Communication Systems

Telephony System

Lecture (1-1) :- Introduction

Historical review for electronic communications

What is TELEPHONE?

Invention of the telephone and History of the telephone

Natural of sound

Human Voice and Auditory systems Properties.

Assignment of Frequency Band for Telephony System

Historical review for electronic communications.

1729 - Stephen Grey discovered that electricity could be transmitted over large distances using wires as long as they were insulated.

1800 - The electric battery was invented by Volta, but at this time there was still no practical use for it

1834- Charles Wheatstone experimented with the rate of transmission of electricity along 1/2 mile of insulated (suspended) copper wire.
Patent issued for the "Electric Printing Telegraph", a very early forerunner of the fax machine

1837- Charles Wheatstone and William Cooke acquire a patent for a 5 needle telegraph, Samuel Morse also showcased his telegraph

1838 - Samuel Morse demonstrated a new and improved telegraph which used Morse Code.

1844 - Charles Wheatstone did submarine telegraph experiments in Swansea bay

1850 - First successful submarine telegraph cable laid between England and France by John Watkins Brett's

1851- A total of 51 telegraph companies are now in operation

1852 - A Manchester cotton mill merchant, John Pender, became directory of the English and Irish Magnetic Telegraph Company.

1854 - It is proven by John Tyndall that light can be sent through a curved stream of water and therefore bent.

1860 - The Morse Inker was invented to allow messages sent in Morse Code to be recorded

1866 - The first successful transatlantic cable was laid using a special ship designed by Brunel.

1870 - Brunel's ship also laid the Bombay Porthcurno line

1872 - Pender merged all of his many telegraph companies to create The Eastern Telegraph Company; it was the first global company of its kind.

1876 - Alexander Graham Bell acquires a patent for the electric telephone

1889 - Almon Strowger patents the direct dial.

1900 - The Eastern Telegraph Company has approximately 150,000km[i] of submarine cables in its network. Porthcurno telegraph station now is connected to India, South Africa, Australia and the Americas[ii].

1901 - Marconi is the first to successfully wirelessly communicate between Cornwall and Newfoundland.

1924 - Marconi is the first to telephone Australia using radio waves; he later gets contracts from the Post Office to setup wireless telegraphy circuits to other countries as well as Australia

1925 - A new system is setup to automatically relay Telegraph messages.

1926 – Commercial availability of the radiofax

1929 - The Eastern Telegraph Company is merged with the Wireless Telegraph Company to form The Imperial and International Communications Company as a result of a government order.

1934 - Imperial and International Communications was renamed Cable and Wireless Limited.
First modern fax machine commercially available (Long Distance Xerography)

1935 - The world's first around the world telephone call using radio and cable happens.

1944 - The first police FM radio communication is setup in Hartford.

1944 - A telephone submarine cable is laid across the English Channel.

1947 - Douglas H. Ring and W. Rae Young of Bell Labs propose a cell-based approach which led to "cellular phones."

1950 - The telegraph becomes obsolete as the telephone takes over.Cable & Wireless is nationalised and becomes part of the Post Office.

1956 - The first transatlantic telephone cable is laid making it easier for the UK to call overseas.

1957 - The USSR launches the world's first artificial satellite Sputnik 1, it transmitted low frequency radio frequencies.

1960 - The Ruby Laser is invented which is considered to be the first successful optical light laser.

1962 - NASA launches its first artificial satellite Telstar. First non-public then later public television images are transmitted via Telstar between Europe and the USA.

1964 - The "Post Office Tower" is built in London to handle ever increasing telephone communications.

1966 - Cable & Wireless built their first earth satellite to assist with the Apollo moon landings.

1972 - Bob Kahn and Vint Cerf invented TCP

1975 - Robert Metcalfe invents Ethernet to wire local computers together.

1975 - The development of their first commercial fiber optic communication system was finished.

1981 - Nordic Mobile Telephone, the world's first automatic mobile phone is put into operation

1991 - TAT-8 was setup and was the first transatlantic optical fibre cable.

1991 - Tim Burners-Lee invents the World Wide Web.
GSM is put into operation
1992 - Neil Papworth sends the first SMS (or text message).
1994 - The WWWC (World Wide Web Consortium) is setup to help set web standards.

What is TELEPHONE

The term (**Telephone**) is derived from Greek: (tele, far) and (phone, voice), together meaning distant voice. A common short form of the term is phone, which came into use almost immediately after the first patent was issued.

A **telephone** is a telecommunications device that permits two or more users to conduct a conversation when they are too far apart to be heard directly. A telephone converts sound, typically and most efficiently the human voice, into electronic signals that are transmitted via cables and other communication channels to another telephone which reproduces the sound to the receiving user.

The electric telephone was invented in the 1870s, based on earlier work with harmonic (multi-signal) telegraphs.

Invention of the telephone and History of the telephone

Alexander Graham Bell held the master patent (on March 7, 1876) for the telephone.

A controversies over the issue have arisen. As with other great inventions such as radio, television, the light bulb, and the digital computer, there were several inventors who did pioneering experimental work on voice transmission over a wire, who then improved on each other's ideas.

However, the key innovators were ***Alexander Graham Bell*** and ***Gardiner Greene Hubbard***, who created the first telephone company, the Bell Telephone Company in the United States, which later evolved into

American Telephone & Telegraph (AT&T), at times the world's largest phone company.

The first commercial telephone services were set up in 1878 and 1879 on both sides of the Atlantic in the cities of New Haven, Connecticut in the US and London, England in the UK.

Telephone technology grew quickly after the first commercial services emerged, with inter-city lines being built and telephone exchanges in every major city of the United States by the mid-1880s.

The first transcontinental telephone call occurred on January 25, 1915. Despite this, transatlantic voice communication remained impossible for customers until January 7, 1927 when a connection was established using radio. However no cable connection existed until TAT-1 was inaugurated on September 25, 1956 providing 36 telephone circuits.

In 1880, ***Bell*** and co-inventor ***Charles Sumner Tainter*** conducted the world's first wireless telephone call via modulated light beams projected by photophones. The scientific principles of their invention would not be utilized for several decades, when they were first deployed in military and fiber-optic communications.

The first transatlantic telephone cable (which incorporated hundreds of electronic amplifiers) was not operational until 1956, only six years before the first commercial telecommunications satellite (1962).

Telegraph System

Bell's first telephone transmitter, ca. 1876, reenacted 50 years later

Bell placing the first New York to Chicago telephone call in 1892

Natural of sound :

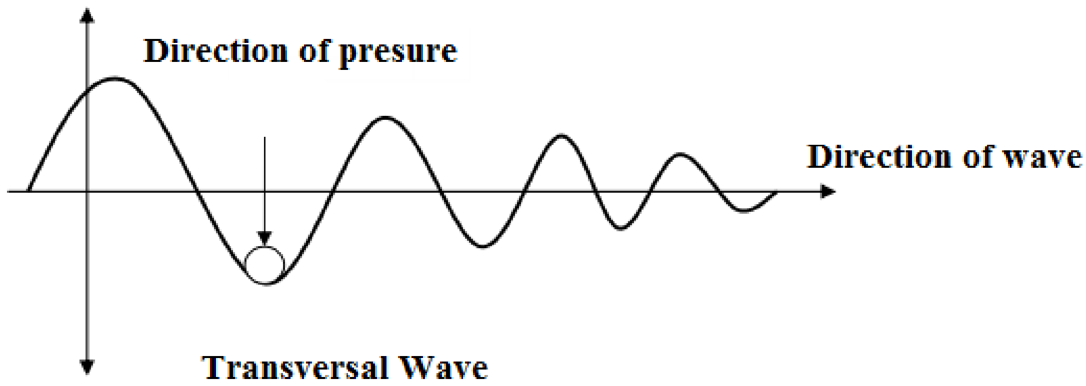
sound is a vibration that propagates as an acoustic wave, through a transmission medium such as a gas, liquid or solid. It is measured in terms of frequency and amplitude.

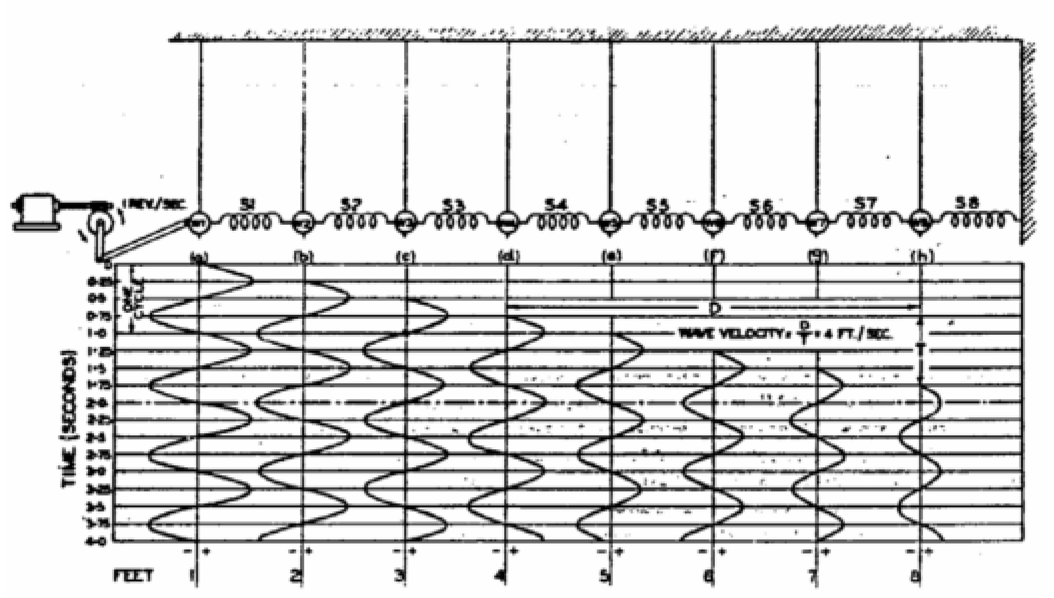
Longitudinal and transverse waves

Sound is transmitted through gases, plasma, and liquids as **longitudinal waves**, also called **compression** waves. It requires a medium to propagate. Through solids, however, it can be transmitted as both **longitudinal** waves and **transverse waves**.

Longitudinal sound waves are waves of alternating pressure deviations from the equilibrium pressure, causing local region of compression and refraction.

While transverse waves (in solids) are waves of alternating shear stress at right angle to the direction of propagation.The energy carried by an oscillating sound wave converts back and forth between the potential energy of the extra compression (in case of longitudinal waves) or lateral displacement strain (in case of transverse waves) of the matter, and the kinetic energy of the displacement velocity of particles of the medium.





Mechanical representation for Longitudinal wave

Sound waves are often simplified to a description in terms of sinusoidal plane waves, which are characterized by these generic properties:

- Frequency, or its inverse, wavelength
- Amplitude, sound pressure or Intensity
- Speed of sound
- Direction

Sound that is perceptible by humans has frequencies from about 20 Hz to 20,000 Hz. In air at standard temperature and pressure, the corresponding wavelengths of sound waves range from 17 m (56 ft) to 17 mm (0.67 in).

Sometimes speed and direction are combined as a velocity vector; wave number and direction are combined as a wave vector.

Frequency

Humans with normal hearing can hear sounds between 20 Hz and 20,000 Hz. Frequencies above 20,000 Hz are known as **ultrasound**. When your dog tilts his head to listen to seemingly imaginary sounds, he is tuning in to ultrasonic frequencies, as high as 45,000 Hz. Bats can hear at among the highest frequencies of any mammal, up to 120,000 Hz. They use ultrasonic vocalizations as sonar, allowing them to pursue tiny insects in the dark without bumping into objects.

At the other end of the spectrum are very low-frequency sounds (below 20 Hz), known as **infrasound**. Elephants use infrasound for communication, making sounds too low for humans to hear. Because low frequency sounds travel farther than high frequency ones, infrasound is ideal for communicating over long distances.

Amplitude

The relative strength of sound waves (Amplitude), is measured in Phone unit and decibels (dB), which refer to the sound pressure level or intensity. The lower threshold of human hearing is 0 dB at 1kHz. Moderate levels of sound (a normal speaking voice, for example) are under 60 dB. Relatively loud sounds, like that of a vacuum cleaner, measure around 70 dB. When workplace sound levels reach or exceed 85 dB, employers must provide hearing protection. A rock concert, at around 125 dB, is pushing the human pain threshold.

$$IL = 10\log_{10}\frac{I}{I_o}$$

IL is a sound intensity level in dB
I_o is a reference intensity power = $10^{-12} w/m^2$
I is a sound intensity power

Speed of sound

The **speed of sound** is the distance travelled per unit of time by a sound wave as it propagates through an elastic medium. At 20 °C (68 °F), the speed of sound in air is about 343 metres per second. It depends strongly on temperature as well as the medium through which a sound wave is propagating. The speed of sound in an ideal gas depends only on its temperature and composition.

Human Voice and Auditory systems Properties.

Human voice system.

The human voice consists of sound made by a human being using the vocal tract, such as talking, singing, laughing, crying, shouting, etc. This system consist of:-

1- Larynx which contain the Vocal cords
2- Oro pharynx and Naso Pharynx, and Laryngeo Pharynx
3- Nasal Cavity and Mouth Cavity, which specify the sound properties.
4- Tongue, Tooth, Lips, Lower and Upper jaw
5- Trachea.

The human voice frequency is specifically a part of human sound production in which the vocal folds (vocal cords) are the primary sound source.

the mechanism for generating the human voice can be subdivided into three parts; the lungs, the vocal folds within the larynx (voice box), and the articulators.

The lungs, the "pump" must produce adequate airflow and air pressure to vibrate vocal folds. The vocal folds (vocal cords) then vibrate to use airflow from the lungs to create audible pulses that form the laryngeal sound source.

The muscles of the larynx adjust the length and tension of the vocal folds to ‘fine-tune’ pitch and tone.

The articulators (the parts of the vocal tract above the larynx consisting of tongue, palate, cheek, lips, etc.) articulate and filter the sound emanating from the larynx and to some degree can interact with the laryngeal airflow to strengthen or weaken it as a sound source.

The vocal folds, in combination with the articulators, are capable of producing highly intricate arrays of sound. The tone of voice may be modulated to suggest emotions such as anger, surprise, fear, happiness or sadness. The human voice is used to express emotion, and can also reveal the age and sex of the speaker.

Articulation Curves:-

The articulation curves are one of the curves that specify the characteristics of the human voice system figure (5) below shows the relationship between the percentages of articulation (Understandability) verse frequency.

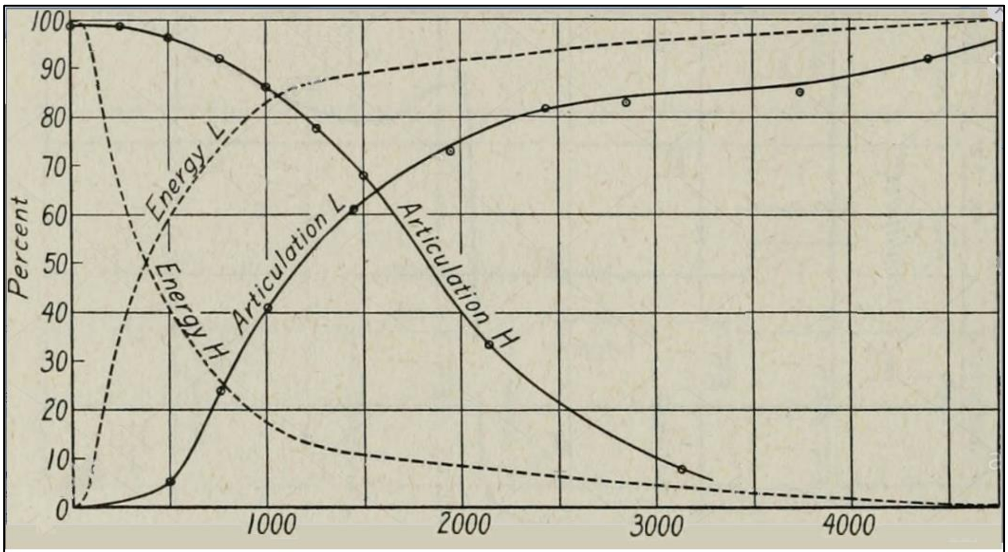


Figure (5) Articulation Curves

One can note that there are two type of articulation curves, the first one is the articulation curve the upper frequencies and other curve for the lower frequencies. In the curve of the lower frequencies, as the frequency increases, the percentage of articulation decreases (the voice of a man), i.e. the owner of a throat that generates low frequencies, and this percentage increases as the frequency decreases. However in the upper frequencies curve, the percentage of articulation increases with increasing frequency and decreases with decreasing frequency (voice of a women). It is deduced from the curves that the ratio of speech clarity varies between owners of higher frequencies and those with lower frequencies in the range between 500 and 1500 hertz.

Human Auditory System.

The auditory system is the sensory system for the sense of hearing. It includes both the sensory organs (the ears) and the auditory parts of the sensory system. This system can be divided into three main parts

1 - The outer ear:- consist of

* - Pinna
* - Audiotry canal
* - Eardrum

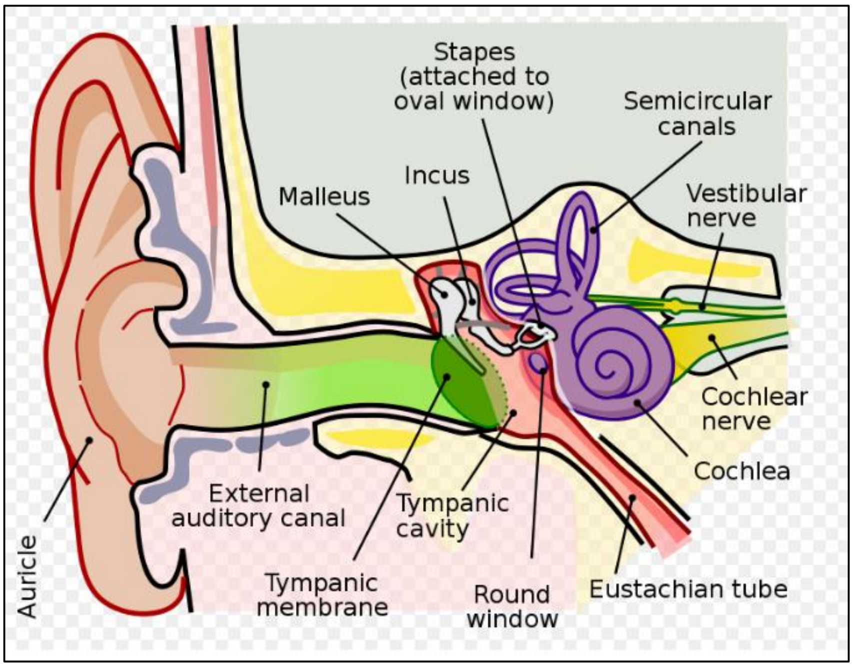
2 - The midle ear:- consist of

* - Hammer
* - Anvil
* - Stirrup

These three bones are connected to each other by supporting muscles and ligaments, while the hammer is held behind the eardrum and the stirrup bone on the oval window of the inner ear. The middle ear vacuum is connected to the throat through Eustachain tube, which is usually closed and only opens during swallowing and yawning in order to equalize the pressure and confirm the balance on both sides of the eardrum.

3 – The inner ear

The inner ear is a bony structure filled with oil (lymph tissue) that has two small openings that lead to the middle ear, which are the oval window and the Round window, and they are closed by two flexible membranes that prevent oil from escaping from the inner ear. The vibrations of the eardrum are transmitted through the three bones in the middle ear to the oval window, and corresponding vibrations arise in the inner ear. The cochlea is the part that is responsible for receiving the sound and converting it into electrical signals sent through the auditory nerve to the brain.



Audibility threshold curves.

The term Audibility threshold is defined as the minimum and perceptible level of field strength for any tone that can be perceived or heard at every frequency that the human ear is sensitive to.

Figure (7) show that the average audibility of the normal ear appears through the lower curve, where the maximum sensitivity of the ear appears near the frequency of 4 kilohertz, and we notice that at frequencies less than 4 kHz, the audibility increases with a decrease in the frequency, and the lowest hearing ability is at about the frequency 30 Hertz and are close to a million times the power at 4 kHz. While the ability to hear at high frequencies increases rapidly until it reaches the cutoff point.

The audibility varies among peoples, especially if the age difference between them is large, as the cutting frequency for a young person reaches 20 kilohertz or perhaps more, while the greater cutting frequency for an older person (up to 40 or 50 year) is less than 10 kHz. Note that the low frequency of the audibility does not depend on the age of the person.

The higher the amplitude of the sound (the intensity of sound signal), the better audibility in the two smaller and larger frequency ranges, but when the intensity of the sound signal is increased to the limits of 120 Phones, the sensation of numbness results, and if the capacity increases to the limits of 140 Phones, the feeling of (tickling sensation) into a form of pain and may lead to permanent damage the ability to hear.

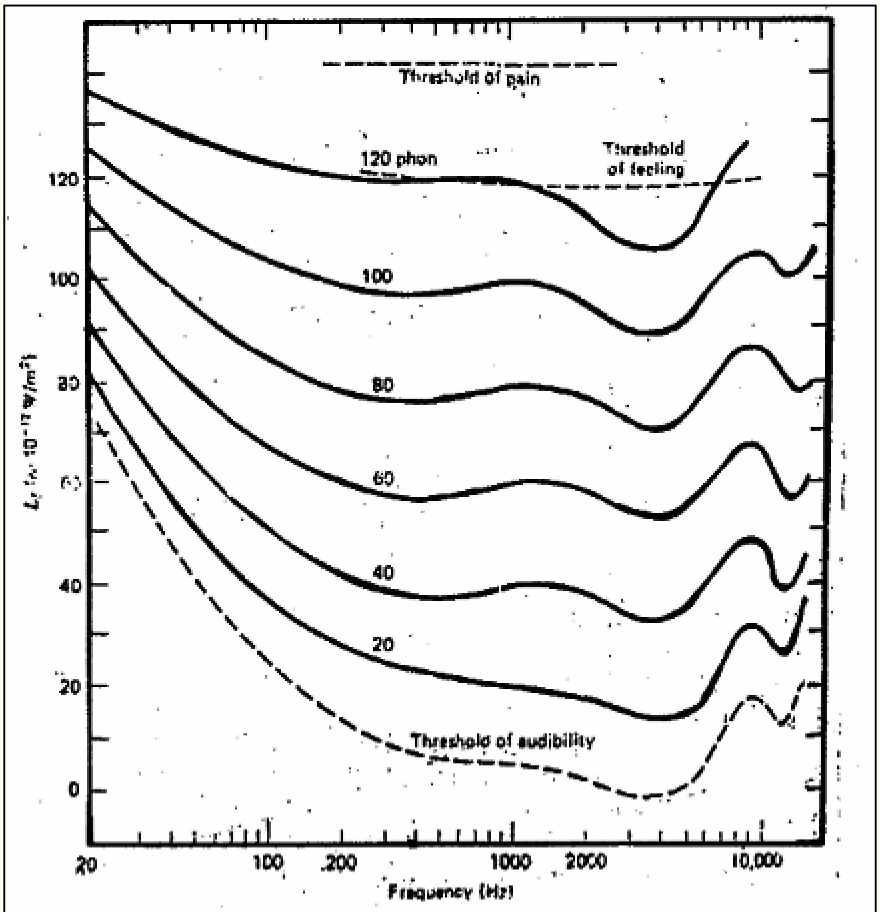


Figure (7) Audibility threshold Curves

Assignment of Frequency Band for Telephony System:-

From the previous study of the human voice and auditory system properties we can conclude:-

1 – The human articulation band can be limited between (500 to 2500 Hz)

2 – The maximum sensitivity of the human audibility lies between (300 to 4000 Hz)

3 – In the beginning the the frequency band (500 to 2500 Hz) was used for to product an economical telephony system.

4 – Earlier in modern economical telephony system use the frequency range (300 to 3400 Hz)

This frequency range (300 to 3400 Hz) offer the following properties:-

1 – Maximum sensitivity of audibility.

2 – High articulation for each the upper and lower frequencies sources.

3 – Low cost and available techniques are available in this range.

Communication Systems

Telephony System

Lecture (2-1) :- Telephony system Structure

- **Telephony system structures**
- **Subscriber set**
  Carbon microphone،
  Electromagnetic Loudspeaker
  Ringing unit.
  Dialing unit (DTMF and pulse tone).

Telephony System structures

Telephony system consist of

1 – Telephone set

2 – Distribution Box Unit (DBU)

3 – Telephony network

4 – Cabinet

5 – Mane distribution Frame (MDF)

6 – Exchangers

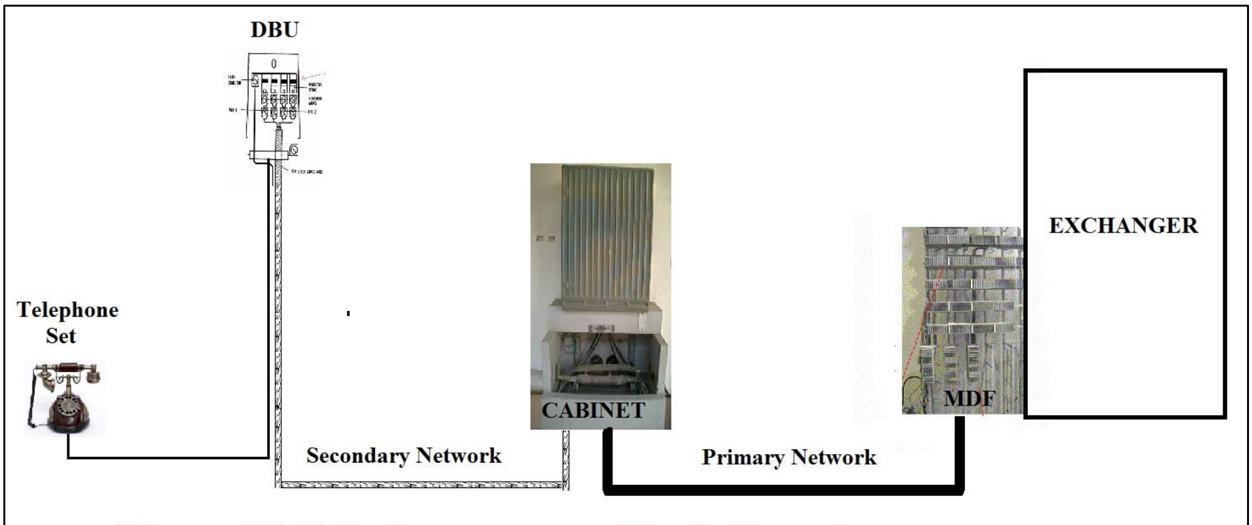


Figure (1) Telephony system block digram

Telephone Set (Subscriber Set):-

The first part from the telephony system is the telephone set, this achieve the following jobs:-

1 – Send the request for call to the local exchanger.

2 – Send the dial number.

3 – Receive the various tones from the local exchanger.

4 – Receive the ring signal.

5 - Send and receive the speech signal.

6 – End the call.

The telephone set consist of:-

1 – The hand set which consist of

A – The carbon microphone

B – The loudspeaker

2 – Dial unit.

3 – Cradle which contain

A- The matched transformer

B- Transmission regulation circuit(balance circuit)

4 – Bell circuit

Figure (2 and 3) show the telephone set part and telephone set schematic circuit.

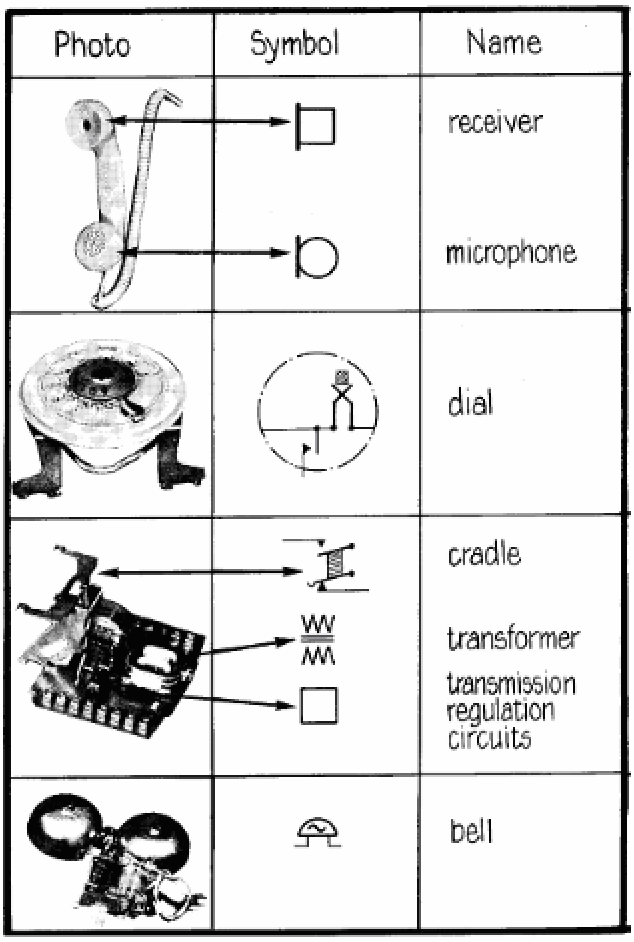
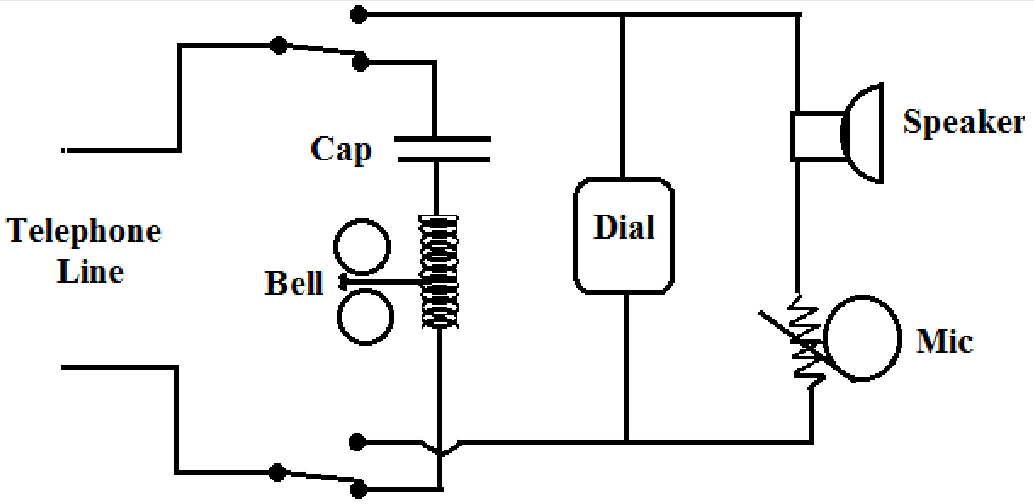


Figure (2)



Circuit Diagram for telephon set

- - The Telephone Handset

  The telephone handset is (usually) handle with the subscriber hand during the call. Its contain from the carbine microphone and loudspeaker, also there is a simple echo circuit that make feedback from the microphone to the speaker to insure the subscriber that the microphone work quite.

- **<u>- The carbon microphone:-</u>**

  The carbon microphone is a transducer used to convert the mechanical signal of sound to electrical signal that can be treated in the next circuit of the telephone set, figure (4) below show the construction of the carbon microphone that early used in the telephone set

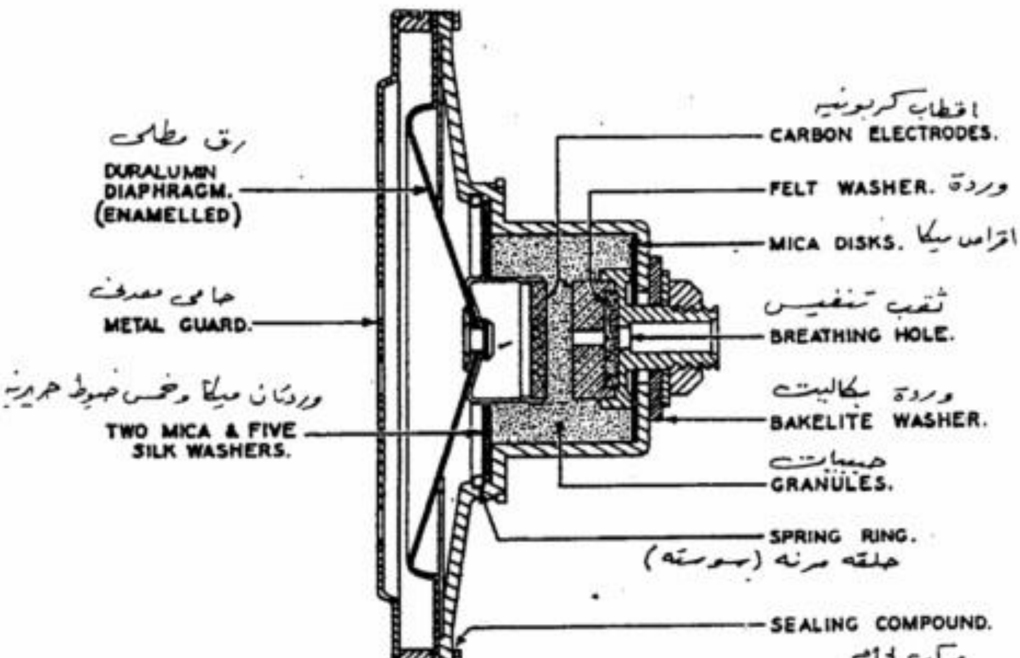


Figure (4) Carbon microphon structure

- **<u>The Loudspeaker :-</u>**

The loudspeaker is a transducer that convert the electrical signal into the sound signal due to the mechanical vibration waves that occur on last part of the loudspeaker (diaphragm). Figure (5) show the construction of the electromagnetic loudspeaker.

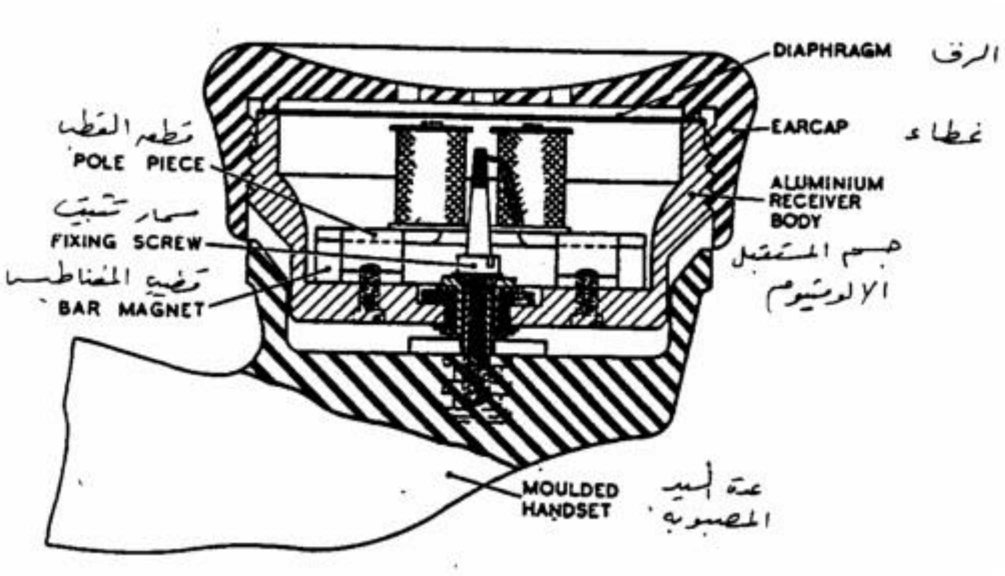


Figure (5) Elecromagnetic Loudspeaker

Communication Systems

Telephony System

Lecture (3-1):- Telephony system Structure

- **Subscriber set**
  - **Dialing unit (DTMF and pulse tone).**
  - **Bell circuit.**
  - **Distribution Box Unit (DBU)**
  - **Secondary network**
  - **Cabinet**
  - **Primary network**
  - **Main Distribution Frame (MDF)**

<u>Dialing unit</u>

The main job of the dialing unit in telephony system is to send the called subscriber number as an electronicl signal from telephone set to the local exchanger in order to achieve the link with the called subscriber. There are two types of the dialing unit that used in In telephone set, the pulse tone dialing unit and dialing dual tone multiple frequency :-

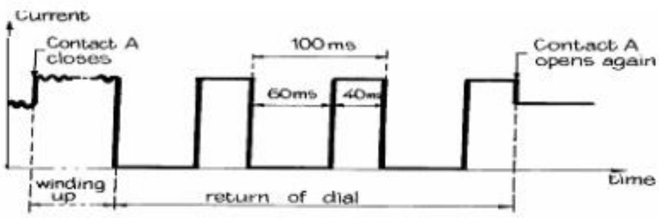
A rotary dial is a component of a telephone or a telephone switchboard that implements a signaling technology in telecommunications known as pulse dialing. It is used when initiating a telephone call to transmit the destination telephone number to a telephone exchange. These two types of dialing model introduced by the rotary dial unit and keypad dial system.

Rotary dial unit.

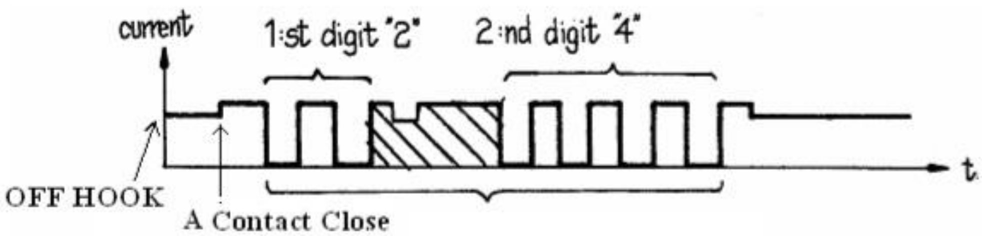
Rotary Dial (Pulse Tone):-

On the rotary phone dial, the digits are arranged in a circular layout so that a finger wheel may be rotated against spring tension with one finger. Starting from the position of each digit and rotating to the fixed finger stop position, the angle through which the dial is rotated corresponds to the desired digit. Compact telephones with the dial in the handset had all holes equally spaced in the dial, and a spring-loaded finger stop with limited travel.

When released at the finger stop, the wheel returns to its home position driven by the spring at a speed regulated by a centrifugal governor device. During this return rotation, the dial interrupts the direct electrical current of the telephone line (local loop) the specific number of times associated with each digit and thereby generates electrical pulses which the telephone exchange decodes into each dialed digit. Each of the ten digits is encoded in sequences to correspond to the number of pulses.



Pulse Tone whwn Send No. 3 in Rotary Dial



Pulse Train for send No. 24 using Rotary Dial System

Keypad Dial (DTMF):-

The telephone keypad is laid out as a matrix of push buttons in which each row represents the low frequency component and each column represents the high frequency component of the DTMF signal. The commonly used keypad has four rows and three columns, but a fourth column is present for some applications. Pressing a key sends a combination of the row and column frequencies. For example, the 1 key produces a superimposition of a 697 Hz low tone and a 1209 Hz high tone. Initial pushbutton designs employed levers, enabling each button to activate one row and one column contact. The tones are decoded by the switching center to determine the keys pressed by the user.

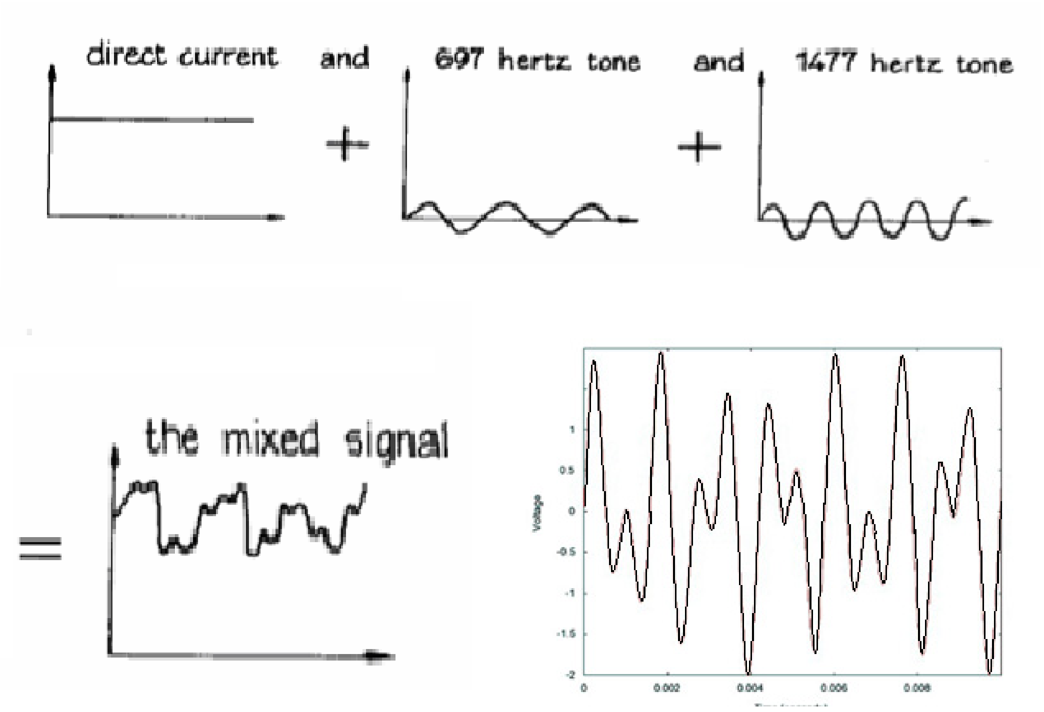
The DTMF system uses a set of eight audio frequencies transmitted in pairs to represent 16 signals, represented by the ten digits, the letters A to D, and the symbols # and *. As the signals are audible tones in the voice frequency range, they can be transmitted through electrical repeaters and amplifiers, and over radio and microwave links, thus eliminating the need for intermediate operators on long-distance circuits.

DTMF was originally decoded by tuned filter banks. By the end of the 20th century, digital signal processing became the predominant technology for decoding. DTMF decoding algorithms typically use the Goertzel algorithm. As DTMF signaling is often transmitted in-band with voice or other audio signals present simultaneously, the DTMF signal definition includes strict limits for timing (minimum duration and interdigit spacing), frequency deviations, harmonics, and amplitude relation of the two components with respect to each other.

DTMF keypad layout.

DTMF keypad frequencies (with sound clips)

	1209 Hz	1336 Hz	1477 Hz	1633 Hz
697 Hz	1	2	3	A
770 Hz	4	5	6	B
852 Hz	7	8	9	C
941 Hz	*	0	#	D

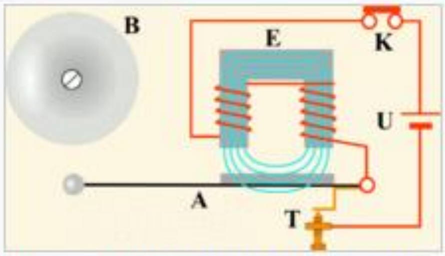


Bell Circuit

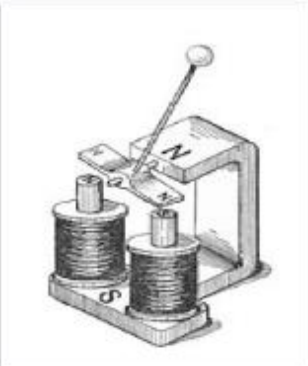
An electric bell is a mechanical or electronic bell that functions by means of an electromagnet. When an electric current is applied, it produces a repetitive buzzing, clanging or ringing sound. But they are now being widely replaced with electronic sounders. An electric bell consists of one or more electromagnets, made of a coil of insulated wire around an iron bar, which attract an iron strip armature with a clapper. When an electric current flows through the coils, the electromagnet creates a magnetic field which pulls the armature towards it, causing the hammer to strike the bell.

Telephone bells were powered by 60 to 105-volt RMS 20-Hertz AC. and a different design, the polarized bell, was used. These have an armature containing a permanent magnet, so that this is alternately attracted and repelled by each half-phase and different polarity of the supply. In practice, the armature is arranged symmetrically with two poles of opposite polarity facing each end of the coil, so that each may be attracted in turn. No contact breaker is required, so such bells are reliable for long service.

Older electric doorbell



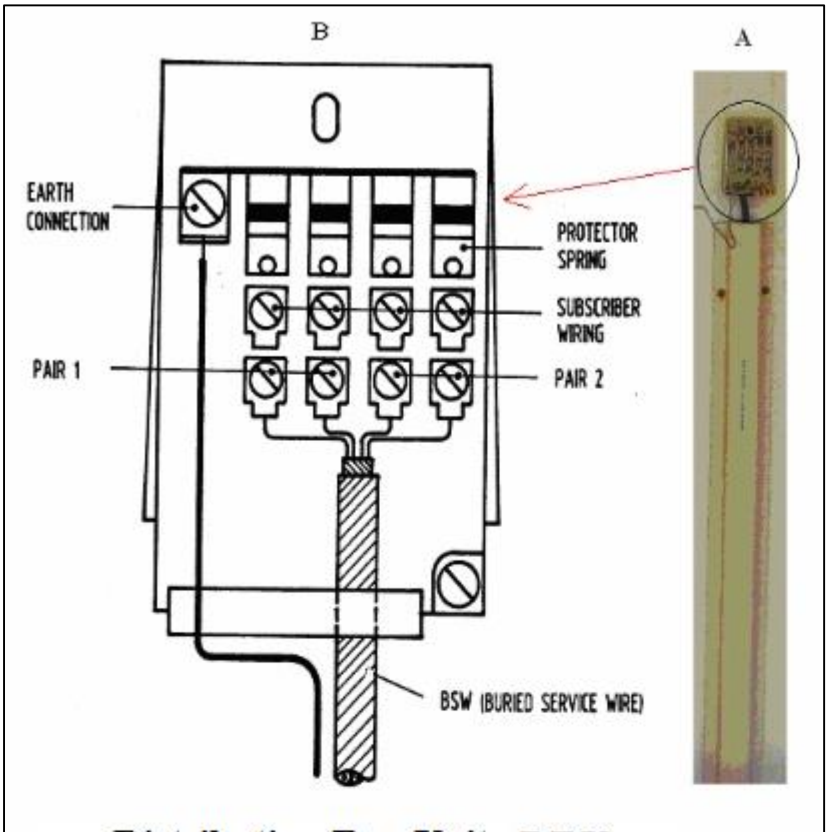
How an interrupter-type electric bell works.



Polarised bell, circa 1903

Distribution Box Unit (DBU)

DBU is a unit for distributing telephone lines to subscribers, as it is installed on one of the buildings or on a pole that specify for the telephone network. A number of DBU are connected with a single cabinet via the secondary network. There are varies capacity of DBUs, some have 10 lines and others have 20 or 30 lines. The DBU contains a Joint Box Unit consisting of two groups of connection point pairs connected each point of one of the groups with the corresponding point from the second group from the bottom. The connection points for one of the groups are connected from the top with the incoming lines in the cabinet, while the other connection group points are connected with the lines going to the phones of the subscribers. There are backup points of more than 20 percent of the capacity of the DBU.



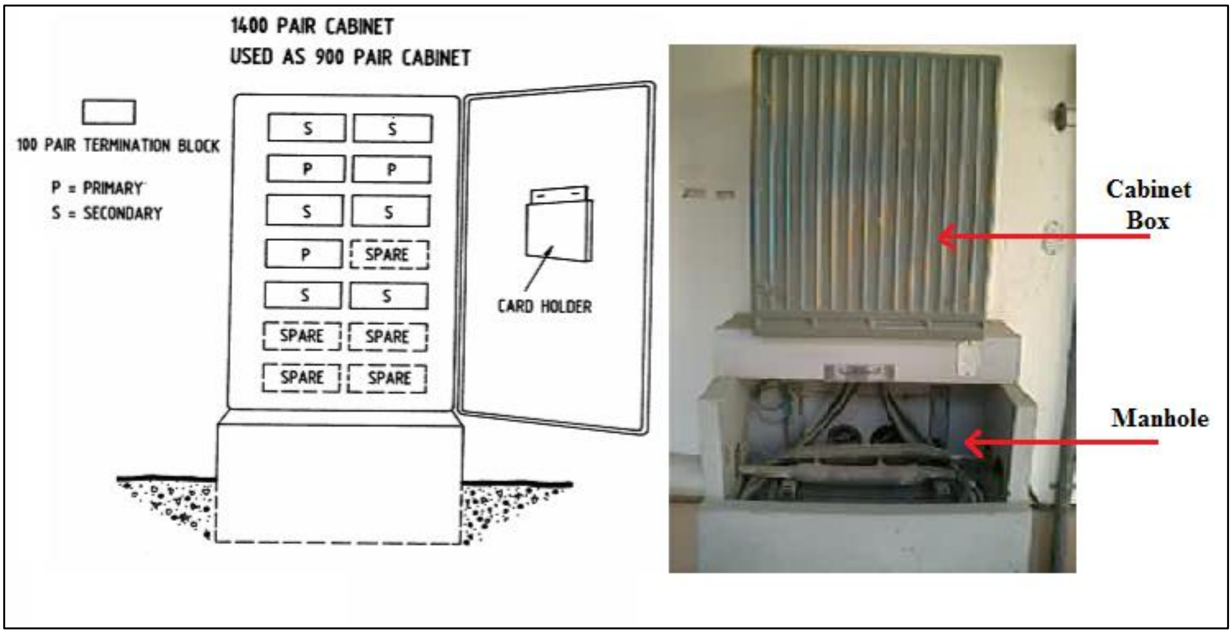
Distribution Box Unit DBU

<u>Secondary Network:-</u>

The secondary network consist of the cables that joint the DBU with the cabinet, these cables have different capacity (number of pairs) and they expanded in underground toward the cabinet, this cables have the capacities 20, 50, 100, 200 and 300 telephone lines .

Cabinet:-

The cabinet is the intermediate part that connect the DBUs with the Maine Distribution Frame (MDF) in the exchanger office. Its lie in the main roads from the city. This consist of a material box with distribution board that have two types of the connectors groups, secondary connectors groups and primary connectors groups. Also there is a spear connectors groups. The primary and the spear connectors groups connected to the MDF with the primary network cables, while the secondary connectors groups connected to the DBUs through the secondary network cables. The connection between the primary and secondary connectors can be done by using short jumper wires according to specified diagram with special tools. Under the cabinet box there is a manhole in which the cables for the primary and secondary network pass through it in the cables ducts. The cabinets may have different capacities, 800, 1200, etc.



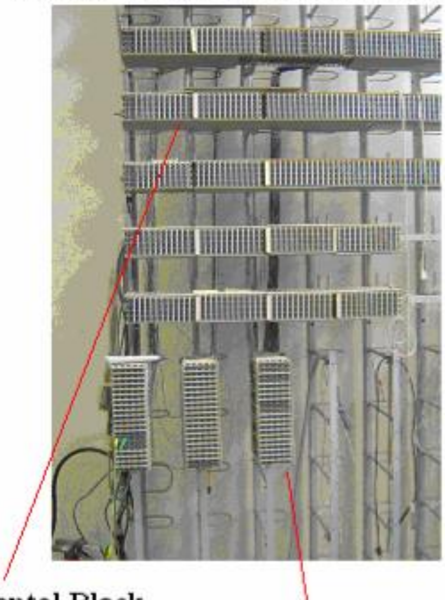
Primary Network:-

This network consist of the cables that connecting the cabinet with MDF, this cables have different capacities like 200, 300, 400, 600, 900, 1000, 1200, 1500, 1800 and 2400 telephone lines.

Main Distribution Frame (MDF):-

The MDF is a big board installed in the Exchanger building office, this contain two type of connectors units, the vertical units and the horizontal units. The horizontal units connected interiorly with the electronic exchanger devices, while the vertical connectors units connected to the subscriber lines that came from the cabinet throw the primary cables. Technicians can connects the vertical and the horizontal connecters with short jumper wires according to specified diagram with special tools. Under this board there is a zero room that found to arranges the entry and leaving cables through the cables ducts.

(MDF) Main Distribution Frame



Zero Room

Communication Systems

Telephony System

Lecture (4-1):- Telephony system Structure

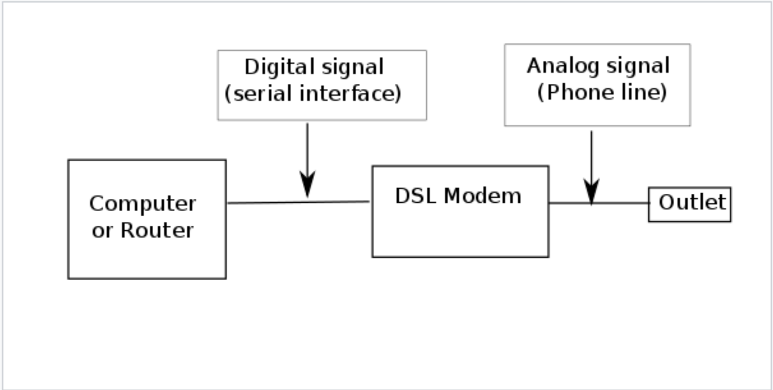
- Digital Subscriber Line (DSL)
  - The Telephone Exchanger Structure
    - Telephone Call procedures:-

Digital subscriber line (**DSL**) is a family of technologies that are used to transmit digital data over telephone lines. In telecommunications marketing, the term DSL is widely understood to mean asymmetric digital subscriber line (ADSL), the most commonly installed DSL technology, for Internet access.

DSL service can be delivered simultaneously with wired telephone service on the same telephone line since DSL uses higher frequency bands for data. On the customer premises, a DSL filter on each non-DSL outlet blocks any high-frequency interference to enable simultaneous use of the voice and DSL services.

The bit rate of consumer DSL services typically ranges from 256 kbit/s to over 100 Mbit/s in the direction to the customer (downstream), depending on DSL technology, line conditions,

The DSL end of the connection consists of a terminal adapter or "DSL modem". This converts data between the digital signals used by computers and the analog voltage signal of a suitable frequency range which is then applied to the phone line.



DSL Modem schematic

Available types of DSL are:

SDSL. Single-pair symmetric high-bit-rate digital subscriber lines operate on a single copper twisted pair. The advantage is a reduction from two wire pairs to just one. 128 kbps–2 Mbps up- and downstream; range 2 miles.

ADSL. Asymmetric digital subscriber lines deliver traffic at different speeds, depending on its direction, and support a wide range of data services, especially interactive video. ADSL provides three information channels: an ordinary telephone (POTS) channel, an upstream channel, and a higher-capacity downstream channel. These are independent, i.e., voice conversation can exist simultaneously with data traffic. These channels can be separated by frequency-division multiplexing. Downstream speed 1.5–7 Mbps; upstream 16–640 kbps; range 2–3.4 miles.

ADSL Lite. A slower version of ADSL designed to run over digital loop carrier systems and over lengths of more than 3 miles. Downstream 384 kbps – 1.5 Mbps; upstream 384–512 kbps.

HDSL. High-bit-rate digital subscriber lines (two twisted pear) provide services in both directions for applications that require communications symmetry, such as voice, corporate intranets, and high-volume email. Typical use is between corporate sites. 1 Mbps up- and downstream; range 2–3.4 miles.

•

ISDN. An international communications standard for sending voice, video, and data over digital telephone lines. Up to 144 kbps up- and downstream; range 3.4–4.5 miles.

VDSL. Very high-bit-rate asymmetric digital subscriber lines provide very high bandwidth downstream, but have distance limitations and require fiber optic cable. Originally developed to provide video-on-demand over copper phone lines. Downstream 13–52 Mbps; range 1000 ft.

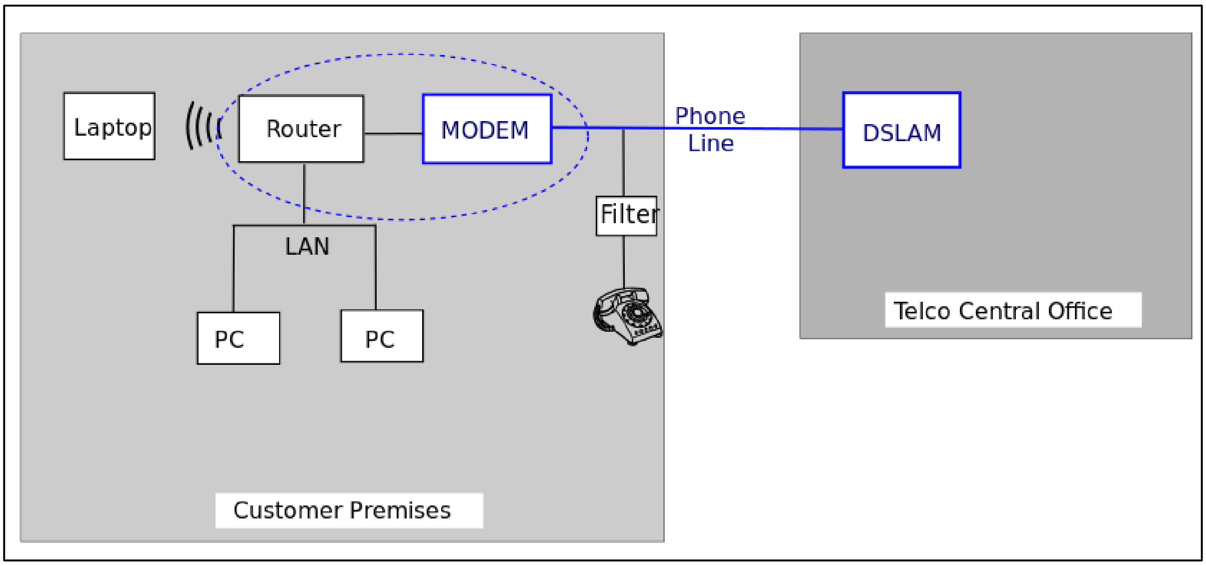
G.DMT (G Discrete Multi-Tone ASDL

A kind of asymmetric DSL technology, based on DMT modulation, that offers a downstream bandwidth of up to 8 Mbit/s, and an upstream bandwidth of up to 1.544 Mbit/s. G.DMT is another name for the standard officially known as ITU-T Recommendation G.992.1.

The frequency layout can be summarized as:

- 30 Hz-4 kHz, voice.
- 4–25 kHz, unused guard band.
- 25–138 kHz, 25 upstream bins (7-31).
- 138–1104 kHz, 224 downstream bins (32-255).

Typically, a few bins around 31-32 are not used in order to prevent interference between upstream and downstream bins either side of 138 kHz.



Telephone Exchanger

A **telephone exchange**, **telephone switch**, or **central office** is a telecommunications system used in the public switched telephone network (PSTN) or in large enterprises. It interconnects telephone subscriber lines or virtual circuits of digital systems to establish telephone calls between subscribers.

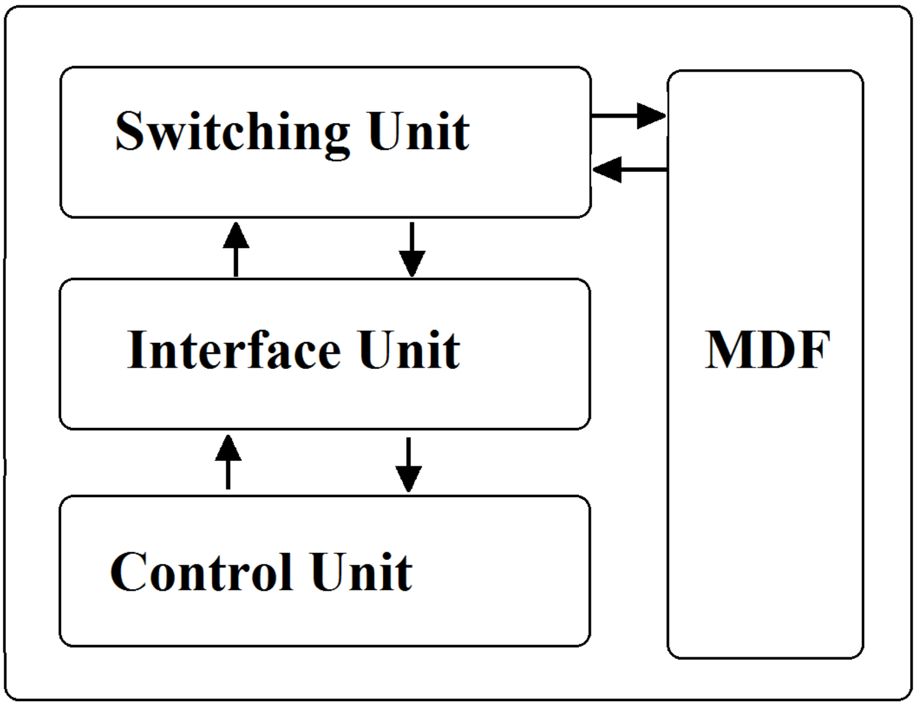
The telephone exchanger consist of the following parts:-

1 – Switching Unit.

2 – Control Unit.

3 – Interface Unit

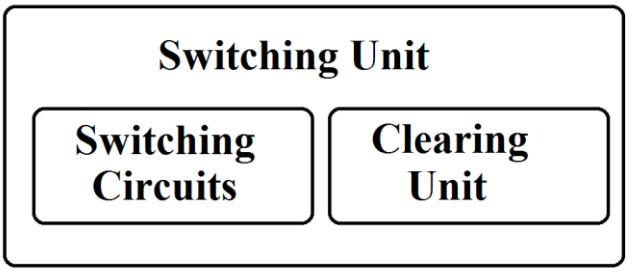
4 – Mane Distribution Frame (MDF)



Telephone Exchanger Block digram

Switching Unit

this unit exchange the digital code that came from each samples from the asking subscriber sound with the that came from the called subscriber sound, detect the hand off signal (wake – up signal), provide the subscriber with life tone, also this unit contain the different signal tone generator and the clearing unit that release the switching circuit in case of ending the call.



Control Unit:-

This job of this unit is to control all the exchanger devices and contain the operating system that arrange the exchanger, the number of the called subscriber are recognized in this unit. This unit provide the suitable paths for the call signal flow, and this unit contain data base for the subscribers assist the class of service of each subscriber and it states. These jobs can be stated as:-

1 – Control the switching unit and all the exchanger devices.

2 - Recognize the asking subscriber states.

3 – Analysis the dialing signal to detect and collect the called subscriber number

4 - Provide the suitable paths for call (between the exchangers)

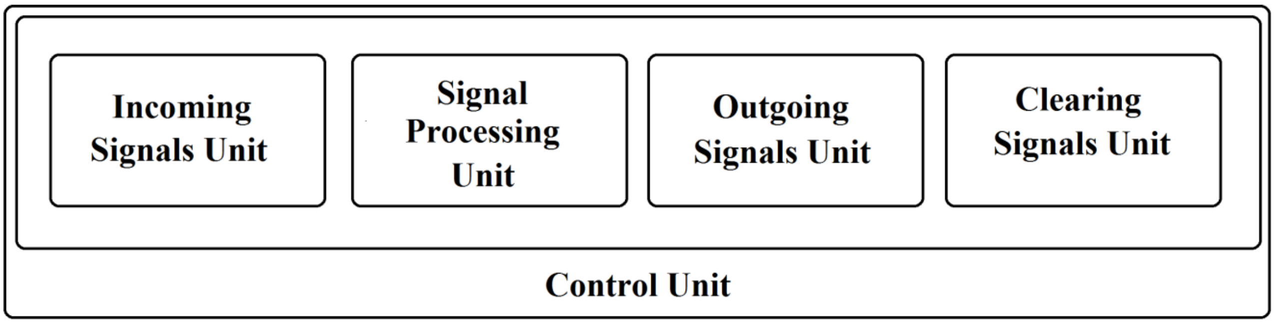
The control unit consist of:-

1 – Incoming Signals unit.

2 – Signal processing unit.

3 – Outgoing signals unit.

4 – Clearing signals unit.



Interface unit:-

This unit arranging and matching the transferring the data between the high speed units (signal processing unit) and the low speed unit (switching unit).

Telephone Call procedures:-

1 – The subscriber rise the hand set from the telephone set, then the wake-up signal will be generated (from -48V DC to -24V DC).

2 – The exchanger (switching unit) recognize the wake-up signal and search in the data base for the asking subscriber state then (if no obstacle) send the life tone signal (-24V DC + 5v sinusoidal signal 440Hz) to the subscriber that rise the hand set.

3- The subscriber dial the called subscriber numbers through the dialing circuits (pulse tone -24V DC + 40V pulse signal 40 µSec on and 60 µSec off) (DTMF -24V DC + 5v Sinusoidal signal specific two tone).

4 – The exchanger (control unit) recognize and collect the called subscriber number and provide the suitable path to achieve the call.

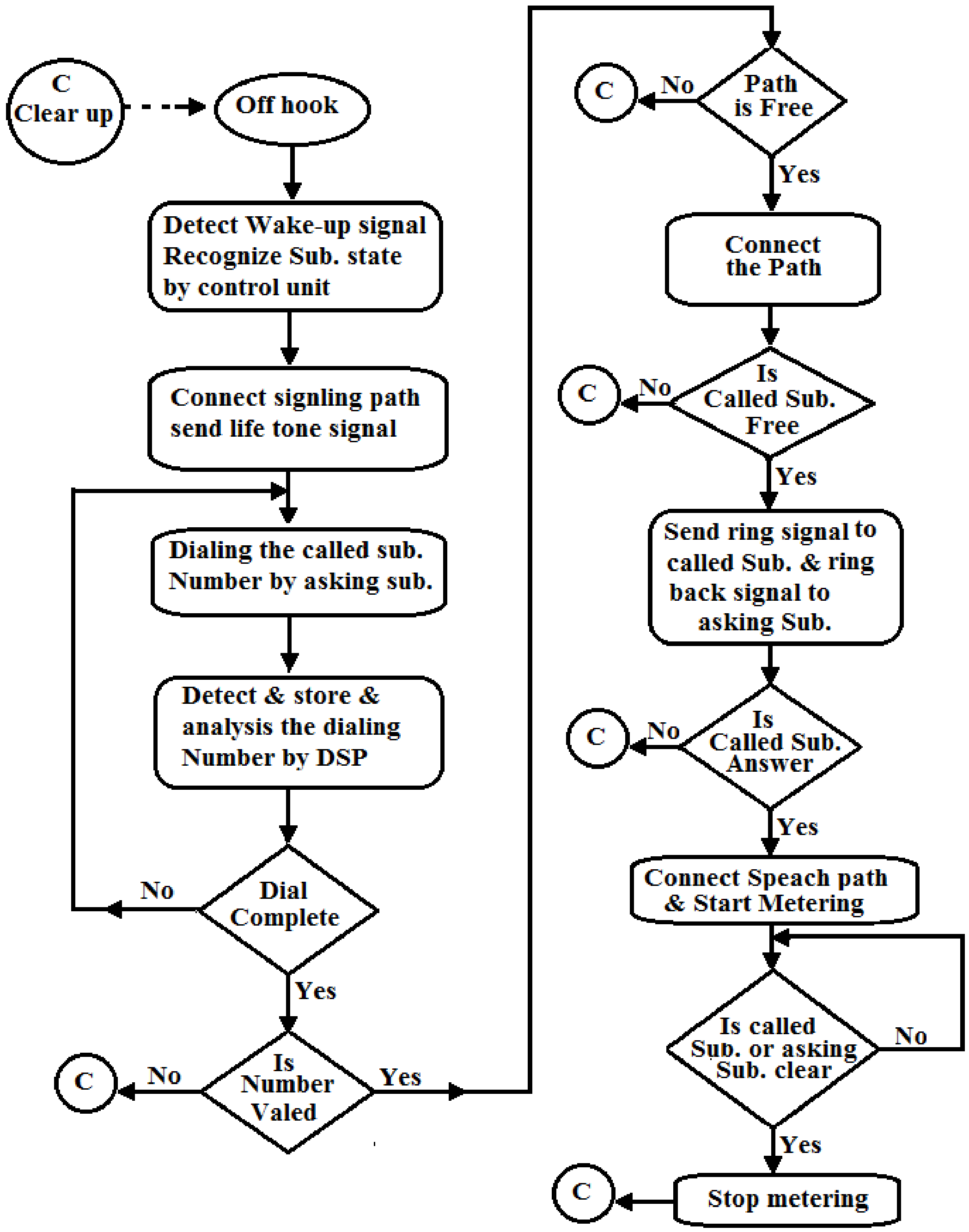
5 – After achieving the path the exchanger send the called subscriber number to corresponding called subscriber.

6 – The corresponding exchanger for the called subscriber search in the data base for the called subscriber to Know it state and (if no obstacle) send the ring signal (- 48V Dc + 40V sinusoidal signal 25Hz) to the called subscriber while the corresponding exchanger for the asking subscriber send the ring-back signal to the asking subscriber.

7 - If the called subscriber answer the call (rise the hand set from the telephone set so wake-up signal will be generated) the each two exchangers provide Speech path (tow time slot one for receive speech data and the second for send the speech data signal), the control unit observe the call and the cost monitoring start at the moment of the speech path provided.

8 – If any one of the two subscribers end the call (put the hand set on the telephone set (hand on) Clear signal (-24V DC to -48V DC) will be generated in the telephone set. Then the exchanger recognize this signal (stop the cost call metering) and the control unit send clear signal to the switching unit to make switching in the release case. Also this exchanger send clear signal to the next exchanger to make the process that lead to end the call in the next side of call.

The flowchart of the above process are shown below



Telephone Call Procedures Flowchart

Communication Systems

Telephony System

Lecture (5-1):- Digital Subscriber Card

Digital Subscriber Card:-

The digital subscriber card or also called (plain old telephone service (POTS)) is the first item that receive the analog speech signal from the telephone set through the telephone network, they connected directly with these lines came from the MDF, several type of digital subscriber cards will be implemented, some of them serve one subscriber other serve many subscriber (2 , 4, 8, 16 etc,,). Each part that serve one subscriber contain the following circuits:-

1- Matched circuits

2- Hybrid circuit

3- PCM Encoder which contain

A- Transmitter Low pass filter

B- Quantizer ((A/D) converter)

C- Compression circuit

D- Parallel to serial

4 – PCM Decoder which contain

A - Serial to parallel

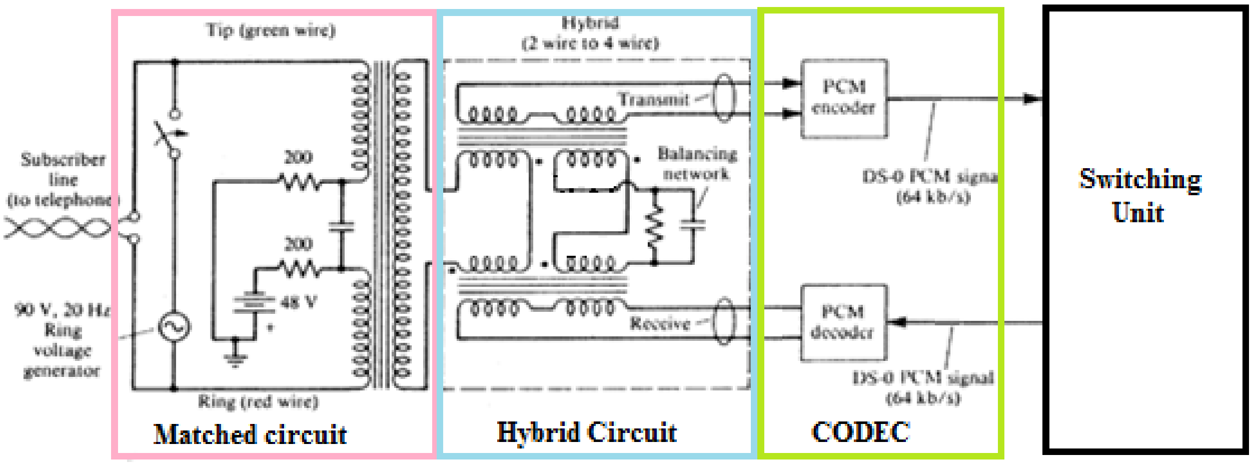
B - Expansion circuit

C - Decoder ((D/A) converter)

D - Receiver low pass filter.

However PCM encoder and PCM decoder can be lie in one electronic chip called CODIC chip which it convert (the samples of the speech signal for asking party) to PCM form and send this PCM code to the switching circuits

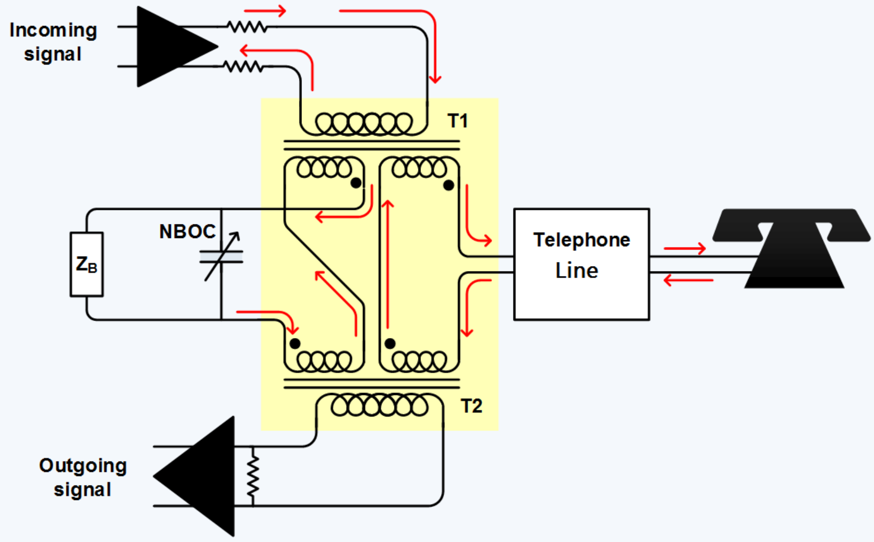
And in other way receive the PCM code that came from the called party and convert this code to an analog signal and send this analog signal to the telephone set through the hybrid circuit and telephony network. The block diagram of the digital subscriber card are shown below



Digital Subscriber Card Block Diagram

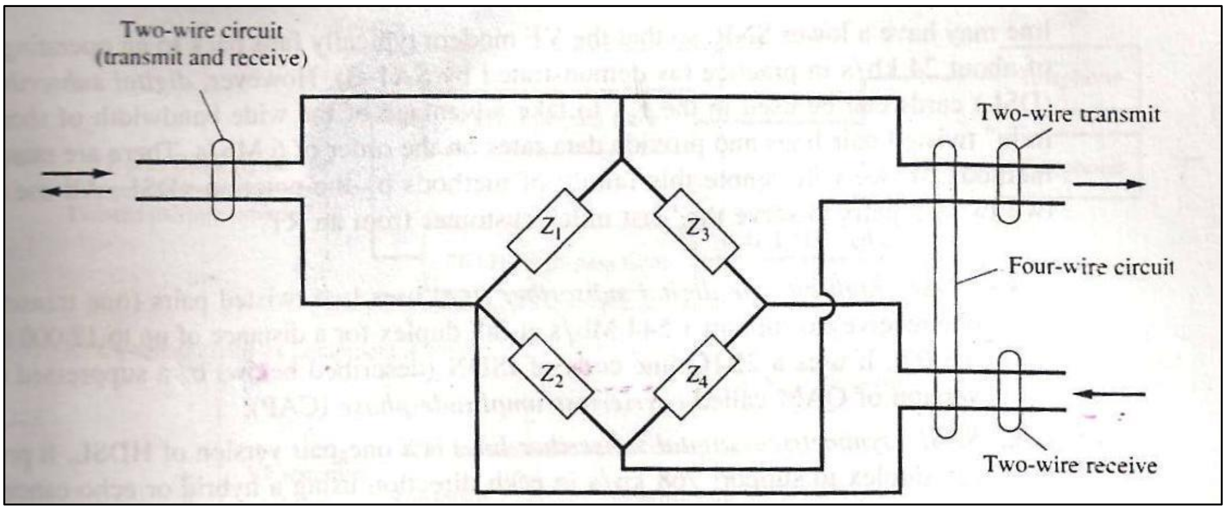
1 - Matched circuits: - The job of this circuit is to insure transferring the speech signal with maximum power efficiency by matching the input impedance of the digital subscriber card with the telephone line impedance, this part of the digital subscriber card may contain signal generator of ring signal or other telephony signals.

2 – Hybrid circuit. The hybrid circuit (also called two to four wire circuit) is a balanced transformer circuit (or its equivalent electronic circuit) that provides isolation for the transmitter and receiver signals.



Hybrid balanced transformer circuit

The equivalent hybrid circuit acts as a balanced Wheatstone bridge where $\frac{Z_1}{Z_3} = \frac{Z_2}{Z_4}$. Thus the voltage on the receive line is balanced out and does not appear on the transmit line (Upper right). Consequently, self-oscillation (ringing feedback) is prevented, even though there may be some coupling of the amplified transmitted signal to the receive line at the distant end of the four-wire line or coupling from the transmit line to the receive line along the four-wire path. As shown



PCM Encoder:-

A - Transmitter Low pass filter:-

CCITT limits telephone signals carrying the human voice to a bandwidth of from 300Hz to 3,400 Hz. Although the human voice contains lower and higher frequencies this limited bandwidth is sufficient for normal intelligibility.

