

#### Avaya Solution & Interoperability Test Lab

Application Notes for Movitas Mobile SIP Apps with Avaya Aura® Session Manager and Avaya Aura® Communication Manager – Issue 1.0

#### **Abstract**

These Application Notes describe configuration steps required for Movitas Mobile SIP Apps with Avaya Aura<sup>®</sup> Session Manager and Avaya Aura<sup>®</sup> Communication Manager.

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

The objective of this compliance test was to validate the interoperability of Avaya Aura<sup>®</sup> Session Manager and Avaya Aura<sup>®</sup> Communication Manger with Movitas's MvPBX system and its respective iPhone, Android and Blackberry applications. The testing included multiple call scenarios including calls between desk phones and Movitas application users and calls to outside PSTN lines through the Avaya Aura<sup>®</sup> Communication Manager.

# 2. General Test Approach and Test Results

The general test approach was to verify that calls are routed to/from Avaya Aura<sup>®</sup> Communication Manager and Avaya Aura<sup>®</sup> Session Manager to MvPBX and PSTN. Avaya Aura<sup>®</sup> Communication Managers Extension to Cellular (EC500) feature was used to send calls to two phones at the same time. EC500 feature provides the ability to route calls to an Avaya Extension and a non-Avaya phone, at the same time. During interoperability testing, calls were routed to an Avaya extension and a SIP phone using EC500.

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.

### 2.1. Interoperability Compliance Testing

During the compliance there were 4 extensions associated with Avaya Aura<sup>®</sup> Communication Manager and Avaya Aura<sup>®</sup> Session Manager; Front Desk, Room 1, Room 2, Room 3; Room extensions were associated with 3 respective SIP Users in the Movitas MvPBX system.

The Movitas SIP User is a SIP application installed on a smart mobile devices; iPhone, Android and Blackberry. These SIP Users are registered to the Movitas MvPBX system. Any call that comes in to one of the three Avaya extensions will also ring their associated SIP User in the smart phone. The call can be accepted either on the extension or the SIP User. Whichever phone answers the call, the other stops ringing and becomes idle.

The focus of this testing was to verify the SIP Trunk interoperability between Avaya Aura<sup>®</sup> Session Manager and Movitas MvPBX in placing multiple call scenarios between two systems. The following test areas were practiced in the compliance testing:

- SIP Trunk registration of Movitas MvPBX to Avaya Aura® Session Manager.
- Calls from PSTN to Avaya Extension and SIP User.
- Calls from SIP User to Avaya Extension.
- Calls from Avaya extension to another Avaya Extension and a SIP User.
- Calls from SIP User to a PSTN Line

- Calls from SIP User to another SIP User and Avaya Extension
- SIP settings compatibility for codec support and packet size negotiation.

#### 2.2. Test Results

All Test Cases Passed

#### 2.3. Support

Technical support for Movitas can be obtained through the following:

Phone: 888-343-3721

• Email: support@movitas.com

# 3. Reference Configuration

Figure 1 illustrates the compliance test configuration consisting of:

- Avaya Aura® Communication Manager
- Avaya Aura® Session Manager
- Avaya G430 and G450 Media Gateway
- Avaya IP Phones
- Movitas SIP Server

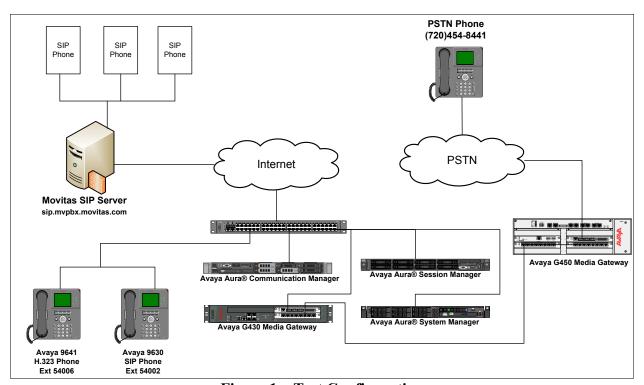


Figure 1 – Test Configuration

# 4. Equipment and Software Validated

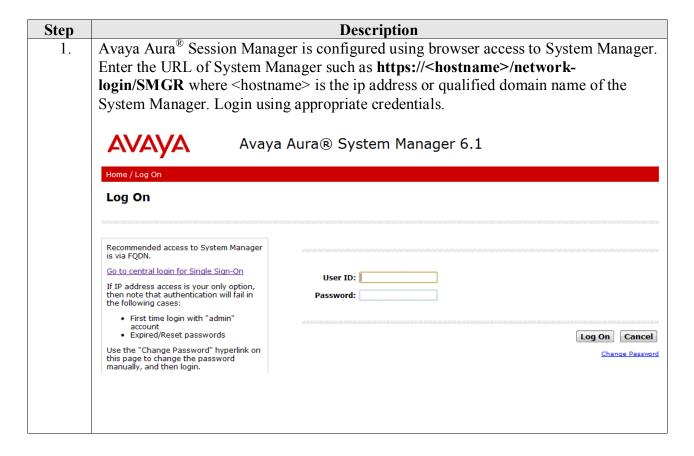
The following equipment and version were used in the reference configuration described above:

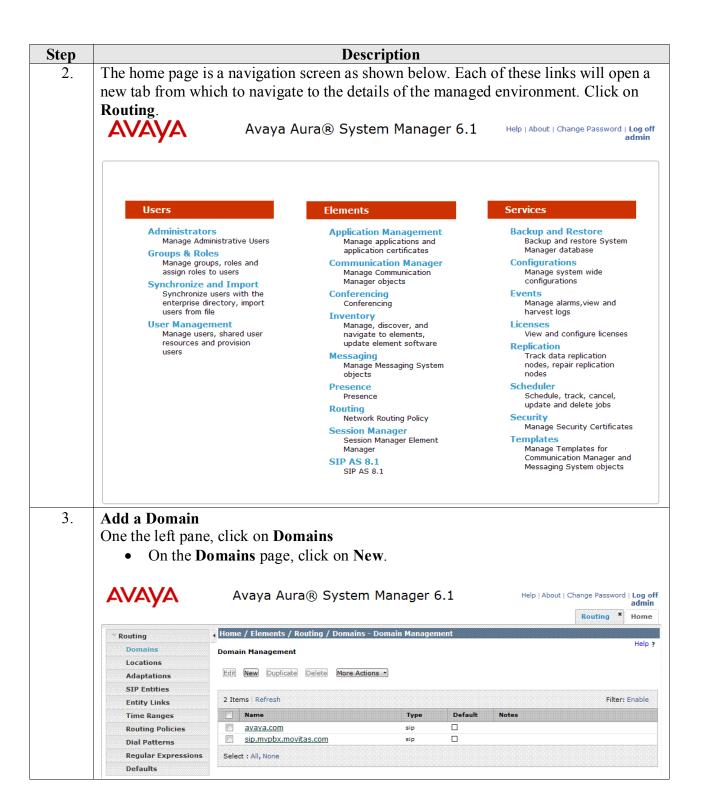
Component	Firmware Version	Description
Dell R610 Server	6.0.1	Runs Avaya Aura®
Avaya Aura® Communication	SP 00.1.510.1-19528	Communication Manager call
Manager		processing software.
Avaya G450 Media Gateway:	HW1, FW 31.20.1	Supports analog, DCP and IP
MM710BP (DS1)	HW11, FW044	phones.
MM712AP (DCP)	HW07, FW009	
Avaya Aura® Session Manager	6.1	SIP Routing Engine
Avaya Aura <sup>®</sup> System Manager	6.1	Administers Session Manager
Movitas MvPBX	ab6eb8c2	Movitas PBX
Blackberry 9700	6.6.0.195	Runs Movitas Phone app
MovitasPhone	1.0	
iPhone 4	5.1	Runs Movitas Phone app
MovitasPhone	1.5.0	
Android Incredible	2.3.4	Runs Movitas Phone app
MovitasPhone	2.0.1	

