Université de Liège

Bachelor in Engineering - 3rd year

11 mai 2020

Applied digital signal processing

Homework 3

BULUT Stephan - s172244
MUKOLONGA Jean-David - s170679
SIBOYABASORE Cédric - s175202

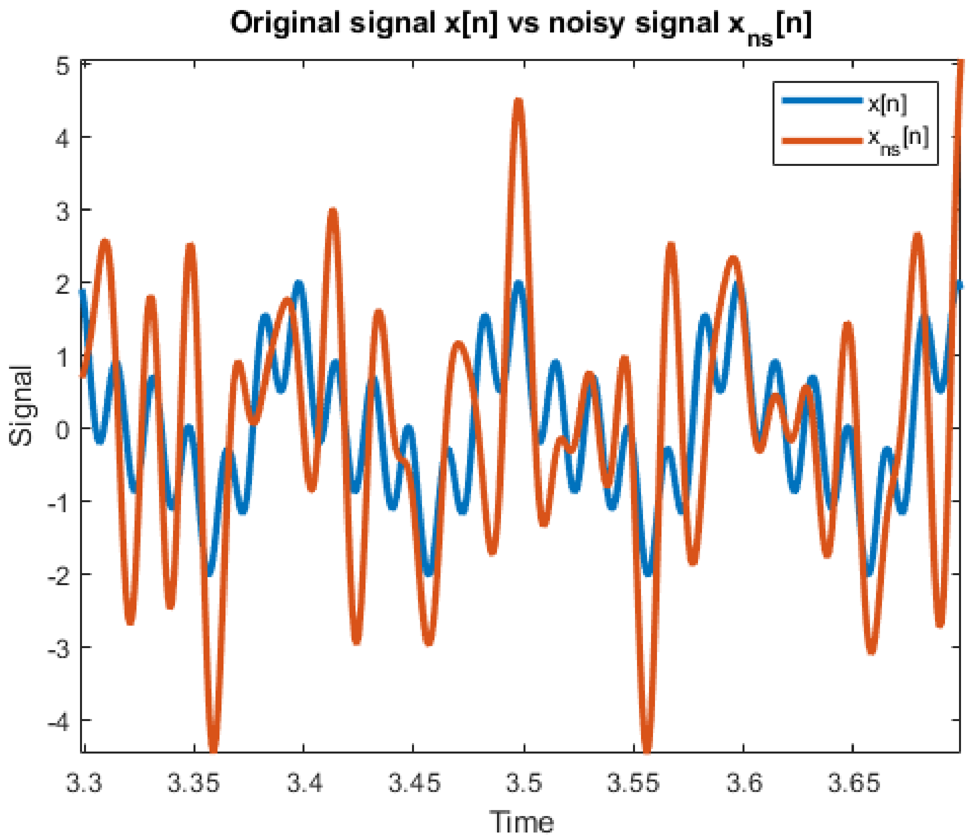
1 Noise filtering

1.a

We load the original signal $x[n]$ and the noisy signal x_{ns}[n] both sampled at 1000 Hz from the *hw3_noisy_signal.mat* file and then we plot them in the range

