



## Avaya Solution & Interoperability Test Lab

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# Application Notes for AEi Communications VM-9200-SMLT(S) SIP Telephone with Avaya Aura® Session Manager and Avaya Aura® Communication Manager - Issue 1.0

## Abstract

These Application Notes describe the steps required to integrate AEi Communications VM-9200-SMLT(S) SIP Telephones with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. VM-9200-SMLT(S) SIP Telephones serve the hospitality industry and provide the following features: speakerphone, hold and message waiting indicator (MWI). In the compliance test, VM-9200-SMLT(S) SIP Telephones successfully registered with Avaya Aura® Session Manager, established calls with the PSTN and other Avaya SIP and H.323 telephones, and executed telephony and hospitality features using Avaya Aura® Communication Manager Feature Access Codes (FACs) and Feature Name Extensions (FNEs).

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

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

# 1. Introduction

These Application Notes describe the steps required to integrate AEi Communications VM-9200-SMLT(S) SIP Telephones with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. VM-9200-SMLT(S) SIP Telephones serve the hospitality industry and provide the following features: speakerphone, hold, wake up calls, house-keeping status and message waiting indicator (MWI). In the compliance test, VM-9200-SMLT(S) SIP Telephones successfully registered with Avaya Aura® Session Manager, established calls with the PSTN and other Avaya SIP and H.323 telephones, and executed telephony and hospitality features using Avaya Aura® Communication Manager Feature Access Codes (FACs) and Feature Name Extensions (FNEs).

# 2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between AEi Communications VM-9200-SMLT(S) SIP Telephones and Avaya SIP and H.323 telephones. Basic telephony features, such as hold, speaker, and hospitality features including wake up calls and updating housekeeping status for a guest's room were also exercised. In addition, other extended telephony features, such as call forwarding and call pickup were also exercised using FACs and FNEs.

The serviceability testing focused on verifying that the AEi Communications VM-9200-SMLT(S) SIP Telephone comes back into service after re-connecting the Ethernet connection or rebooting the SIP phone.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and AEi Communications did not include use of any specific encryption features as requested by AEi Communications.

Avaya's formal testing and Declaration of Conformity is provided only on the headsets/handsets that carry the Avaya brand or logo. Avaya may conduct testing of non-Avaya headset/handset to determine interoperability with Avaya phones. However, Avaya does not conduct the testing of non-Avaya headsets/handsets for: Acoustic Pressure, Safety, Hearing Aid Compliance, EMC regulations, or any other tests to ensure conformity with safety, audio quality, long-term reliability or any regulation requirements. As a result, Avaya makes no representations whether a particular non-Avaya headset will work with Avaya's telephones or with a different generation of the same Avaya telephone.

Since there is no industry standard for handset interfaces, different manufacturers utilize different handset/headset interfaces with their telephones. Therefore, any claim made by a headset vendor that its product is compatible with Avaya telephones does not equate to a guarantee that the headset will provide adequate safety protection or audio quality.

## 2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of VM-9200-SMLT(S) with Session Manager.
- Calls between VM-9200-SMLT(S) and Avaya SIP and H.323 telephones with Direct IP-Media (Shuffling) enabled and disabled (see **Section 5.2**).
- Support of multiple incoming and outgoing calls, using L1 and L2.
- G.711 MU-Law codec support.
- Proper recognition of DTMF tones.
- Long call duration and long hold duration.
- Extended telephony features using Communication Manager FNEs and FACs, such as Hospitality Wakeup calls, Housekeeping Status Access Codes, Call Forwarding and Call Pickup.
- Voicemail coverage, MWI support, and logging into voicemail system to retrieve voicemail messages.
- Proper system recovery after a restart of the VM-9200-SMLT(S) and loss of IP connectivity.

## 2.2. Test Results

All test cases passed with the following observations noted:

- The VM-9200-SMLT(S) SIP Telephone does not support conference.
- The VM-9200-SMLT(S) SIP Telephone does not support transfer however a call can be transferred to it.
- The VM-9200-SMLT(S) SIP Telephone does not support the Long Hold Recall Timer feature.
- The VM-9200-SMLT(S) SIP Telephone does not support Multi-Device Access feature.
- The VM-9200-SMLT(S) SIP Telephone does not support Call Park feature however can Unpark a call if a call is parked by an Avaya telephone.

## 2.3. Support

For technical support on the AEi Communications VM-9200-SMLT(S) SIP Telephone, contact AEi Communications Support via phone, email, or website.

