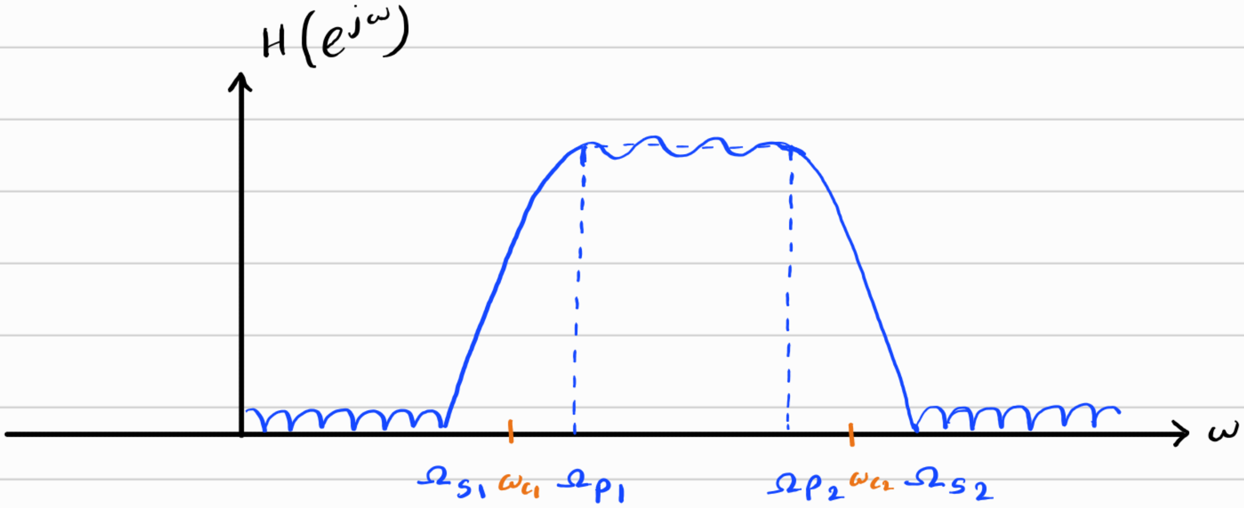
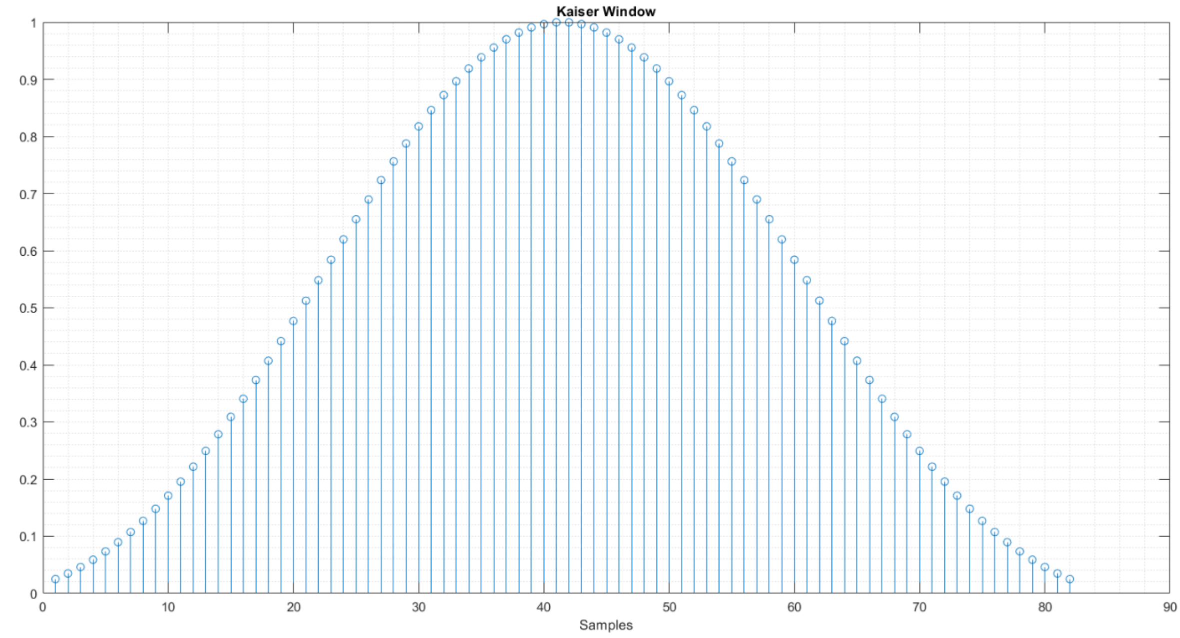
1) Design a FIR bandpass digital filter

Index number : 200488E

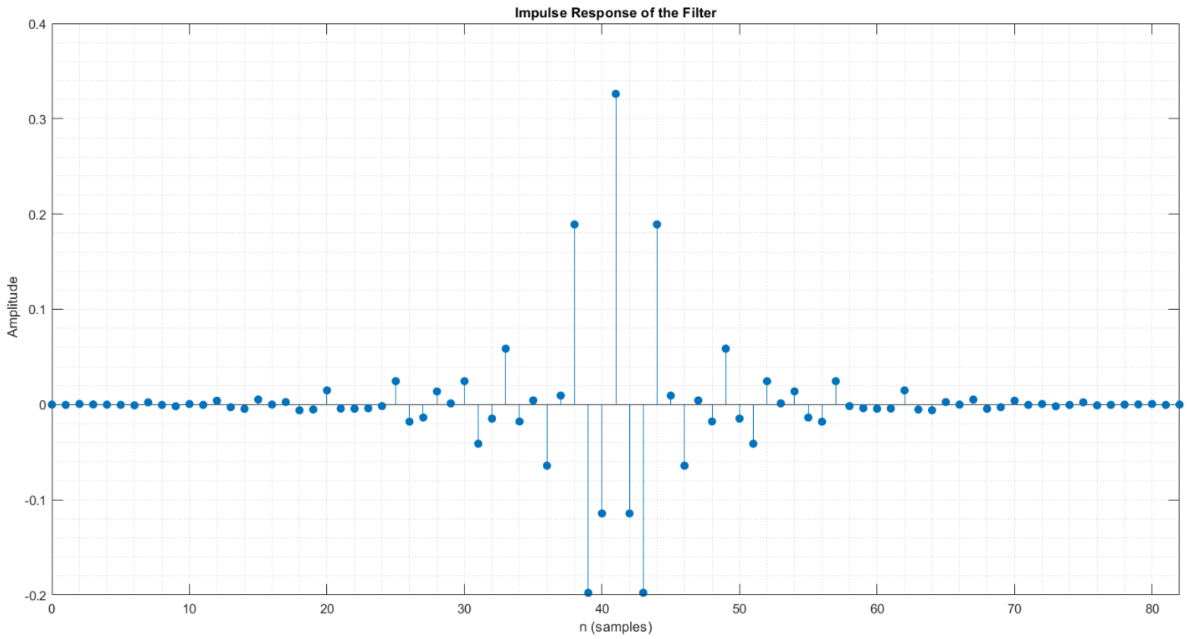
A B C (under 4, 8, 8)

