16BEEC601 **MOBILE COMMUNICATION** **3 2 0 3**

OBJECTIVES

- To learn the fundamental cellular radio concepts
- To learn radio propagation models
- To provide ideas about analog and digital modulation techniques used in mobile communication
- To learn various coders and multiple access techniques used in mobile communication
- To study the architectures of AMPS, GSM, WLL, Bluetooth, DECT, GPRS

INTENDED OUTCOMES:

- Gain adequate knowledge in the fundamentals of cellular radio concepts
- Gain adequate knowledge in radio propagation models
- Ability to provide ideas about analog and digital modulation techniques used in mobile communication

UNIT-I CELLULAR CONCEPT AND SYSTEM DESIGN FUNDAMENTALS

Introduction to wireless communication: Evolution of mobile communications, mobile radio systems- Examples, trends in cellular radio and personal communications.

Cellular Concept: Frequency reuse, channel assignment, hand off, Interference and system capacity, tracking and grade of service, Improving Coverage and capacity in Cellular systems.

UNIT-II MOBILE RADIO PROPAGATION

Free space propagation model, reflection, diffraction, scattering, link budget design, Outdoor Propagation models, Indoor propagation models, Small scale Multipath propagation, Impulse model, Small scale Multipath measurements, parameters of Mobile multipath channels, types of small scale fading, statistical models for multipath fading channels.

UNIT-III MODULATION TECHNIQUES AND EQUILISATION

Modulation Techniques: Minimum Shift Keying, Gauss ion MSK, M-ary QAM, M-ary FSK, Orthogonal Frequency Division Multiplexing, Performance of Digital Modulation in Slow-Flat Fading Channels and Frequency Selective Mobile Channels. Equalization: Survey of Equalization Techniques, Linear Equalization, Non-linear Equalization, Algorithms for Adaptive Equalization. Diversity Techniques, RAKE receiver.

UNIT-IV CODING AND MULTIPLE ACCESS

Coding: Vocoders, Linear Predictive Coders, Selection of Speech Coders for Mobile Communication, GSM Codec, RS codes for CDPD. Multiple Access Techniques: FDMA, TDMA, CDMA, SDMA, Capacity of Cellular CDMA and SDMA.

UNIT V WIRELESS SYSTEMS ANTENNAS AND STANDARDS

AMPS, GSM, WLL, Bluetooth, IS-95 and DECT - RFID antennas – Mobile Antennas - GPRS

TEXT BOOKS:

S.NO.	Author(s) Name	Title of the book	Publisher	Year of publication
1	Rappaport.T.S	Wireless Communications: Principles and Practice	Pearson Education/ Prentice Hall of India, New Delhi	2003
2	Jochen Schiller	Mobile Communication	PHI, New Delhi.	2003

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KARPAGAM ACADEMY OF HIGHER EDUCATION

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Department of Electronics and Communication Engineering

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MOBILE COMMUNICATION

LECTURE NOTES

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UNIT V WIRELESS SYSTEMS ANTENNAS AND STANDARDS

Second Generation and Third Generation Wireless Networks and Standards, WLL, Blue tooth. AMPS, GSM, IS-95 and DECT - RFID antennas – Mobile Antennas - GPRS

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UNIT - 1

1. CELLULAR CONCEPT AND SYSTEM DESIGN FUNDAMENTALS

1.1 INTRODUCTION TO WIRELESS COMMUNICATION

Communication is a vital factor which allows people to connect with each other around the world. From paper based Media to electronic media like telephone, television, radio, the communication was developed rapidly during the last century. As they are wired media, the use is limited to certain distance. To overcome the limitation of the wired communication media. The mobile communication was found to make the communication more efficient and effective.

1.2 EVOLUTION OF MOBILE COMMUNICATION

First Generation Mobile Communication (1g)

1st generation mobile phones were the earliest. Cellular systems that were developed in early 80's. The idea was come up with the development of short ranged radio telephones such as walkie-talkies at early 70's. These systems were used to communicate over small geographical area and the system is fully analog, which means communication is done by switching from sender to receiver. In these telephone systems, the communication was done by transmitting radio signals on specific frequencies through the airwaves.

In the early 80's, the 1st generation phone were developed to increase the efficiency of the mobile technology. 1G phones are also analog and the major improvement done was increasing the range of the transmission. So that the communication can be done over a large area, that the walkie-talkies. 1st Generation mobile phones used a single universal network standard known as Advanced Mobile Phone Systems (AMPS).

In this technology, separate frequencies were used for each conversation, and therefore needed considerable band width for a large number of users. In AMPS, the cell centre could assign channels to handsets based on the signal strength. It also allowed re-using the same frequency in various locations without interference. This allows a large number of phones to be supported over a geographical area. AMPS Cellular Service was operated in the 800-900 MHZ cellular FM band.

Features of 1G mobile Technology

Analog system, Mobility, circuit switched Technology, Basic voice calls only, Limited local and regional coverage, phones were large in size, low capacity.

Second Generation Mobile Communication (2G)

After the 1st Generation Mobile Technology, at early 90's the 2G or 2nd Generation mobile networks were established. It was developed to overcome the problem arose in analog systems.

Therefore digital encryption of the voice calls was used in 2G Technology. 2G Cellular Telecom networks were commercially launched on GSM (Global System for Mobile Communication). Standard in FINLAND in 1991. 2G introduces data services for mobile phones (i.e.SMS & E.mail) SMS is nothing but short message service. This SMS facility made the mobile communication more efficient and effective to use.

2G Technologies can be divided into two standards depending on the type of multiplexing used. TDMA-based (Time Division Multiple Access) and CDMA-based (Code Division Multiple Access) are those two standards.

TDMA operates in 800 MHZ or 1900 MHZ.

CDMA operates both in 800 MHZ or 1900 MHZ bands.

The main 2G standards are follows.

GSM-TDMA based Technology.

CDMA one – CDMA based Technology

2.5G Mobile Technology:

It is the enhanced version of 2G Technology.

The enhancement achieved through implementing a packet switched domain in addition to circuit switched domain. This enhancement gives the user a better service and access the internet at higher data rates. The technology used to access the internet is General Packet Radio Service (GPRS). It provides data rates from 56 Kbps upto 115 kbps. GPRS can be used for services such as WAP, MMS, E-Mail, WWW access.

2.75G Mobile Technology :

Another enhancement of 2G Technology is 2.75G, it is road to 3G Technology. EDGE is a standard developed by AT&T in 2003, It provides increased capacity of GSM/GPRS. Higher data transfer rates are achieved.

Third Generation Mobile Communication (3G)

3rd generation or 3G Mobile Technology is the latest state in the development of wireless communication technology. The 3G Technology was first used in Japan in 2001, as they did not use 2.5G technology. 3G Technology allows the mobile to offer high speed internet access, data and CD quality services. Currently 80% of the world population is using this 3G mobile technology for their communication purposes. Compared to earlier mobile phones, 3G handsets provide many features like, TV streaming, multimedia, video conferencing, web-browsing, e-mail, paging, fax, navigational maps etc, universal Mobile Telephone System is one of the technologies used in 3G mobiles which are based on W-CDMA technology. The 3G technology allows 2 Mbps speed for stationary & 38H kbps for mobile systems. In HSDPA or 3.5G technology, the data is transmitted together with error correction bits that can be corrected without re-transmission.

Fourth Generation Mobile Communication (4G)

4G can be considered as the future of the Mobile technology. Even though 3G is a successful invention, still these are many reasons to go for 4G technology. In 4G mobile phones, the data transfer rate is more than 100 mbps. It is expected to have high quality services for multimedia purposes such as real time audio, high speed data, HDTV, mobile TV etc. A 4G mobile phone can be considered as a fully functional computer with the portability. Wimax connection is using the 4G technology in the mobile phones recently. It uses OFDM (Orthogonal Frequency Division Multiplexing) technology. OFDM allows transferring data more than other forms of multiplexing.

1.3 EXAMPLES OF WIRELESS COMMUNICATION SYSTEMS

The term ‘mobile’ has been used in classify any radio terminal that could be moved during operation. The term ‘mobile’ is used to describe a radio terminal that is attached to a high speed mobile platform.

A subscriber is often called as mobile user or portable user. Each user pays a subscription fee to use the system and each user’s communication device is called a subscription unit.

Mobile radio transmission systems may be classified as simplex, half-duplex or full-duplex. In simplex systems, communication is possible in only one direction, paging systems, in which messages are received but not acknowledged, are simplex systems.

Half-duplex radio systems allows two-way communication. But use the same radio channel for both transmission and reception. This means that at any given time, a user can only transmit or service information. Full duplex, allows simultaneous radio transmission and reception between a subscriber and a base station; by providing two simultaneous but separate channels (Frequency Division Duplex or FDD) or adjacent time slots on a single radio channel (time, division duplex TDD) for communication to and from the user.

1.3.1 Paging system

A communication systems that send brief messages to a subscriber. The message may be either a numeric message, an alphanumeric message, or a voice message. In modern paging systems news headlines, stock quotations and faxes may be sent. A message is sent to a paging subscriber via the paging system access number with a telephone keypad or modem. The issued message is called PAGE, The paging system then transmits the page throughout the service area using base stations. Which broadcast the page on a radio carrier. Paging systems vary widely in their complexity and coverage area.

Simple paging system cover a limited range of 2 to 5 km, or may be within individual building. Wide area paging systems consist of a network of telephone lines, many base station transmitters and large radio towers that simultaneously broadcast a page from each base stations (This is called simul-casting) Simul-cast transmitters are located within the same service area or in different cities or countries paging systems are designed to provide reliable communication to subscribers.

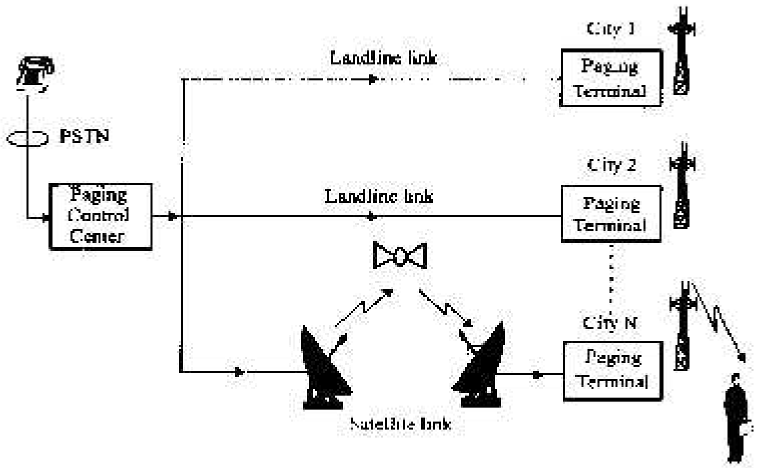


Fig 1.3.1 **Paging system**

A wide area paging system. The paging control, centre dispatches pages received from the PSTN throughout several cities at the same time.

1.3.2 Cordless Telephone System

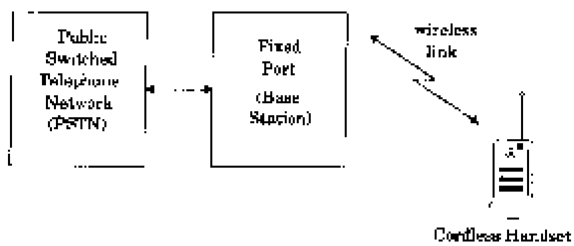


Fig 1.3.2 **Cordless Telephone System**

Cordless telephone systems are full duplex communication systems that use radio to connect a portable handset to a dedicated base station, which is then connected to a dedicated telephone line with a specific telephone number on the PSTN first generation cordless telephone systems (1980s) allows subscribers to use their handsets only over distances of a few ten meters.

Second generation cordless telephones allow subscribers to use their handsets at many outdoor locations within urban centres. Cordless telephone systems provides the user with limited range and mobility. It provides coverage ranges up to a few hundred meters.

1.3.3 Cellular Telephone system

A cellular telephone systems provides a wireless connection to the PSTN. Cellular systems accommodate a large number of users over a large geographical area, with a limited frequency spectrum. It provides high quality service.

Basic block diagram of cellular mobile system:

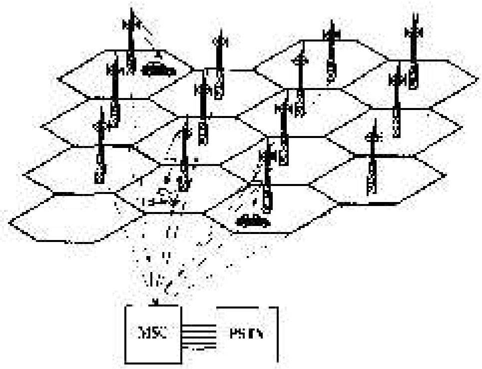


Fig 1.3.3 **Basic block diagram of cellular mobile system**

The elements of a basic cellular system are

1. Mobile Unit

2. Base station

3. Mobile switching centre

4. Public switched Telephone network

The primary element of a cellular mobile system is the 'cell'

Cell

It is a sub area which is allotted a particular frequency.

In other words cell is the smallest area kin a geographical zone

Cell Shape

It can be circle, hexagon or square. When comparing the performance of various cell shapes Hexagonal shape is the optimum cell shape which was selected as a standard shape, it improves the entire mobile system's efficiency.

Mobile Unit

It is available with the subscriber in a cell. It contains a control unit, transceiver and antenna system.

Cell Site

It provides interface between MTSO (MSC-Mobile Switching Centre) and the mobile units. It has a control unit, Radio cabinets, Antennas, power plant and Data terminals.

MTSO

Mobile Telephone Switching office connects all the mobile phones to PSTN (Public Switched Telephone network). The MTSO monitors all the calls initiated, processed and terminated. The hand off mechanism is done by MTSO as per the request from the individual cells. The MTSO can handle more than 1 lakh subscribers and approximately 5000 simultaneous conversations, with all maintenance and billing functions.

Base station

It serves as a connector (bridge) between all mobile subscribers in a cell to MSC through telephone lines or microwave link.

FVC : The channels that are used for voice transmission from base station to mobile units are known as forward voice channels.

RVC : The channels that are used for voice transmission from mobile units to base station are known as Reverse Voice channels.

Control Channel

There are two channels responsible for initiating mobile calls known forward and reverse control channels (RCC and FCC).

Whenever a call is in progress the mobile unit is connected to MSC and PSTN. Each base station in the cell connects the mobile unit to the MSC which in turn is connected to the PSTN. The call is initiated, processed and terminated and this entire process is monitored and controlled by MSC.

Cellular Mobile Telephone Operation

When a subscriber makes a call by dialing the number and pressing the send button.

1. First the cellular phone scans the nearest base station in order to provide strongest signal for its use (21 control channels).

2. An origination message is sent by the cellular phone which includes MIN (Mobile Identification Number) as well as ESN (Electronic Serial Number) and the number that has been dialed.

3. Once the cellular service provides verified that the subscriber is among its customers based on the MIN & ESN. The base station send a channel assignment message to the cellular phone.

4. The cellular phone tunes into the assigned channel and the call begins.

5. All of this happened by the time the subscriber hears the ringing or busy signal on the other end of the phone.

6. Call termination: when the mobile user turns off the transmitter, a particular signal transmits to the cell site and both sides free the voice channel.

7. Hand off procedure.

1.4 TRENDS IN CELLULAR RADIO AND PERSONAL COMMUNICATION

Since 1989, there has been enormous activity throughout the world to develop personal wireless systems. Personal communication Services (PCS) 2 GHz frequency band originated in the united kingdom. Personal communication Network (PCN) is a means of improving international competitiveness in the wireless field while developing new wireless systems and service for citizens. The terms PCN and PCS are often used interchangeably. PCN refers to a wireless networking concept where any user can make or receive calls using a light-weight, personalized communication. PCS refers to new wireless system that incorporate more network feature and more personalized than existing cellular radio systems. Today, cellular and PCS are identical in functionality and different only in frequency Band. IEEE 802.11 is developing standard for wireless access between computers inside buildings. The European Telecommunications standard Institute (ETSI) is also developing the 20 mbps HIPERLAN Standard for indoor wireless networks.

1.5 FREQUENCY REUSE

A radio channel consists of a pair of frequencies, one for each direction of transmission that is used for full duplex operation.

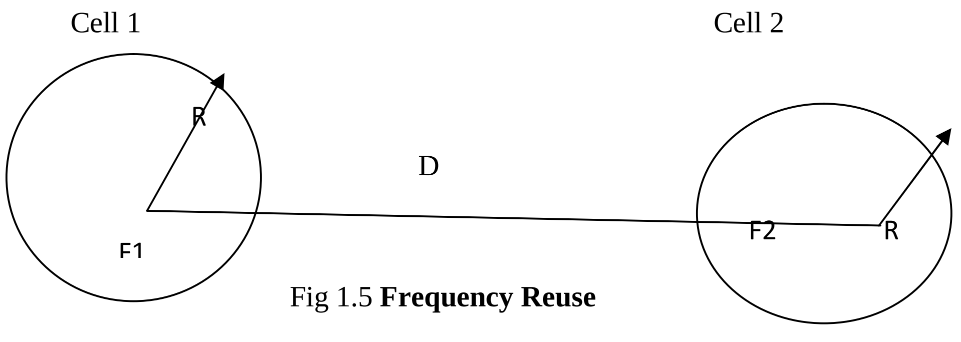


Fig 1.5 **Frequency Reuse**

A particular radio channel, say ‘F’, used in one geographic zone to call a cell say ’C’ with a coverage radius ‘R’ can be used in another cell with the same coverage radius at a distance ‘D’ away.

Frequency reuse is the core concept of the cellular mobile radio system. In this frequency reuse system, users in different geographical locations (different cells) may simultaneously use the same frequency channel.

The frequency reuse system can drastically increase the spectrum efficiency. But if the system is not properly designed, interference may occur. Interference due to the common use or the same channel is called Cochannel Interference and this is major concern in the concept or frequency reuse schemes:

Frequency reuse concept can be used in the time domain and space domain.

Time Domain	-	Same frequency is used in different time slots, it is called Time Division Multiplexing.
Space Domain	-	Frequency reuse in the space domain can be divided into two categories.

1. Same frequency assigned in two different geographic area, such as AM or FM radio stations using the same frequency in different cities.
2. Same frequency repeatedly used in a same geographical general area in one system, the scheme is used in cellular systems. There are many Cochannels in the system.

K reuse pattern - K=4, 7, 12 and 19

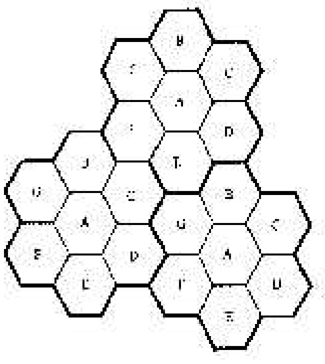


Fig 1.5.1 **K Reuse Pattern**

Frequency Reuse Distance:

The minimum distance which allows the same frequency to be reused will depend on many factors.

1. Number of Cochannel cells.

2. The terrain contour's type
3. The Antenna height
4. The transmitted power at each cell site.

The frequency reuse Distance D is given by

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

K - Frequency reuse pattern

R - Radius

The smallest value of K can be K = 3 when I = j=1

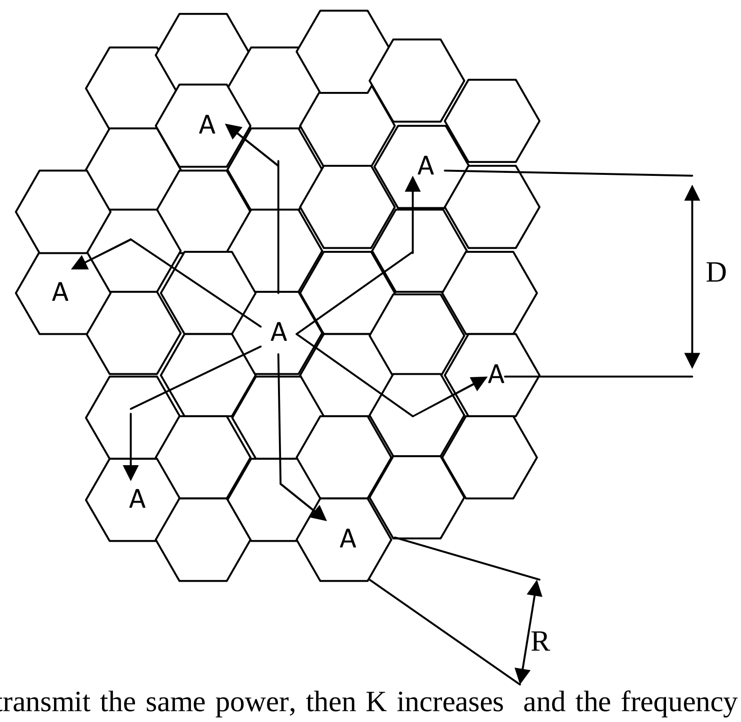
$$K = i^2 + ij + j^2$$

i & j are shift parameters in N cell reuse pattern

N – Cell reuse pattern

$$K = i^2 + ij + j^2 \qquad I = 2 \;\; j = 1$$

$$K = 4 + 2 + 1 = 7$$



If all the cell sites transmit the same power, then K increases and the frequency reuse distance 'D' increases. This increased D reduces the chance that cochannel interference may occur.

1.6 CHANNEL ASSIGNMENT

Channel Assignment defined as allocating channels to the cell sites and also to the mobile unit.Channel Assignment strategies can be classified as either fixed or dynamic. The choice of Channel Assignment strategy impacts the performance of the system, particularly as how calls are managed when a mobile user is handoff from one cell to another. In a fixed assignment strategy, each cell is allocated a predetermined set of voice channels. Any call attempt within the cell can only be served by the unused channels in that particular cell. If all the channels in that cell are occupied, the call is blocked and the subscribed does not receive service. One approach to overcome this situation is Borrowing Strategy, a cell is allowed to borrow channels from the neighboring cell if all of its own channels are already occupied. The MSC supervises such borrowing procedures and ensures that the borrowing of a channel does not interfere with any of the calls in progress in the donor cell.

In a dynamic channel assignment strategy, voice channels are not allocated to different cells permanently. Instead, each time a call request is made, the serving base station requests a channel from the MSC. The switch then allocates a channel to the requested cell following an algorithm that takes into account the livelihood of future blocking. Within the cell, the reuse distance of the channel and other cost functions.

Accordingly the MSC only allocates a given frequency if that frequency is not presently in use in the cell or any other cell which falls within the minimum restricted distance of frequency reuse to avoid cochannel Interference. Dynamic channel assignment reduce the call blocking, which increases the turning capacity of the system. Since all the available channels in the market are accessible to all of the cells. Dynamic channel assignment strategies require the MSC to collect real-time data on channel occupancy traffic distribution and radio signal strength indications of all channels on a continuous basis.

1.7 HAND OFF MECHANISM

Handoff refers to a process of transferring an ongoing call or data session from one channel connected to the core network to another. The channel change due to handoff may be through a time slot, frequency band, codeword, or combination of these for time-division multiple access (TDMA), frequency-division multiple access (FDMA), code-division multiple access (CDMA), or a hybrid scheme. Handoff is also called as ‘Handover’.

Reasons for a Handoff to be conducted:

- To avoid call termination when the phone is moving away from the area covered by one cell and entering the area covered by another cell.
- When the capacity for connecting new calls of a given cell is used up.
- When there is interference in the channels due to the different phones using the same channel in different cells.
- When the user behaviors change

Importance of Handling Handoff:

Customer satisfaction is very important in cellular communication and handling handoff is directly related to customer satisfaction. Effective handling of handoff leads to improved reception and fewer dropped calls and results in customer satisfaction which is very important in Mobile communication. Handoff is very common and most frequently occurred in cellular communication so it should be handled efficiently for desired performance of the cellular network. Handoff is very important for managing the different resources in Cellular Systems. Handoffs should not lead to significant interruptions even though resource shortages after a handoff cannot be avoided completely. Thus handling handoffs is very much important for a desired interruption free cellular communication.

Design Considerations for handoff:

The main goal while designing handoff is to reduce major changes to existing networks esp. at lower levels. This will ensure that existing networks will continue to function as before without requiring current users to change to the new approach.

Types of Handoffs:-

Handoff is the mechanism which transfers an ongoing call from one cell to another cell as users are near to the coverage area of the neighbouring cell. If handoff does not occur quickly, the Quality of Service (QoS) will degarde below an acceptable level and the connection will be lost.

Handoffs are classified into two categories – *hard and soft handoffs*, which are further divided among themselves.

Hard handoff:

A hard handoff is essentially a "*break before make*" connection. Here the link to the prior base station is terminated before or as the user is transferred to the new cell's base station. This means that the mobile is linked to no more than one base station at a given time. A hard handoff occurs when users experience an interruption during the handover process caused by frequency shifting. A hard handoff is perceived by network engineers as event during the call. These are intended to be instantaneous inorder to minimize the disruption of the call. Hard handoff can be further divided as intra and inter-cell handoffs.

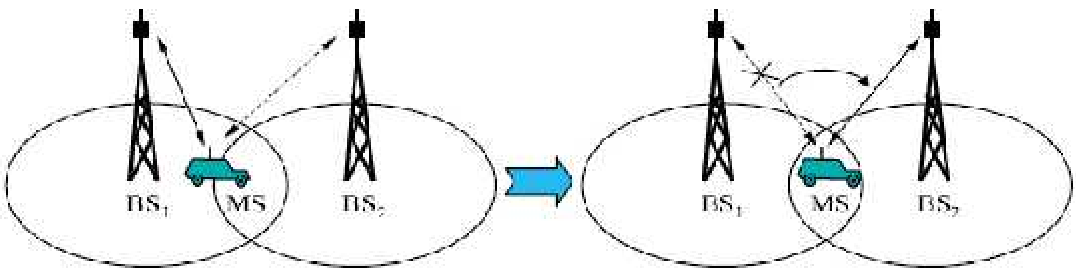


Fig 1.7.1 **Hard Handoff between the MS and BSS**

Intra and inter-cell handoffs: In intra-cell handoff the source and target are one and the same cell and only the used channel is changed during the handoff. The purpose of intra-cell handoff is to change a channel, which may be interfered, or fading with a new clearer or less fading channel. In inter-cell handoff the source and the target are different cells (even if they are on the same cell site). The purpose of the inter-cell handoff is to maintain the call as the subscriber is moving out of the area of the source cell and entering the area of the target cell. Finally, Hard handoff is permitted between members of different softzones, but not between members of the same softzone. This is primarily used in FDMA and TDMA.

Soft handoff:

Soft handoff is also called as Mobile Directed Handoff as they are directed by the mobile telephones. Soft handoff is the ability to select between the instantaneous received signals from different base stations. Here the channel in the source cell is retained and used for a while in parallel with the channel in the target cell. In this the connection to the target is established before the connection to the source is broken, hence this is called *"make-before-break"*. The interval during which the two connections are used in parallel, may be brief or substantial because of this the soft handoff is perceived by the network engineers as state of the call. Soft handoffs can be classified as Multiways and softer handoffs.

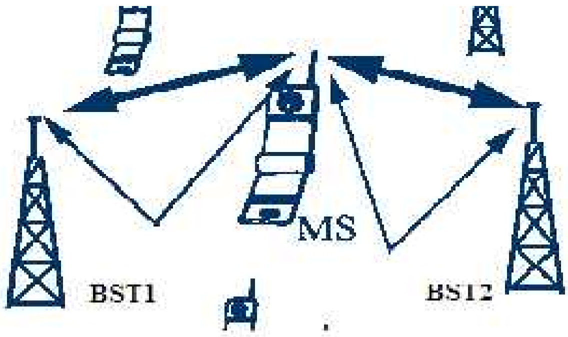


Fig 1.7.2 Soft Handoff between MS and BSTs

Multiways and softer handoffs: A soft handoff which involves using connections to more than two cells is a multiways handoff. When a call is in a state of soft handoff the signal of the best of all used channels can be utilized for the call at a given moment or all the signals can be combined to produce a clear signal, this type is called softer handoff.

In soft handoffs the chance that the call will be terminated abnormally are lower. Call could only fail if all the channels are interfered or fade at the same time. But this involves the use of several channels in the network to support just a single call. This reduces the number of remaining free channels and there by reducing the capacity of the network. Soft handoff is permitted between members of a particular softzone, but not between members of different softzones.

1.8 INTERFERENCE AND SYSTEM CAPACITY

There are two types of Interference they are

1. Cochannel Interference.
2. Non-Cochannel Interference or Adjacent channel Interference.

The frequency reuse concept is applied for cellular communication, where some cells user the same frequency and because of this there are chances of interference between those cochannel cells which is termed as cochannel interference. To avoid cochannel Interference the same frequency channel should be reused with a distance to D. Q is the cochannel reduction ration

$$Q = \frac{D}{R}$$

The interference due to the neighboring channel or adjacent channels which are not cochannel is known as non-cochannel interference. The adjacent channels should be allotted different carrier signals because there is a chance of overlapping of frequency which will lead to interference.

Near end far end ratio interference is an example to non-cochannel interference.

In one cell system

Motor vehicles in a given cell are usually moving some mobile units are close to the cell site and some are not.

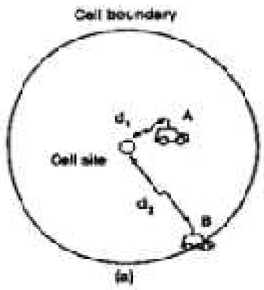


Fig 1.8 Near-end-far end interference a) In one cell System

The close-in mobile unit has a strong signal which causes adjacent channel interference. In this situation, near-end-far-end interference can occur only at the reception point in the cell site. A minimum separation of 5 B (5 channel band width) is required between each adjacent channel used within one cell.

In cells of two systems

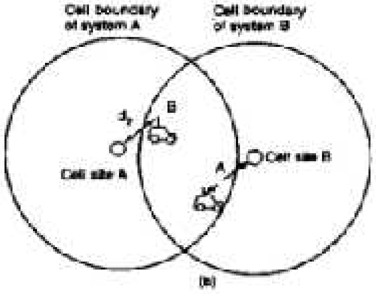


Fig 1.8 Near-end-far end interference b) In two System

Adjacent channel Interference can occur between two systems in a duopoly market system. In this situation, adjacent-channel Interference can occur at both; the cell site and the mobile unit.

For Instance, mobile unit A can be located at the boundary of its own home cell A in the system A but very close to cell B of system B.

The other situation would occur if mobile unit B were at the boundary of cell B of system A but very close to cell A of system A.

1.9 TRUNKING AND GRADE OF SERVICE

Cellular radio systems rely on trunking to accommodate a large number of users in a limited radio spectrum.

The concept of trunking allow a large number of users to share the relatively small number of channels in a cell by providing access to each user, on demand, from a pool of available channels., In a trunked radio system, each user is allocated a channel on a per call basis and upon termination of the call, the previously occupied channel is immediately returned to the pool of available channels. When a particular user requests service and all of the radio channels are already in use, the user is blocked or denied access to the system. In some systems, a queue may be used to hold the requesting users units until a channel become available.

Grade of service is a measure of the ability of a user to access a trunked system during the busiest hour. The busy hour is based upon customer demand at the busiest hour during a week, month or a year. The busy hours for cellular radio systems typically occur during rush hours between 4 PM & 6PM on a Thursday or Friday evening.

