

Digital Signal Processing CSE-356

Lab Report

Title :

Design & Simulation of Different types of Infinite Impulse Response (IIR) Filter.

Submitted by :

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Design & Implementation of Finite Impulse Response (FIR) Filter.

Objective:

FIR filter can provide linear phase response, sharp cut off and stability at the expense of process time and memory storage. In this exp, FIR filter is designed based on window method were ideal impulse response of the filter up is multiplied by a smooth window function w(n) of finite duration to alleviate excess ripples.

In frequency sampling method the FIR filter is represented by desired frequency response instead of impulse response. The coefficient of an FIR filter is evaluated as,

H(n)=IDFT{H(K)}, k=0,1,2,3.....N-1 =
$$\frac{1}{N} \sum_{k=0}^{N-1} H(k) e^{j(\frac{2\pi}{N})dk}$$
;

And $\alpha = (N-1)/2$ (i)

Here, H(k) is the desired freq response of the filter of N samples taken at intervals of KFs/N

Let us, consider a low pass FIR filter of passband : 0-5 kHZ, sampling freq, Fs=18kHz and the number of samples, N=9,

Now, KFs/N=K*18/9 = 2kHz.

For the passband of 0-5 kHz, k=0,1 and 2.For stopband, k=4,5,6,7,8

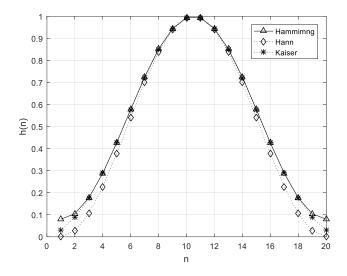
$$H(k) = \begin{cases} 1; & k = 0,1,2 \\ 0; & k = 4,5,6,7,8 \end{cases}$$

In Matlab syntax of FIR filter is based on freq sampling technique is , un = FIR2(N-1,F,H); where H is the vector of desired magnitude response at the corresponding freq points of the vector F. The freq points of the vector F is in normalized form in the range 0 to 1

%plot Hamming, Han & Kaiser window functions used in design of a FIR filter

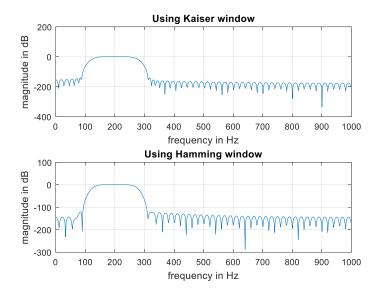
```
beta=5.2;
N=20;
n=1:1:20;
y=hamming(N);
y1=hann(N);
y2=kaiser(N,beta);
plot(n,y,'k^-',n,y1,'kd:',n,y2,'k*:');
```

```
xlabel('n');
ylabel('h(n)');
legend('Hammimng','Hann','Kaiser');
grid on
```



%design a bandpass Fir filter with upper and lower cutoff frequency of 275 %and 125 (use both Hamming and Kaiser window function). $Consider\ sampling\ frequency\ of\ 2KHz.$

```
Fs=2000;
Fn=Fs/2;
N=100; % number of coeffficients
beta =5.65; %beta parameterfor the kaiser window function
fc1=125/Fn; %normalized cutoff frequency of lower side
fc2=275/Fn; %normalized cutoff frequency of upper side
Fc = [fc1 fc2];
hn=fir1(N-1,Fc, kaiser (N,beta)); %fir coefficient;
[H,f]=freqz(hn,1,512,Fs); %freq
mag = 20 *log(abs(H))
subplot(2,1,1)
plot (f, mag)
grid on
xlabel('frequency in Hz')
ylabel('magnitude in dB')
title('Using Kaiser window')
hn=fir1(N-1,Fc,hamming(N));
[H,f]=freqz(hn,1,512,Fs); %freq
mag = 20 *log(abs(H))
subplot(2,1,2)
plot (f, mag)
grid on
xlabel('frequency in Hz')
ylabel('magnitude in dB')
title('Using Hamming window')
```



```
Fs=18; N=9; \\ fts=[0\ 1\ 2\ 3\ 4\ 5\ 6\ 7]/7; \\ Hk=[1\ 1\ 1\ 0\ 0\ 0\ 0\ 0]; \\ b=fir2(N-1,fts,Hk); \\ %b=fir2(N,F,A) \ designs \ an \ nth \ order \ fir \ digital \ filter \ with \ the \ frequency \\ %response \ specially \ by \ vectors \ F
```

%b=fir2(N,F,A) designs an nth order fir digital filter with the frequency
%response specially by vectors F
[h,f]=freqz(b,1,512,Fs);
plot(f*(Fs/N),abs(h));
xlabel('frequency in Hz')
ylabel('magnitude in dB')
grid on

