© 2021 Little Lion Scientific
ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

STUDY OF THE PRINCIPLES OF ERROR CORRECTING CODE IN A MULTIPATH COMMUNICATION CHANNEL WITH INTERSYMBOL INTERFERENCE

MANARA A. SEKSEMBAYEVA[1], NURLAN N. TASHATOV[2], GENNADY V. OVECHKIN[3], DINA ZH. SATYBALDINA[4], YERZHAN N. SEITKULOV[5]

[1,2,4,5] L.N. Gumilyov Eurasian National University, Nur-Sultan, Kazakhstan
[3]Ryazan State Radio Engineering University named after V.F. Utkin, Ryazan, Russian Federation

E-mile: anuarkizi@gmail.com

ABSTRACT

The growth of modern wireless communications has increased the demand for high quality and reliable data services. The purpose of this article stands for a review of the principles and techniques of error correcting codes (ECC) and a study of their practical application to combat intersymbol interference (ISI) in multipath channels. A DVB-T system with MIMO technology developed in the Matlab/Simulink environment is introduced and analyzed. The bit error rate (BER) of the additive white Gaussian noise (AWGN) channel and an ISI-influenced multipath Rayleigh attenuation channel is measured and discussed. The method based on a DVB-T receiver-transmitter with MIMO technology provides a higher quality and speed of data transmission in comparison to a DVB-S2 system with MIMO. Other ways to improve the quality of data transmission are offered, including size and type converters, amplifiers, and gain normalization. Images are transmitted through the system that includes a Rayleigh channel and an AWGN with different signal-to-noise ratio values (SNR, dB) and different gain values. The BER graph is obtained; the results are discussed and compared with similar works.

The system can be used for transmission of radiological images (such as computed tomography or magnetic resonance imaging) of high quality and high resolution to remote healthcare workers using a wireless network.
Keywords: *Intersymbol Interference, MIMO, BER, PSNR, AWGN, SNR, Rayleigh Channel, LDPC Code, BCH Code, Amplifier, Eye Diagrams*

1. INTRODUCTION

The main tasks of error correcting code (ECC) is to control, detect and correct data errors that occur during transmission, processing or storage, in order to ensure reliable communication over unreliable channels [1, p.27].

According to the noisy-channel coding theorem, presented by Claude Shannon in 1948, any code is capable of detecting and correcting errors, if some of the bits in this code carry redundant information, i.e. used for ECC. This redundancy allows detecting a certain number of errors anywhere in the message, and correcting them without retransmission. Needless to say, that in some critical systems, retransmission may be costly, inappropriate or merely impossible.

The first error-correcting code was introduced by Richard Hamming in 1950. It was a linear code that adds three additional check bits to every four data bits of the message. Over the past 70 years, the error correcting techniques have undergone significant development; they are applied everywhere in computing, data storage, transmission protocols in networking, telecommunication, satellite broadcasting etc.

Good ECC should meet the following criteria: fix as many bugs as possible, provide the least feasible redundancy, and ensure easy encoding and decoding. These demands contradict each other; therefore, different conditions require different codes. When solving a real problem, it is often necessary to develop original codes based on priorities: either to

© 2021 Little Lion Scientific
ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

fight against errors, or boost the speed, or keep the simplicity and clearness.

2. THE PRINCIPLES OF ERROR CORRECTING CODE

According to the methods of data processing, error correcting techniques can be divided into block codes and convolutional codes. Block codes divide data into fragments of constant length and process each of them separately; while, convolutional codes treat data as a continuous stream [2, p. 115].

2.1. **Block Codes**

Consider a uniform block code with the radix q and the block size k; the ensemble of messages has the size of $N_p = q^k$. The block code maps the set of messages to $N = q^n$ possible codewords, where n is the number of characters in a codeword after encoding (also called the codeword length). The codewords from the set $N_p < N$, intended for the message transmission, are called the allowed codewords; the rest of them from the set $N_з = N - N_p$ are called the forbidden codewords (not allowed for the transmission) [2, p.12].

Suppose, due to distortions, the codeword $A_i (i = \overline{1, N_p})$ when transmitted in a communication channel, turns into one of the forbidden words $B_j (j = \overline{1, N_з})$; in this case, an error will be detected, since such a word cannot be transmitted (Figure 1). However, no error can be detected if, after transmission, the codeword is read as another allowed word.

Error correction is more difficult than error detection, since it is also necessary to locate the position of a distorted character (as well as the error value for non-binary codes). Correcting errors with ECC requires splitting a subset or the whole $N_з$ set into disjoint subsets $N_з^i (i = \overline{1, N_p})$, in accordance with the number of the allowed codewords. If the received codeword belongs to the subset $N_з^i$, then the allowed codeword A_i is considered as successfully transmitted.

In Figure 1(b), the error in the transmitted forbidden word B_{1j} will be fixed, since $B_{1j} \in N_з^1$, then the word A_1 will be received. However, the errors in the transmitted words B_{2j} и B_{4j} will be overlooked because these words refer to subsets of other allowed words. Only if the transmitted word falls into the subset of the forbidden words, which do not belong to any of the subsets of $N_з^i$, then the error will be detected, but not fixed. The correction is done by another method, for example, by the method of negative acknowledgement [4, p.14].

