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Avaya Solution & Interoperability Test Lab

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.
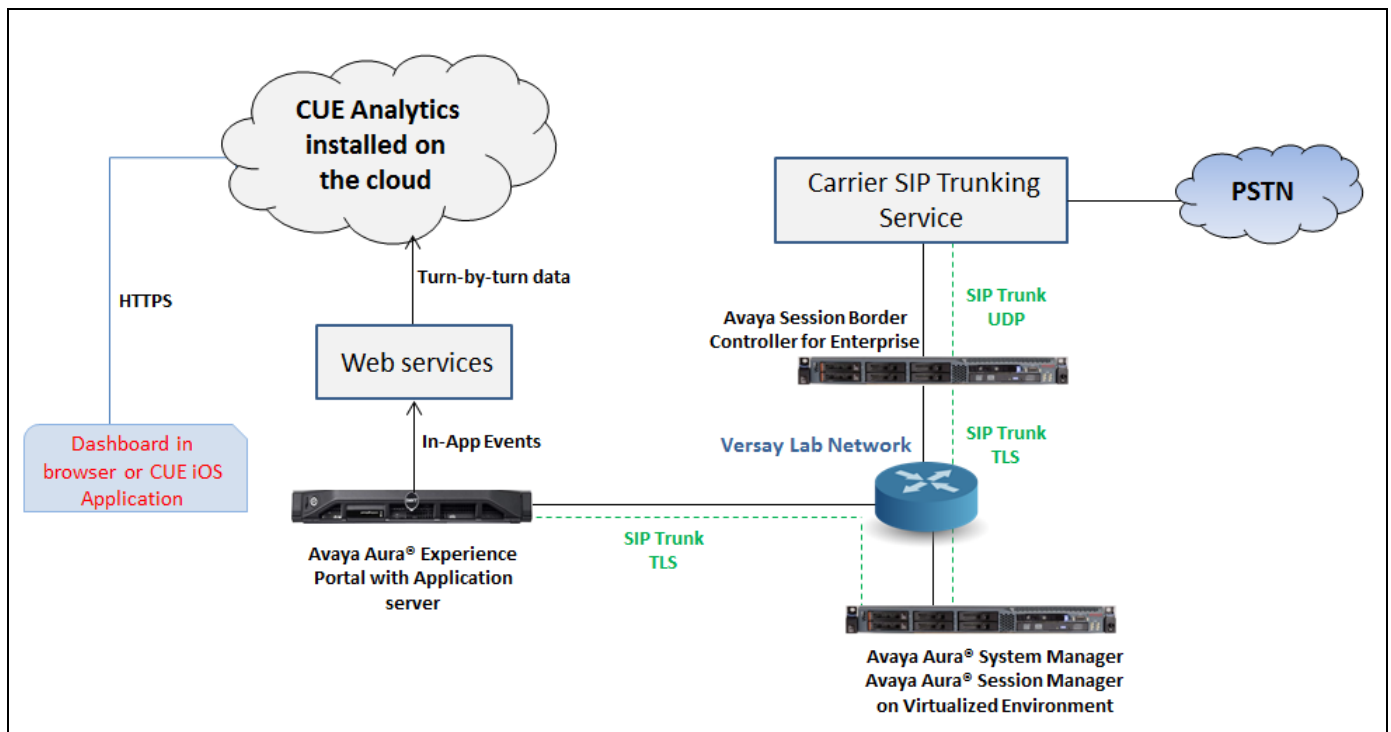
Figure 1: Test Configuration Diagram

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
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.

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.

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

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:	stlasr01a
Enable:	● Yes ○ No
Engine Type:	Nuance ▼
Network Address:	63.209.140.86
Base Port:	5060
Total Number of Licensed ASR Resources:	2
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:	15	seconds
Response Timeout:	4	seconds
Protocol:	MRCP V2 ▼	
Transport Protocol:	TCP ▼	
Listener Port:	5060	

Save   Cancel   Help

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.

