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As per Revised Syllabus of

SAVITRIBAI PHULE PUNE UNIVERSITY

Choice Based Credit System (CBCS)

T.E. (E&Tc) Semester - VI

CELLULAR NETWORKS

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PREFACE

The importance of **Cellular Networks** is well known in computer engineering fields. Overwhelming response to my books on various subjects inspired me to write this book. The book is structured to cover the key aspects of the subject **Cellular Networks.**

The book uses plain, lucid language to explain fundamentals of this subject. The book provides logical method of explaining various complicated concepts and stepwise methods to explain the important topics. Each chapter is well supported with necessary illustrations, practical examples and solved problems. All chapters in this book are arranged in a proper sequence that permits each topic to build upon earlier studies. All care has been taken to make students comfortable in understanding the basic concepts of this subject.

Representative questions have been added at the end of each section to help the students in picking important points from that section.

The book not only covers the entire scope of the subject but explains the philosophy of the subject. This makes the understanding of this subject more clear and makes it more interesting. The book will be very useful not only to the students but also to the subject teachers. The students have to omit nothing and possibly have to cover nothing more.

I wish to express my profound thanks to all those who helped in making this book a reality. Much needed moral support and encouragement is provided on numerous occasions by my whole family. I wish to thank the **Publisher** and the entire team of **Technical Publications** who have taken immense pain to get this book in time with quality printing.

Any suggestion for the improvement of the book will be acknowledged and well appreciated.

Author
V. S. Bagad

Dedicated to God.

SYLLABUS

Cellular Networks - (304192)

Credit	Examination Scheme :
03	In-Sem(Theory) : 30 Marks
	End-Sem(Theory) : 70 Marks

Unit I Introduction of Wireless Channel

Introduction, Free Space Propagation Model, Ground-Reflection Scenario, Hata Model and Receiver-Noise Computation. Channel Estimation techniques and Diversity in wireless communications. **(Chapter - 1)**

Unit II Orthogonal Frequency Division Multiplexing

Introduction, Motivation and Multicarrier basics, OFDM example, bit error rate for OFDM. **Multiple-Input Multiple-Output Wireless Communications :** Introduction to MIMO Wireless Communications, MIMO System Model and MIMO-OFDM. **(Chapter - 2)**

Unit III Introduction to Mobile Communication

Introduction to Cellular Service Progression, Cell Geometry, Overview of Cellular mobile and Network architecture, Cellular radio system design - Frequency assignments, frequency reuse channels, Concept of cell splitting and Cell sectoring. Significance of Handover in cellular systems with Handoff algorithms and roaming. **(Chapter - 3)**

Unit IV Wireless System Planning

Link-Budget Analysis, Tele-traffic Theory, Tele-traffic System Model and Steady State Analysis.
(Chapter - 4)

Unit V Wireless and Mobile Technologies and Protocols and their performance evaluation

Introduction, Wireless and mobile technologies, LTE- advanced, 5G - Architecture, wireless local area network and Simulations of wireless networks. **(Chapter - 5)**

Unit VI Performance Analysis Issues

Introduction to Network coding, basic hamming code and significance of Information Theory. Interference suppression and Power control. MAC layer scheduling and connection admission in mobile communication. **(Chapter - 6)**

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UNIT I

1 Introduction of Wireless Channel

Syllabus

Introduction, Free Space Propagation Model, Ground-Reflection Scenario, Hata Model and Receiver-Noise Computation. Channel Estimation techniques and Diversity in wireless communications.

Contents

1.1 Characteristics of Wireless Model

- The radio channel is different than wired channel. There exists extremely harsh environment compared to "wired" or guided media.
- The radio channel is time variant because of movement of people, switching off and on of interference, movement of mobile terminals, sensitivity to a variety of other factors, fading and multipath. Therefore, there is need of a framework that characterizes the radio channel.

Radio channel characterization

- The radio propagation is modeled as a random phenomenon. Transmission of radio signals are effected by :
  1. Ground terrain
  2. Atmosphere
  3. Objects
  4. Interference with other signals
  5. Distance (path loss)

1.2 Radio Propagation Mechanism

SPPU : April-16, 18, Dec.-17

- The radio propagation can be explained by three basic mechanisms :
  1. Reflection and transmission
  2. Diffraction
  3. Scattering.

1.2.1 Reflection and Transmission

- Reflection occurs when electromagnetic wave impinges on object larger than the wavelength λ. The electromagnetic wave bounces off the object. Examples : Walls, buildings, ground.
- The electromagnetic signal is attenuated by a reflection factor. Attenuation depends on -
  1. Nature of material
  2. Frequency of the carrier
  3. Angle of incidence
  4. Nature of the surface.
- Usually transmission through an object leads to larger losses (absorption) than reflection. Multiple reflections can result in a weak signal.

1.2.2 Diffraction

- Diffraction occurs when radio wave is incident upon the edge of a sharp object. Examples : Wall, roof edge, door.
- Each such object becomes a secondary source of emission. In this case, the losses are much larger than with reflection or transmission.
- Diffraction is important in micro-cells for non-line of sight transmission i.e. propagation into shadowed regions.
- Diffraction is not significant in indoor areas because of large losses in diffracted signal.

1.2.3 Scattering

- Scattering is caused by irregular objects comparable in size to the wavelength. These objects scatter rays in all directions. Each scatterer acts as a source resulting in -
  1. Signal propagates in all directions
  2. Large losses in signal strength
  3. Insignificant except when the transceiver is in very cluttered environments
- Examples of scatterers : Foliage, furniture, lampposts, vehicles.
- Fig. 1.2.1 and Fig. 1.2.2 illustrates all three mechanisms for outdoor and indoor applications.

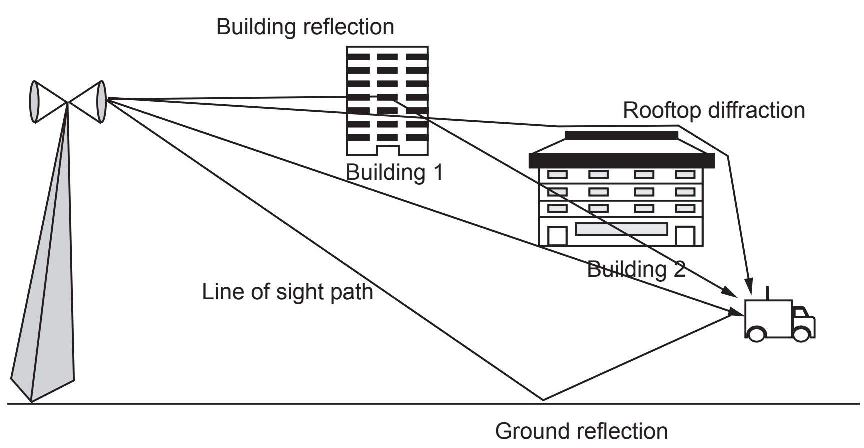


Fig. 1.2.1 Radio propagation mechanisms in an outdoor area

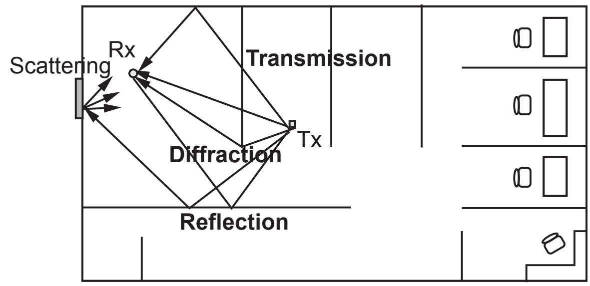


Fig. 1.2.2 Radio propagation mechanisms in an indoor area

University Questions

1. *With the help of neat diagram explain the three basic propagation mechanisms of signal in mobile communication system.* **SPPU : April-16, Marks 4**
2. *Discuss basic propagation mechanisms, reflection and diffraction in wireless communication.* **SPPU : Dec.-17, Marks 5**
3. *Explain : Reflection.* **SPPU : April-18, Marks 6**

1.3 Path - Loss Modeling and Signal Coverage

- Path - loss models are commonly used to estimate link budgets, cell sizes and shapes, capacity, handoff criteria etc.
- The path - loss models used to estimate macroscopic or large scale variation of Received Signal Strength (RSS).

Path-loss = Loss in signal strength as a function of distance

- Path loss is :
  1. Terrain dependent (urban, rural, mountainous), ground reflection, diffraction, etc.
  2. Site dependent (antenna heights for example)
  3. Frequency dependent
  4. Line of sight or not
- By path-loss models, radio engineers calculate the coverage area of wireless base stations and Access Points (APs) also the maximum distance between two terminals in ad-hoc networks.

1.3.1 Free Space Propagation

Path-loss Gradient

- In most environments, the radio signal strength falls as some power α of the distance called as power distance gradient or path-loss gradient.
- The signal strength is proportional to $P_t\, d^{-\alpha}$, where P_t is transmitted power and d is distance in meters.
- When an antenna radiates signal in all direction, the signal strength density at a sphere of radius d is the total radiated signal strength divided by the area of the sphere ($4\pi d^2$). Also, there are losses dependent on frequency.
- The relation between transmitted power (P_t) and received power (P_r) is given by :

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

where,

G_t = Transmitter antenna gain in the direction from transmitter to receiver

G_r = Receiver antenna gain

d = Distance between transmitter and receiver

λ = Wavelength of carrier

1.3.2 Two - Ray Model of Mobile Environment

- In free space, the signal travels from the transmitter to receiver along the single path but in realistic environment, the signal reaches the receiver through several different paths.
- The two-ray model is shown in Fig. 1.3.1.

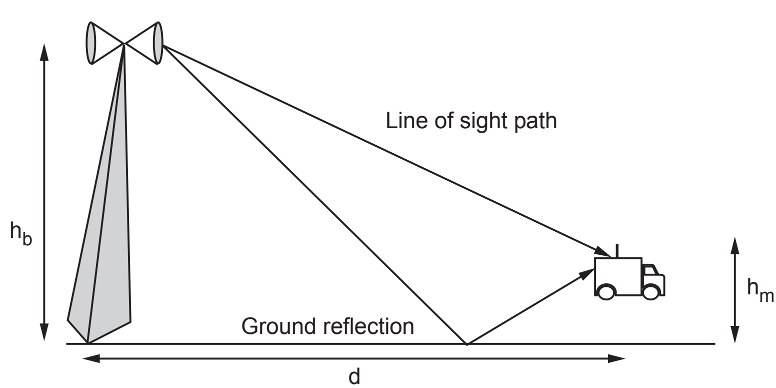


Fig. 1.3.1 Two-ray model

- The Line of Sight (LOS) component between base station and mobile terminal carries the signal similar to as in free space.
- Another path of signal is through the reflection off the earth's surface. These two paths travel different distances based on height of base station antenna (h_b) and height of mobile terminal antenna (h_m).
- At receiver these two signals and added constructively or destructively. The relation between transmit power for two-ray model can be approximated by expression :

$$P_r = P_t G_t G_r \frac{h_b^2 h_m^2}{d^4}$$

- It can be observed that the signal strength falls as the fourth power of distance (d) between the transmitter and receiver. In other words, there is a loss of 40 dB per decade or 12 dB per octave.

1.3.3 Distance Power Relationship and Shadow Fading

- The received signal power (P_r) is proportional to the distance between transmitter and receiver (d), raised to certain exponent α which is referred to as distance-power gradient.

$$P_r = P_0 \, d^{-\alpha}$$

where,

P_0 = Received power at reference distance from transmitter (usually one meter).

α = 2; for free space and

α = 4; for two-path model.

- The distance power relationship in decibels is given by :

$$10 \log (P_r) = 10 \log (P_0) - 10 \, \alpha \log (d)$$

- The last term on right hand side of equation shows the power loss in dBs with respect to received power at one meter.
- Path loss in dB at a distance of at a distance of one meter as :

$$L_0 = 10 \log_{10} (P_t) - 10 \log_{10} (P_0)$$

- The total path loss in dB is given by :

$$L_p = L_0 + 10 \, \alpha \log(d)$$

where,

L_0 is termed the frequency dependent component.

