



Avaya Solution & Interoperability Test Lab

Application Notes for Global BHS FLAvoice with Avaya Aura® Session Manager and Avaya Aura® Communication Manager – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Global BHS FLAvoice to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Global BHS FLAvoice is a hospitality solution.

In the compliance testing, Global BHS FLAvoice used the SIP User interface from Avaya Aura® Session Manager to provide hospitality features including voicemail, wakeup call, room status, and minibar.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required for Global BHS FLAvoice to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. FLAvoice is a hospitality solution that uses the SIP User interface from Session Manager to provide voicemail, wakeup call, room status, and minibar features.

In the compliance testing, FLAvoice emulated six virtual SIP users that registered to Session Manager. The six virtual SIP users were configured as separate members of three hunt groups for handling of voicemail, wakeup call, room status and minibar. FLAvoice used the Abto VoIP SIP SDK for Windows to support all SIP communications.

In the compliance testing, subscribers of FLAvoice voicemail consisted of all staff and guest station users on Communication Manager. The Call Coverage feature from Communication Manager was used to redirect calls to FLAvoice via an available virtual SIP user in the voicemail hunt group. The activation and deactivation of Message Waiting Indicator (MWI) for voicemail users were accomplished by FLAvoice via use of SIP NOTIFY.

Scheduling of wakeup calls were initiated from the staff and guest telephones by calling the wakeup call hunt group, and the delivery of wakeup calls were initiated by FLAvoice via available virtual SIP users associated with the wakeup feature.

Room status and minibar were accomplished by calling the room status and minibar hunt group from the guest telephones.

FLAvoice also supports the Property Management System (PMS) interface for integration with a third-party PMS system for initiation of other hospitality features such as check-in and check-out. In the compliance testing, the FLAvoice CHECK-INS AND CHECK-OUTS MANAGER VIA IP tool was used for setting of necessary check-in, check-out, and room change status for various guests on FLAvoice as part of testing the voicemail, wakeup call, room status and minibar features.

2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were made from local users to the hunt groups for various features. Calls were made from the PSTN to the voicemail hunt group to verify remote retrieval of voice messages.

The serviceability test cases were performed manually by disconnecting/reconnecting the Ethernet connection to FLAvoice.

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 these DevConnect Application Notes 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 these Application Notes, the interface between Session Manager and FLAvoice did not include use of any specific encryption features as requested by Global BHS.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing focused on verifying the following on FLAvoice:

- Proper handling of SIP message exchanges including registration, G.711, G.729, media shuffling, codec negotiation, session refresh, DTMF, REFER, and NOTIFY.
- Voicemail recording, logging, and retrieval, with proper MWI activation/deactivation.
- Scheduling and delivery of wake-up calls, including retried attempts and escalation to staff.
- Proper handling of room status and minibar.

The serviceability testing focused on verifying the ability of FLAvoice to recover from adverse conditions, such as disconnecting and reconnecting the Ethernet connection to FLAvoice.

2.2. Test Results

All test cases were executed. The following was observed on FLAvoice from the compliance testing.

- When a voicemail user calls from the PSTN to perform remote retrieval of voice message, the greeting announcement heard by the caller was slightly chopped off from the beginning. Global BHS shared that a configurable delay will be implemented in a future release to help address this observation.

2.3. Support

Technical support on FLAvoice can be obtained through the following:

- **Phone:** +1 (407) 501-7500
- **Email:** support@globalbhs.com
- **Web :** <http://globalbhs.com/suporte>

3. Reference Configuration

The configuration used for the compliance testing is shown in **Figure 1**, with the domain name used in the testing being “dr220.com”.

The configuration of Session Manager is performed via the web interface of System Manager. The detailed administration of basic connectivity between Communication Manager, System Manager, and Session Manager are not the focus of these Application Notes and will not be described.

The Communication Manager resources used in the compliance testing are shown in the table below.

Device Type	Extension
Staff Station	65001 (H.323)
Guest Station	66002 (SIP), 63001 (Analog)

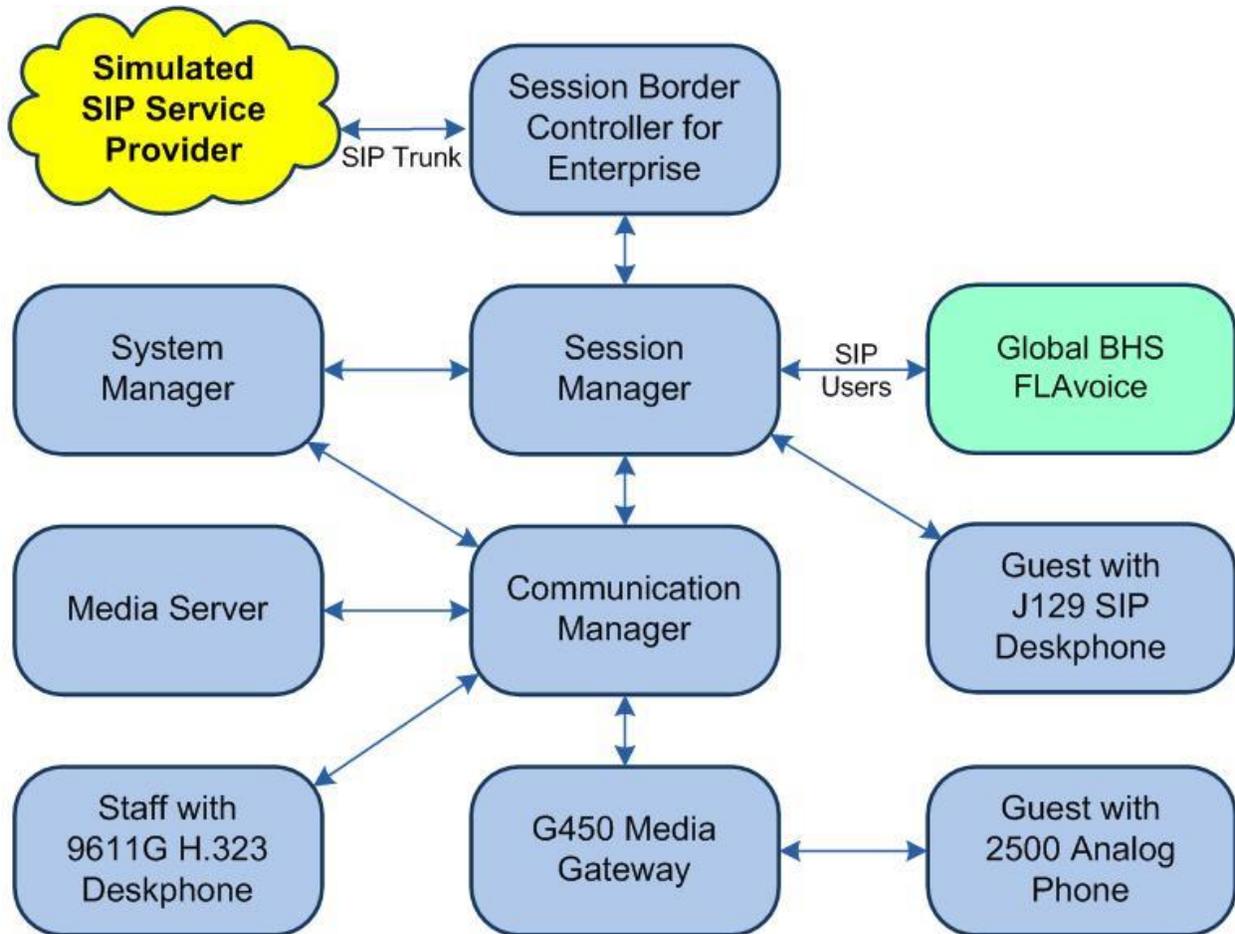


Figure 1: Compliance Testing Configuration

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 Virtual Environment	8.1.1 (8.1.0.1.1.890.25763)
Avaya G450 Media Gateway	41.16.0
Avaya Aura® Media Server in Virtual Environment	8.0.1.121
Avaya Aura® Session Manager in Virtual Environment	8.1.1 (8.1.1.0.811021)
Avaya Aura® System Manager in Virtual Environment	8.1.1 (8.1.1.0.0310912)
Avaya 9611G IP Deskphone (H.323)	6.8202
Avaya J129 IP Deskphone (SIP)	4.0.2.1.3
2500YMGK Analog Phone	NA
Global BHS FLAvoice <ul style="list-style-type: none">Abto VoIP SIP SDK for Windows (SIPVoipSDK.dll)	9.4.7 4.11.406.1

