Performance Analysis of Optimal Error-Protection Schemes for Enabling Quality Mobile Video Transmission

Amaad Khalil[*], Muhammad Abeer Irfan[†], Ishtiaque Ahmed[‡], Yaser Ali Shah[§], Abid Iqbal[**], Irfan Ahmad[††]

Abstract

Multimedia communication systems play a vital role in every field of life, thanks to their improved capabilities and higher customer demands. Multimedia communication is one of the most exciting research areas, as the demand for multimedia content over mobile information systems is increasing with each passing day. However, as wireless networks are susceptible to channel noise and data corruption, novel techniques must be devised for reliable, high-quality communication. Various flavors of forward error correction (FEC) methods can be deployed in mobile video transmission to achieve such a transmission mechanism. A comparison of different channel codes is required to understand the effectiveness of each algorithm. In this research, we have investigated Convolutional codes, Red Muller codes, Polar codes, Low-density parity check codes (LDPC), and Concatenated codes for their error correction capabilities. Furthermore, a mobile video transmission system employing these five channel codes in the presence of channel noise is being developed. The performance of each channel code is then evaluated using quantitative, i.e., peak signal-to-noise ratio (PSNR) and bit error rate (BER), and qualitative metrics. Simulation results show that the concatenated coding scheme outperforms other channel codes in mitigating the errors and distortions which typically arise in wireless transmission. More specifically, the concatenated coding schemes offer about 17 dB gain compared to the convolutional coding scheme at the PSNR degradation point of 40 dB.

Keywords: Low-density parity-check codes; Convolutional codes; Concatenated codes; Polar codes; Reed-Muller codes

Introduction

Researchers have recently paid much attention to digital multimedia communication systems. The need for cellular systems has increased in mobile communication due to 4G and higher wireless standards due to these systems' technological advancements and high-end features, such as mobility, accessibility, and high data rates (Davey & MacKay, 1998). Utilizing the bandwidth effectively

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[*] Department of Computer Systems Engineering – University of Engineering & Technology, Peshawar, 25000, Pakistan, amaadkhalil@uetpeshawar.edu.pk

[†] Department of Computer Systems Engineering – University of Engineering & Technology, Peshawar, 25000, Pakistan, abeer.irfan@uetpeshawar.edu.pk

[‡] National Center of Big Data and Cloud Computing – University of Engineering & Technology, Peshawar, 25000, Pakistan ishtiaque-ncbc@uetpeshawar.edu.pk

[§] Department of Computer Science, COMSATS University Islamabad, Attock Campus, Attock 43600, Pakistan Yaser@cuiatk.edu.pk

[**] Department of Electrical Engineering – University of Engineering & Technology, Peshawar, 25000, Pakistan, abid.iqbal@uetpeshawar.edu.pk

[††] Department of Electrical Engineering – University of Engineering & Technology, Peshawar, 25000, Pakistan, irfanahmed@uetpeshawar.edu.pk

is one of the main components of multimedia transmission (Khan et al., 2009). Military operations, satellite data transfer, and video conferencing will all benefit from improved performance due to the FEC technique. The forward error correction (FEC) technique provides crucial benefits across domains. In military operations, it ensures secure and accurate data exchange. For satellite data transfer, FEC maintains data integrity despite the noise. In video conferencing, it minimizes interruptions, ensuring seamless communication.

However, multimedia transmission over such channels is challenging due to the demand for higher rates and the irrational behavior of wireless channels (Seferoglu & Markopoulou, 2007). The behavior of a wireless channel is random and uncertain; this causes transmission errors in the data. Therefore, techniques are required to avoid such errors in video transmission; these techniques must be able to detect and correct errors at the receiving end. Upon detecting an error at the receiver end, it is corrected by either resending the entire data via an automated, repeated request (ARQ) or reconstructing the data by rectifying the error. The latter is made possible by adding redundant bits to the input data by FEC, which reduces bandwidth utilization and saves time and cost (Proakis, 1995) (Zorzi, 1998).

In recent literature, three well-known color modes—RGB, YUV, and HIS—are used for video transmission. The aerial images are transmitted using RGB colors. However, the final images have a blurring effect. The solution to this issue is to transmit each RGB channel separately in grey levels (Jia et al., 2006) (Süsstrunk et al., 1999); however, this technique results in a very computationally complex. Vector image restoration (Süsstrunk et al., 1999) is another method for blurred images.