Parameter	Value
Maximum passband ripple, $\tilde{A}_p$	0.14 dB
Minimum stopband attenuation, $\tilde{A}_a$	58 dB
Lower passband edge, Ω_{p1}	1200 rad/s
Upper passband edge, Ω_{p2}	1700 rad/s
Lower stopband edge, Ω_{s1}	900 rad/s
Upper stopband edge, Ω_{s2}	1900 rad/s
Sampling frequency, Ω_{sm}	4600 rad/s

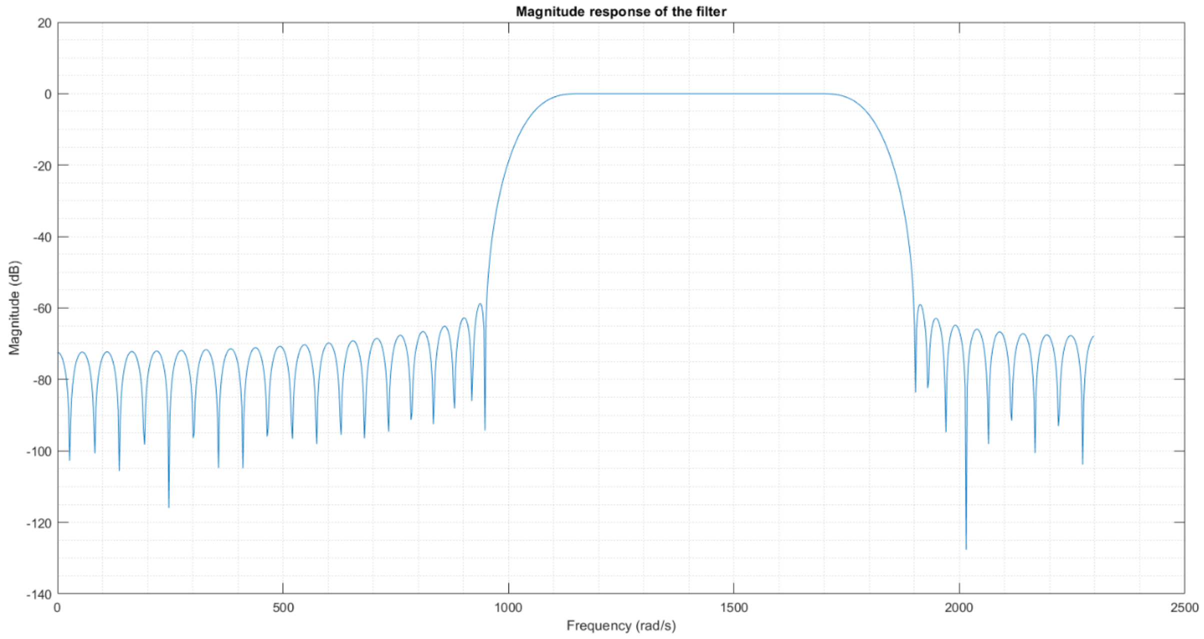




a) Impulse response

