Outline

- 嵌入式應用: 語音助理
  - 語音識別 (Speech recognition)
  - 自動語音辨識 (Automatic Speech Recognition, ASR)
  - 電腦語音識別 (Computer Speech Recognition)
  - 語音轉文字識別 (Speech To Text, STT)
  - 自然語言處理 (Natural Language Processing, NLP)
    - 讓電腦擁有理解人類語言的能力

語音助理流程

使用者 → 麥克風 → 語音識別 → 語音轉文字

→ 關鍵字 → 執行特定任務 → 回覆訊息

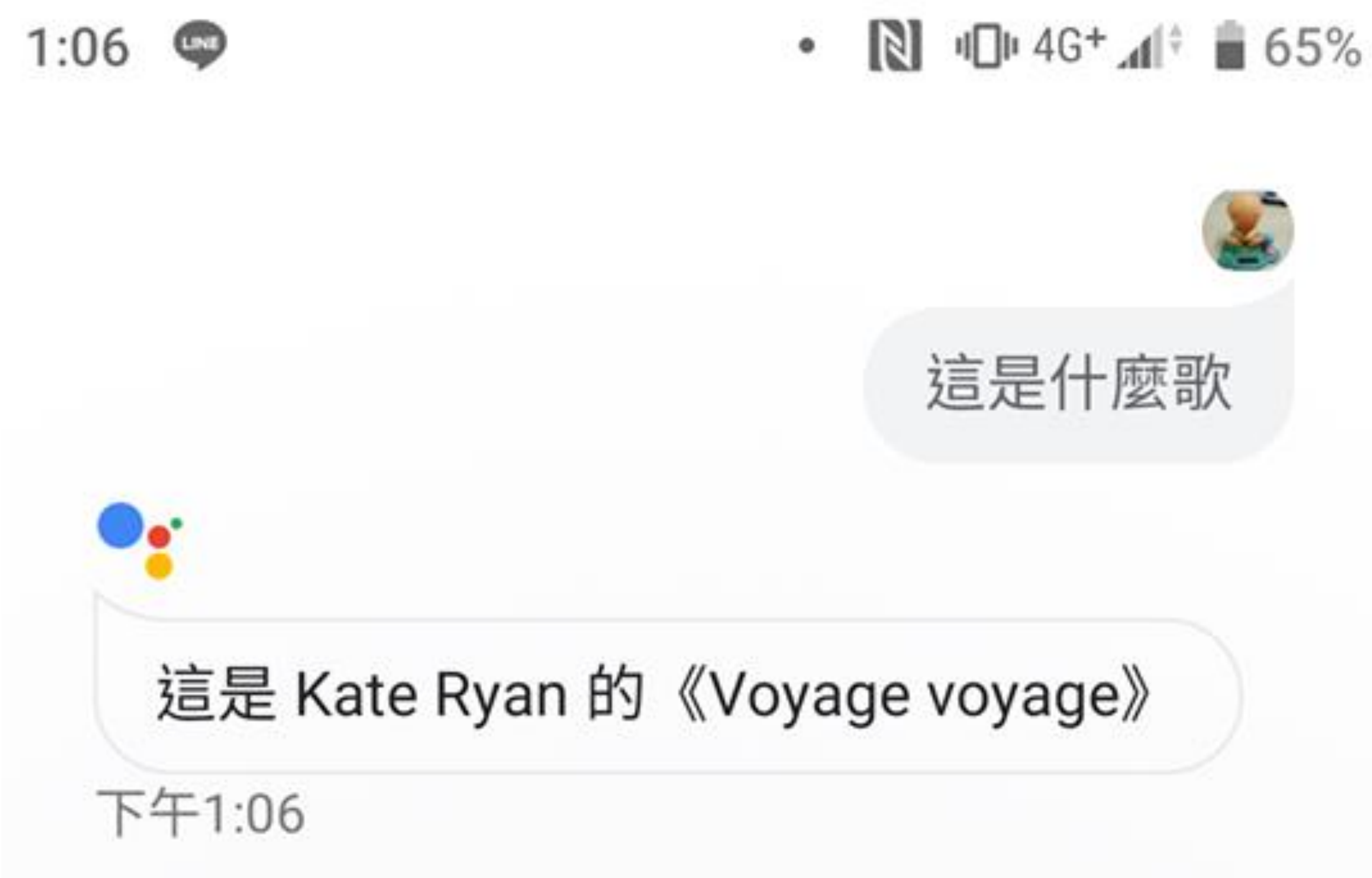
→ 文字轉語音 → 語音波形合成 → 喇叭播放

https://zh.wikipedia.org/wiki/%E6%A2%85%E7%88%BE%E5%80%92%E9%A0%BB%E8%AD%9C
https://en.wikipedia.org/wiki/Mel-frequency_cepstrum

