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# Efficient Inter Symbol Interference and Filtering Model for Digital Telecommunication System Using MATLAB

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## Abstract:
The paper emphasizes the efficient inter symbol interference and filtering model for digital telecommunication system using MATLAB. The problem in this study is to remove the unwanted signals from the input signal in the reality. The solution in this paper is to design the appropriate filtering system in the telecommunication based on the signal processing approaches. The implementation of ISI filtering was developed based on the background theory of the filtering design. The implementation was completed by using MATLAB. The results confirm that the developed filtering design was met the high performance of the digital telecommunication system.

## Keywords:
Inter Symbol Interference, Filtering Model, Digital Telecommunication System, MATLAB, Future Technology

## 1. Introduction

Digital communication involves transmission of messages using finite alphabets (finite symbols) during finite time intervals (finite symbol interval). Any communication system (be it analog or digital in nature) in the electronic consumer market (be it hard disk drives, Compact Discs, telephony, mobile communication systems, etc..,) is made up of the following elements as represented in Figure 1 [1,2,3].

The prime goals of a communication design engineer (one who designs a practical communication system) would be to

(1) Reduce the bandwidth needed to send data.

Bandwidth, a limited and valuable resource, is the difference between the highest and the lowest frequency allocated for transmitting a message in any communication

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system. For example in GSM technology the typical bandwidth allocated for a single user is 200 KHz. More bandwidth provides space to transmit more data as well as more transmission rate (measured in bits per second - "bps"). The goal of reduced bandwidth is needed because of the growing bandwidth demands and the limited availability of communication spectrum. A downloading speed of 56Kbps was felt sufficient few years ago, but now it is not so. Hence it is essential to send more data in lesser bandwidth. This is achieved by compressing the data at the transmitting end and decompressing it at the receiving end. A "Source Encoder" and a "Source Decoder" serve this purpose.

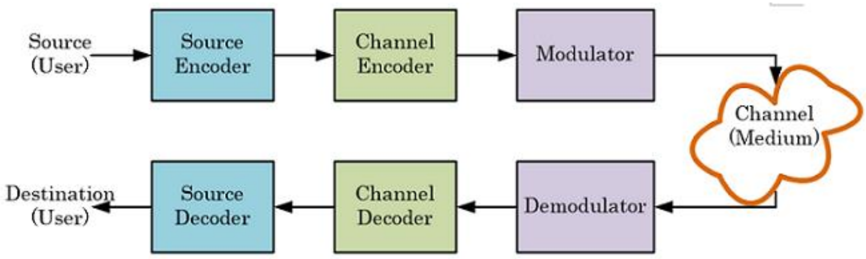


***Figure 1.*** *Ultimate Concepts of Communication System.*

(2) To make data robust against harsh environments

Data will get corrupted when it is sent in harsh media (referred to as "channel"). For example mobile phones are operated in a very noisy environment in which the noise sources may be one or more of the following: interference from other mobile users, ignition noise, thermal noise, multipath interference and other man made noises. Channel coding is a technique to make the transmitted data robust to such noises, meaning that you can still recover your data (using a Channel Decoder) intact even if it is corrupted by certain amount of noise.

(3) Send data over a long distance

Obviously data has to be sent over a long distance through any media used for/by the communication system. The media may be a simple twisted pair copper wires used in telephone networks or the air media in the case of a mobile or satellite communication system. In the physical world, it is not possible to send a signal (carrying data) over infinite distance. According to the inverse square law of distance the intensity of the transmitted signal is inversely proportional to the square of the distance [4-8].

The rest of the paper is organized as follows. The background theory of Inter Symbol Interference and Filtering is mention in the next section. The Implementation of Raised Cosine Filter is presented based on the mathematical expressions. The results and discussions section give the simulation results based on the mathematical model of the developed system. Finally, the conclusion section is presented in the last section of this paper.

