Linear Block Coding and Discrete Wavelet Transform Based Audio Signal Transmission over AWGN and Rician Fading Channel

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 purpose of the channel coding process is to reduce the effect of some disruptive elements that the data is influenced in the transmission phase as much as possible. This procedure ensures that the data is delivered to the receiver with minimum error. In this study, a communication system was established in MATLAB environment for the transmission of five second audio signal. In this communication system, BER and MSE performances of four phase shift keying methods are obtained first. In the second phase of the study, the AWGN and Rician fading channels were individually coded with four different LBC types. Performance evaluation of these coding types was done with BER and MSE criteria. In the last part of the study, it was investigated which wavelet family is suitable for which level, in order to ensure audio transmission over AWGN and Rician fading channels with the least possible error. Four wavelet transform families at different levels were applied to the audio signal for LBC (7, 4), LBC (15, 8), LBC (17, 8) and LBC (23, 12) encoded channels, and the Mean Squared Error (MSE) performances were compared.**

Index Terms—**Audio Signals, Channel Coding, Linear Block Coding, Wavelet Transform.**

I. Introduction

TRADITIONAL ANALOG communication techniques do not perform well in mobile environments because appropriate techniques for reducing the effects of multi-path propagation fading have not been developed [1].

S. KARAGOL, is with Department of Electrical and Electronics Engineering, Ondokuz Mayis University, Samsun, Turkey, (e-mail: serap.karagol@omu.edu.tr)

https://orcid.org/0000-0002-5750-1143

D. YILDIZ, is with Department of Electrical and Electronics Engineering, Ondokuz Mayis University, Samsun, Turkey, (e-mail: dogan.yildiz@omu.edu.tr)

https://orcid.org/0000-0001-9670-4173

Manuscript received September 3, 2018; accepted March 26, 2019.
DOI: 10.17694/bajece.456774

Channel coding, also known as Forward Error Control (FEC) coding, can be defined as the detection and correction of bit errors in digital communication systems. Channel coding is performed on both transmitter and receiver. On the transmitter side, the channel coding is called the encoder. Here, extra bits (parity bits) are added to the raw data before modulation. On the receiver side, the channel coding is called the decoder. Channel coding allows the receiver to detect and correct errors caused by effects such as noise, parasite and fading that occur during transmission [2]. The key benefits of channel coding to communication systems are as follows: maximizing data transmission efficiency to provide high bit rate, low encoder / decoder complexity by limiting the size and cost of transceivers, minimizing the amount of energy required for transmission at the desired signal to noise ratio and thus achieving high Bit Error Rate (BER) performance. Lower-rate channel codes can usually correct more errors [3].

In recent years, many studies have been carried out using different channel coding methods in order to transmit data with the least possible loss. In [3], Dhaliwal et al. analyzed convolutional code's BER performance over Additive White Gaussian Noise (AWGN) channel using fixed Viterbi decoder, different code rates, different restriction lengths and Quadrature Phase Shift Keying (QPSK) modulation. In [4], Baviskar et al. proposed an adaptive Low Density Parity Control Code (LDPC) supported audio processing algorithm to transmit the speech signal over the wireless AWGN channel. Binary Phase Shift Keying (BPSK) technique have helped to achieve better BER performance. In [5], the robustness of the turbo decoding algorithm in a generalized Gaussian noise has been researched. In [6], Haque et al. studied the effect of various combined FEC codes on the performance of a wireless Orthogonal Frequency Division Multiplexing (OFDM) system. In this study, the authors used Reed-Solomon (RS) (255, 239, 8) encoder, Cyclic (15, 11) encoder, Bose-Chaudhuri-Hocquenghem (BCH) (127, 64) encoder and ⅔ and ¾ encoders of convolutional coding for channel coding. QPSK, 8-PSK, 32-Quadrature Amplitude Modulation (32-QAM) and 64-QAM methods are used as the digital modulation method. In [7], Sidhu et al. evaluated the performance of multi-level linear block codes on Rayleigh fading channel. The error correction feature of linear block codes is controlled with different digital modulations. In [8], BER performance of 2 × 2 Full-Rate Linear-Receiver Space

Time Block Code (FRLR-STBC) is evaluated. Using Appell and Gauss hypergeometric functions, new precise and asymptotic closed form BER formulas of BPSK-modulated FRLR-STBC are derived over Rayleigh fading channels, along with the corresponding coding gain and diversity order.

In the literature, Wavelet Transform (WT) is also used in some applications related to audio signal processing and channel coding. In [9], Romano et al. proposed a technique for analyzing and compressing speech signals using biorthogonal wavelet filters. This technique compared human voice with a typical Voice Over Internet Protocol (VoIP) encoding. This study has been emphasized how the wavelet filters can be suitable, especially in terms of the compression ratio, without causing significant degradation of the signal quality for the audiences. In [10], Kudumakis et al. investigated the performance of some different wavelet families for low bit rate coding of audio signals. For evaluating the coding gains of these wavelets, both octave and uniform sub-band coding schemes, fixed and dynamic bit-shares with entropy noiseless Huffman coding are used. In [11], Luo presented a new ultra-low latency wavelet voice encoder for real-time wireless transmission using non-overlapping short block processing and embedded coding. To investigate the correlation of the audio signal, a limited and minimized 2D fast lift WT is developed. In [12], Sinha et al. used a wavelet-based method that uses the median function threshold. By selecting a suitable main wavelet, a nearly unchanged signal quality and a flexible compression ratio are obtained and the performance of the proposed method is evaluated using various analysis methods. In [13], Sarnin et al. presented a performance evaluation of BPSK and QPSK techniques using error correcting code. BCH, Cyclic and Hamming codings are used as a coding method. Random data is passed through the AWGN channel and the BER is calculated. In [14], Bhatti recorded a speech signal of .wav format, represented by 8-kHz sampling frequency, 5-sec and 8 bits per sample in MATLAB environment. The audio signal is encoded in the AWGN channel using the Cyclic (17, 8) Coding Method. BPSK was used for the modulation of the audio signal. In [15], the authors used RS coding and Code Division Multiple Access (CDMA) methods, unlike the study [14].

In this study, a communication system was established in MATLAB environment for the transmission of 5 second audio signal. In this communication system, BPSK modulation is used in the remainder of the study, because the modulation analysis with good performance is performed first and the performance of BPSK modulation is obtained better than the other modulations. In the second phase of the study, the AWGN and Rician fading channels were individually coded with LBC (7, 4), LBC (15, 7), LBC (17, 8) and LBC (23, 12) respectively. Performance evaluation of these coding types was done with BER and MSE criteria. In the last part of the study, it was investigated which wavelet family is suitable for which level, in order to ensure audio transmission over AWGN and Rician fading channels with the least possible error. Four wavelet transform families at different levels were applied to the audio signal for four different types of LBC encoded channels, and the Mean Squared Error (MSE) performances were compared.

II. Audio Transmission Steps

In this study, the five-second length audio signal obtained from the external environment is passed through the steps of sampling, quantization and A/D conversion and then given to AWGN noisy (or AWGN noisy with Rician fading) LBC coded channel. In this section, the basic concepts used in the system for the transmission of the audio signal will be mentioned.

