EXP 10 ***APPLICATION OF IIR FILTERS***

14/8/14

AIM:

To apply the IIR filters in finding the power soectral density of a signal.

PROGRAM:

*LOW PASS FILTER*

clc;

clear all;

close all;

fprintf('\nDIGITAL FILTER SPECIFICATIONS\n');

passfreq=input('Enter the passband frequency(hz) ');

stopfreq=input('Enter the stopband frequency(hz) ');

passatt=input('Enter the passband attenuation(db) ');

stopatt=input('Enter the stopband attenuation(db) ');

FS=input('ENTER THE SAMPLING FREQUENCY ');

TS=1/FS;

opassfreq=(2/TS)\*tan(pi\*passfreq\*TS);

ostopfreq=(2/TS)\*tan(pi\*stopfreq\*TS);

lambda=sqrt((10^(stopatt/10))-1);

ebsi=sqrt((10^(passatt/10))-1);

N=log(lambda/ebsi)/log(ostopfreq/opassfreq);

N=ceil(N);

fprintf('\nThe order of the filter is %d',N);

cf=opassfreq/(ebsi^(1/N));

fprintf('\nThe cut-off frequency is %d',cf);

for k=1:N

phi=(pi/2)+(((2\*k-1)\*pi)/(2\*N));

p(k)=exp(1i\*phi);

end

fprintf('\nTHE POLES ARE ');

disp(p);

p=p\*cf;

k=cf^(numel(p));

z=[];

[nums,dens]=zp2tf(z,p,k);

[numz,denz]=bilinear(nums,dens,FS);

[FR,W]=freqz(numz,denz);

figure

set(gca, 'FontSize', 16)

plot(W./pi,((abs(FR))));

xlabel('FREQUENCY');

ylabel('MAGNITUDE');

title('FREQUENCY RESPONSE OF FILTER');

figure

set(gca, 'FontSize', 16)

plot(W./pi,angle(FR));

ylabel('PHASE IN RADIANS');

xlabel('NORMALISED FREQUENCY');

[IR,t]=impz(numz,denz);

figure

set(gca, 'FontSize', 16)

plot(t/pi,IR);

title('IMPULSE RESPONSE');

Fs = 1000;

T = 1/Fs;

L = 1000;

t = (0:L-1)\*T; % Time vector

inputsig= cos(2\*pi\*300\*t)+ cos(2\*pi\*75\*t)+cos(2\*pi\*50\*t);

figure

set(gca, 'FontSize', 16)

plot(Fs\*t(1:50),inputsig(1:50));

title('Input Signal');

xlabel('time (milliseconds)');

NFFT = 2^nextpow2(L); % Next power of 2 from length of y

Y = fft(inputsig,NFFT)/L;

Y=Y.\*conj(Y);

f = Fs/2\*linspace(0,1,NFFT/2+1);

% Plot single-sided amplitude spectrum.

figure

set(gca, 'FontSize', 16)

plot(f,2\*abs(Y(1:NFFT/2+1)))

title('Amplitude Spectrum of input signal');

xlabel('Frequency (Hz)');

ylabel('|X(f)|');

%PSD OF FILTERED SIGNAL

y=conv(inputsig,IR);

figure

set(gca, 'FontSize', 16)

plot(Fs\*t(1:50),y(1:50));

title('Filtered Signal y(t)');

xlabel('time (milliseconds)');

NFFT = 2^nextpow2(L); % Next power of 2 from length of y

Y = fft(y,NFFT)/L;

Y=Y.\*conj(Y);

f = Fs/2\*linspace(0,1,NFFT/2+1);

% Plot single-sided amplitude spectrum.

figure

set(gca, 'FontSize', 16)

plot(f,2\*abs(Y(1:NFFT/2+1)))

title('Amplitude Spectrum of y(t)')

xlabel('Frequency (Hz)')

ylabel('|Y(f)|')

OUTPUT:

DIGITAL FILTER SPECIFICATIONS

Enter the passband frequency(hz) 100

Enter the stopband frequency(hz) 200

Enter the passband attenuation(db) 0.4

Enter the stopband attenuation(db) 30

ENTER THE SAMPLING FREQUENCY 1000

The order of the filter is 6

The cut-off frequency is 7.896542e+002

THE POLES ARE -0.2588 + 0.9659i -0.7071 + 0.7071i -0.9659 + 0.2588i -0.9659 - 0.2588i -0.7071 - 0.7071i -0.2588 - 0.9659i









*HIGH PASS FILTER*

clc;

clear all;

close all;

fprintf('\nDIGITAL FILTER SPECIFICATIONS\n');

passfreq=input('Enter the passband frequency(hz) ');

stopfreq=input('Enter the stopband frequency(hz) ');

passatt=input('Enter the passband attenuation(db) ');

stopatt=input('Enter the stopband attenuation(db) ');

