



## Avaya Solution & Interoperability Test Lab

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# Application Notes for Versay CUE Analytics with Avaya Aura® Experience Portal Release 7.1 - Issue 1.0

### Abstract

These Application Notes describe the configuration steps required to integrate Versay CUE Analytics with Avaya Aura® Experience Portal.

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

The objective of compliance test was to validate interoperability of Versay CUE Analytics solution with Avaya Aura® Experience Portal (Experience Portal).

Versay CUE is a real-time analytics tool designed to surface the customer experience. In the compliance test, CUE Analytics captures application events in the Experience Portal, which are aggregated, analyzed by the CUE Analytics Contextual Understanding Engine and published in real-time to a secure dashboard and mobile (iOS) application. Currently, the customer applications are IVR's, but the idea is that CUE Analytics can surface data from any type of customer experience (website, mobile app, IVR). During the compliance test, the CUE Analytics system is installed in the cloud.

## 2. General Test Approach and Test Results

General test approach was to verify interoperability of the Versay CUE Analytics solution with Avaya Aura® Experience Portal.

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 this Application Note, the interface between Avaya systems and Versay CUE Analytics did not include use of any specific encryption features as requested by Versay.

This test was conducted in a lab environment simulating a basic customer enterprise network environment. The testing focused on the standards-based interface between the Avaya solution and the third party solution. The results of testing are therefore considered to be applicable to either a premise-based deployment or to a hosted or cloud deployment where some elements of the third party solution may reside beyond the boundaries of the enterprise network, or at a different physical location from the Avaya components.

Readers should be aware that network behaviors (e.g. jitter, packet loss, delay, speed, etc.) can vary significantly from one location to another, and may affect the reliability or performance of the overall solution. Different network elements (e.g. session border controllers, soft switches, firewalls, NAT appliances, etc.) can also affect how the solution performs.

If a customer is considering implementation of this solution in a cloud environment, the customer should evaluate and discuss the network characteristics with their cloud service provider and network organizations, and evaluate if the solution is viable to be deployed in the cloud.

The network characteristics required to support this solution are outside the scope of these Application Notes. Readers should consult the appropriate Avaya and third party documentation for the product network requirements. Avaya makes no guarantee that this solution will work in all potential deployment configurations.

## 2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability tests. Feature tests focused on the ability of CUE Analytics to successfully receive application and call events from Experience Portal through the application server and publish in real-time to a secure dashboard. Following widgets in the CUE dashboard were verified:

- CUE Now Screen – Calls widget: shows the number of active callers currently in the system.
- CUE Now Screen – Timeline widget: shows the active callers in the system by 5-minute intervals for the time period selected. A vertical line is drawn noting the current time.
- CUE Now Screen – Callers widget: shows current callers and repeat callers. The Active tab will show the current callers in the system by ANI with the most recent call being at the top.
- CUE Now Screen – Engagement widget: provides real-time updates for three main averages for the active callers in the system and provides an indication as to how the current callers are tracking as compared to the average for callers in the selected time period.
- CUE Trends Screen – Call Volume widget: plots total number of calls for a specific time period.
- CUE Trends Screen – Dialogs widget: displays information about each dialog state traversed in the application in 2 tabs for the time period selected: Visits - the number of times each dialog state has been traversed by a caller. Duration - the time spent in each dialog state. This widget will update after a call has ended.
- CUE Trends Screen – Recognitions widget: displays the number of times callers traversed each dialog state (Visits) and the percentage of responses that were In Grammar, No Match and No Inputs.
- CUE Trends Screen – Call Outcomes widget: provide two types of view, Total percentage and Over Time. The Over Time view allows for up to three outcomes to be graphed. The Total percentage View is designed to summarize and tabulated call outcomes for a given time period.

- CUE Trends Screen – Hang ups widget: summarize the total number of hangups in the application for the time period selected, what percentage of the total calls were hangups and how many of the hang ups were "short calls" as determined by the threshold.
- CUE Trends Screen – Events widget: displays custom events and metrics that were identified by customer as important to track in CUE.
- CUE Trends Screen – Locations widget: provides a count and percentage of calls by state/country for the time period selected. Location is determined by the 3-digit area code.

Serviceability testing focused on verifying the ability of CUE system to recover from adverse conditions, such as restart, power failures and network disconnects.

## **2.2. Test Results**

All test cases were executed and passed.

