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Performance Evaluation of Channel Codes for High Data Rate Mobile Wireless System

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Abstract

Mobile wireless system incorporates dual sphere in terms of mobility as well as computing. Such systems have revamped momentously resulting in expedient wireless devices. Current wireless communication requires enormous data rates, streaming audio/video, asymmetric access/multiple access, fully-integrated, low cost, reliable seamless systems with the capability of providing voice, high speed data, and video with minimum latency and long battery life to enhance error free communication. However, there are many obstacles and bottlenecks to achieving this in practice. These bottlenecks include limited spectrum, limited energy, multipath fading and propagation loss, cost, standardization processes. This paper focuses on coding and modulation techniques for wireless communications so as to reduce/overcome many of these obstacles. To put the discussion in perspective, a short background on the coding and modulation approaches of wireless communication systems is provided. The turbo coding is mainly chosen to facilitate the wireless communication. The paper enlightens the impetus behind using turbo codes for high data rate mobile wireless systems. Paper explains how better out-turns can be yielded when approaches like cyclic redundancy check and scrambling are embedded with coding. The BER performance and SNR ratio are observed for illustrating variance between convolution and turbo coding for different rates and modulations. Simulation results are plotted using Matlab software.

Index Terms: Convolution code, Turbo code, Modulation, AWGN Channel.

1. Introduction

Rise and shine of advanced radio transmission technologies has intensified speculations of the user in terms of data rate and various efficiencies. Data transferal reinforcing online activities of user is rapidly growing. Certainly some hiccups are there for transferring data in intact form. Factors like atmospheric ducts,

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ionospheric reflection/refractions or reflection from any kind of other body results in multipath propagation. Consequence is interferences or fading corrupting the transmitted data. This flaw can be axed with efficient channel coding approach. This approach is imperative for wireless communication. By appending redundant bits with the information to be transmitted, channel coding refines link performance of wireless set-ups. With the aid of such techniques, errors in the transmitted sequence can be detected as well as corrected [1][2].

Diverse approaches can be utilized according to in-hand application. This paper focuses on two magnificent such techniques namely convolution coding and turbo coding. These codes belong to the class of forward error correction (FEC) codes. Turbo codes have propensity to furnish high performance closer to channel capacity construed by Shannon capacity theorem. Convolution codes (CC) or intuitive codes are elementary units of turbo codes. CCs are extensively used for real time error correction [3]. These codes have some gaps which are filled by its advanced counter-part turbo codes. Near optimum performance can be expected applying turbo codes using various modulation schemes for wireless communication systems.

The paper is devised as follows: Section 2 introduces the necessary concepts about encoding and decoding incorporated in convolution and turbo codes. Cyclic redundancy check and scrambling are explained in sub-section 2.5 and 2.6 which when added to coding schemes enhances the outputs. Modulation schemes used for presented scenario are explained briefly in section 3. Next section gives an overview about AWGN (Additive White Gaussian Noise) channel. Section 5 represents simulation methodology and environment used for investigations and section6 delineates corresponding results of proposed model and their discussions. Lastly, paper is being concluded and future scope is given.

2. Channel Codes

Channel codes inhabit at the heart of digital communication. It's because of these channel codes that receiver becomes proficient enough to detect as well as correct errors itself. Thus retransmission can be avoided which then circumvents unnecessary delays. Successive subsections describe about two of such efficient codes:

2.1. Convolution Code

Elias pioneered convolution codes [4]. These magnificent codes are instinctive due to the verity of several ways to understand them. Only parity bits are transmitted in these codes. Amongst the set of possible transmitted data, decoder retrieves the most likely one. It is one of the most authentic ways for error-free data transferal and receival. Though convolutional codes are intricate with harder implementation and lower code rates (usually below 0.90) [5], as compared with linear block codes but have effective error correcting potential.

