Experiment 8: Linear Phase F I R Filter Design using Windowing Method

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Experiment No.	8

AIM:	Linear Phase FIR Filter design using window function.
OBJECTIVE:	To design the digital filter using windowing technique and study the spectrum of the filter.
INPUT SPECIFICATION:	**For LPF / HPF filter Design :** 1. Pass band Attenuation (Ap) 2. Stop band Attenuation (As > 40 dB) 3. Pass band Frequency (Fp) in Hz 4. Stop band Frequency (Fs) in Hz 5. Sampling Frequency in Hz 6. **For BPF / BSF filter Design :** 1. Pass band Attenuation (Ap) 2. Stop band Attenuation (As) 3. Pass band Frequency (Fp1, Fp2) in Hz 4. Stop band Frequency (Fs) in Hz 5. Sampling Frequency in Hz
PROBLEM DEFINITION:	1. Accept the input specifications for LPF, HPF, BPF and BSF. 2. Design the filter by selecting appropriate window function. 3. Plot magnitude spectrum and phase spectrum and verify the value of Ap and As in pass band and stop band from the magnitude spectrum. 4. Comment on Phase Spectrum.

ALGORITHM :	1. Record Audio Signal in the presence of noise ==>. x[n].
	 2. Play the recorded signal x[n] and observe the quality of sound.
	 3. Design FIR Low Pass Filter using.
	    Assume Ap and As,
	    Fpass = 3000 Hz. Fstop = 4000 Hz Assume Fs
	 4. Filter the audio signal x[n]
	    i.e. Process the input and obtain output signal y[n]
	 5. Play the filtered signal [n] and observe the quality of sound

