**Experiment Name:** Design and implementation of finite impulse response (FIR) filter.

**Objective:**

FIR filters can provide linear phase response , sharp cutoff and stability at the expense of process time and memory storage , They can be realized efficiently in hardware,The filter startup transients have finite duration. In this experiment , FIR filter is designed based on window method where ideal impulse response of the filter hd(n) is multiplied by a smooth window function w (n) of finite duration to alleviate excess ripples .

**Source Code:**

% plot hamming, han and kaiser window functions used in design of a FIR filter .% window function 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('Hamming','Hann','Kaiser')

grid on

**Figure:**



**Problem Name:**

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.

**Source Code:**

Fs=2000;

Fn=Fs/2;

N=100;

beta=5.65;

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 coefficients

[H,f]=freqz(hn,1,512,Fs); % Frequency response

mag=20\*log(abs(H)) %conversion in dB

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)) % Fir coefficient

[H,f]=freqz(hn,1,512,Fs); % frequency response of Fir

mag=20\*log(abs(H)); % conversion in dB

subplot(2,1,2)

plot(f,mag),grid on

xlabel('Frequency in Hz')

ylabel('Magnitude in dB')

title('Using Hamming window')

**Figure:**



**Problem Name:**

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

h(n) = IDFT{H(k)}; k= 0,1,2,3,...N-1

here, H(k) is the desired frequency response (Normalized form) 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 frequency , Fs=18 kHz and the number of samples, N=9.

Now, kFs/N=k\*18/9=2 kHz. For the passband of 0-5 kH , k=0,1 and 2 . For stopband k = 4 , 5 , 6 , 7 and 8.So, H(k)= { 1;k=0,1,2

{ 0,k=4,5,6,7,8

In matlab Syntax of FIR filter based on frequency sampling technique is , hn=fir2(N-1,F,H); where H is the vector of desired magnitude response at the corresponding frequency points of the vector F.The frequency points of the vector F is in normalized form in the range of 0 to 1.

**Source Code:**

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);

[h,f]=freqz(b,1,512,Fs);

plot(f\*(Fs/N),abs(h));

xlabel('Frequency in kHz')

ylabel('Magnitude')

Figure:

