



## **Avaya Solution & Interoperability Test Lab**

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# **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.

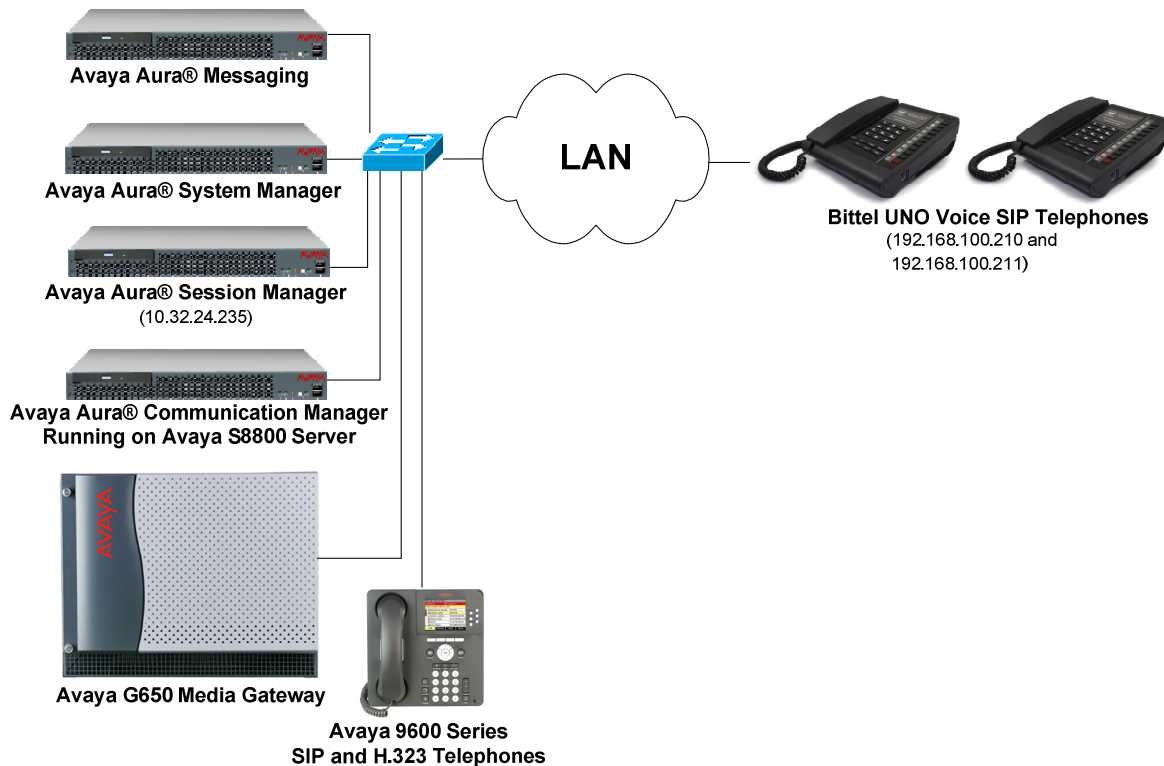
- **Phone:** 86-633-2212103
- **Email:** [info@bittelgroup.com](mailto:info@bittelgroup.com)
- **Web:** <http://www.bittelphone.com/en/products/support.asp>

### 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	
<b>Automatic Wakeup Call Access Code: *70</b>	
<b>Housekeeping Status (Client Room) Access Code: *71</b>	
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	
<b>Idle Appearance Select: 78119</b>	
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	
<b>Room Activated Wakeup With Tones? y</b>	
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 y n
Data Privacy	n n n n n 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
Console Permissions	y n 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
<b>Client Room</b>	y <b>y</b> 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
Call Forwarding Busy/DA	y y 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
Extended Forwarding All	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
Trk-to-Trk Transfer Override	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
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.

The screenshot shows the Avaya Aura System Manager 6.3 login interface. At the top, the Avaya logo is on the left and the title 'Avaya Aura® System Manager 6.3' is on the right. Below the title is a red navigation bar with the text 'Home / Log On'. The main heading is 'Log On'. On the left side, there is a box containing instructions: 'Recommended access to System Manager is via FQDN.' followed by a link 'Go to central login for Single Sign-On'. Below this, it states 'If IP address access is your only option, then note that authentication will fail in the following cases:' followed by a bulleted list: '• First time login with "admin" account' and '• Expired/Reset passwords'. At the bottom of this box, it says 'Use the "Change Password" hyperlink on this page to change the password manually, and then login.' In the center, there are two input fields: 'User ID:' and 'Password:'. To the right of these fields are two buttons: 'Log On' and 'Cancel'. At the bottom right, there is a link 'Change Password'. At the bottom left, it says 'Supported Browsers: Internet Explorer 8.x, 9.x or 10.x or Firefox 15.0, 16.0 or 17.0.'