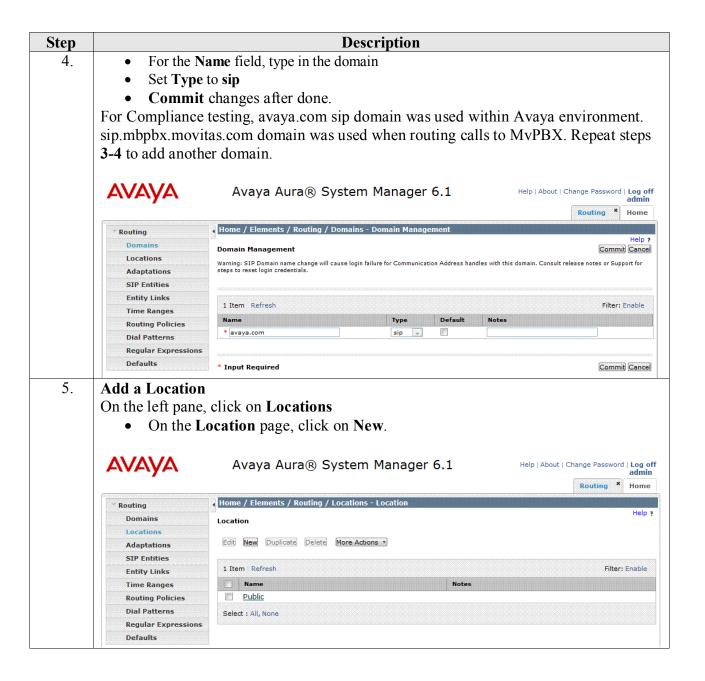
# 5. Configure Avaya Aura® Session Manager

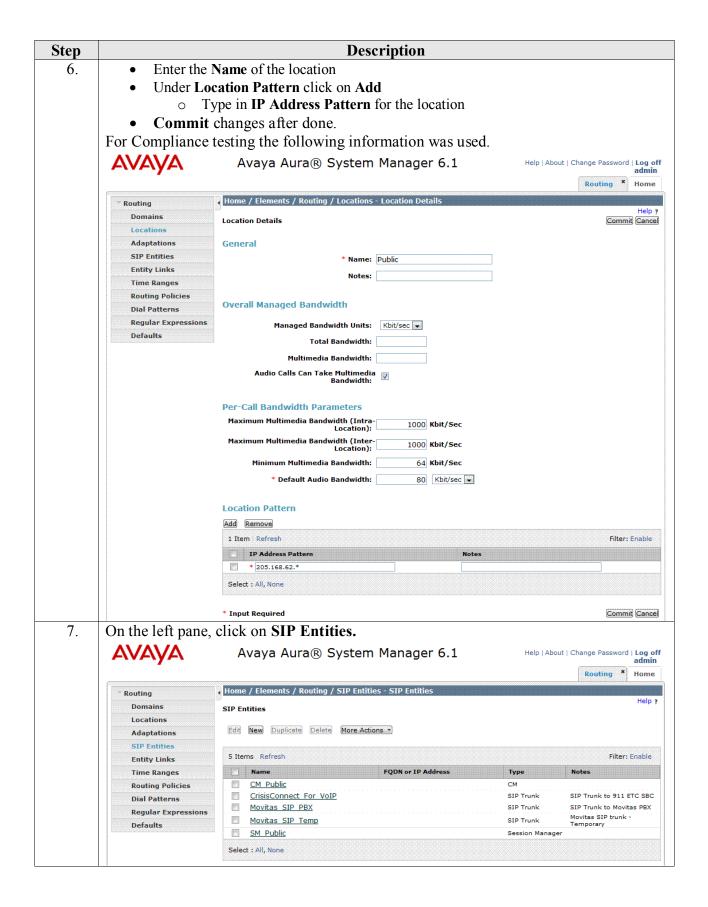
This section provides the steps for configuring Session Manager to communicate with Movitas MvPBX. For more details, see the administration guide.

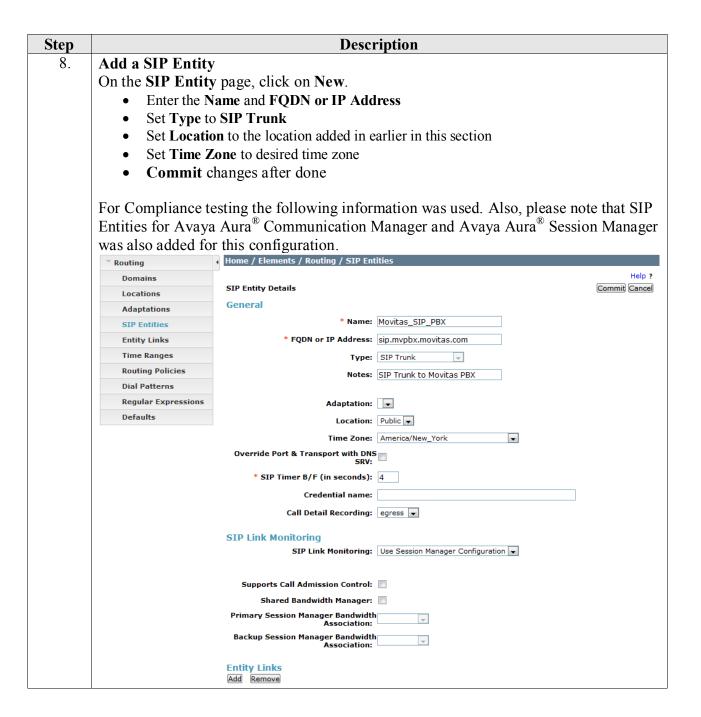
### 5.1. Configuration details

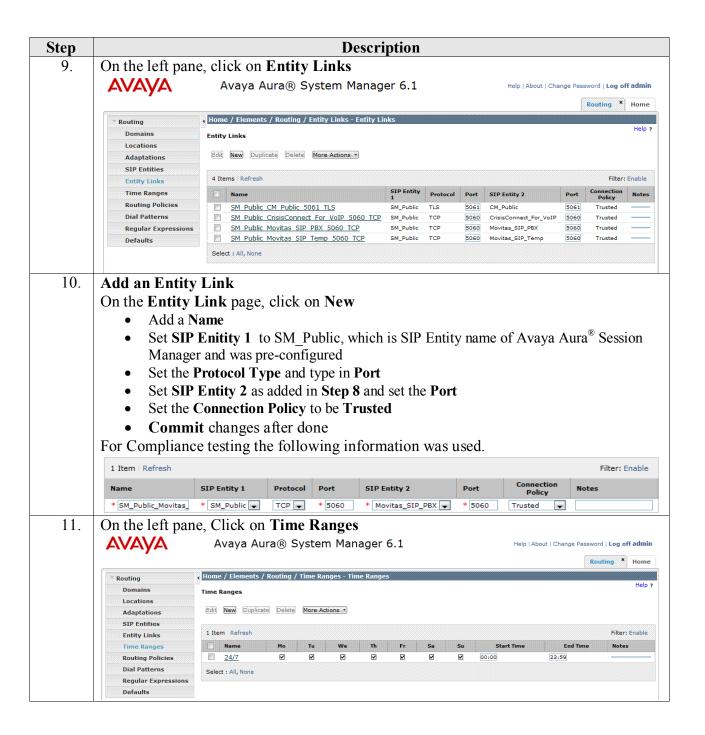


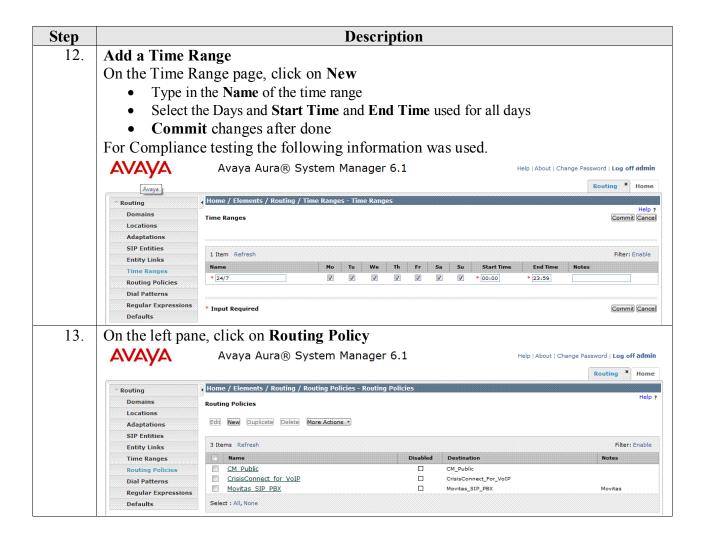


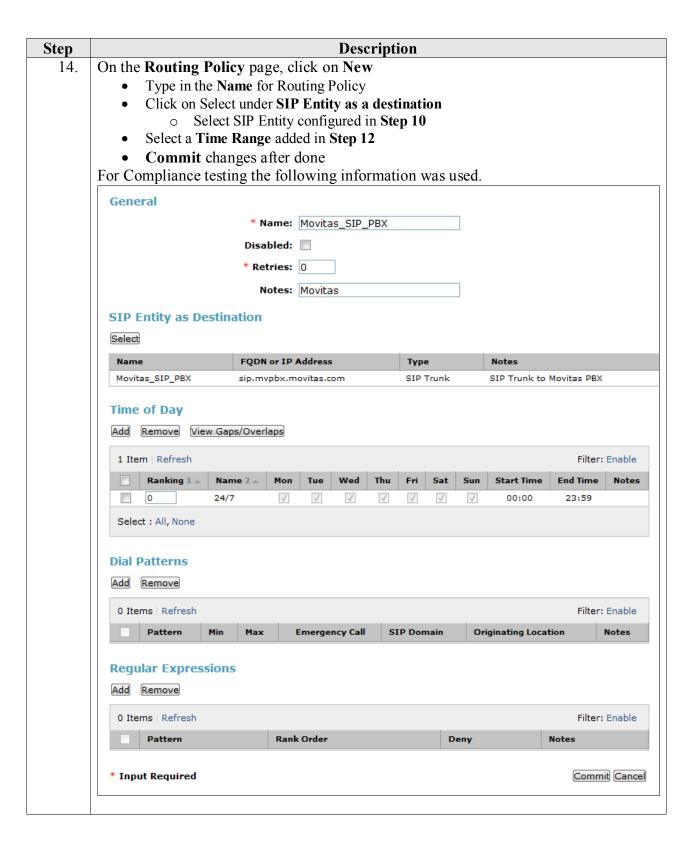


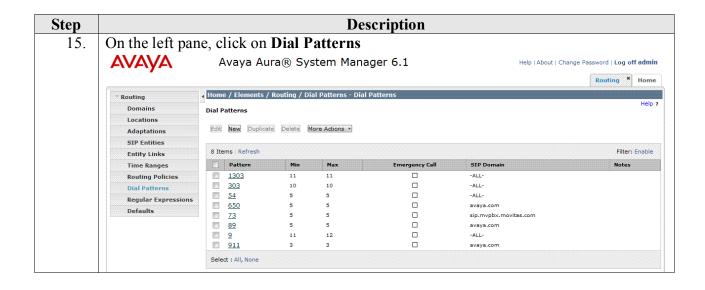










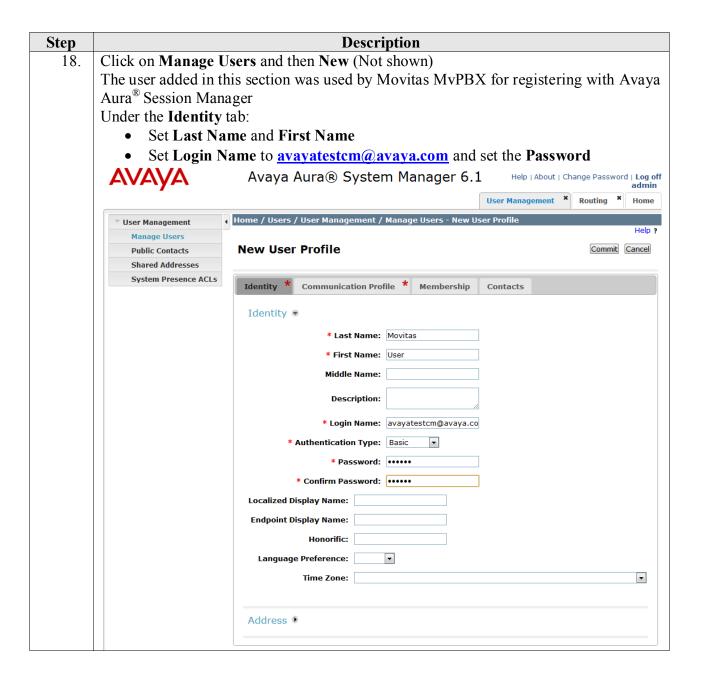


#### Step **Description** On Dial Patterns page, click on New 16. For compliance testing the following pattern was used. All calls dialed with 5 digits and starts with 7 will route to Movitas MvPBX. Set Pattern to 7 Set Min and Max to 5 Set SIP Domain to the domain configured in Step 4, sip.mvpbx.movias.com. Click on Add under Originating Locations and Routing Policies, screenshot not shown for this step. Select location configured in **Step 6** Select Routing Policy configured in Step 14 Click on **Select** once done Commit changes after done Dial Pattern Details Commit Cancel General \* Pattern: 7 \* Min: 5 \* Max: 5 Emergency Call: Emergency Priority: 1 **Emergency Type:** SIP Domain: sip.mvpbx.movitas.com Notes: to Movitas\_SIP\_PBX **Originating Locations and Routing Policies** Add Remove 1 Item | Refresh Filter: Enable Routing Policy Notes Originating Location Routing Policy Originating Location Name 1 Destination Public Movitas SIP PBX Movitas SIP PBX Movitas Select : All, None **Denied Originating Locations** Add Remove 0 Items | Refresh Filter: Enable **Originating Location** Notes