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: <u>Home</u> > System Configuration > <u>Applications</u> > Add Application

Add Application

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

Start With: `<None>` ▾
Name: Avaya Test
Enable: ⦿ Yes ○ No
Type: VoiceXML ▾
Reserved SIP Calls: ⦿ None ○ Minimum ○ Maximum
Requested:

URI

⦿ Single ○ Fail Over ○ Load Balance

VoiceXML URL: https://sampa01.vail/~sergeyn/projects/Avaya_CUE/vxml/Greeting.vxml?   [Verify]

Mutual Certificate Authentication: ○ Yes ⦿ No
Basic Authentication: ○ Yes ⦿ No

Speech Servers

ASR: Nuance ▾   TTS: Nuance ▾

Languages: | English(USA) en-US | Voices: | English(USA) en-US Jill F |
                                                 | English(USA) en-US Tom M |

Application Launch

⦿ Inbound ○ Inbound Default ○ Outbound

⦿ Number ○ Number Range ○ URI
Called Number: [   ]   [Add]

| 8446276300 | [Remove]

Speech Parameters ▶
Reporting Parameters ▶
Advanced Parameters ▶
[Save] [Save & Add Next] [Cancel] [Help]

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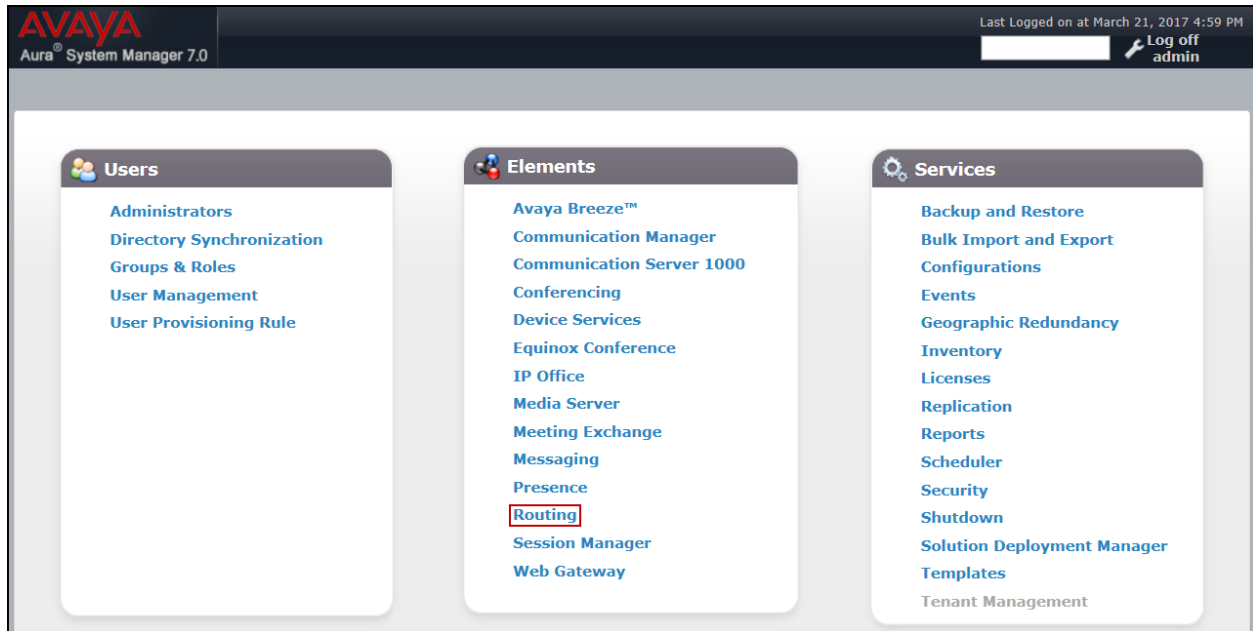
The navigation tree displayed in the left pane below will be referenced in subsequent sections to navigate to items requiring configuration. Most items discussed in this section will be located under the **Routing** link shown below.

