



## Avaya Solution & Interoperability Test Lab

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# 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.

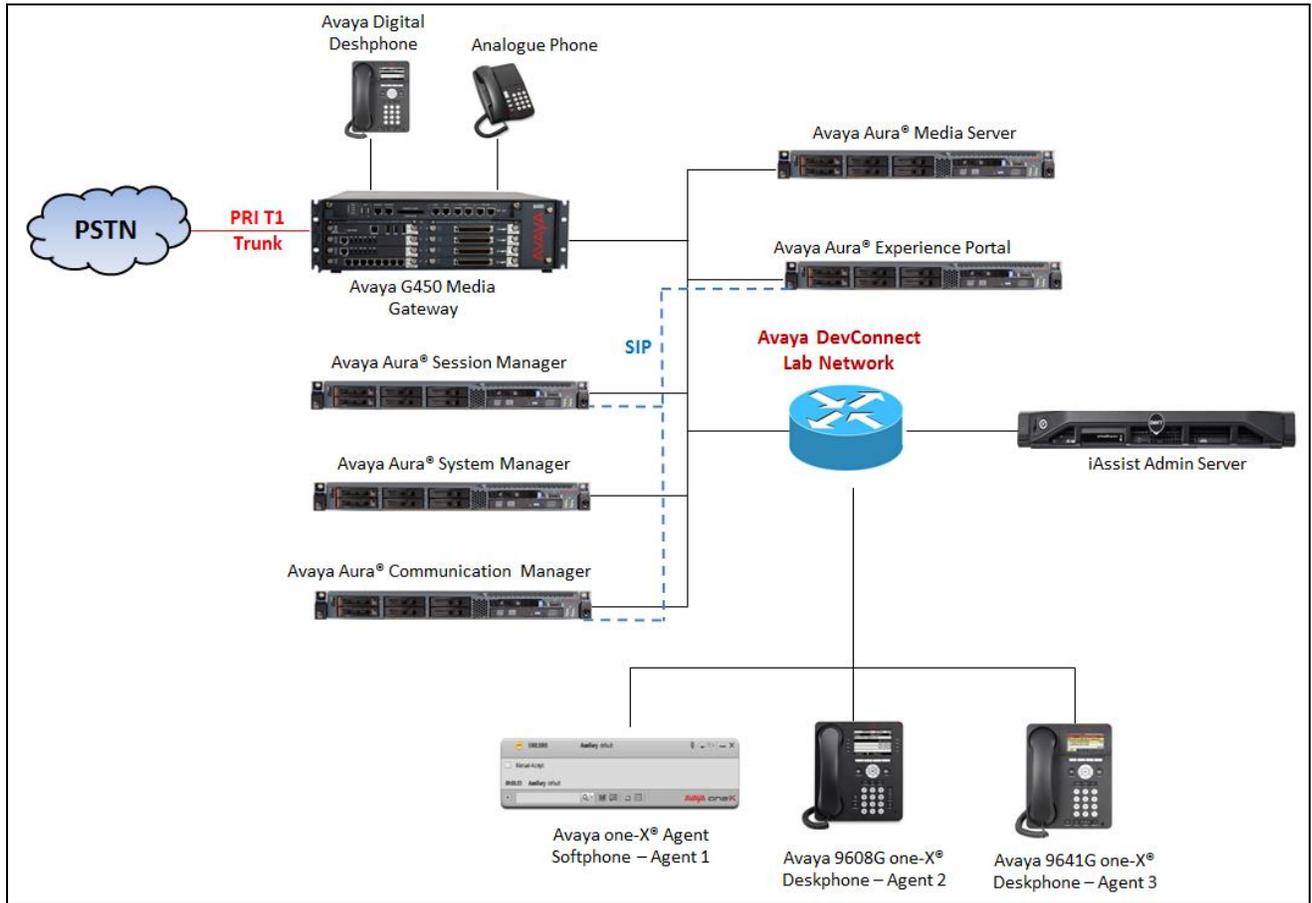
- There is no ring back tone on the agent's phone while the customer's phone ringing for the call back. This is design intent from iAssist Call Back Manager when their application uses Call Control XML from Experience Portal to create a conference call between agent and customer.

