ECE 311 Lab 4

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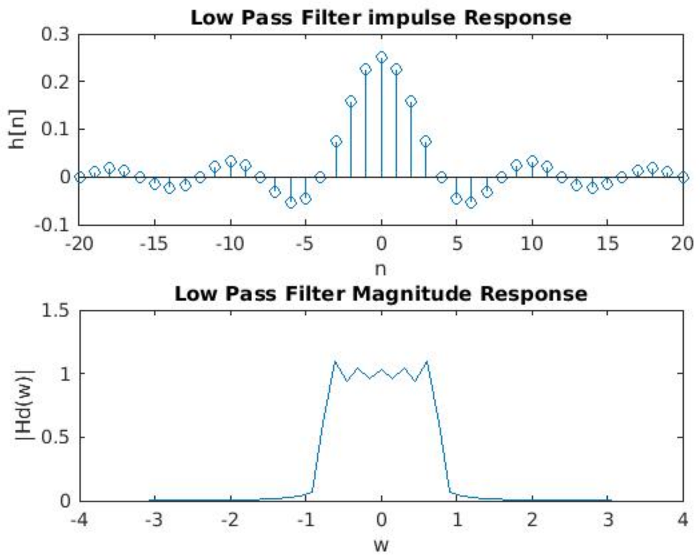
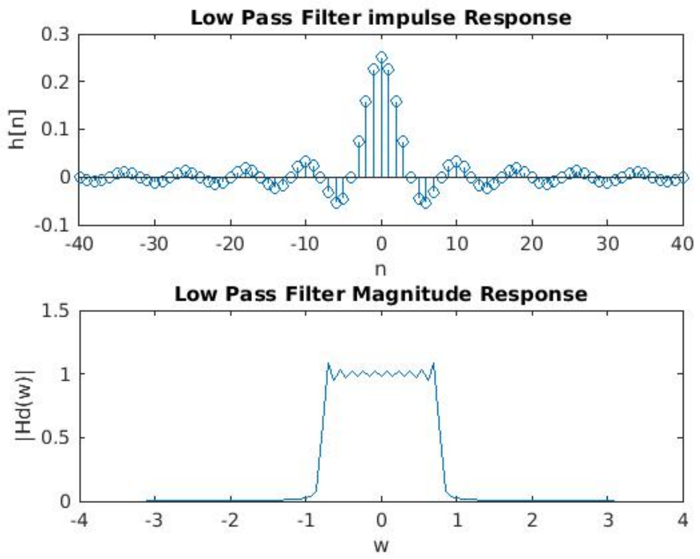
Report Item 1

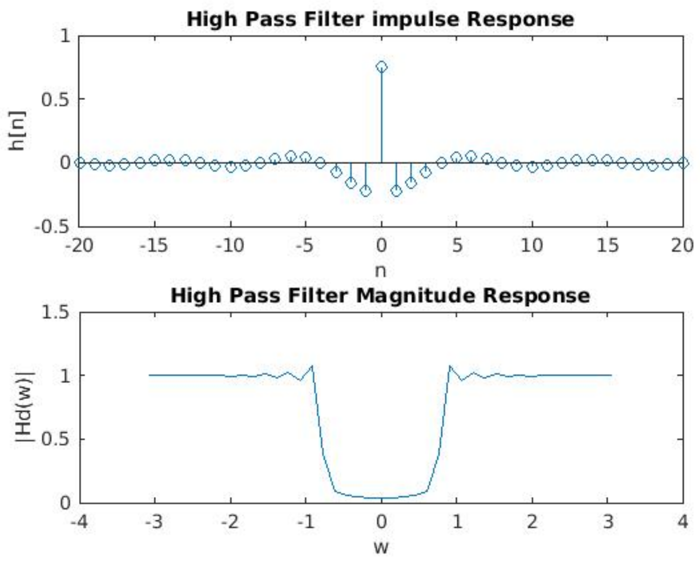
```matlab
function [   ] = filters(N,wc,w0 )
    d = zeros(1,N*2 + 1);
    d(N+1) = 1; % delta function
    n = N*2+1;
    w = fftshift((0:n-1)/n*2*pi);
    w(1:n/2) = w(1:n/2) - 2*pi;
    N = linspace(-N,N,(N*2)+1); % create -N to N array
    lpi = (wc/pi).*sinc(wc.*N./pi);
    lpm = fftshift(fft(lpi));
    hpi = d-lpi;
    hpm = fftshift(fft(hpi));
    bpi = cos(w0.*N).*lpi;
    bpm = fftshift(fft(bpi));

    figure;
    subplot(211);
    stem(N,lpi);
    title('Low Pass Filter impulse Response');
    ylabel('h[n]');
    xlabel('n');
    subplot(212);
    plot(w,abs(lpm));
    title('Low Pass Filter Magnitude Response');
    ylabel('|Hd(w)|');
    xlabel('w');

    figure;
    subplot(211);
    stem(N,hpi);
    title('High Pass Filter impulse Response');
    ylabel('h[n]');
    xlabel('n');
    subplot(212);
    plot(w,abs(hpm));
    title('High Pass Filter Magnitude Response');
    ylabel('|Hd(w)|');
    xlabel('w');

    figure;
    subplot(211);
    stem(N,bpi);
    title('Band Pass Filter impulse Response');
    ylabel('h[n]');
    xlabel('n');
    subplot(212);
    plot(w,abs(bpm));
    title('Band Pass Filter Magnitude Response');
    ylabel('|Hd(w)|');
    xlabel('w');
end
```

