Transmission of Audio Signal from Reed-Solomon AWGN Channel Using Wavelet Transform Families

S. Karagol, and D. Yildiz

Abstract— **Mobile communication has become an important part of our daily lives for voice communication, 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 primary purpose of the channel coding process is to reduce the effect of disturbing influences to the minimum level. In this way, the receiver will be able to receive the data sent by the transmitter with the minimum error. In this research paper, it is required to acquire two audio signals with 8 bits of resolution and 8 kHz sampling frequency for 5 seconds in MATLAB. In the first part of this study, Reed-Solomon (RS) Coding Method was used to transmit two audio data recorded in .wav format with minimum error. Binary Phase Shift Keying (BPSK) modulation was used as the modulation method. The Bit Error Rate (BER) performance curve was plotted 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 transform families at different levels were applied to the audio signal, and the performances of these classes were compared.**

Index Terms— **Audio Signal, Channel Coding, Reed-Solomon Coding, Wavelet Transform Families.**

I. Introduction

TO TRANSMIT data 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 other purpose of the channel coding process is to reduce the effect of some disturbing elements that the data has encountered in the transmission phase as much as possible. The process ensures that the data is delivered to the receiver with minimum error.

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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 comprises of algorithms and techniques 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 data transmission with minimum loss. Miah and Rahman [4] used a wireless communication simulator that included Gray coding, modulation, different channel models (frequency selective fading channels, smooth fading channels, Additive White Gaussian Noise (AWGN)), 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 transmitting over a mobile audio channel based on the M-Phase Shift Keying (M-PSK) 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. In [7], measuring voice quality in VOIP communication was aimed. More specifically, besides the channel coding and compression methods, the effect of wireless channel conditions on the quality of the voice signal received was investigated.

In the study of [8], audio signal that is 5 sec and in .wav format where each sample was expressed with 8 bits that has 8 kHz sampling frequency was recorded in the MATLAB environment. This obtained audio signal was coded with the Cyclic Coding Method by applying to the AWGN noisy channel. Binary Phase Shift Keying (BPSK) is used for the modulation of the audio signal. Finally, the difference between the original signal and the received signal was observed by drawing the Bit Error Rate (BER) performance curve. In [9], Bhatti used Reed-Solomon coding and Code Division Multiple Access (CDMA) methods unlike in [8].

Wavelet transforms have also been used in some topics on channel coding. In the study [10], a wavelet-based voice-coded communication system was applied to compress the size of the audio signal and the performance evaluation of this algorithm was performed. In [11], some applications of Discrete Wavelet Transform (DWT) have been described for the problem of extracting information from non-speech sounds.

In the first part of this study, two audio signals were recorded for 5 sec, in .wav format where each samples were expressed with 8 bits that has 8 kHz sampling frequency in MATLAB environment. These obtained audio signals were coded with the Reed-Solomon Coding Method by applying to the AWGN noisy channel. Binary Phase Shift Keying (BPSK) was used for the modulation of the audio signals. After modulating with BPSK, CDMA encoded signals were spread. The signals were then dispread and demodulated after passing through the AWGN channel. Finally, the differences between the original signals and the received signals were observed by drawing the Bit Error Rate (BER) performance curve. 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.

II. Audio Transmission steps

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

A. Channel Coding

The main aim of the channel coding process, which has an important place in communication systems, is to reduce transmission errors. Therefore, the coding made for this aim 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 receiver decodes each bit independent from the others, it decides on the majority of the information bits. 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

Irving Reed and Gus Solomon published a paper in the Journal of the Society for Industrial and Applied Mathematics in 1960 [12]. These codes are used in many applications today due to their powerful performances. As example of these applications, compact disc players and deep space applications can be shown.

Reed-Solomon (R-S) Codes are cyclic and non-binary codes. They have symbols consisting of m-bit arrays where m is an integer greater than 2. R-S (n, k) codes on m-bit symbols exist for all k and n for which

$$0 < k < n < 2m + 2 \tag{1}$$

In Eq. (1), the symbol k represents the number of data symbols being encoded and n represents the total number of code symbols in the encoded block.