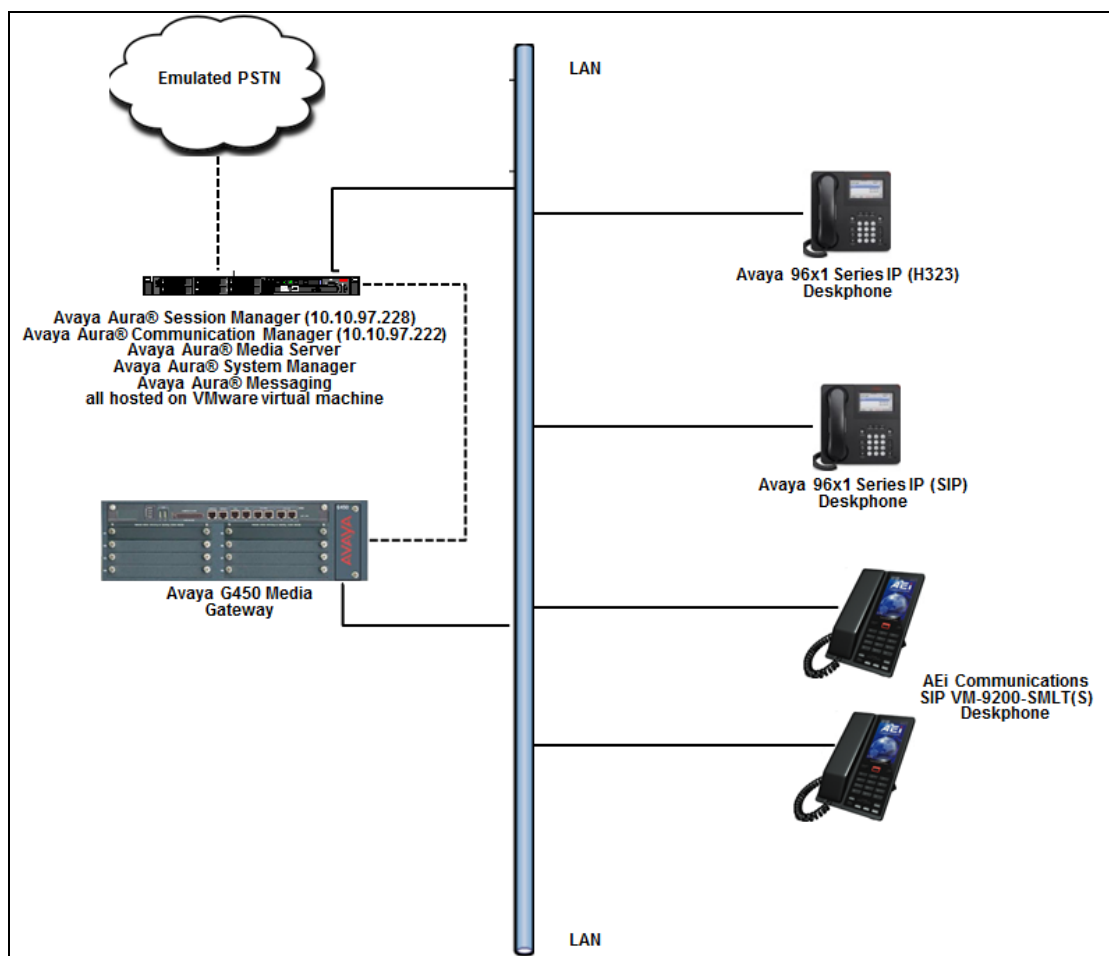
- **Phone:** +1 (650) 552-9416
- **Email:** [sales@aeicomcommunications.com](mailto:sales@aeicomcommunications.com)
- **Web:** <http://www.aeicomcommunications.com/contact.html>

### 3. Reference Configuration

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

- Avaya Aura® Communication Manager running in a virtual environment with an Avaya G450 Media Gateway and/or Avaya Aura® Media Server.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya Aura® Messaging served as the voicemail system.
- Avaya 96x1 Series SIP and H.323 Telephones.
- AEi Communications VM-9200-SMLT(S) SIP Telephones.

AEi Communications VM-9200-SMLT(S) SIP Telephones registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.



**Figure 1: Avaya SIP Network with AEi Communications VM-9200-SMLT(S) SIP Telephones**

## 4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya Aura® Communication Manager in a Virtual Environment	7.1.0.0.532
Avaya Aura® Media-Server in a Virtual Environment	7.8.0.312
Avaya Aura® System Manager in a Virtual Environment	7.1.0.0
Avaya Aura® Session Manager in a Virtual Environment	7.1.0.0.710028
Avaya Aura® Messaging	7.0.1.2.0-FP1SP2
Avaya G450 Media Gateway	38.18.0/1
Avaya 9611 IP Deskphones	6.6401 (H.323)
Avaya 9641GS IP Deskphones	7.0.1.2.9 (SIP)
AEi Communications VM-9200-SMLT(S) SIP Telephone	VM2SLTD_A37

## 5. Configure Avaya Aura® Communication Manager

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

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

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

**Note:** It is assumed that basic configuration of the Communication Manager has already been completed, such as the SIP trunk to Session Manager. The SIP station configuration for AEi Communications VM-9200-SMLT(S) is configured through Avaya Aura® System Manager in **Section 6.2**.

### 5.1. Verify License

Using the SAT, verify that the Off-PBX Telephones (OPS) option is enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

