

Experiment 9: Linear Phase F I R Filter Design using Frequency sampling method

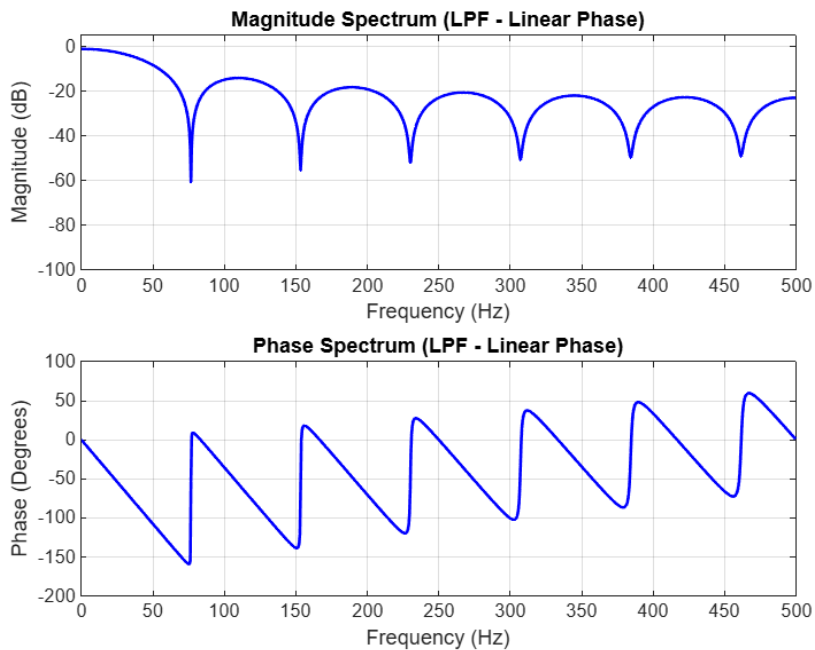
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Experiment No.	9

AIM:	FIR Filter design using Frequency Sampling Method.
OBJECTIVE:	The objective of this experiment is to design the digital filter using frequency sampling method.
INPUT SPECIFICATION:	<p>For LPF / HPF filter Design :</p> <ol style="list-style-type: none"> 1. Pass band Attenuation (A_p) 2. Stop band Attenuation ($A_s > 40$ dB) 3. Pass band Frequency (F_p) in Hz 4. Stop band Frequency (F_s) in Hz 5. Sampling Frequency in Hz <p>For BPF / BSF filter Design :</p> <ol style="list-style-type: none"> 1. Pass band Attenuation (A_p) 2. Stop band Attenuation (A_s) 3. Pass band Frequency (F_{p1}, F_{p2}) in Hz 4. Stop band Frequency (F_s) in Hz 5. Sampling Frequency in Hz
PROBLEM DEFINITION:	<ol style="list-style-type: none"> 1. Accept the input specifications for two cases ; one for LPF/BPF and second for HPF/BSF. Assume any appropriate value for filter order N. 2. Design Linear Phase as well as Non Linear Phase FIR filter. 3. Plot Magnitude Spectrum and Phase Spectrum and verify the value of A_p and A_s in pass band and stop band from the spectrum. 4. If the design parameters are not satisfied from the spectrum, change the value of filter order N adaptively.
ALGORITHM :	<ol style="list-style-type: none"> 1. Record Audio Signal in the presence of noise $\Rightarrow x[n]$. 2. Play the recorded signal $x[n]$ and observe the quality of sound. 3. Design IIR Low Pass Filter using. Assume A_p and A_s, $F_{pass} = 3000$ Hz. $F_{stop} = 4000$ Hz Assume F_s 4. Filter the audio signal $x[n]$ i.e. Process the input and obtain output signal $y[n]$ 5. Play the filtered signal $y[n]$ and observe the quality of sound

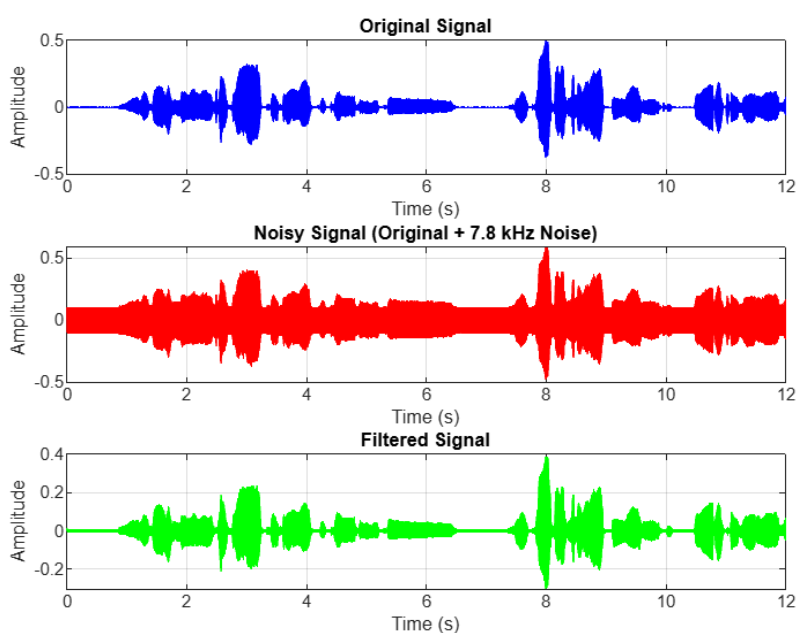
RESULTS OF CODE

```
1000
Enter Pass Band Attenuation (Ap) in dB:
1
Enter Stop Band Attenuation (As) in dB (> 40):
49
Enter Pass Band Frequency (Fp) in Hz:
0.1
Enter Stop Band Frequency (Fs) in Hz:
0.4
Enter filter Order (N):
12
```

Designing Low Pass Filter (LPF)



```
Playing original signal...
Playing noisy signal...
Playing filtered signal...
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RESULT ANALYSIS :	<ol style="list-style-type: none"> 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:	<ol style="list-style-type: none"> 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. Use Frequency Sampling Method when you need precise control over the frequency response. 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.