## Approved Software Combination

<table>
<thead>
<tr>
<th>Version or Approved Combination</th>
<th>Product</th>
<th>Version</th>
<th>notes</th>
<th>PLDS ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>CE Platform OVA</td>
<td>CE Platform OVA</td>
<td>CE 3.0 Beta GA</td>
<td>MD5 Sum - 2ae0dd71342412c12e5c5f d5ebea321fe</td>
<td>CE0000000031</td>
</tr>
<tr>
<td>CE Platform ISO (for upgrades)</td>
<td>CE Platform ISO (for upgrades)</td>
<td>CE 3.0 Beta GA</td>
<td>MD5 Sum - 1fd3b0f7303e38ef8d67cfe 666b199da</td>
<td>CE0000000030</td>
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<tr>
<td>System Manager patch</td>
<td>System Manager patch for CE</td>
<td>CE 3.0 Beta GA</td>
<td>MD5 Sum - fd774403b9942d1739c308 45907df0e3</td>
<td>CE0000000035</td>
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<tr>
<td>CE Element Manager Patch</td>
<td>CE Element Manager Patch</td>
<td>CE 3.0 Beta GA</td>
<td>MD5 Sum - 843e3d4f93b454751488f4 81e0ebdc47</td>
<td>CE0000000032</td>
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<tr>
<td>AMS for CE</td>
<td>AMS for CE</td>
<td>7.6.0.834</td>
<td>Sha1 ca517c3d7c92f3d584d629 2b605c5bb0afcc49d8</td>
<td>CE0000000033</td>
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<tr>
<td>Real-Time Speech Snap-in</td>
<td>Real-Time Speech Snap-in</td>
<td>3.0.0.0.09002</td>
<td>MD5 Sum - 108d6de5f3cb19dcd8f85f75 21654dc8e</td>
<td>RT50000000005</td>
</tr>
</tbody>
</table>

## Collaboration Environment Interoperability

Real-Time Speech Snap-in 3.0 software (Sprint 9) is deployed as a snap-in service on Avaya Aura Collaboration Environment 3.0.

## Additional Supporting Documentation

The *Avaya Real-Time Speech Snap-in Reference Guide* provides an introduction to the snap-in and details on deploying and managing the snap-in.

The Real-Time Speech Snap-in SDK documentation provides a detailed overview of the REST API in addition to providing details on best practices for managing your search queries and how to use the provided sample application.
Known issues and workarounds
Please refer to the CE 3.0 documentation for known platform issues and workarounds.

**Snap-in – Known Issues**

<table>
<thead>
<tr>
<th>Problem</th>
<th>Issue Description</th>
<th>Workaround</th>
<th>Reference</th>
<th>Keywords</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Only one speech search can be initiated per call</td>
<td>No workaround in current release</td>
<td>ASASNAPIN-394</td>
<td>Speech Search</td>
</tr>
<tr>
<td>2</td>
<td>All, AtLeast and First Operators must include a duration</td>
<td>Use a duration value in all of the operators mentioned above. Alternatively, use an ANY operator</td>
<td>ASASNAPIN-473</td>
<td>operating matching</td>
</tr>
<tr>
<td>3</td>
<td>UTF-8 encoded characters are not correctly handled in speech search requests and matches</td>
<td>Use non UTF-8 encoded characters</td>
<td>ASASNAPIN-470</td>
<td>Character formatting output</td>
</tr>
<tr>
<td>4</td>
<td>No Speech Search Stop Events are received from the snap-in</td>
<td>Use CALL_ENDED events as an indicator Speech Search is ended</td>
<td>ASASNAPIN-331</td>
<td>Speech Search Stop</td>
</tr>
<tr>
<td>5</td>
<td>Matches from the Last Operator are not received</td>
<td>None. Last Operator not supported in this release</td>
<td>ASASNAPIN-475</td>
<td>Last Operator</td>
</tr>
<tr>
<td>6</td>
<td>500 Internal Server Error returned when attempting to delete a query with an invalid ID</td>
<td>Ensure your application is configured use the Accept header in HTTP response.</td>
<td>ASASNAPIN-424</td>
<td>500 internal server error</td>
</tr>
<tr>
<td>7</td>
<td>Cannot create a search query with multiple languages</td>
<td>Create a duplicate query with a new language identifier</td>
<td>ASASNAPIN-169</td>
<td></td>
</tr>
</tbody>
</table>
Keywords: Query, language

