Avaya Real-Time Speech Snap-in

Release 3.1

Release Notes

This document contains information on software lineup, known issues and workarounds specific to this release of Real-Time Speech Snap-in
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**Purpose**

This document contains known issues, patches and workarounds specific to this build and does not constitute a quick install guide for Real-Time Speech. Please refer to the information below to identify any issues relevant to the component(s) you are installing and then refer to the Real-Time Speech Reference guide for full installation instructions.

**Publication History**

<table>
<thead>
<tr>
<th>Issue</th>
<th>Change Summary</th>
<th>Author(s)</th>
<th>Date</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.0</td>
<td>Initial release for Beta - 3.1</td>
<td>Real-Time Speech Design Team</td>
<td>14th October 2015</td>
</tr>
<tr>
<td>1.1</td>
<td>Updates for GA</td>
<td>Real-Time Speech Design Team</td>
<td>23rd November 2015</td>
</tr>
</tbody>
</table>
Software Information

Approved Software Line-up and download locations
All Real-Time Speech Snap-in 3.1 Beta Software for all sites is available to download from PLDS:

<table>
<thead>
<tr>
<th>Product</th>
<th>Notes</th>
<th>Version</th>
<th>MDS Checksum</th>
<th>PLDS ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>Real-Time Speech Snap-in</td>
<td>GA build of RTS snap-in</td>
<td>3.1.0.0.109</td>
<td>32df24b5be1a523b34f48f505c02683</td>
<td>RTS00000021</td>
</tr>
<tr>
<td>Speech Services Snap-in</td>
<td>GA build of ASR/TTS snap-in</td>
<td>3.1.0.0.108</td>
<td>1f48affe496d621f31837de7b3cd861</td>
<td>RTS00000018</td>
</tr>
<tr>
<td>Real-Time Speech Tasks</td>
<td>Tasks SVAR required to support ED integration</td>
<td>3.1.0.0.103</td>
<td>96f18ff3cb0eaa634f43d12ab608f04d</td>
<td>RTS00000017</td>
</tr>
<tr>
<td>System Manager 7.0 OVA</td>
<td>GA release of System Manager</td>
<td>7.0</td>
<td>f89380aec4e1fe7eeae9a378118b132</td>
<td>SMGR70GA001</td>
</tr>
<tr>
<td>System Manager SP1 Patch</td>
<td>Patch required on top of SMGR 7.0 to support EDP 3.1.1</td>
<td>7.0.0.1</td>
<td>a86736f1bf7d5764a9ed7b4ea3cb1f</td>
<td>RTS00000011</td>
</tr>
<tr>
<td>EDP 3.1.1 GA ISO</td>
<td>Used when upgrading from 3.1 GA line-up</td>
<td>3.1.1.0.311006</td>
<td>e0f143307e53a1fccc39e5a6112b178f0</td>
<td>CE000000115</td>
</tr>
<tr>
<td>EDP 3.1.1 GA OVA</td>
<td>For new installs</td>
<td>3.1.1.0.311006</td>
<td>51723d1a004a85e7d0b37a2cc39e2c17</td>
<td>CE000000116</td>
</tr>
<tr>
<td>AAMS 7.7 OVA</td>
<td>GA release of AAMS 7.7</td>
<td>7.7.0.226</td>
<td>3d476ce8b74efc1bc32e4e45ef1ea141</td>
<td>MSR00000016</td>
</tr>
<tr>
<td>AAMS 7.7 update 281</td>
<td>Update to GA release which aligns with EDP 3.1.1</td>
<td>7.7.0.281</td>
<td>11382d3c1b7907d382faffc942de59bd</td>
<td>MSR00000025</td>
</tr>
<tr>
<td>AAMS 7.7 System Layer 14</td>
<td>Update to GA release which aligns with EDP 3.1.1</td>
<td>7.7.0.14</td>
<td>6c13700de16ea05ed1583a218b1576b7</td>
<td>MSR00000026</td>
</tr>
</tbody>
</table>

Hardware Appliance

There are no software downloads associated with the Hardware Appliance deployment of the Avaya Aura® Media Server.
**Engagement Development Platform Interoperability**

Real-Time Speech 3.1 aligns with the Engagement Development Platform Release 3.1.1. Real-Time Speech is not supported on earlier versions of EDP.

For further information in the EDP 3.1.1 release, please refer to the latest EDP documentation and release notes on the Avaya Support Site https://support.avaya.com/documents/
Using the SDK

Download the SDK from the DevConnect Portal. The Avaya DevConnect portal can be accessed at: http://www.devconnectprogram.com/site/global/home/p_home.gsp

Unzip the contents of the SDK to a local machine and use the HTML file: “Developing Applications with the Real Time Speech SDK” as the starting point to navigate the contents of the SDK.
Deployment & Configuration Information

New Installation

Upgrading from Real-Time Speech 3.0
Follow the details outlined in the Real-Time Speech Reference Guide to complete an upgrade from Real-Time Speech 3.0 to 3.1.

To preserve queries defined as part of a Real-Time Speech 3.0 solution, ensure that you export the queries to an external location prior to starting the upgrade.

Upgrading from Real-Time Speech 3.1 Beta Versions
Follow the details outlined in the Upgrading Avaya Engagement Development Platform and Real-Time Speech Reference Guide for upgrade and patching instructions for EDP, SMGR and RTS to upgrade from beta to the GA lineup.

Post-Installation Configuration
Follow the details outlined in the Real-Time Speech Reference Guide to complete the post installation configuration of Real-Time Speech.

