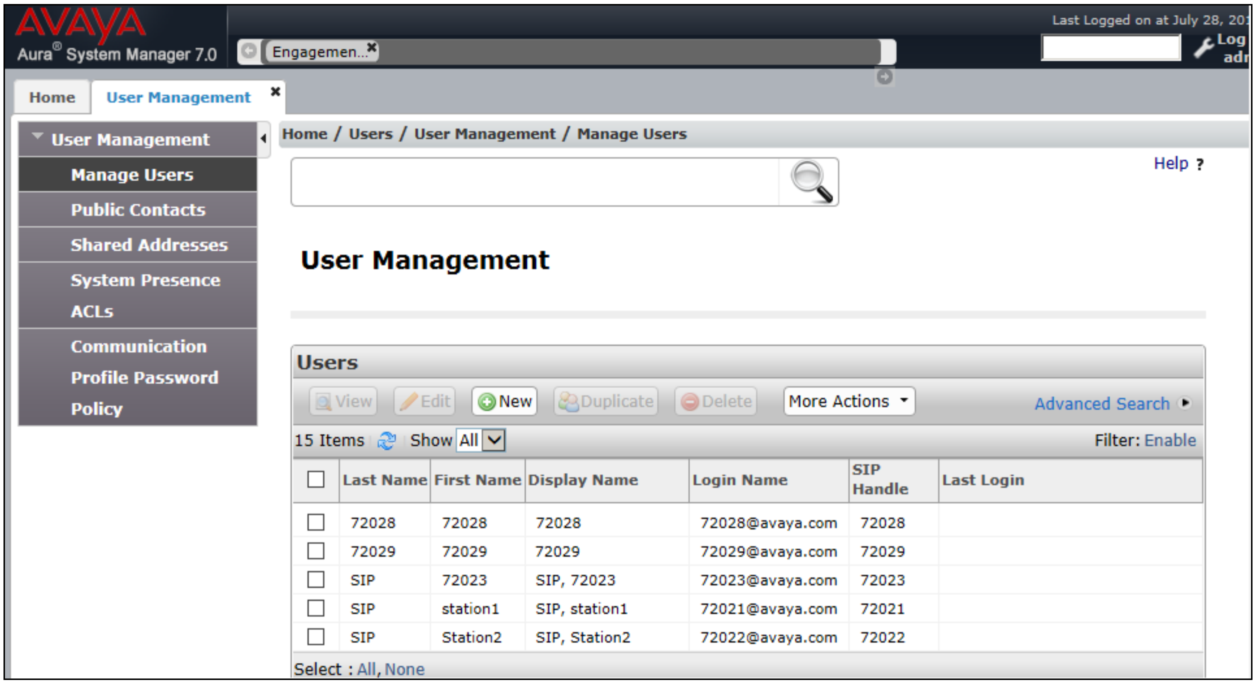
6.1. Launch System Manager

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.



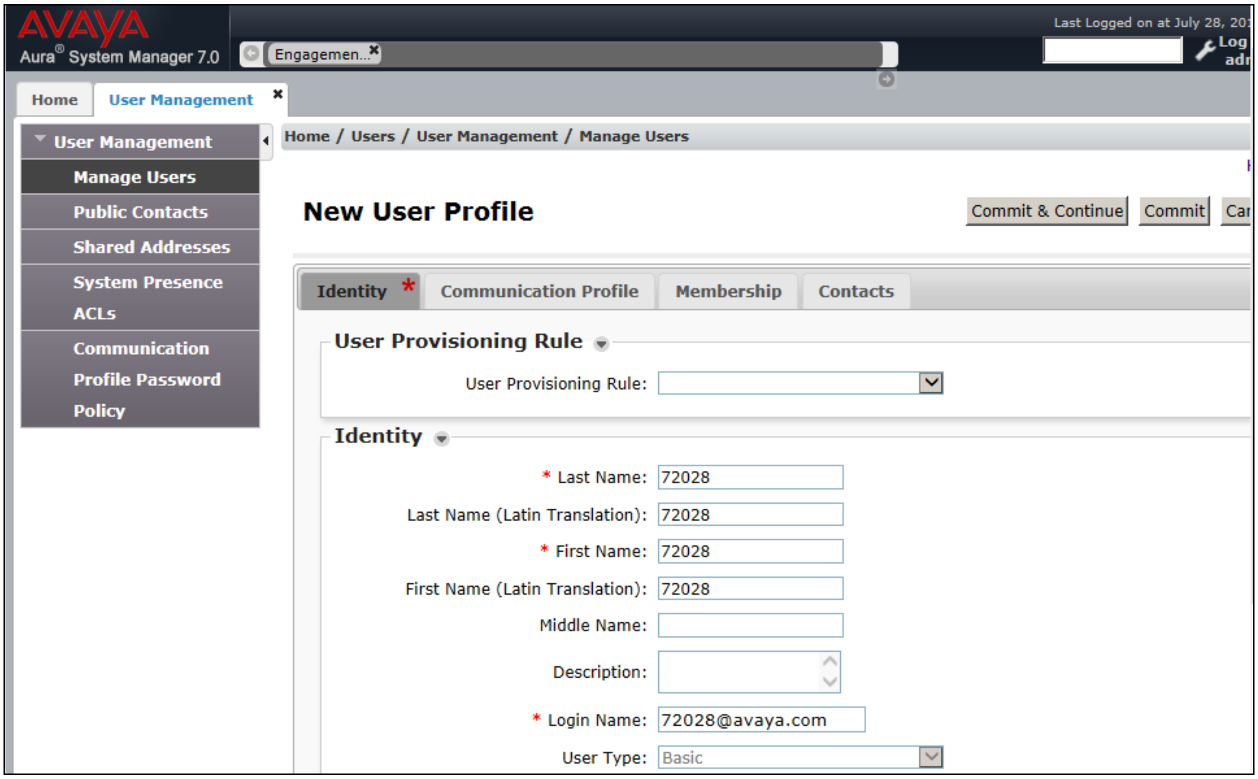
6.2. Administer SIP User

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.



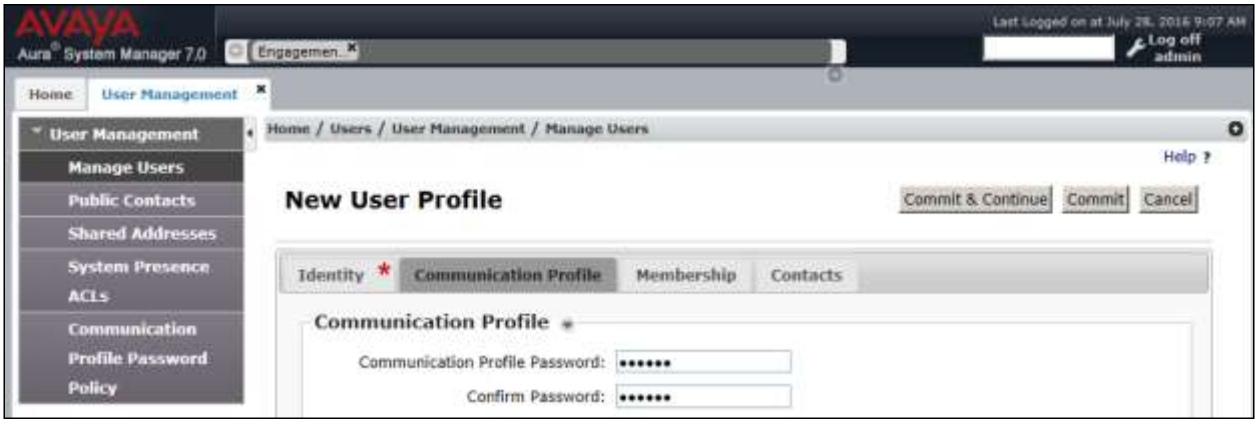
6.2.1. Identity

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 SDB-8000 SIP DECT extension and "*<domain>*" is the applicable SIP domain name from **Section 5.2**. Retain the default values in the remaining fields.



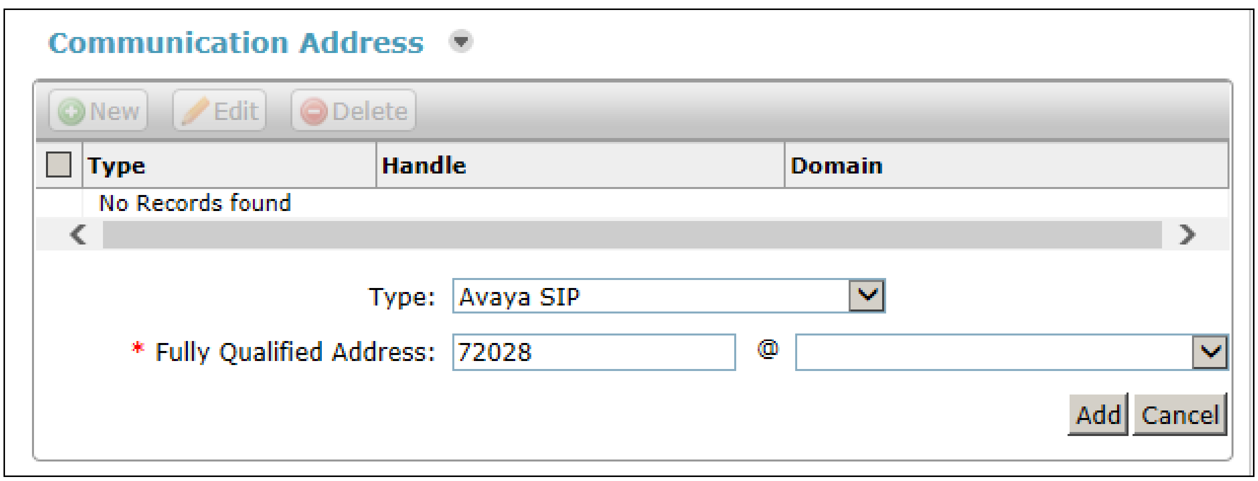
6.2.2. Communication Profile

Select the **Communication Profile** tab. For **Communication Profile Password** and **Confirm Password**, enter the desired password for the SIP user to use for registration.



