NEAR EAST UNIVERSITY

Faculty of Engineering

Department of Electrical and Electronic Engineering

MOBILE SOUND TRANSFORMATION IN GSM

Graduation Project
EE – 400

Student: Mahmoud El-Qasass (20033013)

Supervisor: Msc. Jamal Fathi

Nicosia - 2007

ACKNOWLEDGMENT

First of all I would like to thank ALLAH for guiding me through my study.

More over I want to pay special regards to my family who are enduring these all expenses and supporting me in all events. I am nothing without their prayers. They also encouraged me in crises. I shall never forget their sacrifices for my education so that I can enjoy my successful life as they are expecting.

Also, I feel proud to pay my special regards to my project adviser "Msc. Jamal Fathi". He never disappointed me in any affair. He delivered me too much information and did his best of efforts to make me able to complete my project. I would like to thank Assoc. Prof. Dr. Adnan Khashman for his advices in each stag of our undergraduate program and how to choose the right path in life.

The best of acknowledge, I want to honor those all persons who have supported me or helped me in my project. I also pay my special thanks to my all friends who have helped me and gave me their precious time to complete this project. Also my especial thanks go to my friends, Mohammad Al-Mustafa, Malik Al-Mustafa and

Bassem Al-Soudi.

ABSTRACT

Recently, communicating with others became the most interest in the worldwide. GSM is a global Telecommunication system that is being used in the earlier years, developed in Europe, and enhanced to meet the ultimate specifications in all over the globe. The aim of this project is to explain how the sound transformation process operates in the GSM system, Methodology with full detailed information about the transmission and reception process being accomplished. Also determining an identification of the GSM services that are provided for the subscriber.

CONTENTS

	ACKNOWLEDGMENT	**i**
	ABSTRACT	**ii**
	CONTENTS	**iii**
	INTRODUCTION	**ix**
1.	**INTRODUCTION TO SOUND TRANSFORMATION**	**1**
	1.1 Introduction	1
	1.2 Electromagnetic Spectrum	1
	1.3 What is Antenna?	2
	1.3.1 Dipole Antenna	2
	1.4 Transmission Media	4
	1.4.1 Guided Media	4
	1.4.1.1 Twisted Pair	5
	1.4.1.2 Coaxial Cable	7
	1.4.1.3 Fiber-Optic Cable	8
	1.4.1.3.1 Types Optical fibers:	9
	1.4.1.3.2 How Does an Optical Fiber Transmit Light?	9
	1.4.1.3.3 A Fiber-Optic Relay System	10
	1.4.1.3.4 Advantages of Fiber Optics	10
	1.4.1.3.5 How Are Optical Fibers Made?	11
	1.4.1.3.5.1 Making a Pre-form Glass Cylinder	12
	1.4.1.3.5.2 Drawing Fibers from the Preform Blank	13
	1.4.2 Unguided Media	13
	1.4.2.1 Ground Wave Propagation	14
	1.4.2.2 Sky Wave	14
	1.4.2.3 Line-Of-Sight Propagation	15
	1.5 Summary	16
2.	**GSM OVERVIEW**	**17**
	2.1 Introduction	17
	2.2 History of GSM	17
	2.3 GSM Network Component	18

2.3.1 Switching System (SS) Components 20
2.3.1.1 Mobile Switching Center (MSC) 20
2.3.1.2 Gateway MSC 21
2.3.1.3 Home Location Register (HLR) 21
2.3.1.4 Visitor Location Register (VLR) 21
2.3.1.5 Authentication Center (AUC) 21
2.3.1.6 Equipment Identity Register 22
2.3.2 Base Station System (BSS) Components 22
2.3.2.1 Transcoder Controller (TRC) 22
2.3.2.2 Base Station Controller (BSC) 22
2.3.2.3 Base Transceiver Station (BTS) 22
2.3.3 Operation and Support System (OSS) 23
2.3.3.1 Operation and Maintenance Center (OMC) 23
2.3.3.2 Network Management Center (NMC) 23
2.3.4 Mobile Station (MS) 23
2.4 GSM Geographical Network Structure 24
2.4.1 Cell 24
2.4.2 Location Area 24
2.4.3 MSC Services Area 25
2.4.4 Public Land Mobile Network (PLMN) Service Area 25
2.4.5 GSM Service Area 25
2.5 GSM Frequency Bands 25
2.5.1 GSM 900 25
2.5.2 GSM 1800 25
2.5.3 GSM 1900 26
2.6 Modulation Method 26
2.7 Transmission Problems in GSM Network 27
2.7.1 Path Loss 27
2.7.2 Shadowing 27
2.7.3 Multipath Fading 27
2.7.4 Rayleigh Fading 28
2.7.5 Time Dispersion 28

2.7.6 Time Alignment 28

2.8 Summary 28

3. MODULATION **29**

3.1 Modulation 29

3.2 Amplitude Modulation (AM) 30

3.3 Frequency Modulation (FM) 30

3.4 Phase Modulation (PM) 31

3.5 Coherent and Incoherent Systems 33

3.6 Frequency Shift Keying (FSK) 34

3.7 Minimum Shift Keying (MSK) 35

3.7.1 Gaussian Minimum Shift Keying (GMSK) 35

3.8 Phase Shift Keying (PSK) 36

3.8.1 Binary Phase Shift Keying (BPSK) 37

3.8.2 Quadrature phase shift keying (QPSK) 38

3.8.3 Л/4Quadrature Phase Shift Keying (Л/4-QPSK) 40

3.8.4 Offset Quadrature Phase Shift Keying (O-QPSK) 41

3.9 Summary 41

4. GSM TRANSMISSION PROCESS **42**

4.1 Introduction 42

4.2 Analog to Digital (A/D) Conversion 42

4.2.1 Sampling 43

4.2.2 Quantization 43

4.2.3 Coding 43

4.3 Speech Coding 43

4.3.1 The Dimension of Performance in Speech Compression 44

4.3.1.1 Speech Quality 44

4.3.1.2 Bit Rate 44

4.3.1.3 Communication Delay 45

4.3.1.4 Complexity 45

4.3.2 Speech Coders Categories 45

4.3.2.1 Waveform Coders 46
4.3.2.2 Vocoders 46
4.3.3 Speech Coding in GSM 46
4.4 Channel Coding 47
4.4.1 Convolutional Encoders 47
4.4.2 Channel Coding Process in GSM 48
4.4.2.1 Cyclic Encoding 49
4.5 Interleaving 50
4.6 Burst Format 51
4.6.1 Normal Burst 52
4.7 Ciphering 53
4.8 Modulation 54
4.9 Multipath Channels 56
4.9.1 Fading Types 57
4.9.1.1 Large Scale Fading 57
4.9.1.2 Small Scale Fading 58
4.9.2 The Propagation Mechanism 59
4.9.2.1 Reflection 59
4.9.2.2 Diffraction 59
4.9.2.3 Scattering 59
4.9.3 Noise and Interference in Multipath Channels 59
4.9.4 Doppler Shift 60
4.9.5 Rayleigh and Rician Distribution 61
4.9.5.1 Rayleigh Fading Distribution 61
4.9.5.2 Rician Fading Distribution 61
4.10 Equalizer 62
4.11 Summary 62

5. METHODOLOGY **63**
5.1 Introduction 63
5.2 Transmission Procedure 64
5.2.1 Speech Coding 64

5.2.2 Channel Coding 65
5.2.3 Interleaving 67
5.2.4 Burst Format 68
5.2.5 Modulation 69
5.3 Reception Procedure 71
5.3.1 Demodulation 71
5.3.2 Burst Deformat 72
5.3.3 Deinterleaving 72
5.3.4 Channel Decoding 73
5.3.5 CRC Error Detector 74
5.3.6 Aggregation Process 74
5.4 Summary 75

6. GSM SUBSCRIBER SERVICES 76
6.1 Introduction 76
6.2 Personal Communications Service (PCS) 76
6.3 GSM Services 77
6.3.1 Speech Services 78
6.3.1.1 Telephony 78
6.3.1.2 Emergency Calls (with/without SIM Card Inserted in MS) 78
6.3.1.3 Short Message Service Point to Point 78
6.3.1.4 Short Message Cell Broadcast 79
6.3.1.5 Advanced Message Handling Service 79
6.3.1.6 Duel Personal and Business Numbers 79
6.3.1.7 Dual-Tone Multi frequency (DTMF) 79
6.3.1.8 Voice Mail 79
6.3.1.9 Fax Mail 79
6.3.2 Data services 80
6.3.2.1 Videotex Access 80
6.3.2.2 Teletex 80
6.3.2.3 Alternate Speech and Facsimile Group III 80
6.3.3 Supplementary Services 81

6.3.3.1 Number Identification 81
6.3.3.2 Call Barring 81
6.3.3.3 Call Forwarding 81
6.3.3.4 Call Hold 81
6.3.3.5 Call Waiting 82
6.3.3.6 Advice of Charge (AoC) 82
6.3.3.7 Multi-Party 82
6.3.3.8 Closed User Groups (CUGs) 82
6.4 Summary 82

7. CONCLUSION **83**
8. REFRERENCES **84**

INTRODUCTION

Wireless communications is enjoying its fastest growth period in history, due to enabling technologies. The ability to provide wireless communications to an entire population was not even conceived until Bell Laboratories developed the cellular concept in the 1960s and 1970s. Since the mid 1990s, the cellular communications industry has witnessed explosive growth. The worldwide cellular and personal communication subscriber base surpassed 600million users in late 2001, and the number is projected to reach two billion subscribers at the end of 2006.

First generation wireless networks are based on analog technology and used FM modulation. The network includes the mobile terminals, the base stations, and mobile switching centers (MSC). The MSC performs all the control functions of the network as well as management functions such as billing and call handling and processing.
First generation analog systems provided analog speech and low rate data transmission between the users. Example on first generation systems is the Advanced Mobile Phone System (AMPS).
Second generation wireless systems employ digital modulation and advanced call processing capabilities. Second generation systems have introduced new network architectures that reduce the computational load on the MSC.

In contrast to first generation systems, which were designed for voice, second generation wireless networks have been specifically designed to provide data services. The network controlling structure is more distributed in second generation systems, since mobile stations assume greater control functions.
GSM is the second generation standard that provides many features over the first generation systems. It uses digital modulation and a remarkable feature of GSM is on-the-air privacy which is provided by the system. This privacy is made possible by encrypting the digital bit stream sent by the transmitter. Maybe the most important feature of GSM is the standards it provides, which make it possible for the service providers and the customers to buy different equipment from different manufacturers and still operating the systems quite easily and reliably.

In this project, chapter one is an introductory about how information being transmitted thought a transmission media that are classified into a guided and unguided medias. Guided medias includes twisted pairs, coaxial cables and fiber optics, unguided medias contains types of electromagnetic propagations which are ground wave propagation, sky wave propagation and line of sight propagation.

Chapter two is an introduction to the GSM network, history, network architecture, components, specifications, and some problems in the transmission process are stated clearly and in some detail.

Chapter three gives an over-view of Modulation techniques which includes Amplitude Modulation, Phase Modulation, and Frequency Modulation in order to get a complete knowledge of basic techniques of Modulation; also it includes explanations about the GMSK that is used as a modulation method for the GSM system.

Chapter four talks about the transmission process in GSM, the steps involved in detail, channel conditions and types, as well as speech coding.

Chapter five describes the project in details, the transmission and reception processes and the radio channel environments that are typically similar to those in the real field.

Finally, Chapter six talks about the services that are can be provided for the subscribers by the GSM.

1. INTRODUCTION TO SOUND TRANSFORMATION

1.1 Introduction

In order to communicate with others using cell phone, you're voice must transfer from you're mobile to the other mobile that you are trying to communicate with, there are a specific processes that you're voice is going through to reach you're voice to the other side (the receiver), the voice signal is transferred in a medium that is suitable for making the transformation process, this chapter will discuss electromagnetic spectrum and the types of transmission medias that electromagnetic waves goes through.

1.2 Electromagnetic Spectrum

All electromagnetic radiation is classified by wavelength and frequency in the Electromagnetic Spectrum as shown in figure 1.1. The frequencies are expressed in cycles per second.

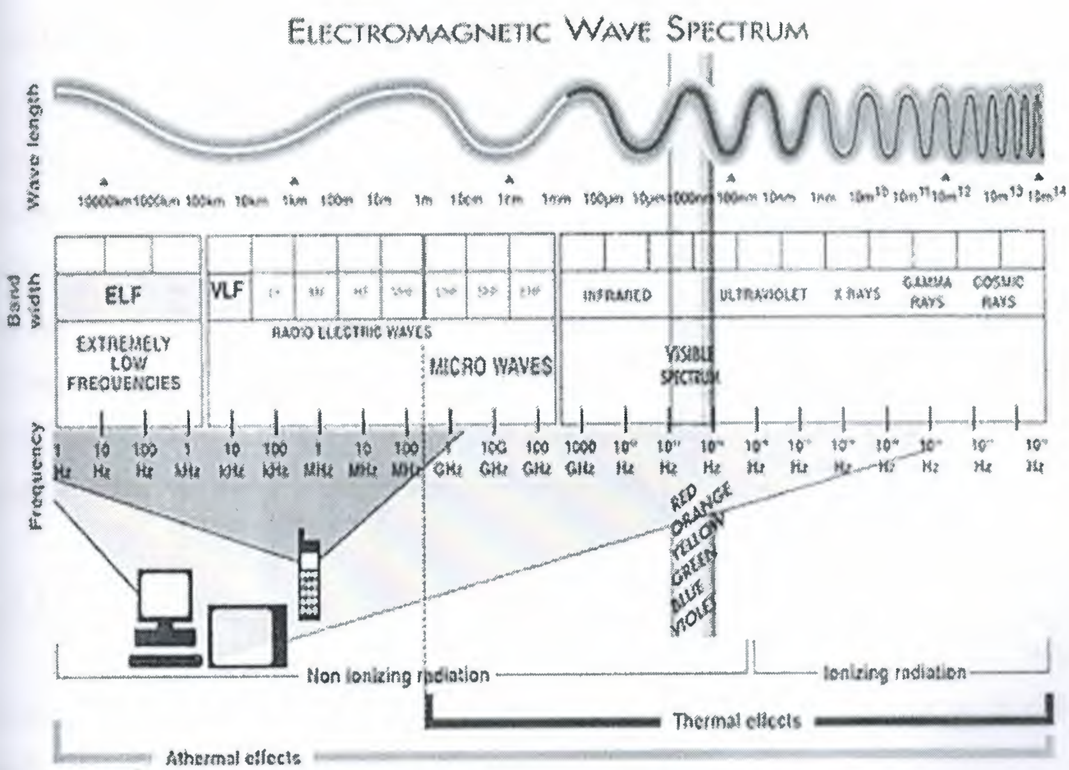


Figure 1.1 Electromagnetic Waves Spectrum

All electromagnetic radiation can be classified as ionizing and non-ionizing radiation. The conventional paradigm holds that ionizing radiation, such as X-rays, causes biological effects through the breaking of molecular bonds, which can damage genetic material such as DNA and non-ionizing radiation can cause effects when the intensity is sufficient to cause heating or thermal effects. The thermal/non-thermal dividing line is used as the basis for present safety standards of electromagnetic radiation. This would mean that EMFs from things such as power lines and cellular phones are safe and have no effect as long as they don't heat you up.

Yet, it is now known that weak electromagnetic fields (weak meaning non-ionizing and below thermal levels) can cause changes in living things. For example, recall that ELF power line AC fields induce weak electrical currents in conducting objects, such as humans and animals. Also, microwave radiation is also known to be dangerous because of its non-thermal effects that produce biological changes. Microwave radiation is emitted by: broadcast radio and TV transmissions, radar, microwave ovens, and cellular phones to name just a few

1.3 What is Antenna?

The beginning and end of a communication circuit is the antenna. The antenna can provide gain and directivity on both transmit and receive. The take-off angle of the antenna is based on the type of antenna, the height of the antenna above ground, and the terrain below and in front of the antenna. The take-off angle will determine the angle of incidence on the ionosphere, which will affect where the signal will be refracted by the ionosphere. There are alots different kinds of Antennas for different positions for example Yagi-Uda Antenna, Horn Antenna, Omni Antenna and the most basic form of the antenna is the Dipole antenna **[12]**.

1.3.1 Dipole Antenna

This is nothing more than a straight piece, as shown in figure 1.2, when voltage is applied to the wire, current flows and the electrical charges pile up in either end.

A balanced set of positive and negative charges separated by some distance is called a dipole. The dipole moment is equal to the charge times the distance by which it is separated.

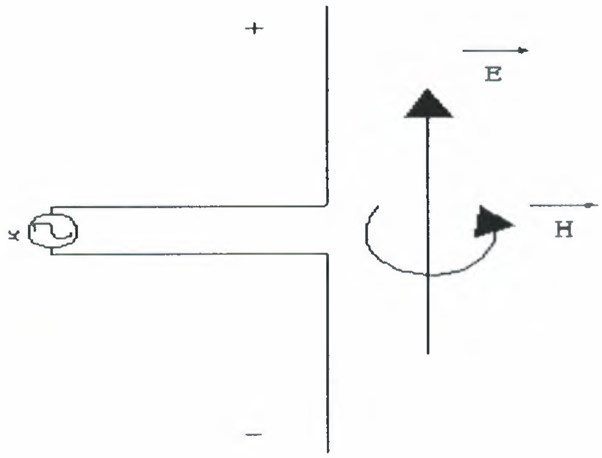


Figure 1.2 Dipole Antennas

When an alternating voltage is applied the antenna, dipole moment oscillates up and down on the antenna, corresponding to the current. The oscillating current creates oscillating electric (E) and magnetic (H) fields which in turn generate more electric and magnetic fields. Thus a outward propagating electromagnetic wave is created. The electric field is oriented along the axis of the antenna and the magnetic field is perpendicular to both the electric field and the direction of propagation. The orientation of the fields is called the polarization as shown in figure 1.3 **[12]**.

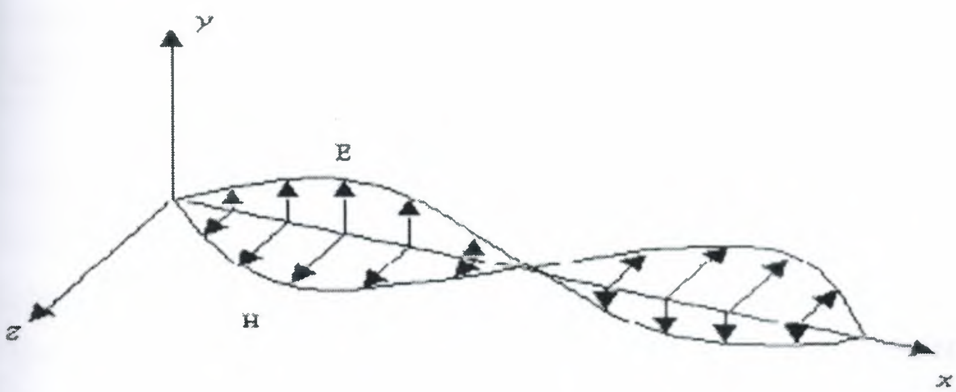


Figure 1.3 Polarizations

1.4 Transmission Media

The transmission media is divided mainly to the following types:

1.4.1 Guided Media

The guided media includes: twisted pair, coaxial cable and fiber-optic cable.

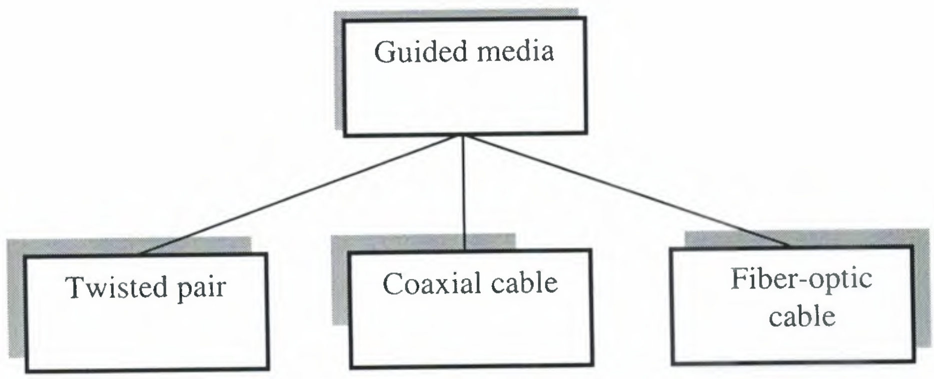


Figure 1.4 Categories of Guided Media

This table contains the typical characteristics for guided media

Table 1.1 Typical Characteristics for Guided Media

Medium Transmission	**Total Data Rate**	**Bandwidth**	**Repeater Spacing**
Twisted Pair	1 - 100 Mbps	100 Hz -5 MHz	2 – 10 km
Coaxial Cable	1 Mbps -1 Gbps	100 Hz – 500 MHz	1 – 10 km
Optical Fiber	2 Gbps	2 GHz	10- 100 km

In the past, two parallel flat wires were used for communications. Each wire is insulated from the other and both are open to free space. This type of line is used for connecting equipment that is up to 50m apart using moderate rate (less than 20 kbps). The signal, typically a voltage or current level relative to some ground reference is applied to one wire while the ground reference is applied to the other. Although a two wire open line can be used to connect two computers directly, it is used mainly for connecting computers with modems. As shown in Figure 1.4 two simple wires more sensitive to noise interference.

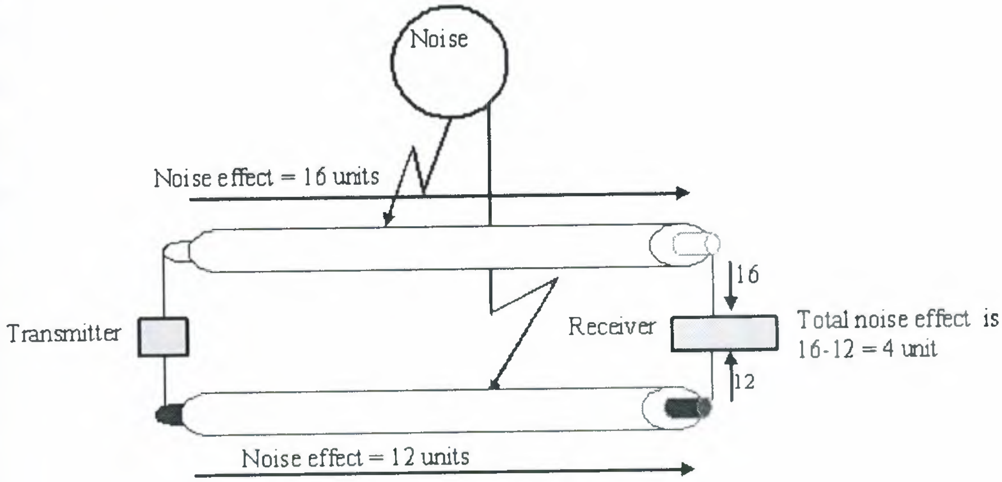


Figure 1.5 Effect of Noise in Parallel Lines

1.4.1.1 Twisted Pair

A twisted pair consists of two insulated copper wires. Over longer distances, cables may contain hundreds of pairs. The twisting of the individual pairs minimizes electromagnetic interference between the pairs.