Parameter α is called the "path loss gradient" or exponent.

The value of α determines how quickly the RSS falls with distance.

1.3.4 Shadow Fading

- Shadowing occurs when line of site is blocked. The actual received signal strength will vary around its mean value. The shadow fading is also called as slow fading.
- The path loss equation can be modified to include this effect by adding a random component.

$$L_p = L_0 + 10\,\alpha \log_{10}(d) + X$$

where,

X is random signal with a distribution that depends on fading component.

1.3.5 Path - Loss Models for Megacellular

- The megacellular area spans over 100s of kilometres. The mega cellular areas are served mostly by LEO satellites.
- The path loss is usually modelled similar to free space but fading characteristics are different.

1.3.6 Path - Loss Models for Macrocellular Areas - Okumura - Hata Model

- Empirical formula calculating the median path-loss for a quasi smooth terrain in an urban area is -

$$L_p = 69.55 + 26.16 \log f_c - 13.82 \log h_b - a(h_m) + [44.9 - 6.55 \log h_b] \log d$$

where,

f_c in MHz : $150 < f_c < 1500$ MHz

h_b in meters - base station antenna height : $30 < h_b < 200$ m

h_m in meters - mobile antenna height : $1 < h_m < 10$ m

d in kilometers - distance : $1 < d < 20$ km

- The correction factor for the mobile antenna height is given by :

1. Small - medium city :

$$a(h_m) = (1.1 \log f_c - 0.7)\, h_m - (1.56 \log f_c - 0.8)$$

2. Large city :

$$a(h_m) = 8.29 (\log 1.54\, h_m)^2 - 1.1, \quad f_c \le 200 \text{ MHz}$$

$$a(h_m) = 3.2 (\log 11.75\, h_m)^2 - 4.97, \quad f_c \le 400 \text{ MHz}$$

3. For a suburban area :

$$L_p = L_p(\text{Urban}) - \left[2\log\left[\frac{f_c}{28}\right]^2 - 5.4\right]$$

For an open area :

$$L_p = L_p(\text{Urban}) - 4.78\ (\log f_c)^2 + 18.33 \log f_c - 40.94$$

1.3.7 Path - Loss Models for Microcellular Areas

- The microcellular area spans from 100 meters to few of kilometres. These are usually supported by base station height which is approximately equals to roof tops or lampposts.
- The propagation characteristics are quite complex with the propagation of signals affected by reflection from buildings, grounds and scattering from vehicles.

1.3.8 Path - Loss Models for Picocellular Indoor Areas

- The picocells are radio cells covering buildings or parts of buildings. The picocell spanning between 30 and 100 m.
- The applications of picocells are : WLANs, wireless PBX, PCS

1.3.8.1 Multifloor Attenuation Model

- Multifloor attenuation model is expressed as -

$$L_p = L_0 + nF + 10 \log(d)$$

where,

F is signal attenuation per floor.

L_0 is the path-loss at first meter.

d is distance in meters.

n is number of floors.

- Typical values for F = 10 dB and 16 dB for measurements at 900 MHz and 1.7 GHz, respectively.
- Furniture objects cause shadowing approximately of 4 dB.

1.3.9 Path - Loss Models for Femtocellular Area

- The femtocellular area span is between 2 and 10s of meters.
- The femtocells exist in individual residences using low power devices for applications : bluetooth, home RF. Since femtocells are usually deployed in

residential areas, JTC model may apply to predict the coverage of femtocell at 1.8 GHz. For operation at 2.4 GHz and 5 GHz (unlicensed bands) -

$$L_p = L_0 + 10\,\alpha\,\log(d)$$

where,

L_0 is the path - loss at first meter.

d is distance in meters.

α is the path - loss exponent.

- Selected measurements of indoor path loss models are shown in Table 1.3.1.

f_c (GHz)	Environment	Scenario	Path Loss at d = 1 m (dB)	Path loss gradient α
2.4	Indoor office	LOS	41.5	1.9
		NLOS	37.7	3.3
5.1	Meeting room	LOS	46.6	2.22
		NLOS	61.6	2.22
5.2	Suburban residences	LOS and same floor	47	2 to 3
		NLOS and same floor		4 to 5
		NLOS and room in the higher floor directly above Tx		4 to 6
		NLOS and room in the higher floor not directly above the Tx		6 to 7

Table 1.3.1 Path-loss models for femtocells at 2.4 GHz, 5.1 GHz and 5.2 GHz

1.4 Effects of Multipath and Doppler

SPPU : April-17

- Radio waves arrive at the receiver from different directions with different delays. At the receiver antenna they combine via vector addition.
- Received signal level varies (10s of dBs) due to short-term (rapid) variations and long-term (slow) variations.

- Rapid fluctuation is caused by :
  1. Movement of mobile terminal toward or away from the base station transmitter is called as Doppler.
  2. Addition of signals arriving via different paths called as multipath fading.

1.4.1 Doppler Spectrum

- Doppler spectrum is the spectrum of fluctuations of received signal strength. Doppler spectrum is used for designing coding schemes and interleaving sizes to achieve efficient performance.
- The Doppler spectrum for a Rayleigh fading channel is modeled by -

$$D(\lambda) = \frac{1}{2\pi f_m} \times [1 - (\lambda / f_m)^2]^{-1/2} \qquad \text{for} \quad -f_m \leq \lambda \leq f_m$$

where, f_m is maximum possible Doppler frequency

$$f_m = v_m / \lambda; \ v_m \text{ is the mobile velocity}$$

- This spectrum is commonly used in mobile radio modeling and is known as classical Doppler spectrum as shown in Fig. 1.4.1.

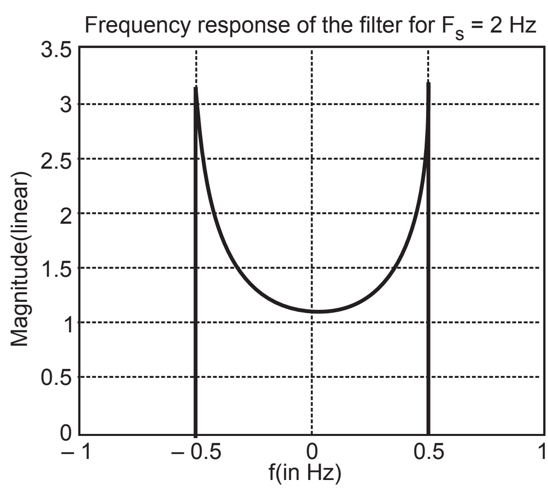


Fig. 1.4.1 Classical doppler spectrum

1.4.2 Channel Measurement and Modeling Techniques

- Practically it is very difficult to derive the expressions for propagation of radio signal through a complex medium. A general modelling procedure is shown in Fig. 1.4.2.
- All narrowband or wideband measurements are done on actual site where wireless network is setup. The measurement results are used via simulations of the

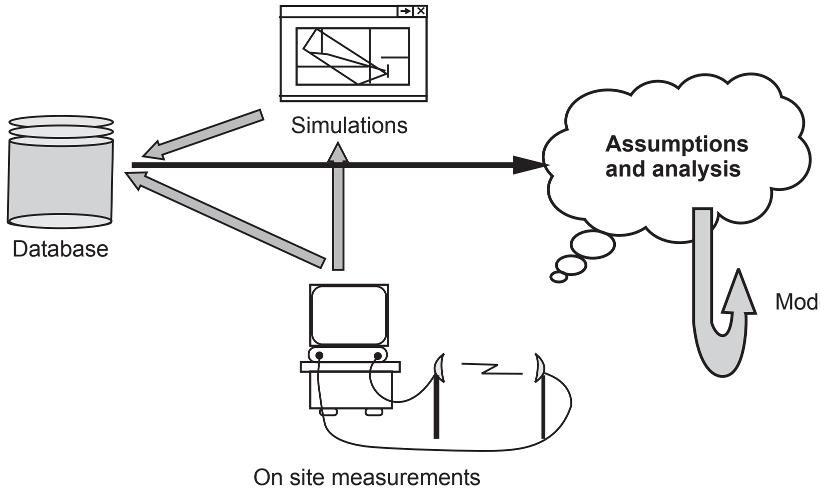


Fig. 1.4.2 General procedure for modeling the radio channel

environment. All the data are included in data base. This data along with realistic assumptions used to construct empirical models. Path loss models are examples of such empirical models.

1.4.3 Impulse Response Model of Multipath Channel

- A mobile radio channel is modelled as a linear filter with a time varying impulse response, where the time variation is due to receiver motion in space. The impulse response is a useful characterization of the channel as it is used to predict and compare the performance of many different mobile communication systems and transmission bandwidths for a particular mobile channel condition.
- Consider the case where time variation is due to receiver motion in space. This is shown in Fig. 1.4.3. In the given figure the receiver moves along the ground at some constant velocity ?. For a foxed position d, the channel between the transmitter and the receiver can be modelled as a linear time invariant system.

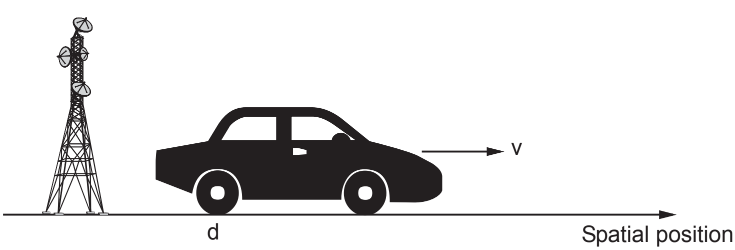


Fig. 1.4.3 The mobile radio channel as a function of time and space

- Due to the different multipath waves which have propagation delays which vary over different spatial locations of the receiver, the impulse response of the linear time invariant channel should be a function of the receiver. That is the channel impulse response can be expressed as h(d, t). Let x(t) represents transmitted signal then the received signal y(d, t) at distance d can be expressed as convolution of x(t) and h(d, t).

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

For a causal system, h (d, t) = 0 for t < 0, thus equation A reduces to,

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

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

Since the receiver moves along the ground at constant velocity, the position of the receiver can be expressed as,

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

$$= x(t) \otimes h(d, t)$$

It is clear that the mobile radio channel can be modeled as a linear time varying channel where the channel changing with time and distance.

- v can be assumed constant over a short time or distance interval
- x(t) represent the transmitted bandpass waveform
- y(t) the received signal waveform
- $h(t,\tau)$ the impulse response of the time varying multipath radio channel
- t represents the time variations due to motion
- τrepresents the channel multipath delay for a fixed value of t

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

- If the multipath channel is assumed to be a band limited bandpass channel, then $h(t,\tau)$ may be equivalently described by a complex baseband impulse response $h_b(t, \tau)$ with the input and output being the complex envelope representations of the transmitted and received signals respectively

Bandpass channel impulse response model

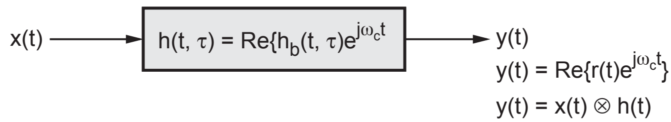


Fig. 1.4.4 Bandpass channel Impulse response model

Baseband equivalent channel impulse response

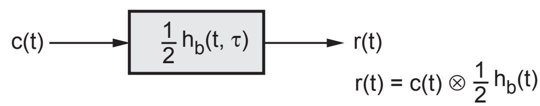


Fig. 1.4.5 Baseband equivalent channel Impulse response model

- The factor 1/2 is due to the properties of the complex envelope in order to represent the passband radio system at baseband.
- The low pass characterization removes the high frequency variations caused by the carrier.
- The average power of a bandpass signal $x^2(t)$ is equal to $0.5|c^2(t)|$.

University Question

1. *Explain impulse response model of a multipath channel.* **SPPU : April-17, Marks 4**

1.5 Fading

SPPU : Dec.-16

- Fading is defined as variation in intensity of received radio signal due to :
  - Variation in propagation time
  - Relative phase differences
  - Change in frequencies
  - Change in characteristics of propagation path with time.
- Fading refers to time variation of received signal power caused by changes in transmission medium fading describes the rapid fluctuation of amplitudes, phases or multipath delays of radio signal over a short period of time or travel distance. The strength of received signal varies with respect to time.
- Various propagation mechanisms caused fading of radio signals such as : refraction, reflection, diffraction, scattering, attenuation and ducting of radio waves. These propagation mechanisms determine amplitude, phase, polarization and frequency of fading.

