T.R.
KAHRAMANMARAŞ SÜTÇÜ İMAM UNIVERSITY
GRADUATE SCHOOL OF NATURAL AND APPLIED SCIENCES

CODING TECHNIQUES IN LONG TERM EVOLUTION (LTE) WIRELESS SYSTEMS

Younus Ameen MUHAMMED

MASTER THESIS

DEPARTMENT OF BIOENGINEERING AND SCIENCES

KAHRAMANMARAŞ 2013

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M.Sc thesis entitled “CODING TECHNIQUES IN LONG TERM EVOLUTION (LTE) WIRELESS SYSTEMS” and prepared by Younus Ameen MUHAMMED, who is a student at Bioengineering and Sciences Department, Graduate School of Natural and Applied Sciences, Kahramanmaraş Sütçü İmam University, was certified by all the majority jury members, whose signatures are given below.

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CODING TECHNIQUE IN LONG TERM EVOLUTION (LTE) WIRELESS SYSTEMS

(YÜKSEK LİSANS TEZİ)

Younus Ameen MUHAMMED

ÖZ

Bu çalışmada LTE and LTE-Gelişmiş kablosuz haberleşme mimarilerinin gereksinimleri araştırılmıştır. Yoğunlaştığımız diğer bir konu LTE ve LTE-Advanced'teki kanal kodlamasıdır. Bu çalışmada, LTE'de hangi kodlama tekniğinin kullanıldığı, bu kodlama tekniğinin seçilme nedeni ve LTE performasının kanal kodlamasıyla nasıl atrrılabileceği sorularına cevap verilmiştir. Ayrıca bu çalışmada turbo kodlama performansının arttırılması için kod blok boyutu maksimum seçilerek farklı bir yaklaşımda bulunulmuş ve MATLAB yazılımı kullanılarak simulasyon ile sonuçlar elde edilmiştir. Sonuçlar önerilen yöntemle performasın daha iyi olduğunu göstermiştir. Ayrıca bu tezde IEEE 802.16e WiMAX'a papralel birleştirilmiş konvolüsyon kodlama ile sistematik konvolüsyon kodlamayı uygulayarak sonuçlarını karşılaştıdık. Simulasyon sonuçları parallel birleştirilmiş konvolüsyon kodlamanın rekürsif sistematik kodlamaya gore daha iyi performans gösterdiğini ortaya koymuştur.

Anahtar Kelimeler: LTE, Turbo Kodlama, Paralel Birleştirilmiş Konvolüsyon Kodlama, Maksimum Kod Blok Büyüklüğü, RS-CC, LTE İletim Bloğu.

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CODING TECHNIQUE IN LONG TERM EVOLUTION (LTE) WIRELESS SYSTEMS

(M.Sc. THESIS)

Younus Ameen MUHAMMED

ABSTRACT

In this thesis, the requirements and architectures of LTE and LTE-Advanced were investigated. The channel coding in LTE and LTE-advanced is another part we focused on. In this study we answered questions such as, which kind of coding technique is used in LTE? Why LTE selected this technique for channel coding? How to improve performance of LTE by channel coding? We use different technique to improve the performance of turbo coding by taking the maximum code block size in transport block segmentation, and results were obtained through simulations performed in the MATLAB software. Results show that LTE performance is improved with the proposed approach. In this thesis, we also applied parallel concatenated convolutional code (LTE-PCCC) and recursive systematic convolutional code to the IEEE 802.16e WiMAX and compared the results. Simulation result show that parallel concatenated convolution code lead to better performance than recursive systematic convolutional code.

Key Words: LTE, Turbo Coding, Parallel concatenated convolutional code (PCCC), Maximum codeblock size, RS-CC, Transport block in LTE.

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CODING TECHNIQUE IN LONG TERM EVOLUTION (LTE) WIRELESS SYSTEMS

(YÜKSEK LİSANS TEZİ)

ÖZET

Bu tezde LTE ve LTE-Gelişmiş mimarilerinin gereksinimleri araştırılmıştır. Ayrıca LTE kanal kodlamasında iletim bloğu ayrıştırılmasına da yoğunlaşılmıştır. LTE yüksek veri hızları için Turbo kod kullanır ve kontrol bilgisi için konvolüsyonel kodlamayı kullanır. İletim bloğunu altparçalara ayırırken maksimum blok uzunluğunu alarak Turbo kodlama performası arttırılabilir. Ayrıca bütün kod blok uzunlukları aynı ise Turbo kodlamanın performansı en iyi olmaktadır. Bu çalışmada, bütün altblokların uzunluğunu maksimum yaparak kodlama performansını LTE tanımlamasındaki altblok oluşturma davranışı ile karşılaştırdık ve sonuçlar uyguladığımız yöntemin daha iyi performans gösterdiğini ortaya koydu. Turbo kodlama performasının iyileştirilmesi LTE kanal kodlamasını iyileştirir ve bu da genel olarak LTE performasının iyileştirilmesine yol açar.

Bu tezde ikinci iş olarak QPSK ve 64QAM modulasyonu ile parallel sıralı konvolüsyon kodu (LTE PCCC) ve IEEE802.16e WiMAX rekursif sistematik konvolüsyon kodu uygulanmiş ve sonuçları karşılaştırılmıştır. Simülasyon sonuçları LTE PCCC`nin QPSK ve 64QAM modülasyonları için daha iyi performans gösterdiğini ortaya koymuştur.

CODING TECHNIQUE IN LONG TERM EVOLUTION (LTE) WIRELESS SYSTEMS

SUMMARY

In this thesis, we investigated the requirements and architectures of LTE and LTE-Advanced. Also we focused on the transport block segmentation in channel coding in LTE. LTE uses the Turbo code for higher data rates and convolutional codes for control information. Taking the maximum codeblock size when segmenting the transport block into a number of subsegments, improves the performance of Turbo code. Also when all codeblock sizes are the same, it also improves the performance of Turbo code. Then we applied the maximum codeblock size to the LTE, and compared to the old system. Results show that new approach improves the performance. Improving the performance of Turbo code improves the channel coding in LTE, which in turn improves performance of the LTE.

IEEE 802.16e WiMAX uses recursive systematic convolutional codes for higher data rate. By applying the parallel concatenated convolutional code (LTE PCCC) and the recursive systematic convolutional code to the IEEE 802.16e WiMAX with the QPSK and 64QAM modulation we compared the performance of both techniques. Simulation results show that LTE PCCC performs better in both the QPSK and 64QAM modulations.

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Younus Ameen MOHAMMED

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LIST OF ABBREVIATIONS

3G:	Third Generation Wireless Networks.
3GPP:	Third Generation Partnership Project.
ACK:	Acknowledgement.
AMC:	Adaptive Modulation Coding Schemes.
AMPS:	Advanced Mobile Phone System.
BER:	Bit Error Rate.
BMSC:	Broadcast Multicast Service Center.
BPSK:	Binary Phase-Shift Keying.
CB:	Code Block.
CD:	Compact Disc.
CDMA2000:	Code Division Multiple Access.
CFI:	Control Format Indicator.
CPC:	Continuous Packet Connectivity.
CS:	Circuit Switched.
DL:	Downlink.
DL-SCH:	Downlink Sheared Channel.
DVD:	Digital Video Disc.
Eb/N0:	Energy per Bit to Noise Power Spectral Density Ratio.
E-DCH:	Enhanced Dedicated Channel.
EDGE:	Exchanged Data Rates for GSM Evolution.
eNB:	Evolved Node B.
EPC:	Evolved Packet Core.
EPS:	Evolved Packet System.
ETSI:	European Telecommunications Standards Institute.
E-UTRAN:	Evolved UMTS Terrestrial Radio Access Network.
EvDO:	Evolution-Data Optimized.
FEC:	Forward Error Correction.
FM:	Frequency Modulation.
G:	Generation.
GERAN:	GSM EDGE Radio Access Network.
GPRS:	General Packet Radio Service.
GSM:	Global System Communications.

GW:	Getaway.
HARQ:	Hyper Automatic Rapid Request.
HOM:	Higher Order Modulations.
HRPD:	High Rate Packet Data.
HSPA:	High Speed Packet Access.
IEEE:	Institute of Electrical and Electronics Engineers.
IMEIs:	International Mobile Equipment Identity.
IMS:	IP Multimedia Subsystem.
IMT:	International Mobile Telecommunications.
IP:	Internet Protocol.
IS:	Information System.
ITU:	International Telecommunication Union.
LTE:	Long Term Evolution.
MAP:	Maximum a Posteriori.
MBMS:	Multimedia Broadcast Multicast Service.
MCE:	Multicast Coordination Entity.
MCH:	Multicast Channel.
MHz:	Mega Hertz.
MIMO:	Multi Input Multi Output.
MME:	Mobility Management Entity.
MTCH:	Multicast Traffic Channel.
NACK:	Negative Acknowledgment.
NAS:	Non- Access Stratum.
NMT:	Nordic Mobile Telephone.
OFDMA:	Orthogonal Frequency Division Multiple Access.
PCCC:	Parallel Concatenated Convolutional Code.
PCH:	Paging Channel.
PDC:	Data Control Protocol.
PDN:	Packet Data Network.
PLMN:	Public Land Mobile Network.
PoC:	Push-to-Talk Over Cellular.
QAM:	Quadrature Amplitude Modulation.
QoS:	Quality Over Server.
QPP:	Quadratic Permutation Polynomials.

RAID:	Redundant Array of Independent Disks.
RAN:	Radio Access Network.
RAT:	Radio Access Technology.
RLC:	Radio Link Control.
RM:	Rate Matching.
RRC:	Radio Resource Control.
RS-CC	Reed–Solomon Convolutional Code.
RSE:	Recursive Systematic Encoder.
Rx:	Receiver.
SAE:	System Architecture Evolution.
SC-FDMA:	Signal Carrier Frequency Division Multiple Access.
SIC:	Successive Interference Cancellation.
SMS:	Short Message Service.
SNR:	Signal to Noise Ratio.
SON:	Self-organizing Networks.
TA:	Tracing Area.
TACS:	Total Access Communications System.
TAI:	Tracking Area Identity.
TAL:	Tracking Area lists.
TB:	Transport Block.
TDMA:	Time Division Multiple Access.
Tx:	Transmitter.
UCI:	Uplink Control Information.
UE:	User Equipment.
UMTS:	Universal Mobile Telecommunication System.
UP:	Uplink.
UP-SCH:	Uplink Sheared Channel.
UTRAN:	UTMS Terrestrial RAN.
VoIP:	Voice Over IP.
WCDMA:	Wideband Code Division Multiple Access.
WiMAX:	Worldwide Interoperability for Microwave Access.

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1. INTRODUCTION

Technology was developed step by step to get to the advanced level. Today there is a number of electronic equipment which plays a great role to make life easier. These equipment also employ a different design approach than most commercial radio and television systems use [1], such as, cellular phone, internet chatting, internet webbing… etc. The people or users are demanding higher data rate with better quality, and more efficiency. (For example easier to use, smaller in size… etc.) In order to obtain this goal a lot of large companies competing with each other to reach this advanced level.

In the rest of this chapter, the history of wireless communication in general will be focused, the mobile system generation will be illustrated and the structure, objects and the research question will be outlined.

1.1. Overview

If we compare the history of wireless communication and other science, it can be seen that wireless communication developed sharply, then the first wireless communication produced before Pre-industrial age, these systems transmitted information in the distance, for instance smoke signals, torch signaling, flashing mirrors, signal flares, or semaphore flags. Some kind of communication that we can provide in wireless communication, such as, in 1600 people used drums to communicate together in the distance as shown in Figure 1.1. People at that time communicated by the sound of the drums [1].

Figure 2.1. Drums in communication [1].

Semaphores is the first visual telegraph system, were built in 1792 and 1794 in France and Sweden, respectively. Samuel Morse proposed an electrical telegraph. This telegraph was the first device to send messages by using electricity. Telegraph messages were sent through special code for each letter of the message. It was changed the dots and dashes of this code into electrical impulses and transmitted them over telegraph wires. A telegraph receiver on the other side of the wire converted the electrical impulses to dots and dashes on a paper tape. Later, the code became universal and known as Morse Code. Figure 1.2 shows the electric telegraph. Most messages that traveled long distances were entrusted to messengers who memorized them or carried them in writing [2].

Figure 1.2. The first telegraph by Samuel Morse [3].

In the United States, the Morse telegraph was succeeded for a number of reasons, including its simple operation and it's relatively low cost. By 1851, the country had over 50 telegraph companies therefore, most business telegraphs were controlled by the magnetic telegraph company, which held the Morse patents.

After Samuel Morse telegraph, Alexander Graham Bell interested in the education of deaf individuals which led him to develop the microphone in 1876, what he referred as an "electrical speech machine", which better to call a telephone. By 1878, Bell had set up a telephone exchange in New Haven and Connecticut as shown in figure (1.3). By 1884, long distance connections have been made between Boston, Massachusetts and New York City. Bell was imagined a great use for his innovations but he never imagined that telephone line could be used for transmitting video images. Since Alexander Graham Bell's death in 1922, the telecommunication industry has undergone an amazing revolution. This revolution has gotten to a point of one's ability to access information which depends on telecommunications technology. Bell's "electrical speech machine" paved the way for the information superhighway [4].

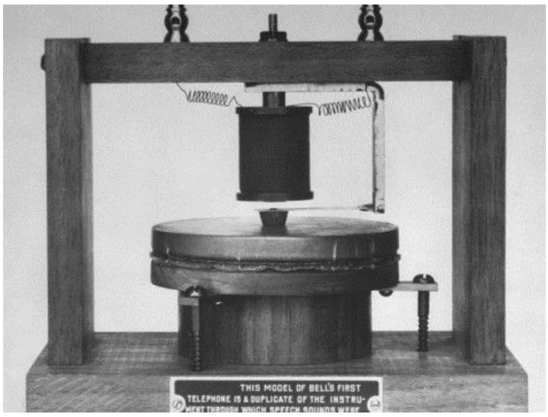

Figure 1.3. First telephone by Graham Bell [5].

Figure 1.4. First mobile radio telephone in 1924 [6].

The ability to provide wireless communication for an entire population was not even conceived until Bell laboratory development the cellular concept in the 1960s and 1970s, with the development of highly reliable, solid-state, miniature, radio frequency hardware in the 1970s. The wireless communication area was born. The recent exponential growth in cellular radio and personal communication systems throughout the world attribute directly to new technologies of the 1970s which are mature today. The future development of consumer-based on mobile and regulatory decisions which affect or support new or extend services, as well as consumer needs a technology advances in the signal processing, access and network areas.

1.2. Mobile System Generation

Mobile communication technologies are often divided into generations, the first-generation of cell phone was introduced in United State in 1979. System is called advanced mobile phone system (AMPS). Several incompatible cellular phone system (TACS, NMT, etc.) In that epoch were in introduced in Weston Europe. Mobile phone being designed for one system could not be used by another system and roaming within a country was not possible [7]. The first generation systems were analog and designed for voice application.

Figure 1.5. Mobile generation evolution [8].

Second generation 2G, was the first system using the digital technology. It was better than first-generation in terms of higher date rate, using the digital voice calling and short message service (SMS). The most popular 2G system was the Global System for Mobile Communications (GSM), which was designed originally as a pan-European and became more popular in the world. In addition, notable IS-95, which is known as CDMAOne designed by Qualcomm, and it became the dominant 2G system in the USA. The technology of the code division multiple access (CDMA), and time division multiple access (TDMA) used in the second generation. This improvement caused the development of the second-generation techniques such as saving power, clearer voice than before by increasing the signal-to-noise ratio. Moreover, the system used a narrow band FM technique which is more secure and safe in the data and voice provide in second-generation. Less than a decade 2.5G added a Packet of Services such as GPRS and EDGE [9].

The main issue with second generation was slow transmission of voice and data. The rapid changes of user expectation also caused development of third generation (3G) network, which has ability to transfer voice data and non-voice data over the same network simultaneously. 3G added the Air Interface: UMTS. It can also provide a number of applications such as internet, email, fax, etc. Moreover, advantage of third-generation is more coverage and growth with less investment. 3G supports both 2G/2.5G and can access to make handover between GSM and UMTS technologies. [9].

Third generation partnership project (3GPP) evolved from a universal mobile telecommunication system (UMTS), which in turn evolved from Global System Communications (GSM). LTE came after both (GSM) and (UTMS). CDMA2000 was developed from IS-95 and it is used in North America. The Impact of the 3G technology was known as CDMA2000 1x radio transmission technology (1xRTT). It was subsequently enhanced to 3.5G system with two alternative names, CDMA2000 high rate packet data (HRPD) or evolution data optimized (EV-DO), where both uses similar techniques to high speed packet access. The specifications of IS-95 and CDMA2000 are produced by a similar collaboration of 3GPP, which is known as the Third Generation Partnership Project 2 (3GPP2) [7].

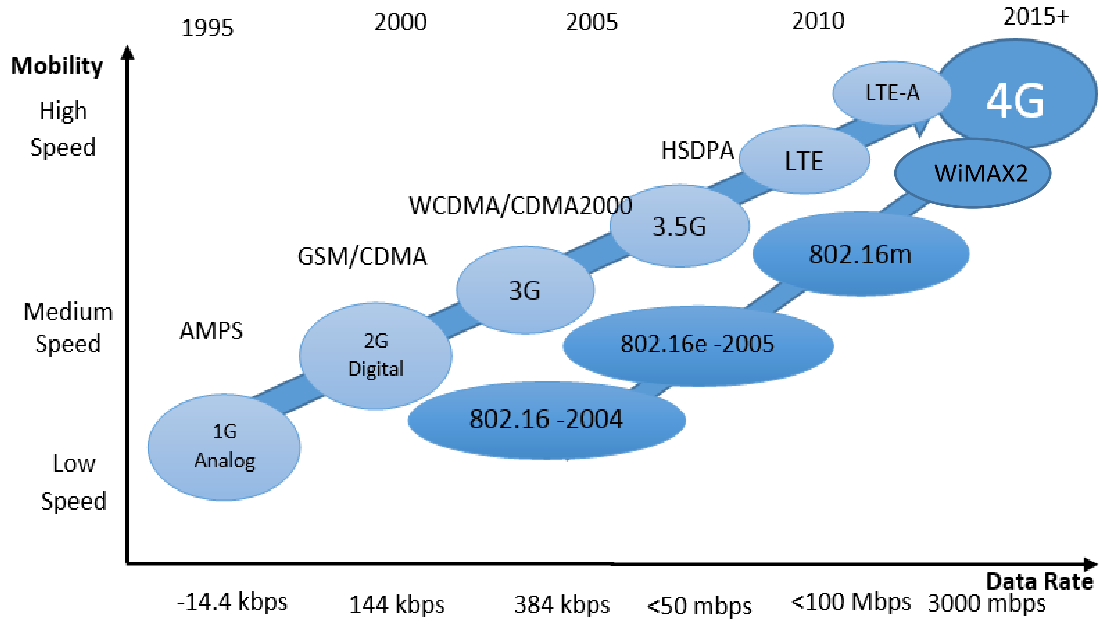

Figure 1.6. Mobile system generation evolution.

The final 3G technology is Worldwide Interoperability for Microwave Access (WiMAX). This was developed by the Institute of Electrical and Electronics Engineers under IEEE standard 802.16 and has a very different history from other 3G systems. The original specification (IEEE 802.16–2001) was for a system that delivered data over point-to point microwave links instead of fixed cables. A later revision, known as fixed WiMAX (IEEE 802.16–2004), supported point-to-multipoint communications between an omnidirectional base station and a number of fixed devices. A further amendment, known as mobile WiMAX (IEEE 802.16e), allowed the devices to move and to hand over their communications from one base station to another. Once these capabilities were all in place, WiMAX started to look like any other 3G communication system, albeit one that had been optimized for data from the very beginning [10].