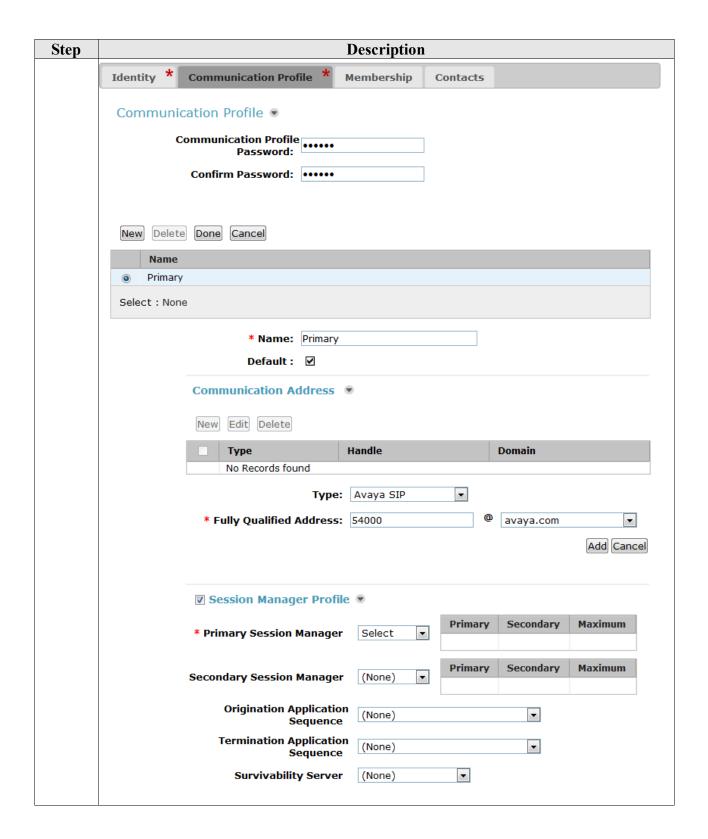
\* Input Required

Commit Cancel

**Description** Step Go back to the **Home** screen by click **Home** tab on the right top corner, and click on 17. **User Management AVAYĂ** Avaya Aura® System Manager 6.1 Help | About | Change Password | Log off Users Elements Services Administrators **Backup and Restore Application Management** Manage Administrative Users Backup and restore System Manage applications and application certificates Manager database Groups & Roles Configurations Manage groups, roles and **Communication Manager** assign roles to users Manage Communication Manage system wide configurations Synchronize and Import Manager objects Synchronize users with the Conferencing enterprise directory, import users from file Conferencing Manage alarms, view and harvest logs Inventory **User Management** Manage, discover, and Licenses Manage users, shared user navigate to elements, update element software View and configure licenses resources and provision Replication users Messaging Track data replication nodes, repair replication Manage Messaging System objects nodes Scheduler **Presence** Presence Schedule, track, cancel, update and delete jobs Routing Security Network Routing Policy Manage Security Certificates Session Manager **Templates** Session Manager Element Manager Manage Templates for Communication Manager and Messaging System objects **SIP AS 8.1** SIP AS 8.1



Step	Description
19.	Under the Communication Profile tab:
	<ul> <li>Set the Communication Profile Password and Confirm Password</li> </ul>
	<ul> <li>Under Confirm Password, click on New</li> </ul>
	<ul> <li>Set the <b>Handle</b> and <b>Domain</b> to an extension that will be used by</li> </ul>
	MvPBX to register
	<ul> <li>For Compliance testing, information shown in the following screen shot was used</li> </ul>
	<ul> <li>Check the Session Manager Profile box and set the Primary Session</li> </ul>
	Manager to configured Session Manager.
	• Check the CM Endpoint Profile box (not shown)
	<ul> <li>Set System to configured Avaya Aura<sup>®</sup> Communication Manager</li> </ul>
	<ul> <li>Note: Avaya Aura<sup>®</sup> Communication Manager was pre-configured</li> </ul>
	<ul> <li>Set Profile Type to Endpoint</li> </ul>
	<ul> <li>Set Extension same as the Handle used in this step</li> </ul>
	<ul> <li>Set Security Code to a desired value. This information will be used</li> </ul>
	when registering MvPBX
	o Set Port to IP
	Commit the changes.
	Screenshot on next page



# 6. Configure Avaya Aura® Communication Manager

This section describes the Avaya Aura<sup>®</sup> Communication Manager configuration to support connectivity to Avaya Aura<sup>®</sup> Session Manager and related functionality.

The configuration of Avaya Aura<sup>®</sup> Communication Manager was performed using the System Access Terminal (SAT). After the completion of the configuration, perform a **save translation** command to make the changes permanent. Please refer that some administration was already performed prior to this test effort. Please refer to *Administering Avaya Aura*® *Communication Manager* for more details, if needed.

# 6.1. Trunk Configuration – for SIP Trunks to Avaya Aura<sup>®</sup> Session Manager

This section summarizes the configuration of the SIP trunk that connects the Avaya Aura® Communication Manager to Avaya Aura® Session Manager

1	Description						
1.	System Parameters – Customer Options						
	Use the display system-parameters customer-options command to verify that t						
	options highlighted below are enabled or						
	options inglingliced below are enabled of	11 450 1.					
	display system-parameters customer-op	tions Page 4 of	11				
	OF110	NAL FEATURES					
	Emergency Access to Attendant? y	IP Stations?	У				
	Enable 'dadmin' Login? y						
	Enhanced Conferencing? y	ISDN Feature Plus?					
	Enhanced EC500? y ISDN/SIP Network Call Redirection?						
	Enterprise Survivable Server? n	ISDN-BRI Trunks?	-				
	Enterprise Wide Licensing? n ISDN-PRI?						
	ESS Administration? n	Local Survivable Processor?					
	Extended Cvg/Fwd Admin? y External Device Alarm Admin? n	Malicious Call Trace?					
	External Device Alarm Admin: n Five Port Networks Max Per MCC? n	Media Encryption Over IP? Mode Code for Centralized Voice Mail?					
	Flexible Billing? n	Mode code for centralized voice mail:	11				
	Forced Entry of Account Codes? n	Multifrequency Signaling?	7.7				
	Global Call Classification? n	Multimedia Call Handling (Basic)?	-				
	Hospitality (Basic)? y	Multimedia Call Handling (Enhanced)?	-				
	Hospitality (G3V3 Enhancements)? n	Multimedia IP SIP Trunking?	-				
	IP Trunks? y		_				
	IP Attendant Consoles?	n					

#### **Description** Step 2. **Node Names** Use the **change node-names ip** command to create node names for Avaya Aura® Session Manager The example below shows the node names and IP addresses used for the compliance test. These node names will be used in the administration of other forms in Avaya Aura® Communication Manager. change node-names ip Page 1 of 2. TP NODE NAMES Name IP Address 0.0.0.0 default 205.168.62.28 procr procr6 205.168.62.18 sm

#### 3. **IP network region**

The Avaya Aura® Communication Manager, Avaya Aura® Session Manager and VoIP (H.323/SIP) endpoints were located in a single IP network region (IP network region 1) using the parameters described below. Use the **change ip-network-region 1** command to view these settings. By default, all elements will also be in IP network region 1 unless specifically placed in a separate region using the **ip-network-map** command. The example below shows the values used for the compliance test.