Mel-Frequency Cepstral Coefficients

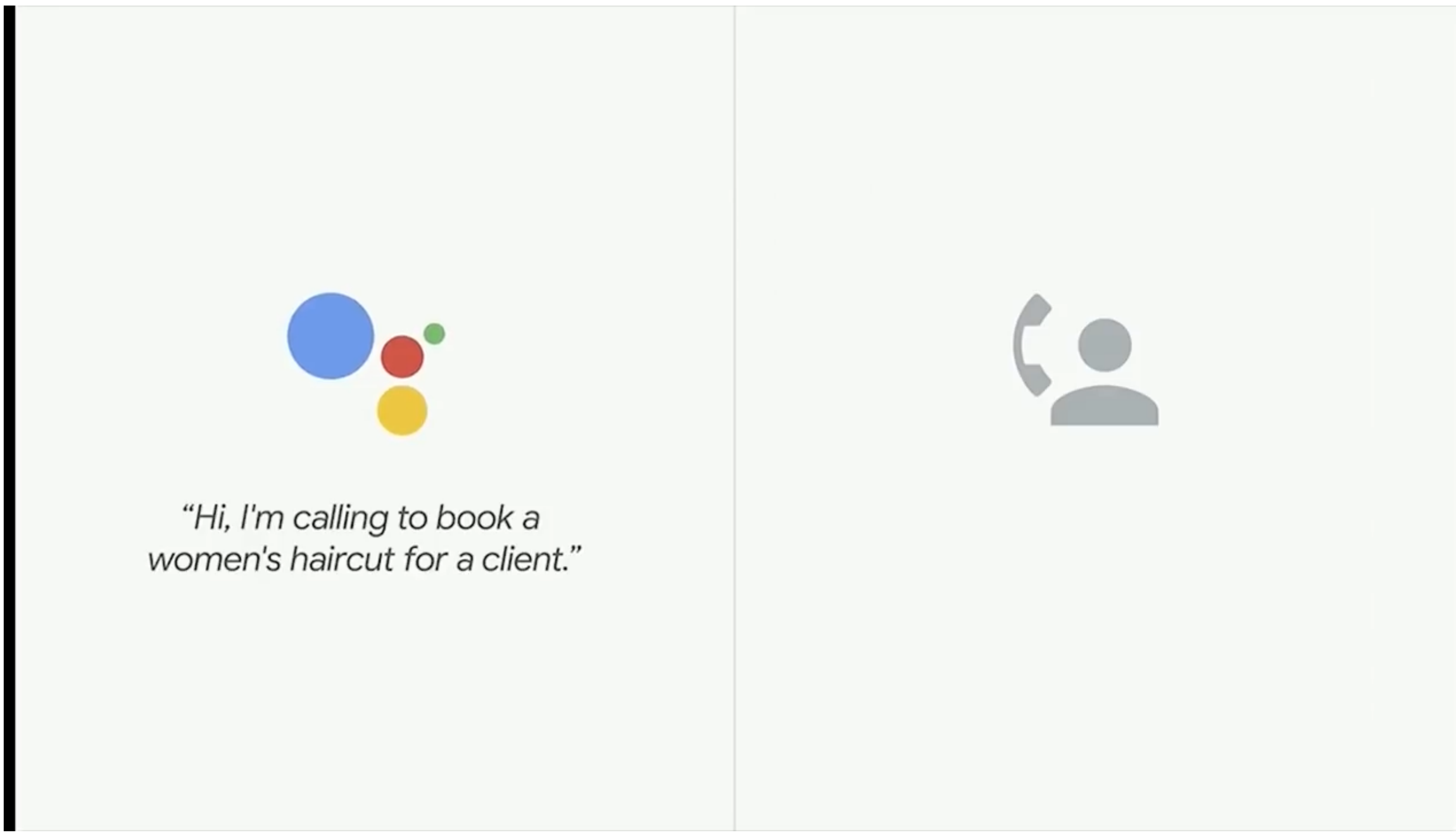
- MFCCs are commonly used as features in speech recognition systems, such as the systems which can automatically recognize numbers spoken into a telephone.
- MFCC(梅爾倒頻譜係數)
  1. Take the Fourier transform of a signal (with sliding window)
  2. Map the powers of the spectrum obtained above onto the mel scale, using triangular overlapping windows.
  3. Take the logs of the powers at each of the mel frequencies.
  4. Take the discrete cosine transform of the list of mel log powers, as if it were a signal.
  5. The MFCCs are the amplitudes of the resulting spectrum.
- Application: music information retrieval
  - audio similarity measures

