

DSP2 Week 10 experiment Report

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Exercise 1

a. (Source Code)

```
9      t = 0:0.01:1;
10     x = sin(4*pi*t);
11     noise = randn(1, length(t));
12     x1 = x + 0.15*noise;
13     h = [1/4 1/2 1/4];
```

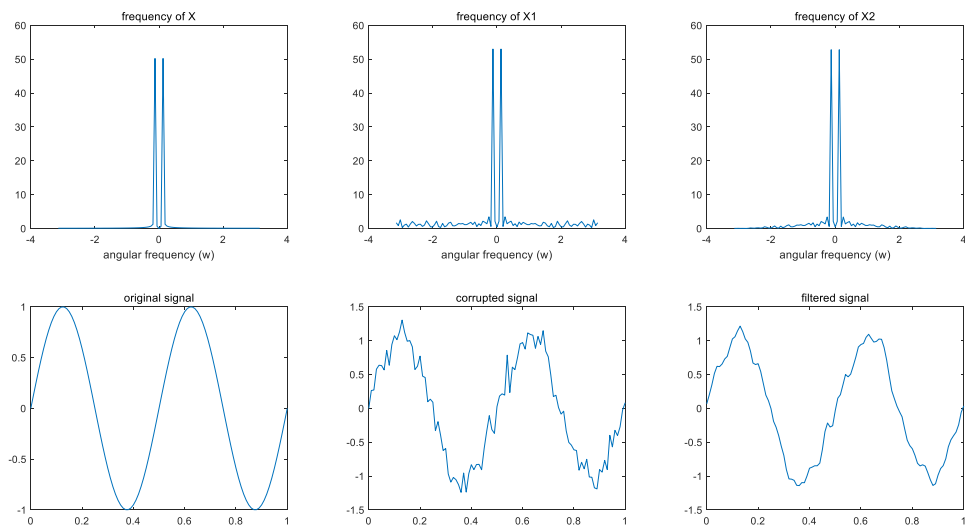
I generated signal x, x1 and a low-pass filter h.

b. (Source Code)

```
21     angular_freq = -pi:2*pi*0.01:pi;
22
23     X = fft(x);
24     X = fftshift(X);
25     magX = abs(X);
26
27     X1 = fft(x1);
28     X1 = fftshift(X1);
29     magX1 = abs(X1);
30
31     X2 = fft(x2);
32     X2 = fftshift(X2);
33     magX2 = abs(X2);
```

```
35     subplot(2,3,1);
36     plot(angular_freq, magX);
37     title('frequency of X')
38     xlabel('angular frequency (w)')
39     subplot(2,3,2);
40     plot(angular_freq, magX1)
41     title('frequency of X1')
42     xlabel('angular frequency (w)')
43     subplot(2,3,3);
44     plot(angular_freq, magX2)
45     title('frequency of X2')
46     xlabel('angular frequency (w)')
47
48     subplot(2,3,4);
49     plot(t, x);
50     title('original signal')
51     subplot(2,3,5);
52     plot(t, x1);
53     title('corrupted signal')
54     subplot(2,3,6);
55     plot(t, x2);
56     title('filtered signal')
```

(Result)



c. (Source Code)

```
48 Err1 = mean((x-x1).^2)
49 Err2 = mean((x-x2).^2)
50 difference = (Err1 - Err2)
```

(Result)

```
Err1 = 0.0214
Err2 = 0.0089
difference = 0.0125
```

- d. x is the original signal. x_1 is a new signal with some noise and x_2 is a filtered signal with low-pass filter. Upper three graphs are frequency of x , x_1 and x_2 . The frequency of x_1 has lots of noise on the overall frequency whereas the frequency of x_2 is more concentrated on the low frequency region. Lower three graphs are the graphs of x , x_1 and x_2 . The graph of x_1 looks very distorted whereas the graph of x_2 looks more similar to the original graph x .

Exercise 2

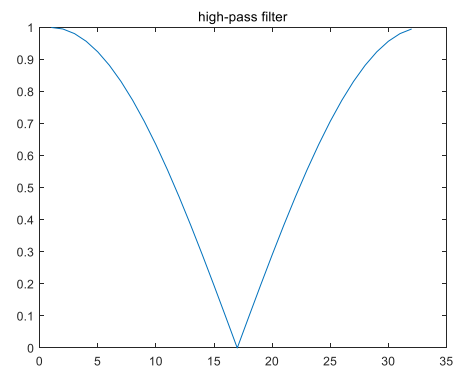
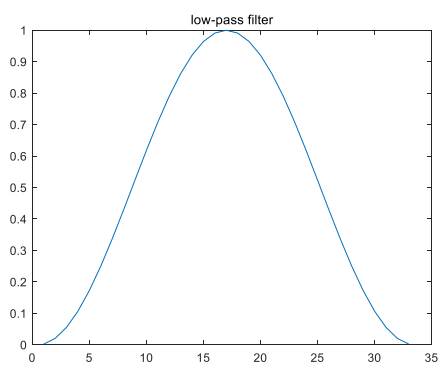
a. (Source Code)

```

1  [y, Fs] = audioread("dsp2_experiment10.wav")
2
3  lpf = [1/4 1/2 1/4];
4  pad = zeros(1, 30);
5  lpf = [lpf, pad];
6  LPF = fft(lpf);
7  LPF = fftshift(LPF);
8  magLPF = abs(LPF);
9  plot(magLPF)
10 title('low-pass filter')
11
12 hpf = [1/2 -1/2];
13 hpf = [hpf, pad];
14 HPF= fft(hpf);
15 HPF = fftshift(HPF);
16 magHPF= abs(HPF);
17 plot(magHPF)
18 title('high-pass filter')
19
20 y_lpf = conv(y, lpf);
21 y_hpf = conv(y, hpf);