FS=input('ENTER THE SAMPLING FREQUENCY ');

TS=1/FS;

temp=passfreq;

passfreq=stopfreq;

stopfreq=temp;

opassfreq=(2/TS)\*tan(pi\*passfreq\*TS);

ostopfreq=(2/TS)\*tan(pi\*stopfreq\*TS);

lambda=sqrt((10^(stopatt/10))-1);

ebsi=sqrt((10^(passatt/10))-1);

N=log10(lambda/ebsi)/log10(ostopfreq/opassfreq);

N=ceil(N);

fprintf('\nThe order of the filter is %d',N);

cf=ostopfreq/(ebsi^(1/N));

fprintf('\nThe cut-off frequency is %d',cf);

for k=1:N

phi=(pi/2)+(((2\*k-1)\*pi)/(2\*N));

p(k)=exp(sqrt(-1)\*phi);

end

fprintf('\nTHE POLES ARE ');

disp(p);

b=[zeros(1,N),1];

syms s b1 a1;

syms f

if rem(N,2)==0

for i=1:(N/2)

f=(s^2 + b1\*s + 1);

a1(i)=2\*sin(((2\*i-1)\*pi)/(2\*N));

hn(i)=subs(f,b1,a1(i));

end

else

for i=1:((N-1)/2)

f=(s+1)\*(s^2+b1\*s+1);

a1(i)=2\*sin(((2\*i-1)\*pi)/(2\*N));

hn(i)=subs(f,b1,a1(i));

end

end

h=1;

for g=1:i

h=h\*hn(g);

end

a=fliplr(sym2poly(coeffs(h,s)));

[bh,ah]=lp2hp(b,a,cf);

[numz,denz]=bilinear(bh,ah,FS);

[FR,W]=freqz(numz,denz);

figure

set(gca, 'FontSize', 16)

plot(W./pi,((abs(FR))));

xlabel('NORMALISED FREQUENCY');

ylabel('MAGNITUDE')

title('FREQUENCY RESPONSE');

figure

set(gca, 'FontSize', 16)

plot(W./pi,angle(FR));

ylabel('PHASE IN RADIANS');

xlabel('NORMALISED FREQUENCY');

[IR,t]=impz(numz,denz);

figure

set(gca, 'FontSize', 16)

plot(t/pi,IR);

title('IMPULSE RESPONSE');

Fs = 1000;

T = 1/Fs;

L = 1000;

t = (0:L-1)\*T; % Time vector

inputsig= cos(2\*pi\*300\*t)+ cos(2\*pi\*75\*t)+cos(2\*pi\*50\*t)+cos(2\*pi\*t\*490);

figure

set(gca, 'FontSize', 16)

plot(Fs\*t(1:50),inputsig(1:50));

title('Input Signal');

xlabel('time (milliseconds)');

NFFT = 2^nextpow2(L); % Next power of 2 from length of y

Y = fft(inputsig,NFFT)/L;

Y=Y.\*conj(Y);

f = Fs/2\*linspace(0,1,NFFT/2+1);

% Plot single-sided amplitude spectrum.

figure

set(gca, 'FontSize', 16)

plot(f,2\*abs(Y(1:NFFT/2+1)))

title('Amplitude Spectrum of input signal');

xlabel('Frequency (Hz)');

ylabel('|X(f)|');

%PSD OF FILTERED SIGNAL

y=conv(inputsig,IR);

figure

set(gca, 'FontSize', 16)

plot(Fs\*t(1:50),y(1:50));

title('Filtered Signal y(t)');

xlabel('time (milliseconds)');

NFFT = 2^nextpow2(L); % Next power of 2 from length of y

Y = fft(y,NFFT)/L;

Y=Y.\*conj(Y);

f = Fs/2\*linspace(0,1,NFFT/2+1);

% Plot single-sided amplitude spectrum.

figure

set(gca, 'FontSize', 16)

plot(f,2\*abs(Y(1:NFFT/2+1)))

title('Amplitude Spectrum of y(t)')

xlabel('Frequency (Hz)')

ylabel('|Y(f)|')

OUTPUT:

DIGITAL FILTER SPECIFICATIONS

Enter the passband frequency(hz) 200

Enter the stopband frequency(hz) 100

Enter the passband attenuation(db) 2

Enter the stopband attenuation(db) 20

ENTER THE SAMPLING FREQUENCY 1000

The order of the filter is 4

The cut-off frequency is 1.553841e+003

THE POLES ARE -0.3827 + 0.9239i -0.9239 + 0.3827i -0.9239 - 0.3827i -0.3827 - 0.9239i