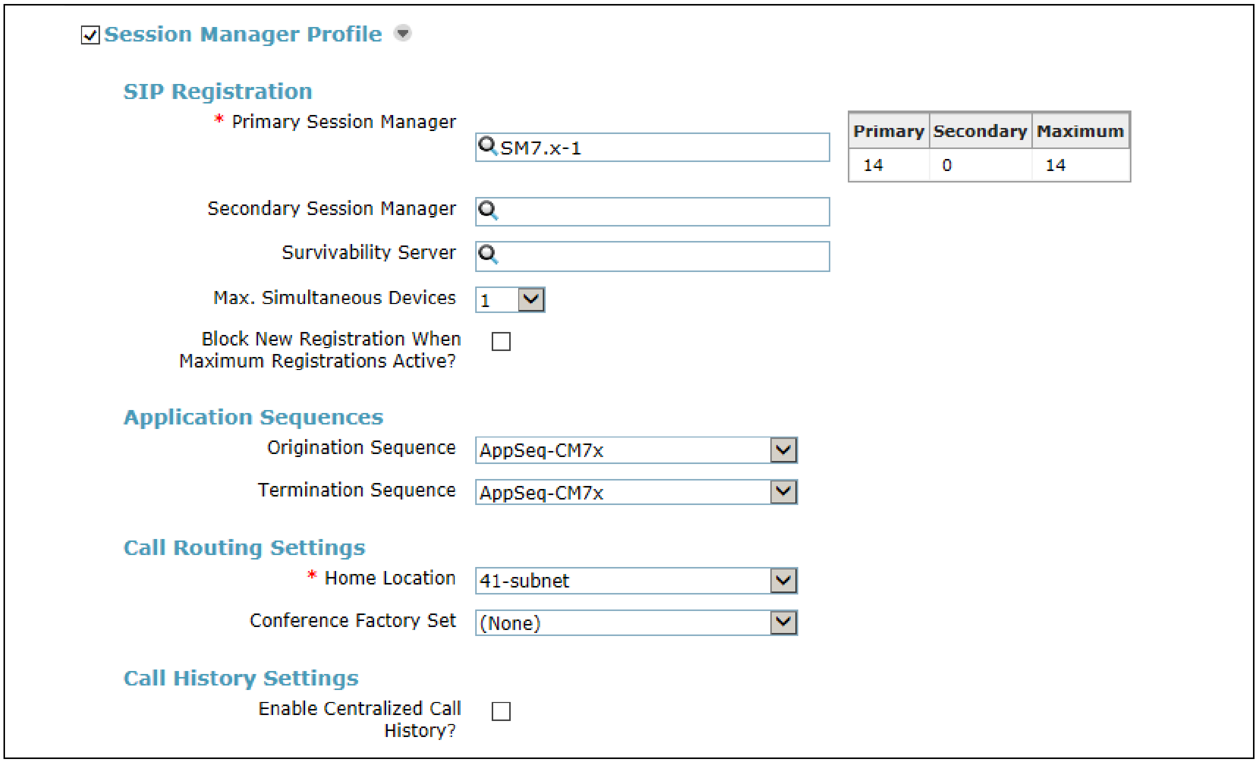
6.2.3. Communication Address

In the **Communication Address** sub-section, click **New** to add a new entry. The **Communication Address** sub-section is updated with additional fields 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 6.2.1**. Click **Add**.



6.2.4. Session Manager Profile

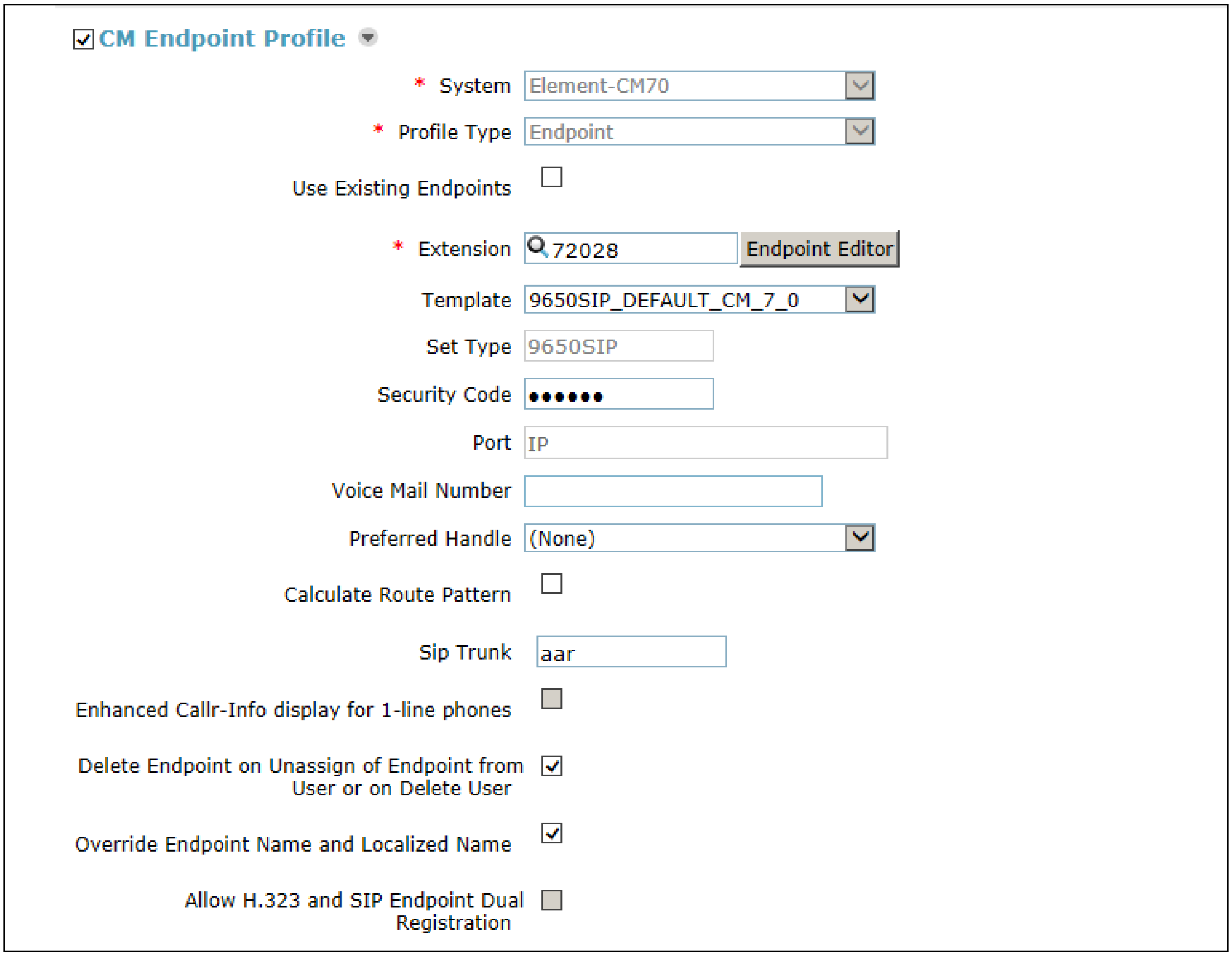
Scroll down to check and expand **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.



