Mufasa Fiffi

Lecture Note

On

Wireless Communication

Contents

Evolution of Mobile Communication

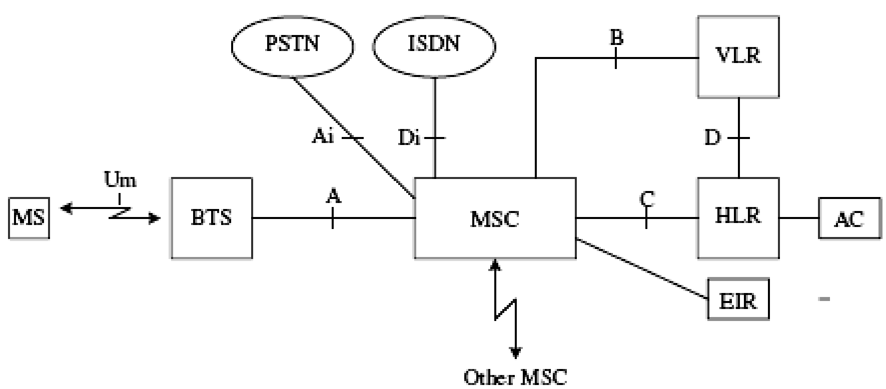
The first version of cellular telephony to be commercially deployed in the 1980s consisted of analog systems, where frequency modulation is used for analog voice and FSK for signaling and control data. The bandwidth of each channel allocated to an individual user is 30 kHz. These systems, which had no user data transport capability, were later followed by TDMA systems, where a channel is divided into a number of synchronized slots, each allocated to a single user. The TDMA systems installed in United States are based on standards IS-54 and IS-136, use a channel spacing of 30 kHz, and provide six slots per frame, eventually tripling the capacity compared to the older analog system. GSM, which is used in much of Europe and many other countries of the world, is also based on the TDMA technology, where each channel has a bandwidth of 200 kHz, and each frame consists of eight slots. A distinctive feature of these systems is their support of SMS and circuit-switched user data. An enhanced data service called GPRS is also now available in GSM. CDMA systems, which use direct sequence spread spectrum technology, has been deployed in many country since 1995. Standards for 3G wireless services were published in 1999. Support for high-speed data at rates from 144 kb/s for urban and suburban outdoor environments to 2,048 Mb/s for indoor or low-range outdoor environments is one of the most important features of 3G. Because of the many advantages that it offers, the CDMA technology forms the basis of 3G systems.

<u>Chronology of important developments in mobile communications</u>

1946 First domestic public land mobile service introduced in St. Louis. The system operated at 150 MHz and had only three channels.
1956 First use of a 450 MHz system. Users had to use a push-to-talk button and always needed operator assistance.
1970 FCC sets aside 75 MHz for high-capacity mobile telecommunication systems.
1974 FCC grants common carriers 40 MHz for development of cellular systems.
1978 First cellular system called AMPS was introduced in Chicago on a trial basis.
1981 Cellular systems deployed in Europe.
1983 First commercial deployment of cellular system in Chicago. It is an analog system and does not have a user data transport capability. Analog systems around 450 and 900 MHz band were also introduced in many countries of Europe during 1981—90.
1989 FCC grants another 10 MHz bandwidth for cellular systems, thus giving a total of 50 MHz.
1991 GSM introduced in Europe and other countries of the world.
1993 TDMA system called IS-54 introduced in the United States. SMS available in GSM.
1995 CDMA cellular and PCS technology introduced in the United States.
1997 ETSI publishes GPRS standard.
1999 Standards for 3G wireless services published.

First-Generation Network

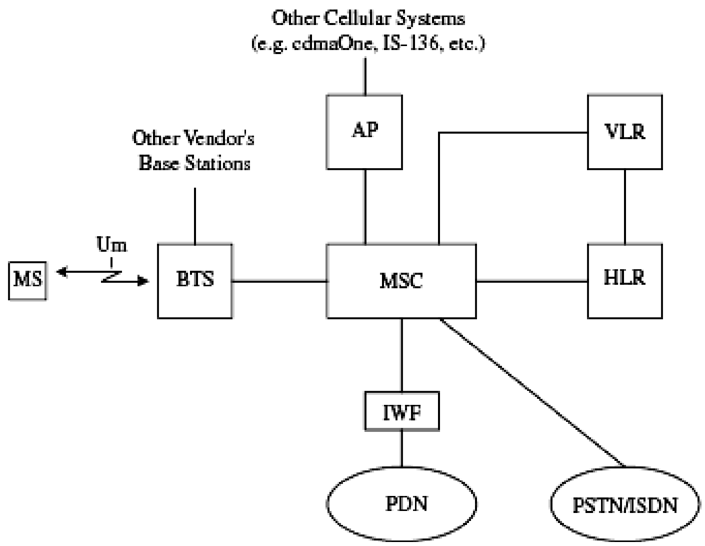
Network reference model of the *Telecommunications Industry Association/Electronics Industry Association* (TIA/EIA) standard IS-41, is shown in Figure below. This also represents the network for the first generation systems that support only voice and no data. This reference model is similar to the GSM architecture. The *mobile switching center* (MSC) performs mobile switching functions and interfaces the cellular network to a PSTN, *Integrated Services Digital Network* (ISDN), or another MSC. *Home Location Register* (HLR) contains a centralized database of all subscribers to the home system. This database includes such information as the *electronic serial number* (ESN), *directory number* (DN), the service profile subscribed by this user (such as roaming restriction, if any, supplementary services that this mobile has subscribed to, and so on), and its current location. Similarly, *Visitor Location Register* (VLR) contains a database of all visitors to this particular system. Whenever a mobile station moves into a Foreign Service area, its MSC saves all the pertinent information of that mobile station in its VLR. The home MSC is also notified so that incoming calls to this mobile can be forwarded to the foreign MSC. The information in the VLR is really the same as that of the HLR. However, when the mobile moves out of this foreign serving area, its MSC removes the database of this visitor from its VLR. The *equipment identity register* (EIR) contains the equipment identification number. The *authentication center* (AC) manages user data-encryption-related functions such as ciphering keys, and so on.



The reference model of a mobile communication network

Second-Generation Networks

An important feature of the *second-generation* (2G) systems is their data service capability. For example, IS-95 supports circuit-switched data and digital fax, IP, mobile IP, and *cellular digital packet data* (CDPD). GSM provides the short messaging service and circuit switched data at rates up to 9.6 kb/s per slot. Figure below shows network architecture that supports these data services as well as voice. Notice that it is very similar to architecture of first generation except for its interface to a *public data network* (PDN). This interface to the PDN is via an interworking function labeled IWF, which actually performs some protocol conversion that might be necessary because of the differences in the protocols used on the mobile stations and the PDN.



2G wireless network with packet data services

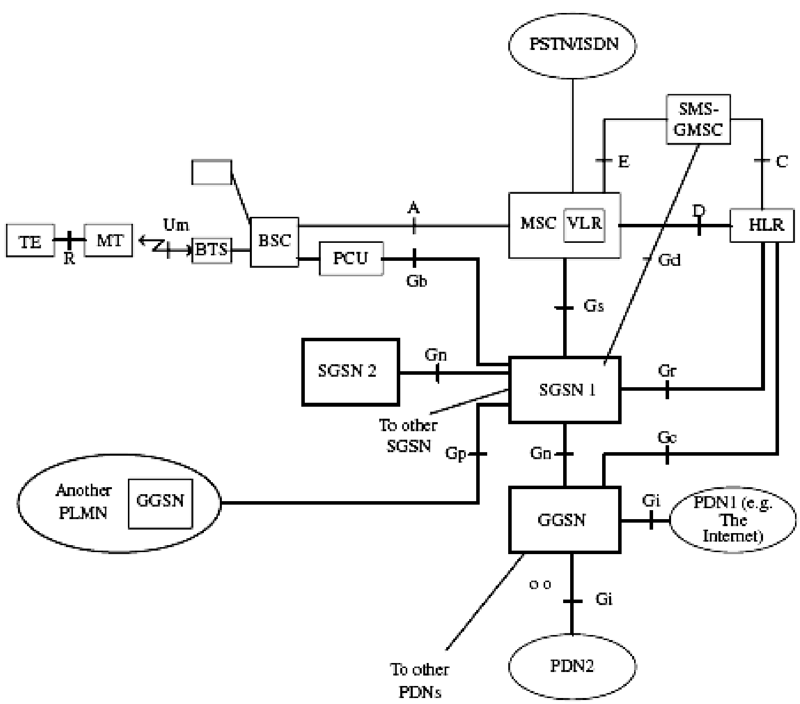
2G+ Networks

In the Figure above, except for the A interface between a BS and an MSC, the core network is circuit-switched. Equipment from many different manufacturers is now available in the market that can support packet mode data in a core network. One possible architecture around which many new networks are being built is shown in Figure below. The salient features of this architecture are the following:

- First, it consists of a backbone network that is based on IP/ATM.
- Second, it interfaces to legacy networks in a rather straightforward way. For example, the media gateway performs the necessary protocol conversion between the backhaul ATM network and the circuit-switched PSTN or ISDN. The IP routers are used to route packets to or from IP-based packet data networks. The mobility server, which is based on IP, supports mobility management, connection control, and signaling gateway functions to help provide seamless roaming capability across different networks with centralized directory management and, if needed, end-to-end security. As such, the functional entities

of the mobility server would include, among other things, call control, HLR and VLR databases, and radio resources management.

- Third, it allows for distributed processing, thus offloading the core network, and provides a platform where new services, features, and applications can be developed, tested, and installed in the network when necessary. Finally, the architecture is compatible with an all-IP network that appears to be the trend of the future.
- GPRS, which has already been introduced in the 2G+ version of GSM, supports packet mode data at rates up to 171 kb/s. The core network consists of a number of *serving GPRS support nodes* (SGSNs), a *gateway GPRS support node* (GGSN), and a *packet control unit* (PCU). The SGSN, which is actually a router, connects to a BSC via a PCU, which implements the link layer protocol. There may be more than one serving GSN in any *public land mobile network* (PLMN) as shown. Two separate PLMNs are connected through a GGSN. The GGSN is also a router and is the first entry point of the core network from any external packet data network (such as the Internet). The *short messaging service gateway MSC* (SMS-GMSC) provides the necessary protocol conversion for handling SMS through the GPRS network (instead of the traditional GSM network).



Third-Generation (3G) Wireless Technology

As mentioned earlier, the first-generation mobile telecommunication systems to be introduced in the 1980s were analog. These systems, which are still in service, do not have any user data transport capability. To provide data services in these analog systems, a new platform— say, *Cellular Digital Packet Data* (CDPD)—has to be overlaid on the cellular system. However, even this arrangement supports only slow-speed data. The second-generation systems—IS-136, cdmaOne, and GSM—are digital and have data transport capabilities but only to a limited extent. For example, GSM supports SMSs
and user data at rates only up to 9.6 kb/s. With IS-95B, it is possible to provide data rates in the range of 64 to 115 kb/s in increments of 8 kb/s over a 1.25 MHz channel. In 1997, to provide for packet mode data services in GSM, ETSI defined a new standard called *General Packet Radio Service* (GPRS), whereby a single time slot may be shared by multiple users for transferring packet mode data. In GPRS, each slot can handle up to 21.4 kb/s. Because each user may be allocated up to 8 slots, data rates up to about 171.2 kb/s per user are possible.
To support high-speed data rates and, more importantly, to be able to provide for multimedia services, the *International Telecommunications Union-Radio Communication Sector* (ITU-R) undertook the task of defining a set of recommendations for *International Mobile Telecommunication in the year 2000* (IMT-2000). Thus, eventually, there were only 4 systems for 3G mobile communications: cdma2000, UWC -136, WCDMA UMTS FDD, and WCDMA UMTS TDD. cdma 2000 is required to comply with EIA/TIA IS-41 and WCDMA UMTS with GSM MAP intersystem networking standards.

3G Requirements

3G systems are required to operate in many different radio environments, such as indoor or outdoor, urban, suburban, or rural. The end users may be fixed or moving at various speeds. For example, services
may involve:

- Stationary users or pedestrian (0 to 10 km/h)
- Ordinary vehicular applications up to 100 km/h
- High-speed vehicular applications up to 500 km/h
- Aeronautical applications up to 1500 km/h
- Satellites up to 27000 km/h.

The infrastructure used to deliver 3G services may be either terrestrial or satellite based. The information types may include speech, audio, data, text, image, and video. Radio interfaces must be designed to provide voice band data and variable bit rate services to end users. Both circuit and packet mode data must be supported. The data rates may be

- 144 kb/s or more in vehicular operations
- At least 384 kb/s for pedestrians
- About 2.048 Mb/s for indoor or low-range outdoor applications

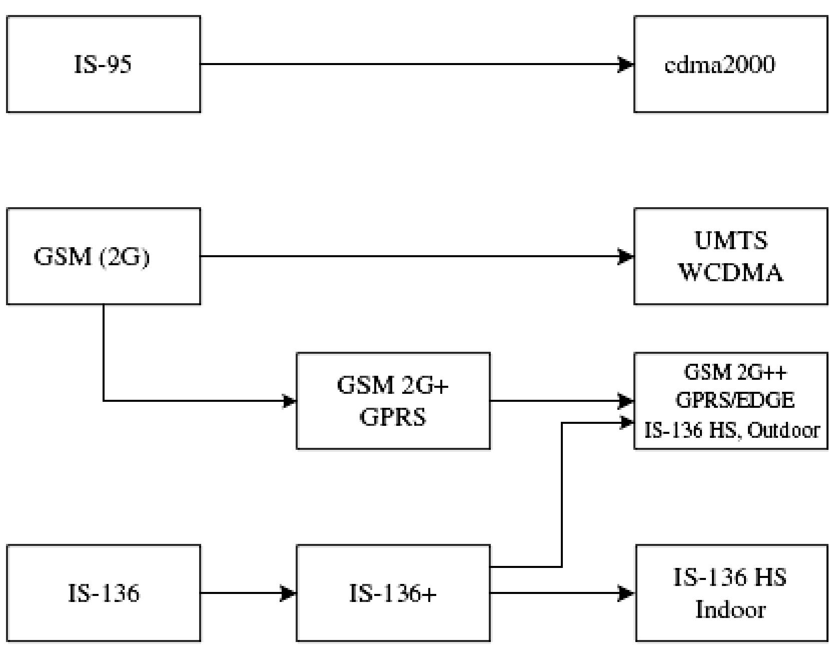
The 3G standards envisage different types of user traffic:

1. Constant bit rate traffic, such as speech, high-quality audio, video telephony, full-motion video, and so on, which are sensitive to delays and, more importantly, delay variations.
2. Real-time variable bit rate traffic, such as variable bit-rate encoded audio, interactive MPEG video, and so on. This type of traffic requires variable bandwidths and is also sensitive to delays and delay variations.
3. Non-real-time variable bit rate traffic, such as interactive and large file transfers, that can tolerate delays or delay variations.

Some possible applications that appear commercially attractive are

- Conversational voice, video phone and video conferencing, interactive games, and two way process control, and telemetry information.
- High-speed Internet access applications, such as web browsing, e-mail, data transfer to or from a server (such as a database download for later analysis), transaction services (that is, e-commerce), and so on.
- Audio streaming, one-way video, still images, large-volume data transfers, and tele metering information for monitoring purposes at an operations and maintenance center.
- Entertainment-quality audio.
- Inquiries/reservation (such as, plane ticket ordering and so on.

The evolution path to 3G systems

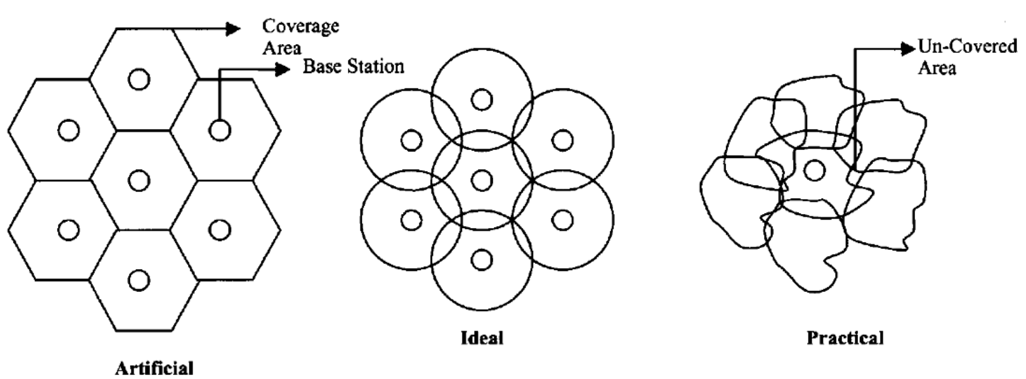


Cellular Concept

Cellular telephone systems must accommodate a large number of users over a large geographic area with limited frequency spectrum, i.e., with limited number of channels. If a single transmitter/ receiver are used with only a single base station, then sufficient amount of power may not be present at a huge distance from the BS. For a large geographic coverage area, a high powered transmitter therefore has to be used. But a high power radio transmitter causes harm to environment. Mobile communication thus calls for replacing the high power transmitters by low power transmitters by dividing the coverage area into small segments, called cells. Each cell uses a certain number of the available channels and a group of adjacent cells together use all the available channels. Such a group is called a cluster. This cluster can repeat itself and hence the same set of channels can be used again and again. Each cell has a low power transmitter with a coverage area equal to the area of the cell. This technique of substituting a single high powered transmitter by several low powered transmitters to support many users is the backbone of the cellular concept.

What is a Cell?

Power of radio signals transmitted from the BS decays as the signals travel away from it. A minimum amount of signal strength (let us say, x dB) is needed in order to be detected by the MS or mobile sets which may be the hand-held personal units or those installed in the vehicles. The region over which the signal strength lies above this threshold value x dB is known as the coverage area of a BS and it must be a somewhat circular region, considering the BS to be isotropic radiator. Such a circle, which gives this actual radio coverage, is called the foot print of a cell (in reality, it is amorphous). It might so happen that either there may be an overlap between any two such side by side circles or there might be a gap between the coverage areas of two adjacent circles. Such a circular geometry, therefore, cannot serve as a regular shape to describe cells. We need a regular shape for cellular design over a territory which can be served by 3 regular polygons, namely, equilateral triangle, square and regular hexagon, which can cover the entire area without any overlap and gaps. Along with its regularity, a cell must be designed such that it is most reliable too, i.e., it supports even the weakest mobile witch occurs at the edges of the cell. For any distance between the center and the farthest point in the cell from it, a regular hexagon covers the maximum area. Hence regular hexagonal geometry is used as the cells in mobile communication.



- Solves the problem of spectral congestion and user capacity.
  - Offer very high capacity in a limited spectrum without major
- Technological changes.
  - Reuse of radio channel in different cells.
  - Enable a fix number of channels to serve an arbitrarily large number of users by reusing the channel throughout the coverage region.

What is Cluster

A group of cells forms cluster when the entire available spectrum is divided equally among the cells. Cells in this group have a disjoint set of frequencies.

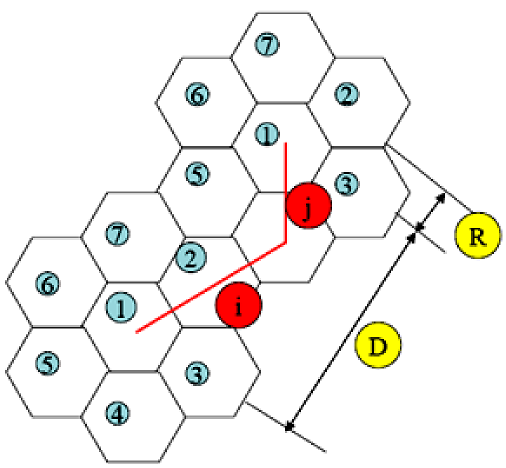
The number of cells in a cluster must be determined so that the cluster can be repeated continuously within the covering area of a service provider. The typical clusters contain 4,7,12 or 21 cells. The smaller the number of cells per cluster is, the bigger the number of channels per cell will be. The capacity of each cell will therefore be increased. However, a balance must be found in order to avoid the interference that could occur between neighbouring clusters.

Only certain cluster sizes and cell layout are possible.

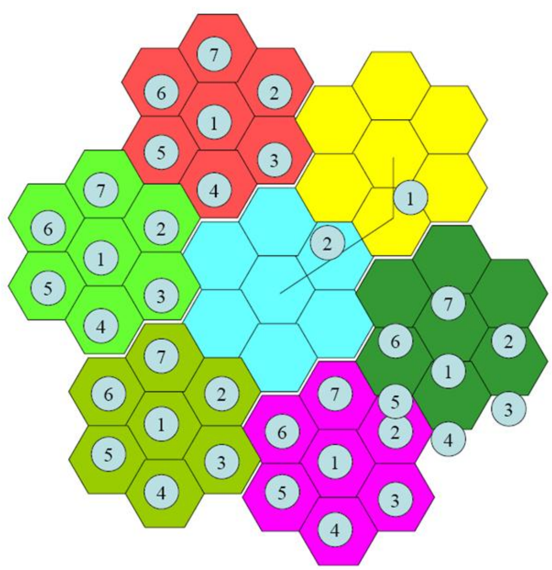
• The number of cells per cluster, N, can only have values which satisfy

$$N = i^2 + ij + j^2$$

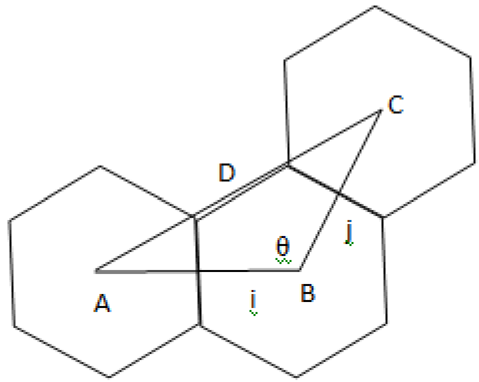
Where, i and j are non-negative integers.



The Cell Structure for N = 7



Derivation of cluster size (N)



Assumptions:

In the diagram above 3 regular hexagonal cells are taken out of which cell A and C is co-channel cells.

Applying cosine law of triangle,

$$D^2 = i^2 - 2ij\cos\theta + j^2$$

As per cluster design rule, θ = 120^0

Thus,

$$D^2 = i^2 + ij + j^2$$

Here, i, is a direction in which there can be a number of cells and j is a direction 60^0 anticlockwise from i. Along j there can be a number of cell. i and j are positive integers.

If, R_p is perpendicular distance from center of a cell to its one of the side, from cell geometry,

$$R_p = \frac{\sqrt{3}}{2}R$$

Then, center to center distance of two cells is,

$$2R_p = \sqrt{3}R$$

If we generalize, we can re-write above equation as,

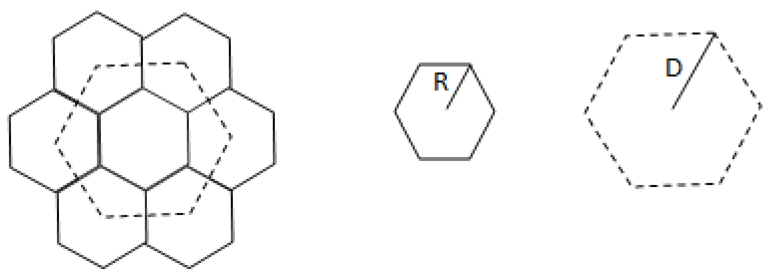
$$D^2 = \left(i \times 2R_p\right)^2 + \left(i \times 2R_p\right)\left(j \times 2R_p\right) + \left(j \times 2R_p\right)^2$$

$$D^2 = \left(2R_p\right)^2 (i^2 + ij + j^2\ )$$

$$D^2 = \left(2 \times \frac{\sqrt{3}}{2} R\right)^2 \ (i^2 + ij + j^2\ )$$

$$D^2 = 3R^2 (i^2 + ij + j^2\ )$$

Now considering area of hexagon,



Area of hexagon with radius R,

$$A_R = \frac{3\sqrt{2}}{2}\ R^2$$

Area of hexagon with radius D,

$$A_D = \frac{3\sqrt{2}}{2}\ D^2$$

Taking the ratio of two expression of area we get,

$$\frac{A_D}{A_R} = \frac{D^2}{R^2}$$

Also,

$$\frac{A_D}{A_R} = number\ of\ cells\ enclosed\ inside\ the\ large\ hexagon$$

From geometry of cells of large and small gexagon above,

$$N + 6\left(\frac{1}{3}N\right) = \frac{D^2}{R^2}$$

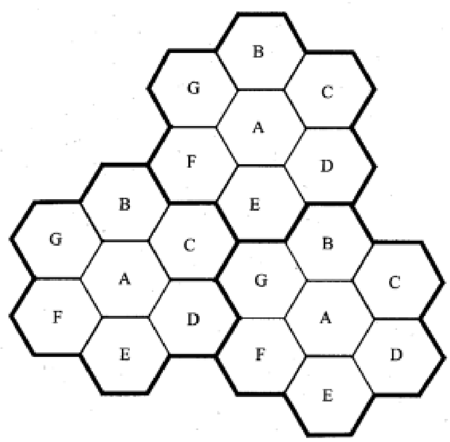
$$3N = \frac{3R^2(i^2 + ij + j^2)}{R^2}$$

Hence we have,

$$N = i^2 + ij + j^2$$

Frequency Reuse

Frequency reuse, or, frequency planning, is a technique of reusing frequencies and channels within a communication system to improve capacity and spectral efficiency. Frequency reuse is one of the fundamental concepts on which commercial wireless systems are based that involve the partitioning of an RF radiating area into cells. The increased capacity in a commercial wireless network, compared with a network with a single transmitter, comes from the fact that the same radio frequency can be reused in a different area for a completely different transmission. Frequency reuse in mobile cellular systems means that frequencies allocated to the service are reused in a regular pattern of cells, each covered by one base station. The repeating regular pattern of cells is called cluster. Since each cell is designed to use radio frequencies only within its boundaries, the same frequencies can be reused in other cells not far away without interference, in another cluster. Such cells are called `co-channel' cells. The reuse of frequencies enables a cellular system to handle a huge number of calls with a limited number of channels.



Consider a cellular system which has a total of *S* duplex channels. $k < S$

• Each cell is allocated a group of *k* channels,
• The *S* channels are divided among *N* cells.
• The total number of available radio channels

$$S = kN$$

• The *N* cells which use the complete set of channels is called *cluster*.

• The cluster can be repeated M times within the system. The total number of channels, C, is used as a measure of capacity

$$C = MkN = MS$$

• The capacity is directly proportional to the number of replication M.
• The cluster size, N, is typically equal to 4, 7, or 12.
• Small N is desirable to maximize capacity.
• The frequency reuse factor is given by $1/N$

Channel Assignment Strategies

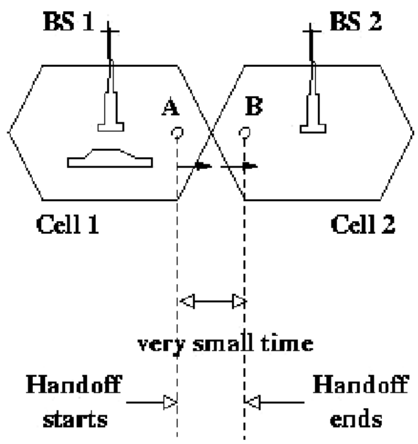
Channel assignment strategies are classified into two types: fixed and dynamic, as given below.

Fixed Channel Assignment (FCA)
In fixed channel assignment strategy each cell is allocated a fixed number of voice channels. Any communication within the cell can only be made with the designated unused channels of that particular cell. Suppose if all the channels are occupied, then the call is blocked and subscriber has to wait. This is simplest of the channel assignment strategies as it requires very simple circuitry but provides worst channel utilization. Later there was another approach in which the channels were borrowed from adjacent cell if all of its own designated channels were occupied. This was named as borrowing strategy. In such cases the MSC supervises the borrowing process and ensures that none of the calls in progress are interrupted.

Dynamic Channel Assignment (DCA)
In dynamic channel assignment strategy channels are temporarily assigned for use in cells for the duration of the call. Each time a call attempt is made from a cell the corresponding BS requests a channel from MSC. The MSC then allocates a channel to the requesting the BS. After the call is over the channel is returned and kept in a central pool. To avoid co-channel interference any channel that in use in one cell can only be reassigned simultaneously to another cell in the system if the distance between the two cells is larger than minimum reuse distance. When compared to the FCA, DCA has reduced the likelihood of blocking and even increased the trunking capacity of the network as all of the channels are available to all cells, i.e., good quality of service. But this type of assignment strategy results in heavy load on switching center at heavy traffic condition.

Handover Process

When a user moves from one cell to the other, to keep the communication between the user pair, the user channel has to be shifted from one BS to the other without interrupting the call, i.e., when a MS moves into another cell, while the conversation is still in progress, the MSC automatically transfers the call to a new FDD channel without disturbing the conversation. This process is called as handoff. A schematic diagram of handover is given below Processing of handoff is an important task in any cellular system. Handovers must be performed successfully and be imperceptible to the users.



Handoff operation

– identifying a new base station

– re-allocating the voice and control channels with the new base station.

Handoff Threshold

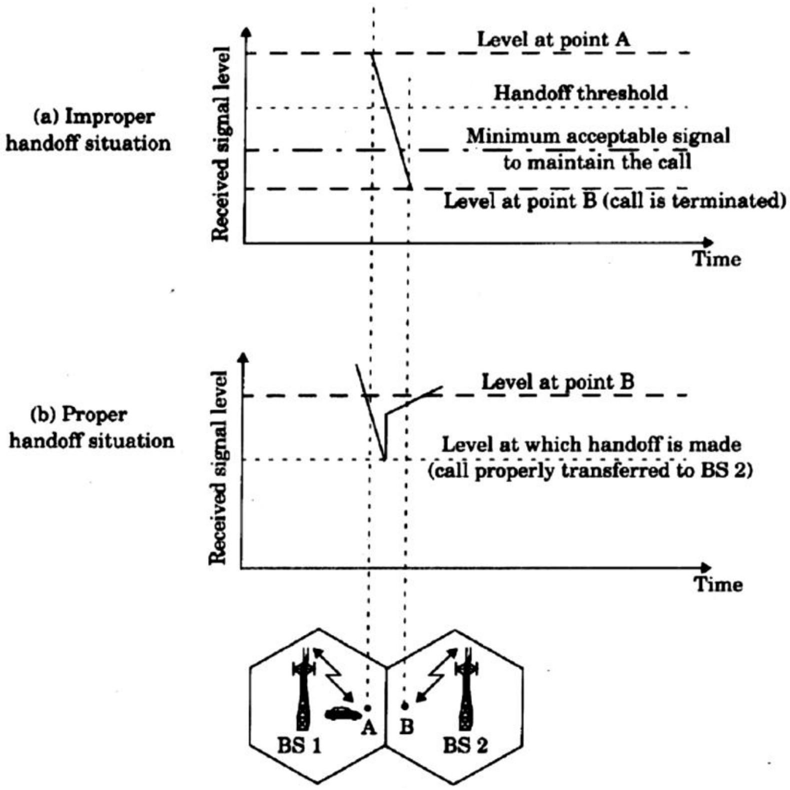
– Minimum usable signal for acceptable voice quality (-90dBm to -100dBm)

– Handoff margin cannot be too large or too small.

$$\Delta = P_{r,handoff} - P_{r,minimum\ usable}$$

– If it is too large, unnecessary handoffs burden the MSC

– If it is too small, there may be insufficient time to complete handoff before a call is lost.



