

Wake Word Detector

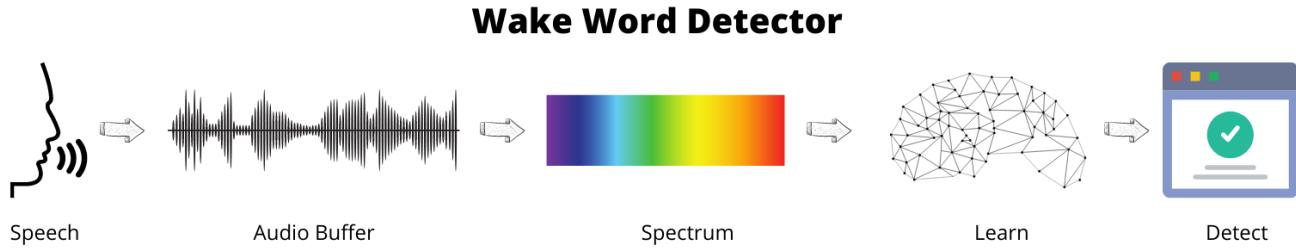


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Background

Personal Assistant devices like Google Home, Alexa and Apple Homepod, will be constantly listening for specific set of wake words like "Ok, Google" or "Alexa" or "Hey Siri", and once these sequence of words are detected it would prompt to user for next commands and respond to them appropriately.

Introduction

To create a open-source custom wake word detector, which will take audio as input and once the sequence of words are detected then prompt to the user.

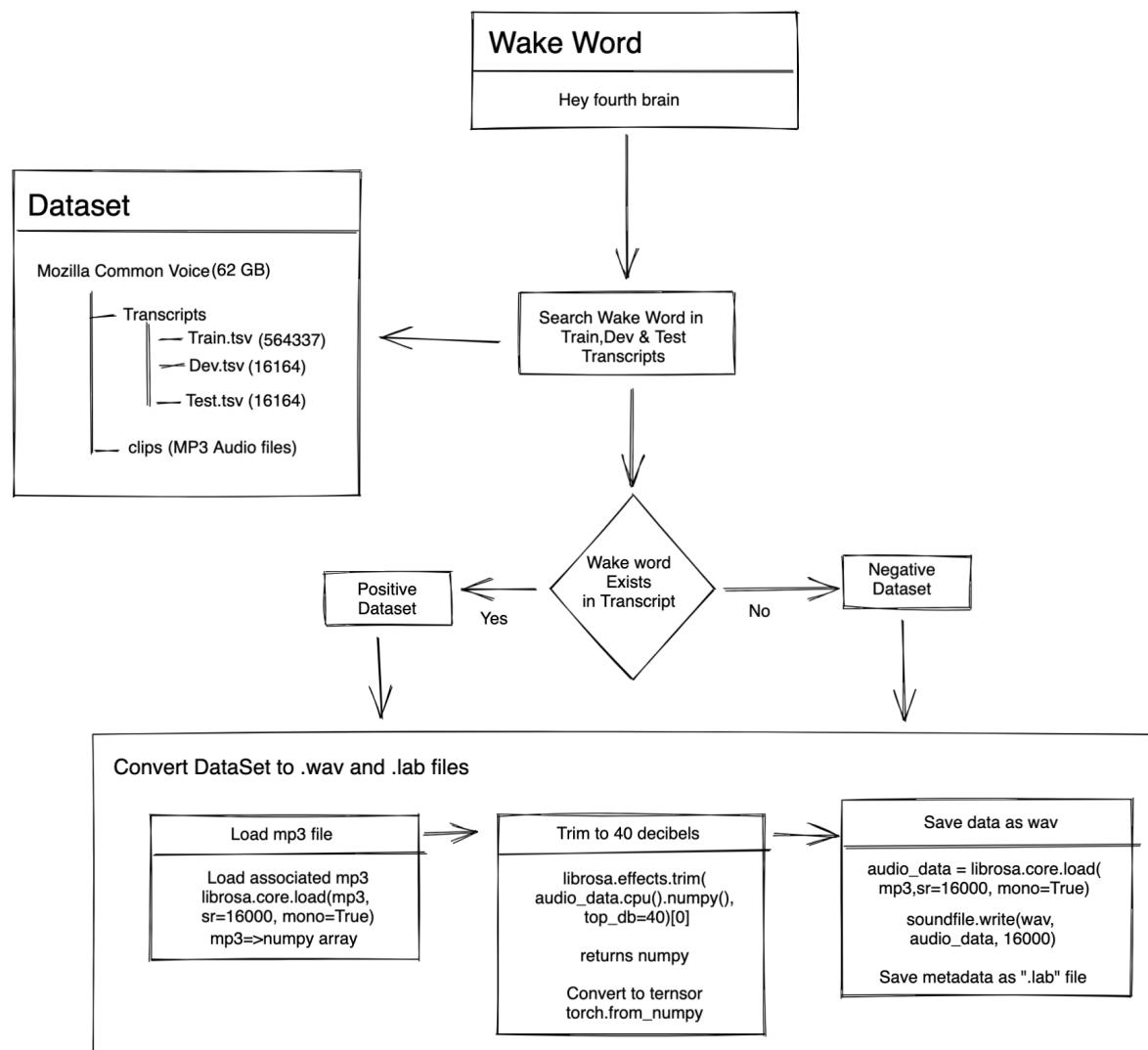
Goal is to provide configurable custom detector so that anyone can use it on their own application to perform operations, once configured wake words are detected.

