



**REDUCING BIT ERROR RATE USING CRC  
VERIFICATION IN TURBO CODES WITH  
OFDM**



**PROJECT REPORT**

*Submitted by*

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## **BONAFIDE CERTIFICATE**

Certified that this project report titled “**REDUCING BIT ERROR RATE USING CRC VERIFICATION IN TURBO CODES WITH OFDM**” is the bonafide work of **G.SOWNDHARYA [15MCO010]** who carried out the research under my supervision. Certified further, that to the best of my knowledge the work reported herein does not form part of any other project or dissertation on the basis of which a degree or award was conferred on an earlier occasion on this or any other candidate.

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## ABSTRACT

Long Term Evolution is a standard for wireless communication which requires high data rate, bandwidth efficiency, and better coverage. So, it needs spectrally efficient techniques like Orthogonal Frequency Division Multiple Access, Multiple Input and Multiple Output and Error Control Codes etc. These techniques are fulfilling some of the features of LTE standard. Here, Orthogonal Frequency Division Multiplexing and Error Control Codes are considered to improve the performance for LTE standard. Orthogonal Frequency Division Multiplexing has become a popular modulation technique in wireless communications in which partitioning a wideband fading channel into flat narrowband channels, it helps to mitigate the effects of multipath fading and Intersymbol Interference using a simple one-tap equalizer. Error control codes have become a vital part of modern digital wireless systems; it will enhance reliable transmission to be achieved over noisy channels. Over the past decade, turbo codes have been widely considered to be the most powerful error control code as they perform close to the Shannon limit.

The main objective of this project is to increase the data transmission and reducing bit error rate by combining OFDM and Turbo Codes. In addition to that, an algorithm is proposed to reduce the bit error rate in turbo codes by applying Cyclic Redundancy Check and Flip and Check algorithm. In order to reduce the bit error rate, the turbo decoder first iterates minimum times without any CRC verification. If the CRC is verified, the process should stop. Otherwise, the process will continue until the CRC is verified. Finally, the decoding process and FC principle is continuing iteratively, until the CRC is verified or until the maximum number of iteration is reached. Here, MAX Log MAP decoding algorithm is applied in turbo decoding process to reduce the computational complexity. The code rate required for this process is  $1/3$ , number of iterations has been increased from 1 to 8 and the size of the block is varied from 40 to 6144. The algorithm has been implemented and simulated using MATLAB software.

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## LIST OF ABBREVIATIONS

ACRONYMS	ABBREVIATIONS
ARQ	Automatic Repeat Request
BER	Bit Error Rate
CRC	Cyclic Redundancy Check Algorithm
CP	Cyclic Prefix
CCSDS	Consultative Committee for Space Data Systems
CDMA	Code Division Multiple Access
DVB	Digital Video Broadcast
FC	Flip and check
FEC	Forward Error Correcting
PCCC	Parallel Concatenated Convolutional Codes
RSC	Recursive Systematic Convolutional
LLR	Log Likelihood Ratio
LTE	Long Term Evolution
ML	Maximum Likelihood Ratio
MAP	Maximum a-Posterior
SNR	Signal to Ratio
TCOFDM	Turbo Coded OFDM System
UMTS	Universal Mobile Telecommunication System
WIMAX	Worldwide Interoperability for Microwave Access

# CHAPTER 1

## INTRODUCTION

Today, wireless communication has become an integral part of everyone's life. Wireless communication applications require error correction schemes in order to maintain high quality of service, reliable transmission, improve safety and provide better bit error rate performance. Basically, a communication system deals with information or data transmission from one point to another. Over the years, there has been a tremendous growth in digital communications. In these digital communication systems; the information is represented as a sequence of binary bits. The binary bits are then mapped (modulated) to analogy signal waveforms and transmitted over a communication channel. The communication channel introduces noise and interference to corrupt the transmitted signal. At the receiver end, the channel corrupted transmitted signal is mapped back to binary bits. The received binary information is an estimate of the transmitted binary information. Normally, during signal transmission through noisy channels, errors can occur on the received data. Number of bit errors depends on the amount of noise and interference in the communication channel. These errors can be detected and corrected using coding techniques. Error correction codes decrease SNR by introducing the coding gain for the communication link. The coding gain measures the amount of additional SNR required to provide the same BER performance for an uncoded message signal.

In wireless communication system, signal can travel from transmitter to receiver over multiple reflective paths. This phenomenon, called multipath fading, can cause fluctuations in the received signal's amplitude, phase, and angle of arrival, giving rise to the multipath fading. In wireless communication, the received signal is not same as the transmitted signal. The received signal may be represented in the complex base band form. Rayleigh fading is a statistical model for the effect of a propagation environment on a radio signal, such as that used by wireless devices. Fading is due to multi path propagation. Fading phenomenon is multiplication of the signal waveform with a time-dependent coefficient which is often modelled as a random variable, making the received Signal to Noise Ratio (SNR) a random quantity. A defining characteristic of the wireless channel are the variations of the channel strength over time and over frequency. The variations can be divided into two types such as, **1. Large-scale fading**, due to path loss of signal as a function of distance and shadowing by large objects such as buildings and hills.

**2. Small-scale fading**, due to the constructive and destructive interference of the multiple signal paths between the transmitter and receiver.

### 1.1 DIGITAL COMMUNICATION

In a digital transmission system, error control is achieved by the use of a channel encoder at the transmitter and a corresponding decoder at the receiver, as shown in Fig.1.1. The main aim of the channel coding is received information is as close as possible to the transmitted information.

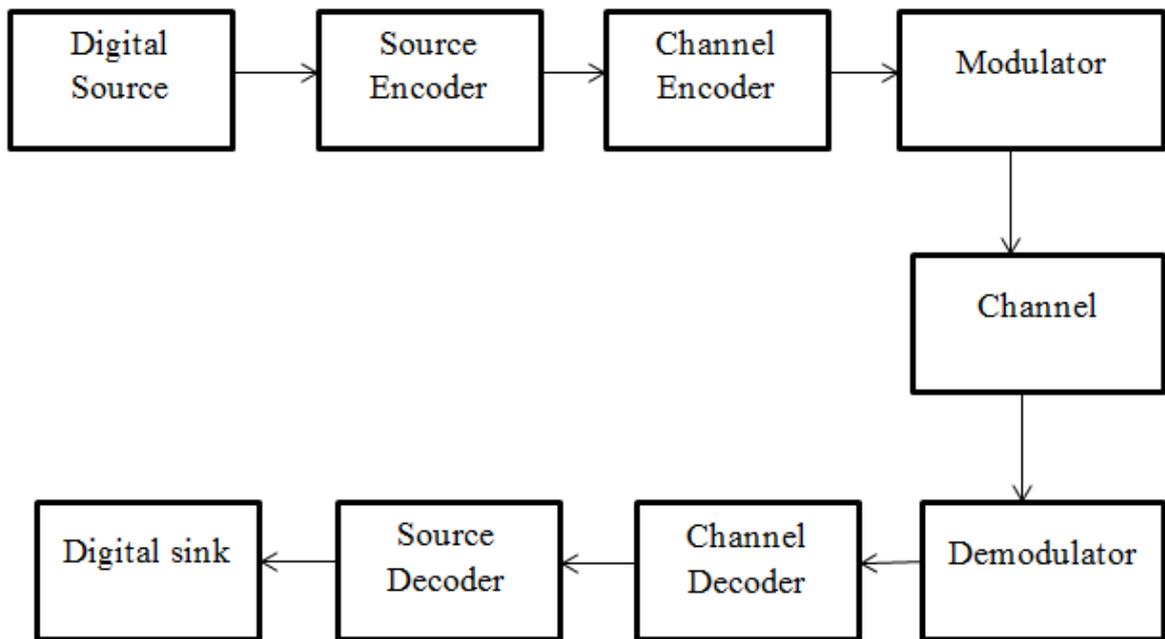


Fig. 1.1 Block Diagram of Digital Communication

In communication systems, the messages generated from the source are converted into a sequence of binary digits. The source output may be either an analog signal, such as an audio or video signal, or a digital signal. The process of converting the output of either an analog or digital source into a sequence of binary digits is called source encoding or data compression. The sequence of binary digits from the source encoder, that information sequence is passed to the channel encoder. The purpose of the channel encoder is to introduce, some redundancy in the binary information sequence that can be used at the receiver to overcome the effects of noise and interference encountered in the transmission of the signal through the channel. This increases the reliability of the received data and improves the fidelity of the received signal. The binary sequence at the output of the channel encoder is passed to the digital modulator, which serves as the interface to the

communication channel. The communication channel is the physical medium that is used to send the signal from the transmitter to the receiver. In wireless transmission, the channel may be the atmosphere (free space). At the receiving end of a digital communication system, the digital demodulator processes the channel corrupted transmitted waveform and reduces the waveforms to a sequence of numbers that represent estimates of the transmitted data symbols (binary or M-ary). This sequence of number s is passed to the channel decoder, which attempts to reconstruct the original information sequence from knowledge of the code used by the channel encoder and the redundancy contained in the received data.

## **1.2 CHANNEL CODING**

Channel coding is one of the most important aspects in digital communication systems, which can be considered as the main difference between analog and digital systems making error detection and correction possible. A way of encoding data in a communications channel that adds of redundancy into the transmission path in order to lower the error rate. A general objective of channel coding is to achieve maximum data transfer, in a minimum bandwidth while maintaining an acceptable quality of transmission. The quality of transmission in digital communications is essentially concerned with the probability of bit error ( $P_e$ ) at the receiver. The Shannon-Hartley law for the capacity of a communications channel demonstrates two things. Firstly it shows how bandwidth (B) and signal power (S/N) may be traded in an ideal system, and secondly it gives a theoretical limit for the transmission rate of data from a transmitter of given power, over a channel with a given bandwidth which operating in a given noisy environment. Moreover, the use of error-control coding adds complexity to the system, especially for the implementation of decoding operations in the receiver. Thus, the design trade-offs in the use of error-control coding to achieve acceptable error performance include considerations of bandwidth and system complexity. The design goal of channel coding is to increase the resistance of a digital communication system to a channel noise.

### **CLASSIFICATION OF CHANNEL CODES**

Channel coding can be classified into two types such as block coding and convolutional coding. The channel encoder adds redundant bits according to the coding method used. The channel decoder transforms the received sequence into an estimated information sequence. Since channel impairments may cause some decoding errors, the channel decoder must be implemented in a way that minimizes the probability of decoding

error. This is done by using statistical information about the channel impairments. Concatenated codes are the combination of two or more encodings of a message, either in serial or in parallel.

### Single-Level Concatenated Codes

Concatenated codes were originally proposed by Forney, the serial concatenation of two channel codes, with one code being the outer code and the other one is inner code. Concatenated codes are used to achieve Shannon channel capacity performance. Figure 1.2 depicts the proposed concatenated code of Forney, which is referred to as a single-level concatenation. The decoding of the proposed concatenated code is to done in stages using a separate inner and outer decoder. In this case, the result of the inner decoder is passed on to the outer decoder to do a second decoding iteration, it will the performance of channel codes. Normally, the inner code is typically a convolutional code and the outer code is typically a Reed-Solomon code, as is the case for the digital video broadcast satellite (DVB-S) standard. However, a concatenated code composed of a LDPC inner code and a BCH outer code is employed by the digital video broadcast satellite for second generation (DVB-S2) standard. The single level concatenated codes as shown in Fig 1.2



Fig. 1.2 Single Level Concatenated Codes

### Multi-Level Concatenated Codes

Single-level concatenation is a special case of multi-level concatenation. In a multi-level concatenation there are more than two encoding and decoding stages used. Multiple stages of encoding are combined to form a multi-level concatenated code. Multi-level concatenations can provide a more flexible, more powerful concatenated code than single-level concatenations. Multi-Level Concatenated Codes resulted from the combination of three ideas that is coding community such as the transforming of commonly used non-systematic convolutional codes into systematic convolutional codes. The utilization of soft input soft output decoding. Instead of using hard decisions, the decoder uses the probabilities of the received data to generate soft output which also contain information about the degree of certainty of the output bits. This is achieved by using an interleaver. Encoders and decoders working on permuted versions of the same information.

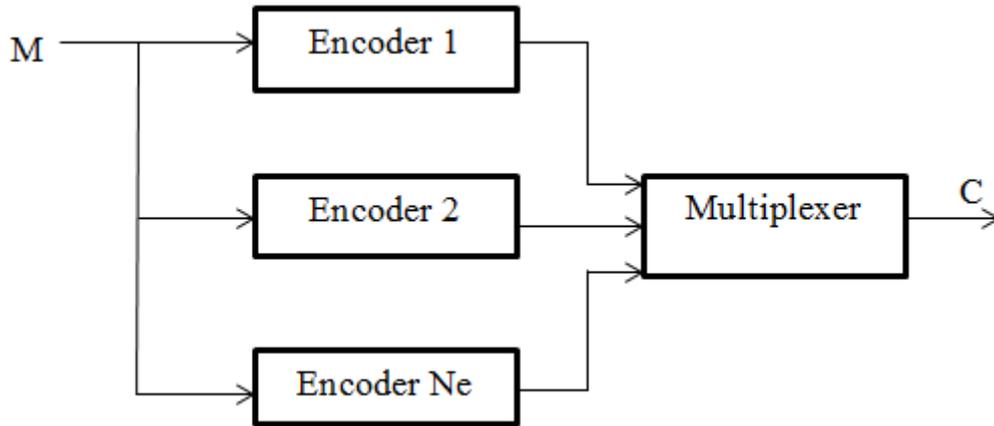


Fig. 1.3 Multilevel Concatenated Codes

In multi-level concatenated codes encoders ( $N_e$ ) are used to encode a message  $\mathbf{m}$ , to create a new code word  $\mathbf{c}$ . In the case of parallel concatenation as shown in Fig 1.3 message  $\mathbf{m}$  is passed to  $N_e$  separate encoders and their results are multiplexed (concatenated) together to create the code word  $\mathbf{c}$ . For both block and convolutional codes are used for more than one stage of parallel encoding, interleaving the input of second interleaver must be done to randomize the bits of  $\mathbf{m}$  in order maintain independence between the outputs of the encoders.

