THE UNIVERSITY OF MELBOURNE — MELBOURNE SCHOOL OF ENGINEERING

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Group Code (if applicable):

Assignment Title:	Matlab Project
Subject Number:	ELEN90058
Subject Name:	Signal Processing
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Lecturer/Tutor:	
Due Date:	03/OCT/2017

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1. Analysis of signal $\mathbf{r_s[n]}$

We choose the ***projsignal0.mat,*** from the table we get the $F_c = 12.288kHz$.

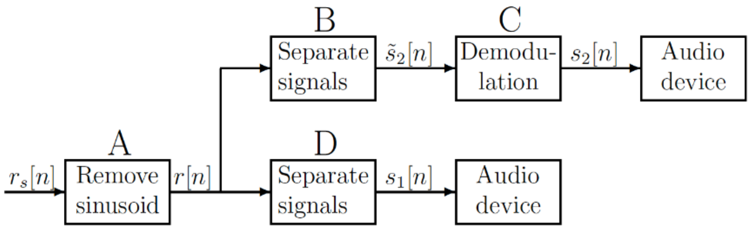


Figure1: Signal removal and separation

Consider the system in Figure 1. The signal $r_s[n]$ is made up of three components

$$r_s[n] = r_1[n] + r_2[n] + w[n]$$

$r_1[n]$ and $r_2[n]$ are two parts of the signal, $\omega[n]$ is the sinusoidal disturbance signal. $r_1[n]$ has a bandwidth of 4096Hz and $r_2[n]$ is a DSB-SC modulated signal. The carrier frequency is F_c=12.288kHz, the bandwidth of the original signal is 4096Hz.

The function of Block A is to remove the sinusoidal disturbance signal. The Block D is a lowpass filter to get the low frequency part of the original signal. And the Block B is a highpass filter to get the high frequency part of the original signal. Block C has two functions, the first is to demodulate the output signal of Block B, and then use a lowpass filter to get the low frequency part of the signal.

After loading the ***projsignal0.mat,*** it can easily be observed that the notch frequency of the disturbance is 0.3π, which is almost 4915Hz in time domain.

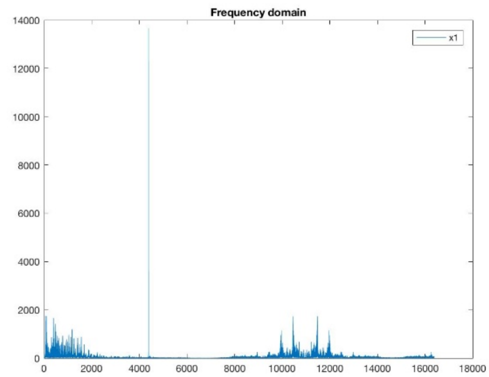


Figure 3: the original signal in frequency domain

Assuming signal $r_2[n]$ is the DSB-SC modulated signal of m(t). Then we get:

$$r_2(t) = m(t) \cdot \cos(2\pi F_c t) \leftrightarrow R_2(f) = \frac{1}{2}[M(f + F_c) + M(f - F_c)]$$

Hence, the bandwidth of $r_2(t)$ is $[F_c - 4096, F_c + 4096]$, since $F_c = 12288Hz$.
So we get the bandwidth is [8192Hz, 16348Hz].

2. FIR Filters Design

2.1 Block A

In this block, we need a FIR notch filter, using the function of the Kaiser window, we can get a FIR notch filter.

Passband: [0 4815Hz] and [5015Hz $+\infty$];
Stopband: [4914Hz 4916Hz];
Passband ripple: 0.001(for further thought about requirements of ripple)
Stopband ripple: 0.001

Using the function of *kaiserord()* to calculate the order,
the lowest order $N_1 = 1200$.

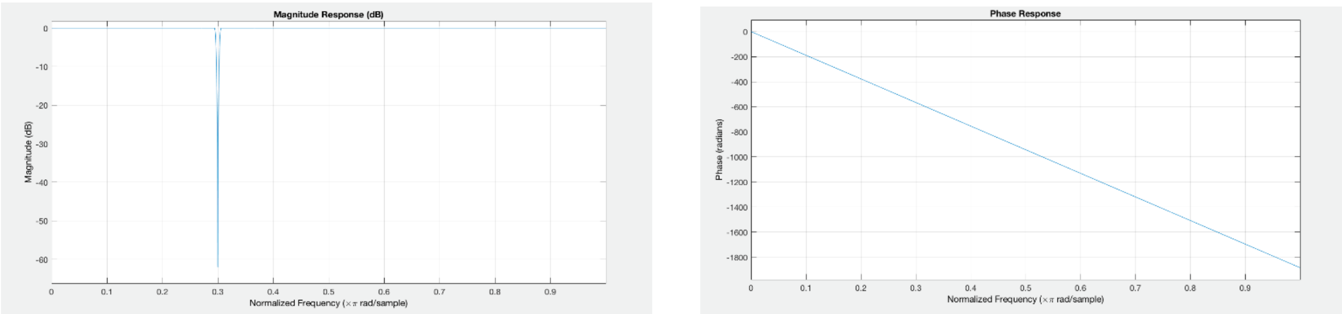


Figure 4: The magnitude and phase response of the notch filter

As figure 4 shows, the gain of the filter at 0.3pi is smaller than -60dB, which satisfy with the design requirement of the stopband, moreover, using the function *islinphase()* in Matlab, the filter has the linear phase. hence the filter satisfies the requirement of linear phase.

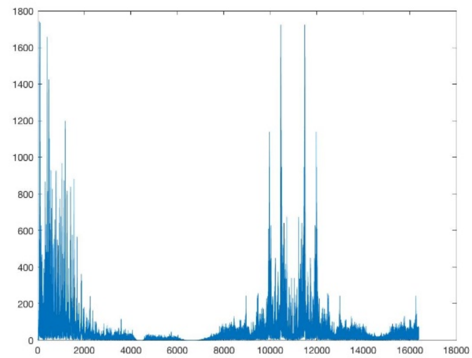


Figure 5: The output of the notch filter

2.2 Block D

Using the Kaiser Window to design a lowpass filter, the requirements are below:
Passband: [0 4096Hz];
Stopband: [4596Hz $+\infty$];
Passband ripple: 0.001
Stopband ripple: 0.001

Using the the function of *kaiserord()* to calculate the order,
the lowest order is $N_2 = 238$.

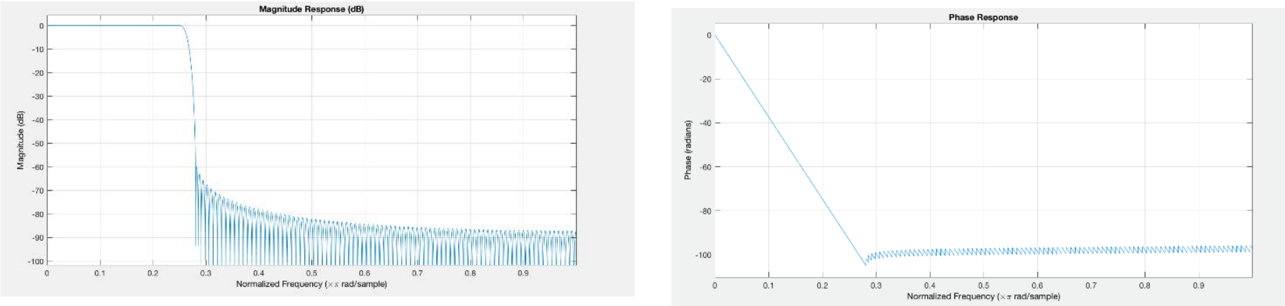


Figure 5: The magnitude and phase response of Block D lowpass filter

islinphase() was used to check if the cascaded filter has the linear phase and the filter satisfies the requirement of linear phase.

