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Comparison Study between 1/2 Rate And 2/3 Rate Convolutional Encoder with Viterbi Decoder

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Abstract--As the need for bandwidth by bandwidth application became extremely alarming, the need to design a good encoders and decoders for the next generation wireless communications system became very imperative. Convolutional Encoder with Viterbi Decoder has been proved to be vital in this area. This research is on comparison study between 1/2 rate and 2/3 rate convolutional encoder with Viterbi decoder. The result of the research shows that an increase in the coding rate 'k/n' brings about a decrease in signal to noise ratio (SNR) gain of both. Also soft decision coding with width of 2 tends to have a better coding gain as compared to soft decision coding with width of 3.
Keyword--Convolutional Encoder, Viterbi Decoder, Communication, signal to noise ratio (SNR).

I. INTRODUCTION

We deal with a severely hostile/harsh environment across the air interface where we encounter channel variations, multipath fading, shadowing etc. which brings about a loss of information meant to be transmitted. This introduces or rather poses a question on how we can guard against this to be able to recover our transmitted signals at the receiving end of the channel. Encoding and Interleaving answers this question. Basically, these are processes used to build redundancy into the signals to be transmitted so that information lost on transit can be recovered. They are built into the transmitting base stations (BTs), phones and probably computers. Though we have simplified encoding scheme known as repetition code, but based on this project, we shall deal exhaustively with a kind of encoding method known as convolutional encoding in which we shall incorporate Viterbi decoders for next generation BWA systems. Each type and nature of a communication channel determines to a great extent which kind of transmitted information that will be affected by noise or deteriorated in any communication system. Many non-wired systems display large characteristic gains when the receiver knows the channel properties. The degree of obtaining these gains relies on the receiver's ability to accurately evaluate the channel parameters [1]. Convolutional encoders with Viterbi decoders are techniques used in correcting errors which are greatly deployed in communication systems to better the BER performance [2].

II. REVIEW OF RELATED WORK

The objective of any communication system is to successfully transmit data from the generating end to the sink with minimal error or degradation. Convolutional encoders with Viterbi decoders are techniques used in correcting errors which are greatly deployed in communication systems to better the BER performance [3]. Channel coding for error detection and correction helps the communication system designers to reduce the effects of a noisy transmission channel. Error control coding theory has been the subject of intense study since the 1940s and now being widely used in communication systems. As illustrated by Shannon in his paper published in 1948 [4], for each physical channel there is a parametric quantity called the channel capacity C that is a function of the channel input output characteristics. Shannon showed that there exist error control codes such that arbitrary small error probability of the data to be transmitted over the channel can be achieved as long as the data transmission rate is less than the capacity of the channel. Benson Ojedayo *et al* [5] conducted a research on Modeling of Convolutional Encoders with Viterbi Decoders for Next Generation Broadband Wireless Access Systems. They simulated four different rates which show similar behaviour of rapid fall in the graph. The fall signifies immunity to noise and how much better the message was decoded. They concluded that the initial simulation results indicate that on a Gaussian noise channel the Viterbi algorithm operating on the un-quantized channel output and at constraint length performs 1–2 dB better than their designed algorithm at similar complexity and with 5 to 7 quantization levels, which gave a BER at 8.6E-007 at 5db on an AWGN channel with BPSK modulation.
Jyoti *et al* [6] conducted a research on Performance Improvement by RS Encoding for Different Modulation Techniques used in WiMAX. They observed that properly chosen error correcting coding scheme can significantly improve the BER performance.
The simulation results clearly show significant decrease in bit error rate for a wide range of Eb/No. They found that BER for all

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tested modulation technique decrease monotonically with increasing values of Eb/No.