### 1.3 ERROR CONTROL CODES

Error control coding is used to detect and often correct symbols which are received in error. Error correction exists in two main forms such as Automatic Repeat Request and Forward Error Correction. With ARQ the receiver requests retransmission of data packets, if errors are detected, using some error detection mechanism. In FEC some redundancy bits are added to the data bits, which are done either block wise which is called block coding or convolutional coding, where the coded bit depends not only on the current data bit but also on the previous bits. In LTE both block codes and convolutional codes are used. There is also an enhanced coding technique used in LTE, called Turbo code, which has performances close to the Shannon's limit. The primary function of an error control code is to enhance the reliability of message during transmission of information carrying symbols through a communication channel. An error control code can also ease the design process of a digital transmission system in multiple ways such as the following:

- a) The transmission power requirement of a digital transmission scheme can be reduced by the use of an error control codes.

- b) Even the size of a transmitting or receiving antenna can be reduced by the use of an error control codec while maintaining the same level of end-to-end performance.
- c) Access of more users to same radio frequency in a multi-access communication system can be ensured by the use of error control technique.
- d) Jamming margin in a spread spectrum communication system can be effectively increased by using suitable error control technique.

### **1.3.1 FORWARD ERROR CORRECTION**

Forward Error Correction adds redundancy to data in the form of parity bits, which is used to detect and correct the errors in transmitted data. Different FEC methods are used to check and correct errors which are close to the limit proposed by Shannon for data transmission. Various types of Forward Error Correction Codes such as Linear block codes, Hamming codes, Reed Solomon codes, Convolutional codes and Turbo codes. Among these turbo codes are powerful error detecting and correcting codes.

#### **Linear Block Codes**

Linear block codes are referred to as  $(n, k)$  codes, where  $n$  is the length of the block, and  $k$  is the number of information bits in the case of a binary code. This means that to transmit data, an LBC uses  $2^k$  codeword's, which make a  $2^n$  possible codeword's. The general form of an LBC is  $C = DG$ , Where  $C$  is the codeword,  $D$  is the  $k$ -bit message, and  $G$  is the generator matrix that creates the parity check bits from the data bits. The code rate of a code is the ratio  $n=R / K$ , the number of input bits over the number of output bits of an FEC encoder. This ratio can be thought of as the percentage of time used to send actual information bits through a channel. The code rate improves with larger data block lengths, but larger block lengths have a proportionally larger number of expected errors.

#### **Hamming Codes**

Hamming codes are the earliest codes capable of actually correcting an error detected at the receiver. However, they are limited so that they can detect and correct only a single error. The  $(7, 4)$  Hamming code is a famous single error correcting code because it belongs to the class of perfect codes. An advantage of hamming codes is that encoding and decoding are easy to implement. They would be effective as a simple and efficient code over a channel where it is known that errors are burst-free. Disadvantages of Hamming codes are that they are very ineffective for low SNR, where the received signal level is very low. These types of conditions will tend to cause more frequent errors. Also, Hamming codes do very poorly against the bursts of errors caused by fading over a channel.

## **Reed Solomon Codes**

Reed-Solomon (RS) codes are non-binary codes; a codeword is defined in terms of multi-bit symbols rather than bits. Such codes can achieve a very large Hamming distance. The RS decoder corrects the entire symbol, whether the error was caused by one bit being corrupted or by all of the bits being corrupted. Thus, if a symbol is wrong, it might as well be wrong in all of its bit positions. This gives RS codes tremendous burst-noise advantages over binary codes. Burst-noise is relatively common in wireless communication due to fading. RS codes are mainly non-binary block codes that are especially effective against burst-errors. The coding efficiency mainly increases with code length. RS codes can be used for long block lengths with less decoding time than other codes because RS codes work with symbol-based arithmetic. This makes RS coding harder to implement, but it provides better throughput.

## **Convolutional Codes**

Convolutional codes are widely used as channel codes in practical communication systems for error correction. The encoded bits depend on the current  $k$  input bits and a few past input bits. Convolutional codes are mainly used in many cellular applications and standards. Turbo codes are concatenation of many turbo encoders because it gives better error performance as well as give low weight code words. The main decoding strategy for convolutional codes is based on the widely used Viterbi algorithm. As a result of the wide acceptance of convolutional codes, there have been several approaches to modify and extend this basic coding scheme. Trellis coded modulation (TCM) and turbo codes are two such examples. In TCM, redundancy is added by combining coding and modulation into a single operation. This is achieved without any reduction in data rate or expansion in bandwidth as required by only error correcting coding schemes.

Convolutional codes are commonly described using two parameters: the code rate and the constraint length. The code rate,  $k/n$ , is expressed as a ratio of the number of bits into the convolutional encoder ( $k$ ) to the number of channel symbols output by the convolutional encoder ( $n$ ) in a given encoder cycle. The constraint length parameter,  $K$ , denotes the "length" of the convolutional encoder, i.e. how many  $k$ -bit stages are available to feed the combinatorial logic that produces the output symbols. Closely related to  $K$  is the parameter  $m$ , which indicates how many encoder cycles an input bit is retained and used for encoding after it first appears at the input to the convolutional encoder. The  $m$  parameter can be thought of as the memory length of the encoder.

## **Turbo Codes**

Turbo codes are part of a class known as convolutional codes. Turbo Codes are well suited for long distance and low power wireless communications because they achieve a very low bit error rate (BER) at very low SNR. Turbo codes have to be implemented in many applications such as deep space communications and third generation cellular standards. The superior performance of turbo codes is due to the unique encoding process, but also the fact that the decoding process is iterative. However, the increase in iterations, improvement over the BER achieved decreases. Also, continually increasing the number of iterations causes decoder delay, and it decreases the throughput that turbo codes are able to achieve; because with this added delay, fewer bits can be decoded per second. Another way to improve the BER is to add more parity bits to each input bit. As the code rate decreases, the BER decreases at an even lower SNR because more errors can be corrected by adding more parity bits. However, decreasing the code rate by too much will significantly decrease the number of information bits a channel can transmit per second because the channel will now be filled with more parity bits. In short, there will always be a trade-off between BER and data throughput.

The presence of two turbo codes increases the probability of high Hamming weight code words, but will also double the number of low Hamming weight code word's present. Although the BER decreases dramatically for low SNR, the presence of low weight code words begins to dominate the performance of turbo codes at higher SNR, causing the BER curve to flatten out at these values. Though the encoding and decoding processes lowers the BER, implementing these practices is costly, and may not be desirable. Furthermore, the decoding algorithm is complex, and more iteration requires greater amounts of memory, causing the implementation of turbo codes to be more expensive than linear block codes. Lastly, in order to calculate the LLRs necessary for the decoder, the channels Characteristics must be known, which means that you must be familiar with the channel over which the data is being transmitted.

Turbo codes are used in many applications. It allow for relatively high data rates. They are mainly intended for satellite communications, wireless internet access, wireless local area networks (LAN's), microwave systems, and mobile communications applications. Specific standards that use turbo codes include CCSDS (for deep-space communications), mobile communications like UMTS and CDMA2000 (3G mobile), satellite telecommunications like DVB-RCS (return channel over satellite), DVB-RCT (return

channel over terrestrial), Inmarsat (M4) and Eutelsat, wireless Internet access like IEEE 802.16 (Wimax), LTE.

## 1.4 OFDM

Frequency Division Multiplexing (FDM) has been used for a long time to carry more than one signal over a channel. FDM divides the channel bandwidth into sub channels and transmits multiple relatively low rate signals by carrying each signal on a separate carrier frequency. To ensure that the signal of one sub channels did not overlap with the signal from an adjacent one, some guard-band was left between the different sub channels. Obviously, this guard-band led to inefficiencies. In order to solve the bandwidth efficiency problem,

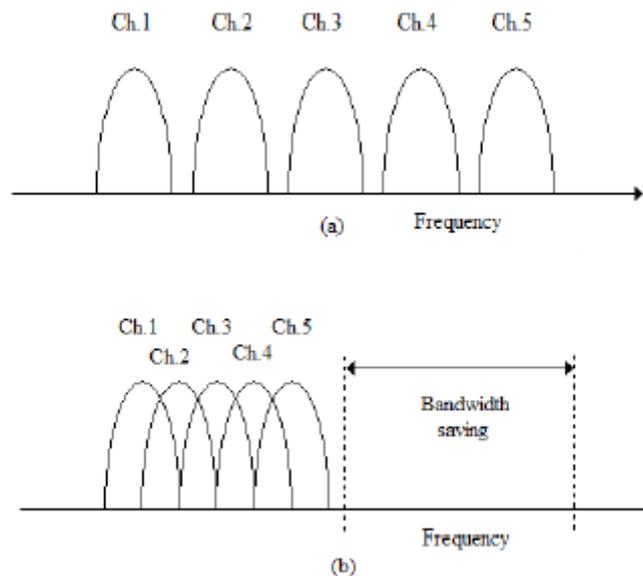


Fig. 1.4 Comparisons of FDM and OFDM

Orthogonal frequency division multiplexing was proposed, where the different carriers are orthogonal to each other. With OFDM, it is possible to have overlapping sub channels in the frequency domain, thus increasing the transmission rate.

## ORGANIZATION OF THE REPORT

- **Chapter 2 (Literature review):** discusses the various types of Error Detection and correction Scheme and OFDM technique. It explains the mathematical requirement of turbo encoder and decoder also discusses about the proposed algorithm such as cyclic redundancy check and flips and check algorithm.

- **Chapter 3 (Turbo codes)** gives the detailed explanation of turbo codes used for error detection and correction in the received code words.
- **Chapter 4 (OFDM)** gives the detailed explanation about Orthogonal Division Multiplexing.
- **Chapter 5 (Methodology)** gives detailed explanation about the LTE turbo encoder and decoder, proposed method of cyclic redundancy check and flip and check algorithm for reducing bit error and computational complexity of the turbo decoding process.
- **Chapter 6 (Results and Discussions)** shows the experimental results of Turbo coded OFDM system as well as comparison between MAP and MAX Log MAP decoding algorithm using cyclic redundancy check and flip and check algorithm.
- **Chapter 7 (Conclusion)** gives conclusion of the project by comparing the obtained results.

# CHAPTER 2

## LITERATURE REVIEW

### 2.1 Lowering the Error Floor of Turbo Codes with CRC verification

Thibaud Tonnellier, Camille Leroux, Bertrand Le Gal propose a method to lower the error floor of turbo codes by using the CRC code during the iterative process. In this paper denoted as the flip and check (FC) algorithm a set of test patterns are generated by identifying the least reliable bits among the frame. This identification is based on the extrinsic information of each bit. Each test pattern is then checked with the CRC code. This method does not require the modification of the interleaver nor the coding scheme as soon as a CRC code is already present. Thus, this method can be already applicable to existing standards. Monte Carlo simulations show that such a method can reduce the frame error rate (FER) by at least one order of magnitude in the error floor region. The proposed FC algorithm compares favourably with other methods in terms of decoding performance and computational complexity.

### 2.2 Complexity Reduced Turbo Decoding with Concatenated Detection Codes

Lei Cao, John Daigle, Chang Wen Chen and Mustafa Matalgah error detection outer codes such as CRC [5] and error correction inner codes such as turbo codes are almost always employed together. The turbo decoding status detected through CRC can be used in two ways. First, it can be feedback to the transmitter to update the transmission strategy such as code rate, modulation type, or power control. Second, this information can be used directly at the receiver. For example, if the CRC match is reported in an early iteration step due to a good channel condition, all the following decoding iterations are no longer needed so that the computational resource is saved.

### 2.3 List Decoding of Turbo Codes

Krishna R. Narayanan and Gordon L. Stuber proposes list of decoding as another technique for reducing the BER floor of turbo codes. List decoding is typically used with concatenated codes. The inner decoder uses a Viterbi algorithm or some other hard-output decoding algorithm and outputs a list of probable paths. The outer decoder then selects one of

these paths. For packetized transmission, the cyclic redundancy check (CRC) code forms a convenient outer code. Various methods can be used to generate the list. When a soft-output inner decoder is used, the list can be generated by the method, applied to turbo codes.

#### **2.4 Enhanced Turbo Decoding for Error Floor Reduction**

Stephan Schwinger and Monica Navarro propose a method for reducing error floor using CRC. The appearance of an error floor in turbo codes is mainly caused by the presence of low-weight code words and therefore the principal line of attack to ameliorate this problem deals with the elimination of these code words, which can be done by appropriate design of the interleaver or e.g. the insertion of dummy bits. However, for many fields of application, like for space communications, these techniques are not applicable since they require modifications at the transmitter side. In this paper, we consider solely methods which do not modify the transmitter and hence concentrate on the decoding algorithm for a given turbo code. Choosing a good interleaver limits the number of frames for which the decoder will exhibit undesirable convergence behaviour, but will not eliminate these frames.

#### **2.5 Effect of Turbo Coding on OFDM Transmission to Improve BER**

Dhiraj G. Agrawal and Roma K. Paliwal propose a technique to improve BER performance with the combination of OFDM and Turbo Codes. The combination of these two techniques enhances the throughput of an existing OFDM system by implementing adaptive modulation and turbo coding. The new system guarantees to reach a target performance BER of  $10^{-2}$  over a slow time-varying fading channel. The system automatically switches from lower to higher modulation schemes on individual subcarriers, depending on the state of the quasi-stationary channel. In conjunction with the adaptive design, forward error correction is performed by using turbo codes. The combination of parallel concatenation and recursive decoding allows these codes to achieve near Shannon's limit performance in the turbo cliff region.

#### **2.6 Simulation of Turbo Coded OFDM System**

Jyoti Chand and Deependra Pandey [7] propose a method to enhance the performance in wireless communication with the combination of turbo codes and OFDM. Orthogonal Frequency Division Multiplexing (OFDM) has been increasing rapidly in a high speed wireless communication systems. In OFDM we basically divide a wideband fading channel into a narrowband channels. OFDM is capable of neglecting or deleting the diverse (bad)

effects of multipath fading with the help of a one tap equalizer. A system is needed that can transmit information wirelessly and accurately. The technique of combining an OFDM which is suitable for high data rate transmission with a Forward Error Correction (FEC) method over a wireless channels by proper Power allocation and efficient use of coding, the performance at the high data rates improves.

## **2.7 CTC Turbo Decoding Architecture for LTE Systems**

Cristian Anghel, Valentin Stanciu [4] proposed a method for Long Term Evolution (LTE) standard using a Max Log MAP. Due to the new LTE interleaver, the decoding performances are improved compared with the ones corresponding to UMTS standard. Moreover, the new LTE interleaver provides support for the parallelization of the decoding process inside the algorithm, taking advantage on the main principle introduced by turbo decoding such as the usage of extrinsic values from one turbo iteration to another. This paper presents an efficient solution for the hardware implementation of a Convolutional Turbo Code (CTC) LTE decoder.