It has been mentioned before that LTE (Long Term Evolution) is evolved from an earlier 3GPP system which is known as (UMTS). Previously voice call dominated the traffic in the mobile telecommunication networks. Then growing data participated to develop mobile networks in 2009 but it was very slow at the beginning, but after that it increased dramatically.

LTE was generated from 3GPP standard and in 2004 a number of studies have been conducted in a long term evaluation of UMTS. Then it's standardized by Release 8 of 3GPP which required a peak data rate 100 Mbps for downlink and 50 Mbps in uplink. It used OFDMA and SC-FDMA for downlink and uplink respectively, and BPSK, 16QAM, 64QAM modulation. With 2×2 multi input multi output (MIMO) developed the deliveries of peak data rate 300Mbps for DL and 75 Mbps for UL. Moreover, it was required to support a special efficiency three or four times higher than release 6 of 3GPP in the DL and three times in the UL. Another requirement of LTE is that latency, the time taken for data to travel from mobile phone and fix network, should be less than five milliseconds. Coverage and mobility are other requirements of LTE which are optimized by cell size up to 5km. In addition, LTE works with degraded performance up to 30 km and supports cell sizes of up to 100 km [7]. It works with high performance with mobile speed up to 120 kmph and supports speed to 350 kmph. The LTE is designed with various different bandwidth between 1.4 MHz and 20 MHz.

LTE-Advanced is another release of 3GPP which had a lot of requirements. Firstly, delivery of a peak data rate of 500 Mbps in the uplink, and 1000 Mbps in the downlink. In practice, the system has been designed so that it can eventually deliver peak data rates of 1500 and 3000 Mbps respectively, using bandwidth of 100MHz that is made from five separate components of 20MHz.

The LTE-Advanced includes targets for the spectrum efficiency in certain test scenarios. The spectral efficiency is 3.5 to 6 times greater than Release 6 of WCDMA on the uplink, and 4.5 to 7 times greater on the downlink. Finally, LTE-Advanced is designed to be backwards compatible with LTE, in the sense that an LTE mobile can communicate with a base of the station that is operating LTE-Advanced and vice-versa [10].

It is expected that LTE-Advanced will first be commercially available in 2014, with wider deployments by 2015. LTE-Advanced will be both backwards- and forwards-compatible with LTE, meaning that LTE devices will operate in newer LTE-Advanced networks, and LTE-Advanced devices will operate in older LTE networks [51].

Basically the IMT-advanced was announced through ITU while 3GPP supported the studies on LTE which finally caused to issue LTE. The requirements of LTE were outlined in previous sections. The requirements which have to be included in fourth generation will be defined in turn. In October 2010, ITU had announced two systems with IMT-advanced. One of these systems was LTE-Advanced and second was a new version of WiMAX in IEEE.8016m which was called WiMAX 2.0. Neither LTE nor WiMAX 1 performed IMT-advanced. Therefore, the experts of wireless communication accounted them as 3.9G systems. In addition, it was caused to reduce sales of LTE and WiMAX in markets [11].

In this thesis, we investigate the requirements and architectures of LTE and LTE-Advanced, we also focus on the channel coding in LTE and LTE- advanced. In addition we answer these questions such as which kind of coding technique is used in LTE? Why LTE selected this technique for channel coding? How to improve performance of LTE by channel coding? Moreover, we use different technique to improve the performance of turbo coding by taking the maximum codeblock size in transport block segmentation. As the second work in this thesis, we compare the effect of parallel concatenated convolution code and recursive systematic convolutional code in IEEE 802.16e WiMAX.

2. ARCHITECTURE OF THE LONG TERM EVOLUTION (LTE)

2.1. Introduction

At the moment, mobile phones are the most indispensable equipment in life. Without them life will become more difficult. Therefore, mobile communication has to be compared with imposing things in the live such as water, topsoil, or electricity. Because the requirements of life are increasing rapidly, the mobile phones should develop as a rival because technologies are involved in most of our activities in our daily tasks. ITU‘s work permeates into every business, office, hospital, school, and home.

Subscriptions of mobile phones are increasing rapidly. The subscriptions were grown from 1 billion in 2000 to 6 billion in June 2012. Ownership of multiple subscriptions is becoming increasingly common. In last 11 years it was reached to almost 6 billion subscriptions. In the near future, it will increase to more than six billion mobile cellular subscriptions and over than 2.4 billion people online.

HSPA had around 900 million global subscriptions in the end of December 2011. This number increased to 1 billion in 2012. It is expected to reach 2.8 billion in the end of 2015. In September 2012 it was reported that there were around 476 commercial HSPA networks in 181 countries worldwide.

The 3G Americas community (4G Americas), published a number of studies which helped to understand 3GPP stander work, started with release (1999) (Rel99) from the beginning 2003 to February 2011, after that it was published 4G Mobile Broadband Evolution which includes 3GPP Release 10 and Beyond - HSPA+, SAE/LTE and LTE-Advanced [12].

The 4G Mobile Broadband Evolution is not only important for its high data rate but also it is speed and easy accessing for a large amount of data in a short time. It can also be used in both poor and reach countries as it may not be as costly as network cabling systems. Therefore, it is more suitable and economical with social networks.

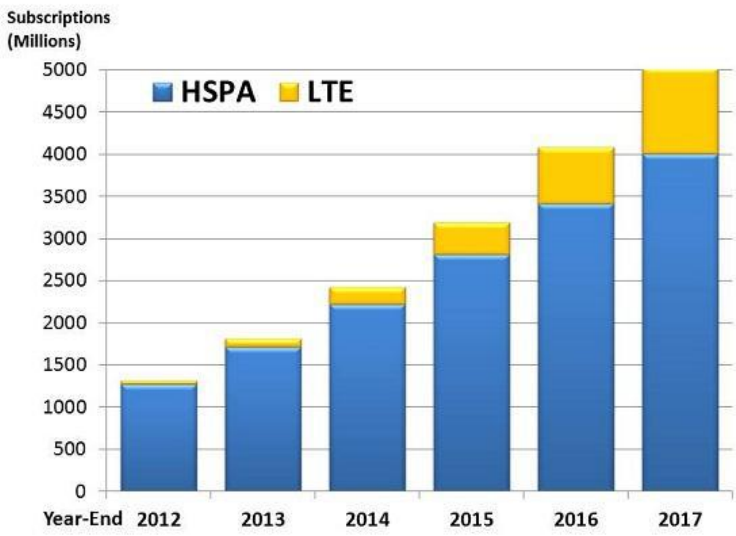

Figure 2.1. Global HSPA-LTE Subscriber Growth Forecast [13].

Understanding the history of 3GPP UMTS development can be helpful in understanding the architecture of LTE. To begin with, the UMTS is first standardized the European Telecommunications Standards Institute (ETSI) in March 2000 when standards were functionally frozen in Rel-99. It first released the Third Generation Partnership Project (3GPP) which had some improvements for high speed in the circuit switched and packet-switched modes. The next release was 5 which released in June 2002. The fifth release of 3GPP had some important enhancement: high speed data rate than previously and improving spectral efficiencies through the HSDPA. Release 5 insert IP Multimedia Subsystem (IMS), also insert IP UTRAN which was important to reduce the network cost and to recognize transport network efficiencies [12].

Release 6 was frozen in March 2005, specified features in Uplink Enhanced Dedicated Channel (E-DCH). E-DCH provided high data speed capacity and user speed on the uplink, with shorter transmission time intervals in the uplink (as low as 2 ms) [14], and the addition of processing of the Hybrid Automatic Retransmission Request (HARQ). E-DCH had another benefit which was improving the end-user for uplink intensive applications such as ending video or email with attachment transfers. Rel-6 supports multicast and broadcast services via Multimedia Broadcast/Multicast Services (MBMS). Through the (MBMS) services, it became more efficient where specific content is meant for a large number of users such as streaming audio or video broadcast [14].

Seventh release of 3GPP was frozen in December 2007. It had a lot of evolutions in its services as it moved from HSPA to the HSPA+. Rel-7 introduced some enhancements in services for instance Push-to-Talk Over Cellular (PoC), video, picture sharing, and Voice and Video over IP (VoIP) other features were: Multiple-Input Multiple-Output (MIMO), Continuous Packet Connectivity (CPC) and Higher Order Modulations (HOMs). Another release of 3GPP was 8 (LTE) which was frozen in December 2008. It introduced some enhancements; Evolved Packet System (EPS) which contents of a flat IP-based all-packet core (SAE/EPC) coupled with a new OFDMA-based RAN (E-UTRAN/LTE). With announcement of Rel-8, planning for content release 9 (Rel-9) and (Rel-10) began. Release 9 was given a lot of LTE/EPC enhancements. Support for non-3GPP accesses, local breakout and CS fallback. It was frozen in December 2009 [14].

Rel-9 started from first submitted to the ITU-R for meeting the IMT-Advanced requirements. At the same time 3GPP studied another item called LTE-Advanced, which defined release 10 of 3GPP. LTE-Advanced was meeting as a candidate technology for IMT-Advanced, also meeting the requirements of ITU-Advanced. LTE-Advanced provided tenth release of 3GPP including HSPA+ enhancements. Release 11 (Rel-11) including LTE-Advanced and Multi-RAT related enhancements and another release independent features for which specifications were frozen in September 2012. And planning for Release 12. Rel-12 is targeted for a functional freeze date by June 2014 [14].

2.2. LTE General Architecture

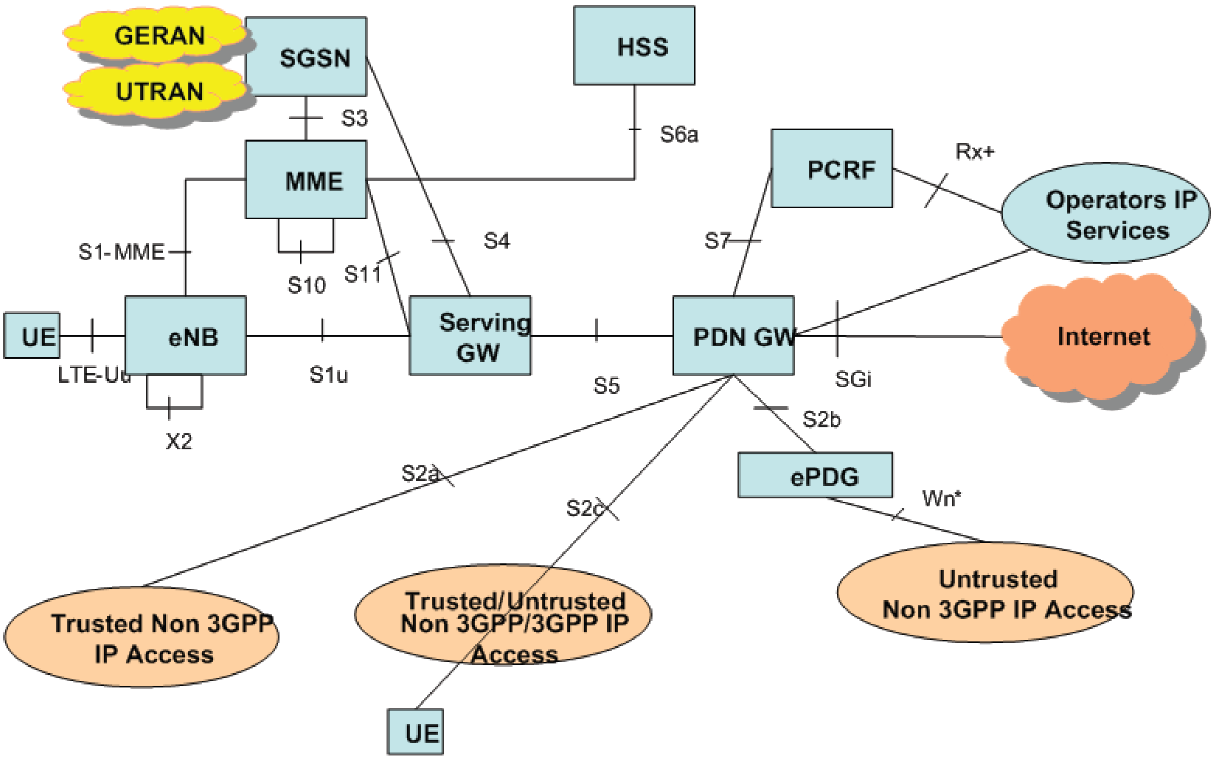

Figure 2.2. general architecture of LTE [15].

2.2.1. System architecture evolution (SAE)

SAE is core network architecture of LTE, which is the evolution of the GPRS Core Network. LTE has a new packet core, known as the Evolved Packet Core (EPC), which functions as the main core of SAE. The SAE enhances packet switched technology in order to cope with rapid growth in Internet Protocol (IP) traffic, which will help provide a more optimized network, higher data rates and lower latency [15].

The EPC, specified by 3GPP standards, will support the Terrestrial Radio Access Network (E-UTRAN), through a reduction in the number of network elements, simpler functionality, improved redundancy but most importantly allowing for connections and hand-over to other fixed line and wireless access technologies, giving the service providers the ability to deliver a seamless mobility experience [7].

2.2.2. Evolved radio access network (RAN)

The evolved RAN for LTE consists of a single node called eNodeB (eNB) that interfaces with the UE. The eNB hosts the Medium Access Control (MAC), physical (PHY), Radio Link Control (RLC), and Packet Data Control Protocol (PDC) layers, these layers include the functionality of header-compression, user-plane and encryption. It also offers Radio Resource Control (RRC) functionality corresponding to the control plane. It performs many functions including radio resource management, scheduling, admission control, enforcement of negotiated UL QoS, cell information broadcast, ciphering/deciphering of user and control plane data, and compression/decompression of DL/UL user plane packet headers [7].

2.2.3. Serving getaway (SGW)

The SGW routes and forwards user data packets, while also acting as the mobility mooring for the user plane during inter-eNB handovers and as the anchor for mobility between LTE and other 3GPP technologies (terminating S4 interface and relaying the traffic between 2G/3G systems and PDN GW). For idle state UEs, the SGW terminates the DL data path and triggers paging when DL data arrives for the UE. It manages and stores UE contexts, e.g. parameters of the IP bearer service, network internal routing information. It also performs replication of the user traffic in case of lawful interception [16].

2.2.4. Mobility management entity (MME)

In a mobile communication network, the locations of the UEs are tracked so that the incoming calls can be delivered to the user equipment's (UEs). Mobility management procedures contain location paging and update. When a user's equipment moves from one location to another location, it reports its new location to the network by the location update process. When an incoming call to the UE arrives, the network identifies the location of the UE via the paging procedure [17].

The MME is the key control-node for the LTE access-network. It is involved in the bearer activation/deactivation process and is also responsible for choosing the SGW for a UE at the initial attach and at time of intra-LTE handover involving Core Network (CN) node relocation. It is responsible for authenticating the user (by interacting with the HSS). The Non- Access Stratum (NAS) signaling terminates at the MME and it is also responsible for generation and allocation of temporary identities to UEs. It checks the authorization of the UE to camp on the service provider's Public Land Mobile Network (PLMN) and enforces UE roaming restrictions. The MME is the termination point in the network for ciphering/integrity protection for NAS signaling and handles the security key management. Lawful interception of signaling is also supported by the MME. The MME also provides the control plane function for mobility between LTE and 2G/3G access networks with the S3 interface terminating at the MME from the SGSN. The MME also terminates the S6a interface towards the home HSS for roaming UEs as shown in figure (2.3). The mobility in LTE is optimized for zero to 15 Km/h, also support with high performance between 15 to 120 Km/h, and supported up to 350km/h or even up to 500Km/h [18].

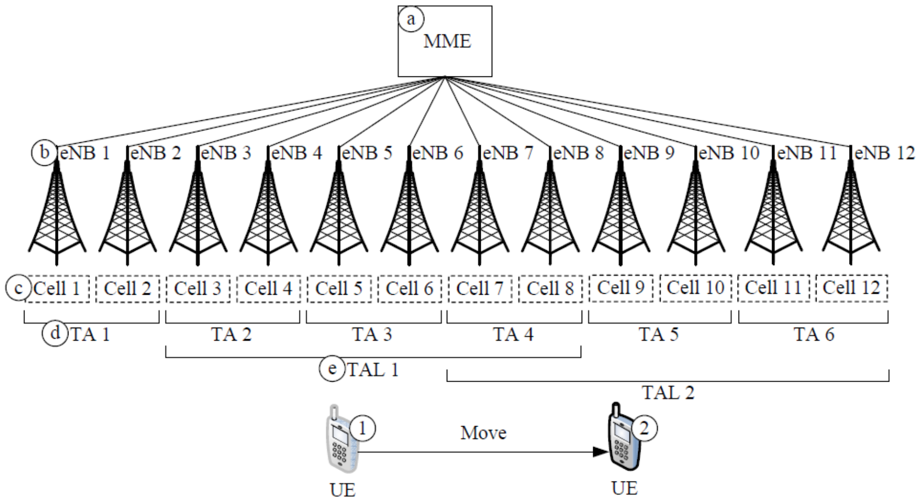

Figure 2.3. MME in LTE [17].

3.2.5. Packet data network getaway (PDN GW)

The PDN GW provides connectivity to the UE to external packet data networks by being the point of exit and entry of traffic for the UE. A UE may have simultaneous connectivity with more than one PDN GW for accessing multiple PDNs. The PDN GW performs policy enforcement, packet filtering for each user, charging support, lawful interception and packet screening. Another key role of the PDN GW is to act as the anchor for mobility between 3GPP and non-3GPP technologies such as WiMAX and 3GPP2 (CDMA 1X and EvDO) [19].

2.3. LTE Technologies

To direct higher data rates and faster connection times, LTE or LTE-Advanced contains several new technologies to reduce the radio interference and noise. Increasing efficiency and performance is another demand of 3GPP (LTE and LTE-Advanced). The significant technologies in the LTE will be discussed and also the way these technologies was chosen by 3GPP will be illustrated.

2.3.1. Orthogonal frequency division multiplexing (OFDM)

OFDM is a technology allowing a base station to split a chunk of radio spectrum into sub-channels. The signal strength of the sub-channels and the number of channels assigned to different devices can be varied as needed, as shown in figure (2.4). OFDM allows high data rates, even far from a base station, and it copes well with the type of radio interference that is common in urban areas, where signals reflect off walls to produce confusing echoes [20].

Increasing the efficiency of power amplifier, reducing the power ratio, saving battery live, treatment of power consumption and latency is one of the main features of LTE. It improves performance by several techniques. One of them is (OFDM). LTE uses different access mode for downlink and uplink, using (OFDMA) in the downlink and Single Carrier – Frequency Division Multiple Access (SC-FDMA) technology in the uplink, as shown in figure (2.4) [18].

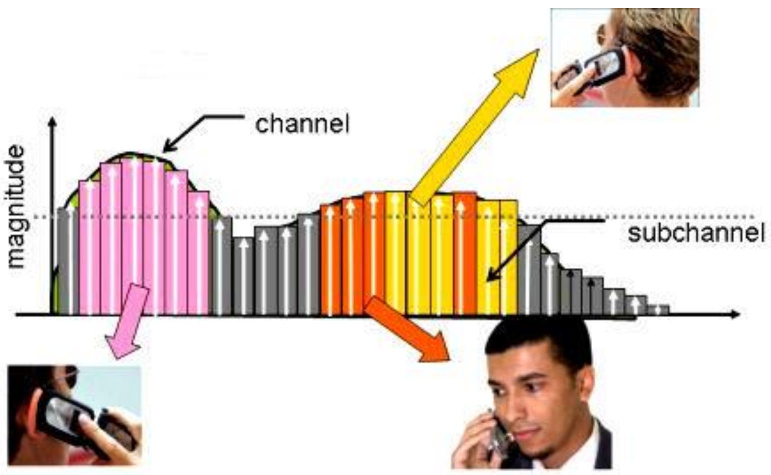

Figure 2.4. OFDM Sub-carriers [20].

