Powered by Universal Speech Solutions LLC

UniMRCP

Azure SR Plugin

Usage Guide

Revision: 15

Created: October 26, 2017

Last updated: February 26, 2021

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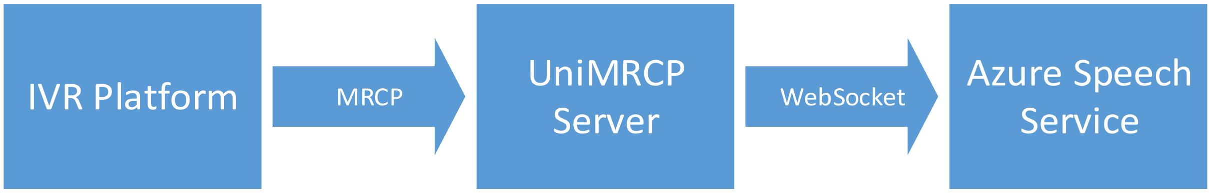
1 Overview

This guide describes how to configure and use the Microsoft Azure Speech Recognition (SR) plugin to the UniMRCP server. The document is intended for users having a certain knowledge of Microsoft Azure Speech API and UniMRCP.



1.1 Installation

For installation instructions, use one of the guides below.

- RPM Package Installation (Red Hat / Cent OS)
- Deb Package Installation (Debian / Ubuntu)

1.2 Applicable Versions

Instructions provided in this guide are applicable to the following versions.

UniMRCP 1.5.0 and above

UniMRCP Azure SR Plugin 1.0.0 and above

2 Supported Features

This is a brief check list of the features currently supported by the UniMRCP server running with the Azure SR plugin.

2.1 MRCP Methods

- ✓ DEFINE-GRAMMAR
- ✓ RECOGNIZE
- ✓ START-INPUT-TIMERS
- ✓ STOP
- ✓ SET-PARAMS
- ✓ GET-PARAMS

2.2 MRCP Events

- ✓ RECOGNITION-COMPLETE
- ✓ START-OF-INPUT

2.3 MRCP Header Fields

- ✓ Input-Type
- ✓ No-Input-Timeout
- ✓ Recognition-Timeout
- ✓ Speech-Complete-Timeout
- ✓ Speech-Incomplete-Timeout
- ✓ Waveform-URI
- ✓ Media-Type
- ✓ Completion-Cause
- ✓ Confidence-Threshold
- ✓ Start-Input-Timers
- ✓ DTMF-Interdigit-Timeout
- ✓ DTMF-Term-Timeout
- ✓ DTMF-Term-Char
- ✓ Save-Waveform
- ✓ Speech-Language
- ✓ Cancel-If-Queue

- ✓ Sensitivity-Level

2.4 Grammars

- ✓ Built-in speech transcription grammar
- ✓ Built-in/embedded DTMF grammar
- ✓ SRGS XML (limited support)

2.5 Results

- ✓ NLSML
- ✓ Microsoft Speech JSON

3 Configuration Format

The configuration file of the Azure SR plugin is located in */opt/unimrcp/conf/umsazuresr.xml*. The configuration file is written in XML.

3.1 Document

The root element of the XML document must be <*umsazuresr*>.

Attributes

Name	Unit	Description
license-file	File path	Specifies the license file. File name may include patterns containing '*' sign. If multiple files match the pattern, the most recent one gets used.
subscription-key-file	File path	Specifies the Microsoft subscription key file to use. File name may include patterns containing '*' sign. If multiple files match the pattern, the most recent one gets used.

Parent

None.

Children

Name	Unit	Description
<ws-streaming-recognition>	String	Specifies parameters of streaming recognition employed via Microsoft Speech WebSocket protocol.
<speech-contexts>	String	Contains a list of speech contexts. Available since Azure SR 1.5.0.
<speech-dtmf-input-detector>	String	Specifies parameters of the speech and DTMF input detector.
<utterance-manager>	String	Specifies parameters of the utterance manager.
<rdr-manager>	String	Specifies parameters of the Recognition Details Record (RDR) manager.

<monitoring-agent>	String	Specifies parameters of the monitoring manager.
<license-server>	String	Specifies parameters used to connect to the license server. The use of the license server is optional.

Example

This is an example of a bare document.

```
< umsazuresr license-file="umsazuresr_*.lic" subscription-key-
file="cognitive.subscription.key">
</ umsazuresr>
```

3.2 Streaming Recognition

This element specifies parameters of Microsoft WebSocket streaming recognition.

