Asterisk
Google SR and SS

Usage Guide

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Author: Arsen Chaloyan
Table of Contents

1 Overview........................................................................................................................................4
  1.1 Applicable Versions....................................................................................................................4
2 Generic Speech Recognition API ..................................................................................................5
  2.1 Overview....................................................................................................................................5
  2.2 Configuration Steps ...................................................................................................................5
  2.3 Usage Examples..........................................................................................................................5
3 Suite of UniMRCP Applications....................................................................................................8
  3.1 Overview....................................................................................................................................8
  3.2 Configuration Steps ...................................................................................................................8
  3.3 Usage Examples..........................................................................................................................9
    MRCPRecog.................................................................................................................................9
    MRCPSynth.................................................................................................................................10
    SynthAndRecog...........................................................................................................................11
1 Overview

This guide describes how to utilize the Google Cloud Speech services with Asterisk.

Note that the Asterisk and the UniMRCP server typically reside on different hosts in a LAN, although both might be installed on the same host.

Installation of the Asterisk and the UniMRCP server with the Google SR and SS plugins is not covered in this document. Visit the corresponding web pages for more information.

http://unimrcp.org/asterisk
http://unimrcp.org/gsr
http://unimrcp.org/gss

1.1 Applicable Versions

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

- UniMRCP Modules for Asterisk 1.5.0 and above
- UniMRCP GSR Plugin 1.1.0 and above
- UniMRCP GSS Plugin 1.0.0 and above
2 Generic Speech Recognition API

2.1 Overview

The module `res_speech_unimrcp.so` provides an implementation of the Generic Speech Recognition API of Asterisk, based on the UniMRCP client library.

2.2 Configuration Steps

This section outlines major configuration steps required for use of the module `res_speech_unimrcp.so` with the UniMRCP server.

Check the name of the profile, referenced in the configuration file `res-speech-unimrcp.conf`, which is located in the configuration directory of Asterisk. Use `uni2` for MRCPv2, and `uni1` for MRCPv1.

```
[general]
unimrcp-profile = uni2 ; UniMRCP MRCPv2 Server
; unimrcp-profile = uni1 ; UniMRCP MRCPv1 Server
```

Open the configuration file `unimrcp-client.xml`, located in the configuration directory of UniMRCP and specify IP addresses of the client and the server. In the following example, the Asterisk/UniMRCP client is located on 10.0.0.1 and the UniMRCP server is on 10.0.0.2.

```
<properties>
  <!-- <ip type="auto"/> -->

  <ip>10.0.0.1</ip>
  <server-ip>10.0.0.2</server-ip>
</properties>
```

2.3 Usage Examples

**Built-in Speech Context**

Make use of a built-in speech grammar `transcribe` for recognition, having no speech contexts defined, by adding the following entries in the Asterisk configuration file `extensions.conf`.

```
[res-speech-unimrcp-1]
exten => s,1,Answer()
exten => s,2,SpeechCreate()
exten => s,3,SpeechActivateGrammar(builtin:speech/transcribe)
exten => s,4,SpeechBackground(beep, 20)
```
In the Asterisk dialplan, associate the provided sample to an extension, for example, 701.

```
exten => 701,1,Goto(res-speech-unimrcp-1,s,1)
```

Place a test call and make sure recognition works as expected.

**Dynamic Speech Context**

Dynamically load a speech context for recognition, by adding the following entries in the Asterisk configuration file `extensions.conf`.

```
[res-speech-unimrcp-2]
exten => s,1,Answer()
exten => s,2,SpeechCreate()
exten => s,3,SpeechLoadGrammar(directory,application/xml:/usr/local/unimrcp/data/directory.xml)
exten => s,4,SpeechActivateGrammar(directory)
exten => s,5,SpeechBackground(beep, 20)
exten => s,6,Verbose(1, "Recognition result count: ${SPEECH(results)}")
exten => s,7,GotoIf("${SPEECH(results)}" = "0")?8:10)
exten => s,8,Playback(error)
exten => s,9,Goto(5)
exten => s,10,Verbose(1, "Recognition result: ${SPEECH_TEXT(0)}, confidence score: ${SPEECH_SCORE(0)}, grammar-uri: ${SPEECH_GRAMMAR(0)}")
exten => s,11,SpeechDestroy()
exten => s,12,Hangup()
```