5. Configure Avaya Aura® Communication Manager

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

- Administer IP codec set
- Administer hunt groups
- Administer coverage path
- Administer stations

5.1. Administer IP Codec Set

Administer a codec set for integration with FLAvoice. Use the “change ip-codec-set n” command, where “n” is an existing codec set number to use for interoperability. In the compliance testing, codec set “1” was used for FLAvoice and for the staff and guest stations.

For **Audio Codec**, enter the pertinent codec variants as shown below. For **Media Encryption** make certain “none” is included. For **Encrypted SRTP**, make certain the value is not “enforce-enc-srtp”.

```
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: G.729        n           2          20
3:
4:
5:
6:
7:

Media Encryption                                Encrypted SRTP: best-effort
1: 1-srtp-aescm128-hmac80
2: aes
3: none
```

5.2. Administer Hunt Groups

Administer three hunt groups for FLAvoice voicemail, wakeup call, and combined room status and mini bar respectively.

5.2.1. Voicemail

Use the “add hunt-group n” command, where “n” is an available hunt group number. This hunt group is used for FLAvoice voicemail. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Group Number:** The available group number.
- **Group Name:** A descriptive name.
- **Group Extension:** An available extension number.
- **Group Type:** “circ”

```
add hunt-group 61                                     Page 1 of 60
                                                    HUNT GROUP
Group Number: 61
Group Name: FLAvoice Voicemail
Group Extension: 62001
Group Type: circ                                     Coverage Path:
TN: 1                                               Night Service Destination:
COR: 1                                             MM Early Answer? n
Security Code:                                     Local Agent Preference? n
ISDN/SIP Caller Display:
```

Navigate to **Page 3** and enter the extension of all FLAvoice virtual SIP users from **Section 6.3** for handling of voicemail, as shown below.

```
add hunt-group 61                                     Page 3 of 60
                                                    HUNT GROUP
Group Number: 61   Group Extension: 62001           Group Type: circ
Member Range Allowed: 1 - 1500   Administered Members (min/max): 0 /0
                                                    Total Administered Members: 0
GROUP MEMBER ASSIGNMENTS
Ext      Name(16 characters)      Ext      Name(16 characters)
1: 66991      14:
2: 66992      15:
3:            16:
```

5.2.2. Wakeup Call

Use the “add hunt-group n” command, where “n” is an available hunt group number. This hunt group is used for FLAvoice wakeup call. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Group Number:** The available group number.
- **Group Name:** A descriptive name.
- **Group Extension:** An available extension number.
- **Group Type:** “circ”

```
add hunt-group 62                                     Page 1 of 60
                                                    HUNT GROUP
Group Number: 62
Group Name: FLAvoice Wakeup
Group Extension: 62002
Group Type: circ                                     Coverage Path:
TN: 1                                               Night Service Destination:
COR: 1                                             MM Early Answer? n
Security Code:                                     Local Agent Preference? n
ISDN/SIP Caller Display:
```

Navigate to **Page 3** and enter the extension of all FLAvoice virtual SIP users from **Section 6.3** for handling of wakeup call, as shown below.

```
add hunt-group 62                                     Page 3 of 60
                                                    HUNT GROUP
Group Number: 62   Group Extension: 62002           Group Type: circ
Member Range Allowed: 1 - 1500   Administered Members (min/max): 0 /0
                                                    Total Administered Members: 0
GROUP MEMBER ASSIGNMENTS
Ext      Name(16 characters)      Ext      Name(16 characters)
1: 66993      14:
2: 66994      15:
3:            16:
```

5.2.3. Room Status and Minibar

Use the “add hunt-group n” command, where “n” is an available hunt group number. This hunt group is used for FLAvoice room status and minibar. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Group Number:** The available group number.
- **Group Name:** A descriptive name.
- **Group Extension:** An available extension number.
- **Group Type:** “circ”

```
add hunt-group 63                                     Page 1 of 60
                                                    HUNT GROUP
Group Number: 63
Group Name: FLAvoice RoomBar
Group Extension: 62003
Group Type: circ                                     Coverage Path:
TN: 1                                               Night Service Destination:
COR: 1                                             MM Early Answer? n
Security Code:                                     Local Agent Preference? n
ISDN/SIP Caller Display:
```

Navigate to **Page 3** and enter the extension of all FLAvoice virtual SIP users from **Section 6.3** for handling of room status and minibar, as shown below.

```
add hunt-group 63                                     Page 3 of 60
                                                    HUNT GROUP
Group Number: 63   Group Extension: 62003           Group Type: circ
Member Range Allowed: 1 - 1500   Administered Members (min/max): 0 /0
                                                    Total Administered Members: 0
GROUP MEMBER ASSIGNMENTS
Ext      Name(16 characters)      Ext      Name(16 characters)
1: 66995                               14:
2: 66996                               15:
3:                                       16:
```

5.3. Administer Coverage Path

Add a coverage path using the “add coverage path n” command, where “n” is an available coverage path number. This coverage path is used for coverage to FLAvoice for voicemail.

For **Point1**, enter “h61” to designate the voicemail hunt group “61” from **Section 5.2.1** as the first coverage point. Retain the default values in the remaining fields.

```
add coverage path 61                                     Page 1 of 1
                                                    COVERAGE PATH
                Coverage Path Number: 61
    Cvg Enabled for VDN Route-To Party? n             Hunt after Coverage? n
                Next Path Number:                   Linkage

COVERAGE CRITERIA
  Station/Group Status   Inside Call   Outside Call
        Active?          n              n
        Busy?            y              y
    Don't Answer?        y              y      Number of Rings: 2
        All?             n              n
    DND/SAC/Goto Cover?  y              y
    Holiday Coverage?    n              n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
Point1: h61           Rng:      Point2:
    Point3:              Point4:
    Point5:              Point6:
```

5.4. Administer Stations

Use the “change station n” command, where “n” is a non-SIP station extension from **Section 3**. Note that similar configuration for SIP station extensions are performed from System Manager in **Section 6.2**.

For **Coverage Path 1**, enter the coverage path number from **Section 5.3**.

For analog stations, the **Message Waiting Indicator** may need modification, depending on the type of analog telephone. In the compliance testing, one analog station with phone **Type** of “2500” was required to have the **Message Waiting Indicator** set to “led” for interoperability.

```
change station 63001                                     Page 1 of 4
                                     STATION
Extension: 63001                                         Lock Messages? n      BCC: 0
  Type: 2500                                           Security Code:       TN: 1
  Port: 001V302                                         Coverage Path 1: 61  COR: 1
  Name:                                                 Coverage Path 2:     COS: 1
Unicode Name? n                                         Hunt-to Station:     Tests? y
STATION OPTIONS
  XOIP Endpoint type: auto                               Time of Day Lock Table:
  Loss Group: 1                                         Message Waiting Indicator: led
  Off Premises Station? n                               Message Lamp Ext: 63001
```

Repeat this section and **Section 6.2** to administer all stations from **Section 3**.

