Avaya Solution & Interoperability Test Lab

Application Notes for AEi Communications VM-2208-S(S) SIP 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 VM-2208-S(S) SIP Telephones with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. VM-2208-S(S) SIP Telephones serve the hospitality industry and provide the following features: speakerphone, hold and message waiting indicator (MWI). In the compliance test, VM-2208-S(S) SIP 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).

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 VM-2208-S(S) SIP Telephones with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. VM-2208-S(S) SIP Telephones serve the hospitality industry and provide the following features: speakerphone, hold, wake up calls, house-keeping status and message waiting indicator (MWI). In the compliance test, VM-2208-S(S) SIP 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

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between AEi Communications VM-2208-S(S) SIP 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 FACs and FNEs.

The serviceability testing focused on verifying that the AEi Communications VM-2208-S(S) SIP Telephone comes back into service after re-connecting the Ethernet connection or rebooting the SIP 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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note 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 this Application Note, the interface between Avaya systems and AEi Communications did not include use of any specific encryption features as requested by AEi Communications.

Avaya's formal testing and Declaration of Conformity is provided only on the headsets/handsets 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/handsets 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 VM-2208-S(S) with Session Manager.
- Calls between VM-2208-S(S) and Avaya SIP and H.323 telephones with Direct IP-IP Media (Shuffling) enabled and disabled (see **Section 5.2**).
- Support of multiple incoming and outgoing calls, using L1 and L2.
- G.711 MU-Law 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.
- Proper system recovery after a restart of the VM-2208-S(S) and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observations noted:

- The VM-2208-S(S) SIP Telephone does not support conference.
- The VM-2208-S(S) SIP Telephone does not support transfer however a call can be transferred to it.
- The VM-2208-S(S) SIP Telephone does not support the Long Hold Recall Timer feature.
- The VM-2208-S(S) SIP Telephone does not support Multi-Device Access feature.
- The VM-2208-S(S) SIP Telephone does not support Call Park feature however can Unpark a call if a call is parked by an Avaya telephone.
- There is no option to enable Caller ID from the GUI. The Caller ID in VM-2208-S(S) SIP Telephone is turned off by default and can only be turned on by provisioning in the configuration file.

2.3. Support

For technical support on the AEi Communications VM-2208-S(S) SIP Telephone, contact AEi Communications Support via phone, email, or website.

- **Phone:** +1 (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 96x1 Series SIP and H.323 Telephones.
- AEi Communications VM-2208-S(S) SIP Telephones.

AEi Communications VM-2208-S(S) SIP Telephones registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.

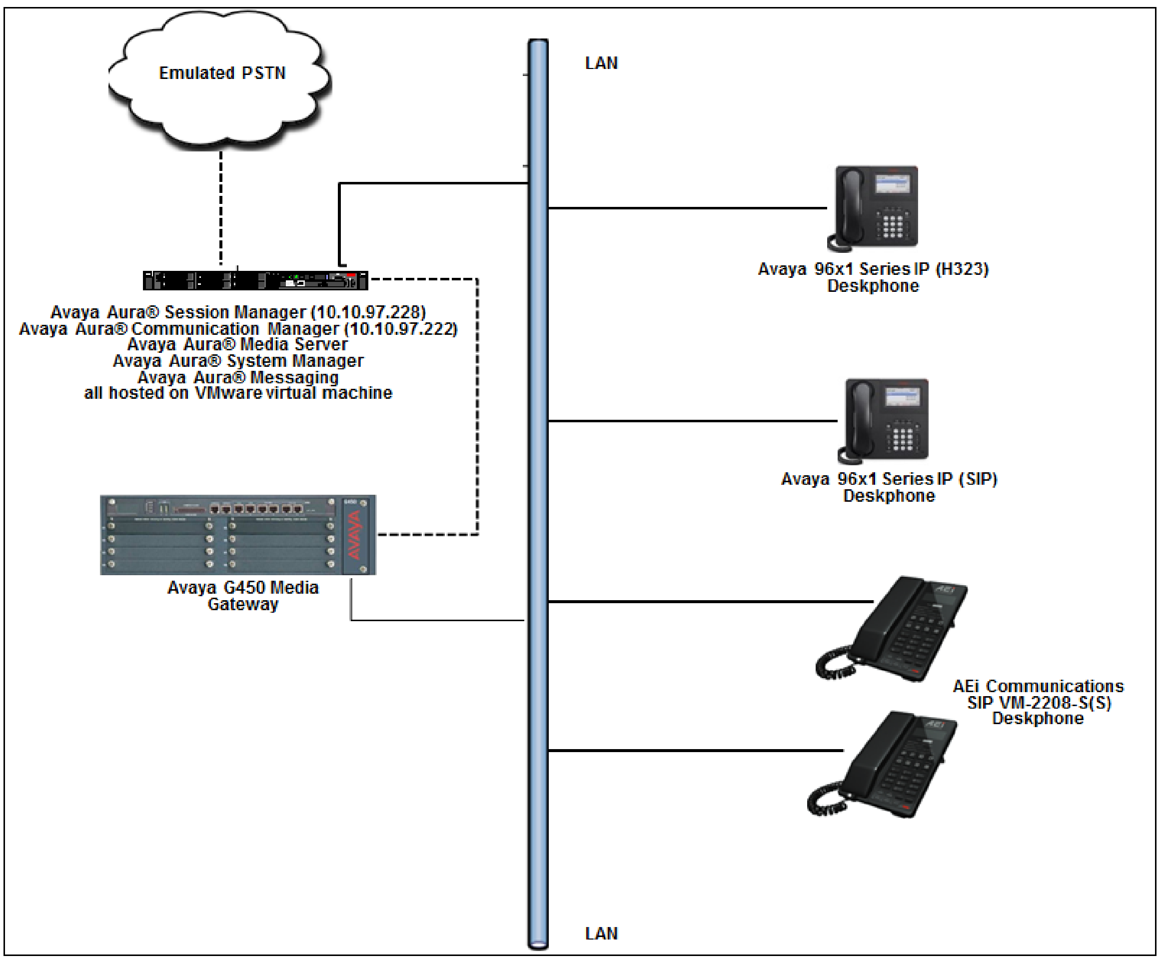


Figure 1: Avaya SIP Network with AEi Communications VM-2208-S(S) SIP Telephones

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	7.1.0.0.532
Avaya Aura® Media-Server in a Virtual Environment	7.8.0.312
Avaya Aura® System Manager in a Virtual Environment	7.1.0.0
Avaya Aura® Session Manager in a Virtual Environment	7.1.0.0.710028
Avaya Aura® Messaging	7.0.1.2.0-FP1SP2
Avaya G450 Gateway	38.18.0/1
Avaya 9611 IP Deskphones	6.6401 (H.323)
Avaya 9641GS IP Deskphones	7.0.1.2.9 (SIP)
AEi Communications VM-2208-S(S) SIP Telephone	VM2SLTD_A37

