Deep Learning for Voice Faker

PROJECT PRESENTATION

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01

INTRODUCTION

An introduction of our project and why we think it is interesting.

PROJECT MOTIVATION and GOAL

Motivation 1

Deepface is popular
we can combine them together

Motivation 2

Help people with visual
or reading impairments

Goal 1

construct simple
identification model

Goal 2

help the lazy guys and the
speaking handicapped
to have a better life

02

ABOUT THE PROJECT

Project Result and the Colab Implementation

PROJECT STAGES

	Classification	Speech-To-Text
Dataset	Urbansound8k	Libasound train-100
Model	Audio to MelSpectrom & eliminate noise	Greedy decoding & CTCloss + deep speech 3
Accuracy	70%	70%
Benchmark	70~79%	80~84%

Colab Result

AI22.ipynb

檔案 編輯 檢視畫面 插入 執行階段 工具 說明 系統不會儲存變更

+ 程式碼 + 文字 複製到雲端硬碟

RAM 磁碟 編輯 共用

[12]

```python
            if (new_channel == 1):
                # Convert from stereo to mono by selecting only the first channel
                resig = sig[:1, :]
            else:
                # Convert from mono to stereo by duplicating the first channel
                resig = torch.cat([sig, sig])

            return ((resig, sr))
def spectro_gram(aud, n_mels=64, n_fft=1024, hop_len=None,top=80):
            sig, sr = aud
            top_db = top

        # spec has shape [channel, n_mels, time], where channel is mono, stereo etc
            spec = transforms.MelSpectrogram(
            sr, n_fft=n_fft, hop_length=hop_len, n_mels=n_mels)(sig)

        # Convert to decibels
            spec = transforms.AmplitudeToDB(top_db=top_db)(spec)
            return (spec)
for response in os.walk(file_path):
    for x in response[2]:
        if x[-3:]=='txt':
            continue
        s,rate=rechannel(torchaudio.load(file_path+'/'+x),1)
        spectrograms = []
        w = valid_audio_transforms(s).squeeze(0).transpose(0, 1)
        spectrograms.append(w)
        s = nn.utils.rnn.pad_sequence(spectrograms, batch_first=True).unsqueeze(1).transpose(2, 3)
        device = torch.device("cuda:0" if torch.cuda.is_available() else "cpu")
        s=torch.squeeze(s,2)
        input=s.to(device)
        output = model(input)
        output = F.log_softmax(output, dim=2)
        output = output.transpose(0, 1) # (time, batch, n_class)
        # Get the predicted class with the highest score
        decoded_preds, decoded_targets = GreedyDecoder(output.transpose(0, 1), [], 1,out=True)
        print(file_path+'/'+x,decoded_preds)
```

```
test/0.flac ['it was nowmided joly and the plag which had cheefly raged at the other end o the town and as i said before in the parisheus of sant chiles saint and drus  hol firn']
```

Test result

Load pretrained model

[8]

```python
import gdown
path="https://drive.google.com/u/1/uc?id=1043wWFXJm5XzFL4wa7wf8nmD2qKZScfw&export=download"
output ='/content/sounds.pth'
gdown.download(path, output,quiet=True)
model.load_state_dict(torch.load(output,map_location=torch.device('cpu')))
```

```
<All keys matched successfully>
```

[9]

```python
path="https://drive.google.com/u/1/uc?id=1dBGrwgOKAQOFi6VVtpAnzLat7CHbB3nb&export=download"
output ='test.7z'
gdown.download(path, output,quiet=True)
```

```
'test.7z'
```

[10]