## **2.8 Interleaver Design for Serially Concatenated Convolutional Codes**

Fred Daneshgaran and Massimiliano Laddomada propose a method for the problem of interleaver design for serially concatenated convolutional codes (SCCCs) tailored to the constituent codes of the SCCC configuration. The theoretical framework for interleaver optimization based on a cost function closely tied to the asymptotic bit-error rate (BER) of the block code resulting from proper termination of the constituent codes in the SCCC code. A canonical form of the interleaving engine denoted as the finite state permuter (FSP) and using its structural property develop a systematic iterative technique for construction of interleavers. The core theoretical results focus on the asymptotic behaviour of a class of cost functions and their martingale property, which is then used to develop an order recursive interleaver optimization algorithm. The issue of the complexity of the interleaver growth algorithm presented in the paper and demonstrate that it has polynomial complexity. Subsequently, we provide details about the application of the proposed technique and present a modification of the algorithm that employs error pattern feedback for improved performance at a reduced complexity.

## **2.9 Convergence and Errors in Turbo-Decoding**

Andrew C. Reid, and T. Aaron Gulliver describes a method of Turbo-codes are decoded using iterative decoding algorithms with somewhat elusive convergence properties. In this paper, the modes of convergence observed in an extensive series of turbo-code simulations are categorized by examining the bit convergence behaviour for each frame. Based on these results, a new method for decoder termination based on average log-likelihood ratios is presented and compared with other methods. As an application Forward error correcting system is considered.

## **2.10 Performance Analysis and Formulation of Turbo Coded OFDM over Rayleigh & Rician Fading Channel**

Avijit Patel and B. S. Rai propose a method for wireless communication, to attain high throughput and better transmission quality; there is a need of better system design. Orthogonal Frequency Division Multiplexing (OFDM) is a technique for parallel transmission. The performance of OFDM can be improved by using some kind of channel coding. This form of OFDM is called coded-OFDM (COFDM). In this paper Turbo code is used as channel coding with OFDM, called Turbo coded OFDM. This paper compares the performance of Turbo coded OFDM using binary phase shift keying (BPSK), Quadrature Phase Shift Keying (QPSK), Quadrature Amplitude Modulation (QAM) modulation technique for different Rayleigh and Rician fading channel. Results is formulated that a combination fading channel and modulation technique gives better performance.

# CHAPTER 3

## TURBO CODES

Digital data transmission channels are subject to various impairments including noise, distortion, and interference. Therefore, the output of a channel differs from its input and decoding errors can result from impaired transmission. Although we can achieve the required performance by simply by using sufficient power, in many cases error-control techniques can provide the required accuracy with less energy.

Error control techniques can be classified into two main categories: forward error correction (FEC) and Automatic Repeat Request (ARQ). In FEC, redundant bits are added to the information data at the transmitting side, by utilizing that redundancy can be able to correct any errors that may have been contributed by the channel at the receiving end. On the other hand, in ARQ, the receiving terminal does not attempt to correct the errors, but attempts to detect them. In the event that an error is detected, the receiver simply requests the re-transmission of the data. The purpose of channel coding is to average the effects of channel impairments over several transmitted bits. In order to achieve this purpose, the channel encoder transforms the information sequence into a longer binary-coded sequence by adding redundant or parity check symbols.

### 3.1 INTRODUCTION OF TURBO CODES

In 1993, a new iterative decoding technique for concatenated channel codes was introduced by Berrou. A Turbo code is a concatenated code that is decoded using an iterative decoding process. It was obtained better performance could be achieved with just two serial or parallel concatenations, which is used in all 3G and 4G telecommunications standards, as well as the DVB standards for satellite. At present concatenated turbo codes have been implemented in many 3G telecommunications standards (e.g. UMTS) and are included as part of the 4G telecommunications standard such as LTE and WiMAX.

### 3.2 TURBO ENCODER

Generally, the turbo consist of parallel concatenation of Recursive Systematic Convolutional (RSC) encoder which is separated by a random interleaver as shown in Fig 3.3. Two component RSC codes are used to encode the same information bits, and the outputs from the two RSC codes are different and then multiplexed. The constraint length of

convolutional codes is usually proportional to complexity of the encoder and decoder. For turbo codes, RSC component codes with relatively short  $K$  are used. The better performance of turbo codes has been obtained even simple component codes with use of a long length-interleaver between the component codes. The purpose of the interleaver is to decorrelate the inputs to the component RSC encoders, so that the parity streams from each encoder are independent. This improves the likelihood that an error pattern will eventually be correctable by one of the two decoders. The interleaving is usually performed in a pseudorandom manner over a fixed length block.

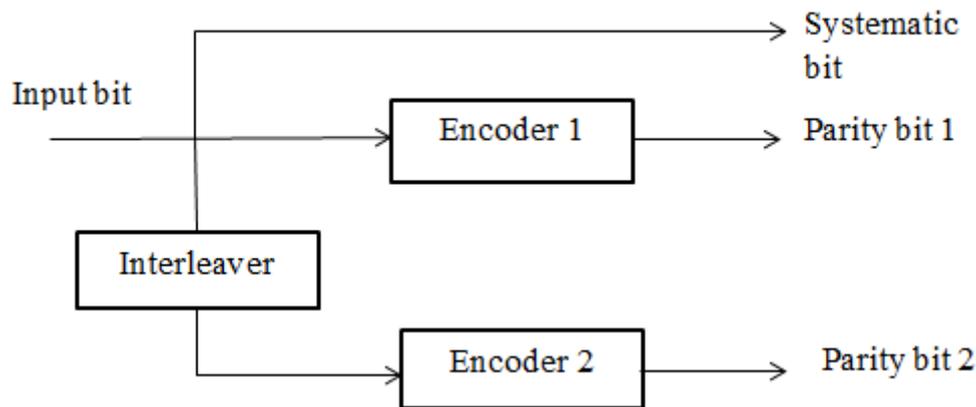


Fig. 3.1 Turbo Encoder

### 3.3 INTERLEAVER

An interleaver in a turbo encoder is used after error control coding and signal mapping to ensure that fading bursts affecting blocks of symbols transmitted over the channel are broken up at the receiver by a de-interleaver, prior to decoding. Most error control codes work much better when errors in the received sequence are spread far apart. Another role is that of the interleaver between component codes in a serially concatenated code scheme; for example, between a Reed Solomon outer code and a convolutional inner code. Turbo code BER performance improves with interleaver length, a phenomenon called interleaver gain. It will enhance the delay to the decoding process. The trellis decoding nature of most convolutional codes means that uncorrected errors at the output of the decoder will tend to occur in bursts. The interleaving between the two component codes then ensures that these bursts are adequately spread before entering the outer decoder. In both these examples, the interleaver is typically implemented as a block interleaver. This is a rectangular matrix such that bits or symbols are written in one row at a time, and then read out one column at a time. Thus bits or

symbols which were adjacent on writing are spaced apart by the number of rows when reading. The de-interleaving process is simply the inverse of this; writing in column by column and reading out row by row, to achieve the original bit or symbol ordering. Block interleavers are not suitable as turbo code interleavers, because they tend to generate large numbers of code words with a relatively low weight, and therefore with a relatively low hamming distance between them, due to the regularity of the spreading process.

### 3.4 TURBO DECODER

Turbo decoding is an iterative decoding process. The turbo decoder consists of two component decoders as shown in Figure 3.2. The iterative decoding is based on that each decoder produces a soft estimate of the original information bits; this estimation is used by the other decoder, to produce a better estimation. The new estimation is used again by the first decoder to enhance its estimation. The estimation is better with the increase of the number of iterations. Each component decoder is based on soft input soft output decoding. The soft representation of the information bits is carried out in a form of a Log Likelihood Ratio (LLR). The soft output of each decoder provides a-priori probability of the information bits to be used by the other decoder. The a-priori information is also called extrinsic information. Each component decoder operation is based on the received systematic, and parity bits from the channel, in addition to the extrinsic information from the other decoder. At the beginning of the first iteration, the decoder has no a-priori information about information bits. It has only channel information on systematic and parity bits. Thus, the input a-priori information is set initially to zero. The extrinsic information generated by each decoder is the key difference among successive iterations.

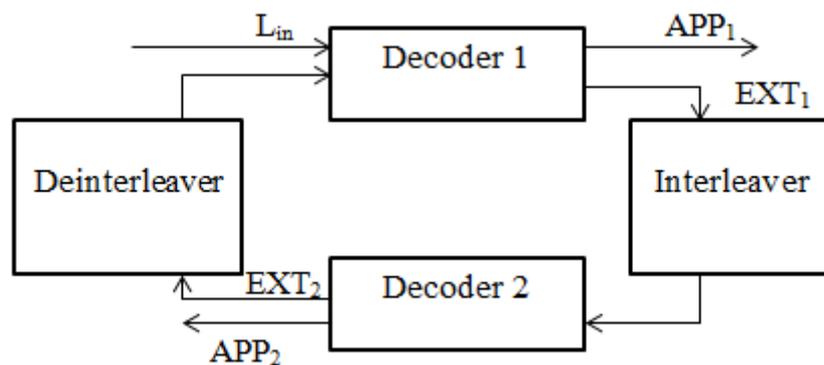


Fig. 3.2 Turbo Decoder

### 3.4.1 Need for Soft Input/Soft Output Algorithm

Decoding of convolutional codes is most frequently achieved using the Viterbi algorithm which makes use of a decoding trellis to record the estimated states of the encoder at a set of time instants. The Viterbi algorithm based on rejecting the least likely path through the trellis at each node, and keeping the most likely one. The Viterbi decoder estimates a maximum likelihood sequence. Making hard decisions in decoding process, leads to loss of valuable information. So we go for soft decisions, until all possible information has been extracted from the received signal values. The turbo decoding relies on passing information about individual transmitted bits from one decoding stage to the next. The interleaving of the received information sequence between decoders limits the estimating maximum likelihood sequences. So, an algorithm is required that can output soft-decision maximum likelihood estimates on a bit-by-bit basis. The decoder should be able to accept soft decision inputs from the previous iteration of the decoding process. Such a decoder is termed a Soft Input-Soft Output (SISO). Berrou and Glavieux used two such decoders in each stage of their turbo decoder. They implemented the decoders using a modified version of an SISO algorithm proposed by Bahl, Cocke, Jelinek and Raviv. Their modified Bahl algorithm is commonly referred to as the Maximum A Posteriori or MAP algorithm, and achieves soft decision decoding on a bit-by-bit basis by making two passes of a decoding trellis, in the case of Viterbi algorithm.

The soft output of each decoder is based on LLR. In binary turbo codes, for a certain data bit  $u_k$ , the LLR  $L(u_k)$  is defined as the logarithm of the ratio of probability that  $u_k=+1$  to the probability that  $u_k=-1$ . This means the ratio between a-priori probabilities.

$$L(u_k) = \ln \left( \frac{p(u_k = +1)}{p(u_k = -1)} \right) \quad (1)$$

Unlike LLR, the conditional LLR  $L(u_k/y)$  commonly used in decoding techniques. It is based on the ratio of a-posteriori probabilities. Its equation is given as follows

$$L(u_k / y) = \ln \left( \frac{p(u_k = +1 / y)}{p(u_k = -1 / y)} \right) \quad (2)$$

Where  $y$  is the received code word. This ratio of the a-posteriori probabilities will be used by the decoder to provide soft representation of the decoded bits.

### 3.4.2 Methods for Decoding Turbo Codes

Iterative decoding of turbo codes requires a decoding algorithm that can produce soft outputs, as the soft outputs of one component decoder become the inputs to another component decoder. The soft output decoding algorithm was first used in MAP algorithm, which is also known as BCJR algorithm. The principle of BCJR decoding is that the metric associated with a particular branch in the trellis is given by the product of probabilities associated with choosing the most appropriate start-node, and following the most appropriate set of branches that arrive at the most appropriate end-node at the end of the trellis.

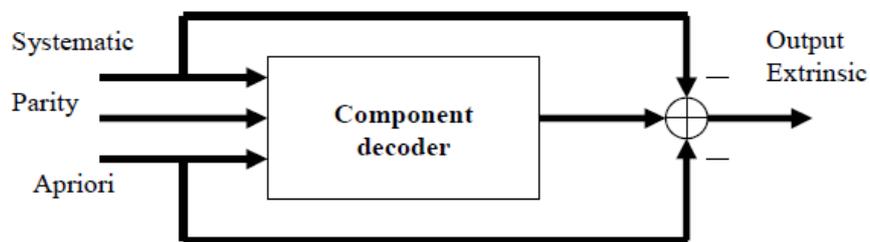


Fig. 3.3 LLR Computation

The soft output for decoded information  $u_k$ , which is the information bit associated with the transition from time  $k-1$  to  $k$  in the trellis, can be estimated by using log likelihood values. The extrinsic LLR should be bypassed to the other component decoder as an a priori probability used in next iteration.

### 3.4.3 Maximum A-Posteriori probability (MAP) algorithm

The Maximum A-Posteriori (MAP) algorithm introduced in 1974 by Bahl, Cocke, Jelinek and Raviv. The MAP algorithm is optimal for Convolutional codes and it minimizes the number of bits decoded incorrectly, while the Viterbi algorithm minimizes the probability of an incorrect path in the trellis. The MAP algorithm provides the decoded bit sequence along with the probability that each bit is decoded correctly. First we need to define the following:

$\alpha_{k-1}(s')$  is the probability that the trellis is in state ( $s'$ ) at time  $k - 1$  given the sequence of bits previous to bit  $k$ ,

$\beta_k(s', s)$  is the probability that the trellis moves from  $s'$  to  $s$  given the received channel value  $y_k$  at time  $k$ ,

$\gamma_{k(s)}$  is the probability that the trellis is in states at time  $k$  given the sequence after bit  $k$ .