## **2.3. Support**

For support on the CUE Analytics Solution visit the corporate Web page at:

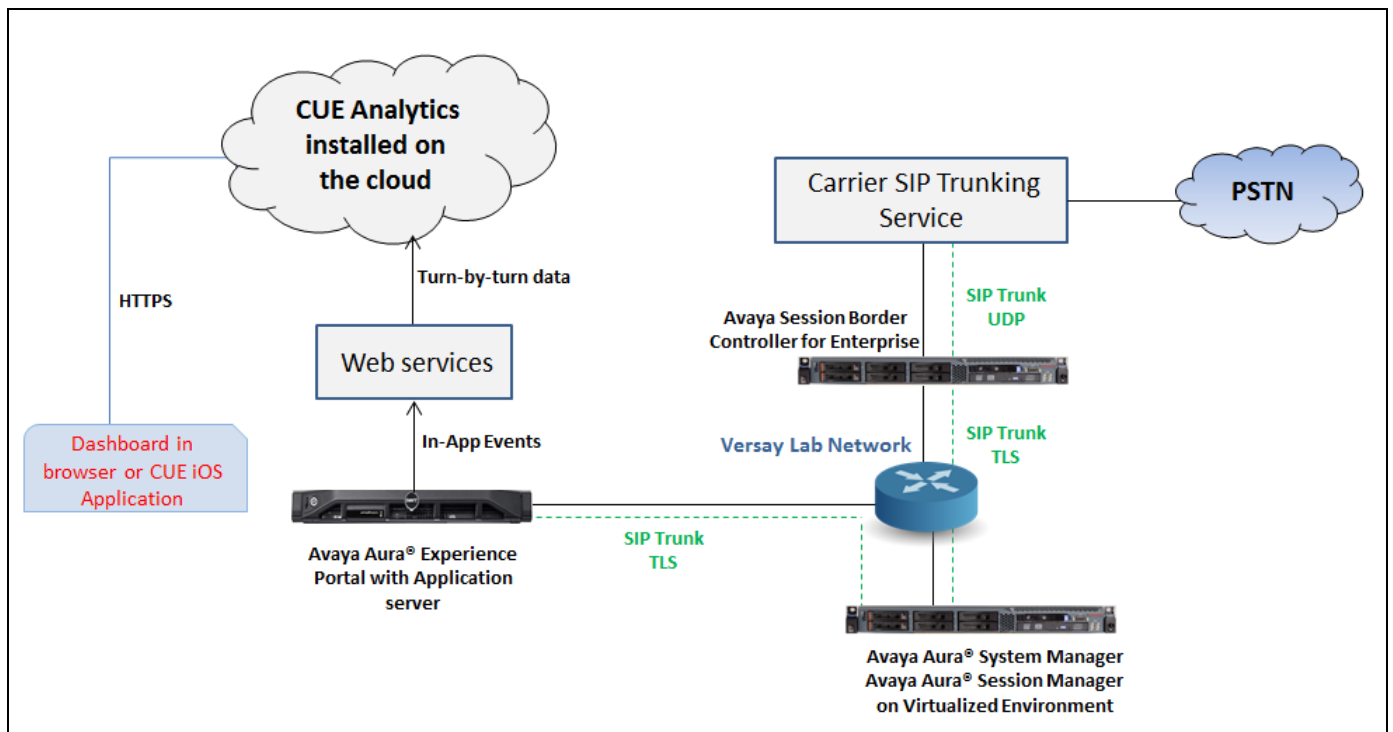
<http://www.versay.com/solutions-1/> or call (888) 869-0121 or send email to sales@versay.com.

### 3. Reference Configuration

**Figure 1** illustrates the sample configuration of how the CUE Analytics solution interacts with Avaya Aura® Experience Portal. For inbound calls, the calls are routed from the service provider to the Avaya SBCE then to Session Manager. Session Manager used the configured dial patterns (or regular expressions) and routing policy to determine the recipient (in this case Experience Portal) and on which link to send the call. Once the call arrived at Experience Portal, further incoming call treatment, such as executing VXML script from the application server was performed.

The Avaya components used to create the simulated enterprise customer site included:

- Avaya Aura® Session Manager.
- Avaya Aura® System Manager.
- Avaya Session Border Controller for Enterprise.
- Avaya Aura® Experience Portal.



**Figure 1: Test Configuration Diagram**

## 4. Equipment and Software Validated

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

<b>Equipment/Software</b>	<b>Release/Version</b>
Avaya Aura® System Manager	7.0.1.0.0.64859
Avaya Aura® Session Manager	7.0.1.0.701007
Avaya Aura® Experience Portal	7.1.0.1117
Avaya Session Border Controller for Enterprise	7.1.0.2
Versay CUE Analytics	1.13.0

## 5. Configure Avaya Aura® Experience Portal

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

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

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

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

Expand All | Collapse All

You are here: Home

### Avaya Aura® Experience Portal Manager

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

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

### Installed Components

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

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

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

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

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

## 5.1. 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 “TLS” if SIP connection to SIP proxy using TLS, otherwise select TCP.
- **Proxy Servers:** Enter the SIP signaling IP address of Session Manager.
- **SIP Domain:** Enter a SIP domain “avayasm.vailsys.com” as configured in **Section 6.1**.
- 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** button to save the changes.

