LAB-8

BL.EN.U4AIE21077

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A1. Use LSTM, Bi-LSTM networks for speech recognition. Use STFT / STCT, MFCC & LPC coefficients.

In [1]:
```
import numpy as np
import librosa
import tensorflow as tf
from tensorflow.keras.models import Sequential
from tensorflow.keras.layers import LSTM, Bidirectional, Dense, Dropout
import matplotlib.pyplot as plt
import librosa.display
```

```
WARNING:tensorflow:From C:\Users\virin\AppData\Roaming\Python\Python39\site-packages
\keras\src\losses.py:2976: The name tf.losses.sparse_softmax_cross_entropy is depreca
ted. Please use tf.compat.v1.losses.sparse_softmax_cross_entropy instead.
```

In [2]:
```
def extract_features(file_path):
    audio, sr = librosa.load(file_path, sr=16000)
    stft = np.abs(librosa.stft(audio, n_fft=400, hop_length=160))
    mfccs = librosa.feature.mfcc(y=audio, sr=sr, n_mfcc=40)
    lpc = librosa.lpc(y=audio, order=12)
    return audio, sr, stft, mfccs, lpc
```

In [3]:
```
def process_data(file_paths):
    audios, srs, stfts, mfccs, lpcs = [], [], [], [], []
    for file_path in file_paths:
        audio, sr, stft, mfcc, lpc = extract_features(file_path)
        audios.append(audio)
        srs.append(sr)
        stfts.append(stft)
        mfccs.append(mfcc)
        lpcs.append(lpc)
    return np.array(audios), np.array(srs), np.array(stfts), np.array(mfccs), np.array
```

In [4]:
```
def build_lstm_model(input_shape):
    model = Sequential([
        LSTM(128, input_shape=input_shape, return_sequences=True),
        Dropout(0.3),
        LSTM(128, return_sequences=True),
        Dropout(0.3),
        Dense(64, activation='relu'),
        Dense(10, activation='softmax')
    ])
    return model
```

In [5]:
```
def build_bilstm_model(input_shape):
    model = Sequential([
        Bidirectional(LSTM(128, input_shape=input_shape, return_sequences=True)),
```

```
        Dropout(0.3),
        Bidirectional(LSTM(128, return_sequences=True)),
        Dropout(0.3),
        Dense(64, activation='relu'),
        Dense(10, activation='softmax')
    ])
    return model
```

In [6]:
```
audio_file_path = "Hari1.wav"
audio, sr, stft_data, mfcc_data, lpc_data = extract_features(audio_file_path)
stft_data = stft_data.reshape(1, stft_data.shape[0], stft_data.shape[1])
mfcc_data = mfcc_data.reshape(1, mfcc_data.shape[0], mfcc_data.shape[1])
lpc_data = lpc_data.reshape(1, lpc_data.shape[0])
```

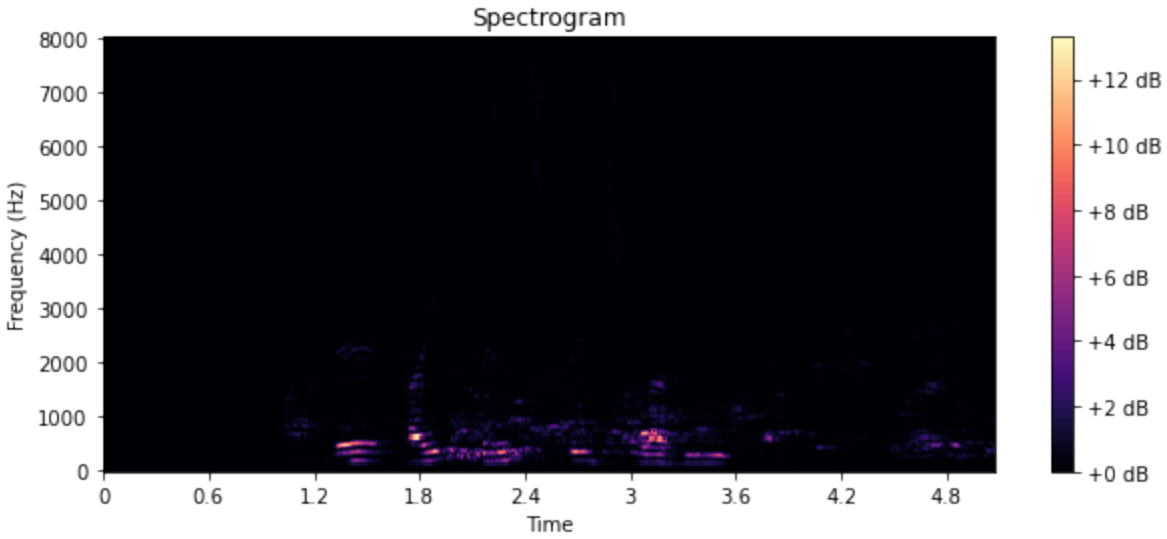
In [7]:
```
lstm_model = build_lstm_model(stft_data.shape[1:])
bilstm_model = build_bilstm_model(stft_data.shape[1:])
lstm_model.compile(optimizer='adam', loss='sparse_categorical_crossentropy', metrics=[
bilstm_model.compile(optimizer='adam', loss='sparse_categorical_crossentropy', metrics
lstm_predictions = lstm_model.predict(stft_data)
bilstm_predictions = bilstm_model.predict(stft_data)
```

```
WARNING:tensorflow:From C:\Users\virin\AppData\Roaming\Python\Python39\site-packages
\keras\src\layers\rnn\lstm.py:148: The name tf.executing_eagerly_outside_functions is
deprecated. Please use tf.compat.v1.executing_eagerly_outside_functions instead.

WARNING:tensorflow:From C:\Users\virin\AppData\Roaming\Python\Python39\site-packages
\keras\src\optimizers\__init__.py:309: The name tf.train.Optimizer is deprecated. Ple
ase use tf.compat.v1.train.Optimizer instead.

1/1 [==============================] - 1s 700ms/step
1/1 [==============================] - 3s 3s/step
```