In order to reduce the complexity of computing the products of several probabilities, we often use logarithmic values, which is often called log-MAP it is represented b

$$L(u_k) = \log\left(\frac{p(u_k = 1/y)}{p(u_k = 0/y)}\right) = \log\left(\frac{\sum_s \sum_{s'} \gamma + 1(y_k, s^1_{k-1}, s^1_{k-1})\alpha_{k-1}(s^1_{k-1})\beta_k(s^1_k)}{\sum_s \sum_{s'} \gamma - 1(y_k, s^0_{k-1}, s^0_k)\alpha_{k-1}(s^0_{k-1})\beta_k(s^0_k)}\right) \quad (3)$$

Where  $y_k$  is the received signal at time  $k$  and  $k-1$  by using branch metric  $\gamma(k)$ , forward state metric  $\alpha_{k-1}$  and backward state metric  $\beta_k$  respectively. The forward and backward path metrics are calculated in a recursive manner as shown in equation

$$\alpha_k(s) = \frac{\sum_{s'} \sum_{i=\pm 1} \gamma^i(y_k, s', s)\alpha_{k-1}(s')}{\sum_s \sum_{s'} \sum_{j=\pm 1} \gamma^j(y_k, s', s)\alpha_{k-1}(s')} \quad (4)$$

$$\beta_k(s) = \frac{\sum_{s'} \sum_{i=\pm 1} \gamma^i(y_{k+1}, s, s')\beta_{k+1}(s')}{\sum_s \sum_{s'} \sum_{j=\pm 1} \gamma^j(y_{k+1}, s, s')\beta_{k+1}(s')} \quad (5)$$

Once the soft outputs are obtained, iterative decoding can be performed by estimating the associated extrinsic information and transferring it to the next decoding stage. Figure 3.5 shows the relationship between soft inputs and outputs of a SISO decoder. Initially, the a priori information  $L(\mathbf{u})$  is set to zero, where  $L_c$  is the channel gain. Assuming that soft outputs are available from the first decoding process of decoder 1, the a priori information going into the decoder 2 is the difference between the output and the input to the first decoding process. The information passed to the next component decoder, called the extrinsic information, is used as a priori information by the next decoding stage in conjunction with the received values. The final step after calculation of the branch metrics and state probabilities at each time slot of the codeword is to calculate the LLRs. These LLRs represent the decoder soft output. Equation of LLR is shown in equation (16),

$$LLR = \ln \left( \frac{\sum_{s^1 \rightarrow s \Rightarrow u_k = +1} \alpha_{k-1}(s^1) \cdot \gamma_{k-1} \rightarrow k(s^1 \rightarrow s) \cdot \beta_k(s)}{\sum_{s^1 \rightarrow s \Rightarrow u_k = -1} \alpha_{k-1}(s^1) \cdot \gamma_{k-1} \rightarrow k(s^1 \rightarrow s) \cdot \beta_k(s)} \right) \quad (6)$$

As turbo decoders are based on iterative decoding, the extrinsic likelihood probabilities are calculated from LLRs. Extrinsic likelihood represents how much information the decoder

adds about the decoded bits. It is obtained by subtracting the input values to the decoder from its output LLRs as follows

$$Le(u_k) = LLR - L_c \cdot y_{ks} - L(u_k) \quad (7)$$

Where LLR is the soft output Log Likelihood Ratio from the decoder.

$L_c$  is the channel reliability.

$y_{ks}$  is the received systematic bit

$L(u_k)$  is the input A-priori probability

### 3.4.4 Max Log MAP Decoding Algorithm

There are iterative SISO algorithms other than the MAP algorithm that can be used for the decoding of convolutional turbo codes. In max log MAP algorithm, large portion of the summation can be eliminated, it will reduce the computational complexity of turbo decoding algorithm. The metrics is calculated by using

$$A_k(s) = \ln(\alpha_k(s)) \quad (8)$$

$$B_k(s) = \ln(\beta_k(s)) \quad (9)$$

$$\Gamma_k(s', s) = \ln(\Gamma_k(s', s)) \quad (10)$$

$$L(u_k) = \max_{(s', s)} \Rightarrow u_k = +1 \left( A_{k-1}(s') + \Gamma_k(s', s) + B_k(s) \right) \\ - \max_{(s', s)} \Rightarrow u_k = -1 \left( A_{k-1}(s') + \Gamma_k(s', s) + B_k(s) \right) \quad (11)$$

The MAP algorithm is a Maximum Likelihood (ML) algorithm it has moderate BER performance and more computational complexity. Information bits returned by the MAP algorithm need not form a connected path through the trellis while for SOVA it will be a connected path. Another factor is that the Max Log MAP algorithm removes the decoder dependency on SNR. The SNR becomes a scaling factor multiplied by another term representing the cross correlation between received data and original data corresponding to each branch. However the MAP algorithm is not easily implementing able due to its complexities. Several approximations on the MAP algorithm are now available, such as the Max-Log-MAP algorithm where computations are largely in the logarithmic domain and hence values and operations are easier to implement.

## CHAPTER 4

### ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING

#### 4.1 INTRODUCTION

Multicarrier modulation divides the information data into many parallel sub-channels of narrow bandwidth. The data rate of each sub-channel is much less than the total data rate. Each sub-channel can be designed to have a bandwidth less than the coherence bandwidth of the channel. It increases wireless capacity without increasing bandwidth. Therefore, it can be assumed that each sub-channel experiences flat fading and the demodulator can be implemented without an equalizer. In parallel-data system, the total signal frequency band is divided into  $N$  non-overlapping frequency sub-channels. Each sub-channel is modulated with a separate symbol, and then the  $N$  sub-channels are frequency multiplexed. It seems good to avoid spectral overlap of channels to eliminate inter-channel interference. However, this leads to inefficient use of the available spectrum, so we go for OFDM.

#### 4.2 NEED FOR MULTI CARRIER SYSTEM

OFDM is especially suitable for high-speed communication due to its resistance to ISI. OFDM overcomes the effects of multipath by breaking the signal into many narrow bandwidth carriers. This results in a low symbol rate reducing the amount of ISI. In addition to this, a guard period is added to the start of each symbol, removing the effects of ISI for multipath signals delayed less than the guard period. As communication systems increase their information transfer speed, the time for each transmission necessarily becomes shorter. Since the delay time caused by multipath remains constant, ISI becomes a limitation in high-data rate communication. OFDM avoids this problem by sending many low speed transmissions simultaneously. For example, the figure below shows two ways to transmit the same four pieces of binary data.

#### 4.3 CONCEPT OF OFDM TECHNIQUES

Orthogonal frequency division multiplexing is also known as multicarrier modulation techniques. In this technique multicarrier signals are used with different frequencies, some of the bits are transmitted in each channel. In the OFDM, high rate data stream are split into  $N$  sub stream using serial to parallel converter. The data rate of each sub stream is  $R/N$  bps and

the data is transmitted by using separate carrier with the adjacent spacing is  $\Delta f$ , so the bit duration is  $N/R$ . The OFDM has various advantages such as frequency selective fading, this type of fading is corrected by using forward error correction method and another major advantage is overcome intersymbol interference (ISI). The intersymbol interference causes due to the smaller distance between the bits or symbols. OFDM is implemented using the Fast Fourier transform (FFT). This transform is used at the OFDM transmitter to map an input signal onto a set of orthogonal sub carriers. Similarly, the transform is used again at the OFDM receiver to process the received sub carriers. The signals from the sub carriers are then combined to form an estimate of the source signal from the transmitter.

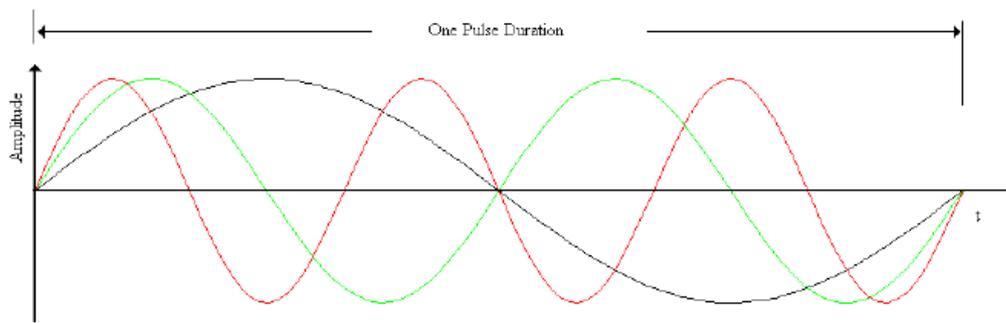


Fig. 4.1 Orthogonality of Sub-Carrier

The orthogonal and uncorrelated nature of the sub carriers is exploited in OFDM with powerful results. The energy from other sub carriers does not contribute because it is uncorrelated. This separation of signal energy is the reason that the OFDM sub carriers' spectrums can overlap without causing interference. Mathematically, each carrier can be represented as a complex wave such as

$$S_C(t) = A_C(t)e^{j(\omega_c(t)+\phi_c(t))} \quad (12)$$

The real signal is the real part of  $s_c(t)$ .  $A_c(t)$  and  $\phi_c(t)$ , denotes amplitude and phase of the carrier, can vary based on symbol by symbol. The values of the parameters are constant over the symbol duration period  $t$ . Thus, the complex signal  $S_s(t)$  is represented by

$$S_S(t) = \frac{1}{N} \sum_{n=0}^{N-1} A_N(t)e^{j[\omega_n t + \phi_n(t)]} \quad (13)$$

Where  $\omega_n = \omega_0 + n\Delta\omega$

This is of course a continuous signal. If we consider the waveforms of each component of the signal over one symbol period, then the variables  $A_c(t)$  and  $\phi_c(t)$  take on fixed values, which depend on the frequency of that particular carrier, it can be rewritten

$$\phi_n(t) = \phi_n \quad (14)$$

$$A_n(t) = A_n \quad (15)$$

If the signal is sampled using a sampling frequency of  $1/T$ , then the resulting signal is represented by

$$S_s(KT) = \frac{1}{N} \sum_{n=0}^{N-1} A_n e^{[j(\omega_0 + n\Delta\omega)KT + \phi_n]} \quad (16)$$

#### 4.4 OFDM TRANSMISSION AND RECEPTION

OFDM transmitters generate both the carrier and the data signal simultaneously. The process signal generation in OFDM is based on the series of mathematical computations known as an Inverse Fourier Transform, and the process results in the formation of a complex modulated waveform at the output of the transmitter.

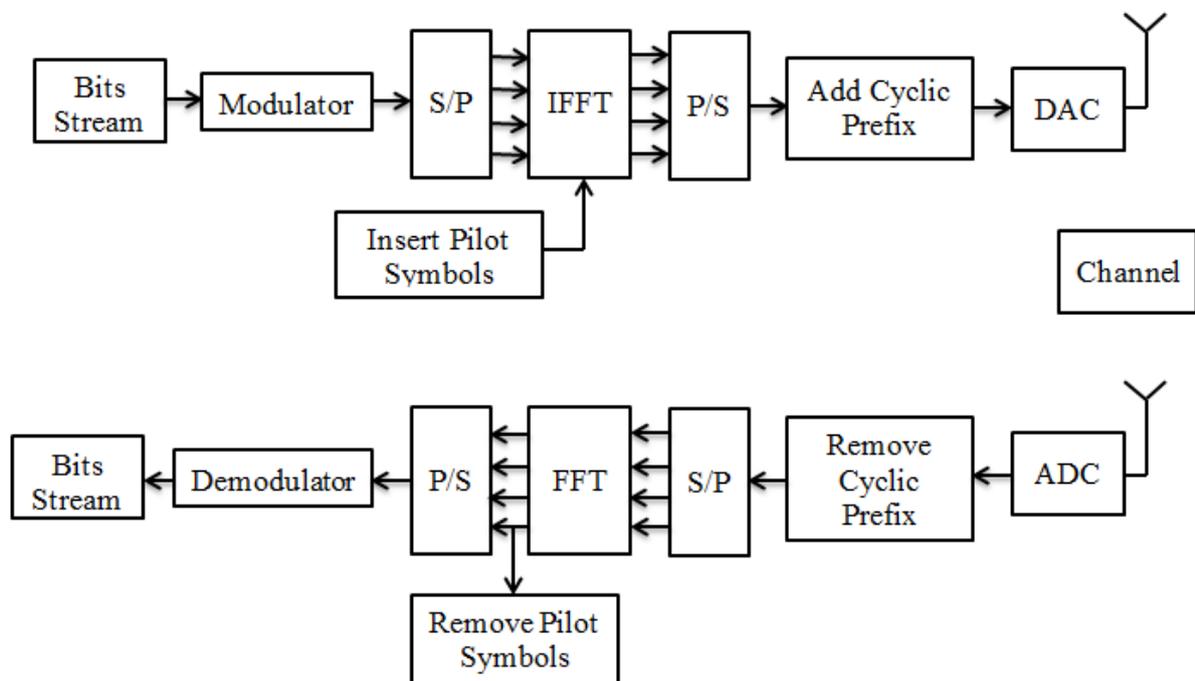


Fig. 4.2 Architecture of OFDM

The incoming serial data is first converted from serial to parallel and grouped into  $x$  bits each to form a complex number. The complex numbers are modulated in a base band by the IFFT and converted back to serial data for transmission. A guard interval is inserted between symbols to avoid intersymbol interference (ISI) caused by multipath distortion. The discrete symbols are converted to analog and lowpass filtered for RF up-conversion. The block diagram of OFDM as shown in Figure 4.2. The receiver performs the reverse operation of the transmitter, mixing the RF signal to base band for processing, then using a Fast Fourier Transform (FFT) to analyze the signal in the frequency domain. The amplitude and phase of the sub carriers is then picked out and converted back to digital data. The IFFT and the FFT are complementary function and the most appropriate term depends on whether the signal is being received or generated.

#### **4.5 SERIAL TO PARALLEL CONVERSION**

Data is transmitted in the form of serial data stream. In OFDM, each symbol transmits 40 - 4000 bits, so in serial to parallel conversion stage need to convert input serial bit stream into parallel bit stream. The data allocated to each symbol depends on modulation schemes and number of subcarriers. At the receiver side, reverse process takes place, data from the sub carriers being converted back to the original serial data stream. For example, consider 16 QAM modulation scheme, each sub carrier carries 4 bits of data, the number of sub carriers becomes 100 so the number of bits per symbol would be 400. When an OFDM transmission techniques is used in a multipath fading environment, in this case frequency selective fading may occur so that more number of bits in error.

#### **4.6 FFT AND IFFT TRANSFORM**

The main components of the OFDM system are IFFT at the transmitter side and FFT at the receiver side. These operations perform between  $N$  complex data symbols and OFDM symbols it will reduces the fading in multipath channel. To overcome the fading in multipath environment, high data stream is split into low data stream during the transmission. Let us consider the data set to be transmitted is,

$$X(1), X(2) \dots X(N)$$

where  $N$  is the total number of sub-carriers.

