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Analysis of Multi-rate filters in Communication system by using interpolation and decimation, filters

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Abstract

This paper introduces the implementation of communication system using the method of interpolation and decimation and also by the help of filters. BER is calculated for the study of communication system. BER is considered as the key parameter which is used in assessing systems that transmit the digital data from one place to another. In this paper at the transmitter side the signal is generated and sent through a communication channel that is having noise in it and at the receiver side same signal is recovered back to get the signal. This signal that is received at the receiver side is having noise indulge in it as it passes through the channel that adds up noise in it.

Keyword: BER, SNR, BPSK

I. INTRODUCTION

In communication system various blocks are used to transmit the information. The transmitted signal has to be passed through the source coding, channel coding, modulator block and then the signal is passed by the channel where the noise get added in signal after this at the receiver end source encoder, channel encoder, demodulator is used these are used to recover back the information signal that has been transmitted by the transmitter side. In communication system the only requirement is the secure communication that means the data - send through the transmitter will transmit through security at the receiver side. Here, the main emphasis is given on the communication system that we are going to choose.

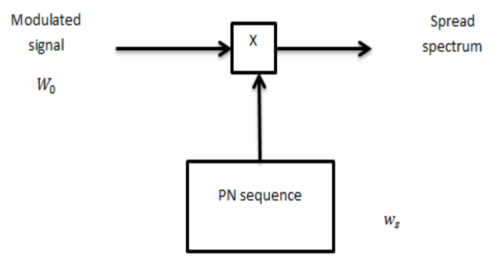
The AWGN channel is added into the system. At the receiver side the filter is used to remove the effect of the AWGN noise. This filter will filter out the undesirable frequencies and provide only those frequencies that are needed in the communication system. After that the calculations of BER are performed as our main goal is to decrease the BER graph. BER can be stated as the number of errors to the total number of bits sent. BER will be less if the medium between the transmitter and receiver is good and the value of SNR is high and the BER will be small and its effect can be neglected. If noise is detected in the system then BER need to be taken into consideration. Errors get introduced into the system when the data is transmitted over a data link that is channel. BER accesses the end to end performance of a system including the transmitter, receiver and the medium between the two. In this way, bit error rate, BER enables the actual performance of a system in operation to be tested, in spite of testing the component parts and assume that they will operate acceptably when in place. The main reasons for the degradation of a data channel and the corresponding bit error rate, BER is noise and changes to the propagation path. BPSK system is used here that generates 1 and -1 by taking 180 degrees phase shift. The pseudo random binary sequence is used in the generation of BPSK.

Sampling means that we can take the series of samples of a continuous varying signal and utilize these values of samples to describe the whole signal without any loss of the available information (data,signal). These samples can be used to reconstruct the original signal.

The pseudo random binary producer is a generator that generates the unique binary sequence for each transmitted signal and this code is different for each transmitted signal. This code is generated using the ex-or operation. The first bit and the last bit is ex-ored and then a sequence is generated.

A DSSS generator requires:

- A modulated signal in the RF band
- A PN sequence to spread it



Central objective of this study is to make the communication more secure and reliable as the demand for communication increasing day by day so in order to provide the users required bandwidth some steps should be taken. As the filters are considered to be an array of band pass filters which separate data i.e., input signal into the multiple components and these multiple components carry a single frequency sub band of original signal.

II. SYSTEM MODEL

BPSK system is taken where the inputs are given at the transmitter side would be 1 or -1. The key perseverance of this report is to make the communication systemmore efficient and secure. To achieve this multi-rate signals are used because they provide an efficient way for communication by reducing cost. Multi-rate signals can also be used in communication system to get the better sampling rate and these having advantages in DSP also, so it is good to use this technique in the communication.

Interpolation is done at the transmitter side to send the data or information signal. So as to make the system secure interpolation is done. Interpolation can also be termed as the information preserving process where all the x[n] samples are exist in the expanded signal y[n]. The expansion process is done along with a unique digital low pass filter which is termed as an anti-aliasing filter. Aliasing is not caused by the process of expansion in the interpolated signal. But this process yields undesirable replicas in the frequency spectrum of the signal.

Let x(n) be original input signal and v(n) is sequence having L-1 zeros introduced, y(n) is output sequence of low-pass filter and let h(0), ..., h(k-1) be constants of low-pass filter, then

$$y(n) = \sum_{k=0}^{K} h(k)v(n-k)$$

However, v(n-k) = 0 except n-k is a multiple of L, interpolation factor. Due to this L-1 zeros were added in sequence x(n) to acquire v(n). Let x(n) be input signals, and h(k) be filter constants. Then output signal y(r) can be represented as:

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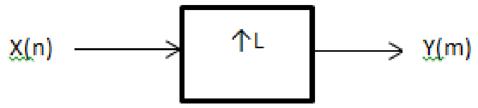
$$y(r) = \sum_{n=0}^{K/L} h(r - Ln)x(n)$$



After this noise is added in the system and inverse process of interpolation is done at the receiver side that is decimation.

