Malayalam Text-to Speech system

Guide: Jiby J P
Project Coordinator: Manilal D.L

Group 16
Kurian Benoy 34
kurianbenoy@mec.ac.in

Objective

- To build a Text to Speech(TTS) system in Malayalam
- Obtain the state of art result

Contents

Introduction

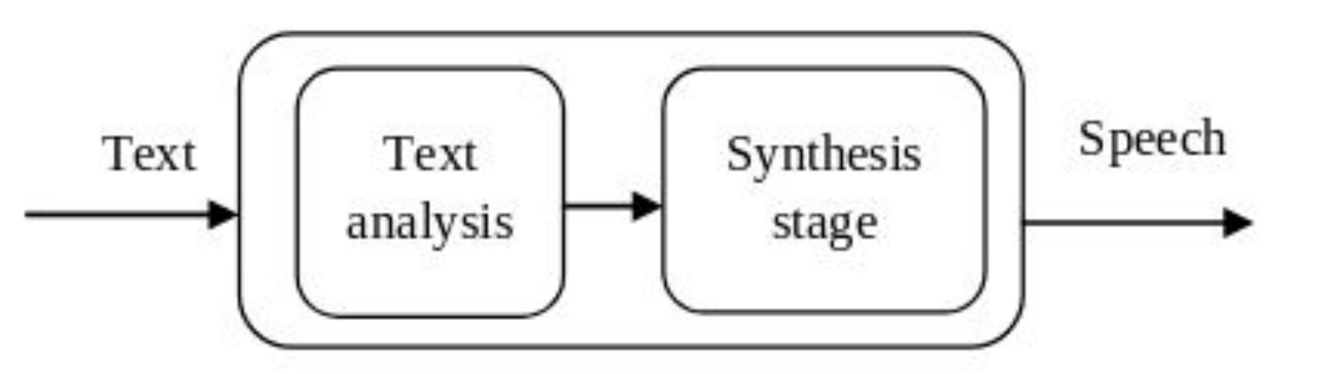
Text to speech

Text to speech systems convert any written text into spoken speech. Text-to-speech systems is a vital step for accessibility to disabled people like blind, and deaf. It can be used in lot of educational applications as well. Most of the text-to-speech systems are currently made for English.

Text to speech

TTS consists of two parts usually:

- The front-end consists of converting a text by text normalization, pre-processing, or tokenization and converting into graphemes.
- The back-end, referred to as the synthesizer, which converts the symbolic linguistic representation into sound.


Text
Text analysis
Synthesis stage
Speech

History

In 1779, the German-Danish scientist Christian Gottlieb Kratzenstein received the first prize in a competition declared by the Russian Imperial Academy of Sciences and Arts for the models he had designed of the human vocal tract that could generate the five long vowel sounds (International Phonetic Alphabet Notation: [a], [e], [I], [o] and [u]). The bellows-operated "acoustic-mechanical speech machine" by Wolfgang von Kempelen of Pressburg, Hungary, described in a 1791 article[2], followed by adding models of tongues and lips. This allowed it to produce consonants as well as voices. Charles Wheatstone created a "talking machine" based on von Kempelen's design in 1837. Wheatstone's model was a bit more complicated and was capable to produce vowels and most of the consonant sounds. Some sound combinations and even full words were also possible to produce. Vowels were produced with vibrating reed and all passages were closed. Resonances were effected by the leather resonator like in von Kempelen's machine. Consonants, including nasals, were produced with turbulent flow through a suitable passage with reed-off. Joseph Faber exhibited the "Euphonia" in 1846. Paget revived Wheatstone's concept in 1923.

History

In the 1930s, Bell Labs developed a vocoder that automatically analyzed speech in its fundamental tones and resonances.

Homer Dudley developed a keyboard-operated voice-synthesizer called The Voder (Voice Demonstrator), which he exhibited at the 1939 New York World Fair. Dr. Franklin S. Cooper and his colleagues at the Haskins Laboratories designed the Pattern Playback in the late 1940s and completed it in 1950. There have been several different versions of this hardware device; only one currently survives. It reconverted recorded spectrogram patterns into sounds, either in original or modified form. The spectrogram patterns were recorded optically on the transparent belt.

