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Faculté de Technologie

قسم الإلكترونيـك
Département d'Électronique

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Presented by :

KUDOYI OMAR JUMA

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Master's Thesis

Error Detection and Correction techniques in LTE Technology.

Proposed by: Mr. Mahdi BERSALI

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Abstract

In this thesis I look at some of the techniques used for error detection and control in transmission to ensure reliable and accurate reception of the transmitted information. I look at the most common method of error detection or checking, CRC.I also look at the techniques with a retransmission mechanism such as ARQ and HARQ. The in depth description of the technique without a retransmission mechanism is also discussed, FEC which is the main mechanism.

In chapter one I look at ARQ retransmission protocol citing some basic elements of this protocol and the three types of ARQ protocol which are Stop and Wait, Go back N and selective repeat ARQ protocols. Chapter two describes the FEC technique, its classification and decoding techniques. Chapter three I discuss the three types of HARQ protocol in depth. The type I HARQ protocol, type II HARQ protocol and type III HARQ protocol.it also discusses the different schemes used in different technologies with an extensive study of HARQ in LTE.

I also take a look at the history of digital communication systems specifically basing on the 4th generation known as the LTE.A brief scope on the 3 GPP and its releases. I look at the LTE architecture and the radio channels present in this technology.

Key words: ARQ, FEC, HARQ, LTE, CRC, CC-HARQ, IR-HARQ, 3GPP.

Résumé

Dans cette thèse, je regarde certaines techniques utilisées pour la détection et le contrôle des erreurs de transmission pour assurer une réception fiable et précise de l'information transmise. Je regarde la méthode la plus courante de détection d'erreur ou de vérification, CRC, aussi je regarder les techniques avec un mécanisme de retransmission comme ARQ et HARQ. La description approfondie de la technique sans un mécanisme de retransmission est également discutée, FEC qui est le principal mécanisme.

Dans le premier chapitre, je détaille le protocole de retransmission ARQ citant certains éléments de base de ce protocole et les trois types de protocole ARQ qui sont SAW, GBN et SR ARQ protocole. Le chapitre deux décrit la technique FEC, ses techniques de classification et de décodage. Chapitre trois que je discuter les trois types de protocole HARQ en profondeur. Le type I, le type II et le type III HARQ Protocol. Illustré également sont les différents systèmes utilisés dans les différentes technologies avec une étude approfondie de HARQ dans LTE.

Je prends aussi un regard sur l'histoire des systèmes de communication numériques basant spécifiquement sur la 4ème génération appelé LTE. Je parle aussi sur la 3 GPP et ses releases. Je regarde l'architecture LTE et les chaînes de radio présentes dans cette technologie.

Mots clés: ARQ, FEC, HARQ, LTE, CRC, CC, IR, 3GPP, SAW, GBN, ET SR-ARQ.

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Last but not least, a thousand gratitude to Allah for giving me the strength to finish my studies. Alhamdulillah. May Allah bless us, Insha'Allah.

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List of acronyms

2G: Second Generation GSM Mobile Networks

2G: Second Generation GSM Mobile Networks

3 GPP. Third Generation Partnership Project

3G: Third Generation UMTS Mobile Networks

3G: Third Generation UMTS Mobile Networks

4G: Fourth Generation LTE Mobile Networks

4G: Fourth Generation LTE Mobile Networks

ACID: Channel Identifier

ACK: Acknowledgment

ACK: positive acknowledgment

A-MAP IE: Advanced Medium Access Protocol Information Element

AMPS: Advanced Mobile Phone System

ARIB: Association of Radio Industry and Business

ARQ: Automatically request

ATIS: automatic terminal information services

ATM: Asynchronous Transfer Mode

AWGN: Additive White Gaussian Noise

BER: Bit Error Rate

BLER: Block Error Rate

BSC: Base Station Controller

BTS: Base Transceiver Station

CA: Carrier Aggregation

CC: Component Carrier

CCH: Common Channel

CC-HARQ: Chase Combining HARQ.

CFI: control format indicator.

CN: Core Network

CoMP: Coordinated Multi Point transmission

C-Plane: Control Plane

CPRI: Common Public Radio Interface

CQI: Channel Quality Indicator

CRC: Cyclic Redundancy Check

CRC: Cyclic Redundancy Check

CSI: Channel State Information

CSR: Cell Selection / Reselection

CTR: Cell Traffic Recording

DAT: digital audiotape

DL: Down Link

DSA: Dynamic Spectrum Allocation

DSL: Digital Subscriber Line

DSP: Digital Signal Processing

EAC: Extended Alamouti Coding

ECC: error-correcting code

ED: Error-Detection

eNodeB: Evolved NodeB

EPC: Evolved Packet Core

ETSI: European Telecommunication Standards Institute.

FDD: Frequency Division Duplexing

FDD: Frequency Domain Duplex

FPGA: Field Programmable Gate Array

FS: Frequency Synthesizer

FSA: Fixed Spectrum Allocation

GSM: Global System for Mobile Communications.

HARQ: Hybrid Automatic Repeat reQuest

HDSL: High Speed Digital Subscriber Line

HSDPA: High Speed Data Packet Access

HS-DSCH: High Speed Downlink Shared Channel

HS-SCCH: High-Speed Shared Control Channels

IF: Inter-Frequency

IM: Inter-modulation

IMT: International Mobile Telecommunications

IP: Internet Protocol

IQ: In-phase and Quadrature

IR-HARQ: Incremental Redundancy HARQ.

ITU: International Telecommunication Union

ITU-R: International Telecommunication Union – Radio communication sector

LNA: Low Noise Amplifier

LO: Local Oscillator

LSB: Least Significant Bit

LTE: Long Term Evolution

LU: Local Unit

MAC: medium access control.

MGW: Media Gateway

MIMO: Multiple Input Multiple Output

MMSE: Minimum Mean Square Error

MPDU: MAC packet data unit.

MSC: Mobile Switch Center

MSDU: MAC service data unit.

MSS: Mobile Soft Switch

NACK: negative acknowledgment

NACK: Negative Acknowledgment

NAMPS: Narrowband AMPS

NMT-900: Nordic Mobile Telephone System.

PDC: Pacific Digital Cellular

PLMN: Public Land Mobile Network

QPSK: Quadrature Phase Shift Keying

RAM: Read Access Memory

RAN: Radio Access Network

RAT: Radio Access Technology

REG: resource element group

RF: Radio Frequency

RFM: Radio Frequency Module

RFT: Radio Frequency Transceiver

SNR: Signal to Noise Ratio

SPID: Sub-Packet Identifier

SSCOP: Service Specific Connection Oriented Protocol

TACS: Total Access Cellular System

TB: transport block

TCP: Transmission Control Protocol

TDD: Time Domain Duplex.

TTI: Transmission Time Interval

TX: Transmitter

TXB: Transmitter Board

UARFCN: UTRA Absolute Radio Frequency Channel Number

UE: User Equipment

UMTS: Universal Mobile Terrestrial System

URAN: UMTS Radio Access Network

USDC: United States Digital Cellular Standards.

WCDMA: Wideband Code Division Multiple Access

General introduction

Error detection and correction has great practical importance in maintaining data (information) integrity across noisy communication networks channels and less reliable storage media. Error correction sends additional information so that incorrect data can be corrected and accepted. Error correction is the additional ability to reconstruct the original, error-free data.

There are several ways to design the channel code and protocol for an error correcting system, we can cite:

Automatic Repeat-Request (ARQ) where the transmitter sends the data and also an error detection code, which the receiver uses to check for errors, and request retransmission of erroneous data. In many cases, the request is implicit; the receiver sends an acknowledgement (ACK) of correctly received data, and the transmitter re-sends anything not acknowledged within a reasonable period of time.

Forward Error Correction (FEC) where the transmitter encodes the data with an error-correcting code (ECC) and sends the coded message. The receiver never sends any messages back to the transmitter. The receiver decodes what it receives into the most likely data. The codes are designed so that it would take an unreasonable amount of noise to trick the receiver into misinterpreting the data.

HARQ (Hybrid ARQ**)** which is a combination of ARQ and FEC. FEC is not LTE specific technology and a kind of generic error correction mechanism.

Error detection which sends additional information so that incorrect data can be detected and rejected. Error detection is the ability to detect the presence of errors caused by noise or other impairments during transmission from the transmitter to the receiver. Repetition Schemes, Parity Schemes, Checksum, Hamming Distance Based Checks and C R C are examples here.

1.1 Introduction

The major functionality of data link layer protocols is ensuring error-free communications between a transmitter and a receiver. In a data-oriented communication system, user data is fragmented into a series of small frames and sent out by transmitter, the data is received and reassembled at the corresponding receiver.

In this chapter I look at the first error control mechanism with a retransmission criteria. Voice is a service where retransmission does not apply. If a piece of information is lost, and is retransmitted, the conversation becomes intelligible. On the other hand, data services practically rely on retransmission, since most allows a certain tolerance to delays.

I introduce ARQ (Automatic Repeat reQuest) by looking at some types and how they deal with error detection and control. In conclusion part I look at the different technologies that uses this mechanism for error control.

Most redundancy methods can only detect errors, they can tell whether a frame is erroneous or not, but they cannot locate the errors. Due to the fact that a frame may be totally lost, error correction is achieved by the retransmission of erroneous frames, and this retransmission must be done automatically through the interactions between the transmitter and the receiver. Such an automatically achieved method is called Automatic Repeat Request (ARQ).

The data link layer has two main functions, flow control and error control.

1.1.1 Flow Control

In the data communication, flow control manages the transmitted data between sender and receiver. First, data will be transmit to the receiver, an acknowledgment will be sent back to the sender, and then the sender sends the next data. The flow control is responsible for adjusting the amount of the transmitted data. But on the receiving side, the receiver is limited, the incoming data is not allowed to overflow the receiver. The receiver needs time to check and process the incoming data then store them before they can be used. This processing speed is much slower than the data transmission speed. If incoming data almost fills up the receiver buffer memory, the receiver informs the sender to slow down the transmission speed or pause for a while.

1.1.2 Error Control

Error control in data link layer means two things, error detection and error correction. Sometimes, data may be lost or damaged during the transmission, error control will inform the sender, specify the frames and ask for retransmission. This process is called automatic repeat request (ARQ).

1.2 Overview of ARQ protocol

Automatic repeat request (ARQ) uses retransmissions to recover data that is lost due to inevitable errors when transmitting over variable and unreliable channels. ARQ is based on the principle that the receiver can inform the transmitter about the transmission failure, to which the transmitter responds by retransmitting the lost data. ARQ is an extremely powerful type of feedback-based communication that is extensively used at different layers of the network stack. The basic ARQ strategy adheres to the pattern of transmission followed by feedback of an ACK or NACK indicating successful or unsuccessful decoding.

In an ARQ scheme, the receiver uses an error-detecting code, typically a Cyclic Redundancy Check (CRC), to detect if the received packet is in error or not. If no error is detected in the received data packet, the received data is declared error-free and the transmitter is notified by sending a positive acknowledgment (ACK). On the other hand, if an error is detected, the receiver discards the received data and notifies the transmitter via a return channel by sending a negative acknowledgment (NACK). In response to a NACK, the transmitter retransmits the same information.

A retransmission timer is specified using the ARQ_RETRY_TIMEOUT parameter, which is the minimum time interval during which a transmitter waits before the retransmission of an unacknowledged ARQ block. ARQ in the MAC can be enabled (figure 1.1).or disabled (figure 1.2).

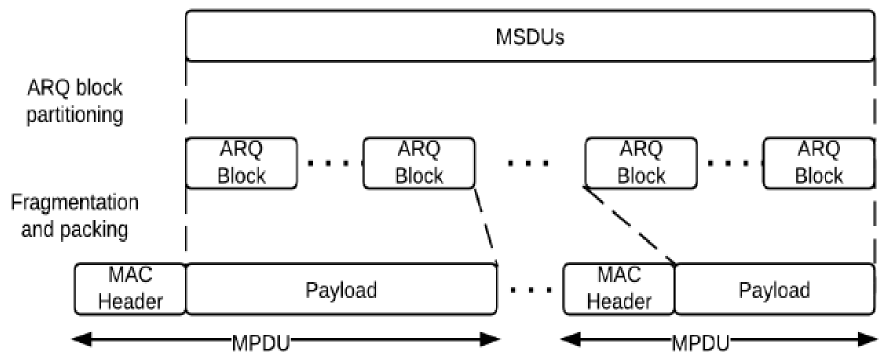

Figure 1.1 MAC layer with ARQ enabled [28]

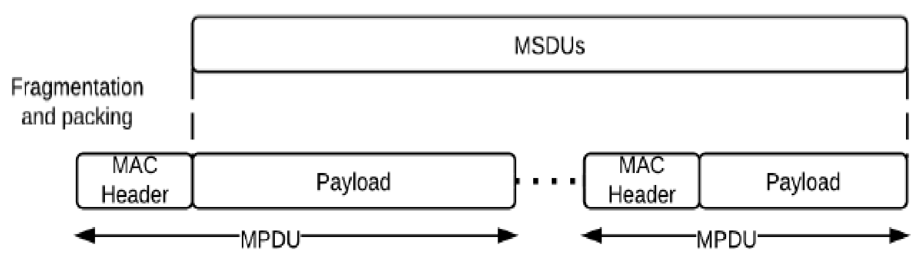

Figure 1.2 MAC layer with ARQ disabled. Ref: [28]

In an ARQ enabled MAC, the MSDUs are first divides into fixed-length ARQ blocks. A BSN (Bloc Sequence Number) is given to each ARQ block. The length of each ARQ block is specified using the ARQ_BLOCK_SIZE parameter. After ARQ blocks partitions, fragmentation and packing are applied, the ARQ blocks are assembled into MPDUs.

In an ARQ disabled MAC, the MSDUs are fragmented and packed into MPDUs without partitioning them into ARQ blocks (figure 1.3). The figures show the ARQ enabled and ARQ disabled MAC.

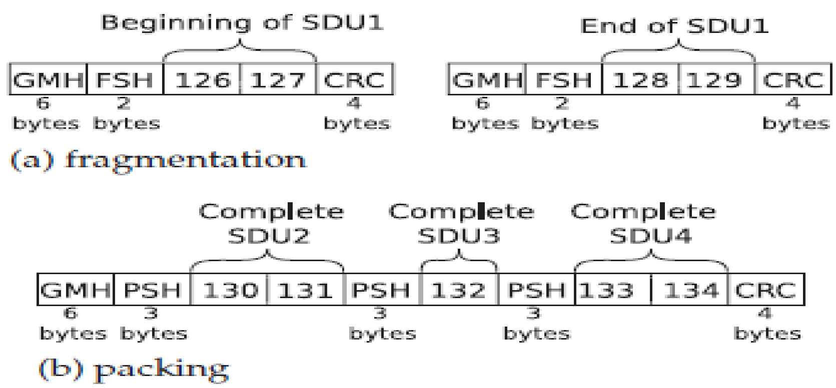

Figure 1.3 ARQ blocks with fragmentation and packing [28]

We can have a slight difference in the ARQ length in 802.16d standard which takes the values from 1 to 2040 bytes. The standard 802.16e takes a double digit value from 16 to 1024 bytes, i.e 16, 18, a power of 2 digits.

1.2.1 Basic elements of ARQ

The elements of the ARQ protocol structure as shown in figure 4 are:

- Information frame (I-Frame): this is used to transfer user packets, control frames, and time-out mechanisms.

- Control fames: they are short binary blocks having a header to provide the control information then followed by a CRC. The control frames includes: ACKs (acknowledge) for a frame well received; the NACKs for erroneous or lost frames.

- Enquiry frame (ENQ): it commands the receiver to report its status of the send frames.

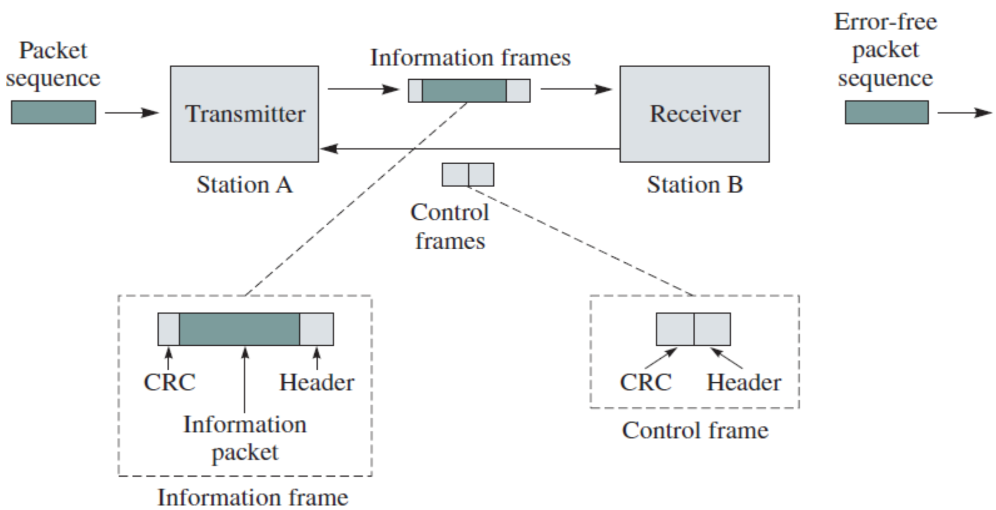

Figure 1.4 elements of ARQ protocols

1.2.2 Types of ARQ schemes

There are different well known ARQ techniques. The use of one or another is defined depending on the kind of data transmitted, the maximum delay supported or the possibility of having buffers in both the transmitter and the receiver. These techniques includes the following:

- Stop-and-wait ARQ
- Go Back N ARQ
- Selective repeat ARQ

A general ARQ scheme can be presented in the figure below:

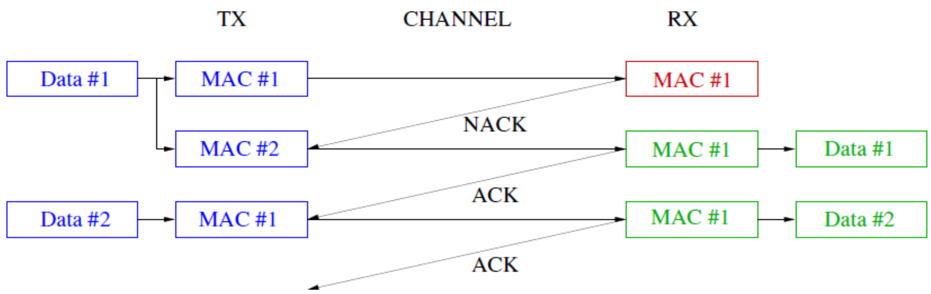

Figure 1.5 general ARQ scheme. Ref [29]

The transmitter of pure ARQ systems encodes the data with the CRC, and then transmits the resulting packet onto the noisy channel. The receiver checks whether the received packet has been corrupted by the channel or not, and replies back to the transmitter with ACK or NACK accordingly. If a NACK occurs, the receiver drops the received packet. In that case, the transmitter receives NACK and retransmits the same packet. Otherwise, the receiver releases the decoded packet and the protocol starts again with the next data packet.

a Stop-and-wait ARQ.

Only one packet can be on the channel at the same time. The transmitter has a one packet size buffer. There is no buffer in the receiver. The transmitter waits for a packet from an upper layer. Once it gets a packet, it is transmitted to the receiver. We have two scenarios. If a packet is received correctly, the ACK (positive acknowledgement) is sent to the transmitter to inform about the successful reception of a packet. On the other hand, if a packet is somehow corrupted or lost, the NACK (negative acknowledgement) is sent to the transmitter to request a retransmission of the lost packet [33].

Sending device sends a frame 0 to the receiving device, keeping a copy of this frame 0. At the same time, the timer starts timing. If this frame 0 is correctly received, a positive ACK 1 will be sends back to the sender. During the transmission, if the frame 0 is lost, the receiving device does not receive anything, so it does nothing [33] If a damaged frame 0 is send to the receiver, the receiver will discard this frame 0 automatically and remain silent. But the sending device is still waiting the ACK 1 back, after the timer times up, the sender assumes that the transmission of frame 0 has failed, so it will retransmit the frame 0 to receiver again. This is the reason why the sender keeps a copy of frame 0. There are control variables for the sender and the receiver, called "S" and "R" variables. S holds the number of the recently sent frame. R holds the number of excepted frame.

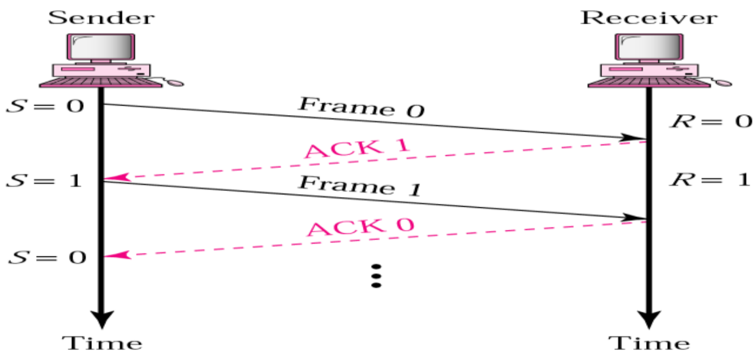

Figure 1.6 Stop-and-wait ARQ normal operation Ref [33]

As we can see in the figure below, frame 1 is lost, but the sender is still expecting the ACK 0 back. After timer time out, frame 1 sends again.

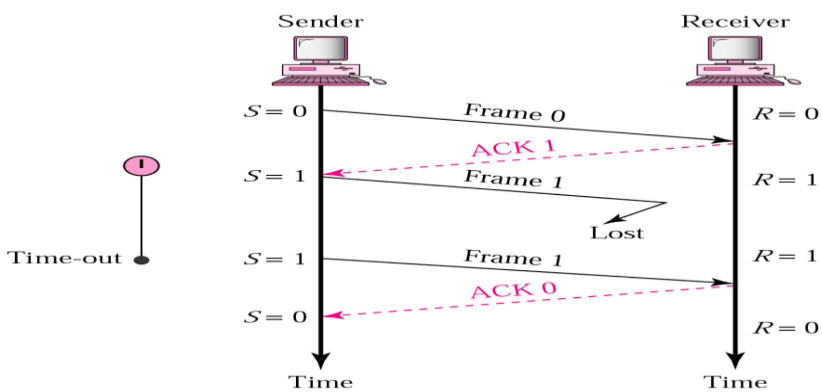

Figure 1.7 Stop-and-wait ARQ lost frame Ref [33]

Now, in the following illustration we see a lost ACK frame. When ACK 0 is sends back to sender, it is lost. So after time out, frame 1 sends again.

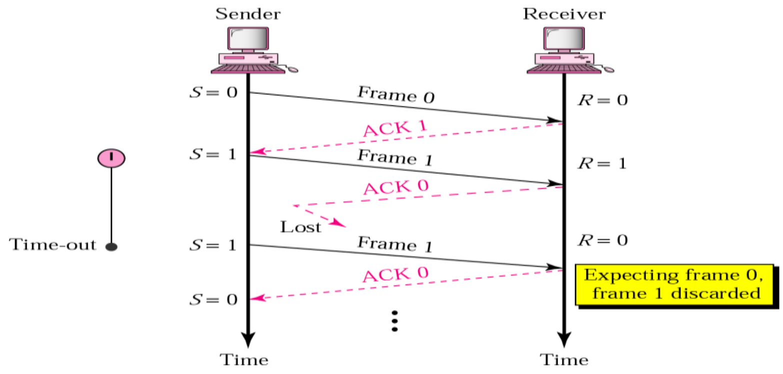

Figure 1.8 Stop-and-wait ARQ lost ACK Ref [33]

We have been looking at unidirectional transmissions. But we could have bidirectional transmission. In that case, both sides are sending and receiving. If the transmissions share the same channel, it is called half-duplex. If they use separate channels, is called full-duplex.
Examples of some applications using stop-and-wait ARQ includes the following:

Bysinc protocol (binary synchronous communication).
Character-oriented data link control that uses the ASCII character set.
Xmodem: A popular file transfer protocol for modems.

b Go-back-N ARQ

Stop-and-wait ARQ protocol is inefficient, because the transmitter is only busy during the transmission time and waits for feedback without doing anything else. Go-Back-N protocol is based on pipelining. However, there is a limit at the number of packets that can be sent until the transmitter changes its mode from transmission to reception. Up to N packets can be travelling through the channel simultaneously. The transmitter has an N packet buffer [33].

To understand well how this protocol works, we need to understand some technical terms used:

- Sequence Numbers: throughout the transmission, there are several frames waiting for transmission, therefore a need to number them sequentially. If the header of frame allows m bits for the sequence number, those frames range will be 0 to 2^m-1. If m=2, sequence number will be 0, 1, 2, 3, 0, 1, 2, 3 repeat in this way. Unlike Stop-and-wait ARQ's 0,1,0,1.

- Sender sliding window: the window size is fixed at 2^m-1. Inside this sliding window, there are copies of the transmission frames. When the correct ACK arrived, the sliding window slid forward.

- Receiver sliding window: Receiver sliding window in Go-Back-N ARQ is always 1. It is always waiting for the correct frame to come in the correct order, then it sends back the ACK and slides forward. If the frame is lost or damaged, the receiver will wait for the resend.

Control variables: in the Go-Back-N ARQ, sender's control variables are S, S_F,S_L. the receiver's variable is R. the Slide window size is W. S is the sequence number of the latest sent frame, S_F is the sequence number of the first frame in the slide window, S_L is the sequence number of the last frame in the slide window. R is the sequence number of the accepted frame [33].

$$W = SL - SF + 1 = 2m - 1 \quad (1\text{-}1)$$

The frame is accept only when R and the sequence number of the received frame are the same, otherwise it is discarded.