Licensing
Due to a commercial licensing issue, customers deploying Real-Time Speech prior to the end of January 2016 will be required to install a temporary license. Customers should continue to order Real-Time Speech through the Avaya commercial tools. On receipt of your order, Avaya will manually generate a temporary license until this issue has been resolved.
Localization
Not Applicable to this release
Known Issues

Real-Time Speech Snap-in

At high traffic rates, SPEECH_SEARCH_STOPPED events received are not timely.

<table>
<thead>
<tr>
<th>Tracking Number</th>
<th>Description</th>
<th>Impact</th>
<th>Workaround</th>
</tr>
</thead>
<tbody>
<tr>
<td>ASASNAPIN-1502</td>
<td>Under higher traffic, SPEECH_SEARCH_STOPPED events may be received approximately 2 minutes late (after call has ended) as there may be an issue producing these events.</td>
<td>RealTimeSpeech will attempt to deliver SPEECH_SEARCH_STOPPED events. CALL_ENDED events are not impacted.</td>
<td>CALL_ENDED events can also be used to indicate the call has completed a speech search. The UCID of SPEECH_SEARCH_STARTED event can be correlated with the UCID the CALL_ENDED event. If a CALL_ENDED event is received, it can be assumed that the speech search has also ended.</td>
</tr>
</tbody>
</table>

Sometimes SPEECH_SEARCH_ERROR events are received with a “Redirect to or from Media Server” error reason when starting or stopping a Speech Search

<table>
<thead>
<tr>
<th>Tracking Number</th>
<th>Related</th>
<th>Description</th>
<th>Impact</th>
<th>Workaround</th>
</tr>
</thead>
<tbody>
<tr>
<td>ASASNAPIN-1501</td>
<td>ZEPHYR-36862: Cannot add a Media Server to a Call for media treatment consistently</td>
<td>In some cases a SPEECH_SEARCH_ERROR event containing the reason “redirect to or from media server failed with status code 504”</td>
<td>The Speech Search will not engage for requests encountering such errors. There are two things to note: 1. If the system is in a high traffic scenario, further speech searches will need to wait until traffic reduces. 2. In normal usage, subsequent request will work.</td>
<td>Follow guidelines for performance usage to ensure systems do not become under load to reduce these errors. In normal usage scenarios, further speech searches will continue to work. The expectation is that this error should occur less than 1% in usage scenarios.</td>
</tr>
</tbody>
</table>
Increasing Speech Search Engine usage within Avaya Aura Media Server

<table>
<thead>
<tr>
<th>Tracking Number</th>
<th>Description</th>
<th>Impact</th>
<th>Workaround</th>
</tr>
</thead>
<tbody>
<tr>
<td>SSE-6525</td>
<td>The Speech Search Engine (SSE) memory usage never decreases from its ‘high water mark’ on AAMS even if all searches are completed. This is particularly an issue under load because temporarily high traffic volumes can result in memory usage nearing 4 GBs - the system limit. At this point AAMS rejects new calls and as SSE memory usage cannot be reclaimed, this error condition cannot be recovered unless AAMS is restarted.</td>
<td>This occurs under high load of speech searches. In our labs we have encountered this issue beyond 800 simultaneous speech searches on a high end physical AAMS deployment. This will occur beyond 85 simultaneous speech searches on lower end virtualized AAMS deployments. RealTimeSpeech will produce a SPEECH_SEARCH_ERROR event with the reason “System Overload”.</td>
<td>There is only one workaround for this issue – reboot AAMS. As this is a system outage scenario, we recommend that customers stay within the capacity limits of RealTimeSpeech. See the Capacity section below for further details.</td>
</tr>
</tbody>
</table>

Media Server not shuffled out when Speech Search is stopped

<table>
<thead>
<tr>
<th>Tracking Number</th>
<th>Description</th>
<th>Impact</th>
<th>Workaround</th>
</tr>
</thead>
<tbody>
<tr>
<td>ZEPHYR-36299</td>
<td>When Media Server Shuffling is enabled for the Real Time Speech snap-in, calls are successfully shuffled onto the AMS when starting a speech search. However, stopping the speech search does not result in the call being shuffled off of the media server again.</td>
<td>This occurs for all speech searches that are stopped when Media Server Shuffling is enabled for the Real Time Speech snap-in. The impact here is that as soon as a speech search is started on a call, then the Media Server resource cannot be relinquished when a speech search is stopped. For many scenarios, it may not impact the customer as they may wish to start other speech searches on the same call. The impact here is only when customers want to relinquish the Media Server resource for other calls. On end of the call, the Media Server resource is relinquished as expected.</td>
<td>There is no workaround for this.</td>
</tr>
</tbody>
</table>
No validation is performed on the EventProperty when subscribing for events using the RealTimeSpeech Event Subscription REST API

<table>
<thead>
<tr>
<th>Tracking Number</th>
<th>Description</th>
<th>Impact</th>
<th>Workaround</th>
</tr>
</thead>
</table>
| ASASNAPIN-1484  | It is possible to subscribe for invalid event types when subscribing for events using the RealTimeSpeech Event Subscription REST API. For example, the following example is accepted: 

```json
{
    "eventFamily": "SpeechSearch",
    "eventType": "SPEECH_SEARCH_SOMETHING_INVALID",
    "serviceName": "SampleApplication",
    "serviceVersion": "1.0.0",
    "callbackUrl": "https://172.16.3.21:7443"
}
```

