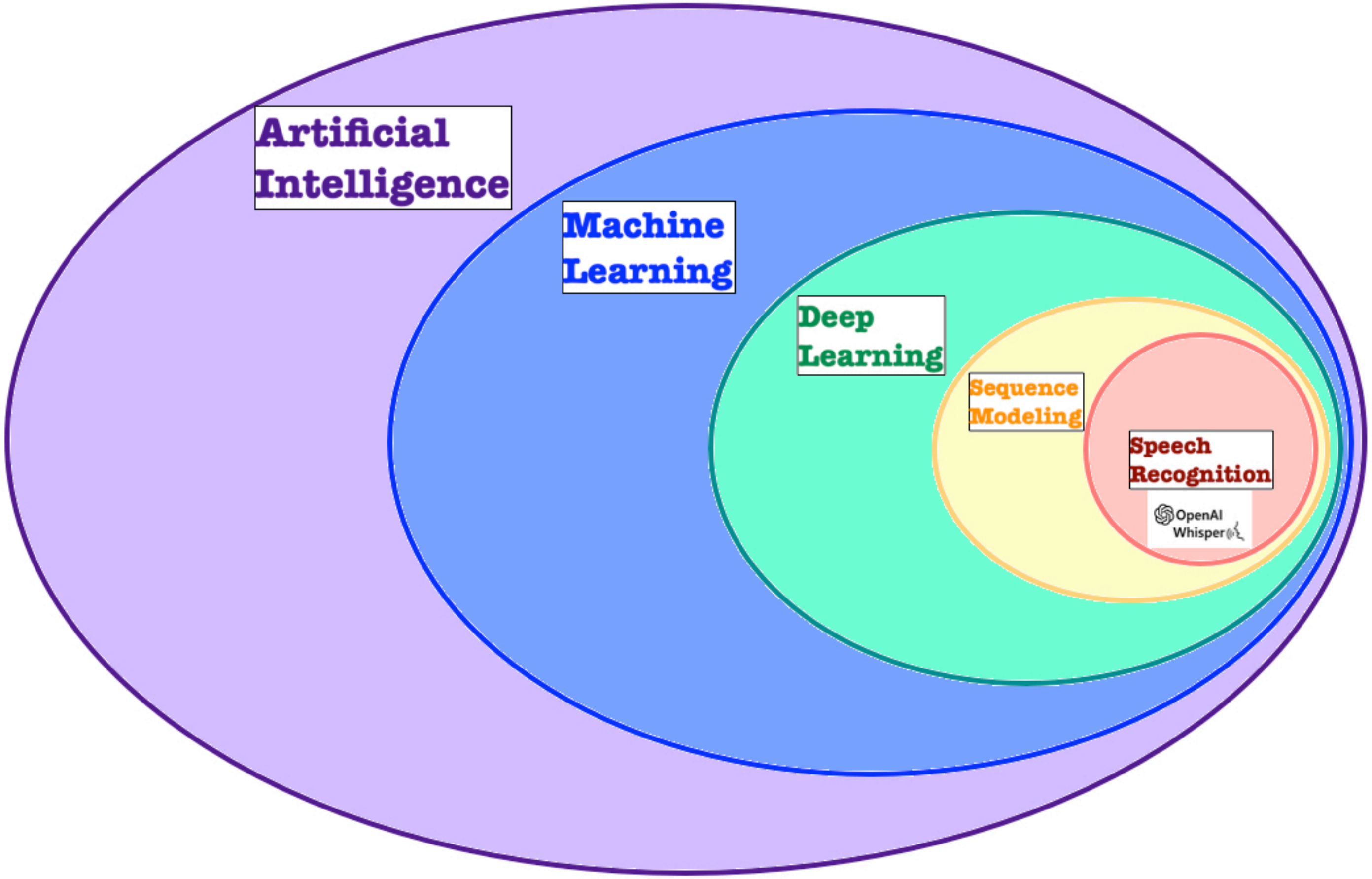

Introduction to
SPEECH
RECOGNITION
with
OpenAI
Whisper
ASR
.mp3 to TXT
.wav to TXT
LIVE
Song LYRICS maker!
No Movie Subtitles No Problem!

Category of Speech Recognition



AI → Machine Learning → Deep Learning → Sequence Modeling → Speech Recognition

Artificial Intelligence

The broad science of creating systems that can perform tasks typically requiring human intelligence.

Machine Learning

A subset of AI where machines learn from data to improve performance on a task without being explicitly programmed.

Deep Learning

A subset of Machine Learning based on neural networks with many layers (hence "deep").

Sequence Modeling

A class of deep learning focused on data that comes in sequences — where order matters.

Speech Recognition

The application of sequence modeling—a branch of deep learning—to convert speech into text.

