Hand in Problem 2 – Applied Signal Processing

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Task 1

```
h = firpm(60,[0 0.1 0.2 1],[0 pi*2*0.05 0 0],'differentiator');
```

The code above is used in this assignment. The explanation for the first input of 60 is due to the fact that in Matlab environment the system recognize the input of N as N+1. Hence, we can obtain 61 order as required by the problem. The second input represents the vectors of the frequency of the passband edges of 0.05 Hz as well as stopband edges of 0.1 Hz due to the fact that Matlab considers the input has the range between 0 and 1 Hz which corresponds to the Nyquist frequency which is fs/2. Hence the input is in fact is equivalent to [0 0.05 0.1 1].

The third input is due to the fact that we want to have the value of passband of $2\pi \cdot 0.05$ as well as value of passband of 0 betweeen 0.1 Hz and 1 Hz. In addition, for the last input, the differentiator function is used as a special weighting scheme for for non-zero amplitude bands. The weight is assumed to be equal to the inverse of frequency times the weight W. It will also makes the filter have a constant +90° shift.

Task 2

the filter coefficients h(n) represents the values of the impulse response of the filter that we are using. It has the length of 61 due to our input of 60 in the '*firpm*' function. The **'h'** function above in Matlab could be represented by the equation in the Z-transform (complex frequency domain) as below.

$$H(z) = h(1) + h(2) \cdot z^{-1} + h(3) \cdot z^{-2} + h(4) \cdot z^{-3} + \cdots + h(n+1)z^{-n}$$

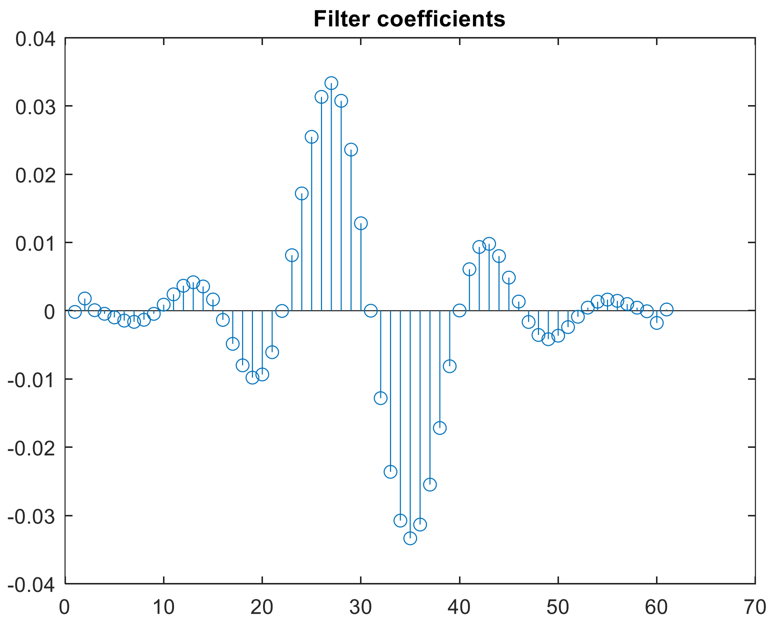

Figure 1. Low Pass FIR Coefficients h(n)

In addition, due to the fact that we want to show the ideal filter magnitude of frequency response. The magnitude is in fact as shown below in Figure 2. Hence setting the maximum value of the magnitude of the pass-band edge frequency which is equal to $2\pi \cdot 0.05$ gives us the plot as shown in figure 3 which is plotted together with the magnitude of the designed filter. Moreover, due to zero padding of 1000 cycles, we get the smoother looking spectrum and longer FFT result.