A. Channel Coding

In the transmitter part of the digital communication system, the data like sound, image etc. produced by the source is converted into binary sequences with the shortest possible length, with no loss of information (or acceptable loss of information) with the aid of the source encoder. The binary k-bit sequences at the output of the source encoder are converted to n-bit sequences by adding n-k control bits with the aid of the channel encoder (n>k). The intent of channel coding is to protect the data sequences from the disturbing effect of the channel. The coded arrays are then divided into blocks of a certain length. These sequences are then mapped to one of the elements of the signal set of the modulation to be used and modulated to give the channel. The type of modulation used varies depending on the channel's feature. On the receiver side, the demodulator allows distorted waves to be processed in the decoder due to disturbing effects in the channel. The channel and source decoders operate inversely to the channel and source encoders, respectively, to find the transmitted data and transfer it to the user. Because of the redundancy control bit insertion in the channel coding, the increase in sequence length reduces the data rate. One way to avoid this fall is to increase the modulation speed. The increase in modulation speed means to shorten the duration of the channel signal, which means that the bandwidth of the channel signal increases. Therefore, this can be done if the bandwidth of the channel is appropriate. The expansion of the signal set (for example, using 8-PSK modulation instead of 4-PSK) is another way of avoiding the drop in the transmission rate in band-limited channels. However, in this method, since the signs are closer to each other, the error performance of the system is reduced. Therefore, coding and modulation together in the design of efficient systems in terms of both error performance and bandwidth are now considered as a whole [16].

In this study, LBC is used for channel coding process and BPSK is used as modulation method.

B. Linear Block Coding

LBC is an encoding method characterized by an (n, k) notation, where k is the length of the message vector and n is the length of the codeword, and LBC is a subclass of similarity control codes. Here, the encoder converts the message block of length k into a longer code word block. The code is a set of vectors created by vectors called code words. k-bit messages constitute 2k different message arrays, referred to as k-arrays. Blocks of n bits form 2n different arrays, expressed as n-arrays As a result of the encoding process, each of the 2k k-arrays is assigned to each of 2n n-arrays. With a block code, each of the 2k message arrays is mapped to 2k different code words in the n-array. This mapping is an one to one correspondence.

V_i and V_j are two separate code words from a set of (n, k) binary block codes. In order for any of these two words to be linear, the necessary and sufficient condition is that $V_i \oplus V_j$ is also a code vector at the same time. In general, a G generator matrix is defined as a k×n matrix:

$$G = \begin{bmatrix} V_1 \\ V_2 \\ \vdots \\ V_k \end{bmatrix} = \begin{bmatrix} V_{11} & V_{12} & \cdots & V_{1n} \\ V_{21} & V_{22} & \cdots & V_{2n} \\ \vdots & \vdots & & \vdots \\ V_{k1} & V_{k2} & \cdots & V_{kn} \end{bmatrix} \quad (1)$$

m message vector is a 1 × k row vector consisting of k message bits:

$$m = [m_1 m_2 \dots m_k] \quad (2)$$

The U code vector is expressed as the product of the m and G sequences as follows:

$$U = mG \quad (3)$$

Hence, it can be said that the code vector of a message is a linear combination of the components of the G generator matrix. With a G generator matrix, a vector of length k is translated into a vector of length n. Here, the (n-k) bits are the parity bits. A systematic linear block code has a generator matrix in the form:

$$G = [P : I_k] \quad (4)$$

Here, since the unit matrix is not required to be stored, the systematic generator matrix will reduce the coding complexity. Eq. (5) could be obtained by combining Eq. (3) and (4).

$$U = [m_1 m_2 \dots m_k] \times \left[\underbrace{\begin{matrix} P_{11} & P_{12} & \dots & P_{1,(n-k)} \\ P_{21} & P_{22} & \dots & P_{2,(n-k)} \\ \vdots & \vdots & & \vdots \\ P_{k1} & P_{k2} & \dots & P_{k,(n-k)} \end{matrix}}_{P} \underbrace{\begin{matrix} 1 & 0 & \dots & 0 \\ 0 & 1 & \dots & 0 \\ \vdots & \vdots & & \vdots \\ 0 & 0 & \dots & 1 \end{matrix}}_{I} \right] \quad (5)$$

The systematic code vector can be expressed as follows.

$$U = [\underbrace{P_1 P_2 \dots P_{n-k}}_{parity\ bits} \underbrace{m_1 m_2 \dots m_k}_{message\ bits}] \quad (6)$$

The parity check matrix (H) provides us with a simple way to know if an error has occurred. It is necessary for decoding in receiver. GH^T has to be equal to zero. Since U = mG, $UH^T = 0$ condition will be provided for all valid codes. U can be said to be a code word generated by the G generator matrix if $UH^T = 0$ is satisfied. The vector r= $[r_1 r_2 \dots r_n]$ is the signal at the receiver and this vector is the consequent of the transmission of the U= $[u_1 u_2 \dots u_n]$ vector over the channel. Eq. (7) gives the vector r.

$$r = U + e \quad (7)$$

Here the vector e = $[e_1\ e_2\ \dots\ e_n]$ is the error vector. The syndrome of vector r is S and it is given with Eq. (8) [17].

$$S = rH^T \quad (8)$$

C. *Types of Channels*

It is much more convenient to construct mathematical models that reflect the characteristics of transmission media during the design of communication systems for the transmission of information over physical channels. Thus, this mathematical model of the channel can be used to design channel coders and modulators in transmitters and channel decoders and demodulators in receivers.

1) *AWGN Channel*

One of the simplest mathematical models of a communication channel is the additive noise channel shown in Fig. 1. In this model, the transmitted signal s(t) is distorted by the additive noise process n(t).

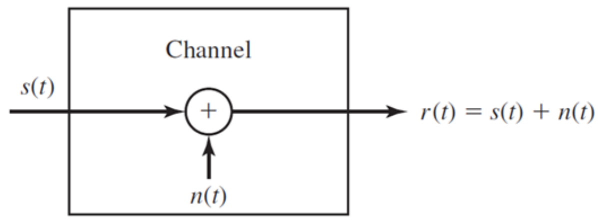


Fig. 1. The additive noise channel

Physically, additive noise process is produced by electronic components and amplifiers in the receiver end of the communication systems as well as in the case of transmission of radio signals. If the noise is mainly produced by the electronic components and the amplifier on the receiver side of the communication system, this noise can be characterized as thermal noise. This type of noise is statistically characterized as a Gaussian noise process. The mathematical model to be obtained for this channel is therefore called Additive White Gaussian Noise (AWGN) Channel. Since this channel model is a valid model for most of the physical communication channels and can be mathematically examined, it is a channel model that is often used in the analysis and design of communication systems. Channel attenuation components can easily be integrated into this model.