In the compliance testing, three stations were configured as shown below.

```
list station 63001 count 3                               Page 1
                                     STATIONS
Ext/      Port/      Name/      Move  Cable  Room/      Cv1/  COR/
 Hunt-to  Type        Surv GK NN  Move  Cable  Jack      Cv2  COS  TN
63001   001V201  Analog Room  no    no     Jack      61  1
          2500
65001   S000103  H323 Staff  no    no     Jack      61  1
          9611
66002   S000068  Avaya, SIP 2  no    no     Jack      61  1
          J129
```

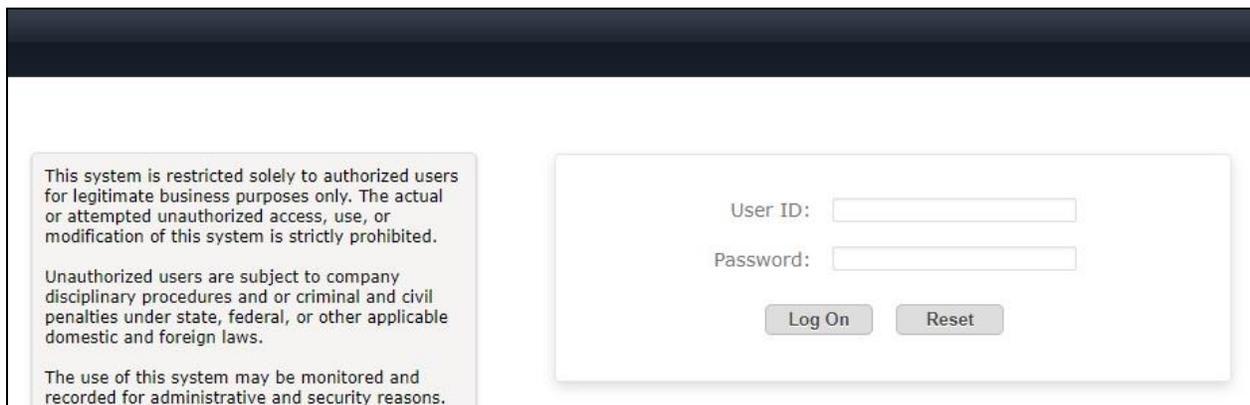
6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager, which is performed via the web interface of System Manager. The procedures include the following areas:

- Launch System Manager
- Administer existing SIP users
- Administer virtual SIP users
- Administer Session Manager entity

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 System Manager. Log in using the appropriate credentials.



This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

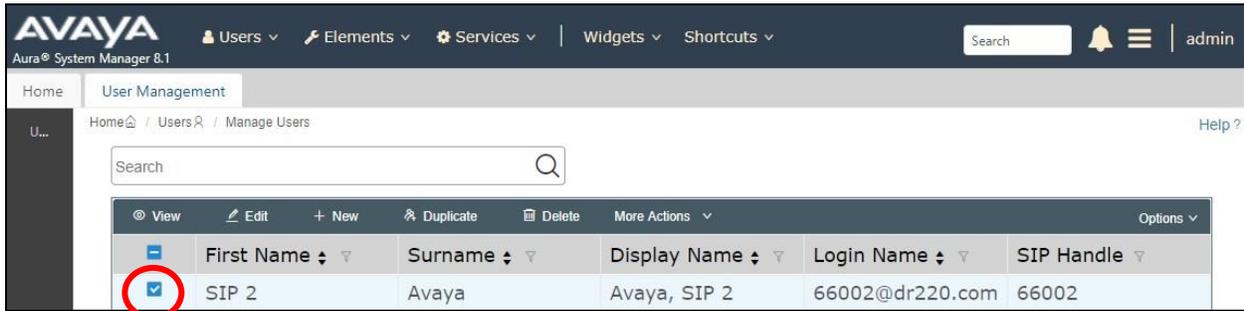
The use of this system may be monitored and recorded for administrative and security reasons.

User ID:

Password:

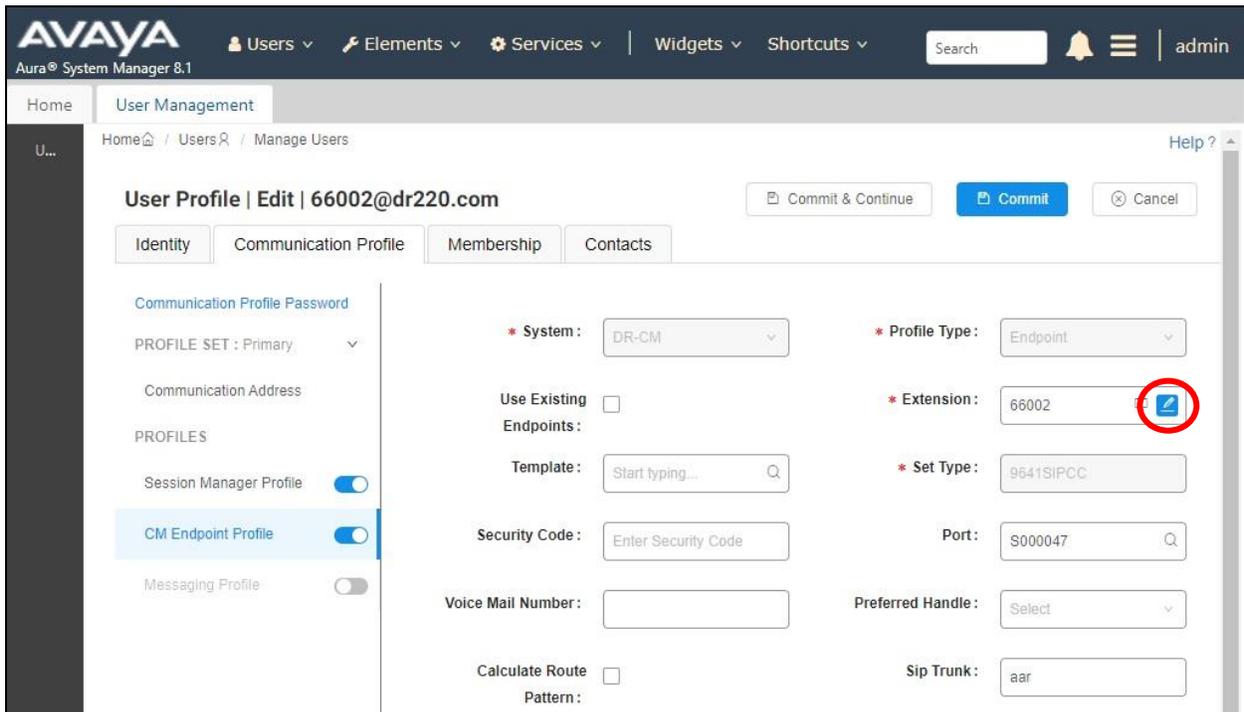
6.2. Administer Existing SIP Users

In the subsequent screen, select **Users** → **User Management** from the top menu. Select **User Management** → **Manage Users** (not shown) from the left pane to display the screen below. Select the entry associated with the first SIP station user from **Section 3**, in this case “66002”, and click **Edit**.



The **User Profile | Edit** screen is displayed. Select the **Communication Profile** tab, followed by **CM Endpoint Profile** to display the screen below.

Click on the **Editor** icon shown below.