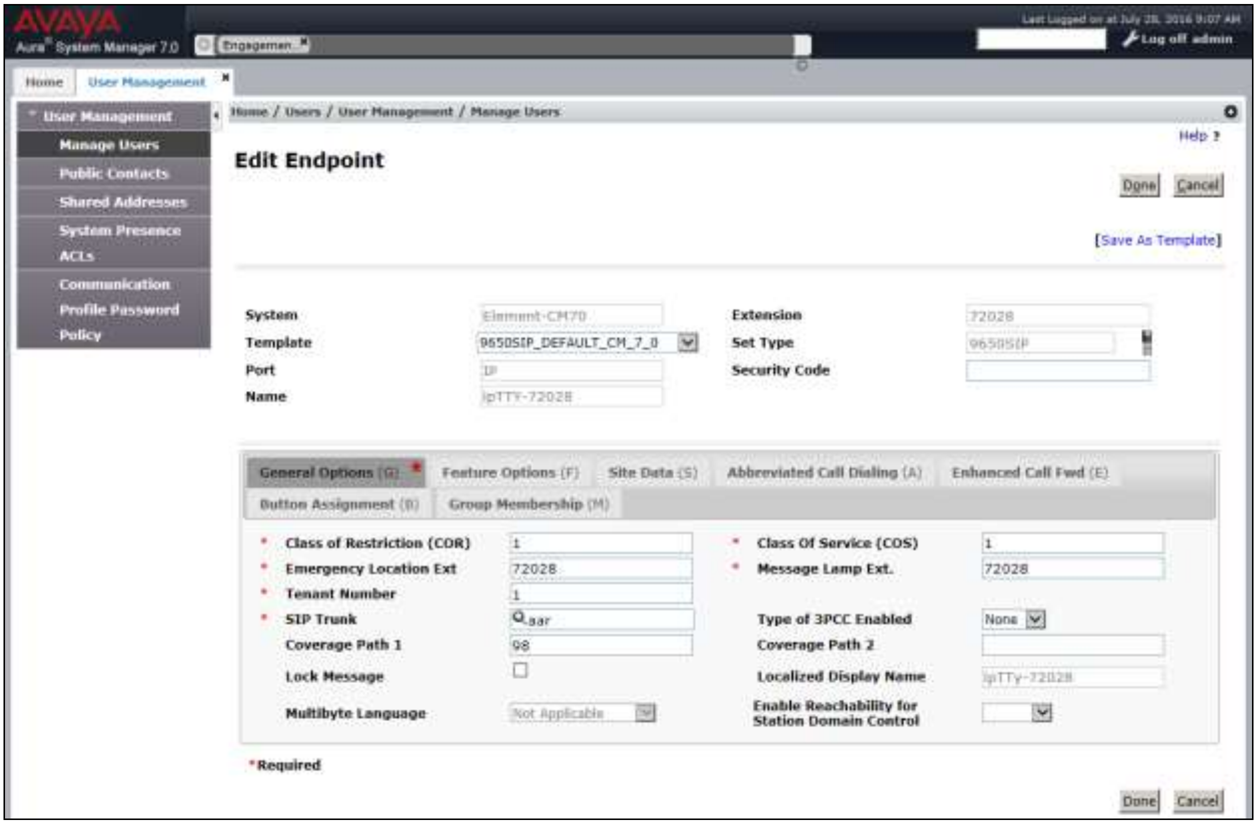
6.2.5. CM Endpoint Profile

Scroll down to check and expand **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager, and select *Endpoint* for **Profile Type**. For **Extension**, enter the SIP user extension from **Section 6.2.1**. For **Template**, select *9650SIP_DEFAULT_CM_7_0*. For **Port**, click and select *IP*. Retain the default values in the remaining fields.

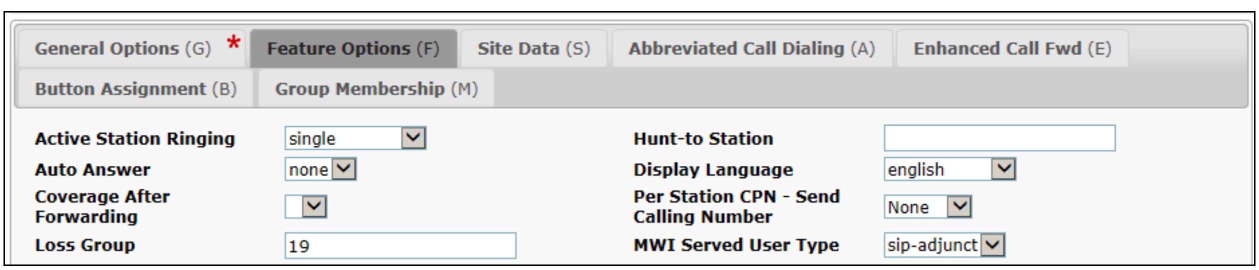
Click **Commit** to save the configuration (not shown).