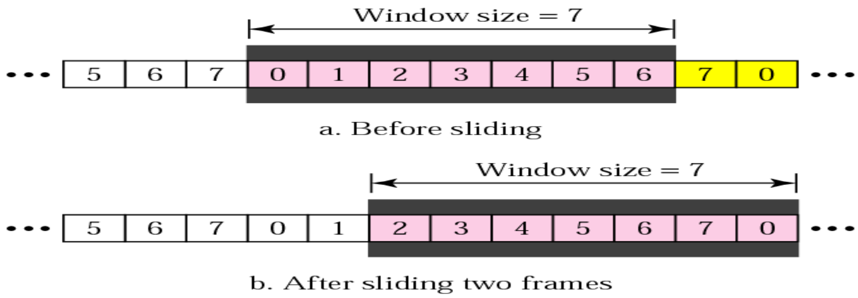

Figure 1.9 Go-Back-N ARQ sender sliding window Ref [33]

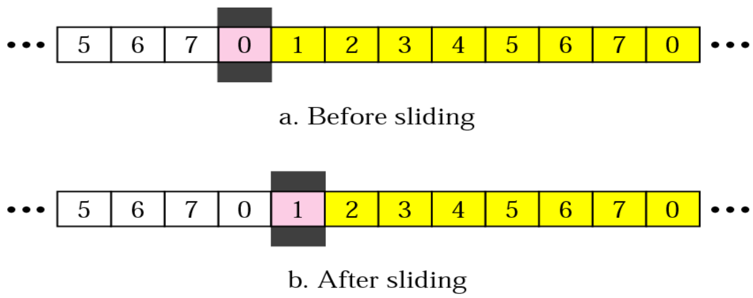

Figure 1.10 Go-Back-N ARQ receiver sliding window Ref [33]

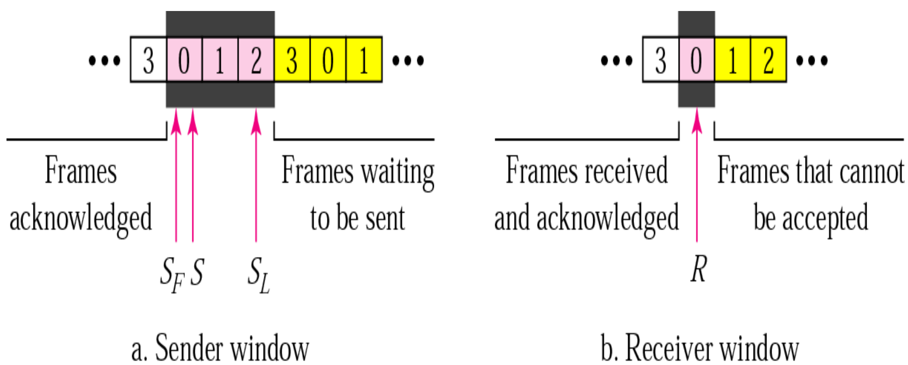

Figure 1.11 Go-Back-N ARQ control variable Ref [33]

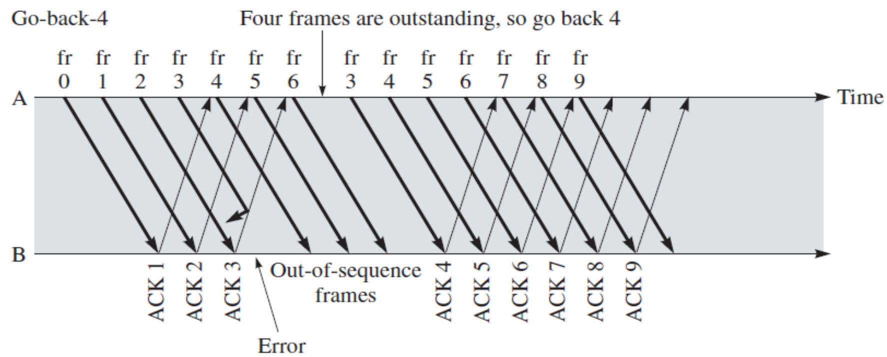

Figure 1.12 Go-Back-N ARQ protocol

This protocol gets its name from the action it takes when an error occurs.as seen in the figure above, the error occurred in frame 3, so the receiver ignores that frame and all the subsequent frames. When the transmitter reaches the maximum number of outstanding frames, it is forced to go back N frames and begin retransmission from frame 3.

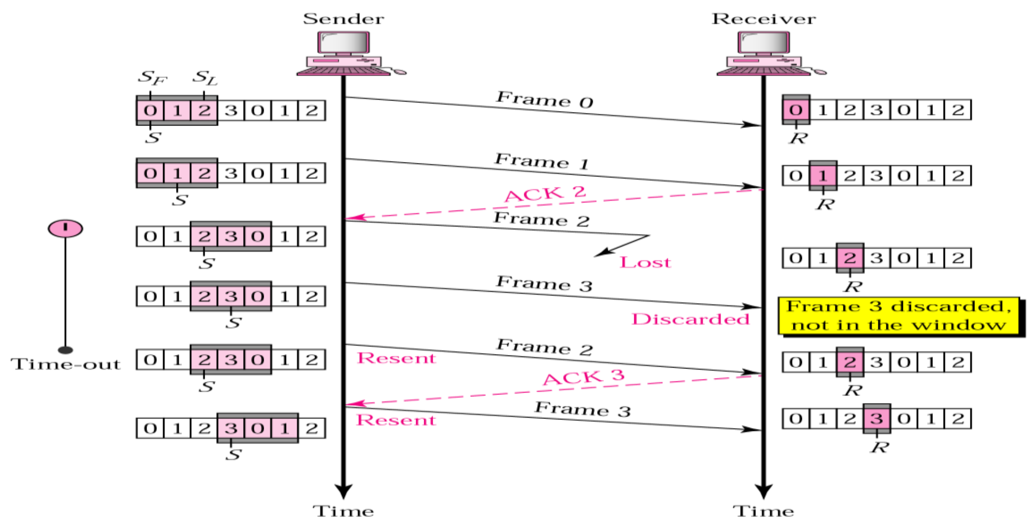

Figure 1. 13 Go-Back-N ARQ lost frame Ref [33]

c Selective Repeat ARQ protocol.

In channels with high error rate, the Go-Back-N ARQ protocol is not effective. Thus there is need of improvement. We add two more features on this protocol to make it more effective in channels with high error rate. First the receive window is enlarged so as to accept out of order bur error free frames. Then secondly, we modify the retransmission mechanism so as to transmit only individual frames. This protocol is referred to as selective repeat ARQ.

In this advanced version protocol, there is no need to resend N frames, only the lost or damaged frame are resend. The slide window size is given by 2^{m-1}. Receiver has 2 control variables, R_F and R_L.

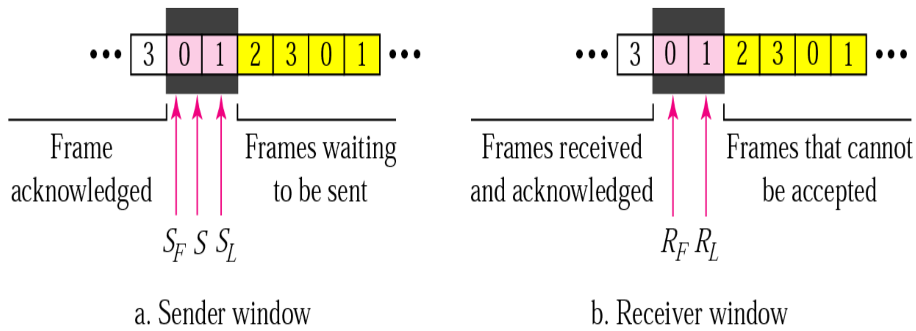

Figure 1.14 Selective Repeat ARQ receiver and sender slide window Ref [33]

NACK means negative acknowledgment, it does only exist in Selective Repeat ARQ. We have some examples of protocols that use selective repeat ARQ protocol which includes the following:

TCP: it's used over internets that use IP to transport packets in connectionless mode. SSCOP: it provides error control messages in ATM networks.

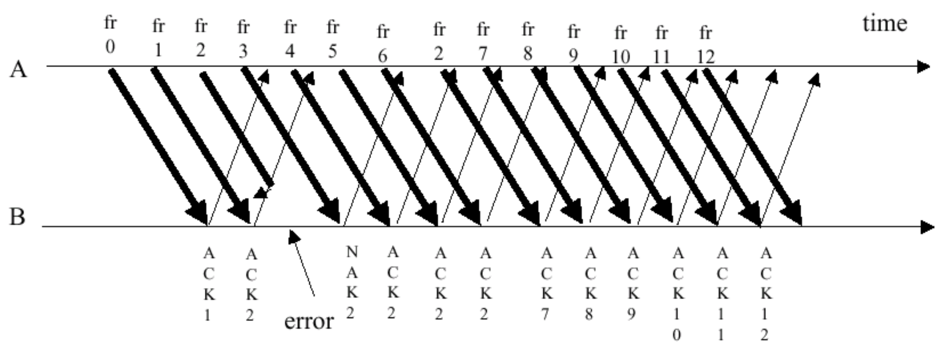

Figure 1.15 selective repeat ARQ protocol

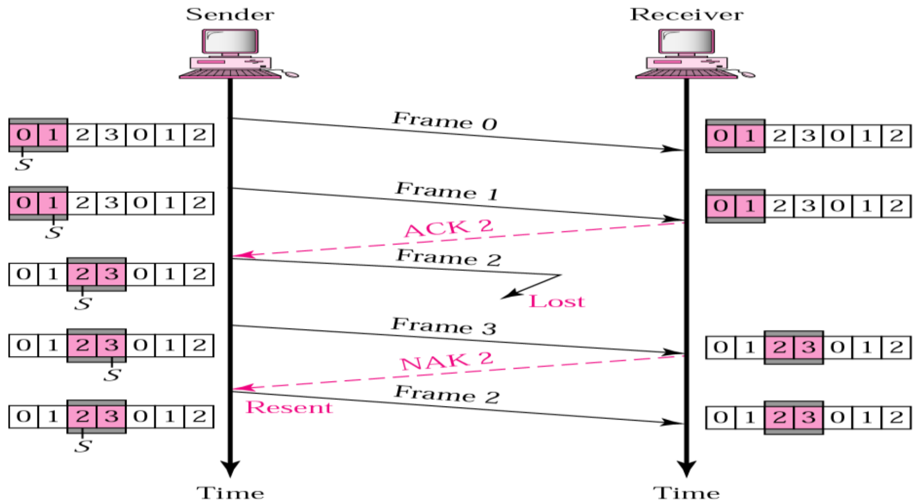

Figure 1.16 Selective Repeat ARQ lost frame Ref [33]

We can look at the comparison in the performance of the three ARQ protocols as seen in the figure below.

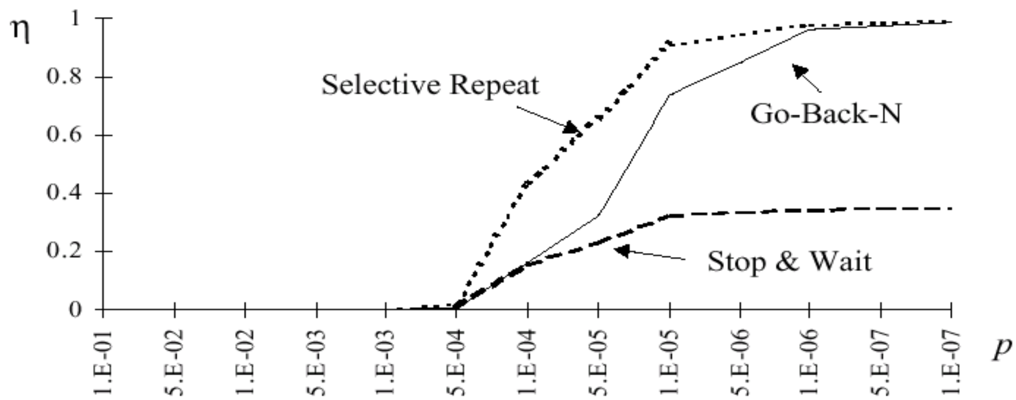

Figure 1.17 transmission efficiency in ARQ protocols

For comparison purposes we consider the case of a 1024 byte frame in the presence of bit errors occurring at a probability *P*. We take a 1.5Mbps channel over a moderate distance link, Go-Back-N and selective repeat ARQ use a window size of 4. It is clear that Stop-and-Wait ARQ cannot achieve the efficiency higher than 0.35 at any given value of *P*.

Go-Back-N ARQ achieves efficiency comparable to that of Selective Repeat ARQ for P less than 10^{-6} and then drops to the performance of Stop-and-Wait ARQ protocol.

Selective repeat ARQ achieves high efficiency over a greater range of *P* but also drops as *P* becomes larger than 10^{-4}. We can summarize in the table below.

	Stop-and-Wait	Go-back-N	Selective Repeat
Sender slide window size.	1	2^m-1	2^{m-1}
Receiver slide window size.	1	1	2^{m-1}
ACK	YES	YES	YES
NAK	NO	NO	YES
Frame sequence	0,1,0,1...	0...2^m-1	0...2^m-1
Bandwidth utilization	Low	Medium	High

Table 1 three ARQ protocols comparing table

1.3 Conclusion

It is not simple to say which protocol is better or faster than the other. It depends on the environment and all the parameters. Choose the proper protocol for suitable situation. The ARQ block size should be the governing parameter, while the ARQ transmission window size should be adapted. The reason is that the ARQ block size has a set of discrete values, while the ARQ transmission window can accept any value within the specified rearrangement range

Now we look at some technologies that use ARQ protocol.

TECHNOLOGY	RANGE	FREQUENCY BANDE.	CHANNEL BANDWIDTH	MOBILITY	ARQ
Wireless Wide Area Networks (WWAN).					
GSM (2G)	3-35KM	900-1800Mhz (TDMA)	200Khz (TDMA). 1.23Mhz (CDMA).	Simless global roaming.	YES
GPRS (2.5G)		800-1900Mhz (CDMA)	200Khz.		YES
EDGE (2.75G)					YES
3G		1900-2025Mhz 2110-2200Mhz	5Mhz		YES
3G LTE			1.25, 2.5, 5, 10, 20 MHz		YES
Wireless Metropolitan Area Network (WMAN).					
IEEE 802.16 (WiMAX)	Up to 50 km	11 – 66 GHz	20, 25, 28 MHz	fixed	NO
IEEE 802.16a		2 – 11 GHz	1.75 – 20 MHz	fixed	YES

Table 2 The applications of ARQ

2.1 Introduction.

Forward error correction (FEC) or channel coding is a technique used for controlling errors in data transmission over unreliable or noisy communication channels. The central idea is the sender encodes his message in a redundant way by using an error-correcting code (ECC). In the simplest terms, FEC involves injecting redundant repair traffic into a data path to enable the recovery of data lost in transit. Data rate to be transmitted is increased. FEC is not restricted to communications only but can also be found in storage applications like compact disks, digital versatile disks, digital audiotape (DAT) tapes and hard disks in personal computers.

FEC eliminates the feedback loop between the receiver and the sender in a communication channel. This key property makes FEC ideally suited for settings where the receiver cannot reply back to the sender like over long-distance links where the receiver's feedback takes too long to arrive at the sender, or in multicast channels where multiple receivers can swamp a sender with feedback.

2.2 Overview of FEC codes

Communication systems that use the FEC don't request a repetition of the transmission of coded information. The source information generates a binary signal representing equally likely symbols at a rate rb. The encoder takes a group of k message bits, and adds to it $n - k$ parity check bits. This is the encoding procedure for a linear block code $Cb\ (n, k)$ whose code rate is $Rc = k/n,$ with $Rc < 1$. The transmission rate r must be higher than the source information rate rb.

$$r = (n/k)rb = rb/Rc \qquad (2\text{-}1)$$

d_{min} for FEC codes is 2t+1.

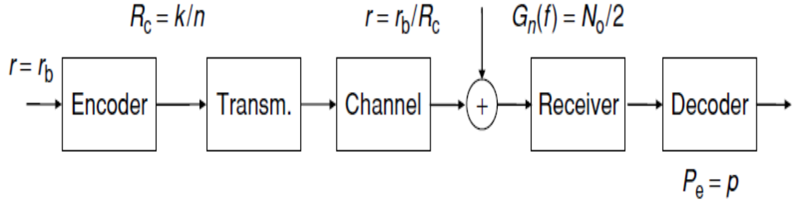

Figure 2.1 transmission bloc with FEC. Ref [30]

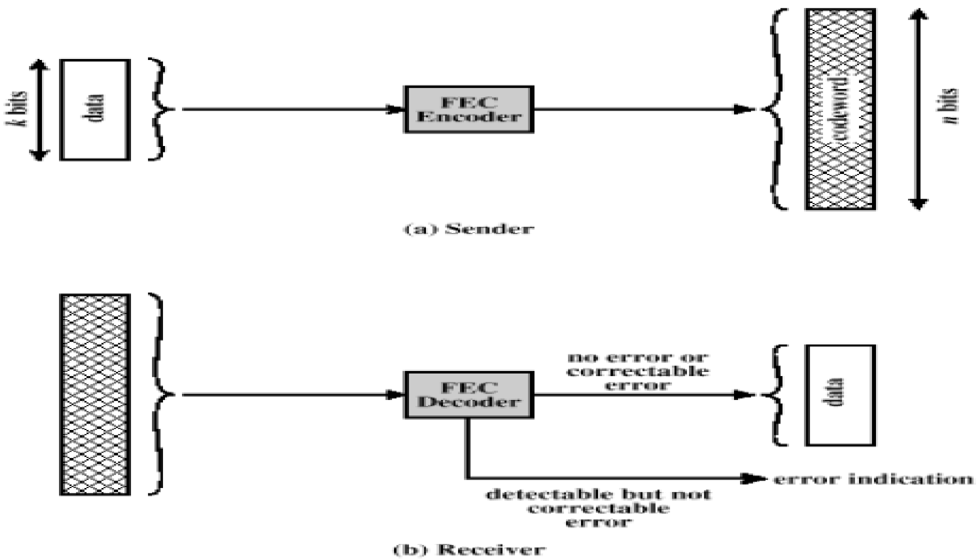

Figure 2.2 FEC process ref [31]

2.3 Classification of FEC codes.

In the figure below is a summary of all the FEC codes that exists.

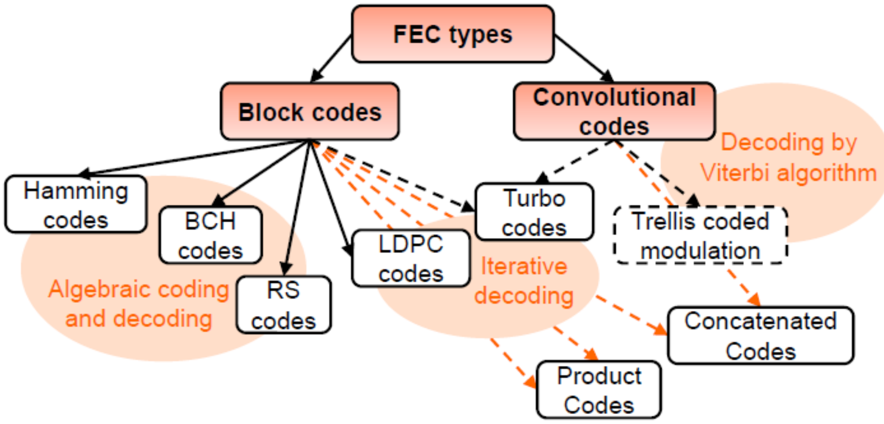

Figure 2.3 FEC codes classification

All FEC codes can be classified as either block or convolutional codes. A block code operates on discrete chunks or blocks of data, from *k* input symbols, it generates *n* output symbols. The decoder can reconstruct the original *k* input symbols from any *k* of the total *n* output symbols. As a result, the block code is resilient to a maximum of *n-k* losses. Note that a symbol could be a single bit or a large set of bits. Examples of block codes include Hamming Codes and Cyclic Codes. A Cyclic Redundancy Check (CRC) code can detect any error burst up to the length of the CRC code itself.

A convolutional code operates on a stream of data. The function used by the encoder to generate n bits of output from m bits of input depends on the last k bits of input. The value k is called the constraint length and the value m/n is the rate of the code. Concatenated codes involve wrapping one code within another; the inner code corrects most of the errors while the outer code mops up the remaining errors. Both block and convolutional codes can be either systematic or nonsystematic. If the k information bits are transmitted unaltered first followed by the transmission of the r check bits it is called a systematic code. A non-systematic block code is one which has the check bits interspersed between the information bits [5].

2.3.1 Block codes

The block coder input is a stream of information bits.

The coder segments this bit stream into blocks of k information bits and for each block it calculates a number of r check bits, or it picks the r check bits from a tabulated set of values. It then transmits the entire block, or code word of n = k + r channel bits. This is called an (n, k) block code.

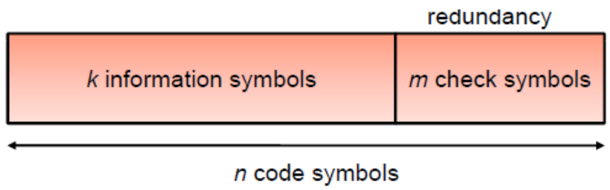

Figure 2.4 block code (ref [3])

Features of a block code.

Code rate: R=k/n.

Code identification: (n, k, $dmin$).

Code capabilities of a linear block code with $dmin$:

- Up to $dmin$ - 1 errors can be detected
- Up to [($dmin$ – 1)/2] errors can be corrected

2.3.2 Convolutional codes

This code generates redundant bits continuously. Error checking and correcting is carried out continuously. The minimum Hamming distance of convolutional codes is termed *free distance* and is denoted df. The minimum free distance of a convolutional code is defined as the minimum Hamming distance between any two codes sequences, which is the minimum weight of all non-zero code sequences of any length. If a soft-decision decoding algorithm is used, the code performance is measured by Euclidean distance. The minimum free euclidean distance is defined as the minimum Euclidean distance between any two code sequences. It depends on both the convolutional code trellis and modulation type.

The minimum number of positions in which any two code-words in any particular block differ from each other is called the Hamming distance $dmin$. A convolutional code with parameters n,

k and K will be denoted as $Cconv\ (n, k, K)$. Input processes k bits at a time. Output produces n bits for every k input bits. K = constraint factor. The constraint length parameter, K, denotes the length of the convolutional encoder, i.e. how many k-bit stages are available to feed the combinatorial logic that produces the output symbols k and n generally very small.

Decoding is done by the Viterbi algorithm (VA): soft decision decoding estimates the most likely sequence of encoder register states based on the channel observation. Convolution codes are well suited for the correction of single bit errors.

a Convolution encoder

Convolutional encoding with Viterbi decoding is a FEC technique that is particularly suited to a channel in which mainly Additive White Gaussian Noise (AWGN) corrupts the transmitted signal. Convolutional codes are suitable when the information symbols to be transmitted arrive serially in long sequences rather than in blocks. To provide the extra bits required for error control, an output rate greater than the message bit is achieved by connecting two or more mod-2 adders to the shift registers.

A binary convolutional code of rate $1/k$ bits per symbol can be generated by a linear finite-state machine consisting of an L-stage shift register, k modulo-2 adders connected to some of the shift registers, and a commutator that scans the output of the modulo-2 adders. The whole system is called a convolutional encoder. The error-correcting capability of a convolutional encoder increases as rate r decreases. However, the channel bandwidth and decoder complexity both increase with K.

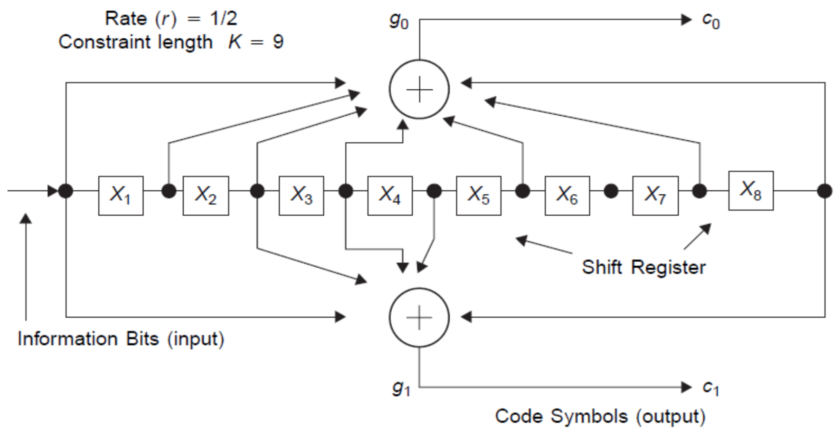

Figure 2.5 convolutional encoder. (Ref [14])

Input raw binary data is encoded by the binary convolutional encoder, with a rate of 1/2. Convolutional encoding the data is accomplished using a shift register and associated combinatorial logic that performs modulo-two addition.

A shift register is a chain of flip-flops wherein the output of the nth flip-flop is tied to the input of the $(n + 1)$ th flip-flop. Every time the active edge of the clock occurs, the input to the flip-flop is clocked through to the output, and thus the data are shifted over one stage. The following steps are followed while designing convolutional encoder.

Initialize the Memory Registers with zeros on reset. Store the incoming bit in memory register m_in. After the input bit has arrived and data in is valid the operation starts and the output is calculated as:

$$x1 = m_{in} + m2 + m4. \qquad (2.2)$$

$$X2 = m\_in + m1 + m3 + m4. \qquad (2.3)$$

$$X3 = m_{in} + m1 + m2 + m3 + m4. \qquad (2.4)$$

Perform shifting operation

$$m4 = m3; \quad m3 = m2; \quad m2 = m1; \quad m1 = m\_in.$$

b Properties of convolution encoder. [15]

The encoder has finite memory. The main method of identifying a convolutional code is to specify its trellis diagram. The trellis diagram is a way to show the transition between various states as the time evolves. The trellis diagram is obtained by specifying all states on a vertical axis and repeating this vertical axis along the time axis. Then, each transition from a state to another state is denoted by a line connecting the two states on two adjacent vertical axes. In a sense, the trellis diagram is nothing but a repetition of the state transition diagram along the time axes.