Implementation

Preparing labelled dataset

Used [Mozilla Common Voice dataset](#),

- Go through each wake word and check transcripts for match
- If found then it will be in positive dataset
- If not found then it will be in negative dataset
- Load appropriate mp3 files and trim the silence parts
- save as .wav file and transcript as .lab file
- Code reference [fetch_dataset_mcv.py](#)



Word Alignment

- For positive dataset, used [Montreal Forced Alignment](#) to get timestamps of each word in audio.
- Download the [stable version](#)

```
wget https://github.com/MontrealCorpusTools/Montreal-Forced-Aligner/releases/download/v1.0.1/montreal-forced-aligner_linux.tar.gz
tar -xf montreal-forced-aligner_linux.tar.gz
rm montreal-forced-aligner_linux.tar.gz
```

- Download the [LibriSpeech Lexicon dictionary](#)

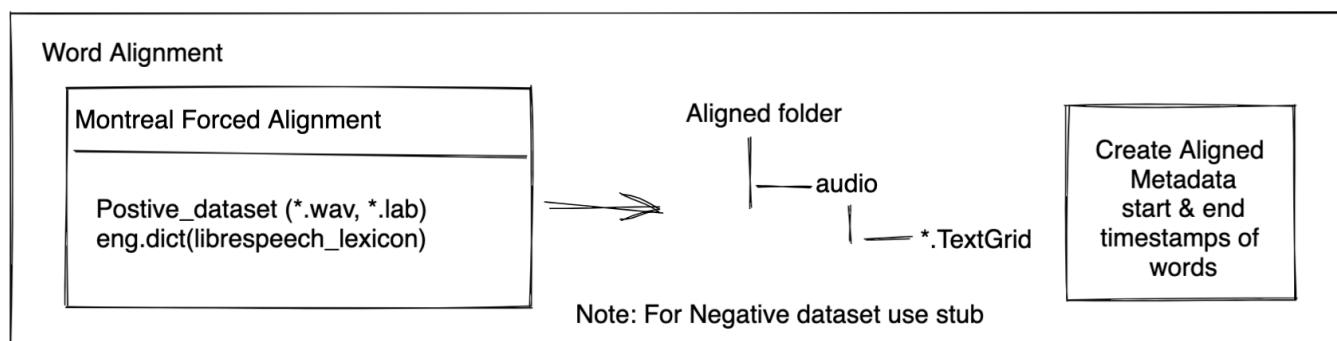
```
wget https://www.openslr.org/resources/11/librispeech-lexicon.txt
```

- Known issues in MFA

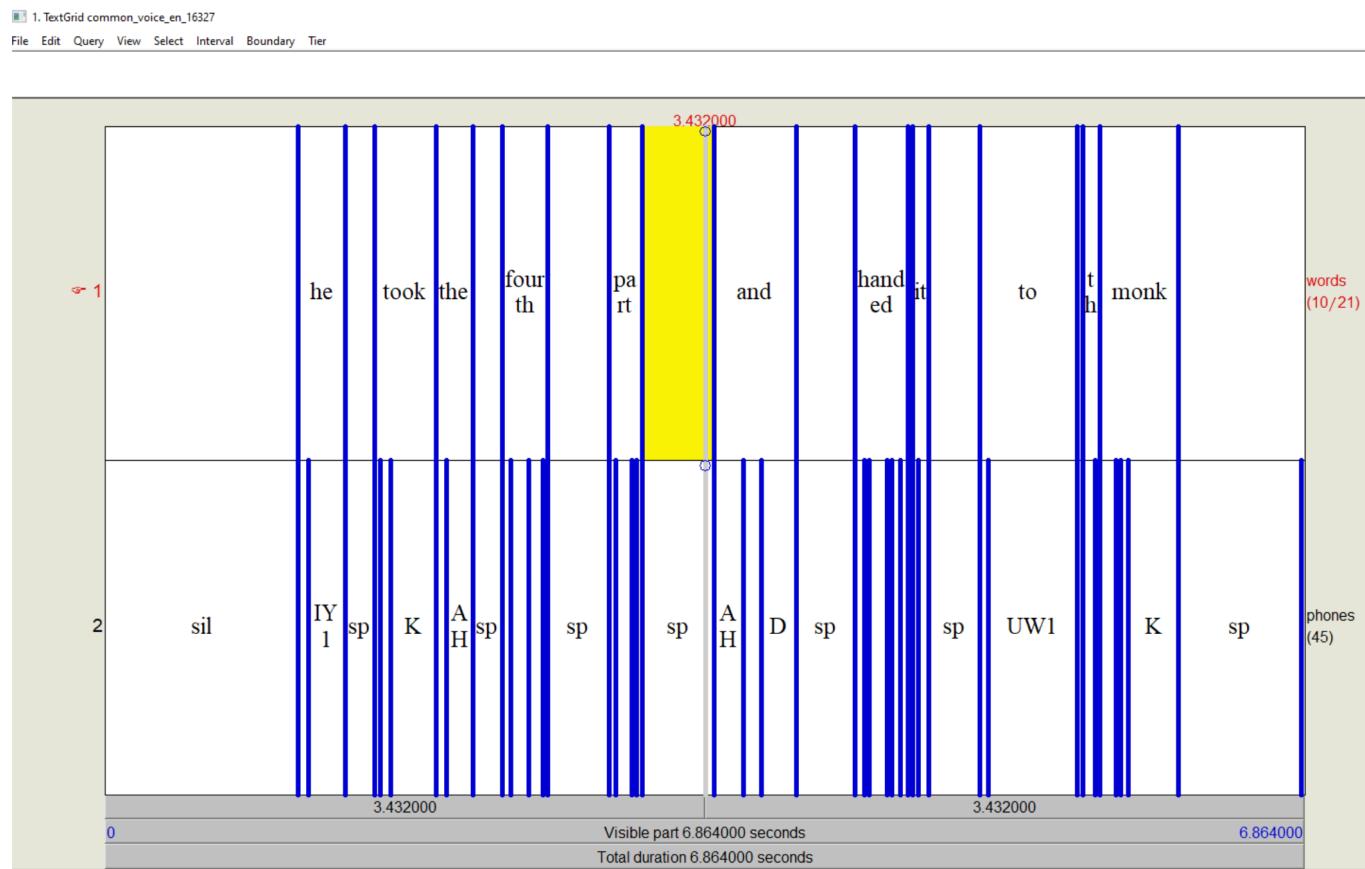
```
# known mfa issue https://github.com/MontrealCorpusTools/Montreal-Forced-Aligner/issues/109
cp montreal-forced-aligner/lib/libpython3.6m.so.1.0 montreal-forced-aligner/lib/libpython3.6m.so
cd montreal-forced-aligner/lib/thirdparty/bin && rm libopenblas.so.0
&& ln -s ../../libopenblas-p0-8dca6697.3.0.dev.so libopenblas.so.0
```