Welcome, epadmin  
Last logged in May 3, 2017 at 8:05:55 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 > Add SIP Connection

### Add SIP Connection

Use this page to add a new SIP connection.

Name:

Enable:  Yes  No

Proxy Transport:

Proxy Servers  DNS SRV Domain

Address	Port	Priority	Weight	
172.20.152.253	5061	0	0	Remove

[Additional Proxy Server](#)

Listener Port:

SIP Domain:

P-Asserted-Identity:

Maximum Redirection Attempts:

Consultative Transfer:  INVITE with REPLACES  REFER

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



Scroll down to **SRTP** section, configure SRTP profile as shown in the **Configured SRTP List** below.

The screenshot shows the Avaya Aura Experience Portal 7.1.0 configuration interface. The left sidebar contains a navigation menu with categories: User Management, Real-time Monitoring, System Maintenance, System Management, System Configuration, and Security. The main content area is titled 'Call Capacity' and includes a 'Maximum Simultaneous Calls' input field set to '2'. Below this are two radio button options: 'All Calls can be either inbound or outbound' (selected) and 'Configure number of inbound and outbound calls allowed'. The 'SRTP' section contains several settings: 'Enable' (Yes selected), 'Encryption Algorithm' (AES\_CM\_128 selected), 'Authentication Algorithm' (HMAC\_SHA1\_80 selected), 'RTCP Encryption Enabled' (No selected), and 'RTP Authentication Enabled' (Yes selected). An 'Add' button is located to the right of the SRTP settings. Below the SRTP settings is a 'Configured SRTP List' which contains a single entry: 'SRTP-Yes,AES\_CM\_128,HMAC\_SHA1\_80,RTCP Encryption-No,RTP Authentication-Yes'. A 'Remove' button is positioned to the right of this list.

## 5.2. Administer Speech Server

The Experience Portal system integrates with two types of third-party speech servers: Automatic Speech Recognition (ASR) technology enables an interactive voice response (IVR) system to collect verbal responses from callers and Text-to-Speech (TTS) technology enables an IVR system to render text content into synthesized speech output according to algorithms within the TTS software. This section demonstrates of how to add Nuance ASR and TTS servers.

### 5.2.1. Administer ASR

On the left pane, navigate to **System Configuration** → **Speech Servers** (not shown). To add an **ASR** server, click on **ASR** tab (not shown) and click **Add** (not shown). Enter a **Name**, set **Enable** to **Yes** and set **Engine Type** to **Nuance**. Fill in the IP address of Nuance speech server in **Network Address**. In **Base Port**, fill in “5060” for **TCP** and “5061” for **TLS**, in this case the port “5060” was used for TCP. Enter appropriate value in **Total Number of Licensed ASR Resources**, set **New Connection per Session** to **Yes**, set **Languages** to **English(USA) en-US**.

In the **MRCP** section, select **MRCP V2** in the **Protocol** dropdown menu, select **TCP** in the **Transport Protocol** dropdown menu and enter the port “5060” in the **Listener Port** field.

You are here: [Home](#) > [System Configuration](#) > [Speech Servers](#) > Add ASR Server

### Add ASR Server

Use this page to configure Experience Portal to communicate with a new ASR server.

Name:

Enable:  Yes  No

Engine Type:

Network Address:

Base Port:

Total Number of Licensed ASR Resources:

New Connection per Session:  Yes  No

Languages:

English(Australia) en-AU  
English(UK) en-GB  
English(India) en-IN  
English(Singapore) en-SG  
English(South\_Africa) en-ZA  
English(USA) en-US

MRCP

Ping Interval:  seconds

Response Timeout:  seconds

Protocol:

Transport Protocol:

Listener Port:

## 5.2.2. Administer TTS

On the left pane, navigate to **System Configuration** → **Speech Servers** (not shown). To add a **TTS** server, click on **TTS** tab (not shown) and click **Add** (not shown).

