Avaya Solution & Interoperability Test Lab

Application Notes for Bittel UNO Voice SIP Telephone with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate Bittel UNO Voice SIP Telephones with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Bittel UNO Voice SIP Telephones are hotel guest phones that provide the following features: speakerphone, hold, redial, message waiting indicator (MWI), and programmable buttons. In the compliance test, Bittel UNO Voice SIP Telephones successfully registered with Avaya Aura® Session Manager, established calls with other Avaya SIP and H.323 telephones, and executed telephony and hospitality features using Avaya Aura® Communication Manager Feature Name Extensions (FNEs).

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 Bittel UNO Voice SIP Telephones with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Bittel UNO Voice SIP Telephones are hotel guest phones that provide the following features: speakerphone, hold, redial, message waiting indicator (MWI), and programmable buttons. In the compliance test, Bittel UNO Voice SIP Telephones successfully registered with Avaya Aura® Session Manager, established calls with other Avaya SIP and H.323 telephones, and executed telephony and hospitality features using Avaya Aura® Communication Manager 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 Bittel UNO Voice and Avaya SIP and H.323 telephones and exercising basic telephony features, such as hold and mute, and hospitality features, including wake up calls and updating housekeeping status for a guest's room. 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 Bittel UNO Voice SIP 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 Bittel UNO Voice with Session Manager.
- Calls between Bittel UNO Voice and Avaya SIP and H.323 telephones with Direct IP-IP Media (Shuffling) enabled and disabled.
- G.711 and G.729 codec support.
- Proper recognition of DTMF tones.
- Basic telephony features including Hold, Mute, and Redial.
- Long call duration and long hold duration.
- Extended telephony features using Communication Manager Feature Name Extensions (FNEs) 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 Bittel UNO Voice.
- Proper system recovery after a restart of the Bittel UNO Voice and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observations noted:

- The Bittel UNO Voice SIP Telephone Model HA9888 (67) TSD-IP, which was used for the compliance test, does not support multiple calls, caller ID display, and initiating a call transfer or conference call.

- While Bittel UNO Voice SIP Telephone is dialing a call, DTMF tones are heard by the caller. After the call is established, DTMF tones are not heard when the keypad is pressed. For example, when logging into voicemail, DTMF tones are not heard by the caller, but the DTMF tones are sent to the voicemail system and the user is able to log in.

- Bittel UNO Voice does not provide an indication to the caller when the Long Hold Recall Timer expires. When Communication Manager sends the re-INVITE message to Bittel UNO Voice when the Long Hold Recall Timer expires, Bittel UNO responds to the message but doesn't provide a reminder to the caller that there is a call on hold.

- If a call attempt fails for whatever reason, such as calling a busy telephone, dialing invalid number, or using an unsupported codec, Bittel UNO Voice plays busy tone for 30 seconds and then hangs up or the caller may hang up.

2.3. Support

For technical support on the Bittel UNO Voice SIP Telephone, contact Bittel Customer Service and Support via phone, email, or website.

- **Phone:** 86-633-2212103
- **Email:** info@bittelgroup.com
- **Web:** http://www.bittelphone.com/en/products/support.asp

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 on an Avaya S8800 Server with a G650 Media Gateway. Communication Manager was configured as an Evolution 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 serving as the voicemail system.
- Avaya 9600 Series SIP and H.323 Telephones.
- Bittel UNO Voice SIP Telephones.

Bittel UNO Voice SIP Telephones registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.

