



Avaya Solution & Interoperability Test Lab

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

Abstract

These Application Notes describe the configuration steps required to integrate the Acqueon iAssist Call Survey Manager with Avaya Aura® Experience Portal. iAssist Call Survey Manager is a performance and quality assurance application that allows customers (or subscribers) to provide feedback about their call center or product experience. iAssist Call Survey Manager is used to create, add, preview, modify, and remove surveys.

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 Survey Manager with Avaya Aura® Experience Portal. iAssist Call Survey Manager is a performance and quality assurance application that allows customers (or subscribers) to provide feedback about their call center or product experience. iAssist Call Survey Manager is used to create, add, preview, modify, and remove surveys.

iAssist Call Survey Manager (CSM) allows callers to participate in an automated survey on Avaya Aura® Experience Portal. Surveys can be created and assigned dynamically and survey reports can be captured and analyzed. This helps the organization to improve quality, customer and employee satisfaction.

CSM provides flexibility in questionnaire design, storage of caller responses in a database, and generating reports for easy analysis of the survey. CSM allows the creation and design of questionnaires with as many questions as desired. The sequencing of the questions may be dynamic based on caller's previous answer selection. Typical surveys cover caller experience on their interaction with agents and product feedback.

Another Acqueon related solution is described in Application Notes for Acqueon iAssist Call Back Manager with Avaya Aura® Experience Portal. This is mentioned here since CSM and Call Back Manager resides on the same server and Cal Back Manager will be mentioned in this Application notes.

2. General Test Approach and Test Results

This section describes the interoperability compliance testing used to verify the iAssist Call Survey Manager (CSM) 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 CSM application to allow the caller to provide agent or product feedback. After the survey was completed, a survey report was generated to review the survey responses. In addition, it was verified that the CSM application handled error conditions, such as entering an invalid response; properly. iAssist CSM has the ability to place outbound calls to PSTN lines for surveys.

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.

For the testing associated with these Application Notes, the interface between Avaya systems utilized enabled capabilities of TLS/SRTP but the Acqueon Call Survey Manager did not include use of any specific encryption features as requested by Acqueon.

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 and running iAssist CSM.
- The ability of the caller to complete the survey successfully.
- Verifying the correct response to invalid entries by the caller.
- Routing outbound calls to PSTN lines via Avaya Aura Communication Manager.

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. Avaya Aura® Experience Portal was successful in running the iAssist CSM applications.

2.3. Support

For technical support on the iAssist Call Survey 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>

3. Reference Configuration

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 hosted the iAssist CSM applications supporting the CSM inbound and modules. The Acqueon iAssist Admin server contained the Microsoft SQL database and also was used to configure the iAssist CSM application.

