Analysis and Performance Evaluation of Convolutional Codes over Binary Symmetric Channel Using MATLAB

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ABSTRACT

The most common concern of any communication system is the data quality. There exist different components that can impact the quality of data during its conveying over the channel as noise, fading, etc. Forward error correcting codes (FEC) play a major role for overcoming this noise as it adds a control bits to the original data for error detection and correction. This paper aims at analyzing convolutional codes with different rates and evaluating its performance. Binary phase shift modulation (BPSK) scheme and binary symmetric channel (BSC) model are used. First a convolution encoder is presented and then additive white Gaussian noise (AWGN) is added. The paper uses maximum likelihood mechanism (Vetribi Algorithm) for decoding process. Simulations are carried out using MATLAB with Simulink tools. Bit error rate (BET) is used as testing parameter and results of system behavior for both coded and encoded are compared.

Keywords: Convolutional codes, Viterbi decoding, AWGN, Code rate and BPSK

1 INTRODUCTION

The fundamental target of communication systems is to involve conveying the information through the channel to be received with as less error as possible.

Digital communications have been adopted to perform such goal due to their capability of processing data faster than the conventional (analogy) communications and potentiality of extremely less error rate. One of the major reasons for the continuous growth in the use of wireless communication is to increase the capability to provide efficient communication links to almost any location, at constantly reducing cost with increasing power efficiency. For this reason, digital communication systems have experienced quick-replacement in the area of telecommunications [1].

To appreciate the wireless version of a digital communication system, firstly it is necessary to consider what the essential components of a digital communications system are. A general block diagram of a digital data communication system is shown below:

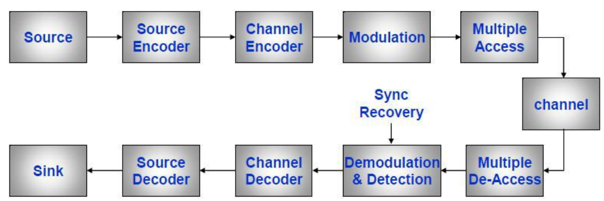

Figure 1 Block diagram of digital communication system

In this system, it is primarily concerned with measuring the probability of error and the mechanisms introduced to minimize it at the receiver side [2]. The source represents any entity that contains information to send such as, audio, image, data…etc. Whereas, the source encoder, provides digitization and compression in order to remove the redundant information that results in reducing the bandwidth [1]. Channel coding evolves adding the controlled redundancy (extra bits) to detect and correct the errors at the receiver side for instance; linear codes, convolution codes and turbo codes. Afterward, the baseband signal needs to be modulated to a conceivable format that matches the physical medium. Multiple access is a mechanism that involves more than one transceiver to share the same medium such as TDMA, FDMA CDMA,…etc, then, the

physical channel where the signal typically experiences the distortion components, either Additive White Gaussian Noise (AWGN) process or multipath.
The receiver side performs the complementary process of the transmitter with the capability of overcoming the interference produced through the channel [1] & [2].

1.1 Coding Theory Overview

Coding theory is a technique used to efficiently and accurately transfer the information from one point to another. This theory has been sophisticated for the purpose of several applications such as, minimizing noise from compact disc recorders, or sending of financial information across telephone line, data transfer among many computers in a networks or from one location in a memory to the central processor, and information transmission from a distance source such as a weather or communications satellite or the Voyager spacecraft which sent pictures of Jupiter and Saturn to Earth. Therefore, the prior task of coding theory is to deal with the issue of detecting and correcting transmission errors resulted from the noise that is introduced through the channel [3]. In practice, the control we have over this noise is the choice of a good channel to be used for transmission and the use of various noise filters to combat certain types of interference which may be encountered [3].

1.2 Block Codes

In block coding, the message is segmented into blocks, each of which has k bits information which is named dataword. Every single dataword **(k)** is appended an extra bits called redundancy **(m)** to form codeword **(v)** with a length of **n** as shown in the following figure**.**

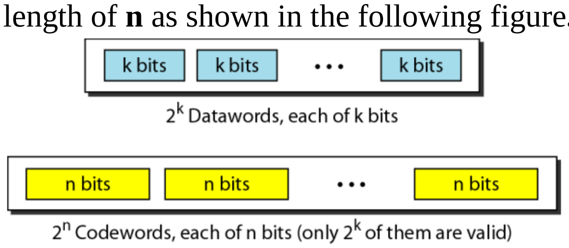

Figure 2 Datawords and codewords in block coding

Each block of message is expressed by the binary of k - tuple $\mathbf{u} = (\mathbf{u_1}, \mathbf{u_2},... \mathbf{u_3})$. In bock coding, the symbol **u** is used to express a k-bit message instead of the entire information sequence.
As a result, there are $\mathbf{2^K}$ different possible messages. The encoder reforms each message **u** independently into an n-tuple by adding redundancy, so that the new form will be as:
$\mathbf{v= (v_1,v_2,\ldots v_k)}$ of discrete symbols called a code word. Therefore, corresponding to the $\mathbf{2^k}$ different possible message, there are $\mathbf{2^n}$ different possible code words at the encoder output. This set of $\mathbf{2^k}$ code words of length **n** is block code called **(n, k)** block code .
This ratio **R =k/n** is called a code rate, and can be interpreted as the number of information bits entering the encoder per transmitted symbol [3].