Among the numerous color model, as anticipated earlier, the YUV color model has also been explored for efficient transmission by various research groups. YUV comprises luma (brightness) and chroma (color) information transmitted independently, which takes less bandwidth than RGB(Shi et al., 2009). The YUV format separates luminance and chrominance in videos, aiding efficient compression. Chosen for evaluation due to its human vision alignment and multimedia relevance, it assesses error correction techniques realistically in video transmission scenarios.

This work offers a comparative assessment of several FEC codes to comprehend the efficiency and accuracy of these coding methods for video transmission. In video transmission, YUV is preferred as the encoding of the luma and chroma is done separately. For these videos, error correction codes such as convolution, Polar, Reed-Muller, LDPC, and concatenated codes have also been investigated. After the thorough simulation, the obtained findings depict that we can reconstruct the video efficiently in the presence of impaired wireless channels at lower Eb/N0 values.

The order in which the paper is organized is as follows: Section 2 provides a concise overview of channel coding and related terminologies. Section 3 describes a system overview of the proposed approach. We assess the performance in Section 4 in light of the simulation outcomes. A conclusion is drawn based on the findings of this research work in Section 5.

Channel Coding techniques

One of the significant difficulties in wireless communication is transmitting safe and timely data to the user across a wireless channel. Typically, channel noise caused by fading and interference affects the sent signal during wireless transmission. Therefore, channel coding is employed in transmission systems to identify or fix problems. The original signal needed for recovery is retransmitted by communication systems using ARQ (Puri & Ramchandran, 1999) (Biersack, 1992).

FEC techniques are used for error detection and correction, where the extra bits are added to the transmitted signal before performing modulation. These bits assist in retrieving the original bit stream using various algorithms at the receiver end. The main objective of the FEC techniques is to increase the minimize the impact of noise in the transmission system. Approaches that use channel coding have a higher chance of recovering the sent signal at the receiver end. High-quality channel codes improve the system's performance since it can manage high data rates.

Additionally, it reduces the transmitting and receiving power of the antennas. FEC eliminates the need for data retransmission, which improves the performance of the entire transmission system (Luby et al., 2002) (Nafaa et al., 2008). FEC offers reliable data transmission, but its downside is; that extra bits are added at the transmitter side. Convolutional and linear block codes are the two main categories of channel codes. They are frequently used for error detection and correction (Ryan & Lin, 2009).

Convolutional Codes

In order to reduce channel noise during real-time transmission, these codes have been used in communication systems. Convolution codes were initially proposed by (Elias, 1955). These codes are widely utilized to exploit rigorous mathematical structures, yielding efficient results. The channel encoder adds the extra bits according to the current input bits and the lastly transmitted information bits. The entire bit stream is considered a single codeword. The convolutional code is denoted as (k,m,n) (Elias, 1955) (Rosenthal et al., 1996). Here, k stands for the bits that go into the Encoder, m stands for the memory used, and n stands for the output of the convolution code. During the transmission, the bits are fed into the m-length shift registers, also known as the constraint length. The longer constraint length indicates many parity bits, making the channel more error resistant. The coding rate r is denoted mathematically as r = k/n.

Recursive Systematic Convolutional Codes

Recursive systematic convolutional (RSC) codes have a better signal-to-noise ratio (SNR) than the rest of the convolution codes, even at higher code rates. These are frequently used because they are effective in robust channel codes, like concatenated and turbo codes. Encoding is done using a finite state machine, while decoding is performed by a trellis diagram (Elias, 1955) (Rosenthal et al., 1996). The convolution encoder can be seen in Figure 1. It consists of a systematic code since the output contains the input bits. In Figure 1, the output of the systematic code is represented by Output 2. The reverse behavior is exhibited in non-systematic coding. The RSC encoder has a feedback structure for the recursive systems, while non-recursive systems lack a feedback mechanism (Rosenthal et al., 1996).

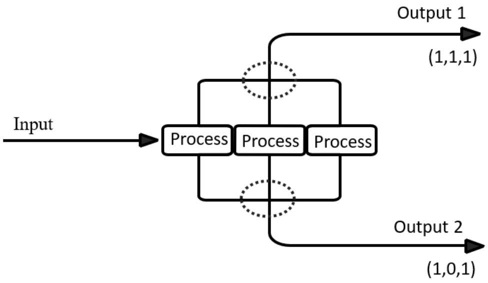


Figure 1: Convolutional Encoder

The RSC encoder contains k − 1-stage shift registers for a constraint length of k. The registers initially have a value of 0; then, they pass through n mod-2 adders to produce n generator polynomial. For instance, the generator polynomials g1 = [111] and g2 = [101] define the convolutional Encoder and can be expressed as G = [g1,g2]. The register m1 accepts input bits and computes outputs 1 and 2 based on the generator polynomials (m0 and m−1). These binary values are shifted to the next register (m1) for the input bit. The final bit is transmitted via m−1, the registers are reset, and the encoding process is resumed.