Highlights

Call blocking

Busy tone etc.

Call delay, can dropping.

1.10 IMPROVING COVERAGE AND CAPACITY IN CELLULAR SYSTEMS

When cellular service providers build their networks, their networks are designed to provide coverage to the area of desire with the expectation of possible increase in population in the near future. For example, a company may design a cellular network to cover a city of area 1000 km2 with population of 1,000,000 people today assuming that 15% of the population will subscribe to their cellular service, or 150,000 people. However, to accommodate possible increase in the percentage of subscribers or the same percentage of subscribers but an increase in population, the network designer may build the network to provide acceptable GOS for 200,000. Such move guarantees that the network will need any expansion for possibly 5 years. In some cases, it may be difficult to predict the need for network expansion or even when network expansion is predictable, the time for network expansion arrives. There are several techniques to expand an already existing network or to add more capacity to a network being built. In the following two techniques were discussed below.

1.10.1 Cell splitting

When the mobile traffic is congested to face the situation the larger cell can be subdivided into smaller cells. The technique is termed as "cell splitting" Installation of micro cell between the existing cells, capacity increases.

Usually the new radius is one-half the original radius.

$$\text{New cell radius} = \frac{\text{Old Cell radius}}{2}$$

$$\text{New cell area} = \frac{\text{Old Cell area}}{4}$$

There are two kinds of cell splitting techniques.

1. Permanent splitting: The installation of every new Split Cell has to be planned ahead of time; the number of channels, the transmitted power, the assigned frequencies, the choosing of the cell-site selection and the traffic load consideration should all be considered.

2. Dynamic splitting : This scheme is based on utilizing the allocated spectrum efficiency in real-time. The algorithm for dynamically splitting cell sites is a tedious job. Since we cannot afford to have one single cell unused during cell splitting.This technique does not require new base terminal station. The existing base station itself reconfigured using omni-directional antennas.

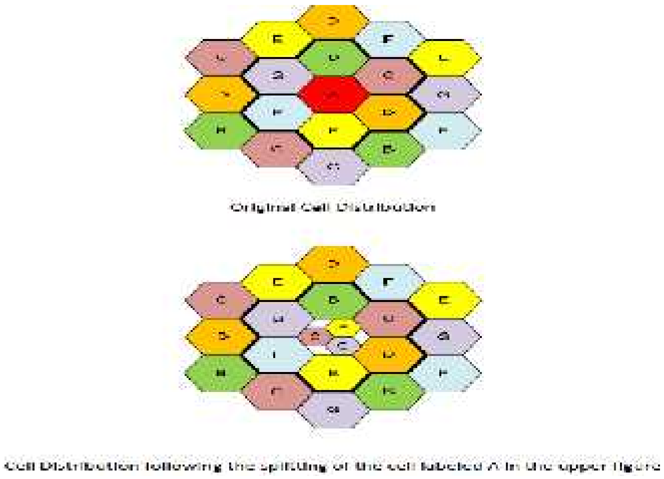


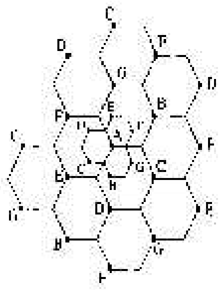
Fig 1.10 Cell Splitting

Note that following cell splitting, the new small cells are reassigned new frequencies that do not cause co-channel interference with adjacent cells as shown in the above figure. In addition, the power transmitted in the small cells is reduced compared to the power transmitted in the large cells as it would require much less power to cover the cell compared to the large cells. In fact the power has to be reduced by a factor of

$$\frac{P_{\text{Transmitted in Small Cell}}}{P_{\text{Transmitted in Large Cell}}} = \left(\frac{R_{\text{Small Cell}}}{R_{\text{Large Cell}}}\right)^{n}$$

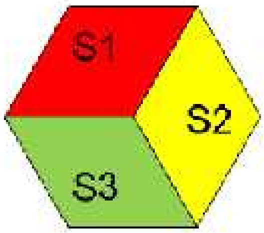
For example, if the cell radius of the small cells is half the radius of the large cell and the path loss exponent $n = 4$, the power transmitted by the tower of the small cell is only 1/16 that of the power transmitted by the tower of the large cell. In addition to the advantage of having a higher network capacity due to cell splitting, the reduced transmitted power, especially by the mobile phone, is another major advantages because it increases the battery life of these mobile phones. The main disadvantage of cell splitting is that it requires the construction of new towers, which is very costly.

Example of cell splitting

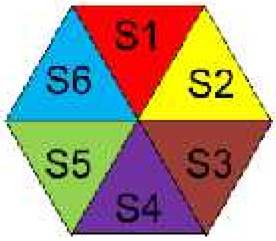


The base stations are placed at corners of the cells and the area served by base station A is assumed to be saturated with traffic. (i.e. the blocking of base station A exceeds acceptable rates) New base stations are needed in the region to increase the number of channels in the area and reduce the area served by the single base station. Note that in the figure that the original base station. A has been surrounded by six new micro cell base station. The smaller cells were added in such a way as to preserve the frequency reuse plan of the system.

Cell sectoring technique increases the capacity via a different strategy. In this method, a cell has the same coverage space but instead of using a single omni-directional antenna that transmits in all directions, either 3 or 6 directional antennas are used such that each of these antennas provides coverage to a sector of the hexagon. When 3 directional antennas are used, 120°sectoring is achieved (each antenna covers 120°), and when 6 directional antennas are used, 60°sectoring is achieved (each antenna covers 60°).



120° Cell Sectoring



60° Cell Sectoring

Fig 1.11 Cell sectoring

Dividing the cells into sectors actually reduces the network capacity because the channels allocated to a cell are now divided among the different sectors. In fact, handoff takes place when a cell phone moves from one sector to another in the same cell. The gain in network capacity is achieved by reducing the number of interfering co-channel cells. If sectoring is done in a way that channels assigned to a particular sector are always at the same direction in the different cells (i.e., group A of channels is assigned to the sector to the left of the tower in all cells, and group B of channels is assigned to the sector at the top of all cells, and so on), each sector causes interference to the cells that are in its transmission angle only. Unlike the case of no sectoring where 6 interfering co-channel cells from the first-tier co-channels cells cause interference, with 120° sectoring, 2 or 3 co-channel cells cause interference and with 60° sectoring, 1 or 2 co-channel cells cause interference. The number of cochannel interfering cells depends on the cluster shape and size. By having less than 6 interfering first-tier co-channel cells causing interference, the SIR is increased for the same cluster size. This allows us to reduce the cluster size and achieve the same original SIR, which directly increases the network capacity.

UNIT - 2

2. MOBILE RADIO PROPAGATION

2.1 FREE SPACE PROPAGATION MODEL

Although EM signals when traveling through wireless channels experience fading aspects due to various aspects, 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

$$P_r(d) = \frac{P_t G_t G_r a^2}{(4®)^2 d^2 L} \quad \rightarrow (2.1)$$

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 depended upon line attenuation 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

$$G = \frac{4® A_e}{a^2} \quad \rightarrow (2.2)$$

Path loss for the free space model with antenna gains

$$PL(dB) = 10\log\frac{P_t}{P_r} = -10\log\left(\frac{G_t G_r a^2}{(4®)^2 d^2}\right) \quad \rightarrow (2.3)$$

The field 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, d_f, which is dependent upon the largest dimension of the antenna as

$$d_f = \frac{2D^2}{a} \quad \rightarrow (2.4)$$

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

$$P_r(d) = P_r(d_0)\left(\frac{d_0}{d}\right)^2 \quad \rightarrow (2.5) \qquad d \geq d_0 \geq d_f$$

2.2 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.

2.2.1 Reflection

Reflection occurs when incident electromagnetic waves are partially reflected when they impinge on obstructions of different electrical properties. A propagating electromagnetic wave impinges on objects the size of which are large compared to its wavelength, such as the surface of the earth, buildings, walls, etc. The electromagnetic radio waves get reflected from tall building structures which have a good amount of conductivity. Reflection can also occur due to metal reinforcement. The extent of reflection of radio waves depends on the composition and surface characteristics of the objects. The angle of reflection is equal to the angle at which the wave strikes the object and is measured by the Fresnel reflection coefficient. Upon reflection, the signal strength of the radio wave gets attenuated that depends on many factors like the frequency of the radio waves, the angle of incidence, and the nature of the medium including its material properties, thickness, homogeneity, etc. Generally, higher frequencies reflect more than lower frequencies.

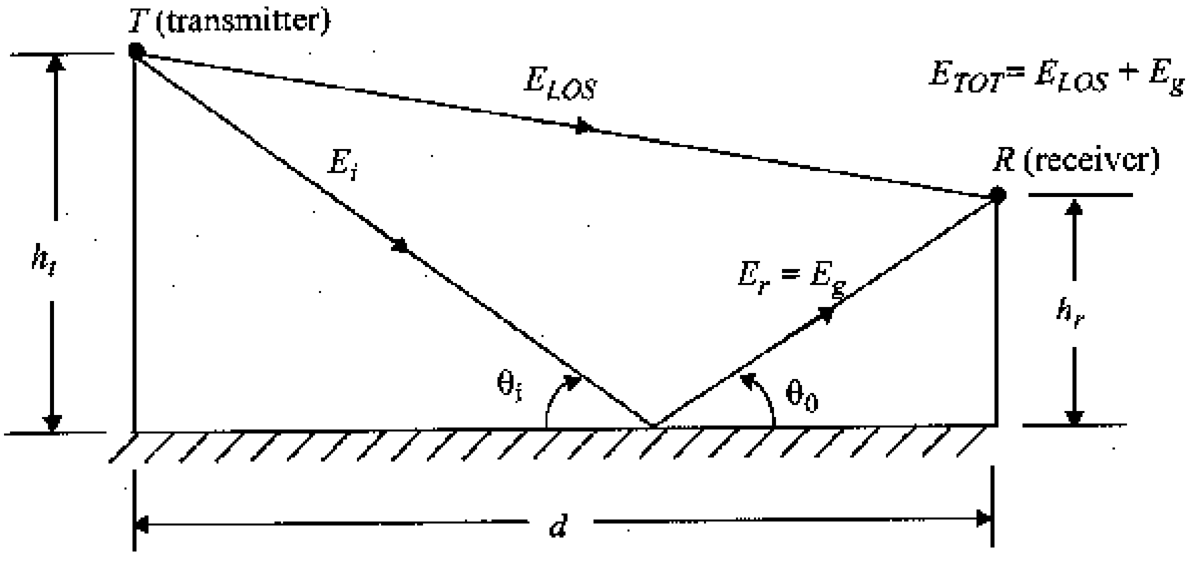


Fig 2.2.1 Reflection

As an instance, let a ground-reflected wave near the mobile unit be received. Because the ground-reflected wave has a 180^{o} phase shift after reflection, the ground wave and the line-of-sight wave may tend to cancel each other, resulting in high signal attenuation. The vector sum of the phases of the multipath received signals may give a resultant zero amplitude at certain time instants and large signal amplitude at some other time. Most of the times, the vectorial addition of these multipath reflected signals produce an undetectable signal. Further, because the mobile antenna is lower than most human-made structures in the operational area, multipath interference occurs. These reflected waves may interfere constructively or destructively at the received. In outdoor urban area, the reflection mechanism often loses its importance because it involves multiple reflections that reduce the strength of the signal to negligible values. However, reflection mechanisms often dominate radio propagation in indoor applications. The reflections are a source of multipath signals which case low strength in signal reception. Reflection results in a large-scale fading of the radio signals.

2.2.2 Diffraction

Diffraction is referred to the change in wave pattern caused by interference between waves that have reflected from a surface or a point. It is based on Huygen's principle which states that all points on a wavefront can be considered as point sources for production of secondary wavelets that can combine to produce a new wavefront in the direction of propation of the signal. Diffraction occurs when the radio path between a transmitted and received is obstructed by a surface with sharp irregular edges. Waves bend around the obstacle, even when a line-of –sight condition does not exist. It causes regions of signal strengthening and weakening irregularly. Diffraction can also occur in different situations such as when radio waves pass through a narrow slit or the edge of a reflector or reflect off from two different surfaces approximately one wavelength apart. At higher frequencies, diffraction depends on the geometry of the object, as well as the amplitude., phase and polarisation of the incident wave at the point of diffraction. Figure:depicts a simple case of diffraction of a radio signal.

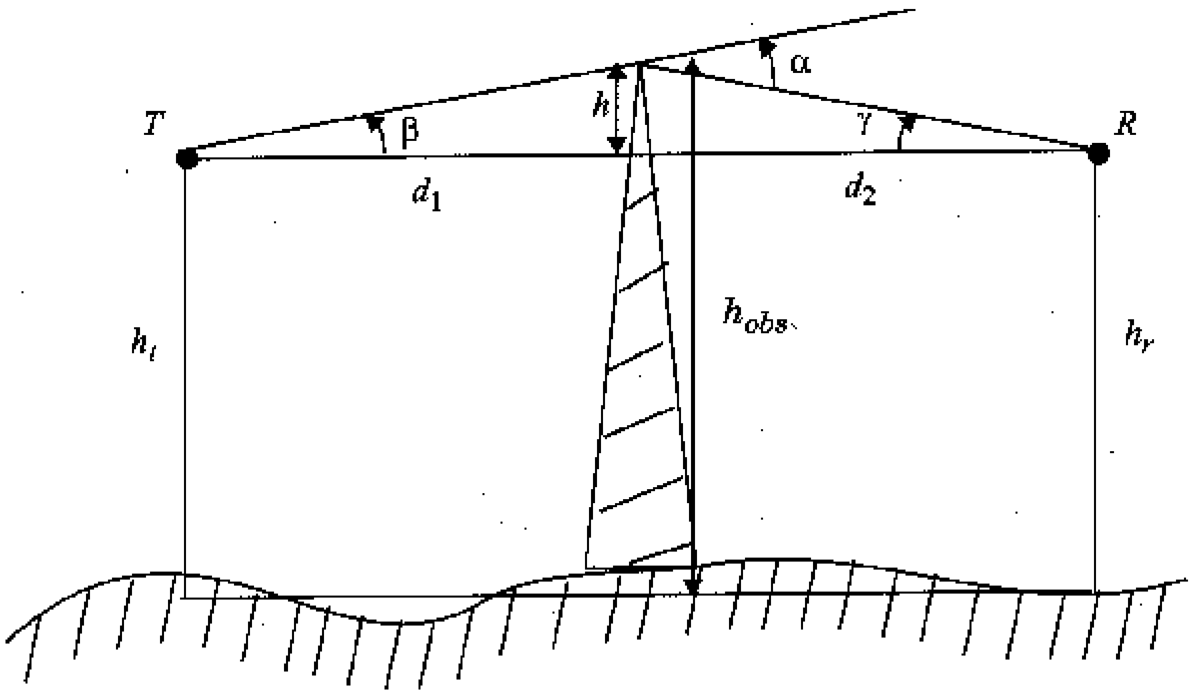


Fig 2.2.2 Diffraction

Diffraction is a description of how a radio signal propages around and over on obstruction, and is measured in dB. Diffraction often results in *small-signal fading.* In effect, diffraction results in propagation into shadow regions because the diffracted field can reach a receiver, which is not in the line-of-sight of the transmitter. Because a secondary wavelet is created, it suffers a signal loss much greater than that experienced via reflection. Although the received signal strength decreases rapidly as a receiver moves deeper into the shadow region, the diffraction field still exists and often produces useful signal strength.

Consequently, diffraction is an important phenomenon of propagation impairment in outdoor applications such as in micro-cellular areas where signal transmission through buildings is virtually impossible. It is less consequential in indoor applications where a diffracted signal is extremely weak compared to a reflected signal or a signal that is transmitted through a relatively thin wall.

In mobile communications systems, diffraction loss occurs from the blockage of secondary waves such that only a portion of the energy is diffracted around an obstacle., Most cellular systems operate in urban areas where there is no direct line-of-sight path between the transmitter and the receiver(either from the cell-site to the mobile unit or vice-versa), and where the presence of high-rise buildings causes severe diffraction loss. In many practical situations, the propagation path may consist of more than one obstruction. For example, in hilly terrains, the total diffraction loss must be computed due to all the obstacles.

2.2.3 Scattering

Scattering is a special case of reflection caused by irregular objects such as walls with rough surfaces, vehicles, foliage, traffic signs, lamp posts, and results in many different angles of reflection and scatter waves in all directions in the form of spherical waves. Thus, due to availability of numerous objects, scattering effects are difficult to predict. Scattering occurs when the size of objects is comparable or smaller than the wave-length of the propagating radio wave, and where the number of obstacles per unit volume is large. Figure:depicts a typical case of scattering of a radio signal.

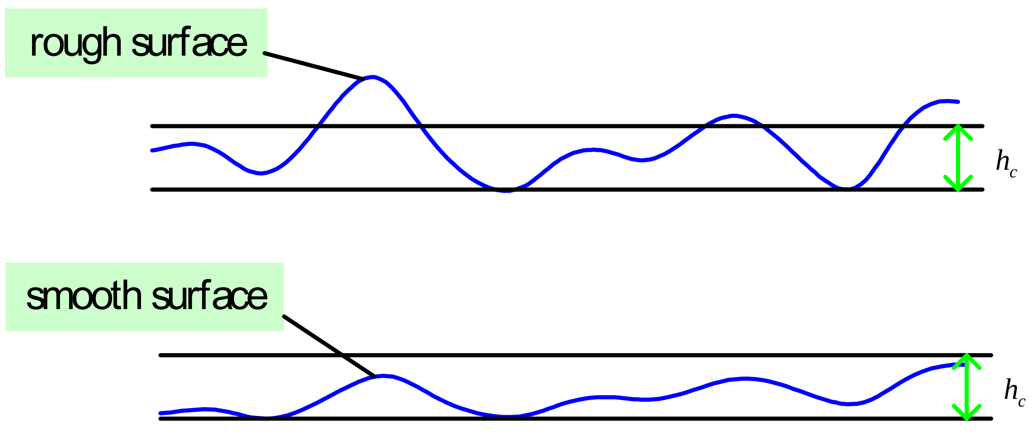


Fig 2.2.3 Scattering

Propagation in many directions results in reduced received-signal power levels, especially far from the scatterer. So an incoming radio signal is scattered into several weaker outgoing radio signals. As a result, the scattering phenomenon is not significant unless the receiver or transmitter is located in a highly noisy environment. In a mobile radio environment, scattering provides additional radio energy level at the receiver to what has been predicted by reflection and diffraction models along. In radio channels, knowledge of the physical location of large distance objects, which induce scattering, can be used to accurately predict scattered signal strength levels. In a mobile radio environment heavy foliage often causes scattering. Scattering too results in small-scale effects.

These three impairments to free-space propagation influence system performance in various ways depending on local conditions and as the mobile unit moving within a cell in a cellular system.

- If a mobile unit has a clear line-of-sight condition with the cell-site then only reflection may wave a significant effect whereas diffraction and scattering have minor effects on the received signal levels.
- If there is no clear line-of-sight condition, such as in an urban area at busy street level, then diffraction and scattering are the primary means of signal reception.

One Major adverse effect of multipath propagation is that multiple copies of a signal may arrive at different phases. If these phases add destructively, the signal level relative to noise declines, making signal detection at the receiver much more difficult and unreliable.

The second major effect of multipath propagation is increase in received data errors due to inter-symbol interference in digital transmission. As the mobile unit moves, the relative location of various objects also changes; hence inter-symbol interference increases to the extent that makes it difficult to design signal processing techniques that will filter out multipath effects in order to recover the intended signal with fidelity.

An extreme form of signal attenuation is blocking or shadowing of radio signals, which is caused by obstacles much larger in size than the wavelengths of the operating signals such as a small wall, trees, or a large vehicle on the street.

Another form of propagation effect is the effect of refraction. Refraction occurs because the velocity of the electromagnetic waves depends on the density of the medium through which it travels. Waves that travel into a denser medium are bent towards the medium. This is the reason for line-of-sight radio waves being bent towards the earth since the density of the atmosphere is higher closer to the earth.

2.3 LINK BUDJET ANALYSIS

Most radio propagation models are derived using a combination of analytical and empirical methods.The empirical approach is based on fitting curves or analytical expressions that recreate a set of measured data.In the empirical approach it has the advantage that considered all propagation factors, both known and unknown, through actual field measurements. It consists of two models they are Log distance path loss model and log normal shadowing. By using path loss models to estimate the received signal level as a function of distance, it becomes possible to predict the SNR for a mobile communication system.

2.3.1Log-distance Path Loss Model

According to this model the received power at distance d is given by.

$$\overline{PL}(d)(dB) = \overline{PL}(d_0) + 10n\log\left(\frac{d}{d_0}\right)$$

The value of n varies with propagation environments. The value of n is 2 for free space. The value of n varies from 4 to 6 for obstruction of building, and 3 to 5 for urban scenarios. The important factor is to select the correct reference distance d_0. For large cell area it is 1 Km, while for micro-cell system it varies from 10m-1m.

Limitations:

Surrounding environmental clutter may be different for two locations having the same transmitter to receiver separation. Moreover it does not account for the shadowing effects

2.3.2Log Normal Shadowing

The equation for the log normal shadowing is given by,

$$PL(d) = \overline{PL}(d) + X_{\pm} = \overline{PL}(d_0) + 10n\log\left(\frac{d}{d_0}\right) + X_{\pm}$$

$X_{\pm}$: zero-mean Gaussian distributed random variable (in dB) with standard deviation ±

The probability that the received signal level will exceed a certain value can be calculated from

$$\Pr[P_r(d) > ¥] = Q\left(\frac{¥ - \overline{P_r(d)}}{\pm}\right)$$

where $\overline{P_r(d)} = P_t(d) - \overline{PL}(d)$

2.4 OUTDOOR PROPAGATION MODELS

There are many empirical outdoor propagation models such as Longley-Rice model, Durkin's model, Okumura model, Hata model etc. Longley-Rice model is the most commonly used model within a frequency band of 40 MHz to 100 GHz over different terrains. Certain modifications over the rudimentary model like an extra urban factor (UF) due to urban clutter near the receiver is also included in this model

2.4.1Okumura 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 200 MHz to 1900

MHz and distances of 1 Km to 100 Km.It can be applicable for base station eective antenna heights (h_t) 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 path-loss formula of Okumura at distance d parameterized by the carrier frequency f_c is given by

$$P_L(d)dB = L(f_c; d) + A_{mu}(f_c; d) - G(h_t) - G(h_r) - G_{AREA}$$

where $L(f_c; d)$ is free space path loss at distance d and carrier frequency f_c, $A_{mu}(f_c; d)$ 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:

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

$$G(h_r) = 10 \log_{10}(h_r/3); \quad h_r <= 3m$$

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

Correlation factors related to terrain are also developed in order to improve the models accuracy. 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.

2.4.2 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 simpli_es the calculation of path loss because it is closed form formula and it is not based on empirical curves for the dierent 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

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

and for larger cities at a frequencies $f_c > 300$ MHz by

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

else it is

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

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

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

Unlike the Okumura model,the Hata model does not provide for any specific path-correlation factors. 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.

2.5 INDOOR PROPAGATION MODELS

An indoor propagation environment is more hostile than a typical outdoor propagation environment. The indoor propagation model estimates the path loss inside a room or a closed area inside a building delimited by walls of any form. Phenomena like lack of a line-of-sight conditions, multipath propagation, reflection, diffraction shadow fading, heavy signal attenuation, close proximity of interference sources, and rapid fluctuations in the wireless channel characteristics have a significant influence on the received power in indoor propagation. Moreover, the ranges involved need to be of the order of 100 metres or less. Typically, multipath propagation is very important in indoor environments. Simple empirical propagation models are therefore not sufficient.,

The indoor propagation models are suitable for wireless devices designed for indoor application to approximate the total path loss an indoor wireless like may experience. Typically, such wireless devices use the lower microwave bands of around 2.4 GHz. However, the model applies to a much wider frequency range. The indoor propagation models can be used for picocells in cellular network planning.

2.5.1 Partition and building Penetration Losses

Generally, an indoor environment comprises of buildings having a variety of partitions and obstacles which cause additional propagation path loss. Different types of particulars are used such as those for forming part of the building structures, called *hard or fixed partitions,* and obstacles inside the building, called *soft or movable partitions.* In such a building in which wireless devices are used, signals may pass through fixed/movable furniture, computers, moving people within a room as well as through walls, doors, and/or windows if the piconet. To determine partition losses, the following assumptions may be made.

- Free space path loss occurs between partitions

- Signal strength may drop suddenly as it passes through partitions.
- The amount of partitions loss depends on the type of partitions.
- Signal losses between different floors of the same building exhibit special behavior and are required to be modeled separately.

Table 1: Typical partition losses in a building

S.No.	Type of Partition	Partition loss (dB)
1	Metalic	26
2	Aluminium wall	20.4
3	Concreter wall	13
4	Foil insulation within wall	3.9
5	Double plaster board wall	3.4
6	Cloth partition	1.4

Another important factor for indoor propagation when the transmitter is located outside the building is *building penetration loss.* Signal penetration within the building is a function of frequency of transmissions, height, and the building materials. Signal strength received inside the building increases with height, whereas penetration loss decreases with increasing frequency. Building penetration loss decreases at a rate of about 1.9 dB per floor from the ground floor up to the 15th floor and then starts increasing due to shadow effects of adjacent tall buildings. Building penetration loss on the ground floor of the building typically varies from 8 dB to 20 dB for a frequency range of 900 MHz to 2 GHz respectively. The number and size of windows as well as the type of materials used for construction of windows in the building have a significant contribution towards total building penetration loss. Building penetration loss behind windows is typically 6 dB less than that behind exterior walls. Moreover, plate glass has penetration loss of about 6 dB as compared to 3-30 dB from metallic lead-lined glass.

2.5.2 Log-distance Propagation Model

The log-distance path loss model is a radio propagation model that predicts the path loss which is encountered by a signal inside a building or densely populated areas over distance. The model is applicable to indoor propagation modeling. Log-distance path loss model is based on distance-power law, and is expressed as

$L_{pLog}(dB) = L_p(r_{0)\ (dB)\ +}\ 10$

Where $L_p(r_0)$ is the path loss at the reference distance r_0 (usually taken as 1 m), and is a function of frequency of transmission, f_c and distance between transmitter and receiver, r.

ນ is the path-loss propagation constant,

R is the distance between the transmitter and receiver in metres,

X_g is a normal random variable with zero mean, reflecting the attenuation caused by flat fading. In case of no fading, X_g is taken as 0. In case of only shadow fading or slow fading, this random variable may have Gaussain distribution with standard deviation in dB, resulting in log-normal distribution of the received power in watts. In case of only fast fading caused by multipath propagation, the corresponding gain may be modeled as a random variable with Rayleigh distribution or Ricean distribution.

Empirical measurements of path loss exponent and standard deviation corresponding to different indoor propagation conditions are shown below

Table 2: Different indoor propagation conditions

Type of indoor structure	Frequency, f_c	Path-loss exponent,	Standard deviation
Vacuum or infinite space	----	2.0	0
Grocery store	914 MHz	1.8	5.2
Retail store	914 MHz	2.2	8.7
Office with soft partition	900 MHz 1.9 GHz	2.4; 2.6	9.6; 14.1
Office with hard partition	1.5 GHz	3.0	7.0
Suburban home or street	900 MHz	3.0	7.0
Metal working factory	1.3GHz	1.6 for LOS; 3.3 for obstructive path	5.8 for LOS; 6.8 for obstructive path

Thus, the log-distance model is a combination of a modified power-distance law and a log-normal fading model that also uses empirical data. It is highly recommended to ultimately take field measurements to verify that the model is accurately characterizing the operation environment. If found necessary, the model can be corrected using the measured data.

2.6 SMALL-SCALE MULTIPATH PROPAGATION

The three most important effects- Rapid changes in signal strength over a small travel distance or time interval, Random frequency modulation due to varying Doppler shifts on different multipath signals, Time dispersion caused by multipath propagation delays. The Factors influencing small-scale fading is

- Multipath propagation: reflection objects and scatters
- Speed of the mobile: Doppler shifts

- Speed of surrounding objects
- Transmission bandwidth of the signal
- The received signal will be distorted if the transmission bandwidth is greater than the bandwidth of the multipath channel.
- Coherent bandwidth: bandwidth of the multipath channel.
- Doppler Shift - A mobile moves at a constant velocity *v*, along a path segment having length *d* between points *X* and *Y*.

Path length difference , $\Delta l = d\cos¯ = v\Delta t\cos¯$

Phase change, $\Delta¤ = \frac{2®\Delta l}{a} = \frac{2®v\Delta t}{a}\cos¯$

Doppler shift, $f_d = \frac{1}{2®}\cdot\frac{\Delta¤}{\Delta t} = \frac{v}{a}\cos¯$

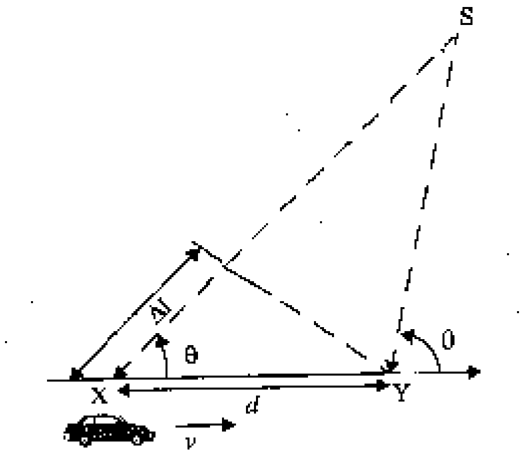


Fig 2.6 Small-Scale Multipath Propagation

2.7 IMPULSE RESPONSE MODEL OF A MULTIPATH CHANNEL

- A mobile radio channel may be modeled as a linear filter with a time varying impulse response
  - time variation is due to receiver motion in space
  - filtering is due to multipath

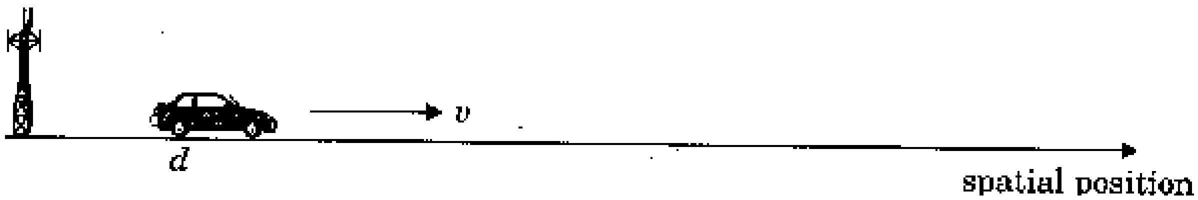


Fig 2.7 Impulse Response Model

- The channel impulse response can be expressed as *h(d,t).* Let *x(t)* represent the transmitted signal, then the received signal *y(d,t)* at position *d* can be expressed as

$$y(d,t) = x(t) \otimes h(d,t) = \int_{-\infty}^{\infty} x(²)h(d,t-²)d²$$

- For a causal system

$$y(d,t)=\int_{-\infty}^{t} x(\tau)h(d,t-\tau)d\tau$$

The position of the receiver can be expressed as

We have $$d=vt$$

$$y(vt,t)=\int_{-\infty}^{t} x(\tau)h(vt,t-\tau)d\tau$$

Since *v* is a constant, $y(vt,t)$ is just a function of *t*.

$$y(t)=\int_{-\infty}^{t} x(\tau)h(vt,t-\tau)d\tau$$

In general, the channel impulse response can be expressed

- t : time variation due to motion
- τ : channel multipath delay for a fixed value of *t*.