Thus the An anti-aliasing transmitter low pass filter must be used to fixed the maximum frequency of the input speech signal to 3400 Hz (Low pass filter with cutoff frequency = 3400Hz).

B – Quantizer (A/D converter):-

According to the Shannon sampling theorem. The sampling rate must be greater than twice the highest signal frequency. An anti-aliasing filter must be used to prevent any frequencies higher than those permissible from reaching the sampling circuit.

In telephony the sampling rate has been set at 8 kHz. This rate even allows for safety gap, as twice the highest voice frequency is two times 3,400 Hz - 6,800 Hz.

Practically the sampling frequency for real PCM system must be

Fs = (2* Fmax) + 0.2* Fmax so Fs must be equal or greater than 7.47kHz

And this approximated to 8kHz.

If sampling values are to be transmitted as numbers, the infinite number of different pulse amplitudes must be reduced to a finite quantity of sampling values. If 256 values are used, as set down according to CCITT, a certain rounding-off error, the so-called quantization error, will result when an intermediate value is rounded off to a lower or higher value.

In order to reduce the quantization error they used A/D converter with N=12 bit (or with 4096 levels).

Using 12 bit PCM code lead to increase the required transmitted bandwidth and the number of digital circuits used in the next stages (multiplexing, shift registers , memory and other digital circuit), therefor the designer of the digital subscriber card thought about reduce these circuits and the required channel bandwidth by compress the 12 bit PCM code to 8 bit PCM code by using digital compression technique.

C - Compression circuit

The CCITT standard was based on the following considerations:

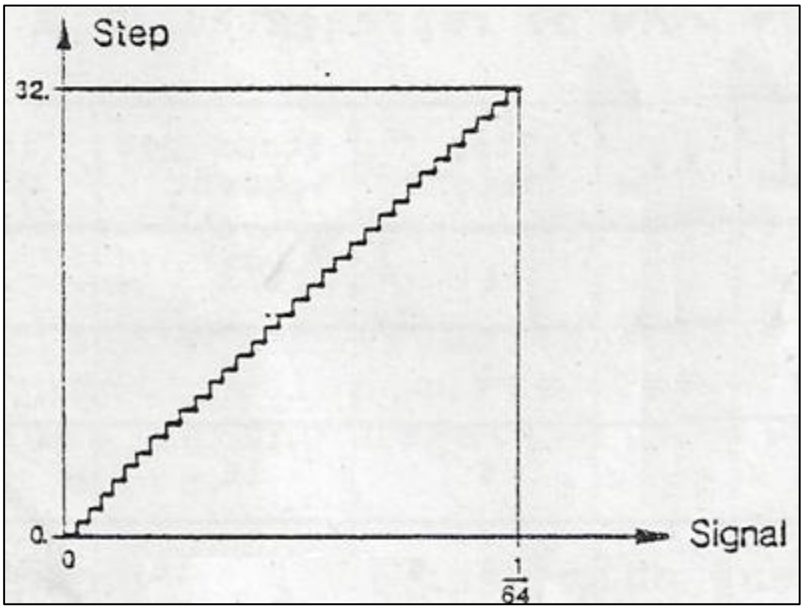
1- Both large and small signals should be sampled in relatively fine steps.

2- Both large and small signals should have approximately the same signal to noise ratio in order to achieve uniform clarity for both large and small signals.

3- Series of tests have shown that acceptable clarity is achieved when the noise level amounts to only 1/64 of the signal. S / N = 36 dB

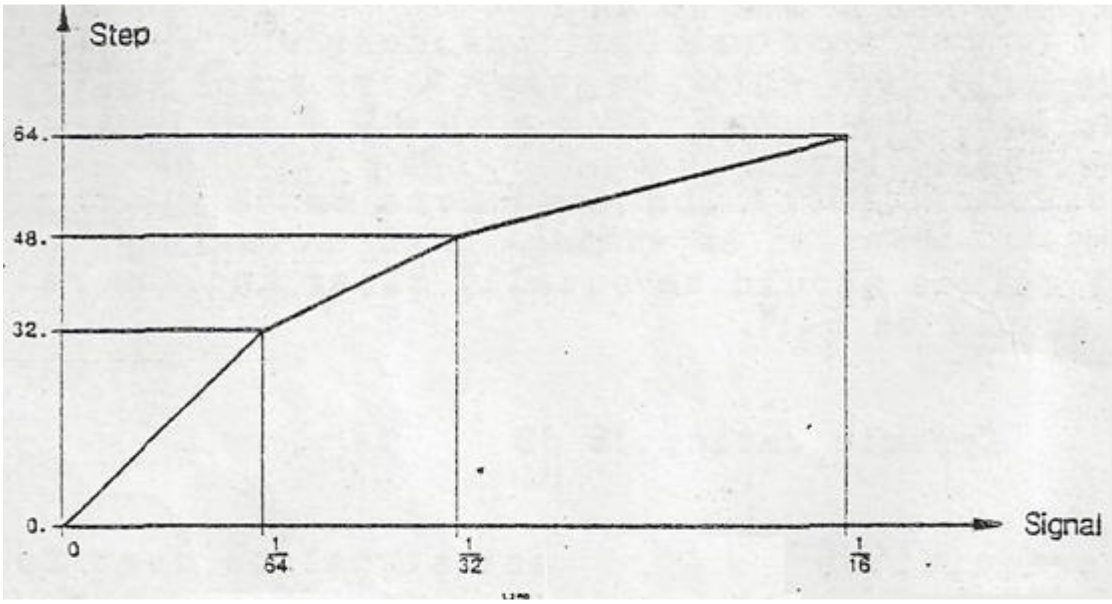
4- Loud and soft voices should be equally clear and the ratio between them should be 64:1. Dynamic ratio: 36 dB

Based on these prerequisites the smallest signal to meet full quality standards will be sampled at 32 steps per half signal.



Signals that are twice as high (i.e. signals with 1/32 of the maximum amplitude) may be sampled in steps twice as large without affecting, clarity, which means that 1 6 steps of double height can be added to the 32 fine steps. The 16 new steps taken together are as high as the 32 fine steps.

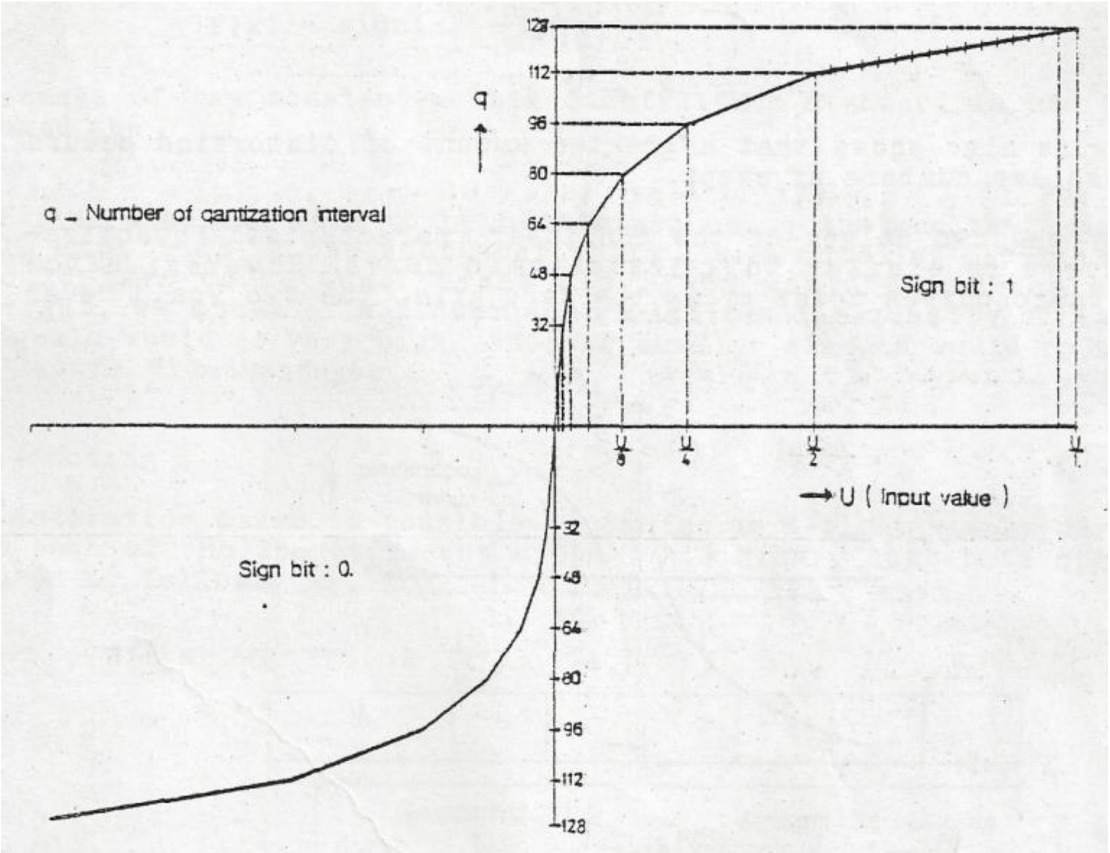
If the signal is doubled again, thus representing 1/16 of the maximum signal amplitude, an additional 1 6 new steps are used which are now four times as high as the finest steps.



Further expansion can now be- represented in a table:

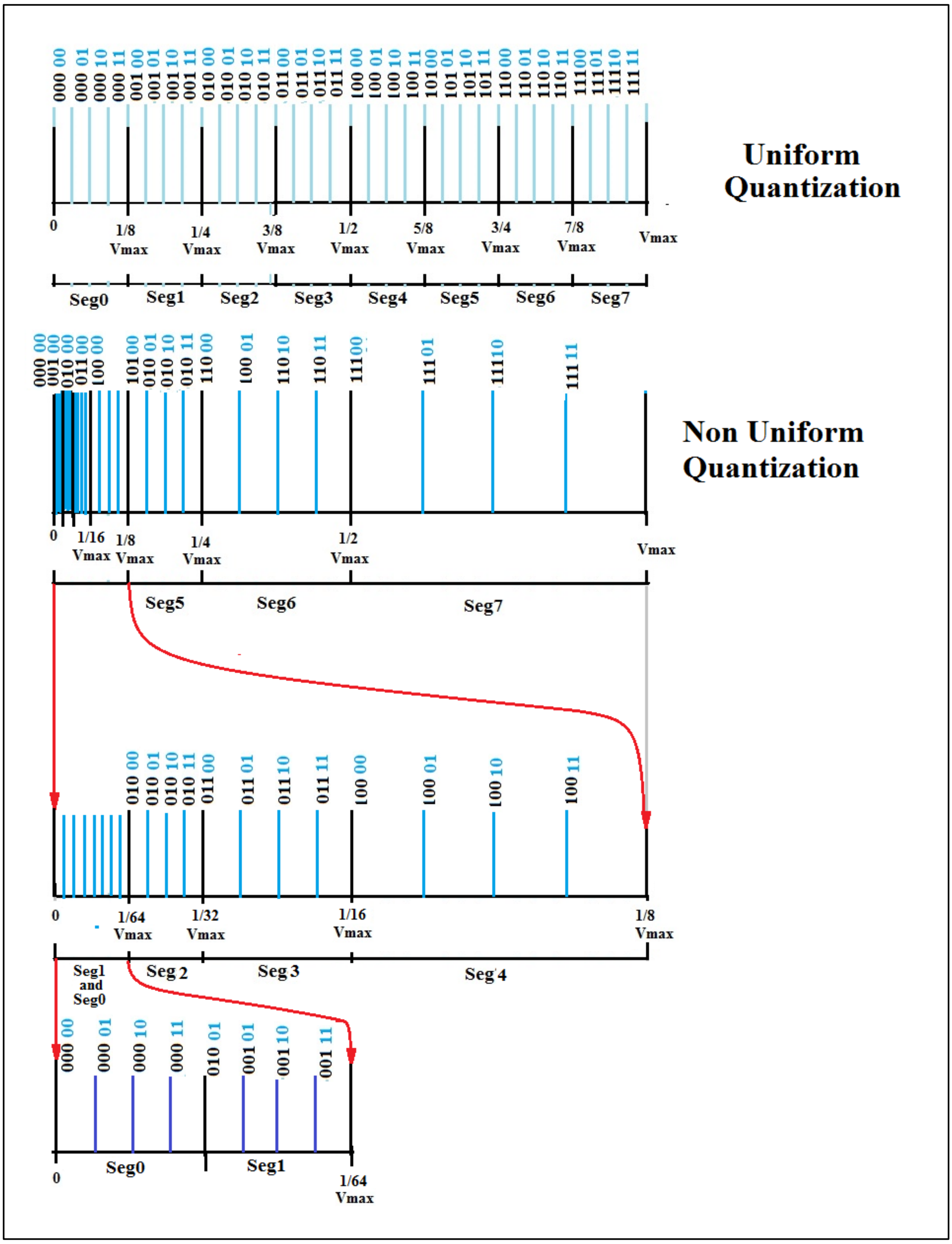
Segment No.	max. amplitude	Step height	Steps per segment	Steps total
1	$\frac{1}{64}$	1	32	32
2	$\frac{1}{32}$	2	16	48
3	$\frac{1}{16}$	4	16	64
4	$\frac{1}{8}$	8	16	80
5	$\frac{1}{4}$	16	16	96
6	$\frac{1}{2}$	32	16	112
7	1	64	16	128

The table has 7 segments each representing a doubling of the height of the steps. Since 16 new steps can be added to each segment, stepping is relatively the same for each signal height, i.e. the quality of quantization (the signal to noise ratio) is uniform over a wide area.



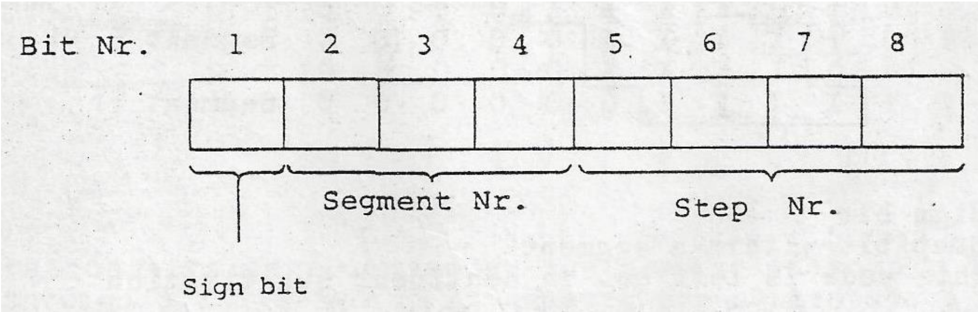
This figure shows both the positive and the negative side of the so called nonlinear quantization curve. The innermost segment (segment no.1) continues on the negative side, yielding a 13 segment characteristic curve.

Measurements of the proportion of distortion due to quantization (quantization distortion) demonstrate the effectiveness of the methods described. The signal to noise is uniform over a wide area.


Uniform Quantization
0
1/8 Vmax
1/4 Vmax
3/8 Vmax
1/2 Vmax
5/8 Vmax
3/4 Vmax
7/8 Vmax
Vmax
Seg0
Seg1
Seg2
Seg3
Seg4
Seg5
Seg6
Seg7
Non Uniform Quantization
0
1/16 Vmax
1/8 Vmax
1/4 Vmax
1/2 Vmax
Vmax
Seg5
Seg6
Seg7
0
1/64 Vmax
1/32 Vmax
1/16 Vmax
1/8 Vmax
Seg1 and Seg0
Seg 2
Seg 3
Seg 4
0
Seg0
Seg1
1/64 Vmax

Coding:

Quantization makes it possible to assign an 8-digit binary number to each of the 256 different steps. This binary number is structured as follows:



Bit 1 determines allocation to the upper (positive) or lower (negative) side of the characteristic curve.

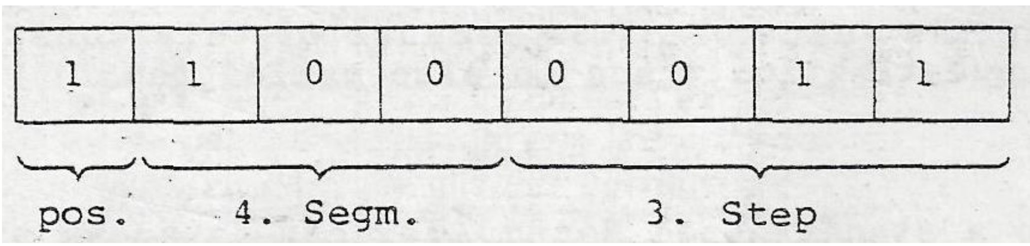
1: Positive

0: Negative

Bits 5, 6, 7, and 8 determine the number of the step within a segment. Since 16 different numbers can be formed with 4 bits, this 4-digit combination is sufficient for the 16 possible steps in a segment. Segment 1, however, which has a total of 64 steps on the positive and negative side, is an exception. This segment is divided into 2 positive and two negative "sub-segments", each of which has only 16 steps.

Thus,, each half, of the characteristic curve has 8 segments (6 segments and 2 "sub-segments") Three bits of the code, bits 2, 3 and 4, are sufficient for marking these 8 segments.

Example:



This code marks the third step in the fourth segment, the segment which has 8 times, as many steps as the innermost segment in the positive half of the characteristic curve.

The following figure shows the coding of a linearly quantized signal with 4096 steps, which requires 12 bits. Standard nonlinear quantization is achieved by leaving out codes.

Bit	1	2	3	4	5	6	7	8	9	10	11	12	
	V	0	0	0	0	0	0	0	X	X	X	X	Segment 1a
	V	0	0	0	0	0	0	1	X	X	X	X	Segment 1b
							(1	0	X	X	X	X)	
	V	0	0	0	0	0	1	X	X	X	X	0	Segment 2
						(1	0	X	X	X	X	0)	
	V	0	0	0	0	1	X	X	X	X	0	0	Segment 3
					(1	0	X	X	X	X	0	0)	
	V	0	0	0	1	X	X	X	X	0	0	0	Segment 4
				(1	0	X	X	X	X	0	0	0)	
	V	0	0	1	X	X	X	X	0	0	0	0	Segment 5
			(1	0	X	X	X	X	0	0	0	0)	
	V	0	1	X	X	X	X	0	0	0	0	0	Segment 6
		(1	0	X	X	X	X	0	0	0	0	0)	
	V	1	X	X	X	X	0	0	0	0	0	0	Segment 7

V: Sign bit

X: Step bit within a segment

( ): This code is left out in nonlinear quantization.

Since a sampling is made 8,000 times per second and every sampling value converted into an 8 digit code, 64,000 bits per second are required for the transmission of a voice signal.

Each of the 8-bit groups is transmitted serially.

MSB							LSB
1	2	3	4	5	6	7	8

Bit 1, the sign bit, is the Most Significant Bit-(MSB) and is transmitted first. The last bit, which is supplied by the encoder' circuit, is the Least Significant Bit (LSB).

Example :-

For the PCM Encoder suppose that the maximum input voltage is 8 V

Find the PCM code for instantaneous input signal with 2.8V in both uniform and non-uniform quantization for N=8.

For uniform quantization:-

$\Delta q = Vmax/ql$ $ql = 2^n = 2^8 = 256$ but in fact the ql = 128 because one bit are occupied for sign. (+ or -)

$\Delta q = 8 / 128 = 0.0625$ volt

The equivalent value for instantaneous input signal = vin / Δq

= 2.8 / 0.0625=44.8   ---- = 45

The binary code for 45 is 10101101 not that the MSB is sign bit (1 for positive)

For non-uniform quantization

The instantaneous input voltage signal 2.8 it is positive polarity so the sign signal bit is 1 also it is lie between the 1/4 Vmax and 1/2 Vmax so it is in the Seg 6

The for MSB are 1110

The equivalent value within seg. 6 are calculated as:-

1/4 v max = 8/4 = 2 volt

The offset value from the 1/4 vmax = 2.8 – 2 = 0.8

The equivalent value within seg6 = the offset value/ Δq

Δq = Vmax/2-Vmax/4 /16 = (4-2) /16 = 0.125 volt

The equivalent value within seg6 = 0.8/0.125 = 6.4 --- = 6

The binary code = 0110

So the overall non-uniform code for 2.8 v is 1110 0110

Other Sol :-

Vin / Δq

Δq = Vmax / 2^{n-1} = $8/2^{11}$ = 8/2048 = 0.00390625 v

The vin equivalent value = Vin / 0.00390625 = 716.8 = 717

The binary code for 717 = 101011010001 not that the MSB is the sign bit

To find the compressed 8 bit code

1 - Copy the sign signal bit from the 12 bit code to the 8 bit code.

2 – Extract the first 1" from the left side of the code to find the segment no.

as the following

s 1 x x x x x x x x x x	seg 7
s 0 1 x x x x x x x x x	seg 6
s 0 0 1 x x x x x x x x	seg 5
s 0 0 0 1 x x x x x x x	seg 4
s 0 0 0 0 1 x x x x x x	seg 3
s 0 0 0 0 0 1 x x x x x	seg 2
s 0 0 0 0 0 0 1 x x x x	seg 1
s 0 0 0 0 0 0 0 x x x x	seg 0

for our example the code is 10<u>1</u>011010001 the first 1" after the sign bit lie in seg 6.

So the four MSB of the compressed code are 1110 not the the MSB is the sign bit and the next 3 bits are the segment no.

The input signal code within the segment can be found by taking the next four bit after the first 1" that represent the Seg. No. In our example this code are 0110

Then the overall compressed code are (1110 0110) which equal to that we find in the first method.

Note :- to find the expansion 12 bit code from the compressed 8 bit code

1 – Copy the sign bit from the MSB of the comp. cod to the MSB of the expand code.

2 – Read the next 3 bits and put 1" in the Seg location as :-

Seg code	Expand code
111	1 x x x x 0 0 0 0 0 0 0
011	0 1 x x x x 0 0 0 0 0 0
101	0 0 1 x x x x 0 0 0 0 0
100	0 0 0 1 x x x x 0 0 0 0
011	0 0 0 0 1 x x x x 0 0 0
010	0 0 0 0 0 1 x x x x 0 0
001	0 0 0 0 0 0 1 x x x x 0
000	0 0 0 0 0 0 0 x x x x

3 – Copy the next four bits (the code within the segment) instead of x x x x bits in the above table.

4 – complete the 12 bits by inserting the zero in the next bits after the code within the segment bits.

The compressed code are 1 1 1 0 0 1 1 0

The expand code are 1 0 1 0 1 1 0 0 0 0 0 0

Communication Systems

Telephony System

Lecture (6-1):-

Telephony signals Multiplexing

Telephony Framing

Switching units

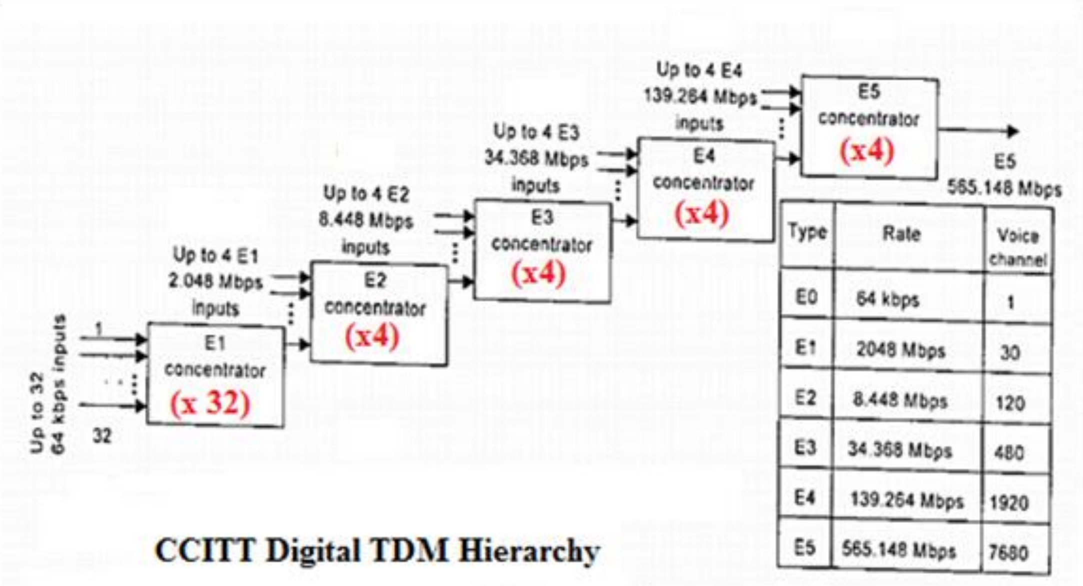
Time switching

Space switching

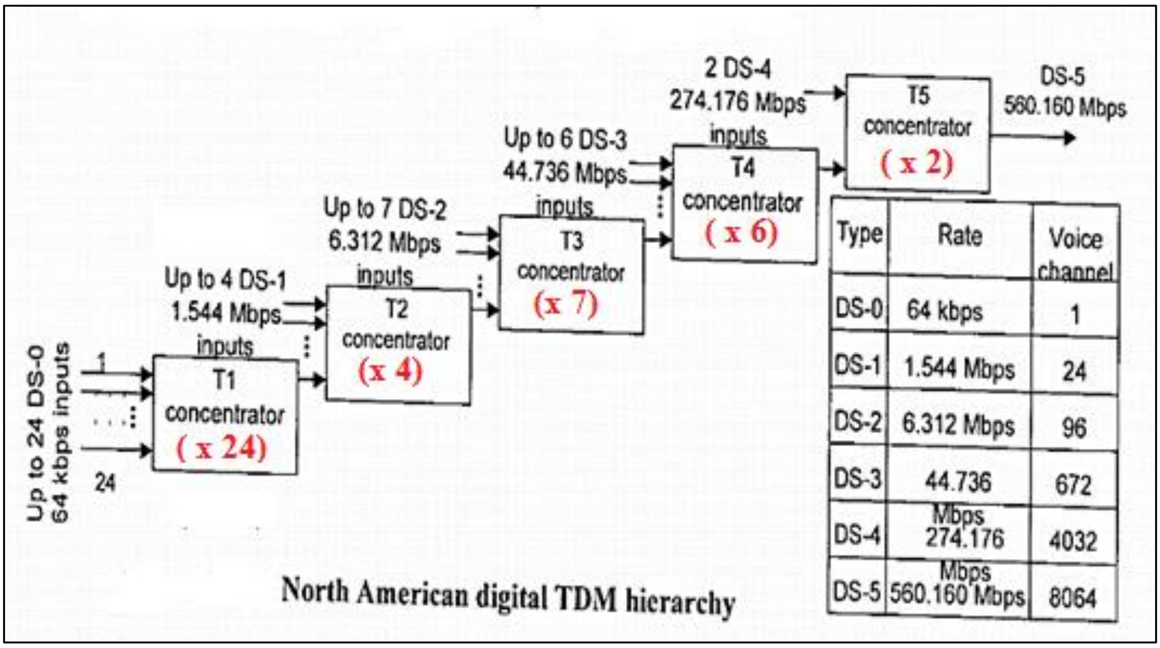
Space-time switching

***<u>Telephony Multiplexing</u>*:-**

After the digitized (PCM) speech signal leave the digital subscriber card it is enter to the multiplexing stage to generate the line telephony form which also called frame construction of PCM speech signals. In general there are two type of PCM multiplexing (PCM-TDM) systems, the CCITT and the North America system. The block diagram of these two systems are shown below.

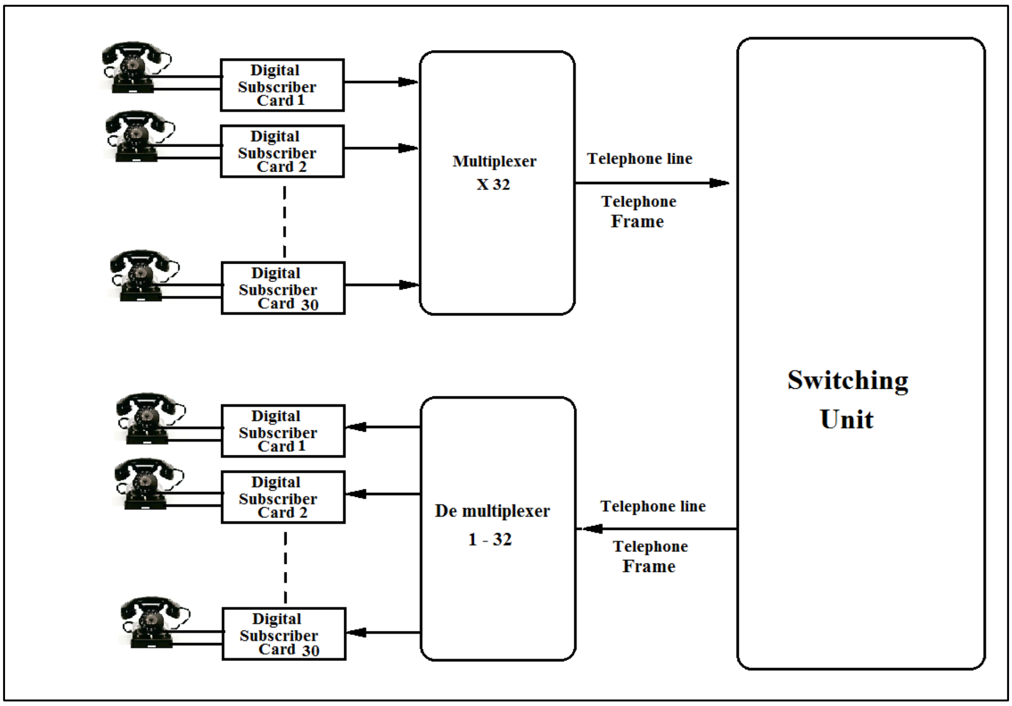


CCITT Digital TDM Hierarchy



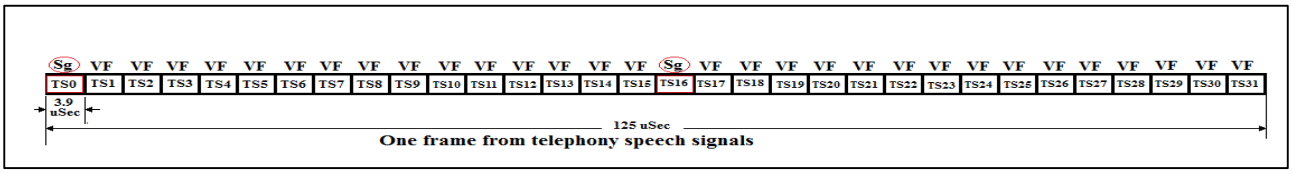
North American digital TDM hierarchy

The block diagram that illustrate the processing stages of the PCM speech signal passed through them are shown:-



Telephony Framing:-

As shown in CCITT TDM hierarchy each 30 voice channel and two signaling channel are multiplexed into one line called telephony speech frame, this frame occupied 125 micro second, in this frame each voice channel occupied one time slot from the 32 time slots, this time slot which occupied 3.9 microsecond carry the 8-bits PCM code for one sample of the speech signal of one subscriber, so that the signaling rate for speech channel rise to 2.05Mbit/Sec, however time slot 0 and time slot 16 are sizing for signaling channels.



Its an important to know that each 16 frame represent one multi-frame in which all the voice channels are served by the two signaling channels, while the first (four bits) from TS16 serve vf1 to vf15 which lie in TS1 to TS15 , and the second four bits) from TS16 serve the VF16 to VF30 which lie in TS17-TS31, however TS are used for frame synchronization.

Switching Unit:-

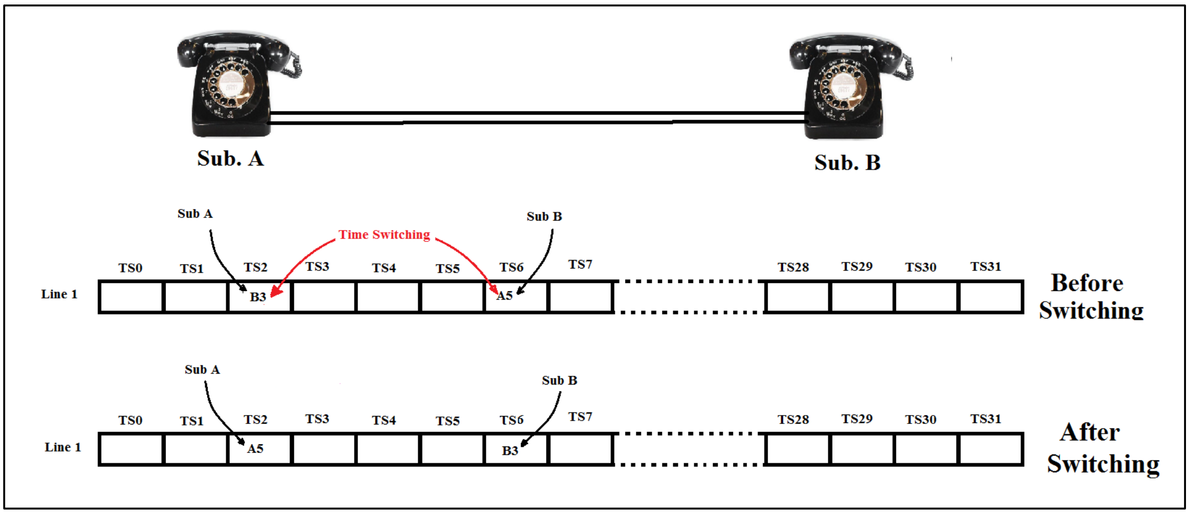
A **telephone switch**, is a telecommunications system used in the public switched telephone network (PSTN) or in large enterprises. It interconnects telephone subscriber lines or virtual circuits of digital systems to establish telephone calls between subscribers.

Different switching technique can be used according to the location of the PCM code of the two subscribers that in call in the time slots and lines. if the PCM code of these two subscriber lie in the same line but with different time slots number, time switching technique is used to achieve exchanging the speech information of these two subscribers, while if the two subscriber PCM codes lies in the same time slot number but within different lines, the space switching technique must be used to achieve the information exchanging.

If the location of the PCM code for the two subscriber within the call lies in different time slot and different lines, the two (space and time) switching techniques must be used to do the exchanging information job.

Time Switching:

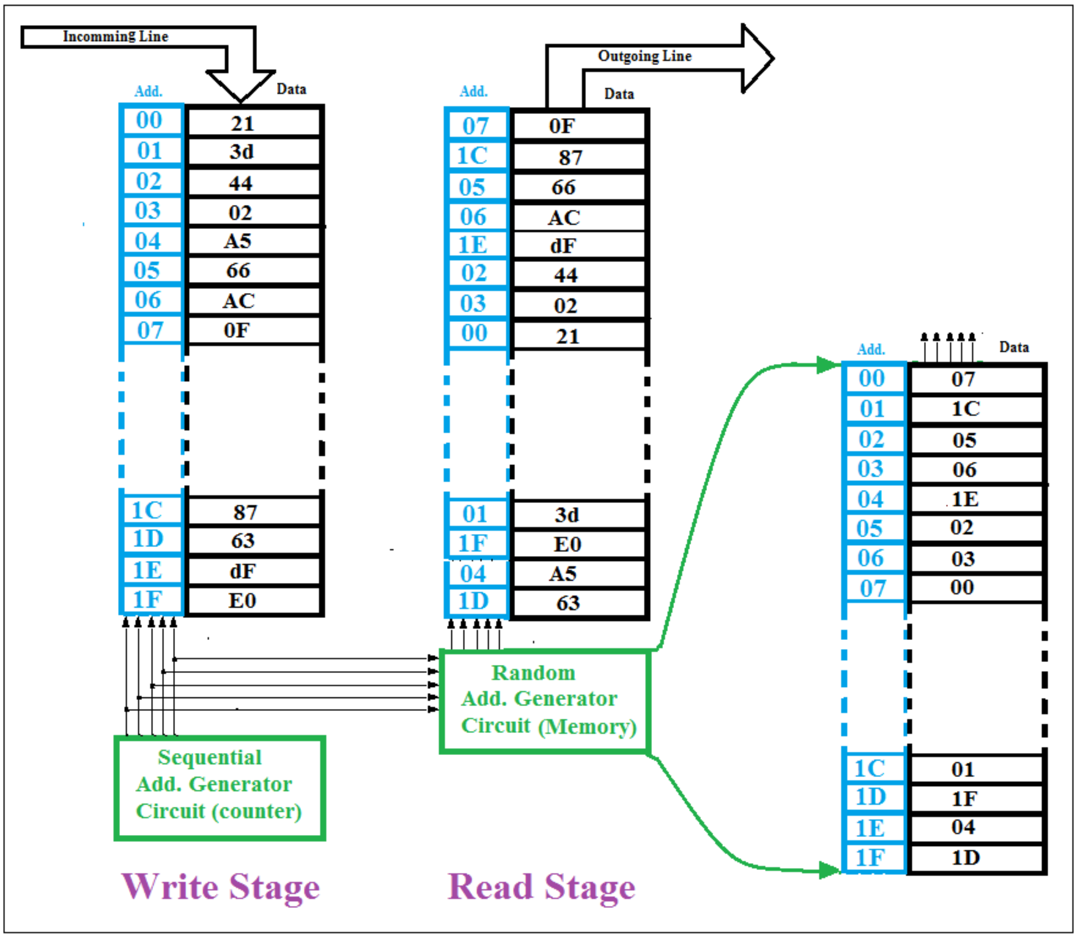
The time switching used to exchange the PCM code between two different time slots but within the same telephone line (PCM line).



In time switching, time-slot interchange is carried out through two stores, each containing a storage address for each channel of the PCM frame. The speech store consists of each data of the incoming time-slots (that is its speech sample) at a matching address. All address of the connection store corresponds to a time slot upon the outgoing line. This contains the number of the time-slot on the incoming lines that sample is to be transmitted in which outgoing time-slots. Information is write in the speech store cyclically in synchronism with the incoming PCM systems.

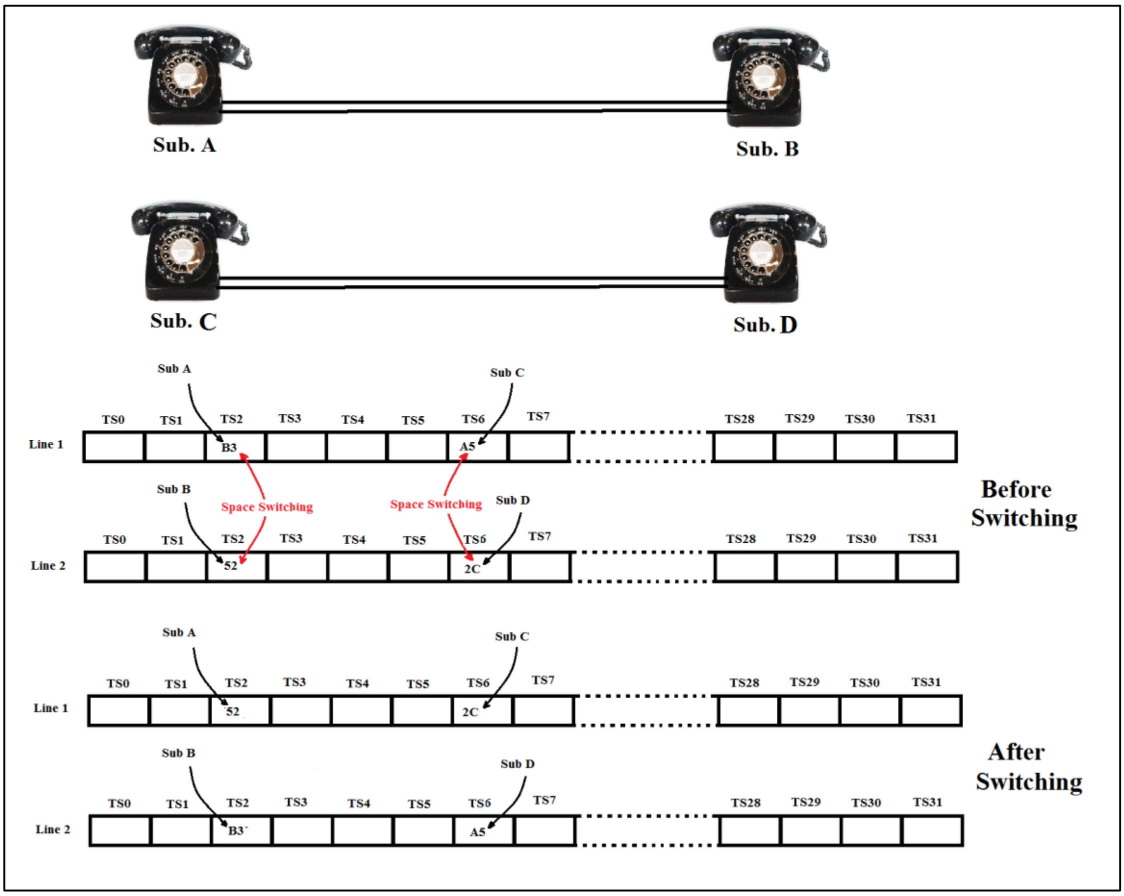
However, random access read out is utilized. ((or, the connection store has cyclic read out, although writing in is non cyclic)). Because the time switching need to store all the information in the 32 time slots, before read them in random sequence, that lead to made delay by one frame (125 micro second).

Note that the read process achieve in fact not in random sequence but with corresponding address of the time slots need to be exchanged, this addresses controlled by the control processor. The time switching process can be illustrated in the following figure.

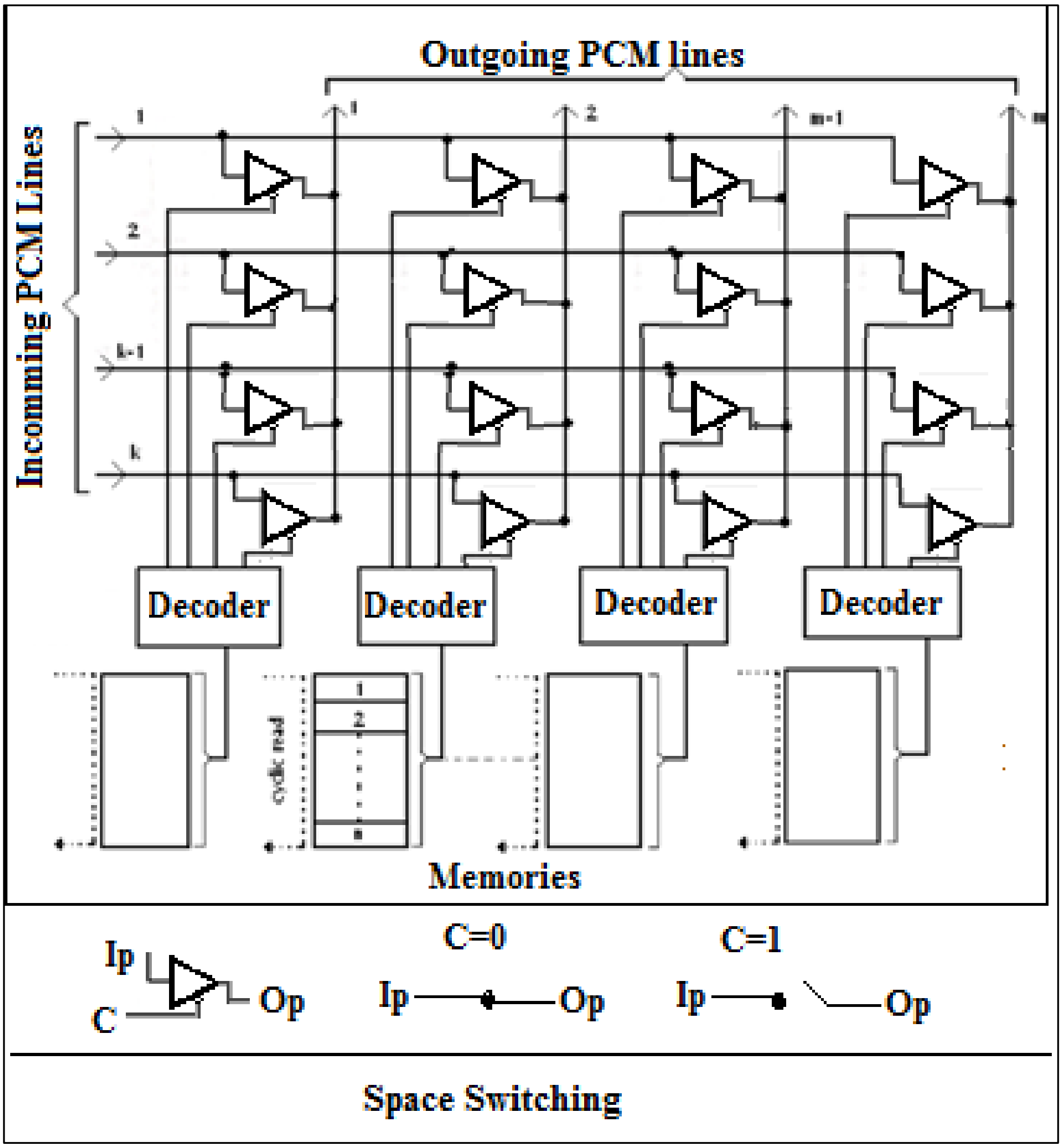


Time Switching

Space Switching: The space switching used to exchange the PCM code between two different PCM lines but with same time slots.



The space switching can be made in between outgoing and incoming PCM lines by a cross point matrix of the form demonstrated in the following figure.

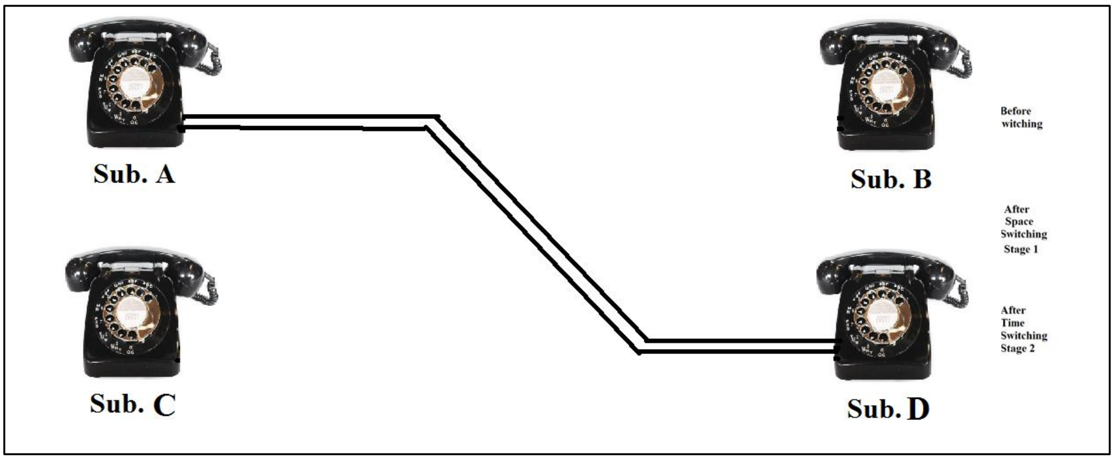


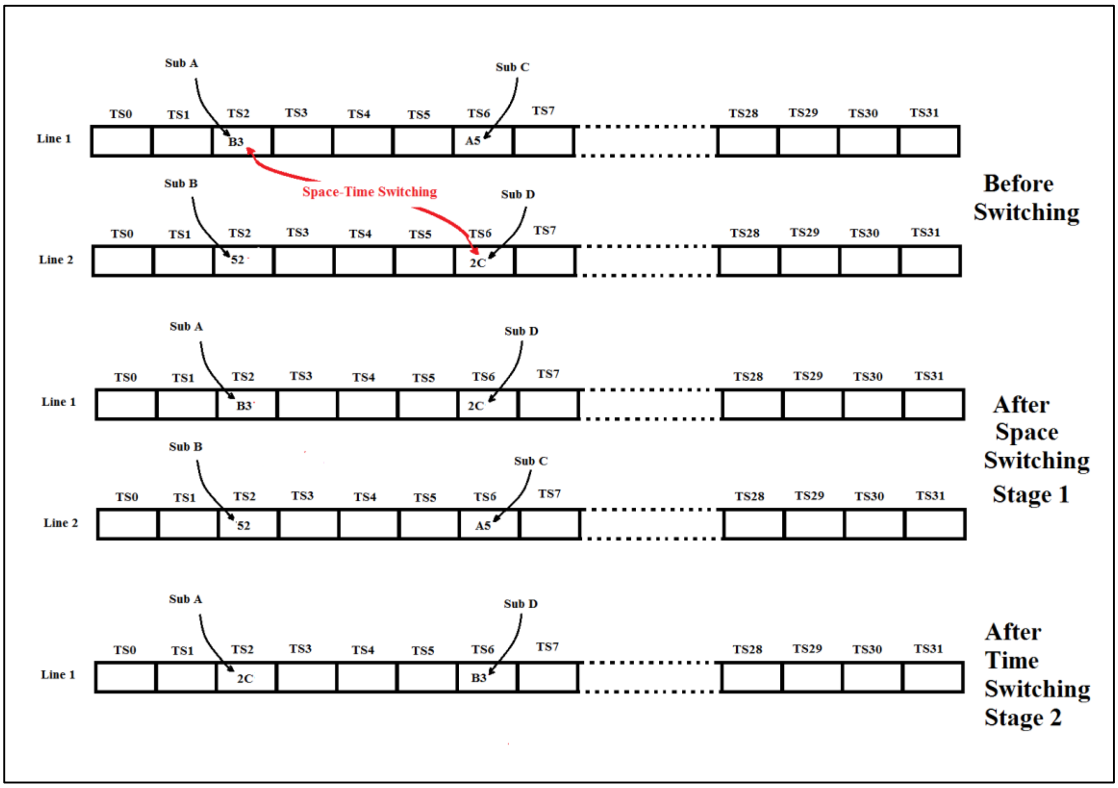
However, various channels of an incoming PCM frame may require to be switched by various cross points in order to reach various destinations. Therefore, the cross point is a two-input AND gate. One input is linked to the incoming PCM highway and another to a connection store which produce a pulse at the needed instant. A group of cross points gates can be implemented like an integrated circuit, for illustration using a multiplexer chip.

In the above figure that demonstrates a space switch with k incoming and m outgoing PCM lines, all carrying n channels. The connections store for every column of cross points is a memory along with an address location for every time-slot that stores the number of the cross points to be operated in which time slot. This number is written in the address by the controlling processor so as to setup the connection. The numbers are read out cyclically, into synchronism along with the incoming PCM frame. In every time slot, the number stored at the equivalent store address is read out and decoding logic converts it in a pulse or a single lead to function the relevant cross point.

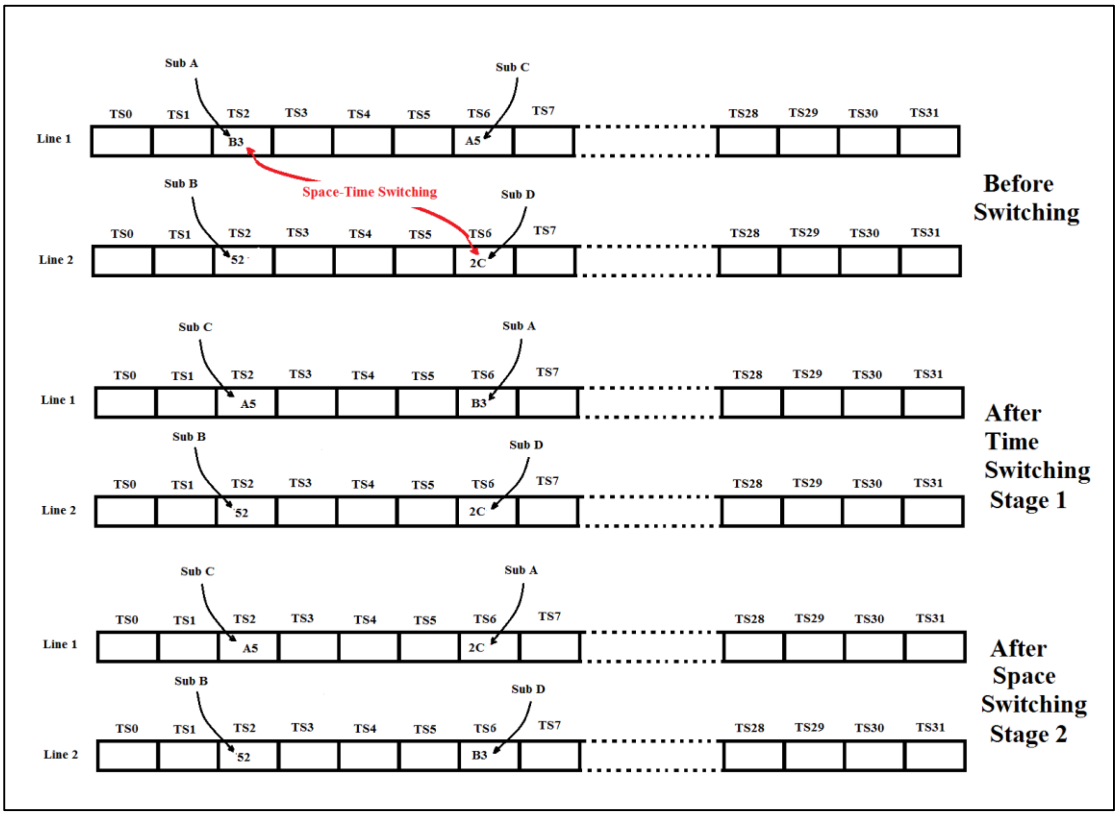
Time-Space Switching:

Time-Space or Space-Time Switching technique must be used when the time slots for the subscriber in call where not in same line and not in the same timeslot number. In this case the two space and time switching are used in cascade. As shawon in figure below.





The above job can also achieved by Time-Space technique as shown:-



Before Switching
Line 1
Sub A
Sub C
TS0 TS1 TS2 TS3 TS4 TS5 TS6 TS7 TS28 TS29 TS30 TS31
B3
A5
Space-Time Switching
Line 2
Sub B
Sub D
52
2C
After Time Switching Stage 1
Line 1
Sub C
Sub A
A5
B3
Line 2
Sub B
Sub D
52
2C
After Space Switching Stage 2
Line 1
Sub C
Sub A
A5
2C
Line 2
Sub B
Sub D
52
B3

Communication Systems

Telephony System

Lecture (7-1):-

Telephony Numbering System

Signaling in Telephony system

<u>Telephony Numbering System</u>:-

A **telephone numbering** is a type of numbering scheme used in telecommunication to assign telephone numbers to subscriber telephones. Telephone numbers are the addresses of participants in a telephone network, reachable by a system of destination code routing. Telephone numbering are defined in each of administrative regions of the public switched telephone network (PSTN) and they are also present in private telephone networks. For public number systems, geographic location plays a role in the sequence of numbers assigned to each telephone subscriber.

Many numbering system subdivide their territory of service into geographic regions designated by a prefix, often called an **area code** or **city code**, which is a set of digits forming the most-significant part of the dialing sequence to reach a telephone subscriber.

Numbering plans may follow a variety of design strategies which have often arisen from the historical evolution of individual telephone networks and local requirements. A broad division is commonly recognized between

<u>A - Closed numbering system</u>, such as found in North America, which feature fixed-length area codes and local numbers.

<u>B- Open (flexible) numbering system</u> that feature a variance in the length of the area code, local number, or both of a telephone number assigned to a subscriber line.

The International Telecommunication Union (ITU) has established a comprehensive numbering plan, designated E.164, for uniform interoperability of the networks of its member state or regional administrations.

It is an open numbering system, however, imposing a maximum length of 15 digits to telephone numbers. The standard defines a country calling code (*country code*) for each state or region which is prefixed to each national numbering plan telephone number for international destination routing.

<u>E.164</u> conformant telephone numbers consist of a country calling code and a national telephone number. National telephone numbers are defined by national or regional numbering plans, such as the European Telephony Numbering Space, the North American Numbering Plan (NANP), or the UK number plan.

Within the national numbering plan, a complete destination telephone number is composed of an area code and a subscriber telephone number. The subscriber number is the number assigned to a line connected to customer equipment. The first few digits of the subscriber number may indicate smaller geographical areas or individual telephone exchanges. Callers in a given area or country do not need to include the particular area prefixes when dialing within the same area.

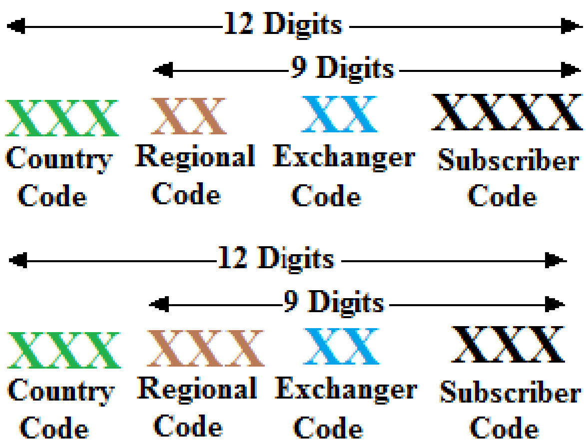
However if the subscriber number size increase, the probability of error in this number increase by humane effects, and the size of registers that needed to save these numbers also must be increased, therefore minimum number size for telephony subscribers by assign the number size in

1 – Local calls

2 – Regional (national) calls

3 – International calls

- - In local calls the call number contain the exchanger code (XX) plus subscriber code (XXXXX or XXXX) as XX XXXXX or XX XXXX that usually consist of 7 digits, or 6 digits
- - In national calls the call number contain the regional code (XX or XXX) plus the exchanger code (XX) plus subscriber code (XXXXX or XXXX) as XX XX XXXXX or XXX XX XXXX that always contain 9 digits this done by increasing the regional number from two digit to three digit if the local number contain 6 digit only.
- In international calls the call number consist of the country code (xxx) Plus regional code plus exchanger code plus subscriber code as



The country code for all country are assigned at 1964 by the ITU each have unique code that not similar to any another code, the world was divided into 9 telephony numbering regions as following

(1) for North American

(2) for Africa

(3 , 4) for Europe

(5) for south American

(6) for Australia

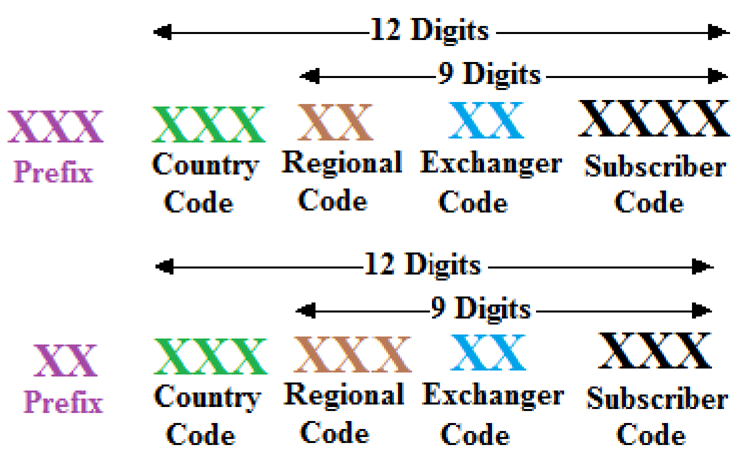
(7) for USSA

(8) for East Asian

(9) for Middle east

The country code for each country started with the telephony region that it lie in it and consist of three digits. The country code for Iraq is (964)

All the international call started with international prefix that used to indicate the call is international in order to get the required link for this call, however this prefix are different from each country to another for example the prefix of many country is "00" and for other is "010" or "009"



Signaling in Telephony system:-

The signaling is an important part from any communication system, because of without the signaling there are no ability to contact any link for transmit and received any massages. The signaling system used to arrange the start of call, the end of call, indicate the life line or blocked line and other signals that responses by the subscribers. The signaling system are also very important to arrange and achieve the call path between the exchangers, in general there are three type of signaling in telephony system;-

1 - Signaling between the subscriber and the exchanger.

2 – Signaling within the exchanger.

3 – Signaling between the exchangers.

1 – Signals between the subscriber and the exchanger

A– The wake-up signal send from subscriber to the exchanger (from -48V DC to -24V DC)

B –The life tone signal send from exchanger to the subscriber (-24V DC + 5v continuous sinusoidal signal 440Hz)

C - The dialing signal send from subscriber to exchanger

1 - (Pulse tone -24V DC + 40V pulse signal 40 µSec on and 60 µSec off)

2 - (DTMF -24V DC + 5v Sinusoidal signal specific two tone).

D – The busy tone send from exchanger to the subscriber (-24V DC + 5v discrete (one second on and one second off) sinusoidal signal 440Hz to 640Hz)

E - Ring signal send from exchanger to the subscriber (- 48V Dc + 40V sinusoidal signal 25Hz)

F - Ring-back signal send from exchanger to the subscriber (-24V DC + 5v discrete sinusoidal signal 440Hz to 640Hz)

G - Clear signal send from subscriber to the exchanger (-24V DC to -48V DC)

2 – Signals within the exchanger

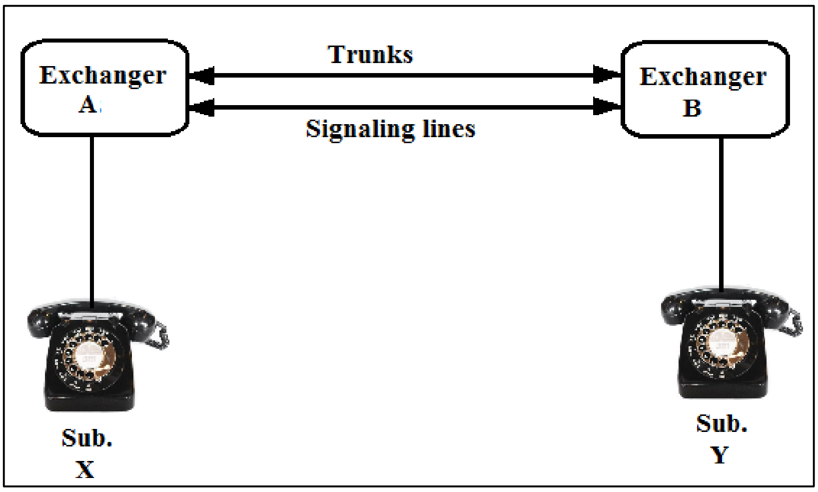
These signals transferred between the exchanger units (DSP, Switching, Control, and interface) to arrange the exchanger work, in fact there is no standard (within the exchanger) signals because the different technique that used for the different companies or factories of exchangers, but we can see that two signal stele used in every exchanger that are

A- Seize signal send from control unit to switching unit in order to seize the specific line (time slot) for call.

B – Clear signal send from control unit to switching unit in order to release the seizing line in case end the call.

3 – Signals between the exchangers

All exchangers in telephony system connected to other by using the trunk lines and special lines to arrange the transferring the coded speech signal from the asking party to the called party through the same or different levels of exchangers.



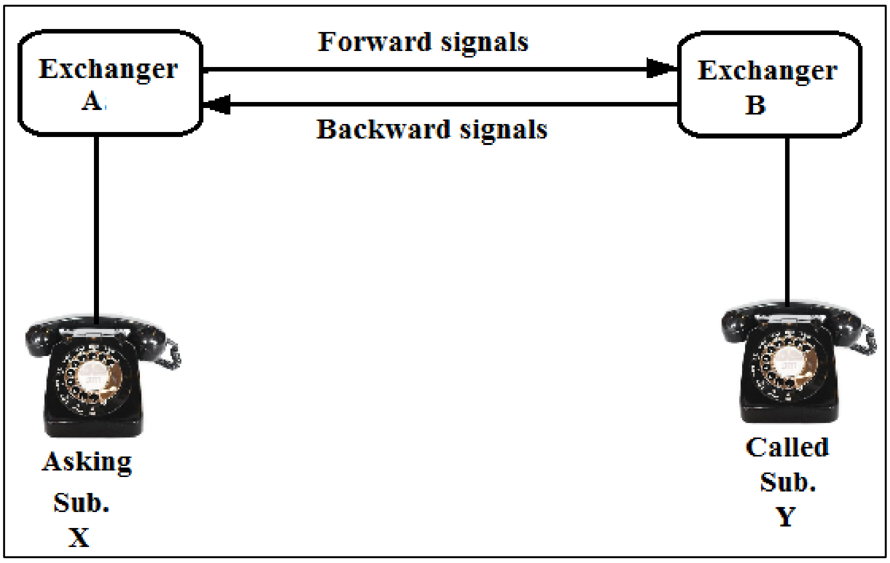
In order to arrange the transferring the speech signals between the different exchangers, there is a need to inform the exchanger about provide and seizing the lines in the exchanger that dell with the call by using signaling lines.

In general these signaling lines may be within the speech lines that called Channel Associated Signal (CAS) or in assigned line (not used by speech channels) that called Common Channel Signaling (CCS).

Anyway the signals between the exchangers can be classified according to the direction of transferring as:-

1 – Forward signals:- these signals transferred from the asking parity exchanger (outgoing side) to the called party exchanger (incoming side).

2 – Backward Signals:- these signals transferred from the called parity (incoming side) to the asking party exchanger (outgoing side).



However these forward and backward signals can be classified according to the state of these signals into two types

1 – Line signals:- that represented by a DC signals or AC signals and dell with the seizing the lines and clear the lines and other orders for lines.

2 – Registering signals:- that represented in digital form and dell with the information (party numbers, state and other), this data can be storage in memories and registers.

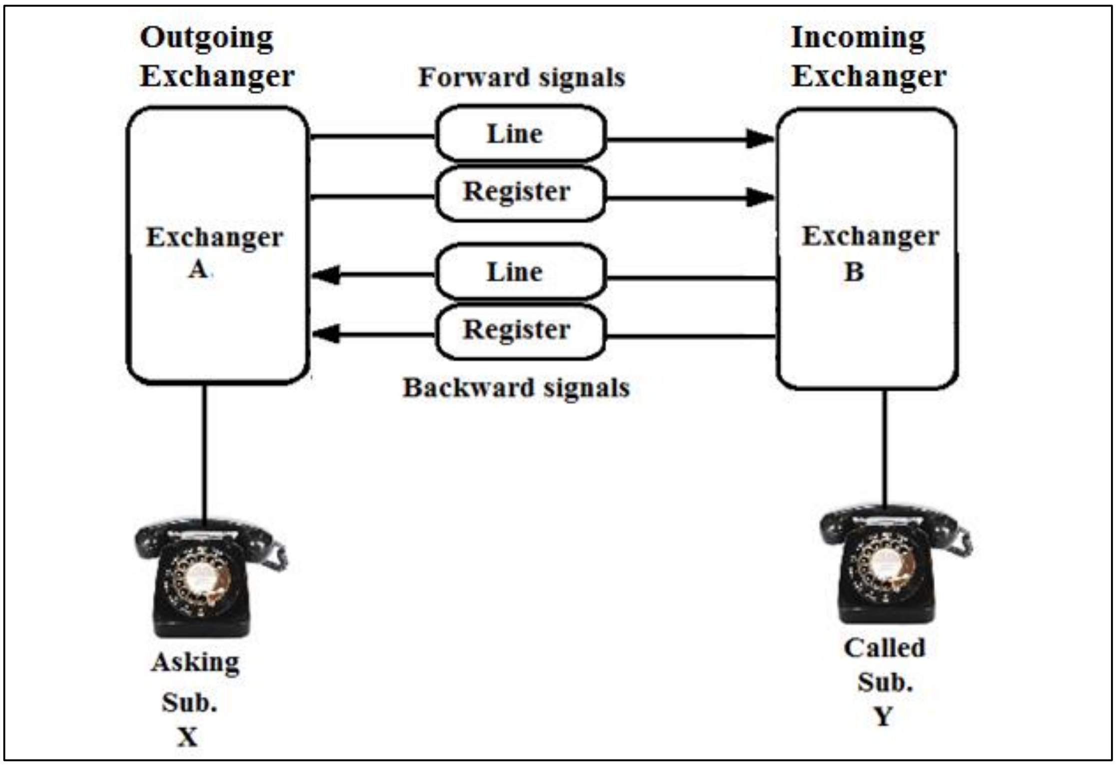
When we take in consideration the above classification modes we can estimates four type of signals between the exchngers:-

1 – Forward Line signals.

2 – Backward Line signals.

3 – Forward Register signals.

4 – Backward Register signals.



1- ***Forward line signals:-***

Forward line signals send from the outgoing side to incoming side (from the exchanger that connected to the asking party to the exchanger that connected to the called party), all of these signals dell with the line state and control. These signals are ;-

1 – **Seizing signal**:- send at the beginning of call to indicate the transition of the circuit at the incoming end from idle to seize state.

2 – **Clear forward signal:-** signal send to incoming end to terminal the call or call attempt.

3 – **Trunk offering signal:-** send by the operator to enter the connection due to new call (the called party not connected to this exchanger but with exchanger in link with it).

4 – **Trunk disconnect signal:-** send by the operator from the outgoing side to inform the incoming side to release the connection after end of call.

5 – **Ringing signal:-** signal send to the incoming end from the exchanger that connected to the asking subscriber, after ending the dialing process and

recognition of the called party number, in order to inform the exchanger that connected to the called subscriber to send ring signal to called subscriber.

2 - Backward line signals

These signals send by the incoming side to the outgoing side to inform it of the line state and the response of the forward signals. These signals are:-

1 – **Seize acknowledge:-** send by the incoming exchanger to the outgoing exchanger as a response of seize signal and to inform the outgoing side the transition of the incoming circuit from idle to seize state.

2 – **Answer Signal:-** signal send by the incoming exchanger to the outgoing exchanger to infirm the outgoing exchanger that the calld party has answer the call in order to :-

A – Start speech link

B – Start metering for cost calculation

C – Start measuring of the duration of call for statistic international purpose.

3 – **Clear back signal:-** signal send by the incoming exchanger to indicate the outgoing exchanger that the called party end the call.

4 – **Release Guard signal:-** signal send to the outgoing exchanger fom the incoming exchanger after the outgoing exchanger send clear forward signal to indicate that the leter has been fully effected and the relative switching circuits have been translate from seize to idle case.

5 – **Blocking Signal:-** signal send by the incoming exchanger as a response of the seize signal if there are no line in idle case in the incoming exchanger(refuse the seize order).

Communication Systems

Telephony System

Lecture (8-1):-

Signaling in Telephony system (2)
Signals Flow graph for telephone call.
Exchangers Classifications

Signaling in Telephony system:-

1 – Signals between the subscriber and the exchanger

2 – Signals within the exchanger

3 – Signals between the exchangers

1 – Forward Line signals.

2 – Backward Line signals.

3 – Forward Register signals.

4 – Backward Register signals.

1- Forward line signals:-

1 – Seizing signal

2 – Clear forward signal

3 – Trunk offering signal

4 – Trunk disconnect signal

5 – Ringing signal

2 - Backward line signals

1 – Seize acknowledge

2 – Answer Signal

3 – Clear back signal

4 – Release Guard signal

5 – Blocking Signal

3 – Forward Register signals:-

Forward register signals send from the outgoing side to incoming side (from the exchanger that connected to the asking (calling) party to the exchanger that connected to the called party), all of these signals dell with the information of the (parties, numbers, state and other). These signals are :-

1 – Address signal:- This include the information about the called party number.

2 – Calling party category signal:- This signal send the information of the calling party state (has special service or priority).

3 – Redirect call:- This signal send when the calling parity want to re contact the call after he end the call.

4- Backward register signals:-

1 – Signal requesting transmission of address signal:- signal send by the incoming exchanger as a request to send the next digit of the called subscriber number, this signal send after the outgoing exchanger send the first digit of the called subscriber number and so on to last digit of the called number.

2 – Signal requesting information about the calling party:- signal send to the outgoing exchanger as a request for the information of the calling subscriber.

3 – Congestion signal :- send by the incoming exchanger to inform the outgoing exchanger that there are no idle line offered in it (all the line in seize case).

4 – Address complete signal. This signal send by the incoming exchanger after the outgoing exchanger send the last digit of the called subscriber

by the outgoing exchanger to indicate it that the incoming exchanger received all the called subscriber number.

5 – Condition of called subscriber:- signal send to the outgoing exchanger to inform it that the called party is in one of the following state:-

A – Special tone to indicate that the called subscriber can not be reached to regions not coverd.

B – Called subscriber busy.

C – Un allocated number.

D – Called subscriber line free and charge.

E - Called subscriber line free but not charge.

F - Called subscriber line not of order.

6 – Information service:- information signal send from the incoming exchanger to inform the outgoing exchanger about the external lines state (in service idle or under maintenance)

Signals flow graph for telephone call.

The following graph represent the signals flow graph for the telephone call that subscriber X which connected to exchanger A is the calling party and subscriber Y which connected to the exchanger B is the called party in three cases

1 – Subscriber Y end the call and re contact the call

2 – Subscriber X end the call.

3 – Subscriber Y end the call.

Notes:-

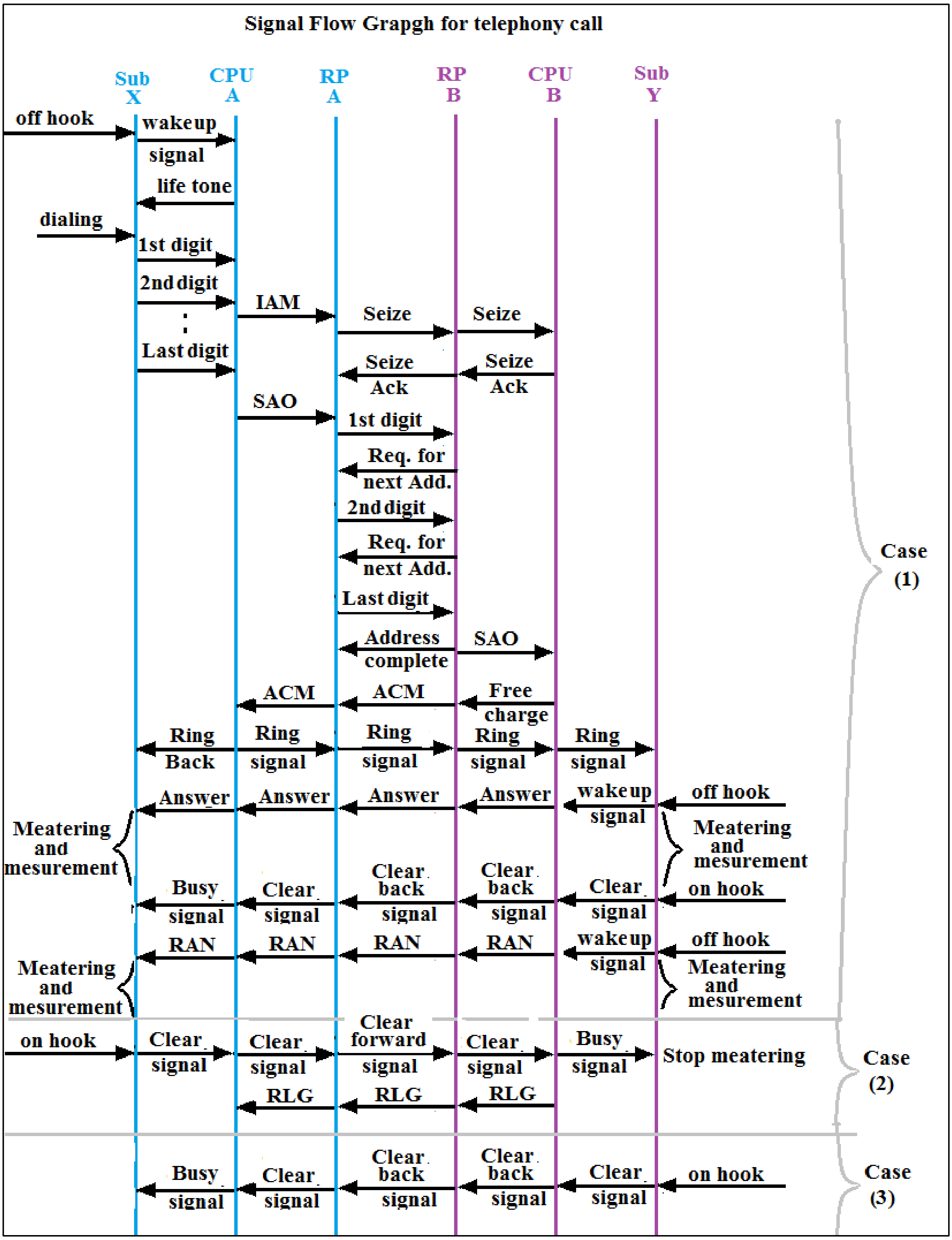
IAM = Initial Address Message.

SAO = Subscriber Address Message.

ACM = Address Condition Message.