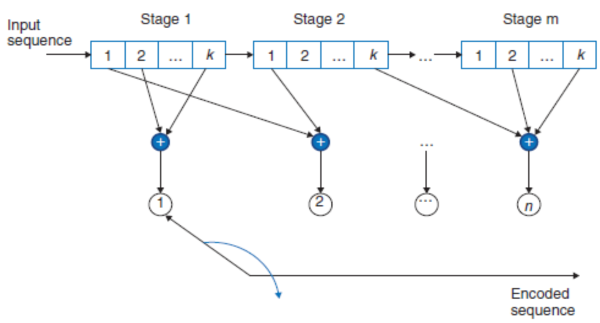

Fig.1. Convolutional Encoder

Three parameters control operation of convolutional codes; number of bits at output/input, constraint length and number of memory registers. Ratio of input to output bits gives the code rate which measures code efficiency. Number of k-bit stages which are present to supplement combinatorial logic giving output symbols is stated by constraint length (‘K’) [6]. There is parameter ‘m’ furnishing information about memory length of encoder. Convolutional encoder composes tapped shift registers and mod-2 adders as shown in Fig.1. Shift registers have tap gains which are basically binary digits. These codes are created with convolution of input sequence with impulse response of encoder. It operates on a continuous data stream. Input samples are fed to encoder where shift registers store them temporarily. Further, they are combined in a known manner applying modulo-2 addition to obtain encoded output. Thus, by operating on current input as well as previous ones desired output is generated.

2.2. Viterbi Decoding

A Viterbi decoder employs the Viterbi algorithm for decoding a bit stream as counter part of forward error correction based on a convolutional code. The process of distinguishing original data from the corrupted received one can be done using the diagram called "trellis". Decoder examines the whole received sequence and suitable path with the smallest hamming distance is discovered. Amongst various survival paths, the Viterbi algorithm utilizes maximum likelihood principle to limit the differentiation. For each node, decoder examines two metrics that is path and branch metric. The decoding complexity goes on increasing linearly with branches and exponentially with encoder’s memory. Viterbi decoding can be based on hard/soft decision depending on whether its input is in the form of bits or log-likelihood ratios (LLRs). LLRs are probability measures are based on the logarithm of the likelihood of correct data reception instead of manipulated data. In hard-decision decoding, quantization of symbols is done up-to one bit precision whereas for soft-decision case multiple bits (three or four) of precision are utilized for quantization of one bit. As performance of link is influenced directly by choice of decision criteria thus, appropriate selection of quantization level must be done [3].

2.3. Turbo Coding

Turbo codes can work efficiently at reduced power levels rendering improved safety and extended battery life, both of which are significant to the mobile equipment. It is theoretically possible to reach the Shannon limit with convolutional codes of larger constraint length. But limitation in terms of processing power needed to decode such long codes creates barrier making such approach impractical. Turbo codes with the help of recursive coders and iterative soft decoders shatter this barrier.

At one end with recursive coder appropriate constraint lengths can be generated and on the other end, with each iteration better output from corrupted input can be obtained. At the turbo encoder, numbers of recursive systematic convolutional (RSC) codes are arranged in parallel concatenation [7] as shown in Fig.2.

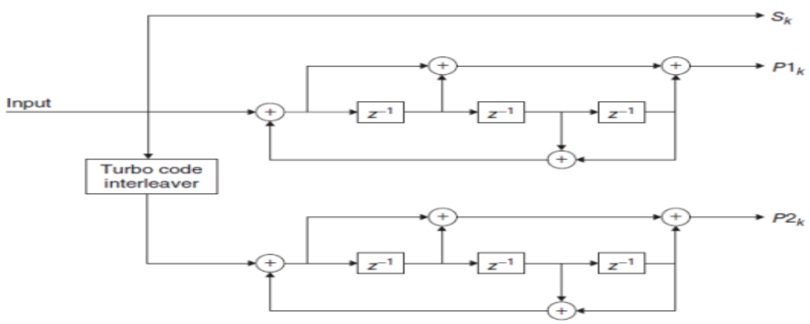

Fig.2. Turbo Encoder

Generally, two RSCs are connected in the set-up. More number of encoders elevates complexity and overhead. At the output of encoder three data streams are procured; one is the input itself and other two are the parity bit streams. One of the RSC is fed with basic data bits and input to other RSC is an interleaved version of the systematic input. The interleaver design has an appreciable effect on code performance [8]. Codes with lower weight result in poor error performance. It becomes significant to construct codes with effective weights which mean if one of the encoder is generating low weight output then it is mandatory that other must generate good code weight to enhance performance of arrangement. Outputs of two encoders are thus time displaced. The outputs of the two coders may be multiplexed into single sequence giving a rate R=1/3 code, or they can be punctured to give a rate of R=1/2 code.