Lin *et al* [7] employs the combination of both FEC and ARQ systems in order to attain an excellent reliability as exhibited by ARQ system and high throughput as also exhibited by FEC systems. With the HARQ bringing together these two properties of FEC and ARQ, the shortcomings experienced when either of the codes is used independently are overcome. Unlike ARQ system which discards signals that were previously received due to the presence of error, HARQ is rather suggested to improve the performance of the signal by adding all the received signals together so as to decode the message transmitted. The main aim of Convolutional Code (CC) is to make available an efficient and reliable error correcting capability over an impairment channel, in the presence of a physically implementable decoder, to complement the success of J. A. Viterbi as proposed in 1967 [8], when he developed a decoding algorithm for convolutional codes. Though the Viterbi algorithm was relatively simple, it still met the criteria of exhibiting behaviour almost like that of a Maximum Likelihood Decoding (MLD) in practical decoders [9]. This Viterbi algorithm invention opened doors for new developments in convolutional codes as more factual research and advancement followed suit since then, which links convolutional coding with Viterbi decoding. Three types of CC exists which are Automatic Repeat request (ARQ), Forward Error Correction (FEC), and Hybrid Automatic Repeat ReQuest (HARQ).

Automatic Repeat reQuest (ARQ) is an error correction technique which combines error detection with the respect to retransmit an erroneous data. The received data block is checked for an existence of error in which if an error is indicated, the system will automatically request a retransmission of the sent data. This process is continued unless the transmitted data is certified to be error free. Due to this repeated request, we can say that ARQ technique needs the presence of a feedback channel and used for transmitting non real-time data [10].

Forward Error Correction (FEC) channel coding technique does not require a feedback and the data to be transmitted is first coded using an error-correcting code prior to its transmission. This extra signal combined with the code is in-turn utilized at the receiving end to recover the transmitted original data. FEC have enjoyed a wide range of application with respect to error-control. The two main types of FEC code are Block codes and Convolutional codes [11].

Block codes are suitable for burst errors and deal with large blocks of about a couple of hundred bytes. They are used in detecting errors only and tend to waste a high rate of bandwidth due to the addition of extra bits to the original transmitting data. Block codes could be classified as Hamming codes, Reed-Solomon codes, Cyclic codes etc. If a block of 'k' signal bits are encoded to yield a code word of 'n' bits (such that 'n' is greater than 'k') therefore, for each 'k' information bit arrays, a unique code word of 'n' bits exists. Convolutional codes sequence of 'n' bits makes use of both the present and previous 'k' information bits unlike the block codes. This makes it suitable for checkmating random errors. Both random and burst errors can be tackled in broadband wireless system (which is the goal of this dissertation) using a combination of block and convolutional codes for IEEE 802.6 – 2009 standard [12].

Hybrid Automatic Repeat ReQuest (HARQ) error correcting codes according to Lin et al [13] employs the combination of both FEC and ARQ systems in order to attain an excellent reliability as exhibited by ARQ system and high throughput as also exhibited by FEC systems. With the HARQ bringing together these two properties of FEC and ARQ, the shortcomings experienced when either of the codes is used independently are overcome. Unlike ARQ system which discards signals that were previously received due to the presence of error, HARQ is rather suggested to improve the performance of the signal by adding all the received signals together so as to decode the message transmitted. Two types of improved HARQ exists which are HARQ with chase combining (HARQ-CC) and HARQ with incremental redundancy (HARQ-IR). It will also be good to mention that many other ECCs exhibiting various tremendous coding gains was discovered alongside CCs and Viterbi decoding. Some types of these codes include: Low-Density Parity-Check (LDPC) codes, Turbo codes and Concatenated Codes.

Low-Density Parity-Check (LDPC) codes which was discovered in the early 1960's by Robert Gallanger in his PhD thesis, but it suffered a setback because of its requirement of high complex computational method and the then further introduction of concatenated Reed-Solomon's and convolutional codes which were all believed to be ideally appropriate for error control coding. LDPC possess a characteristic parity-check matrix that exhibits a few 1's when compared to the number of 0's and can be represented using two different possibilities just like all linear block codes via matrices and graphical representation [14].

