Digital FIR–LP Filter using Window Functions

A L Choodarathnakara

***Abstract*— The concept of analog filtering is not new to the electronics world. But the problems associated with it like attenuation and distortion are really a heavy price to pay. To overcome all these problems a whole new concept of digital filtering came into existence. In this humble effort in realizing one such filtering operation, we would like to, present a brief overview of Low Pass FIR filter to demonstrate the digital filtering operation. The signal processing is done by making use of TMS320C50 DSP Board of Texas Instruments. The DSP, with accuracy being its forte, considerably reduces the distortion, which one might come across during one such endeavor. The concept of design that we are making use of here is the windowing technique. Making use of the process of convolution, we obtain the final frequency response of the Low Pass Digital Filter.**

***Index Terms*—Digital Filters, Recursive Filters, Gibb's Phenomenon, TMS320C50.**

I. Introduction

Signals play a major role in our life. They are used to communicate between humans, and humans and machines. A signal is defined as any physical quantity that varies with time, space or any other independent variables viz., distance, temperature, etc., examples of signal that encounter frequently are speech, music, video and picture signal. Most signals, which we encounter, are generated by natural means. However signal can be generated synthetically or by means of computer simulation. The signal carries valuable information. Our objective of signal processing is to extract entire information carried by the signal. The method of information extraction depends on the type of signal and nature of information being carried by the signal.

Signal processing has played a major role in such diverse field as speech and data communication, biomedical engineering, acoustics, sonar, radar, seismology, instrumentation, robotics & many others. Sophisticated signal processing algorithms & hardware are prevalent in a wider range of systems from highly specialized military systems through industrial applications to low cost, high volume consumer electronics. Much of traditional signal processing involves processing one signal to obtain other signal. Another important class of signal processing problem is signal interpretation, which uses preprocessing steps like filtering, attenuation, amplification, parameter estimation, etc. In such problems, objective of processing is not to obtain an output signal, but to obtain a characterization of the input signal.

The actual implementation of a digital filter could be either in software/hardware form, depending on applications. In both types of implementation, the signal variables and the filter coefficients can't be represented with infinite precision. As a result, a direct implementation of a digital filter may not provide satisfactory performance due to the finite precision arithmetic. It is thus of interest to develop alternative realizations based on other types of time domain representation with equivalent input-output relations, depending on the type of digital filter being implemented, and choose the realization that provides satisfactory performance under finite precision arithmetic. Hence we develop the methods for the analysis of real causal FIR digital filter structures when implemented with finite precision arithmetic and present additional realizations that have been developed to minimize the effects of finite word-length. The structural representation provides the relations between some pertinent internal variables with the input and output that in-turn provide the keys to the implementation.

A variety of software packages are presently commercially available that have made the design of digital filter rather simple to implement on a computer like MATLAB. But the students have limited real world design opportunity. The embedded microprocessors are used to run the real time digital signal processing algorithms, but are not easily reprogrammed to implement new algorithms. Hence, in this work, fixed-point Digital Signal Processor is used to design FIR digital filter which helps in real-time DSP applications. The DSP processor 'C50 is a fixed-point processor having a lesser accuracy as compare to floating-point processor. Hence, higher order floating point processor could be used to improve the accuracy of the filtering operation. In this work the FIR digital filters are designed by using Kaiser and Hamming window methods.

II. DIGITAL SIGNAL PROCESSING (DSP)

Digital Signal Processing is an area of science and engineering that has rapidly developed over past 30 years. This rapid development is a result of the signified advances in digital computer technology and integrated circuit fabrication. Speech analysis was the driving force behind the initial attempts to process signals digitally. The existing tolerances demanded of filters in speech processing systems simply could not be maintained over time with analog techniques, as they are subjected to temperature drift, component tolerances and aging.

In the sixties, DSP hardware used discrete component and consequently, because of the high cost volume its application could only be justified for very specialized requirements. In the seventies, monolithic components for some of the DSP subsystems appeared, primarily, dedicated digital multiplexers and address generators and DSP systems could be implemented using bit slice processors. Breakthrough for mass exploitation of DSP technique came in 1979. When

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Intel introduced 2920, completely self contained signal processing device in a 40-pin package incorporating on board program EPROM, data RAM, A/D and D/A converter, and architecture and instruction set powerful enough to implement a full duplex 1200bps modem, including transmit and receive filters.