2.4 Error Detection via CRC [7]

Cyclic Redundancy Check (CRC) spreads the dependency more uniformly across check digits. Packet structure for error detection has the following format:

1. Header: information about who the data is meant for, what purpose data is being sent, etc.

2. Payload: data that we need to communicate. May be of various lengths.

3. Footer: where the CRC is included. The CRC is always the same length, regardless of the payload length. The transmitter calculates the cyclic redundancy check (CRC) from the header and payload and puts it in the footer.

The receiver, once it has all of the header and footer, calculates the CRC based on the received header and footer, and compares it to the one it received. If they match, the receiver decides that the data was received correctly.

2.4.1 Generation of the CRC

A y-bit CRC is described by a y-order polynomial. The y refers to the maximum exponent in the CRC polynomial. $C(x) = x + 1$, for a first order polynomial. The data we want to send, d, can also be described as a polynomial.

$$D\ (x) = 0x3 + 1x2 + 0x + 1x0 = x2 + 1. \qquad (2.5)$$

for d= [0 1 0 1].

To find the CRC bit, we divide the two polynomials, $d(x) \div c(x)\ (modulo - 2)$ and the CRC bit is the remainder. So in fact the CRC is $d(x)\ mod\ c(x)$.

2.4.2 CRC properties

- If the polynomial $P(x)$ has both first and last terms non zero it will not divide a polynomial of the form $E(x) = xi$. Such a CRC will thus detect all single bit errors.

- If a polynomial P(x) has a prime factor with three terms it will not divide a polynomial of the form $E(x) = xi\ (1 + xj)$. Such a CRC will thus detect all double bit errors.

- If $P(x)$ also has a factor $(x + 1)$ it will not divide a $E(x)$ polynomial with an odd number of terms.

Flow diagram for coding procedure for packet transmission.

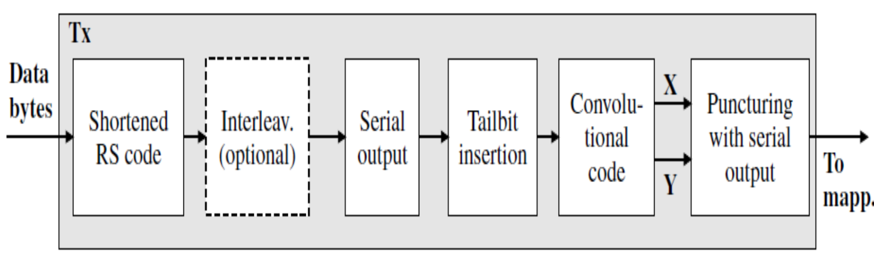

Figure 2.6 coding procedure for packet transmission. (Ref [2])

More information can be found in [4].on other coding schemes used and the mathematical aspect of each code presented here in.

2.5 Turbo Coding

Turbo codes are a special class of concatenated codes where there exists an interleaver between two parallel or serial encoders. The existence of the interleaver results in very large code word lengths with excellent performance, particularly at low SNR. When we apply a systematic recursive convolutional code in an iterative scheme and introduce an interleaver between the two parallel/serial encoders we obtain a convolutional turbo code.

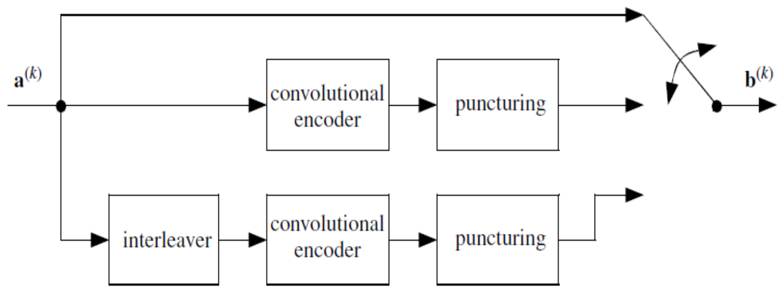

Figure 2.7 Convolution Turbo encoder (Ref [2])

The code structure consists of two parallel recursive systematic punctured convolutional codes. A block of encoded bits consists of three parts. The two parity bit parts and the systematic part which is the same in both code bit streams and, hence, has to be transmitted only once. The code bit sequence at the output of the Turbo encoder is given by the vector b $^{(k)}$.

At the receiver, the decoding is performed iteratively. The component decoders are soft output decoders providing log-likelihood ratios (LLRs) of the decoded bits. The basic idea of iterative decoding is to feed forward/backward the soft decoder output in the form of LLRs, improving the next decoding step. In the initial stage, the non-interleaved part of the coded bits $\mathbf{b}^{(k)}$ is decoded. Only the LLRs given by the vector $\mathbf{l}^{(k)}$ at the input of the Turbo decoder are used.

In the second stage, the interleaved part is decoded. In addition to the LLRs given by $\mathbf{l}^{(k)}$, the decoder uses the output of the first decoding step as a priori information about the coded bits. This is possible due to the separation of the two codes by the interleaver.

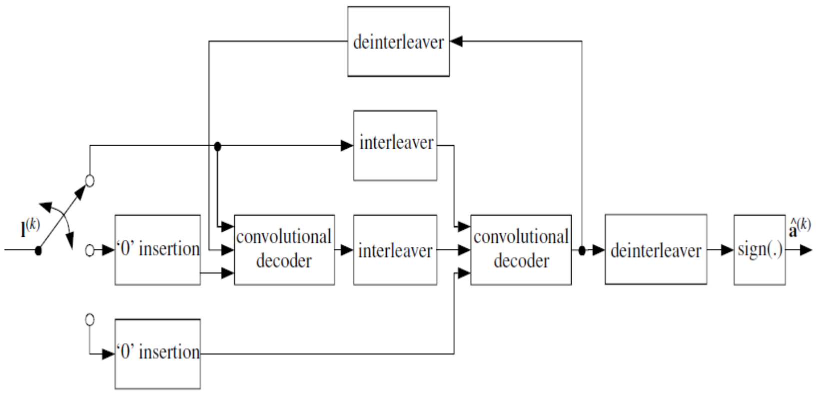

Figure 2.8 Convolutional Turbo decoder. (Ref [2])

2.6 Soft and Hard Decision Decoding

The decoding of a block code can be performed with hard or soft decision input, and the decoder may output hard or soft decision data. In hard decision decoding, each received channel bit is assigned a value of 1 or 0 at the demodulator depending whether the received noisy data is higher or lower than a threshold. The decoder then uses the redundancy added by the encoder to determine if there are errors, and, if possible, to correct the errors. The desired output of the decoder is a corrected code word.

A soft decision decoder receives not only the binary value of 1 or 0, but also a confidence value associated with the given bit. If the demodulator is certain that the bit is a 1, it places very high confidence on it. If it is less certain, it places a lower confidence value. A soft input decoder can output either hard decision data or soft decision data. Example Viterbi decoder.

Chapitre 3 Hybrid automatic repeat request.

3.1 Introduction.

The two categories of techniques for controlling transmission errors in data communication systems discussed in previous chapters have disadvantages. For example, in a FEC communication system, The received code word needs to be decoded even if it has errors, and the decoded message has to be delivered to the user, regardless of whether it is correct or not. To obtain high system reliability, a long powerful code must be used and a large collection of error patterns must be corrected. This makes decoding difficult to implement and expensive. In an ARQ communication system, the throughput is not constant it falls rapidly with increasing channel error rate, this is the primary weakness of the ARQ scheme. Drawbacks of FEC and ARQ schemes can be overcome if the two basic error control schemes are properly combined.

Hybrid Automatic Repeat Request (HARQ) mechanism is an enhancement to the ARQ mechanism. In the standard ARQ mechanism, only error-detection(ED) information is added to data which has to be transmitted. In HARQ, it adds the existing ED bits as well as the Forward Error Correction (FEC) bits. So HARQ performs better than the ARQ in the poor signal conditions. The term Hybrid indicates that both FEC and ARQ are used.

If the packet is received without errors, then the behavior is similar to that of ARQ, however, if the packet contains errors, instead of discarding the packet as with ARQ it is kept in the receiver's buffer. Then the sender will send the packet again, possibly with additional FEC information and the receiver can combine the information of all transmissions. Thus the probability of successful decoding increases with every retransmission.

HARQ uses FEC to correct a subset of errors at the receiver and rely on error detection to detect the remaining errors. Most practical HARQ schemes uses CRC codes for error detection and convolutional or Turbo codes for error correction.

There are two types of HARQ namely Type I and Type II HARQ mechanisms. The simplest version is Type I HARQ, which adds both the ED bits as well as FEC to each message before transmission. Type II HARQ either transmits the ED bits or only the FEC information. HARQ can be used either in selective repeat mode or in stop and wait mode.

HARQ is a retransmission technique to improve packet reception. HARQ spans both MAC and physical layer. The combining bits process is performed by the physical layer. They are widely used in most of the contemporary communication systems, as the delay tolerance of many data services allows using retransmission to recover erroneous packets.

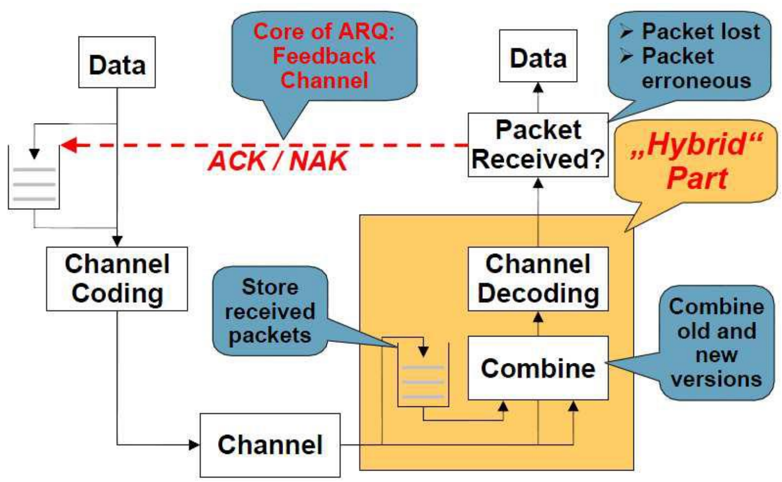

Figure 3.1 HARQ scheme protocol. Ref: [32]

On receiving a packet, the protocol checks if the packet has been sent for the first time or not. In case there are previously stored transmissions of the same packets, they are combined. The resultant packet is sent to the decoder, which tries to recover the information. If the packet arrives for the first time, it is directly sent to the decoder. A copy of the packets is stored in the buffer, which is erased once the decoding process of the packet is successful.

3.2 Types of HARQ schemes

The HARQ scheme can be classified into three categories:

- Type-I HARQ (repetition coding) scheme.
- Type-II HARQ scheme.
- Type-III HARQ (a modified HARQ II) scheme.

3.2.1 Type-I HARQ (repetition coding) scheme

Type-I HARQ describes an ARQ mechanism for which the data packet, after CRC encoding, is encoded by a FEC code of given rate R. Simultaneous error-correction and error detection is used. Type-I HARQ protocol is then equivalent to the truncated ARQ protocol except that Type-I HARQ protocol uses FEC to correct errors.

The throughput of the Type-I HARQ protocol is equal to the throughput of the same FEC code used without ARQ. However, the Type-I HARQ protocol exploits the ARQ to improve the reliability of its transmission compared to pure FEC. Since the code must be able to both detect and correct errors, more parity-check bits are required. This increases the overhead associated with transmissions. Because of this, type-I HARQ schemes result in a lower throughput than the ARQ schemes if the channel conditions are good. However, if the channel conditions are not good and the error correcting capability of the FEC code is good, a type-I HARQ scheme can achieve a higher throughput than an ARQ scheme.

The main interest of Type-I HARQ is to use the correction capability of the FEC in order to recover the information bits in more noisy conditions, and to decrease the retransmission probability of the underlying ARQ.

3.2.2 Type-II HARQ scheme.

The main setback of a Type-I HARQ protocol is that it does not benefit from the different retransmissions. All retransmissions are independent and independently processed at the receiver, making the Type-I HARQ protocol highly inefficient.

In type-II HARQ schemes, the receiver combines the information from different retransmissions to decode the packet. Type-II HARQ gives a satisfying solution to the drawback of Type-I schemes by introducing memory and processing at the receiver.

The main difference between Type-II and Type-I schemes is that Type-II performs combining of the multiple packets received within each ARQ transmission, which allows to increase the correction capability of the code (hence more powerful coding gains). Type-II HARQ thus automatically adapts the code rate to the current channel conditions.

Type-II HARQ method improves the throughput by using the information contained in the first retransmissions to decrease the probability of error in the next retransmissions. These schemes are sometimes called HARQ schemes with soft combining. They do not discard data from a previous attempt received in error (detected but not corrected). They combine the packet received in previous transmission with that of later transmissions to decode the overall packet.

Hybrid ARQ with soft combining is therefore usually categorized into Chase combining and incremental redundancy, depending on whether the retransmitted bits are required to be identical to the original transmission or not.

a Chase Combining (CC) Schemes

These are type-II HARQ schemes in which all the retransmissions carry the same information. The packet is encoded using the same channel code during each transmission and the transmitted signal contains the same coded bits. The blocks of data, along with the CRC code, are encoded using FEC encoder before transmission.

The retransmissions consist of the same set of coded bits as the original transmission. If the receiver is unable to correctly decode the data block, a re-transmission is requested. After each retransmission, the receiver combines each received bit with any previous transmissions of the same bit, and the combined signal is sent to the decoder. Since each retransmission is an identical copy of the original transmission, Chase Combining does not give any additional coding gain but only increases the accumulated received Eb/No i.e. the energy per information bit divided by the noise spectral power density, after each retransmission, improving the likelihood of correct decoding.

In chase combining HARQ, the redundancy version of the encoded bits is not changed from one transmission to the next; therefore, the puncturing pattern remains the same. The code rate remains always the same. The receiver uses the current and all previous HARQ transmissions of the data block in order to decode the information bits. The process continues until either the information bits are correctly decoded and pass the CRC test, or the maximum number of HARQ retransmissions is reached. When the maximum number of retransmissions is reached, the MAC layer resets the process and continues with fresh transmission of the same data block.

The receiver uses maximum-ratio-combining (MRC) to decode the data packet. The combining is usually done after the demodulation but before the decoding.

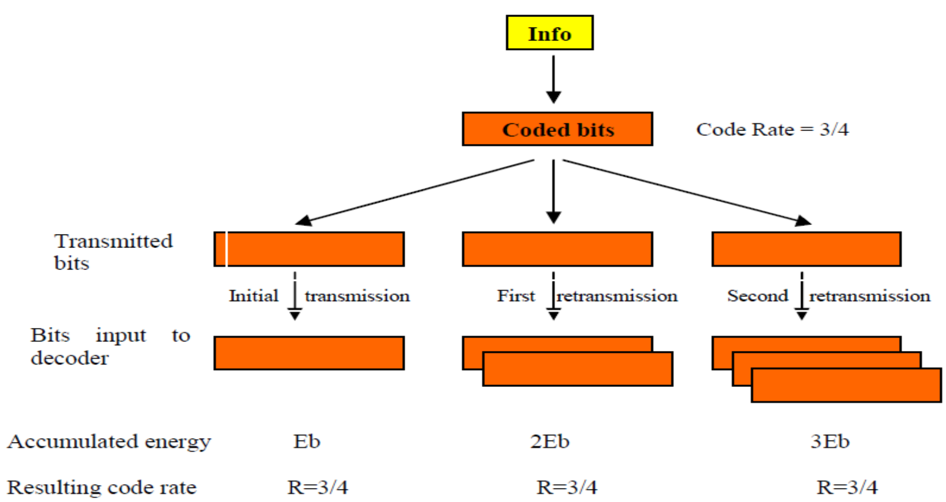

Figure 3.2 Example Chase Combining HARQ [32]

The first step is to perform initial encoding of the information bits with a mother code, which is normally a low-rate code. This mother code is usually a convolutional code, in the case of LTE a Turbo Code is used. In the next step, the bits to be transmitted are obtained from the encoded sequence by either puncturing or repeating it to match the desired code rate.

Mr. Chase developed a practical and effective combining approach, known as Chase combining, for overcoming the problem of obtaining reliable communications when the actual channel capacity is unknown. When combining, packets should be weighed according to their relative reliability. Such a combining strategy can operate in a very high-error environment to achieve error-free results for all channels with finite capacity.

Chase combining has advantages over the other HARQ methods which include the following:
Smaller decoder complexity, smaller memory requirements, and the ability to self-decode every block before joint decoding.

b Incremental redundancy (IR) schemes

The IR-HARQ protocol is different from the Type-I HARQ and the CC-HARQ protocols in the sense that transmitter does not send the same packet over the channel. In an Incremental Redundancy (IR) HARQ scheme, a number of coded bits with increasing redundancy, where each represents the same set of input bits, are generated and transmitted to the receiver when a re-transmission is requested, to assist the receiver with the decoding of the information bits. The different redundancy versions are created by puncturing the code word.

The puncturing process consists of choosing a series of bits from the original packet following a concrete pattern. It is used to try to send the same information but using a higher code rate.

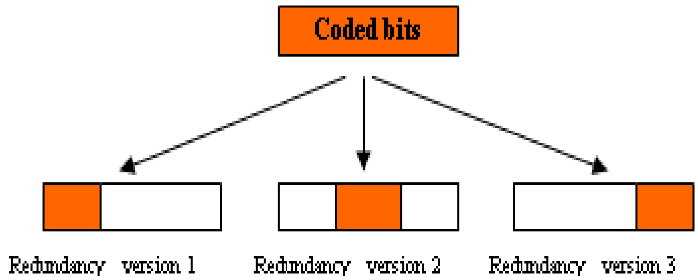

Figure 3.4 creation of the redundancy versions [32]

The receiver combines the retransmission with previous transmission attempts of the same packet. In this technique each re-transmission carries additional parity bits thus the effective code rate is lowered by each re-transmission. The energy per bit only increases in case some bits are transmitted more than once. We have different techniques used to decide which and how many bits will be included in each retransmission.

HARQ IR schemes which do not care about link adaptation: these HARQ protocols rely on the scheduler to correctly decide the code rate that should be used for every transmission depending on the channel prediction. Therefore, the HARQ process just has to worry about which bits send to the scheduler for the next attempt in case a retransmission is required.

HARQ IR schemes with a channel sensing functionality, which adapts the code rate of the next retransmission based on the predicted SINR for the next attempt or the difficulty found to decode the previous packet.

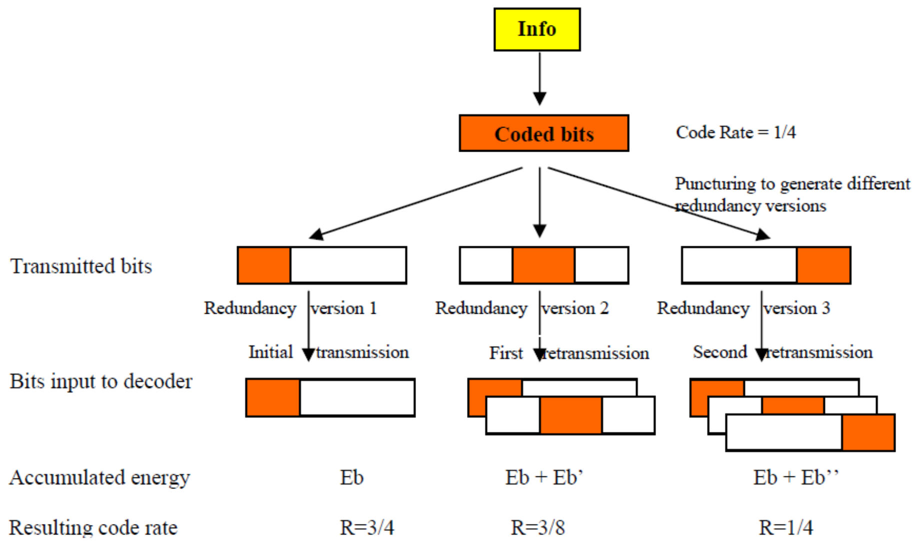

Figure 3.5 Incremental redundancy HARQ scheme [32]

In the first transmission only a limited number of the coded bits are transmitted, effectively leading to a high code rate. In the retransmissions, additional coded bits are transmitted, thus the resultant code rate decreases, providing a coding gain.

The incremental redundancy HARQ was an optional feature in Release 10 of mobile WiMAX systems however, it has been made the mandatory HARQ scheme for IEEE 802.16m systems. It is assumed that the receiver has received all previously transmitted redundancy and that each re-transmission provides some amount of information about the data packet. The puncturing pattern to be used for a given HARQ transmission is indicated by the Sub-Packet Identifier (SPID). By default, the SPID of the first transmission is always zero where all, for instance, turbo code systematic bits and some parity bits are sent. Note that only the parity bits are punctured, and each transmission by itself decodable. The SPID of the subsequent transmissions can be arbitrarily chosen by the system. It must be noted that if the initial transmission with SPID of 0 is not properly received or lost, the re-transmission of only additional parity bits will not typically help decoding of the data packet, and a fresh transmission of the systematic bits might be necessary.

3.2.3 Type-III HARQ schemes

This scheme is similar to IR, and uses puncturing techniques to create the set of bits. However, each packet is self-decodable. This characteristic is achieved by sending the systematic bits in every retransmission required [32].

In a type III HARQ scheme, both user data and complementary parity bits are included in every retransmission. The type III HARQ schemes are less efficient than incremental redundancy schemes due to repeated user data bits. Also in this case, after every retransmission, a richer set of parity bits is available at the receiver, improving the probability of reliable decoding.

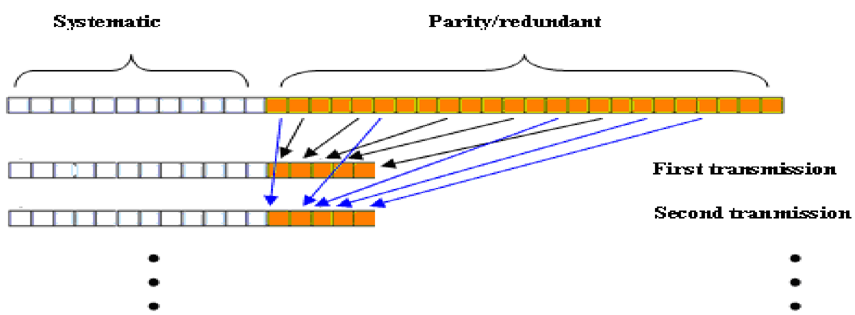

Figure 3.6 type 3 HARQ scheme [32]

In this process the receiver is allowed to decide either to ask for a retransmission including new redundant bits or to ask for a retransmission where systematic bits are included again. However, the type 3 HARQ protocol explained here above is considered without feedback, and just sending ACKs or NACKs is allowed, so the set of bits to be sent are decided independently from the receiver, and the systematic bits are included every time.

It is important to send the systematic bits every time. However, it decreases the system throughput. Therefore, it becomes a tradeoff between the safety measure of sending systematic bits every time and the throughput achieved.

3.3 Different HARQ schemes used in different technologies

HARQ protocols were introduced in the HSPA release.

a HSDPA.

The HARQ functions at both the Media Access Control (MAC) and the PHY layer. Since the MAC is located in the Node B (base station), erroneous transport blocks can be rapidly retransmitted, reducing the delay due to retransmissions compared with Radio Link Control (RLC) retransmissions [32]. The node B decides whether to use Incremental Redundancy or Chase Combining scheme by choosing the puncturing pattern to be used for the retransmission.

It is been proved that the IR scheme provides significant gains when the initial code rate is high. Thus, node B usually decides to use IR rather than CC when the User Equipment (UE) is close to it, and the transmission power does not limit the achievable data rate.

In HSDPA the UE asks for the retransmissions. On receiving a packet, the UE tries to decode it. Depending on the success or failure in decoding, UE confirms to the Base Station (BS) a good reception or asks for a retransmission (by sending a bit).

In HSDPA retransmissions are scheduled as any other data and the Node B is free to schedule the retransmission to the UE at any time instant and using a redundancy version of its choice. This type of operation is often referred to as adaptive asynchronous hybrid ARQ. Adaptive since the Node B may change the transmission format and asynchronous since retransmissions may occur at any time after receiving the ACK/NAK.

The HARQ structure implemented in HSDPA is a stop and wait scheme. In order to support continuous transmission to a single UE, multiple stop and wait structures are used in parallel. Each of these HARQ processes has its own buffer, identified by a number, where erroneous transmissions are stored while the new retransmission arrives. The data is deleted from the concrete buffer once the decoding of this block has been successfully performed.

b HSUPA.

The Node B informs the UE about the successful reception or not, asking for a retransmission if necessary. One of the main differences between the uplink and the downlink HARQ implementation concerns the use of soft handover in the uplink. When a UE is in soft handover, the HARQ protocol needs to be performed by various BS. The data transmitted by the UE can be received in one BS but not in another one.

For the uplink, different from the downlink, retransmission follows a non-adaptive synchronous pattern, where retransmissions take place at a predefined time after the initial transmission, and the different possible puncturing sets of bits are predefined from the moment of the initial transmission. That leads to an affordable control signaling overhead, since there is no need to specify the HARQ process number. HARQ is not applicable for all types of traffic.
Broadcast transmissions, where the information is addressed to multiple users, do not use HARQ.

c IEEE 802.16m. (Mobile WiMAX)

The HARQ mechanism is used for all unicast data traffic in both downlink and uplink. The IEEE 802.16m HARQ scheme is based on an N-process stop-and-wait protocol. In single channel stop-and-wait, the transmitter waits after each transmission until an acknowledgement from the receiver is obtained.
In the case of positive acknowledgement, a new packet is transmitted; otherwise, the same packet is re-transmitted. The N-process stop-and-wait mechanism makes use of the waiting time and transmits other sub-packets. Both BS and MS are required to maintain multiple simultaneous HARQ channels. The DL HARQ channels are identified by a HARQ Channel Identifier (ACID), whereas the UL HARQ channels are identified by both ACID and the index of the UL sub frame in which the UL HARQ data burst is transmitted. Multiple UL HARQ channels in the same UL sub frame are identified by different ACIDs and UL HARQ channels in different UL sub frames are identified by the index of the UL sub frame when they are addressed with the same ACID.