2.4. Turbo Decoder

Operations of turbo encoder are inverted at decoder end. Turbo decoder configuration incorporates two A Posteriori Probability (APP) decoders along with two interleavers in a feedback loop. MAP (Maximum A-Posteriori) and SOVA (Soft Output Viterbi Algorithm) algorithms fits in this category of turbo decoders. Each component decoder outputs extrinsic information with soft bits. This information specifies the likelihood of a bit that whether it is 0 or 1. Decoder operates iteratively. In the first iteration, channel likelihoods are given to each trellis and forward-backward algorithm runs next. Since it is the basic iteration, a-priori information is not known at this step so it is either ignored or set to zero. These likelihoods are forwarded from one trellis to other multiplying channel likelihoods in between. A Log Likelihood Ratio (LLR) simplifies the passage of information between the different components incorporating the decoder. LLR is basically logarithmic ratio of probability of a bit being 1 to that of bit being 0. [9]

The soft output from first encoder computes extrinsic information which is provided as a-priori information to second decoder. Now this second decoder has an advantage of having this a-priori information and using this data respective bits are estimated. From second decoder, output is folded back to first decoder for second iteration. The forward-backward algorithm is followed again in each trellis with updated probabilities. The iterative process repeats until either the decoding has converged or the set number of iterations has been hit.

2.5. Cyclic Redundancy Check (CRC)

For accurate outcomes iterative decoding is necessary. But along with accuracy with increased iterations proportionally decoding complexity expands too. This flaw and that of corresponding delay in turbo coding with increased iterations can be removed using early termination method. This method involves CRC binding using CRC generator polynomial with information to be given to turbo encoder. CRC checking syndrome is attached with data at the input end of turbo encoder [10]. At the decoder side, with ending of each iteration, algorithm can detect whether any error is present or not. If data is error-free then rather following set number of iterations decoder comes out of loop and stops decoding earlier. Thus, we have dual advantage; reduction in computational complexities and time, that too without affecting performance of system.

2.6. Scrambling

The geographical region has been cleaved into umpteen smaller numbers of cells. To indiscriminate the interference between different cells scrambling is the approach followed in current high data rate wireless systems [11]. The positive facet is that intrusion from cells can be avoided prior to decoding using distinct scrambling sequences. Depending upon the physical cell identity, cells utilize unique sequences for scrambling originated with sequence generators. Preferred pseudo random sequence is produced by Gold sequence of varying lengths. After polynomial pair generation, exclusive-or operation is performed between incoming informative bits and Gold sequence bits. At the descrambler two alternatives are there depending on whether hard or soft-decision demodulation is performed.

3. Modulation

The progression to digital modulation furnishes more information capacity, affinity for digital data services, more data security, better dissemination, and expeditious system availability. Bandwidth, admissible power and noise are the factors causing hindrance. More bits per symbol in modulation technique results are more competent in confronting the errors [14]. Diverse digital modulation forms are there like amplitude shift keying (ASK), frequency shift keying (FSK), Phase shift keying (PSK) and quadrature amplitude modulation (QAM). Quadrature Phase Shift Keying (QPSK) can be viewed as effectively two independent BPSK systems (I and Q). Two bit transferal takes place in a single modulation symbol resulting in twice the bandwidth efficiency. It is more spectrally efficient. But its receiver complexity is high. Compared with BPSK, QPSK has potential to either doubling the data rate keeping same bandwidth or preserving data rate halving the bandwidth. Large envelope alterations occur during phase transitions, thus necessitating linear amplification. Its constellation diagram is shown in Fig.3.

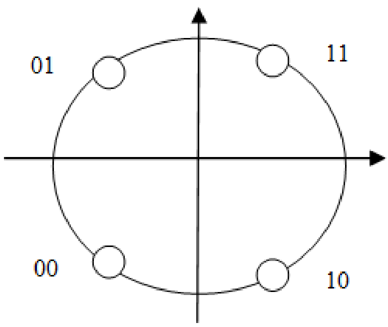

Fig.3. Constellation diagram of QPSK

In M-ary PSK modulation, the amplitude of the transmitted signal was restricted to remain constant, thereby returning a circular constellation. By relaxing these amplitude constraints when both amplitude as well as phase is empowered to vary, a new modulation scheme called Quadrature Amplitude Modulation (QAM) originates. [12] Various QAM classifications are there. Sixteen-state Quadrature Amplitude Modulation (16QAM), integrating four I values and four Q values. Four bits per symbol can be sent. With better spectral efficiency in this modulation format symbol rate is one fourth to that of bit rate. It surpasses BPSK, QPSK, or 8PSK. No doubt higher level M-ary schemes (such as 64-QAM and 256-QAM) are very bandwidth efficient, but due to reduced distance between symbols, signal becomes more vulnerable to noise and linear amplification becomes necessity. Such a signal may have to be transmitted with additional power [15] (to effectively widen the symbols out more). This in turn lowers the power efficiency of higher schemes as compared to other simpler ones.