Recursive and non-recursive code

In contrast to the non-recursive codes, recursive codes accept the output bits of the generator polynomial as input. Recursive codes are also called systematic codes, whereas non-recursive codes are classified as non-systematic codes (Elias, 1955). Due to the higher weights of the codewords the recursive Encoder generates, it has improved error performance (Sharanya & Jayashree, 2016). In systematic codes, the bits from the input go straight to the output, which is impossible in non-systematic codes. Figure 2 shows the generator polynomials g1 and g2 used in the RSC encoder. In this case, g1 is [111], and g2 is [101]. It is expressed as G = [1,g1/g2]. The systematic output, or feedback sent back to the input, is denoted by n1, whereas the feed-forward output is represented by n2.

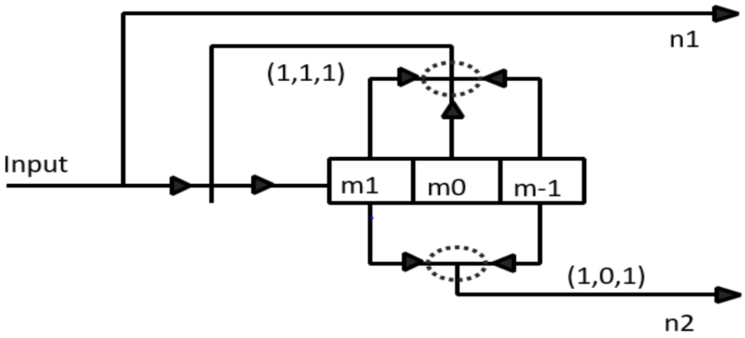


Figure 2: RSC Encoder

Trellis Termination

Trellis termination is a process that determines the state of the Encoder when the bit stream is finished. For example, imagine the Encoder receives (*l*-1) zero bits as input, which usually causes the trellis to end and sets the state to zero. Due to the feedback procedure used in this method, RSC encoders are not appropriate. As a result, an ON/OFF switch is instated to either turn on or off the encoding process, which concludes the trellis, as shown in Figure 3.

Convolutional codes can be decoded using a variety of algorithms. Convolutional codes employ common decoding approaches such as the Maximum-a-Posteriori (MAP) and Viterbi algorithms (Berrou & Adde, 1995). The Viterbi algorithm, a hard-decision decoding procedure, determines the most probable data sequence. By using a trellis diagram, it finds the path. The Code Division Multiple Access (CDMA) cellular networks, the Global System for Mobile (GSM), bioinformatics, and deep satellite communications are among the fields in which the popular Viterbi algorithm has been used. The Soft-Output Viterbi Algorithm (SOVA), a variant of the Viterbi algorithm, generates soft output using probabilistic soft inputs. It decodes more successfully because of its reduced BER but is more expensive and complex than the Viterbi approach (Rosenthal et al., 1996) (Berrou & Adde, 1995). The MAP algorithm uses the Bahl-Cocke-Jelinek-Raviv (BCJR) strategy, a soft-decision decoding technique, to calculate the forward α, backward β, and smoothed probabilities. Modern iterative decoding error-correcting codes like turbo and low-density parity-check codes (LDPC) depend on this method to operate effectively. It yields higher results for concatenated and turbo codes (Hagenauer & Hoeher, 1989).

Low-density parity-check Codes

LDPC codes can be called linear block codes, and their Encoder typically uses bipartite graphs that work on the idea of check and variable nodes. It employs the Message Passing or Belief Propagation algorithm (Gallager, 1962). At small Eb/N0, LDPC code performs significantly better while requiring less complexity to reach Shannon's capacity. The performance of LDPC codes is demonstrated with various parameters in the article (Ryan & Systems, 2004)[21], and the results show that this type of FEC primarily depends on the code rate r and the size of the check and parity nodes.

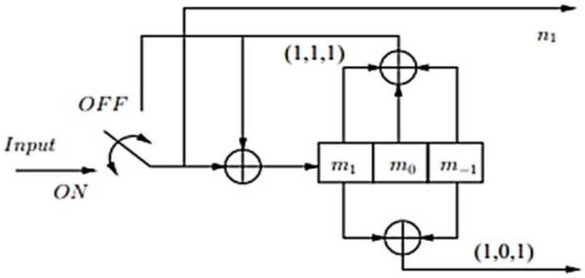


Figure 3: Trellis Termination of ½ rate RSC Encoder

Concatenated codes

Concatenated codes are an iterative serial code known as a "serial concatenated convolution code" (SCCC), primarily consisting of an inner code, an outer code, and a linking interleaved. Figure 4 depicts the working of Concatenated code Encoder. Sharanya et al. introduced inter-leaver for the first time in the SCCC (Sharanya & Jayashree, 2016). This new approach yields a significantly better result with the highest interleaver gain than turbo codes.