A subscription will be created, but subscribers will NOT receive events as there are no valid eventType values. |
Users need to ensure that the EventType is valid. For a list of supported eventType values, refer to the REST API documentation in the RealTimeSpeech SDK. |

Some Unicode characters in a Query phrase cause Speech Search Engage failures on AAMS

<table>
<thead>
<tr>
<th>Tracking Number</th>
<th>Description</th>
<th>Impact</th>
<th>Workaround</th>
</tr>
</thead>
<tbody>
<tr>
<td>ASASNAPIN-1463</td>
<td>In very particular cases some Unicode characters can cause issues in Queries that are submitted for speech search. During testing it was identified that copying Windows 1252 encoded content such as a quote (i.e. ‘) got interpreted as â€™ in UTF-8 encoded content. The following link provides a chart of Windows 1253 characters interpreted as UTF-8 bytes: <a href="http://www.i18nqa.com/debug/utf8-debug.html">http://www.i18nqa.com/debug/utf8-debug.html</a></td>
<td>The Speech Search engine fails to engage on the Avaya Aura Media Server. An error with the message “Internal Media Server” is reported. The speech search fails.</td>
<td>Remove the offending Unicode character(s). Subsequent speech searches will work without the offending Unicode character(s).</td>
</tr>
</tbody>
</table>

Attempts to stop a speech search started on a call (callParty = mixed) are ineffective

<table>
<thead>
<tr>
<th>Tracking Number</th>
<th>Description</th>
<th>Impact</th>
<th>Workaround</th>
</tr>
</thead>
<tbody>
<tr>
<td>ZEPHYR-36769</td>
<td>If a speech search is started using the ‘mixed’ callParty, a stop request is ineffective. The search continues after the stop request and matches can still be made.</td>
<td>Speech search cannot be terminated on a call when it is started on the ‘mixed’ Call Party. Events for recognized phrases may be sent beyond the point in a call where they are desired. New speech searches cannot replace the first one.</td>
<td>Start speech searches using the ‘both’, ‘called’, or ‘calling’ callParty, if it’s likely that a speech search is to be stopped during a call.</td>
</tr>
</tbody>
</table>
Speech Services Snap-in

Second ASR stop request is not effective when started on callParty : 'both'

<table>
<thead>
<tr>
<th>Tracking Number</th>
<th>Description</th>
<th>Impact</th>
<th>Workaround</th>
</tr>
</thead>
<tbody>
<tr>
<td>ASASNPIN-1500</td>
<td>When a SpeechServices Automatic Speech Recognition (ASR) is initiated on both sides of a call, a stop request works to stop ASR on one side of the call only. A second stop request is ineffective, leaving the ASR playing on that side of the call.</td>
<td>A user cannot stop an ASR dialog playing on one side of the call if it has been started using ‘both’ as the call party.</td>
<td>Users will need to start ASR on each side separately using ‘called’ and ‘calling’ as the call party. Each side can be stopped separately if desired.</td>
</tr>
</tbody>
</table>

Missing ASR Exited Event after starting ASR on both parties and 1 party exits triggering Mid Call event

<table>
<thead>
<tr>
<th>Tracking Number</th>
<th>Description</th>
<th>Impact</th>
<th>Workaround</th>
</tr>
</thead>
<tbody>
<tr>
<td>ASASNPIN-1508</td>
<td>When stopping an Automatic Speech Recognition on individual participants after it was started on ‘both’, an EXITED event and a MIDCALL event are received for the first stop request and no further events are received for the second stop request (although it does stop).</td>
<td>There is no indication if the second ASR request has successful stopped or not.</td>
<td>There is no known workaround for this. We provide the following recommendation: If an EXITED event is important in a customer application and both sides are to be included, start ASR on the calling and called parties separately.</td>
</tr>
</tbody>
</table>

Possible to issue stop requests for ASR or TTS requestIds that doesn't exist

<table>
<thead>
<tr>
<th>Tracking Number</th>
<th>Description</th>
<th>Impact</th>
<th>Workaround</th>
</tr>
</thead>
<tbody>
<tr>
<td>ASASNPIN-1239</td>
<td>It is possible to issue stop requests for ASR and TTS with requestIds that don’t match values in the system.</td>
<td>No checking is performed on stop requests to ensure that stop requests are valid. This has no impact on the system.</td>
<td>Care should be taken not to issue invalid stop requests for ASR or TTS that contain invalid requestIds.</td>
</tr>
</tbody>
</table>
## Engagement Designer (Dynamic Tasks)

No Speech Search Matches are received when Starting a Speech Search using ED with AAMS Shuffling enabled.

<table>
<thead>
<tr>
<th>Tracking Number</th>
<th>ASASNAPIN-1503</th>
</tr>
</thead>
<tbody>
<tr>
<td>Related</td>
<td>ZEPHYR-37220</td>
</tr>
</tbody>
</table>

There are some limitations of the Avaya Aura Media Server Shuffling solution on EDP that may cause customers problems. In such cases, RealTimeSpeech provides customers the ability to turn off shuffling.

There is a known issue using RealTimeSpeech and Engagement Designer with AAMS Shuffling enabled. An intermittent Glare Exception on EDP may occur. (This is tracked as ZEPHYR-37220).

There is also an issue related to receiving RealTimeSpeech matches in Engagement Designer scenarios where call forwarding is used.