- Creating aligned data

```
montreal-forced-aligner\bin\mfa_align -q positive\audio librispeech-lexicon.txt montreal-forced-aligner\pretrained_models\english.zip
aligned_data
```



Generated textgrid file



Fix data imbalance

Check for any data imbalance, if the dataset does not have enough samples containing wake words, consider using text to speech services to generate more samples.

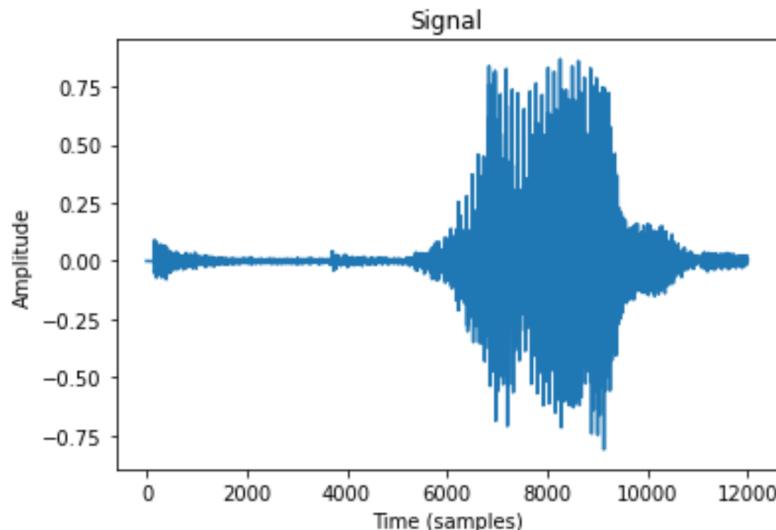
- Used Google Text To Speech Api, set environment variable `GOOGLE_APPLICATION_CREDENTIALS` with your key.
- Used various speed rates, pitches and voices to generate data for wake words.
- Code [generate_dataset_google_tts.py](#)

Extract audio features

- Below is how sound looks like when plotted on time (x-axis) and amplitude (y-axis)

```
import librosa
sounddata = librosa.core.load("hey.wav", sr=16000, mono=True)[0]

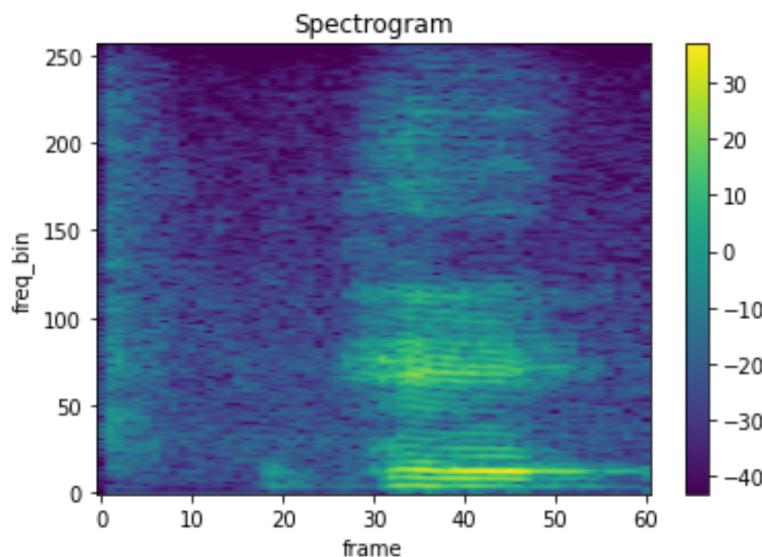
# plotting the signal in time series
plt.plot(sounddata)
plt.title('Signal')
plt.xlabel('Time (samples)')
plt.ylabel('Amplitude')
```