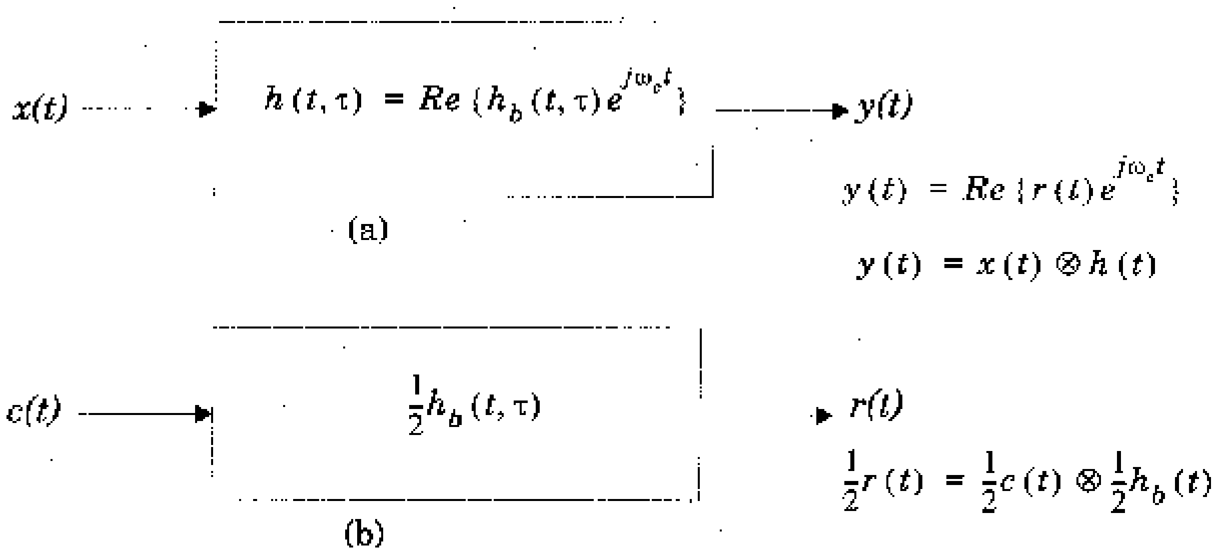
With the channel impulse response $h(t,\tau)$, we may have the output

$$y(t)=\int_{-\infty}^{t} x(\tau)h(t,\tau)d\tau = x(t)\otimes h(t,\tau)$$

For bandlimited bandpass channel, then $h(t,\tau)$ may be equivalently described by a complex baseband impulse response $h_b(t,\tau)$

- The equivalent baseband output

$$\frac{1}{2}r(t)=\frac{1}{2}c(t)\otimes\frac{1}{2}h_b(t,\tau) \quad \text{or} \quad r(t)=\frac{1}{2}c(t)\otimes h_b(t,\tau)$$



$$x(t)=\mathrm{Re}\{c(t)\exp(j\omega_c t)\}$$
$$y(t)=\mathrm{Re}\{r(t)\exp(j\omega_c t)\}$$

$$r(t)=\frac{1}{2}c(t)\otimes h_b(t,\tau)$$

Discretize the multipath delay axis τ into equal time delay segments called *excess delay bins*.

The baseband response of a multipath channel can be expressed as

$$h_b(t,\tau) = \sum_{i=0}^{N-1} a_i(t,\tau)\exp\left(j2\pi f_c \tau_i(t) + j\theta(t,\tau)\right)\delta(\tau - \tau_i(t))$$

$a_i(t,\tau)$ amplitude of the ith multipath component

$\tau_i(t)$ excess delay of ith multipath component

If the channel impulse response is assumed to be time invariant, the channel impulse response may be simplified as

$$h_b(\tau) = \sum_{i=0}^{N-1} a_i \exp\left(j\theta_i\right)\delta(\tau - \tau_i)$$

The impulse response may be measured by using a probing pulse which approximates a delta function.

$$p(t) \approx \delta(t - \tau)$$

2.8 SMALL-SCALE MULTIPATH MEASUREMENT

To determine the small – scale fading effects, a number of wideband channel sounding techniques have been developed

2.8.1 Direct RF Pulse System

A wideband pulsed bistatic radar usually transmits a repetitive pulse of width T_{bb} s, and uses a receiver with a wide bandpass filter ($\mathbf{BW} = \frac{2}{\mathbf{Tbb}}$ Hz). The signal is then amplified, envelope detected, and displayed and stored on a high speed oscilloscope. Immediate measurements of the square of the channel impulse response convolved with the probing pulse can be taken. If the oscilloscope is set on averaging mode, then this system provides a local average power delay profile.

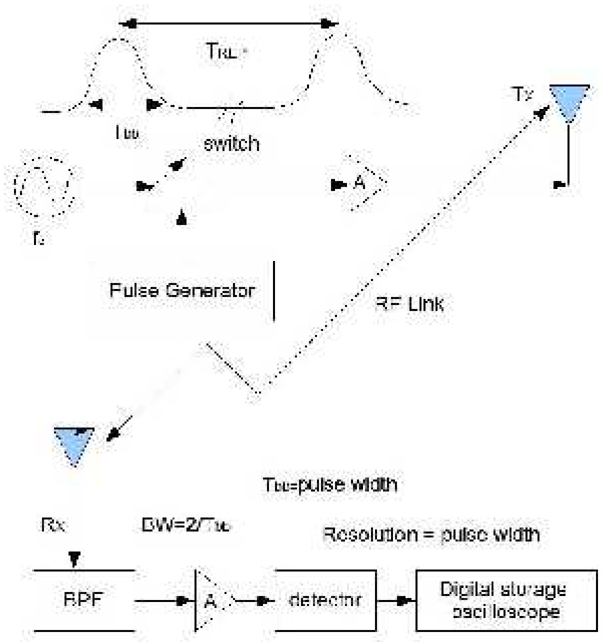


Fig 2.8.1 Direct RF pulsed channel IR measurement.

This system is subject to interference noise. If the first arriving signal is blocked or fades, severe fading occurs, and it is possible the system may not trigger properly.

2.8.2 Frequency Domain Channel Sounding

In this case we measure the channel in the frequency domain and then convert it into time domain impulse response by taking its inverse discrete Fourier transform (IDFT). A vector network analyzer controls a swept frequency synthesizer. An S-parameter test set is used to monitor the frequency response of the channel. The sweeper scans a particular frequency band, centered on the carrier, by stepping through discrete frequencies. The number and spacing of the frequency step impacts the time resolution of the impulse response measurement. For each frequency step, the S-parameter test set transmits a known signal level at port 1 and monitors the received signal at port 2. These signals allow the analyzer to measure the complex response, $S_{21}(\omega)$, of the channel over the measured frequency range. The $S_{21}(\omega)$ measure is the measure of the signal bw from transmitter antenna to receiver antenna (i.e., the channel).

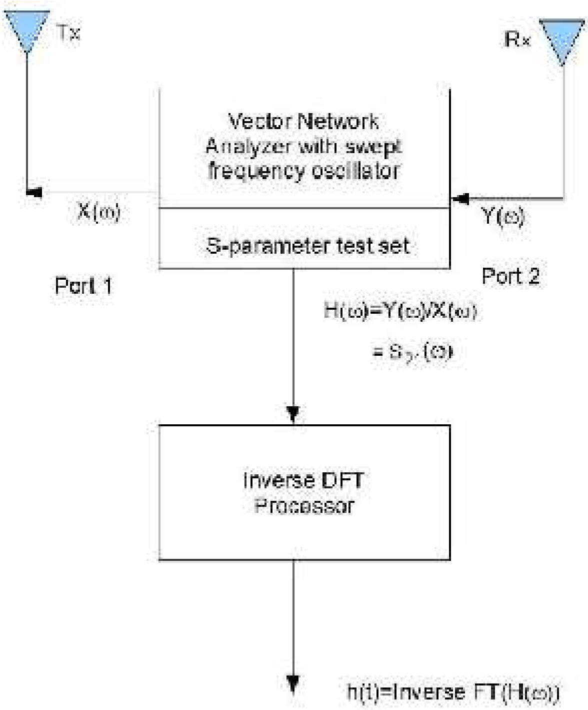


Fig 2.8.2 Frequency domain channel IR measurement

This system is suitable only for indoor channel measurements. This system is also non real-time. Hence, it is not suitable for time-varying channels unless the sweep times are fast enough.

2.9 MOBILE MULTIPATH CHANNEL PARAMETERS

To compare the different multipath channels and to quantify them, we define some parameters. They all can be determined from the power delay profile. These parameters can be broadly divided in to two types.

2.9.1 Time Dispersion Parameters:

These parameters include the mean excess delay, rms delay spread and excess delay spread. The mean excess delay is the first moment of the power delay profile and is defined as

Since the rms delay spread is the square root of the second central moment of the power delay profile

$$\pm_2 = \sqrt{\overline{2^2} - (\bar{2}^2)}$$

where

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

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

2.9.2 Frequency Dispersion Parameters:

To characterize the channel in the frequency domain, we have the following parameters.

(1) Coherence bandwidth: It is a statistical measure of the range of frequencies over which the channel can be considered to pass all the frequency components with almost equal gain and linear phase. When this condition is satisfied then we say the channel to be at.

Practically, coherence bandwidth is the minimum separation over which the two frequency components are affected differently. If the coherence bandwidth is considered to be the bandwidth over which the frequency correlation function is above 0.9, then it is approximated as

$$B_c \approx \frac{1}{50\pm_2}$$

However, if the coherence bandwidth is considered to be the bandwidth over which the frequency correlation function is above 0.5, then it is defined as

$$B_c \approx \frac{1}{5\pm_2}$$

The coherence bandwidth describes the time dispersive nature of the channel in the local area. A more convenient parameter to study the time variation of the channel is the coherence time. This variation may be due to the relative motion between the mobile and the base station or the motion of the objects in the channel.

(2) Coherence time: this is a statistical measure of the time duration over which the channel impulse response is almost invariant. When channel behaves like this, it is said to be slow

faded. Essentially it is the minimum time duration over which two received signals are affected differently. For an example, if the coherence time is considered to be the bandwidth over which the time correlation is above 0.5, then it can be approximated as

$$T_C \approx \frac{9}{16®f_m}$$

f_m : maximum Doppler shift given by $f_m = v/$ ª

2.10 MULTIPATH & SMALL-SCALE FADING

Multipath signals are received in a terrestrial environment, i.e., where different forms of propagation are present and the signals arrive at the receiver from transmitter via a variety of paths. Therefore there would be multipath interference, causing multi-path fading. Adding the effect of movement of either Tx or Rx or the surrounding clutter to it, the received overall signal amplitude or phase changes over a small amount of time. Mainly this causes the fading.

2.10.1 Fading

The term fading, or, small-scale fading, means rapid fluctuations of the amplitudes, phases, or multipath delays of a radio signal over a short period or short travel distance. This might be so severe that large scale radio propagation loss effects might be ignored.

2.10.2 Multipath Fading Effects

In principle, the following are the main multipath effects:

1. Rapid changes in signal strength over a small travel distance or time interval.
2. Random frequency modulation due to varying Doppler shifts on different multipath signals.
3. Time dispersion or echoes caused by multipath propagation delays.

2.10.3 Factors Influencing Fading

The following physical factors influence small-scale fading in the radio propagation channel:

(1) Multipath propagation : Multipath is the propagation phenomenon that results in radio signals reaching the receiving antenna by two or more paths.The effects of multipath include constructive and destructive interference, and phase shifting of the signal.

(2) Speed of the mobile: The relative motion between the base station and the mobile results in random frequency modulation due to different doppler shifts on each of the multipath components.

(3) Speed of surrounding objects : If objects in the radio channel are in motion, they induce a time varying Doppler shift on multipath components. If the surrounding objects move at a greater rate than the mobile, then this effect dominates fading.

(4) Transmission Bandwidth of the signal : If the transmitted radio signal bandwidth is greater than the bandwidth" of the multipath channel (quantified by coherence bandwidth), the received signal will be distorted.

2.11 TYPES OF SMALL-SCALE FADING

The type of fading experienced by the signal through a mobile channel depends on the relation between the signal parameters (bandwidth, symbol period) and the channel parameters (rms delay spread and Doppler spread). Hence we have four different types of fading. There are two types of fading due to the time dispersive nature of the channel.

2.11.1 Fading Effects due to Multipath Time Delay Spread

Flat Fading

Such types of fading occurs when the bandwidth of the transmitted signal is less than coherence bandwidth of the channel. Equivalently if the symbol period of the signal is more than the rms delay spread of the channel, then the fading is at fading.

So we can say that at fading occurs when $B_S << B_C$ $T_S >> \pm_2$

where BS is the signal bandwidth and BC is the coherence bandwidth. Also where TS is the symbol period and $\pm_2$ is the rms delay spread. And in such a case, mobile channel has a constant gain and linear phase response over its bandwidth.

Frequency Selective Fading

Frequency selective fading occurs when the signal bandwidth is more than the coherence bandwidth of the mobile radio channel or equivalently the symbols duration of the signal is less than the rms delay spread.

$B_S > B_C$ And $T_S > \pm_2$

At the receiver, we obtain multiple copies of the transmitted signal, all attenuated and delayed in time. The channel introduces inter symbol interference.

2.11.2 Fading Effects due to Doppler Spread

Fast Fading

In a fast fading channel, the channel impulse response changes rapidly within the symbol duration of the signal. Due to Doppler spreading, signal undergoes frequency dispersion leading to distortion. Therefore a signal undergoes fast fading if

$$T_S > T_C \quad \text{and} \quad B_S < B_D$$

where T_C is the coherence time . where B_D is the Doppler spread. Transmission involving very low data rates suffer from fast fading.

Slow Fading

In such a channel, the rate of the change of the channel impulse response is much less than the transmitted signal. We can consider a slow faded channel a channel in which channel is almost constant over atleast one symbol duration. Hence

$$T_S << T_C \quad \text{and} \quad B_S >> B_D$$

We observe that the velocity of the user plays an important role in deciding whether the signal experiences fast or slow fading.

2.12 STATISTICAL MODELS FOR MULTIPATH PROPAGATION

Several multipath models are used to explain the observed statistical nature of the mobile channel

2.12.1 Ossana Model

The first model was developed by ossana based on interference of waves incident and reflected from the flat sides of randomly located buildings. Ossana's model is inflexible and in appropriate for urban areas where the direct path is almost always blocked by buildings or other obstacles.

2.12.2 Clarke's Models for Flat Fading

- Clark developed a model where the statistical characteristics of the electromagnetic fields of the received signal are deduced from scattering. The model assumes a fixed transmitter with a vertically polarized antenna. The received antenna is assumed to comprise of *N* azimuthal plane waves with arbitrary carrier phase, arbitrary angle of arrival, and each wave having equal average amplitude. Equal amplitude assumption is based on the fact that in the absence of a direct line-of-sight path, the scattered components arriving at a receiver will experience similar attenuation over small-scale distance. Doppler shift due to the motion of the receiver.

  Assume no excess delay due to multipath.

  - Flat fading assumption.

  For the *n*th wave arriving at an angle $\sim_n$ to the *x*-axis, the Doppler shift is given by

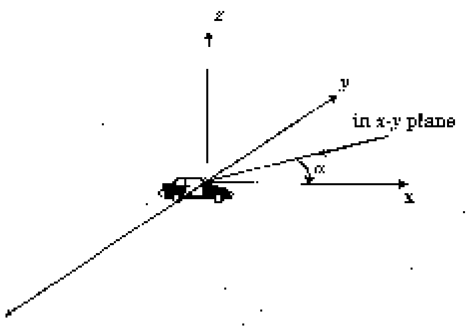


Fig 2.12.2 Doppler Shift

The vertically polarized plane waves arriving at the mobile have E field components given by (assume a single tone is transmitted)

$$E_z(t) = E_0 \sum_{n=1}^{N} C_n \cos(2®f_c t + ¯_n)$$

E_0 : real amplitude of local average E - field (constant)

C_n : real random variable representing the amplitude of nth arriving wave.

f_c : carrier frequency.

$¯_n$: random phase of the nth arriving wave.

The random arriving phase is given by

$$¯_n = 2®f_n t + ¤_n$$

The amplitude of E-field is normalized such that

$$\sum_{i=1}^{N} \overline{C_n^2} = 1$$

$E_z(t)$ can be modeled as a Gaussian random process if N is sufficient large. Since the Doppler shift is very small when compared to the carrier frequency, the three field components may be modeled as narrow band random process.

Where
$$E_z(t) = T_c(t)\cos(2®f_c t) + T_s(t)\sin(2®f_c t)$$

$$T_c(t) = E_0 \sum_{i=1}^{N} C_n \cos(2®f_n t + ¤_n)$$

$$T_s(t) = E_0 \sum_{i=1}^{N} C_n \sin(2®f_n t + ¤_n)$$

$T_c(t)$ $T_s(t)$ are Gaussian random processes which are denoted as T_c and T_s , respectively.

$T_c(t)$ And $T_s(t)$ are uncorrelated zero-mean Gaussian random variable with equal variance given by

$$\overline{T_c^2} = \overline{T_c^2} = \overline{|E_z|^2} = E_0^2 / 2$$

The envelope of the received E-field is given by

$$|E_z(t)| = \sqrt{T_c^2(t) + T_s^2(t)} = r(t)$$

It can be shown that the random received signal envelope r has a Rayleigh distribution given by

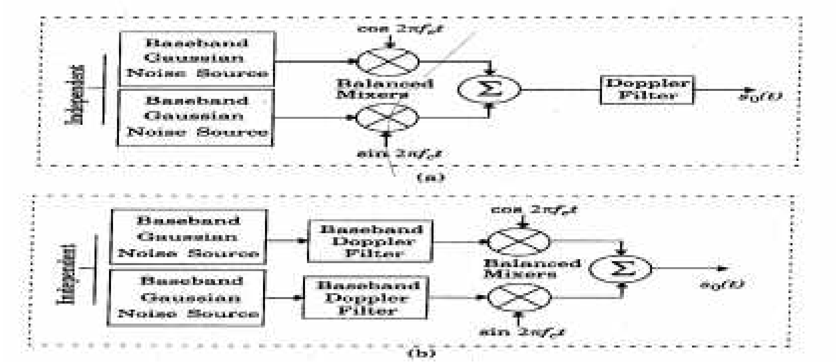
$$p(r) = \begin{cases} \dfrac{r}{±^2}\exp\left(-\dfrac{r^2}{2±^2}\right) & 0 \le r \le \infty \\ 0 & r < 0 \end{cases}$$

where $±^2 = E_0^2 / 2$

2.12.3 Simulation of Clarke Fading Model

Produce a simulated signal with spectral and temporal characteristics very close to measured data. Two independent Gaussian low pass noise are used to produce the in-phase and quadrature fading branches. Use a spectral filter to sharp the random signal in the frequency domain by using fast Fourier transform

(FFT).Time domain waveforms of Doppler fading can be obtained by using an inverse fast Fourier transform (IFFT).



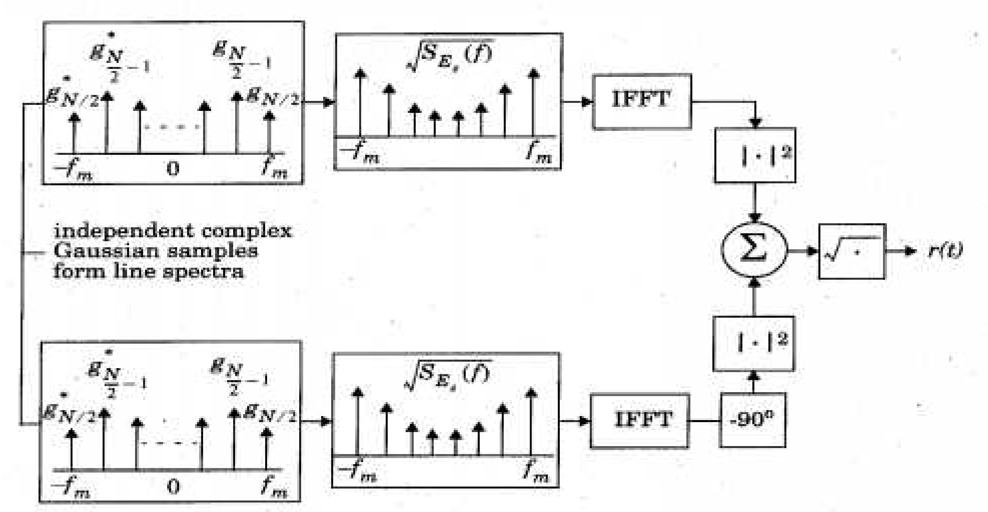


Fig 2.12.3 Smith simulator using N carriers to generate fading signal

1. Specify the number of frequency domain points *N* used to represent $\sqrt{S(f)}$ and the maximum Doppler frequency shift f_m .
2. Compute the frequency spacing between adjacent spectral lines as $\Delta f = 2 f_m /(N-1)$. This defines the time duration of a fading waveform,$T = 1/\Delta f$.
3. Generate complex Gaussian random variables for each of the *N/2* positive frequency components of the noise source.
4. Construct the negative frequency components of the noise source by conjugating positive frequency and assigning these at negative frequency values.
5. Multiply the in-phase and quadrature noise sources by the fading spectrum.
6. Perform an IFFT on the resulting frequency domain signal from the in-phase and quadrature arms, and compute the sum of the squares of each signal.
7. Take the square root of the sum.

UNIT - 3

3.MODULATION TECHNIQUES AND EQUALIZATION

3.1 INTRODUCTION

Modulation may be defined as the process by which some parameters of high frequency signal termed as carrier, is varied in accordance with the signal to be transmitted.Modulation may be done by varying the amplitude, phase or frequency of a high frequency signal (carrier) in accordance with the amplitude of the message signal. Demodulation or detection is the process of recovering the original modulating signal from a modulated wave.

Digital Modulation:Modem mobile communication systems use digital modulation techniques. Advancements in Very Large - Scale Integration (VLSI) Digital Signal Processing (DSP) technology has made digital modulation more cost effective than analog transmission system.

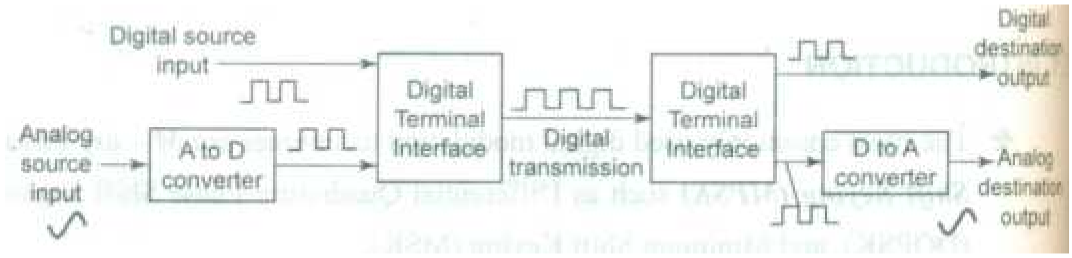


Fig 3.1 Digital Transmission System

Advantages of Digital over Analog Modulation:

(i) Greater noise immunity.

(ii) Robustness to channel impairments

(iii) Easier multiplexing of various forms of information.

(iv) Greater security.

(v) To transmit digital signals on bandpass channel, the amplitude, frequency or phase of the sinusoidal carrier is varied in accords; with the incoming digital data.

(vi) Since the digital data is in discrete steps, the modulation of the bandpass sinusoidal carrier is also done in discrete steps. Therefore this type of modulation (Digital Modulation) is also called switching or signaling

3.2 MINIMUM SHIFT KEYING

Msk is a spectrally efficient modulation scheme and is particularly attractive for use in mobile radio communication systems

Let fI and f2 denote the frequencies of the two FSK signals for symbols "0" and "1" respectively.The frequency separation between the two frequencies is Δ= f2-f1. The modulation index, denoted by h, is defined as the product of the frequency separation and the symbol interval.

$h=\Delta.T_b=1/2$------------→ 1

$f2=fc +1/2\Delta$----------→ 2

$f1= fc-1/2\Delta$----------→ 3

from eqn (1)

$\Delta.T_b=1/2$

$2\Delta=1/ T_b$

$1/2\Delta=1/4\ T_b$ -----------→ 4

Substitute eqn (4) in eqn (2) and (3) we have

$f2 =fc +1/4\ T_b$ for symbol '1'---------------→ 5

$f1 =fc -1/4\ T_b$ for symbol '0'---------------→ 6

3.2.1 MSK Transmitter

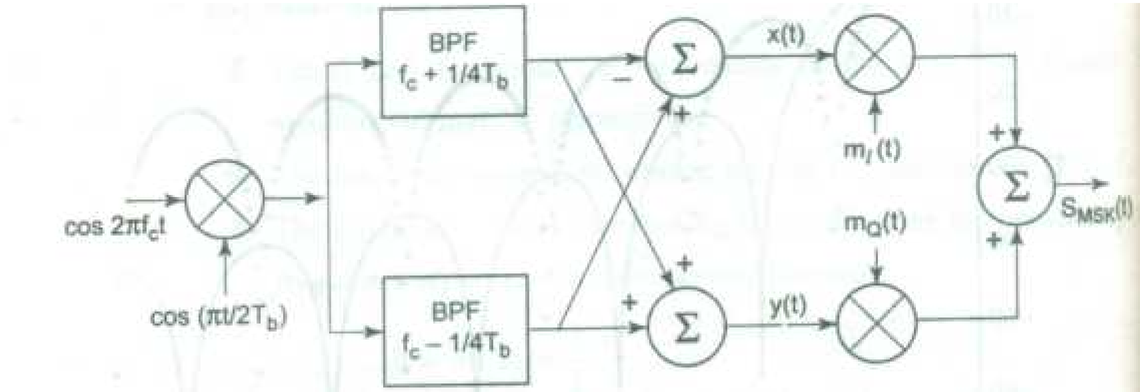


Fig 3.2.1 Block diagram of an MSK Transmitter

The carrier signal (cos 2πfct) is multiplied by cos [πt / 2Tb] to produce two phase coherent signals at fc $+1/4\ T_b$ and fc $-1/4\ T_b$. These two FSK signals are separated using two narrow bandpass filters and appropriately combined to form the in - phase quadrature carrier components x (t) and y (t) respectively.

x(t) and y(t) are multiplied by odd and even bit streams, ml(t) and mQ(t). The outputs of the multipliers are then added to give MSK modulated signal SMSK(t).

3.2.2 MSK Receiver

The MSK signal can be detected noncoherently or coherently.In non-coherent detection, the MSK signal is detected in the same way as an FSK signal by using frequency discriminator,

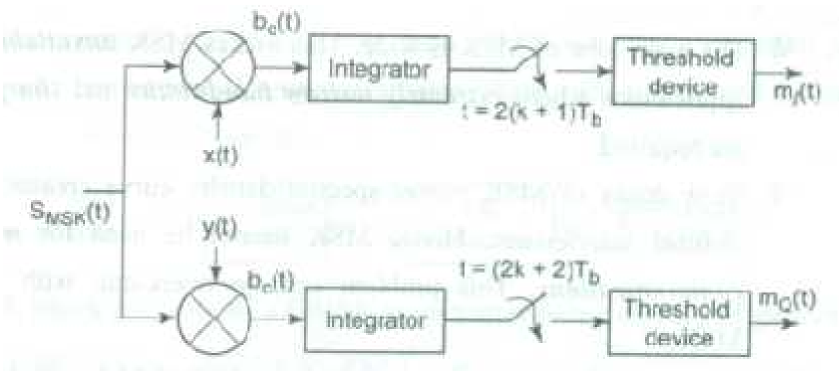


Fig 3.2.2 Block diagram of an MSK Receiver

where the constraint of continuous phase is not used The fig. shows the structure of an MSK receiver with coherent detection which exploits the continuous phase constraint of the signal. The received signal SMSK(t) (in the absence of noise and interference) is multiplied by the respective in - phase and quadrature carriers x(t) and yet).

The outputs of the multipliers are beet) and -bo(t). The integrators integrate over the period of 2T b (Two bit period).For the upper correlator, the sampling switch samples output of integrator at t = (2K +l)T b. Then the decision device decides whether bo(t) is aO or al. Similarly lower correlator output is b(t). The output data streams correspond to ml(t) and mQ(t), which are offset combined to obtain the demodulated signal.

Mertis:

(i) Constant envelope

(ii) Spectral efficiency

(iii) Good BER performance

(iv) Self - synchronizing capability.

(v) MSK is a **spectrally efficient modulation** scheme and is particularly attractive for use in mobile radio communication systems.

3.3 GAUSSIAN MINIMUM SHIFT KEYING

The main lobe of MSK is wide. This makes MSK **unsuitable** for the applications where **extremely narrow bandwidths and sharp cutoffs** are required. Slow decay of MSK power spectral density curve creates adjacent channel interference. Hence MSK cannot be used for **multi-user communications.** This· problem can be overcome with Gaussian MSK. GMSK is a simple binary modulation scheme which may be viewed as a **derivative of MSK.** GMSK is used in European second generation wireless systems such as GSM. The word **Gaussian** refers to the **shape of a filter** that is used before the modulator (transmitter) to **reduce** the **transmitted bandwidth** of the signal. GMSK uses **less bandwidth** than conventional FSK.

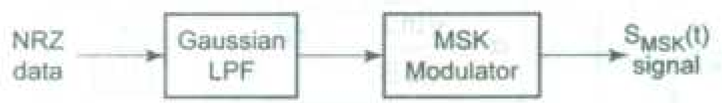


Fig 3.3 Block diagram of Gaussian MSK Transmitter

In GMSK, the sideband levels of the spectrum are further reduced by passing the modulating NRZ data waveform through a premodulation Gaussian pulse - shaping filter. Gaussian pulse - shaping filter which is particularly effective when used in conjunction with Minimum Shift Keying (MSK) modulation, or other modulations which are well suited for power efficient non linear amplifiers. Premodulation Gaussian filtering converts the full response message signal (Where each base band symbol occupies a single bit period T) into a partial response scheme where each transmitted symbol spans several bit periods

Fig 3.3.1Gaussian MSK transmitter using direct FM generation

The spectrum of this LPF has Gaussian shape. The impulse response of Gaussian filter is given by

$$h_G(t) = \frac{\sqrt{\pi}}{\alpha} \exp\left(-\frac{\pi^2}{\alpha^2} t^2\right)$$

The impulse response of the Gaussian filter gives rises to a transfer function that highly dependent upon the 3 - dB bandwidth. The Gaussian low pass filter has a transfer function is given by

$$H_G(f) = \exp\left(-\alpha^2 f^2\right)$$

The parameter α is related to bandwidth B.