In OFDMA, each of the sub-carriers is modulated by different data symbols and these symbols last for relatively long duration. In the SC-FDMA signal carrier is multiple carrier but all subcarrier in the uplink are modulated with the same data as shown in Figure 2.5. The first layer which is a green layer in SC-FDMA modulated by the same data [22].

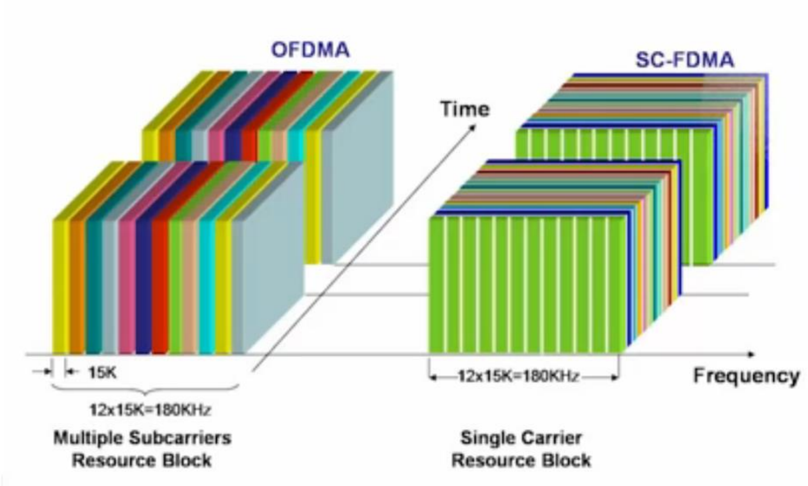

Figure 2.5. OFDMA and SC-FDMA [21].

2.3.2. Multiple antenna techniques (MIMO)

MIMO is refers to the use of more than one transmit antenna in the station and more than one receiver antenna in user equipment’s. MIMO is used in new wireless communication like WIMAX and LTE with different Rx and Tx [24], as shown in Figure 2.6. This is to reduce the noise and to increase the capacity of spectrum. The effect of MIMO appear with applies separate modulation, coding, and spreading for more than two transport blocks transmitted with two parallel streams. The achievable peak data rates in the downlink will become double. The actual radio propagation conditions that the UE experiences determine whether one or two streams can be transmitted [24].

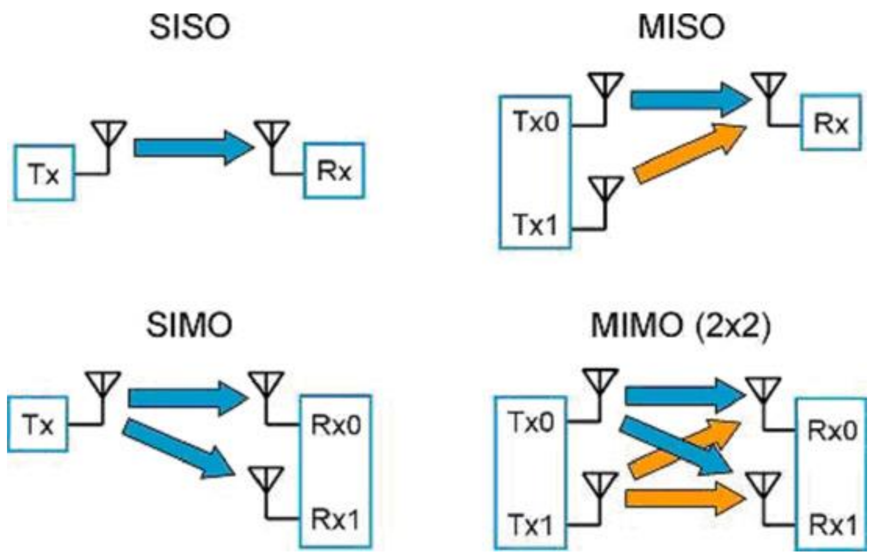

Figure 2.6. MIMO Antenna [24].

Simulations show increased performance of the system with increasing number of antennas. The transmitter sends streams with different transmit antennas in the same frequency band and the same time slot. On the receiver side the receiver separates data out of the original substreams from the mixed signals using the non-correlation of signals on multiple receive antennas results of the multipath in the transmission. This leads to significant increase in data rates and throughput. Last release of 3GPP, which is called LTE-advanced, uses MIMO 8x8 in the DL and 4x4 in the UL. MIMO shall be used when S/N (Signal to Noise ratio) is high, i.e. high quality radio channel. For situations with low S/N it is better to use other types of multi-antenna techniques to improve S/N, e.g. TX-diversity as shown in figure 2.7, figure 2.8 and figure 2.9 [24].

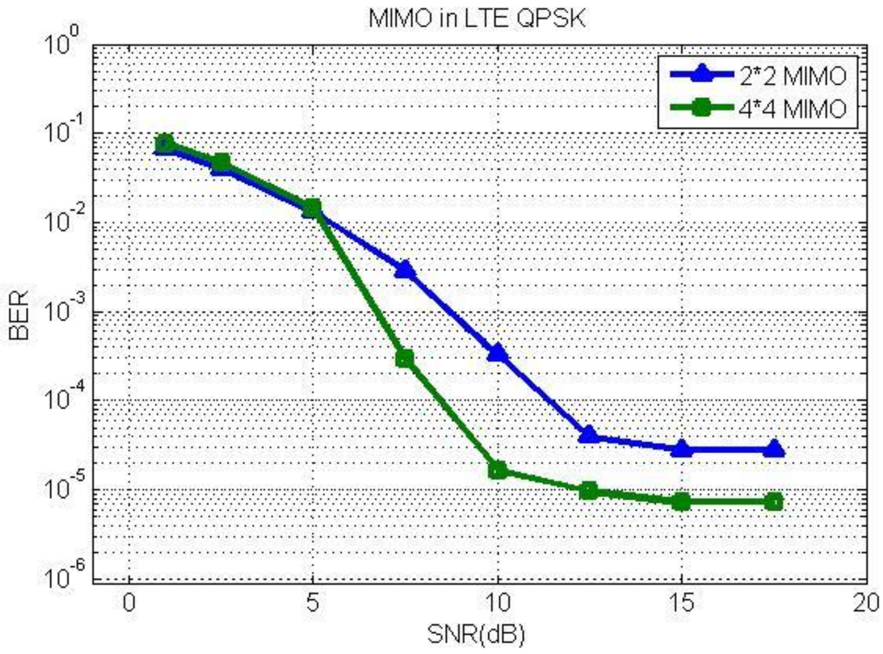

Figure 2.7. Performance of MIMO in LTE using QPSK modulation.

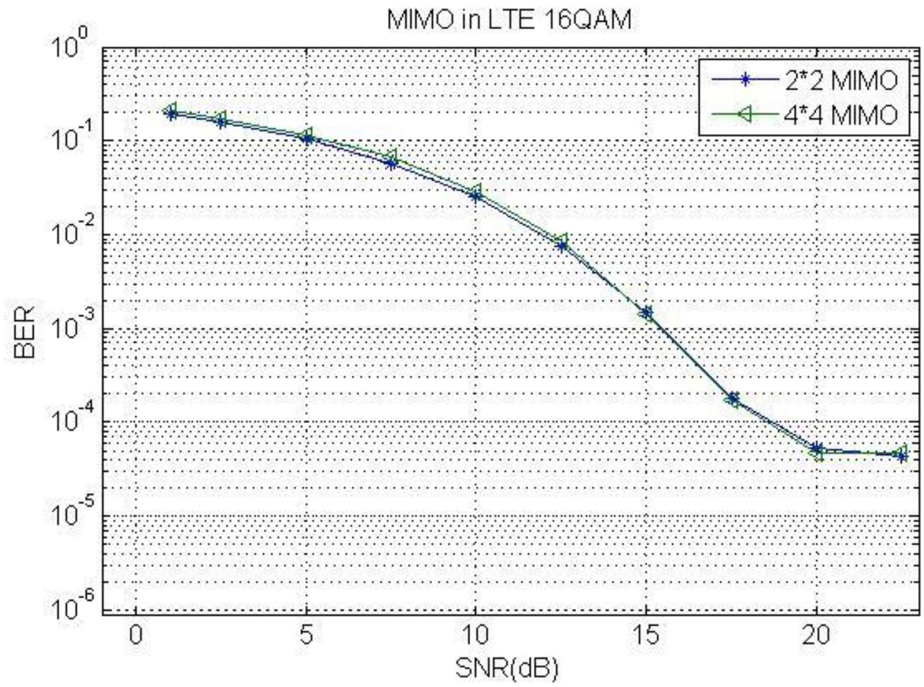

Figure 2.8. Performance of MIMO in LTE using 16QAM modulation.

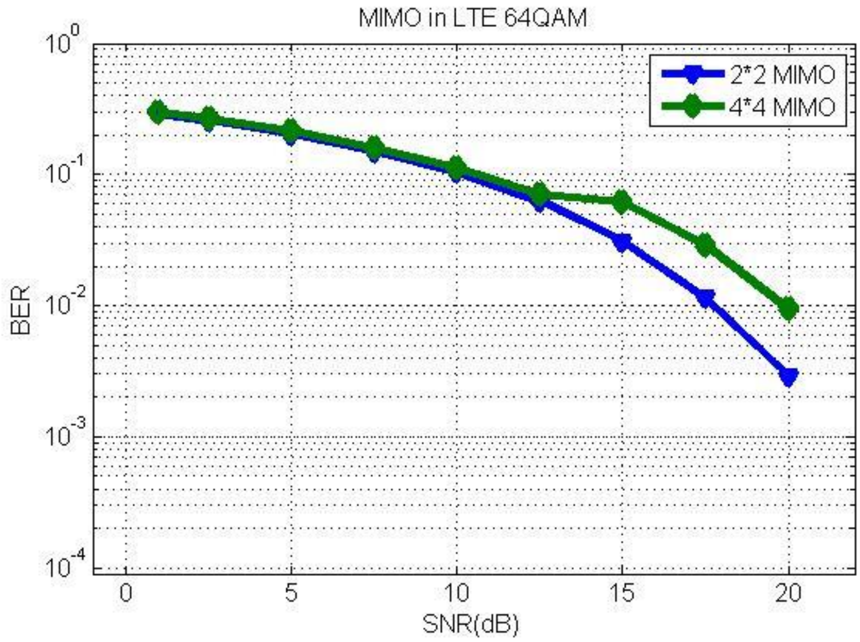

Figure 2.9. Performance of MIMO in LTE using 64QAM modulation.

2.3.3. High rate modulation

Adaptive Modulation with different coding schemes (AMC) is another technique used in LTE to improve the peak data rate, system capacity, and coverage reliability. LTE uses three different modulations with different coding rate (QPSK, 16QAM, 64QAM). High order modulation transmits more bits to symbol with better spectral efficiencies and improving performance of the system. High order modulation also needs better SNR. For instance, when a user is near to a cell site, LTE uses high order modulation due to low noise and interference in shorter ranges LTE uses 64QAM, in the medium ranges uses 16QAM, and longer ranges LTE uses QPSK as shown in Figure 2.10 [25].

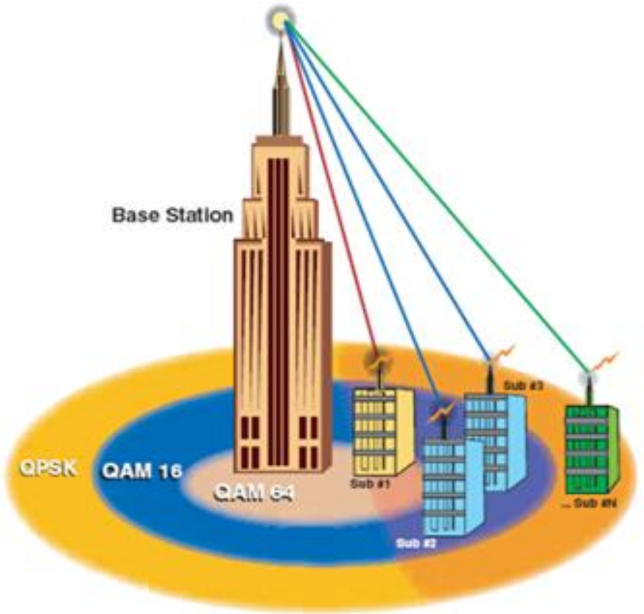

Figure 2.10. Adaptive modulation in LTE [26].

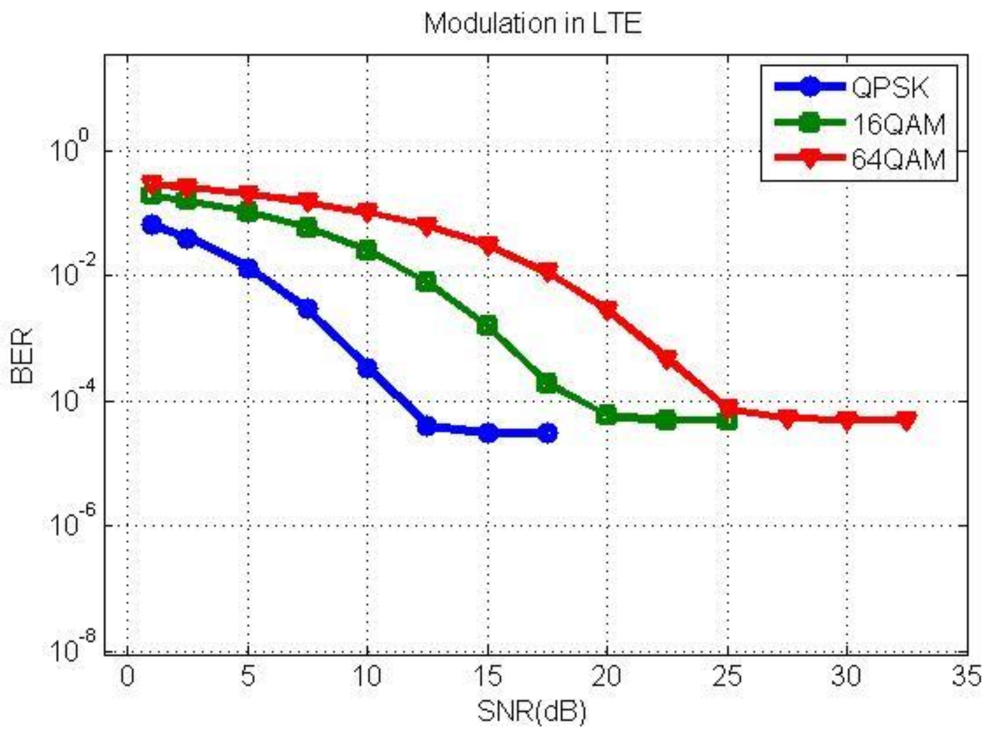

Figure 2.11. Error rate performance for different modulation in LTE.

Figure 2.11 represents the comparison of the packet error rate performance for each of the modulation with channel coding rate (1/3). It can be seen that the PER performance of the 64 QAM offers a significant improvement when compared with 16QAM. Also it can be interpreted from figure 2.11 that 16QAM shows significant improvement over QPSK.

2.3.4. Channel coding with transport block segmentation

LTE uses two different coding techniques for transport channels. The first is called turbo coding and the second is called bit convolutional code. Turbo coding is used for large data packets, and convolutional code for control information. Turbo coding with code rate (1/3) is used in DL-SCH, UP-SCH, PCH, and MCH [27]. LTE uses bit convolutional coding within control information in BCH, DCI, and UCI. Block code with different code rates are used in CFI and UCI [25] as shown in Figure 2.12.

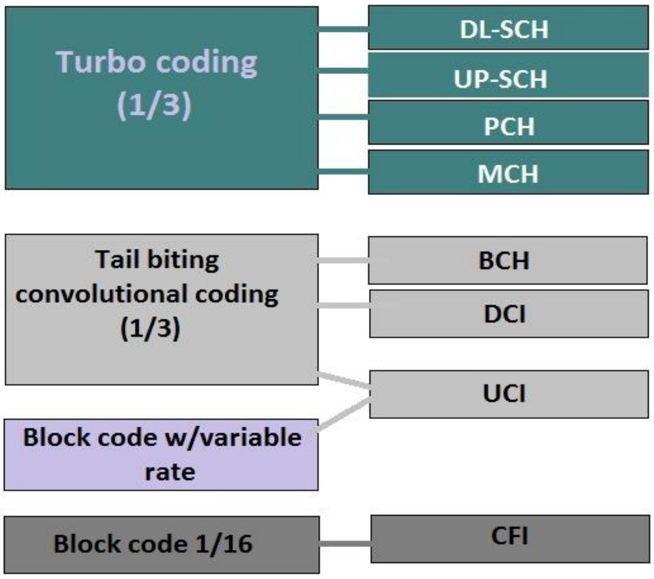

Figure 2.12. Channel coding in LTE.

Processing of the transport channel for lager data packets (DL-SCH, UP-SCH, PCH and MCH) is shown in Figure 2.14. Systems starts with adding CRC attachments to the data bits before proceeding to the next step. It segments transport block with CRC attachment into a number of codeblocks. The next step is turbo coding, rate matching and finally code block concatenation. The processing of the transport blocks for small data packets or control information is shown in Figure 2.13. System is not very complex, it just has three steps which starts with CRC attachment or adding cyclic redundancy check (CRC), without segmentation. Process on the next step is convolution coding and finally rate matching [25]. In the rest of this chapter we will demonstrate all steps of transport blocks in LTE.

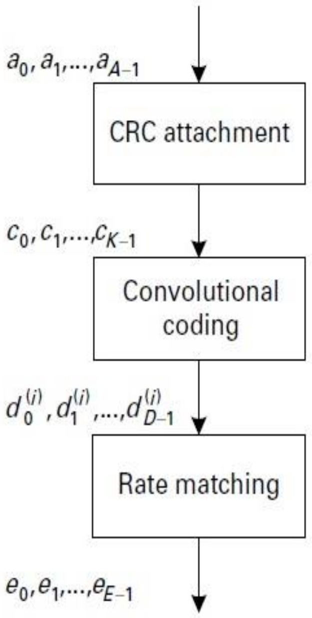

Figure 2.13. Transport block of DCI and BCH [25].

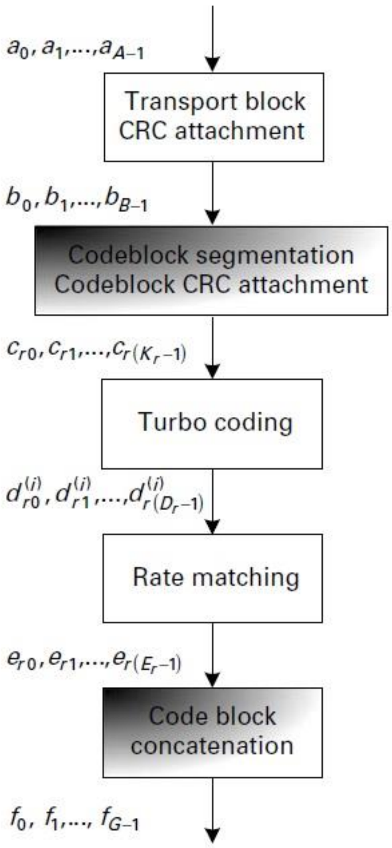

Figure 2.14. Transport block of DL-SCH and UP-SCH [25].