Attributes

Name	Unit	Description
language	String	Specifies the default language to use, if not set by the client. For a list of supported languages, visit https://docs.microsoft.com/en-us/azure/cognitive-services/speech/api-reference-rest/supportedlanguages
max-alternatives	Integer	Specifies the maximum number of speech recognition result alternatives to be returned. Can be overridden by client by means of the header field *N-Best-List-Length*.
alternatives-below-threshold	Boolean	Specifies whether to return speech recognition result alternatives with the confidence score below the confidence threshold. Available since Azure SR 1.5.0.
confidence-format	String	Specifies the format of the confidence score to be returned (use "auto" for a format based on protocol version, "mrcpv2" for a float value in the range of 0..1, "mrcpv1" for an integer value in the range of 0..100). Available since Azure SR 1.5.0.
results-format	String	Specifies the format of results to be returned to the client (use "standard" for NLSML and "transparent" for Microsoft JSON).

start-of-input	String	Specifies the source of start of input event sent to the client (use "service-originated" to rely on service-originated startDetected event and "internal" for plugin-originated event).
skip-unsupported-grammars	Boolean	Specifies whether to skip or raise an error while referencing a malformed or not supported grammar. Available since Azure SR 1.4.0.
transcription-grammar	String	Specifies the name of the built-in speech transcription grammar. The grammar can be referenced as *builtin:speech/transcribe* or *builtin:grammar/transcribe*, where *transcribe* is the default value of this parameter. Available since Azure SR 1.4.0.
auth-validation-period	Integer	Specifies a period in seconds used to re-validate access token based on subscription key. The lifetime of retrieved access token is set to 10 min by Microsoft.
http-proxy	String	Specifies the URI of HTTP proxy, if used. Available since Azure SR 1.8.0.
inter-result-timeout	Time interval [msec]	Specifies a timeout between interim results containing transcribed speech. If the timeout is elapsed, input is considered complete. The timeout defaults to 0 (disabled). Available since Azure SR 1.12.0.
input-token	String	Specifies a token in the JSON structure received from the service used to compose the input in the NLSML results. Defaults to 'Lexical'. Can be overridden by client. Available since Azure SR 1.13.0.
instance-token	String	Specifies a token in the JSON structure received from the service used to compose the instance in the NLSML results. Defaults to 'ITN'. Can be overridden by client. Available since Azure SR 1.13.0.
single-utterance	Boolean	Specifies whether to detect a single spoken utterance or perform continuous recognition. Available since Azure SR 1.15.0.

Parent

<umsazuresr>

Children

None.

Example

This is an example of streaming recognition element.

```
<ws-streaming-recognition
   language="en-US"
   max-alternatives="1"
   alternatives-below-threshold="false"
   confidence-format="auto"
   results-format="standard"
   start-of-input="service-originated"
   skip-unsupported-grammars="true"
   transcription-grammar="transcribe"
   auth-validation-period="480"
/>
```

3.3 Speech Contexts

This element specifies a list of speech contexts.

Availability

>= Azure SR 1.5.0.

Attributes

None.

Parent

<umsazuresr>

Children

<speech-context>

Example

The example below defines a speech contexts *directory*.

```
<speech-contexts>
  <speech-context id="directory" speech-complete="true" enable="true">
    <phrase>call Steve</phrase>
    <phrase>call John</phrase>
    <phrase>dial 5</phrase>
    <phrase>dial 6</phrase>
  </speech-context>
</speech-contexts>
```

3.4 Speech Context

This element specifies a speech context.

Availability

>= Azure SR 1.5.0.

Attributes

Name	Unit	Description
id	String	Specifies a unique string identifier of the speech context to be referenced by the MRCP client.
enable	Boolean	Specifies whether the speech context is enabled or disabled.
speech-complete	Boolean	Specifies whether to complete input as soon as an interim result matches one of the specified phrases.
language	String	The language the phrases are defined for. Available since Azure SR 1.6.0.
scope	String	Specifies a scope of the speech context, which can be set to either *hint* or *strict*. Available since Azure SR 1.7.0.

Parent

<speech-contexts>

Children

<phrase>

Example

This is an example of speech context element.

```
<speech-context id="directory" speech-complete="true" enable="true">
  <phrase>call Steve</phrase>
  <phrase>call John</phrase>
  <phrase>dial 5</phrase>
  <phrase>dial 6</phrase>
</speech-context>
```

3.5 Phrase

This element specifies a phrase in the speech context.

Availability

>= Azure SR 1.5.0.

Attributes

Name	Unit	Description
tag	String	Specifies an optional arbitrary string identifier to be returned as an instance in the NLSML result, if the transcription result matches the phrase.

Parent

<speech-context>

Children

None.

This is an example of a speech context with phrases having tags specified. Available since GSR 1.9.0.

```
<speech-context id="boolean" speech-complete="true" scope="strict" enable="true">
  <phrase tag="true">yes</phrase>
  <phrase tag="true">sure</phrase>
  <phrase tag="true">correct</phrase>
  <phrase tag="false">no</phrase>
  <phrase tag="false">not sure</phrase>
  <phrase tag="false">incorrect </phrase>
</speech-context>
```

3.6 Speech and DTMF Input Detector

This element specifies parameters of the speech and DTMF input detector.

