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An Extensive Study on Noise free Signal (Random Packets) on All Performance Parameters

Harish Chauhan[1]* Prof. Yash Pal Singh[2]

[1] M. Tech (ECE) SITM, Rewari

[2] Director, SITM

Abstract – Now a days ICT has become the backbone of all communication system. Its ranges from education to business, research to industry, health care to automation. The wide coverage and the reliable bandwidth[29] offered by the broadband satellite systems has made them a promising media for IP streaming. But the demand for exploring new solutions concerning this industry has kept increasing to maintain the growth rate require by the various market sectors.

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1. INTRODUCTION

In order to minimize the design time as well as implementation costs, the initial steps in researches concerning satellite systems are often taken in simulation environments. In contrast with the actual satellite system test-beds which are expensive or sometimes not available to the academic research community, the simulation software packages like Math works MATLAB[7][40] are widely used at the early stages of modeling and design process. However, such packages are mostly designed as general purpose tools and therefore, the built-in models provided by these tools are often incomplete or too simplistion.

2. PROBLEM DEFINITIOIN

The objective of this research is to obtain

- noise free signal (Random Packets)
- the performance parameters as
- Bit Error Rate(BER)
- Signal to noise ratio(SNR)
- No. of parity checks
- The FFT spectrum analysis
- Constellation Diagram

3. LITERATURE REVIEW

Irfan Ali [7] introduced the Bit error rate, (BER) simulation using Mat lab. Bit error rate, (BER) is a key parameter that is used in assessing systems that transmit digital data from one location to another. Systems for which bit error rate, is applicable include radio data links as well as fiber optic data systems, Ethernet, or any system that transmits data over a network of some form where noise, interference, and phase jitter may cause quality degradation of the digital signal. Mat lab is an ideal tool for simulating digital communication[29]s systems.

Atul Gautam, Kumar Saurabh, Manish Sharma[4] in their paper proposed that BER performance of communication system with using different modulation techniques i.e. BPSK, QPSK and GMSK in Rayleigh Fading Channel is enhanced by using LPDC encoder. Low density parity check (LDPC) codes are one of the best error correcting codes in today‘s coding world

Gunther M. A. Sessler, Ricard Abello´, Nick James, Roberto Madde, and Enrico Vassallo[17] Th discussed a new implementation of a Gaussian minimum-shift keying (GMSK) modulator and demodulator on the European Space Agency (ESA)‘s common deep-space receiver the Intermediate Frequency Modem System (IFMS) , which is a software radio based platform. The GMSK demodulator is needed for ESA‘s deepspace and near-Earth missions, starting with the Herschel–Planck satellites in 2008. The implementation requiremented and hardware restrictions from the IFMS lead to the need for a significant simplification versus the optimum demodulation approach.

Ann Spriet, Geert Rombouts, Marc Moonen and Jan Wouters [16] in their paper, solutions for combined feedback and noise suppression in hearing aids were developed. The techniques presented were based on the generalized sidelobe canceller (GSC) and adaptive feedback canceller (AFC), with a prediction error method (PEM) adaptation to avoid speech distortion.

4. DIGITAL COMMUNICATION TECHNIQUES

4.1 Communication

Communication is the exchange and flow of information and ideas from one person to another. It involves a sender transmitting a idea, information, or feeling to a receiver. Effective communication occurs only it the receiver understand the exact information or idea that the sender intended to transmit.

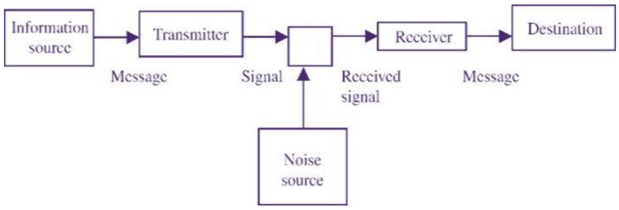

Figure 1: Block Diagram of Communication

Firstly Information is generated from information Source and then pass to the Transmitter .After Transmitter the signal is passed through some channel from where the noise accumulation takes place by Noise source , which could be anything i.e atmospheric noise , thermal noise.

After Channel the signal is reached to the receiver and after modification and removal of noise the signal reaches to the Destination.