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 VM-2208-S(S) 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: 65000 129
                                      Maximum Stations: 41000 42
                              Maximum XMOBILE Stations: 41000 0
                    Maximum Off-PBX Telephones - EC500: 41000 0
                    Maximum Off-PBX Telephones -   OPS: 41000 15
                    Maximum Off-PBX Telephones - PBFMC: 41000 0
                    Maximum Off-PBX Telephones - PVFMC: 41000 0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 313   0




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

5.2. Administer IP 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 *bvwdev.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. Note that this is also dependent on the **Direct IP-IP Audio Connections** value defined in the **signaling-group** form (not shown).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       NR Group: 1
   Location: 1       Authoritative Domain: bvwdev.com
       Name: Region1               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? y
      UDP Port Max: 8001
   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 VM-2208-S(S) SIP 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 VM-2208-S(S) SIP Telephone supports G.711Mu.

```
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: enforce-unenc-srtcp
 1: 1-srtp-aescm128-hmac80
 2: 2-srtp-aescm128-hmac32
 3: none
 4:
```

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   3
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME

       Exclusion (Toggle On/Off):
      Extended Group Call Pickup:
          Held Appearance Select:
          Idle Appearance Select: 56214
              Last Number Dialed:
            Malicious Call Trace:
     Malicious Call Trace Cancel:
             Off-Pbx Call Enable:
            Off-Pbx Call Disable:
                   Priority Call:
                          Recall:
                  Send All Calls:
           Send All Calls Cancel:
               Transfer Complete:
             Transfer On Hang-Up:
          Transfer to Voice Mail:
         Whisper Page Activation:
```

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  11
                               FEATURE ACCESS CODE (FAC)
                                 Hospitality Features

               Automatic Wakeup Call Access Code: *60
   Housekeeping Status (Client Room) Access Code: *61
   Housekeeping Status (Client Room) Access Code: *62
   Housekeeping Status (Client Room) Access Code: *63
   Housekeeping Status (Client Room) Access Code: *64
   Housekeeping Status (Client Room) Access Code: *65
   Housekeeping Status (Client Room) Access Code:
       Housekeeping Status (Station) Access Code:
       Housekeeping Status (Station) Access Code:
       Housekeeping Status (Station) Access Code:
       Housekeeping Status (Station) Access Code:
          Verify Wakeup Announcement Access Code:
                Voice Do Not Disturb Access Code:
```

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: integrated
                         Integrated Announcement Extension: 56003