On **Page 1**, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

```
display system-parameters customer-options                               Page 1 of 12
                                OPTIONAL FEATURES

G3 Version: V17                                     Software Package: Enterprise
Location: 2                                           System ID (SID): 1
Platform: 28                                          Module ID (MID): 1

                                USED
Platform Maximum Ports: 65000 129
Maximum Stations: 41000 42
Maximum XMOBILE Stations: 41000 0
Maximum Off-PBX Telephones - EC500: 41000 0
Maximum Off-PBX Telephones - OPS: 41000 15
Maximum Off-PBX Telephones - PBFMC: 41000 0
Maximum Off-PBX Telephones - PVFMC: 41000 0
Maximum Off-PBX Telephones - SCCAN: 0 0
Maximum Survivable Processors: 313 0

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

## 5.2. Administer IP Network Region and IP Codec Set

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *bvwdev.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G450 Media Gateway or Avaya Aura® Media Server. Note that this is also dependent on the **Direct IP-IP Audio Connections** value defined in the **signaling-group** form (not shown). The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager.

change ip-network-region 1		Page 1 of 20	
IP NETWORK REGION			
Region: 1	NR Group: 1		
Location: 1	<b>Authoritative Domain: bvwdev.com</b>		
Name: Region1	Stub Network Region: n		
MEDIA PARAMETERS	<b>Intra-region IP-IP Direct Audio: yes</b>		
<b>Codec Set: 1</b>	<b>Inter-region IP-IP Direct Audio: yes</b>		
UDP Port Min: 2048	IP Audio Hairpinning? y		
UDP Port Max: 8001			
DIFFSERV/TOS PARAMETERS			
Call Control PHB Value: 46			
Audio PHB Value: 46			
Video PHB Value: 26			
802.1P/Q PARAMETERS			
Call Control 802.1p Priority: 6			
Audio 802.1p Priority: 6			
Video 802.1p Priority: 5			
H.323 IP ENDPOINTS		AUDIO RESOURCE RESERVATION PARAMETERS	
H.323 Link Bounce Recovery? y		RSVP Enabled? n	
Idle Traffic Interval (sec): 20			
Keep-Alive Interval (sec): 5			
Keep-Alive Count: 5			



In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to the VM-9200-SMLT(S) SIP Telephone. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set 1 was specified in IP Network Region 1 shown above. The default settings of the **IP Codec Set** form are shown below. The VM-9200-SMLT(S) SIP Telephone supports G.711Mu.

change ip-codec-set 1

Page 1 of 2

IP MEDIA PARAMETERS

Codec Set: 1

Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)
1: <b>G.711MU</b>	n	2	20
2:			
3:			
4:			
5:			
6:			
7:			

Media Encryption

Encrypted SRTCP: enforce-unenc-srtcp

1: 1-srtp-aescm128-hmac80

2: 2-srtp-aescm128-hmac32

3: none

4:

## 5.3. Administer Hospitality Features

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

### 5.3.1. Administer Feature Name Extensions (FNEs)

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

```
change off-pbx-telephone feature-name-extensions set 1          Page 2 of 3
EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME

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

### 5.3.2. Administer Features Access Codes (FACs)

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

change feature-access-codes	Page 8 of 11
FEATURE ACCESS CODE (FAC) Hospitality Features	
<b>Automatic Wakeup Call Access Code: *60</b>	
<b>Housekeeping Status (Client Room) Access Code: *61</b>	
Housekeeping Status (Client Room) Access Code: *62	
Housekeeping Status (Client Room) Access Code: *63	
Housekeeping Status (Client Room) Access Code: *64	
Housekeeping Status (Client Room) Access Code: *65	
Housekeeping Status (Client Room) Access Code:	
Housekeeping Status (Station) Access Code:	
Housekeeping Status (Station) Access Code:	
Housekeeping Status (Station) Access Code:	
Housekeeping Status (Station) Access Code:	
Verify Wakeup Announcement Access Code:	
Voice Do Not Disturb Access Code:	

### 5.3.3. Allow Wake-up Calls

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

change system-parameters hospitality	Page 2 of 3
HOSPITALITY	
Dual Wakeups? n      Daily Wakeup? n      VIP Wakeup? n	
<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: integrated	
Integrated Announcement Extension: 56003	
Length of Time to Remain Connected to Announcement: 30	
Extension to Receive Failed Wakeup LWC Messages:	
Routing Extension on Unavailable Voice Synthesis:	
Display Room Information in Call Display? n	
Automatic Selection of DID Numbers? n	
Custom Selection of VIP DID Numbers? n	
Number of Digits from PMS:	
PMS Sends Prefix? n	
Number of Digits in PMS Coverage Path: 3	
Digit to Insert/Delete:	

### 5.3.4. Allow Housekeeping Status Updates

To allow housekeeping to change the housekeeping status of a guests room by dialing the appropriate access code, **Client Room** must be enabled on the COS assigned to the SIP phone. In this example, **Client Room** was enabled for COS 1, which was assigned to the AEi VM-9200-SMLT(S) phone.

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

## 6. Configure Avaya Aura® Session Manager

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

- Launch System Manager
- Administer SIP User

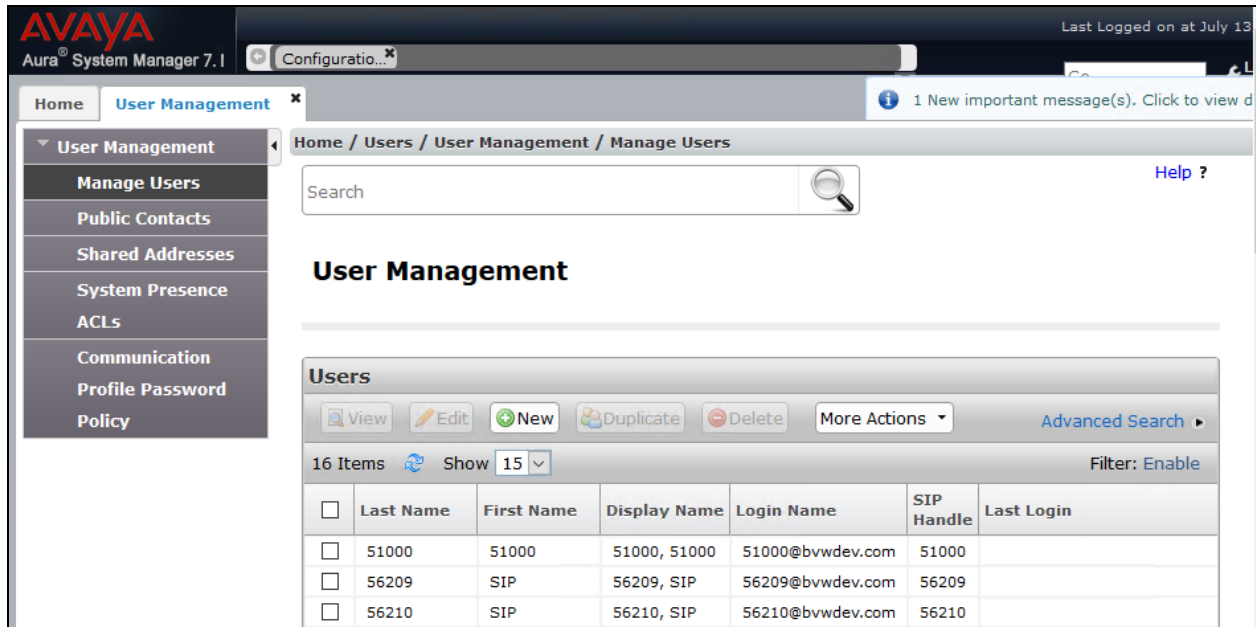
**Note:** It is assumed that basic configuration of Session Manager has already been performed. This section will focus on the configuration of a SIP user for the VM-9200-SMLT(S) SIP Telephone.

