Department of Electronic and Telecommunication Engineering

University of Moratuwa

EN2063—SIGNALS AND SYSTEMS

Filter Design REPORT

Name	Index Number
SAIRISAN.R	200522V

1. a)

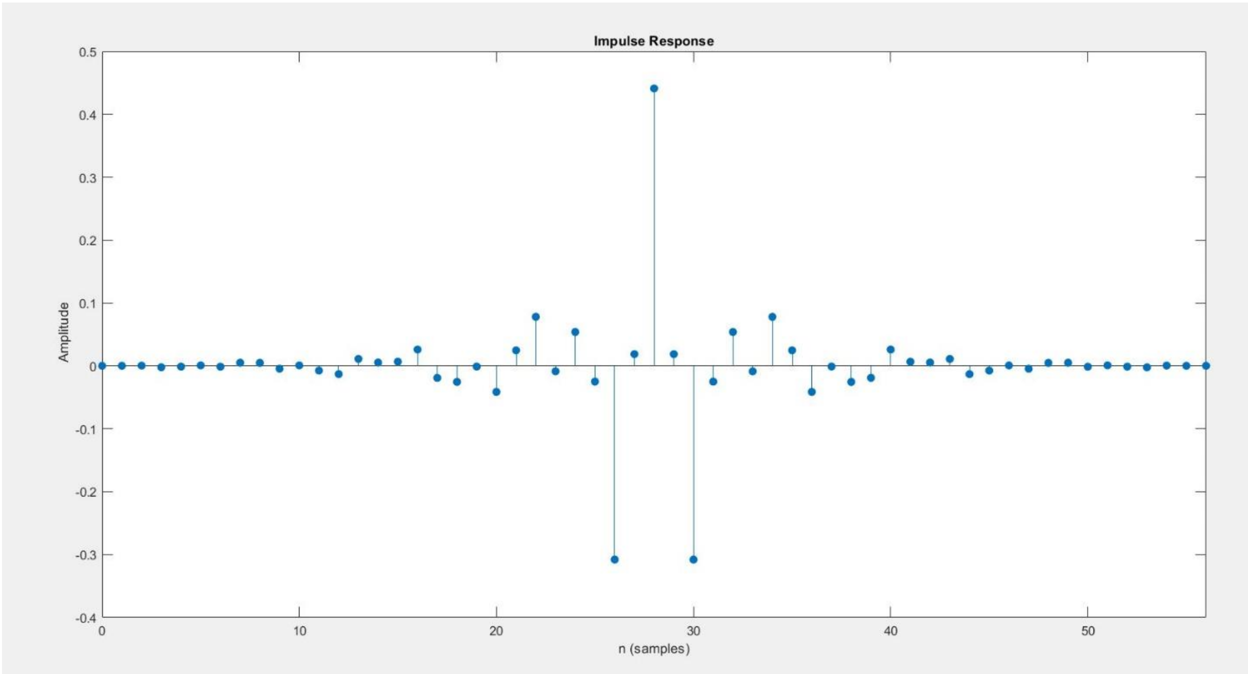

Fig 1.a – Impulse Response of FIR Filter

```
%DESIGN OF FIR FILTER..................................
%----------------------------------------------------------------------
%Index Number - 200552V

% A- First Digit of Index No.
% B- Second Digit of Index No.
% C- Third Digit of Index No.

% Ap- Maximum Passband ripple in DB
% Aa- Minimum Stopband attenuation in DB
% digital_Wp1- Lowe Passband Edge  in hertz
% digital_Wp2- Upper Passband Edge in hertz
% digital_Ws1- Lower Stopband Edge  in hertz
% digital_Ws2- Upper Stopband edge  in hertz
% digital_Wsm -Sampling Frequency  in hertz
clear all,

A=5;
B=5;
C=2;
Ap= 0.1+(0.01*A);      Rp = Ap;
Aa= 50+B;              Rs = Aa;
Wsm= 2*(((C*100)+1500));

Analog_Wp1= ((C*100)+400);
Analog_Wp2= ((C*100)+900);
Analog_Ws1= ((C*100)+100);
Analog_Ws2= ((C*100)+1100);


fcuts = [Analog_Ws1    Analog_Wp1    Analog_Wp2     Analog_Ws2];   %setting bandfrequencies (stop and passbands)
mags = [0 1 0];
devs = [10^(-Rs/20) 10^(-Rp/20) 10^(-Rs/20)];          % converstion of decible attenuation.

[n,Wn,beta,ftype] = kaiserord(fcuts,mags,devs,Wsm);   %to computes the order of an FIR filter and the independent parameter to a Kaiser window
n = n + rem(n,2);
hh = fir1(n,Wn,ftype,kaiser(n+1,beta),'noscale');       %to calculates the coefficients of FIR filters designed using the windowing method
[h,f]=freqz(hh,1,1024,Wsm);        % computes the frequency, magnitude, and phase response of a digital filter

%Code change only here....................
% plot(f,db(h))
% ylim([-75 10])
% % xlim([0 1])
% xlabel("Normalized Frequency ( × \pi rad/sample)");
% ylabel('Magnitute (DB)')
% title('56th order FIR Bandpass Filter')


impz(hh);          %impulse response of the filter..
```

b)

