



# Speech Recognition with Python

By Maciej Morawski

- [in/mpfmorawski](#) (LinkedIn)
- [@mpfmorawski](#) (GitHub)

# # whoami



[mpfmorawski/README.md](#)

**Welcome!** 🙌

- My name is Maciej Morawski! 🐙
- I'm a Python Backend Developer Intern @ Schneider Electric, living in Warsaw, Poland 🏢
- I have graduated from the MSE in Robotics and Automatic Control @ WUT but...
- That's not the end of my university education! I'm still a Computer Science Student @ WUT !
- I'm interested in Robotics, AI and... all those amazing things you can do using Python 🐍  
(that's why I'm here with you today 😊)
- Today is my first time speaking at such a large conference, but I hope to speak many more times!

# Agenda

1. Speech-to-text intro
2. SpeechRecognition
3. AssemblyAI
4. OpenAI's Whisper
5. Transformers
6. Summary

# Slides

Slides can be found **here**



<https://bit.ly/pyconpl-asr-slides>

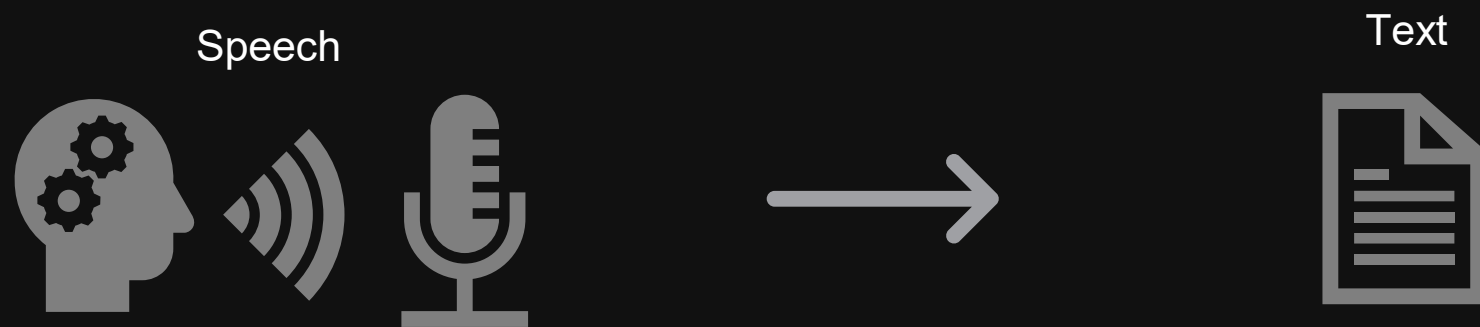
1

# Speech-to-text intro

# Speech-to-text intro

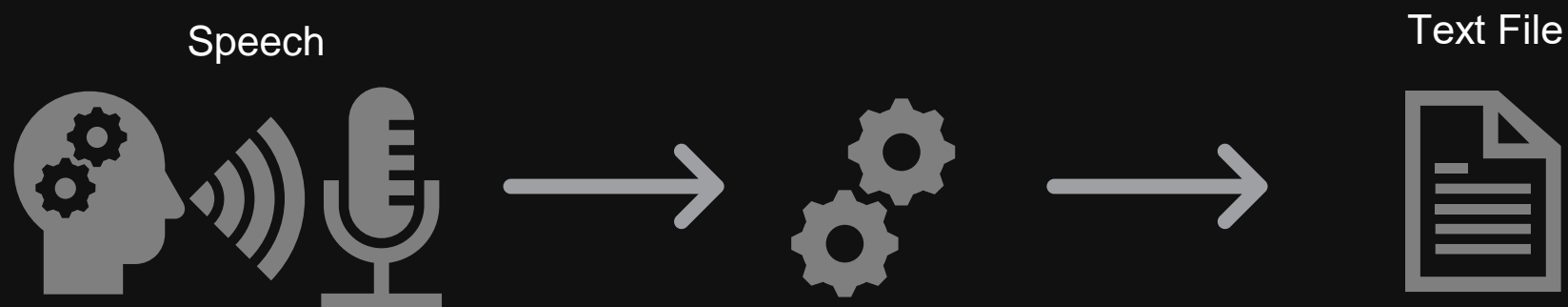
What is... **speech recognition?**

# Speech-to-text intro



**Speech Recognition**

# Speech-to-text intro



**Automatic Speech Recognition (ASR)**

**Speech-To-Text (STT)**



# Speech-to-text intro

Today we will not focus on theory.  
We will answer the question...

What solutions can we use?

# Speech-to-text intro

If your first thought was to make your own  
speech recognition engine

putting it simply...

 **DON'T DO THAT!** 

# Speech-to-text intro

Expertise

Data Requirements

Why it's bad idea to  
build your own  
speech recognition engine?

Time and Cost

Accuracy and Performance

Maintenance and Updates



# Speech-to-text intro

And because...

there are plenty of solutions  
available on the market!

# Speech-to-text intro

Speech recognition engines/APIs can be divided by **their availability** into two main groups:

- tools that can only be used online 
- tools that can be used offline 

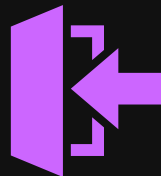
# Speech-to-text intro

But... speech recognition engines/APIs can also be divided by **service providers** into two main groups:

- external solutions



- internal solutions



Hint 💡: Which is much more important from the company's point of view 😊

# Speech-to-text intro

## External solutions



### Advantages (+):

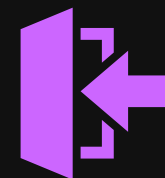
- Easy to use
- Scalability
- No need for infrastructure and maintenance
- Cost (when processing a small amount of audio data)

### Disadvantages (-):

- Privacy concerns
- Dependence on internet connection
- Cost (when processing a large amount of audio data)

# Speech-to-text intro

## Internal solutions



### Advantages (+):

- Data security
- Cost (when processing a large amount of audio data)

### Disadvantages (-):

- Limited scalability
- Maintenance
- Cost (when processing a small amount of audio data)



# Speech-to-text intro

What options are available to us  
as Python Developers 🐍?



