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Real-time Digital Signal Processing Laboratory

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Lab 2

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Contents

Abstract

In this Lab we shall begin with creating an equiripple filter using MATLAB, then we shall test this filter for sine waves generated in different frequencies located in the passband, transition band nnd stopband, we do this with the help of a sine wave generator written with MATLAB.

In the next stage we go on to design a C program to simulate an FIR filter using circular buffer, we do this with the help of the header file generated from the equiripple filter in MATLAB, we test the sine waves in this filter and plot the results.

In the last part we apply a white noise to this filter and plot the output which gives us the response of the designed filter.

1 Design and implementation of FIR filter with circular buffer

1.1 Part 1

Here we design the filter given in the lab manual with FDA tool.

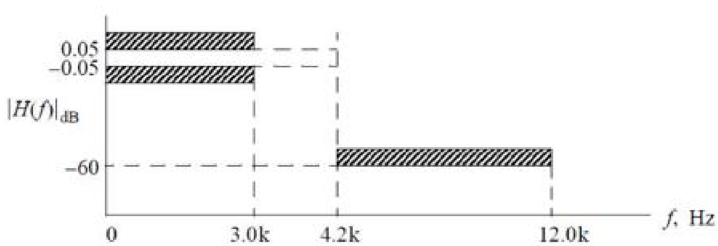

Figure 1: Desired Filter

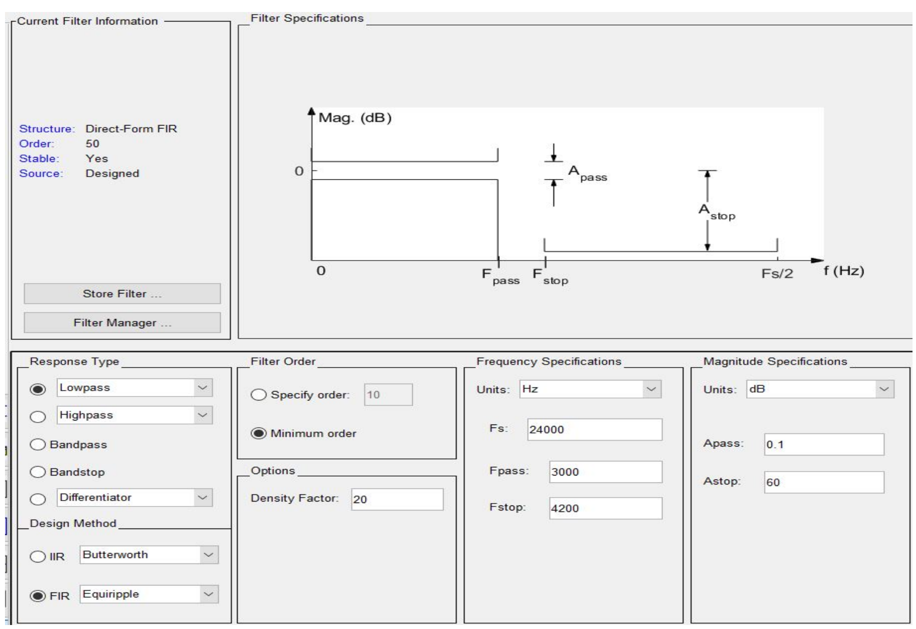

Figure 2: FDA Tool filter designing

1.1.1 Magnitude and Phase response of filter

The magnitude and phase response of the filter can be observed as follows.

