Design and Simulation of Wave Shaping Schemes for a Virtual Data Communication and Impaired Link Environment System for Advanced ICT Education

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Abstract - **Design and Simulation of Waves Shaping Schemes for a Virtual Data Communication and Impaired Link Environment System for Advanced ICT Education is aimed at providing a simulator for the performance of digital filtering of signals for data communication experiments with the aid of MATLAB. A fundamental aspect of signal processing is filtering. Filtering involves the manipulation of the spectrum of a signal by passing or blocking certain portions of the spectrum, depending on the frequency of those portions. This work is designed to provide a flexible platform for teaching the operations of waves shaping schemes (lowpass, bandpass and highpass) modelled over an additive white Gaussian noise (AWGN) channel.**

Keywords **– Lowpass, Bandpass, Highpass, MATLAB.**

I. INTRODUCTION

The entire Simulation and Implementation bothers on creating a software teaching tool for simulating digital communication systems. This is done to study the viability of the communication system, besides, the use of this virtual laboratory will enable Data Communication students view and understand the interplay between the different communication systems.

The idea of simulating physical systems before they are built is not new as it is a common practice in systems engineering. For example, a satellite system was simulated before it was physically built. To summarise, the purpose of a simulation laboratory is not only to test the feasibility of the different parts of the hardware implementation but also to establish the maximum performance of the system. The master simulator design which is required to test the performance of the wave shaping schemes implemented in this project follows processes which include signal generation, Channel Coding and Decoding, Frequency - Time Domain Representation of Signals, Digital Modulation and Demodulation, Communication Channel Impairments. Below is a block diagram of the master simulator.
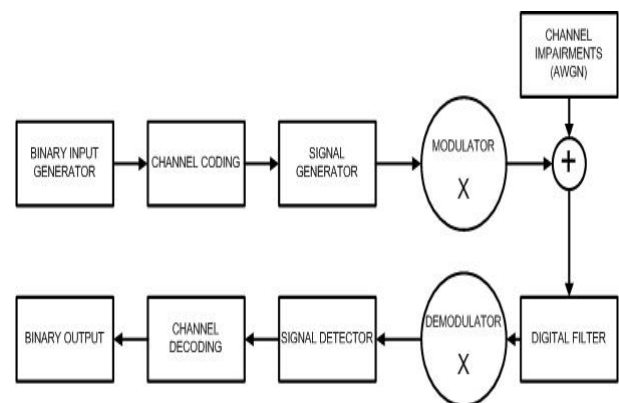

Fig 1.1 Block Diagram of the Master Simulator

Source: Computer Simulation of Digital Modems for Satellite Mobile Communications (PhD Thesis)[1]

II. DESIGN OF THE MASTER SIMULATOR

A. SIGNAL GENERATION

A signal is a function that conveys information about the behaviour or attributes of some phenomenon[2]. In the physical world, any quantity exhibiting variation in time or variation in space (such as an image) is potentially a signal that might provide information on the status of a physical system, or convey a message between observers, among other possibilities[3].

The signal to be first generated is based on the simulator user's input of a 4- bits data into the system. The program written to accept this input must be able to verify the validity of the input string and when not valid the program displays an error message requiring the user another chance to

enter a valid input. The number of trials the user is also limited to ensure system efficiency.

B. CHANNEL CODING AND DECODING

Channel coding schemes are specially designed to improve error performance of signals through communication channels. Channel coding schemes can be grouped into block codes in which specific numbers of bits are added to message bits for error detection and correction.

The Channel coding scheme used in the master simulator is the hamming encoder and decoder. this choice is based on the fact that the system is a just a test system. The encoding of the 4-bits input is computed by multiplying the input by the 7 columns of the hamming encoder G matrix as specified and stored in the program. the output of the encoder is a 7 message bits array and the addition of a check bit making 8-bits which serve as the baseband signal to be modulated before transmission over the communication channel.

C. DIGITAL MODULATION AND DEMODULATION

We modulate to increase the frequency of a signal to enable multiplexing with other signals for efficient transmission. To transmit a signal over a communication channel, a pure carrier is generated at the transmitter, the carrier is modulated with the information to be transmitted. Any reliably detectable change in signal characteristics can carry information. At the receiver the signal modifications or changes are demodulated and detected[4].

Modulation can be computed by either time domain multiplication of the baseband signal and a carrier signal or their frequency domain convolution. This project employs time domain multiplication of the signals to generate the modulated signal. Therefore the program is written to generate the a carrier signal which is sinusoidal in this case. Both the baseband and carrier signal are sampled at equal rates and then multiplied to generate the desired modulated signal.