The Digital Signal Processing involves the replacement of analog elements such as amplifiers, modulators and filters by the subsystem shown in Figure 1. The waveform to be processed is filtered, then sampled, and converted to digital representation, and finally input to the signal processor. All subsequent waveform manipulation is then implemented in software. The processed signal samples in the form of digital words are finally converted back to analog samples and passed through a reconstruction filter to recover the signal as a continuous waveform.

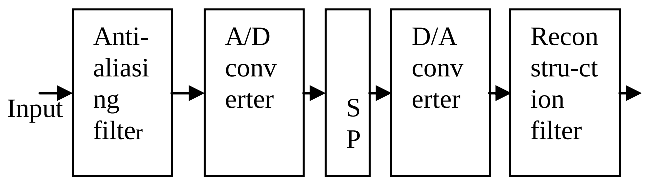

Fig 1. Block diagram representing of Digital Signal Processing

A. Anti-Aliasing Filter

The distortion free recovery of sampled reference signal is possible only when the reference signal is entirely band limited to half the sampling rate. Even when the input signal is naturally band limited, an anti-aliasing filter is still advisable to reduce out of band noise into the wanted frequency band. Ideally the anti-aliasing filter should have flat amplitude and linear phase response over the bandwidth of the signal, and infinite attenuation at the half-sampling rate and beyond. A practical analog filter does not need this specification having, amongst other faults, a finite transmission region. Although the filter transfer function in the transition range need affect the wanted signal, any non-zero response beyond the half sampling frequency will permit high frequency input components and noise to alias into the wanted band.

In the absence of anti-aliasing filter, a white noise filter with power spectral density N_0 and bandwidth nB_s (where B_s is the sampling rate and n is an integer) will result in a recovered signal with noise spectral density of $2nN_0$ occupying a bandwidth $B_s/2$. In other words, the noise power in the recovered, band limited signal is the same as in the entire bandwidth of the input signal, assuming no processing of the sampled signal.

One of the most convenient forms of anti-aliasing filter is switch capacitor type. Since the filter bandwidth can be varied according to the internal or external filter clock. Typically, a clock rate is 50 to 100 times the filter cutoff frequency, and anti-aliasing of the switched capacitor filter can be achieved with a simple RC network.

B. Analog to Digital and Digital to Analog Conversion:

Most signals of practical interest, such as speech, radar signal, and various communication signals are analog. To process analog signals by digital means, it is first necessary to convert them into digital form that is, converting them to sequence of numbers having finite precision. This procedure is called Analog to Digital (A/D) conversion, and the corresponding devices are called A/D converters. Conceptually, we view A/D conversion as three step process.

- *Sampling:* This is the conversion of continuous time varying signal into a discrete time signal obtained by taking samples of continuous time signal at discrete time instants. Thus if Xz(t) is input to the sampler, the output is Xa(nT) = x(n) where T is the sampling interval.
- *Quantization:* This is the conversion of discrete time continuous valued signal into a discrete valued (digital) signal. The value of each signal samples is represented by value selected from a finite set of possible values. The difference between the unquantized sample x(n) and the quantized output Xq(n) is called quantization error.
- *Coding:* In the coding process, each discrete value Xq(n) is represented by a b-bit binary sequence. The process of converting a digital signal into an analog signal is known as digital to analog conversion (D/A). All D/A converters connect the dots in a digital signal by performing some kind of interpolation, whose accuracy depends on the quality of D/A conversion process.

C. Advantages of Digital Signal Processing

- The inexpensive and relatively fast digital circuits have made it possible to construct high sophisticated digital systems capable of performing complex digital signal processing functions and tasks, which are usually too difficult and are too expensive to be performed by analog signal processing systems.
- Digital signal processing allows sharing of processor among different physical signals by means of time-sharing.
- Digital circuits can be easily cascaded without loading effect unlike analog circuits.
- Digital signals can be stored almost infinitely without any loss of information on various storage medias like magnetic disks, optical discs etc., but stored analog data deteriorates rapidly.
- Digital systems for signal processing hardware allow programmable operations. Through software one can more easily modify the signal processing functions to be performed by the hardware. This digital hardware and associated software provided a greater degree of flexibility in system design.
- A higher order of precision achievable with digital hardware and software compare to analog circuits and analog signal processing systems.

