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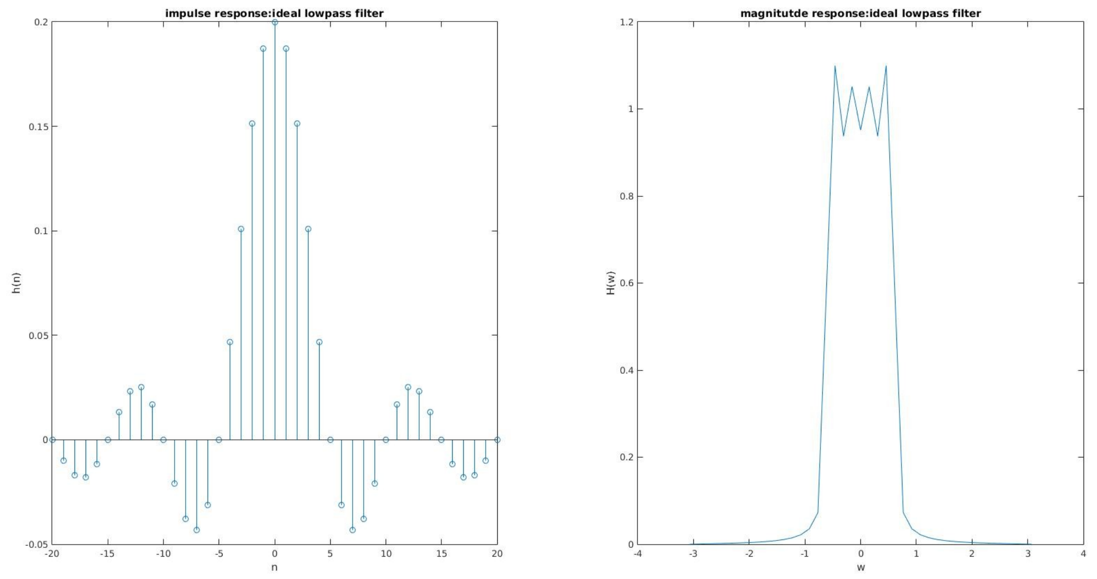
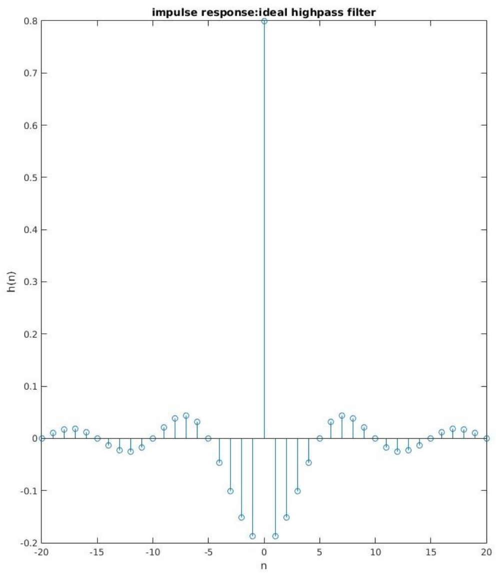
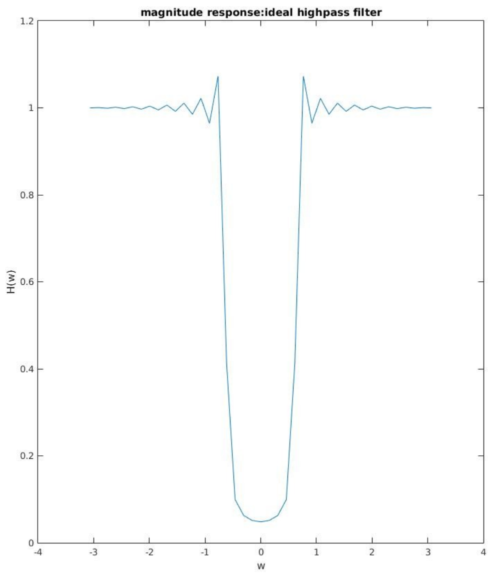
Report Item 1:
code:

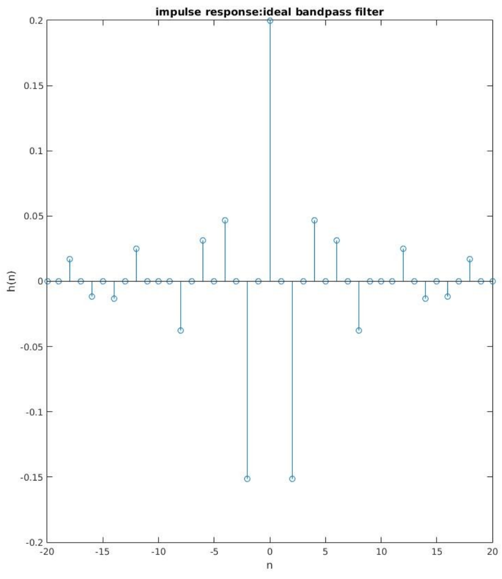
```
N = 20
n = -N:N
wc = pi/5
w0 = pi/2

lowpass = wc/pi * sinc(wc/pi * n)
highpass = [n==0] - lowpass
bandpass = cos(w0*n) .* lowpass

LOWPASS = fft(lowpass)
LOWPASS = fftshift(LOWPASS)
HIGHPASS = fft(highpass)
HIGHPASS = fftshift(HIGHPASS)
BANDPASS = fft(bandpass)
BANDPASS = fftshift(BANDPASS)

size = 2*N+1
w = fftshift((0:size - 1)/size*2*pi);
w(1:size/2) = w(1:size/2) - 2*pi;

figure(1)
subplot(121)
stem(n,lowpass)
title('impulse response:ideal lowpass filter')
xlabel('n')
ylabel('h(n)')
subplot(122)
plot(w,abs(LOWPASS))
title('magnitutde response:ideal lowpass filter')
xlabel('w')
ylabel('H(w)')

 figure(2)
subplot(121)
stem(n,highpass)
title('impulse response:ideal highpass filter')
xlabel('n')
ylabel('h(n)')
subplot(122)
plot(w,abs(HIGHPASS))
title('magnitude response:ideal highpass filter')
xlabel('w')
ylabel('H(w)')

figure(3)
subplot(121)
```

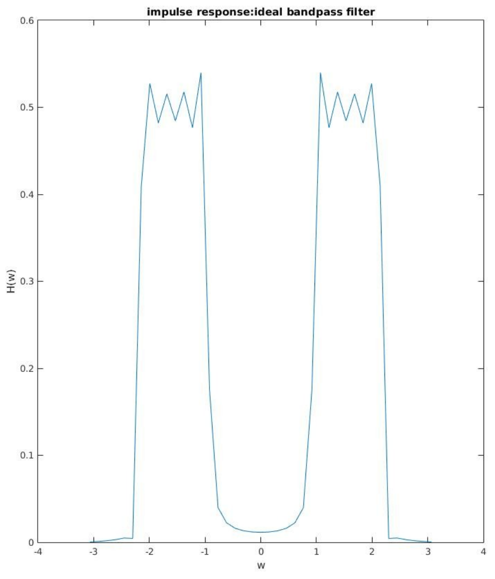
```
stem(n,bandpass)
title('impulse response:ideal bandpass filter')
xlabel('n')
ylabel('h(n)')
subplot(122)
plot(w,abs(BANDPASS))
title('impulse response:ideal bandpass filter')
xlabel('w')
ylabel('H(w)')

figure:
N = 20
```