## 2. Background Theory of Inter Symbol Interference and Filtering

### *2.1. Fundamental Concepts*

Most of the communication channels (data storage devices, optical, wireless channels etc...) can be considered as band-limited linear filters. The channels can be modeled as having the following frequency response (over a given bandwidth W).

$$H(f) = A(f)e^{j\theta(f)}$$

Here A(f) is amplitude response and θ(f) is the phase response of the channel over the given bandwidth W. The envelope or group delay for the given filter model is defined as,

$$\tau(f) = -\frac{1}{2}\frac{d\theta(f)}{df}$$

A channel is considered non-distorting (within the given bandwidth W occupied by the transmitted signal), when the amplitude response is constant and the phase response is a linear function of frequency (within the given bandwidth W). That is, the group delay, given above, is constant for a non-distorting channel.

Amplitude distortion occurs when the amplitude response is no longer a constant. Delay/phase distortion occurs when the phase response is not a linear function of frequency (that is, the envelope or group delay is not a constant).

A non-ideal channel frequency response is caused by amplitude and phase distortion. When a succession of pulses transferred through a non-ideal channel, at a rate 2W symbols/second (R=1/T=2W -'Nyquist Rate'), will get distorted to a point that they are no longer distinguishable from each other. This is called Inter Symbol Interference (ISI). It means that a symbol transmitted across a non-ideal channel will be affected by the other symbols. The symbols smear on to each other and therefore are indistinguishable at the receiver. To minimize the effect of ISI, pulse shaping filters are generally employed at the transmitter and receiver to match the spectral characteristics of the signal with that of the channel.

Generally following strategies are employed to mitigate ISI:

(1) Use ideal rectangular pulse shaping filters to achieve zero ISI:

Maximum transferable data rate that is possible with zero ISI is R=1/T=2W symbols/second, provided the "ideal rectangular transmit and receive pulse shaping filters" are used. Ideal rectangular transmit and receive filters are practically unrealizable. So this option is not viable to achieve zero ISI.

(2) Relax the condition of transmitting at maximum rate R=1/T=2W, to achieve zero ISI:

If the transmission rate is reduced below 2W (i.e. R=1/T<2W), then it is possible to implement practically realizable filters. Raised Cosine and Square Root Raised cosine filters are generally used to achieve zero ISI, if the transmission data rate is reduced below 2W. The signals generated using this method are called full response signals.

(3) Relax the condition of zero ISI and transfer at Nyquist Rate (R=1/T=2W):

In this method, we relax the condition of achieving zero ISI, so that the data can be transferred at maximum possible rate (R=1/T=2W). Instead of achieving zero ISI, this method introduces controlled amount of ISI in the transmitted signal and counters it upon receiving the signal at the receiver. The transmit filter is designed to introduce 'deterministic' or 'controlled' amount of ISI and is counteracted in the receiver side. Methods like duobinary signaling, modified duobinary signaling are employed under this category. The resulting signals are called partial response signals which are transmitted at Nyquist rate of 2W symbols/second. This method is also called "Correlative Coding".

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### *2.2. Correlative coding – Duobinary Signaling*

The condition for zero ISI (Inter Symbol Interference) is

$$\mathrm{p}(nT)=\begin{cases}1, n=0\\0, n\neq 0\end{cases}$$

It states that when sampling a particular symbol (at time instant nT =0), the effect of all other symbols on the current sampled symbol is zero.

As discussed in the previous section, one of the practical ways to mitigate ISI is to use partial response signaling technique (otherwise called as “correlative coding”). In partial response signaling, the requirement of zero ISI condition is relaxed - as a controlled amount of ISI is introduced in the transmitted signal and is counteracted in the receiver side.

By relaxing the zero ISI condition, the above equation can be modified as,

$$\mathrm{p}(nT)=\begin{cases}1, n=0,1\\0, otherwise\end{cases}$$

This states that the ISI is limited to two adjacent samples. Here we introduce a controlled or “deterministic” amount of ISI and hence its effect can be removed upon signal detection at the receiver.

Duobinary Signaling: The following figure shows the duobinary signaling scheme.