RAN = Re-answer

RLG = Released guard



Signal Flow Grapgh for telephony call
Sub X
CPU A
RP A
RP B
CPU B
Sub Y
off hook
wakeup signal
life tone
dialing
1st digit
2nd digit
IAM
Seize
Seize
Last digit
Seize Ack
Seize Ack
SAO
1st digit
Req. for next Add.
2nd digit
Req. for next Add.
Last digit
Address complete
SAO
ACM
ACM
Free charge
Ring Back
Ring signal
Ring signal
Ring signal
Ring signal
Answer
Answer
Answer
Answer
wakeup signal
off hook
Meatering and mesurement
Meatering and mesurement
Busy signal
Clear signal
Clear back signal
Clear back signal
Clear signal
on hook
RAN
RAN
RAN
RAN
wakeup signal
off hook
Meatering and mesurement
Meatering and mesurement
Case (1)
on hook
Clear signal
Clear signal
Clear forward signal
Clear signal
Busy signal
Stop meatering
RLG
RLG
RLG
Case (2)
Busy signal
Clear signal
Clear back signal
Clear back signal
Clear signal
on hook
Case (3)

Signaling techniques:-

There are two techniques used for telephony signaling

- - Channel Associated Signaling (CAS).
- - Common Channel Signaling (CCS).

In the CAS system one or more voice channels are occupied by the signaling channel as illustrated in the CCITT framing paragraph, this system known as signaling system No.5 R1, R2. While the CCS signaling system transfer the signals between the exchangers by specified channel, this system used in North American and known as SS7 signaling system. In the last year more countries used the CCS signaling system because :-

1 – The signaling capacity in CCS is significantly higher than in CAS.

2 – Additional speech channels are valid in CCS system

3 – Information not related to speech channel could be transmitted between the exchangers.

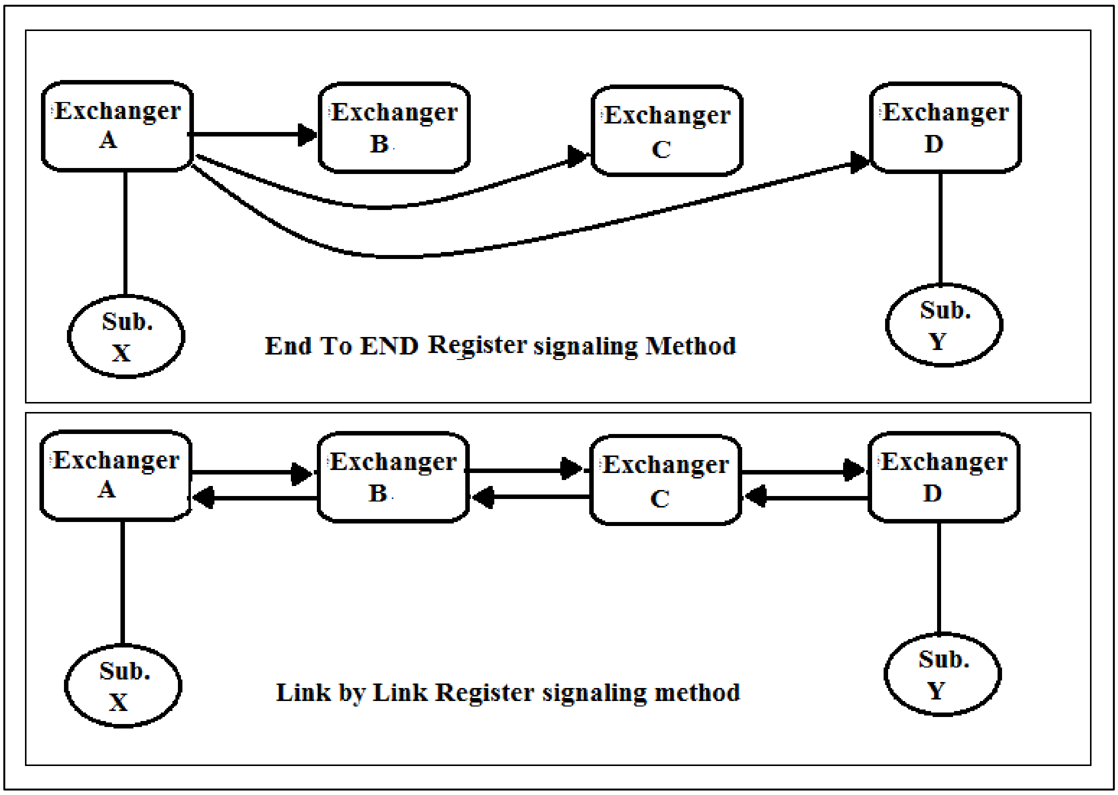
4 – Speech channels used more efficiently.

Register signals transfer methods

There are two types of transmitting register signals between the exchangers:-

- - Link by link method
- - End to End method

In End to End method the exchanger that connected to the calling subscriber make all connections to provides the link with all the exchangers that where within the call link. While in the Link by Link method the exchanger that connected to the calling subscriber make the link with next exchanger only and this exchanger make the link with the next one and so on.



Signaling States ;-

Signals that transferred between the exchangers can be represented in the following state

1 – Continues electrical signals (DC) :- like seize and clear signals

2 – Alternative electrical signal (AC) within the voice frequency:- like

Dial signal (440Hz + 350Hz continuous)

Busy signal (620Hz + 480Hz 0.5Sec on 0.5 Sec off)

Ring back signal (480Hz+440Hz 2 Sec on 4 Sec off),

Congestion signal (620Hz +480Hz 0.2Sec on o.3Sec off)

Recall signal (620Hz +480 0.3Sec on 0.2 Sec off)

3 – Digital signal. Many form of digital signals are used according to signaling type system (R1,R2,D1,D2, etc) but in all of these systems the digital message code consist of 4 bits only, the signals represented by the code value and the duration of appearance on the line for example :-

Signals	Forward	Backward
Idle	100	100
Seize	000	
Seize ACk		110
Answer		010
Clear forward	100	
Blocking		110

…………………………………………………………………..

Exchangers classifications

Exchangers can be classification according to them jobs as

- - Local Exchangers :- The main job for these exchangers are to achieve the speech call between the subscribers.
- - Transit Exchangers :- the main job for these exchangers are to achieve the link between the different Exchangers.

The above two types of exchangers can be execute the two jobs but the maine job of each exchanger specify it type.

In the ITU organization they classify the exchangers according to them levels as

1 – Local exchangers((LE)

2 – Primary Center   (PC)

3 – Secondary Center   (SC)

4 – Tertiary Center   (TC)

5 – Quaternary Center   (QC)

6 – International Center   (IC)

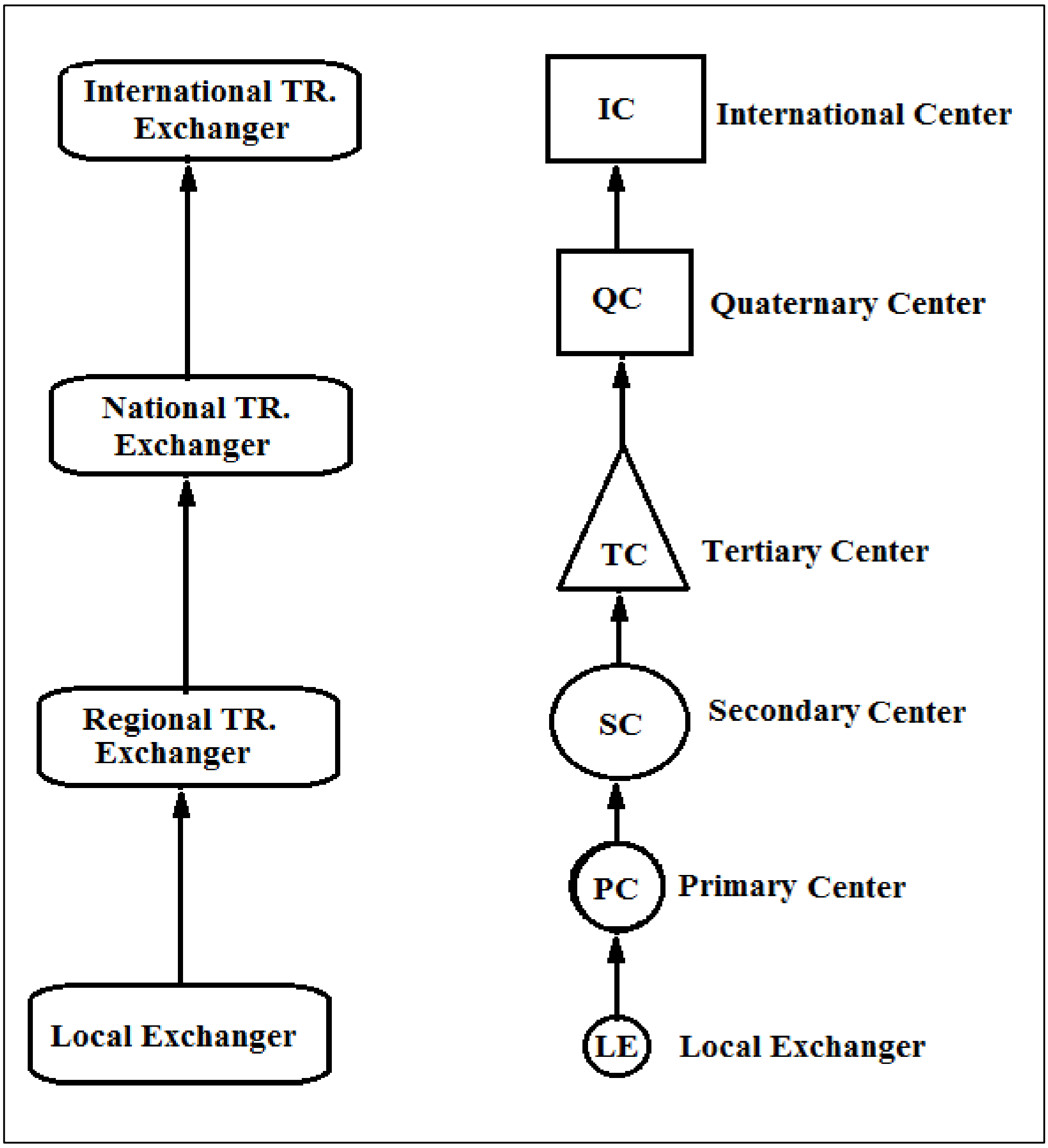
The telephony system practically achieve four levels of exchangers only that are:-

1 – Local Exchangers

2 – Regional Transit Exchangers

3 – National Transit Exchangers

4 – International Transit Exchangers.


International TR. Exchanger
National TR. Exchanger
Regional TR. Exchanger
Local Exchanger
IC
International Center
QC
Quaternary Center
TC
Tertiary Center
SC
Secondary Center
PC
Primary Center
LE
Local Exchanger

Communication Systems

Telephony System

Lecture (9-1):-

A network in Telephony

or Exchanger Network

A network in Telephony

A network in telephony may be defined as a method of connecting exchanges so that any one subscriber in the network can communicate with any other subscriber.

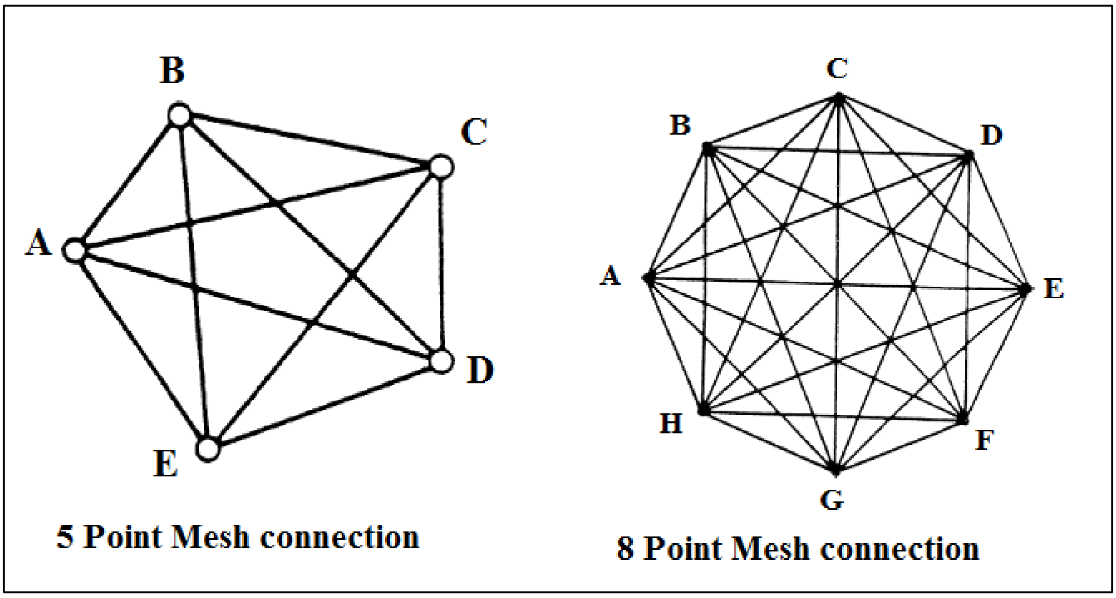
Let us assume that subscribers access the network by a nearby local exchange. Thus the problem is essentially how to connect exchanges efficiently. There are three basic methods of connection in conventional telephony:

(1) Mesh,

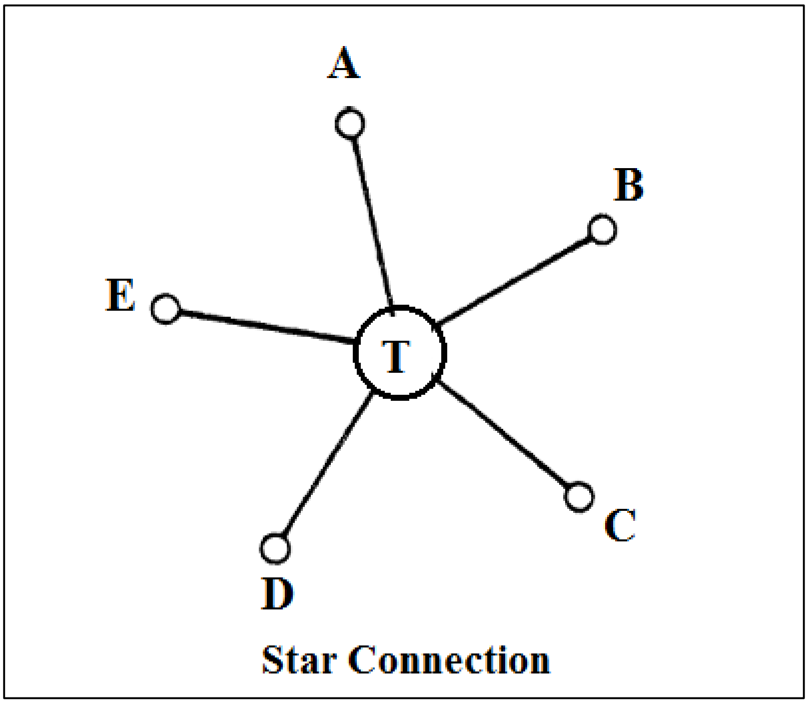
(2) Star.

(3) Double and higher-order star

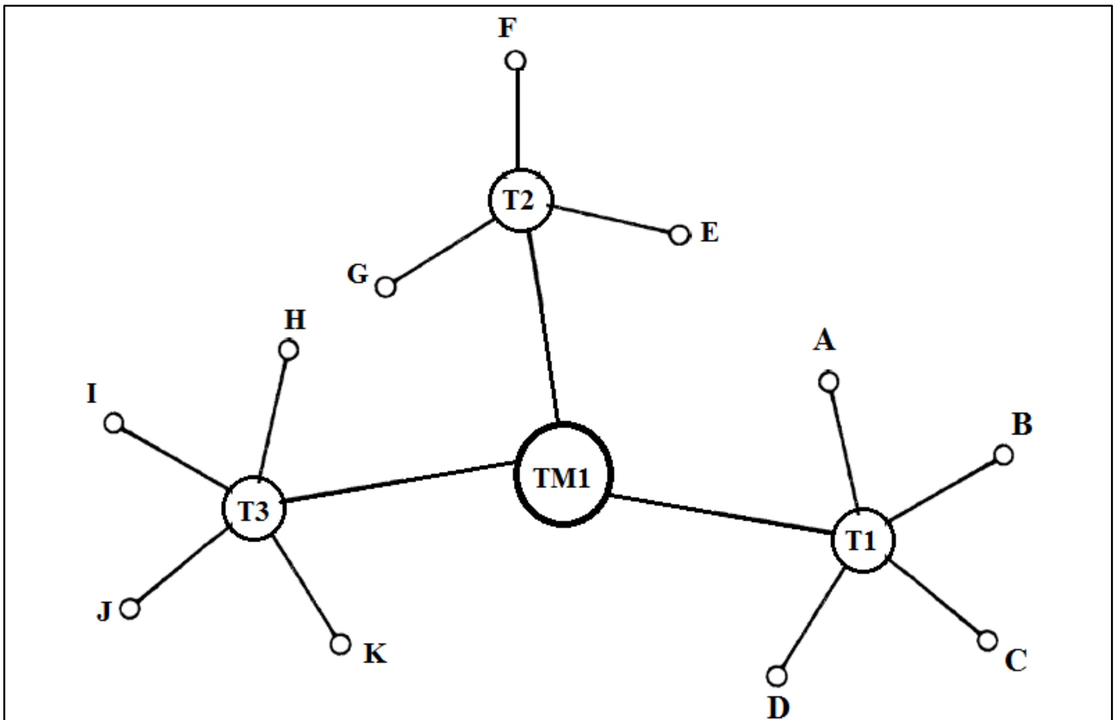
The mesh connection is one in which each and every exchange is connected by trunks (or junctions) to each and every other exchange as shown rein Figure bellow



Star connection utilizes an intervening exchange, called a *tandem exchange*, such that each and every exchange is interconnected via a *single* tandem exchange. An example of a star connection is shown in Figure



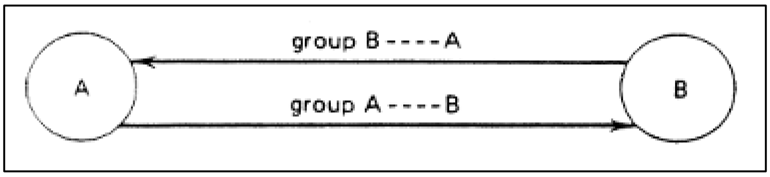
double-star configuration is one where sets of pure star sub networks are connected via higher-order tandem exchanges, and this trend can be carried still further, as shown in Figure



Double Star Connection

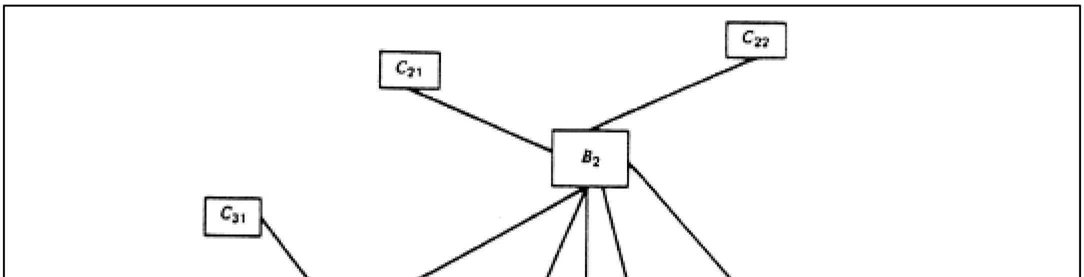
As a general rule we can say that mesh connections are used when there are comparatively high traffic levels between exchanges, such as in metropolitan networks. On the other hand, a star network may be applied when traffic levels are comparatively low.

Another factor that leads to star and multiple-star network configurations is network complexity in the trunking outlets (and inlets) of a switch in a full mesh. For instance, an area with 20 exchanges would require 380 traffic groups (or links), and an area with 100 exchanges would require 9900 traffic groups. This assumes what are called *one-way groups*. A one-way group is best defined considering the connection between two exchanges, A and B. Traffic originating at A bound for B is carried in one group and the traffic originating at B bound for A is carried in another group, as shown in the following diagram:

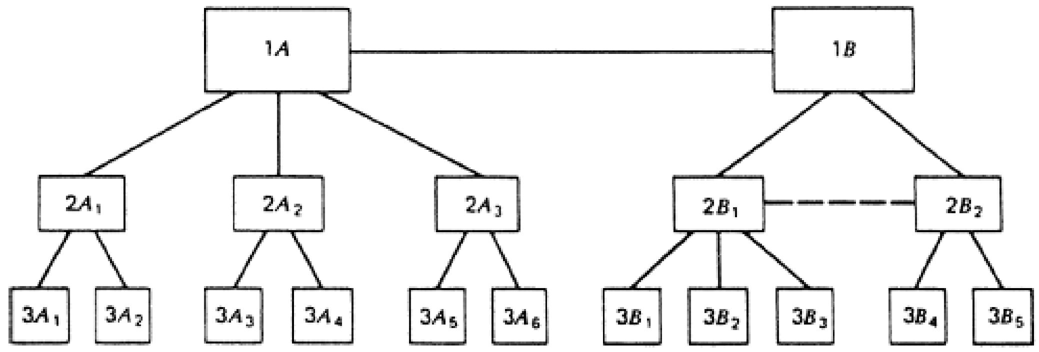


Thus, in practice, most networks are compromises between mesh and star configurations. For instance, outlying suburban exchanges may be connected to a nearby major exchange in the central metropolitan area. This exchange may serve nearby subscribers and be connected in mesh to other large exchanges in the city proper.

Another example is the city's long-distance exchange, which is a tandem exchange looking into the national long-distance network, whereas the major exchanges in the city are connected to it in mesh. An example of a real life compromise among mesh, star, and multiple-star configurations is shown



The figure bellow illustrate the several levels in a high-order star network



High - order star network

There are three levels or ranks of exchanges in the figure. The smallest blocks in the diagram are the lowest-ranked exchanges, which have been marked with a "3" to indicate the third level or rank. Note that there are restrictions or rules of traffic flow. As the figure is drawn, traffic from 3A1 to 3A2 would have to flow through exchange 2A1 . Likewise, traffic from exchange 2A2 to 2A3 would have to flow through exchange 1A.

Carrying the concept one step further, traffic from any A exchange to any B exchange would necessarily have to be routed through exchange 1A.

The next consideration is the **high-usage** (HU) route. For instance, if we found that there were high traffic intensities (e.g., >20 Erlangs) between 2B trunks and switch gear might well be saved by establishing a HU route between the two (shown by a dashed line in Figure above). Thus we might call the high usage route a *highly traveled shortcut*. Of course, HU routes could be established between any pair of exchanges in the network if traffic intensities and distances involved proved this strategy economical. When HU routes are established, traffic between the exchanges involved will first be offered to the HU route, and very flow would take place through a last choice route or, as shown in Figure above, up to the next level and down. If routing is through the highest level of higher-order star network, we call this route the final route, a hierarchical network term.

Communication Systems

Telephony System

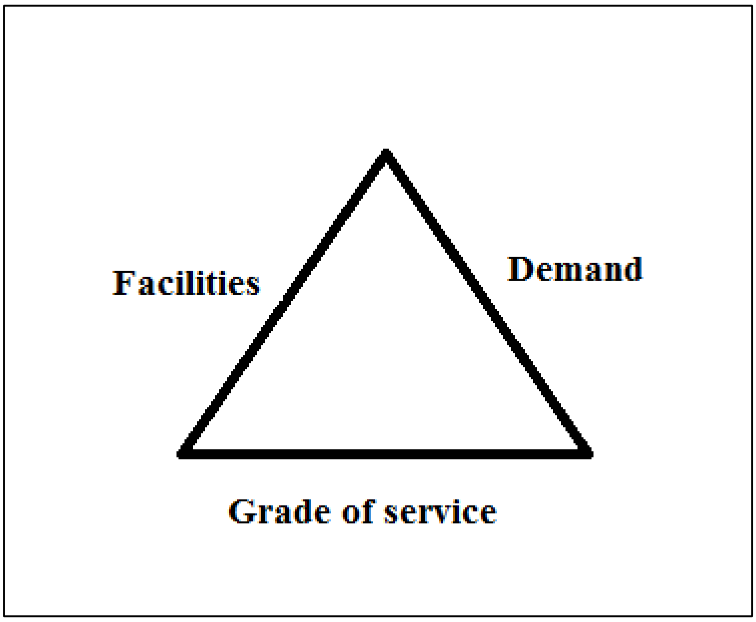
Lecture (10-1):-

Telephony traffic

Telephone traffic engineering, or traffic engineering is the application of traffic engineering theory to telecommunications (telephone). Telephone traffic engineers use their knowledge of statistics including queuing theory, the nature of traffic, their practical models, their measurements and simulations to make predictions and to plan telecommunication networks such as a telephone network or the Internet. These tools and knowledge help provide reliable service at lower cost.

The telephone traffic also can be define as the process of occupancy or the busyness of the telephone link and the possibility of the network exchange during the process of conducting the link or through the calling process.

The telephone traffic can be presented by the three sides from the triangular the first side is the demand, the second is the facilities, and the third side is the grade of service as shown in figure.



The change of any side from the triangular need to change one from the other sides or the two sides.

In this lecture we shall study

- - The probability of offered call on first attempt.
- - The probability of lost call
- - The probability of meeting blockage

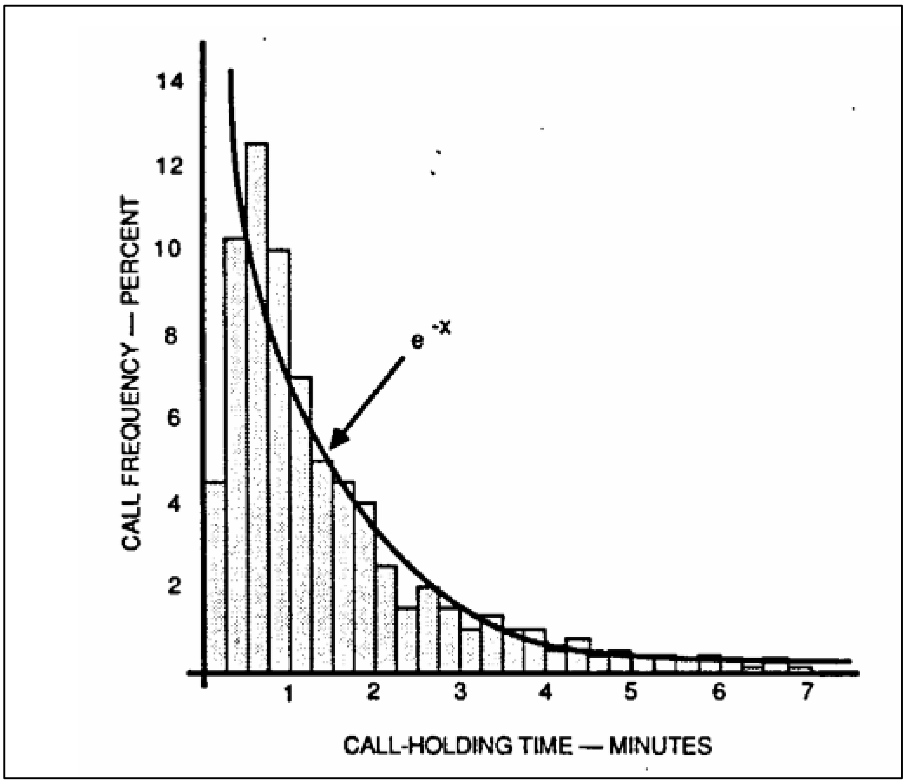
In order to reach the above objects we must study and understand some expressions:-

1 – **Call attempt** :- is the work get by the subscriber to get the telephone services, this attempt may be successful or not. So

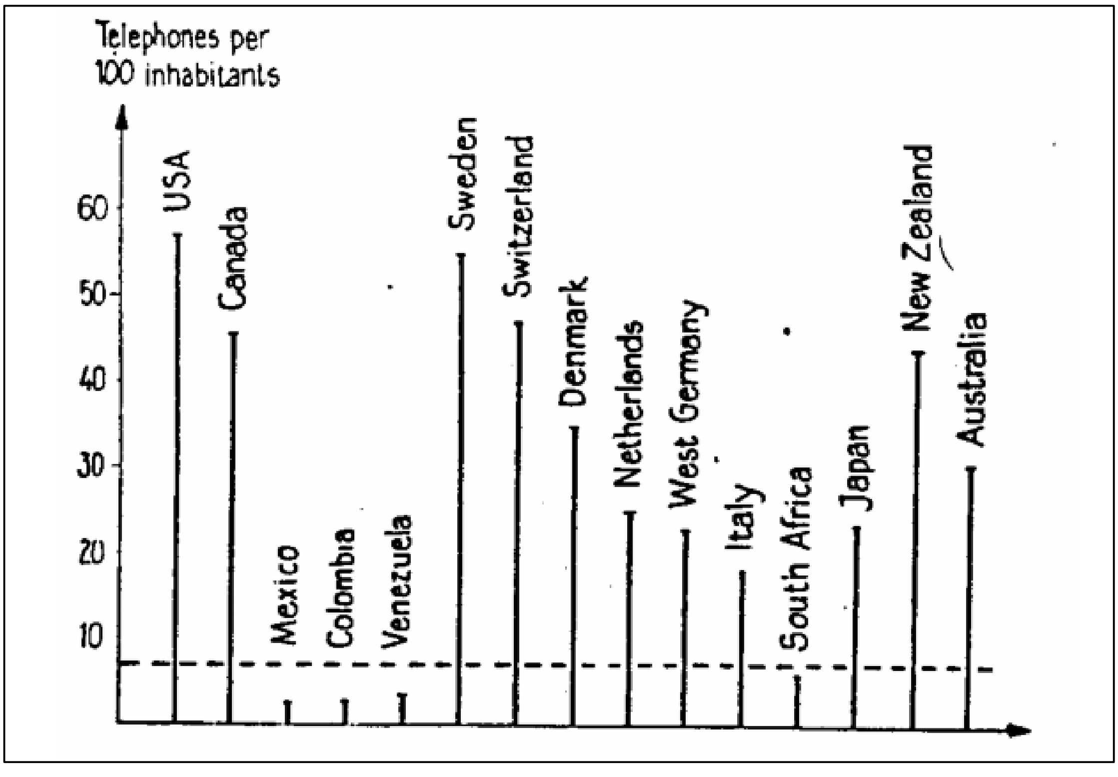
Successful call attempts < or = call attempts.

2 – **Call** :- is the series of attempts to reach the telephone line by dialing the specific calling number, the last attempt ether successful or fail, the number of successful calls < the calls

3 – **Call holding time:-** Is the time that include the connected the subscriber through the telephone channel (the time between the answer signal and clear signal). The figure bellow show the call holding time in some city.



4 – **Telephone density**:- The number of telephone sets per (100) living persons. The telephone density in different countries are shown in the figure bellow:-



5 – **Traffic density**: - The number of simultaneous calls at a given moment.

6 – **Traffic intensity**:- The average traffic density during a 1-h period. The quantity of traffic used in the calculation for dimensioning of switches is the traffic intensity.

7 – **Offered traffic**: - Is the traffic density that the circuits of the exchanger can dealing with it.

8 – **Carried traffic**: - The number of calls that seizing telephone channels.

9 – **Block traffic**:- The number of calls that cannot seize telephone channels.

Block traffic = offered traffic – carried traffic

10 - **Busy hour** :- Is the 60 minutes from the day that carry the maximum telephony traffic. Its one of the traffic variation that depend on

A – Subscriber category.

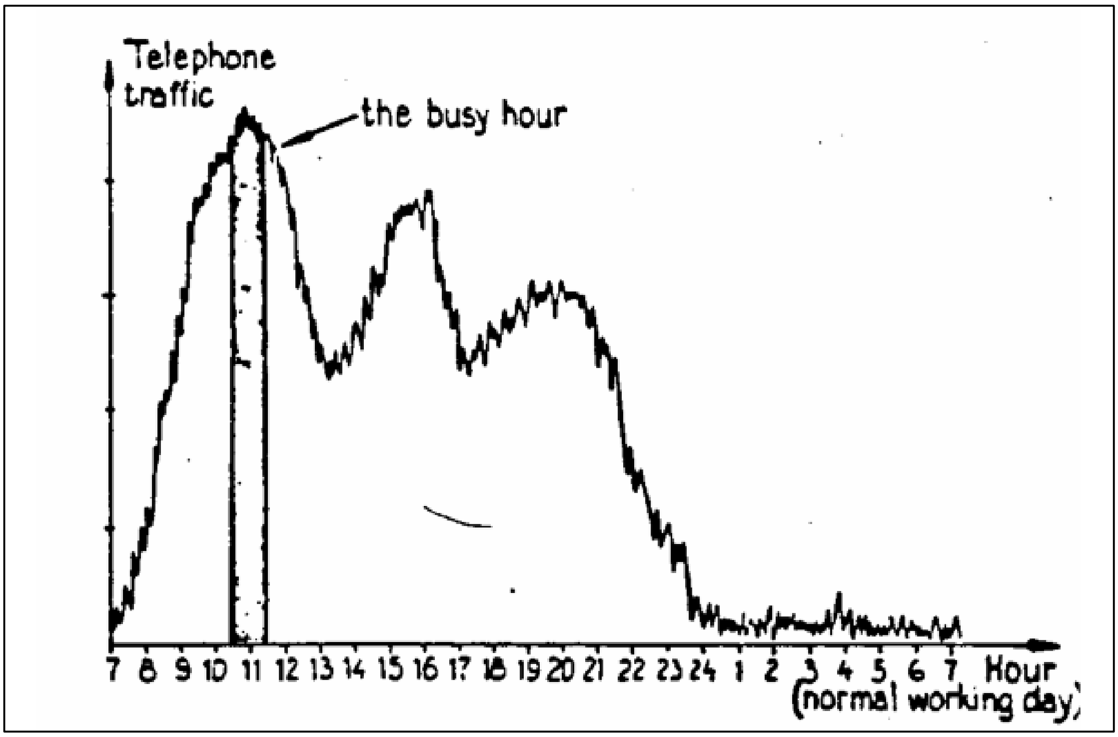
B – Time of the day.

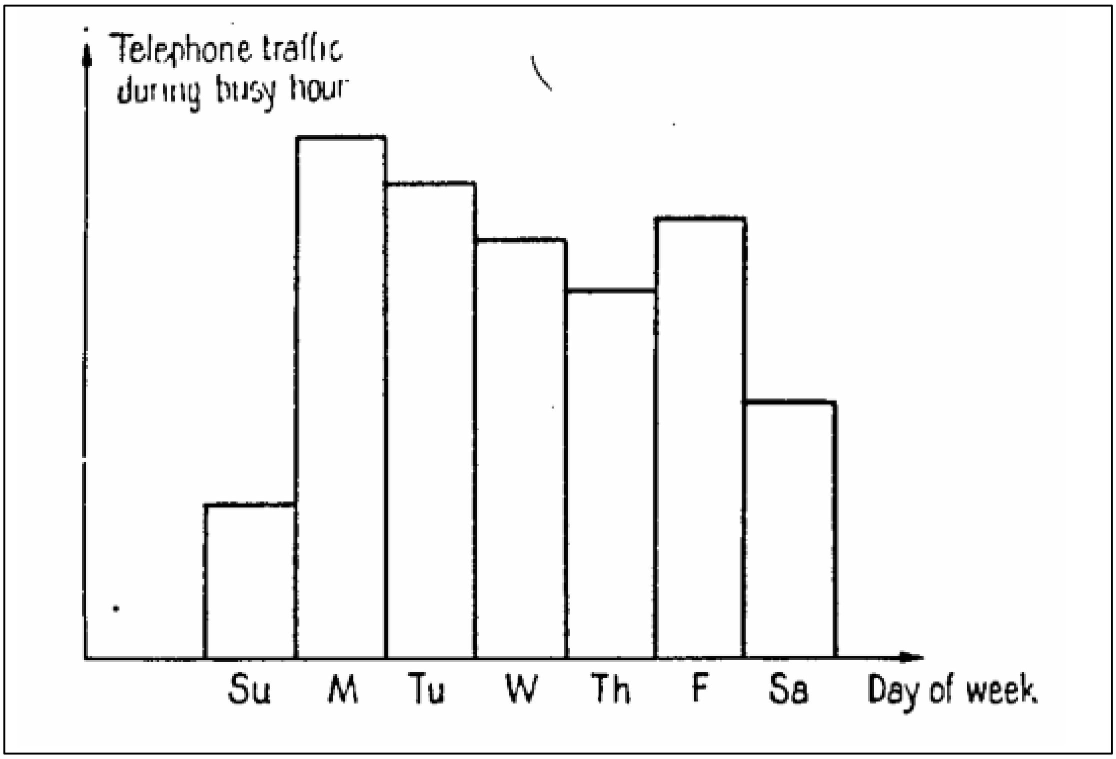
C - Month of the year.

D – Holiday.

E – Price of service.

The two figures bellow illustrate the busy hour of telephone traffic through one day and traffic distribution in different day of week in specific city.


Telephone traffic
the busy hour
7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 1 2 3 4 5 6 7 Hour
(normal working day)


Telephone traffic during busy hour
Su M Tu W Th F Sa Day of week

11 – **Congestion: -** The last of call cannot be linked because of all of the connection groups or switches group are in occupancy case.

12 – **Traffic flow**:- it is the results of multiply the number of call by the average of call holding time at certain time.

Traffic flow = number of call * average call holding time

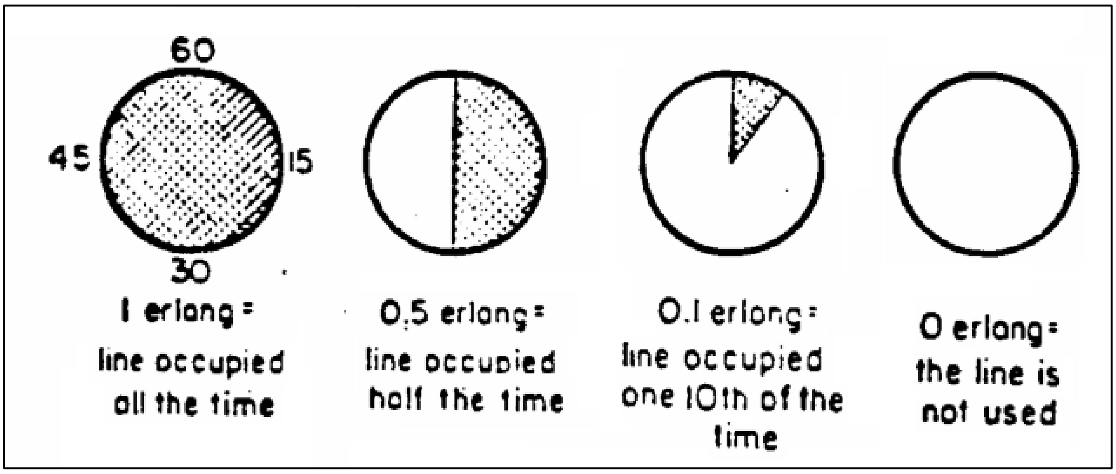
The traffic flow can be measured using

A – Erlang :

B – Centrum Call Second (CCS)

C – Equated Busy Hour Call (EBHC)

- – Erlang : - one Erlang is the occupancy of the one telephone line in average holding time equal one hour.



- - Centrum Call Second CCS : - one CCS is the occupancy of the one telephone line in average holding time equal 100 Sec.

1 Erlang = 36 CCS

- - Equated Busy Hour Call (EBHC):- one EBHC is the occupancy of the one telephone line in average holding time equal 2 minutes

1 Erlang = 30 EBCH = 36 CCS

Example: - an Exchanger with 5000 line capacity have average occupied call equal 0.6 during the busy hour and average holding time equal 250 second, find the traffic flow in CCS, Erlang and EBHC

Sol.: traffic flow = ((5000*0.6)*250)/100 = 7500 CCS

1 Erlang = 36 CCS   so the traffic flow = 7500/36 = 208.33 Erlang

1 Erlang = 30 EBHC   so the traffic flow = 208.33*30 = 6250 EBHC

Example :- a group of 20 subscriber achieved 50 call in average holding time = 200 Sec during the busy hour calculate the traffic flow in CCS and Erlang for each subscriber.

Sol:- over all traffic flow = (50 * 200) /100 = 100 CCS

Traffic flow for each subscriber = 100 / 20 = 5 CCS

1 Erlang = 36 CCS

Traffic flow for each subscriber = 5/36 = 0.139 Erlang

13 – **Grade of Services** :- the probability of meeting blockage during the busy hour.

$$Grade\ of\ service = \frac{Number\ of\ lost\ call}{Total\ number\ of\ offerd\ call}$$

Example:- in specific Exchanger there are 354 seizing line and 6 lost call in busy hour, find the grade of service?

$$Grade\ of\ service = \frac{Number\ of\ lost\ call}{Total\ number\ of\ offerd\ call}$$

$$Grade\ of\ service = \frac{Number\ of\ lost\ call}{Carried\ traffic + Lost\ traffic}$$

$$Grade\ of\ service = \frac{6}{6+354} = 0.017$$

The probability of finding available trunk in the first attempt from the offered calls.

This depend on :-

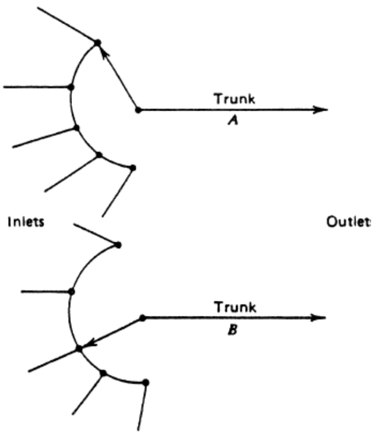
1 – The distribution in time and the duration of offered traffic.

2 – The number of traffic sources.

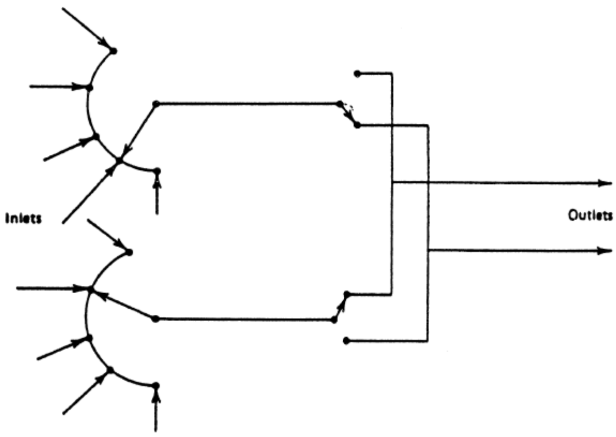
3 – The availability of trunks in group of traffic sources

A – Limit availability

B – Flexible or Full availability



limited availability.



full availability.

4 – The manner in which the lost call are handled.

A –Lost Call Held (LCH) The LCH concept assumes that the telephone user will immediately reattempt the call on receipt of a congestion signal and will continue to redial.

B – Lost Call Clear (LCC). The LCC concept, assumes that the user will hang up and wait some time interval before reattempting if the user hears the congestion signal on the first attempt.

C – Lost Call Delayed (LCD). The LCD concept assumes that the user is automatically put in queue (a waiting line or pool). For example, this is done when the operator is dialed.

Probability-Distribution Curves

Telephone-call originations in any particular area are random in nature. We find that originating calls or call arrivals at an exchange closely fit a family of probability-distribution curves following a Poisson distribution. The **Poisson** distribution is fundamental to traffic theory.

Most of the common probability-distribution curves are two-parameter curves. That is, they may be described by two parameters, (***mean and variance***).

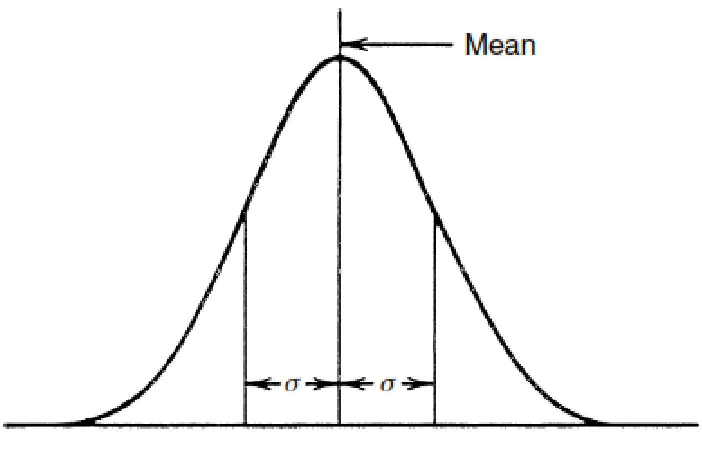
The **mean** is a point on the probability-distribution curve where an equal number of events occur to the right of the point and to the left of the point. *Mean* is synonymous with *average*. We define mean as the x-coordinate of the center of the area under the probability-density curve for the population. The lowercase Greek letter mu (μ) is the traditional indication of the mean; x is also used.

The second parameter used to describe a distribution curve is the **dispersion**, which tells us how the values or population are dispersed about the center or mean of the curve. There are several measures of dispersion. One is the familiar standard deviation, where the standard deviation σ of a sample of n observations

$x_1, x_2, \ldots, x_n$ is

$$\sigma = \sqrt{\frac{1}{n-1}\sum_{i=1}^{n}(x_i - \overline{x})^2}$$

The *variance V* (σ^2)of the sample values is the square of σ. Mean and standard deviation of a normal distribution curve are shown



A normal distribution curve showing the mean and the standard deviation, σ.

We can see that σ^2 is another measure of dispersion, the variance, or essentially the average of the squares of the distances from mean aside from the factor n/(n − 1).

We have introduced two distribution functions describing the probability of distribution, often called the *distribution* of x or just f(x). Both functions are used in traffic engineering. But before proceeding, the variance-to-mean ratio (VMR) is introduced. Sometimes VMR (α) is called the *coefficient of over dispersion*. The formula for VMR is

$$\alpha = \frac{\sigma^2}{\mu}$$

Traffic probability distributions can be divided into three distinct categories:
(1) Smooth: - α is less than 1
(2) Rough: - α is greater than 1
(3) Random: - α = 1

Smooth traffic is characterized by a positive binomial distribution function, perhaps better known to traffic people as the *Bernoulli distribution*. If we assume that subscribers make calls independently of each other and that each has a probability p of being engaged in conversation, then if n subscribers are examined, the probability that x of them will be engaged is

$$B(x) = C_x^n p^x (1-p)^{n-x}, \qquad 0 < x < n$$

$$\text{Its mean} = np \qquad \text{Its variance} = np(1-p)$$

Where
B(x) :- is the probability of x call in progress.
P :- is the probability of each call be in progress
C_x^n Is number of ways that x entities can be taken n at a time.

Smooth traffic is assumed in dealing with small groups of subscribers; the number 200 is often used as the breakpoint.

<u>Rough traffic</u> This is characterized by binomial distribution with a negative index. Therefore, if the distribution parameters are k and q,

$$R'(x, k, q) = \binom{x+k-1}{k-1} q^x (1-q)^k$$

where
R' is the probability of finding x calls in progress for the parameters k and q.
k is a positive number representing a hypothetical number of traffic sources

q represents the occupancy per source and may vary between 0 and 1

Random traffic:-Is represented by the Poisson probability function which can be derived from the binomial distribution, assuming that the number of subscribers (s) is very large and the calling rate per line (h) is low such that the product sh = m remains constant and letting s increase to infinity in the limit

$$P(x) = \frac{m^x}{x!}e^{-m}$$

where $x = 0, 1, 2, \ldots$

The probability of blockage at switch due to congestion or all trunks busy.

The Erlang B loss formula has been widely used today outside of the United States. Loss here means the probability of blockage at the switch due to congestion or to "all trunks busy" (ATB). This is expressed as *grade of service* (E) or the probability of finding x channels busy. The other two factors in the Erlang B formula are the mean of the *offered* traffic and the number of trunks of servicing channels available. Thus

$$E_B = \frac{A^n/n!}{1 + A + A^2/2! + \cdots + A^n/n!}$$

where
n is the number of trunks or servicing channels
A is the mean of the offered traffic
E_B is the grade of service using the Erlang B formula.

This formula assumes the following:

- Traffic originates from an infinite number of sources.
- Lost calls are cleared assuming a zero holding time.
- The number of trunks or servicing channels is limited.
- Full availability exists.

The Poisson formula (the probability of lost call) for the following assumptions:

(1) Infinite sources.

(2) Equal traffic density per source.

(3) Lost calls held (LCH).

The formula is

$$P = e^{-A} \sum_{x=n}^{\infty} \frac{A^x}{x!}$$

While the Erlang B (Grade of service) formula assuming

(1) Infinite sources

(2) Equal traffic density per source

(3) Lost calls cleared (LCC).

The formula is

$$P = \frac{\frac{A^n}{n!}}{\sum_{x=0}^{n} \frac{A^x}{x!}}$$

The Erlang C formula, commonly used with digital switching where one would expect to find queues, assumes

(1) Infinite sources.

(2) Lost calls delayed (LCD).

(3) Exponential holding times.

(4) Calls served in order of arrival.

The formula is

$$P = \frac{\frac{A^n}{n!} \cdot \frac{n}{n-A}}{\sum_{x=0}^{n-1} \frac{A^x}{x!} + \frac{A^n}{n!} \frac{n}{n-A}}$$

The binomial formula assumes

(1) **<u>Finite</u>** sources.

(2) Equal traffic density per source.

(3) Lost calls held (LCH).

The formula is

$$P = \left(\frac{s-A}{s}\right)^{s-1} \sum_{x=n}^{s-1} \binom{s-1}{x} \left(\frac{A}{s-A}\right)^x$$

Communication Systems

Mobile Communication

Lecture (1-2) :- Introduction

After the wide successfulness of the telephony system of connecting the world by a huge network , the communication engineers was dream to provide telephone service for each person, that's can not be possible with wired communication system. After the radio (wireless) communication was came out, this dream approached to achievement. and this increased the hope of the communication engineers to make the call successful with movement of the subscriber, this leaded to appearance of the Mobile Communication.

Note :- Mobile communication is the communication system that carry call successfully with the movement in car speed, while the portable communication is the communication system that carry call successfully with the movement in human speed.

Historical review for the mobile communication systems.

A) - The first approach of the Pre-Prevailing Stage of mobile communication systems

- The first approach of the Pre-Prevailing Stage of mobile communication systems appeared after the second world war (1950 and beyond). The specifications of this system are :-

1 - only car telephone service

2 - Heavy, bulky and expensive equipment

3 - No handover capability

4 - Poor grade of service

5 - Low speech quality

6 - Low capacity

7 - High market saturation

8 - No frequency reuse

9 - Power level is not safe (very high)

10 - Power hungry transceivers

B– The first generation of mobile system (1G)

The firs generation of mobile system started to spreading at the end of 70s and start of 80s from the past century.

●-The main good spots of this system are :-

1 - Cellular system

2 - Frequency reuse

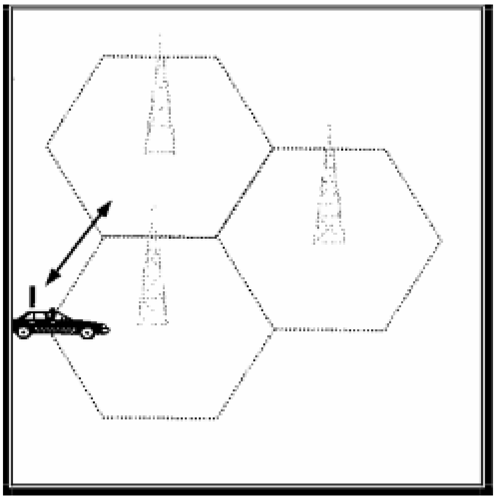
3 - Adaptive power control

4 - (Cell Sectorization)

5 - Cell splitting

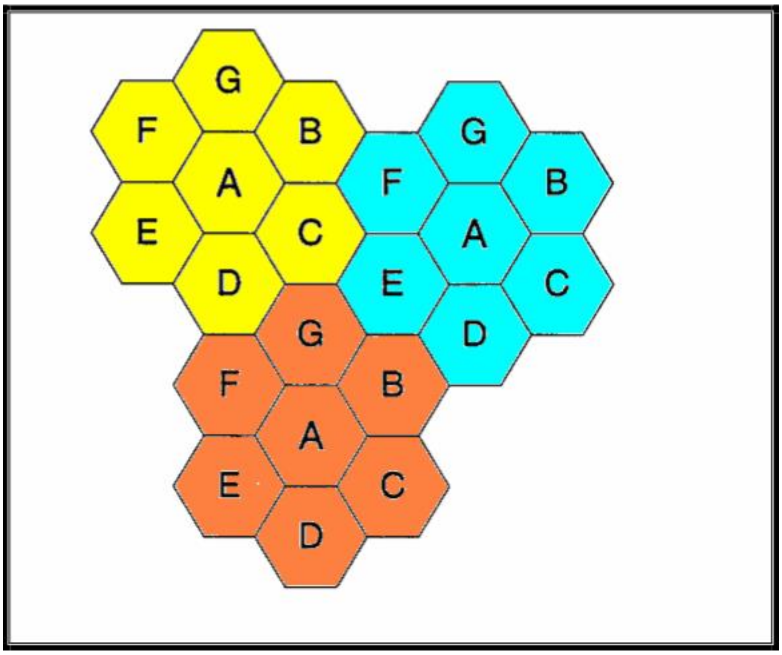
6 - Handover

1 - Cellular system :- This mean that the land that must be serviced by the mobile communication system are divided into sub band each sub band are served with specified sub communication system, these land cells can take different size and different form.



2 - Frequency reuse :- The frequency band which dedicated for the mobile communication system are divided into number of frequency sub band , each sub band contain number of channels called (frequency Group), each

frequency group serve the subscribers within one cell, the number of cells that consume the all frequency band called cluster, this cluster can be repeated in some way to increase the subscribers capacity.



3 - Adaptive power control :- The near subscriber from the base station tower transmit with low power radiation and the far subscriber from the base station tower transmit with high power radiation . and this applying for the down link also. This to increase the S/N ratio and increase the battery life.

4 - (Cell Sectorization) :- Each Cell are divided into multi sector, each sector served by group of antennas , to increase the S/N ratio.

5 - Cell splitting Each sector within the cell can be converted to new cell and rearrange the frequency groups according the new plan of clusters and cells.

6 – Handover :- :- when the subscriber travel from one cell to other during the call process they need to change the channel link because the old cell cannot serve him, this change of channel without disconnect the link and continue the call called (handover).

- **-The main bad spots of the first generation of mobile system are** :-

  1-- Depending on analog technique

  2 - Different operating frequency ranges

  3 – Incompatible (no standardization)

  4 - Suffer from capacity saturation

  5 - Limited to voice service

  6 - Insufficient transmission quality

  7 - No encryption

  8 - Frequency modulation (FM)

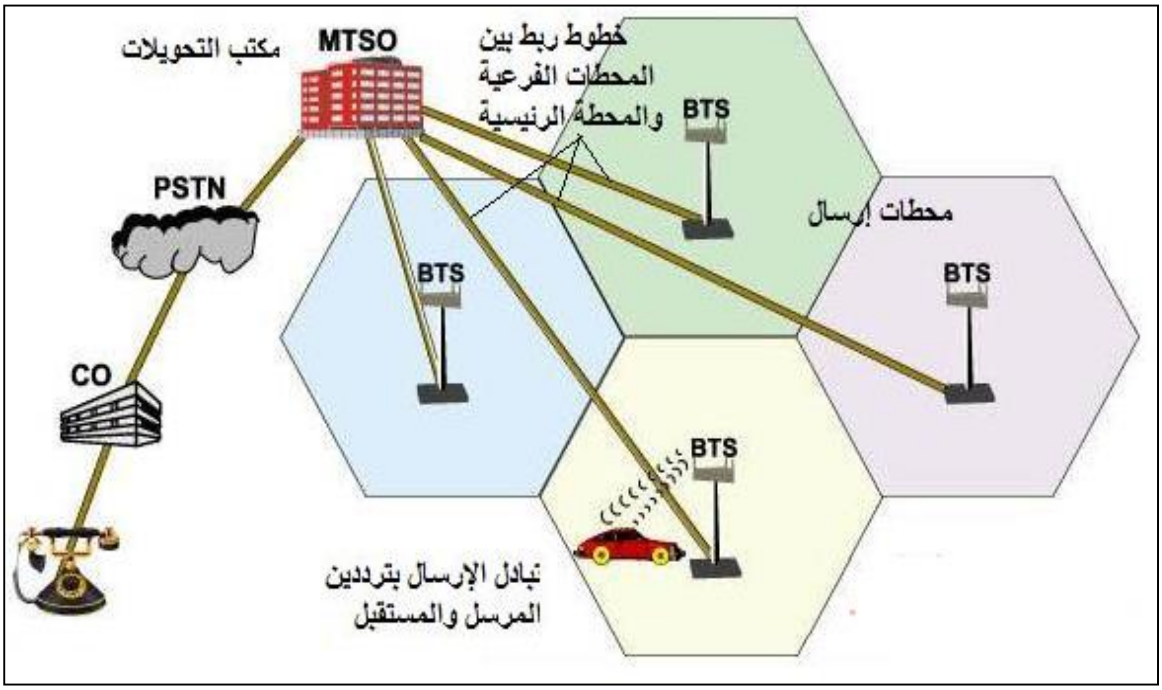
  9 - FDMA transmission technology only

Table (1-1) Illustrate the no standardization and the different operation frequency for the First Generation mobile communication systems.

سنة التشغيل Operational year	مدى التردد(ميجا هيرتز) Frequency range (MHz)	الدولة Country	اسم النظام System
1979	800	اليابان Japan	Nippon Telephone and Telegraphy (NTT-MTS)
1979	800	الولايات المتحدة US	Advanced Mobile Phone Service (AMPS)
1981-85	450 900	اسكندنافيا Scandinavia	Nordic Mobile Telephone (NMT)
1985	900	المملكة المتحدة UK	Total Access Communi. System (TACS)
1985	450	المانيا Germany	C450
1985 1989	450 900	فرنسا France	Radiocom 2000 (NMT)
1985 1990	450 900	ايطاليا Italy	RTMS TACS

- - Construction of the first generation mobile communication system.

The construction of the First generation of mobile system are shown in figure (3) below



We can see that this system consist from the three main parts :-

1 – Mobile Station

2 - Local Service Provider (LS)

3 - Mobile Telephone Switching Office (MTSO)

One example of the first generation is the Advanced Mobile Telephone System (AMPS) which is

1 - Use the frequency range (824MHz to 894MHz)

2 – The channel bandwidth for the speech signal is (30KHz)

3 - Each call need two channel one for the up link and the second for down link.

4 – There is a space guard equal (45MHz) between the up link and the down link

5 – Each local server provider consist of (395) mobile channel and (21) for PSTN with the mobile

6 – Each Local service provider consist of two BTS

C – The Second generation of mobile systems (2G).

After high development of the digital technique and increasing the request for the mobile system, the second generation of mobile system appeared in used at the end of the 80s and the first of the 90s from the past century, this system specified by the high quality services with the low cost of service because using the digital technique.

- - Objectives of the second generation mobile system.

1 – common standard.

The second mobile generation have (5) standard which are :-

a- The Pan-European digital cellular standard (Group Special Mobile) (GSM):

b- Electronic Association interim standard (IS-54) (American standard)

c -Interim Standard (IS-95) (American standard) (using CDMA)

d – JDC (japans standard) and personal handy phone system(PHS)

e – European wireless telephone service DECT,CT-2

2 – International Rooming

3 - Huge capacity

4 - Digital encryption techniques

5 - Noise and interference robust

6 - Enhanced range of services

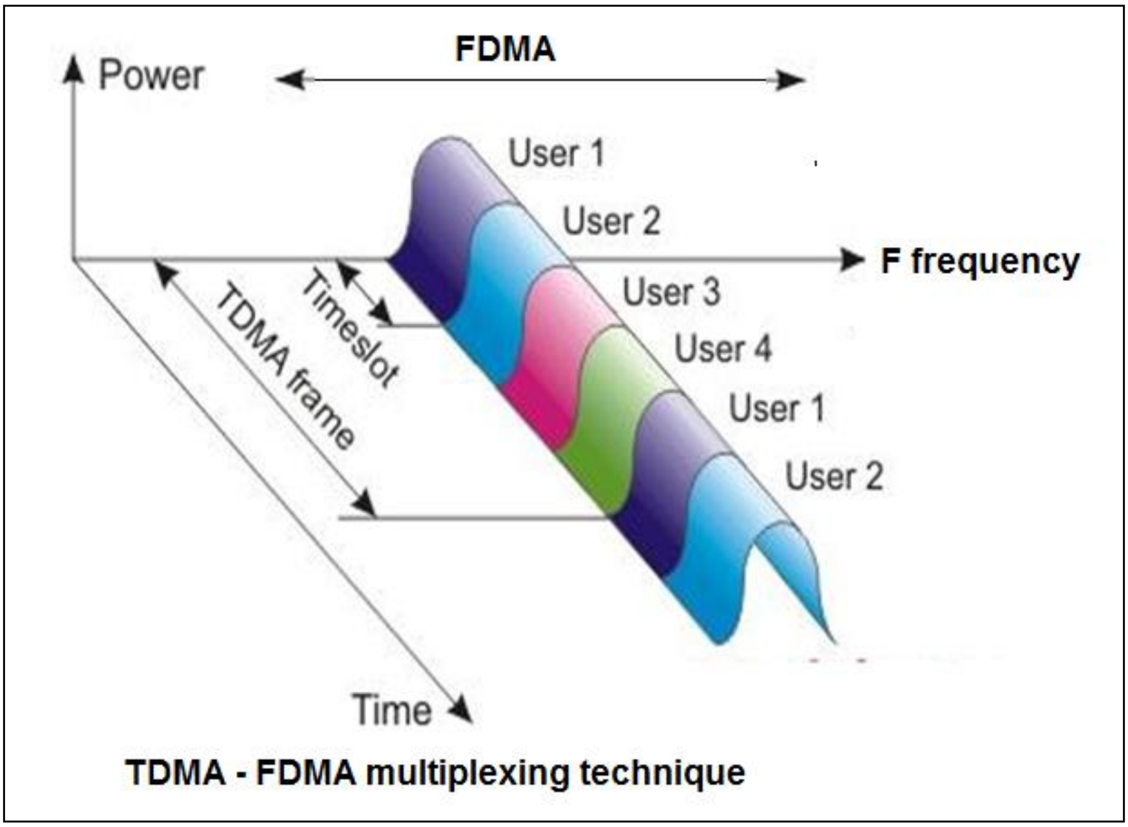
7 - Low cost equipment

8 - Low power consumption

9 - Lightweight, compact, pocket size terminals

10 - TDMA – FDMA digital transmission

11 - Integrated services digital network compatibility



- - Historical Review for the GSM systems

1982 Committee of European Post & Telecoms (CEPT) recommended 2×25MHz in 900 MHz

. 1982 Group special mobile (GSM) was established by the CPET

1987 Essential elements of wireless transmission are specified

1989 European Telecommunication Standards Institute took over the responsibility for GSM specifications

1990 The phase 1 GSM900 specifications are frozen

Adaptation to DCS 1800 commences

1991 First GSM networks lunched

1992 GSM has changed its name to the Global System for Mobile communications for marketing reasons

Most European GSM networks turn commercial

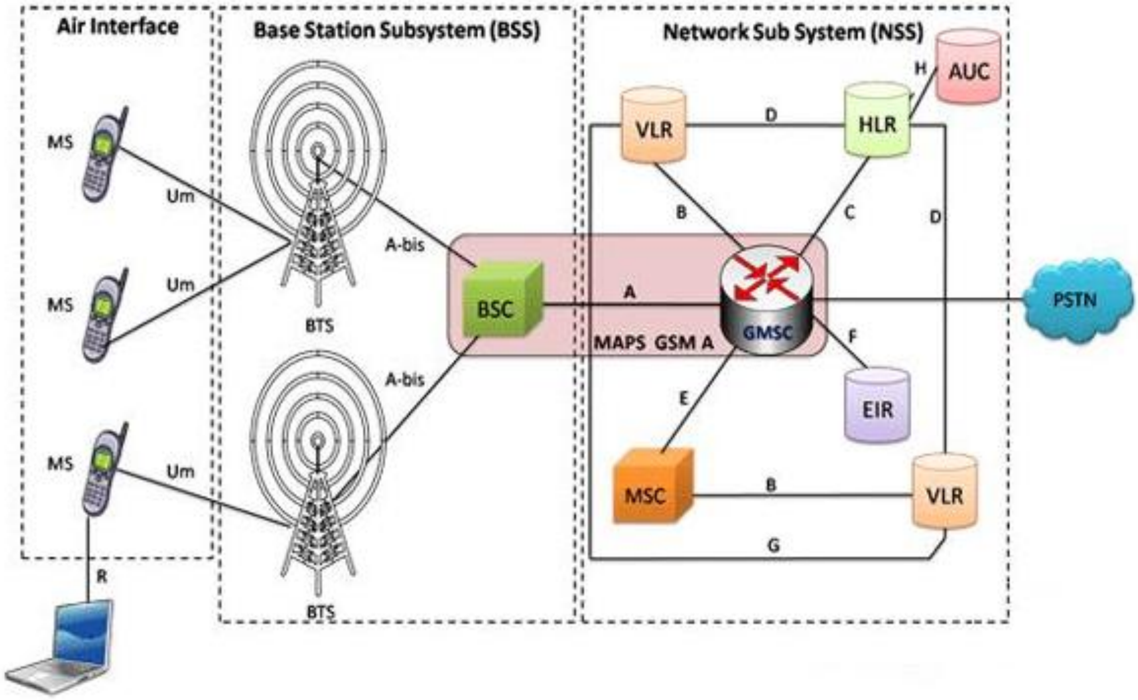
Some 13 networks in 7 countries are "on air" by the end of the year

1993   First roaming agreements in effect

By the end 1993, networks in 18 countries are operational

- - Construction of the GSM system

The construction of the GSM system are shown in figure (4) below



From figure (4) above we can see that the GSM system consist of the following :-

1 – Mobile station which consist of mobile equipment (ME) and Subscriber Identity Model card (SIM card)

2 – Base station system (BSS) consist of Base Transceiver system (BTS), Base station Control (BSC), antennas, Power supply and other component.

3 – Network sub system (NSS) consist of

* - Mobile switching system (MSC)

* - Home Location Register (HLR)

* - Visitor Location Register (VLR)

*- Equipment Identity register (EIR)

*- Authentication Center (AUC)

*- Operation and maintenance center (OMC)

*- Echo canceller unit (ECU)

* - Antennas and power supplies and other.

D) - the third generation of mobile system (3G).

The third generation of mobile appeared in Japan in mid 0f 2001, the 3G produce a high quality service toward the internet and data communication and give the transition toward the packet switch communication system and multimedia services.

- - Objectives of the third generation system.

1 - Global standard

3G produce two standard system which are :-

a- Universal Mobile Telecommunication system (UMTS)

b- International Mobile Telecommunications-2000 (IMT-2000) Conform to International Telecommunication Union. (ITU)

2 - Global roaming

3 - Multimedia services

4 - Unique universal handset

5 - Multiple environment (indoor, outdoor, and vehicular scenarios)

6 - Circuit and packet switching mode of services

- - Third Generation Specifications :-

1 – Voice and High speed data services (up to 14.4 Mbit/sec)

2 - Use (Code and Wideband Code) Division Multiple Access (CDMA and WCDMA)

3 - Multimedia Cell Smart Phone

4 - Easy internet access

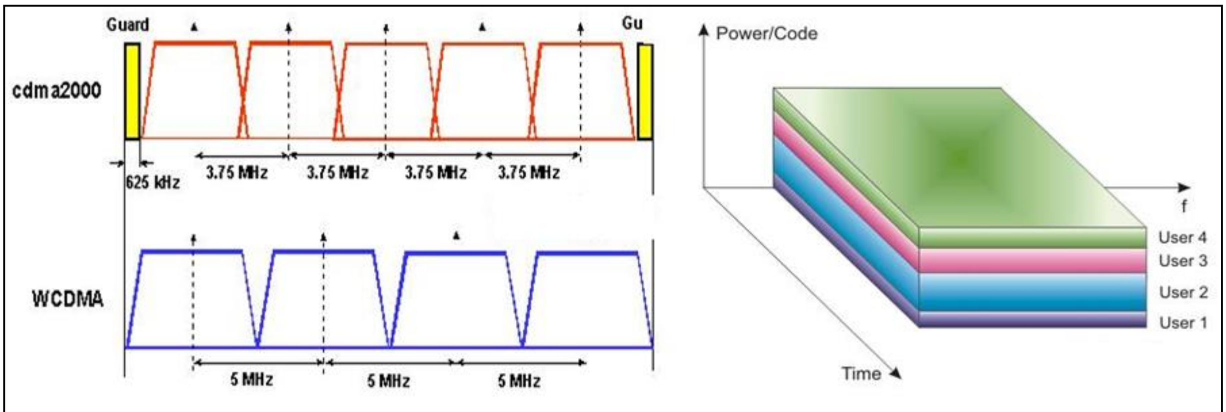
5 – Many components of the GSM core network were reused with a simple software Upgrade

6 – Video call ability and Mobile TV receiver ability

7 – High Security System and more safety with respect to G2

8 - 60 MHz divided to 12 block s of 5MHz channel Bandwidth

9 - Frequency Range - in Europe and Asia 1920 MHz to 1980 MHz for Up Link and 2110 MHz to 2170 MHz for Down Link



Code division and wide band code division multiple access

E- Fourth generation Mobile system (4G).

The fourth generation mobile system are commercially deployed in 2010 in South Korea and Scandinavia (LTE) , the main specification of this version are the high capacity of data rate and the compatibility with the internet network, the fourth generation are completely transferred to the packet switch system,

The specifications of the fourth generation mobile system can be summarized by :-

1 - Integrated System which include Multi Systems that depending on IP protocol

2 - Conjoined Network for multi converging and development networks (WAFI, WIMAX, WLAN)

3 - 100MHz to 1GHz Data Transfer Rate

4 - HD TV Mobile Receiving include

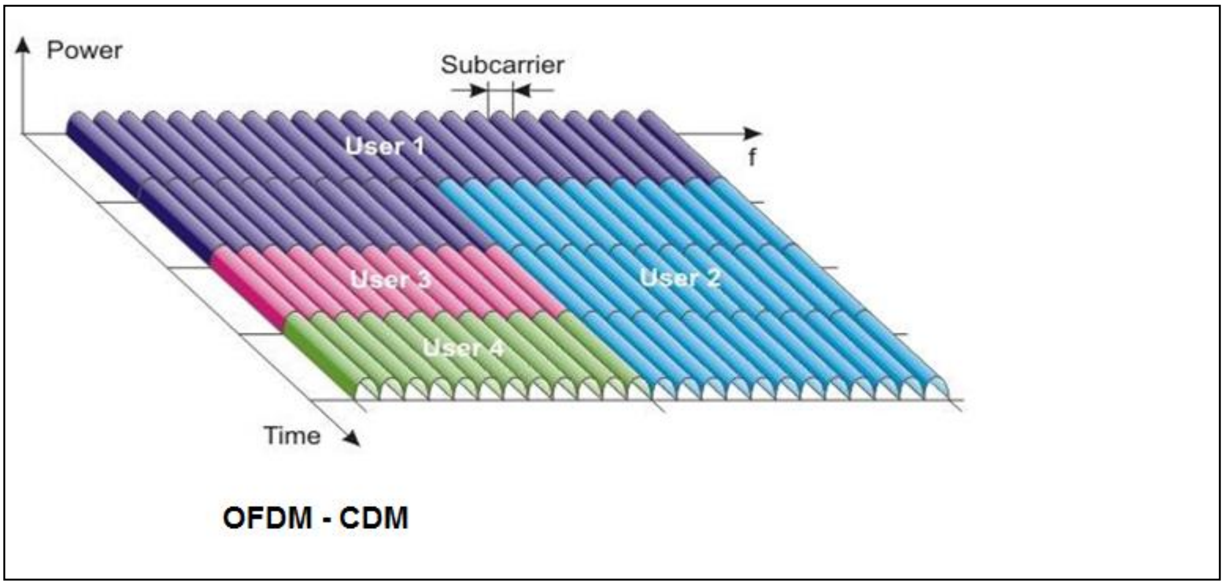
5 - Use Variable Spreading Factor Orthogonal Frequency and Code Division Multiplexing (VSF – OFCDM)

6 - Two 4G standard systems are commercially deployed

a- The Mobile WiMAX standard

b- Long term evolution (LTE) standard

7 - WiMAX smart phones have been available since 2010.



F- Fifth Generation Mobile system (5G) :-

The world looking forward get out the Fifth Generation mobile system (5G) commercially in 2020, the main specifications for this version are the high data rate transmission (up to 10 Gbit/sec) ,the wide compatibility with the internet applications, and high trustworthy (very low probability of error) because of using high efficient channel coding technique.

References:-

1 - Communication Systems for the Mobile Information Society
Martin Sauter

2 - FROM GSM TO LTE-ADVANCED AN INTRODUCTION TO MOBILE NETWORKS AND MOBILE BROADBAND
Martin Sauter

3 - D900/D1800/D1900 Overview
Siemens

4 - GSM SYSTEM ENGINEERING
Asha Mehrotra

5 - GSM Fundamental Course
Asiacell

Communication Systems

Mobile Communication Systems

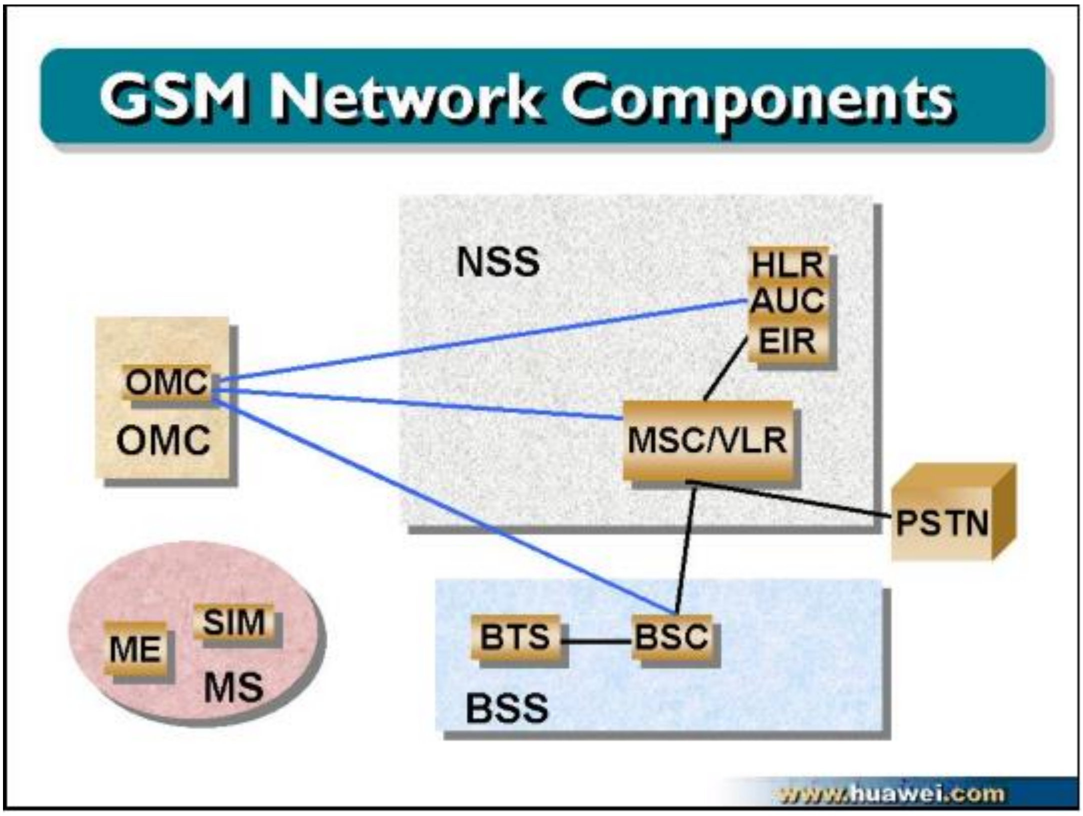
Lecture (2-2) :- GSM System

At the beginning of the 1990s, GSM, the Global System for Mobile Communications triggered a novel change in the way people communicate with each other. While earlier analog wireless systems were used by only a few people, GSM was used by over 1.5 billion subscribers worldwide at the end of 2005. This has mostly been achieved by the steady improvements in all areas of telecommunication technology and due to the steady price reductions for both infrastructure equipment and mobile phones.

in this topic we shall study the architecture of this system in details, (starting from the mobile station and ending to the network sub system), the air interface link (between the mobile station and the base station), the mobility of the signaling for this system, and the modified versions of this system(HSCSD, GPRS, and DEGE).