4. Additive White Gaussian Noise (AWGN) Channel

It is the basic model of deformities that communicating channel may inhere. In communications, AWGN channel model has deterioration which is the linear addition of wideband or white noise. Such noise is equally there with indistinguishable power at every frequency. It has got constant spectral density [15] and amplitude is Gaussian distributed. With respect to signal in a communication channel, AWGN is the statistically random radio noise specified by a wide frequency range. However, it provides amenable mathematical models that are helpful for acquiring apprehension in the underlying behavior of system before consideration of phenomenon like dispersion or multipath fading. Thus, AWGN is utilized for background simulation of noise in the channel under consideration [13]. When signal is passed through this channel white Gaussian noise is added to it. Due to this, transmitted signal gets manipulated [3]. Practically noise is even more complex but still this model is effective when simulations are performed for amplifier/background noise.

5. Simulation Methodology and Environment

Simulations of different combinations of coding, rate and modulation have been executed using MATLAB software. The process has been stated as: Input bits are fed to the encoder. In one case encoder is convolutional and other case includes turbo coder. The convolution encoder maps the incoming data stream into corresponding unique output which we call as code-word. This encoded data is then passed through modulator which in turn produces modulated data stream. Next step is data transmission via AWGN channel (see Fig.4a). According to channel characteristics, information will get manipulated passing through it.

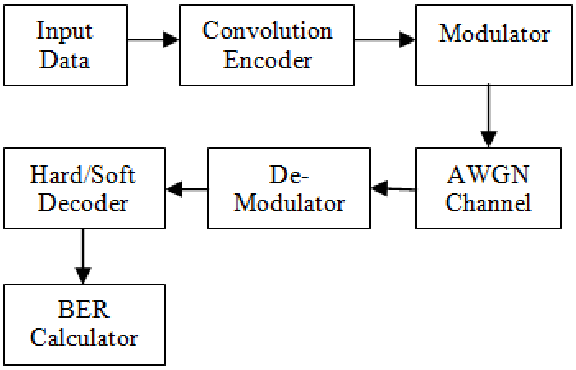

Fig.4a. Proposed simulation model for Convolution codes

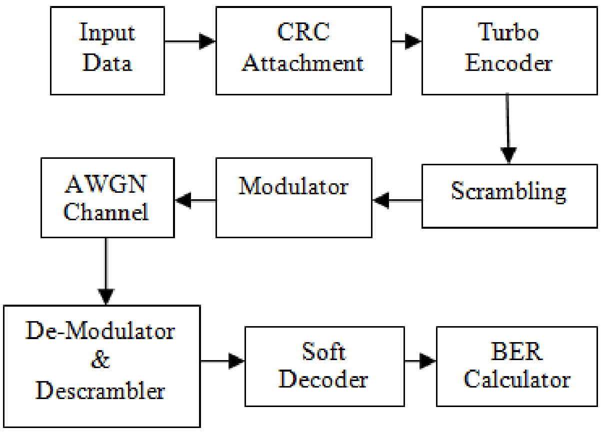

Fig.4b. Proposed simulation model for Turbo codes

While using turbo coding, variation in proposed model lies in the fact that CRC is attached to input data before being applied to encoder. Secondly, after this channel coding, scrambling is exercised which helps in combating interference between cells (see Fig.4b). At the receiver end, inverse operations are performed. After signal detachment from carrier i.e. after demodulation data is given to decoder. Two scenarios exist here; for hard decision Viterbi decoding, demodulator translates received data into bits which are assigned for furnishing relevant information. Whereas in soft decision decoding, log-likelihood ratios are obtained from demodulator for yielding required error-free data. In convolution coding both scenarios are applied and for turbo only soft-decision decoding is utilized. Additionally for turbo decoding descrambler is applied with demodulator. At the end bit error rate is calculated in both proposed models. Parameters wielded for simulations regarding proposed models are given in table1 below.

Table1. Parameters used for simulation

Encoding schemes	Convolution & Turbo encoding
Decoder	Soft and hard-decision Viterbi
Modulations	QPSK, 16QAM, 64QAM
CRC	24-bits
Channel	AWGN
For Scrambling	Gold Sequence
Iterations for turbo decoder	1 to 6
Code rate	1/ 2, 1/3, 2/3, 3/ 4
Block Length	512 and 2048

6. Results and Discussion

In this section plots for various BER vs. Eb/No values have been presented for all the mandatory modulation and coding profiles using proposed model. Plot shown in Fig. 5 represents performance of convolution codes for QPSK modulation using hard-decision Viterbi decoding at the decoder end. Different rates are used for analysis. It can be seen that performance improves as rate increases from 1/3 to 3/4. But at the same time redundant bits are increased too which in turn increases complexity and delay. So there is a trade-off between desired performance and timing/complexity restrictions.