4.2 Types of Communication

- Analog communication
- Digital communication[29]

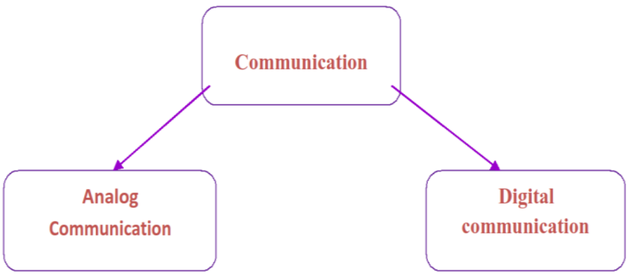

Figure 2 : Types of communication

4.3 Analog Communication

An analog or analogue signal is any continuous signal for which the time varying feature (variable) of the signal is a representation of some other time varying quantity, i.e., analogous to another time varying signal. For example, in an analog audio signal, the instantaneous voltage of the signal varies continuously with the pressure of the sound waves. It differs from a digital signal, in which a continuous quantity is represented by a discrete function which can only take on one of a finite number of values. The term analog signal usually refers to electrical signals; however, mechanical, pneumatic, hydraulic, human speech, and other systems may also convey analog signals.

An analog signal uses some property of the medium to convey the signal's information. For example, an aneroid barometer uses rotary position as the signal to convey pressure information. In an electrical signal, the voltage, current, or frequency of the signal may be varied to represent the information.

Converting an analog signal to digital form introduces a constant low-level noise called quantization noise into the signal which determines the noise floor, but once in digital form the signal can in general be processed or transmitted without introducing additional noise or distortion. Therefore as analog signal processing systems become more complex, they may ultimately degrade signal resolution to such an extent that their performance is surpassed by digital systems. This explains the widespread use of digital signals in preference to analog in modern technology. In analog systems, it is difficult to detect when such degradation occurs. However, in digital systems, degradation can not only be detected but corrected as well.

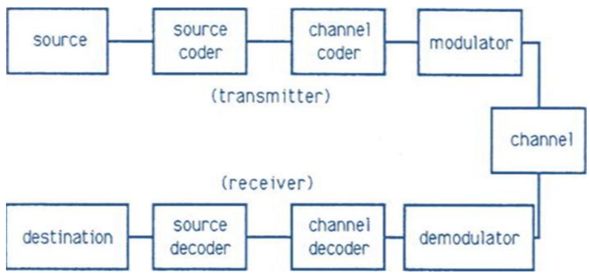

Figure 3: Block Diagram of Analog Communication

- **Anaog to Digital Converter**

Analog to Digital Converter is used to convert Analog Signal to Digital Signal

- **Source Encoder**

The signal produced by source is converted into digital signal consists of 1's and 0's. For this we need source encoder. We should like to use as few binary digits as possible to represent the signal. In such a way this efficient representation of the source output results in little or no redundancy. This sequence of binary digits is called information sequence. Source Encoding or Data Compression The process of efficiently converting the output of analog or digital source into a sequence of binary digits is known as source encoding.

- **Building blocks of Digital communication System**

The source of information can be analog or digital, e.g. analog: audio or video signal, digital: like teletype signal.

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- **Digital Modulator**

The binary sequence is passed to digital modulator which in turns convert the sequence into electric signals so that we can transmit them on channel. The digital modulator maps the binary sequences into signal wave forms, for example if we represent 1 by sin x and 0 by cos x then we will transmit sin x for 1 and cos x for 0.

- **The Channel**

The communication channel is the physical medium that is used for transmitting signals from transmitter to receiver. The modulation and coding used in a digital communication[29] system depend on the characteristics of the channel. The two main characteristics of the channel are BANDWIDTH[29]

- Telephone Channel
- Coaxial Channel
- Optical Fibers
- Microwave, Radio and Satellite[6] Channel

- **The Digital Demodulator**

The digital demodulator processes the channel corrupted transmitted waveform and reduces the waveform to the sequence of numbers that represents estimates of the transmitted data symbols.

- **Channel Decoder**

This sequence of numbers then passed through the channel decoder which attempts to reconstruct the original information sequence from the knowledge of the code used by the channel encoder and the redundancy contained in the received data. The average probability of a bit error at the output of the decoder is a measure of the performance of the demodulator – decoder combination.

- **Source Decoder**

Source decoder tries to decode the sequence from the knowledge of the encoding algorithm. And which results in the approximate replica of the input at the transmitter end.

- **Digital to Analog Converter**

Analog to Digital Converter is used to convert Digital Signal to Analog Signal.