## 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                                     Page 1 of 4
                                                    HUNT GROUP
    Group Number: 1                                ACD? y
    Group Name: Skill-1                            Queue? y
    Group Extension: 3320                          Vector? y
    Group Type: ucd-mia
    TN: 1
    COR: 1
    Security Code:                                MM Early Answer? n
    ISDN/SIP Caller Display:                      Local Agent Preference? n
    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                                Page 2 of 4
                                                    HUNT GROUP
    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      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                                     Page 1 of 3
                                         VECTOR DIRECTORY NUMBER

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 UUI 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	<b>49</b>	<b>4</b>	<b>aar</b>			
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 String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI 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.

AVAYA  
Aura System Manager 7.0

Recommended access to System Manager is via FQDN.  
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

User ID:

Password:

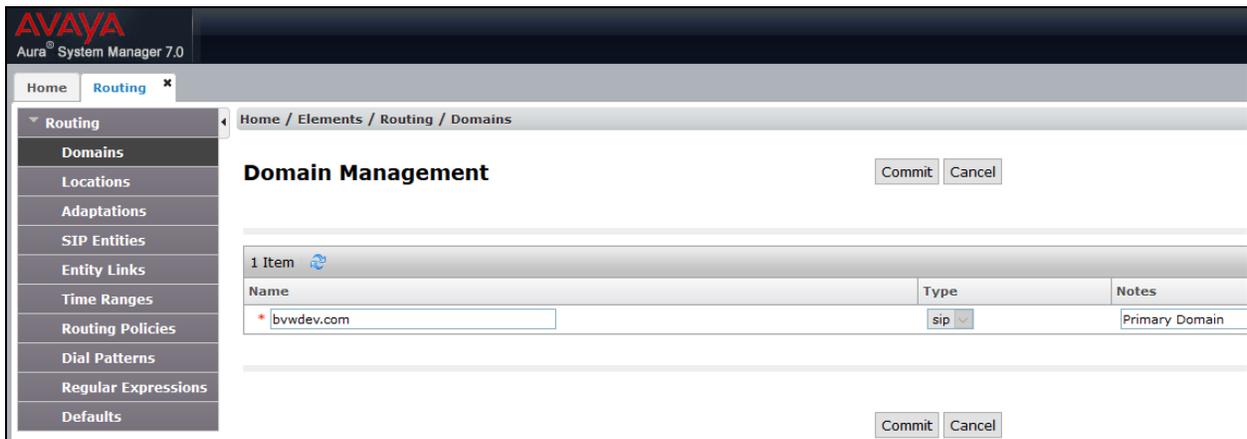
[Change Password](#)

## 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.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The top navigation bar includes the Avaya logo and the text 'Aura System Manager 7.0'. The left sidebar contains a menu with 'Routing' selected, and 'Locations' highlighted. The main content area is titled 'Location Details' and has a 'General' sub-section. The 'Name' field is populated with 'BvwDevSIL' and the 'Notes' field is empty. 'Commit' and 'Cancel' buttons are visible at the top right of the form.

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.

The screenshot shows the 'Location Pattern' sub-section. It features a table with 4 items. The first three rows have IP address patterns: '10.10.5.\*', '10.10.97.\*', and '10.10.98.\*'. The fourth row has a pattern starting with '10.'. There are 'Add' and 'Remove' buttons at the top, and 'Commit' and 'Cancel' buttons at the bottom.

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.10.5.*	
<input type="checkbox"/>	* 10.10.97.*	
<input type="checkbox"/>	* 10.10.98.*	
<input type="checkbox"/>	*	

## 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

The screenshot shows the 'Adaptation Details' configuration page. The 'General' tab is active. The 'Adaptation Name' is 'ChangeFromNumber', the 'Module Name' is 'DigitConversionAdapter', and the 'Module Parameter Type' is 'Name-Value Parameter'. A table below contains one row with 'Name' 'fromto' and 'Value' 'true'. There are also fields for 'Egress URI Parameters' and 'Notes'.