History

The first formant synthesizer, PAT (Parametric Artificial Talker), was introduced by Walter Lawrence in 1953 (Klatt 1987). PAT consisted of three electronic formant resonators connected in parallel. The input signal was either a buzz or noise. A moving glass slide was used to convert painted patterns into six time functions to control the three formant frequencies, voicing amplitude, fundamental frequency, and noise amplitude (track 03). At about the same time Gunnar Fant introduced the first cascade formant synthesizer OVE I (Orator Verbis Electris) which consisted of formant resonators connected in cascade (track 04). Ten years later, in 1962, Fant and Martony introduced an improved OVE II synthesizer, which consisted of separate parts to model the transfer function of the vocal tract for vowels, nasals, and obstruent consonants. Possible excitations were voicing, aspiration noise, and frication noise. The OVE projects were followed by OVE III and GLOVE at the Kungliga Tekniska HÃ¶gskolan (KTH), Swede. (as mentioned in [1])

Modules

- Module1 : EDA, dataset collection
- Module2: Train first TTS system in Malayalam
- Module3: Fine tune TTS system
- Module4: User Interface

Work Done

Dataset collection

1. Malayalam Speech Corpora, which was initiated to create high quality dataset under SMC. The recording platform can be found at https://msc.smc.org and dataset can be downloaded from: https://gitlab.com/smc/msc
2. Crowdsourced high-quality Malayalam multi-speaker speech data set by openslr.org
   Dataset can be found: http://openslr.org/63/ and is licensed under Attribution-ShareAlike 4.0 International

Dataset collection

3. The corpus contains 10 words in Malayalam corresponding to 10digits (0-9) in English. These words are uttered by 10 speakers include 6 females and 4 males of age ranging from 15 to 40. Every speaker gives 10 trials of each word and thus have 100 samples per speaker. Signals are recorded with a sampling frequency of 8 KHz. This dataset was Mini P.P etc. and licensed under CC.4.0

https://data.mendeley.com/datasets/5kg453tsjw

Text to Speech system in English

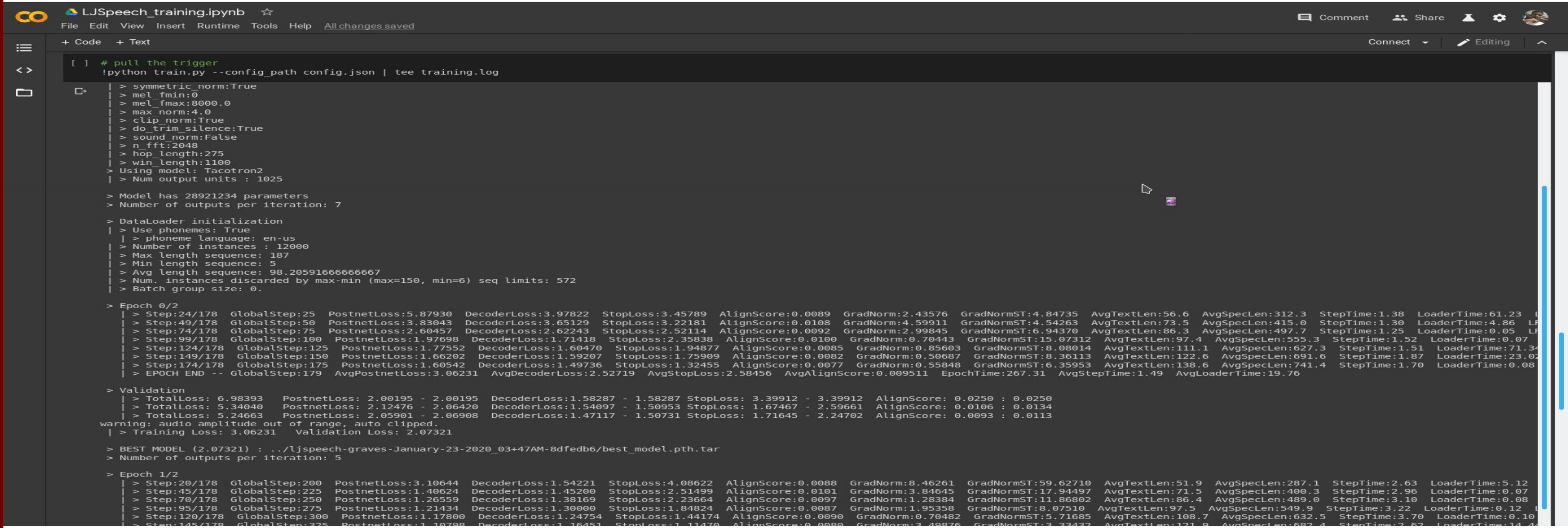
Using Tactron2 architecture made a TTS system in English using pretrained models from Mozilla/TTS. TTS used Tactron2 architecture made a TTS system in English using pretrained models from Mozilla/TTS. TTS aims a deep learning based Text2Speech engine, low in cost and high in quality.

