# Wentworth Institute of Technology

## Electrical Engineering and Technology Department

## ELMC 870

## ELECTROMECHANICAL SYSTEMS II

## Laboratory Exercise # 11

## FIR FILTER #1

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**WENTWORTH INSTITUTE OF TECHNOLOGY**

**ELECTRONICS AND MECHANICAL ENGINEERING DEPARTMENT**
**ELMC 870 ELECTROMECHANICAL SYSTEMS II**
**SPRING SEMESTER**
**LABORATORY # 11**
**FIR FILTER #1**

**Purpose:**

- This the first Lab on FIR Filter.
- In this lab you will observe the characteristic of FIR filter.
- Develop the frequency of a signal by SPTOOL (MATLAB)
- Develop the frequency of FIR Filter
- You will build FIR filters by MATLAB SIMULINK

## Procedure:

### A. Observe the FIR Filter Process.

**Overview of Filtering:**

For this lab, we will define an FIR filter as a discrete time system that converts an input signal x[n] into an output signal y[n] by means of weighted summation:

$$y[n] = \sum_{k=0}^{M} b_k x[n-k] \text{..........(1)}$$

Equation (1) gives a rule for computing the n th value of the output sequence from certain values of input sequence. The filter coefficients [$b_k$] are constants that define the filter behavior.
A complete DSP system is shown below:

**Digital FIR Filter**

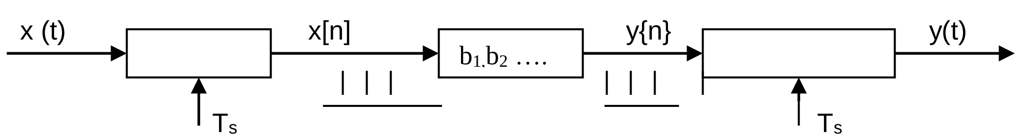



Figure #1

From the figure above we can see that x(t) = Continuous function
x[n] = Discrete function
y[n] = Output from the FIR Filter Discrete Function

y(t) = Filtered signal continuous
$F_s = 1/T_s$ = Sampling Frequency

Problem #1

A linear time invariant system is described by the difference equation:

$y[n] = 2x[n] – 3x[n-1] + 2x[n-2]$

(a) Determine the FIR Filter Coefficients
(b) Determine the response of this system to a unit impulse input, i.e., find the output $y[n] = h[n]$ when the input is $x[n] = \delta [n]$. Plot $h[n]$ as a function of n.

(c) When the input to this system is

$$x[n] = \begin{cases} 0 & n < 0 \\ n+1 & n = 0,1,2 \\ 5-n & n = 3,4 \\ 1 & n \geq 5 \end{cases}$$

(a) Compute the values of $y[n]$ over the range $0 \leq n \leq 10$
Do by convolution method.
(b) Compute by $y[n]$ by MATLAB and compare the results.
(c) From the previous part plot both $x[n]$ and $y[n]$

Problem #2

A DSP system filter is described by the difference equation

$$y[n] = 1/3\ (x[n] + x[n-1] + x[n-2]\ )$$

(a) Determine the filter coefficients.
(b) What is $h[n]$, the impulse response of this system?
(c) Generate a signal in MATLAB workspace with the following specification:
$x(t) = 8\cos( 2\pi\ 100\ t)$ Take 50 samples of $x[n]$
(d) Determine output $y[n]$ by convolution
(e) Plot the output $y(t)$. What is the frequency of the output $y(t)$ ?
(f) Change the frequency to 325 Hz and determine $y[n]$ by convolution and plot the output from the filter.
(g) Conclude what type of filter it is.

**B. Develop the frequency response of a signal by SPTOOL**

Before you design a filter, you need to know the frequency response of the signal you are processing through the filter. After that you need to design a filter for filtering proper frequencies of the incoming signal. Then you need to find the frequency response of a filter you have designed. We will learn all the steps one by one.

**Frequency Response of incoming signal by SPTOOL**

**Problem # 3**

A DSP system filter is described by the difference equation

$y[n] = 1/3\ (x[n] + x[n-1] + x[n-2]\ )$

(a) Generate a signal in MATLAB workspace with the following specification:
    $x(t) = 8\cos(\ 2\pi\ 100\ t)$    Take 1000 samples of x[n]

    Plot x(t) to verify that it is 100 Hz signal

Develop frequency response of the signal by SPTOOL. Instructor will show how to generate frequency response by SPTOOL Check whether the signal is 100 Hz.

(b) Determine output y[n] through the filter given above by convolution
(c) Develop frequency of response the signal by SPTOOL to check whether the signal is filtered.
(d) Change the frequency to 325 Hz and determine y[n] by convolution and plot the output y[n]from the filter.
(e) Develop the frequency response of y[n] to verify the signal is filtered
(f) Comment on your observation

Problem # 4

(a) Create addition of two signals 100 Hz and 325 Hz xx[n] with sample frequency of 1000 Hz and develop frequency response the composite signal to verify the frequencies.
(b) Process the two signals through the filter in Problem # 3 and develop the frequency response by SPTOOL. Write your conclusion.

## C. <u>AUDIO PROCESSING BY MATLAB</u>

### 1. Playback of a Sampled Signal through D/A converter

The file ‘handel’ in the MATLAB contains a segment of a sampled signal. Load this digitized signal in your MATLAB file. Play the file using ‘wavplay’command from inside of MATLAB. Following is the MATLAB command to load and play the segment of the sampled signal.

```
load handel          %load the sampled signal as y vector

y                    % will show the digitized form of y vector.

wavplay(y,8192)      % this will playback the sampled signal. The music was sampled
                     at % 8192 Hz  and played back at a sample rate 8192.
```

If you play back the signal at a rate of 16000 how the sampled signal will sound? Play the signal at the rate of 16000 Hz, listen, and comment. Do the same for sample rate of 4000 Hz and comment on the sound.