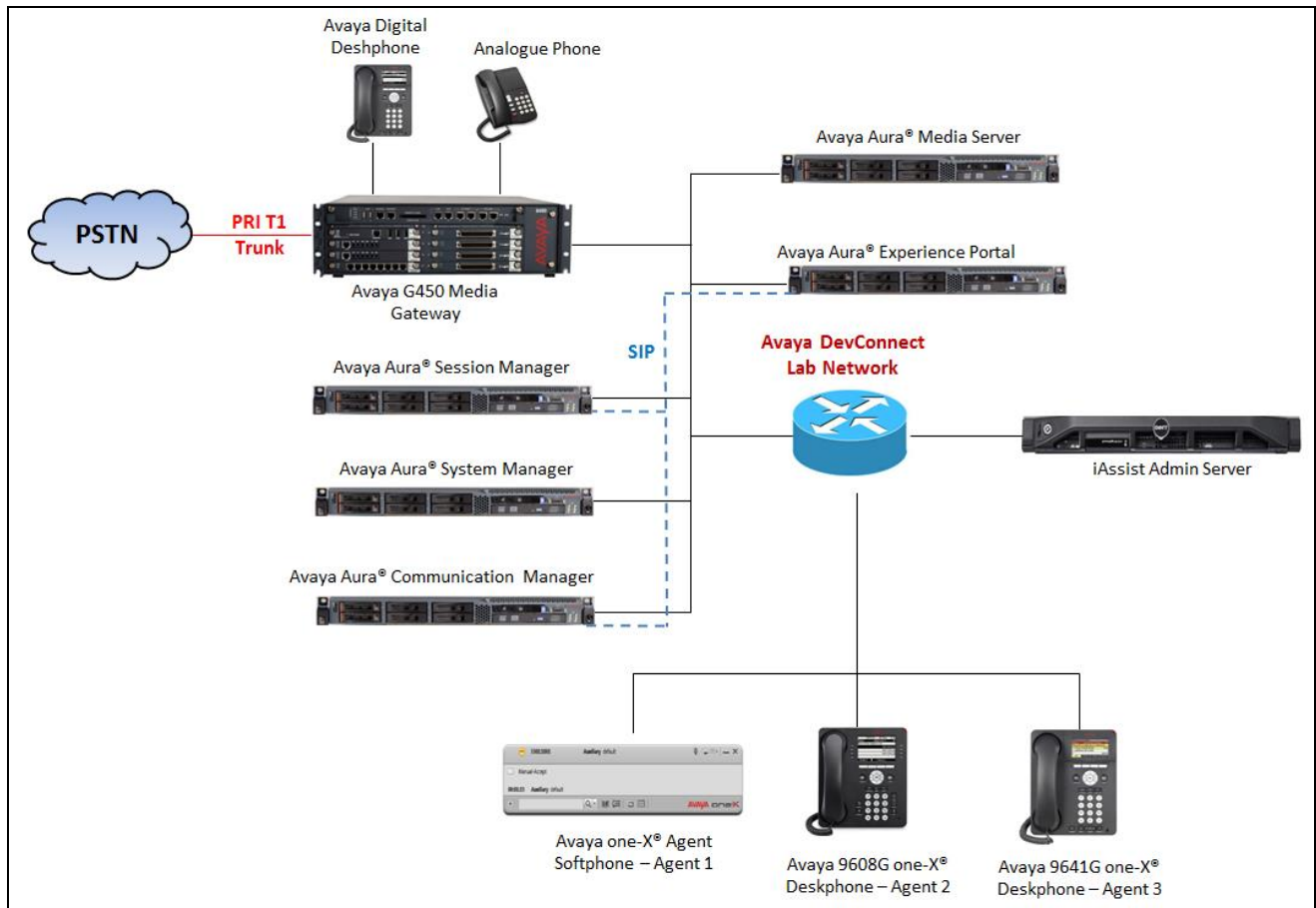


Figure 1: Test Configuration Diagram

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager 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 9641G H323 IP Deskphone	6.6.4
Avaya 9608G SIP IP Deskphone	7.1.29
Acqueon iAssist Callback Manager application running on Windows Server 2012	2.2.1.6

5. 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 SIP Signaling
- Administer SIP Trunk
- Administer Route Pattern
- Administer Dial Plan
- Administer AAR table

5.1. 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:** Enter port TLS 5061.
- **Far-end Listen Port:** The same port number as in **Near-end Listen Port**.
- **Far-end Network Region:** Enter an existing network region
- **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		
H.323 Station Outgoing Direct Media? n	Alternate Route Timer(sec): 6		

5.2. 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 5.1**.
- **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	Night Service:	
Dial Access? n			
Queue Length: 0			
Service Type: tie	Auth Code? n		
Member Assignment Method: auto			
Signaling Group: 1			
Number of Members: 14			

5.3. 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 5.2**
- **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	

5.4. 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			

5.5. 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 5.3**

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

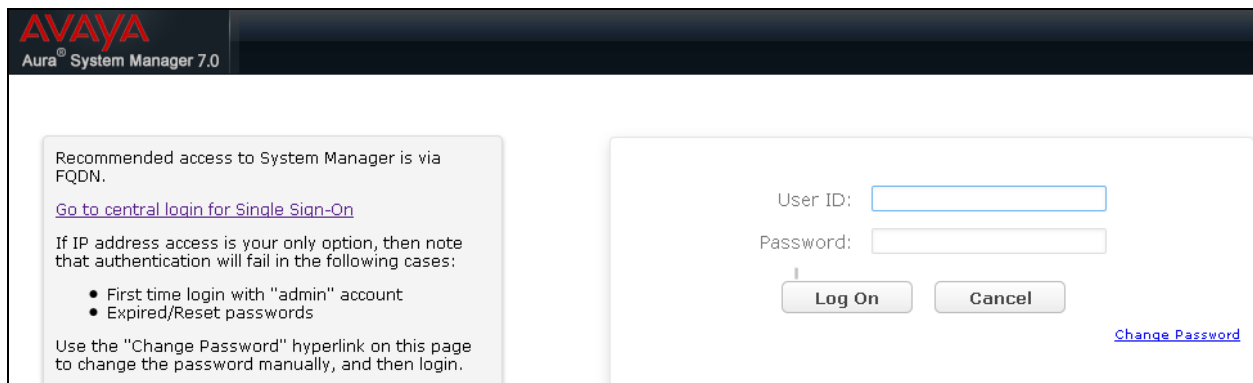
6. Configure Avaya Aura® Session Manager

