



openai / **whisper**



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Robust Speech Recognition via Large-Scale Weak Supervision

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Whisper

[\[Blog\]](#) [\[Paper\]](#) [\[Model card\]](#) [\[Colab example\]](#)

Whisper is a general-purpose speech recognition model. It is trained on a large dataset of diverse audio and is also a multitasking model that can perform multilingual speech recognition, speech translation, and language identification.

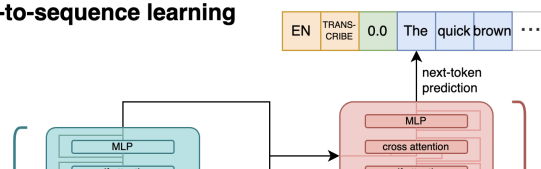
Approach

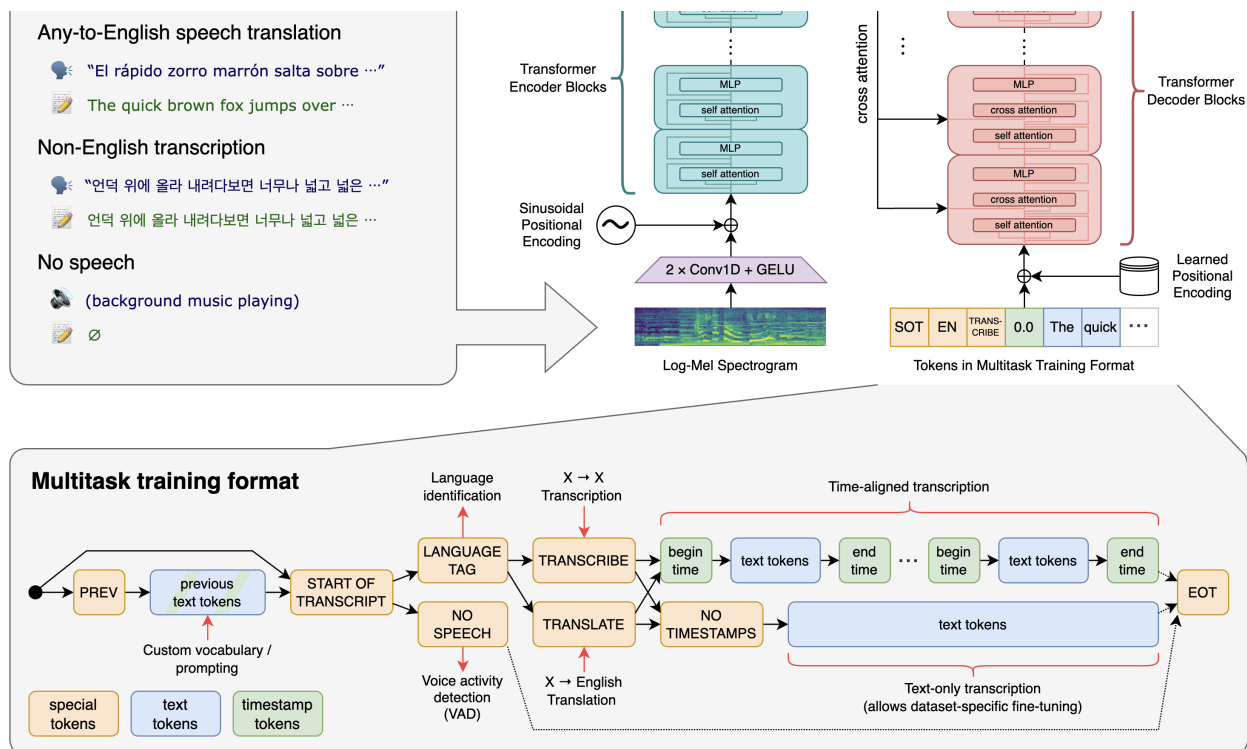
Multitask training data (680k hours)

English transcription

-  "Ask not what your country can do for ..."
-  Ask not what your country can do for ...

Sequence-to-sequence learning





A Transformer sequence-to-sequence model is trained on various speech processing tasks, including multilingual speech recognition, speech translation, spoken language identification, and voice activity detection. These tasks are jointly represented as a sequence of tokens to be predicted by the decoder, allowing a single model to replace many stages of a traditional speech-processing pipeline. The multitask training format uses a set of special tokens that serve as task specifiers or classification targets.