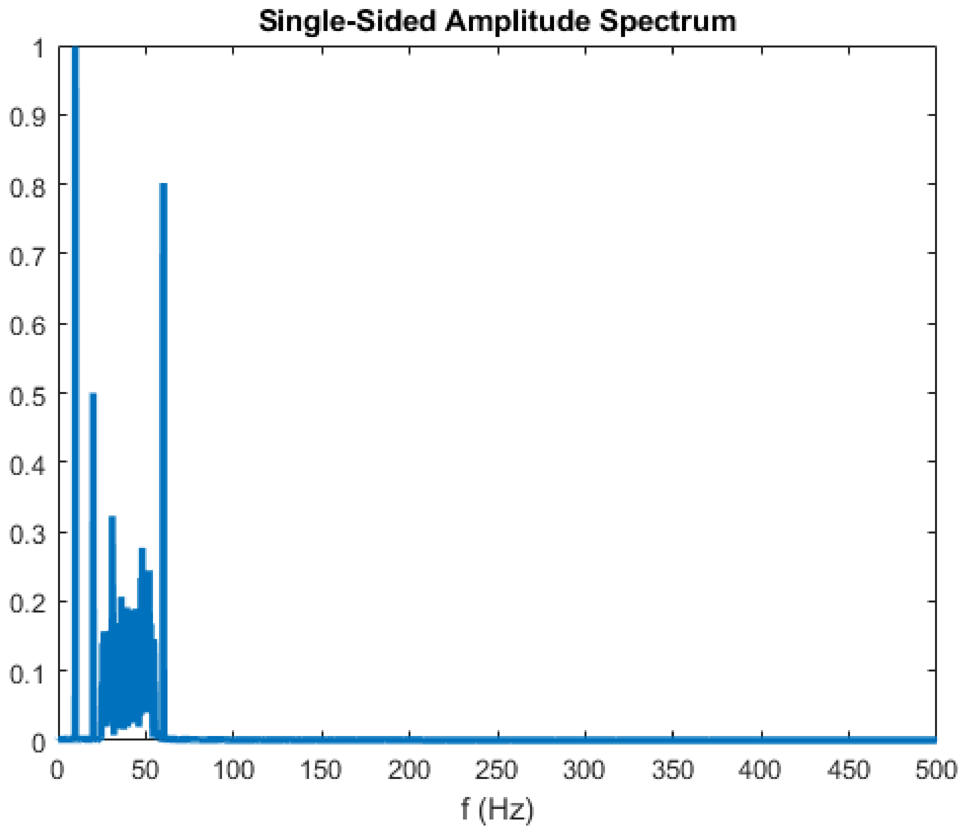
$$[N/2 - 200, N/2 + 200]$$

where $N = 7000$ is the length of both signals.



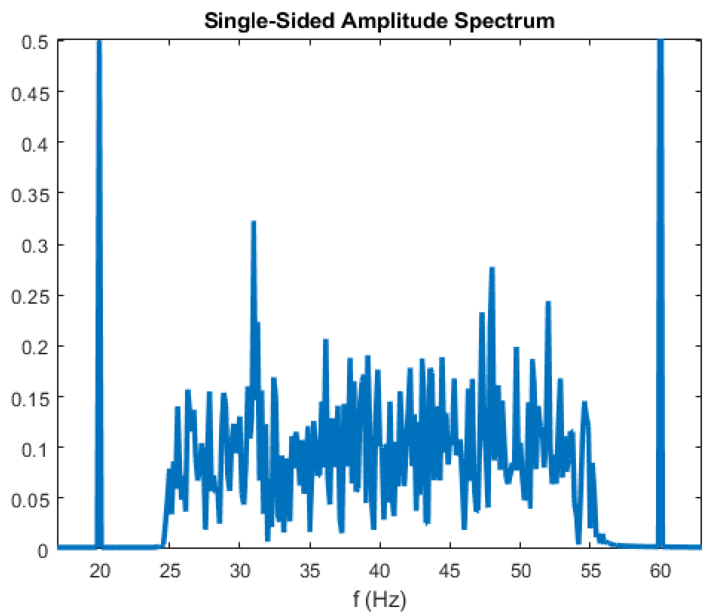
1.b

We plot the single-sided magnitude spectrum of the noisy signal (see graph below) thanks to the fft Matlab function that allows transformation of the signal from time domain to the frequency domain and thanks to a few manipulations afterwards.