2.3.4.1. Cyclic redundancy check (CRC) attachment

CRC is a common technique for error detection; n-bits of CRC applied before data blocks. LTE uses two kinds of CRC in the same number of bits with different generator polynomials. One reason for the popularity of CRC is its simplicity in implementation. The CRC calculation can be easily implemented by a linear feedback shift register (LFSR). The LFSR can be used as a circuit for polynomial division. The maximum code block size in LTE is limited to Z = 6144. A transport block is a data block delivered by the MAC layer to the physical layer for transmission in a single sub frame of one millisecond [28].

The early stopping strategies can be used to further reduce the operational complexity and also the power consumption of turbo decoders. While many detection methods for early stopping have been studied [10], attachment and checking of CRC bits turns out to be a simpler and more reliable approach. A single acknowledgment (ACK) or negative acknowledgment (NACK) per transport block is provided for hybridARQ retransmissions. Therefore, in transmissions with multiple codeblocks, the receiver will NACK the transmission as long as one of the codeblocks is in error after the maximum number of iterations. If we can introduce a CRC per codeblock, the decoder can stop decoding after one codeblock is in error, thus saving power that could have been wasted in decoding the rest of the codeblocks. From a powersaving perspective, a small 8-bit codeblock CRC, which gives a miss detection rate of 0.4%, will be sufficient. Note that even if a miss detection occurs, the only negative impact is the receiver will proceed to decode the rest of the codeblocks and waste the decoding power with 0.4% of probability. Here, we assume that transport block CRC of length 24 bits is available to catch miss detections from the codeblock CRC to ensure transport block integrity [25].

In multi-codeword MIMO transmission in LTE, two MIMO codewords are transmitted, each of which can carry multiple codeblocks. For each MIMO codeword or transport block, a CRC is computed based on all information bits in the transport block, that is based on all the codeblocks in the MIMO codeword. In the case when there is no codeblock CRC, the receiver has to wait until all the codeblocks in the first codeword are decoded before it can cancel this codeword and proceed to decoding of codeblocks in the second codeword, as illustrated in Figure 2.15. This is because transport block CRC check cannot be performed until all the codeblocks within the transport block are received. A CRC check is necessary before canceling the first codeword to make sure that only the successfully received codeword is cancelled. A cancellation of an unsuccessful codeword may degrade successive interference cancellation (SIC) receiver performance due to residual interference. With a codeblock CRC, on the other hand, SIC operation can be performed on a codeblock basis as illustrated in Figure 2.16. This helps to reduce the codewords decoding time and buffering complexity required for SIC.

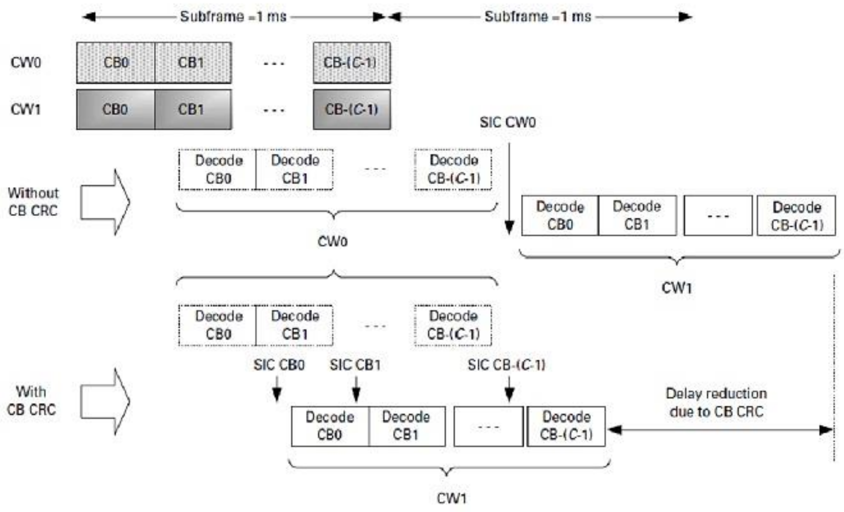

Figure 2.15. Codeblock CRC reduces delays and MIMO SIC complexity [25].

CRC attachment scheme in LTE is shows in Figure 2.16. A transport block CRC is added before a transport block and computed all information bits of that TB is attached to the TB. The transport block segmented in multiple codeblocks (CB), also added CRC before each of this codeblocks and computed each segment independently. The probability of misdetection of an error of the transport block is about:

$$P_m = 2^{-24} = 6 \times 10^{-8}$$

This kind of attachment is very simple implementation of CRC for checking for early stopping of decoding, the maximum CB size is equal to 6144 bits in LTE. More than one codeblock is used for completely large TB sizes. In the worst-case, overhead, due to 24 additional bits, is less than 1%. This small increase in above was considered appropriate given the low misdetection probability.

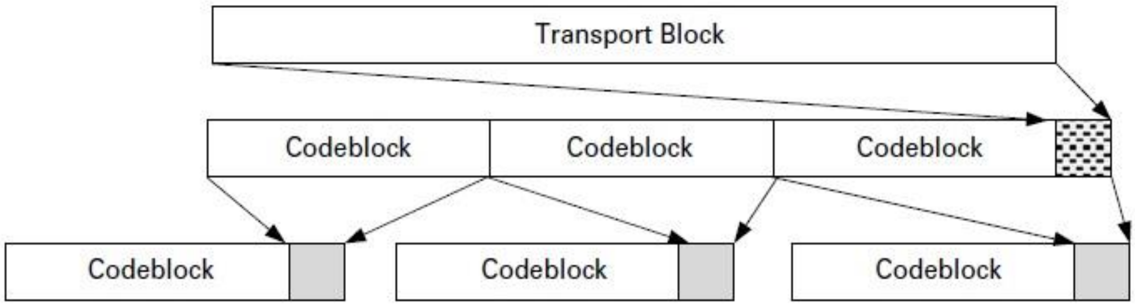

Figure 2.16. CRC Attachment Scheme in LTE [27].

As we discussed before LTE uses 24-CRC or 24 bits for CRC checking in the transport blocks and the codeblocks. The advantage of this number of bits is low misdetection probability for early stopping as well as MIMO SIC [25].

It selects similar number of CRC for codeblocks and the transport blocks with different generator polynomial. If identical generator polynomial are used in both codeblocks and transport blocks, if an error sequence passed in the transports block also may pass in the codeblocks. If different generator polynomial is used, the error sequence path the transport block cannot pass to the codeblock CRC checking. Figure 2.17 shows the plot of miss rates in the case of similar and different transport block and codeblock [25].

Two different generator polynomials for L=24 there are called g_{CRC24A} (D) and g_{CRC24B} (D) for TB-CRC and CB-CRC respectively are given as;

gCRC24A (D) $=D^{24} + D^{23} + D^{18} + D^{17} + D^{14} + D^{11} + D^{10} + D^{7} + D^{6} + D^{5} + D^{4} + D^{3} +D+1$

gCRC24B (D) $= D^{24} + D^{23} + D^{6} + D^{5} + D + 1$.

gCRC16 (D) $= D^{16} + D^{12} + D^{5} + 1$.

gCRC24A (D) is used for transport block CRC calculation on DL-SCH, UP-SCH, PCH and MCH. The gCRC24B (D) polynomial is used for codeblock CRC calculation and the gCRC16 (D) polynomial is used for CRC calculation for DCI and BCH transport channels [28].

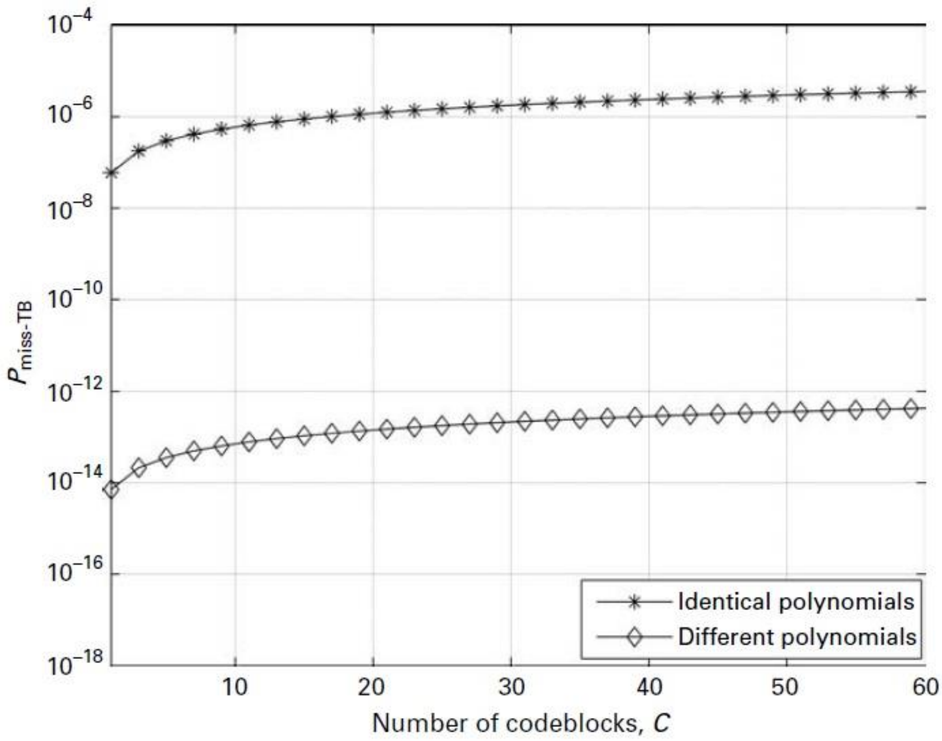

Figure 2.17. Transport error miss rates in the case similar and different transport block and codeblock [25].

2.3.4.2. Transport block segmentation in LTE

It was motioned before that, the maximum codeblock size is limited to Z=6144 and the CRC-checking is limited to 24 bits. 6120 bits of the maximum code block is data bits and 24 for CRC-checking bits. It is assumed that, the transport block size is B. When the transport block size more than 6144 bits, the segmentation of the transport block is performed.

It should be noted that when the size of the transport block (B) do not reach to the maximum code block size in LTE therefore did not add CRC-Checking to the codeblock. But if the size of transport block is larger than the maximum codeblocks size then CRC-checking can be added to the whole of the transport block, after adding CRC-checking for each of the codeblocks [29].

For the purposes of reducing complexity, a certain fixed number of turbo interleaver sizes is supported as given in Table 2.2 below. In particular, the granularity between two adjacent interleaver sizes is 8-bits for small codeblocks and goes up to 64 bits for the largest codeblock size. When the transport block size is not matched to the turbo interleaver size, filler bits are added. The reason for a coarser granularity of interleaver sizes for larger code blocks is that a larger number of filler bits is still a small fraction of the codeblock size when the codeblock size is large. Table 2.2 is uses to divide transport block into codeblocks. Then using adjacent interleaver size a larger K+ and next smaller size K-, the number of codeblocks of size K+ and size K− as C+ and C− respectively.

$\Delta K = K_{+} - K$

For instance in the table (2.1) message size equal to 15805 bits, by using blow equations we illustrate transport block segmentation in LTE.

Table 2.1. Transport block segmentation in LTE.

Message length	15805 bits
C	3
B`	15877
K+	5312
K-	5248
ΔK	64
C-	1
C+	3
Filler Bits	59

Table 2.2. Turbo code internal interleaver parameters in LTE [27].

i	K	f_1	f_2	i	K	f_1	f_2	i	K	f_1	f_2	i	K	f_1	f_2
1	40	3	10	48	416	25	52	95	1120	67	140	142	3200	111	240
2	48	7	12	49	424	51	106	96	1152	35	72	143	3264	443	204
3	56	19	42	50	432	47	72	97	1184	19	74	144	3328	51	104
4	64	7	16	51	440	91	110	98	1216	39	76	145	3392	51	212
5	72	7	18	52	448	29	168	99	1248	19	78	146	3456	451	192
6	80	11	20	53	456	29	114	100	1280	199	240	147	3520	257	220
7	88	5	22	54	464	247	58	101	1312	21	82	148	3584	57	336
8	96	11	24	55	472	29	118	102	1344	211	252	149	3648	313	228
9	104	7	26	56	480	89	180	103	1376	21	86	150	3712	271	232
10	112	41	84	57	488	91	122	104	1408	43	88	151	3776	179	236
11	120	103	90	58	496	157	62	105	1440	149	60	152	3840	331	120
12	128	15	32	59	504	55	84	106	1472	45	92	153	3904	363	244
13	136	9	34	60	512	31	64	107	1504	49	846	154	3968	375	248
14	144	17	108	61	528	17	66	108	1536	71	48	155	4032	127	168
15	152	9	38	62	544	35	68	109	1568	13	28	156	4096	31	64
16	160	21	120	63	560	227	420	110	1600	17	80	157	4160	33	130
17	168	101	84	64	576	65	96	111	1632	25	102	158	4224	43	264
18	176	21	44	65	592	19	74	112	1664	183	104	159	4288	33	134
19	184	57	46	66	608	37	76	113	1696	55	954	160	4352	477	408
20	192	23	48	67	624	41	234	114	1728	127	96	161	4416	35	138
21	200	13	50	68	640	29	80	115	1760	27	110	162	4480	233	280
22	208	27	52	69	656	185	82	116	1792	29	112	163	4544	357	142
23	216	11	36	70	672	43	252	117	1824	29	114	164	4608	337	480
24	224	27	56	71	688	21	86	118	1856	57	116	165	4672	37	146
25	232	85	58	72	704	155	44	119	1888	45	354	166	4736	71	444
26	240	29	60	73	720	79	120	120	1920	31	120	167	4800	71	120
27	248	33	62	74	736	139	92	121	1952	59	610	168	4864	37	152
28	256	15	32	75	752	23	94	122	1984	185	124	169	4928	39	462
29	264	17	198	76	768	217	48	123	2016	113	420	170	4992	137	234
30	272	33	68	77	784	25	98	124	2048	31	64	171	5056	39	158
31	280	103	210	78	800	17	80	125	2112	17	66	172	5120	39	80
32	288	19	36	79	816	127	102	126	2176	171	136	173	5184	31	96
33	296	19	74	80	832	25	52	127	2240	209	420	174	5248	113	902
34	304	37	76	81	848	239	106	128	2304	253	216	175	5312	41	166
35	312	19	78	82	864	17	48	129	2368	367	444	176	5376	251	336
36	320	21	120	83	880	137	110	130	2432	265	456	177	5440	43	170
37	328	21	82	84	896	215	112	131	2496	181	468	178	5504	21	86
38	336	115	84	85	912	29	114	132	2560	39	80	179	5568	43	174
39	344	193	86	86	928	15	58	133	2624	27	164	180	5632	45	176
40	352	21	44	87	944	147	118	134	2688	127	504	181	5696	45	178
41	360	133	90	88	960	29	60	135	2752	143	172	182	5760	161	120
42	368	81	46	89	976	59	122	136	2816	43	88	183	5824	89	182
43	376	45	94	90	992	65	124	137	2880	29	300	184	5888	323	184
44	384	23	48	91	1008	55	84	138	2944	45	92	185	5952	47	186
45	392	243	98	92	1024	31	64	139	3008	157	188	186	6016	23	94
46	400	151	40	93	1056	17	66	140	3072	47	96	187	6080	47	190
47	408	155	102	94	1088	171	204	141	3136	13	28	188	6144	263	480

The number of codeblocks is given as;

$$C = \lceil B/(Z - L) \rceil = \lceil 15805/(6144 - 24) \rceil = 3$$

It can be noted that the B' is a total bits of the new transport blocks after CRC-checking is added for both codeblock and transport block. As:

$$B' = B + C \times L = 15805 + 3 \times 24 = 15877$$

Also K_+ and K_- equal to 5312 and 5248 respectively. It's the result of the C×K ≥ B or from nearest number in table (2.2), when B' is divided by C.

$$C_- = \left\lceil \frac{C.K_+ - B'}{\Delta K} \right\rceil = \left\lceil \frac{3 \times 5312 - 15877}{64} \right\rceil \simeq 0$$

$$C_+ = C - C_- = 3 - 0 = 3$$

The number of filler bits can be represented as shown below.

$$F = C_+ \times K_+ + C_- \times K_- - B'$$

$$= 3 \times 5312 + 1 \times 5248 - 15877 = 59$$

2.3.4.3. Tail biting convolutional coding and turbo coding

LTE uses both tail biting convolutional coding and turbo coding for error correction [29]. Tail biting convolutional coding with constrain length and coding rate 1/3, is shown in Figure 2.18.

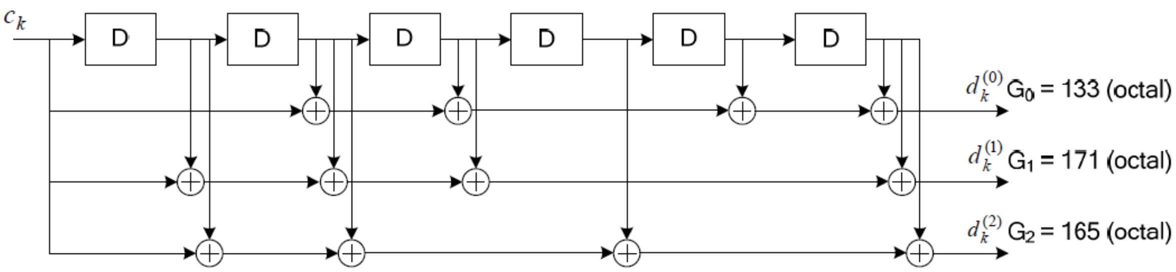

Figure 2.18. Tail biting convolutional encoder [25]

LTE turbo encoding is based on two eight-state constituent encoders and one turbo code internal interleaver. The sequence of original input bits z_k is encoded by the first encoder; the input to the second encoder is an interleaved version of the original sequence z'_k. Therefore, the resulting coded sequence is the result of combining the information sequence of bit (systematic bits, S) with the two sequences of parity bits (P1 and P2).

Hence, the overall code rate is 1/3 even though the rate of each encoder is 1/2, because their outputs are combined with input data bits (see Figure 2.19) [29].

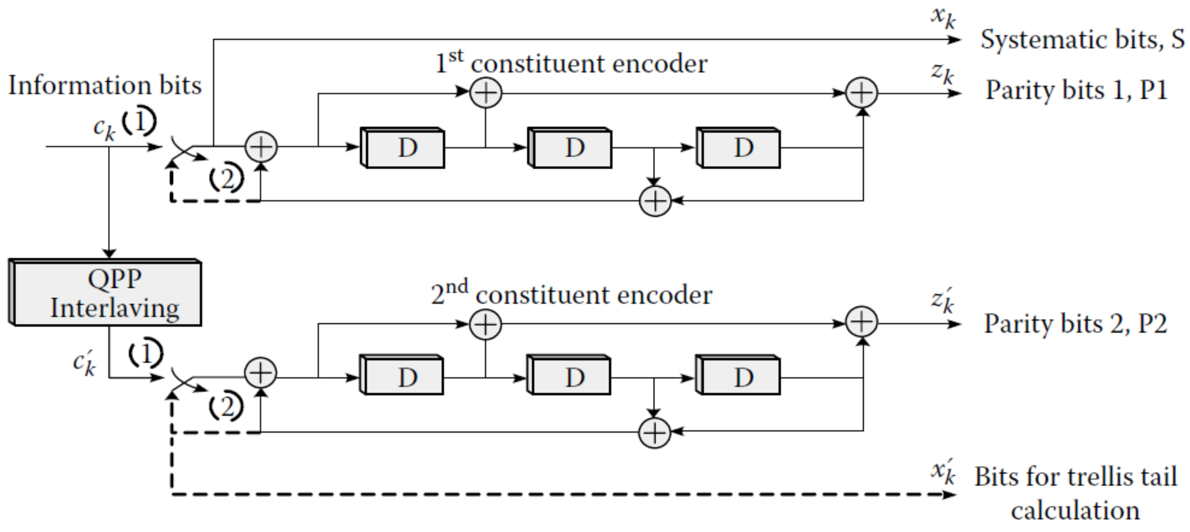

Figure 2.19. Structure of Rate 1/3 Turbo Encoder [25]

2.3.4.4. Rate matching

The purpose of the rate matching process is to adjust the coded block size to the allocated physical resources. The first step in this process consists of interleaving the systematic and parity bits in the coded block provided as an output of the turbo encoder process. This interleaving process is aimed to mitigate the effects of error bursts likely to appear in radio mobile channels due to the time variant fading effects. Consequently, a dual de-interleaving operation is needed at the receiver side to recover original coded blocks. The result of the interleaving process is stored into a circular buffer where a puncturing process is applied to decrease the effective coding rate according to the MCS corresponding to the considered CQI level.

