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Digital Data Transmission System Over the Gaus Noise Channel using a 16-QAM Modulator, Hamming Source Encoding, and Turbo Channel Encoding

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A B S T R A C T

In this study, the author built and evaluated the performance of a digital data transmission system over a Gaussian noise channel. This system uses components like the 16-QAM modulator and demodulator, hamming source encoding and decoding, and turbo channel encoding and decoding. The 16-QAM modulator is a digital signal modulation method that allows multiple bits to be transmitted through a single symbol. It helps optimize bandwidth and transmission speed. Hamming source encoders help improve noise immunity by detecting and correcting bit errors during transmission. This ensures data integrity. By using independent codes and combining information from received signals, Turbo channel coding improves channel performance. The performance evaluation of the system is conducted through bit error rate calculation and analysis. The results show that the proposed system has the ability to improve transmission performance and anti-interference ability.

Keywords: Digital data transmission, Gaus noise, 16-QAM modulator, hamming source coding, turbo channel coding.

1. Introduction

Digital information systems are swiftly replacing their traditional counterparts on a global scale, driven by the need to cater to diverse service requirements, ensure quality, diversify offerings, and reduce costs, particularly amidst surging user numbers. To address these challenges, researchers have integrated a plethora of advanced techniques, prominently featuring digital modulation techniques. Among these techniques, quadrature amplitude modulation (QAM) stands out as a pivotal tool in digital telecommunications. QAM facilitates the efficient transmission of digital signals over transmission channels with high bandwidth. By amalgamating amplitude modulation (AM) and phase modulation (PM) techniques, QAM enables the seamless transmission of digital data across channels, offering efficiency and flexibility. This method finds widespread application across various digital communication systems, including cable networks, mobile networks, the Internet, digital television, and numerous other telecommunications systems.

2. Functional blocks of a digital data transmission system

2.1. The 16-QAM modulator and demodulator

M-QAM modulation is an advanced technology that increases channel efficiency without having to increase transmission power or expand bandwidth.

This modulation process combines two signal components, each with an independent phase and amplitude, creating a new system called quadrature amplitude modulation. QAM is a modulation system in which the carrier wave is modulated in both amplitude and phase. The general form of QAM modulation is defined as follows:

$$S_1(t) = \sqrt{\frac{2E_0}{T}} a_i \cos(2\pi f_c t) - \sqrt{\frac{2E_0}{T}} b_i \sin(2\pi f_c t);\ (0 \le t \le T)$$

In there,

E_0 is the energy of the signal with the lowest amplitude.

a_i, b_i is independent pairs of integers are chosen depending on the message position; i = 1, 2,..., L.

The basic form of the M-QAM signal burst is that of two types of ASK signals with L states. So, the signal $S_i(t)$ consists of two carrier wave components with perpendicular phase modulated by a pair of independent, discrete signals, so it is called perpendicular amplitude modulation.

$S_i(t)$ can be decomposed into a pair of basis functions:

$$\phi_1(t) = -\sqrt{\frac{2}{T}}.b_i \sin(2\pi f_c t) \qquad 0 \le t \le T$$

$$\phi_2(t) = -\sqrt{\frac{2}{T}}.a_i \sin(2\pi f_c t) \qquad 0 \le t \le T$$

The coordinates of the message points are $a_i\sqrt{E_0}$ and $b_i\sqrt{E_0}$ with:

$$(a_i,b_i) = \begin{bmatrix} (-L+1,L-1) & (-L+3,L-1) & (L-3,L-1) & (L-1,L-1) \\ (-L+1,L-3) & (-L+3,L-3) & (L-3,L-3) & (L-1,L-3)\_ \\ (-L+1,-L+3) & (-L-3,-L-3) & (L-3,-L+3) & (L-1,-L+3) \\ (-L+1,-L+1) & (-L+3,L-3) & (L-3,-L+1) & (L-1,-L+1) \end{bmatrix}$$

For 16-QAM we have L = 4:

$$(a_i,b_i) = \begin{bmatrix} (-3,3) & (-1,3) & (1,3) & (3,3) \\ (-3,1) & (-1,1) & (1,1) & (3,1) \\ (-3,-1) & (-1,-1) & (1,-1) & (3,-1) \\ (-3,-3) & (-1,-3) & (1,-3) & (3,-3) \end{bmatrix}$$