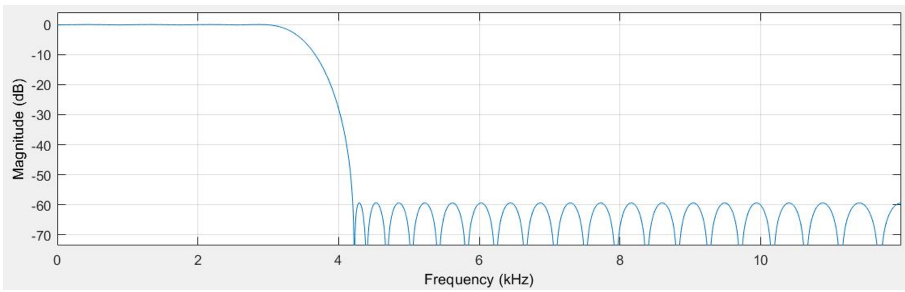

Figure 3: Magnitude response

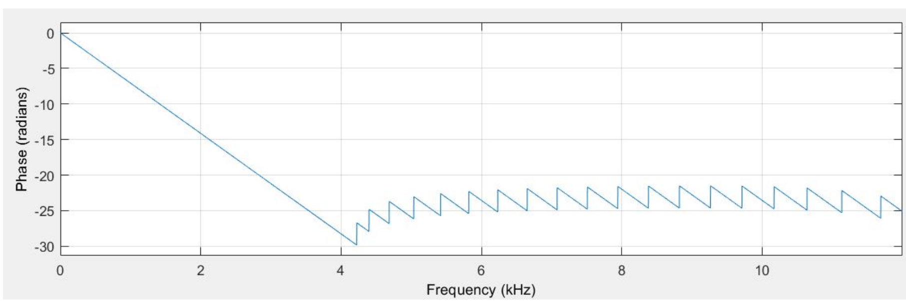

Figure 4: Phase response

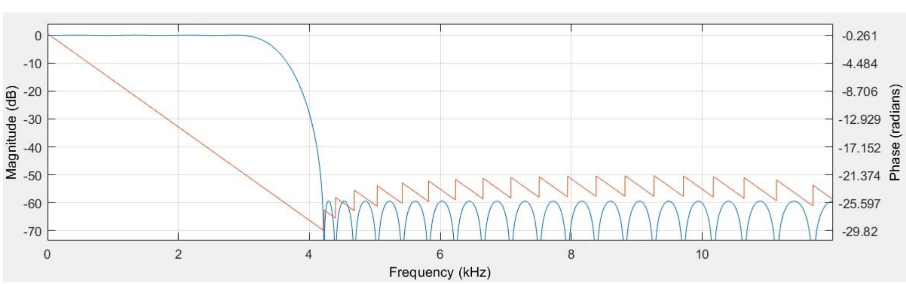

Figure 5: Magnitude and Phase response

1.2 Part 2

Here we generate different sine waves (two in each band of the filter) using MATLAB and filter them with our designed C program then draw the plots, the results are as follows.