D. FREQUENCY-TIME DOMAIN REPRESENTATION OF SIGNALS

As signals moves through the communication channel, it is important that one can view it's behaviour both in time and frequency domains. the

time domain representation of the signal is produced by a simulated oscilloscope and the frequency domain representation is computed by a Fast Fourier Transformation (FFT) of the time domain representation of the signal. The FFT computation is made easy by the used of MATLAB inbuilt function that performs the computation.

E. COMMUNICATION CHANNEL IMPAIRMENTS

With any communications system, it must be recognized that the output signal will differ from the transmitted signal due to various channel transmission impairments. For digital signals, bit errors are introduced: A binary 1 is transformed into a binary 0 and vice versa. For the master simulator, Additive White Gaussian Noise (AWGN) is added to the modulated signal to simulate the communication channel impairment. A program is written in MATLAB to generate the AWGN that is then added to the modulated signal. The process is a linear addition of wideband or white noise with a constant spectral density and a Gaussian distribution of amplitude to an input signal .

III. DESIGN OF WAVES SHAPING SCHEMES (FIR FILTERS)

A filter is a frequency selective device that removes unwanted information from the original message signal. Unwanted signals can be noise or other undesired information. Digital filters are more versatile when compared to the analog filters in their characteristics such as programming flexibility, ability to handle both low as well as high frequency signals accurately.

A finite impulse response filter of length M with input x(n) and output y(n) is described by the difference equation

$$y(n) = b_0 x(n) + b_1 x(n-1) + \cdots + b_{M-1} x(n-M+1)$$

$$= \sum_{k=0}^{M-1} b_k x(n-k)$$

where bk is the set of filter coefficients. The transfer function of this filter in z domain can be represented as

$$H(z) = \sum_{k=0}^{M-1} h(k)\, z^{-k}$$

The desired frequency response specification for linear phase filter is the Fourier transform of the

desired impulse response, and this can be represented as

$$H_d(\omega) = \sum_{n=0}^{\infty} h_d(n) \ e^{-j\omega n}$$

Truncation is performed by multiplying desired sample response with a window function in time domain which gives sample response of filter represented as

$$h(n) = h_d(n)w(n)$$

where w(n) is the window function. The window function used in this project is the hamming window which is defined by

$$w(n) = 0.54 - 0.46 \ \cos\frac{2\pi n}{M-1}$$

filtering of signals can be computed either by the convolution of the time domain signal and the filter coefficients or the multiplication of the frequency domain characteristics of the signal and filter coefficients. This project employs the multiplication process. The process involves computing the desired filter coefficients, computing the FFT of the filter coefficients to get the frequency response of the filter, computing the FFT of the signal , multiplying this frequency responses and finally computing the inverse fourier transform (IFFT) of the multiplied responses to get the desired filtered signal.

A. DESIGN OF FIR LOWPASS FILTER

lowpass filters pass low-frequency signals but attenuates (reduces the amplitude of) signals with frequencies higher than the cut-off frequency. The actual amount of attenuation for each frequency depends on filter design methodology. The MATLAB function fir1 is used to design and generate filter coefficients which is then used to filter the input signal in frequency domain. The filter designed in this project is a 63rd order filter due to the large number of sample of the input signal. A low cut- off frequency is however varied based on application
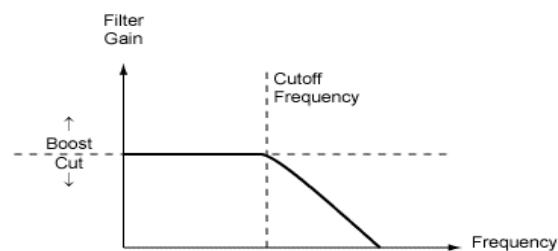

Fig 3.7 Lowpass Filter Diagram.
Source: audiomulch.com

B. DESIGN OF FIR BANDPASS FILTER

Bandpass filters pass frequencies within a certain range and attenuates frequencies outside that range. The MATLAB function fir1 is also used to design and generate filter coefficients which is then used to filter the input signal in frequency domain. The filter designed in this project is a 63rd order filter due to the large number of sample of the input signal. Two cut- off frequencies are   however specified for bandpass filters based on application to only pass this specified frequencies.
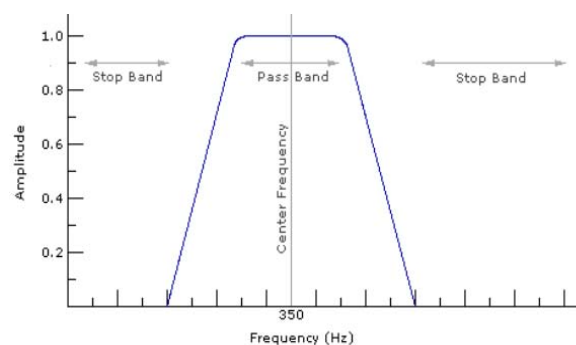