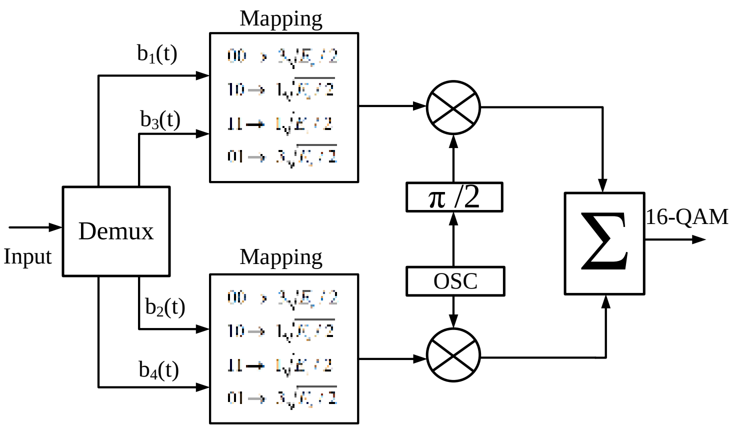

Fig.1. Diagram of the 16-QAM modulator

Modulator operation: The Demux streamer converts the input binary stream b(t) into four independent streams, in which two odd bits are fed to the level converter in the upper branch and two even bits are fed to the lower branch level converter. Level converters convert 2 levels into L levels, generating L-level signals corresponding to the 0 phase and $\pi/2$ phase inputs. After receiving two L-level signals with two carriers with perpendicular phase generated from the OSC internal oscillator and then adding them together, we get a 16-QAM signal.

Conversely, on the receiver side, the 16-QAM received signal is fed into two branches of phase 0 and phase $\pi/2$, then multiplied by the same two orthogonal functions as the transmitter side generated from the internal oscillator. Thanks to the orthogonal property, the two signal components can be separated.

2.2. Source encoding and decoding blocks

The proposed system uses Hamming encoding and decoding methods. Hamming code is a linear error-correcting code used to detect and correct errors in transmitted data. Hamming codes can detect single- and double-bit errors and are also capable of correcting single-bit errors. Typical parameters for Hamming codes include:

- Hamming distance: This is the number of bits that differ between two encoded words in the Hamming code. It is used to calculate the number of errors that can be detected or corrected. The hamming distance can also be called the signal distance.

- Hamming weight: The Hamming weight of a codeword is the number of 1's in that codeword. For example, the codeword 00111010 has a hamming weight of 4. The Hamming weight provides information about the error level of the codeword and plays an important role in the error detection and repair process.

For every positive integer m greater than or equal to 3, there exists a Hamming code with the following parameters:

Codeword length: $N = 2^m - 1$

Message length: $K = 2^m - m - 1$

Test section length: m = N - K;

Correction ability: t = 1 (dmin = 3)

Test matrix H with columns as a nonzero m-dimensional vector

Test matrix:

$$H = \begin{vmatrix} 0\,0\,0\,1\,1\,1\,1 \\ 0\,1\,1\,0\,0\,1\,1 \\ 1\,0\,1\,0\,1\,0\,1 \end{vmatrix}$$

The check bits x, y, z are placed at position 2^i with i = 0, 1, 2,…,

$t = \left(x, y, u_0, z, u_1, u_2, u_3\right)$, where u_0, u_1, u_2, u_3 are the information bits.