The decoder's components of any particular coding scheme exchange mutual information before generating the curves; this is presented in (Ten Brink, 2000). The accuracy of EXIT analysis can be improved by incorporating higher interleavers for higher non-correlation and the Gaussian-distributed Probability Density Function (PDF) (Ten Brink, 2000). EXIT analysis is now done for three-stage systems with several iterations and various source bit coding situations for varied hamming distances.

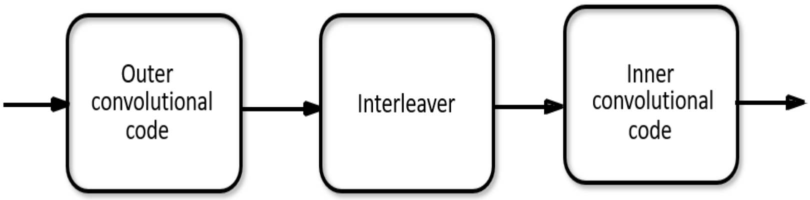


Figure 4: Concatenated code Encoder

Reed-Muller Codes

Reed–Muller (RM) codes are a family of error-correcting codes invented by Irving S. Reed and Gustave Solomon Muller in 1954 (Muller, 1954). They are linear codes that offer a good trade-off between the rate (the ratio between the data size and the code size) and the minimum distance of the code. RM codes are well-suited for applications such as communication channels with limited bandwidth, where a minor rate loss is acceptable, but a small number of errors must be corrected. They are also used in digital circuits and are widely used in error-correction applications such as digital television and digital audio (Muller, 1954) (Reed, 1954). The importance of Reed-Muller codes lies in their ability to offer a good trade-off between rate and distance. They can achieve a rate of up to 1/3 with a minimum distance of 5 and can be used in various applications. They are also linear codes, meaning they are relatively easy to implement and can be decoded quickly, making them ideal for applications such as digital television, digital audio, and communication channels with limited bandwidth (Kudekar et al., 2016).

A Reed-Muller code is an error-correction code used to detect and correct errors in digital communications (Muller, 1954). It is a linear block code consisting of two parts: an encoder and a decoder. The Encoder is responsible for taking a binary data sequence and transforming it into a longer sequence of bits suitable for transmission. The Encoder achieves this by using a generator matrix, which contains a set of coefficients representing the binary data and is used to construct the more extended sequence (Reed, 1954). The decoder takes the longer sequence of received bits and transforms it back into the original binary data sequence.

The decoder performs this task using a parity-check matrix containing a set of coefficients to determine the original data from the received bits (Kudekar et al., 2016). A Reed-Muller decoder works by applying the principles of Boolean algebra to the input values. These values are represented as logic equations, which are then simplified and rearranged to produce a set of equations that can be solved to determine the original values. The decoder transforms the input values into a set of equations that can be solved using Boolean algebra. The decoder then proceeds to solve the equations and deduce the original values. The decoder output is typically a binary number representing the original data (Pedarsani et al., 2011).