Turbo codes was first introduced in 1993 by Berrou, Glavienx and Thitimajshima and can be described to be a repeatedly soft-decoding theory which integrates more than one relatively component code on different interleaved versions of the same information sequence (e.g. block and convolutional coding), to attain a high behavioural decoding specifically in high speed communications using the parallel concatenation of convolutional encoders or to produce a block code that can perform up to a fraction approaching the maximum channel capacity limit stated by Shannon. Unlike the conventional codes which yield hard-decision decoded bits at the final step of the decoder, the decoding algorithm of turbo code must bring about an exchange of soft decision among decoders to work properly rather than hard-decision. Therefore the concept behind turbo codes is to pass soft decision from the output of one decoder to the input of the next decoder and also iterating this process severally to obtain a more reliable decision [15].

Concatenated Codes was first introduced in 1965 by Forney in a bid to resolve a theoretical issue but they enjoyed a wide usage in the 1970's for space communications. They are basically error-correcting codes which are obtained from the combination of two or more simpler codes with the intention of achieving a good performance with a reasonable complexity. Turbo codes including other recent capacity-approaching codes can be referred to as extensions of this approach. As a matter of fact, the probability of decoding error can be set to decrease exponentially as the block length N, of the coding scheme approach infinity. Nevertheless, it has been established that the complexity of a naïve complex decoding scheme that easily compute the likelihood of every feasible transmitted code-word increases exponentially with the block length N, so such an optimum decoder rapidly becomes impracticable. Forney in 1966 demonstrated that concatenated codes could actually be implemented in achieving exponentially reducing error probability at all data rates less than capacity, with decoding complexity that increases only polynomials with code block length, N [16].

The efficiency of convolutional codes anchors on their constraint length 'K'. Codes possessing large constraint length are more powerful but have a set-back due to their high decoding complexity. This can be taken care of by using sequential decoding algorithm instead of Viterbi decoding algorithm.

One of the advantages of convolutional code is its speed and high affinity for error control. They are also quite less expensive when compared to block codes. For the same encoder/decoder level, convolutional codes generally out-perform block codes by providing higher coding gain.

A. Types of Channel

The following channel types discussed below exists in the non-wired communication systems.

1) Raleigh Fading: This models an environment where many objects that scatter the radio signal before it gets to the receiver exist and it is said to take care of the fading in a non-wired communication channel. The two kinds of fading include;

A) Long-Term Fading: This is order wise known as slow or log-normal fading exhibiting an envelope of fading information relating to the distance and received power.

B) Short-Term Fading: This can be called fast or Raleigh fading and it occurs basically as a result of reflection of transmitted signal. It can also be referred to the rapid fluctuations of a received signal that is set over an average value which changes gradually with the receiver movement.

The total information delivered to the receiver is composed of the product of Long-term and Short-term fading behaviors.

2) Gaussian Channel: This is basically employed in modeling the impairment produced at the receiving end for a perfect transmitting path. It is also called the AWGN channel. It describes very well and accurately many physical time-varying channels and also very useful in the provision of the system performance upper bound. It is assumed that the noise here possess a consistent Power Spectra Density, PSD across the channel bandwidth and a Gaussian Probability Density Function, PDF (though this rather is not responsible for fading). The Gaussian channel due to its relevance for both cases of single receiver and transmitter have been given a considerable recognition in multiple access literature though its model did not adequately explain the numerous access channels in which many users transmit frequently. The model which is of great desire here is one in which the active number of the network patronizes is a random variable and the Gaussian signal characterized on the basics that some users are transmitting. The additive noise channel is show in Figure 1 below.

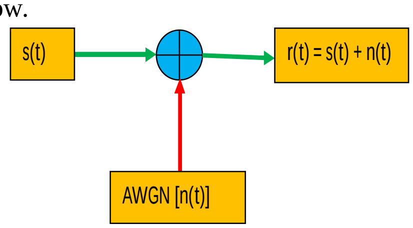

Figure 1: Additive noise channel

s(t) which is the transmitted signal is corrupted by the additive random noise n(t). The source of additive noise could be linked to the thermal noise innate in electronic components and the amplifiers at the receiver end or even arising from interference in the channel.