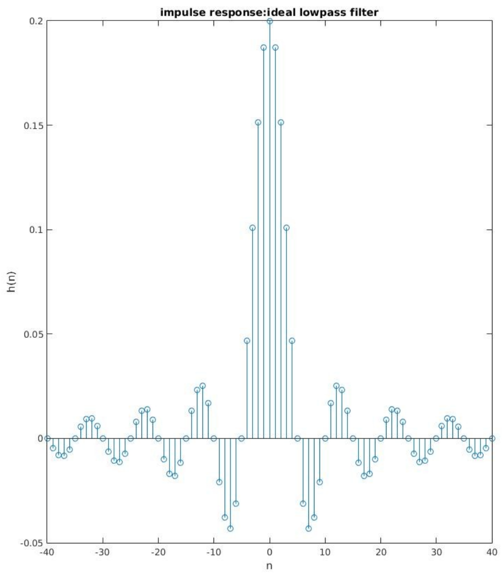

impulse response:ideal highpass filter
h(n)
n

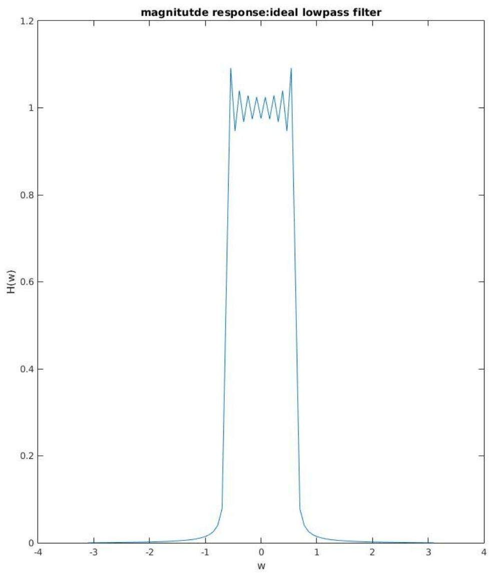

magnitude response:ideal highpass filter
H(w)
w

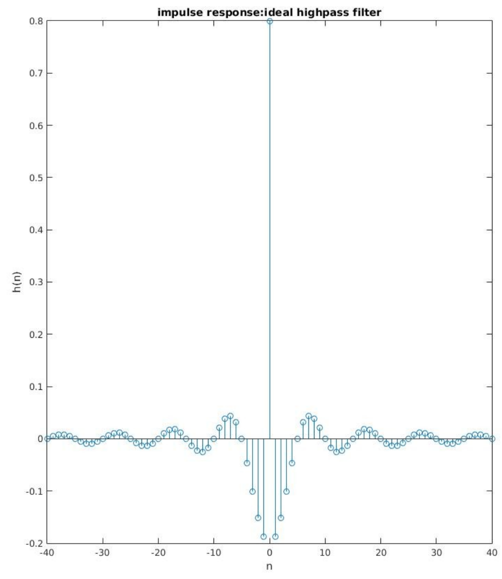

impulse response:ideal bandpass filter
h(n)
n

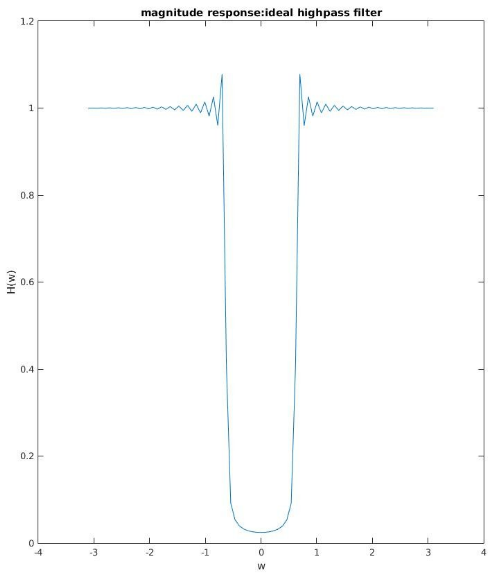

impulse response:ideal bandpass filter
H(w)
w

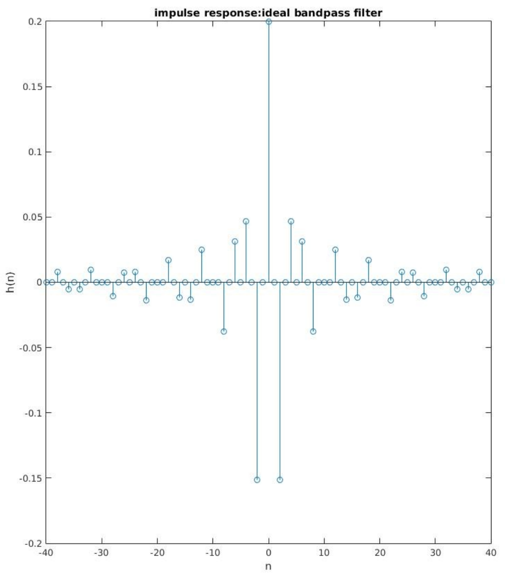
N = 40

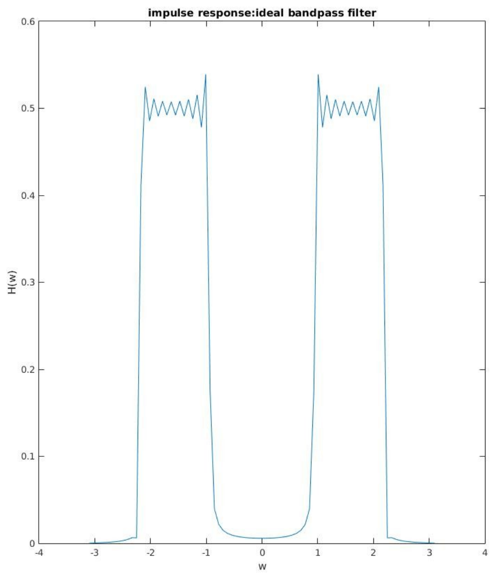

impulse response:ideal lowpass filter
h(n)
n
0.2
0.15
0.1
0.05
0
-0.05
-40
-30
-20
-10
0
10
20
30
40


magnitutde response:ideal lowpass filter
H(w)
w
1.2
1
0.8
0.6
0.4
0.2
0
-4
-3
-2
-1
0
1
2
3
4


impulse response:ideal highpass filter
h(n)
n
0.8
0.7
0.6
0.5
0.4
0.3
0.2
0.1
0
-0.1
-0.2
-40
-30
-20
-10
0
10
20
30
40


magnitude response:ideal highpass filter
H(w)
w
1.2
1
0.8
0.6
0.4
0.2
0
-4
-3
-2
-1
0
1
2
3
4





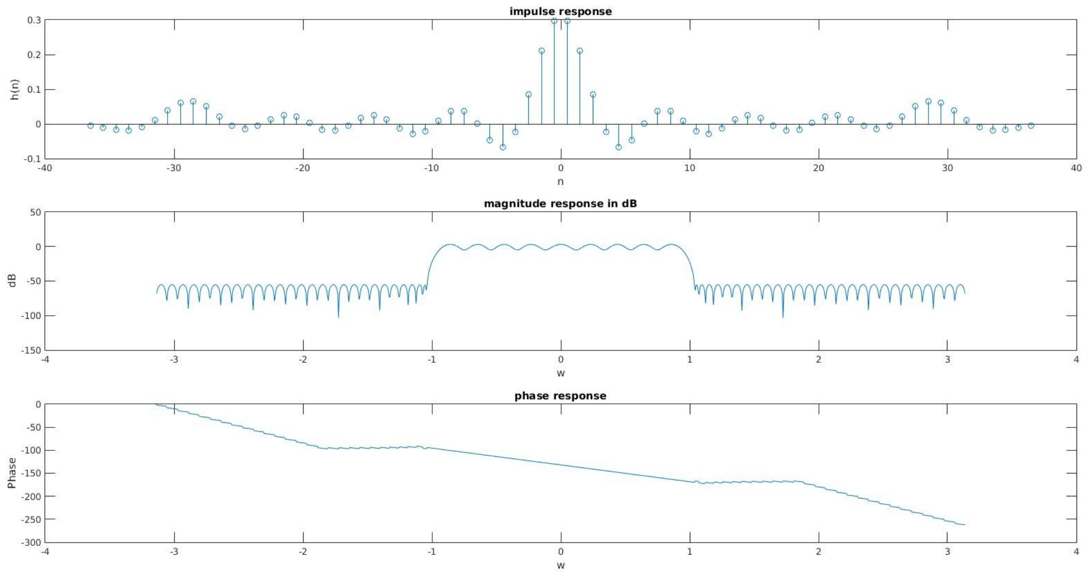
The difference from the ideal magnitude response is that the magnitude responses in the figure have Gibbs phenomenon and there are also ripples in the pass-band. The main reason why this is happening is that the ideal filter is of infinite duration and not causal.