Handoff must ensure that the drop in the measured signal is not due to momentary fading and that the mobile is actually moving away from the serving base station.

• Running average measurement of signal strength should be optimized so that unnecessary handoffs are avoided.

– Depends on the speed at which the vehicle is moving.

– Steep short term average -> the hand off should be made quickly

– The speed can be estimated from the statistics of the received short-term fading signal at the base station

• Dwell time: the time over which a call may be maintained within a cell without handoff.

• Dwell time depends on

– propagation

– interference

– distance

– speed

Handoff measurement

- In first generation analog cellular systems, signal strength measurements are made by the base station and supervised by the MSC.
- In second generation systems (TDMA), handoff decisions are mobile assisted, called mobile assisted handoff (MAHO)
- Intersystem handoff: If a mobile moves from one cellular system to a different cellular system controlled by a different MSC.
- Handoff requests is much important than handling a new call.

Practical Handoff Consideration

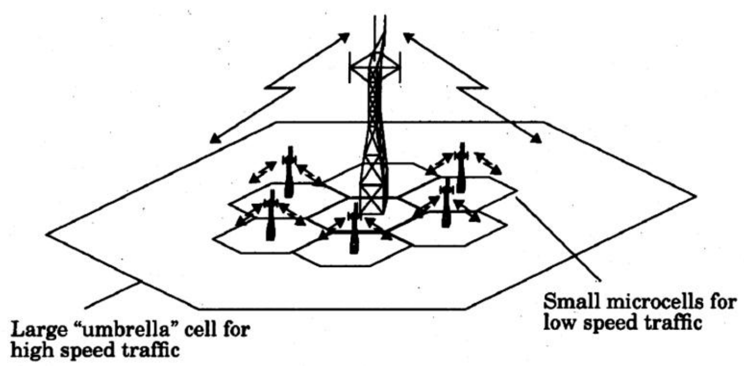
• Different type of users
– High speed users need frequent handoff during a call.
– Low speed users may never need a handoff during a call.

• Microcells to provide capacity, the MSC can become burdened if high speed users are constantly being passed between very small cells.

• Minimize handoff intervention
– handle the simultaneous traffic of high speed and low speed users.

• Large and small cells can be located at a single location (umbrella cell)
– different antenna height
– different power level

• Cell dragging problem: pedestrian users provide a very strong signal to the base station
– The user may travel deep within a neighboring cell



Types of Handover

1. Hard Handover: it occors when carrier frequency changes during handover from one cell to another. Also called *break before make* type.

2. Soft Handover: it occurs when serving cell and neighbour cell has the same carrier frequency and no change in frequency takes place during handover. Also called *make before break* type. Soft handover is characteristic of CDMA cellular system.

3. Softer Handover: it occurs when handover takes place between two sectors of the same base station. Carrier frequency remains unchanged.

4. Soft-Softer Handover: it occurs when handover takes place between two different sectors of two different base stations. Carrier frequency remains unchanged.

Interference & System Capacity

Susceptibility and interference problems associated with mobile communications equipment are because of the problem of time congestion within the electromagnetic spectrum. It is the limiting factor in the performance of cellular systems. This interference can occur from clash with another mobile in the same cell or because of a call in the adjacent cell. There can be interference between the base stations operating at same frequency band or any other non-cellular system's energy leaking inadvertently into the frequency band of the cellular system. If there is an interference in the voice channels, cross talk is heard will appear as noise between the users. The interference in the control channels leads to missed and error calls because of digital signaling. Interference is more severe in urban areas because of the greater RF noise and greater density of mobiles and base stations. The interference can be divided into 2 parts: co-channel interference and adjacent channel interference.

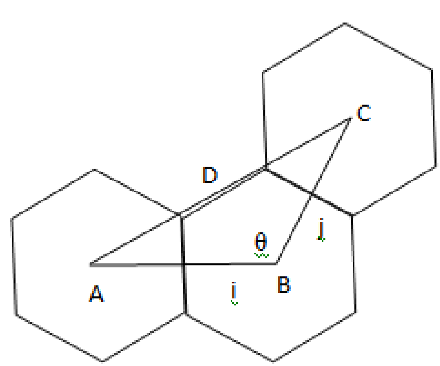
Co-channel interference (CCI)

For the efficient use of available spectrum, it is necessary to reuse frequency bandwidth over relatively small geographical areas. However, increasing frequency reuse also increases interference, which decreases system capacity and service quality. The cells where the same set of frequencies is used are call co-channel cells. Co-channel interference is the cross talk between two different radio transmitters using the same radio frequency as is the case with the co-channel cells. The reasons of CCI can be because of either adverse weather conditions or poor frequency planning or overlycrowded radio spectrum. If the cell size and the power transmitted at the base stations are same then CCI will become independent of the transmitted power and will depend on radius of the cell (R) and the distance between the interfering co-channel cells (D). If D/R ratio is increased, then the effective distance between the co-channel cells will increase and interference will decrease. The parameter Q is called the frequency reuse ratio and is related to the cluster size. For hexagonal geometry

$$Q = D/R = \sqrt{3N}.$$

From the above equation, small of `Q' means small value of cluster size `N' and increase in cellular capacity. But large `Q' leads to decrease in system capacity but increase in transmission quality. Choosing the options is very careful for the selection of `N', the proof of which is given in the first section.

Derivation of Co-channel reuse ratio Q



From the geometry of regular hexagons in the above figure,

$$\boldsymbol{D^2 = i^2 - 2ij\cos\theta + j^2}$$

$$\boldsymbol{D^2 = (i \times 2R_p)^2 + 2(i \times 2R_p)(i \times 2R_p)\cos 120 + \left(j \times 2R_p\right)^2}$$

Where, R_p = center to center distance of hexagonal cells

$$\boldsymbol{D^2 = 4R_p^2(i^2 + ij + j^2)}$$

For regular hexagonal cell,

$$\boldsymbol{R_p = \frac{\sqrt{3}}{2}R}$$

Thus,

$$\boldsymbol{D^2 = 4 \times \frac{3R^2}{4}(i^2 + ij + j^2)}$$

We know from the equation of cluster size,

$$\boldsymbol{N = i^2 + ij + j^2}$$

Hence we have,

$$\boldsymbol{Q = \frac{D}{R} = \sqrt{3N}}$$

The Signal to Interference Ratio (SIR) for a mobile receiver which monitors the forward channel can be calculated as

$$\frac{S}{I} = \frac{S}{\sum_{i=1}^{i_0} I_i}$$

Where, i_0 is the number of co-channel interfering cells, S is the desired signal power from the baseband station and I_i is the interference power caused by the i-th interfering co-channel base station. In order to solve this equation from power calculations, we need to look into the signal power characteristics. The average power in the mobile radio channel decays as a power law of the distance of separation between transmitter and receiver. The expression for the received power P_r at a distance d can be approximately calculated as

$$P_r = P_0(\frac{d}{d_0})^{-n}$$

Where, P_0 is the power received at a close-in reference point in the far field region at a small distance do from the transmitting antenna and `n' is the path loss exponent. Let us calculate the SIR for this system. If D_i is the distance of the i-th interferer from the mobile, the received power at a given mobile due to i-th interfering cell is proportional to $(D_i)^{-n}$ (the value of 'n' varies between 2 and 4 in urban cellular systems).

Let us take that the path loss exponent is same throughout the coverage area and the transmitted power be same, then SIR can be approximated as

$$\frac{S}{I} = \frac{R^{-n}}{\sum_{i=1}^{i_0} D_i^{-n}}$$

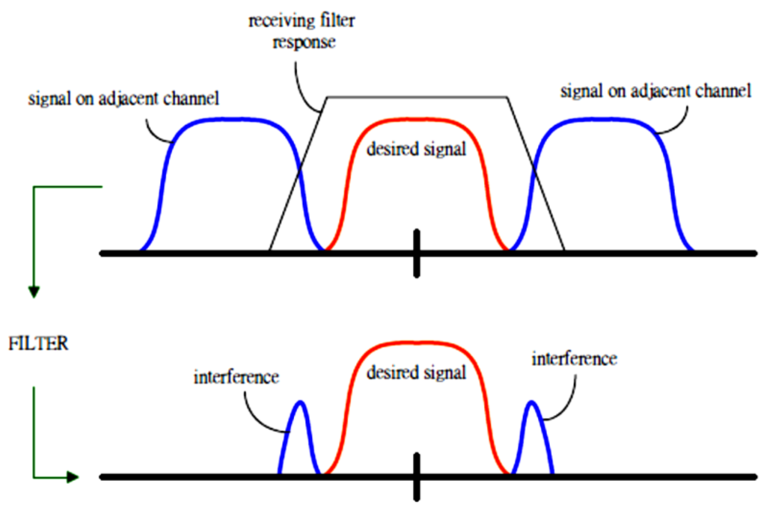
where the mobile is assumed to be located at R distance from the cell center. If we consider only the first layer of interfering cells and we assume that the interfering base stations are equidistant from the reference base station and the distance between the cell centers is 'D' then the above equation can be converted as

$$\frac{S}{I} = \frac{(D/R)^n}{i_0} = \frac{(\sqrt{3N})^n}{i_0}$$

The effect of co-channel interference can be minimized by optimizing the frequency assignments of the base stations and their transmit powers. Tilting the base-station antenna to limit the spread of the signals in the system can also be done.

Adjacent Channel Interference (ACI)

This is a different type of interference which is caused by adjacent channels i.e. channels in adjacent cells. It is the signal impairment which occurs to one frequency due to presence of another signal on a nearby frequency. This occurs when imperfect receiver _lters allow nearby frequencies to leak into the passband. This problem is enhanced if the adjacent channel user is transmitting in a close range compared to the subscriber's receiver while the receiver attempts to receive a base station on the channel. This is called near-far effect. The more adjacent channels are packed into the channel block, the higher the spectral effciency, provided that the performance degradation can be tolerated in the system link budget. This effect can also occur if a mobile close to a base station transmits on a channel close to one being used by a weak mobile. This problem might occur if the base station has problem in discriminating the mobile user from the "bleed over" caused by the close adjacent channel mobile. Adjacent channel interference occurs more frequently in small cell clusters and heavily used cells. If the frequency separation between channel is kept large this interference can be reduced to some extent. Thus assignment of channels is given such that they do not form a contiguous band of frequencies within a particular cell and frequency separation is maximized. Efficient assignment strategies are very much important in making the interference as less as possible.



Adjacent channel interference can be minimized through careful filtering and *channel assignment*.
• Keep the frequency separation between each channel in a given cell as large as possible

Trunking and Grade of Service
Cellular systems use the concept of trunking to accommodate a large number of users in a limited radio spectrum. Resources are shared so that the number of lines is much smaller than the number of possible connections. A line that connects switching o_ces and that is shared among users on an as-needed basis is called a trunk. A call that cannot be completed owing to a lack of resources is said to be blocked. So one important to be answered in mobile cellular systems is: How many channels per cell are needed in a cellular telephone system to ensure a reasonably low probability that a call will be blocked?

In a trunked radio system, a channel is allotted on per call basis. The performance of a radio system can be estimated in a way by looking at how efficiently the calls are getting connected and also how they are being maintained at handoffs. Some of the important factors to take into consideration are

- Arrival statistics
- Service statistics
- Number of available channels

Trunking mainly exploits the statistical behavior of users so that a fixed number of channels can be used to accommodate a large, random user community. As the number of telephone lines decrease, it becomes more likely that all channels are busy for a particular user. As a result, the call gets rejected and in some systems, a queue may be used to hold the caller's request until a channel becomes available. In the telephone system context the term Grade of Service (GoS) is used to mean the probability that a user's request for service will be blocked because a required facility, such as a trunk or a cellular channel, is not available. For example, a GoS of 2 % implies that on the average a user might not be successful in placing a call on 2 out of every 100 attempts. In practice the blocking frequency varies with time. One would expect far more call attempts during business hours than during the middle of the night. Telephone operating companies maintain usage records and can identify a "busy hour", that is, the hour of the day during which there is the greatest demand for service. Typically, telephone systems are engineered to provide a specified grade of service during a speci_ed busy hour.

User calling can be modeled statistically by two parameters: the average number of call requests per unit time λ and the average holding time H. The parameter λ is also called the average arrival rate, referring to the rate at which calls from a single user arrive. The average holding time is the average duration of a call. The product:

$$A_u = \lambda H$$

That is, the product of the average arrival rate and the average holding time{is called the offered traffic intensity or offered load. This quantity represents the average traffic that a user provides to the system. Offered traffic intensity is a quantity that is traditionally measured in Erlangs. One Erlang represents the amount of traffic intensity carried by a channel that is completely occupied. For example, a channel that is occupied for thirty minutes during an hour carries 0.5 Erlang of traffic. Call arrivals or requests for service are modeled as a Poisson random process. It is based on the assumption that there is a large pool of users who do not cooperate in deciding when to place calls. Holding times are very well predicted using an exponential probability distribution. This implies that calls of long duration are much less frequent than short calls. If there are U numbers of user then total offerred traffic intensity is given by

$$A=UA_u$$

Two models are widely used in tra_c engineering to represent what happens when a call is blocked. The blocked calls cleared model assumes that when a channel or trunk is not available to service an arriving call, the call is cleared from the system. The second model is known as blocked calls delayed. In this model a call that cannot be serviced is placed on a queue and will be serviced when a channel or trunk becomes available. Use of the blocked-calls-cleared statistical model leads to the Erlang B formula that relates offered traffic intensity A, grade of service GoS, and number of channels C. The Erlang B formula is:

$$\boldsymbol{P_r[blocking] = \frac{\frac{A^C}{C!}}{\sum_{k=0}^{C}\frac{A^k}{k!}} = GOS}$$

Improving capacity in cellular system

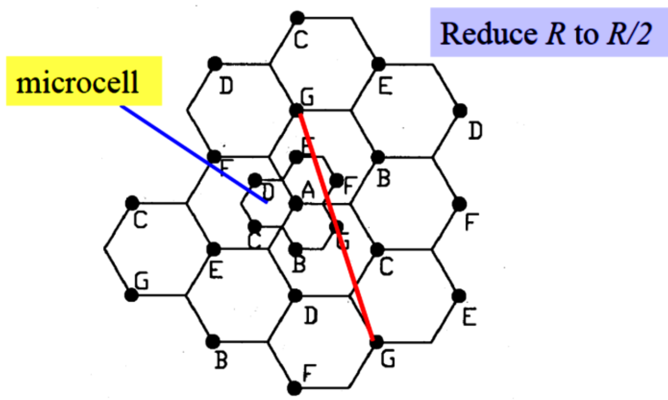
The basic idea of adopting the cellular approach is to allow space for the growth of mobile users. When a new system is deployed, the demand for it is fairly low and users are assumed to be uniformly distributed over the service area. However, as new users subscribe to the cellular service, the demand for channels may begin to exceed the capacity of some base stations.

Methods for improving capacity in cellular systems

- Cell Splitting: subdividing a congested cell into smaller cells.
- Sectoring: directional antennas to control the interference and frequency reuse.

Cell-Splitting

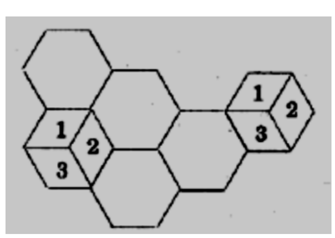
Cell Splitting is based on the cell radius reduction and minimizes the need to modify the existing cell parameters. Cell splitting involves the process of sub-dividing a congested cell into smaller cells, each with its own base station and a corresponding reduction in antenna size and transmitting power. This increases the capacity of a cellular system since it increases the number of times that channels are reused. Since the new cells have smaller radii than the existing cells, inserting these smaller cells, known as microcells, between the already existing cells results in an increase of capacity due to the additional number of channels per unit area. There are few challenges in increasing the capacity by reducing the cell radius. Clearly, if cells are small, there would have to be more of them and so additional base stations will be needed in the system. The challenge in this case is to introduce the new base stations without the need to move the already existing base station towers. The other challenge is to meet the generally increasing demand that may vary quite rapidly between geographical areas of the system. For instance, a city may have highly populated areas and so the demand must be supported by cells with the smallest radius. The radius of cells will generally increase as we move from urban to sub urban areas, because the user density decreases on moving towards sub-urban areas. The key factor is to add as



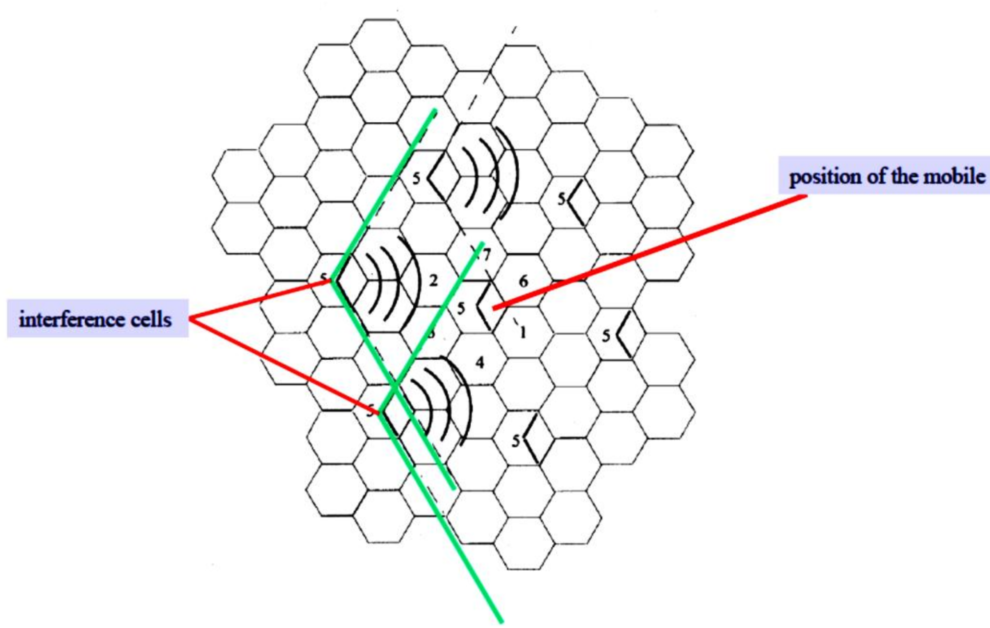
minimum number of smaller cells as possible wherever an increase in demand occurs. The gradual addition of the smaller cells implies that, at least for a time, the cellular system operates with cells of more than one size. Consider that the cells in the center of the diagram are becoming congested, and cell A in the center has reached its maximum capacity. Figure also shows how the smaller cells are being superimposed on the original layout. The new smaller cells have half the cell radius of the original cells. At half the radius, the new cells will have one-fourth of the area and will consequently need to support one-fourth the number of subscribers. Notice that one of the new smaller cells lies in the center of each of the larger cells. If we assume that base stations are located in the cell centers, this allows the original base stations to be maintained even in the new system layout. However, new base stations will have to be added for new cells that do not lie in the center of the larger cells. The organization of cells into clusters is independent of the cell radius, so that the cluster size can be the same in the small-cell layout as it was in the large-cell layout. Also the signal-to-interference ratio is determined by cluster size and not by cell radius. Consequently, if the cluster size is maintained, the signal-to-interference ratio will be the same after cell splitting as it was before. If the entire system is replaced with new half-radius cells, and the cluster size is maintained, the number of channels per cell will be exactly as it was before, and the number of subscribers per cell will have been reduced.

Sectoring

Sectoring is basically a technique which can increase the SIR without necessitating an increase in the cluster size. Till now, it has been assumed that the base station is located in the center of a cell and radiates uniformly in all the directions behaving as an omni-directional antenna. However, it has been found that the co-channel interference in a cellular system may be decreased by replacing a single omni-directional antenna at the base station by several directional antennas, each radiating within a specified sector. If the base station feeds three 120^o directional antennas, each of which radiates into one of the three sectors. The channel set serving this cell has also been divided, so that each sector is assigned one-third of the available number cell of channels. This technique for reducing co-channel interference wherein by using suitable directional antennas, a given cell would receive interference and transmit with a fraction of available co-channel cells is called 'sectoring'.



- Interference Reduction



From the diagram above it can be seen that due to the use of sectral antenna number of interferers for sectors goes lesser. Hence improvement in SIR

$$\frac{S}{I} = \frac{(D/R)^n}{i_0} = \frac{(\sqrt{3N})^n}{i_0}$$

This improvement in SIR allows reduction in cluster size. Thus higher number of cluster reuse increases overall system capacity.

Mobile Radio Propagation

There are two basic ways of transmitting an electro-magnetic (EM) signal, through a guided medium or through an unguided medium. Guided mediums such as coaxial cables and fiber optic cables are far less hostile toward the information carrying EM signal than the wireless or the unguided medium. It presents challenges and conditions which are unique for this kind of transmissions. A signal, as it travels through the wireless channel, undergoes many kinds of propagation effects such as reflection, diffraction and scattering, due to the presence of buildings, mountains and other such obstructions. Reflection occurs when the EM waves impinge on objects which are much greater than the wavelength of the traveling wave. Diffraction is a phenomena occurring when the wave interacts with a surface having sharp irregularities. Scattering occurs when the medium through the wave is traveling contains objects which are much smaller than the wavelength of the EM wave. These varied phenomena's lead to large scale and small scale propagation losses. Due to the inherent randomness associated with such channels they are best described with the help of statistical models. Models which predict the mean signal strength for arbitrary transmitter receiver distances are termed as large scale propagation models. These are termed so because they predict the average signal strength for large Tx-Rx separations, typically for hundreds of kilometers.

Free Space Propagation Model

Although EM signals when traveling through wireless channels experience fading effects due to various effects, but in some cases the transmission is with a direct line of sight such as in satellite communication. Free space model predicts that the received power decays as negative square root of the distance. Friis free space equation is given by

$$P_r(d) = \frac{P_t G_t G_r \lambda^2}{(4\pi)^2 d^2 L}$$

where P_t is the transmitted power, $P_r(d)$ is the received power, G_t is the transmitter antenna gain, G_r is the receiver antenna gain, d is the Tx-Rx separation and L is the system loss factor depend upon line attenuation, filter losses and antenna losses and not related to propagation. The gain of the antenna is related to the effective aperture of the antenna which in turn is dependent upon the physical size of the antenna as given below

$$G = 4\pi A_e/\lambda^2.$$

The path loss, representing the attenuation suffered by the signal as it travels through the wireless channel is given by the difference of the transmitted and received power in dB and is expressed as:

$$PL\ (dB) = 10log\frac{P_t}{P_r} = -10log\left[\frac{\lambda^2}{(4\pi d)^2}\right]$$

The fields of an antenna can broadly be classified in two regions, the far field and the near field. It is in the far field that the propagating waves act as plane waves and the power decays inversely with distance. The far field region is also termed as Fraunhofer region and the Friis equation holds in this region. Hence, the Friis equation is used only beyond the far field distance, df, which is dependent upon the largest dimension of the antenna as

$$d_f = 2D^2/\lambda.$$

Also we can see that the Friis equation is not de_ned for d=0. For this reason, we use a close in distance, do, as a reference point. The power received, Pr(d), is then given by:

$$P_r(d) = P_r(d_o)(d_o/d)^2$$

Basic Methods of Propagation

Reflection, diffraction and scattering are the three fundamental phenomena that cause signal propagation in a mobile communication system, apart from LoS communication. The most important parameter, predicted by propagation models based on above three phenomena, is the received power. The physics of the above phenomena may also be used to describe small scale fading and multipath propagation. The following subsections give an outline of these phenomena.

Reflection

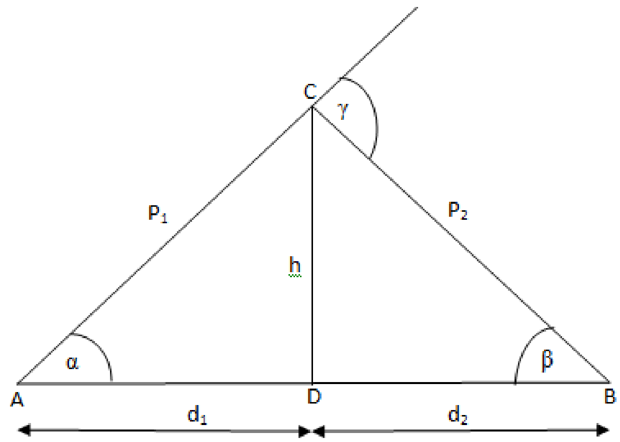
Reflection occurs when an electromagnetic wave falls on an object, which has very large dimensions as compared to the wavelength of the propagating wave. For example, such objects can be the earth, buildings and walls. When a radio wave falls on another medium having different electrical properties, a part of it is transmitted into it, while some energy is reflected back. Let us see some special cases. If the medium on which the e.m. wave is incident is a dielectric, some energy is reflected back and some energy is transmitted. If the medium is a perfect conductor, all energy is reflected back to the first medium. The amount of energy that is reflected back depends on the polarization of the e.m. wave. Another particular case of interest arises in parallel polarization, when no reflection occurs in the medium of origin. This would occur, when the incident angle would be such that the reflection coefficient is equal to zero. This angle is the Brewster's angle.

Diffraction

Diffraction is a phenomenon that takes place when the radio wave strikes a surface and changes its direction of propagation owing to the inability of the surface to absorb it. The loss due to diffraction depends upon the kind of obstruction in the path. In practice, the mobile antenna is at a much lower height than the base station antenna, and there may be high buildings or hills in the area. Thus, the signal undergoes diffraction in reaching the mobile antenna. This phenomenon is also known as 'shadowing' because the mobile receiver is in the shadow of these structures.

Also, diffraction is the phenomenon due to which an EM wave can propagate beyond the horizon, around the curved earth's surface and obstructions like tall buildings. As the user moves deeper into the shadowed region, the received field strength decreases. But the diffraction field still exists and it has enough strength to yield a good signal. This phenomenon can be explained by the Huygen's principle, according to which, every point on a wavefront acts as point sources for the production of secondary wavelets, and they combine to produce a new wavefront in the direction of propagation. The propagation of secondary wavelets in the shadowed region results in diffraction. The field in the shadowed region is the vector sum of the electric field components of all the secondary wavelets that are received by the receiver.

Fresnel Zone Geometry:
Consider a transmitter and receiver located accros a obstruction as shown in the diagram.



Assumptions:

- **$d_1, d_2 >> h$**
- A : Transmitter
- B : Receiver
- CD : Height of obstruction above LOS = h
- C : Diffraction point
- P_1 and P_2 : Path traversed by EMW
- α, β, and γ : various angle made by EMW while travelling from TX to RX
- d_1 and d_2 : distance of obstruction from transmitter and reciever respectively

In the triangle ACD,

$$P_1 = \sqrt{d_1^2 + h^2}$$

$$P_1 = \sqrt{1 + \left(\frac{h}{d_1}\right)^2}$$

Applying Taylor's series,

$$P_1 = d_1\left(1 + \frac{1}{2}\frac{h^2}{d_1^2}\right)$$

Similarly, for triangle BCD,

$$P_2 = \sqrt{d_2^2 + h^2}$$

$$P_2 = \sqrt{1 + \left(\frac{h}{d_2}\right)^2}$$

$$P_2 = d_2\left(1 + \frac{1}{2}\frac{h^2}{d_2^2}\right)$$

Thus path difference is,

$$\Delta = (P_1 + P_2) - (d_1 + d_2)$$

$$\Delta = \boldsymbol{\frac{h^2}{2}\left(\frac{d_1 + d_2}{d_1 d_2}\right)}$$

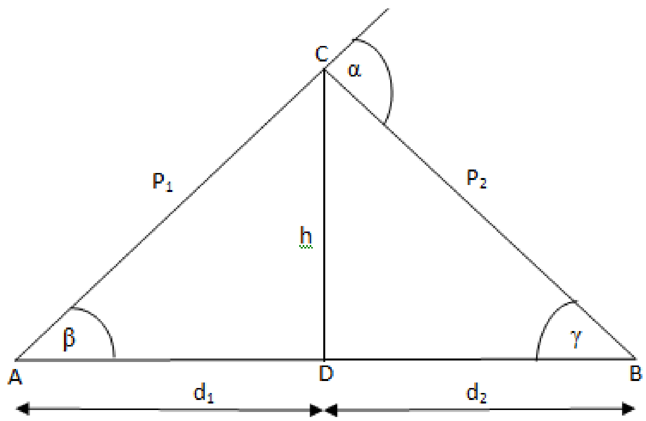
Phase difference is,

$$\boldsymbol{\varphi = \frac{2\pi}{\lambda}\Delta}$$

$$\boldsymbol{\varphi = \frac{2\pi}{\lambda}\frac{h^2}{2}\left(\frac{d_1 + d_2}{d_1 d_2}\right)}$$

Derivation of Fresnel-Kirchoff diffraction parameter relation

In the following diagram,



$$\tan\beta = \frac{h}{d_1} \approx \beta$$

And,

$$\tan\gamma = \frac{h}{d_2} \approx \gamma$$

From the property of triangles,

$$\alpha = \beta + \gamma$$

Therefore,

$$\alpha = \frac{h}{d_1} + \frac{h}{d_2} = h\left(\frac{\boldsymbol{d_1} + \boldsymbol{d_2}}{\boldsymbol{d_1 d_2}}\right)$$

From Fresnel-Kirchoff diffraction condition,

$$\varphi = \frac{\pi}{2}\nu^2$$

Where, ν is Fresnel-Kirchoff diffraction parameter,

So,

$$\frac{2\pi}{\lambda}\frac{h^2}{2}\left(\frac{d_1 + d_2}{d_1 d_2}\right) = \frac{\pi}{2}\nu^2$$

$$\nu = h\sqrt{\frac{2}{\lambda}\left(\frac{d_1 + d_2}{d_1 d_2}\right)}$$

Also,

$$\alpha = h\left(\frac{\boldsymbol{d_1 + d_2}}{\boldsymbol{d_1 d_2}}\right)$$

Thus,

$$\nu = \frac{\alpha(\boldsymbol{d_1 d_2})}{(\boldsymbol{d_1 + d_2})}\sqrt{\frac{2}{\lambda}\left(\frac{d_1 + d_2}{d_1 d_2}\right)}$$

$$\nu = \alpha\sqrt{\left[\frac{(\boldsymbol{d_1 d_2})}{(\boldsymbol{d_1 + d_2})}\right]^2 \times \frac{2}{\lambda}\left(\frac{d_1 + d_2}{d_1 d_2}\right)}$$

Hence,

$$\nu = \alpha\sqrt{\frac{2}{\lambda}\left(\frac{d_1 d_2}{d_1 + d_2}\right)}$$

Scattering

The actual received power at the receiver is somewhat stronger than claimed by the models of reflection and diffraction. The cause is that the trees, buildings and lampposts scatter energy in all directions. Hence, scattering occurs when an electromagnetic wave falls on an object, which has very small dimensions as compared to the wavelength of the propagating wave. Scattering occurs due to the roughness of the surface upon which EM wave impinges. Roughness is tested by a Rayleigh criterion, which defines a critical height h_c of surface protuberances for a given angle of incidence θi, given by,

$$h_c = \frac{\lambda}{8 sin\theta_i}$$

A surface is smooth if its minimum to maximum protuberance h is less than h_c, and rough if protuberance is greater than h_c.