- Fading is caused by certain terrain geometry, attenuation, changes in transmission medium, refraction, multipath propagation, rainfall, obstacles etc.

1.5.1 Types of Fading

- Two types of fading exists in radio signals.
  1. Large scale / Long term fading / Slow fading
  2. Small scale / Short term fading / Fast fading

1.5.2 Large Scale Fading

- When fading duration is very long and signal attenuation is large it is called as **large scale fading**. The large scale fading is due to several factors such as atmosphere, large pathloss, shadowing, trees and foliage. Such kind of fading is usual low observed in rural areas. This slowly changing fading is referred as **slow fading**.
- Large scale fading can be compensated by increasing transmitter power so that received signal can be within certain limits.

1.5.3 Small Scale Fading

- When fading duration is very small i.e. signal strength varies very fast for short distance, it is called as **small scale fading**. Small scan fading mainly because of multipath propagation, speed of surrounding obstacles, transmission bandwidth of signal and doppler shift.
- The changes of amplitude is about 20 or 30 dB over a short distance. This rapidly changing fading phenomenon is referred as **fast fading** :

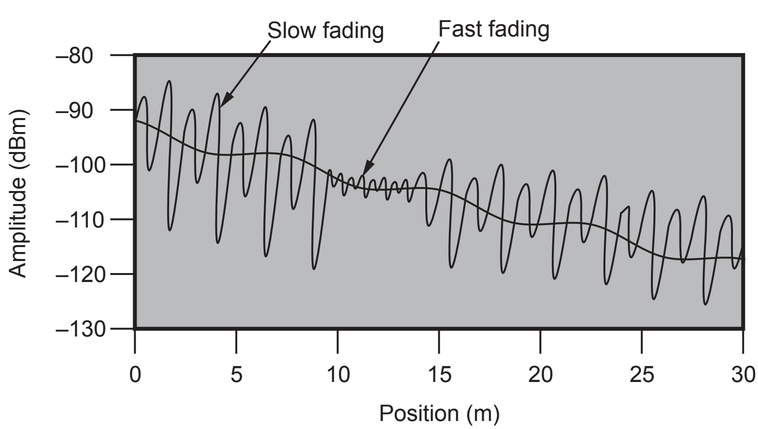


Fig. 1.5.1 Slow and fast fading in an urban mobile environment

1.5.4 Flat Fading

- When fading in all frequency components of received signal fluctuate in the same proportions simultaneously it is called as **flat fading**. It is also referred as non-selective fading.

1.5.5 Selective Fading

- When different spectral components of received radio signal fluctuates unequally it is called as **selective fading**.
- Selective fading is relative to the bandwidth of entire communical channel. Non-selective fading implies that the signal bandwidth of interest is narrower than the entire spectrum affected by fading.

University Question

1. *Explain the factors influencing small scale fading.* **SPPU : Dec.-16, Marks 6**

1.6 Channel Estimation Modeling and Multipath Fading

- For designing a communication system, the effects of multipath fading and noise on the communication channel must be estimated. The multipath fading channel can be divided on the basis of distribution function of the instantaneous power of the channel on the radio environment. Different types of fading multipath channel are as follows :
  1. Additive White Gaussian Noise (AWGN) channel
  2. Log normal fading channel.
  3. Rayleigh fading channel
  4. Rician fading channel

1.6.1 AWGN Channel

- In AWGN channels, the desired signal is degraded by the thermal noise associated with the channel itself and losses in transmitter and receiver circuitry.
- AWGN channel is accurate in specific cases such as space communication and co-axial wire communication. It is not suitable for mobile communication.

1.6.2 Log Normal Fading Channel

- The propagation models a developed to determine the path loss are known as large scale fading models as they all characterizes the received signal power by averaging it over the large distance. But due to trees, foliage, rainfall, and atmospheric condition there is gradual change in local mean power and such type of fading channel is characterized by log normal distribution function.

1.6.3 Rayleigh Fading Channel

- Rayleigh fading exists when there is multiple indirect paths between transmitter and receiver i.e. there is no distinct Line Of Sight (LOS) path.
- Rayleigh fading can be dealt by studying performance characteristics in such critical environment.

1.6.4 Ricing Fading Channel

- Rician fading exists where there is a direct Line Of Sight (LOS) path along with number of indirect multipath signals. This model is mostly applicable in indoor environment whereas Rayleigh fading model characterizes outdoor environment.
- All the channels are characterized by a parameter 'K' where -

$$K = \frac{\text{Power in dominant path}}{\text{Power in scattered paths}}$$

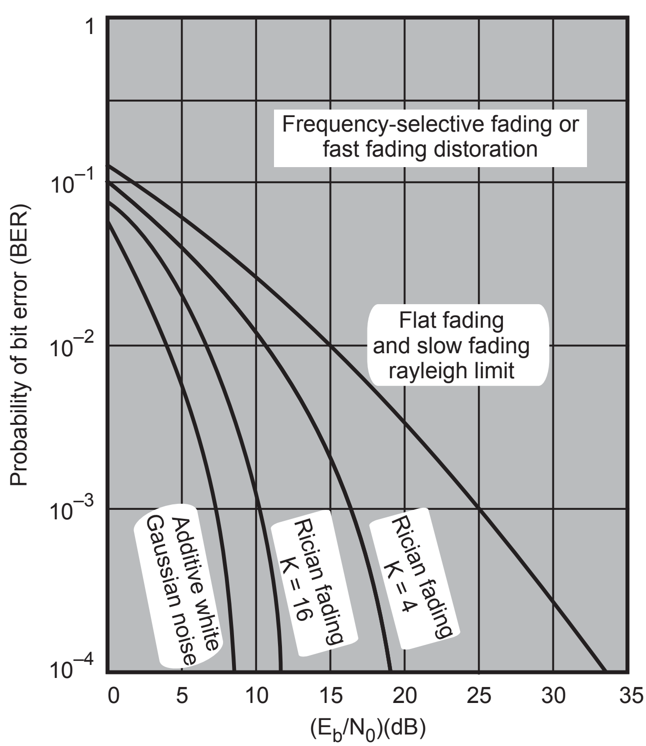


Fig. 1.6.1 Theoretical bit error rate for various fading conditions

i) For Rayleigh channel, numerator is zero therefore $K = 0$

ii) For AWGN channel, denometor is zero therefore $K = \infty$

iii) For Rician channel, with reasonably strong signal $K = 4$ to 16

Bit Error Rate (BER) for various fading conditions is shown in Fig. 1.6.1.

1.7 Diversity in Wireless Communications

- Diversity technique is the most effective technique that can be used to nullify the effect of multipath fading.
- Diversity technique employ some form of time space or frequency diversity either or both the transmission and reception of the desired signal.
- The basic theory behind the use of diversity in a wireless system is as follows : Any fading of the transmitted signal that occurs will not remain the same over time nor will it be the same over different signal paths or be the same for different frequencies. Therefore, if some form of time, space or frequency diversity is used, the effects of signal fading can be mitigated.
- Diversity technique is a method for improving the reliability of a message signal by utilizing two or more communication channels with different characteristics.
- Diversity plays an important role in combating fading, co-channel interference, avoiding error bursts and it may exploit the multipath propagation resulting in a diversity gain.
- The diversity techniques, provide two or more number of inputs at the mobile reception end such that fading among these two inputs are not correlated.

Types of diversity

- There are several different kinds of diversity which are commonly employed in wireless communication systems as follows.

1. Time diversity

- In the time diversity method, the information is transmitted repeatedly at specific time spacings that would exceed the coherence time of the mobile channel, and this will lead to repetition of signals for several times; irrespective of fading conditions.
- When an identical information is sent for different time slots, it is possible to obtain diversity branch signals.
- The time diversity technique is well suited for spread spectrum CDMA system, in which, RAKE receiver is used for reception.

2. Frequency diversity

- Fading is a major problem and in order to reduce it, diversity is being used.
- Diversity techniques are used in wireless communications to mitigate the effect of fading over a radio channel.
- Frequency diversity utilizes transmission of the same signal at two different, spaced, frequency carriers achieving two independently fading versions of a signal.
- Frequency diversity is a costly mechanism to use because of the difficulties to generate several transmitted signals and the combining signals received at several different frequencies simultaneously.
- There are two types of frequency diversity :

  a) Frequency hop spread - spectrum

  b) Direct sequence spread - spectrum

3. Space diversity

- In space diversity technique two antennas are separated by a distance 'd' so as to get the two input signal with low correlation among fading effects.
- The antenna would be at a height of 'h' from ground level at the cell site. As the distance 'd' between antennas varies with change in antenna height for both cell site and mobile antenna.

1.7.1 Comparison of Macroscopic Diversity and Microscopic Diversity Technique

Sr. No.	Macroscopic diversity	Microscopic diversity
1.	Prevents large scale fading.	Prevents small scale fading.
2.	Large Scale fading is caused by shadowing due to variation in both the terrain profile and the nature of the surroundings.	Small Scale fading is caused by multiple reflections from the surroundings.
3.	Large scale fading is log normally distributed signal.	It is characterized by deep and rapid amplitude fluctuations which occur as the mobile moves over distances of a few wavelengths.
4.	This fading is prevented by selecting an antenna which is not shadowed when others are, this allows increase in the signal-to-noise ratio.	This fading is prevented by selecting an antenna which gives a strong signal that mitigates this small signal fading effect.

1.8 Short Answered Questions

Q.1 What are the propagation mechanisms of EM waves ?

Ans. : The four propagation mechanisms of EM waves are

i. Free space propagation

ii. Reflection

iii. Diffraction

iv. Scattering

Q.2 What is the significance of propagation model ?

Ans.: The major significance of propagation model are:

i. Propagation model predicts the parameter of receiver.

ii. It predicts the average received signal strength at a given distance from the transmitter.

Q.3 What do you mean by small scale fading ? What are the factors influencing small scale fading ?

Ans. : Rapid fluctuations of the amplitude, phase as multipath delays of a radio signal over a short period of time is called small scale fading.

- The factors which influence small scale fading are : Multipath propagation, Speed of the mobile, Speed of surrounding objects and the transmission bandwidth of the signal.

Q.4 When does large scale propagation occur ?

Ans. : Large scale propagation occurs due to general terrain and the density and height of buildings and vegetation, large scale propagation occurs.

Q.5 Differentiate the propagation effects with mobile radio.

Ans. :

Sr. No.	Slow fading	Fast fading
1.	Show variance in the signal strength.	Rapiad variation in the signal strength.
2.	Mobile Station (MS) moves slowly.	Local objectes reflect the signal causes fast fading.
3.	It occurs when the large reflectors and diffracting along the tranmission paths are distant from the terminal. Eg. Rayleigh fading, Rician fading and Doppler shift.	It occurs when the user terminal (MS) moves for short distances.

Q.6 Define Doppler shift.

Ans. : If the receiver is moving towards the source, then the zero crossings of the signal appear faster and the received frequency is higher. The opposite effect occurs if the receiver is moving away from the source. The resulting chance in frequency is known as the Doppler shift (f_D).

$$F_D = f_r - f_0 = - f_0 \, V/C$$

Where, f_0 - Transmission frequency

f_r - Received frequency

Q.7 Differentiate time selective and frequency selective channel.

Ans. : The gain and the signal strength of the received signal are time varying means then the channel is described as time selective channel.

- The frequency response of the time selective channel is constant so that frequency flat channel.
- The channel is time invariant but the impulse response of the channel show a frequency - dependent response so called frequency selective channel.

Q.8 Define coherence time and coherence bandwidth.

Ans. : Coherence time is the maximum duration for which the channel can be assumed to be approximately constant. It is the time separation of the two time domain samples.

- Coherence bandwidth is the frequency separation of the two frequency domain samples.

Q.9 What is free space propagation model ?

