

Reconnaissance vocale et transcription (STT) avec Vosk

En recherchant un programme de transcription de l'audio à l'écrit (**STT** pour "Speak To Text") je suis tombé sur un [article intéressant](#).

Parmi les différents programmes proposés sur l'article il a fallu trier ceux qu'on pouvait piloter depuis Python ne garder que ceux qui offraient des modèles entraînés avec la langue française. On aurait très bien pu entraîner notre propre modèle si on avait eu des heures d'extraits audio sous le coude (avec transcription écrite à la main d'humain) et des floppées de GigaFlops.

Comme ce n'est pas le cas, mon choix s'est arrêté sur **VOSK**

Installation

Je n'ai pas réussi à installer **VOSK** avec une version de Python inférieure à la 3.8 donc il est conseillé de mettre à jour votre interpréteur Python si besoin.

Il vous faudra également **PIP**, le récupérateur de packets Python.

```
$ apt install python3-pip
```

C'est une bonne pratique de créer un environnement virtuel avant d'installer les packets nécessaires :

```
$ python3.8 -m venv env
$ source env/bin/activate
$ pip3 install vosk
```

Dans le cas où vous voulez pouvoir utiliser un micro (pour de la transcription en temps réel par exemple) il faudra également installer la librairie `sounddevice`

```
$ pip3 install sounddevice
```

Bien, mais de base VOSK est vierge de tout apprentissage. Il faut donc lui fournir un modèle pré-entraîné sur la langue de votre choix.

Utilisation de modèles pré-entraînés

Quelques modèles sont proposés à l'adresse suivante : <https://alphacephei.com/vosk/models>

J'ai eu l'occasion de tester deux modèles :

<https://alphacephei.com/vosk/models/vosk-model-small-fr-pguyot-0.3.zip> Très léger (<50 Mo) mais peu précis. Conseillé pour les machines peu performantes, téléphonie mobile ou Raspberry Pi.

<https://alphacephei.com/vosk/models/vosk-model-fr-0.6-linto-2.2.0.zip> Plus gros (1,5 Go) mais bien meilleures performances. C'est le modèle qui a été utilisé lors de la résidence "Artificialité Insolente" sur l'installation "Nathalie".

Les modèles sont à décompresser dans le dossier model

Sur la page <https://github.com/alphacep/vosk-api/tree/master/python/example> on peut trouver plusieurs scripts Python pour interroger le modèle de plusieurs façons.

Transcription d'un fichier audio vers fichier texte

Utilisation avec le modèle vosk-model-en-us-0.22

Bizarrement le script plante avec ce modèle, après ce dernier message:

```
LOG (VoskAPI:ReadDataFiles():model.cc:307) Loading RNNLM model from model/vosk-model-en-us-0.22/rnnlm/final.raw
```

La solution consiste a supprimer le dossier rnnlm...

Transcription depuis un microphone

test_microphone.py (cliquer pour afficher le code)

test_microphone.py

```
#!/usr/bin/env python3

import argparse
import os
import queue
import sounddevice as sd
import vosk
import sys

q = queue.Queue()

def int_or_str(text):
    """Helper function for argument parsing."""
    try:
        return int(text)
    except ValueError:
        return text

def callback(indata, frames, time, status):
    """This is called (from a separate thread) for each audio block."""
    if status:
        print(status, file=sys.stderr)
    q.put(bytes(indata))

parser = argparse.ArgumentParser(add_help=False)
parser.add_argument(
    '-l', '--list-devices', action='store_true',
    help='show list of audio devices and exit')
args, remaining = parser.parse_known_args()
if args.list_devices:
    print(sd.query_devices())
    parser.exit(0)
parser = argparse.ArgumentParser(
    description=__doc__,
    formatter_class=argparse.RawDescriptionHelpFormatter,
    parents=[parser])
parser.add_argument(
    '-f', '--filename', type=str, metavar='FILENAME',
    help='text file to store transcriptions')
parser.add_argument(
    '-m', '--model', type=str, metavar='MODEL_PATH',
    help='Path to the model')
parser.add_argument(
    '-d', '--device', type=int_or_str,
    help='input device (numeric ID or substring)')
parser.add_argument(
    '-r', '--samplerate', type=int, help='sampling rate')
args = parser.parse_args(remaining)

try:
    if args.model is None:
        args.model = "model"
    if not os.path.exists(args.model):
        print("Please download a model for your language from https://alphacephei.com/vosk/models")
        print("and unpack as 'model' in the current folder.")
        parser.exit(0)
    if args.samplerate is None:
        device_info = sd.query_devices(args.device, 'input')
        # soundfile expects an int, sounddevice provides a float:
        args.samplerate = int(device_info['default_samplerate'])

    model = vosk.Model(args.model)

    if args.filename:
        dump_fn = open(args.filename, "a")
    else:
        dump_fn = None

    with sd.RawInputStream(samplerate=args.samplerate, blocksize = 1024, device=args.device, dtype='int16',
                           channels=1, latency='high', callback=callback):
        print('#' * 80)
        print('Press Ctrl+C to stop the recording')
```

```

print('#' * 80)

rec = vosk.KaldiRecognizer(model, args.samplerate)
while True:
    data = q.get()
    if rec.AcceptWaveform(data):
        r = eval(rec.Result())
        t = r["text"]
        if t:
            print(t)
            if dump_fn is not None and len(t) > 5:
                dump_fn.write(t+'\n')

except KeyboardInterrupt:
    print('\nDone')
    parser.exit(0)
except Exception as e:
    parser.exit(type(e).__name__ + ': ' + str(e))

