DIGITAL SIGNAL PROCESSING LAB

7. FIR FILTER (LP/HP) USING KAISER WINDOW TECHNIQUE

AIM:

To design a FIR filter using Kaiser windowing technique and verify its frequency response.

APPARATUS:

PC with MATLAB

THEORY:

The window method for a causal linear-phase FIR filter is obtained by multipying an ideal filter that has an infinite-duration impulse response (IIR) by a finite-duration window function:

 $h[n] = h_d[n]w[n]$

where h[n] is the practical FIR filter, $h_d[n]$ is the ideal IIR prototype filter, and w[n] is the finiteduration window function. An important consequence of this operation is that the DTFTs of $h_d[n]$ and w[n] undergo circular convolution in frequency

$$H(e^{j\omega}) = \frac{1}{2\pi} \oint_{2\pi} H_d(e^{j\theta}) W(e^{j(\omega-\theta)}) d\theta$$

Kaiser Window:

A very flexible family of window functions has been developed by Kaiser. These windows are nearly optimum in the sense of having the largest energy in the mainlobe for a given peak sidelobe level.

They are closely related to the prolate spheroidal wavefunctions which are the optimum timelimited, continuous-time functions in a similar sense. The Kaiser windows are of the form,

$$w_{K}(n) = \frac{I_{0} \left[\beta \sqrt{1 - (1 - 2n/M)^{2}}\right]}{I_{0} \left[\beta\right]}, \qquad n = 0, 1, \dots, M,$$

where $I_0[$] is the modified zeroth-order Bessel function of the first kind and β is a shape parameter determining the tradeoff between the mainlobe width and the peak sidelobe level. Typical values for β are in the range 4 < β < 9. $I_0[$] is most easily computed from its power series expansion

$$I_0[x] = 1 + \sum_{m=1}^{\infty} \left[\frac{(x/2)^m}{m!} \right]^2,$$

PROCEDURE:-

- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window \Figure window

PROGRAM:

% To design of FIR filters using Kaiser window techniques.

clc;

```
clear all;
close all;
% Low Pass Filter
n=100;
fp=200;
fq=300;
fs=1000;
fn=2*fp/fs;
beta = input('Enter beta value ') % use 0.5 to 2.5
window=kaiser((n+1), beta);
b=fir1(n,fn,window);
[H W]=freqz(b,1,128);
subplot(2,1,1);
plot(W/pi,abs(H));
title('magnitude response of lpf');
ylabel('gain in db----->');
xlabel('normalized frequency---->');
grid on
subplot(2,1,2);
plot(W/pi, angle(H));
title('phase response of lpf');
ylabel('angle---->');
xlabel('normalized frequency---->');
grid on
% Highpass Filter
n=100;
fp=300;
fq=200;
fs=1000;
fn=2*fp/fs;
window=kaiser((n+1), beta);
b=fir1(n,fn,'high',window);
[H W]=freqz(b,1,128);
figure(2)
subplot(2,1,1);
plot(W/pi,abs(H));
title('mag res of hpf');
ylabel('gain in db---->');
xlabel('normalized frequency---->');
grid on
subplot(2,1,2);
```

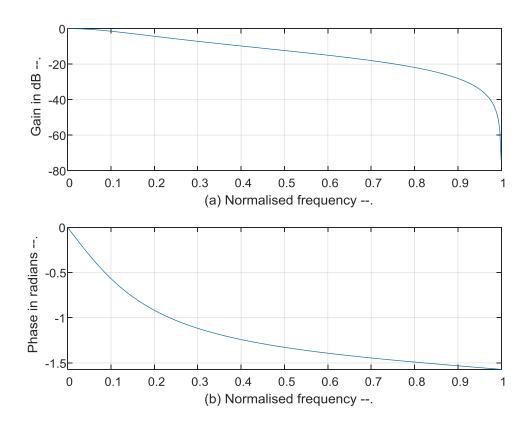
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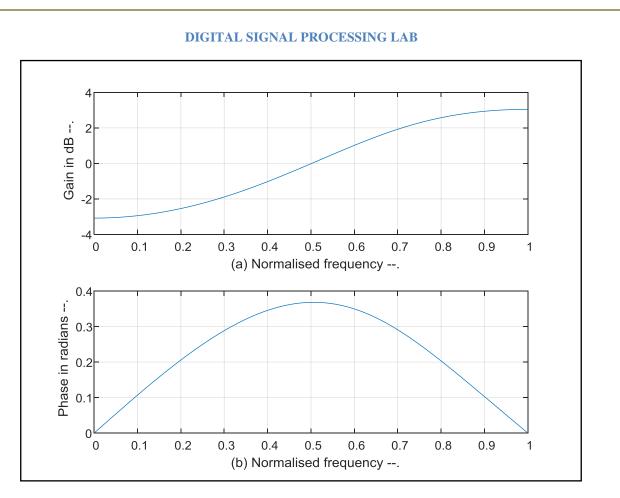
```
plot(W/pi,angle(H));
title('phase res of hpf');
ylabel('angle---->');
xlabel('normalized frequency---->');
grid on
```

INPUT::

enter the passband ripple: 0.02 enter the stopband ripple: 0.01 enter the passband frequency: 1000 enter the stopband frequency: 1500 enter the sampling frequency: 10000 enter the beta value: 1.2

OUTPUT:





RESULT:

Thus the MATLAB program for FIR LP\HP using Triangular window Techniques was executed and its frequency response is also verified.