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

6.1. Launch System Manager

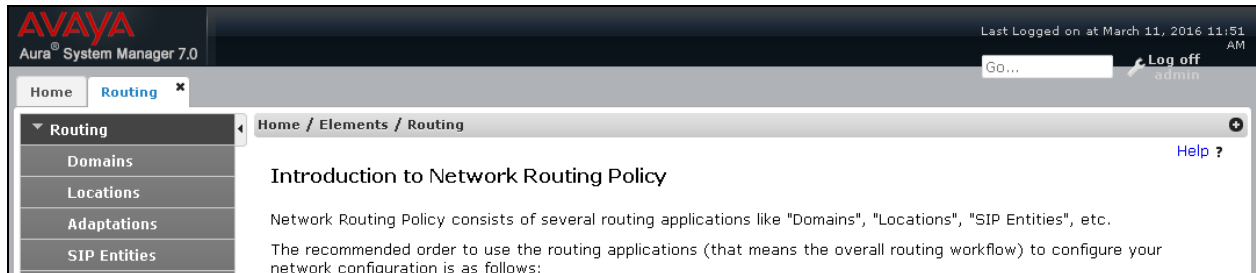
Access the System Manager web interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of System Manager. Log in using the appropriate credentials.



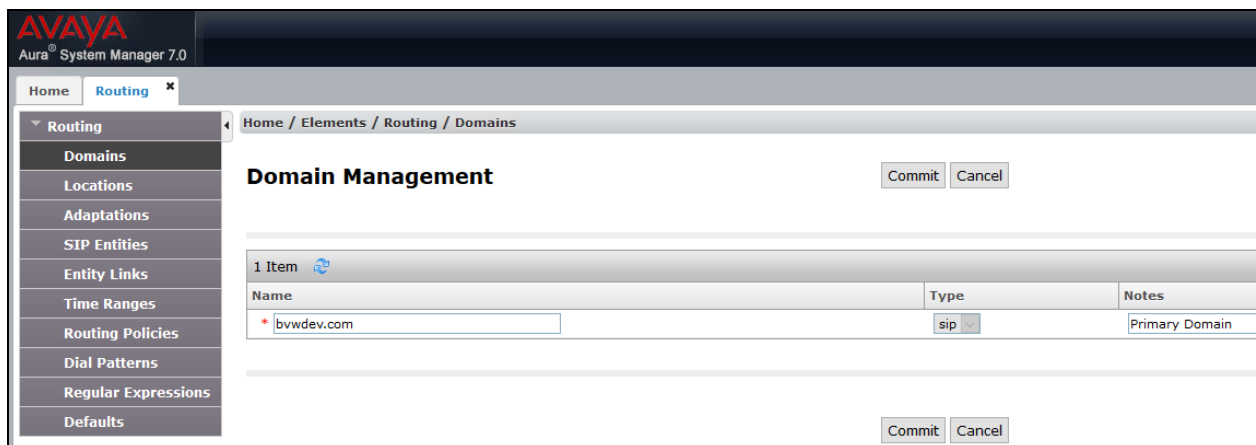
The screenshot shows the Avaya Aura System Manager 7.0 login interface. The header features the Avaya logo and the text "Aura® System Manager 7.0". The main content area is divided into two sections. The left section contains a message: "Recommended access to System Manager is via FQDN." followed by 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.

6.2. 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**.



6.3. Administer Locations

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

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

<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"/>	*	

Select : All, None

Commit Cancel

6.4. Administer Adaptation

The adaptation below was used for iAssist Callback Manager and was not applied for Call Survey Manager since the calling numbers 4905 was configured in Experience Portal to launch the CSM application.

- **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

Name	Value
fromto	true

Select : All, None

Egress URI Parameters:

Notes:

(Continue) the screenshot to show the **Digit Conversion for Outgoing Calls from SM** section.

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

6.5. 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.