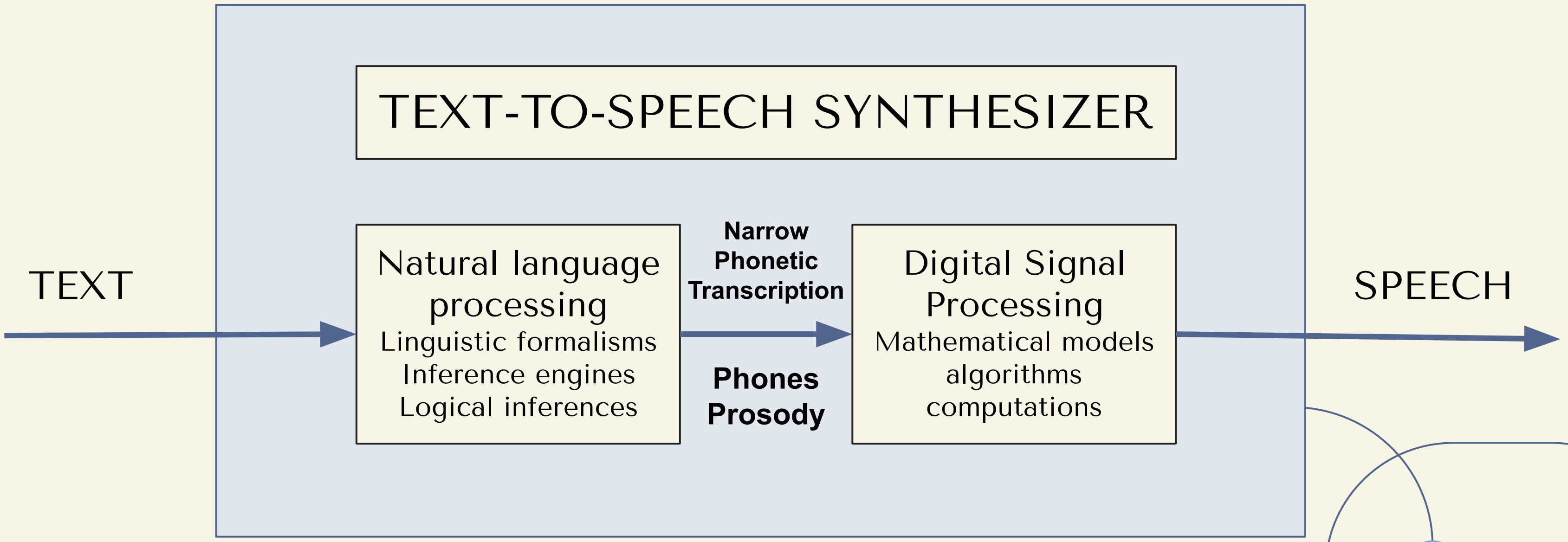
```
!p7zip -d 'test.7z'
```

03

Text-To-Speech

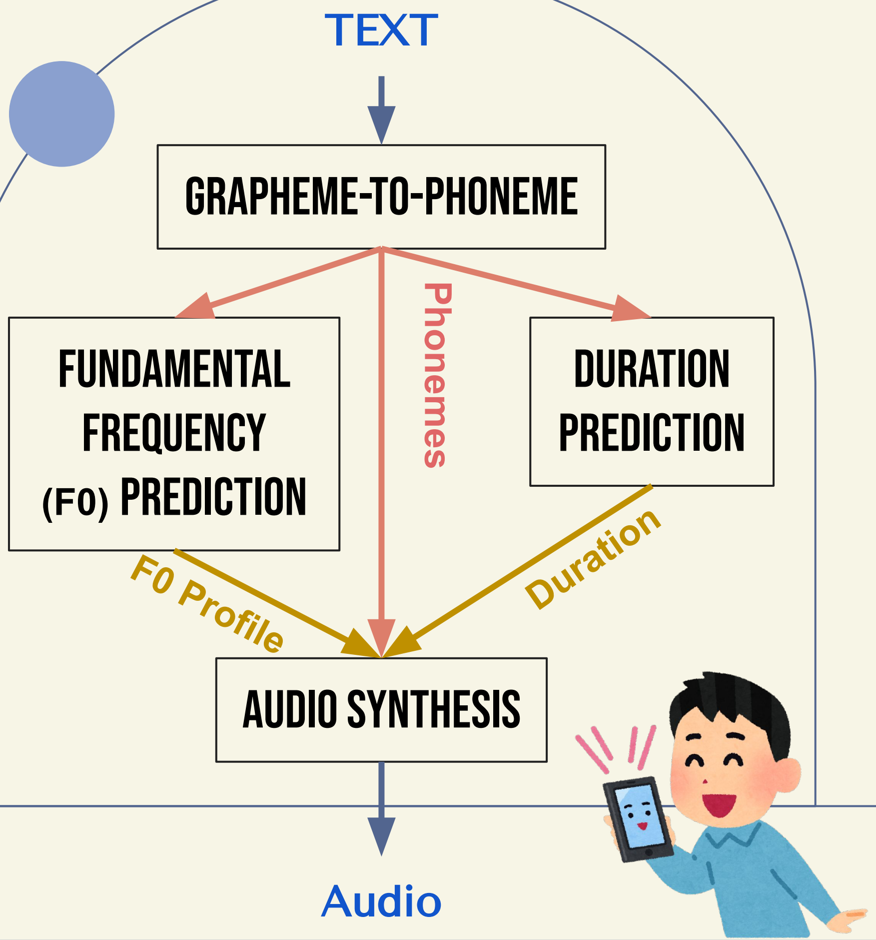
Text-to-speech (TTS) lets your devices read text to you. Text-to-speech is used not only to **help people with visual or reading impairments**, but also **convenient for those who are lazy to talk** .