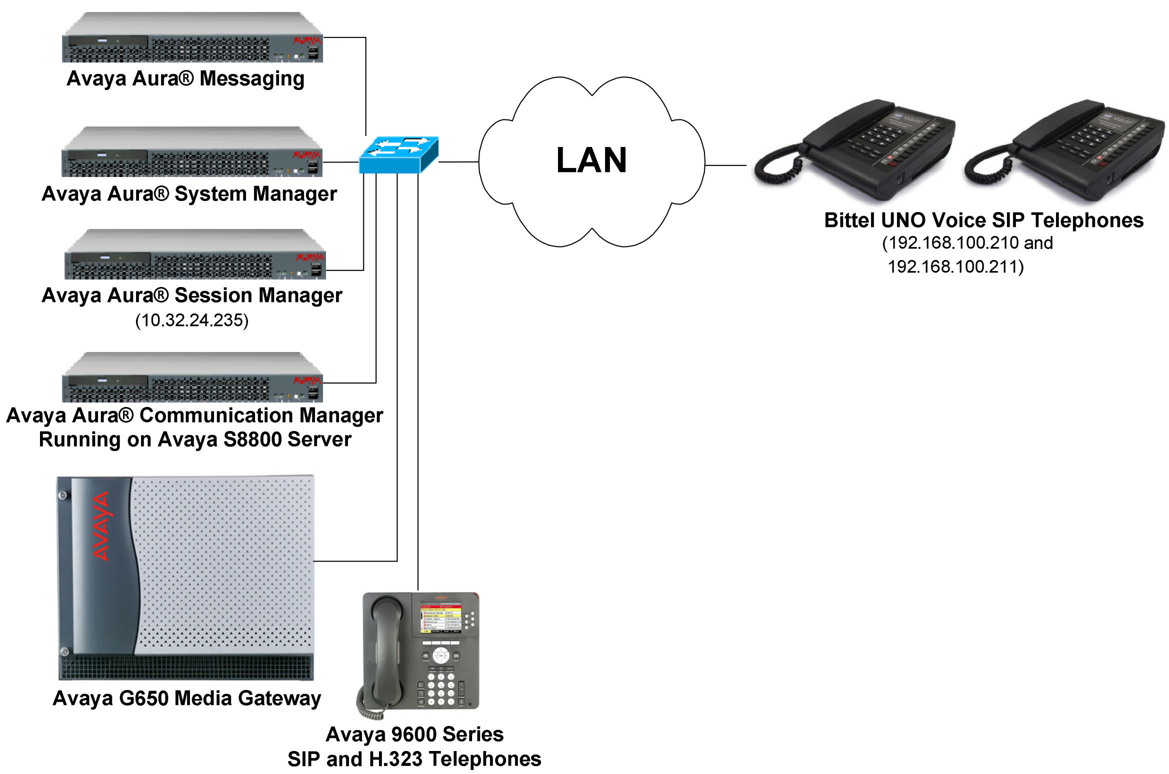


Figure 1: Avaya SIP Network with Bittel UNO Voice SIP Telephones

3.1. SIP Call Flows

Bittel UNO Voice SIP Telephones originate a call by sending a call request (SIP INVITE message) to Session Manager, which then routes the call over a SIP trunk to Communication Manager for origination services. If the call is destined for another local SIP phone, Communication Manager routes the call back over the SIP trunk to Session Manager for delivery to the destination SIP phone. If the call is destined for a H.323 telephone, Communication Manager routes the call directly to the H.323 endpoint.

For a call arriving at Communication Manager that is destined for a Bittel UNO Voice SIP Telephone, Communication Manager routes the call over the SIP trunk to Session Manager for delivery to the Bittel UNO Voice SIP Telephone.

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 running on an Avaya S8800 Server with and G650 Media Gateway	6.3 SP 1 (R016x.03.0.124.0 w/Patch 20850)
Avaya Aura® Session Manager	6.3 SP 3 (6.3.3.0.633004)
Avaya Aura® System Manager	6.3.3 (Build No. – 6.3.0.8.5682-6.3.8.1814, Software Update Revision No: 6.3.3.5.1719)
Avaya Aura® Messaging	6.2 SP 2
Avaya 9600 Series IP Telephones	3.2 (H.323) 2.6.9.1 (SIP)
Bittel UNO Voice SIP Telephone	10.19_0.94.003

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 Bittel UNO Voice 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  11
                                OPTIONAL FEATURES

     G3 Version: V16                           Software Package: Enterprise
       Location: 2                              System ID (SID): 1
       Platform: 28                             Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 65000 118
                                      Maximum Stations: 41000 21
                              Maximum XMOBILE Stations: 41000 0
                    Maximum Off-PBX Telephones - EC500: 41000 0
                    Maximum Off-PBX Telephones -   OPS: 41000 9
                    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 *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 G650 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: 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? y
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 34
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        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 Bittel UNO Voice 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 Bittel UNO Voice SIP Telephone supports G.711 and G.729 codecs.

```
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:
 6:
 7:
```

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 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: *70
   Housekeeping Status (Client Room) Access Code: *71
   Housekeeping Status (Client Room) Access Code: *72
   Housekeeping Status (Client Room) Access Code: *73
   Housekeeping Status (Client Room) Access Code: *74
   Housekeeping Status (Client Room) Access Code: *75
   Housekeeping Status (Client Room) Access Code: *76
       Housekeeping Status (Station) Access Code: *77
       Housekeeping Status (Station) Access Code: *78
       Housekeeping Status (Station) Access Code: *79
       Housekeeping Status (Station) Access Code: *80
          Verify Wakeup Announcement Access Code: *81
                Voice Do Not Disturb Access Code:
```

5.3.2. Administer Feature Name Extensions (FNEs)

Prior to dialing the wakeup call or housekeeping status access codes, 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 or housekeeping status access code may be dialed.

```
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): 78117
      Extended Group Call Pickup:
          Held Appearance Select: 78118
          Idle Appearance Select: 78119
              Last Number Dialed: 78120
            Malicious Call Trace:
     Malicious Call Trace Cancel:
             Off-Pbx Call Enable:
            Off-Pbx Call Disable:
```

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 Bittel SIP 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                  y  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  n  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  n  y  y  y  y  y  y  y  y  n  n  n  y  y  y
 Off-hook Alert                 y  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Client Room                    y  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  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  n  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  y  y  y  n  n  n
```

6. Configure Avaya Aura® Session Manager

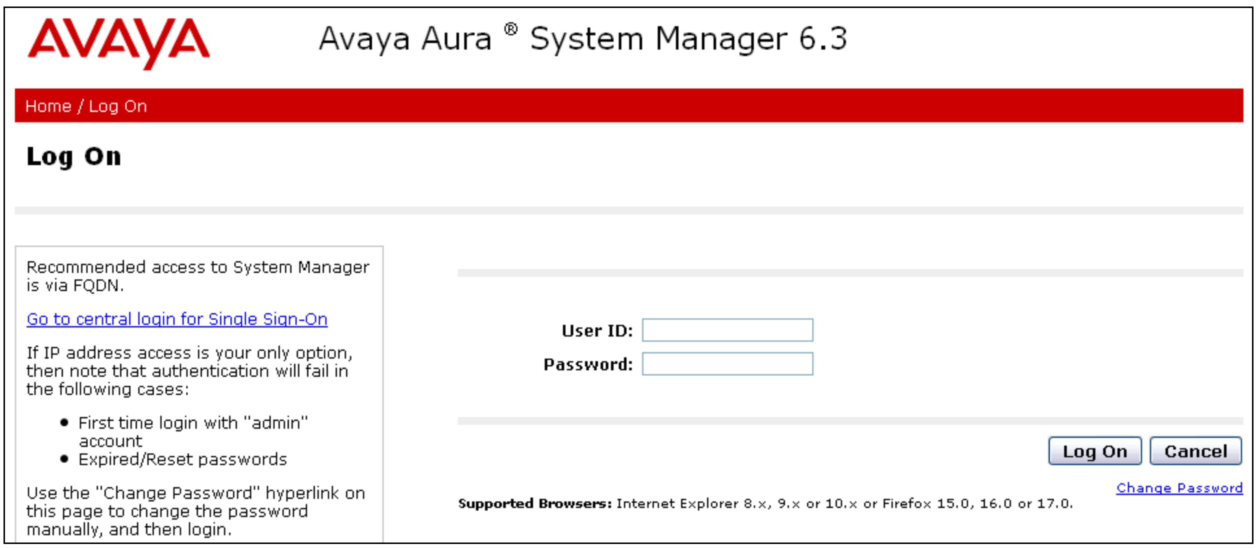
This section provides the procedures for configuring Session Manager. The procedures include the following areas:

- 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 Bittel UNO Voice SIP Telephone.