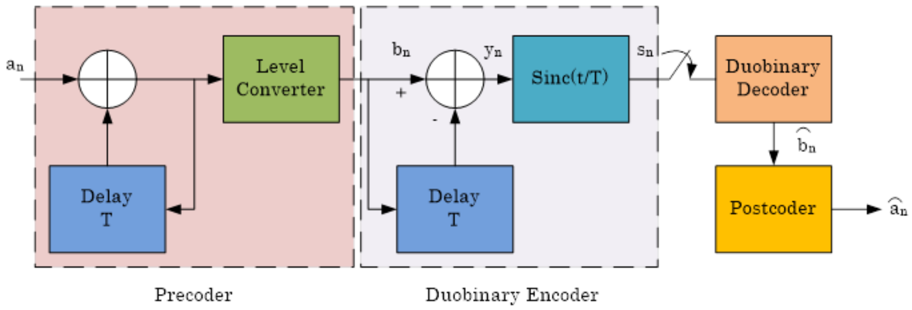


***Figure 2.*** *Duobinary Signaling Scheme.*

*Encoding Process:*

(1) $a_n$ = binary input bit; $a_n \epsilon\{0,1\}$.

(2) $b_n$ = NRZ polar output of Level converter in the precoder and is given by

$$b_n=\begin{cases}-d, if a_n=0\\+d, if a_n=1\end{cases}$$

(3) $y_n$ can be represented as

$$y_n=b_n+b_{n+1}=\begin{cases}+2d, if a_n=a_{n-1}=1\\0, if a_n\neq a_{n-1}\\-2d, if a_n=a_{n-1}=0\end{cases}$$

Note that the samples bn are uncorrelated (i.e either +d for “1″ or -d for “0″ input). On the other-hand, the samples $y_n$ are correlated (i.e. there are three possible values

+2d, 0, -2d depending on $a_n$ and $a_{n-1}$). It means that duobinary encoding correlates the present sample an and the previous input sample $a_{n-1}$.

(4) From the diagram, impulse response of the duobinary encoder is computed as

$$h(t) = \sin c\left(\frac{t}{T}\right) + \sin c\left(\frac{t-T}{T}\right)$$

*Decoding Process:*

(5) The receiver consists of a duobinary decoder and a postcoder (inverse of precoder).The duobinary decoder implements the following equation (which can be deduced from the equation given under step 3 (see above))

$$\bar{b}_n = y_n - \bar{b}_{n-1}$$

This equation indicates that the decoding process is prone to error propagation as the estimate of present sample relies on the estimate of the previous sample. This error propagation is avoided by using a precoder before duobinary encoder at the transmitter and a postcoder after the duobinary decoder. The precoder ties the present sample and the previous sample (correlates these two samples). The postcoder does the reverse process.

6) The entire process of duobinary decoding and postcoding can be combined together as one algorithm. The following decision rule is used for detecting the original duobinary signal samples {$a_n$} from {$y_n$}

$$\begin{aligned} & if y_n < d, then \bar{a}_n = 1 \\ & if y_n > d, then \bar{a}_n = 0 \\ & if y_n = 0, randomly guess \bar{a}_n \end{aligned}$$

### *2.3. Modified Duobinary Signaling*

Modified Duobinary Signaling is an extension of duobinary signaling. Modified Duobinary signaling has the advantage of zero PSD at low frequencies (especially at DC) which is suitable for channels with poor DC response. It correlates two symbols that are 2T time instants apart, whereas in duobinary signaling, symbols that are 1T apart are correlated.

The general condition to achieve zero ISI is given by

$$\mathrm{p}(nT) = \begin{cases} 1, n = 0 \\ 0, n \neq 0 \end{cases}$$

As discussed in a previous article, in correlative coding, the requirement of zero ISI condition is relaxed as a controlled amount of ISI is introduced in the transmitted signal and is counteracted in the receiver side. In the case of modified duobinary signaling, the above equation is modified as

$$\mathrm{p}(nT) = \begin{cases} 1, n = 0, 2 \\ 0, otherwise \end{cases}$$

This states that the ISI is limited to two alternate samples. Here a controlled or "deterministic" amount of ISI is introduced and hence its effect can be removed upon signal detection at the receiver.

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Modified Duobinary Signaling: Figure 3 shows the modified duobinary signaling scheme.

