Institute of Information Technology (IIT)

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Lab Report: 06

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EXP.NO: 06

Name Of the Experiment: Implementation of lp fir filter.

AIM: To implement an LP FIR filter for a given sequence.

SOFTWARE: MATLAB

THEORY: FIR filters are digital filters with a finite impulse response. They are also known as non-recursive digital filters, as they do not have feedback.

An FIR filter has two important advantages over an IIR design:

- First of all, an FIR filter's design does not include a feedback loop. A feedback loop is not present, hence, an FIR filter is by nature stable. In the meantime, we need to examine the stability of an IIR filter.

- An FIR filter's design does not have a feedback loop, to start with. An FIR filter is intrinsically stable since it lacks a feedback loop. In the interim, we must examine the stability of an IIR filter.

FIR Filter Design

An FIR filter is designed by finding the coefficients and filter order that meet certain specifications, which can be in the time domain (e.g., a matched filter) and/or the frequency domain (most common). Matched filters perform a cross-correlation between the input signal and a known pulse shape. The FIR convolution is a cross-correlation between the input signal and a time-reversed copy of the impulse response. Therefore, the matched filter's impulse response is "designed" by sampling the known pulse shape and using those samples in reverse order as the coefficients of the filter.

When a particular frequency response is desired, several different design methods are common:

1. Window design method

2. Frequency Sampling method

3. Weighted least squares design

Window Design Method:

In the window design method, one first designs an ideal IIR filter and then truncates the infinite impulse response by multiplying it with a finite-length window function. The result is a finite impulse response filter whose frequency response is modified from that of the IIR filter.

Window Name	Transition Width (Approximate)	Transition Width (Exact)	Min. Stop Band Attenuation	MATLAB Command
Rectangular	4π/M	8.1π/M	21 dB	B = FIR1(N, Wn, boxcar)
Bartlett	8π/M	1.6π/M	25 dB	B = FIR1(N,Wn, bartlett)
Hanning	8π/M	2.6π/M	44 dB	B=FIR1(N,Wn, hanning)
Hamming	8π/M	6.6π/M	53 dB	B=FIR1(N,Wn,hamming)
Blackman	12π/M	11π/M	74 dB	B=FIR1(N,Wn,blackman)

Methodology

Algorithm:

Step I : Enter the pass band frequency (fp) and stop band frequency (fq).

Step II : Get the sampling frequency (fs), length of window (n).

Step III : Calculate the cut off frequency, fn

Step IV : Use boxcar, hamming, blackman Commands to design window.

Step V : Design filter by using above parameters.

Step VI : Find frequency response of the filter using matlab command freqz.

Step VII : Plot the magnitude response and phase response of the filter.

Flow Chart:

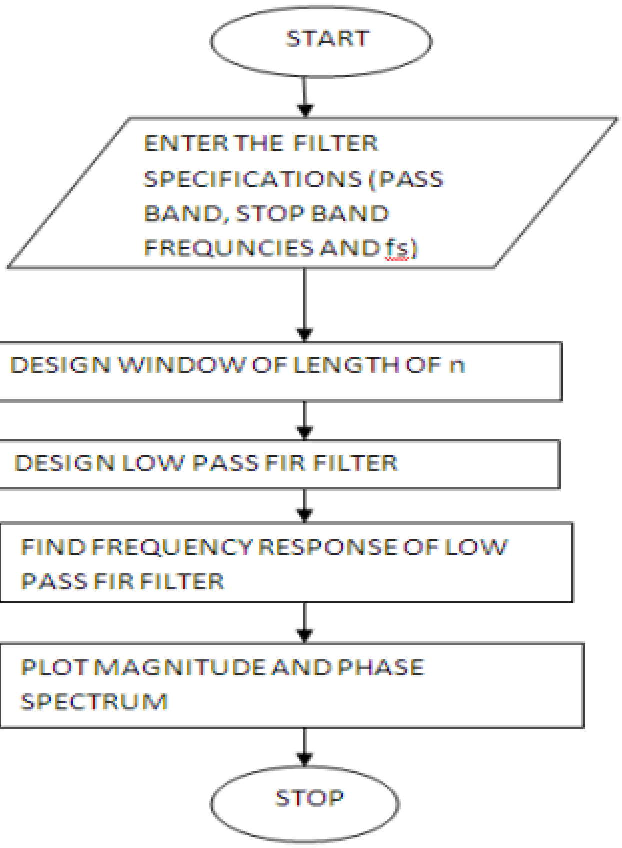

Fig-01: Flowchart

PROGRAM FOR RECTANGULAR WINDOW:

clc;

```
clearvars;
close all;

n = 20;
fp = 200;
fq = 300;
fs = 1000;
fn = 2 * fp / fs;
window = rectwin(n + 1);
 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');
subplot(2, 1, 2);
plot(W/pi, angle(H));
title('Phase Response of LPF');
ylabel('Phase Angle');
xlabel('Normalized Frequency');
```

OUTPUT WAVEFORMS:

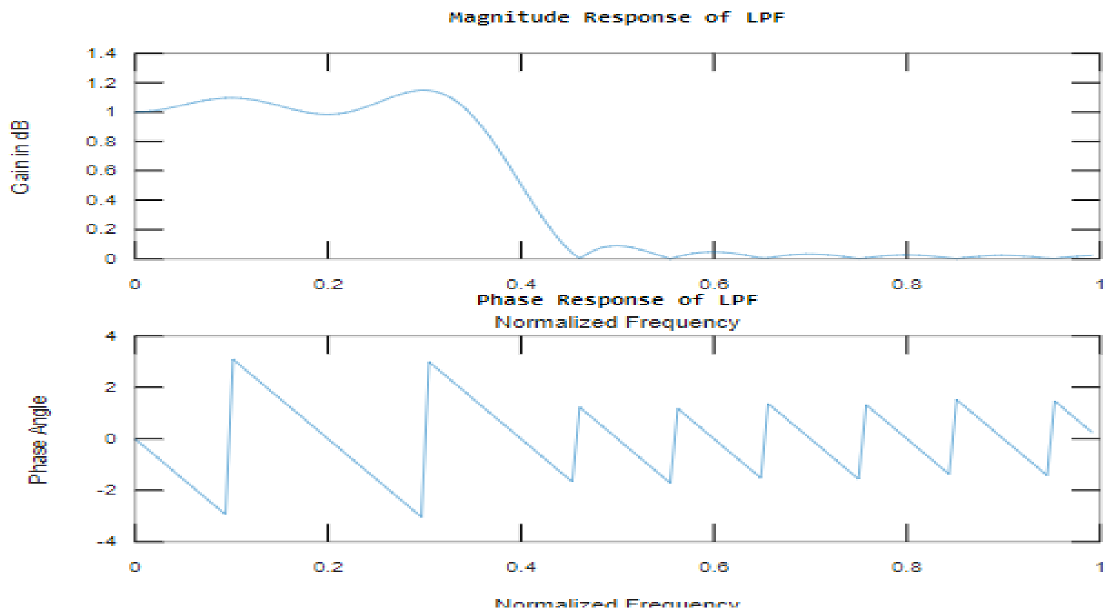

Fig. 02: wavelength for a Rectangular Window

RESULTS:

The amplitude and phase responses of the filter are then plotted by the software. The phase response is displayed in degrees, while the magnitude response is plotted in decibels (dB). In the program's output:

The filter's abrupt cutoff is seen at the lower cutoff frequency, according to the magnitude response plot. When frequency rises, the magnitude response eventually drops to zero.

The filter has a linear phase response, as seen by the phase response figure. At the lower cutoff frequency, the phase response is 0 and grows linearly toward 90 degrees as the frequency rises.