Practical Link Budget Models

OKUMURA MODEL

The Okumura model is used for Urban Areas is a Radio propagation model that is used for signal prediction.The frequency coverage of this model is in the range of 150 MHz to 1920 MHz and distances of 1 Km to 100 Km.It can be applicable for base station effective antenna heights (ht) ranging from 30 m to 1000 m. Okumura used extensive measurements of base station-to-mobile signal attenuation throughout Tokyo to develop a set of curves giving median attenuation relative to free space (A_{mu}) of signal propogation in irregular terrain. The empirical pathloss formula of Okumura at distance d parameterized by the carrier frequency f_c is given by

$$L_{50}(\mathrm{dB}) = L_F + A_{mu}(f, d) - G(h_{te}) - G(h_{re}) - G_{AREA}$$

where L_F is free space path loss at distance d and carrier frequency f_c, $A_{mu}(f, d)$ is the median attenuation in addition to free-space path loss across all environments,$G(h_t)$ is the base station antenna height gain factor,$G(h_r)$ is the mobile antenna height gain factor,G_{AREA} is the gain due to type of environment. The values of $A_{mu}(f_c.d)$ and G_{AREA} are obtained from Okumura's empirical plots. Okumura derived empirical formulas for $G(h_t)$ and $G(h_r)$ as follows:

$$G(h_t) = 20\log_{10}(h_t/200), \qquad 30m < h_t < 1000m$$

$$G(h_r) = 10\log_{10}(h_r/3), \qquad h_r \leq 3m$$

$$G(h_r) = 20\log_{10}(h_r/3), \qquad 3m < h_r < 10m$$

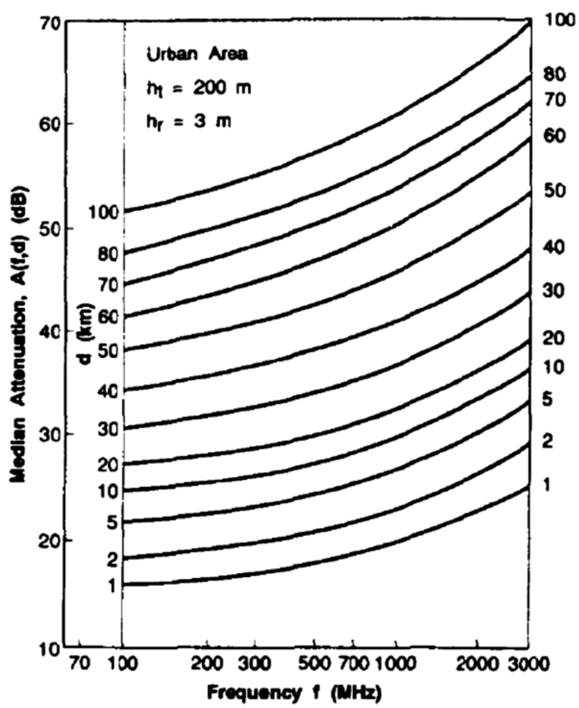


Fig: Okumura Curves

Okumura's model has a 10-14 dB empirical standard deviation between the path loss predicted by the model and the path loss associated with one of the measurements used to develop the model.

HATA MODEL

The Hata model is an empirical formulation of the graphical path-loss data provided by the Okumura and is valid over roughly the same range of frequencies, 150-1500 MHz. This empirical formula simplifies the calculation of path loss because it is closed form formula and it is not based on empirical curves for the different parameters. The standard formula for empirical path loss in urban areas under the Hata model is

$$P_{L,urban}(d)dB = 69.55+26.16\log_{10}(f_c)-13.82\log_{10}(h_t)-a(h_r)+(44.9-6.55\log_{10}(h_t))\log_{10}(d)$$

The parameters in this model are same as in the Okumura model,and $a(h_r)$ is a correction factor for the mobile antenna height based on the size of coverage area. For small to medium sized cities this factor is given by

$$a(h_r) = (1.11\log_{10}(f_c) - 0.7)h_r - (1.56\log_{10}(f_c) - 0.8)dB$$

And, for larger cities at a frequencies fc > 300 MHz by

$$a(h_r) = 3.2(\log_{10}(11.75h_r))^2 - 4.97dB$$

Else, it is

$$a(h_r) = 8.29(\log_{10}(1.54h_r))^2 - 1.1dB$$

Corrections to the urban model are made for the suburban, and is given by

$$P_{L,suburban}(d)dB = P_{L,urban}(d)dB - 2(\log_{10}(f_c/28))^2 - 5.4$$

Corrections to the urban model are made for the open and rural area, is given by

$$L_{50}(\mathrm{dB}) = L_{50}(urban) - 4.78(\log f_c)^2 - 18.33\log f_c - 40.98$$

The Hata model well approximates the Okumura model for distances d > 1 Km. Hence it is a good model for first generation cellular systems, but it does not model propogation well in current cellular systems with smaller cell sizes and higher frequencies. Indoor environments are also not captured by the Hata model.

Longley-Rice Model

Longley-Rice model is applicable to point to point communication systems in the frequency range from 40 MHz to 100 GHz over different kinds of terrain.

- Median transmission loss predicted through terrrain profile and troposheric refractivity
- Two ray model used to predict received signal level
- Fresnel-kirchoff knife edge model used to predict diffraction losses

Modes of operation

1. Point to point mode: used when detail terrain profile is available
2. Area mode: used when detail terrain profile is not available

Drawback: No correction factor for evvironmental factors

Walfisch and Bertoni Model

This model considers the impact of rooftops and building heights by using diffraction to predict average signal strength at street level. The model considers the path loss, S, to be a product of three factors,

$$S = P_0 Q^2 P_1$$

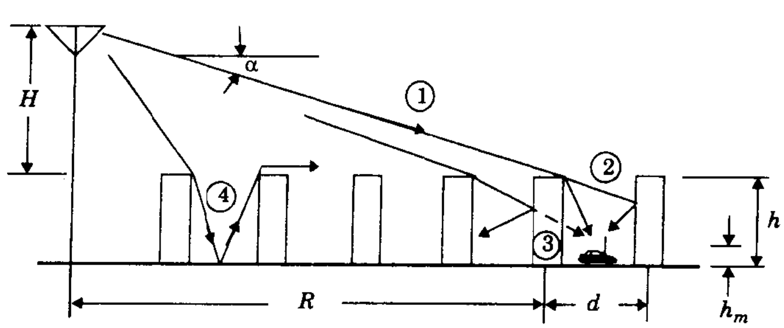
P_0 = Freespace path loss

Q^2 = Shadow effect due to tall buildings (rooftop to street diffraction and scatter factor)

P_1 = Diffraction loss due to tall buildings (multiscreen diffraction due to rows of buildings)

In dB, the path loss is given by,

$$S\ (dB) = L_0 + L_{rts} + L_{ms}$$



Small scale fading and Multipath

Small scale fading or fading is used to describe the rapid fluctuations of the amplitude, phases or multipath delays of a radio signal over a short period of time or travel distances, so that large scale path loss effects may be ignored.

Fading is caused by interference between multipath signals at the receiver. This results into wide variations in the amplitude and phases of resultant signal.

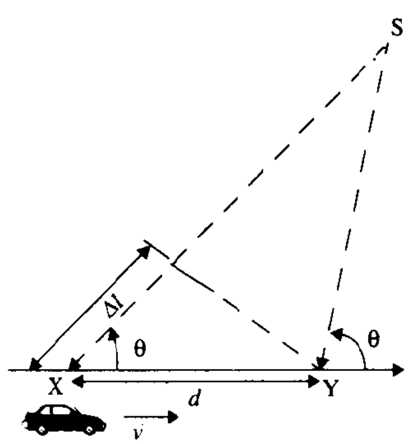
Effects:

1. Rapid changes in signal strength over small travel distance or time interval
2. Random frequency modulation (Doppler Shift)
3. Time dispersion

Factors that influence fading:

1. Multipath propagation
2. Speed of mobile
3. Speed of surrounding object
4. Transmission bandwidth of signal

Doppler Shift



Consider a mobile is moving at a constant velocity v from X to Y separated by a distance d. it receives signal from a remote source S as depicted in the diagram above.

The path difference in this case is given by,

$$\Delta l = d\ cos\theta$$

Now,

$$d = v\ \Delta t$$

$$\Delta l = v \, \Delta t \, cos\theta$$

Phase difference,

$$\Delta\emptyset = \frac{2\pi}{\lambda} \, \Delta l$$

$$\Delta\emptyset = \frac{2\pi}{\lambda} \, v \, \Delta t \, cos\theta$$

Doppler Shift in frequency is given by,

$$\omega_d = \frac{\Delta\emptyset}{\Delta t}$$

$$\omega_d = \frac{2\pi}{\lambda} \frac{v \, \Delta t \, cos\theta}{\Delta t}$$

$$\omega_d = \frac{2\pi}{\lambda} \, v \, cos\theta$$

Also,

$$\omega_d = 2\pi f_d$$

$$2\pi f_d = \frac{2\pi}{\lambda} \, v \, cos\theta$$

Hence,

$$f_d = \frac{v \, cos\theta}{\lambda}$$

Shift in frequency is thus,

$$f = f_c \pm f_d$$

Parameters of mobile multipath channels

1. Time dispersion parameter
2. Coherence bandwidth
3. Doppler spread and coherence time

Time dispersion parameters

1. Mean excess delay ($\bar{\tau}$)
2. RMS delay spread (σ_τ)
3. Maximum excess delay (X dB)

Mean excess delay:

$$\bar{\tau} = \frac{\sum_k a_k^2 \tau_k}{\sum_k a_k^2} = \frac{\sum_k P(\tau_k)\tau_k}{\sum_k P(\tau_k)}$$

RMS delay spread:

$$\sigma_\tau = \sqrt{\overline{\tau^2} - (\bar{\tau})^2}$$

$$\overline{\tau^2} = \frac{\sum_k a_k^2 \tau_k^2}{\sum_k a_k^2} = \frac{\sum_k P(\tau_k)\tau_k^2}{\sum_k P(\tau_k)}$$

Maximum excess delay is the value on time axis of a power delay profile corresponding to latest arriving path that has strength (dB) above specified threshold.

Coherence Bandwidth

Coherence bandwidth is statistical measure of the range of frequencies over which the channel can be considered flat. Flat channel passes all spectral components with approximately equal gain and linear phase.

It is also defined as the range of frequencies over which two frequency components have strong potential for amplitude correlation.

For 90% correlation, coherence bandwidth is expressed as;

$$B_c \approx \frac{1}{50\sigma_\tau}$$

For 50% correlation, coherence bandwidth is expressed as;

$$B_c \approx \frac{1}{5\sigma_\tau}$$

Doppler Spread and coherence time

This parameter describes time varying nature of channel, caused by either relative motion between mobile and base station or the movement of objects in the channel.

Doppler Spread (B_D) : Doppler spread is a measure of spectral broadening. It is defined as range of frequencies over which the received Doppler spectrum is essentially non zero.

Coherence time (T_C) : Coherence time characterizes time varying nature of the frequency dispersive channel in time domain.

$$T_C \approx \frac{1}{f_m}$$

Coherence time is statistical measure of the time duration over which the channel impulse response is essentially invariant.

Also can be defined as time duration over which two received signals have a strong potential for amplitude correlation.

For 50% correlation:

$$T_C \approx \frac{9}{16\pi f_m}$$

Rule of thumb for coherence time calculation:

$$T_C = \sqrt{\frac{9}{16\pi f_m^2}} = \frac{0.423}{f_m}$$

Types of small scale fading

Depending on the relation between the signal parameters and channel parameters transmitted signals will undergo different types of fading.

1. Multipath delay spread leads to time dispersion and frequency selective fading
2. Doppler spread leads to frequency dispersion and time selective fading

Small-Scale Fading
(Based on multipath time delay spread)

Flat Fading
1. BW of signal < BW of channel
2. Delay spread < Symbol period

Frequency Selective Fading
1. BW of signal > BW of channel
2. Delay spread > Symbol period

Small-Scale Fading
(Based on Doppler spread)

Fast Fading
1. High Doppler spread
2. Coherence time < Symbol period
3. Channel variations faster than baseband signal variations

Slow Fading
1. Low Doppler spread
2. Coherence time > Symbol period
3. Channel variations slower than baseband signal variations

Flat Fading:

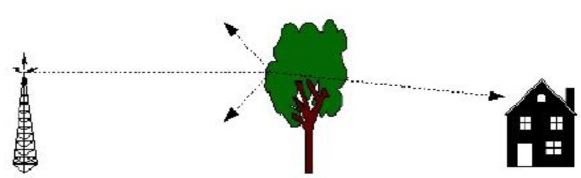
If the mobile radio channel has a constant gain and linear phase response over a bandwidth which is greater than the bandwidth of the transmitted signal then the received signal will undergo flat fading.

A signal undergo flat fading if,

$$B_S \ll B_C$$

and

$$T_S \gg \sigma_\tau$$



Flat Fading is caused by absorbers between the two antennae and is countered by antenna placement and transmits power level.

Frequency selective fading:

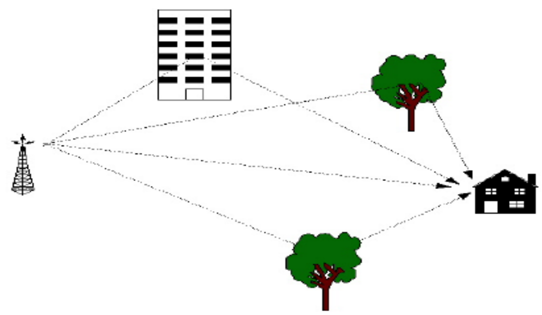
If the channel possesses a constant gain and linear phase response over a bandwidth that is smaller than the bandwidth of transmitted signal then the received signal undergo frequency selective fading.

A signal undergo flat fading if,

$$B_S > B_C$$

And,

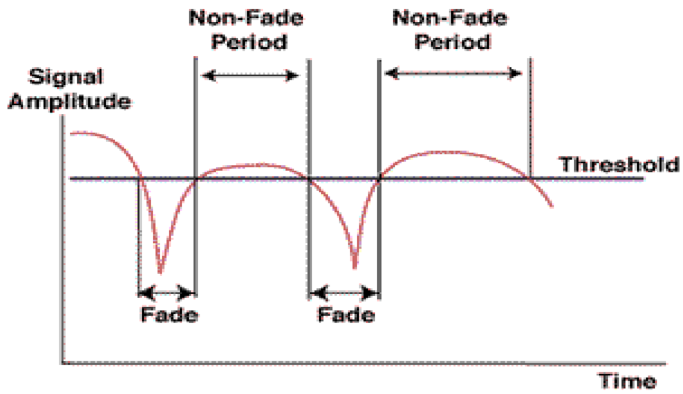
$$T_S < \sigma_\tau$$



Frequency selective fading is caused by reflectors between the transmitter and receiver creating multi-path effects.

Effects of Frequency Selective Fading

- The dips or fades in the response due to reflection cause cancellation of certain frequencies at the Receiver.
- Reflections off near-by objects (e.g. ground, buildings, trees, etc) can lead to multi-path signals of similar signal power to the direct signal.
- This can result in deep nulls in the received signal power due to destructive interference.



When the waves of multi-path signals are out of phase, reduction of the signal strength at the receiver can occur.

Fast fading:

If the mobile radio channel impulse response changes rapidly within the symbol duration the received signal undergo fast fading. In fast fading channel coherence time of the channel is smaller than the symbol period of the transmitted signal and leads to signal distortion.

A signal undergo flat fading if,

$$T_S > T_C$$

And,

$$B_S < B_D$$

Slow fading:

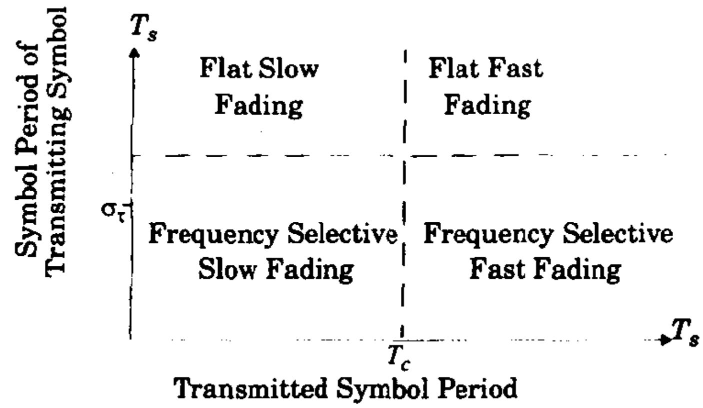
If the mobile radio channel impulse response changes at a rate much slower than the transmitted baseband signal the received signal undergo slow fading. The channel may be assumed to be static over one or several reciprocal bandwidth intervals.

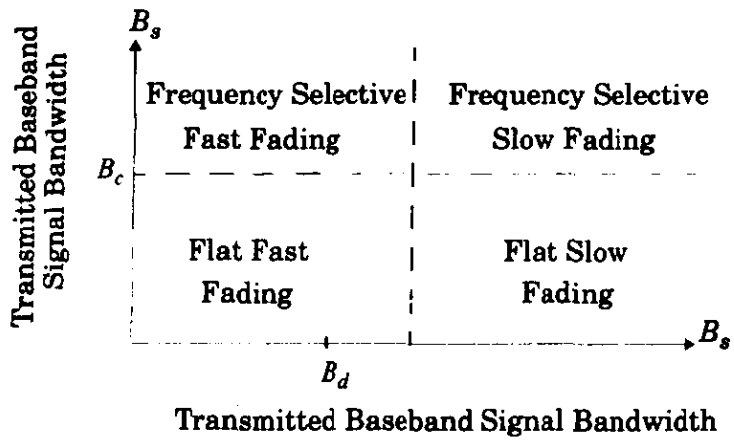
A signal undergo flat fading if,

$$T_S \ll T_C$$

And,

$$B_S \gg B_D$$



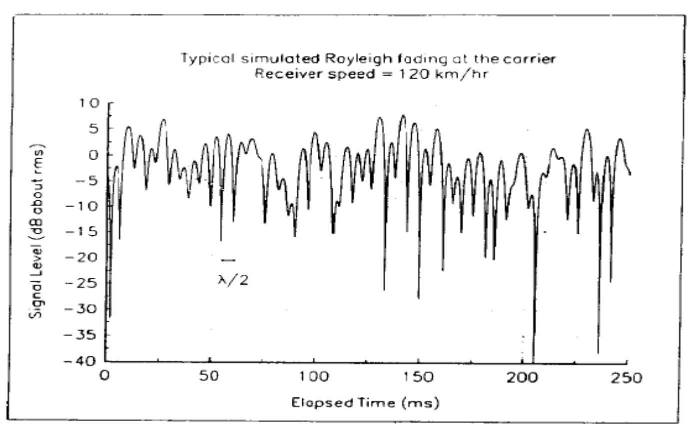


Rayleigh Fading Distribution

Rayleigh distribution is commonly used to describe the statistical time varying nature of the received envelope of a flat fading signal.

Rayleigh distribution also describe envelope of an individual multipath component.

- Rayleigh distributed signal:

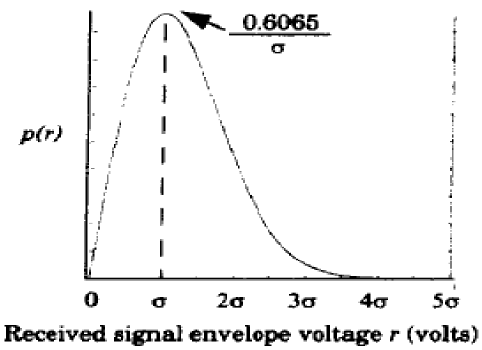


- The Rayleigh distribution has pdf

$$p(r) = \begin{cases} \frac{r}{\sigma^2}\exp\left(-\frac{r^2}{2\sigma^2}\right) & (0 \le r \le \infty) \\ 0 & (r < 0) \end{cases}$$

$\sigma =$ the rms value of the received voltage signal before envelope detection

$\sigma^2 =$ the time-average power of the received signal before envelope detection



- The probability that the envelope of the received signal does not exceed a specified value R is

$$P(R) = Pr(r \le R) = \int_0^R p(r)\,dr = 1 - \exp\left(-\frac{R^2}{2\sigma^2}\right)$$

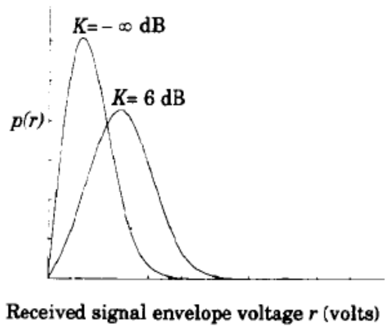
Rician Fading Distribution

When there is a dominant stationary (nonfading) signal component present, such as line-of-sight propagation path, the small-scale fading envelope distribution is Rician.

Random multipath componnets arriving at different angles are superimposed on a stationary dominant signal.

- The Rician distribution is given by

$$p(r) = \frac{r}{\sigma^2} e^{-\frac{(r^2+A^2)}{2\sigma^2}} I_0\left(\frac{Ar}{\sigma^2}\right) \quad \text{for } (A \ge 0, r \ge 0)$$
$$= 0 \quad \text{for } (r < 0)$$



- The Rician distribution is described in terms of a parameter K

$$\boxed{K = \frac{A^2}{2\sigma^2}}$$

$$\boxed{K(dB) = 10\log\frac{A^2}{2\sigma^2}}$$

- When $K = 0$ we have Rayleigh fading
- When $K = \infty$ we have no fading, channel has no multipath, only LOS component

Modulation-Demodulation Methods in Mobile Communications

Review

- Modulation is a process of varying one waveform in relation to another waveform
- Information signals are transported between a transmitter and receiver over some form of transmission medium.
- However the original information signal are seldom in a form that is suitable for Transmission
- Therefore they must be transformed from their original form into a form that is more suitable for transmission.
- The process of impressing low frequency information signals onto a high frequency carrier signal is called modulation.

Analog Modulation

- The process of transforming an analog baseband (lowpass) signal, for example an audio signal, over an analog passband channel, a limited RF band, is called analog modulation
- Eg Amplitude and Angle modulation
- Used in first generation mobile radio

Digital Modulation

- The process of transforming a digital bit stream over an analog passband channel
- Eg BPSK,QPSK,MSK etc
- Used in 2G, 3G mobile radios
- Advancements in VLSI and DSP has made digital modulation more cost efficient than analog transmission systems.

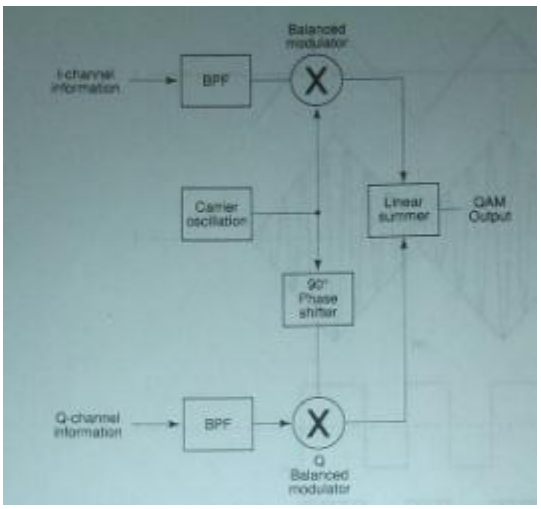
Advantages

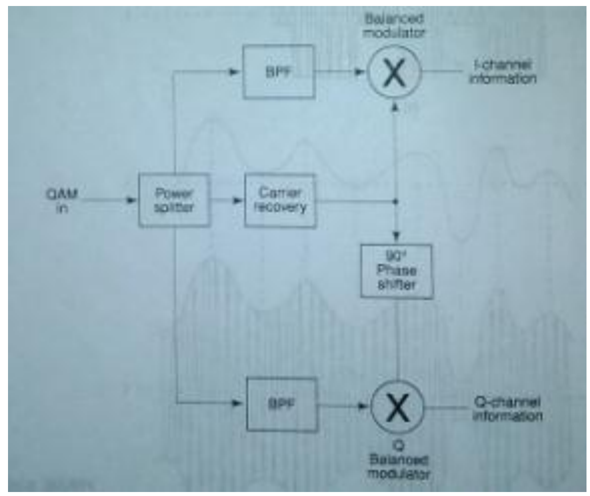
  - Greater noise immunity
  - Easier multiplexing of information (voice, data, video)
  - Can accommodate digital transmission errors, source coding, encryption and equalization.
  - DSP can implement digital modulators, demodulators completely in software.

Quadrature Amplitude Modulaion (QAM)

QAM is a form of amplitude modulation where signals from two separate information sources, that is channels, modulate the same carrier frequency at the same time without interfering with each other.m

The information sources modulate the same carrier after it has been separated into carrier signals that are 90 degree out of phase with each other.





The transmitted signal is given by:

$$s(t) = Ax_1(t)\cos(2\pi f_c) + Ax_2(t)\sin(2\pi f_c)$$

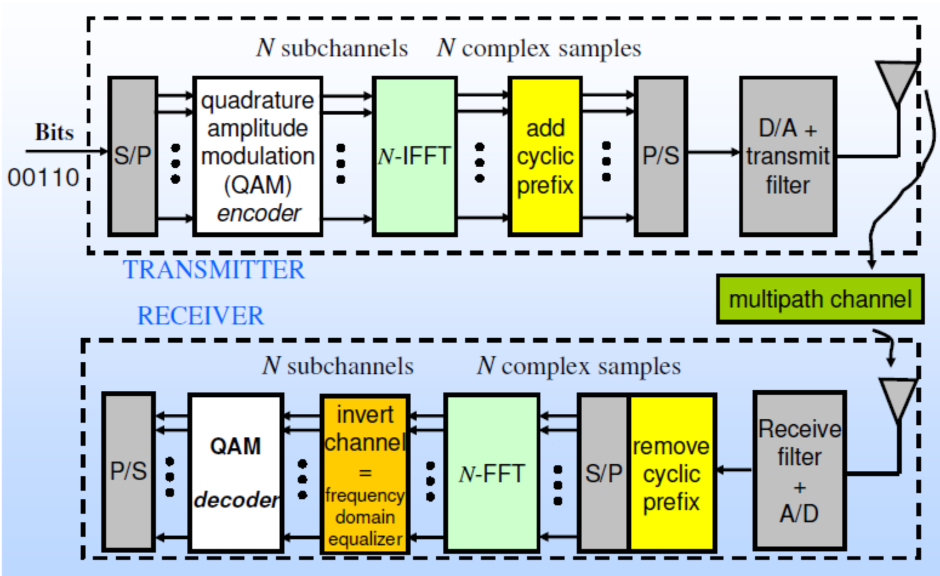
Where, x_1 (t) and x_2 (t) are two different message signals applied to the product modulators. Both message signals are band limited in the interval $-f_m \leq f \leq f_m$, thus s(t) occupy a bandwidth of $2f_m$

Orthogonal Frequency division Multiplexing (OFDM)

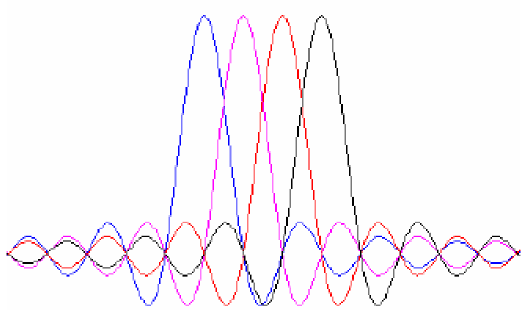
OFDM belongs to a family of transmission schemes called multicarrier modulation, which is based on the idea of dividing a given high-bit-rate data stream into several parallel lower bit-rate streams and modulating each stream on separate carriers often called subcarriers, or tones.

Therefore, in high-data-rate systems in which the symbol duration is small, being inversely proportional to the data rate, splitting the data stream into many parallel streams increases the symbol duration of each stream such that the delay spread is only a small fraction of the symbol duration.

In order to completely eliminate ISI, guard intervals are used between OFDM symbols. By making the guard interval larger than the expected multipath delay spread, ISI can be completely eliminated. Adding a guard interval, however, implies power wastage and a decrease in bandwidth efficiency.



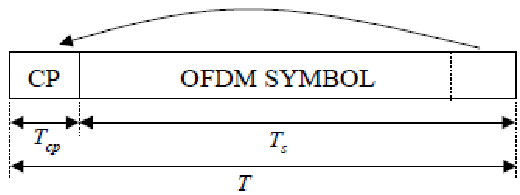
Spectra of individual subcarrier



OFDM is a spectrally efficient version of multicarrier modulation, where the subcarriers are selected such that they are all orthogonal to one another over the symbol duration, thereby avoiding the need to have nonoverlapping subcarrier channels to eliminate intercarrier interference.

Orthogonality between the subcarriers allows their overlapping while disabling the occurrence of crosstalks

Multicarrier modulation schemes eliminate or minimize intersymbol interference (ISI) by making the symbol time large enough so that the channel-induced delays are an insignificant (typically, < 10 percent) fraction of the symbol duration.



Where, T_{cp} denotes the length of the cyclic prefix and $T = T_{cp} + T_s$ is the length of the transmitted symbol.

The length of the cyclic prefix (CP) should be made longer than the experienced impulse response to avoid ISI and ICI (Inter Carrier interference). However, the transmitted energy increases with the length of the cyclic prefix

Advantages:

Immunity to delay spread

- Symbol duration >> channel delay spread
- Guard interval

Resistance to frequency selective fading

- Each subchannel is almost flat fading

Simple equalization

- Each subchannel is almost flat fading, so it only needs a one-tap equalizer to overcome channel effect.

Efficient bandwidth usage

- The subchannel is kept orthogonality with overlap.

Disadvantages:

The problem of synchronization

Symbol synchronization
  - Timing errors
  - Carrier phase noise

Frequency synchronization
  - Sampling frequency synchronization
  - Carrier frequency synchronization

High peak to average power ratio (PAPR)

Binary Phase Shift Keying (BPSK)

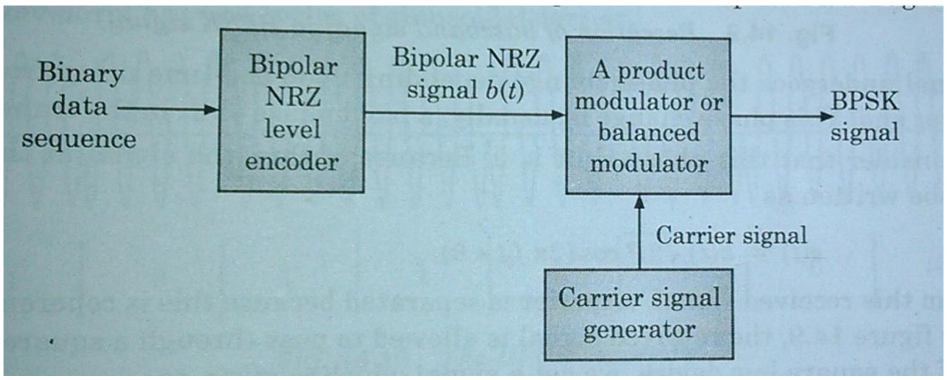
- Phase of a constant amplitude carrier signal is switched between two values according to the two possible signals m1 and m2 corresponding to binary 1 and 0 respectively

$$S_{BPSK}(t) = \sqrt{\frac{2E_b}{T_b}}\cos(2\pi f_c t + \theta_c) \quad 0 \le t \le T_b \text{ (binary 1)}$$

or

$$S_{BPSK}(t) = \sqrt{\frac{2E_b}{T_b}}\cos(2\pi f_c t + \pi + \theta_c)$$

$$= -\sqrt{\frac{2E_b}{T_b}}\cos(2\pi f_c t + \theta_c) \quad 0 \le t \le T_b \text{ (binary 0)}$$

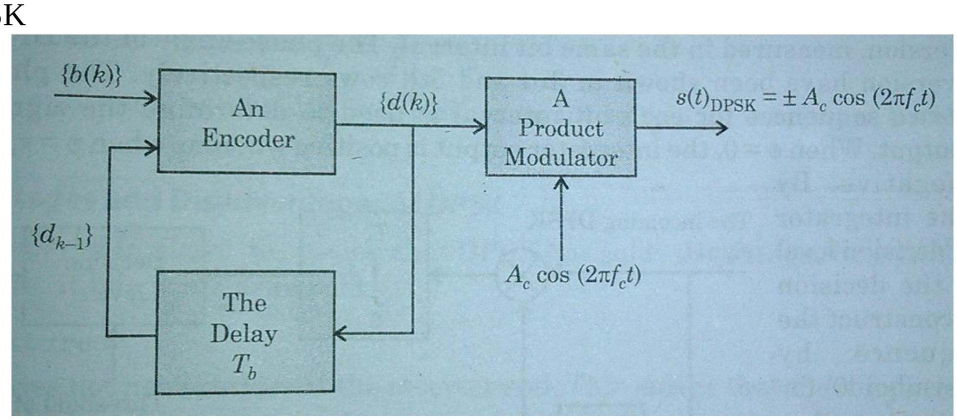


BPSK Transmitter

- BPSK uses coherent or synchronous demodulation
- Require phase and frequency information of the carrier at the receiver
- If pilot carrier signal transmitted along with the BPSK signal phase and frequency can be recovered at the receiver using PLL
- If not then squaring loop may be used to synthesize phase and frequency information of the carrier at the receiver.