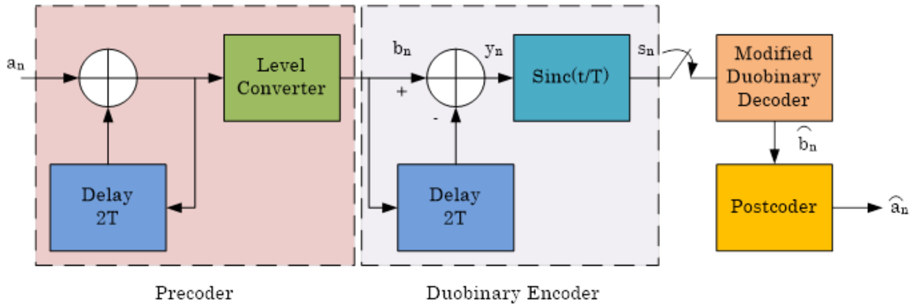


***Figure 3.*** *Modified Duobinary Signaling Scheme.*

*Encoding Process:*

(1) $a_n$ = binary input bit; $a_n \in \{0,1\}$.

(2) $b_n$ = NRZ polar output of Level converter in the precoder and is given by

$$b_n = \begin{cases} -d, if a_n = 0 \\ +d, if a_n = 1 \end{cases}$$

(3) $y_n$ can be represented as

$$y_n = b_n + b_{n-2} = \begin{cases} +2d, if a_n = a_{n-2} = 1 \\ 0, if a_n \neq a_{n-2} \\ -2d, if a_n = a_{n-2} = 0 \end{cases}$$

Note that the samples bn are uncorrelated (i.e. either +d for "1" or -d for "0" input). On the otherhand, the samples yn are correlated (i.e. there are three possible values +2d, 0,-2d depending on $a_n$ and $a_{n-2}$). It means that the modified duobinary encoding correlates present sample $a_n$ and the previous input sample $a_{n-2}$.

(4) From the diagram, impulse response of the modified duobinary encoder is computed as

$$h(t) = \sin c\left(\frac{t}{T}\right) + \sin c\left(\frac{t-2T}{T}\right)$$

*Decoding Process:*

(5) The receiver consists of a modified duobinary decoder and a postcoder (inverse of precoder).The modified duobinary decoder implements the following equation (which can be deduced from the equation given under step 3 (see above))

$$\bar{b}_n = y_n - \bar{b}_{n-2}$$

This equation indicates that the decoding process is prone to error propagation as the estimate of present sample relies on the estimate of the previous sample. This error propagation is avoided by using a precoder before modified duobinary encoder at the transmitter and a postcoder after the modified duobinary decoder. The precoder ties the present sample and previous to previous sample (correlates these two samples). The postcoder does the reverse process.

(6) The entire process of modified duobinary decoding and the postcoding can be combined together as one algorithm. The following decision rule is used for detecting the original modified duobinary signal samples {$a_n$} from {$y_n$}

$$
\begin{aligned}
&if y_n < d, then \bar{a}_n = 1 \\
&if y_n > d, then \bar{a}_n = 0 \\
&if y_n = 0, randomly guess \bar{a}_n
\end{aligned}
$$

## 3. Implementation of Raised Cosine Filter

In communication systems, data is transmitted as binary bits (ones and zeros). It is easier to implement a binary system using switches, where turning on a switch represents '1' and turning it off represents '0'. Such simple binary systems essentially represent ones and zeros as rectangular pulses of finite duration (say τ seconds).

A rectangular pulse of finite duration τ manifests itself as an infinitely extending sinc pulse in the frequency domain (see figure below). This implies that a rectangular pulse requires infinite bandwidth if it is not to be distorted during its transmission. Due to the dual nature of Fourier Transform, the following figure is valid if we alternate the frequency and time domain representations. That is, a symbol with finite bandwidth will extend infinitely in time. This implies that to send a single '1' or '0' or a series of them (for a multi-level signaling), you would need infinite time duration. This is absolutely impractical. In practical terms, signals will not extend infinitely forward and backward in time. But it will definitely be non-zero after the time duration τ. This implies that the residues of adjacent symbols/signals overlap with each other giving rise to Inter Symbol Interference (ISI). If the residual energy from the adjacent symbol is very strong, it becomes impossible to distinguish the present symbol and there is a possibility of it being misinterpreted altogether. To avoid or reduce this effect, "Pulse Shaping" techniques are used to make sure that the data carried by the symbols are not affected by the overlapping effect of adjacent symbols.