Report Item 2:
code:

```
load impulseresponse
N = length(h)
n = -(N-1)/2 : 1 : (N-1)/2;
n = n'

pad = 1024
H = fft(h,pad)
H = fftshift(H)
w = fftshift((0:pad - 1)/pad*2*pi);
w(1:pad/2) = w(1:pad/2) - 2*pi;
db = mag2db(abs(H))

subplot(311)
stem(n,h)
title('impulse response')
xlabel('n')
ylabel('h(n)')
```

```
subplot(312)
plot(w,db)
title('magnitude response in dB')
xlabel('w')
ylabel('dB')


subplot(313)
plot(w,phase(H))
title('phase response')
xlabel('w')
ylabel('Phase')
```

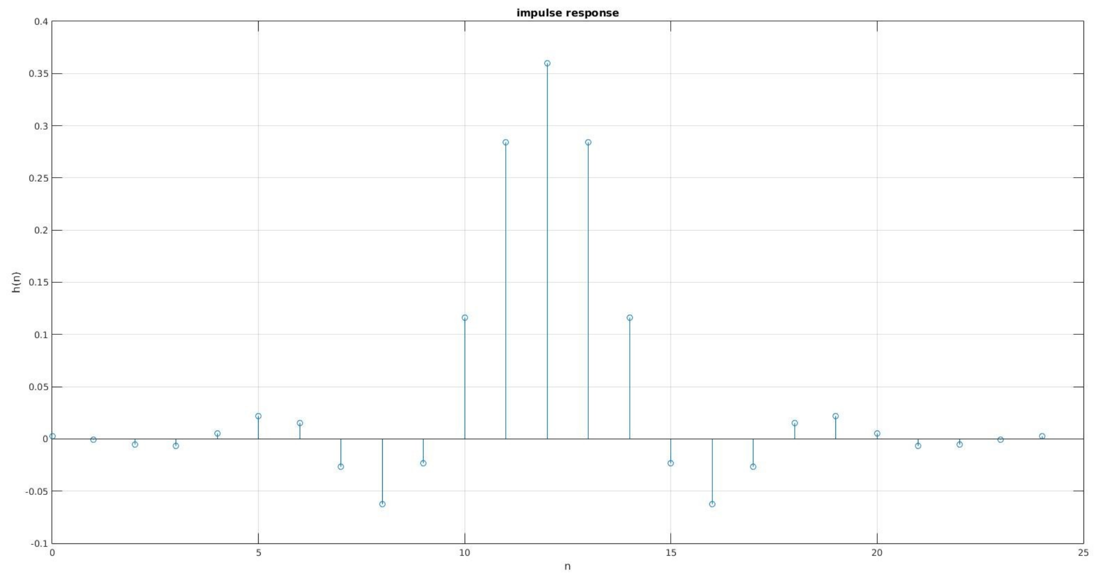
figure:

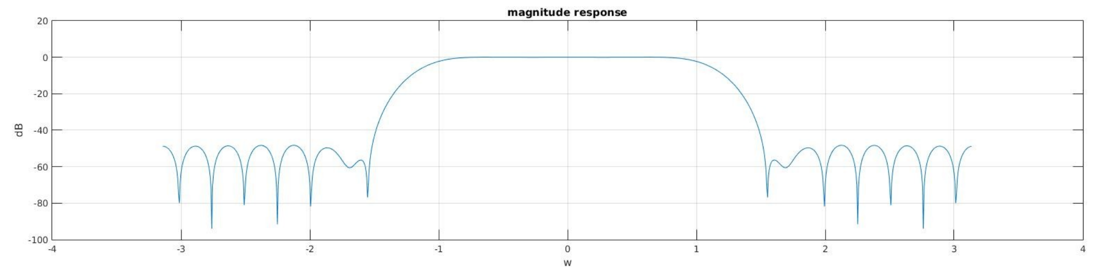


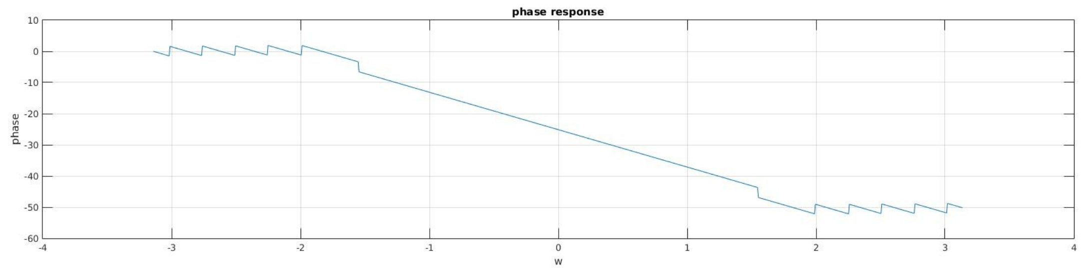
```
stop-band ripple:30dB
pass-band ripple:7dB
transition bandwidth:0.2



Report Item 3:
code:
N = 25;
cutoff = pi/3
M = (N-1)/2
w = fftshift((0:N-1)/N*2*pi);
w(1:N/2) = w(1:N/2)-2*pi;
Dw = zeros(1,length(w))
Dw(9:17) = 1
Gw = Dw.*exp(-i*M*w)
gn = real(ifft(ifftshift(Gw)))
wn = hamming(N)
```

```
hn = wn .* real(gn')
figure(1)
stem([0:N-1],hn)
grid on
title('impulse response')
xlabel('n')
ylabel('h(n)')

size = 1024

H = fft(hn,size)
H = fftshift(H)
o = fftshift((0:size-1)/size*2*pi)
o(1:size/2) = o(1:size/2) - 2*pi;
figure(2)
subplot(211)
plot(o,mag2db(abs(H)))
title('magnitude response')
xlabel('w')
ylabel('dB')
grid on
subplot(212)
plot(o,phase(H))
title('phase response')
xlabel('w')
ylabel('phase')
grid on
```

figure:







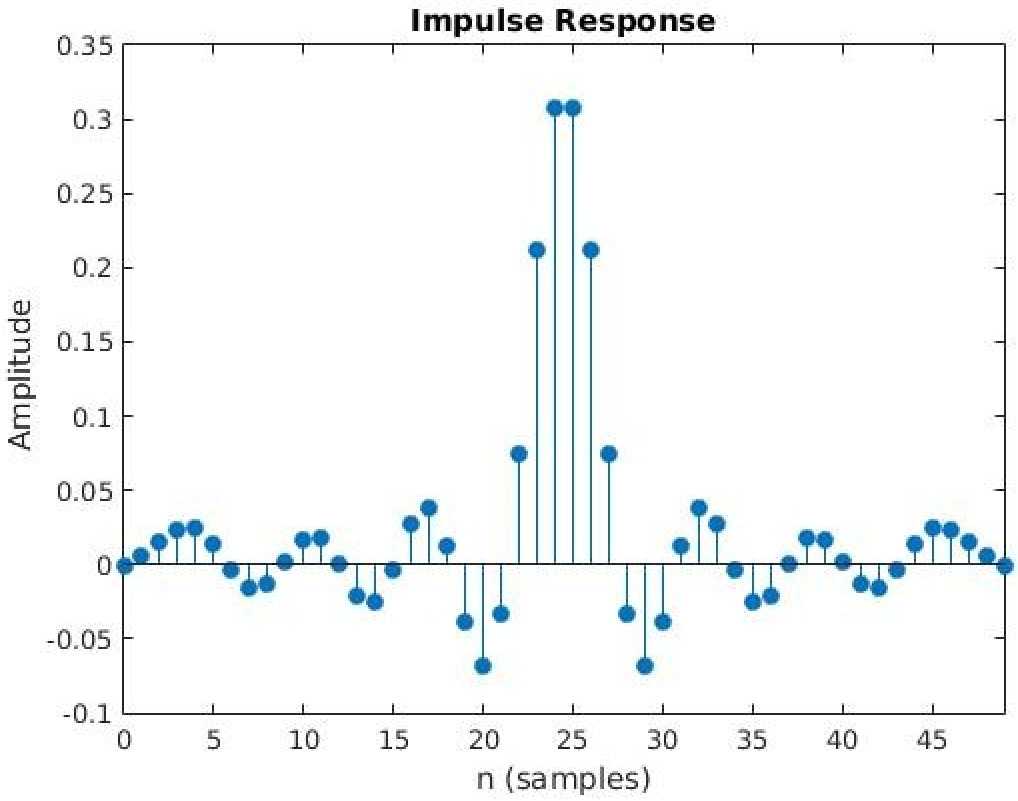
```
pass-band ripple:0.26dB
stop-band attenuation:50dB
pass-band edge frequency:0.9
stop-band edge frequency:1.5
```

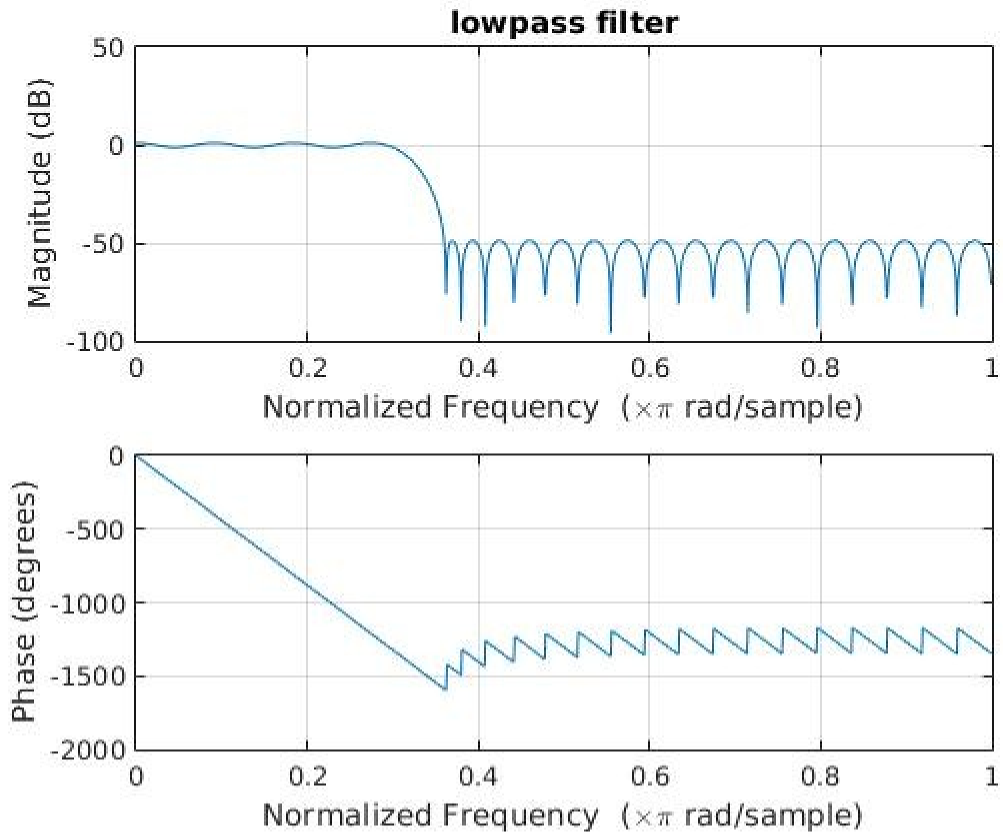
Report Item 4:

code:

```
rp = 2
a = [1,0]
fs = 1
f = [0.3/2,0.36/2]
rs = 50
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,1,1024)
title('lowpass filter')
figure;
impz(b,1)
```

figure:


Impulse Response
Amplitude
n (samples)
0.35
0.3
0.25
0.2
0.15
0.1
0.05
0
-0.05
-0.1
0
5
10
15
20
25
30
35
40
45



Report Item 5
code:

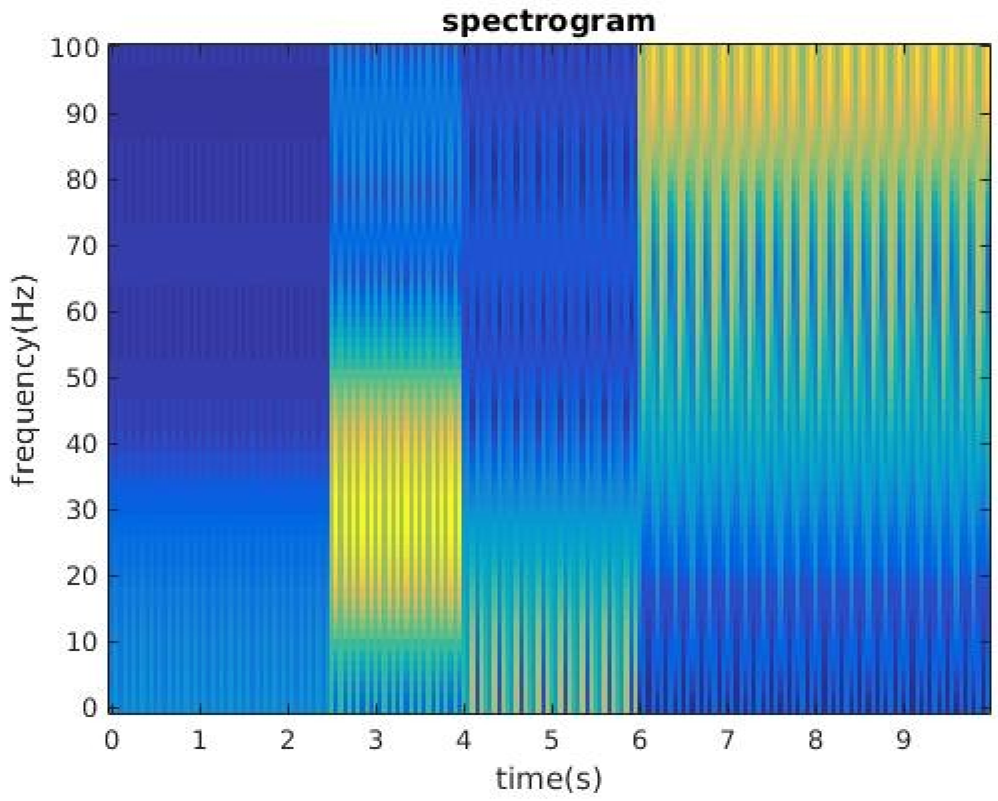
```
function [stdft,Omega,shift] = mySTDFT(x,M,D,P,fs)
N = length(x);
n = 0:length(x)-1;
row = ceil(P/2);
col = floor((N-M)/D)+1;
stdft = zeros(row,col);

for i = 1:col
y = x(((i-1)*D +1):(i-1)*D+M);
Y = fft(y,P);
stdft(:,i) = Y(1:row);
end

%shift

m = [0:col-1];
shift = D*m/fs;

%omega
w = [0:col-1]/col*2*pi;
Omega = fs*w;
end
```

```
load spectrogram
M = 5
P = 128
D = 5
[stdft,analog,shift] = mySTDFT(x,M,D,P,fs)
imagesc(shift,analog/2/pi,abs(stdft))
axis('xy')
title('spectrogram')
xlabel('time(s)')
ylabel('frequency(Hz)')
```