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The noise is characterized statistically as a Gaussian noise process and its mathematical model is therefore known as the additive Gaussian noise channel which can be represented mathematically as

$$r(t) = s(t) + n(t)$$

During the transmission process through the channel, the signal suffers attenuation given the received signal as

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

Where α is the attenuation factor, s(t) the transmitted signal and n(t) additive random noise.

3) Ricean Channel: This is a theoretical model of a non-wired propagation phenomenon in which the prevailing pathway (usually the direct line-of-sight, LOS pathway) exists together with numerous other irregular pathways between the transmitter and receiver. The predominant pathway reduces the delay spread which in turn lowers the fading depth significantly thereby suggesting enough lower fading margins in designing the system.

B. Channel Capacity

In the design of an information transmitting system, a crucial question that must be answered is how much information a particular information system will be able to transmit or process in a stipulated time. This is known as the Channel Capacity.

Shannon in 1948 published an interesting work [29] in which he defined the theoretical limit of communication efficiency over an impaired channel, which lead to the discovery of the theoretical basis of digital communications. He illustrated in his work that the communication channel displays a capacity, C (bps). Therefore, if one transmits at a rate (say Rbps) less than the given capacity, C, there is a possibility of him transmitting an error free signal as long as an appropriate coding is employed. The capacity, C of an AWGN channel of bandwidth, W, as shown in is obtained by;

$$C = W\log2(1 + S/N)$$

Where, C = maximum capacity of the channel (bits/second),

W = Channel bandwidth (Hz),

S = Average input signal power,

N = Average noise power.

Generally speaking, Shannon's information theory gives us an insight into the quantity of information a particular channel can carry at a given time. This theorem can be put together in the following few sentences:

The maximum rate of information, C contained in a given communication system is known as the channel capacity.

By the application of intelligent coding technique, when the information rate, R is less than C, a small error probability can be approached. A lower error probability can be achieved if the encoder is made to perform over a longer signal data block though this causes delay and increase in computational time. Basically, the signal strength, bandwidth and the quantity of noise in the channel determines the maximum information transmitting rate. This can be explained thus:

Increase in the channel bandwidth creates room for a faster change in the signal information thereby increasing the information rate.

Increase in S/N creates room for increase in information rate but checkmating increase in error due to noise.

At the absence of noise i.e. S/N = infinity, an infinite information rate can be achieved irrespective of the bandwidth, therefore as bandwidth tends to infinity, the noise power also increases, so the channel capacity cannot be infinity. Bandwidth can then be traded for S/N.

C. Channel Coding (CC)

The main aim of CC is to make available an efficient and reliable error correcting capability over an impairment channel, in the presence of a physically implementable decoder, to complement the success of J. A. Viterbi as proposed in 1967, when he developed a decoding algorithm for convolutional codes. Though the Viterbi algorithm was relatively simple, it still met the criteria of exhibiting behavior almost like that of a Maximum Likelihood Decoding (MLD) in practical decoders. This Viterbi algorithm invention opened doors for new developments in convolutional codes as more factual research and advancement followed suit since then, which links convolutional coding with Viterbi decoding.

Three types of CC exists which are Automatic Repeat request (ARQ), Forward Error Correction (FEC), and Hybrid Automatic Repeat ReQuest (HARQ).

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automatically request a retransmission of the sent data. This process is continued unless the transmitted data is certified to be error free. Due to this repeated request, we can say that ARQ technique needs the presence of a feedback channel and used for transmitting non real-time data [36]

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FEC have enjoyed a wide range of application with respect to error-control. The two main types of FEC code are Block codes and Convolutional codes.

a) Block Codes: They are suitable for burst errors and deal with large blocks of about a couple of hundred bytes. They are used in detecting errors only and tend to waste a high rate of bandwidth due to the addition of extra bits to the original transmitting data. Block codes could be classified as Hamming codes, Reed-Solomon codes, Cyclic codes etc.