Generation of the HARQ sub-packets follows the channel coding procedures. The received sub packets are combined by the FEC decoder as part of the decoding process. The use of incremental redundancy HARQ is mandatory in IEEE 802.16m. Each sub-packet contains part of a code word identified by an SPID. In order to specify the start of a new transmission, a single-bit HARQ Identifier Sequence Number (AI_SN) is toggled on every new HARQ transmission attempt on the same ACID. If the AI_SN changes, the receiver will treat the corresponding HARQ transmission as a new encoder packet and discard previous HARQ transmissions associated with the same ACID.

The IEEE 802.16m uses adaptive asynchronous HARQ in the downlink.

On receiving a DL Basic Assignment Advanced Medium Access Protocol Information Element (A-MAP IE), the MS attempts to receive and decode the data burst. If the decoding is successful, the MS sends a positive acknowledgement to the BS; otherwise, the MS will send a negative acknowledgement to the BS. The process of re-transmissions is controlled by the BS using the ACID and AI_SN fields in the DL Basic Assignment A-MAP IE. If the AI_SN field for the ACID remains the same between two HARQ burst allocations, it indicates retransmission. The BS may allocate different resource and transmission formats that are signaled through the DL Basic Assignment A-MAP IE for each re-transmission. If the AI_SN field for the ACID is toggled, it indicates the transmission of a new HARQ packet. The maximum number of HARQ channels per MS in the downlink is 16. The delay between two consecutive HARQ transmissions of the same data burst does not exceed the maximum [T_ReTx_Interval ¼ 1, 2... 8]. The number of re-transmissions of the same data packet does not exceed the maximum [N_MAX_ReTx =4].

The HARQ ACK/NACK timing is defined for FDD mode and for TDD mode with certain DL/UL ratios. A failed HARQ burst should be re-transmitted within the maximum re-transmission delay bound. A HARQ burst is discarded if the maximum number of re-transmissions is reached. When persistent allocation is applied to initial transmissions, HARQ re-transmissions are supported in a non-persistent manner, i.e., resources are allocated dynamically for HARQ re-transmissions. An asynchronous HARQ scheme is used in the downlink where the interval between successive transmissions/re-transmissions may vary, providing more flexibility for the downlink scheduler.

The IEEE 802.16m uses a synchronous HARQ scheme in the uplink where the interval between successive transmissions/re-transmissions is the same, resulting in a lower signaling overhead in resource assignment. In synchronous HARQ, resource allocation for the re-transmissions in the uplink can be fixed or adaptive according to control signaling. The default operation mode of HARQ in the uplink is non-adaptive, i.e., the parameters and the resource for the re-transmission are known a priori. The BS can signal an adaptive uplink HARQ mode.

On receiving an UL Basic Assignment A-MAP IE, the MS transmits a HARQ sub-packet in the assigned resource. The BS attempts to decode the data packet. If the decoding is successful, the BS will send a positive acknowledgement to the MS; otherwise, the BS will send a negative acknowledgement to the MS. If re-transmission becomes necessary, and if the MS does not receive a UL Basic Assignment A-MAP IE for the failed HARQ sub packet, the MS transmits the next sub-packet through the resources assigned in the previous sub-packet transmission with the same ACID. An UL Basic Assignment A-MAP IE may be sent to signal the re-transmission with corresponding ACID and unchanged AI_SN. On receiving the UL Basic Assignment A-MAP IE, the MS performs the HARQ re-transmission.
The maximum number of HARQ channels per MS is 16 in the uplink.

d *LTE*

The LTE technology as specified within 3GPP Release 8 was first commercially deployed by end 2009. LTE has become the fastest developing mobile system technology. As other cellular technologies LTE is continuously worked on in terms of improvements. 3GPP groups added technology components into so called releases. Initial enhancements were included in 3GPP Release 9, followed by more significant improvements in 3GPP Release 10, also known as LTE-Advanced.

Long-Term Evolution (LTE) is the fourth generation (4G) air interface for mobile telephony. LTE is the access part of the Evolved Packet System (EPS). The main requirements for the LTE access network are high spectral efficiency, high peak data rates, short round trip time, and frequency flexibility. The EPS is purely IP-based. Both real time services and data services are carried by the IP. The IP address is allocated when the mobile is switched on and released when switched off.

LTE uses IR HARQ with a 1/3 turbo encoder used for FEC. Error detection is done by The Transport Block (TB) CRC. The receiver only receives different punctured versions of the same turbo-encoded data; each of these retransmissions are self-decodable. Thus, it falls into category of a type III Hybrid ARQ. In LTE retransmissions are sent with an initial coding rate of 1/2 or 3/4. The maximum number of simultaneous DL-HARQ processes (number of PDSCH transmissions catered for) is limited to 8.

In LTE, the N-channel stop-and-wait protocol is used as the Hybrid ARQ protocol as it offers low buffering requirements and low ACK/NACK feedback overhead.

3.4 HARQ in LTE.

One of the purpose of LTE to come in is to bring in high data rates. To achieve this UE must embrace some techniques where it can transmit data quickly and reliably.

LTE uses a hybrid automatic repeat request (HARQ) scheme for error correction. The eNodeB sends a HARQ indicator to the UE to indicate a positive acknowledgement (ACK) or negative acknowledgement (NACK) for data sent using the uplink shared channel. The channel coded HARQ indicator code word is transmitted through the Physical Hybrid Automatic Repeat Request Indicator Channel (PHICH).

3.4.1 HARQ Indicator.

A HARQ indicator of ‘0' represents a NACK and a ‘1' represents an ACK.

PHICH Groups

Multiple PHICHs are mapped to the same set of resource elements (REs). This set of REs constitutes a PHICH group. The PHICHs within a PHICH group are separated through different orthogonal sequences. A PHICH resource is identified by the index pair (ngroup PHICH, nseq PHICH). The variable (ngroup PHICH) is the number of the PHICH group and the variable (nseq PHICH) is the orthogonal sequence index within the group.

The number of PHICH groups varies based on whether the frame structure is type one, frequency division duplex (FDD), or type two, time division duplex (TDD).

The HARQ Indicator undergoes repetition coding to create a HARQ indicator code word made up of three bits $\{a_0, a_1, a_2\}$.

0 — Negative acknowledgement (0 0 0).

1 — Positive acknowledgement (1 1 1).

3.4.2 Formation of PHICH.

The HARQ Indicator code word undergoes BPSK modulation, scrambling, layer mapping, precoding, and resource mapping.

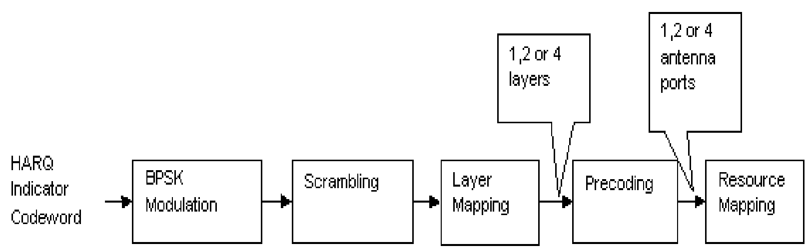

Figure 3.7 PHICH formation. (Ref matlab)

a Modulation

The HARQ indicator code word undergoes BPSK modulation resulting in a block of complex-valued modulated symbols, z_0, z_1, z_2.

b Scrambling

The block of modulated symbols is bitwise multiplied with an orthogonal sequence and a cell specific scrambling sequence to create a sequence of symbols d0… d (M_{symb}-1).The number of symbols M_{symb} is given by the following equation.

$$M_{symb}=3*N_{SF}^{PHICH}$$

Where N_{SF}^{PHICH} is the spreading factor for the PHICH which is given the value 4 when normal cyclic prefix is used or 2 if the extended cyclic prefix is applied. The orthogonal sequence allows multiple PHICHs to be mapped to the same set of resource elements.

Scrambling with a cell-specific sequence serves the purpose of intercell interference rejection. When a UE descrambles a received bit stream with a known cell specific scrambling sequence, interference from other cells will be descrambled incorrectly and therefore only appear as uncorrelated noise.

The three modulated symbols, z_0, z_1, z_2, are repeated N_{SF}^{PHICH} times and scrambled to create a sequence of six or twelve symbols depending on whether a normal or extended cyclic prefix is used. When using a normal cyclic prefix, the first four scrambled symbols are created as shown in the following figure.

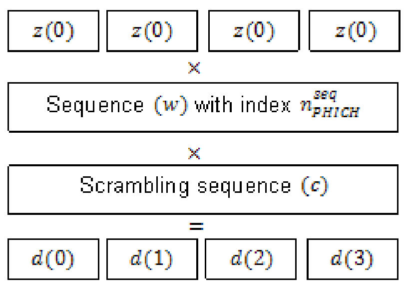

Figure 3.8 scrambled symbols. (Ref matlab)

The variable w is an orthogonal scrambling sequence given in the following table.

Sequence Index	Orthogonal Sequence, $w(0),\ldots,w\left(N_{SF}^{PHICH}-1\right)$	
n_{PHICH}^{seq}	Normal Cyclic Prefix, $N_{SF}^{PHICH}=4$	Extended Cyclic Prefix, $N_{SF}^{PHICH}=2$
0	[+1 +1 +1 +1]	[+1 +1]
1	[+1 −1 +1 −1]	[+1 −1]
2	[+1 +1 −1 −1]	[+j +j]
3	[+1 −1 −1 +1]	[+j −j]
4	[+j +j +j +j]	—
5	[+j −j +j −j]	—
6	[+j +j −j −j]	—
7	[+j −j −j +j]	—

Table 3 orthogonal scrambling sequence. (Ref matlab)

The variable c is a cell-specific pseudo-random scrambling sequence created using a length-31 Gold sequence. The scrambling sequence is initialized using the slot number within the radio frame and the cell ID.

c Resource Group Alignment

As resource element groups (REGs) contain four resource elements (each able to contain one symbol) the blocks of scrambled symbols are aligned to create blocks of four symbols.

In the case of a normal cyclic prefix, each of the original complex modulated symbols, z_0, z_1, z_2, is represented by four scrambled symbols. Therefore, no alignment is required.

In the case of an extended cyclic prefix each of the original complex modulated symbols, z_0, z_1, z_2, is represented by two scrambled symbols. To create blocks of four symbols, zeros are added before or after blocks of two scrambled symbols depending on whether the PHICH index is odd or even. This allows two groups to be combined during the resource mapping stage and mapped to one REG.

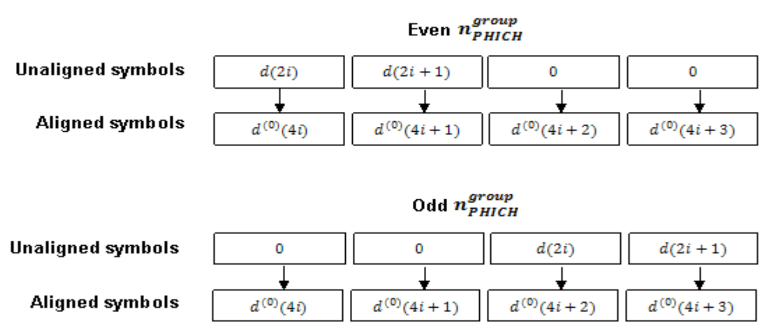

Figure 3.9 creation of block of four symbols. (Ref matlab)

d Layer Mapping

The Antenna mapping block maps the transport block to different antennas. LTE uses up to four transmit antennas. LTE supports different multiple transmit antennas schemes: transmit diversity, beamforming and spatial multiplexing. The goal of the resource block mapping is to map the data to be sent on each antenna to a set of resource blocks assigned by the scheduler. The complex symbols are mapped to one, two, or four layers depending on the number of transmit antennas used.

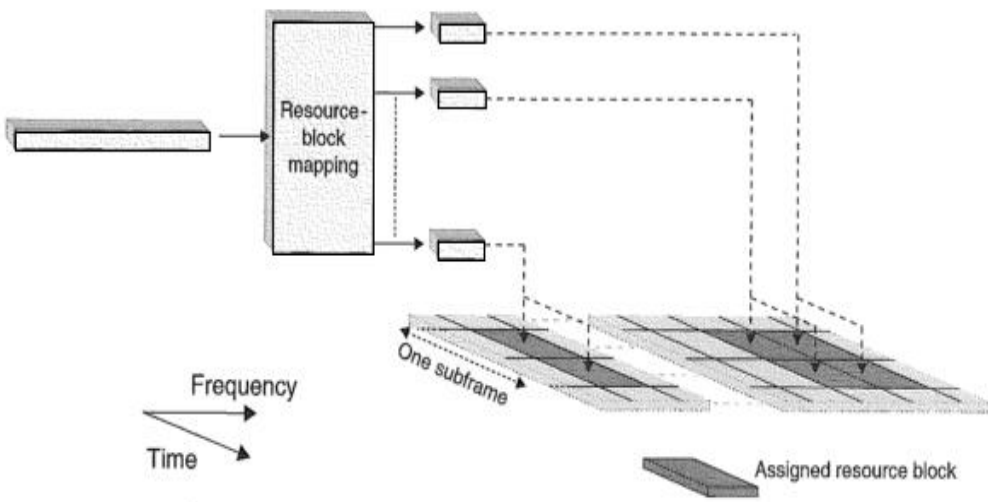

Figure 3.10 downlink resource block mapping Ref [26]

3.4.3 Mapping to Resource Elements

a PHICH Duration

The number of OFDM symbols used to carry the PHICH is configurable by the PHICH duration. The PHICH duration is either normal or extended. A normal PHICH duration causes the PHICH to be present in only the first OFDM symbol. In general an extended PHICH duration causes the PHCH to be present in the first three OFDM symbols but there are some exceptions. The PHICH is present in the first two OFDM symbols under the following exceptions.

- Within sub frame 1 and 6 when frame structure type 2 (TDD) is used.
- Within MBSFN sub frames.

b Relationship between CFI and PHICH Duration.

Since the control format indicator (CFI) configures how many OFDM symbols are used for mapping the physical downlink control channel (PDCCH) and hence which OFDM symbols are available for the physical downlink shared channel (PDSCH), care must be taken when using an extended PHICH duration so the PHICH is not mapped into the same region as the PDSCH.

3.5 Downlink transport channel processing

At the beginning of the transport channel processing, a Cyclic Redundancy Check (CRC) is computed and attached to each transport block (TB) for the detection of errors in the TB by the receiver. After the CRC insertion, the data (TB + CRC) to be sent are turbo coded with a coding rate of 1/3. The task of the hybrid-ARQ is to take care of the retransmission if erroneously received packets are received. Retransmission must represent the same information bits as the initial message but the coded bits used for each retransmission can be different than the original message. Later the information to be transmitted is modulated using one of the following modulation schemes: QPSK, 16QAM, and 64QAM representing two, four, and six bits per modulation symbol, respectively.

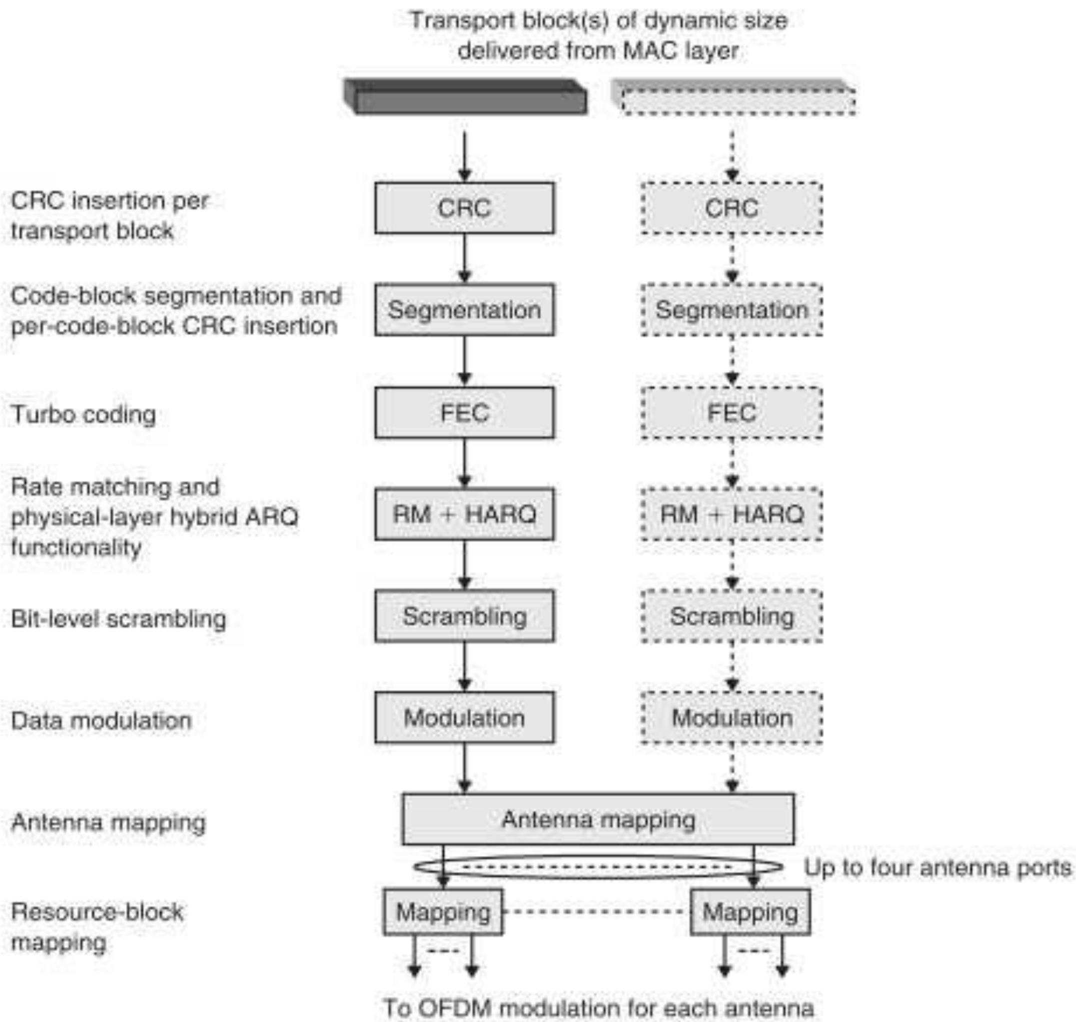

Figure 3.11 LTE downlink transport-channel processing [26]

Chapitre 4 LONG TERM EVOLUTION

4.1 Introduction.

In recent years, operators across the world have seen a rapid growth of mobile broadband subscribers. At the same time, the traffic volume per subscriber is also increasing rapidly; in particular, with the introduction of more advanced mobile devices and real time services such as multimedia telephony and mobile TV. The introduction of these new and demanding services such as audio, video streaming, interactive gaming with rapid response patterns has drawn attention toward possible limitation of the capacity and Quality of Service (QoS). Since these services have different performance requirements, for example in terms of bit-rates and packet delays, under the partnership of the 3GPP, LTE (Long Term Evolution) has been introduced to fulfill this ambitious task. LTE has been deployed in Europe, USA and around the world. Thus, nowadays operators have already started to propose LTE technology to subscribers in order to provide high speed data rates.

4.2 History of digital communication system

4.2.1 First Generation Cellular System

Almost all of the systems from this generation were analog systems where voice was considered to be the main traffic. These systems could often be listened to by third parties. 1G cellular telephone system divided cities into small cells. This division allowed extensive frequency reuse across a city, allowing millions to use cell phones simultaneously. Numerous incompatible analog systems were placed in service around the world in First Generation. This generation uses FM technology for voice transmission and digital signaling for control information. First generation systems include: AMPS, NAMPS, TACS, and NMT-900.

4.2.2 Second Generation Cellular System

All the standards belonging to this generation are commercial centric and they are digital in form. The 2G (second generation) systems designed were still used mainly for voice applications but were based on digital technology, including digital signal processing techniques. These 2G systems provided circuit-switched data communication services at a low speed. This system operates nationwide or internationally and today's mainstream system, al- though the data rate for users in this system is very limited. Around 60% of the current market is dominated by European standards. Second generation includes: USDC (United States Digital Cellular Standards IS-54 and Is-1 36)**,** GSM (Global System for Mobile Communications)**,** PDC (Pacific Digital Cellular)**,** CDMA-1.

4.2.3 Third Generation Cellular System

3G systems promise faster communications services, including voice, fax and Internet, anytime and anywhere with seamless global roaming. ITU's IMT-2000 global standard for 3G has opened the way to enabling innovative applications and services. The first 3G network was deployed in Japan in 2001.

2.5G networks, such as GPRS (Global Packet Radio Service) are already available in some parts of Europe. The systems in this standard are basically a linear enhancement of 2G systems. They are based on two parallel backbone infrastructures, one consisting of circuit switched nodes, and one of packet oriented nodes. The ITU denes a specific set of air interface technologies as third generation, as part of the IMT-2000 initiative. Currently, transition is happening from 2G to 3G systems. For third generation mobile (3G, FOMA) data rates are 384 kbps (download) maximum, typically around 200kbps, and 64kbps upload.

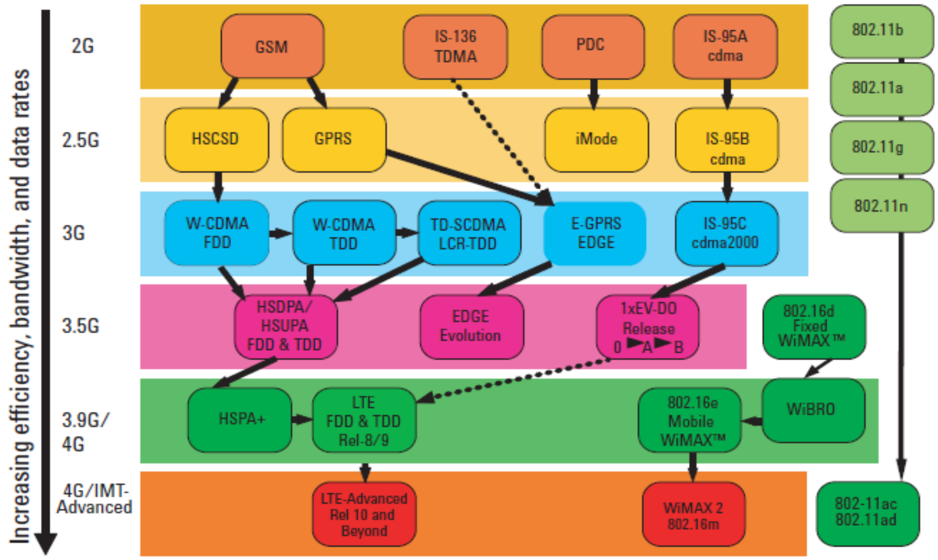

Figure 4.1 evolution of mobile networks. (Ref [17])

To achieve the target data rate of LTE and eventually 4G, many new techniques are necessary. Multiple-Input Multiple-Output (MIMO) is one key technique among them because of its ability to enhance the radio channel capacity of cellular systems at no extra cost of spectrum. The expression MIMO includes both traditional beamforming and diversity techniques as well as spatial multiplexing techniques. The diversity format of MIMO was included in 3G and 3G enhancements already, like Space-Time Transmit Diversity (STTD), Closed-Loop Transmit Diversity (CLTD) mainly targeting at increasing the link quality.

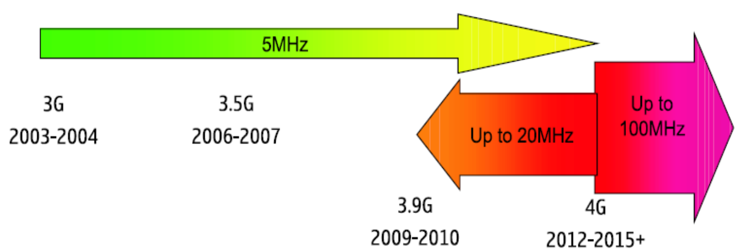

Figure 4.2 roadmap to LTE (ref [18]).

4.3 Scope about 3GPP.

The Third Generation Partnership Project (3GPP) is made up of several telecommunications standard development organizations like (ARIB, ATIS, CCSA, ETSI, etc.) and it is responsible for the development of E-UTRAN specifications aiming at establishing interoperability between multiple vendors, and adapt the system to the regulations of the different countries.

4.3.1 LTE Releases.

Release 99 (Mar. 2000): UTRA in FDD and TDD modes.
Rel-4 (Mar. 2001): TD-SCDMA.
Rel-5 (Mar. 2002): HSDPA with IMS (IP Multimedia Services).
Rel-6 (Mar. 2005): HSUPA with MBMS.
Rel-7 (2007): DL MIMO, optimized real-time services (VoIP, gaming, push-to-talk).

4.3.2 LTE Release 8

LTE release 8 introduced LTE for the first time in 2008. It consisted of a completely new radio interface and core network which enabled substantially improved data performance compared with previous systems. Highlights from release 8 include the following features:

- Up to 300Mbit/s downlink and 75Mbit/s uplink.
- Latency as low as 10ms.
- Bandwidth sized in 1.4, 3, 5, 10, 15, or 20MHz blocks to allow for a variety of deployment scenarios.
- Orthogonal frequency domain multiple access (OFDMA) downlink.
- Single-carrier frequency domain multiple access (SC-FDMA) uplink.
- Multiple-input multiple-output (MIMO) antennas.
- Flat radio network architecture, with no equivalent to the GSM base station controller (BSC) or UMTS radio network controller (RNC), and functionality distributed among the base stations (enhanced NodeBs)

- All IP core network, the System Architecture Evolution (SAE).