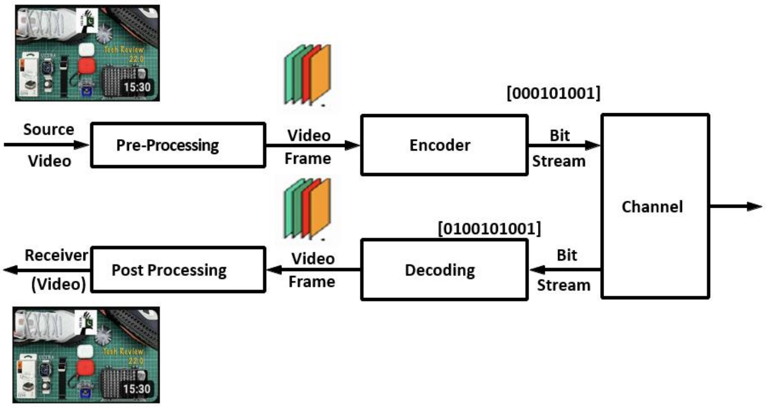


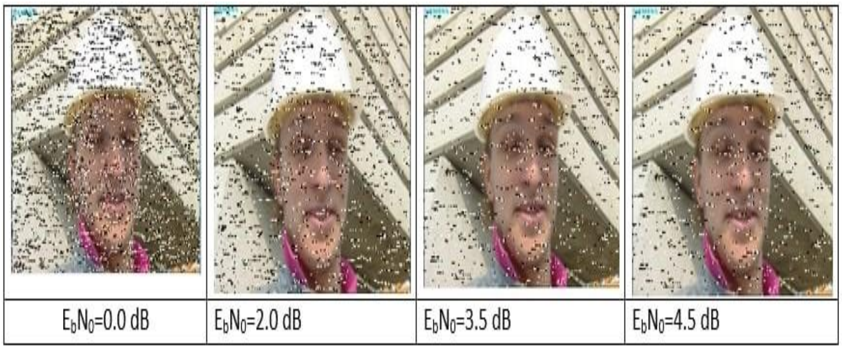
Figure 5: Video Transmission System using different error correction codes

Polar Codes

Polar codes are a new class of error-correcting codes developed by the Norwegian mathematician Arıkan in 2008 (Arikan, 2009). These codes have been proven to achieve the symmetric capacity of a binary-input memoryless channel with a low encoding and decoding complexity. Polar codes are a breakthrough in communication and are gaining significant attention as a potential solution to the problem of communication over a noisy channel. Polar codes have the unique ability to construct codes with capacity approaching the channel capacity, which is the theoretical maximum rate at which information can be sent over a communication channel with a given error rate. Since capacity is the highest rate of information transmission that can be achieved, Polar codes are of great value to communication systems. Polar codes significantly reduce complexity compared to existing codes (Indumathi et al., 2016).

They have a low encoding and decoding complexity, making them suitable for high-speed communication systems. Polar codes are also very efficient regarding their space requirements since they require only a few bits to store the encoding and decoding. In addition, Polar codes have a strong resilience to channel noise, making them ideal for use in communication systems operating in harsh environments (Aung et al., 2019). A successive cancellation list (SCL) decoder is an algorithm used to decode polar codes, which are a type of error-correcting code (ECC). It works by using a list of symbols and their associated likelihoods and then successively canceling out symbols from the list until only a single symbol remains.

This symbol is then the correct symbol chosen by the decoder. The SCL decoder works by first constructing a list of all possible symbols and their associated likelihoods. This list is sorted in descending order of likelihood (Tal & Vardy, 2015). The decoder then starts at the top of the list and successively removes symbols from the list that are not likely to be correct. This process is repeated until only one symbol is left in the list. This symbol is the correct one the decoder chooses (Yao, H. et al. 2023). The SCL decoder effectively decodes polar codes, allowing the decoder to identify the most likely symbol in the list quickly (Moradi, M et al. 2023). It also requires relatively low computational complexity, making it a suitable algorithm for decoding polar codes in hardware and software applications (Tal & Vardy, 2015) (Condo et al., 2018).



E_bN_0=0.0 dB E_bN_0=2.0 dB E_bN_0=3.5 dB E_bN_0=4.5 dB

Fig. 6: Subjective Quality Assessment of Convolutional Coding at Various E_b/N_0

E_bN_0=0.0 dB E_bN_0=2.0 dB E_bN_0=3.5 dB E_bN_0=4.5 dB

Fig. 7: Subjective Quality Assessment of Polar Codes at Various at Various Eb/N_0