### 6.1. Launch System Manager

Access the System Manager Web interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of the System Manager server. Log in using the appropriate credentials.

## 6.2. Administer SIP User

In the subsequent screen (not shown), select **Users** → **User Management** → **Manage Users** to display the **User Management** screen below. Click **New** to add a user.



The screenshot shows the Avaya Aura System Manager 7.1 User Management interface. The top navigation bar includes 'Home' and 'User Management'. The left sidebar lists various management options, with 'Manage Users' selected. The main content area displays a table of users. The table has columns for 'Last Name', 'First Name', 'Display Name', 'Login Name', 'SIP Handle', and 'Last Login'. There are 16 items in the table, and the first three rows are visible.

	Last Name	First Name	Display Name	Login Name	SIP Handle	Last Login
<input type="checkbox"/>	51000	51000	51000, 51000	51000@bvwdev.com	51000	
<input type="checkbox"/>	56209	SIP	56209, SIP	56209@bvwdev.com	56209	
<input type="checkbox"/>	56210	SIP	56210, SIP	56210@bvwdev.com	56210	

### 6.2.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter “<ext>@<domain>”, where “<ext>” is the desired VM-9200-SMLT(S) SIP extension and “<domain>” is the applicable SIP domain name from **Section 5.2**. Retain the default values in the remaining fields.

The screenshot displays the Avaya Aura System Manager 7.1 interface. The top navigation bar includes the Avaya logo and the text 'Aura System Manager 7.1'. A breadcrumb trail shows 'Home / Users / User Management / Manage Users'. The left sidebar contains a 'User Management' menu with options: 'Manage Users', 'Public Contacts', 'Shared Addresses', 'System Presence', 'ACLs', 'Communication', 'Profile Password', and 'Policy'. The main content area is titled 'New User Profile' and features four tabs: 'Identity' (selected), 'Communication Profile', 'Membership', and 'Contacts'. Below the tabs, there is a 'User Provisioning Rule' dropdown menu. The 'Identity' section contains the following fields:

- \* Last Name: 56211
- Last Name (Latin Translation): 56211
- \* First Name: AEi SIP
- First Name (Latin Translation): AEi SIP
- Middle Name: (empty)
- Description: (empty text area)
- Update Time : June 20, 2017 4:14:42 P
- \* Login Name: 56211@bvwddev.com

Buttons for 'Commit & Continue', 'Commit', and 'Cancel' are located at the top right of the form area.

### 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.

The screenshot shows the Avaya Aura System Manager 7.1 web interface. The left sidebar contains a 'User Management' menu with options like 'Manage Users', 'Public Contacts', 'Shared Addresses', 'System Presence', 'ACLs', 'Communication', 'Profile Password', and 'Policy'. The main content area is titled 'New User Profile' and has tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' tab is active, showing fields for 'Communication Profile Password' and 'Confirm Password', both masked with dots. There is a 'Generate' link next to the confirm password field. At the top right, there are buttons for 'Commit & Continue', 'Commit', and 'Cancel'. A notification bar at the top indicates '1 New important message(s). Click to view'.

### 6.2.3. Communication Address

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

The screenshot shows the 'Communication Address' form. At the top, there are buttons for 'New', 'Edit', and 'Delete'. Below these is a table with columns 'Type', 'Handle', and 'Domain'. The table is currently empty, showing 'No Records found'. Below the table, there are input fields for 'Type' (set to 'Avaya SIP'), 'Fully Qualified Address' (with a sub-field for 'Address' containing '56211'), and 'Domain' (set to 'bvwddev.com'). There are 'Add' and 'Cancel' buttons at the bottom right.



## 6.2.4. Session Manager Profile

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

☒ **Session Manager Profile**

**SIP Registration**

\* Primary Session Manager

DevvmSM

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices

4

Block New Registration When  
Maximum Registrations  
Active?

☐

Primary	Secondary	Maximum
15	0	15

**Application Sequences**

Origination Sequence

DevvmCM\_AppSeq

Termination Sequence

DevvmCM\_AppSeq

**Call Routing Settings**

\* Home Location

Belleville

Conference Factory Set

(None)

**Call History Settings**

Enable Centralized Call  
History?

☐

### 6.2.5. CM Endpoint Profile

Scroll down to check and expand **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager, and select *Endpoint* for **Profile Type**. For **Extension**, enter the SIP user extension from **Section 6.2.1**. For **Template**, select *9641SIP\_DEFAULT\_CM\_7\_1*. Retain the default values in the remaining fields.

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

☒ **CM Endpoint Profile** ▼

\* System

DevvmCM ▼

\* Profile Type

Endpoint ▼

Use Existing Endpoints

☐

\* Extension

56211

Endpoint Editor

Template

9641SIP\_DEFAULT\_CM\_7\_1 ▼

Set Type

9641SIP

Security Code

Port

IP

Voice Mail Number

Preferred Handle

(None) ▼

Calculate Route Pattern

☐

Sip Trunk

aar

Enhanced Callr-Info display for 1-line phones

☐

Delete Endpoint on Unassign of Endpoint from User or on Delete User.

