



Avaya Solution & Interoperability Test Lab

Application Notes for Acqueon iAssist Call Back Manager with Avaya Aura® Experience Portal - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Acqueon iAssist Call Back Manager with Avaya Aura® Experience Portal. The iAssist Call Back Manager offers callers queued to a call center the option to continue to wait in queue for an agent or request a call back when either an agent becomes available or schedule a call back for a specified date and time.

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 to integrate the Acqueon iAssist Call Back Manager with Avaya Aura® Experience Portal. The iAssist Call Back Manager offers callers queued to a call center the option to continue to wait in queue for an agent or request a call back when either an agent becomes available or schedule a call back for a specified date and time.

The iAssist Call Back Manager (CBM) consists of two modules: the Inbound Module and the Outbound Module. The Inbound Module is designed to take a call back request from a caller waiting to be serviced by an agent. The Outbound Manager retrieves the call back request based on priority and time of the callback and then dials the agent queue. If the agent is available, the call details are voiced to the agent and then an outbound call to the telephone number specified by the caller is made. The incoming call flow is described below.

- Customer calls the contact center and gets routed to an agent queue.
- If the wait time in queue is more than the threshold set (Expected Wait Time), calls are routed to the inbound CBM application on Avaya Aura® Experience Portal.
- Once the call is answered by the CBM inbound channel on Avaya Aura® Experience Portal, CBM offers various options to leave a call back request. The following are the call back options:
 - Call back as soon as an agent is available
 - Call back on same day at a later time
 - Call back on a future day and time
- CBM then prompts the customer to enter the call back contact number, account information, and appropriate date/time of call back. A request is then registered into the CBM database.

The CBM outbound module running on the iAssist Admin server continuously polls the database on a regular interval to retrieve pending callback requests. The outbound module then calls the appropriate agent group number to get an agent to process the callback. Once the agent answers the call, CBM plays the customer's information to the agent. CBM then dials the customer's number and conferences the call with the agent. If the customer call cannot be completed, CBM reschedules the call based on a pre-defined schedule interval. CBM reschedules the call for a specified number of times. Once the maximum attempts have been made unsuccessfully, the call is marked as failed.

Another Acqueon related solution is described in Application Notes for Acqueon iAssist Call Survey Manager with Avaya Aura® Experience Portal

2. General Test Approach and Test Results

This section describes the interoperability compliance testing used to verify the iAssist CBM applications with Avaya Aura® Experience Portal.

The interoperability compliance test included feature and serviceability testing. The feature testing focused on routing calls to Experience Portal and running the iAssist CBM applications to allow the caller the option to request a call back. All of the call back request options available in the Inbound CBM application were tested. In addition, the Outbound CBM application was also verified. The iAssist Outbound CBM Module initiated the call back to the agent and caller and established a two-way talk path. Conditions where the call back could not be established were also verified. In these cases, the call was either rescheduled or marked as failed, if the number of retries were exceeded. Finally, the registered call back requests and call back status were verified in iAssist reports.

The serviceability testing focused on verifying the ability of iAssist Admin server and Avaya Aura® Experience Portal to recover from adverse conditions, such as power failures and disconnecting cables to the IP network.

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.

2.1. Interoperability Compliance Testing

Interoperability compliance testing included feature and serviceability testing. The feature testing focused on the following functionality:

- Routing incoming calls to Avaya Aura® Experience Portal when the expected wait time for an agent exceeds a configured threshold.
- Experience Portal successfully running the iAssist Inbound CBM application and all of the call back options tested.
- The ability of the caller to continue waiting in queue for an agent.
- The ability of the caller to make a call back request. Call back options described above were tested.
- iAssist CBM servicing pending call back requests and running the iAssist Outbound CBM application.
- Failure conditions, such as the call back failing due to network problems, and verifying that the call back was rescheduled.
- The ability to reschedule a call back if the call to the agent or caller is not completed within a specified timeout value.
- iAssist reports showing the registered call back requests and the call back status.

The serviceability testing focused on verifying the ability of the iAssist Admin server and Experience Portal to recover from adverse conditions, such as power failures and disconnecting cables to the IP network.

2.2. Test Results

All test cases passed with the following observations below.

- There is no ring back tone on the agent's phone while the customer's phone ringing for the call back. This is design intent from iAssist Call Back Manager when their application uses Call Control XML from Experience Portal to create a conference call between agent and customer.

2.3. Support

For technical support on the iAssist Call Back Manager, contact Acqueon via phone, email, or internet.

- **Phone:** +9198403 57893 (or) +1 888 946 6878
- **Email:** support@acqueon.com
- **Web:** <http://acqueon.issuetrak.com>

Reference Configuration **Error! Reference source not found.** illustrates the configuration used for testing. In this configuration, Avaya Experience Portal interfaces with Avaya Aura® Communication Manager via SIP. The iAssist Admin Server server hosted the iAssist CBM applications supporting the CBM inbound and outbound modules. The Acqueon iAssist Admin server contained the Microsoft SQL database and also was used to configure the iAssist CBM application.