```

Entraînement d'un nouveau modèle linguistique

Tutoriaux

http://kaldi-asr.org/doc/kaldi_for_dummies.html

http://kaldi-asr.org/doc/data_prep.html

<https://www.eleanorchodroff.com/tutorial/kaldi/training-acoustic-models.html>

<https://towardsdatascience.com/how-to-start-with-kaldi-and-speech-recognition-a9b7670ffff6>

Installation de Kaldi et initialisation du projet

Kaldi est un kit d'outils pour la création de modèles linguistiques. Les modèles sont ensuite utilisés par VOSK pour faciliter leur utilisation pour la reconnaissance vocale.

Cloner le répo de Kaldi : <https://github.com/kaldi-asr/kaldi>

Créer un nouveau dossier pour le projet dans le dossier egs (mycorpus dans l'exemple ci-dessous)

Recréer l'arborescence ci-dessous à partir du dossier mycorpus (*les lignes rouges pointillées sont des liens symboliques*) :



```

cd mycorpus
ln -s ../wsj/s5/steps .
ln -s ../wsj/s5/utils .
ln -s ../../src .
cp ../wsj/s5/path.sh .

```

Traitement des fichiers son

Détection des silences et des non silences avec Python

<https://librosa.org/>

Création des fichiers du dossier 'data/train'

Les fichiers essentiels à la création d'un modèle kaldi sont : wav.scp, utt2spk, spk2utt et text.

Fichier 'text'

```

s5# head -3 data/train/text
sw02001-A_000098-001156 HI UM YEAH I'D LIKE TO TALK ABOUT HOW YOU DRESS FOR WORK AND
sw02001-A_001980-002131 UM-HUM
sw02001-A_002736-002893 AND IS

```

The first element on each line is the utterance-id, which is an arbitrary text string, but if you have speaker information in your setup, you should make the speaker-id a prefix of the utterance id; this is important for reasons relating to the sorting

of these files. The rest of the line is the transcription of each sentence. You don't have to make sure that all words in this file are in your vocabulary; out of vocabulary words will get mapped to a word specified in the file data/lang/oov.txt.

It needs to be the case that when you sort both the `utt2spk` and `spk2utt` files, the orders “agree”, e.g. the list of speaker-ids extracted from the `utt2spk` file is the same as the string sorted order. The easiest way to make this happen is to make the speaker-ids a prefix of the utterance-id. Although, in this particular example we have used an underscore to separate the “speaker” and “utterance” parts of the utterance-id, in general it is probably safer to use a dash (“-”). This is because it has a lower ASCII value; if the speaker-ids vary in length, in certain cases the speaker-ids and their corresponding utterance ids can end up being sorted in different orders when using the standard “C”-style ordering on strings, which will lead to a crash. Another important file is `wav.scp`. In the Switchboard example,

Fichier 'wav.scp'

```
s5# head -3 data/train/wav.scp
sw02001-A /home/dpovey/kaldi-trunk/tools/sph2pipe_v2.5/sph2pipe -f wav -p -c 1 /export/corpora3/LDC/LDC97S62/swb1/sw02001.sph |
sw02001-B /home/dpovey/kaldi-trunk/tools/sph2pipe_v2.5/sph2pipe -f wav -p -c 2 /export/corpora3/LDC/LDC97S62/swb1/sw02001.sph |
```

The format of this file is

`<recording-id> <extended-filename>`

where the “extended-filename” may be an actual filename, or as in this case, a command that extracts a wav-format file. The pipe symbol on the end of the extended-filename specifies that it is to be interpreted as a pipe. We will explain what “recording-id” is below, but we would first like to point out that if the “segments” file does not exist, the first token on each line of “wav.scp” file is just the utterance id. The files in `wav.scp` must be single-channel (mono); if the underlying wav files have multiple channels, then a `sox` command must be used in the `wav.scp` to extract a particular channel.