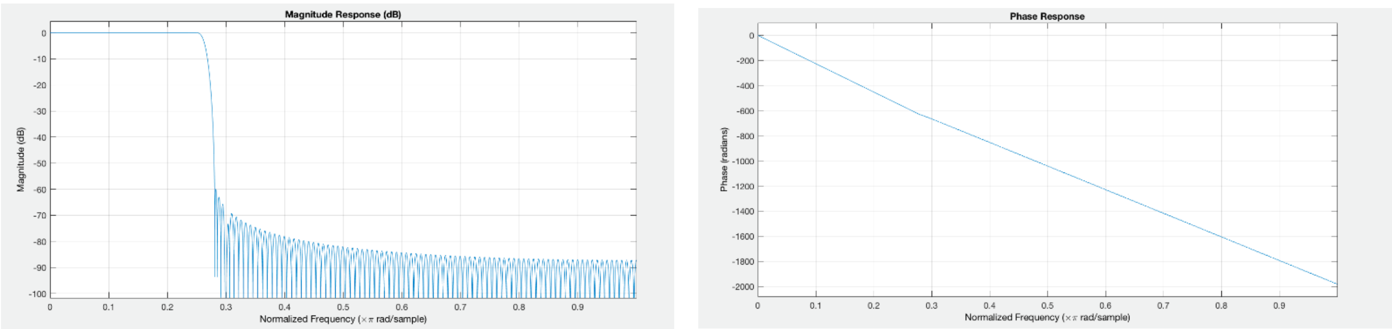
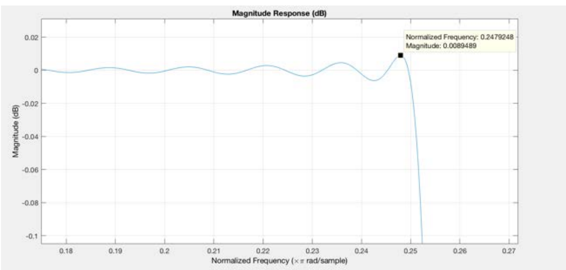


Figure 6: The magnitude and phase response of $H_1(z)$



$$20\log_{10} A = 0.0089489$$
$$\text{A} = 1.0010308111$$
$$\text{A-1} = 0.00103 < 0.01$$

satisfy design requirement.

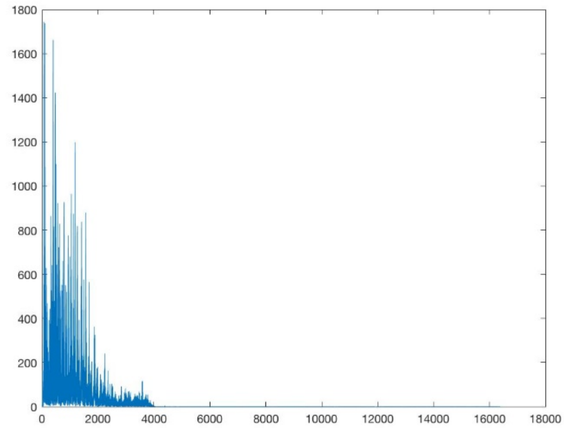


Figure 7: The output of Block D

2.3 Block B

Using the Kaiser window to design a highpass filter.
Passband: [12288Hz $+\infty$];
Stopband:[0 8192Hz];
Passband ripple: 0.001;
Stopband ripple: 0.001;

Using the the function of *kaiserord()* to calculate the order,
the lowest order is $N_3 = 30$.

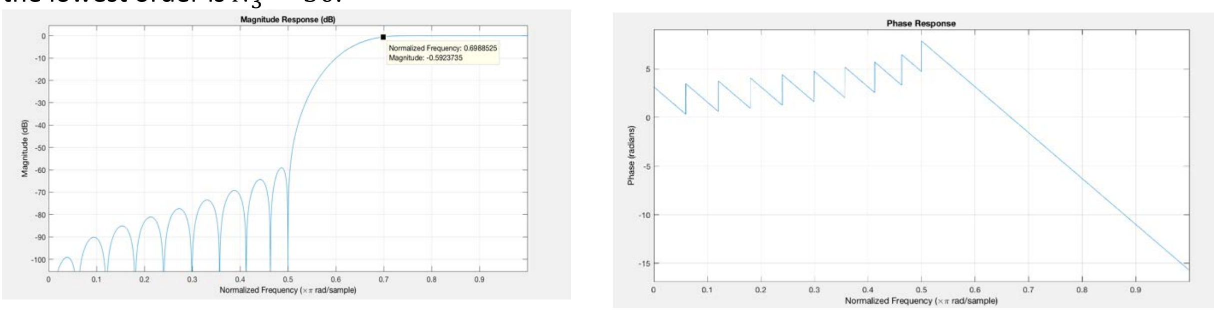


Figure 8: the magnitude and phase response of highpass filter

using the function *islinphase()* in Matlab, the filter has the linear phase. hence the filter satisfies the requirement of linear phase.

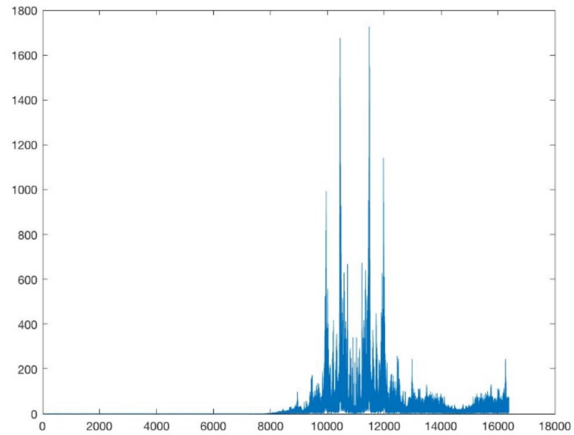
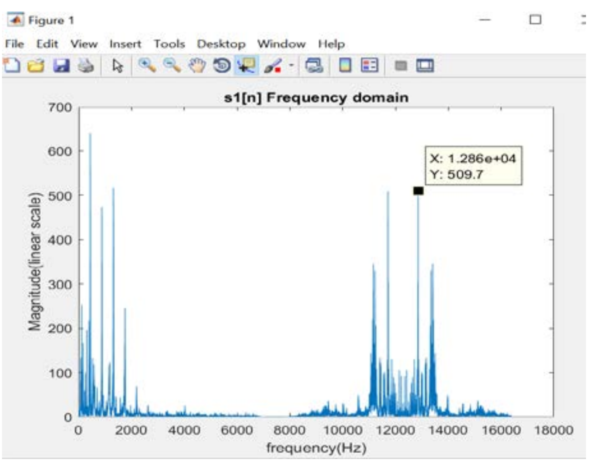
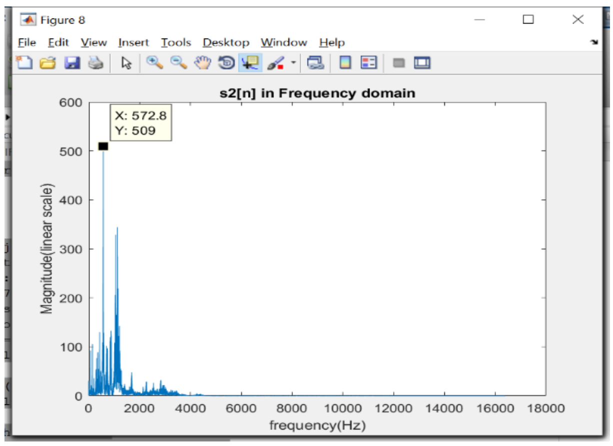


Figure 9: the output of Block B

Demodulation

A DSBSC modulated signal can be demodulated by multiplying its carrier with a phase shift. It is known that r2[n] is the modulated signal, so when it is demodulated by $\cos(2\pi F_c t + \psi)$, a signal $\frac{1}{2}[r2[n] + r2[n] * \cos(2\pi 2 * F_c t)]$ is generated. Take the DTFT of the signal, we expect a scaled version of r2[n] as well as a modulated version of r2[n] centred at 2Fc. To recover the original signal, we will use block C to filter out the spectrum above the frequency of r2[n].
In order to fully recover the signal, a phase offset is estimated by test and error from an iconic peak in the demodulated signal, we measure the magnitude of it and compare it with the same peak shifted back to the lowpass spectrum after demodulation. The phase of the demodulation signal was adjusted so that the magnitude in the demodulated signal is still the same as that in the modulated signal.





As shown in the figures above, after adjusting the phase by pi/3, we have the same magnitude response in the signal before and after modulation.

2.4 Block C

Using the Kaiser Window to design a lowpass filter

Passband: [0 4096Hz];
Stopband: [4596Hz $+\infty$];
Passband ripple: 0.001
Stopband ripple: 0.001

Using the the function of *kaiserord()* to calculate the order,
the lowest order is $N_2 = 238$.