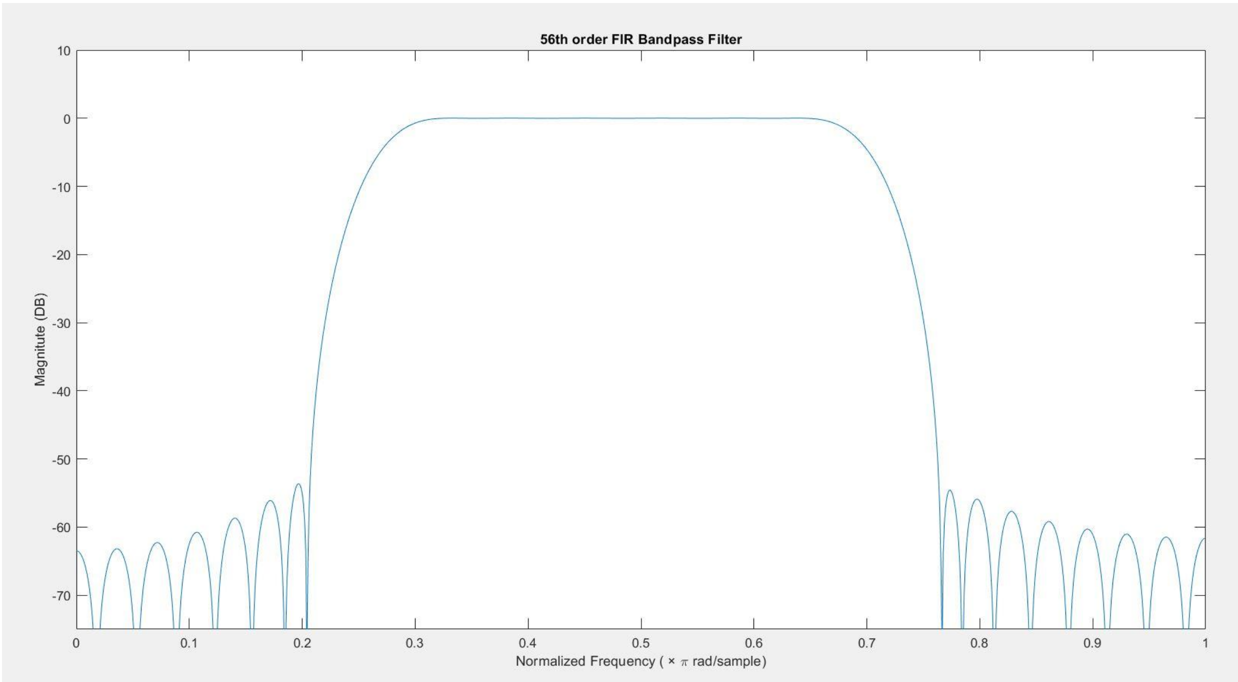

Fig 1.b –Frequency response of the FIR filter in Normalized Frequency (π rad/sample.)

```
%DESIGN OF FIR FILTER.................................
%-------------------------------------------------------------------
%Index Number - 200552V

% A- First Digit of Index No.
% B- Second Digit of Index No.
% C- Third Digit of Index No.

% Ap- Maximum Passband ripple in DB
% Aa- Minimum Stopband attenuation in DB
% digital_Wp1- Lowe Passband Edge  in hertz
% digital_Wp2- Upper Passband Edge in hertz
% digital_Ws1- Lower Stopband Edge  in hertz
% digital_Ws2- Upper Stopband edge  in hertz
% digital_Wsm -Sampling Frequency  in hertz
clear all,

A=5;
B=5;
C=2;
Ap= 0.1+(0.01*A);     Rp = Ap;
Aa= 50+B;             Rs = Aa;
Wsm= 2*(((C*100)+1500));

Analog_Wp1= ((C*100)+400);
Analog_Wp2= ((C*100)+900);
Analog_Ws1= ((C*100)+100);
Analog_Ws2= ((C*100)+1100);


fcuts = [Analog_Ws1    Analog_Wp1    Analog_Wp2     Analog_Ws2];    %setting bandfrequencies (stop and passbands)
mags = [0 1 0];
devs = [10^(-Rs/20) 10^(-Rp/20) 10^(-Rs/20)];          % converstion of decible attenuation.

[n,Wn,beta,ftype] = kaiserord(fcuts,mags,devs,Wsm);   %to computes the order of an FIR filter and the independent parameter to a Kaiser window
n = n + rem(n,2);
hh = fir1(n,Wn,ftype,kaiser(n+1,beta),'noscale');       %to calculates the coefficients of FIR filters designed using the windowing method
[h,f]=freqz(hh,1,1024,Wsm);       % computes the frequency, magnitude, and phase response of a digital filter

%Code change only here...................
plot((2*f)/(Wsm),db(h))
ylim([-75 10])
 xlim([0 1])
xlabel("Normalized Frequency ( × \pi rad/sample)");
ylabel('Magnitute (DB)')
title('56th order FIR Bandpass Filter')


% impz(hh);          %impulse response of the filter..
```

c)