4.4 List of common digital modulation techniques

The most common digital modulation techniques are:

- Phase-shift keying (PSK):
- Binary PSK (BPSK), using M=2 symbols
- Quadrature PSK (QPSK), using M=4 symbols
- 8PSK, using M=8 symbols
- 16PSK, using M=16 symbols
- Differential PSK (DPSK)
- Differential QPSK (DQPSK)
- Offset QPSK (OQPSK)
- π/4–QPSK
- Frequency-shift keying (FSK):
- Audio frequency-shift keying (AFSK)

- Multi-frequency shift keying (M-ary FSK or MFSK)
- Dual-tone multi-frequency (DTMF)
- Amplitude-shift keying (ASK)
- On-off keying (OOK), the most common ASK form
- M-ary vestigial sideband modulation, for example 8VSB
- Quadrature amplitude modulation (QAM) - a combination of PSK and ASK:
- Polar modulation like QAM a combination of PSK and ASK.
- Continuous phase modulation (CPM) methods:
- Minimum-shift keying (MSK)
- Gaussian minimum-shift keying (GMSK)
- Continuous-phase frequency-shift keying (CPFSK)
- Orthogonal frequency-division multiplexing (OFDM) modulation:
- Discrete Multitone (DMT) - including adaptive modulation and bit-loading.
- Wavelet modulation
- Trellis Coded Modulation (TCM), also known as trellis modulation
- Spread-spectrum techniques:
- Direct-sequence spread spectrum (DSSS)
- Chirp spread spectrum (CSS) according to IEEE 802.15.4a CSS uses pseudo-stochastic coding
- Frequency-hopping spread spectrum (FHSS) applies a special scheme for channel release

In Simulink Model we have make use of QPSK and GMSK modulator which will be discussed later now we are going to discussed the advantages and Disadvantages of Analog and Digital communication[29].

- **Bit error rate**

In digital transmission, the number of bit errors is the number of received bits of data stream over a communication channel that have been altered due to noise, interference, distortion or bit synchronization errors.

The bit error rate (BER) is the number of bit errors per unit time. The bit error ratio(also BER) is the number of bit error divided by the total number of transfer bits during a studied time interval. BER is a unitless performance measure, often expressed as a percentage.[1]

The bit error probability, p_e is the expectation value of the bit error ratio. The bit error ratio can be considered as an approximate estimate of the bit error probability. This estimate is accurate for a long time interval and a high number of bit errors.

- **Factors effecting the Bit Error Rate**

In a communication system, the receiver side BER may be affected by transmission channel noise, interference, distortion, bit synchronization problems, attenuation, wireless multipath fading etc.

The BER may be improved by choosing a strong signal strength (unless this causes cross-talk and more bit errors), by choosing a slow and robust modulation scheme or line coding scheme, and by applying channel coding schemes such as redundant forward error correction codes.

The transmission BER is the number of detected bits that are incorrect before error correction, divided by the total number of transferred bits (including redundant error codes). The information BER, approximately equal to the decoding error probability, is the number of decoded bits that remain incorrect after the error correction, divided by the total number of decoded bits (the useful information).

- **Analysis of Bit Error Rate**

The BER may be evaluated using stochastic (Monte Carlo) computer simulations. If a simple transmission channel model and data source model is assumed, the BER may also be calculated analytically. An example of such a data source model is the Bernoulli source.

Examples of simple channel models used in information theory are:

- Binary symmetric channel (used in analysis of decoding error probability in case of non-bursty bit errors on the transmission channel)
- Additive White Gaussian Noise (AWGN) channel without fading.

A worst-case scenario is a completely random channel, where noise totally dominates over the useful signal. This results in a transmission BER of 50% (provided that a Bernoullibinary data.

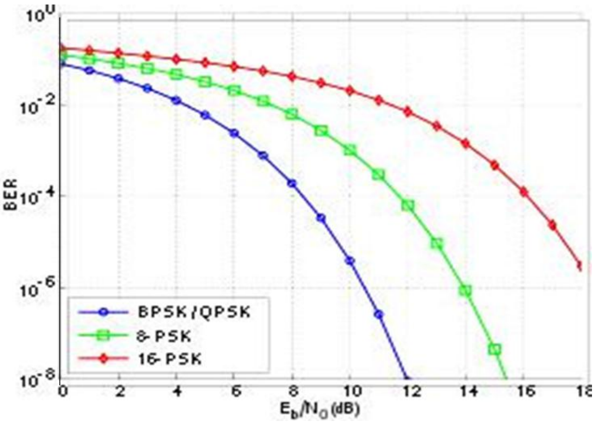

Figure 4. : Bit Error Rate of BPSK, QPSK, 8-PSK, 16-PSK with AWGN channel

In a noisy channel, the BER os often expressed as a function of normalized carrier to noise ratio measure denoted E_b/N_o, (energy per bit to noise power spectral density ratio), or E_s/N_o(energy per modulation symbol to noise spectral density).