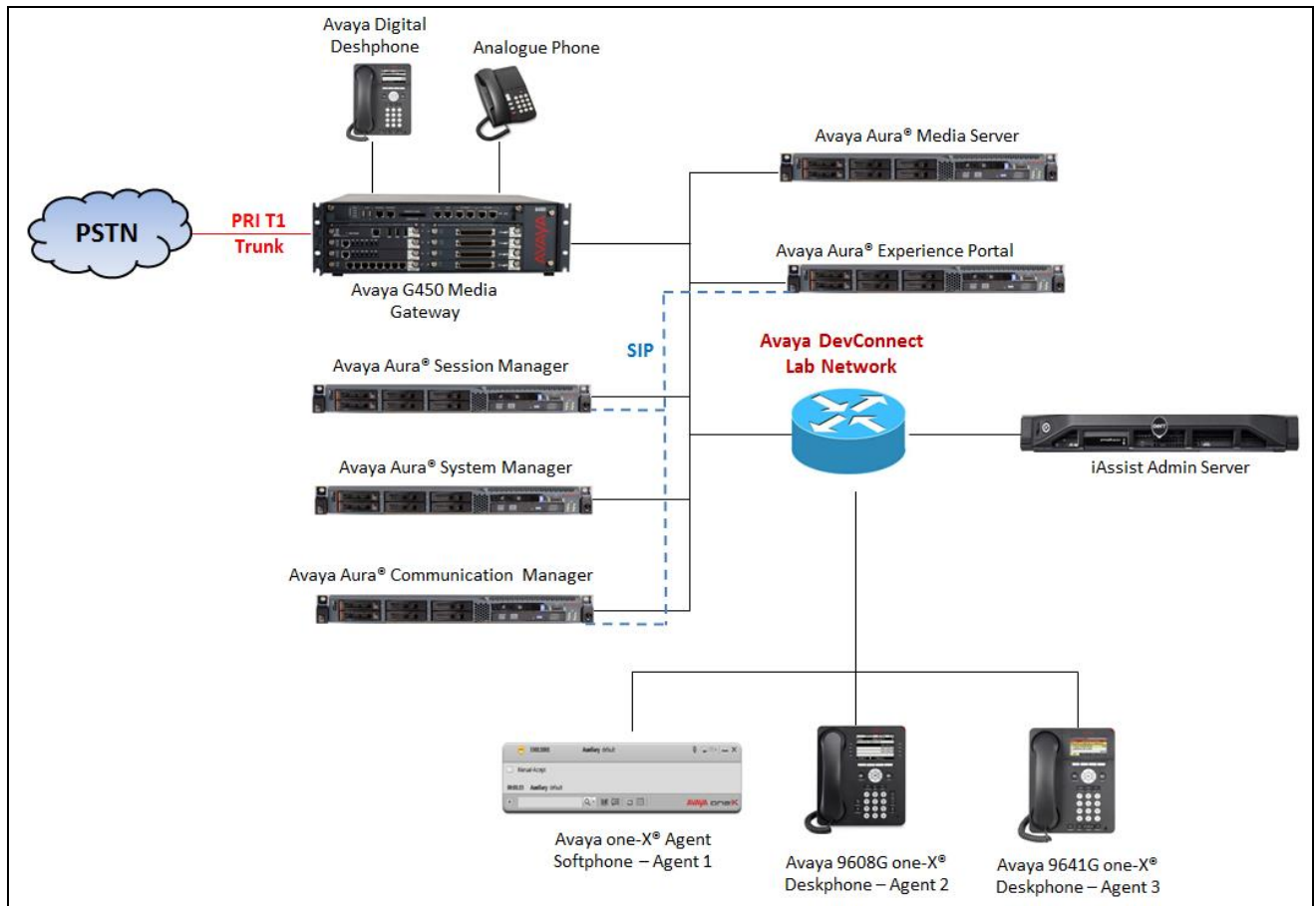


Figure 1: Test Configuration Diagram

3. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on Virtualized Environment	R017x.00.0.441.0 Patch 23523
Avaya Aura® System Manager running on Virtualized Environment	7.1.0.0.116662
Avaya Aura® Session Manager running on Virtualized Environment	7.1.0.0.710028
Avaya Aura® Media Server running on Virtualized Environment	7.8
Avaya Aura® Experience Portal running on Virtualized Environment	7.1.0.0.1107
Avaya G450 Media Gateway	38.19.0
Avaya 9641GS H323 IP Deskphone	6.6.4
Avaya 9621G SIP IP Deskphone	7.1.29
Acqueon iAssist Callback Manager application running on Windows Server 2012	2.2.1.16

4. Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Communication Manager via the System Access Terminal (SAT). The procedures include the following areas:

- Administer Hunt Groups for Agents.
- Administer Agent IDs for Agents.
- Administer Call Vectoring.
- Administer Signaling Group.
- Administer Trunk Group.
- Administer Route Pattern.
- Administer Dial Plan
- Administer AAR Table

4.1. Administer Hunt Groups

This section provides the Hunt Group configuration for the call center agents. Agents will log into Hunt Group 1 configured below. Provide a descriptive name and set the **Group Extension** field to a valid extension. Enable the **ACD**, **Queue**, and **Vector** options. This hunt group will be specified in the **Agent LoginIDs** configured in Section **Error! Reference source not found.**

add hunt-group 1	HUNT GROUP	Page 1 of 4
Group Number: 1	ACD? y	
Group Name: Skill-1	Queue? y	
Group Extension: 3320	Vector? y	
Group Type: ucd-mia		
TN: 1		
COR: 1	MM Early Answer? n	
Security Code:	Local Agent Preference? n	
ISDN/SIP Caller Display:		
Queue Limit: unlimited		
Calls Warning Threshold:	Port:	
Time Warning Threshold:	Port:	

On Page 2 of the Hunt Group form, enable the **Skill** option.

change hunt-group 1	HUNT GROUP	Page 2 of 4
Skill? y	Expected Call Handling Time (sec): 180	
AAS? n	Service Level Target (% in sec): 80 in 20	
Measured: both		
Supervisor Extension:		
Controlling Adjunct: none		
VuStats Objective:		
Multiple Call Handling: none		
Timed ACW Interval (sec):	After Xfer or Held Call Drops? n	

4.2. Administer Agent IDs

This section provides the Agent Login IDs for the agents. Add an **Agent Login ID** for each agent in the call center as shown below. In this configuration, agent login IDs 1000 to 1002 were created for three agents.

add agent-loginID 1000		Page 1 of 2
AGENT LOGINID		
Login ID: 1000	AAS? n	
Name: Agent 1000	AUDIX? n	
TN: 1		
COR: 1		
Coverage Path:	LWC Reception: spe	
Security Code: 1234	LWC Log External Calls? n	
Attribute:	AUDIX Name for Messaging:	
	LoginID for ISDN/SIP Display? n	
	Password:1234	
	Password (enter again):1234	
	Auto Answer: station	
AUX Agent Remains in LOA Queue: system	MIA Across Skills: system	
AUX Agent Considered Idle (MIA): system	ACW Agent Considered Idle: system	
Work Mode on Login: system	Aux Work Reason Code Type: system	
	Logout Reason Code Type: system	
	Maximum time agent in ACW before logout (sec): system	
	Forced Agent Logout Time: :	
WARNING: Agent must log in again before changes take effect		

On Page 2 of the **Agent LoginID** form, set the skill number (SN) to hunt group 1, which is the hunt group (skill) that the agents will log into.

change agent-loginID 1000		Page 2 of 2
AGENT LOGINID		
Direct Agent Skill:	Service Objective? n	
Call Handling Preference: skill-level	Local Call Preference? n	
SN RL SL	SN RL SL	
1: 1 1	16:	31: 46:
2:	17:	32: 47:
3:	18:	33: 48:
4:	19:	34: 49:
5:	20:	35: 50:
6:	21:	36: 51:
7:	22:	37: 52:
8:	23:	38: 53:
9:	24:	39: 54:
10:	25:	40: 55:
11:	26:	41: 56:
12:	27:	42: 57:
13:	28:	43: 58:
14:	29:	44: 59:
15:	30:	45: 60:

4.3. Administer Call Vectoring

This section describes the procedures for configuring call vectoring for the Agent and inbound call to iAssist CallBack Manager

Configure the **Vector Directory Number** (VDN) that will handle incoming customer calls. The VDN invokes a vector that will queue the call to an agent split and also route the call to the iAssist CBM application on Avaya Aura® Experience Portal if the call is queued and the expected wait time exceeds a configured threshold in the associated vector. In this example, VDN 3347 and vector 8 were used.

add vdn 3347	Page 1 of 3
VECTOR DIRECTORY NUMBER	
Extension: 3347	
Name*: Accquen CBM Inbound	
Destination: Vector Number 8	
Attendant Vectoring? n	
Meet-me Conferencing? n	
Allow VDN Override? n	
COR: 1	
TN*: 1	
Measured: none Report Adjunct Calls as	
ACD*? n	
VDN of Origin Annc. Extension*:	
1st Skill*:	
2nd Skill*:	
3rd Skill*:	

Vector 8 queues the call to the agent split (skill 1), checks the expected wait time for the agent split (skill 1), and if it exceeds 30 seconds they will give an option to the caller whether they want to stay in the queue or they want agent to call back. If the caller select #1 they will continue to wait in the queue otherwise the call will be routed to second VDN and from the second VDN the call is routed to Experience Portal via SIP trunk. Experience Portal will then direct the call to the iAssist CBM application.

add vector 8				Page 1 of 6	
CALL VECTOR					
Number: 8		Name: Acqueon Inbound			
Multimedia? n	Attendant Vectoring? n	Meet-me Conf? n	Lock?		
n					
Basic? y	EAS? y	G3V4 Enhanced? y	ANI/II-Digits? y	ASAI Routing?	
y					
Prompting? y	LAI? y	G3V4 Adv Route? y	CINFO? y	BSR? y	Holidays? y
Variables? y	3.0 Enhanced? y				
01 wait-time	5	secs hearing 1100		then silence	
02 goto step	6	if expected-wait		for skill 1	pri m > 30
03 check	skill 1	pri m if unconditionally			
04 queue-to	skill 1	pri m			
05 wait-time	15	secs hearing 1100		then silence	
06 collect	1	digits after announcement 1105		for none	
07 goto step	3	if digits		=	1
08 route-to	number 3349	with cov n if unconditionally			
09 stop					
10 disconnect	after announcement none				

Below is the second VDN 3349 and vector 11 to route the contact center call to Experience Portal.

add vdn 3349	VECTOR DIRECTORY NUMBER		Page	1 of	3
Extension: 3349					
Name*: Second VDN for CBM					
Destination: Vector Number				11	
Attendant Vectoring? n					
Meet-me Conferencing? n					
Allow VDN Override? n					
COR: 1					
TN*: 1					
Measured: both				Report Adjunct Calls as	
ACD*? n					
Acceptable Service Level (sec): 20					
VDN of Origin Annc. Extension*:					
1st Skill*:					
2nd Skill*:					
3rd Skill*:					