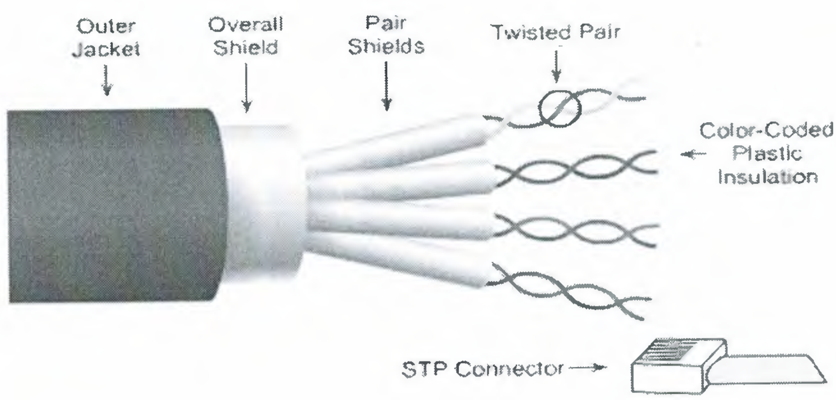


Figure 1.6 Twisted Pair Cable

Wire pairs can be used to transmit both analog and digital signals. For analog signals, amplifiers are required about every 5 to 6 km. For digital signals, repeaters are used at every 2 or 3 km. It is the backbone of the telephone system as well as the low - cost microcomputer local network within a building. In the telephone system, individual telephone sets are connected to the local telephone exchange or "end office" by twisted pair wire. These are referred to as "local loops". Within an office building, telephone service is often provided by means of a Private Branch Exchange (PBX). For modern digital PBX systems, data rate is about 64 kbps. Local loop connections typically require a modem, with a maximum data rate of 9600 bps. However, twisted pair is used for long distance trucking applications and data rates of 100 Mbps or more may be achieved **[1]**.

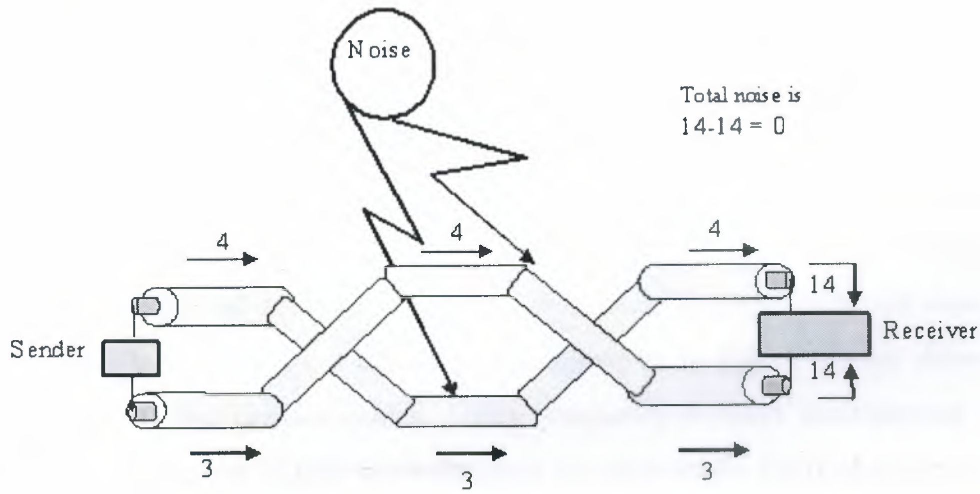


Figure 1.7 Effect of Noise on Twisted-Pair Lines

The twisted pair comes in two forms: shielded (STP) and unshielded (UTP). Figure 1.8 shows STP (a) and UTP (b, c). The metal casing prevents the penetration of electromagnetic noise and eliminates cross-talk. Materials and manufacturing requirements make STP more expensive than UTP but less susceptible to noise. UTP is cheap, flexible, and easy to use.

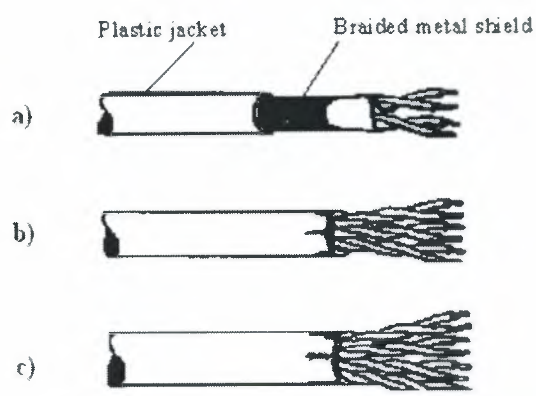


Figure 1.8 (a) STP, (b) and (c) UTP

1.4.1.2 Coaxial Cable

The main limiting factor of a twisted pair line is its capacity and a phenomenon known as the skin effect. As the bit rate increases, the current flowing in the wires tends to flow only on the outer surface of the wire, thus using the less available cross-section. This increases the electrical resistance of the wires for higher frequency signals, leading to the attenuation In addition, at higher frequencies; more signal power is lost as a result of radiation effect.

Coaxial cables, like twisted pairs, consist of two conductors, but are constructed differently to permit it to operate over a wider range of frequencies. Coaxial cables have been perhaps the most versatile transmission medium and are enjoying increasing utilizing in a wide variety of applications. The most important of these are long-distance telephone and television transmission, television distribution, and short-range connections between devices and local area networks. In Figure 1.9 are shown the constructions of the coaxial cables. Using frequency-division multiplexing a coaxial cable can carry over 10,000 voice channels simultaneously. Coaxial cables are used to transmit both analogue and digital signals **[12]**.

The principal constraints on performance are attention, thermal noise, and intermodulation noise.

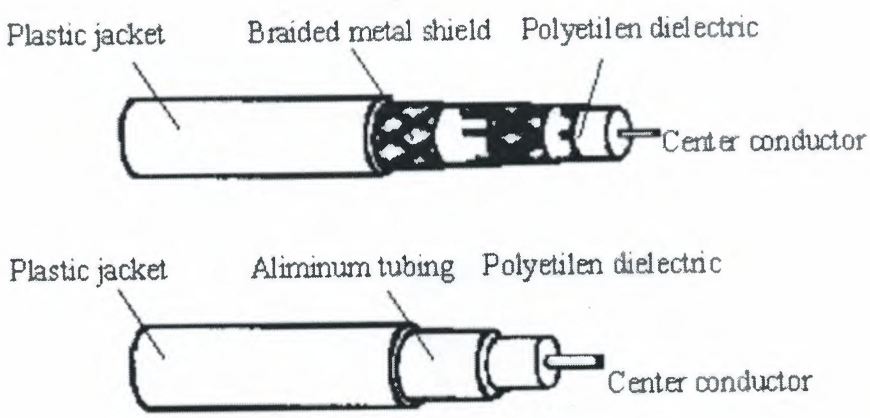


Figure 1.9 Coaxial Cable

1.4.1.3 Fiber-Optic Cable

You hear about fiber-optic cables whenever people talk about the telephone system, the cable TV system or the Internet. Fiber-optic lines are strands of optically pure glass as thin as a human hair that carries digital information over long distances. They are also used in medical imaging and mechanical engineering inspection. In more than 10 years since optical waveguides became a reality for practical applications, there have been tremendous strides in the development of cabling. The goal of cabling is to enable the multitude of advantages of optical waveguides to be fully realized. The benefits of optical cables include such attributes as light weight, small diameter, and excellent transmission characteristics.

Fiber optics (optical fibers) are long, thin strands of very pure glass about the diameter of a human hair. They are arranged in bundles called optical cables and used to transmit light signals over long distances. If you look closely at a single optical fiber shown in figure 1.10, you will see that it has the following parts:

- Core - Thin glass center of the fiber where the light travels.
- Cladding - Outer optical material surrounding the core that reflects the light back into the core.
- Buffer coating - Plastic coating that protects the fiber from damage and moisture.

Hundreds or thousands of these optical fibers are arranged in bundles in optical cables. The bundles are protected by the cable's outer covering, called a jacket.

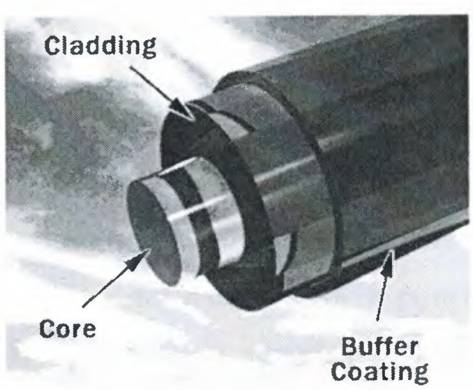


Figure 1.10 Single Optical Fiber

1.4.1.3.1 Types Optical fibers:

Single-mode fibers have small cores (about 3.5 x 10^{-4} inches or 9 microns in diameter) and transmit infrared laser light (wavelength = 1,300 to 1,550 nanometers). And Multi-mode fibers have larger cores (about 2.5 x 10^{-3} inches or 62.5 microns in diameter) and transmit infrared light (wavelength = 850 to 1,300 nm) from light-emitting diodes (LEDs).

Some optical fibers can be made from plastic. These fibers have a large core (0.04 inches or 1 mm diameter) and transmit visible red light (wavelength = 650 nm) from LEDs.

1.4.1.3.2 How Does an Optical Fiber Transmit Light?

Suppose you want to shine a flashlight beam down a long, straight hallway. Just point the beam straight down the hallway, light travels in straight lines, so it is no problem. What if the hallway has a bend in it? You could place a mirror at the bend to reflect the light beam around the corner. What if the hallway is very winding with multiple bends? You might line the walls with mirrors and angle the beam so that it bounces from side-to-side all along the hallway. This is exactly what happens in an optical fiber and its explained in figure 1.11.

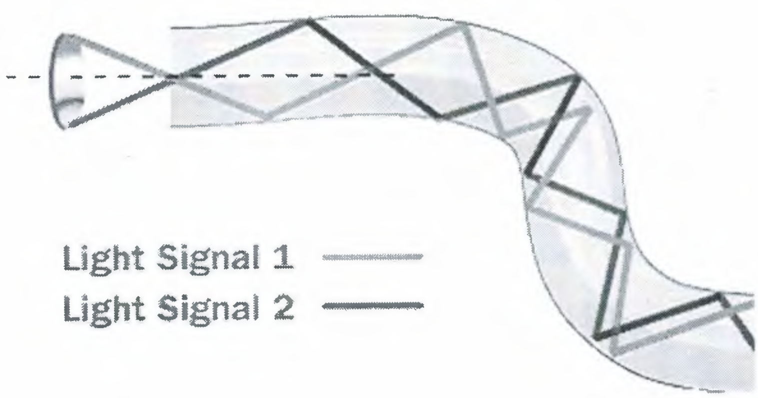


Figure 1.11 Diagram of Total Internal Reflection in an Optical Fiber

The light in a fiber-optic cable travels through the core (hallway) by constantly bouncing from the cladding (mirror-lined walls), a principle called total internal reflection. Because the cladding does not absorb any light from the core, the light wave can travel great distances. However, some of the light signal degrades within the fiber, mostly due to impurities in the glass. The extent that the signal degrades depends on the purity of the glass and the wavelength of the transmitted light (for example, 850 nm = 60 to 75 percent/km; 1,300 nm = 50 to 60 percent/km; 1,550 nm is greater than 50 percent/km). Some premium optical fibers show much less signal degradation, less than 10 percent/km at 1,550 nm **[11]**.

1.4.1.3.3 A Fiber-Optic Relay System

Fiber-optic relay systems consist of the following:

- Transmitter - Produces and encodes the light signals.
- Optical fiber - Conducts the light signals over a distance.
- Optical regenerator - May be necessary to boost the light signal.
- Optical receiver - Receives and decodes the light signals.

1.4.1.3.4 Advantages of Fiber Optics

Why are fiber-optic systems revolutionizing telecommunications? Compared to conventional metal wire (copper wire), optical fibers are:

1. Less expensive - Several miles of optical cable can be made cheaper than equivalent lengths of copper wire. This saves your provider (cable TV, Internet) and you money.
2. Thinner - Optical fibers can be drawn to smaller diameters than copper wire.
3. Higher carrying capacity - Because optical fibers are thinner than copper wires, more fibers can be bundled into a given-diameter cable than copper wires. This allows more phone lines to go over the same cable or more channels to come through the cable into your cable TV box for example.
4. Less signal degradation - The loss of signal in optical fiber is less than in copper wire.
5. Low power - Because signals in optical fibers degrade less, lower-power transmitters can be used instead of the high-voltage electrical transmitters needed for copper wires. Again, this saves your provider and you money.
6. Digital signals - Optical fibers are ideally suited for carrying digital information, which is especially useful in computer networks.
7. Non-flammable - Because no electricity is passed through optical fibers,

8. Flexible - Because fiber optics are so flexible and can transmit and receive light, they are used in many flexible digital cameras for the following:
    - Medical imaging - in bronchoscopes, endoscopes, laparoscopes
    - Mechanical imaging - inspecting mechanical welds in pipes and engines (in airplanes, rockets, space shuttles, cars)
    - Plumbing - to inspect sewer lines

Because of these advantages, you see fiber optics in many industries, most notably telecommunications and computer networks. For example, if you telephone Europe from the United States (or vice versa) and the signal is bounced off a communications satellite, you often hear an echo on the line. But with transatlantic fiber-optic cables, you have a direct connection with no echoes **[11]**.

1.4.1.3.5 How Are Optical Fibers Made?

Optical fibers are made of extremely pure optical glass. We think of a glass window as transparent, but the thicker the glass gets, the less transparent it becomes due to impurities in the glass. However, the glass in an optical fiber has far fewer impurities than window-pane glass. One company's description of the quality of glass is as follows:

If you were on top of an ocean that is miles of solid core optical fiber glass, you could see the bottom clearly.

Making optical fibers requires the following steps:

- Making a pre-form glass cylinder.
- Drawing the fibers from the pre-form.

1.4.1.3.5.1 Making a Pre-form Glass Cylinder

The glass for the preform is made by a process called modified chemical vapor deposition (MCVD).

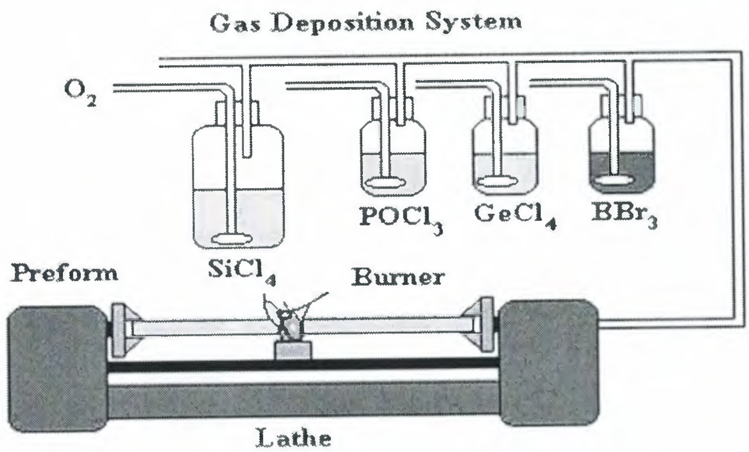


Figure 1.12 MCVD Process for Making the Preform Blank

In MCVD, oxygen is bubbled through solutions of silicon chloride (SiCl4), germanium chloride (GeCl4) and/or other chemicals. The precise mixture governs the various physical and optical properties (index of refraction, coefficient of expansion, melting point, etc.). The gas vapors are then conducted to the inside of a synthetic silica or quartz tube (cladding) in a special lathe. As the lathe turns, a torch is moved up and down the outside of the tube.

The extreme heat from the torch causes two things to happen:

- The silicon and germanium react with oxygen, forming silicon dioxide (SiO2) and germanium dioxide (GeO2).
- The silicon dioxide and germanium dioxide deposit on the inside of the tube and fuse together to form glass.

The lathe turns continuously to make an even coating and consistent blank. The purity of the glass is maintained by using corrosion-resistant plastic in the gas delivery system (valve blocks, pipes, seals) and by precisely controlling the flow and composition of the mixture. The process of making the preform blank is highly automated and takes several hours. After the preform blank cools, it is tested for quality control (index of refraction).

1.4.1.3.5.2 Drawing Fibers from the Preform Blank

Once the preform blank has been tested, it gets loaded into a fiber drawing tower. The blank gets lowered into a graphite furnace (3,452 to 3,992 degrees Fahrenheit or 1,900 to 2,200 degrees Celsius) and the tip gets melted until a molten glob falls down by gravity. As it drops, it cools and forms a thread. After this operation ends, the Finished Optical Fiber must be tested.

.

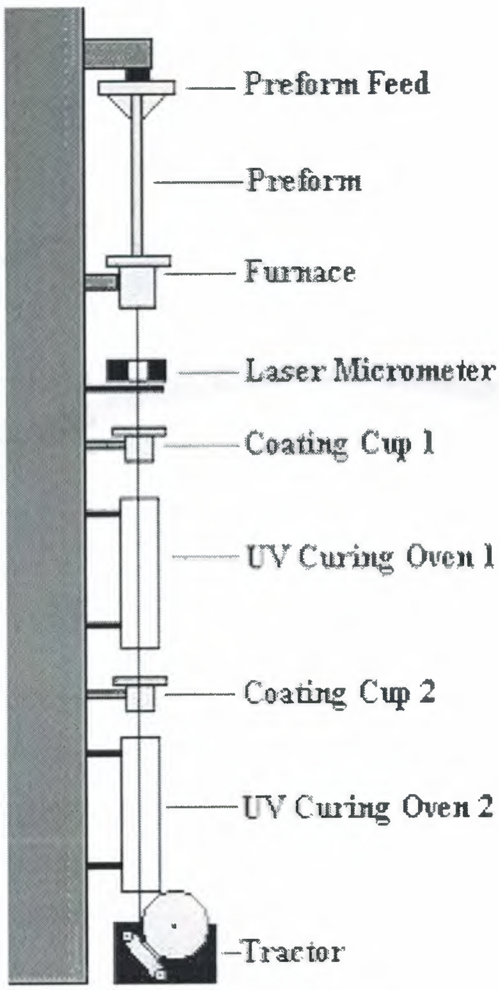


Figure 1.13 Fiber Drawing Tower

1.4.2 Unguided Media

There are three basic modes of getting a radio wave from the transmitting to receiving antenna: ground wave, sky wave and line-of-sight propagations.

1.4.2.1 Ground Wave Propagation

Ground waves are radio waves that follow the curvature of the earth. These waves may be vertically polarized to alleviate short circuiting the electric field through the conductivity of the ground. Since the ground is not a perfect electrical conductor, ground waves are attenuated as they follow the earth's surface. At low frequencies, ground losses are low and become lower at lower frequencies. The VLF and LF frequencies are mostly used for military communications, especially with ships and submarines **[1]**.

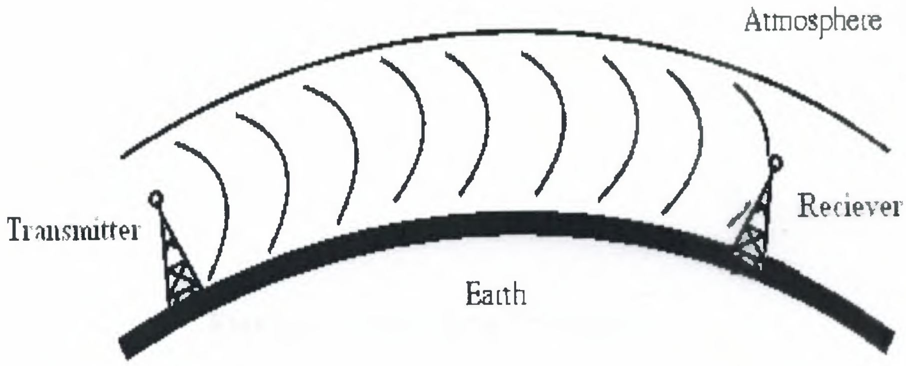


Figure 1.14 Ground Wave Propagation

Early commercial and professional radio services relied exclusively on long wave, low frequencies and ground-wave propagation. To prevent interference with these services, amateur and experimental transmitters were restricted to the higher (HF) frequencies, felt to be useless since their ground-wave range was limited. Upon discovery of the other propagation modes possible at medium wave and short wave frequencies, the advantages of HF for commercial and military purposes became apparent. Amateur experimentation was then confined only to authorized frequency segments in the range **[13]**.

1.4.2.2 Sky Wave Propagation

Radio waves in the LF and MF ranges may also propagate as ground waves, but suffer significant losses, or are attenuated, particularly at higher frequencies. But as the ground wave mode fades out, a new mode develops: the sky wave. Sky waves are reflections from the ionosphere. While the wave is in the ionosphere, it is strongly bent, or refracted, ultimately back to the ground. From a long distance away this appears as a reflection. Long ranges are possible in this mode also, up to hundreds of miles. Sky waves in this frequency band are usually only possible at night, when the concentration of ions is not too great since the ionosphere also tends to attenuate the signal. However, at night, there are just enough ions to reflect the wave but not reduce its power too much [13].

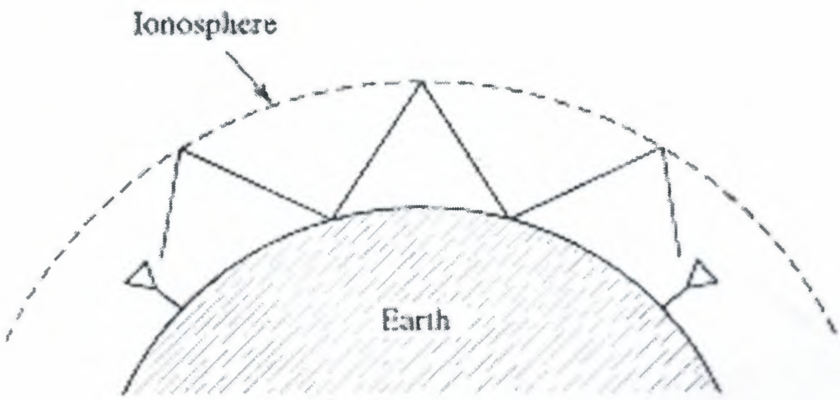


Figure 1.15 Sky Wave Propagation

1.4.2.3 Line-Of-Sight Propagation

The simplest and most easily understood way in which a signal travels from one antenna to another is by 'line-of-sight' propagation. Line-of-sight propagation requires a path where both antennas are visible to one another and there are no obstructions. VHF and UHF communication typically use this path.