In the **CM Endpoint Profile** sub-section (not shown), click the **Endpoint Editor** button to display the page below. In the **General Options** tab, specify that coverage path that points to the voicemail system in the **Coverage Path 1** field. This provides voicemail coverage for the SIP user. In this example, coverage path 98 was used.



In the **Feature Options** tab, set the **MWI Served User Type** field to *sip-adjunct*. This allows MWI to be enabled for the SIP user. The voicemail system was connected via SIP to Session Manager. Once completed, click **Done**.

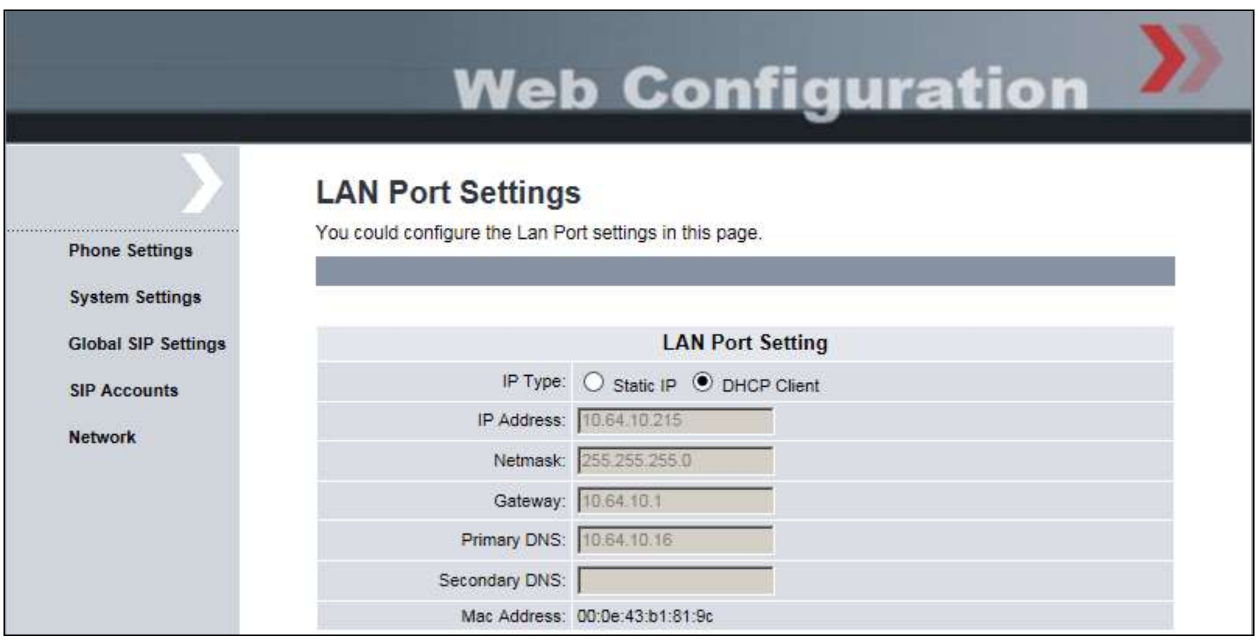


7. Configure AEi Communications SDB-8000 SIP DECT Telephone

Access the SDB-8000 web interface by using the URL "https://ip-address:8000" in an Internet browser window, where "ip-address" is the IP address of the SIP phone. Log in using the appropriate credentials and then click **Login**.

7.1. Administer LAN Port Settings

Select **Network → LAN Port Settings** in the left pane and configure the SIP phone's network settings as shown below. During the compliance test, DHCP was utilized.