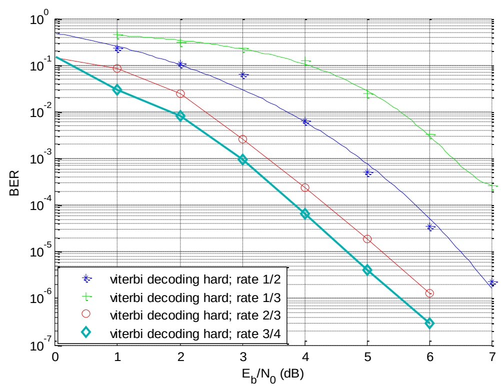

Fig.5. BER of hard-decision Viterbi decoding for different data rates using QPSK modulation

In Fig. 6, performance analysis has been done for soft-decision Viterbi decoding for varying rates. In this case too performance enhancement can be visualized with increased rates. At any target BER, higher rate (3/4) is giving better performance at lower Eb/No than lower rates at same Eb/No. If BER for soft and hard outcomes are compared (see table2) it can be clearly observed that there is a drastic improvement in softer decoding. Difference is more significant for lower rates.

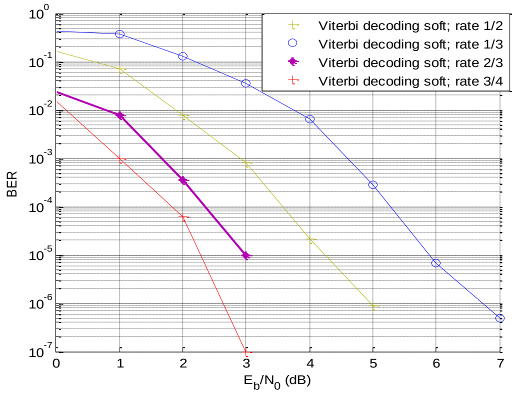

Fig.6. BER of soft-decision Viterbi decoding for different data rates using QPSK modulation

At target BER of 10^{-3} for hard-decision and soft decision Viterbi decoding, required Eb/No for data rate 1/3 is 6.3 and 4.5 dB respectively. For data rate 3/ 4 respectively Eb/No values are 3 and 1 dB. Thus, soft-decision Viterbi decoding has better denouements. (See Table2)

Table 2. Performance comparison of hard and soft decision Viterbi decoding at target BER of 10^{-3}

	Eb/No (dB)	
Data rate	Viterbi hard	Viterbi soft
1/3	6.3	4.5
1/ 2	4.9	2.9
2/3	3.3	1.8
3/4	3	1

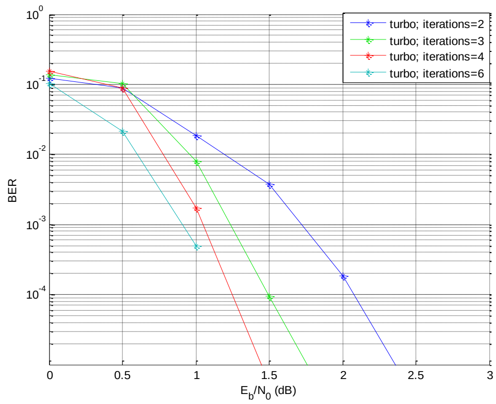

Fig.7. BER of turbo coding for different iterations using QPSK modulation with data rate 1/3

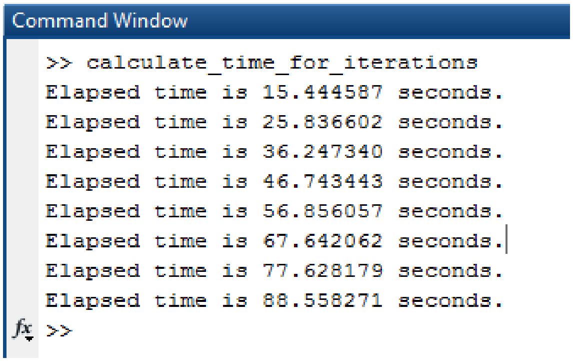

Fig.8. Timing comparison for iterations (1 to 8) used in turbo coding

Using data rate 1/3 and QPSK modulation, BER performance for turbo codes using soft decision decoding has been shown in Fig. 7. Four iterations are used for analysis; 2, 3, 4 and 6. As the numbers of iterations are increased from 2 to 6 there is a mesmerizing refinement in outcomes. But a price has to be paid for such optimum accomplishment. This price is in the form of delay. With increase in iterations proportionally delay is increased too. This can be seen in Fig. 8. From iteration 1 to iteration 8, time elapsed ranges from 15.44 to 88.56 second. Moreover, from the profile view it can be observed that approximately 84% of the total time at the receiver is consumed by decoder. If we go on increasing iterations latency will be increased proportionally.