## 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.

The screenshot shows the Avaya Aura System Manager 6.3 interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura® System Manager 6.3', and a user status bar indicating 'Last Logged on at October 7, 2013 1:07 PM' with links for 'Help', 'About', 'Change Password', and 'Log off admin'. The left sidebar contains a 'User Management' menu with options: 'Manage Users', 'Public Contacts', 'Shared Addresses', and 'System Presence ACLs'. The main content area is titled 'User Management' and shows a breadcrumb trail 'Home / Users / User Management / Manage Users'. Below the title, there are buttons for 'View', 'Edit', 'New' (circled in red), 'Duplicate', 'Delete', and a 'More Actions' dropdown. A table below these buttons shows a list of users with columns: 'Last Name', 'First Name', 'Display Name', 'Login Name', 'SIP Handle', and 'Last Login'. The table contains two rows of data. At the bottom right, there is a 'Filter: Enable' link.

	Last Name	First Name	Display Name	Login Name	SIP Handle	Last Login
<input type="checkbox"/>	46009	Video	46009, Video	46009@br110.com	46009	
<input type="checkbox"/>	78300	A175	78300, A175	78300@avaya.com	78300	

### 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.

The screenshot shows the 'New User Profile' screen in the Avaya Aura System Manager 6.3. The top navigation bar is the same as the previous screenshot. The left sidebar is also the same. The main content area is titled 'New User Profile' and shows a breadcrumb trail 'Home / Users / User Management / Manage Users'. Below the title, there are buttons for 'Commit & Continue', 'Commit', and 'Cancel'. The screen is divided into four tabs: 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Identity' tab is selected, showing fields for 'Last Name', 'First Name', 'Middle Name', 'Description', 'Login Name', 'Authentication Type', 'Password', and 'Confirm Password'. The 'Last Name' field is filled with 'Bittel', the 'First Name' field is filled with 'UNO', the 'Login Name' field is filled with '78010@avaya.com', and the 'Authentication Type' dropdown is set to 'Basic'. The 'Password' and 'Confirm Password' fields are masked with dots.

## 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.

AVAYA Avaya Aura® System Manager 6.3

Last Logged on at June 7, 2013 9:19 AM  
Help | About | Change Password | Log off admin

User Management \* Home

Home / Users / User Management / Manage Users

Help ?

**New User Profile**

Commit & Continue Commit Cancel

Identity \* Communication Profile \* Membership Contacts

Communication Profile

Communication Profile Password: .....

Confirm Password: .....

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**.

New Delete Done Cancel

Name

Primary

Select : None

\* Name: Primary

Default : ☒

Communication Address

New Edit Delete

Type	Handle	Domain
No Records found		

Type: Avaya SIP

\* Fully Qualified Address: 78010 @ avaya.com

Add Cancel

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.

☒ **Session Manager Profile**

**SIP Registration**

\* **Primary Session Manager** 

Primary	Secondary	Maximum
18	0	18

**Secondary Session Manager**

**Survivability Server**

**Max. Simultaneous Devices**

**Block New Registration When Maximum Registrations Active?** ☐

**Application Sequences**

**Origination Sequence**

**Termination Sequence**

**Call Routing Settings**

\* **Home Location**

**Conference Factory Set**

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).

☒ **CM Endpoint Profile**

\* **System**

\* **Profile Type**

**Use Existing Endpoints** ☐

\* **Extension**

\* **Template**

**Set Type**

**Security Code**

**Port**

**Voice Mail Number**

**Preferred Handle**

**Enhanced Callr-Info display for 1-line phones** ☐

**Delete Endpoint on Unassign of Endpoint from User or on Delete User** ☒

**Override Endpoint Name** ☒

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.