- When Short-time Fourier transform (STFT) computed, below is how spectrogram looks like

```
from torchaudio.transforms import Spectrogram
spectrogram = Spectrogram(n_fft=512, hop_length=200)
spectrogram.to(device)

inp = torch.from_numpy(sounddata).float().to(device)
hey_spectrogram = spectrogram(inp.float())
plot_spectrogram(hey_spectrogram.cpu(), title="Spectrogram")
```

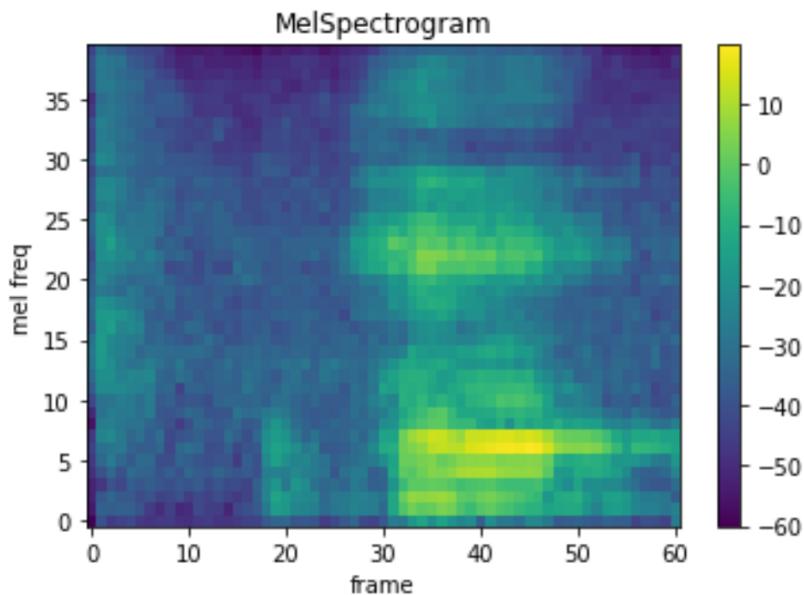


- A mel spectrogram is a spectrogram where the frequencies are converted to the mel scale.

```
from torchaudio.transforms import MelSpectrogram
mel_spectrogram = MelSpectrogram(n_mels=40, sample_rate=16000,
                                 n_fft=512, hop_length=200,
                                 norm="slaney")

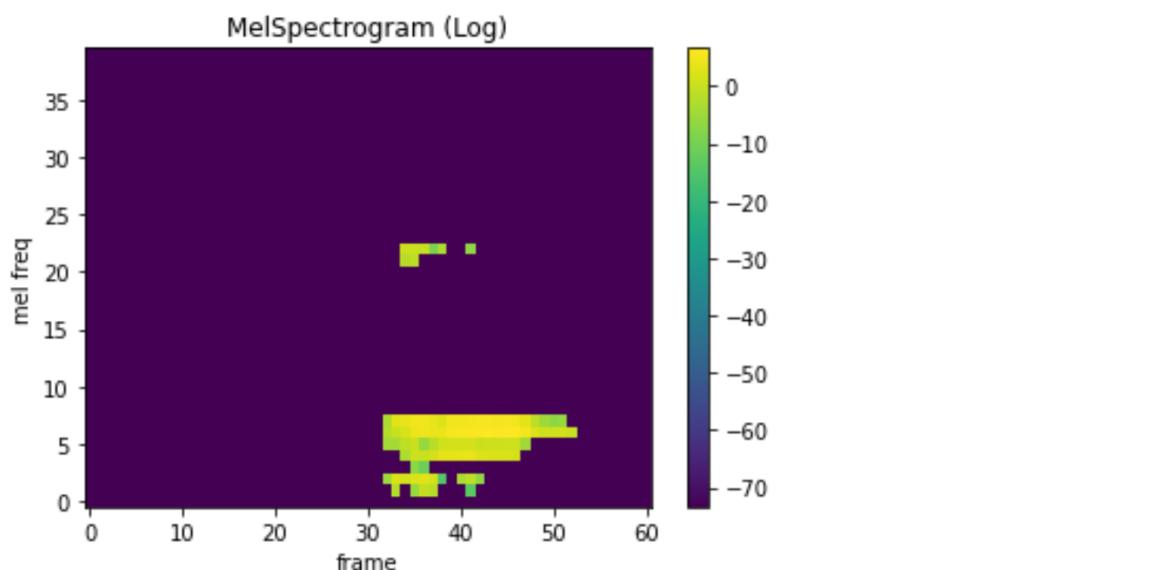
mel_spectrogram.to(device)
inp = torch.from_numpy(sounddata).float().to(device)
hey_mels_slaney = mel_spectrogram(inp.float())
```

```
plot_spectrogram(hey_mels_slaney.cpu(), title="MelSpectrogram",  
ylabel='mel freq')
```



- After adding offset and taking log on mels, below is how final mel spectrogram looks like

```
log_offset = 1e-7  
log_hey_mel_specgram = torch.log(hey_mels_slaney + log_offset)  
plot_spectrogram(log_hey_mel_specgram.cpu(), title="MelSpectrogram  
(Log)", ylabel='mel freq')
```