(Continue) the screenshot shows the adaptation.

The screenshot shows the 'Digit Conversion for Outgoing Calls from SM' configuration page. It displays a table with one row: Matching Pattern '\*4905', Min '\*4', Max '\*4', Phone Context, Delete Digits '\*4', Insert Digits '3349', Address to modify 'both', Adaptation Data, and Notes.

## 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 breadcrumb navigation shows 'Home / Elements / Routing / SIP Entities'. The main content area is titled 'SIP Entity Details' and includes a 'General' section with the following fields:

- Name:** \* Name: ASM70A
- FQDN or IP Address:** \* FQDN or IP Address: 10.33.1.12
- Type:** Type: Session Manager (dropdown menu)
- Notes:** Notes: (text input field)
- Location:** Location: BvwDevSIL (dropdown menu)
- Outbound Proxy:** Outbound Proxy: (dropdown menu)
- Time Zone:** Time Zone: America/Toronto (dropdown menu)
- Credential name:** Credential name: (text input field)

Below the General section is the 'SIP Link Monitoring' section with the following field:

- SIP Link Monitoring:** SIP Link Monitoring: Use Session Manager Configuration (dropdown menu)

Buttons for 'Commit' and 'Cancel' are visible in the top right corner of the form area. The left navigation pane shows 'Routing' expanded with 'SIP Entities' selected.

## 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 displays the Avaya Aura System Manager 7.0 interface. The top navigation bar includes the Avaya logo, 'Aura System Manager 7.0', and a 'Last Logged on at May 23, 201' timestamp. The breadcrumb trail is 'Home / Elements / Routing / SIP Entities'. The left-hand navigation pane is expanded to 'SIP Entities'. The main content area is titled 'SIP Entity Details' and contains a 'General' tab. The configuration fields are as follows:

- Name:** ACM-Trunk1-Private
- FQDN or IP Address:** 10.33.1.6
- Type:** CM
- Notes:** (empty)
- Adaptation:** (empty)
- Location:** BvwDevSIL
- Time Zone:** America/Toronto
- SIP Timer B/F (in seconds):** 4
- Credential name:** (empty)
- Securable:**
- Call Detail Recording:** none

Buttons for 'Commit' and 'Cancel' are visible in the top right corner 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 web interface. The top navigation bar includes the Avaya logo, the text "Aura System Manager 7.1", and a search box with "GO...". A notification bubble indicates "1 New important message(s)". The left sidebar shows a tree view with "Routing" selected, and sub-items: 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 includes "Commit" and "Cancel" buttons. The "General" section contains the following fields:

- 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 text box)
- Securable:**
- Call Detail Recording:** none

## 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 with 1 item is displayed. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, DNS Override, and Port. The row contains: Name: \*ASM70\_ACM\_Trunk1\_50, SIP Entity 1: \*ASM70A, Protocol: TLS, Port: \*5061, SIP Entity 2: \*ACM-Trunk1-Private, DNS Override: , Port: \*5061. A 'Filter: Enable' button is in the top right of the table area. Below the table is a 'Select: All, None' dropdown.

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 with 1 item is displayed. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, and Port. The row contains: Name: \*ASM70A\_AEP71\_5060\_T, SIP Entity 1: \*ASM70A, Protocol: TCP, Port: \*5060, SIP Entity 2: \*AEP71, Port: \*5060. A 'Filter: Enable' button is in the top right of the table area. Below the table is a 'Select: All, None' dropdown.

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.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The top navigation bar includes 'Home', 'Session Manager', and 'Routing'. The left sidebar lists various configuration options, with 'Routing Policies' selected. The main content area displays the 'Routing Policy Details' for a policy named 'To-CM-Trunk1'. The 'General' section includes fields for Name, Disabled, Retries, and Notes. The 'SIP Entity as Destination' section features a 'Select' button and a table listing available SIP entities.

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.

