



Application Notes for configuring Mutare giSTT Speech to Text Connector Snap-in with Avaya Breeze™ R3.2.1 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate Mutare giSTT Speech to Text Connector Snap-in with Avaya Breeze™.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required to integrate Mutare giSTT Speech to Text Connector Snap-in with Avaya Breeze™.

giSTT Speech to Text Connector Snap-in is deployed on Avaya Breeze™. Once deployed, using Avaya Engagement Designer, workflow is created to use giSTT Speech to Text Connector Snap-in. The giSTT Speech to Text Connector for Avaya Breeze™ is an easy-to-use snap-in that developers can embed in any application that requires speech to text functionality. Using Mutare's simple API, embedded voice files can be converted to text with amazing speed.

2. General Test Approach and Test Results

The interoperability compliance testing included feature testing. The feature testing focused on giSTT Speech to Text Connector Snap-ins ability to covert speech to text.

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.

2.1. Interoperability Compliance Testing

Compliance Testing was mainly focused around giSTT snap-ins' ability to use provided input data (audio) and generate relevant results (text). All the audio files were generated using only English language. The testing included:

- Various types of audio files supported by giSTT as input and validating successful conversion
- Various types of Base64 audio supported by giSTT as input and validating successful conversion
- Providing invalid input values and validating appropriate error response
- Types of audio files included:
 - Voicemail .wav files from Avaya Aura® Messaging
 - Voicemail .wav files from Avaya Aura® Communication Manager Messaging
 - .wav files in 8 bits and 16 bits PCM format
 - .MP3 files in Base64 audio format

A sample workflow was generated using Engagement Designer to test the giSTT Speech to Text Connector Snap-in.

2.2. Test Results

The giSTT Speech to Text Connector Snap-in successfully passed compliance testing.

2.3. Support

For giSTT Speech to Text Connector Snap-in support, Mutare can be reached using the following methods:

- **Web:** <http://www.mutare.com/support.asp>
- **Phone:** +1-855-782-3890
- **Email:** snapstore@mutare.com

3. Reference Configuration

Figure 1 illustrates the test configuration used to verify the giSTT Speech to Text Connector Snap-in with Avaya Breeze™. The configuration consists of Avaya Aura® Session Manager, Avaya Aura® System Manager, and an Avaya Breeze™ server.