The popped-up screen below is displayed. For **Coverage Path 1**, enter the coverage path number from **Section 5.3**.

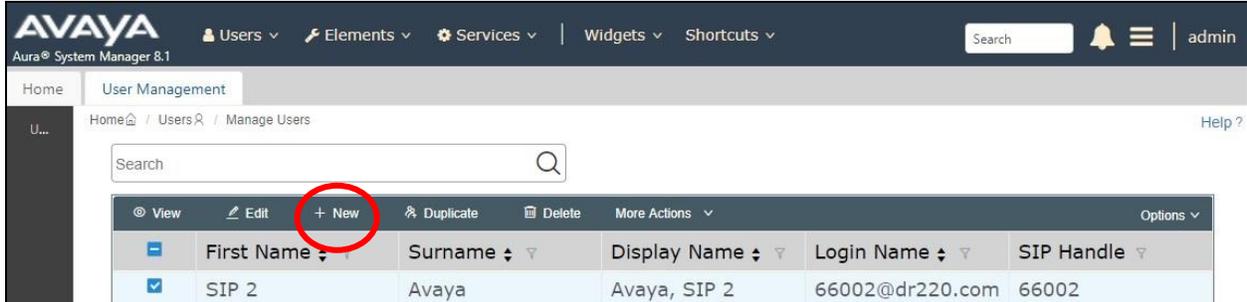
Repeat this section to administer all SIP station users that will be using FLVoice for voicemail. In the compliance testing, one SIP station user was configured as shown below.

The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes the Avaya logo, 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts' menus, along with a search bar and notification icons. The main content area is titled 'User Profile | Edit | 66002@dr220.com' and features several tabs: 'General Options (G)', 'Feature Options (F)', 'Site Data (S)', 'Abbreviated Call Dialing (A)', 'Enhanced Call Fwd (E)', 'Button Assignment (B)', 'Profile Settings (P)', and 'Group Membership (M)'. The 'General Options (G)' tab is active, displaying various configuration fields. A red box highlights the 'Coverage Path 1' field, which contains the value '61'. Other visible fields include 'Class of Restriction (COR)', 'Emergency Location Ext', 'Tenant Number', 'SIP Trunk', 'Class Of Service (COS)', 'Message Lamp Ext.', 'Type of 3PCC Enabled', 'Coverage Path 2', 'Localized Display Name', and 'Enable Reachability for Station Domain Control'.

Field	Value
Class of Restriction (COR)	1
Emergency Location Ext	66002
Tenant Number	1
SIP Trunk	Qaar
Coverage Path 1	61
Lock Message	<input type="checkbox"/>
Multibyte Language	Not Applicable
Class Of Service (COS)	1
Message Lamp Ext.	66002
Type of 3PCC Enabled	Avaya
Coverage Path 2	
Localized Display Name	Avaya, SIP 2
Enable Reachability for Station Domain Control	system

6.3. Administer Virtual SIP Users

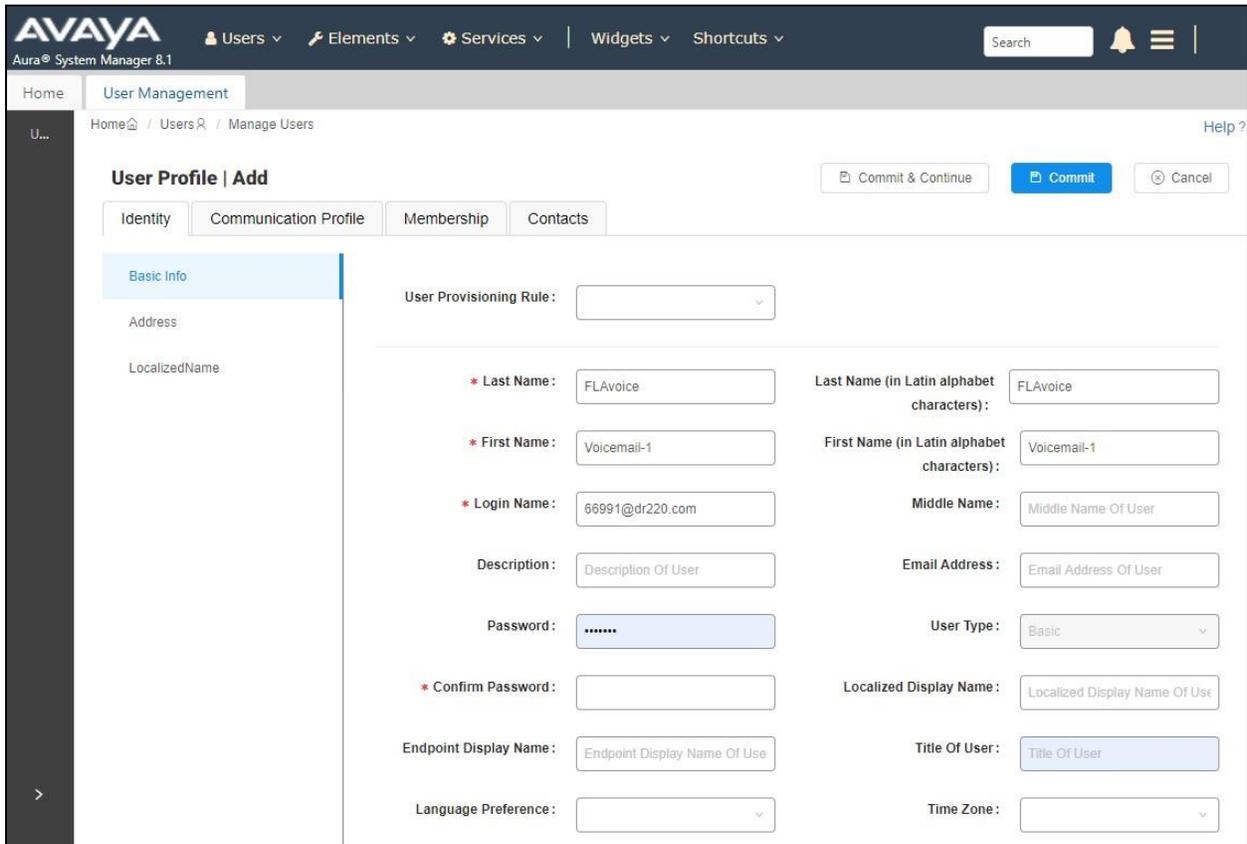
The screen below is displayed again. Click **New** to add a virtual SIP user for handling of FLAvoice voicemail.



6.3.1. Identity

The **User Profile | Add** screen is displayed. Enter desired **Last Name** and **First Name**.