The complete rate matching process is depicted in Figure 2.20. This process allows different coding rates to be achieved (i.e., a higher coding rate to improve robustness against the effects of the channel at the expense of a lower spectral efficiency, or vice versa) [29].

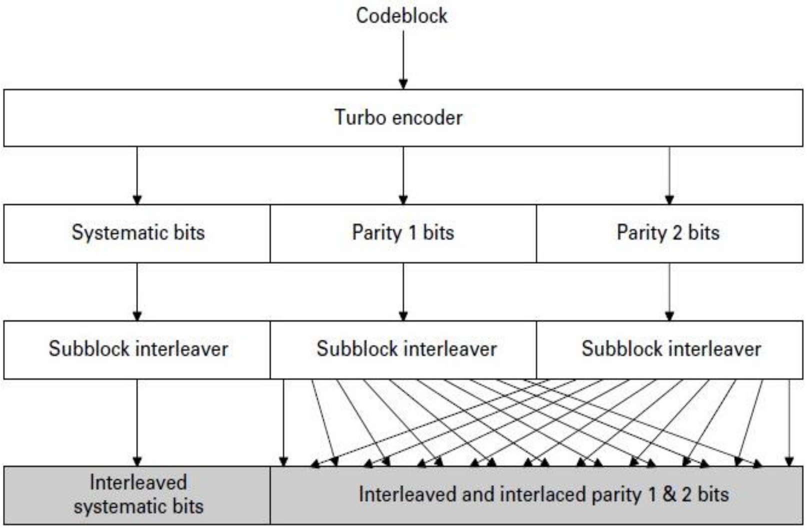

Figure 2.20. Rate Matching in LTE [25].

In this chapter the channel coding technique in LTE which includes parallel turbo coding and convolutional coding is discussed briefly and also there will be a discussion on the Hybrid-ARQ (HARQ). In the next chapter there will be a general discussion on convolutional coding and turbo coding and also the kind of turbo coding and why LTE selects the parallel turbo coding instead of any kind of forward error correction (FEC). Furthermore a detailed discussion will be made in the next chapter about Circular-buffer rate matching, codeblock concatenation of turbo code.

2.3.5. Hybrid automatic repeat request (HARQ) in LTE

HARQ is a combination of automatic repeat request and error correction in the physical layer. Traditional ARQ is only used for error detection and request to the transmitter to repeat data, by which makes use of acknowledgments and timeouts to achieve reliable data transmission [28]. The main difference between the ARQ and Hybrid-ARQ is that, HARQ tries to correct the error by several coding techniques, for example turbo coding and convolutional coding.

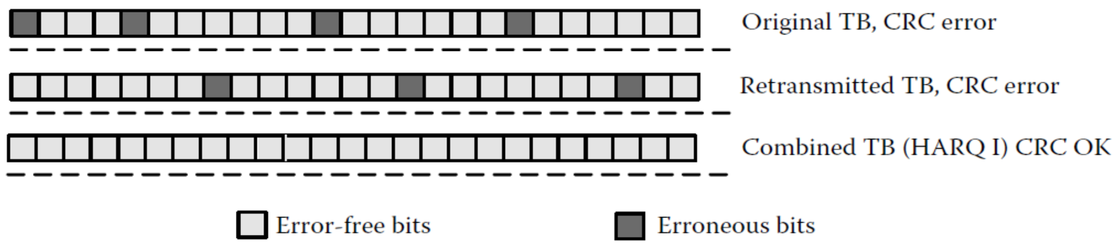

Figure 2.21. Chase combining HARQ [25].

There are two types of HARQ. First one is known as chase combining or type one HARQ, and the second type of HARQ is known as incremental redundancy or type two of HARQ. The working strategies of type one HARQ or chase combing process is once a transport bock (TB) is resaved all transport block is checked by calculating its CRC and comparing it with the CRC sequence. A decoding error raises a retransmission request that is answered by the transmitter by sending again the same transport block. An incorrectly received transport block is stored at the receiver in order to be combined with the retransmitted transport block to increase the probability of successful decoding [29] as shown in Figure 2.21.

In the second type of the HARQ, a code word is compounded by most of the systematic bits and a few parity bits for error correction. If an error occurs in the decoding process, a retransmission request is sent to the transmitter. Additional parity bits are transmitted and combined upon arrival with the previously received version of the coded transport block, resulting in a lower coding rate. The process is repeated until a successful decoding is performed or until the transport block is discarded when the retransmission limit is reached, as it is shown in Figure 2.22.

Both chase combining and incremental redundancy retransmission are used in physical PDSCH and PUSCH in LTE and it is based on multiple parallel stop-and-wait processes. A negative acknowledge (NACK) is issued if an error is detected and unsuccessful decoding implying a retransmission request for the particular transport block. An acknowledgement (ACK) is reported to the transmitter when a transport block is received without errors and successfully decoded. Once an ACK/NACK message is received at the transmitter, its timing is used to associate it with its correspondent HARQ process.

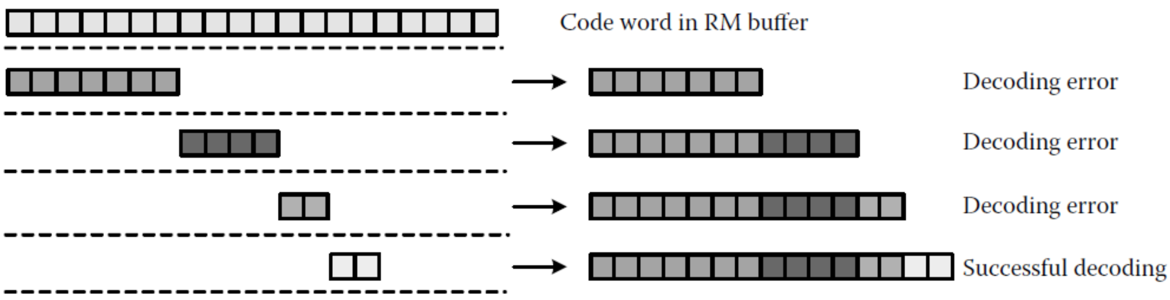

Figure 2.22. Incremental redundancy HARQ [25].

On the other hand PDSCH and PUSCH HARQ operations are based on different protocols. The PDSCH is an asynchronous protocol, retransmissions may occur at any time. Each process is identified with an HARQ process number. But the PUSCH is a synchronous protocol, in which the retransmission instant is fixed by the initial transmission time and therefore the process number can be implicitly derived from it [28].

2.3.6. Quality of service, policy and charging

Applications such as web browsing, VoIP, video streaming and video telephony have special QoS needs. Thus, an important feature of any all-packet network is the providing a QoS mechanism to enable differentiation of packet flows based on QoS requirements. In EPS, QoS flows called EPS bearers are established between the UE and the P-GW as shown in Figure 2.23. A radio bearer transports the packets of an EPS bearer between a UE and an eNB. Each IP flow (e.g. VoIP) is associated with a different EPS bearer and the network can prioritize traffic accordingly. When receiving an IP packet from the Internet, P-GW performs packet classification based on certain predefined parameters and sends it to an appropriate EPS bearer. Based on the EPS bearer, eNB maps packets to the appropriate radio QoS bearer. There is one-to-one mapping between an EPS bearer and a radio bearer [30].

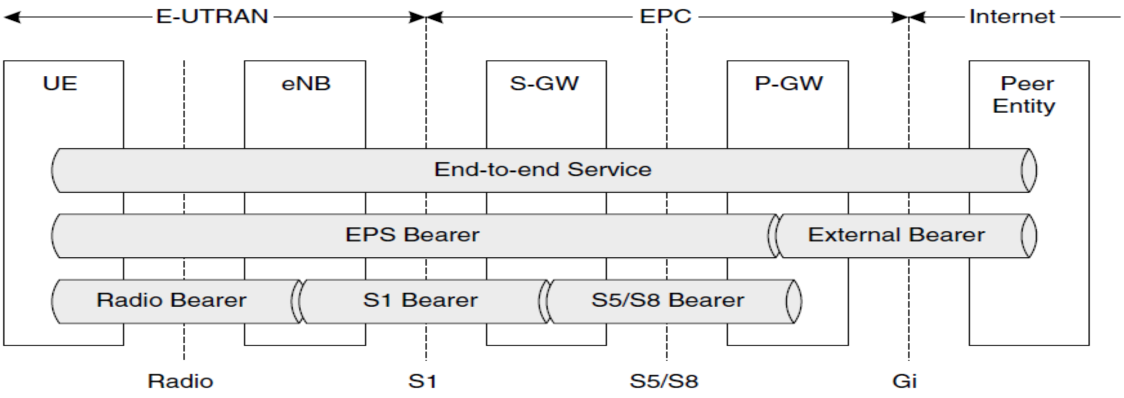

Figure 2.23. EPS bearer service architecture [25].

2.2.7. Multimedia broadcast/multicast service

MBMS is another feature which was introduced in LTE Release 9. MBMS is a bearer service for multicast/ broadcast transmission of data used to transmit the same information or a data to all UEs in an area over common bearer. In Release 9 of LTE MBMS do not require complex control. Single Frequency network (MBSFN) transmission is one of the prime features that were supported in MBMS. With MBSFN transmission eNBs in the MBSFN area transmit the same signal simultaneously using the same time frequency resource. The user equipment receives the combined signals as a signal or storage multi signal in one signal, by applying the MBSFN transmission, improving signal quality, improving coverage and strong signal without much complexity in the UE. By applying the MBSFN transmission the 3GPP provide 95% coverage with packet error rate about 1% and the spatial efficiency limited to 3 bit/s/Hz. [18]. Figure 2.24 shows the logical architecture for MBMS in LTE. Gateway (GW) distributed data received from the Broadcast Multicast Service Center (BMSC). BMSC is a network node which stores content to be transmitted by MBMS and control MBMS transmission to relevant eNBs by IP multicast. The multi Cell Multicast Coordination Entity (MCE) specifies the radio resources to be used by the eNBs comprising the MBSFN and ensures that the content is synchronized. To support MBMS, logical channels, namely Multicast Control Channel (MCCH), Multicast Traffic Channel (MTCH), and a transport channel, namely Multicast Channel (MCH) [31] are used.

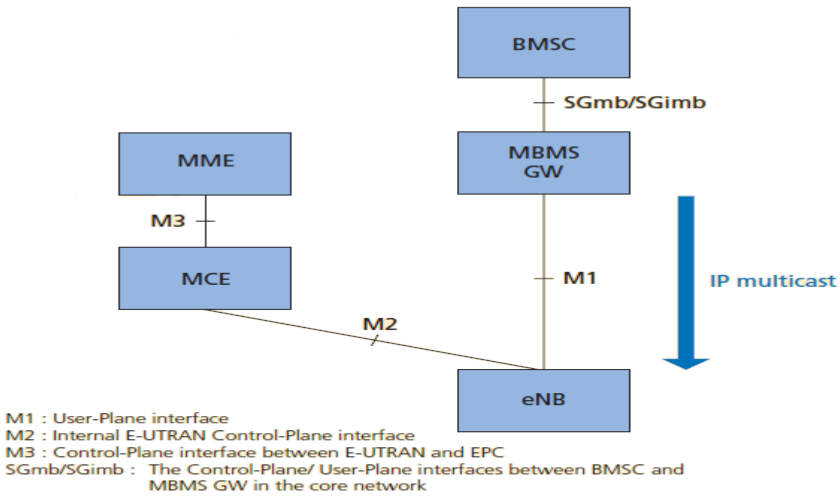

Figure 2.24. MBMS in LTE [31].

2.3.8. Other enhancements

2.3.8.1. IMS emergency calls

In Release 8 of 3GPP IMS Emergency calls were not supported, and it's worked with other radio access technologies supported the Circuit Switched domain (CS). The CS domain is a network domain that provides services based on circuit switching for instance, GERAN by using the CS callback. LTE Release 9 enhanced the emergency calls with IP Multimedia System (IMS) Emergency Calls. This support for emergency calls is wished for across the US and network [32].

First thing to consider is that any UE can make an emergency call provided that the network supports. This means there is no subscription for that UE in Home Subscriber Server (HSS). When UE wants to perform emergency attach, it will send an attach request with attach type as Emergency Attach. Upon receiving this attach MME may or may not contact HSS for Authentication. If MME supports, then call may proceed even if Authentication fails if at all HSS was contacted. If authentication fails then MME may not contact HSS for Update location request. In a nut shell, when Emergency Attach is received Auth/Security procedure becomes optional. LTE can make the emergency call without SIM card. This is another case of improvement in IMS emergency call in LTE. In this case IMEI is required. Basically mobile network is doomed without IMSI or IMEI. This needs SGW/PGW to be prepared for receiving Create Session requests without IMSIs or IMEIs [32].

2.3.8.2. Self-organizing networks (SON)

SON is started in first Release in LTE or Release 8 of 3GPP. This release enhanced SON features in multi-vendor network environment. SON tries to reduce the number and structure of network parameters and to reduce the complexity of network architecture. Combination of the 2G, 3G and EPC is another feature in SON in LTE, because of the SON the number of base station (Home eNB) well increasing. The main functionality of SON includes: self-configuration, self-optimization and self-healing [33].

3. TURBO CODING IN LTE

3.1. Background

Storage and communication have become part of our life's. Without error control coding or channel coding reliable storage and data transmission would be impossible. Error in data storage or data transmission can come from many different sources such as physical defects, interference, random noise or channel fading. Therefore, correcting error coding may reduce channel errors, improving the quality of data storage and the data transmission [34]. There are two strategies which help to improve the quality of the channel or reducing the occurred errors. These strategies are Automatic Repeat Request (ARQ) and Forward Error Correction (FEC) [34].

The ARQ attempts to detect the errors in the received data. It will detect errors and the receiver will report the transmitter any existence of errors and the transmitter will resent the data until it is received correctly. However, the FEC is not only detecting errors but it also attempts to correct the errors, as in many digital communications where retransmission is impossible or difficult. For instance satellite receives only, it cannot transmit reports. In addition, many practical applications cannot retransmit easily. Therefore, it is not exaggeration to say that it is impossible for any receiver in a real-time broadcasting system to request the data and resent it, therefore the FEC is the only suitable solution [35].

There are several techniques for error control coding, such as Hamming Codes, Hamming codes was introduced in 1950 by Richard Hamming. Before Hamming there were some basic techniques for error control coding such as parity checks but they haven't done a very good job. Hamming codes are sometimes called Error-correcting code memory ECC. There are a number uses of hamming codes at today's application such as RAM and RAID2. Moreover, the second technique which released after Hamming codes called Reed-Solomon codes is introduced by the Reed and Solomon in 1960. RS codes uses mathematical technique for detecting errors and correcting. The RS codes were combined with convolutional codes called RSV which is used in many applications (CD, DVD, blue Ray… etc.) [35].

One of the main the advantage of the RS is the burst errors. RS also used in the space acceleration (Galileo n 1989, pathfinder 1996 and MER in 2003) [36]. The RS is very close to the Shannon limits. In 1993 turbo codes was introduced by the group of the communication engineering in French and it's very close to Shannon limits [37].

Two different coding techniques were used in channel coding in LTE. First is tail biting convolutional codes and the second is turbo coding. LTE uses the tail biting convolutional code for control information and turbo coding for data channels [38]. In the rest of this chapter, the structure of the tail convolutional coding in general will be focused and also the convolutional coding in LTE will be discussed. Moreover, we demonstrate the structure of the turbo coding in LTE. Finally, the Circular-buffer rate matching for turbo coding is another subject in this chapter.

3.2. Tail Biting Convolutional Codes

Convolutional codes are introduced by Elias in (1954) [39]. Reed–Solomon or tail biting convolutional codes are used in many communications systems and other applications. The RS error correction works by adding some redundant bits to a digital data sequence. This is done by oversampling a polynomial constructed from the encoded data. The polynomial is evaluated at several points and then these values are sent (or recorded). By sampling the polynomial more often than needed, the receiver can recover the original polynomial in the presence of a relatively low number of errors [50].

3.2.1. Convolutional encoder

Commonly it has three parameters n, k, m, where n is the number of output bits, k is the number of input bits and m is the number of memory registers. The code rate of convolutional codes is the ratio of number of the input bits to output bits (k/n). With the code rate of convolutional codes we can measure the efficiency of the code. Basically, the code rate of convolutional code is from 1/8 to 7/8. Figure 3.1 represents the convolutional encoding with 3 memory registers one input bit and three output bits and code rate is 1/3.

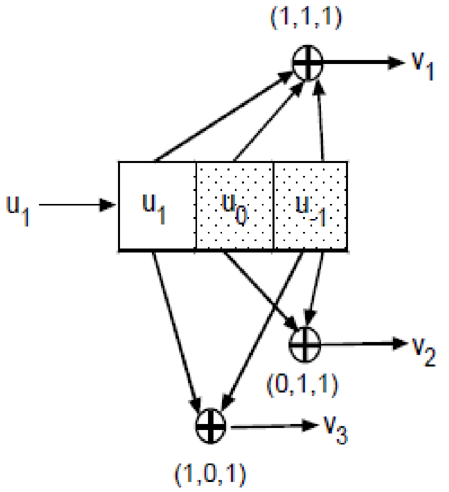

Figure 3.1. Convolutional codes [39].

The output in the Figure 3.1 can be calculated by the selection of the shift registers. For example the output of first generator polynomial is (1, 1, 1), the output of the second generator polynomial equal to (0, 1, 1) and output of the third generator polynomial equal (1, 0, 1). The output is only by the sum of these bits.

$v_1 = mod2\ (u_1 + u_0 + u_{-1})$
$v_2 = mod2\ (u_0 + u_{-1})$
$v_3 = mod2\ (u_1 + u_{-1})$

The puncturing code in convolutional codes comes from combination of two convolutional codes in the individual system and the result is code rate ($^k/_n$) as shown in figure 3.2. By using two rates $^1/_2$ codes together and using only one of the two outputs of the first $^1/_2$ codes, the result is two input and three output or code rate of the convolutional code is ($^2/_3$). The advantage of the puncturing is that rates can be changed dynamically depending on the channel condition.

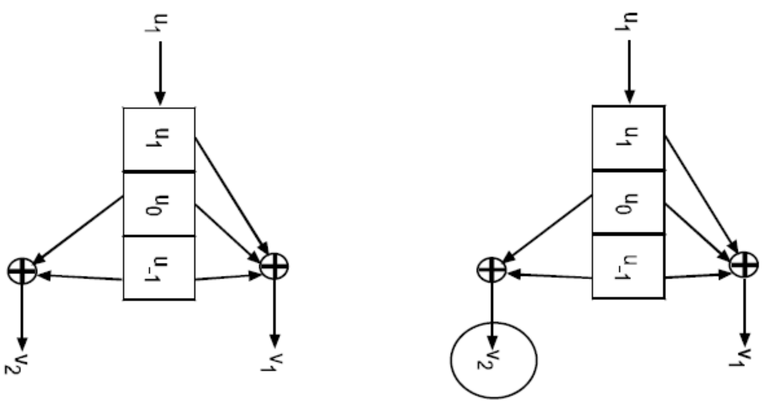
Figure 3.2. Puncturing in convolutional code with rate (2/3) [39].