Three cases are compared in Fig. 9: QPSK, 16-QAM and 64-QAM applying 2 and 6-iteration scenario. It has been observed that for any BER say 10^{-3} in the case of QPSK requisite Eb/No is 1.65dB and for 16-QAM and 64-QAM Eb/No is 3.5dB and beyond 5dB respectively. That means QPSK is performing better than higher modulations. For cleaner channels, higher modulations are more efficient. But as the channel impairments increase there is a downfall in output quality for higher modulations. Efficiency of lower modulations is more for deteriorating channels.

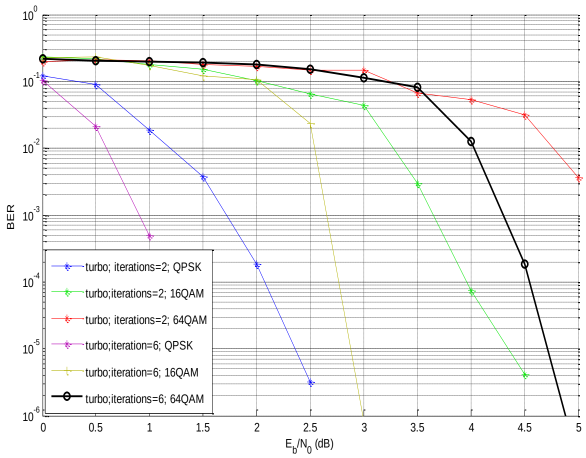

Fig.9. BER of turbo coding for 2 and 6 iterations using QPSK, 16-QAM & 64-QAM modulation schemes with data rate 1/3

Table3 represents clearly the difference between using lower and higher modulation schemes at different iterations. For 2 iterations and target BER of 10^{-3}, QPSK requires lesser Eb/No in dB than higher QAMs. Further with increase in iterations, more improvement can be seen.

Table 3. Performance comparison of different modulation schemes for various iterations using Turbo coding at target BER of 10^{-3}

Iterations	Modulation	Eb/No (dB)
2	QPSK	1.65
2	16QAM	3.6
2	64QAM	Beyond 5
6	QPSK	0.95
6	16QAM	2.65
6	64QAM	4.35

Further when outcomes for convolution with soft Viterbi decoding and turbo codes are compared for 1/3 data rate, certainly turbo codes outperforms convolution codes which is evident from Fig. 10. With increase in number of iterations, performance gap increases and turbo codes start reaching near Shannon Limit.

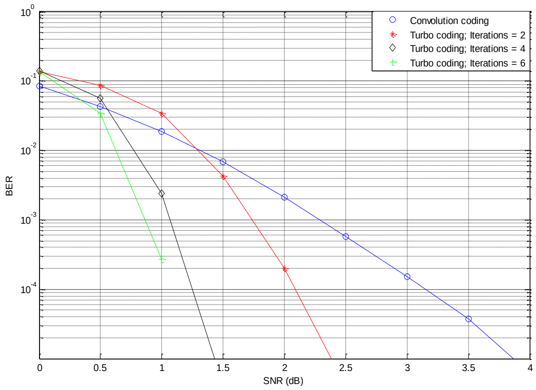

Fig.10. Comparison of BER of convolution and turbo coding using QPSK modulation and 1/3 rate

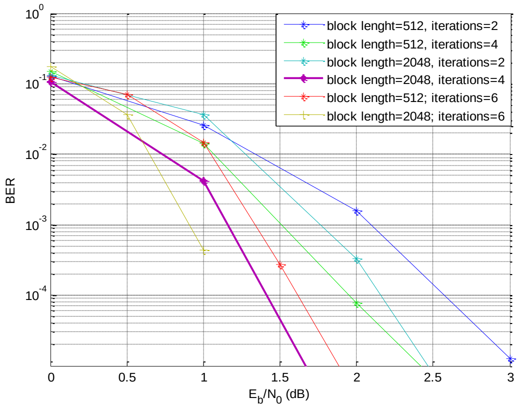

Fig.11. BER of turbo coding for different iterations and block lengths using QPSK modulation with data rate 1/3