Google assistant



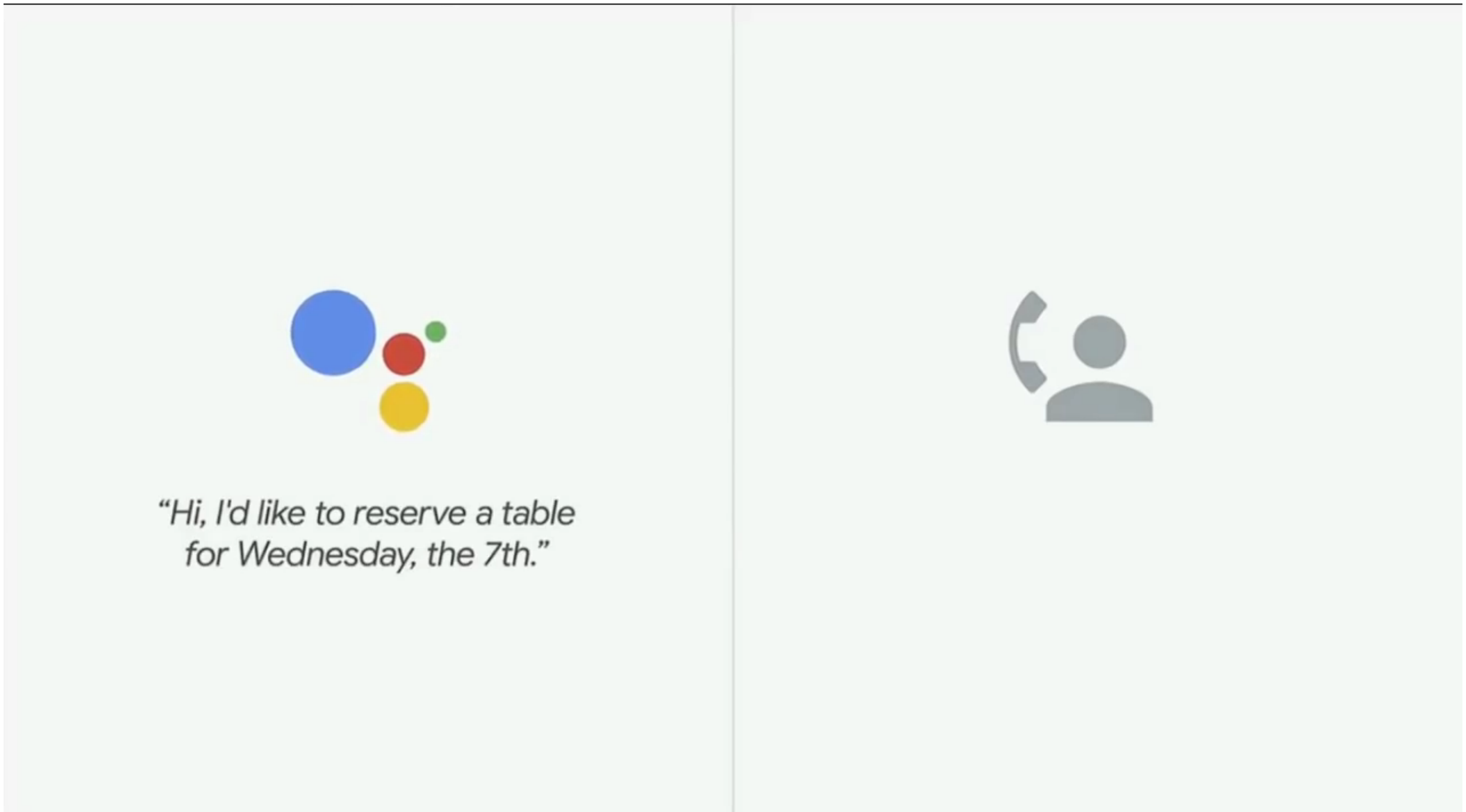
Google assistant

- Google Assistant will soon be able to call restaurants and make a reservation for you (2018/5/9)