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b) Convolutional Codes: For this codes case, its coded sequence of ‘n’ bits makes use of both the present and previous ‘k’ information bits unlike the block codes. This makes it suitable for checkmating random errors. Both random and burst errors can be tackled in broadband wireless system (which is the goal of this dissertation) using a combination of block and convolutional codes for IEEE 802.6 – 2009 standard as is discussed below.

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a) Turbo Codes: This was first introduced in 1993 by Berrou, Glavienx and Thitimajshima and can be described to be a repeatedly soft-decoding theory which integrates more than one relatively component code on different interleaved versions of the same information sequence (e.g. block and convolutional coding), to attain a high behavioral decoding specifically in high speed communications using the parallel concatenation of convolutional encoders or to produce a block code that can perform up to a fraction approaching the maximum channel capacity limit stated by Shannon. Unlike the conventional codes which yield hard-decision decoded bits at the final step of the decoder, the decoding algorithm of turbo code must bring about an exchange of soft decision among decoders to work properly rather than hard-decision. Therefore the concept behind turbo codes is to pass soft decision from the output of one decoder to the input of the next decoder and also iterating this process severally to obtain a more reliable decision.

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This combination of two codes has an advantage of providing a stronger error protection. The inner code which can either be a binary or Reed-Solomon code takes care of the random errors while the outer code which is usually Reed-Solomon code takes care of burst errors to prevent it from overloading the inner code. Care should be taken to balance the relative capabilities of the combined codes as an imbalance can increase the bit error rate of the system.

Concatenated codes and turbo codes are employed in the standard of WiMAX, W-CDMA and space communication system just like

Viterbi algorithm are seen to be used presently in our signal processing activities of many digital communication receivers.

D. Error Detection Schemes

Error detection is most commonly realized using a suitable harsh function (or checksum algorithm). A hash function adds a fixed-length tag to a message, which enables receivers to verify the delivered message by re-computing the tag and comparing it with the one provided.

There exists a vast variety of different hash function designs. However, some are of particularly widespread use because of either their simplicity or their suitability for detecting certain kinds of errors (e.g., the cyclic redundancy check's performance in detecting burst errors). Random- error-correcting codes based on minimum distance coding can provide a suitable alternative to hash functions when a strict guarantee on the minimum number of errors to be detected is desired. Repetition codes, described below, are special cases of error-correcting codes: although rather inefficient, they find applications for both error correction and detection due to their simplicity.

1) Repetition Codes: A repetition code is a coding scheme that repeats the bits across a channel to achieve error-free communication. Given a stream of data to be transmitted, the data is divided into blocks of bits. Each block is transmitted some predetermined number of times. For example, to send the bit pattern "1011", the four-bit block can be repeated three times, thus producing "1011 1011 1011". However, if this twelve-bit pattern was received as "1010 1011 1011" – where the first block is unlike the other two – it can be determined that an error has occurred.

Repetition codes are very inefficient, and can be susceptible to problems if the error occurs in exactly the same place for each group (e.g., "1010 1010 1010" in the previous example would be detected as correct). The advantage of repetition codes is that they are extremely simple, and are in fact used in some transmissions of number stations.

2) Parity Bits: A parity bit is a bit that is added to a group of source bits to ensure that the number of set bits (i.e., bits with value 1) in the outcome is even or odd. It is a very simple scheme that can be used to detect single or any other odd number (i.e., three, five, etc.) of errors in the output. An even number of flipped bits will make the parity bit appear correct even though the data is erroneous. Extensions and variations on the parity bit mechanism are horizontal redundancy checks, vertical redundancy checks, and "double," "dual," or "diagonal" parity.

3) Checksums: A checksum of a message is a modular arithmetic sum of message code words of a fixed word length (e.g., byte values). The sum may be negated by means of a ones'-complement operation prior to transmission to detect errors resulting in all-zero messages. Checksum schemes include parity bits, check digits, and longitudinal redundancy checks. Some checksum schemes, such as the Damm algorithm, the Luhn algorithm, and the Verhoeff algorithm, are specifically designed to detect errors commonly introduced by humans in writing down or remembering identification numbers.