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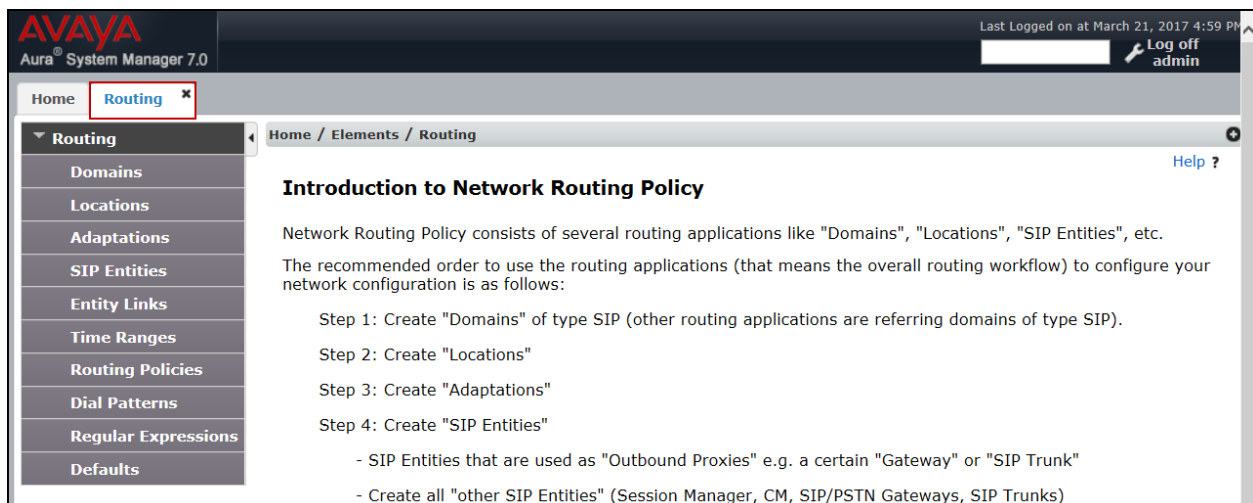
6.1. SIP Domain

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.

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.

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 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 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.
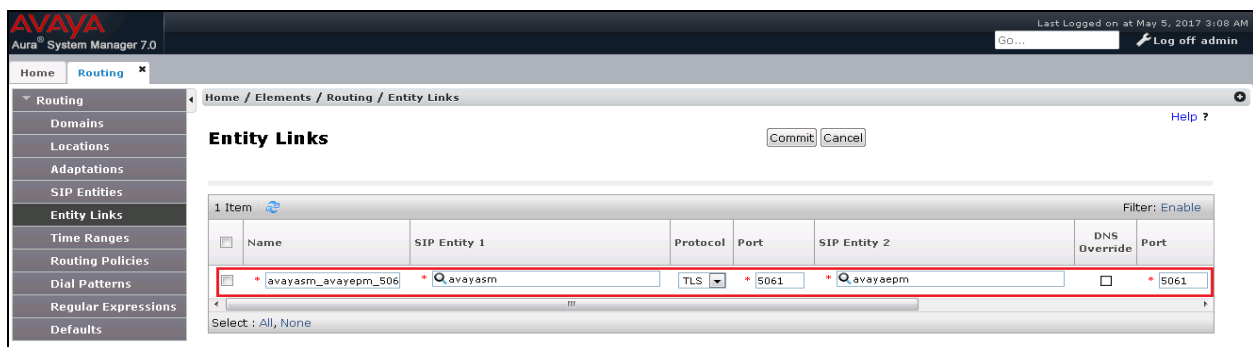- Select the location that applies to the SIP Entity being created, defined in **Section 6.2**.
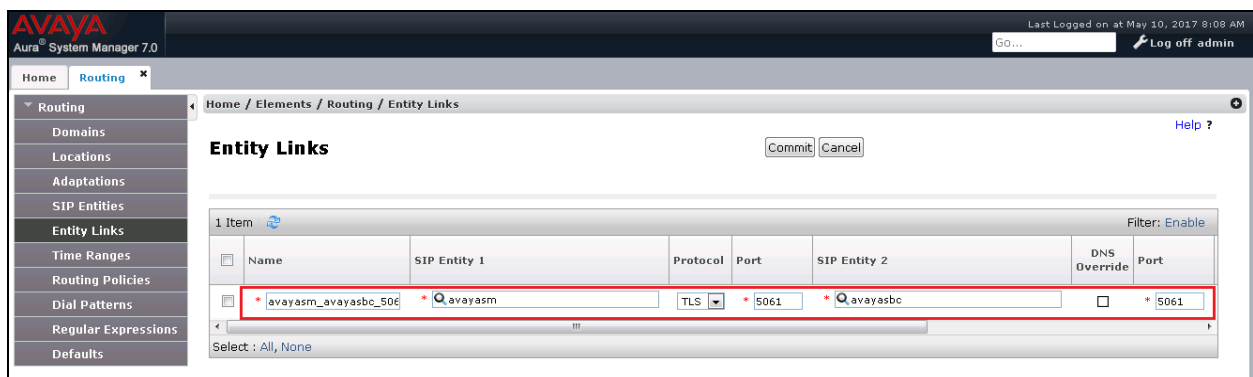
6.4. Entity Links