Setup

We used Python 3.9.9 and [PyTorch](#) 1.10.1 to train and test our models, but the codebase is expected to be compatible with Python 3.8-3.11 and recent PyTorch versions. The codebase also depends on a few Python packages, most notably [OpenAI's tiktoken](#) for their fast tokenizer implementation. You can download and install (or update to) the latest release of Whisper with the following command:

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



Alternatively, the following command will pull and install the latest commit from this repository, along with its Python dependencies:

```
pip install git+https://github.com/openai/whisper.git
```



To update the package to the latest version of this repository, please run:

```
pip install --upgrade --no-deps --force-reinstall git+https://github.com/openai/whisper.git
```



It also requires the command-line tool [ffmpeg](#) to be installed on your system, which is available from most package managers:

```
# on Ubuntu or Debian
sudo apt update && sudo apt install ffmpeg
```



```
# on Arch Linux
sudo pacman -S ffmpeg
```

```
# on MacOS using Homebrew (https://brew.sh/)
```

```
# on macOS using Homebrew (https://brew.sh/)
brew install ffmpeg

# on Windows using Chocolatey (https://chocolatey.org/)
choco install ffmpeg

# on Windows using Scoop (https://scoop.sh/)
scoop install ffmpeg
```

You may need [rust](#) installed as well, in case [tiktoken](#) does not provide a pre-built wheel for your platform. If you see installation errors during the `pip install` command above, please follow the [Getting started page](#) to install Rust development environment. Additionally, you may need to configure the `PATH` environment variable, e.g. `export PATH="$HOME/.cargo/bin:$PATH"`. If the installation fails with `No module named 'setuptools_rust'`, you need to install `setuptools_rust`, e.g. by running:

```
pip install setuptools-rust
```