☒

Override Endpoint Name and Localized Name

☒

Allow H.323 and SIP Endpoint Dual Registration

☐

In the **CM Endpoint Profile** sub-section (not shown), click the **Endpoint Editor** button to display the page below. In the **General Options** tab, specify that coverage path that points to the voicemail system in the **Coverage Path 1** field. This provides voicemail coverage for the SIP user. In this example, coverage path 2 was used.

## Edit Endpoint

DoneCancel

[Save As Template]

SystemDevvmCM

Extension56211

Template9641\_DEFAULT\_CM\_7\_1

Set Type9641

Port500068

Security Code

Name56211, AEi SIP

General Options (G) \*

Feature Options (F)

Site Data (S)

Abbreviated Call Dialing (A)

Enhanced Call Fwd (E)

Button Assignment (B)

Group Membership (M)

\* Class of Restriction (COR)

1

\* Class Of Service (COS)

1

\* Emergency Location Ext

56211

\* Message Lamp Ext.

56211

\* Tenant Number

1

Coverage Path 2

Coverage Path 1

2

Localized Display Name

56211, AEi SIP

Lock Message

☐

Multibyte Language

Not Applicable

\*Required

DoneCancel

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

<b>General Options (G) *</b>		<b>Feature Options (F)</b>		<b>Site Data (S)</b>		<b>Abbreviated Call Dialing (A)</b>	
<b>Enhanced Call Fwd (E)</b>		<b>Button Assignment (B)</b>		<b>Group Membership (M)</b>			
<b>Active Station Ringing</b>	single	<b>Multimedia Mode</b>	enhanced	<b>MWI Served User Type</b>	sip-adjunct		
<b>Auto Answer</b>	none						

## 7. Configure AEi Communications VM-9200-SMLT(S) SIP Telephone

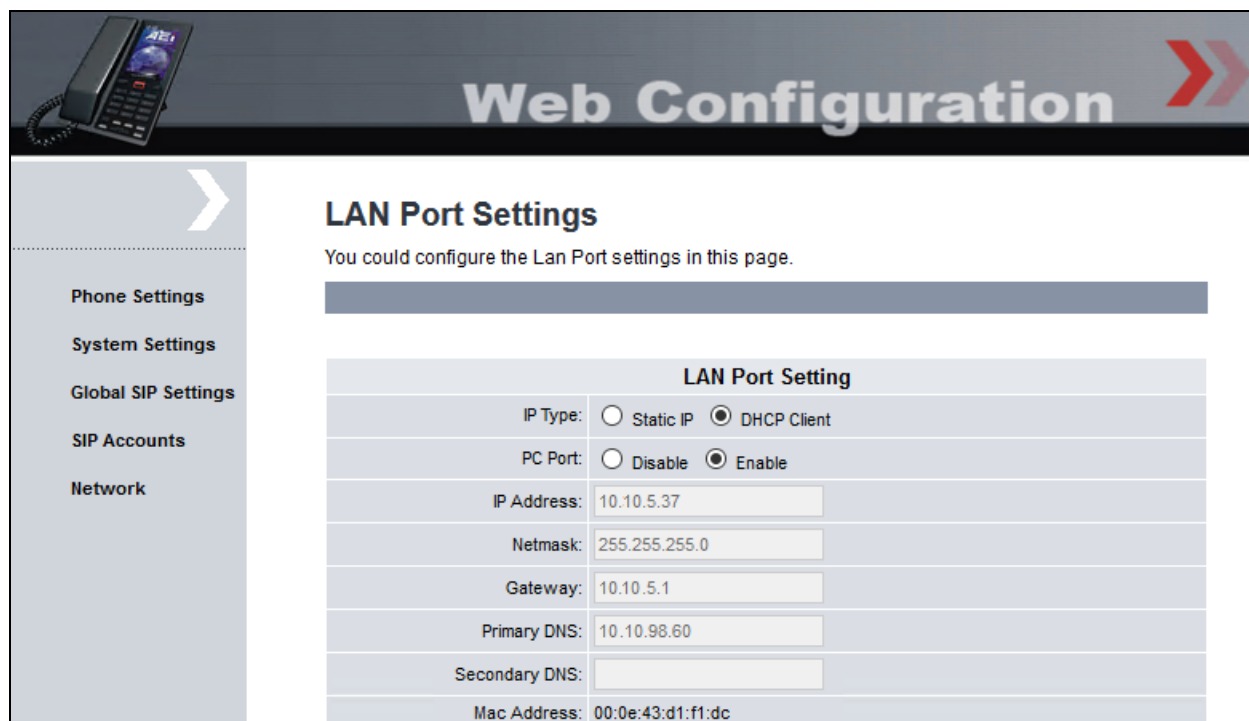
Access the VM-9200-SMLT(S) web interface by using the URL “https://ip-address:8000” in an Internet browser window, where “ip-address” is the IP address of the SIP phone. Log in using the appropriate credentials and then click **Login**.



The image shows a login screen for a VOIP PHONE. It has a dark header with the text "VOIP PHONE" in white. Below the header, the word "Login:" is in red. There are two input fields: "Username:" and "Password:". A "login" button is located at the bottom right of the form.