# Speech-to-text intro

Let's go straight to practice **together!**




<https://bit.ly/pyconpl-asr-notebook>

2

# SpeechRecognition

# SpeechRecognition

Source: <https://pypi.org/project/SpeechRecognition/>



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## SpeechRecognition 3.10.0

`pip install SpeechRecognition`

 [Latest version](#)

Released: Mar 13, 2023

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

### Navigation

 [Project description](#)

 [Release history](#)

### Project description

pypi

v3.10.0

status

stable

python

3.8 | 3.9 | 3.10

license

BSD



 [Continuous Integration Test Results](#)


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









**UPDATE 2022-02-09:** Hey everyone! This project started as a tech demo, but these days it needs more time than I


# SpeechRecognition

Source: [https://github.com/Uberi/speech\\_recognition](https://github.com/Uberi/speech_recognition)




 Uberi / **speech\_recognition**

Type  to search



 **Code**  Issues **262**  Pull requests **33**  Actions  Projects  Wiki  Security  Insights







 **speech\_recognition** Public

Watch **284** Fork **2.3k** Starred **7.3k**

 master  3 branches  47 tags

[Go to file](#) [Add file](#) [Code](#)

 **ftnext** [chore] Bump up ✓ 8b07762 on Mar 13  **454** commits


	.github	[bugfix] whisper uses walrus, so drop Python 3.7 (su...	3 months ago
	examples	[docs] Add Whisper API example	3 months ago
	reference	[docs] recognize_whisper_api SetupError	3 months ago
	speech_recognition	[chore] Bump up	3 months ago
	tests	[bugfix] Update whisper expected value	3 months ago
	third-party	Bump version and update third party archives	6 years ago


## About


Speech recognition module for Python, supporting several engines and APIs, online and offline.

[pypi.python.org/pypi/SpeechRecognition/](https://pypi.python.org/pypi/SpeechRecognition/)

[audio](#) [python](#) [speech-recognition](#) [speech-to-text](#)


 [Readme](#)

 [BSD-3-Clause, GPL-2.0 licenses found](#)

 [Activity](#)

# SpeechRecognition

Speech recognition engine/API support:

- CMU Sphinx (works offline)
- Google Speech Recognition
- Google Cloud Speech API
- Wit.ai
- Microsoft Azure Speech
- Houndify API
- IBM Speech to Text
- Snowboy Hotword Detection (works offline)
- Tensorflow
- Vosk API (works offline)
- OpenAI whisper (works offline)
- Whisper API  ✨ added in the last release in March ✨

# SpeechRecognition

```
pip install SpeechRecognition
```

# SpeechRecognition

```
import speech_recognition as sr

r = sr.Recognizer()
    with sr.AudioFile(audio_path) as source:
        audio = r.record(source)

r.recognize_google(audio)
```



# SpeechRecognition

Google Speech Recognition (using the default API key)

```
import speech_recognition as sr

r = sr.Recognizer()
with sr.AudioFile(audio_path) as source:
    audio = r.record(source)

try:
    print( r.recognize_google(audio) )
except sr.UnknownValueError:
    print( "Could not understand audio" )
except sr.RequestError as e:
    print( f"Could not request results service; {e}" )
```

# SpeechRecognition

## Google Cloud Speech

```
import speech_recognition as sr

r = sr.Recognizer()
with sr.AudioFile(audio_path) as source:
    audio = r.record(source)

GOOGLE_CLOUD_SPEECH_CREDENTIALS = "CREDENTIALS"

try:
    print( r.recognize_google_cloud(audio, credentials_json=GOOGLE_CLOUD_SPEECH_CREDENTIALS) )
except sr.UnknownValueError:
    print( "Could not understand audio" )

except sr.RequestError as e:
    print( f"Could not request results from service; {e}" )
```

# SpeechRecognition

Wit.ai

```
import speech_recognition as sr

r = sr.Recognizer()
with sr.AudioFile(audio_path) as source:
    audio = r.record(source)

WIT_AI_KEY = „KEY“

try:
    print( r.recognize_wit(audio, key=WIT_AI_KEY) )
except sr.UnknownValueError:
    print( "Could not understand audio" )
except sr.RequestError as e:
    print( f"Could not request results from service; {e}" )
```

# SpeechRecognition

## Houndify

```
import speech_recognition as sr

r = sr.Recognizer()
with sr.AudioFile(audio_path) as source:
    audio = r.record(source)

HOUNDIFY_CLIENT_ID = "ID"
HOUNDIFY_CLIENT_KEY = "KEY"

try:
    print( r.recognize_houndify(audio,
                                client_id=HOUNDIFY_CLIENT_ID,
                                client_key=HOUNDIFY_CLIENT_KEY)
except sr.UnknownValueError:
    print( "Could not understand audio" )
except sr.RequestError as e:
    print( f"Could not request results from service; {e}" )
```

# SpeechRecognition

## IBM Speech to Text


```
import speech_recognition as sr

r = sr.Recognizer()
with sr.AudioFile(audio_path) as source:
    audio = r.record(source)

IBM_USERNAME = „USERNAME"
IBM_PASSWORD = „PASSWORD"

try:
    print( r.recognize_ibm(audio,
                           username=IBM_USERNAME,
                           password=IBM_PASSWORD) )
except sr.UnknownValueError:
    print( "Could not understand audio" )
except sr.RequestError as e:
    print( f"Could not request results from service; {e}" )
```

# SpeechRecognition

Whisper API  ✨ added in the last release in March ✨

```
import speech_recognition as sr

r = sr.Recognizer()
with sr.AudioFile(audio_path) as source:
    audio = r.record(source)

OPENAI_API_KEY = „KEY”

try:
    print( r.recognize_whisper_api(audio, api_key=OPENAI_API_KEY) )
except sr.UnknownValueError:
    print( "Could not understand audio" )
except sr.RequestError as e:
    print( f"Could not request results from service; {e}" )
```

# Speech-to-text intro

Let's get back to Google Colab!



3

AssemblyAI



# AssemblyAI API

Source: <https://www.assemblyai.com/>



AssemblyAI

[Why AssemblyAI](#)

[Models](#)

[Use Cases](#)

[Pricing](#)

[Developers](#)

[Playground](#)

[Your dashboard >](#)

NEW • [Introducing LeMUR, our new framework for applying powerful LLMs to transcribed speech >](#)

## Access powerful AI models to transcribe and understand speech

Our simple API exposes AI models for speech recognition, speaker detection, speech summarization, and more. We build on the latest state-of-the-art AI research to offer production-ready, scalable, and secure AI models through a simple API. Used by thousands of breakthrough startups and dozens of global enterprises for mission-critical workloads.