4.3.3 LTE Release 9

LTE release 9 brought refinements to release 8 LTE as well as introducing some new service features and network architecture improvements. Highlights from release 9 include the following noteworthy modifications:

- Evolved multimedia broadcast and multicast service (eMBMS) for the efficient delivery of the same multimedia content to multiple destinations
- Location services (LCS) to pinpoint the location of a mobile device
- Dual layer beamforming

4.3.4 LTE Release 10

LTE release 10 is considered to be the onset of LTE-Advanced. It significantly improved data throughput and extended cell coverage. Highlights from release 10 include the following features:

- Higher order MIMO antenna configurations supporting up to 8×8 downlinks and 4×4 uplinks.
- Data throughput of up to 3Gbit/s downlink and 1.5Gbit/s uplink.
- Carrier aggregation (CA), allowing the combination of up to five separate carriers to enable bandwidths up to 100MHz.
- Relay nodes to support Heterogeneous Networks (HetNets) containing a wide variety of cell sizes.
- Enhanced inter-cell interference coordination (eICIC) to improve performance towards the edge of cells.

4.3.5 LTE Release 11 and Beyond

LTE release 11 includes such features as enhancements to Carrier Aggregation, MIMO, relay nodes, and eICIC, introduction of new frequency bands, coordinated multipoint transmission and reception to enable simultaneous communication with multiple cells, and advanced receivers.

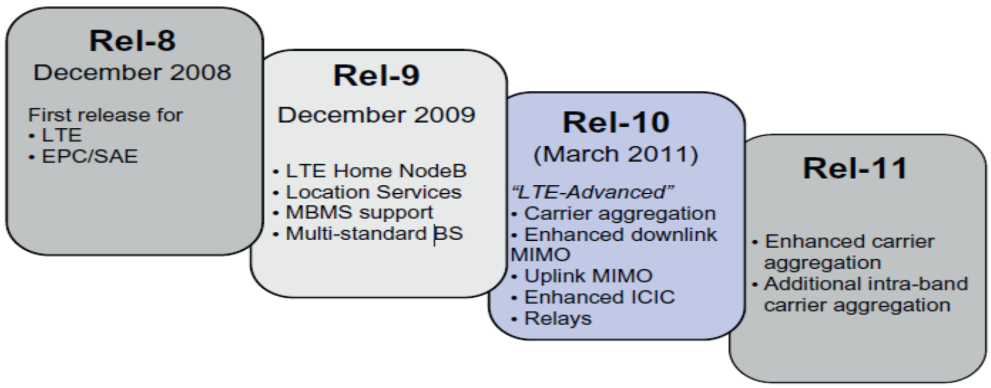

Figure 4.3 summery of Releases of 3GPP specifications for LTE [25]

4.4 LTE architecture.

4.4.1 General architecture of the network.

The global architecture not only include the core network and access network but also other blocs showing the relation between them. I look at different interfaces used for signalization and data and also the interfaces dedicated on control which supports only signalization. LTE is developed under the assumption that all the services are packet based and therefore the radio access technology, as well as the core network (EPC: Evolved Packet Core), are fully packet-switched (PS) with IP connectivity.

EPS (Evolved Packet System) is a new bloc in LTE comprised of EPC (Evolved Packet Core) and E-UTRAN (Evolved UTRAN). The figure below shows the general architecture of LTE.

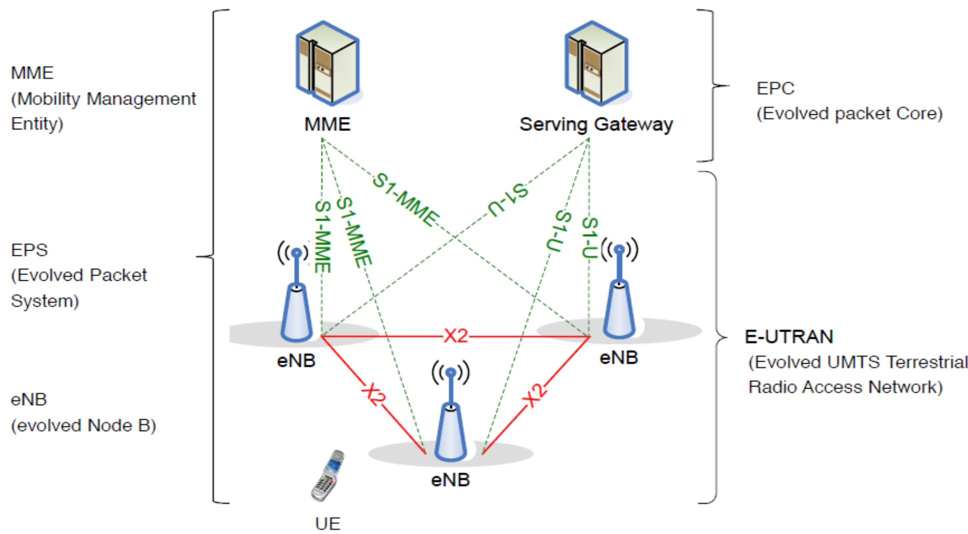

Figure 4.4 simple architecture LTE. [25]

Now, we can take a closer look at the LTE architecture and the interfaces between the different entities in the following diagram.

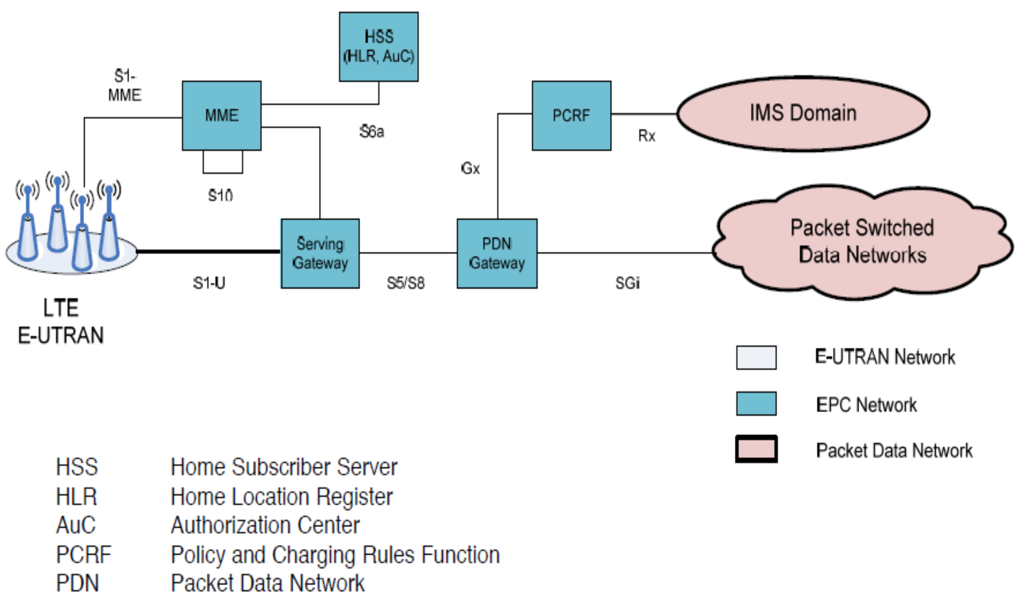

Figure 4.5 block diagram for LTE architecture. Ref [34]

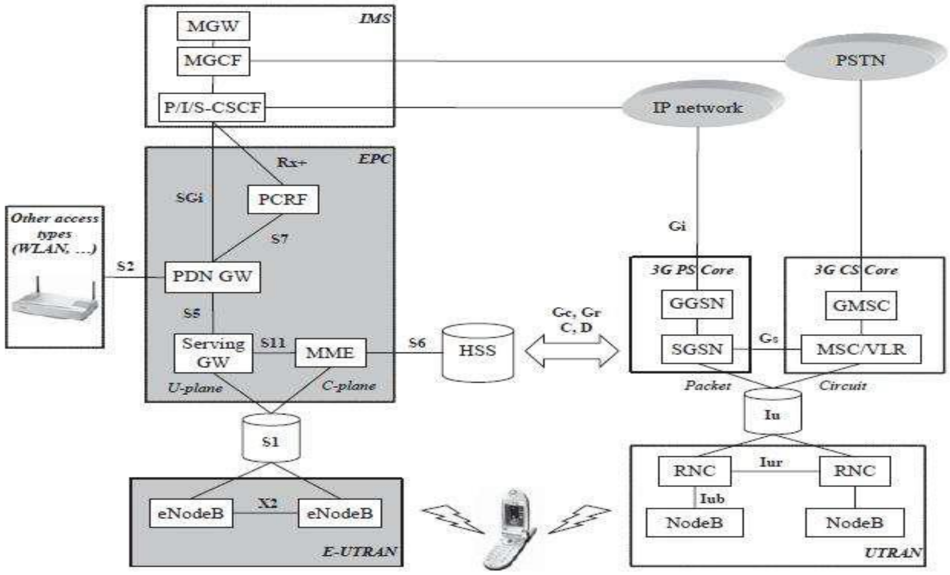

Figure 4.6 general LTE architecture Ref [25]

4.4.2 ACCESS NETWORK.

I look at some of the characteristics of the access network which include the following:

a Radio interface bandwidth.

E-UTRAN can support up to 100Mbps with the frequency band of 20 MHz in downlink and up to 50Mbps in uplink with the same frequency band.
It uses OFDMA in downlink and SC-FDMA in uplink.

b Connectivity

The UE pass from the idle mode to active mode for it to send and receive traffic. This takes less than 100ms.

c Delay in data transmission

The delay is less than 5ms between UE and access network. If only one UE is active on the radio interface, this delay can reduce to 25ms which makes LTE good in real time IP transmissions.

d Mobility

This is assured with the speed between 120km/h and 350km/h.in LTE hard handover is used the situations where the UE is moving at a high speed.

e Band usage flexibility.

A range of frequency bands are supported in LTE including the following.
1.25, 2.5, 5, 10, 15 and 20MHz.

4.4.3 CORE NETWORK.

In 2G and 3G networks, we used to distinguish CS (Circuit Switched) from PS (Packet S witched) in the core network. In LTE the core network is purely packet switched (Evolved Packet Core).

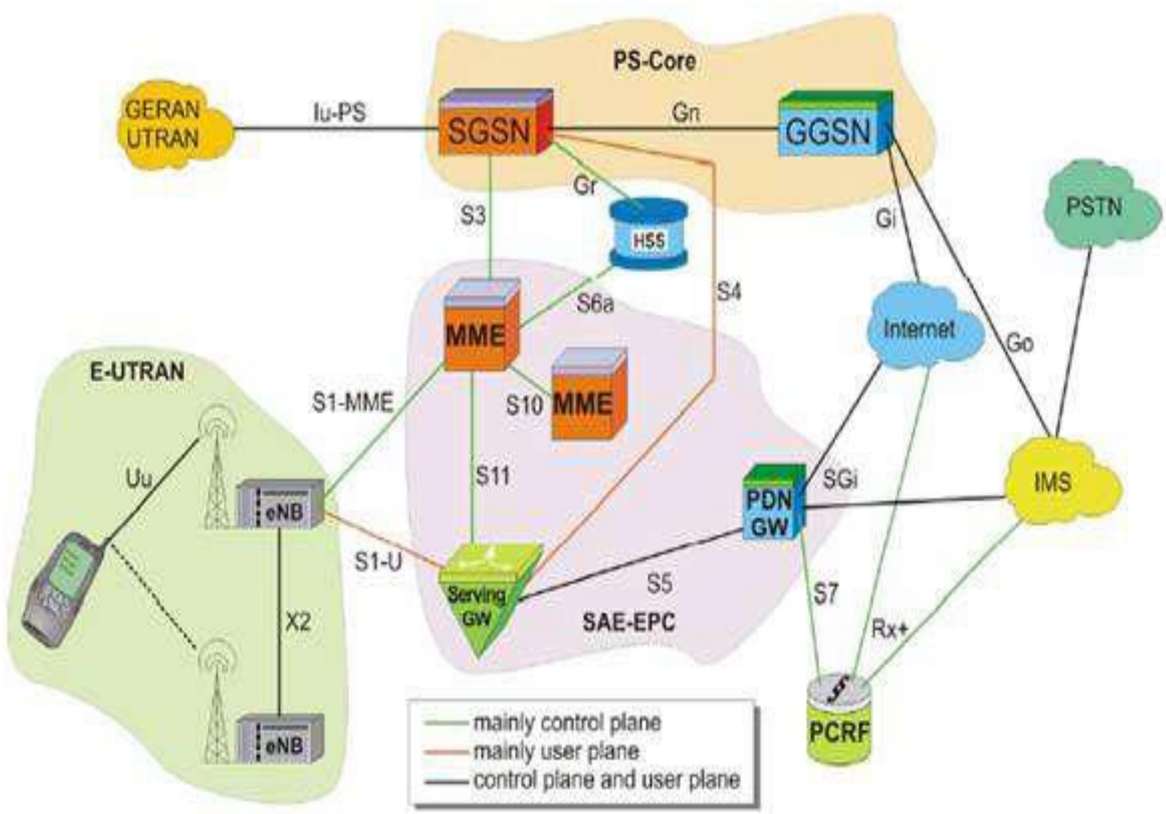

Figure 4.7 core network Ref [34]

4.4.4 Core network entities

EPC supports only access to the packet switched domain and it consists of the following entities as shown in the figure:

- **Mobility Management Entity (MME):** It’s the principal element of the EPC. Its functions include the following:

Mobility Management: In Idle and Active mode, UE tracking, MME selection and mobility between 3GPP access networks, Authentication and security through NAS signaling, Management of subscription profile and service connectivity, Packet core Bearer management functions including dedicated bearer establishment, Paging procedures and intra-LTE handover.

- **Serving Gateway (S-GW):** provides tunneling management between P-GW and eNodeB and switching.
- **PDN Gateway (P-GW):** is the router that looks to the outside world (Internet).

Its main functions are: User Equipment IP allocation and routing, per user packet filtering, charging for UL/DL per UE, per PDN and QoS Class Identifier and Mobility to non-3GPP RATs.

- **Home Subscriber Server (HSS):** database that stores subscriber data such as User identification, addressing and the user-specific security credentials needed for authentication and ciphering.[32]

- **Policy and Charging Rules Function (PCRF):** responsible for quality-of-service (QoS) handling, Interfaces with the PDN gateway to convey policy decisions to it and charging.

4.5 Overview of bandwidth demand in 4G. [15]

Mobile network system enabling true broadband wireless access with peak data rates of 100 Mbps in high-mobility and one 1 Gbps in low-mobility applications.

Spectrum allocation as of today is fixed with several dispersed band fragments across geographical regions that are used by different standards.

4.5.1 Spectrum Allocation Methodologies.

The current method of assigning spectrum to different radio systems is based on fixed spectrum allocation schemes. Dynamic Spectrum Allocation (DSA) concepts represent the state-of-the-art for enabling higher data throughput and more efficient ways of handling the spectrum.

a Fixed Spectrum Allocation

A block of radio spectrum is allocated for a particular Radio Access Technology (RAT). Normally, such a block of resources is for individual operator use. These spectrum blocks are of fixed size and are separated by fixed guard bands, and are reserved solely for the use of the radio license owner until the end of the license period. The fixed allocation of guard bands controls interference between different networks accessing contiguous spectrum and it is thus a simple and easily regulated way of managing spectrum resources.

ITU-R is responsible for the global regulation of the frequencies allocated on a geographical base, whereas the use of spectrum in each country is nationally regulated by the corresponding government agencies with the power to make spectrum available for particular use within their operational area. Spectrum extensions for the LTE and LTE-Advanced systems is thus defined in WRC-2000 and WRC-07.

b Dynamic Spectrum Allocation Principles (DSA).

DSA creates a balance between the data pipe and the available spectrum. As spectrum availability is scarce, IMT-A capacity requirements need frequency agile radio technologies that are able to aggregate bandwidth on demand, enabling flexible usage of the spectrum and to operate across any band. There are two main methods of DSA:

- **Spectrum Sharing and Coexistence (SSC).** The purpose of this method is to simplify the coexistence of different wireless systems within the same frequency band. SSC may enable different operators to share resources within the allocated band across the same standards and offer services to users using higher bandwidths.

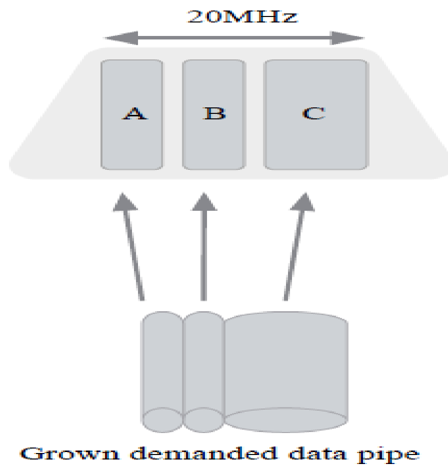

Figure 4.8 SSC concept

A demand of an increased data pipe may be redirected across multiple carriers A, B or C within the same RAT. An example it could be LTE 20 MHz spectrum band divided into 10+5+5 MHz carriers.

- **Flexible Spectrum Use (FSU).**

This is aimed at regulating spectrum access between different radio access networks (RAN) within the same radio access technology (RAT). FSU allows the sharing of spectrum within the same radio access technology.

It is based on the assumption that when one network operator needs spectrum another network operator might have spectrum available. Therefore the use of the unused spectrum or sharing of the spectrum from a common pool leads to a more efficient utilization of spectrum. There can be a single operator and a multi operator FSU. The purpose of single operator FSU is to reach a high performance in uncoordinated deployments, and enables different Uplink/Downlink

(UL/DL) switching points for time division duplex (TDD) operations to enhance spectrum usage. The multi operator FSU simplifies different operators to access the common spectrum pool in flexible manner based on their individual traffic requirements.

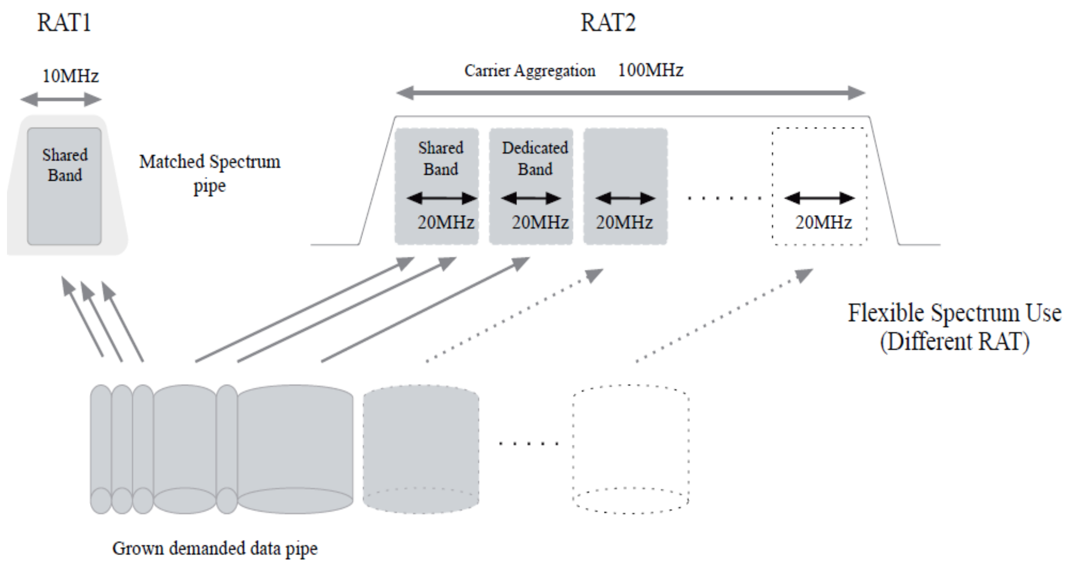

Figure 4.9 Flexible Spectrum use (FSU) concept

Band	MHz	Uplinks MHz	Downlink MHz		Region or typical name
1	2x60	1920-1980	2110-2170		UMTS core, "2.1GHz"
2	2x60	1850-1910	1930-1990		US PCS, "1900MHz"
3	2x75	1710-1785	1805-1880		"1800MHz"
4	2x45	1710-1755	2110-2155		US AWS
5	2x25	824-849	869-894		"850MHz"; US, Korea, APAC, MEA, Africa
7	2x70	2500-2570	2620-2690		"2.6GHz"
8	2x35	880-915	925-960		GSM 900
9	2x35	1749-1784	1844-1879		Japan 1700
10	2x60	1710-1770	2110-2170		Extended AWS
11	2x20	1427.9-1447.9	1475.9-1495.9		Japan 1500
12	2x17	699-716	729-746		US 700 MHz Lower (Band A,B,C)
13	2x10	777-787	746-756		US 700 MHz Upper (Band C) – Verizon
14	2x10	788-798	758-768	FDD	US 700 MHz Upper (Band D+)
17	2x12	704-716	734-746		US 700 MHz Lower (Band B, C) – AT&T
18	2x15	815-830	860-875		Japan 800 – new
19	2x15	830-845	875-890		Japan 800 – new
20	2x30	832-862	791-821		„800MHz"; European Digital Dividend band
21	2x15	1448-1463	1496-1511		Japan (upper 1500)
22	2x80	3410-3490	3510-3590		3,5 GHz band FDD
23	2x20	2000-2020	2180-2200		US S-band
24	2x34	1626.5-1660.5	1525-1559		US L-band
25	2x65	1850-1915	1930-1995		US ext. 1900
26	2x35	814-849	859-894		Korea, US: Extended 850
27	2x17	807-824	852-869		Latin America, 850
28	2x45	703-748	758-803		"APAC 700"; mainstream
29	1x11		717-728		Carrier Aggregation

Figure 4.10 LTE frequency spectrum

4.6 LTE interfaces and protocols.

We first look at the simple architecture of EPS with all network interfaces and nodes presented.it shows the interaction and signalization at the user level.

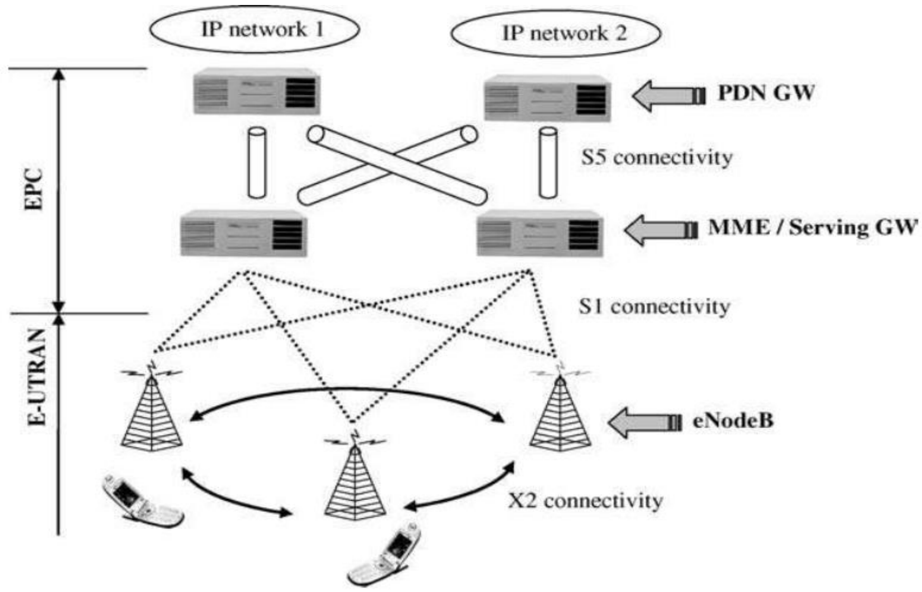

Figure 4.11 simple EPS architecture Ref [25]

For the interface S1:

S1-U (user plane)-transports the user data between the eNodeB and Serving GW.

S1-C (control plane)-transport only signalization information between eNodeB and MME.

4.6.1 Network interfaces of E-UTRAN.

We describe extensively the interfaces S1 and X2.

E-UTRAN is composed of two layers:

- Radio Network-consists of upper layer protocols of the interface.
- Transport Network-refers to the manner in which data is transported in the radio network layer.

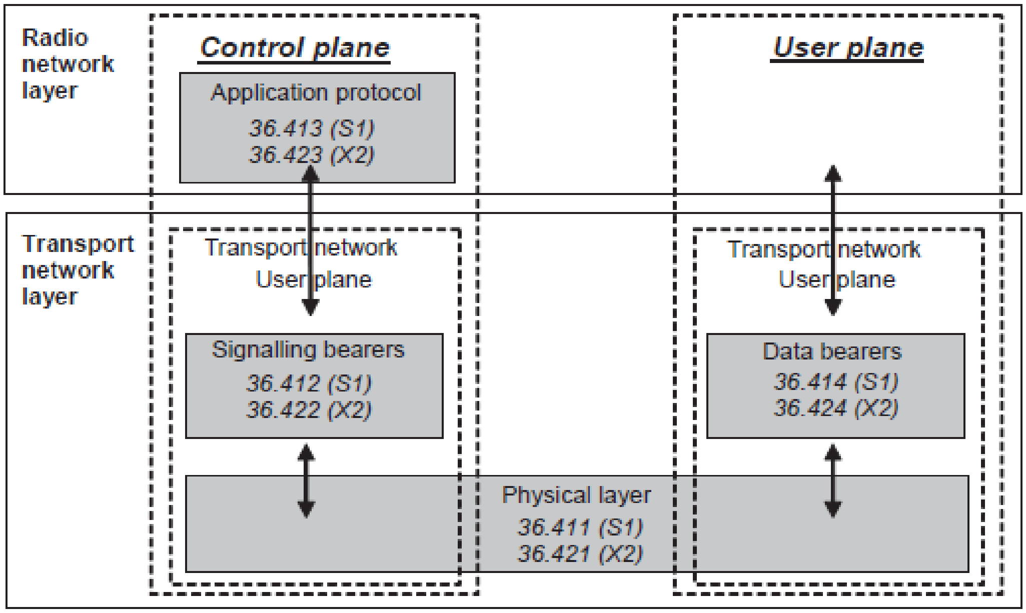

Figure 4.12 interface E-UTRAN Ref [25]

The eNodeB (eNB) holds all the network functionalities and therefore there is no need for a centralized controller as the RNC (Radio Network Controller) in UMTS. The E-UTRAN also define a separation between User Plane and Control Plane which make them independent from each other. This fact influence the latency of the system by lowering it and allows a better scalability.