Unless you are VERY close to your destination, you need to keep the antenna as high as possible. Because radio waves follow a straight-line in this mode, they simply go off into space as the curvature of the earth causes the ground to drop away beneath the radio waves.

As we elevate the antenna, the distance to the horizon gets further and further away. With enough power to reach the other antenna and a high enough antenna to see it, we can talk without problems. VHF repeaters are usually mounted on high buildings or mountain tops for this very reason. When you are operating with a small VHF hand held, your signal must be able to travel in a straight-line to the repeater or your signal will be lost to someone beyond line-of-sight **[13]**.

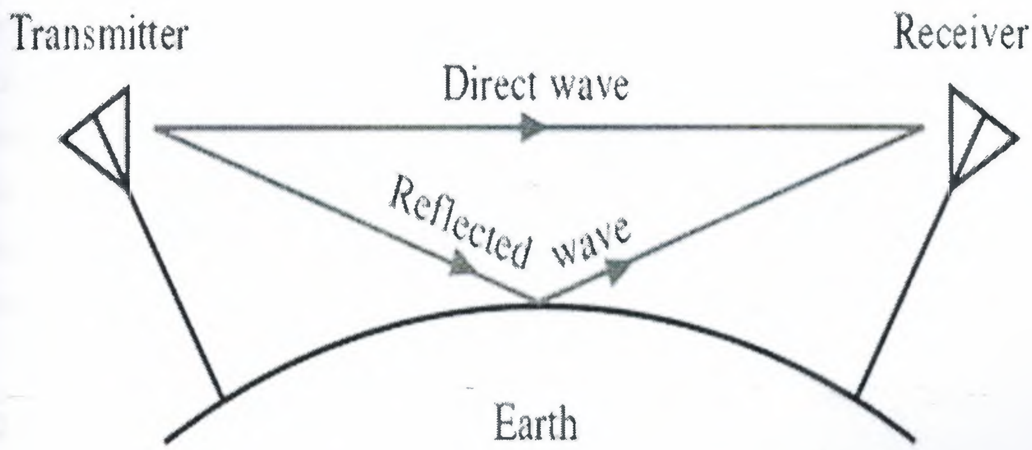


Figure 1.16 Line-Of-Sight Propagation

1.5 Summary

In this introductory chapter, we have seen the types of transmission medias that information can be sent through, Guided media which is subdivided into three types, Twisted Pairs, Coaxial Cables and Fiber Optic cables. The other type of transmission medias is Un-Guided media, in this type, electromagnetic waves are propagated through free space in three categories which are Ground Wave Propagation, Sky Wave Propagation and Line of Sight Propagation.

2. GSM OVERVIEW

2.1 Introduction

This chapter will provide the reader with a broad and clear understanding of the GSM system. It also acts as a base for other studies in radio and wireless systems architectures and applications. The main functions of each device (node) on a GSM network will be studied with specific emphasis on the relevance to the overall system. The GSM overview is a natural starting point in the overall wireless program. This chapter will introduce the GSM history, system components, geographical network structure, frequency bands, modulation method, and transmission problems.

2.2 History of GSM

During the early 1980s, analog cellular telephone systems were experiencing rapid growth in Europe, particularly in Scandinavia and the United Kingdom. Each country developed its own system, which was incompatible with everyone else's in equipment and operation. This was an undesirable situation, because not only was the mobile equipment limited to operation within national boundaries, which in a unified Europe were increasingly unimportant, but there was also a very limited market for each type of equipment, so economies of scale and the subsequent savings could not be realized.

The Europeans realized this early on, and in 1982 the Conference of European Posts and Telegraphs (CEPT) formed a study group called the GSM (Group Special Mobile) to study and develop a pan-European public land mobile system. The proposed system had to meet certain criteria:

- Good subjective speech quality
- Low terminal and service cost
- Support for international roaming
- Ability to support handheld terminals
- Support for range of new services and facilities
- Spectral efficiency

- ISDN compatibility

In 1989, GSM responsibility was transferred to the European Telecommunication Standards Institute (ETSI), and phase I of the GSM specifications were published in 1990. Commercial service was started in mid-1991, and by 1993 there were 36 GSM networks in 22 countries. Although standardized in Europe, GSM is not only a European standard. Over 200 GSM networks (including Digital Cellular Service (DCS1800) at 1800 MHz and Personal Communication Service at 1900 MHz (PCS1900)) are operational in 110 countries around the world. In the beginning of 1994, there were 1.3 million subscribers worldwide, which had grown to more than 55 million by October 1997. With North America making a delayed entry into the GSM field with a derivative of GSM called PCS1900, GSM systems exist on every continent, and the acronym GSM now aptly stands for Global System for Mobile communications.

The developers of GSM chose an unproven (at the time) digital system, as opposed to the then-standard analog cellular systems like AMPS in the United States and TACS in the United Kingdom. They had faith that advancements in compression algorithms and digital signal processors would allow the fulfillment of the original criteria and the continual improvement of the system in terms of quality and cost. The over 8000 pages of GSM recommendations try to allow flexibility and competitive innovation among suppliers, but provide enough standardization to guarantee proper interworking between the components of the system. This is done by providing functional and interface descriptions for each of the functional entities defined in the system. [5]

2.3 GSM Network Component

The GSM technical specifications define the different entities that form the GSM network by specifying their functions and interface requirements. The GSM network can be divided into three main systems:

- Switching System (SS)
- Base Station System (BSS)
- Mobile Station (MS)

In addition to these systems, there exist the Operation and Maintenance Center (OMC) and the Network Management System (NMC), which is used to operate, maintain, and manage the GSM network.

SS is responsible for performing call processing and subscriber related functions. It includes the following functional units:

- Mobile services Switching Center (MSC)
- Home Location Register (HLR)
- Visitor Location Register (VLR)
- Authentication Center (AUC)
- Equipment Identity Register (EIR)

BSS performs all the radio-related functions. It is comprised from the following functional units:

- Transcoder Controller (TRC)
- Base Station Controller (BSC)
- Base Transceiver Station (BTS)

Figure 2.1 shows the GSM system architecture, which consists of the switching system, the base station system and the user equipment. [5]

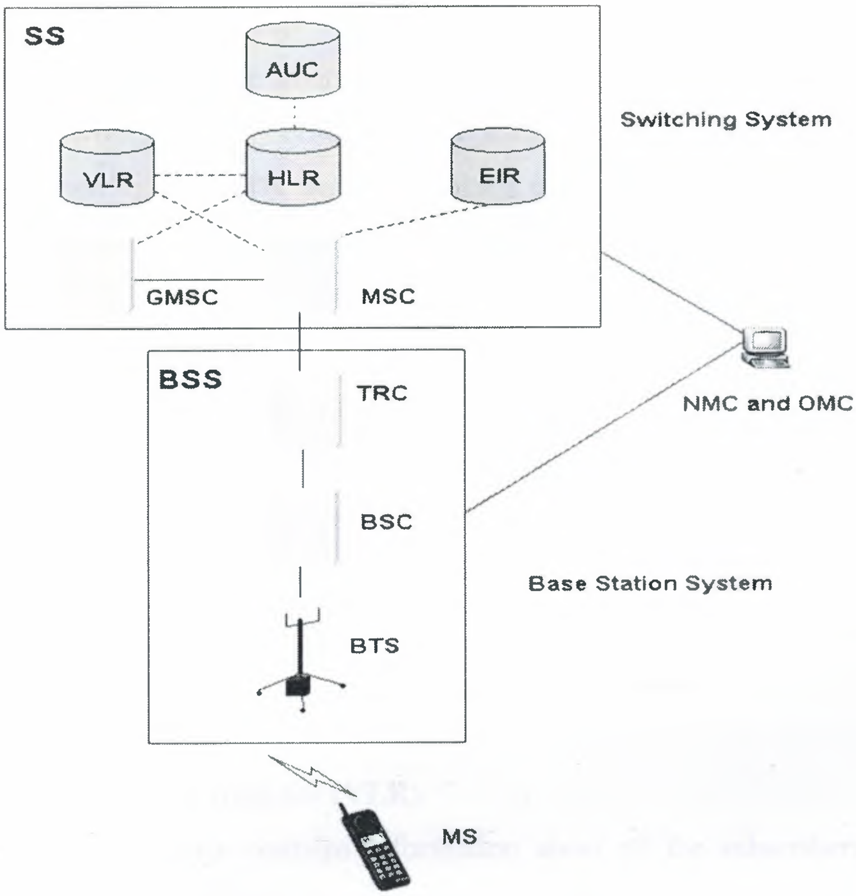


Figure 2.1 GSM System Model

2.3.1 Switching System (SS) Components

Its main role is to manage the communications between the mobile users and other users, such as mobile users, ISDN users, fixed telephony users, etc. It also includes data bases needed in order to store information about the subscribers and to manage their mobility. The different components of the SS are described below.

2.3.1.1 Mobile Switching Center (MSC)

It is the central component of the SS. MSC performs the telephony switching functions for the mobile network. It also provides connection to other networks. MS call setup, reestablishment, and routing, digit translation, call control and signaling, billing data capture, formatting and teleprocessing, handovers, management of facilities for calls, echo control, and short message service support [1].

2.3.1.2 Gateway MSC (GMSC)

A gateway is a node interconnecting two networks. It enables an MSC to interrogate networks HLR in order to route a call to a Mobile Station (MS). If a person connected to the PSTN wants to make a call to a GSM subscriber, then the PSTN exchange will access the GSM network by first connecting the call to the GMSC.

2.3.1.3 Home Location Register (HLR)

It is a centralized network database that stores and manages all mobile subscriptions belonging to a specific operator. It acts as a permanent store for a person's subscription information until that subscription is canceled. The information stored includes subscriber identity, subscriber supplementary services, subscriber location information, and subscriber authentication information.

2.3.1.4 Visitor Location Register (VLR)

The VLR database contains information about all the subscribers currently located in an MSC service area. It temporarily stores the subscription information so that the MSC can serve all the subscribers currently visiting the MSC service area. When a subscriber roams into a new MSC service area, VLR sends a request to the subscriber's HLR to get information about the subscriber. HLR then sends a copy of the information to the VLR. It is always implemented together with a MSC; so the area under control of the MSC is also the area under control of the VLR. Functions are to allocate Mobile Station Roaming Number (MSRN) for incoming call setups and Temporary Mobile Subscriber Identity (TMSI) for identification.

2.3.1.5 Authentication Center (AUC)

AUC register is used for security purposes. Its main function is to authenticate the subscriber's attempting to use a network. It is a database connected to the HLR, which provides it with the authentication parameters and ciphering keys used to ensure network security.

2.3.1.6 Equipment Identity Register (EIR)

EIR is also used for security purposes. It is a database containing mobile equipment identity information, which helps to block calls from stolen, unauthorized, or defective MSs. A terminal is identified by its International Mobile Equipment Identity (IMEI). The EIR allows them to forbid calls from stolen or unauthorized terminals.

2.3.2 Base Station System (BSS) Components

BSS connects the Mobile Station and the SS. It is in charge of the transmission and reception. It can be divided into three parts:

2.3.2.1 Transcoder Controller (TRC)

The main function of a TRC is to multiplex network traffic channels from multiple BSCs onto one 64 kbps Pulse Code Modulation (PCM) channel. Another function is to perform rate adaptation, which involves the conversion of the information arriving from the MSC at a rate of 64 kbps to a rate of 16 kbps for transmission to a BSC.

2.3.2.2 Base Station Controller (BSC)

The BSC controls a group of BTSs and manages all the radio related functions of a GSM network. It is a high capacity switch that provides functions such as handovers, radio channel assignment, and the collection of cell configuration data. A number of BSCs may be controlled by a MSC.

2.3.2.3 Base Transceiver Station (BTS)

BTS controls the radio interface to the MS. It holds the radio equipment such as transceivers and antennas, which are needed to serve each cell in the network. A BTS is usually placed in the center of a cell. Its transmitting power defines the size of a cell. Each BTS has between one and sixteen transceivers depending on the density of users in the cell.

2.3.3 Operation and Support System (OSS)

OSS is connected to the different components of the SS and to the BSC, in order to control and monitor the GSM system. It is also in charge of controlling the traffic load of the BSS.

2.3.3.1 Operation and Maintenance Center (OMC)

It is a computerized monitoring center, which is connected to other network components such as MSCs and BSCs. It provides information about the status of the network and can monitor and control some system parameters.

2.3.3.2 Network Management Center (NMC)

It is responsible for the centralized control of a network. Only one NMC is required for a network and this controls the subordinate OMCs.

2.3.4 Mobile Station (MS)

MS is used to communicate with the mobile network. It is the physical equipment used by a subscriber, most often a normal hand-held cellular telephone. The range or coverage area of an MS depends on its output power.

MSs consist of a mobile terminal and a Subscriber Identity Module (SIM). The SIM is a smart card that identifies the terminal. By inserting the SIM card into the terminal, the user can have access to all the subscribed services. Without the SIM card, the terminal is not operational.

The SIM card is protected by a four-digit Personal Identification Number (PIN) in order to identify the subscriber to the system; the SIM card contains some parameters of the user such as its International Mobile Subscriber Identity (IMSI).

It contains identification numbers of the user and a list of available GSM networks, and tools needed for authentication and ciphering, depending on the type of the card there is also a storage space for messages and a phone numbers, etc [1].

2.4 GSM Geographical Network Structure

Every telephone network needs a specific structure to route incoming calls to the correct exchange and then on to the subscriber. In a mobile network, this structure is very important to monitor the subscriber's location.

The GSM network is made up of geographic areas as shown in Figure 2.2, these areas include cells, location areas (LAs), MSC/VLR service areas, and public land mobile network (PLMN) areas. [5]

2.4.1 Cell

A cell is the basic unit of a cellular system and is defined as the area of radio coverage given by one BTS. Each cell is assigned a unique number called the Cell Global Identity (CGI).

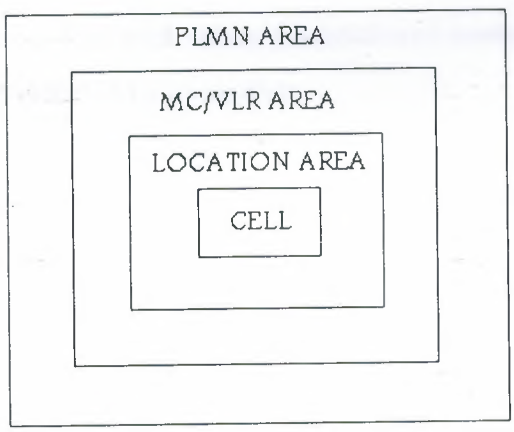


Figure 2.2 GSM Network Areas

2.4.2 Location Area

Location area is defined as a group of cells. Within the network a subscriber's location is known by the LA, which they are in. The identity of the LA in which an MS is currently located is stored in the VRL. When a MS crosses a boundary from a cell belonging to another LA, it must report its new location to the network. When a MS crosses a cell boundary within a LA, it doesn't need to report its new location to the network.

2.4.3 MSC Services Area

A MSC service area is made up of a number of LAs and represents the geographical part of the network controlled by one MSC. The subscribers MSC service area is also recorded, monitored and stored in the HLR.

2.4.4 Public Land Mobile Network (PLMN) Service Area

A PLMN service area is the entire set of cells served by one network operator and is defined as the area in which an operator offers radio coverage and access to its network. In one country there may be several PLMN service areas, one for each mobile operator's network.

2.4.5 GSM Service Area

The GSM service area is the entire geographical area in which a subscriber can gain access to a GSM network. The GSM area increases as more operators agree to make roaming agreements with each other. International roaming is the term applied when a MS moves from one PLMN to another.

2.5 GSM Frequency Bands

As GSM has grown worldwide, it has expanded to operate at three frequency bands: 900, 1800, and 1900 MHz.

2.5.1 GSM 900

The original frequency band specified for GSM was 900 MHz. Most GSM networks worldwide use this band. In some countries extended version of GSM 900 can be used, which provides extra network capacity. This extended version of GSM is called E-GSM, while primary version is called P-GSM.

2.5.2 GSM 1800

In 1990, in order to increase competition between operators, the UK requested the start of new version of GSM adapted to the 1800 MHz frequency band. By granting licenses for GSM 1800 in addition to GSM 900, a country can increase the number of operators.

2.5.3 GSM 1900

In 1995, the Personal Communications Services (PCS) concept was specified in the US. The basic idea was to enable "person to person" communication rather than "station to station". The frequencies available for PCS are around 1900 MHz.

2.6 Modulation Method

In the GSM-900 system, two frequency bands have been made available:

- 890 - 915 MHz for uplink (direction MS to BS).
- 935 - 960 MHz for downlink (direction BS to MS).

The 25 MHz bands are then divided into 124 pairs of frequency duplex channels with 200 kHz carrier spacing using Frequency Division Multiple Access (FDMA). Since it is not possible for a same cell to use two adjacent channels, the channel spacing can be said to be 200 KHz interleaved.

The frequency that is used in GSM 900, to transfer the information over the air interface as shown above is around 900 MHz, since this is not the frequency at which the information is generated; modulation techniques are used to translate the information into the usable frequency band. In GSM the modulation technique used is GMSK, it enables the transmission of 270 kbps within a 200 KHz channel.

As in most digital cellular systems GSM use the technique of Time Division Multiple Access (TDMA) to split this 200 KHz radio channel into 8 time periods. These periods of time are referred as time slots (which creates 8 logical channels). A logical channel is therefore defined by its frequency and the TDMA frame time slot number. Each mobile station on a call is assigned one time slot on the uplink frequency and one on the downlink frequency. By employing eight time slots, each channel transmits the digitized speech in a series of short bursts. A GSM terminal is transmitting for one eighth of the time. Figure 2.3 illustrates the TDMA frame. **[6]**

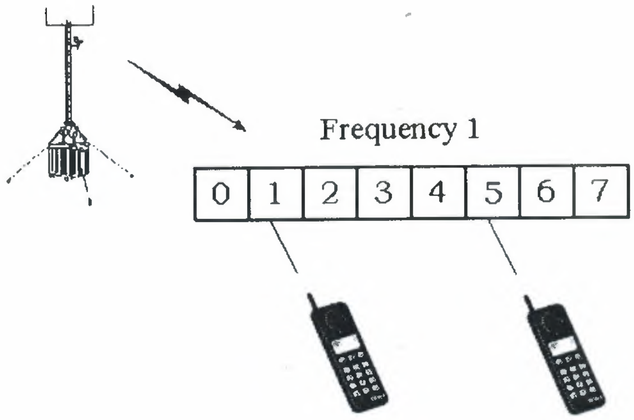


Figure 2.3 TDMA Frame in the Downlink

2.7 Transmission Problems in GSM Network

Many problems occur during the transmission of a radio signal. Some of the problems are described below **[5]**.

2.7.1 Path Loss

Path loss occurs when the received signal becomes weaker and weaker due to increasing the distance between MS and BTS, even if there are no obstacles between the transmitter and receiver.

2.7.2 Shadowing

Shadowing occurs when there are physical obstacles including hills and buildings between BTS and MS. The obstacles create a shadowing effect, which can decrease the received signal strength. When the MS moves, the signal strength fluctuates depending on the obstacles types.

2.7.3 Multipath Fading

Multipath fading occurs when there is more than one transmission path to the BTS or MS, and therefore more than one signal arriving at the receiver. This may be due to buildings or mountains, which reflect the transmitted signal.

2.7.4 Rayleigh Fading

This occurs when a signal takes more than one path between the MS and BTS. In this case the signal is not received on a line of sight path directly from the transmitter. Rather it is reflected off buildings and other objects and is received from several different paths, and then the received signal is the sum of many identical signals that differ only in phase.

2.7.5 Time Dispersion

Time dispersion is another problem relating to multipath. It causes Inter-Symbol Interference (ISI) where consecutive bits interfere with each other making it difficult for the receiver to determine which bit is the correct one. If the reflected signal arrives one bit time after the direct signal, the receiver will not know which one is correct.

2.7.6 Time Alignment

Each MS is allocated a time slot on a TDMA frame. This is an amount of time during which the MS transmits information to the BTS. The information must also arrive at the BTS within that time slot. The time alignment problem occurs when the part of information transmitted by a MS doesn't arrive within the allocated time. Instead, that part may arrive during the next time slot, and may interfere with information from another MS using that other time slot. A large distance between the MS and the BTS causes time alignment. Effectively, the signal cannot travel over the large distance within the given time.

2.8 Summary

in this chapter, we have seen the GSM system and we have discussed all the working principles of every part contains in it, and we have known that GSM system uses GSMK to modulate the signals into a suitable form that matches the channel conditions.

In the next chapter, we will discuss the most important modulation methods including the GMSK.

3. MODULATION

3.1 Modulation

Modulation is the process of impressing a low-frequency information signal onto a higher frequency carrier signal. Modulation is done to bring information signals up to the Radio Frequency (or higher) signal Some systems even have two stage Modulation, where the information is brought up to an Intermediate Frequency (IF), and then increased to the transmission frequency, and then increased to the transmission frequency. Base band Signal is a term used to describe the unmodulated signal or in other words, the information signal Carrier Signal is what the information signal is combined with to form the new modulated signal. The frequency of the carrier is described as the center frequency of the signal. Both the base band and carrier have bandwidth that matters for AM, but not for FM/PM modulated band width.

Automatic modulation recognition is a rapidly evolving area of signal exploitation with applications in DF confirmation, monitoring, spectrum management, interference identification and electronic surveillance. Generally stated, a signal recognizer is used to identify the modulation type (along with various parameters such as baud rate) of a detected signal for the purpose of signal exploitation. For example, a signal recognizer could be used to extract.

Signal information useful for choosing a suitable counter measure, such as jamming .In recent years interest in modulation recognition algorithms has increased with the emergence of new communication technologies.

In particular, there is growing interest in algorithms that treat quadrature amplitude modulated (QAM) signals, which are used in the HF, VHF, and UHF bands for a wide variety of applications including FAX, modem, and digital.

3.2 Amplitude Modulation (AM)

Information signal is added and subtracted to and from a carrier signal. Amplitude modulation means a carrier wave is modulated in proportion to the strength of a signal. The carrier rises and falls instantaneously with each high and low of the conversation. Check out the diagram below. See how the voice current produces an immediate and equivalent change in the carrier.