8 Problem: Cannot create a search query with more than 35 phrases
Workaround: Create a query with less than 35 phrases.
Reference: ASASNAPIN-317
Keywords: Query

9 Problem: Some global attributes are not applied to cluster and profile sections
Workaround: User cluster or profile attribute settings
Reference: ASASNAPIN-418
Keywords: Global attributes

10 Problem: Incorrect Alarm triggered when no queries configured
Workaround: None
Reference: ASASNAPIN-489
Keywords: alarms

11 Problem: Time format differences between speech and call events.
Workaround: None
Reference: ASASNAPIN-486
Keywords: Eventing

12 Problem: No error or speech match is received after exceeding allowable phrase limit
Workaround: Create a query with less than 35 phrases.
Reference: ASASNAPIN-490
Keywords: Query

13 Problem: Issues when attempting to auto-start speech search when originating and terminating sequence is enabled
Workaround: Only auto-start with originating or terminating sequencing configured
Reference: ASASNAPIN-494
Keywords: Auto Start Speech Search

14 Problem: Under continuous high traffic, CE performance can degrade
Workaround: Under high traffic, schedule regular restarts of the CE application server
Reference: ASASNAPIN-515
Keywords: High Traffic

15 Problem: Unable to filter call events for a specific user/end-point
Workaround: Subscribed users will receive call events for all end-points
Reference: ASASNAPIN-525
Keywords: Call Filtering

16 Problem: Incorrect operator data sent when using an inline query
Workaround: Use tagged queries.
Reference: ASASNAPIN-521 / ASASNAPIN-516
Keywords: High Traffic

17 Problem: Incorrect language applied to calls. Sub-languages are not correctly applied. For example, if en-us is requested, US English will be applied to the call. If en-uk is requested, the US English is incorrectly applied to the call
Workaround: None
Reference: ASASNAPIN-527
Keywords: Speech Search / Languages

18 Problem: Match for top level ALL operator not produced even if all child operators are matched
Workaround: Use a small query
Reference: ASASNAPIN-522
Keywords: ALL operator matching

Sample Application – Known Issues

1. Problem: Using the REST interface, duplicate queries can be created.
   Workaround: None. The Query Name can be duplicated, but the ID is for each query is unique
   Reference: ASASNAPIN-326
   Keywords: query

2. Problem: Intermittent issue with deleting queries
   Workaround: Reselect the query and select the delete option again. Verify that no other queries have been deleted.
   Reference: ASASNAPIN-520
   Keywords: Query

Collaboration Designer Integration – Known Issues

1. Problem: Only a single speech search match can be processed within a Collaboration Designer block.
   Workaround: Limit the search query to a scenario where only a single match is required.
   Reference: WORKFLOW-919
   Keywords: Collaboration Designer
Post Install Steps

In addition to the setup steps documented in Real-Time Speech Snap-in Guide the following setup instructions should be completed post Snap-In installation.

**Update Signalling Group on Communication Manager**

To ensure no PRACK messages are sent to CE (which can cause calls not to get established correctly) please ensure the following values are set to YES on all CM signalled groups used by CE (including remote CM’s, if applicable).

- **Direct IP-IP Audio Connections? = y**
- **Initial IP-IP Direct Media? = y**
- **Outgoing Direct Media? = y**

**Ensuring Users are correctly configured for CE sequencing**

To ensure both H.323 and SIP users are successfully sequenced into CE, please ensure that the following configuration steps are adhered to:

**Creating E.164 and Avaya SIP addresses for each user**

For each user, ensure that an E.164 and Avaya SIP address is created.
From System Manager, navigate to: Users > User Management > Manage Users
Select the required user and select “Edit”
Under the “Communication Profile”, add the E.164 and Avaya SIP address.

**Ensure Implicit User Sequencing is enabled on Session Manager**

From System Manager, navigate to: Home > Elements > Session Manager > Session Manager Administration.