Fig. 8: Subjective Quality Assessment of Reed Muller Codes at Various E_b/N_0

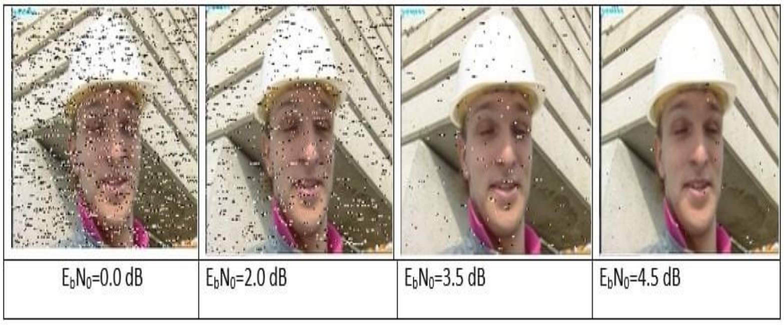


Fig. 9: Subjective Quality Assessment of LDPC Coding at Various E_b/N_0

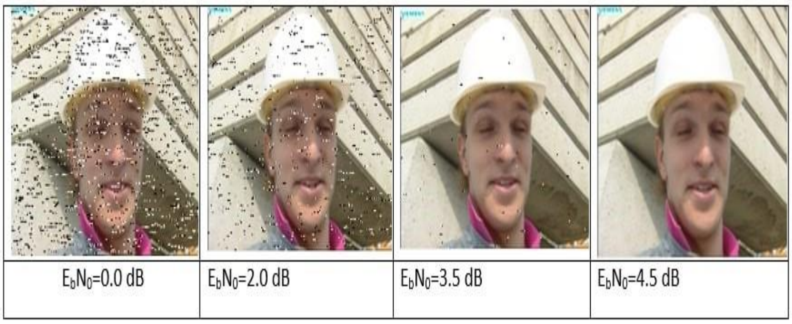


Fig. 10: Subjective Quality Assessment of Concatenated Coding at Various E_b/N_0

System Overview

The" Foreman" video sequence in the YUV format has been considered to evaluate the systems. The schematic diagram of the proposed transmission system using various error correction codes is shown in Figure 5. The video is first transformed into a standard bitstream. The Encoder of convolutional code contains three fundamental parameters, namely *k*, *m*, and *n*: *k*, which represents the input bits, *m*, which is for the memory registers, and *n*, which is the output of the convolutional code. The Encoder uses a half-code rate, resulting in an output message twice as large as the input. The

Encoder employs a half-code rate, resulting in an output signal twice as large as the input. The LDPC encoder adds parity bits, or extra redundant bits, to the input bitstream; these bits record any faults introduced into the transmission channel. The interleaved creates randomness in the bits, which helps with error correction.

After the encoding, the modulation process is carried out using various FEC codes. The digital data in the video bitstream has been sent for modulation, translating it into an analog signal. This makes it compatible with the channel, transmitting the waveform without interruptions. Our system employs the Binary Phase Shift Key (BPSK) modulation technique, where the phase of the carrier wave is switched from one state to another. The phase is shifted between the angles of 0° and 180°.

The signal set space is represented as:

$$s_1(t) = \sqrt{\frac{2E}{T}} \cos(\omega_0 t + 0) \quad (1)$$

$$s_2(t) = \sqrt{\frac{2E}{T}} \cos(\omega_0 t + \pi) \quad (2)$$

At the receiver end, demodulation occurs, which adheres to the same rules as modulation. The decoder receives the bitstream with errors due to various factors. Errors are detected and corrected by removing both the incorrect and unnecessary bits. The Viterbi algorithm has some drawbacks, one of which is that it uses too many resources; it follows the maximum likelihood decoding technique. This algorithm is typically used where the constraint length is shorter than 10. In the LDPC method, the decoder uses the belief propagation (BP) technique, which helps in effectively reducing the errors caused by a noisy transmission channel. The soft input-soft output (SISO) method decodes concatenated codes. It focuses on the MAX-Log MAP method and determines the best BER and decoding complexity trade-offs. The concatenated codes that rely on the trellis termination method are decoded using the BCJR algorithm. This algorithm is highly popular in iterative decoding because it recovers corrupted data and produces effective results.

System Performance and Simulation Results

This section describes the performance of the proposed video transmission technology. To simulate our schemes, we took advantage of IT++'s built-in library. As anticipated in the section, the" Foreman" video sequence was considered in the experiments; It contains 45 frames with a resolution of 176x144 pixels per frame. Five FEC codes—the LDPC, Polar, Reed-Muller, concatenated, and convolutional codes were the test subjects. The system used the variable E_b/N_0 to analyze the given input video sequence.