Attributes

Name	Unit	Description
vad-mode	Integer	Specifies an operating mode of VAD in the range of [0 ... 3]. Default is 1.
speech-start-timeout	Time interval [msec]	Specifies how long to wait in transition mode before triggering a start of speech input event.

speech-complete-timeout	Time interval [msec]	Specifies how long to wait in transition mode before triggering an end of speech input event. The complete timeout is used when there is an interim result available.
speech-incomplete-timeout	Time interval [msec]	Specifies how long to wait in transition mode before triggering an end of speech input event. The incomplete timeout is used as long as there is no interim result available. Afterwards, the complete timeout is used. Available since Azure SR 1.2.0.
noinput-timeout	Time interval [msec]	Specifies how long to wait before triggering a no-input event.
input-timeout	Time interval [msec]	Specifies how long to wait for input to complete.
dtmf-interdigit-timeout	Time interval [msec]	Specifies a DTMF inter-digit timeout.
dtmf-term-timeout	Time interval [msec]	Specifies a DTMF input termination timeout.
dtmf-term-char	Character	Specifies a DTMF input termination character.
speech-leading-silence	Time interval [msec]	Specifies desired silence interval preceding spoken input.
speech-trailing-silence	Time interval [msec]	Specifies desired silence interval following spoken input.
speech-output-period	Time interval [msec]	Specifies an interval used to send speech frames to the recognizer.

Parent

<umsazuresr>

Children

None.

Example

The example below defines a typical speech and DTMF input detector having the default parameters set.

```
<speech-dtmf-input-detector
  vad-mode="2"
```

```
      speech-start-timeout="300"
      speech-complete-timeout="1000"
      speech-incomplete-timeout="3000"
      noinput-timeout="5000"
      input-timeout="10000"
      dtmf-interdigit-timeout="5000"
      dtmf-term-timeout="10000"
      dtmf-term-char=""
      speech-leading-silence="300"
      speech-trailing-silence="300"
      speech-output-period="200"
    />
```

3.7 Utterance Manager

This element specifies parameters of the utterance manager.

Attributes

Name	Unit	Description
save-waveforms	Boolean	Specifies whether to save waveforms or not.
purge-existing	Boolean	Specifies whether to delete existing records on start-up.
max-file-age	Time interval [min]	Specifies a time interval in minutes after expiration of which a waveform is deleted. Set 0 for infinite.
max-file-count	Integer	Specifies the max number of waveforms to store. If reached, the oldest waveform is deleted. Set 0 for infinite.
waveform-base-uri	String	Specifies the base URI used to compose an absolute waveform URI.
waveform-folder	Dir path	Specifies a folder the waveforms should be stored in.
file-prefix	String	Specifies a prefix used to compose the name of the file to be stored. Defaults to 'umsazuresr-', if not specified.
use-logging-tag	Boolean	Specifies whether to use the MRCP header field Logging-Tag, if present, to compose the name of the file to be stored. Available

		since Azure SR 1.12.0.

Parent

<umsazuresr>

Children

None.

Example

The example below defines a typical utterance manager having the default parameters set.

```
<utterance-manager
  save-waveforms="false"
  purge-existing="false"
  max-file-age="60"
  max-file-count="100"
  waveform-base-uri="http://localhost/utterances/"
  waveform-folder=""
/>
```

3.8 RDR Manager

This element specifies parameters of the Recognition Details Record (RDR) manager.

Attributes

Name	Unit	Description
save-records	Boolean	Specifies whether to save recognition details records or not.
purge-existing	Boolean	Specifies whether to delete existing records on start-up.
max-file-age	Time interval [min]	Specifies a time interval in minutes after expiration of which a record is deleted. Set 0 for infinite.
max-file-count	Integer	Specifies the max number of records to store. If reached, the oldest record is deleted. Set 0 for infinite.
record-folder	Dir path	Specifies a folder to store recognition details records in. Defaults to

		${UniMRCPInstallDir}/var.
file-prefix	String	Specifies a prefix used to compose the name of the file to be stored. Defaults to 'umsazuresr-', if not specified.
use-logging-tag	Boolean	Specifies whether to use the MRCP header field Logging-Tag, if present, to compose the name of the file to be stored. Available since Azure SR 1.12.0.

Parent

<umsazuresr>

Children

None.

Example

The example below defines a typical utterance manager having the default parameters set.

```
<rdr-manager
  save-records="false"
  purge-existing="false"
  max-file-age="60"
  max-file-count="100"
  waveform-folder=""
/>
```

3.9 Monitoring Agent

This element specifies parameters of the monitoring agent.

Attributes

Name	Unit	Description
refresh-period	Time interval [sec]	Specifies a time interval in seconds used to periodically refresh usage details. See <usage-refresh-handler>.

Parent

<umsazuresr>

Children

<usage-change-handler>
<usage-refresh-handler>

Example

The example below defines a monitoring agent with usage change and refresh handlers.

```
<monitoring-agent refresh-period="60">

  <usage-change-handler>
    <log-usage enable="true" priority="NOTICE"/>
  </usage-change-handler>

  <usage-refresh-handler>
    <dump-channels enable="true" status-file="umsazuresr-channels.status"/>
  </usage-refresh-handler >

</monitoring-agent>
```

3.10 Usage Change Handler

This element specifies an event handler called on every usage change.

Attributes

None.

Parent

<monitoring-agent>

Children

<log-usage>
<update-usage>
<dump-channels>

Example

This is an example of the usage change event handler.

```
<usage-change-handler>
  <log-usage enable="true" priority="NOTICE"/>
  <update-usage enable="false" status-file="umsazuresr-usage.status"/>
  <dump-channels enable="false" status-file="umsazuresr-channels.status"/>
</usage-change-handler>
```

3.11 Usage Refresh Handler

This element specifies an event handler called periodically to update usage details.

Attributes

None.