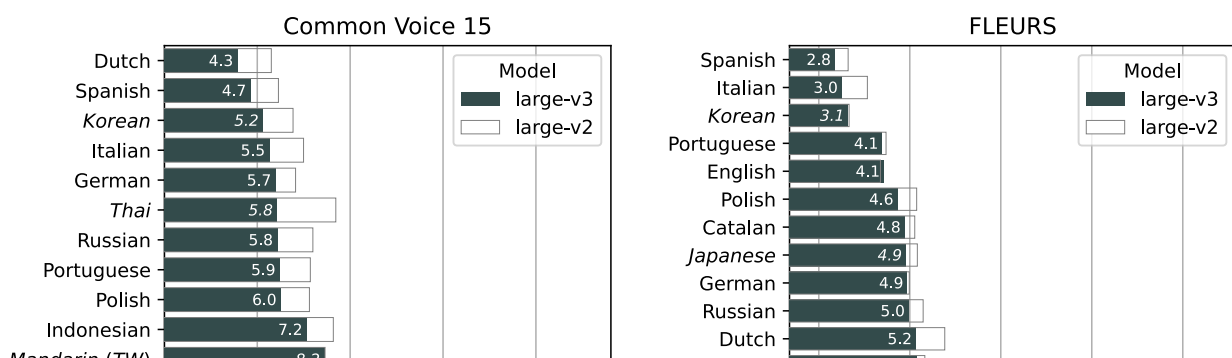
Available models and languages

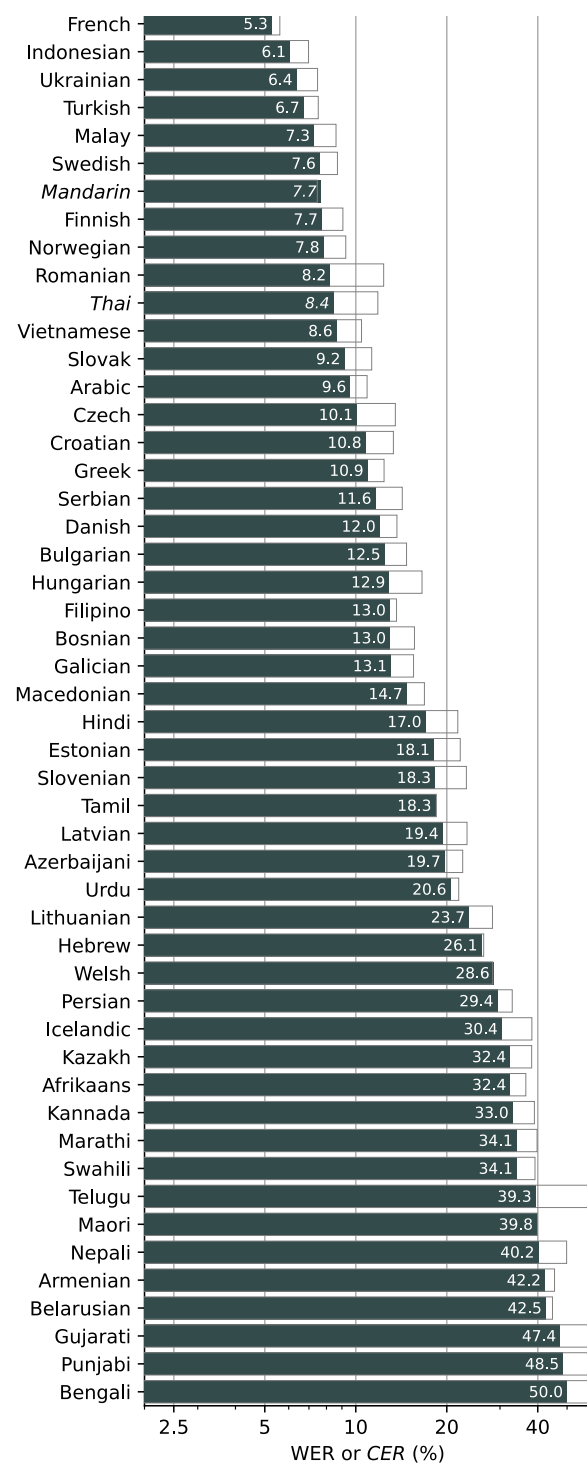
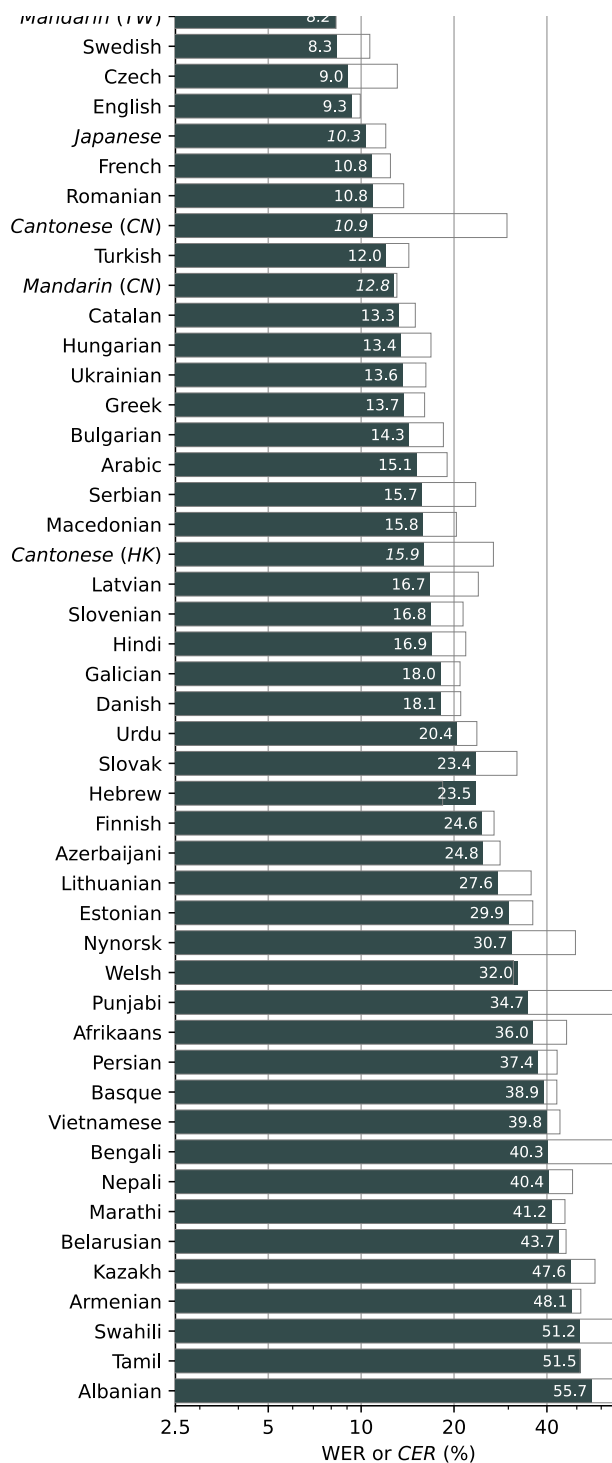
There are six model sizes, four with English-only versions, offering speed and accuracy tradeoffs. Below are the names of the available models and their approximate memory requirements and inference speed relative to the large model. The relative speeds below are measured by transcribing English speech on a A100, and the real-world speed may vary significantly depending on many factors including the language, the speaking speed, and the available hardware.