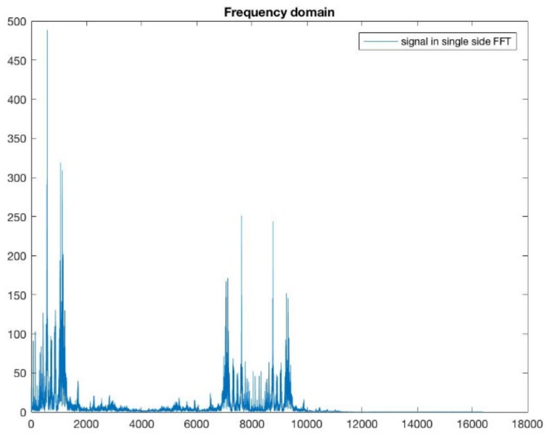


Figure 10: after demodulation

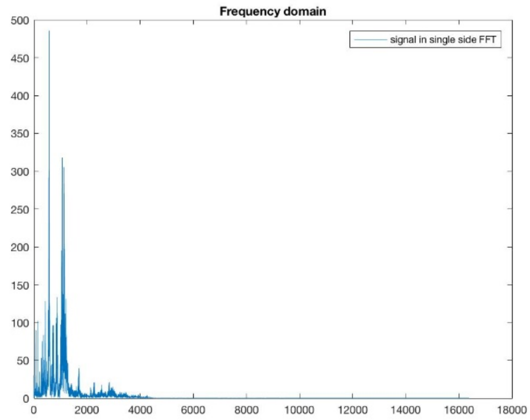


Figure 11: the output of demodulated signal after lowpass filter

Using the function *islinphase()* in Matlab to check if the filter has the linear phase. The result is logic, hence the lowpass filter in Block C satisfies the requirement of linear phase.

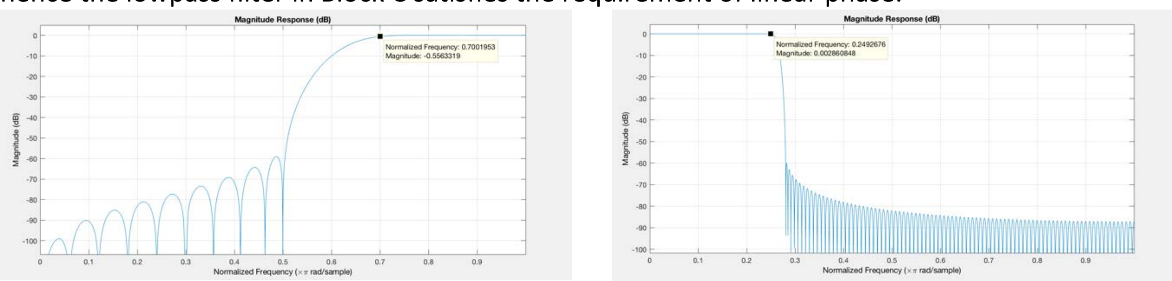


Figure 12: the response of cascade of filters A and B and the response of filter C

From the figure 12, we know

$$-20\log_{10} A_1 = -0.5563319$$
$$20\log_{10} A_2\ 0.002860848$$
$$A_1 = 1.006614, A_2 = 1$$

Hence,
$|1 - 1.006614 \times 1| = 0.6614\% < 2\%$, which means the cascading A B C filter satisfy the ripple design requirement.

3. FIR Filters Optimization

3.1 Total Group Delay

If a filter has linear phase, its frequency response can be written as

$$H\left(e^{-j\omega}\right) = e^{-j\frac{N}{2}\omega}e^{-j\beta}\breve{H}(\omega)$$

where N is the order of the filter and $\breve{H}(\omega)$ is a real function of ω.

So the Group Delay can be calculated by

$$\tau_p(\omega) = -\frac{d\theta(\omega)}{d\omega} = -\frac{d}{d\omega}\left(\beta - \frac{N}{2}\omega\right) = \frac{N}{2}$$

The total group delay for cascade $H_1(z)$

$$\frac{1200 + 238}{2} = 719$$

The total group delay for the cascade of the filters in blocks A, B and C

$$\frac{1200 + 30 + 238}{2} = 734$$

3.2 Minimise the Total Group Delay

By increasing the transition region and making full use of ripple, we optimize the filters with new specifications.

Block A

Passband: [0 4015Hz] and [5815Hz $+\infty$];
Stopband: [4915.4Hz 4915.6Hz];
Passband ripple: 0.001
Stopband ripple: 0.001
Now the order of the notch filter is 134.

Block D and Block C

Passband: [0 4096Hz];
Stopband: [5415Hz $+\infty$];
Passband ripple: 0.01
Stopband ripple: 0.01
Now the order of the lowpass filter is 60.
From the magnitude response we can know the maximum ripple of $H_1(z)$ is

$$20\log_{10} A = 0.0648808$$

A = 1.0074976
A-1=0.0074976<0.01, satisfy design requirement.

Block B

Passband: [8192Hz $+\infty$];
Stopband:[0 4096Hz];
Passband ripple: 0.001;
Stopband ripple: 0.001;
Now the order of the highpass filter is 30.

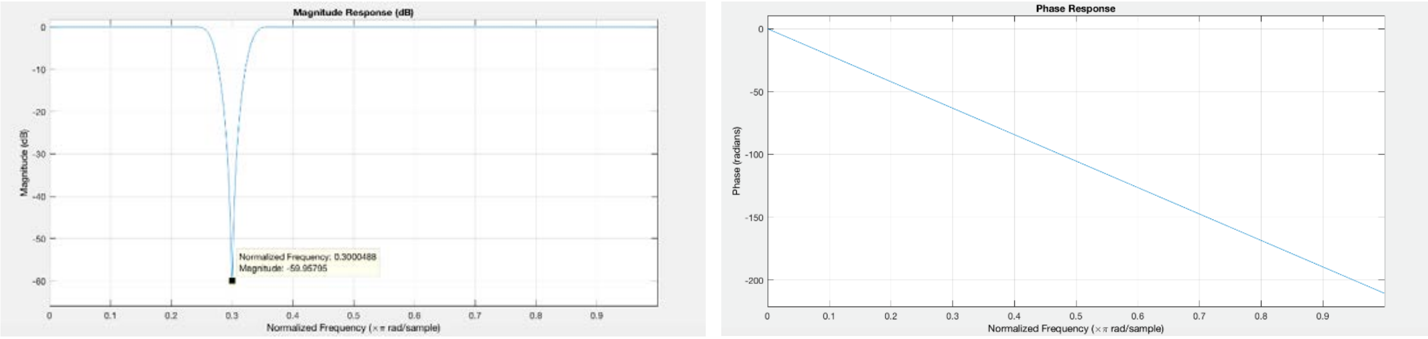


Figure 13: the new notch filter bode diagram

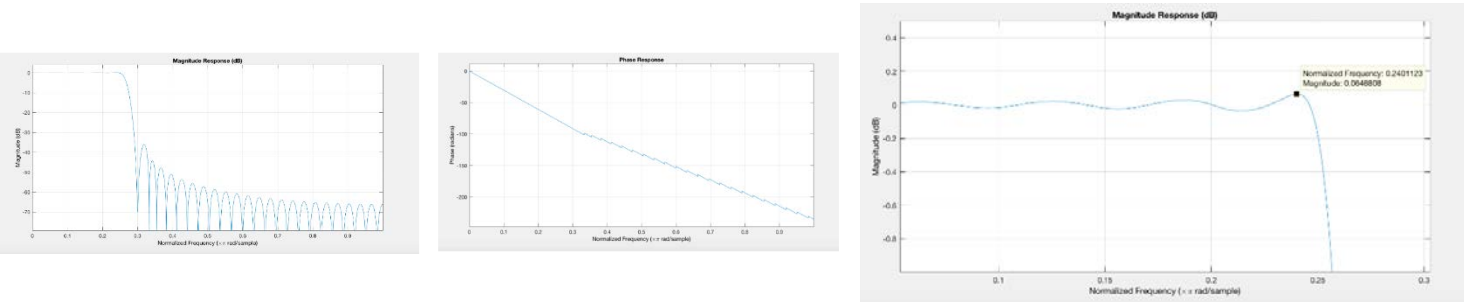


Figure 14: the response of $H_1(z)$

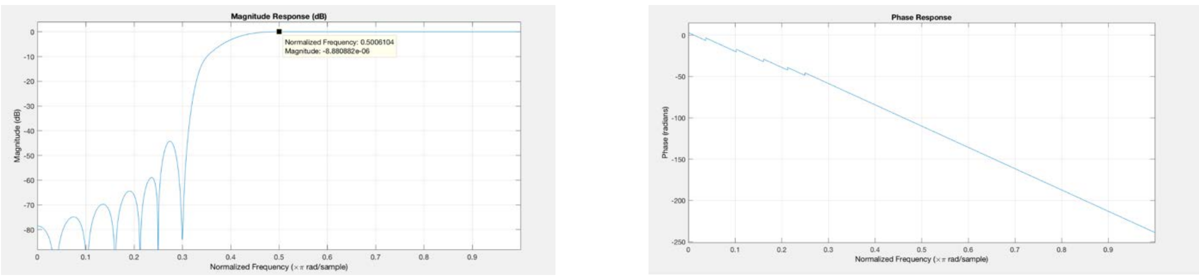


Figure 15: the response of cascade of filters A and B

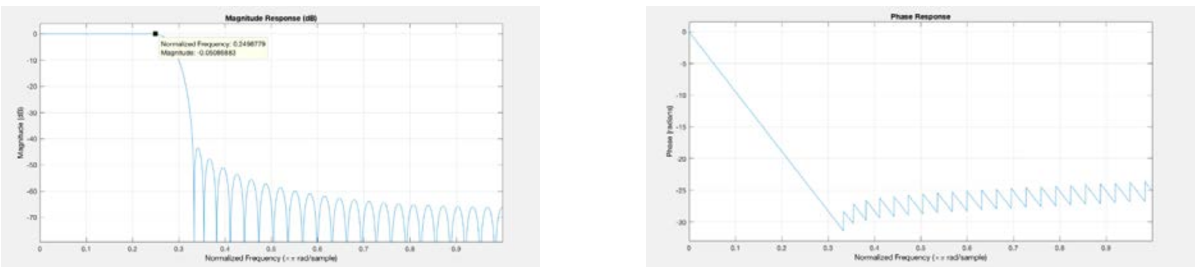


Figure 16: the response of filter in Block C

From the figure 13 it can be observed that the gain at the notch frequency 0.3π is less than -60dB.