The 3 - dB baseband bandwidth of $H_G(t)$ is given by

$$\alpha = \frac{\sqrt{\ln 2}}{\sqrt{2}\,B}$$

$$\boxed{\alpha = \frac{0.5887}{B}}$$

Normally GMSK is defined by BT product, where B is bandwidth and T is baseband symbol duration.

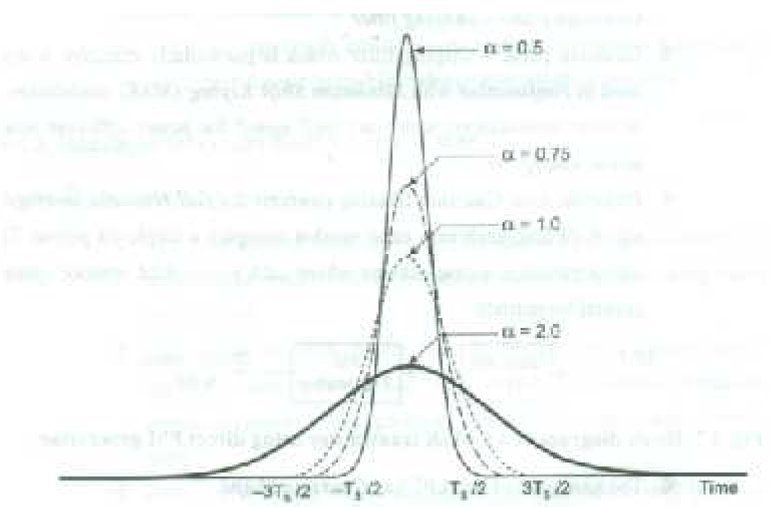


Fig 3.3.2 Impulse response of a Gaussian pulse shaping filter

GMSK Receiver:

GMSK signals can be detected using orthogonal coherent detectors or with simple noncoherent detectors such as standard PM discriminators. Carrier recovery is sometimes performed using a method suggested by De Buda.

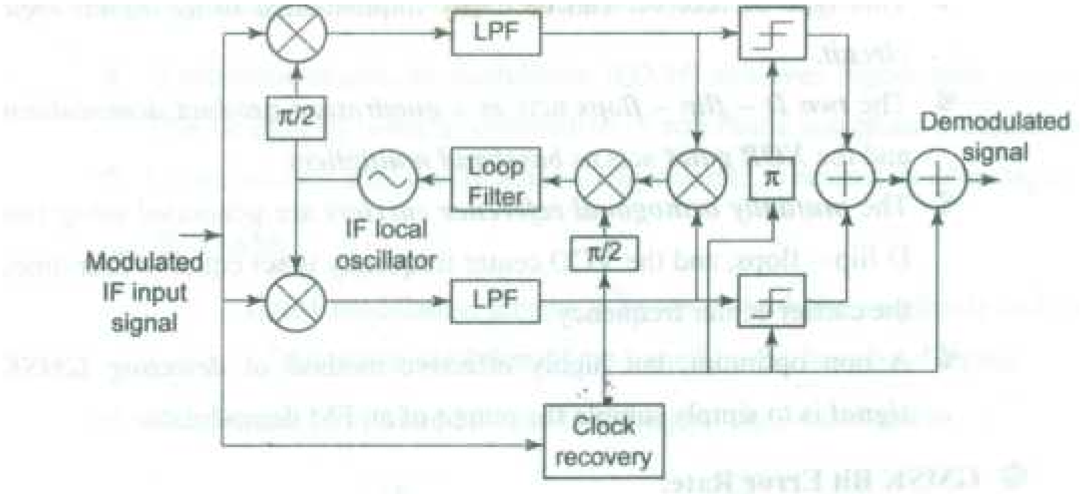


Fig 3.3.4 Block diagram of Gaussian MSK Receiver

The sum of the two discrete frequency components contained at the output of a frequency doubler is divided by four. De Buda's method is similar to the Costa's loop and is equivalent to that of a PLL with a frequency doubler.

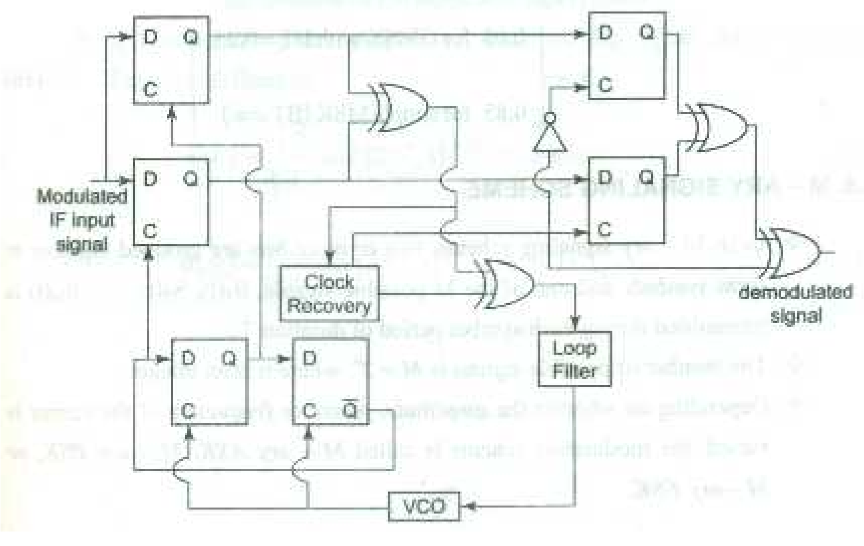


Fig 3.3.5 Digital logic circuit for GMSK demodulation

This type of receiver can be easily implemented using digital logic circuit. The two D - flip - flops acts as a quadrature product demodulator and the XOR gates acts as baseband multipliers. The mutually orthogonal reference carriers are generated using two D flip - flops, and the vco center frequency is set equal to four times the carrier center frequency. A non-optimum, but highly effective method of detecting GMSK signal is to simply sample the output of an FM demodulator.

GMSK Bit Error Rate:

The bit error probability for GMSK is given by

$$P_e = Q\left\{\sqrt{\frac{2\gamma E_b}{N_0}}\right\}$$

Where, y is a constant related to BT.

$$\gamma = \begin{cases} 0.68 & \text{for GMSK with } BT = 0.25 \\ 0.85 & \text{for simple MSK } (BT = \infty) \end{cases}$$

3.4 M -ARY SIGNALING SCHEME

In an M - ary signaling scheme, two or more bits are grouped together to form symbols and one of the M possible signals, SI(t), S2(t), ... SM(t) is transmitted during each symbol period of duration Ts.The number of possible signals is M = 2", where n is an integer. Depending on whether the amplitude, phase or frequency of the carrier is varied, the modulation scheme is called M - ary ASK, M - ary PSK, or M-ary FSK.

3.4.1 M- ary Quadurature Amplitude Modulation (QAM)

Quadrature amplitude modulation (QAM) achieves higher data rates than FSK or PSK by using a combination of amplitude and phase modulation. QAM requires a relatively noise - free channel to realize its advantages.

QAM: QAM modulation scheme is in which both the amplitude and phase of the transmitted signals are varied by the base band signal.

The general form of an M - ary QAM signal can be defined

$$S_i(t) = \sqrt{\frac{2E_{min}}{T_s}}\, a_i \cos(2\pi f_c t) + \sqrt{\frac{2E_{min}}{T_s}}\, b_i \sin(2\pi f_c t) \quad \ldots (1)$$

$$0 \le t \le T \qquad i = 1, 2, \ldots M$$

Where, Emin is the energy of the signal with the lowest amplitude. a, and b, are a pair of independent integers chosen according to the location of the particular signal point. If rectangular pulse shapes are assumed, the signal Si(t), the basic functions defined as

$$\phi_1(t) = \sqrt{\frac{2}{T_s}} \cos(2\pi f_c t) \qquad 0 \le t \le T_s \qquad \dots (2)$$

$$\phi_2(t) = \sqrt{\frac{2}{T_s}} \sin(2\pi f_c t) \qquad 0 \le t \le T_s \qquad \dots (3)$$

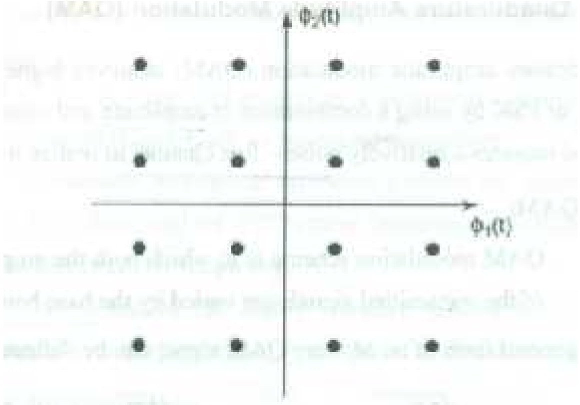


Fig 3.4.1 Constellation diagram of an M-ary QAM (M=16) signal set

The coordinates of the i^{th} message point are ai, $\sqrt{(Emin)}$ and bi $\sqrt{(Emin)}$, where (ai, bi) is an element of the L by L matrix given by

$$\{a_i, b_i\} = \begin{bmatrix} (-L+1, L-1) & (-L+3, L-1) & \dots & (L-1, -L-1) \\ (-L+1, L-3) & (-L+3, L-3) & \dots & (L-1, L-3) \\ \dots & \dots & \dots & \dots \\ (-L+1, -L+1) & (-L+3, -L+1) & \dots & (L-1, -L+1) \end{bmatrix} \dots (4)$$

Where, L $=\sqrt{(M)}$

L = $\sqrt{(16)}$ = 4, the L by L inatrix is given by

For example, 16– QAM

$$\{a_i, b_i\} = \begin{bmatrix} (-3,3) & (-1,3) & (1,3) & (3,3) \\ (-3,1) & (-1,1) & (1,1) & (3,1) \\ (-3,-1) & (-1,-1) & (1,-1) & (3,-1) \\ (-3,-3) & (-1,-3) & (1,-3) & (3,-3) \end{bmatrix} \qquad \dots (5)$$

Probability of Error: The average probability of error in an AWGN channel for M - ary QAM using coherent detection is given by

$$P_e \approx 4\left(1 - \frac{1}{\sqrt{M}}\right) Q\left(\sqrt{\frac{2E_{min}}{N_o}}\right) \qquad \dots (6)$$

In terms of the average signal energy Eav

$$P_e \approx 4\left(1-\frac{1}{\sqrt{M}}\right) Q\left(\sqrt{\frac{3E_{av}}{(M-1)N_o}}\right) \qquad \ldots (7)$$

The power spectrum and bandwidth efficiency of QAM modulation is identical to M - ary PSK modulation. In power efficiency, QAM is superior to M - ary PSK.

3.5 M-ary FREQUENCY SHIFT KEYING (MFSK)

M-ary frequency shift keying

It is an extension of a binary FSK. In M-ary system, $M=2^N$ different symbols are used. N – number of bits per symbol. Every symbol was separate frequency for transmission. Such system is called M-ary FSK system.

Transmitter

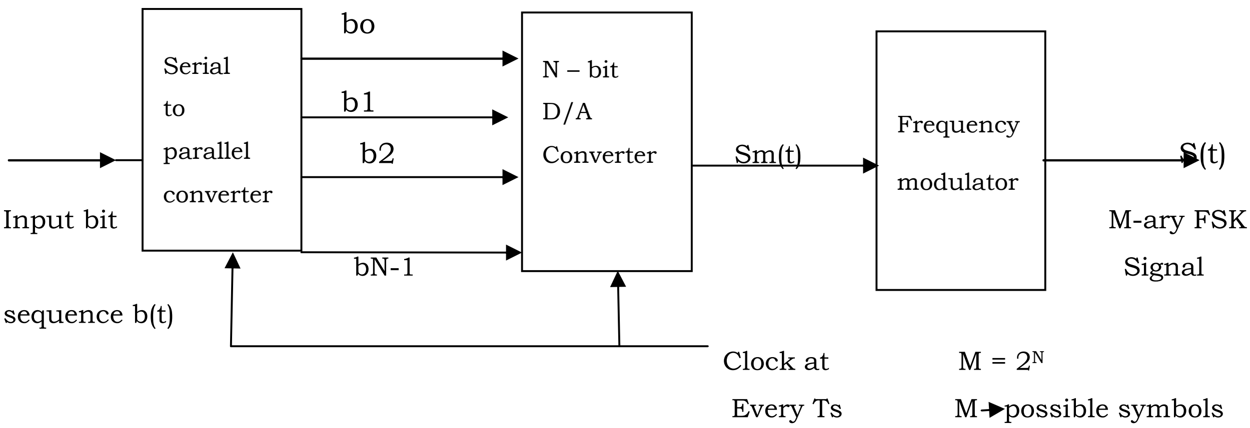


Fig 3.5.1 M-ary FSK Transmitter

The input bit sequence b(t) is converted into parallel by serial to parallel converter. These N bits are applied to the digital to analog converter which forms a symbol at the output . These bits are applied after every Ts seconds. Ts is the symbol period Ts =NTb period. The output of digital to analog converter is given to a frequency modulator. Thus depending upon the value of the symbol, the frequency modulator generates the output frequency. For every symbol, the frequency modulator produce different frequency output. This particular frequency signal remains at the output for one symbol duration. Thus for ‘M’ symbols, there are ‘M’ frequency signals at the output of modulation. Thus the transmitted frequencies are f_0, f_1, f_2, f_3 -------- f_M-1 depending upon the input symbol to the modulator.

Receiver

The M-ary FSK signal is given to the set of ‘M’ band pass filters. The center frequencies of those filters are f_0, f_1, f_2, -------- f_M-1

These filters pass their particular frequency and attenuate others. The envelope detector outputs are applied to a decision device. The decision device produces its output depending upon the highest input.

Depending upon the particular symbol, only one envelope detector will have higher output. The outputs of other detectors will be very low.

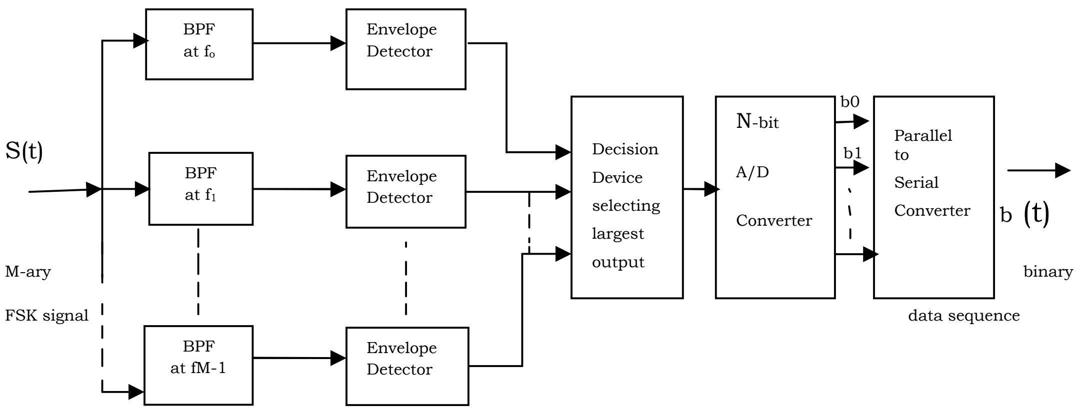


Fig 3.5.2 M-ary FSK Receiver

The output of the decision device is given to N bit analog to digital 'N' bit symbol in parallel. These bits are then converted to serial bit stream by parallel to serial converter. In some cases the bits appear in parallel. Then there is no need to use serial to parallel and parallel to serial converters.

3.6 ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING

OFDM is of great interest by researchers and research laboratories all over the world. It has already been accepted for the new wireless local area network standards IEEE 802.11a, High Performance LAN type 2 (HIPERLAN/2) and Mobile Multimedia Access Communication (MMAC) Systems. Also, it is expected to be used for wireless broadband multimedia communications. Data rate is really what broadband is about. The new standard specify bit rates of upto 54 Mbps. Such high rate imposes large bandwidth, thus pushing carriers for values higher than UHF band. For instance, IEEE802.11a has frequencies allocated in the 5- and 17- GHz bands. OFDM can be seen as either a modulation technique or a multiplexing technique. One of the main reasons to use OFDM is to increase the robustness against frequency selective fading or narrowband interference. In a single carrier system, a single fade or interferer can cause the entire link to fail, but in a multicarrier system, only a small percentage of the subcarriers will be affected. Error correction coding can then be used to correct for the few erroneous subcarriers. The concept of using parallel data transmission and frequency division multiplexing was published in the mid-1960s.Some early development is traced back to the 1950s. A U.S. patent was filed and issued in January 1970. In a classical parallel data system, the total signal frequency band is divided into N nonoverlapping frequency subchannels. Each subchannel is modulated with a separate symbol and then the N subchannels are frequency-multiplexed. It seems good to avoid spectral overlap of channels to eliminate interchannel interference. However, this leads to inefficient use of the available spectrum. To cope with the inefficiency, the ideas proposed from the mid-1960s were to use parallel data and FDM with overlapping subchannels, in which, each carrying a signaling rate b is spaced b apart in frequency to avoid the use of high-speed equalization and to combat impulsive noise and multipath distortion, as well as to fully use the available bandwidth

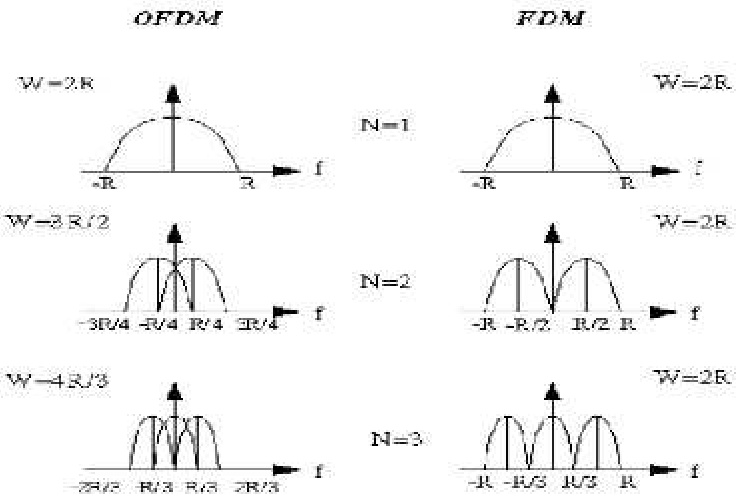


Fig 3.6 Concept of OFDM signal: orthogonal multicarrier technique versus conventional multicarrier technique.

The word orthogonal indicates that there is a precise mathematical relationship between the frequencies of the carriers in the system. In a normal frequency-division multiplex system, many carriers are spaced apart in such a way that the signals can be received using conventional filters and demodulators. In such receivers, guard bands are introduced between the different carriers and in the frequency domain, which results in a lowering of spectrum efficiency. It is possible, however, to arrange the carriers in an OFDM signal so that the sidebands of the individual carriers overlap and the signals are still received without adjacent carrier interference. To do this, the carriers must be mathematically orthogonal. The receiver acts as a bank of demodulators, translating each carrier down to DC, with the resulting signal integrated over a symbol period to recover the raw data. If the other carriers all beat down the frequencies that, in the time domain, have a whole number of cycles in the symbol period T, then the integration process results in zero contribution from all these other carriers. Thus, the carriers are linearly independent (i.e., orthogonal) if the carrier spacing is a multiple of 1/T.

The OFDM transmission scheme has the following key advantages:

- Makes efficient use of the spectrum by allowing overlap
- By dividing the channel into narrowband flat fading sub channels, OFDM is more resistant to frequency selective fading than single carrier systems.
- Eliminates ISI and IFI through use of a cyclic prefix.
- Using adequate channel coding and interleaving one can recover symbols lost due to the frequency selectivity of the channel.
- Channel equalization becomes simpler than by using adaptive equalization techniques with single carrier systems.

- It is possible to use maximum likelihood decoding with reasonable complexity, as discussed in OFDM is computationally efficient by using FFT techniques to implement the modulation and demodulation functions.
- In conjunction with differential modulation there is no need to implement a channel estimator.
- Is less sensitive to sample timing offsets than single carrier systems are.
- Provides good protection against cochannel interference and impulsive parasitic noise.

In terms of drawbacks OFDM has the following characteristics:

- The OFDM signal has a noise like amplitude with a very large dynamic range.Therefore it requires RF power amplifiers with a high peak to average power ratio.
- It is more sensitive to carrier frequency offset and drift than single carrier systems are due to leakage of the DFT.

3.7 PERFORMANCE OF DIGITAL MODULATION IN SLOW FLAT FADING CHANNELS

Flat - fading channels cause a multiplicative (gain) variation in the transmitted signal s(t) The slow flat fading channels change much slower than the applied modulation; it can be assumed that the attenuation and phase shift of the signal is constant over at least one s ymbol interval.The received signal ret) is expressed as

$$r(t) = \alpha(t)\exp\left(-j\theta(t)\right)s(t)+n(t) \qquad 0 \le t \le T \qquad \ldots (1)$$

Where, $\alpha(t)$ - Gain of the channel

$\theta(t)$ - Phase shift of the channel

$n(t)$- Additive Gaussian noise.

For the accurate estimate of the phase $\theta(t)$, coherent or noncoherent matched filter detection may be employed at the receiver. The probability of error in AWGN channels is viewed as a conditional error probability where the condition is that a is fixed. The probability of error in slow flat fading channels can be obtained by averaging the error in AWGN channels over the fading probability density function. The probability of error in a slow flat fading channel is given by

$$P_e = \int_0^\infty P_e(X)P(X)dX \qquad \ldots (2)$$

Where, Pe(X) is the probability of error for an arbitrary modulation at a specific value of signal - to - noise ratio X,

$$X=\frac{\alpha^2 E_b}{N_0} \qquad \ldots (3)$$

P(X) = Probability density function of X due to the fading channel.

E_b = Average energy per bit.

N_o = Noise power density in a non - fading AWGN channel.

The random variable α^2 is used to represent instantaneous power values of the fading channel, with respect to the non-fading E_b / N_o. For a unity gain fading channel $\overline{\alpha^2} = 1$ For Rayleigh fading channels, the fading amplitude a has a Rayleigh distribution, so the fading power α^2 and consequently X has a chi - square distribution with two degrees of freedom. Therefore

$$P(X) = \frac{1}{\Gamma}\exp\left(-\frac{X}{\Gamma}\right) \qquad X \geq 0 \qquad \ldots (4)$$

Where, $\Gamma = \frac{E_b}{N_o}\overline{\alpha^2}$ is the average value of the signal to noise ratio

For $\overline{\alpha^2} = 1$, Γ corresponds to the average E_b / No for the fading channel. By using equation (4) and the probability of error of a particular modulation scheme in AWGN, the probability of error in a slow flat -fading channel can be evaluated.

$$P_{e,PSK} = \frac{1}{2}\left[1 - \sqrt{\frac{\Gamma}{1+\Gamma}}\right] \text{ (coherent binary PSK)} \qquad \ldots (5)$$

$$P_{e,FSK} = \frac{1}{2}\left[1 - \sqrt{\frac{\Gamma}{2+\Gamma}}\right] \text{(coherent binary FSK)} \qquad \ldots (6)$$

In a slow, flat, Rayleigh fading channel,

$$P_{e,DPSK} = \frac{1}{2(1+\Gamma)} \text{ (differential binary PSK)} \qquad \ldots (7)$$

$$P_{e,NCFSK} = \frac{1}{2+\Gamma} \text{ (non coherent orthogonal binary FSK)} \qquad \ldots (8)$$

For large values of Eb / No the error probability equations may be simplified as

Pe,PSK = 1/4 Γ (coherent binary FSK) ... (9)

Pe,FSK = 1/2 Γ (coherent FSK) ... (10)

Pe,DPSK= 1/2 Γ (differential PSK) ... (11)

Pe,NCFSK = 1/ Γ (noncoherent orthogonal binary FSK) ….(12)

$$P_{e,GMSK} = \frac{1}{2}\left(1 - \sqrt{\frac{\delta\Gamma}{\delta\Gamma+1}}\right) \approx \frac{1}{4\delta\Gamma} \text{ (coherent GMSK)} \qquad \ldots (13)$$

$$\text{Where, } \delta = \begin{cases} 0.68 & \text{for BT} = 0.25 \\ 0.85 & \text{for BT} = \infty \end{cases}$$

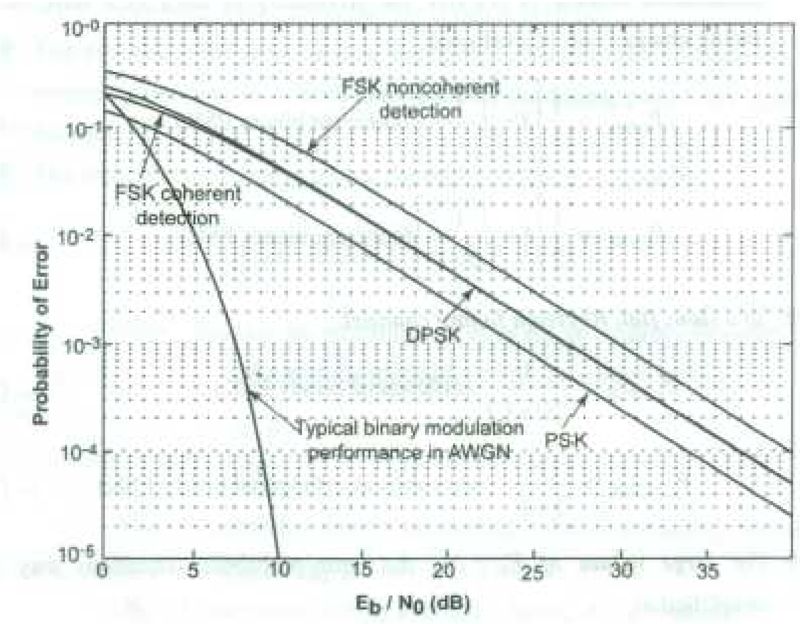


Fig 3.7 Bit error rate performance of binary modulation schemes in a Rayleigh flat - fading channel as compared to a typical performance curve in AWGN

3.8 DIGITAL MODULATION IN FREQUENCY SELECTIVE MOBILE CHANNELS

Frequency selective fading caused by multipath time delay spread causes Inter Symbol Interference (lSI) which results in an irreducible BER floor for mobile systems. If a mobile channel is not frequency selective, the time - varying Doppler spread due to motion creates an irreducible BER floor due to the random spectral spreading. Simulation is the major tool used for analyzing frequency selective fading effects. The irreducible error floor in a frequency selective channel is primarily caused by the errors due to the inter symbol interference, which interferes with the signal component at the receiver sampling instants. This occurs when

(a) the main (undelayed) signal component IS removed through multipath cancellation.

(b) a non - zero value of d causes lSI

(c) the sampling time of a receiver is shifted as a result of delay spread.

The parameter d is the rms delay spread normalized by the symbol period.

From the figure, it is seen that the BER performance of BPSK is the best among all the modulation schemes compared. This is due to symbol offset interference does not exist in BPSK.Both filtered and unfiltered BPSK, QPSK, OQPSK and MSK modulation schemes were suitable for frequency selective mobile channels.

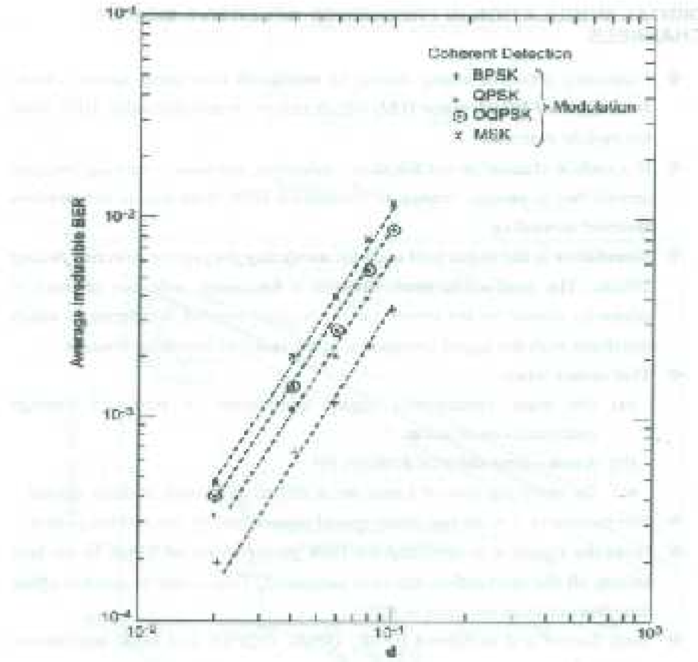


Fig 3.8 The irreducible BER performance for different modulations with coherent detection for a channel with a Gaussian shaped power delay profile

3.9 EQUALIZATION

Equalization can be used to compensate the Inter Symbol Interference (lSI) created by multi path within time dispersion channel.

3.9.1 Fundamentals of Equalization

Equalizer:The device which equalizes the dispersive effect of a channel is referred to as an equalizer.

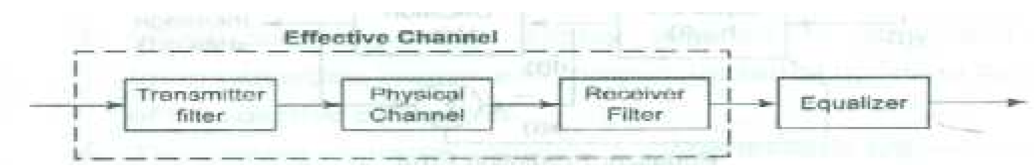


Fig 3.9.1 The Equalized System

For channel equalization at baseband, we can design an equalizer and place it between the demodulator and decision device such that the output of the equalizer (i.e., the input of the decision device) is lSI free. lSI is the major obstacle to high speed data transmission over wireless channels, equalization is a technique used to compensate intersymbol interference.

3.10 SURVEY OF EQUALIZATION TECHNIQUES

The major classification of equalization techniques are linear and nonlinear equalization.

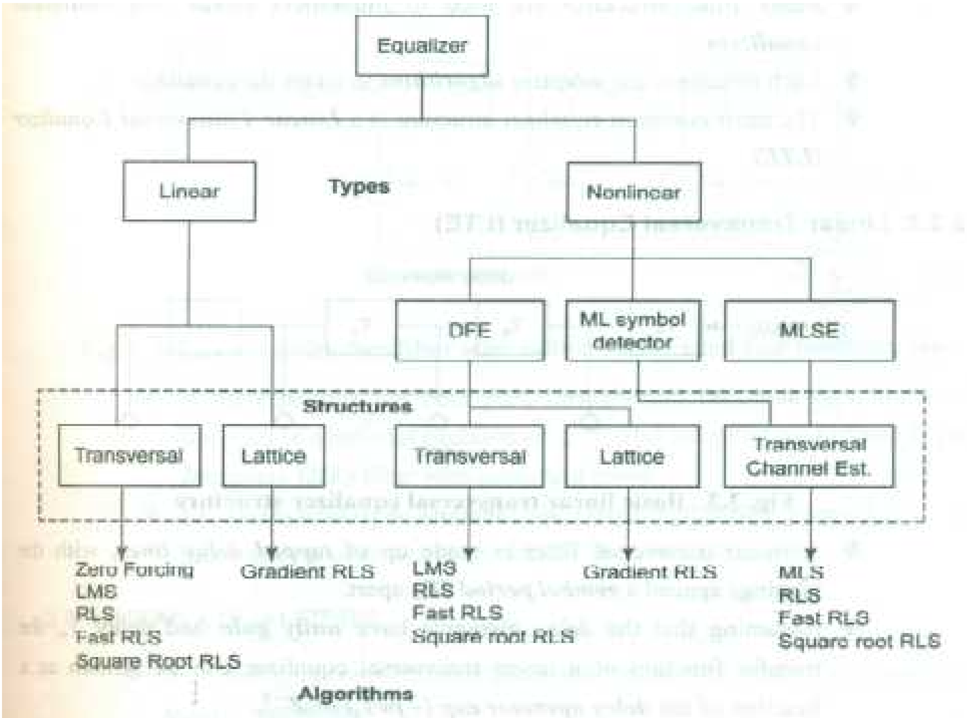


Fig 3.10 Classification of equalizers

3.11 LINEAR EQUALIZERS

A linear equalizer can be implemented as an FIR filter. So it also called transversal filter.In this equalizer, the present and past values of the received signal are linearly weighted by the filter coefficient and summed to produce the output. The output of the transversal filter before a decision is made (threshold detection) is

$$\hat{d}_K = \sum_{n=-N_1}^{N_2} (C_n^*) Y_{K-n} \qquad \dots (1)$$

Where, $\hat{d}_K$ - Output at time index K

C_n^* - Complex filter coefficients or tap weights.