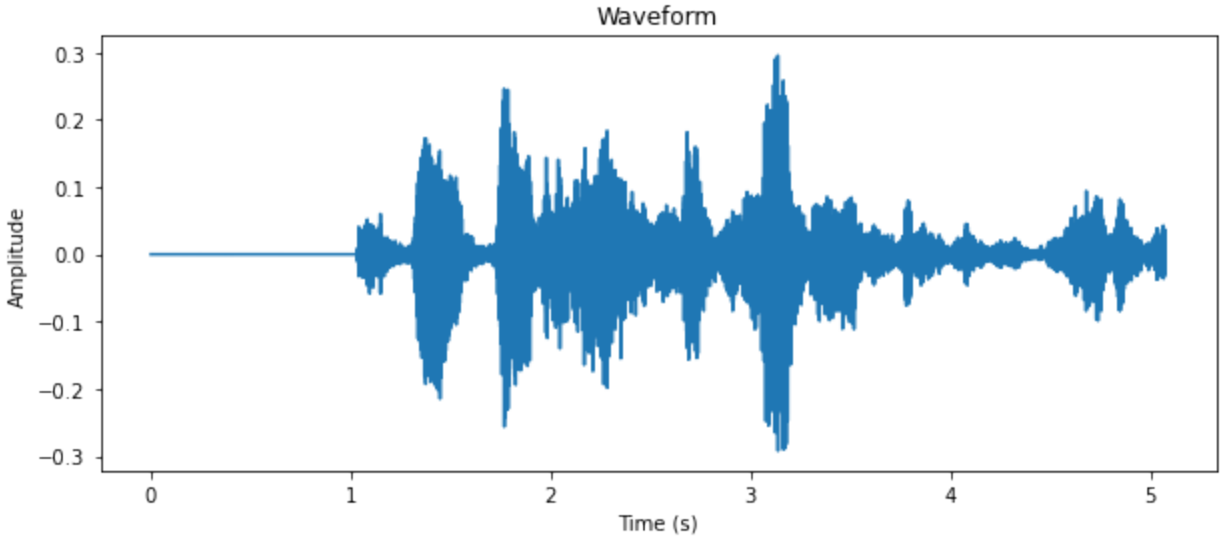
In [8]:
```
print("Sampling rate (sr):", sr)
print("Number of FFT points (n_fft):", stft_data.shape[1])
print("Shape of stft_data:", stft_data.shape)
```

```
Sampling rate (sr): 16000
Number of FFT points (n_fft): 201
Shape of stft_data: (1, 201, 508)
```

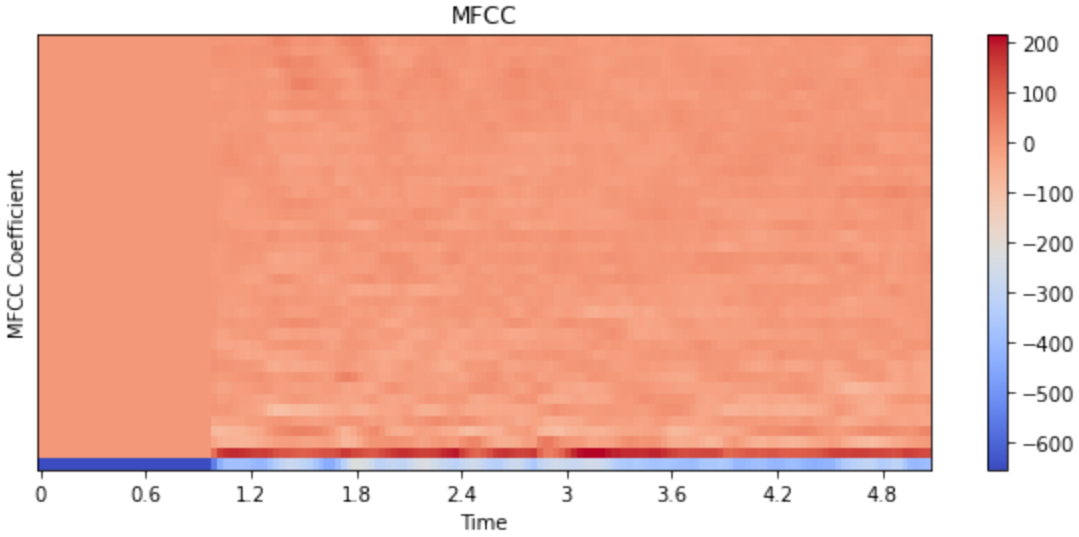
In [9]:
```
# Spectrogram
plt.figure(figsize=(10, 4))
librosa.display.specshow(stft_data[0], sr=sr, hop_length=160, x_axis='time', y_axis='l
plt.colorbar(format='%+2.0f dB')
plt.title('Spectrogram')
plt.xlabel('Time')
plt.ylabel('Frequency (Hz)')
plt.show()
```