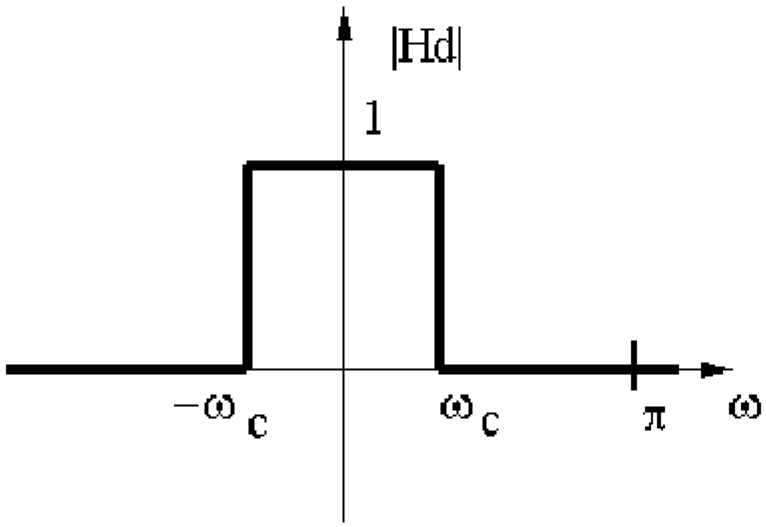

Figure 2. Magnitude of an ideal Low Pass Filter [1]

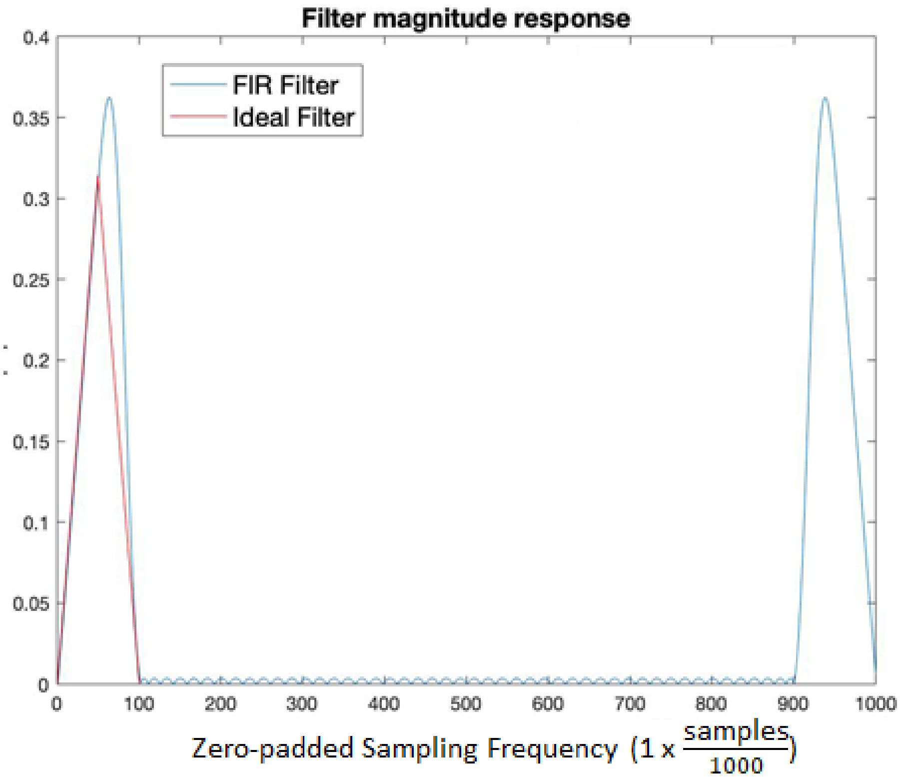

Figure 3. Magnitude of the Frequency Function |H(ω)| (Designed + Ideal Case)

Task 3

The group delay that we have in the frequency function is due to the fact that the use of FIR filter which has the value of $\frac{N-1}{2} \cdot f_s$. Since the filter that we use is odd symmetric FIR filter, we will have the phase response and output as shown below and the plot is shown in Figure 4.

$$\Phi(\omega) = \angle H(\omega) = \begin{cases} -\omega \dfrac{M-1}{2} + \dfrac{\pi}{2} & H_r(\omega) > 0 \\ -\omega \dfrac{M-1}{2} - \dfrac{\pi}{2} & H_r(\omega) < 0 \end{cases}$$