From the figure 14 we can know the maximum ripple of $H_1(z)$ is

$$20\log_{10} A = 0.0648808$$

A = 1.0074976
A-1=0.0074976<0.01

From the figure 15 and 16 we can know the maximum ripple of filters A, B and C

$$-20\log_{10} A_1 = -8.88 \times 10^{-6}$$
$$-20\log_{10} A_2 = -0.005086883$$
$$A_1 = 1, A_2 = 1.00587$$

Hence,

$$|1 - 1 \times 1.00587| = 0.587\% < 2\%$$

The new filters satisfy all the requirements.

The total group delay after minimizing,

For $H_1(z)$

$$\frac{134 + 60}{2} = 97$$

For cascade of the filters in blocks A, B and C

$$\frac{134 + 60 + 30}{2} = 112$$

3. IIR filter Design

Block A: Second order Notch filter

Central frequency $\omega_w = 2\pi * \frac{4915}{fs} = 0.3\pi$
Filter parameters

$$\beta = \cos(\omega_w) = 0.58778$$
$$bandwidth = \cos^{-1}\left(\frac{2\alpha}{1 + \alpha^2}\right)$$

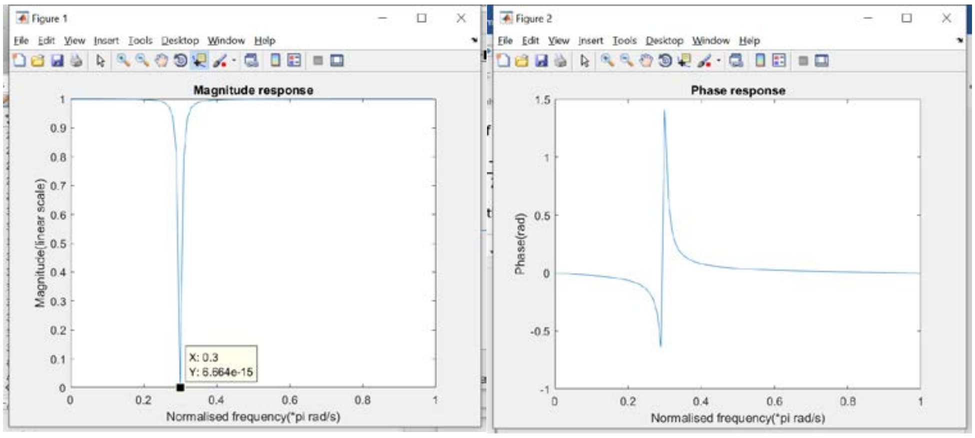
By trial and error in Matlab, we found that when the bandwidth = 0.015 π, the gain of the filter and the distortion of the filter is the most suitable for our application, further analysis will be done in the later part of this report about this.

Therefore $\alpha = \frac{1}{\cos(bandwidth)} - \sqrt{\frac{1}{\cos(bandwidth)^2 - 1}} = 0.9540.$

The notch filter has transfer function of

$$H_A(z) = \frac{1 + \alpha}{2} * \frac{1 - 2\beta z^{-1} + z^{-2}}{1 - \beta * (1 + \alpha)z^{-1} + \alpha z^{-2}}$$

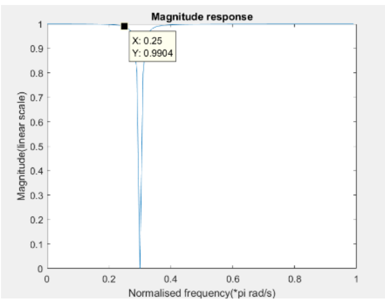
The magnitude and phase response of the filter has been plotted below



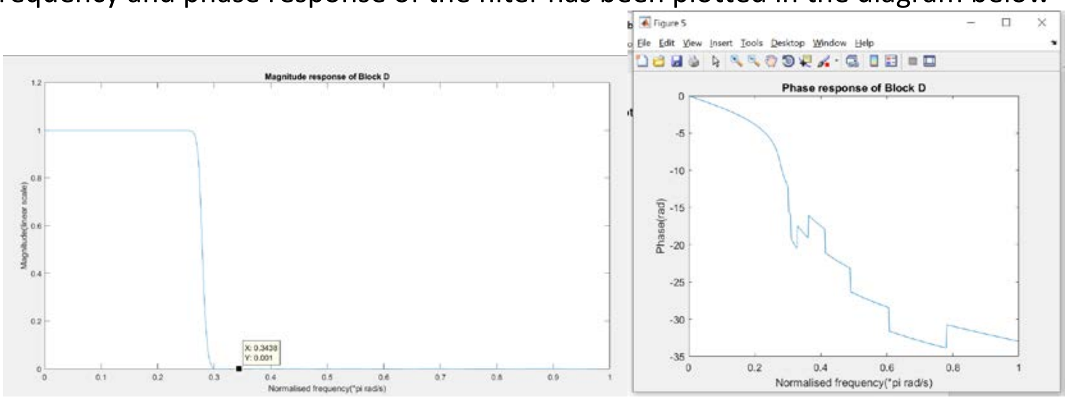
As the magnitude response shows, the gain of the filter at ω_W is 20log(6.664e-15) = -283.52dB << -60dB, therefore this filter will size up the design requirement.
The phase response of the block A filter is not linear, which is different from FIR filters.

Block D: LPF
Passband: [0, 4096]
Stopband: (4096, inf)
Passband ripple: <4E-4
Stopband ripple: 0.001

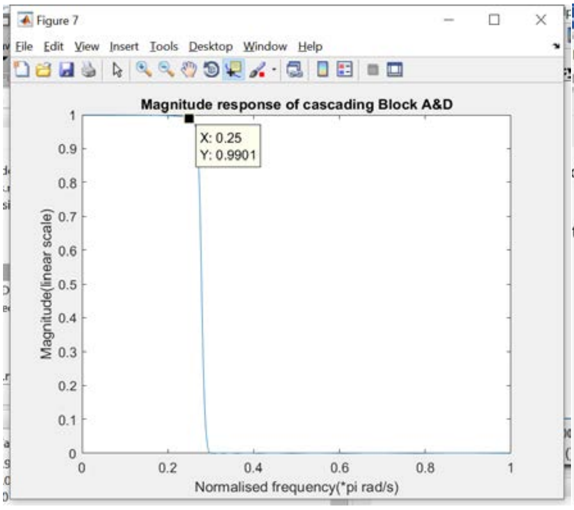
Ripple and filter type analysis

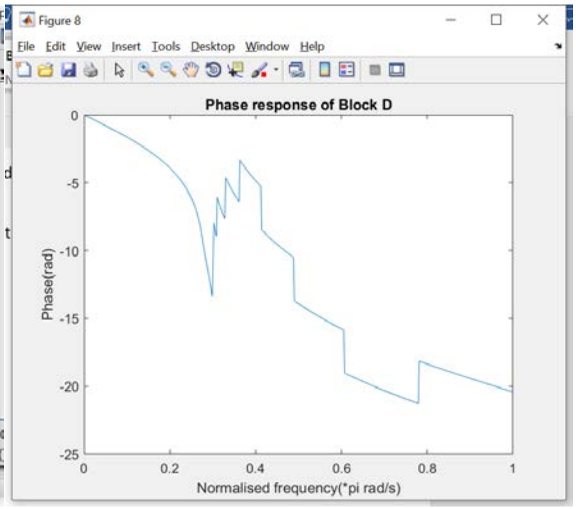


Ripple in passband, cascading with Block A: <0.01
It was known from the magnitude response of the filter A that it will introduce a ripple at the passband edge ($2\pi * \frac{4096}{fs}$ = 0.25 π) of filter D, and the ripple has been estimated to be 1-0.9904 = 9.6E-3. Therefore, $1-|H_A(Z)| * |H_D(Z)| = 1-(0.9904) * (1-\mathrm{rp_D}) < 0.01$, rp_D<0.000403877, rp_D = 4E-4 was set to satisfy this condition. So, it's preferably to have a flat passband from the low pass filter in this case. Therefore, a **Chebyshev Type 2 filter** has been implemented.
The frequency and phase response of the filter has been plotted in the diagram below