In [10]:
```
# Waveform
plt.figure(figsize=(10, 4))
plt.plot(np.arange(len(audio)) / float(sr), audio)
plt.title('Waveform')
plt.xlabel('Time (s)')
plt.ylabel('Amplitude')
plt.show()
```



In [11]:
```
# MFCC
plt.figure(figsize=(10, 4))
librosa.display.specshow(mfcc_data[0], sr=sr, x_axis='time')
plt.colorbar()
plt.title('MFCC')
plt.xlabel('Time')
plt.ylabel('MFCC Coefficient')
plt.show()
```



A2. Speak the sentence “Bhanumathi weds Rajat” (भानुमति वेड्स रजि || ಭಾನುಮತಿ ವೆಡ್ಸ್ ರಜತ್). Construct speech for the word “Bharat” (ಭಾರತ್ || भारत || ભરત્ || பாரத்) by combining segmented phonemes taken from “Bhanumathi weds Rajat” speech recording. Listen to this word speech generated and note down the issues associated with this approach of speech synthesis. The string written Indian scripts may have some error. Please validate before using.

In [12]:
```
import os
import tempfile
import speech_recognition as sr
import pyttsx3
import matplotlib.pyplot as plt
import numpy as np
import librosa
```

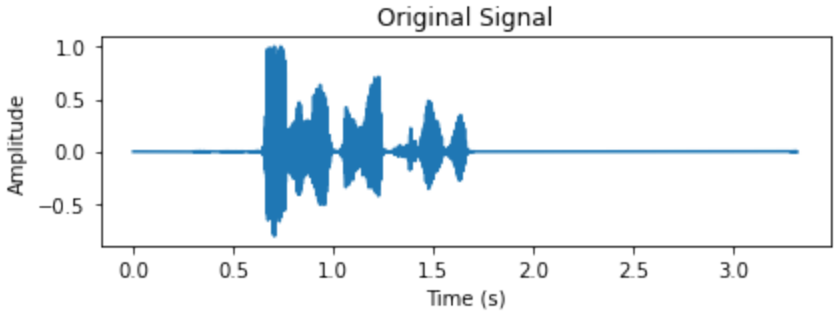
In [13]:
```
custom_temp_dir =  r"C:\Users\virin\Downloads\LAB-8"
def segment_phonemes(transcription, target_word):
    words = transcription.split()
    for word in words:
        if word.lower() == target_word.lower():
            return word
def synthesize_word(phonemes, output_file):
    engine = pyttsx3.init()
    engine.save_to_file(phonemes, output_file)
    engine.runAndWait()
```

In [14]:
```
audio_file = "Hari-77.wav"
recognizer = sr.Recognizer()
with sr.AudioFile(audio_file) as source:
    audio_data = recognizer.record(source)
transcription = recognizer.recognize_google(audio_data)
```

In [15]:
```
phonemes = segment_phonemes(transcription, "Bharat")
output_file = os.path.join(custom_temp_dir, "bharat_speech.wav")
synthesize_word(phonemes, output_file)
plt.figure(figsize=(10, 6))
```

Out[15]:
```
<Figure size 720x432 with 0 Axes>

<Figure size 720x432 with 0 Axes>
```

In [20]:
```
# Original signal
plt.subplot(2, 1, 1)
plt.title('Original Signal')
audio_data_np, _ = librosa.load(audio_file, sr=source.SAMPLE_RATE)
plt.plot(np.linspace(0, len(audio_data_np) / source.SAMPLE_RATE, num=len(audio_data_np
plt.xlabel('Time (s)')
plt.ylabel('Amplitude')
plt.tight_layout()
plt.show()
```



In [21]:
```
# Reconstructed signal
reconstructed_audio_data, _ = librosa.load(output_file, sr=source.SAMPLE_RATE)
plt.subplot(2, 1, 2)
plt.title('Reconstructed Signal')
plt.plot(np.linspace(0, len(reconstructed_audio_data) / source.SAMPLE_RATE, num=len(re
plt.xlabel('Time (s)')
plt.ylabel('Amplitude')
plt.tight_layout()
plt.show()
```