For example, in the case of QPSK modulation and AWGN channel, the BER as function of the Eb/N0

People usually plot the BER curves to describe the performance of a digital communication[29] system. In optical communication, BER(dB) vs. Received Power(dBm) is usually used; while in wireless communication, BER(dB) vs. SNR(dB) is used.

Measuring the bit error ratio helps people choose the appropriate forward error correction codes. Since most such codes correct only bit-flips, but not bit-insertions or bit-deletions, the Hamming distance metric is the appropriate way to measure the number of bit errors. Many FEC coders also continuously measure the current BER.

5. SIMULINK MODEL

The wide coverage and the reliable bandwidth[29] offered by the broadband satellite systems has made them a promising media for IP streaming. But the demand for exploring new solutions concerning this industry has kept increasing to maintain the growth rate require by the various market sectors.

In order to minimize the design time as well as implementation costs, the initial steps in researches concerning satellite systems are often taken in simulation environments. In contrast with the actual satellite system test-beds which are expensive or sometimes not available to the academic research community, the simulation software packages like Mathworks MATLAB[7][40] are widely used at the early stages of modeling and design process. However, such packages are mostly designed as general purpose tools and therefore , the built-in models provided by these tools are often incomplete or too simplistic.

This thesis is based on suppression of Noise during communication for Random Packets . As noise is a critical issue during communication from source to the destination .It corrupts the actual signal coming from source. The received signal at the receiver has to be noise free so to remove the noise some strategies has to be use. The noise from the signal can be removed by channel codings. Channel coding is also called Forward Error Correction(FEC).

To accomplish this research we have made use of the DVB(Digital Video Broadcasting) model and its coding. In this model instead of using MPEG-TS ,we have use the Random Numbers Generated By Bernoulli Sequence Generator.

5.1. QPSK WITH LDPC

The block diagram of the Model has been shown in thesis report further. But here we are showing some of the initialized values that are obtained from the coding structure of DVB to start the simulation

5.2.1 Initialization

Es/No dB: 1

Modulation Type: 'QPSK'

Number of Bytes Per Packet: 188

Number of Bits Per Packet: 1504

BCH Codeword Length: 32400

BCH Message Length: 32208

BCH Generator Polynomial: [1x193 double]

BCH Primitive Polynomial: [1 0 0 0 0 0 0 0 0 0 0 0 1 0 1 1 0 1]

Number of Packets Per BBFrame: 21

Number of Information Bits Per Codeword: 31584

Bit Period: 3.1662e-05

LDPC Codeword Length: 64800

LDPC Parity Check Matrix: [32400x64800 logical]

LDPC Number of Iterations: 50

Interleave Order: [64800x1 double]

Constellation: [4x1 double]

Phase Offset: 0.7854

Bits Per Symbol: 2

Number of Symbols Per Codeword: 32400

Noise Variance : 0.7943

The model simulation take place according to these initialized values.

5.2.2 Model

Block Diagram in model in MATLAB[7][40] –Simulink is given below and the function of each block is also given

6. CONCLUSION AND FUTURE SCOPE

This research developed a system that gives the noiseless signal at the receiver. This research worked upon the Forward Error Correction (FEC) or Channel coding called LDPC with the use of the GMSK and QPSK Modulation. This research shows the relations between the SNR,BER,Eb/No, coding Rate to get the noiseless random packets by the use of LDPC with GMSK and QPSK. Most mobile products are designed with Class C power amplifiers, which offer the highest power efficiency, yet because they are nonlinear, require the amplified signal to have a constant envelope. This reduces the desirability of implementing QPSK in this situation. However, QPSK effectively utilizes band width[29]; whereas, GMSK requires more bandwidth[29] to effectively recover the carrier. Furthermore, due to its frequency modulating characteristic, GMSK shows a greater immunity to signal fluctuations. QPSK and GMSK each provide beneficial features, and although neither dominates the other, both contribute to the advancement of wireless telecommunication systems.

As SNR increases modulation techniques BER value reduces. And we find that LPDC encoder performance is good with QPSK. But in case of GMSK we cannot reduce the BER rate to sufficient value. Because in GMSK; the data is much more than BPSK and QPSK. Therefore the value BER does not reduce to very less value.

- These Systems could be used for some Digital communication[29] as in Digital video broadcasting. Transmitter and implementation could be helpful in submarine communication at a higher data rate and low probability of error.
- Also useful in best utilization of bandwidth[29] in military communication in under water submarines.

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Corresponding Author

Harish Chauhan*

M. Tech (ECE) SITM, Rewari