Ensure that “Enable Implicit Users Applications For SIP Users” is enabled.

**Session Manager Administration**

This page allows you to administer Session Manager instances and configure their global settings.

**Global Settings**

- Allow Unauthenticated Emergency Calls
- Allow Unsecured PPM Traffic
- Failback Policy
- ELIN SIP Entity
- Better Matching Dial Pattern or Range in Location ALL Overrides Match in Originator’s Location
- Ignore SDP for Call Admission Control
- Disable Call Admission Control Threshold Alarms
- Disable Loop Detection Alarms
- Loop Detection Alarms Threshold (hours)
- Enable TLS Endpoint Certificate Validation
- Enable Dial Plan Ranges
- Enable Implicit Users Applications for SIP users
Configuration and Usage Limitations and Recommendations

**No Support for AAEP interaction for CE sequenced users**

In this release, there is no support for sequenced users integrated with Avaya Aura Experience Portal.
Troubleshooting

Following should be validated when troubleshooting an issue with the Real-Time Speech Search Snap-in. Reference the Real-Time Speech Snap-in guide for required process steps:

1.) Validate that the Real-Time Speech Snap-in has been successfully installed and deployed on Collaboration Environment
2.) Validate that the CallEventControl and EventingConnector have been successfully installed and deployed on the same CE cluster as the Real-Time Speech Snap-in
3.) Validate that you can navigate to the snap-in REST API, by accessing the following URL: https://<ce-fqdn-or-ip>/services/RealTimeSpeech/queries
4.) Validate that at least one query has been configured in the system (if not using the system example query)
5.) Validate that the AMS associated with the CE cluster is configured to include the Speech Search Engine

Snap-in Attributes

The following attributes are available for configuration:

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Party Target</td>
<td>Indicates which party(ies) on a two party call should have speech searching initiated for them.  By default, speech search is started on both the calling and called parties. Due to engineering limitations in the initial release, starting speech search on both parties will use 2 separate speech search sessions on the AMS. This reduces the overall system capacity.</td>
</tr>
<tr>
<td>Enable Automatic Start of Speech Search</td>
<td>When enabled, speech search will automatically be started for any call that is sequenced into the snap-in by CE.</td>
</tr>
<tr>
<td>Enable Example Query</td>
<td>By default, the example query is disabled, but can be enabled to support validation of a system install.</td>
</tr>
<tr>
<td>Search Language</td>
<td>This is the language to be applied when automatically starting a speech search request. This attribute is not used when programatically starting</td>
</tr>
</tbody>
</table>
Ensure Implicit User Sequencing is configured correctly

Please ensure that implicit user sequencing does not include a “+” at the start of the sequence:

<table>
<thead>
<tr>
<th>Pattern</th>
<th>Min</th>
<th>Max</th>
<th>SIP Domain</th>
<th>Origination Application</th>
</tr>
</thead>
<tbody>
<tr>
<td>633xxxx</td>
<td>7</td>
<td>7</td>
<td>-ALL-</td>
<td>SGCE3 AppSeq</td>
</tr>
<tr>
<td>+633xxxx</td>
<td>8</td>
<td>8</td>
<td>-ALL-</td>
<td>SGCE3 AppSeq</td>
</tr>
</tbody>
</table>

The first entry in the above screen-shot is the correct format.
The second entry is the NOT the correct format for entry implicit user sequencing.

Using the Real-Time Speech Snap-in log file

The Real-Time Speech snap-in generates a log file that can be used for troubleshooting purposes. You must connect to the CE server via SSH to access the log. Most log output is not generated by default and will only appear when lower level logging is enabled for the snap-in. You can enable this via the Collaboration Environment page in System Manager, or alternatively, directly on the CE server using the following command:

```
ce dlogon RealTimeSpeech
```

The log file is generated at the following location, with logs rotated to .1, .2 etc. extensions over time:

```
/var/log/Avaya/services/RealTimeSpeech/RealTimeSpeech.log
```

You can use vi or tail to view or monitor the log accordingly, or alternatively, use the following commands (for which you do not need to remember the file location).