          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 VM-2208-S(S) phone.

```
change cos                                                      Page   1 of   2
CLASS OF SERVICE


                                 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              y  y  y  y  y  y  y  y  y  y  y  y  y  y  y  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             y  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         y  y  y  y  y  y  y  y  y  y  y  y  y  y  y  y
 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  n  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

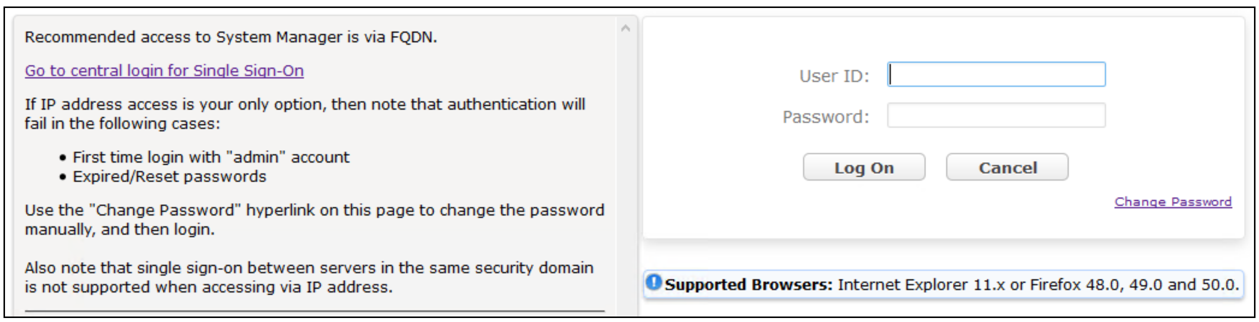
This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- 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 VM-2208-S(S) SIP Telephone.

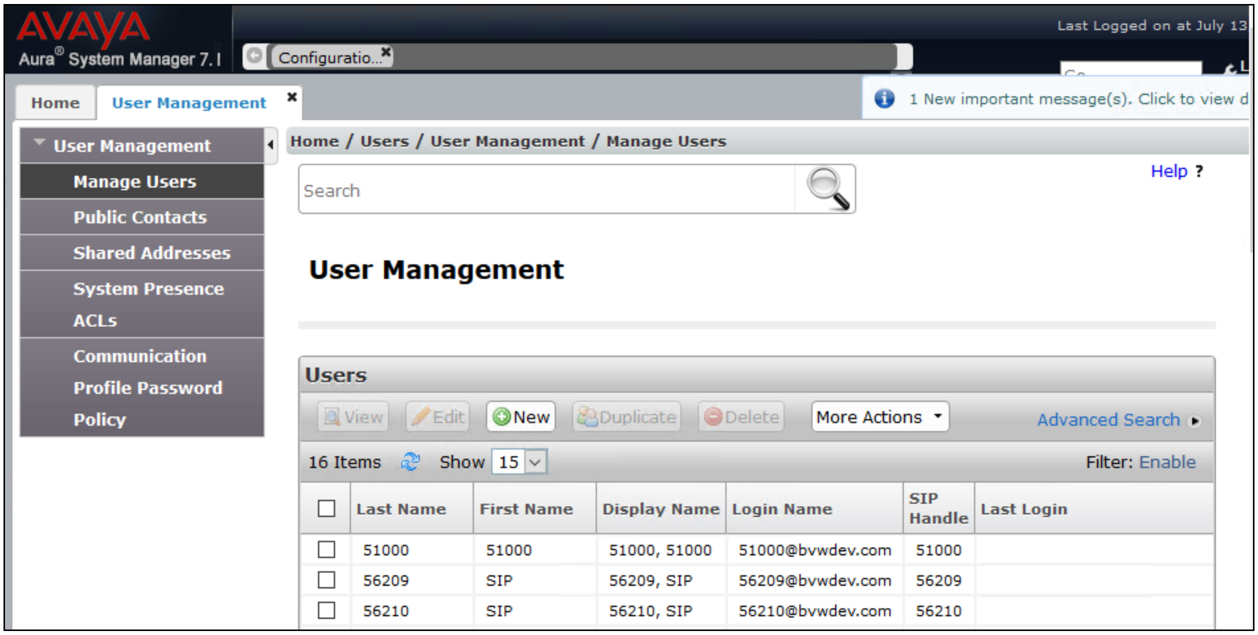