The screenshot shows the 'Routing Policy Details' configuration page for a policy named 'To-AEP'. The page is divided into several sections:

- General:** Contains fields for Name (To-AEP), Disabled (checkbox), Retries (0), and Notes (route to EP system 10.33.1.25).
- SIP Entity as Destination:** A table listing destination entities.
- Time of Day:** A table defining the time range for the policy.

**SIP Entity as Destination Table:**

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

**Time of Day Table:**

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

## 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:**

\* **Min:**

\* **Max:**

**Emergency Call:**

**Emergency Priority:**

**Emergency Type:**

**SIP Domain:**

**Notes:**

**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.

The screenshot shows the 'Dial Pattern Details' configuration page. The left sidebar is expanded to 'Dial Patterns'. The main content area is titled 'Dial Pattern Details' and has 'Commit' and 'Cancel' buttons. Under the 'General' section, the following fields are visible:

- \* Pattern:** 33
- \* Min:** 4
- \* Max:** 4
- Emergency Call:**
- Emergency Priority:** 1
- Emergency Type:** (empty)
- SIP Domain:** bwvdev.com
- Notes:** Dial pattern to CM71 from all locations

Below the 'General' section is the 'Originating Locations and Routing Policies' section. It has 'Add' and 'Remove' buttons. A table below shows 1 item:

<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	

At the bottom of the table, it says '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.

The screenshot shows the Avaya Aura® Experience Portal Manager (EPM) web interface. The top navigation bar includes the Avaya logo, user information (Welcome, eadmin), and a red bar with the text "Avaya Aura® Experience Portal 7.1.0 (ExperiencePortal)" and navigation links for Home, Help, and Logoff. The left sidebar contains a navigation menu with categories like User Management, Real-time Monitoring, System Maintenance, System Management, System Configuration, Security, Reports, and Multi-Media Configuration. The main content area displays the title "Avaya Aura® Experience Portal Manager" and a description of the EPM interface. A red warning message indicates the license grace period for Experience Portal will end on Jan 16, 2017 10:46:53 AM PST. Below this, the "Installed Components" section lists Media Processing Platform, Email Service, HTML Service, and SMS Service with brief descriptions.

## 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.

The screenshot displays the Avaya Aura Experience Portal 7.1.0 (ExperiencePortal) interface. The top navigation bar includes the Avaya logo, user information (Welcome, epadmin, Last logged in today at 7:22:48 AM PDT), and navigation links (Home, Help, Logoff). 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 shows the 'Change SIP Connection' page for a connection named 'ASM70'. The page includes a breadcrumb trail: Home > System Configuration > VoIP Connections > Change SIP Connection. The configuration fields are as follows:

- Name: ASM70
- Enable:  Yes  No
- Proxy Transport: TCP
- Proxy Servers:  Proxy Servers  DNS SRV Domain
- Table of Proxy Servers:

Address	Port	Priority	Weight	
10.33.1.12	5060	0	0	Remove

- Additional Proxy Server: (empty)
- Listener Port: 5060
- SIP Domain: bvwdev.com
- P-Asserted-Identity: (empty)
- Maximum Redirection Attempts: 0
- Consultative Transfer:  INVITE with REPLACES  REFER
- SIP Reject Response Code:  ASM (503)  SES (480)  Custom 503
- SIP Timers:

T1:	250	milliseconds
T2:	2000	milliseconds
B and F:	4000	milliseconds

- Call Capacity:

Maximum Simultaneous Calls:	10
-----------------------------	----

## 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 shows the 'Change Application' configuration page in the Avaya Aura Experience Portal 7.1.0. The page title is 'Change Application' and the breadcrumb trail is 'Home > System Configuration > Applications > Change Application'. 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 includes the following configuration options:

- Name:** iAssist\_CBM
- Enable:**  Yes  No
- Type:** VoiceXML
- Reserved SIP Calls:**  None  Minimum  Maximum
- Requested:** [Empty text box]
- URI:**
  - Single  Fail Over  Load Balance
  - VoiceXML URL:** http://10.10.98.2:8080/Inbound\_CBM/Start [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:** [Empty text box] [Add]
  - [List containing 3349] [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 shows the Avaya Aura Experience Portal 7.1.0 configuration interface. The breadcrumb path is: Home > System Configuration > Applications > Change Application. The page title is "Change Application".