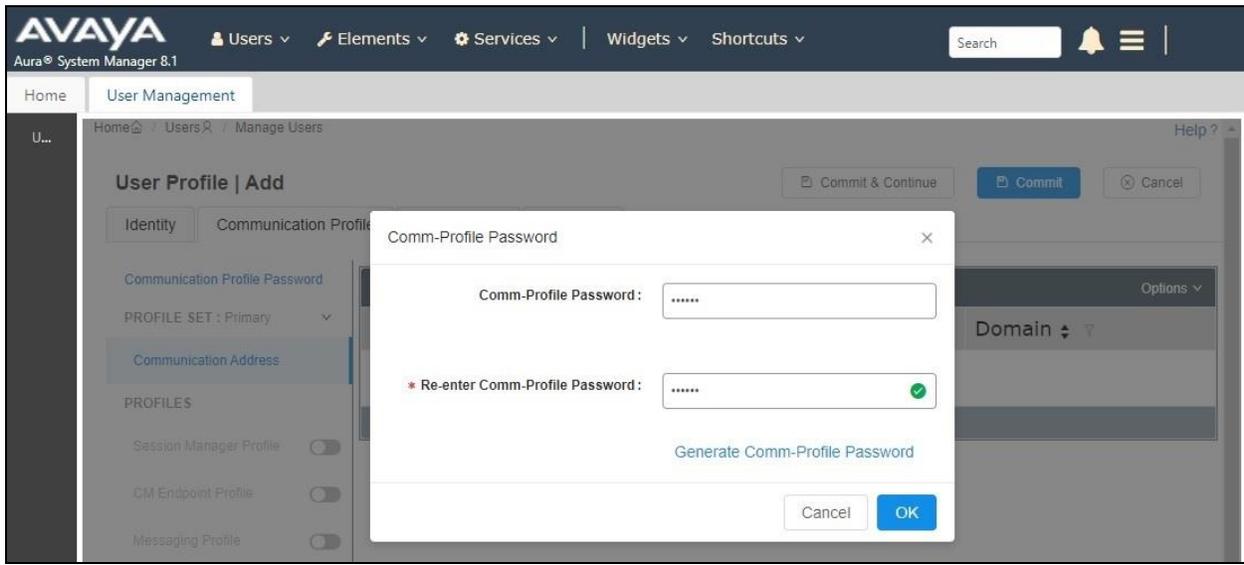
For **Login Name**, enter “n@x”, where “n” is the desired user extension and “x” is the applicable domain name from **Section 3**. Retain the default values in the remaining fields.



6.3.2. Communication Profile

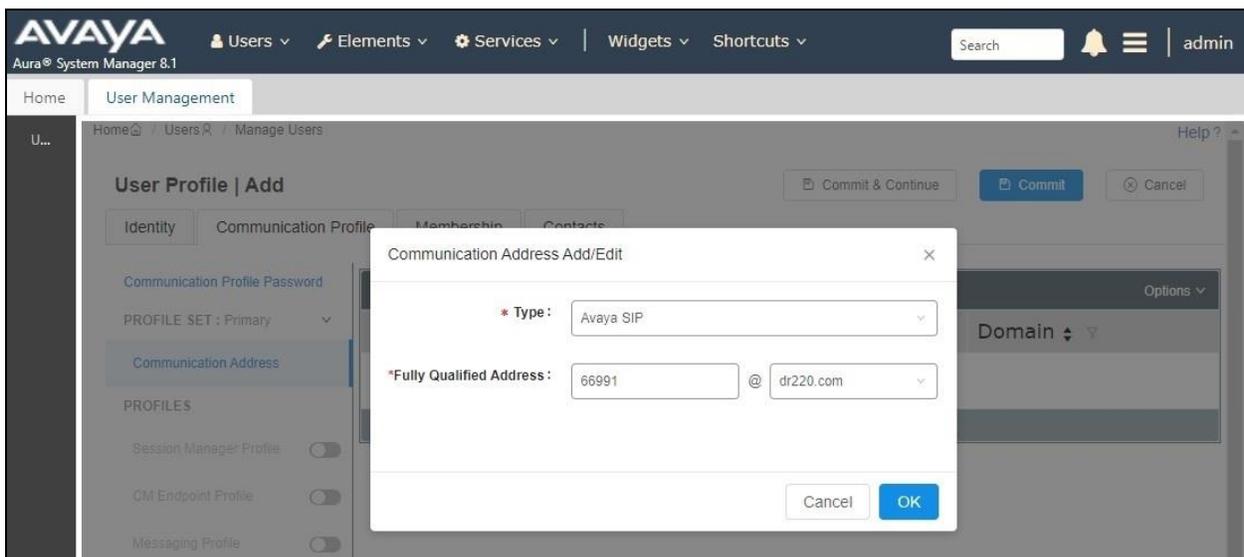
Select the **Communication Profile** tab, followed by **Communication Profile Password** to display the **Comm-Profile Password** pop-up box.

For **Communication-Profile Password** and **Re-enter Comm-Profile Password**, enter the desired password for the virtual SIP user to use for registration.



Select **Communication Address** from the left, followed by **New** to display the **Communication Address Add/Edit** pop-up box.

For **Type**, select “Avaya SIP”. For **Fully Qualified Address**, enter and select the SIP user extension and domain name to match the login name from **Section 6.3.1**.



Select **Session Manager Profile**. For **Primary Session Manager**, **Origination Sequence**, **Termination Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager and Communication Manager as shown below. Retain the default values in the remaining fields.

The screenshot displays the Avaya Aura System Manager 8.1 interface for adding a user profile. The left sidebar shows the 'Session Manager Profile' selected under the 'PROFILES' section. The main configuration area is divided into several sections:

- SIP Registration:**
 - Primary Session Manager: DR-SM
 - Secondary Session Manager: Start typing...
 - Survivability Server: Start typing...
 - Max. Simultaneous Devices: Select
- Block New Registration When Maximum Registrations Active?:**
- Application Sequences:**
 - Origination Sequence: DR220-CM-APP-Sequence
 - Termination Sequence: DR220-CM-APP-Sequence
- Emergency Calling Application Sequences:**
 - Emergency Calling Origination Sequence: Select
 - Emergency Calling Termination Sequence: Select
- Call Routing Settings:**
 - Home Location: NJ-Loc

Select **CM Endpoint Profile** from the left. For **System**, select the value corresponding to the applicable Communication Manager. For **Template**, select “9620SIP_DEFAULT_CM_8_1”. For **Extension**, enter the SIP user extension from **Section 6.3.1**. Retain the default values in the remaining fields.

The screenshot shows the 'User Profile | Add' form in Avaya Aura System Manager 8.1. The 'Communication Profile' tab is selected. The form contains the following fields and values:

- System:** DR-CM
- Profile Type:** Endpoint
- Extension:** 66991
- Template:** 9620SIP_DEFAULT_CM_8_1
- Set Type:** 9620SIP
- Port:** IP
- Preferred Handle:** Select
- Sip Trunk:** aar
- Calculate Route Pattern:**
- SIP URI:** Select
- Delete on Unassign from User or on Delete User:**
- Override Endpoint Name and Localized Name:**
- Allow H.323 and SIP Endpoint Dual Registration:**

Repeat **Section 6.3** to add the desired number of virtual SIP users for handling of FLAvoice voicemail, wakeup, and combined room status and minibar. In the compliance testing, two SIP users were created for voicemail, two for wakeup call, and two for room status and minibar, as shown below.

	First Name	Surname	Display Name	Login Name	SIP Handle
<input checked="" type="checkbox"/>	SIP 2	Avaya	Avaya, SIP 2	66002@dr220.com	66002
<input type="checkbox"/>	Voicemail-1	FLAvoice	FLAvoice, Voicemail-1	66991@dr220.com	66991
<input type="checkbox"/>	Voicemail-2	FLAvoice	FLAvoice, Voicemail-2	66992@dr220.com	66992
<input type="checkbox"/>	Wakeup-1	FLAvoice	FLAvoice, Wakeup-1	66993@dr220.com	66993
<input type="checkbox"/>	Wakeup-2	FLAvoice	FLAvoice, Wakeup-2	66994@dr220.com	66994
<input type="checkbox"/>	RoomBar-1	FLAvoice	FLAvoice, RoomBar-1	66995@dr220.com	66995
<input type="checkbox"/>	RoomBar-2	FLAvoice	FLAvoice, RoomBar-2	66996@dr220.com	66996

6.4. Administer Session Manager Entity

Select **Elements** → **Routing** → **SIP Entities** from the top menu to display the **Routing** tab, followed by the applicable SIP entity for Session Manager from the left pane (not shown), in this case “DR-SM”. The **SIP Entity Details** screen is displayed.