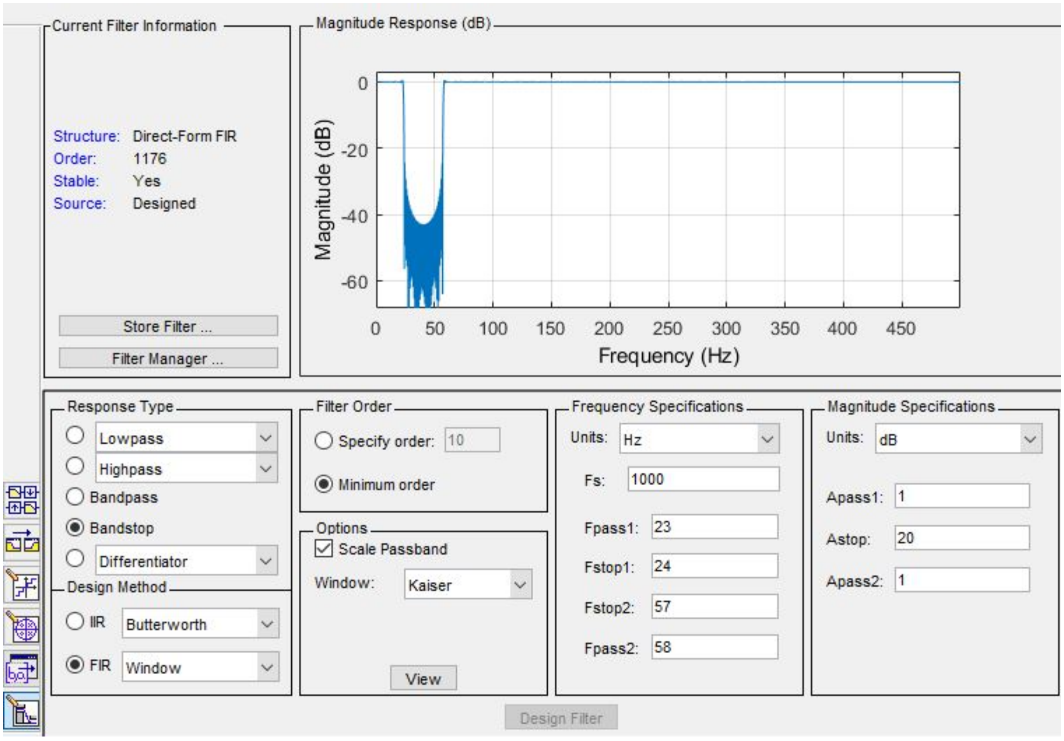
1.c

On the single-sided amplitude spectrum above we can identify the frequency range of the noise because we have 3 sinusoidal components in the original signal $x[n]$ ($f_1 = \frac{20\pi}{2\pi}$ =10 Hz, $f_2 = \frac{40\pi}{2\pi}$ =20 Hz, $f_3 = \frac{120\pi}{2\pi}$ =60 Hz) so anything else is the noise.
The approximate frequency range of the noise is [24, 57] Hz.