And vector 11

add vector 11						Page 1 of 6	
CALL VECTOR							
Number: 11		Name: To route call to CBM					
Multimedia? n	Attendant Vectoring? n		Meet-me Conf? n			Lock?	
n							
Basic? y	EAS? y	G3V4 Enhanced? y	ANI/II-Digits? y		ASAI Routing?		
y							
Prompting? y	LAI? y	G3V4 Adv Route? y	CINFO? y	BSR? y	Holidays? y		
Variables? y	3.0 Enhanced? y						
01 route-to	number 4905		with cov n if unconditionally				

4.4. Administer Signaling Group

Use the “add signaling-group n” command, where “n” is an available signaling group number, in this case “1”. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** “sip”
- **Transport Method:** “tls”
- **Near-end Node Name:** An existing C-LAN node name or “procr”.
- **Far-end Node Name:** The existing node name for Session Manager.
- **Near-end Listen Port:** An available port for integration with Session Manager
- **Far-end Listen Port:** The same port number as in **Near-end Listen Port**.
- **Far-end Network Region:** An existing network region to use with Session Manager.
- **Far-end Domain:** The applicable domain name for the network.
- **Direct IP-IP Audio Connections:** “y”

change signaling-group 1		Page 1 of 3	
SIGNALING GROUP			
Group Number: 1		Group Type: sip	
IMS Enabled? n		Transport Method: tls	
Q-SIP? n			
IP Video? n		Enforce SIPS URI for SRTP? n	
Peer Detection Enabled? n		Peer Server: SM	
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y			
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n			
Alert Incoming SIP Crisis Calls? n			
Near-end Node Name: procr		Far-end Node Name: interopASM	
Near-end Listen Port: 5061		Far-end Listen Port: 5061	
		Far-end Network Region: 1	
Far-end Domain: bvwdev.com			
		Bypass If IP Threshold Exceeded? n	
Incoming Dialog Loopbacks: eliminate		RFC 3389 Comfort Noise? n	
DTMF over IP: rtp-payload		Direct IP-IP Audio Connections? y	
Session Establishment Timer(min): 3		IP Audio Hairpinning? n	
Enable Layer 3 Test? y		Initial IP-IP Direct Media? n	

4.5. Administer Trunk Group

Use the “add trunk-group n” command, where “n” is an available trunk group number, in this case “1”. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** “sip”
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** “tie”
- **Signaling Group:** The signaling group number from **Section 5.4**.
- **Number of Members:** The desired number of members, in this case “14”.

```
add trunk-group 1                                     Page 1 of 22
                                     TRUNK GROUP
Group Number: 1                                     Group Type: sip          CDR Reports: y
  Group Name: Private Trunk                         COR: 1          TN: 1          TAC: #01
  Direction: two-way                               Outgoing Display? n
Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: tie                                Auth Code? n
                                                Member Assignment Method: auto
                                                Signaling Group: 1
                                                Number of Members: 14
```

Go to the page 3, set **UI Treatment** as “shared” and **Send UCID?** to **y**. The iAssist Callback Manager application needs to obtain the UCID information of incoming call from Communication Manager to Experience Portal.

```
add trunk-group 1                                     Page 3 of 22
TRUNK FEATURES
  ACA Assignment? n                               Measured: none
                                                Maintenance Tests? y
  Suppress # Outpulsing? n   Numbering Format: private
                                                UI Treatment: shared
                                                Maximum Size of UI Contents: 128
                                                Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? y
                                                Hold/Unhold Notifications? y
  Send UCID? y                                   Modify Tandem Calling Number: no
Show ANSWERED BY on Display? y
```

4.6. Administer Route Pattern

Use the “change route-pattern n” command, where “n” is an existing route pattern number to be used to reach Experience Portal, in this case “1”. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern Name:** A descriptive name.
- **Grp No:** The SIP trunk group number from **Section 5.4**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.

change route-pattern 1													Page 1 of 3		
Pattern Number: 1													Pattern Name: SIP-TLS-To-SM		
SCCAN? n		Secure SIP? n		Used for SIP stations? n											
Grp		FRL	NPA	Pfx	Hop	Toll	No.	Inserted					DCS/	IXC	
No				Mrk	Lmt	List	Del	Digits					QSIG		
								Dgts					Intw		
1: 1		0											n	user	
2:											n	user			
3:											n	user			
4:											n	user			
5:											n	user			
6:											n	user			
BCC		VALUE		TSC	CA-TSC		ITC		BCIE	Service/Feature		PARM	Sub	Numbering	LAR
0		1	2	M	4	W	Request					Dgts	Format		
1:		Y	Y	Y	Y	Y	n	n	rest				lev0-pvt	next	
2:		Y	Y	Y	Y	Y	n	n	rest					none	
3:		Y	Y	Y	Y	Y	n	n	rest					none	
4:		Y	Y	Y	Y	Y	n	n	rest					none	
5:		Y	Y	Y	Y	Y	n	n	rest					none	
6:		Y	Y	Y	Y	Y	n	n	rest					none	

4.7. Administer Dial Plan

This section provides a sample dial plan used for routing calls with dialed digits 49xx to Experience Portal. Use the “change dialplan analysis 0” command, and add an entry to specify the use of digits pattern 49, as shown below.

change dialplan analysis						Page 1 of 12		
DIAL PLAN ANALYSIS TABLE								
Location: all						Percent Full: 4		
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type
0	3	fac	43	4	aar			
1	4	ext	49	4	aar			
13	5	aar	46	4	aar			
14	5	aar	50	5	aar			
20	4	aar	546	5	aar			
23	5	aar	56	5	udp			
24	5	aar	60	5	udp			
28	5	aar	8	1	fac			
30	4	aar	9	1	fac			
33	4	ext	*	3	dac			

4.8. Administer AAR Table

Use the “change aar analysis 0” command, and add an entry to specify how to route calls to 49xx. In the example shown below, calls with digits 49xx will be routed as an AAR call using route pattern “1” from **Section 5.6**

change aar analysis 49							Page	1 of	2
AAR DIGIT ANALYSIS TABLE									
Location: all							Percent Full: 2		
	Dialed	Total		Route	Call	Node	ANI		
	String	Min	Max	Pattern	Type	Num	Reqd		
49		4	4	1	aar		n		

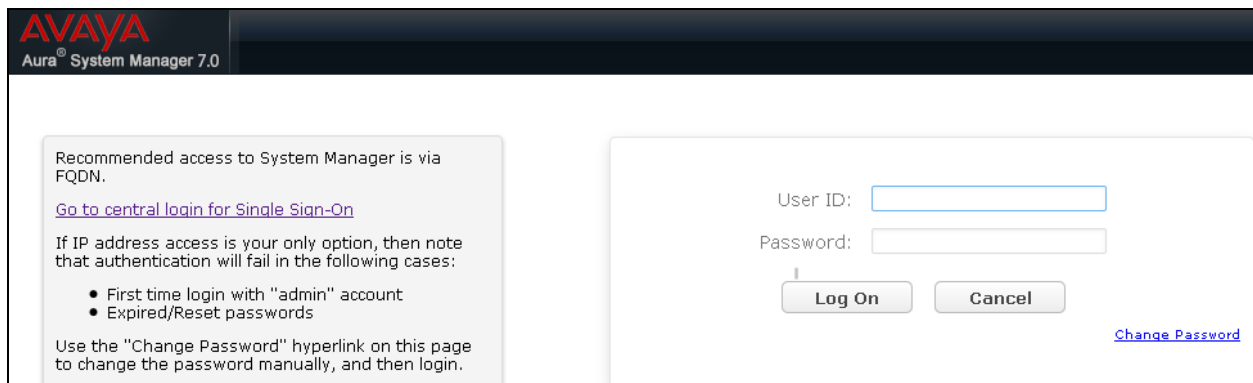
5. Configure Avaya Aura® Session Manager

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

- Launch System Manager
- Administer Domain
- Administer locations
- Administer Adaptation
- Administer SIP entities
- Administer routing policies
- Administer dial patterns

5.1. Launch System Manager

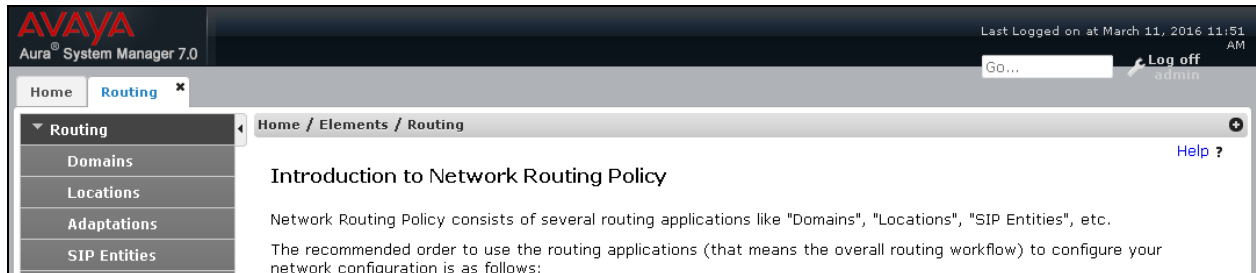
Access the System Manager web interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of System Manager. Log in using the appropriate credentials.