6.5.1. 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 6.3**.
- **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'. The left-hand navigation pane lists various configuration options: Domains, Locations, Adaptations, SIP Entities (which is 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) is set to 'ASM70A'; 'FQDN or IP Address' (with a red asterisk) is set to '10.33.1.12'; 'Type' is a dropdown menu set to 'Session Manager'; 'Notes' is an empty text area; 'Location' is a dropdown menu set to 'BvwDevSIL'; 'Outbound Proxy' is an empty text field; 'Time Zone' is a dropdown menu set to 'America/Toronto'; and 'Credential name' is 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.

6.5.2. 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.

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.

AVAYA
Aura® System Manager 7.0

Last Logged on at May 23, 2017

Home / Elements / Routing / SIP Entities

SIP Entity Details

Commit Cancel

General

* Name: ACM-Trunk1-Private

* FQDN or IP Address: 10.33.1.6

Type: CM

Notes:

Adaptation:

Location: BywDevSIL

Time Zone: America/Toronto

* SIP Timer B/F (in seconds): 4

Credential name:

Securable: ☐

Call Detail Recording: none

6.5.3. 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 top navigation bar shows 'Home' and 'Routing' (selected). The left-hand menu lists various configuration options, with 'SIP Entities' highlighted under the 'Routing' section. 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.

6.6. 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 Experience Portal. 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). Configure the following values:

- **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** (not shown) 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'. The page title is 'Entity Links'. There are 'Commit' and 'Cancel' buttons. Below the title is a table with 1 item. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, DNS Override, and Port. The data row shows: Name: ASM70_ACM_Trunk1_50, SIP Entity 1: ASM70A, Protocol: TLS, Port: 5061, SIP Entity 2: ACM-Trunk1-Private, DNS Override: (unchecked), Port: 5061. There is a 'Filter: Enable' button and a 'Select: All, None' dropdown at the bottom.

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'. The page title is 'Entity Links'. There are 'Commit' and 'Cancel' buttons. Below the title is a table with 1 item. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, and Port. The data row shows: Name: ASM70A_AEP71_5060_T, SIP Entity 1: ASM70A, Protocol: TCP, Port: 5060, SIP Entity 2: AEP71, Port: 5060. There is a 'Filter: Enable' button and a 'Select: All, None' dropdown at the bottom.

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

6.7. Administer Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.5**. 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 screen shows the Routing Policy for Communication Manager.

The screenshot displays the Avaya Aura System Manager 7.0 web interface. The top navigation bar includes the Avaya logo, 'Aura System Manager 7.0', and a 'Last Logged on at May 23, 2017' timestamp. The left sidebar shows a tree view with 'Routing' expanded, listing 'Domains', 'Locations', 'Adaptations', 'SIP Entities', 'Entity Links', 'Time Ranges', 'Routing Policies' (selected), 'Dial Patterns', 'Regular Expressions', and 'Defaults'. The main content area is titled 'Routing Policy Details' and contains two sections: 'General' and 'SIP Entity as Destination'. The 'General' section has fields for 'Name' (To-CM-Trunk1), 'Disabled' (checkbox), 'Retries' (0), and 'Notes'. The 'SIP Entity as Destination' section has 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 screen shows 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

Commit

Cancel

Help ?

Routing Policy Details

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

Add

Remove

View Gaps/Overlaps

1 Item

Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/> 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

6.8. 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.

6.8.1. 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 Trio Enterprise. 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 signaling group 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 6.7** 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: bvwdev.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 Policy Notes
<input type="checkbox"/>	-ALL-		To-AEP	0	<input type="checkbox"/>	AEP71	route system 10.33

6.8.2. 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 signaling group 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 6.7** 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

7. 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 EPM. Log in using the appropriate credentials.

Note: Some of the screens in this section are shown after the Experience Portal had been configured. Ensure to save the screen parameters as you configure Avaya Aura® Experience Portal.

The screenshot displays the Avaya Aura® Experience Portal Manager (EPM) web interface. The top header features the Avaya logo on the left and the text "Welcome, eadmin" and "Last logged in today at 1:21:13 PM PST" on the right. Below the header is a red navigation bar with the text "Avaya Aura® Experience Portal 7.1.0 (ExperiencePortal)" and links for "Home", "Help", and "Logoff". The left sidebar contains a list of navigation links: "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", and "Multi-Media Configuration". The main content area shows the "You are here: Home" breadcrumb and the "Avaya Aura® Experience Portal Manager" section. This section includes a description of the EPM interface and a red warning message: "License grace period for Experience Portal will end on Jan 16, 2017 10:46:53 AM PST." Below this is the "Installed Components" section, which lists "Media Processing Platform", "Email Service", "HTML Service", and "SMS Service" with brief descriptions of each.