Audio transformations

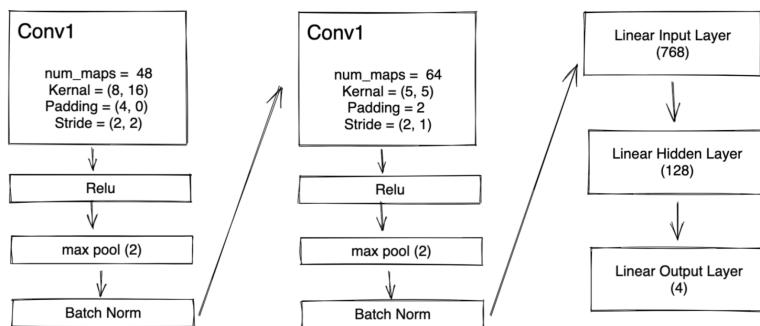
- Used [MelSpectrogram](#) from Pytorch audio to generate mel spectrogram
- Hyperparameters

Sample rate = 16000 (16kHz)
 Max window length = 750 ms (12000)
 Number of mel bins = 40
 Hop length = 200
 Mel Spectrogram matrix size = 40 x 61

- Used Zero Mean Unit Variance to scale the values
- Code [transformers.py](#) and [audio_collator.py](#)

Define model architecture

- Given above transformations, Mel spectrogram of size **40x61** will be fed to model
- Below is the CNN model used



- Code [model.py](#)
- Below is the CNN model summary

```
: summary(model, input_size=(1,40,61))
```

Layer (type)	Output Shape	Param #
<hr/>		
Conv2d-1	[-1, 48, 31, 13]	6,192
ReLU-2	[-1, 48, 31, 13]	0
MaxPool2d-3	[-1, 48, 15, 6]	0
MaxPool2d-4	[-1, 48, 15, 6]	0
BatchNorm2d-5	[-1, 48, 15, 6]	96
Conv2d-6	[-1, 64, 8, 6]	76,864
ReLU-7	[-1, 64, 8, 6]	0
MaxPool2d-8	[-1, 64, 4, 3]	0
MaxPool2d-9	[-1, 64, 4, 3]	0
BatchNorm2d-10	[-1, 64, 4, 3]	128
Linear-11	[-1, 128]	98,432
ReLU-12	[-1, 128]	0
Dropout-13	[-1, 128]	0
Linear-14	[-1, 4]	516
<hr/>		

Total params: 182,228

Trainable params: 182,228

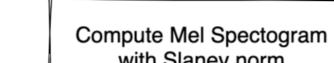
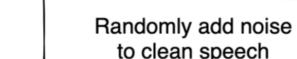
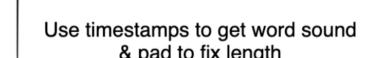
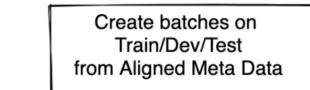
Non-trainable params: 0

Input size (MB): 0.01

Forward/backward pass size (MB): 0.46

Params size (MB): 0.70

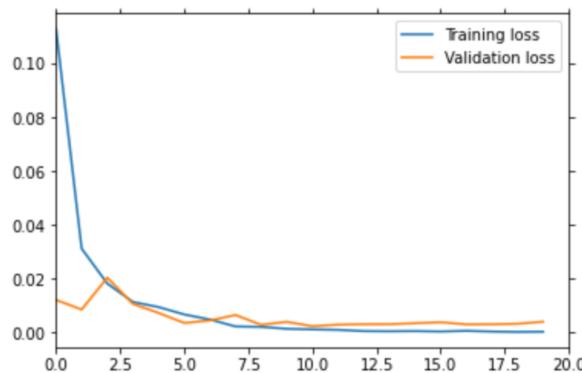
Estimated Total Size (MB): 1.17



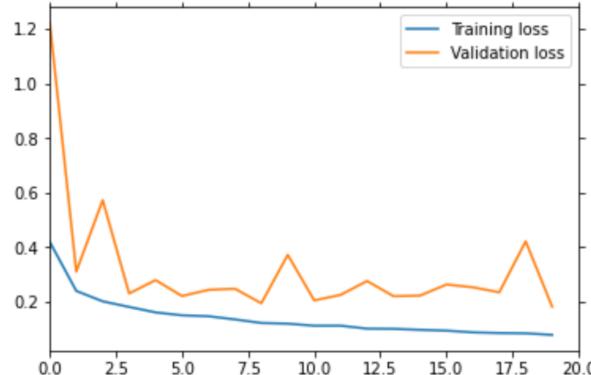
Train model

- Used batch size as 16, Tensor of size **[16, 1, 40, 61]** will be fed to Model

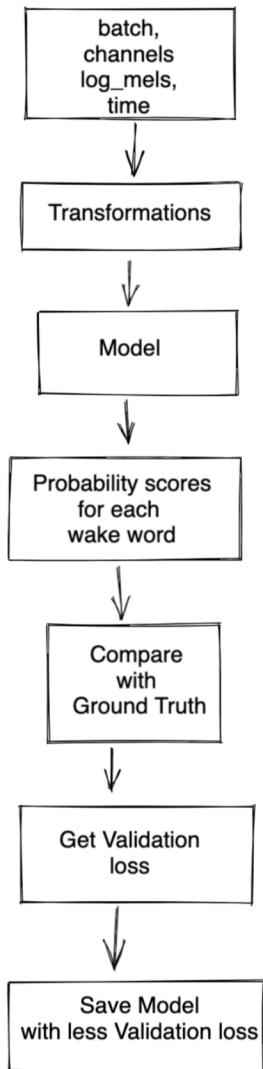
- Used 20 epochs, below is how the train vs validation loss looks like without noise



- As you can see, without noise, there is overfitting problem
- Its resolved after adding noise, below is how the train vs validation loss looks like