There are two methods of the convolutional encoding and the previous section showed mathematically what happens in an encoder. The encoding hardware is more simple because does no math. Table 3.1 represents the uses of the convolutional encoding. This table consists of four items: input bits, the state of the encoder, output bits and the output state which will be the input state for next bit. There are three graphical ways: state diagram, tree diagram and trellis diagram. We focus in the trellis diagram, they represent linear time sequencing of the events. The discrete time in the x-axis and the y-axis shows all possible states. Figure 3.3 shows the convolutional encoder of the input bits 101100 and the output bits 11 11 01 11 01 11 01 01 11 [39].

Input Bit	Input State			Output Bits		Output State		
I_1	s_1	s_2	s_3	O_1	O_2	s_1	s_2	s_3
0	0	0	0	0	0	0	0	0
1	0	0	0	1	1	1	0	0
0	0	0	1	1	1	0	0	0
1	0	0	1	0	0	1	0	0
0	0	1	0	1	0	0	0	1
1	0	1	0	0	1	1	0	1
0	0	1	1	0	1	0	0	1
1	0	1	1	1	0	1	0	1
0	1	0	0	1	1	0	1	0
1	1	0	0	0	0	1	1	0
0	1	0	1	0	0	0	1	0
1	1	0	1	1	1	1	1	0
0	1	1	0	0	1	0	1	1
1	1	1	0	1	0	1	1	1
0	1	1	1	1	0	0	1	1
1	1	1	1	0	1	1	1	1

Table 3.1. Convolutional encoder (2, 1, 4) [39].

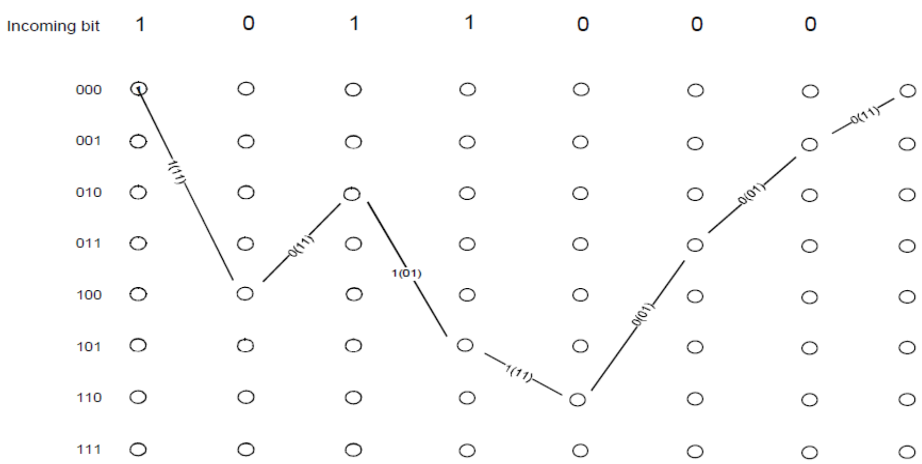

Figure 3.3. Convolutional encoder triller diagram [39].

3.2.2. Convolutional decoder

There are two different decoding of convolutional codes: Sequential decoding (Fano algorithm) and Maximum likely-hood decoding (Viterbi decoding). We just focus on the Maximum likelihood (Viterbi) decoding. Viterbi decoding is the best known implementation of the maximum likelihood decoding. In this kind of decoding the probability of error is small and the probability of two errors is much smaller than the single error [40].

Figure 3.4 shows the Viterbi decoding, where the Viterbi decoder examines an entire received sequence of a given length. The decoder computes a metric for each path and makes a decision based on this matrix. All paths are followed until two paths converge onto one node. Then the path with the higher metric is kept and the one with lower metric is discarded and the path selected are called the survivors.

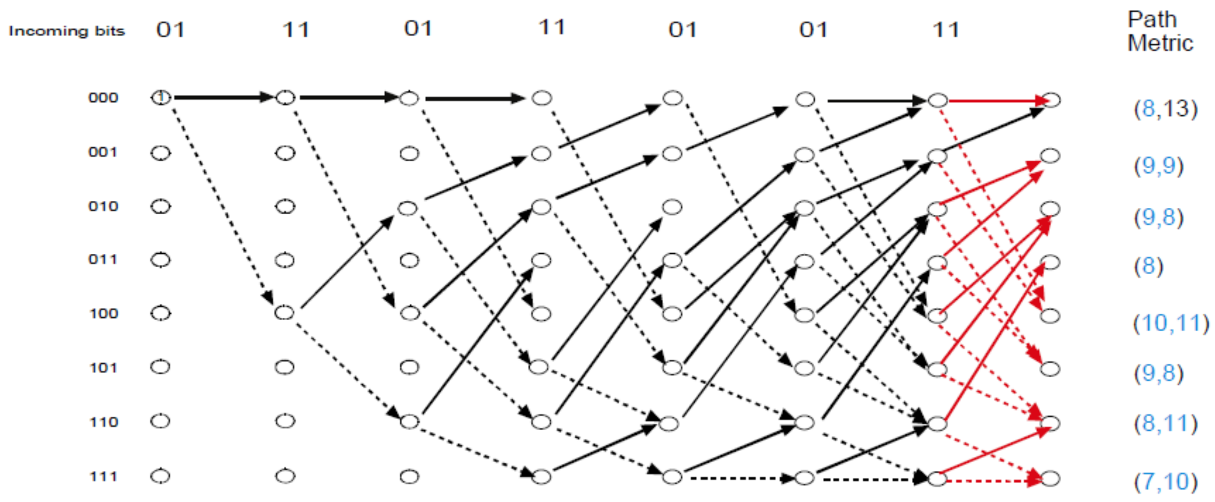

Figure 3.4. Viterbi Decoding [39].

As we motioned in the chapter two LTE uses one third code rate with constraint length 8 convolutional code in the channel coding for control information. The structure of convolutional code in LTE is shown in Figure 3.5 and the three output of the convolutional code encoder is $d_k^{(0)}, d_k^{(1)}$ and $d_k^{(2)}$ represent the first, second and third parity streams respectively [27].

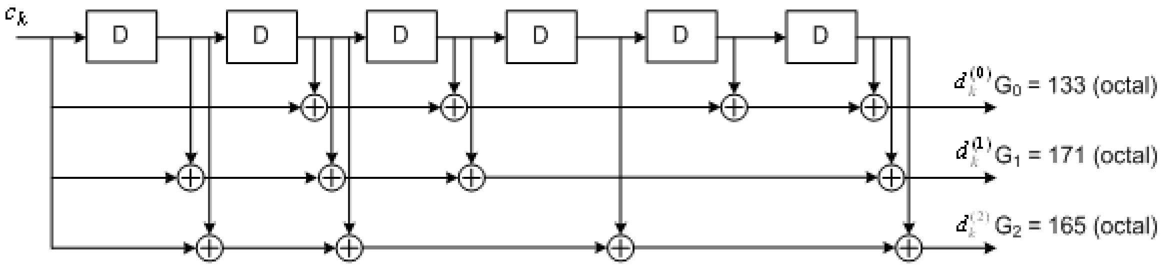

Figure 3.5.Convolutional encoding in LTE [25].

3.3. Turbo Coding

Turbo code was introduced by Glavieux, Berrou and Thitimahshuma in 1993 in French, the initial result showed that turbo code could achieve energy efficiencies within half decibel of the Shannon capacity in 1948. Turbo codes have been proposed for low power system like satellite communication or interference limited like third generation circular technologies [36].

3.3.1. Turbo encoder

Turbo code encoder in LTE consists of two identical recursive systematic convolutional codes RSC by parallel concatenation. The code rate of the RSC encoder is 1/2, the input of the second RSC is internal turbo encoder interleaver. Only one of the systematic output from two component encoder is used because the systematic output from the other component encoder is just a permuted version of the chosen systematic output as can be seen in Figure 3.6.

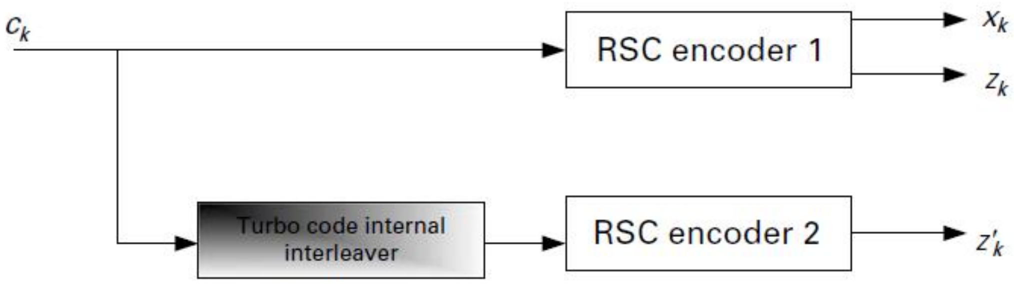

Figure 3.6. Turbo code in LTE [25].

The Turbo coding in LTE system is parallel concatenated convolutional code (PCCC) with two 8 -state rate 1/3 constituent encoders, the transfer function is given as:

$$G(D) = \left[1, \frac{g_1(D)}{g_0(D)}\right], \qquad g_0(D) = 1 + D^2 + D^3, \qquad g_1(D) = 1 + D + D^3$$

The initial value of the shift registers of the 8-state constituent encoders is set to all zeros state before starting to encode the input bits as shown in Figure 3.7. The output of the LTE turbo encoder consists of three parts: one is systematic bits and two parity bits. The Turbo encoder output is given as:

$$d_k^{(0)} = x_k, \qquad d_k^{(1)} = z_k, \qquad d_k^{(2)} = z'_k \qquad k = 0, 1, 2, \ldots, K-1$$

The systematic bits x_k is untouched input bits, and parity z_k, z'_k is the output of the first and second convolutional encoder respectively.

Trellis termination is another part of the turbo encoder in LTE. The Trellis termination performed by taking the tail bits from the shift registers feedback after all information bits are encoded. Tail bits are padded after the encoding of information bits. The first three tail bits shall be used to terminate the first constituent encoder (upper switch of Figure 3.7 in lower position) while the second constituent encoder is disabled. The last three tail bits shall be used to terminate the second constituent encoder (lower switch of Figure 3.7 in lower position) while the first constituent encoder is disabled [25]. The transmitted bits for trellis termination are then:

X_K, Z_K, X_{K+1}, Z_{K+1}, X_{K+2}, Z_{K+2}, X'_K, Z'_K, X'_{K+1}, Z'_{K+1}, X'_{K+2}, Z'_{K+2}

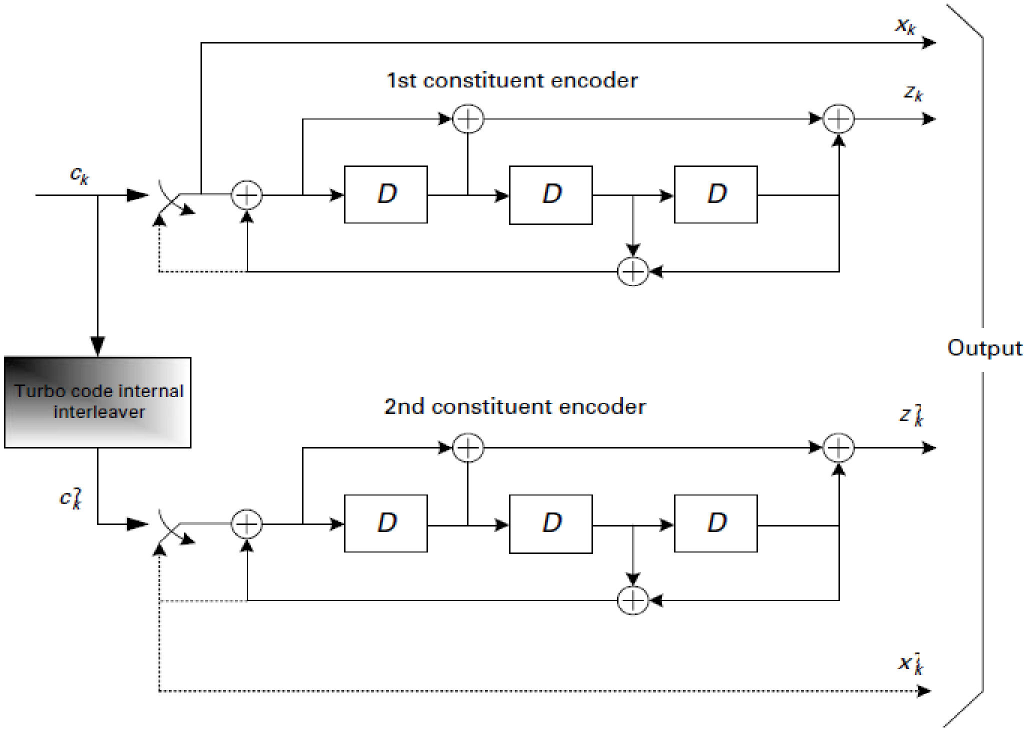

Figure 3.7. PCCC in LTE [25].

3.3.2. Turbo code internal interleaver

The internal interleaver is the critical part of the turbo encoder in LTE, and it's the key difference between the turbo encoder in LTE and UTMS. The input bits to the internal interleaver in LTE is denoted by $c_0, c_1, \ldots, c_{K-1}$, where K is the number of input bits, and the output of the turbo encoder internal interleaver is denoted by $c'_0, c'_1, \ldots, c'_{K-1}$. The relationship between the input and output of the turbo encoder internal interleaver is given as:

$$c'_i = c_{\Pi(i)}, \quad i = 0, 1, \ldots, (K-1)$$

The relationship between the input index Π (i) and the output index i satisfies the following quadratic form:

$$\Pi_{(i)} = (f_1 . i + f_2 . i^2) mod\ K$$

where the parameter f_1 and f_2 depend on the block size K in table (2.2) [26].

The contention free interleaver used in the LTE system is based on the QPP principle [42]. The QPP interleaver size granularity becomes coarser as the interleaver size increases. For $40 \leq K \leq 512$, K contains all multiples of 8. For $512 \leq K \leq 1024$, K contains all multiples of 16. For $1024 \leq K \leq 2048$, K contains all multiples of 32 and for $2048 \leq K \leq 6144$, K contains all multiples of 64. The advantage of the QPP interleaver is its simplicity and de-interleaver is inverse of the same algorithm and hardware can be used for both interleaving and de-interleaving. Figure 3.8 shows the interleaver and de-interleaver of QPP, for block size 40 [25].

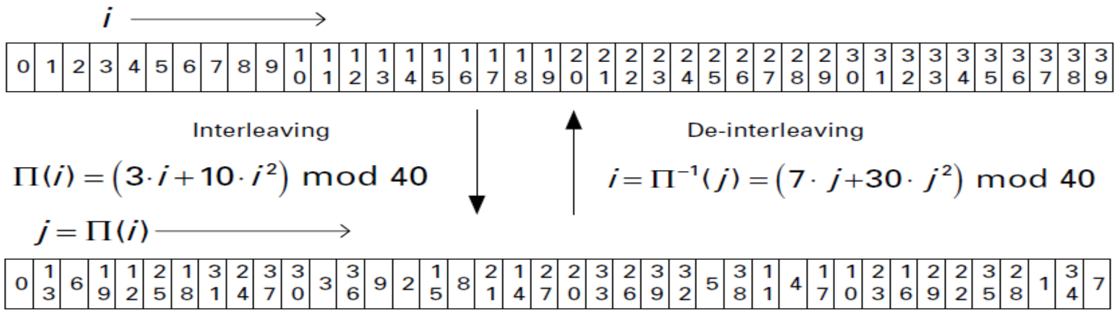

Figure 3.8. Interleaver and de-interleaver for QPP with block size 40 in LTE turbo code encoder [25].

3.3.3. Circular-buffer rate matching for turbo code in LTE

A synchronous adaptive hybrid ARQ is used for uplink transmissions in LTE, also adaptive hybrid ARQ and an asynchronous in the downlink transmissions, with adaptive modulation scheme, resource and coding can change in retransmissions. It requires different number of coded bits to be transmitted from original transmission. Another things constraint in LTE turbo coding is that systematic bits should preferably be carried in the first transmission attempt as turbo code performance is sensitive to the systematic bits. This require adaptable rate matching and highly flexible scheme. So as to meet this goal, a circular-buffer-based rate-matching scheme is used in LTE which is shown in Figure 3.9. As we motioned before the turbo code rate in LTE is 1/3, and the three outputs of turbo codes consist by one stream of systematic bits and two streams of parity bits. Each of these three streams is interleaved separately by subblock interleavers. After that the interleaved parity one and parity two are interlaced. During the HARQ rate-matching procedure, each transmission reads bits from the buffer, starting from an offset position and increasing the bit index. If the bit index reaches the maximum number, the bit index is reset to the first bit in the buffer. In other words, the buffer is circular [25].

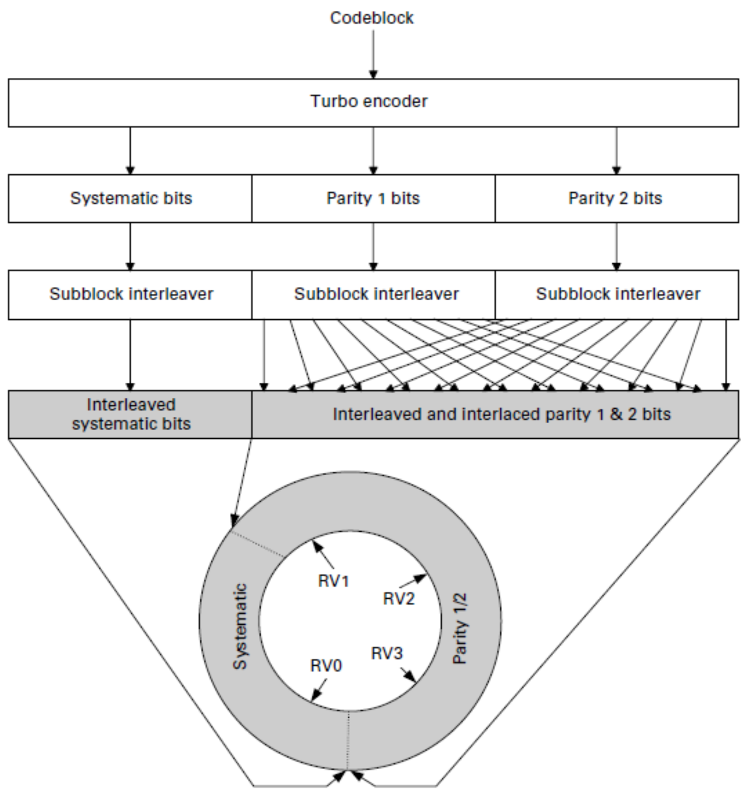

Figure 3.9. Circular Buffer rate matching in LTE [25].

3.3.4. Sub block interleaving

The input bits to the block interleaver are denoted by:

$d_0^{(i)}, d_1^{(i)}, d_2^{(i)}, \dots d_{D-1}^{(i)}, i = 0,1,2$

The number of bits for systematic, parity 1 and parity 2 streams is $D = K + 4$. K Is the number of bits within code block with $x_k, k = 0, 1, 2, \dots, K - 1$ and trellis termination adds 4 bits to each of systematic, parity 1 and parity 2 streams. Sunblock interleaver is arrange the sub block by writing row-wise in rectangular matrix, therefor, permute the rectangular matrix and reading from the matrix column-wise. As can be seen in Figure 3.10 [27].

