

Discussion #3

Design an FIR filter with linear phase that removes a 60 kHz interferer.

Use one of the existing filters you know. Don't use any computer-based design algorithms.

Give the sampling rate you're using.

Approach #1: Use an averaging filter with L coefficients.

$$y[n] = \frac{1}{L} (x[n] + x[n-1] + \dots + x[n-(L-1)]) \text{ for } n \geq 0.$$

This filter has linear phase because the filter coefficients (which equal $\frac{1}{L}$) are even symmetric about the midpoint.

~~of the input~~ To be LTI, the initial conditions $x[-1], x[-2], \dots, x[-(L-1)]$ must be zero. We have to have

LTI properties satisfied before we can talk about linear phase.

An L -point FIR averaging filter zeros out frequencies $\frac{2\pi}{L}m$ where $m=1, 2, \dots, L-1$. We'll let $m=1$.

$$\frac{2\pi}{L} = 2\pi \frac{f_0}{f_s} \rightarrow f_s = L f_0$$

To satisfy the Sampling Theorem, $f_s \geq 2f_0$. So, $L \geq 3$.

For $f_0 = 60$ kHz, f_s can be 180 kHz ($L=3$), 240 kHz ($L=4$)...

Approach #2: Same as Approach #1 but we allow another frequency zeroed out by the FIR averaging filter to be used; i.e., $2 \leq m \leq L-1$:

$$\frac{2\pi}{L} m = 2\pi \frac{f_0}{f_s} \leadsto f_s = \frac{L}{m} f_0$$

To satisfy the Sampling Theorem, $f_s > 2 f_0$.

$$\frac{L}{m} f_0 > 2 f_0 \leadsto \frac{L}{m} > 2$$

$\frac{L}{m}$	m	f_s
3	1	180 KHz
4	1	240 KHz
5	{1, 2}	{300, 250} KHz
6	{1, 2}	{360, ¹⁸⁰ ~~250~~ } KHz
7	{1, 2, 3}	{420, 210, 140} KHz
⋮	⋮	⋮

Reference: Prof. Brian Evans, "Designing Averaging Filters", EE 4455 Handout, Fall 2026, <https://tinyurl.com/5dsu8pzu>.