*BAND PASS FILTER*

clc;

clear all;

close all;

fprintf('\nDIGITAL BAND PASS FILTER SPECIFICATIONS\n');%75 150 230 280

w1=input('ENTER THE VALUE FOR w1(hz) ');

wL=input('ENTER THE VALUE FOR wL(hz) ');

wU=input('ENTER THE VALUE FOR wU(hz) ');

w2=input('ENTER THE VALUE FOR w2(hz) ');

FS=input('ENTER THE SAMPLING FREQUENCY');

passatt=input('Enter the passband attenuation(db) ');

stopatt=input('Enter the stopband attenuation(db) ');

TS=1/FS;

lambda=sqrt((10^(stopatt/10))-1);

ebsi=sqrt((10^(passatt/10))-1);

ow1=(2/TS)\*tan(pi\*w1\*TS);

owL=(2/TS)\*tan(pi\*wL\*TS);

owU=(2/TS)\*tan(pi\*wU\*TS);

ow2=(2/TS)\*tan(pi\*w2\*TS);

a3=owL\*owU;

b3=owU-owL;

A=(-(ow1^2)+a3)/(ow1\*b3);

B=((ow2^2)-a3)/(ow2\*b3);

owr=min(abs(A),abs(B));

opassfreq=1;

ostopfreq=owr;

N=log10(lambda/ebsi)/log10(ostopfreq/opassfreq);

N=ceil(N);

fprintf('\nThe order of the filter is %d',N);

cf=ostopfreq/(ebsi^(1/N));

fprintf('\nThe cut-off frequency is %d',cf);

for k=1:N

phi=(pi/2)+(((2\*k-1)\*pi)/(2\*N));

p(k)=exp(sqrt(-1)\*phi);

end

fprintf('\nTHE POLES ARE ');

disp(p);

b=[zeros(1,N),1];

syms s b1 a1;

syms f

if rem(N,2)==0

for i=1:(N/2)

f=(s^2 + b1\*s + 1);

a1(i)=2\*sin(((2\*i-1)\*pi)/(2\*N));

hn(i)=subs(f,b1,a1(i));

end

else

for i=1:((N-1)/2)

f=(s+1)\*(s^2+b1\*s+1);

a1(i)=2\*sin(((2\*i-1)\*pi)/(2\*N));

hn(i)=subs(f,b1,a1(i));

end

end

h=1;

for g=1:i

h=h\*hn(g);

end

a=fliplr(sym2poly(coeffs(h,s)));

[bh,ah]=lp2bp(b,a,sqrt(a3),b3);

[numz,denz]=bilinear(bh,ah,FS);

[FR,W]=freqz(numz,denz);

figure

set(gca, 'FontSize', 16)

plot(W./pi,((abs(FR))));

xlabel('NORMALISED FREQUENCY');

ylabel('MAGNITUDE')

title('FREQUENCY RESPONSE');

figure

set(gca, 'FontSize', 16)

plot(W./pi,angle(FR));

ylabel('PHASE IN RADIANS');

xlabel('NORMALISED FREQUENCY');

[IR,t]=impz(numz,denz);

figure

set(gca, 'FontSize', 16)

plot(t/pi,IR);

title('IMPULSE RESPONSE');

Fs = 1000;

T = 1/Fs;

L = 1000;

t = (0:L-1)\*T; % Time vector

inputsig= cos(2\*pi\*300\*t)+ 1.22\*cos(2\*pi\*475\*t)+cos(2\*pi\*50\*t)+cos(2\*pi\*200\*t);

figure

set(gca, 'FontSize', 16)

plot(Fs\*t(1:50),inputsig(1:50));

title('Input Signal');

xlabel('time (milliseconds)');

NFFT = 2^nextpow2(L); % Next power of 2 from length of y

Y = fft(inputsig,NFFT)/L;

Y=Y.\*conj(Y);

f = Fs/2\*linspace(0,1,NFFT/2+1);

% Plot single-sided amplitude spectrum.

figure

set(gca, 'FontSize', 16)

plot(f,2\*abs(Y(1:NFFT/2+1)))

title('Amplitude Spectrum of input signal');

xlabel('Frequency (Hz)');

ylabel('|X(f)|');

%PSD OF FILTERED SIGNAL

y=conv(inputsig,IR);

figure

set(gca, 'FontSize', 16)

plot(Fs\*t(1:50),y(1:50));

title('Filtered Signal y(t)');

xlabel('time (milliseconds)');

NFFT = 2^nextpow2(L); % Next power of 2 from length of y

Y = fft(y,NFFT)/L;

Y=Y.\*conj(Y);

f = Fs/2\*linspace(0,1,NFFT/2+1);