To find x, y, z: we have $tH^T = 0 \Rightarrow$ x, y, z

To generate code:

$$t.H^T = (x, y, u_0, z, u_1, u_2, u_3)x \begin{vmatrix} 0 & 0 & 1 \\ 0 & 1 & 0 \\ 0 & 1 & 1 \\ 1 & 0 & 0 \\ 1 & 0 & 1 \\ 1 & 1 & 0 \\ 1 & 1 & 1 \end{vmatrix} = 0$$

$$x.0 + y.0 + u_0.0 + z.1 + u_1.1 + u_2.1 + u_3.1 = 0 \;\Rightarrow z = u_1 + u_2 + u_3$$

$$x.0 + y.1 + u_0.1 + z.1 + u_1.0 + u_2.1 + u_3.1 = 0 \;\Rightarrow y = u_0 + u_2 + u_3$$

$$x.1 + y.0 + u_0.1 + z.1 + u_1.1 + u_2.0 + u_3.1 = 0 \;\Rightarrow x = u_0 + u_1 + u_3$$

Receiving codeword $r = (r_0, r_1, r_2, r_3, r_4, r_5, r_6)$, to check, we calculate Syndrome S:

$$s = r.H^T = (r_0, r_1, r_2, r_3, r_4, r_5, r_6) \begin{vmatrix} 0 & 0 & 1 \\ 0 & 1 & 0 \\ 0 & 1 & 1 \\ 1 & 0 & 0 \\ 1 & 0 & 1 \\ 1 & 1 & 0 \\ 1 & 1 & 1 \end{vmatrix} = (S_0, S_1, S_2)$$

$$s_0 = r_0.0 + r_1.0 + r_2.0 + r_3.1 + r_4.1 + r_5.1 + r_6.1 = r_3 + r_4 + r_5 + r_6$$

$$s_1 = r_0.0 + r_1.1 + r_2.1 + r_3.0 + r_4.0 + r_5.1 + r_6.1 = r_1 + r_2 + r_5 + r_6$$

$$s_2 = r_0.1 + r_1.0 + r_2.1 + r_3.0 + r_4.1 + r_5.0 + r_6.1 = r_0 + r_2 + r_5 + r_6$$

2.3. Channel encoding and decoding blocks

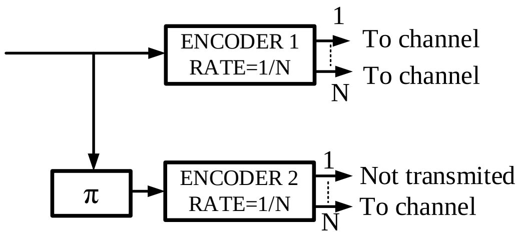

Fig.2. Turbo channel encoder diagram

The Turbo encoder performs encoding of the binary input signal using a parallel-coupled encoding scheme. This encoding scheme uses two identical convolutional encoders. Each component encoder is independently terminated by tail bits. Turbo channel coding is an encoding and decoding method based on combining multiple separate codecs. The encoder in turbochannel encoding usually consists of two convolutional encoders operating in parallel. The goal of turbochannel coding is to create a code with better interference resistance and higher transmission efficiency. The Turbo channel coding structure is built to take advantage of the combination of information from separate codecs.

The Turbo Decoding block performs decoding of the input signal using a decoding structure connected in parallel. The iterative decoding structure in the Turbo Decoding system uses the following components:

Posterior probability decoding: decoding based on the naturally determined probability of the input bits of the encoder. Through updating this probability, the decoder can make decisions about the resulting bits.

Stacker: performs naturally deterministic and probabilistic updates of input bits.

The stacking solver: performs inverse permutation on the stacker. This helps generate variations of the natural deterministic probabilities to improve the decoding process.

Both decoding components in the Turbo Decoding system use the same trellis structure and decoding algorithm. The trellis structure helps keep track of the states of the encoding and decoding systems, while the decoding algorithm determines how to update the natural deterministic probabilities of the input bits to make the final decision about the bits. The combination of trellis structure and decoding algorithm makes the Turbo Decoding system effective in decoding signals.

A block diagram illustrating APP decoding (marked as SISO modules) outputs an update sequence of the natural deterministic probabilities of the encoder's input bits, π(u;O). This sequence is based on the naturally determined probability sequence of the channel bits, π(c;I), and the encoding parameters.