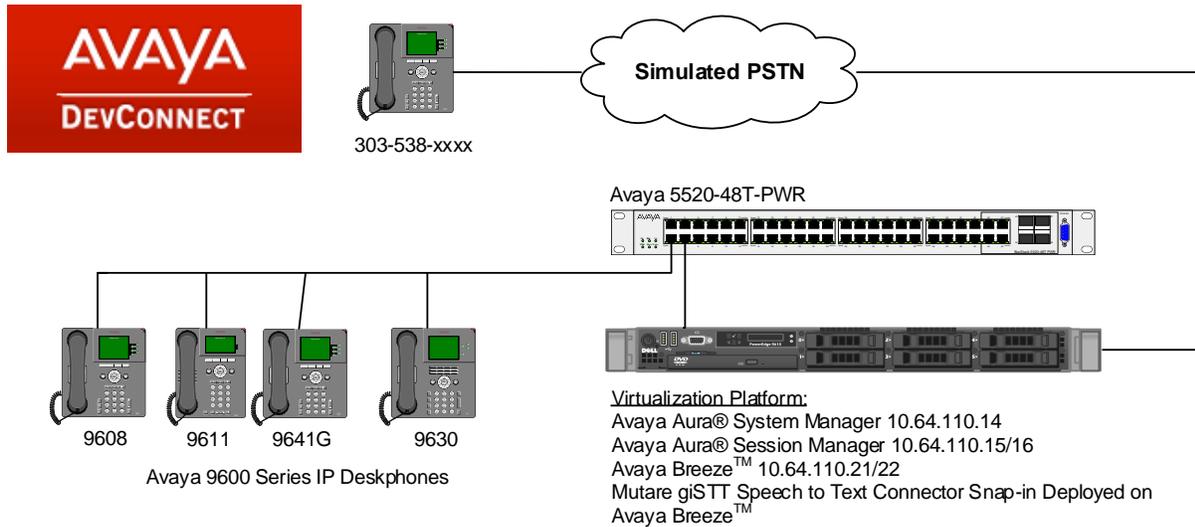


Figure 1: Mutare giSTT Speech to Text Connector Snap-in Lab Diagram

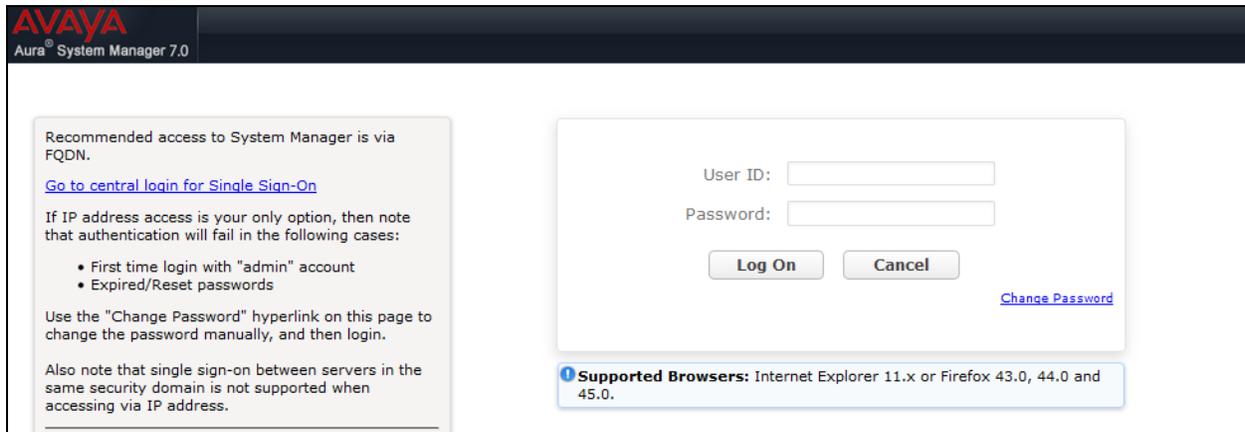
4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® System Manager	7.0.1.2
Avaya Aura® Session Manager	7.0.1.2.701230
Avaya Breeze™	3.2.1.0.1.320111
Avaya Engagement Designer	3.2.1.0.00033
giSTT Speech to Text Connector Snap-in deployed on Avaya Breeze™	1

5. Configure Avaya Breeze™ and Avaya Aura® Session Manager

Configuration of Avaya Breeze™ and Avaya Aura® Session Manager is performed via Avaya Aura® System Manager. Access the System Manager Administration web interface by entering <https://<ip-address>/SMGR> as the URL in a web browser, where <ip-address> is the IP address of System Manager. Log in using appropriate credentials.



AVAYA
Aura System Manager 7.0

Recommended access to System Manager is via FQDN.
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

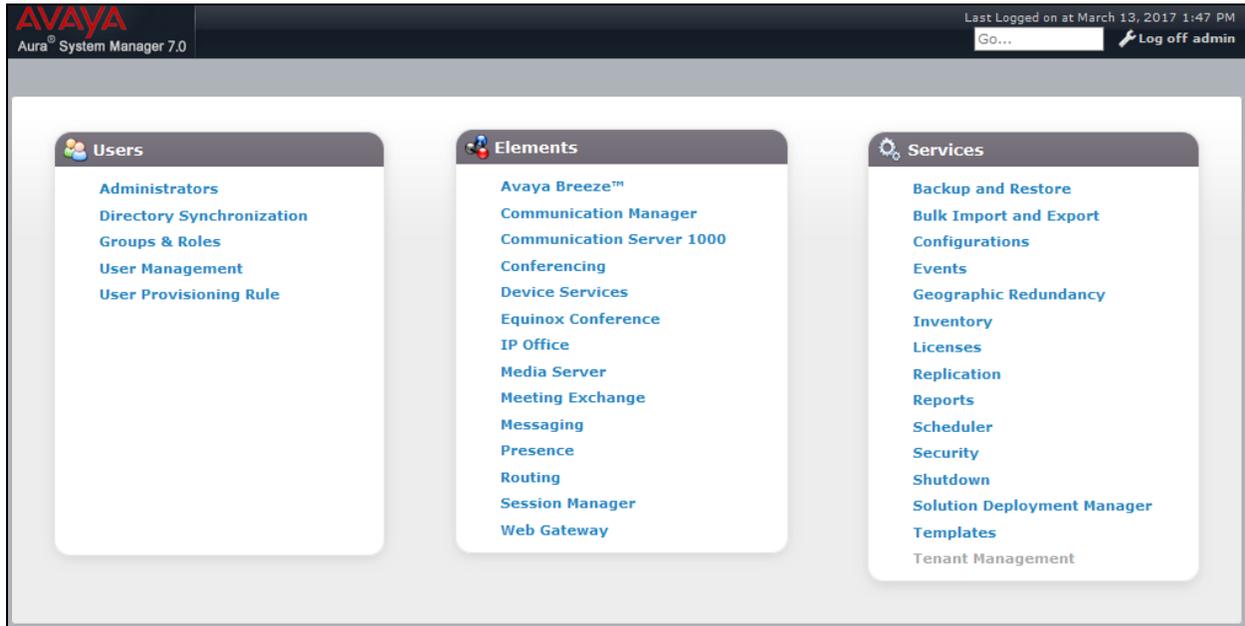
User ID:

Password:

[Change Password](#)

Supported Browsers: Internet Explorer 11.x or Firefox 43.0, 44.0 and 45.0.

Once logged in, the following screen is displayed.



5.1. Configure SIP Entities

Create a SIP Entity for Avaya Breeze™. Navigate to **Home** → **Elements** → **Routing** → **SIP Entities** and click the **New** button (not shown).

Enter a descriptive **Name** for the Avaya Breeze™ server and provide the **FQDN or IP Address** in the textbox. Select *Avaya Breeze* for **Type**. Default values may be used for the remaining fields.

SIP Entity Details

General

* Name:

* FQDN or IP Address:

Type:

Notes:

Scroll down to the **Entity Links** section. Enter a descriptive **Name**. Select the Session Manager SIP Entity for **SIP Entity 1**, and this Avaya Breeze™ SIP Entity for **SIP Entity 2**. Set the **Protocol** and **Port** (i.e. **TLS/5061**). Set the Connection Policy to *trusted*. Click **Commit**.

Entity Links

Override Port & Transport with DNS
SRV:

1 Item Filter:

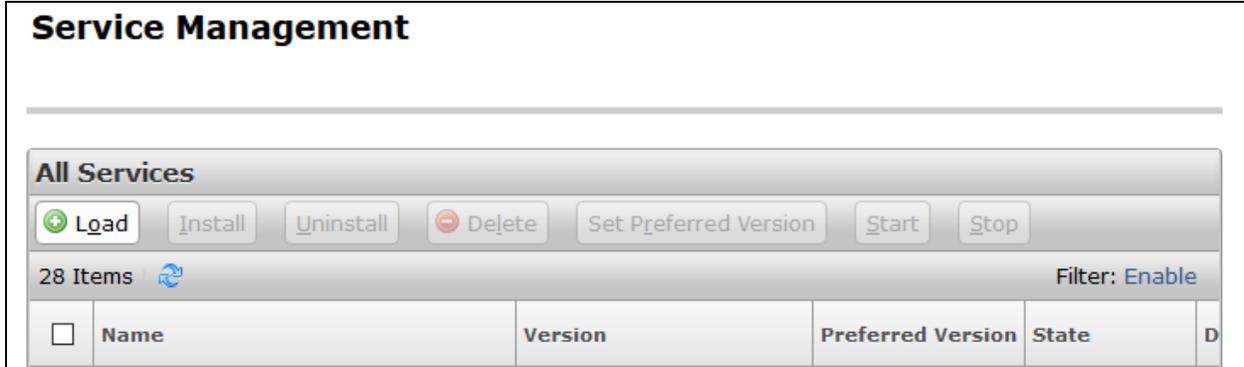
<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
<input type="checkbox"/>	* asm_abrz_5061_TLS	asm	TLS	* 5061	abrz	* 5061	trusted