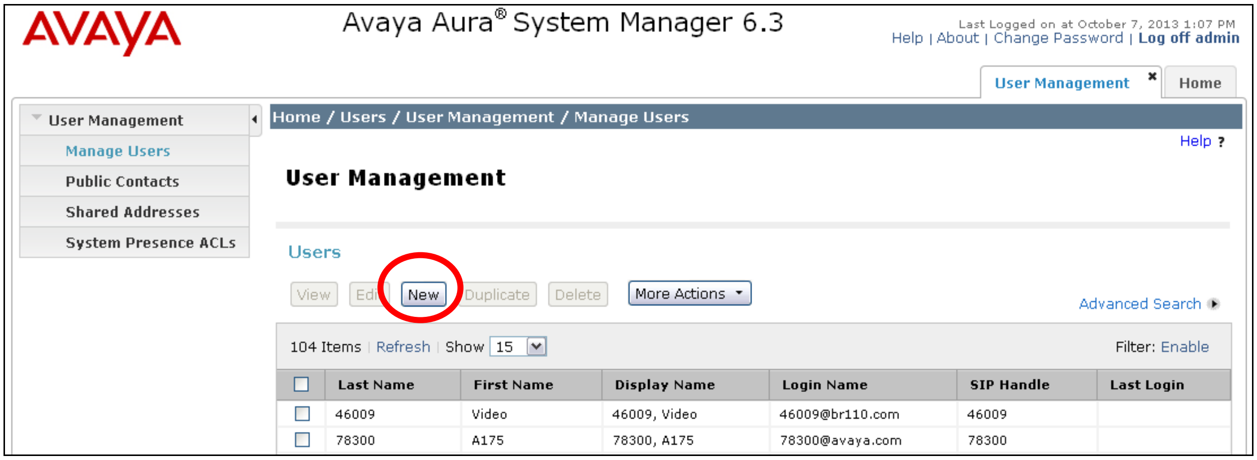
6.1. Launch System Manager

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.



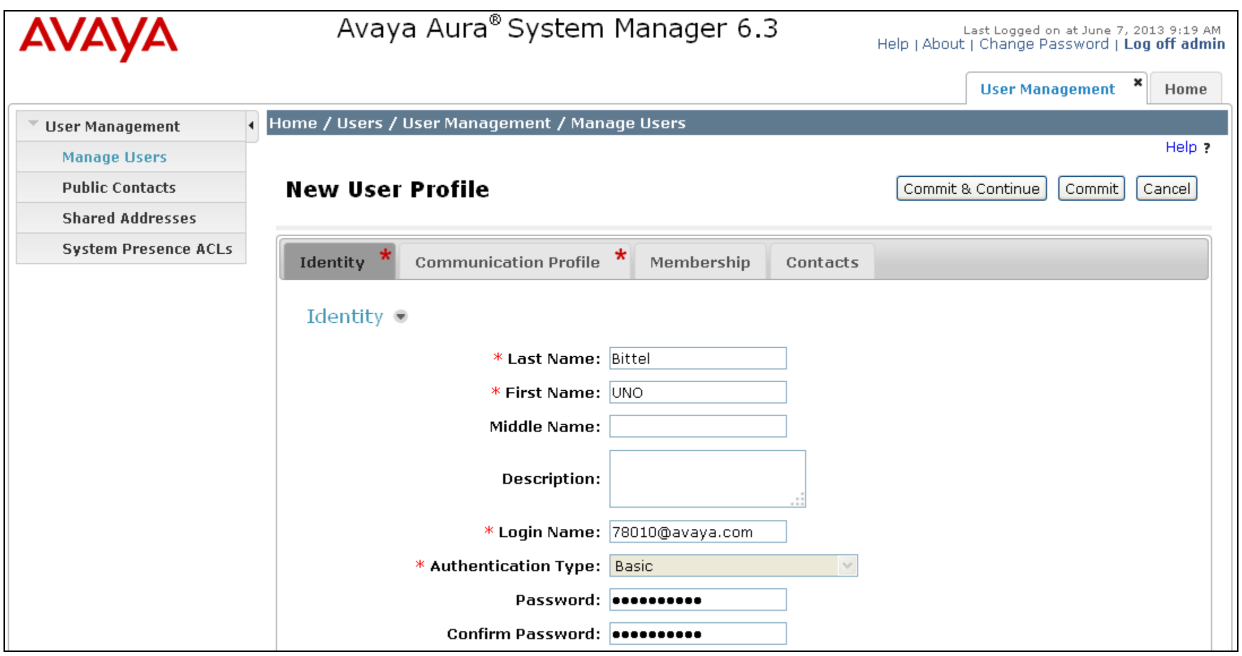
6.2. Administer SIP User

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.