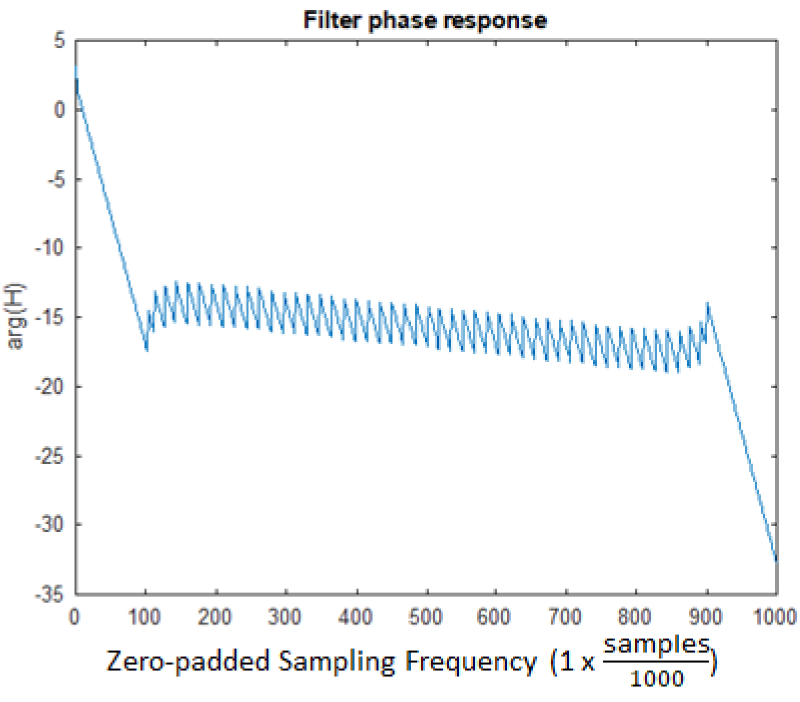

Figure 4. Phase Response of the Filter

Task 4

The result of the filtered signal of the position will give us the velocity of the truck since the filter that we are using has the differentiator. Moreover, we will have very high oscillation at the end of the signal due to the fact that the signal was discontinued abruptly which will result in side-lobe interference. Ideal frequency response theoretically should ideally have infinite impulse response. This phenomena is widely known as **Gibbs phenomenon**. The frequency content truncation will be causing a time domain ringing artifact in the signal. The output (Vehicle Speed) result of the FIR-PM filtering is shown in Figure 5 with compensated delay of 30 samples. However, we could try to describe the behavior of the signal with infinite/long enough frequency content as shown in Figure 6. By flipping the original length signal, we will have continuous signal for 1000 seconds.