If a signal is attenuated during transmission along the channel, the received signal will be in the form of

$$r(t) = \alpha s(t) + n(t) \quad (9)$$

Here, α represents the attenuation factor [18]. It can be said that a random X variable has a normal distribution if it has a probability density function (pdf) with μ average and σ^2 variance values as in the following Eq. (10) [19].

$$P(x) = \frac{1}{\sqrt{2\pi\sigma}} \exp\left[\frac{-(x-\mu)^2}{2\sigma^2}\right], \qquad -\infty < x < \infty \quad (10)$$

2) *Rician Channel*

The amplitude envelope of the small-scale faded channel is Rician distributed if there is a predominantly non-attenuated signal component (such as LOS) [20]. At the output of the envelope detector, the dominant signal causes the formation of a direct current component. The Rayleigh distribution is obtained when the dominant signal component in the Rician distribution is damped [21]. This channel can be seen in Fig. 2.

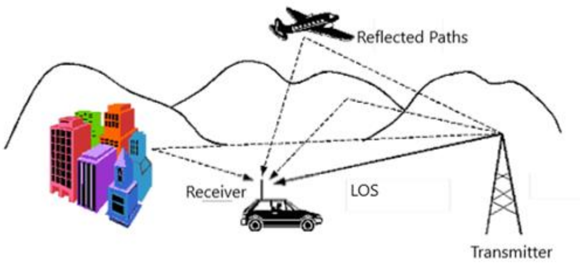


Fig. 2. Rician fading channel [22]

The Rician distribution is as follows.

$$P_{Rician}(r) = \frac{r}{\sigma^2} e^{\frac{-(r^2+A^2)}{2\sigma^2}} I_0\left(\frac{rA}{\sigma^2}\right), \quad A \geq 0 \; and \; r \geq 0 \tag{11}$$

Here A – is the maximum value (envelope) of a dominant signal

$I_0(.)$ – is the degree 0, first modified Bessel function

σ^2 – is the power of the perpendicular components of the channel's complex damping variable.

In Fig. 3, the Rician distribution is given for different A values [22].

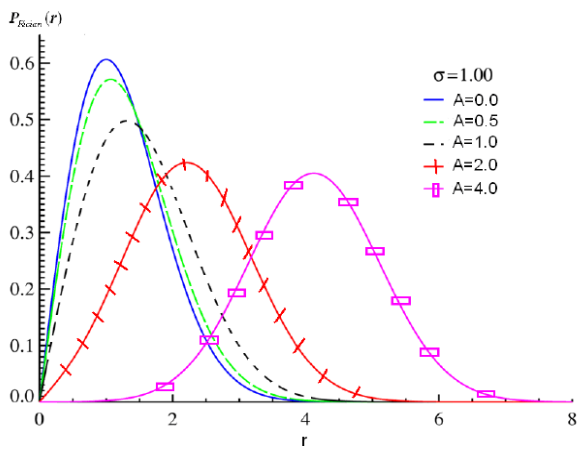


Fig. 3. Probability density function of Rician distribution [22]

The Rician distribution is usually expressed using the K parameter. The K parameter is calculated in dB:

$$K(dB) = 10\log\frac{A^2}{2\sigma^2} \; [dB] \tag{12}$$

The average power of the Rician distribution is as follows:

$$\Omega = E[r^2] = A^2 + 2\sigma^2 = 2\sigma^2(1+K) \tag{13}$$

Using these parameters, the probability density function can be rewritten as in Eq. (14).

$$P_{Rician}(r) = \frac{2r(K+1)}{\Omega} e^{-K-\frac{r^2(K+1)}{\Omega}} I_0\left(2r\sqrt{\frac{K(K+1)}{\Omega}}\right) \tag{14}$$

If K = 0 is written in Eq. (14), the Rician distribution becomes the Rayleigh distribution. The increase in the value of the K parameter means that the effect of the fading in the channel is reduced. In the case of K → ∞, there is no fading in the channel and in this case, this channel is converged to the AWGN channel [22].

D. Wavelet Transform

Wavelets are small waves with an average value of 0, with a limited time interval. Wavelets are mathematical functions that localize a set of data or a function in both the frequency domain and the time domain. It can be said that the beginning of wavelet functions is Haar's thesis in 1909. In the 1930's, scale-varying functions were developed in physics, mathematics, seismology and electronics engineering. The broad concept of wavelet was introduced by Grossman and Morlet (1984) in the mid 80s [23].

WT gives successful results in data analysis where the Fourier transform is used ahead of time and some aspects are inadequate. For example, signals with alternating small or irregular details are generally better analyzed using wavelets compared to the conventional Fourier transform. WT is used extensively in one-dimensional and two-dimensional signal processing applications such as voice and audio signal processing, communication, geophysics, economics and medicine [24].

There are two types of wavelet analysis: Continuous Wavelet Transform (CWT) and Discrete Wavelet Transform (DWT). In CWT, the signal being processedi is matched and transformed with the wavelet fundamental function in continuous frequency and time increases. The original signal is expressed as a weighted integral of the continuous basis wavelet function. If f(t) is a square integrable function of time, t, then the CWT of f(t) is defined as

$$W_{a,b} = \int_{-\infty}^{\infty} f(t) \frac{1}{\sqrt{|a|}} \psi^*(\frac{t-b}{a}) dt \tag{15}$$

where $a, b \; \epsilon R, a \neq 0$, R - is the set of real numbers;
'*' – denotes the complex conjugation; and the wavelet function is defined as

$$\psi_{a,b}(t) = \frac{1}{\sqrt{|a|}} \psi(\frac{t-b}{a}) \tag{16}$$

Here: a – is called the scaling parameter which captures the local frequency content;

b – is called the translation parameter which localizes the wavelet basis function at time t = b and its neighborhood.

If the scale and translation parameters a and b are taken at discrete values, DWT is obtained. In this case, the parameters a and b are often based on powers of two and called dyadic scales and translations:

$$a_j = 2^j, b_{j,k} = k2^j \quad \text{for all} \quad j,k \; \epsilon \; Z \tag{17}$$

In this situation, Eq. (15) becomes

$$\psi_{j,k}(t) = 2^{-\frac{j}{2}} \psi(2^{-j} \times (t-k)) \text{ for all } j,k \; \epsilon \; Z \tag{18}$$

The DWT discards all redundant information in CWT by employing a set of orthogonal basis functions. The multi-resolution decomposition formula for DWT of the original function as follows:

$$f(t) = \sum_{k=-\infty}^{\infty} c_k \Phi(t-k) + \sum_{k=-\infty}^{\infty} \sum_{k=-\infty}^{\infty} d_{j,k} \Psi(2^j.t-k) \qquad (19)$$

where $\Phi(.)$ – is the scaling function;
$d_{j,k}$ – are the wavelet coefficients
c_k – are the scaling coefficients.

In the right hand side of Eq. (18), the first term represents an approximation to the general trend of the original function and the second term represents the local details in the original function. The wavelet coefficients $d_{j,k}$ times the dilated and translated wavelet function can be interpreted as the local residual error between successive signal approximations at scales j-1 and j and Eq. (20) is the detail signal at scale j [24].

$$r_j(k) = \sum_{k=-\infty}^{\infty} d_{j,k} \Psi(2^j.t-k) \qquad (20)$$

There are many wavelets in the literature proposed by different researchers. Daubechies (db), Symlet (sym), Coiflet (coif) and Biorthogonal (bior) wavelets are used in this study. Some types of these wavelets are shown in Fig. 4.