The discrete-time representation of the signal after IFFT is

$$x(n) = \frac{1}{\sqrt{N}} \sum_{k=0}^{N-1} X(k) e^{j2\pi k \frac{n}{N}} \quad (17)$$

At the receiver side, the data is recovered by performing FFT on the received signal is

$$Y(k) = \frac{1}{\sqrt{N}} \sum_{n=0}^{N-1} x(n) e^{-j2\pi k \frac{n}{N}} \quad k=0 \dots N-1 \quad (18)$$

An N-point FFT only requires  $N \log(N)$  multiplications, which is much more computationally efficient than an equivalent system with equalizer in time domain.

#### 4.7 CYCLIC PREFIX

In an OFDM system, the channel has a finite impulse response. The maximum delay ( $t_{\max}$ ) of all reflected paths of the OFDM transmitted signal, shown in Figure 4.3 it will causes the delay dispersion. The delay dispersion increases the Inter Symbol Interference (ISI) between each subcarrier due to overlapping. By adding cyclic prefix to each symbol, overcome the problem of ISI and ICI. Let us consider a cyclic prefix of length  $N_{cp}$ , where ( $N_{cp} < N$ ), is formed by copying the last  $N_{cp}$  values from the input vector of  $X$  and adding those  $N_{cp}$  values to the front part of the same  $X$  vector.

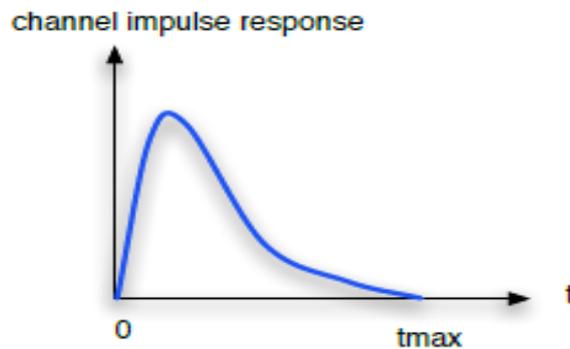


Fig. 4.3 Channel Response with Delay ( $t_{\max}$ )

The idea behind is to convert the linear convolution (between signal and channel response) to a circular convolution. In the FFT of circularly convolved signals is equivalent to a multiplication in the frequency domain. However, in order to preserve the orthogonality property,  $t_{\max}$  should not exceed the duration of the time guard interval. Once the above

condition is satisfied, there is no ISI since the previous symbol will only have effect over samples within  $[0, t_{\max}]$ . And it is clear that orthogonality is maintained so that there is no ICI.

$$r(k) = s(k) \otimes h(k) + e(k) \quad (19)$$

$$Y(n) = DFT(y(k)) = DFT(IDFT(X(n)) \otimes h(k) + e(k)) \quad (20)$$

$$Y(n) = X(n).H(n) + E(n); 0 \leq k \leq N-1 \quad (21)$$

Where  $\otimes$  denotes circular convolution and  $E(n) = DFT(e(k))$ .

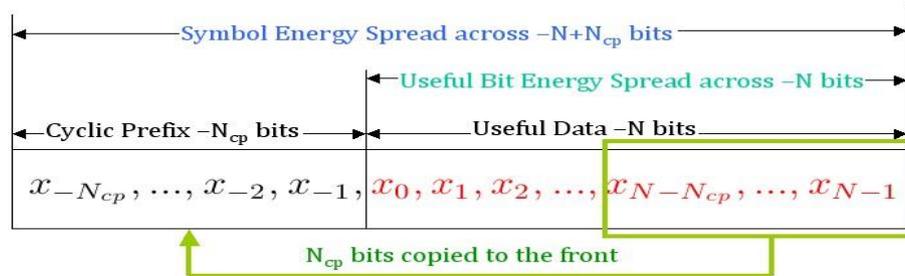


Fig. 4.4 Cyclic Prefix

Another advantage with the cyclic prefix is that it serves as a guard between consecutive OFDM frames. This is similar to adding guard bits, which means that the problem with inter frame interference also will disappear. The representation of cyclic prefix as shown in Fig 4.4. If  $T$  is the duration of OFDM symbol in one seconds, addition of cyclic prefix length is  $N_{cp}$ , the total duration of the OFDM symbol becomes  $T + T_{cp}$ ,  $T_{cp}$  calculated by using  $N_{cp}$  and  $N$ , so  $T_{cp} = N_{cp} * T / N$ . Number of samples consider for cyclic prefix is  $N_{cp} = T_{cp} * N / T$ , Where  $N$  denotes the FFT/IFFT length and  $T$  is the IFFT/FFT period and  $T_{cp}$  is the duration of cyclic prefix.

## 4.8 SUB CARRIER MODULATION

One way to communicate a message signal whose frequency spectrum does not fall within that fixed frequency range, and also unsuitable for the channel, is to change a transmittable signal according to the information in the message signal. This alteration is called modulation, and the modulated signal that is transmitted. The receiver then recovers the original signal through a process called demodulation. Modulation is a process by which a carrier signal is altered according to information in a message signal. The carrier frequency, denoted  $F_c$ , is the frequency of the carrier signal. The sampling rate,  $F_s$ , is the rate at which the message signal is sampled during the simulation. The frequency of the carrier signal is usually much greater than the highest frequency of the input message signal. The Nyquist sampling theorem requires that the simulation sampling rate  $F_s$  be greater than two times the

sum of the carrier frequency and the highest frequency of the modulated signal, in order for the demodulator to recover the message correctly.

QAM is a method for sending two separate (and uniquely different) channels of information. The carrier is shifted to create two carriers namely the sine and cosine versions. The outputs of both modulators are algebraically summed and the result of which is a single signal to be transmitted, containing the In-phase (I) and Quadrature (Q) information. The set of possible combinations of amplitudes is a pattern of dots known as a QAM constellation. Once each subcarrier has been allocated bits for transmission, they are mapped using a modulation scheme to a subcarrier amplitude and phase, which is represented by a complex Inphase and Quadrature-phase (IQ) vector. Fig 4.5 shows an example of subcarrier modulation mapping. This example shows 16-QAM, which maps 4 bits for each symbol. Each combination of the 4 bits of data corresponds to a unique IQ vector, shown as a dot on the figure 4.5. A large number of modulation schemes are available allowing the number of bits transmitted per carrier per symbol to be varied.

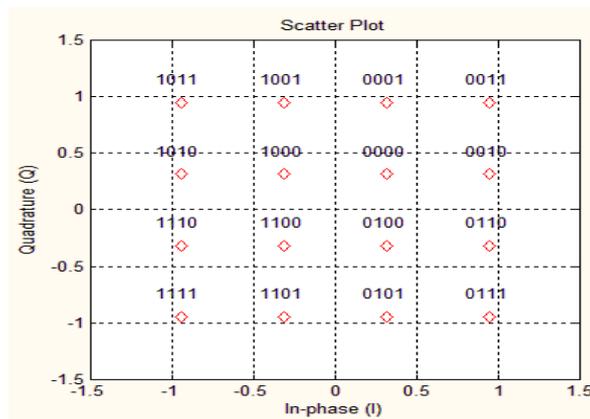


Fig. 4.5 Constellation of QAM Modulation

Subcarrier modulation can be implemented using a lookup table, making it very efficient to implement. In the receiver, mapping the received IQ vector back to the data word performs sub carrier demodulation.

## 4.9 FREQUENCY TO TIME DOMAIN CONVERSION

After the subcarrier modulation stage each of the data sub carriers is set to amplitude and phase based on the data being sent and the modulation technique. All unused sub carriers

are set to zero. This sets up the OFDM signal in the frequency domain. An IFFT is used to convert this signal to the time domain, allowing it to be transmitted. In the frequency domain, before applying the IFFT, each of the discrete samples of the IFFT corresponds to an individual sub carrier. Most of the sub carriers are modulated with data. The outer sub carriers are unmodulated and set to zero amplitude.

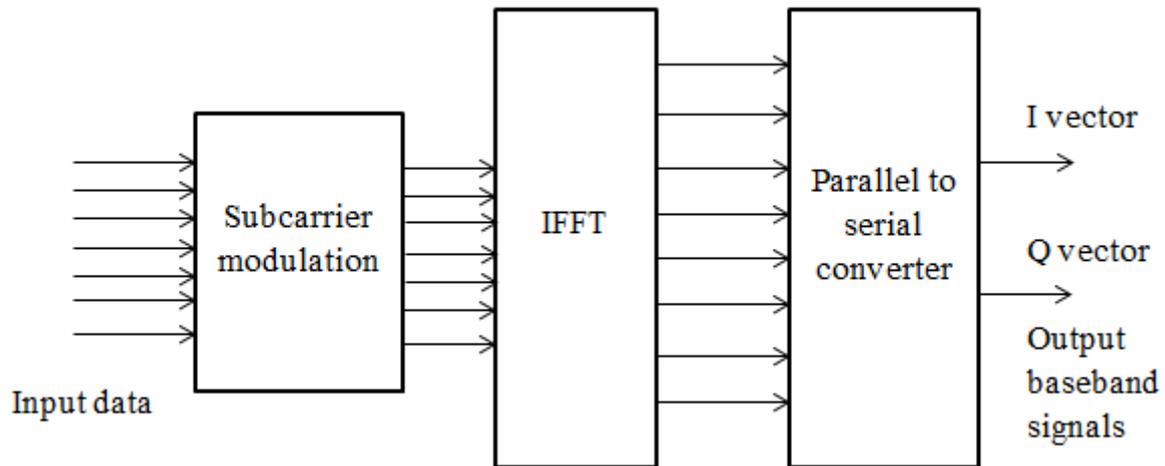


Fig. 4.6 OFDM generation in IFFT stage

In a digital communications system, the symbols that arrive at the receiver have been convolved with the time domain channel impulse response of Length  $L_c$  samples. Thus, the effect of the channel is convolution. In order to undo the effects of the channel, another convolution must be performed at the receiver using a time domain filter known as an equalizer. The length of the equalizer needs to be on the order of the time span of the channel. The equalizer processes symbols in order to adapt its response in an attempt to remove the effects of the channel. Such an equalizer can be expensive to implement in hardware and often requires a large number of symbols in order to adapt its response to a good setting. In OFDM, the time domain signal is still convolved with the channel response. However, the data will ultimately be transformed back into the frequency-domain by the FFT in the receiver. Because of the periodic nature of the cyclically-extended OFDM symbol, this time domain convolution will result in the multiplication of the spectrum of the OFDM signal with the frequency response of the channel. The result is that each sub carrier's symbol will be multiplied by a complex number equal to the channel's frequency response at that sub carrier's frequency. Each received sub carrier experiences a complex gain due to the channel.

## **DESIGN PARAMETERS**

The basic design parameter choose for OFDM generation such as

- FFT Length
- Number of used subcarrier
- Bandwidth
- Baseband Sampling Frequency
- Symbol Duration
- Subcarrier Spacing
- Guard Interval/Cyclic Prefix
- Guard Band

## **ADVANTAGES**

- Makes efficient use of the spectrum by allowing overlap.
- By dividing the channel into narrowband flat fading sub channels, OFDM is more resistant to frequency selective fading than single carrier systems.
- Eliminates ISI and IFI through use of a cyclic prefix.
- OFDM is computationally efficient by using FFT techniques to implement the modulation and demodulation functions.
- Using adequate channel coding and interleaving one can recover symbols lost due to the frequency selectivity of the channel.
- Channel equalization becomes simpler than by using adaptive equalization techniques with single carrier systems. It is possible to use maximum likelihood decoding with reasonable complexity.
- Is less sensitive to sample timing offsets than single carrier systems are.
- Provides good protection against co-channel interference and impulsive parasitic noise.

## **DISADVANTAGES**

- The OFDM signal has a noise like amplitude with a very large dynamic range; therefore it requires RF power amplifiers with a high peak to average power ratio.
- It is more sensitive to carrier frequency offset and drift than single carrier systems are due to leakage of the DFT.

# CHAPTER 5

## PROPOSED SYSTEM

### 5.1 CHANNEL CODING IN LTE STRANDARD

One of the most important issues in digital communication systems is error correction. Error correction can be considered in two main categories such as ARQ (Automatic Repeat Request) and FEC (Forward Error Correction) technique. In ARQ, the receiver requests retransmission of the data packets, if errors are detected or until the received packets becomes error-free or a maximum number of retransmissions are reached. For FEC redundancy bits are added to data bits, making error correction possible. In LTE both block codes and convolutional codes are used. CRC (Cyclic Redundancy Check) which is a cyclic linear block code is used for an error detection technique. Control channels and data channels in LTE use convolutional and Turbo codes respectively. A turbo code is formed by concatenating two or more codes. These basic codes can be convolutional codes or block codes. In the proposed system, turbo codes are used in LTE standard for improving the bit error rate performance, in addition to that OFDM technique are used to improve the data transmission as well as spectral efficiency. The combination of OFDM and recursive turbo decoding achieves near Shannon's limit channel capacity.

### 5.2 LTE TURBO ENCODER

The turbo code discussed here is used by the long term evolution (LTE) specification, as standardized by the Third-Generation Partnership Project (3GPP). The scheme of turbo encoder is a Parallel Concatenated Convolutional Code (PCCC) with two 8-state constituent encoders and one turbo code internal interleaver. The coding rate of turbo encoder is  $1/3$ . The structure of turbo encoder is shown in Fig. 5.1. A turbo encoder is a parallel concatenation of two codes. The two codes used are RSC Codes which are discussed in chapter 3. As the two codes work parallel on the input data, they are called as parallel concatenated convolutional code encoder. The interleaver used is a pseudo random interleaver which randomizes the input data in an irregular manner. More generally, encoders used by LTE turbo code is comprised of a pair of constraint length  $k = 4$  RSC encoders, each one identical to the other. The overall code rate is  $r = 1/3$ . The size of the input data word is varied in the range of 40 bits to 6,144 bits.