Ans. : The free space propagation model is used to predict received signal strength, when unobstructed line - of - sight path between transmitter and receiver.

- Friis free space equation is given by,

$$P_{RX(d)} = P_{TX} G_{TX} G_{RX} \left(\frac{\lambda}{4 \pi d} \right)^2$$

- The factor $(\lambda / 4\lambda d)^2$ is also known as the free space loss factor.

Q.10 Define EIRP.

Ans. : EIRP (Equivalent Isotropically Radiated Power) of a transmitting system in a given direction is defined as the transmitter power that would be needed, with an isotropic radiator, to produce the same power density in the given direction.

$$EIRP = P_t G_t$$

Where, P_t - Transmitted power in W

G_t - Transmitting antenna gain

Q.11 What is path loss ?

Ans. : The path loss is defined as the difference (in dB) between the effective transmitted power and the received power. Path loss may or may not include the effect of the antenna gains.

Path - loss = Loss in signal strength as a function of distance.

Q.12 Define indoor propagation models.

Ans. : The indoor propagation models are used to characterizing radio propagation inside the buildings.

- The distances covered are much smaller and the variability of the environment is much greater for smaller range of transmitter and receiver separation distances.
- Features such as lay - out of the building, the construction materials and the building type strongly influence the propagation within the building.
- Some of the indoor propagation models are :
  i. Long - distance path loss model.
  ii. Ericession multiple break point model.
  iii. Attenuation factor model.

Q.13 What is the necessity of link budget ?

Ans. : The necessities of link budget are :

i. A link budget is the clearest and most intuitive way of computing the required transmitter power. It tabulates all equations that connect the Transmitter power to the received SNR.

ii. It is reliable for communications.

iii. It is used to ensure the sufficient receiver power is available.

iv. To meet the SNR requirement link budget is calculated.

Q.14 Define diffraction.

Ans. : Diffraction is the spreading of a beam through gaps and around corners. The maximum effect occurs when the gap has a similar size to the wavelength.

- Radio waves of about 5 m are diffracted by large buildings. Radio waves of 1 km are be diffracted around hills and through valleys, so they are able to reach most areas and are suitable for broadcasting.
- Microwave beams of a few centimetres do not spread round corners or around hills. This is why the transmitters and receivers must be in line of sight. When microwaves are transmitted from a satellite dish the wavelength must be small compared to the dish diameter to reduce diffraction. This

means that, compared to radio waves, microwaves can be sent as a thin beam.

Q.15 Classify the various types of fading.

Ans. :

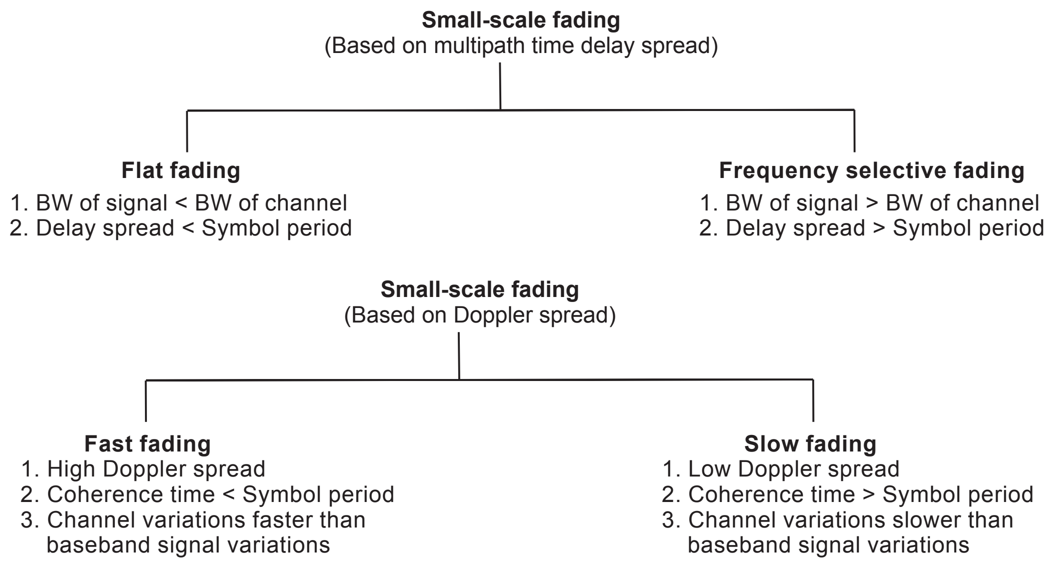


Fig. 1.8.1

Q.16 What is diversity technique ?

Ans. : **Diversity** is the technique used to compensate for fading channel impairments.

- Diversity is implemented by using two or more receiving antennas. While equalization is used to counter the effects of ISI, diversity is usually employed to reduce the depth and duration of the fades experienced by a receiver in a flat fading channel.
- These techniques can be employed at both base station and mobile receivers. Spatial diversity is the most widely used diversity technique.

Q.17 What is spatial diversity technique ?

Ans. : Spatial diversity :

- In spatial diversity technique multiple antennas are strategically spaced and connected to common receiving system. While one antenna sees a signal null, one of the other antennas may see a signal peak and the receiver is able to select the antenna with the best signal at any time.
- The CDMA systems use rake receivers which provide improvement through time diversity.

1.9 Multiple Choice Questions

Q.1 In cellular system, if cluster size N is greater than 7 then spectrum efficiency _____.

a reduces
b increases
c remains same
d none of the above

Q.2 The use of omnidirectional antenna ______ cochannel and long distance Interferences.

a reduces
b increases
c remains same
d none of the above

Q.3 The antenna coverage can be controlled by using multiple antennas and _______of power pattern.

a synthesis
b increasing
c complexing
d lowering

Q.4 The process of allocating setup channels within the available frequency spectrum is called _____.

a frequency management
b channel assignment
c handoff
d cell splitting

Q.5 The process of allocating specific channels to cell sites and mobile units is called _______.

a frequency management
b channel assignment
c handoff
d cell splitting

Q.6 As the subscriber moves between cells, the communication with base station departing cell ceases and communication with base station of the entering cell commences. This process is known as _____.

a frequency management
b channel assignment
c handoff
d cell splitting

Answer Keys for Multiple Choice Questions :

Q.1	a	**Q.2**	a	**Q.3**	a	**Q.4**	a
Q.5	b	**Q.6**	c				

❑❑❑

Notes

UNIT II

2 Orthogonal Frequency Division Multiplexing

Syllabus

Introduction, Motivation and Multicarrier basics, OFDM example, bit error rate for OFDM. ***Multiple-Input Multiple-Output Wireless Communications :*** *Introduction to MIMO Wireless Communications, MIMO System Model and MIMO-OFDM.*

Contents

2.1 Multicarrier Basics

- Multi Carrier Modulation (MCM) is a method of sending data by splitting it into several parallel sub-streams and sends each sub stream on a different frequency known as a subcarrier.
- The inverse fast fourier transform itself acts as a multi carrier modulator and the fast fourier transform serves as a multi carrier demodulator.
- OFDM, also known as multicarrier transmission or modulation, uses multiple carrier signals at different frequencies, sending some of the bits on each channel.
- OFDM is the basic multi carrier modulation technique for both wireless and cellular communications.
- OFDM is a perfect choice for point-to-point communication, which offers minimum complexity and achieves very high bandwidth.

2.2 Orthogonal Frequency Division Multiplexing (OFDM)

- OFDM is a Multi Carrier Modulation (MCM) scheme, in which many parallel data streams are transmitted at the same time over a channel, with each transmitting only a small part of the total data rate. Similar to FDM.
- However, in the case of OFDM, all the sub-channels are dedicated to a single source.
- With OFDM, a high-speed digital message is divided into a large number of separate carrier waves.
- OFDM was chosen as a modulation scheme to support high speed packet data transfer.
- OFDM is a form of multi carrier, multi symbol, multirate FDM in which the user gets to use all the FDM channels together. It possesses the property of orthogonality.
- An OFDM modulation system uses several to many carriers that are all transmitted simultaneously. Each carrier transmits a sub-symbol that may encode one to many bits of data. The entire transmitted symbol consists of the sum of all the sub-symbols.
- OFDM is the modulation scheme for the IEEE 802.11 LAN wireless standard.

Illustrative Example

Example 2.2.1 *If an OFDM system transmits 32 kbps over each carrier and uses 16 carriers, what is the overall data rate ?*

Solution : The overall data rate for an OFDM system = 32 kbps × 16

= 512 kbps **Ans.**

2.3 OFDM Concept

- OFDM is a multi-carrier transmission scheme. OFDM transform high-speed serial transmission to low-speed parallel transmission. It increases symbol duration, robust to multipath interference.
- Multiplexing is an important signal processing operation in which a number of signals are combined and transmitted parallelly over a common channel.
- In order to avoid interference during parallel transmission, the signals can be separated in frequency and then the resulting technique is called Frequency Division Multiplexing (FDM).
- In FDM, the adjacent bands are non overlapping but if overlap is allowed by transmitting signals that are mutually orthogonal (that is, there is a precise mathematical relationship between the frequencies of the transmitted signals) such that one signal has zero effect on another, then the resulting transmission technique is known as **Orthogonal Frequency Division Multiplexing (OFDM).**
- OFDM is a technique of transmitting high bit rate data into several parallel streams of low bit rate data. At any instant, the data transmitted simultaneously in each of these parallel data streams is frequency modulated by carriers (called subcarriers) which are orthogonal to each other.
- Fig. 2.3.1 shows the basic concept of OFDM.

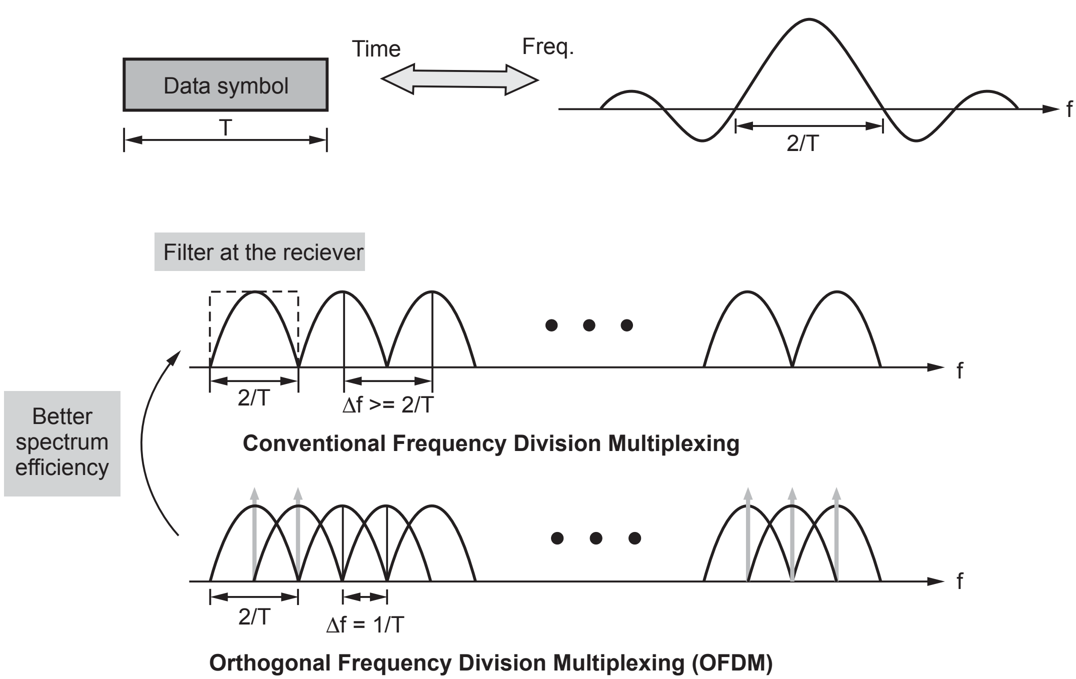


Fig. 2.3.1 OFDM concept

- For high data rate communication the bandwidth (which is limited) requirement goes on increasing as the data rate increases or the symbol duration decreases.
- Thus in OFDM, instead of sending a particular number of symbols, say P, in T seconds serially, the P symbols can be sent in parallel with symbol duration now increased to T seconds instead of T/P seconds as was previously.
- This offers many advantages in digital data transmission through a wireless time varying channel. The primary advantage of increasing the symbol duration is that the channel experiences fading instead of frequency selective fading since it is ensured that in the time domain the symbol duration is greater than the r.m.s. delay spread of the channel. Viewed in the frequency domain this implies that than the bandwidth of the OFDM signal is less than coherent bandwidth of the channel.