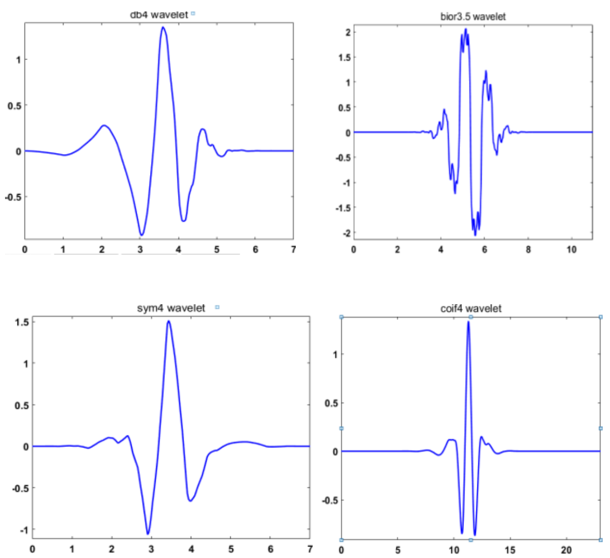


Fig. 4. Some types of Daubechies (db), Symlet (sym), Coiflet (coif) and Biorthogonal (bior) wavelets

III. Simulation Results

In this section, the performances of different coding types on different channels and in different conditions are obtained using MATLAB simulations. LBC (15,8) is used first to determine the performance of phase shift keying modulations. The created communication system is also explained for LBC (15, 8). Similar logic applies for other types of LBC encoding.

The speech signal (original audio signal) used in simulation is sampled at f_s = 8 kHz frequency and recorded in MATLAB environment for 5 seconds with 8 bits resolution. The original audio signal is shown in Fig. 5.

In the second step, the amplitude values of -1 to 1 are shifted to the range of 0 to 255 and these values are expressed by the 8 bit binary number system. In the third step, the H parity check matrix and the G generator matrix are formed appropriately. The audio signal is encoded as a 40,000 x 8 matrix using these matrices. AWGN noise was first added in the fourth stage (then Rician fading). The S (7x15 matrix) syndrome test matrix, which will be used in the recovery phase of the audio signal, has been produced. Here, since the AWGN is added to the original audio signal at 0.8, the largest amplitude value increases from about 0.25 to about 1 and the smallest amplitude value decreases from about -0.25 to about -1.

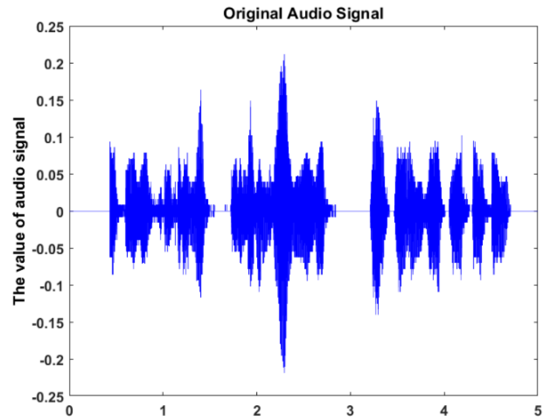


Fig. 5. Original audio signal

The difference between the signal obtained in the coding end resultant signal and the first original audio signal, namely error, is found for different modulation methods. In the course of determining the modulation with the best performance, AWGN channel and Rician fading channel are used for both coded and non-coded scenarios. BER-EbNo graphs for different EbNo values were traced and the performances of channel-coded and non-channel-coded modulation techniques were compared. For channel-coded scenario, LBC (15, 8) coding technique is used. Fig. 6 shows the performance without channel coding of the modulations and Fig. 7 shows the channel-coded performances of the modulations in AWGN channel.

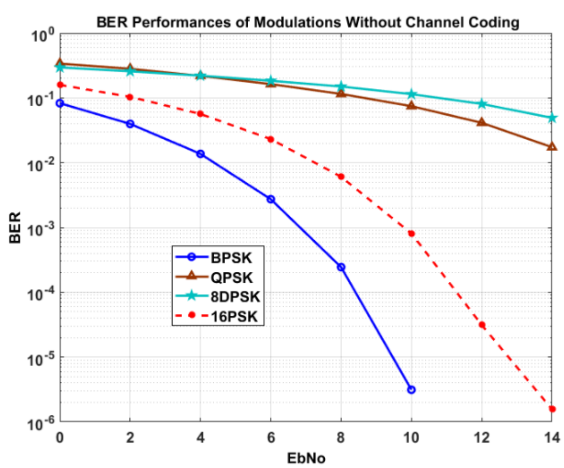


Fig. 6. BER performances of modulations without channel coding in AWGN channel

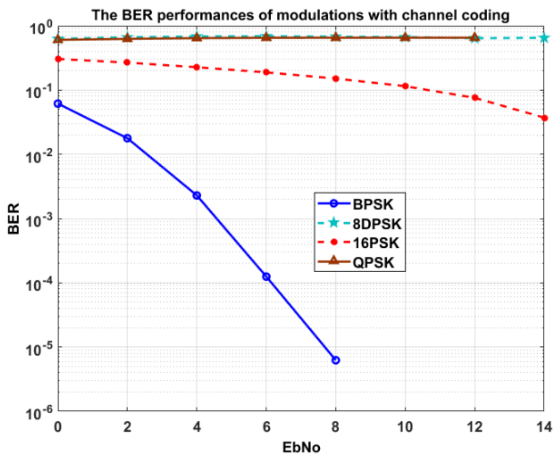


Fig. 7. BER Performances of modulations with channel coding in AWGN channel

Fig. 8 shows the non-channel-coded performances of the modulations and Fig. 9 shows the channel-coded performances of the modulations in Rician fading channel. For this simulation, K-factor =10 value is used.

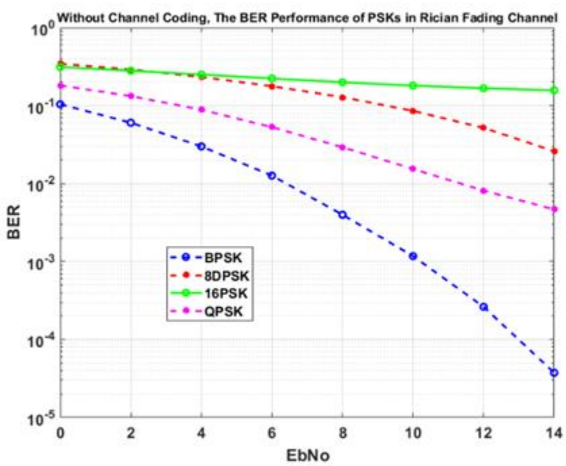


Fig. 8. BER performances of modulations without channel coding in Rician fading channel

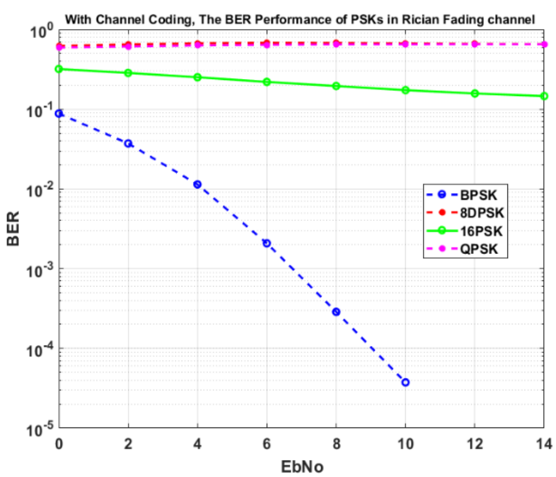


Fig. 9. BER performances of modulations with LBC (15,8) channel coding in Rician fading channel

The BER varies inversely with the signal to noise ratio. As can be seen from the figures, when the BER performances of the modulations are compared, the performance of the BPSK modulation is better than that of the other modulations for both channel coded and non-channel coded situations and for both of the channel types. The MSE performances of the modulations for AWGN channel and for LBC (15, 8) coded version are given in Table I.