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.

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

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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.

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.

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- **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*.

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**.

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).

3. 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.
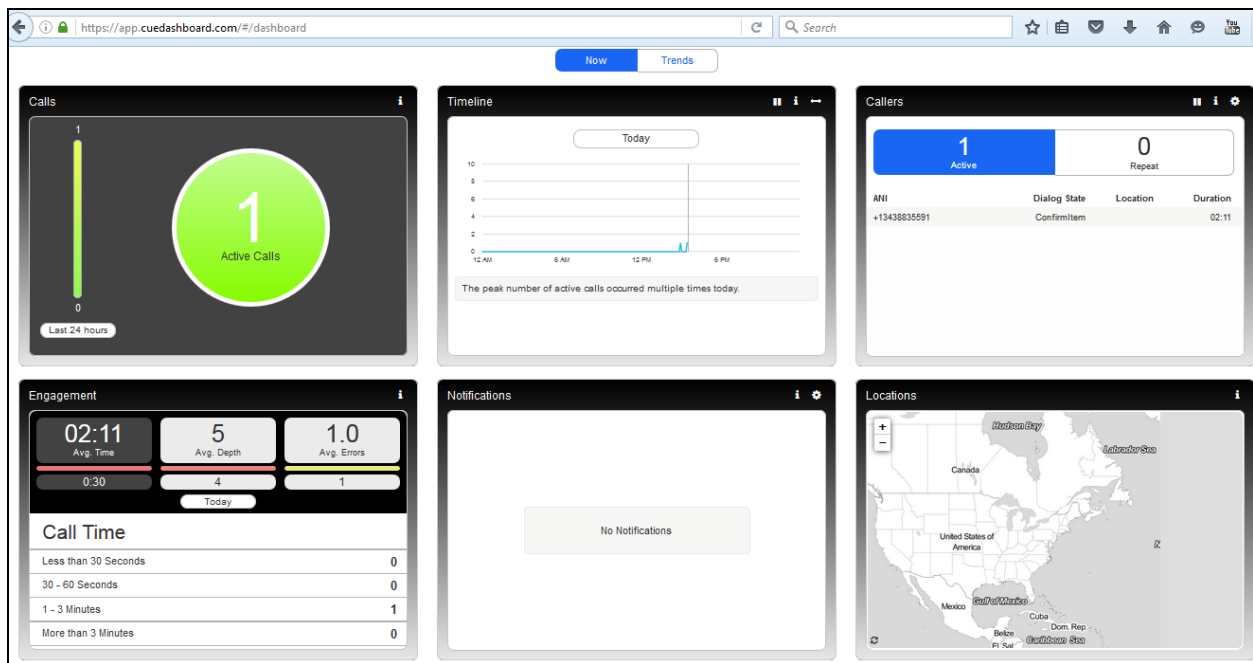
9. Conclusion

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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