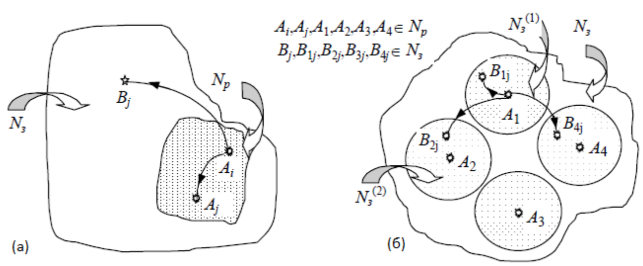

Figure 1 - Detection (a) And Correction (b) Of Errors [3, p.13]

Thus, in a block code, a group of k characters from the source is matched with n characters transmitted over the channel. Such a block code is denoted as a (n, k) code. When the code is transmitted without an excess in each word, $N_p = 2^k$ combinations are allowed, and an error just in one character turns one allowed combination into another, i.e., leads to an error in the message received. The introduction of redundant symbols increases the number of combinations to $N = 2^n$, where some of them $N_з = 2^n - 2^k$ are forbidden and can occur only in case of transmission errors in the communication channel.

To evaluate the difference between two code combinations, we use the Hamming distance. For binary code, it is defined as:

$$d(B_i, B_j) = \sum_{k=1}^{n} (b_{ik} \oplus b_{jk}) \quad (1)$$

where b_{ik} and b_{jk} are the k^{th} characters of the code combinations B_i and B_j, respectively. The smallest Hamming distance for a given code is called the code distance. In order for the code to detect a single error, it is obvious that the code distance must be $d = 2$. To detect g errors, the code distance of $d = g + 1$ is the least. In general, to detect and correct g errors, it is necessary and sufficient that the code distance satisfies the condition $d \geq 2g + 1$ [6, p. 7].

The greater is the code redundancy, the greater is the code distance between the allowed combinations. Otherwise, trying to improve the transmission time, we increase the probability of an

Journal of Theoretical and Applied Information Technology
30th September 2021. Vol.99. No 18
© 2021 Little Lion Scientific

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

error when receiving the signal. Thus, the problem of constructing a noise-resistant code is reduced to providing the necessary code distance d with the minimum possible number of the check characters $r = n - k$.

To determine the proximity of the code to the optimal parameters, a range of assessments are used. For example, this optimal interval for r:

$$\log_2\left(\sum_{i=0}^{\left\lfloor\frac{d-1}{2}\right\rfloor} C_n^i\right) \le r \le \log_2\left(\sum_{i=0}^{d-2} C_n^i\right) \quad (2)$$

where the logarithm on the left is the upper Hamming bound (the necessary condition for the code existence), the logarithm on the right is the lower Varshamov-Hilbert bound (the sufficient condition for the code existence) [16, p. 32]. The boundaries (2) allow evaluating the effectiveness of the block code and the feasibility of its application.

Block codes do a good job when dealing with rare, random and large errors. Their disadvantage is low efficiency with frequent and small errors (for example, in channels with white noise).

The family of block codes includes: Hamming codes, CRC codes (cyclic redundancy check), BCH codes, Reed-Solomon codes, McDonald's codes, Varshamov codes, Berger codes and many others.

2.2. **Convolutional codes**

Unlike block codes, convolutional codes process data as a continuous stream, without dividing it into fragments. At each clock cycle of the coder, the following happens [5, p. 224]:

– k characters of the input semi-infinite sequence are converted to $n > k$ characters of the output sequence;
– m previous characters also get involved in the conversion;
– the linear property is satisfied (if two encoded sequences $\boldsymbol{x}$ and $\boldsymbol{y}$ correspond to the code sequences $\boldsymbol{X}$ and $\boldsymbol{Y}$, then the encoded sequence $a\boldsymbol{x} + b\boldsymbol{y}$ corresponds to $a\boldsymbol{X} + b\boldsymbol{Y}$).

The general scheme of a non-recursive convolutional code is shown in Figure 2. It consists of k-ary shift registers with the lengths of $m_1, m_2, \dots, m_k$, Several (sometimes, all) register inputs and several memory locations are connected to multiple n modulo adders q. The number of adders is larger than the number of shift registers, $n > k$.

At each clock cycle of the coder, k characters enter the input and, together with the characters stored in the shift registers, are sent to the inputs of the corresponding adders. After adding, n code characters are ready for transmission.

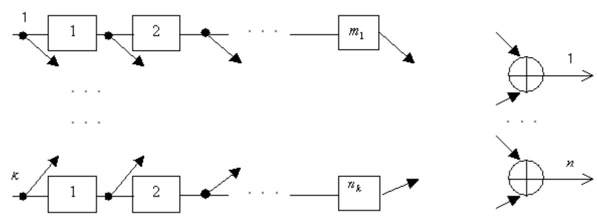

Figure 2 - General Scheme Of ECC Using A Non-recursive Convolutional Code [6, P.227]

In the end, a shift occurs in each register: all cells are shifted one bit to the right; the leftmost cells are filled with input characters, and the rightmost cells are wiped. Then, the cycle repeats.

The maximum length $m = \max\{m_1, \dots, m_k\}$ is called the code memory. The values of the shift registers at each moment of time are called the encoder state [5, p. 224]. The convolutional code with memory m and speed $1/n$ can be described by a state diagram with the total number of states equal to 2^m (Fig.3).