D. Disadvantages of Digital Signal Processing

- One obvious disadvantage is the increased system complexity in the digital signal processing of analog signals because of the need for additional pre and post processing devices such as the A/D & D/A converters and their associated filter and complex digital circuitry.
- A second disadvantages associated with digital signal processing is the limited range of frequencies available

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for processing. This property limits its applications particularly in the digital processing of analog signals.

- Third disadvantages stems from the fact that digital systems are constructed using active devices that consumes electrical power.

III. DIGITAL FILTERS

Digital filters are implemented using either a digital logic circuit or a computer program and they operate on a sequence of numbers that are obtained by sampling the continuous waveforms. The use of digital filters is wide spread today because of easy availability of advanced computer software. There are two basic types of digital filters, which are listed as below.

A. Recursive Filters

The transfer function of a recursive filter is expressed as the ratio of two polynomials:

$H(z) = (\Sigma a_i z^{-i} / \Sigma b_j z^{-j})$ Where, $0 < i < n$ and $0 < j < n$

B. Non-recursive Filters

The transfer function contains a finite number of elements and is in the form of a polynomial.

$H(z) = \Sigma h_i z^{-i}$ Where, $0<i<n$

A digital filter is a mathematical algorithm implemented in either hardware or software that operates on a digital input signals to produce a digital output signals for the purpose of achieving a filtering objective. The term digital filter refers to the specific hardware or software routine that performs the filtering algorithm. Digital filters often operates on digitized analog signals or just numbers representing some variables, stored in a computer memory or even the space within the device which we are using for filtering operation. The band limited analog signal is sampled periodically and converted into a series of digital samples x (n) where n=0,1,2,3, …… The digital processor implements the filtering operation, mapping the input sequence x (n) into the output sequence y(n) in accordance with a computational algorithm for the filter.

Digital filters can be classified into:

1. Infinite Impulse Response (IIR)
2. Finite Impulse Response (FIR)

Here in IIR filters the impulse response of the filter is of infinite duration where as for FIR filters it is of finite duration. In practice it is not feasible to compute the output of the IIR filter because the length of its impulse response is too long. But in FIR filter the impulse response is finite and has only N values.

C. Advantages of Digital Filters

- The values of resistor, capacitor and inductor used in analog filter changes with temperature. Since the digital filter does not have these components, they have high thermal stability.
- In the digital filters precision of the filter depends on the length (size) of the register used to store the filter coefficients. However by increasing the register bit-length (in hardware) the performance characteristics of the filter like accuracy, dynamic range, stability, frequency response and tolerance can be enhanced.
- The digital filters are programmable. Hence the filter co-efficient can be changed any time to implement to adaptive features.
- A single digital filter can be used to process multiple signals by using the technique of multiplexing.

D. Disadvantages of Digital Filters

- The bandwidth of the discrete signal is limited by the sampling frequency. The bandwidth of real discrete signal is half the sampling frequency.
- The performance of digital filters depends on the hardware (i.e., depends on the bit length of the register in the hardware used to implement the filter)

E. Comparison of FIR & IIR Filters

TABLE 1: COMPARISON OF IIR & FIR FILTERS

IIR FILTER	FIR FITER
1. The filter can be designed by considering the infinite samples of impulse response.	1. The filter can be designed by considering the finite samples of impulse response.
2. Impulse response cannot be directly converted to digital filter transfer function.	2. Impulse response can be directly converted to digital filter transfer function.
3. The digital filter can be obtained by first design the analog filter and then transferring analog to digital filter.	3. The digital filter can be designed directly to achieve the desired specification.
4. Linear phase characteristics cannot be achieved.	4. Linear phase filter can be easily designed.
5. It is preferred when some phase distortion can be tolerated.	5. It is preferred when there is a need for linear phase.
6. For a given amplitude response requires lesser order filter and requires less time for computation.	6. Requires relatively larger order filter and more computation time and hence more memory.
7. Realized recursively hence stability cannot be ensured.	7. Realized non-recursively hence inherently stable.

D. Advantages of FIR Filters

- FIR filters with exactly linear phase can be easily designed.
- With FIR, it is easier to synthesize filters of arbitrary frequency responses.
- Efficient realization of FIR filter exist as both recursive and non-recursive structures.
- FIR filters realized non-recursively i.e. by direct convolution are always stable.
- Roundoff noise, which is inherent in realization with finite precision arithmetic can easily be made small for non-recursive realization of FIR filters

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E. Disadvantages of FIR Filters

- A large value of N, the impulse response duration, is required to adequately approximate sharp cutoff filters. Hence a large amount of processing is required to realize such filters when realized via slow convolution.
- The delay of linear phase FIR filters need not always be an integer number of samples. This non-integral delay can lead to problem in some signal processing applications.