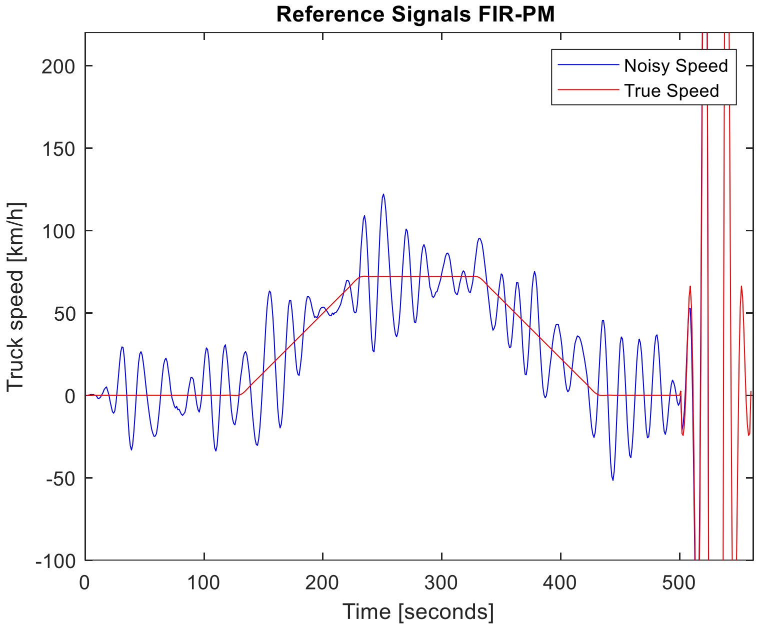

Figure 5. The Measured vs True Signal of the Truck Velocity

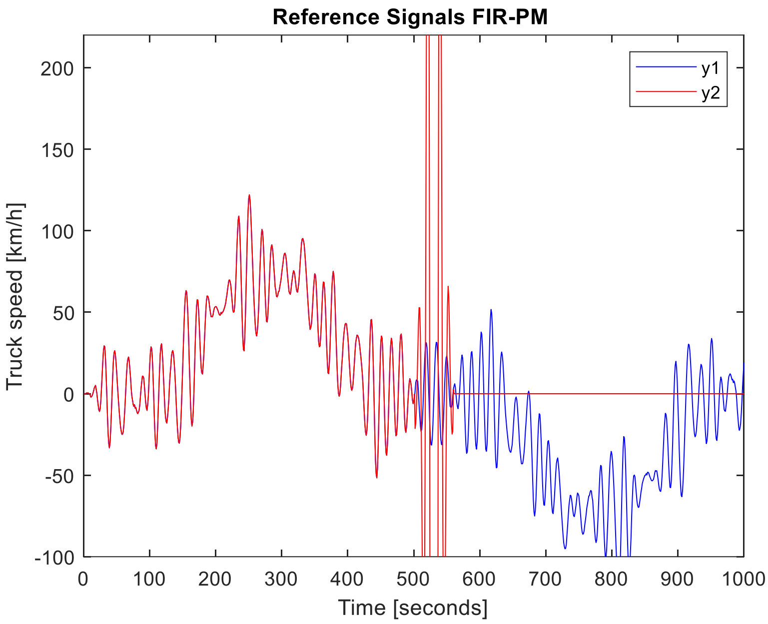

Figure 6. The result of **flipud** and **zero** function signal (y1 and y2)

If we observe figure 6 above, we will see that the y1 signal will have symmetric output response with the y2 except for the fact that it does not include the very large oscillation observed in y2 signal due to the elimination of the Gibbs Phenomenon.

Task 5

By using the Euler filter which is described mathematically below,

$$h_{\mathrm{Euler}}(n) = \begin{cases} \dfrac{1}{\Delta t}, & n = 0 \\ -\dfrac{1}{\Delta t}, & n = 1 \\ 0, & \text{otherwise} \end{cases}$$

we get the result as shown in Figure 7. If we observed the result between using the optimized FIR-PM filter and the trivial Euler filter, we will observe the same true speed due to the Euler approximation of derivative as the result of the FIR-PM filter (72 km/h). However, if we observe the measured signal, it will result in significantly worse output compared to the use of FIR-PM filter. Hence, we could conclude that using FIR-PM is more optimal.

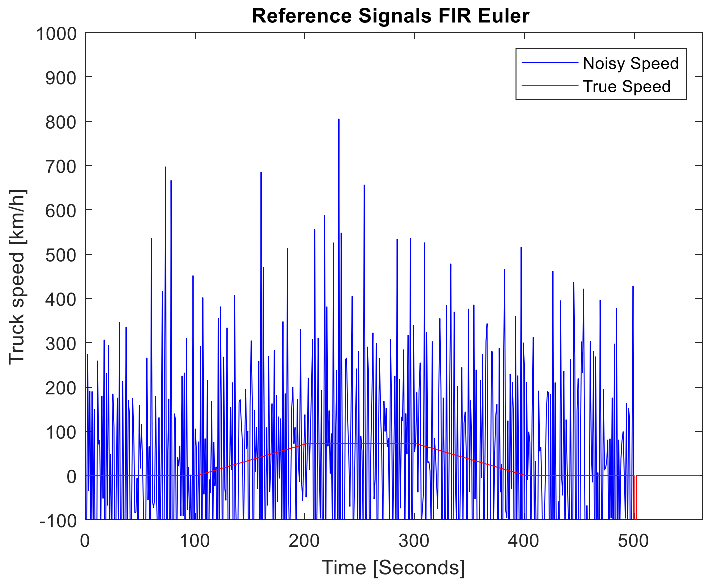

Figure 7. The Measured and True signal filtered with Euler Filter

Task 6

The maximum speed of the vehicle using "true" signal will give us 72 km/h whereas using the measured signal with FIR-PM filter will give us 122 km/h excluding the large oscillation at the end of the signal. On the other hand, using the Euler trivial filter will give us significantly larger maximum speed of 806 km/h.

References

[1] The Io-sinh function, calculation of Kaiser windows and design of FIR filters (2015). Augusto, José Soares.

Accessed from : http://publications.lib.chalmers.se/records/fulltext/74710.pdf (2019/12/09)