4.6.2 User plane and control plane.

a User plane

This plane does not necessarily contain user profile but it also contain information on signalization associated with application services like SIP or RTCP.

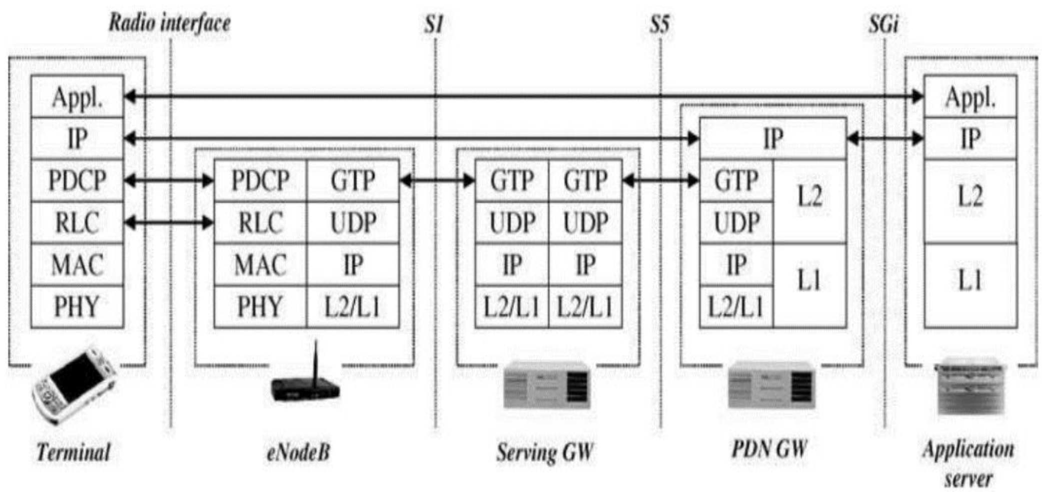

Figure 4.13 protocol pile for user plane Ref [25]

As seen in the figure above the packets from the application layer are routed through Packet Core Gateways to their final destination. Application layer can have a large number of protocols like TCP and UDP for end to end transportation, RTP (Real Time Protocol) for data transportation and signalization protocol at application layer like SIP, SDP, and RTCP.

b Control plane

It corresponds to the information flux by the E-UTRAN and EPC as signalization flux.it includes signalization message from RRC (Radio Resource Control).

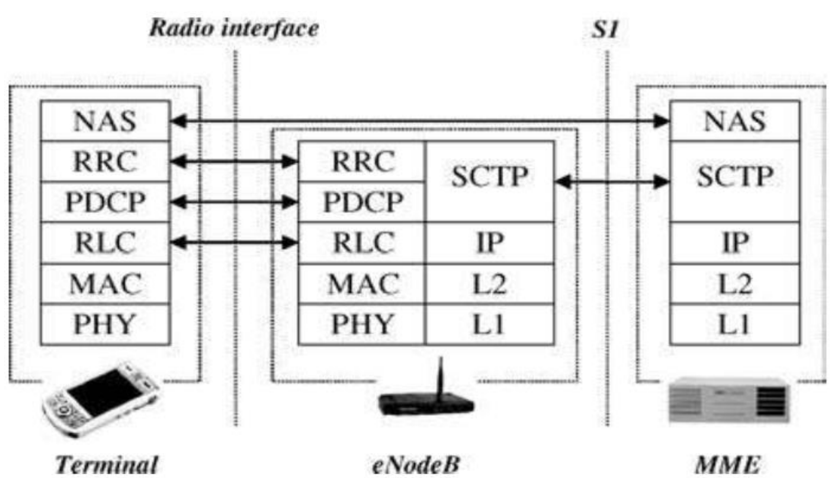

Figure 4.14 protocol pile for control plane Ref [25]

4.6.3 Radio interface protocols

The protocols in this interface are very specific and include the following layers:
PHY (Physical Layer), MAC (Medium Access Control) in charge of packet order and fast repetitions, RLC (Radio Link Control) is responsible for weak data transmission and PDCP (Packet Data Convergence Protocol) which provides the compression protocol and ciphering of data.

The PDCP layer whose role is to provide the compression protocol for the headers and implement data ciphering. It supports radio carriers. The RLC layer provides the PDCP layer with basic services from layer 2 of OSI model like data packets segmentation and ARQ (Automatic Repeat Request) for error correction mechanism.

The MAC layer maps and multiplexes the logical channels onto the transport channels.
The MAC layer also supports HARQ (Hybrid ARQ).

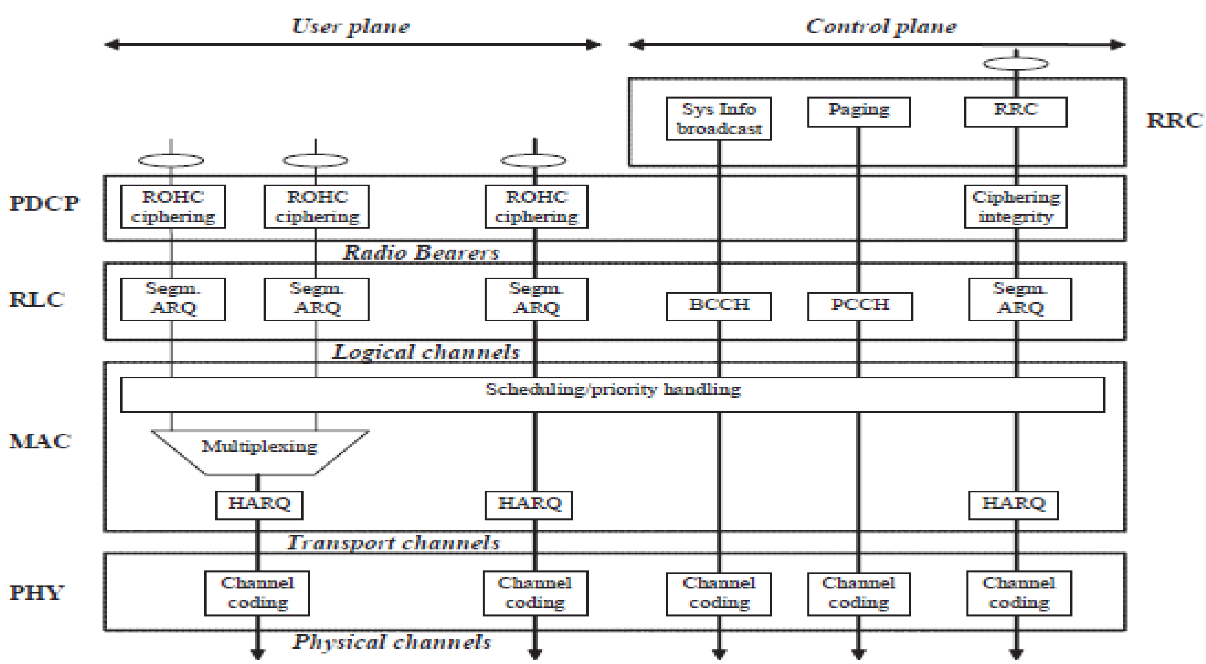

Figure 4.15 protocol layer structure for downlink in eNodeB Ref [34]

4.7 RADIO CHANNELS.

E-UTRAN has different types of channels which include the following:

- Logical channels-what is transmitted.
- Transport channels-how it is transmitted.
- Physical channels.

Now before we look at these channels, we need to understand how the MAC layer sees the PHY layer. We should understand that The LTE PHY layer is typically full duplex, it is designed primarily for full duplex operation in paired spectrum.
LTE can support TDD operation in unpaired spectrum. The PHY layer operates continuously for downlink with interspersed synchronization, providing multiple channels simultaneously with varying modulation. The downlink channel operates as a continuous stream.

LTE uses the concept of a resource block, which is a block of 12 subcarriers in one slot. A transport block is a group of resource blocks with a common modulation/coding. The physical interface is a transport block, which corresponds to the data carried in a period of time of the allocation for the particular UE. Each radio sub frame is 1 millisecond (ms) long; each frame is 10 milliseconds. Multiple UEs can be serviced on the downlink at any particular time in one transport block. The MAC controls what to send in a given time.

4.7.1 LOGICAL CHANNELS.

There are two types of logical channels
There exists Logical control channels and Logical traffic channels.

a Logical control channels

- BCCH (Broadcast Control Channel): A downlink channel for broadcasting system control information.

- PCCH (Paging Control Channel): A downlink channel that transfers paging information. This channel is used when the network does not know the location cell of the UE.
- CCCH (Common Control Channel): Uplink channel for transmitting control information between UEs and network. This channel is used by the UEs having no RRC connection with the network.
- MCCH (Multicast Control Channel):
- DCCH (Dedicated control Channel): A point-to-point bi-directional channel that transmits dedicated control information between a UE and the network. Used by UEs that have an RRC connection.

b Logical traffic channels

- DTCH (Dedicated Traffic Channel): A point-to-point channel, dedicated to one UE, for the transfer of user information. A DTCH can exist in both uplink and downlink.
- MTCH (Multicast Traffic Channel):

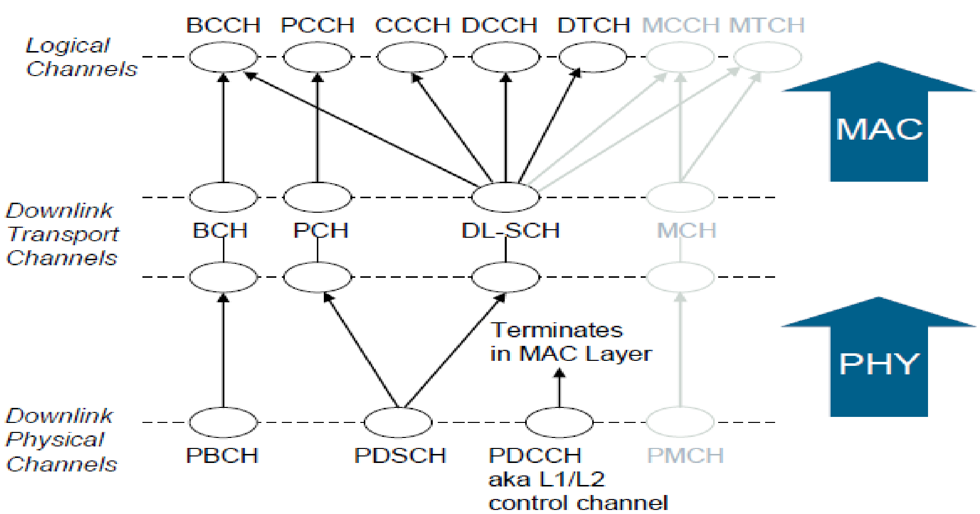

Figure 4.16 MAC downlink mapping

4.7.2 PHYSICAL CHANNELS

- **Physical broadcast channel (PBCH)**

- The coded BCH transport block is mapped to four sub frames within a 40 ms interval.
- 40 ms timing is blindly detected, i.e. there is no explicit signaling indicating 40 ms timing.
- Each sub frame is assumed to be self-decodable, i.e. the BCH can be decoded from a single reception, assuming sufficiently good channel conditions.

- **Physical control format indicator channel (PCFICH)**

  - Informs the UE about the number of OFDM symbols used for the PDCCHs.
  - Transmitted in every sub frame.

- **Physical downlink control channel (PDCCH):**
  - Informs the UE about the resource allocation of PCH and DL-SCH, and Hybrid ARQ information related to DL-SCH
  - Carries the uplink scheduling grant.

- **Physical Hybrid ARQ Indicator Channel (PHICH):**

Carries Hybrid ARQ ACK/NACKs in response to uplink transmissions.

- **Physical downlink shared channel (PDSCH):** Carries the DL-SCH and PCH.

- **Physical multicast channel (PMCH):** Carries the MCH.

- **Physical uplink control channel (PUCCH):**
  - Carries Hybrid ARQ ACK/NACKs in response to downlink transmission.
  - Carries Scheduling Request (SR).
  - Carries CQI reports.

- **Physical uplink shared channel (PUSCH):** Carries the UL-SCH.

- **Physical random access channel (PRACH):** Carries the random access preamble.

4.7.3 TRANSPORT CHANNELS.

Transport channels describe how data is protected from the transmission errors, the type of canal coding, and the CRC protection used, packet size of the data send to the radio interface.

a Downlink transport channels include the following:

- **BCH (Broadcast Channel)** it's associated with the logical channel BCCH. It has a fixed and predefined Transport Format
- **PCH (Paging Channel)** associated with the BCCH.
- **DL-SCH (Downlink Shared Channel).**
- **MCH (Multicast Channel)** associated with MBMS to control the transport information.

b The uplink transport channels include the following:

- **UL-SCH (Uplink Shared Channel).** Which is equivalent to DL-SCH in uplink.
- **RACH (Random Access Channel).**

4.8 LTE frame structure

Each radio frame has the length of 10 ms which contains ten equal sub frames of 1 ms length. The basic time unit is specified as $T_s = 1/30720000s$, hence the time of frame length can be specified as $T_{frame} = 307200T_s$ and sub frame as $T_{subframe} = 30720T_s$.

LTE can operate in both FDD and TDD. Basically the frames are equal in the structure, furthermore TDD has a special sub frame inserted. This special frame is used for the required guard time to switch from uplink to downlink transmission and vice versa.

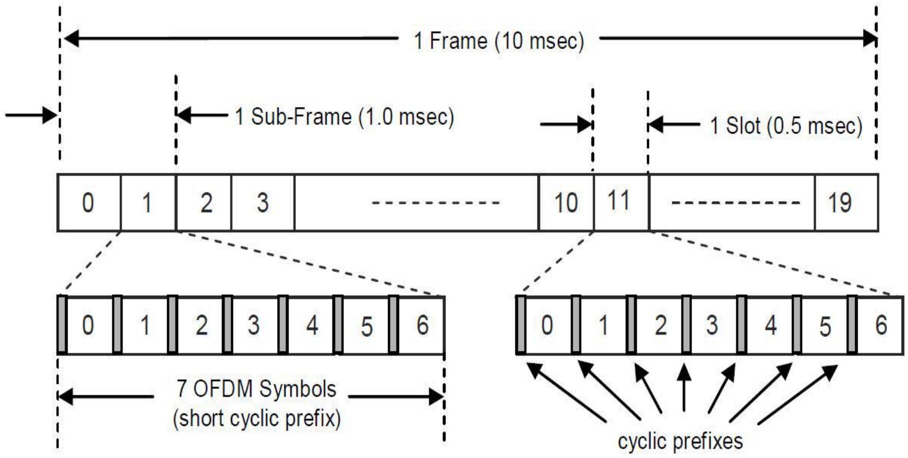

Figure 4.17 LTE generic frame structure

4.9 Types of LTE frames.

4.9.1 Type 1 (FDD).

Designed for FDD and can be used for both half and full duplex FDD transmission modes. One sub frame includes two slots. One half of the frames is used for downlink and the other half is used for uplink transmission, they are separated in the frequency domain.

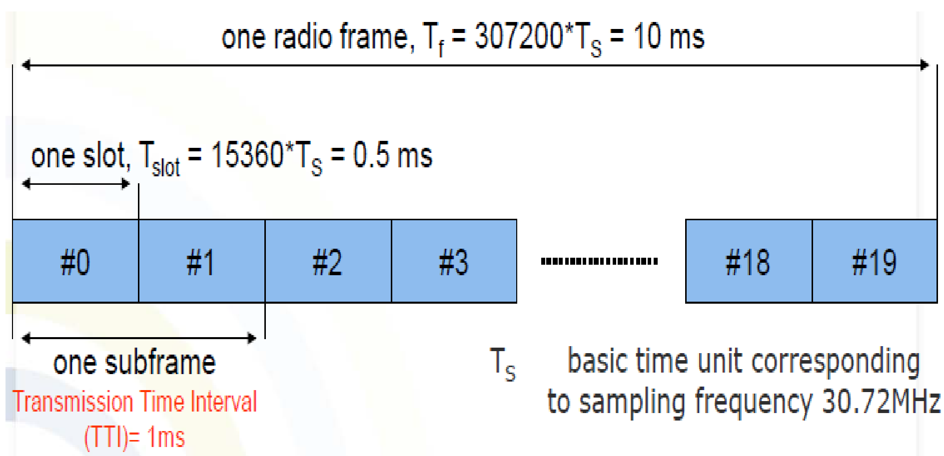

Figure 4.18 LTE type 1 frame structure

4.9.2 Type 2 (TDD).

In TDD mode type 2 frame is used, where it consists two same sized half frames with duration of 5 ms. Special sub frame is constructed by three fields named Downlink Pilot Timeslot (DwPTS), Guard Period (GP) and Uplink Pilot Timeslot (UpPTS).

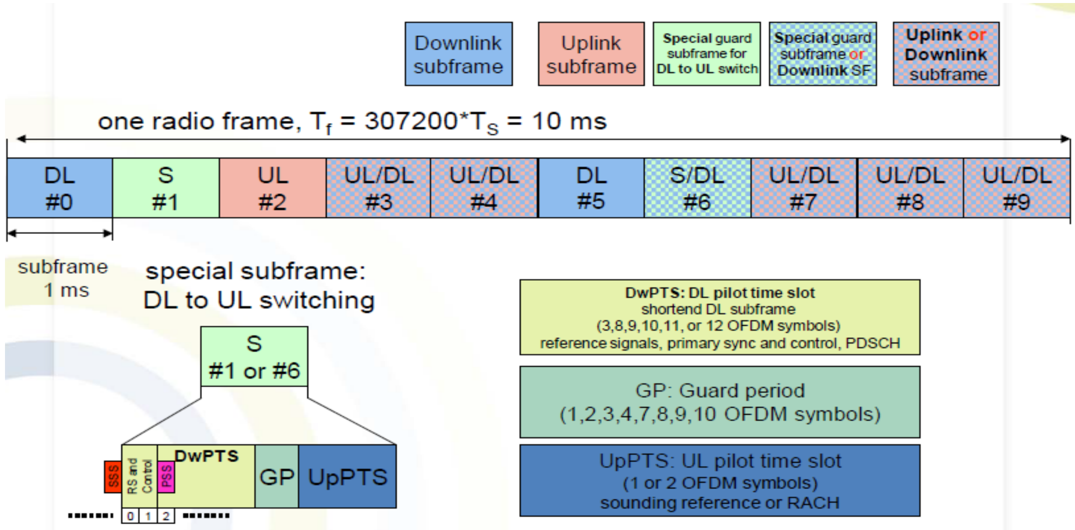

Figure 4.19 LTE type2 frame structure

4.10 Slot structure.

Every transmitted slot can physically be seen as a time frequency grid, where each resource element (RE) matches one OFDM sub carrier during OFDM symbol interval. The transmission bandwidth gives the number of sub carriers used for the transmission. LTE has two lengths of CP specified, a normal and extended CP. For normal CP, each slot has seven OFDM symbols and for extended six OFDM symbols are placed in each slot.

In LTE downlink transmission the subcarrier spacing is 15 kHz. So these 12 subcarriers are grouped together to one resource block (RB) in frequency domain.

Each RB uses 180 kHz bandwidth in one slot duration. For normal CP, each RB has 84 RE's, and for the extended CP it has 74 RE’s, due the longer CP length.

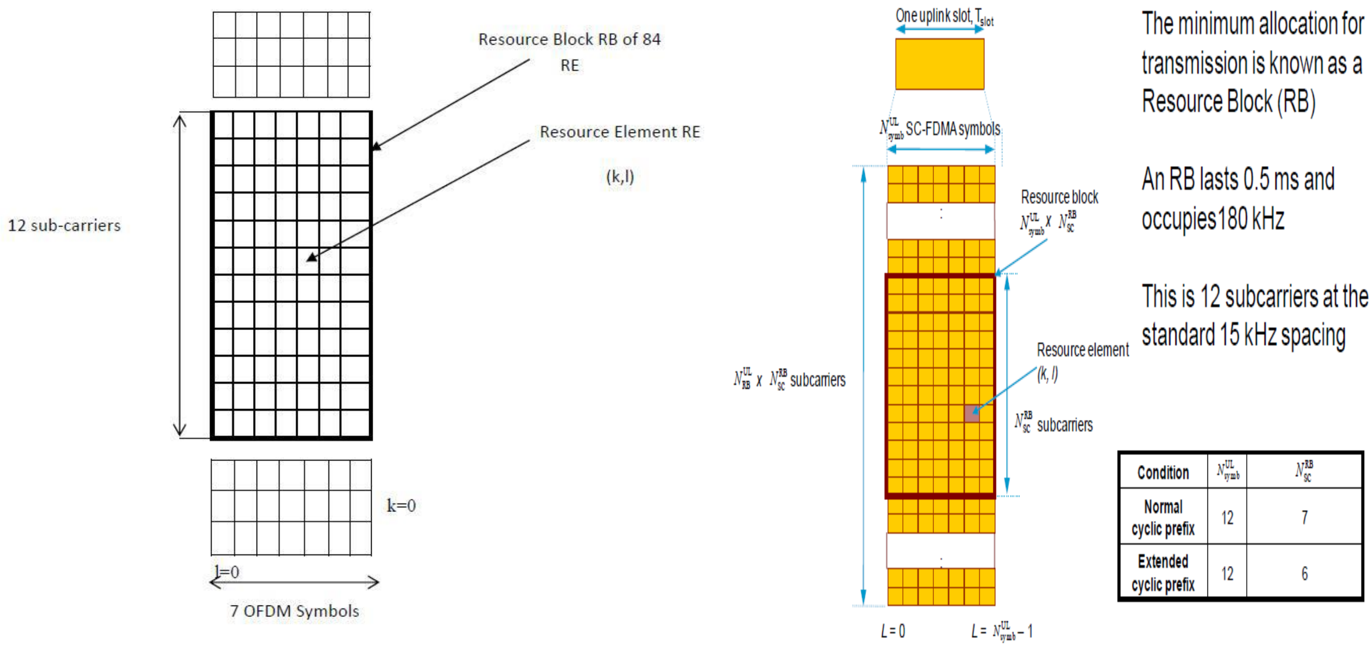

Figure 4.20 LTE resource grid for normal (a) downlink (b) uplink CP. Ref

There are two reasons why LTE has defined two CP lengths:

- ✓ For very large cells less efficient longer CP can be advantageous over normal one, due the large delay spread. On the other hand, if the channel performance is limited more by noise than by signal corruption, it may not recover the losses in terms of reduced received signal energy.

- ✓ For multicast or broadcast transmissions, cyclic prefix not only covers the time dispersion, but it also used to distinguish the timing difference between the transmissions from the cells involved.

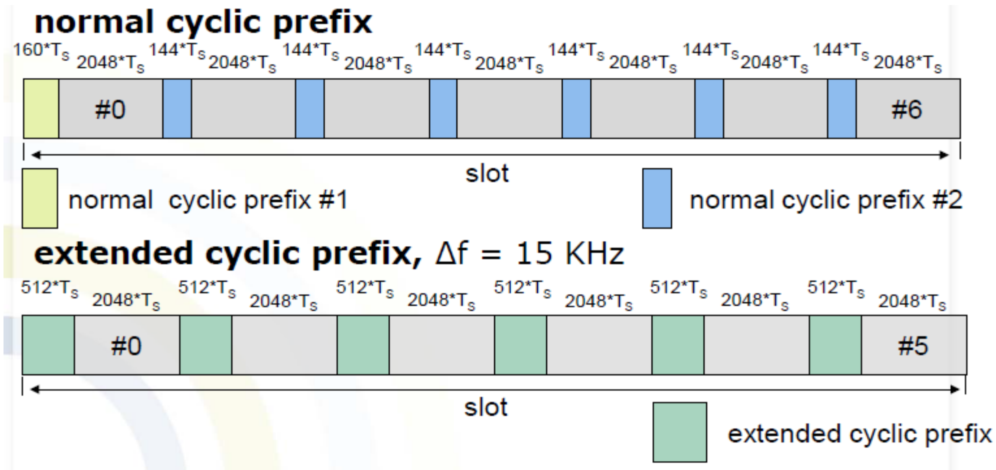

Figure 4.21 slot structures with normal CP and extended CP

4.11 Frames and Packet Timelines: LTE Downlink

A full frame is 10 ms but we normally think in terms of the 1-ms sub frame, which contains the transport block. In the transport block we have MAC header and space (padding). Within that there is the RLC header, then within the RLC header there can be a number of PDCPs. There is an arbitrary relationship between the IP packets coming in, which form the SDUs, and how the RLC PDUs are formed.