[Try the API >](#)

[Contact sales >](#)

# AssemblyAI API

Source: <https://www.assemblyai.com/>

Try the API >

Contact sales >

Python

TypeScript

PHP

Ruby

C#

```
1 import requests
2
3 endpoint = "https://api.assemblyai.com/v2/transcript"
4
5 json = {
6     "audio_url": "https://storage.googleapis.com/bucket/b2c31290d9d8.wav"
7 }
8
9 headers = {
10     "Authorization": "c2a41970d9d811ec9d640242ac12",
11     "Content-Type": "application/json"
12 }
13
14 response = requests.post(endpoint, json=json, headers=headers)
15 AssemblyAI nse)
```

# AssemblyAI API

```
import requests
```

# AssemblyAI API

```
endpoint = "https://api.assemblyai.com/v2/transcript"

json = {
    "audio_url": "https://storage.googleapis.com/bucket/b2c31290d9d8.wav"
}

headers = {
    "Authorization": "c2a41970d9d811ec9d640242ac12",
    "Content-Type": "application/json"
}

response = requests.post(endpoint, json=json, headers=headers)
parse(response)
```

# AssemblyAI API

## Output

```
NameError                                Traceback (most recent call last)
<ipython-input-2-3072985cb8ce> in <cell line: 13>()
      11
      12 response = requests.post(endpoint, json=json, headers=headers)
--> 13 parse(response)

NameError: name 'parse' is not defined
```

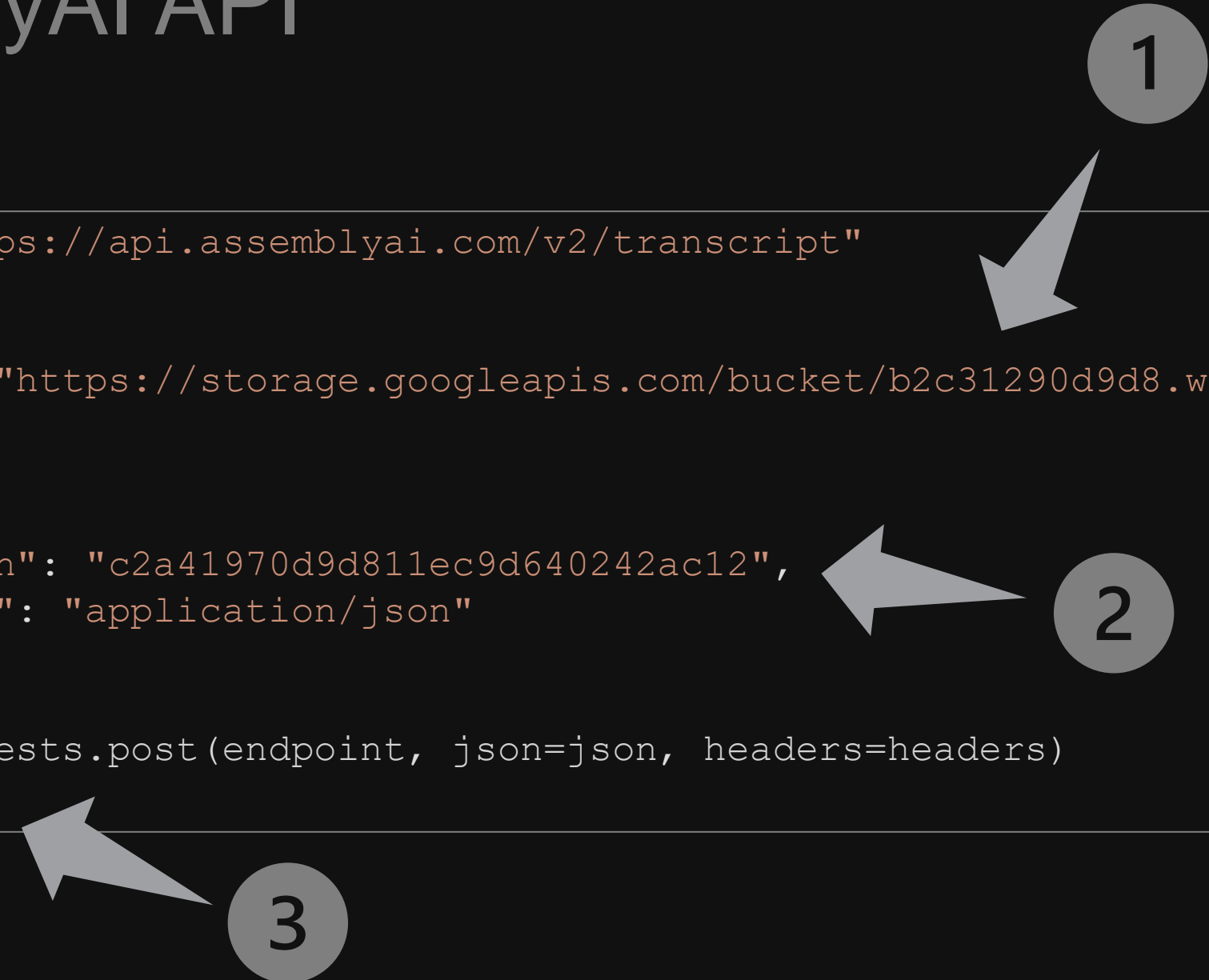
# AssemblyAI API

```
endpoint = "https://api.assemblyai.com/v2/transcript"

json = {
    "audio_url": "https://storage.googleapis.com/bucket/b2c31290d9d8.wav"
}

headers = {
    "Authorization": "c2a41970d9d811ec9d640242ac12",
    "Content-Type": "application/json"
}

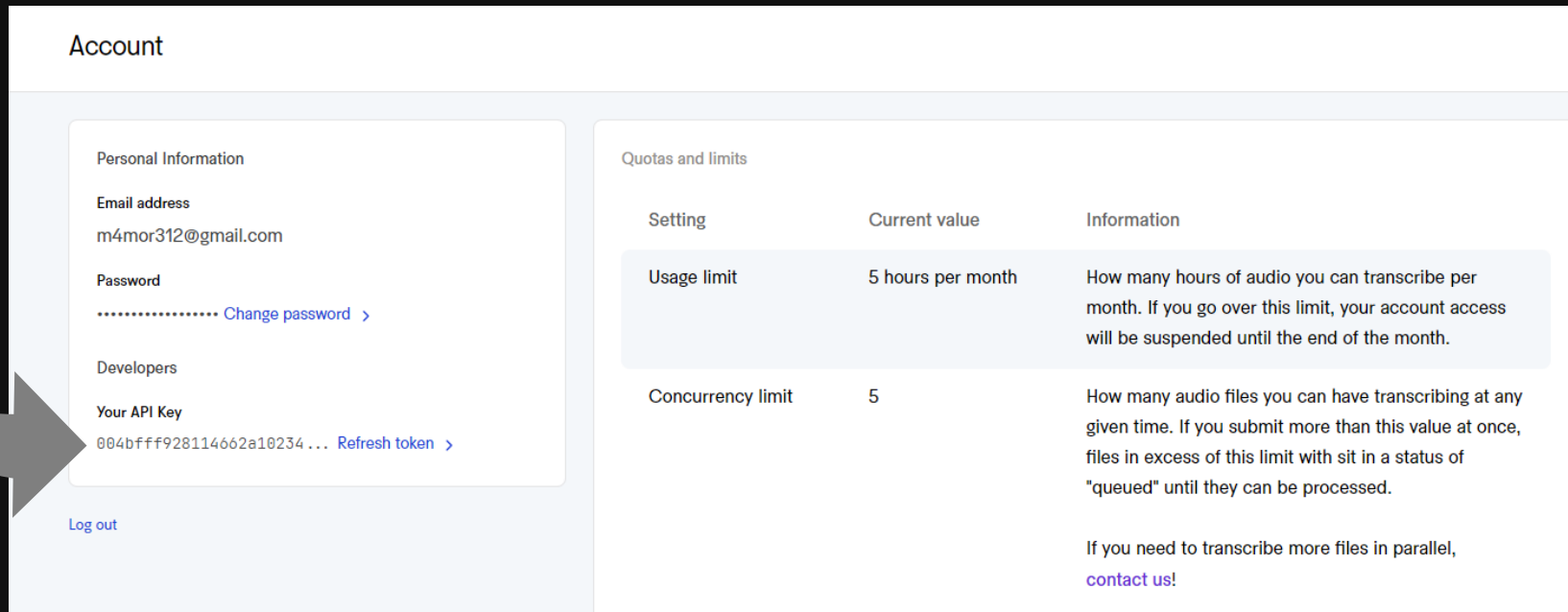
response = requests.post(endpoint, json=json, headers=headers)
parse(response)
```