Use this page to change the configuration of an application.

Name: IASSIST\_CBM\_OUTBOUND  
Enable:  Yes  No  
Type: CCXML  
Reserved SIP Calls:  None  Minimum  Maximum  
Requested:

URI

Single  Fail Over  Load Balance  
CCXML URL:    
Mutual Certificate Authentication:  Yes  No  
Basic Authentication:  Yes  No

Speech Servers

ASR:  TTS:

Application Launch

Inbound  Inbound Default  Outbound

Speech Parameters ▶  
Reporting Parameters ▶  
Advanced Parameters ▶

### 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.

The screenshot shows the 'EPM Settings' configuration page in the Avaya Aura Experience Portal 7.1.0. The page has a red header with the title 'Avaya Aura® Experience Portal 7.1.0 (ExperiencePortal)' and a 'Home' button. A breadcrumb trail indicates the path: Home > System Configuration > EPM Servers > EPM Settings. The left sidebar contains a navigation tree with categories like User Management, Real-time Monitoring, System Maintenance, System Management, System Configuration, Security, Reports, and Multi-Media Configuration. The main content area is titled 'EPM Settings' and includes a description: 'Use this page to configure system parameters that affect the Experience Portal system.' Below this are several configuration sections: 'Experience Portal Name' (set to 'ExperiencePortal'), 'Number of Application Server Failover Logs' (set to '10'), and 'Commands to Retain in Configuration History' (set to '50'). The 'Resource Alerting Thresholds (%)' section includes 'HTML Units' (80) and 'Disk' settings with 'High Water' (90) and 'Low Water' (80) thresholds. The 'Web Service Authentication' section contains 'Application Reporting' fields for 'User Name' (set to '<Default>'), 'Password', and 'Verify Password'. Below that is the 'Outcall' section with fields for 'User Name' (set to 'outcall'), 'Password', and 'Verify Password'. At the bottom, there is a 'Miscellaneous' section and four buttons: 'Save', 'Apply', 'Cancel', and 'Help'.