1.3 Error correction mechanisms

Wireless technology has experienced a considerable enhancement in terms of the achievement of the fast deployment at low expenses. Whereas, the utilization of poor quality of the channel usually demands retransmissions, that are administrated by using Automatic Repeat Request (ARQ) [4]. Although, the ARQ is an advantageous tool to mitigate the errors of the packet, it has some drawbacks including the increment of the power expenditure due to retransmission and also latency. However, since the attention was gradually being shifted to the forward error correction (FEC), it has provided several advantages compared to the use of ARQ scheme [3]. One of its advantages is the development of the performance of the error control which is obtained by applying the error correction code that is based on adding extra bits to the original data that is known as redundancy. This redundancy is the principle of forward error correction that provides the capability of error detection and also correction at the destination side without the need to retransmission or waiting for acknowledgment return compared to ARQ. Also, in case of

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propagation delay, the FEC has become more suitable selection than other schemes [4].

Therefore, appending the check bits will typically increase the bandwidth used for transmission or causing the packet delay or might be both, however to achieve multiple bits correction, it is advantageous to add more redundant bits. This strategy requires higher bandwidth of forward channel as well as more cost. Furthermore, the FEC techniques can be divided into two main categories which are the block coding and convolutional coding. The block coding consists of several coding shames such as, Reed Solomon, BCH and Hamming codes [3] & [4].

2 CONVOLUTION CODES

Convolution codes have been applied for several systems encompassing today's common wireless standard as well as satellite communications. The adoption of these codes is the recent approach that express the well building block in more powerful and modern codes that are applied for wide area cellular structures such as, 3G, LTE, LTE-A, 4G. The main purpose of using convolution coding technique is to minimize the probability of errors over any noisy communication channel. It assists in recovering the most likely message from among the set of all possible transmitted messages [4].

2.1 Implementation of convolutional code

The convolution code encoder accepts k- bit block of information sequence **u** and generates an encoded sequence (codeword) **v** of n-symbol blocks.

However, each encoded block depends on both the corresponding k-bit message block at the same time unit, and also on **m** previous message blocks. Hence, the encoder has a memory order of **m**. The number of encoded symbols is called an (n, k, m) convolution code. The ratio R=k/n is called the code rate. Block codes contains no memory which results in an independency for the consecutive codewords [5]. On the other hand, because of dealing with data blocks, buffering memory and latency overheads are always associated with block codes. Block codes, as opposed to convolution codes, can be cross interleaved for reliable storage of data. Block codes can be concatenated with convolution codes or mapped together onto an iterative (turbo) configuration for higher performance over some channels [5].

Convolution encoders are typically simple state techniques to be implemented in hardware. Whereas, decoding part is complex because it is based on involving searching for a best path to reconstruct or recover the transmitted information, while it is more amenable to soft-decision decoding compared to block codes, thereby better coding performance can be resulted. Hence convolution codes are conceivable for lower SNR channels, as well as transmitters use simple low power devices. Because the encoder consists of memory, it is essential to be constructed with the circuit of sequential logic. For binary convolution codes, control bits (redundant bits) are appended to data sequence to combat the channel noise in case of k<n or R<1. Basically, k and n are both small integers and more redundant bits are appended. This can be achieved by increasing the memory order, while keeping k and n, this result in the code rate R, fixed. In fact, the serious issue of implementing the code is the method of utilizing the memory to obtain more reliability to transmit the information through a noisy channel [6]. A binary convolution encoder with (2,1,2), where, k=1, n=2, and m=2

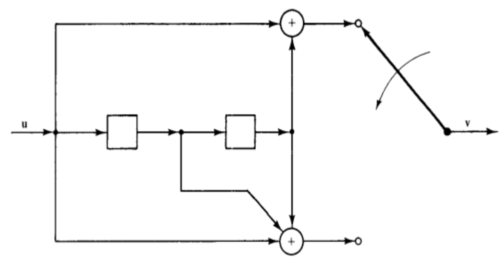


Figure 3 Binary convolution encoder with (2, 1, 2).

is implemented as shown in Figure (3), which demonstrates the technique of generating the codeword if the information source is given as (1 1 0 1 0 0 0…..). For this example, the most left bit is supposed to be the first input bit to the encoder. All operations are performed using exclusive-or (XOR) operation. As a result of that, the encoded sequence is

$V = (1\,1\ \ 1\,0\ \ 1\,0\ \ 0\,0\ \ 0\,1\ \ 1\,1\ \ 0\,0\ \ 0\,0\ \ 0\,0)$

Where, the first encoded bit is assumed to be the top output.