# AssemblyAI API

## Authorization

1. Go to: <https://www.assemblyai.com/dashboard/signup>
2. Sign up
3. Go to: <https://www.assemblyai.com/app/account>
4. Copy your API Key



The screenshot shows the 'Account' page of the AssemblyAI dashboard. On the left, under 'Personal Information', the 'Your API Key' is displayed as '004bfff928114662a10234 ...' with a 'Refresh token' link. A large grey arrow points from this API key to the 'Authorization' instructions above. On the right, under 'Quotas and limits', a table shows the usage and concurrency limits.

Setting	Current value	Information
Usage limit	5 hours per month	How many hours of audio you can transcribe per month. If you go over this limit, your account access will be suspended until the end of the month.
Concurrency limit	5	How many audio files you can have transcribing at any given time. If you submit more than this value at once, files in excess of this limit will sit in a status of "queued" until they can be processed.

If you need to transcribe more files in parallel, [contact us!](#)

# AssemblyAI API

```
ASSEMBLY_AI_API_KEY = "your_api_key"
```



# AssemblyAI API

## Uploading audio

```
UPLOAD_ENDPOINT = "https://api.assemblyai.com/v2/upload"

headers = {"authorization": ASSEMBLY_AI_API_KEY}
with open(filename, "rb") as f:
    response = requests.post(UPLOAD_ENDPOINT,
                             headers=headers,
                             data=f)

print(response.json())
```

# AssemblyAI API

Output

```
{'upload_url': 'https://cdn.assemblyai.com/upload/random-letters-and-numbers'}
```

# AssemblyAI API

Making request for transcription

```
TRANSCRIPT_ENDPOINT = "https://api.assemblyai.com/v2/transcript"

json = {
    "audio_url": response.json()["upload_url"]
}

headers = {
    "authorization": ASSEMBLY_AI_API_KEY
}

response = requests.post(TRANSCRIPT_ENDPOINT,
                        json=json,
                        headers=headers)

print(response.json())
```

# AssemblyAI API

## Output

```
json full of data!
```

But if we take a closer look...

```
response.json()['text']=None  
response.json()['status']='queued'
```

# AssemblyAI API

## Polling

```
polling_endpoint = f"{TRANSCRIPT_ENDPOINT}/{response.json()['id']}"

while True:
    response = requests.get(polling_endpoint, headers=headers).json()
    if response['status'] == 'completed':
        break
    elif response['status'] == 'error':
        raise RuntimeError(f"Transcription failed: {response['error']}")
    else:
        time.sleep(3)

print(response["text"])
```

# AssemblyAI API

Output

```
Our transcription!
```

# AssemblyAI API

Let's get back to Google Colab!