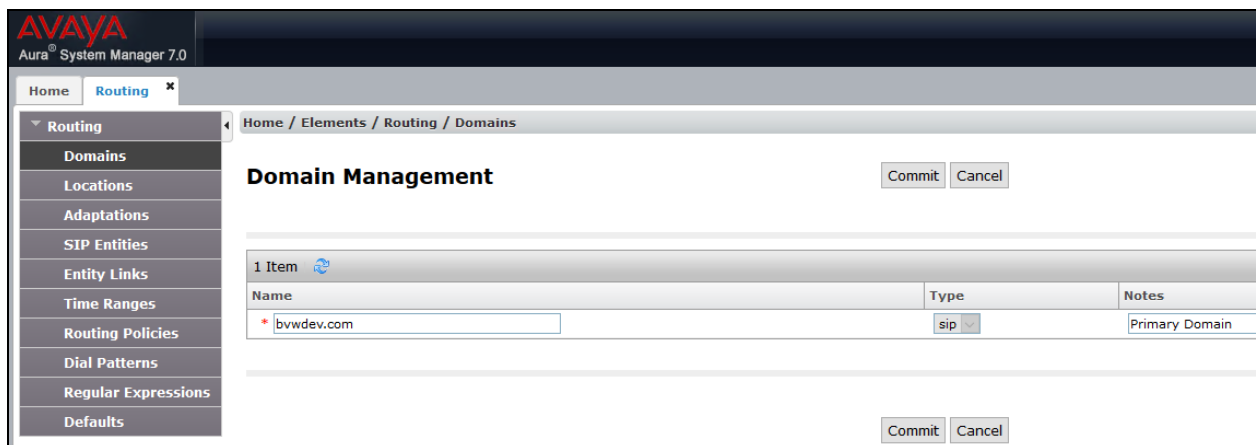
The screenshot shows the Avaya Aura System Manager 7.0 login interface. The header features the Avaya logo and the text "Aura® System Manager 7.0". The main content area is divided into two sections. The left section contains a message: "Recommended access to System Manager is via FQDN." followed by a link "Go to central login for Single Sign-On". Below this, it states: "If IP address access is your only option, then note that authentication will fail in the following cases:" followed by a bulleted list: "• First time login with 'admin' account" and "• Expired/Reset passwords". It also includes a note: "Use the 'Change Password' hyperlink on this page to change the password manually, and then login." The right section contains the login form with fields for "User ID:" and "Password:", a "Log On" button, a "Cancel" button, and a "Change Password" link.

5.2. Administer Domain

In the subsequent screen (not shown), select **Elements → Routing** to display the **Introduction to Network Routing Policy** screen below. Select **Routing → Domains** from the left pane, and click **New** in the subsequent screen (not shown) to add a new domain



The **Domain Management** screen is displayed. In the **Name** field enter the domain name, select *sip* from the **Type** drop down menu and provide any optional **Notes**.



5.3. Administer Locations

Select **Routing** → **Locations** from the left pane, and click **New** in the subsequent screen (not shown) to add a new location for Trio Enterprise.

The **Location Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name** and optional **Notes**. Retain the default values in the remaining fields.

AVAYA
Aura® System Manager 7.0

Last Logged on at May 23, 201

Home Routing

Home / Elements / Routing / Locations

Location Details

Commit Cancel

General

* Name: BvwDevSIL

Notes:

Scroll down to the **Location Pattern** sub-section, click **Add** and enter the IP address of all devices involved in the compliance testing in **IP Address Pattern**, as shown below. Retain the default values in the remaining fields.

Location Pattern

Add Remove

4 Items Filter: Enable

IP Address Pattern	Notes
* 10.10.5.*	
* 10.10.97.*	
* 10.10.98.*	
*	

Select : All, None

Commit Cancel

5.4. Administer Adaptation

During compliance test, the dial pattern 4905 was used to route the contact center call to Experience Portal when the caller decides to have agent call them back. When the call leaves Session Manager and arrives in Experience Portal the number 4905 in From header will be replaced by the number 3349 which is second VDN configured in **Section 5.3**. The iAssist Callback application monitors this VDN and they need to receive this number in their callback application. Here are the steps to create an Adaptation. Select **Adaptations** on the left panel menu and then click on the **New** button in the main window (not shown). Enter the following for the newly adaptation.

- **Adaptation Name** An informative name (e.g., ChangeFromNumber)
- **Module Name** Select **DigitConversionAdapter**
- **Module Parameter Type** Select Name-Value Parameter

Click **Add** to add a new row for the following values as shown below table:

Name	Value
fromto	true

Home / Elements / Routing / Adaptations

Adaptation Details [Commit] [Cancel] [Help ?]

General

* Adaptation Name: ChangeFromNumber

* Module Name: DigitConversionAdapter

Module Parameter Type: Name-Value Parameter

Add Remove

Name	Value
fromto	true

Select : All, None

Egress URI Parameters: []

Notes: []

(Continue) the screenshot shows the adaptation.

Digit Conversion for Outgoing Calls from SM

Add Remove

1 Item [Filter: Enable]

Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
* 4905	* 4	* 4	[]	* 4	3349	both	[]	[]

Select : All, None

5.5. Administer SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to it, which includes Communication Manager and Experience Portal.

5.5.1. SIP Entity for Session Manager

Navigate to **Routing** → **SIP Entities** in the left navigation pane and click on the **New** button in the right pane (not shown). In the **General** section, enter the following values. Use default values for all remaining fields:

- **Name:** Enter a descriptive name.
- **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity that is used for SIP signaling.
- **Type:** Select *Session Manager* for Session Manager.
- **Adaptation:** This field is only present if **Type** is not set to **Session Manager** if Adaptations were to be created, here is where they would be applied to the entity.
- **Location:** Select the location that applies to the SIP Entity being created, defined in **Section 7.3**.
- **Time Zone:** Select the time zone for the location above.

The following screen shows the addition of the *Session Manager* SIP Entity for Session Manager. The IP address of the Session Manager Security Module is entered in the **FQDN or IP Address** field.

The screenshot displays the Avaya Aura System Manager 7.0 web interface. The top navigation bar includes the Avaya logo, the text 'Aura System Manager 7.0', and a 'Last Logged on at May 23, 2017' timestamp. Below the navigation bar, there are tabs for 'Home', 'Session Manager', and 'Routing'. The 'Routing' tab is active, and the breadcrumb trail shows 'Home / Elements / Routing / SIP Entities'. On the left, a vertical navigation pane lists various configuration options: Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'SIP Entity Details' and has a 'General' sub-section. It contains several input fields: 'Name' (with a red asterisk) containing 'ASM70A', 'FQDN or IP Address' (with a red asterisk) containing '10.33.1.12', 'Type' (a dropdown menu set to 'Session Manager'), 'Notes' (an empty text area), 'Location' (a dropdown menu set to 'BvwDevSIL'), 'Outbound Proxy' (an empty text field), 'Time Zone' (a dropdown menu set to 'America/Toronto'), and 'Credential name' (an empty text field). At the bottom, there is a 'SIP Link Monitoring' section with a dropdown menu set to 'Use Session Manager Configuration'. 'Commit' and 'Cancel' buttons are located at the top right of the form area.