Orthogonality of signals

- Orthogonal signals can be viewed in the same perspective as we view vectors which are perpendicular/orthogonal to each other. The inner product of two mutually orthogonal vectors is equal to zero. Similarly the inner product of two orthogonal signals is also equal to zero.

2.4 OFDM Transmitter and Receiver

- The OFDM transmitter and receiver block diagram is shown in Fig. 2.4.1.

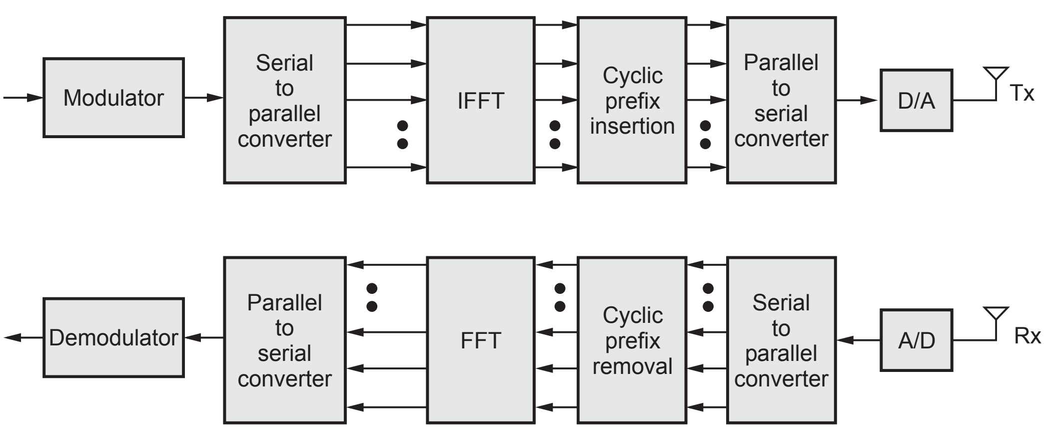


Fig. 2.4.1

Advantages of OFDM

1. Easy to mitigate the adverse effects of channel dispersion by the use of cyclic prefixes.
2. Low-complexity implementation based on FFT/IFFT.
3. Support high-rate transmission at a low implementation cost.

Disadvantages of OFDM

1. High peak-to-average power ratio, so that *highly linear* power amplifiers are required at the transmitters in order to avoid intermodulation interference.
2. The use of cyclic prefixes reduces transmission efficiency. Some power is wasted by transmitting cyclic prefixes, which are redundant.

2.5 MIMO

- Systems with Multiple Element Antennas (MEAs) at both link ends are called MIMO systems. The MEAs of a MIMO system can be used for four different purposes :

1. Beamforming,
2. Diversity,
3. Interference suppression and
4. Spatial multiplexing (transmission of several data streams in parallel).

- The first three concepts are the same as for smart antennas. Spatial multiplexing allows direct improvement of capacity by simultaneous transmission of multiple data streams.

2.5.1 Spatial Multiplexing

- Spatial multiplexing uses Multiple Element Antennas (MEAs) at the transmitter (TX) for transmission of parallel data streams as shown in Fig. 2.5.1.

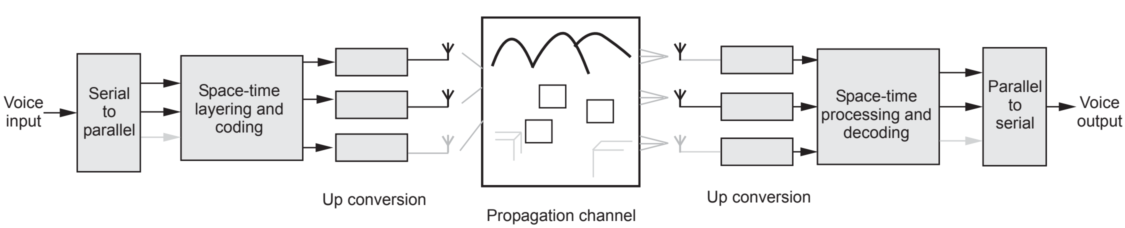


Fig. 2.5.1 Spatial multiplexing principle

- An original high-rate data stream is multiplexed into several parallel streams, each of which is sent from one transmit antenna element. The channel "mixes up" these data streams, so that each of the receive antenna elements sees a combination of them.
- If the channel is well behaved, the received signals represent linearly independent combinations. In this case, appropriate signal processing at the receiver (RX) can separate the data streams.

- A basic condition is that the number of receive antenna elements is at least as large as the number of transmit data streams. It is clear that this approach allows the data rate to be drastically increased - namely, by a factor of min (N_t, N_r).

2.5.2 System Model

- Block diagram of a Multiple-Input Multiple-Output (MIMO) system is shown in Fig. 2.5.2.

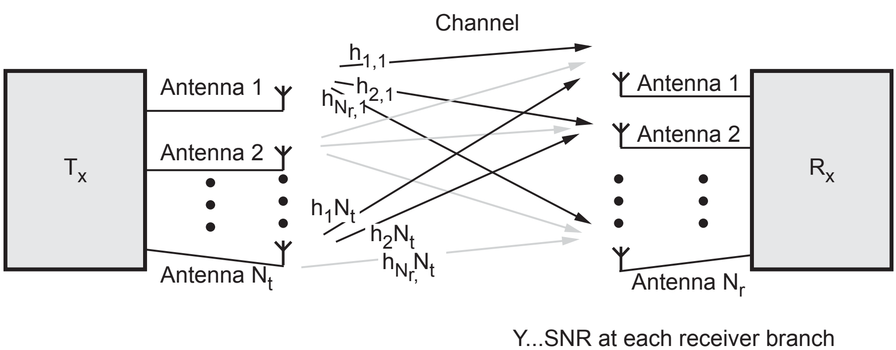


Fig. 2.5.2 MIMO block diagram

At the transmitter (TX), the data stream enters an encoder, whose outputs are forwarded to N_t transmit antennas. From the antennas, the signal is sent through the wireless propagation channel, which is assumed to be quasi-static and frequency-flat if not stated otherwise. The quasi-static means that the coherence time of the channel is so long that a large number of bits can be transmitted within this time.

The channel matrix is expressed as :

$$H = \begin{pmatrix} h_{11} & h_{12} & \dots & h_{1N_t} \\ h_{21} & h_{22} & \dots & h_{2N_t} \\ \vdots & \vdots & \dots & \vdots \\ h_{N_r 1} & h_{N_r 2} & \dots & h_{N_t N_t} \end{pmatrix}$$

Where,

hij are complex channel gains (transfer functions) from the j^{th} transmit to the i^{th} receive antenna.

- The received signal vector :

$$\mathbf{r} = \mathbf{Hs} + \mathbf{n} = \mathbf{x} + \mathbf{n}$$

It contains the signals received by N_r antenna elements, where **s** is the transmit signal vector and **n** is the noise vector.

2.5.3 Capacity in Non-Fading Channels

- To understand MIMO systems, the derivation of the capacity equation for MIMO systems in non-fading channels is necessary. This capacity equation is often known as Foschini's equation.
- For a capacity equation for normal (single-antenna) Additive White Gaussian Noise (AWGN) channels, Shannon showed, the information-theoretic (ergodic) capacity of such a channel is expressed as -

$$C_{Shannon} = \log_2(1+\gamma \cdot |H|^2)$$

Where,

γ is the SNR at the RX,

H is the normalized transfer function from the TX to the RX.

- The capacity of channel **H** is thus given by the sum of the capacities of the eigen modes of the channel :

$$C = \sum_{k=1}^{R_H} \log_2 \left[1+\frac{P_k}{\sigma_n^2}\sigma_k^2\right]$$

Where,

$\sigma^2 n$ is noise variance,

P_k is the power allocated to the k^{th} eigen mode.

- The equivalent capacity expression can be expressed as -

$$C = \log_2 \left[\det\left(I_{Nr} + \frac{\bar{\gamma}}{N_t} HR_{ss}H^*\right)\right]$$

Where,

$\mathbf{I_{Nr}}$ is the $N_r \times N_r$ identity matrix,

$\bar{\gamma}$ is the mean SNR per RX branch,

$\mathbf{R_{ss}}$ is the correlation matrix of the transmit data (if data at the different antenna elements are uncorrelated, it is a diagonal matrix with entries that describe the power distribution among antennas).

No Channel State Information at the Transmitter and Full CSI at the Receiver

- When the RX knows the channel perfectly, but no CSI is available at the TX, it is optimum to assign equal transmit power to all TX antennas, $P_k = P/N_t$ and use uncorrelated data streams. Capacity thus takes on the form :

$$C = \log_2 \left[\det\left(I_{Nr} + \frac{\bar{\gamma}}{N_t} HH^*\right)\right]$$

2.5.4 Layered Space-Time Structure

- There are information-theoretic limits of MIMO systems. In order to realize these capacities in practice, a possibility is joint encoding of the data streams that are to be transmitted from different antenna elements, combined with maximum likelihood detection.
- When this technique is combined with capacity-achieving codes, it can closely approximate the capacity of a MIMO system. For a small number of antenna elements and for a small modulation alphabet (BPSK or Quadrature-Phase Shift Keying (QPSK)), such a scheme can actually be employed. But, in most practical cases, the complexity of a joint MLSE is high. For this reason, space-time structures have been proposed.
- The space-time structure allows breaking up the demodulation process into several separate pieces with fewer complexities. These structures are also widely known under the name of **BLAST architectures.**

2.5.4.1 Horizontal BLAST

- H-BLAST is the simplest possible layered space-time structure. The transmitter (TX) first demultiplexes the data stream into N_t parallel streams, each of which is encoded separately.
- Each encoded data stream is then transmitted from a different transmit antenna. The channel mixes up the different data streams; the receiver (RX) separates them out by successive nulling and interference subtraction.
- Fig. 2.5.3 shows block diagram of a horizontal BLAST transceiver.
  (See Fig. 2.5.3 on next page.)

2.5.4.2 Diagonal BLAST

- The main drawback of H-BLAST is that it does not provide diversity. The first stream, which has diversity order 1, dominates the performance at high SNRs. A better performance can be achieved with the so-called D-BLAST scheme.
- In this D-BLAST, streams are cycled through the different transmit antennas, such that each stream sees all possible antenna elements i.e. each single transmit stream is subdivided into a number of subblocks. The first subblock of stream 1 is transmitted from antenna 1, the next subblock from antenna 2 and so on.
- Fig. 2.5.4 shows assignment of bit-streams to different antennas for horizontal BLAST and diversity BLAST. (See Fig. 2.5.4 on page no. 2 - 10)
- Decoding can be done stream by stream. Each decoded block can be subtracted from signals at the other antenna elements and therefore enhances the quality of the residual signal.