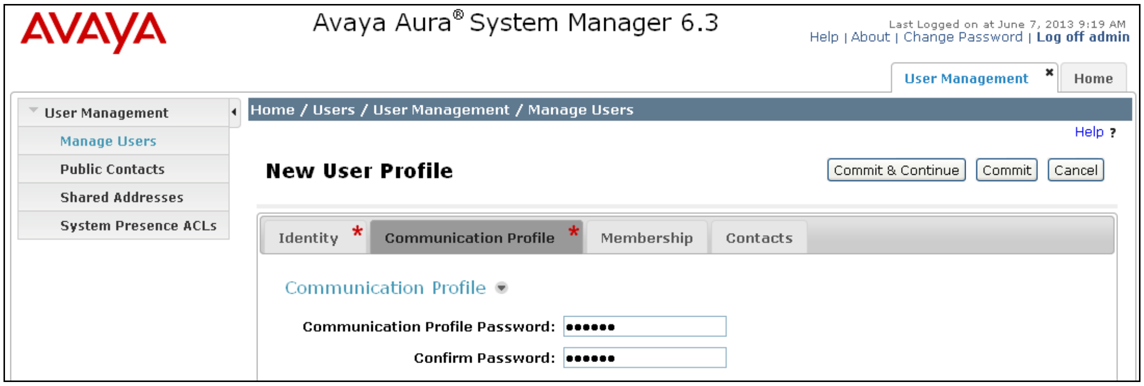
6.2.1. Identity

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 Bittel UNO Voice SIP extension and "*<domain>*" is the applicable SIP domain name from **Section 5.2**. For **Password** and **Confirm Password**, enter the appropriate credentials for System Manager. Retain the default values in the remaining fields.

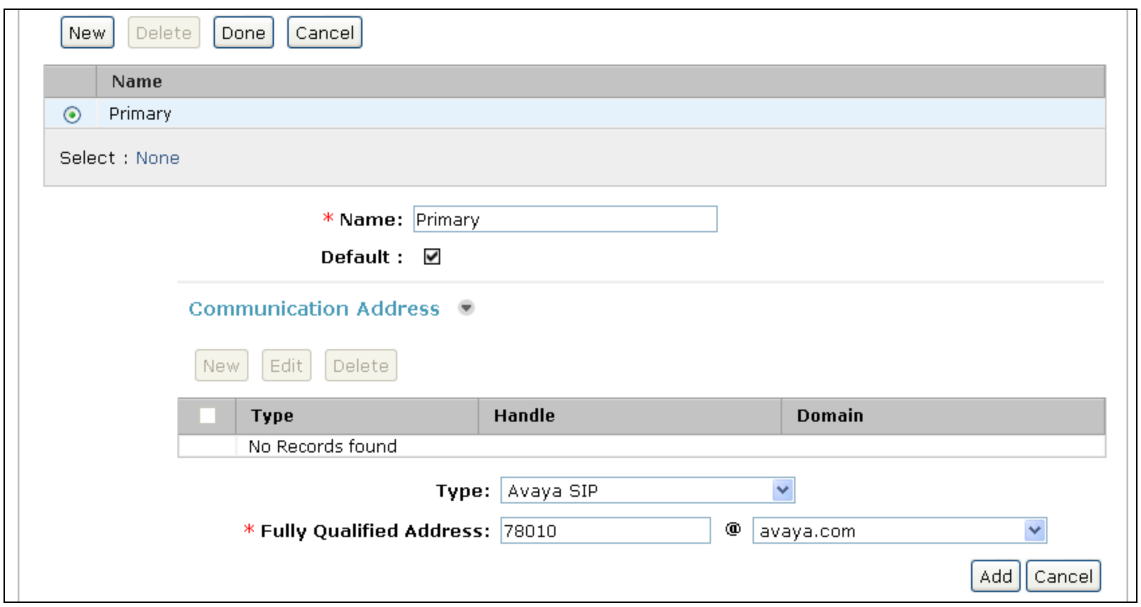


6.2.2. Communication Profile

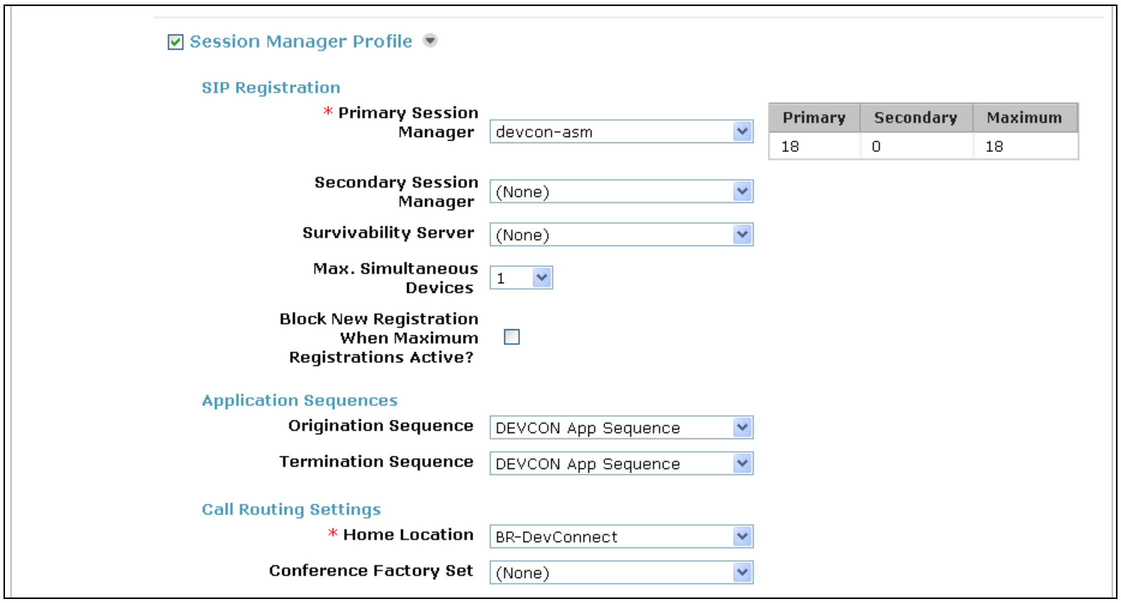
Select the **Communication Profile** tab. For **Communication Profile Password** and **Confirm Password**, enter the desired password for the SIP user to use for registration.