5.5.2. SIP Entity for Communication Manager

Select **Routing** → **SIP Entities** from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for Communication Manager. Note that this SIP entity is used for integration with Trio Enterprise.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of an existing CLAN or the processor interface.
- **Type:** Select “CM” in the dropdown list.
- **Notes:** Any desired notes.
- **Location:** Select the applicable location for Communication Manager.
- **Time Zone:** Select the applicable time zone.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The top header includes the Avaya logo and 'Aura System Manager 7.0'. The left navigation pane has a tree view with 'Routing' expanded and 'SIP Entities' selected. The main content area is titled 'SIP Entity Details' and includes a 'General' tab. The form contains the following fields:

- Name:** Text input field with value 'ACM-Trunk1-Private'.
- FQDN or IP Address:** Text input field with value '10.33.1.6'.
- Type:** Dropdown menu with 'CM' selected.
- Notes:** Text input field.
- Adaptation:** Dropdown menu.
- Location:** Dropdown menu with 'BvwDevSIL' selected.
- Time Zone:** Dropdown menu with 'America/Toronto' selected.
- SIP Timer B/F (in seconds):** Text input field with value '4'.
- Credential name:** Text input field.
- Securable:** Check box.
- Call Detail Recording:** Dropdown menu with 'none' selected.

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

5.5.3. SIP Entity for Experience Portal

Select **Routing** → **SIP Entities** from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for Experience Portal.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of the Experience Portal server.
- **Type:** Select “Voice Portal” in the dropdown list.
- **Notes:** Any desired notes.
- **Adaptation:** Select the adaptation configured in **Section 6.4**
- **Location:** Select the applicable location from **Section 6.3**.
- **Time Zone:** Select the applicable time zone.

The screenshot displays the Avaya Aura System Manager 7.1 interface. The left-hand navigation pane shows the 'Routing' menu expanded, with 'SIP Entities' selected. The main content area is titled 'SIP Entity Details' and includes a 'General' tab. The form contains the following fields and values:

- Name:** AEP71
- FQDN or IP Address:** 10.33.1.25
- Type:** Voice Portal
- Notes:** AEP System 7.1
- Adaptation:** ChangeFromNumber
- Location:** BvwDevSIL
- Time Zone:** America/Toronto
- SIP Timer B/F (in seconds):** 4
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty field)
- Securable:** (unchecked checkbox)
- Call Detail Recording:** none

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

5.6. Administer Entity Links

A SIP trunk between Session Manager and a telephony system is described by an Entity Link. Two Entity Links were created; one to the Communication Manager and one to Trio Enterprise. To add an Entity Link, select to **Routing** → **Entity Links** in the left navigation pane and click on the **New** button in the right pane (not shown). Fill in the following fields in the new row that is displayed:

- **Name:** Enter a descriptive name.
- **SIP Entity 1:** Select the Session Manager from the drop-down menu.
- **Protocol:** Select applicable transport protocol.
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end.
- **SIP Entity 2:** Select the name of the other systems from the drop-down menu.
- **Port:** Port number on which the other system receives SIP requests from Session Manager.
- **Connection Policy:** Select **Trusted** to allow calls from the associated SIP Entity.

The screens below show the Entity Link to Communication Manager and Experience Portal. During the compliance test, **TLS** transport with port **5061** was used between Session Manager and Communication Manager.

The screenshot shows the 'Entity Links' configuration page. The breadcrumb is 'Home / Elements / Routing / Entity Links'. There are 'Commit' and 'Cancel' buttons. A table lists one item with the following details:

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port
ASM70_ACM_Trunk1_50	ASM70A	TLS	5061	ACM-Trunk1-Private	<input type="checkbox"/>	5061

The Entity Link to Experience Portal is shown below; **TCP** transport and port **5060** were used.

The screenshot shows the 'Entity Links' configuration page. The breadcrumb is 'Home / Elements / Routing / Entity Links'. There are 'Commit' and 'Cancel' buttons. A table lists one item with the following details:

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port
ASM70A_AEP71_5060_T	ASM70A	TCP	5060	AEP71	5060

5.7. Administer Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 7.5**. Two routing policies were added: an incoming policy with Communication Manager as the destination, and an incoming policy to Experience Portal. To add a routing policy, select to **Routing → Routing Policies** in the left navigation pane and click on the **New** button in the right pane (not shown). The following screen is displayed:

- In the **General** section, enter a descriptive **Name** and add a brief description under **Notes** (optional).
- In the **SIP Entity as Destination** section, click **Select**. The **SIP Entity List** page opens (not shown). Choose the appropriate SIP entity to which this routing policy applies (**Section 6.5**) and click **Select**. The selected SIP Entity displays on the **Routing Policy Details** page as shown below.
- Use default values for remaining fields.
- Click **Commit** to save.

The following screens show the Routing Policy for Communication Manager.

AVAYA
Aura® System Manager 7.0

Last Logged on at May 23, 2017

Home / Elements / Routing / Routing Policies

Routing Policy Details

Commit Cancel

General

* Name:

Disabled: ☐

* Retries:

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
ACM-Trunk1-Private	10.33.1.6	CM	

The following screens show the Routing Policy for Experience Portal.

Home

Routing

1 New important message(s). Click to view details.

Home / Elements / Routing / Routing Policies

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Routing Policy Details

CommitCancel

Help?

General

Name: To-AEP

Disabled: ☐

Retries: 0

Notes: route to EP system 10.33.1.25

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
AEP71	10.33.1.25	Voice Portal	AEP System2 10.33.1.25

Time of Day

AddRemoveView Gaps/Overlaps

1 Item

Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None

5.8. Administer Dial Patterns

Dial Patterns are needed to route specific calls through Session Manager. For the compliance test, dial patterns were needed to route calls from Communication Manager to Experience Portal and vice versa. Dial Patterns define which route policy will be selected for a particular call based on the dialed digits, destination domain and originating location.

5.8.1. Dial Pattern for Experience Portal

Select **Routing** → **Dial Patterns** from the left pane, and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach Experience Portal. The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern:** A dial pattern to match, in this case “49”.
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** The domain name from **Section 6.2**.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching Experience Portal. In the compliance testing, the entry allowed for all call originations in the location “ALL”. The Experience Portal routing policy from **Section 6.5.3** was selected as shown below.

Dial Pattern Details Commit Cancel

General

* **Pattern:** 49

* **Min:** 4

* **Max:** 4

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: bwvdev.com

Notes: Route to Experience Portal R7.1

Originating Locations and Routing Policies

Add Remove

1 Item Filter

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Notes
<input type="checkbox"/>	-ALL-		To-AEP	0	<input type="checkbox"/>	AEP71	route system 10.33

5.8.2. Dial Pattern for Communication Manager

Select **Routing** → **Dial Patterns** from the left pane, and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach Communication Manager. The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern:** A dial pattern to match, in this case “33”.
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** The domain name from **Section 6.2**.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching Communication Manager. In the compliance testing, the entry allowed for all originating locations “ALL”. The Communication Manager routing policy from **Section 6.5.2** was selected as shown below.

Dial Pattern Details [Commit] [Cancel]

General

* Pattern: 33

* Min: 4

* Max: 4

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: bwvdev.com

Notes: Dial pattern to CM71 from all locations

Originating Locations and Routing Policies

[Add] [Remove]

1 Item [Filter: Enable]

<input type="checkbox"/>	Originating Location Name ▲	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Note
<input type="checkbox"/>	-ALL-		To-CM-Trunk1	0	<input type="checkbox"/>	ACM-Trunk1-Private	

Select : All, None

6. Configure Avaya Aura® Experience Portal

Avaya Aura® Experience Portal is configured via the Experience Portal Manager (EPM) web interface, to access the web interface, enter **http://<ip-addr>/** as the URL in a web browser, where <ip-addr> is the IP address of the EPM. Log in using the appropriate credentials.

Note: Some of the screens in this section are shown after the Experience Portal had been configured. Don't forget to save the screen parameters as you configure Avaya Aura® Experience Portal.