7.1. Administer VoIP Connection

On the left pane, click on the **VoIP Connections** under **System Configuration** (not shown). To add a **SIP Connection**, click on the **SIP** tab on **VoIP Connections** page (not shown).

- **Name:** Enter a descriptive name.
- **Enable:** Select “Yes” radio button.
- **Proxy Transport:** Select “TCP” if SIP connection uses TCP.
- **Proxy Servers:** Enter the SIP signaling IP address of Session Manager.
- **SIP Domain:** Enter a SIP domain “bvwddev.com” as configured in **Section 7.3**.
- In the **Call Capacity** section, enter a number of SIP call that in the **Maximum Simultaneous Calls** and select “All call can be either inbound or outbound” option. All other fields can be left at default.

Click **Save** (not shown) 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

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: ASM70

Enable: ☒ Yes ☐ No

Proxy Transport: TCP

☒ Proxy Servers ☐ DNS SRV Domain

Address	Port	Priority	Weight	
10.33.1.12	5060	0	0	Remove

[Additional Proxy Server](#)

Listener Port: 5060

SIP Domain: bvwddev.com

P-Asserted-Identity:

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

7.2. Configure iAssist CSM Applications

CSM application is configured in Avaya Aura® Experience Portal to handle inbound calls that launch the CSM application in the iAssist server.

7.2.1. Configure the Inbound CSM Application

Navigate to **System Configuration → Applications**. In the **Applications** page, add an Experience Portal application to handle incoming calls. Configure the application as shown below.

- **Name:** Enter a descriptive name, e.g. iAssist_CSM.
- **Enable:** Select **Yes**.
- **Type:** Select VoiceXML from the dropdown menu.
- **URI:** Select **Single** radio button and enter a CSM link in the VoiceXML URL.
- **Application Launch:** Select **Inbound** and add the number **4906** as configured in **Section 6.8.1.**

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

Change Application

Use this page to change the configuration of an application.

Name: iAssist_CSM

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: TTS:

Application Launch

☒ Inbound ☐ Inbound Default ☐ Outbound

☒ Number ☐ Number Range ☐ URI

Called Number:

Add

4906

Remove

7.3. 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 which is a sub section of **Web Service Authentication**, 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

▼ 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](#) > [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:

High Water

Low Water

Disk:

Web Service Authentication ▼

Application Reporting

User Name:

Password:

Verify Password:

Outcall

User Name:

Password:

Verify Password:

Miscellaneous ▶

Save

Apply

Cancel

Help

KP; Reviewed:
SPOC 11/8/2017

Solution & Interoperability Test Lab Application Notes
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iAssist-AEP71