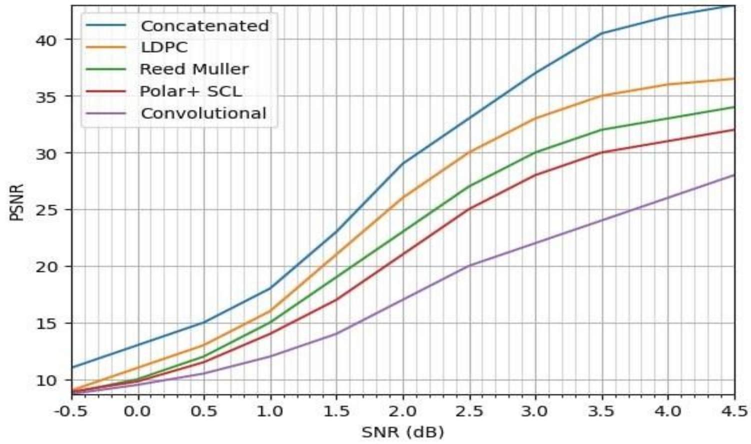


Fig. 11: PSNR Graph of the FEC Codes

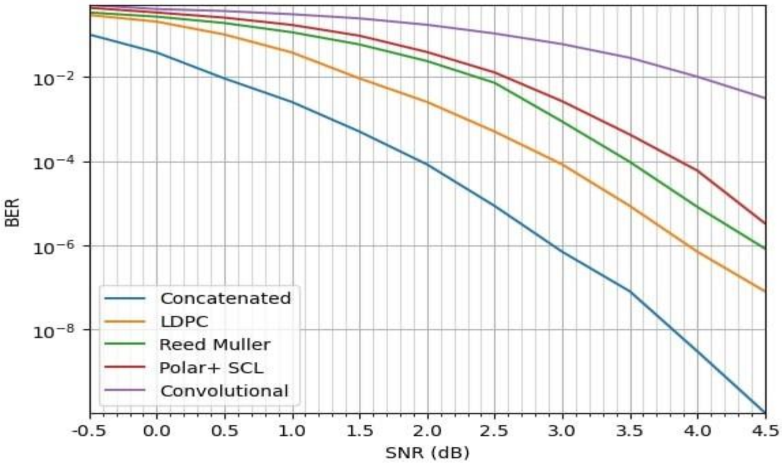


Fig. 12: BER Graph of the FEC Codes

The subjective quality evaluations for various E_b/N_0 are determined for each FEC code under consideration. The initial E_b/N_0 value was set to zero for the experiments. The value of E_b/N_0 is gradually raised until error-free video transmission is achieved. The subjective quality assessments for the FEC code are visually displayed in Figures 6, 7, 8, 9, and 10, respectively. The outcomes demonstrated that, compared to other codes under consideration, concatenated codes achieved more excellent video quality at a lower E_b/N_0 with fewer errors, as shown in Figure 10.

We compute the peak signal-to-noise ratio (PSNR) for various E_b/N_0 for all the FEC codes for the objective evaluation. The comparative analysis of these codes is shown in Figure 11. It can be seen here that the concatenated code performed better than the other codes. The concatenated code has the highest PSNR, followed by the LDPC, Reed-Muller, polar code, and the last convolution codes. Moreover, the concatenated code shows no distortion at E_b/N_0= 4.5 dB, unlike the other codes that show distortion at 4.5 Db.

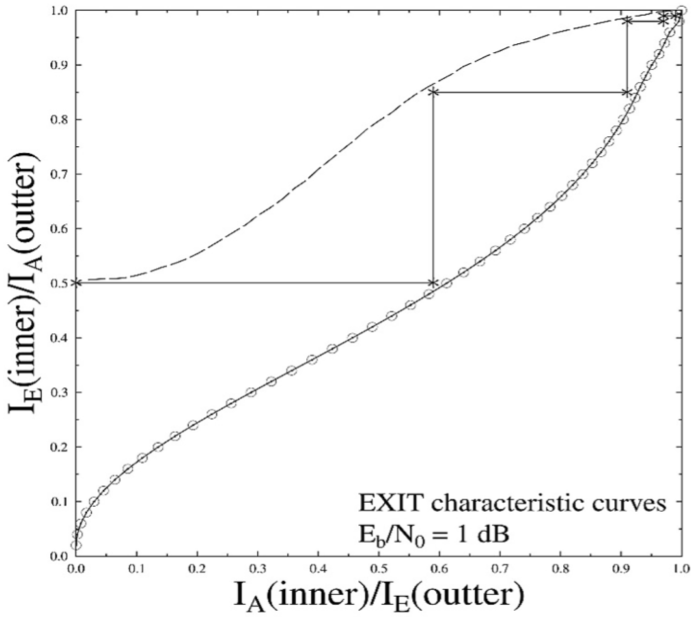


Fig. 13: EXIT Chart Analysis of Serial Concatenated Code at E_b/N_0=1 dB

The BER graph provides additional insight into the performance of the FEC algorithm. Figure 12 depicts the BER curve for various E_b/N_0 levels for the anticipated FEC codes. These findings support the earlier PSNR conclusions. The outcomes present substantial evidence that concatenated codes outperformed other codes in performance. The convolutional codes had the most minor performance, followed by the LDPC codes. The BER graph provides additional insight into the performance of the FEC algorithm. Figure 12 depicts the BER curve for various E_b/N_0 levels for the anticipated FEC codes. These findings support the earlier PSNR conclusions. The outcomes present substantial evidence that concatenated codes outperformed other codes in performance. The convolutional codes had the most minor performance, followed by the LDPC codes.