AVAYA Welcome, eadmin
Last logged in today at 1:21:13 PM PST

Avaya Aura® Experience Portal 7.1.0 (ExperiencePortal) Home ? Help Logoff

Expand All | Collapse All

You are here: Home

Avaya Aura® Experience Portal Manager

Avaya Aura® Experience Portal Manager (EPM) is the consolidated web-based application for administering Experience Portal. Through the EPM interface you can configure Experience Portal, check the status of an Experience Portal component, and generate reports related to system operation.

License grace period for Experience Portal will end on Jan 16, 2017 10:46:53 AM PST.

Installed Components

Media Processing Platform
Media Processing Platform (MPP) is an Avaya media processing server. When an MPP receives a call from a PBX, it invokes a VoiceXML (or CCXML) application on an application server. It then communicates with ASR and TTS servers as necessary to process the call.

Email Service
Email Service is an Experience Portal feature which provides e-mail capabilities.

HTML Service
HTML Service is an Experience Portal feature which supports web applications with HTML5 capabilities. It includes support for browser based services for mobile devices.

SMS Service
SMS Service is an Experience Portal feature which provides SMS capabilities.

- ▼ **User Management**
 - Roles
 - Users
 - Login Options
- ▼ **Real-time Monitoring**
 - System Monitor
 - Active Calls
 - Port Distribution
- ▼ **System Maintenance**
 - Audit Log Viewer
 - Trace Viewer
 - Log Viewer
 - Alarm Manager
- ▼ **System Management**
 - Application Server
 - EPM Manager
 - MPP Manager
 - Software Upgrade
 - System Backup
- ▼ **System Configuration**
 - Applications
 - EPM Servers
 - MPP Servers
 - SNMP
 - Speech Servers
 - VoIP Connections
 - Zones
- ▼ **Security**
 - Certificates
 - Licensing
- **Reports**
- **Multi-Media Configuration**

6.1. Administer VoIP Connection

On the left pane, click on the VoIP Connections under System Configuration (not shown). To add a **SIP Connection**, click on **SIP** tab on **VoIP Connections** page (not shown). Fill in **Name**, in the **Address** and **Port** boxes, select “TCP” in the **Proxy Transport** dropdown menu, fill the SM signaling IP address and Port of the SIP Proxy used for call transport, in this case Avaya Aura® Session Manager was used, in **SIP Domain**, fill in the domain and set the **Maximum Simultaneous Calls**. All other values can be left as **Default**. Click **Save** to save changes.

AVAYA

Welcome, epadmin
Last logged in today at 7:22:48 AM PDT

Avaya Aura® Experience Portal 7.1.0 (ExperiencePortal)

Home Help Logoff

Expand All | Collapse All

▼ **User Management**

Roles

Users

Login Options

▼ **Real-time Monitoring**

System Monitor

Active Calls

Port Distribution

▼ **System Maintenance**

Audit Log Viewer

Trace Viewer

Log Viewer

Alarm Manager

▼ **System Management**

Application Server

EPM Manager

MPP Manager

Software Upgrade

System Backup

▼ **System Configuration**

Applications

EPM Servers

MPP Servers

SNMP

Speech Servers

VoIP Connections

Zones

▼ **Security**

Certificates

Licensing

▼ **Reports**

Standard

Custom

Scheduled

▼ **Multi-Media Configuration**

Email

HTML

SMS

You are here: [Home](#) > [System Configuration](#) > [VoIP Connections](#) > [Change SIP Connection](#)

Change SIP Connection

Use this page to change the configuration of a SIP connection.

Name:

Enable: ☒ Yes ☐ No

Proxy Transport:

☒ Proxy Servers ☐ DNS SRV Domain

Address	Port	Priority	Weight	
10.33.1.12	5060	0	0	Remove

[Additional Proxy Server](#)

Listener Port:

SIP Domain:

P-Asserted-Identity:

Maximum Redirection Attempts:

Consultative Transfer: ☒ INVITE with REPLACES ☐ REFER

SIP Reject Response Code: ☒ ASM (503) ☐ SES (480) ☐ Custom

SIP Timers

T1: milliseconds

T2: milliseconds

B and F: milliseconds

Call Capacity

Maximum Simultaneous Calls:

6.2. Configure iAssist CBM Applications

Two applications are configured in Avaya Aura® Experience Portal, one to handle inbound calls that are queued to the agent split and the second one to handle the call back request (i.e., outbound calls to agent and caller).

6.2.1. Configure the Inbound CBM Application

In the **Applications** page, add an Experience Portal application to handle incoming calls that are queued to the agent split. This application will provide the caller the option to either continue waiting in the agent queue or to request a call back. Configure the application as shown below.

The screenshot displays the Avaya Aura® Experience Portal 7.1.0 (ExperiencePortal) interface. The left sidebar contains a navigation menu with categories: User Management, Real-time Monitoring, System Maintenance, System Management, System Configuration, Security, Reports, and Multi-Media Configuration. The main content area is titled 'Change Application' and shows the configuration for the 'iAssist_CBM' application. The configuration includes fields for Name, Enable (Yes/No), Type (VoiceXML), Reserved SIP Calls (None/Minimum/Maximum), Requested, URI (Single/Fail Over/Load Balance), VoiceXML URL, Mutual Certificate Authentication, Basic Authentication, Speech Servers (ASR/TTS), and Application Launch (Inbound/Inbound Default/Outbound, Number/Number Range/URI). A list of called numbers is shown at the bottom, with '3349' currently selected and an 'Add' button next to it.

Avaya Aura® Experience Portal 7.1.0 (ExperiencePortal)

Expand All | Collapse All

▼ User Management
Roles
Users
Login Options

▼ Real-time Monitoring
System Monitor
Active Calls
Port Distribution

▼ System Maintenance
Audit Log Viewer
Trace Viewer
Log Viewer
Alarm Manager

▼ System Management
Application Server
EPM Manager
MPP Manager
Software Upgrade
System Backup

▼ System Configuration
Applications
EPM Servers
MPP Servers
SNMP
Speech Servers
VoIP Connections
Zones

▼ Security
Certificates
Licensing

▼ Reports
Standard
Custom
Scheduled

▼ Multi-Media Configuration
Email
HTML
SMS

You are here: [Home](#) > [System Configuration](#) > [Applications](#) > Change Application

Change Application

Use this page to change the configuration of an application.

Name: iAssist_CBM

Enable: ☒ Yes ☐ No

Type: VoiceXML

Reserved SIP Calls: ☒ None ☐ Minimum ☐ Maximum

Requested:

URI

☒ Single ☐ Fail Over ☐ Load Balance

VoiceXML URL: **Verify**

Mutual Certificate Authentication: ☐ Yes ☒ No

Basic Authentication: ☐ Yes ☒ No

Speech Servers

ASR: No ASR TTS: No TTS

Application Launch

☒ Inbound ☐ Inbound Default ☐ Outbound

☒ Number ☐ Number Range ☐ URI

Called Number: **Add**

Remove

6.2.2. Configure the Outbound CBM Application

In the **Applications** page, add another Experience Portal application to handle the outbound calls to the agent and caller. Configure the application as shown below.

The screenshot displays the Avaya Aura Experience Portal 7.1.0 (ExperiencePortal) interface. The top navigation bar is red with 'Avaya Aura® Experience Portal 7.1.0 (ExperiencePortal)' on the left and 'Home' and 'Help' on the right. A breadcrumb trail indicates the current location: 'You are here: Home > System Configuration > Applications > Change Application'.

The left sidebar contains a tree view of system components, including User Management, Real-time Monitoring, System Maintenance, System Management, System Configuration (highlighted), Security, Reports, and Multi-Media Configuration.