<table>
<thead>
<tr>
<th>Impact</th>
<th>Media Server Shuffling may not always work. Some scenarios for receiving RealTimeSpeech matches in ED may not work.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Workaround</td>
<td>Do not use AAMS Shuffling in such scenarios. Disable AAMS Shuffling for RealTimeSpeech by setting the “Enable Media Server Shuffling” value to “false” in System Manager on appropriate Attribute Configuration page at Elements -&gt; Engagement Development Platform -&gt; Configuration -&gt; Attributes. See the RealTimeSpeech deployment documentation for further details.</td>
</tr>
<tr>
<td></td>
<td>There is no known workaround for not receiving RealTimeSpeech matches in ED. This is a known limitation of the solution. Customers can instead use the dispatching solution from RealTimeSpeech to receive match results.</td>
</tr>
</tbody>
</table>
Sample Application (Query Management and Event viewer)

Intermittently, the Sample Application cannot start a speech search for extensions using Auto-Answer

<table>
<thead>
<tr>
<th>Tracking Number Description</th>
<th>ASASNAPIN-1470</th>
</tr>
</thead>
<tbody>
<tr>
<td>Description</td>
<td>In some cases using extensions with Auto Answer fails when using the Sample Application. This is related to the way that the Sample Application handles the order of events and in some cases drops events. The Sample Application is not designed for systems under load and therefore implements a throttle on handling of events. Sometimes the order and receipt of events are affected by this design.</td>
</tr>
<tr>
<td>Impact</td>
<td>There is no impact to the Auto Answer feature. It is only the case where the Sample Application is used to start a speech search on phones supporting Auto Answer. This issue can also occurs when using the sample application without auto answer but manually answering calls very quickly after set is ringing. The sample application does not work well when multiple events are generated in and around the same time – particularly with call answered events, as this impacts the ability to start search using the Sample Application.</td>
</tr>
<tr>
<td>Workaround</td>
<td>Do not use the Sample Application to start a speech search, if you want to use the Auto Answer feature on your phone or if trying to handle many calls.</td>
</tr>
</tbody>
</table>

Metadata editor is missing a scrollbar for greater than 10 items

<table>
<thead>
<tr>
<th>Tracking Number Description</th>
<th>ASASNAPIN-1375</th>
</tr>
</thead>
<tbody>
<tr>
<td>Description</td>
<td>The metadata panel in the Sample Application’s query wizard lacks a scroll-bar when attaching more than ~10 items of metadata to a single concept or phrases.</td>
</tr>
<tr>
<td>Impact</td>
<td>Greater than 10 metadata items do not work in the Sample Application.</td>
</tr>
<tr>
<td>Workaround</td>
<td>Do not use greater than 10 metadata items in the metadata panel of the Sample Application.</td>
</tr>
</tbody>
</table>

Real-Time Dashboard

3.1 Release Details

What's New in Real Time Speech 3.1?

Improved Speech Search Events

We have enhanced Speech Search Events produced during a Speech Search with supplementary information that supports building of richer client applications. We have also extended the Speech Search REST API to support application identification when starting a speech search. This is included as part of the speech search events and can be used to correlate the events to the application that started the speech search. We’ve also included a match reporting level option to control the amount of match events received. Furthermore, we have also introduced new Speech Search Query Events.

For further information, see the REST API documentation that is part of the Real Time Speech SDK.

Introducing Search Query Metadata

In previous releases of RealTimeSpeech, speech match events generated by the RealTimeSpeech snap-in contain the phrase or concept matched as well as confidence and temporal information. This by itself is somewhat useful, but in many cases, applications consuming this data need additional contextual information to make effective use of the match data provided. In 3.1, we provide the ability to associate arbitrary additional data points with speech search query concepts and phrases that can be acted upon by specific customer applications is needed. This means customers can now add semantic meaning to match results.

For further information, see the REST API documentation that is part of the Real Time Speech SDK.

Support for Mixed Audio Speech Searches

The RealTimeSpeech snap-in allows a speech search to be initiated that uses a mixed audio stream on the Avaya Aura Media Server. This is supported by the REST interfaces or by configuring a Service Profile parameter during auto start mode. It allows a user of the snap-in to search for terms for all parties on a shared (mixed) stream. Using a mixed stream has a major advantage in that it does reduce the overall number of speech search sessions required. However, there is a tradeoff. Match results will not be as accurate in mixed mode as using dedicated stream searches for each party.

Support for Larger Search Queries

In RealTimeSpeech 3.0, the size and number of queries transmitted to the Avaya Aura Media Server was restricted by a 6K limit on SIP message sizes. In the 3.1 release, the RealTimeSpeech snap-in now facilitates the passing of queries by reference to overcome this limitation. A URI is built and passed to the Avaya Aura Media Server. The Avaya Aura Media Server then retrieves queries over a REST interface
using a HTTPS GET request to the URI. This increases the overall size of queries supported. See the Query Sizing section in this document for further details.

**NOTE:** The 6k restriction still remains if a speech search uses the *inline query* mechanism over the REST API (See REST API documentation for further details). It is, therefore, recommended that customers use stored queries with tags, and start speech searches with the desired tags when using larger queries.

**Management Statistics for System Administrators**

The RealTimeSpeech snap-in supports interfaces that allow administrators to receive real time statistics about the overall state of the snap-in and supported speech search statistics. For further information, see the REST API documentation that is part of the Real Time Speech SDK.

**Support for Persisted Queries**

In 3.0, the RealTimeSpeech snap-in stored speech search queries in-memory in the EDP Data Grid (shared memory in an EDP cluster). In 3.1, EDP introduced a new feature that supported Data Grid persistence that enabled speech search queries to be persisted to a store.