Fichier 'segments'

In the Switchboard setup we have the “segments” file, so we'll discuss this next.

```
s5# head -3 data/train/segments
```

```
sw02001-A_000098-001156 sw02001-A 0.98 11.56
sw02001-A_001980-002131 sw02001-A 19.8 21.31
sw02001-A_002736-002893 sw02001-A 27.36 28.93
```

The format of the “segments” file is:

`<utterance-id> <recording-id> <segment-begin> <segment-end>`

where the segment-begin and segment-end are measured in seconds. These specify time offsets into a recording. The “recording-id” is the same identifier as is used in the “wav.scp” file– again, this is an arbitrary identifier that you can choose.

Fichier 'utt2spk'

The last file you need to create yourself is the “utt2spk” file. This says, for each utterance, which speaker spoke it.

```
s5# head -3 data/train/utt2spk
sw02001-A_000098-001156 2001-A
sw02001-A_001980-002131 2001-A
sw02001-A_002736-002893 2001-A
```

The format is

`<utterance-id> <speaker-id>`

Note that the speaker-ids don't need to correspond in any very accurate sense to the names of actual speakers– they simply need to represent a reasonable guess. In this case we assume each conversation side (each side of the telephone conversation) corresponds to a single speaker. This is not entirely true – sometimes one person may hand the phone to another person, or the same person may be speaking in multiple calls – but it's good enough for our purposes. If you have no information at all about the speaker identities, you can just make the speaker-ids the same as the utterance-ids, so the format of the file would be just `<utterance-id> <utterance-id>`. We have made the previous sentence bold because we have

encountered people creating a “global” speaker-id. This is a bad idea because it makes cepstral mean normalization ineffective in training (since it's applied globally), and because it will create problems when you use `utils/split_data_dir.sh` to split your data into pieces.

Fichier 'reco2file_and_channel' (optionnel)

The file “reco2file_and_channel” is only used when scoring (measuring error rates) with NIST's “sclite” tool:

```
s5# head -3 data/train/reco2file_and_channel
sw02001-A sw02001 A
sw02001-B sw02001 B
sw02005-A sw02005 A
```

The format is:

<recording-id> <filename> <recording-side (A or B)>

The filename is typically the name of the .sph file, without the suffix, but in general it's whatever identifier you have in your “stm” file. The recording side is a concept that relates to telephone conversations where there are two channels, and if not, it's probably safe to use “A”. If you don't have an “stm” file or you have no idea what this is all about, then you don't need the “reco2file_and_channel” file.

Fichier 'spk2gender' (optionnel)

There is another file that exists in some setups; it is used only occasionally and not in the Kaldi system build. We show what it looks like in the Resource Management (RM) setup:

```
s5# head -3 ../../rm/s5/data/train/spk2gender
adg0 f
ahh0 m
ajp0 m
```

This file maps from speaker-id to either “m” or “f” depending on the speaker gender.

Une fois tous les fichiers créés, lancer la commande :

```
$ utils/fix_data_dir.sh data/train
```

Fichiers du dossier 'data/local/lang'

Fichiers de données liés au langage. On doit fournir : `lexicon.txt`, `nonsilence_phones.txt`, `optional_silence.txt`, `silence_phones.txt` et `extra_questions.txt` (optionnel).

Fichier 'lexicon.txt'

Le fichier `lexicon.txt` contient la liste de chaque mot du corpus (en majuscule), suivi de sa prononciation phonétique.

Exemple:

```
WORD W ER D
LEXICON L EH K S IH K AH N
```

The pronunciation alphabet must be based on the same phonemes you wish to use for your acoustic models. You must also include lexical entries for each “silence” or “out of vocabulary” phone model you wish to train.

<https://en.wikipedia.org/wiki/ARPABET>

On peut s'aider de cet outil en ligne pour la construction du fichier `lexicon.txt` (mais se limite à la prononciation anglaise) : <http://www.speech.cs.cmu.edu/tools/lextool.html>

Liens pour la phonétique de la langue bretonne :

- <https://en.wikipedia.org/wiki/Help:IPA/Breton>
- https://en.m.wikiversity.org/wiki/Breton/Breton_pronunciation
- http://www.kervarker.org/en/courseintro_03_noid.html
- https://en.wikipedia.org/wiki/Breton_language#Geographic_distribution_and_dialects

Fichier 'nonsilence_phones.txt'

Liste de tous les phonèmes utilisés dans notre corpus

Article extrait de : <http://lesporteslogiques.net/wiki/> - **WIKI Les Portes Logiques**

Adresse : <http://lesporteslogiques.net/wiki/ressource/logiciel/vosk/start?rev=1635683432>

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