Designing an Appropiate Order of Fir Filter Using Windowing Method

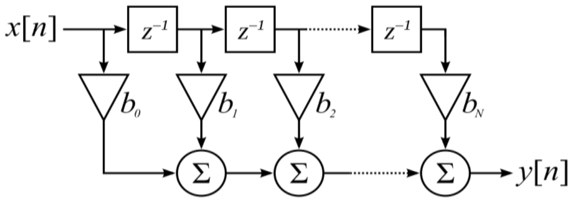
Tasher Ali Sheikh, Arindum Mukherjee

Abstract- ***This paper the authors mainly concern with the best order of the FIR filter using Boxcar windowing method. Simulation results give encouraging results regarding best order of the FIR Filters among the orders. A new method of implementation of best order of FIR Filter is proposed using MATLAB SIMULINK. Interesting results are obtained by comparing the different order filter's cut-off frequency to the assumption fixed 1000 Hz using Boxcar window. This novel concept of finding of the best order of the FIR filter using different order in Boxcar windows method that has been verified through MATLAB simulation results.***

Keywords- ***FIR, MATLAB SIMULINK, DSK5416 Kit, TMSC32054 hardware.***

I. INTRODUCTION

In signal processing, a finite impulse response (FIR) filter is a filter whose impulse response is of finite duration, because it settles to zero in finite time. The impulse response of an Nth order discrete-time FIR [5][2] filter last for N + 1 samples, and then settles to zero. FIR filters can be discrete-time or continuous time, and digital or analog. The output y of a linear time invariant system is determined by convolving its input signal x with its impulse response b.For a discrete-time FIR filter, the output [6] is a weighted sum of the current and a finite number of previous values of the input. The operation is described by the following equation, which defines the output sequence y[n] in terms of its input sequence x[n].



Fig(1):Block diagram of FIR filter

$$y[n] = b_0 x[n] + b_1 x[n-1] + \cdots + b_N x[n-N]$$
$$= \sum_{i=0}^{N} b_i x[n-i]$$

Where,
x(n) is the input signal,
y(n) is the output signal,
b_i Are the *filter coefficients*, also known as *tap weights*, that make up the impulse response is the order 0f the filter; an N^{th}order filter has (N+1) terms on the right-hand side.

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An FIR filter has a number of useful properties which sometimes make it preferable to an *infinite impulse response* (IIR) filter. FIR filters require no feedback. This means that any [1][3] rounding errors are not compounded by summed iterations. The same relative error occurs in each calculation. This also makes implementation simpler. Are inherently stable. All the poles are located at the origin and thus are located within the unit circle (the required condition for stability in a discrete, linear-time [2[4]] invariant system), also known as *all-zero system*. They can easily be designed to be *linear phase* by making the coefficient sequence symmetric; linear phase, or phase change proportional to frequency, corresponds to equal delay at all frequencies. This property is sometimes [4] desired for phase-sensitive applications, for example data communications, crossover filters, and mastering.

II. LINEAR PHASE FIR FILTER:

Linear phase is a property of a filter, where the *phase response* of the filter is a linear function of frequency, excluding the possibility of wraps at $\pm\pi$. In a causal system, perfect [1] linear phase can be achieved with a discrete-time FIR filter. Linear phase system has the property of the true time delay.
Since a linear phase filter has constant group delay, all frequency components have equal delay times. That is, there is no distortion due to the time delay of frequencies relative to one another; in many applications, this constant[2] group delay is advantageous. By contrast, a filter with *non-linear phase* has a group delay that varies with frequency, resulting in distortion. A filter with linear phase may be achieved by an FIR filter which is either symmetric or anti-symmetric.
A necessary but not sufficient condition is

$$\sum_{n=-\infty}^{\infty} h[n] \cdot \sin(\omega \cdot (n-\alpha) + \beta) = 0$$

for some α, β

III.WINDOWING METHOD:

In particular, the method used to convert an 'ideal impulse response' of infinite duration, such as a sink function to a finite impulse response (FIR) filter design is termed as the 'Windowing Method'.
In this method by multiplying the infinite impulse response with a finite length [1] sequence which is called the window function, a finite sequence can be obtained. In signal processing, the windowing function is a mathematical function that is zero valued outside some chosen interval. It can be written as