We have made changes in RealTimeSpeech to support this. This feature is transparent to clients using the Queries REST API. The advantage is that when a single node or the entire cluster is rebooted, the queries are still available as they are retrieved from the store after the reboot.

**Media Server Improvements**

**Physical Servers**

EDP 3.1 introduced support for the deployment of Avaya Aura Media Servers on physical servers in addition to the existing VMWare OVA deployments offered. The RealTimeSpeech support the usage of the deployment of Avaya Aura Media Servers on physical servers as it allows for greater capacity. See the *Capacity* section in this document for further details.

**Media Server Shuffling**

EDP 3.1 introduced support for dynamically sequencing the Avaya Aura Media Server into an existing call. This reduces the need to have the Avaya Aura Media Server *always* on every call. Using Media Server Shuffling allows customers to add Media Server support to a call *as needed*. In this scenario, when a customer initiates a Speech Search the Media Server is automatically invited into the call. For further details, see the *AAMS Shuffling* section in this document.
Capacity

The following table describes the overall capacities of the Avaya Aura Media Servers.

<table>
<thead>
<tr>
<th>AAMS Server Type</th>
<th>Max Concurrent Channels per Server</th>
<th>Max Servers in a Cluster</th>
</tr>
</thead>
<tbody>
<tr>
<td>Virtual</td>
<td>85</td>
<td>8</td>
</tr>
<tr>
<td>Avaya Small Appliance</td>
<td>200</td>
<td>8</td>
</tr>
<tr>
<td>Avaya Large Appliance</td>
<td>600</td>
<td>8</td>
</tr>
</tbody>
</table>

**NOTE:** This table is based on monitoring of single speech search sessions on a call. The stated max concurrent channels per server would vary depending on the query size, quantity and expected speech matches of the query/queries in use. These benchmarks are based on tests using a single query with twenty phrases.

If customers exceed these figures they are liable to encounter the issues:

- ASASNAPIN-1501: Sometimes SPEECH_SEARCH_ERROR events are received with a “Redirect to or from Media Server” error reason when starting or stopping a Speech Search

Including Speech Search is a media intensive operation. Based on traffic testing in a lab configuration, exclusively running Real-Time Speech searches, EDP nodes using the General Profile configuration were individually capable of 350 calls with a speech search. To successfully achieve 630 speech searches on the Avaya Aura Media Server Large Appliance configuration, two EDP deployments of nodes running the General Purpose profile were required.
Certificate issue with System Manager
If you encounter issues logging into System Manager on Chrome or Firefox due to a "weak ephemeral Diffie-Hellman public key", this can be disabled on Firefox. (There is no known workaround for Chrome)

Firefox:
Type in your browser  about:config
Search for
- security.ssl3.dhe_rsa_aes_128_sha
- security.ssl3.dhe_rsa_aes_256_sha

Set them both to false (just double click to set them to false or true).

AAMS and CM interoperability
If you are encountering a SIP 488 Status error and you are using Communication Manager, you will need to configure
- Enforce SIPS URI for SRTP? y

AAMS Shuffling
RealTimeSpeech 3.1 supports shuffling of the Avaya Media Server into a call and is enabled by default. This means that AAMS is only invited into the call as needed and supports dynamic utilization of the Media Server resources. There are some limitations of this solution and it may not work for all customer scenarios. RealTimeSpeech provides a mechanism to disable this using System Manager.

To disable AAMS Shuffling for RealTimeSpeech by setting the “Enable Media Server Shuffling” value to “false” in System Manager on appropriate Attribute Configuration page at Elements -> Engagement Development Platform -> Configuration -> Attributes. See the RealTimeSpeech deployment documentation for further details.
Pass by Reference Solution requires trust between AAMS and EDP

In the 3.1 release, RealTimeSpeech now provides support for a larger queries solution using what we call a Pass by Reference implementation. This solution requires trust between AAMS and EDP as queries are now fetched directly from the RealTimeSpeech snap-in.

NOTE: The Pass by Reference solution is not applicable for inline queries. The full query will be passed to AAMS if using inline queries with a RESTful Start Speech Search request, and therefore subject to the 6K limitation size from prior releases. Customers are recommended to use stored queries and start speech searches using tags in order to avail of the Pass by Reference solution and the ability to support larger query sizes.

The details on how to setup RealTimeSpeech are provided in the deployment documentation.

Determining if you have an issue with EDP / AAMS trust not set up correctly

When trust is incorrectly setup, you will receive a SPEECH_SEARCH_ERROR after the speech search start is initiated (after the SPEECH_SEARCH_STARTED event). If you receive the following reason message in the event then you may have encountered a trust issue:

Speech Search failed:Internal media server error

To determine the exact cause of the Internal Media Server issue

- Enable AAMS debug logs
- Logging to AAMS and navigate to the /opt/avaya/app/amsinst/ma/MAS/common/log directory.
- Place a call and start a speech search
- Open the ivrmpDebug.txt file have a look at the output

Look for an (IVRMP_AUDIOMONITOR_ERROR related to an IVRMP_FETCH_FAILED to the RealTimeSpeech snap-in) error such as the following:

(10-15 09:45:39.192)<I,IvrMP,ESP-2,00000000-0000-0000-0000-000000000000> NWMsgTrace Message Sent to [127.0.0.1]
   RTYPE: IVRMP_EVENT
   RSTAT: IVRMP_AUDIOMONITOR_ERROR
   RTID : 1386
   RSID : 25423:1947:1405:0:
   PLSZ : 252
   PAYLOAD :
   aev_res=15:UrlRequestError^M
   aev_rslt=18:IVRMP_FETCH_FAILED^M
   aev_rsrc=176:https://10.135.63.62/services/RealTimeSpeech-3.1.0.0.78/speechsearch/fetch?id=tA8n_W6BR90f311qhpPRew&id=k9oPzDLWRpyEKDFY1LSYOA&id=CCsd3k2WQ9CziKhswT82bg&timestamp=1434015623662
If you find this, look for the corresponding IVRMP_COMMAND_RESPONSE earlier in the log:

(10-15 09:45:39.171)<I,IvrMP,IEP-2,00000000-0000-0000-0000-000000000000> NWMsgTrace Message Sent to [127.0.0.1]
  RTYPE: IVRMP_EVENT
  RSTAT: IVRMP_COMMAND_RESPONSE
  RTID : 1386
  RSID : 25423:1947:14050:
  PLSZ : 30
PAYLOAD :
aev_res=2:OK^M
aev_rslt=2:OK^M
(10-15 09:45:39.171)<I,IvrMP,IEP-2,6118fbee-268d-3ffe-a800-1c4d9e6feaa2>
(10-15 09:45:39.171)<I,IvrMP,IEP-2,6118fbee-268d-3ffe-a800-1c4d9e6feaa2>
(10-15 09:45:39.173)<E,IvrMP,NWCM,00000000-0000-0000-0000-000000000000> GET: failure after 0 bytes received for https://10.135.63.62/services/RealTimeSpeech-3.1.0.0.78/speechsearch/fetch?id=tA8n_W6BR90f31ljhPRew&id=k9oPzDLWRpyEKDFY1LSYOA&id=CCsd3k2WQ9Cz1KhsWT82bg&timestamp=1434015623662

If you see the following:

(10-15 09:45:39.173)<E,IvrMP,NWCM,00000000-0000-0000-0000-000000000000> CURLMSG = 1, result = Peer certificate cannot be authenticated with given CA certificates
(10-15 09:45:39.173)<E,IvrMP,NWCM,00000000-0000-0000-0000-000000000000> GET: failure after 0 bytes received for https://10.135.63.62/services/RealTimeSpeech-3.1.0.0.78/speechsearch/fetch?id=tA8n_W6BR90f31ljhPRew&id=k9oPzDLWRpyEKDFY1LSYOA&id=CCsd3k2WQ9Cz1KhsWT82bg&timestamp=1434015623662
GET: failure after 0 bytes received for
https://10.135.63.62/services/RealTimeSpeech-
3.1.0.0.78/speechsearch/fetch?id=tA8n_W6BR9of311qhpPRew&id=k9oPzDLWRpyEKDFY1LSYOA&id=CCsd3k2WQ9CzikhswT82bg&timestamp=1434015623662

This indicates you have a trust issue between EDP and AAMS.
**Query Sizing**

**Limitations on Overall Query Sizes**

Real Time Speech supports up to **25** individual queries for a call, including up to **200** phrases and a maximum depth of **5** levels of concepts. It is preferable to group small queries using tags, rather than using a top-level 'ALL' or 'AT_LEAST' operator concept.

**NOTE:** Inline Queries (queries included directly in Speech Search Start REST API request i.e. non stored queries) have a limit of **35** phrases.

**Limitations on Match Results**

RealTimeSpeech 3.1 currently provides the constituent matches for concept level match events. This means that for every higher level match (ALL, AT_LEAST, ANT, FIRST concepts) the child matches (phrases, or concepts) that asserted it are also reported.

There is a 4K limit (including SIP information and payload details) on delivery on match results from AAMS. The outcome of this is that match events from AAMS containing approximately more than 20 elements will not be reported to the RealTimeSpeech snap-in. Each element corresponds to either a single concept or single phrase match from a nested query.

This effects constituent (or child) matches from queries containing nested hierarchical concepts. Take a scenario, whereby, a query writer has created a query with a top level ALL concept containing more than 20 sub-phrases. When all 20 sub-phrases are matched by the engine, phrase match events are reported (when using all match reporting mode - summary match reporting mode will not report this). However, the top level ALL may not be reported as the constituent matches exceed that capacity allowed. The number of actual matches supported is a sizing issue and may or may not be smaller than the 20 match number. This is a guideline only.

It is recommended to use limited hierarchies, restricting ALL and AT_LEAST operators or multiple smaller queries can all work around this problem.

**Sample Application and Dashboard Usage**

The RealTimeSpeech Sample Application and Dashboard examples included in the SDK are intended to showcase features from RealTimeSpeech only. The Sample Application does not support multiple user sessions and is recommended for single browser usage only. Neither the Sample Application nor the Dashboard is engineered for high traffic scenarios. They are intended as example, and not production level, applications. The Sample Application Query Manager will work on a cluster with multi node EDP deployments. Users may experiences issues with using the Event and Search Manager of the Sample Application in EDP multi node deployments.
Problems with Avaya One X Communicator 6.2.6 as Softphone with EDP and ASR/TSS functionality

At the time of release, some issues were encountered using the Avaya One X Communicator 6.2.6 with EDP 3.1.1 and in particular using the Automatic Speech Recognition and Text to Speech features provided by the SpeechServices snap-in. This was encountered when AAMS Shuffling was enabled on RealTimeSpeech. The SDP from One X Communicator has an invalid “m=video” line in the SDP when EDP is trying to shuffle the AAMS into the call. Enabling or disabling video calling in One X Communicator does not appear to have any effect. It is recommended that customers use a different softphone such as the Avaya Communicator software, or if this is not possible, to disable AAMS Shuffling when using the Avaya One X Communicator softphone. This issue was verified on the 6.2.6.03-FP6 release of One X Communicator. Customers should also check for Avaya One X Communicator updates to address this issue.