PROGRAM FOR HANNING WINDOW:

```
clc;

clear all;

close all;

n=20;

fp=200;
```

```
fq=300;

fs=1000;

fn=2*fp/fs;

window=hamming(n+1);

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------>');

 subplot(2,1,2);

plot(W/pi,angle(H));

title('phase response of lpf');

ylabel('angle-------->');

xlabel('normalized frequency------>');
```

OUTPUT WAVEFORMS:

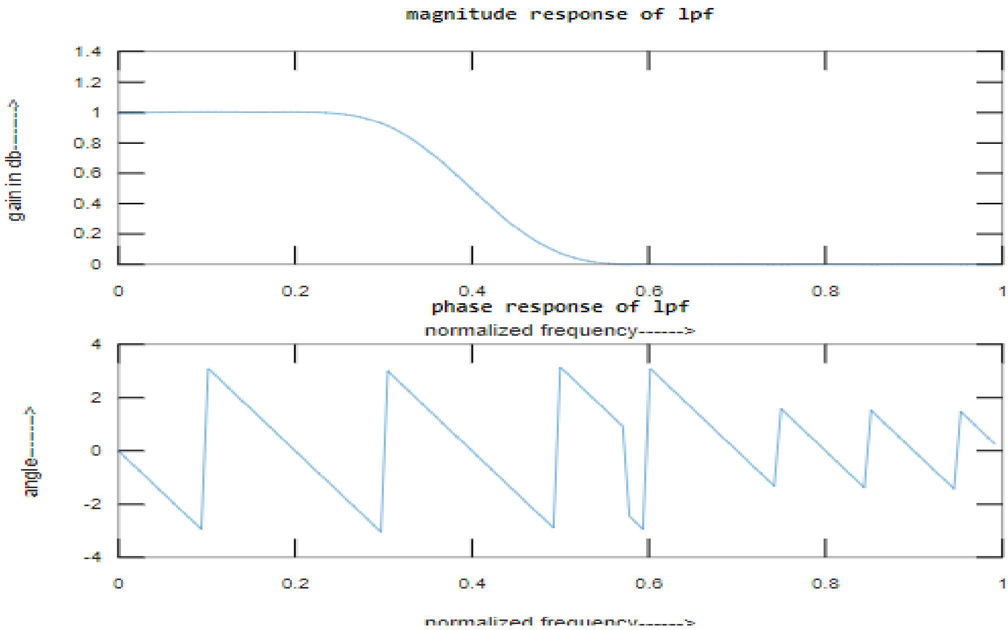

Fig. 03: wavelength for Hammering window

RESULTS:

The amplitude and phase responses of the filter are then plotted by the software. The phase response is displayed in degrees, while the magnitude response is plotted in decibels (dB). In the program's output:

The filter's abrupt cutoff is seen at the lower cutoff frequency, according to the magnitude response plot. When frequency rises, the magnitude response eventually drops and approaches zero.

The filter has a linear phase response, as seen by the phase response figure. At the lower cutoff frequency, the phase response is zero, and as the frequency rises, it grows linearly toward 90 degrees.

DISCUSSION:

The program's findings demonstrate that the rectangular window filter has larger sidelobes than the Hamming window filter. The energy in the sidelobes is decreased as a result of the Hamming window's tapering edges. In general, for applications where aliasing is a concern, the Hamming window is preferable to the rectangular window. For situations where computing complexity is a factor, the rectangular window could be a superior option due to its simpler and quicker implementation.

VIVA QUESTIONS:

1. Define the filter.

A filter is a device or system used in signal processing that modifies a signal's frequency content. Filters have the ability to selectively let some frequencies pass while attenuating or blocking others.

2. What are the different types of filters?

Different filters include:

· **Low-pass filter:** allows high-frequency components to be attenuated while allowing low-frequency components to flow.

·**High-pass filter:** Low-frequency components are attenuated while high-frequency components are allowed to pass.

·**Band-pass filter:** Allows a specific range of frequencies (a band) to pass while attenuating frequencies outside that range.

·**Band-stop filter (notch filter):** restricts the passage of frequencies outside of a certain range while attenuating those frequencies inside that range.

· **All-pass filter:** This doesn't affect the magnitude of the signal but shifts its phase.