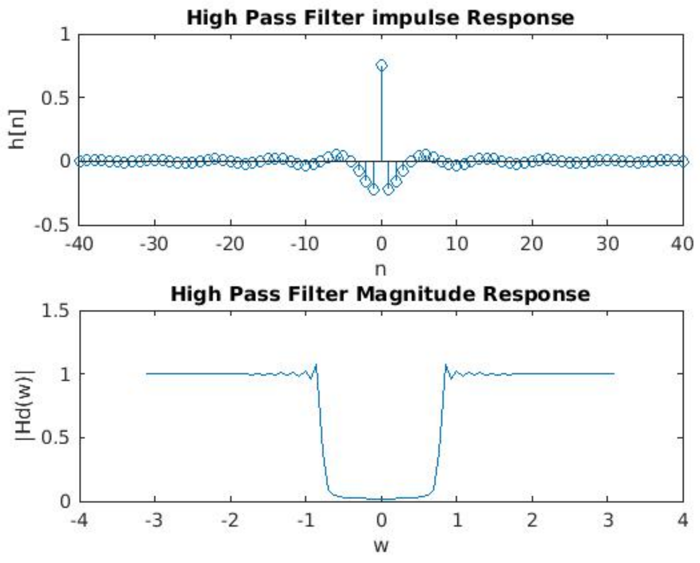
filters.m

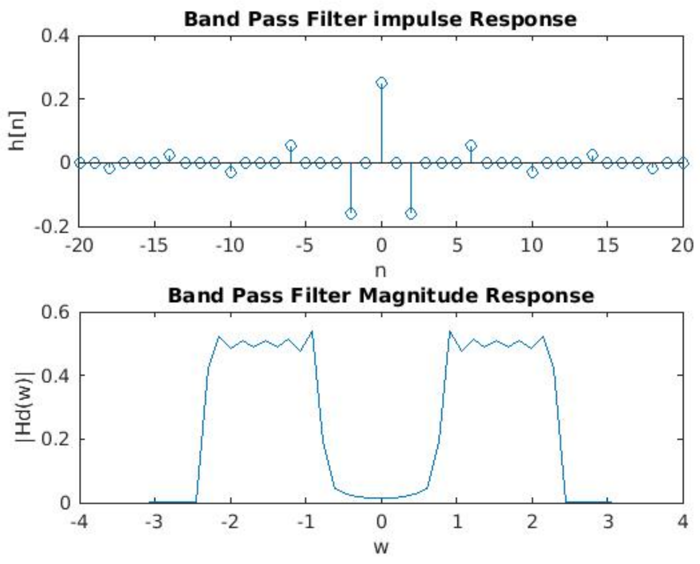


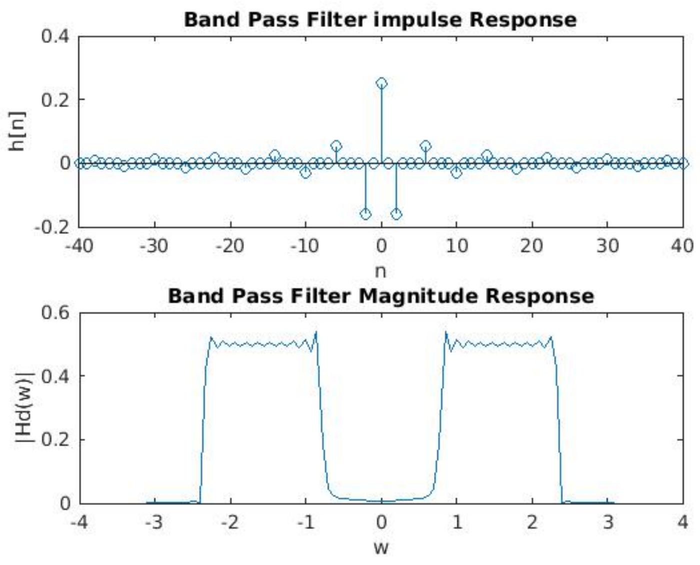


Notice that the magnitude response shows some attenuation and non-vertical edges that would not be present in an ideal model of a filter. This is not possible in a setting with a discrete amount of samples because the jump from low to high or vice versa happens instantaneously.