```

(Result)



b. (Source Code)

```

22 sound(y, Fs)
23 %sound(y_lpf, Fs)
24 %sound(y_hpf, Fs)

```

(Result)

The original signal is a mixture of low-pitched voice and high-pitched noise. The low-pass filtered signal sounds less high-pitched noise. However, the high-pass filtered signal sounds very small voice.

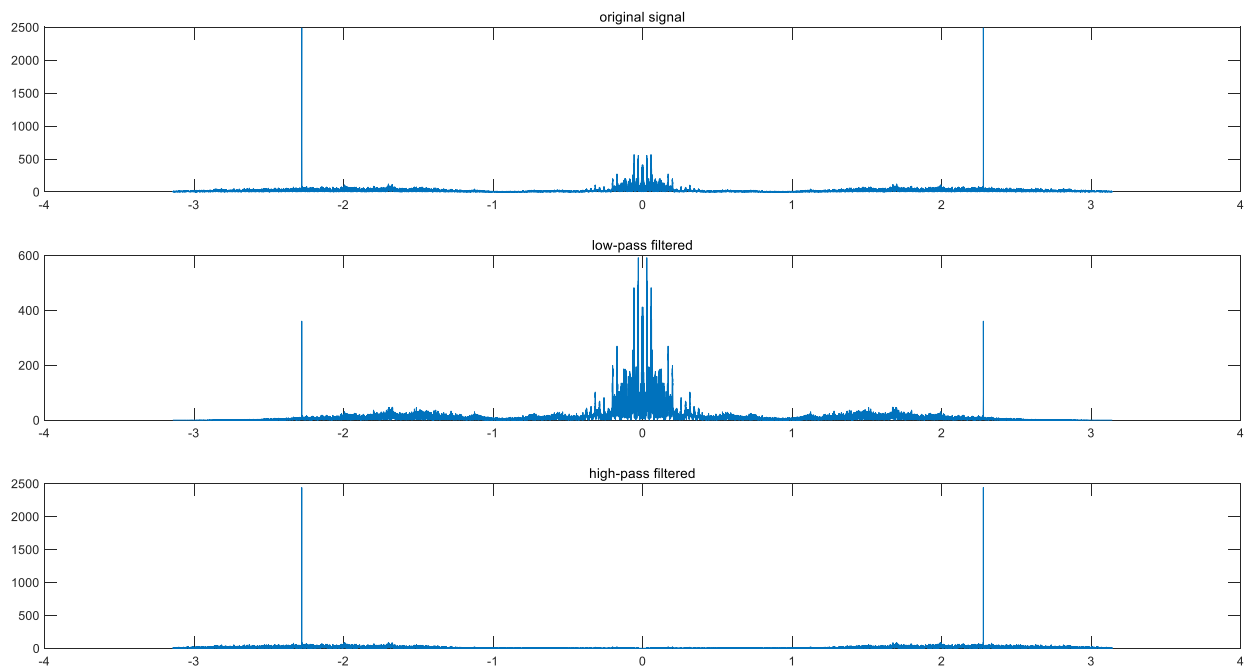
c. (Source Code)

```

25 Y = fft(y);
26 Y = fftshift(Y);
27 Y_lpf = fft(y_lpf);
28 Y_lpf = fftshift(Y_lpf);
29 Y_hpf = fft(y_hpf);
30 Y_hpf = fftshift(Y_hpf);
31
32 angular_freq = -pi:2*pi/(length(y)-1):pi;
33 subplot(3,1,1);
34 plot(angular_freq, abs(Y));
35 title('original signal')
36
37 angular_freq = -pi:2*pi/(length(y_lpf)-1):pi;
38 subplot(3,1,2);
39 plot(angular_freq, abs(Y_lpf));
40 title('low-pass filtered')
41
42 angular_freq = -pi:2*pi/(length(y_hpf)-1):pi;
43 subplot(3,1,3);
44 plot(angular_freq, abs(Y_hpf));
45 title('high-pass filtered')

```

(Result)



Low-pass filtered frequency graph shows that the magnitude of the high frequency has decreased, and the magnitude of the low frequency is almost the same as the original signal. High-pass filtered frequency graph shows that the magnitude of the low frequency has decreased, and the magnitude of the high frequency is almost the same as the original signal.