< >

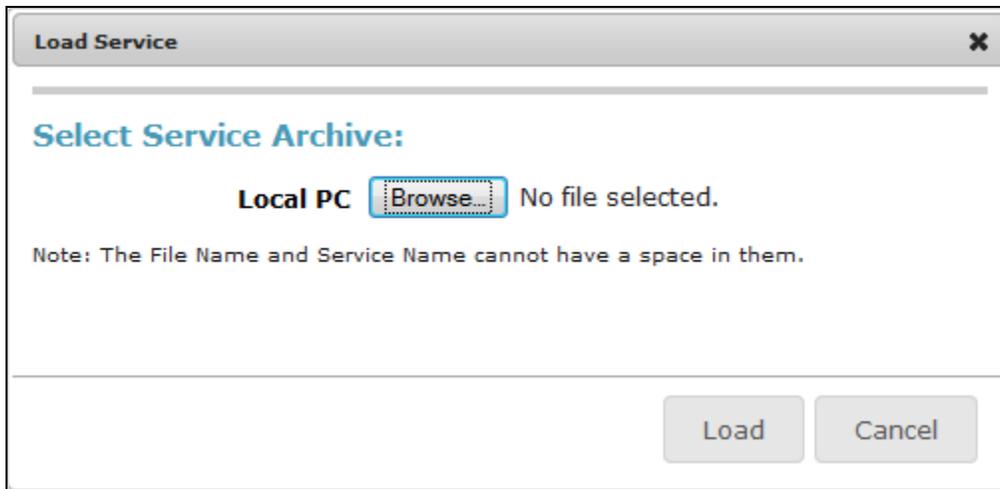
Select : All, None

5.2. Deploy giSTT Speech to Text Connector Snap-in

Obtain the giSTT Speech to Text Connector Snap-in, save the file to a local system. Navigate to **Home → Elements → Avaya Breeze™ → Service Management**. Click the **Load** button.



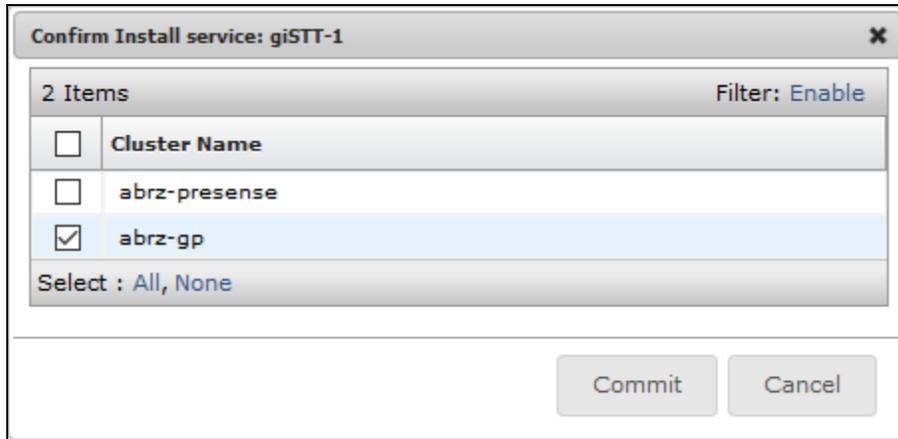
Click the **Browse** button, navigate to the giSTT Speech to Text Connector Snap-in svar file saved on the local system, and select it. Click the **Load** button to load the service.



The screen below shows **giSTT** version **1** has been loaded. Select the radio button to the left of the service and then click the **Install** button.

All Services						
<input type="button" value="Load"/> <input type="button" value="Install"/> <input type="button" value="Uninstall"/> <input type="button" value="Delete"/> <input type="button" value="Set Preferred Version"/> <input type="button" value="Start"/> <input type="button" value="Stop"/>						
21 Items						Filter: Enable
<input type="checkbox"/>	Name	Version	Preferred Version	State	Deployment Type	License Mode
<input type="checkbox"/>	PresenceServices	7.0.1.0.846		✓ Installed	Java	Not Applicable
<input type="checkbox"/>	Postgres	3.2.0.1.320110		✓ Installed	JDBC Provider	Not Applicable
<input type="checkbox"/>	mysql	3.2.0.1.320110		✓ Installed	JDBC Provider	Not Applicable
<input checked="" type="checkbox"/>	giSTT	1		✓ Loaded	Workflow	Not Applicable
<input type="checkbox"/>	EventingConnector	3.1.0.0.310007		✓ Loaded	Java	Not Applicable
<input type="checkbox"/>	EventingConnector	3.1.1.0.311008		✓ Installed	Java	Not Applicable

During compliance testing, the service was installed on a single Breeze Server within a cluster named **abrz-gp**; SIP Entity referenced in **Section 5.1** (i.e. *abrz*). Select the cluster of server where the service will be installed and click the **Commit** button.