IV. WINDOW FUNCTIONS

Windowing technique is the straight forward method to obtain the filter impulse response with minimal computational effort and relative success of windows is due to their simplicity and in ease of use. The Finite Impulse Response sequence used in FIR filter design (ref.6) are obtained from infinite duration impulse response by truncating the infinite series at $n = \pm N$. But, this results in undesirable oscillations in the pass band and stop band of the digital filter. This is due to slow convergence of the Fourier series near the points of discontinuity. These undesirable oscillations can be reduced by using a set of time limited weighting functions, w(n), referred to as window functions, and to modify the Fourier coefficients.

A. Obtaining finite length impulse response by Truncation:

One way to obtain a finite length impulse response is to simply truncate an infinite impulse response. Suppose that the desired frequency response is $H_d(e^{jw})$ with a Fourier series representation.

$$H_d(e^{jw}) = \Sigma h_d(n)\, e^{-jwn} \quad -\infty < n < \infty \quad \text{------------ } 1$$

The impulse response $h_d(n)$ has infinite length and this frequency response is that of an IIR system. Also the system is not reliable since $h_d(n)$ is not equal to zero for n<0. The realizability problem can be easily solved by simply shifting the sequence to the right if it has only a finite number of non-zero terms for n<0.

We examine some ways to obtain a new frequency response $H(e^{jw})$ from Equation 4 that will represent a FIR system. Once this is done the realizability problem can be easily solved. One obvious way to achieve this desired representation is to simply truncate the Fourier Series of Equation 1.

In order to obtain a realizable filter, we can truncate the impulse response sequence to obtain a new sequence having length N with defining relation.

$$h(n) = \begin{cases} h_d(n) & 0 <= n <= N-1 \\ 0 & \text{otherwise} \end{cases} \quad \text{------------- } 2$$

The frequency response corresponding to this finite duration sequence is:

$$H(e^{jw}) = \Sigma h_d(n)\, e^{-jwn} \quad -\infty < n < \infty$$
$$= \Sigma h(n)\, e^{-jwn} \quad 0 < n < N-1 \quad \text{------------- } 3$$

It can also be obtained from Equation 1 by truncation since we use only the N values of $h_d(n)$, $0 <= n <= N-1$, We could also think of obtaining Equation 3 by looking through a window and seeing only these terms of $h_d(n)$. For this reason the process of obtaining Equation 3 from Equation 1 is referred to as windowing.

B. Effects of Windowing

The major effects of windowing are:

- Discontinuities in $H(e^{jw})$ are converted into transition bands between values on either side of the discontinuity and the width of these transition bands depends on the width of the main lobe of $W(e^{jw})$.
- Ripples from the side lobes of $W(e^{jw})$ produces approximation errors for all W.

C. Desirable Characteristics of Windowing

- The Fourier transform of the window function $W(e^{jw})$ should have a small width of main lobe containing as much of the total energy as possible.
- The Fourier transform of the window function $W(e^{jw})$ should have side lobes that decrease in energy rapidly as W tends to π.

D. Gibb's Phenomenon

In the Figure 2 oscillations or ringing effect takes place near band edge (i.e. w_c) of the filter. These oscillations or ringing is generated because of side lobes in the frequency response w(n) of the window function. This oscillatory behaviour near the band edge of the filter is called Gibb's phenomenon. These side lobes are generated because of abrupt discontinuity of the window function. In case of rectangular window the side lobes are larger in size since the discontinuity is abrupt. Hence ringing effect is maximum in rectangular window.