- Code - [train.py](#)



Test Model

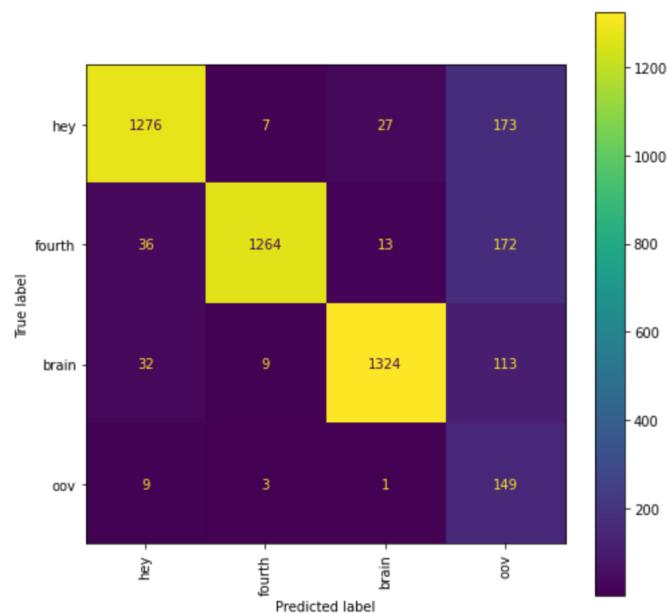
Below is how model performed on test dataset, model achieved 87% accuracy

	precision	recall	f1-score	support
brain	0.97	0.90	0.93	1478
fourth	0.99	0.85	0.91	1485
hey	0.94	0.86	0.90	1483
oov	0.25	0.92	0.39	162
accuracy			0.87	4608
macro avg	0.79	0.88	0.78	4608
weighted avg	0.94	0.87	0.90	4608

Test Loss: 0.606232

Test Accuracy of hey: 86% (1276/1483)
 Test Accuracy of fourth: 85% (1264/1485)
 Test Accuracy of brain: 89% (1324/1478)
 Test Accuracy of oov: 91% (149/162)

Test Accuracy (Overall): 87% (4013/4608)



Inference

Below are the methods used on live streaming audio on above model.

Using Pyaudio

- Used [Pyaudio](#), to get input from microphone
- Capture 750ms window of audio buffer
- After n batches, do transformations and infer on model
- Code - [infer.py](#)
 - * listening ..

```
C:\Users\rajas\Anaconda3\lib\site-pa
the builtin `float`. To silence this
ou specifically wanted the numpy sca
Deprecated in NumPy 1.20; for more d
['hey']
['hey', 'fourth']
['hey', 'fourth', 'brain']

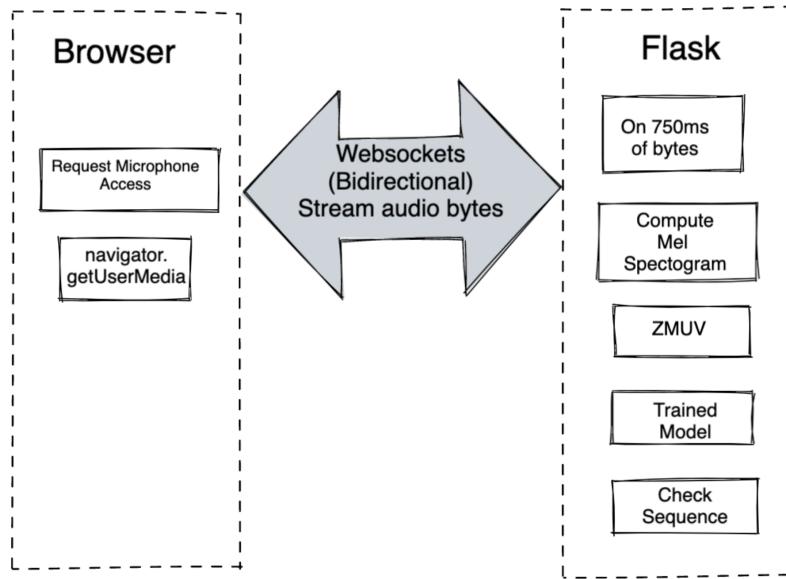
```

- Wake word hey fourth brain detected

Using web sockets

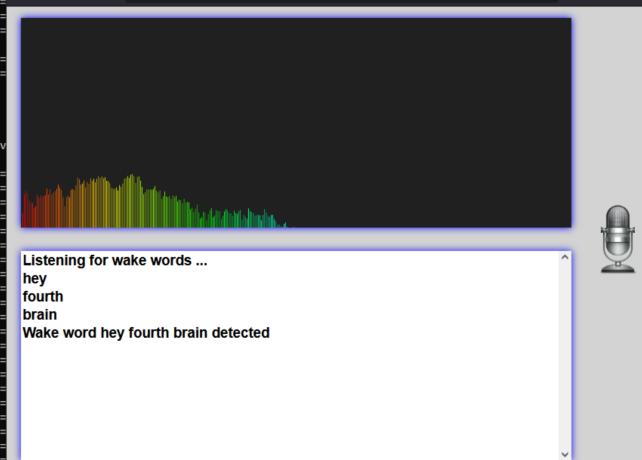
- Used [Flask Socketio](#) at server level to capture audio buffer from client.
- At Client, used [socket.io](#) at client level to send audio buffer through socket connection.
- Capture audio buffer using [getUserMedia](#), convert to array buffer and stream to server.
- Inference will happen at server, after n batches of 750ms window

- If sequence detected, send detected prompt to client.



