Transmission of Audio Signal from Linear Block Coded AWGN Channel Using Different Modulation Techniques and Wavelet Transform Families

Mekhla Dadhich[1], Aabhas Mathur[2], Dr. Hemant Dhabhai [3]

[1]Scholar, M. Tech., Digital Communication, Aravali Institute of Technical Studies, Udaipur, India

[2]Associate Professor and Head, Department of Electronics & Communication, [3]Director

[23]Aravali Institute of Technical Studies, Udaipur, India

Abstract- Mobile communication has become an important part of our daily lives for voice communication and data sharing and access over the Internet. Mobile communication is an open network, so maintaining the privacy and reliability of data has always been anxiety. The reliability of the data against channel noise can be achieved by various error correction codes. The purpose of the channel coding process is to reduce the effect of some disruptive elements that the information is influenced in the transmission phase as much as possible, that is to ensure that the data is delivered to the receiver with minimum error. In the first part of this work, Linear Block Coding Method is used to transmit the audio data recorded in .wav format for 5 seconds in MATLAB environment with minimum error. Four different types of modulation were used as the modulation method and the best performance was obtained by using Binary Phase Shift Keying (BPSK). The Bit Error Rate (BER) performance curve was drawn and the difference between the original and the received signal was observed. According to the BER performance curve, as the signal to noise ratio increases, less false data is received at the receiver. In the second part of the study, four wavelet transforms at different levels were applied to the audio signal, and the performances of these families were compared.

Keywords- Linear block coding, channel coding, wavelet transform, audio signals.

I. INTRODUCTION

To move information along the channel, the digital information is converted into electrical impulses or waveforms. This process is called channel coding. Channel coding aims to bring a digital signal into conformity with the physical characteristics of the transmission channel and to ensure synchronization in the receiver [1]. The purpose of the channel coding process is to reduce the effect of some disturbing elements that the information has encountered in the transmission phase as much as possible that is to ensure that the data is delivered to the receiver with minimum error. The Reed-Solomon (RS) coding is one of the error correction methods that can be used to obtain the message properly. This form of transmission is commonly known as the Forward Error Correction (FEC) [2]. An error correction code; Consists of techniques and algorithms that include two basic operations which are encoding and decoding. [3]. Recently, several studies have been carried out using different channel coding methods in order to realize information transmission with minimum loss. Miah and Rahman [4] used a wireless communication simulator that included Gray coding, modulation, different channel models Additive White Gaussian Noise (AWGN), smooth fading and frequency selective fading channels), channel estimation, adaptive equalization and demodulation. Subsequently, they tested the effects of different channel models on transmitted data using Quadrature Phase Shift Keying (QPSK) schemes at the receiver. In [5], a system for data transfer over a mobile audio channel based on the M-Phase Shift Keying (M-FSK) modulation with optimized parameters was proposed. Rashed et al. [6] investigated the effects of Cyclic Residual Codes and convolutional codes on the transmission performance of audio signals from AWGN and Rayleigh, Rician faded channels.

II. AUDIO TRANSMISSION STEPS

In this study, the five second length audio signals was passed through the steps of sampling, quantization and A/D conversion, and AWGN noisy RS coded channel.

A. Channel Coding

The main purpose of channel coding is to reduce transmission errors. Therefore, the coding made for this purpose can also be called error control coding. To minimize the error in communication systems, control bits which contain no information and only provide error checking are added to the data directory. Since the added control bits do not contain information, redundancy occurs in the transmitted data. If the recipient is to resolve each bit independently of the others, the recipient will decide on information bit according to the majority. This is also a method for reducing the error to the minimum. In this study, Reed-Solomon coding method is used for channel coding process.