Differential Phase shift Keying (DPSK)

- In order to eliminate the need for phase synchronization of coherent receiver with PSK, a differential encoding system can be used with PSK
- The digital information content of the binary data is encoded interms of signal transitions
- The symbol 0 may be used to represent transition in agiven binary sequence with respect to the previous encoded bit while symbol 1 to indicate no transition
- This new signalling technique which combines differential encoding with PSK is known as DPSK

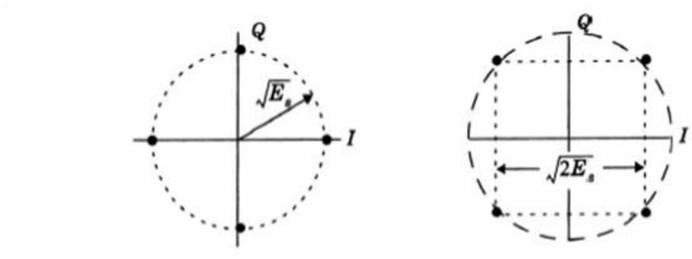


Quadrature Phase Shift Keying (QPSK)

In communication systems, we have two main resources, transmission power and channel bandwidth. Since channel bandwidth depends upon information bit rate, if two or more bits are combined into a symbol then signalling rate will reduce and thus transmission channel bandwidth.

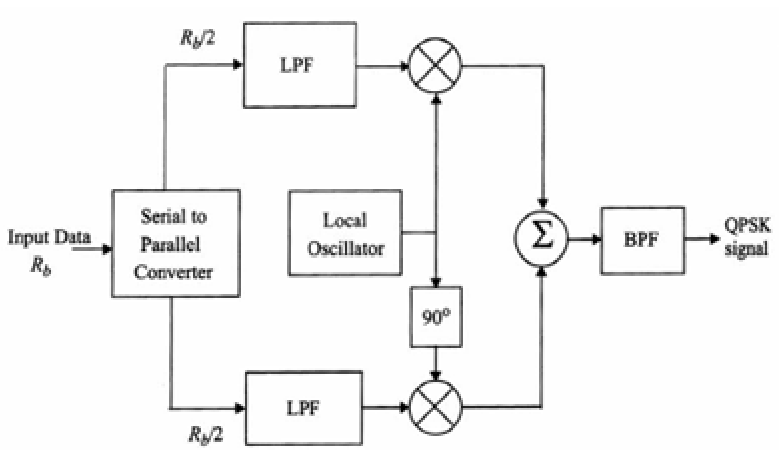
In QPSK two successive bits in the data sequence are grouped together. This reduces the bit rate or signalling rate and thus reduces the channel bandwith requirement.

$$S_{QPSK}(t)=\sqrt{\frac{2E_s}{T_s}}\cos(2\pi f_c t+(i-1)\frac{\pi}{2})\quad 0\le t\le T_s\quad i=1,2,3,4.$$



(a) (b)

- The input bit stream is split in to in-phase and quadrature phase while passing through serial to parallel converter
- The two binary streams are separately modulated by two carriers which are in quadrature

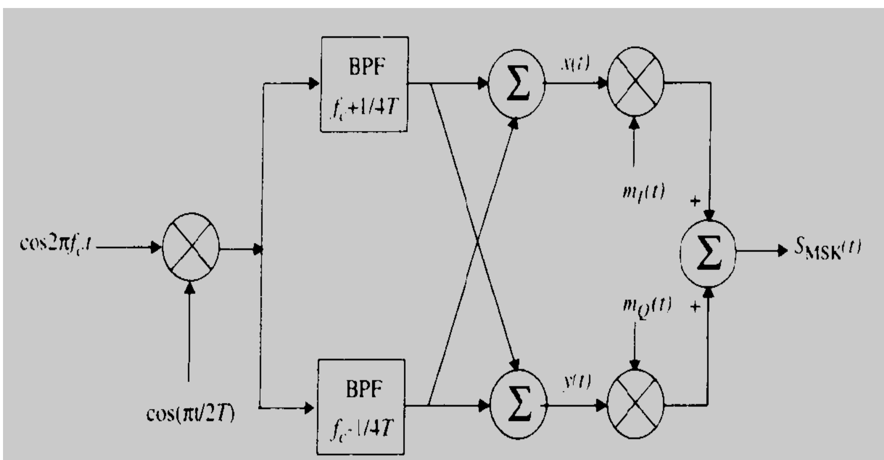


Minimum Shift Keying (MSK)

- MSK is a special type of continuous phase frequency shift keying
- Peak frequency deviation is 1/4th of bit rate i.e $\Delta F = \frac{R_b}{4}$
- Modulation index, $\mu = \frac{2\Delta F}{R_b}$, $i.e\ \mu = 0.5$
- A modulation index of 0.5 corresponds to the minimum frequency spacing that allows two FSK signals to be coherently orthogonal

$$\int_0^T v_H(t)v_L(t)dt = 0$$

MSK Transmitter



MSK for N-bit stream is given by:

$$S_{MSK} = m_I(t)p(t-2T_b)\cos 2\pi f_c t + m_Q(t)p(t-2T_b)\sin 2\pi f_c t$$

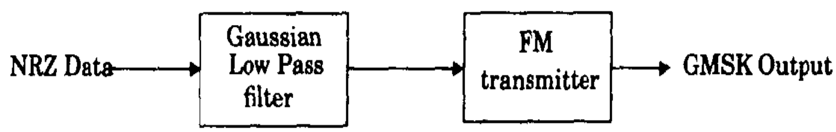
Where,

$$p(t) = \begin{Bmatrix} cos\left(\frac{\pi t}{2T_b}\right) & 0 \le t \le 2T_b \\ 0 & elsewhere \end{Bmatrix}$$

MSK is spectrally efficient modulation scheme and is particularly attractive for use in mobile radio communication systems. It possessesproperties such as constant envelope, spectral efficiency, good BER performance and self synchronizing capability

Gaussian Minimum Shift Keying (GMSK)

- Simple binary modulation scheme which may be viewed as derivative of MSK
- Side lobe levels of the spectrum are further reduced by passing the modulating NRZ data waveform through a premodulation Gaussian pulse shaping filter
- Premodualtion gaussian filtering converts the full response message signal into a partial response scheme where each transmitted symbol spans several bit period
- The premodulation gaussian filtering introduces ISI in the transmitted signal, but it can be shown that the degradation is not severe if the 3-dB bandwidth-bit duration product (BT_b) is greater than 0.5
- GMSK sacrifices the irreducible error rate caused by partial response in exchange for extremely good spectral efficiency



Note:

BT_b is product of 3-dB bandwidth of gaussian pulse shaping filter and bit duration. As BT_b increases number of side lobes in power spectral density of GMSK increase and when BT_b tends to infinty th PSD is similar to MSK. When BT_b is close to zero side lobes vanishes but ISI is a major constraint.

Thus in GSM $BT_b = 0.3$ is taken as a compromise between bit error probability and bandwidth conservation.

Spread Spectrum Modulation Techniques

- Spread Spectrum techniques employ a transmission bandwidth that is several orders of magnitude greater than the minimum required signal bandwidth.
- Bandwidth inefficient for a single user
- Multiple users can use the bandwidth without interfering one another
- Signal is Pseudorandom in nature thus noise like property in comparison to information data
- The spreading waveform is pseudo-noiose code (PN sequence)
- PN sequence is binary sequence which appears random but are deterministic to the intended receiver
- Spread spectrum signals demodulated at the receiver through cross correlation with the locally generated version of the PN Sequence
- Cross correlation with the correct PN sequence despreads the SS signal and restores the modulated message in the same narrow band as the original data.
- Cross correlation with the undesired SS signal results in small wideband noise

Properties of Spread Spectrum

- Inherent interference rejection capability
- Frequency planning not required
- Resistance to Multipath fading
- Exploit multipath components to improve performance through RAKE receiver

Pseudo-noise (PN) Sequence

- Binary sequence with an autocorrelation that resembles, over a period, the autocorrelation of a random binary sequence.
- In order to protect the signal, the chip sequence code used is pseudo-random. It appears random, but is actually deterministic, so that the receiver can reconstruct the code for synchronous detection. This pseudo-random code is also called pseudo-noise (PN).

PN Codes

PN codes are maximum length Pseudorandom Binary Sequences

- **Pseudorandom:** Of, relating to, or being random numbers generated by a *deterministic* process.
- **Binary:** Takes on one of two values.
- **Maximum Length:** Maximum achievable period of a generated sequence (not arbitrary).

Properties:

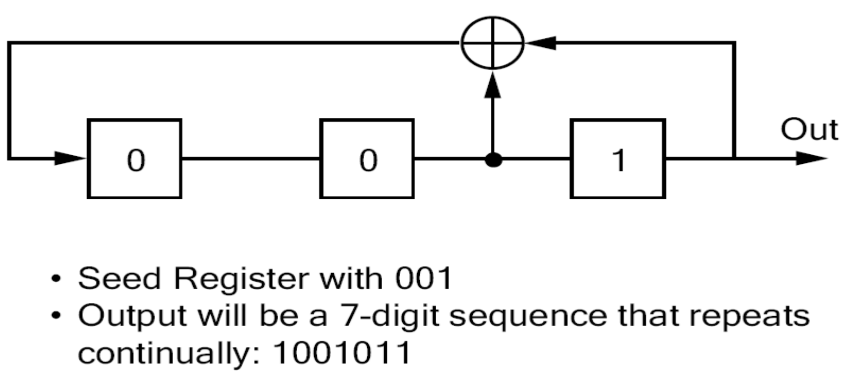
- **Balance**
- **Run-Length**
- **Shift and Add**
- **Autocorrelation**

Balance property: The output sequence will have an almost equal number of zeros and ones

Run-length property: In any period, half of the runs of consecutive zeros or ones are of length one, one-fourth are of length two, one-eighth are of length three, etc.

Shift and add property: The chip-by-chip sum of the output sequence C_k and any shift of itself C_{k+t} is a time-shifted version of the same sequence.

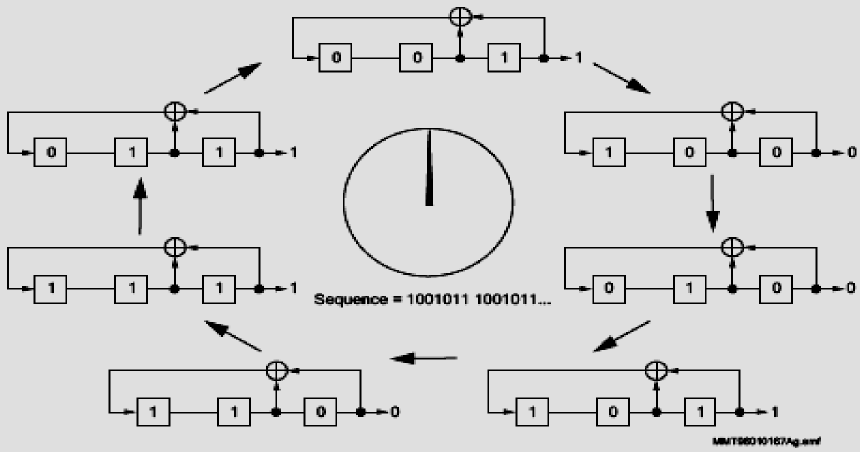
PN Code Generation



PN Code Generation

- PN codes are generated from prime polynomials using modulo 2 arithmetic.
- The state machines generating these codes are very simple and consist of shift registers and XOR gates.

- PN codes are maximum length. In general, if there are N shift registers (N = number of shift registers), the length of the PN code is equal to 2^N-1.
- In this example, the number of distinct states in the shift registers is $(2^3-1) = 7$.



Clock Pulse	D3	D2	D1
0	0	0	1
1	1	0	0
2	0	1	0
3	1	0	1
4	1	1	0
5	1	1	1
6	0	1	1
7	0	0	1

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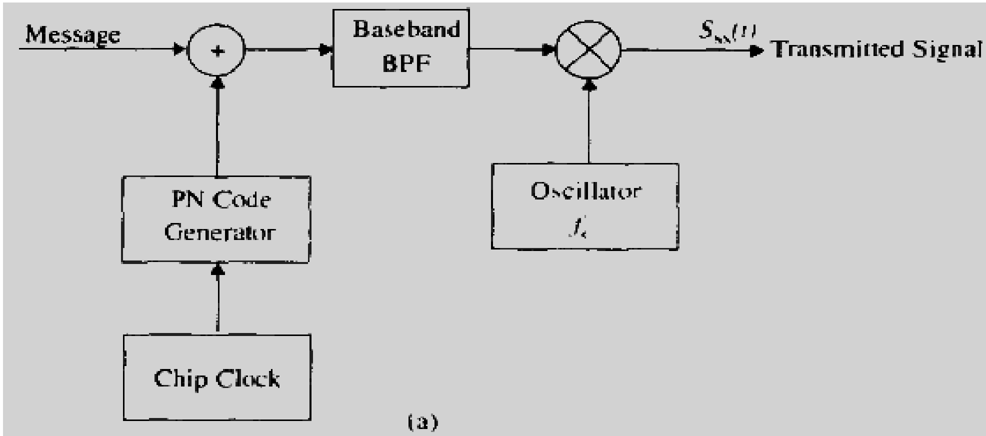
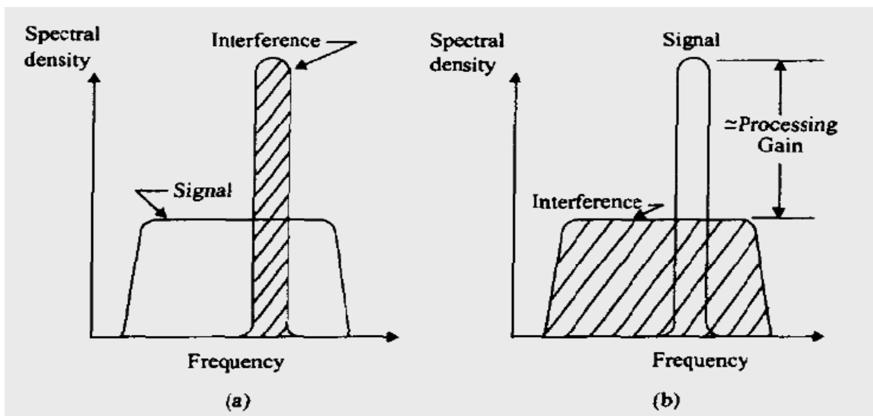
Direct Sequence SS

- Spreads the baseband data by directly multiplying the baseband data pulses with a pseudo-noise sequence that is produced by a PN code generator
- Synchronized data symbols, which may be information bit or binary channel code symbols, are added in modulo-2 fashion to the chips before being phase modulated



$$S_{ss}(t) = \sqrt{\frac{2E_s}{T_s} m(t)p(t)\cos(2\pi f_c t + \theta)}$$



a. Received signal at the o/p of wideband filter

b. Received signal after multiplying with the PN code

Processing Gain, $PG = \frac{W_{SS}}{B} = \frac{T_S}{T_C} = \frac{R_C}{R_S}$

The greater the processing gain of the system, the greater will be its ability to suppress in band interference

Frequency Hopping SS

- Periodic change of transmission frequency
- Modulation by time varying Pseudorandom carrier frequencies
- Set of carrier frequencies called *Hopset*
- Hopping occurs over a frequency band that includes a number of channels.

Each channel is a spectral region with central frequency in the Hopset and band width large enough to include most of the power in narrowband modulation burst having the corresponding carrier frequency

Instantaneous bandwidth:

The bandwidth of the channel used in the Hopset, B

Total hopping bandwidth:

The bandwidth of the spectrum over which the hopping occurs, B_{ss}

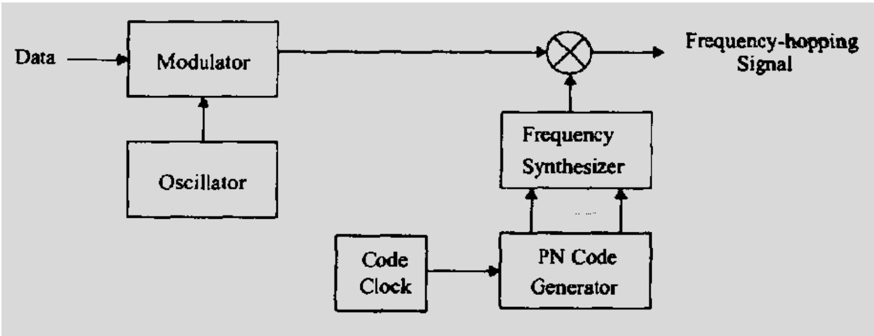
Processing Gain (PG) = B_{ss} / B (for FHSS)

Fast FH:

More than one frequency hop during each transmitted symbol

Slow FH:

One or more symbols transmitted in the time interval between frequency hops



(a) Transmitter

Data is sent by hopping the transmitter carrier to seemingly random channels which are known only to desired receiver.

On each channel small burst of data are sent using conventional narrowband modulation before the transmitter hops again

Equalization and Diversity Techniques

Need for equalization:

- Equalization is used for compensating intersymbol interference (ISI)
- An equalizer within a receiver compensates for the average range of expected channel amplitude and delay characteristics
- Equalizers must be adaptive as the channel is generally unknown and time varying

Fading Counter Measures:

Fading	Flat	Frequency Selective
Slow	**Diversity Coding Interleaving**	**Equalization**
Fast	**Diversity Coding Interleaving**	**Diversity Coding Interleaving**

Fundamentals of equalization;

1. ISI has been recognised as a major obstacle to high speed data transmission over mobile radio channels.
2. Equalization is a technique used to combat ISI.
3. As the mobile fading channels are random and time varying, equalizers must track the time varying characteristics of the mobile channel, and thus are called adaptive equalizers.

Adaptive Equalization

The operating modes of an adaptive equalizer are:

1. Training
2. Tracking

Training mode of adaptive equalization:

- Initially a known fixed length training sequence is sent by the ttransmitter so that the receiver’s equalizer may average to a proper setting.
- Training sequence is pseudorandom sequence or a fixed prescribed bit pattern
- Immediately following the training sequence user data is sent

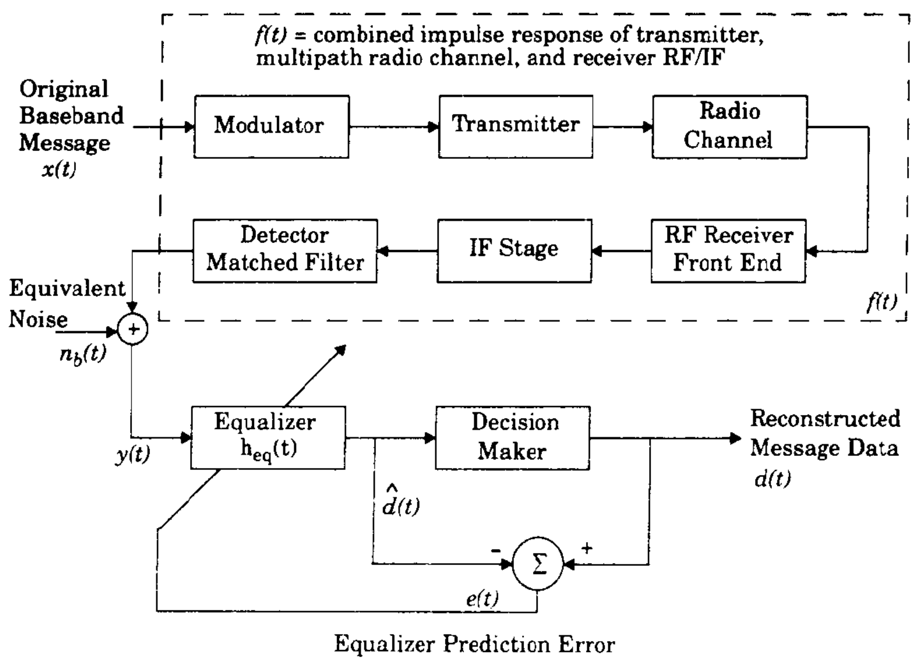
The training sequence:

- The training sequence is designated to permit an equalizer at the receiver to acquire the proper filter coefficients in the worst possible channel conditions (maximum delay, deepest fades, maximum ISI)
- Therefore when the training sequence is finished the filter coefficients are near optimal
- An adaptive equalizer at the receiver uses a recursive algorithm to evaluate the channel and estimate filter coefficients to compensate for the channel

Tracking Mode:

- When the data of the user are received the adaptive algorithm of the equalizer tracks the changing channel
- As a result of this the adaptive equalizer continuously changes the filter characteristics over time i.e weights of the filter changes over time
- Equalizers are widely used in TDMA system

Block diagram of adaptive equalizer:



f(t) = combined impulse response of transmitter, channel and receiver

e(t) = Equalizer prediction error

Working of the adaptive equalizer:

The signal received by the equalizer is given by

$$y(t) = x(t) \otimes f^*(t) + n_b(t)$$

Where,

$$x(t) = original\ signal$$

$$f(t) = combined\ impulse\ response$$

$$n_b(t) = baseband\ noise$$

If the impulse response of equalizer is heq (t), the output of equalizer is given by,

$$d(t) = x(t) \otimes f^*(t) \otimes h_{eq}(t) + n_b(t) \otimes h_{eq}(t)$$

$$d(t) = x(t) \otimes g(t) + n_b(t) \otimes h_{eq}(t)$$

The desired output is x(t) which is the original source data. Assuming that nb = 0 then, in order that d(t) = x(t)

$$g(t) = f^*(t) \otimes h_{eq}(t) = \delta(t)$$

The main goal of equalization is to satisfy the above equation. In frequency domain it is given by,

$$H_{eq}(f^*)\, F^*(-f) = 1$$

It implies that the equalizer is an inverse filter of the channel.

If the channel is frequency selective, the equalizer

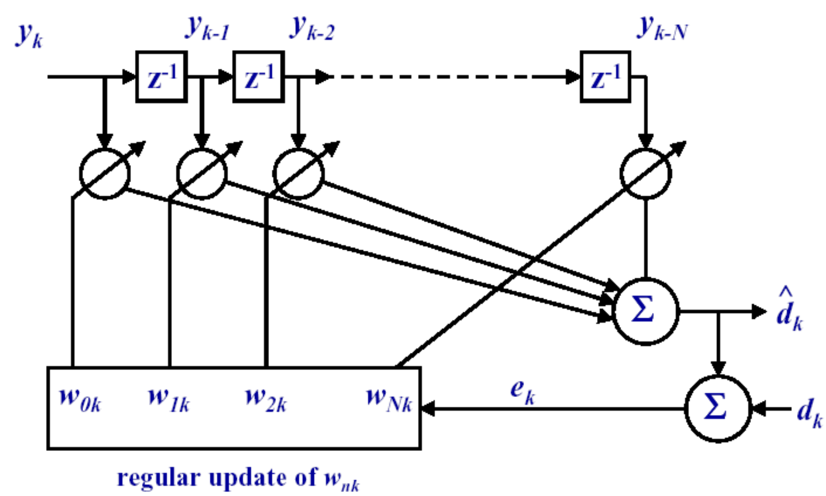
1. Enhances the frequency components with small amplitude and
2. Attenuates the strong frequencies in the received frequency spectrum

In order to provide,

1. Flat, composite, received frequency response and
2. A linear phase response

For a time varying channel, the equalizer is designed to track the channel variations so that the above equation is approximately satisfied.

A Generic Adaptive Equalizer



$\hat{d}_k$ = output of equalizer

d_k = x_k or represent a known property of the transmitted signal

Adaptive Equalizer

1. An adaptive equalizer is a time varying filter which must constantly be retuned
2. In the block diagram subscript k represent discrete time index
3. It can be seen from the block diadgram that there is a single input at anytime instant
4. The value of y_k depends upon the instantaneous state of adio channel and specific value of noise
5. The block diagam shown is called transversal filterand in this case has N delay elements, N+1 taps and N+1 tunable multiplierscalled weights
6. These weights have second subscript k to explicitely show that they vary with time and are updated on sample by sample basis.
7. The adaptive algorithm is controlled by the error signal e_k
8. The error signal is derived by comparing the output of the equalizer with some signal d_k which is either replica of transmitted signal x_k or which represents a known property of the transmitted signal
9. The adaptive algorithm uses e_k to minimize the cost function and uses the equalizer weights in such a manner that it minimizes the cost function iteratively
10. The Least Mean Square algorithm searches for the optimum or near optimum weights

Survey of equalization techniques

Equalization techniques can be classified into:

1. Linear
2. Non-Linear

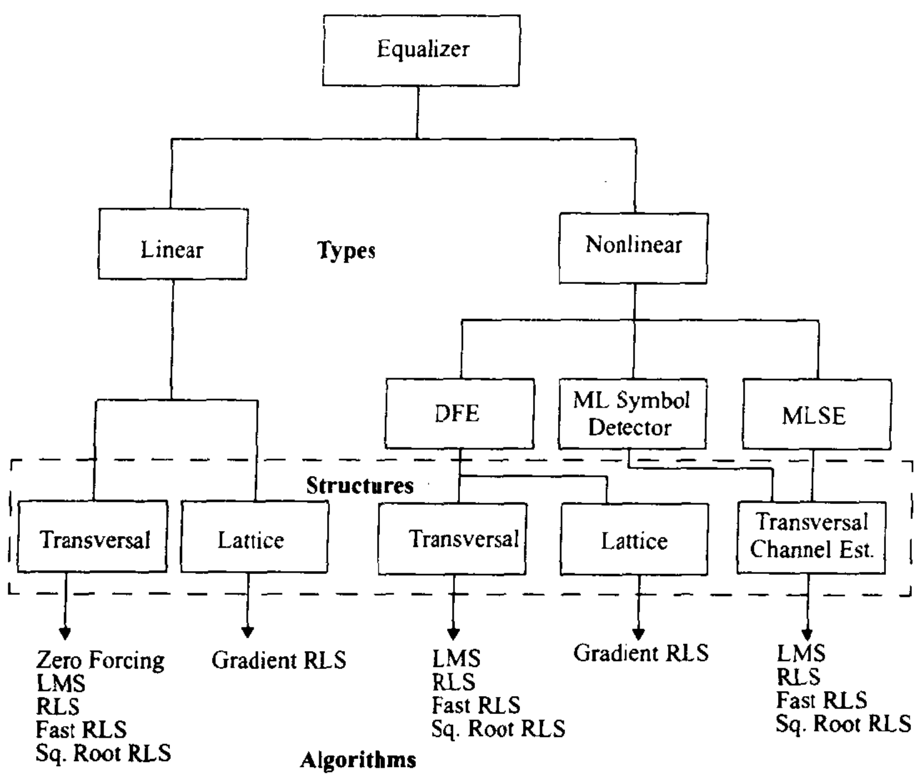
These are classified based on how the output of an adaptive equalizer is used for subsequent control of the equalizer.

In general, the decision maker determines the value of the digital data bit being received and applies thresholding operation.

In linear equalization the output signal d(t) is not used in feedback path to adapt the equalizer.

In non-linear equalization the output signal d(t) is feedback to change the subsequent output of the equalizer.

There are many filter structures that are used to implement linear and non-linear equalizers and numerous algorithms for each structure to adapt the equalizer.



Diversity Techniques

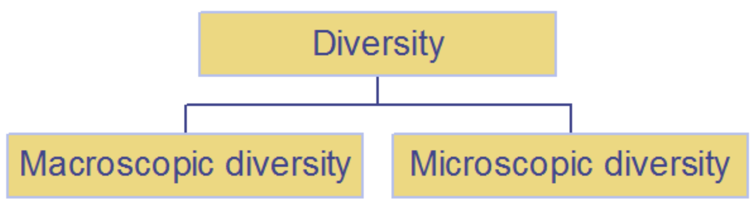
A *diversity scheme* is a method that is used to develop information from several signals transmitted over independent fading paths.

- Diversity exploits the random nature of radio propagation by finding independent (uncorrelated) signal paths for communication.
- Diversity is employed to reduce depth and duration of the fades experienced by a receiver in a flat fading (narrowband) channel
- No Training and Tracking (no training sequence)

Objective:

Combining multiple signals in such a fashion so as to reduce the effects of excessive deep fades in the received signal

Types:



<u>MACROSCOPIC DIVERSITY</u>

- Prevents Large Scale fading.
- Large Scale fading is caused by shadowing due to variation in both the terrain profile and the nature of the surroundings.
- Large Scale fading is log normally distributed signal.
- This fading is prevented by selecting an antenna which is not shadowed when others are, this allows increase in the signal-to-noise ratio.

MICROSCOPIC DIVERSITY

- Prevents Small Scale fading.
- Small Scale fading is caused by multiple reflections from the surroundings. It is characterized by deep and rapid amplitude fluctuations which occur as the mobile moves over distances of a few wavelengths.
- This fading is prevented by selecting an antenna which gives a strong signal that mitigates this small signal fading effect.

Commonly used diversity techniques are

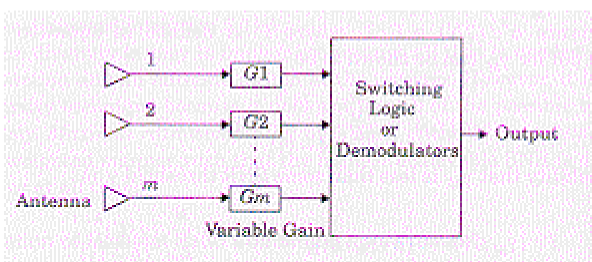
1. Space diversity
2. Frequency diversity
3. Polarization diversity
4. Time diversity

Space Diversity

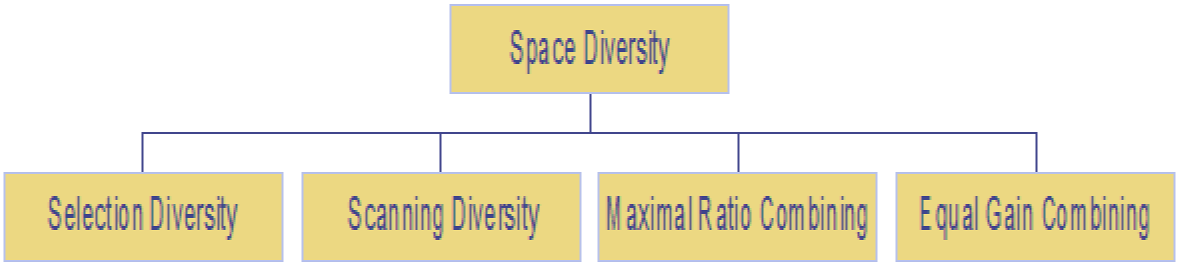
A method of transmission or reception, or both, in which the effects of fading are minimized by the simultaneous use of two or more physically separated antennas, ideally separated by one half or more wavelengths.