For conventional R-S (n, k) code,

$$(n,k) = (2m - 1, 2m - 1 - 2t) \tag{2}$$

where t is the symbol-error correcting capability of the code and n - k = 2t is the number of parity symbols.

The distance between two code words in non-binary codes is defined as the number of different bits in the sequences. For Reed-Solomon codes, the code minimum distance is given by

$$d_{\min} = n - k + 1 \tag{3}$$

The code can correct t bits or fewer errors than t bits and any combination of these errors, where t can be stated as

$$t = \left\lfloor \frac{d_{\min} - 1}{2} \right\rfloor = \left\lfloor \frac{n-k}{2} \right\rfloor \tag{4}$$

and $\lfloor x \rfloor$ represents the largest integer less than x. Eq. (4) for R-S codes shows that up to 2t bit parity symbols are needed to correct the t bit symbol error. The decoder has n-k redundant symbols for error correction. A redundant symbol is used to locate each error. Another redundant symbol is used to find the correct value of the faulty 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).

$$(n,k) = (2^m - 1, 2^m - 1 - 2t) \tag{5}$$

RS Codes focus on the evaluation of polynomials over the elements in a finite field called as Galois Field (GF) [9]. The primitive polynomial for m = 8 is given by

$$f(x) = x^8 + x^4 + x^3 + x^2 + 1 \tag{6}$$

The RS code (255,251) is used in this research study.

C. Pseudo Noise (PN) Sequences and CDMA Implementation

A Pseudo-random Noise (PN) sequence which only includes binary numbers 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 study. The BPSK modulated code functions are used for (255,251) code as

Pseudo noise 1 = [0 0 0 1 0 0 1 1 0 1 0 1 1 1 1];
Pseudo noise 2 = [1 0 0 0 1 1 0 0 1 0 1 1 1 1 0];

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

III. Wavelet Transform

Wavelets are a method that can be applied to many different fields including applied mathematics, signal processing techniques, audio and image compression techniques. The wavelets were first used by Jean Morlet and A. Grossman for geographic information systems. In fact, the basic beginning of the wavelets extends to Joseph Fourier and to his Fourier transformation. With the emergence of Fourier equations after 1807, mathematicians tented to frequency analyzes for signal recognition. In 1977, Esteban and Galand introduced a new filter concept, but in this way the error was high in regaining the main signal [13]. The wavelet term was first used in 1984 by Morlet and Grossman in quantum physics studies [14]. In 1987 Mallat emerged the relation between wavelet and filter groups. Meyer revealed the first wave named his name [15]. This was a function that could be used in continuous applications, as opposed to Haar wavelets. Over the years, Ingrid Daubhecies has become the basis for many of today's application by coming up with set of vertical base wavelet series [16]. Wavelet functions are derived from the main wavelet by varying various parameters.

IV. Simulation Results

The five 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. 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 dispreaded and demodulated