In this example, the speech context is loaded from a file located at `/usr/local/unimrcp/directory.xml` and having the following sample content.

```
<speech-context>
  <phrase>call Steve</phrase>
  <phrase>call John</phrase>
  <phrase>dial 5</phrase>
  <phrase>dial 6</phrase>
</speech-context>
```
In the Asterisk dialplan, associate the provided sample to an extension, for example, 702.

```
  exten => 702,1,Goto(res-speech-unimrcp-2,s,1)
```

Place a test call and make sure recognition works as expected.
3 Suite of UniMRCP Applications

3.1 Overview

The module app_unimrcp.so provides a suite of speech recognition and synthesis applications for Asterisk.

3.2 Configuration Steps

This section outlines major configuration steps required for use of the module app_unimrcp.so with the UniMRCP server.

Open the configuration file mrcp.conf, located in the configuration directory of Asterisk, and add two profiles for MRCPv2 and MRCPv1 respectively. In the following example, the Asterisk/UniMRCP client is located on 10.0.0.1 and the UniMRCP server is on 10.0.0.2.

```
[uni2]
  ; MRCP settings
  version = 2
  ;
  ; SIP settings
  server-ip = 10.0.0.2
  server-port = 8060
  ; SIP user agent
  client-ip = 10.0.0.1
  client-port = 25097
  sip-transport = udp
  ;
  ; RTP factory
  rtp-ip = 10.0.0.1
  rtp-port-min = 28000
  rtp-port-max = 29000
  ;
  ; Jitter buffer settings
  playout-delay = 50
  max-playout-delay = 200
  ; RTP settings
  ptime = 20
  codecs = PCMU PCMA L16/96/8000 telephone-event/101/8000
  ; RTCP settings
  rtcp = 0
```

```
[uni1]
```
3.3 Usage Examples

MRCPRecog

Built-in Speech Context

Make use of a built-in speech grammar *transcribe* for recognition, having no speech contexts defined, by adding the following entries in the Asterisk configuration file *extensions.conf*.

```plaintext
[app-unimrcp-1]
exten => s,1,Answer
exten => s,2,MRCPRecog("builtin:speech/transcribe", spl=en-US&f=beep&p=uni2)
exten => s,3,Goto(mrcprecog-output,s,1)
```

Note the macro *mrcprecog-output*, used in the example above, is defined in the file *mrcp_sample_apps.conf*.

In the Asterisk dialplan, associate the provided sample to an extension, for example, 801.
Dynamic Speech Context
Dynamically load a speech context for recognition, by adding the following entries in the Asterisk configuration file `extensions.conf`.

```
[app-unimrcp-2]
exten => s,1,Answer
exten => s,2,MRCPRecog("/usr/local/unimrcp/data/directory.xml", spl=en-US&p=uni2)
exten => s,3,Goto(mrcprecog-output,s,1)
```

In this example, the speech context is loaded from a file located at `/usr/local/unimrcp/directory.xml` and having the following sample content.

```
<speech-context>
  <phrase>call Steve</phrase>
  <phrase>call John</phrase>
  <phrase>dial 5</phrase>
  <phrase>dial 6</phrase>
</speech-context>
```

In the Asterisk dialplan, associate the provided sample to an extension, for example, 802.

```
exten => 802,1,Goto(app-unimrcp-2,s,1)
```

Place a test call and make sure recognition works as expected.

MRCPSynth

Use the application MRCPSynth for speech synthesis.

```
[app-unimrcp-2]
exten => s,1,Answer
exten => s,2,MRCPSynth(Welcome to Asterisk,l=en-US&p=uni2)
```

In the Asterisk dialplan, associate the provided sample to an extension, for example, 802.
exten => 802,1,Goto(app-unimrcp-2,s,1)

Place a test call and listen to the synthesized message.

**SynthAndRecog**

Use the application SynthAndRecog for speech synthesis and recognition.

```
[app-unimrcp-3]
exten => s,1,Answer
exten => s,2, SynthAndRecog(Please say something, builtin:speech/transcribe, spl=en-US&p=uni2)
exten => s,3,Goto(mrcprecog-output,s,1)
```

In the Asterisk dialplan, associate the provided sample to an extension, for example, 803.

```
exten => 803,1,Goto(app-unimrcp-3,s,1)
```

Place a test call and listen to the synthesized prompt and say something. Make sure recognition works as expected.