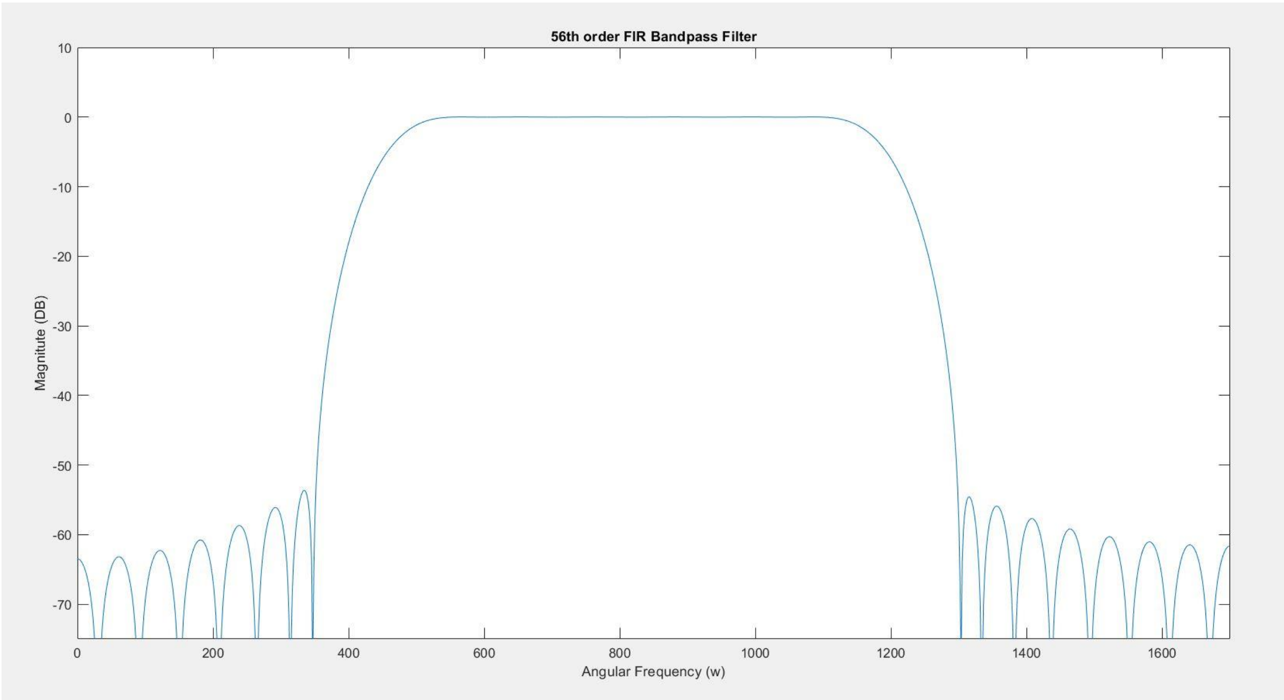

Fig 1.c –Frequency response of the FIR filter in Passband Angular Frequency (Wp1<W<Wp2)

```
%DESIGN OF FIR FILTER..................................
%----------------------------------------------------------------------
%Index Number - 200552V

% A- First Digit of Index No.
% B- Second Digit of Index No.
% C- Third Digit of Index No.

% Ap- Maximum Passband ripple in DB
% Aa- Minimum Stopband attenuation in DB
% digital_Wp1- Lowe Passband Edge  in hertz
% digital_Wp2- Upper Passband Edge in hertz
% digital_Ws1- Lower Stopband Edge  in hertz
% digital_Ws2- Upper Stopband edge  in hertz
% digital_Wsm -Sampling Frequency  in hertz
clear all,

A=5;
B=5;
C=2;
Ap= 0.1+(0.01*A);     Rp = Ap;
Aa= 50+B;             Rs = Aa;
Wsm= 2*(((C*100)+1500));

Analog_Wp1= ((C*100)+400);
Analog_Wp2= ((C*100)+900);
Analog_Ws1= ((C*100)+100);
Analog_Ws2= ((C*100)+1100);


fcuts = [Analog_Ws1    Analog_Wp1    Analog_Wp2     Analog_Ws2];   %setting bandfrequencies (stop and passbands)
mags = [0 1 0];
devs = [10^(-Rs/20) 10^(-Rp/20) 10^(-Rs/20)];          % converstion of decible attenuation.

[n,Wn,beta,ftype] = kaiserord(fcuts,mags,devs,Wsm);   %to computes the order of an FIR filter and the independent parameter to a Kaiser window
n = n + rem(n,2);
hh = fir1(n,Wn,ftype,kaiser(n+1,beta),'noscale');       %to calculates the coefficients of FIR filters designed using the windowing method
[h,f]=freqz(hh,1,1024,Wsm);       % computes the frequency, magnitude, and phase response of a digital filter

%Code change only here...................
plot(f,db(h))
ylim([-75 10])
 xlim([0 1700])
xlabel("Normalized Frequency ( × \pi rad/sample)");
ylabel('Magnitute (DB)')
title('56th order FIR Bandpass Filter')


% impz(hh);          %impulse response of the filter..
```

2. a)

```
>> Untitled
    0.0390   -0.0169   -0.1578    0.0381    0.3519   -0.0365   -0.5093   -0.0000    0.5093    0.0365   -0.3519   -0.0381    0.1578    0.0169   -0.0390

    1.0000   -1.1058    0.7713   -0.6977    1.4648   -0.9663    0.3761   -0.2744    0.3919   -0.1374    0.0061   -0.0135    0.0207   -0.0017   -0.0011
```