A **Notch filter:** is usually employed to eliminate specific interference or noise since it attenuates a fairly limited range of frequencies.

3. Why are FIR filters generally preferred over IIR filters in multirate (decimating

and interpolating) systems?

In multirate systems like decimating and interpolating systems, FIR (Finite Impulse Response) filters are typically favored over IIR (Infinite Impulse Response) filters since they have linear phase characteristics and don't cause phase distortion to the signal. Phase distortion in multirate systems can result in issues such as signal distortion, loss of signal quality, and even information loss. FIR filters are better suited for situations where phase precision is important since they may be made to have acceptable linear phase characteristics.

4. What is the difference between IIR and FIR filters?

IIR Filters (Infinite Impulse Response Filters): These filters employ feedback in their construction, which enables them to need fewer filter coefficients to provide the appropriate frequency response. Compared to FIR filters, they are typically more computationally effective for generating a certain response. They could be less stable than FIR filters and can produce phase distortion. These filters employ feedback in their construction, which enables them to need fewer filter coefficients to provide the appropriate frequency response. Compared to FIR filters, they are typically more computationally effective for generating a certain response. They could be less stable than FIR filters and can produce phase distortion.

FIR Filters (Finite Impulse Response Filters): These filters only use feedforward coefficients and have no feedback. They are inherently stable and can have linear phase responses, making them suitable for applications where phase accuracy is important. However, they might require more filter coefficients to achieve the same response as an IIR filter.

5. Differentiate between ideal filter and practical filter responses.

Ideal Filter Response: These filters use feedback in their design, allowing them to produce the desired frequency response with fewer filter coefficients. They are often more computationally efficient for producing a certain response when compared to FIR filters. They have the potential to exhibit phase distortion and be less stable than FIR filters.

Practical Filter Response: Both FIR and IIR practical filters have limits brought on by things like filter length, implementation restrictions, and non-ideal components. They show some degree of stopband attenuation, passband ripple, and transition bandwidth. It takes time for the passband to switch over to the stopband.

6. What are the filter specifications required to design the analog filters?

The filter specifications required to design analog filters typically include:

· Filter type (low-pass, high-pass, etc.)

· Cutoff frequency (or frequencies for band-pass/stop filters)

· Passband ripple and/or stopband attenuation

· Transition bandwidth

· Filter order or length

· Phase requirements (for linear phase filters)

· Tolerances and design constraints

7. What is meant by the frequency response of a filter?

The behavior of a filter with regard to various frequencies in the input signal is represented by the frequency response of the filter. It demonstrates how the filter's effects on the input signal's components' amplitude and phase vary with frequency.

8. What is meant by magnitude response?

Magnitude response: The frequency response of a filter illustrates how it responds to different frequencies contained in the input signal. It illustrates the frequency dependence of the filter's effects on the amplitude and phase of the input signal components.

9. What is meant by phase response?

Phase response: A filter's phase response is a graph showing the phase shift it introduces as a function of frequency. It illustrates how the phase connections between various frequency components of the input signal are impacted by the filter.

10. Difference between FIR low-pass filter and high-pass filter.

1. The difference between an FIR low-pass filter and an FIR high-pass filter lies in the frequency ranges they pass and attenuate:

· **FIR Low-Pass Filter:** By attenuating high-frequency components, this filter lets low-frequency components flow through. It is intended to filter out higher frequencies while retaining a signal's low-frequency content.

· **FIR High-Pass Filter:** Low-frequency components are muted while high-frequency components can travel through this filter. The high-frequency component of a signal is extracted or emphasized, while the lower frequencies are removed.

REFERENCE:

[1]Geeksforgeeks, “phenomenon-rectangular-and-hamming-window-implementation,”, Aug. 27, 2021. Available: https://www.geeksforgeeks.org/gibbs-phenomenon-rectangular-and-hamming-window-implementation/#:~:text=The%20Rectangular%20is%20not%20widely,of%20the%20signal's%20frequency%20spectrum. [Accessed: Aug. 28, 2023]