High Pass Filter impulse Response
h[n]
n
High Pass Filter Magnitude Response
|Hd(w)|
w


High Pass Filter impulse Response
h[n]
n
High Pass Filter Magnitude Response
|Hd(w)|
w

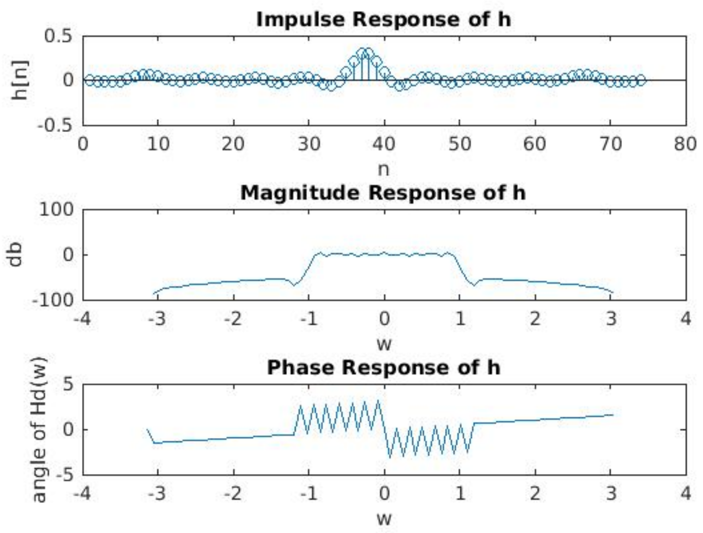

Band Pass Filter impulse Response
h[n]
n
0.4
0.2
0
-0.2
-20
-15
-10
-5
0
5
10
15
20
Band Pass Filter Magnitude Response
|Hd(w)|
w
0.6
0.4
0.2
0
-4
-3
-2
-1
0
1
2
3
4


Band Pass Filter impulse Response
h[n]
n
0.4
0.2
0
-0.2
-40
-30
-20
-10
0
10
20
30
40
Band Pass Filter Magnitude Response
|Hd(w)|
w
0.6
0.4
0.2
0
-4
-3
-2
-1
0
1
2
3
4

Report Item 2

```
load impulseresponse.mat
% variable name is h
figure;
subplot
subplot(311);
stem(h);
n = 74;
w = fftshift((0:n-1)/n*2*pi);
w(1:n/2) = w(1:n/2) - 2*pi;
title('Impulse Response of h');
xlabel('n');
ylabel('h[n]');
subplot(312);
h_m = abs(fftshift(fft(h)));
h_m = mag2db(h_m);
plot(w,h_m);
title('Magnitude Response of h');
xlabel('w');
ylabel('db');
subplot(313);
h_p = angle(fftshift(fft(h)));
plot(w,h_p);
title('Phase Response of h');
xlabel('w');
ylabel('angle of Hd(w)');

%find pass band ripple
top = max(h_m);
bottom_range = h_m(28:48);
bottom = min(bottom_range);
passband_ripple = top - bottom;
% result is 8.0126
%passband edge is approximately .75 rad to 1.25 rad so .5 rad
```

impresp.m

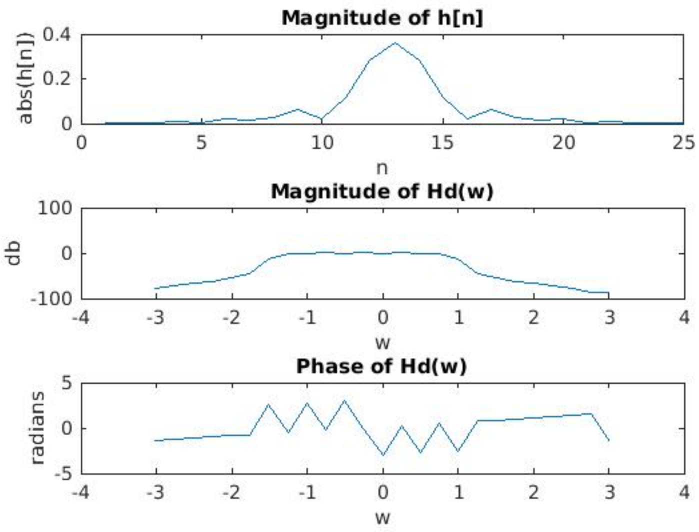


The transition bandwidth is around .5 radians wide. I calculated the maximum passband ripple(max - min) to be around 8db. The stopband ripple(max - min) seems to be around 20db.