Y_i - Input received signal at time $t_0 + iT$

t_0 - Equalizer starting time and

$N = N_1 + N_2 + 1$ - Number of taps.

The values N 1 and N2 denote the number of taps used in the forward and reverse portions of the equalizer, respectively.

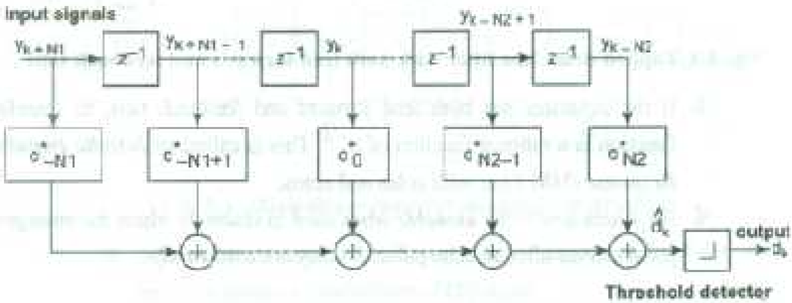


Fig 3.11 Structure of a linear transversal equalizer

The minimum mean squared error of linear transversal equalizer is given by

$$E\left[|e(n)|^2\right] = \frac{T}{2\pi}\int_{-\pi/T}^{\pi/T} \frac{N_0}{\left|F\left(e^{j\omega T}\right)\right|^2 + N_0}\, d\omega \qquad \dots (2)$$

Where, $F(ei^{eoT})$-Frequency response of the channel

No - Noise power spectral density

Lattice Equalizer: The linear equalizer can also be implemented by lattice filter.

In a lattice filter, the input signal YK is transformed into a set of N intermediate forward and backward error signals fn(K) and bn(K) respectively, which are used as inputs to the tap multipliers and are used to calculate the updated coefficients.

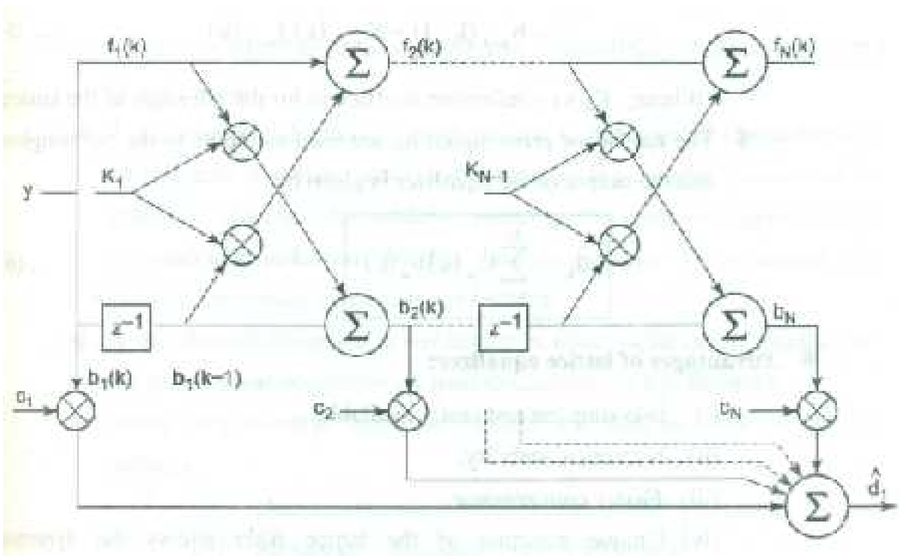


Fig 3.11.1 The structure of a lattice equalizer

The each stage of the lattice is characterized by the following

$$f_1(k) = b_1(k) = y(k) \qquad \dots (3)$$

$$f_n(k) = Y(k) - \sum_{i=1}^{n} K_i\, y(k-i)$$

$$= f_{n-1}(k) + K_{n-1}(k)\, b_{n-1}(k-1) \qquad \dots (4)$$

$$b_n(k) = Y(k-n) - \sum_{i=1}^{n} K_i\, y(k-n+i)$$

$$= b_{n-1}(k-1) + K_{n-1}(k)\, f_{n-1}(k) \qquad \dots (5)$$

Where, Kn(k) - reflection coefficient for the nth stage of the lattice

The backward error signal bn, are used as inputs to the tap weights, and the output of the equalizer is given by

$$\boxed{\hat{d}_k = \sum_{n=1}^{N} C_n(k) b_n(k)} \qquad \ldots (6)$$

Advantages of lattice equalizer:

(i) It is simplest and easily available

(ii) Numerical stability. (iii) Faster convergence

(iv) Unique structure of the lattice filter allows the dynamic assignment of the most effective length of the lattice equalizer .

(v) When the channel becomes more time dispersive, the length of the equalizer can be increased by the algorithm without stopping the operation of the equalizer.

Disadvantages of lattice equalizer:

(i) If the channel is not very time dispersive, only a fraction of the stages are used.

(ii) It is more complicated than a linear transversal equalizer.

3.12 NON LINEAR EQUALIZATION

The linear equalizers are very effective in equalizing channels where ISI is not severe. The severity of the ISI is directly related to the spectral characteristics. In this case that there are spectral nulls in the transfer function of the effective channel, the additive noise at the receiver input will be dramatically enhanced by the linear equalizers. To over come this problem, non linear equalizers can be used. If the channel distortion is too severe in linear equalizer to handle , then the non-linear equalizers are used to compensate the distortion.

3.12.1 Decision Feedback Equalization (DFE)

The Decision Feedback Equalizer (DFE) IS particularly useful for channels with severe amplitude distortions and has been widely used in wireless communications. The basic idea is that, if the value of the symbols already detected are known (past decisions are assumed to be correct), then the lSI contributed by these symbols can be canceled exactly, by subtracting the past symbol values with appropriate weighting from the equalizer output. The DFE can be realized in either the direct transversal form or as a lattice filter.

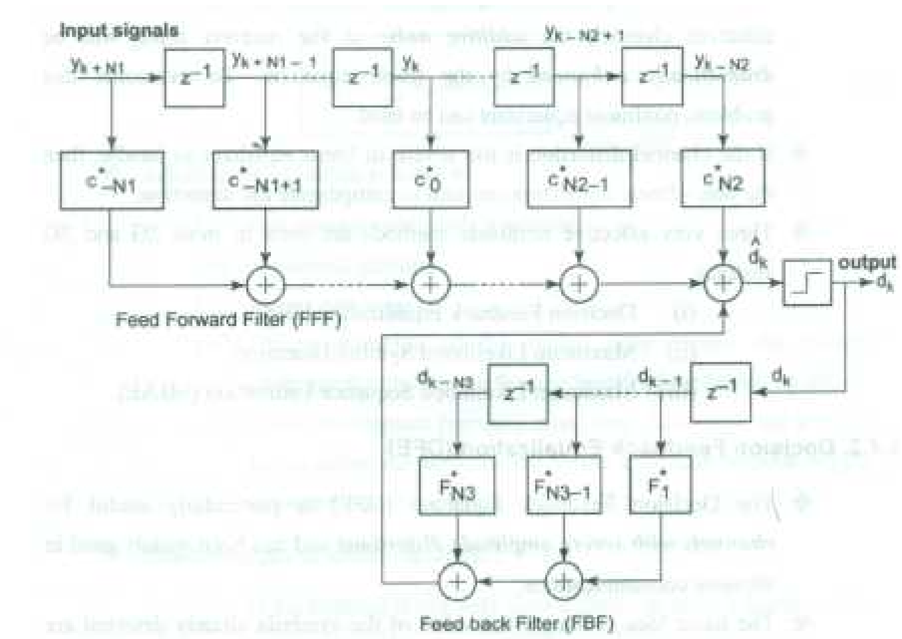


Fig 3.12.1.1 Decision Feedback Equalizer (DFE)

The equalizer has N1 + N2 + 1 taps in the feed forward filter and N3 taps in the feedback filter and its output can be expressed as

$$\hat{d}_K = \sum_{n=-N_1}^{N_2} C_n^* Y_{K-n} + \sum_{i=1}^{N_3} F_i d_{K-i} \quad \ldots (1)$$

Where, C_n^* → Tap gains of the forward filter.

Yn -> Inputs to the forward filter.

F_i^* -> Tap gains for the feedback filter

di(i < k) -> Previous decision made on the detected signal

Once $\hat{d}_K$ is obtained then dK is decided from it. Then dK along with previous decisions dK _ 1, dK _ 2, ••• are fed back into the equalizer.

- Minimum Mean Square Error (MMSE)

The minimum mean square error of DFE is given by

$$E\left[|e(n)|^2\right]_{min} = \exp\left\{\frac{T}{2\pi}\int_{-\pi/T}^{\pi/T} \ln\left[\frac{N_0}{\left|F(e^{j\omega T})\right|^2 + N_0}\right] d\omega\right\} \quad \ldots (2)$$

A DFE has significantly smaller minimum MSE than an LTE. For severely distorted wireless channels, DFE is much better than Linear Transversal Equalizer (LTE).

- Predictive DFE

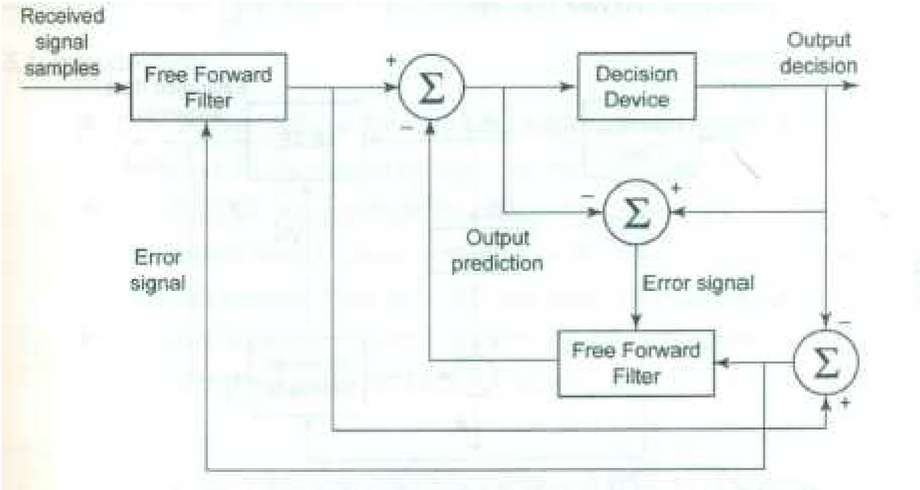


Fig 3.12.1.2 Predictive decision feedback equalizer

The lattice implementation of the DFE is equivalent to a transversal DFE having a feed forward filter of length N 1 and a feedback filter of length N2, where N 1 > N2. Another form of DFE proposed by Belfiore and park is called a predictive DFE. This consists of a Feed Forward Filter (FFF) as in the conventional DFE. The Feed Back Filter (FBF) is driven by an input sequence formed by the difference of the output of the detector and the output of the feed forward filter. Here FBF is called as a noise predictor because it predicts the noise and the residual lSI contained in the signal at the FFT output and subtracts from it the detector output after some feedback delay.

3.12.2 Maximum Likelihood Sequence Estimation (MLSE) Equalizer

In this type of equalizers use various forms of the classical maximum likelihood receiver structure

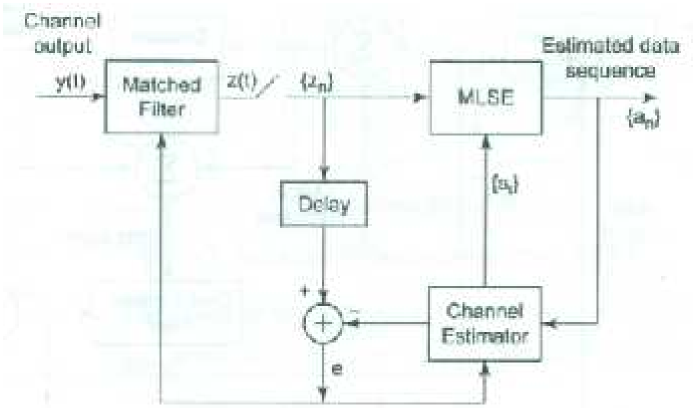


Fig 3.12.2 The structure of MLSE with an adaptive matched filter

Here, using a channel impulse response simulator within the algorithm, the MLSE tests all possible data sequences and chooses the data sequence with the maximum probability as the output. An MLSE usually has a large computational requirement, especially when the delay spread of the channel is large. Using the MLSE as an equalizer was first proposed by Forney, he setup a basic MLSE estimator structure and implemented it with the Viterbi algorithm. The MLSE is optimal, to minimizes the probability of a sequence error. The MLSE requires knowledge of the channel characteristics in order to compute the metrics for making decisions. The MLSE also requires knowledge of the statistical distribution of the noise corrupting the signal. The matched filter operates on the continuous time signal, whereas the MLSE and channel estimator rely on discretized (non - linear) samples.

3.13 ALGORITHMS FOR ADAPTIVE EQUALIZATION

Adaptive Algorithms

The choice of algorithm and its corresponding rate of convergence depend on the channel data rate and coherence time Three basic algorithms are used for adaptive equalization. There are

(i) Zero forcing (ZF) algorithm.
(ii) Least Mean Squares (LMS) algorithm
(iii) Recursive Least Square (RLS) algorithm

3.13.1 Zero Forcing Algorithm

In the zero force equalizer, the equalizer coefficients en are chosen to force the samples of the combined channel and equalizer impulse response to zero.The number of coefficients increases without bound an infinite length equalizer with zero lSI at the output can be obtained.When each of the delay elements provide a time delay equal to the symbol duration T, the frequency response Heq(f) of the equalizer is periodic with a period equal to the symbol rate 1 / T. The combined-response of the channel with the equalizer

$$H_{ch}(f)\,H_{eq}(f) = 1, \qquad |f| < \frac{1}{2T} \qquad \dots (1)$$

Where, Hch (f) - Folded frequency response of the channel.

Heq (f) - Frequency response of the equalizer.

Advantage:

It performs well for static channels with high SNR, such as local wired telephone lines.

Disadvantage:

The inverse filter in the equalizer, may excessively amplify noise at frequencies where the folded channel spectrum has high attenuation

3.13.2 Least Mean Square (LMS) Algorithm

LMS equalizer is used to minimize of the Mean Square Error (MSE) between the desired equalizer output and the actual equalizer output.

The adaptive algorithm uses prediction error e_K to minimize a cost function and updates the equalizer weights in a manner that iteratively reduces the cost function

The input signal to the equalizer as a vector y_k

$$Y_K = [y_K \ y_{K-1} \ y_{K-2} \cdots y_{K-N}]^T \qquad \ldots (2)$$

The output of the adaptive equalizer is a scalar given by

$$\hat{d}_k = \sum_{n=0}^{N} W_{nK} \ Y_{K-n} \qquad \ldots (3)$$

Advantages:

(i) The LMS equalizer maximizes the signal to distortion at its output within the constraints of the equalizer filter length.

(ii) Low computational complexity and

(iii) Simple program

Disadvantages:

(i) Slow convergence and

(ii) Poor tracking

3.13.3 Recursive Least Squares (RLS) Algorithms

The convergence rate of the gradient based LMS algorithm is very slow. In order to achieve faster convergence, complex RLS algorithms used which involve additional parameters RLS significantly improves the convergence rate of adaptive equalizers. The least square error based on the time average is defined as

$$J(n) = \sum_{i=1}^{n} \lambda^{n-i} e^*(i,n)\, e(i,n)$$

Where λ is the weighting factor close to 1

$Y_N(i)$ is the data input vector at time i

$$Y_N(i) = [y(i), y(i-1), \ldots y(i-N+1)]^T$$

Error signal e(i,n) is given by

$$e(i,n) = x(i) - Y_N^T(i) W_N(n) \qquad 0 \le i \le n$$

Where, e* (i,n) is the complex conjugate of e (i, n) and

$W_N(n)$ is the new tap gain vector at time n.

J(n) is the cumulative squared error of the new tap gains on all the old data. To obtain the minimum of least square error J(n), the gradient of J(n) is set to zero.

$$\frac{\partial}{\partial W_N} J(n) = 0$$

Advantages:

(i) Fast convergence

(ii) Good tracking ability. If smaller value of weighting coefficient A, the equalizer has better tracking ability

Disadvantages:

(i) High computational requirement.

(ii) Complex program structure and

(iii) If A is too small, the equalizer will be unstable

3.14 DIVERSITY TECHNIQUES

Diversity is a powerful communication receiver technique that provides wireless link improvement at relatively low cost.

Diversity concept:

If one radio path undergoes a deep fade, another independent path may have a strong signal. By having more than one path to select from, both the instantaneous and average SNRs at the receiver may be improved, often by as much as 20dB to 30dB.

Macroscopic diversity : By selecting a base station which is not shadowed when other are, the mobile can improve substantially the average signal - to - noise ratio on the forward link. This is called macroscopic diversity. Macroscopic diversity is also useful at the base station receiver.

Types of Diversity Techniques (or) Diversity Mechanisms

The diversity techniques are classified into following categories.

(i) Space or Antenna diversity

(ii) Angle or direction diversity

(iii) Polarization diversity

(iv) Time diversity

3.14.1 Space Diversity

Space diversity has been widely used because it can be implemented simply and economically.

Space diversity, also known as antenna diversity, is one of the most popular forms of diversity used in wireless systems. The desired message is transmitted by using multiple transmitting antennas and or receiving antennas. The space separation between adjacent antennas should be large enough to ensure that the signals from different antennas are independently faded. In a Rayleigh fading environment, it can be shown that, if two antennas are separated by half of the carrier wavelength, the corresponding two signals experience independent fading. The concept of antenna space diversity is also used in base station design .Spacing between adjacent receiving antennas is chosen so that multipath fading appearing in the diversity branches (paths) becomes uncorrelated.

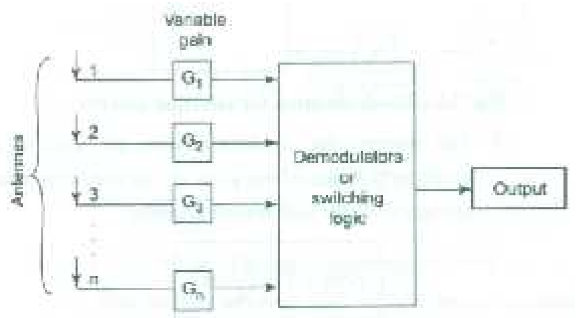


Fig 3.14.1 Generalized block diagram for space diversity

Classification of Space Diversity:

Space diversity reception method can be classified into four categories

(i) Selection diversity

(ii) Feedback diversity

(iii) Maximal ratio combining and

(iv) Equal gain diversity

(i) Selection (or) Switching diversity:

Selection diversity is more suitable for mobile radio applications because of its simple implementation. In this scheme, the receiver monitors the SNR value of each diversity channel and chooses the one with the maximum SNR value for signal detection.

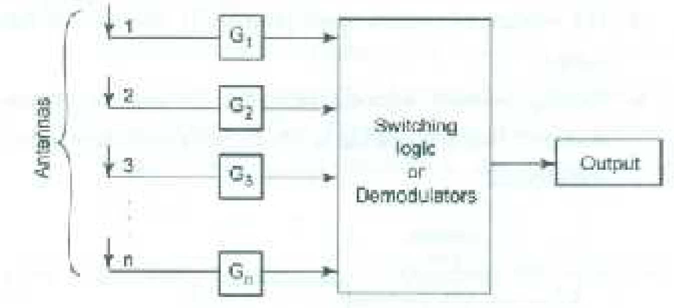


Fig 3.14.1.1 Block diagram for selection diversity

The receiver has m demodulators are used to provide m diversity branches whose gains are adjusted to provide the same average SNR for each branch given by

$$SNR = \Gamma = \frac{E_b}{N_0}\overline{\alpha}^2 \quad \text{Where, assume } \overline{\alpha}^2 = 1$$

The receiver branch having the highest instantaneous SNR is connected to the demodulator.The antenna signals themselves could be sampled and the best one sent to a single demodulator.

Feedback or Scanning Diversity:

Scanning diversity is very similar to selection diversity except that instead of always using the best of M signals, the M signals are scanned in a fixed sequence until one is found to be above a predetermined threshold.

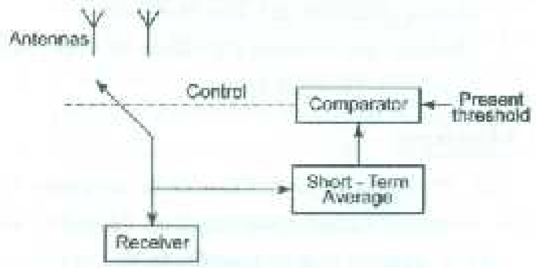


Fig 3.14.1.2 Block diagram of scanning diversity

Maximal Ratio Combining:

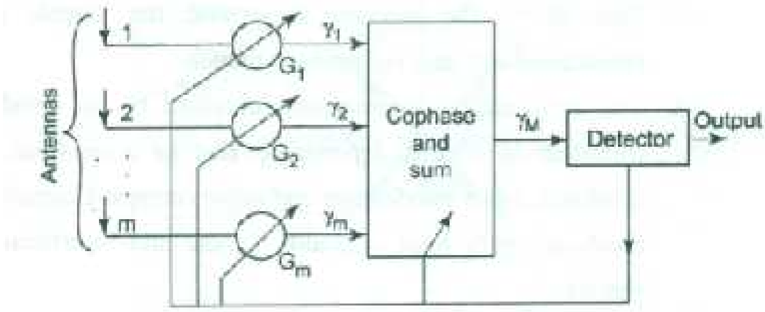


Fig 3.14.1.3 Maximal ratio combiner

In this method, the signals from all of the M branches are weighted according to their individual signal voltage to noise power ratios and then summed. Here, the individual signals must be co - phased before being summed which generally requires an individual receiver and phasing circuit for each antenna element. Maximal ratio combining produces an output SNR equal to the sum of the individual SNRs.

Equal Gain Diversity:

It is similar to maximal - ratio combining, except that the weighting circuits are omitted.

In this method, the branch weights are all set to unity but the signals from each branch are co-phased to provide equal gain combining diversity. This allows the receiver to exploit the signals that are simultaneously received on each branch. The performance improvement obtained by an equal - gain combiner is slightly inferior to that of a maximal - ratio combiner, since interference and noise corrupted signals may be combined with high - quality (noise and interference free) signals. Equal- gain combiner is superior to selection diversity.

3.14.2 Polarization Diversity

Polarization diversity, in which only two diversity branches are available, is effective because the signals transmitted through two orthogonally polarized propagation paths have un correlated fading. statistics in the usual VHF and UHF land mobile radio environment. In this diversity, the transmitted signal with horizontal or vertical polarization is received by antenna with two elements. One element is used for horizontal polarization and the other is used for vertical polarization. Circular and linear polarized antennas have been used to characterize multi path inside buildings. When the path was obstructed, polarization diversity was found to dramatically reduce the multi path spread without significantly decreasing the received power. Polarization diversity is likely to become more important for improving link margin and capacity.

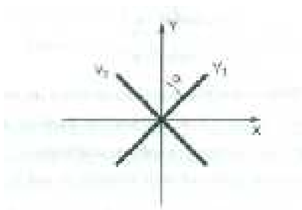


Fig 3.14.2 Theoretical model for base station polarization diversity based on x –y plane

3.14.3 Frequency Diversity

The desired message IS transmitted simultaneously over several frequency slots.The separation between adjacent frequencies slots should be larger than the channel coherence bandwidth such that channel fading over each slot is independent of that is any other slot. Frequency diversity is implemented by transmitting information on **more than one carrier** frequency. Frequency diversity is often employed in microwave line - of - sights links which carry several channels in a **Frequency Division Multiplex** mode (FDM).

3.14.4 Time Diversity

The desired message is transmitted repeatedly over several time periods. The time separation between adjacent transmissions should be larger than the channel coherence time such that the channel fading experienced by each transmission is independent of the channel fading experienced by all of the other transmission. One modem implementation of the time diversity involves the use of the RAKE receiver for spread spectrum CDMA, where the multi path channel provides redundancy in the transmitted message.

3.15 RAKE RECEIVER

One of the advantages of the CDMA system is that multi path interference can be reduced by combining direct and reflected signals in the receiver. The receivers used are called rake receivers. CDMA spreading codes are designed to provide very low correlation between successive chips. Propagation delay spread in the radio channel provides multiple versions of the transmitted signal at the receiver If these multipath components are delayed In time by more than chip duration, they appear like un-cor-rel-ate-d -noise at a CDMA receiver, and equalization is not required. There is useful information in the multipath components, in CDMA, the RAKE receiver may combine the time delayed versions of the original signal transmission in order to **improve** the **signal - to - noise ratio** at the receiver.

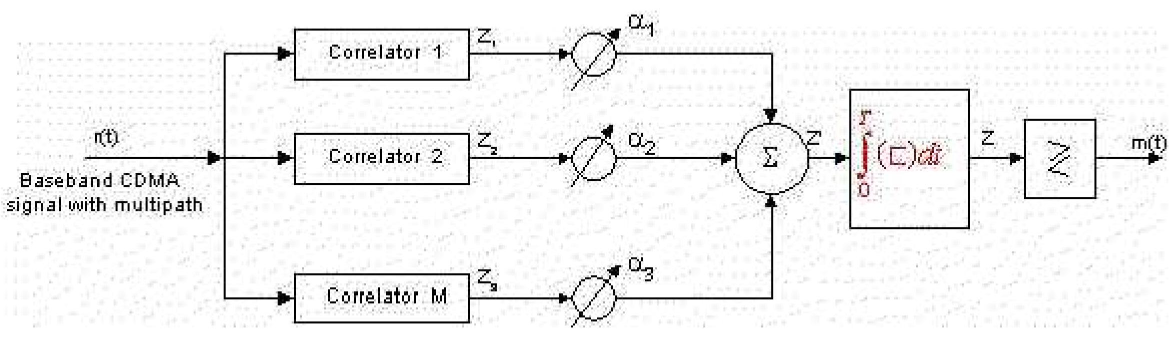


Fig 3.15 An M-branch (M - finger) RAKE receiver implementation.

Each correlator detects a time shifted version of the original CDMA transmission.

A RAKE receiver utilizes **multiple** correla tors to separately detect the M strongest multipath components. The outputs of each correlator are then weighted to provide a better estimate of the transmitted signal then is provided by a single component. Demodulation and bit decisions are then based on the weighted outputs of the M correlators.

3.15.1 Performance of a RAKE Receiver

Assume M correlators are used in a CDMA receiver to capture the M strongest multi path components. A weighting network is used to provide a linear combination of the correlator output for bit detection. Correlator 1 is synchronized to the strongest multipath m1. Multipath component m2 arrives 'tl later than component m1 where.t2 – tl is assumed to be--greater than chip duration. The second correlator is synchronized to m2. It correlates strongly with m2, but has low correlation with m1. In a RAKE receiver, if the output from one correlator is corrupted by fading, the others may not be and the corrupted signal may be discounted through the weighting process. Decisions based on the combination of the M separate decision statistics offered by the RAKE provide a form of diversity which can overcome fading and thereby improve CDMA reception. The outputs of the M correlators are denoted as ZI, Z2, . .. and Zm. They are weighted by αi, α2, ... and αm respectively.The weighting coefficients are based on the power or the SNR from each correlator output.In the case of a maximal ratio combining diversity scheme, the overall signal Z' is given by

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

The weighting coefficients am, are normalized to the output signal power of the correlator in such a way that the coefficients sum to unity.

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

Choosing weight coefficients based on the actual outputs of the correlators yields better RAKE performance.

UNIT - 4

4.CODING AND MULTIPLE ACCESS TECHNIQUES

CODING

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

Encoder: The analog to digital converter located in the transmitter is also known as the encoder or coder.

Decoder: The digital to analog converter located in the receiver is known as the decoder.

The word codec is derived from “coder/decoder”.

4.1VOCODERS

Vocoders is a circuit used for digitizing voice at a low data rate by using knowledge of the way in which voice sounds are produced .A vocoder is an example of lossy compression applied to human speech. Lossy compression compromises signal quality in order to reduce the bit rate. For voice transmissions, vocoders arc often used to achieve great reductions in bit rate.

Vocoders are a class of speech coding systems that analyze the voice signal at the transmitter, transmit parameters derived from the analysis, and then synthesize the voice at the receiver using those parameters.All vocoder systems attempt to model the speech generation process as a dynamic system and try to quantify certain physical constraints of the system. These physical constraints are used to provide a parsimonious description of the speech signal. The most popular among the vocoding systems is the Linear Predictive Coder (LPC).

Speech Generation Model:

The fig. shows the traditional speech generation model that is the basis of all vocoding systems.The sound generating mechanism forms the source and is linearly separated from the intelligence modulating vocal tract filter which forms the system.

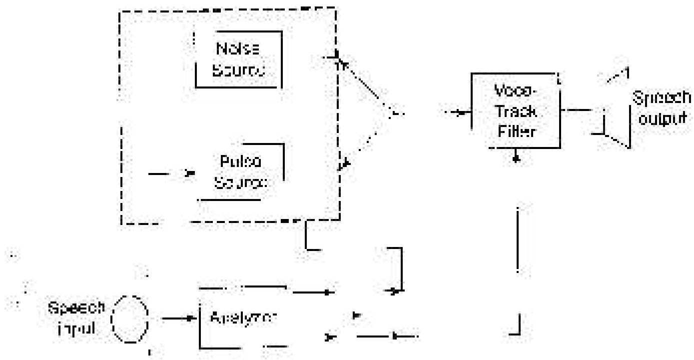


Fig 4.1 Speech generation Model

Types of Speech Signal:

The speech signal is assumed to be of two types.

(i) Voiced

(ii) Unvoiced

Voiced Sound : ("m", "n", "v" pronunciations) are a result quasiperiodic vibrations of the vocal chord .

Unvoiced Sound: ("f", "s", "sh" pronunciations) are produced by turbulent air flow through a constriction.