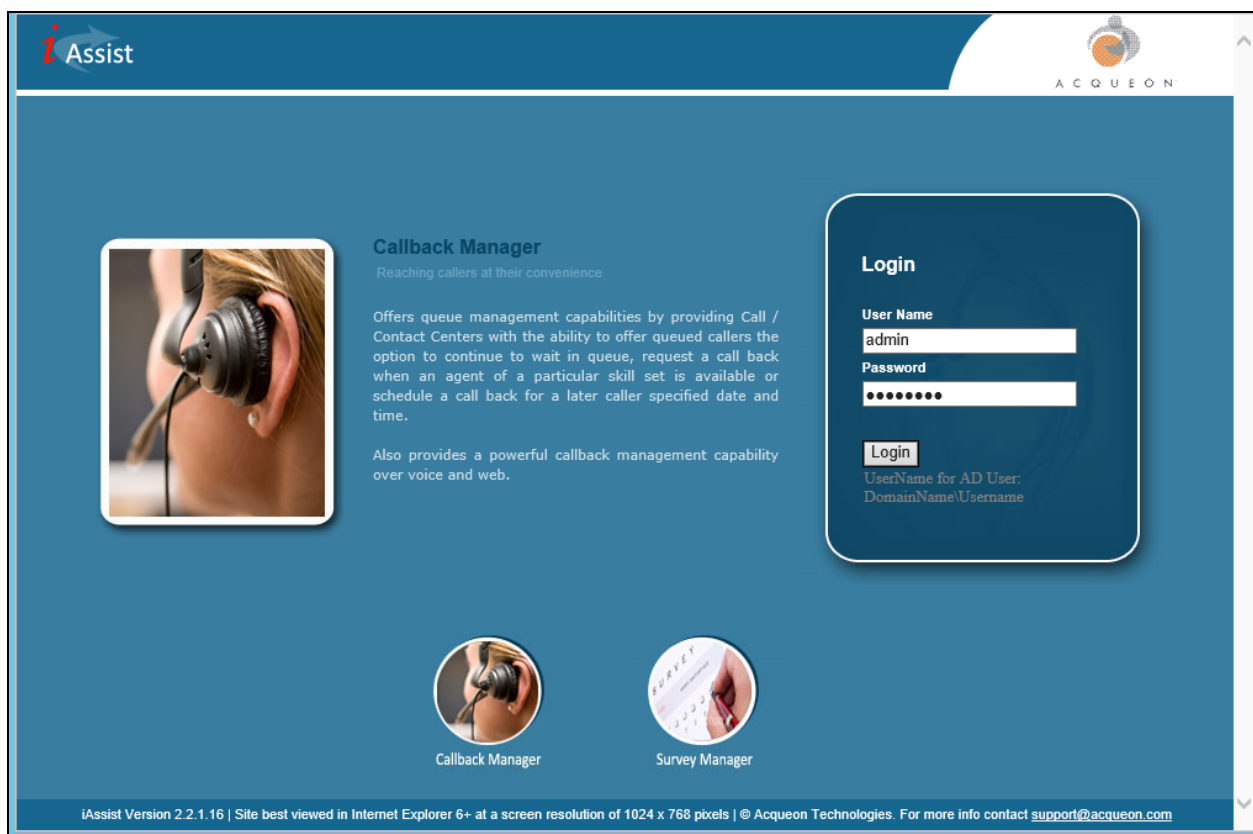
8. Configure Acqueon iAssist Call Survey Manager

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

8.1. 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 **Password**.

Note: The current version of iAssist Call Survey Manager only supports Microsoft Internet Explorer.



8.2. 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., 4906).
- 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 the iAssist logo, a menu with 'Home', 'Manage', 'General', 'CBM', 'CSM', and 'License', and a user status area showing 'Welcome admin | Logout'.

The main content area is divided into two sections:

- BusinessGroup Management:** This section contains a form for configuring a business group. The form fields are:
 - Business Group Name ***: A text input field containing 'AACC_CSM_4906'.
 - Incoming Number ***: A text input field containing '4906'.
 - Site**: A dropdown menu with 'AACC_Site1' selected.
 - Language**: A dropdown menu with 'US English' selected.
 - IVR Configuration Template**: A dropdown menu with 'DEFAULT_CSM_CONFIG' selected.
 Below the form are two buttons: 'Update Business Group' and 'Cancel'. A red asterisk and the word 'Mandatory' are positioned to the right of the form fields.
- Defined Business Group(s):** This section contains a table listing existing business groups.

Business Group	Edit	Delete
AACC_CBM_3349		
AACC_CSM_4906		

8.3. Configuring Business Group

From the menu, select the **CSM → 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 a valid **Business Group Name**.
- **IVR IP Address** [Voice Portal Management System's (VPMS) IP that has been used for dialing the agent and/ or customer].
- Time Zone (not shown) (Time zone of system in which iAssist application is deployed).
- Enter **Customer Timeout** in second.

CSM - Business Group Configurations [AACC_CSM_4906]

Mandatory

Business Group Name	AACC_CSM_4906
IVR IP address *	10.33.1.25
Customer Timeout*	30

Defined Business Group(s)

Business Group	Edit
AACC_CSM_4906	

8.4. 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.