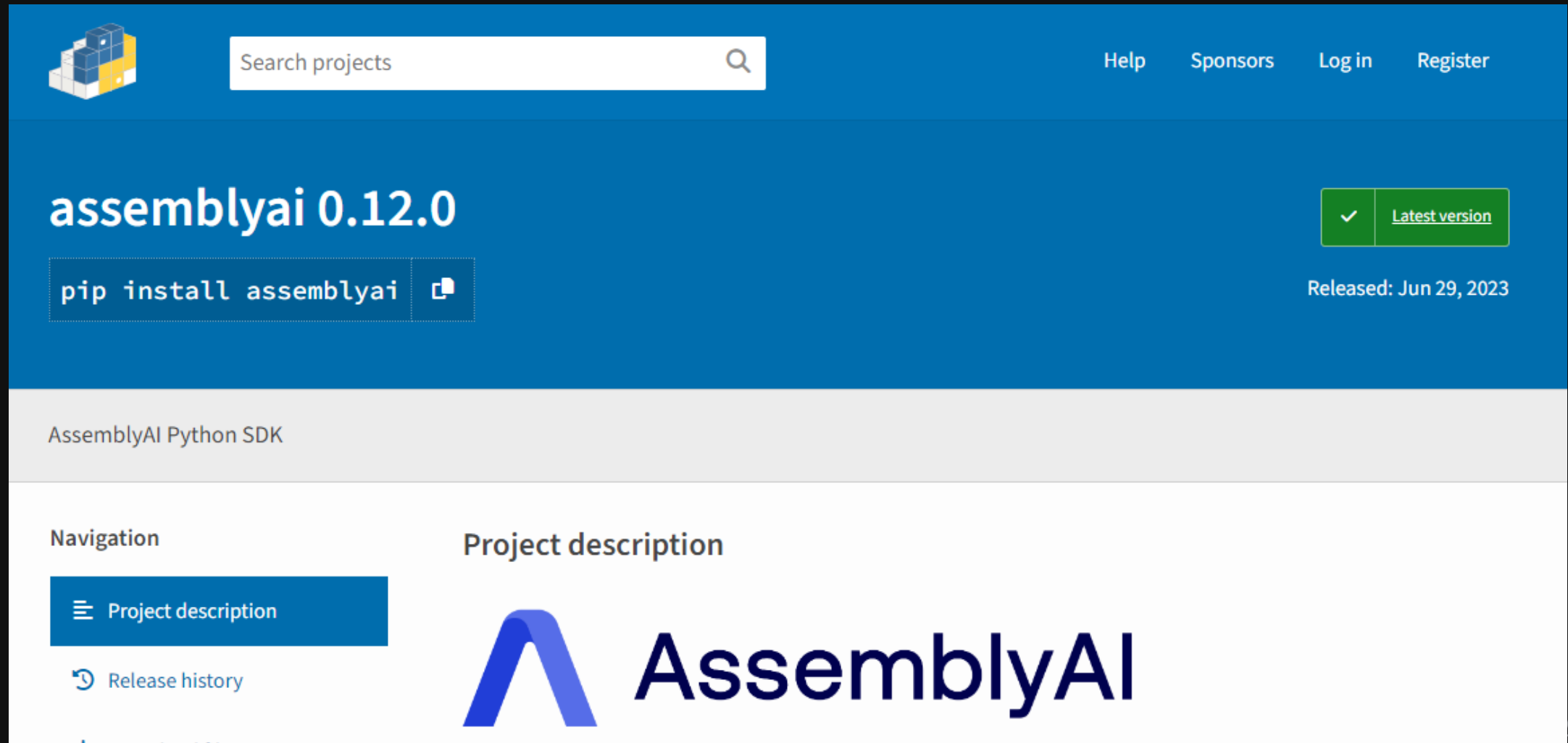
# AssemblyAI API

On the other hand, we can use...



# AssemblyAI API

Source: <https://pypi.org/project/assemblyai/>




The screenshot shows the PyPI project page for AssemblyAI. The header is blue with the AssemblyAI logo (a stack of cubes) on the left, a search bar in the center, and links for Help, Sponsors, Log in, and Register on the right. The main content area has a blue background. It features the project name 'assemblyai' and version '0.12.0' in large white text. To the right of the version is a green button with a checkmark and the text 'Latest version'. Below the version is a white box containing the command 'pip install assemblyai' and a copy icon. To the right of this box is the release date 'Released: Jun 29, 2023'. Below the blue section is a light gray bar with the text 'AssemblyAI Python SDK'. The bottom section has a white background. On the left is a 'Navigation' sidebar with a blue button for 'Project description' and a link for 'Release history'. To the right of the sidebar is the 'Project description' section, which features the AssemblyAI logo (a blue stylized 'A') and the text 'AssemblyAI' in a large, bold, dark blue font.

AssemblyAI Python SDK

Navigation



- Project description
- Release history






Project description

 **AssemblyAI**


# AssemblyAI API

Source: <https://github.com/AssemblyAI/assemblyai-python-sdk>

 AssemblyAI / **assemblyai-python-sdk**




[Code](#) [Issues 2](#) [Pull requests](#) [Discussions](#) [Actions](#) [Projects](#) [Security](#) [Insights](#)

 **assemblyai-python-sdk** Public

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[master](#) [2 branches](#) [15 tags](#) [Go to file](#) [Add file](#) [Code](#)

 **s0h3yl** build: bump version to **0.12.0** ✓ a835be8 2 days ago 🕒 44 commits

📁 .github/workflows	feat: add <b>real-time</b> functionality (#20)	2 weeks ago
📁 assemblyai	fix: allow import of base sdk without extras installed	2 days ago
📁 tests	fix: allow import of base sdk without extras installed	2 days ago
📄 LICENSE	refactor!: update SDK to AssemblyAI API v2 (#1)	2 months ago
📄 README.md	feat: enable retrieval of existing transcripts	2 days ago
📄 assemblyai.png	docs: add <b>AssemblyAI</b> logo (#6)	2 months ago

## About

AssemblyAI's Official Python SDK

[assemblyai.com](#)

- 📖 Readme
- 📄 MIT license
- 📈 Activity
- ★ 40 stars
- 👁 4 watching
- 🔗 0 forks

[Report repository](#)

# AssemblyAI API

```
pip install assemblyai
```

# AssemblyAI API

```
import assemblyai as aai

aai.settings.api_key = "YOUR API KEY"

transcriber = aai.Transcriber()
transcript = transcriber.transcribe(audio_path)

print( transcript.text )
```

# AssemblyAI API

Source: <https://www.assemblyai.com/app>



Upgrade your account

Access real-time transcription, increase your API limits, and more.

Upgrade now >


The banner features a 2x4 grid of eight icons: a document, a smartphone with a speech bubble, a line graph, a clock, a lightning bolt, a speech bubble, a gear, and a wrench. The background of the banner is dark blue with a subtle pattern of horizontal lines.