The main content area is titled 'Change Application' and includes the instruction: 'Use this page to change the configuration of an application.' The configuration fields are as follows:

- Name:** IASSIST_CBM_OUTBOUND
- Enable:** ☒ Yes ☐ No
- Type:** CCXML (dropdown menu)
- Reserved SIP Calls:** ☒ None ☐ Minimum ☐ Maximum
- Requested:** (empty text field)
- URI:**
 - ☒ Single ☐ Fail Over ☐ Load Balance
 - CCXML URL:** http://10.10.98.2:8080/iAssistOutboundCBM_SIP_CCXML/ccxml/start.jsp (with a 'Verify' button)
 - Mutual Certificate Authentication:** ☐ Yes ☒ No
 - Basic Authentication:** ☐ Yes ☒ No
- Speech Servers:**
 - ASR:** No ASR (dropdown menu)
 - TTS:** No TTS (dropdown menu)
- Application Launch:**
 - ☐ Inbound ☐ Inbound Default ☒ Outbound
- Speech Parameters** (expandable section)
- Reporting Parameters** (expandable section)
- Advanced Parameters** (expandable section)

At the bottom of the form are four buttons: 'Save', 'Apply', 'Cancel', and 'Help'.

6.3. Configure the Outcall Authentication

Configure the Outcall User Name and Password that will be sent by iAssist CBM. Click on **EPM Servers** in the left pane, in the resulting page, click on **EPM Settings** to display the page below. Under the **Outcall** section, configure the **User Name** and **Password** used by iAssist CBM when it makes an outcall request to Experience Portal.

Avaya Aura® Experience Portal 7.1.0 (ExperiencePortal) [Home](#)

Expand All | Collapse All

You are here: [Home](#) > System Configuration > [EPM Servers](#) > EPM Settings

EPM Settings

Use this page to configure system parameters that affect the Experience Portal system.

Experience Portal Name:

Number of Application Server Failover Logs :

Commands to Retain in Configuration History:

Resource Alerting Thresholds (%) ▼

HTML Units:	<input type="text" value="80"/>
Disk:	<input type="text" value="90"/> High Water <input type="text" value="80"/> Low Water

Web Service Authentication ▼

Application Reporting

User Name:

Password:

Verify Password:

Outcall

User Name:

Password:

Verify Password:

Miscellaneous ▶

Save **Apply** **Cancel** **Help**

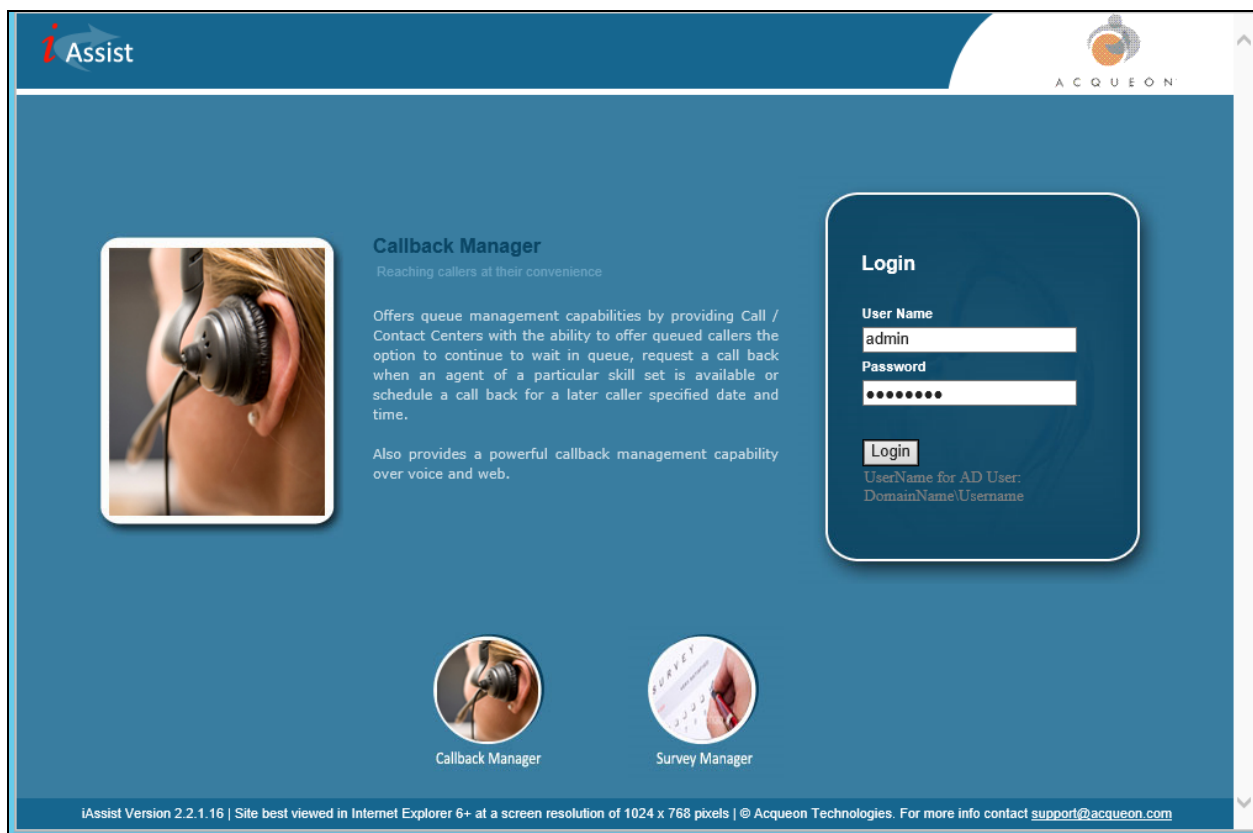
7. Configure Acqueon iAssist Call Back Manager

The configuration of iAssist Callback Manager system is done by Acqueon engineer and is outside of the scope of these Application Notes. This section covers the information on how to use the iAssist Admin application to administer the Callback Manager (CBM).

7.1. Steps to configure the Business Group

Type the URL: <http://10.10.98.2/iAssist> to login into the admin page followed by the User Name and the Password.

Note: The current version of iAssist Callback Manager only supports Microsoft Internet Explorer. The default Username is admin and the password is admin123.



7.2. Configure the business group

Business Group refers to the type of business the application caters. Each business group will have a language and a unique number where the call will be routed to so that the application can identify the caller.

Business Group Management enables configuration and management of a business group. Use the Business Group option under the General tab to add, modify or delete a business group.

- Enter a valid Business Group Name.
- Set the Incoming Number to the number that routes calls to EP (e.g., 3349).
- Select a Site to from the dropdown menu to associate the business group to a site.
- Select the appropriate Language.
- Select the required IVR Configuration Template.

The screenshot displays the iAssist Business Group Management interface. The top navigation bar includes links for Home, Manage, General, CBM, CSM, and License. The user is logged in as 'admin'. The main content area is divided into two panels. The left panel, titled 'BusinessGroup Management', contains a form with the following fields: Business Group Name (AACC_CBM_3349), Incoming Number (3349), Site (AACC_Site1), Language (US English), and IVR Configuration Template (DEFAULT_CBM_CONFIG). The form has 'Update Business Group' and 'Cancel' buttons. The right panel, titled 'Defined Business Group(s)', contains a table with the following data:

Business Group	Edit	Delete
AACC_CBM_3349		
AACC_CSM_4906		

7.3. Configuring Business Group

From the menu, select the **CBM → Business Group Configuration** tab. Click the **Edit** icon of the desired business group to edit the Defined Business Group(s) displayed in the right pane. The Business Group Name will be populated automatically.

- Enter the Outgoing Number (VDN number configured to reach the available agent who is configured/ logged into a particular skill).
- Select the High Priority Queue check box, if required. If there is a separate high priority queue created to handle outbound callback requests, select the High Priority Queue checkbox.
- Provide the High Priority Queue VDN Number.

- IVR IP Address [Voice Portal Management System's (VPMS) IP that has been used for dialing the agent and/ or customer].
- Time Zone (Time zone of system in which iAssist application is deployed).
- Priority can be set as High, Medium, or Low. (Priority that needs to be set for the particular business group. If calls from many business groups are scheduled for the same time, then they will be dialed out based on the Business Group Priority set here).