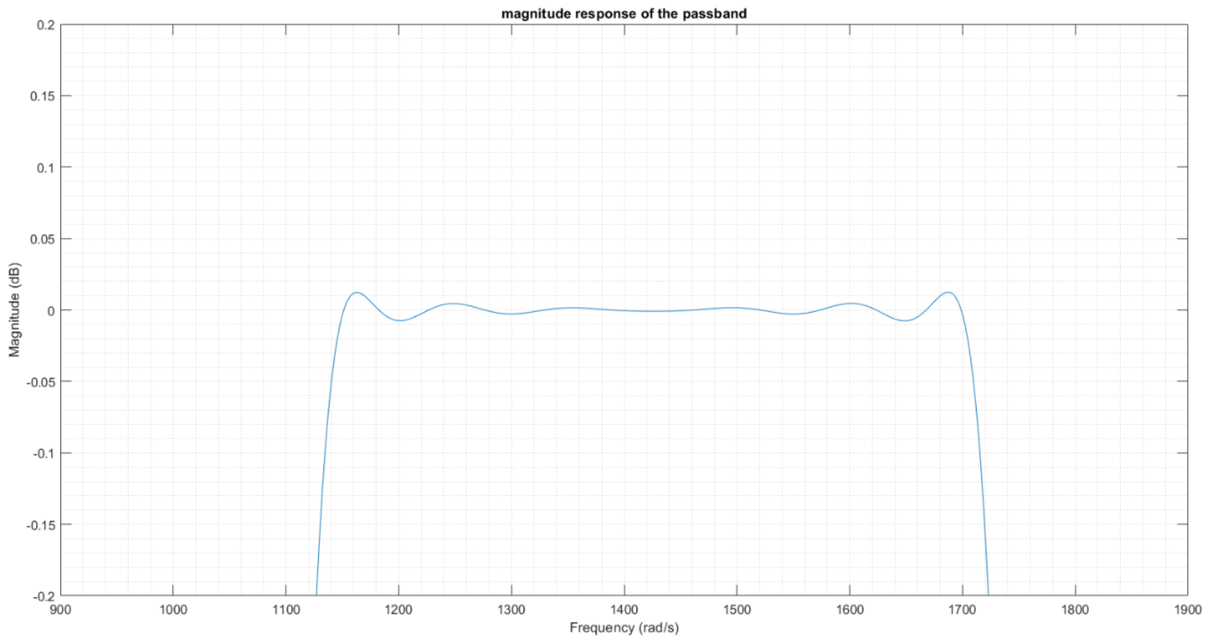


b) Magnitude response



c) Magnitude response

1200rad/s < ω < 1700 rad/s



* Here we can see a smooth frequency transition because we have used a smooth window function which produces a smooth transition between the passband and the stopband.

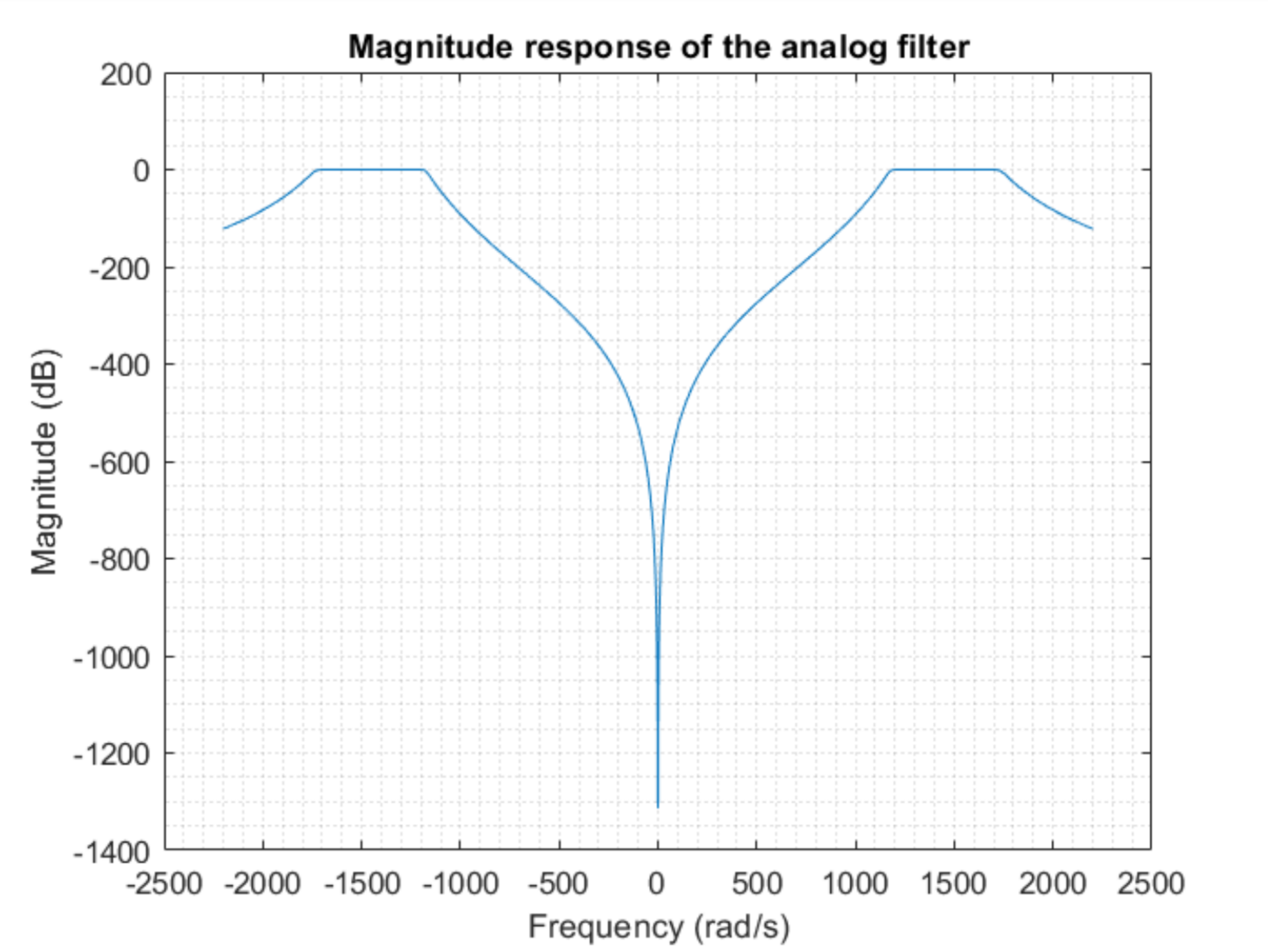
2) Design an IIR bandpass filter by using bilinear transformation