TABLE I
THE MSE PERFORMANCES OF THE MODULATIONS WITH LBC (15,8) CHANNEL CODING

Modulation	**BPSK**	**8-DPSK**	**16-PSK**	**QPSK**
MSE	1,49E-05	0,98169	0,041216	0,061413

For simulations done on the rest of the study, BPSK is used as modulation only because BPSK has good performance as the above figures and table show.

In the second phase of the study, the performance of different types of LBC codes on AWGN and Rician fading channels was investigated. For this purpose, firstly, MSE performance evaluation of LBC coding types in AWGN channel was performed. Table II shows the MSE performances of the LBC coding types in the AWGN channel. As can be seen in the table, the best MSE performance is achieved with the LBC (23, 12) coding type, which has a higher code length.

TABLE II
THE MSE PERFORMANCES OF LBC CODING TYPES ON AWGN CHANNEL

LBC Coding Types			
LBC (7,4)	**LBC (15,8)**	**LBC (17,8)**	**LBC (23,12)**
0,003	1,49E-05	1,49E-05	5,77E-08

The BER performance of the LBC (23, 12) coding type is shown in Fig. 10. As can be seen from the figure, high BER performance is achieved even with very low EbNo values using LBC (23, 12).

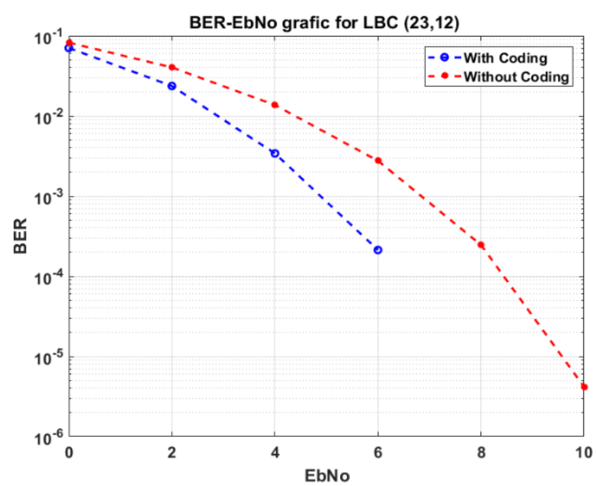


Fig. 10. BER performance of LBC (23,12) in AWGN channel

Secondly, MSE performance evaluation of LBC coding types in Rician fading channel was performed. Table III shows the

MSE performances of the LBC coding types in the Rician fading channel. As can be seen in the table, the best MSE performance is achieved with the LBC (23, 12) coding type same as AWGN channel. As seen in the table, the coding types in the Rician fading channel converge on the AWGN channel with different K-factor values.

In the Rician fading channel, a not-channel coded BER curve is also drawn in order to see the difference between the channel coded version and the not-channel coded version for the best MSE performance (LBC (23,12)). Fig. 11 shows the BER performance for the LBC (23, 12), K-factor = 14 case in Rician fading channel. Both not-coded and coded situations can be seen from the figure.

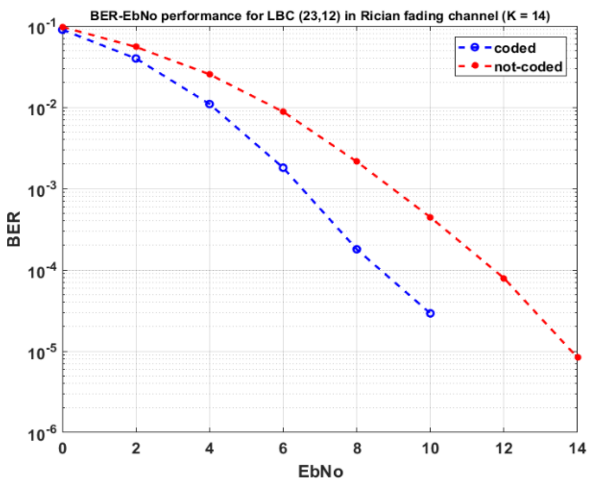


Fig. 11. BER performance of LBC (23,12) in Rician fading channel (K =14)

At the end of the study, four WT families at different levels were applied to the audio signal, and the performance of these families was first compared for the AWGN channel and then for the Rician fading channel. Using DWT, data compressed at different levels instead of the original data are put into the designed communication system and the performance obtained at the output is observed. For example, while the number of samples in the original dataset is 40,000, the number of samples in the dataset obtained from the first-level decompression is 20,000. In Table IV, the MSE performances of different wavelet transform families for LBC (15, 8) obtained by compression in different levels are given in the AWGN channel.

In these conditions, MSE performance was best obtained at db1 part of the Daubechies wavelet family at level 5, sym1 part of the Symlet wavelet family at the 5th level and bior1.1 part of the biorthogonal wavelet family at the 5th level. As can be seen, the MSE performance improves as the compression level increases.

The performances of different LBC species in Rician fading channels with different K-factor values was then obtained. These performances were obtained using DWT at the first, second, and third level compression levels, respectively. Table V, Table VI and Table VII show the performances of the wavelet families at these levels, respectively. As can be seen, as the code size increases, the K-factor value that converge to the AWGN channel of the Rician fading channel also increases. And generally, as the DWT compression level increases, the K-factor value converging the AWGN value also shrinks. As the compression level increases, the MSE performance improves, and as a result no other compression levels need to be provided. The relationship between compression level and MSE for LBC (7, 4) is shown in Fig. 12.

IV. CONCLUSION

According to the obtained results, the LBC method is a suitable method which can be used for channel coding for both AWGN and Rician fading channels. For the channels and coding method considered, BER performance of 4 types of modulation is obtained for the best performance modulation detection. The BER performance of BPSK modulation is the best obtained from these modulations. The BER performance curve is obtained inversely proportional to the signal-to-noise ratio. The coding of the channel by the LBC method reduces the error effect of the channels in the transmitted signal. LBC (7, 4), LBC (15, 7), LBC (17, 8) and LBC (23, 12) used in this study and LBC (23, 12) was obtained as the best transmission performance from the LBC types studied for both of the channel types. The coding types in the Rician fading channel converge on the AWGN channel with different K-factor values.