- A descriptive name was entered for the Name field.
- The Codec Set field was set to the IP codec set to be used for calls within this IP network region. In this case, IP codec set 1 was selected. This is the codec set that will be used for calls between the Movitas and Avaya Aura<sup>®</sup> Communication Manager, via Avaya Aura<sup>®</sup> Session Manager since all components are in IP network region 1.
- The default values were used for all other fields.

```
change ip-network-region 1
                                                              Page 1 of 20
                              TP NETWORK REGION
 Region: 1
               Authoritative Domain: avaya.com
Location: 1
   Name: Public Domain
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? n
  UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
       Audio PHB Value: 46
       Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
      Audio 802.1p Priority: 6
       Video 802.1p Priority: 5
                                    AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS
                                                       RSVP Enabled? n
 H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
  Keep-Alive Interval (sec): 5
           Keep-Alive Count: 5
```

Step			D	escription			
4.	Codecs Use the <b>change ip-codec-set 1</b> command to define the codecs used by IP codec set 1. Please note that G.711MU and G.729A are the only supported codecs for this configuration.						
	change ip-code		Codec Set		Page	1 of	2
	Codec Set:		codec set				
	Audio Codec 1: G.711MU	Silence Suppression n	Frames Per Pkt 2 2	Packet Size(ms) 20 20			

### **Description** Step 5. **Signaling Group** Use the add signaling-group n command, where n is an unused signaling group, to create a new signaling group for each SIP trunk to Avaya Aura® Session Manager. For compliance test, signaling group 4 was created for the trunk to Avava Aura® Session Manager. Signaling group 4 was configured using the parameters highlighted below. Default values were used for all other fields. Set the Group Type to sip. Set the Near-end Node Name to procr. This node name maps to the IP address of the Avaya Aura® Communication Manager. Node names are defined using the change node-names ip command (Step 2). Set the Far-end Node Name to sm. This node name maps to the IP address of the Avaya Aura<sup>®</sup> Session Manager as defined using the **change node-names ip** command (Step 2). Set the Near-end Listen Port and Far-end Listen Port to 5061. Set the **Far-end Network Region** to 1. This is the IP network region which contains the Avaya Aura® Session Manager. Set DTMF over IP to in-band The default values were used for all other fields.

```
add signaling-group 4
                                   SIGNALING GROUP
 Group Number: 2
IMS Enabled? n
                            Group Type: sip
                        Transport Method: tls
       O-SIP? n
                                                                  SIP Enabled LSP? n
     IP Video? n
                                                        Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y Peer Server: SM
   Near-end Node Name: procr
                                                 Far-end Node Name: sm
                                          Far-end Listen For Star-end Network Region: 1
 Near-end Listen Port: 5061
                                               Far-end Listen Port: 5061
Far-end Domain: avaya.com
                                                 Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate

DTMF over IP: in-band

Session Establishment Timer(min): 3
                                                  RFC 3389 Comfort Noise? n
                                                 Direct IP-IP Audio Connections? y
                                                            IP Audio Hairpinning? n
                                                      Initial IP-IP Direct Media? n
Alternate Route Timer(sec): 6
         Enable Layer 3 Test? y
H.323 Station Outgoing Direct Media? n
```

#### **Description** Step 6. Trunk Group Use the add trunk-group n command, where n is an unused trunk group, to create a new trunk group for each SIP trunk to Avaya Aura® Session Manager. For the compliance test, trunk group 4 was created for the trunk to Avaya Aura® Session Manager. Trunk group 4 was configured using the parameters highlighted below. On Page 1: Set the **Group Type** to *sip*. Enter a descriptive name for the **Group Name**. Enter an available trunk access code (TAC) that is consistent with the existing dial plan in the **TAC** field. Set the **Service Type** to *tie*. Set the Member Assignment Method to auto. Set the **Signaling Group** to the signaling group shown in the previous step. Set the **Number of Members** field to the number of channels available in this trunk. For the compliance test, the number of members was chosen to be 10. Set Outgoing Display to v The default values were used for all other fields. TRUNK GROUP roup Number: 4 Group Type: sip CDR Reports: y Group Name: Movitas COR: 1 TN: 1 TAC: \*004 Direction: two-way Outgoing Display? y Dial Access? n Group Number: 4 Dial Access? n Night Service: Queue Length: 0 Service Type: tie Auth Code? n Member Assignment Method: auto Signaling Group: 4 Number of Members: 10

Step	Description				
7.	Trunk Group – continued On Page 3: ■ It is required that the Send Name field is set to y and the Send Calling Numbe field is set to y.				
	• The default values were used for all other fields.				
	add trunk-group 31 Page 3 of 21 TRUNK FEATURES				
	ACA Assignment? n Measured: none  Internal Alert? n Maintenance Tests? y  Data Restriction? n NCA-TSC Trunk Member:  Send Name: y Send Calling Number: y				
	Used for DCS? n Send EMU Visitor CPN? n Suppress # Outpulsing? n Format: public  UUI IE Treatment: service-provider				
	Replace Restricted Numbers? n Replace Unavailable Numbers? n Send Connected Number: n Hold/Unhold Notifications? n				
	Send UUI IE? y Modify Tandem Calling Number? n Send UCID? n Send Codeset 6/7 LAI IE? y				
8.	Automatic Route Selection (ARS)  For the compliance test, ARS was used to route calls to EC500 extension that was 5 digits long and leading digit was 7. Use the <b>change ars analysis</b> command to create an entry in the ARS table.				
	• Set Dialed String to 7				
	<ul> <li>Set Total Min and Max to 5</li> <li>Set Route Pattern to 4</li> </ul>				
	<ul> <li>Set Route Pattern to 4</li> <li>Set Call Type to svcl</li> </ul>				
	change ars analysis 7  ARS DIGIT ANALYSIS TABLE  Location: all Percent Full: 2				
	Dialed Total Route Call Node ANI String Min Max Pattern Type Num Reqd 7 5 5 4 svcl n				

#### **Description** Step 9. **Route Patterns** Use the **change route pattern** n command, where n is an unused route pattern. The example below shows route pattern 4 used in the compliance test. Route pattern 1 was accessed when ARS matches on a dialed string of 7xxxx. For the first entry, set the **Grp No.** field to the trunk group of Avaya Aura® Session Manager (trunk group 4). Set the Facility Restriction Level (FRL) of the trunk to an appropriate level to allow authorized users to access the trunk. The level of $\theta$ is the least restrictive. change route-pattern 4 Page 1 of Pattern Number: 4 Pattern Name: SCCAN? n Secure SIP? n Grp FRL NPA Pfx Hop Toll No. Inserted No Mrk Lmt List Del Digits DCS/ IXC OSIG Dgts Intw n user 2: n user 3: n user 4: n user 5: user 6: n user BCC VALUE TSC CA-TSC ITC BCIE Service/Feature PARM No. Numbering LAR 0 1 2 M 4 W Request Dgts Format Subaddress rest 1: yyyyyn n none 2: y y y y y n n none 3: y y y y y n n rest none 4: yyyyyn n rest none 5: y y y y y n n 6: y y y y y n n rest none rest none

# 6.2. Trunk Configuration – for ISDN/PRI to PSTN

ISDN calls to PSTN were also tested. For PSTN interconnections.