digit c = 8. ⇒ 8 % 4 = 0

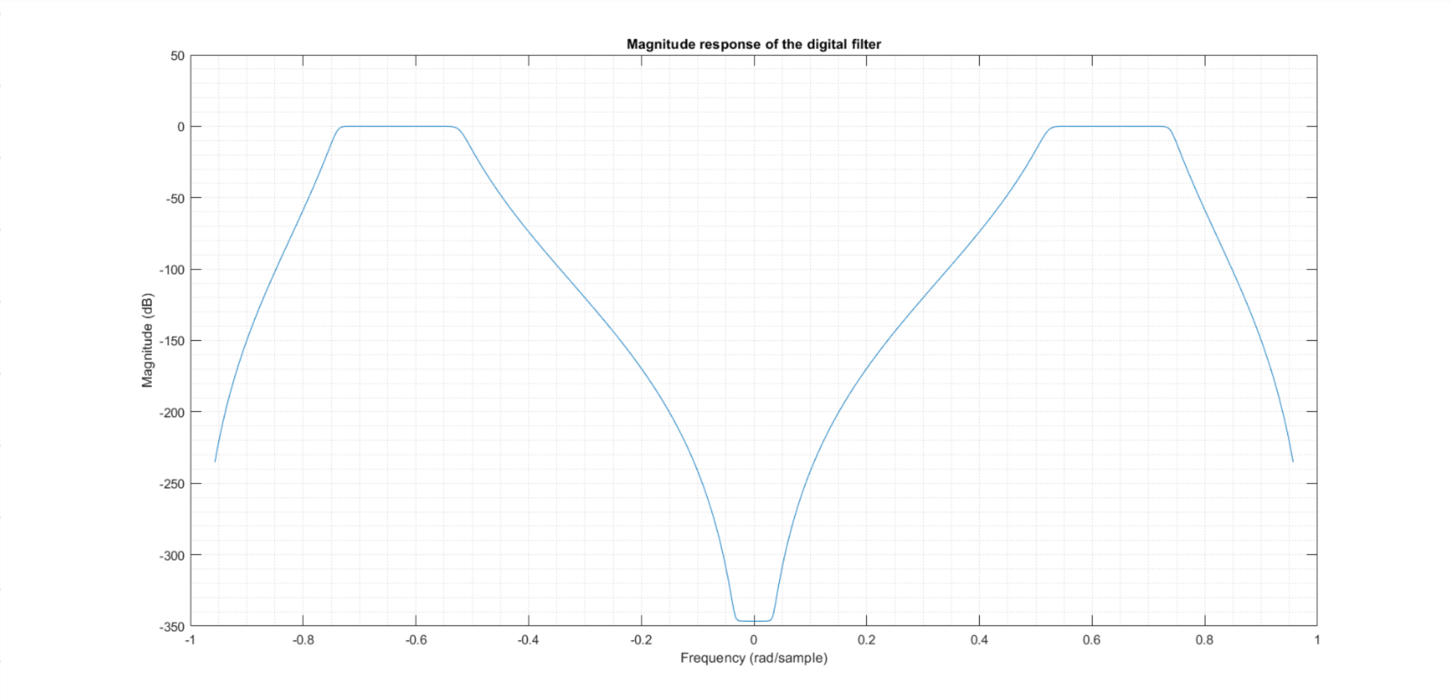
∴ Approximation method - Butterworth

a)

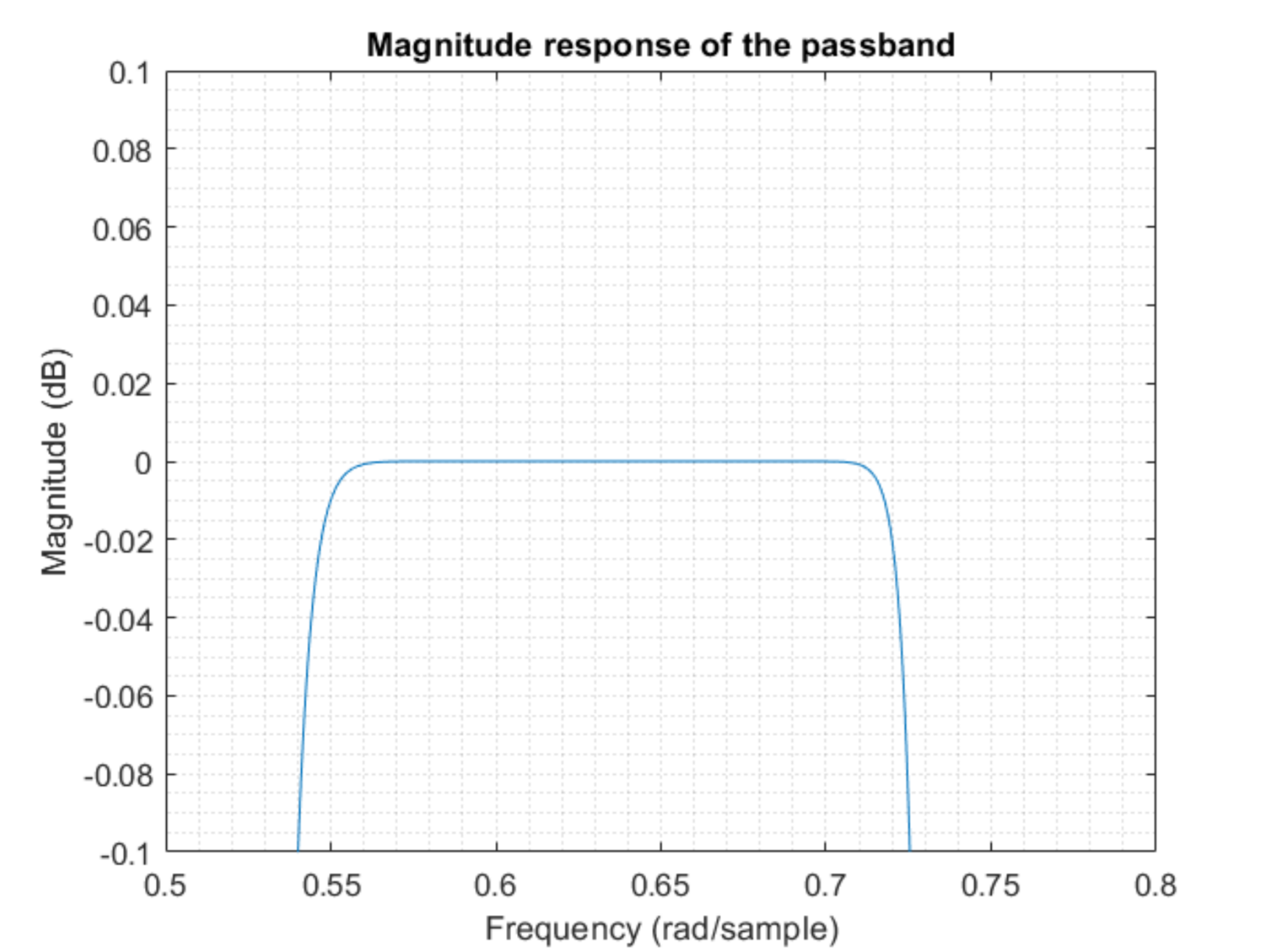
Numerator	Denominator
9.73E-07	1
-7.00E-17	7.2748
-1.07E-05	30.382
-7.89E-16	90.047
5.35E-05	209.25
-3.11E-15	399.58
-0.00016061	646.53
-5.97E-15	902.91
0.00032121	1102.7
-6.20E-15	1187.4
-0.0004497	1133.6
-3.40E-15	962.23
0.0004497	726.93
-6.89E-16	488.11
-0.00032121	290.42
2.17E-16	152.21
0.00016061	69.683
1.42E-16	27.495
-5.35E-05	9.1799
2.14E-17	2.5148
1.07E-05	0.5398
4.19E-19	0.0822
-9.73E-07	0.0073