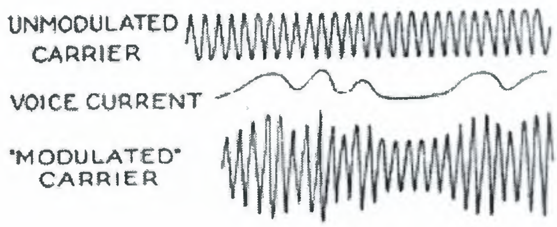


Figure 3.1 Loading the Voice on a Carrier

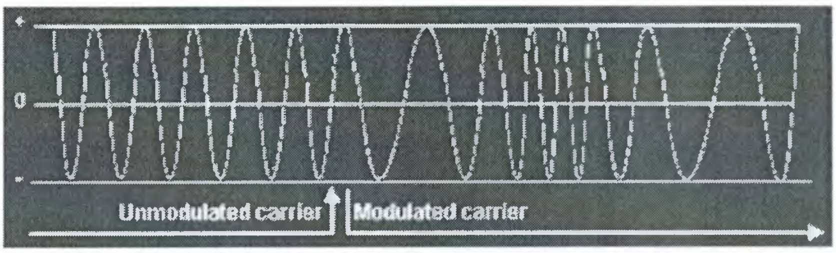
Low frequency commercial broadcast stations in the "A.M band" use amplitude modulation. Most C.B. or citizens band radios use it too. It's a simple, robust method to form a radio wave but it suffers from static and high battery power requirements, reasons enough that few personal communications devices use it.

3.3 Frequency Modulation (FM)

Information signal varies a constant Amplitude carrier signal's frequency directly in proportion to the information's frequency.

Frequency modulation confuses many people but it shouldn't. FM is not limited to the FM band. It is not frequency dependent, that is, it can be used at high or low frequencies. That's because it is a modulation technique, a way to shape a radio wave, not a service by itself. The word frequency in FM relates, instead, to the rate at which this method varies a carrier wave, not to any particular radio frequency it is used on. This will become clearer as it goes on. The virtues of an FM signal are readily apparent by listening to the FM band low distortion, little static, good voice quality and immunity from electrical and atmospheric interference. It's why television audio and analog cellular use it. FM also exhibits a capture effect, whereby the receiver seizes on the strongest signal and rejects any others. No other signals fading in and out like with A.M. What's more, F.M. needs far less power to transmit a signal the same distance than A.M.

It doesn't have the modulated carrier varying in amplitude, as with A.M., but in the number of cycles or rate. Although perhaps not obvious at first, the right hand side does differ from the left hand side.



. **Figure 3.2** The Difference in the Waveform

Frequency modulation varies the carrier at a rate of 440 cycles per second, matching the original signal. This differs dramatically from A.M. as it is seen above, where a wildly swinging sine wave would be produced instead. In F.M. a quick change in audio frequency results in a quick rate change to the carrier. Despite this seemingly complicated operating method, F.M. circuitry after sixty years is now well established, cheap and simple.

3.4 Phase Modulation (PM)

Information signal varies a constant amplitude carrier signal's phase directly in proportion to the information's frequency. Three ways exists to modulate a signal: by amplitude, frequency or phase. And although there are dozens of modulation techniques, under the most confusing names possible, all of them will fit into one of these categories. As looked at amplitude modulation, which changes the carrier wave by signal strength, and frequency modulation, which converts the originating signal into cycles? Now if looked at phase modulation, which changes the angle of the carrier wave. Phase modulation is strictly for digital working and is closely related to F.M. Phase in fact enjoys the same capture effect as F.M.

A digital signal means an ongoing stream of bits, 0s and 1s, on and off pulses of electrical energy. Like those signals running around the inside computer. Well, how do it is transmitted that staccato beat of electrical pulses? One put it on a carrier wave.

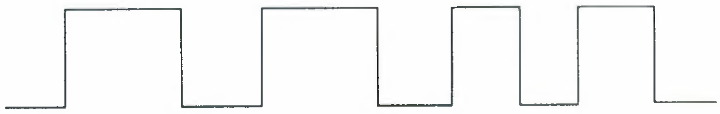


Figure 3.3 Scale Diagram of a Digital Signal

One might think that it could send digital without a carrier wave, like the earliest wireless telegraphs but results wouldn't be good.

Radio technology is built on carrier waves. No matter how one transmits RF energy, there is always some type of 'carrier' involved. Ever hear an A.M. radio station go silent for a minute or two? If they are off the air completely would be heard as static. But if they have simply lost audio for a while one will hear a silence. That's the carrier wave.

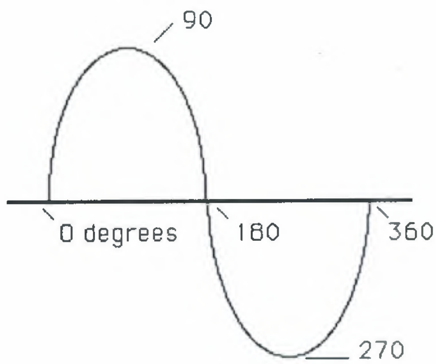


Figure 3.4 Transmission of Analog or Digital Signal

A continuous wave produced to transmit analog or digital information. The phases or angles of a sine way give rise to different ways of sending information.

3.5 Coherent and Incoherent Systems

The terms coherent and incoherent are frequently used when discussing the generation and reception of digital modulation. When linked to the process of modulation the term coherence relates to the ability of the modulator to control the phase of the signal, not just the frequency. For example Frequency Shift Keying (FSK) can be generated both coherently with an IQ modulator and incoherently with simply a Voltage Controlled Oscillator (VCO) and a digital voltage source, as shown below.

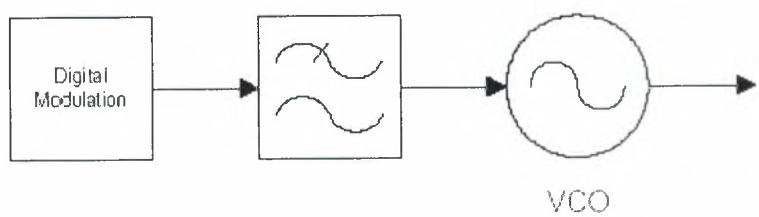


Figure 3.5 In-coherent Generation of FSK

With the system in figure 3.5 the instantaneous frequency of the output waveform is determined by the modulator (within a tolerance set by the VCO and data amplitude etc) but the instantaneous phase of the signal is not controlled and can have any value. Alternatively coherent generation of modulation is achieved as shown in figure 3.6. Here the phase of the signal is controlled, rather than the frequency.

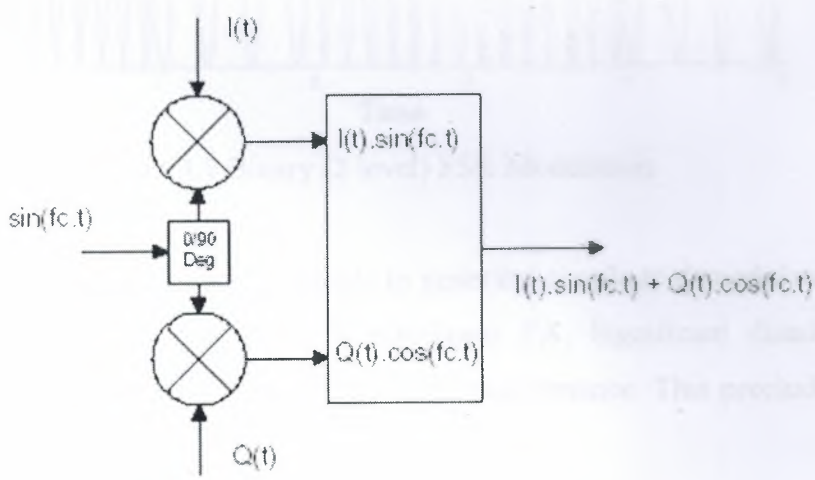


Figure 3.6 Coherent Generation of FSK

When a coherent modulator is used to generate FSK the exact signal frequency and phase are controlled. The modulator shown above offers the possibility to shape the resultant carrier phase trajectory at base band either with analogue filtering or digital signal processing and a DAC. This can be used to generate both constant amplitude and amplitude modulated signals. Use of the term coherent with respect to the act of demodulation refers to a system that makes a demodulation decision based on the received signal phase, not frequency. The high level of digital integration now possible in semiconductor devices has made digitally based coherent demodulators common in mobile communications systems.

3.6 Frequency Shift Keying (FSK)

As previously stated applying modulation in wireless communications involves modifying the phase or amplitude, or both, of a sinusoidal carrier. One of the simplest, and widest used system, is frequency modulation. This exists in a great variety of forms, as will be discussed later, but in essence involves making a change to the frequency of the carrier to represent a different level. The generic name for this family of modulation is Frequency Shift Keying (FSK).

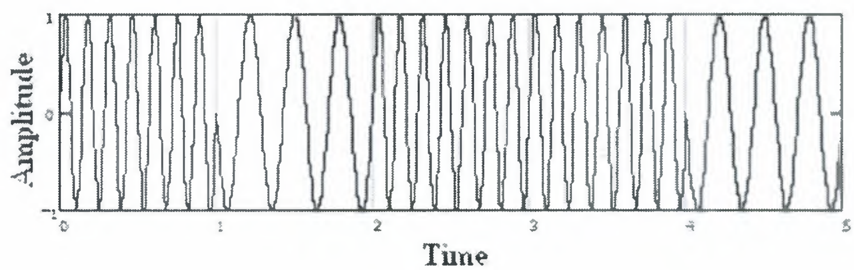


Figure 3.7 Binary (2 level) FSK Modulation

FSK has the advantage of being very simple to generate, simple to demodulate and due to the constant amplitude can utilize a non-linear PA. Significant disadvantages, however, are the poor spectral efficiency and BER performance. This precludes its use in this basic form from cellular and even cordless systems.

3.7 Minimum Shift Keying (MSK)

Minimum Shift Keying is FSK with a modulation index of 0.5. Therefore the carrier phase of an MSK signal will be advanced or retarded 90° over the course of each bit period to represent either a one or a zero. Due to this exact phase relationship MSK can be considered as either phase or frequency modulation. The result of this exact phase relationship is that MSK can't practically be generated with a voltage controlled oscillator and a digital waveform. Instead an IQ modulation technique, as for PSK, is usually implemented. Coherent demodulation is usually employed for MSK due to the superior BER performance. This is practically achievable, and widely used in real systems, due to the exact phase relationship between each bit.

3.7.1 Gaussian Minimum Shift Keying (GMSK)

A variant of MSK that is employed by some cellular systems (including GSM) is Gaussian Minimum Shift Keying. Again GMSK can be viewed as either frequency or phase modulation. The phase of the carrier is advanced or retarded up to 90° over the course of a bit period depending on the data pattern, although the rate of change of phase is limited with a Gaussian response. The net result of this is that depending on the Bandwidth Time product (BT), effectively the severity of the shaping, the achieved phase change over the bit may fall short of90°. This will obviously have an impact on the BER, although the advantage of this scheme is the improved bandwidth efficiency. The extent of this shaping can clearly be seen from the' eye' diagrams in Figure3.8 below for BT=0.3, BT=0.5 and BT=1.

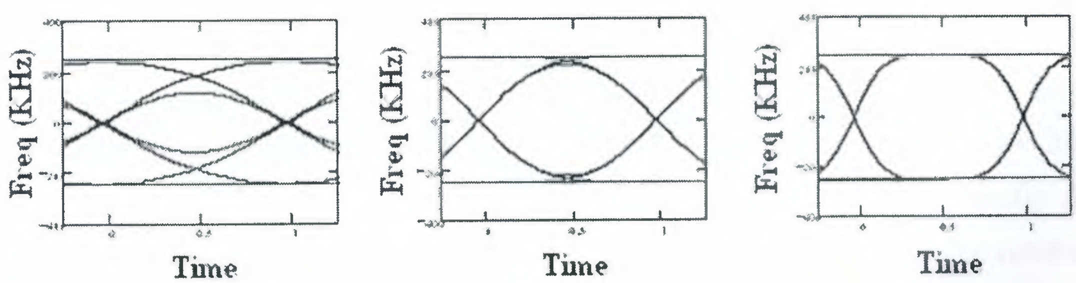


Figure 3.8 Eye Diagrams for GMSK with BT=0.3 (left), BT=0.5 (centre) and BT=1 (right)

This resultant reduction in the phase change of the carrier for the shaped symbols (i.e. 101 and 010) will ultimately degrade the BER performance as less phase has been accrued or retarded therefore less noise will be required to transform a zero to a one and vice versa. The principle advantages of GMSK, however, are the improved spectral efficiency and constant amplitude. The resulting signal spectra's for BT= 0.3, 0.5, 1 and MSK are shown below in Figure 3.9.

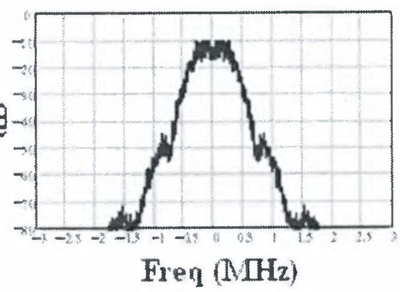


Figure 3.9 (a). BT=0.3

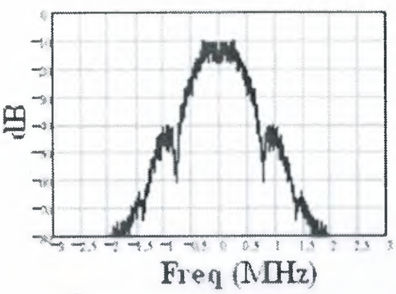


Figure 3.9 (b). BT=0.5

All the waveforms displayed above (GMSK and MSK) have constant amplitude. That is to say that their quadrature phase trajectory never leaves the unit circle. This can be a significant property, particularly as it allows the Power Amplifier device to be operated further into compression yielding improved efficiency and increased output power, without significant spectral re-growth.

3.8 Phase Shift Keying (PSK)

An alternative to imposing the modulation onto the carrier by varying the instantaneous frequency is to modulate the phase. This can be achieved simply by defining a relative phase shift from the carrier, usually equi-distant for each required state. Therefore a two level phase modulated system, such as Binary Phase Shift Keying, has two relative phase shifts from the carrier, + or - 90°. Typically this technique will lead to an improved BER performance compared to MSK. The resulting signal will, however, probably not be constant amplitude and not be very spectrally efficient due to the rapid phase discontinuities. Some additional filtering will be required to limit the spectral occupancy. Phase modulation requires coherent generation and as such if an IQ modulation technique is employed this filtering can be performed at base band.

3.8.1 Binary Phase Shift Keying (BPSK)

The simplest form of phase modulation is binary (two level) phase modulation. With theoretical BPSK the carrier phase has only two states, +/- JI/2. Obviously the transition from a one to a zero, or vice versa, will result in the modulated signal crossing the origin of the constellation diagram resulting in 100% AM. Figure below shows the theoretical spectrum of a 1 Mbits BPSK signal with no additional filtering. Several techniques are employed in real systems to improve the spectral efficiency. One such method is to employ Raised Cosine filtering. Figure 3.10(b) below shows the improved spectral efficiency achieved by applying a raised cosine filter with JI=0.5 to the base band modulating signals.

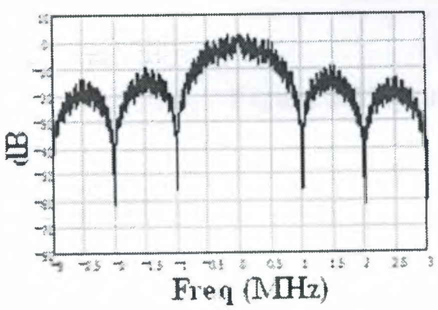


Figure 3.10 (a). Theoretical BPSK

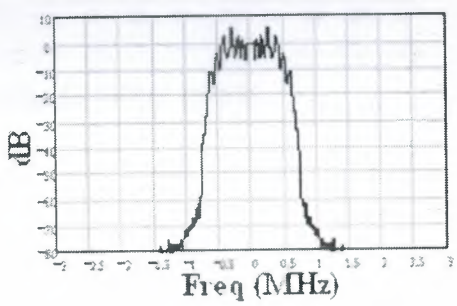


Figure 3.10 (b). Raised Cosine BPSK β=0.5

The improved spectral efficiency will result in some closure of the eye as can be seen in figure3.11 (a) and 3.11(b).

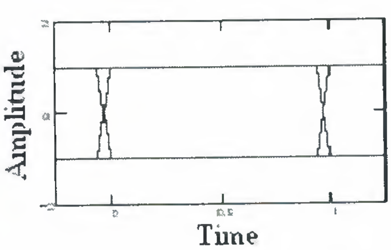


Figure 3.11 (a). Theoretical BPSK

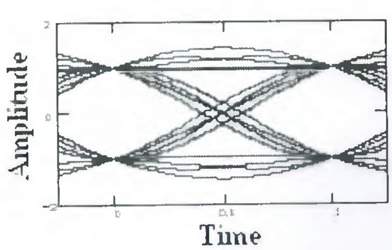


Figure 3.11 (b). Raised Cosine BPSK β=0.5

One potentially undesirable feature of BPSK that the application of a raised cosine filter will not improve is the 100% AM. In a real system the shaped signal will still require a linear PA to avoid spectral re-growth. Further hybrid versions of BPSK are used in real systems that combine constant amplitude modulation with phase modulation. One such example would be Constant Amplitude '50%' BPSK, generated with shaped I and Q vectors designed to rotate the phase around the unit circle between the two constellation points. For a 010 data sequence the trajectory spends 25% of the time traveling from one point to other, 50% of the time at the required point and 25% of the time returning. The resulting carrier phase shift is shown in Figure 3.12 below.

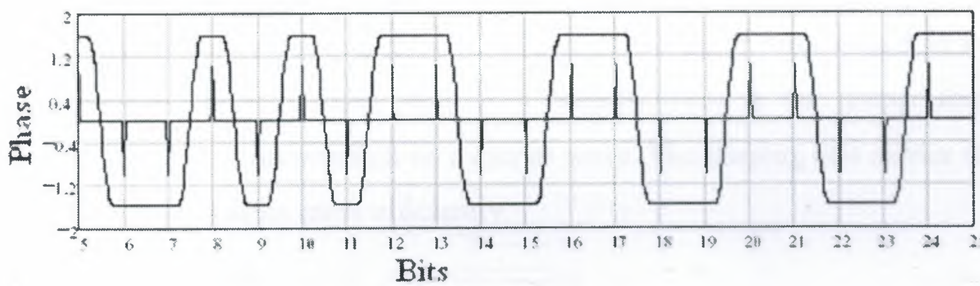


Figure 3.12 Constant Amplitude '50%' BPSK.

3.8.2 Quadrature phase shift keying (QPSK)

Let's discuss the awesomely titled quadrature phase shift keying or QPSK. This scheme, used by most high speed modems, allows quicker data transfer than FSK. And it gives at least four states to send information. There's a good chance we have heard this type as our modem makes a dial up connection. IS-136 uses this technology to enable its digital control channel, allowing PCS like services for conventional cellular. GSM also uses a variation, called, Gaussian Minimum Shift Keying, Quadrature phase shift keying changes a sine wave's normal pattern. It shifts or alters a wave's natural fall to rest or 0 degrees.

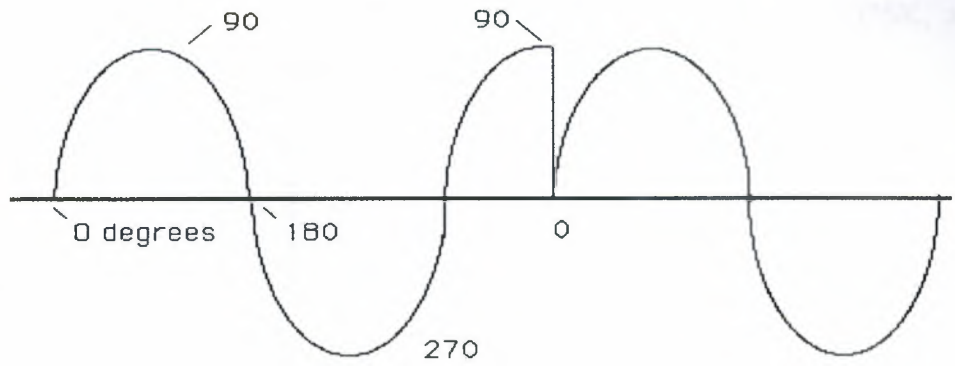


Figure 3.13 As an Example, 90 degrees, 0 degrees, 180 degrees, and 270 degrees might be represented by binary digits 00, 01, 10, and 11 respectively.

When arrange the circuit that at each point, it transmits a bit of force a shift in the sine wave. The receiver expects these shifts and decodes them in the proper sequence. Again, by putting digital information on a carrier wave. The shaping of a carrier wave to do this, to carry more pulses more efficiently.

Wireless services use amplitude, frequency, and phase modulation to send both analog and digital radio signals. But what converts an analog signal to digital in the first place? An encoding scheme. Pulse amplitude modulation first measures or samples the strength of an analog signal. Pulse code modulation encodes these plots into binary words, namely 0s and 1s. These binary digits are represented by on and off pulses of electrical energy.

A digital signal thus produced usually modulates the current carrying the signal within a landline. Modulation and pulses, therefore, get digital messages going. Once completed, the resulting digital signal can be sent over the air with another modulation technique for doing just that.

Higher order modulation schemes, such as QPSK, are often used in preference to BPSK when improved spectral efficiency is required. QPSK utilizes four constellation points, as shown in figure below, each representing two bits of data. Again as with BPSK the use of trajectory shaping (raised cosine, root raised cosine etc) will yield an improved spectral efficiency, although one of the principle disadvantages of QPSK, as with BPSK, is the potential to cross the origin, hence generating 100% AM.

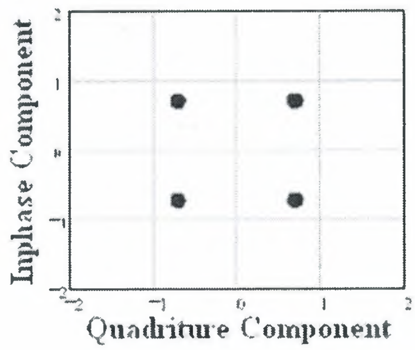


Figure 3.14 Constellation Points for QPSK.

3.8.3 Л/4Quadrature Phase Shift Keying (Л/4-QPSK)

A variant of QPSK that is employed in several digital systems is Л /4-QPSK. As with QPSK two bits are coded onto each symbol, although the quadrature constellations for adjacent bits are offset by Л /4 radians. The two sets of constellation points are shown in figure 3.15.