In a band-limited system, when we try to increase the data rate, it may lead to Inter Symbol Interference (ISI). There are two criteria that must be satisfied for a non-interference system when pulse shaping is employed.

(1) The pulse shape exhibits a zero crossing at the sampling point of all pulse intervals

(2) The shape of the pulses is such that the amplitude decays rapidly outside of the pulse interval.

A rectangular pulse satisfies the first criterion (where it contains zero crossing – see figure above) but not the second criterion (the energy of the rectangular pulse does not decay rapidly outside the pulse interval and in fact it extends to infinite bandwidth). Pulse shapes filters like raised cosine filters, square root raised cosine filters and matched filters are employed to shape the transmitted pulses so that they will satisfy the above two criteria of providing an ISI free system.

*Raised Cosine Filters/Pulses:*

A Raised Cosine looks more like a modified sinc pulse in time domain and is given by the following function (This equation is apt for digital domain and Matlab simulation, it is obtained from its analog form by substituting "t" by n*TS)

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$$p(t)=\begin{cases}\frac{\pi}{4}\sin c\left(\frac{nT_s}{\tau}\right), if\left(1-\left[\frac{2\alpha nT_s}{T}\right]^2\right)=0\\ \frac{\cos\left(\frac{\alpha\pi nT_s}{\tau}\right)}{1-\left(\frac{2\alpha\pi nT_s}{\tau}\right)^2}, ifn=0\\ \frac{\sin c\left(\frac{nT_s}{\tau}\right)-\cos\left(\frac{\alpha\pi nT_s}{\tau}\right)}{1-\left(\frac{2\alpha\pi nT_s}{\tau}\right)^2}, otherwise\end{cases}$$

Here TS is the sampling period, n is the sample number, α is a parameter that governs the bandwidth occupied by the pulse and the rate at which the tails of the pulse decay. A value of α = 0 offers the narrowest bandwidth, but the slowest rate of decay in the time domain. When α = 1, the bandwidth is 1/τ, but the time domain tails decay rapidly. For more details on investigation of raised cosine characteristics and its implementation.

## 4. Results and Discussion

Figure 4 shows the Impulse Response of Duobinary Encoder. Figure 5 illustrates the Response of Duobinary Filter at Transmitter Side.

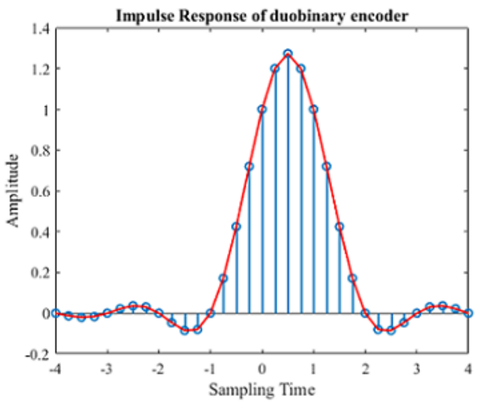


***Figure 4.*** *Impulse Response of Duobinary Encoder.*

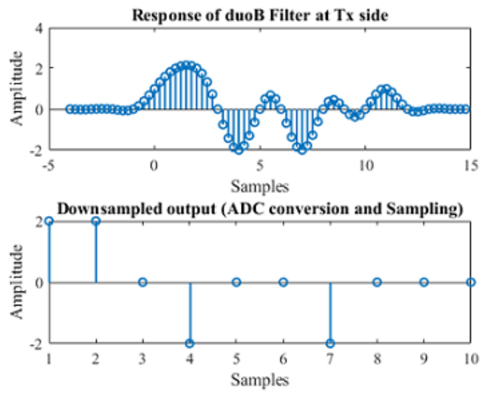


***Figure 5.*** *Response of Duobinary Filter at Transmitter Side.*

Figure 6 demonstraes the Impluse Response of Modified Duobinary Encoder. Figure 7 highlights the Response of Modified Duobinary Filter at Transmitter Side. Figure 8 shows the Impulse Response for α=0.5. Figure 9 mentions the Input Data and Response of Raised Cosine Filter for α=0.5. Figure 10 illustrates the Frequency Domain Response of Raised Cosine Filter for α=0.5. Figure 11 demonstrates the Impulse Response for α=1.