### 7.1. Administer LAN Port Settings

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



The image shows the "Web Configuration" interface for a SIP phone. The left sidebar contains a menu with the following items: Phone Settings, System Settings, Global SIP Settings, SIP Accounts, and Network. The "Network" item is selected. The main content area is titled "LAN Port Settings" and contains the text "You could configure the Lan Port settings in this page." Below this text is a table with the following settings:

LAN Port Setting	
IP Type:	<input type="radio"/> Static IP <input checked="" type="radio"/> DHCP Client
PC Port:	<input type="radio"/> Disable <input checked="" type="radio"/> Enable
IP Address:	10.10.5.37
Netmask:	255.255.255.0
Gateway:	10.10.5.1
Primary DNS:	10.10.98.60
Secondary DNS:	
Mac Address:	00:0e:43:d1:f1:dc

## 7.2. Administer SIP Accounts

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

**Web Configuration**

**SIP Accounts**

You could set information of service domains in this page.

Display Name	Registration Server	Status	Registration	Select
				<input type="checkbox"/>
				<input type="checkbox"/>
				<input type="checkbox"/>

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

- **Transport:** Select the transport protocol (e.g., *UDP*).
- **Registration ID:** Specify the Registration ID (e.g., *56211*, the SIP extension).
- **Display Name:** Specify the Display Name (e.g., *56211*, the SIP extension).
- **Authentication Name:** Specify the SIP extension of the VM-9200-SMLT(S) SIP Telephone (e.g., *56211*).
- **Password:** Specify the SIP password configured in **Section 6.2.2**.
- **Registration Server:** Set to the domain name and port (e.g., *bvwddev.com:5060*).
- **Proxy Server:** Set to the Session Manager IP address and port (e.g., *10.10.97.228:5060*).
- **Voice Mail:** Specify the voicemail pilot number if needed.
- **MWI:** Set to *Enable*.
- Retain the default values in the remaining fields.

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

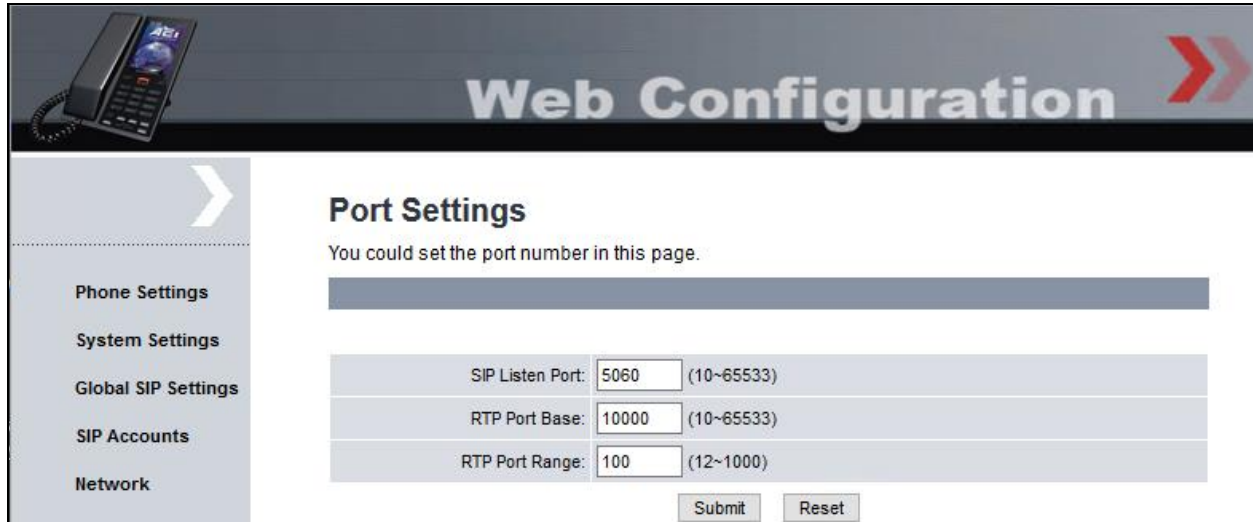
The image shows a web configuration interface for a SIP account. At the top, there is a header with a phone icon and the text "Web Configuration". Below this is a sidebar with navigation links: "Phone Settings", "System Settings", "Global SIP Settings", "SIP Accounts", and "Network". The main content area is titled "SIP Account Settings" and includes a sub-header "SIP Account 1". Below this, there is a form with various settings for the SIP account. The status at the bottom is "registered".

SIP Account 1	
Transport :	UDP ▾
Registration:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Registration ID:	56211
Display Name:	56211
Authentication Name:	56211
Password:	••••
Registration Server:	bvwdev.com:5060
Proxy Server:	10.10.97.228:5060
Proxy Address:	
Voice Mail:	
Expire Time:	60 ▾
DTMF Type:	RFC2833 ▾
Send KeepAlive:	Disable ▾
MWI:	Enable ▾
Mode:	Multi ▾
DNSSRV:	Disable ▾
SRTP :	Disable ▾
Status:	registered