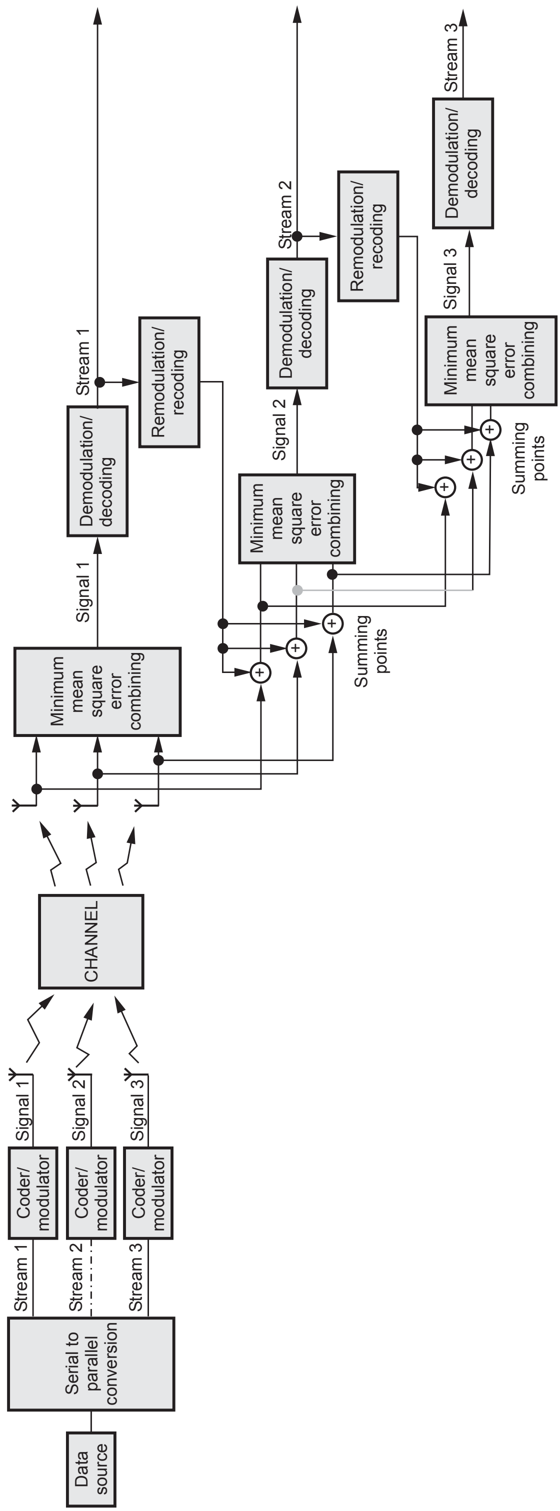


Fig. 2.5.3 Horizontal BLAST transceiver

Fig. 2.5.4 Bit-streams assignment to different antennas for horizontal BLAST and diversity BLAST

- The difference from H-BLAST is that each stream is sometimes in a "good" position in the sense that the other streams have already been subtracted and thus the SINR is very high.

2.5.5 Diversity

- Multiple antennas can also be used to provide pure diversity. There is distinction between systems that have Channel-State Information Transmitter (CSIT) and systems that do not. The former can achieve beamforming gain in addition to diversity gain, while the latter is restricted to achieving better resistance to fading.
- Two variations of diversity are :

1. Diversity with Channel State Information at the Transmitter.
2. Diversity Without Channel State Information at the Transmitter - Space-Time Coding.

2.5.6 Tradeoffs between Diversity, Beamforming Gain and Spatial Multiplexing

- MIMO systems can be used to achieve spatial multiplexing, diversity and/or beamforming. However, it is not possible to attain all of those goals simultaneously.

1. Tradeoff between Beamforming and Diversity Gain

- There is a tradeoff between beamforming and diversity gain; this tradeoff also depends on the environment in which we are operating.
- Consider first an LOS scenario. In this case, it is obvious that the achievable beamforming gain is N_t N_r. Beams are formed at the TX (with gain N_t) and at the RX (with gain N_r) and point them at each other. The gains thus multiply. On the other hand, there is obviously no diversity gain, since there is no fading in an LOS scenario - in other words, the slope of the SNR distribution curve does not change.

- The diversity order is the slope of the BER versus SNR curve for very high SNRs :

$$d_{div} = -\lim_{\bar{\gamma} \to \infty} \frac{\log[\mathrm{BER}(\bar{\gamma})]}{\log(\bar{\gamma})}$$

2. Tradeoff between Spatial Multiplexing and Diversity

- There is also a fundamental tradeoff between spatial multiplexing and diversity. The optimum tradeoff curve between diversity order and rate r is piecewise linear, connecting the points :

$$d_{div}(r) = (N_t - r)(N_t - r), \quad r = 0, \ldots, \min(N_t, N_r)$$

- This implies the maximum diversity order $(N_t N_r)$ and maximum rate, $\min(N_t N_r)$, cannot be achieved simultaneously.

2.6 Multiuser MIMO

- In a cellular scenario where the BS communicates with multiple users at the same time, multiuser MIMO system is employed. The system model is outlined in Fig. 2.6.1.

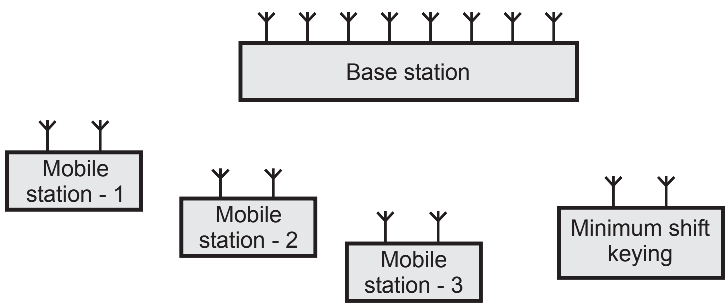


Fig. 2.6.1 Multiuser MIMO system setup

- A single BS with N_{BS} antenna elements communicates with $N_{MS}^{(k)}$ with antenna elements each. This is the case normally occurring in a cellular network and also the one with the most interesting effects for multiuser MIMO.

2.6.1 Performance Limits

- There are fundamental limits of multiuser MIMO about number of users can be supported and at what rate.

Uplink

- The capacity of the uplink system grows with N_{BS}, the number of BS antenna elements.
- The RX at the BS can be an H-BLAST RX that combines MMSE reception with serial interference cancellation.

Downlink

- The signal transmitted to the users consists of a superposition of the data intended for all the different users are linearly or nonlinearly encoded versions of the signal streams intended for the different users.
- The MSs cannot cooperate for the decoding of the datastreams. This has much more significant consequences for the practical implementation than the inability to cooperate for the encoding : Access of the decoder to all the data streams is critical for interference suppression. Consequently, the only way to avoid interference is via a TX precoding that eliminates interference at the RX (and which requires CSIT).

2.6.2 Scheduling

- If the number of data streams for users is larger than N_{BS}, then the BS has to determine which users it wants to serve at a given time. The criteria for the serving is for the maximization of the overall throughput (without regard to fairness) or to ensure that each user is served with a certain minimum data rate.
- Different ways of selecting users (and possibly, grouping users together in certain timeslots) lead to different overall capacity and quality of service.
- Optimizing such criteria in principle requires an exhaustive search over all scheduling assignments which becomes infeasible if the number of users is large.

2.6.3 Linear Precoding - Uplink

- The simplest practical implementation of multiuser MIMO processing is based on linear processing at TX and RX. The RX (BS) has all signals available and can thus perform optimum (linear) processing for interference suppression.
- The optimum transmit and receive weights depend on each other (changing the TX weights at the MS changes the amount of useful power and interference for all the users at the BS) therefore an iterative determination of the optimum weights is necessary. The details depend on the optimization criteria; in the following we consider the minimization of the overall Mean Square Error (MSE).

2.6.4 Linear Precoding - Downlink

- For downlink, the linear precoding is the method that can be implemented most easily. In the case of beamforming (linear precoding) at the TX, the total transmit signal intended for the k^{th} user is the source signal, multiplied with a beamforming matrix $\mathbf{T}^{(\mathbf{k})}$.

- Fig. 2.6.2 shows the capacity for various types of precoding when each MS has only one antennas for $N_t = 10$, $\gamma = 10$ dB.

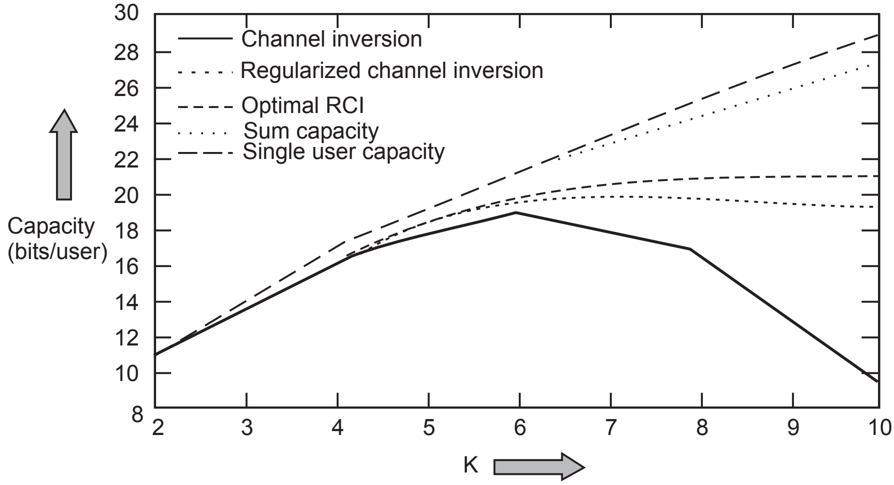


Fig. 2.6.2 Multiuser MIMO capacity as function of the number of users K for different precoding methods for one MS antenna

- It can be seen that with a pure channel inversion, the capacity first increases as the number of users increases, but then starts to decrease again as K starts to approach the number of antenna elements at the BS. This is due to the increased probability of an ill-conditioned channel matrix.
- When the RX has more antennas, the difference between channel inversion and regularized channel inversion becomes much smaller.
- Fig. 2.6.3 shows the capacity for various types of precoding when each MS has multiple antennas for $N_t = 10$, $\gamma = 10$ dB.

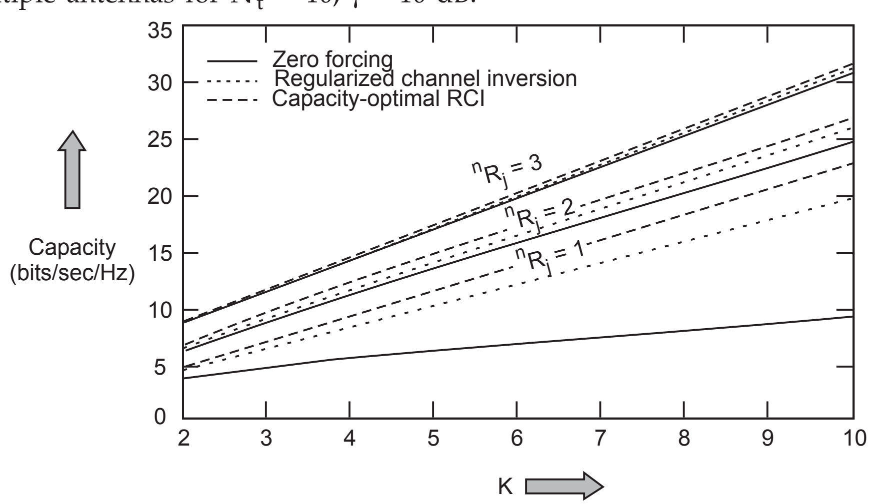


Fig. 2.6.3 Multiuser MIMO capacity as function of the number of users K for different precoding methods for multiple MS antenna

2.6.5 Closed-Loop Systems and Quantized Feedback

- The limited-feedback codebooks provide a good way of reducing the feedback overhead for transmit beamforming. For multiuser MIMO this technique is much more difficult.
- Firstly, the quantization of transmit precoding vector has to be much finer than in the single-user case. This can be explained as : In a single-user case the BS forms a beam pattern that shows a maximum in the direction of the targeted MS. Slight deviations from the optimum beam pattern, due to the quantization effects, do not lead to a significant loss of SNR.
- In the multiuser case, in order to achieve good SINR at the MSs, each transmit beam pattern needs to place nulls toward the users it should not cover. The nulls are much more sensitive to perturbations of the antenna weights than main beams. Therefore, quantization in multiuser settings must be finer.

2.6.6 Base Station Cooperation

- In a cellular system, the existence of multiple users and hence interferers influences the capacity, and decreases the data rate that is possible for a single user.
- For a cellular MIMO system with $N_r = N_t$, the degrees of freedom created by the multiple BS antennas are all used for the separation of the multiple data streams from a single user and none for the suppression of interfering users. Therefore, neither SDMA nor SFIR can be used for the increase of the cellular capacity.