6.1. Launch System Manager

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.



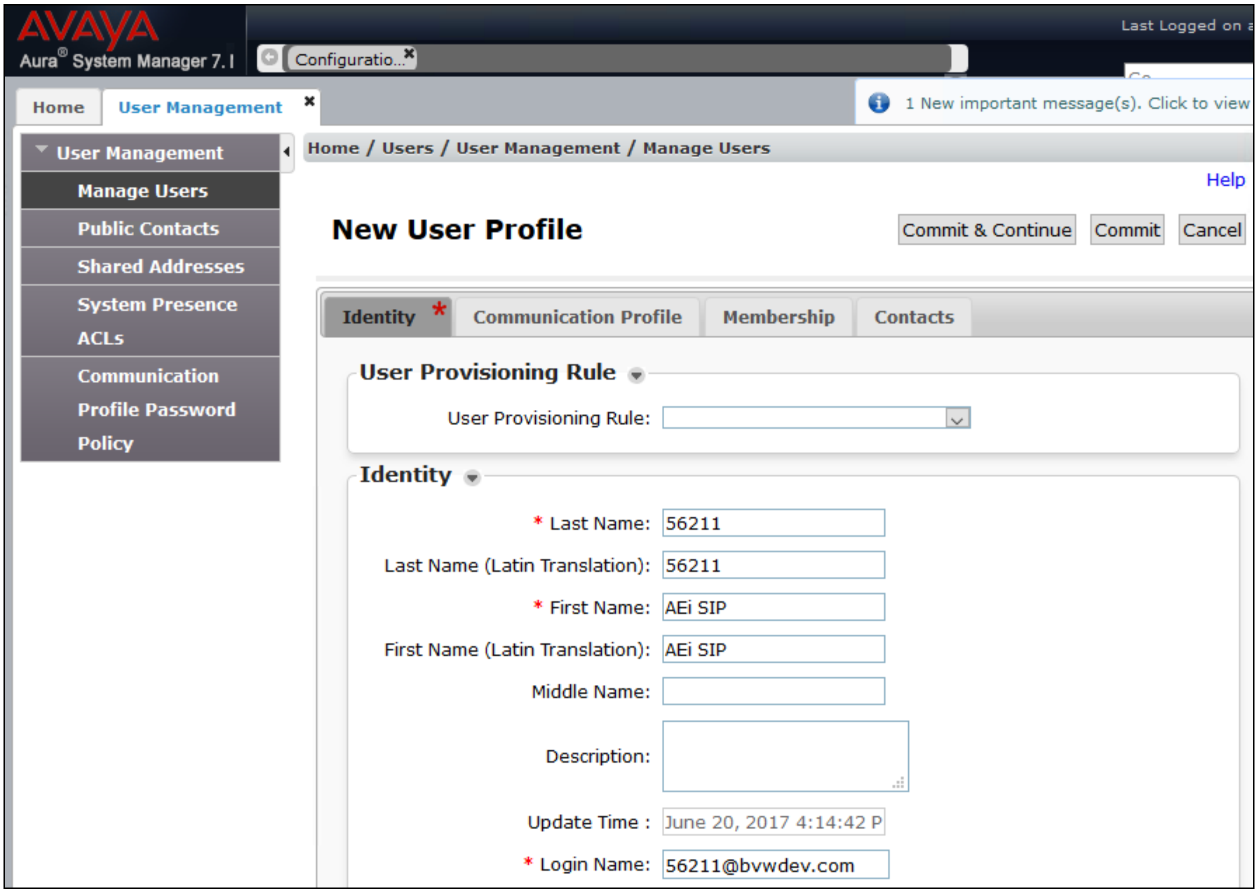
6.2. Administer SIP User

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.



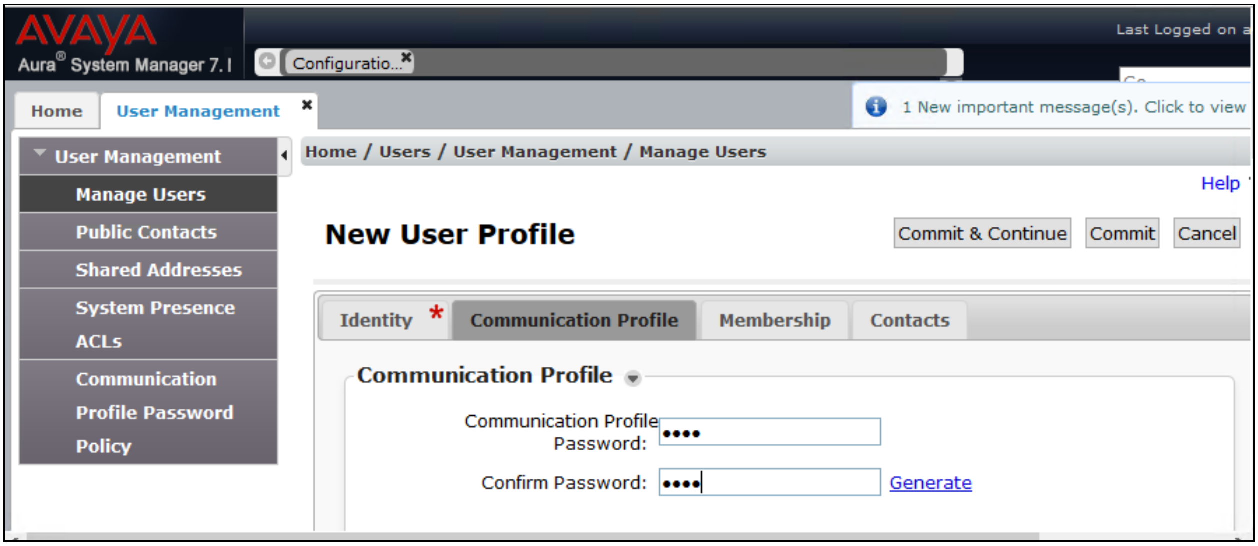
6.2.1. Identity

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 VM-2208-S(S) SIP extension and “*<domain>*” is the applicable SIP domain name from **Section 5.2**. Retain the default values in the remaining fields.



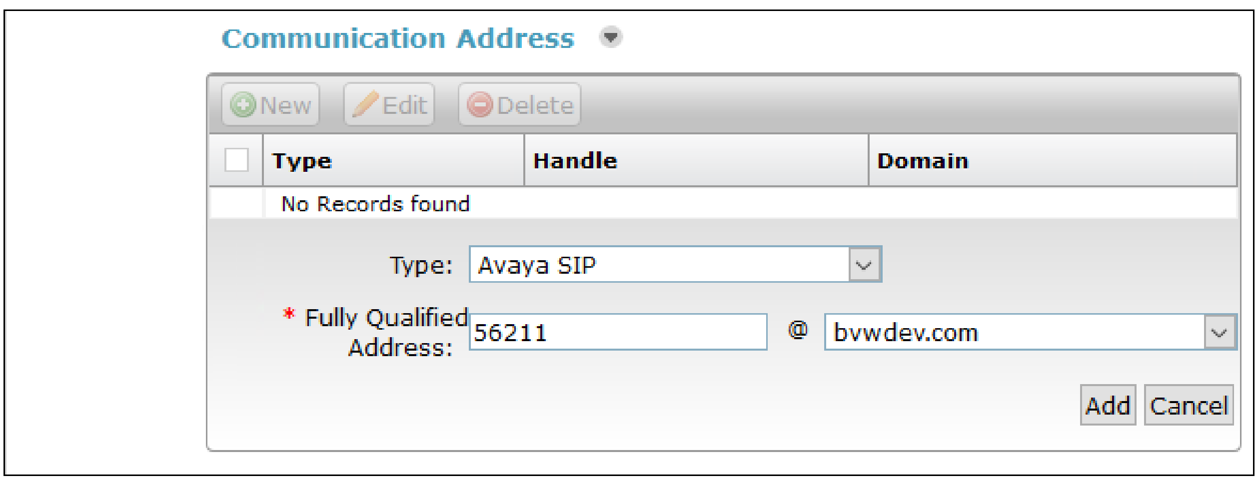
6.2.2. Communication Profile

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



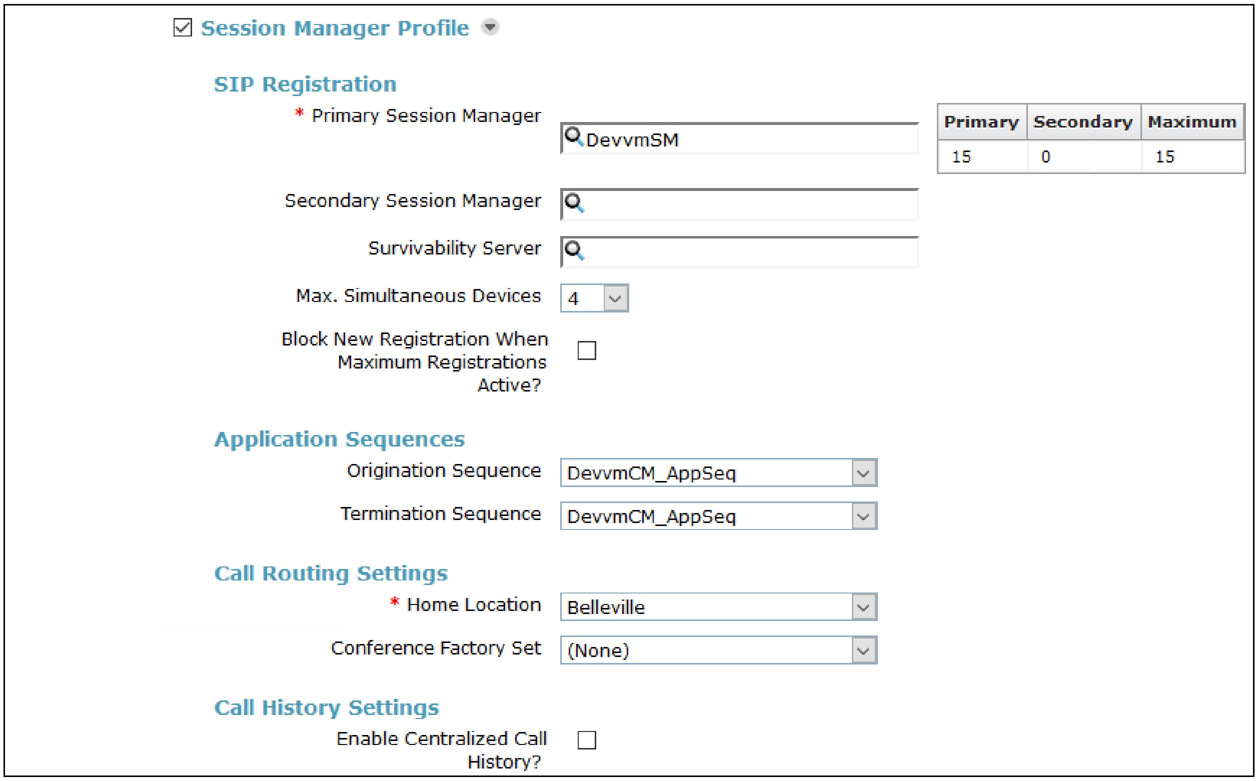
6.2.3. Communication Address

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

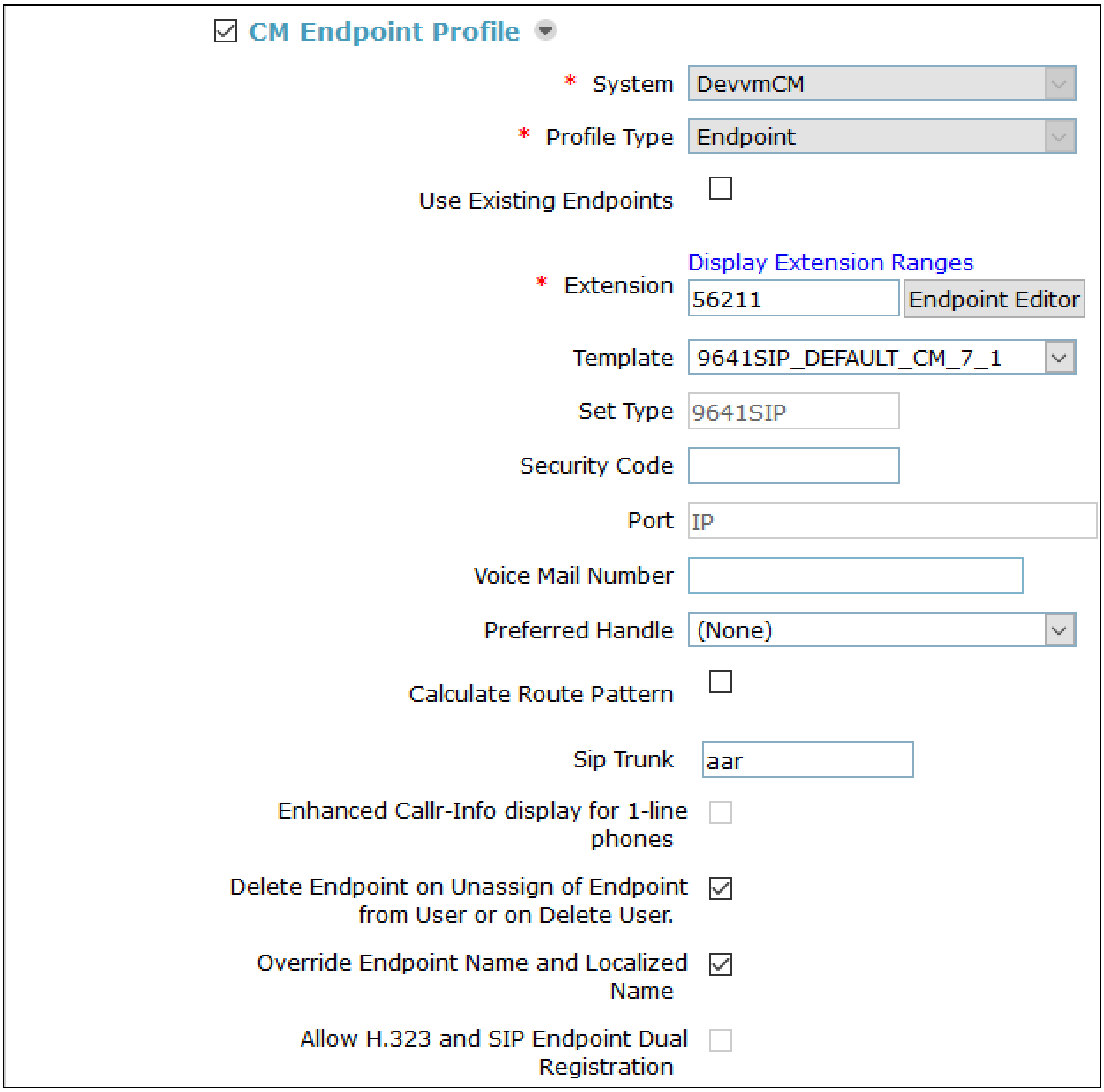