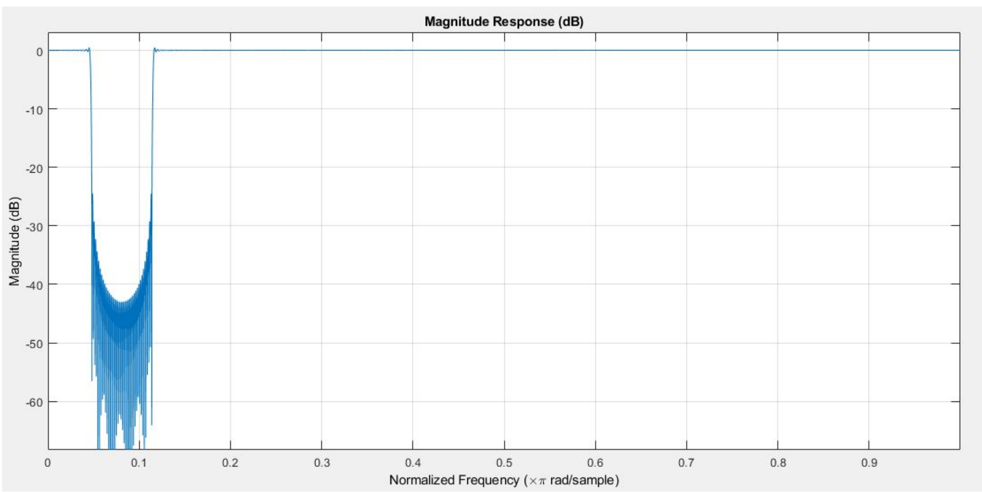


1.d

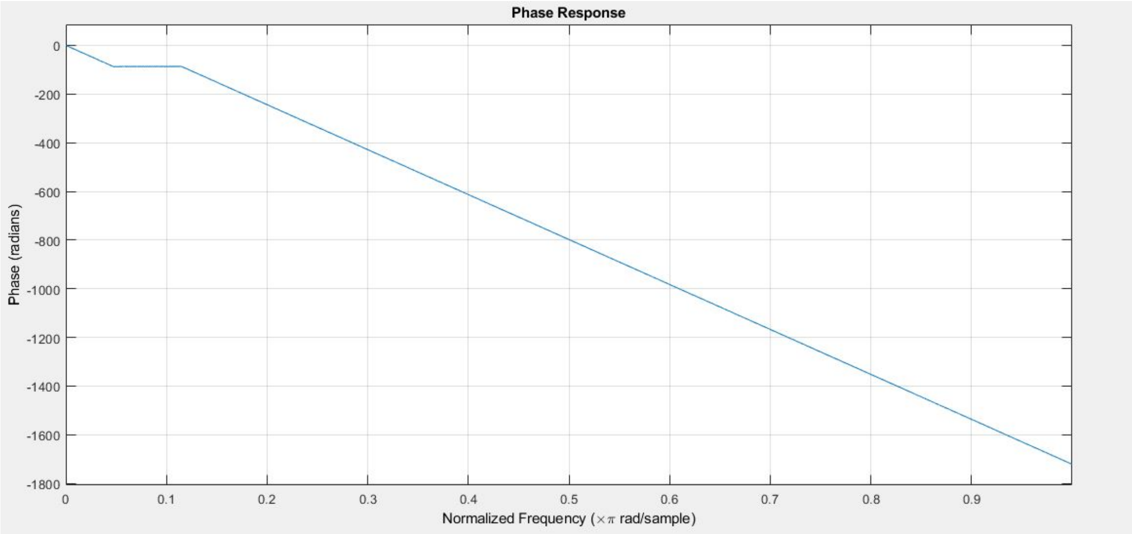
Since the goal is to have a filtered signal $x_{filt}[n]$ that has the same shape as the original signal $x[n]$, we should design a FIR filter. This type of filter preserves the shape of the signal because of its linear phase response unlike IIR filters which have nonlinear phase response which causes distorted amplitudes.
We create the FIR filter with Matlab. We're not interested in a high filter order so we have to make a balance between the order and the attenuation in the stopband. We choose a 20 dB stopband attenuation (in the stopband the frequency component should be divided by 10). The order is still high (1176) but that's the lowest we can do.



Thanks to the fvtool Matlab function we can analyze the FIR filter we designed. We can see the stopband region in the magnitude response :