4) Cyclic Redundancy Checks (CRC): Cyclic redundancy check (CRC) is a single-burst-error-detecting cyclic code and non-secure hash function designed to detect accidental changes to digital data in computer networks. It is not suitable for detecting maliciously introduced errors. It is characterized by specification of a so-called generator polynomial, which is used as the divisor in a polynomial long division over a finite field, taking the input data as the dividend, and where the remainder becomes the result.

Cyclic codes have favorable properties in that they are well suited for detecting burst errors. CRCs are particularly easy to implement in hardware, and are therefore commonly used in digital networks and storage devices such as hard disk drives.

Even parity is a special case of a cyclic redundancy check, where the single-bit CRC is generated by the divisor $x + 1$.

5) Cryptographic Hash Functions: The output of a cryptographic hash function, also known as a message digest, can provide strong assurances about data integrity, whether changes of the data are accidental (e.g., due to transmission errors) or maliciously introduced. Any modification to the data will likely be detected through a mismatching hash value. Furthermore, given some hash value, it is infeasible to find some input data (other than the one given) that will yield the same hash value. If an attacker can change not only the message but also the hash value, then a keyed hash or message authentication code (MAC) can be used for additional security. Without knowing the key, it is infeasible for the attacker to calculate the correct keyed hash value for a modified message.

III. METHODOLOGY

In this research, we shall explore the use of MATLAB in modeling of convolutional encoder with Viterbi decoders for next generation broadband wireless access system. Using the MATLAB software as required and employing the knowledge of analytical theory of the coding fundamental principles, the convolutional encoder and Viterbi decoder was modelled as shown in Figure 2.

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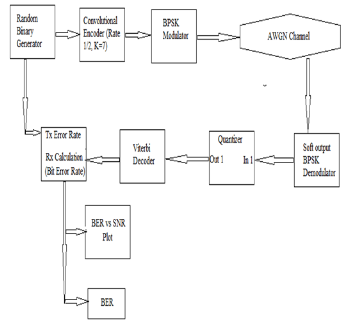

Figure 2: A communication system model block diagram exhibiting the Convolutional Encoder and Viterbi Decoder

The steps involved in simulating a communication channel using convolutional encoding and Viterbi decoding are as follows:

A. Generating the Data

The data to be transmitted through the channel is generated using *randn* function of Matlab in combination with the sign function. We have generated 1000000 bits.

Below is a piece of MATLAB functional code cut out from the full code that performs this action. The multiple zeros sent in at the end of the sequence are used to flush out the bits.

```
clc;
clear;
Rate = 1/2;          %Rate can be 1/2, 2/3
N_Data_bits = 1000000;
Data_Input = [randi([0,1],[1,N_Data_bits-7]), 0, 0, 0, 0, 0, 0, 0];
```

Due to the randomness in the data generation, a different data array is got for each different simulation of the code, giving us a somewhat different plot though with each of the curves maintaining the same plotting trend due to the evenly distribution of the overall data.

B. Convolution ally Encoding The Data

Our convolutional encoder as shown in Figure 3.2 below is made up of a data input generator, a pair of modulo-2 adder with corresponding pair of outputs (first and second) and 6 memory shift registers. A 'k' number of bits/second goes into the input and an 'n' output bits equivalent to '2k' symbols/second got for each output, thus giving a code rate value of 'k/n' = 1/2.