Size	Parameters	English-only model	Multilingual model	Required VRAM	Relative speed
tiny	39 M	tiny.en	tiny	~1 GB	~10x
base	74 M	base.en	base	~1 GB	~7x
small	244 M	small.en	small	~2 GB	~4x
medium	769 M	medium.en	medium	~5 GB	~2x
large	1550 M	N/A	large	~10 GB	1x
turbo	809 M	N/A	turbo	~6 GB	~8x

The `.en` models for English-only applications tend to perform better, especially for the `tiny.en` and `base.en` models. We observed that the difference becomes less significant for the `small.en` and `medium.en` models. Additionally, the `turbo` model is an optimized version of `large-v3` that offers faster transcription speed with a minimal degradation in accuracy.

Whisper's performance varies widely depending on the language. The figure below shows a performance breakdown of `large-v3` and `large-v2` models by language, using WERs (word error rates) or CER (character error rates, shown in *Italic*) evaluated on the Common Voice 15 and Fleurs datasets. Additional WER/CER metrics corresponding to the other models and datasets can be found in Appendix D.1, D.2, and D.4 of [the paper](#), as well as the BLEU (Bilingual Evaluation Understudy) scores for translation in Appendix D.3.





Command-line usage

The following command will transcribe speech in audio files, using the `turbo` model:

```
whisper audio.flac audio.mp3 audio.wav --model turbo
```



The default setting (which selects the `turbo` model) works well for transcribing English. However, **the `turbo` model is not trained for translation tasks**. If you need to **translate non-English speech into English**, use one of the **multilingual models** (`tiny`, `base`, `small`, `medium`, `large`) instead of `turbo`.

For example, to transcribe an audio file containing non-English speech, you can specify the language:

```
whisper japanese.wav --language Japanese
```



To **translate** speech into English, use:

```
whisper japanese.wav --model medium --language Japanese --task translate
```



Note: The `turbo` model will return the original language even if `--task translate` is specified. Use `medium` or `large` for the best translation results.

Run the following to view all available options:

```
whisper --help
```



See [tokenizer.py](#) for the list of all available languages.

Python usage

Transcription can also be performed within Python:

```
import whisper

model = whisper.load_model("turbo")
result = model.transcribe("audio.mp3")
print(result["text"])
```



Internally, the `transcribe()` method reads the entire file and processes the audio with a sliding 30-second window, performing autoregressive sequence-to-sequence predictions on each window.

Below is an example usage of `whisper.detect_language()` and `whisper.decode()` which provide lower-level access to the model.

```
import whisper

model = whisper.load_model("turbo")

# load audio and pad/trim it to fit 30 seconds
audio = whisper.load_audio("audio.mp3")
audio = whisper.pad_or_trim(audio)

# make log-Mel spectrogram and move to the same device as the model
mel = whisper.log_mel_spectrogram(audio, n_mels=model.dims.n_mels).to(model.device)

# detect the spoken language
_, probs = model.detect_language(mel)
print(f"Detected language: {max(probs, key=probs.get)}")

# decode the audio
options = whisper.DecodingOptions()
result = whisper.decode(model, mel, options)

# print the recognized text
print(result.text)
```



More examples

Please use the 🗣️ [Show and tell](#) category in Discussions for sharing more example usages of Whisper and third-party extensions such as web demos, integrations with other tools, ports for different platforms, etc.

License

Whisper's code and model weights are released under the MIT License. See [LICENSE](#) for further details.

Releases 13

 **v20250625** Latest
on Jun 25

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Languages

● **Python** 100.0%