BER AND COST ANALYSIS OF FIR FILTERS FOR DIGITAL TRANSMISSION SYSTEM

MANISH TRIKHA, RAJESH MEHRA
M.E Scholar, Associate Professor, Department of Electronics & Communication Engineering, NITTR, Chandigarh, UT, India
Email: manish.trikha@gmail.com

Abstract: In this paper a low pass FIR filter design has been presented using different approaches for digital transmission systems. The developed FIR filter has been design using Raised cosine, Nyquist equiripple and halfband approaches. The design filters have been compared in term of their performance and implementation cost. All the three design are simulated using MATLAB. It can be observed from the simulated result the performance of Raised cosine filter is slightly better as compared to other designs on the other hand implementation cost of Nyquist equiripple filter is better by 75% as compared to raised cosine filter whereas for halfband filter is better by 85% as compared to Raised cosine filter.

Keywords: BER, Raised cosine filter, Nyquist equiripple FIR Filter, Multistage halfband filter.

I. INTRODUCTION

Digital filtering is one of the most powerful tools of DSP. Apart from the obvious advantages of virtually eliminating errors in the filter associated with passive component fluctuations over time and temperature etc., digital filters are capable of performance specifications that would, at best, be extremely difficult to achieve with an analog implementation. In addition, the characteristics of a digital filter can be easily changed under different software interface. Therefore, they are widely used in adaptive filtering applications in communications such as Pulse shaping, echo cancellation in modems, noise cancellation, and speech recognition. Digital filters cannot be applied directly to the real world signal as real world signals are continuous in nature such as voice, speech so the some sort of signal conversion before applying digital filtering is required. There are two fundamental types of digital filters: finite impulse response (FIR) and infinite impulse response (IIR).

As the terminology suggests, these classifications refer to the filter’s impulse response. By varying the weight of the coefficients and the number of filter taps, virtually any frequency response characteristic can be realized with an FIR filter. FIR filters can achieve performance levels which are not possible with analog filter techniques (such as perfect linear phase response). However, high performance FIR filters generally require a large number of multiply-accumulates and therefore require fast and efficient DSPs. While, IIR filters tend to mimic the performance of traditional analog filters and make use of feedback. Therefore their impulse response extends over an infinite period. Because of feedback, IIR filters can be implemented with fewer coefficients than for an FIR filter.

The low pass filter can be represented by some of the filter characteristics that is the transition band represents frequencies in the middle, which may receive some attenuation but are not removed completely from the output signal. In Fig. 1, which shows the frequency response of a lowpass filter, ωp is the passband ending frequency, ωs is the stopband beginning frequency, and As is the amount of attenuation in the stopband. Frequencies between ωp and ωs fall within the transition band and are attenuated to some lesser degree.

Fig 1 Frequency response of a low pass filter

The difference equation of M order of the recursive digital filters (FIR) can be represented as:

\[ B(n) = \sum_{k=0}^{M-1} h(n)A(n-k) = \sum_{k=0}^{M-1} b_k A(n-k) \] (1)

Where, B (n) is the output signal, h (n) is the filter coefficients and k is the order of the filters. Can express the output signal in frequency domain by convolution of the input signal A (n) and the impulse response h (n) [3].

\[ B(n) = A(n) * h(n) \] (2)

The output signal is determined as,

\[ B(n)=A(0)*h(n)+A(1)*h(n-1)+A(2)*h(n-2)+\ldots\ldots.+A(n)h(0) \] (3)

In differential equation, the coefficient \( b_k \) equals to the successive value h (n) of unit-sample response.

\[ H(z) = \sum_{k=0}^{N-1} b_k z^{-k} \] (4)

BER And Cost Analysis Of Fir Filters For Digital Transmission System
II. DIGITAL TRANSMISSION SYSTEM

As digital technology grows up very rapidly, an ever-increasing number of RF applications will involve the transmission of digital data from one point to another Fig 2. Some examples include cable modems, mobile phones and high-definition television (HDTV). The task is to transmit these bits between source and destination [4].

In modern data transmission systems, bits or groups of bits (symbols) are typically transmitted in the form of individual pulses of energy. Sometimes Rectangular pulse is probably the most fundamental. It is easy to implement in a real-world system because it can be directly compared to opening and closing a switch, which is synonymous with the concept of binary in information. For example, a “1” bit might be used to turn on an energy source for the duration of one pulse interval (τ seconds), Fig 3 which would produce an output level, “A”. Alternately, a “0” bit would turn off the energy source, producing an output level of zero during one pulse interval.

Fig 2 Digital data transmission system

Pulses are sent by the transmitter are detected by the receiver in any data transmission system. At the receiver, the goal is to sample the received signal at an optimal point in the pulse interval by the matched filter Fig 2 to maximize the probability of correct decision. This implies that the fundamental shapes of the pulses be such that they do not interfere with one another at the optimal sampling point. There are two criteria that ensure non-interference.

i. The pulse shape exhibits a zero crossing at the sampling point of all pulse intervals except its own. That is Minimized intersymbol interferences (ISI).
ii. The shape of the pulses be such that the amplitude decays rapidly outside of the pulse interval. That is high stop band attenuation

The rectangular pulse, meets first requirement because it is zero at all points outside of the present pulse interval. It cannot cause interference during the sampling time of other pulses. The trouble with the rectangular pulse, however, is that it has significant energy over a fairly large bandwidth as indicated by its Fourier transform shown in Fig 3. In fact, because the spectrum of the pulse is given by the familiar sin (px)/px (sinc x) response, its bandwidth actually extends to infinity. The unbounded frequency response of the rectangular pulse makes it unsuitable for modern transmission systems. This is where pulse shaping filters come into play. If the rectangular pulse is not the best choice for band-limited data transmission, then what pulse shape will, decay quickly, and provide zero crossings at the pulse sampling times. There are several choices that have but in most systems Raised cosine filter is used to shape the input pulse. In this process of digital pulse shaping the impulse signals are applied to the pulse shaping filter at the instant of sampling points so the output pulses are of same shape of impulse response of the filter.