The screenshot shows the 'New Endpoint' form in the Avaya Aura System Manager 6.3. The 'General Options (G)' tab is selected. The form contains the following fields and values:

- System:** devcon13-CM-ES
- Template:** 9620SIP\_DEFAULT\_CM\_6\_3
- Port:** IP
- Name:** Bittel,UNO
- Extension:** 78010
- Set Type:** 9620SIP
- Security Code:** (empty)
- Class of Restriction (COR):** 1
- Emergency Location Ext:** 78010
- Tenant Number:** 1
- SIP Trunk:** aar
- Coverage Path 1:** 20
- Lock Message:** ☐
- Multibyte Language:** Not Applicable
- Class Of Service (COS):** 1
- Message Lamp Ext.:** 78010
- Type of 3PCC Enabled:** None
- Coverage Path 2:** (empty)
- Localized Display Name:** Bittel,UNO

Buttons at the top right include 'Done', 'Cancel', and '[Save As Template]'. A 'Required' legend is at the bottom left.

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.

The screenshot shows the 'New Endpoint' form in the Avaya Aura System Manager 6.3, with the 'Feature Options (F)' tab selected. The form contains the following fields and values:

- Active Station Ringing:** single
- MWI Served User Type:** sip-adjunct
- Per Station CPN - Send Calling Number:** None
- AUDIX Name:** None
- Remote Soft Phone Emergency Calls:** as-on-local
- LWC Reception:** spe
- IP Phone Group ID:** (empty)
- Speakerphone:** 2-way
- Auto Answer:** none
- Coverage After Forwarding:** system
- Display Language:** english
- Hunt-to Station:** (empty)
- Loss Group:** 19
- Survivable COR:** internal
- Time of Day Lock Table:** None

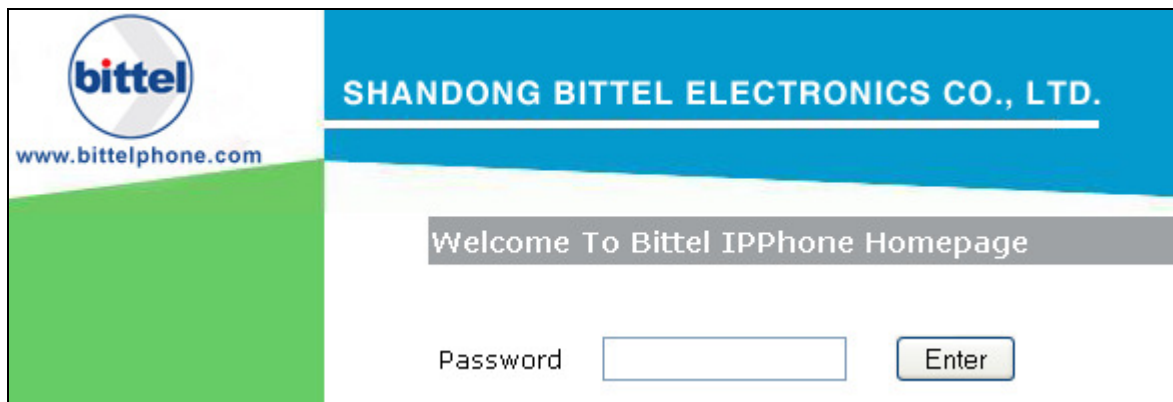
Buttons at the top right include 'Done', 'Cancel', and '[Save As Template]'. A 'Required' legend is at the bottom left.

## 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.



The screenshot shows the Bittel IPPhone Homepage login interface. On the left, there is a logo for 'bittel' with the website 'www.bittelphone.com' below it. The main header area is blue and contains the text 'SHANDONG BITTEL ELECTRONICS CO., LTD.'. Below the header, a grey banner reads 'Welcome To Bittel IPPhone Homepage'. At the bottom, there is a login form with the label 'Password' next to a text input field, and an 'Enter' button to the right of the field.

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**.

The screenshot displays the configuration web interface for a Bittell SIP phone. The interface has a blue header with the Bittell logo and company name, and a green sidebar with navigation links. The main content area is divided into sections: Basic Information, System Settings, and Network Settings. The Network Settings section is currently active, showing fields for Connection Type (Static IP), IP Address, Subnet Mask, Default Gateway, DNS server settings, and Voice/Data VLAN IDs. The 'OK' button is highlighted.

**SHANDONG BITTEL ELECTRONICS CO., LTD.**

**Basic Information**

Device Model: **BITTEL PHONE**  
MAC Address: **F0-07-86-10-02-02**  
Version No.: **10.19\_0.94.003**

**System Settings**

Administrator Password:

**Network Settings**

Connection Type: **Static IP** ▼

IP Address:

Subnet Mask:

Default Gateway:

☒ Automatically get DNS server IP  
☐ Use following DNS server IP

Primary DNS:

Secondary DNS:

Voice VLAN ID:

Data VLAN ID:

Admin Server:

**OK** **Cancel**



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**.