C. The BPSK Modulator

The BPSK modulation technique is utilised here in modulating the transmitted data sequence. The 'zeros' and 'ones' got from the encoders output are mapped onto the antipodal baseband signalling scheme using the BPSK block maps. By this we mean that the 'zero' output values of the encoders are converted to 'ones (1)' and the corresponding 'ones' converted to 'negative ones (-1)'. This is actualized by carrying out a simple MATLAB iteration process involving the use of 'Modulated = 1 - 2*Code' equation on the encoders output as shown in the box below. 'Code' represents the convolutional encoders output and 'Modulated' being the result of the modulation**.**

D. The AWGN Channel

In modeling the AWGN channel, we first of all generated Gaussian random numbers which was further scaled based on the transmitter energy per symbol in comparison to the noise density ratio, i.e. E_s/N_o. This is a function of SNR per bit, E_b/N_o and code rate, k/n which can be represented mathematically as:

$$E_s/N_o = E_b/N_o + 10\log_{10}(k/n)$$

For the code rate of an uncoded channel, $E_s/N_o = E_b/N_o$, making it equivalent to unity. Based on this finding, the rate 1/2 encoder exhibits an energy per symbol to noise density ratio of $E_b/N_o + 10\log_{10}(1/2) = E_b/N_o - 3.01$dB.

The uncoded signal over the AWGN channel has its theoretical BER written as

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$$P_b = 1/2 \text{ erfc}\sqrt{\left(\frac{E_s}{N_o}\right)}$$

E. Demodulation

The Additive White Gaussian Channel gives out its sequence in a complex form ranging from 'negative ones' to 'positive ones' (-1 to +1) but this is not in the form the Viterbi decoder can act on it. Therefore, the function of the BPSK demodulator as employed here is to convert these complex data sequence to real data so it can be acted upon by the Viterbi decoder. The demodulator simply carries out on the complex data an operational function 'y = real(x) > 0' for the case of hard decision decoding and 'y = real(x)' for both cases of soft decision and un-quantized decoding.

F. Quantization

A perfect Viterbi decoder should be able to operate perfectly well with an infinitely quantized sequence, but unfortunately, this has a way of increasing the complexity of the Viterbi algorithm and data sequence decoding time, so a few bits of precision in practice is employed in the quantization of the channel symbol to checkmate this. Since quantisation level can change from 1-signal bit to infinity, we have chosen 1-bit (for hard decision), 2-bit, 3-bit, 4-bit (for soft decision) and unquantized level for this work. Any bit less than or equal to zero is mapped to '0' and ones greater than zero mapped to '1' for the case of '1-bit' quantization level. The input values for the '2-bit', '3-bit' and '4-bit' quantization is being set by the block from '0 to 2^n-1' where 'n' takes the values of '2', '3' and '4' for the respective bit decision decoding, making the numbers range from '0 – 3', '0 – 7' and '0 – 15' respectively. For '3-bit', the Viterbi decoder interprets '0' as the most confident '0' (strongest) and '7' as the most confident '1', while decision values lying between '0 – 7' are at extreme of the respective values.

G. Viterbi Decoding The Encoded Data

Viterbi decoder modelling among the other elements in the whole system is the most tasking. Their modelling process involves some major stages which include: - De-puncturing, Branch Metric Computation BMU, Add-Compare and Select ACS, and finally the TraceBack Decoding TBD.

IV. RESULTS

Comparing soft decision coding with width of 2 and soft decision coding with width of 3, it can be seen from the graph that an increase in the coding rate 'k/n' brings about a decrease in SNR gain of both. Also soft decision coding with width of 2 tends to have a better coding gain as compared to soft decision coding with width of 3

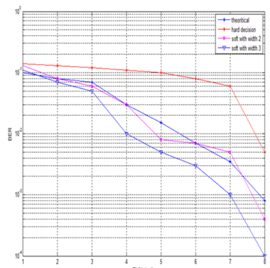

Figure3: Comparison soft decision coding with width 2 and soft decision coding with width 3

V. CONCLUSION

Convolutional Encoder with Viterbi Decoder has been proved to be vital in this area. This research compared study 1/2 rate and 2/3 rate convolutional encoder with Viterbi decoder. The result of the research shows that an increase in the coding rate 'k/n' brings about a decrease in signal to noise ratio (SNR) gain of both. Also soft decision coding with width of 2 tends to have a better coding gain as compared to soft decision coding with width of 3.