Google assistant

- Google Assistant calling a restaurant for a reservation

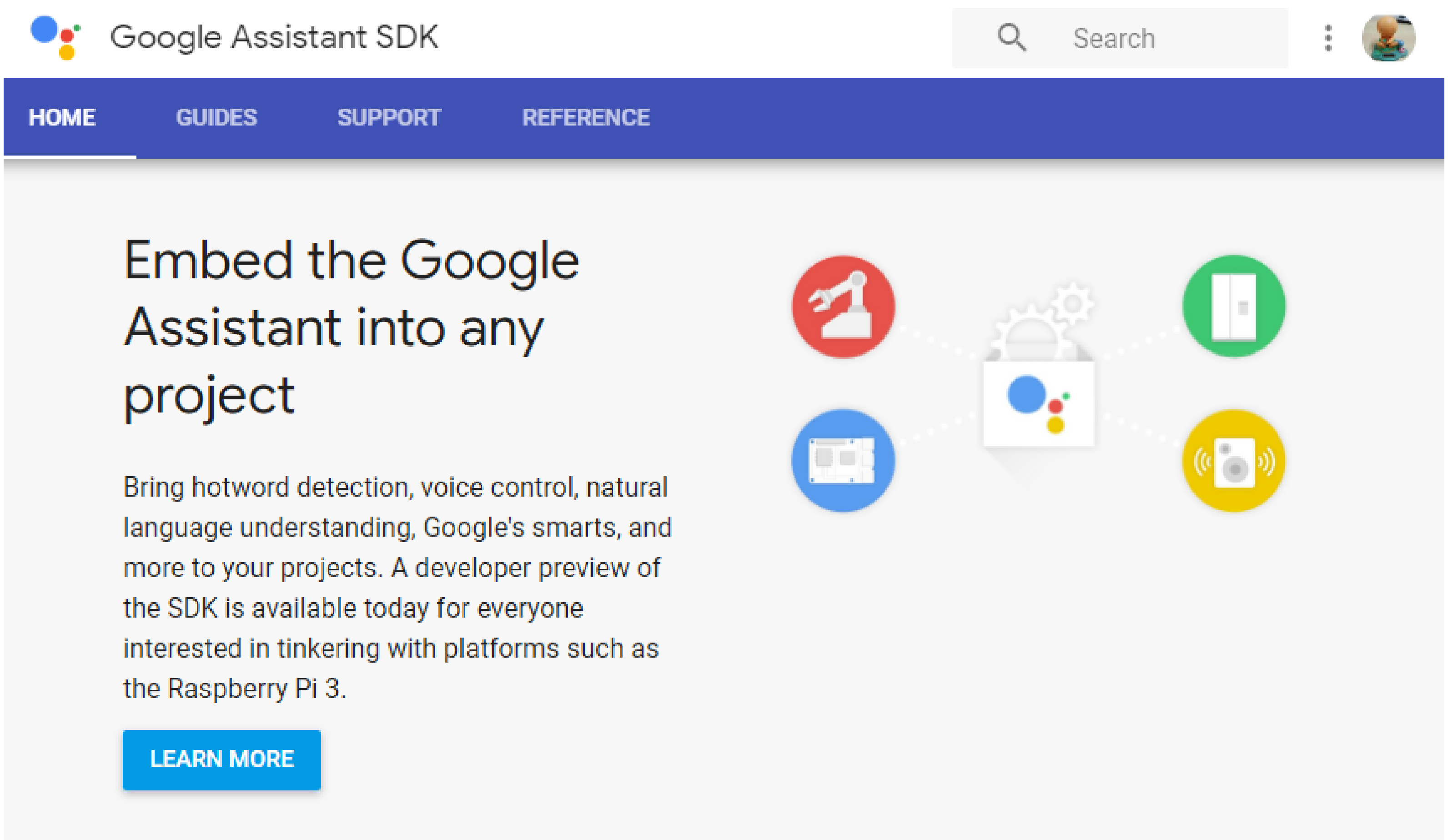


https://www.youtube.com/watch?v=3kODsHcrs2c

Google assistant

- Google IO 2019 Next Gen Google Assistant (2019/5/7)