Managing Subscriptions

Real Time Speech provides a subscription mechanism allowing customers to create subscriptions using the snap-in REST APIs. The snap-in also maintains a faulty subscription / endpoint policy wherein failures to send events to a particular callback URL over a period of 3 minutes consecutively would be marked as faulty. After a period of 30 minutes the callback URL is considered dead and the subscription is terminated.

However, if the callback URL is always available, subscriptions will be always renewed. It is recommended that customers unsubscribe from subscriptions which are no longer used, firstly to avoid confusion of multiple events, and secondly for snap-in performance reasons.

Furthermore, care must be taken with subscription types. RealTimeSpeech provides the ability to create general subscriptions such as subscribe for all calls and speech search events. In production scenarios, receiving this volume of events may be problematic, and the snap-in will not support many subscribers in this scenario. It is therefore, recommended to subscribe for only the required events. Subscriptions using filters are preferred rather than general subscriptions.

NOTE: Use subscription filters to receive only relevant events.

Active / Standby Database Failover for Real Time Speech Queries

While Real Time Speech provides persistence for queries in an EDP cluster with an Active / Standby database, there is a known issue whereby it is not possible to switch over to a standby database that is not ready. Customers need to power up the node with the active database first and let it synchronize. Also if both nodes fail at the same time, query data may be lost. It is therefore recommended, that customers export queries using the Export REST API of RealTimeSpeech to keep a backup of queries.

IMPORTANT: Customer should regularly export Real Time Speech queries.
Speech Search Preservation during Failover is not supported

RealTimeSpeech does not support Speech Search Preservation in the scenario an EDP or AAMS node fails. Speech Searches active on these nodes are lost. New calls are directed to other nodes in a HA scenario, and speech search function as expected. Calls and speech searches active on other AAMS and EDP nodes are unaffected.

Interoperability

Call Service Invocation Order for RealTimeSpeech and other snap-ins

RealTimeSpeech is a Call Intercept snap-in. If a customer wants to use RealTimeSpeech with other Call Intercept snap-ins, the order of the snap-in service invocation is important. This can be configured using the Service Profile Editor in System Manager. (Home / Elements / Engagement Development Platform / Configuration / Service Profiles). On the Service Invocation Details tab of a Service Profile it is possible to configure the Calling and Called Service Invocation Order.

It is important to consider where on your call flow you want to order the invocation of RealTimeSpeech. Take, for example, an Engagement Designer workflow. If you are using RealTimeSpeech tasks it is important to configure RealTimeSpeech first (or earlier) in the invocation order. This way the RealTimeSpeech service is ordered before the Engagement Designer workflow.

The same is true for other Avaya or third party developed Call Intercept snap-ins.

Called vs Answered Party Details in Events

RealTimeSpeech provides customers support for using EDP Call Events. Both the EDP Call Events and the RealTimeSpeech Speech Search Events include called party details in events. In some cases (for example, call forward scenarios), the called party details included in the events may be different to the actual answering party. This is a known limitation in the solution.

Documentation Errata

The following are a list of subsequent changes to official documentation.

HTTP or HTTPS limit on connections

In our documentation we recommended when creating an EDP cluster to use a HTTP or HTTPS limit on connections setting of 6000. This setting specifies the maximum number of connections that a single client (source IP address) can open. It turns out that connections are reused and that for most deployment scenarios the default of 3 is adequate.
Real Time Speech 3.0 Users: Using Real Time Speech 3.1 Event and REST API functionality

Updates for using the Speech Search Event Schema

RealTimeSpeech 3.1 snap-in provides improved Speech Search Events in JSON format. As a result, the event schema has been updated from 1.0 to 1.1. The snap-in no longer produces the 1.0 event format. The 1.1 schema has extended the 1.0 schema meaning that same 1.0 JSON properties are valid in the 1.1 schema. It is not expected to break existing customer use of the schema. However, depending on some JSON implementations, customers may find that parsing code may need to be updated to handle the newer 1.1 event schema features. The new 1.1 schema is provided as part of the SDK.

The 1.1 schema adds the following properties:
- **application** - A new property that contains the application identifier
- **tags** - This is an existing property included in the previous release 3.0 of RTS, but it was not used. This property is now populated in 3.1 release by the tags of all the queries undergoing a speech search, or in the case of match events, the tag of the match.
- **timestamp** - A new property that provides the UTC timestamp of when the event occurred. This is a long integer representation of the time.
- **queryIds** - A new property that contains the Ids of queries stored in the system used by the speech search, or in the case of match events, the queryId if the query where the match occurred.
- **eventingParty** - A new property that provides information from which party the event originated. For example, if the calling party uttered the phrase, then the calling party details will be included. The eventingParty will be similar to the existing callingParty and calledParty details and includes
  - **handle** of the eventing party
  - **domain** of the eventing party
  - **display** of the eventing party
- **The conceptMatches array property introduces two new properties:**
  - **metadata** - A new property that is an array of name and value attributes associated with the query. It supports the new Speech Search Query Metadata feature.
  - **id** - A new property that uniquely identifies the concept within the search query.