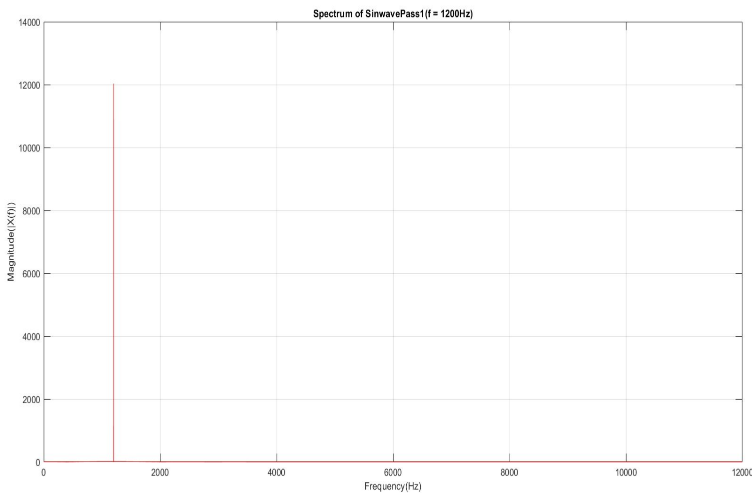

Figure 6: Wave 1 (Passband) Magnitude

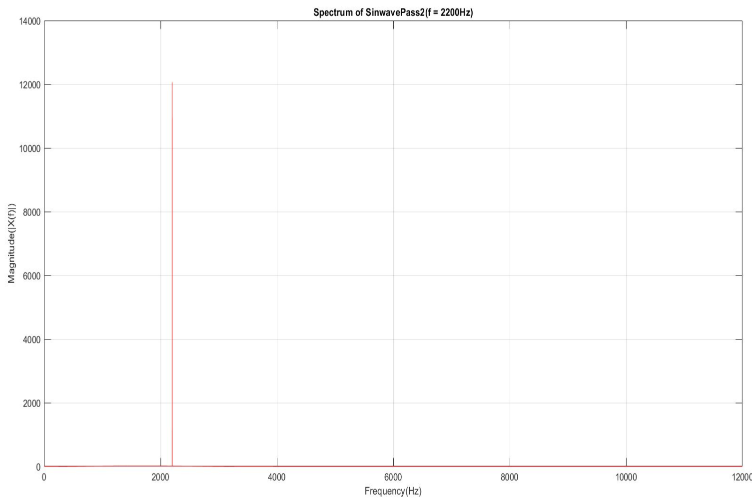

Figure 7: Wave 2 (Passband) Magnitude

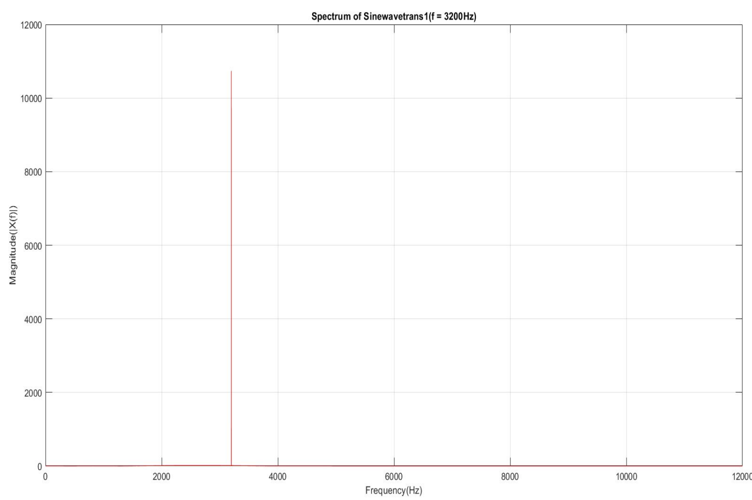

Figure 8: Wave 3 (Transition band) Magnitude

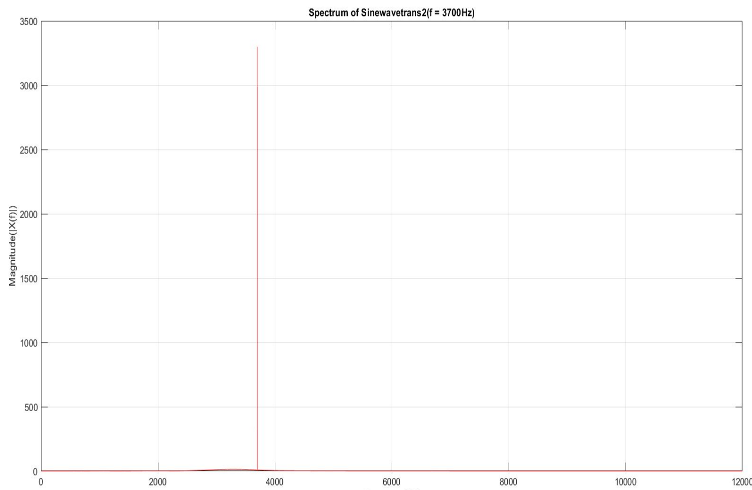

Figure 9: Wave 4 (Transition band) Magnitude

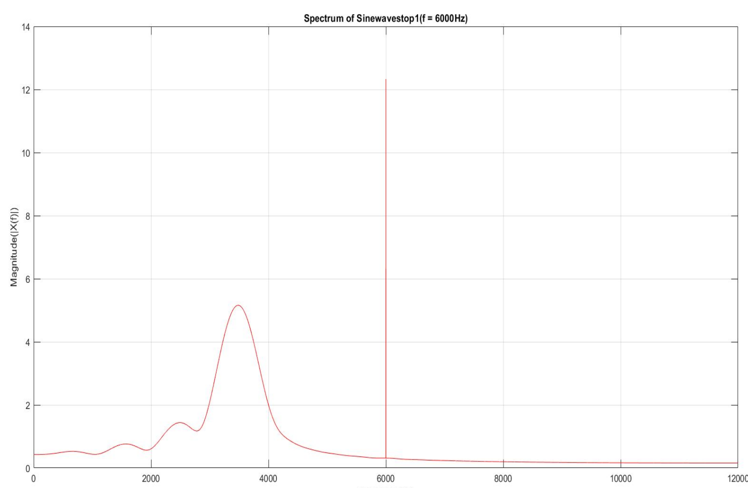

Figure 10: Wave 5 (Stopband) Magnitude

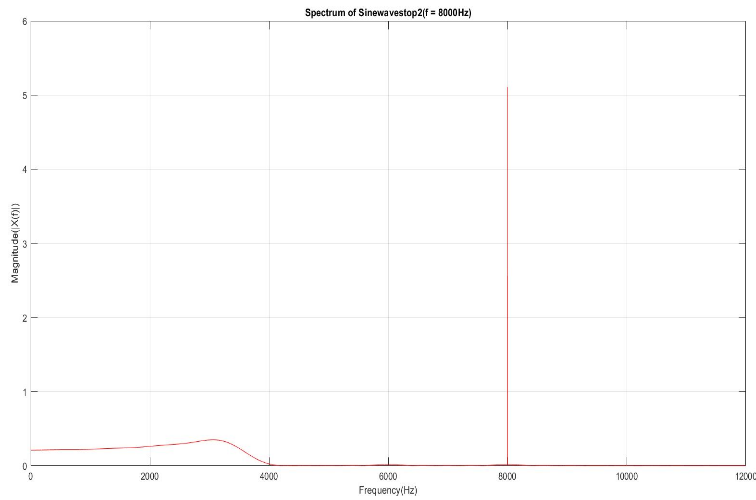

Figure 11: Wave 6 (Stopband) Magnitude

As it is beautifully observed, the peaks generated from the sine wave in the passband are extremely powerful which is expected, the peaks decrease in size and power in the transition band which completely matches our theoretical analysis.