Speech Synthesis (TTS) Technology

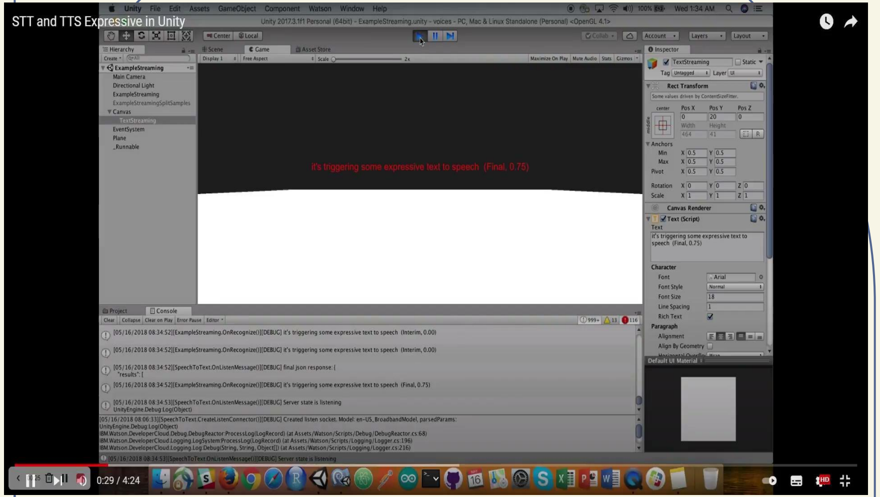


How to implement ?

1. Deep voice 3 open source code and well-trained model for single and multiple speaker TTS https://github.com/r9y9/deepvoice3_pytorch can be run on colab
2. Deep Voice: Real-time Neural Text-to-Speech https://proceedings.mlr.press/v70/arik17a/arik17a.pdf



From STT to TTS



STT and TTS Demo in IBM Watson SDK for Unity | Github Source :
https://github.com/rustyoldrake/ibm_watson_unity/blob/master/ExampleStreaming_plus_Text_to_Speech_expressive.cs

After STT completes the process, there are at least **four further processes** to be carried out.

1. **ASR (speech recognition)**
2. **NLU (natural language understanding)**
3. **DST (dialogue state control)**
4. **NLG (dialogue generation)**