Parent

<monitoring-agent>

Children

<log-usage>
<update-usage>
<dump-channels>

Example

This is an example of the usage change event handler.

```
<usage-refresh-handler>
  <log-usage enable="true" priority="NOTICE"/>
  <update-usage enable="false" status-file="umsazuresr-usage.status"/>
  <dump-channels enable="false" status-file="umsazuresr-channels.status"/>
</usage-refresh-handler>
```

3.12 License Server

This element specifies parameters used to connect to the license server.

Attributes

Name	Unit	Description
enable	Boolean	Specifies whether the use of license server is enabled or not. If enabled, the license-file attribute is not honored.
server-address	String	Specifies the IP address or host name of the license server.
certificate-file	File path	Specifies the client certificate used to connect to the license server. File name may include patterns containing a '*' sign. If multiple files match the pattern, the most

		recent one gets used.
ca-file	File path	Specifies the certificate authority used to validate the license server.
channel-count	Integer	Specifies the number of channels to check out from the license server. If not specified or set to 0, either all available channels or a pool of channels will be checked based on the configuration of the license server.
http-proxy-address	String	Specifies the IP address or host name of the HTTP proxy server, if used. Available since 1.11.0.
http-proxy-port	Integer	Specifies the port number of the HTTP proxy server, if used. Available since 1.11.0.

Parent

<umsazuresr>

Children

None.

Example

The example below defines a typical configuration which can be used to connect to a license server located, for example, at 10.0.0.1.

```
<license-server
  enable="true"
  server-address="10.0.0.1"
  certificate-file="unilic_client_*.crt"
  ca-file="unilic_ca.crt"
/>
```

For further reference to the license server, visit

http://unimrcp.org/licserver

4 Configuration Steps

This section outlines common configuration steps.

4.1 Using Default Configuration

The default configuration should be sufficient for the general use.

4.2 Specifying Recognition Language

Recognition language can be specified by the client per MRCP session by means of the header field *Speech-Language* set in a *SET-PARAMS* or *RECOGNIZE* request. Otherwise, the parameter *language* set in the configuration file *umsazuresr.xml* is used. The parameter defaults to *en-US*.
For supported languages and their corresponding codes, visit the following link.

> https://docs.microsoft.com/en-us/azure/cognitive-services/speech-service/language-support#speech-to-text

Since Azure SR 1.6.0, the recognition language can also be set by the attribute *xml:lang* specified in the SRGS grammar. For example:

```
<?xml version="1.0" encoding="UTF-8"?>
<grammar mode="voice" root="transcribe" version="1.0"
        xml:lang="en-AU"
        xmlns="http://www.w3.org/2001/06/grammar">
  <meta name="scope" content="builtin"/>
  <rule id="transcribe"><one-of/></rule>
</grammar>
```

4.3 Specifying Sampling Rate

Sampling rate is determined based on the SDP negotiation. Refer to the configuration guide of the UniMRCP server on how to specify supported encodings and sampling rates to be used in communication between the client and server.

Since the Azure Speech API supports PCM audio sampled at 16 kHz only, if the RTP session is established with 8 kHz, then the audio data is upsampled to 16 kHz by the plugin.

4.4 Specifying Speech Input Parameters

While the default parameters specified for the speech input detector are sufficient for the general use, various parameters can be adjusted to better suit a particular requirement.

- speech-start-timeout

This parameter is used to trigger a start of speech input. The shorter is the timeout, the sooner a *START-OF-INPUT* event is delivered to the client. However, a short timeout may also lead to a false positive. Note that if the *start-of-input* parameter in the *ws-streaming-recognition* is set to *service-originated*, then a START-OF-INPUT event is sent to the client at a later stage, upon reception of a *speech.startDetected* response from the service.

- speech-complete-timeout

This parameter is used to trigger an end of speech input. The shorter is the timeout, the shorter is the response time. However, a short timeout may also lead to a false positive.
Note that both events, an expiration of the speech complete timeout and a *speech.endDetected* response delivered from the service, are monitored to trigger an end of speech input, on whichever comes first basis. In order to rely solely on an event delivered from the speech service, the parameter *speech-complete-timeout* needs to be set to a higher value.

- vad-mode

This parameter is used to specify an operating mode of the Voice Activity Detector (VAD) within an integer range of [0 … 3]. A higher mode is more aggressive and, as a result, is more restrictive in reporting speech. The parameter can be overridden per MRCP session by setting the header field *Sensitivity-Level* in a *SET-PARAMS* or *RECOGNIZE* request. The following table shows how the *Sensitivity-Level* is mapped to the *vad-mode*.

Sensitivity-Level	Vad-Mode
[0.00 … 0.25)	0
[0.25 … 0.50)	1
[0.50 … 0.75)	2
[0.75 … 1.00]	3

4.5 Specifying DTMF Input Parameters

While the default parameters specified for the DTMF input detector are sufficient for the general use, various parameters can be adjusted to better suit a particular requirement.

- dtmf-interdigit-timeout

This parameter is used to set an inter-digit timeout on DTMF input. The parameter can be overridden per MRCP session by setting the header field *DTMF-Interdigit-Timeout* in a *SET-PARAMS* or *RECOGNIZE* request.