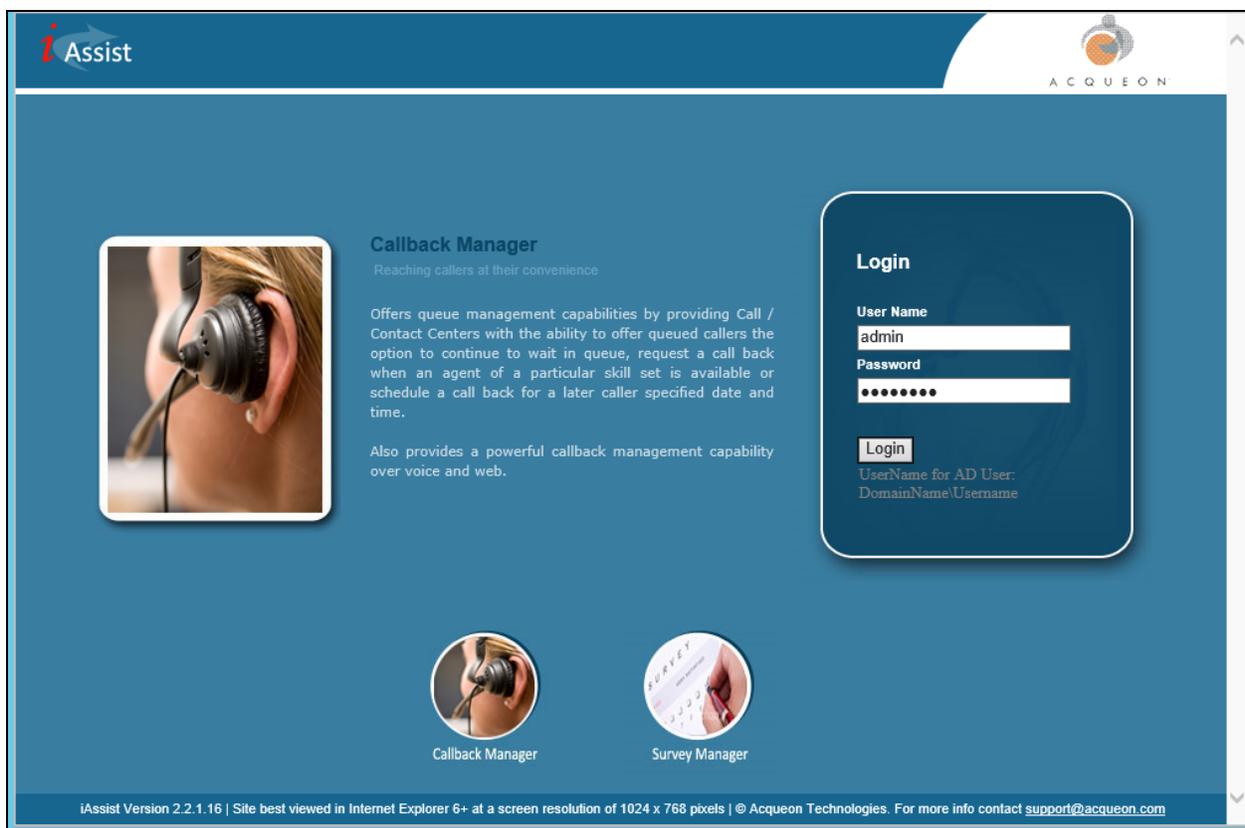
## 7. Configure Acqueon iAssist Call Back Manager

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.



The screenshot displays the Acqueon iAssist admin interface. At the top left is the 'iAssist' logo, and at the top right is the 'ACQUEON' logo. The main content area features a 'Callback Manager' section on the left, which includes a photo of a person wearing a headset and a text description: 'Reaching callers at their convenience. Offers queue management capabilities by providing Call / Contact Centers with the ability to offer queued callers the option to continue to wait in queue, request a call back when an agent of a particular skill set is available or schedule a call back for a later caller specified date and time. Also provides a powerful callback management capability over voice and web.' To the right of this section is a 'Login' form with fields for 'User Name' (containing 'admin') and 'Password' (containing seven dots), a 'Login' button, and a note: 'UserName for AD User: DomainName\Username'. At the bottom of the interface, there are two circular icons: 'Callback Manager' (showing a headset) and 'Survey Manager' (showing a hand holding a survey card). A footer at the very bottom reads: 'iAssist Version 2.2.1.16 | Site best viewed in Internet Explorer 6+ at a screen resolution of 1024 x 768 pixels | © Acqueon Technologies. For more info contact [support@acqueon.com](mailto:support@acqueon.com)'.

## 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 shows the 'BusinessGroup Management' interface. The form on the left has the following fields:

- Business Group Name \***: AACC\_CBM\_3349
- Incoming Number \***: 3349
- Site**: AACC\_Site1 (dropdown)
- Language**: US English (dropdown)
- IVR Configuration Template**: DEFAULT\_CBM\_CONFIG (dropdown)

Buttons: Update Business Group, Cancel

The right pane, titled 'Defined Business Group(s)', contains a table:

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 | Logout

CBM - Business Group Configuration [AACC\_CBM\_3349] \* Mandatory

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:

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

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	
<input type="button" value="Add"/>	

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_CallFlo		
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 \* --SELECT--

VoiceFileName

Value

Add Element

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

Move Up  
Move Down  
Delete  
Delete All

Update CallFlow Cancel

## 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		
<a href="#">EPM / mpp</a>	EPM/MPP	Online	Running	OK	10	10	50	0	0	1	✔
<b>Summary</b>					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
<a href="#">10</a>	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.

- 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.
- 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:  Business Group:   
 From Date:  End Date:   
 Call Status:  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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