Report Item 3

```
N = 25;
M = (N-1)/2;
w = fftshift((0:N-1)/N*2*pi); % 1. define omega as you would for
    FFT
w(1:N/2) = w(1:N/2) - 2*pi;   %
i=sqrt(-1);
for j=1:N
    if(abs(w(j)) < pi/3), % 2.
        g_w(j) = 1 * exp(-i*M*w(j));
    else
        g_w(j) = 0;
    end
end

g_n = ifft(fftshift(g_w)); % 3. find g[n], should be shifted
w_n = hamming(N)'; % window (transposed)
h_n = g_n .* w_n;% h_n is impulse response
figure;
subplot(311);
plot(abs(h_n));
title('Magnitude of h[n]');
xlabel('n');
ylabel('abs(h[n])');
subplot(312);
plot(w,mag2db(abs(fftshift(fft(h_n)))));
title('Magnitude of Hd(w)');
xlabel('w');
ylabel('db');
subplot(313);
plot(w,angle(fftshift(fft(h_n))));
title('Phase of Hd(w)');
xlabel('w');
ylabel('radians');
```

FIR_FILTER.m

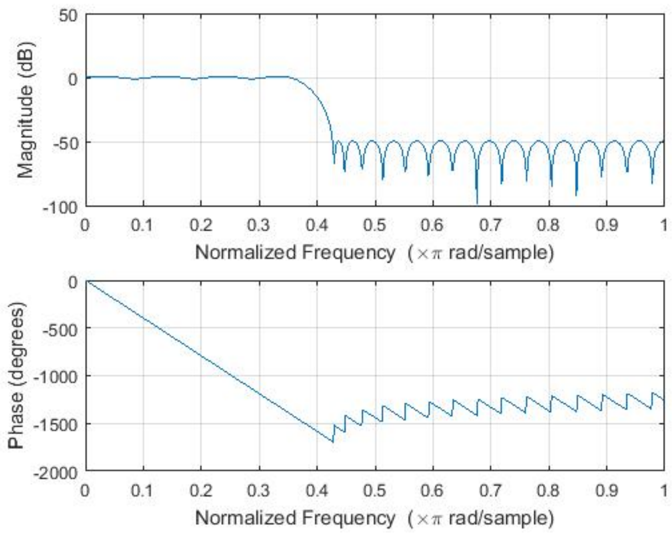


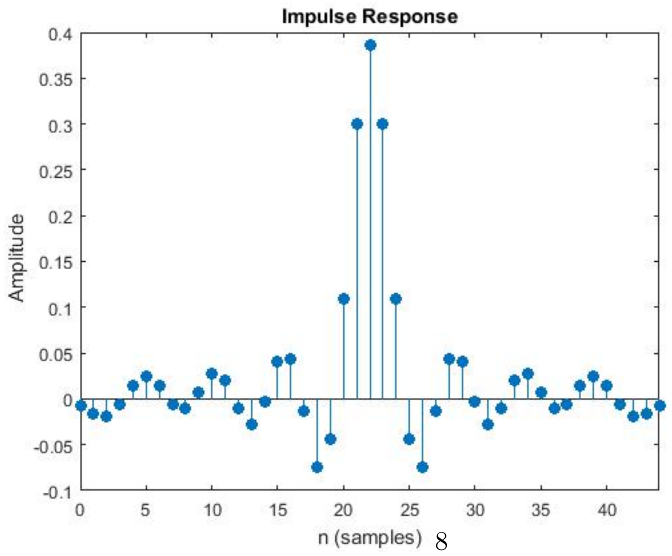
The passband ripple apears to be about 5db, stopband, about 25db. The passband edge frequency is 0.9 radians and the stopband edge frequency is about 1.2 radians.