4

# OpenAI's Whisper

# OpenAI's Whisper

Source: <https://openai.com/research/whisper>

 [Research](#) [Product](#) [Developers](#) [Safety](#) [Company](#)

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September 21, 2022






[Read paper](#)  
[View code](#)  
[View model card](#)


[Speech recognition](#), [Transformers](#), [Open source](#), [Whisper](#), [Milestone](#), [Publication](#), [Release](#)


# OpenAI's Whisper


Source: <https://openai.com/research/whisper>


 openai / **whisper**





 **Code**


 Pull requests **30**


 Discussions


 Actions


 Security


 Insights


 **whisper** Public


 Watch **365**

 Fork **4.5k**

 Starred **40.2k**


 main

 5 branches




 6 tags







Go to file

Add file


 Code


About


 **funboarder13920** fix condition\_on\_previous\_text (#1...  248b6cb on May 5  115 commits


 .github/workflows	Python 3.11 (#1171)	last month
 data	initial commit	9 months ago
 notebooks	Use ndimage.median_filter instead of signal.medfilt...	5 months ago
 tests	Fix truncated words list when the replacement char...	3 months ago
 whisper	fix condition_on_previous_text (#1224)	last month
 flake8	apply formatting with black (#1038)	3 months ago


Robust Speech Recognition via Large-Scale Weak Supervision


 Readme

 MIT license

 Activity

 40.2k stars

 365 watching

 4.5k forks

Report repository



# OpenAI's Whisper

```
pip install -U openai-whisper
```

# OpenAI's Whisper

```
import whisper

model = whisper.load_model(model_name)
transcription = model.transcribe(audio_path)
print(transcription["text"])
```

# OpenAI's Whisper

Source: <https://github.com/openai/whisper>

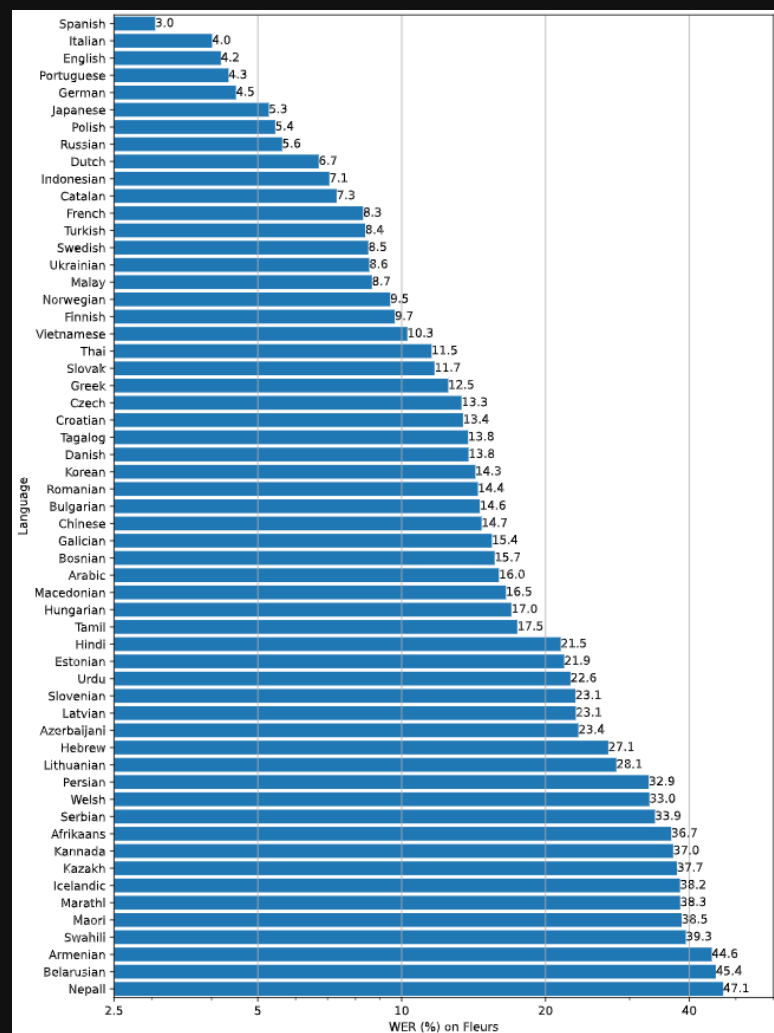
## Available models and languages

There are five model sizes, four with English-only versions, offering speed and accuracy tradeoffs. Below are the names of the available models and their approximate memory requirements and relative speed.

Size	Parameters	English-only model	Multilingual model	Required VRAM	Relative speed
tiny	39 M	<code>tiny.en</code>	<code>tiny</code>	~1 GB	~32x
base	74 M	<code>base.en</code>	<code>base</code>	~1 GB	~16x
small	244 M	<code>small.en</code>	<code>small</code>	~2 GB	~6x
medium	769 M	<code>medium.en</code>	<code>medium</code>	~5 GB	~2x
large	1550 M	N/A	<code>large</code>	~10 GB	1x

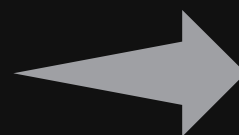
# OpenAI's Whisper

Source: <https://github.com/openai/whisper>



# OpenAI's Whisper

Let's get back to Google Colab!






# Transformers

# Transformers

Source: <https://pypi.org/project/transformers/>



[Help](#) [Sponsors](#) [Log in](#) [Register](#)

## transformers 4.30.2

`pip install transformers`

 [Latest version](#)

Released: Jun 13, 2023

State-of-the-art Machine Learning for JAX, PyTorch and TensorFlow

### Navigation



-  Project description
-  Release history
-  Download files






### Project description




# Transformers

Source: <https://github.com/huggingface/transformers>

 huggingface / transformers










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 transformers Public


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[main](#) [187 branches](#) [129 tags](#) [Go to file](#) [Add file](#) [Code](#)

 **Rocketknight1** Speed up TF tests by reducing ... [134caef](#) 2 hours ago [13,332 commits](#)

 .circleci	Update <a href="#">huggingface_hub</a> commit sha (#24527)	3 days ago
 .github	Unpin DeepSpeed and require DS >= 0.9.3 (#24541)	2 days ago
 docker	pin <a href="#">apex</a> to a speicfc commit (for DeepSpeed CI d...	2 weeks ago
 docs	Check all objects are equally in the main <a href="#">__init__</a> ...	yesterday
 examples	Udate link to RunHouse hardware setup documenta...	7 hours ago
 model_cards	Update URL for Hub PR docs (#17532)	last year

## About

 Transformers: State-of-the-art Machine Learning for Pytorch, TensorFlow, and JAX.