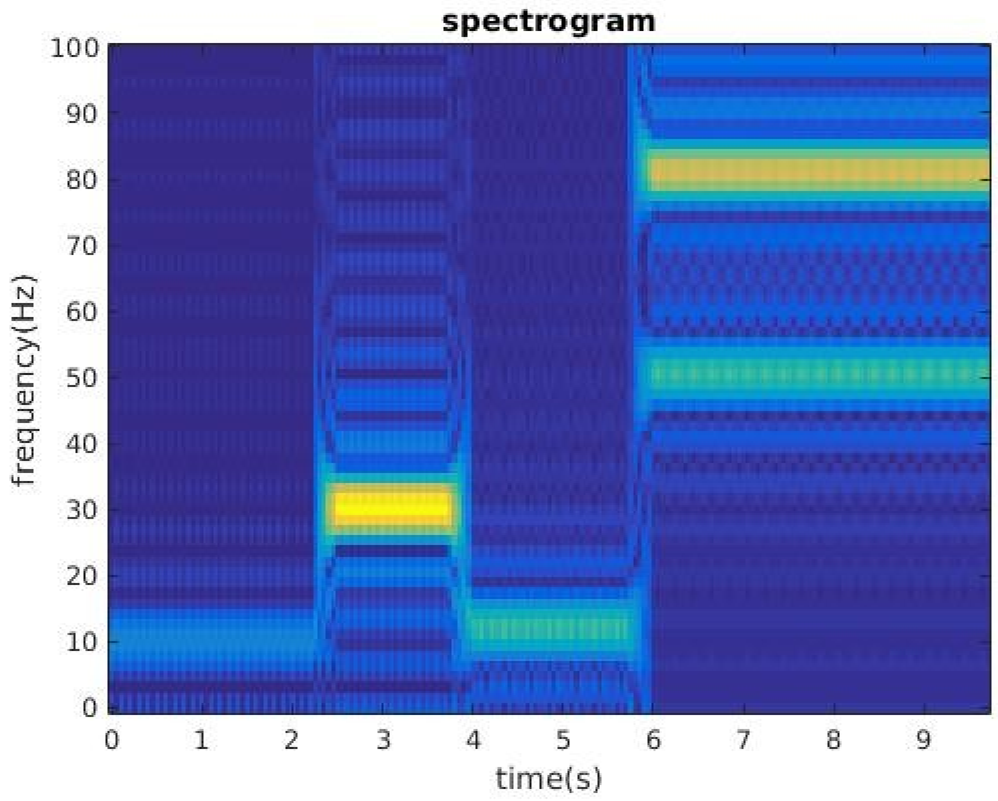
```
load spectrogram
M = 30
P = 128
D = 5
[stdft,analog,shift] = mySTDFT(x,M,D,P,fs)
imagesc(shift,analog/2/pi,abs(stdft))
axis('xy')
title('spectrogram')
xlabel('time(s)')
ylabel('frequency(Hz)')
```

figure:

M = 5:


spectrogram
frequency(Hz)
time(s)
0
10
20
30
40
50
60
70
80
90
100
1
2
3
4
5
6
7
8
9

```
M = 30
```



o to
100 Hz is present in the spectrogram
10 to 50 Hz occurs from 2.5 to 4s
60 to 100 Hz occurs after 6s

When M = 30
The frequency resolution increases. The minimum frequency captured is lower.
Rayleigh limit states:

$$M/fs < 1/fmin$$
$$fmin > fs/M$$

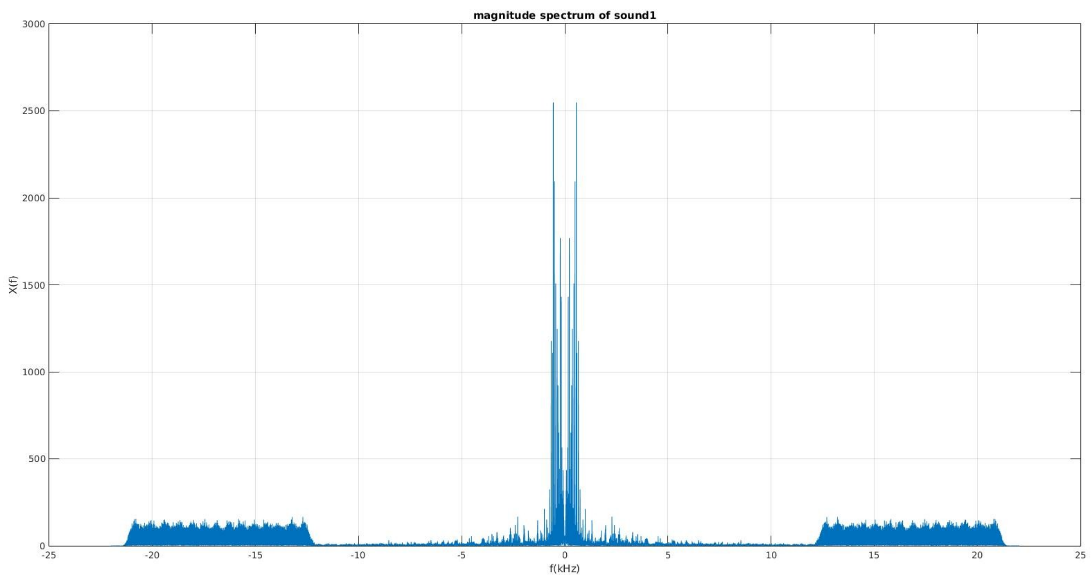
10Hz component is not captured when M = 5 but it is present when M = 30

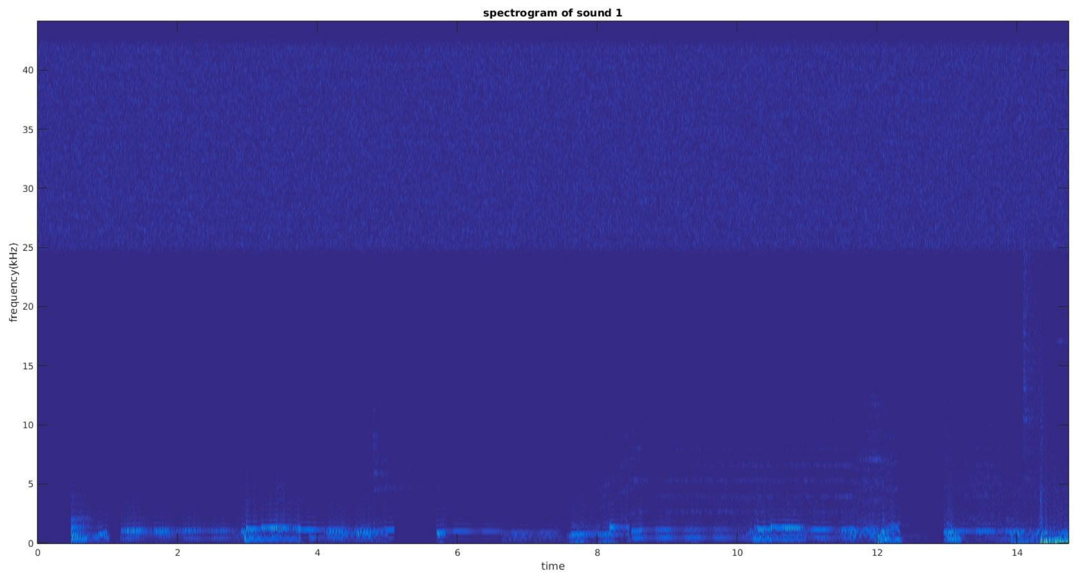
Report Item 6:
code:

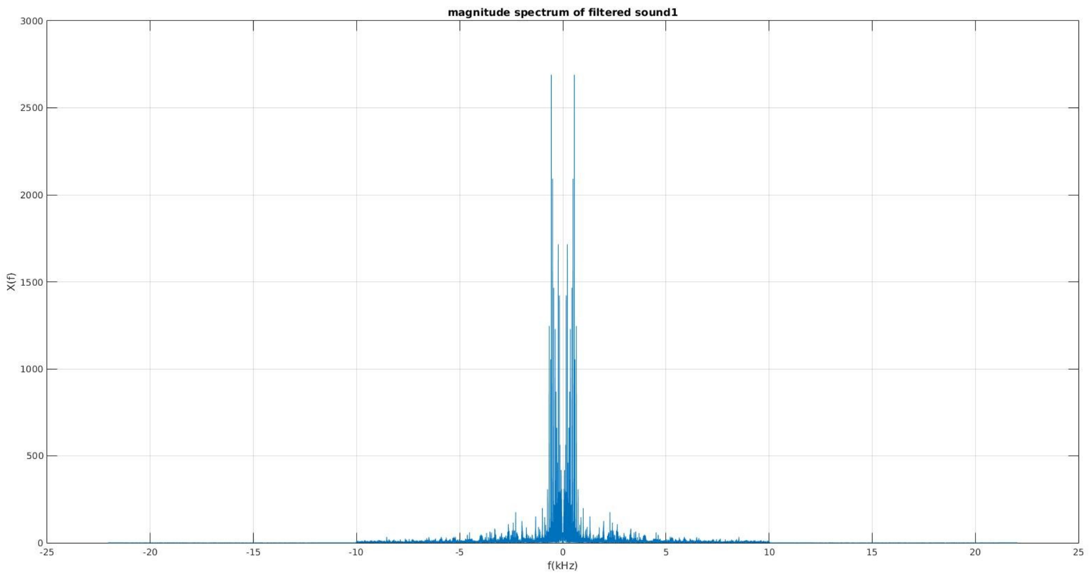
```
[Y,fs] = wavread('sound1')
N = length(Y);
time = length(Y)/fs;
Y_transform = fft(Y);
Y_transform = fftshift(Y_transform);
w = fftshift((0:N - 1)/N*2*pi);
w(1:N/2) = w(1:N/2) - 2*pi;
omega = fs*w/1000/2/pi;%analog kHz

plot(omega,abs(Y_transform))
```

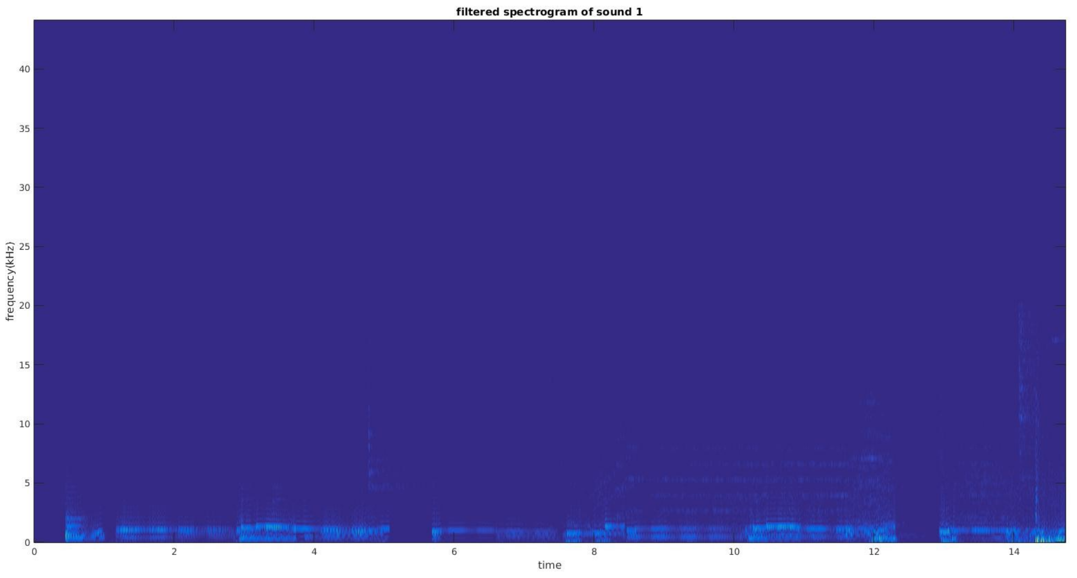
```
title('magnitude spectrum of sound1')
xlabel('f(kHz)')
ylabel('X(f)')
grid on

M = 200
P = 2048
D = 5
[stdft,analog,shift] = mySTDFT(Y,M,D,P,fs);
figure
imagesc(shift,analog/2/pi/1000,abs(stdft));
axis('xy')
title('spectrogram of sound 1')
xlabel('time')
ylabel('frequency(kHz)')



%filter
rp = 1
a = [1 0];
fs = 44100
f = [10000 10050];
rs = 60
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);



B = fft(b,N);
newY = B'.*fft(Y);
newy = ifft(newY);
figure
plot(omega,abs(fftshift(newY)))
title('magnitude spectrum of filtered sound1')
xlabel('f(kHz)')
ylabel('X(f)')
grid on
soundsc(newy,fs)

[stdft,analog,shift] = mySTDFT(newy,M,D,P,fs);
figure
imagesc(shift,analog/2/pi/1000,abs(stdft));
axis('xy')
title('filtered spectrogram of sound 1')
xlabel('time')
ylabel('frequency(kHz)')
```

figure:


magnitude spectrum of sound1
X(f)
f(kHz)
3000
2500
2000
1500
1000
500
0
-25
-20
-15
-10
-5
0
5
10
15
20
25


spectrogram of sound 1
frequency(kHz)
time
40
35
30
25
20
15
10
5
0
2
4
6
8
10
12
14


magnitude spectrum of filtered sound1
X(f)
f(kHz)
3000
2500
2000
1500
1000
500
0
-25
-20
-15
-10
-5
0
5
10
15
20
25


filtered spectrogram of sound 1
frequency(kHz)
time
40
35
30
25
20
15
10
5
0
0
2
4
6
8
10
12
14

M = 200

```
P = 2048
D = 5
fs = 44100
```

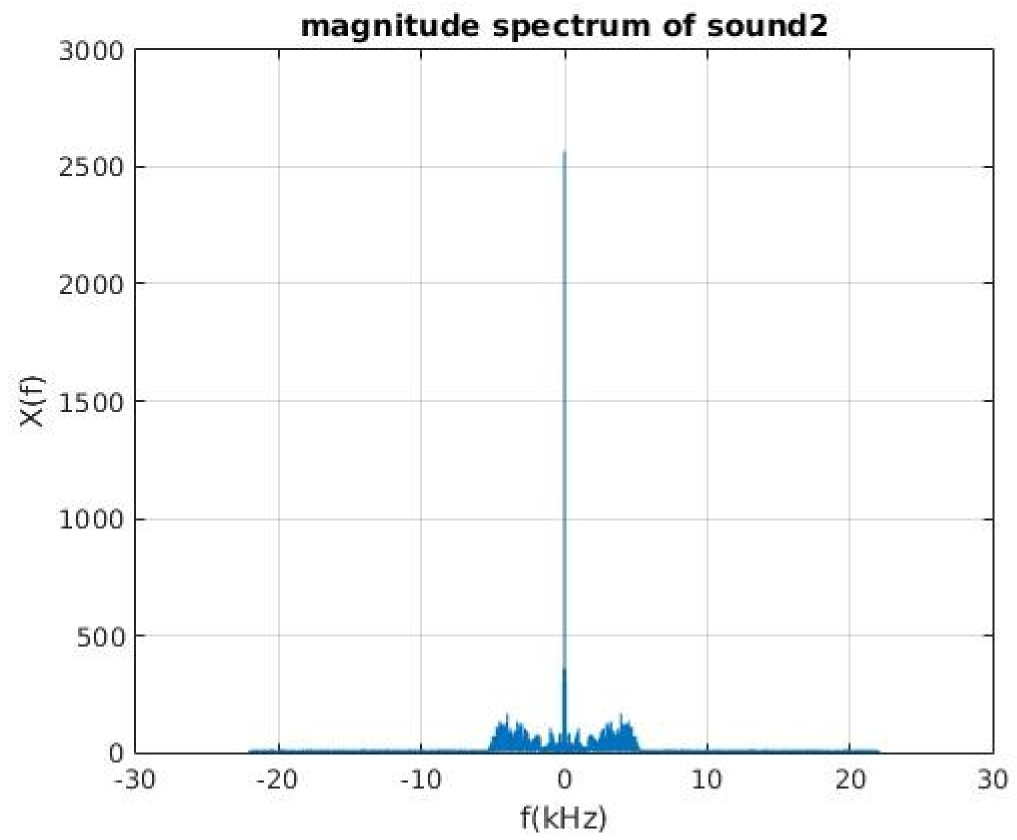
Effectiveness:
White noise is mostly gone. The sound is more pleasant after the filter