**i. Audio Scaling**

To scale an audio file the soundsc() command is used. This allows for the modification of an audio signal's amplitude or frequency.

```
soundsc(y,Fs);
```

To increase the volume of the audio track you can multiple the variable it is stored in by a scalar. To slow down or speed up the track played you can adjust the sampling rate.
Try these

```
wavplay(6*y,Fs)
walplay(y,Fs/2)
```

**ii. Playing a Track Backwards**

The command to reverse the order of the samples in a matrix is flipud(). Experiment with this command.

```
z=flipud(y)
wavplay(z,Fs)
```

Problem #4

Create a frequency response of the music 'handel' and observe the range of frequencies.

## D. Method for developing frequency response of a filter.

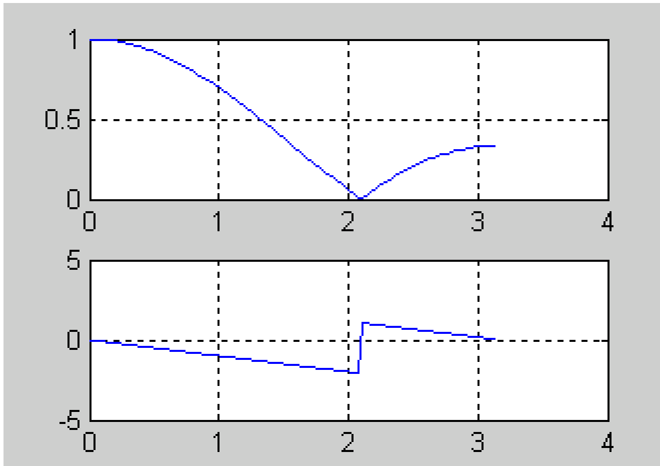
Following two methods can be used to plotting the frequency response of the filter given above.

**Method#1 (Preferable method)** This method plots the absolute value of amplitude vs. normalized frequency $\overset{\Lambda}{\omega}$ in radians and phase vs. normalized frequency $\overset{\Lambda}{\omega}$ . Normalized frequency $\overset{\Lambda}{\omega} = 2\pi \frac{f_0}{f_s}$

MATLAB code is given below:

```
b=[1/3,1/3,1/3]; % filter coefficients for problem # 3
w=0:(pi/100):pi;
H=freqz(b,1,w); %for FIR filter you have to use 1
subplot(2,1,1)
plot(w,abs(H));
grid on
subplot(2,1,2);
plot(w,angle(H))
grid on
```

The frequency response curve is shown below:



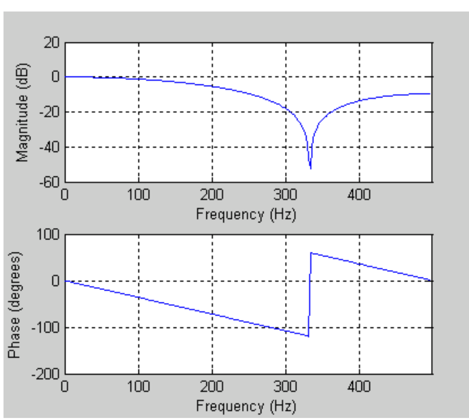
1. From the frequency response curve can you tell what type of filter it is:

2. **Method#2** b=[1/3,1/3,1/3];

```
freqz(b,1,256,1000)
```

filter coeff. ← b
1 for FIR filt. ← 1
256 → 8 bit resolution
1000 → Sampling frequency

Frequency response curve is given below:




3. From the above two method plots can you determine what type of filter it is.

4. Method#1 is the preferable method because this plot is independent of sampling frequency. Review the plots and try to find the correlations between the plots.

5. Now find the frequency response of the filter given by b = [1 –3 3 –1] for sample frequency 2000 Hz. Can you tell what type of filter it is?

### E. Building a FIR filter by SIMULINK

In this lab, we are going to build two digital filters with sampling frequency = 2000 Hz.

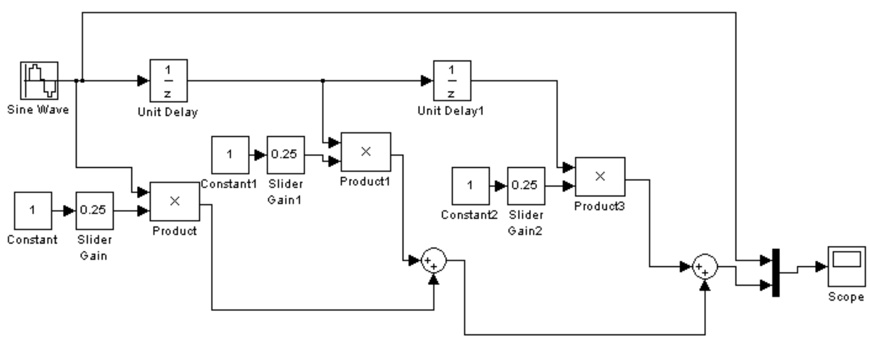
Plot the frequency response curves for the following filters.

Filter #1 b = [1/3 1/3 1/3] Can you tell what type of filter it is? What is the null frequency?

Filter #2 b = [1 –3 3 –1] Can you tell what type of filter it is?

Build these filters by SIMULINK (one example is given below) and prove that the filters are designed as required.

Note: Make sure that you set the sampling frequency correctly.




### F. Laboratory Report: Write a lab report including the following:

- Attach MATLAB Simulink diagrams.
- Some discussions about the filters.