RESULTS OF CODE

```
Step 1: Accept user input specifications
Fs = 1000
Fp = 0.1000
Fs_stop = 0.4000
Ap = 1
As = 46
N = 9
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Step 2: Select appropriate window function
Selected window: hamming
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```
Step 3: Normalize frequencies
Wp = 2.0000e-04
Ws = 8.0000e-04
```
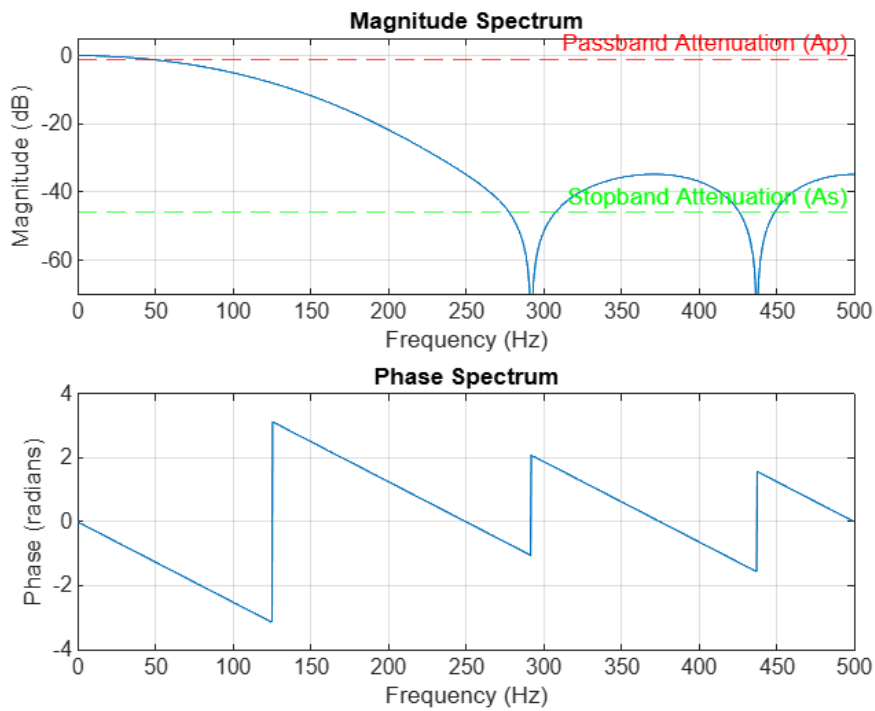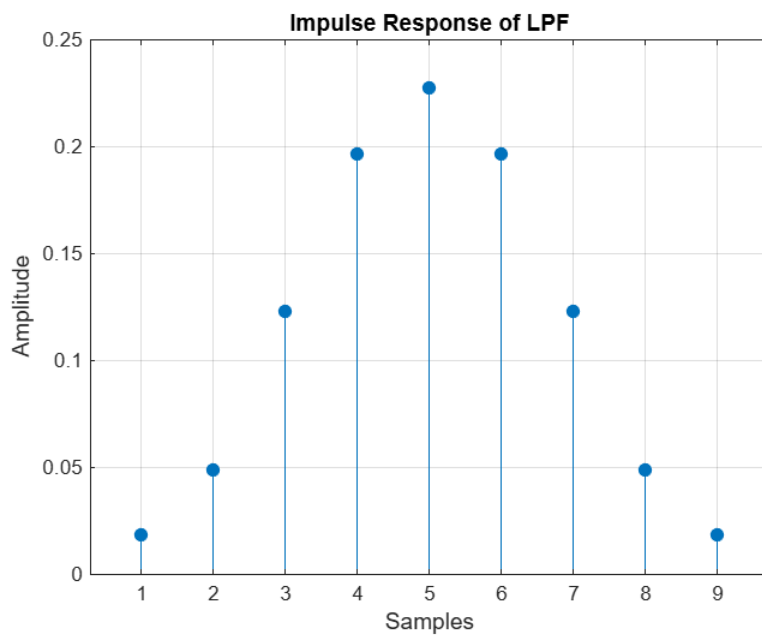
```
Step 4: Design the LPF using the selected window
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```
Step 5: Frequency response
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Step 6: Impulse response

Impulse Response of LPF

Observe the phase spectrum for linearity. Linear phase indicates symmetry in the impulse response.
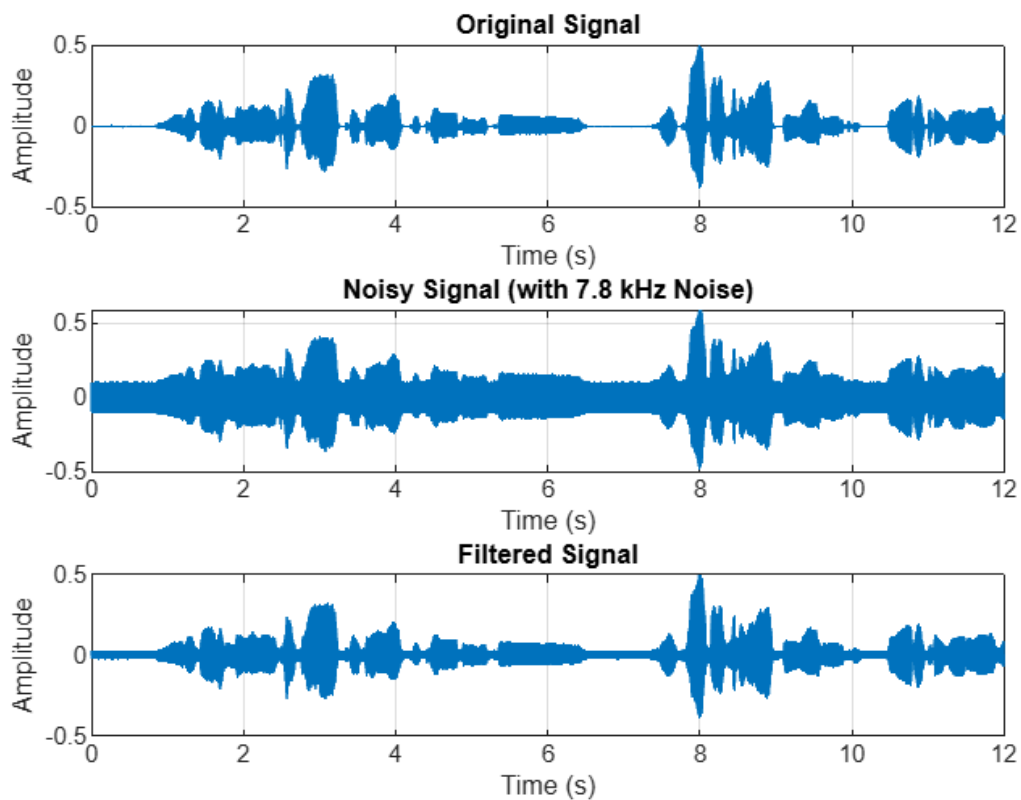Step 1: Load recorded audio
Step 2: Add 7.8 kHz noise
Step 3: noisy audio for 12 seconds
Step 4: Pass the noisy signal through the LPF
Step 5: Play filtered audio for 12 seconds
Step 6: Plot the signals

Original Signal / Noisy Signal (with 7.8 kHz Noise) / Filtered Signal

Observe the plots to analyze the effect of filtering.

RESULT ANALYSIS :	1. By following the procedure given in the document we created a FIR filter of order 9 2. The phase of the filter that we created is linear in nature that means we have created correct filter 3. The graphs at the last demonstrated that the filter is working properly because it eliminated the noise and recovered the original signal 4. If we increase the order of the filter then we will see better result
CONCLUSION:	1. The designed FIR filter exhibited a linear phase response, validating the correctness of the filter design process. This ensures minimal phase distortion in the filtered signal. 2. The selection of window was done on the basis of value of As and the window selection plays a vital role in filter design as it controls the width of main and side lobs 3. The FIR low-pass filter effectively attenuated the 7.8 kHz noise while preserving the original signal, demonstrating its capability to remove high-frequency interference. 4. Increasing the filter order results in sharper transition bands and improved attenuation, enhancing the filter's performance at the cost of increased computational complexity. 5. The experiment showcased the practical application of windowing techniques in FIR filter design for real-world audio signal processing, ensuring high-quality signal reconstruction.