Report Item 4

```
clc, clear all, close all
f = [54,64];
a = [1,0];
rp = [2];
rs = 50;
fs = 300;
dev = [(10^(rp/20)-1)/(10^(rp/20)+1)  10^(-rs/20)];
[n,fo,mo,w] = firpmord(f, a, dev, fs);
b = firpm(n,fo,mo,w);
freqz(b,1);
figure
impz(b,1);
```

report4.m

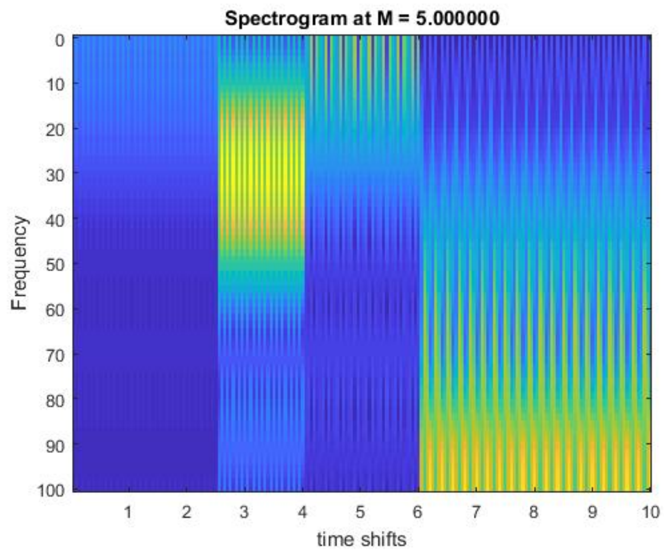


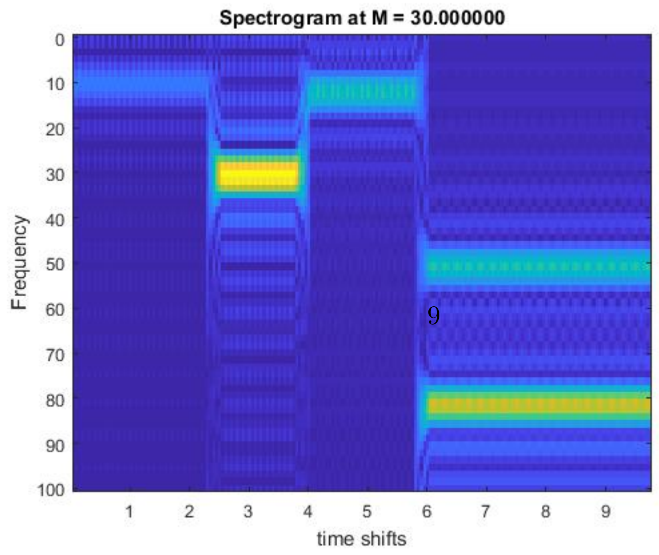


Report Item 5

```
function [a, b, c] = mySTDFT(x, M, D, P, f_s )
a_dim1 = ceil(P/2);
a_dim2 = floor(((length(x) - M)/D) + 1);
dft_mat = zeros(a_dim1,a_dim2);
time_shifts = zeros(1,a_dim2);
for i=1:a_dim2,
    time_shifts(i) = (i*D)/f_s; % vector of shifts
    time_slice = x((1+D*(i-1)):(((i-1)*D)+M)); % get time slice
    time_slice = fft(time_slice,P); % dft zero padded to P
    time_slice = time_slice(1,1:a_dim1); % only take half of P
    dft_mat(:,i) = time_slice; %assign column in dft_mat
end
    a = dft_mat; % output 1
    b = linspace(0,f_s,a_dim1); % output 2
    c = time_shifts; % output 3
    imagesc(c,b,abs(a));
    ylabel('Frequency');
    xlabel('time shifts');
    str = sprintf('Spectrogram at M = %f',M);
    title(str);
end
```

mySTDFT.m





By the spectrogram we can see that(M = 5) there are frequencies ranging from 10 to 50 hz at 2.5 to 4 seconds. Then there are also 80 to 100 hz at 6 to 10 seconds. With M =30, We see frequencies 10hz at 1:2 seconds, 20hz at 2.5:4 seconds, 10hz at 4:6 seconds, and 50 and 85 hz at 6:10 seconds. When we increase M to 30 our minimum frequency to be distinguished goes down and therefore the frequency bands are more distinguishable. Note that $N\Delta t = \frac{1/fmin}{}$, increasing N will decrease fmin.