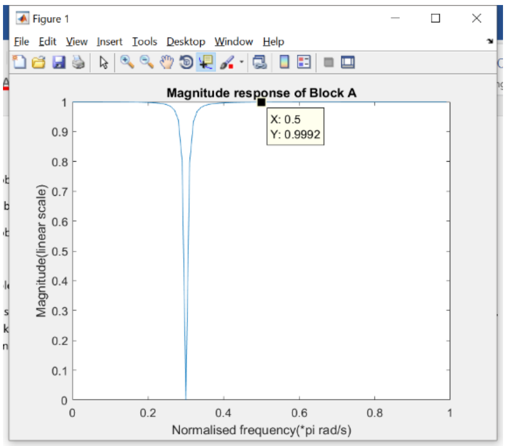
As the magnitude plot shows, the filter has a flat passband and stopband with ripple of 0.001, as we designed.
By cascading A and D together, we have phase and magnitude response as shown below





As analysed before, the passband ripple of filter A+D is 1-0.9901 = 9.9E-3 < 1%, therefore, the design of A and D satisfy the requirements.

Block B: HPF
Passband: [8192, inf), because the modulated signal has been shifted and centred at fc = 12288Hz, and the lower bound of this signal is 12288-bandwidth = 12288-4096 = 8192.
Stopband: [0, 8192)
Passband ripple: 4.5E-3
Stopband ripple: 0.001



Ripple and filter type analysis

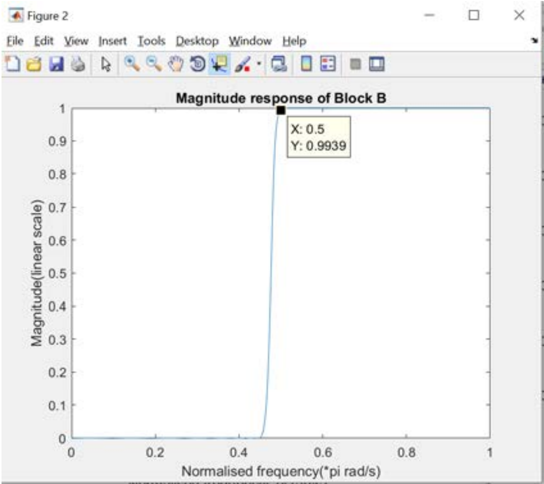
It has been required that cascading A B and C should have ripple less than 2% of the original signal, and knowing that the gain of block A at the passband edge of block B, f = 8192, i.e, 0.5 π at normalised angular velocity, is 0.9992. Therefore, the ripple allowance for cascading B and C will be calculated as below.

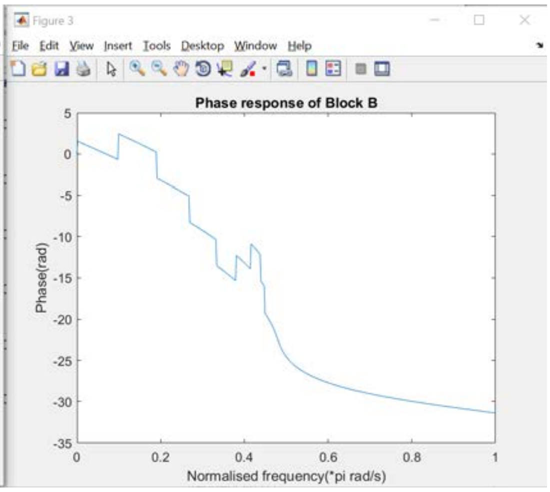
$$1-|H_A(Z)|*|H_B(Z)|*|H_C(Z)| = 1-(0.9992)*(1-\mathrm{rp_B})*(1-\mathrm{rp_C}) < 0.02$$

For the convenience of calculation, we set $\mathrm{rp_B} = \mathrm{rp_C}$, therefore, $\mathrm{rp_B} = \mathrm{rp_C} < 9.65E-3$

Therefore, the passband ripple for Block B is 9E-3. Since a flat response in the passband is always the best option to minimise the ripple, therefore we still choose **Chebyshev Type 2 filter** to implement Block B.

The magnitude and phase response of the filter has been plotted as below.

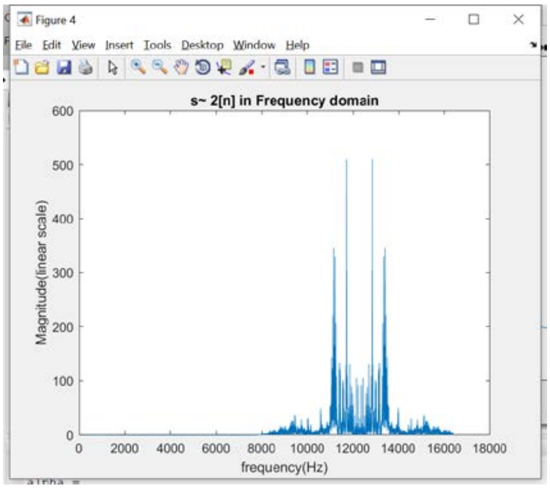




As expected, the stopband has ripple of 0.001 and the magnitude response for the passband is flat.

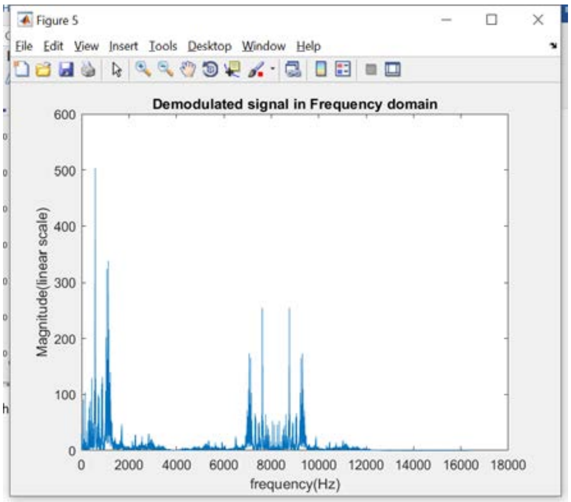
Further analysis on designing Block C
As the magnitude response show, the ripple for this filter at 0.5pi (passband edge) has gain of 0.9939, that means the ripple 1 - 0.9939 = 6.1E-3, recall the magnitude at the 0.5pi for block A is 0.9992, This means the ripple allowance for cascading block C is 1-0.9992*0.9939*(1-rp_C)<0.02.
Rp_C<0.01319. This is a sanity check for the result of Rp_C<9E-3, it means that the performance of block B is better than expected.

Continue with the verification of block B, the signal after Block B HPF is shown as below

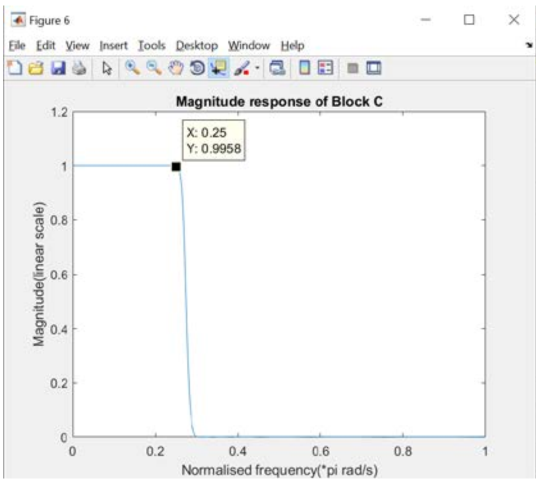


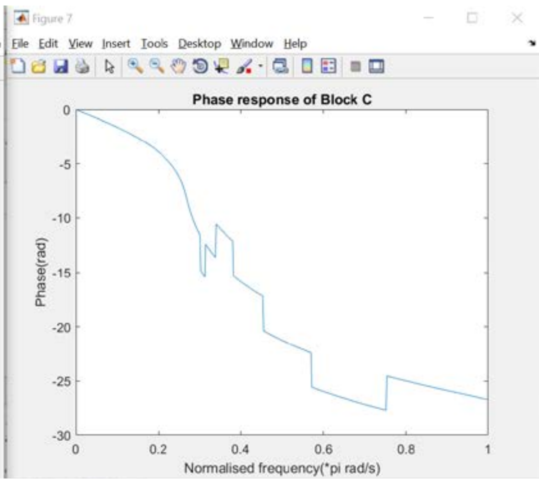
This is the modulated signal bandpass signal.