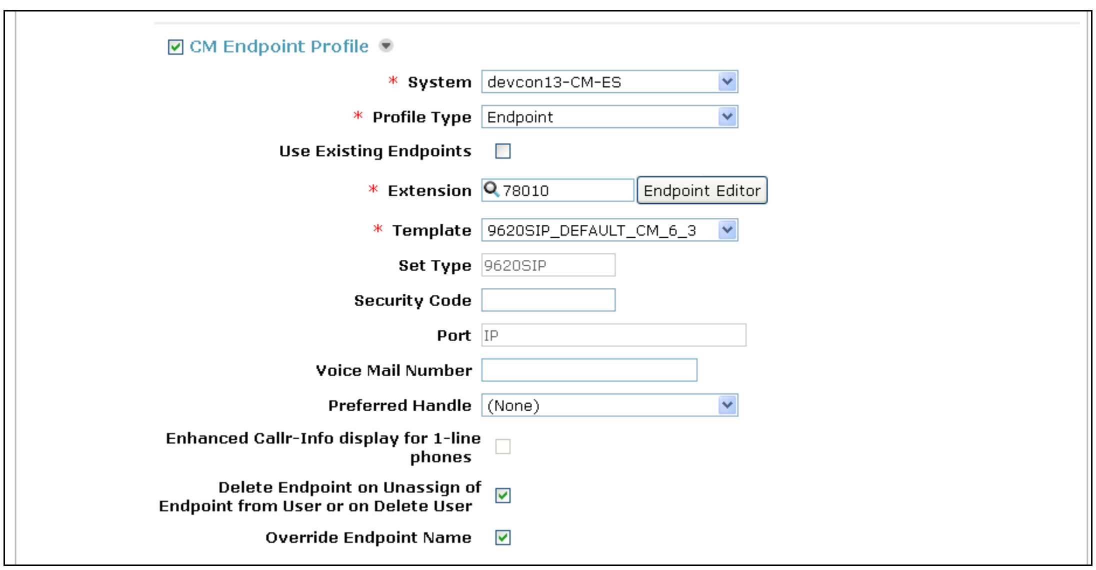
In the **Communication Address** sub-section, click **New** to add a new entry. The **Communication Address** sub-section is updated with additional fields, as shown below. For **Type**, retain "Avaya SIP". For **Fully Qualified Address**, enter and select the SIP user extension and domain name to match the login name from **Section 6.2.1**. Click **Add**.



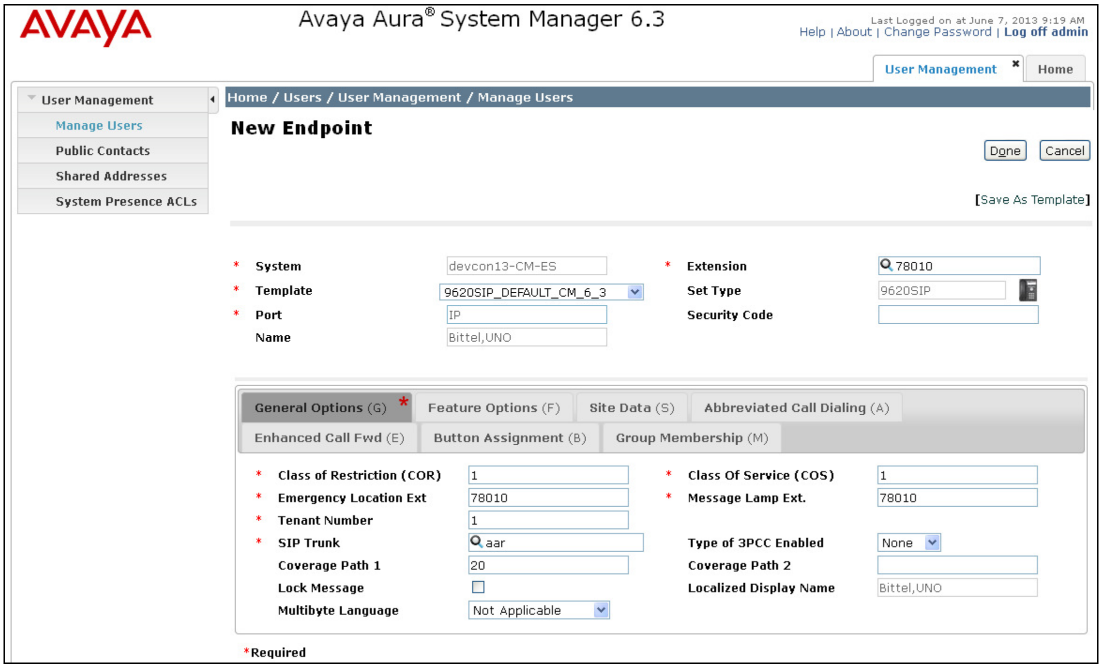
Scroll down to check and expand **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default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