The BER and PSNR graphs in Figure 11 and Figure 12 clearly show that concatenated coding offers the best performance compared to the LDPC, Reed-Muller, Polar, and convolutional codes. So, we continue to estimate the convergence of our concatenated scheme using the EXIT analysis. Figure 13 shows that the two constituent curves of the serially concatenated scheme extend to the point in the top right corner of (1, 1). According to EXIT chart analysis, the concatenated scheme can achieve convergence and near-capacity performance. The staircase curve of the Monte-Carlo-based simulation trajectory confirms it too. More particularly, the curves represent the iterative exchange of mutual information between the components of decoders until the refined output is achieved at (1, 1); hence, the total convergence is attained at E_b/N_0 = 1 dB. If we increase the E_b/N_0, the EXIT tunnel becomes fully open, and the decoders require fewer iterations to reach convergence. Figure 14 shows that the concatenated coding scheme shares the mutual information in fewer iterations at a higher E_b/N_0 of 1.4 dB.

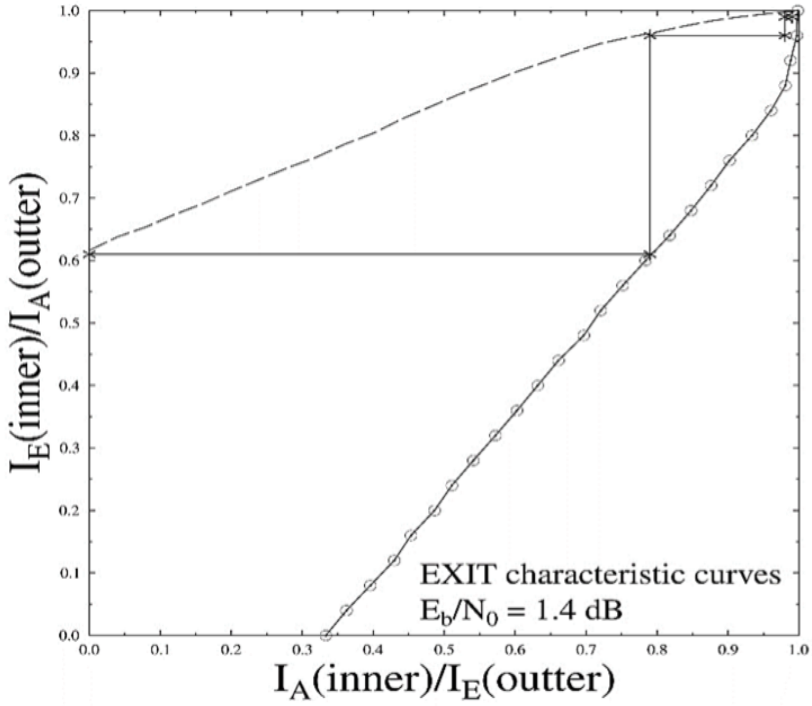


Fig. 14: EXIT Chart Analysis of Serial Concatenated Code at E_b/N_0=1.4 dB

Conclusion

The primary objective of this study is to evaluate various coding methodologies. Over the wireless channel, video transmission employs different FECs, including Reed-Muller codes, Polar codes, LDPC codes, convolutional codes, and concatenated codes. For varied Eb/N0, the concatenated codes outperformed the rest of the FEC coding techniques. The objective analysis using PSNR and BER at lower Eb/N0 depicts that the concatenated codes performed better than the Polar, Reed-Muller, LDPC, and convolutional codes. The EXIT analysis recommends the concatenated technique since it provides optimal convergence at 1 dB. The subjective evaluation also shows that the concatenated coding offers sharper visual quality at a lower value of Eb/N0. From both the objective and subjective analyses, it is inferable that the concatenated coding performs significantly well than its counterpart error-protection schemes. Conclusively, it can be seen that when considering a persistent simulation environment, the performance of FEC schemes decreases in the order of concatenated, LDPC, Reed-Muller, Polar, and convolutional coding. More specifically, at a BER of 10−4, the concatenated scheme outperforms the LDPC code by about 1 dB, the Reed-Muller code by 1.5 dB, and the Polar code by 2 dB. 1.

Future avenues for research involve the investigation of adaptive error correction, hybrid coding techniques, real-time deployment, assessing diverse network scenarios, merging with adaptive modulation, utilizing machine learning to enhance error correction, devising energy-efficient approaches, exploring quantum error correction, fostering industry partnerships for standardization, and examining the potential of error correction in enhancing security and privacy.

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