[huggingface.co/transformers](https://huggingface.co/transformers)

[python](#) [nlp](#) [machine-learning](#) [natural-language-processing](#) [deep-learning](#) [tensorflow](#) [pytorch](#) [transformer](#) [speech-recognition](#) [seq2seq](#) [flax](#) [pretrained-models](#) [language-models](#) [nlp-library](#) [language-model](#)



# Transformers

```
pip install transformers
```

# Transformers

```
from transformers import pipeline

transcriber = pipeline("automatic-speech-recognition",
                        model="facebook/wav2vec2-base-960h")
transcription = transcriber(audio_path)
print( transcription["text"] )
```

# Transformers

Let's get back to Google Colab!



# Transformers

Source: <https://huggingface.co/docs/transformers/main/tasks/asr>

Transformers

Search documentation Ctrl+K

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## Automatic speech recognition

Open in Colab Open Studio Lab



Automatic speech recognition (ASR) converts a speech signal to text, mapping a sequence of audio inputs to text outputs. Virtual assistants like Siri and Alexa use ASR models to help users everyday, and there are many other useful user-facing applications like live captioning and note-taking during meetings.

This guide will show you how to:

1. Finetune [Wav2Vec2](#) on the [MnDS-14](#) dataset to transcribe audio to text.
2. Use your finetuned model for inference.

The task illustrated in this tutorial is supported by the following model architectures:  
[Data2VecAudio](#), [Hubert](#), [M-CTC-T](#), [SEW](#), [SEW-D](#), [UniSpeech](#), [UniSpeechSat](#), [Wav2Vec2](#), [Wav2Vec2-Conformer](#), [WavLM](#)

Automatic speech recognition

Load MnDS-14 dataset

Preprocess

Evaluate

Train

Inference

# Transformers

Source: <https://huggingface.co/docs/transformers/main/tasks/asr>

The task illustrated in this tutorial is supported by the following model architectures:

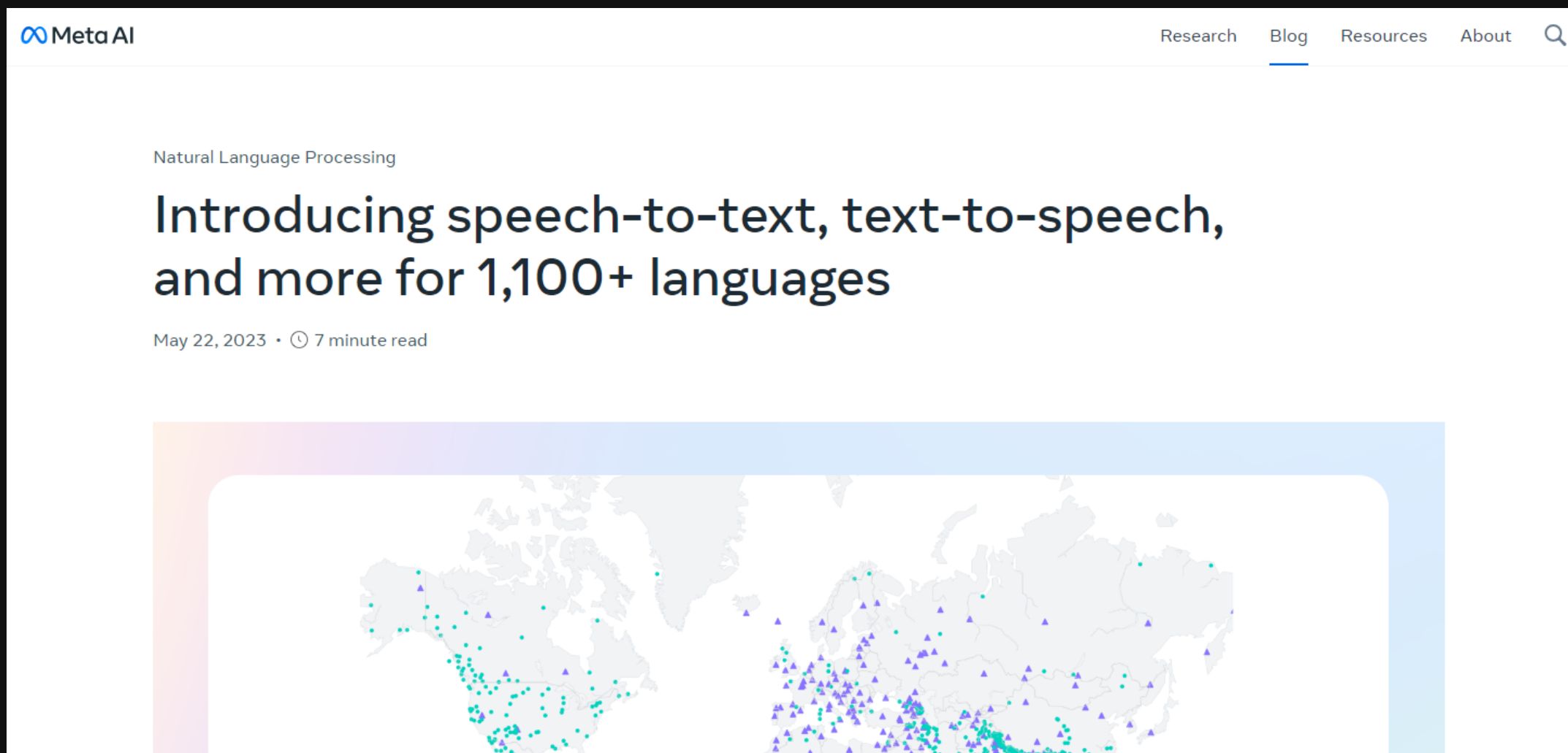
[Data2VecAudio](#), [Hubert](#), [M-CTC-T](#), [SEW](#), [SEW-D](#), [UniSpeech](#), [UniSpeechSat](#), [Wav2Vec2](#), [Wav2Vec2-Conformer](#), [WavLM](#)