2.2 Trellis Diagram in Convolution Codes

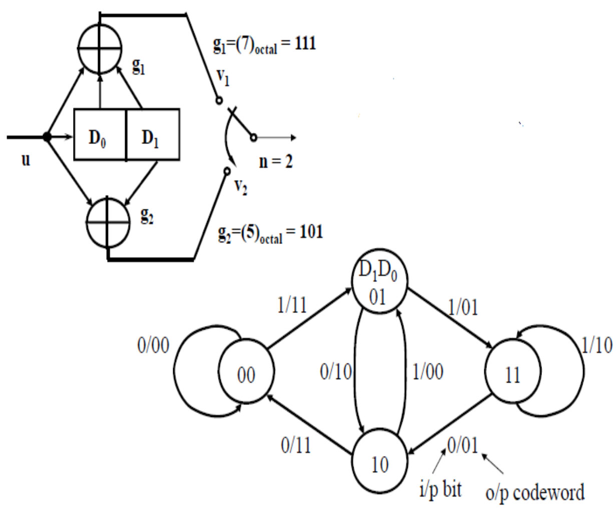
One of the useful means to decode convolution codes is to compare the received codeword with all other possible codes (Maximum Likelihood Decoding). The function of MPA is basically based on choosing the hypothesis that has the highest posteriori probability for the present observation. For M-ary detection, where the selection will be between M hypotheses,

$$\widehat{H}_{MAP}(y) = \arg\max\{P(H_i \backslash y)\} \qquad (1)$$

The function of (argmax) is to return the value of the index i for which its argument is maximum [7].

Trellis diagram is a graphical representation of convolution encoder encompassing the state diagram information with a time dimension cross the horizontal axis.

A (2,1,2) is convolution code with code rate of ½ is encoded as shown in figure (4) [7].



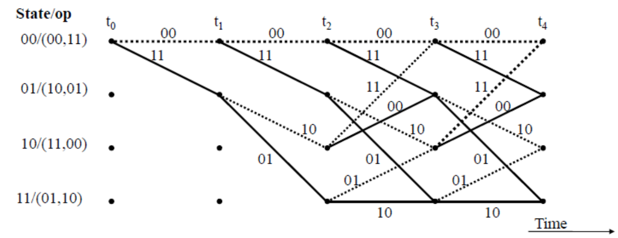


Figure 4 convolution code of (2, 1,2) with 1/2 code rate

If the input data is given as (1011), this maps to 00 01 10 11 path as illustrated in figure 5.

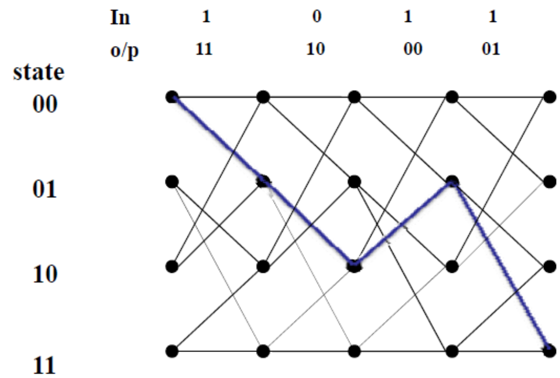


Figure 5 Code Trellis for input (1011)

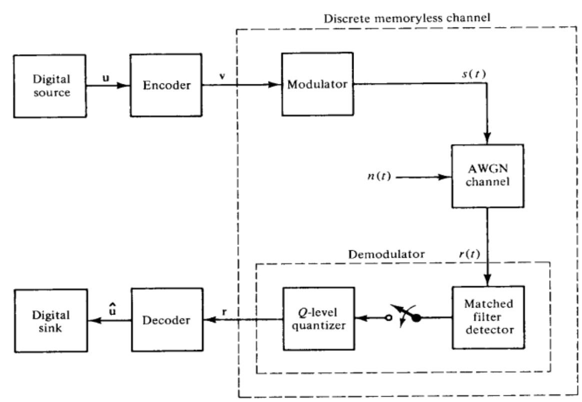


Figure 6 Coded systems over AWGN channel

2.3 Maximum Likelihood Decoding

A schematic diagram of a coded system for an AWGN channel with finite output quantization is show in Figure 6 [5].

The figure shows that the output **u** represents a k- bit message; the encoder output **v** represents an n- symbol code word. In a demodulator side The output **r** points to the corresponding Q-ary received n-tuple, and decoder output $\hat{\boldsymbol{u}}$ represents the estimated copy of the original message. In a convolution coded system, **u** represents **KL** data bit sequence and **v** refers to a code word consisting of

$$N \triangleq nL + nm = n(L + m) \qquad (2)$$

Where kL is the length of the information sequence and N is the length of the codeword. The additional nm encoded symbols are generated after the last block of information bits has inserted the encoder [5].

It is necessary for the decoder to generate an estimate **u** of data sequence which is depends on the received sequence r. Equivalently, where there exists a one–to-one correspondence between the data sequence u and the codeword v, the decoder will has capability of generating an expectation of the code word v [8]. Cleary u = u only if (∇ = v). The rule of decoding is based on a mechanism of selecting an estimated code word for each possibly received sequence **r**. If the code word **v** was sent, a decoding error appears if and only if ∇ ≠ v. The conditional error probability of the decoder is defined as

$$P\left({}^{E}/_{r}\right) \triangleq P(\nabla \neq {}^{v}/_{r}) \qquad (3)$$

The following formula shows who to calculate the decoding error probability which is given as

$$P(E) = \sum_r P\left({}^{E}/_{r}\right)P(r) \qquad (4)$$

Where P(r) is independent of the decoding ruled used since **r** is generated prior to the decoding [5]. As a result, the optimal decoding must reduce $P\left({}^{E}/_{r}\right) = P(\nabla \neq {}^{v}/_{r})$ for all r.