b) Magnitude response of the digital filter.



c) Magnitude response in the passband



* The Butterworth filter has a smooth frequency response, which means that it has a monotonically increasing passband gain and no ripples in the passband or stopband.

* And we can observe that this has a linear phase delay. i.e. the phase delay is constant across the passband of the filter

* And this is a stable filter. Its impulse response decays over time and it doesn't oscillate.

3) Comparison

	FIR Filter	IIR Filter
Order	82	22
No. of Multiplications	$\frac{84}{2} = 42$ (Symmetry)	45
No. of Additions	82	44

FIR filter details

```
Discrete-Time FIR Filter (real)
-------------------------------
Filter Structure    : Direct-Form II Transposed
Numerator Length    : 83
Denominator Length  : 1
Stable              : Yes
Linear Phase        : Yes (Type 1)

Implementation Cost
Number of Multipliers            : 84
Number of Adders                 : 82
Number of States                 : 82
Multiplications per Input Sample : 84
Additions per Input Sample       : 82
```

IIR filter details

```
Discrete-Time IIR Filter (real)
-------------------------------
Filter Structure    : Direct-Form II Transposed
Numerator Length    : 23
Denominator Length  : 23
Stable              : Yes
Linear Phase        : No

Implementation Cost
Number of Multipliers            : 45
Number of Adders                 : 44
Number of States                 : 22
Multiplications per Input Sample : 45
Additions per Input Sample       : 44
```

MATLAB code

Specification of the Filter

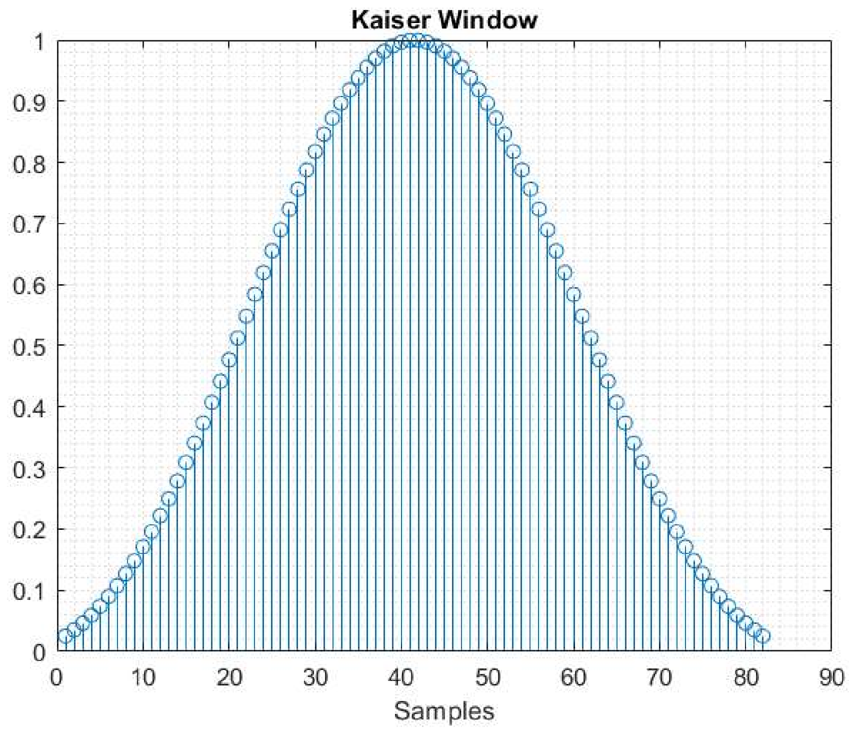
```
fsamp = 4600;  % sampling frequency is  4600 rad/s
fcuts = [900 1200 1700 1900]; % passband and stopband edges
mags = [0 1 0];
devs = [db2mag(-58) db2mag(0.14) db2mag(-58)]; % passband ripple = 0.14dB & stopband ripple = 58dB
```

Desining the Kaiser window

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

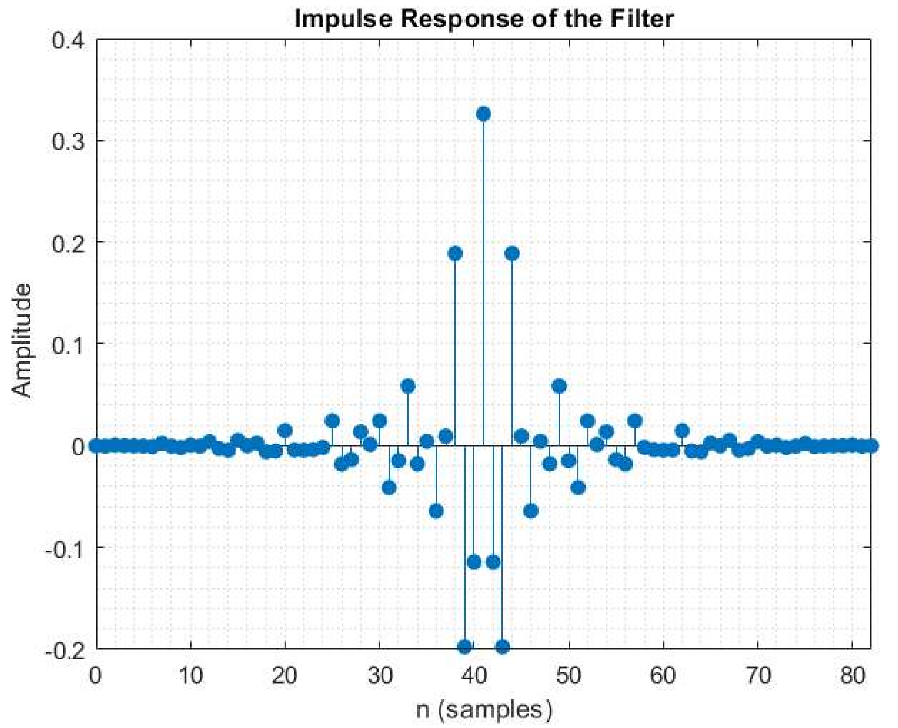
```
n = 82
```

```
figure;
stem(kaiser(n,beta))
title("Kaiser Window")
xlabel("Samples")
grid minor
```