# The latest announcement

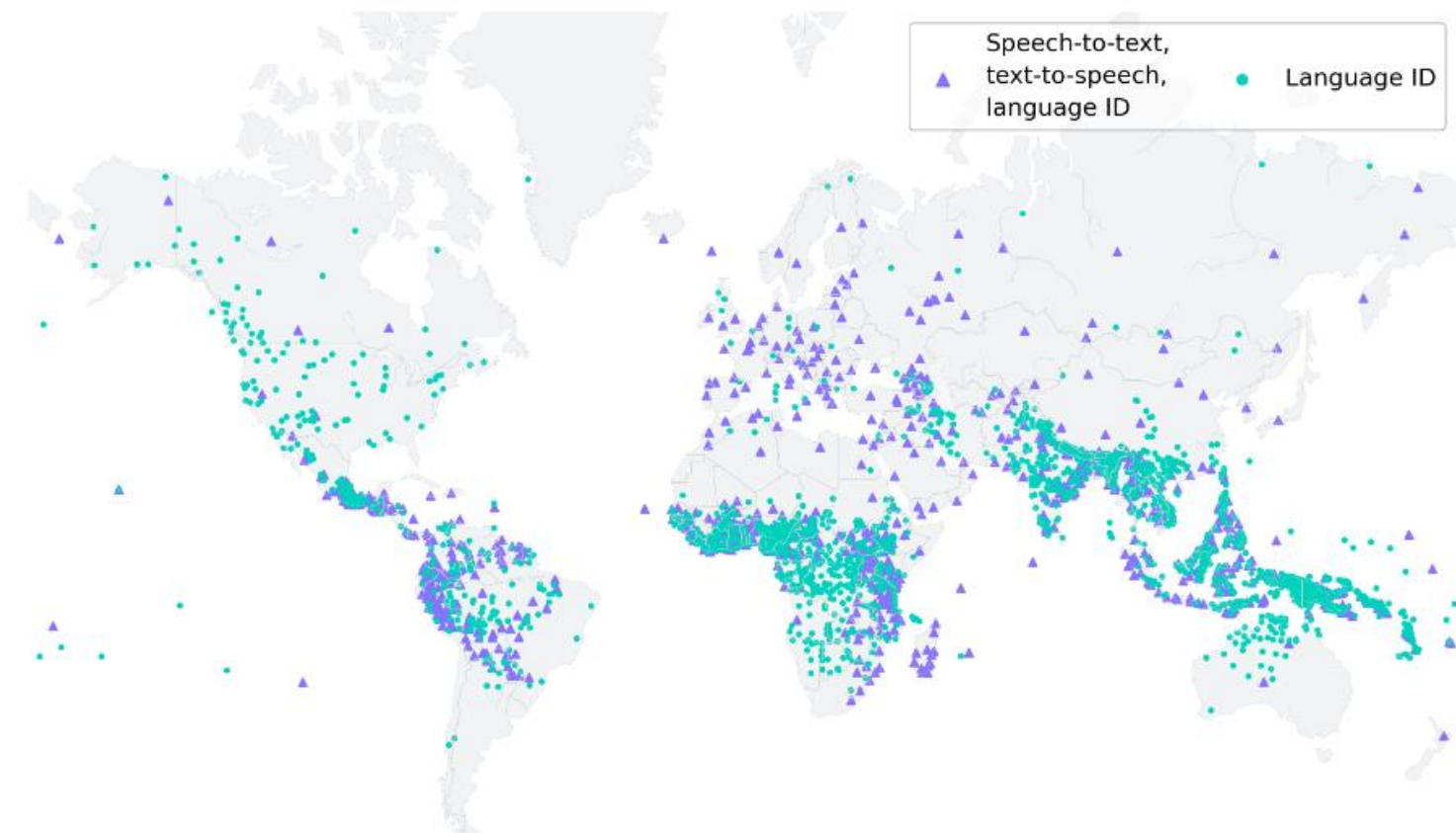
# The latest announcement

Source: <https://ai.facebook.com/blog/multilingual-model-speech-recognition/>



# The latest announcement

Source: <https://ai.facebook.com/blog/multilingual-model-speech-recognition/>



— Illustration of the languages the Massively Multilingual Speech (MMS) recognition model supports. MMS supports speech-to-text and text-to-speech for 1,107 languages and language identification for over 4,000 languages.



# The latest announcement

Source: <https://ai.facebook.com/blog/multilingual-model-speech-recognition/>

We trained multilingual speech recognition models on over 1,100 languages using a 1B parameter wav2vec 2.0 model. As the number of languages increases, performance does decrease, but only very slightly: Moving from 61 to 1,107 languages increases the character error rate by only about 0.4 percent but increases the language coverage by over 18 times.

# The latest announcement

Source: <https://github.com/facebookresearch/fairseq/tree/main/examples/mms>

README.md



## MMS: Scaling Speech Technology to 1000+ languages

The Massively Multilingual Speech (MMS) project expands speech technology from about 100 languages to over 1,000 by building a single multilingual speech recognition model supporting over 1,100 languages (more than 10 times as many as before), language identification models able to identify over [4,000 languages](#) (40 times more than before), pretrained models supporting over 1,400 languages, and text-to-speech models for over 1,100 languages. Our goal is to make it easier for people to access information and to use devices in their preferred language.

You can find details in the paper [Scaling Speech Technology to 1000+ languages](#) and the [blog post](#).

An overview of the languages covered by MMS can be found [here](#).

### Transformers

MMS has been added to Transformers. For more information, please refer to [Transformers' MMS docs](#).

Click [here](#) to find all MMS checkpoints on the Hub.

Checkout the demo here [Open in HF Spaces](#)

# The latest announcement

Source: [https://huggingface.co/docs/transformers/main/en/model\\_doc/mms](https://huggingface.co/docs/transformers/main/en/model_doc/mms)

## Automatic Speech Recognition (ASR)

The ASR model checkpoints can be found here : [mms-1b-fl102](#), [mms-1b-l1107](#), [mms-1b-all](#). For best accuracy, use the `mms-1b-all` model.

### Tips:

- All ASR models accept a float array corresponding to the raw waveform of the speech signal. The raw waveform should be pre-processed with [Wav2Vec2FeatureExtractor](#).
- The models were trained using connectionist temporal classification (CTC) so the model output has to be decoded using [Wav2Vec2CTCTokenizer](#).
- You can load different language adapter weights for different languages via [load\\_adapter\(\)](#). Language adapters only consists of roughly 2 million parameters and can therefore be efficiently loaded on the fly when needed.



# Summary

# Summary

1. Check out the different tools and choose the one that best suits your needs!
2. After this lecture, you can start using speech recognition in your projects right away - just use the examples presented today.

Thank you!

Feel free to ask questions **now**  
or contact me **later!**



# How to contact me?

You can contact me via LinkedIn:

<https://www.linkedin.com/in/mpfmorawski/>



Or check out more on my GitHub:

<https://github.com/mpfmorawski>





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Hope to see you soon!