- Server Code - `application.py`
 - Client Code - `main.js`
 - To run this locally

```
cd server  
python -m venv .venv  
pip install -r requirements.txt  
FLASK_ENV=development FLASK_APP=application.py .venv/bin/flask run --  
port 8011
```



- Use [Dockerfile](#) & [Dockerrun.aws.json](#) to containerize the app and deploy to [AWS Elastic BeanStalk](#)
 - Elastic Beanstalk initialize app

```
eb init -p docker-19.03.13-ce wakebot-app --region us-west-2
```

- Create Elastic Beanstalk instance

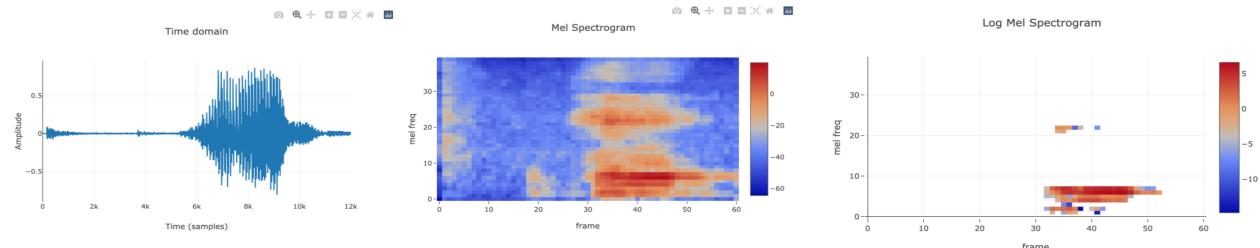
```
eb create wakebot-app --instance_type t2.large --max-instances 1
```

- Disadvantage of above method might be of privacy, since we are sending the audio buffer to server for inference

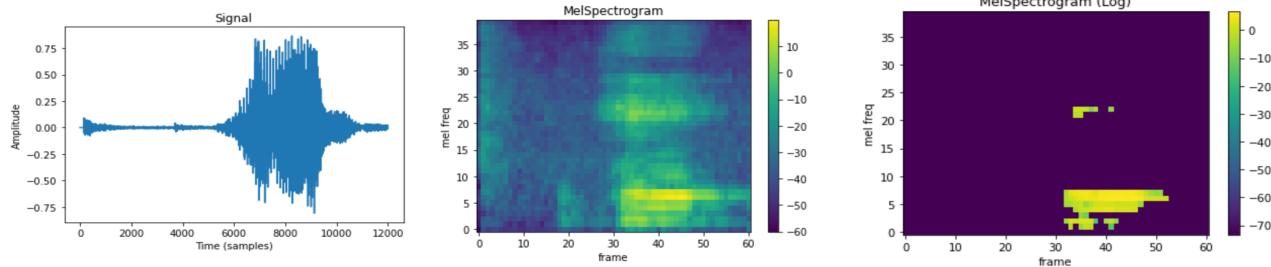
Using ONNX

- Used [Pytorch onnx](#) to convert pytorch model to onnx model
- Pytorch to onnx convert code - [convert_to_onnx.py](#)
- Once converted, onnx model can be used at client side to do inference
- Client side, used [onnx.js](#) to do inference at client level
- Capture audio buffer at client using [getUserMedia](#), convert to array buffer
- Used [fft.js](#) to compute [Fourier Transform](#)
- Used methods from [Meganta.js audio utils](#) to compute audio transformations like Mel spectrograms
- Below is the comparision of client side vs server side audio transformations

Client side plots (Using Plotly.js)



Server side plots (Using Matplotlib)



- Client side code - [main.js](#)
- To run locally

```
cd standalone
python -m venv .venv
pip install -r requirements.txt
FLASK_ENV=development FLASK_APP=application.py .venv/bin/flask run --
port 8011
```

(.venv) (base) C:\Users\rajas\Documents\workspace\TriggerWakeWordDetection\standalone>set FLASK_APP=app.py
(.venv) (base) C:\Users\rajas\Documents\workspace\TriggerWakeWordDetection\standalone>python app.py
* Serving Flask app 'app.py' (lazy loading)
* Environment: development
* Debug mode: on
* Restarting with stat
* Debugger is active!
* Debugger PIN: 142-552-417
* Running on http://127.0.0.1:5000/ (Press CTRL+C to quit)
127.0.0.1 - - [25/Aug/2021 17:59:59] "GET /socket.io/?EIO=4&transport=polling&t=Nj_-8Xa HTTP/1.1" 404 -
127.0.0.1 - - [25/Aug/2021 18:00:04] "GET /socket.io/?EIO=4&transport=polling&t=Nj_-9lr HTTP/1.1" 404 -
127.0.0.1 - - [25/Aug/2021 18:00:07] "GET / HTTP/1.1" 200 -
127.0.0.1 - - [25/Aug/2021 18:00:07] "GET /static/audio/audio_utils.js HTTP/1.1" 200 -
127.0.0.1 - - [25/Aug/2021 18:00:07] "GET /static/audio/main.js HTTP/1.1" 304 -
127.0.0.1 - - [25/Aug/2021 18:00:07] "GET /static/audio/styles.css HTTP/1.1" 200 -
127.0.0.1 - - [25/Aug/2021 18:00:07] "GET /static/audio/rfft.js HTTP/1.1" 200 -
127.0.0.1 - - [25/Aug/2021 18:00:07] "GET /static/audio/mic128.png HTTP/1.1" 200 -
127.0.0.1 - - [25/Aug/2021 18:00:14] "GET /static/audio/mic128.png HTTP/1.1" 200 -
127.0.0.1 - - [25/Aug/2021 18:00:14] "GET / HTTP/1.1" 200 -
127.0.0.1 - - [25/Aug/2021 18:00:14] "GET /static/audio/styles.css HTTP/1.1" 200 -
127.0.0.1 - - [25/Aug/2021 18:00:14] "GET /static/audio/rfft.js HTTP/1.1" 200 -
127.0.0.1 - - [25/Aug/2021 18:00:14] "GET /static/audio/main.js HTTP/1.1" 200 -
127.0.0.1 - - [25/Aug/2021 18:00:14] "GET /static/audio/audio_utils.js HTTP/1.1" 200 -
127.0.0.1 - - [25/Aug/2021 18:00:14] "GET /static/audio/mic128.png HTTP/1.1" 200 -
127.0.0.1 - - [25/Aug/2021 18:00:14] "GET /favicon.ico HTTP/1.1" 404 -
127.0.0.1 - - [25/Aug/2021 18:00:35] "GET /static/audio/prompt.mp3 HTTP/1.1" 200 -