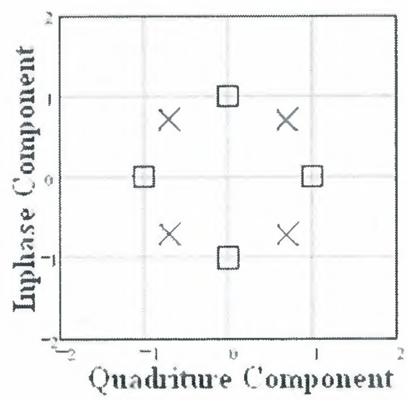


Figure 3.15 Constellation Points for Л/4-QPSK

One advantage of Л/4-QPSK is the improved spectral efficiency, compared to MSK and GMSK, particularly when used with raised cosine phase trajectory shaping due to coding two bits per symbol. Additionally the phase trajectory will no longer cross the origin, avoiding the generation of 100% AM, allowing a harder saturation mode of operation for the PA.

3.8.4 Offset Quadrature Phase Shift Keying (O-QPSK)

The final variant of QPSK to be considered is Offset Quadrature Shift Keying, or O-QPSK. As previously discussed the potential for a 180° phase shift in QPSK results in the requirement for better linearity in the PA and the potential for spectral re-growth due to the 100% AM. O-QPSK reduces this tendency by adding a time delay of one bit period (half a symbol) in the Quadrature arm of the modulator. The result is that the phase of the carrier is potentially modulated every bit (depending on the data), not every other bit as for QPSK, hence the phase trajectory never approaches the origin. The ability of the modulated signal to demonstrate a phase shift of 180° is therefore removed. As with the other phase modulation schemes considered, shaping of the phase trajectory between constellation points is typically implemented with a raised cosine filter to improve the spectral efficiency. Due to the similarities between QPSK and O-QPSK similar signal spectrum and probability of error are achieved. O-QPSK is utilized in the North American IS-95 CDMA cellular system for the link from the mobile to the base station.

Table 3.1 Distance Properties of PSK Modulations

Modulation	P_{ave}	d^2_{min} *Normalized*	*SNR Increase*
BPSK	1	4.00	–
QPSK	1	2.00	3.00 dB
8-PSK	1	0.5858	5.33 dB
16 PSK	1	0.1522	5.85 dB

3.9 Summary

In this chapter, we known that modulation is used to transform signals into a form that is suitable for the channel, we have seen FM, AM and PM and explanations concerning all types of modulation methods, and we have seen how GMSK is used to modulate signals in GSM system.

4. GSM TRANSMISSION PROCESS

4.1 Introduction

The basic components associated with a GSM transmitter and receivers are represented in Figure 4.1. These blocks are implemented in order to transmit and receive data with the least possible number of errors.

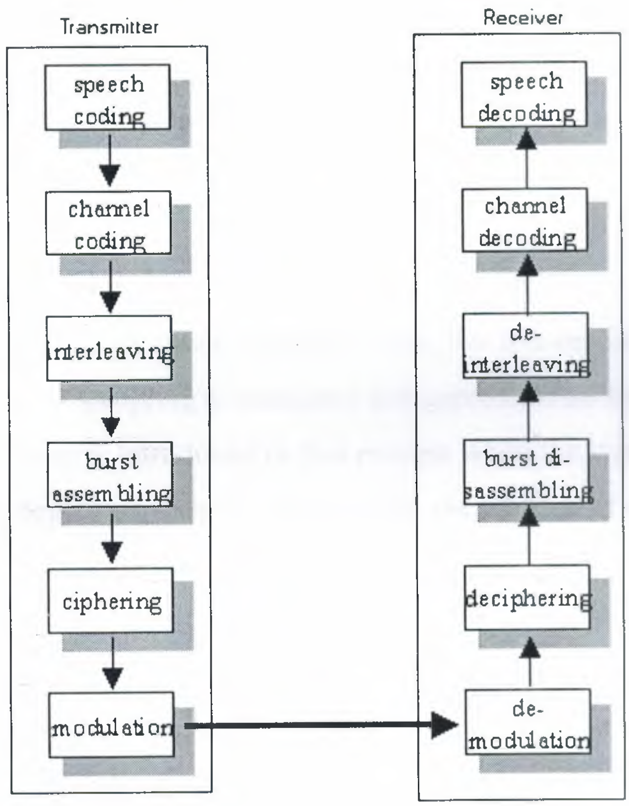


Figure 4.1 GSM Transmission Process.

4.2 Analog to Digital (A/D) Conversion

One of the primary functions of a MS is to convert the analog speech information into digital form for transmission using digital signal. The analog to digital (A/D) conversion process outputs a collection of bits: binary ones and zeros which represent the speech input. The A/D conversion is performed by using a process called Pulse Code Modulation (PCM). PCM involves the three main steps which are sampling, quantization, and coding.

4.2.1 Sampling

The accuracy of describing the analog signal in digital terms depends on how often the analog signal is sampled, among other things. This is expressed as the sampling frequency. The sampling theory states that:"To reproduce an analog signal without distortion, the signal must be sampled with at least twice the frequency of the highest frequency component in the analog signal".

Normal speech mainly contains frequency components lower than 3400 Hz. applying the sampling theory to analog speech signals; the sampling frequency should be at least 2 * 3.4 kHz = 6.8 kHz. Telecommunication systems use a sampling frequency of 8 kHz, which is acceptable based on the sampling theory.

4.2.2 Quantization

The next step is to give each sample a value. For this reason, the amplitude of the signal at the time of sampling is measured and approximated to one of a finite set of values. A slight error is introduced in this process when the signal is quantized or approximated. The degree of accuracy depends on the number of quantization levels used. In GSM 8192 levels are used.

4.2.3 Coding

Coding involves converting the quantized values into binary. Every value is represented by a binary code of 13 bits ($2^{13} = 8192$).

4.3 Speech Coding

In mobile communication systems, the design and subjective tests of speech coders have been extremely difficult. Without low data rate speech coding, digital modulations schemes offer little in the way of spectral efficiency for voice traffic. To make speech coding practical, implementations must consume little power and provides tolerable, if not excellent, speech quality.

The goal of all speech coding systems is to transmit speech with the highest possible quality using the least possible channel capacity. This has to be accomplished while maintaining certain required levels of complexity of implementation and communication delay. In general, there is a positive correlation between coder bit-rate efficiency and the algorithmic complexity required to achieve it. The more complex an algorithm is, the more its processing delay and cost of implementation. A balance needs to be struck between these conflicting factors. And it is the aim of all speech processing developments to shift the point at which this balance is made toward even low bit rates.

The GSM speech coder takes advantage of the fact that in a normal conversation each person speaks on average for less than 40% of the time. By incorporating a voice activity detector (VAD) in the speech coder, GSM systems operate in a discontinuous transmission mode (DTX) which provides a longer subscriber battery life and reduces instantaneous radio interference since the GSM transmitter is not active during silent periods. A comfort noise subsystem (CNS) at the receiving end introduces a background acoustic noise to compensate for the annoying switched muting which occurs due to DTX.

4.3.1 The Dimension of Performance in Speech Compression

Speech coders attempt to minimize the bit rate for transmission or storage of the signal while maintaining required levels of speech quality, communication delay, and complexity of implementation (power consumption). The following criteria represent the performance of speech compression:

4.3.1.1 Speech Quality

Speech quality is usually evaluated on a five-point scale, known as the mean-opinion score (MOS) scale, in speech quality testing an average over a large number of speech data, speakers, and listeners. The five points of quality are: bad, poor, fair, good, and excellent. Quality scores of 3.5 or higher generally imply high levels of intelligibility, speaker recognition and naturalness.

4.3.1.2 Bit Rate

The coding efficiency is expressed in bits per second (bps).

4.3.1.3 Communication Delay

Speech coders often process speech in blocks and such processing introduces communication delay. Depending on the application, the permissible total delay could be as low as 1 ms, as in network telephony, or as high as 500 ms, as in video telephony. Communication delay is irrelevant for one-way communication, such as in voice mail.

4.3.1.4 Complexity

The complexity of a coding algorithm is the processing effort required to implement the algorithm, and it is typically measured in terms of arithmetic capability and memory requirement, or equivalently in terms of cost. A large complexity can result in high power consumption in the hardware.

4.3.2 Speech Coders Categories

Speech coders differ widely in their approaches to achieve signal compression, based on the means by which they achieve compression, speech coders are broadly classified into two categories as shown in Figure 4.2: waveform coders and Vocoders.

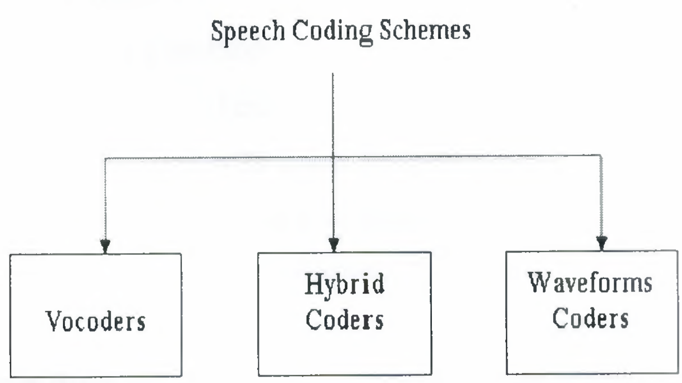


Figure 4.2 Speech Coders Categories.

4.3.2.1 Waveform Coders

Waveform coders essentially strive to reproduce the time waveform of the speech signal as closely as possible. They are designed to be source independent and hence code equally well a variety of signals. They have the advantage of being robust for a wide range of speech characteristics and for noisy environment .The waveform coders are able to produce high-quality speech at high enough bit rates.

4.3.2.2 Vocoders

Vocoders achieve very high economy in transmission bit rate, and it's in general more complex. They are based on using a priori knowledge about the signal to be coded, and for this reason, they are, in general, signal specific.
Vocoders produce intelligible speech at much lower bit rates and the applications of Vocoders so far, have been limited to low-bit-rate digital communication channels.

4.3.3 Speech Coding in GSM

GSM is a digital communications standard, but the voice is analog, and therefore it must be converted to a digital bit stream. GSM uses Pulse Coded Modulation (64kbps) to digitize voice, and then uses the Full-Rate speech codec to remove the redundancy in the signal and achieve a bit rate of 13 kbps.

In order to send our voice across a radio network, we have to turn our voice into a digital signal. GSM uses a method called RPE-LPC (Regular Pulse Excited - Linear Predictive Coder with a Long Term Predictor Loop) to turn our analog voice into a compressed digital equivalent with lower bit rate. Once we have a digital signal we have to add some sort of redundancy so that we can recover from errors when we tram our digital voice over the radio channel.

The LPC encoder fits a given speech signal against a set of vocal characteristics. The best-fit parameters are transmitted and used by the decoder to generate synthetic speech that is similar to the original. Information from previous samples is used to predict the current sample. The coefficients of the linear combination of the previous samples, plus an encoded form of the residual, the difference between the predicted and actual sample, represent the signal. Speech is divided into 20 millisecond samples, each of which is encoded as 260 bits, giving a total bit rate of 13 kbps. See Figure 4.3 for a representation of RPE-LPC.

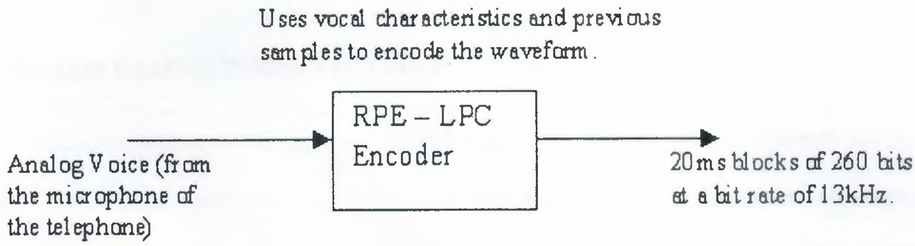


Figure 4.3 RPE-LPC Encoder.

4.4 Channel Coding

Channel coding refers to the class of signal information's designed to improve communications performance by enabling the transmitted signal to better withstand the effects of various channel impairment, such as noise, interference, and fading. It introduces redundancy into the data flow in order to allow the detection or even the correction of bit errors introduced during the transmission. In order to create a reliable connection we need to protect the data as much as possible at that bit error rate (BER) we cannot rely on error detection and retransmission. So in order to get over this you have to use channel coding.

4.4.1 Convolutional Encoders

A convolutional code which is used in GSM is described by three integers, n, k, and K, where the ratio k/n has the same code rate significance (information by coded bit) that it has for block codes; however, n does not define a block or codeword length as it does for block codes. The integer K is a parameter known as the constraint length; it represents the number of k-tuple stages in the encoding shift register, the n-tuple emitted by the convolutional encoding procedure is not only a function of an input k-tuple, but is also a function of the previous K-1 input k-tuples.

At each unit of time, k bits are shifted into the first k stages of the register; all bits in the register are shifted k stages to the right, and the outputs of the n adders are sequentially sampled to yield the binary code symbol or code bits. These code symbols are then used by the modulator to specify the waveforms to be transmitted over the channel. GSM choose to employ a convolutional encoder due to its ability to efficiently correct errors.

4.4.2 Channel Coding Process in GSM

The speech coding produces a 260 bit block for every 20 ms speech sample. From subjective testing, it was found that some bits of this block were more important for perceived speech quality than others. The bits are thus divided into three classes (See Figure 4.4):

- Class Ia 50 bits - most sensitive to bit errors.
- Class Ib 132 bits - moderately sensitive to bit errors.
- Class II 78 bits - least sensitive to bit errors.

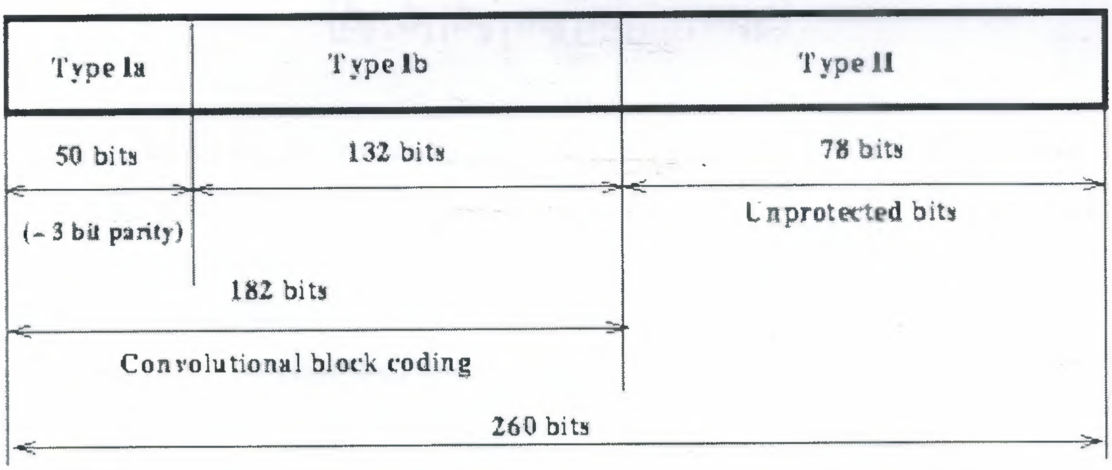


Figure 4.4 Channel Coding in GSM.

As a result of some bits being more important than others, GSM adds redundancy bits to each of the three classes differently. The class Ia bits are encoded in a cyclic encoder. The class Ib bits (together with the encoded class Ia bits) are encoded using convolutional encoding. Finally, the class II bits are merely added to the result of the convolutional encoder. The Figure below is the operation of each encoder as related to each class of bits.

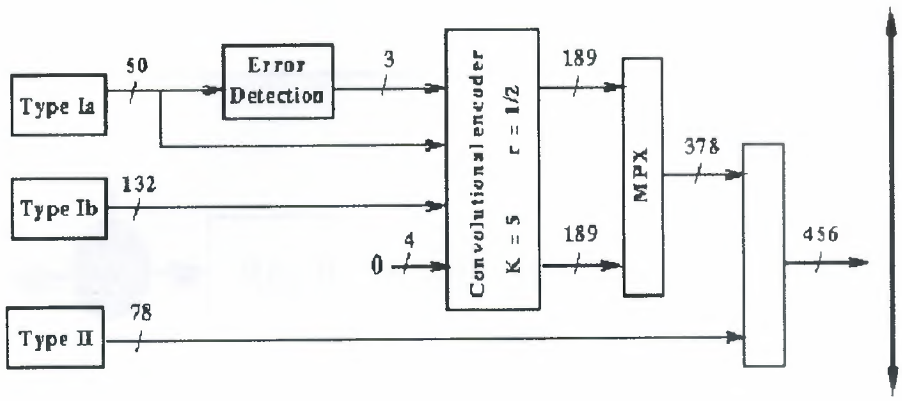


Figure 4.5 A Block Diagram of a Convolutional Encoder.

4.4.2.1 Cyclic Encoding

The class Ia bits are encoded using a cyclic encoder to add three bits of redundancy. The resulting class Ia bits are of the form:

$$(b_0, b_1, b_2, m_0, m_1, \ldots, m_{49})$$

Where b_0, b_1, b_2 are the three redundancy bits added by the cyclic encoder, and $m_0 \ldots m_{49}$ are the original Class Ia bits. The cyclic encoder produces 50+3=53 bits.

Cyclic codes are linear codes (the sum of any two codes is also a codeword). In addition to being linear, a cyclic shift, or rotate, of a codeword produces another codeword, since the code used in GSM is a (53, 50) code, the generator polynomial used in the encoding is of degree 53-50 = 3. The following block diagram can produce the codeword. In Figure 4.6 once the data has been completely shifted through the system, the contents of Reg0 through Reg2 will contain the three additional bits.

GSM chooses to use cyclic encoding due to the ability to quickly determine if errors are present. The three redundancy bits produced by the cyclic encoder enable the receiver to quickly determine if an error was produced. If an error was produced the current 53 bit frame is discarded and replaced by the last known "good" frame.

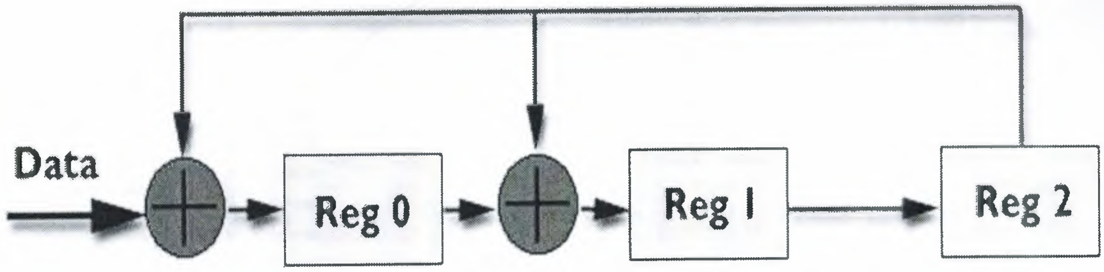


Figure 4.6 Cyclic Encoding.

4.5 Interleaving

Interleaving is meant to decorrelate the relative positions of the bits respectively in the codewords and in the modulated radio bursts. The aim of the interleaving algorithm is to avoid the risk of losing consecutive data bits.

Interleaving is used in:

- Time-Division Multiplexing (TDM).
- Computer Memory.
- Disk Storage.

GSM blocks of full rate speech are interleaved on 8 bursts: the 456 bits of one block are split into 8 bursts in sub-blocks of 57 bits each. A sub-block is defined as either the odd- or the even-numbered bits of the coded data within one burst. Each sub-block of 57 bit is carried by a different burst and in a different TDMA frame. So, a burst contains the contribution of two successive speech blocks A and B. In order to destroy the proximity relations between successive bits, bits of block A use the even positions inside the burst and bits of block B, the odd positions as shown in Figure 4.7.

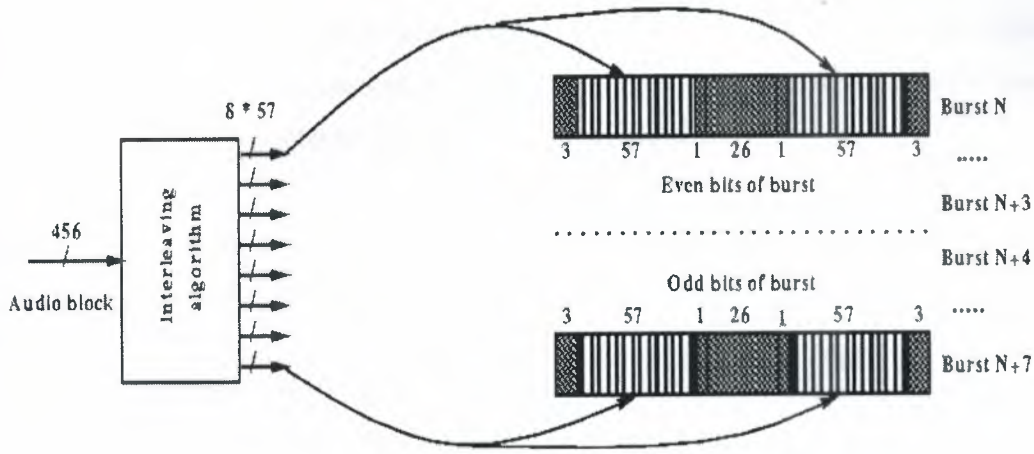


Figure 4.7 Interleaving Operation.

4.6 Burst Format

GSM uses both FDMA and TDMA for multiple accesses. GSM900 uses the 890 MHz to 960 MHz frequency band for communications (Uplink 890-915 MHz, Downlink 935-960 MHz). GSM900 has two frequency bands 45 MHz. The 890 MHz to 915 MHz frequency band is used for uplink from a MS to a BS and 935 MHz to 960 MHz frequency band is used for the downlink from a BS to a MS. Each of the two bands is subdivided into 124 single carrier channels of 200 KHz each. A guard band of 200 KHz is left on both sides of uplink and the downlink bands (See Figure 4.8).

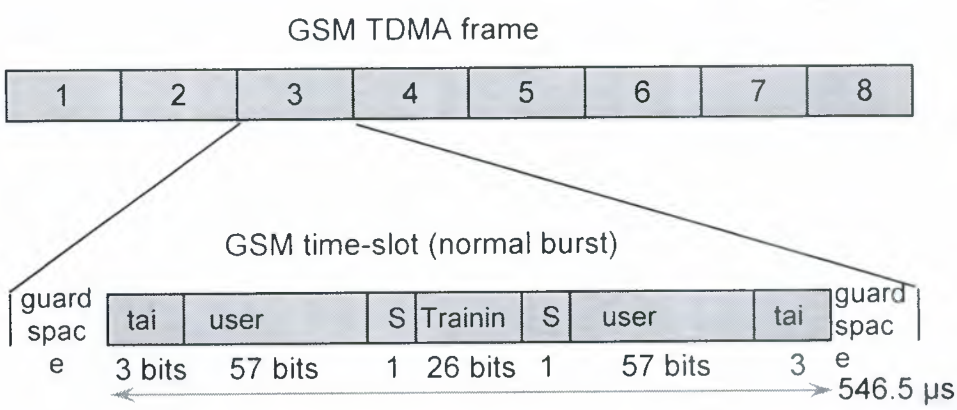


Figure 4.8 Burst Formats in GSM.