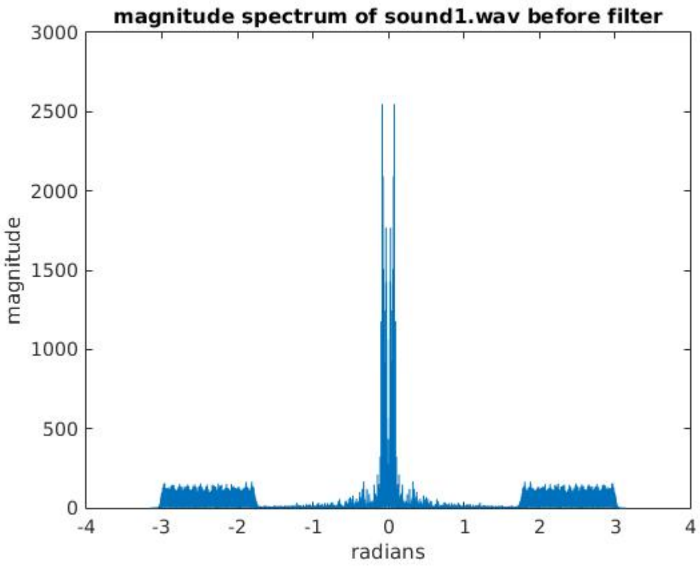
Report Item 6

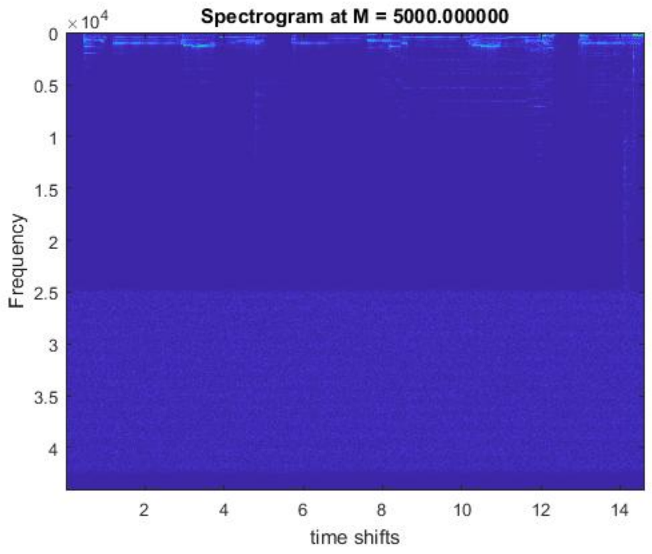
```
[y fs] = audioread('sound1.wav');

N = length(y);
w = fftshift((0:N-1)/N*2*pi); % define omega as you would for FFT
w(1:N/2) = w(1:N/2) - 2*pi;   %
y_w = fftshift(fft(y));
figure;
plot(w,abs(y_w));
ylabel('magnitude');
xlabel('radians');
title('magnitude spectrum of sound1.wav before filter');
m = 5000;
d = 5;
p = 1024;
mySTDFT(y',m,d,p,fs);



f= [0 .4 .5 1];
a = [1 1 0 0];
b = firpm(50,f,a);
b_w = fftshift(fft(b,length(y)))'; % get filter

y_w = b_w .* y_w; % apply filter
figure;
plot(w,abs(y_w));
ylabel('magnitude');
xlabel('radians');
title('magnitude spectrum of sound1.wav after filter');

y = ifft(ifftshift(y_w));
soundsc(y);
filename = 'filtered1.wav';
audiowrite(filename,y,fs);
mySTDFT(y',m,d,p,fs);
```