Since to reduce P is equivalent to rise

$$P\ (\nabla = v/r\ ), p\ (E\ /\ r\ ) \qquad (5)$$

That is declined for a given **r** by selecting as the code word **v** which can maximize

$$p\left(\frac{v}{r}\right) = \frac{p\left(\frac{r}{v}\right)p(v)}{p(r)} \qquad (6)$$

Since, ∇ is selected to be the most likely code word given that **r** is received. If all data sequences, and all codewords, are equally likely [i.e., p (v) is the same for all v], increasing $P\ (E\ /\ r)$ is equivalent to increasing $p(r/v)$. Far a DMC,

$$P(r/v) = \Pi\, P\left({}^{r_i}/_{v_i}\right) \qquad (7)$$

Where, for a memoryless channel every received symbol relies just on the corresponding symbol sent. Consequently, a maximum likelihood decoder (MLD) is obtained when the decoder selects its estimation in order to maximize $P(r/v)$ [7]. Maximizing $P(r/v)$ is equivalent to increasing the log-likelihood function.

$$Log\ P\left(\frac{r}{v}\right) = \sum r \log p\left(\frac{\mathrm{ri}}{\mathrm{vi}}\right) \qquad (8)$$

Maximum Likelihood Decoder is then not essentially optimum, in case of the codewords are not equally likely. Since the conditional probabilities are $p(r/v)$ must be weighted by the code word probabilities $p(v)$ to specify which of possible codewords has capability of maximizing$p(v/r\ )$. On the other hand, for several systems, the probabilities of codeword are not exactly recognized at the receiver end, resulting in an optimum decoding becomes impossible, and an MLD then can be the best feasible of decoding rule [7].

2.4 Viterbi decoding of convolution codes

Channel decoding is known as the technique of recovering the received information at the destination side once sent through the physical channel. Sequential decoding and MLD or Viterbi decoding are the most common schemes of channel decoding for convolutoinal codes. Viterbi decoding is used to recover the original codeword instead of applying the technique of comparing the received data sequence with each possible sequence which requires a huge number of comparisons to be done. However with (n,k,m) codes, $2^{(k+m-1)(L-1)}$ paths exist over the trellis algorithm, where L is the code frames number which has been considered. The decoding algorithm of convolution codes infers the input information values sequence form the stream of the received distorted output symbols. Three essential families are adopted for decoding algorithms of convolution codes which are, Sequential, Viterbi and Maximum posterior (MAP). Viterbi decoding techniques realizes that there is no need to consider all paths, instead only 2^{m-1} needs to be obtained [9].

Viterbi is typically a method that implements the Maximum Likelihood decoding. The goal of the Viterbi algorithm is to find the transmitted sequence (or codeword) that is closest to the received sequence. As long as the distortion is not too severe, this will be the correct sequence [9].

The Viterbi decoder block works according to the maximum likelihood decoding. It means finding the most probable transmitted symbol stream from the received codeword.
Moreover, The Viterbi decoder defines a metric for each path and makes a decision based on this metric. The most common metric is the Hamming distance metric. When two paths come together on a single node, the shortest hamming distance is kept. The number of trellis branches is defined as trace back depth [10]. To define a convolution decoder in MATLAB®2014 simulation, a poly2trellis function is used to convert convolution code to trellis description.
Trellis = poly2trellis (Constraint Length, Code Generator) [10].

3 MODULATION SCHEME

The modulator must select a wave format with duration of T-sec, which is conceivable for every encoded output symbol. For binary system, the modulator must generate each of signals $s_0(t)$ or $s_1(t)$ to represent 0 and 1 respectively [11]. Therefore, the optimum selection of signal in case of using wide channel is

$$S_0\,(t) = \sqrt{\frac{2E}{T}}\,\sin\left(2\pi f_0\,t + \frac{\pi}{2}\right) \quad 0 \le t \le T \qquad (9)$$

$$S_1\,(t) = \sqrt{\frac{2E}{T}}\,\sin\left(2\pi f_0\,t - \frac{\pi}{2}\right) \quad 0 \le t \le T \qquad (10)$$

This is known as Binary Phase Shift-Keying (BPSK) in which the difference between 0 and 1 is 180 degree as shown in figure 7. For this type of modulation the symbol rate is the same as bit rate.

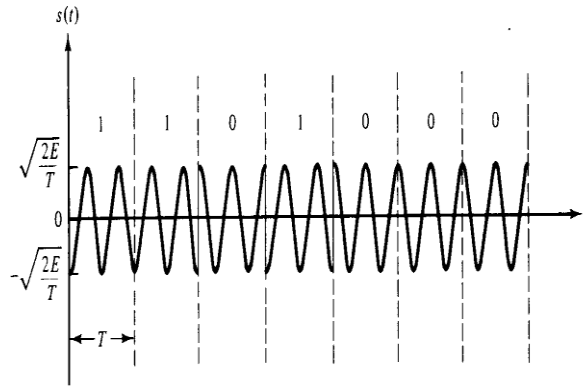


Figure 7 BPSK modulated waveform corresponding to the code word v = (11000)

4 AWGN & DSC CHANNEL

The AWGN channel is a good model for many satellite and deep space communication links. It is not a good model for most terrestrial links because of multipath, terrain blocking, interference, etc. However, for terrestrial path modelling, AWGN is commonly used to simulate background noise of the channel under study, in addition to multipath, terrain blocking, interference, ground clutter and self-interference that modern radio systems encounter in terrestrial operation.