The screenshot shows the AVAYA Aura System Manager 8.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts' menus, along with a search box. The 'Routing' tab is selected. The main content area is titled 'SIP Entity Details' and includes 'Commit' and 'Cancel' buttons. The 'General' section contains the following fields:

- Name:** DR-SM
- IP Address:** 10.64.101.238
- SIP FQDN:** (empty)
- Type:** Session Manager
- Notes:** TLT DR SM
- Location:** DR-Loc
- Outbound Proxy:** (empty)
- Time Zone:** America/New_York
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty)

Scroll down to **Listen Ports** sub-section and verify that the transport protocol used by FLAvoice is specified in the list, in this case “UDP”. Also verify that the corresponding **Endpoint** column is checked, as shown below.

The screenshot shows the AVAYA Aura System Manager 8.1 interface. The top navigation bar is the same as in the previous screenshot. The 'Routing' tab is selected. The main content area is titled 'Listen Ports' and includes 'Add' and 'Remove' buttons. Below the buttons, it says '3 Items'. A table lists the listen ports:

<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Endpoint	Notes
<input type="checkbox"/>	5060	TCP	dr220.com	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	5060	UDP	dr220.com	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	5061	TLS	dr220.com	<input checked="" type="checkbox"/>	

Select : All, None

7. Configure Global BHS FLAvoice

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

- Launch FLAvoice
- Administer PBX
- Administer channel functions
- Administer mailboxes
- Administer wakeup call
- Administer room status and minibar

7.1. Launch FLAvoice

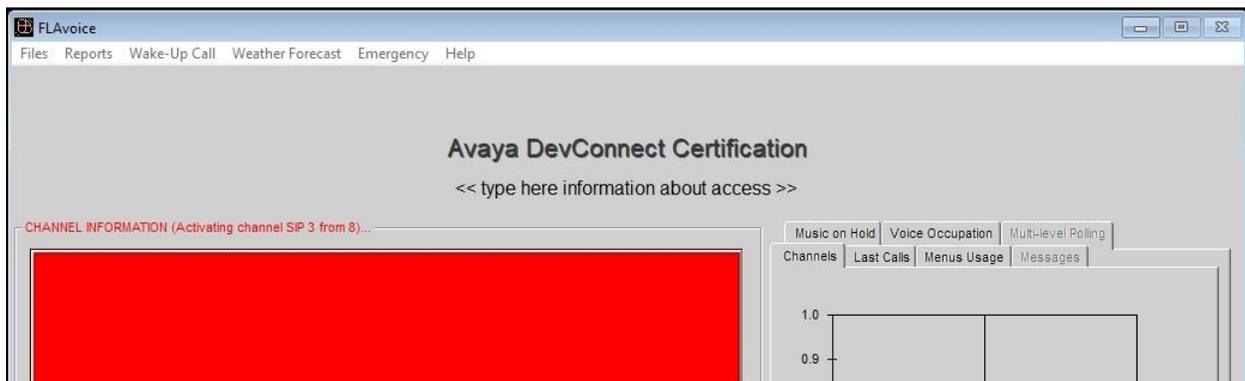
From the FLAvoice server, double-click on the **FLAvoice** icon shown below, which was created as part of installation.



The **FLAvoice** screen below is displayed, where “Avaya DevConnect Certification” is the customer name picked up from the applied license file as part of installation.

Select **Files** → **Unlock Access to Settings** from the top menu and enter the appropriate credentials in the subsequent screen (not shown) to unlock settings. The **FLAvoice** screen is displayed again and updated to allow access to settings.

Select **File** → **Settings** from the top menu.



7.2. Administer PBX

The **Settings** screen is displayed. Select the **PBX** tab. For **PBX Model**, select the value shown below. Retain the default values in the remaining fields and click **Configure SIP parameters**.

The screenshot shows the 'Settings - v.9.4.7' application window. At the top, there is a navigation menu with various tabs including 'Hotel - Express Check-out', 'Hotel - Room Hygienization', 'Hotel - Emergency Calls', 'Hotel - Integration with PMS', 'Hotel - Call Accounting', 'Hotel - Take My Tray', 'Apps', 'SNMP', 'Tools via IP', 'Hotel - Broadband Internet', 'Hotel - Room Status and Minibar', 'Hotel - Room Service', 'Hotel - Wake-up Call', 'Hotel - Weather Forecast', 'Messages Loop', 'Security Module', 'Music on Hold', 'Pop-up Notification', 'Call queuing', 'Fax Mail and Server', 'Outside Line Request', 'Messages, Notifications and E-Mails', 'Messages and Alarms', 'General settings', 'Pre-Paid Telephony', 'Access Control', 'Polling', 'Technical Support IVR', 'PBX', 'Channel Functions', 'Mailboxes', 'Auto-Attendant', 'Redirections', 'Main Window and Owner', and 'Voice Channels'. The 'PBX' tab is selected.

The main content area is titled 'PBX model (ordered by manufacturer and model):' and shows a dropdown menu with the selected value 'AVAYA model Avaya Aura Communication Manager v.8.1 (SIP)'. To the left of the dropdown is an image of an Avaya PBX device with a 'DEVCONNECT TESTED' badge. To the right of the dropdown is a text box containing the following information: 'Does not accept multi-function channels. Analog extensions connected to the PBX may require that they be configured as 'LED' for their Message Waiting Indicator (MWI) to work correctly. Version certified by Avaya DevConnect. PBX accepts control of MWI via SIP NOTIFY.' Below this text box are two buttons: 'Set up PBX Integration...' and 'Configure SIP parameters...'.

Below the PBX model section, there is a 'Dialing type' section with two radio buttons: 'Use tone dialing for redirection.' (selected) and 'Use pulse dialing to transfer and resume calls.'. Under 'Use tone dialing for redirection.', there is a 'Flash time:' field with a value of '450' ms (originally: 450 ms) and a note '[Recommended: 100 ms]'. Below this is a 'Pause time:' field with a value of '1200' ms (originally: 1200 ms). There is also an 'Outside line access code:' field which is currently empty. Below these fields is a button labeled 'Out-of-band Caller ID Information Settings...'.

At the bottom left, there is an information icon and a checkbox labeled 'Show number in tab names.'. At the bottom center, there is a button labeled 'END SETTINGS'.

The **SIP account settings** screen is displayed. For **SIP Proxy**, enter the IP address of the Session Manager signaling interface. For **UDP Port**, enter “5060”. In the channel entries sub-section, enter the extension and password for each virtual SIP user from **Section 6.3**. Retain the default values in the remaining fields.

SIP account settings - v.9.4.7

PBX: AVAYA model Avaya Aura Communication Manager v.8.1 (SIP)

SIP Proxy (IPv4):

Port:

Transport:

Listen IP:

Outbound Proxy:

Realm:

IP port for CSTA:

Dialing detection

- Accept DTMF dialing.
- Accept dialing via RFC 2833.
- Accept SIP INFO dialing.

Channel	Extension number	Password	Authentication ID
▶ 1	66991	123456	
2	66992	234567	
3	66993	345678	
4	66994	456789	
5	66995	567890	
6	66996	123456	
7			
8			

CODEC to be used by SIP extensions (only 1 can be in use at a time)

CODEC: [Recommended: G711u]

OK

7.3. Administer Channel Functions

The **Settings** screen is displayed again. Select the **Channel Functions** tab to display the screen shown below.