Enter a **Name**, set **Enable** to **Yes** and set **Engine Type** to **Nuance**. Fill in the IP address of Nuance speech TTS in **Network Address**. In **Base Port**, fill in **5060**, enter appropriate value in **Total Number of Licensed ASR Resources**, set **New Connection per Session** to **No**, set **Voices** to **English(USA) en-US Jill F**. In the MRCP section, select “MRCP V2” and “TCP” in the **Protocol** and **Transport Protocol** dropdown menu, enter “5060” in the **Listener Port** field and keep other values at default.

You are here: [Home](#) > [System Configuration](#) > [Speech Servers](#) > Add TTS Server

### Add TTS Server

Use this page to configure Experience Portal to communicate with a new TTS server.

Name:

Enable:  Yes  No

Engine Type:

Network Address:

Base Port:

Total Number of Licensed TTS Resources:

New Connection per Session:  Yes  No

Voices:

English(USA) en-US Ethan M  
English(USA) en-US Evan M  
English(USA) en-US Evelyn F  
English(USA) en-US Jennifer F  
English(USA) en-US Jill F  
English(USA) en-US Kate F

**MRCP**

Ping Interval:  seconds

Response Timeout:  seconds

Protocol:

Transport Protocol:

Listener Port:

### 5.3. Administer Applications

Applications are needed to drive calls in Experience Portal. To add a new application, from the left pane, navigate to **System Configurations** → **Applications** and in the Application page click Add button (not shown). Below are sample of application used during the compliance test. In the **Speech Server** section, select the ASR and TTS servers as configured in **Section 5.2**.

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

### Add Application

Use this page to deploy and configure a new application on the Experience Portal system.

Start With:

Name:

Enable:  Yes  No

Type:

Reserved SIP Calls:  None  Minimum  Maximum

Requested:

#### URI

Single  Fail Over  Load Balance

VoiceXML URL:

Mutual Certificate Authentication:  Yes  No

Basic Authentication:  Yes  No

#### Speech Servers

ASR:  TTS:

Languages:  Voices:

#### Application Launch

Inbound  Inbound Default  Outbound

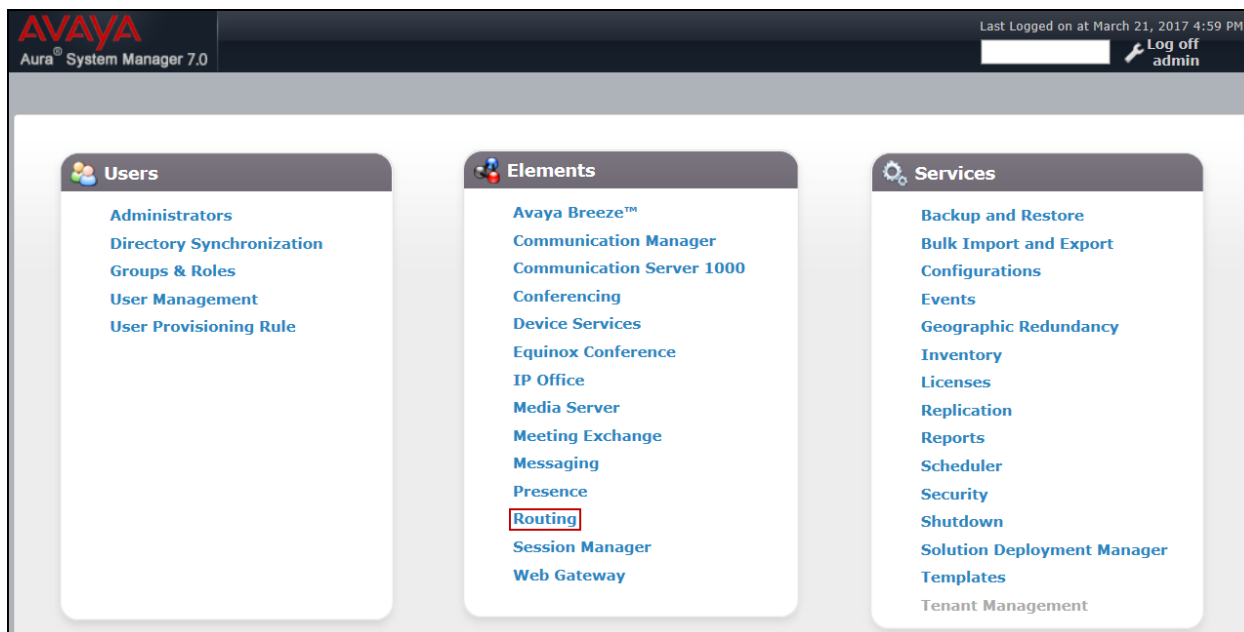
Number  Number Range  URI

Called Number:

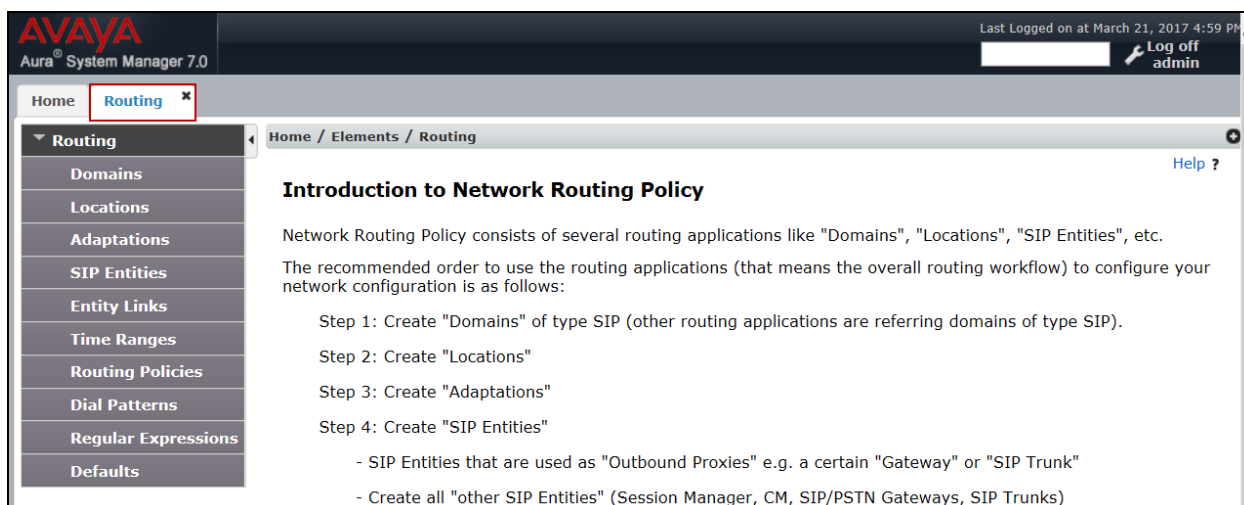
Speech Parameters ▶  
Reporting Parameters ▶  
Advanced Parameters ▶

## 6. Configure Avaya Aura® Session Manager

Session Manager configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL “https://<ip-address>/SMGR”, where “<ip-address>” is the IP address of System Manager. Log in with the appropriate credentials and click on **Log On** (not shown). The screen shown below is then displayed; click on **Routing**.



The navigation tree displayed in the left pane below will be referenced in subsequent sections to navigate to items requiring configuration. Most items discussed in this section will be located under the **Routing** link shown below.



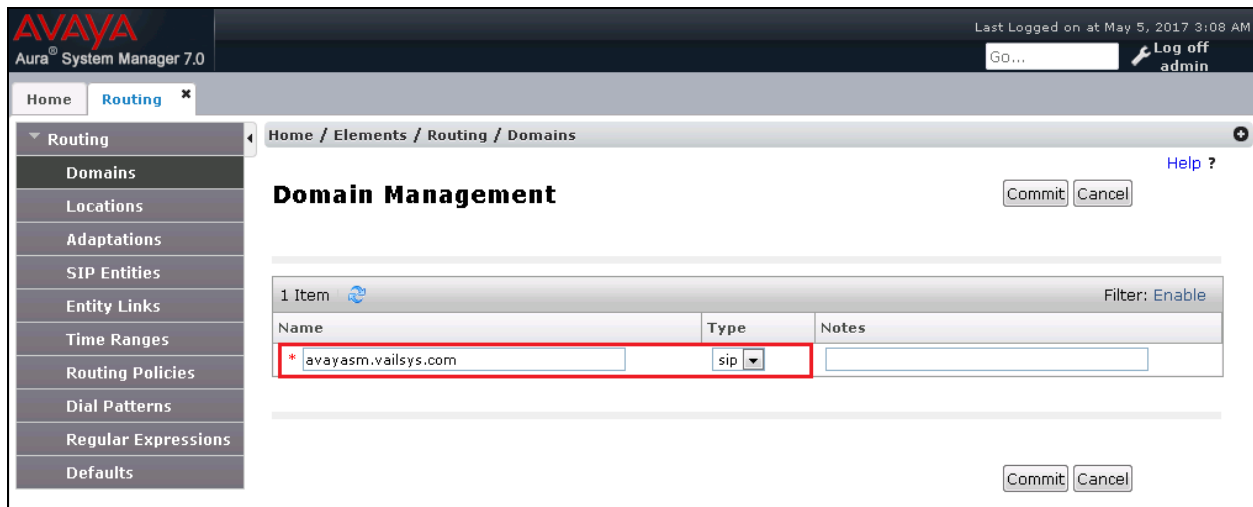
## 6.1. SIP Domain

Create an entry for each SIP domain for which Session Manager will need to be aware in order to route calls. For the compliance test, this was the enterprise domain, *avayasm.vailsys.com*.