To view the log file:

```
ce dlogv RealTimeSpeech
```

To monitor the log output:

```
ce dlogw RealTimeSpeech
```

Of particular interest in this log, look for the SPEECH_SEARCH_STARTED and SPEECH_SEARCH_MATCH events to ensure Real-Time Speech is working as expected.
Exceeding the Speech Search Query Size

In the initial release of the Snap-in, there is a capacity limitation on the size of the speech search query that can be passed from the snap-in to the core Speech Search Engine.

In the event that a defined speech search query is applied to a call, the following error message will be displayed in the Real-Time Speech log files:

```
Event[family=SpeechSearch,type=SPEECH_SEARCH_MATCH,body={"eventType":"SPEECH_SEARCH_MATCH","ucid":"00025001514055547780","appid":"local.1405518432122_12826","searchid":"QorIx7QQMu-06F876xZWg","tags":[]},"conceptMatches":[]{"name":"good afternoon","operator":"phrase","starttime":24.20675,"endtime":24.85675,"confidence":27.65948},{"name":"Basic Greeting","operator":"any","starttime":24.20675,"endtime":24.85675,"confidence":27.65948},"callingParty":{"handle":"10049","domain":"snapin.avaya.com"},"calledParty":{"handle":"20049","domain":"snapin.avaya.com","display":"USDV20049, first"},"timeOfEvent":"2014-07-16 15:56:51.006"},version=1.0.0
```

When creating a speech search query, use the table below to validate the length of the overall query. Substitute “x” values with the number of operators and phrases in your query. Multiply this by the overhead to give the approximate message size. If the sum of all messages is greater than 6000, the message size is too large.

<table>
<thead>
<tr>
<th>Message</th>
<th>Quantity</th>
<th>Overhead</th>
<th>Size</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Message Overhead</td>
<td></td>
<td></td>
<td>2632</td>
</tr>
<tr>
<td>Operator</td>
<td>x</td>
<td>50</td>
<td></td>
</tr>
<tr>
<td>Phrase</td>
<td>x</td>
<td>92</td>
<td></td>
</tr>
<tr>
<td>Total</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Validating AMS is being successfully invoked for Speech Search

Log into the AMS Element Manager and view active sessions:

From the menu tree, select: System Status > Monitoring > Active Sessions

When a call is placed through the system where speech search is being invoked, you should be able to see active sessions on this page. Selecting individual sessions allows you to see the SIP messaging between CE and AMS. (Note that no caller/callee identifying information will be visible here.)
Speech Search Schema Changes

From the initial Beta release, there have been some changes in the event schema for the SpeechSearch event family. The schema has been modified to support integration to Collaboration Designer. The modified schema is detailed in the updated REST API documentation.

Mandatory requirement for HTTPS

With the GA build of CE and the Real-Time Speech Snap-in, HTTPS is mandatory for all REST requests. For details on certificate management, please see the Sample Application User Guide.

Supported Languages

<table>
<thead>
<tr>
<th>Language Name</th>
<th>Language Pack Code</th>
<th>ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>Arabic (Saudi Arabia)</td>
<td>1025</td>
<td>ar_SA</td>
</tr>
<tr>
<td>German (Standard)</td>
<td>1031</td>
<td>de</td>
</tr>
<tr>
<td>English (United States)</td>
<td>1033</td>
<td>en_US</td>
</tr>
<tr>
<td>French (Standard)</td>
<td>1036</td>
<td>fr</td>
</tr>
<tr>
<td>Dutch</td>
<td>1043</td>
<td>nl</td>
</tr>
<tr>
<td>Portuguese (Brazil)</td>
<td>1046</td>
<td>pt_BR</td>
</tr>
<tr>
<td>Russian</td>
<td>1049</td>
<td>ru</td>
</tr>
<tr>
<td>English (United Kingdom)</td>
<td>2057</td>
<td>en_GB</td>
</tr>
<tr>
<td>Spanish (Mexico)</td>
<td>2058</td>
<td>es_MX</td>
</tr>
<tr>
<td>Arabic (Egypt)</td>
<td>3073</td>
<td>ar_EG</td>
</tr>
<tr>
<td>French (Canada)</td>
<td>3084</td>
<td>fr_CA</td>
</tr>
</tbody>
</table>

Note earlier known issue in relation to how language ID’s are managed in this release.