Block length is the parameter next whose variation in turbo coding has been shown in Fig. 11. Just like iterations, with increase in block lengths too superior outputs are expected. Two block lengths have been used 2048 and 512. Using combinations of block lengths and iterations BER performance is shown. For target BER of 10^{-3}, blend of higher block length (2048) as well as iteration (4) surpasses other compositions used for analysis. But again, one cannot increase block length indefinitely. With block length size, processing becomes bulky and time-hungry. Thus, depending on transmission conditions and prerequisites block length can be chosen. Size of block length must not upsurge that of interleaver. Input data to encoder is fed in the form of blocks. The block length must not exceed to that of interleaver's.

Table 4. Performance comparison with different block lengths and iterations using turbo coding at target BER of 10^{-3}

Iterations	Block Length	Eb/No (dB)
2	512	2.1
2	2048	1.75
4	512	1.5
4	2048	1.15
6	512	1.35
6	2048	0.9

Table4 is representing the gap in performance when different block lengths were applied for different iterations. It can be observed that for higher block lengths as well as iterations, better results are obtained.

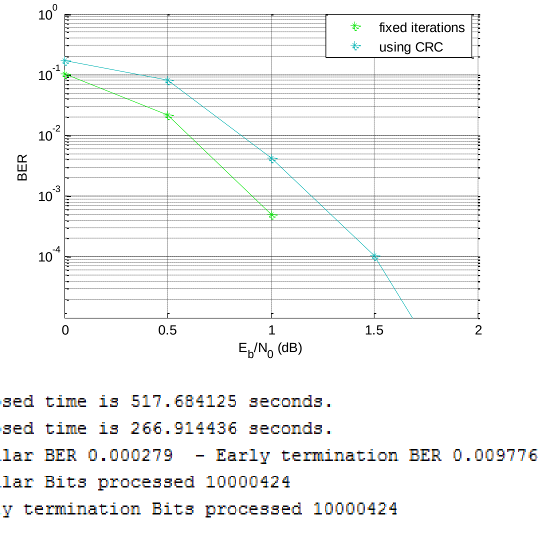

Fig.12. BER and elapsed timing of turbo codes using CRC and without CRC

Fig. 12 illustrates BER performance in the presence and absence of CRC. It can be seen that for same number of bits processed with CRC and without CRC, no doubt BER for fixed number of iterations is better but elapsed timing improves greatly for CRC (266.91 sec) approach. Choice of applying either scenario depends on pre-requisites that whether speed is more important or the quality. Trade-off is there. Decoder does not undergo specified number of iterations and comes out of loop as soon as error-free data is obtained. This is how this approach of early termination helps decoder in reducing its iterations, computational complexity as well as time.

From Fig. 13 it is evident that with increase in length of first polynomial or that of unit impulse function of first polynomial of Gold-sequence there is an improvement in BER. At same Eb/No, for increased length of 30 from 15 gives better performance. Though the gap is not very large for lower Eb/No values but certainly there is a significant performance upswing with increased values of Eb/No. that means dual advantage is what one can get from scrambling. System can circumvent interference from cells as well as increase in length of first sequence ameliorates BER.

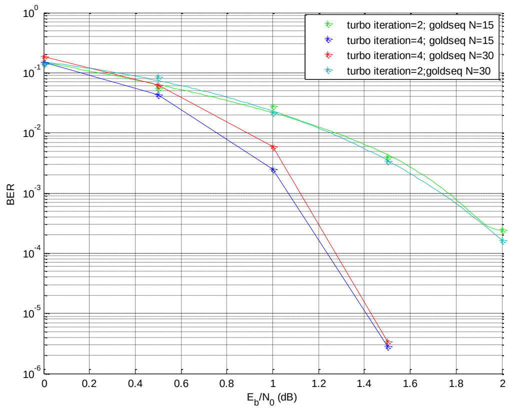

Fig.13. Comparison of BER of turbo coding using QPSK modulation for different gold sequence and iterations

7. Conclusion and Future Scope

In this paper performance of convolution and turbo codes was simulated through the AWGN channel. Different data rates, iterations, block lengths and modulations have been used for analysis. Multiple annotations are there. Firstly, amongst hard and soft decision Viterbi decoding, latter excels the former one with increase in data rate. It has been discerned that turbo codes transcend convolution codes with increase in number of iterations. But simultaneously there is magnification in timing too. Thus, trade-off remains between stipulations. Amidst various modulations for noise constraints, QPSK is preferred over higher schemes. This is because for higher modulations it becomes difficult for decoder to distinguish between relevant symbols and noise. With block length escalation along with iterations Eb/No refines even more. Turbo codes have some inherent gaps of complexity and delay which can be reduced to good extent using methods like early-termination. Withal, fastening the scrambling approach helps further in alleviating interference from diverse cells. So, for wireless systems turbo codes give magnificent outcomes. In future, research can be extended for concatenated codes which are definitely expected to provide better upshots.