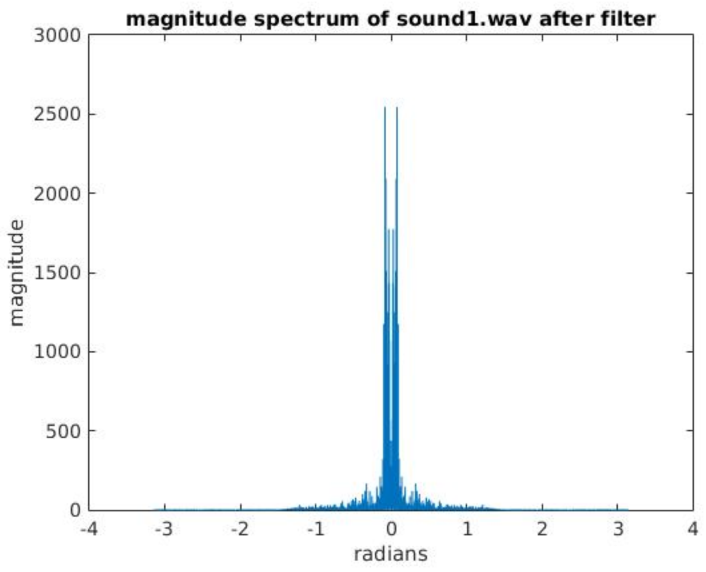
report6.m

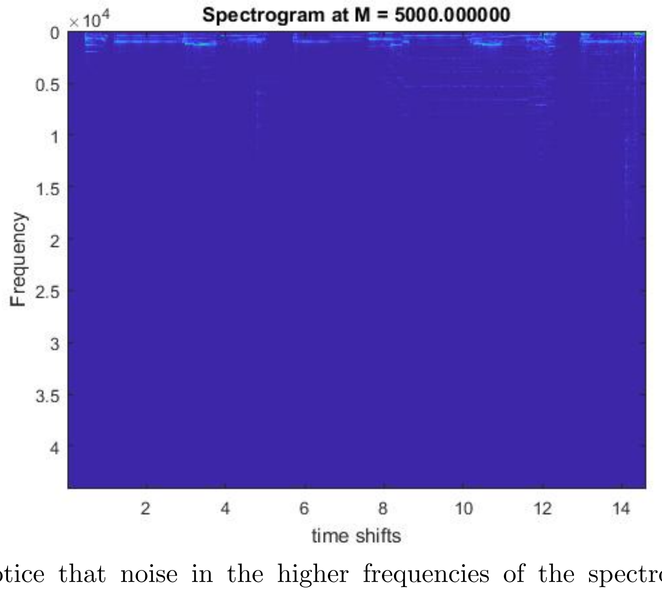
The audio file when played at real time is 15 seconds.

Before filter, I used paramaters $M = 5000$, $P = 1024$, $D = 5$. The signal and sampling frequency are given to us.







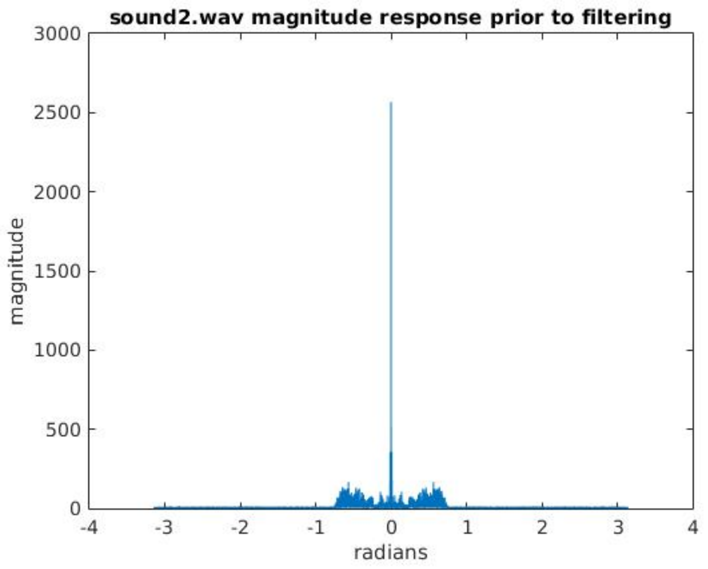


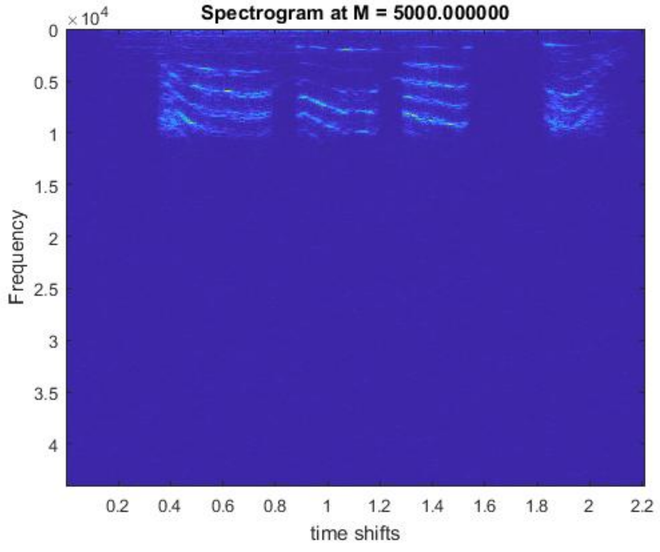
Notice that noise in the higher frequencies of the spectrogram are cleaned. Because the frequencies are mutally exclusive, the filter works well.