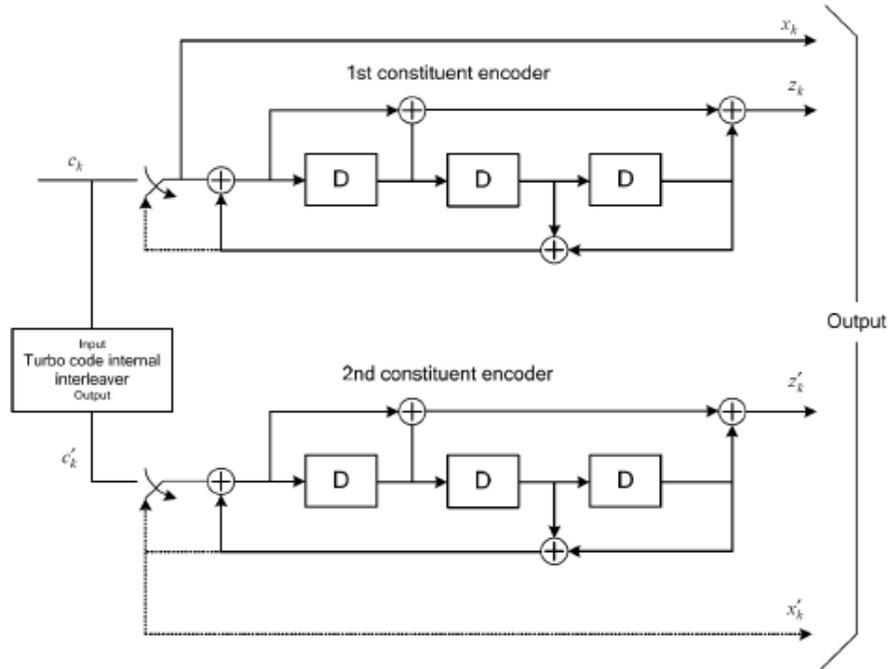


Fig. 5.1 LTE Turbo Encoder

The nature of turbo codes is to find the minimum Hamming distance of the code can be determined by comparing each possible code word with the all-zeroes code word. This process simplifies analysis of the code somewhat, and the minimum Hamming distance is then equal to the minimum code weight (number of '1's) which occurs in any code word. The minimum Hamming distance tends to determine the BER performance of the code at high SNR the asymptotic performance. A high minimum Hamming distance results in a steep rate of fall of BER as SNR becomes large, whereas a low value results in a slow rate of fall. In the case of a turbo code, this rate of fall at high SNR is so slow; it is termed an error floor. At low SNR, however, code words with code weights larger than the minimum value must be considered.

### 5.3 DESIGN OF INTERLEAVER

The interleaver is an important part of the turbo codes. Interleaver is a simple device that rearranges the ordering of data in a pseudo random way. The outputs of the two codes are actually the same, but because of the presence of the interleaver the order of these is changed. Due to the presence of the interleaver, one of the encoder outputs low weight code words. Information data bits are first divided into rectangular matrix with same row and column to reduce the minimum hamming distance by rearranging the encoded bits. After permutes the input bits take modulus of each bit to find the modulation index of the received bits.

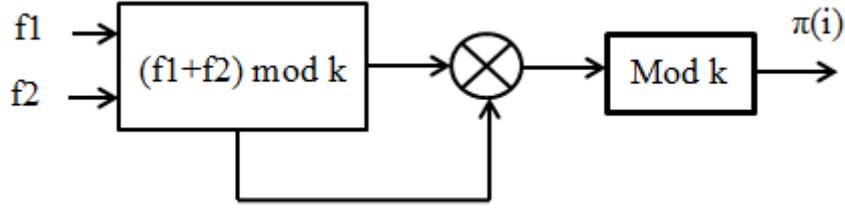


Fig. 5.2 Interleaver Design

From the Fig. 5.2, the input bits to the turbo code internal interleaver are  $c_k$ , where  $K$  is the number of input bits. The output bits from the turbo code internal interleaver are  $c'_k$ . The relationship between the input and output bits is as follows,

$$C_i = C_{\Pi(i)}, i = 0, 1, \dots, (K - 1) \quad (22)$$

I	K	f1	f2
1	40	3	10
4	64	7	16
12	128	15	32
20	192	23	48
60	512	31	64
86	928	15	58
92	1024	31	64
124	2048	31	64

Fig.5.3 Interleaved Bits

Where the relationship between the outputs indexes  $i$  and the input index  $\Pi(i)$  satisfies the following quadratic form:

$$\Pi_{(i)} = (f1.i + f2.i^2) \quad (23)$$

The parameters  $f1$  and  $f2$  depend on the block size  $K$  and are summarized in table 5.3.

## 5.4 TURBO DECODER

A turbo decoder consists of two single soft-in-soft-out (SISO) decoders, which work iteratively. The output of the first decoder feeds into the second to form a turbo decoding iteration. The interleaver and de-interleaver blocks reorder input data in this process. The decoding of turbo codes is an iterative process. The turbo decoder is shown in Fig 5.4. This is performed by decoding each constituent code and sharing the information in an iterative

manner. Each decoder produces a posteriori information, which is used as a priori information by the other decoder. The type of decoding used by turbo codes is called the Maximum A-Posteriori (MAP) algorithm which is same as the SISO decoding algorithm. In turbo codes, the MAP algorithm is used iteratively to improve the performance. Max-log-Map is a simplified version of Log MAP and has reduced BER performance compared to the MAP algorithm and it has a less computational complexity.

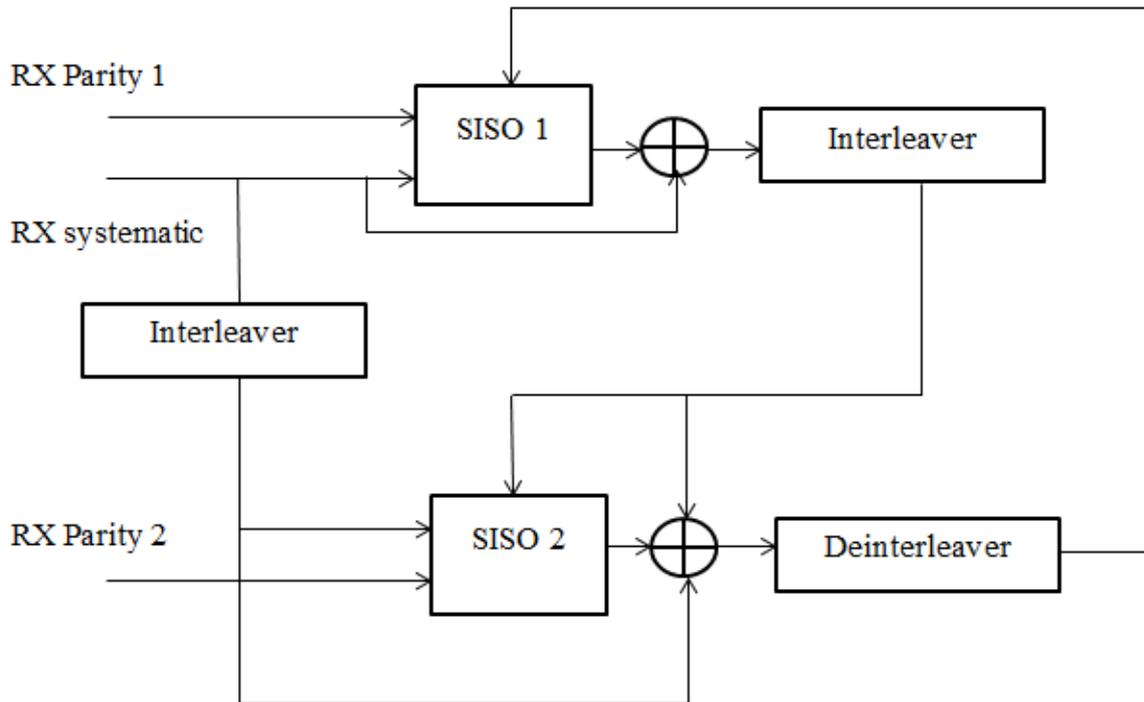


Fig.5.4 Turbo Decoder

## 5.5 CYCLIC REDUNDANCY CHECK ALGORITHM

Cyclic Redundancy Check is one of the most versatile error checking algorithms used in various digital communication systems. Normally, for the error detection in digital communication systems; a checksum is computed on the message that needs to be transmitted. The computed checksum is then appended at the end of the message stream and it is transmitted. At the receiving end, the message stream checksum is computed and compared with the transmitted checksum. If both are equal, then the message received is treated as error free. CRC works in a similar way, but it has greater capabilities for error detection than the conventional forms. Different CRC polynomials are employed for error detection. The size of CRC depends upon the polynomial chosen.

## CRC Operation

All CRC calculations are carried out in the GF (2) (Galois field for 2 elements); in which we can perform addition, subtraction, multiplication and division and the 2 elements specifies the field in which only two values, either 1 or 0. Modulo 2 arithmetic uses binary addition or subtraction with no carry, which is equivalent to the XOR operation. Multiplication and division are similar to binary multiplication and division, respectively. The message to be transmitted is treated as a polynomial and divided by an irreducible (prime) polynomial known as the generator polynomial. The degree of the generator polynomial should be less than that of the message polynomial. For an  $n+1$  bit generator polynomial, the remainder will not be greater than  $n$  bits. The CRC checksum of the data is the binary equivalent of the remainder after the division.

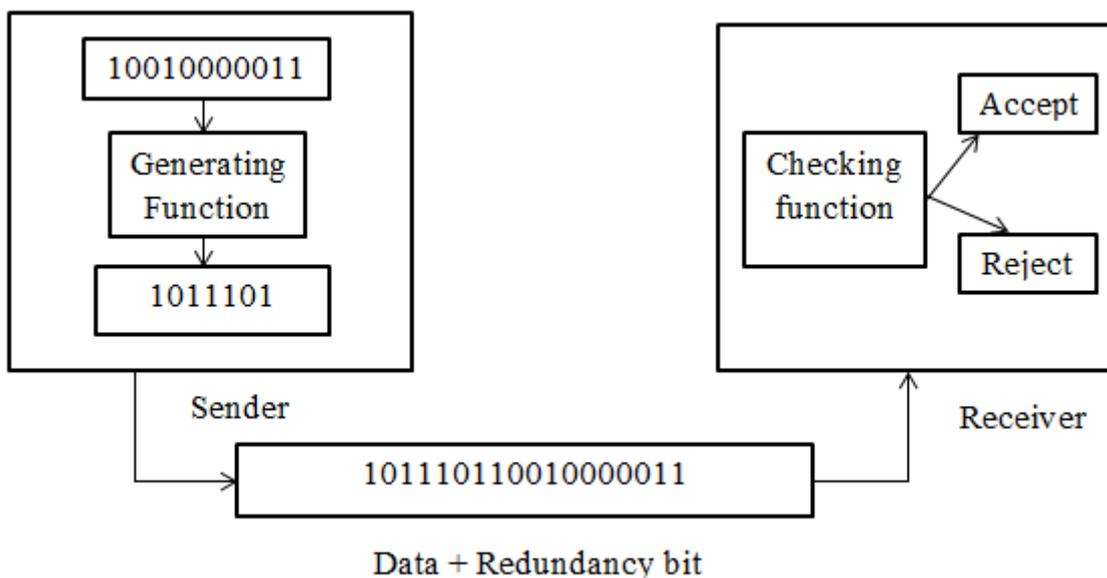


Fig. 5.5 CRC Generator and Checker

Consider a message of  $M$  of  $k$  bits and generator polynomial  $G$  of  $n + 1$  bits. Dividing the message by the generator will yield a remainder  $R$  of  $n$  bits. Therefore,  $M = GQ + R$ , where  $Q$  is the quotient obtained when  $M$  is divided by  $G$ . Now,  $M + R = GQ$ , where the checksum is embedded into the transmitted message ( $M + R$ ). Here, by adding the checksum to the message, we are corrupting the last  $n$  bits of the message. Instead of embedding the checksum in the message, we append it to the message, thus avoiding the corruption of message bits. When the remainder is appended to the message for transmission, it is equivalent to shifting the message bits by the number of remainder bits. Initially, the message stream is appended with zeros. After the checksum calculation, zeros are replaced with the actual checksum

computed, which is the binary equivalent of the remainder of the division. The number of appended zeros is dependent on the degree of generator polynomial.

### CRC Algorithm in Turbo Codes

In the transmitter side, the message sequence consists of  $k+n$  bits that should be divided by some CRC polynomial. At the receiver side, incoming data should be divided by the same CRC polynomial, if get the remainder is one, the receiver assumes error occur in the received data. Suppose the remainder is zero, receiver assumes no error occurs in the received data. In proposed system, CRC is used in decoding process to improve performance of the turbo codes. The turbo decoder iterates until CRC is verified, once if the CRC is verified the decoding process should stops.