Decimation is treated as the discrete time complement of the sampling. In the sampling process our main goal is to convert a continuous time signal x(t) into arrangement of samples x[n]. But in the case of decimation process we begin through the discrete time signal x[n] and change it into an additional discrete time signal y[n] and this discrete time signal contains the subsamples of x[n]. An anti-aliasing filter is used before the decimator or down-sampler to prevent aliasing that can occur due to the lower sampling rate.

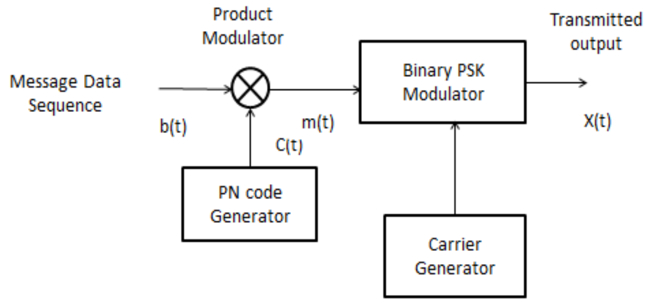
Let x(m) is input signal, h(k), 0 <= k <= K be constants of an assumed low-pass filter and z(m) be output signal before decimating by an amount of M, then:

$$Z(m) = \sum_{k=0}^{K} h(k)x(m - k)$$

Currently, if we assume that output signal afterward decimator is y(r) = z(rM) where sampling rate is decreased with an amount of M. Then, y(r) = z(rM) if e output signal is decimated with an amount of M.

$$y(r) = \sum_{k=0}^{K} h(k)x(rM - k)$$

After this the BER of the system is calculated on the basis of the received signal or data.



Transmitter

III. OTHER PROPOSED MODEL

Instead of using the interpolation and decimation at the transmitter and receiver side other model is also proposed with the use of filters. And comparison is done on the basis of received BER graph.

Filters play an important role in this process. Chebyshev and Butterworth filters are used here to get the BER graph.

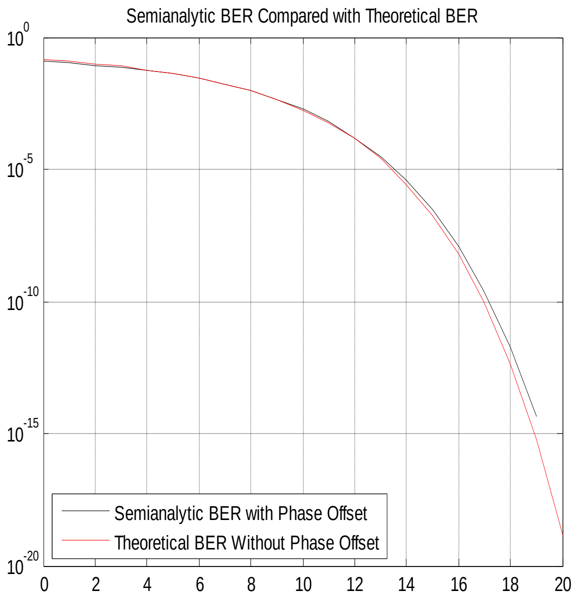
Chebyshev filters reduce the amplitude of signal below or above a particular frequency, provide a steeper roll-off, magnitude response of chebyshev filter show ripples. They provide better attenuation beyond pass band. On the other hand Butterworth filters are used in high quality audio applications due to their flat response in pass band and stop band.To filter the noise from a system low pass filter is used. As noise is high frequency signal but when it is passed by a low pass filter most of the noise is rejected and a clear sound is created. The magnitude response decreases with increase in frequency. They have a wide

transitionband.They has better pulse response.Maximally flat magnitude response in the pass-band. All-around performance is good.

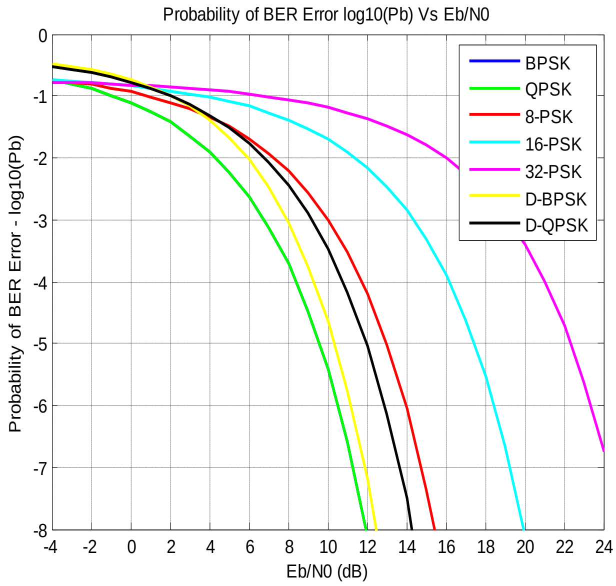
I. RESULTS

The results that we get during the simulation are:

Comparison of theoratical BER with semianalytic BER:-
This graph represents the BPSK modulations Bit error rate in this coding is done to get the required result. The code perfroms following operations that includes the generation of random BPSK modulated symbols that is +1 and -1. After that these symbols are passed through the AWGN channel and then demodulation of the received symbols is done on the basis of the location in the constellation and then number of errors are counted and to get the multiple values of Eb/No this process is repeated again and again.