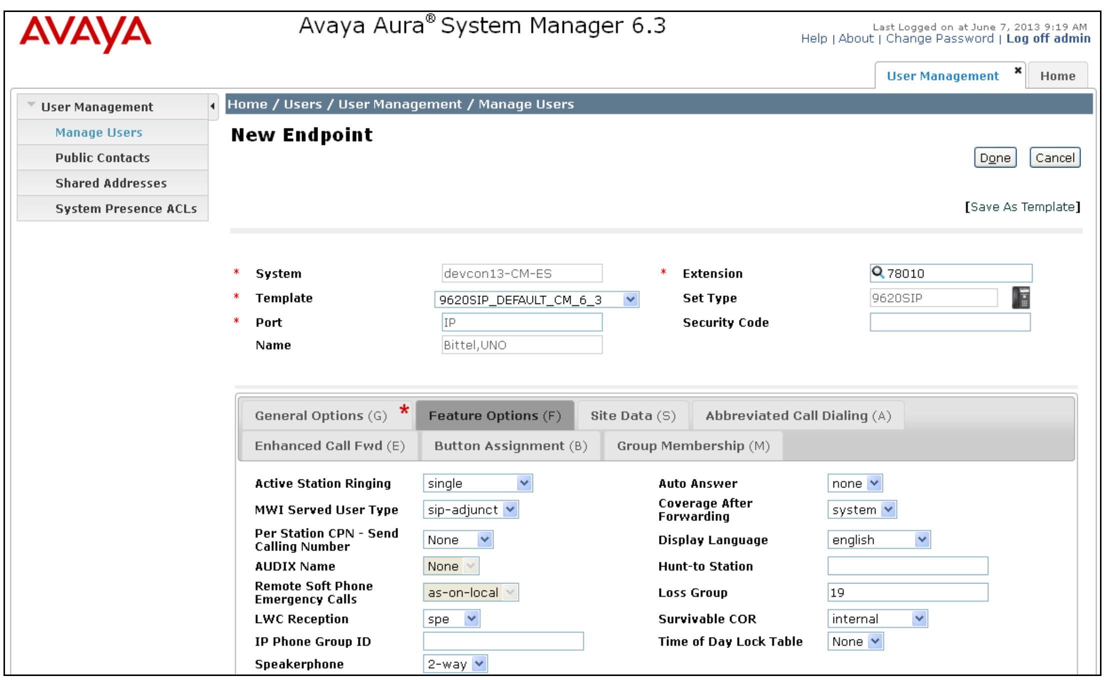
Scroll down to check and expand **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.2.1**. For **Template**, select *DEFAULT_9620SIP_CM_6_3*. For **Port**, click and select *IP*. Retain the default values in the remaining fields. Click **Commit** to save the configuration (not shown).



In the **CM Endpoint Profile** sub-section, 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 20 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. Click **Done** when complete, followed by **Commit** on the previous page.

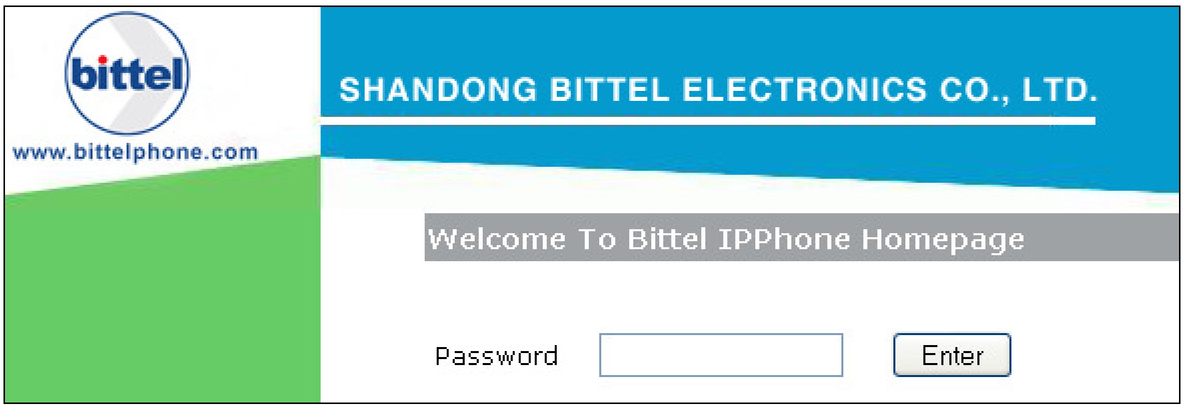


7. Configure Bittel UNO Voice SIP Telephone

Access the Bittel UNO Voice web interface by using the URL “https://ip-address” 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 **Enter**.

Note 1: To access the Bittel UNO Voice web interface initially, hold down the **Mute** button on the SIP phone to hear the the phone’s IP address. Use this IP address to first access the configuration screens. Afterwards, the IP address may be changed.

Note 2: When configuration changes are made to the Bittel UNO Voice SIP Telephone, the phone will reboot.