Signals received from spatially separated antennas on the mobile would have essentially uncorrelated envelopes for antenna separations of one half wavelengths or more.

Generalized block diagram of space diversity.

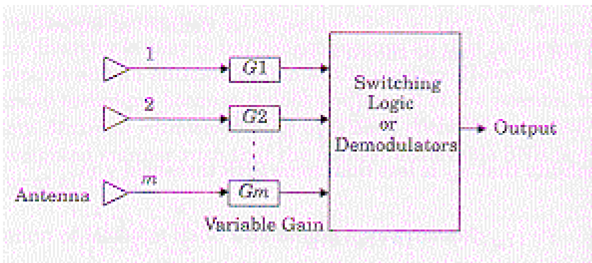


Space diversity reception methods:



Selection Diversity

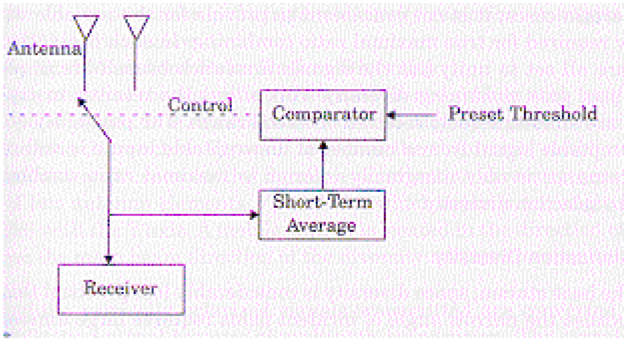
- *Selecting the best signal among all the signals received from different braches at the receiving end.*



- *In the diagram above m demodulators are used to provide m diversity branches*
- *The reciver branch having the heighest instantaneous SNR is connected to the demodulator*
- *Selection diversity offers an average improvement in the link margin without requiring additional transmitter power or sophisticated receiver circuitry.*
- *Selection diversity is easy to implement because all that is needed is a side monitoring station and an antenna switch at the receiver.*
- *However it is not an optimal diversity technique because it does not use all of the possible branches simultaneously.*
- *In practice the SNR is measured as (S+N)/N, since it is difficult to measure SNR.*

Feedback or Scanning Diversity

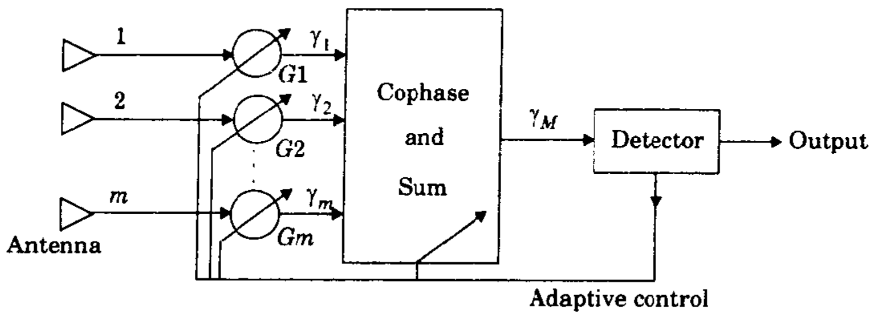
- *Scanning all the signals in a fixed sequence until the one with SNR more than a predetermined threshold is identified.*



- *Signal is then received continuously untill it falls below threshold from the branch having signal above predetermined threshold*
- *Scanning process is re-initiated one the signal falls below preset threshold*
- *This method is very simple to implement, requiring only one receiver.*
- *The resulting fading statistics are somewhat inferior to those obtained by the other methods.*

Maximal Ratio Combining

Combining all the signals in a co-phased and weighted manner so as to have the highest achievable SNR at the receiver at all times.



- Maximal ratio combining produces an output SNR equal to the sum of the individual SNRs
- Thus, it has the advantage of producing an output with an acceptable SNR even when non of the individual signals are themselves acceptable
- This technique gives the best statistical reduction of fading of any known linear diversity combiner

Equal Gain Combining

Combining all the signals in a co-phased manner with unity weights for all signal levels so as to have the highest achievable SNR at the receiver at all times.

- In certain cases it is not convenient to provide for the variable weighting capability.
- This allows the receiver to exploit signals that are simultaneously received on each branch.
- The probability of producing an acceptable signal from a number of unacceptable inputs is still retained.
- The performance is marginally inferior to maximal ratio combining and superior to selection Diversity.

Polarization Diversity

Effective Diversity is obtained with a Correlation Coefficient below 0.7

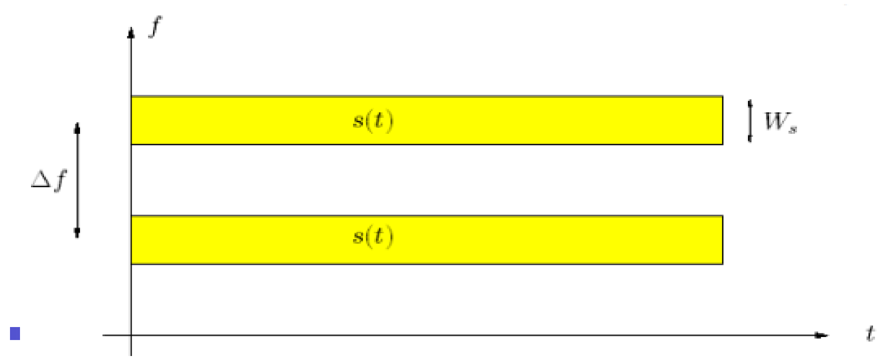
In order to keep the correlation at this level

- Space diversity at a base station requires antenna spacing of up to 20 wavelengths for the broadside case, and even more for the inline case.
- Polarization diversity at a base station does not require antenna spacing.
- The comparatively high cost of using space diversity at the base station prompts the consideration of using orthogonal polarization.
- Polarization diversity provides two diversity branches and allows the antenna elements to be considered.
- Antenna orientation to achieve orthogonal polarization are used
- The decorrelation for the signals in each polarization is caused by multiple reflections in the channel between the mobile and base station antennas
- Circular and linear polarized antennas have been used to characterize multipath inside buildings
- Dramatically reduce multipath delay spread

Frequency Diversity

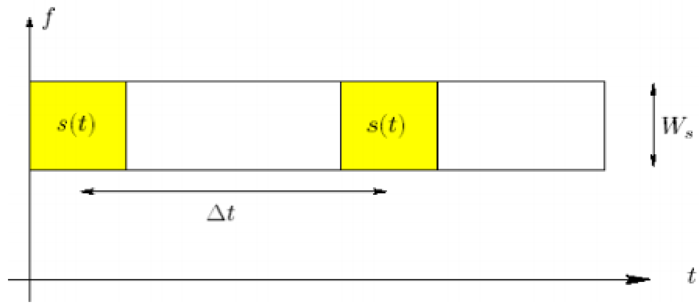
- Modulate the signal through M different carriers i.e transmits information on more than one carrier frequency
- The separation between the carriers should be at least the coherent bandwidth ($\Delta f = B_C$)
- Uncorrelated frequencies suffer different level of fading in frequency selective channels i.e different copies undergo independent fading
- Only one antenna is needed

The total transmitted power is split among the carriers. Often employed in microwave line of sight links



Time Diversity

- Transmit the desired signal in M different periods of time (each symbol is transmitted M times) i.e repeatedly transmits information at time spacing that exceeds the coherence time of the channel
- The interval between transmission of same symbol should be at least the coherence time ($\Delta t = T_C$)
- Different copies undergo independent fading
- Multiple repetition of the signal is received with independent fading conditions
- Reduction in efficiency (effective data rate < real data rate)
- RAKE receiver for spread spectrum CDMA is based on time diversity

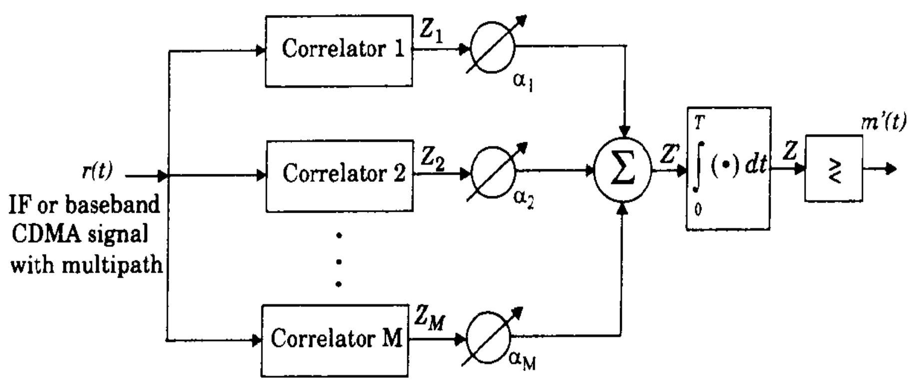


RAKE Receiver

- Multipath components of the signal carry important information
- Multipath components are time delayed versions of the original signal

RAKE receiver collects the time-shifted versions of the original signal by providing a separate correlation receiver for each of the multipath signals

Definition: A receiver technique which uses several baseband correlators to individually process several multipath components. The correlator outputs are combined to achieve improved communications reliability and performance.



Where,

$$Z' = \sum_{m=1}^{M} \alpha_m Z_m$$

$$\alpha_m = \frac{Z_m^2}{\sum_{m=1}^{M} Z_m^2}$$

- RAKE receiver utilizes multiple correlators to separately detect the M strongest multipath components

- Output of each correlator are weighted to provide a better estimate of the transmitted signal than is provided by a single component
- Weighting coefficients are based on the power or the SNR from each correlator output
- If the power or SNR is small from a correlator, small weighting factor will be assigned to it

Each correlator in a RAKE receiver is called a RAKE-receiver finger. The base station combines the outputs of its RAKE-receiver fingers non coherently. i.e., the outputs are added in power. The mobile receiver combines its RAKE-receiver finger outputs coherently, i.e., the outputs are added in voltage.

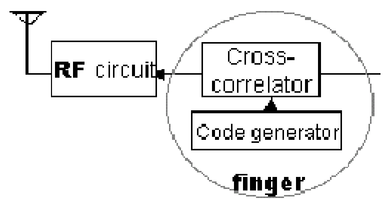


Fig: Rake Finger

The rake receiver consists of multiple correlators, in which the receive signal is multiplied by time-shifted versions of a locally generated code sequence. The intention is to separate signals such that each finger only sees signals coming in over a single (resolvable) path. The spreading code is chosen to have a very small autocorrelation value for any nonzero time offset. This avoids crosstalk between fingers. The rake receiver is designed to optimally detect a DS-CDMA signal transmitted over a dispersive multipath channel. It is an extension of the concept of the matched filter.

Interleaving

Interleaving is used to obtain time diversity in a digital communication system without adding overheads.

It is typical for many speech coders to produce several "important" bits in succession. These bits when pass through erroneous channels may face error in burst. In such cases, when bit errors occur in contiguous bits channel decoders are unable to correct it.

It is the function of interleaver to spread these bits out in time so that if there is a deep fade or noise burst, the important bits from a block of source data are not corrupted at the same time. By spreading the source bits over time, it becomes possible to make use of error control coding which protects the source data from corruption by the channel. Since error control codes are designed to protect against channel errors that may occur randomly.

Types:

1. Block Interleaver
    - Places the encoded data into a rectangular array of m rows and n columns
    - Each row consists of a word of source data having n bits
    - Source bits are placed into the interleaver by sequentially increasing the row number and filling the column
    - The data from the matrix is read out sequentially increasing the column number and transmitted over the channel
2. Convolutional Interleaver
    - Convolutional interleaver can be used in place of block interleaver in much the same fashion.
    - Ideally suited for use with convolutional codes

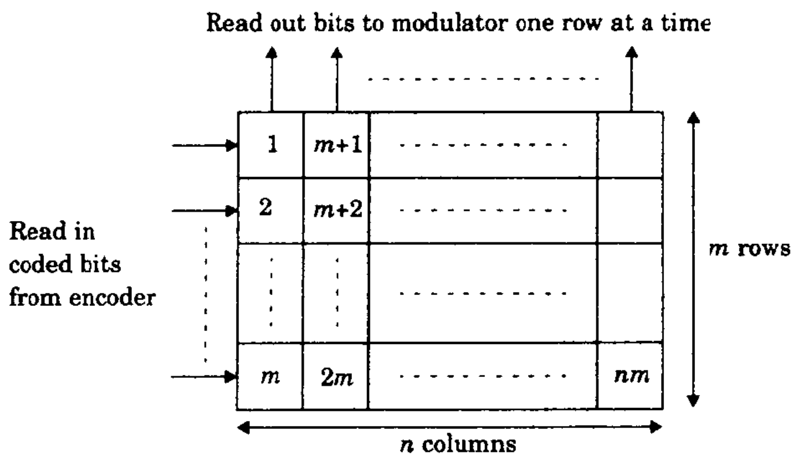


Fig: Block Interleaver

Speech and Channel Coding Fundamentals

Speech Coding:

The goal of speech coding is to transmit speech with highest possible quality using the least possible channel capacity.

Constraints of speech coding:

- Algorithmic complexity
- Communication delay
- Cost of implementation

Classification:

Based on approach of achieving compression speech coders are classified into following two types:

1. Waveform coders:
    a. Waveform coders strive to reproduce time waveform of speech signal as closely as possible
    b. Waveform coders are source independent
    c. Waveform coders have minimal design complexity
    d. Waveform coders have moderate economy in transmission bit rate

    Examples: PCM, DPCM, ADPCM etc.

2. Vocoders:
    a. Vocoders require a priori knowledge about the signal to be coded
    b. Vocoders are source dependent
    c. Vocoders are more complex in design than waveform coders
    d. Vocoders have highest economy in transmission bit rate

    Examples: CELP, RELP etc.

Characteristics of speech signals:

1. Probability Density Function (PDF)
    i. Very high probability of non-zero amplitudes
    ii. Significant probability of very high amplitudes
    iii. Monotonically decreasing function of amplitude between these extremes

2. Autocorrelation Function (ACF)
    i. In every sample speech there is a large component that is easily predicted from the value of the previous samples with a small random error
    E.g. DPCM, ADPCM, LPC

    ii. Autocorrelation function is a measure of similarity between samples of a speech signal as a function of their time separation

3. Power Spectral Density function (PSD)
    i. Non uniform distribution of power in speech signal among different frequency bands
    ii. High frequency component contribute very little to the total speech energy
    iii. However high frequency speech components are important carriers of speech information

Frequency domain coding of speech:

- Speech signal is divided into a set of frequency components which are quantitized encoded separately
- Different frequency bands are encoded preferentially according to some perceptual criteria for each band
- The quantization noise is limited to that particular band only

E.g. Sub band coding (SBC), Adaptive transfer coding (ATC)

Sub- band Coding:

- Speech is typically divided into 4 or 8 sub bands
- Bank of filters are used to divide speech
- Each sub band is sampled at Nyquist rate
- Sampled speech are encoded with different accuracy in accordance to perpetual criteria
- Used for bit rate ranging from 9.6 Kbps to 32 Kbps
- Advantageous for coding below 16 Kbps due to complexity and relative speech quality at low bit rates

Adaptive Transform Coding:

- ATC is frequency domain coding
- Encoding rate ranges from 9.6 Kbps to 20 Kbps
- Input signal segmented into blocks of window size
- Segments are represented by transform coefficients

- Each segment is quantized and transmitted separately
- At receiver quantized coefficients are inverse transformed to reconstruct speech
- Bits are allocated dynamically among frames while keeping bit rate constant

Vocoders

- ❖ Vocoders perform voice signal analysis by breaking down speech signal into small time segments
- ❖ Derive parameters from analysis
- ❖ Transmit parameters and sysnthesize at the receiver (Amplitude, pitch and poles)
- ❖ Vocoders work by modelling speech generation process

Types of Vocoder:

1. Channel vocoder
    a. Determine envelope of speech signal for number of frequency bands
    b. Sample, encode and multiplex voice, unvoice and pitch frequency

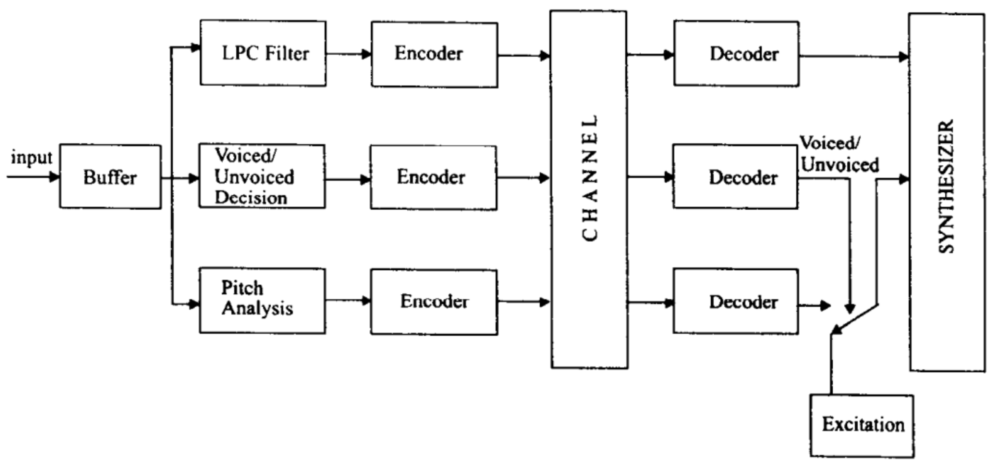
2. Formant vocoder
    a. Do not send sample of whole power spectrum envelope
    b. Transmit positions of peak (formants) of the spectral envelope
    c. Need atleast 3 formants to represent speech
    d. Can operate below 1.2 Kbps

3. Ceptrum vocoder
    a. Separates excitations from vocal tract spectrum
    b. High frequency: excitation coefficient and low frequency: spectral envelope
    c. Periodic pulse train is formed at multiples of sampling period

4. Voice excited vocoder
    a. Eliminate need for pitch extraction and voicing decision
    b. PCM transmission for low frequency band
    c. Channel vocoding for high frequency band
    d. Pitch is generated at the synthesizer
    e. Range: 7.2 Kbps to 9.6 Kbps
    f. Quality: Superior

Linear Predictive Coders

- Linear predictive coders belong to the time domain class of vocoders. This class of vocoders attempts to extract the significant features of speech from the time waveform.
- Transmit good quality voice at 4.8 Kbps
- The reduction principles used in LPC are similar to those in ADPCM coders, however instead of transmitting quantized value of the error signal representing the difference between the predicted and actual waveform, LPC system transmits only selected characteristics of the error signal.
- The parameters include the gain factor, pitch information and voiced/unvoiced decision information, which allow approximation of the correct error signal.



Multi-pulse Excited LPC

Excitation through single pulse per pitch period produces audible distortions at the receiver. Thus, typically 8 pulses per period are used with adjustable amplitude and pulse position. Results in better speech quality because of several pulses per pitch period and multi-pulse algorithm that does not require pitch detection.

Code Excited LPC

The coder and decoder have a predetermined code book of stochastic (zero-mean white Gaussian) excitation signals. For each speech signal the transmitter searches through its code book for the one that gives the best perpetual match to the sound when used as an excitation to the LPC filter.

The index of the code book where the best match was found is transmitted.

CDMA (IS-95) uses a variable rate CELP codec at 1.2 to 14.4 Kbps

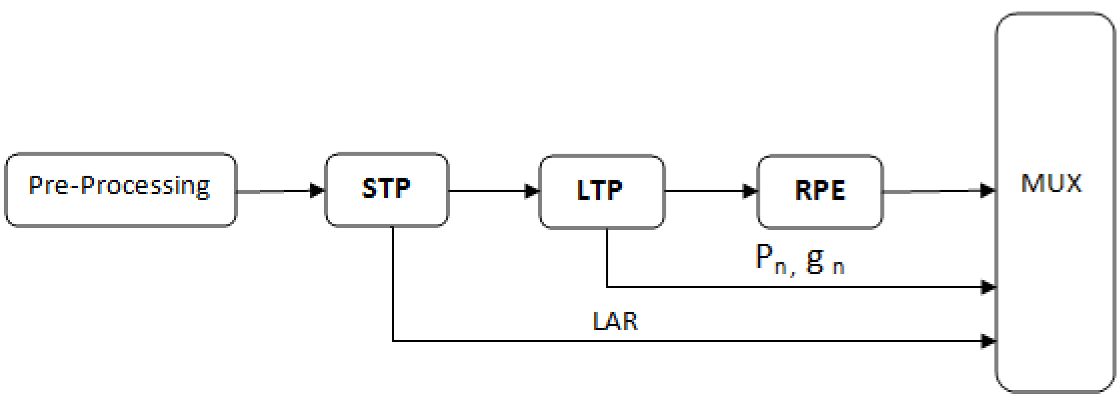
Residual Excited LPC

In this class of LPC coders, after estimating the model parameters (LP coefficients) and excitation parameters (voiced/unvoiced, pitch and gain) from a speech frame,

- the speech is synthesized at the transmitter and substracted from the original speech signal to form a residual signal
- residual signal is quantized, coded and transmitted to the receiver along with LPC model parameters
- at the receiver, the residual error signal is added to the signal generated using the model parameters to synthesize an approximation of the original speech signal

GSM Codec:

The analog speech signal at the transmitter is samapled at a rate of 8000 samples/s, and the samples are quantized with a resolution of 13 bits. This corresponds to a bit rate of 104 kbps for the speech signal. At the input to the speech codec, a speech fram containing 160 samples of 13 bits arrive every 20 ms. The speech codec compresses this speech signal into a source coded speech signal of 260 bit blocks at a bit rate of 13 Kbps. Thus the GSM speech coder achieves a compression ratio of 1 to 8.



The speech compression takes place in the speech coder. The GSM speech coder uses a procedure known as ***Regular Pulse Excitation - Long Term Prediction*** (RPE-LTP) Linear Predictive Coder. This procedure belongs to the family of hybrid speech coders. This hybrid procedure transmits part of the speech signal as the amplitude of a signal envelope, a pure waveform encoding, whereas the remaining part is encoded into a set of parameters. Examples of envelope encoding are PCM and ADPCM. A pure vocoder procedure is LPC.

A simplified block diagram of the RPE-LTP coder is shown in the figure above.

1. Speech data generated with a sampling rate of 8000 samples/s and 13 bit resolution arrive in blocks of 160 samples at the input of the coder
2. The speech signal is then decomposed into three components:
    a. A set of parameters for the adjustment of the short-term-analysis filter (STP) also called reflection coefficients
    b. An excitation signalfor the RPE part with the irrelevant portions removed and highly compressed
    c. A set of parameters for the control of the LTP long-term-analysis filter.
3. The STP and LTP analyses supply 36 filter parameters for each sample block
4. The RPE coding compresses the sample block to 188 bits of RPE parameters

A. Reflection Coefficients	**36 bit/20 ms**
B. RPE Parameter	**188 bit/20 ms**
C. LTP Parameter	**36 bit/20 ms**

This results in the generation of a frame of 260 bits every 20 ms, equivalent to a 13 Kbps GSM speech signal rate.

The speech data preprocessing of the coder removes the DC portion of the signal if present and uses a pre-emphasis filter to emphasize the higher frequencies of the speech spectrum. The preprocessed speech data is run through a nonrecursive lattice filter to reduce the dynamic range of the signal.

Since this filter has a memory of about 1 ms, it is called short-term prediction filter. The coefficients of this filter, called reflection coefficients, are calculated during LPC analysis and transmitted in a logarithmic representation as part of the speech frame, *Log area Ratios (*LARs).

Further processing of the speech data is preceded by a recalculation of the coefficients of tha long-term prediction filter. The new prediction is based on the previous and current blocks of speech data.

The resulting estimated block is finally subtracted from the block to be processed, and the resulting difference signal is passed on to the RPE coder.

After STP and LTP filtering the speech signal has been redundancy reduced. The irrelevance contained in the speech signal is reduced by the RPE coder. This irrelevance represents speech information that is not needed for the understandability of the speech signal, since it is hardly noticeable to human hearing and thus can be removed without loss of quality.

Review of Block Codes

Code Word: The code word is block of n encoded bits. It contains message bits and parity or redundant bits

Code Rate: The code rate is defined as the ratio of the number of message bits (k) to the total number of bits (n) in a code word.

$$code\ rate\ (r) = \frac{k}{n}$$

Hamming Distance: The hamming distance or simply distance between the two code words is defined as the number of positions in which their respective bits differ.

Code #1	11000101
Code #2	10011011

In the above two codes if we compare bits in their respective positions, there are 5 places where bits differ with each other. Thus, Hamming distance between these codes is 5.

Hamming Weight of a code word: The Hamming weight of a code word is the number all non-zero elements in the code word. It can also be defined as the Hamming distance of a code word with all zero code word. In the above example code #1 has Hamming weight 4 and code #2 has Hamming weight 5.

Code Efficiency: The code efficiency is defined as the ratio of message bits to the number of transmitted bits per block

Thus,

$$code\ efficiency = code\ rate\ (r) = \frac{k}{n}$$

Minimum Distance (d_{min}): The minimum distance, d_{min} , of a linear block code is defined as the smallest Hamming distance between any pair of code words the code.

Error detection and correction ability of a code based on d_{min}

A. To detect S errors in a code,

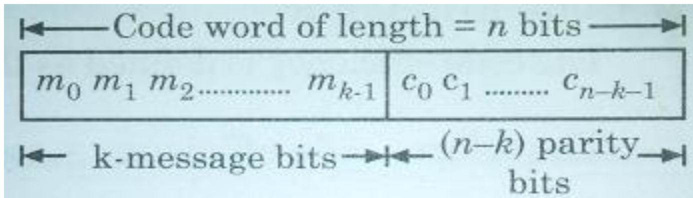
$$d_{min} \geq (S + 1)$$

B. To correct T errors in a code,

$$d_{min} \geq (2T + 1)$$

Linear Block Codes

- A linear block code (n,k) has n encoded bits out of which k bits are parity bits in every code word.



- These parity bits are computed from the message bits following prescribed encoding rule
- A block code generator generates the parity bits required to be added to the message bits to build the code word

X = MG

Where, X = Code vector of 1 x n size

M = Message vector of 1 x k size

G = Generator matrix of k x n size

- The generator matrix is generally represented as:

$[G] = [I_k \mid P]$

Where, I_k = Identity matrix of k x k size

P = Coefficient matrix of k x (n-k)

Hamming Codes

Hamming codes are linear block codes. The family of (n, k) hamming codes for $q \geq 3$ is defined by the following expressions:

- Block length: $n = 2^q - 1$
- Number of message bits $k = 2^q - q - 1$
- Number of parity bits $(n - k) = q$

Where, $q \geq 3$, i.e. minimum number of parity bit 3

- The minimum distance, $d_{min} = 3$
- $code\ efficiency = code\ rate\ (r) = \frac{k}{n} = \frac{2^q - q - 1}{2^q - 1} = 1 - \frac{q}{2^q - 1}$

Hadamard Codes

- Hadamard codes are in the form of a matrix called Hadamard matrix. Each row of the matrix represents a code word.
- A hadamard matrix is an N x N matrix of 1s and 0s such that each row differs from any other row in exactly N/2 locations.
- One row contains all zeros with the remainder containing N/2 zeros and N/2 ones. The minimum distance, d_{min} = N/2 for these codes.
- For N =2, the Hadamard matrix A is

$$A = \begin{bmatrix} 0 & 0 \\ 0 & 1 \end{bmatrix}$$

Golay Codes

Golay codes are linear binary (23, 12) codes. This code is capable of correcting any combination of 3 or less than 3 random errors in the block of 23 bits. The number of message bits out of 23 is 12. This code has minimum distance of 7. Golay code is generated by one of the following two polinomials.

$$g_1 = 1 + p^2 + p^4 + p^5 + p^6 + p^{10} + p^{11}$$

Or,

$$g_2 = 1 + p + p^5 + p^6 + p^7 + p^9 + p^{11}$$

Cyclic Codes

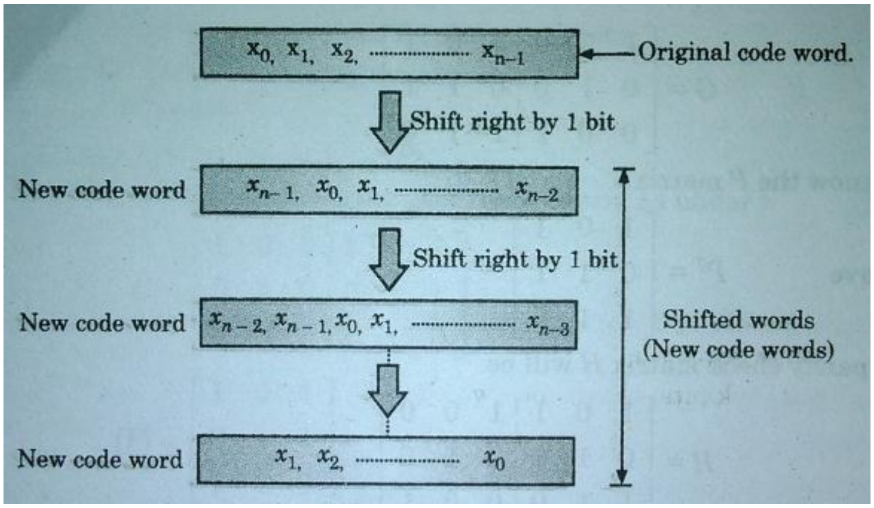
Cyclic codes are linear block codes. A binary code is said to be cyclic if it exhibits the following properties:

1. Linearity property

   A code is said to be linear if sum of any two code words also is a code word. This property states that the cyclic codes are linear block codes

2. Cyclic property

   A code is said to be cyclic if any cyclic shift of a code word results in the formation of another code word.



A cyclic code can be generated by using a generator polynomial g(p) of degree (n-k). The generator polynomial of an (n, k) cyclic code is a factor of p^n+1 and has the general form

$$g(p) = p^{n-k} + g_{n-k-1}p^{n-k-1} + \cdots + g_1 p + 1$$

BCH Codes

BCH codes are one of the most important and powerful linear block codes. The BCH codes are cyclic codes with a wide variety of parameters. The most common BCH codes are characterized as:

For any positive integer $m \geq 3$ and $t < \frac{2^m-1}{2}$ there exist a BCH code with the following parameters

- Block length $n = 2^m - 1$
- Number of message bits $k \geq n - mt$
- Minimum distance $d_{min} \geq 2t + 1$
- BCH codes can detect and correct upto t random errors per code word
- The Hamming code with single error correcting capability is a BCH code
- Block length and code rate of BCH codes are variable and thus provide high flexibility

Reed-Solomon Codes

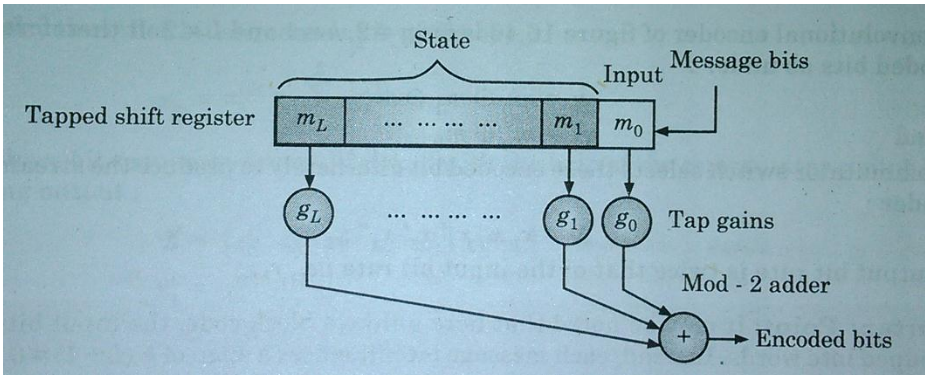
Reed-Solomon codes are non-binary codes which are capable of correcting errors which appear in burst and are commonly used in concatenated codeing systems.