3.3.5. Sub block interlacing

The length of circular buffer is $K_w = 3K_\Pi$, when the number of interleaved bits in each of systematic, parity one and parity two streams isK_Π, when w_1, $w_2, \dots, w_{(K_w - 1)}$ is bit stream in the circular buffer and given as:

$$w_k = v_k^{(0)} \qquad k = 0, 1, \dots, (K_\Pi - 1)$$

$$w_{K_\Pi+2k} = v_k^{(1)} \qquad k = 0, 1, \dots, (K_\Pi - 1)$$

$$w_{K_\Pi+2k+1} = v_k^{(2)} \qquad k = 0, 1, \dots, (K_\Pi - 1)$$

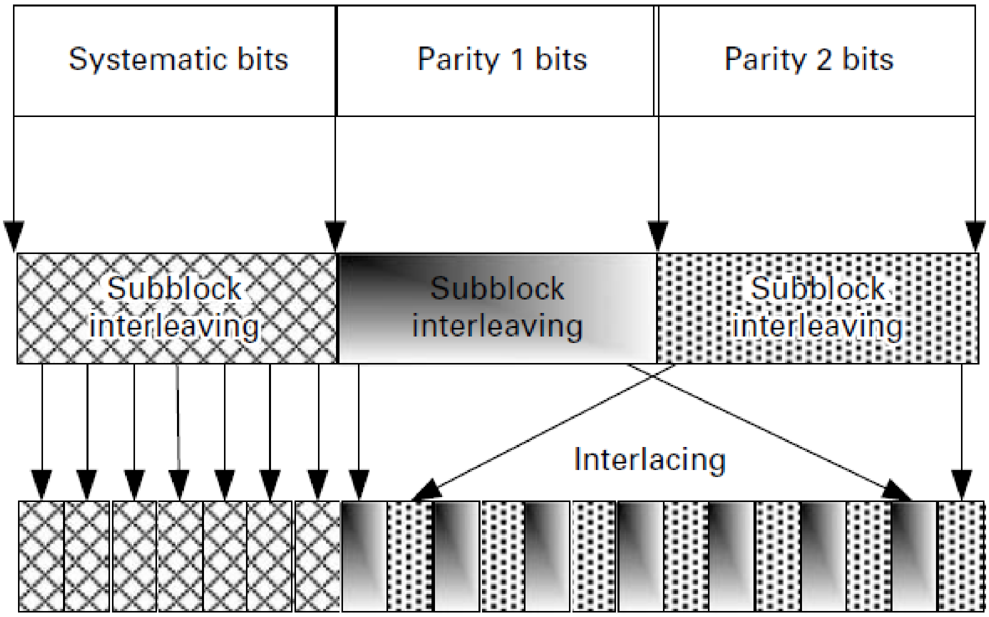

Figure 3.10. Subblock interleaving and interlacing in LTE [25].

It should be noted that parity one and parity two are performed with sub block interlacing shown in Figure 3.10. Systematic bits are generally part of the first HARQ transmission. In response to HARQ NACK, for example, sub block interlacing guarantees that an equal amount of parity one and parity two bits are transmitted [25].

3.4. Turbo Decoder in LTE

Turbo encoder has three outputs: one parity bits and two systematic bits, as well as consist of a feed-through, with two 8-state RSC and an interleaver. LTE secreted PCCC structure and turbo interleaver because perform turbo algorithm in the decoder [43]. The decoding of turbo codes is worked on an iterative structure constructed from two MAP decoders, for each one of constituent encoder. As shown in Figure 3.11 each MAP decoder has three inputs: a prior LLR for systematic bits, channel LLR for systematic bits and channel LLR for parity bits. The output of each MAP decoder is called 'extrinsic information or LLR denoted Z_x. First LLR for parity bits is to be decoded by MAP1 decoder and second parity bits by MAP2 decoder. Moreover, the first LLR as extrinsic information decoded from MAP1 is fed back to input of MAP2. MAP2 then performs second decoding by extrinsic information and second LLR for parity bits. As well as MAP1, the second LLR resulted from MAP2 is feedback to MAP1 as extrinsic information [49].

The desired effect is that the extrinsic LLRs increase in absolute value after each iteration. This effectively improves reliability of the decoding, and decrease bit error rate (BER). In general, since turbo code is a kind of convolution codes, BER performances is decided by constraint length of encoder. The input sequences of MAP1 shall be also uncorrelated with input sequences of MAP2 by turbo interleaver to improve the value of extrinsic information [25].

QPP interleaver has been gaining since it provides contention-free interleaving functionality for high speed parallel turbo decoders. By using QPP interleaver, parallel window of MAP can be parallel operated in each window MAPs. We would like to implement high speed parallel turbo decoder which have 64-parallel window MAP [45].

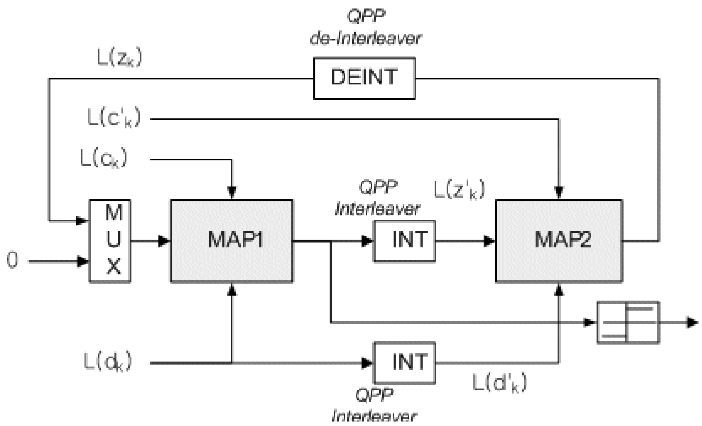

Figure 3.11. General Turbo decoder architecture in LTE [43].

4. PERFORMANCE ANALYSIS OF TURBO CODING IN LTE

4.1. Turbo Code Performance with Different Transport Block Size

4.1.1. Turbo code encoder performance with different transport block size

The required input data ordering for a block of size K is: X_0, X_1, X_2 . . . X_{K-1}. The output data is three bits wide. Table 4.1 lists the ordering for a block of size K.

Table 4.1. Turbo Encoder Output Data Ordering.

output data	source data		
	2	1	0
0	Z'_0	Z_0	X_0
1	Z'_1	Z_1	X_1
…	…	…	…
K-1	Z'_{K-1}	Z_{K-1}	X_{K-1}
K	X_{K+1}	Z_K	X_K
K+1	Z_{K+2}	X_{K+2}	Z_{K+1}
K+2	X'_{K+1}	Z'_K	X'_K
K+3	Z'_{K+2}	X'_{K+2}	Z'_{K+1}

The encoding delay D is the number of clock cycles consumed to encode an entire block of data. If K is the block size, D = K + 14. The encoding delay does not include the loading delay, which requires the same number of clock cycles as the block size K to load the input data to the input buffer. For example:

When K = 6144, D = 6144 +14 = 6158

When K = 40, D = 40 + 14 = 54

Then the encoding latency (the time taken by the encoder to encode an entire block) can be calculated using the following formula:

$$s = \frac{D}{f_{MAX}}$$

Where f_{MAX} is the system clock speed. For the above examples, L = 0.22µs and 25.13µs respectively for f_{MAX} = 245 MHz.

The throughput can be calculated using the following formula:

$$T = \frac{K * f_{MAX}}{D}\ bps$$

For the examples in the previous section, T = 249.43 Mbps and 185.18 Mbps respectively.

4.1.2. Turbo code decoder performance with different transport block size

The decoding delay D is the number of clock cycles consumed to decode an entire block of data. This is dependent on the block size, the number of iterations to perform, and the number of engines available in the decoder. The following calculations assume no early termination is taking place, so are the worst case latency. Let K be the block size, I be the number of decoding iterations and N be the number of engines specified in the decoder. Then the decoding delay D can be calculated using one of the following formula:

If K < 264, D = 26 + (2 × f (K, N) + 14) × 2 × I

If K > 264, D = 26 + (f (K, N) + 46) × 2 × I

Where:

$$f(K,N) = \begin{cases} K/N & \text{if K is divisible by N} \\ K/8 & \text{if K is not divisible by N} \end{cases}$$

For example:

If K = 6144, N = 8, I = 8, D = 26 + (6144/8 + 46) × 2 × 8 = 13,050

If K = 40, N = 8, I = 8, D = 26 + (2 × 40/8 + 14) × 2 × 8 = 410

The decoding latency (the time the decoder takes to decode an entire block to the decoded data is ready for output) can be calculated using the following formula:

$$s = \frac{D}{f_{MAX}}$$

Where f_{MAX} s the system clock speed. For the above examples, L = 52.2µs and 1.64µs respectively for f_{MAX} = 250 MHz. These calculations are for running the Turbo decoder for 8 iterations (I = 8) and assuming no early termination has occurred.

The throughput can be calculated using the following formula:

$$T = \frac{K * f_{MAX}}{D}\ bps$$

For the examples in the previous section, T = 117.7 Mbps and 24.38 Mbps respectively.

These analyses show that turbo code improving his performance with increasing codeblock size in both encoder and decoder [53].

4.2. Performance of Turbo Coding with Different Transport Block Segmentation Procedure

We discussed earlier that the maximum codeblock size in LTE denoted as Z is limited to 6144 bits. When the transport block size denoted as B (or total size after transport block concatenation) is larger than 6144 bits, segmentation of the input bit sequence is performed and an additional codeblock CRC (CB-CRC) sequence of L = 24 bits is attached to each codeblock as shown in figure (4.1)

Figure 4.1. Maximum Codeblock size in LTE.

The input bit sequence to the codeblock segmentation is denoted by;

$b_0, b_1, b_2, b_3, \ldots\ b_{B-1}$ B>0

The output code block segmentation is;

$c_{r0}, c_{r1}, c_{r2}, c_{r3}, \ldots\ c_{r(Kr-1)}$

Where r represents the code block number and K_r indicates the number of bits for code block number (r).

If the transport block is bigger than 6120, then transport block will be divided into a number of codeblocks because the maximum codeblock size in internal turbo code interleaver equal 6144. Therefore 6120 bits is data bit and 24-bits is CRC, as given below:

$L = 0, C = 1$ $\quad B' = B,$ $\quad B \leq Z.$

$L = 24, C = \lfloor B/(Z - L) \rfloor$ $\quad B' = B + C \times L,$ $\quad B > Z.$

Where B’ is transport block after segmentation, L is CRC bits and C is number of codeblock.

Let us assume a transport block size of the message size is equal to 18000 bits. Then 24 bits of CRC is added to the transport block and the result will be equal to 18024. In this example since the transport block is larger than Z, transport block segmentation is performed.

$$B = M + L$$

$$C = \lceil B/(Z - L) \rceil$$

$$C = [18024/\,(6144 - 24)] = 3$$

The transport block is segmented into three blocks which can be seen in Figure 4.2

$$B' = B + C.L$$

$$= 18024 + (3 \times 24)$$

$$= 18096$$

$$Mod\ (18096, 6144) = 5808,$$

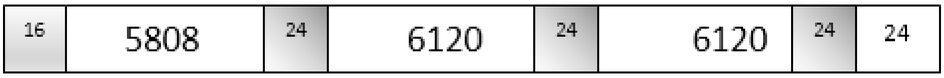

Figure 4.2. Codeblock segmentation with CRC.

First segment is equal to 5808 and matched to the nearest interleaver size of 5824 in the table 2.2, and the difference is the filer bits.

$$5824 - 5808 = 16\ bits$$

Filler bits equal to 16 bits and last transport block size is equal to 5824 bits. The transport block consists four sub blocks. First three sub block size are 6144 bits, and the last one is 5824 bits.

There were problems with this segmentation approach which was driven by different code block size and different code block performance. This is because the turbo code performance will improve with increasing codeblock size, for a given SNR; it would always be more likely that code block with the smallest size is an error. The ACK/NACK is provided per transport block and a single ACK/NACK is provided per transport block (set of code blocks). Therefore, the error performance of transport block would be limited by the performance of the smallest code block size because the receiver will NACK the transport block as long as there is a single code block in error [44].

In order to reduce the number of filler bits while keeping the code block sizes approximately the same, LTE uses two adjacent interleaver sizes, a larger size K_+ and the next smaller size K_-. The first segmentation size denoted as K_+ is minimum K. such that:

$$C \times K \geq B'$$

The second segmentation size denoted K_- is maximum K in the Table 2.2 such that K < K+. The differences in the two adjacent segmentation sizes that are also turbo interleaver sizes denoted as ΔK is given as:

$$\Delta K = K_+ - K_-$$

Let us denote the number of codeblocks of size K_+ and size K_- as C_+ and C_- respectively. When the total number of codeblocks C = 1, the codeblock size is K_+ and C_+ = C = 1. The number of filler bits K is obtained as:

$$F = C_+ \times K_+ + C_- \times K - B'$$

The filler bits are added to the beginning of the first codeblock with r = 0. An example of codeblock segmentation is shown in Figure 4.3. The total number of bits input to the code block segmentation is assumed as B = 19 000 bits that include 18976 data bits and 24 bits transport block CRC (TB-CRC) [25]. Since B > Z = 6144 the total number of code blocks C is determined as below:

$$C = [B/(Z - L)] = [19\,000/(6144 - 24)] = 4$$

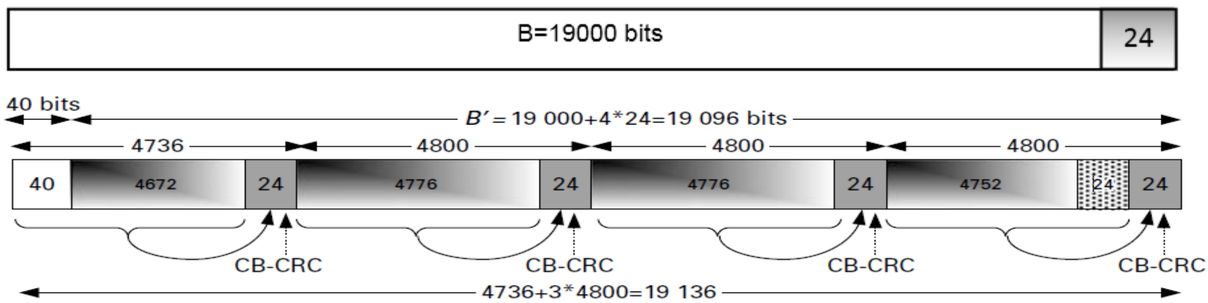

Figure 4.3. Codeblock segmentation in LTE.

The new total size B' is given as:

$B' = B + C \times L = 19\,000 + 4 \times 24 = 19096.$

Since C > 1, the first and second segment sizes are determined by using the procedure described above. The first segmentation size K_+ = 4800 is minimum K in LTE interlever Table. That results in C×K ≥ B. The second segment size is one size smaller than K_+= 4800 which is K_-= 4736 from Table 2.2,

The difference between the two segment sizes is

$\Delta K = K_+ - K_- = 4800 - 4736 = 34$

The number of codeblocks of size $K_-$and K_+ are given as:

$$C_- = \left\lceil \frac{C.\ K_+ - B'}{\Delta K} \right\rceil = \left\lceil \frac{4 \times 4800 - 19096}{64} \right\rceil = 1$$

$C_+ = C - C_- = 4 - 1 = 3$

The number of filler bits F is:

$$F = C_+ \times K_+ + C_- \times K_- - B'$$

$$= 3 \times 4800 + 1 \times 4736 - 19096 = 40$$

We note that the number of filler bits is more than the segmentation in the first example. However, as pointed out earlier the scheme of the figure (4.2) suffers from the problem of vastly different code block sizes. We also note that if two adjacent interleaver sizes were not allowed and a single interleaver size with K = 4800 was used, the number of filler bits would have been greater as calculated below:

$F = C \times K = 4 \times 4800 - 19096 = 104.$

As show in figure (4.4) the transmitter send transport block (19000 bits) with two different code block size 4736 and 4800 bits.

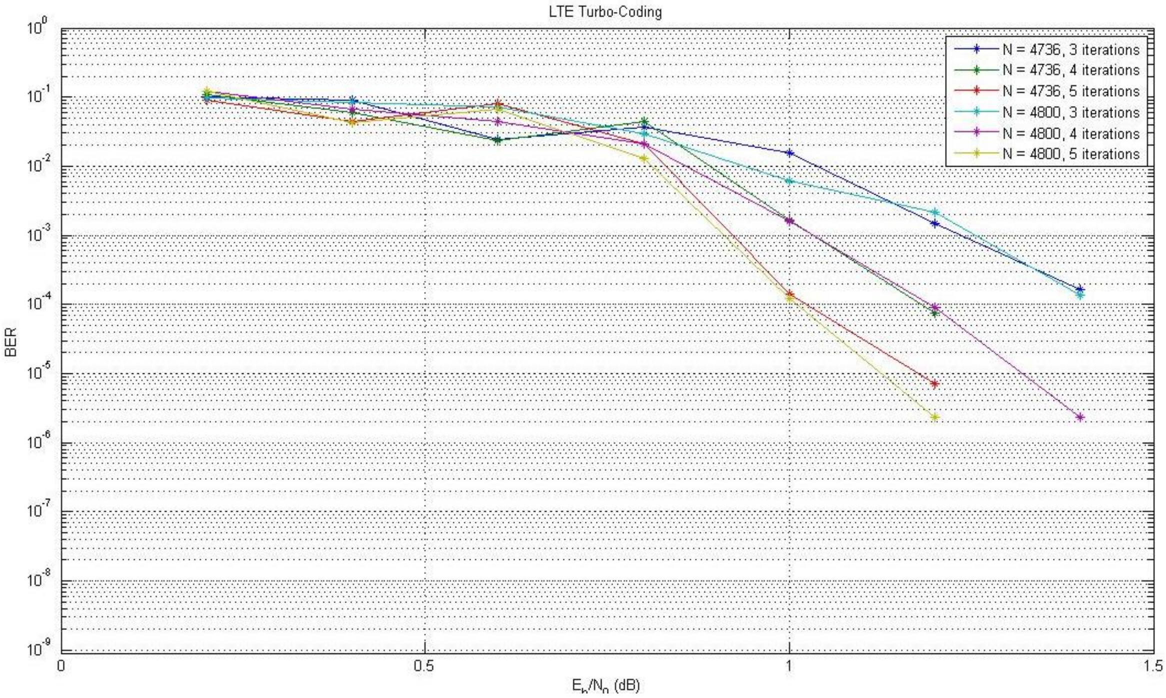

Figure 4.4. Turbo code performance in LTE with different iteration and different codeblock size, BPSK modulation

Therefore, using two adjacent interleaver sizes allows keeping the code block size within a transport block approximately the same while reducing the number of filler bits. However, the two adjacent interleaver size decision was made under the assumption that these bits are transmitted over the air and hence represent an unnecessary overhead. In the later stages of LTE standard development, it was agreed that filler bits are mostly removed after channel coding. Therefore, the overhead represented by filler bits is effectively smaller than if all the coded filler bits are transmitted over the air. The filler bits that go into systematic bits $d_k^{(0)} = x_k$ and second stream from first constituent encoder $d_k^{(1)} = z_k$ are removed before transmission. This is achieved by setting:

$C_k \quad = 0$

$d_k^{(0)} = NULL$ For $k = 0, \dots (F-1)$

$d_k^{(1)} = NULL$

Where C_k, $d_k^{(0)}$, and $d_k^{(1)}$ is the output of first, second and third outputs turbo encoder respectively. And F is the total number of filler bits.

We note that the filler bits that go into the second constituent encoder are not removed. It is difficult to track these bits due to the fact that the input to the second constituent encoder is the interleaved version of the input stream to turbo code [46].