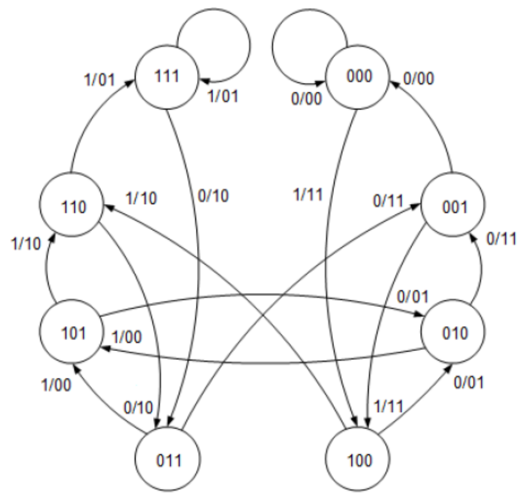

Figure 3-State Diagram Of A Convolutional Encoder With Memory M=3 And Speed R=1/2 [8, P. 89]

Thus, the development of an iterative convolutional coding algorithm is a relatively simple operation. At the same time, the disadvantage of most methods of convolutional coding is rapid increase in complexity with iterations. To overcome this problem, non-binary cyclic algorithms are used to construct non-recursive convolutional codes. Many solutions are based on recursive convolutional codes as parts of more complex turbo codes [17, p. 168].

Three methods are used to decode convolutional codes: the threshold decoding method, the sequential decoding method, and the maximum likelihood method (Viterbi algorithm).

In the threshold decoding, syndromes (error signs) are calculated; their linear convolution is fed to the inputs of the threshold element, where a decision is made on the value of the decoded symbol using a "vote" (the majority method) and comparing the result with the threshold value. A convolutional code that can be decoded by the threshold method must be orthogonal and systematic [9, p. 50]. The advantage of this decoding method is the simplicity, both of the algorithm and the devices implementing it.

The sequential decoding method is probabilistic; the number of operations required to decode a single character is a random variable. Convolutional codes can be represented using a tree, where each sequence of characters corresponds to a certain path. The task of the decoder is to find the true path that was actually generated by the encoder. The method involves searching the code tree for a path from the m-th edge (m is the number of bits of the encoder shift), with the corresponding code sequence located at a certain Hamming distance (1) from the transmitted sequence, and decoding the 1st transmitted character (the 1st edge of the path found). Then the 2nd, 3rd and all subsequent characters of the original message are decoded; that is why, the algorithm is called the sequential decoding.

Decoding convolutional codes with the Viterbi algorithm tries to restore the transmitted sequence according to the maximum likelihood criterion [1, p.201]. The algorithm is based on the following assumptions:

– hidden and observed events create a sequence that is ordered in time;
– each hidden event corresponds to only one observed event;
– the calculation of the most probable hidden sequence up to the moment t depends on the observed event at this moment and the most probable sequence up to the moment $t-1$ only.

The latter assumption is similar to the division of the final problem into subtasks in dynamic programming.

We introduce the following notation:

$\mathrm{O} = \{\mathrm{o}_1, \mathrm{o}_2, \ldots, \mathrm{o}_\mathrm{N}\}$ – the space of observations

$\mathrm{S} = \{\mathrm{s}_1, \mathrm{s}_2, \ldots, \mathrm{s}_\mathrm{K}\}$ - the space of states

$Y = \{\mathrm{y}_1, \mathrm{y}_2, .., \mathrm{y}_\mathrm{T}\}$ - the sequence of observations

$A = (\mathrm{a}_{\mathrm{ij}}), \mathrm{K} \times \mathrm{K}$ - the transition matrix from the i-th to j-th state

$B = (\mathrm{b}_{\mathrm{ij}}), \mathrm{K} \times \mathrm{N}$ – the emission matrix, the probability of o_j to be observed from the state s_i

$X = \{\mathrm{x}_1, \mathrm{x}_2, .., \mathrm{x}_\mathrm{T}\}$ - the sequence of states that led to the sequence of observations Y

Thus, we can write the following pseudocode for the Viterbi algorithm:

Journal of Theoretical and Applied Information Technology
30th September 2021. Vol.99. No 18
© 2021 Little Lion Scientific
ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

```
Viterbi(O, S, P, Y, A, B)
    for j = 1 to K
        TState[j, 1] = P[j] * B[j, Y[1]]
        TIndex[j, 1] = 0
    for i = 2 to T
        for j = 1 to K
            TIndex[j, i] = k ∈ K : (TState[k, i − 1] * A[k, j] * B[j, Y[i]]) ⇝ max
            TState[j, i] = TState[TIndex[j, i], i − 1] * A[k, j] * B[j, Y[i]]
    X[T] = arg max_{1⩽k⩽K} (TState[k, T])
    for i = T downto 2
        X[i − 1] = TIndex[X[i], i]
    return X
```
(3)

where
$P = (p_i), 1 \times K$ - the probability vector of the initial state s_i
$TState = (TS_{ij}), K \times T$ - the probability matrix that the state $s_i$occurs at the j-th step
$TIndex = \left(TI_{ij}\right), K \times T$ - the matrix of indices of the most probable states at the $(j-1)$-th step

The function (3) returns the vector X - a sequence of numbers of the most probable states that lead to these observations.