In the stopband, the peaks generated from the sine waves are extremely weak, this is completely understandable due to the performance of the filter, there is also some ripple in the plots in this case which is also because of the filter.

1.3 Part 3

In this part we create a stochastic white noise using MATLAB, the plots in normal and logarithmic scale are as follows.

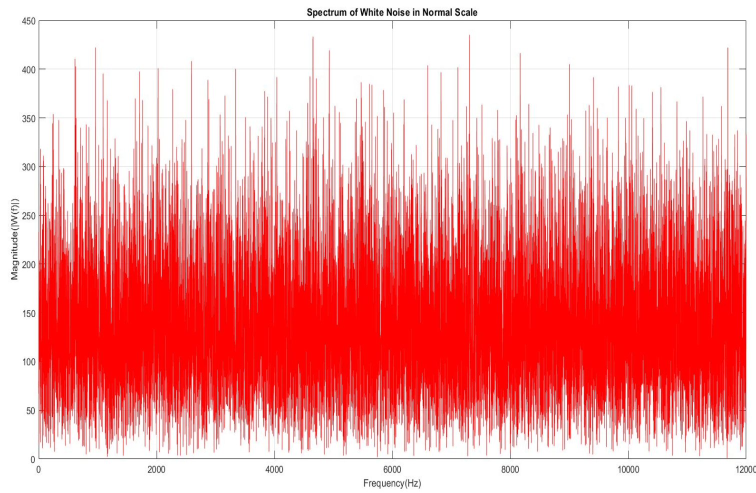

Figure 12: White Noise spectrum

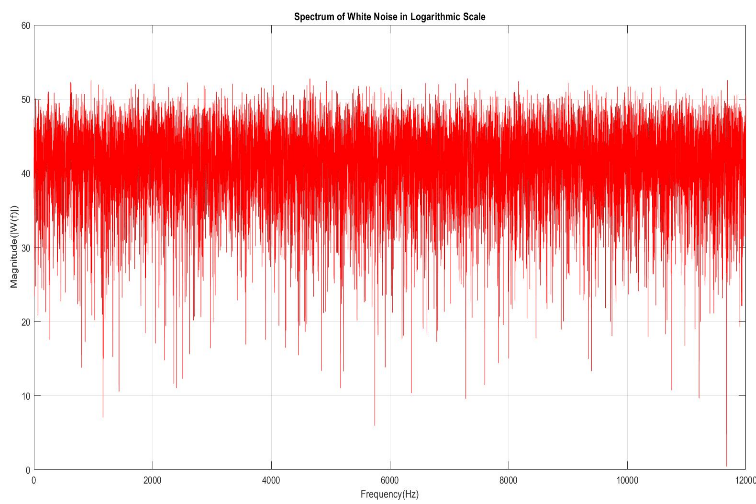

Figure 13: White Noise spectrum (logarithmic)

In a manner similar to testing the sine waves, we apply this white noise to the C program and filter it, then we go on to plot the results in MATLAB.

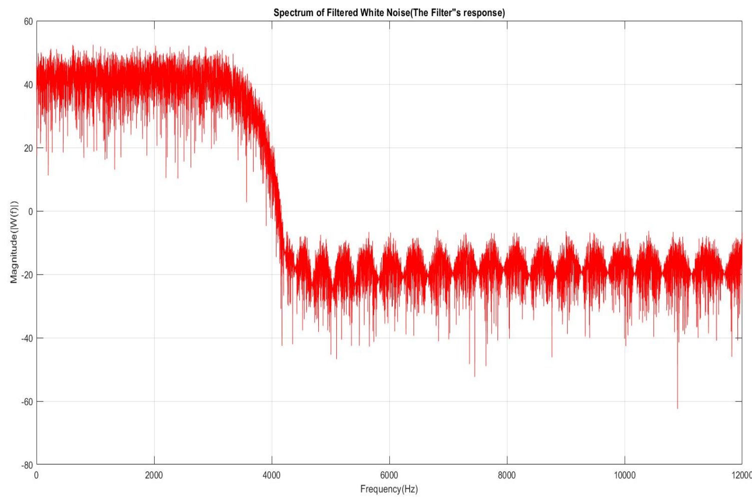

Figure 14: Filtered white noise spectrum

1.3.1 Why is the following method an estimation of the filter's response?

As we can see the spectrum of the filtered white noise looks like the magnitude response of the designed FIR filter, now we shall attempt to explain this phenomenon.