2.7 Software-Defined Radio (SDR)

- An SDR is a radio communication system where components that have typically been implemented in hardware, such as mixers, filters, amplifiers, modulators, demodulators and detectors, are instead implemented in software.
- An SDR can provide software control of a variety of modulation techniques, wideband or narrowband operations, security mechanisms and waveform requirements of current and evolving wireless standards over a broad frequency range.
- Software-defined radios have significant utility for cell phone services, which must serve a wide variety of changing radio protocols in real time.
- A simplified SDR is shown in Fig. 2.7.1.
- In the ideal SDR, the antenna connects directly to the Low Noise Amplifier (LNA) and Analog-to-Digital Converter (ADC) for the reception path or the Power Amplifier (PA) and Digital-to-Analog Converter (DAC) for the transmission path.

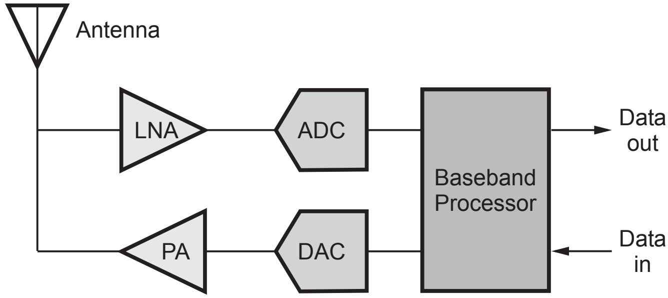


Fig. 2.7.1 Simplified SDR

- The processor handles all radio functions, such as filtering, up/down-conversion, modulation and demodulation and the digital baseband. This is a paradigm change from traditional radio engineering.
- With SDR, the standards are implemented in software, to be loaded and executed in a common hardware platform consisting of FPGA and DSP chips.
- SDR represents a modern approach to radio engineering and will certainly have significant impacts in our society.
- The important SDR capabilities are :

1. The SDR module is capable of reconfiguring RF and can be programmed to tune to a wide spectrum range and operate on any frequency bands within the range.
2. SDRs can be quickly and easily upgraded with enhanced features.
3. SDRs can talk and listen to multiple channels simultaneously.
4. Diverse needs for different systems can now be satisfied with a single generic hardware, thus achieving great interoperability among various wireless networks.
5. The potential of SDR is maximized when combined with spectrum sensing and the so-called cognitive engine. The combination, termed **Cognitive Radio (CR)**, is a frequency-agile wireless communication device that enables dynamic spectrum access. A CR can sense the spectrum to detect unused frequency bands as well as its radio environment and make decisions on how to reconfigure the SDR to achieve improved performance.

- SDR has a broad spectrum of applications for both military and civilian wireless networks, many of which may require supporting a wide variety of evolving wireless protocols in real time.

2.7.1 Advantages of SDR

1. **Open architecture -** SDR is based on open architecture and consists of a common, generic hardware platform, which allows for flexible installation of different software applications as required for signal transmission.

2. **Interoperability -** Support of multiple standards through multimode, multiband radio capabilities.
3. **Flexibility -** Efficient shift of technology and resources.
4. **Adaptability -** Faster migration towards new standards and technologies through programmability and reconfiguration.
5. **Sustainability -** Increased utilization through generic hardware platforms.
6. **Reduced ownership costs -** Less infrastructure, less maintenance, easier deployment.
7. The SDR can receive and transmit various modulation methods using a common set of hardware.
8. The ability to alter functionality by downloading and running new software at will.
9. SDR provides a more efficient use of the spectrum.
10. The possibility of adaptively choosing an operating frequency and a mode best suited for prevailing conditions.
11. SDR can recognize and avoid interference with other communications channels.
12. Elimination of analog hardware resulting in simplification of radio architectures and improved performance.

2.8 Radio Resource Management (RRM)

- In cellular segments of wireless IP networks, there is a need for simple radio resource and teletraffic management solutions that allow operators to customize various network and user profiles, optimize the Grade of Service (GoS) while fulfilling the QoS requirements.
- Using the radio resource management framework, solutions for future network operators to customize and optimize the performance of their multimedia networks serving various user and network profiles can be estimated.
- The Radio Resource Manager (RRM) contains a number of sub-blocks like the connection admission controller, the traffic classifier, the radio resource scheduler and the interference and noise measurements.
- The main role of the RRM is to manage the different available resources to achieve a list of target QoS.

2.8.1 Radio Resource Scheduler (RRS)

- The Radio Resource Scheduler (RRS) is an essential part of the RRM.

- The RRS has two important radio resources to control : MS transmitting power and transmitted data rate.
- The RRS uses those two resources to achieve different objectives like maximizing the number of simultaneous users, reducing the total transmitting power or increasing the total throughput.

2.8.1.1 Objectvies of RRS

- The objectives of the RRS are :

1) Minimize the total transmitting power.
2) Achieve the target SINR in order to achieve a certain BER level (depends on the application).
3) Maximize the fairness between the users. In our definition, the system is fair as long as each user is supported by at least its minimum required QoS. In this sense, minimizing the outage probability leads to maximizing the fairness.
4) Maximize the total transmitted data rate or at least achieve the minimum required data rate.

- The conventional way to achieve these objectives is to select one of them as a target to optimize and use other objectives as constraints. More sophisticated algorithms based on multiobjective (MO) optimization and Kalman filter techniques have been also proposed.
- The modern communication systems (3G or 4G) are supporting the multirate data communication because they are designed not only for voice communication but also for data and multimedia communication.
- An efficient combining algorithm for the power control and the rate control is required for 4G system.
- The term 'efficient' means optimization of the transmitted power and data rate to meet the necessary specifications.
- There are many algorithms for combining power and rate control. The objectives of all algorithms are different.
- Some algorithms suggest maximizing the throughput; others minimizing the packet delay or minimizing the total power consumption.
- There are mainly two methods to achieve the multi-rate transmission,

1. Multi-Code (MC) scheme.
2. Variable-Spreading Length (VSL) scheme.

2.9 QoS Requirements

- Next generation wireless communication systems (4G), will provide a wide range of services to the users. These services, ranging from legacy applications, such as data transfer, to voice and multimedia calls and advanced value-added services, must be supported across a great diversity of network access technologies and by operators targeting very different market segments.
- In order to satisfy user requirements, proper end-to-end QoS must be provided to the application flows. The requirements of seamless mobility of users and scalability further complicate the issue.
- The provision of seamless end-to-end QoS in such a demanding and heterogeneous scenario is still a major challenge in networks research.
- In order to provide end-to-end QoS to the application flows, enough resources must be available along the entire flow path.
- There can be scenario, where the mobile terminals communicating are attached to different access domains, this path comprises :

1) The access networks of both terminals,

2) The core network of the access domains where the access networks belong and

3) The inter-domain path, consisting of all the transit domains traversed by the flows.

- The QoS parameters for different sections of the network are identified and suitable solution is worked out to meet the desired application.

2.10 Short Answered Questions

Q.1 What is OFDM ?

Ans. : OFDM (Orthogonal Frequency Division Multiplexing)

- OFDM is a broadband multicarrier modulation method that offers superior performance and benefits over older, more traditional single-carrier modulation methods because it is a better fit with today's high-speed data requirements and operation in the UHF and microwave spectrum.
- OFDM is based on the concept of Frequency-Division Multiplexing (FDD), the method of transmitting multiple data streams over a common broadband medium. That medium could be radio spectrum, coax cable, twisted pair or fiber-optic cable.

Q.2 What is MIMO and what is the functionalities of MIMO ?

Ans. : (MIMO) stands for Multiple Input Multiple Output.

- The functionalities of MIMO it have multiple antennas at the transmitter side and multiple antennas also have at the receiver side.

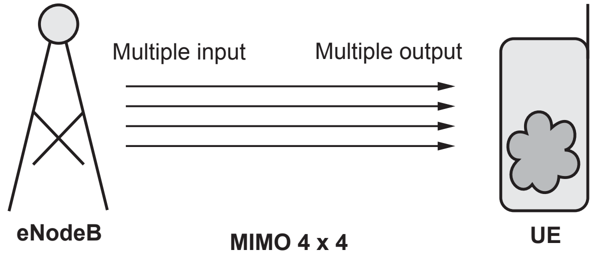


Fig. 2.10.1

2.11 Multiple Choice Questions

Q.1 Orthogonal Frequency Division Multiplexing (OFDM) is used in which of the following wireless technologies _______.

a Long Term Evolution (LTE)

b Worldwide Interoperability for Microwave Access (WiMAX)

c 802.11a for WLAN (Wireless Local Area Network)

d all of the above

Q.2 Difference between Multi-Carrier Modulation (MCM) and Orthogonal Frequency Division Multiplexing (OFDM) is _______.

a multi-carrier transmission

b IFFT/ FFT operation

c MCM does not overcome intersymbol interference

d MCM can only be used for a large bandwidth

Q.3 OFDM converts a frequency selective channel into ____________.

a a single flat fading channel

b N parallel flat-fading channels, where N is the number of subcarriers

c time selective channel

d stationary channel

Q.4 The diversity of the BER across each OFDM subcarrier is _______.

a 0

b 1

c ∞

d L

Answer Keys for Multiple Choice Questions :

Q.1	d	**Q.2**	b	**Q.3**	b	**Q.4**	b

❑❑❑

Notes

UNIT III

3 Introduction to Mobile Communication

Syllabus

Introduction to Cellular Service Progression, Cell Geometry, Overview of Cellular mobile and Network architecture, Cellular radio system design - Frequency assignments, frequency reuse channels, Concept of cell splitting and Cell sectoring. Significance of Handover in cellular systems with Handoff algorithms and roaming.

Contents

3.1 Cellular Service

SPPU : May-17

- The cellular topology is dominant topology that is used in all large-scale terrestrial and satellite wireless networks.
- A **cellular system** is a combination of a modulation and multiple access techniques, this method is equally applicable to both analog and digital systems.
- Cellular systems are complex but a much more efficient. It includes several disciplines of engineering and has taken much enterprise and development to assemble into global systems. Cellular radio requres combination of many large scale technology e.g. HF semiconductor technologies, radio transmission planning and global fixed telecommunications networks.
- In cellular systems, improved spectral efficiency and ability to handle heavy traffic demands can be achieved by frequency reuse and cell splitting techniques. **Frequency reuse** refers to the use of radio channels on the same carrier frequency to cover different areas which are separated from one another by a sufficient distance so that co-channel interference is not objectionable.
- **Cell splitting** is further dividing a cell into smaller cells a set of channel frequencies is reused more often, leading to a higher spectral efficiency.
- Higher spectral efficiency leads to more subscribers, cheaper equipment due to mass production, low call charges and, overall lower cost per subscriber.

3.1.1 Cellular Concept

- The overall service area is divided into small cell, ideally with no gaps or overlaps, each cell being served by its own base station and a set of channel frequencies. The power transmitted by each station is controlled in such a way that the local mobile stations in the cell are served, while co-channel interference, in the cells using the same set of radio channel frequencies is kept minimum.
- As shown in Fig. 3.1.1 the cells are hexagons with the repeater and base station at the centre. The N cells which collectively use the complete set of available frequencies is called a **cluster.**
- Cell sizes are made smaller at the centre of the city or area of occupation of most subscribers.