The computational complexity of the algorithm is $O(T \times K^2)$.

With the Viterbi algorithm, the two techniques are applied: decoding with a "hard" solution (HOVA - Hard Output Viterbi Algorithm) and decoding with a "soft" solution (SOVA - Soft Output Viterbi Algorithm) [17, p. 186]. In the first case, the metric for making a decision is the Hamming distance (1); in the second case, the ordinary Euclidean distance is used. To choose a particular solution, you should take into account the noise statistics in the communication channel. Decoding with a "hard" solution is based on a well-developed theoretical basis, which guarantees that a certain number of errors can be detected for a given code. On the other hand, the expediency of using a "soft" solution may be dictated by the continuous (not discrete) nature of the noise in the channel.

The advantage of convolutional codes is their efficiency in channels with white noise; however, they poorly handle a series of errors. In addition, if the decoder is wrong, a series of errors also appears at its output. Hence, in practice, convolutional codes are used together with block codes, for example, with the Reed-Solomon code.

2.3. **Serial Cascade Codes**

Combining the advantages of block and convolutional codes is achieved in cascade codes.

In the scheme of serial cascade coding, two codecs optimized for correcting errors of different structures are linked up successively.

For example, the transmitted data can be first processed by the Reed-Solomon code; then, it is interleaved so that close characters are located far from each other; and finally, the data is encoded by a convolutional code. On receiving, the convolutional code is decoded first, then the back-interleaving is implemented (so that the series of errors at the output of the convolutional coder fall into different code words of the Reed-Solomon code), and finally, the Reed-Solomon code is decoded (Fig. 4) [2, p. 327].

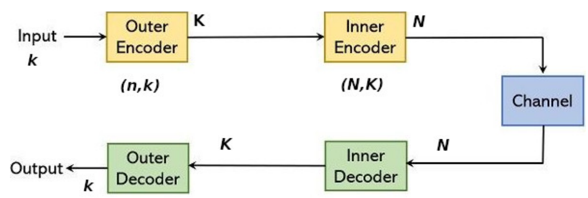

Figure 4 – Structure Scheme Of A Communication Channel Using A Serial Cascade Code

Let the initial sequence consist of K_{inf} binary characters. It is divided into k external sub-blocks of K characters each. The binary Galois field of the degree of extension K forms a group of $q = 2^K$ information symbols of the external code. The check characters are formed by the external code based on k. For the RS code, the corrective capabilities are determined by the equation $d = n - k + 1$. All q-point combinations of the (n, k) code are encoded by the internal (N, K) code. As a result, we get a binary block code of the length $n \times N$, which contains $k \times K$ binary characters of the message with a total minimum distance $d \times D$, where D is the minimum distance of the internal code.

Thus, the decoding of a long (n, N, k, K) code can be replaced by decoding two significantly shorter codes: the internal binary (N, K) code and the external (n, k) code. Depending on the multiplicity of the errors corrected, the decoder complexity increases linearly, which is an undeniable advantage of cascade codes. Another advantage is the ability to use not only constructive methods, but also optimal iterative methods to correct errors with the internal code, if (N, K) is a relatively low-power code [8, p.104].

The advantage of cascade codes is their high correction ability; the disadvantage is a significant redundancy of data. To reduce the redundancy of a sequential cascade code, high-speed and low-redundant convolutional codes are recommended for use as the internal codes.

Journal of Theoretical and Applied Information Technology
30th September 2021. Vol.99. No 18
© 2021 Little Lion Scientific
ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

2.4. **Turbo Codes**

Some cascade codes are based on iterative decoding, performed in several steps; each step uses the information gathered from the previous iteration. It provides greater efficiency, but requires more resources. These codes are: turbo codes and LDPC codes (Gallager codes) [5, p.412].

A turbo code is a parallel cascade block systematic code capable of correcting errors that occur when transferring digital data over a noisy communication channel [6, p.72]. Convolutional codes, Hamming codes, Reed-Solomon codes, Bose-Chawdhuri-Hocquenghem codes, and others can be used as component codes. Depending on the choice of component code, turbo codes are divided into turbo convolutional codes (Turbo Convolutional Codes, TCC) and block turbo product codes (Turbo Product Codes, TPC) [17, p. 214]. For decoding, the algorithm of the maximum a posteriori probability (Maximum of A-posteriori Probability, MAP) is used [6, p.73].

Turbo encoders based on recursive systematic convolutional codes (RSC) have been widely used in information transmission systems [19, p. 439]. A recursive convolutional code is a type of convolutional codes in which the input symbols are transmitted to the encoder, and check (redundant) symbols are generated by a logical chain containing a shift register with feedback (Figure 5).

In the interleaver of the turbo encoder, according to a pseudo-random law, the incoming bits are mixed, similarly to random permutations [17, p. 227]. Then, during the decoding operation, the law of interleaving is considered to be known. The task of the interleaver is to transform the original sequence so that the combinations of bits corresponding to code words with a low weight (the weight is the number of non – zero bits) at the output of the first encoder are converted into combinations giving codewords with a high weight at the outputs of the remaining encoders. Thus, the coders receive the codewords with different weights at the output. The codewords are formed so that the average Hamming distance (1) between them tends to the maximum, which increases the coding efficiency.