**SHANDONG BITTEL ELECTRONICS CO., LTD.**

**Basic SIP Proxy Settings**

SIP Registration: ☐ No ☒ Yes

SIP Proxy:

SIP Server Port:  (Default 5060)

SIP User ID:

SIP Authentication ID:

SIP Authentication PIN:

User Name:

**Advanced SIP Settings**

Local SIP Port:  (Default 5060)

Local RTP Port:  (Between 1024 and 65535, default 6000)

Register Expiration:  (In seconds, default 60s)

Keep Alive Interval:  (In seconds, default 600s)

Send DTMF:

DTMF Payload Type:  (Between 96 and 127, default 101)

Subscribe for MWI: ☐ No ☒ Yes

Message Service Number:

Use Hotline: ☒ No ☐ Yes

Hotline Number:

OK Cancel

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**.

The screenshot displays the Bittel UNO Voice configuration web interface. On the left is a green sidebar with navigation links: Network, SIP Proxy, Voice (highlighted in yellow), Phonebook, Upgrade Files, and About Bittel. The top header is blue with the Bittel logo and the text 'SHANDONG BITTEL ELECTRONICS CO., LTD.'. The main content area is divided into two sections: 'Voice Codec Settings' and 'Dial Rules Settings'. In the 'Voice Codec Settings' section, 'Preferred Voice Codec: (In listed order)' is set to 'PCMU' for Codec 1, with 'Frames per TX 1' set to '1'. Codecs 2, 3, and 4 are all set to 'None', each with 'Frames per TX' set to '1'. The 'G.723 Rate' is set to '6.3kbps' (selected with a radio button) and '5.3kbps' (unselected). The 'Dial Rules Settings' section shows 'Use # to dial immediately:' set to 'No' (selected radio button) and 'Yes' (unselected). 'The time wait to dial out:' is set to '1' (input field) with '(Default 3)' in parentheses. 'The pause time(1-5s):' is set to '3' (input field) with '(Default 3)' in parentheses. At the bottom are 'OK' and 'Cancel' buttons.

Voice Codec Settings			
Codec 1:	PCMU	Frames per TX 1:	1
Codec 2:	None	Frames per TX 2:	1
Codec 3:	None	Frames per TX 3:	1
Codec 4:	None	Frames per TX 4:	1
G.723 Rate:	<input checked="" type="radio"/> 6.3kbps <input type="radio"/> 5.3kbps		

Dial Rules Settings	
Use # to dial immediately:	<input checked="" type="radio"/> No <input type="radio"/> Yes
The time wait to dial out:	1 (Default 3)
The pause time(1-5s):	3 (Default 3)

OK Cancel

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.

Phone Book	
M10:	<input type="text"/>
M9:	77301
M8:	<input type="text"/>
M7:	<input type="text"/>
M6:	<input type="text"/>
M5:	<input type="text"/>
M4:	<input type="text"/>
M3:	<input type="text"/>
M2:	78119p*70
M1:	26000

OK Cancel

## 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.

The screenshot displays the Avaya Aura System Manager 6.3 interface. The left sidebar shows the navigation menu with 'System Status' expanded and 'User Registrations' selected. The main content area is titled 'User Registrations' and includes a table of 103 items. The table has the following columns: Details, Address, First Name, Last Name, Actual Location, IP Address, Remote Office, Shared Control, Simult. Devices, and AST Device. The row for '78010@avaya.com' is highlighted, showing it is registered to 'Bittel' with IP address '192.168.100.210:5060'.

Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device
▶ Show	---	sipera	test102	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>
▶ Show	---	Mathias	Kiwanuka	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>
▶ Show	---	Jonathan	Goff	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>
▶ Show	---	UT136	Panasonic	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>
▶ Show	---	Linda	Sip	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>
▶ Show	---	SIP	Hammer	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>
▶ Show	---	sipera	test119	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>
▶ Show	---	SIP	Hammer	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>
▶ Show	---	test	Sipera1	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>
▶ Show	---	9601SIP	Martin	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>
▶ Show	78010@avaya.com	UNO	Bittel	---	192.168.100.210:5060	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>
▶ Show	---	Eli	Manning	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>
▶ Show	---	Allan	SIP0	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>
▶ Show	---	A175	78300	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>
▶ Show	---	SIP	Hammer	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>

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.

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