Unlike the rectangular pulses, the raised cosine pulses takes on the shape of a sinc pulse Fig 4. Unfortunately, the name “raised cosine” is misleading. It actually refers to the pulse’s frequency spectrum, not to its time domain shape. The raised-cosine filter is an implementation of a low-pass Nyquist filter, i.e., one that has the property of vestigial symmetry. This means that its spectrum exhibits odd symmetry about 1/T, where T is the symbol-period of the communications system. Characterised by two values β the roll-off factor, and T the reciprocal of the symbol-rate. The impulse response of such a filter is given by equation 5.

\[ h(t) = \text{sinc}(\frac{t}{T}) \frac{\cos(\frac{\beta \pi t}{2})}{\frac{\pi^2 t^2}{4T^2}} \]  

Fig 4 Frequency response of raised-cosine filter

Under certain conditions a low pass filter can be designed to have a number of zero-valued coefficients. When used as interpolation filter these filter preserves the non-zero sample of the up sampler output at the interpolator output. Moreover due to the presence of these non-valued coefficients, these filters are computationally more efficient than other low pass filter of the same order. Nyquist filters can replace raised cosine filters for a fraction of the cost as they have an optimal equiripple response. The same stop-band attenuation and transition width can
be obtained with a much lower order. The transition width requirement can be deduced from the roll-off and interpolation factors. Nyquist filters considered in this paper have a minimax stopband behavior. For a well attenuated stopband, the passband will be flat and the transition region will have the appropriate symmetry to guarantee the zero crossing property. For an input sequence \( x(n) \) and a filter response \( h(n) \), the output can be written in terms of a convolution

\[
y(n) = x(n) * h(n)
\]

In the design of sharp cutoff FIR filters, a multistage design based on half-band filters is very efficient. The efficiency of half band filters derives from the fact that about 50% of the filter coefficients are zero thus cutting down the implementation cost. Half band filters can be used for signal oversampling by 2. The resulting signal contains the original samples and the interpolated point between the original samples filtered by the non-zero coefficients [7]-[9]. The two main properties of half band filter:

a. Number of non-zero valued coefficients is nearly half of the filter length
b. The non-zero coefficients exhibits symmetry property

### III. MATLAB BASED SIMULATION

For the simulation of filter in MATLAB the value of stop-band attenuation is 62dB which is well below [8] the required range for digital transmission system that is between (60-80 dB) and roll off factor is 0.23 which is also between the specified range (0.1 to 0.5). Raised cosine filter with Interpolation factor 6, Roll-off factor 0.23 and stop band Attenuation 62 dB is shown in Fig 5.

The magnitude response and impulse response (pulse shape) of the Nyquist Equiripple FIR Filter with Interpolation factor 6, Roll-off factor 0.23 and stop band Attenuation 62 dB is shown in Fig 6.

The magnitude response and impulse response (pulse shape) of the Multistage Halfband filter with Interpolation factor 6, Roll-off factor 0.23 and stop band Attenuation 62 dB is shown in Fig 7.

### IV. RESULT ANALYSIS

As can be seen from Table 1, Raised cosine filter provide 300 multipliers and 295 Adders while at the same time Nyquist Equiripple and Halfband filters provide 76 and 38 Multipliers and 71 and 35 Adder
respectively. Which provide significant savings in hardware implementation.

### Table 1: Filter Cost Comparison Table

<table>
<thead>
<tr>
<th>Filter Type</th>
<th>Multiplications / Input Sample</th>
<th>Additions / Input Sample</th>
</tr>
</thead>
<tbody>
<tr>
<td>Raised Cosine</td>
<td>300</td>
<td>295</td>
</tr>
<tr>
<td>Nyquist FIR-Equiripple</td>
<td>76</td>
<td>71</td>
</tr>
<tr>
<td>FIR-Halfband</td>
<td>38</td>
<td>35</td>
</tr>
</tbody>
</table>

The BER- SNR chart (Fig 8), shows that when 16-QAM modulation scheme and an additive white Gaussian noise channel are used for transmitting-receiving the digital data.

Then for 2dB SNR Raised Cosine filter provided 0.2712 BER while FIR-Nyquist Equiripple and FIR-Halfband provided 0.2774 and 0.2765 respectively and also for different SNR values that is for 5dB and 10dB Raised Cosine filter shows 0.2035 and 0.0714 BER which shows better BER performance over FIR-Nyquist Equiripple and FIR-Halfband over wide range of SNR.

### CONCLUSION

In this paper, the simulation of different pulses shaping digital filters are presented such as raised cosine filter, Nyquist equiripple FIR filter and multistage FIR-Halfband filter and see that multistage FIR-Halfband filter and Nyquist Equiripple filter provide more saving in terms the hardware and operation per sample then that of conventionally used raised cosine filter but If compare the BER performance then Raised cosine filter is better than Nyquist equiripple and FIR halfband, so for the case where little performance degradation can be tolerated in terms of BER then there is another alternatives which provides significant improvement in cost.

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