Fig 2.a –Filter Coefficients of IIR filter b1, a1

b)

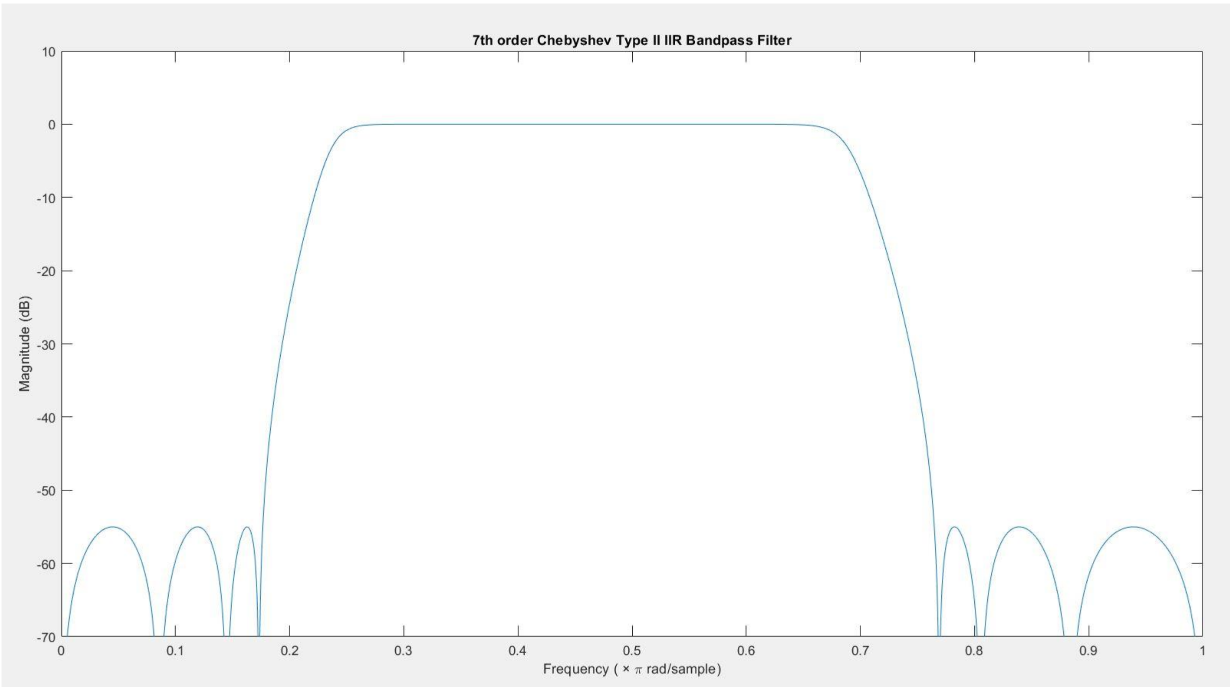

Fig 2.b – Frequency response of the IIR Inverse Chebyshev filter in Normalized Frequency (π rad/sample.)

```
%DESIGN OF IIR FILTER..................................
%----------------------------------------------------------------------
%Index Number - 200552V
% A- First Digit of Index No
% B- Second Digit of Index No
% C- Third Digit of Index No

% Ap- Maximum Passband ripple in DB
% Aa- Minimum Stopband attenuation in DB
% digital_Wp1- Lowe Passband Edge  in hertz
% digital_Wp2- Upper Passband Edge in hertz
% digital_Ws1- Lower Stopband Edge  in hertz
% digital_Ws2- Upper Stopband edge  in hertz
% digital_Wsm -Sampling Frequency  in hertz
clear all,

A=5;
B=5;
C=2;
Ap= 0.1+(0.01*A);     Rp = Ap;
Aa= 50+B;             Rs = Aa;
Wsm= 2*(((C*100)+1500));

%Needed Digital Angluar Frequencies...........................................
digital_Wp1= ((C*100)+400)/(2*pi);
digital_Wp2= ((C*100)+900)/(2*pi);
digital_Ws1= ((C*100)+100)/(2*pi);
digital_Ws2= ((C*100)+1100)/(2*pi);

%Prewarping Analog Frequency for Respective Frequencies.