References

[1] Samridhi and Dr. Malhotra Jyoteesh; Investigation of Channel Coding Techniques for High Data Rate Mobile Wireless Systems; International Journal of Computer Applications (0975 – 8887) Volume 115 – No. 3, April 2015.

[2] Samridhi and Dr. Jyoteesh Malhotra; Performance bounds in terms of QoS parameters in LTE mobile wireless system; International Journal of Advanced Research in Computer and Communication Engineering; Vol. 4, Issue 1, January 2015.

[3] Sanjeev Kumar, Ragini Gupta. Performance Comparison of Different Forward Error Correction Coding

Techniques for Wireless Communication Systems; IJCST Vol. 2, Issue 3, September 2011.
[4] Ripple Dhingra and Danvir Mandal. Convolutional Code Optimization for Various Constraint Lengths using PSO; International Journal of Electronics and Communication Engineering. ISSN 0974-2166 Volume 5, Number 2 (2012), pp. 151-157.
[5] Mr.Vishal G. Jadhao, Prof.Prafulla D. Gawande. Performance Analysis of Linear Block Code, Convolution code and Concatenated code to Study Their Comparative Effectiveness; IOSR Journal of Electrical and Electronics Engineering (IOSRJEEE) Volume 1, Issue 1 (May-June 2012).
[6] Mr.Sandesh Y.M and Mr.Kasetty Rambabu Implementation of Convolution Encoder and Viterbi Decoder for Constraint Length 7 and Bit Rate ½; Int. Journal of Engineering Research and Applications ISSN: 2248-9622, Vol. 3, Issue 6, Nov-Dec 2013, pp.42-46.
[7] Patel SnehaBhanubhai, Grace Mary Shajan, Dalal D. Upena; Performance of Turbo Encoder and Turbo Decoder for LTE; International Journal of Engineering and Innovative Technology (IJEIT) Volume 2, Issue 6, October 2012.
[8] Nimbalker Ajit, Blankenship Yufei, Classon Brian; ARP and QPP Interleavers for LTE Turbo Coding; WCNC 2008.
[9] Alaa Eldin.S. Hassan, Moawad Dessouky, Atef Abou Elazm and Mona Shokair; Evaluation of Complexity versus Performance for Turbo Code and LDPC under Different Code Rates; SPACOMM 2012: The Fourth International Conference on Advances in Satellite and Space Communications.
[10] Jagdish D. Kene and Dr. Kishor D. Kulat. Iterative Decoding Termination Schemes for Turbo Code Performance Optimization In Mobile Wi-Max Environment; IJCSI International Journal of Computer Science Issues, Vol. 10, Issue 4, No 1, July 2013.
[11] Sara M. Hassan, A. Zekry, M. A. Bayomy, G. Gomah; Software Defined Radio Implementation of LTE Transmitter Physical Layer; International Journal of Computer Applications (0975 – 8887) Volume 74– No.8, July 2013.
[12] Dilip Mandloi, Prateek Paliwal and Deepali Yadav. Performance Evaluation of ½ Rate Convolution Coding with Different Modulation Techniques for DS-CDMA System over Rician Channel; International Journal of Computer Applications (0975 – 8887) Volume 55– No.12, October 2012.
[13] Syed MD. Asif, Abdullah - Al - Maruf, S. M. Anisul Islam, Amitavo Tikader, MD. Abdul Alim. Comparison of ber between uncoded signal and coded signal (using convolution code) over slow rayleigh fading channel; Journal of Theoretical and Applied Information Technology, Jan 2010.
[14] Harjot Kaur, Bindiya Jain, Amit Verma. Comparative Performance Analysis of M-ary PSK Modulation Schemes using Simulink; IJECT Vol. 2, Issue 3, Sept. 2011.
[15] Nandaniya S. Jayshree, Kalani B. Nilesh and Dr. Kulkarni G.R.; Comparative Analysis of Different Channel Coding Techniques; IRACST – International Journal of Computer Networks and Wireless Communications (IJCNWC) ISSN: 2250-3501 Vol.4, No2, April 2014.

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