At the end of the study, four WT families at different levels were applied to the audio signal, and the performance of these families was first compared for the different length LBC coded AWGN channels and then for the different length of LBC coded Rician fading channel. And generally, as the DWT compression level increases, the K-factor value converging the AWGN value also shrinks. As the compression level increases, the MSE performance improves. If both LBC (15, 8) and LBC (17, 8) performances are compared for both with wavelet and non-wavelet conditions, it can be said that the increase in the number of parity bits does not have much effect on the transmission performance. As a general evaluation, it can be said that Biorthogonal Wavelet family is the most suitable family among the families handled for audio transmission.

The difference between the original audio signal transmitted from the channels and the audio signal received at the receiver is not observed except for some acoustic differences. In order to obtain better performance as a result of voice transmission from the used channels by LBC method, sampling frequency can be increased and each sample can be represented by more bits. However, these increases will lead to a longer transmission of the audio signal from the channel.

TABLE III
THE MSE PERFORMANCES OF LBC CODING TYPES ON RICIAN FADING CHANNEL

K-factor	LBC Types							
	LBC (7,4)		LBC (15,8)		LBC (17,8)		LBC (23,12)	
	not-coded	coded	not-coded	coded	not-coded	coded	not-coded	coded
1	0,1751	0,061	0,1547	0,088	0,1551	0,0794	0,1525	0,0977
2	0,0611	0,014	0,051	0,0165	0,0521	0,0136	0,052	0,0175
3	0,0213	0,005	0,0152	0,0036	0,0158	0,0031	0,016	0,0036
4	0,0094	0,003	0,0052	7,50E-04	0,0054	7,72E-04	0,005	8,31E-04
5	0,0053	0,003	0,0021	1,84E-04	0,0021	2,24E-04	0,0021	7,60E-05
6	0,004	0,003	6,48E-04	3,67E-05	7,88E-04	6,07E-05	7,82E-04	1,27E-05
7	0,0035	0,003	3,00E-04	2,21E-05	3,48E-04	5,66E-05	2,93E-04	3,87E-06
8	0,0032	0,003	1,77E-04	1,49E-05	9,81E-05	3,49E-05	1,67E-04	1,57E-06
9	**0,003**	**0,003**	1,77E-04	1,49E-05	9,34E-05	1,49E-05	4,10E-05	5,77E-08
10	0,0031	0,003	4,27E-05	1,49E-05	6,82E-05	1,49E-05	2,30E-05	5,77E-08
11			2,80E-05	1,49E-05	4,05E-05	1,49E-05	2,23E-05	5,77E-08
12			2,78E-05	1,49E-05	4,19E-05	1,49E-05	1,27E-05	5,77E-08
13			2,12E-05	1,49E-05	**1,49E-05**	**1,49E-05**	**5,77E-08**	**5,77E-08**
14			2,13E-05	1,49E-05	1,49E-05	1,49E-05	5,77E-08	5,77E-08
15			**1,49E-05**	**1,49E-05**				
16			1,49E-05	1,49E-05				

TABLO IV
MSE PERFORMANCES OF WAVELET TRANSFORM FAMILIES

Wavelet Familiy		MSE Values				
		1st order	2nd order	3rd order	4th order	5th order
Daubechies	db1	4,22E-06	2,44E-06	1,02E-06	5,21E-07	**2,46E-07**
	db2	4,17E-06	2,02E-04	2,01E-04	2,01E-04	2,00E-04
	db3	4,15E-06	2,06E-06	3,83E-04	3,82E-04	3,82E-04
	db4	4,13E-06	1,83E-04	7,88E-04	7,87E-04	7,87E-04
	db5	4,11E-06	2,05E-06	9,93E-07	3,58E-04	3,58E-04
Symlet	sym1	4,22E-06	2,44E-06	1,02E-06	5,21E-07	**2,46E-07**
	sym2	4,17E-06	2,02E-04	2,01E-04	2,01E-04	2,00E-04
	sym3	4,15E-06	2,06E-06	3,83E-04	3,82E-04	3,82E-04
	sym4	4,14E-06	1,82E-04	7,90E-04	7,89E-04	7,89E-04
	sym5	4,14E-06	2,04E-06	9,99E-07	3,65E-04	3,65E-04
Coiflet	coif1	4,14E-06	2,06E-06	3,90E-04	3,89E-04	3,89E-04
	coif2	4,13E-06	1,82E-04	1,81E-04	6,86E-04	6,86E-04
	coif3	4,12E-06	2,06E-06	1,01E-06	4,99E-07	1,46E-04
	coif4	4,11E-06	1,78E-04	7,87E-04	7,87E-04	9,93E-04
	coif5	4,11E-06	2,06E-06	3,64E-04	9,52E-04	1,17E-03
Biorthogonal	bior1.1	4,22E-06	2,44E-06	1,02E-06	5,21E-07	**2,46E-07**
	bior1.3	4,19E-06	2,09E-06	5,44E-04	5,44E-04	5,43E-04
	bior1.5	4,16E-06	2,06E-06	1,01E-06	6,06E-04	6,06E-04
	bior3.1	3,49E-06	1,69E-04	1,68E-04	1,67E-04	1,73E-04
	bior3.3	3,48E-06	1,74E-04	7,49E-04	7,48E-04	7,48E-04
	bior3.5	3,44E-06	1,77E-04	1,76E-04	6,86E-04	6,86E-04

TABLE V
THE MSE VALUES OBTAINED AFTER THE TRANSMISSION OF THE 1ST LEVEL WT COMPRESSED SIGNAL FROM THE RICIAN CHANNEL

Wavelet Family		LBC (7,4)	LBC (15,8)	LBC (17,8)	LBC (23, 12)
		MSE (K=10)	**MSE (K=14)**	**MSE (K=14)**	**MSE(K=17)**
Db	db1	0,0013	4,03E-06	4,03E-06	1,56E-08
	db2	0,0013	3,98E-06	3,98E-06	1,54E-08
	db3	0,0013	3,95E-06	3,95E-06	1,51E-08
	db4	0,0013	3,93E-06	3,93E-06	1,52E-08
	db5	0,0013	3,90E-06	3,90E-06	1,51E-08
Sym	sym1	0,0013	4,03E-06	4,03E-06	1,56E-08
	sym2	0,0013	3,98E-06	3,98E-06	1,54E-08
	sym3	0,0013	3,95E-06	3,95E-06	1,51E-08
	sym4	0,0013	3,92E-06	3,92E-06	1,52E-08
	sym5	0,0013	3,91E-06	3,91E-06	1,50E-08
Coif	coif1	0,0013	3,92E-06	3,92E-06	1,52E-08
	coif2	0,0013	3,92E-06	3,92E-06	1,51E-08
	coif3	0,0013	3,92E-06	3,92E-06	1,51E-08
	coif4	0,0013	3,92E-06	3,92E-06	1,48E-08
	coif5	0,0013	3,92E-06	3,92E-06	1,50E-08
Biort	bior1.1	0,0013	4,03E-06	4,03E-06	1,56E-08
	bior1.3	0,0013	3,99E-06	3,99E-06	1,52E-08
	bior1.5	0,0013	3,97E-06	3,97E-06	1,52E-08
	bior3.1	0,0011	3,26E-06	3,26E-06	1,27E-08
	bior3.3	0,0011	3,24E-06	3,24E-06	1,24E-08
	bior3.5	0,0011	3,21E-06	3,21E-06	**1,23E-08**