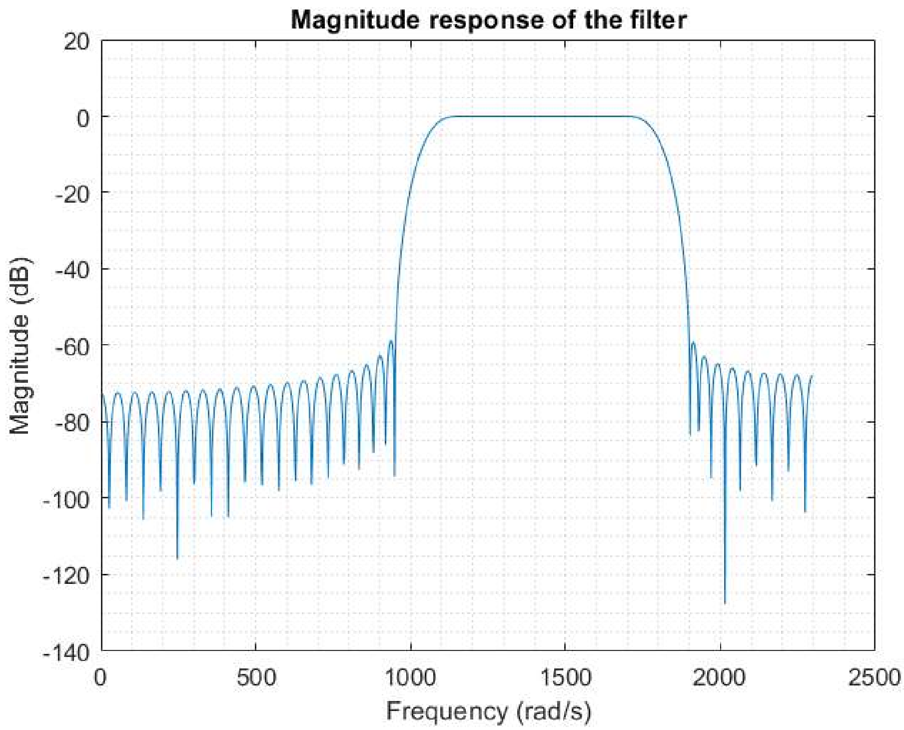
Impilse response of the filter

```
figure;
impz(hh)
title("Impulse Response of the Filter")
grid minor
```



Magnitude response of the filter

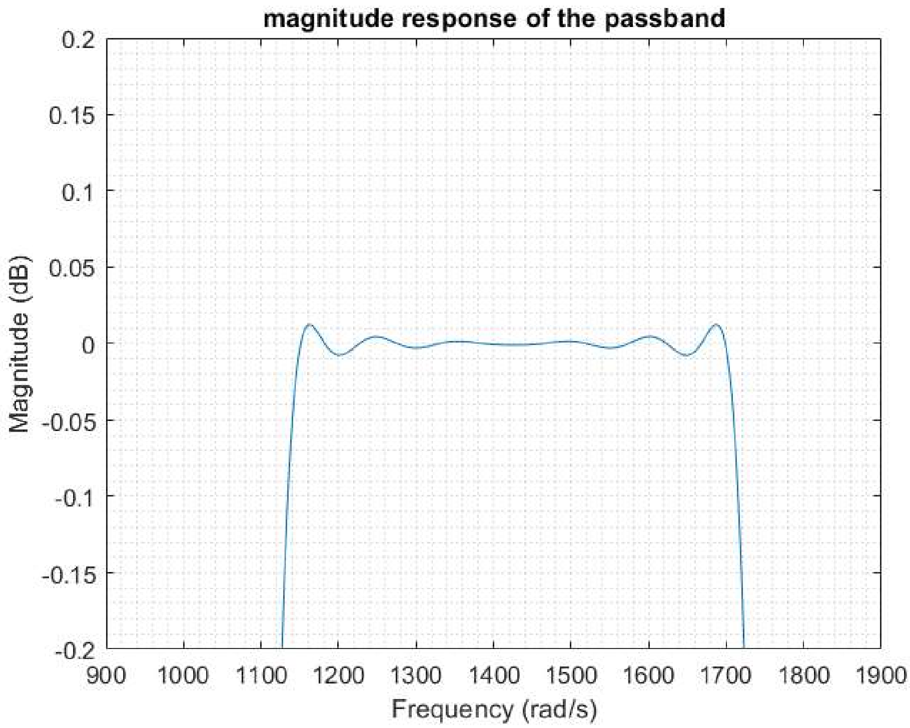
```
figure;
[H,f] = freqz(hh,1,1024,fsamp);
plot(f,(mag2db(abs(H))))
title("Magnitude response of the filter")
xlabel("Frequency (rad/s)")
ylabel("Magnitude (dB)")
grid minor
```