TTS includes two different model implementations which are based on Tacotron and Tacotron2. Tacotron is smaller, efficient and easier to train but Tacotron2 provides better results, especially when it is combined with a Neural vocoder. Therefore, choose depending on your project requirements.

Text to Speech system in English

Training Notebook -

https://colab.research.google.com/drive/1Raaiaqs-RkFako1HJ0tXSW-EnWKvX84x



Text to Speech system in English

Inference Notebook [(https://colab.research.google.com/drive/1pyS5yQAe3UIpCV7q1boTqgyBgHCHID)](https://colab.research.google.com/drive/1pyS5yQAe3UIpCV7q1boTqgyBgHCHID)

```
[ ] SENTENCE = 'Bill got in the habit of asking himself “Is that thought true?” And if he wasn’t absolutely certain it was, he just let it go.'
```

Synthetize

```
[ ] align, spec, stop_tokens, wav = tts(model, SENTENCE, CONFIG, use_cuda, ap, speaker_id=0, use_gl=False, figures=False)

180000/181500 -- batch_size: 15 -- gen_rate: 12.2 kHz -- x_realtime: 0.6   >  Run-time: 23.488433837890625
0:00:00 / 27:03:12

[ ] subin = '3, 2,1 and go.The biggest news in 2019 has to be Janayugoms move to Scribus this is a landmark event since this is the first time an entirely free software based stack is used to produce a newspap

[ ] %%time
align, spec, stop_tokens, wav = tts(model, subin, CONFIG, use_cuda, ap, speaker_id=1, use_gl=False, figures=False)

   | > Decoder stopped with 'max_decoder_steps
576000/580800 -- batch_size: 48 -- gen_rate: 40.0 kHz -- x_realtime: 1.8   >  Run-time: 37.85575842857361
0:00:04 / 27:03:12
CPU times: user 37.2 s, sys: 683 ms, total: 37.9 s
Wall time: 38 s
```

Exploratory Data Analysis

SampleRate, Signal length, duration

In [12]:

```
y, sr = librosa.load("data/slr/male/mlm_00269_00156195788.wav")
tempo, beat_frames = librosa.beat.beat_track(y=y, sr=sr)

print('Estimated tempo: {:.2f} beats per minute'.format(tempo))

beat_times = librosa.frames_to_time(beat_frames, sr=sr)
print(beat_times)
```

```
Estimated tempo: 161.50 beats per minute
[0.23219955 0.60371882 0.9752381  1.34675737 1.69505669 2.04335601
 2.41487528 2.7631746  3.13469388 3.50621315 3.87773243 4.22603175]
```

In [13]:

```
print(f"Sample rate  :", sr)
print(f"Signal Length:{len(y)}")
print(f"Duration     : {len(y)/sr}seconds")
```

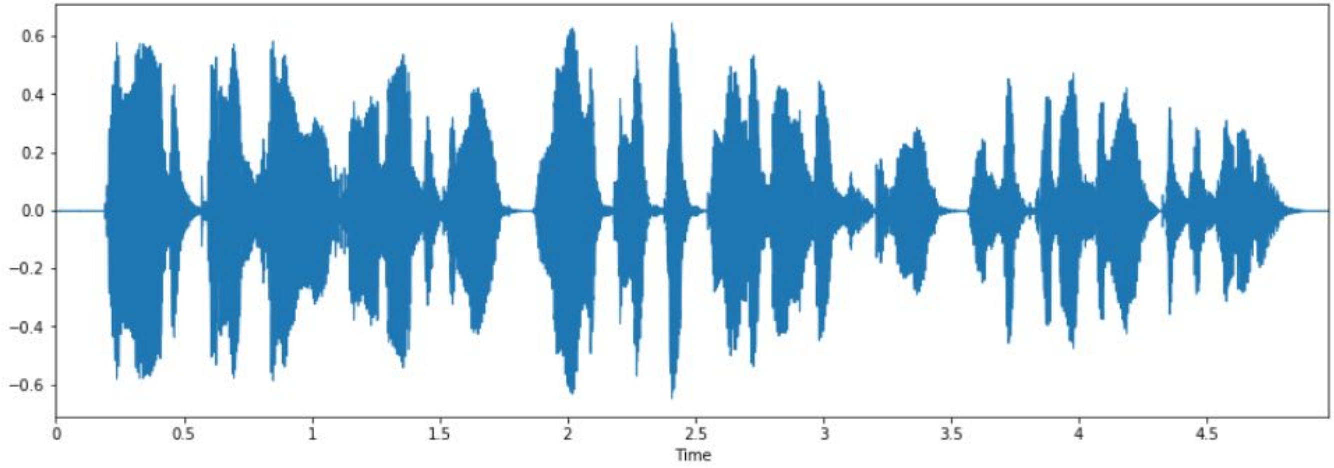
```
Sample rate  : 22050
Signal Length:109778
Duration     : 4.97859410430839seconds
```