Fig 3.8 Bandpass Filter Diagram.
Source: beausievers.com

C. DESIGN OF FIR HIGHPASS FILTER

Highpass filters pass high-frequency signals but attenuates signals with frequencies lower than the cut-off frequency. They offer easy passage of a high-frequency signal and difficult passage to a low-frequency signal. The MATLAB function fir1 is used to design and generate filter coefficients which is then used to filter the input signal in frequency domain. The filter designed in this project is a 63rd order filter due to the large number of sample of the input signal. A high cut- off frequency is however varied based on application.
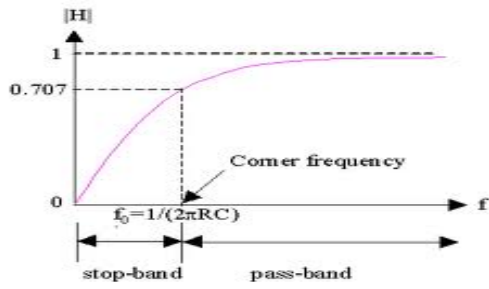
Fig 3.9 Highpass Filter Diagram
Source: cs.wright.edu

IV. DIGITAL FILTERING TESTING RESULTS

A graphical user interface (GUI) has been created for the simulator users to interact and perform experiments. A diagram of this GUI is shown in the figure below
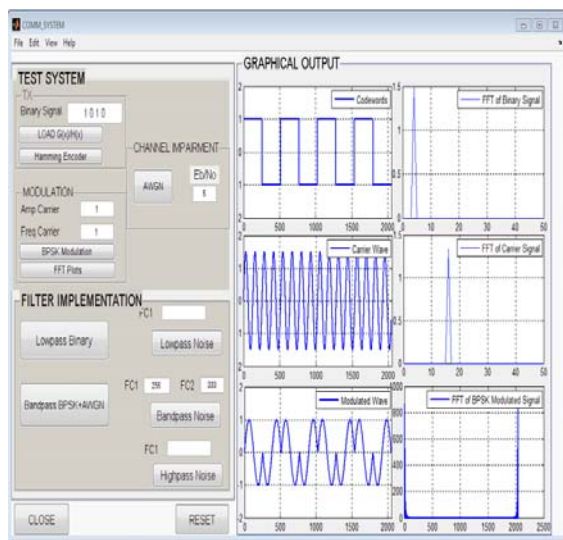

Fig 4.1 Master Simulator GUI Diagram

The generated 8-bits baseband signal, the carrier signal and the modulated signal and their spectral representations:
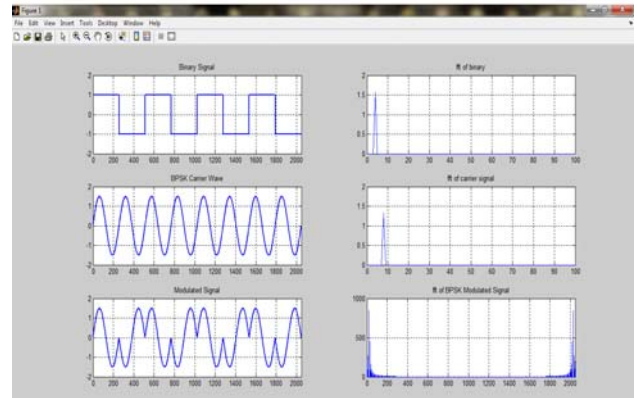

fig 4.2: 8-bits baseband signal, the carrier signal and the modulated signal and their spectrum representations.

Channel impairment (Additive White Gaussian Noise) Representations at different noise levels:
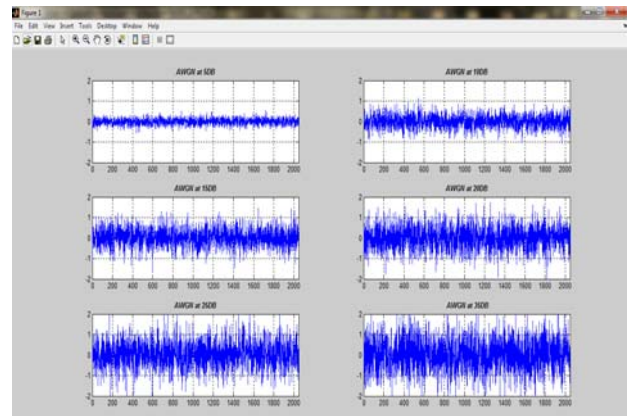

