Experiment 4

Title: Real time speech signal and Spectral analysis

The speech signal has frequency components in the audio frequency range 300 Hz to 3400 Hz of the electromagnetic spectrum. Record the male and female voice speech Signal. Write a program to record the speech signals and sketch it in time domain, its amplitude spectrum and phase spectrum.

Learning Objectives

i) To understand basic concept of sampling of speech signal
ii) To study frequency range of speech signal, audio signals
iii) To study magnitude and phase spectrum of speech signal
iv) To have hands on simulation using python language

Prerequisites

i) Basic understanding of mathematics
ii) Basic understanding of signal representation
iii) Basics of signal sampling
iv) Basic understanding of Python language

Theory

Sampling Theorem:

According to Nyquist principle of sampling, in order to remove alising effect signals are sampled with sampling frequency equals to twice of the frequency of signal to be sampled.

If F = frequency of signal to be sampled
Then, sampling frequency = Fs $\geq$ 2F

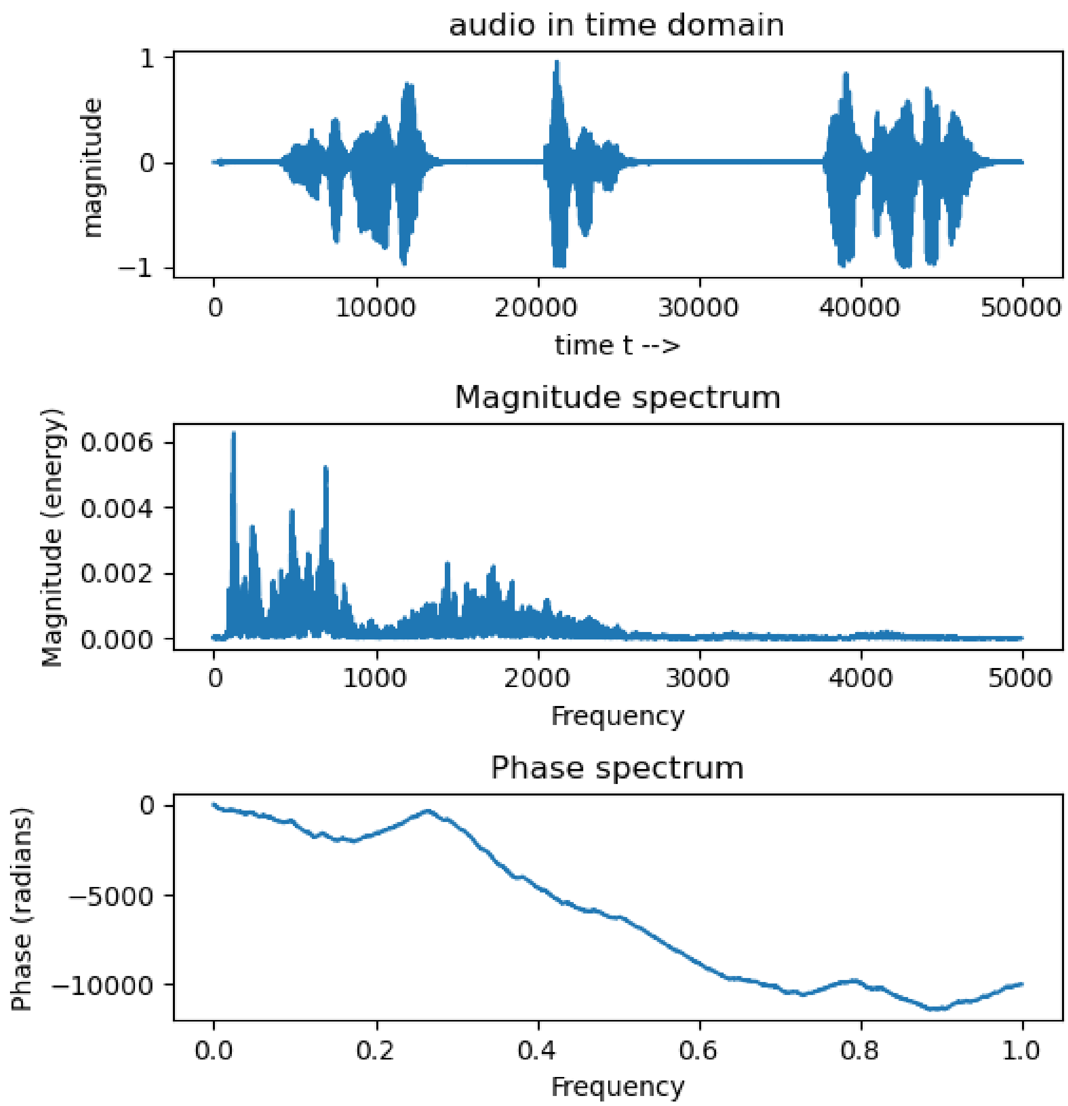
Sampling Speech signal:

The frequency of speech signal lies in the range of 300Hz to 3400Hz.
So the maximum frequency in the speech signal is 3400Hz, i.e. information is going to be present in the this frequency and we must sample this frequency.
So in generalized way we will assume that there is maximum frequency of 4000Hz is present in the speech signal, and in order to sample this frequency we will follow Nyquist principle so the sampling frequency will be double of 4000Hz i.e equals to 8000Hz.
But sampling frequency is greater that or equals to twice of frequency of signal to be sampled, so we will select greater frequency than 8000Hz which is 10000Hz.

For this experiment we will take sampling frequency equals to 10000Hz or 10kHz.
Hence there will be 10000 samples per seconds.

Simulation Code

```
import sounddevice as sd
from scipy.io.wavfile import read, write
import matplotlib.pyplot as plt

# Sampling frequency
freq = 10000
# Recording duration
duration = 5

# Start recorder with the given values of
# duration and sample frequency
recording = sd.rec(int(duration * freq), samplerate=freq, channels=2)

# Record audio for the given number of seconds
sd.wait()

# This will convert the NumPy array to an audio
# file with the given sampling frequency
write("recording0.wav", freq, recording)

Fs, data  = read("recording0.wav")

data = data[:, 0]
print("sampling frequency = ", Fs)

plt.subplot(3,1,1)
plt.plot(data)
plt.title('audio in time domain')
plt.xlabel('time t -->')
plt.ylabel('magnitude')

plt.subplot(3,1,2)
plt.magnitude_spectrum(data, Fs)
plt.title('Magnitude spectrum')

plt.subplot(3,1,3)
plt.phase_spectrum(data)
plt.title('Phase spectrum')

plt.show()
```



Assignment
Task1: Write description for code **Task2: Vary the parameters of the signals and observe the magnitude and phase spectrum**

Conclusion