- Block length $n = 2^m - 1$
- Parity symbols $n - k = 2e$ [for e error correction]
- Minimum distance $d_{min} = 2e + 1$
- RS codes achieve the largest possible d_{min} of any linear code.
- The most popular value of m is 8. RS code with m = 8 is extremely powerful
- This encoder operate on multiple bits rather than individual bits
- RS codes provide a wide range of code rates. The code rates can be chosen to obtain the optimum performance

Convolutional Codes

Convolutional codes are fundamentally different from block codes in that the information sequences are not grouped into distinct blocks and encoded. Instead a continuous sequence of information bits are mapped onto a continuous sequence of encoder output bits.

The encoder design of convolutional encoding is a tap shift register with (L+1) stages. g_0, g_1, g_2,… g_L are the tap gains which are nothing binary digits 0s or 1s. A tap gain of 0 represents open circuit whereas a tap gain of 1 represents a short circuit.



The message bits enter one by one into the tapped shift register, which are then combined by Mod-2 addition to form the encoded bit x

$$x = m_L g_L \oplus \ldots \oplus m_1 g_1 \oplus m_0 g_0$$

Or, $x = \sum_{i=0}^{L} m_i g_i$ [mod-2 addition]

From the equation it is clear that a encoded bit x depends on the current message bit m_0 and states of the shift register defined by the previous L message bits.

Coding Gain

The advantage of error control codes, whether they are block codes or convolutional codes, is that they provide a coding gain for the communication link.

- The coding gain describes how much better the decoded message performs as compared to the raw bit error performance of the coded transmission.
- Coding gain allows 1000 times improvement in bit error performance of channel decoding
- Each error control code has a particular coding gain
- Coding gain depends on
    - Code type
    - Decoder implementation and
    - Channel BER probability
- Coding gain measures the amount of additional SNR that would be required to provide the same BER performance for an uncoded message signal

$$G\ (dB) = \left(\frac{E_b}{N_0}\right)_u - \left(\frac{E_b}{N_0}\right)_c$$

Viterbi: Decoding Algorithm

The Viterbi algorithm operates on the principle of maximum likelyhood decoding and achieves optimum performance. The maximum likelyhood decoder has to combine the entire received sequence and find a valid path which has the smallest Hamming distancefrom the received sequence.

There are 2^N possible paths for a message sequence of N bits.

Viterbi algorithm applies the principle of maximum likelyhood to limit the comparisions of all the paths to surviving paths only

Identifying surviving path:

- Each path, from first bit of code trellis, is assigned a branch metric
- Metric is defined as Hamming distance of each branch with respect to received signal
- By summing the branch metrices path metric is obtained
- A survivor path at each node is identifird through path metric. That is out of multiple paths arriving at a node, the path with lowest path metric, is a survivor path

- Each node has one survivor path. Out of all survivor path one has the lowest path metric and that survivor path gives the encoded sequence at the decoder output

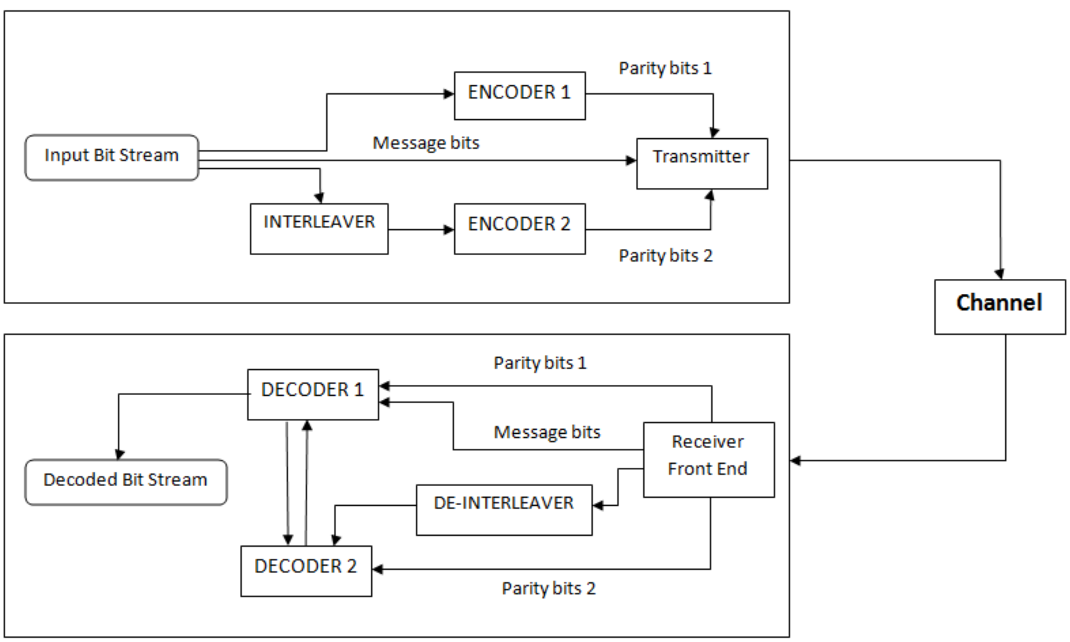
Trellis Coded Modulation

- Trellis coded modulation (TCM) is a technique which combines both coding and modulation to achieve significant coding gains without compromising bandwidth efficiency.
- TCM schemes employ redundant nonninary modulation in combination with a finite state encoder which decides the selection of modulation signal to generate coded signal sequence
- In the receiver, the signals are decoded through maximum likelyhood sequence decoder.
- Coding gain as large as 6 dB can be obtained without any bandwidth expansion or reduction in the effective information rate

Turbo codes:

Turbo codes are a new family of coding that combines the capabilities of convolutional codes with channel estimation theory, and can be thought of as nested or parallel convolutional codes.

Turbo codes allow coding gains which are far superior to all previous error correcting codes.



- Turbo codes use two encoders at the transmitter and two decoders at the receiver. With this devide-and-conquer approach, turbo codes outperform all previous error correcting codes

- Data bits enter the transmitter and are copied to ENCODER 1 and ENCODER 2. Before entering ENCODER 2 data bits are scrambled by the INTERLEAVER
- Each encoder generates a string of parity bits
- The original data bits plus the two strings of parity bits are combined into a single block and then sent over the channel
- The received analog signal is sampled and assigned integers indicating how likely it is that a bit is a 0 or a 1
- Each decoder takes the noisy data and respective parity information and computes how confident it is about each decoded bit. The two decoders exchange this confidence information repeatedly, and after a number of iterations, typically 4 to 10, they begin to agree on all decoded bits
- The decoded data is the sum of the noisy data plus the two final strings of confidence values. The output is converted back to binary bit stream.

Multiple Access Techniques for Wireless Communication

In wireless communication systems it is often desirable to allow the subscriber to send simultaneously information to the base station while receiving information from the base station. A cellular system divides any given area into cells where a mobile unit in each cell communicates with a base station. The main aim in the cellular system design is to be able to increase the capacity of the channel i.e. to handle as many calls as possible in a given bandwidth with a sufficient level of quality of service. There are several different ways to allow access to the channel. These include mainly the following:

1) Frequency Division Multiple Access (FDMA)
2) Time Division Multiple Access (TDMA)
3) Code Division Multiple Access (CDMA)
4) Space Division Multiple Access (SDMA)
5) Frequency Hopped Multiple Access (FHMA)

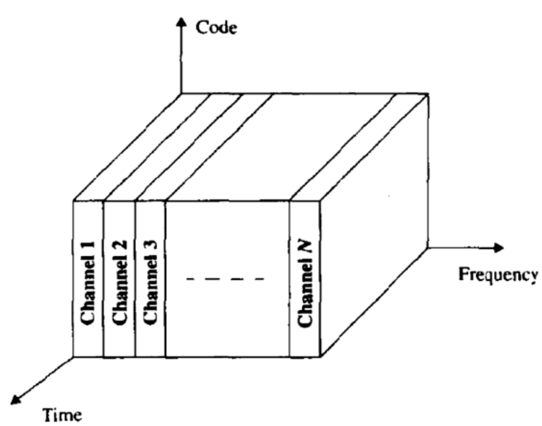
FDMA

This was the initial multiple-access technique for cellular systems in which each individual user is assigned a pair of frequencies while making or receiving a call. One frequency is used for downlink and one pair for uplink. This is called frequency division duplexing (FDD). That allocated frequency pair is not used in the same cell or adjacent cells during the call so as to reduce the co channel interference. Even though the user may not be talking, the spectrum cannot be reassigned as long as a call is in place.

The features of FDMA are as follows:

The FDMA channel carries only one phone circuit at a time. If an FDMA channel is not in use, then it sits idle and it cannot be used by other users to increase share capacity. After the assignment of the voice channel the BS and the MS transmit simultaneously and continuously.

The bandwidths of FDMA systems are generally narrow i.e. FDMA is usually implemented in a narrow band system The symbol time is large compared to the average delay spread. The complexity of the FDMA mobile systems is lower than that of TDMA mobile systems. FDMA requires tight _ltering to minimize the adjacent channel interference.



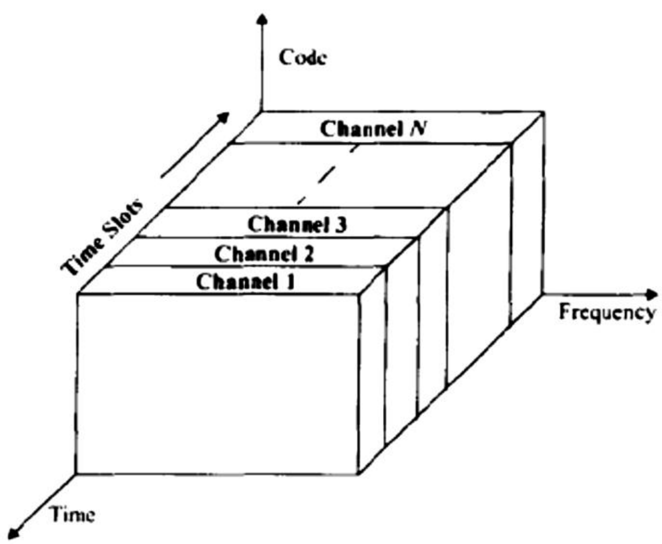
The number of channels that can be simultaneously supported in a FDMA system is given by

$$N = \frac{B_t - 2B_{guard}}{B_c}$$

Where, B_t is the total spectrum allocation, B_{guard} is the guard band allocated at the edge of the allocated spectrum, and B_c is the channel bandwidth.

TDMA

In digital systems, continuous transmission is not required because users do not use the allotted bandwidth all the time. In such cases, TDMA is a complimentary access technique to FDMA. Global Systems for Mobile communications (GSM) uses the TDMA technique. In TDMA, the entire bandwidth is available to the user but only for a FInite period of time. In most cases the available bandwidth is divided into fewer channels compared to FDMA and the users are allotted time slots during which they have the entire channel bandwidth at their disposal.



TDMA requires careful time synchronization since users share the bandwidth in the frequency domain. The number of channels are less, inter channel interference is almost negligible. If, TDMA uses different time slots for transmission and reception. This type of duplexing is referred to as Time division duplexing (TDD).

The features of TDMA include the following:

TDMA shares a single carrier frequency with several users where each users makes use of non overlapping time slots. The number of time slots per frame depends on several factors such as modulation technique, available bandwidth etc. Data transmission in TDMA is not continuous but occurs in bursts. This results in low battery consumption since the subscriber transmitter can be turned OFF when not in use. Because of a discontinuous transmission in TDMA the handoff process is much simpler for a subscriber unit, since it is able to listen to other base stations

during idle time slots. TDMA uses different time slots for transmission and reception thus duplexers are not required. TDMA
has an advantage that is possible to allocate different numbers of time slots per frame to different users. Thus bandwidth can be supplied on demand to different users by concatenating or reassigning time slot based on priority.

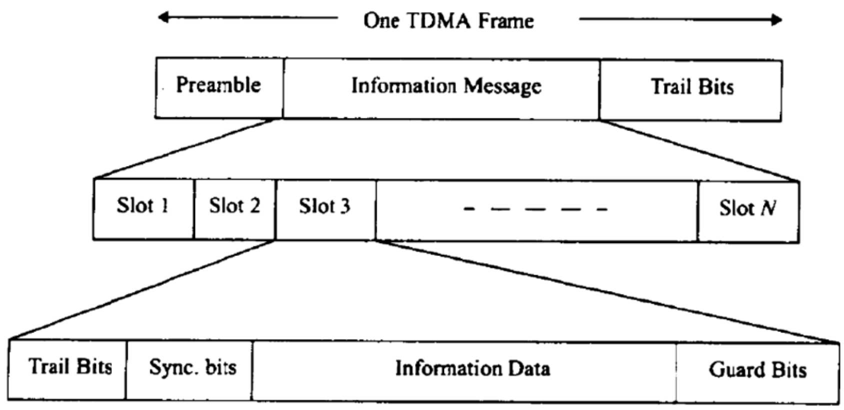


Fig: TDMA Frame Structure

Efficiency of TDMA

The efficiency of a TDMA system is a measure of the percentage of transmitted data that contains information as opposed to providing overhead for the access scheme. The frame efficiency, η_f, is the percentage of bits per frame which contains transmitted data.

$$\eta_f = \left(1 - \frac{b_{OH}}{b_T}\right) \times 100\%$$

Numbers of channels in TDMA system

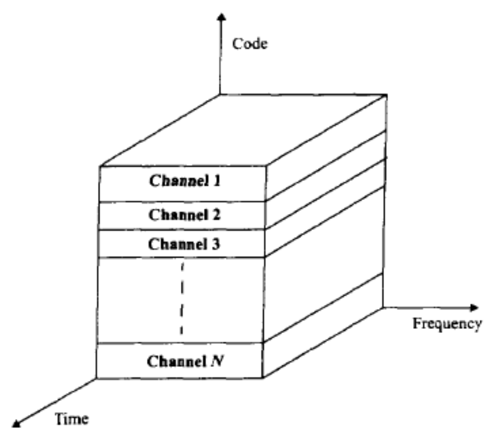
$$N = \frac{m\,(B_{tot} - 2B_{guard})}{B_c}$$

Where, m is the maximum number of TDMA users supported on each radio channel.

CDMA

- Users of a CDMA system share the same frequency. TDD or FDD may be used for duplexing.
- Unlike TDMA or FDMA, CDMA has soft capacity limit. Increasing the number of users in a CDMA system raises the noise floor in a linear fashion. Thus there is no absolute limit on the number of users in CDMA.

- Baseband signal is spreaded over a large spectrum. Spread spectrum reduces multipath fading.
- RAKE receiver is used to improve reception by collecting time delayed versions of the required signal.
- CDMA suffers from Self-jamming. Self-jaming arises from the fact that the spreading sequences of different users are not exactly orthogonal, hence in the despreading of a particular PN code, non-zero contributions to the receiver decision statistic for a desired user arise from the transmission of other users in the system.
- Near-far problem occurs at a CDMA receiver if an undesired user has a high detected power as compared to the desired user. Power control mechanism is used to counter near-far effect.

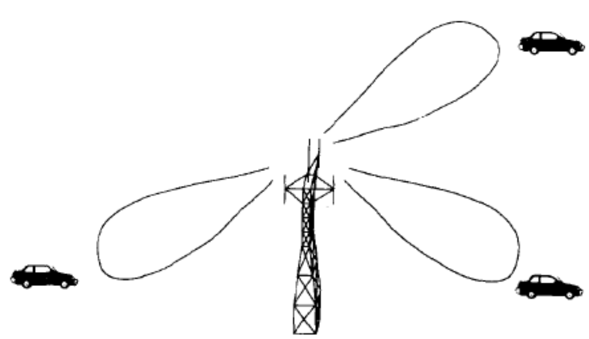


SDMA

Space division multiple access controls the radiated energy for each user in space. SDMA serves different users by using spot beam antennas. Spot beams may be formed on same or different frequency.

To implement SDMA, antenna forming spot beam should:

1. have infinite tracking speed and
2. have infinitesimal beam width



FHMA

In frequency hopped multiple access the carrier frequency of the individual users are varied in a pseudorandom fashion within a wideband channel.

The digital data is broken into uniform sized bursts which are transmitted on different carrier frequencies.

The pseudorandom change of the carrier frequencies of the user randomizes the occupancy of a specific channel at any given time, thereby allowing for multiple access over a wide range of frequencies.

Types:

1. *Fast frequency hopping system*: if the rate change of the carrier frequency is greater than the symbol rate then the system is referred to as a fast frequency hopping system.
2. *Slow frequency hopping system*: if the rate change of the carrier frequency is less than or equal to the symbol rate then the system is referred to as a fast frequency hopping system.

A frequency hopped system provides a level of security, especially when a large number of channels are used, since an unintended receiver that does not know the pseudorandom sequence frequency slots must retune rapidly to search for the signal it wishes to intercept.

Wireless System and Standards

GSM

Global System for Mobile Communication is a second generation cellular system standard that was developed to solve the fragmentation problems of the first generation cellular system. GSM is the world's first cellular system to specify digital modulation and network level architectures and services.

Before GSM networks there were public mobile radio networks (cellular). They normally used analog technologies, which varied from country to country and from manufacturer to another. These analog networks did not comply with any uniform standard. There was no way to use a single mobile phone from one country to another. The speech quality in most networks was not satisfactory. GSM became popular very quickly because it provided improved speech quality and, through a uniform international standard, made it possible to use a single telephone number and mobile unit around the world. The European Telecommunications Standardization Institute (ETSI) adopted the GSM standard in 1991.

The benefits of GSM include:

1. Support for international roaming
2. Distinction between user and device identification
3. Excellent speech quality
4. Wide range of services
5. Interworking (e.g. with ISDN, DECT)
6. Extensive security features

GSM also stands out from other technologies with its wide range of services:

1. Telephony
2. Asynchronous and synchronous data services (2.4/4.8/9.6 kbit/s)
3. Access to packet data network (X.25)
4. Telematic services (SMS, fax, videotext, etc.)
5. Many value-added features (call forwarding, caller ID, voice mailbox)
6. E-mail and Internet connections

GSM System Architecture

The best way to create a manageable communications system is to divide it into various subgroups that are interconnected using standardized interfaces. A GSM network can be divided into three groups:

1. Mobile Station (MS),
2. Base Station Subsystem (BSS) and
3. Network Subsystem (NSS)

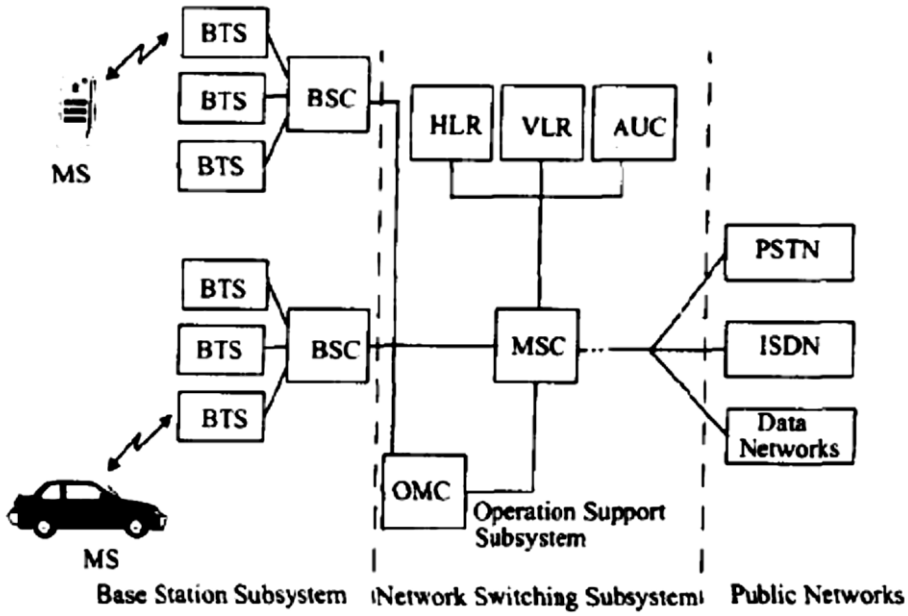


Fig: GSM Architecture

The Mobile Station (MS)

A mobile station may be referred to as a ªhandsetº, a ªmobileº, a ªportable terminalº or ªmobile equipmentº ME). It also includes a subscriber identity module (SIM) that is normally removable and comes in two sizes. Each SIM card has a unique identification number called IMSI (international mobile subscriber identity). In addition, each MS is assigned a unique hardware identification called IMEI (international mobile equipment identity). In some of the newer applications (data communications in particular), an MS can also be a terminal that acts as a GSM interface, e.g. for a laptop computer. In this new application the MS does not look like a normal GSM telephone. The seemingly low price of a mobile phone can give the (false) impression that the product is not of high quality. Besides providing a transceiver (TRX) for transmission and reception of voice and data, the mobile also performs a number of very demanding tasks such as authentication, handover, encoding and channel encoding.

The Base Station Subsystem (BSS)

The base station subsystem (BSS) is made up of the base station controller (BSC) and the base transceiver station (BTS).

The base transceiver station (BTS): GSM uses a series of radio transmitters called BTSs to connect the mobiles to a cellular network. Their tasks include channel coding/decoding and encryption/decryption. A BTS is comprised of radio transmitters and receivers, antennas, the interface to the PCM facility, etc. The BTS may contain one or more transceivers to provide the required call handling capacity. A cell site may be omnidirectional or split into typically three directional cells.

The base station controller (BSC): A group of BTSs are connected to a particular BSC which manages the radio resources for them. Today's new and intelligent BTSs have taken over many tasks that were previously handled by the BSCs. The primary function of the BSC is call maintenance. The mobile stations normally send a report of their received signal strength to the BSC every 480 ms. With this information the BSC decides to initiate handovers to other cells, change the BTS transmitter power, etc.

The Network Subsystem (NSS)

The mobile switching center (MSC): Acts like a standard exchange in a fixed network and additionally provide all the functionality needed to handle a mobile subscriber. The main functions are registration, authentication, location updating, handovers and call routing to a roaming subscriber. The signaling between functional entities (registers) in the network subsystem uses Signaling System 7 (SS7). If the MSC also has a gateway function for communicating with other networks, it is called Gateway MSC (GMSC).

The home location register (HLR): Adatabase used formanagement of mobile subscribers. It stores the international mobile subscriber identity (IMSI), mobile station ISDN number (MSISDN) and current visitor location register (VLR) address. The main information stored there concerns the location of each mobile station in order to be able to route calls to the mobile subscribers managed by each HLR. The HLR also maintains the services associated with each MS.One HLR can serve several MSCs.

The visitor location register (VLR): Contains the current location of the MS and selected administrative information from the HLR, necessary for call control and provision of the subscribed services, for each mobile currently located in the geographical area controlled by the VLR. A VLR is connected to one MSC and is normally integrated into the MSC's hardware.

The authentication center (AuC): A protected database that holds a copy of the secret key stored in each subscriber's SIM card, which is used for authentication and encryption over the radio channel. The AuC provides additional security against fraud. It is normally located close to each HLR within a GSM network.

The equipment identity register (EIR): The EIR is a database that contains a list of all valid mobile station equipment within the network, where each mobile station is identified by its international mobile equipment identity (IMEI). The EIR has three databases:

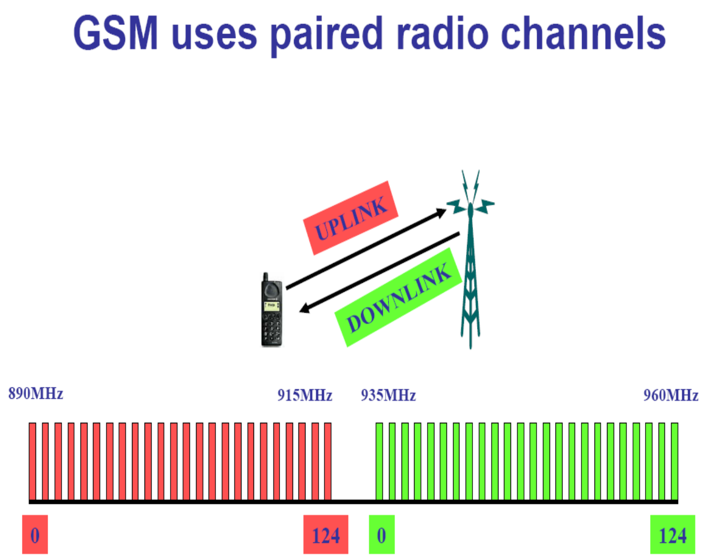
1. White list: for all known, good IMEIs
2. Black list: for bad or stolen handsets
3. Grey list: for handsets/IMEIs that are uncertain

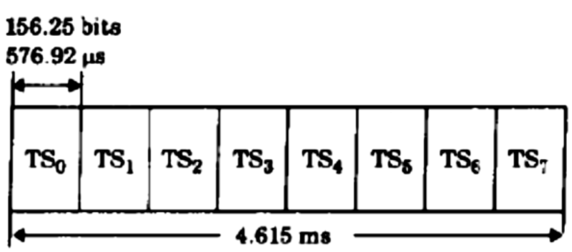
Operation and Maintenance Center (OMC)

The OMC is a management system that oversees the GSM functional blocks. The OMC assists the network operator in maintaining satisfactory operation of the GSM network. Hardware redundancy and intelligent error detection mechanisms help prevent network down-time. The OMC is responsible for controlling and maintaining the MSC, BSC and BTS. It can be in charge of an entire public land mobile network (PLMN) or just some parts of the PLMN.

GSM Radio Subsystem

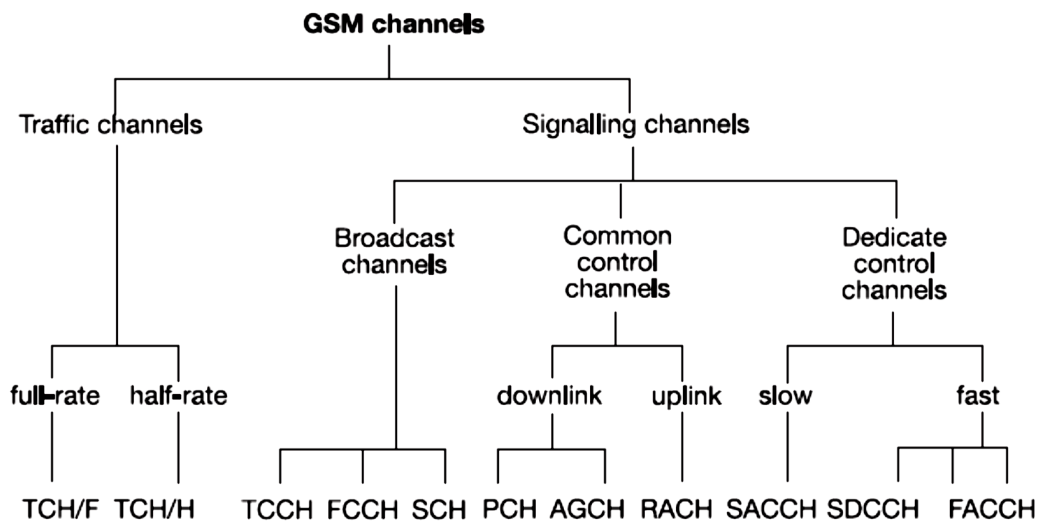
- 890-915 MHz band is used for susbscriber to base station transmission (Reverse Link) and 935-960 MHz band is used for base station to subscriber (MS) (Forward Link)
- The available frequency band is divided into 200 KHz channels called ARFCN (Absolute Radio Frequency Channel Number)
- The ARFCN channel pair is separated by 45 MHz and each channel is time shared between eight subscribers using TDMA.
- Radio transmissions are made at a channel data rate of 270.833 Kbps
- Each TS has an equivalent time allocation of 156.25 bits
- But 8.25 bits of guard time and six totaal start and stop bits are provided to prevent overlap
- Each TS has a time duration of 576.92 micro seconds and frame span of 4.615 milli seconds





GSM Channel Types

There are two types of GSM logical channels, called traffic channel (TCH) and control cahnnel (CCH). Traffic channels carry digitally encoded user speech or user data and control channels carry signalling and synchronizing commands between the base station and the mobile station. Control channels are also called signalling channels.



GSM divides up each ARFCN into 8 time slots. These 8 timeslots are further broken up into logical channels. Logical channels can be thought of as just different types of data that is transmitted only on certain frames in a certain timeslot. Different time slots will carry different logical channels, depending on the structure the BSS uses. There are two main categories of logical channels in GSM:

1. **Signaling/Control Channels**
2. **Traffic Channels (TCH)**

Signaling/Control Channels

These are the main types of signaling Channels:

Broadcast Channels (BCH) - Transmitted by the BTS to the MS. This channel carries system parameters needed to identify the network, synchronize time and frequency with the network, and gain access to the network.

Common Control Channels (CCH) - Used for signaling between the BTS and the MS and to request and grant access to the network.

Standalone Dedicated Control Channels (SDCCH) - Used for call setup

Associated Control Channels (ACCH) - Used for signaling associated with calls and call-setup. An ACCH is always allocated in conjunction with a TCH or a SDCCH.

The above categories can be divided into the following logical channels:

Broadcast Channels (BCH)

1. Broadcast Control Channel (BCCH)
2. Frequency Correction Channel (FCCH)
3. Synchronization Channel (SCH)
4. Cell Broadcast Channel (CBCH)

Common Control Channels (CCCH)

1. Paging Channel (PCH)
2. Random Access Channel (RACH)
3. Access Grant Channel (AGCH)

Standalone Dedicated Control Channel (SDCCH)

1. Associated Control Channel (ACCH)
2. Fast Associated Control Channel (FACCH)
3. Slow Associated Control Channel (SACCH)

Broadcast Channels (BCH)

Broadcast Control Channel (BCCH) - DOWNLINK - This channel contains system parameters needed to identify the network and gain access to the network. These parameters include the Location Area Code (LAC), the Mobile Network Code (MNC), the frequencies of neighboring cells, and access parameters.

Frequency Correction Channel (FCCH) - DOWNLINK - This channel is used by the MS as a frequency reference. This channel contains frequency correction bursts.

Synchronization Channel (SCH) - DOWNLINK - This channel is used by the MS to learn the Base Station Information Code (BSIC) as well as the TDMA frame number (FN). This lets the MS know what TDMA frame they are on within the hyper-frame.

Cell Broadcast Channel (CBCH) - DOWNLINK - This channel is not truly its own type of logical channel. The CBCH is for *point-to-omnipoint* messages. It is used to broadcast specific information to network subscribers; such as weather, traffic, sports, stocks, etc. Messages can be of any nature depending on what service is provided. Messages are normally public service type messages or announcements. The CBCH isn't allocated a slot for itself, it is assigned to an SDCCH. It only occurs on the downlink. The CBCH usually occupies the second sub slot of the SDCCH. The mobile will not acknowledge any of the messages.