analog_Wp1  =  2*Wsm*tan(digital_Wp1*(2*pi/(Wsm/(2*pi)))/2);
analog_Wp2  =  2*Wsm*tan(digital_Wp2*(2*pi/(Wsm/(2*pi)))/2);
analog_Ws1  =  2*Wsm*tan(digital_Ws1*(2*pi/(Wsm/(2*pi)))/2);
analog_Ws2  =  2*Wsm*tan(digital_Ws2*(2*pi/(Wsm/(2*pi)))/2);

[n,Ws] = cheb2ord([analog_Wp1 analog_Wp2],[analog_Ws1 analog_Ws2],Rp,Rs,'s');       % return minimum order and array of stopband angular freq
[b,a] = cheby2(n,Rs,[analog_Ws1 analog_Ws2],'bandpass','s');                        % for calculating analog filter cofficients
% disp(n);
[b1,a1] = bilinear(b,a,Wsm);                                                        % for the for converting analog filter cofficients to digital IIR.
% disp(b1);
% disp(b1);

%Code change only here....................

[H,f]=freqz(b1,a1,2048,Wsm);                         % computes the frequency, magnitude, and phase response of a digital filter
plot(2*f/Wsm,db(H));                              % Normalized the frequency respect to nyquist sampling rate.
ylim([-70 10])
xlabel("Frequency ( × \pi rad/sample)");
ylabel("Magnitude (dB)");
title('7th order Chebyshev Type II IIR Bandpass Filter')
```

c)