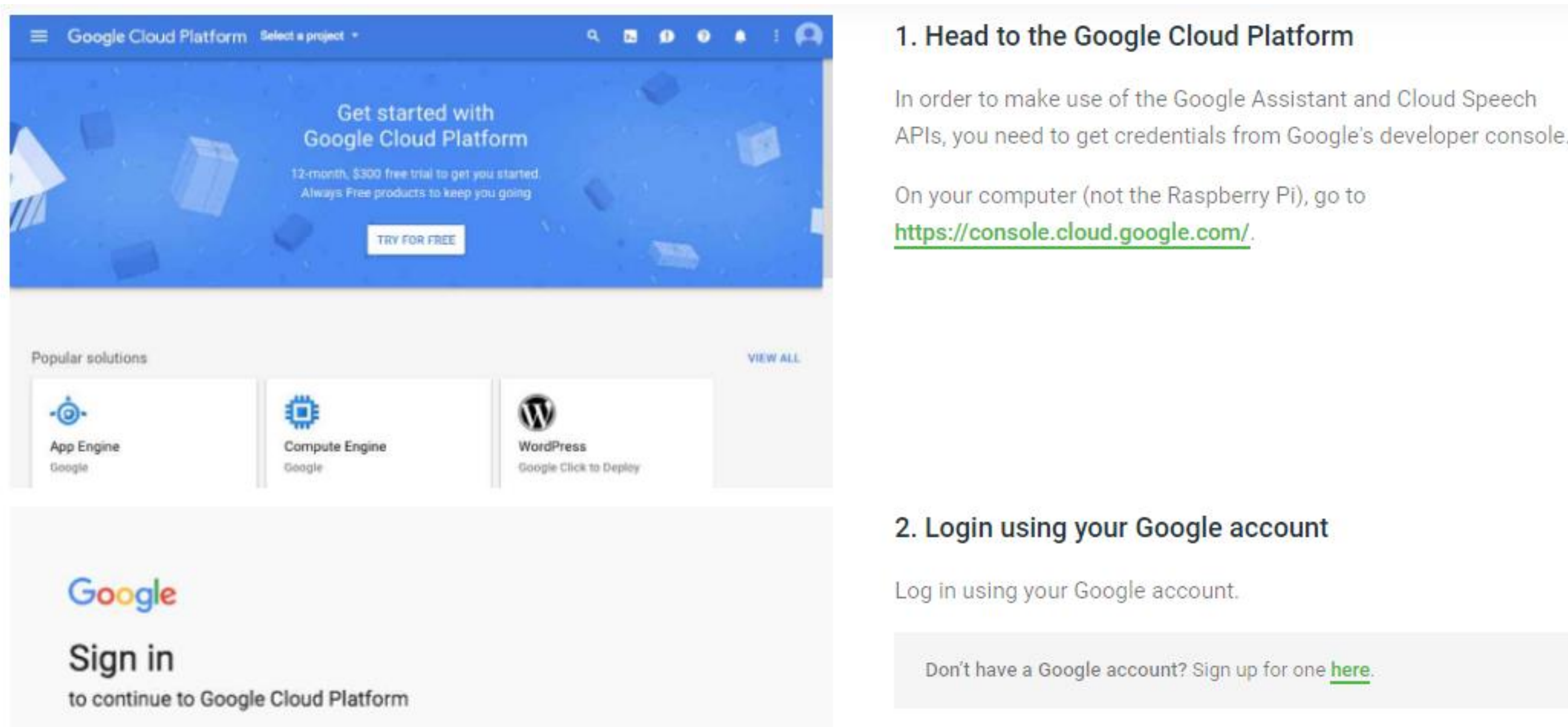
Google assistant SDK



https://developers.google.com/assistant/sdk/

Voice Kit

- Do-it-yourself intelligent speaker. Experiment with voice recognition and the Google Assistant.



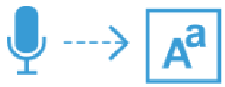
https://aiyprojects.withgoogle.com/voice/

Azure

語音轉換文字 - 將語音轉換成文字以取得直覺式互動

輕鬆將即時語音轉換文字的功能新增到您的應用程式之中，以應用在語音命令、即時轉譯、自動會議記錄，或是話務中心的記錄分析等等。

調整您的語音辨識模型，以適應使用者的說話方式、措辭或獨特詞彙，並根據您的情況配合特定背景雜音、口音和聲紋。

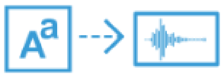


深入了解 >

文字轉換語音 - 為您的應用程式提供自然語音

建置智慧型應用程式和服務，使用文字轉換語音服務自然地與使用者交談。近乎即時地將文字轉換成音訊，並根據說話速度、音調、音量等變化進行調整。

使用自訂語音模型，為您的應用程式提供獨特且可辨識的品牌語音。只要錄製並上傳定型資料，服務就會建立專為您的錄音調整的獨特語音效果。



深入了解 >

語音翻譯

為您的應用程式提供任何支援語言的即時語音翻譯功能，並接收文字或語音翻譯。語音翻譯模型是以尖端語音辨識和神經機器翻譯系統 (NMT) 技術為基礎。這些模型已經過最佳化，能夠理解人們在真實生活中的說話方式，並產生絕佳品質的翻譯。