We know that in signal processing a white noise is a stochastic signal with equal intensity in all frequencies, this means that it shall have a constant PSD(power spectral density).

Due to the fact that white noise is available in all frequencies and because it has a constant spectral density, when we apply it to our filter we are essentially using a statistical approach to obtain the response of our filter, it is better to use a WSS (Wide Sense Stationary) white noise to obtain better results.

1.4 Part 4

Here we shall analyze what happens if we zero pad the FIR filter. We know that when we perform zero padding we add zeros to the filter, time domain. we must pay attention that padding zeros only to the end of the filter has no effect on the spectrum but it becomes more smooth in the time domain.

So we reasoned that by padding we shall have a more accurate answer in the time domain and our plots shall be more smooth.

It is also good to pad zeros so that the number of elements in the signal takes the form of a power of two, somewhat like what we used in performing FFT in butterfly diagrams, doing this shall increase the efficiency of our system.

1.4.1 What happens to the filter paramaters if we change fs to 48KHz?

In this part we change the sampling frequency to $f_s = 48KHz$. We have some changes in the magnitude and phase response which are depicted as follows.

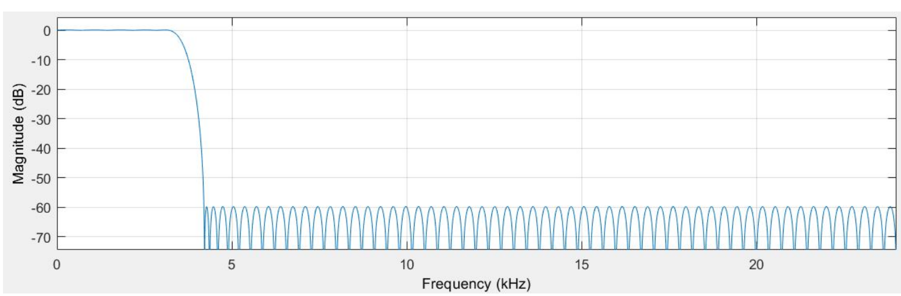

Figure 15: Magnitude response

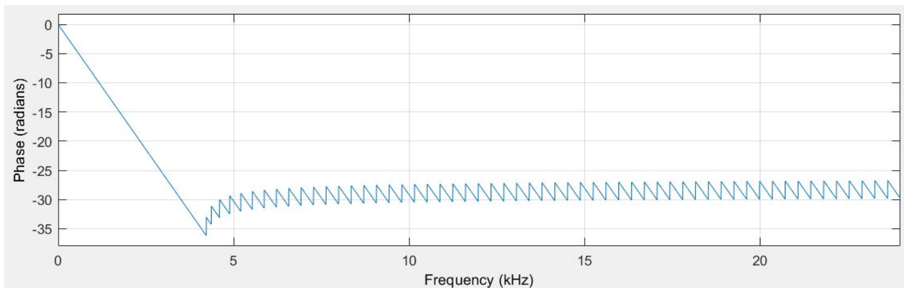

Figure 16: Phase response

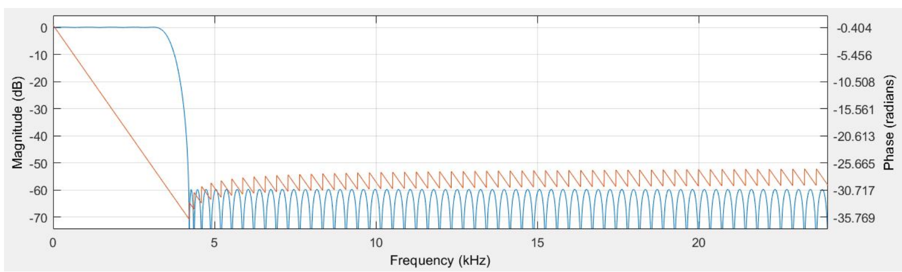

Figure 17: Magnitude and Phase response

It is also observed that the order of the filter changes, in the previous version the order was 54 but here the order changes to 131.

Figure 18: Order of the filters for $f_s = 24KHz \text{and} f_s = 48KHz$

References

[1] Vahid Shah-Mansouri, *Real-time Digital Signal Processing Laboratory lab notes, Fall 01*

[2] Mohammad Ali Akhaee, *Digital Signal Processing lecture notes, Spring 01*