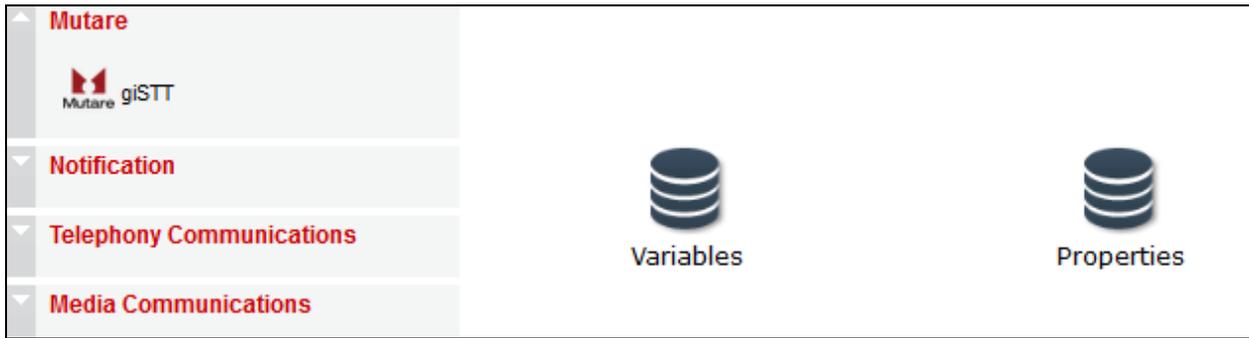


The screen below shows **giSTT** version **1** has been installed.

All Services						
<input type="button" value="Load"/> <input type="button" value="Install"/> <input type="button" value="Uninstall"/> <input type="button" value="Delete"/> <input type="button" value="Set Preferred Version"/> <input type="button" value="Start"/> <input type="button" value="Stop"/>						
21 Items Filter: Enable						
<input type="checkbox"/>	Name	Version	Preferred Version	State	Deployment Type	License Mode
<input type="checkbox"/>	PresenceServices	7.0.1.0.846		✓ Installed	Java	Not Applicable
<input type="checkbox"/>	Postgres	3.2.0.1.320110		✓ Installed	JDBC Provider	Not Applicable
<input type="checkbox"/>	mySQL	3.2.0.1.320110		✓ Installed	JDBC Provider	Not Applicable
<input type="checkbox"/>	giSTT	1		✓ Installed	Workflow	Not Applicable
<input type="checkbox"/>	EventingConnector	3.1.0.0.310007		✓ Loaded	Java	Not Applicable

5.2.1. Generate a workflow

Create a workflow that uses the giSTT Speech to Text Connector Snap-in using the Avaya provided Engagement Designer snap-in. Via a browser, log onto the Engagement Designer portal. Once logged in, note that the giSTT Speech to Text Connector Snap-in is displayed on the left.



Generate a workflow and configure the giSTT Speech to Text Connector Snap-in as follows:

- **Account ID:** The account ID provided by Mutare.
- **Account Token:** The account token provided by Mutare.
- **Audio URL:** Full URL of location of sound file to be transcribed. This should be a full HTTP URL linking to a sound file. Not necessary if using **Base-64 Audio**.
- **Base-64 Audio:** A base 64 encoded string of the sound file to be decoded. Not necessary if using **Audio URL**.
- **Base-64 Type:** The format of sound files passed by base 64 needs to be identified. Enter .MP3 or .WAV. Default is .WAV. Required if using **Base-64 Audio**.
- **External ID:** Type in a desired ID.
- **Language:** Type in **en-US**.
- **Timeout:** Default value of **30**.

The screenshot shows a dialog box titled "giSTT properties". At the top, there is a "Label:" field with the value "giSTT". Below this is a section titled "Properties" containing several input fields:

- Account ID: AccountID
- Account Token: AccountToken
- Audio URL: http://www.mutare.com/images/r/
- Base-64 Audio: UklGRjJWBABXQVZFZm10IBIAAAA
- Base-64 Type: .WAV
- External ID: DevConnectTest
- Language: en-US
- Timeout (Sec): 30

Below the "Properties" section are two expandable sections: "Input Mapper Variables" and "Output Mapper Variables". At the bottom of the dialog are four buttons: "Input Mapping", "Output Mapping", "OK", and "Cancel".