Workflow:

1. Audio input (e.g., microphone)
2. Preprocessing
3. Sequence model analyzes the waveform over time
4. Decodes speech → words/text

From Audio to Text: Let's Build Automatic Speech Recognition (ASR) now!

Environment Setting

Step 1: Create a Virtual Environment

```
python -m venv whisper-env
# Then activate it:

# macOS / Linux
source whisper-env/bin/activate

# Windows
whisper-env\Scripts\activate
```

Step 2: Install Required Libraries: Torch, Whisper, and FFmpeg

```
pip install torch

pip install faster-whisper

#For macOS Users:    brew install ffmpeg

#For Linux Users:    sudo apt install ffmpeg

#For Windows Users:  Use installer from ffmpeg.org and add to PATH
```

With OpenAI Whisper we get a harmless warning when using CPU:

```
UserWarning: FP16 is not supported on CPU; using FP32 instead
  warnings.warn("FP16 is not supported on CPU; using FP32 instead")
```

If you don't hate warnings, you can just disregard it or suppress it.
But if you're allergic about warnings, you can use Faster-Whisper:

Use compute_type="float32" or "int8" in Faster-Whisper to avoid CPU warnings.

```python
from faster_whisper import WhisperModel # type: ignore

# ==============================
# 1. INPUT
# ==============================
# Specify the input audio file
input_audiofile = "sample1.mp3"

# ==============================
# 2. LOAD MODEL (Sequence Modeling Engine)
# ==============================
# Load the Faster-Whisper model with optimized compute_type for CPU (no FP16 warning)
# compute_type options: "int8", "int8_float16", "float16", "float32"
model = WhisperModel("base", compute_type="float32")

# ==============================
# 3. FULL PIPELINE: Preprocessing + Sequence Modeling + Decoding
# ==============================
# Transcribe the audio file:
# - Preprocessing (audio loading, resampling) is handled automatically
# - Sequence modeling is done using optimized CPU inference
# - Decoding converts audio into readable text
print(f"Transcribing '{input_audiofile}'...")
segments, info = model.transcribe(input_audiofile, language="en")

# Combine all segments into a single string
lyrics = ""
for segment in segments:
    lyrics += segment.text.strip() + "\n"

# ==============================
# 4. OUTPUT
# ==============================
# Specify the output text filename
output_textfile = input_audiofile.replace(".mp3", ".txt")
# Save the transcribed text to a .txt file
with open(output_textfile, "w", encoding="utf-8") as f:
    f.write(lyrics)

print(f"Transcription complete. Lyrics saved to '{output_textfile}'")
```

What happens when you run the program for the first time?

Faster-Whisper model components are downloaded from the cloud.

config.json describes the model's architecture (e.g., number of layers, hidden size, attention heads)

vocabulary.txt contains a list of tokens/words the model recognizes – essential for decoding

tokenizer.json contains rules for converting raw audio/text <-> token IDs.

It handles both directions — it's the bridge between text and the model's numerical world.

```
gabriel@MacBookAir-2 speech-recognition % python 01_audio_to_text_file.py
config.json: 100%|██████████████████████████████| 2.31k/2.31k [00:00<00:00, 12.3MB/s]
vocabulary.txt: 100%|██████████████████████████████| 460k/460k [00:13<00:00, 35.3kB/s]
tokenizer.json: 100%|██████████████████████████████| 2.20M/2.20M [01:17<00:00, 28.4kB/s]
tokenizer.json: 100%|██████████████████████████████| 2.20M/2.20M [01:17<00:00, 28.4kB/s]
model.bin: 100%|██████████████████████████████████| 145M/145M [01:55<00:00, 621kB/s]
Transcribing 'sample1.mp3'...
Transcription complete. Lyrics saved to 'sample1.txt'
```

Two Key Roles of tokenizer.json:

1) Converts text (e.g., "muraho neza") into a list of token IDs the model understands (e.g., [4852, 2091, 1245])

Why? Neural networks can't understand words — only numbers. So we "tokenize" the text into IDs.

2) Converts model's predicted token IDs (e.g., [4852, 2091, 1245]) back into readable text ("muraho neza")

Why? After the model "thinks" in token IDs, we need to turn those IDs back into something humans can read.

To explore more, I changed the directory to the cache (check the path to the cache on your PC):

cd /Users/gabriel/.cache/huggingface/hub/models--Systran--faster-whisper-base/snapshots/ebe41f70d5b6dfa9166e2c581c45c9c0cfc57b66'

Hugging Face

Hugging Face is the platform powering modern AI models, making it easy to use, train, and share speech recognition systems — including Whisper.

It's "Model Hub" can be thought as a GitHub for AI models

We can:
🔍 Search for existing models (Whisper, Wav2Vec2, XLS-R, etc.)
📥 Download pretrained weights (what our script does the first time)
☁️ Upload your fine-tuned models so our students (or the world) can use them!
🔗 Example: https://huggingface.co/openai/whisper-large

For Code and Documents

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