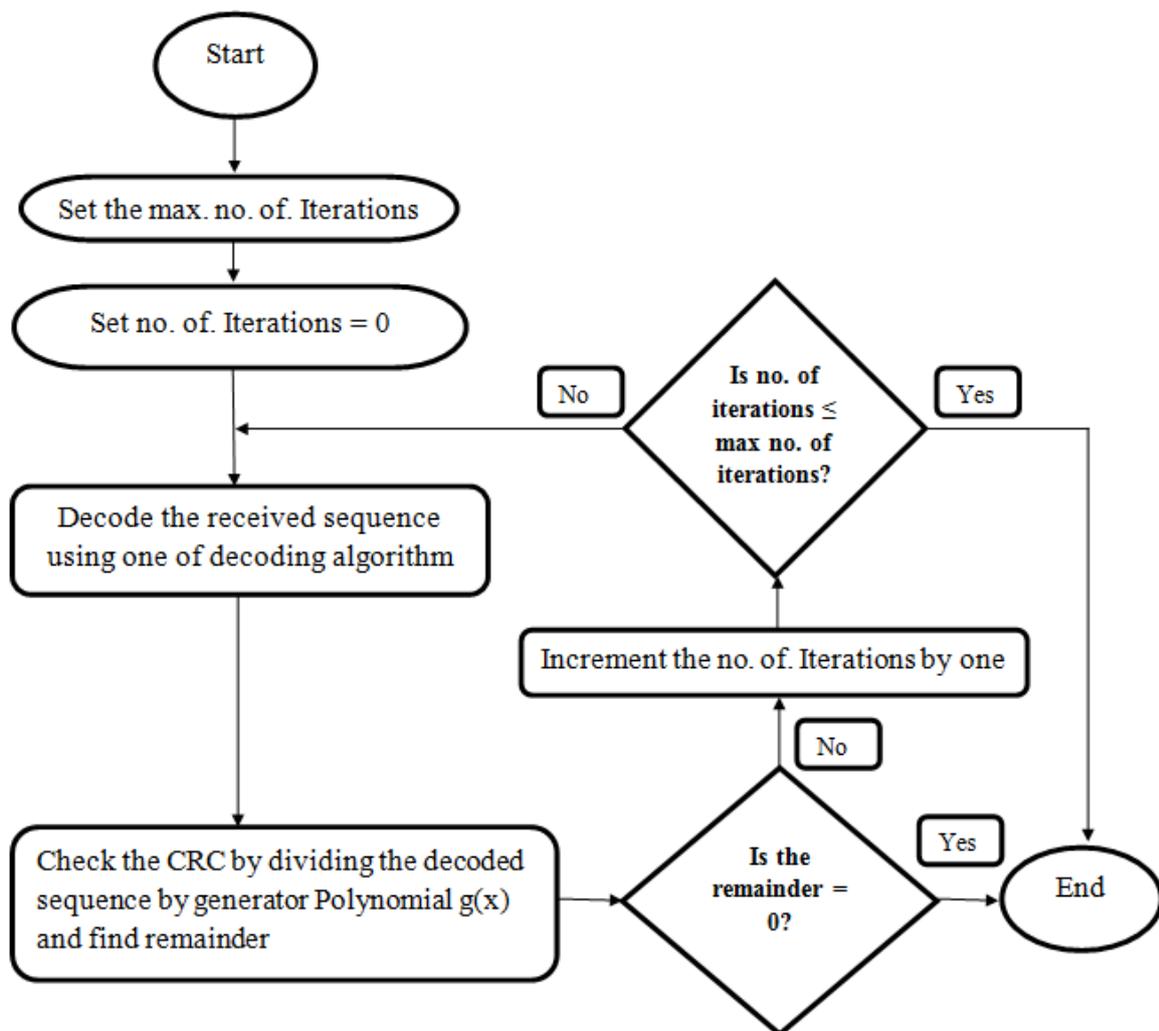


Fig. 5.6 Flow Chart of CRC Algorithm

### CRC Attachment in Turbo Codes

The maximum and minimum code block size are supported by the turbo interleaver where minimum code block size = 40 bits and maximum code block size,  $Z = 6144$  bits. If the length of the input block  $B$  is more than the maximum code block size, the input block is segmented. When the input block is segmented, it is segmented into  $C = \lceil B / (Z - L) \rceil$ , where  $L$  is 24. Therefore,  $C = \lceil B / 6120 \rceil$  code blocks. After segmentation, each code block has a 24-bit Cyclic Redundancy Check (CRC) attached to the end as shown in figure 5.7.

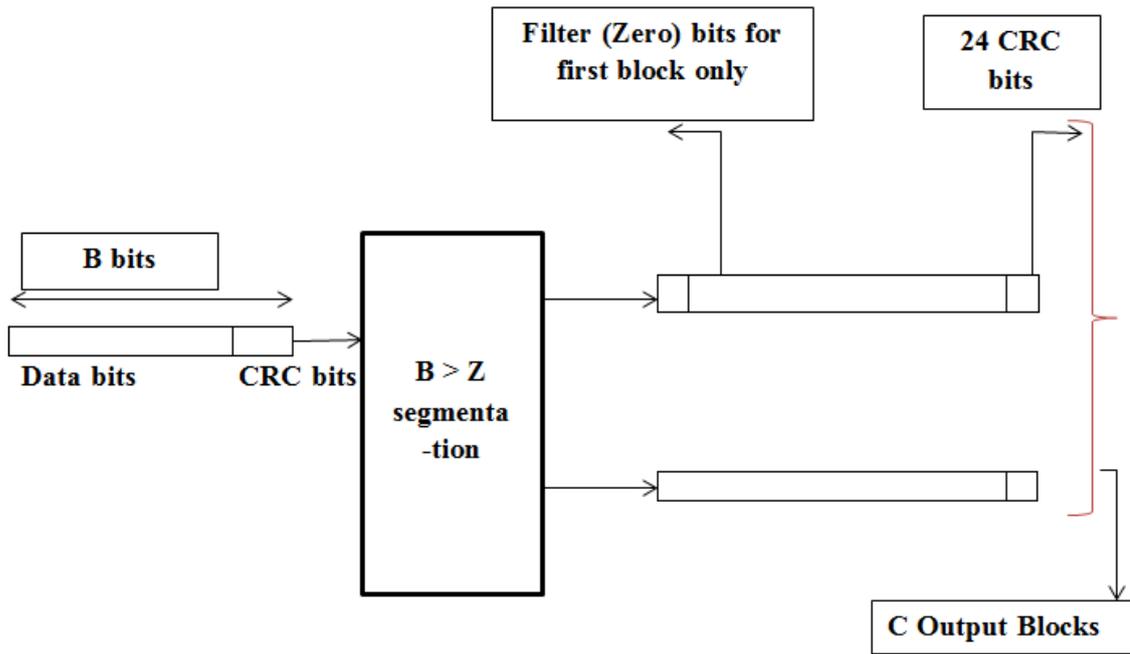


Fig. 5.7 CRC Attachment in Turbo Codes

The generator polynomial for CRC attachment is defined by 3GPP standard,

$$g_{CRC24}(D) = [D^{24} + D^{23} + D^6 + D^5 + D + 1] \quad (24)$$

In CRC algorithm, for a given received bit sequence the turbo decoder may exceed the maximum number of iterations without correcting all error bits and for other received bit sequence the turbo decoder may correct all the error bits before the maximum number of iteration executed. Check the errors in end of the every iteration and if on error detected, stop the decoding process even that the maximum number of iterations has not been executed. In this paper by checking the CRC that attached in the transmitter by LTE standard mainly to help in detecting errors and measure the throughput in the receiver to detect the presence or

absence of any bit errors at the end of every iteration. The flowchart in figure 5.8 explains the procedure of early stopping turbo decoding mechanism.

## 5.6 FLIP AND CHECK ALGORITHM

In this proposed system, Flip and Check Algorithm is used to find the exact position of the least reliable bits present in the generated codeword. The least bits is identified by using state and branch metric values of the extrinsic information. In the proposed system, Flip and check algorithm is used to improve the performance and decreasing the time taken by decoding process. In MAX Log MAP gives better BER perormance after using this algorithm in decoding process.  $Le_1$  represents the extrinsic information of first SISO decoder and  $Le_2$  implies that second decoder output. The reliable bits identification based on Eq. 25. The FC algorithm first iterates minimum times without any CRC verification. The process continues until the CRC is verified. Once, the CRC is verified automatically stops the process. The code word identification is based on presence of reliable bits in the extrinsic information.

$$Ae(k) = Le_1(k) + Le_2(k) \quad (25)$$

## 5.7 IMPLEMENTATION OF TURBO CODES IN OFDM

The code block segmentation and CRC attachment splits the code stream into smaller code blocks if the code stream is longer than the maximum code bock size allowed by the turbo interleaver then it adds 24 CRC bits at the end of code. Turbo encoder encodes the segmented sequence with code rate 1/3. The code block concatenation is sequentially concatenating the outputs from the turbo encoder. The signal mapper modulates the sequence using QPSK modulation. The OFDM mapper splits the spectrum into narrower parallel channels named as subcarriers and the information is transmitted on these parallel channels at a reduced rate. The channel is simulated as Rayleigh channel. The implementation of OFDM in Turbo codes as shown in Fig 5.9. The combination of OFDM and recursive turbo decoding allows these codes to achieve near Shannon limit performance in the wireless communication systems. At the receiver end, the receiver will perform the inverse operations of the transmitter operations. The turbo decoder will decodes the received sequence using max log map algorithms that mentioned previously with different number of iterations.

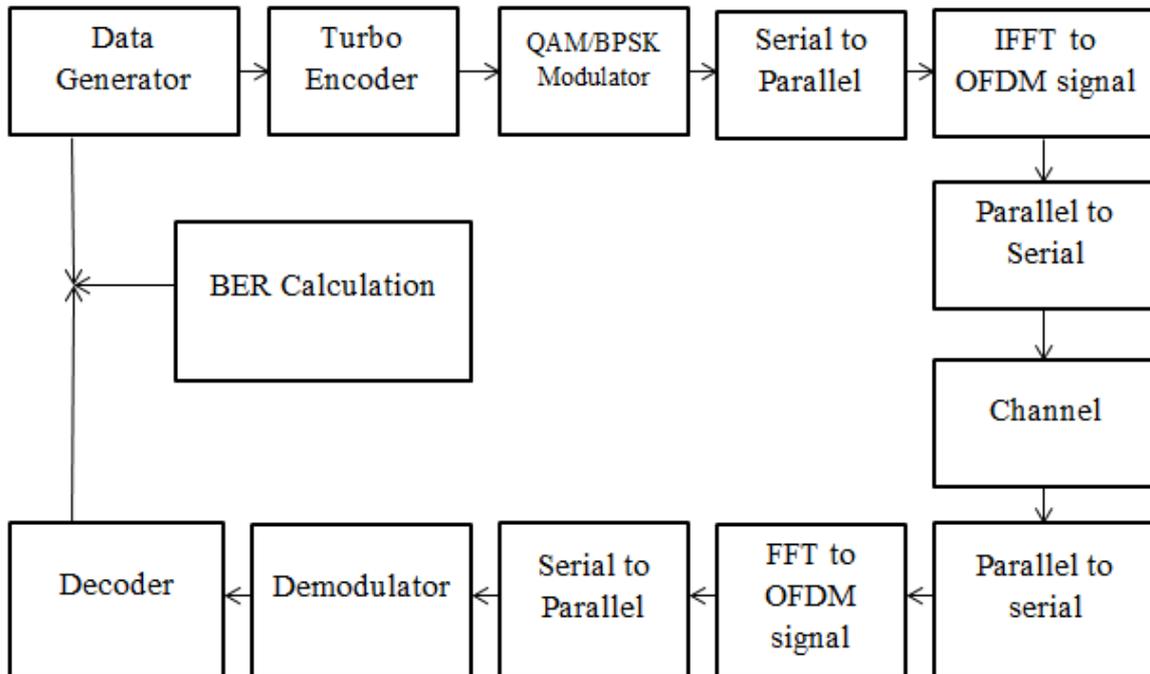


Fig. 5.8 Combining Turbo codes in OFDM

### Algorithm for simulation

For measured the performance of the turbo coded OFDM through MATLAB simulation. The simulation follows the procedure listed below:

1. Generate the information bits randomly.
2. Encode the information bits using a turbo encoder with the specified generator matrix.
3. Use QPSK or different QAM modulation to convert the binary bits, 0 and 1, into complex signals (before these modulation use zero padding)
4. Performed serial to parallel conversion.
5. Use IFFT to generate OFDM signals, zero padding is being done before IFFT.
6. Use parallel to serial convertor to transmit signal serially.
7. Introduce noise to simulate channel errors. Here, the signals are transmitted over an Rayleigh Fading channel.
8. At the receiver side, perform reverse operations to decode the received sequence.
9. Count the number of erroneous bits by comparing the decoded bit sequence with the original one.
10. Calculate the BER and plot.

## CHAPTER 6

### RESULTS AND DISCUSSIONS

The simulated results were obtained using Matlab software. The following results are analysed for TCOFDM with MAP and MAX Log MAP decoding algorithm in the presence of Rayleigh channel and the BPSK modulation scheme is used. The proposed CRC and FC algorithm is used in turbo decoding algorithm to improve BER performance of the turbo codes. To improve the performance of the LTE standard, FEC code can be used. Turbo coding in OFDM can give performance improvement on Rayleigh Fading channel. Here turbo codes are based on the code rate  $1/3$ , constraint length 3 and (7, 5) generators matrix has been used. In this method BPSK and QAM modulation technique are used. The choice of the interleaver becomes Pseudo random interleaver. The turbo codes has various turbo decoding algorithm here MAP and Max Log algorithm has been considered to compare the computational complexity of that algorithm. Compared to MAP algorithm, Max Log MAP algorithm has a less computational complexity because it performs in logarithmic domain. From the simulated analysis, inferred that Max Log MAP gives BER performance better than MAP decoding algorithm. So, choice of the decoding algorithm for LTE standard becomes MAX Log MAP algorithm. After using the proposed CRC and FC algorithm in decoding process achieves better performance and also reduce the time delay.

The turbo coded OFDM system gives better performance for LTE standard by using the proposed CRC and FC algorithm in turbo decoding process. While using the CRC in turbo codes, the time latency becomes reduced as well as maintains the better BER performance. Before reaching the fixed number of iterations, the turbo decoding process stops by verifying CRC. Interleaver is also important in turbo codes which used to find the low weight code words for improving the bit error rate performance. The coding gain for BER  $10^{-3}$ , the SNR becomes 15dB between the coded and uncoded OFDM system. The TCOFDM gives better performance at low SNR values.

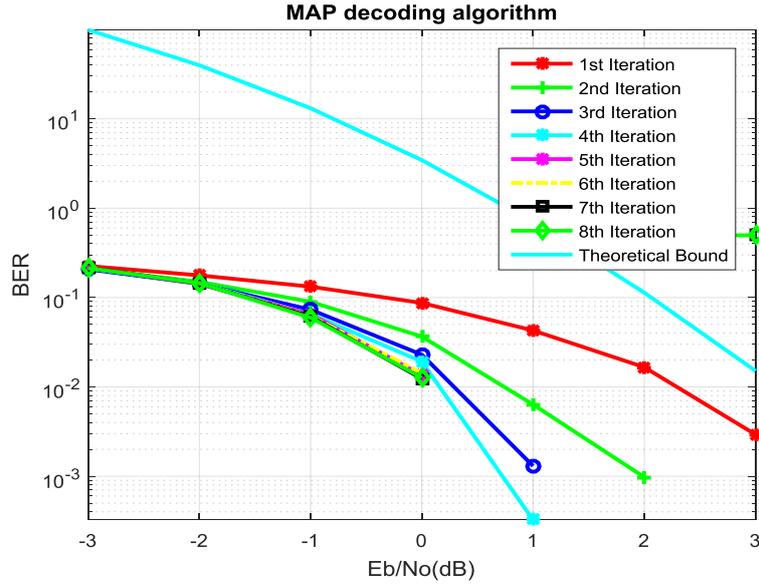


Fig. 6.1 Turbo codes with MAP decoding algorithm using CRC and FC algorithm.

In Fig 6 shows the BER performance of MAP decoding algorithm over Rayleigh fading channel. The code rate is 1/3; the number of iteration up to 8 and the block size is 6144 bits. MAP decoding has a poor BER performance and high computational complexity so it is not suited for many wireless communications.

