

video-summarizer-anne-joan-benita

April 24, 2024

1 Download YouTube Video's Audio

```
[ ]: [!] pip install pytube -q
```

57.6/57.6 kB

832.3 kB/s eta 0:00:00

```
[ ]: from pytube import YouTube
```

```
[ ]: VIDEO_URL = 'https://www.youtube.com/watch?v=h-JVjs9AAmQ' #batman
```

```
[ ]: yt = YouTube(VIDEO_URL)
```

```
[ ]: yt.streams \
    .filter(only_audio=True, file_extension='mp4') \
    .first() \
    .download(filename='ytaudio.mp4')
```

```
[ ]: '/content/ytaudio.mp4'
```

```
[ ]: [!] ffmpeg -i ytaudio.mp4 -acodec pcm_s16le -ar 16000 ytaudio.wav
```

ffmpeg version 4.4.2-0ubuntu0.22.04.1 Copyright (c) 2000-2021 the FFmpeg developers

built with gcc 11 (Ubuntu 11.2.0-19ubuntu1)

configuration: --prefix=/usr --extra-version=0ubuntu0.22.04.1

--toolchain=hardened --libdir=/usr/lib/x86_64-linux-gnu

--incdir=/usr/include/x86_64-linux-gnu --arch=amd64 --enable-gpl --disable-stripping --enable-gnutls --enable-ladspa --enable-libaom --enable-libass

--enable-libbluray --enable-libs2b --enable-libcaca --enable-libcdio --enable-

libcodec2 --enable-libdav1d --enable-libflite --enable-libfontconfig --enable-

libfreetype --enable-libfribidi --enable-libgme --enable-libgsm --enable-libjack

--enable-libmp3lame --enable-libmysofa --enable-libopenjpeg --enable-libopenmpt

--enable-libopus --enable-libpulse --enable-librabbitmq --enable-librubberband

--enable-libshine --enable-libsnappy --enable-libsoxr --enable-libspeex

--enable-libsrt --enable-libssh --enable-libtheora --enable-libtwolame --enable-

libvidstab --enable-libvorbis --enable-libvpx --enable-libwebp --enable-libx265

```

--enable-libxml2 --enable-libxvid --enable-libzimg --enable-libzmq --enable-
libzvbi --enable-lv2 --enable-omx --enable-openal --enable-openc1 --enable-
opengl --enable-sdl2 --enable-pocketsphinx --enable-librsvg --enable-libmfx
--enable-libdc1394 --enable-libdrm --enable-libiec61883 --enable-chromaprint
--enable-frei0r --enable-libx264 --enable-shared
libavutil      56. 70.100 / 56. 70.100
libavcodec     58.134.100 / 58.134.100
libavformat    58. 76.100 / 58. 76.100
libavdevice    58. 13.100 / 58. 13.100
libavfilter    7.110.100 / 7.110.100
libswscale     5.  9.100 / 5.  9.100
libswresample  3.  9.100 / 3.  9.100
libpostproc   55.  9.100 / 55.  9.100
Input #0, mov,mp4,m4a,3gp,3g2,mj2, from 'ytaudio.mp4':
  Metadata:
    major_brand      : dash
    minor_version    : 0
    compatible_brands: iso6mp41
    creation_time    : 2022-03-03T20:20:21.000000Z
  Duration: 00:04:15.65, start: 0.000000, bitrate: 48 kb/s
  Stream #0:0(eng): Audio: aac (HE-AAC) (mp4a / 0x6134706D), 44100 Hz, stereo,
  fltp, 1 kb/s (default)
    Metadata:
      creation_time    : 2022-03-03T20:20:21.000000Z
      handler_name      : ISO Media file produced by Google Inc.
      vendor_id         : [0][0][0][0]
Stream mapping:
  Stream #0:0 -> #0:0 (aac (native) -> pcm_s16le (native))
Press [q] to stop, [?] for help
Output #0, wav, to 'ytaudio.wav':
  Metadata:
    major_brand      : dash
    minor_version    : 0
    compatible_brands: iso6mp41
    ISFT              : Lavf58.76.100
  Stream #0:0(eng): Audio: pcm_s16le ([1][0][0][0] / 0x0001), 16000 Hz, stereo,
  s16, 512 kb/s (default)
    Metadata:
      creation_time    : 2022-03-03T20:20:21.000000Z
      handler_name      : ISO Media file produced by Google Inc.
      vendor_id         : [0][0][0][0]
      encoder          : Lavc58.134.100 pcm_s16le
size= 15978kB time=00:04:15.65 bitrate= 512.0kbits/s speed= 258x
video:0kB audio:15978kB subtitle:0kB other streams:0kB global headers:0kB muxing
overhead: 0.000477%