Navigate to **Routing** → **Domains** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Name:** Enter the domain name **avayasm.vailsys.com**.
- **Type:** Select **sip** from the pull-down menu.
- **Notes:** Add a brief description (optional).
- Click **Commit** to save.

The screen below shows the entry for the enterprise domain.



The screenshot shows the Avaya Aura System Manager 7.0 interface. The left-hand navigation pane is expanded to show the 'Routing' section, with 'Domains' selected. The main content area is titled 'Domain Management' and contains a table with one entry. The entry is highlighted with a red border. The table has columns for Name, Type, and Notes. The Name column contains 'avayasm.vailsys.com', the Type column contains 'sip', and the Notes column is empty. There are 'Commit' and 'Cancel' buttons at the top right and bottom right of the table area.

Name	Type	Notes
* avayasm.vailsys.com	sip	

## 6.2. Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management, call admission control and location-based routing. To add a location, navigate to **Routing** → **Locations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the **General** section, enter the following values:

- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description (optional).
- Click **Commit** to save.

The following screen shows the location details for the location named *SF Dev*. Later, this location will be assigned to the SIP Entity. Other location parameters (not shown) retained the default values.

The screenshot displays the Avaya Aura System Manager 7.0 interface. The top navigation bar includes the Avaya logo, the text 'Aura System Manager 7.0', and a user session summary: 'Last Logged on at May 5, 2017 3:04', a 'GO...' search field, and a 'Log off admin' button. A breadcrumb trail reads 'Home / Elements / Routing / Locations'. The left sidebar contains a menu with 'Routing' selected, and sub-items: Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Location Details' and includes a 'Help ?' link and 'Commit' and 'Cancel' buttons. Under the 'General' section, the 'Name' field is highlighted with a red box and contains the text 'SF Dev'. Below it is an empty 'Notes' field. The 'Dial Plan Transparency in Survivable Mode' section has an 'Enabled' checkbox that is unchecked. At the bottom, there are empty input fields for 'Listed Directory Number' and 'Associated CM SIP Entity'.

### 6.3. SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to it, which includes Experience Portal and the Avaya SBCE. 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, *Voice Portal* for Experience Portal and *SIP Trunk* (or *Other*) for the Avaya SBCE.
- **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 6.2**.
- **Time Zone:** Select the time zone for the location above.
- Click **Commit** to save.

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 shows the Avaya Aura System Manager 7.0 interface. The top navigation bar includes the Avaya logo, 'Aura System Manager 7.0', and a 'Log off admin' button. The left navigation pane shows a tree view with 'Routing' selected. The main content area displays the 'SIP Entity Details' form. The form has a 'General' section with the following fields: 'Name' (avayasm), 'FQDN or IP Address' (172.20.152.253), 'Type' (Session Manager), 'Notes' (empty), 'Location' (SF Dev), 'Outbound Proxy' (empty), 'Time Zone' (America/Chicago), and 'Credential name' (empty). The 'SIP Link Monitoring' section has a dropdown set to 'Use Session Manager Configuration'. The 'Name', 'FQDN or IP Address', and 'Location' fields are highlighted with red boxes.



The following screen shows the addition of the *Experience Portal* SIP Entity. Select the location that applies to the SIP Entity being created, defined in **Section 6.2**.

The screenshot shows the AVAYA Aura System Manager 7.0 interface. The left sidebar contains a navigation menu with 'SIP Entities' selected. The main content area is titled 'SIP Entity Details' and shows the 'General' tab. The form fields are as follows:

- \* Name: avayaepm
- \* FQDN or IP Address: 172.20.152.251
- Type: Voice Portal
- Notes: (empty)
- Adaptation: (empty)
- Location: SF Dev
- Time Zone: America/Chicago
- \* SIP Timer B/F (in seconds): 4
- Credential name: (empty)
- Securable:
- Call Detail Recording: none

The following screen shows the addition of the *Avaya SBCE* SIP Entity for the Avaya SBCE:

- The **FQDN or IP Address** field is set to the IP address of the SBC private network interface.
- Select the location that applies to the SIP Entity being created, defined in **Section 6.2**.