TABLE VI
THE MSE VALUES OBTAINED AFTER THE TRANSMISSION OF THE 2ND LEVEL WT COMPRESSED SIGNAL FROM THE RICIAN CHANNEL

Wavelet Family		LBC(7, 4)	LBC(15, 8)	LBC(17, 8)	LBC(23, 12)
		MSE (K=7)	**MSE (K=9)**	**MSE (K=9)**	**MSE (K=9)**
Db	db1	7,28E-04	2,54E-06	2,54E-06	1,43E-08
	db2	8,12E-04	7,87E-05	7,87E-05	7,59E-05
	db3	7,37E-04	2,22E-06	2,22E-06	8,44E-09
	db4	7,97E-04	7,05E-05	7,05E-05	6,82E-05
	db5	7,32E-04	2,18E-06	2,18E-06	8,22E-09
Sym	sym1	7,28E-04	2,54E-06	2,54E-06	1,43E-08
	sym2	8,12E-04	7,87E-05	7,87E-05	7,59E-05
	sym3	7,37E-04	2,22E-06	2,22E-06	8,44E-09
	sym4	8,02E-04	7,26E-05	7,26E-05	6,98E-05
	sym5	7,26E-04	2,22E-06	2,22E-06	8,38E-09
Coif	coif1	7,30E-04	2,26E-06	2,26E-06	8,38E-09
	coif2	7,99E-04	7,16E-05	7,16E-05	6,89E-05
	coif3	7,32E-04	2,23E-06	2,23E-06	8,41E-09
	coif4	7,98E-04	7,13E-05	7,13E-05	6,90E-05
	coif5	7,32E-04	2,24E-06	2,24E-06	8,49E-09
Biort	bior1.1	7,28E-04	2,54E-06	2,54E-06	1,43E-08
	bior1.3	7,34E-04	2,22E-06	2,22E-06	9,29E-09
	bior1.5	7,32E-04	2,20E-06	2,20E-06	**8,32E-09**
	bior3.1	6,23E-04	6,33E-05	6,33E-05	6,17E-05
	bior3.3	6,33E-04	6,52E-05	6,52E-05	6,31E-05
	bior3.5	6,39E-04	6,67E-05	6,67E-05	6,48E-05

TABLE VII
THE MSE VALUES OBTAINED AFTER THE TRANSMISSION OF THE 3RD LEVEL WT COMPRESSED SIGNAL FROM THE RICIAN CHANNEL

Wavelet Family		**LBC (7,4)**	**LBC (15,8)**	**LBC (17,8)**	**LBC (23,12)**
		MSE (K=6)	**MSE (K=8)**	**MSE (K=8)**	**MSE (K=8)**
Db	db1	0,00047	1,55E-06	1,55E-06	5,95E-09
	db2	0,00053	5,62E-05	5,62E-05	5,49E-05
	db3	0,00055	7,54E-05	7,54E-05	7,35E-05
	db4	0,00064	1,69E-04	1,69E-04	1,67E-04
	db5	0,00048	1,52E-06	1,52E-06	5,00E-05
Sym	sym1	0,00047	1,55E-06	1,55E-06	5,95E-09
	sym2	0,00053	5,62E-05	5,62E-05	5,49E-05
	sym3	0,00055	7,54E-05	7,54E-05	7,35E-05
	sym4	0,00065	1,71E-04	1,71E-04	1,70E-04
	sym5	0,00047	1,50E-06	1,50E-06	3,18E-08
Coif	coif1	0,00056	7,65E-05	7,65E-05	7,52E-05
	coif2	0,00051	4,75E-05	4,75E-05	9,61E-05
	coif3	0,00047	1,52E-06	1,52E-06	5,89E-09
	coif4	0,00064	1,73E-04	1,73E-04	1,72E-04
	coif5	0,00055	7,38E-05	7,38E-05	2,71E-04
Biort	bior1.1	0,00047	1,55E-06	1,55E-06	5,95E-09
	bior1.3	0,00058	1,07E-04	1,07E-04	1,06E-04
	bior1.5	0,00047	1,51E-06	1,51E-06	**5,84E-09**
	bior3.1	0,00046	3,99E-05	3,99E-05	3,85E-05
	bior3.3	0,00058	1,51E-04	1,51E-04	1,49E-04
	bior3.5	0,00047	4,33E-05	4,33E-05	4,17E-05

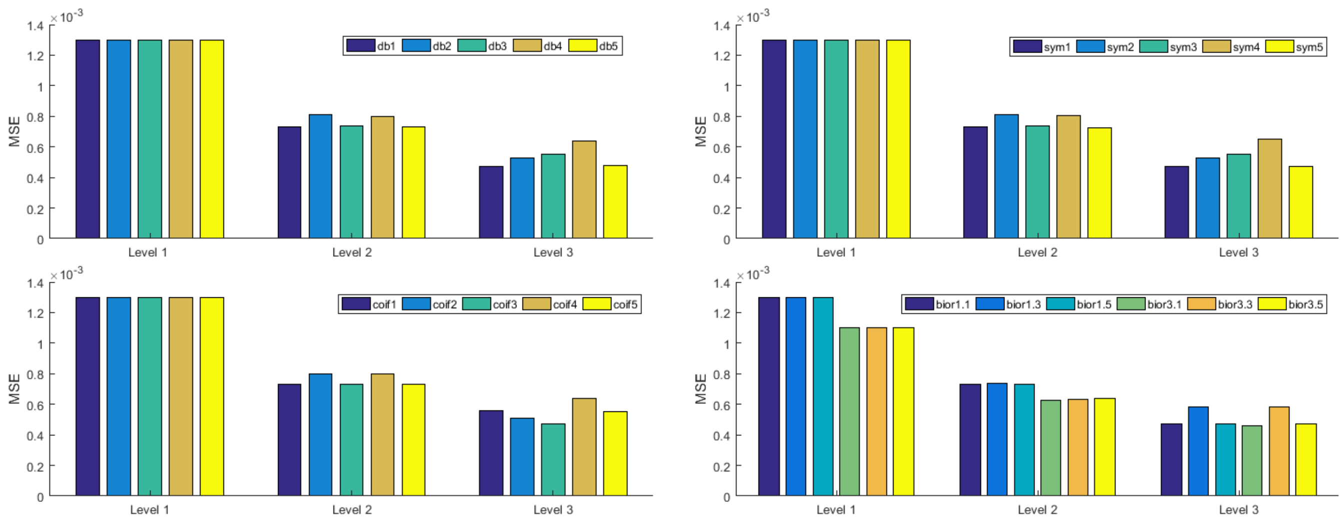


Fig. 12. Comparison of compression level and MSE for LCB(7,4) in Rician fading channel

REFERENCES

[1] N. Jacob, U. Sripati, “Bit Error Rate Analysis of Coded OFDM for Digital Audio Broadcasting System, Employing Parallel Concatenated Convolutional Turbo Codes”, in 2015 IEEE International Conference on Signal Processing, Informatics, Communication and Energy Systems (SPICES), IEEE, 2017. https://doi.org/10.1109/SPICES.2015.7091461

[2] S. Faruque, Introduction to Channel Coding. In: Radio Frequency Channel Coding Made Easy, Springer Briefs in Electrical and Computer Engineering. Springer, Cham, 2016.