Scroll down to check and expand **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default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



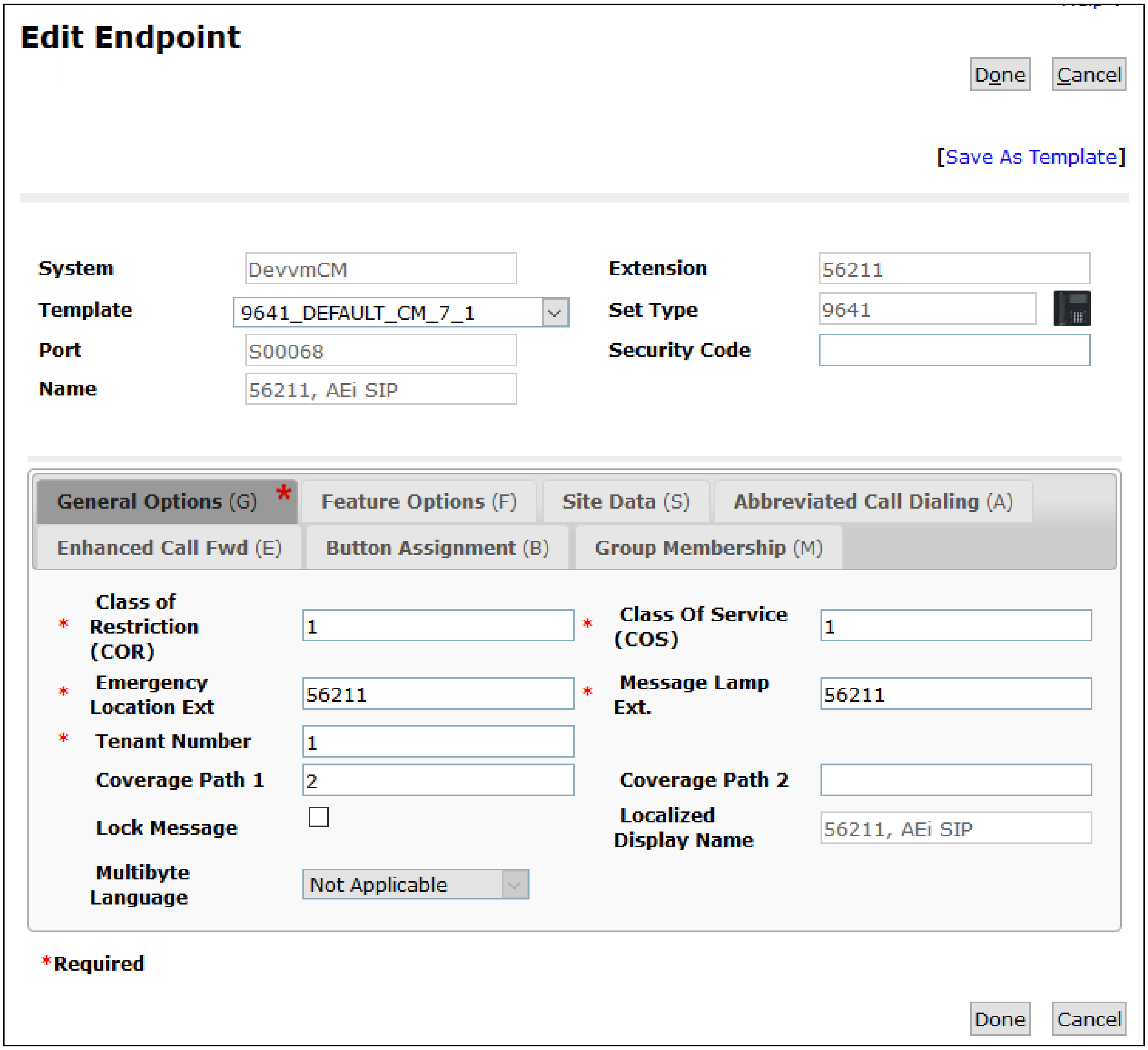
6.2.5. CM Endpoint Profile

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 *9641SIP_DEFAULT_CM_7_1*. Retain the default values in the remaining fields.

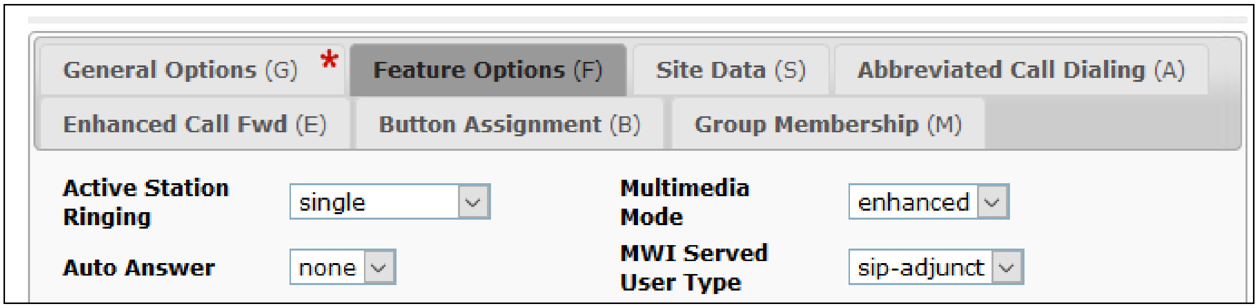
Click **Commit** to save the configuration (not shown).



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 *2* 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**.

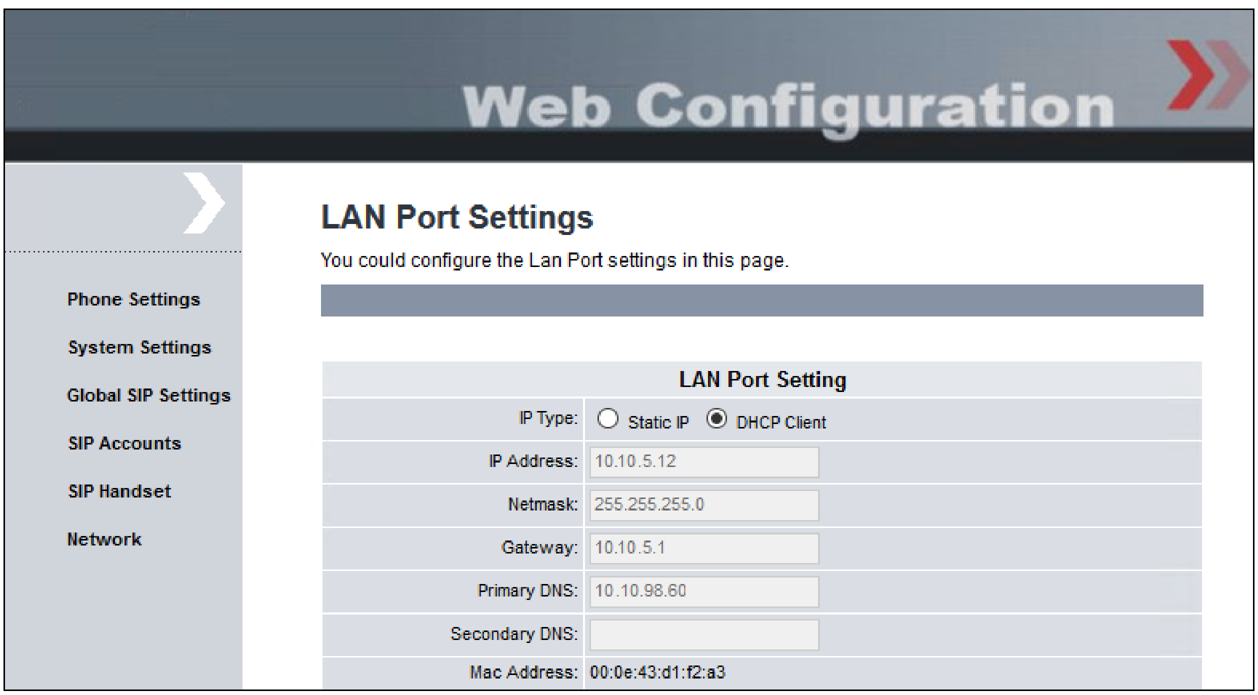


7. Configure AEi Communications VM-2208-S(S) SIP Telephone

Access the VM-2208-S(S) 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.



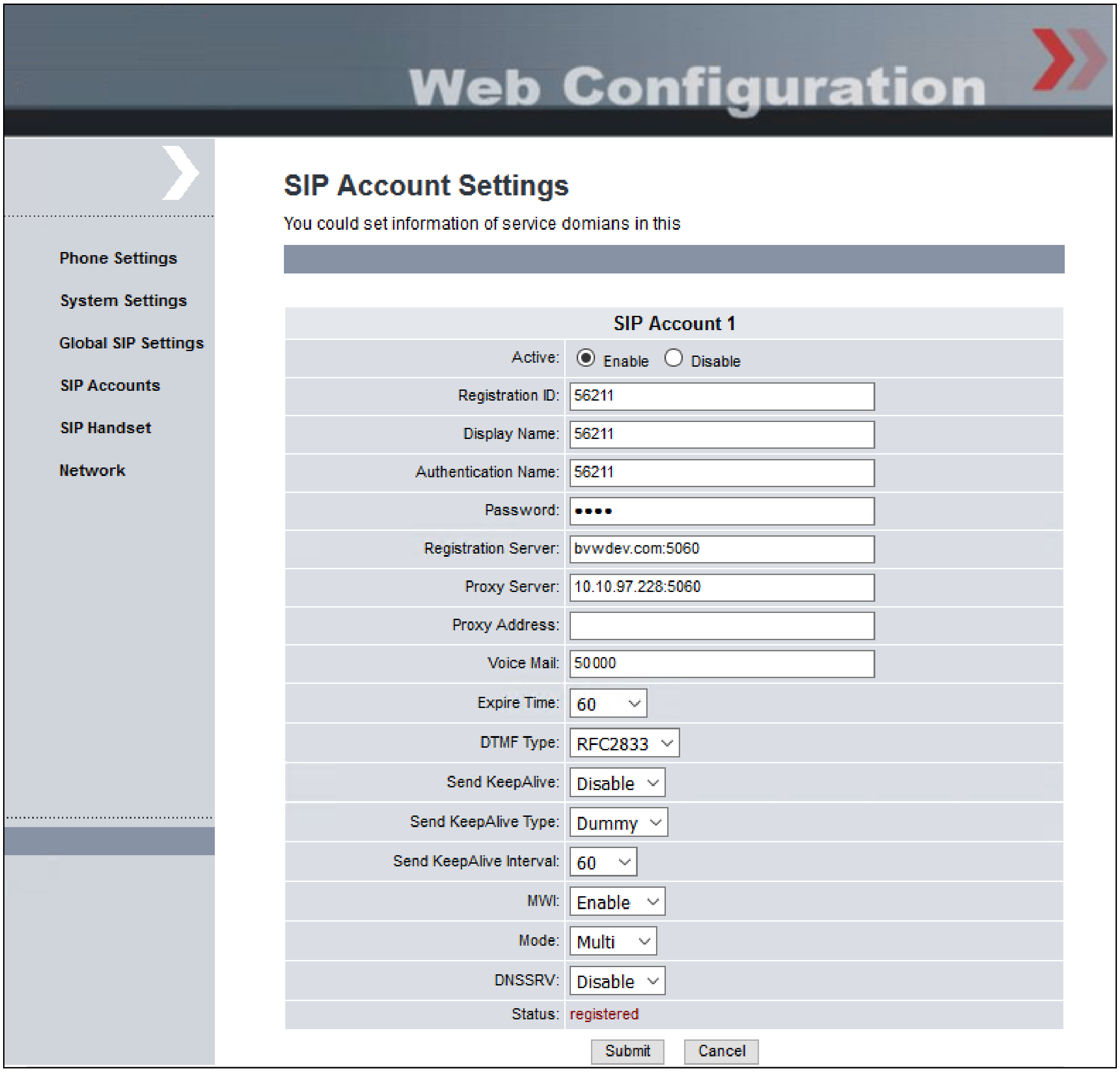
7.2. Administer SIP Accounts

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.