Using the same demodulation method as discussed in FIR filter design, we have the signal as below show

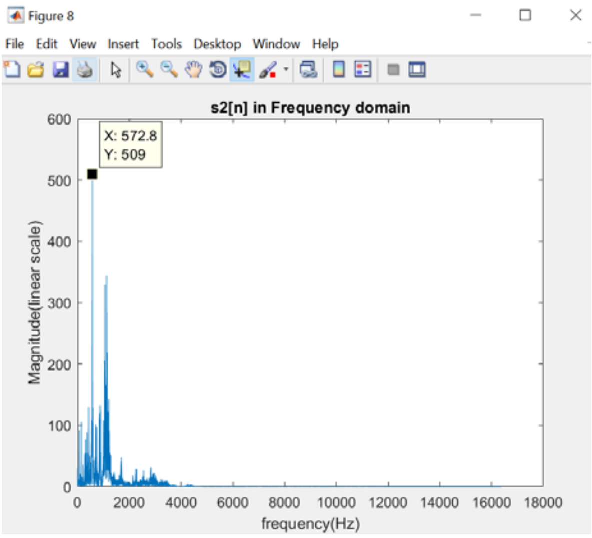


Block C: LPF
Passband: [0, 4096),
Stopband: (4096, inf)
Passband ripple: 9E-3, as discussed previously
Stopband ripple: 0.001
As the diagram below is the magnitude and phase response of the block C lowpass filter





The total ripple from block A B and C can be calculated by 1-0.992*0.9939*0.9958 = 0.0182<0.02, therefore the design satisfies the requirements.



As expected, the signal below is the Signal s2[n] after Block C,

The phase offset has been set to +pi/3 to obtain the original magnitude, by test and error.

2b. Minimise the filter orders

Its known that Elliptic filter has the narrowest transition band as well as the lowest required order among different type of filters.

Therefore, we replace the Chebyshev type 2 filters in block B, C and D to complete this task.

Recall the design requirement for Block B, C and D are shown below

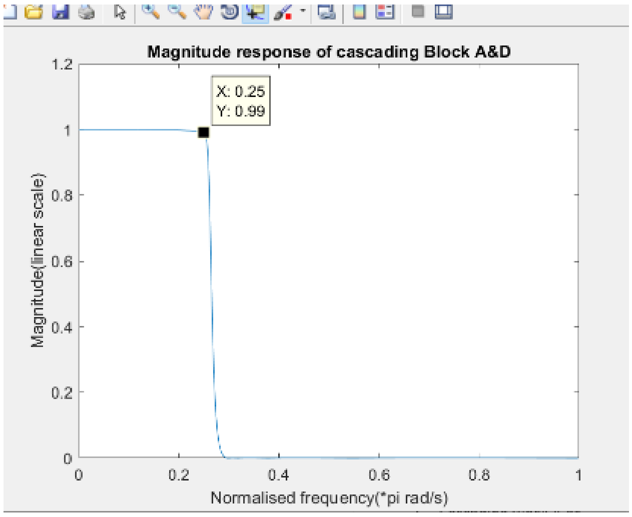
	Block B	**Block C**	**Block D**
Passband	[8192, inf)	[0, 4096),	[0, 4096]
Stopband	[0, 8192)	(4096, inf)	(4096, inf)
Passband ripple	9E-3	9E-3	<4E-4
Stopband ripple	0.001	0.001	0.001
Filter Type	Chebyshev Type 2	Chebyshev Type 2	Chebyshev Type 2
Minimum Order	17	15	17

Optimised order design

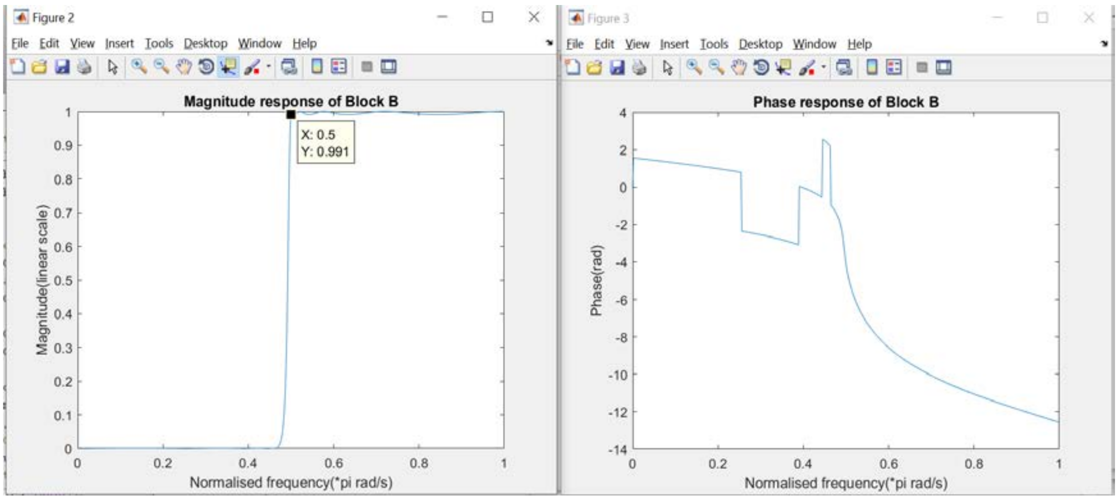
	Block B	**Block C**	**Block D**
Filter Type	Elliptic	Elliptic	Elliptic
Minimum Order	9	8	9

Requirement check

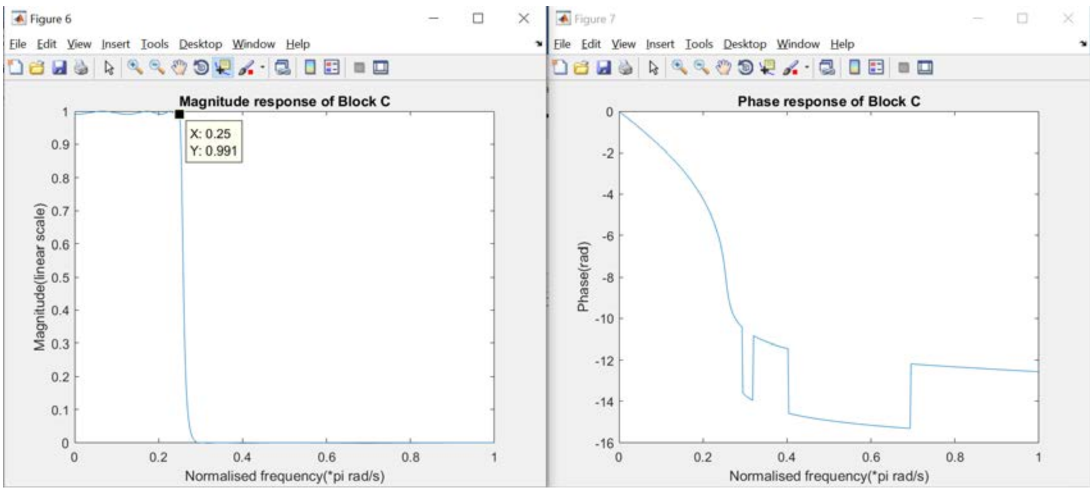
1. The filters are implemented to have minimum order
2. Cascading Block A and D has ripple of 1-0.99 = 0.1 at 0.25pi(the edge of passband)
3. The ripple of Block B has been estimated to be (1-0.991), and that in C is (1-0.991), therefore the total ripple of cascading A B and C is $1 - |H_A(Z)| * |H_B(Z)| * |H_C(Z)| = 1 - 0.9992 * 0.991^2 = 0.018 < 0.02$