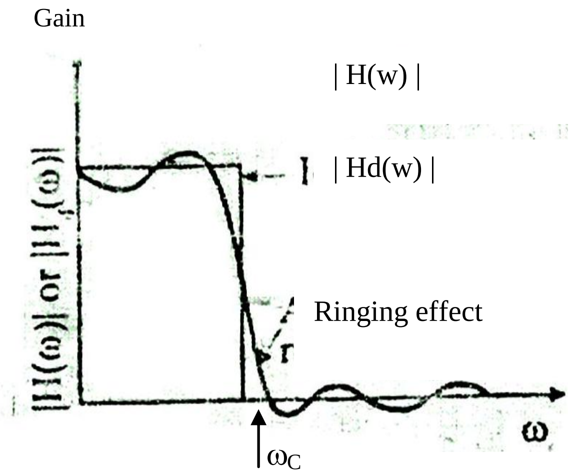

Fig.2: Magnitude Response of a Low-Pass Filter

E. How to Overcome Gibb's Phenomenon?

To overcome the Gibb's phenomenon i.e. effect of ringing in pass band and stop band we go through the use of,

- Window that tapers smoothly to zero at each end.
- Providing a smooth transition from the pass band to stop band.

By tapering the window smoothly to zero at each end the height of the side lobe can be diminished; however, this is achieved at the expense of a wider main lobe and thus a wider transition at the discontinuity.

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D. Types of Window Functions

To reduce Gibb's phenomenon which is more in rectangular window, we go for tapering of window smoothly to zero i.e., triangular and raised cosine windows. The triangular (Bartlett) window has been chosen such that it has tapered sequences from the middle on either side. In the stop band the response is smoother but the attenuation is less that produced by the rectangular window, so it is usually not a good choice.

The raised cosine windows are smoother at the ends, but closer to one at the middle. The smoother ends and the broader middle section produces less distortion of $h_d(n)$ around n=0.Here the particular window among them is selected depending upon the application.

The different types of window functions are Bartlett (Triangular), Blackman, Hamming, Hanning, Kaiser and Rectangular.

TABLE 2: TIME-DOMAIN SEQUENCE OF DIFFERENT WINDOW FUNCTIONS

Name of the Window	***Time-Domain Sequence***
1. Rectangle	$W(n) = 1; \quad 0 \le n \le N-1$
2. Bartlett	$W(n) = \begin{cases} 2n / N-1, & 0 \le n \le N-1 \\ 2 - 2n/N-1, & N-1/2 \le n \le N-1 \end{cases}$
3. Hamming	$W(n) = 0.54 - 0.46 \cos(2\pi n / N-1)\ ; \quad 0 \le n \le N-1$
4. Hanning	$W(n) = ½ [1-\cos(2\pi n / N-1)]; \quad 0 \le n \le N-1$
5. Blackman	$W(n) = 0.42 - 0.5 \cos(2\pi n / N-1 + 0.08 \cos(4\pi n / N-1); \quad 0 \le n \le N-1$
6. Kaiser	$W(n) = \dfrac{I_0 [\ \alpha \sqrt{(N-1/2)^2 - (n - (N-1/2))^2}]}{I_0 [w_a(N-1/2)]}$

The Figure 3 shows the time-domain sequence shapes of different window functions.

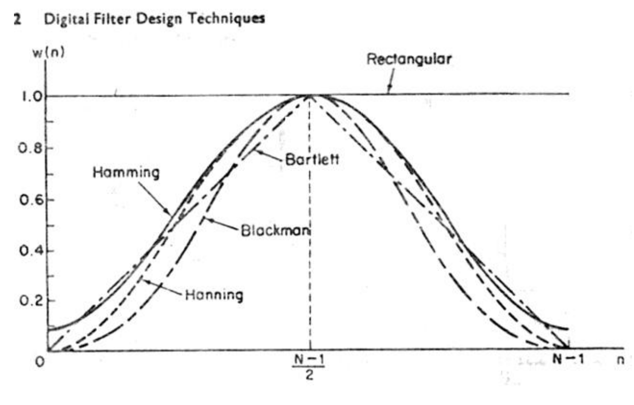
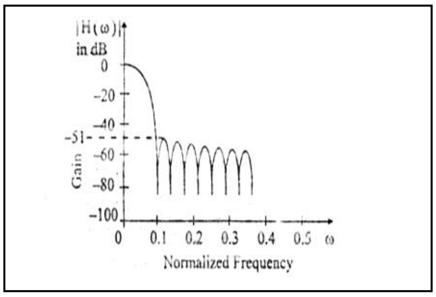
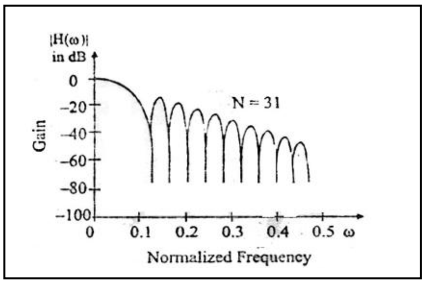