ep	De	scription			
1.	1 0 0 1	<b>tomer-options</b> command to verify that the if this option is disabled please check with			
	display system-parameters customer-options Page 4 of 1 OPTIONAL FEATURES				
	Emergency Access to Attendant? y Enable 'dadmin' Login? y Enhanced Conferencing? y Enhanced EC500? y Enterprise Survivable Server? n Enterprise Wide Licensing? n ESS Administration? n Extended Cvg/Fwd Admin? y External Device Alarm Admin? n Five Port Networks Max Per MCC? n Flexible Billing? n Forced Entry of Account Codes? n Global Call Classification? n Hospitality (Basic)? y Hospitality (G3V3 Enhancements)? n IP Trunks? y  IP Attendant Consoles?	IP Stations? y  ISDN Feature Plus? y ISDN/SIP Network Call Redirection? n ISDN-BRI Trunks? y ISDN-PRI? y Local Survivable Processor? n Malicious Call Trace? y Media Encryption Over IP? y Mode Code for Centralized Voice Mail? n  Multifrequency Signaling? y Multimedia Call Handling (Basic)? y Multimedia Call Handling (Enhanced)? y Multimedia IP SIP Trunking? y			

#### **Description** Step

#### Add DS1 2.

Use the add ds1 Board-location command to add a DS1. In this case, board V2 was used. The gateway used for this testing, was connected to another Avaya Media Gateway which had access to PSTN. This configuration pertains to the Media Gateway G430 as show in the **Test Configuration** diagram.

- Set Signaling Mode to isdn-pri
- Set Name to a desired name
- Set Line Coding to b8zs
- Set **Protocol Version** to **b**

```
add ds1 01V2
                              DS1 CIRCUIT PACK
           Location: 001V2
                                                   Name: PSTN
          Bit Rate: 1.544
                                           Line Coding: b8zs
  Line Compensation: 1
                                          Framing Mode: esf
    Signaling Mode: isdn-pri
           Connect: network
  TN-C7 Long Timers? n
                                      Country Protocol: 1
Interworking Message: PROGress
                                      Protocol Version: b
Interface Companding: mulaw
                                                   CRC? n
          Idle Code: 11111111
                           DCP/Analog Bearer Capability: 3.1kHz
                                        T303 Timer(sec): 4
     Slip Detection? n
                                     Near-end CSU Type: other
  Echo Cancellation? n
                               Block Progress Indicator? n
```

#### 3. **Signaling Group**

Use the add signaling-group n command, where n is an unused signaling group, to create a new signaling group for each ISDN to PSTN Gateway. For the compliance test, signaling group 3 was created for the trunk to the PSTN Gateway.

- Set the Group Type to isdn-pri.
- Set the **Trunk Group for Channel Selection** field to the trunk group created in the next step. This cannot be done until the trunk group is created. Thus, initially this field is left blank and later changed to the correct value after the trunk group is created. A separate trunk group will be created for each signaling-group. Set **Primary D-Channel** as the 24<sup>th</sup> channel for ds1 added in **Step 2**.
- The default values were used for all other fields.

```
add signaling-group 3
                                                                    Page 1 of 5
                                   SIGNALING GROUP
Group Number: 3
                               Group Type: isdn-pri
                        ssociated Signaling? y

Primary D-Channel: 001V224

Max number of NCA TSC: 0

Max number of CA TSC: 0

Trunk Group for NCA TSC:
                   Associated Signaling? y
       Trunk Group for Channel Selection:
                                                    X-Mobility/Wireless Type: NONE
      TSC Supplementary Service Protocol: a Network Call Transfer? n
```

#### **Description** Step 4. **Trunk Group** Use the add trunk-group n command, where n is an unused trunk group, to create a new trunk group for each ISDN/PRI to PSTN gateway. For the compliance test, trunk group 3 was created for the trunk to the Media Gateway as shown in the Test Configuration diagram. On Page 1: Set the **Group Type** to *isdn*. Enter a descriptive name for the **Group Name**. Enter an available trunk access code (TAC) that is consistent with the existing dial plan in the **TAC** field. Set the Carrier Medium to PRI/BRI. Set the **Service Type** to *public-ntwrk*. The default values were used for all other fields. add trunk-group 3 Page 1 of 21 TRUNK GROUP Group Number: 3 Group Type: isdn CDR Reports: r Group Name: PSTN COR: 1 TN: 1 TAC: \*003 Direction: two-way Outgoing Display? y Carrier Medium: PRI/BRI Dial Access? y Busy Threshold: 255 Night Service: Queue Length: 0 Service Type: public-ntwrk Auth Code? n TestCall ITC: rest Far End Test Line No: TestCall BCC: 4

On Page 3:  Set Send Name to y  Set Send Calling Number to y  Set Format to Public  Set Connected Number to y
add change trunk-group 3 Page 3 of 21
ACA Assignment? n Measured: none Wideband Support? n Maintenance Tests? y
Data Restriction? n NCA-TSC Trunk Member:  Send Name: y Send Calling Number: y
Used for DCS? n Suppress # Outpulsing? n Format: public Outgoing Channel ID Encoding: preferred UUI IE Treatment: service-provider
Replace Restricted Numbers? n Replace Unavailable Numbers? n Send Connected Number: y
Network Call Redirection: none Hold/Unhold Notifications? n  Send UUI IE? y Modify Tandem Calling Number: no  Send UCID? n
Send Codeset 6/7 LAI IE? y  Ds1 Echo Cancellation? n
Apply Local Ringback? n US NI Delayed Calling Name Update? n Show ANSWERED BY on Display? y Network (Japan) Needs Connect Before Disconnect? n

**Description** 

Step

On Page 5, type in define all 23 ports for the ds1 added in Step 3. Ports are defined in the format such that **Port** value is set as board name followed by channel number. In our case the board was 001V2 and channel numbers 01-04 were used, as seen the box below.

```
change trunk-group 3
                                                               Page
                                                                      5 of 21
                                TRUNK GROUP
                                     Administered Members (min/max):
                                                                       1/4
GROUP MEMBER ASSIGNMENTS
                                         Total Administered Members:
                                                                       4
              Code Sfx Name
                                   Night
      Port
                                                   Sig Grp
  1: 001V201 MM710
  2: 001V202 MM710
                                                     3
  3: 001V203 MM710
                                                     3
  4: 001V204 MM710
                                                     3
```

# Step Description

#### 7. Automatic Route Selection (ARS)

For the compliance test, an entry was added to route call to PSTN by dialing an 11 digit Telephone Number (TN) and 10 digit TN. The entry is highlighted below which is used to route calls to PSTN by dialing 1303xxxxxxx. This dialed string is mapped to route pattern 3 which routes calls to trunk 3 connected to the PSTN.

```
change ars analysis 13

ARS DIGIT ANALYSIS TABLE
Location: all Percent Full: 2

Dialed Total Route Call Node ANI
String Min Max Pattern Type Num Reqd
1303 11 11 3 hnpa n
```

#### 8. **Route Pattern – PSTN Trunk**

This route pattern is used in cases where the SIP Users or Avaya Extensions need to be used to calls the PSTN number. Avaya Aura® Communication Manager will then route the call out the PSTN trunk.