Block B magnitude and phase response



Block C magnitude and phase response



Appendix

FIR fliters

```
clear;
close all;
load projsignal0.mat;


rs = rs(1:25E3);%only take the first 25000 data point
fvtool(rs);

fsamp = 32768;
% block A notch filter
Bw1 = 4096;
fcuts = [4015 4915.4 4915.6 5815];
mags = [1 0 1];
devs = [0.001 0.001 0.001];

[n,Wn,beta,ftype] = kaiserord(fcuts,mags,devs,fsamp);
n = n + rem(n,2);
hbs = fir1(n,Wn,ftype,kaiser(n+1,beta),'noscale');

Ybs = filter(hbs,1,rs);

fvtool(Ybs);

%block D lowpass filter

fcuts = [4096 5415];
mags = [1 0];
devs = [0.01 0.008];
[n,Wn,beta,ftype] = kaiserord(fcuts,mags,devs,fsamp);
n = n + rem(n,2);
hlp = fir1(n,Wn,ftype,kaiser(n+1,beta),'noscale');
Ylp = filter(hlp,1,Ybs);
fvtool(Ylp);



%Ylp is the output signal of block D, s1[n];

%block B highpass filter
fcuts = [4096 8192];
mags = [0 1];
devs = [0.01 0.001];
[n,Wn,beta,ftype] = kaiserord(fcuts,mags,devs,fsamp);
n = n + rem(n,2);
hhp = fir1(n,Wn,ftype,kaiser(n+1,beta),'noscale');

Yhp = filter(hhp,1,Ybs);
```

```
fvtool(Yhp);

% Yhp is the output signal of block B, use Yhp to demodulation

fs1 = fsamp;
t = (0:1:24999).* (1/fs1);
demod = zeros(1, 25000);
for i = 1:1:25000
   demod(i) = Yhp(i).*cos(2*pi*12288*t(i)+ 0.3*pi );
end

figure;

N=25000;
X1_mags = abs(fft(demod));
fax_bins = [0 : N-1]; %frequency axis in bins
N_2 = ceil(N/2);
plot(fax_bins(1:N_2)*fs1/N, X1_mags(1:N_2));
legend('signal in single side FFT');
title('Frequency domain');


%block C lowpass filter

fcuts = [4096 4596];
mags = [1 0];
devs = [0.001 0.001];
[n,Wn,beta,ftype] = kaiserord(fcuts,mags,devs,fsamp);
n = n + rem(n,2);
hlp = fir1(n,Wn,ftype,kaiser(n+1,beta),'noscale');
Ylp_1 = filter(hlp,1,demod);

figure;

N=25000;
X1_mags = abs(fft(Ylp_1));
fax_bins = [0 : N-1]; %frequency axis in bins
N_2 = ceil(N/2);
plot(fax_bins(1:N_2)*fs1/N, X1_mags(1:N_2));
legend('signal in single side FFT');
title('Frequency domain');
```

IIR A AND D

```
%second order notch filter, block A
clc
clear all
close all

load('projsignal0.mat');
%this is the input
rs = rs(1:25E3);%only take the first 25000 data point
fs1 = 32.768e3;
beta = cos(2*pi*(4915.2/fs1));
%beta = cos(2*pi*(7372.8/fs1));
notch_bw = 0.015*pi;%changed to a wider bandwidth
alpha = (1/cos(notch_bw)) - sqrt((1)/((cos(notch_bw))^2)-1)

num_A  = ((1+alpha)/2).*[1 -2*beta 1];
dem_A = [1 -beta*(1+alpha) alpha];

%filter characteristic
[hA, wA] = freqz(num_A, dem_A, 1E2);


%magnitude plot
plot(wA/pi, abs(hA));
title('Magnitude response of Block A');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Magnitude(linear scale)');


%plot phase
figure;
plot(wA/pi, phase(hA));
title('Phase response of Block A');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Phase(rad)');

%now
r1r2 = filter(num_A, dem_A, rs);

figure;
N=25000;
X1_mags = abs(fft(r1r2));
fax_bins = [0 : N-1]; %frequency axis in bins
N_2 = ceil(N/2);
plot(fax_bins(1:N_2)*fs1/N, X1_mags(1:N_2));
title('Single Side FFT of r[n](r1r2)');
xlabel('frequency(Hz)');
ylabel('Magnitude(linear scale)');


%BLOCK D implementaion
```

```
figure;
%the only requirement here is the passband of r1
dp=4E-4; ds=0.001;
%peak passband ripple and minimum stopband attenuation in dB
ap=-20*log10(1-dp); as=-20*log10(ds);
wp = 2* 4096/fs1;
ws = wp +0.05;%right after passband

[ordD, ~] = cheb2ord(wp, ws, ap, as);
[num_D, dem_D] = cheby2(ordD, as, ws, 'low');
[hD, wD] = freqz(num_D, dem_D);
%MAG
plot(wD/pi, abs(hD));
title('Magnitude response of Block D');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Magnitude(linear scale)');
%PHASE

figure;
plot(wD/pi, phase(hD));
title('Phase response of Block D');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Phase(rad)');


r1 = filter(num_D, dem_D, r1r2);

figure;
N=25000;
X1_mags = abs(fft(r1));
fax_bins = [0 : N-1]; %frequency axis in bins
N_2 = ceil(N/2);
plot(fax_bins(1:N_2)*fs1/N, X1_mags(1:N_2));
title('s1[n] Frequency domain');
xlabel('frequency(Hz)');
ylabel('Magnitude(linear scale)');
sound(r1, fs1);


%H1 analysis
figure;
num_H1 = conv(num_A, num_D);
dem_H1 = conv(dem_A, dem_D);
[hH1, wH1] = freqz(num_H1, dem_H1);
%MAG
plot(wH1/pi, abs(hH1));
title('Magnitude response of cascading Block A&D');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Magnitude(linear scale)');
%PHASE
figure;
```

```
plot(wH1/pi, phase(hH1));
title('Phase response of Block D');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Phase(rad)');

fprintf('The order of Block D is %d\n', ordD);
```

iir abc

```
%second order notch filter, block A
clc
clear all
close all

load('projsignal0.mat');
%this is the input
rs = rs(1:25E3);%only take the first 25000 data point
fs1 = 32.768e3;
beta = cos(2*pi*(4915.2/fs1));
%beta = cos(2*pi*(7372.8/fs1));
notch_bw = 0.015*pi;%changed to a wider bandwidth
alpha = (1/cos(notch_bw)) - sqrt((1)/((cos(notch_bw))^2)-1)

num_A  = ((1+alpha)/2).*[1 -2*beta 1];
dem_A = [1 -beta*(1+alpha) alpha];

%filter characteristic
[hA, wA] = freqz(num_A, dem_A, 1E2);

%now
r1r2 = filter(num_A, dem_A, rs);

figure;
N=25000;
X1_mags = abs(fft(r1r2));
fax_bins = [0 : N-1]; %frequency axis in bins
N_2 = ceil(N/2);
plot(fax_bins(1:N_2)*fs1/N, X1_mags(1:N_2));
title('s1[n] Frequency domain');
xlabel('frequency(Hz)');
ylabel('Magnitude(linear scale)');



figure;
%%%highpass filter, block B
dp_B=9E-3; ds_B=0.001;
%peak passband ripple and minimum stopband attenuation in dB
ap_B=-20*log10(1-dp_B); as_B=-20*log10(ds_B);
```

```
ws_B = (2* 8192/fs1)-0.05;
wp_B = 2* 8192/fs1;%passband

[ordB, ~] = cheb2ord(wp_B, ws_B, ap_B, as_B);
[num_B, dem_B] = cheby2(ordB, as_B, ws_B, 'high');
[hB, wB] = freqz(num_B, dem_B);

%MAG
plot(wB/pi, abs(hB));
title('Magnitude response of Block B');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Magnitude(linear scale)');

%PHASE
figure;
plot(wB/pi, phase(hB));
title('Phase response of Block B');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Phase(rad)');



%apply the filter
r2 = filter(num_B, dem_B, r1r2);


figure;
N=25000;
X1_mags = abs(fft(r2));
fax_bins = [0 : N-1]; %frequency axis in bins
N_2 = ceil(N/2);
plot(fax_bins(1:N_2)*fs1/N, X1_mags(1:N_2));
title('s~ 2[n] in Frequency domain');
xlabel('frequency(Hz)');
ylabel('Magnitude(linear scale)');


%demodulation

t = (0:1:24999).* (1/fs1);
demod = zeros(1, 25000);
for i = 1:1:25000
   demod(i) = r2(i).*cos(2*pi*12288*t(i)+ pi/3);
end

figure;
N=25000;
X1_mags = abs(fft(demod));
fax_bins = [0 : N-1]; %frequency axis in bins
N_2 = ceil(N/2);
plot(fax_bins(1:N_2)*fs1/N, X1_mags(1:N_2));
```

```
title('Demodulated signal in Frequency domain');
xlabel('frequency(Hz)');
ylabel('Magnitude(linear scale)');


%the only requirement here is the passband of r1
dp_C=9E-3; ds_C=0.001;
%peak passband ripple and minimum stopband attenuation in dB
ap_C=-20*log10(1-dp_C); as_C=-20*log10(ds_C);

wp_C = 2* 4096/fs1;
ws_C = wp_C +0.05;%right after passband

[ordC, ~] = cheb2ord(wp_C, ws_C, ap_C, as_C);
[num_C, dem_C] = cheby2(ordC, as_C, ws_C, 'low');
[hC, wC] = freqz(num_C, dem_C);
%MAG
figure;
plot(wC/pi, abs(hC));
title('Magnitude response of Block C');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Magnitude(linear scale)');

%PHASE
figure;
plot(wC/pi, phase(hC));
title('Phase response of Block C');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Phase(rad)');



%DEMOD AND LPFILTERED
r1 = filter(num_C, dem_C, demod);



figure;
N=25000;
X1_mags = abs(fft(r1));
fax_bins = [0 : N-1]; %frequency axis in bins
N_2 = ceil(N/2);
plot(fax_bins(1:N_2)*fs1/N, X1_mags(1:N_2));
title('s2[n] in Frequency domain');
xlabel('frequency(Hz)');
ylabel('Magnitude(linear scale)');

sound(r1, fs1);

fprintf('The order of Block B is %d\n', ordB);
fprintf('The order of Block C is %d\n', ordC);
```