Submit Cancel

### 7.3. Administer Global SIP Settings

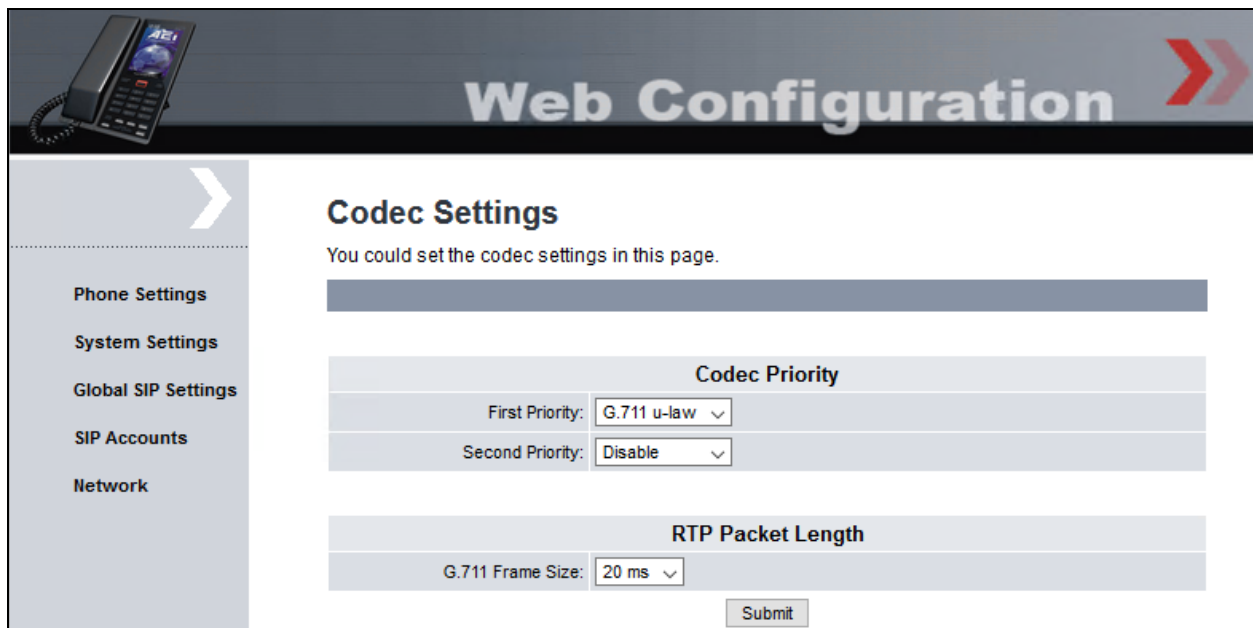
Navigate to **Global SIP Settings** → **Port Settings** and verify the SIP Listen Port being used (e.g., 5060).



The screenshot shows the 'Web Configuration' interface for 'Port Settings'. On the left is a navigation menu with 'Phone Settings', 'System Settings', 'Global SIP Settings', 'SIP Accounts', and 'Network'. The main area is titled 'Port Settings' and includes a sub-header 'You could set the port number in this page.' Below this is a table with three rows: 'SIP Listen Port' with a value of 5060 and a range of (10~65533), 'RTP Port Base' with a value of 10000 and a range of (10~65533), and 'RTP Port Range' with a value of 100 and a range of (12~1000). At the bottom right are 'Submit' and 'Reset' buttons.

Port Settings	
You could set the port number in this page.	
SIP Listen Port:	5060 (10~65533)
RTP Port Base:	10000 (10~65533)
RTP Port Range:	100 (12~1000)
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

Navigate to **Global SIP Settings** → **Codec Settings** to verify the codec priority. In this example, the first priority is *G.711u-law*. AEi Communications VM-9200-SMLT(S) SIP Telephones supports G.711.