深入了解 >

SpeechRecognition

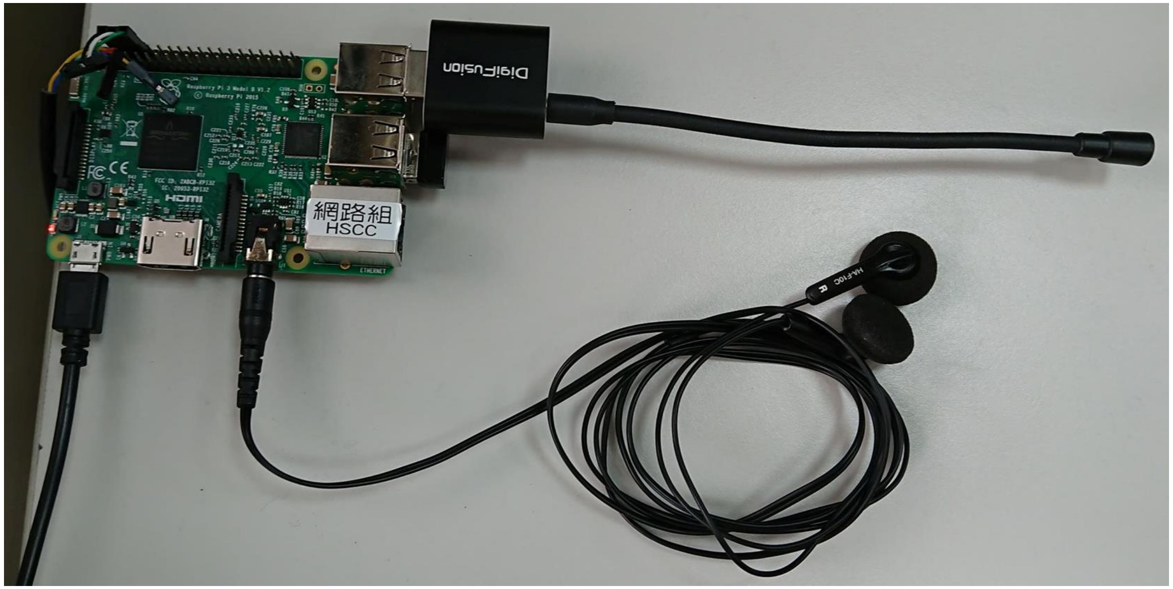
- Library for performing speech recognition, with support for several engines and APIs, online and offline.

- Speech recognition engine/API support:
  - CMU Sphinx (works offline) (卡内基大學)
  - **Google Speech Recognition**
  - **Google Cloud Speech API**
  - Wit.ai (Facebook, Messenger ChatBot)
  - Microsoft Bing Voice Recognition
  - Houndify API (SoundHound,音樂識別平台)
  - IBM Speech to Text
  - Snowboy Hotword Detection (works offline)

https://pypi.org/project/SpeechRecognition/

Dependecy

- sudo pip install SpeechRecognition
- sudo pip install gTTS
- sudo apt-get install libasound2-dev
- sudo apt-get install python-pyaudio
- sudo apt-get install flac



Test and play microphone

- In terminal
  - Check device
    - aplay -l
    - arecord -l
  - Record you voice
    - arecord -D plughw:1 -f cd Filename.mp3
    - use "ctrl + c" to stop recording
  - Play audio
    - omxplayer -o local -p Filename.mp3

Check your device

- aplay -l

(COM8) [80x24]

連線(C) 編輯(E) 檢視(V) 視窗(W) 選項(O) 說明(H)

```
pi@raspberrypi:~$ aplay -l
**** List of PLAYBACK Hardware Devices ****
card 0: ALSA [bcm2835 ALSA], device 0: bcm2835 ALSA [bcm2835 ALSA]
  Subdevices: 7/7
  Subdevice #0: subdevice #0
  Subdevice #1: subdevice #1
  Subdevice #2: subdevice #2
  Subdevice #3: subdevice #3
  Subdevice #4: subdevice #4
  Subdevice #5: subdevice #5
  Subdevice #6: subdevice #6
card 0: ALSA [bcm2835 ALSA], device 1: bcm2835 ALSA [bcm2835 IEC958/HDMI]
  Subdevices: 1/1
  Subdevice #0: subdevice #0
card 1: Device [USB Audio Device], device 0: USB Audio [USB Audio]
  Subdevices: 1/1
  Subdevice #0: subdevice #0
pi@raspberrypi:~$
```