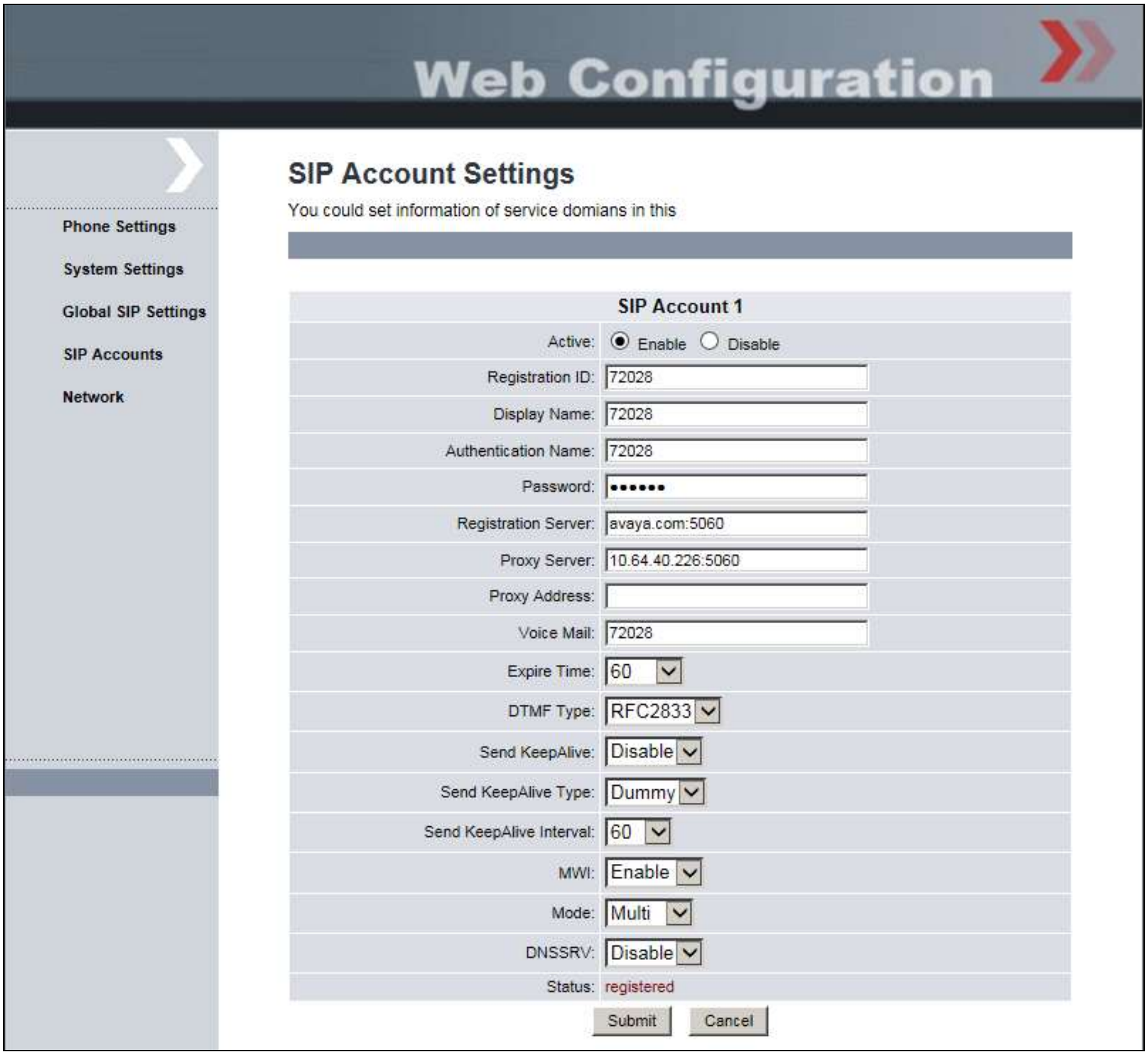
7.2. Administer SIP Accounts

Navigate to **SIP Accounts** in the left pane and click **Add** to add a SIP account.

Navigate to the **SIP Proxy** webpage as shown below. Under the **Basic SIP Proxy Settings** section, configure the following parameters.

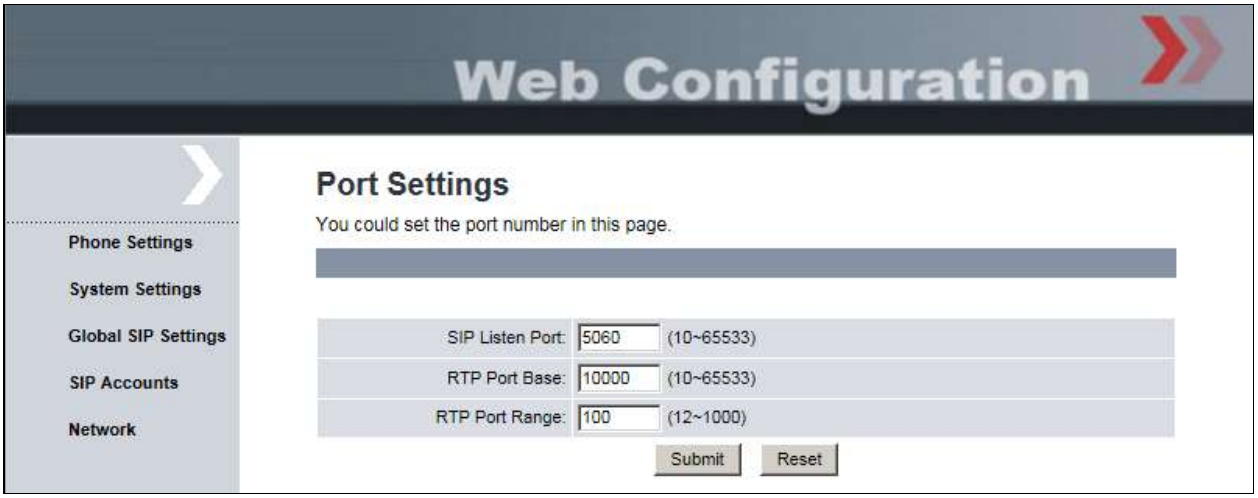
- **Registration ID:** Specify the Registration ID (e.g., *72028*, the SIP extension).
- **Display Name:** Specify the Display Name (e.g., *72028*, the SIP extension).
- **Authentication Name:** Specify the SIP extension of the SDB-8000 SIP DECT Telephone (e.g., *72028*).
- **Password:** Specify the SIP password configured in **Section 6.2.2**.
- **Registration Server:** Set to the domain name and port (e.g., *avaya.com:5060*).
- **Proxy Server:** Set to the Session Manager IP address and port (e.g., *10.64.40.226:5060*).
- **Voice Mail:** Specify the voicemail pilot number (e.g., *72028*).
- **Expire Time:** Specify the frequency of SIP re-registrations (e.g., *60*).
- **DTMF Type:** Select *RFC2833*.
- **MWI:** Set to *Enable*.
- Retain the default values in the remaining fields.

Notice at the bottom of the screen that the status is *registered with Session Manager*.