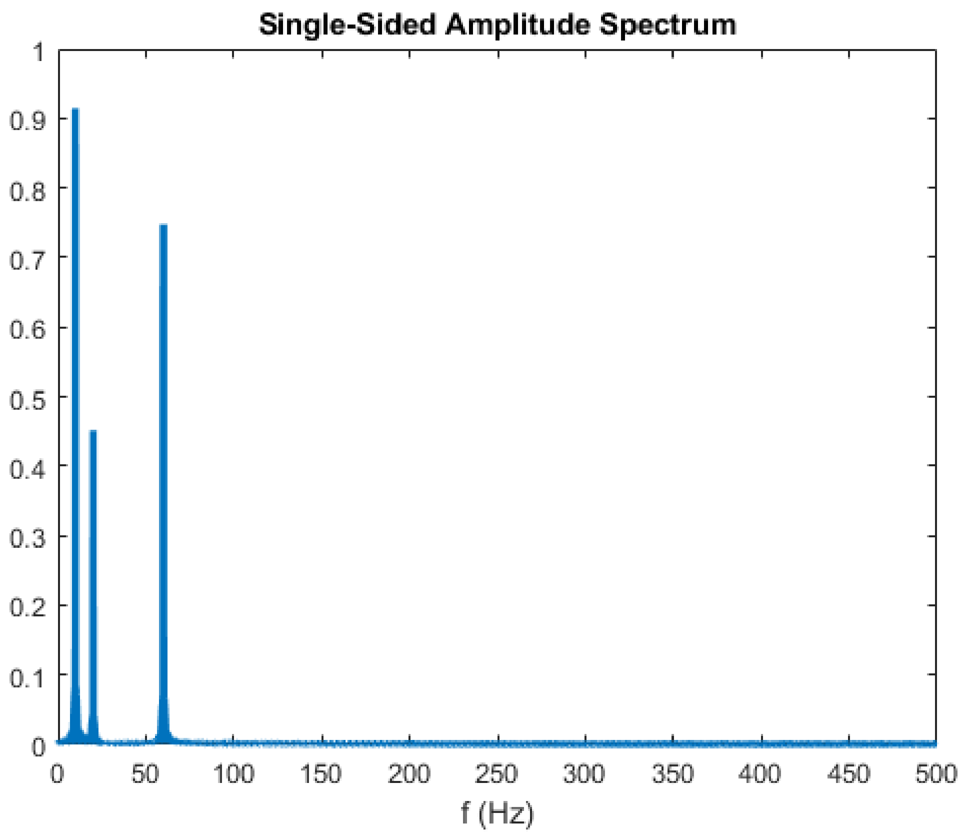


The phase is a linear function of the (normalized) frequency except in the rejection band. It passes from the origin :



1.e

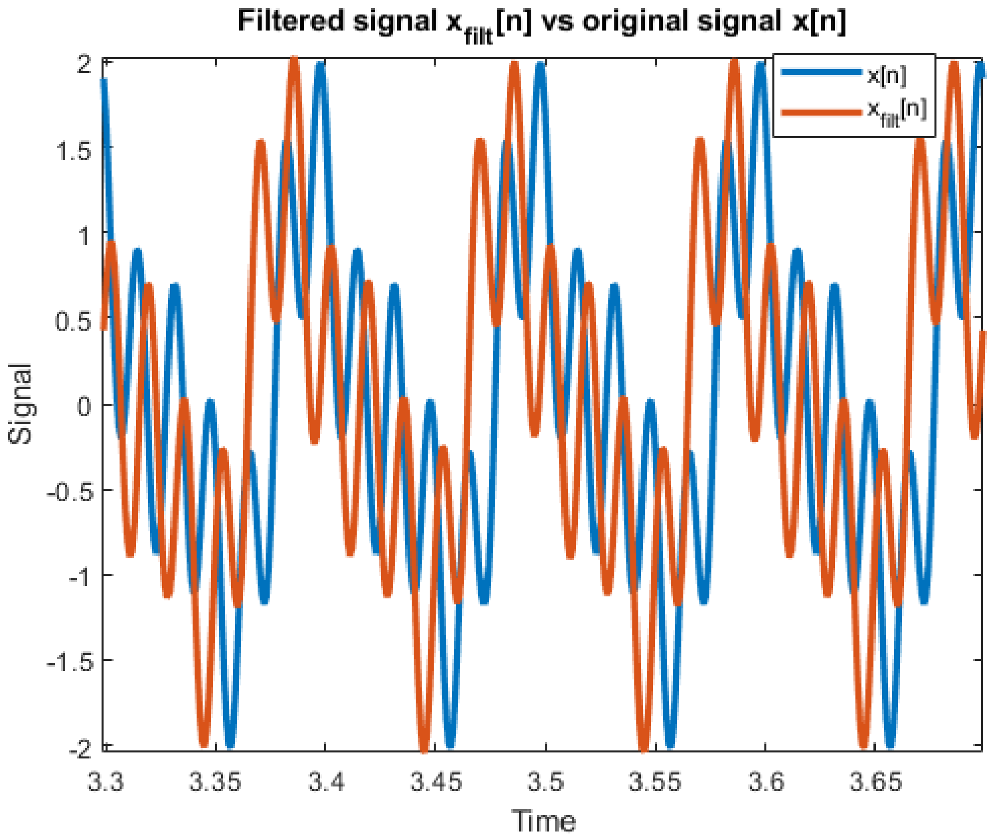
We apply our FIR filter to the $x_{ns}[n]$ signal to obtain the filtered signal $x_{filt}[n]$. Then we plot the single-sided amplitude spectrum of $x_{filt}[n]$:



We can see that the noise has been removed.

1.f

Now that the noise has been removed, we can plot $x[n]$ and $x_{filt}[n]$ in the same axis to see the result of our FIR filter :



Both signals have the same shape, the filtered signal is a delayed version of the original signal. The noise has been removed without distortion thanks to the FIR filter we designed and its linear phase response.