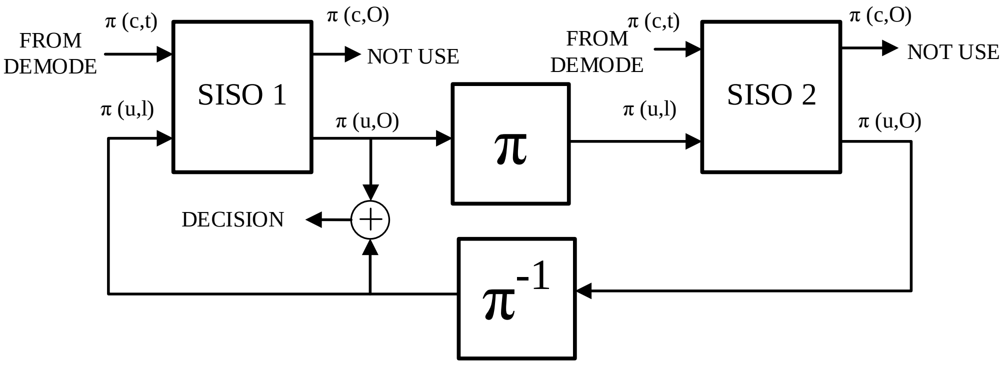

Fig.3. Turbo decoder block diagram

The decoding block iteratively updates these probabilities sequentially in a fixed number of decoding loops and then makes decisions about the resulting bits. The stack (π) used by the decoder is identical to the stack used by the encoder. The stacking solver (π^{-1}) performs the inverse permutation on the stacker. Decoding does not assume knowledge of tail bits and removes these bits from loops.

3. Building a digital data transmission system

The digital communication system uses the 16-QAM digital modulation method, hamming source coding, and turbo channel coding and operates through the AWGN noise channel built as shown in Figure 4, including:

Block that generates input data signals: Bernoulli Binary Generator;

QAM quadrature amplitude digital modulation block: rectangular QAM modulator baseband;

Transmission channel affected by Gaussian noise: AWGN channel;

16-QAM demodulator block: rectangular QAM demodulator baseband (parameters set as 16-QAM modulation blocks);

BER bit error rate calculation block: Error Rate Calculation;

Bit error rate display block: Display.

Input data encoding block: Hamming encoder;

Turbo Channel Encoder Block: Turbo Encoder;

Turbo channel decoding block: Turbo Decoder;

Hamming source decoder block: Hamming Decoder.