B. Reed-Solomon Codes

In 1960, Irving Reed and Gus Solomon published a paper in the Journal of the Society for Industrial and Applied

Mathematics [12]. These codes have great power and utility, and are today found in many applications from compact disc players to deep-space applications. Reed-Solomon codes are non-binary cyclic codes with symbols made up of m-bit sequences, where m is any positive integer having a value greater than 2. R-S(n, k) codes on m-bit symbols exist for all n and k for which

$$0 < k < n < 2m+2 \qquad .....(1)$$

where k is the number of data symbols being encoded, and n is the total number of code symbols in the encoded block. For the most conventional R-S (n, k) code,

$$(n, k) = (2m-1, 2m-1-2t) \qquad .....(2)$$

where t is the symbol-error correcting capability of the code, and n - k = 2t is the number of parity symbols. An extended R-S code can be made up with n = 2m or n = 2m + 1, but not any further.

Reed-Solomon codes achieve the largest possible code minimum distance for any linear code with the same encoder input and output block lengths. For nonbinary codes, the distance between two codewords is defined (analogous to Hamming distance) as the number of symbols in which the sequences differ. For Reed-Solomon codes, the code minimum distance is given by

$$d_{min} = n-1+k \qquad ......(3)$$

The code is capable of correcting any combination of t or fewer errors, where t can be expressed as

$$t = [dmin - 1/2] = [n - k/2] \qquad ....(4)$$

where $\lfloor x \rfloor$ means the largest integer not to exceed x. Eq. (4) illustrates that for the case of R-S codes, correcting t symbol errors requires no more than 2t parity symbols. Eq. (4) lends itself to the following intuitive reasoning. One can say that the decoder has n - k redundant symbols to "spend," which is twice the amount of correctable errors. For each error, one redundant symbol is used to locate the error, and another redundant symbol is used to find its correct value. In implementation of (255,251) code, it is desired to correct up to 3 symbol errors i.e. t =2 and m=8 which is the number of data bits per symbol. Numerically, it can be expressed as from Eq. (5).

$$(n,k) = (2^m - 1, 2^m - 1 - 2t) \qquad \text{for m = 2, n =255, k=251} \qquad ....(5)$$

Thus, it can be said that the RS code (255,251) is used in this study [12].

RS's initial definition focuses on the evaluation of polynomials over the elements in a finite field (Galois Field (GF)) [9]. The primitive polynomial for m = 8 is given by

$$f(x) = x^8 + x^4 + x^3 + x^2 + 1 \qquad ....(6)$$

C. Pseudo Noise (PN) Sequences and CDMA Implementation

A Pseudo-random Noise (PN) sequence which is binary numbers, e.g. 0,1. is used in this paper. The receiver should be able to reject other interfering the Spread Spectrum (SS) signals and/or prevent false correlation with CDMA systems [9]. In CDMA implementation, there are only two users i.e. N = 2 for this research. The BPSK modulated code functions are used in this paper for (255,251) code as

pseudonoise 1 = [0 0 0 1 0 0 1 1 0 1 0 1 1 1 1];

pseudonoise 2 = [1 0 0 0 1 1 0 0 1 0 1 1 1 1 0];

In this study, two audio signals in the form of .wav file are recorded since there are two different data requirements for CDMA application.

III. SIMULATION RESULTS

The three second length audio signals were recorded in MATLAB environment with fs = 8 kHz sampling frequency, and 8 bits resolution. The first and second audio signals are shown in Fig. 1 and Fig. 2 respectively.

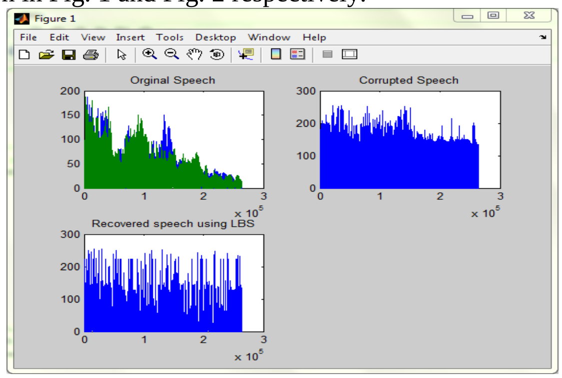

Fig.1: Simulation result on MATLAB

In the second step, amplitude values between -1 and 1 are mapped to the range of 0 to 255 and these values are expressed by the 8 bit binary number system. The signals obtained in the second step are transformed into Galois space and the G generator matrix is formed appropriately. RS channel coding is performed with these components. AWGN is added and the noisy signals are modulated with BPSK in the fourth step. After modulation process through, both audio signals are spreaded by using CDMA. Then, both signals are dispreadedand demodulated. In the second part of the study, four wavelet transform families at different levels were applied to the audio signals and the performances of these classes were compared. Using wavelet transform, instead of the first data, compressed data at different levels is inserted into the designed communication system and obtained signals at the output are observed.