The Additive white Gaussian noise (AWGN) is a common model of noise disturbance exists in any communication system, whereas, in case of the channel experiences the components of multipath, interference or terrain blocking, AWGN does not have the capability to introduce better performance. AWGN is the model which has become familiar with simulating background noise for the channel which is introduced in this paper. If the signal is sent as s(t)=[$s_0(t)$ or $s_1(t)$], the received signal is

$$r(t) = s(t) + n(t) \qquad (11)$$

Where $n(t)$ is Gaussian random process with one-sided power spectral density (PSD).
the demodulator must generate an output corresponding to the received signal for every T-second interval [11].This output may be a real number or one of a discrete set of preselected symbols , according to the design of the demodulator. An optimum demodulator often consists of a matched filter or correlation

detector with a sampling switch. For BPSK modulation with coherent detection the sampled output is a real number,

$$\rho = \int_0^T r\,(t) \int \frac{2E}{T} \sin(2\pi f_0\, \mathrm{t} + \frac{\pi}{2}) dt. \quad (12)$$

Here, for memoryless channel, the symbols are treated independently, meant the output of the detector at an interval will rely on just the sending signal at that interval without dependency of the pervious transmission [11]. For such case, combining M-ary input modulator has the physical channel (DMC). A DMC is totally prescribed by a number of transition probabilities $P(k\backslash i)$, since$0 \le i \le M-1$, 0 ≤ k ≤ N-1, where $\boldsymbol{i}$ is denoted as an input symbol of modulator , **k** represents the output symbols of demodulator . $P(k\backslash i)$ is interpreted as the probability to receive **k** if $\boldsymbol{i}$ was sent. If a communication system in which binary modulator is used (M=2) is considered as an example, the magnitude noise distribution is symmetric, and the demodulator output is quantized to Q=2 levels. In this case, a specific simple and essential channel model, called the Binary Symmetric Channel (BSC) is used. The probability diagram of transmitting for a BSC is shown in Figure 8, taking into account that the transition probability P totally prescribes the channel [11].

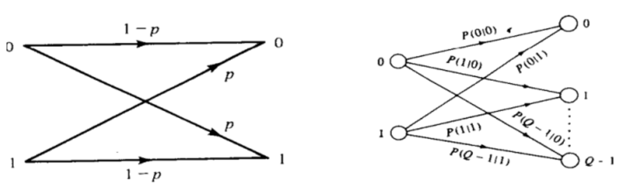


Figure 8 Transition probability diagram

When BPSK modulation is applied over an AWGN channel with optimum coherent detection and binary quantized output, the BSC transition probability is just the BPSK bit error probability for equally likely signal given by

$$p = Q(\sqrt{\frac{2}{N_0}}) \quad (13)$$

Where $Q(X) \triangleq \frac{1}{\sqrt{2\pi}} \int_X^\infty e^{-y^2/2}$

This is the complementary error function of Gaussian statistics. An upper bound on Q(X) is

$$Q(X) \le \frac{1}{2}\, e^{1/-x2} \qquad x \ge 0 \quad (14)$$

When binary coding is applied, the modulator has only binary input (M=2). Similarly, when binary demodulator output quantization is used (Q=2), the decoder has only binary inputs.

In this case, the demodulator is known to have hard decisions. Majority of coded digital communication systems, either block or convolution, use binary coding with hard decision decoding which results in simplicity of implementation when it is compared to non-binary systems [11].

On contrast**,** when Q > 2 (or the output is left un-quantized), the demodulator will perform soft decisions. Therefore, decoder must accept multilevel (or analogy) inputs. Although this will result in complexity for decoder to implement, soft –decision decoding introduces noticeable performance improvement.

The channel is known to have memory when the detector has an output at the interval relies on the pervious transmitted signal.

A fading channel is better example to describe a channel with memory, since in case of multipath transmission there is no independency from interval to interval.

If one encoded symbol is transmitted every T seconds, then the symbol transmission rate (baud rate) is 1/T. In a coded system, if the code rate is R= k/n, where k is information bits correspond to the transmission of n symbols, then data rate of transmission can be calculated as R/T bits per second (bps) [12].

All communication channels can experience signal distortion because of limitations in bandwidth. To reduce the impact of this distortion, the channel should have a bandwidth W of roughly 1/2T Hertz (Hz).

In an un-coded system (R=1), the data rate is 1/T= 2W, and is restricted by bandwidth of channel [12].

In a binary-coded system, with a code rate R<1, the data rate is R/T = 2RW, and is declined by the factor R compared to an un-coded system.

Hence, to keep the same data rate of un-coded system, the binary coded system requires an expansion in bandwidth by a factor of 1/R. This is one of the characteristics of binary coded systems. They therefore need some bandwidth expansion to keep a constant data rate. If there is no availability for additional bandwidth, then binary coding is not feasible, and other ways of reliable communication must be sought [12].