A- GSM structure

- The block diagram for the simplified GSM system are shown in figure (1) below.



From figure (1) above we can see that the GSM system consist of the following :-

1 – Mobile station which consist of mobile equipment (ME) and Subscriber Identity Model card (SIM card)

2 – Base station system (BSS) consist of Base Transceiver system (BTS), Base station Control (BSC), antennas, Power supply and other component.

3 – Network sub system (NSS) consist of

* - Mobile switching system (MSC)
* - Home Location Register (HLR)
* - Visitor Location Register (VLR)
*- Equipment Identity register (EIR)
*- Authentication Center (AUC)

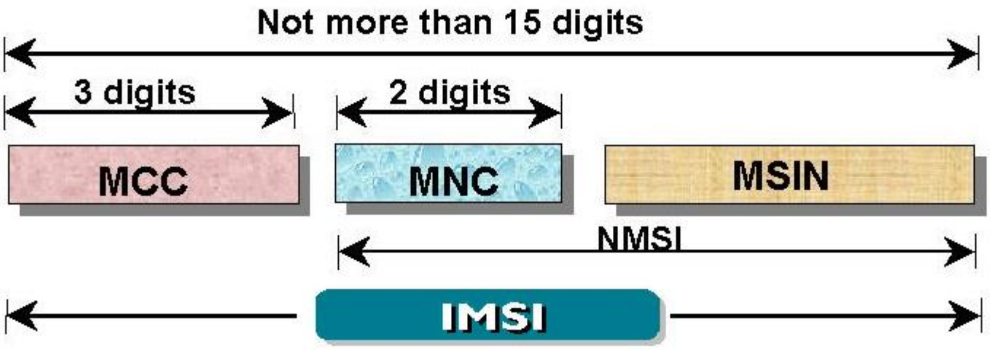
4 - Operation and maintenance center (OMC)

1) – The Mobile Station.

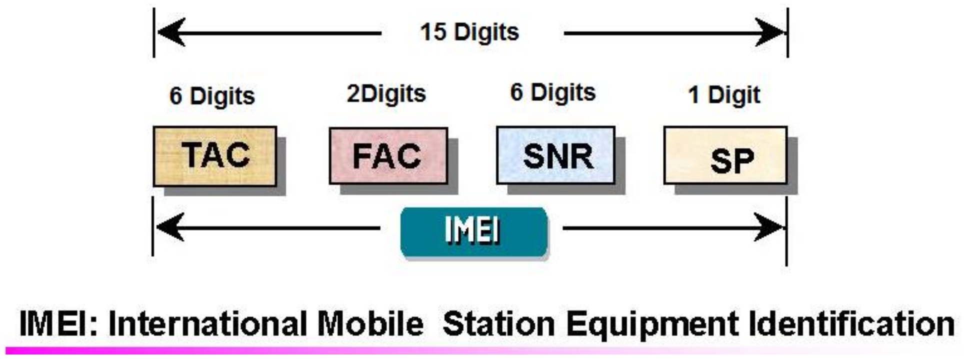
The mobile station consist of two parts, the Mobile Equipment (ME) and the an electronic smart card called a Subscriber Identity Module (SIM card), the mobile equipment is the hardware used by the subscriber to access the network , this may be a telephone, Fax machine,, etc. The hardware has an identity number which is unique for that device and permanently stored in it. this identity number is called an International Mobile Equipment Identity (IMEI).

The SIM is a card which plug into the mobile equipment. This card identified the mobile subscriber, the subscriber is identified by an identity number called the International Mobile Subscribe Identity (IMSI).

The IMSI and the IMEI format are as shown :-



MCC :- Mobile Country Code (for example (964) for Iraq
MNC :- Mobile Network Code (for example (77) for Asia Cell
MSIN :- Mobile Subscriber Identification Number.

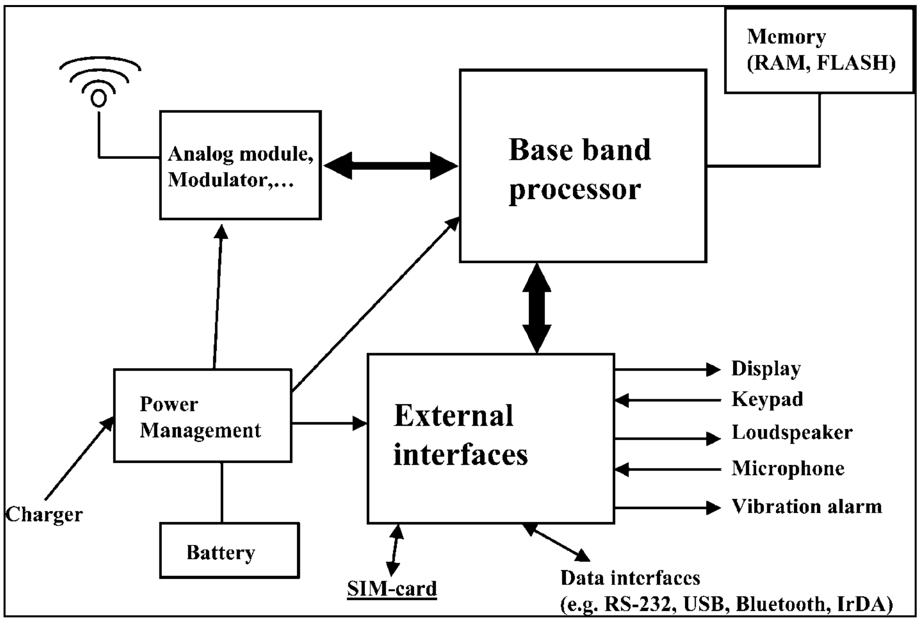


TAC :- Type Approval Code
FAC:- Final Assembly Manufacturer Code
SNR :- Serial Number
SP :- spare digit.

A) – Mobile Equipment (ME)

Due to the progress of miniaturization of electronic components during the mid-1980s, it was possible to integrate all components of a mobile phone into a single portable device.

Independent of the size and variety of different functionalities for the Mobile Equipment , the basic architecture of all mobile phones are shown in Figure (2) bellow.



The core of the mobile phone is the base band processor which contains a RISC (reduced instruction set) CPU and a digital signal processor (DSP). The RISC processor is responsible for the following tasks:

• Handling of information that is received via the different signaling channels (BCCH, PCH, AGCH, PCH, etc.).
• Call establishment (DTAP).
• GPRS management and GPRS data flow.
• Parts of the transmission chain: channel coder, interleaver, cipherer (dedicated hardware component in some designs).
• Mobility management (network search, cell reselection, location update, handover, timing advance, etc.).
• Connections via external interfaces like Bluetooth, RS-232, IrDA, USB.
• User interface control (keypad, display, graphical user interface).

The processor capacity (speed and cash Ram) of the RISC processor is the main factor when deciding which applications and features to implement in a mobile phone. For applications like recording and displaying digital pictures or videos for example, fast processing capabilities are required.

One of the RISC architectures that is used for high-end GSM and UMTS mobile phones is the (ARM-9) architecture. This processor architecture allows CPU speeds of over 200 MHz and provides sufficient computing power for calculation intensive applications.

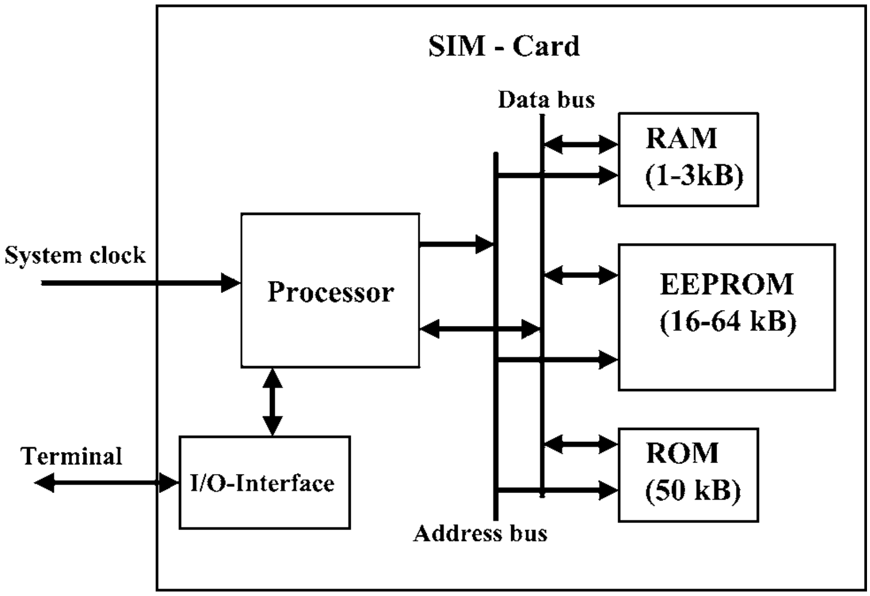
The downside of fast processors, however, is higher power consumption, which forces designers to increase battery capacity while trying at the same time to maintain the physical dimensions of a small mobile phone. Therefore, intelligent power-saving mechanisms are required in order be able to reduce power consumption during times of inactivity. The DSP is another important component of a GSM and UMTS chipset. Its main task is FR, EFR, HR, or AMR speech compression. Furthermore, the DSP is used in the receiver chain to help decode the incoming signal. This is done by the DSP analyzing the training sequence of a burst (we shall explain next lecture). As the DSP is aware of the composition of the training sequence of a frame, the DSP can calculate a filter which is then used to decode the data part of the burst. This increases the probability that the data can be correctly reconstructed. The DSP (56600) architecture with a processor speed of 104 MHz is often used for these tasks.

As millions of mobile phones are sold every year, there is a great variety of chipsets available on the market. The chipset is in many cases not designed by the manufacturer of the mobile phone. While Motorola design its own chipsets, Nokia relies on chipsets of STMicroelectronics and Texas Instruments. Other GSM chipset developers include Infineon, Analog Devices, and Philips, as well as many Asian companies.

The mobile equipment contain three types of memory, RAM , Flash ROM and internal ROM, the job of the internal ROM is store the IMEI and a part from the operating system and this can not editing by the subscriber, while the flash ROM store the operating system, the subscribers number, message, video, sound and other. The third type (RAM) used by the CPU in order to achieve the tasks, this type can not keeping the information when the device power off.

B) – Subscriber Identity Module Card (SIM Card)

The SIM card is one of the most important parts of a GSM network because it contains all the subscription information of a subscriber. Since it is standardized, a subscriber can use any GSM or UMTS phone by simply inserting the SIM card. The block diagram of SIM card components are shown in figure (3) below.



The most important parameters on the SIM card are the IMSI and the secret key (Ki), which is used for authentication and the generation of ciphering keys (Kc). With a number of tools, it is possible to read out most parameters from the SIM card, except for sensitive parameters that are read protected.
Protected parameters (like Ki) can only be accessed with a special unlock code that is not available to the end user.
The SIM card is much more than just a simple memory card as it contains a complete microcontroller system that can be used for a number of additional purposes. The typical properties of a SIM card are shown in Table (1).

Table 1 SIM card properties

CPU	8- or 16-bit CPU
ROM	40–100 kbyte
RAM	1–3 kbyte
EEPROM	16–64 kbyte
Clock rate	10 MHz, generated from clock supplied by mobile phone
Operating voltage	3 V or 5 V

The mobile phone cannot access the information on the EEPROM directly, but has to request the information from the SIM's CPU. Therefore, direct access to sensitive information is prohibited. The CPU is also used to generate the SRES during the network authentication procedure based on the RAND which is supplied by the authentication center (we shall explain in next lecture). It is imperative that the calculation of the SRES is done on the SIM card itself and not in the mobile phone in order to protect the secret Ki key.
From table (1) above we can see that the SIM card work on 10 MHz clock supplied by the mobile equipment through the connection point between them, also we can see that three type of memory lies in the SIM card (RAM, ROM, and EEPROM) , the EEPROME used to store limited subscribers numbers and names (with limited characters) , some massages and notes, this can be edited by the subscriber in opposite of the ROM which contain the protected parameters and the authentication algorithms. While the RAM used by the microprocessor to calculate the KC and other tasks.

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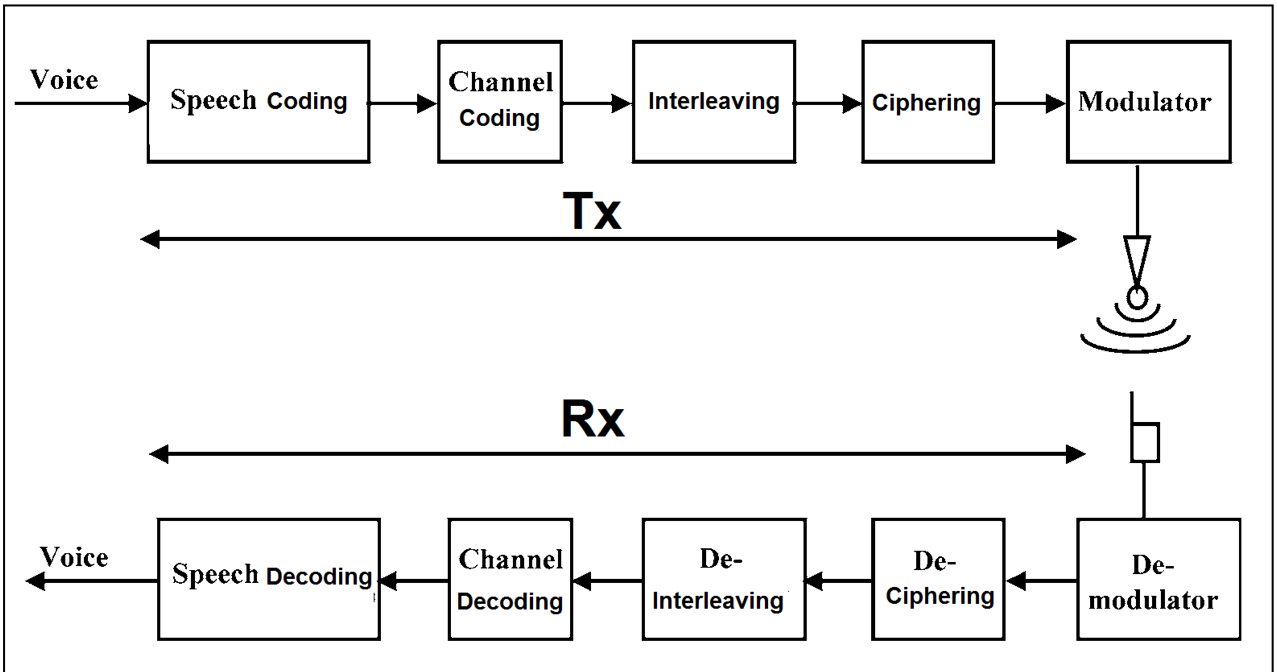
Lecture (3-2) :- The Digital Signal Processing for Speech signal in the mobile station.

As we know that the main specification of the second generation of mobile system depend on the digital technique to transmit the speech signal between all parts of the system, this technique provide an additional properties such as compression, ciphering ,error detection and other techniques, the bandwidth for the speech signal that can be recognized has been limited (from telephony system) between 300Hz- to 3400Hz , this signal sampled by8000 bit per second and converted to PCM (digital form) and passed through an additional functional blocks as shown in Figure(1) below



Digital Signal Processing For Speech Signal in the Mobile Station

From figure (1) above one can be see that the speech signal pass through the following functional Processing blocks in the transmission side:-

1 – Speech Coding

2 – Channel Coding

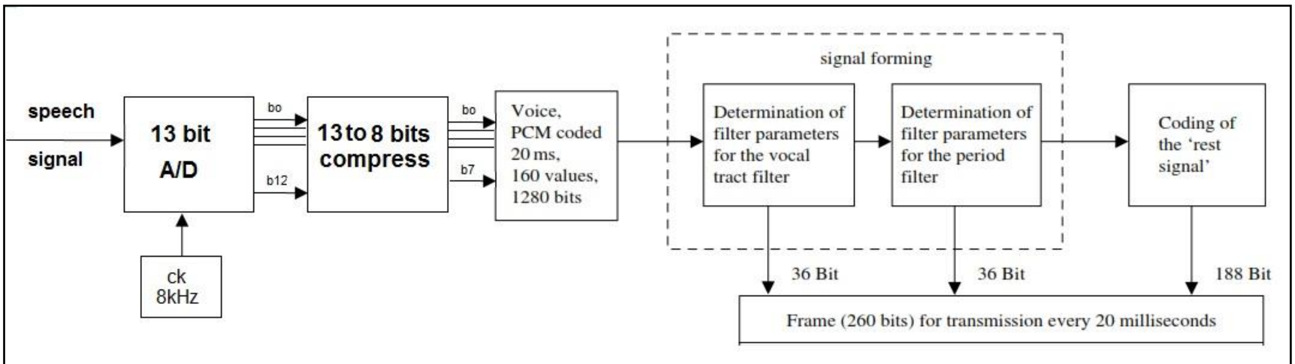
3 – Interleaving

4- Ciphering

5 – Modulator

1 – Speech coding :-

While the PCM algorithm digitizes analog volume levels by statically mapping them to digital values, the GSM speech digitization is much more complex to reach the desired compression rate. At the first the speech signal has been sampled by 8kb/sec an then each sample converted to the 13 bit PCM signal, these digital codes then compressed to 8-bit per sample, after that another compression is achieved by emulating the human vocal system. This is done by using a source-filter model as shown in Figure (2)



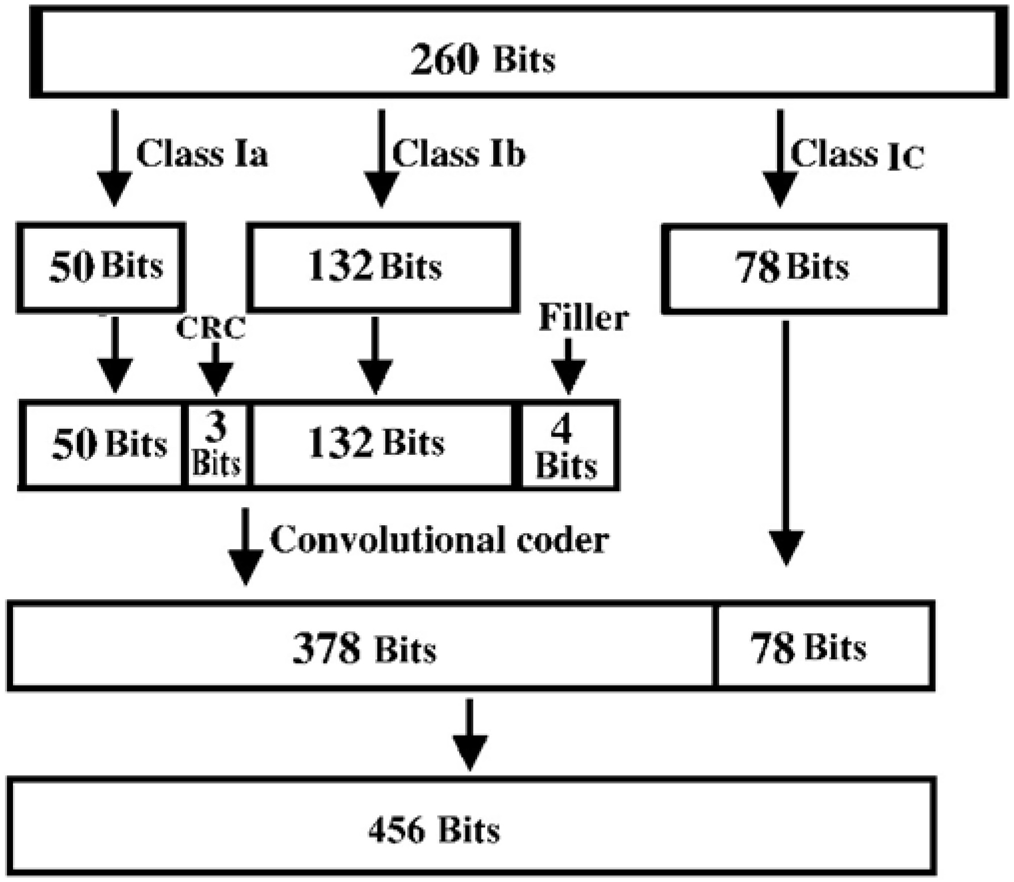
Block diagram for the Speech coding process

The speech forming is simulated by using two time-invariant filters. The period filter creates the periodic vibrations of the human voice while the vocal tract filter simulates the envelope. The filter parameters are generated from the human voice, which is the input signal into the system. The system is simplified by generating a pair of filter parameters for an interval of 20 milliseconds. As an input to the algorithm, a speech signal is used that has previously been converted into an 8- from 13-bit PCM codec. As the PCM algorithm delivers 8000 values per second, the FR codec requires 160 values for a 20 ms interval to calculate the filter parameters. As eight bits are used per value, 8 bits × 160 values = 1280 input bits are used per 20 ms interval. For the period filter, the input bits are used to generate a filter parameter with a length of 36 bits. Afterwards, the filter is applied to the original input signal. The resulting signal is then used to calculate another filter parameter with a length of 36 bits for the vocal tract filter. Afterwards, the signal is again sent through the vocal tract filter with the filter parameter applied. The signal, which is thus created, is called the 'rest signal' and coded into 188 bits.

Once all parameters have been calculated, the two 36-bit filter parameters and the rest signal, which is coded into 188 bits, are sent to the receiver. Thus, the original which was coded in 1280 bits, has been reduced to 260 bits. In the receiver, the filter procedure is applied in reverse order on the rest signal and thus the original signal is recreated. As the procedure uses a lossy compression algorithm, the original signal and the recreated signal at the other end are no longer exactly identical. For the human ear, however, the differences are almost inaudible.

2 – Channel Coding :-

Before a 260-bit data frame is transmitted over the air interface every 20 ms, it traverses a number of additional functional blocks . In a first step, the voice frames are processed in the channel coder unit, which adds error detection and correction information to the data stream. This channel coder process are shown in figure (3) below.



Channel Coder Process

The channel coder separates the 260 bits of a voice data frame into three different classes as shown in Figure (3) above.

- - Fifty **(50)** bits from the 260 bits of a speech frame are **class Ia** bits and extremely important for the overall reproduction of the voice signal, such bits are for example the higher order bits of the filter parameters, a three-bit CRC checksum is calculated and added to the data stream.
- - Another **(132)** bits from the frame are also quite important and are thus put into **class Ib**. However, no checksum is calculated for them. In order to generate the exact amount of bits that are necessary to fill a GSM burst, four (4) filler bits are inserted.

- - The remaining **(78)** bits from the original 260-bit data frame belong to the third class which is called **class Ic**. These are not protected by a checksum and no redundancy is added for them. Errors which occur during the transmission of these bits can neither be detected nor corrected.

Afterwards, the class Ia bits, checksum, class Ib bits, and the four filler bits which represent **(189 bits)** are treated by a convolutional coder which adds redundancy to the data stream. For each input bit, the convolutional decoder calculates two output bits, the resultant outputs of the convolutionl channel coder are **(378 bits)**.

The final output of the channel coder contain **(456 bits)** which are came from the **(378 bits)** and the **(78 bits)** that are classified as class II.

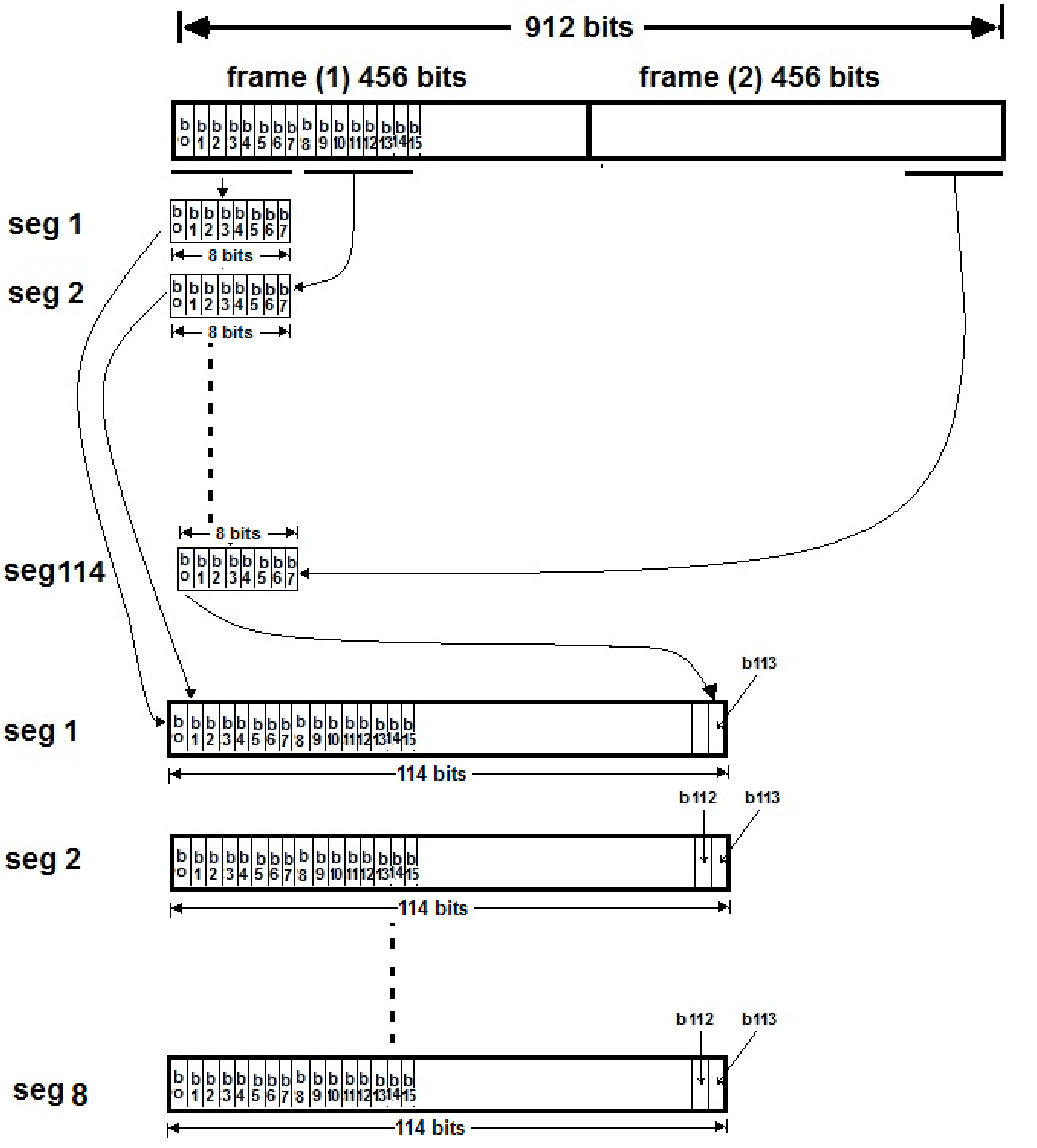
3 – Interleaving

As has been shown from the previous paragraph , the channel coder uses the 260-bit input frame to generate 456 bits on the output side. If several consecutive bits are changed during the transmission over the air interface, the convolutional decoder on the receiver side is not able to correctly reconstruct the original frame. This effect is often observed as air interface disturbances usually affect several series bits.

In order to decrease this effect, the interleaver changes the bit order of a 456-bit data frame in a specified pattern over eight bursts Figure (4).

The interleaving process can be illustrated as the following steps :-

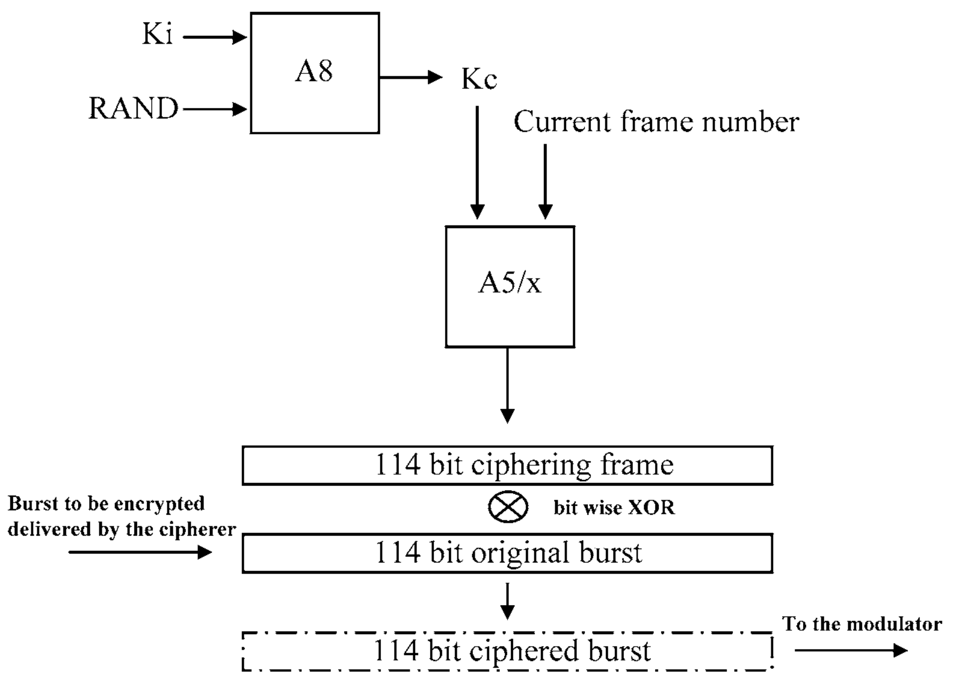
- - Combine two channel coding outputs frames (each contain 456 bits) together to generate one frame that contain (912 bits) which represents 40 mSec from speech signal.
- - Divides the resultant frame to 114 segments each segment contain 8 bits .
- - Generate 8 segments from the above 114 segments by combining b0 from each 114 segment to generate the anew segment (1), and combine b1 from each 114 segments to generate the new segment (2) and so on.



Interleaving Process

4 – Ciphering :-

The next module of the transmission chain is the cipherer (Figure(5)), which encrypts the data frames it receives from the interleaver.



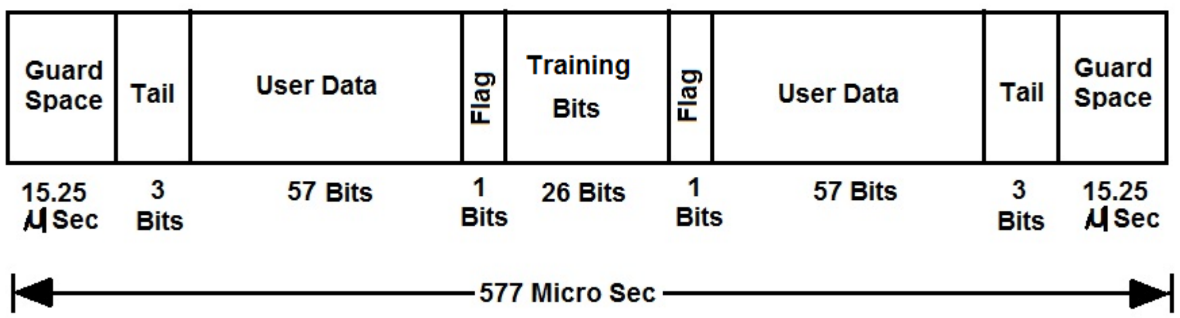
Ciphering Process

GSM uses, like most communication systems, a stream cipher algorithm. In order to encrypt the data stream, a ciphering key (Kc) is calculated in the authentication center and on the SIM card by using a random number (RAND) and the secret key (Ki) as input parameters for the A8 algorithm. Together with the GSM frame number, which is increased for every air interface frame, Kc is used as input parameter for the A5 ciphering algorithm.

The A5 algorithm computes a 114-bit sequence which is XOR combined with the bits of the original data stream. As the frame number is different for every burst, it is ensured that the 114-bit ciphering sequence also changes for every burst which further enhances security.

5 –Modulation :-

Before modulated the frames of the digital voiced ciphered frames they must formatted into specified structure called traffic burst (Tch burst), in this stage each (114) bits ciphered frame are divide to two part each contain (57) bits and assisted with additional sections of bits as a tail, flags and training bits section, the burst is encapsulated by two guard time in which no data is sent, each guard time with duration equal (15.25) micro second lies in the start and the end of the burst data, the overall duration of the burst in GSM is 577 micro second, the Tch burst structure are shown in figure (6) below.



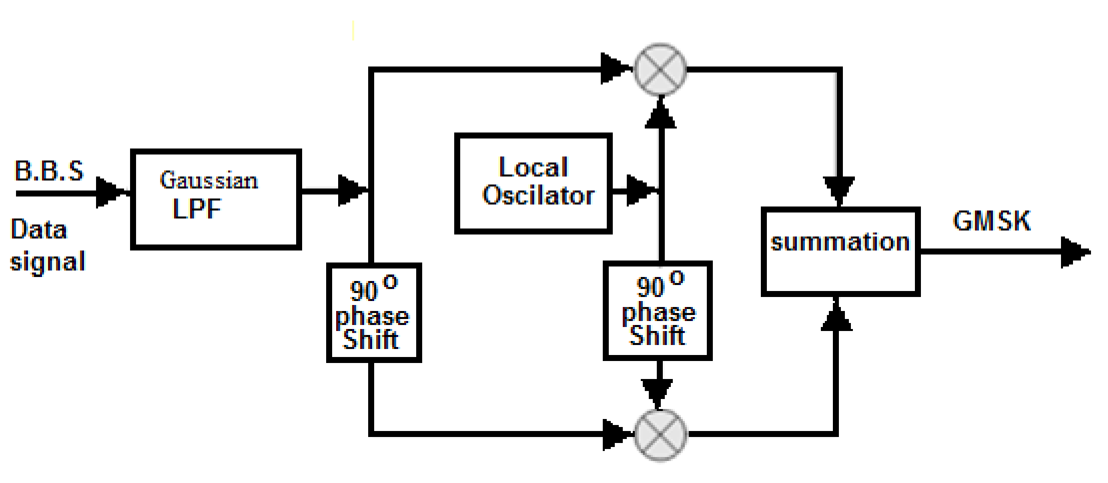
TCH Burst Structure

At the end of the transmission chain, the modulator maps the digital data onto an analog carrier, which uses a bandwidth of 200 kHz. This mapping is done by encoding the bits into changes of the carrier frequency. As the frequency change takes a finite amount of time, a method called Gaussian minimum shift keying (GMSK) is used, which smoothes the flanks created by the frequency changes. GMSK has been selected for GSM as its modulation and demodulation properties are easy to handle and implement into hardware and due to the fact that it interferes only slightly with neighboring channels.

GMSK modulation is based on MSK, which is itself a form of continuous-phase frequency-shift keying, CPFSK. One of the problems with standard forms of PSK is that sidebands extend out from the carrier. To overcome this, MSK and its derivative GMSK can be used.

There are two main ways in which GMSK modulation can be generated. The most obvious way is to filter the modulating signal using a Gaussian filter and then apply this to a frequency modulator where the modulation index is set to 0.5. This method is very simple and straightforward but it has the drawback that the modulation index must exactly equal 0.5. In practice this analogue method is not suitable because component tolerances drift and cannot be set exactly.

A second method is more widely used. The quadrature modulator uses one signal that is said to be in-phase and another that is in quadrature to this. In view of the in-phase and quadrature elements this type of modulator is often said to be an I-Q modulator. Using this type of modulator the modulation index can be maintained at exactly 0.5 without the need for any settings or adjustments. This makes it much easier to use, and capable of providing the required level of performance without the need for adjustments. For demodulation the technique can be used in reverse.



GMSK modulator using an I-Q modulator

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Lecture (4-2) :-The Radio Interface (UM) for GSM

The previous lecture firmed in the consideration that the speech signal constructed into burst form after treated of the speech coding, channel coding and ciphering process are modulated using GMSK technique and became ready to transmit it through the wireless channels, also we know that each burst have the duration of **(577 µSec)** and need **(200 kHz)** physical channel bandwidth.

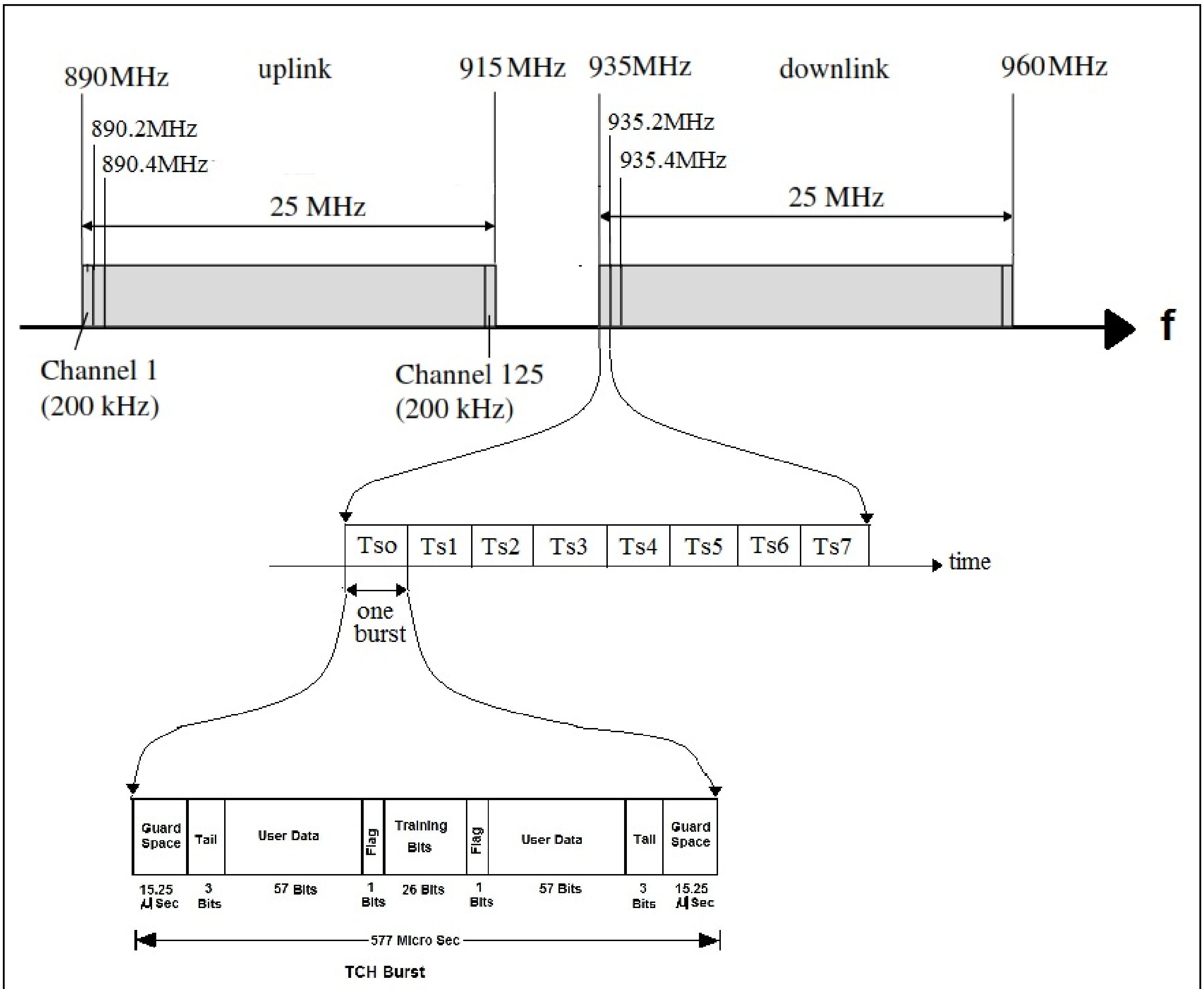
in this lecture and beyond we shall explain the wireless interface between the mobile station and the next unit of the mobile system (Base station).

This subject include the Usable frequency bandwidth for the GSM systems, physical distribution for channels , the classification of the channels, the framing in GSM, types of channels in GSM, the location of each channel in the frame structure, and the burst formats for each type.

- ***Usable Frequency Bandwidth for GSM Systems.***

The GSM systems uses two separated frequency bands, one for up-link and the other for down-link , each band divided using FDMA technique to multi physical channels, each one of these channels occupies (200kHz) from the assigned bandwidth , this small bandwidth (200kHz) called (physical Channel).

Each physical channel are divided in the time domain to 8 time slot, each time slot which called (Logical channel) can be serve one subscriber (inserted of one burst), therefore eight (8) subscribers can be sharing into one physical channel using the TDMA technique. Figure (1) below illustrate the channel distribution in the frequency bands for the Primary GSM 900 system.



channel distribution in the frequency bands for the Primary GSM 900 system.

Note :- some time slots can be carry not only traffic burst but other types like signals burst or other.

Practically there are four GSM systems uses different frequency bands which are :-

1 – Primary GSM 900 (PGSM 900)

2 – Enhanced GSM 900 (EGSM 900)

3 – GSM 1800

4 – GSM 1900

The specifications for these GSM systems are shown in table (1) below

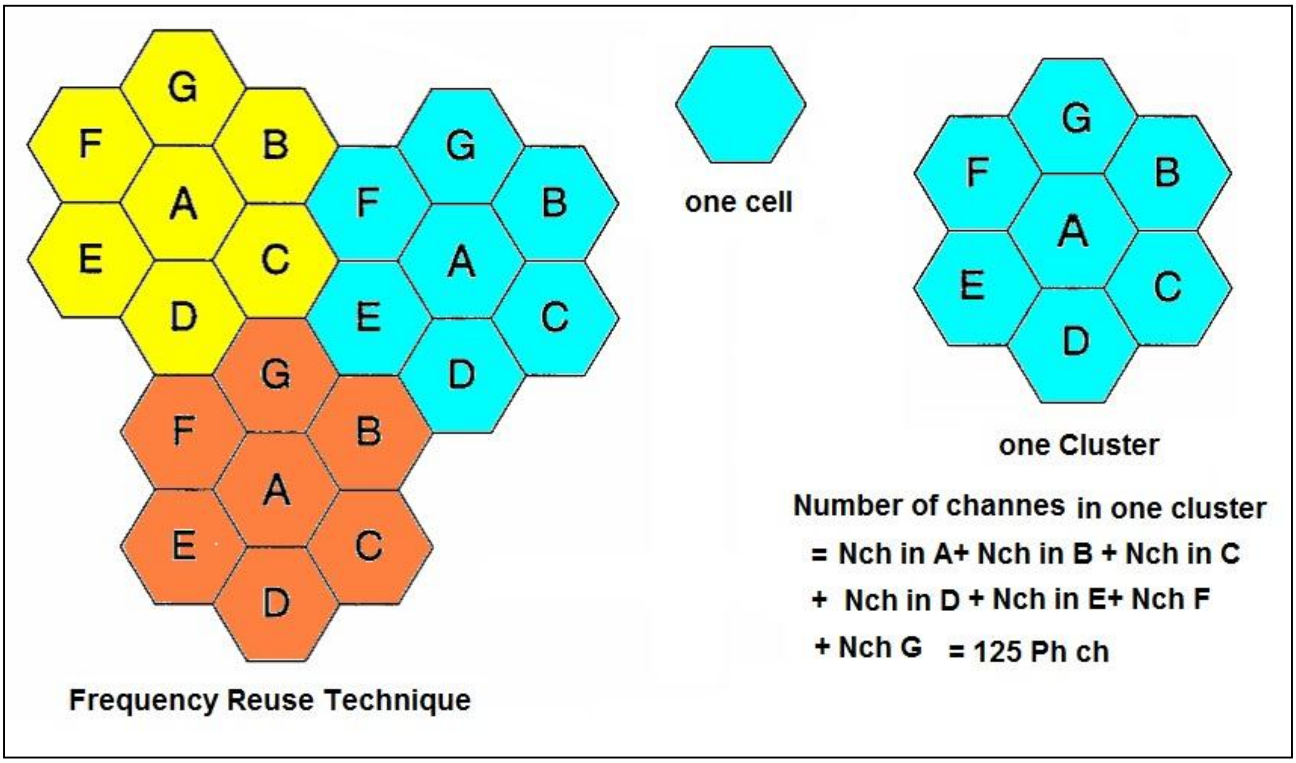
Table (1) GSM SYSTEMS

GSM system	U LINK MHz	D Link MHz	U / D BW	Ph. Ch
PGSM 900	890 – 915	935 – 960	25MHz	125
EGSM 900	880- 915	925 - 960	35MHz	175
GSM 1800	1710 – 1785	1805 - 1880	75 MHz	375
GSM 1900	1850 – 1910	1930 - 1990	60MHz	300
Speech and data transfer systems				
Physical channel Bandwidth = 200KHz				
Each physical Cannel carry 8 traffic channels using (TDMA)				
Signaling rate = 270 Kbit/Sec				
Maximum data bit-rate = 9.6Kbit/Sec				

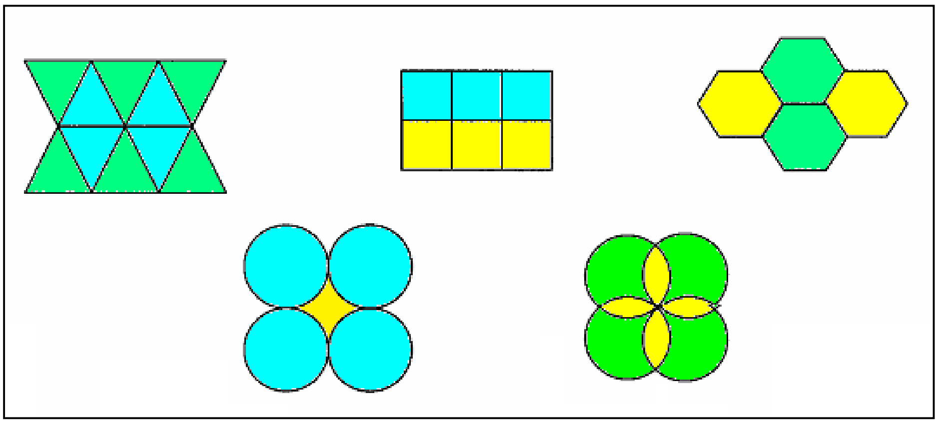
- ***- Channel distribution in GSM system***

As we understand previously there are (1000) subscribers can be contact in the network at the same time (in PGSM), this quite not enough, the efficient solution to increase the capacity of subscribers is the use of frequency reuse technique. This technique done by divide the 125 physical channel into multi groups each group contain specific number of physical channel (not necessary to be equal in each group).

in the other hand, the served land divided into multi clusters, each cluster contain multi cells , each group of channels serve one cell within the cluster, and the each cluster occupied the all 125 physical channels.



The shape of cell can be take many form, may be circular ,rectangular , square, triangular, hexagon or other forms, the shape of the cell assigned according to electromagnetic topography, and this done by practical measurements of the electromagnetic signal radiation pattern on the land.



the number of cells per cluster (K) and the distance between two cells that carry the same frequency group are illustrated below

$K = i^2 + j^2 + i\,j$

Where i and j are an integer number

K is Number of cell per Cluster

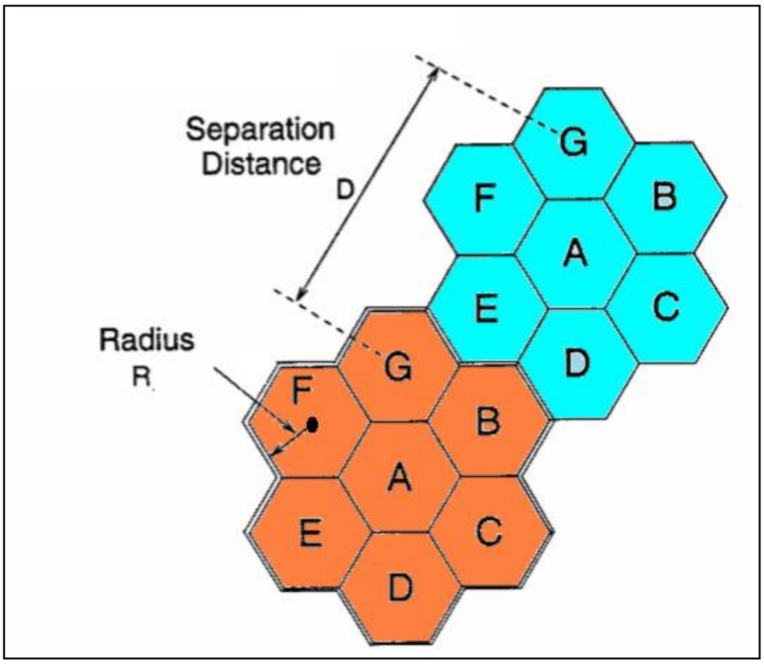
$i = 1,2,3,4,\ldots$

$j = 1,2,3,4,\ldots$

$$\frac{D}{R} = \sqrt{3K}$$

Where **D is the Separation distance between two cells that cary same frequency Group**

R is the Radius of the cell

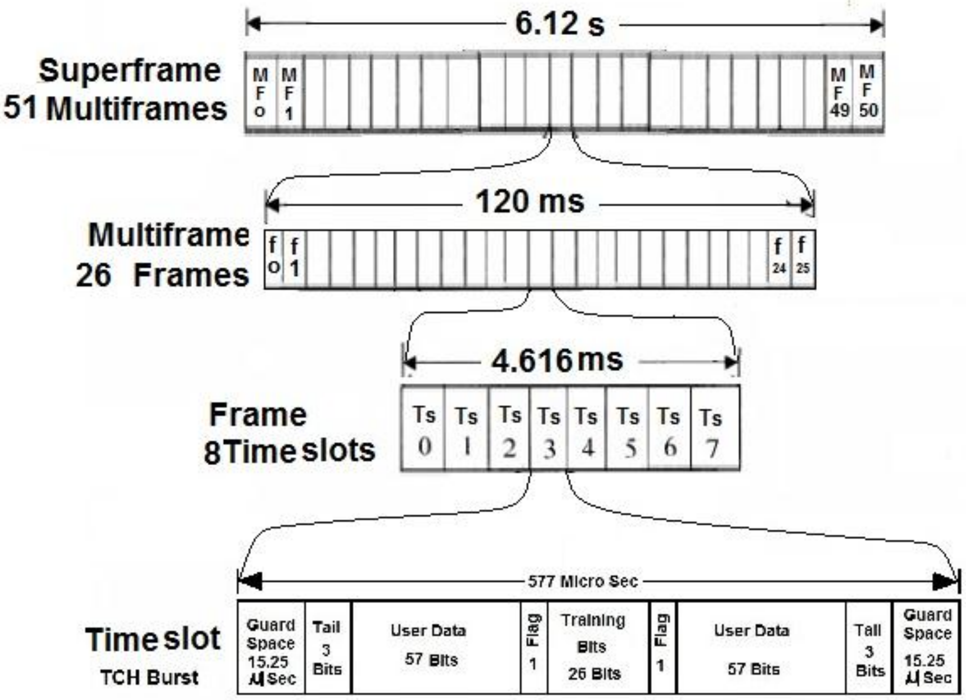


- ***<u>- Framing structure in GSM system</u>***

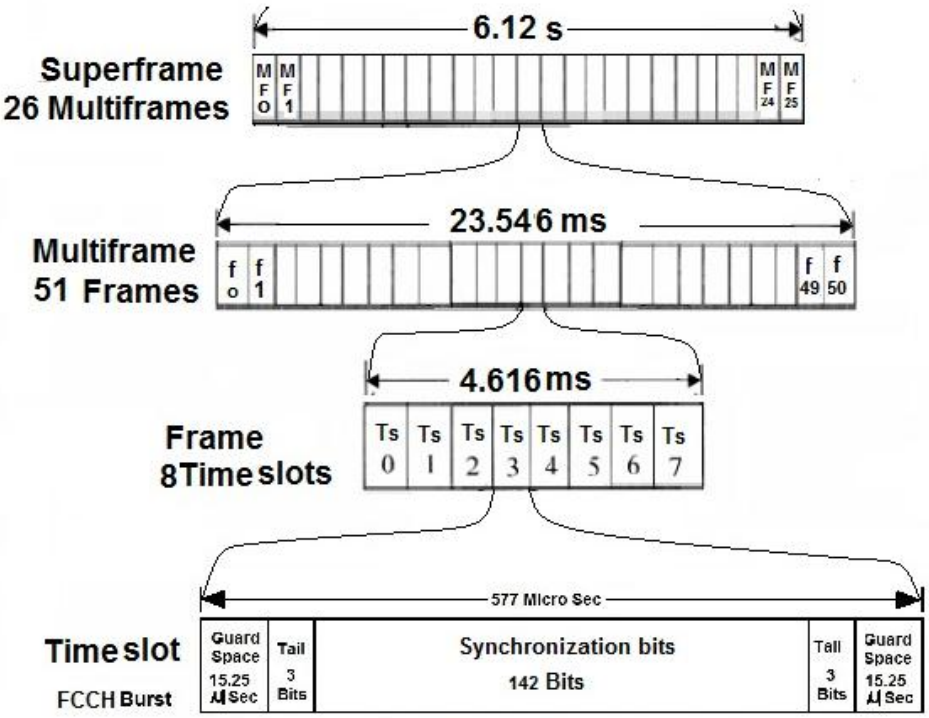
As we illustrate previous, each physical channel contain 8 time slot, and each time slot may be occupied by the traffic burst or other , the non traffic burst which may be occupied the time slots are the signaling burs, therefore there are two type of framing structures in GSM system , the traffic framing structure and signaling framing structure.

.a – **Traffic framing structure** : - each 8 time slot represent one frame which have time duration equal to (8*577µSec = 4.615 m Sec), each **<u>(26)</u>** frame represent one multi frame , and each **<u>(51)</u>** multi frame represent one Super frame.

.b – **Signaling framing structure** : - each 8 time slot represent one frame which have time duration equal to (8*577µSec = 4.615 m Sec), each **<u>(51)</u>** frame represent one multi frame , and each **<u>(26)</u>** multi frame represent one Super frame.



Traffic framing structure



Signaling framing structure

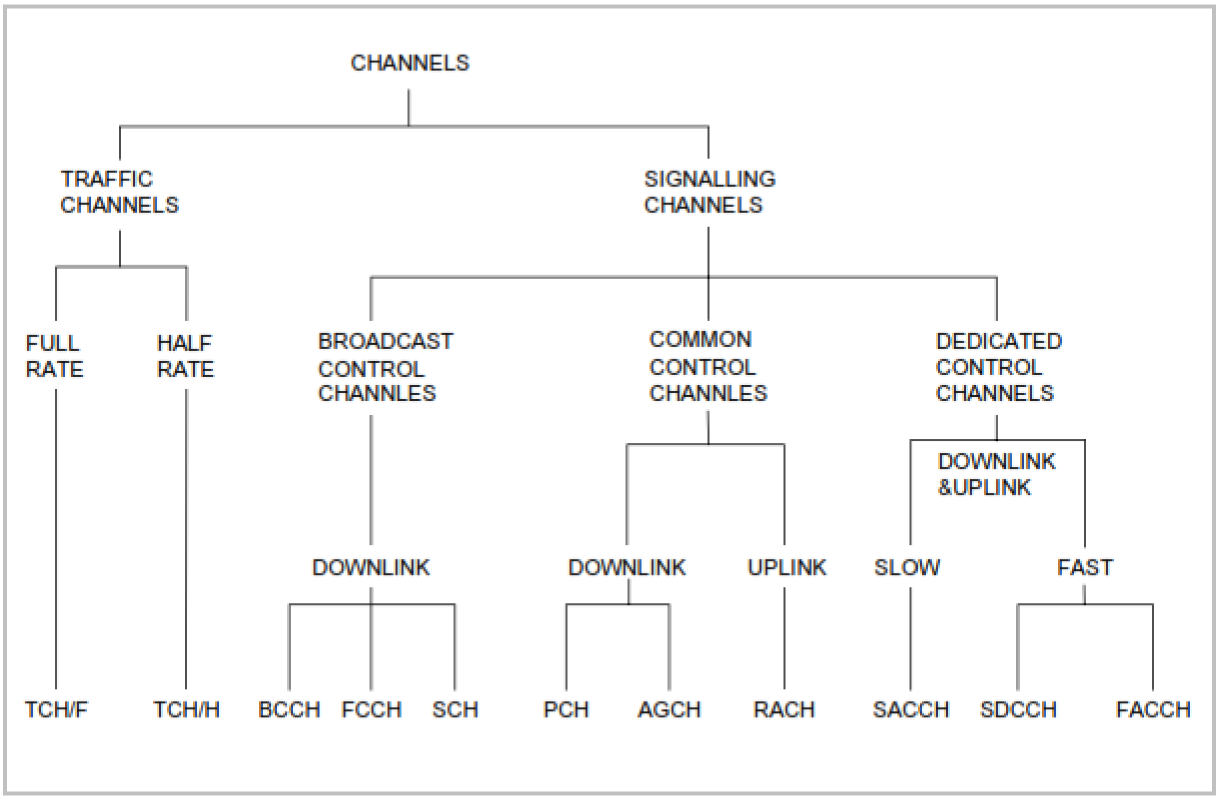
- ***- Classification of radio interface channels***

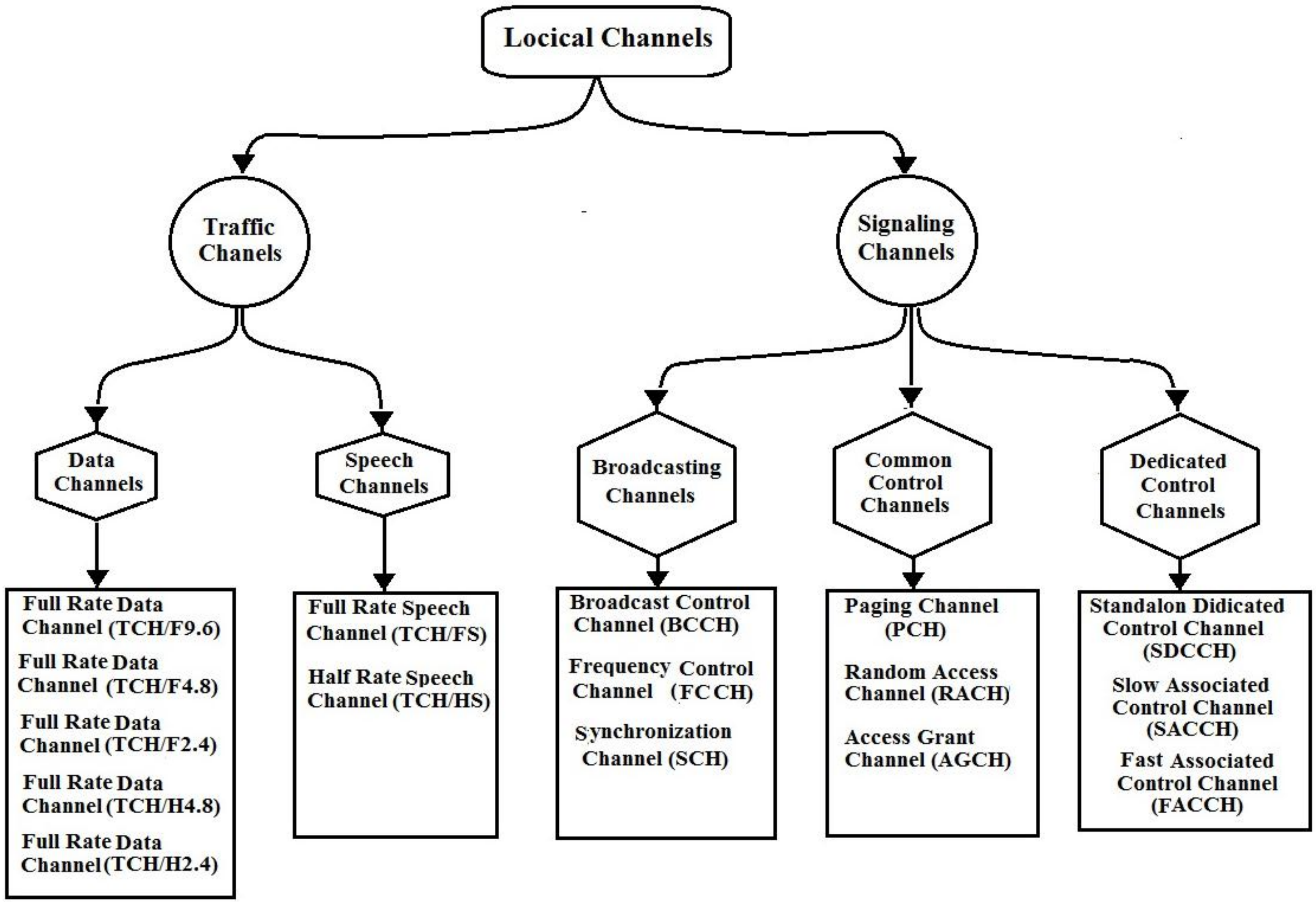
We can understand from the previous sections in this lecture that the usable GSM bandwidth (25 MHz in PGSM) are divided to 125 physical channels, each physical occupied 200KHz and each physical channel are divided in the time domain to 8 time slot, each time slot called Logical channel. These logical channels can be presented into two form

a- Traffic channel which represent the speech signal for the subscriber which formatted as (Tch burst), this type can be appear into two main types,
  - - Speech channels
  - - Data channels.

b- Signaling channels which transfer the necessary signals that used by the GSM system to arrange the mobile communication link. These signaling channels can be classified into three types
  - - Broadcasting Channels
  - - Common Control Channels
  - - Dedicated Control Channels

The block diagram for the logical channels in GSM air interface are shown below





Logical channel for The Radio Interface (UM) for GSM

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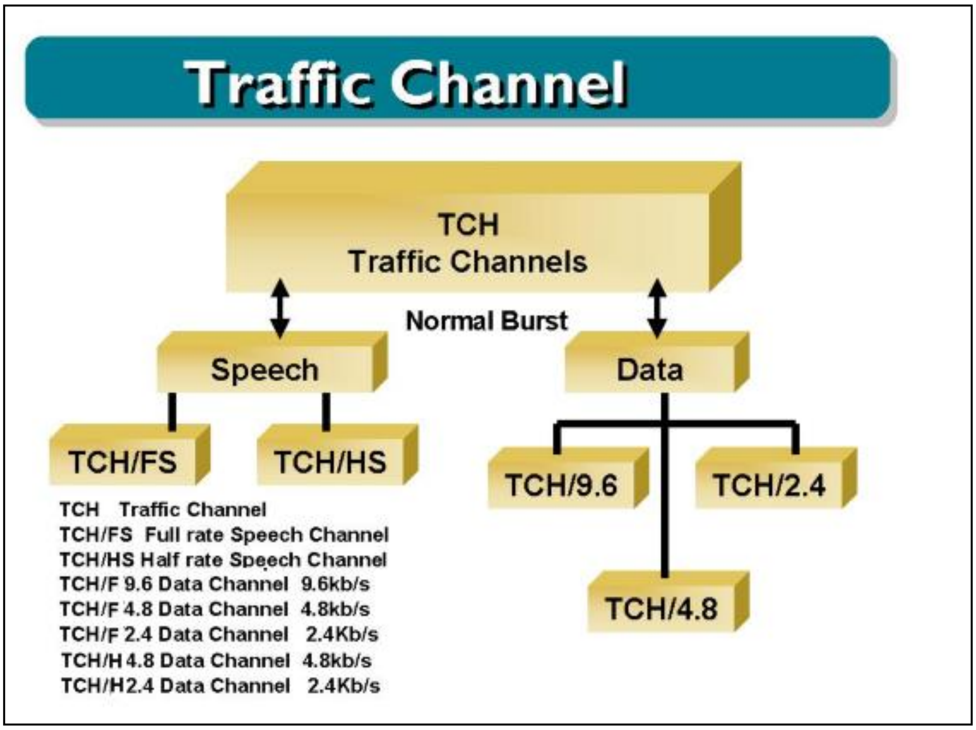
Lecture (5-2) :-The Radio Interface (UM) for GSM Part (2)-

The previous lecture we explain the Usable frequency bandwidth for the GSM systems, physical distribution for channels , the framing in GSM ,the classification of the channels, in this part we shall explain the types of channels in GSM, the location of each channel in the frame structure, and the burst formats for each type. At the end of the previous lecture we know that there are two groups of logical channels, the traffic channels and the signaling channels in this lecture we shall illustrate in details each type of the logical channels and specified the location of each channel in traffic framing or the signaling framing and state the burst form for each channel types.

1 The Traffic Channels (TCH Channels)

The traffic channels can carries speech or the data as shown



- - **Speech Cannels:-**

Speech channels are defined for both full rate and half rate traffic channels When transmitted as full rate, the user data is occupied within TS per frame. When transmitted as half rate, the user data is occupied into the same time slot but sent in alternate frames.

A. **Full - rate Speech Channel)TCH/Fs):** At 16 kbps the full rate speech channel is digitized. The full rate speech channel caries 55.8kbps after adding the GSM channel coding to the digitized speech.

B. . **Half Rate Speech Channels (TCH/HS):** The half rate speech channel can carry digitized speech that is sampled at a rate half that of full rate channel. GSM anticipates the availability of speech coder. It can digitize speech at about 6.5 kbps. After adding GSM channel coding to the digitized speech, the half rate Speech channel will carry 11.4 kbps.

- **-Data Channels :-**

Data channels support a variety of data rates (2.4, 4.8 and 9.6 Kb/s) on both half and full rate traffic channels. The 9.6 Kb/s data rate is only for full rate application.

A. **Full-rate Data Channel for 9600 bps (TCH/F9.6):** The full rate traffic data channel contains raw data that is transmitted at 9.6 kbps. After the application of additional forward error correction coding with the GSM standards, 9600 kbps is transferred at 22.8 kbps.

B. **Full-rate Data Channel for 4500 bps(TCH/F4.8):** The full rate traffic date channel contains data that is transmitted at 4.8 Kbps. After the application of additional forward error correction coding with GSM standards, the 4.8 kbps is transferred at 22.8 kbps.

C. **Full Rate Data Channel for 2400 bps (TCH/F2.4):** The full rate traffic data channel contains raw data that is transmitted at 2.4 kbps. After the application of additional forward error correction coding with GSM standards, the 2.4 kbps data is transferred at 22.8 kbps.

D. **Half Rate Data Channel for 4800 bps (TCH/H4.8):** The half rate traffic data channel carries raw data that is sent at 4800 bps. After the application of forward error correction using GSM standards, 4800 bps data is sent at 11.4 kbps.

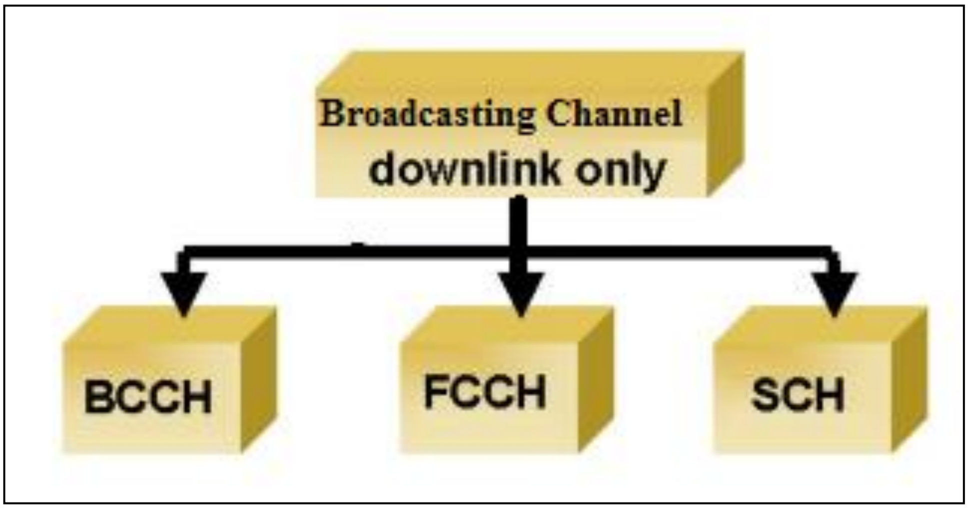
E. **Half Rate Data Channel for 2400 kbps (TCH/H 2.4):** The half rate traffic data channel carries raw user data that is sent at 2400 bps. After application of additional forward error correction using GSM standards, 2400 bps data is sent to 11.4 bps.

2 – Signaling Channels:-

. Signaling channels carry three types of signaling information between an MS and a BTS, broadcasting channels, common control channels, and dedicated control channels.

A) **Broadcasting channels:**

Broadcasting channels are transmitted in downlink direction only i.e. only transmitted by BTS.. The broadcast channels are used to broadcast synchronization and general network information to all the mobile stations (MSs) within a cell. It has three types:



- - **BROADCAST CONTROL CHANNEL (BCCH):**

  The BROADCAST CONTROL CHANNEL (BCCH) is used to broadcast control information to every MS within a cell. This information includes details of the control channel configuration used at the BTS, a list of the BCCH carrier frequencies used at the neighboring BTSs and a number of parameters that are used by the mobile station (MS) when accessing the BTS. The BCCH monitored by the mobile station periodically (at least every 30 Sec), when it is switched on and not in a call. BCCH carries the following information

  1 – Location Area identity (LAI)

  2 – List of neighboring cells that should be monitored by the mobile.

  3 – List of frequencies used in the cell

  4 – Cell identity

  5 – Power Control indicator

  Finally the BCCH is transmitted at constant power at all times and its signal strength is measured by all mobile stations which may seek to use it.

  BCCH Occupied Time slot zero (TS0) of the frames (2,3,4,and 5) from each multi frame.

- - **FREQUENCY CORRECTION CHANNEL (FCCH):**

. is transmitted frequently Used for the frequency (repeated (every 10 sec)) and is more easily detection by the mobile. When FCCH is detected , the mobile station corrects the frequency.

  FCH Occupied Time slot zero (TS0) of frames (0.10,20,30,40) fro each Multi frame.

- - **SYNCHRONISATION CHANNEL (SCH):**

  Allows the mobile station to synchronize time wise with the BTS.

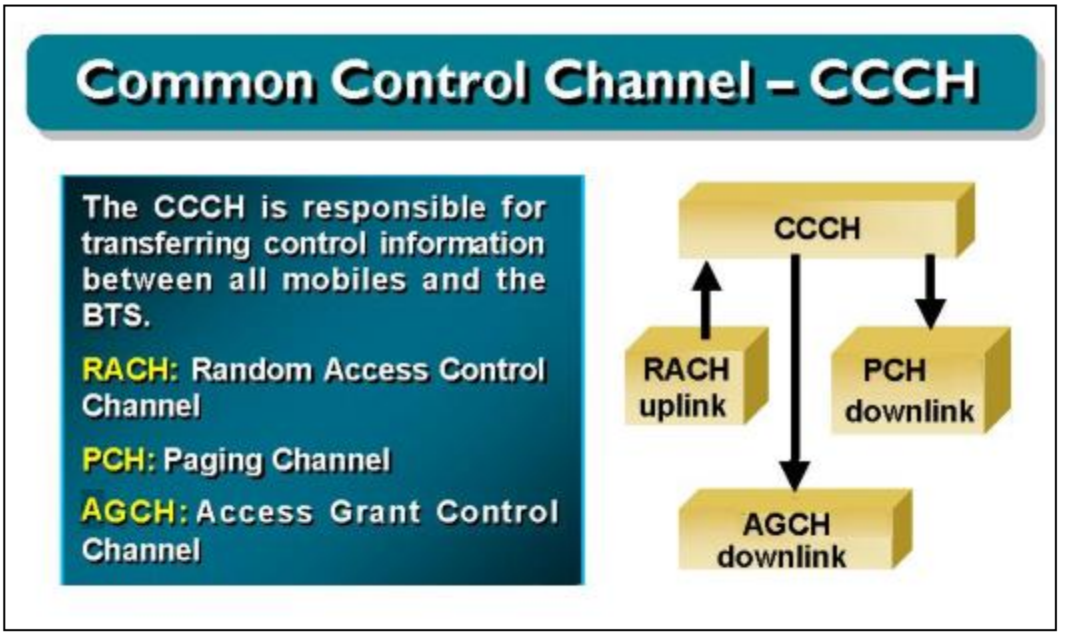
  Repeated broadcast (every 10 frames) of Synchronization Bursts is called

(SCH). It carries the information to enable the mobile to synchronization to the TDM frame structure. (Frame Number and Base Site Identity Code (BSIC) are also carried by this channel.

SCH Occupied Time slot zero (TS0) of frames (1,11,21,31,41) from each multi frame.

B - Common Control Channels:

The common control channels are used by an MS and the BTS during the paging and access procedures. Common control channels are of three types some of these signals are work in downlink and other work in up- link.



- **(PCH) PAGING CHANNEL:**

  . Within certain time intervals the MS will listen to the Paging channel, PCH, to see if the network wants to get in contact with the MS. The reason could be an incoming call or an incoming Short Message. It is thin work in the down link

- **- (RACH) RANDOM ACCESS CHANNEL:**

  If listening to the PCH, the MS will realize it is being paged. The MS answers, requesting a signaling channel, on the Random Access channel, RACH.

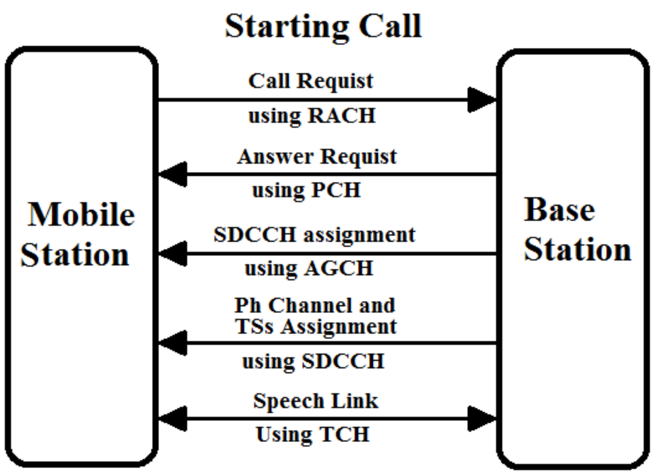
RACH can also be used if the MS wants to send request for contact with the network, e/g. when setting up a mobile originated call.

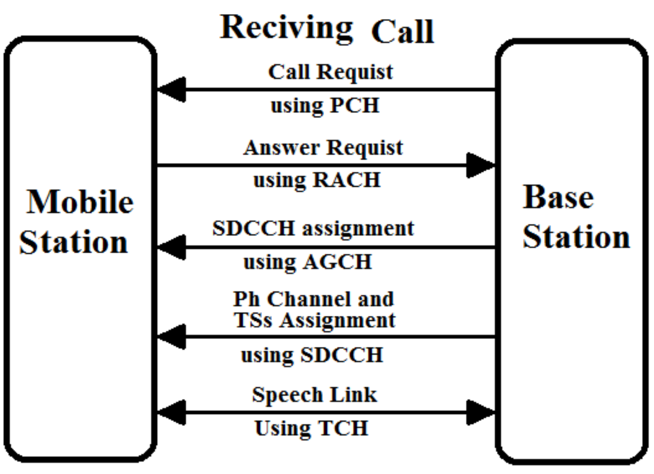
So the RACH work in up-link and it is occupied Time slot zero (TS0) of any frames from the multi frame (this signal are in up-link so it may came in TS0 of all frames within the signaling multi frame)

- **- (AGCH) ACCESS GRANTED CHANNEL:**

The access grant channel (AGCH) is carried data which instructs the mobile to operate in (the dedicated control channel) that transmit the (Time slot and the physical channels for the up-link and down link)).

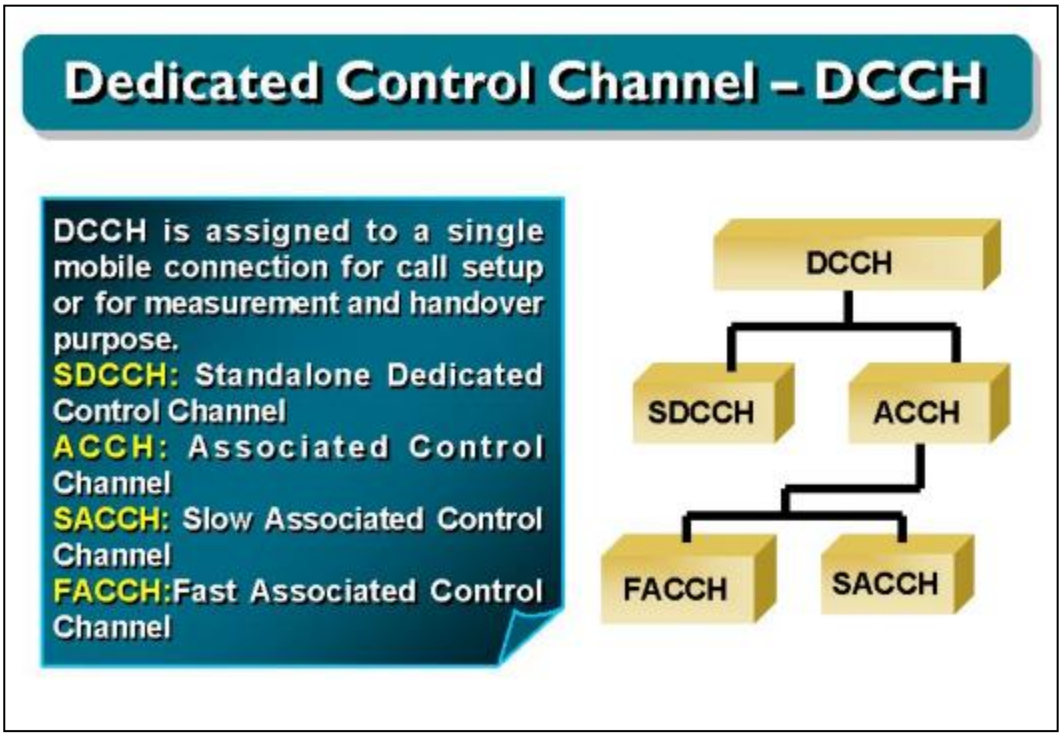
ACCH occupied Time slot zero (TS0) of any frame that not used by (broadcasting channels and the paging channel).