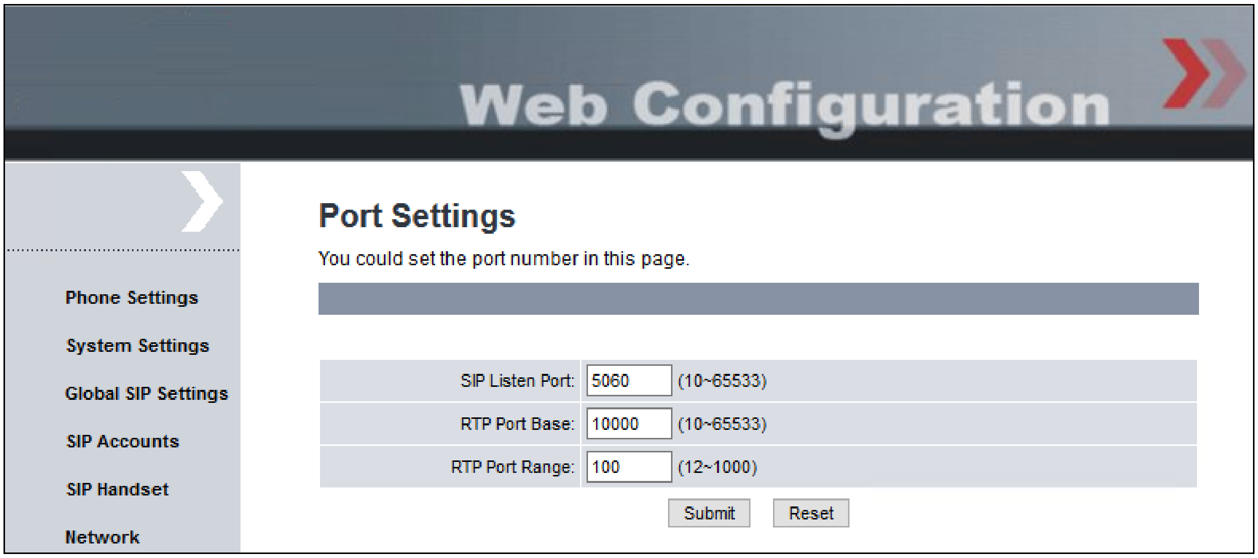
- **Registration ID:** Specify the Registration ID (e.g., *56211*, the SIP extension).
- **Display Name:** Specify the Display Name (e.g., *56211*, the SIP extension).
- **Authentication Name:** Specify the SIP extension of the VM-2208-S(S) SIP Telephone (e.g., *56211*).
- **Password:** Specify the SIP password configured in **Section 6.2.2**.
- **Registration Server:** Set to the domain name and port (e.g., *bvwdev.com:5060*).
- **Proxy Server:** Set to the Session Manager IP address and port (e.g., *10.10.97.228:5060*).
- **Voice Mail:** Specify the voicemail pilot number (e.g., *50000*).
- **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.

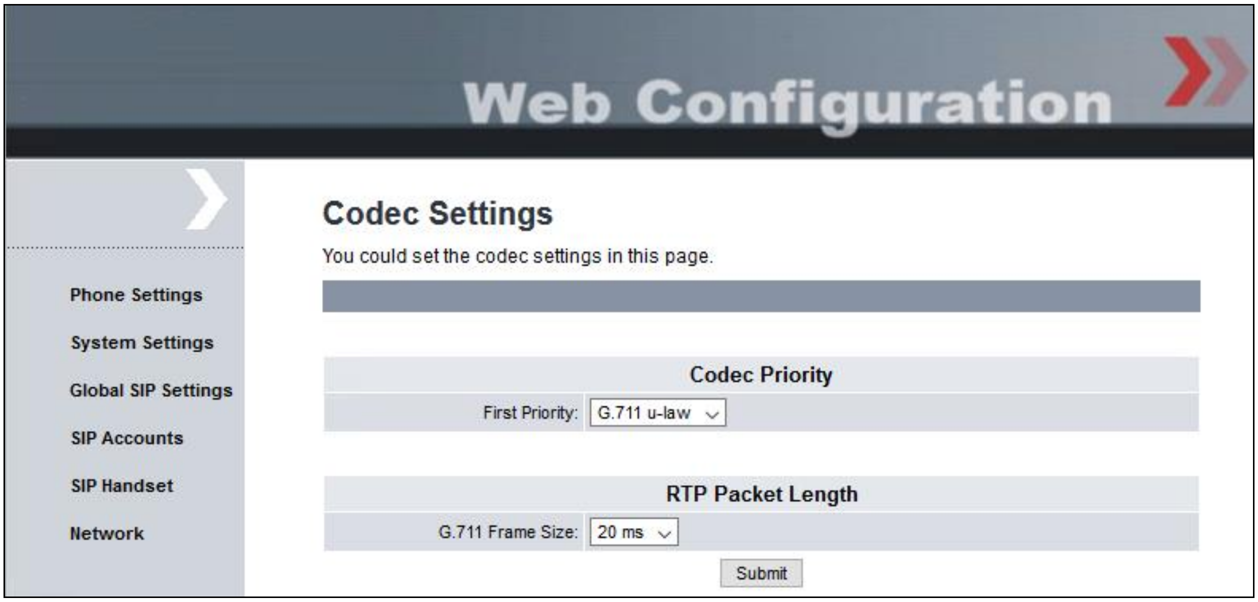


7.3. Administer Global SIP Settings

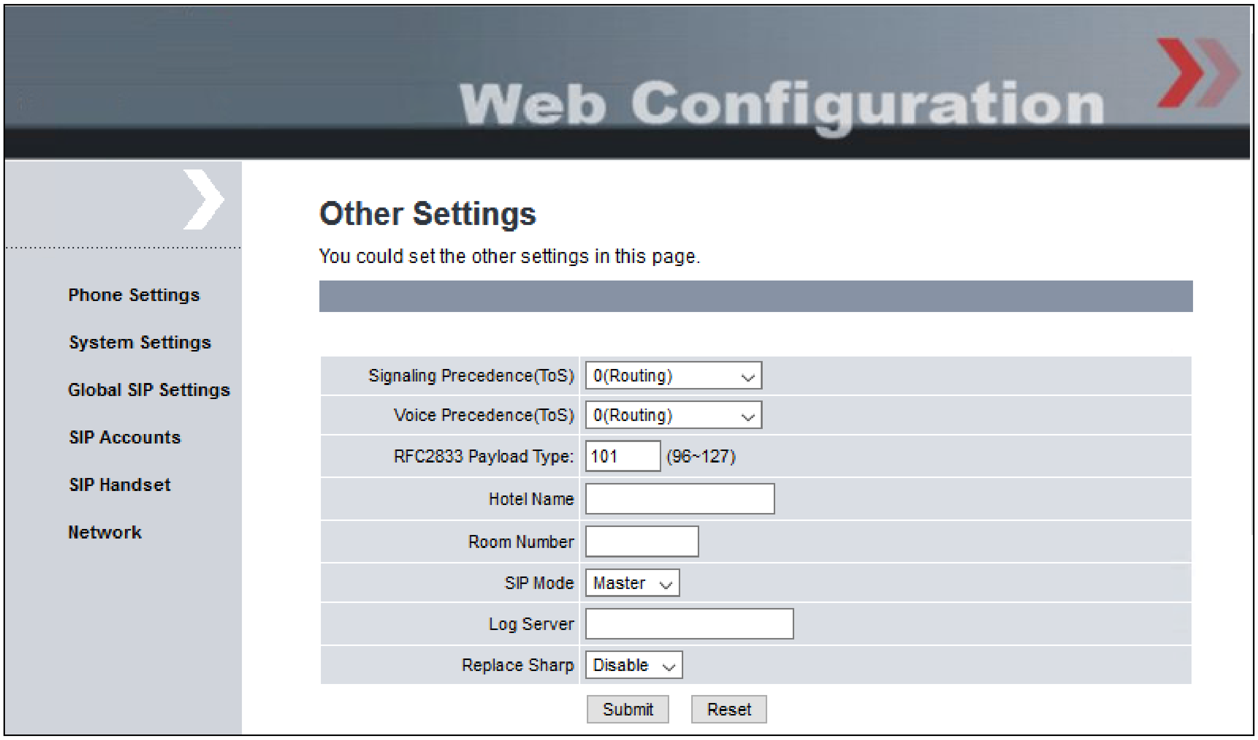