Settings - v.9.4.7

Hotel - Express Check-out | Hotel - Room Hygienization | Hotel - Emergency Calls | Hotel - Integration with PMS | Hotel - Call Accounting
 SNMP | Tools via IP | Hotel - Broadband Internet | Hotel - Room Status and Minibar | Hotel - Room Service | Hotel - Wake-up Call | Hotel -
 Messages Loop | Security Module | Music on Hold | Pop-up Notification | Call queuing | Fax Mail and Server | Outside Line Request |
 Messages, Notifications and E-Mails | Messages and Alarms | General settings | Pre-Paid Telephony | Access Control | Polling | Technical S
 PBX | Channel Functions | Mailboxes | Auto-Attendant | Redirections | Main Window and Owner | Voice Channels |

PBX Model: AVAYA model Avaya Aura Communication Manager v.8.1 (SIP)

CHANNEL	Function of the channel	Language
1	CHANNEL WITH NO FUNCTION ASSIGNED YET	English
2	CHANNEL WITH NO FUNCTION ASSIGNED YET	English
3	CHANNEL WITH NO FUNCTION ASSIGNED YET	English
4	CHANNEL WITH NO FUNCTION ASSIGNED YET	English
5	CHANNEL WITH NO FUNCTION ASSIGNED YET	English
6	CHANNEL WITH NO FUNCTION ASSIGNED YET	English
7	CHANNEL WITH NO FUNCTION ASSIGNED YET	English
8	CHANNEL WITH NO FUNCTION ASSIGNED YET	English

Click the **Function of the channel** field for the first channel, and select the value shown in the screen below from the associated drop-down list. Repeat as necessary for remaining channels.

In the compliance testing, the first two channels corresponded to the two virtual SIP users in **Section 6.3** for handling of voicemail; the next two channels corresponded to the two virtual SIP users in **Section 6.3** for handling of wakeup call, and the last two channels corresponded to the two virtual SIP users in **Section 6.3** for handling of room status and minibar.

Settings - v.9.4.7

Hotel - Express Check-out | Hotel - Room Hygienization | Hotel - Emergency Calls | Hotel - Integration with PMS | Hotel - Call Accounting
 SNMP | Tools via IP | Hotel - Broadband Internet | Hotel - Room Status and Minibar | Hotel - Room Service | Hotel - Wake-up Call | Hotel -
 Messages Loop | Security Module | Music on Hold | Pop-up Notification | Call queuing | Fax Mail and Server | Outside Line Request |
 Messages, Notifications and E-Mails | Messages and Alarms | General settings | Pre-Paid Telephony | Access Control | Polling | Technical S
 PBX | Channel Functions | Mailboxes | Auto-Attendant | Redirections | Main Window and Owner | Voice Channels |

PBX Model: AVAYA model Avaya Aura Communication Manager v.8.1 (SIP)

CHANNEL	Function of the channel	Language
1	Voice mail + Auto-attendant (incoming)	English
2	Voice mail + Auto-attendant (incoming)	English
3	Wake-up Call and Wake-up Programming (incoming and outgoing)	English
4	Wake-up Call and Wake-up Programming (incoming and outgoing)	English
5	Room Status and Minibar (incoming)	English
6	Room Status and Minibar (incoming)	English
7	CHANNEL WITH NO FUNCTION ASSIGNED YET	English
8	CHANNEL WITH NO FUNCTION ASSIGNED YET	English

7.4. Administer Mailboxes

From the **Settings** screen, select the **Mailboxes** tab to display the screen below. Click **Mailboxes Settings**.



Follow reference [3] to create a mailbox for each staff and guest station from **Section 3**. Set **Type of the extension** to “guest” for guest stations and to “service” for staff stations, as shown below.

The screenshot shows the 'Mailboxes Settings - v.9.4.7' interface. It displays a table with the following columns: Mailbox, Enabled, Extension, Extension is digital, Password, User's full name, Type of the extension, Language, and Bilingual per. The table contains three rows of data:

Mailbox	Enabled	Extension	Extension is digital	Password	User's full name	Type of the extension	Language	Bilingual per
63001	<input checked="" type="checkbox"/>	63001	<input type="checkbox"/>		Avaya Analog	guest	English	
65001	<input checked="" type="checkbox"/>	65001	<input type="checkbox"/>	65001	STAFF	service	English	
66002	<input checked="" type="checkbox"/>	66002	<input type="checkbox"/>		Avaya SIP	guest	English	

7.5. Administer Wakeup Call

From the **Settings** screen, select the **Hotel – Wake-up Call** tab to display the screen below. Configure the **Wake-up call settings** sub-section as desired.

In the **How to alarm if the guest doesn't answer the wake-up call** sub-section, check **Call the extension**, and enter the staff extension from **Section 3** as shown below. Retain the default values in the remaining fields.

The screenshot shows the 'Settings - v.9.4.7' interface for 'Hotel - Wake-up Call'. The top navigation bar includes tabs for Messages Loop, Security Module, Music on Hold, Pop-up Notification, Call queuing, Fax Mail and Server, Outside Line Request, Messages, Notifications and E-Mails, Messages and Alarms, General settings, Pre-Paid Telephony, Access Control, Polling, Technical Support IVR, PBX, Channel Functions, Mailboxes, Auto-Attendant, Redirections, Main Window and Owner, Voice Channels, Hotel - Express Check-out, Hotel - Room Hygienization, Hotel - Emergency Calls, Hotel - Integration with PMS, Hotel - Call Accounting, Hotel - Take My Tray, Apps, SNMP, Tools via IP, Hotel - Broadband Internet, Hotel - Room Status and Minibar, Hotel - Room Service, Hotel - Wake-up Call (selected), and Hotel - Weather Forecast.

The main configuration area is divided into several sections:

- Wake-up call settings:** Includes spinners for 'Attempts to set the alarm clock' (3, Recommended: 3), 'Number of rings for wake-up call' (4, Recommended: 4), 'Number of attempts for wake-up' (3, Recommended: 3), and 'Interval between 2 attempts' (2 min, Recommended: 2).
- Time interval (min) for reinforcement call:** A slider set to 'off' with options from 2 to 9.
- When restarting the computer:** A checked checkbox 'Turn off wake-up call if already overdue'.
- Assisted wake-up call:** A text field for 'Extension of the assisted wake-up call' with a note: '(This extension will receive a call from the system asking it to manually wake a guest up)'. There is also a 'Wake-up self test' section with an unchecked checkbox 'Enable wake-up self test' and a 'Voice-mail's group number' field.
- How to alarm if the guest doesn't answer the wake-up call:** A radio button selection between 'Alarm is independent of the PBX identification in the network' (selected) and 'Alarm depends on the PBX identification in the network'. Under the selected option, there are checkboxes for 'Send an e-mail to the operator' (unchecked), 'Call the extension' (checked, with value 65001 and a 'Set alternative extension...' button), 'Print a warning at the default printer' (unchecked), and 'Send SMS to:' (with an empty field). There is also a 'Print warning only if the operator doesn't answer the wake-up alarm call' checkbox (unchecked).
- Groups of extensions for the Wake-up call:** Includes 'Assign group...' and 'Set the wake-up call for a group...' buttons.
- Mailbox types (guest, service, transit, business):** Includes 'Assign type...' and 'Define permissions...' buttons.
- Wake-up calls report:** Includes an 'Erase report...' button.
- Special settings:** Includes a 'Special settings...' button.

At the bottom, there is a checkbox 'Show number in tab names', an 'END SETTINGS' button, and a red text notification: 'Voice channels will need to reload'.