- dtmf-term-timeout

This parameter is used to set a termination timeout on DTMF input and is in effect when dtmf-term-char is set and there is a match for an input grammar. The parameter can be overridden per MRCP session by setting the header field *DTMF-Term-Timeout* in a *SET-PARAMS* or *RECOGNIZE* request.

- dtmf-term-char

This parameter is used to set a character terminating DTMF input. The parameter can be overridden per MRCP session by setting the header field *DTMF-Term-Char* in a *SET-PARAMS* or *RECOGNIZE* request.

4.6 Specifying No-Input and Recognition Timeouts

- noinput-timeout

This parameter is used to trigger a no-input event. The parameter can be overridden per MRCP session by setting the header field *No-Input-Timeout* in a *SET-PARAMS* or *RECOGNIZE* request.

- input-timeout

This parameter is used to limit input (recognition) time. The parameter can be overridden per MRCP session by setting the header field *Recognition-Timeout* in a *SET-PARAMS* or *RECOGNIZE* request.

4.7 Specifying Speech Recognition Mode

By default, the speech recognition mode is derived from the service endpoint path, which is composed in the following format and defaults to the single-utterance mode.

Service Endpoint	URI
Speech Service (regional)	https://$region.stt.speech.microsoft.com/speech/recognition/**interactive**/cognitiveservices/v1

For the continuous speech recognition, the path is composed using the token *dictation* instead of *interactive* as follows.

Service Endpoint	URI
Speech Service (regional)	https://$region.stt.speech.microsoft.com/speech/recognition/**dictation**/cognitiveservices/v1

The recognition mode can also be specified by the configuration parameter *single-utterance*.

If the parameter *single-utterance* is set to true, then the service endpoint path is composed using the token *interactive*; otherwise, the token *dictation* is used.

Single Utterance Mode

In the single utterance mode, recognition is terminated upon an expiration of the speech complete timeout or an end of utterance event is delivered from the service.

Continuous Recognition Mode

In the continuous mode, recognition is terminated upon an expiration of the speech complete timeout, which is recommended to be set in the range of 1500 msec to 3000 msec. The service may return multiple results (sub utterances), which are concatenated and sent back to the MRCP client in a single RECOGNITION-COMPLETE event.

The parameter *single-utterance* can be overridden per MRCP session by setting the header field *Vendor-Specific-Parameters* in a *SET-PARAMS* or *RECOGNIZE* request, where the parameter name is *single-utterance* and acceptable values are *true* and *false*.

4.8 Specifying Vendor-Specific Parameters

The following parameters can optionally be specified by the MRCP client in *SET-PARAMS*, *DEFINE-GRAMMAR* and *RECOGNIZE* requests via the MRCP header field *Vendor-Specific-Parameters*.

Name	Unit	Description
start-of-input	String	Specifies the source of start of input event sent to the client (use "service-originated" for an event originated based on a first-received interim result and "internal" for an event determined by plugin). Available since Azure SR 1.9.0.
alternatives-below-threshold	Boolean	Specifies whether to return speech recognition result alternatives with the confidence score below the confidence threshold. Available since Azure SR 1.9.0.
speech-start-timeout	Time interval [msec]	Specifies how long to wait in transition mode before triggering a start of speech input event. Available since Azure SR 1.9.0.
interim-result-timeout	Time interval [msec]	Specifies a timeout between interim results containing transcribed speech. If the timeout is elapsed, input is considered complete. The timeout defaults to 0 (disabled). Available since Azure SR 1.12.0.

All the vendor-specific parameters can also be specified at the grammar-level via a built-in or SRGS XML grammar.

The following example demonstrates the use of a built-in grammar with the vendor-specific parameters *alternatives-below-threshold* and *speech-start-timeout* set to *true* and *100* correspondingly.

```
builtin:speech/transcribe?alternatives-below-threshold=true;speech-start-timeout=100
```

The following example demonstrates the use of an SRGS XML grammar with the vendor-specific parameters *alternatives-below-threshold* and *speech-start-timeout* set to *true* and *100* correspondingly.

```
<grammar mode="voice" root="transcribe" version="1.0" xml:lang="en-US"
xmlns="http://www.w3.org/2001/06/grammar">
  <meta name="scope" content="builtin"/>
  <meta name="alternatives-below-threshold" content="true"/>
  <meta name="speech-start-timeout" content="100"/>
  <rule id="transcribe">
    <one-of ><item>blank</item></one-of>
  </rule>
</grammar>
```

4.9 Maintaining Utterances

Saving of utterances is not required for regular operation and is disabled by default. However, enabling this functionality allows to save utterances sent to the service and later listen to them offline.

The relevant settings can be specified via the element *utterance-manager*.

- save-waveforms

  Utterances can optionally be recorded and stored if the configuration parameter *save-waveforms* is set to true. The parameter can be overridden per MRCP session by setting the header field *Save-Waveforms* in a *SET-PARAMS* or *RECOGNIZE* request.

- purge-existing

  This parameter specifies whether to delete existing waveforms on start-up.

- max-file-age

  This parameter specifies a time interval in minutes after expiration of which a waveform is deleted. If set to 0, there is no expiration time specified.