4.3. Discussion and Result

We motioned before the problems in the previous segmentation methods (different code block size and different code block performance, the error performance of the transport block would be limited by the performance of the smallest code block size).

In order to fix these problems, we present new approach to transport segmentation. The new approach work with the transport block segmentation procedure, when the transport block size larger than 6120 bits, transport block segmented into a number codeblocks. Transport block in the new approach divides into 6120, if the remaining bits are not equal to zero then filler bits are added to the beginning of the first codeblock till achieving 6120 and 24-CRC bits will be added to get maximum block size with (6144). These bits are called filler bits. Let us assume a transport block size of the message size equal to 41696 bits.

B = message bite + CRC-TR

B = 41720 + 24 = 41720 bits

$C = B/(Z - L)$

$$= \left\lceil \frac{41720}{(6144 - 24)} \right\rceil = 7$$

$B' = B + C \times L$

$= 41720 + 7 \times 24$ = 41888 bits

$F = Mod\ (41888, 6144) - 6144$ = 1120

Transport block segmentation is shown in figure (4.5).

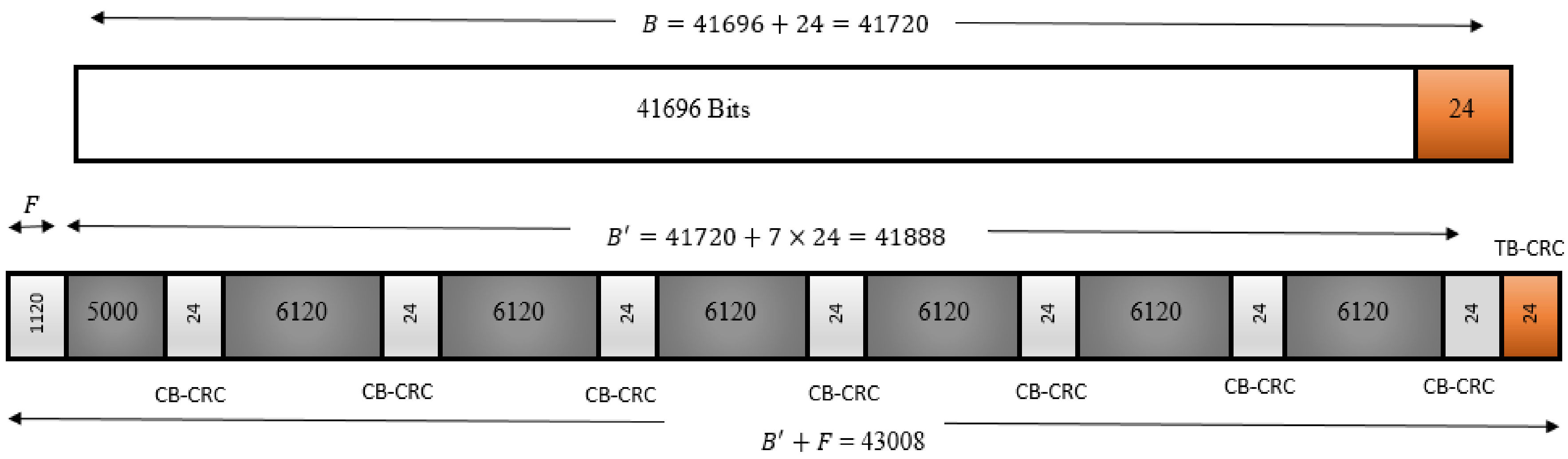

Figure 4.5. Transport block segmentation into equal size codeblocks.

Figure 4.6 shows the performance of maximum codeblock size with new approach in LTE. The message size is equal to 41696 bits. And filler bits are equal to 1120 bits.

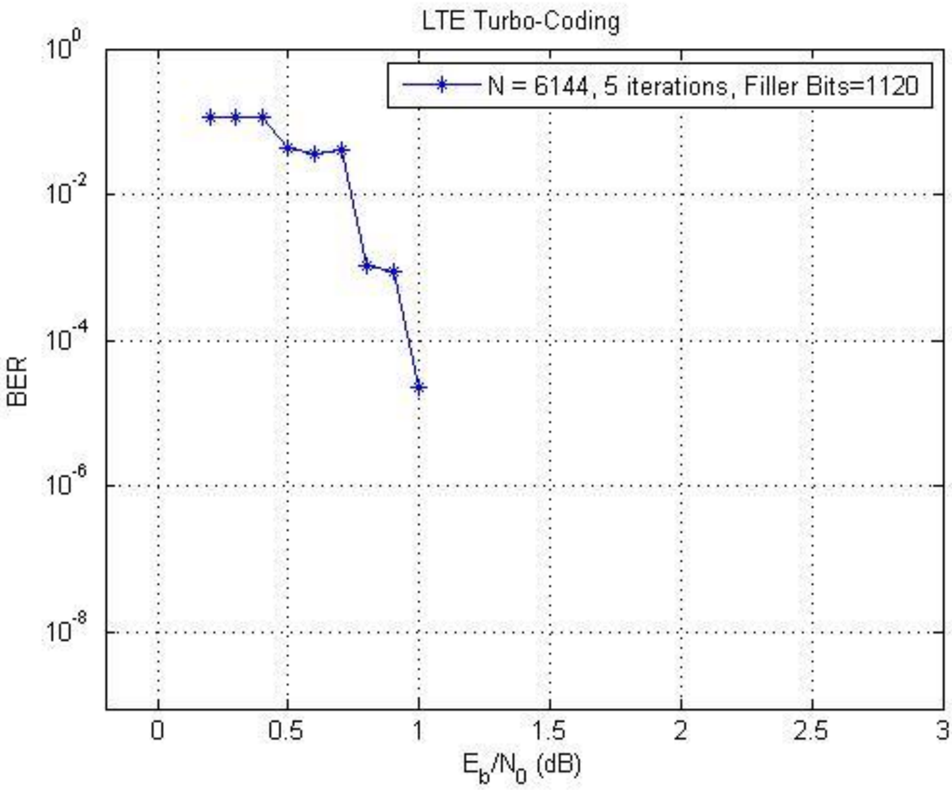

Figure 4.6. Performance of LTE turbo coding with maximum CB size, BPSK modulation.

In order to demonstrate the result of the new approach, we selected 100 random messages and segmented these messages by using two methods we mentioned before, and took the SNR per bit (E_b/N_o) for all the massages by using the MATLAB simulation. Tables (4.3) and (4.4) are average E_b/N_0 with BER of the 100 random message by using Matlab simulation. Matlab parameters shown in the table (4.2). Figure 4.7 represents a comparison of the average (100) random messages with two modules.

Table 4.2. Parameters of the LTE turbo coding simulation.

Modulation	BPSK
Code rate turbo coding	1/3
Generator polynomial	G1= 13 G2= 15
T	360
Constraint length	8
Number of iteration	5
Channel	AGWN

Table 4.3. E_b/N_0 With bit error rate in LTE.

E_b/N_0	BER
0.2	0.104785238
0.3	0.090125556
0.4	0.070784569
0.5	0.049560444
0.6	0.027378503
0.7	0.011965972
0.8	0.003991371
0.9	0.00112488
1	0.000507862

Table 4.4. E_b/N_0 With bit error rate in LTE with maximum codeblack size.

E_b/N_0	BER
0.2	0.105064
0.3	0.082327
0.4	0.063991
0.5	0.042469
0.6	0.031555
0.7	0.013908
0.8	0.005414
0.9	0.000908
1	5.54E-05

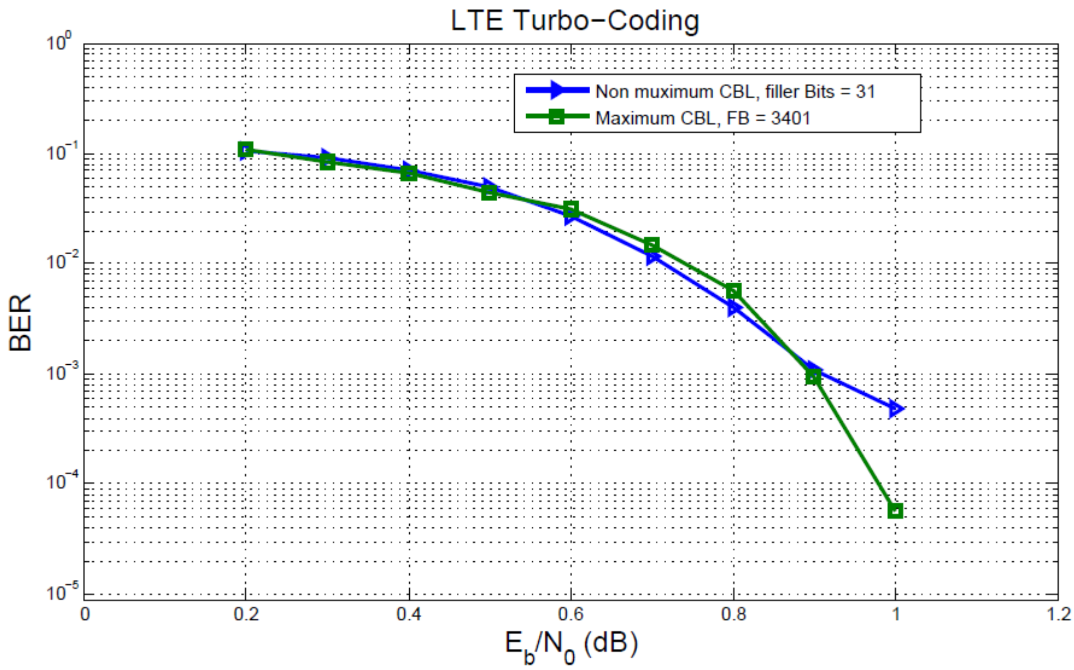

Figure 4.7. Performance analyses of the CB segmentation by using two methods, BPSK modulation.

Simulation result show that new approach performs better than the standard system for SNR per bit values greater than 0.85. For other values, it performs equal or comparable to the standard approach as shown in figure 4.7. In our approach all block sizes are the same and maximum. Different block sizes reduces performance of Turbo Coding. If all codeblock sizes has maximum size, turbo code improves its performance. For example, 1000-bit blocks have an inherent advantage of about 1.4 dB as compared with 100-bit blocks. An additional gain of about 0.5 dB potentially is obtained by going from 1000-bit blocks to 10,000-bit blocks, and another 0.2 dB by going to 100,000-bit blocks. After that, there is less than another 0.1 dB of improvement available before the ultimate capacity limit for unlimited block sizes is reached [52]. Then the overall gain for codeblock size (6144) is going to be 1.81 dB. Another advantage of this new approach is to reduce the complexity in QPP interleaver. New approach has just one disadvantage. This disadvantage is that in some cases large number of filler bits is added. To fix this problem in the later stages of LTE standard development, it was agreed that filler bits are mostly removed after channel coding before transmitted over the air [45].

4.4. Performance Analysis of Parallel Turbo Coding in IEEE 802.16e WiMAX

Concatenated Reed–Solomon Convolutional Code (RS-CC) is used in FEC in WiMAX. This code is used on both the uplink and downlink in IEEE 802.16e WiMAX. It consists of the concatenation of a Reed–Solomon outer code and a rate-compatible convolutional inner code [50].

The physical layer of WiMAX is simulated by the RS-CC and LTE PCCC in MATLAB software, by using the Viterbi algorithm decoding in RS-CC and Max-algorithm decoding in PCCC, with the parameters shown in the table (4.5). Note that parameters used in the PCCC are the same parameters used in PCCC in LTE and the number of decoding iterations are six. QPSK is used when channel have bad conditions and both of them (QPSK and 64QAM) can be used in high E_b/N_o.

Table 4.5. MATLAB parameters of WiMAX with QPSK modulation.

T= 360	Code rate		Constraint length	N	Generator polynomial
RS-CC	1/2		7	384	G1= 177 G2= 133
PCCC	1/3	6 Iterations	4	6144	G1= 13 G2= 15

Simulation results shows the PCCC in WiMAX lead to better performance than RS-CC. Figure 4.8 shows the performance of PCCC compared with RS-CC in WiMAX.

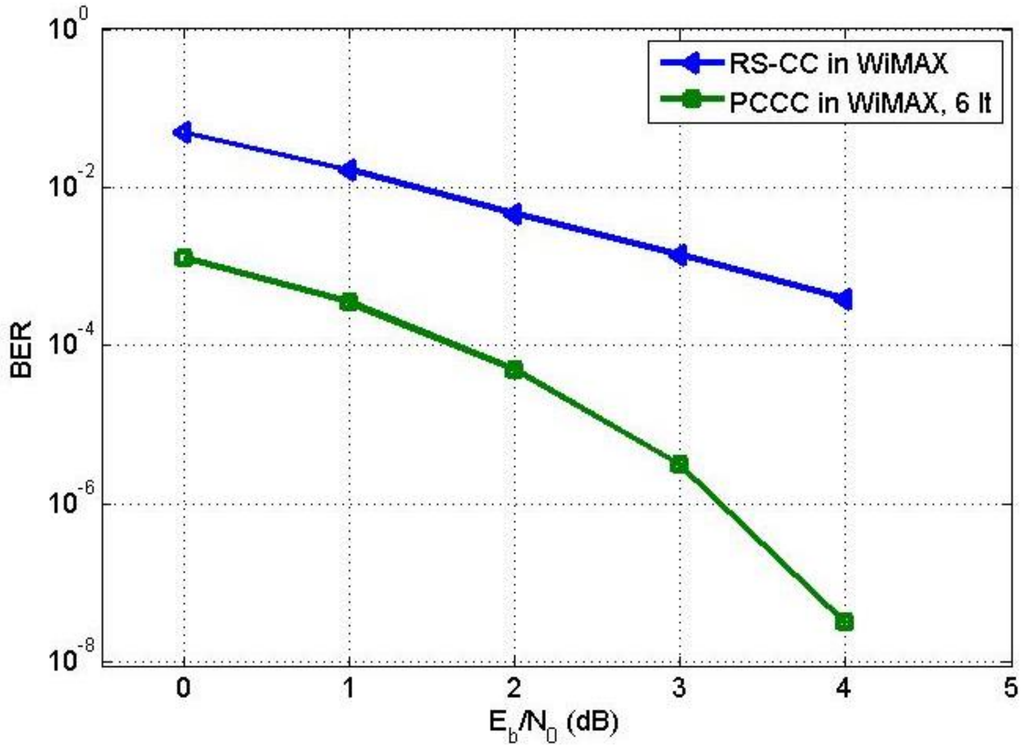

Figure 4.8. LTE PCCC and RS-CC in WiMAX with QPSK modulation.

Figure 4.9 shows the effect of LTE PCCC and RS-CC in WiMAX with 64QAM modulation, with parameters shows in table (4.6). Simulation results shows that PCCC has better performance than RS-CC, because this parameter using in the PCCC and iterative decoding algorithm, the effect of PCCC in 64QAM modulation more effective in QPSK modulation.

Table 4.6. MATLAB parameters of WiMAX with 64QAM modulation.

T= 360	Code rate		Constraint length	N	Generator polynomial
RS-CC	1/2		7	1152	G1= 177 G2= 133
PCCC	1/3	6 Iterations	4	6144	G1= 13 G2= 15

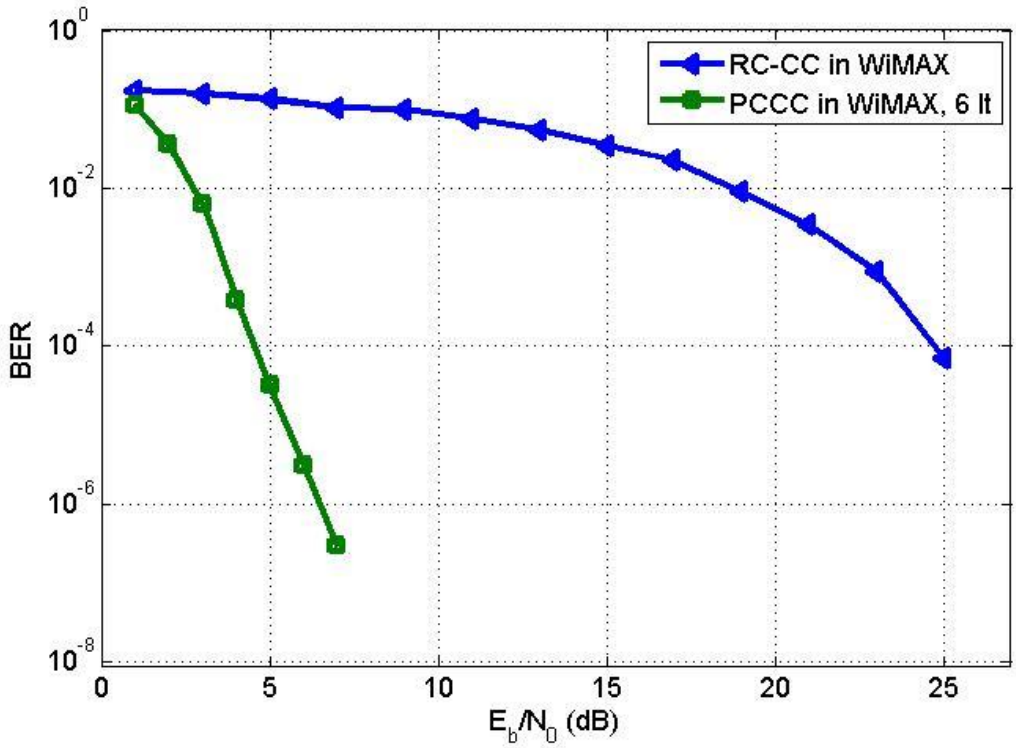

Figure 4.9. LTE PCCC and RS-CC in WiMAX with 64QAM modulation.

5. CONCLUSION

In this study we investigated the architecture and enhancements of the releases of the 3GPP, especially release 8 to release 12. Previous studies are not complete and did not discuss all of the possible scenarios and enhancements of the 3GPP releases. In this study, the channel coding in LTE and LTE-Advanced was focused, and this study also investigated the structure technique of the Convolutional code and Turbo codes.

LTE uses parallel concatenated convolutional code in large data rates (downlink and uplink) and convolution code in control information. The interleaver in the turbo coding in LTE is QPP. In order to combine the channel coding and modulation scheme LTE uses puncturing and circular buffer after turbo coding and convolutional code in channel coding, because LTE using three different Modulation (QPSK, 16QAM, 64QAM) and each modulation needs a specific code rate of the turbo is coding and convolutional code.

Transport block in the LTE is segmented into a number of segmentations in the case TB bigger than the maximum CB length (6144). In this study we presented a new approach to the LTE codeblock segmentation procedure. We investigated and compared new approach with the standard segmentation in LTE. Simulation results show that new approach performs better or similar compared to the standard approach, because new approach uses every time maximum CB length. This improves the performance of turbo coding with maximum CB length, another advantage of new approach is every CB has the same length. Also turbo code improves performance with same CB length. But the disadvantage of new approach to the LTE codeblock segmentation procedure is some time high number of the filler bits. If there are high number of filler bits then more energy is consumed. In order to reduce the number of filler bits in later releases of LTE the filler bits are removed before transmitting the data. Removing filler bits increases the efficiency of the system. The overall gain in the new approach is about 1.83 dB

WiMAX uses RC-CC for error correction in the channel coding. We applied PCCC with the same parameters in LTE like type of interleaver, constraint length, etc. to the IEEE 802.16e WiMAX and compared with the RC-CC with two kinds of modulation (QPSK and 64QAM). Simulation results show that PCCC had better performance than RS-CC, because of the parameters used in the PCCC and iterative decoding algorithm. The PCCC in 64QAM modulation is more effective than QPSK modulation.

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