Exploratory Data Analysis

Waveform display

```
In [15]: import librosa.display
         plt.figure(figsize=(15, 5))
         librosa.display.waveplot(y, sr=sr)

Out[15]: <matplotlib.collections.PolyCollection at 0x7fcc5d60dbe0>
```



Exploratory Data Analysis

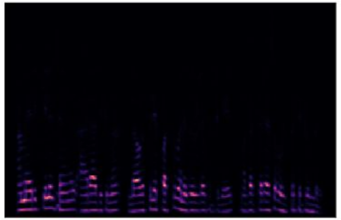
Mel Spectrogram

```
In [16]: sg0 = librosa.stft(y)
         sg_mag, sg_phase = librosa.magphase(sg0)
         display(librosa.display.specshow(sg_mag))
```

```
<matplotlib.axes._subplots.AxesSubplot at 0x7fcc5c561860>
```

```
In [17]: sg1 = librosa.feature.melspectrogram(S=sg_mag, sr=sr)
         display(librosa.display.specshow(sg1))
```

```
<matplotlib.axes._subplots.AxesSubplot at 0x7fcc5c0f1ac8>
```

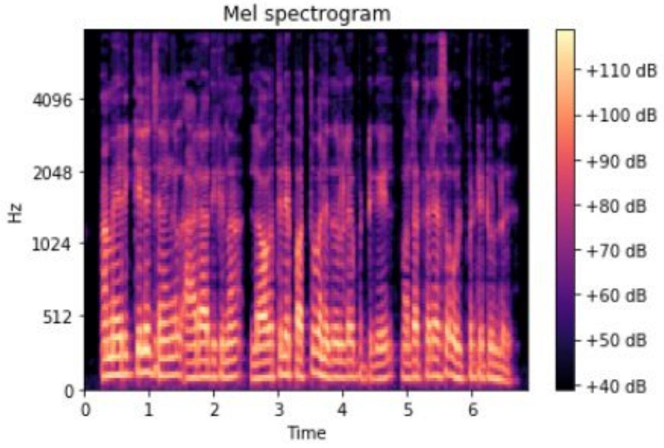


Exploratory Data Analysis

Mel Spectrogram vs Amplitude Plot

```
In [20]: # code adapted from the librosa.feature.melspectrogram documentation
         librosa.display.specshow(sg2, sr=16000, y_axis='mel', fmax=8000, x_axis='time')
         plt.colorbar(format='%+2.0f dB')
         plt.title('Mel spectrogram')

Out[20]: Text(0.5, 1.0, 'Mel spectrogram')
```

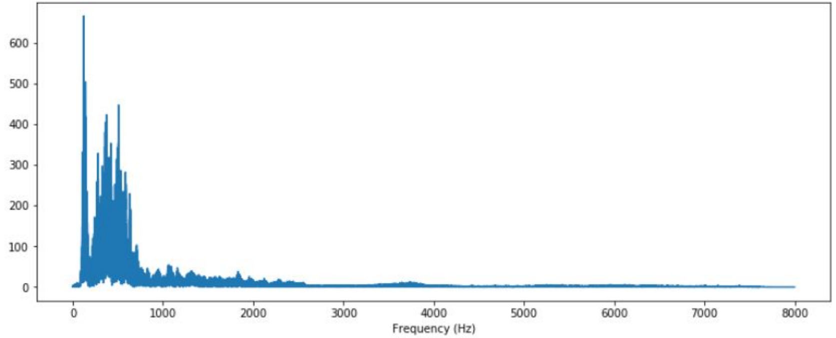


Exploratory Data Analysis

Fourier series transform

In [28]:
```python
# Code adapted from https://musicinformationretrieval.com/fourier_transform.html and the original
# implementation of fastai audio by John Hartquist at https://github.com/sevenfx/fastai_audio/
def fft_and_display(signal, sr):
    ft = scipy.fftpack.fft(signal, n=len(signal))
    ft = ft[:len(signal)//2+1]
    ft_mag = np.absolute(ft)
    f = np.linspace(0, sr/2, len(ft_mag)) # frequency variable
    plt.figure(figsize=(13, 5))
    plt.plot(f, ft_mag) # magnitude spectrum
    plt.xlabel('Frequency (Hz)')
```

In [29]:
```python
y, sr = librosa.load("data/slr/male/mlm_00269_00156195788.wav", sr=16000)
fft_and_display(y, sr)
```



Research paper

A study on Text to speech systems for Non-English languages

Thank you!