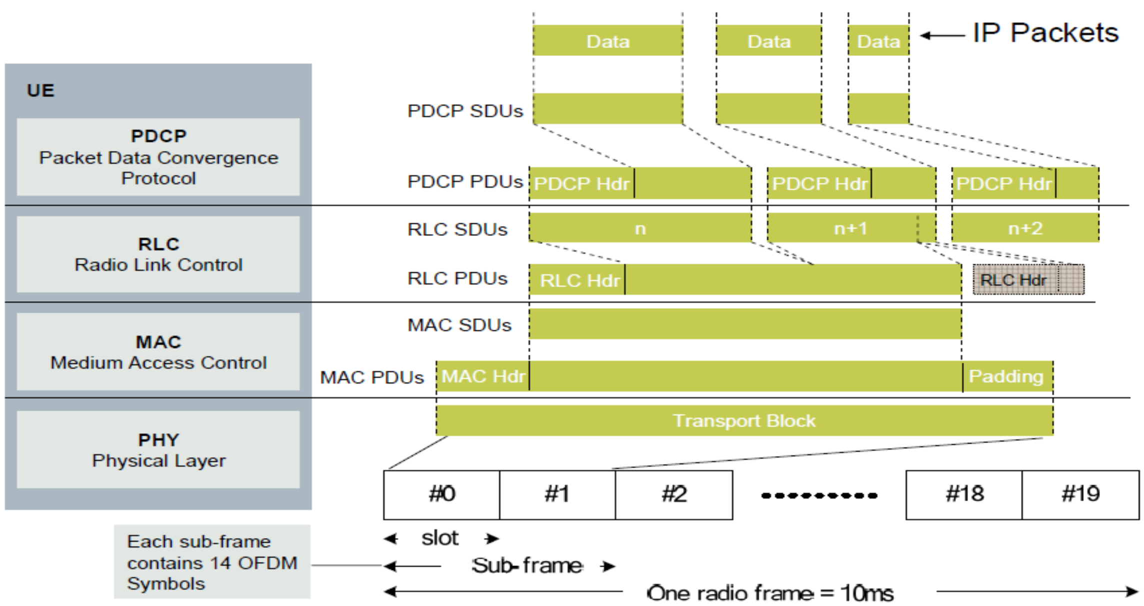

Figure 4.22 Time domain view of the LTE downlink [35]

4.12 Life of an LTE Packet

Here we trace the flow of a packet through the sub-layers of the LTE stack.

4.12.1 The downlink direction (from network to terminal).

The flow starts by delivering a transport block from the physical layer to the MAC layer that contains the information that was decoded off the air in the previous radio sub frame. There can be an arbitrary relationship between what is in the transport block and the actual packets that are being delivered to higher layers.[35] The transport block, delivered from the PHY to the MAC, contains data from the previous radio sub frame. It may contain multiple or partial packets, depending on scheduling and modulation. The figure below shows this flow.

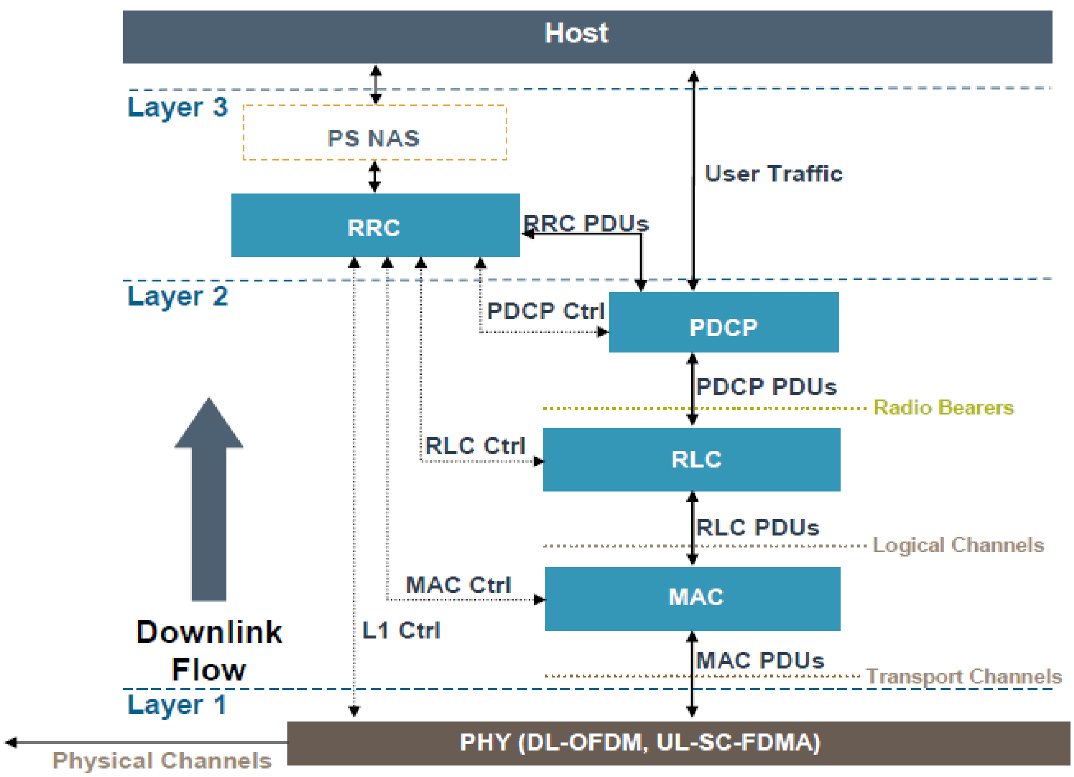

Figure 4.23 Life of an LTE Packet: Downlink [35]

a Layer 2 MAC sub Layer

The Medium Access Control (MAC) radio protocol sublayers main purpose is to provide an efficient coupling between RLC services and the physical layer. It is responsible for managing the hybrid ARQ function, which is a transport-block level automatic retry. It also performs the transport as a logical mapping which can be defined as a function that breaks down different logical channels out of the transport block for the higher layers. Its main functions include the following:

- Multiplexing of Radio Bearers (Signaling and Data Bearers) : Mapping between Logical channels and transport channels, logical channel identification and transport format selection, reference signals, synchronization signals, broadcast channel and HARQ indicator channel.
- Dynamic Scheduling (UL/DL): decides when, where and what kind of data is scheduled and to which UE the data is sent.
- Timing Advance: for synchronization of the mobile transmission.
- MAC Control Messaging: PDCCH indicate which resource blocks are allowed to use in the uplink direction (Uplink packet scheduling).

b RLC Sublayer [25]

The main function of the Radio Link Control sublayer is to receive and deliver data packet to its peer RLC entity. There are three different transmission modes assigned to different logical channels depending on the type of information they carry:

- Transparent Mode (TM): used for general information, it does not alter the upper layer data, no RLC header, just forwards the data.
- Unacknowledged Mode (UM) used for signaling and Acknowledge Mode (AM) used for user data.

The main functions of the RLC layer are: **[25]**

- Segmentation: decides on PDU sizes depending on QoS and available resources
- Automatic Repeat request (ARQ): ensures the correct delivery of data for AM mode over the air interface.

- Reassembly: (UM and AM) it needs a RLC header in the PDU to know the order of the sequence and Status Report (AM): indicates if retransmission was lost.

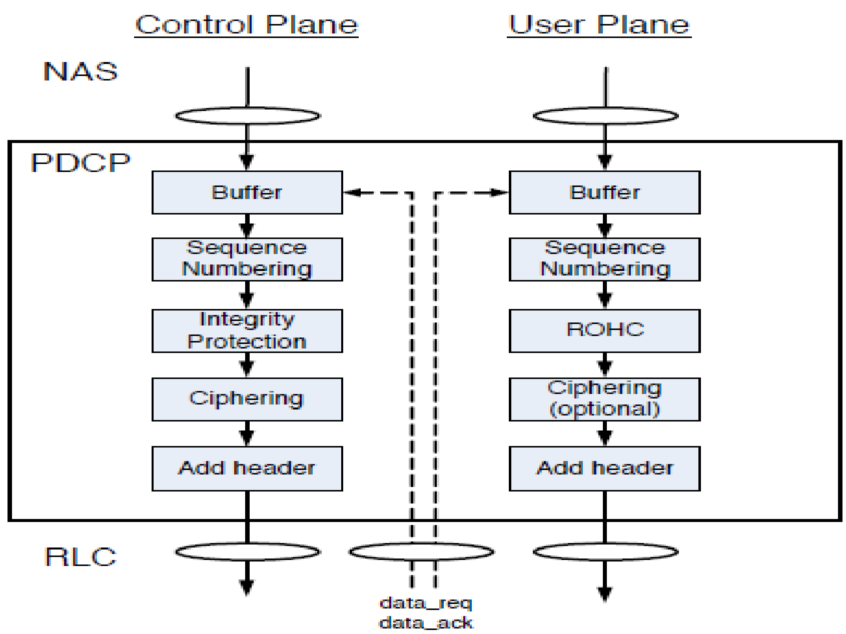

Figure 4.24 PDCP Layer main packet operations Ref [25]

c PDCP Sublayer.

The main functions of the PDCP Layer are:

- Encapsulation of higher layer protocols
- Packet handling: Buffers packets until they are scheduled by lower layers
- Packet forwarding: Lossless Retransmission of PDCP SDU to support Handover

4.12.2 Hybrid ARQ

The Hybrid Automatic Repeat request (HARQ) process, which is done in combination between the MAC and the PHY layers, it retransmits transport blocks (TBs) for error recovery. The PHY layer performs the retention and re-combination (incremental redundancy) and the MAC layer performs the management and signaling. The MAC layer indicates a NACK when there’s a transport block CRC failure; the PHY layer usually indicates that failure.

Retransmission is done by the eNodeB or the sender on the downlink using a different type of coding. The coding is sent and maintained in buffers in the eNodeB. Eventually, after one or two attempts, there will be enough data to reconstruct the signal. In HARQ operation, the retransmission does not have to be fully correct. It has to be correct enough that it can be combined mathematically with the previous transport block in order to produce a good transport block.

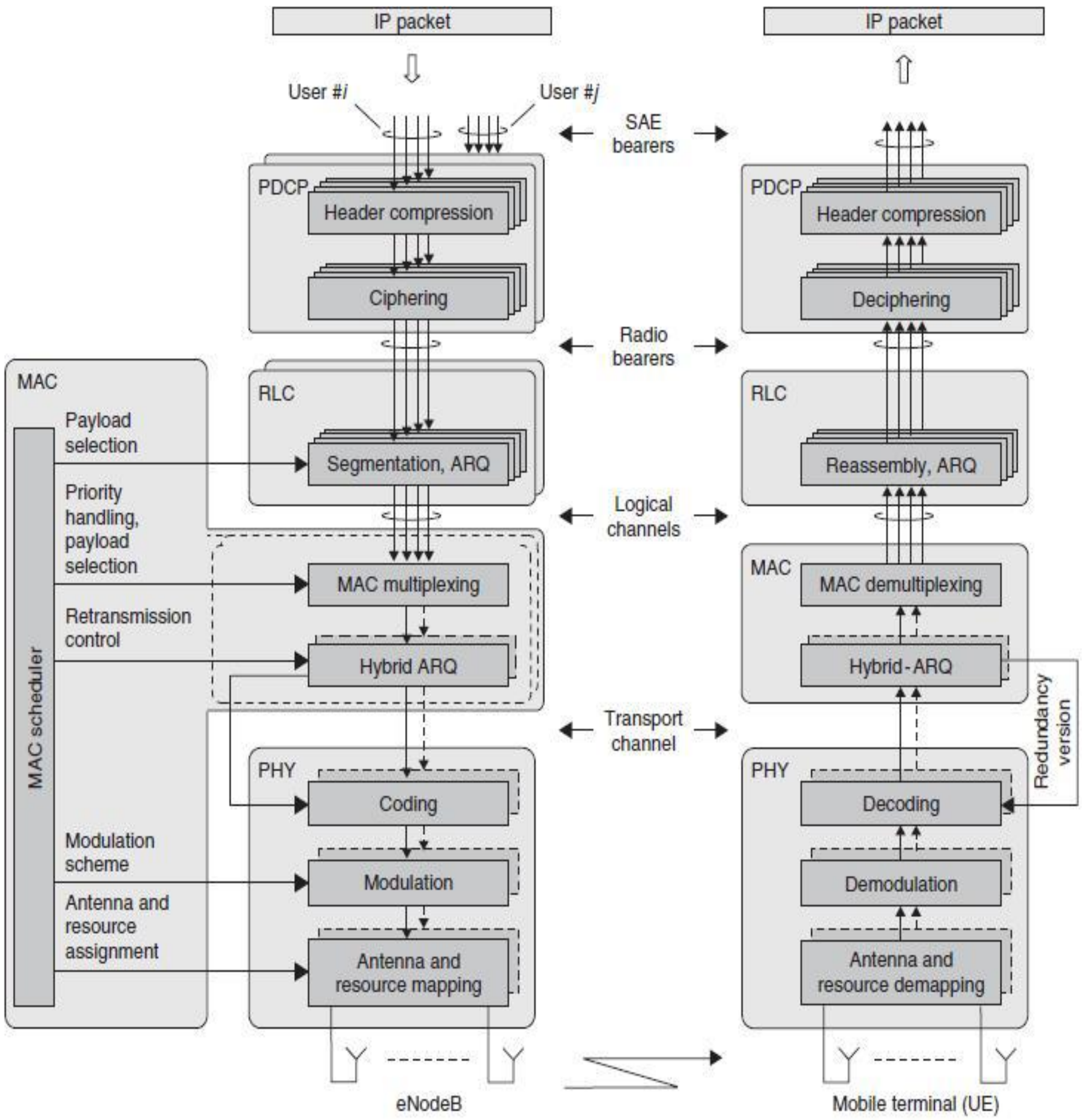

Figure 4.25 LTE downlink general architecture [34]

4.12.3 Life of an LTE Packet: Uplink.

LTE processes on the uplink side are often similar to processes on the downlink side. Only that the peak data rate is half that of downlink; access is granted by the eNodeB; there are changes in logical channels and transport channels; and random access is used for initial transmissions.
The PHY uses SC-FDMA for the uplink because it has a lower peak-average ratio, which allows a more power-efficient transmitter in the UE.

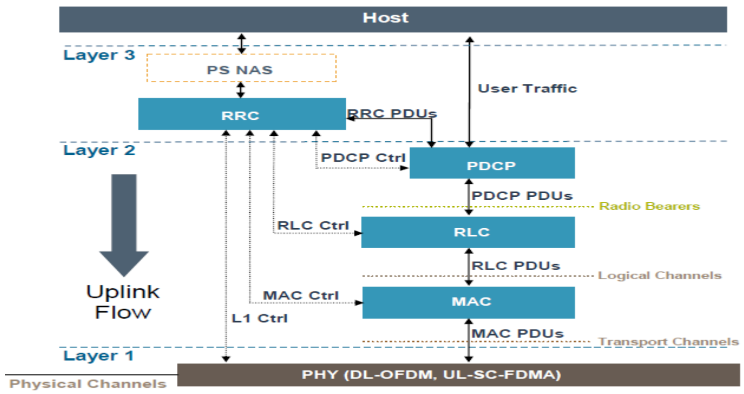

Figure 4.26 Life of an LTE Packet: Uplink

PDCP Layer- Uplink processing includes header compression and encryption.
RLC Layer- headers are applied. There is still a need to support transparent mode, unacknowledged mode and acknowledged mode. The uplink process concatenates rather than segments the SDUs into transport blocks.
Segmentation is only done when it is needed to fit SDUs into a transport block.

4.12.4 MAC Uplink Channel Mapping

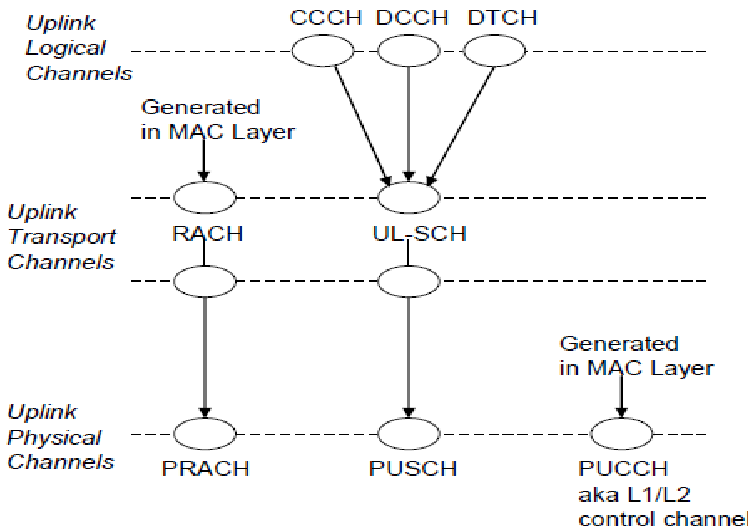

Figure 4.27 uplink channel mapping [35].

All MAC transmissions on the UL-SCH must be scheduled by the Random Access Channel (RACH) procedure. When the UE is not connected, no transmit slots are ever scheduled. The RACH provides a means for disconnected devices to transmit. Transmitting on the UL-SCH requires a resource allocation from the eNodeB, and time alignment to be current. Otherwise the RACH procedure is required.

a Layer 3: RRC Layer.

Radio Resource Control (RRC) is a protocol of the control plane layer that handles the UE management and controls Layer 2 and Layer 1 parameters as well as UE - eNodeB signaling. RRC Layer interact with the Lower and Upper layers in an intelligent manner as shown in the figure below.

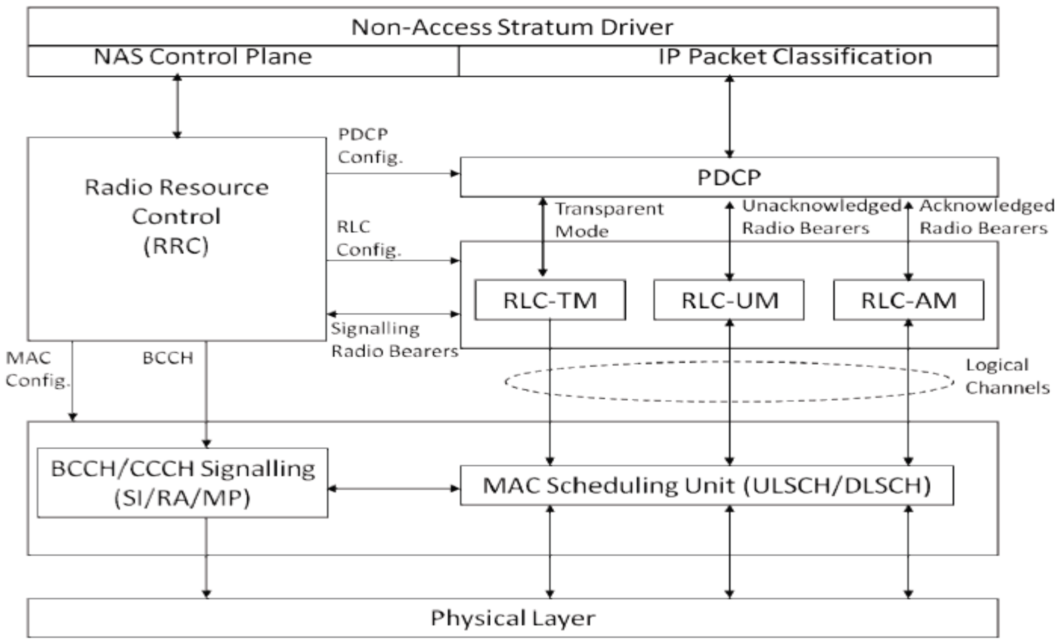

Figure 4.28 RRC control over the different Layer [34]

The functions of RLC depends on its state in relation to the UE.

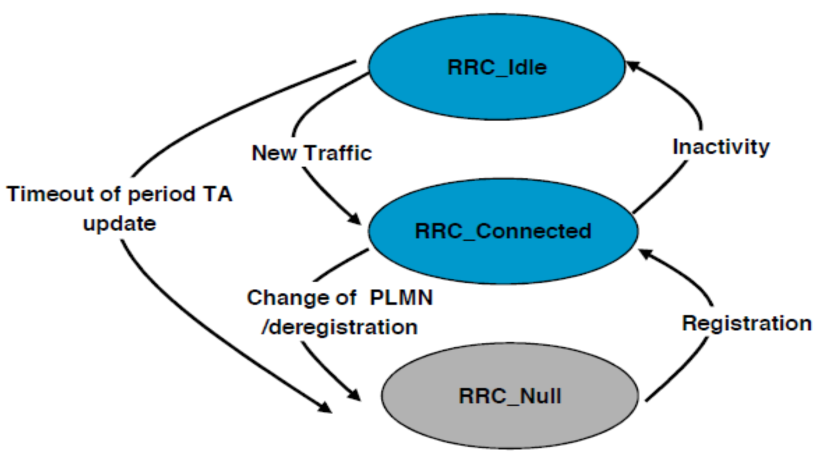

Figure 4.29 RLC states [34]

b Applicable to both States: [34]

Broadcast of System information (SI): informs the UE about the different configuration parameters necessary to use the transport channels and for mobility purpose.

c RRC Idle mode: [34]

Paging: allows the UE to detect an incoming call by monitoring the paging channel.
UE cell selection and re-selection, controlled by the parameters of the SI.

d RRC Connected mode: [34]

RRC Connection management between UE and eNodeB: Radio Resource allocation for the UE and configuration of signaling Radio Bearer (SRB) to send over the control channels.
Security functions: such as Key management.
Quality of service (QoS) management: establishment, maintenance and release of Radio bearers (point to point, MBMS services).

I start by comparing the convolution codes and forward error correction (FEC) error control techniques.

Parameters:

Huffman coding algorithm is used, the code dictionary is generated using the HUFFMANDICT function.
After convolution encoding, we also add the BCH encoding of the data.
We use QAM modulation with M=16 then pass through AWGN channel.

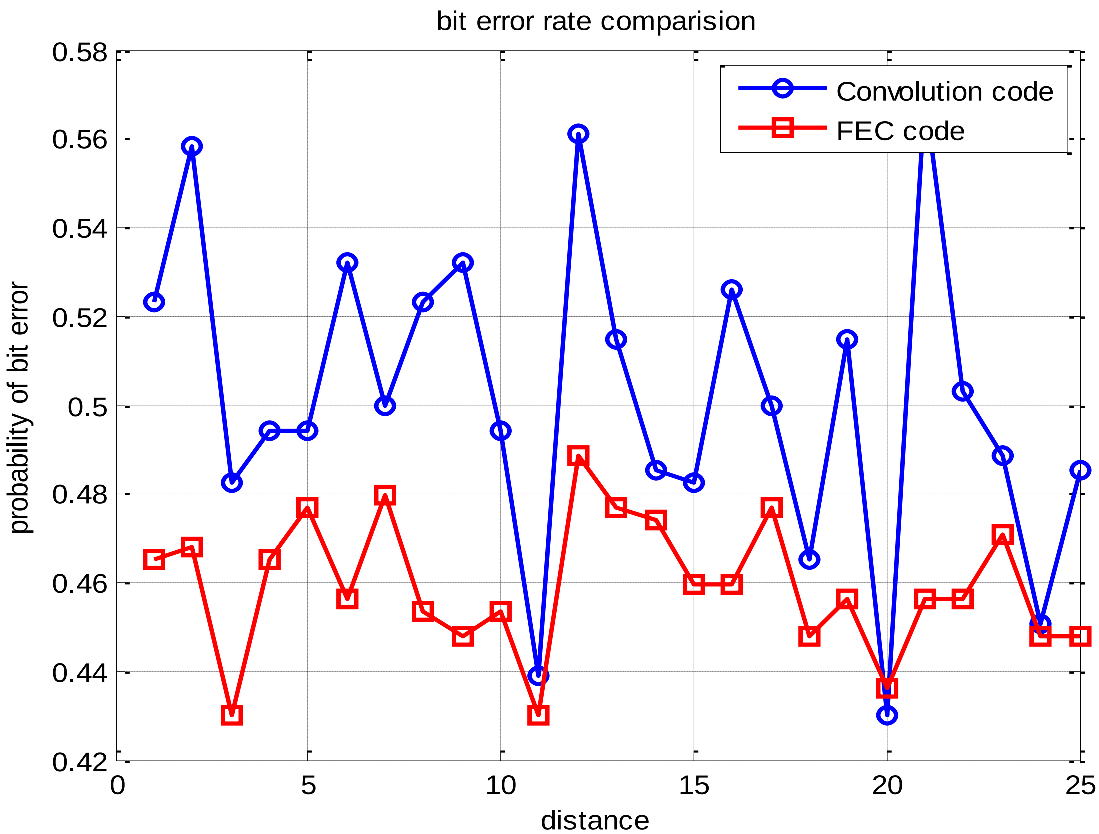

Figure 5.1 bit error comparison of FEC and Convolution

We clearly see that for the same signal power, the bit error rate for FEC is less than that of the simple convolutional encoder which makes FEC more preferred to convolutional encoder due to less error for a given signal power.

I now look at the simulation of the ARQ protocol before applying it in the communication chain for comparison purposes. The number of frames is 8 and the number of bits per frame is 4, in that we create 32 binary message to be transmitted in form of 0s and 1s.the message is transmitted through a binary symmetric channel.10 messages are received and acknowledged the rest are not as the sliding window slides forwards as the messages are send from the transmitter's buffer. The whole process is shown in the figure that follows.

Parameters:

You enter the number of frames and the number of bits per frame as you wish.

The sliding window size is 1.A binary symmetric channel is used in this case.