In figure 4.9, each of the 200 KHz channels is divided into eight time slots and carries eight TDMA channels. The eight time slots form a TDMA frame. A MS uses the same time slot number in the uplink and the downlink. Each time slot of a TDMA frame lasts for 576.9 microseconds and contains data, which is also called a burst. The transmission rate is 270.83 kbps per carrier frequency.

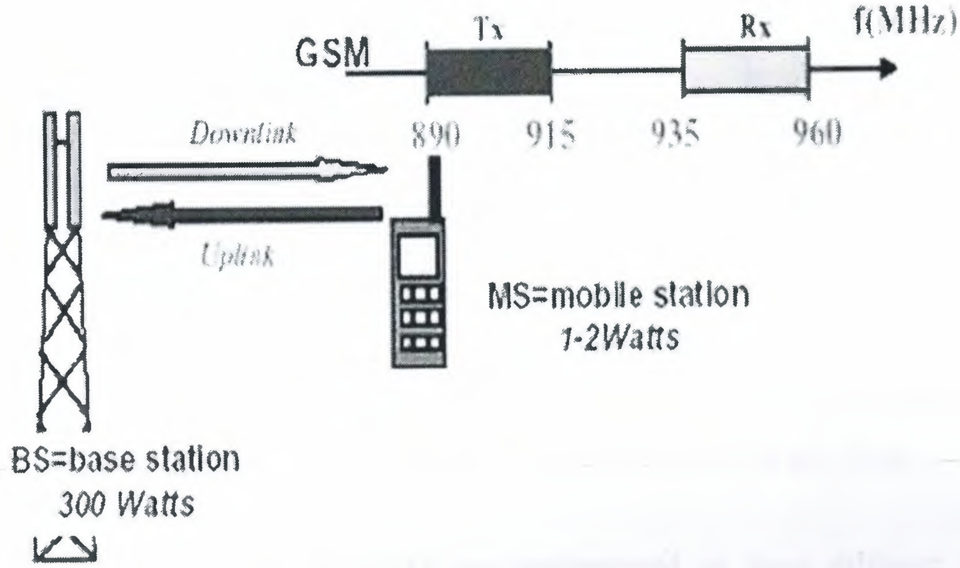


Figure 4.9 GSM900 Frequency Allocation.

4.6.1 Normal Burst

A normal burst consists of 148 bits, which is a mix of information. A normal burst contains 26 training sequence bits termed Equalization Bits. This sequence is one of the fixed combinations of bits that are known to the MS and the BS. The existence of multipath causes the original signal to fade. Therefore, it is difficult to exactly identify the transmitted signal at the receiver. The training bit sequence helps in estimating the channel response, optimize reception and reduce inter-symbol interference caused by propagation time differences in the multipath channel. The structure of a normal burst is represented in Figure 4.10.

3	57	1	26	1	57	3
Tail bits	Data bits	CTRL bit	Equalization Bits	CTRL bit	Data bits	Tail bits
{000}		{1/0}		{1/0}		{000}

Figure 3.10 Normal Burst.

Stealing flag bits reside on both sides of the training sequence. They can either be '1' or '0'. They are signaling bits that indicate whether the burst contains traffic data or signaling data. In addition to the stealing flags there are two blocks of 57 bits each containing channel-coded user data. Therefore, the actual voice data consists of 114 bits of the 148 total bits transmitted. Toward the end there are 3 tail bits on each side, which are set to '0'. The tail bits are required for the demodulation process. There is also a guard period that equals a bit period of 8.25 bits or 30.4 microseconds.

4.7 Ciphering

The purpose of ciphering is to encode the burst so that it cannot be interpreted by any device other than the intended receiver. The ciphering algorithm in GSM is called A5 algorithm. It does not add bits to the burst, meaning that the input and the output to the ciphering process are the same as the input: 456 bits per 20ms.

The security mechanisms of GSM are implemented in three different system elements; the Subscriber Identity Module (SIM), the GSM handset or MS, and the GSM network. The SIM contains the IMSI, the individual subscriber authentication key (Ki), the ciphering key generating algorithm (A8), the authentication algorithm (A3), as well as a Personal Identification Number (PIN). The GSM handset contains the ciphering algorithm (A5). The encryption algorithms (A3, A5, and A8) are present in the GSM network as well.

Distribution of security information is among the three system elements, the SIM, the MS, and the GSM network. Within the GSM network, the security information is further distributed among the authentication centre (AUC), the home location register (HLR) and the visitor location register (VLR). The AUC is responsible for generating the sets of RAND, SRES, and Kc, which are stored in the HLR and VLR for subsequent use in the authentication and encryption processes.

4.8 Modulation

GSM uses GMSK modulation, which stands for Gaussian Minimum Shift Keying; GMSK uses a Gaussian filter from which its name was derived. The frequency separation utilized is the minimum frequency separation required for the two modulating signals to be orthogonal over a signaling interval of length T. One of the advantages of GMSK modulation is that it does not produce any Intersymbol Interference (ISI). GMSK modulation is continuous phase modulation, which achieves a good compromise between power and bandwidth efficiency while maintaining a specified level of BER at the expense of only reasonable increase of system complexity.

The relation between premodulation bandwidth, B, and the bit period, T, defines bandwidth of the system. GSM uses BT as 0.3 with the modulating symbol rate of 270.833 Kbps. Prior to the first bit entering the modulator, the modulator resides in an internal state that represents the modulation bit stream consisting of consecutive ones had been entered to a differential encoder. After the last bit of the time slot the modulator returns to an internal state that represents the modulation of bit stream consisting of ones. These bits are called dummy bits and define the start and the stop of the active and useful part of the burst as presented in Figure 4.11.

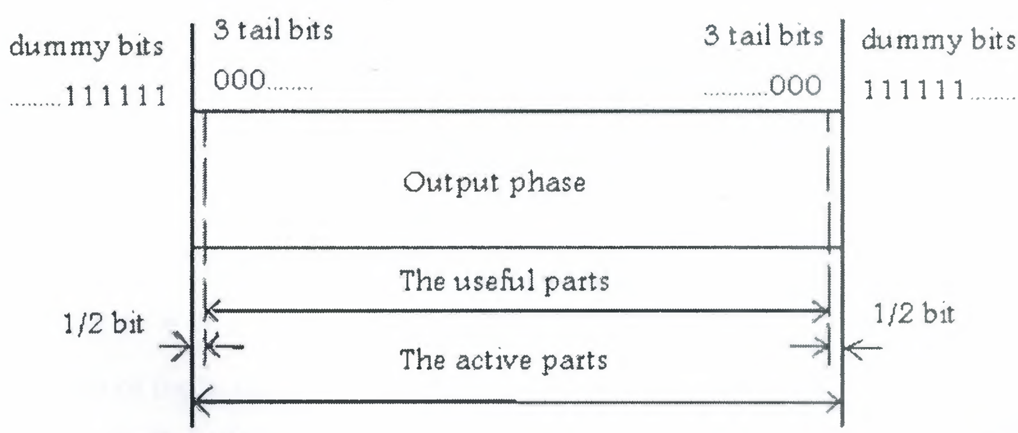


Figure 4.11 Relations between Active Part of Burst, Tail Bits and Dummy Bits

Before the burst enters the modulator, it is differentially coded. The differential encoder sequence is then applied to the Gaussian filter in the modulator. The GMSK modulator is shown in Figure 4.12.

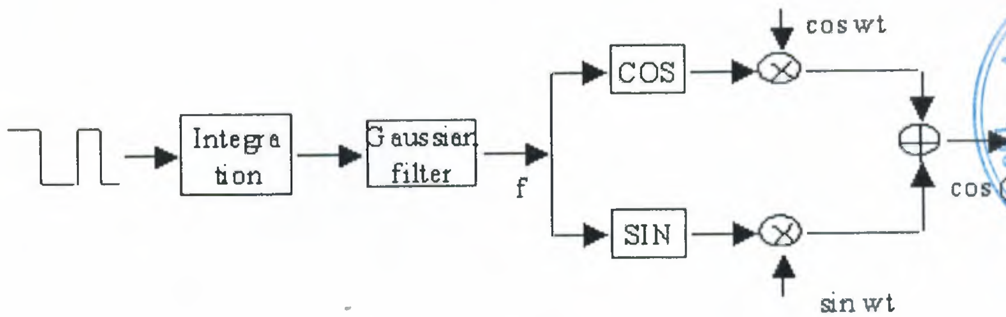


Figure 4.12 GMSK Modulation.

The Gaussian low pass filter has an impulse response given by the following equation

$$h(t) = (\pi)^{1/2}/a \ \ \exp(-\ (\pi^2 / a^2)\, t^2) \qquad (4.1)$$

The parameter (a) is related to B, 3-dB bandwidth of base band Gaussian shaping filter,

$$a = 0.5887/B \qquad (4.2)$$

The filter response is represented graphically in the following Figure 4.13.

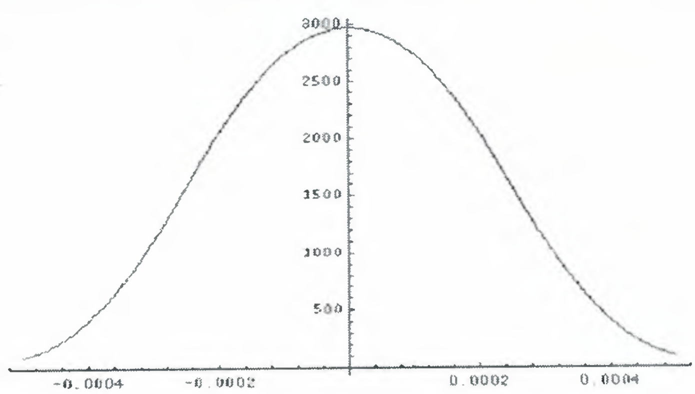


Figure 4.13 The Response of Gaussian Filter

The output of the modulator is given by

$$m(t)= \sin(2\pi f_c t)I(t) + \cos(2\pi f_c t)Q(t) \qquad (4.3)$$

Where f_c is the carrier frequency used from the oscillator. The final output of the modulator, which is sent through the channel, is shown in figure 4.14.

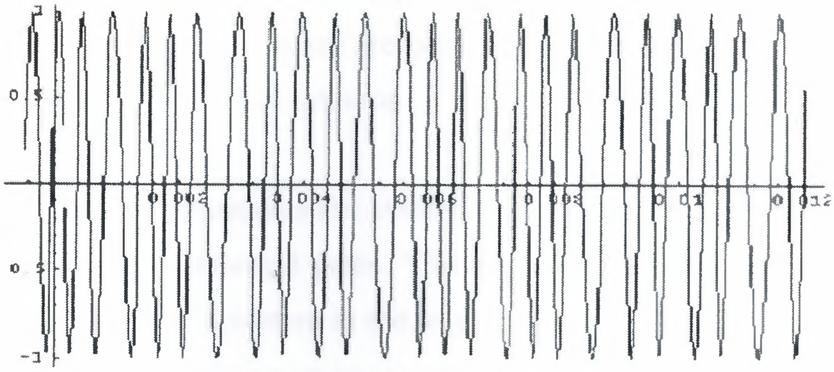


Figure 4.14 Final Output of the Modulator.

4.9 Multipath Channels

In the study of communication systems, the classical (ideal) additive-white-Gaussian-noise (AWGN) channel, with statistically independent Gaussian noise samples corrupting data samples free of intersymbol interference (ISI), is the usual starting point for understanding basic performance relationships. The primary source of performance degradation is thermal noise generated in the receiver. Often, external interference received by the antenna is more significant than the thermal noise .The thermal noise usually has a flat power spectral density over the signal band and a zero-mean Gaussian voltage probability density function (pdf).

If a radio channel's propagating characteristics are not specified, one usually infers that the signal attenuation vs. distance behaves as if propagation takes place over ideal free space. The model of free space treats the region between the transmitter and receiver antennas as being free of all objects that might absorb or reflect radio frequency (RF) energy. It also assumes that, within this region, the atmosphere behaves as a perfectly uniform and non absorbing medium. Furthermore, the earth is treated as being infinitely far away from the propagating signal (or, equivalently, as having a reflection coefficient that is negligible). Basically, in this idealized free-space model, the attenuation of RF energy between the transmitter and receiver behaves according to an inverse-square law.

For most practical channels, where signal propagation takes place in the atmosphere and near the ground, the free-space propagation model is inadequate to describe the channel and predict system performance.

In a wireless mobile communication system, a signal can travel from transmitter to receiver over multiple reflected paths. This phenomenon, referred to as multipath propagation, can cause fluctuation in the received signal amplitude, phase, and angle of arrival, giving rise to the terminology multipath fading (See Figure 4.15).

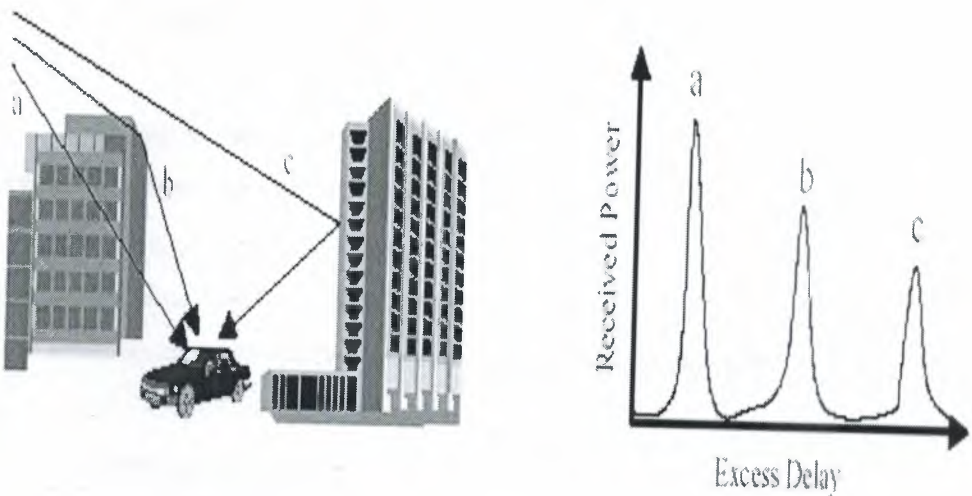


Figure 4.15 Multipath Fading

4.9.1 Fading Types

There are two types of fading that characterize mobile communication. These types are:

4.9.1.1 Large Scale Fading

Represents the average signal power attenuation or the path loss due to motion over large areas. This phenomenon is affected by prominent terrain contours (e.g., hills, forests, billboards, clumps of building, etc.) between the transmitter and receiver. The statistics of large scale fading provide a way of computing an estimate of path loss as a function of distance. This is often described in terms of a mean-path loss (*n*th-power law) and a log-normally distributed variation about the mean.

4.9.1.2 Small Scale Fading

Refers to the dramatic changes in signal amplitude and phase that can be experienced as a result of small changes in the spatial positioning between a receiver and a transmitter.

Small scale fading manifests itself in two mechanisms, time spreading of the signal and time variant behavior of the channel. For mobile radio application, the channel is time-variant because motion between the transmitter and receiver results in propagation path changes. The rate of change of these propagation conditions accounts for the fading rapidity (rate of change of the fading impairments).

Small scale fading is called Rayleigh fading if there are multiple reflective paths that are large in number, and if there is no line-of-sight signal component: the envelop of such a received signal is statistically described by a Rayleigh pdf. When there is a dominant nonfading signal component present, such as a line-of-sight propagation path, the small-scale fading envelope is described by a Rician pdf.

Figure 4.16 illustrates the relationship between large-scale and small-scale fading. For the case of a mobile radio, Small-scale fading superimposed on large-scale fading.

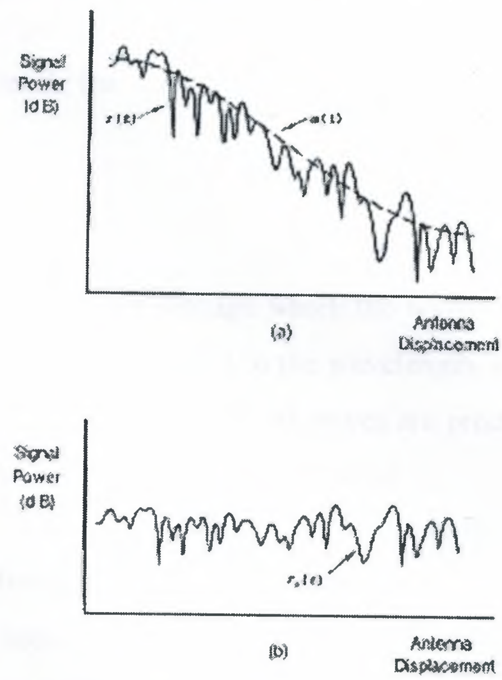


Figure 4.16 Large and Small Scale Fading.

4.9.2 The Propagation Mechanism

There are three basic mechanisms that impact signal propagation in a mobile communication system. They are reflection, diffraction, and scattering.

4.9.2.1 Reflection

Occurs when a propagating electromagnetic wave impinges upon an object which has very large dimensions compared to the wavelength of the propagating wave.

4.9.2.2 Diffraction

Occurs when the radio path between the transmitter and receiver is obstructed by a surface that has sharp irregularities (edges), The secondary waves resulting from the obstructing surface are present throughout the space and even behind the obstacle, giving rise to a bending of waves around the obstacle, even when a line-of -sight path does not exist between transmitter and receiver. At high frequencies, diffraction, like reflection, depends on the geometry of the object, as well as the amplitude, phase, and polarization of the incident wave at the point of diffraction.

It's a phenomenon that accounts for RF energy traveling from transmitter to receiver without a line-of-sight path between them. It is often termed shadowing because the diffracted field can reach the receiver even when shadowed by an impenetrable obstruction.

4.9.2.3 Scattering

Occurs when the medium through which the wave travels consists of objects with dimension that are small compared to the wavelength, and where the number of obstacles per unit volume is large. Scattered waves are produced by rough surfaces, small objects, or by other irregularities in the channel.

4.9.3 Noise and Interference in Multipath Channels

The received signal in a multipath channel exhibits large variations in magnitude as shown in Figure 4.17. Although the mean SNR (or C/I) might be acceptable, the variations experienced in the multipath channel mean that occasionally the noise will be far more significant. At these times the system will experience a large number of errors.

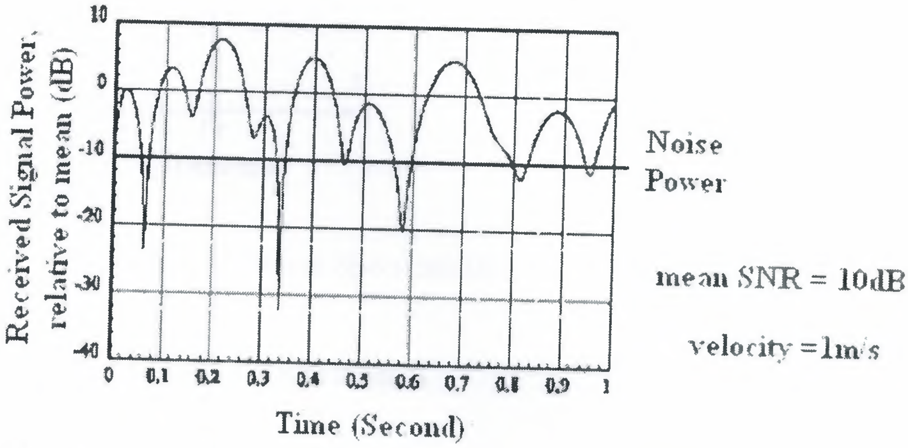


Figure 4.17 Noise and Interference in the Multipath Channel

4.9.4 Doppler Shift

Motion of the mobile causes periodic phase shifts which change with time. A typical spectrum for a Rayleigh channel is shown in the figure below. The rate of change of phase gives rise to a Doppler frequency (*Fd*), which varies with mobile speed (v) and the angle of arrival of the rays (θ).

$$Fd = (v/\lambda) \cos \theta \quad (\lambda = \text{wavelength}) \tag{4.4}$$

From equation (4.4) if the mobile is moving toward the direction of arrival of the wave, the Doppler shift is positive (i.e., the apparent received frequency is increased), and if the mobile is moving away from the direction of arrival of the wave, the Doppler shift is negative (i.e., the apparent received frequency is decreased) (See Figure 4.18).

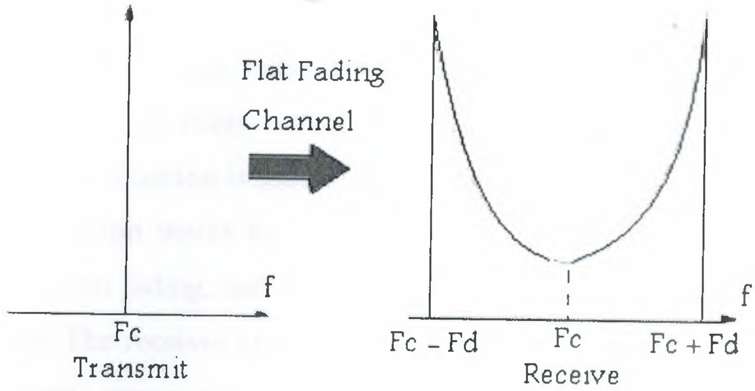


Figure 4.18 Power Spectrum Density for Rayleigh Fading.

4.9.5 Rayleigh and Rician Distribution

4.9.5.1 Rayleigh Fading Distribution

In mobile radio channel, the Raleigh distribution is commonly used to describe the statistical time varying nature of the received envelope of a flat fading signal, or the envelope of individual multipath components.

A multipath channel without a significant deterministic component can be approximated to a Rayleigh distribution where the received signal experiences large variations in magnitude.

4.9.5.2 Rician Fading Distribution

When there is a dominant stationary (nonfading) signal component present, such as a line-of-sight propagation path. The small-scale fading envelope distribution is Rician. In such a situation, random multipath components arriving at different angles are superimposed on a stationary dominant signal. At the output of an envelop detector, this has the effect of adding a dc component to the random multipath.