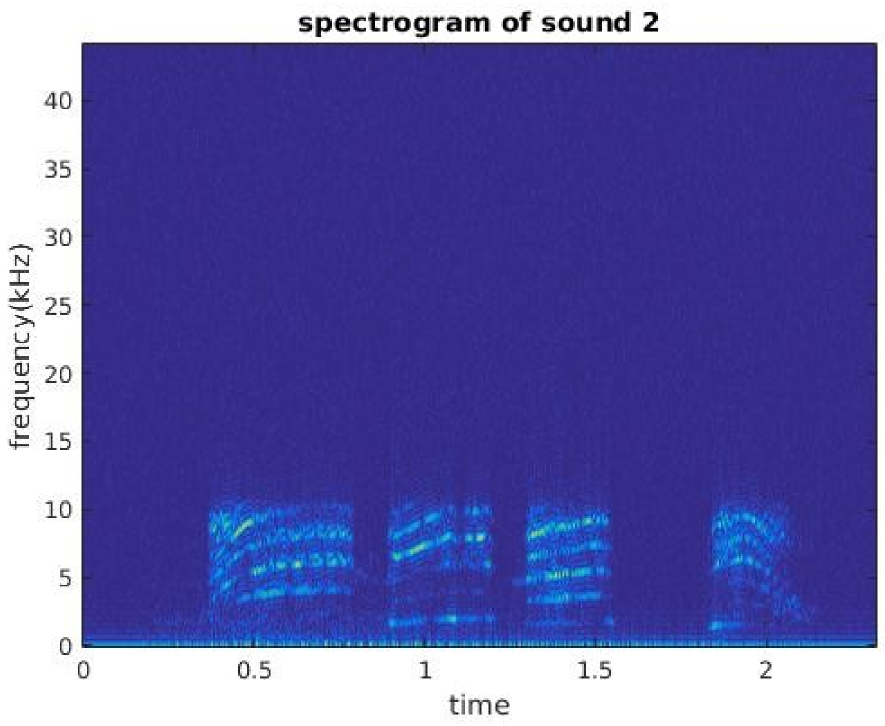
Report Item 7:
code:

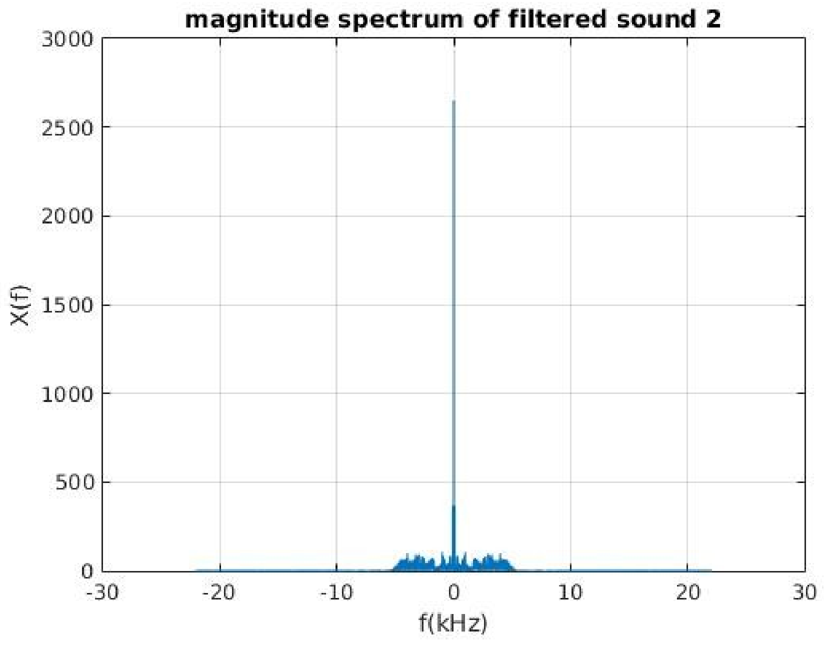
```
[Y,fs] = wavread('sound2')
N = length(Y);
time = length(Y)/fs;
Y_transform = fft(Y);
Y_transform = fftshift(Y_transform);
w = fftshift((0:N - 1)/N*2*pi);
w(1:N/2) = w(1:N/2) - 2*pi;
omega = fs*w/1000/2/pi;%kHz

plot(omega,abs(Y_transform))
title('magnitude spectrum of sound2')
xlabel('f(kHz)')
ylabel('X(f)')
grid on

M = 200
P = 2048
D = 1
[stdft,analog,shift] = mySTDFT(Y,M,D,P,fs);
figure
imagesc(shift,analog/2/pi/1000,abs(stdft));
axis('xy')
title('spectrogram of sound 2')
xlabel('time')
ylabel('frequency(kHz)')

rp = 1
a = [1 0];
fs = 44100
f = [2000,10000];
rs = 60
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)

B = fft(b,N);
newY = B'.*fft(Y);
newy = ifft(newY);
figure
plot(omega,abs(fftshift(newY)))
title('magnitude spectrum of filtered sound 2')
xlabel('f(kHz)')
ylabel('X(f)')
```

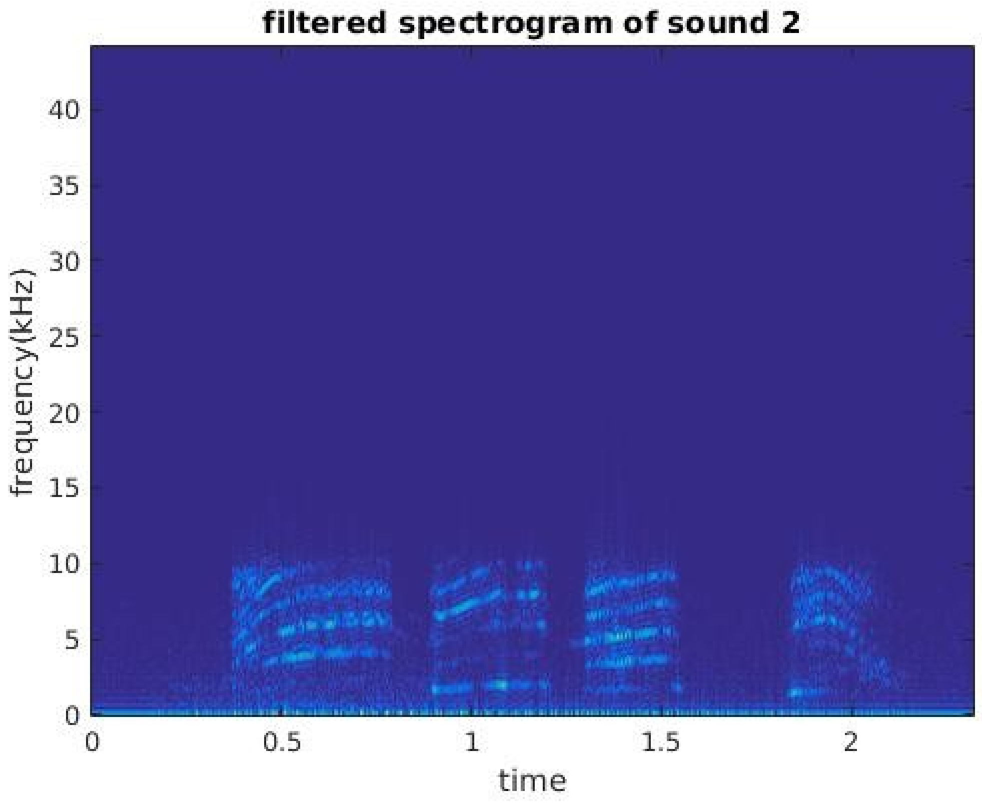
```
grid on
soundsc(newy,fs)

[stdft,analog,shift] = mySTDFT(newy,M,D,P,fs);
figure
imagesc(shift,analog/2/pi/1000,abs(stdft));
axis('xy')
title('filtered spectrogram of sound 2')
xlabel('time')
ylabel('frequency(kHz)')
```