D - Dedicated Control Channels (DCCHs):

Signaling information is carried between an MS and a BTS using associated and dedicated control channels during or not during a call, They are of three types:



- **- (SDCCH STAND ALONE DEDICATED CONTROL CHANNEL:**
  Stand alone dedicated control channel is a physical channel that dedicated by the system to the specified subscriber to transmit alone some important information like the frequencies for the traffic channel and the time slot (for up- link and down link) that specified for the subscriber to access the call, after end this job this channel are released.
- **- (SAACH)SLOW ASSOCIATED CONTROL CHANNEL:**
  Non-urgent information, e.g. transmitter power control, is transmitted using the slow associated control channel (SACCH). On the uplink MS sends averaged measurements on own base station (signal strength and quality) and neighboring base stations (signal strength). On the downlink the MS receives system information, which transmitting power and what timing advance to use.

It is transmitted at 13thFrame of TCH. As seen, SACCH is transmitted on both up-and downlink, point-to-point. It uses normal burst.

- **- (FACCH) FAST ASSOCIATED CONTROL CHANNEL:**
  More urgent information, e.g. a handover command, is sent using time slots that are 'stolen' from the traffic channel. If, suddenly, during the conversation a handover must be performed the Fast Associated Control channel, FACCH, is used. FACCH works in stealing mode, meaning that one 2. ms segment of speech is exchanged for signaling information necessary for the handover.

Note :- SACCH and FACCH are work during the call process while the SDCCH work before the starting the call.

4 – Burst Structures for signaling channels.

Each type of signing channels have burst structure different from the traffic burst structure that illustrated previously. These burst structures are shown below

Normal

3 start bits	58 bits of encrypted data	26 training bits	58 bits of encrypted data	3 stop bits	8.25 bits guard period

FCCH burst

3 start bits	142 fixed bits of all zeroes	3 stop bits	8.25 bits guard period

SCH burst

3 start bits	39 bits of encrypted data	64 bits of training	39 bits of encrypted data	3 stop bits	8.25 bits guard period

RACH burst

8 start bits	41 bits of synchronization	36 bits of encrypted data	3 stop bits	68.25 bit extended guard period

Dummy burst

3 start bits	58 mixed bits	26 training bits	58 mixed bits	3 stop bits	8.25 bits guard period

The diagram below illustrate the location of the signaling channels in the signaling framing structure .

FN	TS-0	TS-1
0	FCCH	SDCCH/0
1	SCH	SDCCH/0
2	BCCH	SDCCH/0
3	BCCH	SDCCH/0
4	BCCH	SDCCH/1
5	BCCH	SDCCH/1
6	AGCH/PCH	SDCCH/1
7	AGCH/PCH	SDCCH/1
8	AGCH/PCH	SDCCH/2
9	AGCH/PCH	SDCCH/2
10	FCCH	SDCCH/2
11	SCH	SDCCH/2
12	AGCH/PCH	SDCCH/3
13	AGCH/PCH	SDCCH/3
14	AGCH/PCH	SDCCH/3
15	AGCH/PCH	SDCCH/3
16	AGCH/PCH	SDCCH/4
17	AGCH/PCH	SDCCH/4
18	AGCH/PCH	SDCCH/4
19	AGCH/PCH	SDCCH/4
20	FCCH	SDCCH/5
21	SCH	SDCCH/5
22	SDCCH/0	SDCCH/5
23	SDCCH/0	SDCCH/5
24	SDCCH/0	SDCCH/6
25	SDCCH/0	SDCCH/6
26	SDCCH/1	SDCCH/6
27	SDCCH/1	SDCCH/6
28	SDCCH/1	SDCCH/7
29	SDCCH/1	SDCCH/7
30	FCCH	SDCCH/7
31	SCH	SDCCH/7
32	SDCCH/2	SACCH/0
33	SDCCH/2	SACCH/0
34	SDCCH/2	SACCH/0
35	SDCCH/2	SACCH/0
36	SDCCH/3	SACCH/1
37	SDCCH/3	SACCH/1
38	SDCCH/3	SACCH/1
39	SDCCH/3	SACCH/1
40	FCCH	SACCH/2
41	SCH	SACCH/2
42	SACCH/0	SACCH/2
43	SACCH/0	SACCH/2
44	SACCH/0	SACCH/3
45	SACCH/0	SACCH/3
46	SACCH/1	SACCH/3
47	SACCH/1	SACCH/3
48	SACCH/1	free
49	SACCH/1	free
50	free	free

Signaling Frame
Down link

FN	TS-2	. . .	TS-7
0	TCH		TCH
1	TCH		TCH
2	TCH		TCH
3	TCH		TCH
4	TCH		TCH
5	TCH		TCH
6	TCH		TCH
7	TCH		TCH
8	TCH		TCH
9	TCH		TCH
10	TCH		TCH
11	TCH		TCH
12	SACCH		SACCH
13	TCH		TCH
14	TCH		TCH
15	TCH		TCH
16	TCH		TCH
17	TCH		TCH
18	TCH		TCH
19	TCH		TCH
20	TCH		TCH
21	TCH		TCH
22	TCH		TCH
23	TCH		TCH
24	TCH		TCH
25	free		free
0	TCH		TCH
1	TCH		TCH
2	TCH		TCH
3	TCH		TCH
4	TCH		TCH
5	TCH		TCH
6	TCH		TCH
7	TCH		TCH
8	TCH		TCH
9	TCH		TCH
10	TCH		TCH
11	TCH		TCH
12	SACCH		SACCH
13	TCH		TCH
14	TCH		TCH
15	TCH		TCH
16	TCH		TCH
17	TCH		TCH
18	TCH		TCH
19	TCH		TCH
20	TCH		TCH
21	TCH		TCH
22	TCH		TCH
23	TCH		TCH
24	TCH		TCH
25	free		free

Traffic Frame
Down link

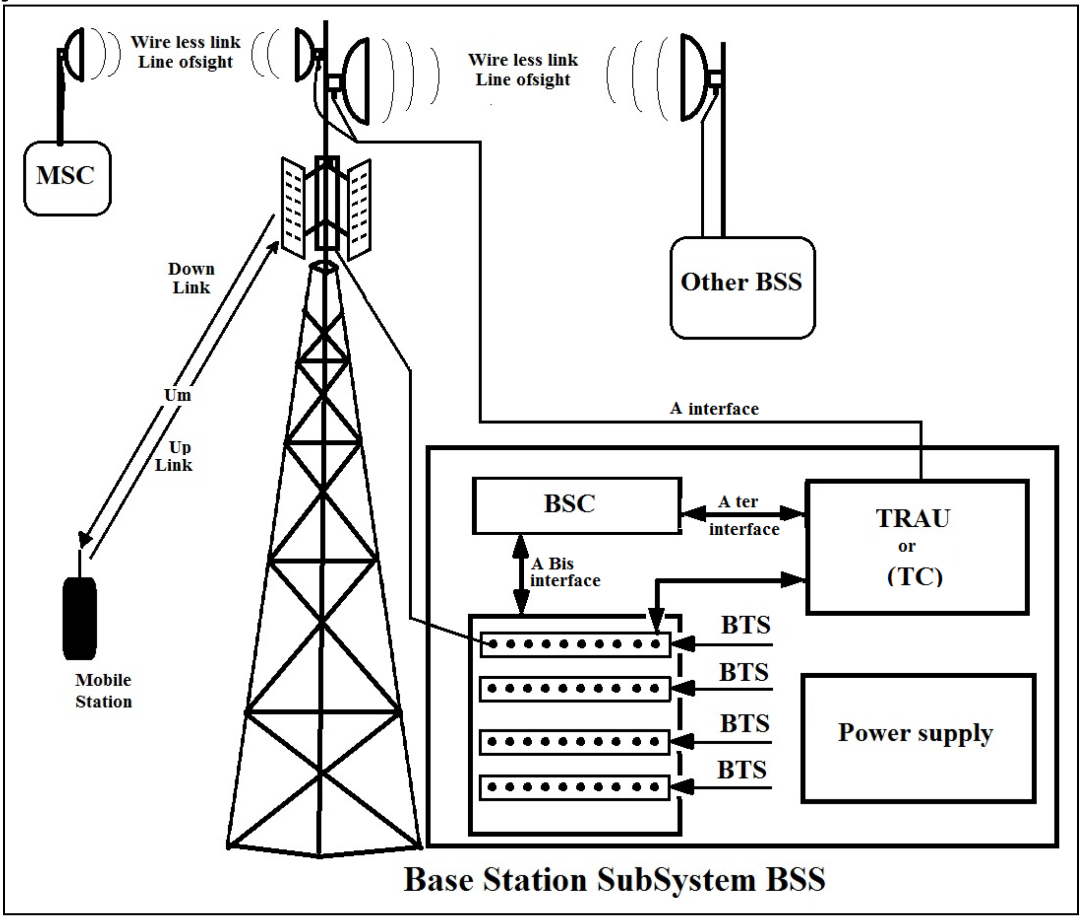
Communication Systems

Mobile Communication Systems

Lecture (6-2) :-Base Station Subsystem (BSS)

The Base Station Subsystem is responsible for managing the radio network, and it is controlled by an MSC. Typically, one MSC contains several BSSs. A BSS itself may cover a considerably large geographical area consisting of many cells (a cell refers to an area covered by one or more frequency resources). The BSS consists of the following elements:

- **BSC** Base Station Controller
- **BTS** Base Transceiver Station
- **TRAU** Transcoder and Rate Adaptation Unit (often referred to as TC (Transcoder))
- **Antennas**
- **Power Supply**



- ***-The Base Station Controller (BSC)***

Each Base station subsystem (BSS) contain only one Base Station Controller (BSC), the BSC is the central network element of the BSS and it controls the radio network. It has several important tasks, some of which are presented in the following:

1 - BTS and TRAU control

2 - Connection establishment between the MS and the NSS

3 - Mobility management

4 - Statistical raw data collection

5 - Air- and A-interface signalling support

- **- BTS and TRAU control**

Inside the BSS, all the BTSs and TRAUs are connected to the BSC(s). The BSC maintains the BTSs. In other words, the BSC is capable of separating (barring) a BTS from the network and collecting alarm information. TRAUs are also maintained by the BSC, that is, the BSC collects alarms related to the transcoders.

- **- Connection establishment between the MS and the NSS**

All calls to and from the MS are connected through the switching functionality of the BSC.

- **- Mobility management**

The BSC is responsible for initiating the vast majority of all handovers, and it makes the handover decision based on, among others, measurement reports sent by the MS during a call.

- **- Statistical raw data collection**

Information from the Base Transceiver Stations, Transcoders, and BSC are collected in the BSC and forwarded via the DCN (Data Communications Network) to the NMS (Network Management Subsystem), where they are post-processed into statistical views, from which the network quality and status is obtained.

- **-Air- and A-interface signalling support**

  In the A-interface, (Common Channel Signalling System) is used as the signalling language, while the environment in the air interface allows the usage of a protocol adapted from ISDN standards, namely LAPDm (Link Access Protocol on the ISDN D Channel, modified version). Between the Base Transceiver Station and the BSC (Abis interface), a more standardised LAPD protocol is used. The BSC also enables the transparent signalling connection needed between the MSC/VLR and the MS.

- ***- The Base Transceiver Station (BTS)***

The BTS is the network element responsible for maintaining the air interface and minimizing the transmission problems, it is houses the radio transceivers that define a cell and handles the radio link protocols with the MS. In a large urban area, a large number of BTSs may be deployed.

The BTS corresponds to the transceivers and antennas used in each cell of 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 1 and 16 transceivers, depending on the density of users in the cell. The BTS also connected to one or more antennas, which are capable of transmitting and receiving information to/from one or more TRXs. The antennas are either **omnidirectional or sectorised**. It also has control functions for Operation and Maintenance (O&M), synchronisation and external alarms, etc. Each BTS serves as a single cell or single sector from cell. It also includes the following functions:

1 - Air interface signalling

2 – Ciphering and Deciphering.

3 – Speech Processing.

4 - **Modulation and De-modulation**

- **- Air interface signalling** .

  A lot of both call and non-call related signalling must be performed in order for the system to work. One example is that when the MS is switched on for the very first time, it needs to send and receive a lot of information with the network (more precisely with the VLR) before we can start to receive and make phone calls. Another example is the signalling required to set up both mobile originated and mobile terminated calls. A third very important signalling in mobile networks is the need to inform the MS when a handover is to be performed (and later when the MS sends a message in the uplink direction telling the network that the handover is completed.

- **- Ciphering and Deciphering**

  Both the BTS and the MS must be able to cipher and decipher information in order to protect the transmitted speech and data in the air interface.

- **Speech processing**

  Speech processing refers to all the functions the BTS performs in order to guarantee an error-free connection between the MS and the BTS. This includes tasks like speech coding (digital to analogue in the downlink direction and vice versa), channel coding (for error protection), interleaving (to enable a secure transmission), and burst formatting (adding information to the coded speech / data in order to achieve a well-organised and safe transmission).

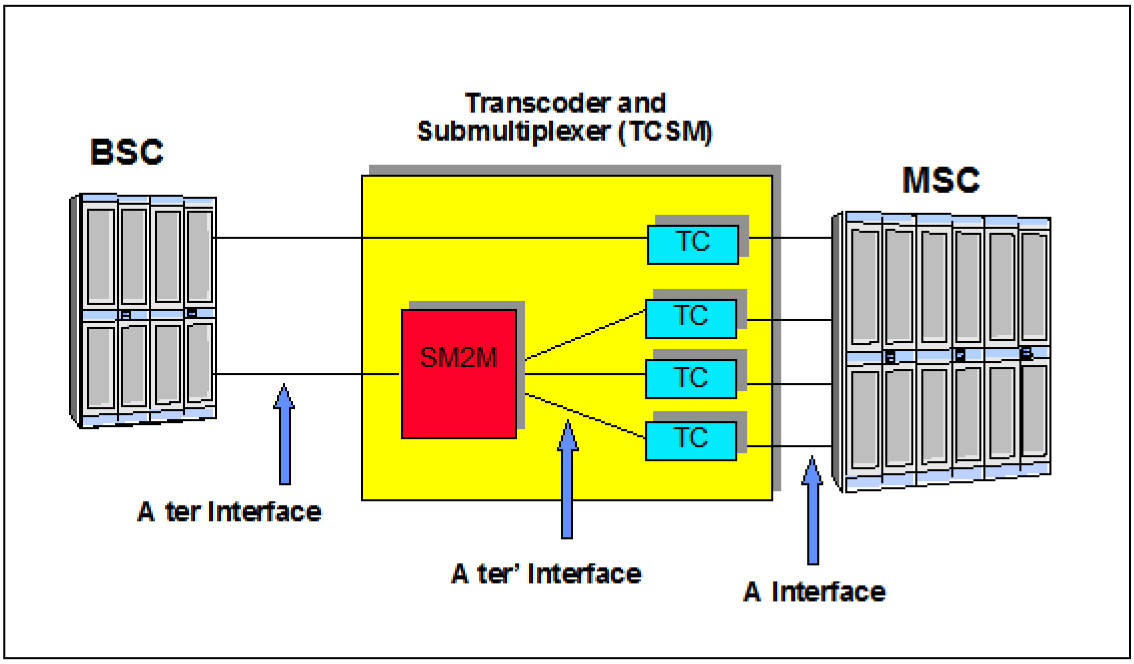
- **- Modulation and De-modulation**

  User data is represented with digital values 0 and 1. These bit values are used to change one of the characteristics of an analogue radio signal in a predetermined way. By altering the characteristic of a radio signal for every bit in the digital signal, we can "translate" an analogue signal into a bit stream in the frequency domain. This technique is called **modulation**. In GSM, Gaussian Minimum Shift Keying (GMSK), is applied.

- ***<u>- Transcoder and Rate Adaptation Unit (TRAU)</u>***

In the air interface (between MS and BTS), the media carrying the traffic is a radio frequency. To enable an efficient transmission of digital speech information over the air interface, the digital speech signal is compressed. We must however also be able to communicate with and through the fixed network, where the speech compression format is different. Somewhere between the BTS and the fixed network, we therefore have to convert from one speech compression format to another, and this is where the Transcoder comes in.

For transmission over the air interface, the speech signal is compressed by the mobile station to **13 kbit/s (Full Rate and Enhanced Full Rate), 5.6 kbit/s (Half Rate),** or **12.2 kbit/s (Enhanced Full Rate).** A more modern speech codec is the **AMR (Adaptive Multirate Coding)** which is more flexible since it produces speech with bitrates similar to older solutions but adapted to link conditions**.** However, the standard bit rate for speech in the PSTN is **64 Kbits/s**. The modulation technique is called **"Pulse Code Modulation" (PCM)**. This requires the GSM network to perform bit rate adaptation of speech.



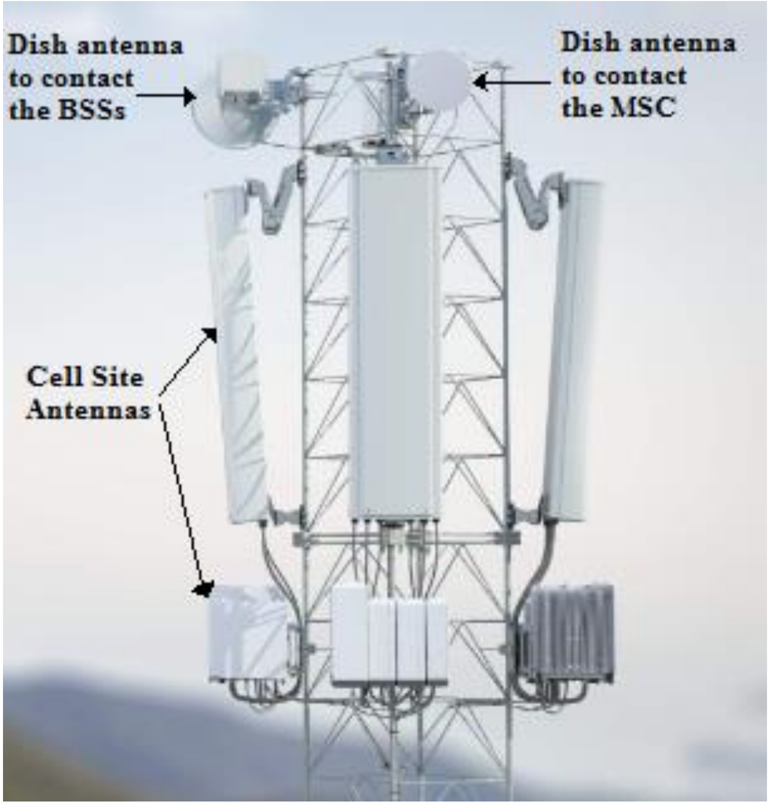
The TRAU thus takes care of the change from one bit rate to another. If the TC is located as close as possible to the MSC with standard PCM lines connecting the network elements, we can, in theory, multiplex four traffic channels in one PCM channel. This increases the efficiency of the PCM lines, and thus lowers the costs for the operator. When we connect to the MSC, the multiplexed lines have to be de-multiplexed. For that reason, the Nokia solution of the TRAU is called **Transcoder and Submultiplexer (TCSM**).

- ***- Antenna System***

The antenna system is one of the most crucial element in a radio access network. The performance of the components in the antenna system ultimately determines the effectiveness of the radio site, while a poor antenna system can severely reduce the radio network performance. Today's complex antenna systems with multiple frequencies and multiple technologies on the same radio site, put great demands on the ability to create effective antenna system solutions.

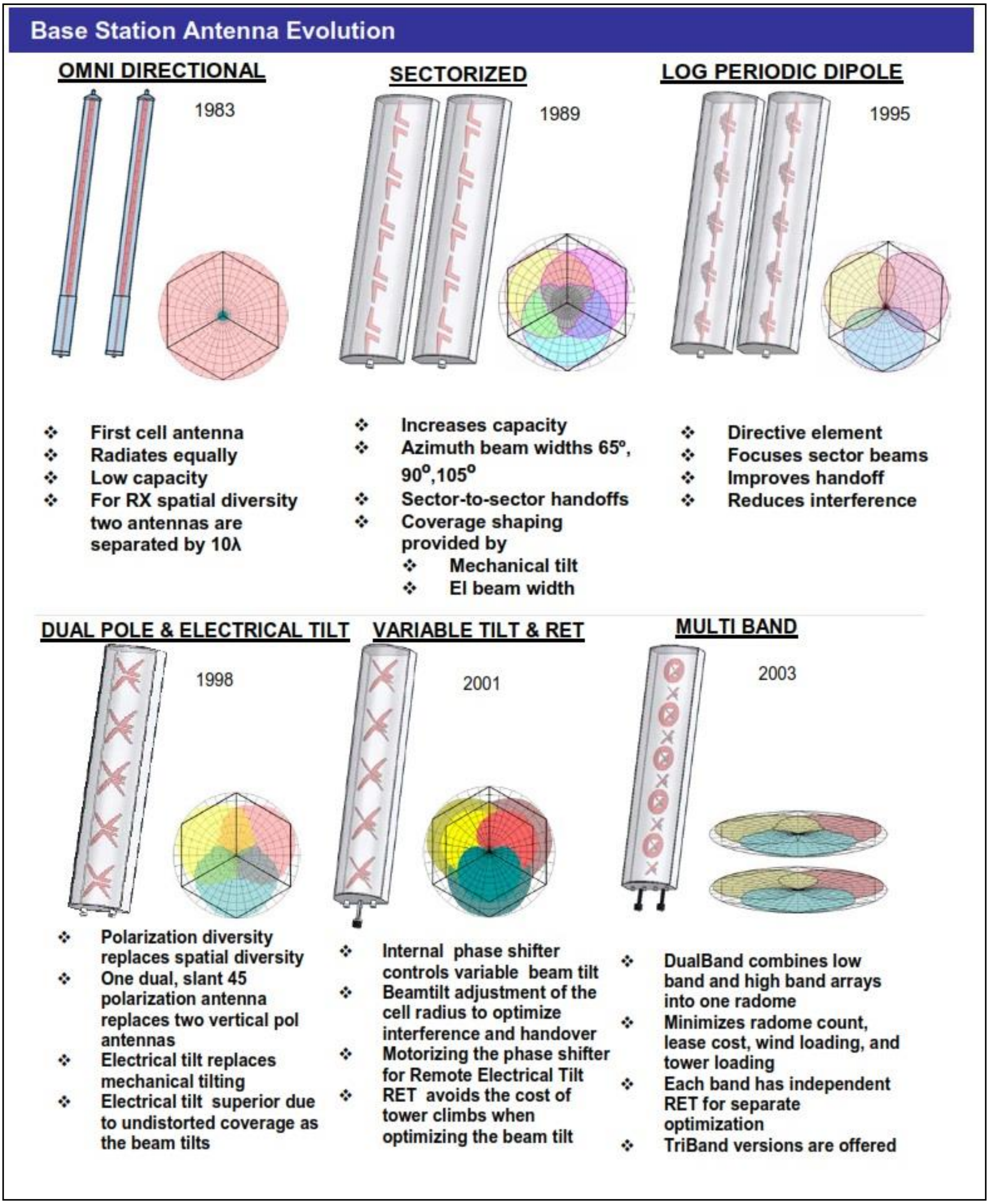
Antennas are part of every aspect of the GSM process, whether in the phones themselves, in signal towers or in satellites. In the United States and Canada, 850 and 1,900 MHz communication bands are used, while in other parts of the world, 900 and 1,800 MHz bands are the norm. Antennas used in the GSM process must conform to the bandwidths employed in the areas where they'll operate, with spectrum usage being negotiated by the ITU (International Telecommunications Union).

Three types of antennas or more can be found at the tower of the base station system, One of them are the antennas that conform with the link between the mobile station and base station that called cell site antennas. The other are used to access the contact between two base stations or between the base station and the mobile switching center (MSC).



Two size of dish antenna can be seen in the above of the BSS tower the big size are for achieve the link between the base station with other base stations ,, and this work at 20 GHz , while the small size dish antenna use to achieve the link between the BSS and the MSC and they work at 40 GHz carrier frequency .

The cell site antenna presents in two main types , the big size antenna are for 850 or 900 GSM, while the small size antenna are for 1800 0r 1900 GSM ..,these antennas may be omidirectional antenna that serve all the cell with radiation pattern 360 degree or sector antennas that coverage 60, 90, 120 or 180 degree, the sector antenna in fact is a two dimension array antennas and can be tilting in different angles, usually we can a group of three antennas in the cell center, used to coverage the cell area, each of this antenna cover 120 degree in the azimuth and about 36 degree in the elevation, this antennas usually have tilting angle down toward the earth. The modern design of these antennas are multiband and dual polarization radiation, and has electrical tilt ability.



- ## Antenna Diversity

The antenna diversity is one of main methods to avoid the fading phenomena in mobile communication that appear due to multipath effect in the receiver (mobile station). Antenna diversity, also known as **space diversity** or **spatial diversity**, is any one of several wireless diversity schemes that uses two or more antennas to improve the quality and reliability of a wireless link.

Antenna diversity can be realized in several ways. Depending on the environment and the expected interference, designers can employ one or more of these methods to improve signal quality. In fact multiple methods are frequently used to further increase reliability.

- **Spatial diversity** employs multiple antennas, usually with the same characteristics, that are physically separated from one another. Depending upon the expected incidence of the incoming signal, sometimes a space on the order of a wavelength is sufficient. Other times much larger distances are needed.
- **Pattern diversity** consists of two or more co-located antennas with different radiation patterns. This type of diversity makes use of directional antennas that are usually physically separated by some (often short) distance. Collectively they are capable of discriminating a large portion of angle space and can provide a higher gain versus a single omnidirectional radiator.
- **Polarization diversity** combines pairs of antennas with orthogonal polarizations (i.e. horizontal/vertical, ± slant 45°, Left-hand/Right-hand circular polarization etc.). Reflected signals can undergo polarization changes depending on the medium through which they are traveling. A polarization difference of 90° will result in an attenuation factor of up to 34 dB in signal strength. By pairing two complementary polarizations, this scheme can immunize a system from polarization mismatches that would otherwise cause signal fade.

- ***- Power Supply***

  Three or more types of power supply must be provided to the base station system in order to continues services of this station. The main feeder provide the station from the electrical city establishment, special electrical generator, and battery base power supply are work together in order to insure continue station work.

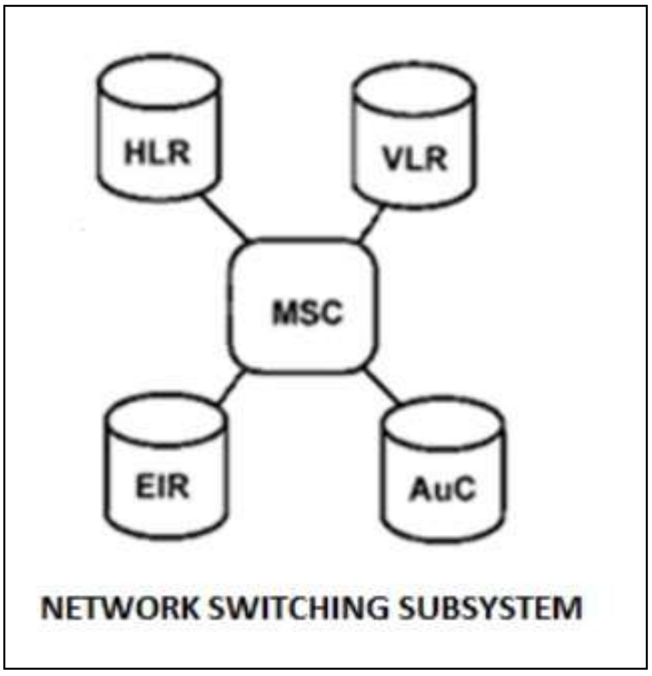
Communication Systems

Mobile Communication Systems

Lecture (7-2) :-NETWORK SWITCHING SUBSYSTEM (**NSS)**

The GSM system architecture contains a variety of different elements, and is often termed the core network. It provides the main control and interfacing for the whole mobile network. The major elements within the core network include:

1. MSC (Mobile Services Switching Centre)
2. HLR (Home Location Register)
3. VLR (Visitor Location Register)
4. EIR (Equipment Identity Register)
5. AUC (Authentication Centre)



1. ***Mobile Services Switching Centre (MSC):***

The main element within the core network area of the overall GSM network architecture is the Mobile switching Services Centre (MSC). The MSC acts like a normal switching node within a PSTN or ISDN, but also provides additional functionality to enable the requirements of a mobile user to be supported. These include registration, authentication, call location, inter-MSC handovers and call routing to a mobile subscriber. It also provides an interface to the PSTN so that calls can be routed from the mobile network to a phone connected to a landline. Interfaces to other MSCs are provided to enable calls to be made to mobiles on different networks. The main jobs for that MSC include

A - Call handling that copes with mobile nature of subscribers

B - Management of required logical radio link channel during calls

C - Management of MSC-BSS signaling protocol

D - Handling location registration and ensure interworking between MS and VLR

E - Control of inter-BSS handovers

F - Acting as a gateway mobile switching center to interrogate the HLR and VLR

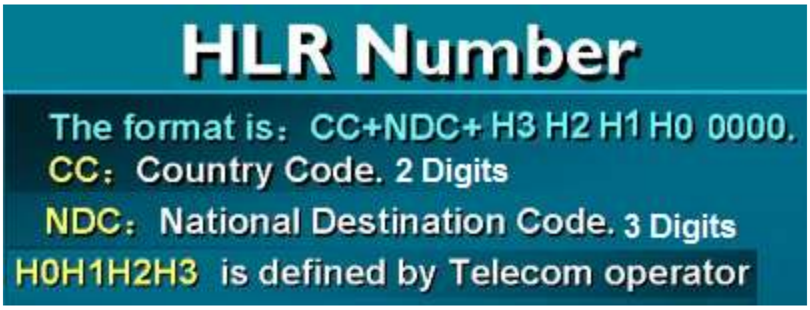
G – Exchange of signaling information with other system entities

H - Other normal functions of a local exchange switch in the fixed network

2. ***Home Location Register (HLR):***

The HLR are the huge capacity and very speedy Server that contain database about all the administrative information for each subscriber along with their last known location. In this way, the GSM network is able to route calls to the relevant base station for the MS. When a user switches on their phone, the phone registers with the network and from this it is possible to determine which BTS it communicates with so that incoming calls can be routed appropriately. Even when the phone is not active (but switched on) it re-registers periodically to ensure that the .network (HLR) is aware of its latest position. There is one HLR per network

The HLR identity number are shown below:-



The information that recorded in HLR include :-

A - Subscriber identity number (IMSI, MSISDN)

B – Current Subscriber VLR (current Location)

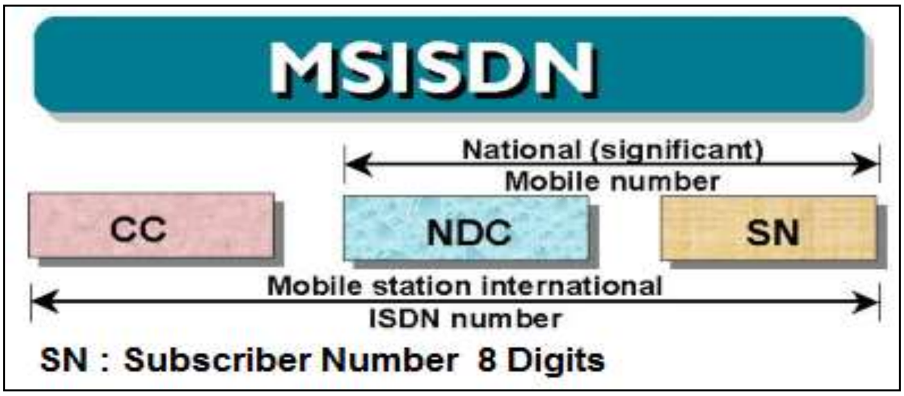
C – Supplementary service information.

D – Subscriber Status (registered , not registered)

E – Authentication Key generator for the subscribers.

F – Temporary mobile subscriber identity (TMSI)

G – Mobile Subscriber Roaming Number (MSRN)



3. ***Visitor Location Register (VLR):***

The VLR contains selected information for the visitor subscribers from the HLR that enables the selected services for the individual subscriber to be provided. The VLR can be implemented as a separate entity, but it is commonly realized as an integral part of the MSC, rather than a separate entity. In this way access is made faster and more convenient. The information that recorded in the VLR are

A - Identity of mobile subscriber

B - Any temporary mobile subscriber identity

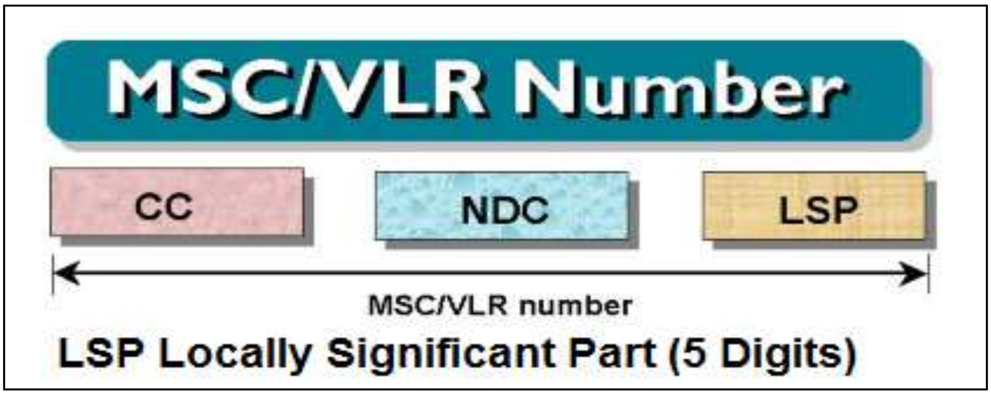
C - ISDN directory number of mobile

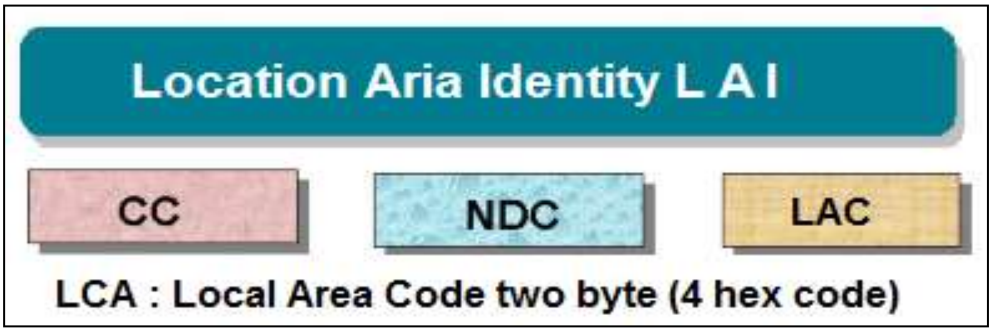
D - A directory number to route calls to a roaming station

E - Location area where mobile is registered (LAI)

F - Copy of subscriber data from HLR

The VLR identity number are shown below





4. ***Equipment Identity Register (EIR):***

The EIR is the entity that decides whether a given mobile equipment may be allowed onto the network. Each mobile equipment has a number known as the International Mobile Equipment Identity. This number, as mentioned previously, is installed in the equipment and is checked by the network during registration. Dependent upon the information held in the EIR, the mobile may be allocated one of three states - allowed onto the network, barred access, or monitored in case its problems. so the each mobile are listed into one of the following lists:

A - White list (Valid list)

B - Gray list (Suspect list)

C - Black list (Fraudulent list)

5. ***Authentication Centre (AUC):***

The authentication center unit (AUC) is a key component of the global system for mobile communications (GSM). The AUC authenticate any subscriber (SIM) card which attempting to access the network are real subscriber and not spy or hacker on the network also this unit achieve the ciphering of the speech data before transmit it in the air space.

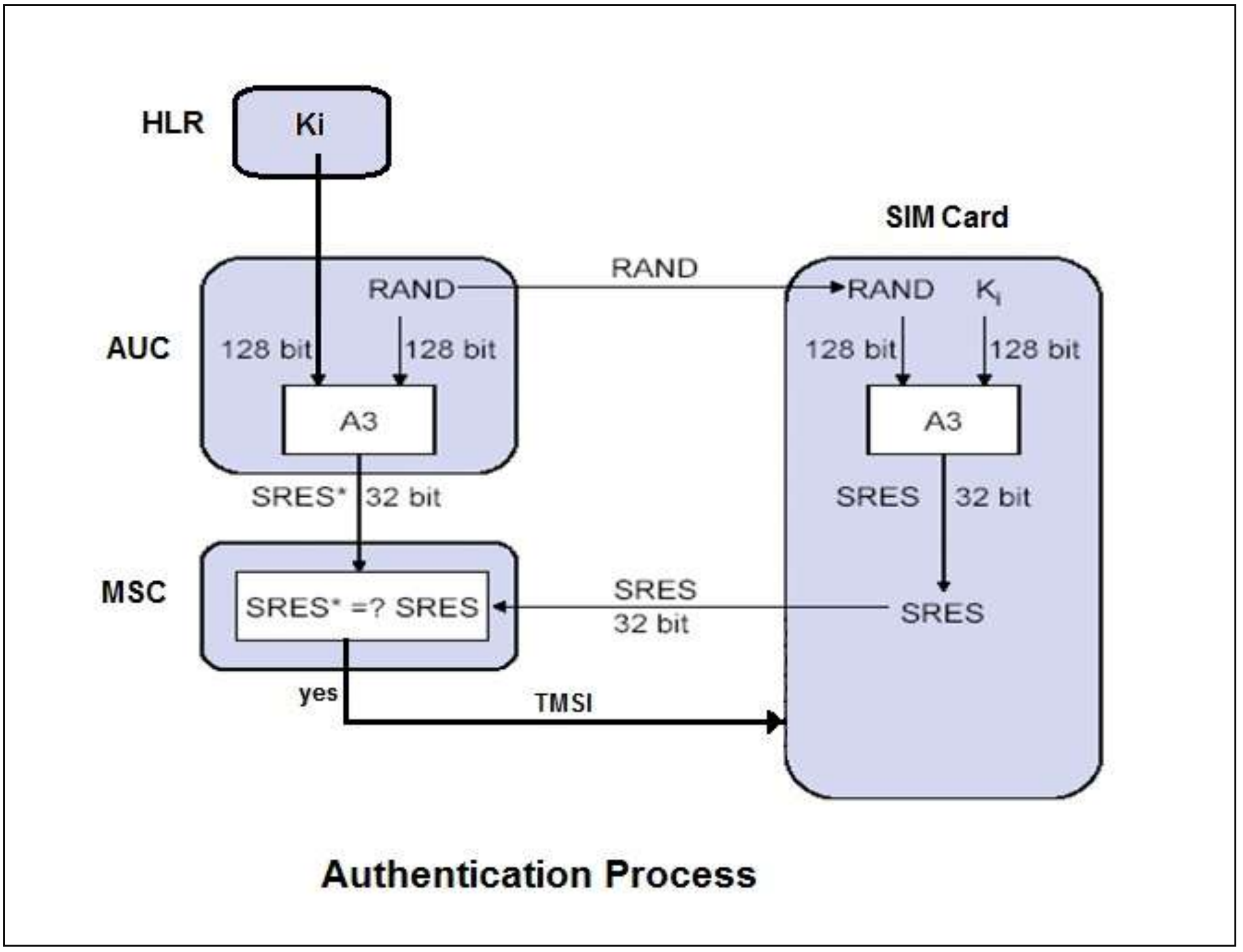
- ### - The Authentication Process

the authentication process are happened at each access to the GSM network by the subscriber, the network allow the subscriber to access the network in case of authentication process are successful, else the access request shall refused and the system end the connection immediately.

The authentication process done by comparing two sets of serial numbers by the MSC, one generated in the authentication unit and the other generated by the SIM card within the mobile station and send to the MSC through the BSS system and the wireless link(UM).

The AUC work together with the HLR and the MSC to generate the first set of (32 bits) serial numbers , the authentication unit get the(Ki) number for the subscriber is get from the HLR when the HLR receive a request from the MSC (this request include the IMSI number)while the authentication unit contain the necessary secret algorithms for authentication (A3) also it contain the random numbers generator.
the (A3) algorithm use the (Ki) and the (128 bits) random numbers which generated by the authentication unit it self as an inputs to generate the first set of serial numbers and send to the MSC in order to compression with the second set that shall come from the SIM card.
The second (32 bit) serial numbers set are generated in the SIM card, also the SIM card contain the same secret algorithm (A3) and the key (Ki) but its need to the(128 bits) Random number to complete the generation of the second set of the (32 bits) serial numbers, the SIM card get the random numbers from the authentication unit through the MSC and generate the (32 bits) serial number, send them to the MSC and the wireless (UM) link and compare this set with that generated in the authentication unit as illustrate previously.

If the two sets of the (32 bits) serial numbers that arrived to the MSC are equal, then authentication process are successful and the MSC send the Temporary mobile subscriber identity (TMSI) as a sign of the successful authentication process, and to use this (TMSI) in the ciphering the speech signal during the call. Else if the two sets of serial numbers are not identical the system refuse the subscriber request of access the network and end the connection directly.



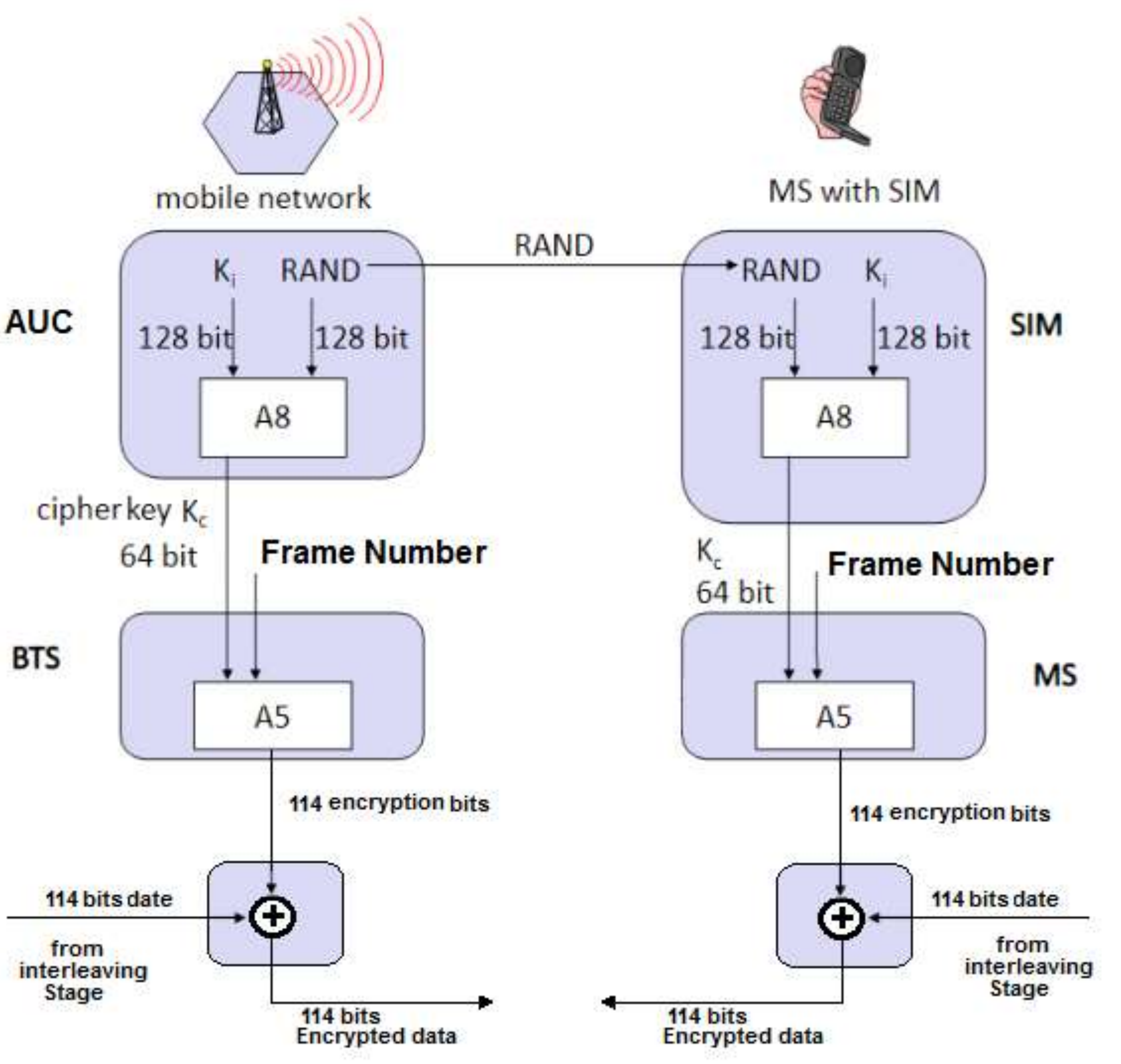
- ## - Ciphering Process

After authentication process complete successfully the ciphering process start immediately to generate (114 encryption bits that used for encrypted and de encrypted the speech signal as illustrate in the speech signal processing section in the previous lectures.

this (114 encryption bits) are treated using (modulo 2 addition) with the 114 bit for the speech signal that came from the interleaving stage. the ciphering process done in both the Mobile switching system MSS and the SIM card.

The generation of the (114 encryption bits) are achieved in the authentication unit and the SIM Card using the same proceedings steps using the two ciphering algorithms (A8 and A5) that stored in the authentication unit and the SIM card together in addition of the (A3 Authentication algorithm).

The (A8) algorithm use the same (128 bits) random number (that used in the authentication process) and the (Ki) as an inputs to generate the new (64 bits Ciphering Key (KC)), the (A5) algorithm use this (64 bits (KC)) with the TDMA Frame number as an inputs to generate the (114 encryption bits). this mean that encryption bits are changed with every frame.



Ciphering Process

Communication Systems

Mobile Communication Systems

Lecture (8-2) :-The OMC, GSM Additional Functional Elements, GSM Interface, Abis Interface, and Battery Life.

- ***- Operations And Maintenance Center (OMC)***

The operation and maintenance center is connected to all equipment in the GSM system except the mobile station (MS), it is

They can be represented into two parts, Operation and maintenance center switching part(OMC-S) and Operation and maintenance center radio part(OMC-R). in some where the OMC is called the operation and support system (OSS). Here are some of the OMC functions:

A. Planning The Network

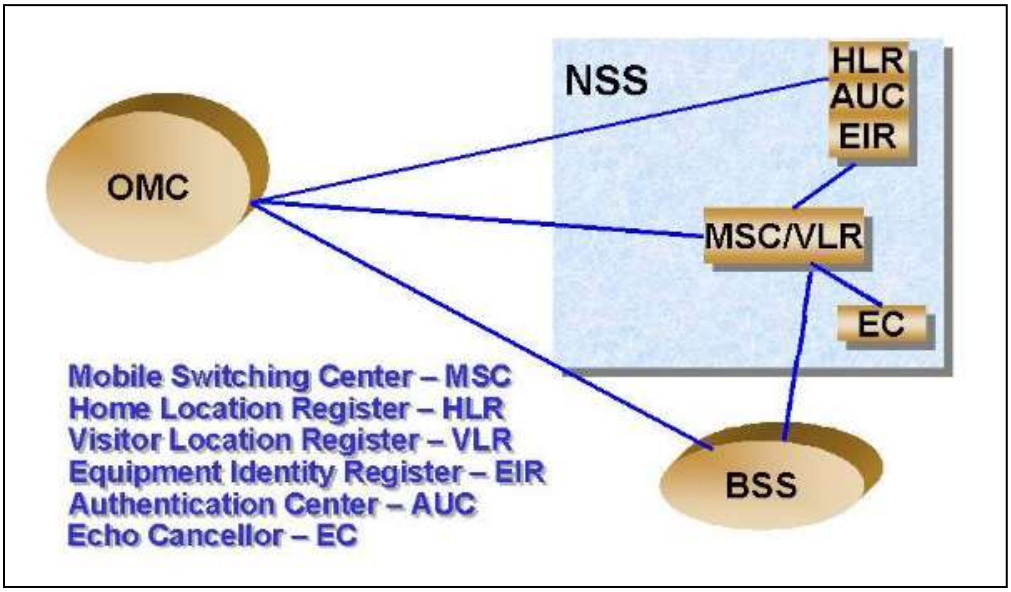
B. Operating The Network

C. Maintenance The Network

D. Supervising The Network

E. Developing The Network

F.Security Management.



- ***- GSM Additional Functional Elements***

1 – Massage Center (MXE)

The MXE is a node that provides integrated voice, Fax, and data messaging. Specifically, the MXE handles short message service, cell broadcast, voice mail, fax mail, email, and notification.

2 – Mobile Service Node (MSN)

The MSN is the node that handles the mobile intelligent network (IN) services..

3 – Gateway Mobile Services Switch Center (GMGC)

A gateway is a node used to integrate two networks, the gateway is often implemented in a MSC. The MSC is then referred to as the GNS

4 – GSM Interworking Unit (GIWU)

The GIWU consist of both hardware and software that provides an interface to various networks for data communications. Through the GIWU, users can alternate between speech and data during the same call. The GIWU hardware equipment is physically located at the MSC/VLR. This unit may called interworking function (IWF). The basic feature for this unit is data rate conversion function and protocol adaptation function.

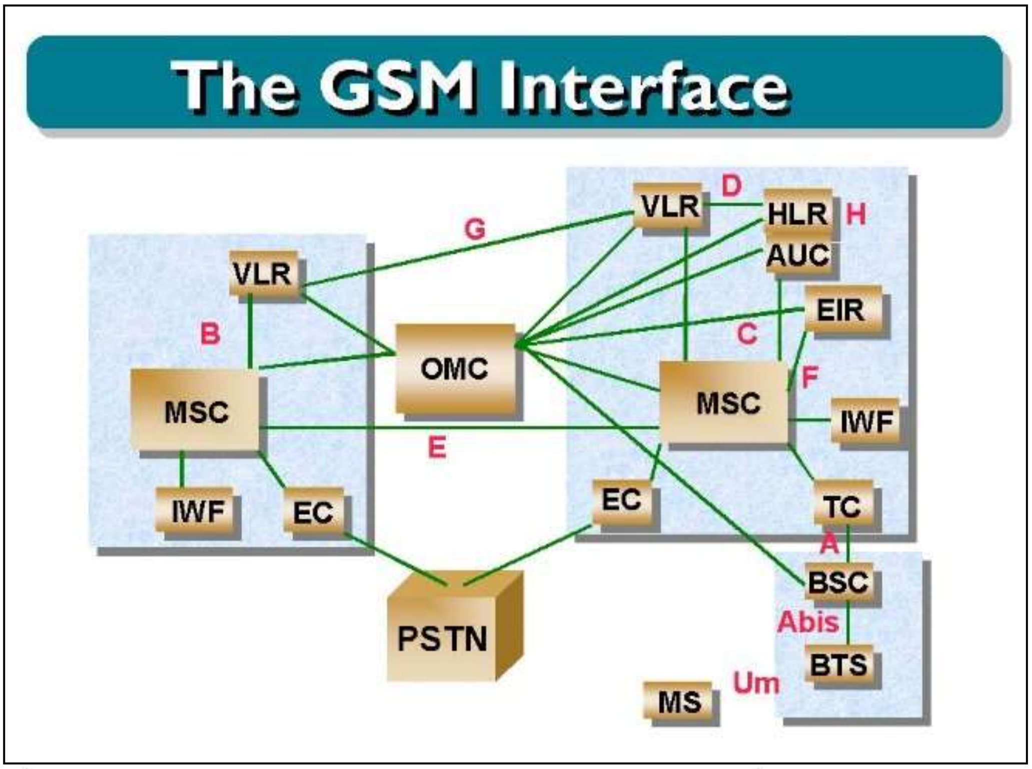
5 – Echo canceller (EC)

An Echo canceller is used between PSTN and MSC for all voice circuits. Echo canceller is required at the MSC because the GSM system delay can cause an unacceptable echo condition even on a short distance with PSTN.

The GSM system delay may be the call processing, speech encoding and decoding etc. the total round trip delay is approximately 180 ms. This would not be apparent to the call between MSs. But this case will be very different for the call between MS and PSTN subscriber. Thus Echo canceller is required between the MS and the PSTN.

- ## *- The GSM Interface*

Each GSM component is designed to communicate over an interface specified by the GSM standards. This provides flexibility and enables a system operator to adopt system component from different manufactures. Each interface within the GSM system has a specified name associated with it. this figure and table below illustrates the name of all the interfaces specified by GSM.

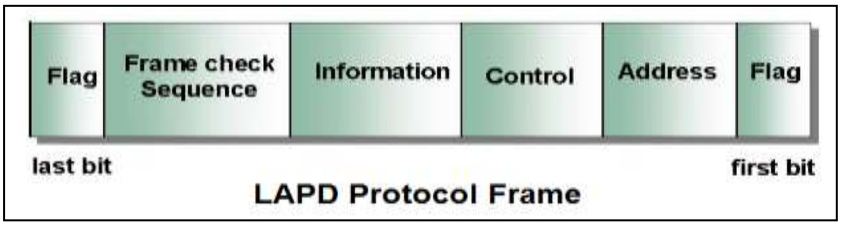


Name	Interface
Um	MS ⇔ BTS
Abis	BTS ⇔ BSC
A	BSC ⇔MSC
B	MSC⇔VLR
C	MSC⇔HLR
D	VLR ⇔HLR
E	MSC⇔MSC
F	MSC⇔EIR
G	VLR ⇔VLR
H	HLR ⇔AUC

- Abis Interface

Abis is the interface between BSC and BTS. Because the GSM specifications for this interface are not very specific, therefore Abis interface is not an open and standard interface. Thus, the Abis interface of different manufactures is varied. This means that one manufacturer's BTS will not work with another manufacture's BSC. That is to say BSC and BTS must come from the same manufacture.

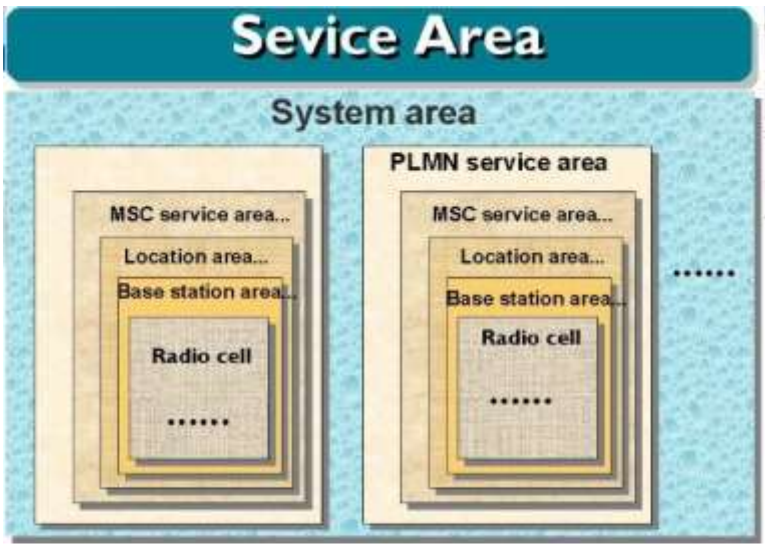
Through Abis interface, a different type of protocol is required GSM has specified the use of LAPD protocol in Abis interface.The LAPD protocol uses the standard frame structure shown below.



The first and last bit are flag bits. Between them, there are frame check, sequence. Information, control and address.

- Service Area

Mobile system service area is subdivided into following six levels:- System area, Public Land Mobile Network (PLMN) service area, MSC service area, Base Station area, and radio cell service area.



1 – System Area

System area is comprises one or more international PLMN service area, in this area, the other users from PLMN,PSTN and ISDN can connect with the MS without knowing where it is.

2 – PLMN (public land mobile network) service area

PLMN service area is served by one operator. The PLMN service area is served by one ore more HLR. There are the same numbering plan and the routing plan in PLMN service area comprises one or more MSC service area.

3 – MSC service area

MSC service area is served by one SC. It may be apart of a city or an entire country area. MSC service area Comprises one or more location area.

4 – Location area

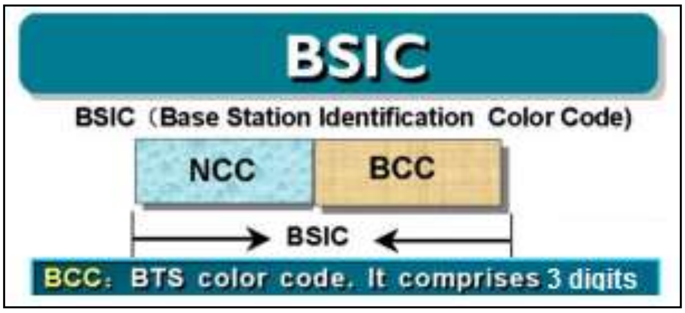
Location area is determined by the operator to fulfill the requirements imposed by traffic and flow, population and subscriber mobility. Location area comprises one or more Base station area

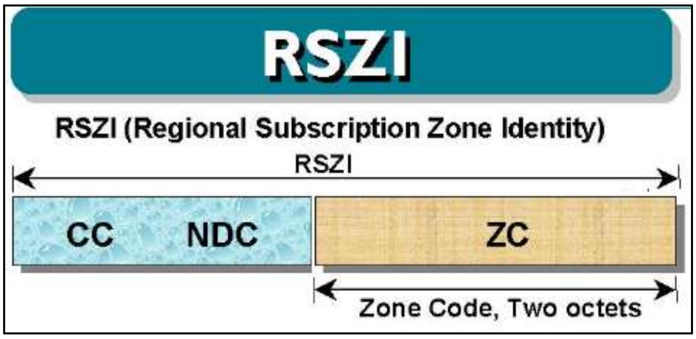
5 – Base station area

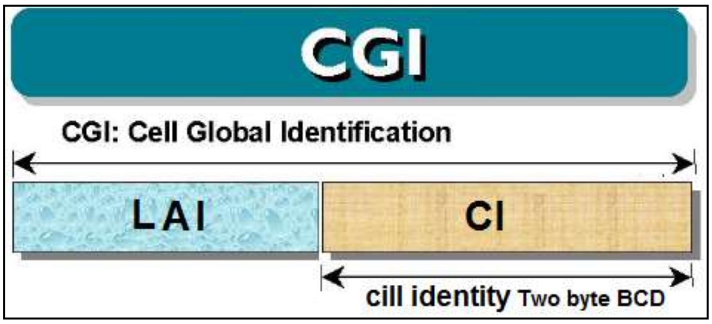
Base station area is served by one or more BTS. It consists one cell if the antenna is omni-directional antenna, or more than one cell if the antenna is directional. Base station area comprises one or more radio cell.

6 – Radio cell service area

Radio cell is served by a directional or omni-directional antenna. It is the smallest service area in GSM system. In radio cell specific radio channel devices are used for each radio connection.







- ***-Battery Life***

In GSM system, one of the main factors which restrict reducing the size of a mobile station is the battery.

A battery must be large enough to maintain a telephone call for an acceptable amount of time without needing to be recharged. Since there is demand for mobiles to become smaller and lighter the battery must also become smaller and lighter. For features which enable the life of a GSM mobile battery to be extended.

1 – power control

2 – Voice activity detection (VAD)

3 – Discontinuous Transmission (DTX)

4 – Discontinuous Reception (DRX)

- - Power control is a feature of GSM air interface which allows operator to not only compensate for distance from mobile to BTS (the case of short distance between the MS and BTS need low transmitted power, while the large distance between them need high transmitted power) as a regards timing, but can also cause the BTS and mobile to adjust their power output. This feature saves the radio battery power at the mobile, and help to reduce co-channel and adjust channel interface.
- - Voice Activity Detection(VAD) is mechanism whereby the source transmitter equipment identifies the present or absence of speech.
- - Discontinuous Transmission (DTX).Each subscriber occupied one time slot from the frame (8 time slots), so the real time for transmitting process are 1/8.
- - Discontinuous reception (DRX). Each subscriber occupied one time slot from the frame (8 time slots), so the real time for reception process are 1/8.

Communication Systems

Mobile Communication Systems

Lecture (9-2) :- HANDOVER IN GSM

when an active mobile moves within the coverage area of a network. That will lead to situations where the MS leaves the coverage area of a single cell, in this case there is a need for a feature to transfer an ongoing call from a physical channel to another without dropping the call, this feature is the handover.

In fact the traveling between two coverage area is not the unique reason to requesting handover, however there are many reasons that causes handover requests, these reasons are :-

- Power lever in handset is too low
- Signal/noise ratio is too low
- Bit-error-rate is too high
- Traveling from one cell to another
- During frequency hopping process
- During cell splitting

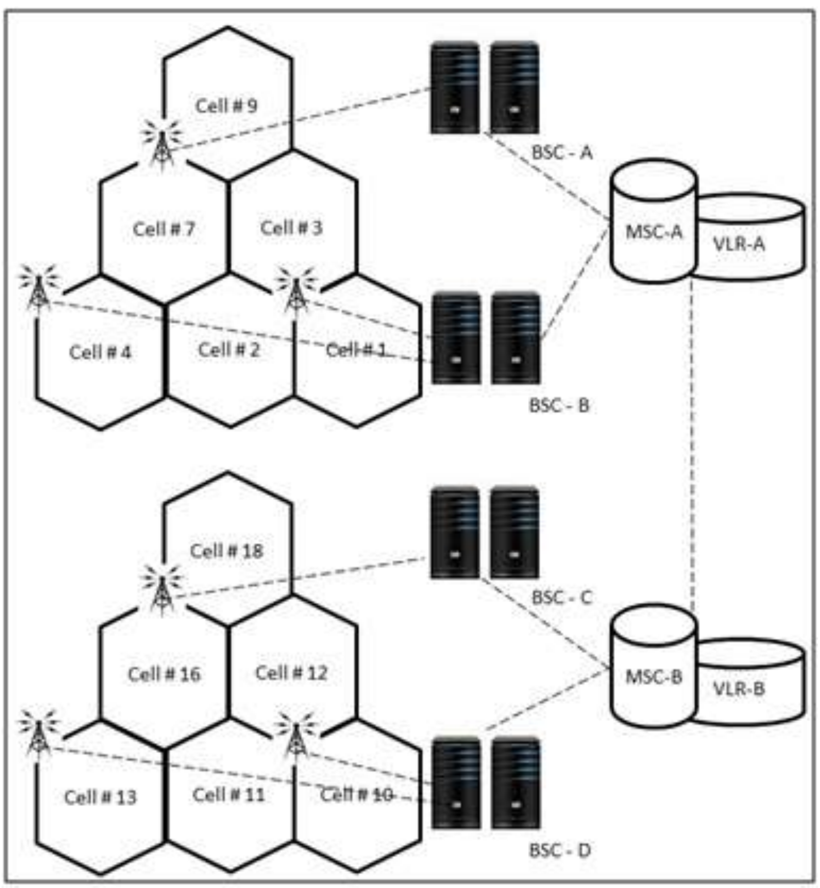
Handover is one of the basic features for cellular communication. It can be categorized into two types: Horizontal handover where it is performed in one network of the same radio access technology (GSM to GSM), and vertical where the handover is performed between different networks or between different radio access technologies (e.g. 3G to LTE). Moreover, the horizontal handover has two types: a hard handover where at any given time the call is handled by only one connection, this is the type used in GSM, and the default in LTE. The second type is the soft handover, where the MS is connected to more than one cell at the same time during the handover process.

Before the handover happens, certain conditions have to be met. Such conditions can be on a radio-environment nature, like the received signal level, and speech quality. It can be as well a network criterion, like cell traffic load. An active MS is continuously sending measurement reports to its serving cell, the need to a handover is then decided.

- ***-Handover classifications***

The handover in GSM is classified as a hard handover, where the established call is carried out by one radio channel, when the handover occurs the MS is ordered to release the current radio channel before connecting to a new one.

There are four classifications of horizontal handover in GSM, From Fig (1) the call data/signaling path is what distinguish them:



Figure(1) Handover diagram in GSM

1. **Intra-cell handover:**

In this special handover, while the MS is connected to a certain cell, a handover request is triggered, however not to another cell, but to the same cell with another physical channel (time slot) in the same carrier or another. The main objective behind this type of handover is due to high co-channel interference which can be seen as a low signal quality at the receiver, if the signal level is acceptable, then moving the call to another time slot, and/or frequency carrier can solve the problem giving the time division property of GSM signals (i.e. the co-channel transmitter might be using different time slots and/or carriers, therefore avoiding the interference).

As an example, in Fig (1) BSC-B sees an MS in cell number one with bad signal quality, however, no other cells can provide a better signal level, therefore BSC-B orders the MS – BTS in cell number one to switch to another radio channel.

2. **Inter-cell handover under the same BSC:**

Using **Fig (1) above,** BSC-B detects that an MS served by cell number (2) can have a better signal level if it is to be served by cell number (4), BSC-B controls both cells and has real time formation about them, therefore, it orders cell number (4) to assign a radio channel to an upcoming connection from an MS, then it sends information to the MS to be able to tune to cell number (4). Although the MS will use only one radio channel at any given time, the radio channel to cell number (1) will be kept reserved until the handover is successful to cell number (4), otherwise in case the handover procedure fails, the MS will try to reconnect to cell number (2) using the same old channel.

3. Inter-cell handover between different BSCs in the same MSC

This scenario can be seen in Fig (1) by an MS moving from cell number (3) which is controlled by BSC–B, to cell number (7)which is controlled by BSC-A, however both BSCs are connected to the same MSC-A. The handover procedure is the same in principle, however, BSC-B does not know which BSC controls cell number (7), and therefore it asks the MSC-A to find the target BSC. The MSC-A plays a switching role between the BSCs without taking any decision regarding the handover itself. Beside the switching role of the MSC, the handover procedure is identical to the one in the previous case.

4. Inter-cell handover between different BSCs in different MSCs

This case can be seen in Fig (1) as an MS moving from cell number(2) towards cell number (18),and a handover is required. BSC-B asks MSC-A to help finding which BSC controls cell number (18), however MSC-A does not have this information, therefore it sends a request to other MSCs, until it receives a positive message from MSC-B with the information that BSC-C controls cell number (18), then the handover procedure continues the same way as in the second way but involves additional switching nodes. It is worth noting that when more entities are involved in the handover procedure, the probability of a handover failure increases.

- ***Handover Initiation***

Handover initiation is the process of deciding when to request a handover. Handover decision is based on received signal strengths (RSS) from current BS and neighboring BSs. In Figure (2) below, RSSs of current BS (BS1) and one neighboring BS (BS2) is examined. The RSS gets weaker as MS goes away from BS1 and gets stronger as it gets closer to the BS2 as a result of signal propagation. The received signal is averaged over time using an averaging window to remove momentary fading due to geographical and environmental factors.

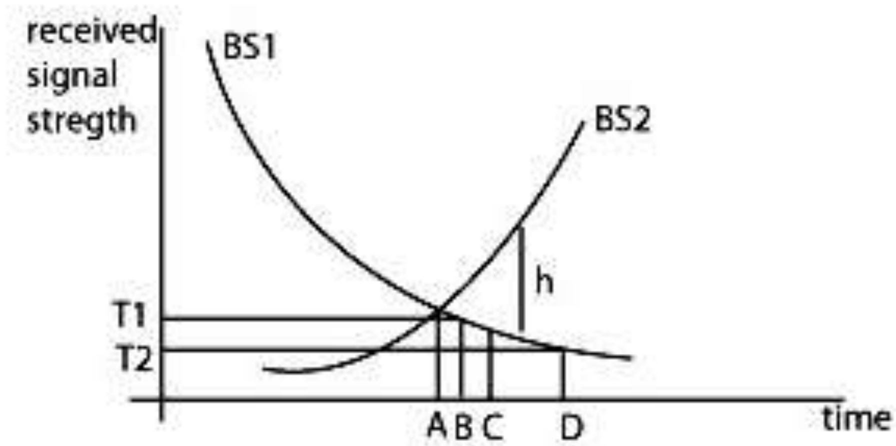


Figure (2) Movement of a MS in the Handover zoon

There are four main handover initiation techniques in GSM System:-

1. Relative Signal Strength

In relative signal strength, the RSSs are measured over time and the BS with strongest signal is chosen to handover. In Figure (2) BS2's RSS exceeds RSS of BS1 at point A and handover is requested. Due to signal fluctuations, several handovers can be requested while BS1's RSS is still sufficient to serve MS. These unnecessary handovers are known as ping-pong effect. As the number of handovers increase, forced termination probability also increases. So, handover techniques should avoid unnecessary handovers.

2. Relative Signal Strength with Threshold

Relative signal strength with threshold introduces a threshold value (T1 in Figure (2)) to overcome the ping-pong effect. The handover is initiated if BS1's

RSS is lower than the threshold value and BS2's RSS is stronger than BS1's.The handover request is issued at point B in Figure (2).

3. Relative Signal Strength with Hysteresis

This technique uses a hysteresis value (h in Figure (2)) to initiate handover. Handover is requested when the BS2's RSS exceeds the BS1's RSS by the hysteresis value h (point C in Fig. (2)).

4. Relative Signal Strength with Hysteresis and Threshold

The last technique combines both the threshold and hysteresis values concepts to come with a technique with minimum number of handovers. The handover is requested when the BS1's RSS is below the threshold (T1 in Figure (2)) and BS2's RSS is stronger than BS1's by the hysteresis value h (point C in Figure (2)). If a lower threshold than T1 (but higher than T2) is chosen then the handover initiation will be somewhere at the right of point C.

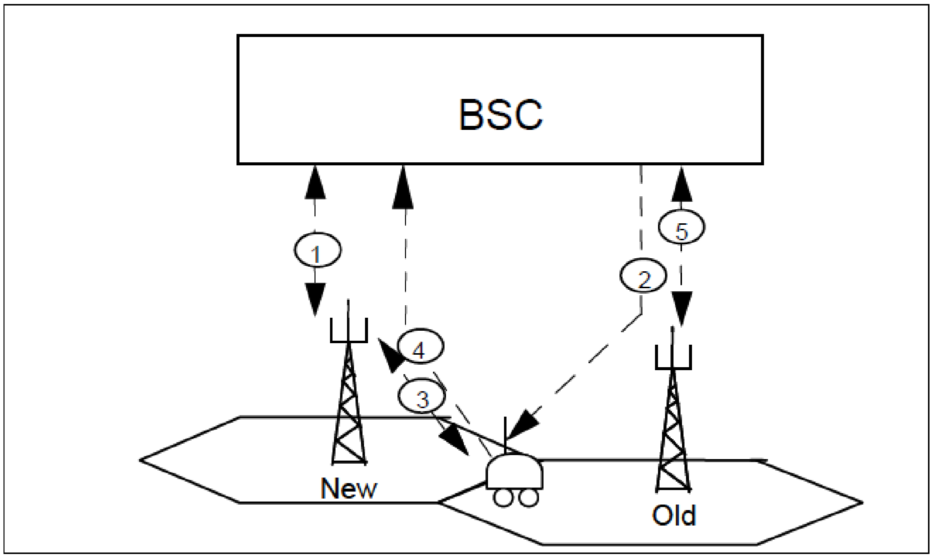
All the techniques discussed above initiate handover before point D where it is the "receiver threshold". Receiver threshold is the minimum acceptable RSS for call continuation (T2 in Fig. (2)). If RSS drops below receiver threshold, the ongoing call is than dropped. The time interval between handover request and receiver threshold enable cellular systems to delay the handover request until the receiver threshold time is reached when the neighboring cell does not have any empty channels.

- ***- Handover Procedure***

The measurement reports from the BTS and MS, together with system parameters set by the operator, are used in the preparation algorithm in the BSC. The outcome could be a handover if this is judged necessary. Three handover cases will be discussed. The difference between them is due to where the cells are located in the network structure, and thus how many nodes will be involved in the handover.

A - Intra- BSC Handover

In this case the handover is controlled by the BSC internally and the MSC will only be informed for statistical reasons (Figure (3)).



1. Activation of New Channel : BSC allocates a TCH in the new cell and orders that BTS to activate it. The chosen HO will be part of the activation message. The BTS in the new cell will acknowledge that the TCH has been activated.

2. Handover Command : After the activation the BSC commands (Figure (3)) the MS to change to the new channel. The message is sent on FACCH and will contain a full description of the new channel and the HO.

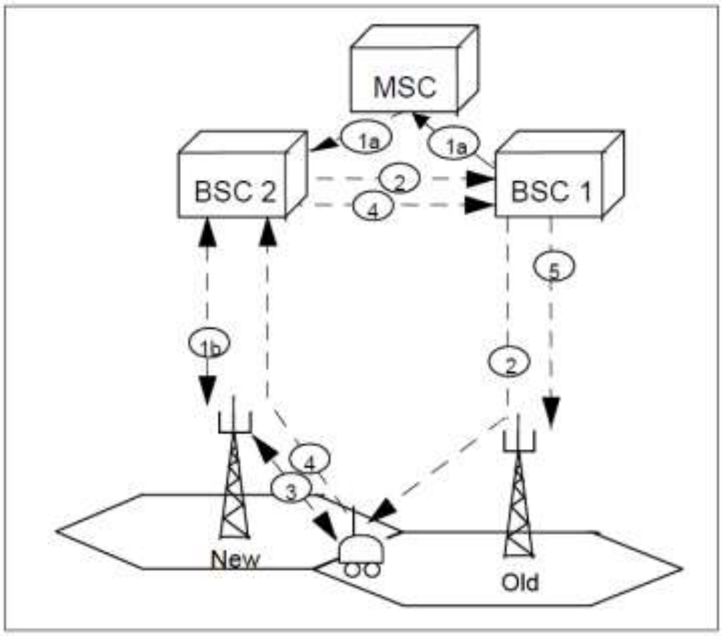
3. Handover Bursts : The MS will tune in to the new channel and send handover bursts on the new channel. The bursts are as short as the access bursts, since the MS does not know the new Timing Advance (TA) value yet. On the detection of the handover bursts, the new BTS will send the new TA to the MS.

4. Handover Complete : Now the MS is ready to continue the traffic and will send a handover complete message addressed to the BSC.

5. Release of Old Channel : When the BSC receives the handover complete message from the MS, the BSC will know that the handover was successful. The BSC orders the old BTS to release the TCH and the BTS will acknowledge.

B. Inter- BSC Handover

In this case BSC1, (old BSC) does not control the letter cell which is the target for the handover. This means that the MSC (Figure (4)) will be part of the link procedure between BSC1 and BSC2 (new BSC)



1. Activation of New Channel : BSC2 will allocate a TCH in the target cell and then order the BTS to activate it. The chosen HO will be part of the activation message. The BTS will acknowledge that the activation has been made.

2. Handover Command : After the activation the new BSC commands the MS to change to the new channel. The message is sent on FACCH via the old channel and will contain a full description of the new channel and the HO.

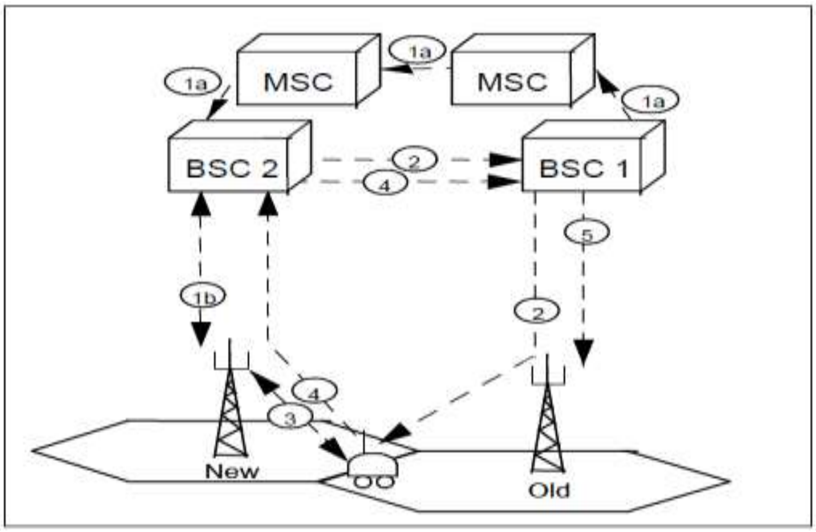
3. Handover Bursts :When the MS has changed to the new channel, it will send handover bursts on the new channel. The information content is the HO. The bursts are as short as the access bursts. This is because the MS does not know the new Timing Advance (TA) value yet. On the detection of the handover bursts, and check of HO, the new BTS will send the new TA.