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$$w(n) = w(-n) = \begin{cases} nonzero\,; & |n| \le \left(\frac{N-1}{2}\right) \\ 0\;; & |n| > \left(\frac{N-1}{2}\right) \end{cases}$$

Where N represents the width of the window function w (n), n is an integer of value, $0 \le n \le N-1$,

And the infinite sequence is truncated at $n = \pm\left(\frac{N-1}{2}\right)$

Various windows are designed here for a particular order to compare and get the better cut-off frequency. Such as
Boxcar or Rectangular Window
Bartlett Window
Hamming Window
Henning Window

Rectangular/Boxcar Window: It is the simplest window as though the waveform suddenly turns on and off and is given as

$$\text{w (n)}= \begin{cases} 1\,; & -\left(\frac{N-1}{2}\right) \le n \le \left(\frac{N-1}{2}\right) \\ 0\;; & elsewhere \end{cases}$$

IV. THE SPECTRAL SPREADING AND LEAKAGE OF FIR FILTER

The truncation operation done in this method increases the signal bandwidth, thus causing its spectrum to spread called spectral spreading. As we know that the signal bandwidth is inversely proportional to the signal duration (width), hence the wider the window, the smaller is its bandwidth and the smaller the spectral spreading which should cause smaller distortion. Smaller window width cause more spectral spreading (more distortion) even into the band where the spectrum is supposed to be zero causing the leakage effect. The leakage tends to be worst near 'w'(desired frequency) and least at frequencies farthest from 'w'.

V. EXPERIMENTAL TMS320VC5416 KIT BLOCK DIAGRAM

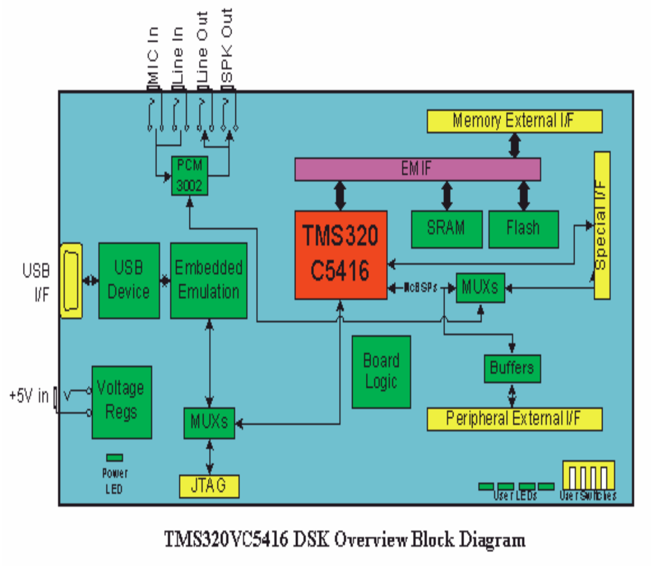

Fig (2):TMS320VC5416 DSK Block diagram

a. TMS320C5416 DSK KIT OPERATION AND SETUP:

The input signal is generated using a function generator. The generated signal is feed into ADC of DSK kit, which converts the input analog signal to digital signal. In our realization the chosen sampling frequency is fc= 8000 Hz. The sampled input signal and the MATLAB generated coefficient of the linear-phase FIR filter are convolved. The output from the system (after the DAC) can be observed through an oscilloscope.

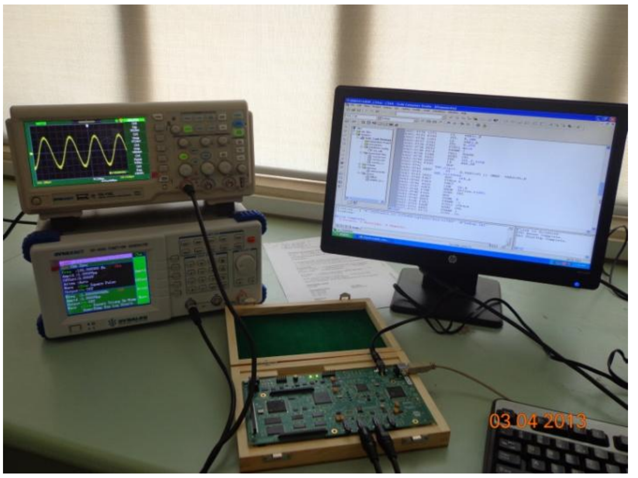

Fig (3): DSK Kit Operation Block Diagram.