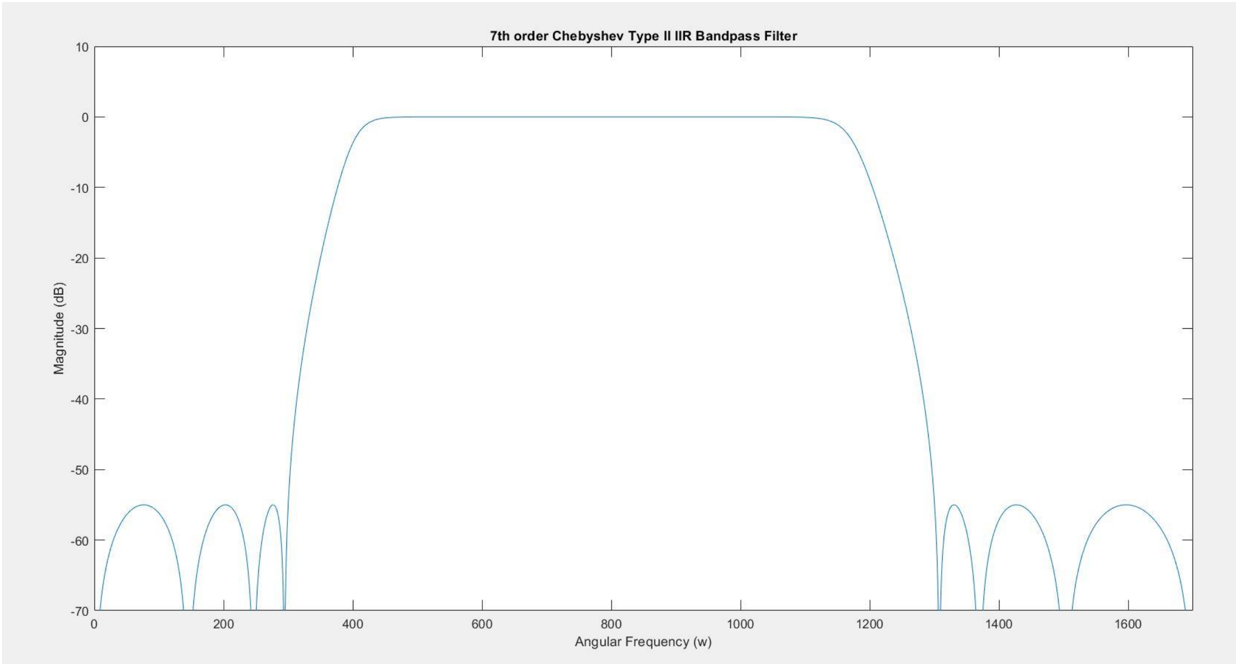

Fig 2.c –Frequency response of the IIR filter in Passband Angular Frequency (Wp1<W<Wp2)

```
%DESIGN OF IIR FILTER...................................
%---------------------------------------------------------------------
%Index Number - 200552V
% A- First Digit of Index No
% B- Second Digit of Index No
% C- Third Digit of Index No

% Ap- Maximum Passband ripple in DB
% Aa- Minimum Stopband attenuation in DB
% digital_Wp1- Lowe Passband Edge  in hertz
% digital_Wp2- Upper Passband Edge in hertz
% digital_Ws1- Lower Stopband Edge  in hertz
% digital_Ws2- Upper Stopband edge  in hertz
% digital_Wsm -Sampling Frequency  in hertz
clear all,

A=5;
B=5;
C=2;
Ap= 0.1+(0.01*A);      Rp = Ap;
Aa= 50+B;              Rs = Aa;
Wsm= 2*(((C*100)+1500));

%Needed Digital Angluar Frequencies............................................
digital_Wp1= ((C*100)+400)/(2*pi);
digital_Wp2= ((C*100)+900)/(2*pi);
digital_Ws1= ((C*100)+100)/(2*pi);
digital_Ws2= ((C*100)+1100)/(2*pi);

%Prewarping Analog Frequency for Respective Frequencies.
analog_Wp1  =  2*Wsm*tan(digital_Wp1*(2*pi/(Wsm/(2*pi)))/2);
analog_Wp2  =  2*Wsm*tan(digital_Wp2*(2*pi/(Wsm/(2*pi)))/2);
analog_Ws1  =  2*Wsm*tan(digital_Ws1*(2*pi/(Wsm/(2*pi)))/2);
analog_Ws2  =  2*Wsm*tan(digital_Ws2*(2*pi/(Wsm/(2*pi)))/2);

[n,Ws] = cheb2ord([analog_Wp1 analog_Wp2],[analog_Ws1 analog_Ws2],Rp,Rs,'s');       % return minimum order and array of stopband angular freq