Common Control Channels (CCCH)

Paging Channel (PCH) - DOWNLINK - This channel is used to inform the MS that it has incoming traffic. The traffic could be a voice call, SMS, or some other form of traffic.

Random Access Channel (RACH) - UPLINK This channel is used by a MS to request an initial dedicated channel from the BTS. This would be the first transmission made by a MS to access the network and request radio resources. The MS sends an *Access Burst* on this channel in order to request access.

Access Grant Channel (AGCH) - DOWNLINK - This channel is used by a BTS to notify the MS of the assignment of an initial SDCCH for initial signaling.

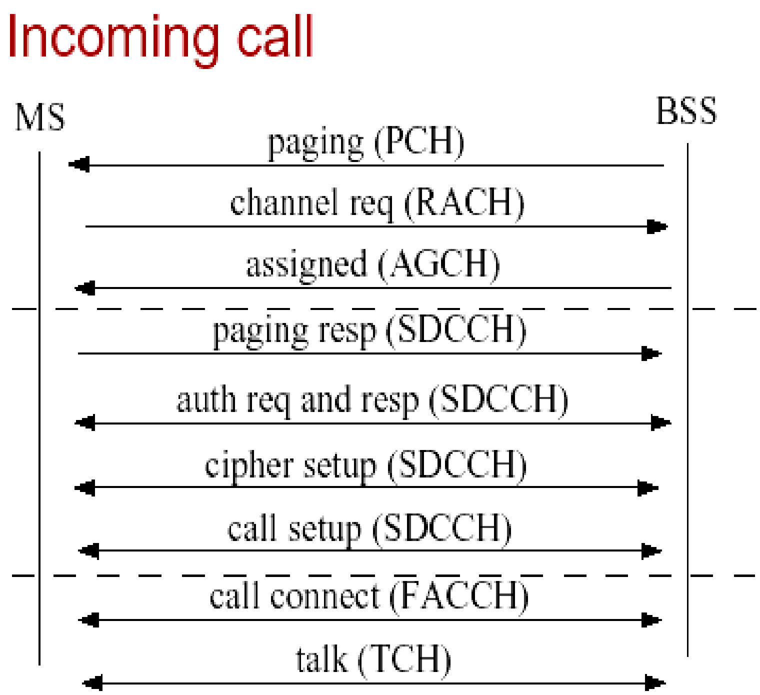
Standalone Dedicated Control Channel (SDCCH) - UPLINK/DOWNLINK - This channel is used for signaling and call setup between the MS and the BTS.

Associated Control Channels (ACCH)

Fast Associated Control Channel (FACCH) - UPLINK/DOWNLINK - This channel is used for control requirements such as handoffs. There is no TS and frame allocation dedicated to a FAACH. The FAACH is a burst-stealing channel, it steals a Timeslot from a Traffic Channel (TCH).

Slow Associated Control Channel (SACCH) - UPLINK/DOWNLINK - This channel is a continuous stream channel that is used for control and supervisory signals associated with the traffic channels.

Example of GSM Call



For outgoing call paging procedure do not take place and other procedure remains unchanged.

Frame Structure for GSM

In GSM frequency band of 25 MHz is divided into 200 KHz of smaller bands, each carry one RF carrier, this gives 125 carriers. As one carrier is used as guard channel between GSM and other frequency bands 124 carriers are useful RF channels. This division of frequency pool is called FDMA. Now each RF carrier will have eight time slots. This division time wise is called TDMA. Here each RF carrier frequency is shared between 8 users hence in GSM system, the basic radio resource is a time slot with duration of about 577 micro second. As mentioned each time slot has 15/26 or 0.577ms of time duration. This time slot carries 156.25 bits which leads to bit rate of 270.833 kbps.

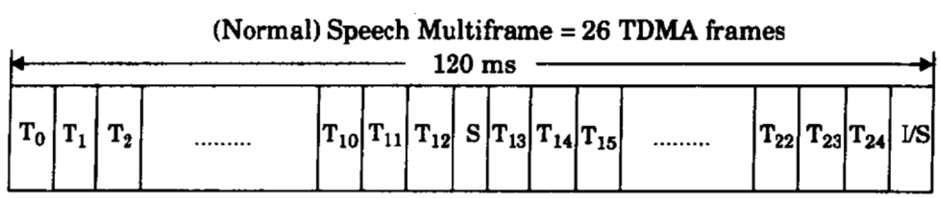
The GSM frame structure is designated as hyperframe, superframe, multiframe and frame. The minimum unit being frame (or TDMA frame) is made of 8 time slots.

One GSM hyperframe composed of 2048 superframes. Each GSM superframe composed of multiframes (either 26 or 51). Each GSM multiframe composed of frames (either 51 or 26 based on multiframe type). Each frame composed of 8 time slots. Hence there will be total of 2715648 TDMA frames available in GSM and the same cycle continues.There are two variants to multiframe structure.

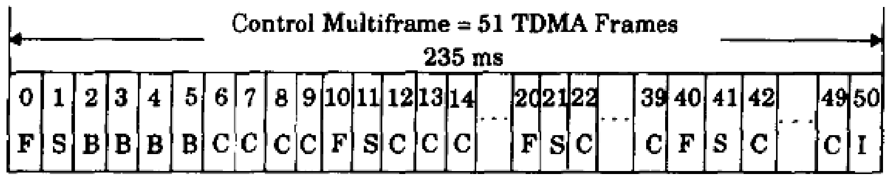
(1) 26 frame multiframe - Called traffic multiframe, composed of 26 bursts in a duration of 120ms, out of these 24 are used for traffic, one for SACCH and one is not used.

(2) 51 frame multiframe- Called control multiframe, composed of 51 bursts in a duration of 235.365 ms.

This type of multiframe is divided into logical channels. These logical channels are time sheduled by BTS. Always occur at beacon frequency in time slot 0.

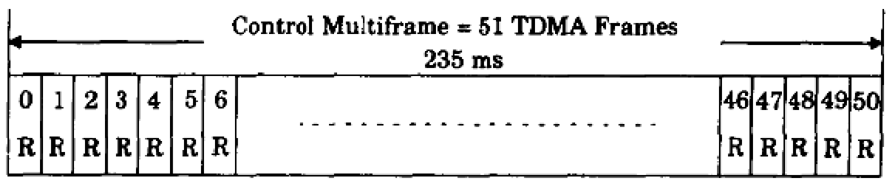


T_n: n^{th} TCH frame
S: Slow Associated Control Channel frame
I: Idle frame



F : FCCH burst (BCH)
S : SCH burst (BCH)
B : BCCH burst (BCH)
C : PCH/AGCH burst (CCCH)
I : Idle

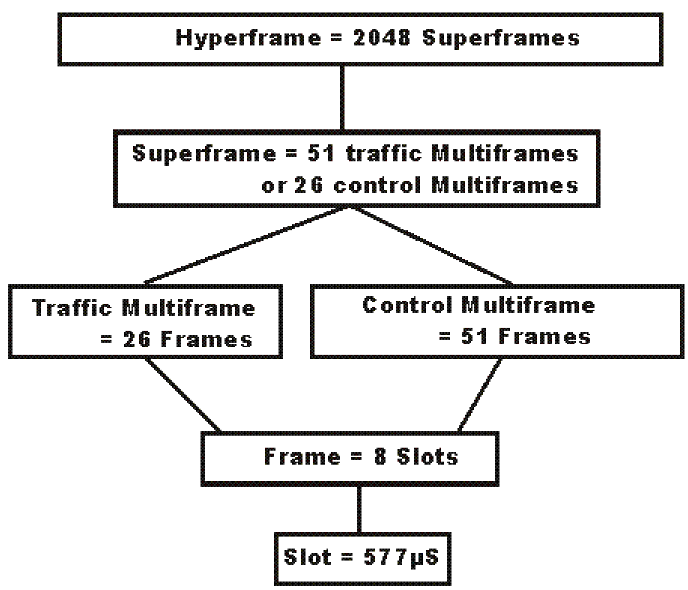
(a) **Downlink Multiframe**

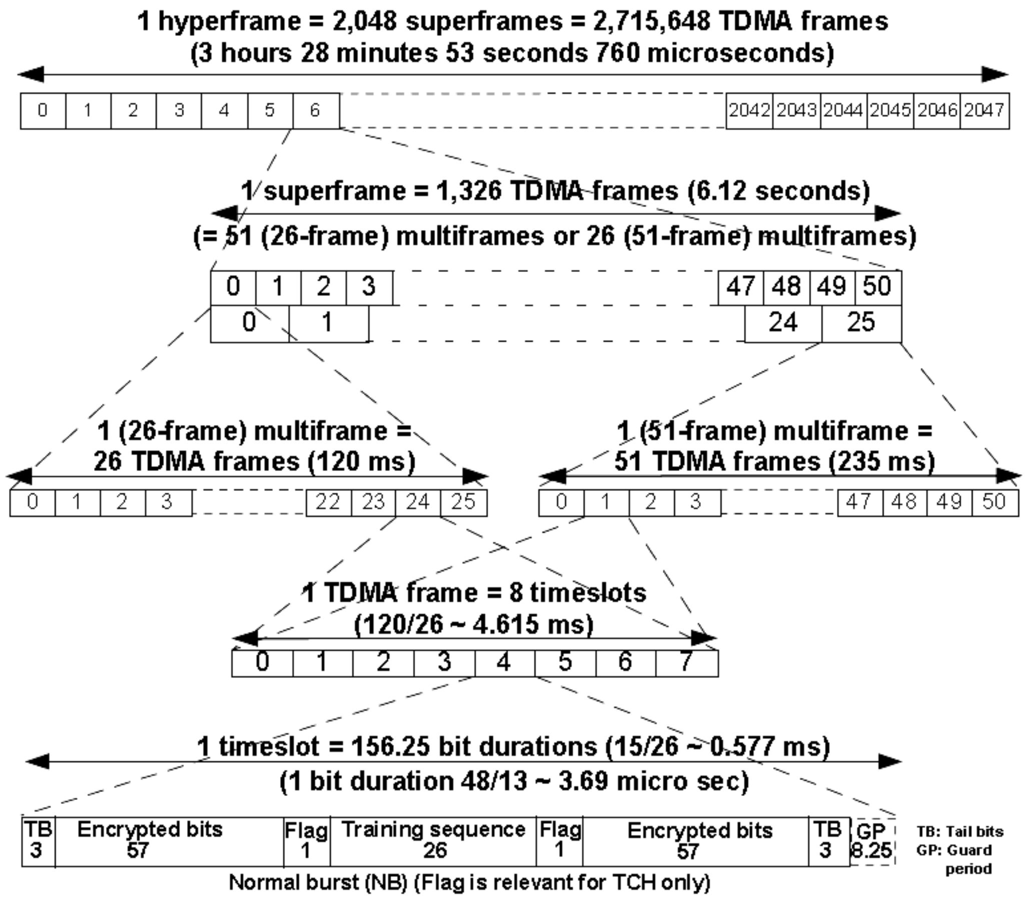


R : Reverse RACH burst (CCCH)

(b) **Uplink Multiframe**

GSM Frame Hierarchy





GSM Burst types

There are various bursts as mentioned below in GSM normal burst.

1. Normal burst (NB)
The fields in each slot are described below.

Trail bits (TB)- These are 3 bits at beginning and end of each time slot. Used for synchronization.

Coded data- This encrypted data coded data are placed in two 57-bit fields in each time slot. Stealing bit- It is used to indicate decoder at receiver whether the incoming burst is carrying signaling data or it is carrying user data.

Training sequence- It is used for multipath equalization, this is used to extract the desired signal from unwanted reflections. This training sequence also used to determine channel the burst has travelled, this helps in correcting rest of the frame and hence helps in decode the transmitted information properly.

Guard bits- These are about 8.25 bits, used to avoid overlap of different bursts.

2. **Frequency Correction Burst (FB)**

   TB (8 bits)
   Fixed bit sequence (142 bits)
   TB (3 bits)
   Guard Time (8.25 bits)

3. **Synchronization Burst(SB)**

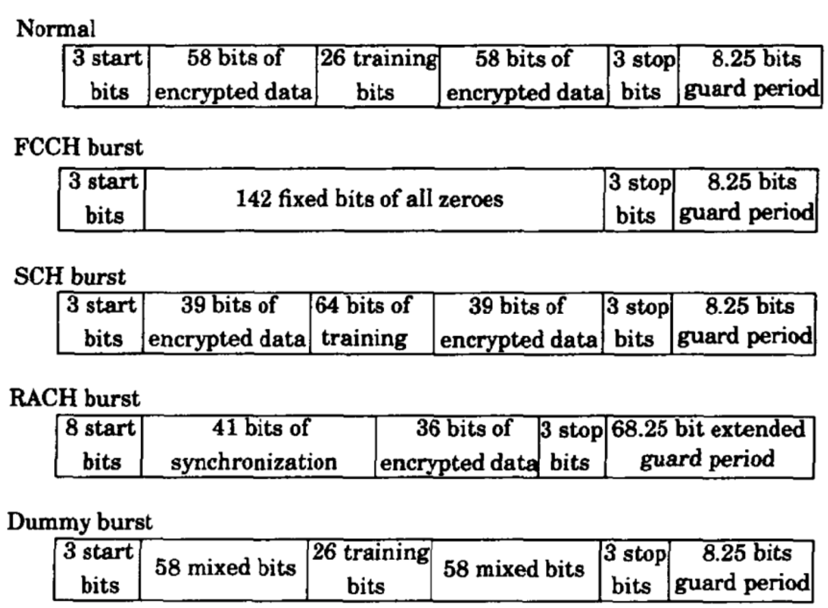
   TB (3 bits)
   Coded Data (39 bits)
   Synchronization sequence (264 bits)
   Coded Data (39 bits)
   TB (3 bits)
   Guard Time (8.25 bits)

4. **Dummy Burst**

   TB (3 bits)
   Mixed (142 bits)
   TB (3 bits)
   Guard period (8.25 bits)

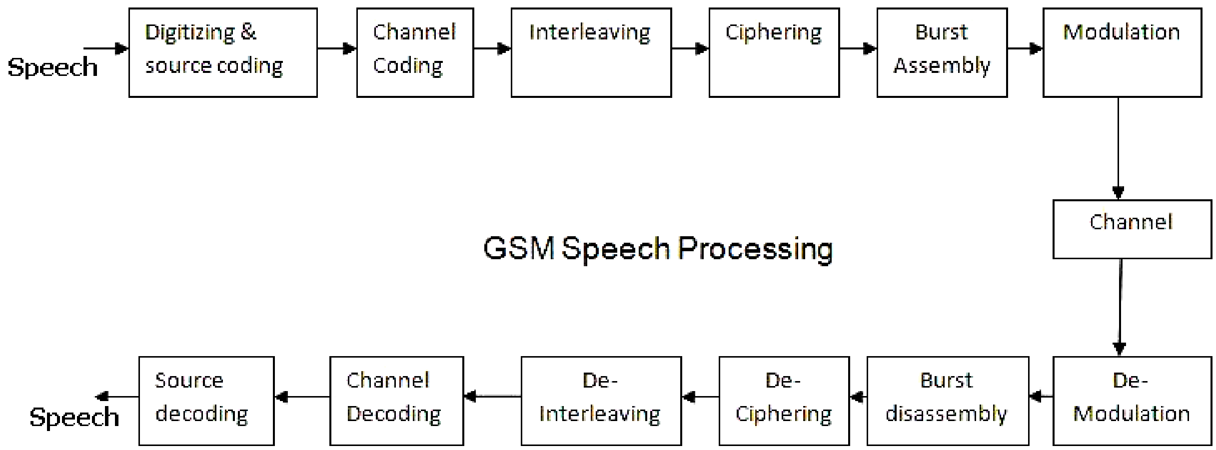
5. **Access Burst (AB)**

   Extended TB (8 bits)
   Synch sequence bits(41)
   Encrypted bits (36)
   TB (3 bits)
   Extended Guard period (68.25 bits)



Signal Processing in GSM

GSM physical layer is nothing but the modules through which speech will pass through before they are transmitted in the air.



These modules are speech coding, channel coding, interleaving, ciphering, burst assembly, modulation. Speech coding block uses 13kbps RELP (Residually Excited Linear Predictive coder). Channel coding block uses convolution coding of rate 1/2 with constraint length of 5. Interleaving block does diagonal interleaving, after 456 encoded bits in 20ms duration are broken into 57 bits sub-blocks. There will be about total 8 sub blocks of 57 bits each. Ciphering block uses A3 and A5 encryption algorithms. Encryption is changed call by call to enhance privacy. Burst assembly block frames the burst as required by GSM frame structure. The same is modulated and Gaussian filtered. Modulation block minimizes the occupied BW using GMSK modulation with BT of 0.3.

CDMA digital cellular standard (IS-95)

- IS-95 is second generation digital wireless cellular comminication system
- It allows each user within a cell to use the same radio channel, and users in the adjacent cells also use the same radio channels, since this is a direct sequence spread spectrum (DSSS) system.
- Band of operation: 800 MHz
  - Reverse Link (Uplink) : 824 MHz to 849 MHz
  - Forward Link (Downlink) : 869 MHz to 894 MHz
- Forward and reverse link separation: 45 MHz
- Duplexing: FDD
- User data is spread to a channel chip rate of 1.2288 Mcps
- Speech coder: Qualcomm Code Excited Linear Predictive (QCELP) Coder
- Channel coder: Convolutional encoder
  - Forward Link: (K=9, R=1/2)
  - Reverse Link: (K=9, R=1/3)

- Modulation technique:
  - Forward Link: QPSK
  - Reverse Link: Offset QPSK
- RAKE receivers are used to resolve and combine multipath components, thereby reducing the degree of fading.
- A combination of open and closed loop power control is used to adjust the transmit power of users so that the base station receives each user with the same received power. This solves the problem of near-far effect

Channels in CDMA (IS-95)

The IS-95 CDMA system is unique in that its forward and reverse links have different link structures. This is necessary to accommodate the requirements of a land-mobile communication system. The forward link consists of four types of logical channels: pilot, sync, paging, and traffic channels. There is one pilot channel, one sync channel, up to seven paging channels, and several traffic channels. Each of these forward-link channels is first spread orthogonally by its Walsh function, then it is spread by a quadrature pair of short PN sequences. All channels are added together to form the composite SS signal to be transmitted on the forward link.

The reverse link consists of two types of logical channels: access and traffic channels. Each of these reverse-link channels is spread orthogonally by a unique long PN sequence; hence, each channel is identified using the distinct long PN code. The reason that a pilot channel is not used on the reverse link is that it is impractical for each mobile to broadcast its own pilot sequence.

Forward CDMA Channel

The IS-95 CDMA system uses a 64 by 64 Hadamard matrix to generate 64 Walsh functions that are orthogonal to each other, and each of the logic channels on the forward link is identified by its assigned Walsh function.

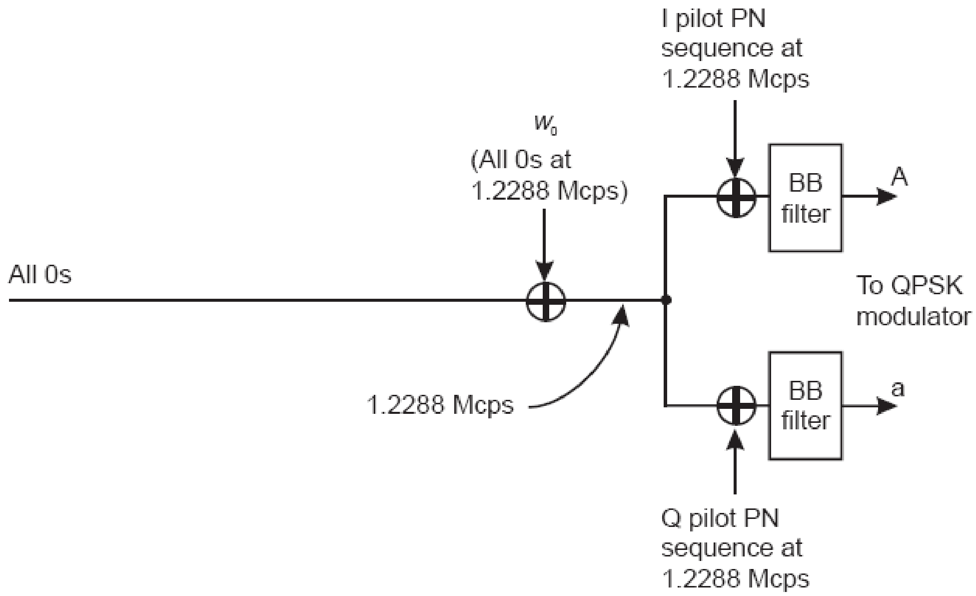
Walsh Codes

Various physical channels may exist at any time on a radio interface. To separate these channels at the receiver, they are spread with Walsh codes at the transmitter. These codes are formed by the rows of an *NXN* square matrix, whose entries are either 0 or 1. Usually, *N*=2*n* where *n* is an integer. They are orthogonal because if a 0 is mapped to -1 and a 1 to 1, then the sum of the term-by-term products of any two rows of this matrix is 0. This matrix, also known as the *Hadamard matrix*.

IS-95 uses a set of 64 fixed-length Walsh codes to spread forward physical channels. For example, Walsh code 0 is assigned to the pilot channel, code 32 to the sync channel, codes 1—7 to paging channels, and the rest to the forward traffic channels. In the reverse direction, they are used for orthogonal modulation where every six symbols from the block interleaver output are modulated as one of 64 Walsh codes.

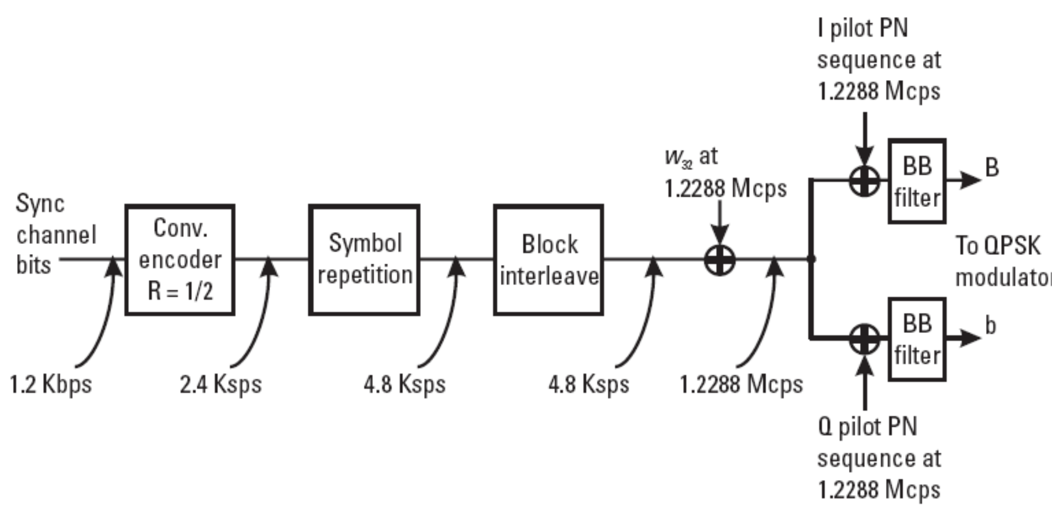
1. Pilot Channel

The pilot channel is identified by the Walsh function 0 (*w*0). The channel itself contains no baseband information. The baseband sequence is a stream of 0s that are spread by Walsh function 0, which is also a sequence of all 0s. The resulting sequence (still all 0s) is then spread, or multiplied, by a pair of quadrature PN sequences. Therefore, the pilot channel is effectively the PN sequence itself. The PN sequence with a specified offset uniquely identifies the particular sector that is transmitting the pilot signal. Note that both Walsh function 0 and the PN sequence are running at a rate of 1.2288 Mcps. After PN spreading, baseband filters are used to shape the digital pulses. These filters effectively low pass filter the digital pulse stream and control the baseband spectrum of the signal. This way, the signal bandwidth may have a sharper roll-off near the band edge. The pilot channel is transmitted continuously by the base station sector. The pilot channel provides the mobile with timing and phase reference. The mobile's measurement of the signal-to-noise ratio (i.e., *Ec* /*I*0) of the pilot channel also gives an indication of which is the strongest serving sector of that mobile.



2. Sync Channel

Unlike the pilot channel, the sync channel carries baseband information. The information is contained in the *sync channel message* that notifies the mobile of important information about system synchronization and parameters. Figure shows that the baseband information is error protected and interleaved. It is then spread by Walsh function 32 and further spread by the PN Sequence that is identified with the serving sector. The baseband information is at a rate of 1.2 Kbps.

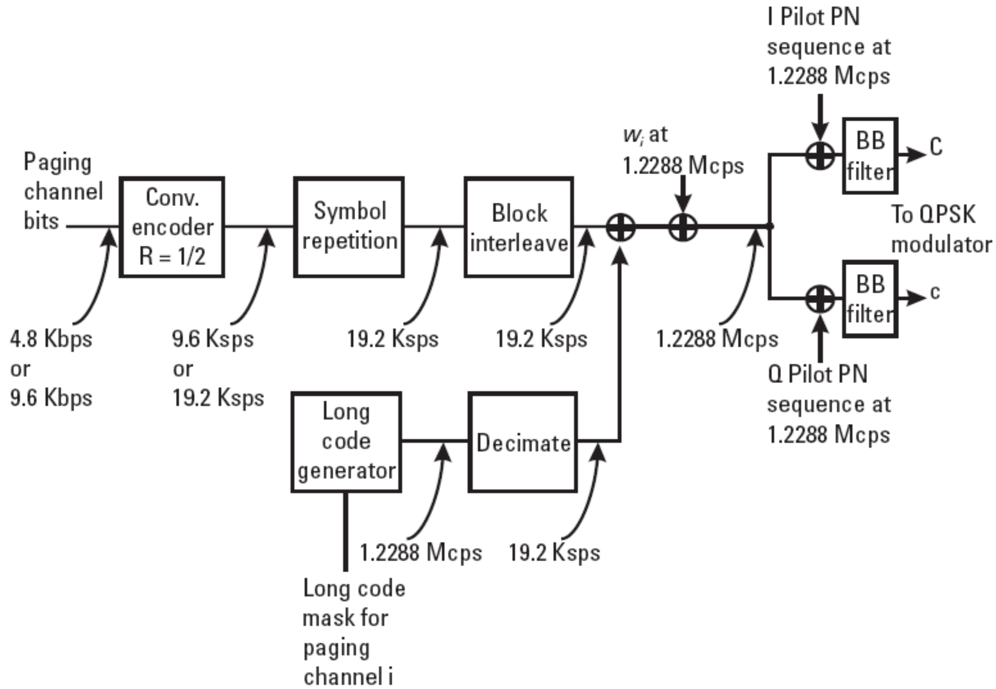


Scrambling Codes

In CDMA, each bit time is subdivided into *m* short intervals called chips. Typically there are 64 or 128 chips per bit. Each station is assigned a unique *m*-bit chip sequence. To transmit a 1 bit, a station sends its chip sequence. To transmit a 0 bit, it sends the one's complement of its chip sequence. No other patterns are permitted. Thus for *m* = 8, if station A is assigned the chip sequence 00011011, it sends a 1 bit by sending 00011011 and a 0 bit by sending 11100100. Increasing the amount of information to be sent from *b* bits/sec to *mb* chips/sec can only be done if the bandwidth available is increased by a factor of *m*, making CDMA a form of spread spectrum communication (assuming no changes in the modulation or encoding techniques). In order to protect the signal, the chip sequence code used is pseudo-random, it appears random, but is actually deterministic, so that the receiver can reconstruct the code for synchronous detection. This pseudo-random code is also called pseudo-noise (PN).

3. Paging Channel

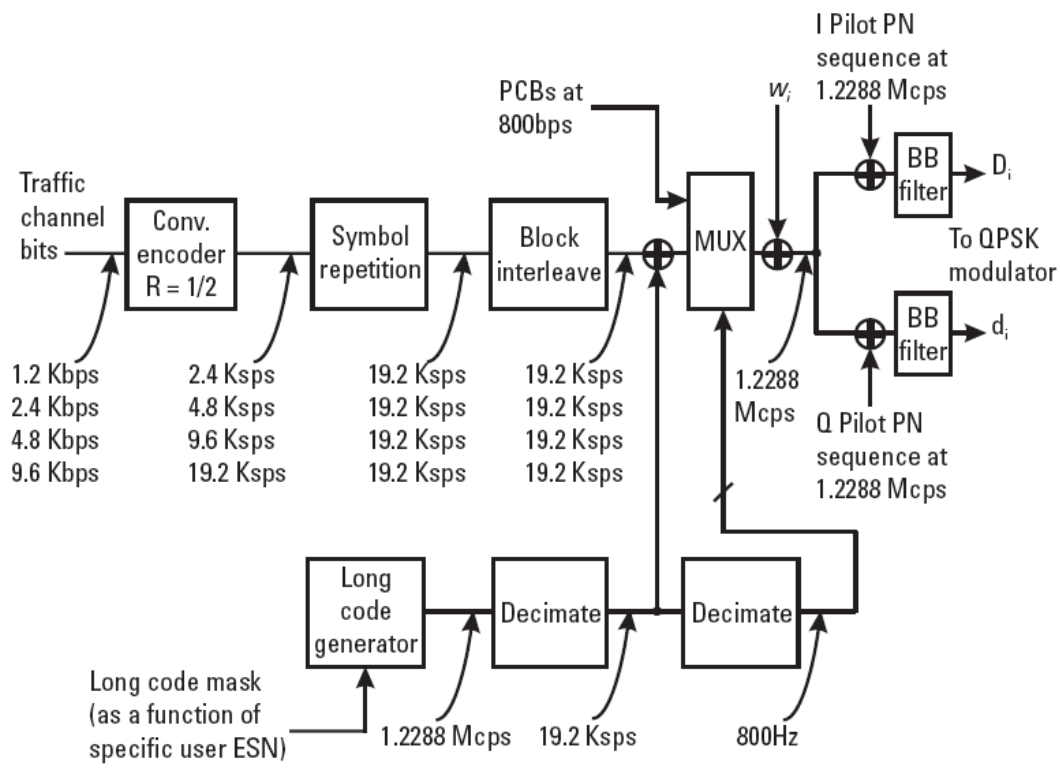
Similar to the sync channel, the paging channel also carries baseband information. But unlike the sync channel, the paging channel transmits at higher rates; it can transmit at either 4.8 or 9.6 Kbps. The PRAT field in the sync channel message informs the mobile of the data rate of the paging channel. Once the mobile acquires timing and synchronization using the sync channel, the mobile begins to monitor the paging channel. Although there can be up to seven paging channels per sector, each mobile only monitors one paging channel.