Fig. 3: A comparison of different Window Functions

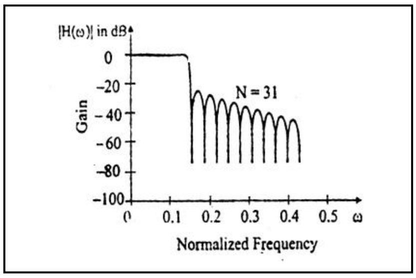
Log-magnitude response of FIR lowpass filter designed using different window functions.



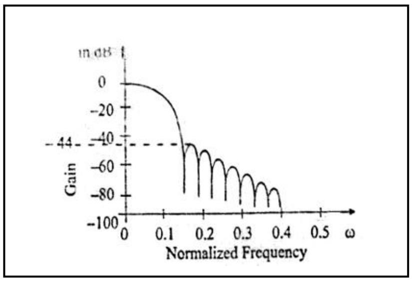
Hamming window



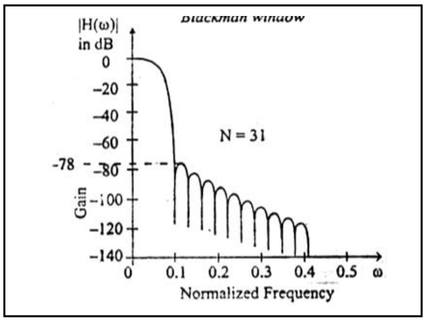
Triangular window



Rectangular window



Hanning window



Blackman window

Fig. 4: Log-magnitude response of FIR lowpass filters designed using different window functions

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The basic parameters for lowpass filter design are summarized in the following Table. 3.

Table. 3: COMPARISON OF DIFFERENT WINDOWS

WINDOW	Peak amplitude of Side Lobe (dB)	Transition width of Main Lobe	Min. Stopband Attenuation(dB)
Rectangular	-13	4π / N	-21
Bartlett	-25	8π / N	-25
Hanning	-31	8π / N	-44
Hamming	-41	8π / N	-53
Blackman	-57	12π / N	-74

For the windows the width of the main lobe is inversely proportional to N, that is, increasing the window length decreases the main lobe width, which results in a decrease in the transition band of the filter. However, the minimum stop band attenuation is independent of the window length and is a function of the selected window. Therefore, to achieve a desired stop band attenuation the designer must find a window that meets the design specifications. It should also be emphasized that windows with low side lobe levels have broader main lobe widths requiring an increase in the order of the filter N to achieve the desired transition width.

1) Hamming Window:

Hamming noted that a reduction in the first side lobe level can be achieve by adding a small constant value to the cosine window. This produces the Hamming window, denoted by $w_h(n)$ and given by:

$$W_h(n)=\begin{cases} 0.54-0.46\cos(2\pi n/N-1); & -(N-1/2)<=n<=(N-1/2) \\ 0 & ; \quad \text{Otherwise} \end{cases}$$

The low pass filter magnitude and log-magnitude responses when designed using the Hamming window are shown in Figure 4. The stop band attenuation in the low pass filter magnitude response is limited by the side lobe level of the window function. Even though the Hamming window achieved an attenuation of 51dB (or gain of –51dB) in the stop band in the low pass filter it may not be sufficient for some applications. Hamming reduce the side lobe magnitude while maintaining the main lobe width.

2) Kaiser Window:

Kaiser window is developed by the scientist Kaiser in terms of zeroth order modified Bessel functions of the first kind. In a Kaiser window, the side lobe levels can be controlled with respect to the main lobe peak by varying a parameter, α. Similar to the other windows the main lobe width can be varied by adjusting the length of the window, which in turn adjusts the transition width of the filter. Therefore, FIR digital filters can be efficiently designed using the Kaiser window function.