b. OUTPUT SIGNAL:

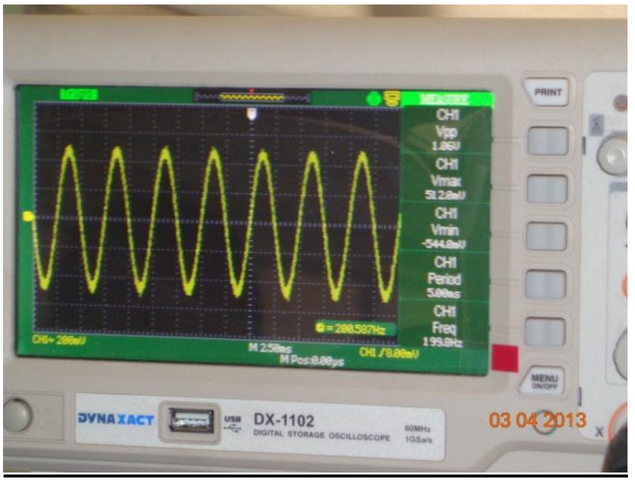

Fig (4): Output of the DSK Kit

VI. DETAIL ANALYSIS AND DESCRIPTIONS OF RESULTS

FREQUENCY RESPONSE OF THE FIR LPF OF DIFFERENT ORDER:

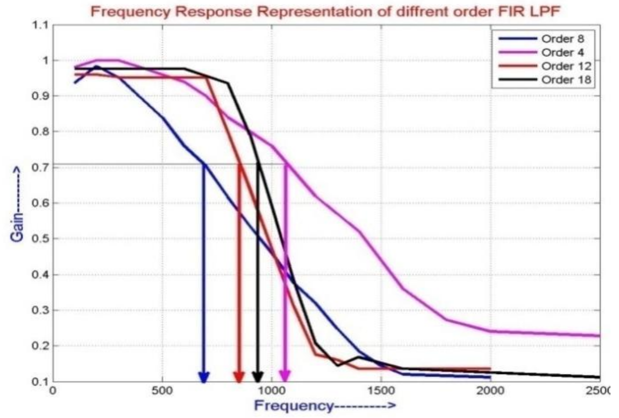

Fig (5): Frequency response of different order FIR LPF

From the frequency response plot in fig (5), it is observed that when the

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order of the filter increases the magnituderesponse approaches the given cut-off-frequency. In our study, the chosen cut-off-frequency is fc=1000Hz.

The comparison for cut-off-frequency vs. order is depicted below:

Order	Cut-off-Frequency(Hz)
4	1075.7039
8	694.6396
12	958.1261
18	942.4

Table .1

From the above comparison it evident that the 12th order filter's cut-off-frequency is nearer to the chosen frequency. The idea behind this is that when cut-off-frequency approaches the desired one, the signal power is maximum and noise power is negligible which does not affects the original message signal

FREQUENCY RESPONSE OF THE FIR HPF OF DIFFERENT ORDER:

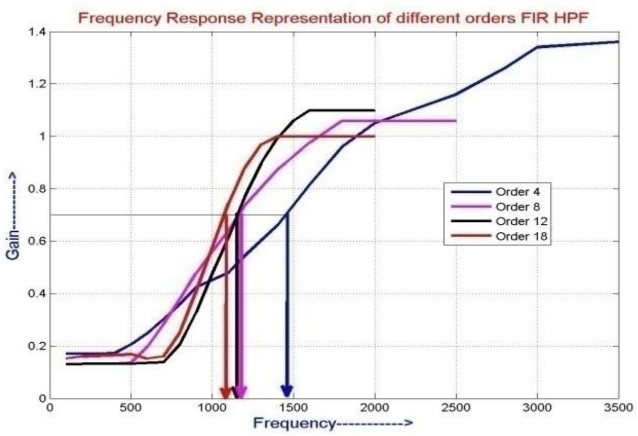

Fig (6): Frequency response of different order FIR HPF

From the frequency response plot in fig (6), it is observed that when the order of the filter increases the magnitude response approaches the given cut-off-frequency. In our study, the chosen cut-off-frequency is fc=1000Hz.