Report Item 7

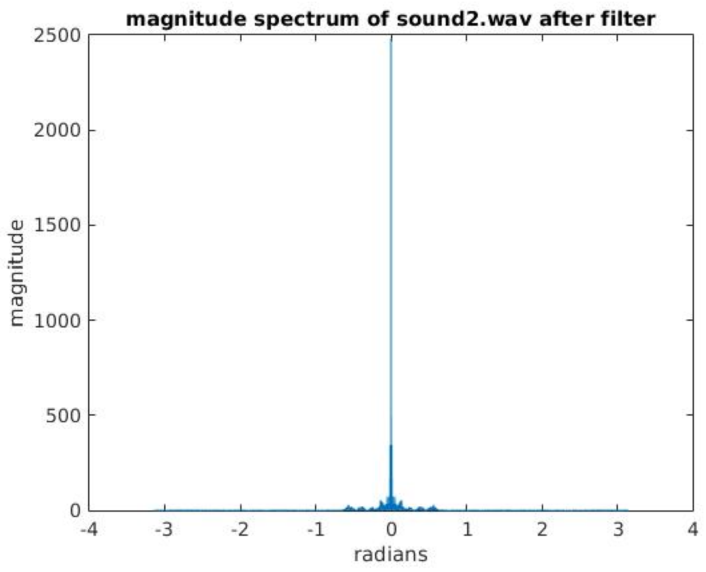
```
[y fs] = audioread('sound2.wav');
%soundsc(y);
N = length(y);
w = fftshift((0:N-1)/N*2*pi); % define omega as you would for FFT
w(1:N/2) = w(1:N/2) - 2*pi;   %

yw = fftshift(fft(y));
figure;
plot(w,abs(yw));
title('sound2.wav magnitude response prior to filtering');
xlabel('radians');
ylabel('magnitude');
figure;
m = 5000;
d = 5;
p = 1024;
mySTDFT(y',m,d,p,fs);

f= [0 .1 .2 1];
a = [1 0 0 0];
b = firpm(50,f,a);
b_w = fftshift(fft(b,length(y)))'; % get filter

yw = b_w .* yw; % apply filter
figure;
plot(w,abs(yw));
ylabel('magnitude');
xlabel('radians');
title('magnitude spectrum of sound2.wav after filter');


y = ifft(ifftshift(yw));
figure;
mySTDFT(y',m,d,p,fs);
soundsc(y);
filename = 'filtered2.wav';
audiowrite(filename,y,fs);
```

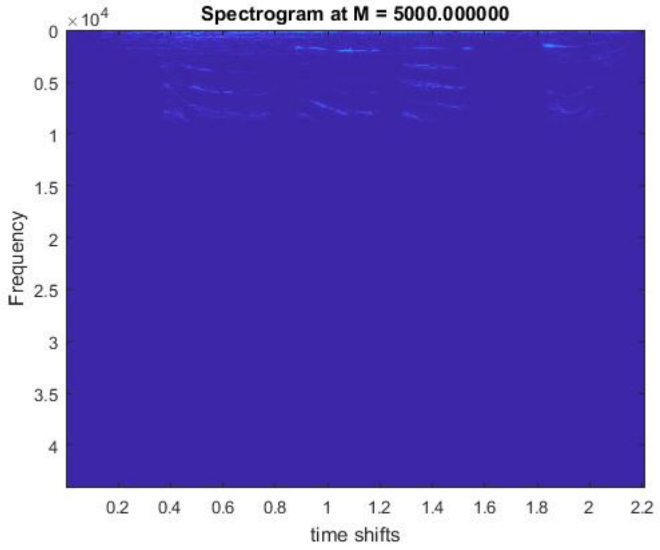
report7.m





Before filter, I used paramaters $M = 5000$, $P = 1024$, $D = 5$. The signal and sampling frequency are given to us. In the spectrogram we can see noise distributed among the low frequency bands.





I tried to filter out all of the noise outside of the lowest frequency but with such a small band of desired frequency, creating a sinc of that small width is hard. Therefore some noise still remains. Also, the desired sound is quieter after filtering. Overall, the filter helped, but could be better.

```matlab

y = load('speechsig.mat');
x = y.x;
xnoise = y.xnoise;

N = length(xnoise);
w = fftshift((0:N-1)/N*2*pi); % define omega as you would for FFT
w(1:N/2) = w(1:N/2) - 2*pi;   %

xnoisew = fftshift(fft(xnoise));

h = hamming(N);
xnoisewh = xnoisew .* h;

xnoise = ifft(ifftshift(xnoisewh),N);
%soundsc(real(xnoise));

r = zeros(N,1);
r(2000:4999,1) = rectwin(3000);

xnoisewr = xnoisew .* r;
plot(abs(xnoisew));
xnoise = ifft(ifftshift(xnoisewr));


%soundsc(real(xnoise));
```
report8.m