- arecord -l

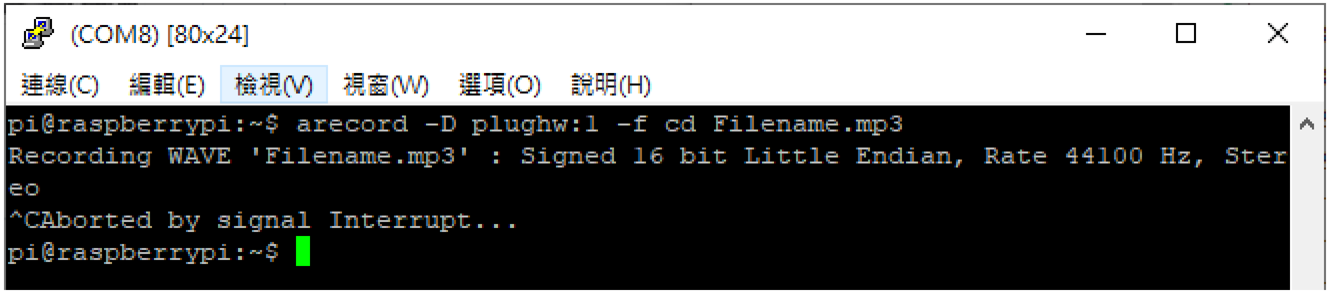
(COM8) [80x24]

連線(C) 編輯(E) 檢視(V) 視窗(W) 選項(O) 說明(H)

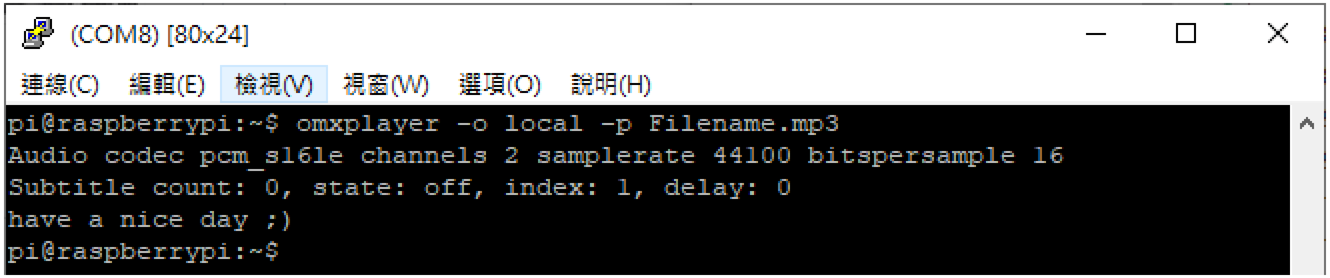
```
pi@raspberrypi:~$ arecord -l
**** List of CAPTURE Hardware Devices ****
card 1: Device [USB Audio Device], device 0: USB Audio [USB Audio]
  Subdevices: 1/1
  Subdevice #0: subdevice #0
pi@raspberrypi:~$
```

Test and play microphone

- Record you voice
  - arecord -D plughw:1 -f cd Filename.mp3
  - use “ctrl + c” to stop recording



- Play audio
  - omxplayer -o local -p Filename.mp3



Speech to text (microphone)

```
import speech_recognition as sr

#obtain audio from the microphone
r=sr.Recognizer()

with sr.Microphone() as source:
    print("Please wait. Calibrating microphone...")
    #listen for 1 seconds and create the ambient noise energy level
    r.adjust_for_ambient_noise(source, duration=1)
    print("Say something!")
    audio=r.listen(source)

# recognize speech using Google Speech Recognition
try:
    print("Google Speech Recognition thinks you said:")
    print(r.recognize_google(audio))
except sr.UnknownValueError:
    print("Google Speech Recognition could not understand audio")
except sr.RequestError as e:
    print("No response from Google Speech Recognition service: {0}".format(e))
```

Speech to text (audio file)

```
import speech_recognition as sr

#obtain audio from the microphone
r=sr.Recognizer()

myvoice = sr.AudioFile('hello.flac')
with myvoice as source:
   print("Use audio file as input!")
   audio = r.record(source)

# recognize speech using Google Speech Recognition
try:
   print("Google Speech Recognition thinks you said:")
   print(r.recognize_google(audio))
except sr.UnknownValueError:
   print("Google Speech Recognition could not understand audio")
except sr.RequestError as e:
   print("No response from Google Speech Recognition service: {0}".format(e))
```

Input format: PCM WAV, AIFF/AIFF-C, or Native FLAC

Speech to text (result)

You can ignore the ALSA warning messages

(COM8) [80x24]

