

Homework #4

Finite Impulse Response (FIR) Filters

Assigned on Saturday, October 5, 2024

Due on Friday, October 11, 2024, by 11:59 pm via Canvas submission

Late homework is subject to a late penalty of two points per minute late.

Reading: McClellan, Schafer and Yoder, *Signal Processing First*, 2003, Chapter 5 (all).

See the book's [companion Web site](#) for demos and other supplemental information.

Web site contains solutions to hundreds of worked problems.

E-mail address for Mr. Elyes Balti (TA) is ebalti@utexas.edu. Please consider posting questions on [Ed Discussion](#)., which can be answered by anyone in the class. You can post anonymously.

Lecture and office hours follow:

<i>Time Slot</i>	<i>Monday</i>	<i>Tuesday</i>	<i>Wednesday</i>	<i>Thursday</i>	<i>Friday</i>
11:00 am		Evans (ECJ 1.204)		Evans (ECJ 1.204)	
11:30 am		Evans (ECJ 1.204)		Evans (ECJ 1.204)	
12:00 pm		Evans (ECJ 1.204)		Evans (ECJ 1.204)	
12:30 pm					
1:00 pm			Balti (EER 5.652)		
1:30 pm			Balti (EER 5.652)		
2:00 pm	Evans (EER 6.882; Zoom)		Balti (EER 5.652)		Balti (EER 4.650)
2:30 pm	Evans (EER 6.882; Zoom)		Balti (EER 5.652)		Balti (EER 4.650)
3:00 pm	Evans (EER 6.882; Zoom)				Balti (EER 4.650)
3:30 pm			Evans (EER 6.882; Zoom)		Balti (EER 4.650)
4:00 pm			Evans (EER 6.882; Zoom)		
4:30 pm			Evans (EER 6.882; Zoom)		
5:00 pm				Balti (EER 4.702)	
5:30 pm				Balti (EER 4.702)	
6:00 pm				Balti (EER 4.702)	
6:30 pm				Balti (EER 4.702)	

[ECE 313 tutoring](#) is available on Monday and Thursdays 7-10pm in person in EER 0.814.

As stated on the course descriptor, “Discussion of homework questions is encouraged. Please be sure to submit your own independent homework solution.”

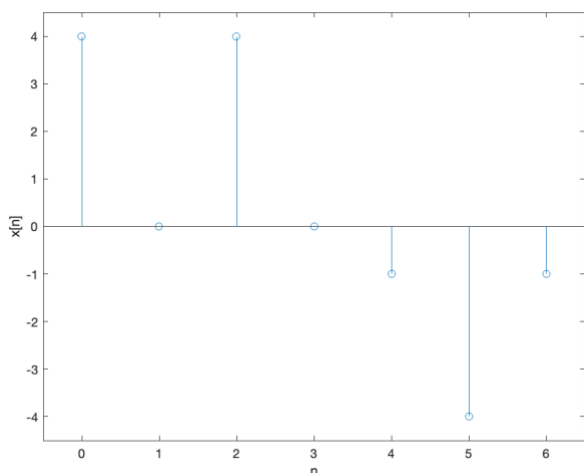
NOTE: In your solutions, please put all work for problem 1 together, then all work for problem 2 together, etc. Please see additional homework guidelines on the [homework page](#).

1. Sampling & Aliasing. 28 points.

Signal Processing First, problem P-4.12, page 98.

2. Finite Impulse Response (FIR) Filter. 36 points.

Signal Processing First, problem P-5.3, page 126, but use the following input signal $x[n]$:



```
x = [ 4 0 4 0 -1 -4 -1 ];  
n = 0 : length(x) - 1;  
stem(n, x);  
xlim( [-0.5, 6.5] );  
ylim( [-4.5, 4.5] );  
xlabel( 'n' );  
ylabel( 'x[n]' );
```

Note that $x[n] = 0$ for $n < 0$ or $n > 6$

3. System Identification. 36 points.

A common problem that arises in audio and other systems is characterizing an unknown system. Consider an audio system with a speaker transmitting sound in a concert hall that is recorded by a microphone, and we'd like to characterize the acoustic response in the concert hall.

If we model an unknown system as a linear time-invariant finite impulse response (FIR) filter, we can try to infer its FIR filter coefficients b_k (i.e. the impulse response of the FIR filter). In practice, the unknown system will likely not be linear and time-invariant, but sometimes, a linear time-invariant system model captures useful information about the unknown system for the application at hand.

For the concert hall acoustics example, if we know the FIR filter coefficients for the linear time-invariant model of the concert hall acoustics, then we can mimic the effect of the concert hall by using

its FIR coefficients to filter a music track and the output will sound as if it was played in the concert hall. Several audio systems with a “concert hall” effect will list several different concert halls.

Here is the input-output relationship for a linear time-invariant FIR filter with N coefficients with input $x[n]$ and output $y[n]$:

$$y[n] = b_0 x[n] + b_1 x[n - 1] + b_2 x[n - 2] + \dots + b_{N-1} x[n - (N - 1)]$$

In advance, we don't know the value of N .

If we could input a known test signal for $x[n]$ and observe $y[n]$, we can attempt to compute the FIR coefficients b_k by *deconvolution*. In the test setup, we would assume that a laptop is sending the test signal to the speakers and the same laptop is receiving the output from the microphone. We'll assume that the test will begin at time $n = 0$ and perform deconvolution in the time domain.

The first output value (i.e. when $n = 0$) is

$$y[0] = b_0 x[0] + b_1 x[-1] + b_2 x[-2] + \dots + b_{N-1} x[-(N - 1)]$$

For linear time-invariant systems, it is a necessary (but not sufficient) condition for the system to be “at rest”, which means that all the initial conditions $x[-1], x[-2], \dots, x[-(N - 1)]$ must be zero.

$$y[0] = b_0 x[0] + b_1 x[-1] + b_2 x[-2] + \dots + b_{N-1} x[-(N - 1)]$$

Since know $x[n]$ and $y[n]$ in our test setup, we have one equation and one unknown at $n = 0$:

$$y[0] = b_0 x[0]$$

and we can compute

$$b_0 = \frac{y[0]}{x[0]}$$

For this calculation to be valid, the first value of the test signal, $x[0]$, cannot be zero.

(a) Develop an algorithm to compute the remaining values of b_k assuming you know the value of N .

(b) By hand, compute the values of b_k given and

- i. input signal $x[n]$ with non-zero values [1 2 3 4 5]
- ii. output signal $y[n]$ with non-zero values [1 1 1 1 1 -5]

Use the MATLAB command `conv` to convolve $x[n]$ and b_k to make sure that the result is $y[n]$.

(c) Write a MATLAB program for your algorithm in (a) and apply it to the signals in part (b) to compute b_k . In your algorithm, stop computing values of b_k when $|b_k - b_{k-1}| \leq 10^{-7} |b_k|$. That way, we don't need to know the value of N in advance. I have the stopping criterion as

$$|b_k - b_{k-1}| \leq 10^{-7} |b_k| \text{ instead of } \left| \frac{b_k - b_{k-1}}{b_k} \right| \leq 10^{-7} \text{ to avoid a possible division by zero error.}$$

Returning to the room acoustics example, the input test signal could be a chirp signal that sweeps all audible frequencies.