Magnitude response in the passband

```
figure;
```

```
[H,f] = freqz(hh,1,1024,fsamp);
plot(f,(mag2db(abs(H))))
axis([900 , 1900 , -0.2 , 0.2]);
title("magnitude response of the passband")
xlabel("Frequency (rad/s)")
ylabel("Magnitude (dB)")
grid minor
```



Disign a IIR bandpass digital Filter

Filter specification

```
wp = [1200 1700];
ws = [900 1900];
Rp = 0.14;
Rs = 58;
```

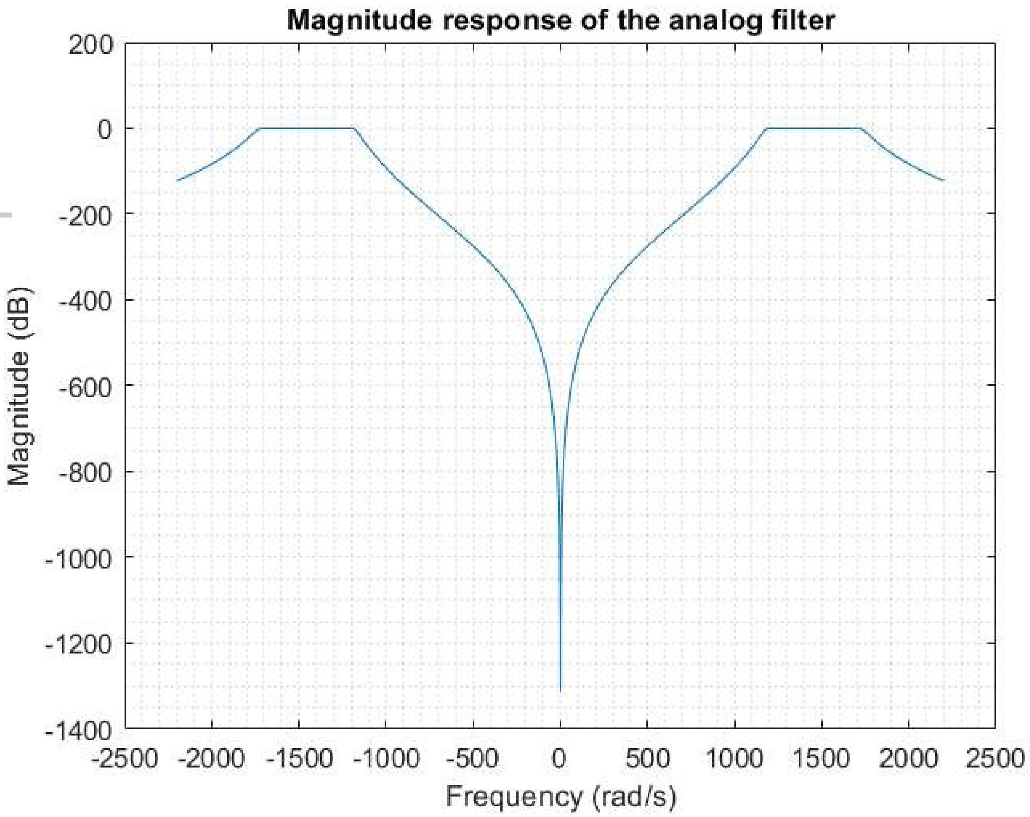
Designing the analog filter

```
[n,wn] = buttord(wp,ws,Rp,Rs,"s");
[b,a] = butter(n,wn,"s");
filter = tf(b,a);
w_sam = 4600;
t_sam = 2*pi/w_sam;
f_sam = 1/t_sam;
```

Analog filter magnitude response

```
w = linspace(-2200,2200,4400);
h = freqs(b, a, w);
mag = abs(h);

figure(1)
plot(w,mag2db(mag))
title("Magnitude response of the analog filter")
xlabel("Frequency (rad/s)")
ylabel("Magnitude (dB)")
grid minor
```



Prewrapping frequencies

```
wp(1) = 2/t_sam*tan(wp(1)*t_sam/2);
wp(2) = 2/t_sam*tan(wp(2)*t_sam/2);
ws(1) = 2/t_sam*tan(ws(1)*t_sam/2);
ws(2) = 2/t_sam*tan(ws(2)*t_sam/2);
```

Frequancy Normalizing

```
wp = [wp(1)/(w_sam/2) wp(2)/(w_sam/2)];
ws = [ws(1)/(w_sam/2) ws(2)/(w_sam/2)];
```

Transforming to a Digital filter

```
[n,wc] = buttord(wp,ws,Rp,Rs,'s');
[z,p,k] = buttap(n);
[A,B,C,D] = zp2ss(z,p,k);
[At,Bt,Ct,Dt] = lp2bp(A,B,C,D,sqrt(wp(1)*wp(2)),wp(2)-wp(1));

w=linspace(-2200/(w_sam/2),2200/(w_sam/2),4400);
[Ad,Bd,Cd,Dd] = bilinear(At,Bt,Ct,Dt,1/pi); % Bilinear Transformation here
filter = ss2sos(Ad,Bd,Cd,Dd);
[b,a] = sos2tf(filter);

filter = tf(b,a); % Coefficients of the transfer function
[num,den] = tfdata(filter);
num
```

```
num = 1×1 cell array
    {[9.7337e-07 -7.0003e-17 -1.0707e-05 -7.8852e-16 5.3535e-05 -3.1071e-15 -1.6061e-04 -5.9745
```

```
den
```