Owing to the use of two systematic convolutional encoders, the message part and the check part can be clearly distinguished in the code block. The following two code blocks are transmitted over the communication channel: the first code block with the message and the check parts of the upper encoder; the second code block with the mixed message and check parts of the lower encoder.

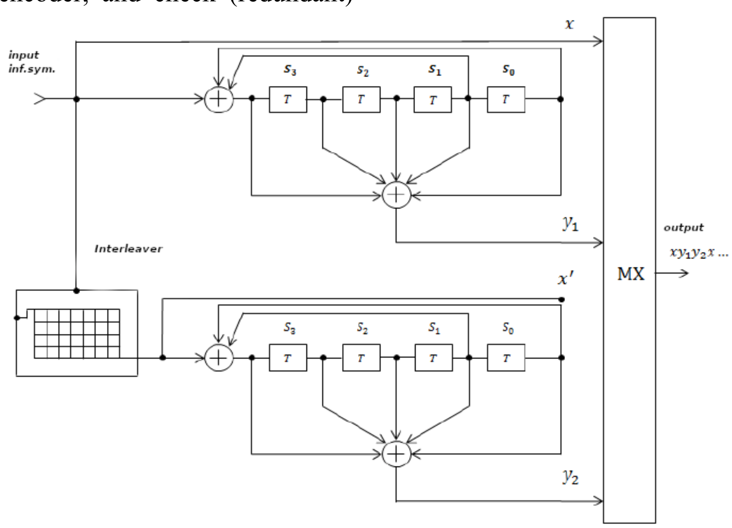

Figure 5 - Generalized Functional Diagram Of A Turbo Encoder Based On 2 RSCs [10, P. 9]

When decoding, the code block can be divided again into two code parts, where the message parts are identical up to the interleaving. Therefore, two decoders can be used. The decoded message of the first (second) decoder will serve as a priori to clarify the decoding result from the second (first) decoder. The refinement operation can be performed repeatedly, which is the essence of iterative decoding (Fig. 6).

© 2021 Little Lion Scientific

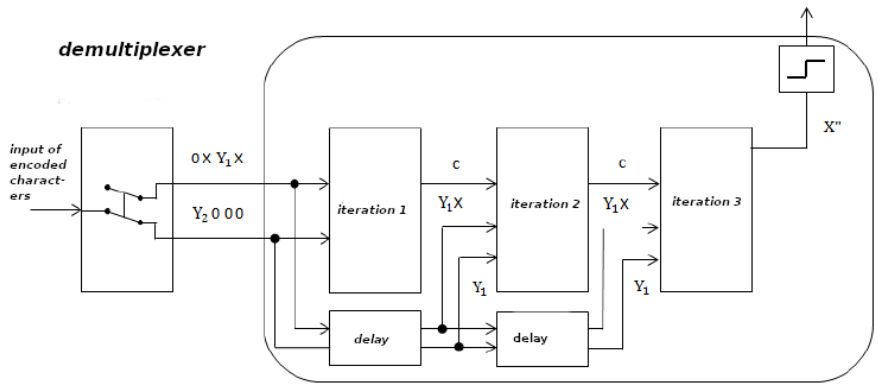

Figure 6 - Channel Turbo Decoder With Three Decoding Iterations [10, P. 20]

The end of the decoding process occurs either after the specified number of iterations is completed, or after the acceptable value of the decoding error probability is reached. The value of the decoding error probability depends on the length of the interleaving interval, the number of iterative cycles, and the decoding algorithms used.

The following probabilistic decoding algorithms are most often used in channel turbo decoders [17, p. 235]:

- MAP (Maximum A Posteriori) - the maximum of a posteriori probability;
- Log-MAP – the logarithmic maximum of a posteriori probability;
- Max-Log-MAP – the maximum of the logarithmic maximum of a posteriori probability;
- SOVA (Soft Output Viterbi Algorithm) - the soft Viterbi decoding algorithm.

The computational complexity of the turbo decoder does not depend on the length of the message block and is comparable to the complexity of the Viterbi algorithm (3) for convolutional codes [10, p.36].

2.5. Gallagher Codes

LDPC code is a kind of block linear codes with parity check. Its feature is the low density of significant elements in the check matrix, which greatly simplifies the implementation of coding tools. This approach was proposed by R. Gallagher in 1963, so LDPC codes are also called Gallagher codes.

An LDPC code is called an (n, J, K) code if its check matrix H of dimension $(n - k) \times n$ has J ones in each column and K ones in each row.

Since $K\ (n-k) = Jn$, the code transmission rate is given by the equation:

$$R = 1 - \frac{J}{K} \tag{4}$$

In his work [11], Gallagher used codes with the block length equal to 126, 252, 504, 1008 bits, and values J=3 or 4, K=6 or 9, at a code transfer rate $\frac{k}{n} = \left\{\frac{1}{3}; \frac{1}{2}; \frac{2}{3}\right\}$.

It is proven that LDPC codes provide an exponential decrease in the error probability with an increase in the code length, and a logarithmic increase in the number of operations required to decode a single character of the codeword [16, p. 128].

Currently, two methods are used to construct the parity-check matrix for an LDPC code. The first is based on generating the initial parity-check matrix using a pseudo-random generator; the codes obtained in this way are called random-like codes. In the second case, certain transformations based on the theory of groups and finite fields are used; these codes are called structured codes. Random-like codes resolve errors better, but structured codes allow you to optimize storage, encoding and decoding procedures, as well as to obtain more predictable results [7].

