



Signals and Systems Laboratory **(EC2P002)**

END SEMESTER EXAMINATION

**Speech Signal
Processing**

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AIM:

- To read the given speech signal in MATLAB.
- To extract the clean voice and store the final processed signal in .wav format.
- To suggest a systematic, practical engineering solution to obtain the desired output

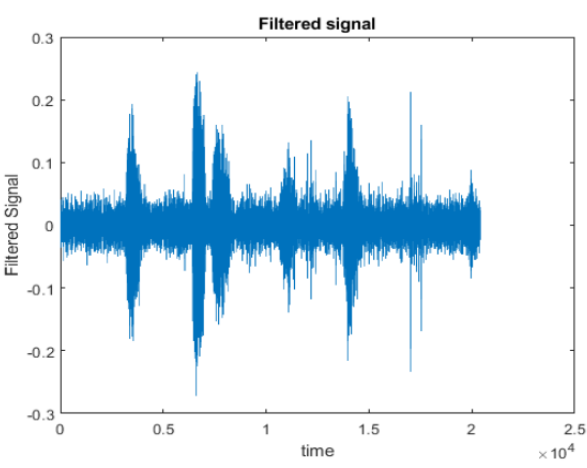
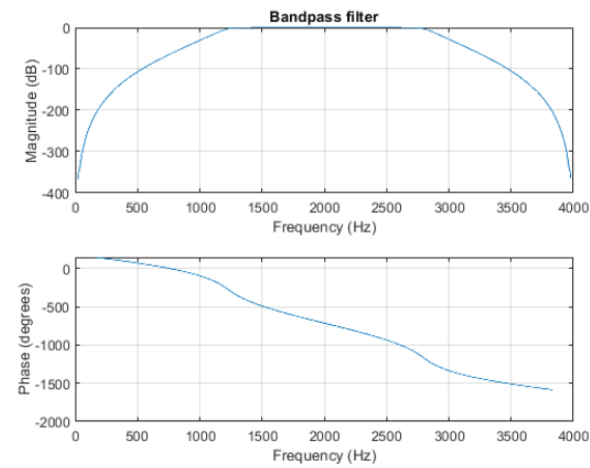
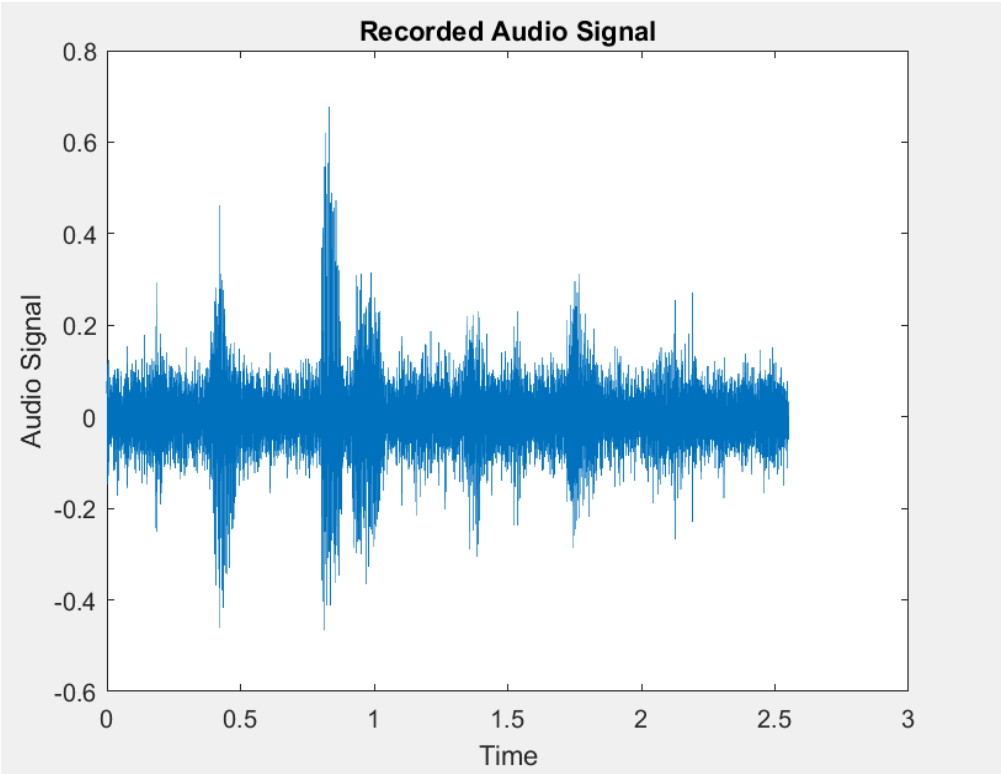
THEORY:

A band-pass filter or bandpass filter (BPF) is a device that passes frequencies within a certain range and rejects (attenuates) frequencies outside that range. An example of an analogue electronic band-pass filter is an RLC circuit (a resistor–inductor–capacitor circuit). These filters can also be created by combining a low-pass filter with a high-pass filter. The bandwidth of the filter is simply the difference between the upper and lower cutoff frequencies. The shape factor is the ratio of bandwidths measured using two different attenuation values to determine the cutoff frequency, e.g., a shape factor of 2:1 at 30/3 dB means the bandwidth measured between frequencies at 30 dB attenuation is twice that measured between frequencies at 3 dB attenuation. A band-pass filter can be characterized by its Q factor. The Q-factor is the reciprocal of the fractional bandwidth. A high-Q filter will have a narrow passband and a low-Q filter will have a wide passband. These are respectively referred to as narrow-band and wide-band filters.

Our aim is to reduce the noise of the given signal using the technique of band pass filters in MATLAB.

RESULTS:

1) Recorded Speech Signal



DISCUSSION

So we have removed some of the background voice and got the cleaner signal by converting the sound signal in the frequency domain using fast Fourier transform and then passing the transfer function in a second order lowpass digital Butterworth filter and then filtered it to smoothen out the high frequency fluctuations in data or remove periodic trends of a specific frequency from data. Then, we have plotted the sound and saved it in .wav format.

CONCLUSION

Sound is also a signaling system whether it be for notifying or warning us so it is very important that it is clean and noise is not interfering with it. So the extraction of a clean voice is very important and has many applications in our daily

APPENDIX

```
1  clc
2  close all
3  clear all
4  [y,fs] = audioread('C:\Users\Shory\OneDrive\Documents\MATLAB\test1.wav');
5  sound(y,fs);
6  totaltime = length(y)./fs;
7  t = 0:(totaltime/length(y)):(totaltime-totaltime/length(y));
8  plot(t,y);
9  title("Original Signal"); xlabel("time"); ylabel("Filtered Signal");
10 l = length(y);
11 NEFT = 2^nextpow2(l);
12 dft = abs(fft(y,NEFT));
13 freq = fs/2*linspace(0,1,NEFT/2+1);
14 plot(freq, dft(1:length(freq)));
15 title("DFT of the signal"); xlabel("f"); ylabel("DFT(y)");
16 grid on;
17 o=10;
18 wn = [350 800]*7/fs;
19 [b,a] = butter(o, wn, 'bandpass');
20 freqz(b,a, 1024, fs); title("Bandpass filter");
21 filt = filter(b,a,y);
22 plot(filt); title("Filtered signal"); xlabel("time"); ylabel("Filtered Signal");
23 sampledSignal = [tempname(), '.wav'];
24 audiowrite(sampledSignal, filt,fs);
25 audiowrite("19EE01017.wav",filt,fs);%storing the rectified audio to this new file
26 [y_new,fs1] = audioread(sampledSignal);
27 sound(y_new,fs1);
```