and eventually arrive at **TTS (Text To Speech)**

04

ABOUT

Future Work

Outlook of Other techniques

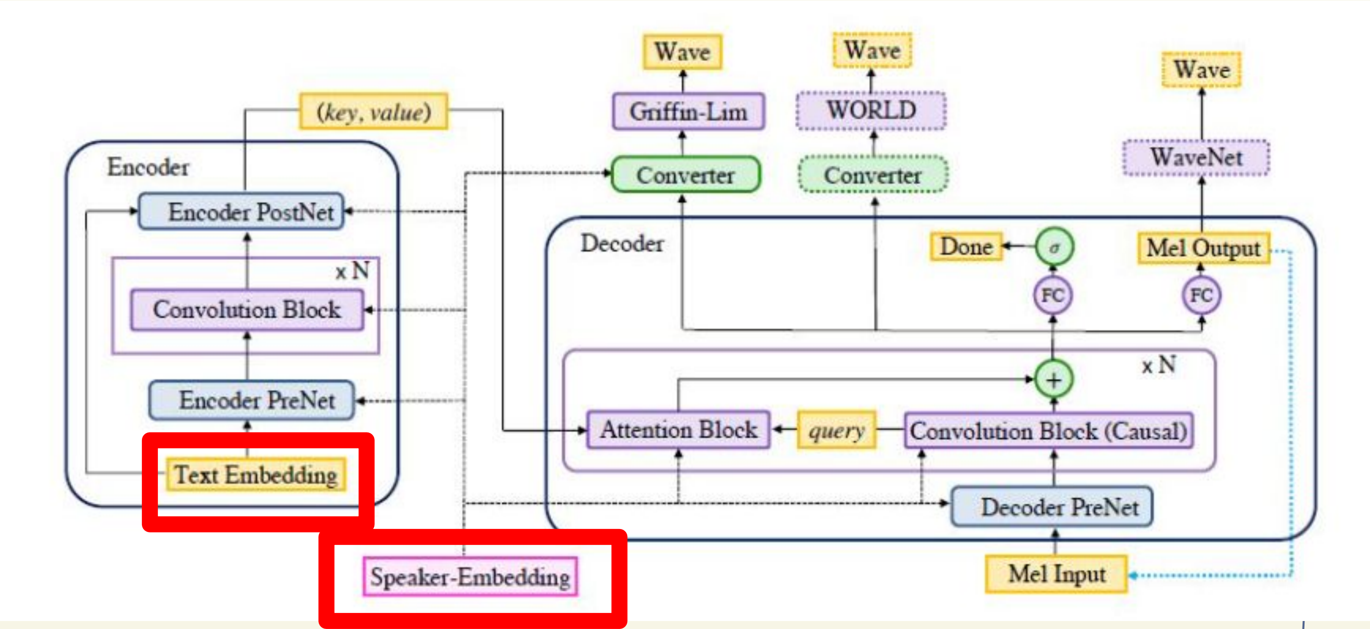
Voice cloning technology

Research on a practical technique for training TTS (Text to Speech) model with **few samples**

How to do?

pretrained + fine-tone

Multi-speaker model—Deep voice 3



cloning

Speaker adaptation

fine-tune a trained multi-speaker model for an unseen speaker using a few audio-text pairs

Speaker encoding

Train a separate network to generate new speaker embeddings, which are then used for multi-speaker generation models

conclusion

For the TTS task, the generation network of a single speaker requires nearly 20 hours of training data, but when voice cloning an unseen new person speaks, it only takes a few minutes or a few seconds.

Reference

- Neural Voice Cloning with a Few Samples / Jitong Chen,Kainan Peng,and Wei Ping
  In NIPS 2018 https://arxiv.org/pdf/1802.06006.pdf
- Deep4SNet: deep learning for fake speech classification / Dora M.,Yohanna,Diego,Gonzalo
  https://www.sciencedirect.com/science/article/pii/S0957417421008770
- Audio Deep Learning Made Simple: Sound Classification, Step-by-Step | by Ketan Doshi
- Audio Signal Processing and Recognition
- AFE characteristics proposed by the European Telecommunications Standards Institute
  https://www.etsi.org/deliver/etsi_es/201100_201199/201108/01.01.03_60/es_201108v010103p.pdf
- GitHub - mozilla/DeepSpeech
- The state-of-art benchmark accuracy of our project's datasets
  - Urbansound8k (Classification) / 79%
  - UrbanSound8K Benchmark (Environmental Sound)Classification | Papers With Code
  - Libasound train-100 (Audio-to-text) / 84%
    Global speech-to-text transcript error rating 2020 | Statista

THANKS

This is the end of our presentation

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