The comparison for cut-off-frequency vs. order is depicted below:

Order	Cut-off-Frequency(Hz)
4	1460.26
8	1169.7894
12	1159.8684
18	1077.97662

Table .2

From the above comparison it evident that the 18th order filter's cut-off-frequency is nearer to the chosen frequency. The idea behind this is that when cut-off-frequency approaches to the desired one, the signal power is maximum and noise power is negligible which does not affects the original message signal.

FREQUENCY RESPONSE OF THE FIR BPF OF DIFFERENT ORDER:

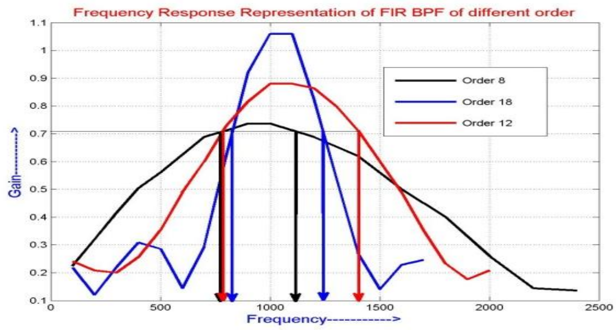

Fig (7): Frequency response of different order FIR BP

From the frequency response plot in fig(7), it is observed that when the order of the filter increases the magnitude response approaches the given cut-off-frequency. In our study, the chosen cut-off-frequency is fc=1000Hz.

The comparison for cut-off-frequency vs order is depicted below:

Order	Frequency (Hz)
8	779.1667- 1120.925
12	783.593-1404.4643
18	829 – 1241.7839

Table. 3

From the above comparison it evident that the 18th order filter's cut-off-frequency is nearer to the chosen frequency. The idea behind this is that when cut-off-frequency approaches the desired one, the signal power is maximum and noise power is negligible which does not affects the original message signal.

FREQUENCY RESPONSE OF THE FIR BRF OF DIFFERENT ORDER:

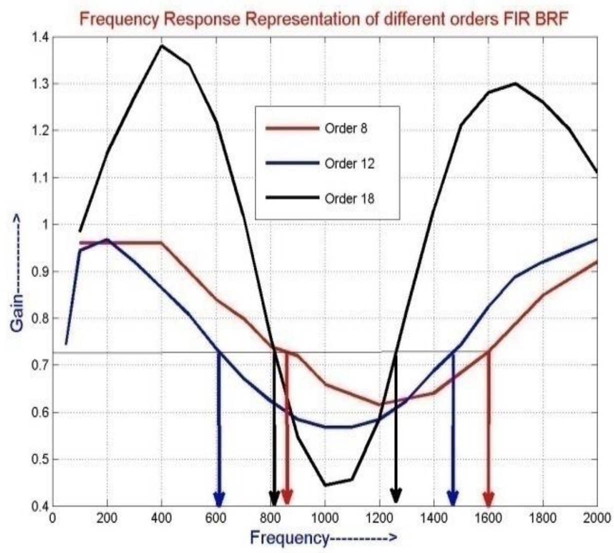

Fig (8): Frequency response of different order FIR BRF

From the frequency response plot in fig (8), it is observed that when the order of the filter increases the magnitude response approaches the given cut-off-frequency. In our study, the chosen cut-off-frequency is fc=1000Hz.

The comparison for cut-off-frequency vs order is depicted below:

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Order	Frequency (Hz)
8	821.6666 -1253.0254
12	745.2812–1433.9286
18	824.9958 - 1253

Table. 4

From the above comparison it evident that the 18th order filter's cut-off-frequency is nearer to the chosen frequency. The idea behind this is that when cut-off-frequency approaches the desired one, the signal power is maximum and noise power is negligible which does not affects the original message signal.

VII. BEST ORDER FILTER

From the above FIR LPF, HPF, BPF and BRF of different orders with rectangular window and specified cut off frequency we have observed that *order 18th* isalmost equal to the specified cut off frequency. Hence we can say that *order 18th* is the *best order filter* among them.

VIII. CONCLUSIONS

In this paper we have studied FIR system order and the effects of different order are implementing in FIR systems. The effect of Total Harmonic Distortion (THD) on the digital filters is also being studied. Real-time digital filters using TMSC32054 hardware and software DSK 5416 Kit and result are obtained for FIR filter.

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