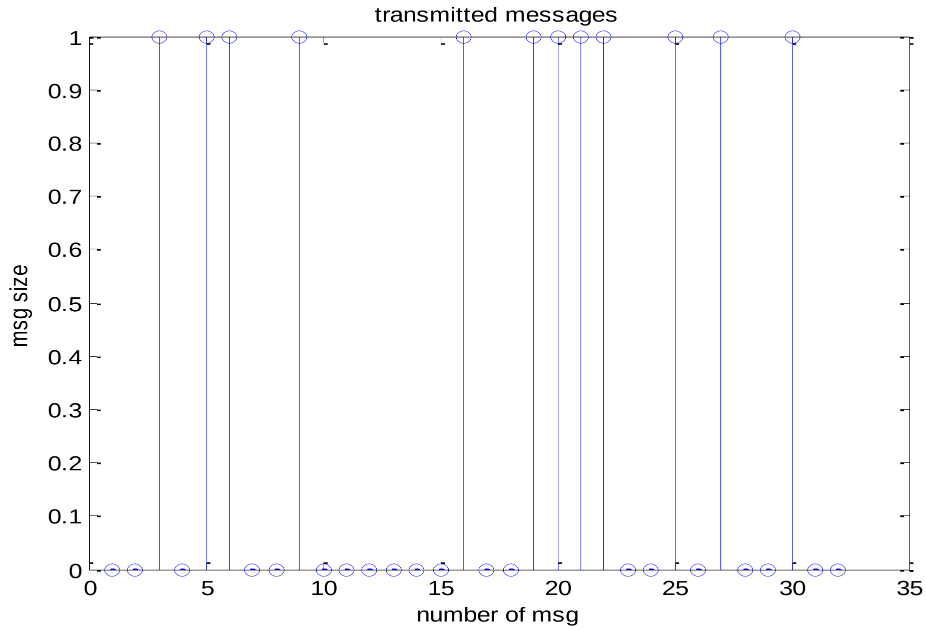

Figure 5.2 ARQ transmitted messages

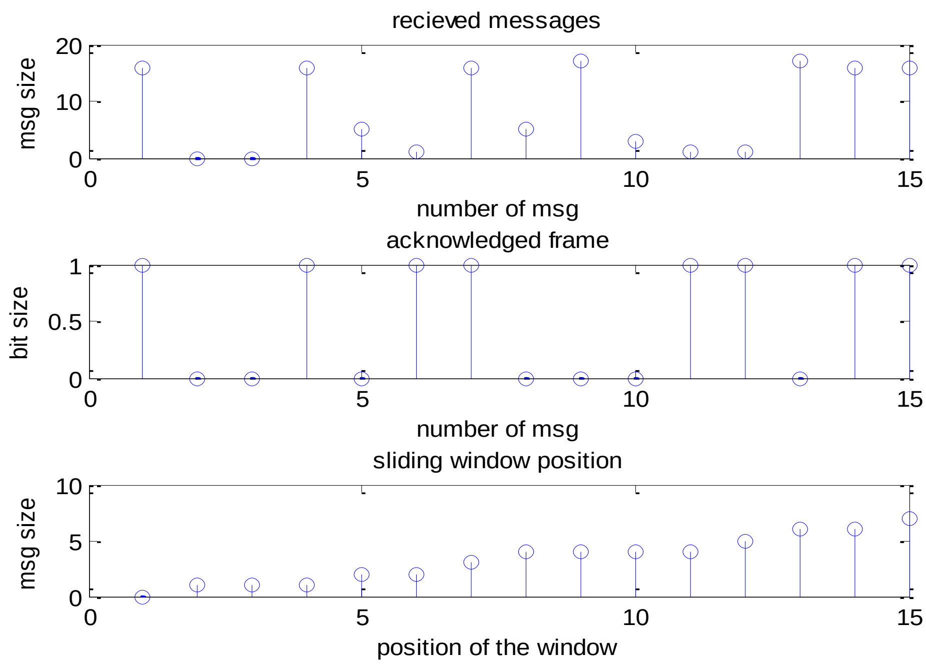

Figure 5.3 ARQ received, acknowledged and the sliding window position after transmission

In the next simulation I look at the application of ARQ protocol with changing the number of maximum retransmissions.as it will be seen in the diagram the quality of information received depends on the maximum number of retransmission specified.as the number of retransmission increases the quality of information received gets better.

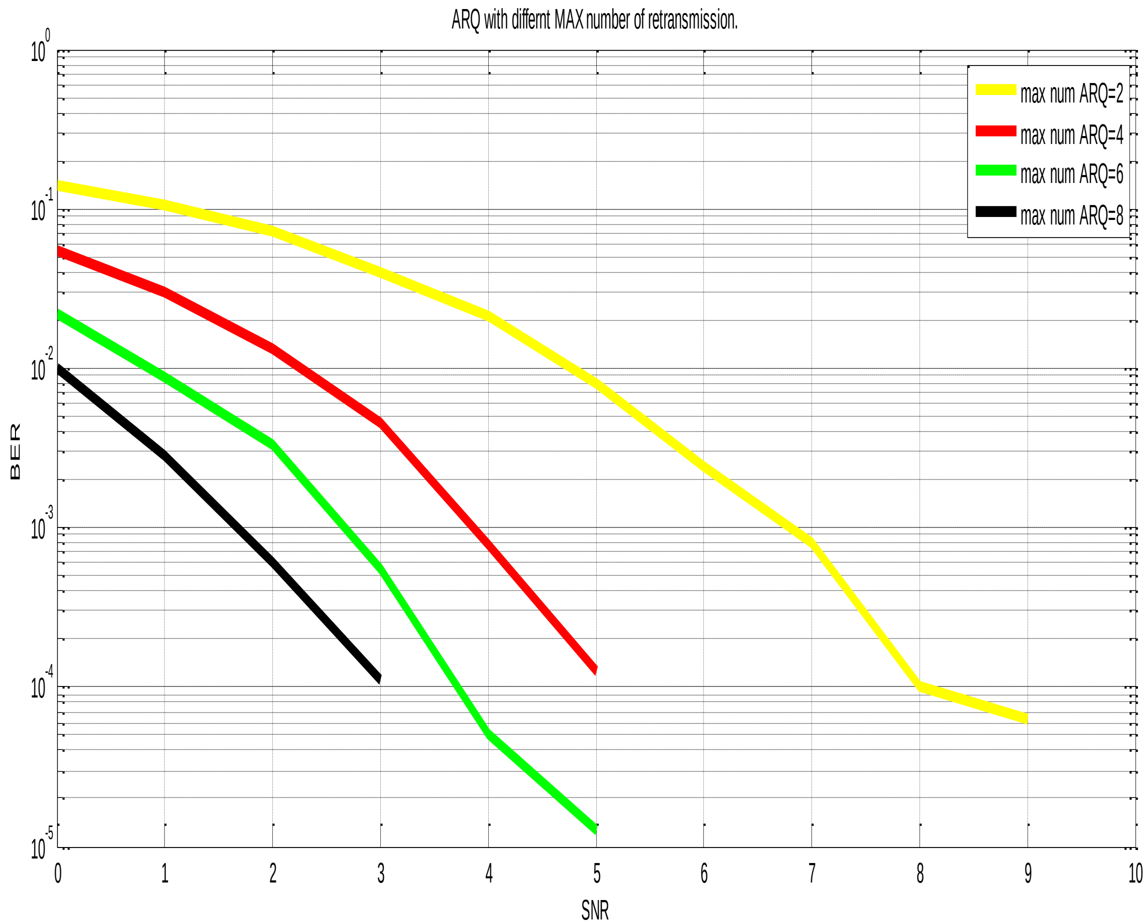

Figure 5.4 ARQ with different number of maximum retransmission

In the next simulation I look at the comparison when transmission is done while applying ARQ and with simple convolution codes.

Parameters:

k = 4; % Lengths of code words and messages.

Dmin = 7; % Minimum distance.

We also use BPSK modulation

We perform the constellation mapping and then call the function: found in the soft copy of this thesis. This functions define the ARQ protocol which is simply a loop.

```
testHARQfunction(msg_orig,msg_tx,num_block,length_block,num_maxARQ,snr_array(
i),n,k);
```

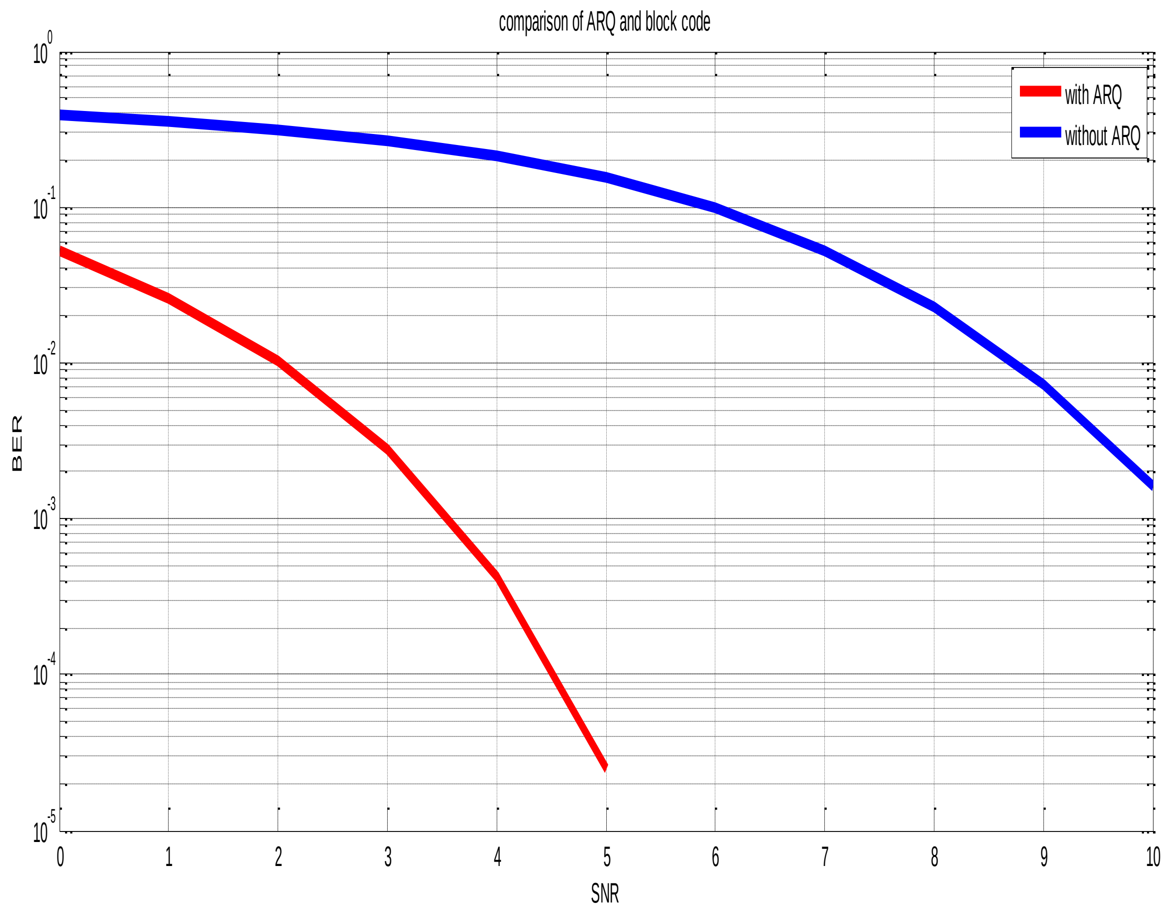

Figure 5.5 comparison of transmission with and without ARQ

In the following simulation I look at the comparison between the ARQ scheme and the HARQ scheme in general. I will compare two main things, the Bit Error Rate and the rate of transmission which determines how fast the information is processed and transmitted when we apply the two transmission schemes.

Parameters used are:

```
SNR=-5:5:25; you can define your own range for a better comparison.
 Nt =2; transmitting antenna in the case of MIMO.
 Nr =2; receiving antennas.
 M=4; number of constellation
SymTime =10;
PacketNum =1000;
```

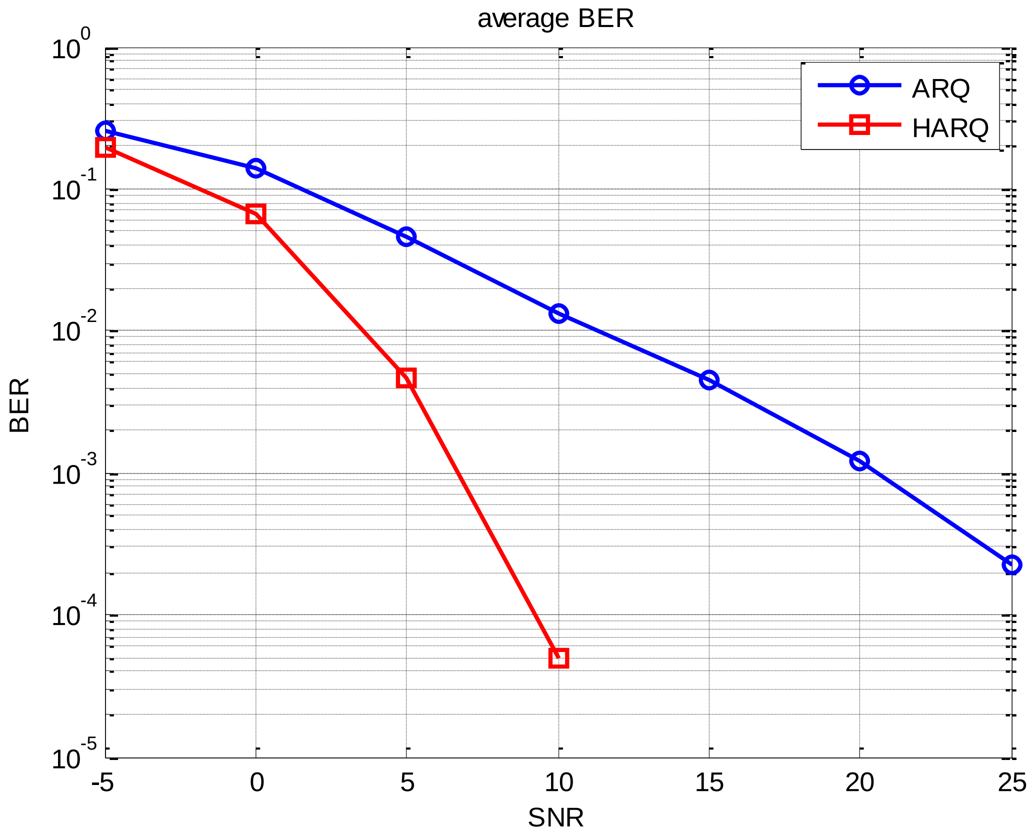

Figure 5.6 average BER comparison between ARQ and HARQ schemes

As we can clearly see in the figure above, HARQ schemes are far much better than the ARQ schemes in terms of error par bit. For a given SNR the BER for HARQ is much better than the BER for the ARQ protocol. This makes it favorable to the LTE technology which requires minimum errors or no errors or sometimes negligible errors during transmission.

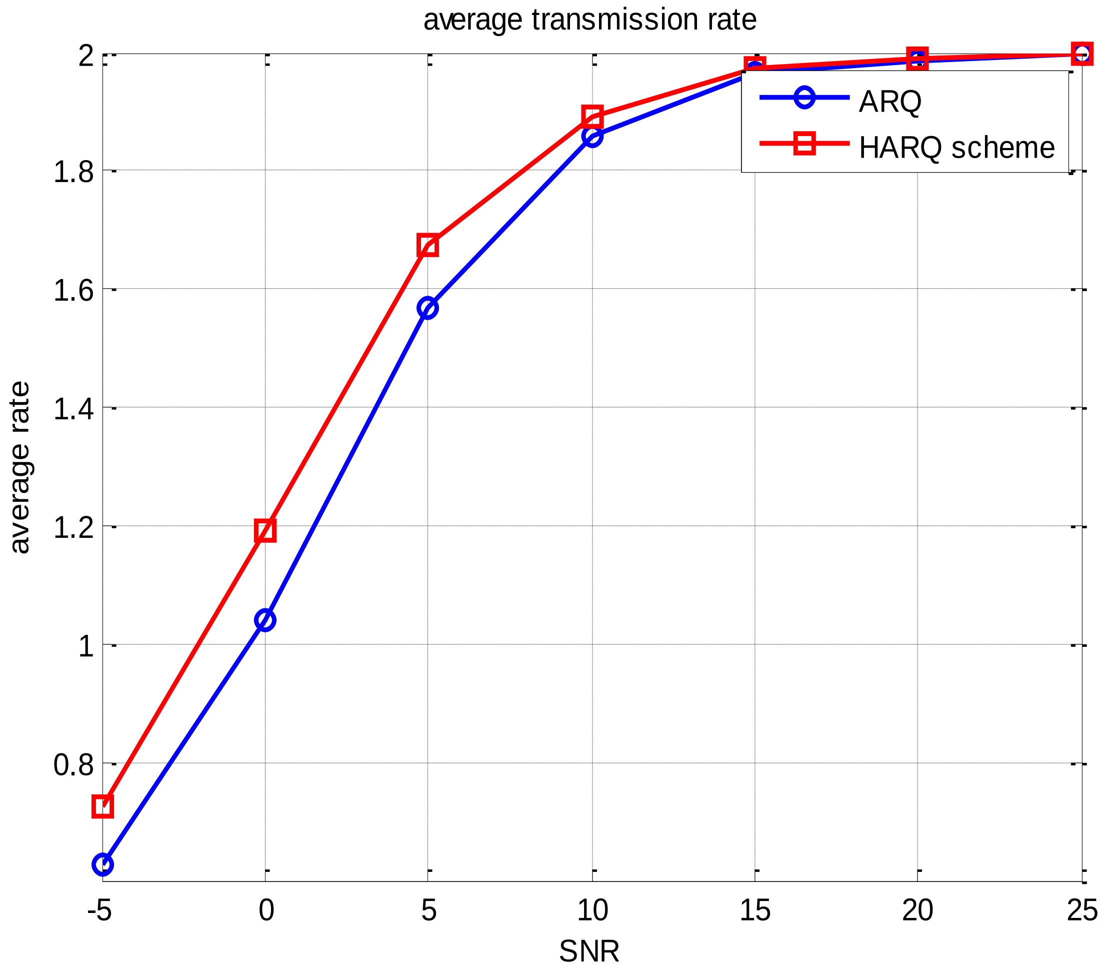

Figure 5.7 average rate transmission of ARQ and HARQ scheme

As seen in the figure the transmission rate of HARQ is faster than that of ARQ protocol. This is due to the combined efforts of both ARQ as a retransmission mechanism and FEC as the anchor in faster detection of the errors.

In this thesis I have looked at the various techniques used in the detection and correction of errors during transmission of the information from the source to the destination. The first methods I looked at was FEC which in addition to the convolution encoder and the Viterbi decoding it adds the CRC check to detect the errors. This is why in the simulation part while I compared the BER of simple convolution code and that of the FEC, we found that FEC performs much better due to this additional utility of the CRC check.

The following methods that I looked at were the ARQ protocol and the HARQ protocol. In my case I considered the general case using the general algorithm of these protocols. I did not consider the types but in future if one wants more convincing results it will be better if they consider simulating the different types that exist of these protocols.

It has been seen that the HARQ protocol is better than the ARQ protocol in that it gives a better BER and its transmission rate is better that that of the ARQ protocol. I also described the HARQ process as it is applied in the LTE technology. As to the simulation part in the LTE technology, there are built in HARQ functions, for uplink and downlink simulation.

I also described the LTE technology according to the 3GPP specifications in its releases. It is a very simple and very interesting technology based on its architecture that has been really simplified and allows a variety of applications that are user friendly and simple to understand for a better connectivity on the internet for a speed and reliable communications.it is indeed a milestone in the technological evolution bringing information at our doorstep. Just a click away.

Figure 1 shows an example of a digital communication chain, including the main steps that take place during the process.

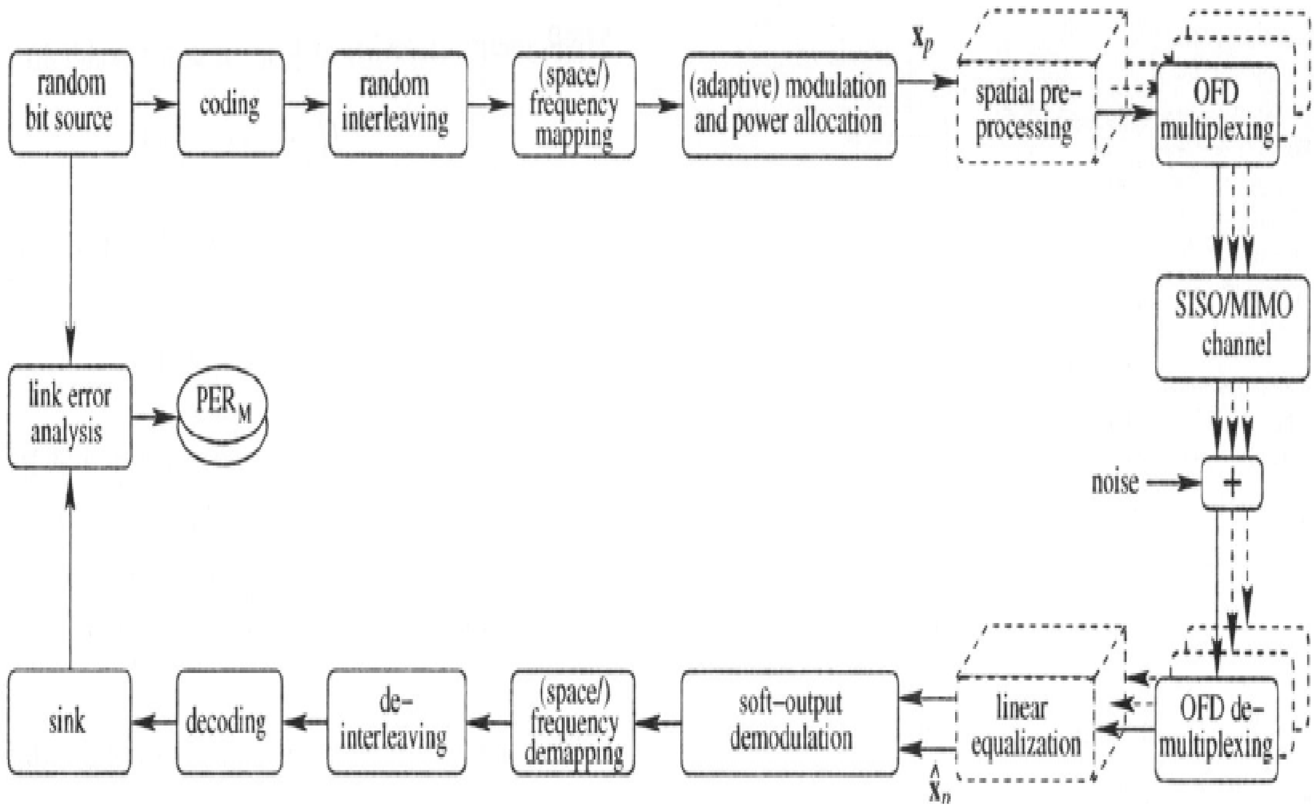

Figure A.1: Example of a digital communication chain [36]

This section analyzes with more detail how the mutual information value is calculated.

Mutual Information Calculation for a SISO/MIMO System

The continuous definition of MI is:

$$MI \equiv I(X;Y) = \sum_i \int p(y \mid x_i) P(x_i) \log_2 \frac{p(y \mid x_i)}{p(y)} dy \qquad \text{(A.1)}$$

According to this, mutual information per symbol can be written as:

$$MI = \log_2 N - \frac{1}{N}\sum_{i=1}^{N} E\left\{\log_2\left(1+\sum_{K=1,k\neq i}^{N}\exp\left[-\frac{\left|x_i - x_k + w\right|^2 - \left|w\right|^2}{\sigma^2}\right]\right)\right\} \tag{A.2}$$

Multiple-input Multiple-output Technique

Multiple-input multiple-output communication techniques have been an important area of focus for the next-generation wireless systems because of their potential for high capacity, increased diversity, and interference suppression [36].

A MIMO channel with *nT* transmitters and *nR* receivers is typically represented as a matrix H of dimension $nR \times nT$, where each of the coefficients $[H]i;j$ represents the transfer function from the *jth* transmitter to the *ith* receiver, as shown below:

$$\mathbf{H} = \begin{bmatrix} h_{1,1} & h_{1,2} & \cdots & h_{1,nT} \\ h_{2,1} & h_{2,2} & \cdots & h_{2,nT} \\ \cdots & & & \\ h_{nR,1} & h_{nR,2} & \cdots & h_{nR,nT} \end{bmatrix} \tag{A.3}$$

We denote the signal or symbol transmitted from the *jth* transmitter as *xj*, and collect all such symbols into an *nT* -dimensional vector x. With this notation, the matrix model of the channel is

$$y = Hx + n; \tag{A.4}$$

Where n is a vector of additive noise, and y is the vector of received data, with an element in n and y for each receive antenna.

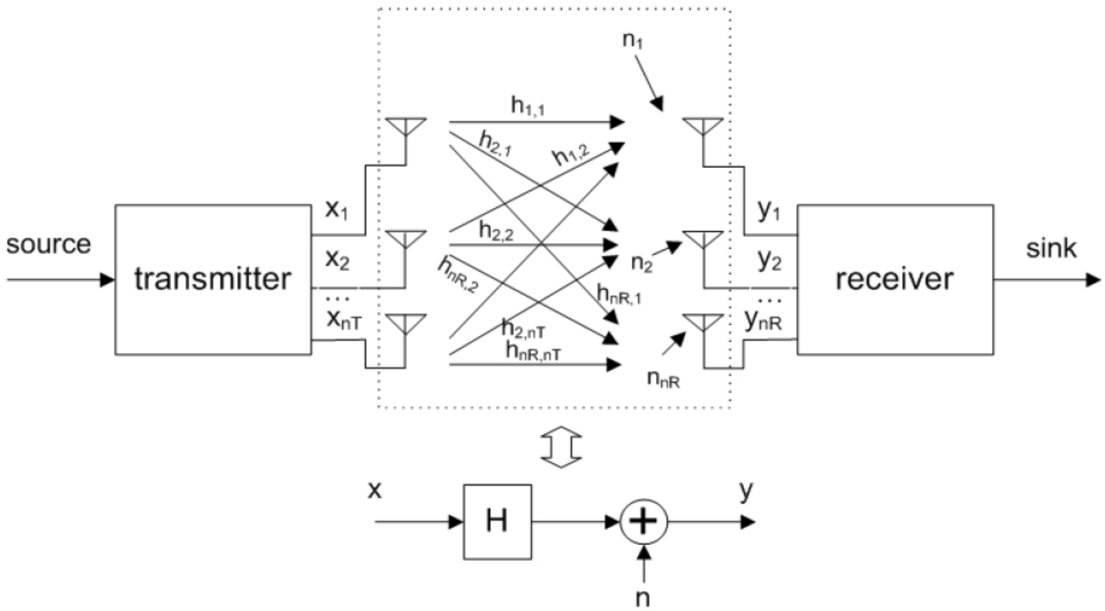

Figure: mimo system.

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