4. Handover Complete :Now the MS is ready to continue the traffic and will send a handover complete message, which will be addressed to the old BSC as a clear command.

5. Release of Old Channel : When the old BSC receives the clear command from the MSC, the BSC knows that the handover was successful. The BSC orders the BTS to release the TCH and the BTS will acknowledge.

C - Inter- MSC Handover

In this case the old BSC is connected to a different MSC than the BSC that controls the target cell. This means that a new MSC will be part of the procedure figure (5). The old MSC will be called anchor-MSC and the new MSC will be called the target MSC.



1.Handover Request: The old BSC will use the anchor-MSC to send a request to the new BSC for a handover to the target cell. The anchor-MSC knows which MSC to contact and the target-MSC in turn knows which BSC that controls the target cell.

2. Activation of New Channel : The new BSC allocates a TCH in the target cell and order the BTS to activate it. The chosen HO will be part of the activation message. The BTS will acknowledge that the activation has been made.

3. Handover Command : After the activation the new BSC commands the MS to change to the new channel. The message is sent on FACCH via the old channel and will contain a full description of the new channel and the HO. In order to reroute the call, the target- MSC will also send a handover number, similar to the MSRN, to the anchor-MSC.

4. Handover Bursts ; When the MS has changed to the new channel, it will send handover bursts on the new channel. The information content is the HO. The bursts are as short as the access bursts as the MS does not know the new Timing Advance (TA) value yet. On the detection of the handover bursts, and check of HO, the new BTS will send the new TA.

5. Handover Complete : Now the MS is ready to continue the traffic and will send a handover complete message, which will be addressed to the old BSC (Figure 3.4) as a clear command.

6. Release of Old Channel : When the old BSC receives the clear command from the anchor MSC, the BSC knows that the handover was successful. The BSC orders the BTS to release the TCH and the BTS will acknowledge.

- **Reasons of handover failure:**

Handover failure can occur for a number of reasons. Some of them are listed below:

1 The network takes too long to set the handover after the handover has been initiated.

2 There is no available channel on the target base stations.

3 The target link fails in some way during the execution of handoff.

4 Handoff is denied by the network, either for lack of resources or because the portable has exceeded some limit on the number of handoffs which may be attempted in some period of time.

5 In some other systems, handoffs can fail due to resource blocking.

Communication Systems

Mobile Communication Systems

Lecture (10-2):- GSM mobility & signals flow.

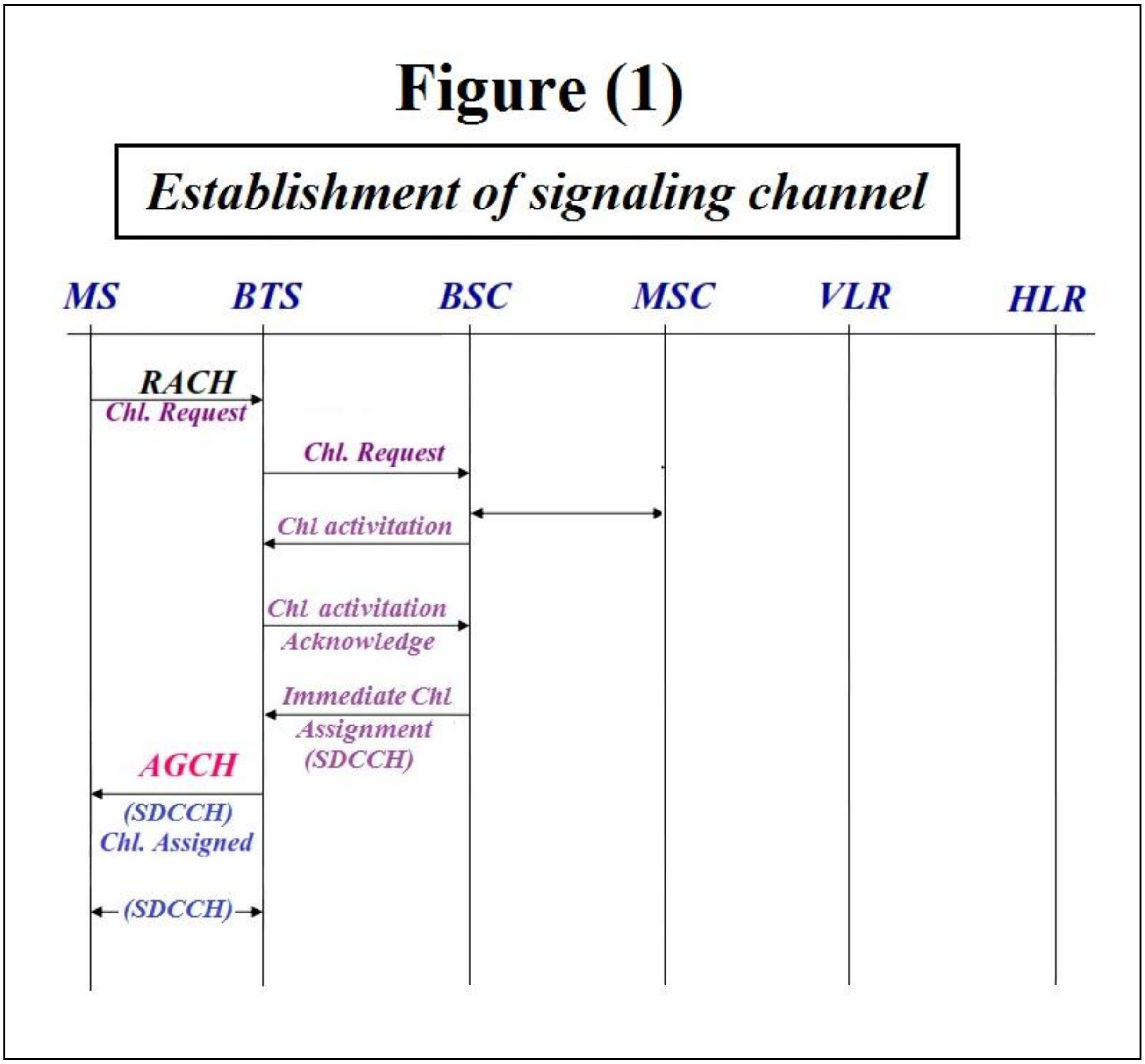
Mobility is a process that allows transferring the mobile service as the mobile station moves from one cell to another. Mobility management in GSM networks has the purpose of monitoring subscribers and their location (in terms of the base station they are registered to), to enable them to receive voice calls, SMSs or data services when an active mobile moves within the coverage area of a network. That will lead to situations where the MS leaves the coverage area of a single cell, in this case there is a need for a feature to transfer an ongoing call from a physical channel to another without dropping the call, and this feature is the handover.

In this lecture the following signal flow graph will be illustrated:-

1 – Establishment of signaling channel.
2 – Authentication Process
3 – Establishment of traffic channel.
4 – Registration Process.
5 - Location update
6 – Mobile starting call.
7– Mobile receiving call.
8 – Handover process

1 – Establishment of signaling channel.

Any access to the GSM system must be started with the establishment of signaling channel process, so you can see this process in all signal flow grapes that will illustrate in next processes, the signals flow graph for establishment of signaling channel are shown in figure (1) below.

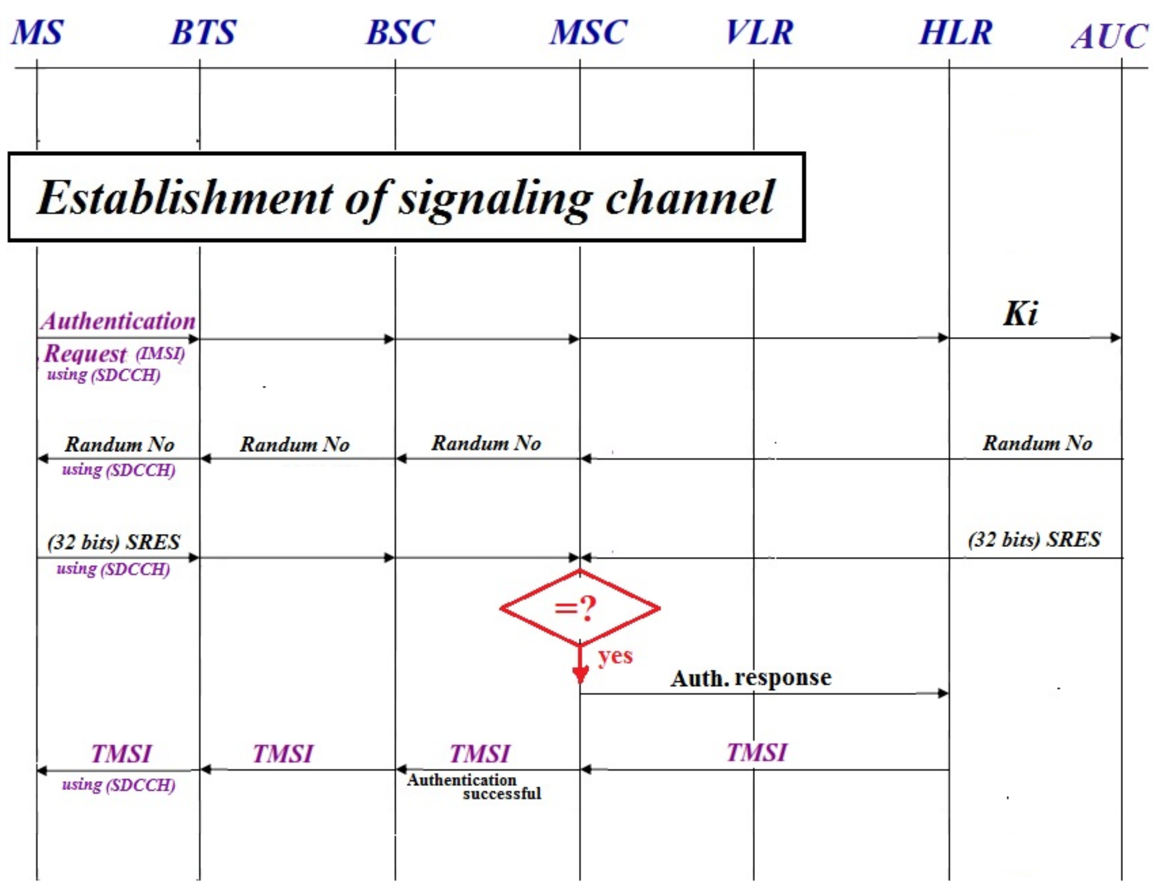


2 – Authentication Process.

The authentication process is first defense line to protect the GSM system from the spies and prevent the un-subscribers of the network from access to it. Also you will see this process in all signal flow graph for all the processes that will illustrate in next paragraphs, the signal flow graph for authentication process are shown in figure (2) below.

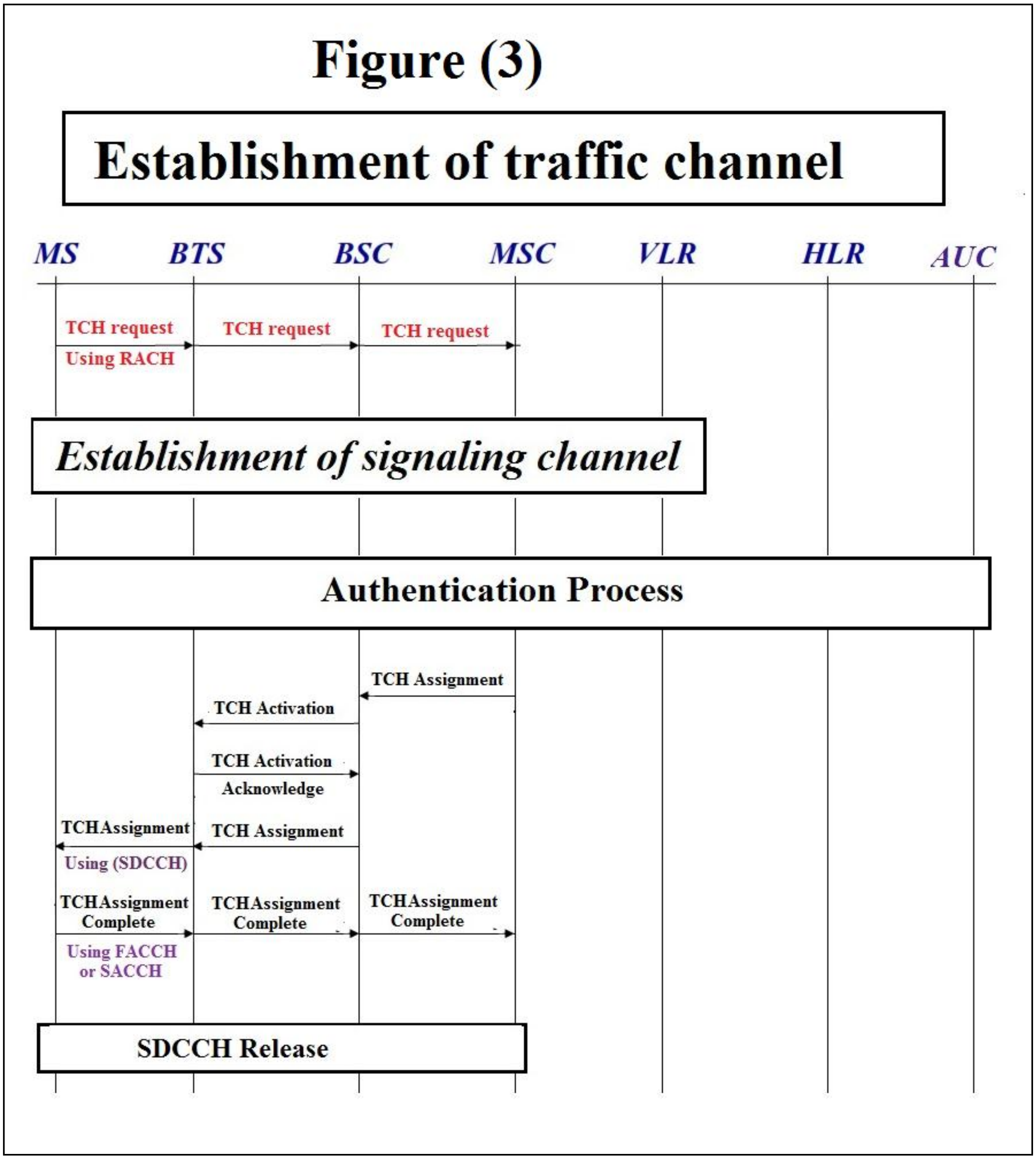
Figure (2)

Authentication Signal Flow Graph



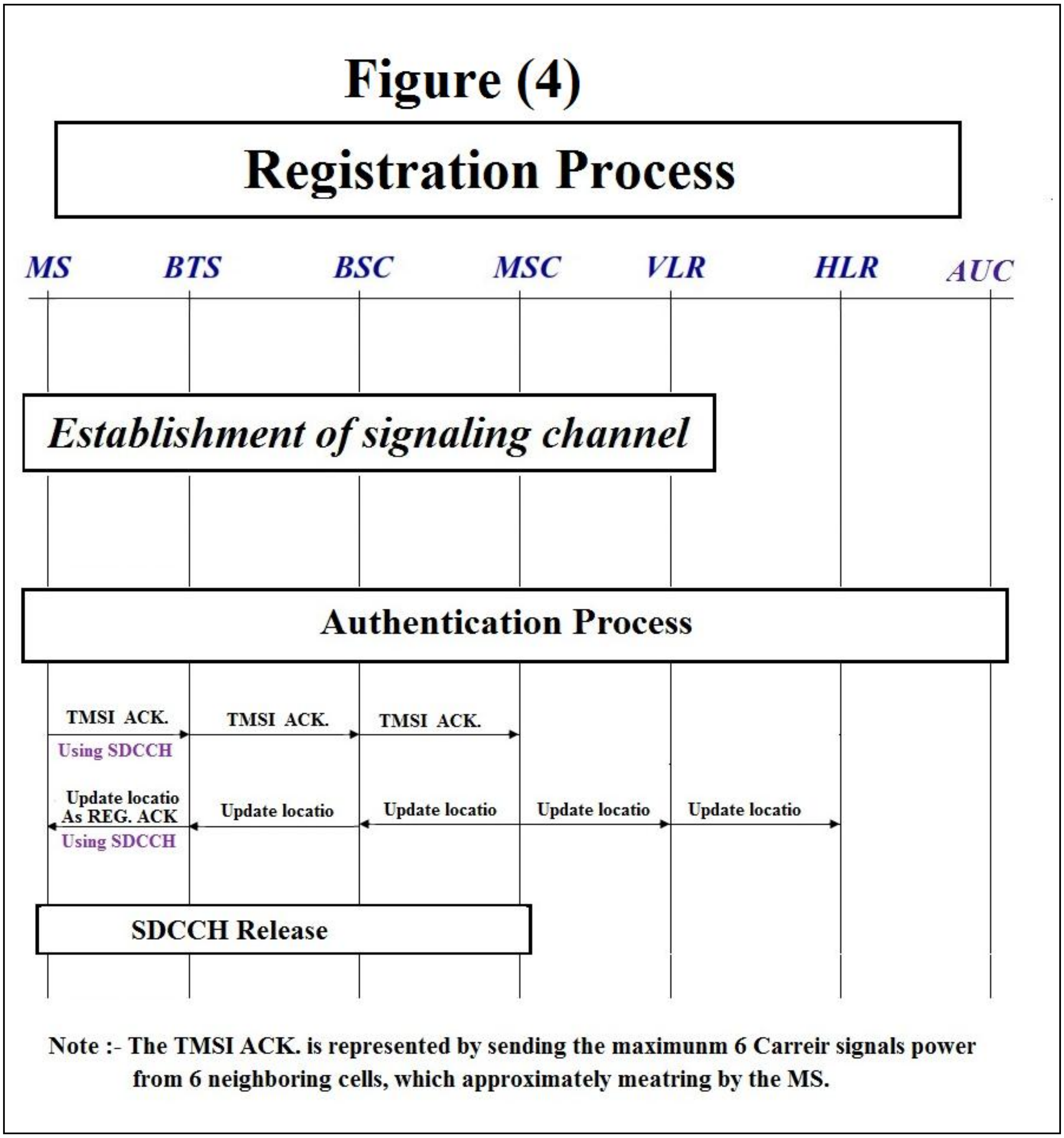
3 – Establishment of Traffic channel:-

The steps that needed to achieve the establishment of traffic channel are shown in figure (3)below.



4 – Registration Process.

The registrations process taken place at the first power on for the mobile equipment after inserting the subscriber identity model (SIM card), the signal flow graph for this process are illustrated in figure (4)below.



5 - Location update

The location update process are periodically happened, at all time from the day, the broadcasting channels for the all the cell of the GSM system are transmit the information of the cell and network, the mobile station are scan this broadcasting channels and make an approximated power measurement for these signals, then send a 6 signals that have maximum power and them information to the base station through the common control channels (RACH) if the mobile station are non in link but its active (power on), else if its in call link, these 6 signals and them information are send by using either (SACCH or FACCH) dedicated control channel. These signals powers and them information are arrived to the MSC, the MSC make analysis by using specific algorithm, and estimate the location of the mobile station within the network and within the cell.

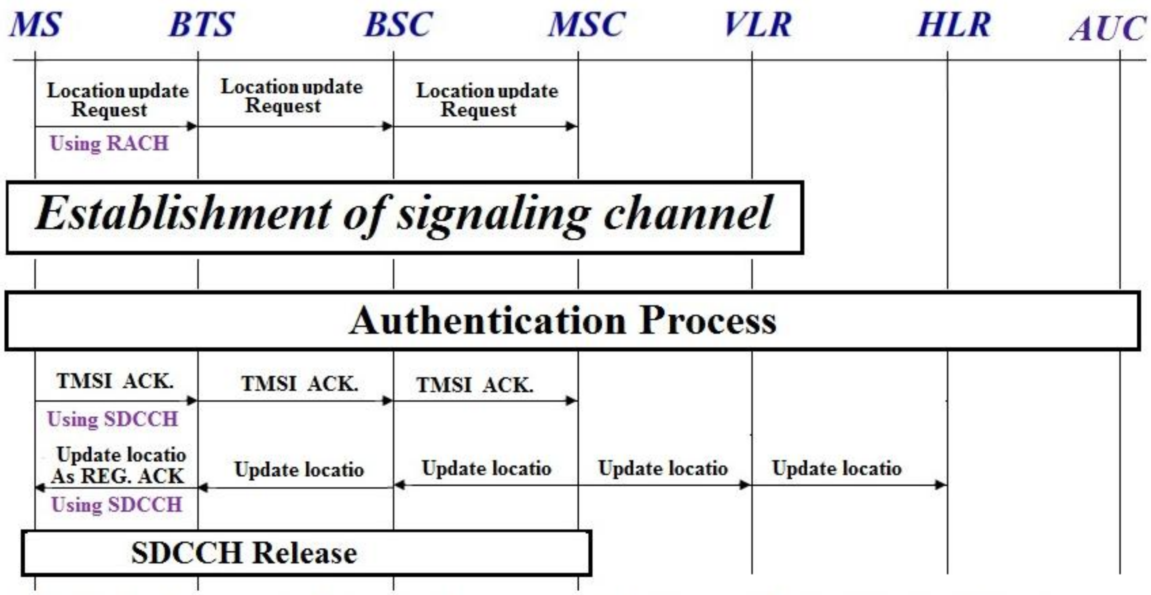
The location update process can be classified into two types

A – Intra MSC location update. The mobile station are within the network

B – Inter MSC location up date . The mobile station go out of the network.

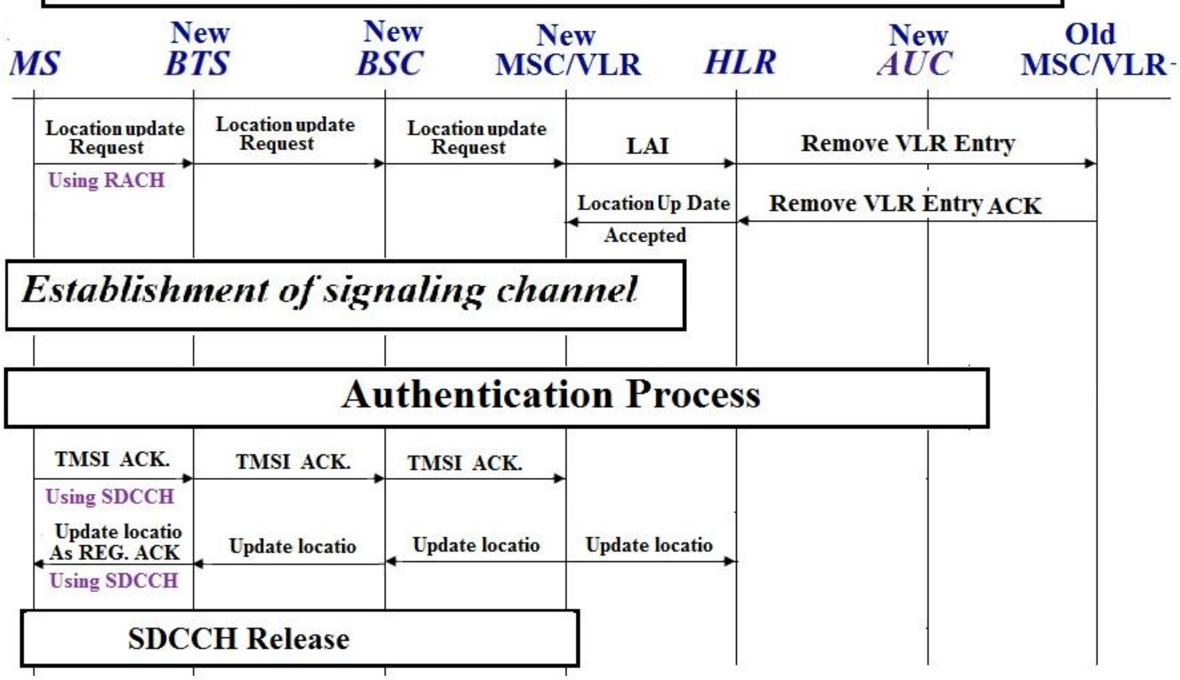
The signal flow graph for the two types of location up date process are shown in next two figures (5 and 6).

Figure (5) Intra MSC Location update process



Note :- The Location up date request is represented by sending the maximunm 6 Carreir signals power from 6 neighboring cells, which approximately meatring by the MS.

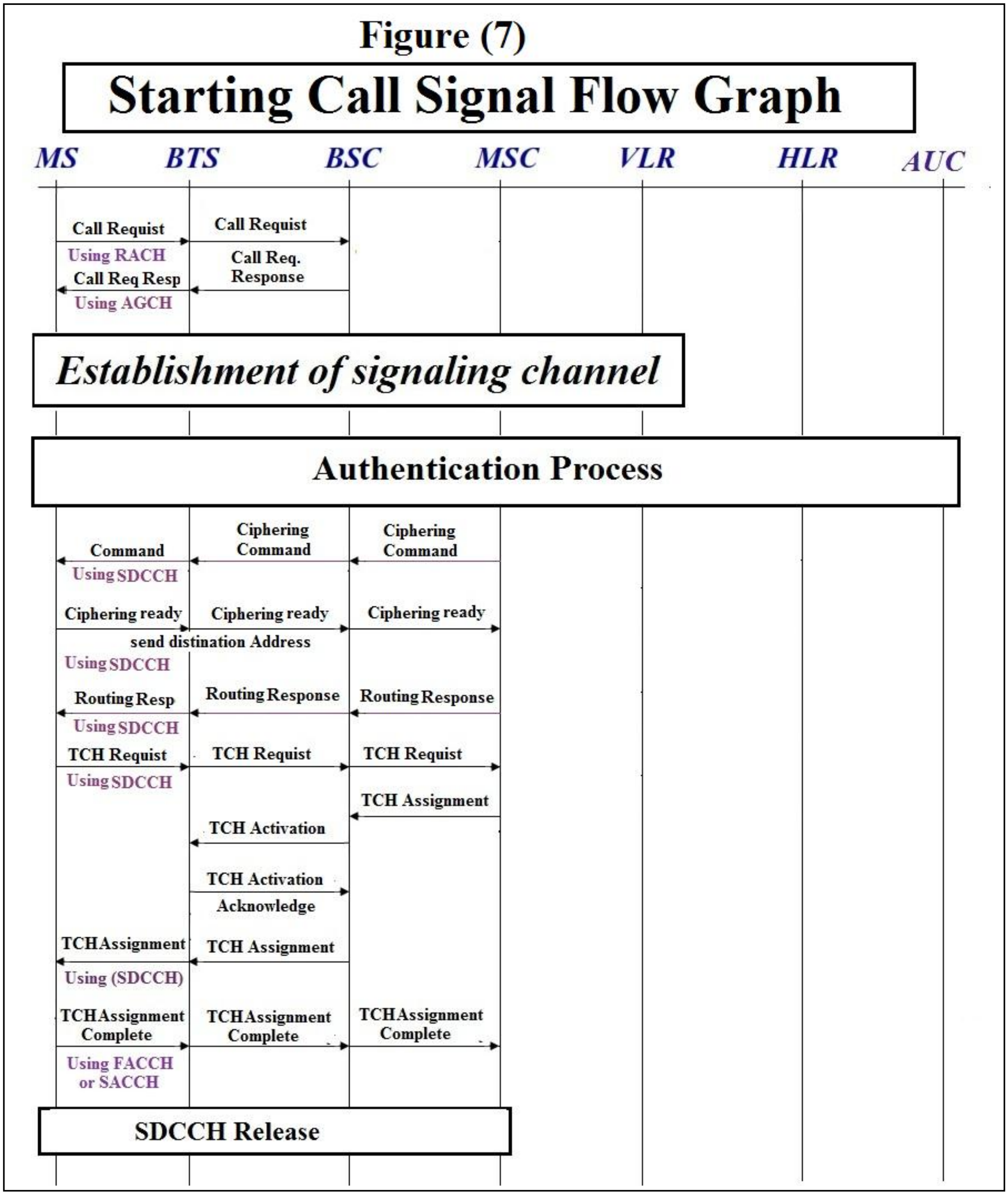
Figure (6) Inter MSC Location update Process



Note :- The Location up date request is represented by sending the maximunm 6 Carreir signals power from 6 neighboring cells, which approximately meatring by the MS.

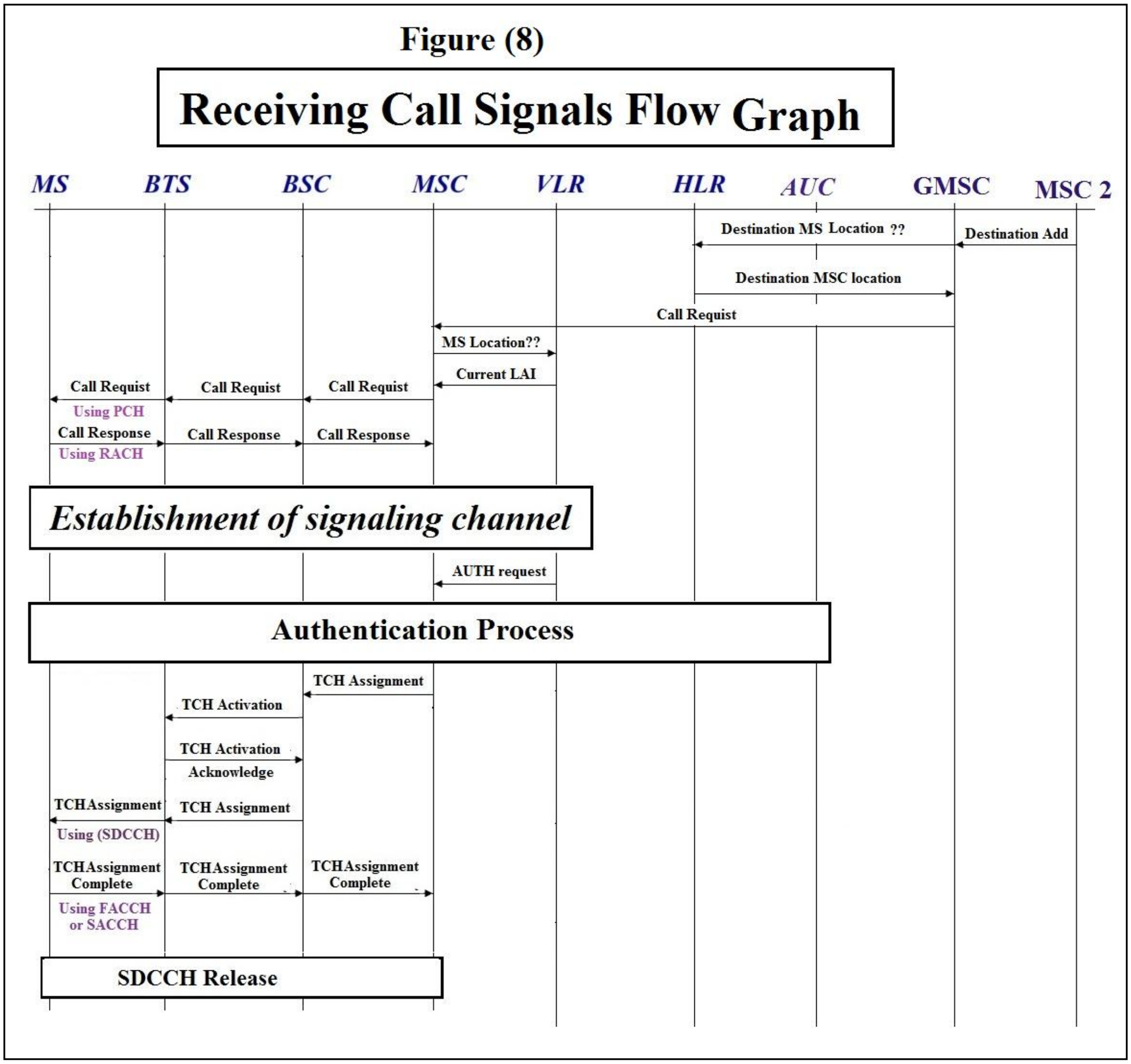
6 – Mobile starting call.

The signal flow graph for starting call process are shown in figure (7) below.



7– Mobile receiving call.

The signal flow graph for starting call process are shown in figure (8) below.



8 – Handover process

In GSM System There are four types of Handover process, Inter Cell Handover, Intra Cell Handover, Inter MSC Handover, and Intra MSC Handover.

The signal flow graph for intra and inter BSC handover are shown in figure (9 and 10) below.

Figure (9) Intra BSC handover Process

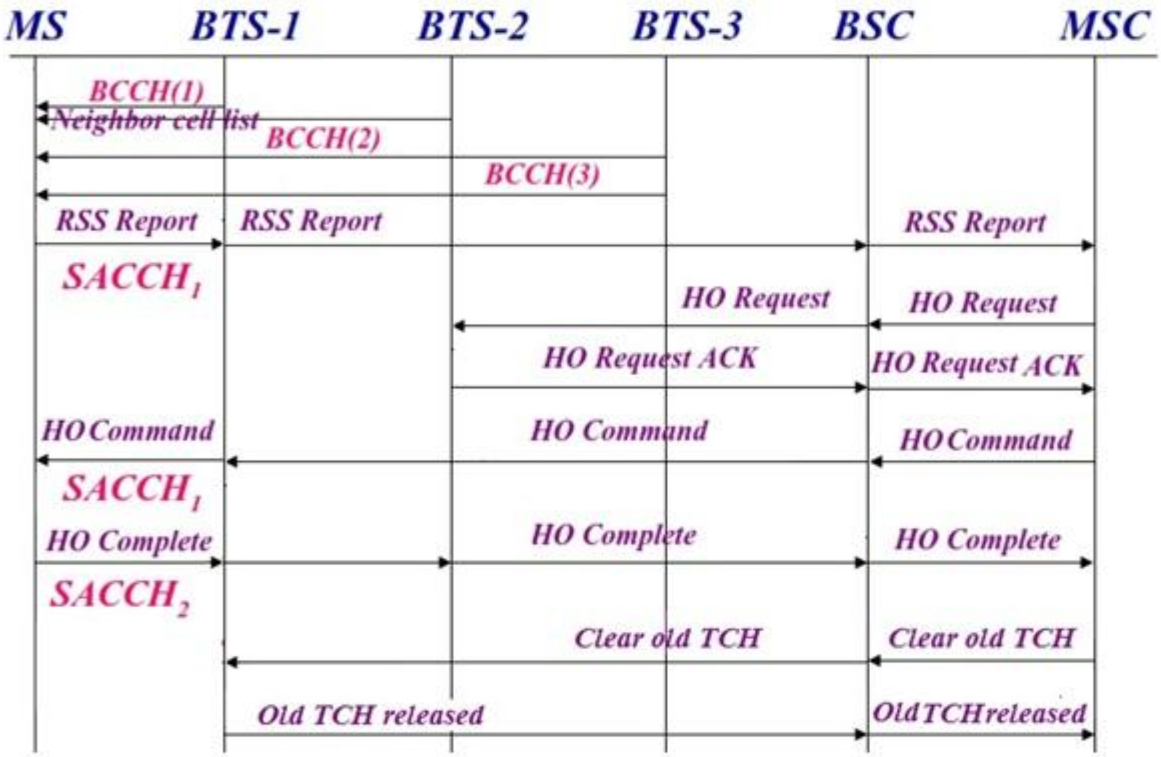
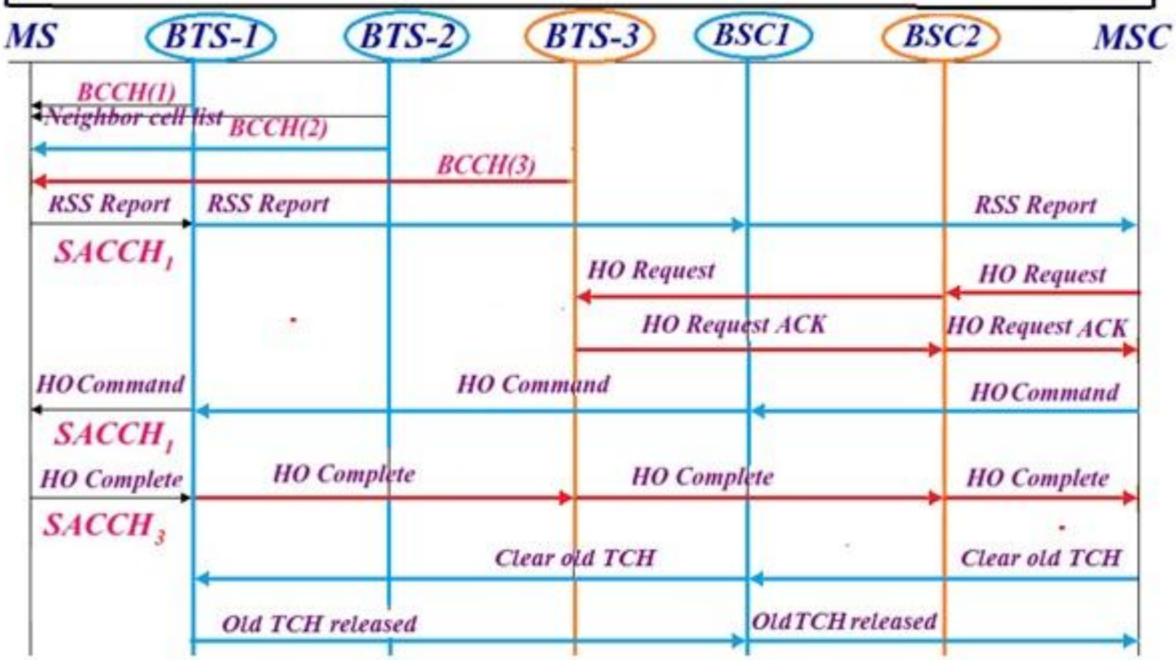


Figure (10) Inter BSC Handover Process



Communication Systems

Mobile Communication Systems

Lecture (11-2):- GSM modified versions.

To increase the data transmission rates, in GSM, new bearer services with rates comparable to or higher as ISDN are developed:

1_ HSCSD (High Speed Circuit Switched Data)

2_ GPRS (General Packet Radio Services)

3_ EDGE (Enhanced Data Rates for the GSM Evolution)

1- High Speed Circuit Switched Data HSCSD

HSCSD is a circuit switched data service (only point-to-point) for applications with higher band width demands and continuous data stream, e.g. motion pictures or video telephony. The higher band width is achieved by combining 1-8 physical channels for one subscriber. Additionally, the data transmission codec was changed such that a maximum of 14.4 kbit/s instead of 9.6 kbit/s can be transmitted per physical channel. In this way, HSCSD theoretically enables transmission rates up to 115.2 kbit/s. In order to implement HSCSD merely the GSM PLMN software must be modified.

High Speed Circuit Switched Data provides Symmetric and pseudo-asymmetric services:

In symmetric service the timeslot allocation for downlink and uplink is symmetric and the same data rates are used in down- and uplink direction.

In pseudo-asymmetric service the timeslot allocation for downlink and uplink is symmetric but the data rate used in uplink direction is lower than in downlink direction.

The following conditions characterize an HSCSD connection:

- _ All n radio timeslots are located on the same TRX.
- _ The same frequency hopping sequence and training sequence is used.
- _ Only TCH/F are used.
- _ The quality measurements reported on the main channel are based on the worst quality measured on the main and the unidirectional downlink timeslots used.
- _ In both symmetric and asymmetric HSCSD configuration the neighbor cell measurements are reported on each uplink channel used.

2- General Packet Radio Service GPRS

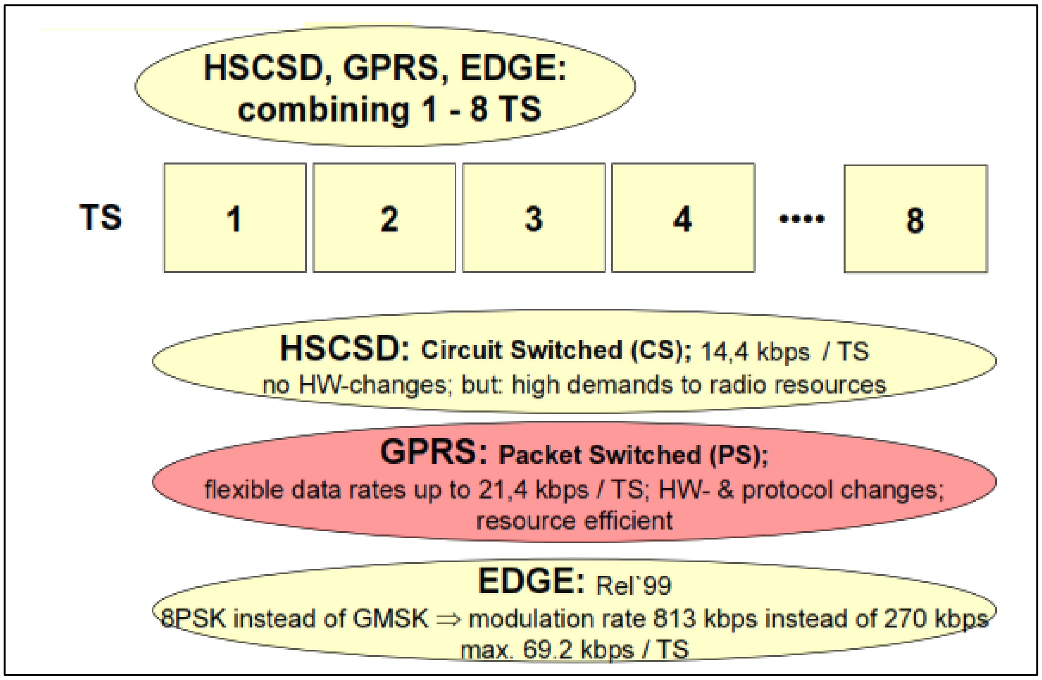
With GPRS it is possible to combine 1-8 physical channel for one user, just as with HSCSD. Various new coding schemes with transmission rates of up to 21.4 kbit/s per physical channel enable theoretical transmission rates up to 171.2 kbit/s. Opposite to HSCSD,

GPRS is a packet-switched bearer service, meaning that the same physical channel can be used for different subscribers. GPRS is resource efficient for applications with a short-term need for high data rates (e.g. surfing the Internet, Email, ...).

GPRS also enables point-to-multipoint transmission and volume dependent charging. Extensions of the GSM network and protocol architecture are necessary for GPRS implementation.

3- Enhanced Data rates for the GSM Evolution EDGE

EDGE is able to realize up to 69.2 kbit/s per physical channel though the change of the GSM modulation procedure (8PSK instead of GMSK). Theoretically, transmission rates of up to 553.6 kbit/s would be possible by combining up to 8 channels.



General Packet Radio Service GPRS

GPRS is a 2.5G technology implemented by GSM network operators, to provide higher data rates for subscribers, to address the increased need for high-speed connections to the the internet and corporate networks, and also as an intermediate step on the path towards 3G systems.

GPRS is an extension of GSM. Instead of requiring a phone number to be dialed and a permanent circuit to be created until the user disconnects, GPRS is packet- based. When the mobile station registers on the network, it is assigned an IP address which enables data to be routed to it by other nodes on the network. The mobile station is then able to send and receive data almost immediately.

GPRS Architecture

For introducing GPRS, the logical GSM architecture is extended by the following functional units:

1 _ The Serving GPRS Support Node **SGSN** is on the same hierarchic level as MSC and has functions comparable to those of a Visited MSC (VMSC).

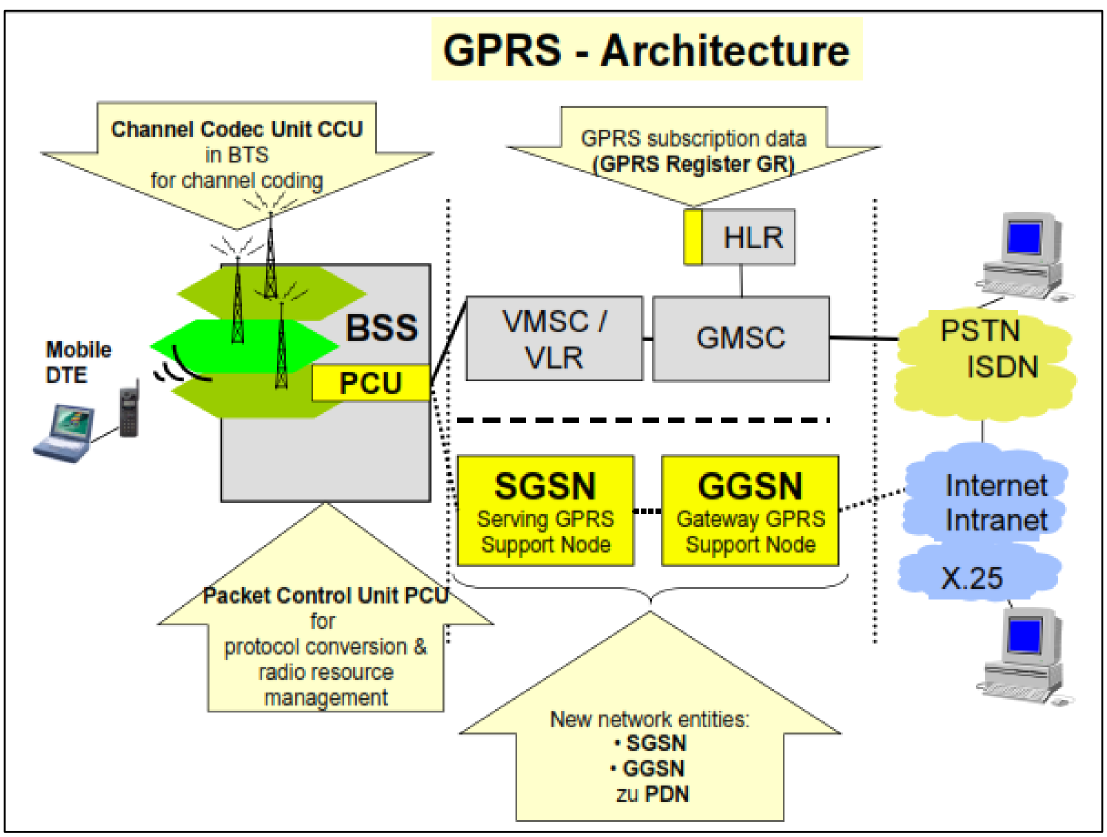
2 _ The Gateway GPRS Support Node **GGSN** has functions comparable with those of a Gateway MSC (GMSC) and offers interworking functions for establishing contact between the GSM/GPRS-PLMN and external packet data networks PDN

3_ A GPRS Support Node **GSN** includes the central functions required to support the GPRS. One PLMN can contain one or more GSNs.

4_ In the BSS a Packet Control Unit **PCU** ensures the reception/adaptation of packet data from SGSN into BSS and vice versa.

5 - Channel Codec Unit (**CCU**): The CCU is realized in the BTS to perform the Channel Coding (including the coding scheme algorithms), power control and timing advance procedures

6 - GPRS subscriber data are added to the HLR. On the following pages of this script this extension will be termed GPRS Register **GR**.



Serving GPRS Support Node (SGSN)

SGSN realizes a large number of functions for performing GPRS services. SGSN is on the same hierarchic level as an MSC and handles many functions comparable to a Visited MSC (VMSC). SGSN

1_ is the node serving GPRS mobile stations in a region assigned to it;

2_ traces the location of the respective GPRS MSs

3_ is responsible for the paging of MS;

4_ performs security functions and access control (authentication/cipher);

5_ has routing/traffic-management functions;

6_ collects data connected with fees/charges;

7_ realizes the interfaces to GGSN (Gn), PCU (Gb), other PLMNs (Gp). Further interfaces may be added (Gr, Gs, Gf).

Gateway GPRS Support Node (GGSN)

GGSN realizes functions comparable to those of a gateway MSC.

1_ is the node allowing contact/interworking between a GSM PLMN and a packet data network PDN (realization Gi-interface);

2_ contains the routing information for GPRS subscribers available in the PLMN.

3_ has a screening function;

4_ can inquire about location information from the HLR via the optional Gc interface.

Packet Control Unit PCU The PCU serves

_ For the management of GPRS radio channels, e.g. power control, congestion control, broadcast control information

_ For the temporal organization of the packet data transfer for uplink and downlink

_ It has channel access control functions, e.g. access request and grants

_ It serves for converting protocols from the Gb interface to the radio interface Um. Three options for positioning the PCU are provided:

A: In the BTS B: in the BSC C: In spatial connection with the SGSN

Channel Codec Unit CCU

The CCU contains the following functions:

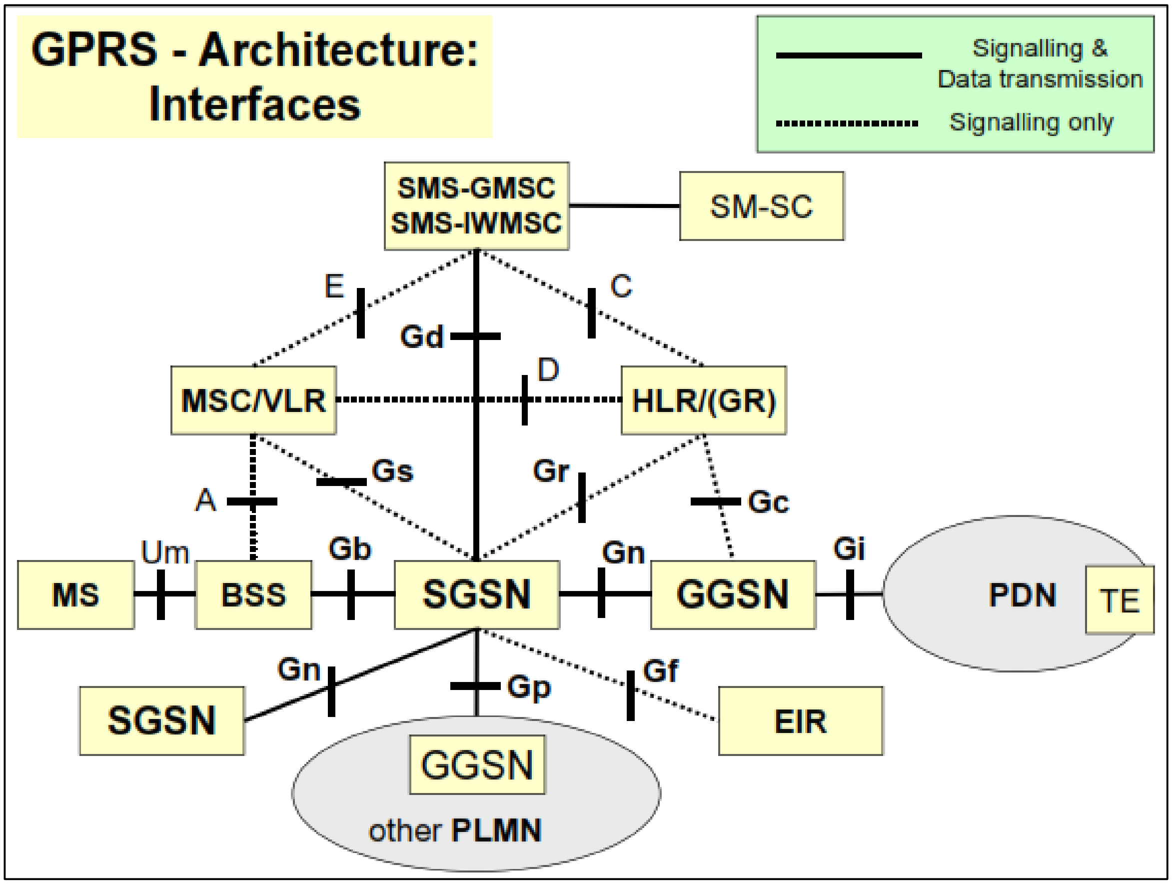
_ Channel coding, including forward error correction FEC and interleaving

_ Radio channel measurements, including received quality and signal level, timing advance measurements

GPRS Interfaces

Integration of functions GGSN and SGSN (which are necessary for GPRS) into a GSM-PLMN makes it necessary to provide names for a series of new interfaces in addition to interfaces A-G already defined in the GSM-PLMN:

_ Gb - between an SGSN and a BSS;
_ Gc - between a GGSN and an HLR
_ Gd - between an SMS-GMSC / SMS-IWMSC and an SGSN
_ Gf - between an SGSN and an EIR
_ Gi - between GPRS and an external packet data network PDN
_ Gn - between two GPRS support nodes GSN within the same PLMN
_ Gp - between two GSN located in different PLMNs.
_ Gr - between an SGSN and an HLR
_ Gs - between an SGSN and an MSC/VLR;



Logical GPRS Radio Channels

In addition to the nine existing logical radio channels used for signaling (BCCH, SCH, FCCH, PCH, RACH, AGCH as well as SDCCH, SACCH and FACCH) and the Traffic Channel (TCH) for circuit switched user information, a new set of logical channels was defined for GPRS.

1 - Packet traffic is realized by means of the Packet Traffic Channel (PTCH) which includes the following :

_ Packet Data Traffic Channel PDTCH.

_ Packet Associated Control Channel PACCH

_ Packet Timing Advance Control Channel PTCCH

The PDTCH is temporarily assigned to the mobile stations MS. Via the PDTCH, user data (point-to-point or point-to-multipoint) or GPRS mobility management and sessionmanagement GMM/SM information is transmitted.

The PACCH was defined for the transmission of signaling (low level signaling) to a dedicated GPRS-MS. It carries information relating to data confirmation, resource allocation and exchange of power control information.

The Packet Common Control Channel PCCCH has been newly defined. It consists of a set of logical channels which are used for common control signaling to start the connection set-up:

_ Packet Random Access Channel PRACH

_ Packet Paging Channel PPCH

_ Packet Access Grant Channel PAGCH

_ Packet Notification Channel PNCH

PRACH and PAGCH fulfil GPRS-MS functions which are analog to the "classical" logical channels RACH and AGCH for non-GPRS-users. The PNCH is used for the initiation of point-to-multipoint multicast (PtM multicast).

For the transmission of system information to the GPRS mobile stations, the

_ Packet Broadcast Control Channel PBCCH was defined analog to the "classical" BCCH.

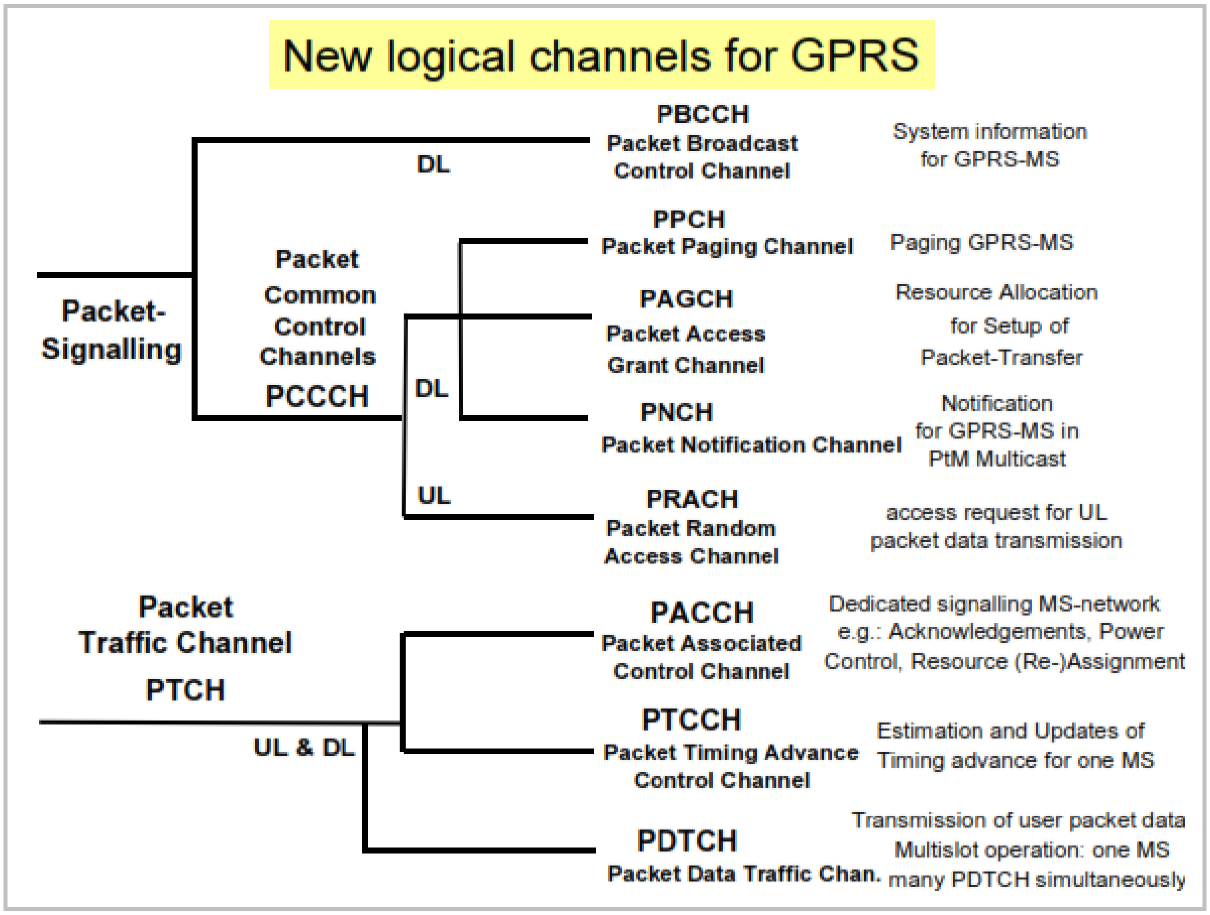
The allowed channel combinations on one time slot are:

_ PBCCH + PCCCH + PDTCH + PACCH + PTCCH

_ PCCCH + PDTCH + PACCH + PTCCH

_ PDTCH + PACCH + PTCCH

Where PCCCH=PPCH + PRACH + PAGCH + PNCH.



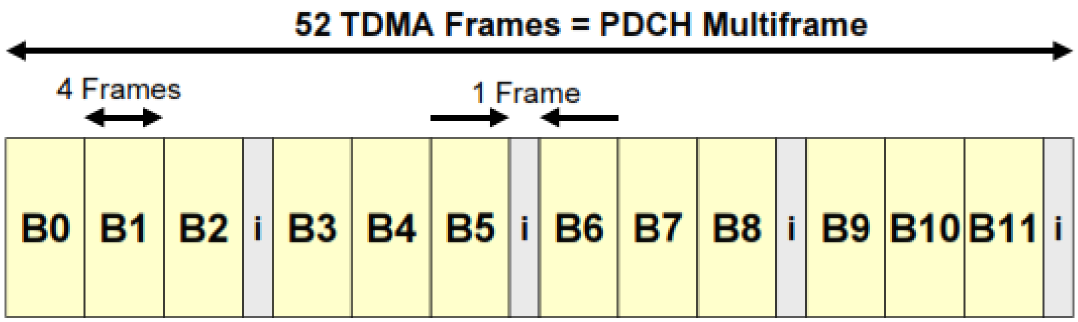
GPRS Framing

The GPRS packet data traffic is arranged in 52-type multiframes 52 TDMA frames in each case are combined to form one GPRS traffic channel multiframe which is subdivided into 12 blocks with 4 TDMA frames each. One block (B0-B11) contains one radio block in each case (4 normal bursts, which are related to each other by means of convolutional coding). Every thirteenth TDMA frame is idle.

The idle frames are used by the MS to be able to determine the various base station identity codes BSIC, to carry out timing advance updates procedures or interference measurements for the realization of power control.

New multiframe for GPRS

- **PDCH** follows **52 multiframe** structure
- 52 Multiframe: **12 Blocks** à **4 TDMA-frames**
- **PCCCHs:** „classical" 51er Multiframes
  or 52er Multiframes



B0 - B11 = Radio Blocks (Data / Signalling)
i = Idle frame

Idle frame:
- **Identification of BSICs**
- **Timing Advance Update Procedure**
- **Interference measurements for Power Control**

- **BCCH** indicates **PDCH** with **PBCCH (in B0)**
- **DL**: this PDCH bears PDCCH & PBCCH
  PBCCH in B0 (+ max. 3 further blocks; indicated in B0)
  PBCCH indicates PCCCH blocks & further PDCHs with PCCCH
- **UL**: PDCH with PCCCH: all blocks to be used for PRACH, PDTCH, PACCH
  PDCH without PCCCH: PDTCH & PACCH only

COMMUNICATION SYSTEMS

RADAR SYSTEMS

Lecture (1-3) : *Historical Review of RADAR*
Fundamental of radar theory
Basic Radar Block Diagram
Basic Elements Of Radar

Radar is an electromagnetic device primarily used for detection and location of objects (targets). The word radar is an acronym derived from the phrase RAdio Detection And Ranging.

In general, radar operates by transmitting an electromagnetic energy and detects the returned echo to extract target information such as range, velocity, and position.

- ***Historical Review of RADAR***

1865 The English physicist James Clerk **Maxwell** developed his electromagnetic light theory (Description of the electro-magnetic waves and Propagation).

1886 The German physicist Heinrich Rudolf **Hertz** discovers the electromagnetic waves and proves the theory of Maxwell with that.

1904 The German high frequency engineer Christian Hülsmeyer invents the "Telemobiloskop" to the traffic supervision on the water. He measures the running time of electro-magnetic waves to a metal object (ship) and back. A calculation of the distance is thus possible. This is the first practical radar test. Hülsmeyer registers his invention to the patent in Germany and in the United Kingdom.

1917 The French engineer Lucien Lévy invents the **super heterodyne receiver**. He uses as first the denomination "Intermediate Frequency", and allowed the possibility of double heterodyning.

1921 The invention of the **Magnetron** as an efficient transmitting tube by the US-American physicist Albert Wallace Hull.

1922 The American electrical engineers Albert H. Taylor and Leo C. Young of the Naval Research Laboratory (USA) **locate a wooden ship** for the first time.

1930 Lawrence A. Hyland (also of the Naval Research Laboratory), **locates an aircraft** for the first time.

1931 A ship is equipped with radar. As antennae are used parabolic dishes with horn radiators.

1936 The development of the **Klystron** by the technicians George F. Metcalf and William C. Hahn, both from General Electric. This will be an important component in radar units as an amplifier or an oscillator tube.

1937 The first practical radar system, known as Chain Home (CH) radar stations operating at 5 megahertz (MHz), was successfully demonstrated.

1940 Different radar equipment's are developed in Second World War as the ideal technique for detecting the enemy, both day and night in the USA, Russia, Germany, France and Japan. As an example in 1940 the **cavity magnetron**, devised by John T. Randall and Henry A.H. Boot at the University of Birmingham was a breakthrough, having the capacity to operate at much higher power and frequencies (3.3 GHz), thus allowing much shorter wavelengths (9.1 cm), suitable for small antennas of airborne radars.

1943 The first model, **British (H2S) airborne radar**, was operative and played a key role in the bomber raids on German cities. H2S was the first airborne, ground scanning radar system. It was developed for the (Royal Air Force's Bomber) command during World War II to identify targets on the ground for night and all-weather bombing.

1950s New and better radar systems emerged. One of these was a highly accurate **mono-pulse tracking radar** designated the AN/FPS-16, which was capable of an angular accuracy of about 0.1 mille radian (roughly 0.006 degree). There also appeared large, high-powered radars designed to **operate at 220 MHz (VHF) and 450 MHz (UHF).**

Another notable development was the **klystron amplifier**, which provided a source of stable high power for very-long-range radars.

Synthetic aperture radar first appeared in the early 1950s, but it took almost 30 more years to reach a high state of development, with the introduction of digital processing and other advances.

The **airborne pulse Doppler radar** also was introduced in the late 1950s in the air-to-air missile.

The decade of the 1950s also saw the publication of important theoretical concepts that helped put radar design on a more quantitative basis. These included the statistical theory of detection of signals in noise; the so-called **matched filter theory**

1960s The first large **electronically steered phased-array radars** were put into operation.

Airborne MTI radar for aircraft detection was developed for the U.S. Navy's Grumman E-2 airborne-early-warning (AEW) aircraft at this time.

Many of the attributes of **HF over-the-horizon radar** were demonstrated during the 1960s, as were the first radars designed for detecting ballistic missiles and satellites.

1970s Digital technology underwent a tremendous advance in signal and data processing, which made modern radars possible. Advances in airborne pulse Doppler radar greatly enhanced its ability to detect targets in the presence of ground clutter.

1980s A series of developments in the production of phased-array radars made possible the air defense (the Patriot and Aegis systems), airborne bomber radar (B-1B aircraft), and ballistic missile detection (PAVE PAWS) systems.

Solid-state technology and integrated microwave circuitry permitted

1990s Advances in computer technology allowed increased information about the nature of targets and the environment to be obtained from radar echoes.

The introduction of Doppler weather radar systems utilized computer technology to measure the wind speed as well as the rate of precipitation.

Terminal Doppler weather radars (TDWR) were installed at or near major airports to facilitate safe takeoff and landing.

High frequency (HF) over-the-horizon radar systems were operated by several countries, primarily for the detection of aircraft at very long ranges.

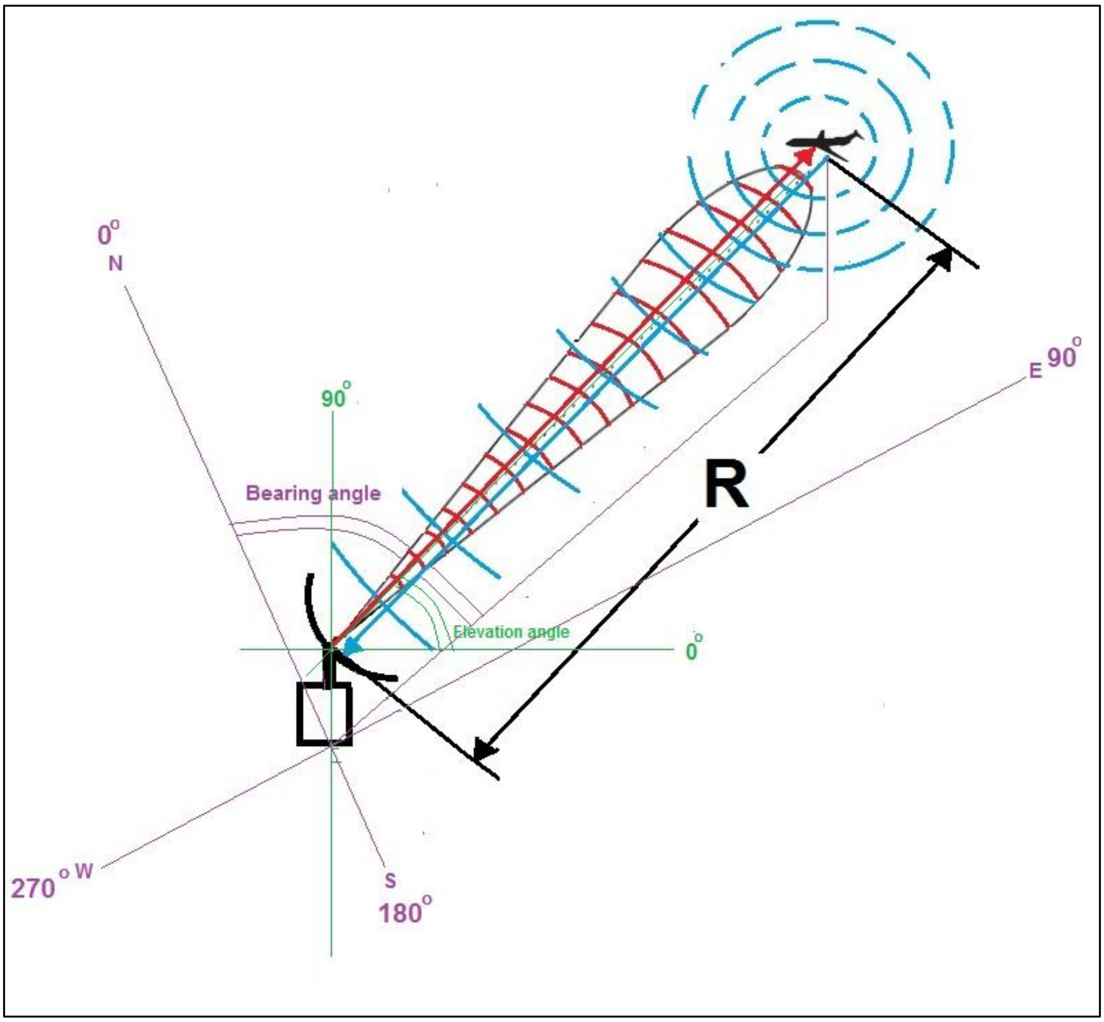
The use of tactical ballistic missiles during the Gulf War (1990–1991) brought back the need for radars for defense against such missiles. Russia and Israeli continually enhanced their powerful radar-based air-defense systems to stay engaged in tactical ballistic missiles.

Radar systems such as altimeters, scatterometers, and imaging radar systems are now widely recognized as highly successful tools for earth observation from aircraft and satellites.

More advances in digital technology in the area of signal and data processing led to further improvement and development of radars to meet the demands of mankind.

- ***Fundamental of radar theory***

The basic radar concept is to transmit radiated the radio frequency (RF) energy by the transmitting antenna, if this RF radiated beam hit any conducting body in the space, the conducting body effected with the electromagnetic wave and regenerate an electromagnetic energy radiated in all direction, small amount of this energy reflected toward the receiving antenna, they collected by the receiving antenna, and detected in the radar receiver, this an indication to found a target in the direction of radiation beam, and so if no received signal no target in this direction.



The direction of the target relative to the radar may be determined by using an antenna with a directional pattern, and observing the direction from which the peak of this pattern is pointing when the received signal is maximized.

Since the electromagnetic energy travels with the speed of light (designated c), the range from the radar to the target, R, may be determined by measuring the time interval, t, between the transmitted signal and the received signal:

$$velocity = \frac{distance}{time} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots(1)$$

The electromagnetic propagation velocity in the atmosphere is nearly the same as that in a vacuum, and the approximation c =3× 10^8 m/s is sufficiently accurate for most analyses.

The distance that the electromagnetic wave travel is distance from transmitting antenna to the object (R), plus the distance between the object to the receiving antenna (R), (usually the transmitting antenna itself is the receiving antenna)

Distance = 2R   Velocity = C = 3*10^8 time = t

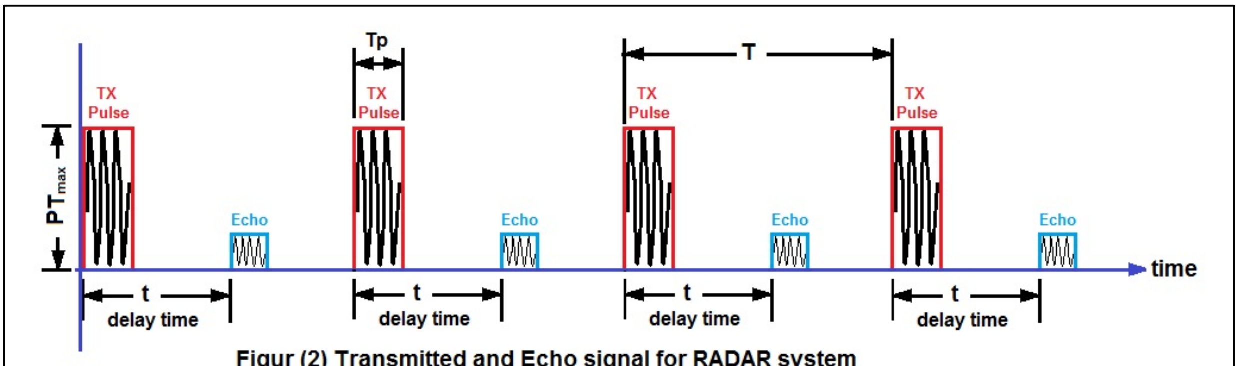
$$C = \frac{2R}{t} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots(2)$$

$$R = \frac{C*t}{2}$$

If the time measured in micro-second

$$R = 150 * t_{\mu\text{-sec}}$$

The question that must be asked now is how can be measure the time delay (t), in order to measure the time delay the radar must be send a modulated pulse signal and receive the echo signal that returned from the target, the time delay is the time from raised edge of the transmitted pulse signal to the raised edge of the received echo signal as illustrated in figure (2)



Figur (2) Transmitted and Echo signal for RADAR system

In fact the radar transmitter dos not send only one pulse but it send train of pulses (periodically) separated by duration time (T) and pulse width (TP).

In Radar system the pulse period time related to other term which is the **p**ulse **R**epetition **F**requency (PRF), where the PRF equal (1/T), the radar that send a periodic pulses called (Pulse Radar). In this type of Radar, the two terms (TP and

PRF) effects on the main specification of radar like the maximum detected range for targets, the average transmitted power and the range resolution

Where Range resolution is the radar metric that describes its ability to identify separate multiple targets at the same angular position, but at different ranges. Range resolution depends largely on the radar's pulse width (TP). The narrower pulse width results in a better resolution. It can be easily demonstrated that the targets to be resolved must be separated by at least the range equivalent of the width of the processed echo pulse. Accordingly,

Maximum Range $\mathbf{R_{max} = T * C/2 = T_{in\ \mu Sec} * 150}$

Range resolution $\mathbf{dR_{min} = TP * C /2 = TP_{in\ \mu Sec} * 150}$

Average Transmitted Power $\mathbf{PT_{av} = (PT_{max} * TP) / T = PT_{max} * TP * PRF}$

- ***Basic Radar Block Diagram***

An elementary form of a typical radar system is described in the block diagram shown in Figure (3) that omits many details. It consists of a transmitter routing electromagnetic energy generated by an oscillator of some sort, to the antenna via a duplexer, (a microwave device that enables the same antenna to be used for both transmission and reception). The antenna serves as transducer to couple EM energy into free space, where it propagates at the speed of light (approximately 3 × 108 m/s). A portion of the backscattered energy, being reflected by an object, is intercepted by the radar antenna, and then detected by the receiver and display the detected targets on the radar indicator (display).

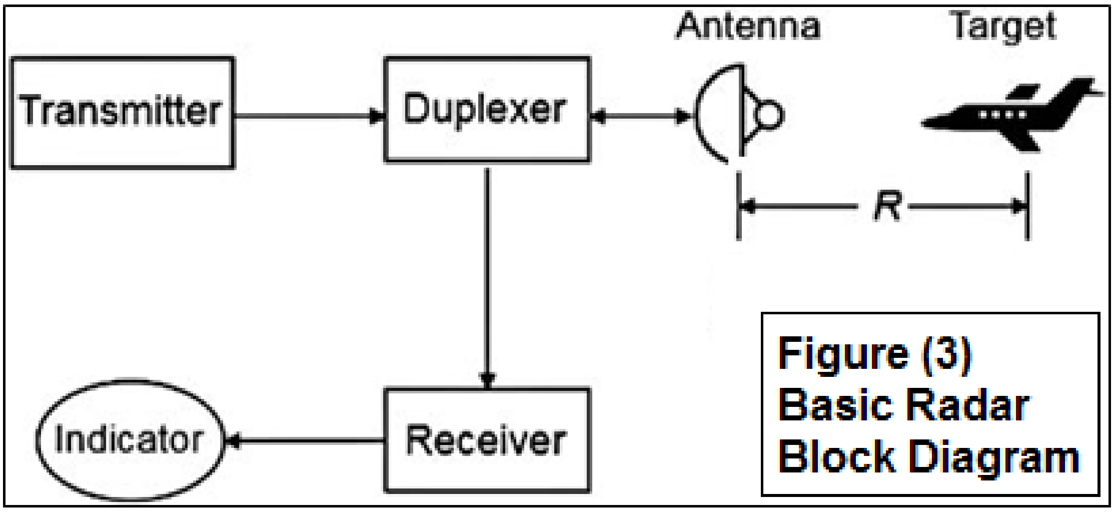


Figure (3) Basic Radar Block Diagram

- ## *Basic Elements Of Radar*

Transmitter: The transmitter is one of the basic elements of a radar system, and generate the *radio frequency* (RF) power signal to illuminate the target. The RF signal generated may be *continuous wave* (CW) or pulsed, and its amplitude and frequency are usually designed to meet the specific requirements of the radar system. There are two methods of generating RF power: in the power oscillator approach, the signal is generated at the required level suitable for applying directly to the antenna; and in the master oscillator approach, the oscillator generates a relatively low-power RF signal and then amplified to the appropriate level.