- Set Grp No to trunk group in Step 4, 3
- Set FRL to 0

```
change route-pattern 4
                                                     Page
             Pattern Number: 4 Pattern Name:
                     SCCAN? n Secure SIP? n
  Grp FRL NPA Pfx Hop Toll No. Inserted
No Mrk Lmt List Del Digits
                                                          DCS/ IXC
                                                          QSIG
                                                          Intw
                       Dats
1: 3
     0
                                                           n
                                                              user
2:
                                                           n
                                                              user
3:
                                                           n
                                                              user
4:
                                                           n user
5:
                                                           n user
6:
                                                              user
   0 1 2 M 4 W Request
                                                Dgts Format
                                               Subaddress
1: y y y y y n n
                       rest
                                                             none
                       rest
2: y y y y y n n
                                                              none
3: y y y y y n n
                        rest
                                                              none
4: y y y y y n n
                       rest.
                                                              none
5: yyyyyn n
                       rest
                                                              none
6: y y y y y n n
                        rest
                                                              none
```

# 6.3. EC500 Configuration

Step	Description				
1.	Use <b>change off-pbx-telephone station-mapping</b> <i>n</i> to map an EC500 extension where				
	<i>n</i> is the Avaya Extension.				
	<ul> <li>Set Application to EC500</li> </ul>				
	<ul> <li>Set Phone Number to the SIP User extension on MvPBX. Note that this phone number must comply ars analysis for calls starting with digit 7 in Section 6.2.</li> <li>Set Trunk Selection to ars</li> </ul>				
	change off-pbx-telephone station-mapping 54006 Page 1 of 3 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION				
	Station Application Dial CC Phone Number Trunk Config Dual				
	Extension Prefix Selection Set Mode				
	54006 EC500 - 73306 ars 1				
	_				
	-				
	-				

# 7. Configure Movitas

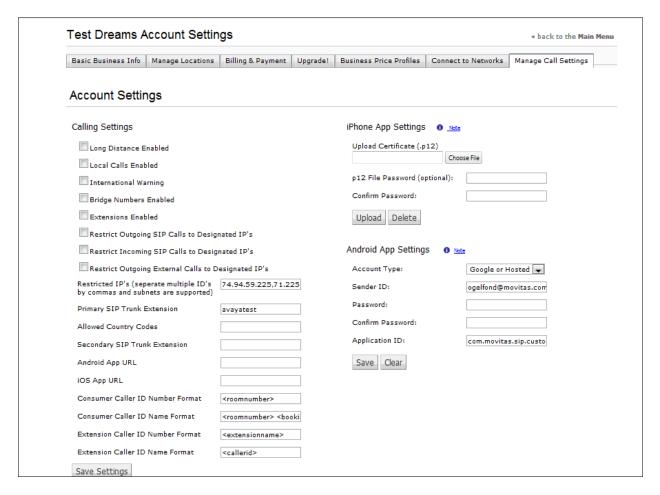
Configuration for Movitas MvPBX was performed by a Movitas Engineer. This document assumes that the Movitas MvPBX system was properly installed and configured by a Movitas Engineer. This section provides the steps of how to configure Movitas MvPBX working with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

# 7.1. Configure Business Account

As a partner on Movitas.com, a business account is established by signing up, choosing a name for the account and providing any necessary billing information. Once this is established the content, users, designs and business extensions can be managed via the Movitas interface.

### 7.2. Enable the Business Account for Calling

After a business account is established, a Movitas admin can set up the business to enable calling functionality on the Account Settings Price Profile page. This can be done by logging into the Movitas system, clicking on Account Settings in the top right, and then clicking on the Price Profile tab. Note: This is only available to Movitas Admins and not all admins. Once calling is enabled, this will result in an additional account setting tab where the SIP trunk can be set, IP addresses can be setup to restrict calls to certain sites, push notifications can be configured and more. For Avaya Aura® Communication Manager testing, the only required field is the SIP Trunk, which will be assigned in step 7.3 below.



# 7.3. Configure SIP Trunk in MvPBX

Once a business is setup, Movitas can assign a SIP Trunk username and password for the Avaya system. The required settings for this are:

- Name: a short name to use as an identifier for the trunk
- Fullname: a human-friendly used to describe the trunk
- Secret: the password used for authentication from the CS1K

- Host: the IP address of the Avaya Aura<sup>®</sup> Communication Manager that we will be connecting to as a user
- Transport: should be set to UDP
- Directmedia: should be set to nonat
- Register: registration entry for MvPBX (ex. 54000@avaya.com:movitas:54000@205.168.62.18/54000~3600)
- Default county code: the numerical country code for outgoing international call parsing
- PSTN prefix: prefix for outbound calls
- From Domain: from domain where calls will come from (ex. avaya.com)

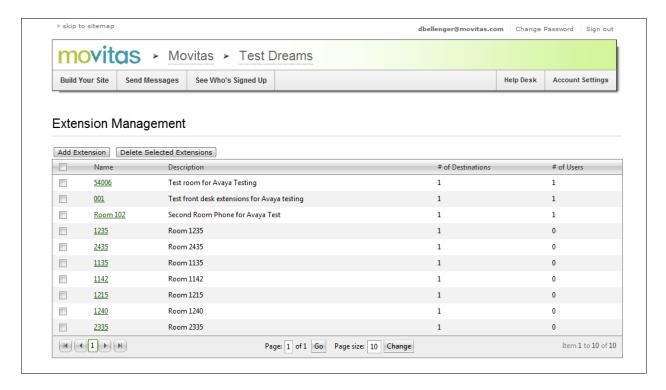
Once established the credentials are provided to complete the setup in Avaya Aura<sup>®</sup> Session Manager and Avaya Aura<sup>®</sup> Communication Manager.

# 7.4. Assign SIP Trunk to Business Account

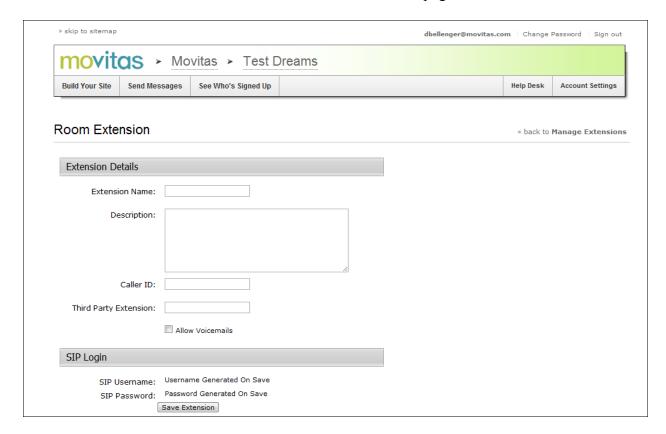
Following the trunk creation, the account administrator can return to the business account for the application and enter the SIP Trunk name into the Primary SIP Trunk field.

### 7.5. Setup Business Extensions

Once the trunk is established, administrators can navigate to the room extensions page where they can create and manage extensions.