A thorough study of the LDPC and BCH codes intended to combat ISI in multipath channels can be found in [12]. A DVB-S2 system with MIMO technology was developed in the Matlab/Simulink environment and used in that research.

An important feature of the check matrix is the absence of loops of a certain length. A loop of length 4 is understood as a rectangle in the check matrix with "1" in the corners. Loops of longer length (6, 8, 10, etc.) correspond to a graph whose vertices are ones, and whose edges are horizontal and vertical lines connecting the vertices. The presence of loops reflects the connectivity, i.e. the dependence of the check vectors that make up the rows of the check matrix. Special methods have been developed and applied to find and remove loops of minimum lengths in the check matrices of LDPC codes [18, p. 587].

Compared to turbo codes, LDPC codes are more complicated to encode, but in some aspects are easier to decode. Although iterative decoding methods are also used here, they can be performed in parallel, which reduces the complexity of the decoder and increases its speed [17, p.307].

The decoding is based on the orthogonality of the generating matrix G and the transposed check matrix H:

$$G \odot H^T = 0 \qquad (5)$$

where $\odot$ is the modulo 2 multiplication.

Then, for each codeword received without errors, the following statement is valid:

$$s = v \odot H^T = 0 \qquad (6)$$

and, for a codeword received with an error:

$$s = v \odot H^T \neq 0 \qquad (7)$$

where v is the received vector, s is the syndrome.

Thus, if condition (6) is met, the block is considered accepted without errors. Otherwise, special methods are used to determine the location of the error and correct it. The exact determination of the error location is an NP-complete problem and does not apply. Instead, a probabilistic iterative decoding method is used, correcting most of the errors beyond half of the code distance. If the check matrix H does not contain loops, then the iterative decoding converges to the exact value x of the original message.

In conclusion, we should note that modern data transmission and storage systems require the development and implementation of new codes that possess not only high correction rate, but also the ability to optimize high-speed data flows. Therefore, the development of error correcting codes is moving towards increasingly complex algorithms, gradually approaching the Shannon limit in efficiency.

3. RESEARCH METHODOLOGY

Improving the reliability of signal reception in digital television also requires the use of error correcting codes. The digital video broadcasting project (DVB) was founded in 1993 to create a framework for digital video services [19, p. 444]. DVB-T, short for "digital video broadcasting – terrestrial", is the standard that was first published in 1997.

According to the DVB-T standard, one of the schemes for the traffic and control channel is a serial cascade code with the Reed-Solomon code as the external part, and a non-systematic convolutional code as the internal part.

To explore the benefits of combining a block code and a convolutional code, a DVB-T system with MIMO technology was designed in the Matlab/Simulink environment (Figure 7).

A DVB-T-based receiver-transmitter with MIMO technology provides high-quality and high-speed data transmission, and supports different means to improve the results, namely: converters of the size and type, amplifiers, and gain normalization.

Also, the advantage of the DVB-T system over DVB-S2 is the use of OFDM technology (orthogonal frequency-division multiplexing) for transmission and encoding digital data on multiple carrier frequencies. For more information about OFDM technologies, see [13; 20].

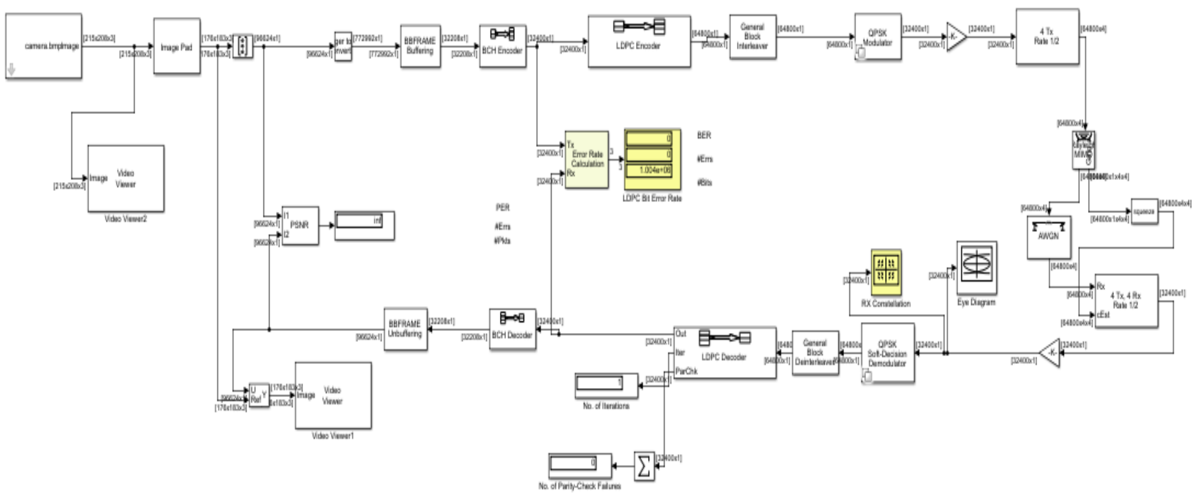

Figure 7. Block diagram of DVB-T transmitter-receiver with MIMO developed in Simulink

4. RESULTS AND DISCUSSION

In this paper, we test the hypothesis that using the additional means of improvement - converters of the size and type, amplifiers, and gain normalization – we can decrease the ISI. Then, using 1) the DVB-T system with the cascade error correcting codes; 2) the MIMO antenna to additionally mitigate the ISI, - we can receive a high-quality image at rather poor values of SNR.