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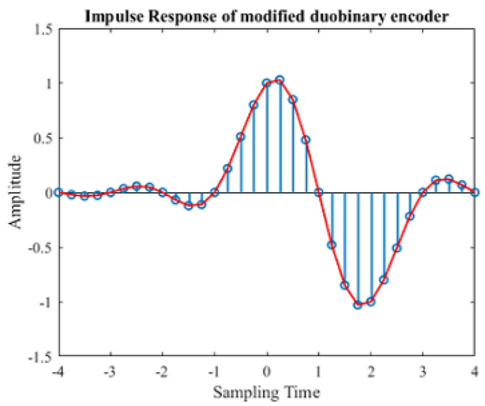


***Figure 6.*** *Impluse Response of Modified Duobinary Encoder.*

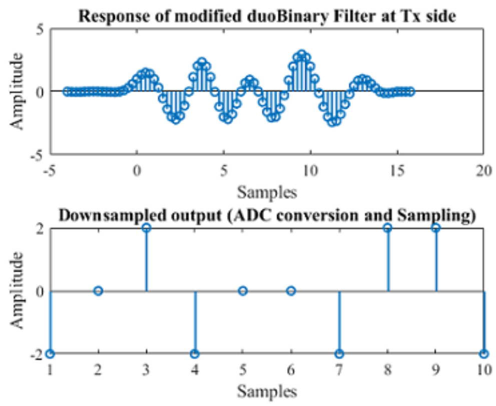


***Figure 7.*** *Response of Modified Duobinary Filter at Transmitter Side.*

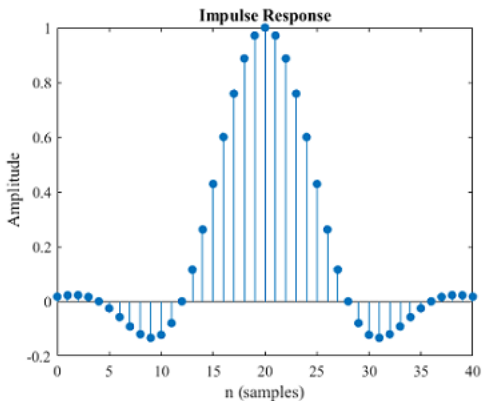


***Figure 8.*** *Impulse Response for α=0.5.*

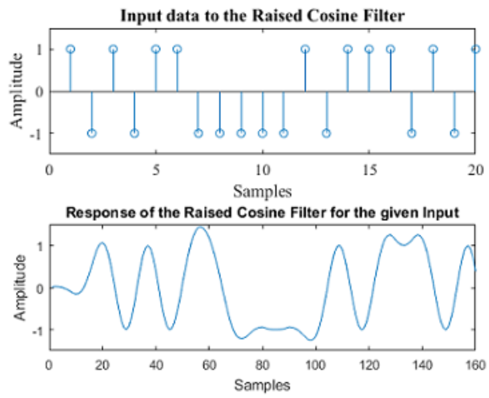


***Figure 9.*** *Input Data and Response of Raised Cosine Filter for α=0.5.*

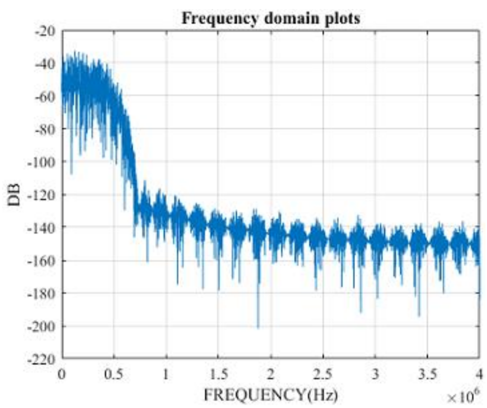


***Figure 10.*** *Frequency Domain Response of Raised Cosine Filter for α=0.5.*

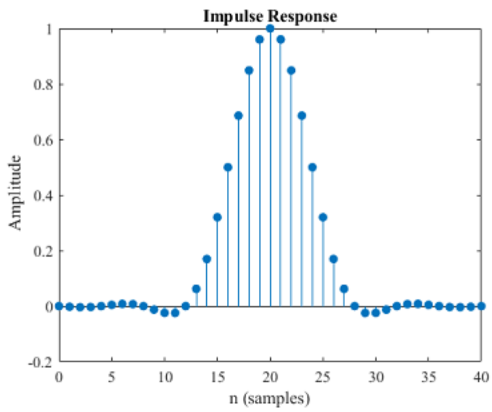


***Figure 11.*** *Impulse Response for α=1.*

Figure 12 gives the Input Data and Response of Raised Cosine Filter for α=1. Figure 13 presents the Frequency Domain Response of Raised Cosine Filter for α=1. The RC filtered outputs for α = 1 and α =0.5 are plotted next. As we can see that as α increases (from 0 to 1), the pulse duration decreases (in frequency domain it will consume excess bandwidth). We must strike an optimum balance between the value of α and the bandwidth it occupies, for the pulse to travel undistorted in a given band-limited channel.

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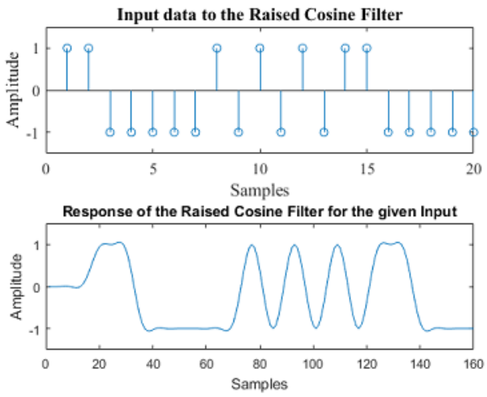


***Figure 12.*** *Input Data and Response of Raised Cosine Filter for α=1.*

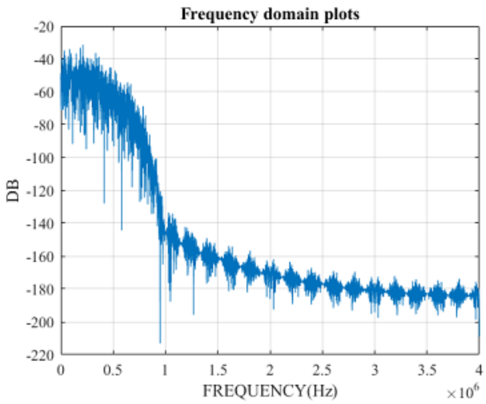


***Figure 13.*** *Frequency Domain Response of Raised Cosine Filter for α=1.*

The frequency response for the Raised Cosine Filtered output for α =1 and α =0.5 are plotted below for comparison. The frequency response shows minimized side lobes for α =1, whereas for the case α =0.5 the side lobes' energy is higher when compared to α =1. Also, for α =1, the eye opening is clearer when compared to α =0.5. This implies that as we increase the value of α from 0 to 1, the ISI decreases whereas the excess bandwidth of the signal increases.

## 5. Conclusions

The first portion of this work has shown the mathematical expression of the ultimate concepts of the digital communication system based on filtering of the unwanted signal from the inputs signal. The mathematical model of IIS filtering approaches has designed by using the signal processing theory. The Frequency Domain Responses of Raised Cosine Filter for α=0.5 and 1 have proved the proposed filtering design based on the input signals. An optimum balance between the value of α and the bandwidth of the band-limited channel is necessary for ISI free reliable communication between the transmitter and receiver.

## Conflicts of Interest

The authors declare that there is no conflict of interest regarding the publication of this article.

## Author Contributions

This study is mainly focused on the Efficient Inter Symbol Interference and Filtering Model for Digital Telecommunication System Using MATLAB. There have been several literature background ideas for implementing the mathematical modeling of the filtering techniques for experimental studies. This work emphasizes on the numerical analysis to develop the applied filtering techniques such as Inter Symbol Interference and Filtering for communication system related subjects. This work could be provided the fundamental concepts with the help of numerical analysis especially computer supported applied design for implementation of mathematical expressions for undergraduate students.

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