[b,a] = cheby2(n,Rs,[analog_Ws1 analog_Ws2],'bandpass','s');                        % for calculating analog filter cofficients
% disp(n);
[b1,a1] = bilinear(b,a,Wsm);                                                 % for the for converting analog filter cofficients to digital IIR.
% disp(b1);
% disp(b1);

%Code change only here....................

[H,f]=freqz(b1,a1,2048,Wsm);              % computes the frequency, magnitude, and phase response of a digital filter
plot(f,db(H));                            % Normalized the frequency respect to nyquist sampling rate.
ylim([-70 10])
xlim([0 1700])
xlabel("Frequency ( × \pi rad/sample)");
ylabel("Magnitude (dB)");
title('7th order Chebyshev Type II IIR Bandpass Filter')
```

3.

The order of the FIR filter (=56) which designed in direct method is higher than the Order of the IIR filter (=7) which designed in indirect method based on analog inverse Chebyshev filter. In other words, for a same specification FIR filter needs higher order to give performance like low order IIR filter. This is a drawback of FIR filter compare to IIR Filter. Few filter coefficients are enough to design IIR filters.

FIR Filter

Filter Coefficients $=n+1=57$

The number of multiplies = number of coefficients × number of samples per second.

The number of multiplies = 57 × 3400samples/s = 193800/s

IIR Filter

Filter Coefficients $=2n+1=15$

The number of multiplies = number of coefficients × number of samples per second.

The number of multiplies = 15 × 3400samples/s = 51000/s