Depending on the workflow, **Input Mapping** or **Output Mapping** can be configured to pass in values to another event or task. The input and output parameters provided by giSTT snap-in are:

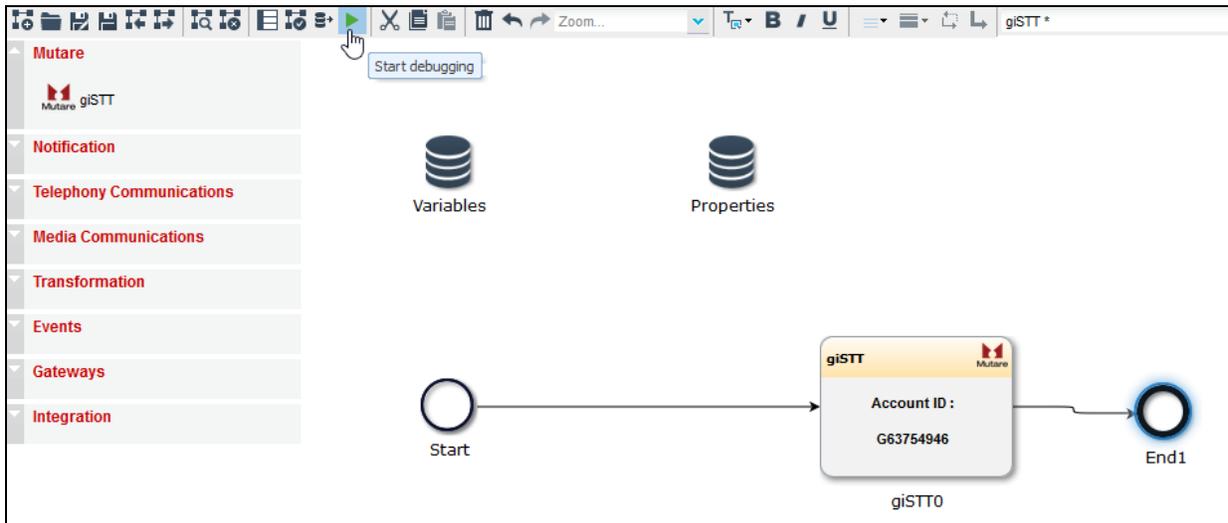
InputSchema	
• InputSchema:object	
- accountId:	string
- accountToken:	string
- audioURL:	string
- audioBase64:	string
- base64Type:	string
- externalId:	string
- language:	string
- timeoutSec:	string

OutputSchema	
• OutputSchema:object	
- STTId:	string
- Message:	string
- ExternalId:	string
- TranscriptionText:	string
- MessageLength:	number
- Status:	number
- Confidence:	number

6. Verification Steps

This section includes steps that can be followed to verify the configuration.

Log onto Engagement Designer portal and create a sample workflow. To verify if giSTT snap-in is able to successfully convert speech to text, provide an **Audio URL** or **Base-64 Audio** supported by giSTT snap-in. Select **Start Debugging**.



Once the debugging is completed, verify the converted text in the **Variables** section of **Debugging Console**.

Debugger Console

Instance status :Completed

Selected node status

Name	giSTT0
State	Completed

Variables:

system	system <input type="button" value="Expand"/>
TextMessage	TextMessage <input type="button" value="Collapse"/> <div style="border: 1px solid gray; padding: 5px; margin-top: 5px;"> <p>NewString1</p> <p>Hi, this is Marsha. I'm di</p> </div>

Output data:

Output	OutputSchema <input type="button" value="Collapse"/> <div style="border: 1px solid gray; padding: 5px; margin-top: 5px;"> <p>STTId</p> <p>ShLMm1geGDy</p> <p>ExternalId</p> <p>DevConnectTest</p> <p>TranscriptionText</p> <p>Hi, this is Marsha. I'm di</p> <p>MessageLength</p> <p>35520</p> <p>Status</p> <p>102</p> </div>
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7. Conclusion

The giSTT Speech to Text Connector Snap-in passed compliance testing. These Application Notes describe the procedures required for the giSTT Speech to Text Connector Snap-in to interoperate with Avaya Breeze™ to support the reference configuration shown in **Figure 1**. Refer to **Section 2.2** for testing result details and any observations noted during testing.

8. Additional References

Product documentation for Avaya products may be found at: <http://support.avaya.com>.

[1] Administering Avaya Aura® Avaya Breeze™, Release 3.2, Issue 1, October 2016.

[2] Administering Avaya Aura® Session Manager, Release 7.0.1, May 2016.

Product information for giSTT may be obtained by contacting Mutare directly.

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