The screenshot shows the AVAYA Aura System Manager 7.0 interface. The left sidebar contains a navigation menu with 'SIP Entities' selected. The main content area is titled 'SIP Entity Details' and shows the 'General' tab. The form fields are as follows:

- \* Name: avayasbc
- \* FQDN or IP Address: 172.20.152.254
- Type: SIP Trunk
- Notes: (empty)
- Adaptation: (empty)
- Location: SF Dev
- Time Zone: America/Chicago
- \* SIP Timer B/F (in seconds): 4

## 6.4. 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 Experience Portal and one to the Avaya SBCE. To add an Entity Link, navigate 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 (**Section 6.3**).
- **Protocol:** Select the TLS 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 system from the drop-down menu (**Section 6.3**).
- **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 screen below shows the Entity Link to Experience Portal. The protocol and ports defined here must match the values used on the Experience Portal VoIP Connection section in **Section 5.1**. *TLS* transport and port *5061* were used.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port
*avayasm_avayepm_506	*Q.avayasm	TLS	*5061	*Q.avayepm	<input type="checkbox"/>	*5061

The Entity Link to the Avaya SBCE is shown below; *TLS* transport and port *5061* were used.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port
*avayasm_avayasbc_506	*Q.avayasm	TLS	*5061	*Q.avayasbc	<input type="checkbox"/>	*5061

## 6.5. Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.3**. One routing policy was added: an incoming policy with Experience Portal as the destination. To add a routing policy, navigate 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 6.3**) 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 Experience Portal.

The screenshot shows the 'Routing Policy Details' page in the AVAYA Aura System Manager 7.0. The page is titled 'Routing Policy Details' and has a breadcrumb path 'Home / Elements / Routing / Routing Policies'. The 'General' section contains the following fields:

- Name:** SMtoEPM
- Disabled:**
- Retries:** 0
- Notes:** (empty)

The 'SIP Entity as Destination' section shows a table with one row:

Name	FQDN or IP Address	Type	Notes
avayaepm	172.20.152.251	Voice Portal	

## 6.6. Dial Patterns

Dial Patterns are needed to route specific calls through Session Manager. For the compliance test, a dial pattern was needed to route calls from Avaya SBCE to the Experience Portal. Dial Patterns define which route policy will be selected for a particular call based on the dialed digits, destination domain and originating location. To add a dial pattern, navigate to **Routing → Dial Patterns** in the left navigation pane and click on the **New** button in the right pane (not shown). Fill in the following, as shown in the screens below:

In the **General** section, enter the following values:

- **Pattern:** Enter a dial string that will be matched against the Request-URI of the call.

- **Min:** Enter a minimum length used in the match criteria.
- **Max:** Enter a maximum length used in the match criteria.
- **SIP Domain:** Enter the destination domain used in the match criteria, or select “**ALL**” to route incoming calls to all SIP domains.
- **Notes:** Add a brief description (optional).
- In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate originating location for use in the match criteria (**Section 6.2**).
- Lastly, select the routing policy from the list that will be used to route all calls that match the specified criteria (**Section 6.5**). Click **Select** (not shown).
- Click **Commit** to save.

The following screen illustrates an example dial pattern used to verify inbound PSTN calls to the Experience Portal. In the example, calls to 10 digit numbers starting with **844**, the area code assigned to the DID numbers provided by service provider, arriving from location **SF Dev**, used route policy **SMtoEPM**.

The screenshot shows the 'Dial Pattern Details' configuration page. The 'General' section includes the following fields:

- Pattern:** 8446276300
- Min:** 10
- Max:** 36
- Emergency Call:**
- Emergency Priority:** 1
- Emergency Type:** (empty)
- SIP Domain:** avayasm.vailsys.com
- Notes:** (empty)

The 'Originating Locations and Routing Policies' section contains a table with the following data:

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
SF Dev		SMtoEPM	0	<input type="checkbox"/>	avayaepm	

## 7. Configure Versay CUE Analytics System

The configuration of Versay CUE Analytics system is done by Versay engineer and is outside of the scope of these Application Notes. To obtain further information on Versay CUE Analytics system configuration please contacts an authorized Versay representative.

## 8. Verification Steps

This section provides the verification steps that may be performed to verify that Avaya Aura® Experience Portal.

1. From the EPM web interface, verify that the MPP servers are online and running. On the left pane, navigate to **System Management** → **MPP Manager**.

You are here: [Home](#) > System Management > MPP Manager

### MPP Manager (May 22, 2017 8:04:34 AM PDT) Refresh

This page displays the current state of each MPP in the Experience Portal system. To enable the state and mode commands, select one or more MPPs. To enable the mode commands, the selected MPPs must also be stopped.