Updates for using the Real Time Speech REST APIs

RealTimeSpeech 3.1 snap-in provides improved Speech Search and Query Management functionality using the REST APIs. The JSON format has also been extended to support this for the
- Start Speech Search REST API Request,
- Stop Speech Search REST API Request
- Query format (request/response)

The JSON Start Speech Search REST API Request adds the following properties:
- **application** - A new property that set the application identifier provided in events
• **matchReporting** - A new property that determines the level of events received (supports: all or summary levels)

The JSON Queries format adds the following properties:
• **id** - A new property that uniquely identifies the concept within the search query. This is automatically generated when creating a query
• **metadata** - A new property that is an array of name and value attributes associated with the query. It supports the new Speech Search Query Metadata feature.

It is not expected to break existing customer use of the Speech Search Start and Queries API. However, depending on some JSON implementations, customers may find that parsing code may need to be updated to handle the newer 3.1 API JSON properties.

The JSON Stop Speech Search REST API Request has changed in 3.1. Due to an issue caused trying to stop a speech search in a cluster scenario it was necessary to locate the call information. Therefore, a **callId** property is now mandatory, and **searchIds** can be provided optionally in the Stop Speech Search REST API request.

**NOTE:** RealTimeSpeech 3.0 customers using the Stop Speech Search REST API feature will need to now include a **callId** in their REST requests.

For further information on events and REST API usage, refer to the SDK documentation.

**Real Time Speech 3.0 Users: Change to the Automatic Start of Speech Search System Manager Attributes**

The RealTimeSpeech 3.1 snap-in by default has the *Enable Automatic Start of Speech Search* attribute set to false. This is a change from 3.0, where automatic starting of speech search was enabled by default. 3.0 customers using this feature will need to enable this flag.

In addition, RealTimeSpeech 3.1 provides a new layout for attributes. Attributes are now grouped in sections. There is a known issue, for users of RealTimeSpeech 3.0 and 3.1 snap-ins installed – attributes will be duplicated. It is recommended to uninstall and delete the RealTimeSpeech 3.0 snap-in to remove the older attributes.
Troubleshooting / FAQ

The following should be validated when troubleshooting an issue with the Real-Time Speech Search Snap-in. Consult the Real-Time Speech Snap-in Reference guide for required steps. Here is an outline of what to do:

1.) Validate that the snap-in has been successfully installed, and that you can access the launch page.
2.) Validate that at least one query has been configured in the system before attempting to start a speech search.
3.) Validate that you can access all elements in the solution, and that all elements are operational (EDP, AAMS, SM).
4.) Validate that the required user sequencing has been configured for monitored endpoints. If sequencing has not been correctly configured, EDP will not be sequenced into the call for the purposes of orchestrating speech search.

General Tips

1.) Ensure that EDP is correctly configured for alarms. Test with the sample alarm.
2.) Check all logs (see further details below) for error and exception messages.
3.) Look at the SIP trace to confirm that EDP is sequenced in the call (4xx and 5xx messages will indicate an issue.

Using the Real-Time Speech Log Files

The Real-Time Speech snap-in generates a log file that can be used for troubleshooting purposes. You must connect to the EDP 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 Engagement Developer Platform page in System Manager, or alternatively, directly on the EDP server using the following command to turn the log level to ALL:

```
ce dlogon RealTimeSpeech
```

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

```
/var/log/Avaya/services/RealTimeSpeech/RealTimeSpeech.log
/var/log/Avaya/services/SpeechServices/SpeechServices.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
```
To turn the log level to default (INFO level):
  ce dlogoff 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

You may find relevant information in EDP general logs. The two most valuable log file commands are:
  • ce dlogv
  • ce alogv

Beyond this, using the ce-report command will collect all relevant logs from the system into a temporary directory.

**Validating AAMS is being successfully invoked for Speech Search**

Log into the AAMS 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 AAMS. (Note that no caller/called party identifying information will be visible here.)

**Error Accessing the Reference Applications**

If you see the following message when accessing either of the reference applications, there is likely a security issue:

Check the setting: “Client Certificate Challenge Enabled” in the following area:
Engagement Development Platform > Configuration > HTTP Security

Navigate to the cluster where Real-Time Speech is installed and un-check the “Client Certificate Challenge Enabled” tick box, before selecting commit.
Sequencing on EDP
For a complete set of details in relation to supported sequencing configurations on EDP, please see the following document (http://avayacollateral.netlabs.net/pdfoverview/UC7368.pdf) on call intercept services.

It is important to note that call intercept services will only work where one of the call parties calls comes in over a SIP trunk (e.g., PSTN trunk). You cannot sequence calls between two endpoints configured on the same CM. For lab simulations, there are two options:

1.) Set-up two independent Communication Managers (connected via SM trunks) and use endpoints connected to each to simulate a PSTN environment
2.) Have the calling party configured as a 3rd party SIP endpoint on Session Manager. When configuring the user on System Manager, ensure no application sequences or CM endpoint profiles are created:
### Session Manager Profile

**SIP Registration**
- Primary Session Manager: SGCESM
- Secondary Session Manager: 
- Survivability Server: 
- Max. Simultaneous Devices: 1
- Block New Registration When Maximum Registrations Active?: 

**Application Sequences**
- Origination Sequence: (None)
- Termination Sequence: (None)

**Call Routing Settings**
- Home Location: SIN
- Conference Factory Set: (None)

**Call History Settings**
- Enable Centralized Call History?: 

### Engagement Development Platform Profile

### CM Endpoint Profile
## Supported Language Packs

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