- max-file-count

  This parameter specifies the maximum number of waveforms to store. If the specified number is reached, the oldest waveform is deleted. If set to 0, there is no limit specified.

- waveform-base-uri

  This parameter specifies the base URI used to compose an absolute waveform URI returned in the header field *Waveform-Uri* in response to a RECOGNIZE request.

- waveform-folder

  This parameter specifies a path to the directory used to store waveforms in. The directory defaults to *${UniMRCPInstallDir}/var*.

4.10Maintaining Recognition Details Records

Producing of recognition details records (RDR) is not required for regular operation and is disabled by default. However, enabling this functionality allows to store details of each recognition attempt in a separate file and analyze them later offline. The RDRs ate stored in the JSON format.

The relevant settings can be specified via the element *rdr-manager*.

- save-records

  This parameter specifies whether to save recognition details records or not.

- purge-existing

  This parameter specifies whether to delete existing records on start-up.

- max-file-age

  This parameter specifies a time interval in minutes after expiration of which a record is deleted. If set to 0, there is no expiration time specified.

- max-file-count

  This parameter specifies the maximum number of records to store. If the specified number is reached, the oldest record is deleted. If set to 0, there is no limit specified.

- record-folder

  This parameter specifies a path to the directory used to store records in. The directory defaults to *${UniMRCPInstallDir}/var*.

The following is the content of a sample RDR.

```
{"recog-details-record": {
  "datetime": "2019-01-19 12:58:50",
  "language": "en-US",
  "sampling-rate": "8000 Hz",
  "max-alternatives": 1,
  "websocket": {
          "connection-start-ts": "0 ms",
          "connection-complete-ts": "317 ms",
          "speech-start-ts": "844 ms",
          "speech-end-ts": "2228 ms",
          "sent": "43448 bytes"
          "turns": "1"
  },
  "transcripts": [
          {"transcript": "call steve", "confidence": 0.945554}
  ],
  "completion-cause": "success",
  "completion-ts": "2228 ms"
}}
```

where the stored attributes are:

- datetime

  This attribute denotes the date and time captured when the corresponding MRCP RECOGNIZE request is received.

- language

This attribute denotes the speech language used with the request.

- sampling-rate

This attribute denotes the sampling rate used with the request.

- max-alternatives

This attribute denotes the number of alternative transcription results returned by the service.

- connection-start-ts

This attribute denotes a time interval in milliseconds, elapsed since initiation of the request, captured when a Websocket connection to the service is originated.

- connection-complete-ts

This attribute denotes a time interval in milliseconds, elapsed since initiation of the request, captured when the Websocket connection to the service is fully established.

- speech-start-ts

This attribute denotes a time interval in milliseconds, elapsed since initiation of the request, captured when streaming of audio data to the service is started.

- speech-end-ts

This attribute denotes a time interval in milliseconds, elapsed since initiation of the request, captured when streaming of audio data to the service is ended.

- sent

This attribute denotes the number of bytes of audio data sent to the service.

- turn

This attribute denotes the number of turns initiated.

- transcripts

This attribute denotes the array of transcripts returned by the service as a result of completion of the request.

- completion-cause

This attribute denotes the completion cause of the request.

- completion-ts

This attribute denotes a time interval in milliseconds, elapsed since initiation of the request, captured upon completion of the request (RECOGNITION-COMPLETE is sent).

5 Recognition Grammars and Results

5.1 Using Built-in Speech Transcription

For generic speech transcription, having no speech contexts defined, a pre-set identifier *transcribe* must be used by the MRCP client in a RECOGNIZE request as follows:

```
builtin:speech/transcribe
```

The name of the identifier *transcribe* can be changed from the configuration file *umsazuresr.xml*, since Azure SR 1.5.0.

Speech contexts are defined in the configuration file *umsazuresr.xml* and available since Azure SR 1.5.0. A speech context is assigned a unique string identifier and holds a list of phrases.

Below is a definition of a sample speech context *directory*:

```
<speech-context id="directory" speech-complete="true">
  <phrase>call Steve</phrase>
  <phrase>call John</phrase>
  <phrase>dial 5</phrase>
  <phrase>dial 6</phrase>
</speech-context>
```

Which can be referenced in a RECOGNIZE request as follows:

```
builtin:speech/directory
```

The prefixes *builtin:speech* and *builtin:grammar* can be used interchangeably as follows:

```
builtin:grammar/directory
```

Since Azure SR 1.7.0, a speech context can be referenced by means metadata in SRGS XML grammar. For example, the following SRGS grammar references a built-in speech context *directory*.

```
<grammar mode="voice" root="directory" version="1.0"
        xml:lang="en-US"
        xmlns="http://www.w3.org/2001/06/grammar">
    <meta name="scope" content="builtin"/>
    <rule id="directory"><one-of/></rule>
```

```
</grammar>
```

Where the root rule name identifies a speech context.

5.2 Using Built-in DTMF Grammars

Pre-set built-in DTMF grammars can be referenced by the MRCP client in a RECOGNIZE request as follows:

```
builtin:dtmf/$id
```

Where *$id* is a unique string identifier of the built-in DTMF grammar.

Note that only a DTMF grammar identifier *digits* is currently supported.