4.10 Equalizer

At the 900 MHz range, radio waves bounce off everything (buildings, hills, cars, airplanes, etc.). Thus many reflected signals, each with a different phase, can reach an antenna. Equalization is used to extract the desired signal from the unwanted reflections. Equalization works by finding out how a known transmitted signal is modified by multipath fading, and constructing an inverse filter to extract the rest of the desired signal. The receiver knows which training sequence it must wait for. The equalizer will then, comparing the received training sequence with the training sequence it was expecting, compute the coefficients of the channel's impulse response. This known signal is the 26 bit training sequence transmitted in the middle of every time slot burst.

4.11 Summary

In this chapter, we have discussed the transmission process step by step starting with the first block which is speech coding, The goal of all speech coding systems is to transmit speech with the highest possible quality using the least possible channel capacity, before that done an analog to digital conversion is being applied, after that it passes through a channel coding block where some redundant information being added to obtain detection and correction in the receiver side, Interleaving used to avoid the risk o losing consecutive data bits and to bring independency positions respectively in the codewords and in the modulated radio bursts, then the burst format are used to gain some allocated place in burst after determinig guard band between each single GSM time slot, Ciphering process are applied, The purpose of ciphering is to encode the burst so that it cannot be interpreted by any device other than the intended receiver, in the end of ciphering the input and the output will be the same, finally a Modulation process is applied for the signal in order to transmit in a compatible form that matches the channel properties.

5. METHODOLOGY

5.1 Introduction

This chapter introduces the methodology of the transmission and reception process of the GSM system, explained in chapter four.

Figure 5.1 shows the transmission process from MS to BTS (uplink) which includes different stages, starts when speech signal inters the MS and finishes when the speech signal is modulated. These stages which the speech signal passed through them are: A\D Conversion, Segmentation, Speech Coding, Channel Coding, Interleaving, Burst Format, and Modulation. The input data rate for one user is 13 kbps and the output data rate for one user is 33.8 kbps. But in GSM one carrier frequency can support eight users so the data rate for eight users equal to 270.4 kbps the signal is transmitted through the channel that has affects a signal by changing phase, amplitude of the signal, and makes delay in the signal.

The reception process from BTS to MS (downlink) includes different stages, starts when the transmitted signal enters the MS and finishes when the transmitted signal is converted to speech. These stages which the transmitted signal passed through them are: Demodulation, Burst Deformats, DeInterleaving, Channel Decoding, Speech Decoding, and D\A Conversion, each step will be explained in details in this chapter.

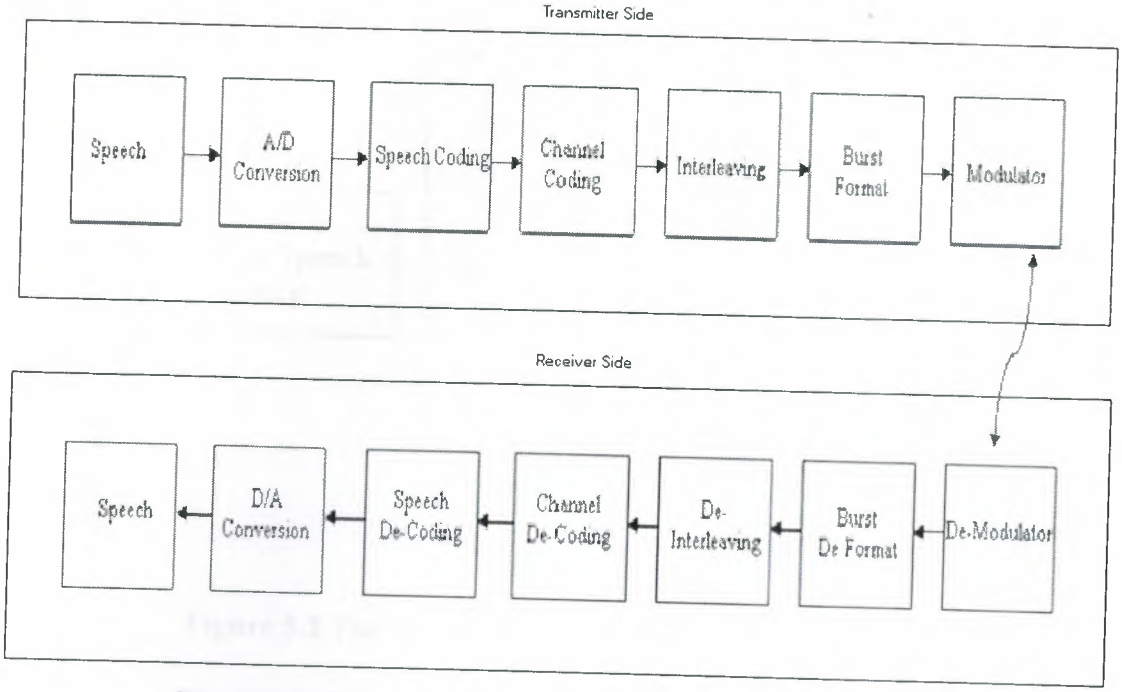


Figure 5.1 The GSM Transmission and Reception Process

5.2 Transmission Procedure

5.2.1 Speech Coding

The speech signal is framed into a 20 ms frames of 13 bits which is coded into 260 bits frames that gives an overall data rate of 13 Kbps for each user.

These bits arrange in a matrix, the size of this matrix is (260*1); The 260 bits are split according to their relative importance:

- Block 1: 50 very important bits.
- Block 2: 132 important bits.
- Block 3: 78 not so important bits.

The split operation is done by using Sub-Matrix block. Actually three Sub Matrix Blocks are needed to achieve the three blocks as seen in Figure 5.2 **[11]**.

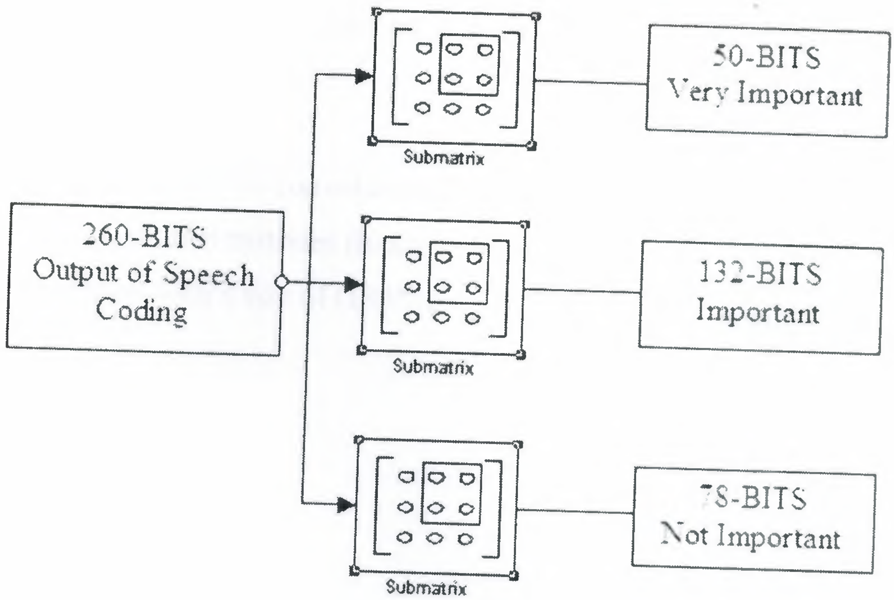


Figure 5.2 The Split Operation Using Sub-Matrix Block

5.2.2 Channel Coding

The first block of 50 bits is sent through a block coder (CRC), which adds three parity bits to result 53 bits as shown in the Figure 5.3. These three bits are used to detect the errors in the received message at the receiver side.

To add three bits, a generator polynomial is set to equal (x3 + x2 + 1) and it's represented as ([1 1 0 1]) in the CRC block.

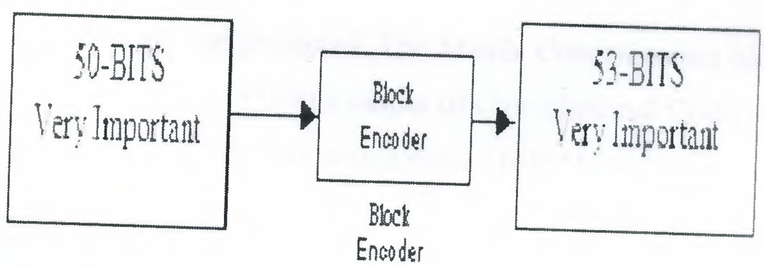


Figure 5.3 The CRC Coding for Very Important Bits

These 53 bits, the 132 bits in the second block and 4 tail bits (total =189) are sent to a 1:2 convolutional coder which result a total number of 378 bits as shown in Figure 5.4. The bits added by convolutional coder enable the error correction when the message is received at the receiver. The Bernoulli Binary Generator block is used to generate a 4 tail bits used to make separation between the important bits and not important bits.

Before the 189 bits enter the convolutional coder, The Matrix Concatenation block is used to concatenate input matrices (blocks: 53 bits, the 132 bits and 4 tail bits) so, the output is one matrix with a size of (189*1).

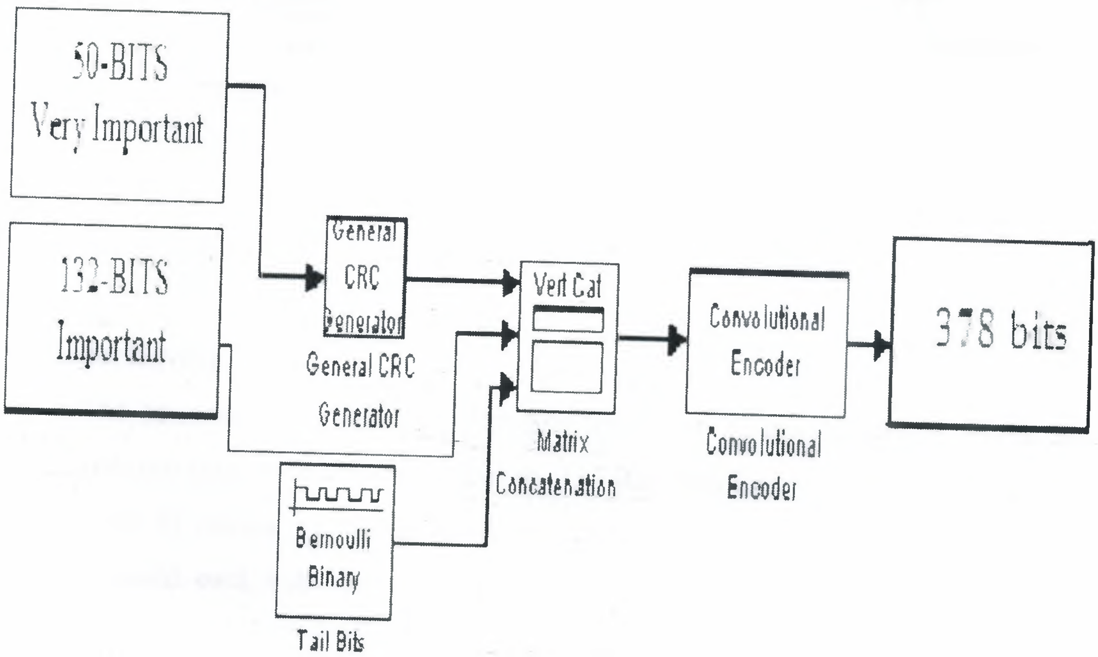


Figure 5.4 Convolutional Coder Operation

The remaining 78 bits are not protected. The Matrix Concatenation block used again to concatenate input matrices (378 bits output of Convolutional Coder, and 78 bits) to produce one matrix contains 456 bits with a size of (456*1) as shown in Figure 5.5.

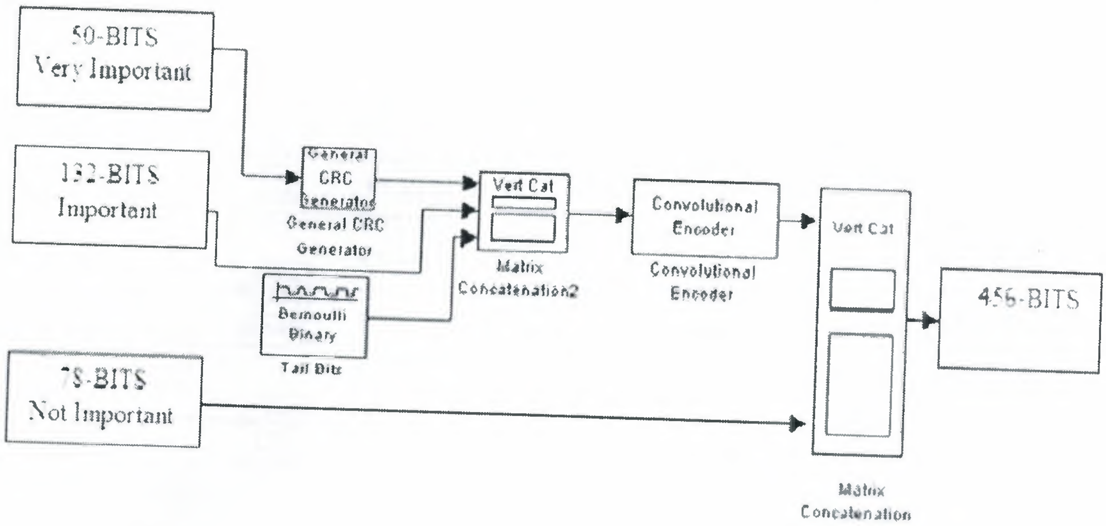


Figure 5.5 The Channel Coding

5.2.3 Interleaving

The channel coder provides 456 bits for every 20 ms of speech. These are interleaved into eight blocks of 57 bits each. The Random Interleaver block rearranges the 456 bits to reduce the error as shown in Figure 5.6. Then, Eight Sub-Matrix Blocks are used, each Sub-Matrix select 57 bits as shown in Figure 5.7.

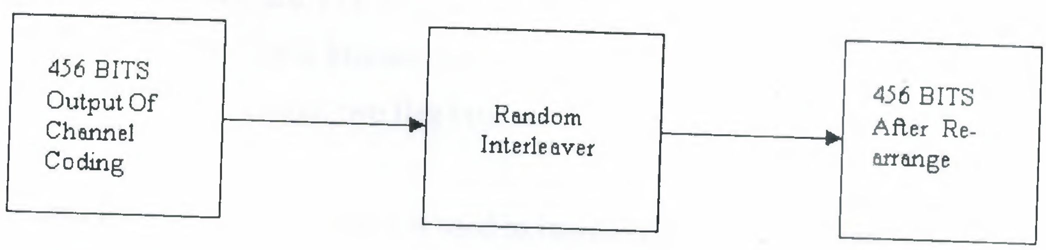


Figure 5.6 The Output of Random Interleaver

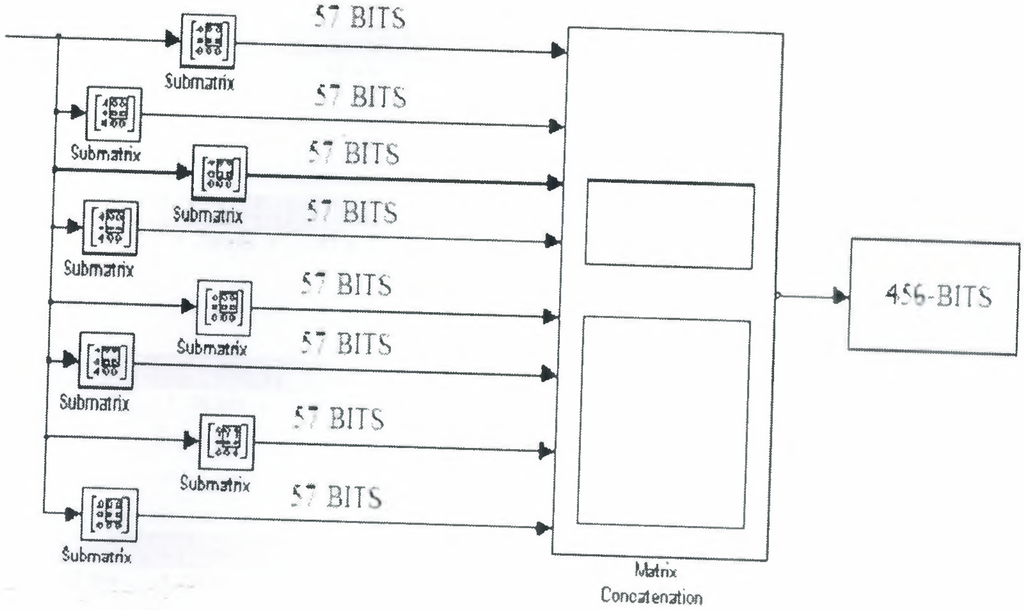


Figure 5.7 The Eight Sub-Matrix Blocks

5.2.4 Burst Format

A normal burst consists of 148 bits, which is a mix of information bits and signaling bits.

The information bits are 114 bits (two blocks contain 57 bits). The Signaling bits divided into 26 Bits, these bits are called Training Sequence, which is known for the mobile and the base station, two flag bits, and six tail bits.

Bernoulli Binary Generator block is used to represent each frame. From Figure 5.8 the results are four frames, each one contain 148 bits. These frames are entered the matrix concatenation block to concatenate the four frames to produce one frame contains 456 bits (4*148) **[1]**.

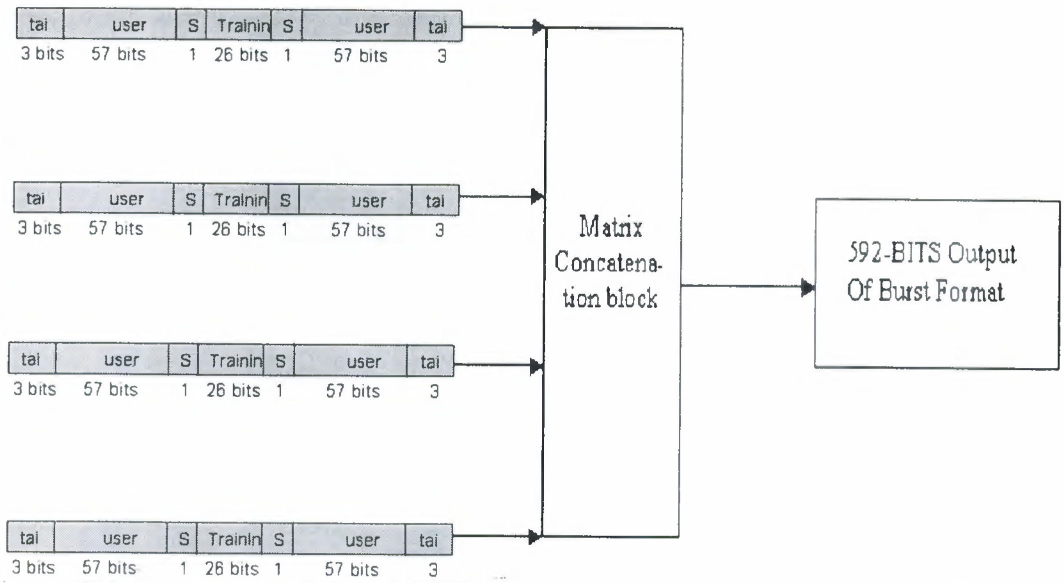


Figure 5.8 The Output of Burst Format

5.2.5 Modulation

The input to the modulation is 456 bits, and these bits will be transmitted through the channel.

In order to illustrate modulation procedure see Figure 5.9, and consider the following sequence, which repeats after every 12 bits.

1,1,-1,1,1,-1,-1,1,-1,1,-1,-1,1,1,-1,1,1,-1,-1,1,-1,1,-1,-1----------

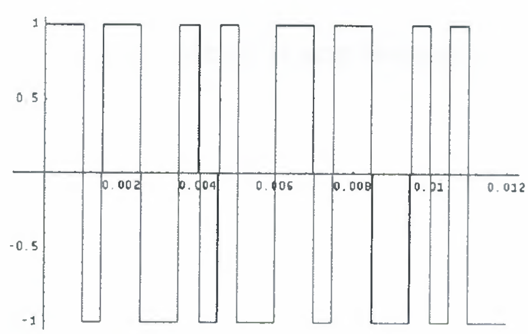


Figure 5.9 Data Stream Being Sent

The GMSK modulator is presented in the Figure 5.10 for more information about the GMSK Modulator block see appendix.

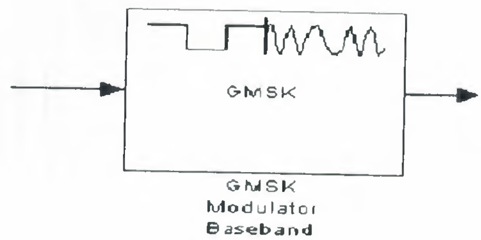


Figure 5.10 GMSK Modulation

The first few Gaussian shaped pulses are represented graphically in the Figure 5.11.

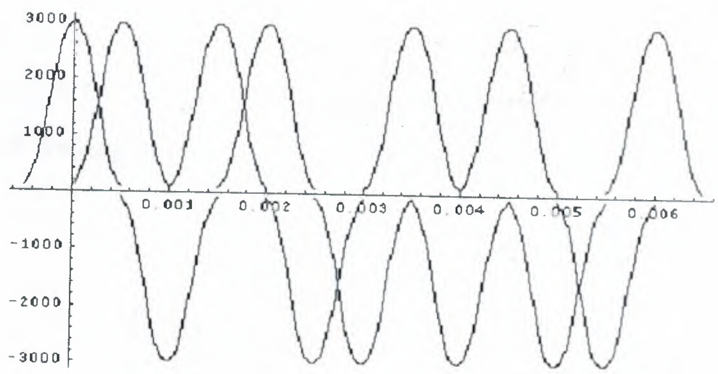


Figure 5.11 Individual Shapes Pulses Representing the Data Stream

The final output of the modulator, which is sent through the channel, represented in Figure 5.12.

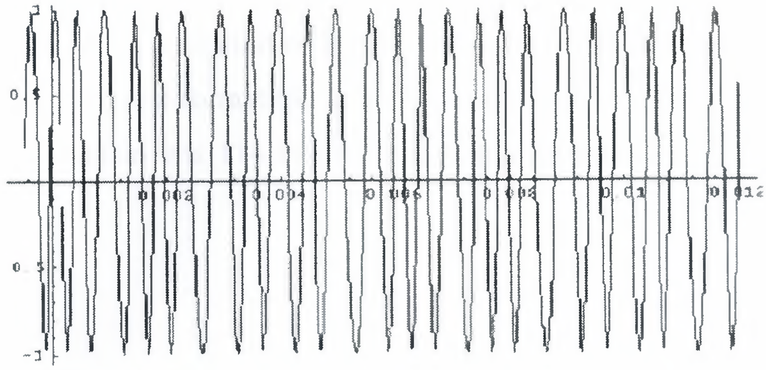


Figure 5.12 Final Output of the Modulator

5.3 Reception Procedure

5.3.1 Demodulation

The Demodulation used to recover the baseband signal that was modulated using by GMSK modulation. We used the GMSK Demodulator Baseband block shown in figure 4.14 demodulates a signal that was modulated using the Gaussian minimum shift keying method. The output of demodulator is shown in the figure below **[12]**.