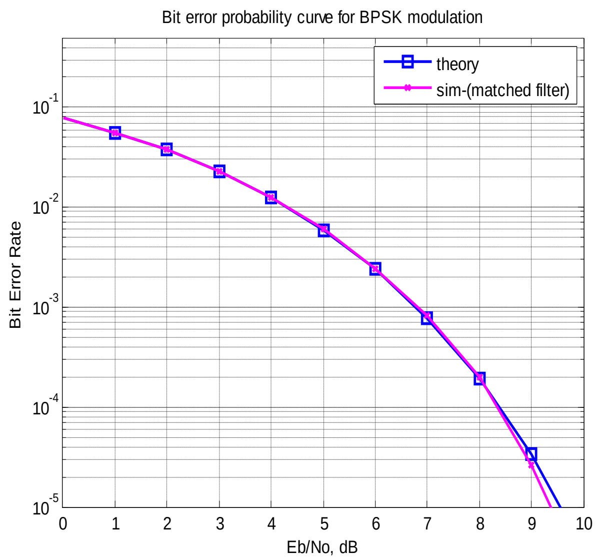


Graph showing comparison of various modulation techniques:-



In this graph the comparison of various modulation techniques is done which results in the BER graph. The BER graph reduces according to the modulation technique that has been used.

Graph for bit error probability curve for BPSK modulation:-



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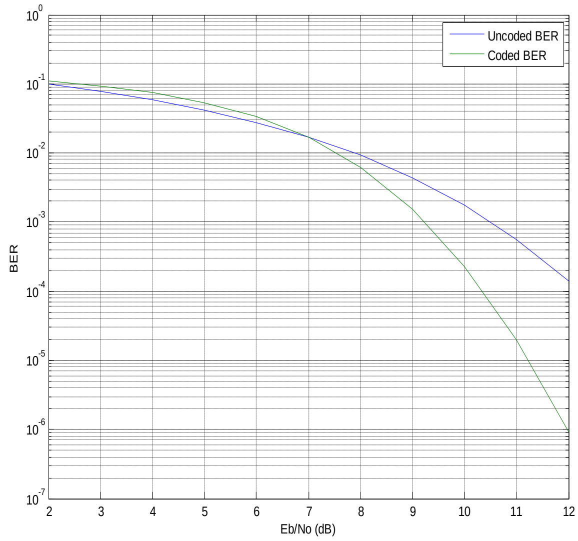
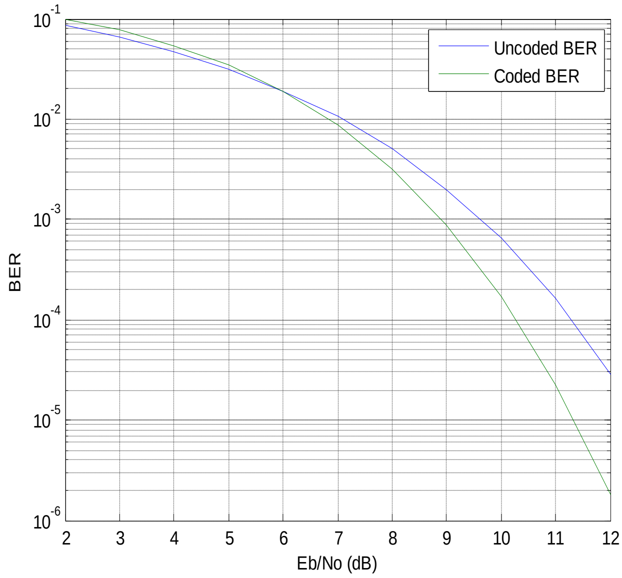
Graph of BER by using the Cehbyshev Filter:
This graph shows that by using the filter according to the noise added we can decrease the BER of the system. Here Cehbyshev Filter is used at the receiver side which due to which errors introduced in the system and when these errors are calculated then the corresponding BER graph can be seen. Which shows the sharp transitions and the corresponding BER reduces.



Graph of BER by using the Butterworth Filter:

This graph shows that by using the filter according to the noise added BER of the system can be minimized. Here butterworth filter is used at the receiver side which due to which errors introduced in the system and when these errors are calculated then the corresponding BER graph can be seen. Which shows the sharp transitions and the corresponding BER reduces. The BER of the system decreases when we use the Butterworth Filter as in the graph the transitions can be seen.



IV. CONCLUSION

This study contributes in the part of multi-rate signal processing and also describes the convergence in between multi-rate signal processing techniques and multicarrier communication systems. The system performance will noticeably degrade if the BER rises too high. If it is within restrictions then the system will work adequately. The BER can be controlled by some factors, the interference level existing in a system is generally set by exterior features and these can't be changed by the system design.By dropping bandwidth level of interference can be decreased. By the use of interpolation and decimation BER can be reduced but filters also proved to be helpful in decreasing the BER.

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