5 DESGIN PARAMETERS AND MEASURMENTS

Bit Error Rate (BER) is considered as the main factor of assessing and evaluating the code performance. It is therefore the common parameter that can assess error correction capability of the implemented codes in wireless communications. This can be obtained by defining the ratio of the number of the bits in error to the total number of the transmitted bits. The bit error rate is impacted by several factors including the noise introduced by the physical channel, noise due to a quantization process, code rate (R), the level of the transmitted power (P_t) as well as the ratio energy per symbol to noise that is introduced as ($\frac{E_s}{N_0}$) and $\frac{k}{n}$ to measure the percentage of code rate [13].

The BER is demonstrated to have a forward proportional to the code rate and an inverse proportional to energy per symbol noise ratio and transmitted power level.

The process is performed as follows; firstly, the encoder encodes the data with code rate ***R*** and sends it through the noisy channel. If the transmitter power level P_t does not vary, this can affect the incoming energy per symbol **E** that reduces to $R \times E$. Hence, the BER is typically measured at the decoder input is bigger than the BER of the data sent without coding. The increment in BER is treated by applying a decoder which can correct errors. Minimizing the BER to more orders of Amplitude can be achieved by selecting the specific forward error correction code.

Coding gain is another factor can evaluate the performance of the code used. It is achieved by calculating the difference in BER obtained by applying error correction codes to that of un-coded transmission [14].

To assess the convolution codes performance over the noisy channel, an Additive White Gaussian Noise (AWGN) channel is used. White noise is a random signal (or process) with a flat power spectral density. Gaussian noise is the noise that statistically has a probability density function (PDF) of the normal distribution. In other words, the values that the noise can take on are Gaussian-distributed. It is most commonly used as additive white noise to obtain (AWGN) [7].

The addition of Gaussian noise to the encoded information is gained by producing Gaussian random numbers with wanted energy per symbol to noise ratio.

The variance σ^2 additive Gaussian noise which has the power spectrum of No/2*(Watts*/Hz is given by

$$\sigma^2 = N_0/2 \qquad (15)$$

If the energy per symbol $\mathbf{E_s}$ is set to 1,
Then:

$$\frac{E_s}{N_0} = \frac{1}{2}\sigma^2 \qquad (16)$$

As a result, the standard deviation is given by:

$$\sigma = \sqrt{\left(\frac{1}{2}\left(\frac{E_s}{N_0}\right)\right)} \qquad (17)$$

The incoming symbols are fed to the Viterbi decoder to achieve the data bits. The decoded data is then compared with the corresponding input given to the encoder and then BER is measured [15].

To measure the probability of bit error rate P_b of a convolution code, the estimate of equation is:

$$P_b < \sum_{d=f}^{\infty} C_d \, P_d \qquad (18)$$

Where $\mathbf{C_d}$ is the summation of bit errors for error events with distance **d**, and **f** is the free distance of the code. The quantity $\mathbf{P_d}$ is the pair wise error probability, given by:

$$P_d = \frac{1}{2} erfc[\sqrt{dR\frac{E_b}{N_0}}\ ] \tag{19}$$

Where **R** is the code rate and **erfc** is the complementary error function, defined by:

$$erfc(x) = \frac{2}{\sqrt{\pi}} \int_x^{\infty} e^{-a} \tag{20}$$

By calculating the shortest free distance of the code, the performance of the convolution code can be determined, where d_{free} is the minimum Hamming distance between any two codewords. It is also called free distance [15] & [16].

5.1 Performance Analysis

The performance analysis generally depends on the error event. Suppose that the transmitted code sequence is **y (D)** and the received code sequence is **y' (D).** Therefore, each sequence determines a unique path through a minimal code trellis. These routes will agree for long periods of time, however will have disagreement over particular finite intervals. An error event corresponds to one of these finite intervals. It starts at the time when the path **y' (D)** first diverges from the path **y (D),** and ends when these two paths merge again. The error sequence is the difference (over this interval):

$$e(D) = y'(D) - y(D)$$

The BER is used to calculate the coding gain, which is the measurement of the difference between signal- to-noise ratios (SNR) between the un-coded systems and the coded system to hit the same level of BER [5]. Convolution coding is performed by simulations in MATLAB®2014 by following the steps as illustrated in Figure 9.

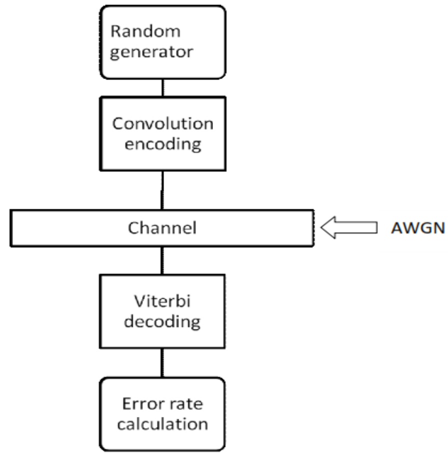


Figure 9 Convolution Coding Flowchart

- Data generation: The information to be sent over the channel is produced using random integer generator. It produced a uniform random distribution integers in the range of [0, M-1]
- Convolution encoding: This is obtained by the recall the function known poly2trellis. It accepts a polynomial description of a convolution encoder and returns the corresponding trellis structure description. The parameters for the poly2trellis function are "constraint length" and "code generator polynomial".
- Adding noise to the transmitted symbols: The AWGN function adds White Gaussian Noise to the channel symbols produced by the encoder. The parameters for this function are the coded symbols, SNR (signal to noise ratio), the state, and the power type (whether in "dB" or "linear").