4. **Traffic Channel**

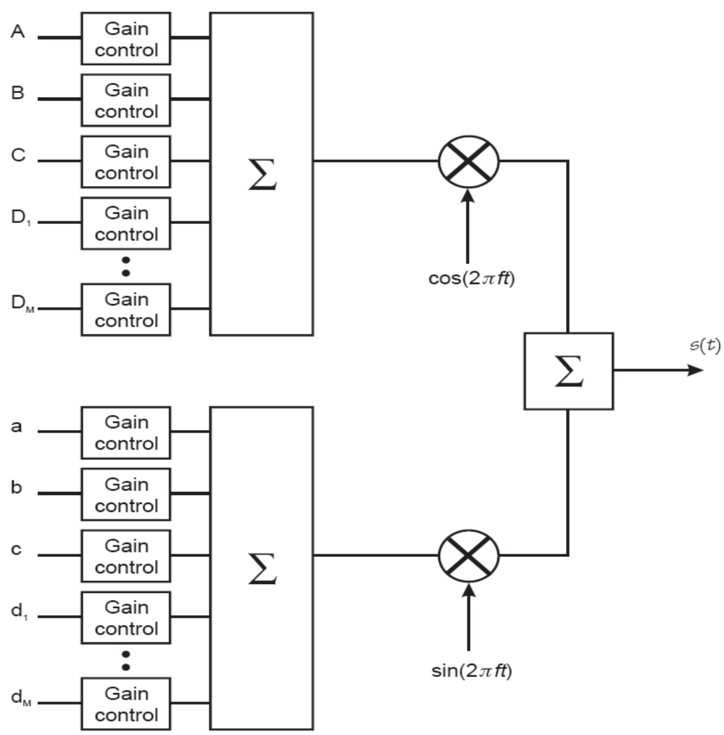
The forward traffic channel is used to transmit user data and voice; signaling messages are also sent over the traffic channel. The structure of the forward traffic channel is similar to that of the paging channel. The only difference is that the forward traffic channel contains multiplexed PCBs

The base station sends the power-control commands to the mobile using the forward link. These power-control commands are in the form of *power control bits* (PCBs). The amount of mobile power increase and power decrease per each PCB is nominally +1 dB and -1 dB



Modulator

The output of the logical channels is fed into the modulator. The gain of each logical channel, including pilot, sync, paging, and all traffic channels, is first adjusted by the gain control function. The gain of each channel dictates how much power is to be transmitted for that channel. The gains for the individual traffic channels are dynamically changing (i.e., they are controlled by the forward Power-control process). After the channel gains are adjusted, the signals are coherently added together to form the composite spread-spectrum signal. After the summation, both the I and the Q paths are up-converted by their respective carriers. The up-converted signals then are added together to form the final pass band QPSK signal.

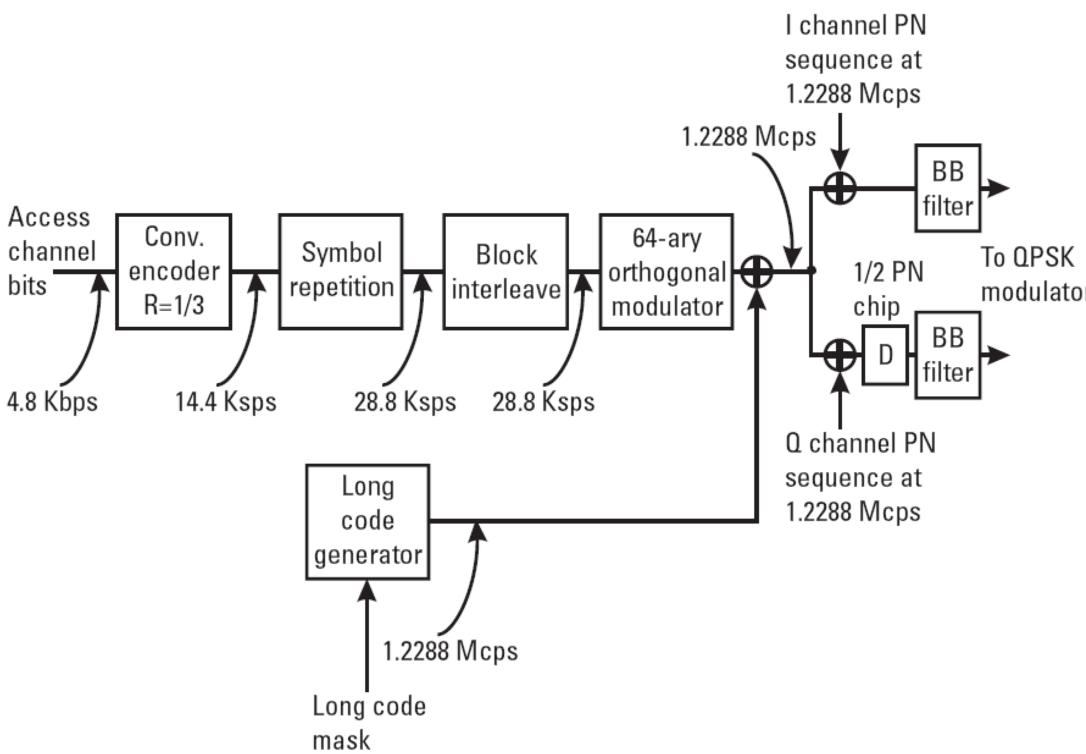


Reverse CDMA Channel

The reverse link supports two types of logical channels: access channels and traffic channels. Because of the noncoherent nature of the reverse link, Walsh functions are not used for channelization. Instead, long PN sequences are used to distinguish the users from one another.

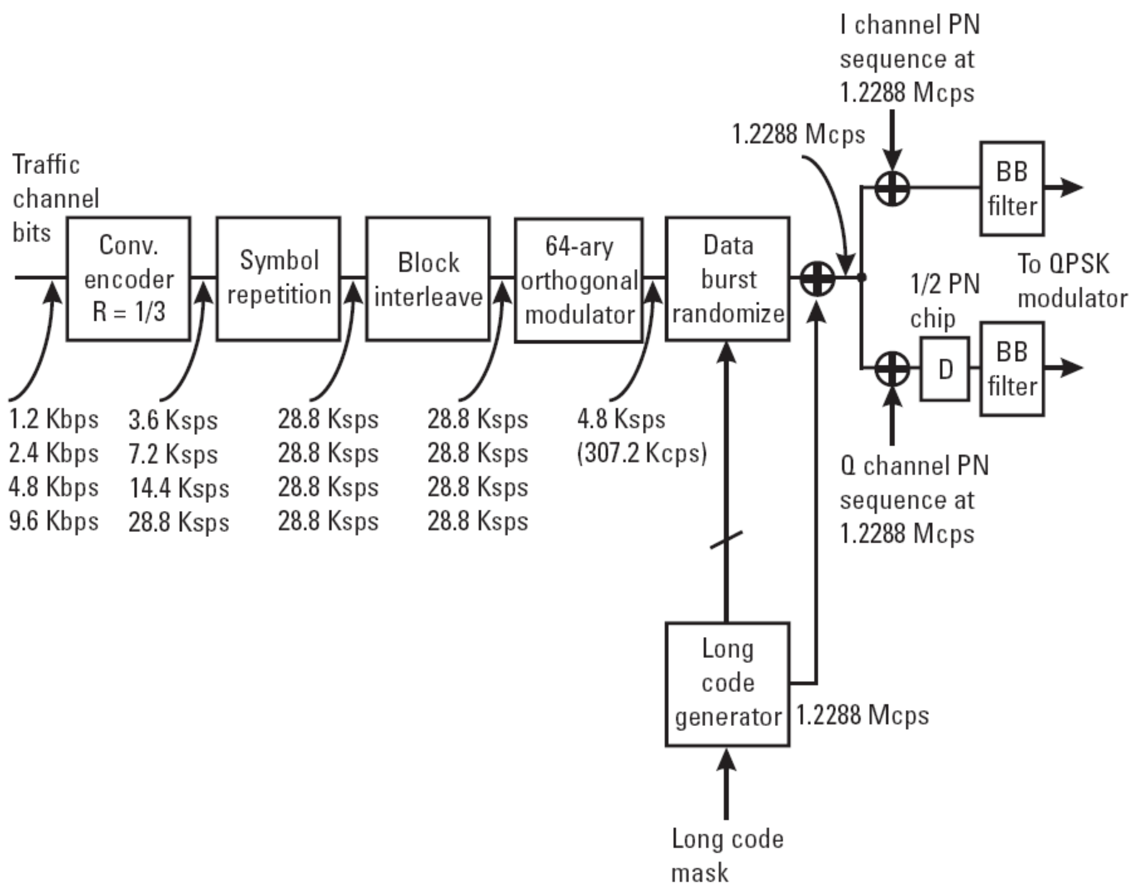
1. Access Channel

The access channel is used by the mobile to communicate with the base station when the mobile doesn't have a traffic channel assigned. The mobile uses this channel to make call originations and respond to pages and orders. The baseband data rate of the access channel is fixed at 4.8 Kbps. The baseband information is first error protected by an R = 1/3 convolutional encoder. The lower encoding rate makes error protection more robust on the reverse link, which is often the weaker of the two links. The symbol repetition function repeats the symbol once, yielding a code symbol rate of 28.8 Ksps. The data is then interleaved to combat fading. Following interleaving, the data is coded by a 64-ary orthogonal modulator. The set of 64 Walsh functions is used, but here the Walsh functions are used to modulate, or represent, groups of six symbols. The reason for orthogonal modulation of the symbols is again due to the noncoherent nature of reverse link. When a user's transmission is not coherent, the receiver (at the base station) still has to detect each symbol correctly. Making a decision of whether or not a symbol is +1 or -1 may be difficult during one symbol period. However, if a group of six symbols is represented by a unique Walsh function, then the base station can easily detect six symbols at a time by deciding which Walsh function is sent during that period. The receiver can easily decide which Walsh function is sent by correlating the received sequence with the set of 64 known Walsh functions. Note that on the forward link, Walsh functions are used to distinguish among the different channels. On the reverse link, Walsh functions are used to distinguish among the different symbols (or among groups of six symbols).



2. Reverse Traffic Channel

The reverse traffic channel is used to transmit user data and voice; signaling messages are also sent over the traffic channel. The structure of the reverse traffic channel is similar to that of the access channel. The major difference is that the reverse traffic channel contains a data burst randomizer. The orthogonally modulated data is fed into the data burst randomizer. The function of the data burst randomizer is to take advantage of the voice activity factor on the reverse link. Forward link uses a different scheme to take advantage of the voice activity factor—when the vocoder is operating at a lower rate, the forward link transmits the repeated symbols at a reduced energy per symbol and thereby reduces the forward-link power during any given period.



Data in each 20 ms frame are divided into 16 power control groups (PCG) each with a period of 1.25 ms. Some PCG are gated-ON while others are gated-OFF while passing through Data Burst Randomizer.

EIRP (Gate-OFF) = EIRP (Gate-ON) – 20 dB

Or

Noise floor level whichever is lower.

- If the user data rate is 9600 bps, transmission occurs on all 16 PCG
- If the user data rate is 4800 bps, transmission occurs on 8 PCG
- If the user data rate is 2400 bps, transmission occurs on 4 PCG and
- If the user data rate is 1200 bps, transmission occurs on all 2 PCG

Wireless Technologies

Wi-Fi

Wi-Fi stands for **Wi**reless **Fi**delity. Wi-Fi is based on the IEEE 802.11 family of standards and is primarily a local area networking (LAN) technology designed to provide in-building broadband coverage.

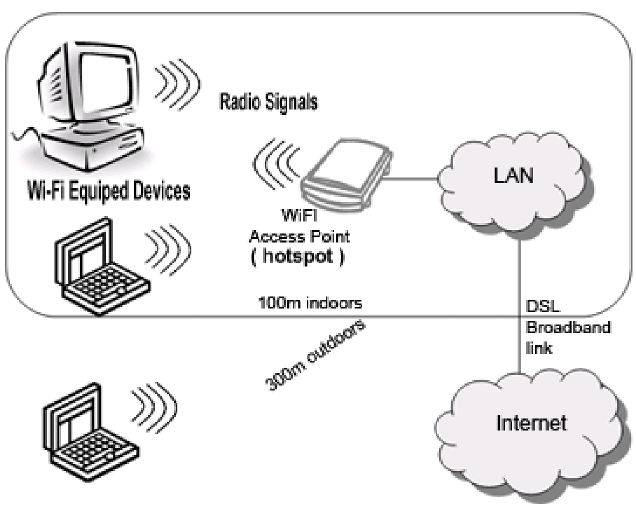
Current Wi-Fi systems based on IEEE 802.11a/g support a peak physical-layer data rate of 54Mbps and typically provide indoor coverage over a distance of 100 feet.

Wi-Fi has become the defacto standard for *last feet* broadband connectivity in homes, offices, and public hotspot locations. systems can typically provide a coverage range of only about 1,000 feet from the access point.

Wi-Fi offers remarkably higher peak data rates than do 3G systems, primarily since it operates over a larger 20MHz bandwidth but Wi-Fi systems are not designed to support high-speed mobility.

There are three most important items which makes Wi-Fi operational. These are:

- Radio Signals
- Wi-Fi Card which fits in your laptop or computer.
- Hotspots which create Wi-Fi Network.



Radio Signals:

Radio Signals make WiFi networking possible. These radio signals transmitted from Wi-Fi antennas are picked up by WiFi receivers such as computers and cell phones that are equipped with WiFi cards. Whenever a computer receives any of the signals within the range of a WiFi network which is usually 300 - 500 feet for antennas, the WiFi card will read the signals and thus create an internet connection between the user and the network without the use of a cord.

Access points which consist of antennas and routers are the main source which transmit and receive radio waves.

Wi-Fi Cards:

WiFi cards can be thought of as being an invisible cord that connects computer to the antenna for a direct connection to the internet.

WiFi cards can be external or internal. Eg on board built in card, USB dongle, PCMCIA card etc

Wi-Fi Hotspots:

A Wi-Fi hotspot is created by installing an access point to an internet connection. The access point transmits a wireless signal over a short distance. Typically, covering around 300 feets. When a Wi-Fi enabled device, such as a Pocket PC, encounters a hotspot, the device can then connect to that network wirelessly.

Security Features

- **Wired Equivalent Privacy (WEP)**
- **Wi-Fi Protected Access (WPA)**
- **IEEE 802.11i/WPA2**

WiMAX

WiMax is a standardized wireless version of Ethernet intended primarily as an alternative to wire technologies (such as Cable Modems, DSL and T1/E1 links) to provide broadband access to customer premises.

WiMAX operate similar to WiFi but at higher speeds, over greater distances and for a greater number of users. WiMAX has the ability to provide service even in areas that are difficult for wired infrastructure to reach and the ability to overcome the physical limitations of traditional wired infrastructure.

WiMAX is:

- Acronym for Worldwide Interoperability for Microwave Access.
- Based on Wireless MAN technology.
- A wireless technology optimized for the delivery of IP centric services over a wide area.
- A scaleable wireless platform for constructing alternative and complementary broadband networks.
- A certification that denotes interoperability of equipment built to the IEEE 802.16 or compatible standard. The IEEE 802.16 Working Group develops standards that address two types of usage models:
    - A fixed usage model (IEEE 802.16-2004).
    - A portable usage model (IEEE 802.16e).

The 802.16a standard for 2-11 GHz is a wireless metropolitan area network (MAN) technology that will provide broadband wireless connectivity to Fixed, Portable and Nomadic devices.

It can be used to connect 802.11 hot spots to the Internet, provide campus connectivity, and provide a wireless alternative to cable and DSL for last mile broadband access.

Feature:

OFDM-based physical layer:

The WiMAX physical layer (PHY) is based on orthogonal frequency division multiplexing, a scheme that offers good resistance to multipath, and allows WiMAX to operate in NLOS conditions.

Very high peak data rates:

WiMAX is capable of supporting very high peak data rates. In fact, the peak PHY data rate can be as high as 74Mbps when operating using a 20MHz wide spectrum.

More typically, using a 10MHz spectrum operating using TDD scheme with a 3:1 downlink-to-uplink ratio, the peak PHY data rate is about 25Mbps and 6.7Mbps for the downlink and the uplink, respectively.

Scalable bandwidth and data rate support:

WiMAX has a scalable physical-layer architecture that allows for the data rate to scale easily with available channel bandwidth.

For example, a WiMAX system may use 128, 512, or 1,048-bit FFTs (fast fourier transforms) based on whether the channel bandwidth is 1.25MHz, 5MHz, or 10MHz, respectively. This scaling may be done dynamically to support user roaming across different networks that may have different bandwidth allocations.

Support for TDD and FDD:

IEEE 802.16-2004 and IEEE 802.16e-2005 supports both time division duplexing and frequency division duplexing, as well as a half-duplex FDD, which allows for a low-cost system implementation.

Quality-of-service support:

The WiMAX MAC layer has a connection-oriented architecture that is designed to support a variety of applications, including voice and multimedia services.

WiMAX system offers support for constant bit rate, variable bit rate, real-time, and non-real-time traffic flows, in addition to best-effort data traffic.

WiMAX MAC is designed to support a large number of users, with multiple connections per terminal, each with its own QoS requirement.

IP-based architecture:

The WiMAX Forum has defined a reference network architecture that is based on an all-IP platform. All end-to-end services are delivered over an IP architecture relying on IP-based protocols for end-to-end transport, QoS, session management, security, and mobility.

WiMAX building blocks

A WiMAX system consists of two major parts:

- A WiMAX base station.
- A WiMAX receiver.

WiMAX Base Station:

A WiMAX base station consists of indoor electronics and a WiMAX tower similar in concept to a cell-phone tower. A WiMAX base station can provide coverage to a very large area up to a radius of 6 miles. Any wireless device within the coverage area would be able to access the Internet.

The WiMAX base stations would use the MAC layer defined in the standard . a common interface that makes the networks interoperable and would allocate uplink and downlink bandwidth to subscribers according to their needs, on an essentially real-time basis.

Each base station provides wireless coverage over an area called a cell. Theoretically, the maximum radius of a cell is 50 km or 30 miles however, practical considerations limit it to about 10 km or 6 miles.

WiMAX Receiver:

A WiMAX receiver may have a separate antenna or could be a stand-alone box or a PCMCIA card sitting in your laptop or computer or any other device. This is also referred as customer premise equipment (CPE).

WiMAX base station is similar to accessing a wireless access point in a WiFi network, but the coverage is greater.

Backhaul:

A WiMAX tower station can connect directly to the Internet using a high-bandwidth, wired connection (for example, a T3 line). It can also connect to another WiMAX tower using a line-of-sight, microwave link.

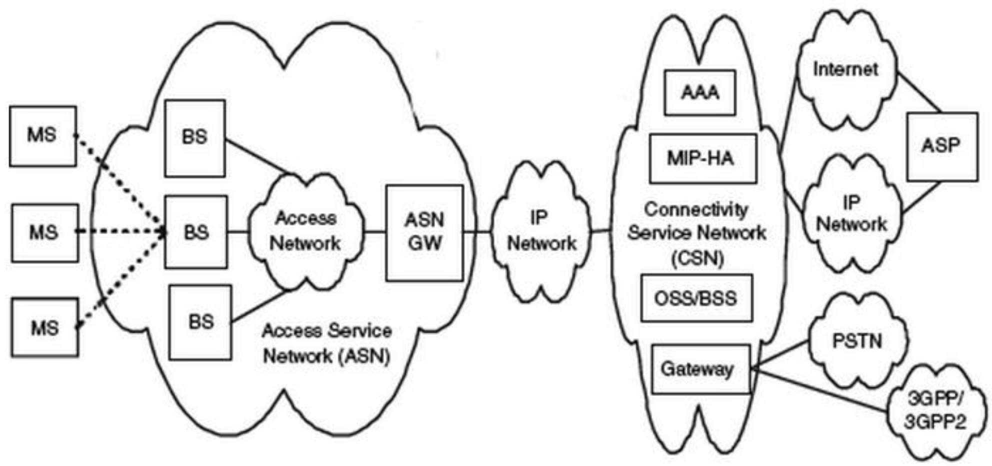
Backhaul refers both to the connection from the access point back to the base station and to the connection from the base station to the core network.

It is possible to connect several base stations to one another using high-speed backhaul microwave links. This would also allow for roaming by a WiMAX subscriber from one base station coverage area to another, similar to the roaming enabled by cell phones.

WiMAX network architecture

The overall network may be logically divided into three parts:

1. Mobile Stations (MS) used by the end user to access the network.
2. The access service network (ASN), which comprises one or more base stations and one or more ASN gateways that form the radio access network at the edge.
3. Connectivity service network (CSN), which provides IP connectivity and all the IP core network functions.



UMB (Ultra-Mobile Broadband)

Ultra-Mobile Broadband, UMB is the next evolution for the cdma2000 cellular telecommunications system. The UMB cellular system promises to provide very much faster data transfer speeds, and enables the system to compete with other mobile broadband systems including WiMAX and Wi-Fi.

The aims for UMB,

- Significant increase in user data rates as compared to the existing cdma2000 cellular technologies
- Increase in the system capacity
- Lowering the cost of per bit data transfer
- Enhancements to the existing services
- Possibility of new applications and
- Ability to use new spectrum opportunities.

UMB salient features

The UMB, Ultra-Mobile Broadband cellular telecommunications system offers many new features and techniques that enable it to fulfil the high expectations for it, and to enable it to compete with other new and emerging technologies.

- Data rates of over 275 Mbit/s in the downlink (base station to mobile) and over 75 Mbit/s in the uplink (mobile to base station).
- Uses an OFDM / OFDMA air interface
- Uses frequency division duplex (FDD).
- Possesses an IP network architecture
- Has a scalable bandwidth between 1.25 - 20 MHz (NB - OFDM / OFDMA systems are well suited for wide and scalable bandwidths)

EV-DO

CDMA2000 1xEV-DO cell phone system is a standard that has evolved from the CDMA2000 mobile phone system and it is now firmly established in many areas of the world.

1xEV-DO, also called EV-DO, EVDO, or just EV, was initially developed to meet the IMT-2000 requirements for a greater-than-2-Mbit/s downlink for stationary communications. For a while, the standard was called HDR (High Data Rate) but was renamed as 1xEV-DO after it was ratified by the ITU. Originally, 1xEV-DO stood for "1x Evolution-Data Only", referring to its being the next evolutionary step after the 1xRTT (“1x”) standard, with its channels carrying only data traffic. Later, due to the negative connotations of the word "only" in its name, the "DO" part of EV-DO was changed to represent "Data Optimized".

Revision 0: EV-DO Rev 0 was the original version of 1xEV-DO and the first to be widely deployed. Revision 0 offered data rates up to 2.4 mbps, averaging 300-600 kbps in the real world. This is much faster than the 50-80 kbps typically offered by 1xRTT technology. Other key features introduced by Revision 0 include:

- **supports IP-based network connectivity and software applications**
- Supports broadband data applications, such as broadband Internet or VPN access, MP3 music downloads, 3D gaming, TV broadcasts, video and audio downloads. In many countries, it has been deployed as a DSL substitute.

Revision A: EV-DO Rev. A offers fast packet establishment on both the forward and reverse links along with air interface enhancements that reduce latency and improve data rates. In addition to an increase in the maximum downlink rate from 2.45 Mb/s to 3.1 Mb/s, Rev. A incorporates 12 time improvement in the maximum uplink data rate, from 225 Kb/s to 1.8 Mb/s. EV-DO Rev. A supports low latency services (as low as 50ms) including VoIP and video telephony on the same carrier with traditional Internet packet data services. EV-DO Rev. A air-interface latency specifications have not been published, however several Qualcomm documents note latency in the "low double digit" range with the highest RSVP settings. Latency of 50ms is claimed to be possible. This compares favorably with Rev. 0 latencies of 150-200 ms.

Revision B: EV-DO Rev B is the next evolutionary step up from EV-DO Rev A. It adds the following enhancements:

- Higher rates per carrier (up to 4.9 Mbps on the downlink).
- Higher rates by bundling multiple channels together enhance user experience and enables new services such as high definition video streaming.
- Utilizes statistical multiplexing across channels to further reduce latency, enhancing the experience for latency-sensitive services such as gaming, video telephony, remote console sessions and web browsing.
- Hybrid frequency reuse which reduces interference from the adjacent sectors and improves the rates that can be offered, especially to users at the edge of the cell.

The Universal Mobile Telecommunication System (UMTS)

International Telecommunication Union (ITU) started the process of defining the standard for third generation systems, referred to as International Mobile Telecommunications 2000 (IMT-2000). In Europe European Telecommunications Standards Institute (ETSI) was responsible of UMTS standardisation process. In 1998 Third Generation Partnership Project (3GPP) was formed to continue the technical specification work.

UMTS services

UMTS offers teleservices (like speech or SMS) and bearer services, which provide the capability for information transfer between access points. It is possible to negotiate and renegotiate the characteristics of a bearer service at session or connection establishment and during ongoing session or connection. Both connection oriented and connectionless services are offered for Point-to-Point and Point-to-Multipoint communication.

Bearer services have different QoS parameters for maximum transfer delay, delay variation and bit error rate. Offered data rate targets are:

- 144 kbits/s satellite and rural outdoor
- 384 kbits/s urban outdoor
- 2048 kbits/s indoor and low range outdoor

UMTS network services have different QoS classes for four types of traffic:

- Conversational class (voice, video telephony, video gaming)
- Streaming class (multimedia, video on demand, webcast)
- Interactive class (web browsing, network gaming, database access)
- Background class (email, SMS, downloading)

UMTS Architecture

A UMTS network consist of three interacting domains; Core Network (CN), UMTS Terrestrial Radio Access Network (UTRAN) and User Equipment (UE). The main function of the core network is to provide switching, routing and transit for user traffic. Core network also contains the databases and network management functions.

The basic Core Network architecture for UMTS is based on GSM network with GPRS. All equipment has to be modified for UMTS operation and services. The UTRAN provides the air interface access method for User Equipment. Base Station is referred as Node-B and control equipment for Node-B's is called Radio Network Controller (RNC).

Core Network

The Core Network is divided in circuit switched and packet switched domains. Few of the

A. Circuit switched elements are

- Mobile services Switching Centre (MSC),
- Visitor location register (VLR) and
- Gateway MSC.

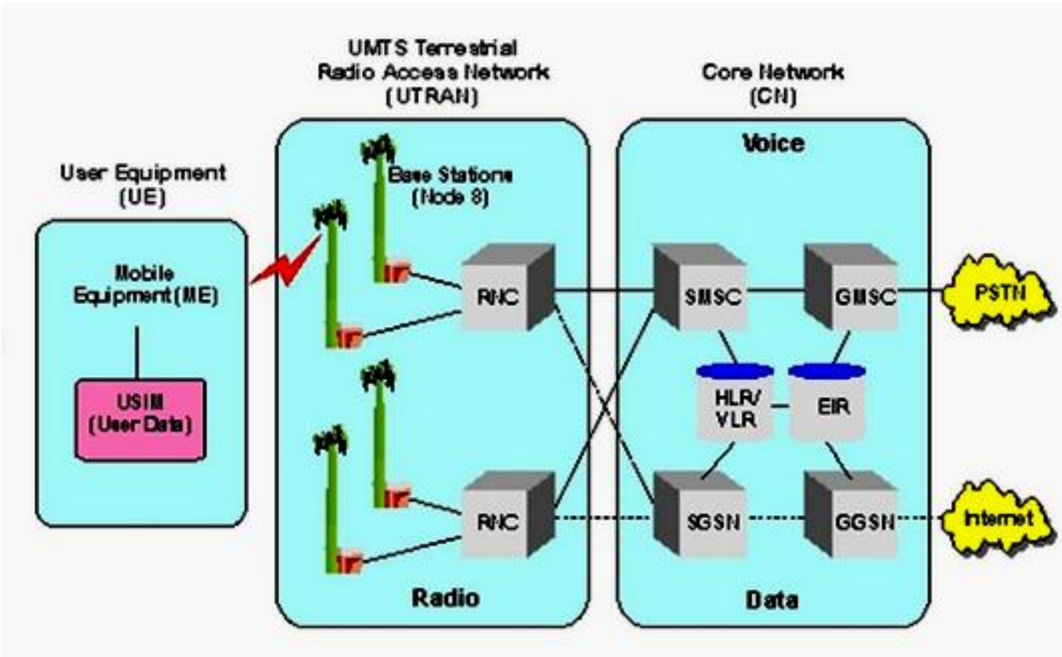
B. Packet switched elements are

- Serving GPRS Support Node (SGSN) and
- Gateway GPRS Support Node (GGSN).

C. Some network elements, like EIR, HLR, VLR and AUC are shared by both domains.

Radio Access Network

Wideband CDMA technology was selected for UTRAN air interface. UMTS WCDMA is a Direct Sequence CDMA system where user data is multiplied with quasi-random bits derived from WCDMA Spreading codes. In UMTS, in addition to channelisation, Codes are used for synchronisation and scrambling. WCDMA has two basic modes of operation: Frequency Division Duplex (FDD) and Time Division Duplex (TDD).



Long Term evolution (LTE)

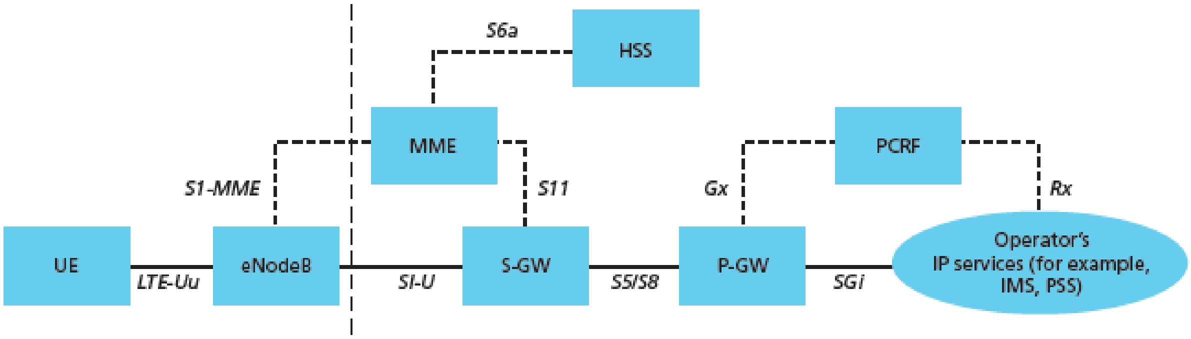
In contrast to the circuit-switched model of previous cellular systems, Long Term Evolution (LTE) has been designed to support only packet-switched services. It aims to provide seamless Internet Protocol (IP) connectivity between user equipment (UE) and the packet data network (PDN), without any disruption to the end users' applications during mobility.

While the term "LTE" encompasses the evolution of the Universal Mobile Telecommunications System (UMTS) radio access through the Evolved UTRAN (E-UTRAN), it is accompanied by an evolution of the non-radio aspects under the term "System Architecture Evolution" (SAE), which includes the Evolved Packet Core (EPC) network. Together LTE and SAE comprise the Evolved Packet System (EPS).

LTE performance requirements

Metric	Requirement
Peak data rate	DL: 100Mbps UL: 50Mbps (20 MHz spectrum)
Mobility support	Upto 500 Kmph
Control plane latency	< 100 ms (for idle to active)
User plane latency	< 5 ms
Control plane capacity	> 200 users per cell (for 5 MHz spectrum)
Coverage	5 to 100 with slight degradation after 30 Km
Spectrum flexibility	1.25, 2.5, 5, 10, 15 and 20 MHz

Architecture



Evolved Radio Access Network (RAN)
The evolved RAN for LTE consists of a single node, i.e., the eNodeB (eNB) that interfaces with the UE.

Serving Gateway (SGW)
The SGW routes and forwards user data packets, while also acting as the mobility anchor for the user plane during inter-eNB handovers and as the anchor for mobility between LTE and other 3GPP technologies

Mobility Management Entity (MME)
The MME is the key control-node for the LTE access- network. It is responsible for idle mode UE tracking and paging procedure including retransmissions. It is involved in the bearer activation/deactivation process and is also responsible for choosing the SGW for a UE at the initial attach and at time of intra-LTE handover involving Core Network (CN) node relocation.

Packet Data Network Gateway (PDN GW)
The PDN GW provides connectivity to the UE to external packet data networks by being the point of exit and entry of traffic for the UE.

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