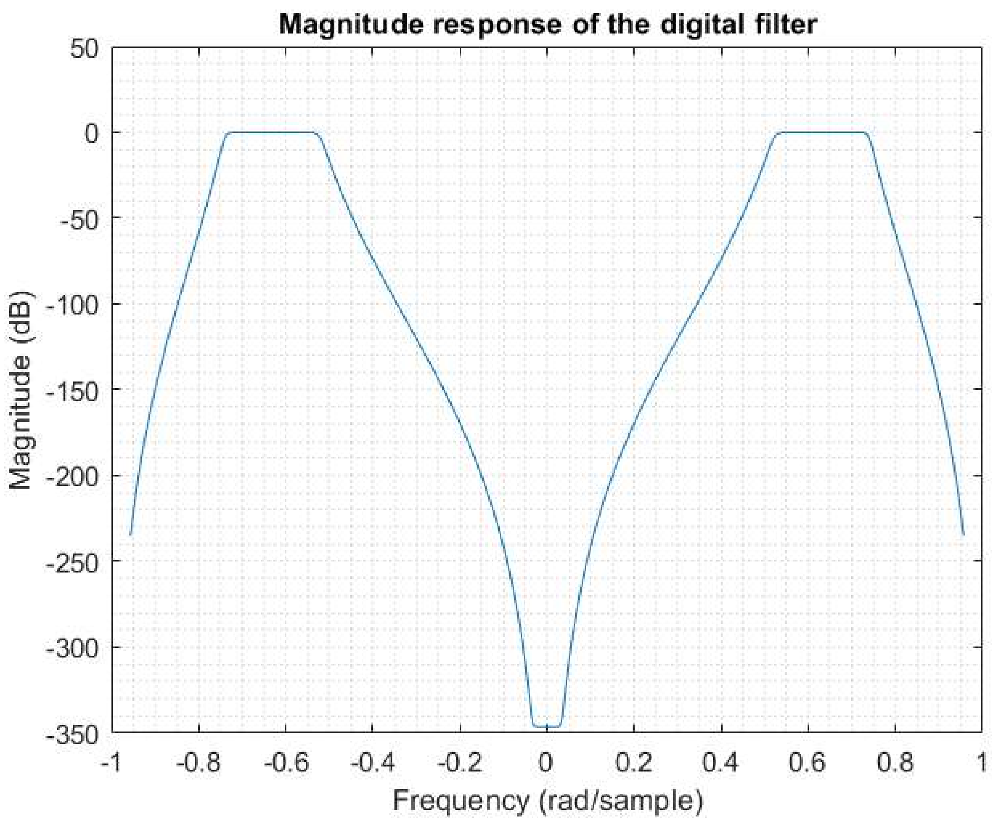
```
den = 1×1 cell array
    {[1 7.2748 30.3821 90.0473 209.2506 399.5753 646.5323 902.9116 1.1027e+03 1.1874e+03 1.1336
```

```
[hd,f] = freqz(b,a,w,2);
```

Digital filter magnitude response

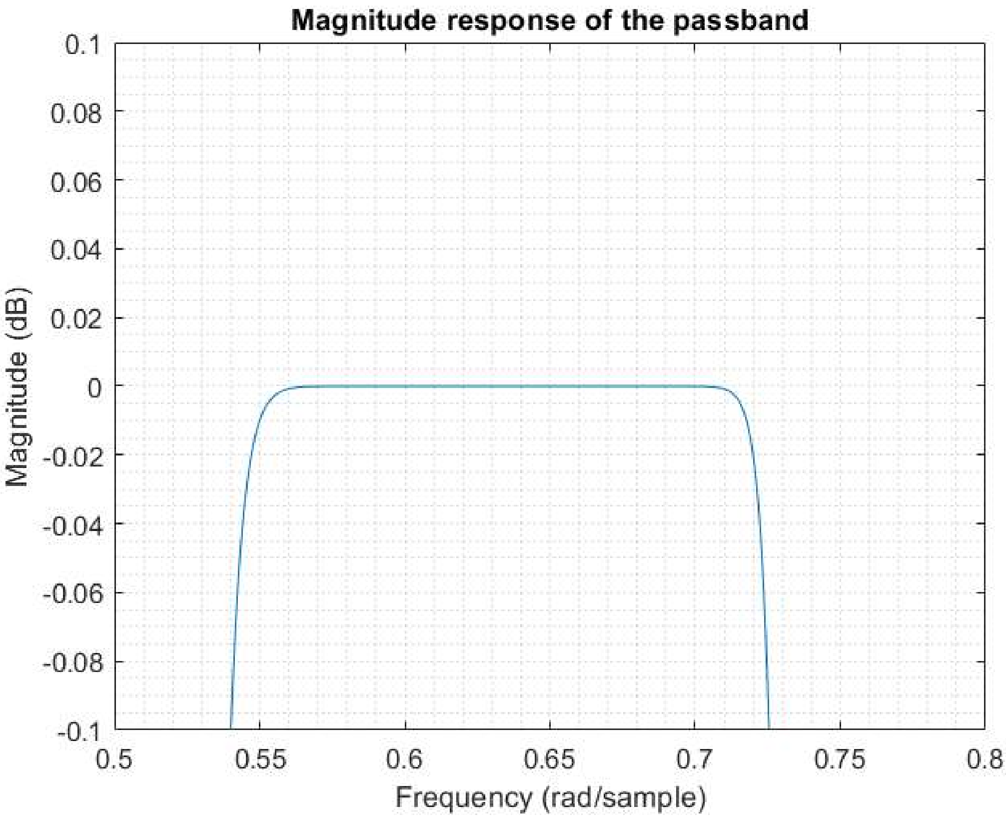
```
magn = abs(hd);
wp=[1200/(w_sam/2) 1700/(w_sam/2)];
ws=[900/(w_sam/2) 1900/(w_sam/2)];
figure(2)

plot(w,mag2db(magn))
title("Magnitude response of the digital filter")
xlabel("Frequency (rad/sample)")
ylabel("Magnitude (dB)")
grid minor
```



Magnitude response of the passband

```
figure(3)
plot(w,mag2db(magn))
axis ([ 0.5 , 0.8 , -0.1 , 0.1]);
title("Magnitude response of the passband")
xlabel("Frequency (rad/sample)")
ylabel("Magnitude (dB)")
grid("minor")
```



Magnitude response of the passband
Magnitude (dB)
0.1
0.08
0.06
0.04
0.02
0
-0.02
-0.04
-0.06
-0.08
-0.1
0.5
0.55
0.6
0.65
0.7
0.75
0.8
Frequency (rad/sample)