figure:




spectrogram of sound 2
frequency(kHz)
40
35
30
25
20
15
10
5
0
0
0.5
1
1.5
2
time


magnitude spectrum of filtered sound 2
X(f)
3000
2500
2000
1500
1000
500
0
-30
-20
-10
0
10
20
30
f(kHz)



White noise is present from 0Hz t0 40KHz
The spectrum fails to capture the noise under 10kHz since it's hidden inside the desired signal and spectrum cannot show how frequency varies over time while spectrogram can do so.

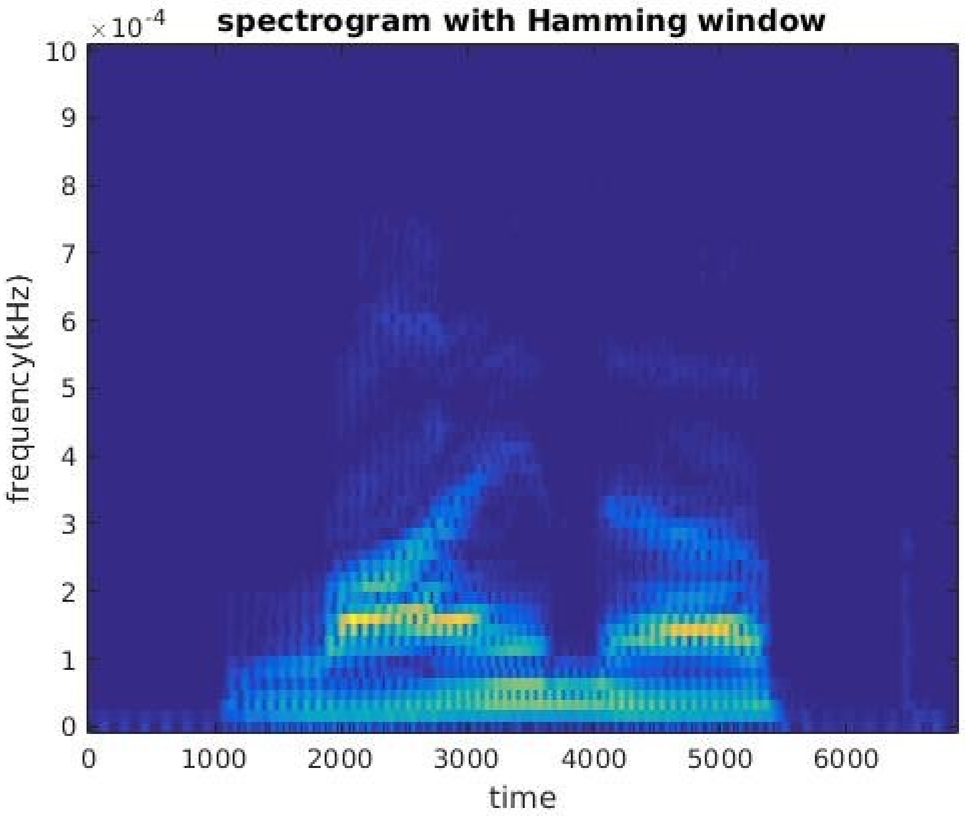
Effectiveness
Filter removes most of the signal while there is still some noise present under 10kHz

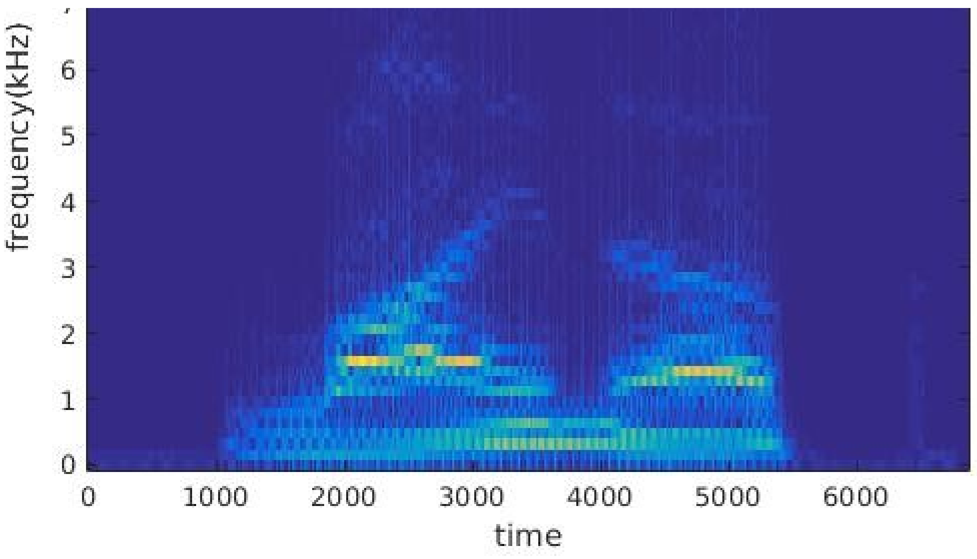
Report Item 8:
code:

```
function [stdft,Omega,shift] = mySTDFTHamming(x,M,D,P,fs)
N = length(x);
n = 0:length(x)-1;
row = ceil(P/2);
col = floor((N-M)/D)+1;
stdft = zeros(row,col);

for i = 1:col
y = x(((i-1)*D +1):(i-1)*D+M);
w = hamming(M)
Y = fft(w.*y,P);
stdft(:,i) = Y(1:row);
end
```

```
%shift

m = [0:col-1];
shift = D*m/fs;

%omega
w = [0:col-1]/col*2*pi;
Omega = fs*w;
end


load speechsig
P = 128
step = 2

M = 128
fftlength = 128
stepsize = 2
fs = 1
[stdftH,analogH,shiftH] = mySTDFTHamming(x,M,stepsize,fftlength,fs)
imagesc(shiftH,analogH/2/pi/1000,abs(stdftH))
axis('xy')
title('spectrogram with Hamming window')
xlabel('time')
ylabel('frequency(kHz)')

[stdft,analog,shift] = mySTDFT(x,M,stepsize,fftlength,fs)
figure
imagesc(shift,analog/2/pi/1000,abs(stdft))

axis('xy')
title('spectrogram with rectangular window')
xlabel('time')
ylabel('frequency(kHz)')
```

```
figure:
```





Spectrogram created by Hamming window does not show the leakage as the one created by rectangular window as Hamming window attenuates the data in the future. Also the main component around 2kHz is clearer when Hamming window is applied