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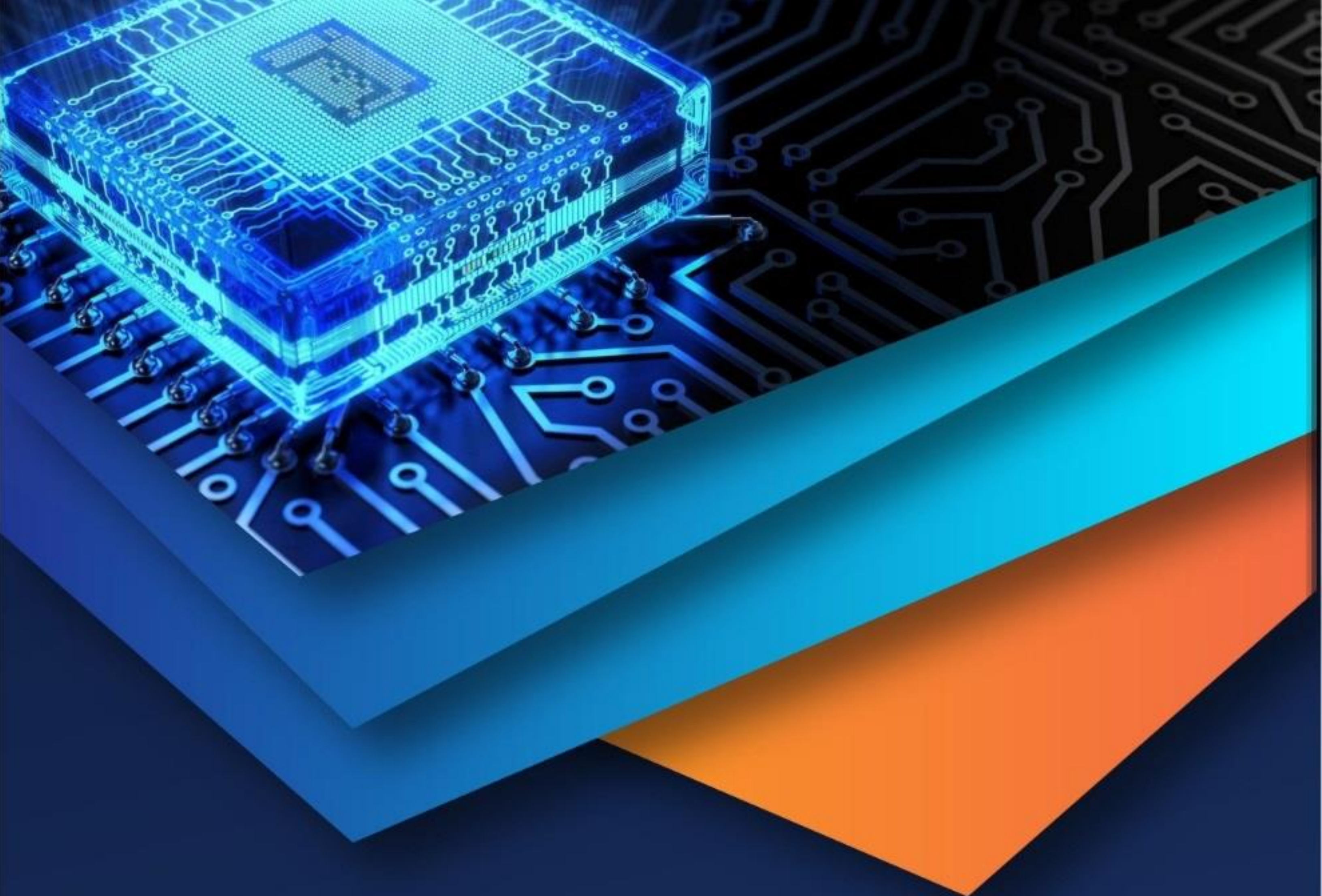

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45.98

IMPACT FACTOR: 7.129

IMPACT FACTOR: 7.429