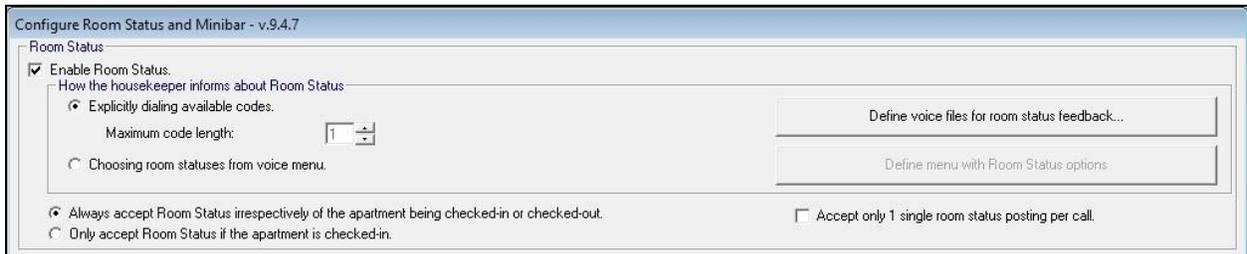
7.6. Administer Room Status and Minibar

From the **Settings** screen, select the **Hotel – Room Status and Minibar** tab to display the screen below. Click **Room Status and Minibar Settings**.



7.6.1. Room Status

The **Configure Room Status and Minibar** screen is displayed. In the **Room Status** subsection, check **Enable Room Status** and retain the default values in the remaining fields. Select **Define voice files for room status feedback**.



The **Voice file settings for Room Status** screen is displayed next. Follow reference [3] to create desired room status and corresponding voice files. In the compliance testing, three room status entries were created as shown below.

The screenshot shows the 'Voice file settings for Room Status - v.9.4.7' window. It contains a table with the following data:

Code	Description	Voice file	Exit
1	Clean and vacant	C:\Program Files (x86)\FLAvoice\Sound Files\Phrases for Room status\Clean and Vacant.wav	
2	Clean and occupied	C:\Program Files (x86)\FLAvoice\Sound Files\Phrases for Room status\Clean and Occupied.wav	
3	Dirty and vacant	C:\Program Files (x86)\FLAvoice\Sound Files\Phrases for Room status\DIV.wav	

7.6.2. Minibar

The **Configure Room Status and Minibar** screen is displayed again. In the **Minibar** sub-section, check **Enable Minibar** and retain the default values in the remaining fields. Select **Define voice files for Minibar Products feedback**.

Configure Room Status and Minibar - v.9.4.7

Room Status

Enable Room Status.

How the housekeeper informs about Room Status

Explicitly dialing available codes.

Maximum code length:

Choosing room statuses from voice menu.

Always accept Room Status irrespectively of the apartment being checked-in or checked-out.

Only accept Room Status if the apartment is checked-in.

Accept only 1 single room status posting per call.

Minibar

Enable Minibar.

Maximum code length: Maximum quantity length:

Always accept Minibar irrespectively of the apartment being checked-in or checked-out.

Only accept Minibar if the apartment is checked-in.

Accept only 1 single minibar posting per call.

Define voice files for room status feedback...

Define menu with Room Status options

Define voice files for Minibar Products feedback...

The **Voice file settings for Minibar products** screen is displayed next. Follow reference [3] to create desired minibar products and corresponding voice files. In the compliance testing, two minibar entries were created as shown below.

Voice files settings for Minibar products - v.9.4.7

Set the codes for products below, as well as their voice files

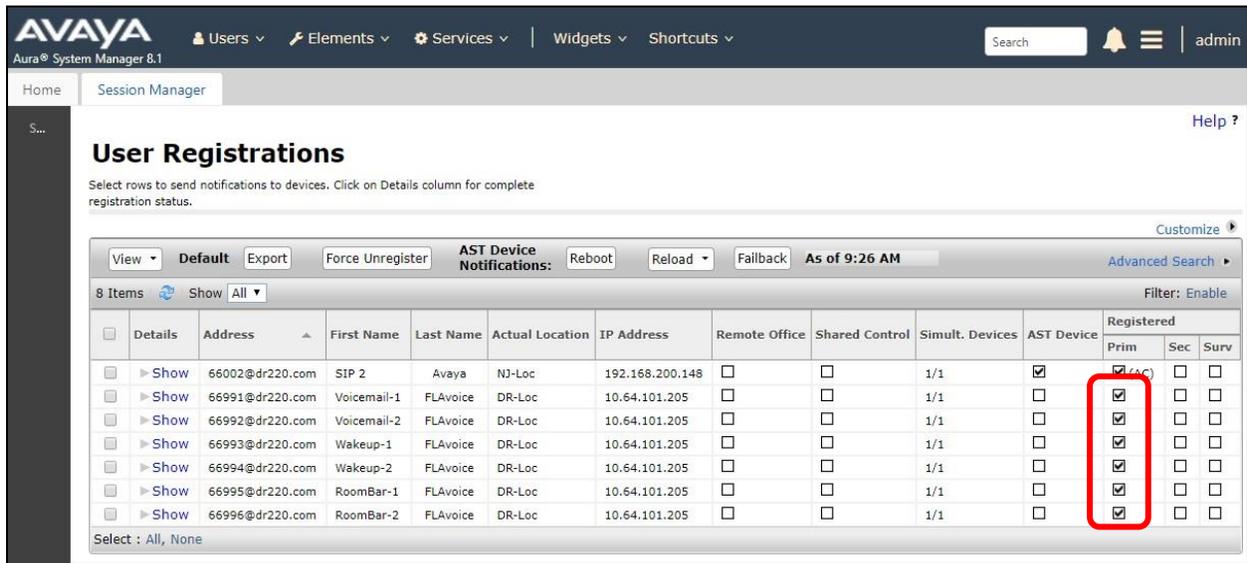
Code	Product	Price	Voice file (singular)
1	Chocolate bar	0	C:\Program Files (x86)\FLVoice\Sound Files\Products for Minibar\Chocolate bar.wav
2	Bottle of water	0	C:\Program Files (x86)\FLVoice\Sound Files\Products for Minibar\Bottle of water.wav

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager, Session Manager, and FLAvoice.

From the System Manager web-based interface, select **Elements** → **Session Manager** → **System Status** → **User Registrations** (not shown) to display the **User Registrations** screen.

Verify that all virtual SIP users from **Section 6.3** are registered, as shown below with a check in the **Registered Prim** column.



	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered		
											Prim	Sec	Surv
<input type="checkbox"/>	▶ Show	66002@dr220.com	SIP 2	Avaya	NJ-Loc	192.168.200.148	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	▶ Show	66991@dr220.com	Voicemail-1	FLAvoice	DR-Loc	10.64.101.205	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	▶ Show	66992@dr220.com	Voicemail-2	FLAvoice	DR-Loc	10.64.101.205	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	▶ Show	66993@dr220.com	Wakeup-1	FLAvoice	DR-Loc	10.64.101.205	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	▶ Show	66994@dr220.com	Wakeup-2	FLAvoice	DR-Loc	10.64.101.205	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	▶ Show	66995@dr220.com	RoomBar-1	FLAvoice	DR-Loc	10.64.101.205	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	▶ Show	66996@dr220.com	RoomBar-2	FLAvoice	DR-Loc	10.64.101.205	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

9. Conclusion

These Application Notes describe the configuration steps required for Global BHS FLVoice to successfully interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

10. Additional References

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

1. *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 6, March 2020, available at <http://support.avaya.com>.
2. *Administering Avaya Aura® Session Manager*, Release 8.1, Issue 4, May 2020, available at <http://support.avaya.com>.
3. *FLVoice USER'S MANUAL*, 9.3.1, available from <http://www.globalbhs.com/suporte/>.

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