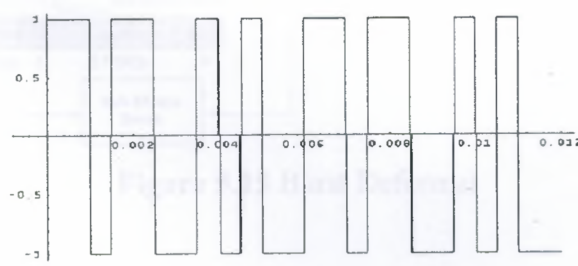


Figure 5.13 The Output of Demodulator

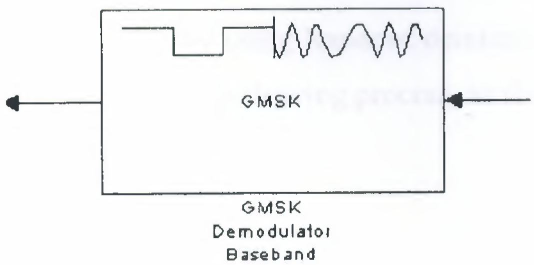


Figure 5.14 GMSK Demodulator

5.3.2 Burst Deformat

It is the reverse operation of Burst Format it is used to remove the additional bits which added in burst format (32 bits) for each frame. We used Sub-Matrix block to select only the main bits 114 bits (before addition operation) as shown in Figure 5.15 **[1]**.

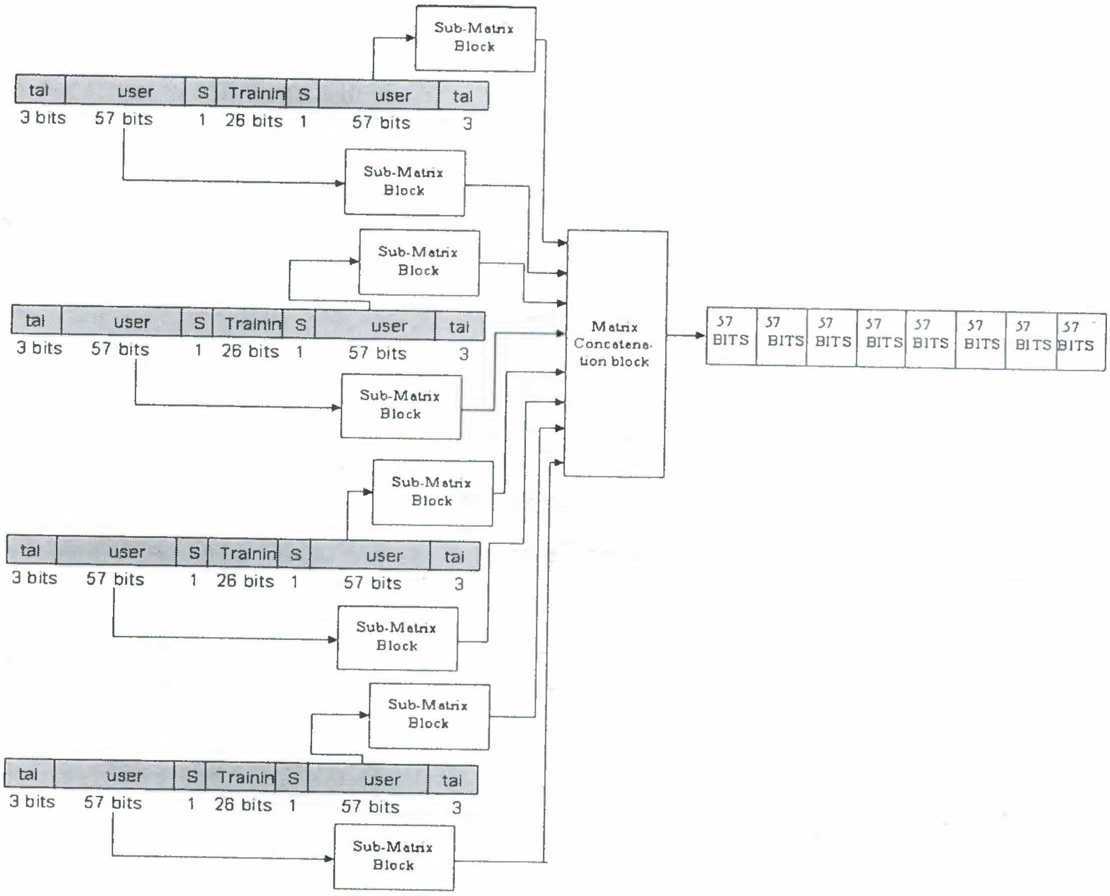


Figure 5.15 Burst Deformat

5.3.3 Deinterleaving

In this stage the order of the 456 bits returns to the original order before Interleaving. This process is done by using Random deinterleaver Block that returns the order of bits as it was before the interleaving process, as shown in Figure 5.16.

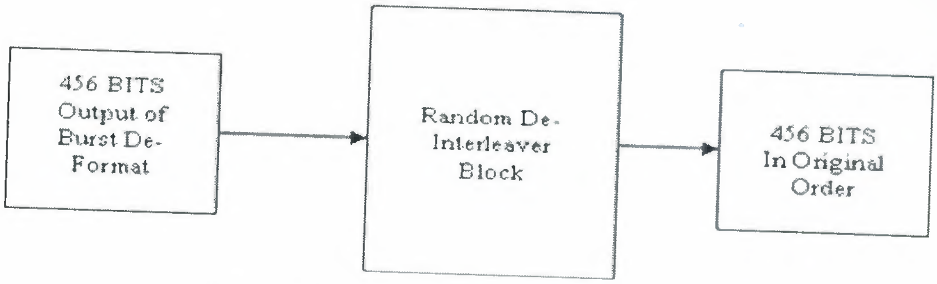


Figure 5.16 The Output of Deinterleaving

5.3.4 Channel Decoding

The Viterbi decoder block is used to remove the addition bits that added by the convolutional coder at the transmission process. The input to this block is 378 bits from 456 bits that get out from deinterleaver. Also, the Sub-Matrix Block is used to split the 456 bits into 378 and 78 bits (Not Important Bits) as shown in Figure 5.17. Figure 5.18 represent the output of this decoder which is 189 bits **[11]**.

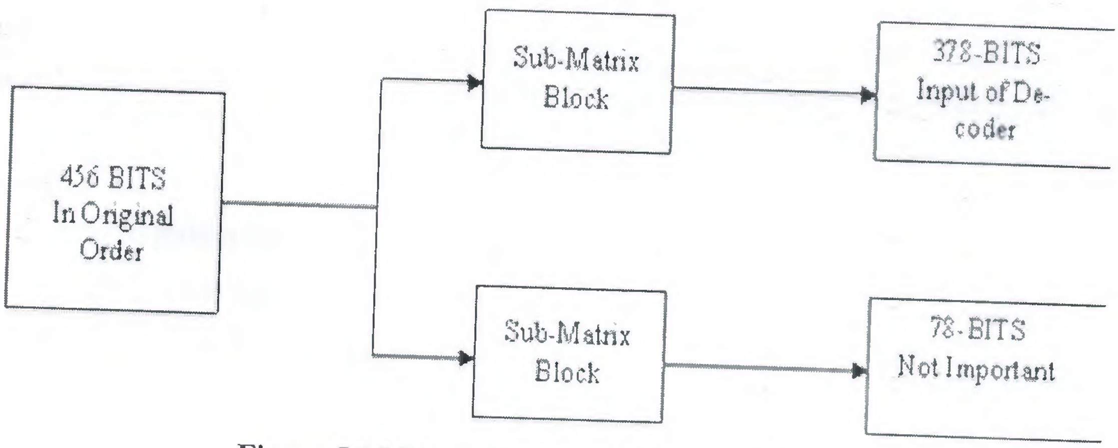


Figure 5.17 The Split Operation before Decoder

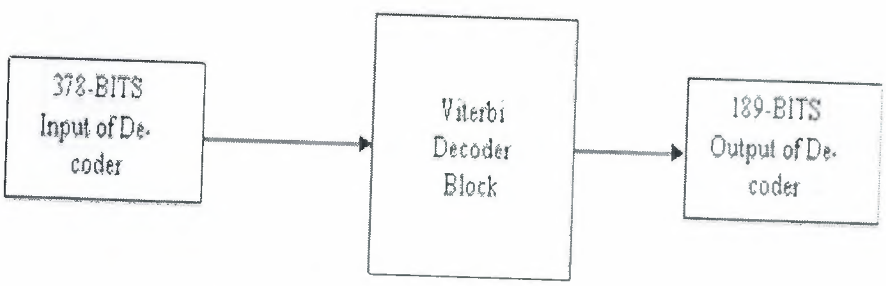


Figure 5.18 The Output of Decoder

5.3.5 CRC Error Detector

The general CRC syndrome error detector block is used to detect the errors that occurs for the very important 50-bits and corrects these errors. The Sub Matrix divide the 189-bits output of Decoder to 50 bits(Very Important Bits) and 132 bits(Important Bits) as shown in Figure 5.19. In the Figure 5.20 the 50 bits entered to the CRC detector to fix errors that occur to these bits **[12]**.

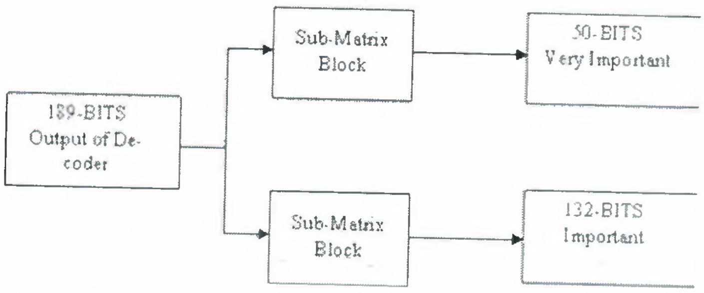


Figure 5.19 The Split Operation for Output of Decoder

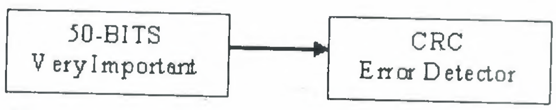


Figure 5.20 The Correction of the Very Important Bits

5.3.6 Aggregation Process

In Figure 5.21 the Matrix Concatenation Block used to aggregate the three blocks to produce 260 received bits.

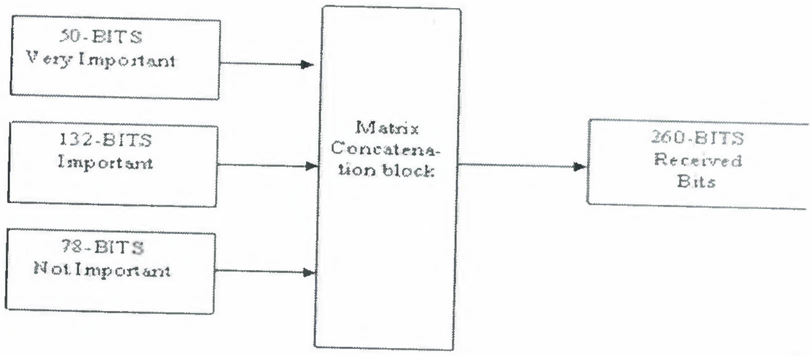


Figure 5.21 The Aggregation Operation

5.4 Summary

In this chapter we have determined all features of transmission and reception process in GSM in details, the transmission process has been explained in the previous chapter, the reception process starts when the received signal has been demodulated and then its passes through burst de-format, it is used to remove the additional bits which added in burst format (32 bits) for each frame, In this stage the order of the (456 bits) the outcome of the burst de-format which returns to the original order before Interleaving after that we use channel decoding to remove the redundant bits that being added by the convolution coder at the transmission process, then speech decoding process, at the end, the digital signal is converted into an analogue signal.

6. GSM SUBSCRIBER SERVICES

6.1 Introduction

GSM has the potential to offer a greatly enhanced range of services compared to existing analogue cellular systems. As well as a full range of data transmission options and fax, there will be a wide range of supplementary services.

The basic call services, which are already provided within analogue systems such as Call Forwarding, Voice Massage Services etc, are already available in some operational systems. Whether these services and others are provided as part of the basic service or at additional cost to the subscriber will depend on the network provider.

The services available to a subscriber will be determined by three factors:

- The level of service provided by the network provider.
- The level of service purchased by the subscriber.
- The capabilities of the subscriber's mobile equipment.

6.2 Personal Communications Service (PCS)

The future of telecommunications includes PCS. PCS at 1900 MHz (PCS 1900) is the North American implementation of digital cellular system (DCS) 1800 (GSM). Trial networks were operational in the United States by 1993, and in 1994 the Federal Communications Commission (FCC) began spectrum auctions. As of 1995, the FCC auctioned commercial licenses. In the PCS frequency spectrum, the operator's authorized frequency block contains a definite number of channels. The frequency plan assigns specific channels to specific cells, following a reuse pattern that restarts with each nth cell. The uplink and downlink bands are paired mirror images. As with AMPS, a channel number implies one uplink and one downlink frequency (e.g., Channel 512 = 1850.2-MHz uplink paired with 1930.2-MHz downlink) **[1]**.

6.3 GSM Services

From the beginning, the planners of GSM wanted ISDN compatibility in terms of the services offered and the control signaling used. However, radio transmission limitations, in terms of bandwidth and cost, do not allow the standard ISDN B-channel bit rate of 64 kbps to be practically achieved.

There are two basic types of services offered through GSM: telephony (also referred to as tele-services) and data (also referred to as bearer services). Telephony services are mainly voice services that provide subscribers with the complete capability (including necessary terminal equipment) to communicate with other subscribers. Data services provide the capacity necessary to transmit appropriate data signals between two access points creating an interface to the network. In addition to normal telephony and emergency calling **[11]**.

Using the ITU-T definitions, telecommunication services can be divided into bearer services, tele-services, and supplementary services. The most basic tele-service supported by GSM is telephony. As with all other communications, speech is digitally encoded and transmitted through the GSM network as a digital stream. There is also an emergency service, where the nearest emergency-service provider is notified by dialing three digits (similar to 911).

A variety of data services is offered. GSM users can send and receive data, at rates up to 9600 bps, to users on POTS (Plain Old Telephone Service), ISDN, Packet Switched Public Data Networks, and Circuit Switched Public Data Networks using a variety of access methods and protocols, such as X.25 or X.32. Since GSM is a digital network, a modem is not required between the user and GSM network, although an audio modem is required inside the GSM network to inter work with POTS.

Other data services include Group III facsimile, as described in ITU-T recommendation T.30, which is supported by use of an appropriate fax adaptor. A unique feature of GSM, not found in older analog systems, is the Short Message Service (SMS). SMS is a bi-directional service for short alphanumeric (up to 160 bytes) messages. Messages are transported in a store-and-forward fashion. For point-to-point SMS, a message can be sent to another subscriber to the service, and an acknowledgement of receipt is provided to the sender. SMS can also be used in a cell-broadcast mode, for sending messages such as traffic updates or news updates. Messages can also be stored in the SIM card for later retrieval.

Supplementary services are provided on top of tele-services or bearer services. In the current (Phase I) specifications, they include several forms of call forward (such as call forwarding when the mobile subscriber is unreachable by the network), and call barring of outgoing or incoming calls, for example when roaming in another country. Many additional supplementary services will be provided in the Phase II specifications, such as caller identification, call waiting, multi-party conversations **[1]**.

6.3.1 Speech Services

The following services listed involve the transmission of speech information and would make up the basic service offered by a network provider:

6.3.1.1 Telephony

Provides for normal MS originated/terminated voice calls.

6.3.1.2 Emergency Calls (with/without SIM Card Inserted in MS)

The number "112" has been agreed as the international emergency call number. This should place you in contact with the emergency services (Police, Fire, Ambulance) whichever country you are in.

6.3.1.3 Short Message Service Point to Point

A convenient facility of the GSM network is the short message service. A message consisting of a maximum of 160 alphanumeric characters can be sent to or from a mobile station. This service can be viewed as an advanced form of alphanumeric paging with a number of advantages. If the subscriber's mobile unit is powered off or has left the coverage area, the message is stored and offered back to the subscriber when the mobile is powered on or has reentered the coverage area of the network. This function ensures that the message will be received **[12]**.

6.3.1.4 Short Message Cell Broadcast

Provides the transmission of an unacknowledged short message (75 bytes maximum) from a service center in the fixed network to all MSs within one cell. This may carry information from the network provider, for example traffic information or advertising.

6.3.1.5 Advanced Message Handling Service

Provides message submission and delivery from the storage from a public Message Handling System (MHS) for example, electronic mail.

6.3.1.6 Duel Personal and Business Numbers

Permits the allocation of duel telephone numbers to a single subscriber. This will allow calls to be made and be billed either "business" or "personal" numbers.

6.3.1.7 Dual-Tone Multi frequency (DTMF)

DTMF is a tone-signaling scheme often used for various control purposes via the telephone network, such as remote control of an answering machine. GSM supports full-originating DTMF.

6.3.1.8 Voice Mail

This service is actually an answering machine within the network, which is controlled by the subscriber. Calls can be forwarded to the subscriber's voice-mail box and the subscriber checks for messages via a personal security code.

6.3.1.9 Fax Mail

With this service, the subscriber can receive fax messages at any fax machine. The messages are stored in a service center from which the subscriber via a personal security code to the desired fax number can retrieve them.

6.3.2 Data services

Data can be sent over the air using some of the present systems, but this requires specially designed to protect the data content in the harsh environment of the air interface.

Special provision is made in the GSM technical specifications for data transmission. Therefore, like ISDN, GSM is "specially designed" for data transmission. GSM can be considered as an extension of ISDN info the wireless environment **[11]**.

Text files, images, messages and fax may all be sent over the GSM network. The data rates available are 2.4 kbits/s and 9.6 kbits/s.

Below is a list of the various forms of data service that GSM will support.

6.3.2.1 Videotex Access

Provides access to computer-based information stored in databases, utilizing public transmission networks, where the requested information is generally in the form of text and/or pictures.

6.3.2.2 Teletex

Provides for data transfer in a circuit or packet-switched network (ITU-TSS X.200) (that is, document transmission).

6.3.2.3 Alternate Speech and Facsimile Group III

GSM supports CCITT Group III facsimile. As standard fax machines are designed to be connected to a telephone using analog signals, a special fax converter connected to the exchange is used in the GSM system. This enables a GSM–connected fax to communicate with any analog fax in the network.

6.3.3 Supplementary Services

A supplementary service is a modification of, or a supplement to, a basic telecommunication service. The network provider will probably charge extra for these services or use them as an incentive to join their network.

Here is a list of some of the optional supplementary subscriber services that could be offered to GSM subscriber.

6.3.3.1 Number Identification

- Receiving party requests calling number to be shown.
- Calling party requests calling number not to be shown.

6.3.3.2 Call Barring

- Bar all incoming or all outgoing calls, This function allows the subscriber to prevent incoming calls. The following two conditions for incoming call barring exist: baring of all incoming calls and barring of incoming calls when roaming outside the home PLMN.
- Bar specific incoming or outgoing calls, This service makes it possible for a mobile subscriber to prevent all outgoing calls.

6.3.3.3 Call Forwarding

- Forward all calls.
- Forward calls when subscriber is busy.
- Forward calls if subscriber does not answer.
- Forward calls if subscriber cannot be located.

6.3.3.4 Call Hold

- This service enables the subscriber to interrupt an ongoing call and then subsequently reestablish the call. The call holds service is only applicable to normal telephony.

6.3.3.5 Call Waiting

- This service enables the mobile subscriber to be notified of an incoming call during a conversation. The subscriber can answer, reject, or ignore the incoming call. Call waiting is applicable to all GSM telecommunications services using a circuit-switched connection.

6.3.3.6 Advice of Charge (AoC)

- Display current cost of call, The AoC service provides the mobile subscriber with an estimate of the call charges. There are two types of AoC information: one that provides the subscriber with an estimate of the bill and one that can be used for immediate charging purposes. AoC for data calls is provided on the basis of time measurements.

6.3.3.7 Multi-Party

- Three party service.
- Conference calling.

6.3.3.8 Closed User Groups (CUGs)

- CUGs are generally comparable to a PBX. They are a group of subscribers who are capable of only calling themselves and certain numbers.

6.4 Summary

In this chapter, we have seen the services being provided by the GSM, main three services are speech service, which includes Telephony, Emergency Calls, Short Message Service Point to Point and Cell Broadcast and others, the second type is data service such as Videotex Access, Teletex, Alternate Speech and Facsimile Group III and the third type is Supplementary Services which includes the following, Number Identification, Call Barring, Call Forwarding, Call Hold, Call Waiting, Multi-Party, Advice of Charge, and Closed User Groups.

7. CONCLUSION

The development of GSM is the first step towards a true personal communication system that will allow communication anywhere, anytime, and with anyone. The functional architecture of GSM, employing intelligent networking principles, and its ideology, which provides enough standardization to ensure compatibility, but still allows manufacturers and operators freedom, has been widely adopted in the development of future wireless systems.

Knowing every step in the whole hierarchy from the beginning till the end of the transmission and reception process mentioned, we figured that sounds can be transmitted through transmission media; it could be physical channel or non-physical channel which is free space, a specific transactions are applied for input data in order to implement a certain signal that is suitable for transmission and can be reconstructed in the receiver side with high quality more efficient and more faster and that process makes our life much easier and more comfortable.

8. REFERENCES

[1]. Mamedov F. S., *Telecommunications*, Lecture Notes, Near East University Press, Nicosia, 2000.

[2]. Chia, "Beyond the Second-Generation Mobile Radio Systems, "British Telecom Engineering 10 (January 1992): 326-335.

[5]. Ericsson, "GSM System Survey", 1999.

[6]. T. S. Rappaport, "Wireless Communications: Principle and Practice", Second Edition, Prentice Hall.

[7]. William C. H. Mok, "Background of Communication System", 1998

[8]. B. Sklar, "Digital Communications: Fundamental and Applications", Second Edition, Prentice Hall.

[9]. Etienne Sicard, Sonia D. Bendhia, "Introduction to GSM", INSA/DGEI, Online Symposium for Electronics Engineers, 2003]

[10] K. Pahlavan and A. H. Levesque, "Wireless Information Networks", John Wiley & Sons, Inc., 1995

[11]. http://www.wikipedia.com

[12]. http://www.howstuffworks.com

[13]. http://www.ham-shak.com