- Decoding: Viterbi algorithm is used to extract convolutionally encoded information.

Calculation of error rate: it calculates the number of errors, symbols and the bit error rate.

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It calculates the error rate of the received information by comparing it to a delayed copy of the data sent [14].

6 SIMULATION RESULTS

Computer simulation was applied in order to calculate and evaluate the BER. Simulation displays the result of the error correction by implementing convolution codes which are based on Bit Error Rate (BER) performance over a range of signal-to-noise ratios (Eb/No). The error rates of the received data were computed by comparing it to a delayed copy of the transmitted data. This comparison (as in Table 1) was done for the encoded system and others with code rates 1/2 and 1/3 respectively.

Table 1: constraint length and generator polynomials of 1/2 and 1/3 code rates

Rate	Constraint Length	Generator Polynomial 1 (octal form)	Generator polynomial 2 (octal form)	Generator polynomial 3 (octal form)
1/2	3	6	7	–
1/3	3	4	5	7

Table2: Simulation results

	Coded System						Uncoded System		
S/N0	Code rate ½			Code rate 1/3					
	BER	No. of Errors Detecte	No. of Symbols Compare	BER	No. of Errors Detecte	No. of Symbols Compare	BER	No. of Errors Detected	No. of Symbols Compared
1	0.05	2	40	0	0	40	0.575	23	40
2	0.0488	2	41	0.02	1	41	0.585	24	41
3	0.0476	2	42	0.05	2	42	0.571	24	42
4	0.0465	2	43	0.05	2	43	0.581	25	43
5	0.0455	2	44	0.05	2	44	0.568	25	44
6	0.0444	2	45	0.04	2	45	0.578	26	45
7	0.0435	2	46	0.04	2	46	0.565	26	46
8	0.0426	2	47	0.04	2	47	0.574	27	47
9	0.0417	2	48	0.06	3	48	0.583	28	48
10	0.0408	2	49	0.06	3	49	0.571	28	49

To discuss the above results, it is clearly demonstrated that an increase in number of symbols results in an increase in errors. The initial values of the BER for the 1/2 and 1/3 codes are 0.05 and 0.0 respectively. These decrease moderately to the end points of 0.0408 and 0.06. As compared to the encoded system which has a BER start point of 0.575 and end point of 0.571, this is about 50% more than the coded system. Monte-Carlo technique was used in draw diagrams/ graphs of bit error-rate (BER) versus signal to noise ratio (E_b/N_o) in order to further characterize the performance of the convolution codes. The simulation results are shown in the Figure 10 and Figure 11. As demonstrated, the vertical axis represents the bit error rate (BER) performance; whereas the ratio of energy bit to noise spectral density (E_b/N_o) is plotted on horizontal axis. The (E_b/N_o) varies between 0 and 20 dB. It can be seen that the curves depend upon E_b /N_o. The curves show that BER decreases with an increase in E_b/N_o. the significance of the simulation results can be illustrated as follows: With a rate 1/2 and 1/3 convolution coding and a constraint length of 3, a data signal can be transmitted with at least 3 dB less power. This in turn contributes to reducing the transmitter or antenna expenses or permits increased data rates for the same transmitter power and antenna size.

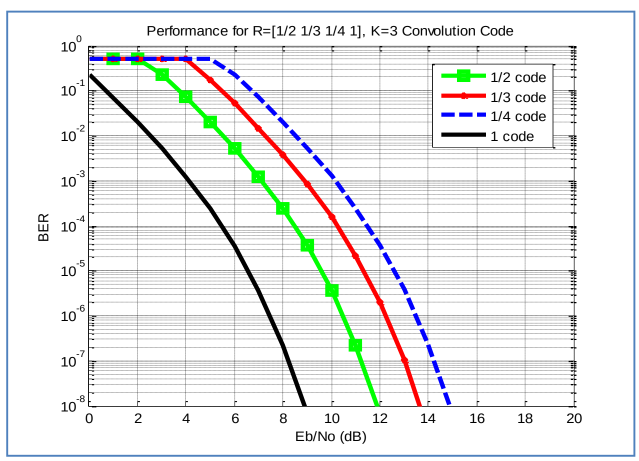

Figure 10 Simulation result for1/2, 1/3, 1/4, 1 convolution code using BPSK modulation