Fig 4.3 Channel impairment (Additive White Gaussian Noise) Representations

Bandpass filtered noisy BPSK modulated signal response:
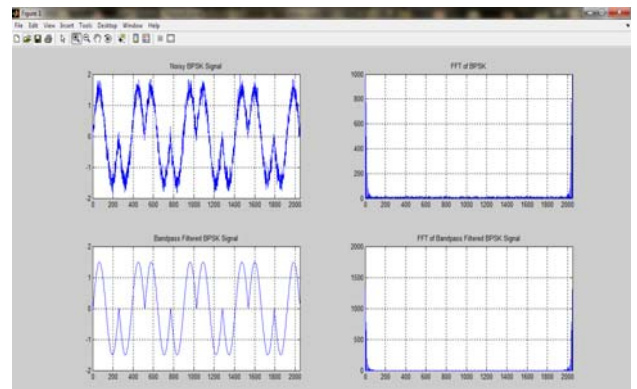

Fig 4.4 Bandpass Filtered Noisy BPSK Signal Representation

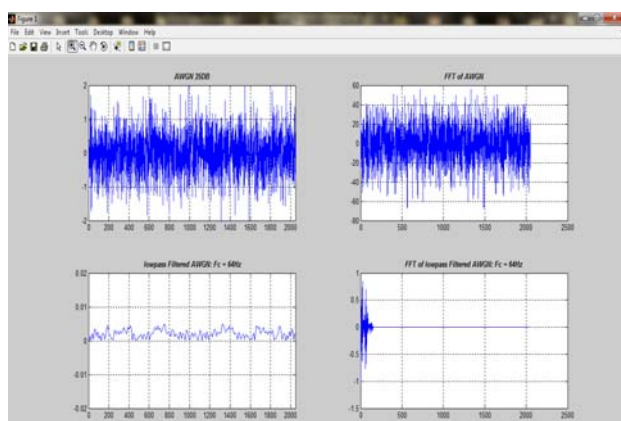Lowpass filtered noise response:

Fig 4.5 Lowpass Filtered Noise Representation

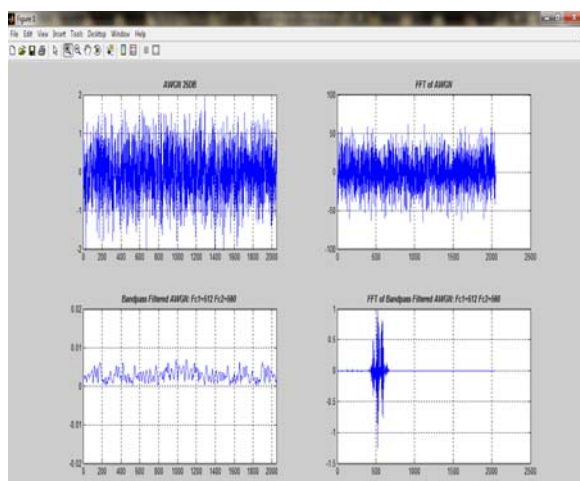Bandpass filtered noise response at two selected cut-off frequencies:

Fig 4.6 Bandpass Filtered Noise Representation
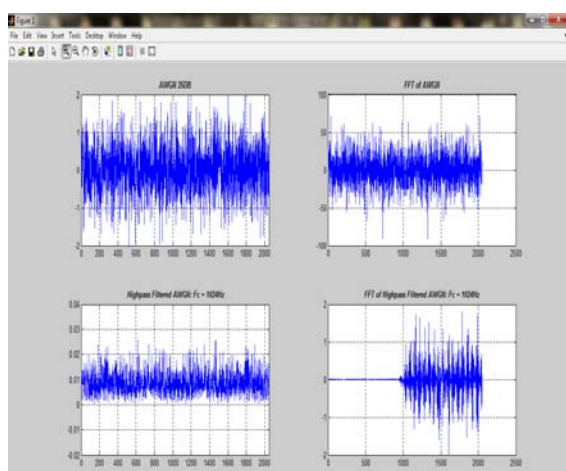
Highpass filtered noise representation:

Fig 4.7 Highpass Filtered Noise Representation

V.   CONCLUSION

The project design has taken into consideration the fact that digital signal processing is a field requiring a high level of precision and system accuracy if correct and acceptable results must be obtained. Utmost adherence to system specification and requirements have to be taken in the selection of software libraries, algorithms and functions used; those chosen must be thoroughly tested and peer-reviewed. This paper presents a software implementation of digital filters using MATLAB GUI in a user friendly environment. The developed software is useful for students in performing digital filters experiments. The evaluated performance using the specific theoretical filter parameters match the performance with the software. The software is working properly towards accurate results upon testing.

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