CSM - Business Group Configurations [AACC_CSM_4906]

Business Hour and Break Hour

	Business Hour [24 Hrs Format]		Break Hour [24 Hrs Format]	
Monday	09:00	18:00	00:00	00:00
Tuesday	09:00	18:00	00:00	00:00
Wednesday	09:00	18:00	00:00	00:00
Thursday	09:00	18:00	00:00	00:00
Friday	09:00	18:00	00:00	00:00
Saturday	09:00	18:00	00:00	00:00
Sunday	09:00	18:00	00:00	00:00

Defined Business Group(s)

Business Group	Edit
AACC_CSM_4906	

8.5. Config Options

Under the **Config Options** menu, the description of the outcome, its reschedule value and number of retries are given. To edit a defined config option:

- Select the Edit icon corresponding to the business group whose configurations are to be modified.
- Enter the desired rescheduled value in minutes under the **Rescheduled Value** field.
- Enter **Number Of Retries** to be made against the outcomes. (By default the value will be 3 retries).
- Select the **Leave Message** option to leave a message if the call is forwarded to voice mail.
- Select the **Close if Answering Machine** box if the call is diverted to the answering machine.
- By enabling this option, the contact will be closed without further retries

Config Options

Mandatory

Outcome Description	Reschedule Value	Number Of Retries
Busy *	<input type="text" value="30"/>	<input type="text" value="3"/>
Not Reachable *	<input type="text" value="30"/>	<input type="text" value="3"/>
No Response *	<input type="text" value="30"/>	<input type="text" value="3"/>
Default *	<input type="text" value="30"/>	<input type="text" value="3"/>
Maximum Retries *	<input type="text" value="3"/>	
	<input type="checkbox"/> Leave Message	<input type="checkbox"/> Close if Answering Machine

Update Business Group

Cancel

8.6. Create Call Survey

To configure the survey, in the iAssist admin home screen, click CSM Survey.

- Enter a **Survey Name**.
- Select a **Survey Type** from the dropdown menu.
- Select a **BusinessGroupName** entitled to handle the survey from the dropdown menu.
- Enter the **Active Start Time** in MM/DD/YYYY HH:MM:SS format. This is the survey start time.
- Enter the **Active End Time** in MM/DD/YYYY HH:MM:SS format. This is the survey end time.

Two types of surveys can be configured in the CSM application.

- An IVRSurvey is a survey of a product, a service, an issue, etc.
- An Agent Survey is an assessment of the performance of an agent attending a customer call.
- However, at any given time, only one survey can be hosted for one business group

The screenshot displays the iAssist admin interface. The top navigation bar includes links for Home, Manage, General, CBM, CSM, and License, along with a user welcome message and a Logout link. The main content area is divided into two sections: 'Create New Survey' and 'Defined Survey'.

The 'Create New Survey' section contains a form with the following fields:

- SurveyName ***: Text input field containing 'AACCLabTest'.
- SurveyType ***: Dropdown menu showing 'AgentSurvey'.
- BusinessGroupName ***: Dropdown menu showing 'AACC_CSM_4906'.
- Active Start Time (MM/DD/YYYY) ***: Text input field containing '7/24/2017'.
- Active End Time (MM/DD/YYYY) ***: Text input field containing '11/30/2017'.

At the bottom of the form are 'Update Survey' and 'Cancel' buttons. A red asterisk and the word 'Mandatory' are positioned above the form fields.

The 'Defined Survey' section displays a table with the following data:

Survey	BusinessGroup	Edit	Delete
AACCLabTest	AACC_CSM_4906		

8.7. Create Survey Questions

- Enter a name for the question in the **Question** field. For e.g. IVRSatisfaction, Agent Performance etc.
- Enter the audio file name of the question in the **Question File Name** field. This is the file that will be played out to the caller as a question.
- Select the **Question Type** from the dropdown menu. There are four Question Types available for selection:

Yes/ No – The answer to this question can only be Yes or No, to be selected by the caller by pressing appropriate keys as prompted.

Choice – The answer to this question will be played out as multiple choices, to be selected by the caller by pressing the appropriate keys as prompted. Select the number of choices to be offered to the caller from the dropdown menu along side the Question type dropdown menu. The minimum number of Choices offered to the caller is 3 and the maximum number is 5.

Number – The answer to this question will be a number. The caller has to press the appropriate number on the telephone keypad. Enter the Minimum Digits and Maximum Digits fields to complete selection of this type of question.

Date – The answer to this question will be date. The caller has to press the appropriate numbers on the telephone keypad in the format as prompted by the application. Select the date format as MM/DD/YYYY or DD/MM/YYYY from the dropdown alongside the Question Type menu.