連線(C) 編輯(E) 檢視(V) 視窗(W) 選項(O) 說明(H)

```
uealsa.device'
ALSA lib conf.c:4528:(_snd_config_evaluate) function snd_func_refer returned err
or: No such file or directory
ALSA lib conf.c:4996:(snd_config_expand) Args evaluate error: No such file or di
rectory
ALSA lib pcm.c:2495:(snd_pcm_open_noupdate) Unknown PCM bluealsa
ALSA lib confmisc.c:1281:(snd_func_refer) Unable to find definition 'defaults.bl
uealsa.device'
ALSA lib conf.c:4528:(_snd_config_evaluate) function snd_func_refer returned err
or: No such file or directory
ALSA lib conf.c:4996:(snd_config_expand) Args evaluate error: No such file or di
rectory
ALSA lib pcm.c:2495:(snd_pcm_open_noupdate) Unknown PCM bluealsa
Cannot connect to server socket err = No such file or directory
Cannot connect to server request channel
jack server is not running or cannot be started
JackShmReadWritePtr::~JackShmReadWritePtr - Init not done for -1, skipping unloc
k
JackShmReadWritePtr::~JackShmReadWritePtr - Init not done for -1, skipping unloc
k
Say something!
Google Speech Recognition thinks you said:
pneumonoultramicroscopicsilicovolcanoconiosis
pi@raspberrypi:~$
```

https://pypi.org/project/SpeechRecognition/
https://github.com/Uberi/speech_recognition/blob/master/examples/audio_transcribe.py

SpeechRecognition

Speech recognition engine/API support:

- CMU Sphinx (works offline)
- **Google Speech Recognition**
- Google Cloud Speech API
- Wit.ai
- Microsoft Bing Voice Recognition
- Houndify API
- IBM Speech to Text

- r.recognize_sphinx(audio)
- **r.recognize_google(audio)**
- r.recognize_google_cloud(audio, credentials_json=GOOGLE_CLOUD_SPEECH_CREDENTIALS)
- r.recognize_wit(audio, key=WIT_AI_KEY)
- r.recognize_azure(audio, key=AZURE_SPEECH_KEY)
- r.recognize_bing(audio, key=BING_KEY)
- r.recognize_houndify(audio, client_id=HOUNDIFY_CLIENT_ID, client_key=HOUNDIFY_CLIENT_KEY)
- r.recognize_ibm(audio, username=IBM_USERNAME, password=IBM_PASSWORD)

Speech Recognition Library Reference
https://github.com/Uberi/speech_recognition/blob/master/reference/library-reference.rst

Text to speech

```
from gtts import gTTS
import os

tts = gTTS(text='hello', lang='en')
tts.save('hello.mp3')

os.system('omxplayer -o local -p hello.mp3 > /dev/null 2>&1')
```

The output format is mp3!

Parameters:

- **text** (*string*) – The text to be read.
- **lang** (*string, optional*) – The language (IETF language tag) to read the text in. Defaults to 'en'.
- **slow** (*bool, optional*) – Reads text more slowly. Defaults to `False`.
- **lang_check** (*bool, optional*) – Strictly enforce an existing `lang`, to catch a language error early. If set to `True`, a `ValueError` is raised if `lang` doesn't exist. Default is `True`.

gTTS (Google Text-to-Speech)

- An interface to Google Translate’s Text-to-Speech API.

gTTS (`gtts.gTTS`)

class `gtts.tts.gTTS`(*text, lang='en', slow=False, lang_check=True, pre_processor_funcs=[<function tone_marks>, <function end_of_line>, <function abbreviations>, <function word_sub>], tokenizer_func=<bound method Tokenizer.run of re.compile('(?<=\？).|(?<=\！).|(?<=\ ?).|(?<=\ !).|(?<!\.[a-z])\. |(?<!\.[a-z])\, |(?<!\d)\:|\;|\—|\，|\...|\：|\n|\¿|\(|\、|\[|\)|\¡|\،|\..|\]|\。', re.IGNORECASE) from: [<function tone_marks>, <function period_comma>, <function colon>, <function other_punctuation>]>*) [source]

Parameters:

- **text** (*string*) – The text to be read.
- **lang** (*string, optional*) – The language (IETF language tag) to read the text in. Defaults to 'en'.
- **slow** (*bool, optional*) – Reads text more slowly. Defaults to `False`.
- **lang_check** (*bool, optional*) – Strictly enforce an existing `lang`, to catch a language error early. If set to `True`, a `ValueError` is raised if `lang` doesn't exist. Default is `True`.