IIR optimised AD

```
%second order notch filter, block A
clc
clear all
close all

load('projsignal0.mat');
%this is the input
rs = rs(1:25E3);%only take the first 25000 data point
fs1 = 32.768e3;
beta = cos(2*pi*(4915.2/fs1));
%beta = cos(2*pi*(7372.8/fs1));
notch_bw = 0.015*pi;%changed to a wider bandwidth
alpha = (1/cos(notch_bw)) - sqrt((1)/((cos(notch_bw))^2)-1)

num_A  = ((1+alpha)/2).*[1 -2*beta 1];
dem_A = [1 -beta*(1+alpha) alpha];

%filter characteristic
[hA, wA] = freqz(num_A, dem_A, 1E2);


%magnitude plot
plot(wA/pi, abs(hA));
title('Magnitude response of Block A');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Magnitude(linear scale)');


%plot phase
figure;
plot(wA/pi, phase(hA));
title('Phase response of Block A');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Phase(rad)');

%now
r1r2 = filter(num_A, dem_A, rs);

figure;
N=25000;
X1_mags = abs(fft(r1r2));
fax_bins = [0 : N-1]; %frequency axis in bins
N_2 = ceil(N/2);
plot(fax_bins(1:N_2)*fs1/N, X1_mags(1:N_2));
title('Single Side FFT of r[n](r1r2)');
xlabel('frequency(Hz)');
ylabel('Magnitude(linear scale)');
```

```
%BLOCK D implementaion
figure;
%the only requirement here is the passband of r1
dp=4E-4; ds=0.001;
%peak passband ripple and minimum stopband attenuation in dB
ap=-20*log10(1-dp); as=-20*log10(ds);
wp = 2* 4096/fs1;
ws = wp +0.05;%right after passband

[ordD, ~] = ellipord(wp, ws, ap, as);
[num_D, dem_D] = ellip(ordD, ap, as, wp,  'low');
[hD, wD] = freqz(num_D, dem_D);
%MAG
plot(wD/pi, abs(hD));
title('Magnitude response of Block D');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Magnitude(linear scale)');
%PHASE

figure;
plot(wD/pi, phase(hD));
title('Phase response of Block D');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Phase(rad)');


r1 = filter(num_D, dem_D, r1r2);

figure;
N=25000;
X1_mags = abs(fft(r1));
fax_bins = [0 : N-1]; %frequency axis in bins
N_2 = ceil(N/2);
plot(fax_bins(1:N_2)*fs1/N, X1_mags(1:N_2));
title('s1[n] Frequency domain');
xlabel('frequency(Hz)');
ylabel('Magnitude(linear scale)');
sound(r1, fs1);


%H1 analysis
figure;
num_H1 = conv(num_A, num_D);
dem_H1 = conv(dem_A, dem_D);
[hH1, wH1] = freqz(num_H1, dem_H1);
%MAG
plot(wH1/pi, abs(hH1));
title('Magnitude response of cascading Block A&D');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Magnitude(linear scale)');
```

```
%PHASE
figure;
plot(wH1/pi, phase(hH1));
title('Phase response of Block D');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Phase(rad)');

fprintf('The order of Block D is %d\n', ordD);
```

IIR optimised ABC

```
%second order notch filter, block A
clc
clear all
close all

load('projsignal0.mat');
%this is the input
rs = rs(1:25E3);%only take the first 25000 data point
fs1 = 32.768e3;
beta = cos(2*pi*(4915.2/fs1));
%beta = cos(2*pi*(7372.8/fs1));
notch_bw = 0.015*pi;%changed to a wider bandwidth
alpha = (1/cos(notch_bw)) - sqrt((1)/((cos(notch_bw))^2)-1)

num_A  = ((1+alpha)/2).*[1 -2*beta 1];
dem_A = [1 -beta*(1+alpha) alpha];

%filter characteristic
[hA, wA] = freqz(num_A, dem_A, 1E2);

%now
r1r2 = filter(num_A, dem_A, rs);

figure;
N=25000;
X1_mags = abs(fft(r1r2));
fax_bins = [0 : N-1]; %frequency axis in bins
N_2 = ceil(N/2);
plot(fax_bins(1:N_2)*fs1/N, X1_mags(1:N_2));
title('s1[n] Frequency domain');
xlabel('frequency(Hz)');
ylabel('Magnitude(linear scale)');



figure;
%%%highpass filter, block B
dp_B=9E-3; ds_B=0.001;
%peak passband ripple and minimum stopband attenuation in dB
ap_B=-20*log10(1-dp_B); as_B=-20*log10(ds_B);
```

```
ws_B = (2* 8192/fs1)-0.05;
wp_B = 2* 8192/fs1;%passband

[ordB, ~] = ellipord(wp_B, ws_B, ap_B, as_B);
[num_B, dem_B] = ellip(ordB, ap_B, as_B, wp_B,'high');
[hB, wB] = freqz(num_B, dem_B);

%MAG
plot(wB/pi, abs(hB));
title('Magnitude response of Block B');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Magnitude(linear scale)');

%PHASE
figure;
plot(wB/pi, phase(hB));
title('Phase response of Block B');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Phase(rad)');



%apply the filter
r2 = filter(num_B, dem_B, r1r2);


figure;
N=25000;
X1_mags = abs(fft(r2));
fax_bins = [0 : N-1]; %frequency axis in bins
N_2 = ceil(N/2);
plot(fax_bins(1:N_2)*fs1/N, X1_mags(1:N_2));
title('s~ 2[n] in Frequency domain');
xlabel('frequency(Hz)');
ylabel('Magnitude(linear scale)');


%demodulation

t = (0:1:24999).* (1/fs1);
demod = zeros(1, 25000);
for i = 1:1:25000
   demod(i) = r2(i).*cos(2*pi*12288*t(i)+ pi/3);
end

figure;
N=25000;
X1_mags = abs(fft(demod));
fax_bins = [0 : N-1]; %frequency axis in bins
N_2 = ceil(N/2);
```

```
plot(fax_bins(1:N_2)*fs1/N, X1_mags(1:N_2));
title('Demodulated signal in Frequency domain');
xlabel('frequency(Hz)');
ylabel('Magnitude(linear scale)');


%the only requirement here is the passband of r1
dp_C=9E-3; ds_C=0.001;
%peak passband ripple and minimum stopband attenuation in dB
ap_C=-20*log10(1-dp_C); as_C=-20*log10(ds_C);

wp_C = 2* 4096/fs1;
ws_C = wp_C +0.05;%right after passband

[ordC, ~] = ellipord(wp_C, ws_C, ap_C, as_C);
[num_C, dem_C] = ellip(ordC, ap_C, as_C, wp_C,  'low');
[hC, wC] = freqz(num_C, dem_C);
%MAG
figure;
plot(wC/pi, abs(hC));
title('Magnitude response of Block C');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Magnitude(linear scale)');

%PHASE
figure;
plot(wC/pi, phase(hC));
title('Phase response of Block C');
xlabel('Normalised frequency(*pi rad/s)');
ylabel('Phase(rad)');



%DEMOD AND LPFILTERED
r1 = filter(num_C, dem_C, demod);


figure;
N=25000;
X1_mags = abs(fft(r1));
fax_bins = [0 : N-1]; %frequency axis in bins
N_2 = ceil(N/2);
plot(fax_bins(1:N_2)*fs1/N, X1_mags(1:N_2));
title('s2[n] in Frequency domain');
xlabel('frequency(Hz)');
ylabel('Magnitude(linear scale)');

sound(r1, fs1);

fprintf('The order of Block B is %d\n', ordB);
fprintf('The order of Block C is %d\n', ordC);
```