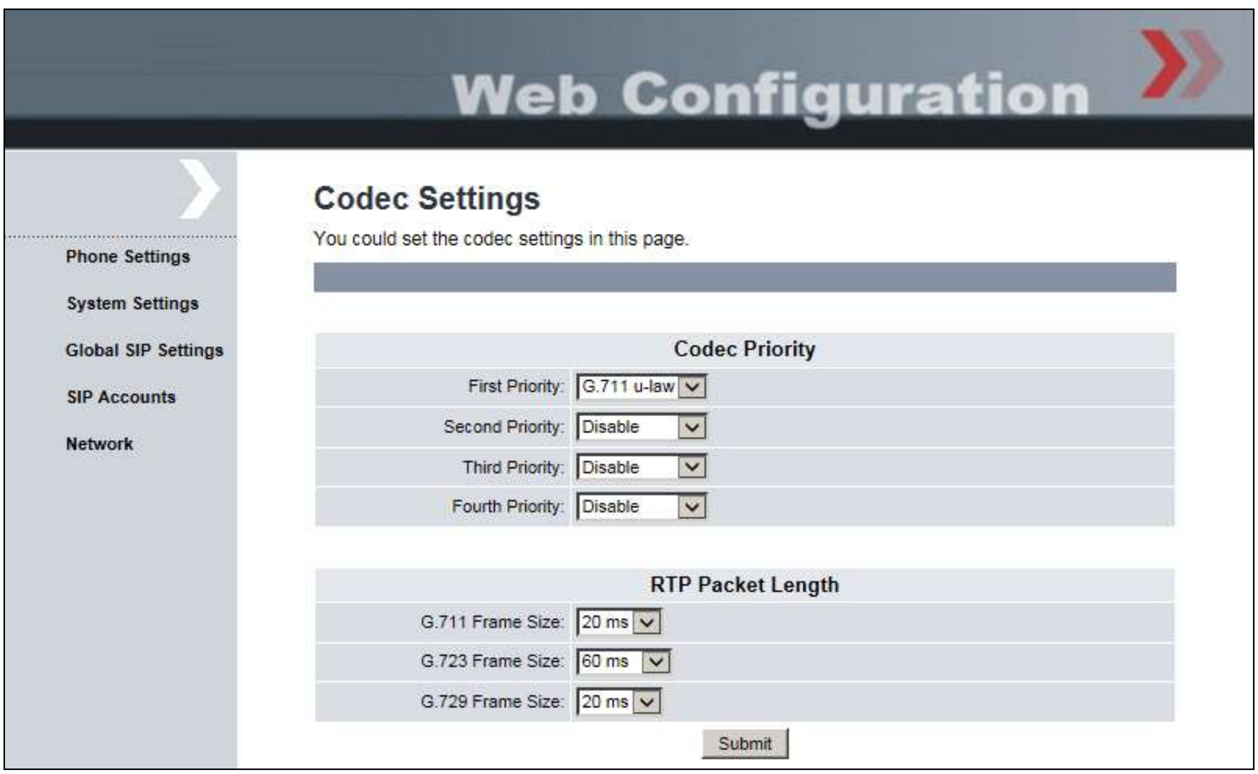

Web Configuration
Phone Settings
System Settings
Global SIP Settings
SIP Accounts
Network
SIP Account Settings
You could set information of service domians in this
SIP Account 1
Active: Enable Disable
Registration ID: 72028
Display Name: 72028
Authentication Name: 72028
Password: ••••••
Registration Server: avaya.com:5060
Proxy Server: 10.64.40.226:5060
Proxy Address:
Voice Mail: 72028
Expire Time: 60
DTMF Type: RFC2833
Send KeepAlive: Disable
Send KeepAlive Type: Dummy
Send KeepAlive Interval: 60
MWI: Enable
Mode: Multi
DNSSRV: Disable
Status: registered
Submit
Cancel

7.3. Administer Global SIP Settings

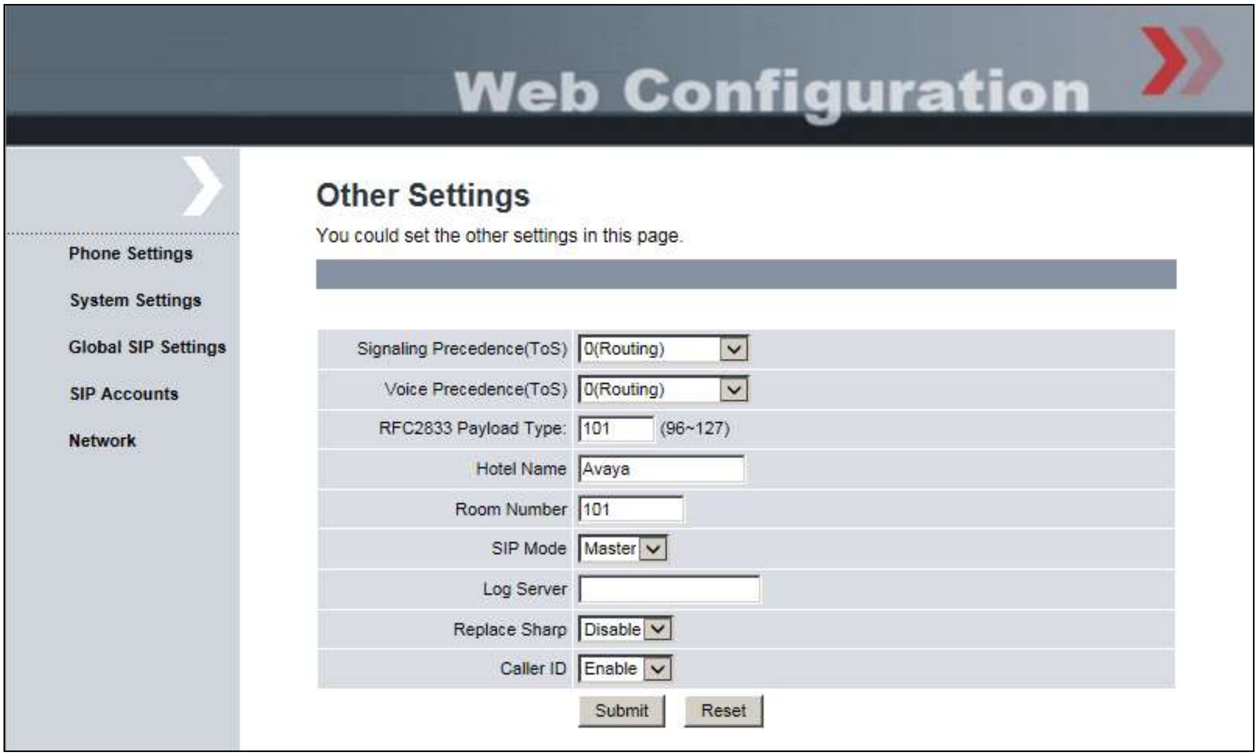
Navigate to **Global SIP Settings → Port Settings** and verify the SIP Listen Port being used (e.g., *5060*).