Since Azure SR 1.7.0, built-in DTMF digits can also be referenced by metadata in SRGS XML grammar. The following example is equivalent to the built-in grammar above.

```
<grammar mode="dtmf" root="digits" version="1.0"
        xml:lang="en-US"
        xmlns="http://www.w3.org/2001/06/grammar">
    <meta name="scope" content="builtin"/>
    <rule id="digits"><one-of/></rule>
</grammar>
```

Where the root rule name identifies a built-in DTMF grammar.

5.3 Retrieving Results

Results received from the speech service are either transformed to the NLSML format or used transparently in the Microsoft Speech JSON format. This behavior is specified via the *results-format* parameter in the *ws-streaming-recognition* element.

6 Monitoring Usage Details

The number of in-use and total licensed channels can be monitored in several alternate ways. There is a set of actions which can take place on certain events. The behavior is configurable via the element *monitoring-agent*, which contains two event handlers: *usage-change-handler* and *usage-refresh-handler*.

While the *usage-change-handler* is invoked on every acquisition and release of a licensed channel, the *usage-refresh-handler* is invoked periodically on expiration of a timeout specified by the attribute *refresh-period*.

The following actions can be specified for either of the two handlers.

6.1 Log Usage

The action *log-usage* logs the following data in the order specified.

- The number of currently in-use channels.
- The maximum number of channels used concurrently. Available since Azure SR 1.5.0.
- The total number of licensed channels.

The following is a sample log statement, indicating 0 in-use, 0 max-used and 2 total channels.

```
[NOTICE] AZURESR Usage: 0/0/2
```

6.2 Update Usage

The action *update-usage* writes the following data to a status file *umsazuresr-usage.status*, located by default in the directory *${UniMRCPInstallDir}/var/status*.

- The number of currently in-use channels.
- The maximum number of channels used concurrently. Available since Azure SR 1.5.0.
- The total number of licensed channels.
- The current status of the license permit.
- The license server alarm. Set to *on*, if the license server is not available for more than one hour; otherwise, set to *off*. This parameter is maintained only if the license server is used. Available since Azure SR 1.7.0.

The following is a sample content of the status file.

```
in-use channels: 0
```

```
        max used channels: 0
        total channels: 2
        license permit: true
        licserver alarm: off
```

6.3 Dump Channels

The action *dump-channels* writes the identifiers of in-use channels to a status file *umsazuresr-channels.status*, located by default in the directory *${UniMRCPInstallDir}/var/status*.

7 Usage Examples

7.1 Speech Transcription

This examples demonstrates how to perform speech recognition by using a RECOGNIZE request.

C->S:

```
MRCP/2.0 336 RECOGNIZE 1
Channel-Identifier: 6e1a2e4e54ae11e7@speechrecog
Content-Id: request1@form-level
Content-Type: text/uri-list
Cancel-If-Queue: false
No-Input-Timeout: 5000
Recognition-Timeout: 10000
Start-Input-Timers: true
Confidence-Threshold: 0.87
Save-Waveform: true
Content-Length: 25

builtin:speech/transcribe
```

S->C:

```
MRCP/2.0 83 1 200 IN-PROGRESS
Channel-Identifier: 6e1a2e4e54ae11e7@speechrecog
```

S->C:

```
MRCP/2.0 115 START-OF-INPUT 1 IN-PROGRESS
Channel-Identifier: 6e1a2e4e54ae11e7@speechrecog
Input-Type: speech
```

S->C:

```
MRCP/2.0 498 RECOGNITION-COMPLETE 1 COMPLETE
Channel-Identifier: 6e1a2e4e54ae11e7@speechrecog
Completion-Cause: 000 success
Waveform-Uri: <http://localhost/utterances/utter-6e1a2e4e54ae11e7-
1.wav>;size=20480;duration=1280
Content-Type: application/x-nlsml
```

```
Content-Length: 214

<?xml version="1.0"?>
<result>
  <interpretation grammar="builtin:speech/transcribe" confidence="0.95">
    <instance>what's the weather like</instance>
    <input mode="speech">what's the weather like</input>
  </interpretation>
</result>
```

7.2 DTMF Recognition

This examples demonstrates how to reference a built-in DTMF grammar in a RECOGNIZE request.

C->S:

```
MRCP/2.0 266 RECOGNIZE 1
Channel-Identifier: d26bef74091a174c@speechrecog
Content-Type: text/uri-list
Cancel-If-Queue: false
Start-Input-Timers: true
Confidence-Threshold: 0.7
Speech-Language: en-US
Dtmf-Term-Char: #
Content-Length: 19

builtin:dtmf/digits
```

S->C:

```
MRCP/2.0 83 1 200 IN-PROGRESS
Channel-Identifier: d26bef74091a174c@speechrecog
```

S->C:

```
MRCP/2.0 113 START-OF-INPUT 1 IN-PROGRESS
Channel-Identifier: d26bef74091a174c@speechrecog
Input-Type: dtmf
```

S->C:

```
MRCP/2.0 382 RECOGNITION-COMPLETE 1 COMPLETE
Channel-Identifier: d26bef74091a174c@speechrecog
Completion-Cause: 000 success
Content-Type: application/x-nlsml
Content-Length: 197

<?xml version="1.0"?>
<result>
  <interpretation grammar="builtin:dtmf/digits" confidence="1.00">
    <input mode="dtmf">1 2 3 4</input>
    <instance>1234</instance>
  </interpretation>
</result>
```

7.3 Speech and DTMF Recognition

This examples demonstrates how to perform recognition by activating both speech and DTMF grammars. In this example, the user is expected to input a 4-digit pin.