The Kaiser window function is given by

$$Wk(n) = \begin{cases} I_0(\beta) / I_0(\alpha); & \text{for } |n| <= N-1/2 \\ 0 & ; \quad \text{Otherwise} \end{cases}$$

Where, α is an independent variable. The parameter β is expressed by:

$$\beta = \alpha\,[1-[2n / N-1]^2]^{0.5}$$

The modified Bessel of the first kind, $I_0(x)$ can be computed from its power series expansion given by:

$$I_0(x) = 1 + \Sigma\,[1/k![x/2]^k]^2$$

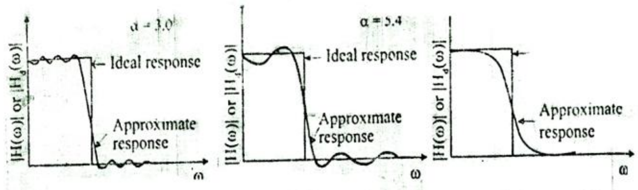

Fig d : Magnitude response of Kaiser window for different values of α (when N = 51)

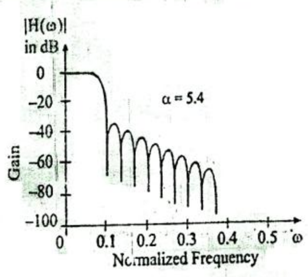

Fig. 5: Log-magnitude response of FIR low pass filter designed using Kaiser window

a) Advantages of Kaiser Window:

- It has good stop band attenuation.
- Minimal computational effort compared with other window function.
- Varying N can achieve desired transition width.
- It has variable parameter that controls the trade off between side lobe amplitude and side lobe width.

V. SOFTWARE IMPLEMENTATION

FIR filters can be designed using any of the following methods:

- Fourier series method.
- Window method
- Frequency sampling method
- Optimal or Minimax method

- **Fourier Series Method**: In Fourier series analysis, any periodic function (period equal to the sampling frequency) can be expressed as a linear combination of complex exponential. Therefore, the desired frequency response of an FIR filter can be represented by the Fourier series.
- **Window Method**: The window method involves a straightforward analytical procedure however in some cases iteration is necessary to obtain the desired result.
- **Frequency Sampling Method**: The third method of design is frequency sampling. A desired frequency response is uniformly sampled and filter coefficients are then determined from these samples using inverse discrete Fourier transform.
- **Optimal or Minimax Design**: Optimal FIR filters are based on the representations of the frequency response. Efficient computer software is available for designing optimal filters, making this technique very attractive.

Before starting down the algorithm for our Assembly Language Program, we would like to present the design considerations and the parameters involved therein. First of all, the sampling frequency, which we have selected, is 8KHz. Using this we go ahead and calculate the timer interrupt rate, which is given by.

TINT = CLKOUT / (TDDR +1) (PRD +1)

The CLKOUT is calculated using the relation

CLKOUT = System clock / 2

System clock is a 40 MHz crystal, therefore the CLKOUT is: CLKOUT = 40 MHz / 2

= 20 MHz

Now our timer interrupt rate is obtained as follows

TINT = 20 MHz / (1+1) (0+1)

The content of TDDR is made equal to 1 and the content of PRD is made equal to zero.

The sampling frequency is given by

Sampling frequency = TINT / (2*A*B)

Where, A and B denote the transmit and receive buffers. Therefore A*B is equal to 625. We will choose a value for A and B such that A*B > = 625. Let A = 18 and B = 36

Therefore the transmit buffer TA = 18 and RA =36. Similarly TB = 18 and RB = 36.