Vocoders Parameters:

The parameters associated with this model are the voice pitch. pole frequencies of the modulating filter, and the corresponding amplitude parameters.

The pitch frequency for most speakers is below 300Hz. extracting this information from the signal is very difficult.

Advantages:

(i) It achieves very high economy in transmission bit rate.

(ii) Less robust.

Disadvantages:

(i) In general more complex

(ii) Their performance tends to be talker dependent.

Channel Vocoders

The channel vocoders model the vocal - tract filter by means of a bank of 12 to 32 non - overlapping, adjacent Band - Pass Filter (DPFs). Each filter has a separate adjustable gain. Channel vocoders are frequency domain vocoders that determine the envelope of the speech signal for a number of frequency bands and then sample, encode, and multiplex these samples with the encoded outputs of the other filters.The sampling is done synchronously every 10ms to 30ms. Along with the energy information about each band, the voiced / unvoiced decision, and the pitch. frequency for voiced speech are also transmitted.

Formant (Peak) Vocoders

The formant vocoder is similar in concept to the channel vocoder. Theoretically, this can operate at lower hit rates than the channel vocoder because it uses fewer control signals. A formant vocoder must be able to identify atleast three formants tor representing the speech sounds, and it must also control the intensities of the formants. Formant vocoders can reproduce speech at bit rates lower than 1200 bits/so.

Demerit:

Due to difficulties in accurately computing the location of formants and formant transitions from human speech, they have not been very successful.

Cepstrum Vocoders:

The cepstrum vocoder separates the excitation and vocal tract spectrum by inverse fourier transforming of the log magnitude spectrum to produce the cepstrum of the signal. Linear filtering is performed to separate the vocal tract cepstral coefficients from the excitation coefficients. In the receiver, the vocal cepstral coefficients are fourier transformed to produce the vocal tract impulse response. By convolving this impulse response with a synthetic excitation signal. the original speech is reconstructed.

Voice - Excited Vocoder:

Voice - excited vocoders eliminate the need for pitch extraction and voicing detection operations. This system uses a hybrid combination of PCM transmission for the low frequency hand of speech, combined with channel vocoding of higher frequency bands.Voice excited vocoders have been designed for operation at 7200bits/s to 9600bits/s.

4.2 LINEAR PREDICTIVE CODERS

4.2.1 LPC Vocoders

Linear Predictive Coders (LPCs) are belongs to the time domain class of vocoders. Linear Predictive Coding (LPC) vocoders model the vocal - tract filter by means of a single, linear all-pole filter. This vocoders attempt to extract the significant features of speech from the time waveform.LPC can able to transmit good quality voice at 4.8kbps and poorer quality voice at even lower rates.The linear predictive coding system models the vocal tract as an all pole linear filter with a transfer function of

$$H(z) = \frac{G}{1+\sum_{k=1}^{M} b_k z^{-k}}$$

Where, G - Gain of the filter and

Z^{-1}- Unit delay operation.

Depending on speech segment (i.e.) voiced and unvoiced, the excitation of this linear titter is either a pulse at the pitcb frequency or random white noise. The coefficients of the all pole filter are obtained in the time domain using linear prediction techniques.

Working Principle of the LPC:

The LPC system transmits only selected characteristics of the error signal, instead of transmitting quantized values of the error signal representing the difference between the predicted and actual waveform.

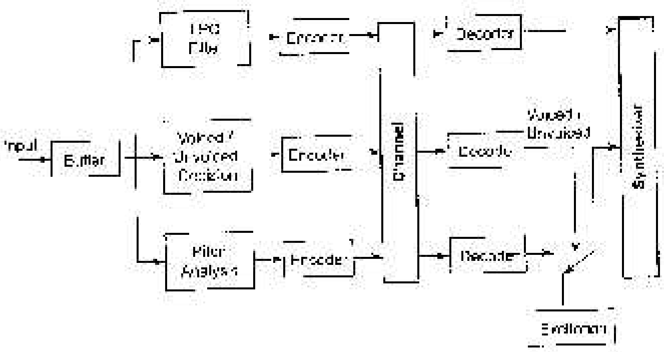


Fig 4.2.1 Block diagram of a LPC coding system

The selected characteristics parameters include the gain/actor, pitch information, and the voiced / unvoiced decision information which allow approximation of the correct error signal. At the receiver, the received information about the error signal IS used to determine the appropriate excitation from the synthesis filter The synthesis filter is designed at the receiver using the received predictor coefficients. In practical, many LPC coders transmit the filter coefficients which already represent the error signal and can be directly synthesized by the receiver.

4.2.2 Multipulse Excited LPC (MPE - LPC)

By using, more than one pulse typically eight per period, and adjusting the individual pulse positions and amplitudes sequentially to minimize a spectrally weighted mean square error. This technique is called the Multiple Excited LPC.

Advantages:

(i) The results from MPE - LPC having better speech quality and
(ii) The number of pulses used can be reduced, in particular for high pitched voices, by incorporating a linear filter with a pitch loop in the synthesizer.

4.2.3 Code - Excited LPC

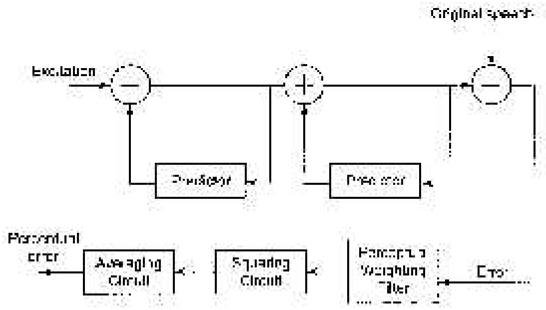


Fig 4.2.3 Block diagram illustrating the CELP code book search

In this method, 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 of stochastic signals for the one that gives the best perceptual 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 then transmitted. The receiver uses this index to pick the correct excitation signal for its synthesizer filter

Operation:

For example, consider the coding of a short 5ms block of speech signal. At a sampling frequency of 8 KHz, each block consists of 40 speech samples. A bit rate of 1/4 bit per sample corresponds to 10 bits per block .Therefore, there are 2^{10} = 1024 possible sequences of length 40 for each block. Each member of the code book provides 40 samples of the excitation signal with a scaling factor that is changed every 5ms block. The scaled samples are passed sequentially through two recursive filters, which introduce voice periodicity and adjust the spectral envelope. The regenerated speech samples at the output of the second filter and compared with samples of the original speech signal to form a difference signal. The difference signal represents the objective error in the regenerated speech signal.This error signal is further processed through a linear filter which amplifies the perceptually more important frequencies and attenuates the perceptually less important frequencies.

Applications:

(i) Advanced DSP and VLSI technology, real-time implementation of CELP codec's are possible.

(ii) The CDMA digital cellular standard(IS-95) proposed by QUALCOMM uses a variable rate CELP codec at 1.2 to 14.4 kbps.

Advantages:

(i) CELP can provide high quality even when the excitation is coded at only 0.25 bits per sample.

(ii) These coders can achieve transmission bit rates as low as 4.8 kbps.

Disadvantages:

(i) The CELP coders are extremely complex.

(ii) It can require more than 500 million multiply and add operations per second.

4.2.4 Residual Excited LPC

In this type of LPC coders, after estimating the model parameters (LP coefficients or related parameters) and excitation parameters (voiced! unvoiced decision, pitch, gain) from a speech frame, the speech is synthesized at the transmitter and subtracted from the original speech signal to from a residual signal. The residual signal is quantized, coded and transmitted to the receiver along with the LPC model parameters.

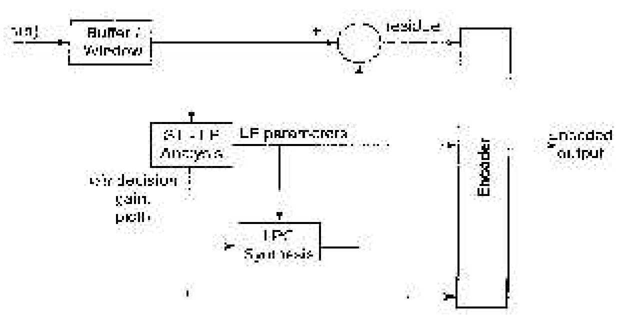


Fig 4.2.4 Block diagram of a RELP encoder

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.

Advantage:

The quality of the synthesized speech is improved due to the addition of the residual error.

Disadvantage:

The residual signal is too complex to transmit exactly with the available bit rate.

4.3 SELECTION OF SPEECH CODEC'S FOR MOBILE COMMUNICATION

Choosing the right speech codec is an important step in the design of a digital mobile communication system. The codec's is required to compress speech to maximize the number of users on the system to utilize the available limited bandwidth. Factors must be considered are

- Compression
- Overall system cost
- Capacity
- End - to -end delay
- The algorithmic complexity of the coder
- The de power requirements
- Compatibility with existing standards and
- Robustness of the encoded speech to transmission errors.

Depending on the technique used, different speech coders show varying degree of immunity to transmission errors.

Cell Size:

The choice of the speech coder will also depend on the cell size used When the cell size is sufficiently small such that high spectra efficiency is achieved through frequency reuse, it may be sufficient to use a simple high rate speech codec. Cellular systems operating with much larger cells and poorer channel conditions need to use error correction coding, thereby requiring the speech codec's to operate at lower bit rates. In mobile satellite communications, the cell sizes are very large and the available bandwidth is very small. In order to accommodate realistic number of users the speech rate must be of the order 3kbps, requiring the use of vocoder techniques.

Multiple Access Technique:

The type of multiple access technique used, being an important in determining the spectral efficiency of the system, strongly influences choice of speech codec.

Type of Modulation:

The type of modulation employed also has considerable impact on the choice of speech codec.

4.4 The GSM CODEC

The original speech coder used in the pan - European digital cellular standard GSM, regular pulse excited long - term prediction (RPE - LTP) codec. This codec has a net bit rate of) 3kbps and was chosen after conducting exhaustive subjective tests on various competing codec .Most recent GSM upgrades have improved upon the original codec specification.

4.4.1 GSM Encoder

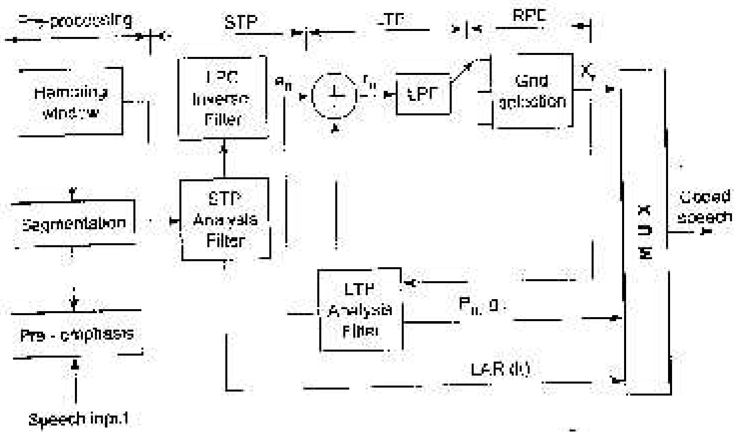


Fig 4.4.1 Block diagram of GSM speech encoder

The GSM speech encoder consists of four major processing blocks.

(i) Pre - processing

(ii) Short - Term Processing (STP)

(iii) Long - Term Prediction (LTP) and

(iv) Regular Pulse Excited (RPE)

Pre - Processing:

The speech sequence is first pre - emphasized ordered into segments of 20ms and then Hamming - windowed .

Short - Term Processing (STP):

In the Short - Term Prediction (STP) filtering analysis where the Logarithmic Area Ratios (LARs) of the reflection coefficients r_n (k) (eight in number) are computed. The eight LAR parameters have different dynamic and probability distribution functions, and hence all of them are not encoded with the same number of bits of transmission. The LAR parameters are also decoded by the LPC inverse filter so as to minimize the error e_n.

Long - Term Prediction (LTP):

LTP analysis which involves finding the pitch period P_n, gain factor g_n is then carried out such the LPT residual r_n is minimized.To minimize r_n, pitch extraction is done by the LTP by determining that value of delay D, which maximizes cross - correlation between the current STP error and a previous error sample e_n- D.

Regular Pulse Excited (RPE):

The extracted pitch P_n and gain g_n are transmitted and encoded at a rate of 3.6kbps. The LTP residual r_n is weighted and decomposed into three candidate excitation sequences. The energies of these sequences are identified, and the one with the highest energy is selected to represent the LTP residual. The pulses in the excitation sequence are normalized to the highest amplitude, quantized, and transmitted at a rate of 9.6kbps.

4.4.2 GSM Speech Decoder

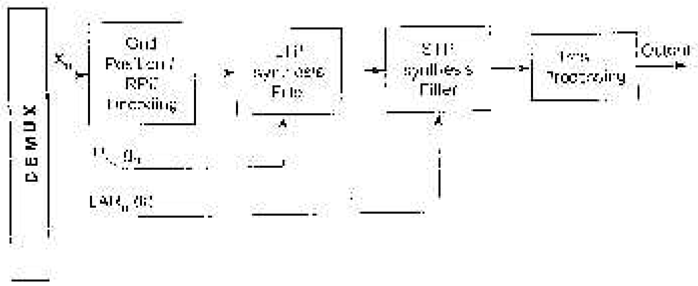


Fig 4.4.2 Block diagram of GSM speech decoder

It consists of four blocks.

(i) RPE decoding

(ii) LTP synthesis filter

(iii) SIP Synthesis filter

(iv) Post – processing

The received excitation parameters are RPE decoded and passed to the LTP synthesis filter which uses the pitch and gain parameter to synthesis the long - term signal. Short - term synthesis is carried out using the received reflection coefficients to recreate the original speech signal.

4.5 REED-SOLOMON (RS) CODES FOR CDPD

RS coding is a type of FEC. It has been widely used because of its relatively large error correction capability when weighed against its minimal added overhead. RS codes are also easily scaled up or down in error correction capability to match the error rates expected in a given system. It provides a robust error control method for many common types of data transfer mediums, particularly those that are one-way or noisy and sure to produce errors. RS codes are an example of a block coding technique. The data stream to be transmitted is broken up into blocks and redundant data is then added to each block. The size of these blocks and the amount of redundant data added to each block is either specified for a particular application or can be user-defined for a closed system. Within these blocks, the data is further subdivided into a number of symbols, which are generally from 6 to 10 bits in size. The redundant data then consists of additional symbols being added to the end of the transmission. The system-level block diagram for an RS codec is shown in Figure

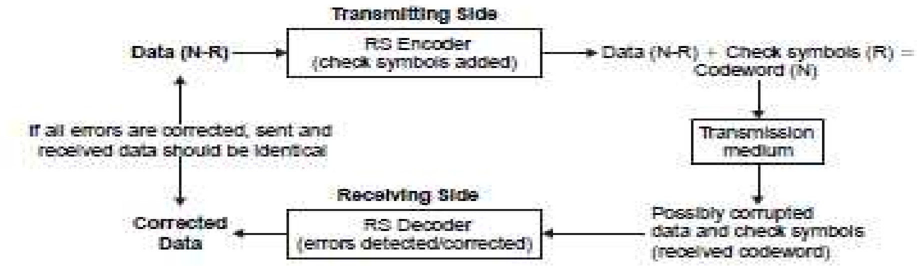


Fig 4.5.1 RS system-level block diagram

The original data, which is a block consisting of N-R symbols, is run through an RS encoder and R check symbols are added to form a code word of length N. Since RS can be done on any message length and can add any number of check symbols, a particular RS code is expressed as RS (N, N-R) code. N is the total number of symbols per code word; R is the number of check symbols per code word, and N-R is the number of actual information symbols per code word. RS encoding consists of the generation of check symbols from the original data. The process is based upon finite field arithmetic. The variables to generate a particular RS code include field polynomial and generator polynomial starting roots. The field polynomial is used to determine the order of the elements in the finite field. Another system-level characteristic of RS coding is whether the implementation is systematic or nonsystematic. A systematic implementation produces a code word that contains the unaltered original input data stream in the first R symbols of the code word. In contrast, in a nonsystematic implementation, the input data stream is altered during the encoding process. Most specifications require systematic coding.

A typical RS decode algorithm consists of several major blocks. The first of these blocks is the syndrome calculation, where the incoming symbols are divided into the generator polynomial, which is known from the parameters of the decoder. The check symbols, which form the remainder in the encoder section, will cause the syndrome calculation to be zero in the case of no errors. If there are errors, the resulting polynomial is passed to the Euclid algorithm, where the factors of the remainder are found The result is evaluated for each of the

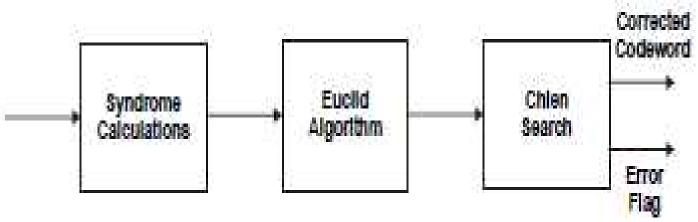


Fig 4.5.2 RS decoder block diagram

In coming symbols over many iterations, and any errors are found and corrected. The corrected code word is the output from the decoder. If there are more errors in the code word than can be corrected by the RS code used, then the received code word is output with no changes and a flag is set, stating that the error correction has failed for that code word. The error correction capability of a given RS code is a function of the number of check bits appended to the message. In general, it may be assumed that correcting an error requires one check symbol to find the location of the error, and a second check symbol to correct the error. In general then, a given RS code can correct R/2 symbol errors, where R is the number of check symbols in the given RS code. Since RS codes are generally described as an RS (N, N-R) value, the number of errors correctable by this code is [N- (N-R)]/2. This error control capability can be enhanced by use of erasures, a technique that helps to determine the location of an error without using one of the check symbols. An RS implementation supporting erasures would then be able to correct up to R errors. Since RS codes work on symbols (most commonly equal to one 8-bit byte) as opposed to individual data bits, the number of correctable errors refers to symbol errors. This means that a symbol with all of the bits corrupted is no different than a symbol with only one of its bits corrupted, and error control capability refers to the number of corrupted symbols that can be corrected. RS codes are more suitable to correct consecutive bits. RS codes are generally combined with other coding methods such as Viterbi, which is more suited to correcting evenly distributed errors. The effective throughput of an RS decoder is a combination of the number of clock cycles required to locate and correct errors after the code word has been received and the speed at which the design can be clocked. Knowing the latency and clock speed allows the user to determine how many symbols per second may be processed by the decoder. In the RS code, there are two RS decoder choices: a high-speed decoder and a low-speed decoder. The trade-off is that the low-speed decoder is usually approximately 20% smaller in device utilization. Note that both decoders operate at the same clock rate, but the low-speed decoder has a longer latency period, resulting in a slower effective symbol rate. As the number of check symbols decreases, the complexity of the decoder decreases, resulting in a smaller design and an increase in performance.

4.6 MULTIPLE ACCESS TECHNIQUES

Multiple access techniques are used to allow a large number of mobile users to share the allocated spectrum in the most efficient manner. As the spectrum is limited, so the sharing is required to increase the capacity of cell or over a geographical area by allowing the available bandwidth to be used at the same time by different users. And this must be done in a way such that the quality of service doesn’t degrade within the existing users.

4.6.1 Frequency Division Multiple Access (FDMA)

Each user is allocated a unique frequency band or channel. These channels are assigned on demand, and can not be shared.

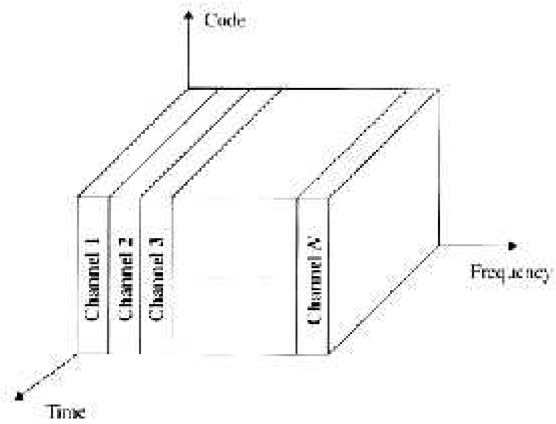


Fig 4.6.1 Frequency Division Multiple Access

The features of FDMA:

The FDMA channel carries only one phone circuit at a time. If an FDMA channel is not in use, then it sits idle and cannot be used by other users to increase or share capacity. It is essentially a wasted resource. After the assignment of a voice channel, the base station and the mobile transmit simultaneously and continuously. The bandwidths of FDMA channels are relatively narrow (30 kHz) as each channel supports only one circuit per carrier. That is, FDMA is usually implemented in narrowband systems. The symbol time is large as compared to the average delay spread. This implies that the amount of inter-symbol interference is low and, thus, little or no equalization is required in FDMA narrowband systems. The complexity of FDMA mobile systems is lower when compared to TDMA systems, though this is changing as digital signal processing methods improve for TDMA. Since FDMA is a continuous transmission scheme, fewer bits are needed for overhead purposes (such as synchronization and framing bits) as compared to TDMA. FDMA systems have higher cell site system costs as compared to TDMA systems, because of the single channel per carrier design, and the need to use costly band pass filters to eliminate spurious radiation at the base station. The FDMA mobile unit uses duplexers since both the transmitter and receiver operate at the same time. This results in an increase in the cost of FDMA subscriber units and base stations. FDMA requires tight RF filtering to minimize adjacent channel interference.

Nonlinear Effects in FDMA:

In FDMA, Many channels share the same antenna at the base station. The power amplifiers or the power combiners, when operated at or near saturation for maximum power efficiency, are nonlinear. The nonlinearities cause signal spreading in the frequency domain and generate inter modulation (IM) frequencies. Intermodulation distortion products occur at frequencies $mf1 + nf2$ for all integer values of m and n.Some of the possible intermodulation frequencies that are produced by a nonlinear device are $(2n+1)f1-2nf2$, $(2n+2)f1-(2n+1)f2$, $(2n+1)f1-2nf2$, $(2n+2)f2-(2n+1)f1$, etc. for $n = 0, 1, 2, \ldots$

Time Division Multiple Access

4.6.2 Time Division Multiple Access (TDMA)

Each user occupies a cyclically repeating time slot, Transmit data in a buffer-and-burst method, the transmission for any user is non continuous. The transmission from various users is interlaced into a repeating frame structure.

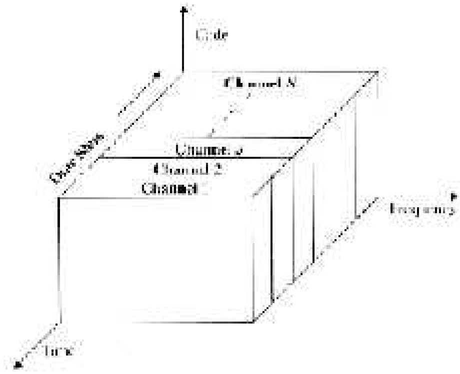
Frame consists of a number of slots (information message), together with a preamble, and tail bits.

Preamble contains the address and synchronization information that both the base station and the subscribers use to identify each other.

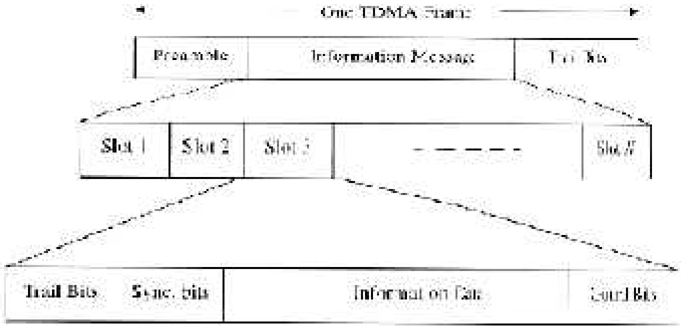
Guard times allow synchronization of the receivers between different slots and frames.

Features of TDMA:

TDMA shares a single carrier frequency with several users, where each user makes use of nonoverlapping time slots. The number of time slots per frame depends on several factors, such as modulation technique, available bandwidth, etc. Data transmission for users of a TDMA system 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 (which is most of the time). Because of discontinuous transmissions in TDMA, the handoff process is much simpler for a subscriber unit, since it is able to listen for other base stations during idle time slots. An enhanced link control, such as that provided by mobile assisted handoff (MAHO) can be carried out by a subscriber by listening on an idle slot in the TDMA frame. TDMA uses different time slots for transmission and reception, thus duplexers are not required. Even if FDD is used, a switch rather than a duplexer inside the subscriber unit is all that is required to switch between transmitter and receiver using TDMA. Adaptive equalization is usually necessary in TDMA systems, since the transmission rates are generally very high as compared to FDMA channels. In TDMA, the guard time should be minimized. If the transmitted signal at the edges of a time slot are suppressed sharply in order to shorten the guard time, the transmitted spectrum will expand and cause interference to adjacent channels. High synchronization overhead is required in TDMA systems because of burst transmissions. TDMA transmissions are slotted, and this requires the receivers to be synchronized for each data burst. In addition, guard slots are necessary to separate users, and this results in the TDMA systems having larger overheads as compared to FDMA. TDMA has an advantage in that it 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 slots based on priority.



a)



b)

Fig 4.6.2 a)Time Division Multiple Access b)frame structure

Efficiency of TDMA:

The frame efficiency, is the percentage of bits per frame which contain transmitted data.

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

It is a measure of the percentage of transmitted data that contains information as opposed to providing overhead for the access scheme. The transmitted data may include source and channel coding bits, so the raw end-user efficiency of a system is generally less than frame efficiency.

Number of channels In TDMA system: It can be found by multiplying the number of TDMA slots per channel by the number of channels available

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

4.6.3 Code Division Multiple Access (CDMA)

In CDMA, the narrowband message signal is multiplied by a very large bandwidth signal called the spreading signal. The spreading signal is a pseudo-noise code sequence that has a chip rate which is orders of magnitudes greater than the data rate of the message. All users use the same carrier frequency and may transmit simultaneously. Each user has its own pseudorandom codeword which is approximately orthogonal to all other code words. The receiver performs a time correlation operation to detect only the specific desired codeword. The receiver needs to know the codeword used by the transmitter.

Near-far problem:

The near-far problem occurs when many mobile users share the same channel. In general, the strongest received mobile signal will capture the demodulator at a base station. In CDMA, stronger received signal levels raise the noise floor at the base station demodulators for the weaker signals, thereby decreasing the probability that weaker signals will be received. The power of multiple users at a receiver determines the noise floor after decorrelation.

Power control:

Power provided by each base station in a cellular system and assures that each mobile within the base station coverage area provides the same signal level to the base station receiver. Power control is implemented at the base station by rapidly sampling the radio signal strength indicator (RSSI) levels of each mobile and then sending a power change command over the forward radio link.

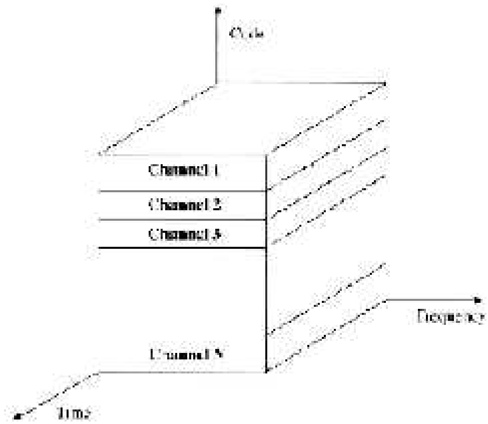


Fig 4.6.3 Code Division Multiple Access

Features of CDMA:

Many users of a CDMA system share the same frequency. Either TDD or FDD may be used.Unlike TDMA or FDMA, CDMA has a soft capacity limit. Increasing the number of users in a CDMA system raises the noise floor in a linear manner. Thus, there is no absolute limit on the number of users in CDMA. Rather, the system performance gradually degrades for all users as the number of users is increased, and improves as the number of users is decreased. Multipath fading may be substantially reduced because the signal is spread over a large spectrum. If the spread spectrum bandwidth is greater than the coherence bandwidth of the channel, the inherent frequency diversity will mitigate the effects of small-scale fading. Channel data rates are very high in CDMA systems. Consequently, the symbol (chip) duration is very short and usually much less than the channel delay spread. Since PN sequences have low autocorrelation, multipath which is delayed by more than a chip will appear as noise. A RAKE receiver can be used to improve reception by collecting time delayed versions of the required signal. Since CDMA uses co-channel cells, it can use macroscopic spatial diversity to provide soft handoff. Soft handoff is performed by the MSC, which can simultaneously monitor a particular user from two or more base stations. The MSC may chose the best version of the signal at any time without switching frequencies. Self-jamming is a problem in CDMA system. Self-jamming 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 transmissions of other users in the system. The near-far problem occurs at a CDMA receiver if an undesired user has a high detected power as compared to the desired user.

4.6.4 Space Division Multiple Access (SDMA)

SDMA utilizes the spatial separation of the users in order to optimize the use of the frequency spectrum. A primitive form of SDMA is when the same frequency is reused in different cells in a cellular wireless network. The radiated power of each user is controlled by Space division multiple access. SDMA serves different users by using spot beam antenna. These areas may be served by the same frequency or different frequencies. However for limited co-channel interference it is required that the cells be sufficiently separated. This limits the number of cells a region can be divided into and hence limits the frequency re-use factor. A more advanced approach can further increase the capacity of the network. This technique would enable frequency re-use within the cell. In a practical cellular environment it is improbable to have just one transmitter fall within the receiver beam width. Therefore it becomes imperative to use other multiple access techniques in conjunction with SDMA. When different areas are covered by the antenna beam, frequency can be re-used, in which case TDMA or CDMA is employed, for different frequencies FDMA can be used.

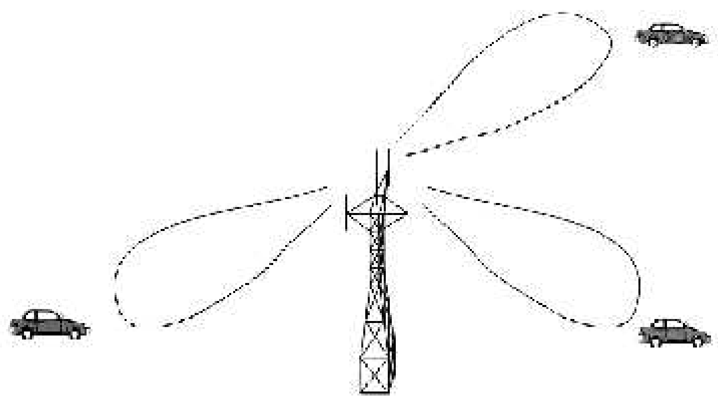


Fig 4.6.4 Space Division Multiple Access

4.7 CAPACITY OF CELLULAR CDMA

CDMA cellular systems typically employ universal frequency reuse, where the bandwidth is shared by all the cells and transmissions are distinguished through the assignment of unique spreading sequences. For such systems, multiple access interference from neighboring cells must be carefully accounted for. With cellular CDMA systems, any technique that reduces multiple access interference translates into a capacity gain. Since cellular CDMA systems use speech coding, the multiple access interference can be reduced by using voice activity detection along with variable rate speech transmission. This technique reduces the rate of the speech coder when silent periods are detected in the speech waveform. Voice activity detection has often been cited as an advantage of CDMA systems over TDMA systems. Cell sectoring is another very effective method for reducing multiple access interference, where each cell is sectored by using directional antennas. 120^o cell sectoring reduces the multiple access interference by roughly a factor of three (on average).