C->S:

```
MRCP/2.0 275 RECOGNIZE 1
Channel-Identifier: 6ae0f23e1b1e3d42@speechrecog
Content-Type: text/uri-list
Cancel-If-Queue: false
Start-Input-Timers: true
Confidence-Threshold: 0.7
Speech-Language: en-US
Content-Length: 47

builtin:dtmf/digits?length=4
builtin:speech/pin
```

S->C:

```
MRCP/2.0 83 2 200 IN-PROGRESS
Channel-Identifier: 6ae0f23e1b1e3d42@speechrecog
```

S->C:

```
MRCP/2.0 115 START-OF-INPUT 2 IN-PROGRESS
Channel-Identifier: 6ae0f23e1b1e3d42@speechrecog
Input-Type: speech
```

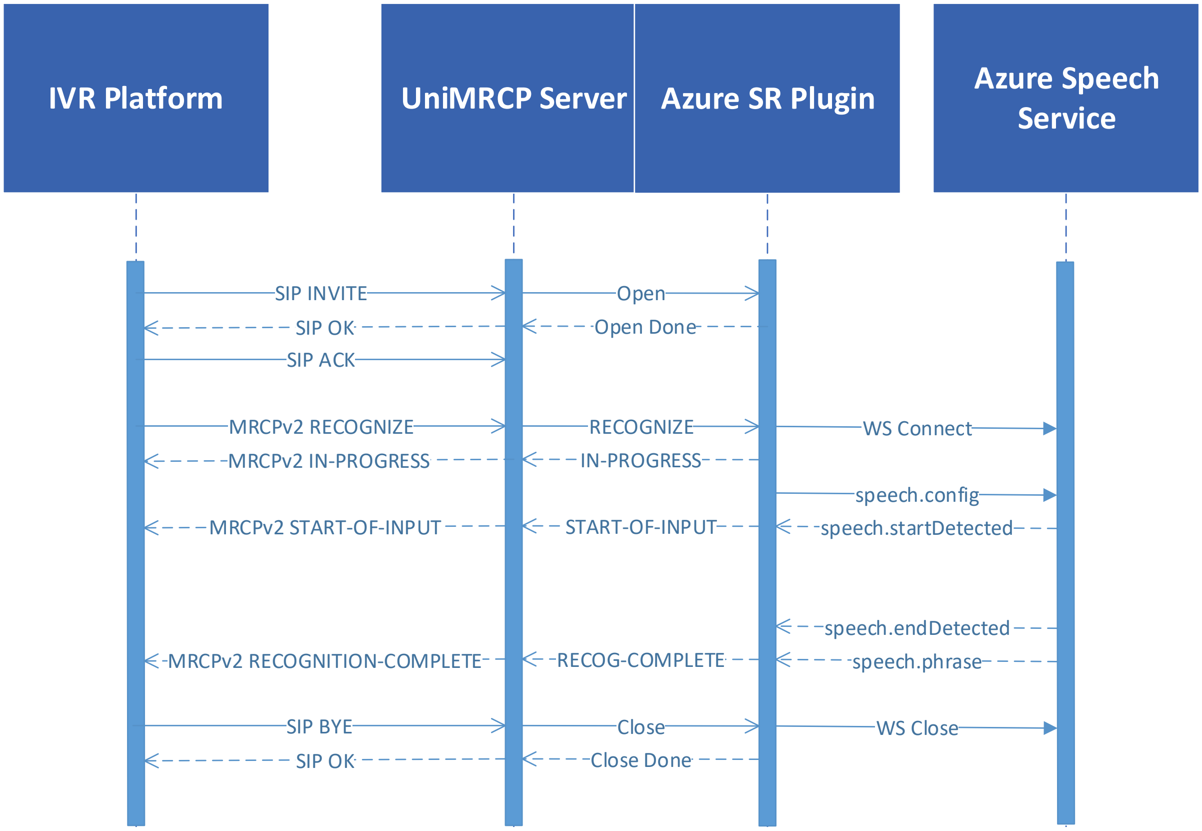
S->C:

```
MRCP/2.0 399 RECOGNITION-COMPLETE 2 COMPLETE
Channel-Identifier: 6ae0f23e1b1e3d42@speechrecog
Completion-Cause: 000 success
Content-Type: application/x-nlsml
Content-Length: 214

<?xml version="1.0"?>
<result>
  <interpretation grammar=" builtin:speech/pin" confidence="1.00">
    <instance>one two three four</instance>
    <input mode="speech">one two three four</input>
  </interpretation>
</result>
```

8 Sequence Diagram

The following sequence diagram outlines common interactions between all the main components involved in a typical recognition session performed over MRCPv2.



9 References

9.1 Microsoft Azure

- Speech WebSocket Protocol
- Basic Concepts
- Authentication

9.2 Specifications

- Speech Recognizer Resource
- NLSML Results