Duplexer: The duplexer enables the same antenna of mono-static radars to be used for both transmission and reception. The duplexer consists of two gas discharge devices, one known as a *transmit-receive* (TR) and the other as an *anti-transmit-receive* (ATR). In this fast-acting RF electronic switch, the TR protects the receiver from a high-power radar signal during the transmission mode and the ATR directs the echo signal to the receiver during the reception mode. Solid-state ferrite circulators are also used.

Antenna: Antennas used in the radar system are mostly highly directional. A common form of radar antenna is a parabolic dish antenna fed from a feeding antenna at its focus. The beam may be scanned in space by mechanical pointing of the antenna. Phased-array antennas have also been used for radar. In a phased-array antenna the antenna beam is scanned electronically by introducing phases to the phase shifters connected to elements. The functions of the antenna are to concentrate the transmitting signal into a narrow beam in a single preferred direction, intercept the target echo signal from the same direction. Received weak energy is then sent via the transmission line to the receiver.

Receiver: The receiver is basically a super heterodyne type consisting of low-noise radio frequency amplifier, a mixer, an *intermediate frequency* (IF) amplifier, a video amplifier, and a display unit. The front-end RF amplifier is usually a parametric amplifier or a low- noise transistor. The mixer and *local oscillator* (LO) convert the RF signal into an IF signal having a center frequency 30 or 60 MHz. The intermediate frequency amplifier is primarily designed as a matched filter and the pulse modulation is extracted by the second detector. The demodulated signal is then amplified by the video amplifier to a level suitable for displaying in an indicator, usually a *cathode-ray tube* (CRT).

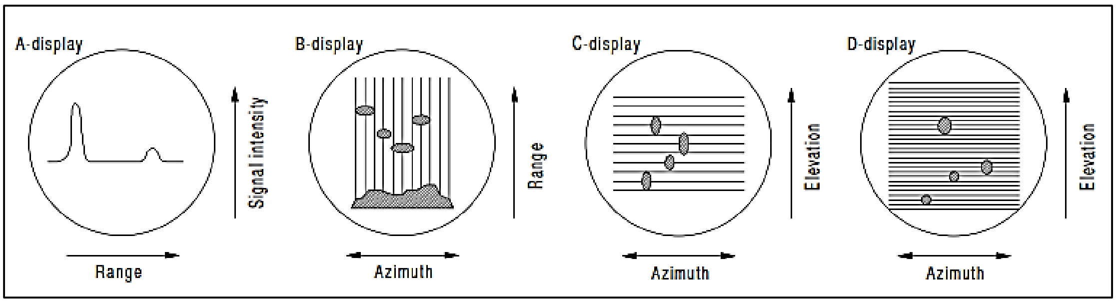
Display: A radar display is an electronic instrument for visual representation of radar data. Radar displays can be classified from the standpoint of their functions, the physical principles of their implementation, type of information displayed, and so forth. From the viewpoint of number of displayed coordinates, they can be one-dimensional (1D), two-dimensional (2D), or three-dimensional (3D). An example of a 1D display is the range display (A-scope). Most widely used are 2D displays, represented by the altitude-range display (range-height indicator, or RHI), azimuth-elevation display (C-scope), azimuth-range display (B-scope), elevation-range display (E-scope), and plan-position indicator (PPI). These letter descriptions date back to World War II, and many of them are obsolete. From the viewpoint of physical implementation, active and passive displays are distinguished.

The former are represented mainly by cathode-ray-tube (CRT) displays and semiconductor displays. Passive displays can be of liquid crystal or ferroelectric types. In most radar applications CRT displays remain the best choice because of their good performance and low cost.

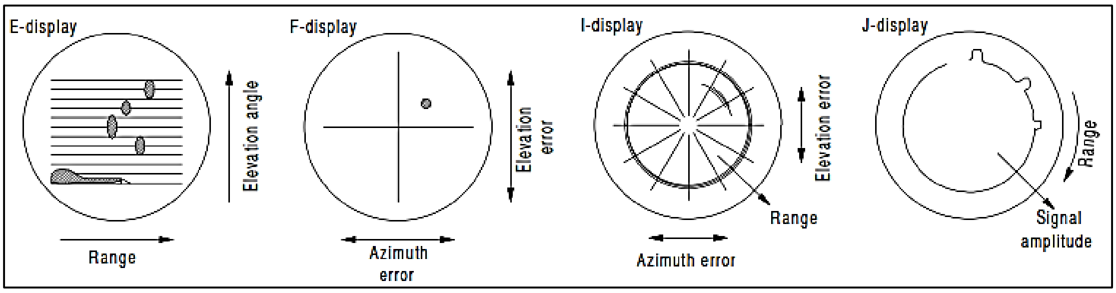
From the viewpoint of displayed information, displays can be classified as presenting radar signal data, alphanumeric, or combined displays. A combined display simultaneously presents various data (dynamic and statistical). Combined radar displays usually display coordinate (signal) data simultaneously with alphanumeric data concerning a large number of objects, and statistical data (map contours, routes, etc.). Combined displays employ special CRTs with several guns operating on a common screen.

Displays in modern radar are typically synthetic-video combined displays, often using the monitors of computer-based work stations. Semiconductor displays are active displays based on the effect of an injection luminescence that takes place when the carriers are recombined on a junction of the semiconductor crystal switched in the forward direction. Semiconductor displays are also known as light-emitting diode displays. They are characterized by a low operating voltage, the ability to overlap with semiconductor logic circuits, small dimensions, a long service life, a high degree of pixel brightness, and a capability for multiplex addressing. Matrix displays with 6,000 to 40,000 elements have an 0.8mm space between the elements. Following some types of radar display units.

- **A-(scope) display** is one "in which targets appear as vertical deflections from a horizontal line representing a time base, target distance [range] is indicated by the horizontal position of the deflection from one end of the time base. The amplitude of the vertical deflection is a function of the signal intensity."
- **B-(scope) display** is "a rectangular display in which each target appears as an intensity-modulated blip, with azimuth indicated by the horizontal coordinate and range by the vertical coordinate"
- **C-(scope) display** is "a rectangular display in which each target appears as an intensity-modulated blip with azimuth indicated by the horizontal coordinate and angle of elevation by the vertical coordinate.
- **D-(scope) display** is "similar to a C-display, but composed of a series of horizontal stripes representing successive elevation angles.

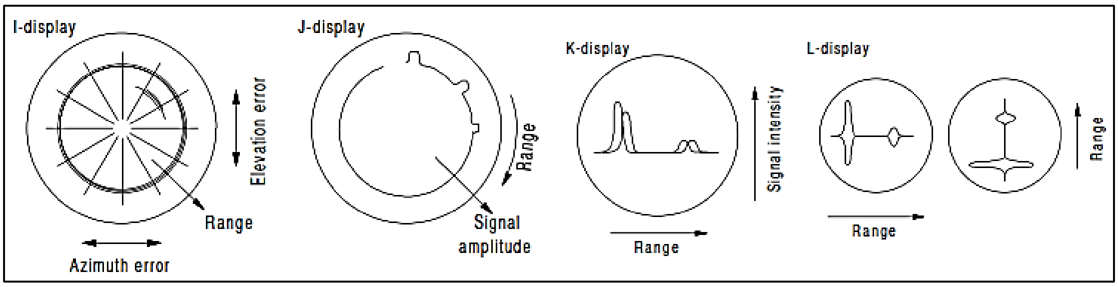


- **E-(scope) display** is "a rectangular display in which targets appear as intensity-modulated blips with range indicated by the horizontal coordinate and elevation angle by the vertical coordinate.
- **F-(scope) display** is "a rectangular display in which a target appears as a centralized blip when the radar antenna is aimed at it. Horizontal and vertical aiming errors are respectively indicated by horizontal and vertical displacement of the blip.
- **G-(scope) display** is a modified F-display in which wings appear to grow on the blip, the width of the wings being inversely proportional to target range
- **H-(scope) display** is "a B-display modified to include an indication of angle of elevation. The target appears as two closely spaced blips approximating a short bright line, the slope of which is in proportion to the tangent of the angle of target elevation.



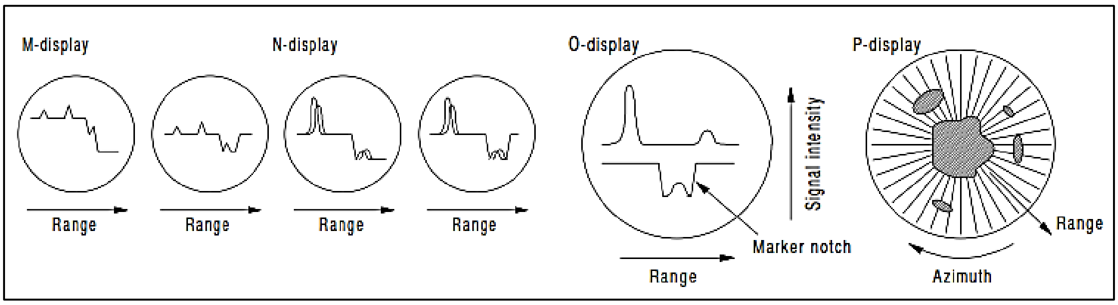
- **I-(scope) display** is used "in a conical-scan radar, in which a target appears as a complete circle when the radar antenna is pointed at it and in which the radius of the circle is proportional to target range.
- **J-(scope) display** is a modified A-display in which the time base is a circle, and targets appear as radial deflections from the time base.
- **K-(scope) display** is "a modified A-display used with a lobe-switching antenna, in which a target appears as a pair of vertical deflections. When the radar antenna is correctly pointed at the target, the deflections (blips) are of equal height, and when not so pointed, the difference in the blip height is an indication of the direction and magnitude of pointing error.

- **L-(scope) display** is "similar to a K-display, but signals from the two lobes are placed back to back. A target appears as a pair of deflections, one on each side of a central time base representing range. Both deflections are of equal amplitude when radar antenna is pointed directly at the target, any inequality representing relative pointing error. The time base (range scale) can be vertical, as in the L-display illustration, or horizontal. The L-display is also known as a bearing deviation indicator.



- **M-(scope) display** is "a type of A-display in which one target range is determined by moving an adjustable pedestal, notch, or step along the baseline until it coincides with the horizontal position of the target-signal deflection; the control that moves the pedestal is calibrated in range. The use of the term M-display is uncommon. More often this display is defined as a variant of an A-display.
- **N-(scope)** display is a K-display having an adjustable pedestal, notch, or step, as in the M-display, for the measurement of range. This display is usually regarded as a variant of an A-display or K-display rather than a separate type.
- **O-(scope) display** is "an A-display modified by the inclusion of an adjustable notch for measuring range.
- **P-(scope) display** commonly known as a **Plan-position indicator (PPI)** is "a display in which target blips are shown on plan position, thus forming a map-like display, with radial distance from the center representing range and with the angle of the radius vector representing azimuth angle. There are various types of PPI displays implementation; the primary ones are: azimuth-stabilized PPI (the reference direction remains fixed with respect to the indicator regardless of the vehicle orientation), delayed PPI (the initiation of time base is delayed), off-center

PPI (the zero position of the time base is located at a point other than the center of display, so equivalent of a larger display can be provided for a selected portion of the service area), open-center PPI (the display of the initiation of the time base precedes that of the transmitted pulse), and north-stabilized PPI (an azimuth-stabilized PPI in which the reference detection is north). This is the type of radar display normally associated with volume search or surveillance radars that allows a human operator to assess the tactical situation within the radius of coverage of the radar. The PPI presents a map-like circular presentation, usually on a CRT (CRT) of azimuth versus range. The display is refreshed at the radar azimuth scan rate. Targets and clutter remain visible from scan-to-scan due to the long-persistence properties of the phosphor coating on the CRT's inside surface. The PPI display may show a full 360° view, or may be restricted to displaying a particular sector during each scan or may reflect operation of the radar in a more limited sector-scan mode.



PPI display for maritime radar.

COMMUNICATION SYSTEMS

RADAR SYSTEMS

Lecture (2-3) Radar Parameters

Radar Measurements.

- ***<u>Radar Parameters</u>***

The performance of radar is determined by choice of a number of parameters these parameters are

<u>1 – Peak power</u>

Obviously the transmission of adequately high power is necessary for ensuring large enough echo levels for satisfactory detection in noise background. The peak power for wide range radar must be higher than the low range radar, the peak power with the other parameters (antenna beam width, modulation type and data rate) effected on the maximum range detection and the target cross-section can be detected. Some radars transmit with peak power reached up to few mega volt amperes (MVA) within the duration time about 1 micro second while other radars transmit with some hundreds of kilo watt within the duration time about ten micro seconds.

<u>2 – Modulation types</u>

The transmitted signal for radar system can be modulated by two main types (pulse modulation and continuous wave modulation), the pulse modulation is widely used form of signal with constant amplitude during the pulse and with various form of pulse duration which can provide improved range resolution. While the pulse repetition frequency (PRF) determined the maximum unambiguous range.

The second type of modulation is the continuous wave modulation (CW), this type enable the radar system to discriminate with considerable precision in the Doppler frequency domain that give an information about the speed of targets but with poor or no range information. The CW radar generally required separated antennas for transmission and reception. One improvement of the CW radar that lead to enable this type for determined the range of the targets is the FM-CW modulation. This type of radar transmit the carrier signal with constant level but with no constant frequency, the frequency were changed according to specific pattern that repeated periodically.

3 – Antenna Beam width

An antenna which concentrates the radiated energy in narrow beam will give increased power density incident on the target and it will also be more effective in receiving the target echoes. A narrow antenna beam will also ensure that the radar will correctly identify the presence of target at same range but with small directional separation. Thus a narrow beam will provide good angular resolution which is a desirable capability.

4 – Frequency bands

Although radars generally use the radio frequency portion (normally 220–35,000 MHz) of the electromagnetic spectrum, they can also function in any spectral region. A few types of radar are found in lower band, except in cases where special propagation and target characteristics dictate their use. At the lower end, over-the-horizon radar uses the band from 6 to 30 MHz to utilize earth's ionosphere for reflecting a radar signal beyond the horizon.

Early in the development of radar, a letter code was employed to designate the radar frequency band. At the upper frequency end of the spectrum, L-band, S-band, C-band, and X-band radars are used where the size of the antenna constitutes a physical limitation. The other higher frequency bands (Ku, K, and Ka) suffer severe weather and atmospheric attenuation. Further information on common usage and applications of the radar frequency bands, are summarized in Table 1.1.

TABLE 1.1
Radar Frequency Bands and Usages

Usage	Frequency (GHz)	Band
OTH surveillance	0.003–0.03	HF
Very long-range surveillance	0.03–0.3	VHF
Very long-range surveillance	0.3–1.0	UHF
Long-range military and air traffic control search	1.0–2.0	L
Moderate-range ground-based and shipboard search	2.0–4.0	S
Search and fire control radars, weather detection	4.0–8.0	C
Short-range tracking, missile guidance, marine radar	8.0–12.5	X
High-resolution mapping, satellite altimetry	12.5–18.0	K_u
Police speed-measuring, airport surface detection	18.0–26.5	K
Very high-resolution mapping, airport surveillance	26.0–40.0	K_a
Laser range finders and optical targeting systems	40.0–300.0	MM- Wave

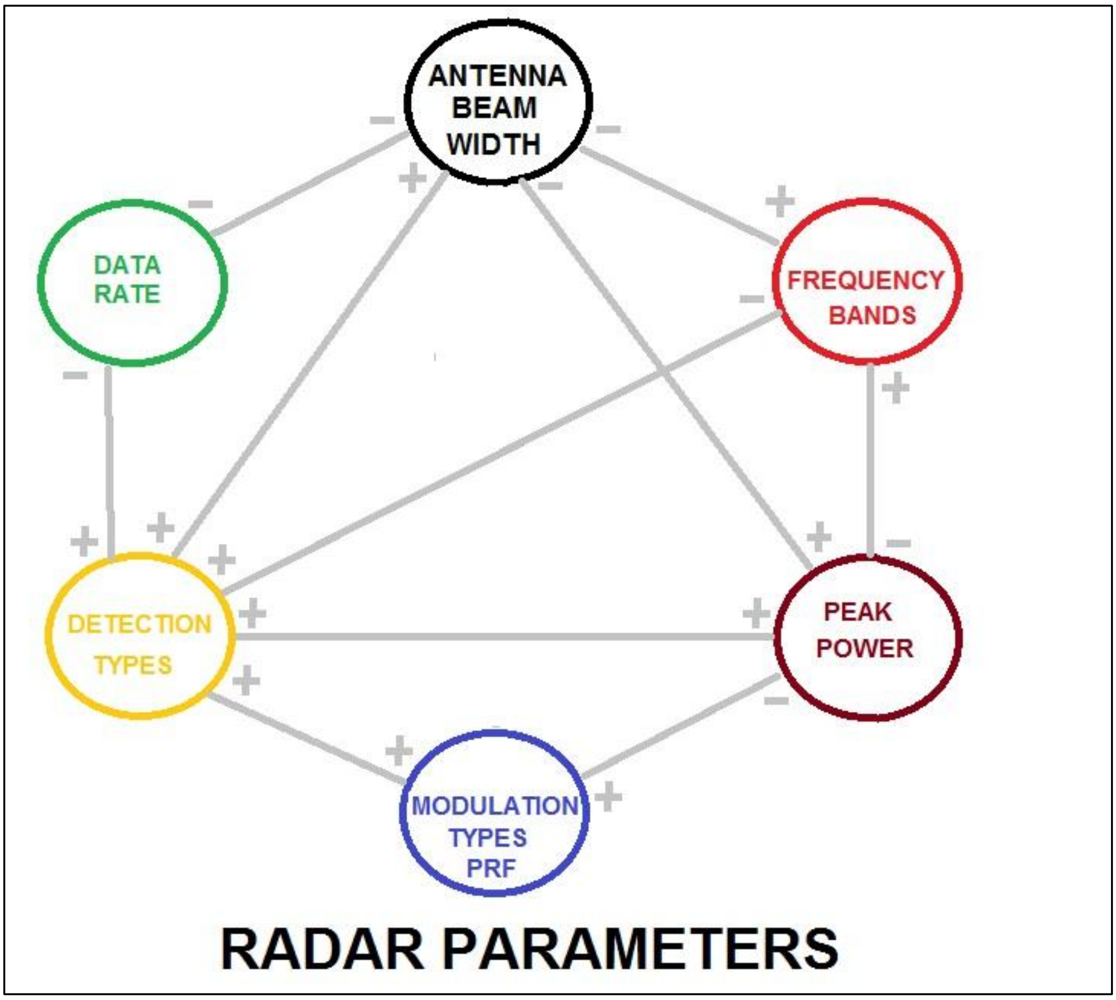
5 – Data rate

This is the rate at which fresh indications of target position are provided by the radar. With a rotating antenna the data rate is determined by the rotation rate. Although a high data rate may be thought desirable the rapid rotation of a narrow beam antenna will result in the transmission and reception of a very small number of pulses as the beam scans past the target and consequently MTI and detection performance is adversely affected. Thus the rate should be made as low as is feasible.

6 – Detection types.

The detection of echoes in the noise and clutter, background is best achieved by the use of matched filtering, possibly done in the receiver, followed by threshold comparison. With good echo signal to noise ratios say (20 dB or more) reliable detection can be achieved with the use of a single radar transmission. Echo's from many transmissions will be used if signal to noise ratio is not good. Target movement which results in change of range from the radar will produce Doppler frequency shift of the echo signal but for typical radar signal durations and target velocities the frequency shift cannot be determined or utilized with single transmission.

Consequently moving target indicator (MTI) required the use of a number of successive transmissions with appropriate digital filtering for the preferential selection and detection of echoes with specified Doppler frequency shift.



- ***<u>Radar Measurements.</u>***

The measurements needed for radar can vary considerably depending on the job to be done and the type or radar to be characterized. The maximum and minimum range, the range resolution, and the pulse repetition frequency, are related to in pulse radar, while the target location in azimuth and elevation the Doppler shift and the signal power received for echo signal are related to both continuous and pulse radar. In previous lecture we illustrated the maximum range, the pulse repetition frequency and the range resolution measurements, in this section the received signal power (radar equation) and the Doppler shift measurement will be illustrated.

1 – The Radar equation.

Radar range equation represents the physical dependences of the transmit power, which is the wave propagation up to the receiving of the echo signals. The power P_R returning to the receiving antenna is given by the radar equation, depending on the transmitted power P_T, the slant range R, and the reflecting characteristics of the aim (described as the radar cross-section σ). At the known sensibility of the radar receiver, the radar equation determines the achieved by a given radar theoretically maximum range. Furthermore one can assess the performance of the radar set with the radar range equation (or shorter: the radar equation).

1 - First, we assume, that electromagnetic waves propagate under ideal conditions, i.e. without dispersion. If high-frequency energy is emitted by an isotropic radiator then the energy propagates uniformly in all directions. Areas with the same power density, therefore, form spheres ($A = 4\pi R^2$) around the radiator. The same amount of energy spreads out on an incremented spherical surface at an incremented spherical radius. That means: the power density on the surface of a sphere is inversely proportional to the square of the radius of the sphere.

So we get the equation to calculate the Non-directional Power Density Su

$$Su = \frac{P_T}{4\pi R_1^2} \quad \text{...} \qquad (1.1)$$

P_T = transmitted power [W]
S_u = nondirectional power density
R_1 = range from transmitter antenna to the aim [m]

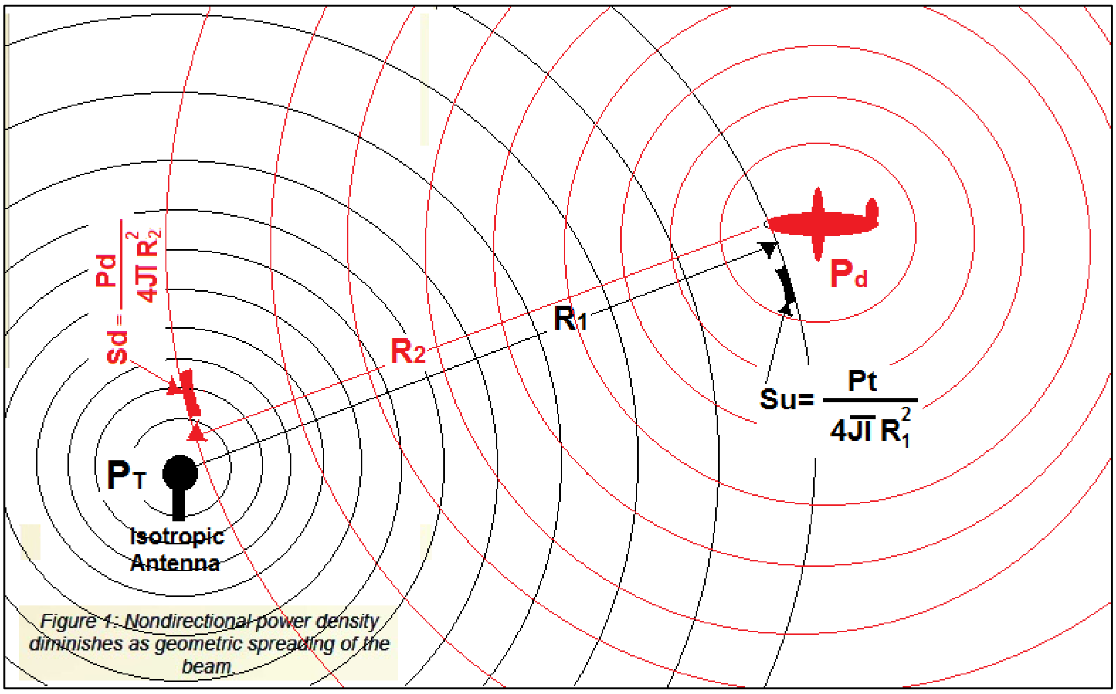


Figure 1: Nondirectional power density diminishes as geometric spreading of the beam.

Since a spherical segment emits equal radiation in all direction (at constant transmit power) if the power radiated is redistributed to provide more radiation in one direction, then this results in an increase of the power density in direction of the radiation. This effect is called antenna gain. This gain is obtained by directional radiation of the power. So, from the definition, the directional power density is:

$$Sg = Su \cdot Gt = \frac{P_T * Gt}{4\pi R_1^2} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots \quad (1.2)$$

S_g = directional power density
Gt = transmitted antenna gain

Of course, in reality, radar antennas aren't “partially radiating” isotropic radiators. Radar antennas must have a small beamwidth and an antenna gain up to 30 or 40 dB. (e.g. parabolic dish antenna or phased array antenna).

The target detection isn't only dependent on the power density at the target position, but also on how much power is reflected in the direction of the radar. In order to determine the useful reflected power, it is necessary to know the radar cross-section σ. This quantity depends on several factors. But it is true to say that a bigger area reflects more power than a smaller area. That means: An Airbus offers more radar cross-section than a sporting aircraft at the same flight situation. Beyond this the reflecting area depends on the design, surface composition and materials used.

If the previously mentioned is summarized, the **absorbent power** P_d by the target results from the power density S_u, the antenna gain G, and the target cross-section σ:

$$Pd = Sg * \sigma = \frac{P_T * Gt * \sigma}{4\pi R_1^2} \quad \text{.................................} \quad (1.3)$$

The target can be regarded as a radiator in turn due to the absorbent power. In this case, the absorbent power Pd is the emitted power.

Since the echoes encounter the same conditions as the transmitted power, the power density yielded at the receiver Sd is given by:

$$Sd = \frac{Pd}{4\pi R_2^2} \quad \text{..} (1.4)$$

S_d= power density at receiving place
P_d= reflected power [W]
R_2 = range aim - receiving antenna [m]

At the radar antenna, the received power P_R is dependent on the power density at the receiving site s_d and the effective received antenna aperture A_r.

$$P_r = S_d \cdot A_r \quad \text{...} (1.5)$$

P_R = received power [W]
A_r = effective received antenna aperture [m²]

The effective antenna aperture arises from the fact that an antenna suffers from losses, therefore, the received power at the antenna is not equal to the input power. As a rule, the efficiency of the antenna is around 0.6 to 0.7 (Efficiency).

Applied to the geometric antenna area, the effective antenna aperture is:

$A_r = A * K_a$ …………………………………………… (1.6)

A = geometric antenna area [m²]
K_a = efficiency

$$P_R = \frac{Pd*Ar}{4\pi R_2^2} \quad \text{………………………………………………..(1.7)}$$

The antenna gain G in terms of the wavelength λ.

$$Gr = \frac{4\pi*Ar}{\lambda^2} \quad \text{……………………………………………..(1.8)}$$

λ = wave length for RF signal [m]
G_r = received antenna Gain
Equation (1.8) can be written in the form

$$Ar = \frac{Gr*\lambda^2}{4\pi} \quad \text{………………………………………………(1.9)}$$

Substituting eq (1.9) in equation (1.7) P_R can be written as

$$P_R = \frac{Pd*\frac{Gr*\lambda^2}{4\pi}}{4\pi R_2^2} = \frac{Pd*Gr*\lambda^2}{(4\pi)^2 R_2^2} \quad \text{…………………………(1.10)}$$

But from Eq (1.3), $Pd = \frac{Ps*Gt*\sigma}{4\pi R_1^2}$

Substituting eq (1.3) in equation (1.10) P_R can be written as

$$P_R = \frac{\frac{P_T*Gt*\sigma}{4\pi R_1^2}*Gr*\lambda^2}{(4\pi)^2 R_2^2} \quad \text{……………………………………(1.11)}$$

$$P_R = \frac{P_T*Gt*\sigma*Gr*\lambda^2}{(4\pi)^3*R_1^2*R_2^2} \quad \text{…………………………….... (1.12)}$$

If the receiving antenna is itself the transmitting antenna

$Gr = Gt = G$ and $R_1 = R_2 = R$

$$P_R = \frac{P_T*G^2*\lambda^2*\sigma}{(4\pi)^3*R^4} \quad \text{………………..……………… (1.13)}$$

For given radar equipment most sizes (*Pt, G, λ*) can be regarded as constant since they are only variable parameters in very small ranges. The radar cross-section, the smallest received power that can be detected by the radar is called P_{Rmin}. Smaller powers than P_{Rmin} aren't usable since they are lost in the noise of the receiver. The minimum power is detected at the maximum range R_{max} as seen from the equation.

$$P_{Rmin} = \frac{P_T * G^2 * \lambda^2 * \sigma}{(4\pi)^3 * (R_{max})^4} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (1.14)$$

The maximum range that can be detected can be written depending on Eq. (1.14)

$$R_{max} = \sqrt[4]{\frac{PT * G^2 * \lambda^2 * \sigma}{(4\pi)^3 * P_{Rmin}}} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (1.15)$$

Since the P_{Rmin} specify the **sensitivity** of the radar receiver (S_{min}) equation (1.15) can be written in the term of radar receiver sensitivity as :-

$$R_{max} = \sqrt[4]{\frac{PT * G^2 * \lambda^2 * \sigma}{(4\pi)^3 * S_{min}}} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (1.16)$$

Also Eq.(1.16) can be written with respect to Ar instead of G as:

$$R_{max} = \sqrt[4]{\frac{PT * Ar^2 * \sigma}{(4\pi) * \lambda^2 * S_{min}}} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (1.17)$$

2 - Doppler effects.

The Doppler effects is “the effective change of frequency of a received signal due to the relative velocity between the transmitter (frequency source) and the receiver.” In radar it primarily manifests itself in the effect that the carrier frequency fr of received signal differs from the carrier frequency fo of the transmitted signal when reflected (or retransmitted) by the moving target. The Doppler shift is illustrated with the help of Figure (2.1). The field peak A departs at time t=t0 when the target is at range R0 and reaches the target after a travel time Δt, during which the target has advanced an additional distance $v\Delta t$, where v is the velocity of the target hence:

$$C_p \Delta t = R_0 + v\Delta t \quad \text{.....(2.1)}$$

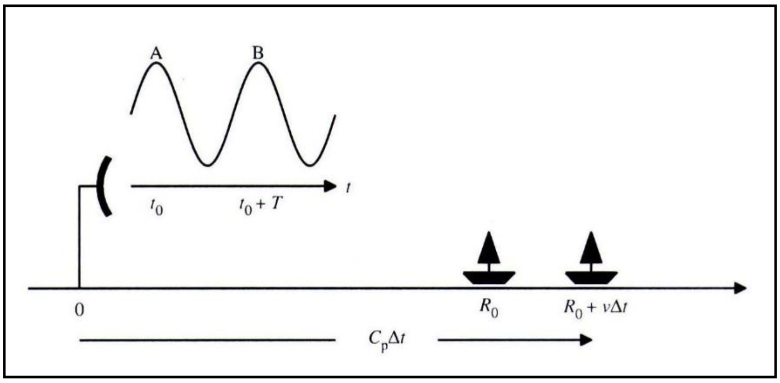


Figure (2.1) Timing in Doppler scene

where R_0 is the target location when peak A leaves the radar at ($t = t_0$), Δt is the travel time of peak A to reach the target, and $v\Delta t$ is the distance advanced by target during Δt , C_p is the velocity of light and T is the period of transmitted sinusoidal waveform.. Rewriting Eq. (2.1), the travel time is given by :

$$\Delta t = \frac{R_0}{C_p - v} \quad \text{....(2.2)}$$

The moment t_1 in which peak A returns to the radar is given by :

$$t_1 = t_0 + 2\Delta t = t_0 + \frac{2R_0}{C_p - v} \quad \text{.....(2.3)}$$

Similar expressions can be worked out for the second peak B, which left the radar T seconds after peak A and returned at t_2 :

$$t_2 = t_0 + T + \frac{2R_1}{C_p - v} \quad \text{.....(2.4)}$$

Where R_1 is the target location when peak B leaves the radar at ($t = t_0+T$) t_2 is the time of return of peak B to radar. Note that R_1 in Eq. (2.4) can be replaced by:

$$R_1 = R_0 + v\,T \qquad \text{……(2.5)}$$

The period of the received waveform T_R is equal to the difference between the arrival times of the two peaks:

$$T_R = t_2 - t_1 = t_0 + T + \frac{2(R_0 + vT)}{C_p - v} - \left(t_0 + \frac{2R_0}{C_p - v}\right) = T\frac{(C_p + v)}{(C_p - v)} \qquad \text{……(2.6)}$$

The ratio between the received and transmitted periods is therefore

$$\frac{T_R}{T} = \frac{C_p + v}{C_p - v} \qquad \text{……(2.7)}$$

And the ratio between the corresponding frequencies is :

$$\frac{f_R}{f_0} = \frac{C_p - v}{C_p + v} = \frac{1 - v/C_p}{1 + v/C_p} \qquad \text{……(2.8)}$$

Yielding the received frequency,

$$f_R = f_0\frac{1 - v/C_p}{1 + v/C_p} \qquad \text{……(2.9)}$$

In electromagnetic propagation the expected target velocities are always much smaller than the velocity of propagation $v << C_p$, yielding the approximation:

$$\frac{1}{1 + v/C_p} = 1 - \frac{v}{C_p} + \frac{v^2}{C_p^2} - \text{.....} \qquad \text{……(2.10)}$$

using equation (2.10) in (2.9) yields :

$$f_R = f_0\left(1 - \frac{v}{C_p}\right)\left(1 - \frac{v}{C_p} + \frac{v^2}{C_p^2} - \ldots\right)$$

$$= f_0\left(1 - \frac{2v}{C_p} + \ldots\right) \approx f_0\left(1 - \frac{2v}{C_p}\right) \qquad \text{……(2.11)}$$

Where $v << C_p$ rewriting Eq.(2.11) to get:-

$$f_R \approx f_0 - \frac{2v}{C_p / f_0} = f_0 - \frac{2v}{\lambda} \quad \text{...(2.12)}$$

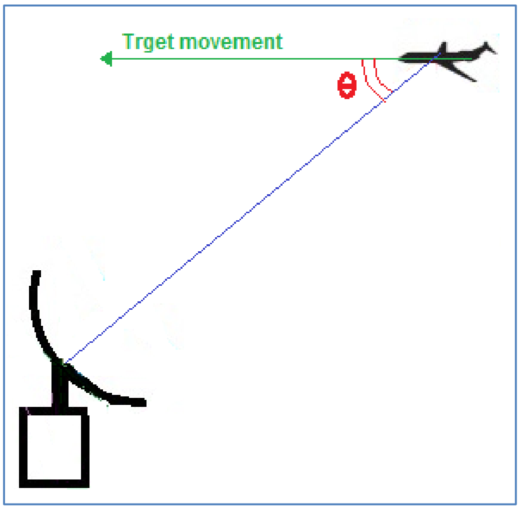
Where λ is the transmitted wavelength. The Doppler shift frequency is defined as:

$$f_d = f_R - f_0 = -\frac{2v}{\lambda} \quad \text{...(2.13)}$$

Note that Figure(2.1) is a special case in which the velocity is exactly in the radial direction, hence equal to range rate, $v = \frac{dR}{dt}$ where R is the radial distance. The more general approximation of Doppler shift is [36] :

$$f_d \approx -\frac{2\,dR/dt}{\lambda} \quad \text{...(2.14)}$$

the Doppler frequency shift can be written in the form of the direction angle between the target movement and the center of radar antenna (ɵ)



as:

$$fd = \frac{-2v\cos(\theta)}{\lambda} \quad \text{..(2.15)}$$

This lead to fd = 0 when ɵ = 90° and fd be +ve when ɵ large than 90° And vice versa.

COMMUNICATION SYSTEMS

RADAR SYSTEMS

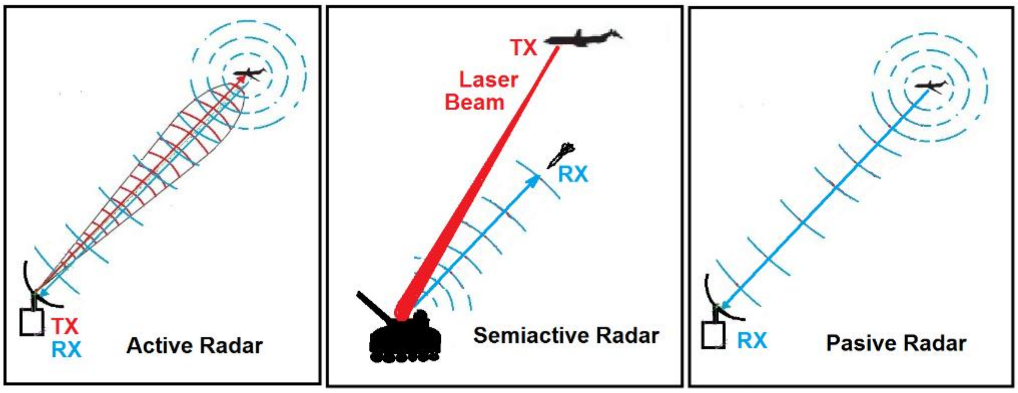
Lecture (3-3) **RADARS CLASSIFICATIONS**

When we start reading about radar, we come across various terms which are explained differently. There are various kinds of Radar classified in different ways. In this lecture the various radar classification ways will explained clearly. Radars can be classified by the following

1 – Classification based on operations technique or construction.
2 – Classification based on Physical configuration
3 - Classification by frequency band
4 - Classification based on specific function
5 - Classification by specific applications

1 – Classification based on operations technique or construction.

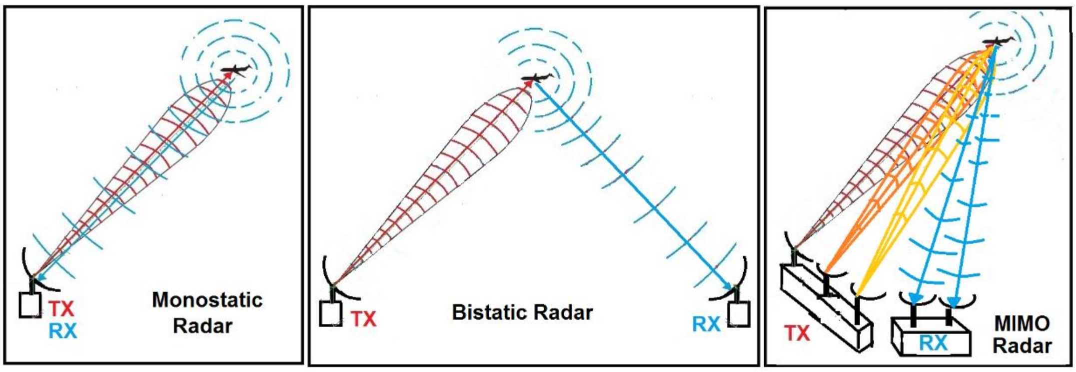
Radars can be classified based on operation technique into three types: - ***Active radars***, ***passive radars*** and ***semi-active radars***. Active radars those depend on transmit high power radio wave toward the target and received the reflected echo signal from the target, passive radar have only receiver that detect the radiated signal from the target like airplane radar or any sources of radio signal that transmitted from the target. While the semi-active radar receive the reflected echo signal that emitted by another source toward the target, these three types are illustrated in figure (1)



2- *Classification based on Physical configuration*

The physical configuration of the transmit and receive antennas also classify radars into monostatic, bistatic, and multistatic radars. Those radars where the same antenna is used for both transmission and reception are in essentially the same location are monostatic. In bistatic/multistatic radars the transmit and receive antennas are geographically placed in two/more different locations where the distance(s) of separation between them is (are) significant. The functions of the elements in bistatic/ multistatic radars are the same as monostatic radars, with the major difference being in the absence of the duplexer. A synchronization link between the transmitter and the receiver is necessary to maximize the receiver knowledge of the transmitted signal. Frequency and phase reference synchronization can also be maintained.

A *Multiple Input Multiple Output* (MIMO) radar system, a subset of multistatic radar, is a system of multiple antennas in which each transmit antenna radiates an arbitrary waveform independently of the other transmitting antennas and each receiving antenna can receive these signals. Due to the different waveforms, the echo signals can be reassigned to the single transmitter, these three types of radar (monoststic, bistatic and MIMO radars) are shown in figure (2) below.



3- *Classification by frequency band*

Although radars generally use the radio frequency portion (normally 220–35,000 MHz) of the electromagnetic spectrum, they can also function in any spectral region. The range of radar frequencies outside either end of the limit is shown in Figure (3). A few types of radar are found in lower band, except in cases where special propagation and target characteristics dictate their use. At the lower end, over-the-horizon radar uses the band from 6 to 30 MHz to utilize earth's ionosphere for reflecting a radar signal beyond the horizon.

Early in the development of radar, a letter code was employed to designate the radar frequency band. At the upper frequency end of the spectrum, L-band, S-band, C-band, and X-band radars are used where the size of the antenna constitutes a physical limitation. The other higher frequency bands (Ku, K, and Ka) suffer severe weather and atmospheric attenuation. Further information on common usage and applications of the radar frequency bands, are summarized in Table (1).

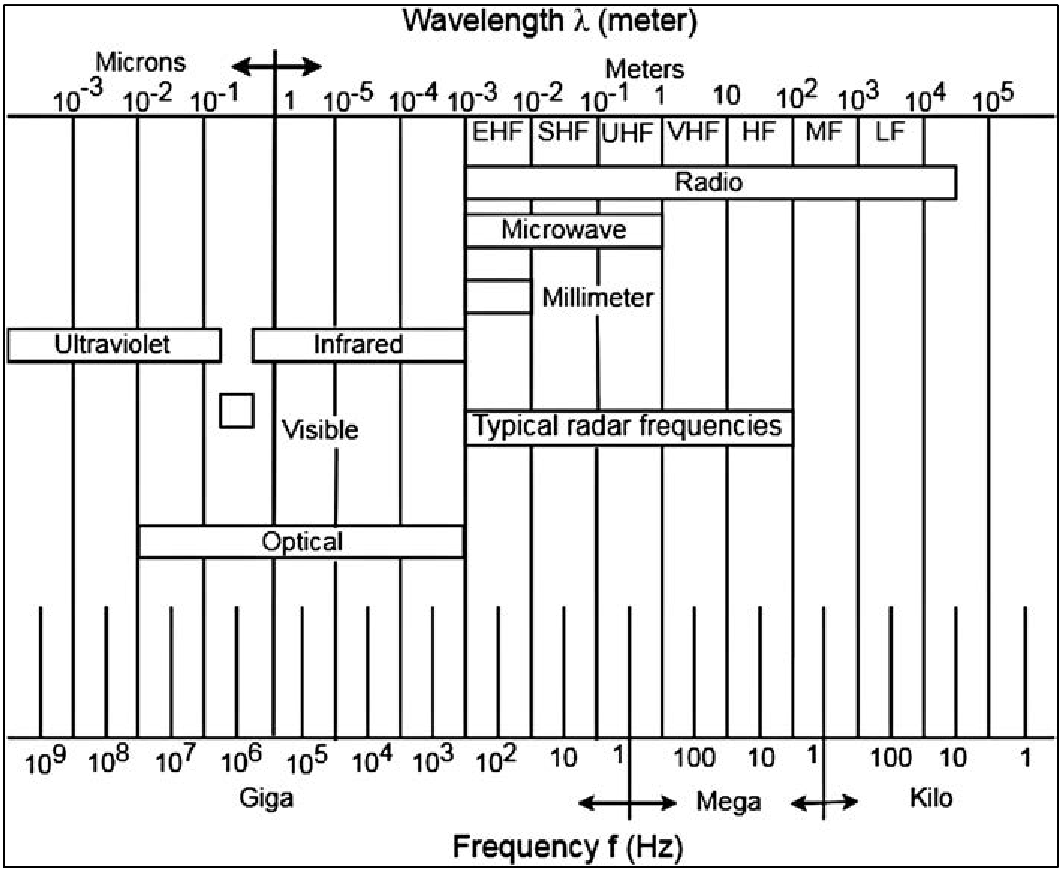


Figure (3) Electromagnetic spectrum & radar bands

TABLE 1 Radar Frequency Bands and Usages

Radar bands	Frequency in MHz	Usage
HF	**3-30**	**OTH surveillance**
VHF	**30-300**	**Very long-range surveillance**
UHF	**300-1000**	**Very long-range surveillance**
L	**1000-2000**	**Long-range military and air traffic control search**
S	**2000-4000**	**Moderate-range ground-based and shipboard search**
C	**4000-8000**	**Search and fire control radars, weather detection**
X	**8000-12500**	**Short-range tracking, missile guidance, marine radar**
KU	**12500-18000**	**High-resolution mapping, satellite altimetry**
K	**18000-26500**	**Police speed-measuring, airport surface detection**
Ka	**26500- 40000**	**Very high-resolution mapping, airport surveillance**
MM	**40000-300000**	**Laser range finders and optical targeting systems**

4 - Classification based on specific function

Classification based on the primary function of radar is shown in the following figure (4)

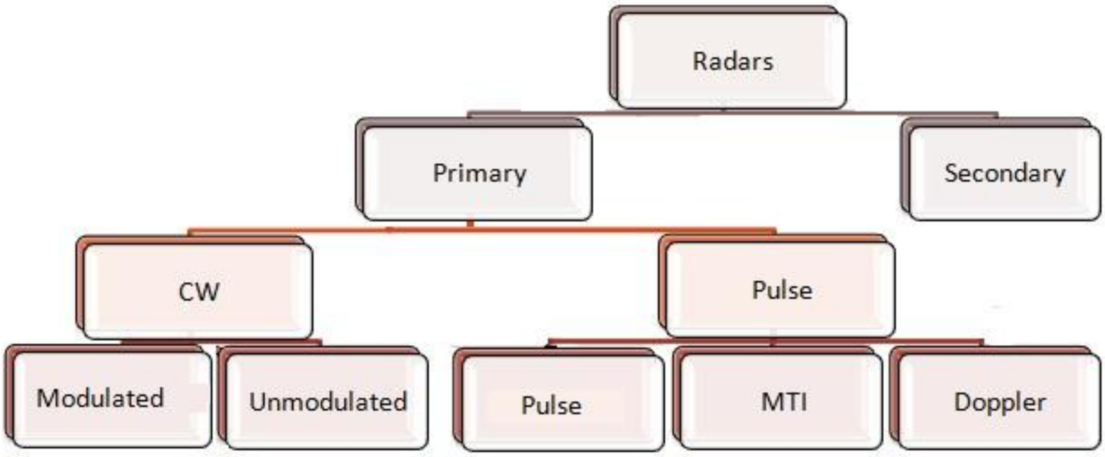


Fig. 4 Block Diagram Representing Different types of RADAR on the Basis of the Primary Function of RADAR

- ***- Secondary Radar:***

Secondary radar units work with active answer signals. In addition to primary radar, this type of radar uses a transponder on the airborne target/object. A simple block diagram of secondary radar is shown in figure (5) below

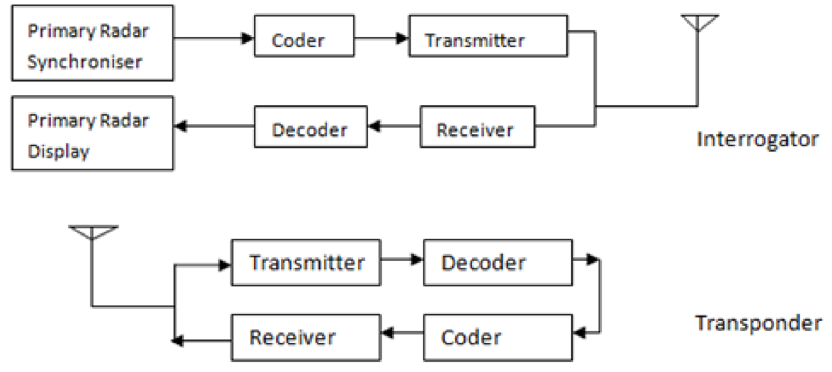


Fig. 5: A Block Diagram Demonstrating a Secondary Radar

The ground unit, called interrogator, transmits coded pulses (after modulation) towards the target. The transponder on the airborne object receives the pulse, decodes it, induces the coder to prepare the suitable answer, and then transmits the interrogated information back to the ground unit. The interrogator/ground unit demodulates the answer. The information is displayed on the display of the primary radar.

The secondary radar unit transmits and also receives high-frequency impulses, the so called interrogation. This isn't simply reflected, but received by the target by means of a transponder which receives and processes. After this the target answers at another frequency.

Various kinds of information like, the identity of aircraft, position of aircraft, etc. are interrogated using the secondary radar. The type of information required defines the MODE of the secondary radar.

- ***- Primary Radar:***

A Primary Radar transmits high-frequency signals toward the targets. The transmitted pulses are reflected by the target and then received by the same radar. The reflected energy or the echoes are further processed to extract target information.

Pulsed Radar:

Conventional pulsed radar transmits high power, high-frequency pulses toward the target. Then it waits for the echo of the transmitted signal for some time before it transmits a new pulse. Choice of pulse repetition frequency decides the range and resolution of the radar.

Target Range and bearings can be determined from the measured antenna position and time-of-arrival of the reflected signal.

The Pulse radar uses low pulse repetition frequency (PRF) to avoid range ambiguities, but these radars can have Doppler ambiguities.

Pulse Doppler radar:-

A pulse radar determines the speed of moving targets by measuring the Doppler frequency shift in the return signal. The pulse Doppler radar consists of the same components as conventional pulse radar with the addition of a CW oscillator. Like the conventional pulse radar, the pulse Doppler radar can determine presence of target, range, and direction to target. Additionally, with the aid of the CW oscillator, the speed of the target can be computed. The speed of the target is determined by comparing the frequency of the return signal to the transmitted signal. However, when the speed of the target causes a Doppler frequency equal to a multiple of the PRF, the radar cannot resolve the target's speed. Such speeds are termed blind speeds. To avoid this problem, the Doppler radar has a high PRF that avoids blind speeds, but high PRF causes ambiguities in range. Pulse Doppler radars are frequently used as airborne intercept, missile seeker, and fire-control radars.

Moving Target Indicator (MTI) Radar:

The MTI radar uses medium pulse repetition frequency (PRF) to avoid range ambiguities and Doppler ambiguities. The main difference between a pulse Doppler radar and an MTI radar is the desired information. Desired information from a pulse Doppler radar is the speed of the target. The PRFs are selected so as to avoid having blind speeds near the target's expected speed. For pulsed-MTI radars, the desired information is the detection and range of the target even in the presence of background clutter. In the pulsed-MTI radar, the PRF is chosen low enough to provide much longer unambiguous ranges than the PRF of a pulse Doppler radar but results in blind speeds and a frequency measurement that is ambiguous. MTI radars, therefore, detect moving targets but do not measure target speed. Pulsed-MTI radars are often used as air surveillance radars.

An MTI radar compares the phase of the target return to the phase of the previous target return and detects any difference in the two signals. The radar stores a sample of each target return for a time equal to the pulse repetition interval (PRI). Each target return travels over two signal paths in the receiver. One goes directly to the phase detector and then goes to the canceler circuit where it is compared with the stored copy of the previous return. The second goes through the delay line to the compression circuit. The delay is such that the target return appears at the end of the delay line just as the return from the next pulse arrives at the canceler circuit.

For the MTI radar to compare the phases of the target return signals, it must have a transmitted signal that is phase stable, or coherent, with respect to some reference. The MTI system maintains coherence by the use of a reference oscillator. If a power oscillator (e.g., a magnetron) is used, the phase of the reference oscillator is locked to the phase of each transmitted pulse. When a coherent oscillator (COHO) is used, the transmitted signal is produced by keying a highly stable signal from the continuously running COHO.

Continuous Wave Radar:

CW radars continuously transmit a high-frequency signal and the reflected energy is also received and processed continuously. These radars have to ensure that the transmitted energy doesn’t leak into the receiver (feedback connection). CW radars may be bistatic or monostatic; measures radial velocity of the target using Doppler Effect. CW radars are divided into two types ;-

A. Unmodulated

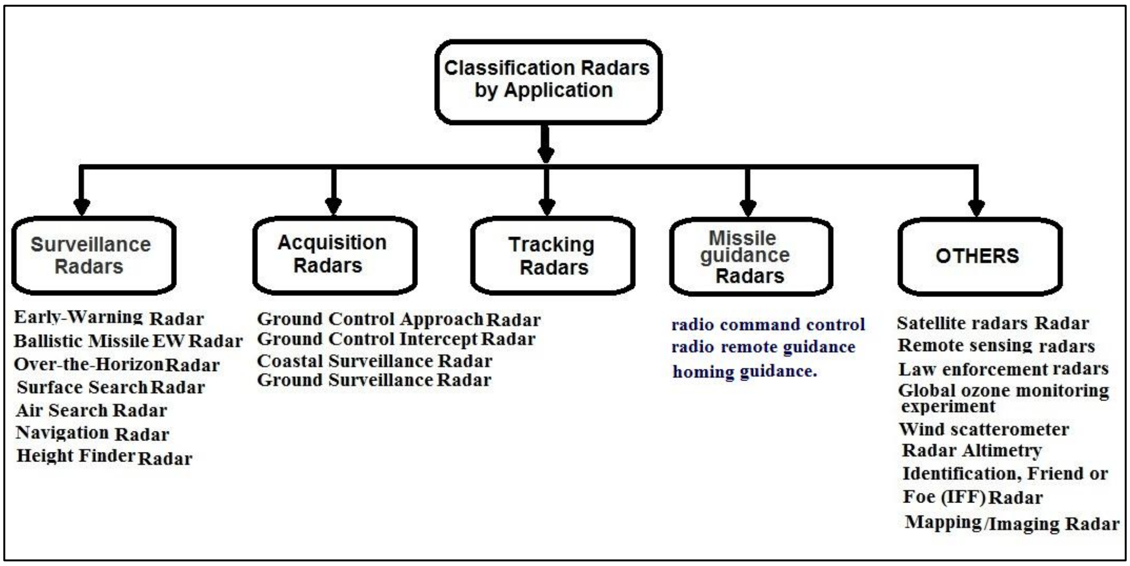
An example of unmodulated CW radar is speed gauges used by the police. The transmitted signal of these equipment's is constant in amplitude and frequency. CW radar transmitting unmodulated power can measure the speed only by using the Doppler-effect. It cannot measure a range and it cannot differ between two reflecting objects.

B. Modulated

Unmodulated CW radars have the disadvantage that they cannot measure range, because run time measurements is not possible (and necessary) in unmodulated CW-radars. This is achieved in modulated CW radars using the frequency shifting method. In this method, a signal that constantly changes in frequency around a fixed reference is used to detect stationary objects. Frequency is swept repeatedly between f1 and f2. On examining the received reflected frequencies (and with the knowledge of the transmitted frequency), range calculation can be done.

5- Classification by specific applications

Radar systems may also be classified according to specific mission and application into two main types Surveillance (or search) radars, acquisition radars, tracking radars, and missile guidance radars.



1 - Surveillance (search) Radars

Primary application of radar is surveillance. These radars typically use high power, scanning antenna and have moderate resolution. The search radar is designed to give warning of approaching targets. Search radars typically use circular, sector, or electronic scans. The search radar may be limited to a single dimension, range only; or may be capable of two dimensions (2D), bearing and range; or three dimensions (3D), bearing, range, and elevation. Other characteristics of the search radar are low illumination rate, large scan volume, and high transmitter power. Examples of search radars are early warning (EW), surface search, airborne search, navigation, and height finder. They are deployed for

- Detection and Tracking of Aircraft, Missiles or Space Objects
- Detection of Fixed or Moving Surface Targets
- Moderate precision tracking of multiple targets

Some of the important applications of Surveillance radars are

- **- Early-Warning Radar**

The primary function of the early-warning (EW) radar is to alert the defense forces of approaching targets as early as possible. This radar function is also termed long-range search. Critical operational features of an EW radar are range and power. Also, an EW radar should resolve the position of the target within a reasonable margin of error. However, the accuracy of range and bearing should not be sacrificed at the expense of maximum range. Therefore, EW radars are normally designed with high power, wide beam angles, fairly long pulse duration, and low PRF. Since it is unnecessary to frequently update the position data on a long-range target, relatively slow circular scans are used. In effect, this slow scan enables a greater number of pulses to illuminate the target, thus providing a better chance of detection at great distances. The nature of the terrain is also important. In rugged mountain areas, the EW radar would be situated to provide maximum coverage. Since mountain ranges produce "blind spots" or gaps in coverage, additional EW radars are required to cover these areas. These radars would be linked to the main radar site by communications links. An EW radar site usually has a separate height finder to refine target position if the EW radar does not have 3D capability. The maximum operational range of an EW radar is usually between 400 and 800 kilometers (kms), but may extend to extremely long ranges. Typical EW radars would have the following parameters:

- RF: 100-4000 MHz
- PRF: 50-400 pulses per second (pps)
- PD: 1-14 microseconds (µ)
- Scan: slow (circular at 4-30 seconds per revolution (SPR))

- **- Ballistic Missile Early-Warning Radar**

The ballistic missile EW (BMEW) radar, a subcategory of the EW radar, detects ballistic missiles in mid-trajectory at ranges of 3000 to 15,000 kms and at altitudes of 1000 to 2000 kms. The BMEW also monitors satellite trajectories for monitoring orbital decay of satellites and the estimated time that a satellite will remain in orbit. Typical BMEWS radars would have the following parameters:

- RF: 200-500 MHz
- PRF: CW-50 pps
- PD: 100-600µ

- Scan: electronic

- **- Over-the-Horizon Radar**

Another radar system with extremely long detection ranges is the over-the-horizon (OTH) radar. The curvature of the earth limits the maximum detection range of normal ground-based radar sites. The OTH radar was developed to overcome this problem. The OTH radar uses the layers of the ionosphere to refract the radio waves, enabling it to detect and track targets beyond the radio horizon. The OTH radar can detect aircraft, missiles, and ships, as well as nuclear explosions, earth surface features (e.g., mountains, islands, and cities), seas, auroras, meteors, and satellites. The following are two types of OTH radars:

- Back scatter—Uses the same radar to transmit and receive. The signal is radiated up to the ionosphere where it is refracted. The target is detected when the signal reflects off of it and is re-radiated up to the ionosphere where it is again refracted to the radar antenna.
- Forward scatter—Uses a transmitter and receiver that are spaced far apart. The signal is radiated up to the ionosphere where it is refracted back to earth. It reflects from the earth and strikes a target. The energy from the target changes direction and continues to radiate until it reaches the receiver.

Typical OTH radars would have the following parameters:

- RF: 3-40 MHz
- PRF: CW-180 pps
- PD: 160-21000μ
- Scan: frequency steering, irregular

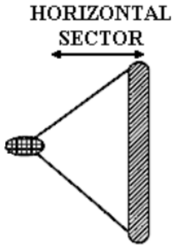
- **- Surface Search Radar**

The primary function of a surface search radar is the detection and determination of accurate range and bearing information of surface targets and low-flying aircraft. Surface search radars maintain a 360° search pattern for all objects within line-of-sight (LOS) from the radar antenna. Search radar information can be used as an input to the weapons system. Shipboard surface search radar is also used extensively as a navigation aid in coastal waters and in poor weather conditions.

The maximum range ability of a surface search radar is primarily limited by the radar horizon LOS. Higher RF permits maximum reflection from small target reflecting areas, such as ship masthead structures and submarine periscopes. Narrow pulse duration permits a high degree of range resolution at short ranges and has greater range accuracy. High pulse repetition rates permit maximum illumination of targets. Appropriate peak powers permit detection of small objects at LOS distances. Wide, vertical beamwidths, usually between 12° to 16°, permit

Compensation for pitch and roll of the ship and detection of low-flying targets, while narrow, horizontal beamwidth, approximately 1.5° , permits accurate bearing determination and good bearing resolution. Typical surface search radars would have the following parameters:

- RF: 5450-5825 MHz
- PRF: 625-650 pps
- PD: .25 – 1.3μ
- Scan: circular and sector



• **- Air Search Radar**

This radar detects and determines the range and bearing of aircraft within a 360° surveillance area. The maximum range can exceed 300 miles. Air search radars typically use low RF to permit long-range transmissions with minimum attenuation. Wide PD and high peak power is used to aid in detecting small objects at longer ranges. Low PRFs provide for greater maximum ranges. A wide, vertical beamwidth is used to ensure detection of objects from the surface to relatively high altitudes and to compensate for the pitch and roll of the ship. Medium horizontal beamwidths provide fairly accurate bearing information.

In addition to providing range and bearing information, some air search radars also provide height information for air contacts. These three-dimensional (3-D) radars provide range, bearing, and altitude information. Two-dimensional (2-D) radars provide only range and bearing information. Air search radars are used to provide early warning and to guide combat air patrol (CAP) aircraft to intercept enemy aircraft.

- **- Navigation Radar**

The primary mission of a navigation radar is to guide an aircraft or ship from one location to another. Used to keep them from colliding with other obstacles, navigation radars also permit operations in less than optimal weather and enable the pilot to navigate by instruments. Navigation radars provide continuous information as to a pilot's position with respect to a desired course, his proximity to other craft and ground obstacles, and miscellaneous meteorological information. Since navigation radars on aircraft or ships provide only a part of the information needed, other navigational aids (e.g., beacons) are necessary to provide safe travel.

Airborne navigation radars perform many functions. They are used for terrain avoidance or terrain following. The weather avoidance radar aids the pilot in avoiding areas of unfavorable weather. Although it may not always be thought of as a radar, the radio altimeter is used as a navigational aid.

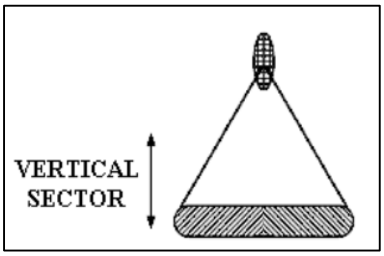
Shipborne navigation radar is also used for enhancing the safety of ship travel by warning of potential collision with other ships and for detecting navigation buoys, especially in poor visibility. Automatic detection and tracking equipment is available for use with radar for the purpose of collision avoidance. Shore-based radar of moderate resolution is also available for the surveillance of harbors as an aid to navigation.

Global navigation systems are a great aid to navigators to keep their craft from becoming lost or allowed to stray into restricted areas. Nearly always, aviation uses short-range radio navigation aids such as very high frequency

(VHF) omnidirectional range/distance measuring equipment (VOR/DME). Marine uses include non-line-of-sight radio systems such as LORAN-C, Decca, and Transit. These systems aid in position determination of the craft with a relatively high degree of accuracy.

- **- Height Finder Radar**

Height finder radars provide altitude information about airborne targets and are normally co-located with EW radars. The height finder radar uses a fan beam that is wide in the horizontal plane and narrow in the vertical plane. By moving (or scanning) this beam vertically, the radar measures the elevation angle and range of a target. The height finder radar can be scanned in elevation mechanically by physically moving the feed point or the entire antenna assembly, or by electrically positioning the beam through the use of electronic phase shifters. The range of this radar is generally about 150 to 450 kms and provides an elevation sector of between 0° and 60° . Typical height finder radars would have the following parameters:



- RF: 2500-5500 MHz
- PRF: 100-900 pps
- PD: 1-4μ
- Scan: sector at 1-10 seconds per sector (SPS)

2 <u>- ACQUISITION RADAR</u>

The acquisition function usually occurs following the search mode after a target is detected. The acquisition radar takes a specific target and performs fine grain measurements of target range, azimuth, elevation, and speed to hand off to a target tracker radar without further search. During the handoff from the search radar to the acquisition radar a change to a higher RF occurs. The target acquisition radar generally has an operating range of 100 to 400 kms and an azimuth resolution of 01° . Other characteristics include higher target illumination rates, smaller search volume, and moderate transmitter power. Typical scan patterns for an acquisition radar are circular, sector, and raster scans. Other scan patterns that can be used are spiral, helical, palmer, and nodding scans, though they are not often used.

Acquisition radars cover short to medium ranges and serve as position indicators for other radar sets. Examples of acquisition radars are Ground Control Approach/Precision Approach, Ground Control Intercept, Coastal Surveillance, and Ground Surveillance.

- **- Ground Control Approach/Precision Approach Radar**

The ground control approach (GCA) radar is a short-range acquisition radar used to provide control of aircraft in the vicinity of an airfield. The range of this radar is generally between 30 and 260 kms. The scan is a circular scan of 2 to 10 SPS. As the aircraft approach within approximately 35 kms of the airfield, the precision approach radar (PAR) takes over. This radar obtains data on the position of aircraft relative to the runway and guides the aircraft to a safe landing. This system is extremely helpful in bad weather. The PAR has a range of up to 35 kms and has a circular scan of less than 2 SPS.

- **- Ground Control Intercept Radar**

The Ground Control Intercept (GCI) radar is similar to, and can function as, an EW radar; however, the parameters of the GCI radar have a shorter maximum range. Usually located at military airfields, the GCI radar site provides target position information to vector defense aircraft to hostile targets. The parameters of the radars in this category are generally selected so that a compromise results between the range of the radar and the accuracy of the control system. The range of the GCI radar is typically 200 to 400 kms. Typical GCI radars would have the following parameters:

- RF: 100-4000 MHz
- PRF: 300-500 pps
- PD: 2-6m
- Scan: circular or sector at less than 20 SPR/SPS

- **- Coastal Surveillance Radar**

The coastal surveillance (CS) radar provides information about ship movement for either coastal defense or ship movement control. A height finder radar may be co-located to provide information on aircraft.

- **- Ground Surveillance Radar**

There are two basic types of ground surveillance radars: the battlefield surveillance (BS) radar and the mortar locating radar (also known as the shell-tracking radar). The BS radar detects and locates enemy vehicles, weapons, and personnel in the immediate area. The mortar locating radar determines the location of enemy weapons by tracking their projectiles. Mortar locating radars provide inputs to direct return fire and are, therefore, called field artillery radars.

BS radars are divided into medium-range radars with a detection range for large targets (e.g., moving trucks or tanks) of 15 to 20 kms. These radars are installed on trucks or trailers, while short-range radars are transported or carried in manpacks. Medium-range BS radars can detect traffic by detecting dust, smoke, or heat given off by vehicles. The battalion or equivalent units use this class of radars. Short-range ground reconnaissance radars usually have a maximum detection range of 5 to 10 kms and detect targets as small as an individual soldier. Radars of this type are used by companies, platoons, and reconnaissance patrols, as well as guard posts. Short-range radars generally use a manual or steady scan and are capable of detecting only moving targets by relying on the Doppler effect.

3 - TRACKING RADAR

Tracking radars (Also called fire control radars) are used to provide range and bearing information of a single target continuously. These radars use very high PRF, very narrow pulse width as well as beam width. This allows these radars to have high accuracy, limited range, and initial detection of the target a bit difficult. They typically take range and bearing information from the search radars. Until a target is located, they remain in acquisition phase searching for the target. Once a target is located, they enters track phase and automatically follow target motions.

. Tracking radars provide continuous positional data on a target and are designed to precisely locate targets and track location of the target for possible input into weapon systems. Since these radars are designed for shorter ranges, the output power is lower. Radars of this type are characterized by short emitter times, small target illumination rate, and small scan volume. A tracking radar system provides range, speed, and angle to achieve fine-grain positional data that may be used to determine the path and future location of the target.

When the radar is used for tracking, it usually cannot detect other potential targets. Therefore, many tracking systems use a separate search radar to provide the tracker with the target position. A tracking radar's range is generally short and limited to the range of its corresponding weapons system. Range resolution is on the order of a fraction of a kilometer and with an azimuth resolution of l° . Typical tracking radars would have the following parameters:

- RF: 2000-50,000 MHz
- PRF: 500-300,000 pps
- PD: .1-2µ
- Scan: complex

4 - MISSILE GUIDANCE RADAR

A radar that provides information used to guide a missile to a hostile target is called a guidance radar. A typical missile guidance signal has a medium to high RF, low to high PRF, a narrow PD, and may have a steady scan or no scan. Other characteristics include a small target illumination rate, small scan volume, and pulse compression. Missiles use radar to intercept targets in three basic ways: radio command control, radio remote guidance, and homing guidance.

Also there are other types of radars that cannot classified with the above radars types.

. **Satellite radars**: Radars used by satellite for rendezvous and docking and large ground- based radars are used for detection and satellite tracking.

. **Remote sensing**: Radars provide information about the geophysical objects, and are used in astronomy to probe the moon and the planets.

. Law enforcement: Radars are used by highway police to measure the speed of automobile traffic, and to detect the intruders.

. Global ozone monitoring experiment (GOME): Atmospheric available ozone monitoring is sometimes needed for many applications. GOME products can be used for retrieving other trace gases relevant to the ozone chemistry as well as other atmospheric constituents. Furthermore, it can be used for climatic variable clouds, solar index, and aerosols. All these are crucial for assessing climate change.

. Wind scatterometer (WSC): Wind scatterometers are used for accurate measurements of the radar backscatter from the ocean surface when illuminated by a microwave signal with a narrow spectral bandwidth to derive information on ocean surface wind velocity.

. Radar Altimetry FMCW radars are used for measurement of altitude above ground level. Principle of operation of FMCW, these radars provide very precise altitude information.

. Identification, Friend or Foe (IFF) Radar

Identification, friend or foe (IFF) is a radar beacon system designed to identify an airborne platform as a friendly or unknown aircraft. The IFF system consists of an interrogator and a transponder subsystem. An IFF system normally operates in conjunction with a parent radar. The pulses transmitted by the IFF interrogator are synchronized with those of the aircraft search radar (parent radar), and the IFF antenna is mounted on the search radar's antenna.

IFF is similar to conventional pulse radars except that the receiver processes a signal radiated from a transmitter and returned by the target rather than the energy returned by the target. An aircraft's IFF system transmits interrogating pulses to which transponders carried in all friendly aircraft respond with coded replies. This results in the challenged transponder providing location and coded information in the returned RF pulse. The received coded RF pulses are converted into decoded video for display.

Advantages to IFF include enhanced target returns, a means of target identification, and a way to provide information about the target to the radar. Since there is only a one-way path from the interrogator to the transponder and from the transponder to the interrogator, the output power of both transmitters remains relatively low while achieving acceptable performance. Separate transmitting and reply frequencies eliminate ground clutter and weather return problems. Dependence upon target size (as with a radar echo signal) is virtually eliminated. Finally, data can be included in the interrogation and reply process, thereby providing target identification, altitude readout, fuel remainder, as well as other information.

- **-MAPPING/IMAGING RADAR**

Mapping radars provide surface images with nearly photographic resolution. They are employed in many nonmilitary applications for remote sensing of the environment, mapping of regions of the world under cloud cover too heavy for optical photography, mapping of agriculture, geological exploration, location and density of sea ice, and detecting mineral and oil resources. For military applications, they reproduce replica of ground terrain for navigation and identification of military targets.

To obtain images of the earth's surface or individual targets, high resolution in both range and azimuth is needed. To achieve a good resolution in azimuth, a narrow beam is used. To achieve a good resolution in range, pulse compression is used.

One type of ground mapping radar that achieves very high azimuth resolution is the synthetic aperture radar (SAR). SARs use a coherent RF and the Doppler phenomenon to effectively process returns over a segment of the radar platform flight path. This produces an effectively large (synthetic) antenna and improves the angular resolution. Pulse compression techniques are often used to improve range resolution. High-resolution radars may also use inverse synthetic aperture radar (ISAR) techniques. Instead of depending on the movement of the platform as in SAR techniques, the ISAR depends on the movement of the target with respect to a stationary radar antenna that is constantly illuminating the target. Another type of mapping radar is the Real Aperture Mapper. This radar is also known as the brute force real aperture. Real Aperture Mappers use standard radar techniques to measure the distance to the ground. The pulse duration and beamwidth of the radar, therefore, limits the resolution.

Mapping radars are normally fitted aboard aircraft or satellites. The parameters of mapping radars are matched to the platform performance and intended coverage. Most mapping radars have no active scan; instead they use the platform's motion to move the radar beam along the earth's surface. Also, the angle of incidence must be accounted for when determining the radar's resolution. The satellite mapping radar has the advantage of being able to do mapping in extreme weather conditions and during nighttime hours, when mapping from an aircraft is much more difficult.

COMMUNICATION SYSTEMS

RADAR SYSTEMS

LECTURE (4 - 3) RADAR SCAN TYPES

Scanning is the process of searching a large volume of space by moving the antenna or shifting the radiation pattern while looking for, or tracking, a target. The type and method of scanning used depends on the purpose and type of radar and on the antenna size and design. Knowledge of the type of scan used by a radar provides valuable information about the purpose of a radar. As a general rule, scans requiring less than 1 second to complete a full cycle are reported in hertz (Hz) or SPS, and scans requiring more than 1 second are reported in seconds or SPR.

Most radars transmit a continuous pulse train of relatively narrow, high energy pulses shaped into a directional beam by the antenna reflector. The antenna is pointed toward the desired volume of space. When illuminated, any sufficiently large object will return energy to the radar antenna. The returned energy is detected and displayed by the radar. The purpose of the directional beam and the scan motion is to provide azimuth and elevation angle information.

Operational requirements, for example search mode, place restrictions on the maximum scan time. The physical requirements cause the space around the radar to be divided into resolution cells that depend on the antenna beamwidth and pulse duration. The antenna scan rate is normally a compromise between the rate at which target position information is desired (the scan rate) and the data rate (the number of pulses on the target). The slower the scan rate, the more pulse returns and the better the detection capability but the more inaccurate the target position. Increased scan rate reduces the error in estimates of target position, but also reduces the probability of detection.

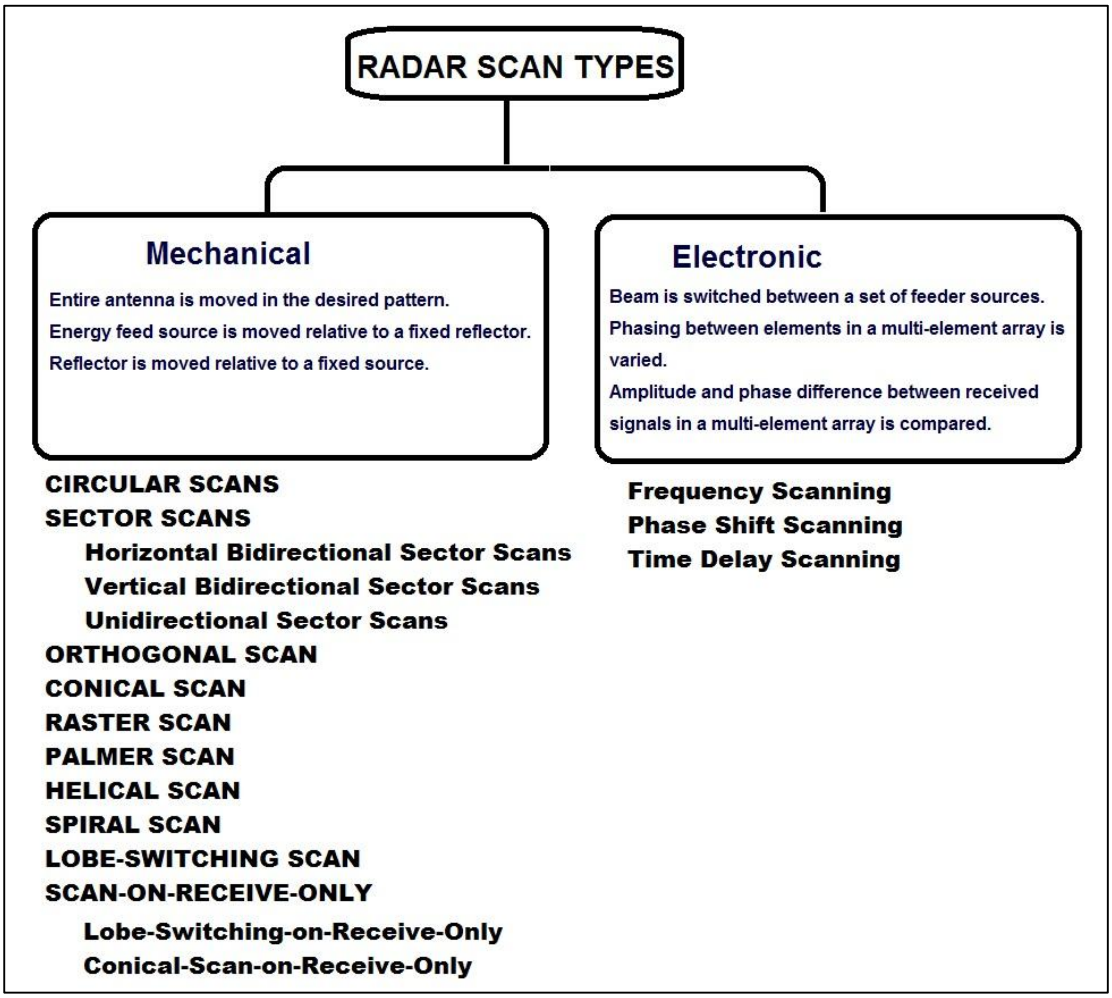
The beam shape of the radar is formed by the configuration and design of the antenna. The beam shape is one of the parameters that serve as a general indicator of the emitter function. Search radars generally use a fan beam (one that is wide in one plane and narrow in the plane perpendicular to it); target tracking and acquisition radars most often use a pencil beam (one that is of nearly equal beamwidth in both azimuth and elevation). For the pencil beam to be used effectively, the radar must be capable of rapidly aiming the beam in the desired direction. Some search radars employ a pencil beam that is swept electronically in one direction and mechanically in the other, thereby effecting a fan beam in the direction of the swept beam. The two basic methods of beam scanning are mechanical and electronic. In each method, the beam is moved in various ways as described in the following:

Mechanical

- Entire antenna is moved in the desired pattern.
- Energy feed source is moved relative to a fixed reflector.
- Reflector is moved relative to a fixed source.