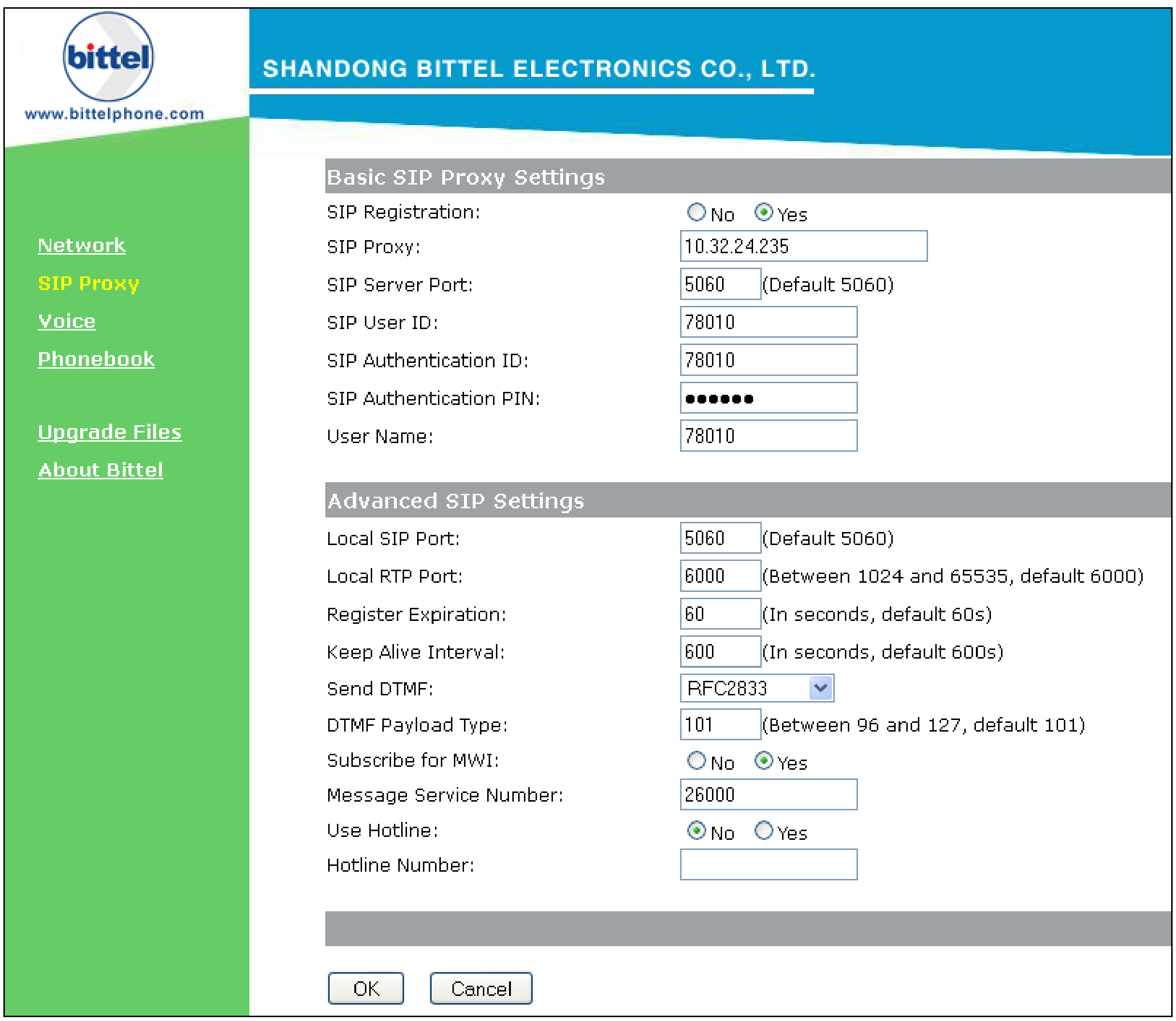


Select **Network** in the left pane and configure the SIP phone's network settings as shown below. In this example, a static IP is configured. Click **OK**.

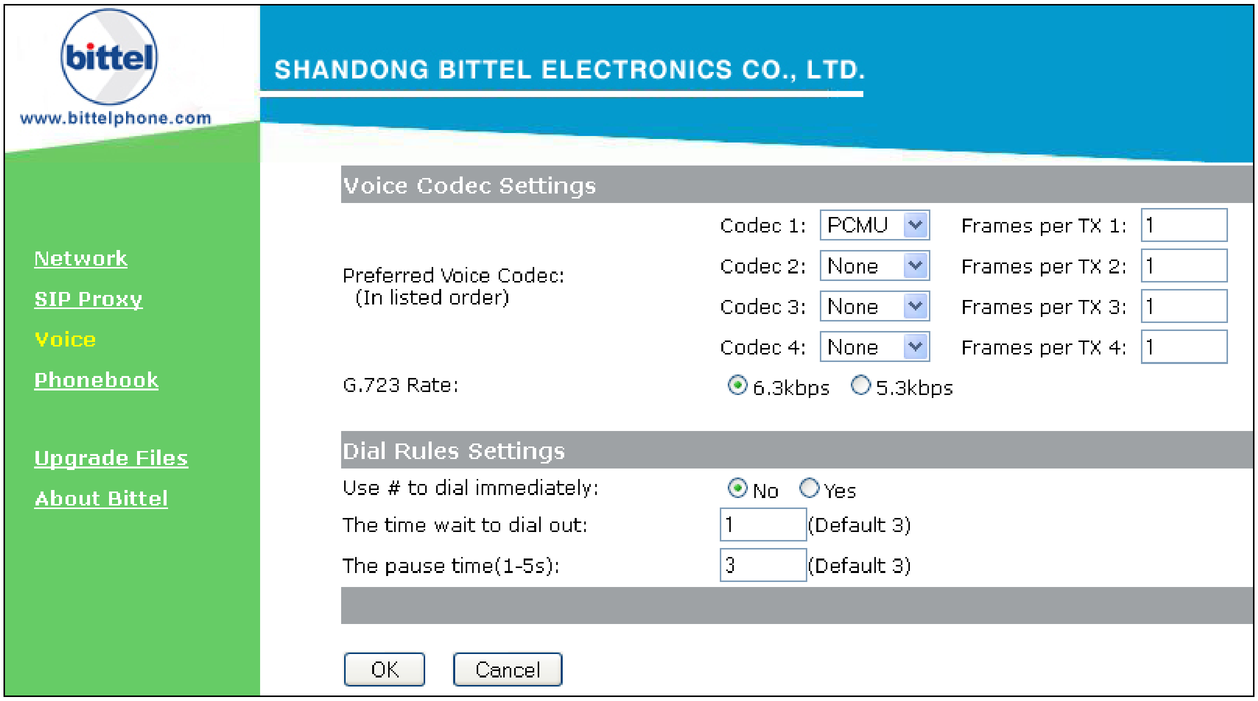
Navigate to the **SIP Proxy** webpage as shown below. Under the **Basic SIP Proxy Settings** section, configure the following parameters.

- **SIP Registration:** Set to *Yes*.
- **SIP Proxy:** Set to the Session Manager IP address (e.g., *10.32.24.235*).
- **SIP Server Port:** Set to the SIP port the phone listens on (e.g., *5060*).
- **SIP User ID:** Specify the User ID (e.g., *78010*, the SIP extension).
- **SIP Authentication ID:** Specify the SIP extension used to register with Session Manager from **Section 6.2.1** (e.g., *78010*).
- **SIP Authentication PIN:** Specify the **Communication Profile Password** configured in Session Manager in **Section 6.2.2**. This password is used for the SIP phone to register with Session Manager.
- **User Name:** Specify any user name for the SIP phone.

Under the **Advanced SIP Settings** section, enable **Subscribe for MWI** and set the **Message Service Number** to the voicemail pilot number (e.g., *26000*). Click **OK**.