The impact of the intersymbol interference (ISI) was investigated for image transmission over the DVB-T system that includes a Rayleigh channel and an AWGN with different signal-to-noise ratio values (SNR, dB) and different gain values. The bit error rate (BER) was evaluated in accordance with the data received.

The results for two amplifier factors (k = 1 and k = 648000) are gathered in Table 1 and illustrated with the plot in Figure 9.

When designing mobile devices for data transmitting and receiving, the goal is to minimize the ISI impact and thus deliver digital data to its destination with the lowest possible error rate. In case of images, delayed or distorted signals can be seen as spots of brightness and colour, specks of different size, and other glitches. The ISI induced in the model system causes errors in the decision making device at the receiver output.

Table 1. The Bit Error Rate (BER) vs. The Signal-To-Noise Ratio (SNR) For Two Amplifier Factors (K)

SNR (dB)	BER
K = 648000	
-40	0.4479
-37	0.2614
-34	7e-4
-33	0
K = 1	
75	0.4742
77	0.4235
80	0.1667
82	0.005787

In Figure 8, the images received after transmission over the DVB-T system with different values of the SNR are shown. As can be seen, the quality of reception improves with the growth of the SNR. At SNR = -33 dB, the original and received images become identical, BER ≈ 0.

Therefore, due to the amplifier and converters, the cascade error correcting codes of the developed DVD-T system overcomes the rest of ISI and improves the reception.

Journal of Theoretical and Applied Information Technology
30th September 2021. Vol.99. No 18
© 2021 Little Lion Scientific
ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

Original image

Image received at *SNR* = -37

Image received at *SNR* = -35

Image received at *SNR* = -33

Figure 8. Original Image And Images Received After Transmission At Different Values Of SNR

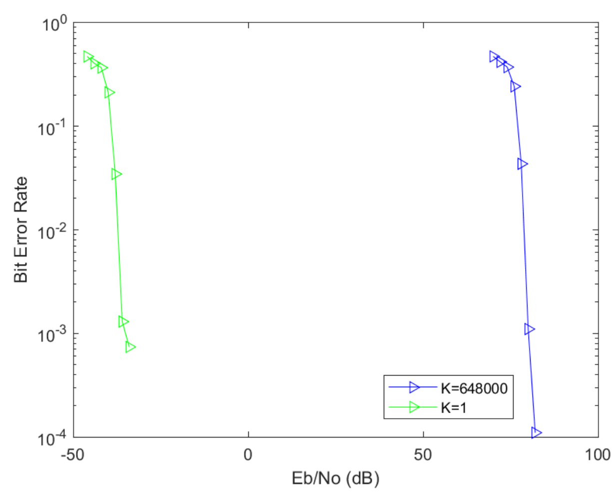

Figure 9. Bit Error Rate In Rayleigh And Gaussian Channels For Different Amplifier Factors K

Comparing out results with the paper [14], we should notice that the high amplifier factor and the cascade error correcting codes allow us to receive an undistorted image at a lower SNR. In [14], the authors describe all the functional blocks of a DVB-T transmitter and receiver and present a simulation model of the DVB-T system in accordance with ETSI EN 300 744 V1.6.1 in the Simulink environment. The implemented model contains all channel coding and modulation building blocks with the parameters: 8K

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

OFDM-mode, code rate (1/2, 2/3, and 3/4), and modulation (64-QAM, 16-QAM). The performance analysis of the video transmission system in the AWGN channel with only Gaussian noise is performed, and BER ≈ 0 is reached at the value of 10 < SNR<15 dB.

In our work, we consider a Rayleigh channel, and, due to the amplifier factor, the size and type convertors, ease the task for the cascade error correcting codes, and get to BER ≈ 0 at SNR = -33 dB, which is a better result.

In [15], a simulation of ECG signals transmission over a communication system using DVB-T technologies was performed, and the BER vs. the SNR was estimated. In order to analyze the improvement achieved using MIMO, the performance of ECG transmission using DVB-T technology over the fading channel is compared with the performance with the MIMO included. In this paper, the value of BER ≈ 0 is obtained at SNR = 20 dB.

In our system, we also use the MIMO antenna technology due to its mitigation effect on intersymbol interference. As already stated, the improvements of the DVB-T system allow us to achieve BER ≈ 0 at a lower SNR.

5. CONCLUSION

The development of modern communication networks primarily depends on the development of error correcting codes, both in theoretical and applied aspects. Historically, each of the offered schemes has had its own advantages and disadvantages. When introducing new communication standards, ECC schemes are selected based on the tasks assigned to the communication system, on the set of its criteria and priorities.

In data transmission systems, where maximum transmission rate with limited bandwidth is required, it is efficient to use LDPC codes. BCH and LDPC codes are more efficient than cascades of RS and convolutional codes, their use allows to reduce the signal-to-noise ratio for reliable system operation. With LDPC and BCH, gains of several decibels can be reached with the same signal/noise. But on the other hand, the LDPC and BCH codes require more efficiency, first of all, from the receiving equipment, which leads to a higher cost of receivers and decoders.