4.8 CAPACITY OF CELLULAR SDMA

In SDMA a number of users share the same available resources and are distinguished only in the spatial dimension. In traditional cellular systems the base station radiate the signal in all direction to cover the entire area of the cell, due to this we have both a waste of power and the transmission, in the directions where there are no mobile terminals to reach, of a signal which will be seen as interfering for co-channel cells, i.e. those cells using the same group of radio channels. Analogously, in reception, the antenna picks up signals coming from all directions, including noise and interference. These considerations have lead to the development of the SDMA technique, which is based on deriving and exploiting information on the spatial position of mobile terminals. In particular, the radiation pattern of the base station, both in transmission and reception is adapted to each different user so as to obtain the highest gain in the direction of the mobile user. Thus SDMA is recognized as one of the most useful techniques for improving the capacity of cellular systems. This technique allows different users to be served on the same frequency channel at the same time thus improving the spectral efficiency.

5. WIRELESS SYSTEM ANTENNAS AND STANDARDS

5.1 2G: SECOND GENERATION NETWORKS AND STANDARDS

Digital modulation formats were introduced in this generation with the main technology as TDMA/FDD and CDMA/FDD. The 2G systems introduced three popular TDMA standards and one popular CDMA standard in the market namely:

5.1.1 TDMA/FDD Standards:

(a) Global System for Mobile (GSM): The GSM standard, introduced by Group Special Mobile, was aimed at designing a uniform pan-European mobile system. It was the first fully digital system utilizing the 900 MHz frequency band. The initial GSM had 200 KHz radio channels, 8 full-rate or 16 half-rate TDMA channels per carrier, encryption of speech, low speed data services and support for SMS for which it gained quick popularity.

(b) Interim Standard 136 (IS-136): It was popularly known as North American Digital Cellular (NADC) system. In this system, there were 3 full-rate TDMA users over each 30 KHz channel. The need of this system was mainly to increase the capacity over the earlier analog (AMPS) system.

(c) Pacific Digital Cellular (PDC): This standard was developed as the counter- part of NADC in Japan. The main advantage of this standard was its low transmission bit rate which led to its better spectrum utilization.

5.1.2 CDMA/FDD Standard

Interim Standard 95 (IS-95): The IS-95 standard, also popularly known as CDMA-One, uses 64 orthogonally coded users and code words are transmitted simultaneously on each of 1.25 MHz channels. Certain services that have been standardized as a part of IS-95 standard are: short messaging service, slotted paging, over-the-air activation (meaning the mobile can be activated by the service provider without any third party intervention), enhanced mobile station identities etc.

5.1.3 2.5G Mobile Networks

The 2G standards for compatibility with increased throughput rates to support modern Internet application, the new data centric standards were developed to be overlaid on 2G standards and this is known as 2.5G standard. Here, the main upgradation techniques are:

- supporting higher data rate transmission for web browsing supporting e-mail traffic
- enabling location-based mobile service

2.5G networks also brought into the market some popular application, a few of which are: Wireless Application Protocol (WAP), General Packet Radio Service (GPRS), High Speed Circuit Switched Dada (HSCSD), Enhanced Data rates for GSM Evolution (EDGE) etc.

5.2 3G: THIRD GENERATION NETWORKS AND STANDARDS

3G is the third generation of mobile phone standards and technology, suppresing 2.5G. It is based on the International Telecommunication Union (ITU) family of standards under the International Mobile Telecommunications-2000 (IMT-2000).ITU launched IMT-2000 program, which, together with the main industry and standardization bodies worldwide, targets to implement a global frequency band that would support a single, ubiquitous wireless communication standard for all countries, to provide the framework for the definition of the 3G mobile systems. Several radio access technologies have been accepted by ITU as part of the IMT-2000 frame work.3G networks enable network operators to oer users a wider range of more advanced services while achieving greater network capacity through improved spectral efficiency. Services include wide-area wireless voice telephony, video calls, and broadband wireless data, all in a mobile environment. Additional features also include HSPA data transmission capabilities able to deliver speeds up to 14.4Mbit/s on the down link and 5.8Mbit/s on the uplink. 3G networks are wide area cellular telephone networks which evolved to incorporate high-speed internet access and video telephony. IMT-2000 defines a set of technical requirements for the realization of such targets, which can be summarized as follows:

(1) high data rates: 144 kbps in all environments and 2 Mbps in low-mobility and indoor environments

(2) symmetrical and asymmetrical data transmission

(3) circuit-switched and packet-switched-based services

(4) speech quality comparable to wire-line quality

(5) improved spectral efficiency

(6) several simultaneous services to end users for multimedia services

(7) seamless incorporation of second-generation cellular systems

(8) global roaming

(9) open architecture for the rapid introduction of new services and technology.

5.3 WIRELESS LOCAL LOOP

Wired technologies responding to need for reliable, high-speed access by residential, business, and government subscribers(ISDN, xDSL, cable modems). Increasing interest shown in competing wireless technologies for subscriber access. Wireless local loop (WLL) is broadly classified into

- Narrowband – offers a replacement for existing telephony services
- Broadband – provides high-speed two-way voice and data service

Advantages of WLL over Wired Approach

- Wireless systems are less expensive due to cost of cable installation that's avoided
- WLL systems can be installed in a small fraction of the time required for a new wired system
- Radio units installed for subscribers who want service at a given time
  - With a wired system, cable is laid out in anticipation of serving every subscriber in a given area

WLL Configuration

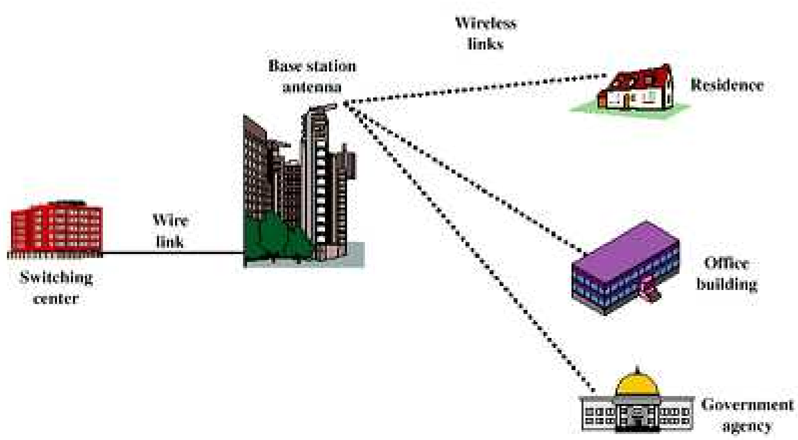


Fig 5.3 WLL Configuration

Propagation Considerations for WLL

WLL has been allocated a frequency band of 2 GHz to 40 GHz , especially the unused frequency bands available above 25 GHz There are wide unused frequency bands available above 25 GHz. At these high frequencies, wide channel bandwidths can be used, providing high data rates. Small size transceivers and adaptive antenna arrays can be used.Free space loss increases with the square of the frequency; losses are much higher in millimeter wave range. Above 10 GHz, attenuation effects due to rainfall and atmospheric or gaseous absorption are large. Multipath losses can be quite high i.e., trees near subscriber sites can lead to multipath fading. Multipath effects from the tree canopy are diffraction and scattering. Measurements in orchards found considerable attenuation values when the foliage is within 60% of the first Fresnel zone. Multipath effects highly variable due to wind.

5.4 BLUETOOTH

Bluetooth was founded in 1998 by : Ericsson, Intel, IBM, Toshiba and Nokia. A cable-replacement technology that can be used to connect almost any device to any other device. Radio interface enabling electronic devices to communicate wirelessly via short range (10 meters) ad-hoc radio connections.A standard for a small , cheap radio chip to be plugged into computers, printers, mobile phones, etc. Bluetooth uses the radio range of 2.45 GHz.Theoretical maximum bandwidth is 1 Mb/s.Several Bluetooth devices can form an ad hoc network called a “piconet”.In a piconet one device acts as a master (sets frequency hopping behavior) and the others as slaves. For example: A conference room with many laptops wishing to communicate with each other.

5.4.1 Bluetooth Architecture

Piconet: Each piconet has one master and up to 7 simultaneous slaves. Master is a device that initiates a data exchange. Slave is a device that responds to the master. All devices in a piconet hop together. Master gives slaves its clock and device ID. Non-piconet devices are in standby M=Master P=Parked S=Slave SB=Standby

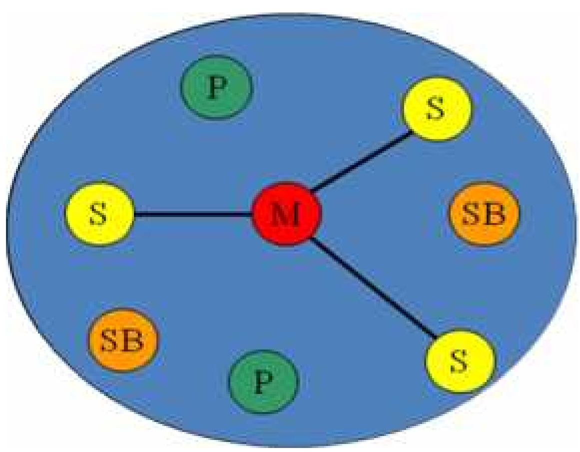


Fig 5.4.1.1 Piconet

Scatternet :Linking of multiple piconets through the master or slave devices. Bluetooth devices have point-to-multipoint capability to engage in Scatternet communication.

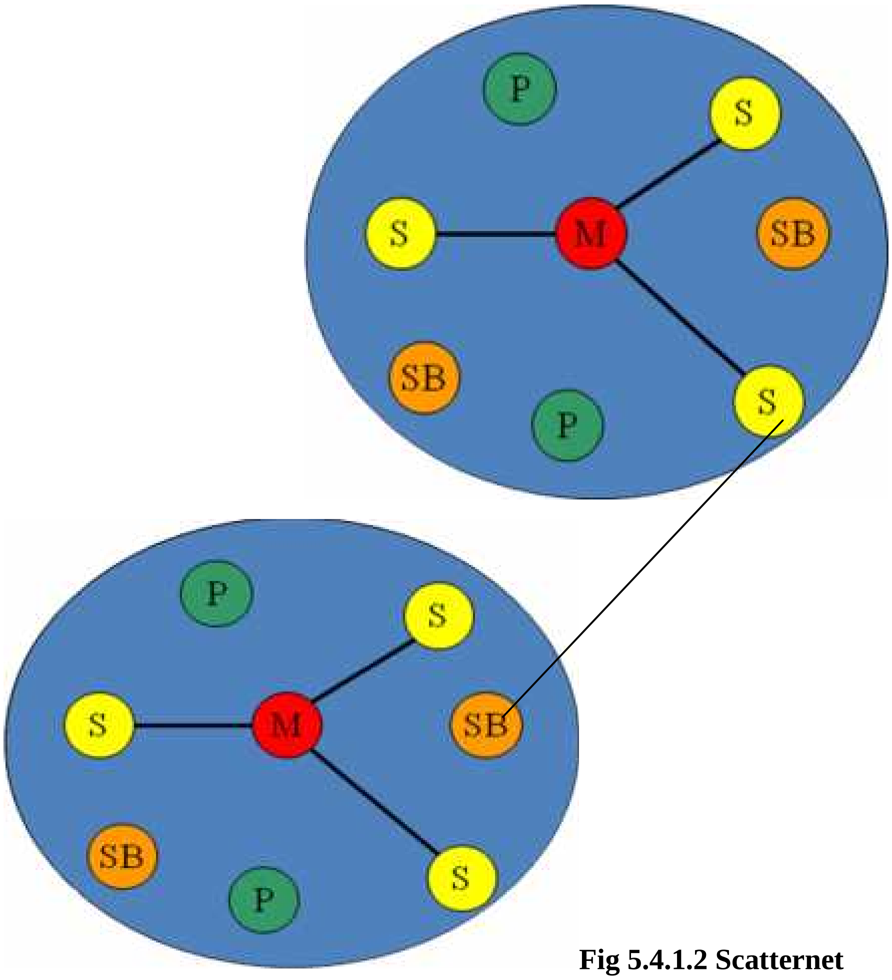


Fig 5.4.1.2 Scatternet

5.4.2 Physical links

Between master and slave(s), different types of links can be established. Two link types have been defined:

Synchronous Connection-Oriented (SCO) link :

- Support symmetrical, circuit-switched, point-to-point connections
- Typically used for voice traffic.
- Data rate is 64 kbit/s.

Asynchronous Connection-Less (ACL)

- Support symmetrical and asymmetrical, packet-switched, point-to-multipoint connections.

– Typically used for data transmission .
– Up to 433.9 kbit/s in symmetric or 723.2/57.6 kbit/s in asymmetric

5.4.3 Bluetooth Protocol Stack

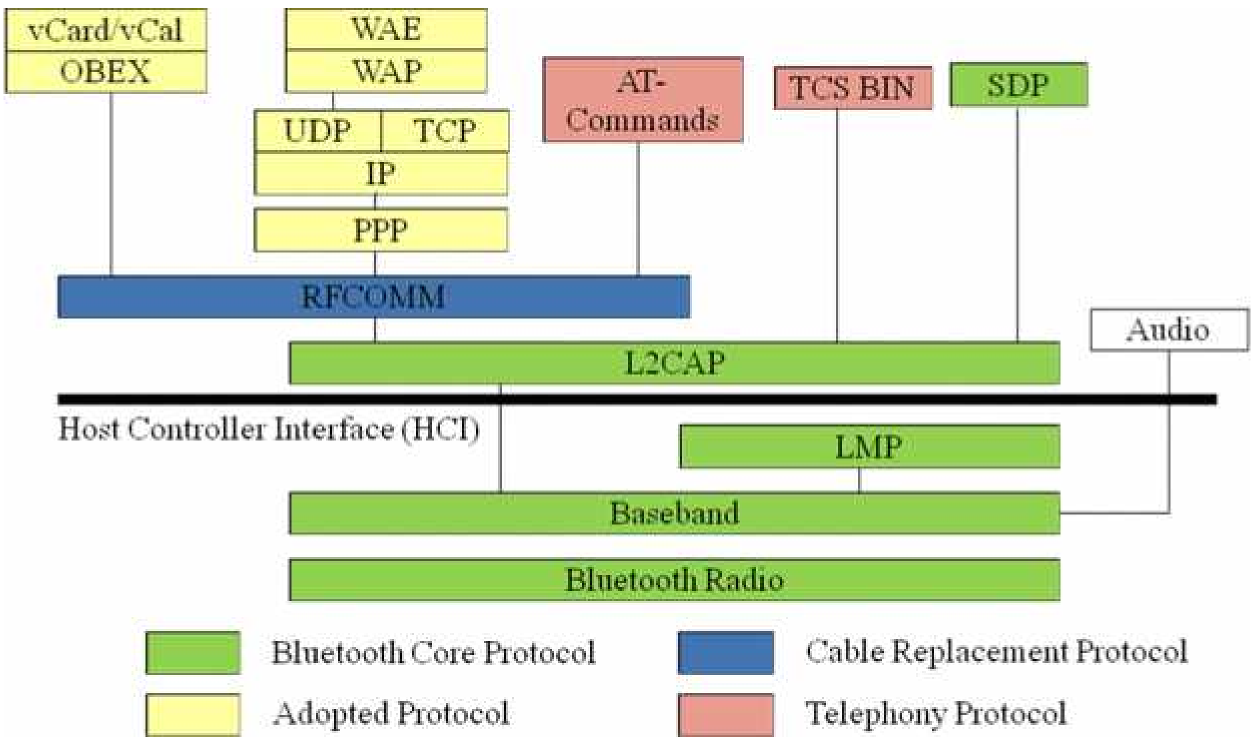


Fig 5.4.3 Bluetooth Protocol Stack

- **Bluetooth Radio** :It specifics details of the air interface, including frequency, frequency hopping, modulation scheme, and transmission power.
- **Baseband**: It is concerned with connection establishment within a piconet, addressing, packet format, timing and power control.
- **Link manager protocol (LMP)**: Establishes the link setup between Bluetooth devices and manages ongoing links, including security aspects (e.g. authentication and encryption), and control and negotiation of baseband packet size
- **Logical link control and adaptation protocol (L2CAP)**: Adapts upper layer protocols to the baseband layer. Provides both connectionless and connection-oriented services.
- **Service discovery protocol (SDP)**: Handles device information, services, and queries for service characteristics between two or more Bluetooth devices.

- **Host Controller Interface (HCI) :** Provides an interface method for accessing the Bluetooth hardware capabilities. It contains a command interface, which acts between the Baseband controller and link manager
- **TCS BIN (Telephony Control Service) :** bit-oriented protocol that defines the call control signaling for the establishment of voice and data calls between Bluetooth devices.
- **OBEX(OBject EXchange)** : Session-layer protocol for the exchange of objects, providing a model for object and operation representation
- **RFCOMM**: A reliable transport protocol, which provides emulation of RS232 serial ports over the L2CAP protocol
- **WAE/WAP**: Bluetooth incorporates the wireless application environment and the wireless application protocol into its architecture.

5.4.4 Connection Establishment States

Standby: State in which Bluetooth device is inactive, radio not switched on, enable low power operation.

Page : Master enters page state and starts transmitting paging messages to Slave using earlier gained access code and timing information.

Page Scan: Device periodically enters page state to allow paging devices to establish connections.

Inquiry: State in which device tries to discover all Bluetooth enabled devices in the close vicinity.

Inquiry scan :Most devices periodically enter the inquiry scan state to make themselves available to inquiring devices.

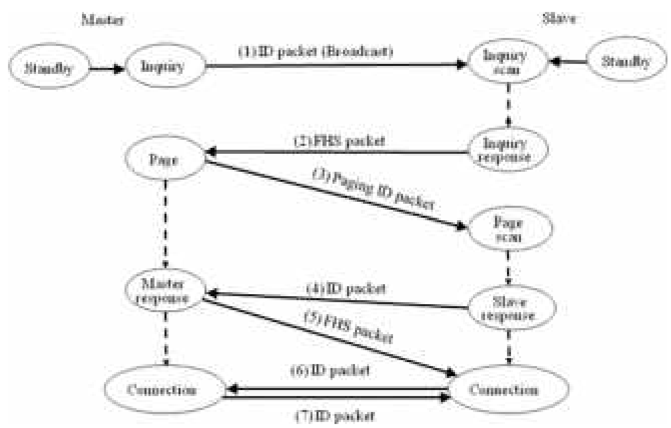


Fig 5.4.4 Inquiry and Page

5.4.5 Bluetooth Security

The following are the three basic security services specified in the Bluetooth standard

Authentication :Verifying the identity of communicating devices. User authentication is not provided natively by Bluetooth.

Confidentiality :Preventing information compromise caused by eavesdropping by ensuring that only authorized devices can access and view data.

Authorization :Allowing the control of resources by ensuring that a device is authorized to use a service before permitting it to do so.

5.5 ADVANCED MOBILE PHONE SYSTEM (AMPS)

AMPS is an analog mobile cell phone system standard developed by Bell Labs, and officially introduced in the Americas in 1978. It was the primary analog mobile phone system in North America (and other locales) through the 1980s and into the 2000s. There were no longer required to support AMPS and companies such as AT&T and Verizon have discontinued this service permanently. AMPS was discontinued in Australia in September 2000.

AMPS cellular service operated in the 850 MHz Cellular band. For each market area, the United States (FCC) allowed two licensees (networks) known as "A" and "B" carriers. Each carrier within a market used a specified "block" of frequencies consisting of 21 control channels and 395 voice channels. Originally, the B (wireline) side license was usually owned by the local phone company, and the A (non-wireline) license was given to wireless telephone providers.

At the inception of cellular in 1983, the FCC had granted each carrier within a market 333 channel pairs (666 channels total). By the late 1980s, the cellular industry's subscriber base had grown into the millions across America and it became necessary to add channels for additional capacity. The additional frequencies were from the band held in reserve for future (inevitable) expansion. These frequencies were immediately adjacent to the existing cellular band. These bands had previously been allocated to UHF TV channels 70–83.

Each duplex channel was composed of 2 frequencies. 416 of these were in the 824–849 MHz range for transmissions from mobile stations to the base stations, paired with 416 frequencies in the 869–894 MHz range for transmissions from base stations to the mobile stations. Each cell site used a different subset of these channels than its neighbors to avoid interference. This significantly reduced the number of channels available at each site in real-world systems. Each AMPS channel had a one way bandwidth of 30 kHz, for a total of 60 kHz for each duplex channel.

Laws were passed in the US which prohibited the FCC type acceptance and sale of any receiver which could tune the frequency ranges occupied by analog AMPS cellular services. Though the service is no longer offered, these laws remain in force.

DIGITAL AMPS:

Later, many AMPS networks were partially converted to D-AMPS, often referred to as **TDMA** (though TDMA is a generic term that applies to many cellular systems). D-AMPS was a digital, 2G standard used mainly by AT&T Mobility . In most areas, D-AMPS is no longer offered and has been replaced by more advanced digital wireless networks.

SUCCESSOR TECHNOLOGIES:

AMPS and D-AMPS have now been phased out in favor of either CDMA2000 or Global System for Mobile Communications(GSM) which allow for higher capacity data transfers for services such as WAP, Multimedia Messaging System (MMS), and wireless Internet access. There are some phones capable of supporting AMPS, D-AMPS and GSM all in one phone (using the GAIT standard).

5.6 GSM (Global System for Mobile Communications)

GLOBAL SYSTEM FOR MOBILE (GSM):

CEPT, a European group, began to develop the Global System for Mobile TDMA system in June 1982.GSM has two objectives: pan-European roaming, which offers compatibility throughout the European continent, and interaction with the integrated service digital network (ISDN), which offers the capability to extend the single-subscriber-line system to a multiservice system with various services currently offered only through diverse telecommunications networks. System capacity was not an issue in the initial development of GSM, but due to the unexpected, rapid growth of cellular service, 35 revisions have been made to GSM since the first issued specification. The first commercial GSM system, calledD2, was implemented in Germany in 1992.

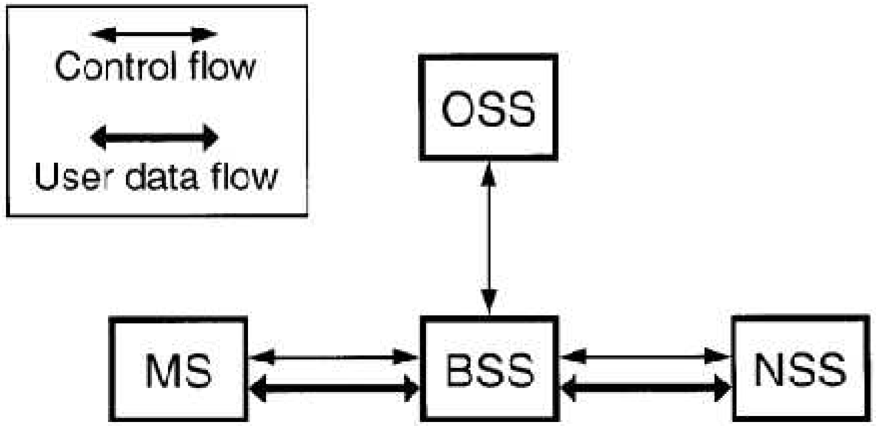


Figure 5.6.1 The external environment of BSS

GSM Architecture :

GSM consists of many subsystems, such as the mobile station (MS), the base station sub system (BSS), the network and switching subsystem (NSS), and the operation subsystem (OSS) in figure 5.6.1

1. The Mobile Station: The MS may be a stand-alone piece of equipment for certain services or support the connection of external terminals, such as the interface for a personal computer or fax. The MS includes mobile equipment (ME) and a subscriber identity module (SIM). ME does not need to be personally assigned to one subscriber. The SIM is a subscriber module which stores all the subscriber-related information. When a subscriber's SIM is inserted into the ME of an MS, that MS belongs to the subscriber, and the call is delivered to that MS. The ME is not associated with a called number—it is linked to the SIM. In this case, any ME can be used by a subscriber when the SIM is inserted in the ME.

2. Base Station Subsystem: The BSS connects to the MS through a radio interface and also connects to the NSS. The BSS consists of a base transceiver station (BTS) located at the antenna site and a base station controller (BSC) that may control several BTSs. The BTS consists of radio transmission and reception equipment similar to the ME in an MS. A transcoder/rate adaption unit (TRAU) carries out encoding and speech decoding and rate adaptation for transmitting data. As a subpart of the BTS, the TRAU may be sited away from the BTS, usually at the MSC. In this case, the low transmission rate of speech code channels allows more compressed transmission between the BTS and the TRAU, which is sited at the MSC. GSM uses the open system interconnection (OSI). There are three common interfaces based on OSI (Fig 5.6.2): a common radio interface, called air interface, between the MS and BTS, an interface A between the MSC and BSC, and an A-bis interface between the BTS and BSC. With these common interfaces, the system operator can purchase the product of manufacturing company A to interface with the product of manufacturing company B. The difference between interface and protocol is that an interface represents the point of contact between two adjacent entities (equipment or systems) and a protocol provides information flows through the interface. For example, the GSM radio interface is the transit point for information flow pertaining to several protocols.

3. Network and Switching Subsystem: NSS (see Fig5.6.3) in GSM uses an intelligent network (IN). The IN's attributes will be described later. A signaling NSS includes the main switching functions of GSM. NSS manages the communication between GSM users and other telecommunications users. NSS management consists of:

Mobile service switching center (MSC): Coordinates call set-up to and from GSM users. An MSC controls several BSCs.

Interworking function (IWF): A gateway for MSC to interface with external networks for communication with users outside GSM, such as packet-switched public data network (PSPDN) or circuit-switched public data network (CSPDN).The role of the IWF depends on the type of user data and the network to which it interfaces.

Home location register (HLR): Consists of a stand-alone computer without switching capabilities, a database which contains subscriber information, and information related to the subscriber's current location, but not the actual location of the subscriber. A subdivision of HLR is the authentication center (AUC). The AUC manages the security data for subscriber authentication. Another sub-division of HLR is the equipment identity register (EIR) which stores the data of mobile equipment (ME) or ME-related data.

Visitor location register (VLR): Links to one or more MSCs, temporarily storing subscription data currently served by its corresponding MSC, and holding more detailed data than the HLR. For example, the VLR holds more current subscriber location information than the location information at the HLR.

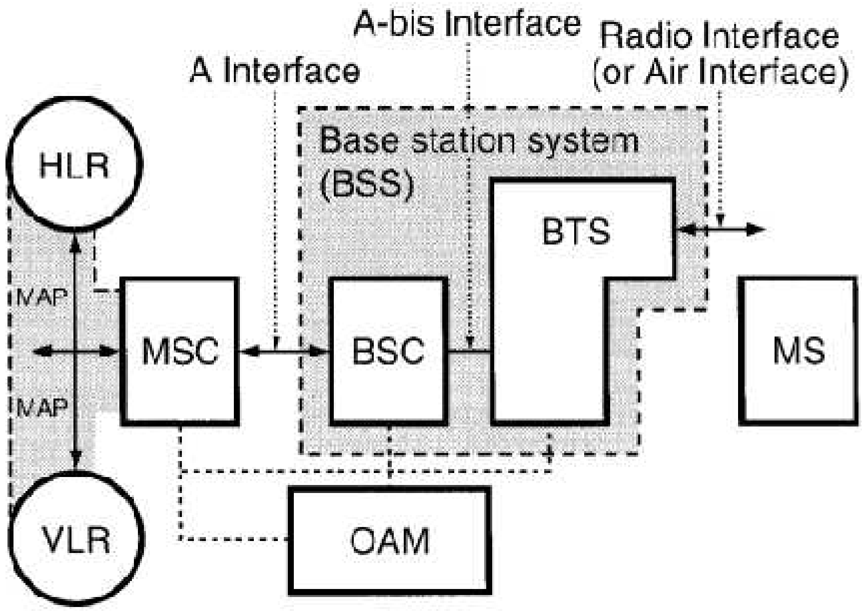
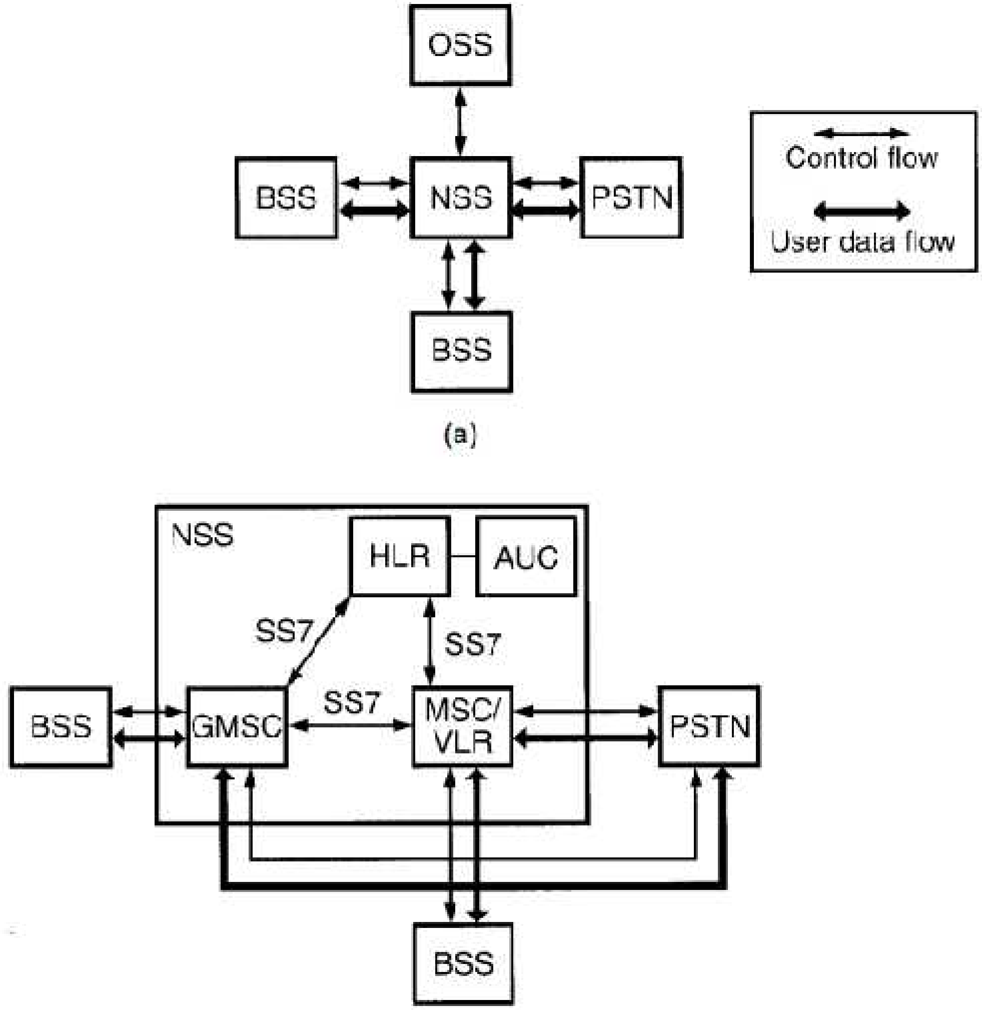


Fig 5.6.2. Functional architecture and principal interfaces

Gateway MSC (GMSC): In order to set up a requested call, the call is initially routed to a gateway MSC, which finds the correct HLR by knowing the directory number of the GSM subscriber. The GMSC has an interface with the external network for gatewaying, and the network also operates the full Signaling System 7 (SS7) signaling between NSS machines.