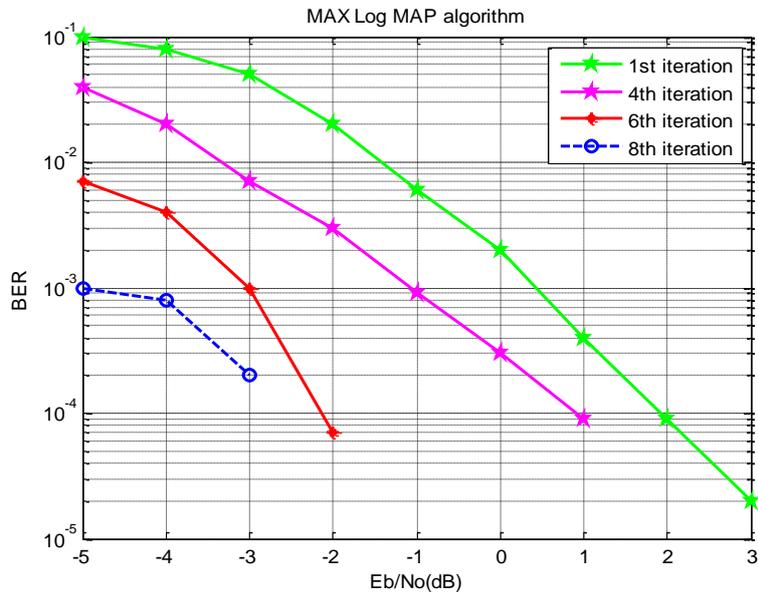


Fig. 6.2 Turbo codes with Max Log MAP decoding algorithm using CRC and FC algorithm.

In Fig 6.2 shows the BER performance of Max Log MAP decoding algorithm over Rayleigh fading channel. The number of iteration up to 8 and the block size is 1024 bits. It has a better BER performance and less computational complexity so it is well suited for many wireless communications such as LTE standard.

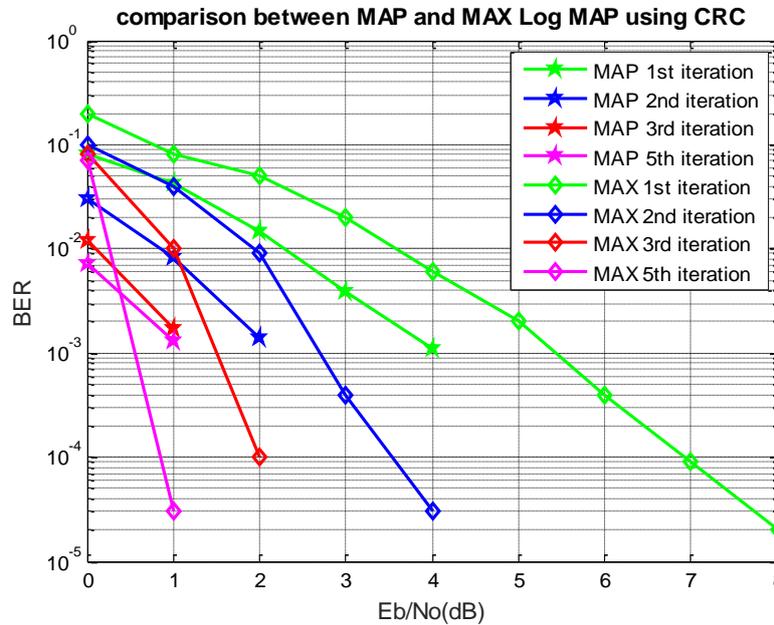


Fig. 6.3 Performance comparison between the MAP and MAX -Log MAP d algorithm of turbo codes using CRC and FC

In Fig 6.3 shows the comparison of BER performance for MAP and Max Log MAP decoding algorithm using proposed CRC and FC algorithm. After using the CRC and FC algorithm in decoding process the bit error rate get decreased. From the simulation analysis, the MAX log MAP decoding algorithm is the best choice for LTE standard.

**Table 6.1** Shows the computational complexity of turbo decoding algorithm. MAP algorithm is twice as complex as compared to Max-Log Map algorithm for memory  $M = 3$ .

OPERATION	MAP ALGORITHM	Max-Log-MAP ALGORITHM
Maximization	$2M-1= 5$	$5M-2= 13$
Addition	$4M= 12$	$10M= 30$
Multiplication	$10M= 30$	0
<b>Total operation</b>	<b><math>14M= 42</math></b>	<b><math>10M-2= 28</math></b>

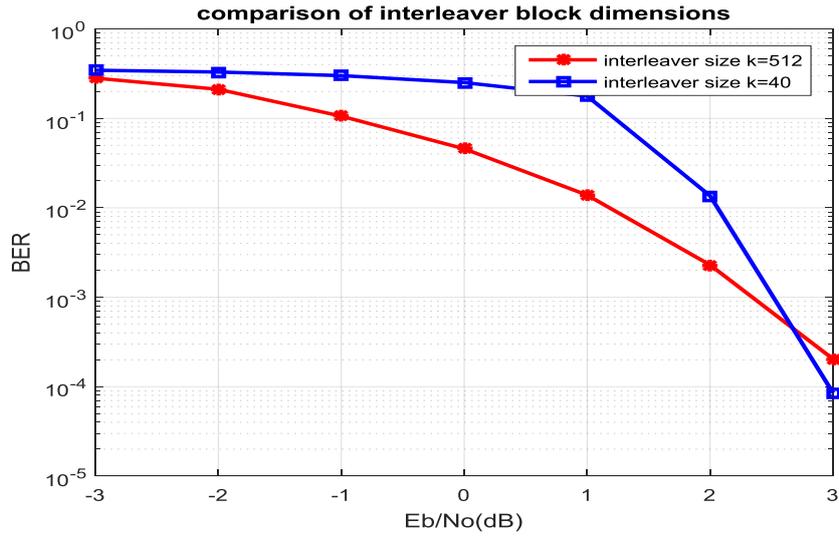


Fig.6.4 Performance comparison between interleaver block dimensions before using CRC and FC algorithm in decoding process.

In Fig 6.4 shows the performance of interleaver block dimensions by varying the length of the interleaver. Length of interleaver is large it will randomize the errors and improve the error performance in turbo codes. Here, the length of the interleaver is 40 and 512 and the number of iterations required up to 3.

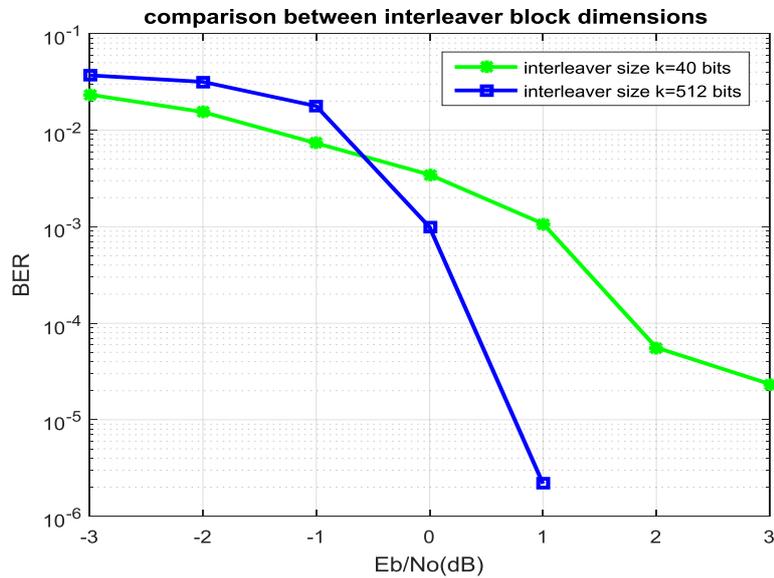


Fig. 6.5 Performances comparison between interleaver block dimensions after using CRC and FC algorithm in decoding process.

In Fig 6.5 shows the performance of interleaver block dimensions by varying the length of the inter-leaver. Length of interleaver is large it will improve the error performance in turbo codes. Here, the length of the interleaver is 40 and 512 and the number of iteration required

up to 3. After using CRC and FC algorithm in decoding process, the bit error rate get decreased when increasing the size of the interleaver.

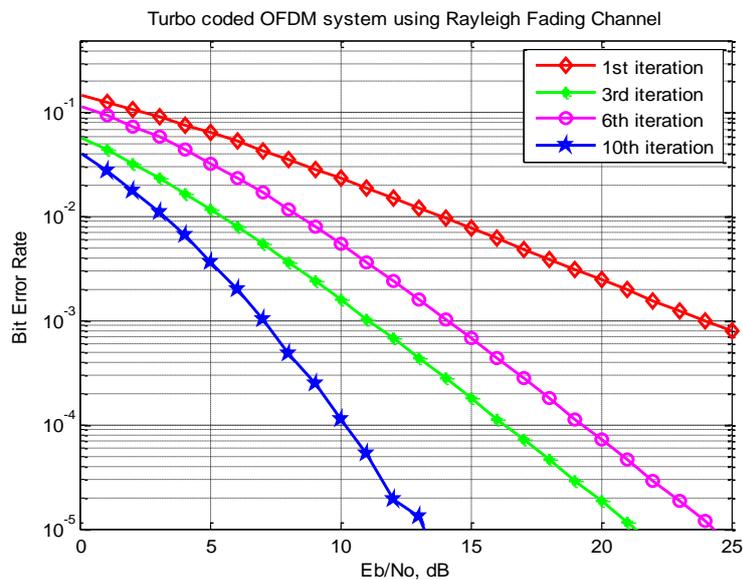


Fig. 6.6 BER performance of Turbo Coded OFDM system

In Fig 6.6 shows the performance of Turbo Coded OFDM system by varying the number of iterations. After combining the OFDM and turbo codes the bit error rate and signal to noise ratio get decreased. Here, size of the block becomes  $10^{-3}$  and the number of iteration required up to 10. After using CRC and FC algorithm in decoding process, the bit error rate get decreased and as well as SNR becomes very low.

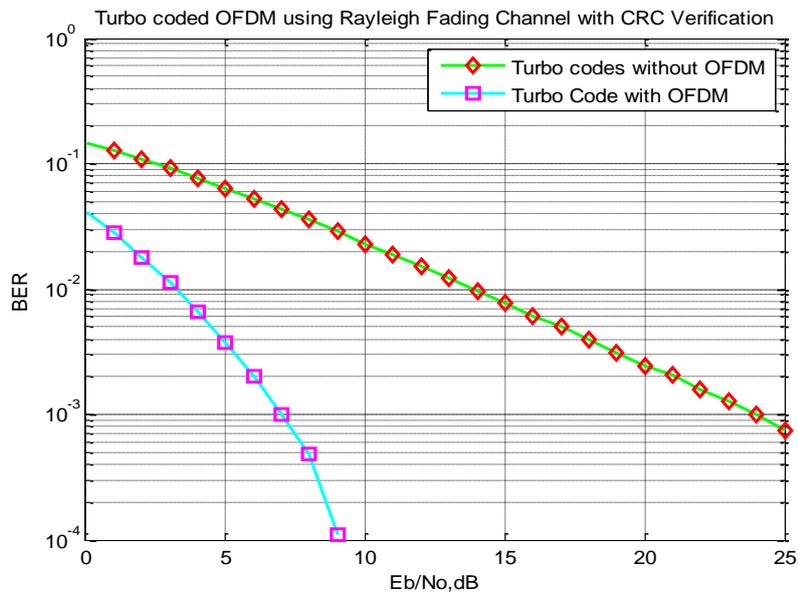


Fig. 6.7 Performance comparisons between with and without Turbo Codes in OFDM

In Fig 6.7 shows the performance of uncoded and Turbo Coded OFDM system. From the simulation analysis, after combining the OFDM with turbo codes the bit error rate and signal to noise ratio get decreased. From the simulated graph, inferred that turbo codes gives better performance while the number of iterations becomes minimum (8 to 10). After using the CRC and FC algorithm in the turbo decoding process it gives better BER performance compared to general turbo code process. Turbo codes have been implemented in the OFDM multiple access technique for increasing the data transmission and minimizing the error occurred in the received data. For the BER  $10^{-3}$  the SNR becomes 15 dB between uncoded OFDM and turbo coded OFDM system.

## CHAPTER 7

### CONCLUSION

Thus the BER performance of LTE TCOFDM is evaluated using CRC and FC algorithm in decoding process over Rayleigh fading channel. For many received sequences the turbo decoder will correct all the errors before achieving the maximum number of iterations. So, using early stopping mechanism by checking the CRC at the end of every decoding iteration will reduce the decoding time without affecting on the BER performance. When analysing the simulated results, comparing with MAP decoding algorithm, MAX Log MAP algorithm provides better BER performance and it has less computational complexity. From the analysis of simulated results, for the BER  $10^{-3}$  the coding gain becomes 15 dB between uncoded OFDM and turbo coded OFDM system. As a result, the TCOFDM system with least number of decoding iterations, 8 to 10 iterations are shown to be sufficient to provide good BER performance. It is concluded that turbo codes provide better error performance and sufficient coding gain as compared with other coding schemes. The type of interleaver and choice of decoding algorithm is important to improve the performance in turbo codes.

Orthogonal frequency division multiplexing used for high data rate transmission and it suffers from high Inter symbol interference (ISI) and Inter channel interference (ICI). ISI and ICI introduces more burst error in the data transmission, it will degrades the performance of the system. To solve this problem, FEC techniques like turbo codes are used. LTE turbo encoder and decoder consist of parallel of concatenation of convolutional codes which is iteratively decoded by Max Log MAP algorithm. From the simulated results, it is concluded that combining recent technology of OFDM and Turbo codes gives better performance for LTE standard in wireless communication.

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## LIST OF CONFERENCE

- Presented a paper titled **“Reducing Bit error rate using CRC verification in turbo codes”** in **IEEE International Conference on Wireless Communications, Signal Processing and Networking (WiSPNET)** held at SSN College of Engineering, Chennai.

Certificate



IEEE International Conference on  
Wireless Communications, Signal Processing  
and Networking (WiSPNET)

22-24 March 2017, Chennai, India

2022

This is to certify that

G. SOWNDHARYA  
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KUMARAGURU COLLEGE OF TECHNOLOGY  
participated and presented a paper titled

REDUCING BIT ERROR RATE USING CRC VERIFICATION IN TURBO CODES

in the IEEE International Conference on Wireless Communications, Signal Processing and Networking  
(WiSPNET) held at SSN College of Engineering, Chennai, India during 22-24 March 2017.

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