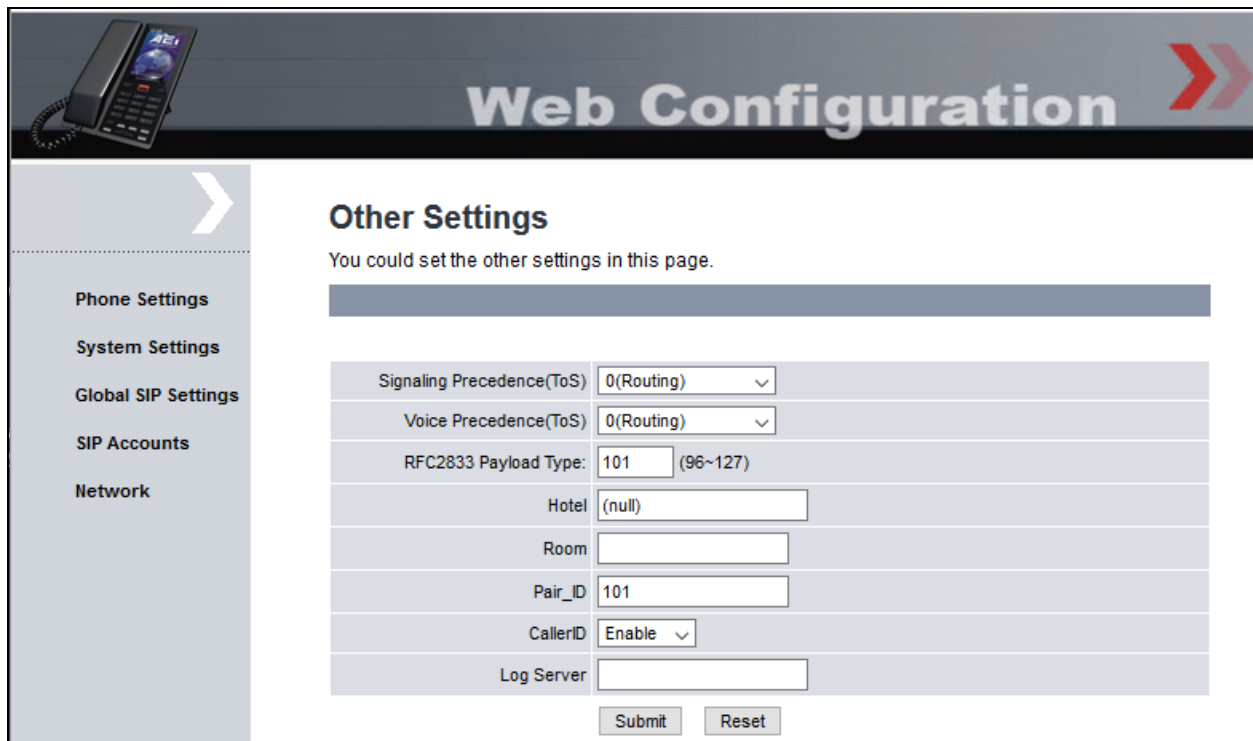


The screenshot shows the 'Web Configuration' interface for 'Codec Settings'. On the left is a navigation menu with 'Phone Settings', 'System Settings', 'Global SIP Settings', 'SIP Accounts', and 'Network'. The main area is titled 'Codec Settings' and includes a sub-header 'You could set the codec settings in this page.' Below this is a table with two sections: 'Codec Priority' and 'RTP Packet Length'. The 'Codec Priority' section has two rows: 'First Priority' with a dropdown menu showing 'G.711 u-law' and 'Second Priority' with a dropdown menu showing 'Disable'. The 'RTP Packet Length' section has one row: 'G.711 Frame Size' with a dropdown menu showing '20 ms'. At the bottom right is a 'Submit' button.

Codec Settings	
You could set the codec settings in this page.	
<b>Codec Priority</b>	
First Priority:	G.711 u-law
Second Priority:	Disable
<b>RTP Packet Length</b>	
G.711 Frame Size:	20 ms
<input type="button" value="Submit"/>	



Navigate to **Global SIP Settings** → **Other Settings**, and verify the CallerID was set to *Enable*, so that the calling party ID can be displayed during the conversation. Retain default values for all other fields.



The image shows a web configuration interface for a device. At the top, there is a header with a telephone icon on the left, the text "Web Configuration" in the center, and a red double arrow icon on the right. Below the header, on the left side, is a vertical navigation menu with a white arrow pointing right. The menu items are: "Phone Settings", "System Settings", "Global SIP Settings" (which is highlighted), "SIP Accounts", and "Network". The main content area is titled "Other Settings" and contains the text "You could set the other settings in this page." Below this text is a table of settings. The table has two columns: the setting name and the value. The settings are: "Signaling Precedence(ToS)" with value "0(Routing)", "Voice Precedence(ToS)" with value "0(Routing)", "RFC2833 Payload Type:" with value "101" and a range "(96~127)", "Hotel" with value "(null)", "Room" with an empty field, "Pair\_ID" with value "101", "CallerID" with value "Enable", and "Log Server" with an empty field. At the bottom of the table are two buttons: "Submit" and "Reset".

Setting	Value
Signaling Precedence(ToS)	0(Routing)
Voice Precedence(ToS)	0(Routing)
RFC2833 Payload Type:	101 (96~127)
Hotel	(null)
Room	
Pair_ID	101
CallerID	Enable
Log Server	

## 8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the AEi Communications VM-9200-SMLT(S) SIP Telephone with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

1. Verify that the VM-9200-SMLT(S) SIP Telephone has successfully registered with Session Manager. In System Manager, navigate to **Elements → Session Manager → System Status → User Registrations** to check the registration status. The SIP registration status can also be seen in the SIP Account page of the VM-9200-SMLT(S) web interface seen in **Section 7.2**.

The screenshot shows the Avaya Aura System Manager 7.1 web interface. The top navigation bar includes the Avaya logo, the text 'Aura® System Manager 7.1', and a 'Configurati...' tab. The right side of the top bar shows 'Last Logged on at July 14, 2017 3:04 PM' and a 'Log off admin' button. The main navigation menu on the left includes 'Session Manager', 'Dashboard', 'Session Manager Administration', 'Communication Profile Editor', 'Network Configuration', 'Device and Location Configuration', and 'Application Configuration'. The main content area is titled 'User Registrations' and includes a breadcrumb trail: 'Home / Elements / Session Manager / System Status / User Registrations'. Below the title, there is a 'Select rows to send notifications to devices. Click on Details column for complete registration status.' instruction. The table below has columns: 'View' (Default), 'Force Unregister', 'AST Device Notifications' (Reboot, Reload, Failback), 'As of 4:10 PM', 'Customize', and 'Advanced Search'. The table itself has columns: 'Details', 'Address', 'First Name', 'Last Name', 'Actual Location', 'IP Address', 'Remote Office', 'Shared Control', 'Simult. Devices', 'AST Device', and 'Registered' (Prim, Sec, Surv). The first row of data shows a device with address '56211@bvwdev.com', first name 'AEi SIP', last name '56211', and actual location 'Belleville'. The 'Registered' column shows 'Prim' as checked, 'Sec' as unchecked, and 'Surv' as unchecked.

2. Verify basic telephony features by establishing calls between a VM-9200-SMLT(S) SIP Telephone with another VM-9200-SMLT(S) SIP Telephone and also with Avaya deskphones.

## 9. Conclusion

These Application Notes have described the administration steps required to integrate the AEi Communications VM-9200-SMLT(S) SIP Telephone with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The AEi Communications VM-9200-SMLT(S) SIP Telephone successfully registered with Session Manager and basic telephony and hospitality features were verified. All test cases passed with observations noted in **Section 2.2**.

## 10. References

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

1. *Administering Avaya Aura® Session Manager*, Release 7.1, Issue 1 May 2017
2. *Deploying Avaya Aura® System Manager*, Release 7.1, Issue 1 May 2017
3. *Administering Avaya Aura® System Manager for Release 7.1*, Release 7.1, Issue 2 May 2017
4. *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 7.1, Issue 1 May 2017

The following document was provided by AEi Communications.

1. *Configuring Hospitality SVM-9x00-SMG SIP IP Phone*, Version 1.0, Date: 28/09/16

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