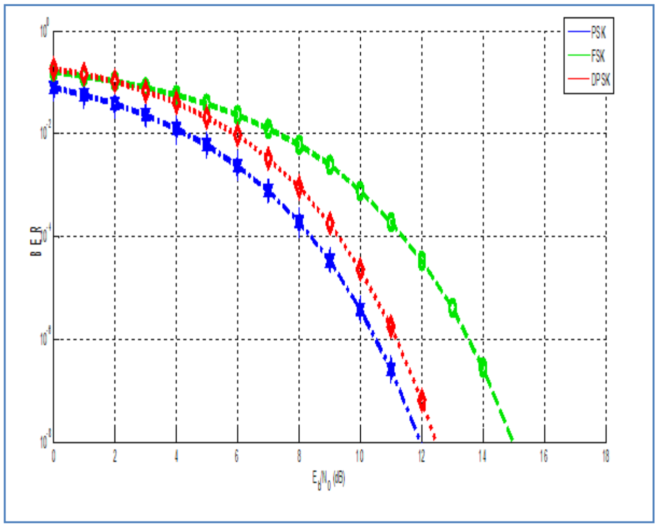

Figure 11 Simulation result for 1/2, 1/3 and uncoded system for BPSK

Higher coding rate results in a less BER at the expense of more bits being processed and transmitted. When more bits are being processed, the processing time will be larger. However, a rate 1/2 and 1/3 coding results in an increase of bandwidth by a factor of 2 and 3 respectively. In general, the bandwidth expansion factor of a convolution code is simply n/k, where k/n is the code rate which is defined as the ratio of the number of bits in the convolutional encoder (*k*) to the number of channel symbols output by the encoder (*n*) in a given encoder cycle. Slower transmission (rate <1) is a high price paid for almost error-free transmission and a code defines a trade-off between the two.

7 CONCLUTION

It is clearly summed up that the forward error correction codes are techniques which work properly over AGWN channel. Convolution code is commonly used to detect and correct errors of the received signal in digital wireless communication system. Comparison has been made between the coded and encoded bit error rate which results in validating the convolution codes simulation. According to the results from the graphs, a 70% and 74% improvement shown for coding gains between reference points of 10^{-2} and 10^{-4} for the two code rates used. This improvement can be attributed to the introduction of the convolution codes. In general, it is noticed that the introduction of the convolution coding has helped to significantly decrease the bit error rate which will result in the transmission of signals of specified quality with a smaller transmit power. This in turn leads to higher power efficiency, whereas the bit rate is half or third that of the encoded scheme thus lowering bandwidth efficiency. As a result, a coded system offers better channel efficiency than the encoded system.

8 REFERENCES

[1] G. Andrea , *Wireless Communications,* Cambridge University Press, United Kingdom, 2005.

[2] J. G. Proakis, *Digital Communication*, fourth Edition, McGraw Hill, 1998.

[3] B. A. Forouzan, Data Communications and Networking, Fuorth Edition, McGraw Hill

[4] O. Eriksson, E. Bjornemo, A. Ahlén, and M. Gidlund, "On hybrid arq adaptive forward error correction in wireless sensor networks," in *IECON 2011-37th Annual Conference on IEEE Industrial Electronics Society*, 2011, pp. 3004-3010.

[5] H. Kumawat and S. Sharma, " An Implementation of a Forward Error Correction Technique using Convoltuion Encoding with Viterbi Decoding", International Journal of Soft Computing and Engineeirng, vol. 2, no.5, November 2012 [ISSN:2231-2307].

[6] Balakishnan, Gopinath, et al, "Performance analysis of error control codes for wireless sensor networks". Information technology, 2007. ITNG'07. Fourth international conference on. IEEE, 2007.

[7] Yunghsiang S.Han Craduate ''introduction to binary convolution code'' institute of communication engineering National Taipei university Taiwan.

[8] John M. Geist, " Viterbi Decoder Performance in Gaussian Noise and Periodic Erasure Bursts," *IEEE Transactions on Communication,* vol. com-28, no. 8, pp.1417-1422, August1980.

[9] Shu Lin and Daniel J. Costello Jr., "error control coding, fundamentals and applications", Second Edition, Prentice-Hall,Inc., 1983 .

[10] Nabeel Arshad1 and Abdul Basit ,"Implementation and analysis of convolutional codes using MATLAB", International Journal of multidisplinary Sciences engineering, VOL. 3, NO. 8, August 2012 [ISSN: 2045-7057**]** www.ijmse.org 9

[11] A. J. Viterbi, "Convolutional Codes and their Performance in Communication Systems", IEEE Transactions on information Theory, Vol. IT-19, No. 5, PP 751-772, Oct 1971.

[12] M. Divya,"Bit Error Rate Performance of BPSK Modulation and OFDM-BPSK with Rayleigh Multipath Channel" International Journal of Engineeirng and Advanced Technology, vol.2, no.4, April 2013 [ISSN:2249- 8958].

[13] Ali Calhan, Celal Ceken and Ismail Ertruk, "Comparative Performance Analysis of Forward Error correction Techniques used in Wireless Communications", Proceedings of the third International Conference on wireless and Mobile Communications,0-7695-2796-5/07.

[14] Davoud Arasteh, " Teaching Convolutional Coding Using MATLAB in Communications Systems Course" Proceedings of the 2006 ASEE Gulf-Southwest Annual Conference.

[15] Sneha Bawane and V.V. Gohokar, " Simulation of Convolutional Encodes", IJRET: International Journal of Research in Engineering and Technology ISSN: 2319-1163 | pISSN: 2321-7308.

[16] Y. Yorozu, M. Hirano, K. Oka, and Y. Tagawa, "Electron spectroscopy studies on magneto-optical media and plastic substrate interface," IEEE Transl. J. Magn. Japan, vol. 2, pp. 740-741, August 1987 [Digests 9th Annual Conf. Magnetics Japan, p. 301, 1982].