In cases where lower transmission latency is important, MIMO technology can be used together with DVB-T to mitigate the effects of multipath and improve the performance.

In this work, using 1) the DVB-T system with the cascade error correcting codes; 2) the MIMO antenna to mitigate the ISI; 3) additional means of improvement, such as converters of the size and type, amplifiers, and gain normalization, to decrease the ISI even more, - we received a high-quality image at rather poor values of SNR. The obtained results met the expectations and were better in comparison to similar studies. Thus, the hypothesis of the work was proven to be correct.

The proposed system can be used to transmit high-quality and high-resolution radiological images (such as computed tomography or magnetic resonance imaging) wirelessly to remote medical workers.

One of the future research directions is application of similar approaches for data transmission over hydroacoustic channels.

In addition, the developed model can be used as a training tool for studying various features of digital broadcasting systems.

REFERENCES

[1] Rao K.D. Channel Coding Techniques for Wireless Communications. - Springer - 2015 - 407 p. - ISBN 978-81-322-2292-7
[2] Declerq D., Fossorier M., Biglieri E.(eds.) Channel Coding: Theory, Algorithms, and Applications. - Elsevier - 2014 - 667 p. - ISBN 978-0-12-396499-1
[3] Makarov A.A., Pribylov V.P. [Pomekhoustoichivoe kodirovanie v sistemah telekommunikatsiy] Noise-resistant coding in telecommunication systems: Textbook / Sib. state University of Telecommunications and Informatics. — 2nd ed., Stereotype. — Novosibirsk, 2017. — 140 p.
[4] Luzin V.I., Nikitin N.P., Gadzikovsky V.I. [Osnovi formirovaniya, peredachi i priema tsifrovoy informatsii] Fundamentals of the formation, transmission and reception of digital information. — M .: Solon-Press, 2014 . — 320 p. — ISBN 978-5-321-01961-0
[5] Goldsmith A. [Besprovodnie kommunikatsii] Wireless communications. [Osnovi teorii i tehnologii besprovodnoy sviazi] Fundamentals of theory and technology of wireless

© 2021 Little Lion Scientific

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

communication. — M .: Technosphere. 2011. — 904 p. – ISBN 978-5-94836-176-5

[6] Justesen J., Høholdt T. A Course In Error-Correcting Codes. — Zurich: European Mathematical Society, 2017. — 228 p. — ISBN 978-3037191798

[7] Nguyen Tram Thi Bao, Tan Tuy Nguyen, Lee Hanho. Efficient QC-LDPC Encoder for 5G New Radio. // Electronics 2019, 8(6), 668; https://doi.org/10.3390/electronics8060668

[8] Gladkikh A. A. Fundamentals of the theory of soft decoding of redundant codes in an erasing communication channel. - Ulyanovsk: UlSTU, 2010. - 379 p.

[9] Nikitin G. I. Convolutional codes. - Pb.: SPbGUAP, 2001. - 80 p.

[10] Korolev A. I. Turbo codes and iterative decoding. - Minsk : BSUIR, 2015. - 74 p.

[11] Gallagher R. Codes with a low density of parity checks. - Trans. from English-M.: Mir, 1966 — - 144 p.

[12] Seksembayeva M. A., Tashatov N.N., Ovechkin G.V., Satybaldina D.Zh., Seitkulov Y.N., Turkish Journal of Computer and Mathematics Education Vol.12 No.10 (2021), 2015-2024

[13] Ovechkin G.V., Seksembayeva M. A., Tashatov N.N., ESTIMATION OF THE BIT ERROR PROBABILITY FOR OFDM WITH M-PSK MODULATION IN MATLAB// ACTUAL PROBLEMS OF MODERN SCIENCE AND PRODUCTION, materials of the V All-Russian Scientific and Technical Conference Ryazan State Radio Engineering University named after V. F. Utkin (25-27 november 2020)//p.173-178

[14] Mahmoud Elsharief, AbdelHaliem Zekry, Mohamed Abouelatta. Implementing a Standard DVB-T System using MATLAB Simulink// International Journal of Computer Applications (0975 – 8887) Volume 98– No.5, July 2014

[15] Waleed Alsaadia , Ali Serenera. Use of DVB-T and DVB-S2 in telecardiology// Procedia Computer Science 120 (2017) 99–103

[16] 16. Ball S. A Course in Algebraic Error-Correcting Codes. - Birkhäuser, 2020. — xiii, 177 p. — (Compact Textbooks in Mathematics). — ISBN 978-3-030-41152-7

[17] 17. Berrou C. Codes and Turbo Codes. - Springer-Verlag France, Paris - 2010 г. - 434 с. - ISBN : 978-2-8178-0038-7

[18] 18. Cancellieri G. Polynomial Theory of Error Correcting Codes. - Springer, 2015. — 736 p. ISBN 978-3-319-01727-3 (eBook)

[19] 19. Schlegel C.B., Perez L.C. Trellis and Turbo Coding. Iterative and Graph-based Error Control Coding. - New York: Wiley-IEEE, 2014. — 507 p. - ISBN 978-1-118-083161

[20] 20. Schulze H., Luders C. Theory and Applications of OFDM and CDMA. - John Wiley & Sons Inc., 2005. — 421 p. — ISBN: 978-0-470-85069-5.