Navigate to **Global SIP Settings → Codec Settings** to verify the codec priority. In this example, the first priority is G.711u-law. AEi Communications SDB-8000 SIP DECT Telephones supports G.711.



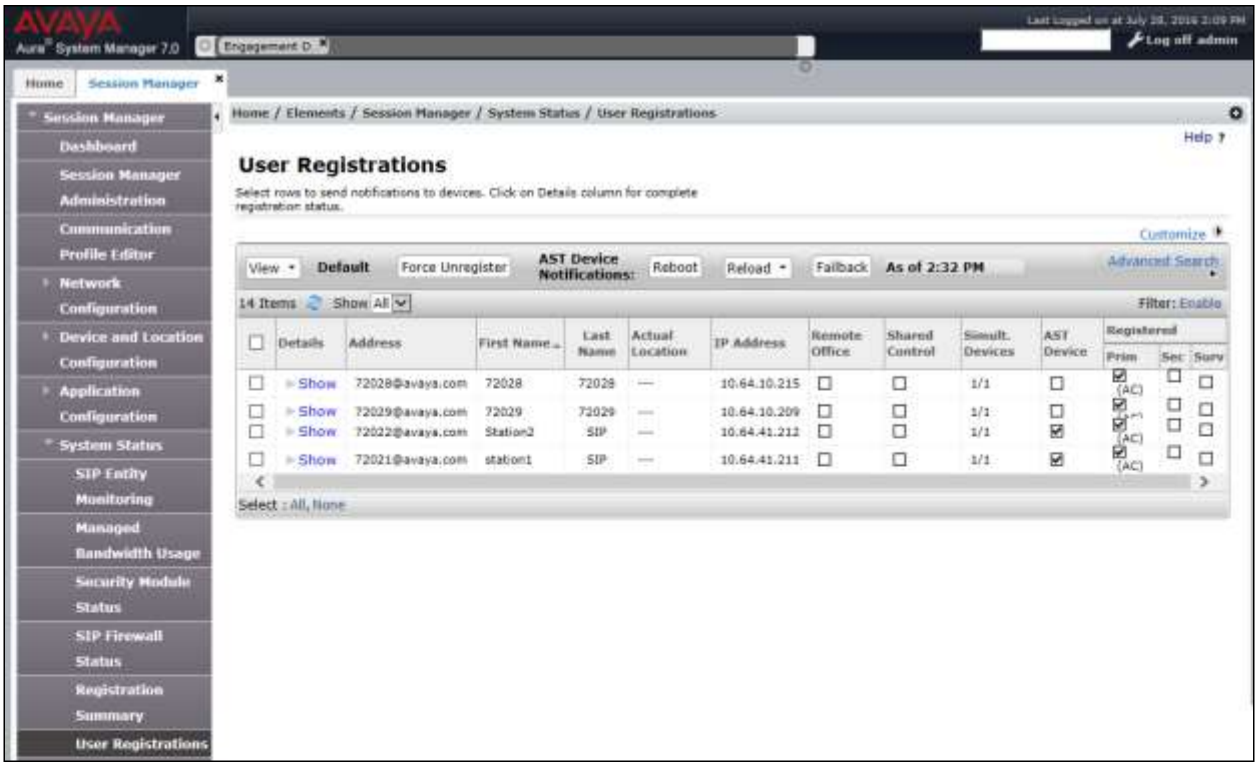
Navigate to **Global SIP Settings → Other Settings**, and verify the Caller ID was set to *Enable,* so that the calling party ID can be displayed during the conversation.



8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the AEi Communications SDB-8000 SIP DECT Telephone with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

1. Verify that the SDB-8000 SIP DECT Telephone has successfully registered with Session Manager. In System Manager, navigate to **Elements → Session Manager → System Status → User Registrations** to check the registration status. The SIP registration status can also be seen in the SIP Account page of the SDB-8000 web interface seen in **Section 7.2**.



2. Verify basic telephony features by establishing calls between a SDB-8000 SIP DECT Telephone with and another phone.

9. Conclusion

These Application Notes have described the administration steps required to integrate the AEi Communications SDB-8000 SIP DECT Telephone with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The AEi Communications SDB-8000 SIP DECT Telephone successfully registered with Session Manager and basic telephony and hospitality features were verified. All test cases passed with observations noted in **Section 2.2**.

10. References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at http://support.avaya.com.

[1] *Administering Avaya Aura® Communication Manager*, Release 7.0.1, Issue 2, May 2016, Document Number 03-300509.

[2] Administering Avaya Aura® System Manager for Release 7.0.1, Release 7.0.1, Issue 2, June 2016

The following document was provided by AEi Communications.

[3] SDB-8000 Configuration Guide 1.0, May 2016

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.