Last Poll: May 22, 2017 8:04:12 AM PDT

	Server Name	Mode	State	Config	Auto Restart	Restart Schedule		Active Calls	
						Today	Recurring	In	Out
<input type="checkbox"/>	avayaepm.vail	Online	Running	Restart needed	Yes	No	None	0	0

**State Commands**

Start Stop Restart Reboot Halt Cancel

**Restart/Reboot Options**

One server at a time  
 All servers

**Mode Commands**

Offline Test Online

2. Verify that the ports on the MPP server are in service. On the left lane, click on **Port Distribution**. Select the MPP server and click **OK** (not shown).

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

### Port Distribution Report (May 22, 2017 7:57:43 AM PDT) Refresh

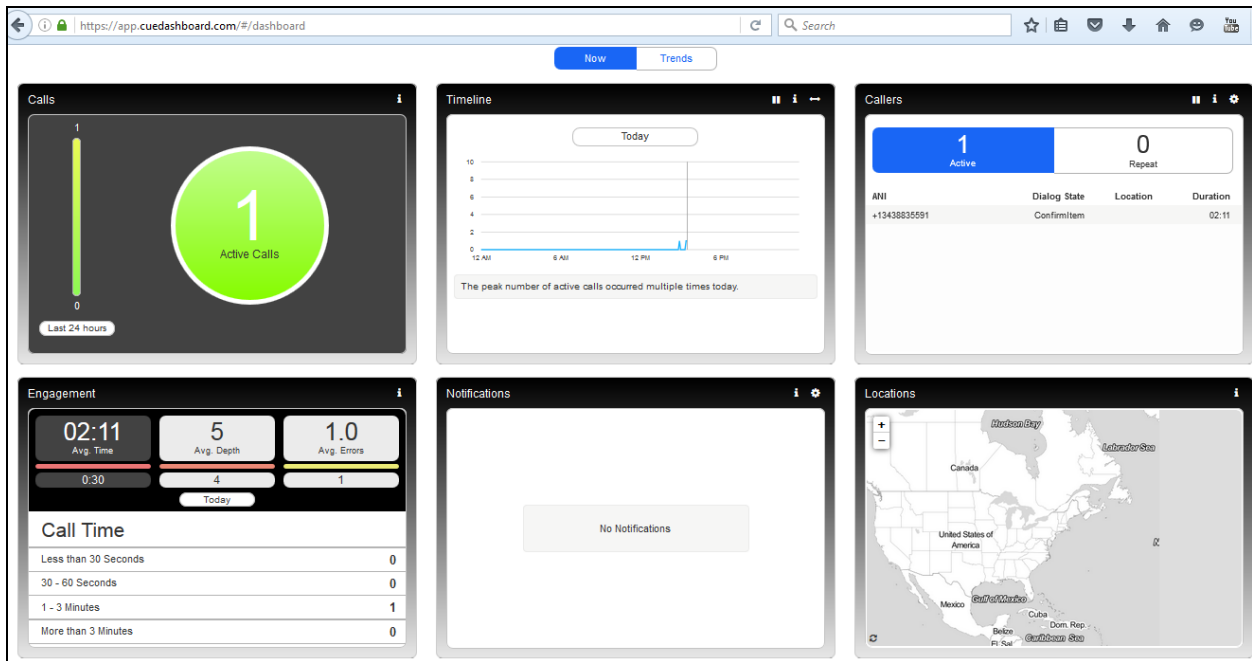
This page displays information about how the telephony resources have been distributed to the MPPs. You configure the telephony resources on the VoIP Connections page.

Total Ports: 2 Last Poll: May 22, 2017 7:57:39 AM PDT

Port	Mode	State	Port Group	Protocol	Current Allocation	Base Allocation
2	Online	In service	ASM70	SIP_Trunk	avayaepm.vail	

**Help**

- Open Firefox or Google Chrome browser to launch the URL <https://app.cuedashboard.com/>, enter an appropriate username and password provided by Versay to log in the CUE dashboard. The CUE dashboard is displayed, select **Now** tab to check real-time information from the dashboard. Place a call to the number 8446276300 as configured in the Experience Portal, the CUE dashboard shows the call information in the real time with 6 widgets: Calls, Timeline, Callers, Engagement, Notifications and Locations.



## 9. Conclusion

These Application Notes describe the configuration steps required to integrate Versay CUE Analytics 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® Session Manager*, Release 7.0, Issue 7, May 2016
- [2] *Administering Avaya Aura® Experience Portal*, Release 7.1, May 2016

Versay CUE Analytics documentation is always available from <http://www.versay.com/cue-analytics>

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