For example, the number of samples after compression with first order wavelet transform became 20000 (40000 samples before compression) samples.

IV. CONCLUSION

According to the obtained results, RS coding method is a suitable method which can be used for channel coding. The

BER performance curve is obtained inversely proportional to the signal-to-noise ratio. The coding of the channel by the RS coding method has reduced the error effect of the AWGN in the transmitted operation. In the second part of this study, four wavelet transform families at different levels were applied to the audio signals and the performances of these families were compared. MSE performances were obtained at db1 of the Daubechies wavelet family at the 4th best level for both signals. After the first audio signal is transmitted from the AWGN channel, there is no difference between the transmitted and received audio signal except for some acoustic differences. With RS coding, the sampling frequency can be increased and each sample can be expressed with more bits in order to achieve better performance. However, these increases will lead to a longer transmission of the audio signal from the channel.

V. REFERENCES

[1]. John G. Proakis, and Masoud Salehi, Digital Communications, Fifth Edition, McGraw-Hill Inc.2008.

[2]. Lin and Daniel J Costello,Error control coding: fundamentals and applications, Englewood Cliffs, Prentice Hall, NJ, 1983.

[3]. Moreira JC and Farrell PG Essentials of error-control coding. John Wiley &Sons, England, (2006).

[4]. Md. Sipon Miah, M. Mahbubur Rahman, T. K Godder, Bikash Chandra Singh and M. Tania Parvin,"Performance Comparison of AWGN, Flat Fading and Frequency Selective Fading Channel for Wireless Communication System using 4QPSK", IJCIT, Vol. 01, Issue 02, pp. 82-90, 2011.

[5]. Béchir TALEB ALI , Geneviève BAUDOIN, Olivier VENARD, "Data transmission over mobile voice channel based on M-FSK modulation", Wireless Communications and Networking Conference (WCNC) Proc., 2013.

[6]. Md. Golam Rashed, M. Hasnat Kabir, Md. Selim Reza, Md. Matiqul Islam, Rifat Ara Shams, Saleh Masum, Sheikh Enayet Ullah, "Transmission of Voice Signal: BER Performance Analysis of Different FEC Schemes Based OFDM System over Various Channel", International Journal of Advanced Science and Technology, vol. 34, pp.89-100, 2011.

[7]. Tiantioukas Nikolaos, "Effects of the Wireless Channel, Signal Compression and Network Architecture on Speech Quality ın VOIP Networks", Master of Science in Electrical Engineering Thesis, Naval Postgraduate School Monterey, June 2007.

[8]. Ali Tariq Bhatti, "Cyclic Channel Coding algorithm for Original and Received Voice Signal at 8 KHz using BER performance through Additive White Gaussian Noise Channel", IJISET - International Journal of Innovative Science, Engineering & Technology, Vol. 2 Issue 4, April 2015.

[9]. Ali Tariq Bhatti, Dr. Jung H. Kim,"Implementation of Reed-Solomon (RS) and CDMA for signaling a voice through AWGN at 8 KHz sampling frequency using BPSK", International Journal of Advent Research in Computer and Electronics (IJARCE), vol.2, no.8, pp.13-27, August 2015.

[10]. Thuraya Mahmood Qaradaghi, Fatima Kamil Faek, Diana Hayder Hussein, "Investigating the Effect of Error Correcting Codes on the Compressed Speech Signals", 1st International Conference on Engineering and Innovative Technology, SU-ICEIT 2016, Salahaddin University-Erbil, Iraq April 12-14, 2016.