Home
Manage
General
CBM
CSM
License
Welcome admin |

CBM - Business Group Configuration [AACC_CBM_3349]

Business Group Name

AACC_CBM_3349

Outgoing Number *

3348

High Priority Queue

☒

High Priority Queue VDN

3348

IVR IP Address *

10.33.1.25

Time Zone

(UTC-05:00) Eastern Time (US & Canada)

Priority

HIGH

UUI Data processing

☐

Defined Business Group(s)

Business Group	Edit
AACC_CBM_3349	

7.4. Business Hours and Break Hours

Business hours and break hours have to be configured in the **Business Hours and Break Hours** tab. It should be entered in the 24-hour format, the break hour is an interval within the business hours, for example, lunch break. Callback request options will be offered to the callers based on the business hours and will not be allowed outside of this schedule. Business hours and break hours should be configured for each day of the week separately as shown.

Home
Manage
General
CBM
CSM
License
Welcome admin | Logout

CBM - Business Group Configuration [AACC_CBM_3349]

RealTime Queue

Business Hour and Break Hour

	Business Hour [24 Hrs Format]			Break Hour [24 Hrs Format]	
	StartTime	Inbound-EndTime	Outbound-End Time(Dialing)	Start Time	End Time
Monday	09:00	18:00	18:15	00:00	00:00
Tuesday	09:00	18:00	18:15	00:00	00:00
Wednesday	09:00	18:00	18:15	00:00	00:00
Thursday	09:00	18:00	18:15	00:00	00:00
Friday	00:00	18:00	18:15	00:00	00:00
Saturday	09:00	18:00	18:15	00:00	00:00
Sunday	09:00	18:00	18:15	00:00	00:00

Defined Business Group(s)

Business Group	Edit
AACC_CBM_3349	

7.5. Time Slots

Time Slot is a defined interval, or slot of time that is offered to callers to choose the call back time. If this is configured, the Inbound CBM will offer the caller the list of configured time slots and the caller can choose one. If this is not configured, the caller will be prompted to enter a time to receive the call back. Timeslots will be played to the caller for the callback options (S- same date and later time and F- Future date and time), if configured.

CBM - Business Group Configuration [AACC_CBM_3349]	
RealTime Queue	
Business Hour and Break Hour	
Holiday	
Timezones	
Time Slots	
Start Time & End Time	09:00 18:00
Max Threshold	
<button>Add</button>	

Defined Business Group	
Business Group	
AACC_CBM_3349	

7.6. Config Options

In Config Options, the **Callback Options** tab allows setting of the various options to be offered to the caller to log a callback request and receive a callback. These options will be dynamically offered based on the settings like Business Hours and Holidays, which are configured.

- As soon as agent available
- Same date later time
- Future date and time

Config Options	
Callback Options	Duplicate Filter
Outbound Configuration	Failure Outcomes
Hidden	
As soon as agent available	<input checked="" type="checkbox"/>
Immediate Callback	<input type="checkbox"/>
Same date later time	<input checked="" type="checkbox"/>
Future date and time	<input checked="" type="checkbox"/>
After 1 hour	<input type="checkbox"/>
Route back to Agent Queue	<input type="checkbox"/>

7.7. Call Flow Generator

From the menu, select General → **CallFlow Generator**. Under this section, call flows can be generated for a business group or business group collection.

- Specify a Call Flow Name.
- Select the required Site.
- Select the desired application from the drop down list in the Application field.
- Select the Filter Type.
- Select a Business Group.

CallFlow	Edit	Delete
CSM_Inbound_CallFlow		
CBM_Outbound_CallFlow		
Inbound_CBM_QP		

In the **Defined Elements** section, select the **Element Name** and click on the **Add Element** button to be displayed below.

Use Template ☐

Element Name

VoiceFileName


Value

CBOptions | - | -
ContactNumber | - | -
RecordName | - | -
Date | - | -
Time | - | -

8. Verification Steps

This section provides the verification steps that may be performed to verify that Experience Portal can run iAssist CBM applications.



1. From the EPM web interface, verify that the MPP server is online and running in the **System Monitor** page shown below.

System Monitor (Aug 22, 2017 1:55:33 PM PDT)  [Refresh](#)

This page displays the current state of the local Experience Portal system plus any remote Experience Portal systems that you have configured. For information about the colored alarm symbols, click Help.


[Summary](#) [ExperiencePortal Details](#)

Last Poll: Aug 22, 2017 1:55:28 PM PDT

Server Name	Type	Mode	State	Config	Call Capacity			Active Calls		Calls Today	Alarms
					Current	Licensed	Maximum	In	Out		
EPM / mpp	EPM/MPP	Online	Running	OK	10	10	50	0	0	1	
Summary					10	10	50			1	

[Help](#)

2. From the EPM web interface, verify that the ports on the MPP server are in-service in the **Port Distribution** page shown below.

You are here: [Home](#) > [Real-Time Monitoring](#) > [Port Distribution](#) > Port Distribution Report  [Refresh](#)

Port Distribution Report (Aug 22, 2017 1:56:51 PM PDT)

This page displays information about how the telephony resources have been distributed to the MPPs. You configure the telephony resources on the VoIP Connections page.

Total Ports: 10

Last Poll: Aug 22, 2017 1:56:35 PM PDT

Port	Mode	State	Port Group	Protocol	Current Allocation	Base Allocation
10	Online	In service	ASM70	SIP_Trunk	mpp	

3. Log out all agents from the skillset or put them in Not Ready status, place calls to the VDN that handles incoming contact center calls and queues them to the agent skillset so that the expected wait time exceeds the threshold configured in the vector. The caller will be prompted to enter an option for call back.

4. As soon as the caller selects the option for agent call back, the caller will be routed to iAssist Callback Manager application in Experience Portal. From there, the caller will enter the information when they want to receive a call back.
5. To check the status of callback, select **General → Status Management**, in the Call Status field (not shown) select a status to display, e.g. “Pending”, “Completed”, or “Failed”. The screenshot below shows the “Completed” call back.

Home Manage General CBM CSM License Welcome admin | [Logout](#)

Status Management

Site:

From Date:

Call Status:

Business Group:

End Date:

Total no of Records:

SI No	Call ID	BusinessGroup	Request Time	Customer Number	<input type="checkbox"/> Select
1	20170818123658	AACC_CBM_3349	8/18/2017 12:37:37 PM	16139671295	<input type="checkbox"/>
2	20170818121410	AACC_CBM_3349	8/18/2017 12:15:19 PM	16139671295	<input type="checkbox"/>
3	20170818120734	AACC_CBM_3349	8/18/2017 12:08:23 PM	4323	<input type="checkbox"/>
4	20170817145655	AACC_CBM_3349	8/17/2017 2:57:44 PM	4323	<input type="checkbox"/>
5	20170817123045	AACC_CBM_3349	8/17/2017 12:31:21 PM	3300	<input type="checkbox"/>
6	20170817122222	AACC_CBM_3349	8/17/2017 12:23:29 PM	94224684602	<input type="checkbox"/>
7	20170817113148	AACC_CBM_3349	8/17/2017 11:32:54 AM	16139671295	<input type="checkbox"/>
8	20170817105416	AACC_CBM_3349	8/17/2017 10:55:46 AM	16139671295	<input type="checkbox"/>
9	20170817100531	AACC_CBM_3349	8/17/2017 10:07:15 AM	94224684602	<input type="checkbox"/>
10	20170816110600	AACC_CBM_3349	8/16/2017 11:07:04 AM	16139671295	<input type="checkbox"/>
11	20170816104416	AACC_CBM_3349	8/16/2017 10:45:49 AM	16139671220	<input type="checkbox"/>

9. Conclusion

These Application Notes describe the configuration steps required to integrate the Acqueon iAssist Call Back Manager application with Avaya Aura® Experience Portal. All feature and serviceability test cases were completed successfully refer to **Section 2.2** for details.

10. Additional References

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

- [1] Administering Avaya Aura® Communication Manager, Release 7.0.3, Document 03-300509, Issue 10, June 2016
- [2] Administering Avaya Aura® Session Manager, Release 7.0, Issue 7, Jan 2016
- [3] Administering Avaya Aura® Experience Portal, Release 7.0.1, April 2015

Product Documentation for Acqueon iAssist Callback Manager can be obtained at <http://www.acqueon.com/avaya-products/iassist-for-avaya-aura-experience-portal/>

- [4] iAssist CBM 2.0 Admin Guide
- [5] iAssist CBM 2.0 IVR Installation Guide

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