Navigate to **Global SIP Settings → Port Settings** and verify the SIP Listen Port being used (e.g., *5060*).



Navigate to **Global SIP Settings → Codec Settings** to verify the codec priority. In this example, the first priority is *G.711u-law*. AEi Communications VM-2208-S(S) SIP Telephones supports G.711.



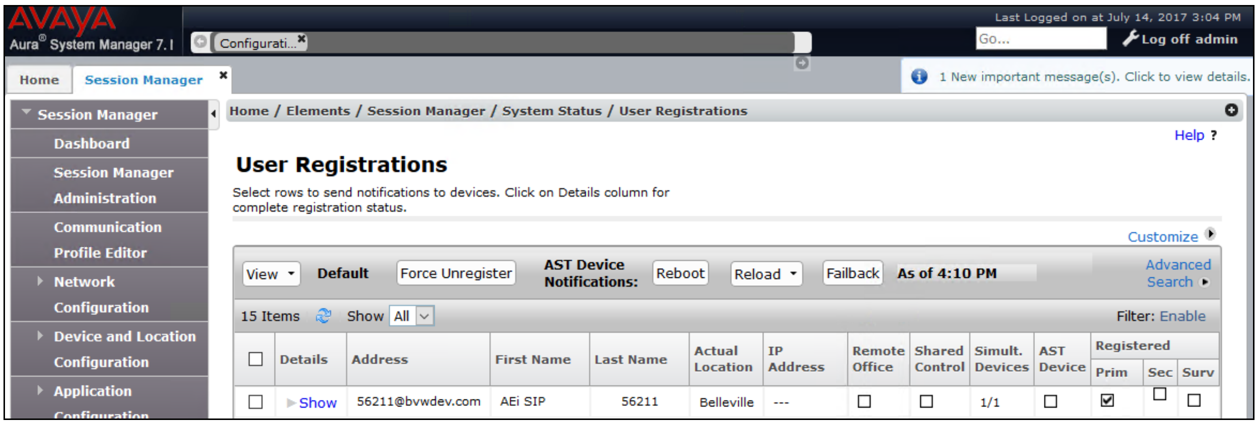
Navigate to **Global SIP Settings → Other Settings**, and retain default values.



8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the AEi Communications VM-2208-S(S) SIP Telephone with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

1. Verify that the VM-2208-S(S) SIP 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 VM-2208-S(S) web interface seen in **Section 7.2**.



2. Verify basic telephony features by establishing calls between a VM-2208-S(S) SIP Telephone with another VM-2208-S(S) SIP Telephone and also with Avaya deskphones.

9. Conclusion

These Application Notes have described the administration steps required to integrate the AEi Communications VM-2208-S(S) SIP Telephone with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The AEi Communications VM-2208-S(S) SIP 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® Session Manager*, Release 7.1, Issue 1 May 2017
2. *Deploying Avaya Aura® System Manager*, Release 7.1, Issue 1 May 2017
3. *Administering Avaya Aura® System Manager for Release 7.1*, Release 7.1, Issue 2 May 2017
4. *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 7.1, Issue 1 May 2017

The following document was provided by AEi Communications.

1. *Configuring Hospitality SVM-2x08-S SIP IP Phone*, Version 1.0, Date: 28/07/16

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.