[3] S. Dhaliwal, N. Singh, G. Kaur, "Performance analysis of convolutional code over different code rates and constraint length in wireless communication." I-SMAC (IoT in Social, Mobile, Analytics and Cloud)(I-SMAC), 2017 International Conference on. IEEE, 2017. https://doi.org/10.1109/I-SMAC.2017.8058393

[4] J. Baviskar, A. Mulla, M. Gulati, P. Vaswani, A. Baaviskar, "LDPC based error resilient audio signal processing for wireless communication." Pervasive Computing (ICPC), 2015 International Conference on. IEEE, 2015. https://doi.org/10.1109/PERVASIVE.2015.7087164

[5] T. Majoul, F. Raouafi, M. Jaidane, "Turbo decoding performance in non-Gaussian noise channels." Systems, Signals and Devices, 2009. SSD'09. 6th International Multi-Conference on. IEEE, 2009. https://doi.org/10.1109/SSD.2009.4956730

[6] Md D. Haque, S. E. Ullah, Md R. Ahmed, "Performance evaluation of a wireless Orthogonal Frequency Division Multiplexing system under various concatenated FEC channel-coding schemes." Computer and Information Technology, 2008. ICCIT 2008. 11th International Conference on. IEEE, 2008. https://doi.org/10.1109/ICCITECHN.2008.4802994

[7] K. K. Sidhu, G. Kaur. "Performance evaluation of multilevel linear block codes on Rayleigh fading channel." Electrical, Electronics, and Optimization Techniques (ICEEOT), International Conference on. IEEE, 2016. https://doi.org/10.1109/ICEEOT.2016.7755080

[8] H. Lee, "Exact and asymptotic BER analysis of 2× 2 FRLR-STBC with BPSK modulation." Electronics Letters. 53(11) 2017: 720-722. http://dx.doi.org/10.1049/el.2016.4194

[9] N. Romano, A. Scivoletto, D. Polap. "A real-time audio compression technique based on fast wavelet filtering and encoding." Computer Science and Information Systems (FedCSIS), 2016 Federated Conference on. IEEE, 2016.

[10] P. E. Kudumakis, M. B. Sandler. "On the performance of wavelets for low bit rate coding of audio signals." Acoustics, Speech, and Signal Processing, 1995. ICASSP-95., 1995 International Conference on. Vol. 5. IEEE, 1995. https://doi.org/10.1109/ICASSP.1995.479498

[11] G. Luo, "Ultra low delay wavelet audio coding with low complexity for real time wireless transmission." Intelligent Signal Processing and Communication Systems, 2005. ISPACS 2005. Proceedings of 2005 International Symposium on. IEEE, 2005. https://doi.org/10.1109/ISPACS.2005.1595516

[12] A. K. Sinha, S. K. Dutta, B. Dev, Jaydipta, R. Ranjan, R. Kumari, "Wavelet based Speech Coding technique using median function thresholding." Electronics and Communication Systems (ICECS), 2014 International Conference on. IEEE, 2014. https://doi.org/10.1109/ECS.2014.6892538

[13] S. S. Sarnin, N. F. Nairn, W. N. S. W. Muhamad, "Performance evaluation of phase shift keying modulation technique using BCH code, Cyclic code and Hamming code through AWGN channel model in communication system." Information Sciences and Interaction Sciences (ICIS), 2010 3rd International Conference on. IEEE, 2010. https://doi.org/10.1109/ICICIS.2010.5534715

[14] Bhatti, Ali Tariq. "Cyclic channel coding algorithm for original and received voice signal at 8 khz using ber performance through additive white gaussian noise channel"." International journal of innovative science, engineering & technology (IJISET), **2**(4), 843-852, 2015.

[15] Bhatti, Ali Tariq, and 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), **2**(8), 13-27, 2015.

[16] E. Göse, “Zamanla değişen kanallarda turbo kodlama*”*, PhD Thesis, Yıldız Technical University, Institute of Science and Technology, 2005.

[17] B. Sklar, Digital Communications Fundamentals and Applications Second Edition, Prentice Hall P T R Upper Saddle River, New Jersey 07458, 2001.

[18] J. Proakis, M. Salehi, Fundamentals of Communication Systems. Pearson Education, 2007.

[19] P. R. Kamala, R. V. S. Satyanarayana, "Optimal linear block code modeling and performance analysis over various channels for use in wireless communications", Emerging Trends in Engineering, Technology and Science (ICETETS), International Conference on. IEEE, 2016. https://doi.org/10.1109/ICETETS.2016.7603038

[20] B. Sklar, "Rayleigh fading channels in mobile digital communication systems. I. Characterization." IEEE Communications magazine 35(7), 90-100, 1997. https://doi.org/10.1109/35.601747

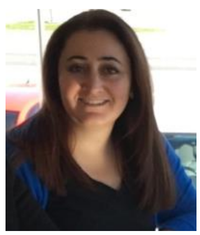
[21] G. L. Stüber, Principles of mobile communication. Norwell, Mass, USA: Kluwer Academic, 1996.

[22] A. Mahmutoğlu, “Uzay Zaman Blok Kodlarını Kullanan Röleli Sistemlerin Genelleştirilmiş Sönümlemeli Kanallardaki Hata Performans Analizi”, Diss. M. Sc. Thesis, İstanbul Technical University, Institute of Science and Technology, 2009.

[23] Adeli, Hojjat, Ziqin Zhou, and Nahid Dadmehr. "Analysis of EEG records in an epileptic patient using wavelet transform." Journal of neuroscience methods 123(1), 69-87, 2003. https://doi.org/10.1016/S0165-0270(02)00340-0

[24] A. Haşiloğlu, "Dalgacık dönüşümü ve yapay sinir ağları ile döndürmeye duyarsız doku analizi ve sınıflandırma" Turkish Journal of Engineering and Environmental Sciences (Turk J Engin Environ Sci.) 25, 405-413. 2001.

BIOGRAPHIES



SERAP KARAGOL received the B.E. degree in Computer Engineering from the Karadeniz Technical University, Trabzon, Turkey, in 2002, and the M.S. and Ph.D. degrees in electrical engineering from North Carolina A&T State University, Greensboro, NC USA, in 2006 and 2011, respectively. She is currently an Assistant Professor in the Department of Electrical and Electronics Engineering, Ondokuz Mayis University, Samsun Turkey. She is engaged in teaching and conducting research in circuits and systems, signal processing, modeling, system identification and wirelesss sensors networks. She is the advisor for more than 10 M.S. and 3 Ph.D. students.

DOGAN YILDIZ born in Istanbul in 1989. He received the B.S. degree in Electrical-Electronics engineering and Mathematics from Fatih University, Istanbul, in 2013 and the M.S. degree in electrical-electronics engineering from Ondokuz Mayis University, Samsun, in 2016.

Since 2013, he has been working as research assistant with the Electrical-Electronics Engineering Department in Ondokuz Mayıs University. His research interests include wireless sensor networks, localization, signal processing, telecommunication and applications, electromagnetic waves and electromagnetic radiation measurements.