% Plot single-sided amplitude spectrum.

figure

set(gca, 'FontSize', 16)

plot(f,2\*abs(Y(1:NFFT/2+1)))

title('Amplitude Spectrum of y(t)')

xlabel('Frequency (Hz)')

ylabel('|Y(f)|')

OUTPUT:

DIGITAL BAND PASS FILTER SPECIFICATIONS

ENTER THE VALUE FOR w1(hz) 75

ENTER THE VALUE FOR wL(hz) 150

ENTER THE VALUE FOR wU(hz) 230

ENTER THE VALUE FOR w2(hz) 480

ENTER THE SAMPLING FREQUENCY1000

Enter the passband attenuation(db) 3

Enter the stopband attenuation(db) 20

The order of the filter is 2

The cut-off frequency is 4.388528e+000









*BAND STOP FILTER*

clc;

clear all;

close all;

fprintf('\nDIGITAL BAND STOP FILTER SPECIFICATIONS\n');

wL=input('ENTER THE VALUE FOR wL(hz) ');

w1=input('ENTER THE VALUE FOR w1(hz) ');

w2=input('ENTER THE VALUE FOR w2(hz) ');

wU=input('ENTER THE VALUE FOR wU(hz) ');

FS=input('ENTER THE SAMPLING FREQUENCY');

passatt=input('Enter the passband attenuation(db) ');

stopatt=input('Enter the stopband attenuation(db) ');

wL=(2\*pi\*wL)/FS;

w1=(2\*pi\*w1)/FS;

w2=(2\*pi\*w2)/FS;

wU=(2\*pi\*wU)/FS;

Ws =[w1 w2];

Wp =[wL wU];

[N,Wn] = buttord(Wp/pi,Ws/pi,passatt,stopatt);

[b,a] = butter(N,Wn, 'stop');

[FR,W]=freqz(b,a);

figure

set(gca, 'FontSize', 16)

plot(W./pi,((abs(FR))));

xlabel('NORMALISED FREQUENCY');

ylabel('MAGNITUDE')

title('FREQUENCY RESPONSE');

figure

set(gca, 'FontSize', 16)

plot(W./pi,angle(FR));

ylabel('PHASE IN RADIANS');

xlabel('NORMALISED FREQUENCY');

[IR,t]=impz(b,a);

figure

set(gca, 'FontSize', 16)

plot(t/pi,IR);

title('IMPULSE RESPONSE');

title('frequency response of bandreject butterworth filter(IIR)' )

Fs = 1000;

T = 1/Fs;

L = 1000;

t = (0:L-1)\*T; % Time vector

inputsig=cos(2\*pi\*200\*t)+cos(2\*pi\*50\*t)+1.22\*cos(2\*pi\*t\*475);

figure

set(gca, 'FontSize', 16)

plot(Fs\*t(1:50),inputsig(1:50));

title('Input Signal');

xlabel('time (milliseconds)');

NFFT = 2^nextpow2(L); % Next power of 2 from length of y

Y = fft(inputsig,NFFT)/L;

Y=Y.\*conj(Y);

f = Fs/2\*linspace(0,1,NFFT/2+1);

% Plot single-sided amplitude spectrum.

figure

set(gca, 'FontSize', 16)

plot(f,2\*abs(Y(1:NFFT/2+1)))

title('Amplitude Spectrum of input signal');

xlabel('Frequency (Hz)');

ylabel('|X(f)|');

%PSD OF FILTERED SIGNAL

y=conv(inputsig,IR);

figure

set(gca, 'FontSize', 16)

plot(Fs\*t(1:50),y(1:50));

title('Filtered Signal y(t)');

xlabel('time (milliseconds)');

NFFT = 2^nextpow2(L); % Next power of 2 from length of y

Y = fft(y,NFFT)/L;

Y=Y.\*conj(Y);

f = Fs/2\*linspace(0,1,NFFT/2+1);

% Plot single-sided amplitude spectrum.

figure

set(gca, 'FontSize', 16)

plot(f,2\*abs(Y(1:NFFT/2+1)))

title('Amplitude Spectrum of y(t)')

xlabel('Frequency (Hz)')

ylabel('|Y(f)|')

OUTPUT:

DIGITAL BAND STOP FILTER SPECIFICATIONS

ENTER THE VALUE FOR wL(hz) 75

ENTER THE VALUE FOR w1(hz) 150

ENTER THE VALUE FOR w2(hz) 230

ENTER THE VALUE FOR wU(hz) 280

ENTER THE SAMPLING FREQUENCY1000

Enter the passband attenuation(db) 1

Enter the stopband attenuation(db) 40









RESULT:

Thus IIR filters have been implemented to obtain the PSD of a signal.