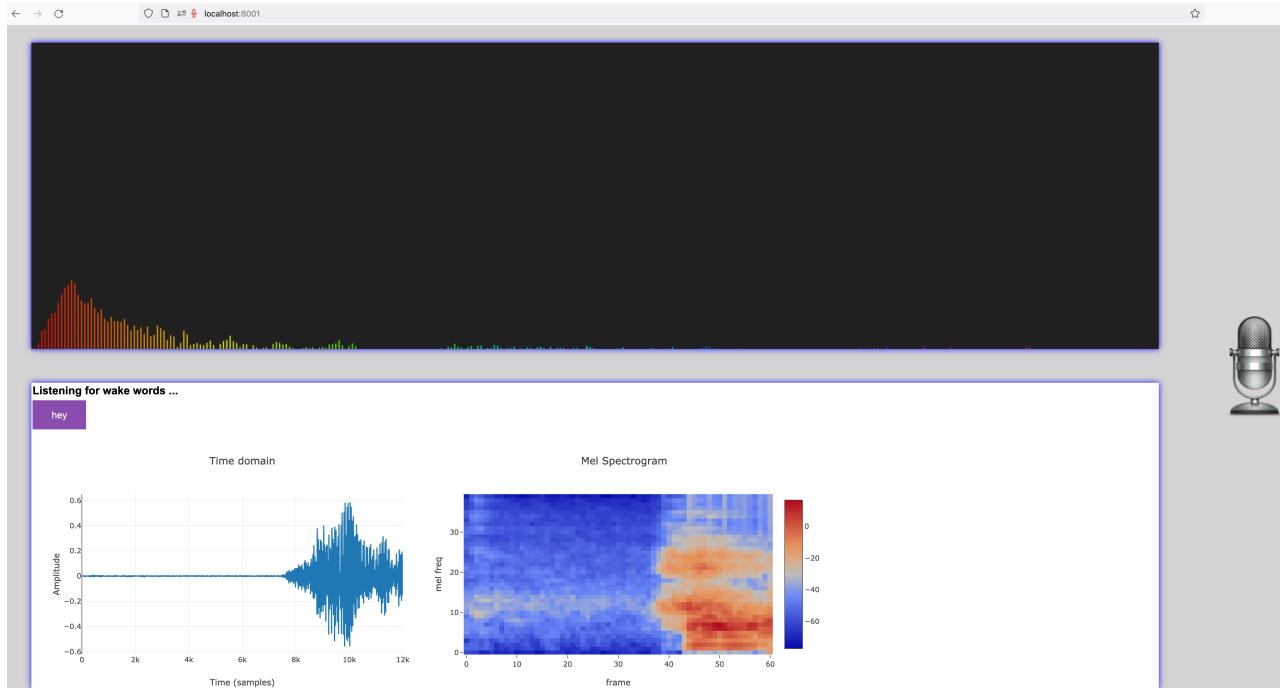
- To deploy to AWS Elastic Beanstalk, first initialize app

```
eb init -p python-3.7 wakebot-std-app --region us-west-2
```

- Create Elastic Beanstalk instance

```
eb create wakebot-std-app --instance_type t2.large --max-instances 1
```

- Refer [standalone_no_flask](#) for client version without flask, you can deploy on any static server, you can also deploy to [IPFS](#)
- Recent version will show, plots and audio buffer for each wake word which model inferred for, click on wake word button to know what buffer was inferred for that word.



Conclusion

In this project, we have went through how to extract audio features from audio and train model and detect wake words by using end to end example with source code. Go through [wake_word_detection.ipynb](#) jupyter notebook for complete walk through of this project.

Enhancements

- Explore different number of mels, in this project we used 40 as number of mels, we can use different number to see whether this will improve accuracy or not, this can be in range of 32 to 128.
- Use RNN or LSTM or GRU or attention to see whether we can get better results
- Check by computing MFCCs (which is computed after Mel spectrograms) and see if we see any improvements.
- Use different audio augmentation methods like [TimeStrech](#), [TimeMasking](#), [FrequencyMasking](#)