```

2 English ASR with HuggingSound

```
[ ]: !pip install torch==2.2.1
      !pip install huggingsound -q
```

```
Requirement already satisfied: torch==2.2.1 in /usr/local/lib/python3.10/dist-
packages (2.2.1+cu121)
Requirement already satisfied: filelock in /usr/local/lib/python3.10/dist-
packages (from torch==2.2.1) (3.13.3)
Requirement already satisfied: typing-extensions>=4.8.0 in
/usr/local/lib/python3.10/dist-packages (from torch==2.2.1) (4.10.0)
Requirement already satisfied: sympy in /usr/local/lib/python3.10/dist-packages
(from torch==2.2.1) (1.12)
Requirement already satisfied: networkx in /usr/local/lib/python3.10/dist-
packages (from torch==2.2.1) (3.2.1)
Requirement already satisfied: Jinja2 in /usr/local/lib/python3.10/dist-packages
(from torch==2.2.1) (3.1.3)
Requirement already satisfied: fsspec in /usr/local/lib/python3.10/dist-packages
(from torch==2.2.1) (2023.6.0)
Collecting nvidia-cuda-nvrtc-cu12==12.1.105 (from torch==2.2.1)
  Downloading nvidia_cuda_nvrtc_cu12-12.1.105-py3-none-manylinux1_x86_64.whl
(23.7 MB)
23.7/23.7 MB
36.8 MB/s eta 0:00:00
Collecting nvidia-cuda-runtime-cu12==12.1.105 (from torch==2.2.1)
  Downloading nvidia_cuda_runtime_cu12-12.1.105-py3-none-manylinux1_x86_64.whl
(823 kB)
823.6/823.6
kB 60.6 MB/s eta 0:00:00
Collecting nvidia-cuda-cupti-cu12==12.1.105 (from torch==2.2.1)
  Downloading nvidia_cuda_cupti_cu12-12.1.105-py3-none-manylinux1_x86_64.whl
(14.1 MB)
14.1/14.1 MB
49.9 MB/s eta 0:00:00
Collecting nvidia-cudnn-cu12==8.9.2.26 (from torch==2.2.1)
  Downloading nvidia_cudnn_cu12-8.9.2.26-py3-none-manylinux1_x86_64.whl (731.7
MB)
731.7/731.7
MB 1.4 MB/s eta 0:00:00
Collecting nvidia-cublas-cu12==12.1.3.1 (from torch==2.2.1)
  Downloading nvidia_cublas_cu12-12.1.3.1-py3-none-manylinux1_x86_64.whl (410.6
MB)
141.7/410.6 MB 2.0 MB/s eta 0:02:17
```

```
[ ]: from huggingsound import SpeechRecognitionModel
```

```
[ ]: import torch
device = "cuda" if torch.cuda.is_available() else "cpu"

[ ]:

[ ]: model = SpeechRecognitionModel("jonatasgrosman/wav2vec2-large-xlsr-53-english",
↳ device = device)
```

OUT OF MEMORY (OOM) error

3 Audio Chunking

```
[ ]: import librosa

[ ]: input_file = '/content/ytaudio.wav'

[ ]: print(librosa.get_samplerate(input_file))

# Stream over 30 seconds chunks rather than load the full file
stream = librosa.stream(
    input_file,
    block_length=30,
    frame_length=16000,
    hop_length=16000
)

[ ]: import soundfile as sf

[ ]: for i,speech in enumerate(stream):
    sf.write(f'{i}.wav', speech, 16000)

[ ]:
```

4 Audio Transcription / ASR / Speech to Text

```
[ ]: audio_path = []
for a in range(i+1):
    audio_path.append(f'/content/{a}.wav')

[ ]: audio_path

[ ]: transcriptions = model.transcribe(audio_path)

[ ]: full_transcript = ' '
```

```
[ ]: for item in transcriptions:
      full_transcript += ''.join(item['transcription'])
```

```
[ ]: len(full_transcript)
```

```
[ ]:
```

5 Text Summarization

```
[ ]: from transformers import pipeline
```

```
[ ]: summarization = pipeline('summarization')
```

```
[ ]: summarized_text = summarization(full_transcript)
```

```
[ ]: summarized_text[0]['summary_text']
```

Text Chunking before Summarization

```
[ ]: num_iters = int(len(full_transcript)/1000)
      summarized_text = []
      for i in range(0, num_iters + 1):
          start = 0
          start = i * 1000
          end = (i + 1) * 1000
          #print("input text \n" + full_transcript[start:end])
          out = summarization(full_transcript[start:end], min_length = 5, max_length=20)
          out = out[0]
          out = out['summary_text']
          # print("Summarized text\n"+out)
          summarized_text.append(out)

      #print(summarized_text)
```

```
[ ]:
```