Electronic

- Beam is switched between a set of feeder sources.
- Phasing between elements in a multi-element array is varied.
- Amplitude and phase difference between received signals in a multi-element array is compared.



- **- CIRCULAR SCANS**

A circular scan is the most common form of mechanical scanning. In circular scanning antennas, the antenna continuously rotates 360° about a vertical axis in either a clockwise or counterclockwise direction. The time between successive signal sweeps is called the scan time. For a circular scan, this time must be constant. If it is not, the scan may be a special case of circular scan or a sector scan. See figure (1)

Search radars commonly use circular scans. Long scan times are associated with long-range search and acquisition radars. Medium scan times are associated with ground-controlled intercept and surface search. Fast scan times are associated with airborne navigation or airport surveillance.

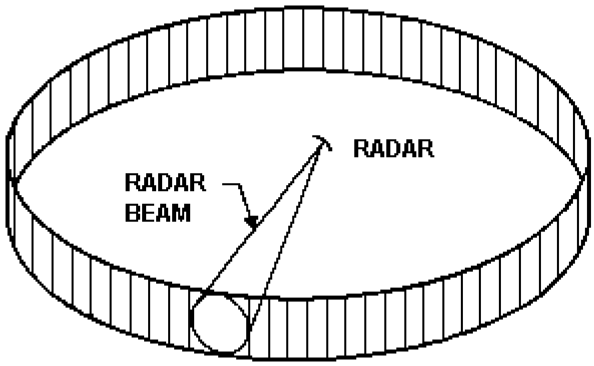


Figure 1: Circular scan

Search type scans, like the circular, normally use an antenna that is wide in the horizontal plane and narrow in the vertical plane. This results in a fan beam that is wide in the vertical plane and narrow in the horizontal plane. See figure (2). Some radars have only a circular scan that provides target range and azimuth information. These radars are often co-located with height-finding radars to provide elevation information. Other radars are designed to provide range, azimuth, and elevation information with varying degrees of accuracy in a single radar set by using multiple beams or by superimposing a vertical scan onto the circular rotation.

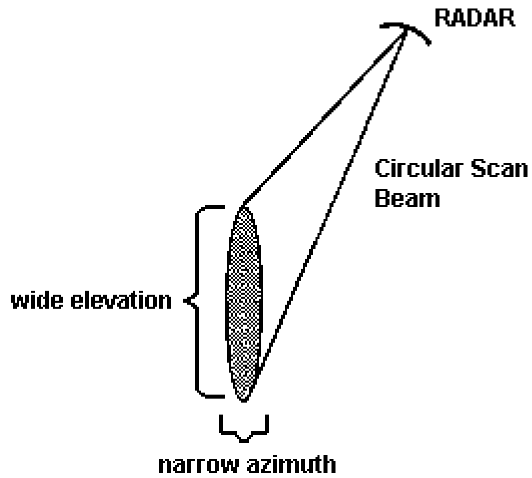


Figure 2: Circular scan antenna output

The speed of the antenna rotation must be compatible with the PRF and the beamwidth of the radar. To obtain optimum resolution of a target, the scan and beamwidth must allow several pulses to strike each target as the beam crosses. Therefore, the scan period, beamwidth, and PRF are interdependent. The horizontal beamwidth of a circular scan is normally narrow to provide the radar with good azimuth resolution.

- **- SECTOR SCANS:-**

There are two types of sector scans, bidirectional and unidirectional. A bidirectional sector scans back and forth (either horizontally or vertically) through a desired sector. A unidirectional sector scans in only one direction. The main identifying feature of a bidirectional sector scan is the uneven or irregular time intervals at which the signal bursts are appeared.

The easiest way to distinguish between the two scan types is to look at the lobe structure on alternate scans of figure (3). As the radar beam moves from the right sector limit to the left sector limit, a signal burst will be heard when the radar beam strikes intercept point A. The radar beam continues until it reaches the left sector limit, where it starts back to the right sector limit. As the beam moves toward the right limit, it again strikes intercept point A. The beam continues until it reaches the right limit, then begins back toward the left limit.

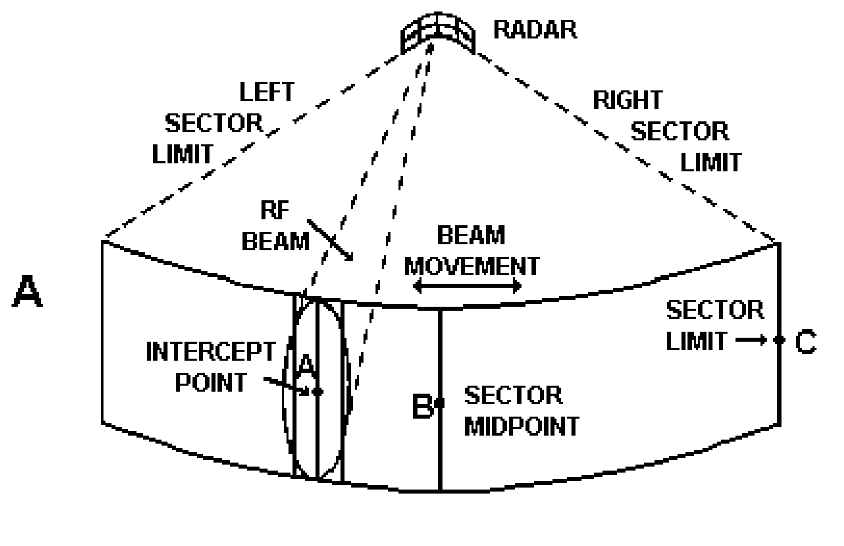


Figure 3. Horizontal sector scan.

Horizontal Bidirectional Sector Scans

A horizontal bidirectional sector scan sweeps its beam back and forth horizontally through a desired sector. The amount of sector sweep may be either variable or fixed, depending on the radar. The antenna scans in both directions. The horizontal sector provides search and early-warning radars with coverage of a smaller area than that of a circular scan, but the coverage area may be illuminated at a faster rate.Examples of this scan are airborne navigation, weather avoidance, and target acquisition. Horizontal sector scans are also used for navigation by ships and some coastal surveillance radars. A horizontal bidirectional sector scan normally has a scan period of 1 to 5 seconds.

Vertical Bidirectional Sector Scans

A vertical bidirectional sector scan sweeps its beam back and forth vertically through a desired sector. The antenna scans in both directions. The vertical sector is commonly used for height finding. Vertical sector scans operate and are measured in the same fashion as horizontal sector scans. The only difference is that the radar antenna moves up and down for the vertical sector scan, instead of back and forth horizontally. A vertical bidirectional sector scan typically has periods of 2 to 4 seconds.

Many land-based, height-finding radars use the vertical bidirectional sector scan with antennas that mechanically move or nod in an up and down motion. In some cases, the antennas rotate in a circular pattern while simultaneously nodding.

Unidirectional Sector Scans

A unidirectional sector scan radar moves its beam through a desired sector, usually at a high rate (i.e., from 10 to 50 Hz). These scans can be horizontal or vertical. However, since an ES operator cannot determine the plane of orientation, all unidirectional sector scans are usually classified "plane undetermined." The unidirectional sector scanning radar produces a rapid scan by electronically shifting through a series of multiple feeds or by rapid rotation of a single feed (figure (4). The beam is reflected from a fixed antenna. The entire antenna can be rotated to change the area being scanned or to follow a detected target. As the name unidirectional implies, the scan is moved in one direction, unlike the bidirectional which reverses direction at the end of the scan limit. The unidirectional scan is used when the radar must provide rapid and precise location updates on fast-moving targets. The unidirectional scan can be differentiated from the circular or sector scan by its high scan rate (usually greater than 10 Hz) and the absence of back lobes.

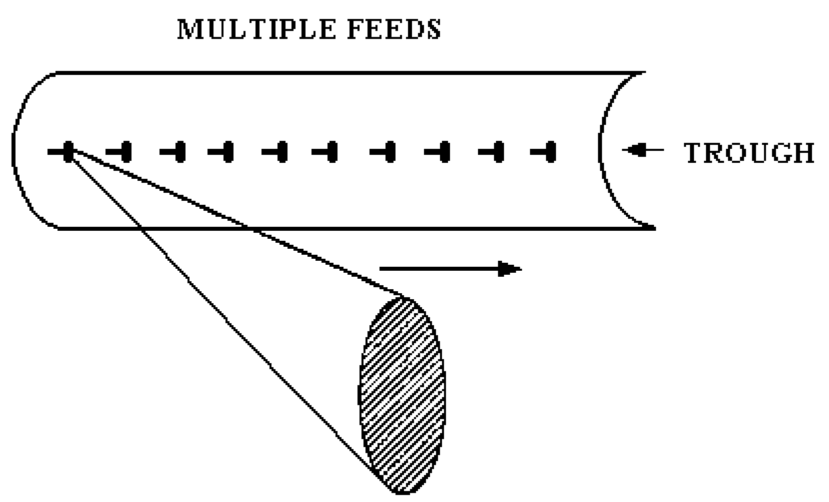


Figure 4 : Unidirectional sector scan

- **<u>- ORTHOGONAL SCAN</u>**

An orthogonal scan (often referred to as a track-while-scan) has two separate unidirectional or bidirectional scans, one in the vertical and the other in the horizontal plane. The beam swept horizontally is wide in the vertical dimension, and the beam swept vertically is wide in the horizontal dimension. An orthogonal scan using bidirectional scans may either interlace the scans (i.e., one horizontal sweep, one vertical sweep, the opposite horizontal sweep, followed by the opposite vertical sweep) or simply complete the vertical bidirectional scan and then the bidirectional sector scan (figure(5).

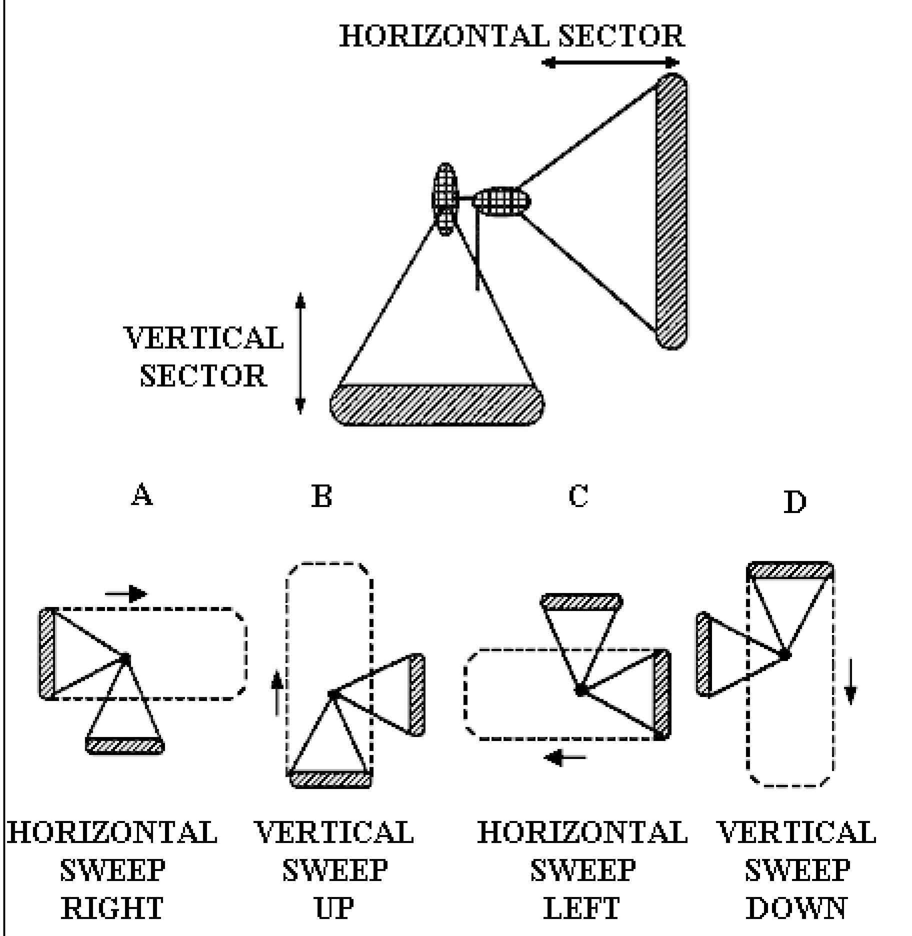


Figure 5: Orthogonal scan.

The horizontally scanned beam provides azimuth data, and the vertical scan provides the radar with elevation data. Targets are tracked in both azimuth and elevation without interruption of the search scan.

An orthogonal scan can track several targets at the same time and is most commonly associated with airport precision approach systems.

The orthogonal scan appears as two unidirectional or bidirectional scans. The scan should be measured between points on similar lobes of consecutive scans. An orthogonal scan normally has a scan period comparable to those of a sector (i.e., 1 to 5 seconds).

•- CONICAL SCAN

A conical scan rotates a feed horn about an axis within a parabolic dish in a circular motion at a rapid rate, usually from 10 to 100 Hz. See view A of figure (6). This type of scan is usually associated with target tracking and gun-laying radars, and is widely used for fire control. A conical scan can be superimposed on other scan types.

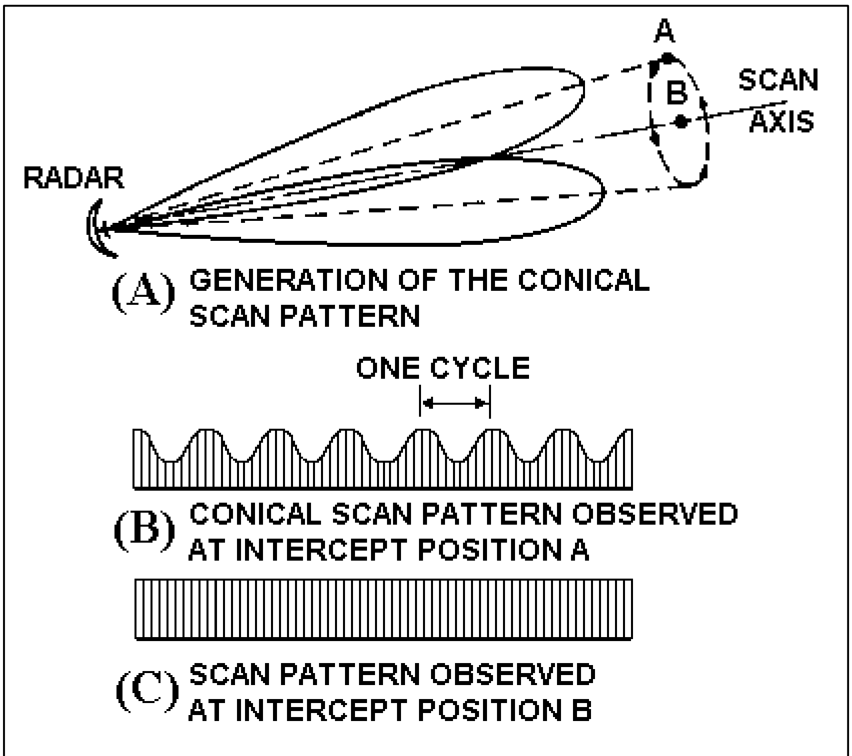


Figure 6 : Conical Scan

There are two basic mechanical methods used to generate a conical scan, nutated feed system and rotated feed system. Both of these methods require the feedhorn to rotate at an offset around the boresight of a parabolic antenna. Differences in the two methods are as follows:

- Nutated feed. Maintains a fixed plane of polarization as it rotates. The signal output consists of a beam with the polarization remaining constant.

- Rotated feed. Rotates the feed system at an offset from the cone axis. The signal output consists of a beam with the polarization continuously constantly.

- **- RASTER SCAN**

A raster scan covers a rectangular-shaped sector by scanning back and forth while changing level (angle of elevation) after each sweep. Raster scans are either horizontal raster or vertical raster. There are two primary types of scanning motions used in creating a raster scan, unidirectional and bidirectional. In both methods, each sweep of a sector by the radar beam is referred to as a bar. The unidirectional raster sweeps across a complete bar then returns to the beginning of the bar where it changes the scan elevation (or azimuth) and sweeps another bar. The bidirectional raster sweeps across a complete bar, changes the scan elevation (or azimuth), and sweeps another bar in the opposite direction. In both the unidirectional and bidirectional raster scans, a raster frame contains two or more bars. Six and eight bar raster frames are common. The pattern of the bars ensures total coverage of a desired area. At the end of a frame (raster pattern), the radar beam returns to the starting position and resumes the raster scan. This time between the end of the frame and the beginning of the next frame is called the fly-back time. The bars do not have to be scanned sequentially (i.e., highest to lowest), and one bar may be scanned several times prior to completion of a frame (7). The raster scan is used for intensive search of a small rectangular area. Raster scans are used by airborne intercept radars; airborne fire-control radars often use raster scans for search and acquisition.

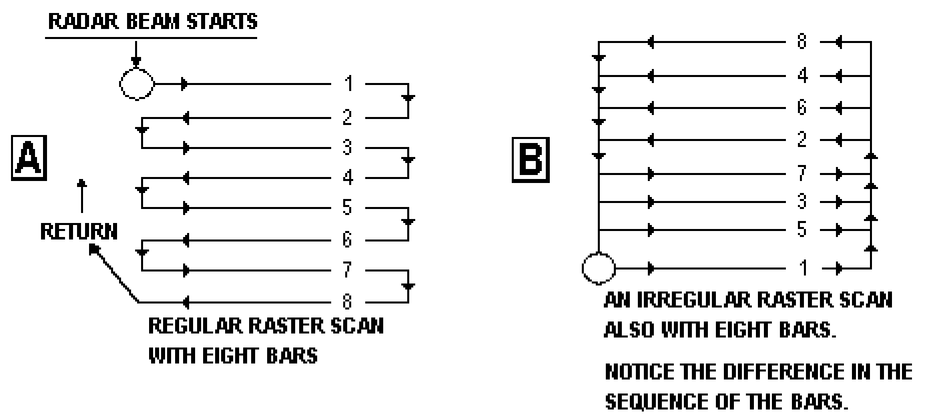


Figure 7 : Eight-bar raster scan.

Raster scans are classified in accordance with the number of bars and the number of levels (elevation or azimuth).

- **- PALMER SCAN**

A palmer scan is a conical scan superimposed on a circular, sector, or raster scan. The addition of the conical scan gives the radar more uniform coverage and can be used for tracking once acquisition is accomplished. A palmer scan will have the same amplitude characteristics of both scans. The most common type of palmer scan is the palmer raster. Several airborne fire-control systems use the palmer raster for performing search functions. See figure (8). In this system, there is a continuous conical scan; in addition, the antenna mount is rotated alternately in azimuth and in elevation.

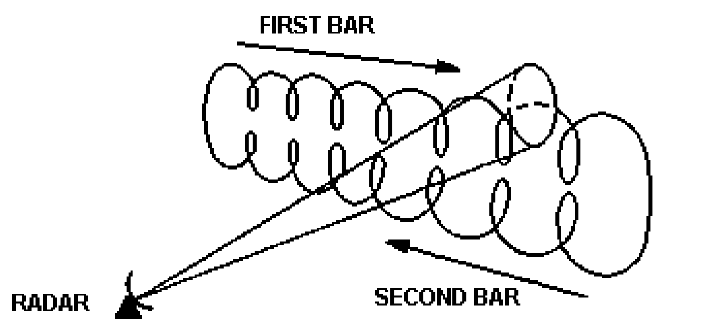


Figure 8 : Palmer Scan

The palmer scan measurement of scan rate involves two measurements: conical (Hz) and the type scan on which it is superimposed. The normal range for the conical portion of this combination scan is usually between 20 and 70 Hz.

- **- HELICAL SCAN**

A helical scan is a circular scan in which the elevation angle changes in steps. It is a combination of a circular scan and a vertical sector scan. It is used by land-based mobile units to search a large portion of the surrounding sky area with a narrow, fire-control radiation beam for aircraft targets. The antenna is continuously rotated in azimuth, while it is slowly raised or lowered in elevation. See figure (9). When the uppermost elevation is reached, the beam is immediately re-oriented to the original azimuth and elevation. The method used for the beam retrace determines the name of the type of helical scan. If the elevation angle increases to a peak then flies back to the starting point, the scan is a sawtooth helical. If the elevation goes up to its peak then back down in the same pattern, the scan is a sinusoidal helical. Usually a maximum of five or six different elevations can be used by a helical scan and still allow data update from each elevation angle within a reasonable amount of time.

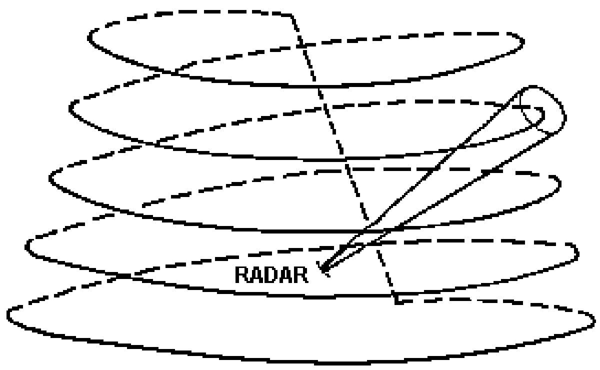


Figure 9 : Helical Scan

The helical scan normally has a scan period of 10 to 60 seconds with rotation periods of 2 to 20 seconds per 360° . Measuring scan rate involves two measurements, the circular scan rate in SPR and the framing rate in seconds.

- **<u>- SPIRAL SCAN</u>**

A spiral scan is a special case of conical scan. The axis of the radar beam varies its angle with respect to the antenna boresight axis in either a sawtooth manner or sinusoidal manner. After completing a full rotation, the angle between the beam and the fixed axis is increased. Starting from the center, the beam spirals outward until it reaches a maximum desired angle and then rotates back to the center. See figure (10). The spiral scan is used by older airborne radars for target acquisition and fire-control radars. When the target is acquired, the system switches to a conical scan. If the target is lost while in a conical scan, the system may automatically switch back to a spiral scan to reacquire the target.

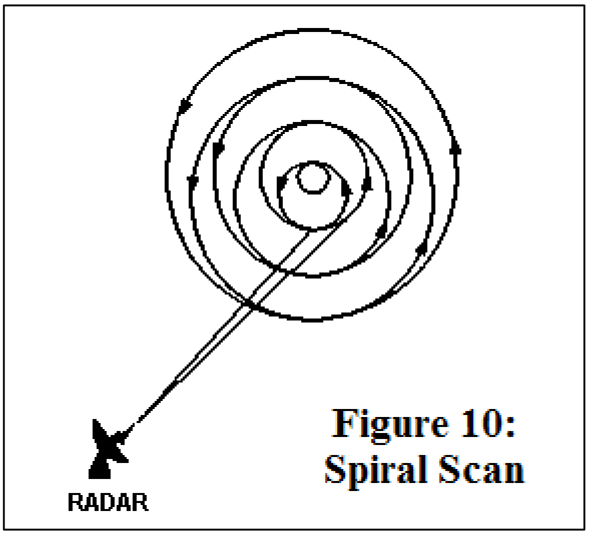


Figure 10: Spiral Scan

A spiral scan usually has a scan period of 1 to 5 seconds with a concurrent rotational rate of 20 to 60 Hz. The number of different amplitude levels observed can often be associated with a specific radar.

- ## - LOBE-SWITCHING SCAN

A lobe-switching scan uses two or more overlapping beams of equal amplitude to scan an area. See figure (11). The system switches between elements at a fast rate. Normal switching rates are 4 to 50 Hz. Target information is gained by comparing the amplitude of the target return at each antenna/element. This system keeps the radar centered on the target by comparing the strength of target returns from each beam. When four overlapping beams are used, the technique is called four-way lobe-switching.

Four-way lobe switching gives angular accuracy in both the horizontal and vertical planes at the same time. It is considered a precision radar and is usually used for such applications as fire-control and airborne intercept.

If the receiving site is in the exact center of either a two-way or four-way lobe-switching radar, there would be no amplitude change and the scan would appear as steady and the intercept operator would hear the PRF tone. If the receiving site is not located in the exact center, it will receive unequal energy from each of the lobes and the intercept operator will hear chatter in the earphones. The scan rate of lobe switching is measured the same as the conical scan rate one complete cycle through each of the lobes, whether two or four.

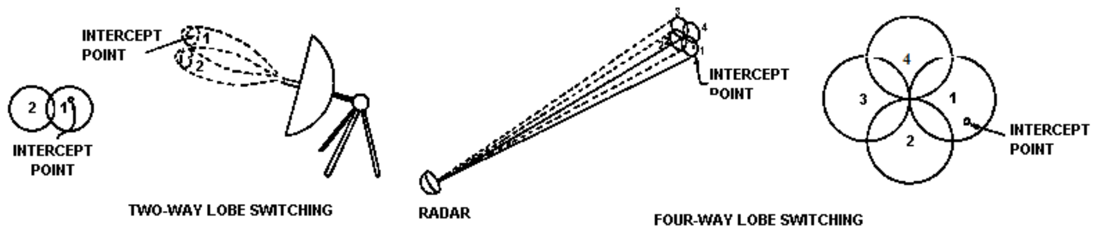


Figure 11: Lobe-switching scan.

SCAN-ON-RECEIVE-ONLY

The scan-on-receive-only (SORO) radar transmits a nonscanning beam. The receiver antenna or feeds are scanned or sampled in elevation and azimuth to perform target tracking and acquisition. The receiving antenna is doing the scanning, not the transmitting antenna. This process prevents a sampled scan pattern from being duplicated and then returned as a decoy by a hostile jammer. This type of scanning technique is virtually impossible to detect by an intercept operator since the scan appears to be steady.

There are four basic types of SORO: lobe-switching-on-receive-only (LORO), conical-scan-on-receive-only (COSRO), monopulse, and hybrid monopulse.

- **- ELECTRONIC SCANS**

In electronic scan systems, the beam direction is controlled electronically and is moved nearly instantaneously from one position to another. Antennas that rely on mechanical scans cannot provide rapid shifts in the beam position. With mechanical scans, there is a practical limit on how fast a system can scan and what the radiated scan pattern can be. There is no such limitation imposed on an electrical scan system since its beam position is controlled by electronic means. Electronic scan systems pose more serious problems for the analyst than the mechanical scan systems. The characteristics of electronic scans determine the information available for scan analysis.

An array antenna must be used to accomplish electronic scanning. Array antennas that perform electronic scanning search in one or more directions without physically moving the antenna. An array antenna is a large number of simple antennas, usually dipoles or horns, each serving as an array element and operating together under some form of mutual control. The elements in an array antenna are arranged in front of the reflector and are identical for easier beam steering. The shape of the array is one of the important factors in determining the azimuth and the elevation scan possibilities as well as the shape of the beam. The transmitted beam is formed by phase addition and subtraction and by amplitude addition and subtraction. By controlling the amplitude and phase of the signal fed to each element in the array, narrow beams are formed. Varying the phase, and possibly the amplitude, of the signal input to the array element causes the individual radiation patterns to coincide in phase in some desired direction, thus "steering" the beam. The antenna beam can be steered as fast as the phase and amplitude of the signal at each array element can be adjusted. Usually the primary method of control is by phase changes, with amplitude variations used to shape the side lobes. Thus, these antennas are called phase array antennas and the array configuration can be one of several geometric shapes. Some commonly used configurations are linear arrays, planar arrays, and spherical arrays. As shown in figure (12).

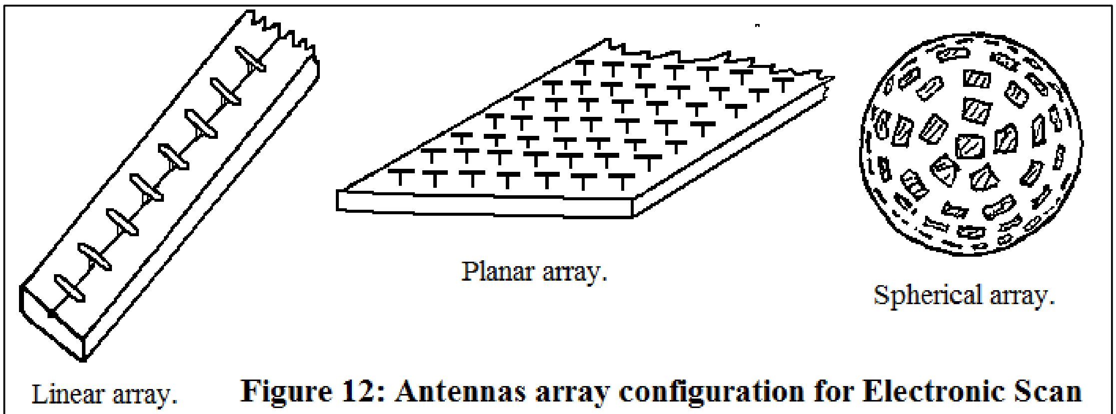


Figure 12: Antennas array configuration for Electronic Scan

Linear Array

The linear array, can be scanned in only one plane and the total scan coverage is somewhat less than 180° . Additional coverage can be achieved by rotating the entire array. The beamwidth of the main lobe depends on the number of elements used to form the array and the spacing between elements. Linear arrays usually produce a fan beam.

Planar Array

All elements in a planar array lie in one plane and scan less than 180° . Planar arrays allow scanning in azimuth and elevation. The beam can be a pencil beam, a fan beam, or a shaped beam. For more than 180° of coverage, the entire planar array must be rotated.

Spherical Array

Spherical array elements are placed on the surface of a sphere. This array allows complete 360° in azimuth and elevation coverage. The beam can be pencil, a fan, or some other shape. The spherical array requires more antenna elements for a given beamwidth and is the most complex to feed and control.

METHODS OF ELECTRONIC SCANNING

Electronic scanning through an array allows a radar to be programmed to perform any number of scan patterns. This versatility makes electronic scanning an ideal system for air search, early warning, target engagement, weapons control and airborne intercept. These radars may display the characteristics of a circular, raster, or other scan pattern, but since the scan is electronic it will not display the side/back lobes or the same aural signature of a mechanically generated scan.

Frequency Scanning

In frequency scanning (FRESCAN), the beam position is determined by the carrier frequency. For a given frequency, the beam will always be pointed in a given direction in space. Changing the frequency changes the relative phase at each of the elements and, therefore, the position of the beam. The scan can be accomplished only in one direction, either azimuth or elevation.

Phase Shift Scanning

Phase shift scanning is accomplished by electrically changing the phase of each phase shift network and thus the phase of the signal fed to each element. This method is slower than FRESCAN because of the time required to send the control signal to the required phase shifters. Usually, this type of system operates on one RF, although it is possible to have a system that changes in frequency. If the array antenna allows it, scan in azimuth and elevation is possible (e.g., a planar array would allow azimuth and elevation scanning but a linear array would not).

Time Delay Scanning

Time delay scanning is similar to phase shift scanning, but uses time delay networks to accomplish phase shift in the signal fed to the elements. It can scan in azimuth and elevation at the same time if the array antenna allows.

COMMUNICATION SYSTEMS

Radar Systems

Lecture (5-3) Pulse Radar

Coherent Pulse Radar

Continuous Wave Radar

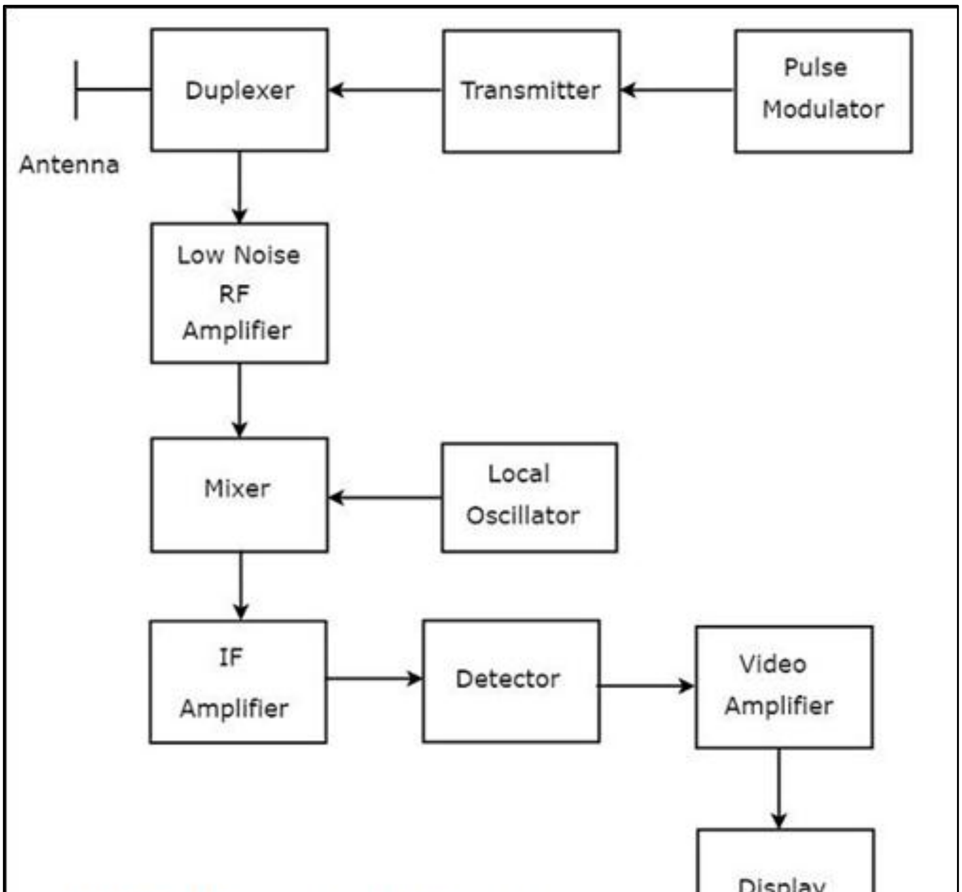
FM- CW Radar

Pulse Radar

The Radar, which operates with pulse signal for detecting stationary targets is called Basic Pulse Radar or simply, Pulse Radar.

Block Diagram for Pulse Radar.

Pulse Radar uses single Antenna for both transmitting and receiving of signals with the help of Duplexer. Following is the block diagram of Pulse Radar:



Block Digram for Pulse Radar

- **Pulse Modulator:** It produces a pulse-modulated signal and it is applied to the Transmitter.
- **Transmitter:** It transmits the pulse-modulated signal, which is a train of repetitive pulses.
- **Duplexer:** It is a microwave switch, which connects the Antenna to both transmitter section and receiver section alternately. Antenna transmits the pulse- modulated signal, when the duplexer connects the Antenna to the transmitter. Similarly, the signal, which is received by Antenna will be given to Low Noise RF Amplifier, when the duplexer connects the Antenna to Low Noise RF Amplifier. (The Duplexer will be illustrated with details in the next paragraph).
- **Low Noise RF Amplifier:** It amplifies the weak RF signal, which is received by Antenna. The output of this amplifier is connected to Mixer.
- **Local Oscillator:** It produces a signal having stable frequency. The output of Local Oscillator is connected to Mixer.
- **Mixer:** We know that Mixer can produce both sum and difference of the frequencies that are applied to it. Among which, the difference of the frequencies will be of Intermediate Frequency (IF) type.
- **IF Amplifier:** IF amplifier amplifies the Intermediate Frequency (IF) signal. The IF amplifier shown in the figure allows only the Intermediate Frequency, which is obtained from Mixer and amplifies it. It improves the Signal to Noise Ratio at output.
- **Detector:** It demodulates the signal, which is obtained at the output of the IF Amplifier.
- **Video Amplifier:** As the name suggests, it amplifies the video signal, which is obtained at the output of detector.
- **Display:** In general, it displays the amplified video signal on CRT screen.

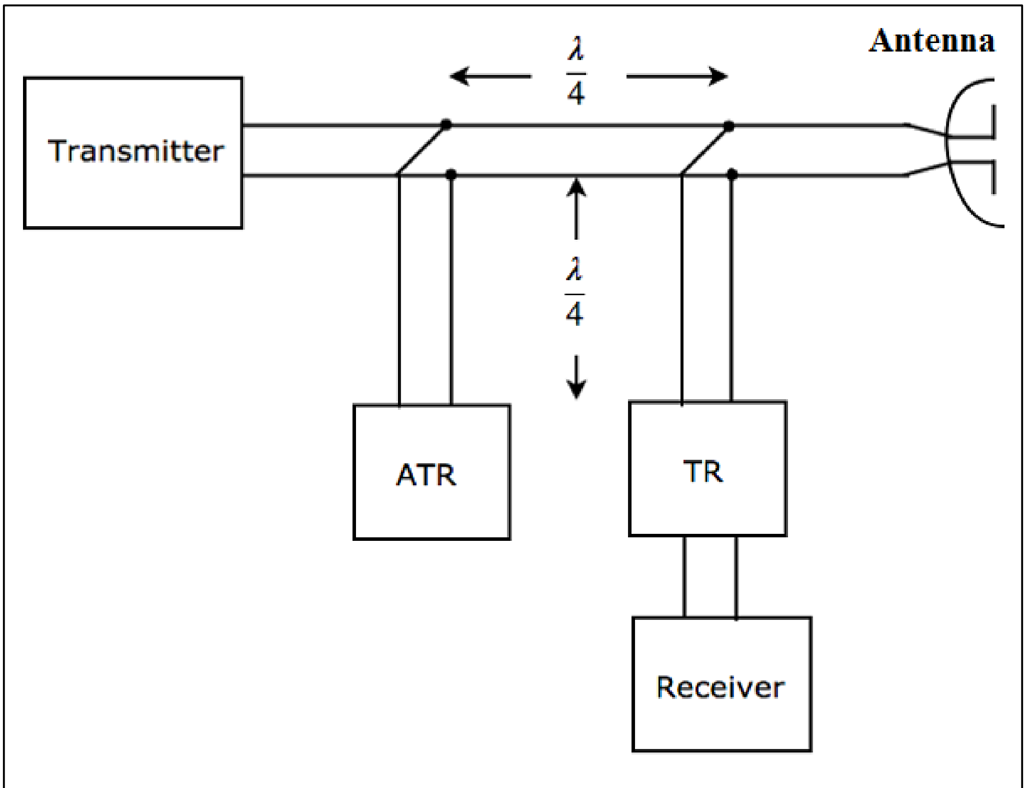
DUPLEXER:

In two-way communication, if we are supposed to use the same Antenna for both transmission and reception of the signals, then we require Duplexer. Duplexer is a microwave switch, which connects the Antenna to the transmitter section for transmission of the signal. Therefore, the Radar cannot receive the signal during transmission time.

Similarly, it connects the Antenna to the receiver section for the reception of the signal. The Radar cannot transmit the signal during reception time. In this way, Duplexer isolates both transmitter and receiver sections. Duplexers can be classified into the following three types.

- Branch-type Duplexer
- Balanced Duplexer
- Circulator as Duplexer

Branch-type Duplexer: Branch-type Duplexer consists of two switches Transmit-Receive (TR) switch and Anti Transmit-Receive (ATR) switch. The following figure shows the block diagram of Branch- type Duplexer:



Block Diagram of Branch - type Duplexer

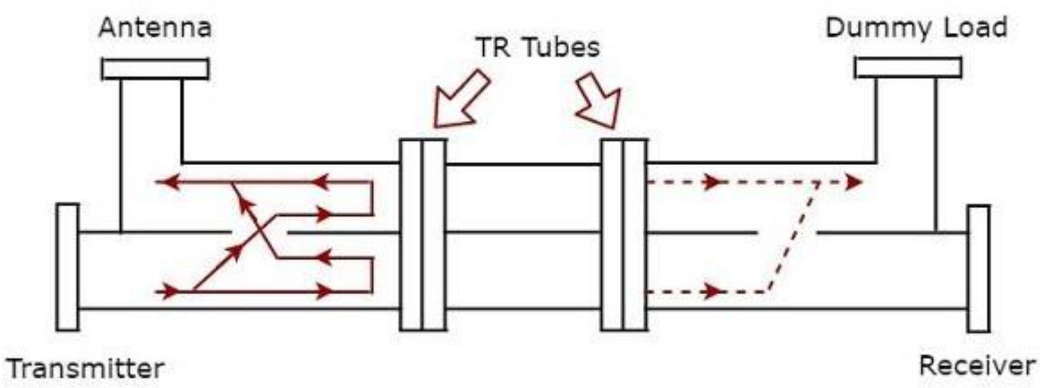
As shown in the figure, the two switches, TR & ATR are placed at a distance of $\lambda/4$ from the transmission line and both the switches are separated by a distance of $\lambda/4$. The working of Branch-type Duplexer is mentioned below.

- During transmission, both TR & ATR will look like an open circuit from the transmission line. Therefore, the Antenna will be connected to the transmitter through transmission line.
- During reception, ATR will look like a short circuit across the transmission line. Hence, Antenna will be connected to the receiver through transmission line. The Branch-type Duplexer is suitable only for low cost Radars, since it is having less power handling capability.

Balanced Duplexer

We know that a two-hole Directional Coupler is a 4-port waveguide junction consisting of a primary waveguide and a secondary waveguide. There are two small holes, which will be common to those two waveguides.

The Balanced Duplexer consists of two TR tubes. The configuration of Balanced Duplexer for transmission purpose is shown in the following figure.

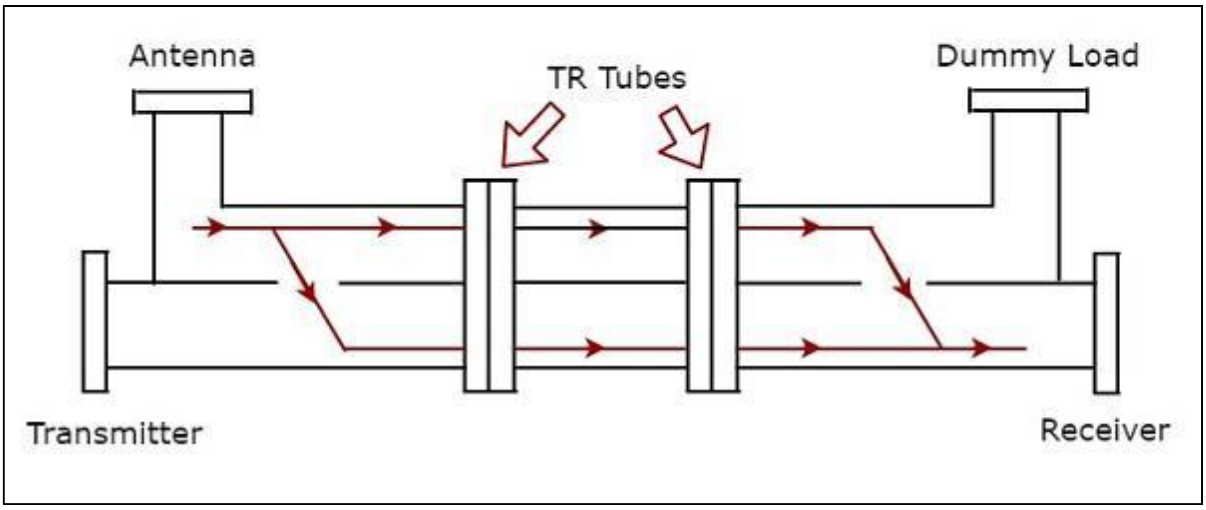


Block Diagram of Balanced Duplexer

The signal, which is produced by the transmitter has to reach the Antenna for the Antenna to transmit that signal during transmission time. The solid lines with arrow marks shown in the above figure represent how the signal reaches Antenna from transmitter.

The dotted lines with arrow marks shown in the above figure represent the signal which is leaked from the Dual TR tubes; this will reach only the matched load. So, no signal has been reached to the receiver.

The configuration of Balanced Duplexer for **reception** purpose is shown in figure given below.



We know that Antenna receives the signal during reception time. The signal which is received by the Antenna has to reach the receiver. The solid lines with arrow marks shown in the above figure represent how the signal is reaching the receiver from Antenna. In this case, Dual TR tubes pass the signal from the first section of waveguide to the next section of waveguide.

The Balanced Duplexer has high power handling capability and high bandwidth when compared to Branch-type Duplexer.

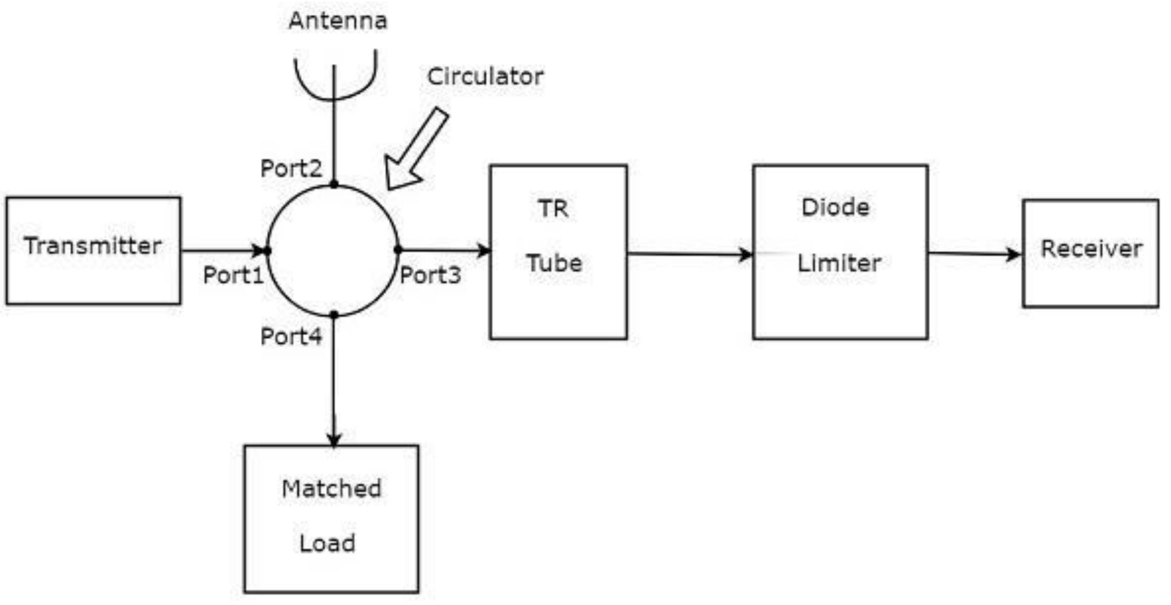
Circulator as a Duplexer

We know that the functionality of the circulator is that if we apply an input to a port, then it will be produced at the port, which is adjacent to it in the clockwise direction. There is no output at the remaining ports of the circulator. So, consider a 4-port circulator and connect the transmitter, Antenna, receiver and matched load to port1, port2, port3 and port4 respectively. Now, let us understand how the 4-port circulator works as Duplexer.

The signal, which is produced by the transmitter has to reach the Antenna for the Antenna will transmit that signal during transmission time. This purpose will be achieved when the transmitter generates a signal at port1.

The signal, which is received by the Antenna has to reach the receiver during reception time. This purpose will be achieved when the Antenna present at port2 receives an external signal.

The following figure shows the block diagram of circulator as Duplexer:



Circulator as a Duplexer

The above figure consists of a 4-port circulator - Transmitter, Antenna and the matched load is connected to port1, port2 and port4 of circulator respectively as discussed in the beginning of the section. The receiver is not directly connected to port3. Instead, the blocks corresponding to the passive TR limiter are placed between port3 of circulator and receiver. The blocks, TR tube & Diode limiter are the blocks corresponding to passive TR limiter.

Actually, the circulator itself acts as Duplexer. It does not require any additional blocks. However, it will not give any kind of protection to the receiver. Hence, the blocks corresponding to passive TR limiter are used in order to provide the protection to the receiver.

COHERANT PULSE RADAR:

A form of pulse radar in which the radio frequency oscillations in a recurrent pulse bear a constant pulse relation to those of a continuous oscillation called coherent radar. The pulse Doppler radar and moving target indicator radar are also are in this form. In this section the MTI Radars are illustrated as a sample from the coherent pulse radars.

In practical applications, Radar receives the echo signals due to stationary objects in addition to the echo signal due to that movable target.

The echo signals due to stationary objects (places) such as land and sea are called clutters because these are unwanted signals. Therefore, we have to choose the Radar in such a way that it considers only the echo signal due to movable target but not the clutters.

For this purpose, Radar uses the principle of Doppler Effect for distinguishing the non- stationary targets from stationary objects. This type of Radar is called Moving Target Indicator Radar or simply, MTI Radar.

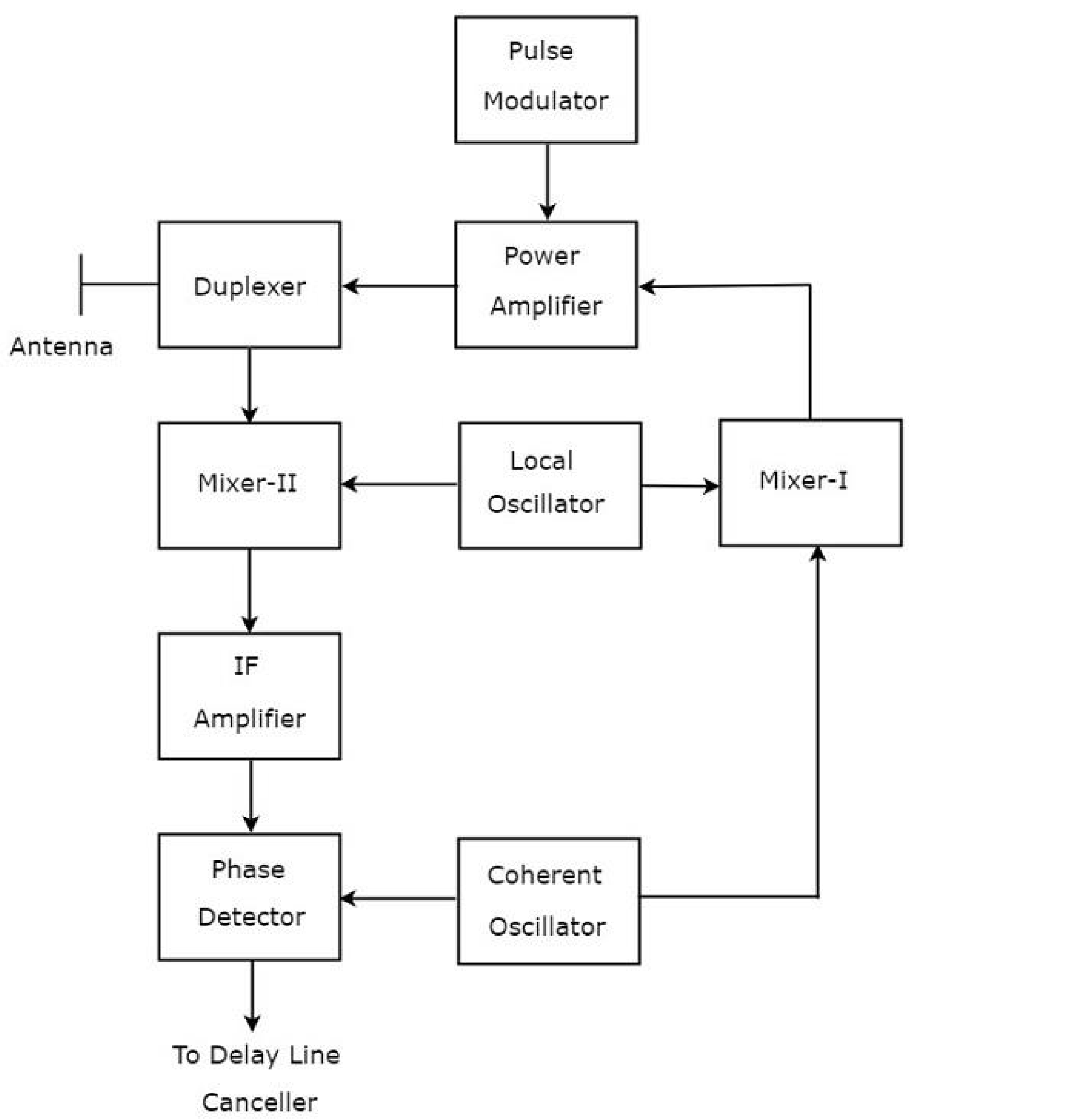
According to Doppler Effect, the frequency of the received signal will increase if the target is moving towards the direction of Radar. Similarly, the frequency of the received signal will decrease if the target is moving away from the Radar.

We can classify the MTI Radars into the following two types based on the type of transmitter that has been used.

- MTI Radar with Power Amplifier Transmitter
- MTI Radar with Power Oscillator Transmitter

MTI Radar with Power Amplifier Transmitter

MTI Radar uses single Antenna for both transmission and reception of signals with the help of Duplexer. The block diagram of MTI Radar with power amplifier transmitter is shown in the following figure.



Block Diagram of MTI Radar With Power Amplifier Transmitter

The function of each block of MTI Radar with power amplifier transmitter is mentioned below.

- Pulse Modulator: It produces a pulse modulated signal and it is applied to Power Amplifier.
- Power Amplifier: It amplifies the power levels of the pulse modulated signal.
- Local Oscillator: It produces a signal having stable frequency f_1. Hence, it is also called stable Local Oscillator. The output of Local Oscillator is applied to

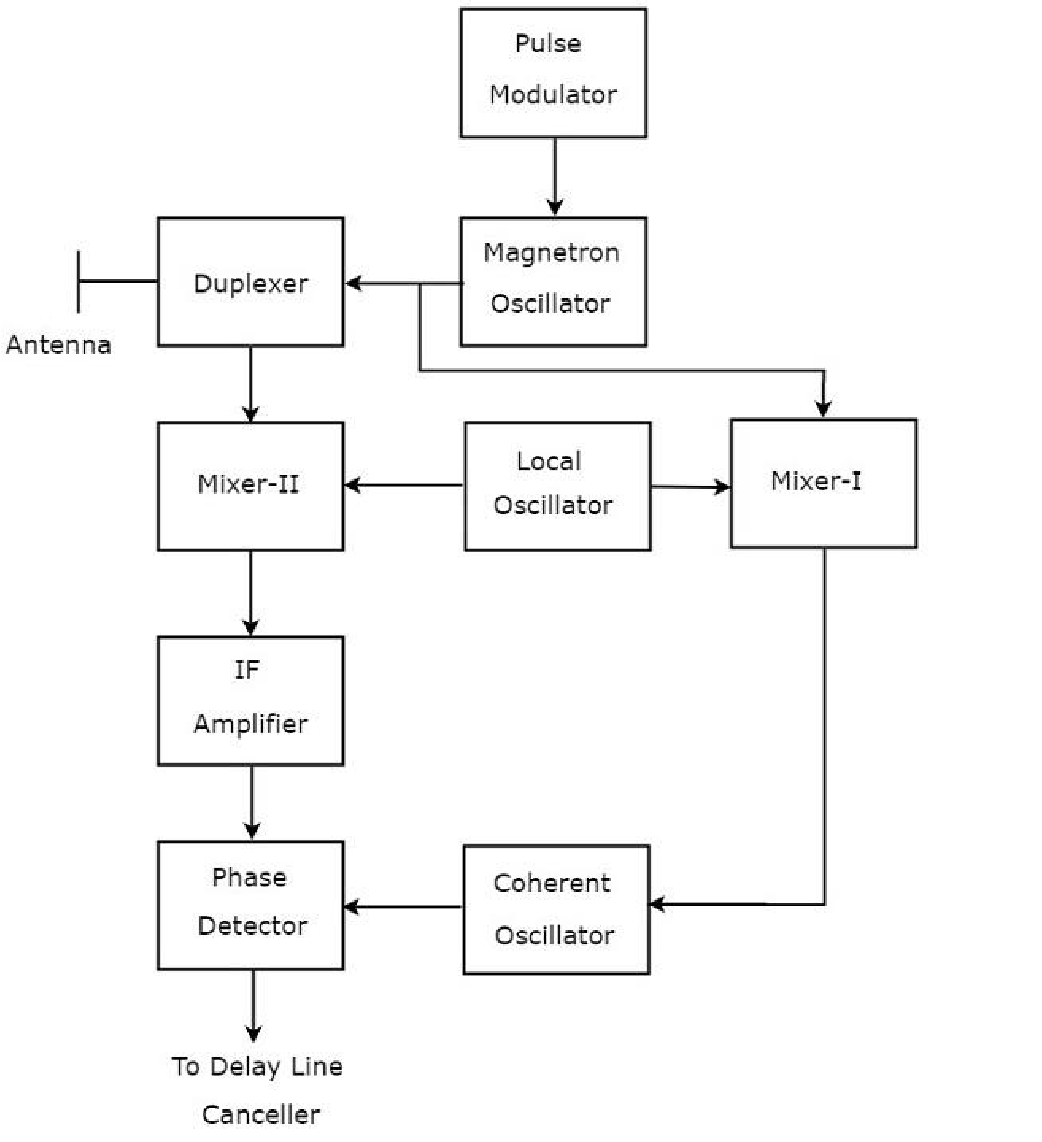
both Mixer-I and Mixer-II.

- Coherent oscillator: produces a signal having an Intermediate Frequency f_c. This signal is used as the reference signal. The output of Coherent oscillator is applied to both Mixer-I and Phase Detector.
- Mixer-I: Mixer can produce either sum or difference of the frequencies that are applied to it. The signals having frequencies of f_1 and f_c are applied to Mixer-I. Here, the Mixer-I is used for producing the output, which is having the frequency $f_1 + f_c$.
- Duplexer: It is a microwave switch, which connects the Antenna to either the transmitter section or the receiver section based on the requirement. Antenna transmits the signal having frequency $f_1 + f_c$ when the duplexer connects the Antenna to power amplifier. Similarly, Antenna receives the signal having frequency of $f_1 + f_c \pm f_d$ when the duplexer connects the Antenna to Mixer-II.
- Mixer-II: Mixer can produce either sum or difference of the frequencies that Are applied to it. The signals having frequencies $f_1 + f_c \pm f_d$ and f_1 are applied to Mixer-II. Here, the Mixer-II is used for producing the output, which is having the frequency $f_c \pm f_d$.
- IF Amplifier: IF amplifier amplifies the Intermediate Frequency (IF) signal. The IF amplifier shown in the figure amplifies the signal having frequency $f_c + f_d$. This amplified signal is applied as an input to Phase detector.
- Phase Detector: It is used to produce the output signal having frequency f_d from the applied two input signals, which are having the frequencies of $f_c + f_d$ and f_d . The output of phase detector can be connected to Delay line canceller.

MTI Radar with Power Oscillator Transmitter

The block diagram of MTI Radar with power oscillator transmitter looks similar to the block diagram of MTI Radar with power amplifier transmitter.

The blocks corresponding to the receiver section will be same in both the block diagrams. Whereas, the blocks corresponding to the transmitter section may differ in both the block diagrams. The block diagram of MTI Radar with power oscillator transmitter is shown in the following figure.



Block Diagram of MTI Radar with power oscillator transmitter

As shown in the figure, MTI Radar uses the single Antenna for both transmission and reception of signals with the help of Duplexer. The **operation** of MTI Radar with power oscillator transmitter is mentioned below.

- The output of Magnetron Oscillator and the output of Local Oscillator are applied to Mixer-I. This will further produce an **IF signal**, the phase of which is directly related to the phase of the transmitted signal.

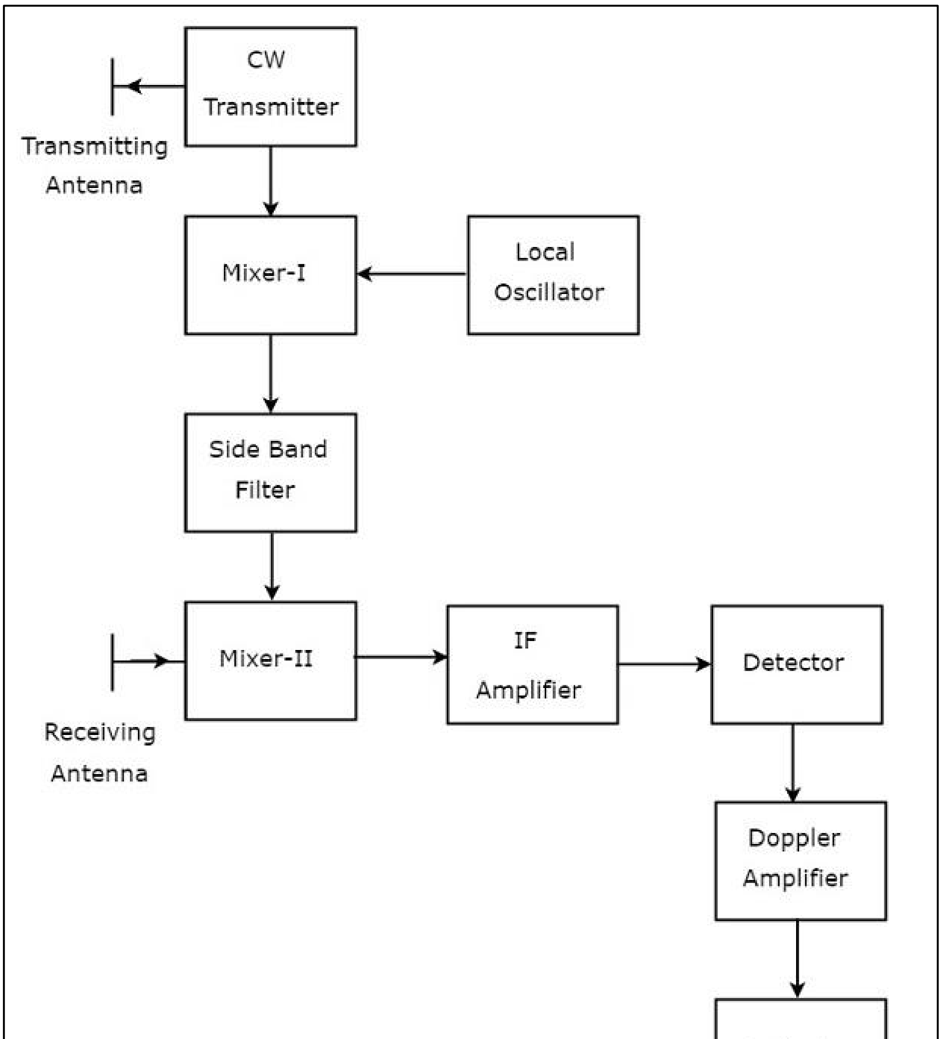
- The output of Mixer-I is applied to the Coherent Oscillator. Therefore, the phase of Coherent Oscillator output will be locked to the phase of IF signal. This means, the phase of Coherent Oscillator output will also directly relate to the phase of the transmitted signal.
- The output of Coherent Oscillator can be used as reference signal for comparing the received echo signal with the corresponding transmitted signal using phase detector.

Continuous Wave (CW) Radar

The Radar, which operates with continuous signal (wave) for detecting non-stationary targets, is called Continuous Wave Radar or simply CW Radar. This Radar requires two Antennas. Among which, one Antenna is used for transmitting the signal and the other Antenna is used for receiving the signal.

Block Diagram for CW Radar

We know that CW Doppler Radar contains two Antennas — transmitting Antenna and receiving Antenna. Following figure shows the block diagram of CW Radar:



Block Diagram for Continous Wave Radar

- CW **Transmitter:** It produces an analog signal having a frequency of f_0. The output of the CW Transmitter is connected to both transmitting antenna and Mixer – I
- Local **Oscillator:** It produces a signal having a frequency of f_1. The output of Local oscillator is connected to Mixer-I.
- **Mixer-I:** Mixer can produce both sum and difference of the frequencies that are applied to it. The signals having frequencies of f_0 and f_1 are applied to Mixer-I. So, the Mixer-I will produce the output having frequencies $f_0 + f_1$ or $f_0 - f_1$.
- **Side Band Filter:** As the name suggests, side band filter allows a particular

side band frequencies - either upper side band frequencies or lower side band frequencies. The side band filter shown in the above figure produces only upperside band frequency, i.e., f0 + f1

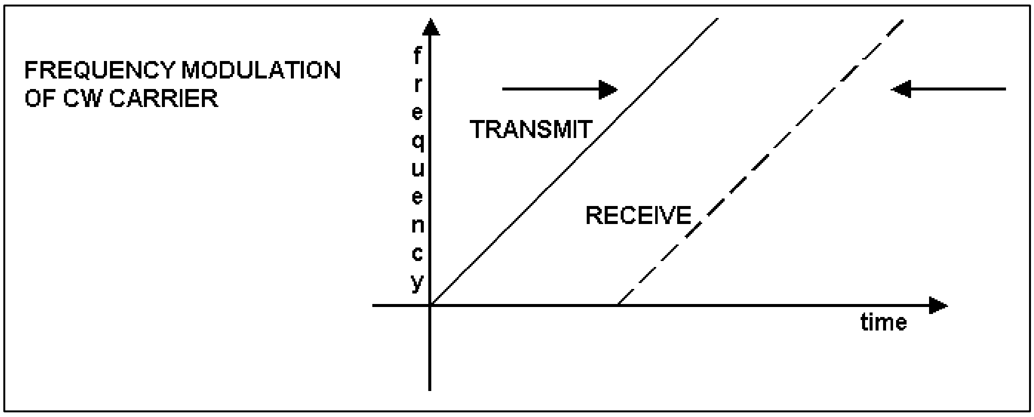
- **Mixer-II:** Mixer can produce both sum and difference of the frequencies that are applied to it. The signals having frequencies of $f_0 + f_1$ and $f_0 \pm f_d$ are applied to Mixer-II. So, the Mixer-II will produce the output having frequencies of $2 f_0 + f_1 \pm f_d$ or $f_1 \pm f_d$.
- **IF Amplifier:** IF amplifier amplifies the Intermediate Frequency (IF) signal. The IF amplifier shown in the figure allows only the Intermediate frequency, $f_1 \pm f_d$ and amplifies it.
- **Detector:** It detects the signal, which is having Doppler frequency, f_d.
- **Doppler Amplifier:** As the name suggests, Doppler amplifier amplifies the signal, which is having Doppler frequency, f_d .
- **Indicator:** It indicates the information related relative velocity and whether the target is inbound or outbound.

CW Doppler Radars give accurate measurement of **relative velocities**. Hence, these are used mostly, where the information of velocity is more important than the actual range.

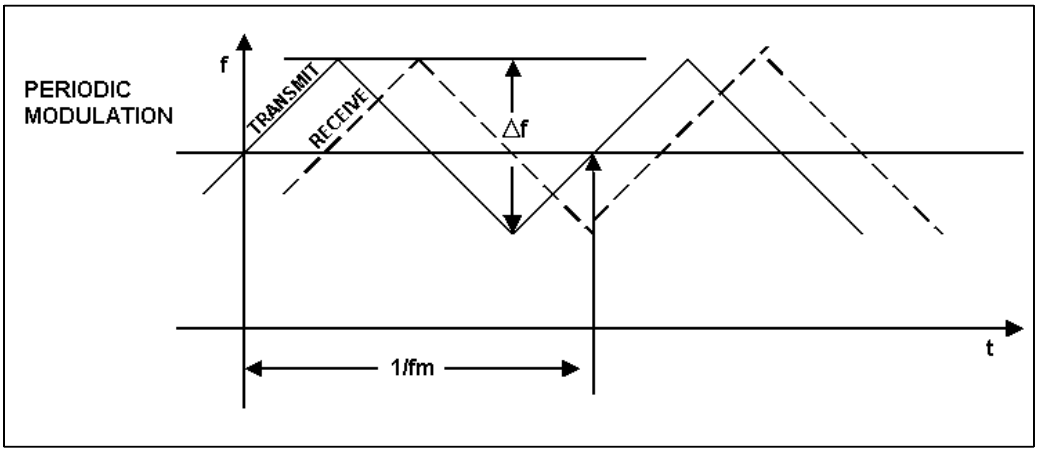
Frequency modulated Continuous Wave (FM- CW) Radar

If CW Doppler Radar uses the Frequency Modulation, then that Radar is called FMCW **Doppler Radar** or simply, **FMCW Radar**. It is also called Continuous Wave Frequency Modulated Radar or CWFM Radar. It measures not only the speed of the target but also the distance of the target from the Radar (Range).

The inability of the CW radar to measure range is related to the relatively narrow bandwidth of its transmitted waveform. Some sort of timing mark is required to measure the elapsed time between transmission and the return signal. A widely used technique to broaden the spectrum of the CW radar transmission is to frequency modulate the carrier. The timing mark is provided by the changes in the transmitted frequency. In an FM-CW radar, the transmitter frequency is changed as a function of time in a known manner as shown in figure below.



Since it is impossible to change frequency in one direction only, a periodic form of modulation is used. Triangular modulation, an example of periodic modulation, is shown in the figures below. Other forms of periodic modulation are sinusoidal and sawtooth.



In the receiver, the return signal is heterodyned with a portion of the transmitted signal producing a beat frequency. If there is no Doppler shift (target motion), the beat frequency is a measure of the target's range. After amplification, this beat frequency is fed to a frequency analyzer for range determination. If there is sufficient target motion, a Doppler frequency will be superimposed on the beat frequency and erroneous ranges will be measured. To determine the true beat frequency and thus the true range, the Doppler frequency must be taken into consideration.

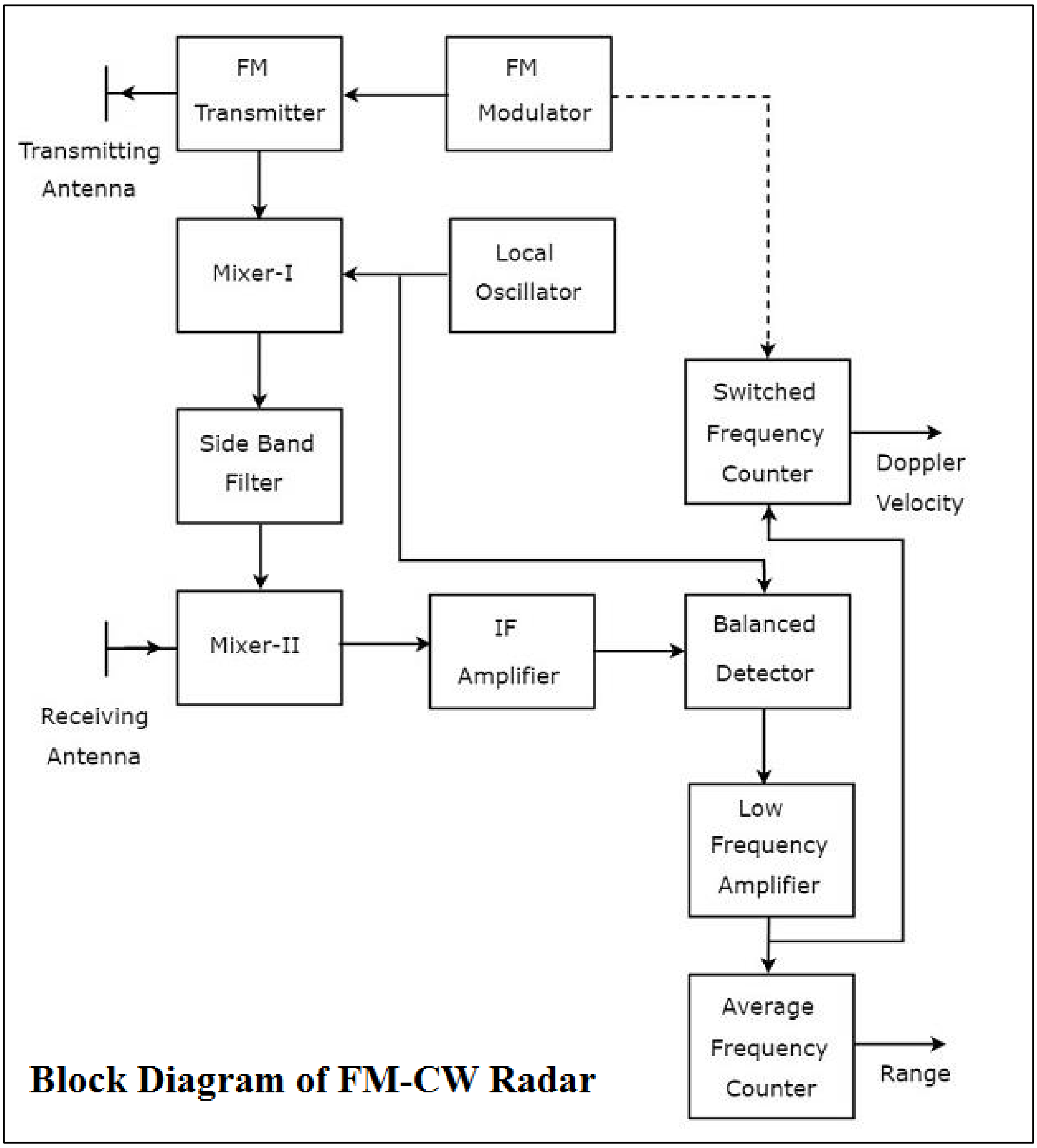
When more than one target is within view of the radar, the mixer output will contain more than one difference frequency. If the system is linear, there will be a frequency corresponding to each target. To determine individual target ranges, the individual beat frequencies must be separated from each other. This can be done with filters or other devices that complicate the radar. Since this adds significantly to the cost of the FM-CW radar, most do not have them incorporated. If the system is nonlinear, the problem of resolving targets and measuring the range of each becomes even more complicated. FM-CW radar is mainly used as an altimeter, an application for which there is only one target, the earth.

Block Diagram of FM-CW Radar

FMCW Radar is mostly used as Radar Altimeter in order to measure the exact height while landing the aircraft. The following figure shows the block diagram of FMCW Radar:

FMCW Radar contains two Antennas — transmitting Antenna and receiving Antenna as shown in the figure. The transmitting Antenna transmits the signal and the receiving Antenna receives the echo signal.

The block diagram of the FMCW Radar looks similar to the block diagram of CW Radar. It contains few modified blocks and some other blocks in addition to the blocks that are present in the block diagram of CW Radar.



Block Diagram of FM-CW Radar

- FM Modulator: It produces a Frequency Modulated (FM) signal having variable frequency, $f_0(t)$ and it is applied to the FM transmitter.
- FM Transmitter: It transmits the FM signal with the help of transmitting antenna. The output of FM Transmitter is also connected to Mixer-I.
- Local Oscillator: In general, Local Oscillator is used to produce an RF signal. But, here it is used to produce a signal having an Intermediate Frequency, f_{IF}. The output of Local Oscillator is connected to both Mixer-I and Balanced Detector.
- Mixer-I: Mixer can produce both sum and difference of the frequencies that

are applied to it. The signals having frequencies of $f_0(t)$ and f_{IF} are applied to Mixer-I. So, the Mixer-I will produce the output having frequency either $f_0(t) + f_{IF}$ or $f_0(t) - f_{IF}$.

- Side Band Filter: It allows only one side band fequencies, i.e., either upper side band frequencies or lower side band frequencies. The side band filter shown in the figure produces only lower side band frequency. i.e., $f_0(t) - f_{IF}$.
- Mixer-II: Mixer can produce both sum and difference of the requencies that are applied to it. The signals having frequencies of $f_0(t) - f_{IF}$ and $f_0(t\text{-}T)$ are applied to Mixer-II. So, the Mixer-II will produce the output having frequency either $f_0(t\text{-}T) + f_0(t) - f_{IF}$ or $f_0(t\text{-}T) - f_0(t) + f_{IF}$.
- IF Amplifier: IF amplifier amplifies the Intermediate Frequency (IF) signal. The IF amplifier shown in the figure amplifies the signal having frequency of $f_0(t\text{-}T) - f_0(t) + f_{IF}$. This amplified signal is applied as an input to the Balanced detector.
- Balanced Detector: It is used to produce the output signal having frequency of $f_0(t\text{-}T) - f_0(t)$ from the applied two input signals, which are having frequencies of $f_0(t\text{-}T) - f_0(t) + f_{IF}$ and f_{IF} . The output of Balanced detector is applied as an input to Low Frequency Amplifier.
- Low Frequency Amplifier: It amplifies the output of Balanced detector to the required level. The output of Low Frequency Amplifier is applied to both switched frequency counter and average frequency counter.
- Switched Frequency Counter: It is useful for getting the value of Doppler velocity.
- Average Frequency Counter: It is useful for getting the value of Range.