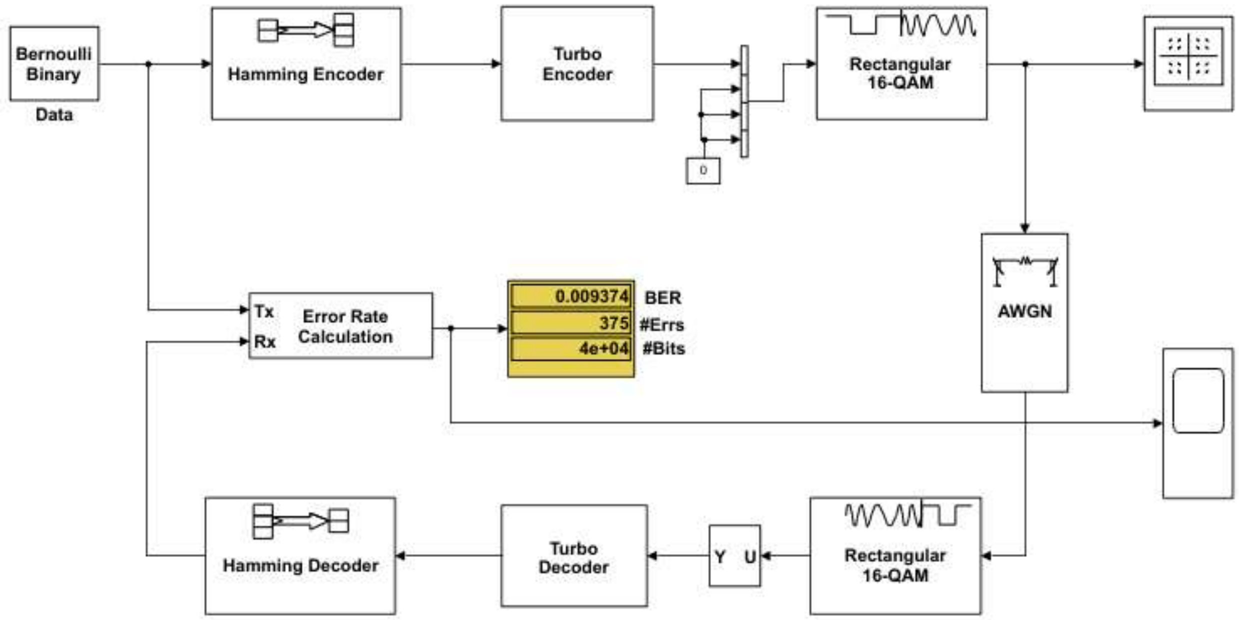

Fig.4. 16-QAM modulated digital data transmission system with Hamming source encoding and Turbo channel encoding via Gaussian noise channel

4. Simulate and evaluate the results

In conventional communication channels, such as wireless channels, cable channels, or optical channels, noise is often modeled as Gaussian noise (AWGN channel). In this case, the signal-to-noise ratio (SNR) is often used to measure the performance of the channel. In a Gaussian noise environment, the SNR value is often determined by the formula:

$$SNR = \frac{P^2}{\sigma^2}$$

In there:

P^2 is the power of the signal, expressed as the signal's energy per bit multiplied by the transmission rate.

σ^2 is the variance of the Gaussian noise.

Recipe: $$BER = \frac{N_e}{N_t} = \frac{N_e}{D.t_0}$$

With Gaussian noise, SNR is usually expressed in dB. SNR values can range from a few dB to tens of dB. For example, in wireless applications, SNR typically ranges from about 10 dB to 40 dB, depending on the transmission environment and operating conditions.

To evaluate the performance of an information transmission system, people use the bit error ratio index.

The transmission accuracy parameter is evaluated through the BER, which is the ratio of the number of erroneous received bits to the total number of transmitted bits during a certain observation time.

As the observation time approaches infinity, the BER approaches the bit error probability.

In there:

N_t is the total number of bits transmitted during time t_0.

N_e Is the number of erroneous received bits in time period t_0

D is the bit rate of the digital stream at the time of observation.

t_0 is the observation time.

In the case BER = 0, the digital system is not faulty.

In the case BER #0, the smaller the BER value, the higher the quality, and vice versa. Each digital system needs different BER criteria.

The survey results are shown in Figures 5 and 6.

According to Figure 5, with the digital information transmission system model without source coding and channel coding, the resulting bit error rate is 7.48%. According to Figure 6, the digital information transmission system model with Hamming source coding and Turbo channel coding results in a bit error rate of 1.01%. Thus, using source coding and channel coding in digital information transmission has significantly reduced the bit error rate.