Navigate to the **Voice** webpage as shown below and configure the desired codecs to be supported in the **Voice Codec Settings** section. In this example, PCMU was allowed. In the Dial Rules Settings section, specify **The time wait to dial out**. In this example, the timer was set to 1 second. Using this setting, Bittel UNO Voice will wait 1 second for more digits and if not received, it will dial the call. If a higher inter-digit timeout is required, increase this field. Click **OK**.



Navigate to the **Phonebook** webpage as shown below. This page is used to configure the programmable button. In this example, the following programmable buttons were configured.

M9 (Operator button): This button was set to the operator extension number (e.g., *77301*).

M2 (Wake Up button): This button was configured to allow a guest to set the wake up feature. In this example, *78119* is the FNE for **Idle Appearance Select**. Dialing this FNE allows the guest to receive dial tone. The 'p' instructs the SIP phone to wait 1 seconds prior to dialing the *70, which is the Automatic Wakeup Call access code configured in **Section 5.3.1**. Recall that **The time wait to dial out** was to 1 second in the webpage above. After the Automatic Wakeup Call access code is dialed, 3-burst confirmation is played, and the guest should then specify the wakeup time in *hhmm* format, where *hh* is the hour and *mm* are the minutes.

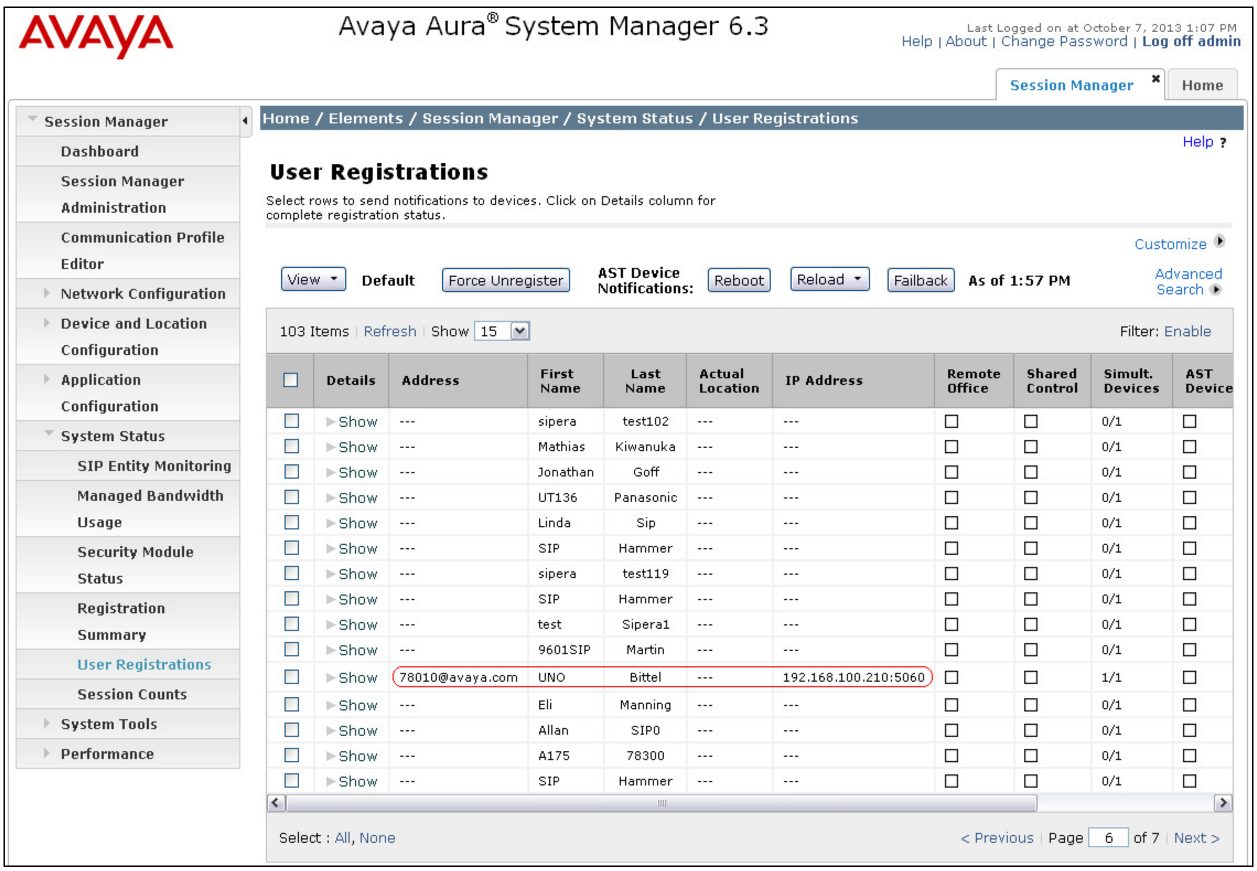
M1 (Messages): Specify the voicemail system pilot number (e.g., *26000*). When this button is pressed, the voicemail system will be dialed to allow the guest to retrieve their messages.

Click **OK** when done.

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the Bittel UNO Voice SIP Telephone with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

1. Verify that the Bittel UNO Voice 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.



2. Verify basic telephony features by establishing calls between a Bittel UNO Voice SIP Telephone with and another phone.

9. Conclusion

These Application Notes have described the administration steps required to integrate the Bittel UNO Voice SIP Telephone with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Bittel UNO Voice SIP Telephone with 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 6.3, Issue 9, October 2013, Document Number 03-300509.

[2] *Administering Avaya Aura® Session Manager*, Release 6.3, Issue 3, October 2013.

[3] *Bittel User Manual Cover Models HA9888(66) TSD-IP, HA9888(67) TSD-IP, HS9888(68) TSD-IP, HS9888(69) TSD-IP,* Version V1.

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.