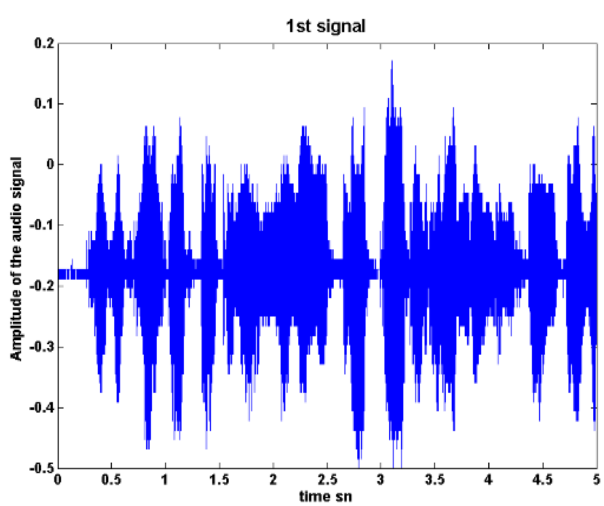

Fig. 1. 1st original signal

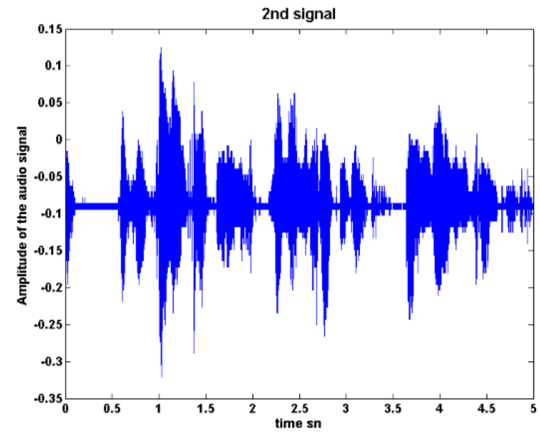

Fig. 2. 2nd original signal

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 original 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. According to Table I and Table II, the Mean Square Error (MSE) performances of Daubechies wavelet family db1 are the best at level 4 for both signals.

TABLE I.
MSE PERFORMANCES OF WAVELET TRANSFORM FAMILIES FOR SIGNAL 1

Wavelet classes		**MSE values**			
		1. order	**2. order**	**3. order**	**4. order**
Daubec.	db1	1.03E-05	2.83E-06	1.56E-06	**4.40E-07**
	db2	4.77E-05	1.46E-04	6.49E-05	7.35E-06
	db3	1.17E-04	1.04E-04	-	-
Symlet	sym1	9.31E-06	4.82E-06	2.26E-06	8.39E-07
	sym2	4.89E-05	1.33E-04	6.10E-05	7.62E-06
	sym3	1.22E-04	1.03E-04	-	-
Coiflet	coif1	1.18E-04	9.95E-05	-	-
	coif2	2.02E-04	9.68E-05	3.24E-05	-
	coif3	1.51E-04	1.01E-04	-	-
Biorth.	bior1.1	1.29E-05	4.59E-06	8.27E-07	4.86E-07
	bior1.3	1.23E-04	1.17E-04	5.26E-05	-
	bior1.5	2.45E-04	1.34E-04	-	-

TABLE II.
MSE PERFORMANCES OF WAVELET TRANSFORM FAMILIES FOR SIGNAL 2

Wavelet classes		MSE values			
		1. order	2. order	3. order	4. order
Daub.	db1	7.39E-06	1.66E-06	2.91E-07	**9.64E-08**
	db2	1.15E-05	8.78E-06	3.21E-06	6.04E-07
	db3	1.57E-05	7.50E-06	-	-
Symlet	sym1	7.02E-06	2.21E-06	6.03E-07	2.92E-07
	sym2	1.15E-05	7.79E-06	2.68E-06	6.83E-07
	sym3	1.48E-05	6.90E-06	-	-
Coiflet	coif1	1.47E-05	7.65E-06	-	-
	coif2	1.81E-05	5.35E-06	1.66E-06	-
	coif3	1.68E-05	7.42E-06	-	-
Bior.	bior1.1	7.67E-06	2.09E-06	8.54E-07	1.68E-07
	bior1.3	1.60E-05	6.65E-06	2.67E-06	-
	bior1.5	1.92E-05	8.11E-06	-	-

The differences between the received signals and original signals are examined by using BER vs EbNo plot as shown in Fig. 3. The bit error rate varies inversely with the signal noise value.

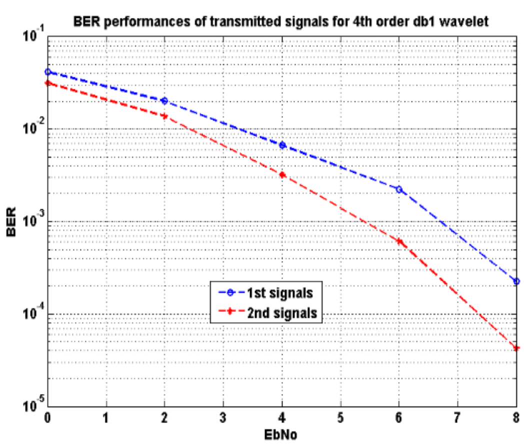

Fig. 3. BER performances of transmitted signals for 4th order db1 wavelet

V. 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.

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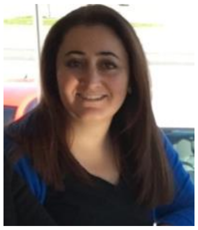
BIOGRAPHIES



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