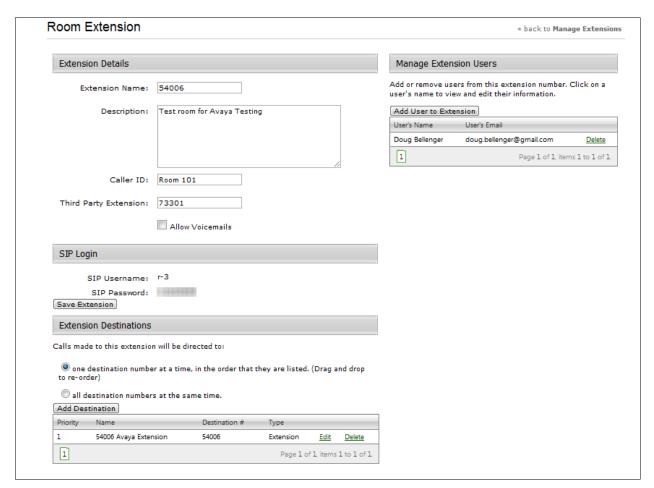
#### Then click "Add Extension" to be taken to the new extension page shown below:



#### The information that should be added is

- Extension Name a familiar name that can be used to identify the extension (ex. Room 101)
- Description a description of the name if required
- Caller ID if a different caller ID is required for the extension it can be entered here. (Note: caller ID and caller Number for consumer extensions and business extensions can also be formatted on the call settings page using variables such as room number, first name, last name, booking last name, caller ID, extension name, etc).
- Third Party Extension for Avaya implementations, the PCA number should be added here in order to map calls to these extensions. This number must be unique for each business extension

Once all the settings are complete, click **Save Extnesion** and the extension will now be enabled and call destinations can be configured such as users, DID's, SIP extensions, or Avaya Aura<sup>®</sup> Communication Manager extensions. An example of the updated page is below.



# 7.6. Setup Call Destinations for the Extension

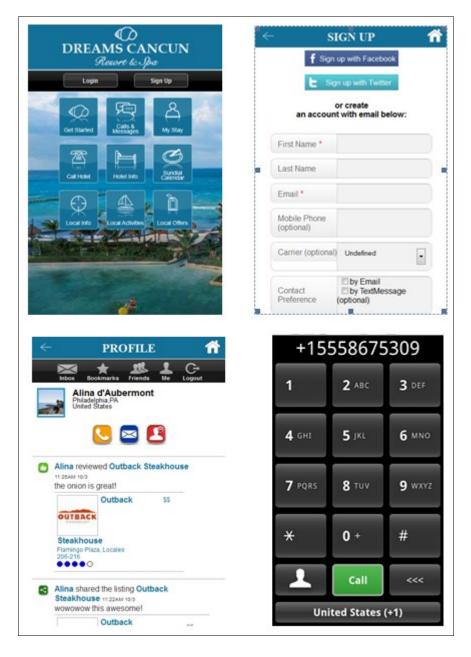
In order to forward calls from a user to the room, in addition to setting up the Third Party Extension with the appropriate redirect number, the Direct Number (DN) also should be added as an extension as shown above. (Warning: setting up the redirect number as a call destination is not recommended due to the creation of a loop between the two systems). In addition to hotel extensions, a call destination can be pointed at a SIP address, a DID, or a user in the system.

# 7.7. Deploy Applications Via App Stores

In order to complete the setup of the applications, Movitas will package the content and calling functionality into the required apps and enter them into the appropriate app store. Once completed the application will be available for download and the content and extensions can be updated via the Movitas portal. Any changes to the SIP Trunk for the business, IP addresses or other settings can be updated as well.

# 7.8. User Creates Account on Application

In order to place calls via the application following the setup, a user will signup for an account, log in to the application and then they will be registered with MvPBX automatically. Once they are logged in, they can navigate to the dialpad where they can dial an extension direct or dial another user account.

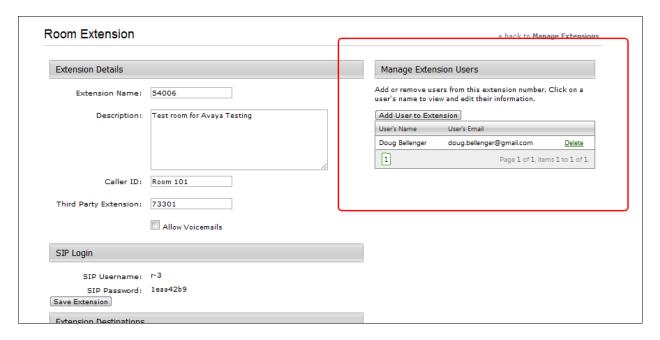


Note: The above shows an example hotel website with the following sections:

- Top Left: Home screen with icons related to hotel services, local information, and guest communication tools
- Top Right: An example of a sign up page for the hotel application
- Bottom Right: An example profile for a user from the application
- Bottom Left: An example dial pad from the Android application for the hotel

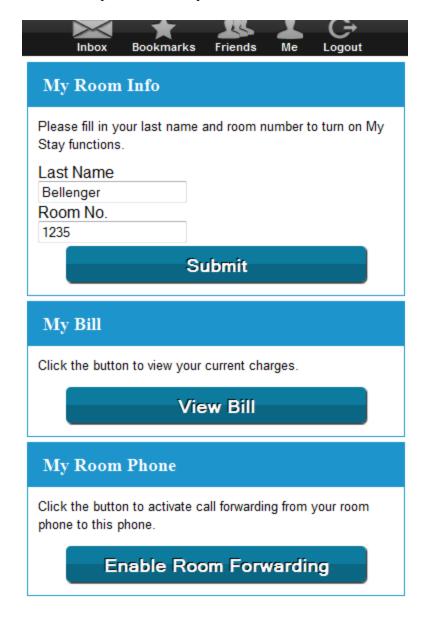
# 7.9. Assign User Such as Front Desk to Extension

Once a user account is setup, administrators can assign users to extensions by accessing the business extension from the extension management page in section 7.5 and the clicking on the extension to manage. When on the extension management page, an administrator can assign a user to an extension by adding their email address under the Manage Extension Users section on the right side of the page. A user can also be deleted from this page by clicking the "delete" link by their name once they have been added.



### 7.10. Assigning User to Room Phone From Application

Pending the availability of web services into a Property Management System (PMS) for a specific client, for example a hotel, a page may be setup to enable forwarding of the room phone as outlined below. This will associate the user with the appropriate extension that calls can be forwarded or received from. This can also be handled by web services and will need to be established custom for each implementation by Movitas.



Note: This is an example of a page setup for hotels, as mentioned in 7.1.10, that allows their guests to enable room forwarding from the CM to the application for the length of their stay. At the end of their stay calls will no longer be forwarded to them based on their checkout date.

# 8. Verification Steps

The following are typical steps to verify the interoperability between the Movitas system and Avaya Aura® Communication Manager and Avaya Aura® Session Manager

- Place a call to Front Desk phone which is one of three Avaya extensions associated with Movitas SIP users
- The Front Desk phone in the Avaya Aura® Communication Manager rings and the Movitas SIP user associated with the Front Desk phone also rings.
- Accept the call on the Movitas SIP user, the Front Desk phone in Avaya Aura® Communication Manager stops ringing and become idle.
- Check two-way audio path between the caller and Movitas SIP user

#### 9. Conclusion

All of the executed test cases have passed and met the objectives as outlined in Section 2. The Movitas SIP MvPBX system and its respective iPhone, Android and Blackberry applications are considered compliant with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

#### 10. Additional References

Product documentation for Avaya products may be found at <a href="http://support.avaya.com">http://support.avaya.com</a>. Avaya

- [1] *Administering Avaya Aura*® *Communication Manager*, Doc # 03-603558, Release 6.0.1, Issue 1.3, December 2010.
- [2] Administering Avaya Aura® Session Manager, Doc # 03-603324, Release 6.1 November 2010

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