The screenshot displays the iAssist application interface. The top navigation bar includes the iAssist logo, a menu with 'Home', 'Manage', 'General', 'CBM', 'CSM', and 'License', and a user greeting 'Welcome admin | Logout'. The main content area is divided into two panels. The left panel, titled 'Create Survey Questions', contains a form with three mandatory fields: 'Question', 'Question File Name', and 'Question Type' (a dropdown menu currently showing 'YesNo'). Below these fields is a 'Create Survey Question' button. Underneath, there is a 'Bulk Upload' section with a 'Question File' input, a 'Browse...' button, and an 'Upload' button. The right panel, titled 'Defined Survey Questions', contains a table with three columns: 'Questions', 'Edit', and 'Delete'. The table lists eight questions (question1 through question8), each with an edit icon (pencil) and a delete icon (X).

Questions	Edit	Delete
question1		
question10		
question2		
question3		
question4		
question5		
question6		
question7		
question8		

8.8. Select Questions

A number of survey questions are included in the application. However, all survey questions need not be part of a specific survey; some survey questions may be skipped. The survey questions for a specific survey can be selected. This will result in only the selected survey questions being played out to the caller; the other questions are simply ignored by the application.

Using this menu you can do the following:

- Select various questions for the survey.
- Arrange the questions in the desired sequence.
- Dynamic routing of the questions, and Enabling/disabling record option for each question

The screenshot displays a web application interface with a top navigation bar containing links: Home, Manage, General, CBM, CSM, License. On the right of the bar, it says 'Welcome admin | Logout'.

The main content area is divided into two panels:

- Select Questions:** This panel is titled 'AACCLabTest'. It contains a table with the following columns: 'Select', 'Question', 'Question Type', and 'Allow recording'. The table lists 10 questions, all of which are selected (checked in the 'Select' column). The 'Allow recording' column has checkboxes, all of which are currently unchecked.
- Defined Survey:** This panel contains a table with three columns: 'Survey', 'Edit', and 'Remove Mapping'. The first row shows 'AACCLabTest' under the 'Survey' column, with an edit icon (pencil) under 'Edit' and a remove icon (X) under 'Remove Mapping'.

At the bottom of the 'Select Questions' panel, there are four buttons: '< Back', 'Next >', 'Finish >>|', and 'Cancel'.

8.9. Call Flow Generating

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

- Specify a **CallFlow Name**.
- Select the required **Site**.
- Select the desired application from the drop down list in the **Application** field.
- Select the **FilterType**.
- Select a **Business Group**.

Call Flow Generator

CallFlow Name *

CSM_Inbound_CallFlow

Site *

AACC_Site1

Appilication

CSM - Inbound

FilterType *

By Business Group Collection

By BusinessGroupID

Business Group *

Select All

AACC_CSM_4906

Mandatory

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

Defined Elements

Use Template

☐

Element Name *

--SELECT--

VoiceFileName

Value

Add Element

Welcome | - | -
DIRECT_TO_SURVEY | - | -
Thank You | - | -

Move Up

Move Down

Delete

Delete All


Update CallFlow

Cancel

9. Verification Steps

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



1. From the EPM web interface, verify that the Media Processing Platform (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 Report** page shown below.

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

Port Distribution Report (Aug 22, 2017 1:56:51 PM PDT)  [Refresh](#)

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	

- Place a call to 4906 which is the number configured for CSM application in Experience Portal. Navigate to **Real-Time Monitoring → Active Calls** to check the active calls being handled by Experience Portal.

You are here: Home > Real-Time Monitoring > Active Calls Report										
Active Calls Report (Aug 27, 2017 3:37:44 PM PDT)										 Refresh
This page displays the status of the active calls being handled by the servers.										
Total Calls: 1					Last Poll: Aug 27, 2017 3:37:34 PM PDT					
Port	Port Group	Protocol	Call Type	MPP Server	Start Time	Calling Number/URI	Called Number/URI	Application	ASR Server	TTS Server
1	ASM70	SIP_Trunk	Inbound	mpp	Aug 27, 2017 3:37:32 PM PDT	sip:3406@bvwdev.com	sip:4906@bvwdev.com	iAssist_CSM		

10. Conclusion

These Application Notes describe the configuration steps required to integrate the Acqueon iAssist Call Survey Manager application with Avaya Aura® Experience Portal. All feature and serviceability test cases were completed successfully with any observations detailed in **Section 2.2**.

11. 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 7.1.1, Document 03-300509, Issue 2, Aug 2017
- [2] Administering Avaya Aura® Session Manager, Release 7.1.1, Issue 2, Aug 2017
- [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 CSM 2.0 Admin Guide
- [5] iAssist CSM 2.0 IVR Installation Guide

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