A. Flow Chart for Real Time Implementation of FIR Filter:

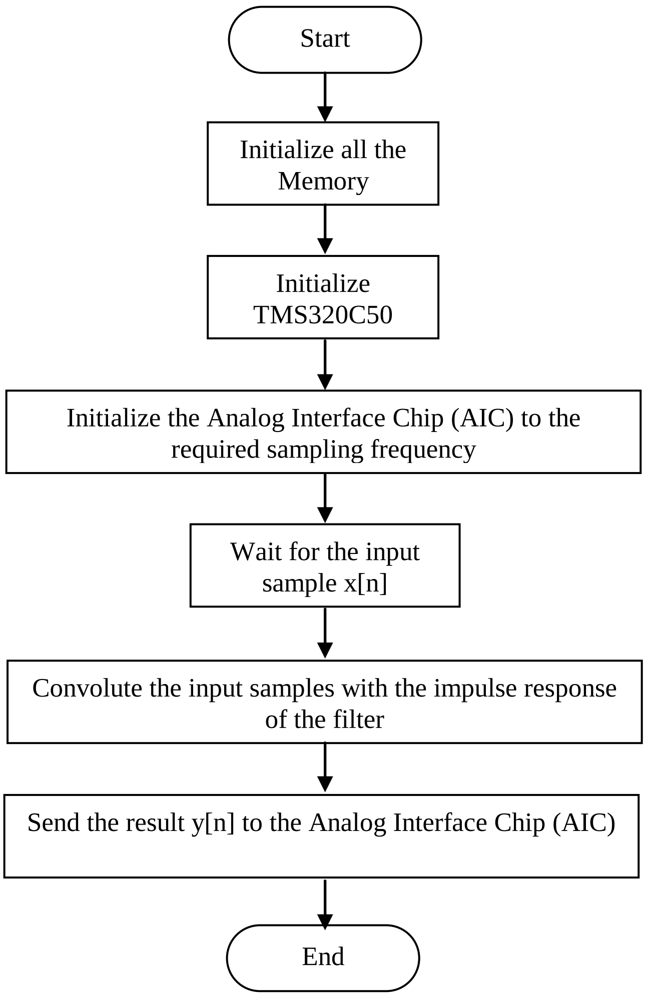

Fig. 6: Flow chart for Real Time Implementation of Digital FIR filter

The flow chart is a visual or graphical representation of an algorithm. It indicates the process of solution, the relevant operations and computations, the point of decisions and other information as part of the solution. By drawing a flow chart, the algorithm can be better understood since it is in pictorial form. Flow charts are normally introduced as an intermediate step to assist the preparation of a computer program.

B. Algorithm to Obtain Filter Coefficients

In order to obtain the coefficients needed for the FIR filter, we make use of 'C' language and MATLAB functions. We use the specifications like pass band and stop band frequencies, pass band ripple, minimum stop band attenuation and sampling frequency.

The following section present the algorithm required to implement the program:

Step 1: The sampling frequency fs is chosen to be 8KHz for the purpose of a successful design.

Step 2: Similarly the cutoff frequency fc is chosen to be 1500Hz.

Step 3: The pass band ripple and stop band attenuation are selected for a proper design.

Step 4: The order of the filter is calculated using the relation.

$N>=F_sD / \Delta F + 1$; Where $\Delta F = f_s - f_p$

The order of the filter, which is obtained, is rounded off to the nearest odd value.

Step 5: The window coefficients are calculated for the specifications.

Step 6: The filter coefficients are calculate using the relation

$H (n) = \Sigma h_d (n) w (n)$

Step 7: The resulting coefficients, which are found to be symmetrical, are stored in an .asm file.

C. Algorithm for Programming

Step 1: The first step involves the initialization of all the memory devices.

Step 2: The DSP TMS320C50 is then initiated for serial port operation.

Step 3: The analog interface chip is initialized to the required sampling frequency.

Step 4: The processor is put in the wait state until the input signal X[n] is not received by it.

Step 5: Then the complex convolution of the input sample takes place with the impulse response of the filter.

Step 6: The result y[n] is send to the analog interface chip for D/A conversion.

VI. CONCLUSIONS & SCOPE FOR FUTURE WORK

A. Conclusion

We have implemented the Low Pass FIR filter using Kaiser and Hamming Window Techniques by making use of TMS320C50 DSP kit. The cutoff frequency we have designed and the practical response we have obtained are happens to be one and the same. The TMS320C50 is a fixed-point processor having a lesser accuracy as compared to floating point processor. The FIR filter has certain inherent disadvantages as discussed in earlier Section III (f). Due to above mentioned limitations, the low pass FIR filter we have designed does not exactly have a cut-off frequency of 1.5 KHz as mentioned in our design, but exhibits a cut-off around 1.35 KHz to 1.55 KHz.

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B. Scope for Further Improvements

We have attempted to implement Kaiser window for this particular design. Optimal linear phase FIR filters could be used for the same purpose to get the precise control of the critical frequency. Further, a floating-point processor and higher order processor could be used to improve the accuracy of the filtering operations so that the filter would make a transition into the stop band at the designed cut off frequency. It could also be used as a part of work involving filtering operations like speech processing, voice recognition and the like.

C. Applications

- This module is useful in Aircraft where along with pilot's voice; the engine sound will also be present.
- This also useful in military applications for transmitting the exact message signal without noise to different stations.
- Digital filters plays an important role in medical applications where most of the signals have low frequencies that might be distorted due to the drift in an analog circuit.

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