Signaling transfer point (STP): Is an aspect of the NSS function as a stand-alone node or in the same equipment as the MSC. STP optimizes the cost of the signaling transport among MSC/VLR, GMSC, and HLR.

As mentioned earlier, NSS uses an intelligent network. It separates the central data base (HLR) from the switches (MSC) and uses STP to transport signaling among MSC and HLR.



(b)

Fig 5.6.3 NSS and its environment (a) the external environment; (b) the internal structure

4. Operation Subsystem: There are three areas of OSS, as shown in Fig 5.6.4 (1) network operation and maintenance functions, (2) subscription management, including charging and billing, and (3)mobile equipment management. These tasks require interaction between some or all of the infrastructure equipment. OSS is implemented in any existing network.

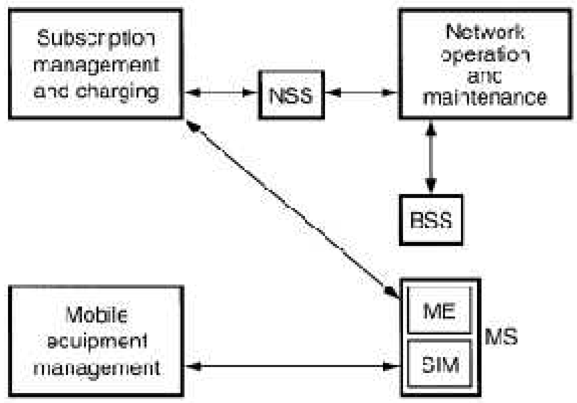


Fig 5.6.4 OSS organization

5.7 INTERIM STANDARD 95 (IS-95)

IS-95 is the first CDMA-based digital cellular standard by Qualcomm. The brand name for IS-95 is **cdmaOne**. IS-95 is also known as TIA-EIA-95. It is a 2G mobile telecommunications standard that uses CDMA, a multiple access scheme for digital radio, to send voice, data and signaling data (such as a dialed telephone number) between mobile telephones and cell sites. CDMA or "code division multiple access" is a digital radio system that transmits streams of bits (PN codes). CDMA permits several radios to share the same frequencies. Unlike TDMA "time division multiple access", a competing system used in 2G GSM, all radios can be active all the time, because network capacity does not directly limit the number of active radios. Since larger numbers of phones can be served by smaller numbers of cell-sites, CDMA-based standards have a significant economic advantage over TDMA-based standards, or the oldest cellular standards that used frequency-division multiplexing. In North America, the technology competed with Digital AMPS(IS-136, a TDMA technology). It is now being supplanted by IS-2000 (CDMA2000), a later CDMA-based standard.

CAPACITY:

IS-95 and its use of CDMA techniques, like any other communications system, have their throughput limited according to Shannon's theorem. Accordingly, capacity improves with SNR and bandwidth. IS-95 has a fixed bandwidth, but fares well in the digital world because it takes active steps to improve SNR.

With CDMA, signals that are not correlated with the channel of interest (such as other PN offsets from adjacent cellular base stations) appear as noise, and signals carried on other Walsh codes (that are properly time aligned) are essentially removed in the de-spreading process. The variable-rate nature of traffic channels provide lower-rate frames to be transmitted at lower power causing less noise for other signals still to be correctly received. These factors provide an inherently lower noise level than other cellular technologies allowing the IS-95 network to squeeze more users into the same radio spectrum.

Active (slow) power control is also used on the forward traffic channels, where during a call, the mobile sends signaling messages to the network indicating the quality of the signal. The network will control the transmitted power of the traffic channel to keep the signal quality just good enough, thereby keeping the noise level seen by all other users to a minimum.

The receiver also uses the techniques of the rake receiver to improve SNR as well as perform soft handoff.

Advantages of CDMA:

- Capacity is CDMA's biggest asset. It can accommodate more users per MHz of bandwidth than any other technology.
    - 3 to 5 times more than GSM
- CDMA has no built-in limit to the number of concurrent users.
- CDMA uses precise clocks that do not limit the distance a tower can cover.
- CDMA consumes less power and covers large areas so cell size in CDMA is larger.
- CDMA is able to produce a reasonable call with lower signal (cell phone reception) levels.
- CDMA uses Soft Handoff, reducing the likelihood of dropped calls.
- CDMA's variable rate voice coders reduce the rate being transmitted when speaker is not talking, which allows the channel to be packed more efficiently.
- Has a well-defined path to higher data rates.

Disadvantages of CDMA:

- Most technologies are patented and must be licensed from Qualcomm.
- *Breathing* of base stations, where coverage area shrinks under load. As the number of subscribers using a particular site goes up, the range of that site goes down.
- Currently CDMA covers a smaller portion of the world as compared to GSM which has more subscribers and is in more countries overall worldwide.

5.8 DIGITAL ENHANCED CORDLESS TELECOMMUNICATIONS

(Digital European Cordless Telecommunications), usually known by the acronym **DECT**, is primarily used for creating cordless phone systems. It originated in Europe, where it is the universal standard, replacing earlier cordless phone standards, such as 900 MHz CT1 and CT2.North American adoption was delayed by United States radio frequency regulations. This forced development of a variation of DECT, called **DECT 6.0** using a slightly different frequency range. The technology is nearly identical, but the frequency difference makes the technology incompatible with systems in other areas, even from the same manufacturer. DECT has almost universally replaced other standards in most countries where it is used, with the exception of North America. DECT is used primarily in home and small office systems, but is also available in many PBX systems for medium and large businesses. DECT can also be used for purposes other than cordless phones. Voice applications, such as baby monitors, are becoming common. Data applications also exist, but have been eclipsed by Wi-Fi. 3G & 4G cellular also competes with both DECT and Wi-Fi for both voice and data. Nowadays you can find DECT as well in special applications like Remote Controls for industrial applications. Dialog Semiconductor announced the first commercially available DECT ULE devices. Unlike standard DECT, the low power variant enables this standard to be used in battery powered devices such as smartphone app controllable home automation or security systems.

DECT handsets and bases from different manufacturers typically work together at the most basic level of functionality: making and receiving calls. The DECT standard includes a standardized interoperability profile for simple telephone capabilities, called GAP, which most manufacturers implement. The standard also contains several other interoperability profiles, for data and for radio local-loop services.

Characteristics

- Frequency: 1880-1990 MHz
- Channels: 120 full duplex
- Duplex mechanism: TDD (Time Division Duplex) with 10 ms frame length
- Multplexing scheme: FDMA with 10 carrier frequencies, TDMA with 2x 12 slots
- Modulation: digital, Gaussian Minimum Shift Key (GMSK)
- Power: 10 mW average (max. 250 mW)
- Range: ca 50 m in buildings, 300 m open space

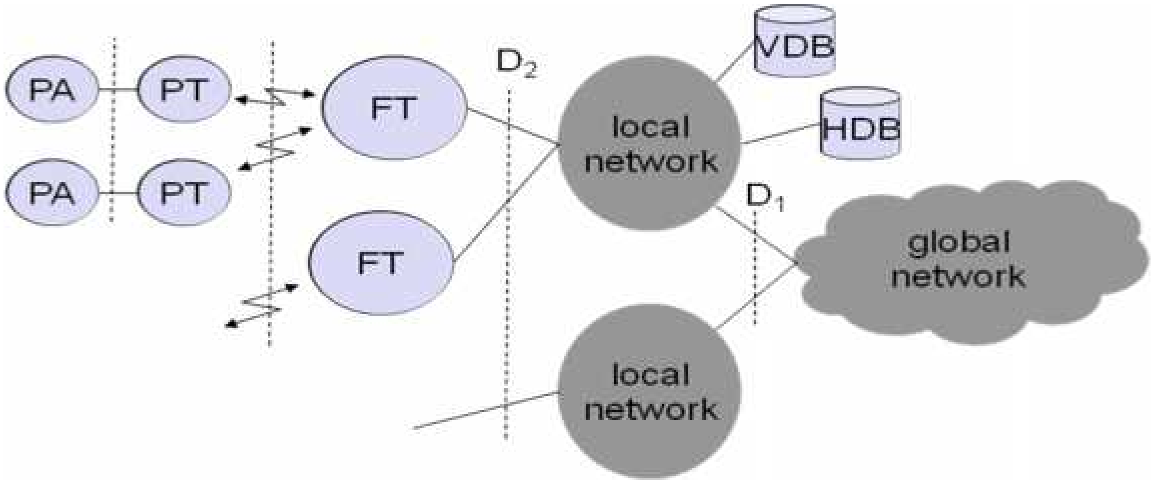


Fig 5.8 DECT system architecture reference model

Global network : PSTN,ISDN,PLMN(Public Land Mobile Network)

Local network: simple Switching, Intelligent call forwarding, address translation(LAN)

HDB: Home data base

VDB: Visitor Data Base

The local network has fixed Termination (Ft) and Portable radio Termination(PT)

APPLICATION:

The DECT standard fully specifies a means for a portable unit, such as a cordless telephone, to access a fixed telecoms network via radio. But, unlike the standards, does not specify any internal aspects of the fixed network itself. Connectivity to the fixed network (which may be of many different kinds) is done through a "Radio Fixed Part" to terminate the radio link, to connect calls to the fixed network. In most cases the gateway connection is to PSTN or telephone jack, although connectivity with newer technologies such as IP has become available. There are also other devices such as DECT, and in these devices there is no gateway functionality.

The DECT standard originally envisaged three major areas of application:

- Domestic cordless telephony, using a single base station to connect one or more handsets to the public telecoms network.

- Enterprise premises cordless PABXs and wireless LANs, using many base stations for coverage. Calls continue as users move between different coverage cells, through a mechanism called handover. Calls can be both within the system and to the public telecoms network.
- Public access, using large numbers of base stations to provide high capacity building or urban area coverage as part of a public telecoms network.

Of these, the domestic application (cordless home telephones) has been extremely successful.

5.9 RADIO-FREQUENCY IDENTIFICATION (RFID) ANTENNAS

RFID Antennas is the wireless non-contact use of radio-frequency electromagnetic fields to transfer data, for the purposes of automatically identifying and tracking tags attached to objects. The tags contain electronically stored information. Some tags are powered by and read at short ranges (a few meters) via magnetic fields (electromagnetic induction, and then act as a passive transponder to emit microwaves or UHF radio waves (i.e., electromagnetic radiation at high frequencies). Others use a local power source such as a battery, and may operate at hundreds of meters. Unlike a bar code, the tag does not necessarily need to be within line of sight of the reader, and may be embedded in the tracked object.

RFID tags are used in many industries. An RFID tag attached to an automobile during production can be used to track its progress through the assembly line. Pharmaceuticals can be tracked through warehouses. Livestock and pets may have tags injected, allowing positive identification of the animal. On off-shore oil and gas platforms, Since RFID tags can be attached to clothing, possessions, or even implanted within people the possibility of reading personally-linked information without consent has raised privacy concerns.

DESIGN:

A radio-frequency identification system uses *tags*, or *labels* attached to the objects to be identified. Two-way radio transmitter-receivers called *interrogators* or *readers* send a signal to the tag and read its response.

RFID tags can be either passive, active or battery-assisted passive. An active tag has an on-board battery and periodically transmits its ID signal. A battery-assisted passive (BAP) has a small battery on board and is activated when in the presence of an RFID reader. A passive tag is cheaper and smaller because it has no battery. However, to start operation of passive tags, they must be illuminated with a power level roughly three magnitudes stronger than for signal transmission. That makes a difference in interference and in exposure to radiation.

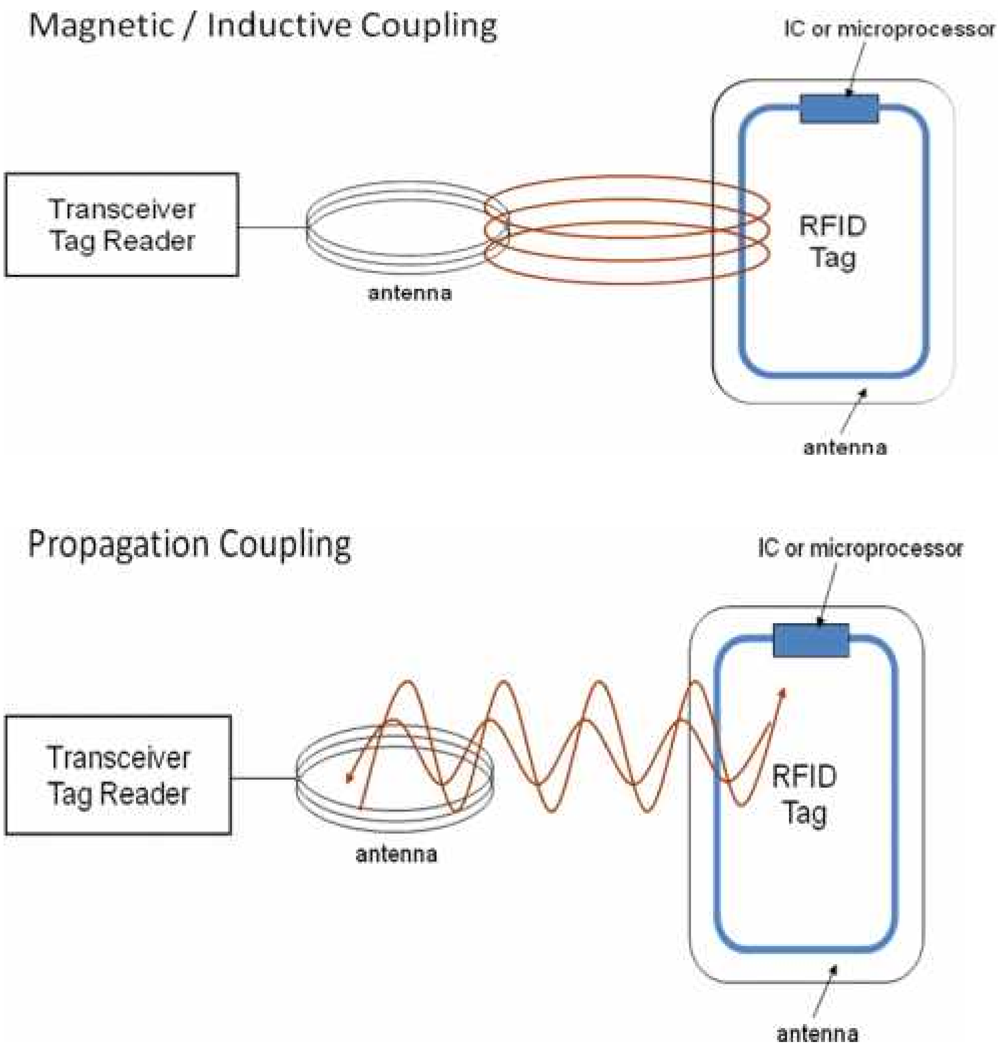


Fig 5.9 RFID Antenna

Tags may either be read-only, having a factory-assigned serial number that is used as a key into a database, or may be read/write, where object-specific data can be written into the tag by the system user. Field programmable tags may be write-once, read-multiple; "blank" tags may be written with an electronic product code by the user. A tag with no inherent identity is always vulnerable to manipulation.

RFID tags contain at least two parts: an integrated circuit for storing and processing information, modulating and demodulating a radio-frequency (RF) signal, collecting DC power from the incident reader signal, and other specialized functions; and an antenna for receiving and transmitting the signal.

The tag information is stored in a non-volatile memory. The RFID tag includes either a chip-wired logic or a programmed or programmable data processor for processing the transmission and sensor data, respectively.

An RFID reader transmits an encoded radio signal to interrogate the tag. The RFID tag receives the message and then responds with its identification and other information. This may be only a unique tag serial number, or may be product-related information such as a stock number, lot or batch number, production date, or other specific information.

USES:

The RFID tag can be affixed to an object and used to track and manage inventory, assets, people, etc. For example, it can be affixed to cars, computer equipment, books, mobile phones, etc.

RFID offers advantages over manual systems that The tag can be read if passed near a reader, even if it is covered by the object or not visible. The tag can be read inside a case, carton, box or other container, and unlike barcodes, RFID tags can be read hundreds at a time. Bar codes can only be read one at a time using current devices.

RFID can be used in a variety of applications:

- Access management
- Tracking of goods
- Tracking of persons and animals
- Toll collection

5.10 MOBILE ANTENNAS

The requirement of a mobile (motor-vehicle–mounted) antenna is an omnidirectional antenna that can be located as high as possible from the point of reception. However, the physical limitation of antenna height on the vehicle restricts this requirement. Generally, the antenna should at least clear the top of the vehicle. Patterns for two types of mobile antenna are shown in Figure below.

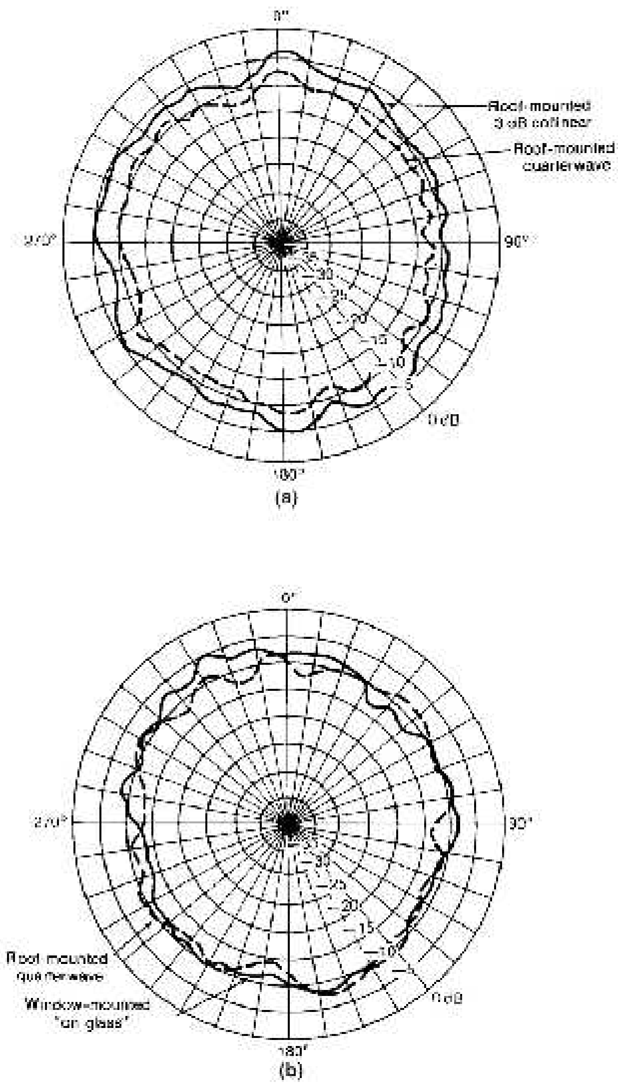


Figure 5.10 Mobile antenna patterns (a) Roof mounted 3-dB-gain collinear antenna versus roof-mounted quarter-wave antenna, (b) Window- mourned “on-glass” gain antenna versus roof-mounted quarter-wave antenna.

5.10.1 Roof-Mounted Antenna

The antenna pattern of a roof-mounted antenna is more or less uniformly distributed around the mobile unit when measured at an antenna range in free space as shown in Figure below. The 3-dBhigh- gain antenna shows a 3-dBgain over the quarter-wave antenna. However, the gain of the antenna used at the mobile unit must be limited to 3 dB because the cell-site antenna is rarely as high as the broadcasting antenna and out-of-sight conditions often prevail. The mobile antenna with a gain of more than 3 dB can receive only a limited portion of the total multipath signal in the elevation as measured under the out-of-sight condition.

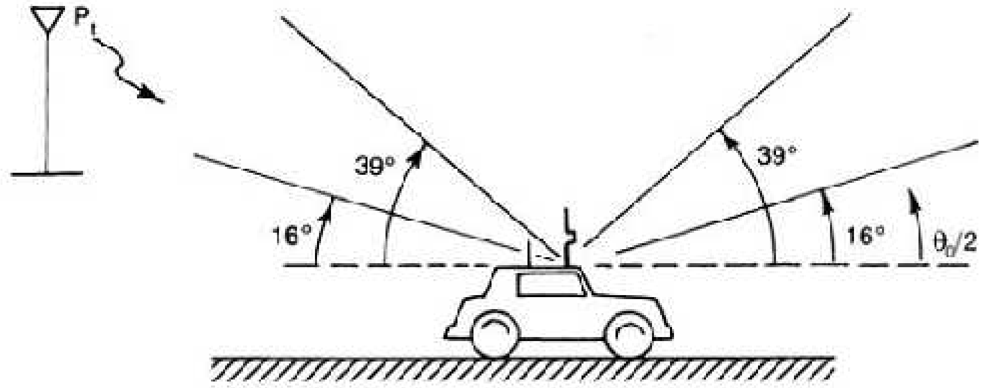


Fig 5.10.1 Vertical angle of signal arrival

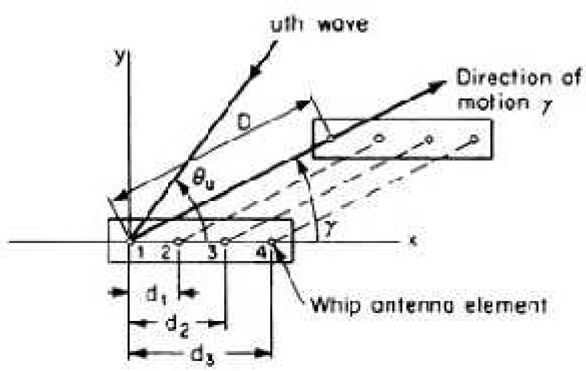
5.10.2 Glass-Mounted Antennas

There are many kinds of glass-mounted antennas. Energy is coupled through the glass; therefore, there is no need to drill a hole. However, some energy is dissipated on passage through the glass. The antenna gain range is 1 to 3 dB depending on the operating frequency. The position of the glass-mounted antenna is always lower than that of the roof-mounted antenna; generally there is a 3-dBdifference between these two types of antenna. Also, glass mounted antennas cannot be installed on the shaded glass found in some motor vehicles because this type of glass has a high metal content.

5.10.3 Mobile High-Gain Antennas

A high-gain antenna used on a mobile unit has been studied. This type of high-gain antenna should be distinguished from the directional antenna. In the directional antenna, the antenna beam pattern is suppressed horizontally; in the high-gain antenna, the pattern is suppressed vertically. To apply either a directional antenna or a high-gain antenna for reception in a radio environment, we must know the origin of the signal. If we point the directional antenna opposite to the transmitter site, we would in theory receive nothing. In a mobile radio environment, the scattered signals arrive at the mobile unit from every direction with equal probability. That is why an omnidirectional antenna must be used. The scattered signals also arrive from different elevation angles. Lee and Brandt used two types of antenna, one $\lambda/4$ whip antenna with an elevation coverage of 39° and one 4-dB-gain antenna (4-dB gain with respect to the gain of a dipole) with an elevation coverage of 16° and measured the angle of signal arrival in the suburban Keyport-Matawan area of New Jersey. There are two types of test: a line-of-sight condition and an out-of-sight condition. In Lee and Brandt's study, the transmitter was located at an elevation of approximately 100 m (300 ft) above sea level. The measured areas were about 12 m (40 ft) above sea level and the path length about 3 mi. The received signal from the 4-dB-gain antenna was 4 dB stronger than that from the whip antenna under line-of-sight conditions. This is what we would expect. However, the received signal from the 4-dB-gain antenna was only about 2 dB stronger than that from the whip antenna under out-of-sight conditions. This is surprising. The reason for the latter observation is that the scattered signals arriving under out-of- sight conditions are spread over a wide elevation angle. A large portion of the signals outside the elevation angle of 16◦ cannot be received by the high-gain antenna. We may calculate the portion being received by the high-gain antenna from the measured beamwidth. For instance, suppose that a 4:1 gain (6 dBi) is expected from the high-gain antenna, but only 2.5:1 is received. Therefore, 63 percent of the signal is received by the 4-dB-gain antenna (i.e., 6 dBi) and 37 percent is felt in the region between 16 and 39◦

	Gain, dBi	Linear ratio	$\theta_0/2$, degrees
Whip antenna (2 dB above isotropic)	2	1.58:1	39
High-gain antenna	6	4:1	16
Low-gain antenna	4	2.5:1	24



Therefore, a 2- to 3-dB-gain antenna (4 to 5 dBi) should be adequate for general use. An antenna gain higher than 2 to 3 dB does not serve the purpose of enhancing reception level. Moreover, measurements reveal that the elevation angle for scattered signals received in urban areas is greater than that in suburban areas.

5.11 GENERAL PACKET RADIO SERVICE (GPRS)

GPRS is a packet oriented mobile data service on the 2G and 3G cellular communication system's global system for mobile communications (GSM). GPRS was originally standardized by European Telecommunications Standards Institute (ETSI) in response to the earlier CDPD and i-mode packet-switched cellular technologies. It is now maintained by the (3GPP). GPRS usage is typically charged based on volume of data transferred, contrasting with circuit switched data, which is usually billed per minute of connection time. Usage above the bundle cap is either charged per megabyte or disallowed.

GPRS is a best-effort service, implying variable throughput and latency that depend on the number of other users sharing the service concurrently, as opposed to circuit switching, where a certain quality of service (QoS) is guaranteed during the connection. In 2G systems, GPRS provides data rates of 56–114 kbit/second.2G cellular technology combined with GPRS is sometimes described as *2.5G*, that is, a technology between the second (2G) and third (3G) generations of mobile telephony. It provides moderate-speed data transfer, by using unused (TDMA) channels in, for example, the GSM system.

GPRS supports the following protocols:

- (IP). In practice, built-in mobile browser use IPv4 since IPv6 was not yet popular.
- Point-to-point protocol (PPP). In this mode PPP is often not supported by the mobile phone operator but if the mobile is used as a modem to the connected computer, PPP is used to tunnel IP to the phone. This allows an IP address to be assigned dynamically (IPCP not DHCP) to the mobile equipment.
- X.25 connections. This is typically used for applications like wireless payment terminals, although it has been removed from the standard. X.25 can still be supported over PPP, or even over IP, but doing this requires either a network-based router to perform encapsulation or intelligence built into the end-device/terminal; e.g., user equipment (UE).

When TCP/IP is used, each phone can have one or more IP addresses allocated. GPRS will store and forward the IP packets to the phone even during handover. The TCP handles any packet loss (e.g. due to a radio noise induced pause).

GPRS SYSTEM ARCHITECTURE

Devices supporting GPRS are divided into three classes: Class A Can be connected to GPRS service and GSM service (voice, SMS), using both at the same time. Such devices are known to be available today. Class B Can be connected to GPRS service and GSM service (voice, SMS), but using only one or the other at a given time. During GSM service (voice call or SMS), GPRS service is suspended, and then resumed automatically after the GSM service (voice call or SMS) has concluded. Most GPRS mobile devices are Class B.Class C Are connected to either GPRS service or GSM service (voice, SMS). Must be switched manually between one or the other service.

A true Class A device may be required to transmit on two different frequencies at the same time, and thus will need two radios. To get around this expensive requirement, a GPRS mobile may implement the dual transfer mode (DTM) feature.

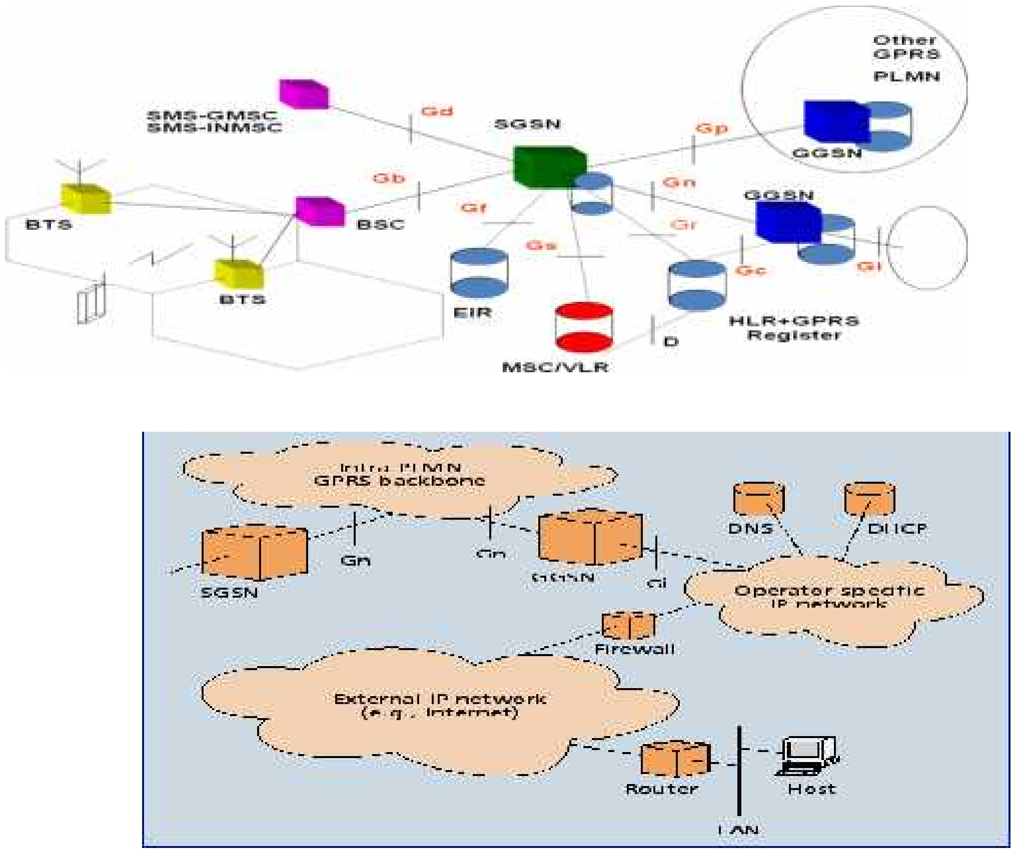


Fig 5.11 GPRS Internet Connection

A DTM-capable mobile may use simultaneous voice and packet data, with the network coordinating to ensure that it is not required to transmit on two different frequencies at the same time. Such mobiles are considered pseudo-Class A, sometimes referred to as "simple class A". Some networks support DTM since 2007.

Usability

The maximum speed of a GPRS connection offered in 2003 was similar to a modem connection in an analog wire telephone network, about 32–40 kbit/s, depending on the phone used. Latency is very high; round-trip time (RTT) is typically about 600–700 ms and often reaches 1s. GPRS is typically prioritized lower than speech, and thus the quality of connection varies greatly.

Devices with latency/RTT improvements (via, for example, the extended UL TBF mode feature) are generally available. Also, network upgrades of features are available with certain operators. With these enhancements the active round-trip time can be reduced, resulting in significant increase in application-level throughput speeds.