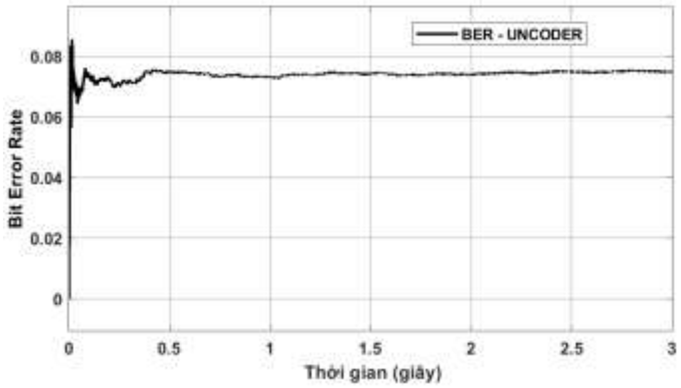

Fig.5. BER bit error rate without encryption

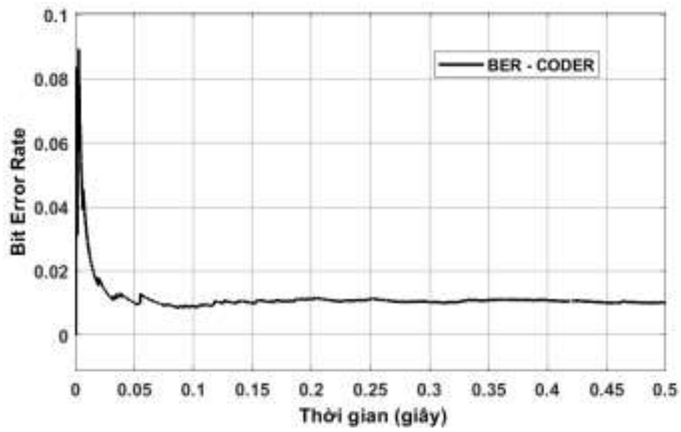

Fig.6. BER Bit Error Rate with Encryption

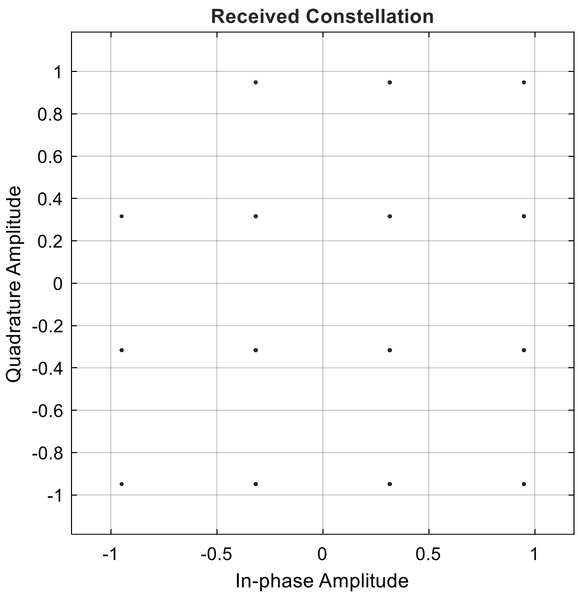

Fig.7. Constellation distribution when the encoder is not used

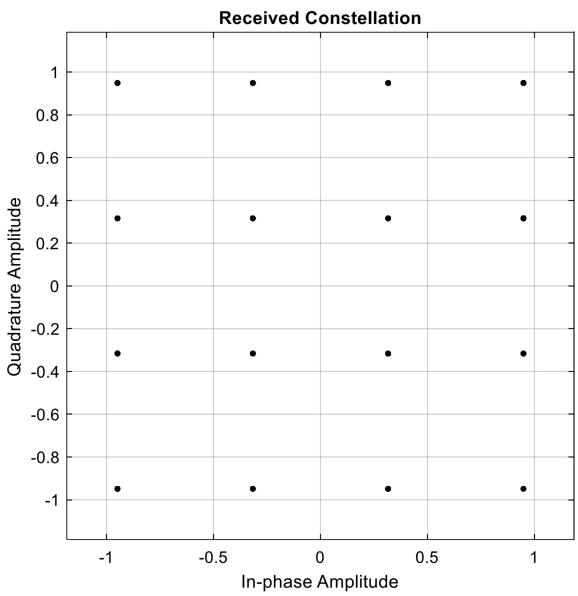

Fig.8. Constellation distribution when an encoder is used

Figures 7 and 8 simulate the constellation distribution of a digital data transmission system without and with a hamming source encoder and a turbo channel encoder when transmitting over a noisy Gauss channel.

The constellation distribution of 16-QAM when using the Hamming Source Encoder and Turbo Channel Encoder will be square with regularly arranged stars. However, when Hamming Source Encoder and Turbo Channel Encoder are not used, the constellation distribution is changed; constellation points are no longer regular and are missing. This demonstrates that noise in the transmission channel affects data transmission, causing a change in the constellation distribution and increasing the bit error rate when a hamming source encoder and a power coder are not used. Turbo channel encoding.

4. Conclusion

The paper presents in detail the digital data communication system over a Gaussian noise channel, using the 16-QAM modulator and demodulator, the hamming source encoder and decoder, and the Turbo channel encoder and decoder. Hamming codes are used to improve the system's anti-jamming capabilities by detecting and correcting bit errors during communication. Turbo codes help improve channel performance by using independent codes and combining information from received signals. The combination of source coding and channel coding reduces the bit error rate significantly. In the future, research may focus on optimizing and improving encoding and decoding methods to enhance the performance and reliability of communication systems.

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