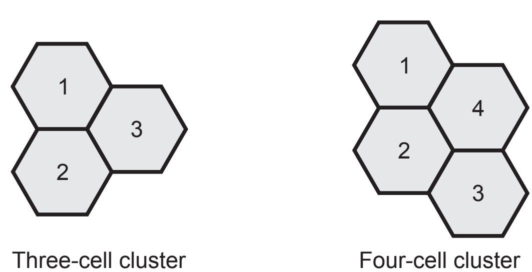
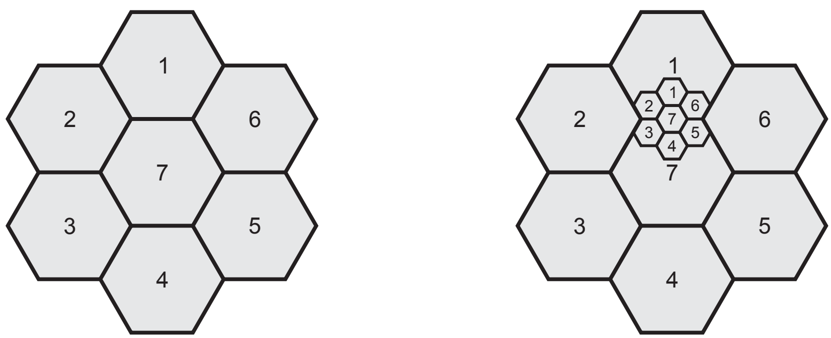


Fig. 3.1.1

- Cells are arranged in clusters. Only certain cluster sizes are possible, principally due to geometry of a hexagon and the allowable cluster sizes of 3, 4, 7 and 12 are shown by way of illustration.
- The hexagon shape permits easy and manageable analysis of a cellular system. The actual radio coverage of a cell is known as **footprint**. The footprint of cell is determined by field measurements or propagation prediction models.
- The hexagonal cells fit together to form a honey comb pattern. Fig. 3.1.2 shows seven cell honey comb pattern or seven cell cluster.



(a) Honey comb pattern or seven cell cluster **(b) Two sizes of cell**

Fig. 3.1.2 The concept of cell

- The hexagonal shape of cell ensures the most effective transmission but in reality the antenna patterns will not achieve this pattern, the cells more likely takes the circular pattern with some overlap.
- When planning a system the aim must be to achieve the maximum use of the available radio spectrum. Also there must be low interference, good quality speech and an acceptable grade of service.
- The frequency channels are full duplex hence each conversation requires a pair of frequencies. The forward and reverse directions from the base station to the mobile are to different frequency band and the two frequencies are separated by 45 MHz.
- The factors affecting number of channels in a particular area :
  1. The available frequency spectrum.
  2. The cell size or transmitter power.
  3. The reduction in the quality of the link that can be tolerated due to co-channel interference.

- The normal maximum number of channels operating in a cell is limited to 120 and this occurs in places where the traffic is highest. The capacity of a system in an area is determined by the number of channels in a cell and the cell size.
- The number of simultaneous users is given by expression :

$$n = \frac{m\,(W / N)}{B}$$

Where,

W is total available spectrum,

B is bandwidth needed per user,

N is frequency reuse factor,

m is number of cell required to cover an area.

- Above expression indicates the capacity of network can be increased by :
  i) Increasing m
  ii) Decreasing frequency reuse factor

3.1.2 Cellular Hierarchy

- Hierarchical cellular infrastructures of different sizes are used in cellular network because of following reasons :
  i) To extend the coverage to the areas those are difficult to cover by a large cell.
  ii) To increase the capacity of the network for those areas that has a higher density of users.
  iii) To provide coverage for specific application.
- For deployment of cellular network numbers of cell sizes are used to provide a comprehensive coverage supporting traffic fluctuations in different geographic areas and supporting a variety of applications.
- Different cell sizes are defined as following.

1. Femtocells

- Femtocells are the smallest unit of hierarchy used for connection of personal equipment such as laptops and cellular telephones.
- The femtocells cover only few meters where these devices are used within physical range of users.

2. Picocells

- Picocells are the small cells inside a building that support local indoor networks. For example, wireless LANs, Wi-Fi networks.
- The size of these networks is in the range of a few tens of meters.

3. Microcells

- The microcells cover the interiors of streets and its antenna is located at the heights lower than the rooftop of the building.
- The microcell covers range of few hundreds of meter. It is used for personal communication systems.

4. Macrocells

- Macrocells cover metropolitan areas and its antennas are mounted above the rooftop of the buildings in the coverage area.
- The macrocells cover areas on the order of several kilometres.

5. Megacells

- Megacells cover nationwide areas with satellites. It usually covers ranges of hundreds of kilometers.

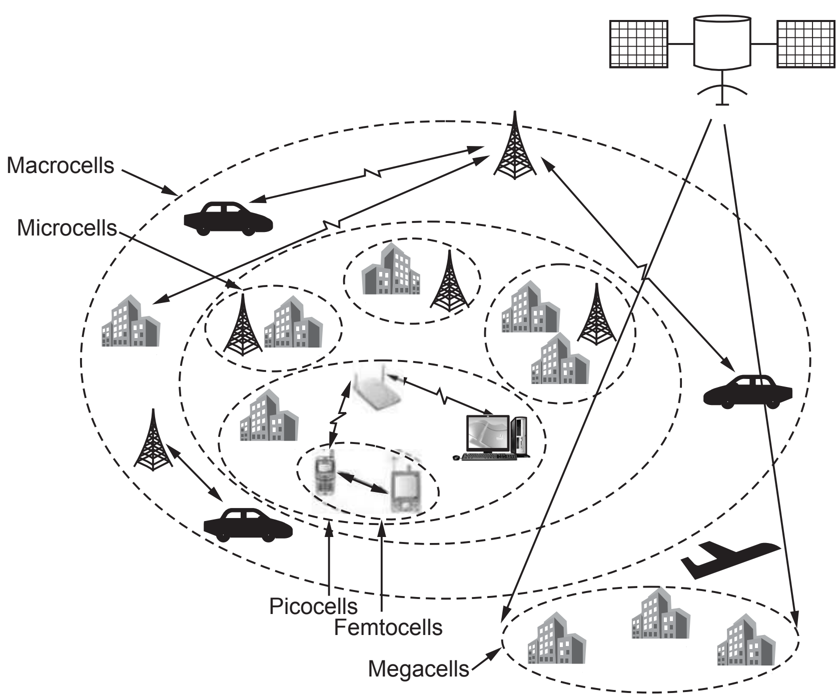


Fig. 3.1.3 Coverage areas of different cells

Solved Example

Example 3.1.1 *A service provider wants to provide cellular communication to a particular geographic area. The total bandwidth the service provider licensed is 5 MHz and system subscriber requires 10 kHz of bandwidth. Determine the system capacity, if the service provider implements a cellular system with 35 transmitter sites and cluster size of 7.*

Solution : Given : Cluster size N = 7

$$\text{Total system bandwidth B} = 5 \text{ MHz} = 5000 \text{ kHz}$$

$$\text{Bandwidth per subscriber} = 10 \text{ kHz}$$

$$\text{Total cell transmitter} = 35$$

$$\text{Bandwidth per cell} = \frac{B}{N} = \frac{5000 \text{ kHz}}{7} \approx 714 \text{ kHz}$$

$$\text{Capacity of each cell} = \frac{714 \text{ kHz}}{10 \text{ kHz / user}} = 71 \text{ user}$$

$$\text{Total system capacity} = \text{Total number of cells} * \text{Capacity of each cell}$$

$$\text{Total system capacity} = 35 * 71 \text{ users} = \mathbf{2485 \text{ users}}$$

3.1.3 Cell Fundamental

- The use of hexagon allows for the complete theoretical coverage of an area without any overlapping cells or gaps in the coverage area.
- The use of hexagons makes the theoretical calculations of system parameters much easier.
- There are a few geometrical figures which ensure full coverage of a given area without either overlapping or holes. These are equilateral triangles, squares and hexagons. Hexagons best approximate the circular shape of base station coverage in a flat terrain without obstacles and the hexagonal edges well approximate the borders between cells of the same size.
- In reality, the base station coverage does not have a regular circular shape because the coverage is a result of terrain architecture and obstacles such as houses, trees, etc.

University Question

1. *What is Microcell zone concept ? How is it used to improve capacity ?*

SPPU : May-17, Marks 5

3.2 Frequency Reuse

SPPU : April-16, 17, 18, May-16

- In a cellular system, the frequency space allocated is insufficient. For a 7 cell cluster arrangement the allocation of frequencies into seven sets is required. The same frequency band or channel used in a cell can be reused in another cluster or

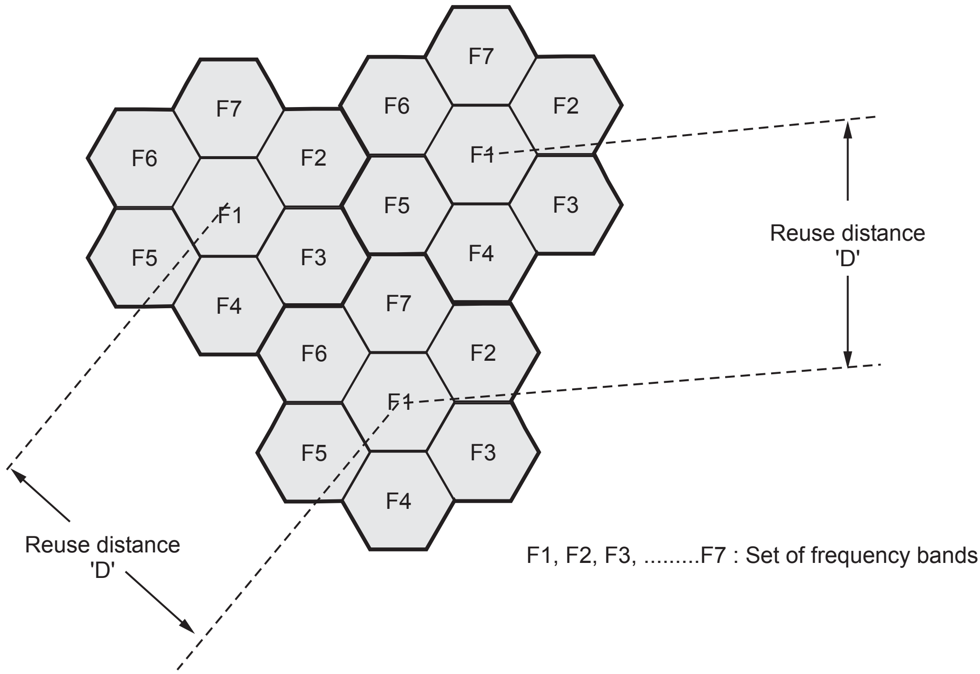


Fig. 3.2.1 Frequency reuse for 7 cell cluster

cell i.e. frequency to be used for multiple simultaneous conversations. This is referred to as **frequency reuse.**

- Frequency reuse is the process of using the same set of frequencies to more than one cell.
- However frequency reuse depends on various factors such as transmitter power of base station, antenna gain and height, distance between cells. The distance between the two cells using the same frequency is known as **reuse distance**, is denoted by D. A typical cluster of seven cells shows frequency reuse pattern and reuse distance.

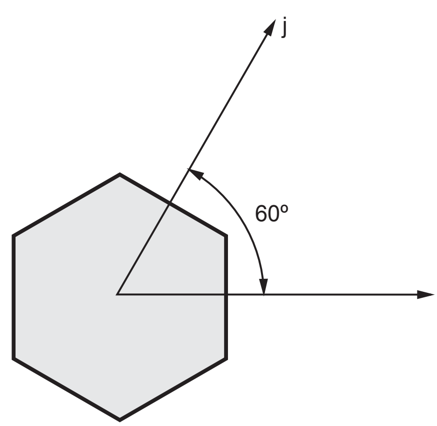


Fig. 3.2.2

- Frequency reuse distance is decided by cluster size 'N'. In hexagonal cell pattern the cluster size (number of cells per cluster) is given by,

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

where N represents the cluster size.

i represents the number of cells to be transversed along direction qi from center of cell.

j represents number of cells in direction 60° to the direction of i.

Substituting different values of i and j (nonnegative integers)

N = 1, 3, 4, 7, 9, 12, 13 ....

Most popular value of N are 4 and 7.

- Due to hexagonal geometry, there are six equidistant neighbours and each neighbor is separated by multiples of 60°.

3.2.1 Frequency Reuse Factor

- The relationship between frequency reuse distance 'D', radius and cell 'R' and number of cells per cluster 'N' is represented by,

$$\boxed{D = \sqrt{3N}\,R}$$

The ratio $\frac{D}{R}$ is known as **reuse factor**.

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

- Fig. 3.2.3 shows various reuse patterns.
- If the system is not properly designed with respect to the number of cells in a cluster, topographic cell distribution and channel assignment, then it will experience excessive interference between the channels in different cells which use the same carrier frequencies.

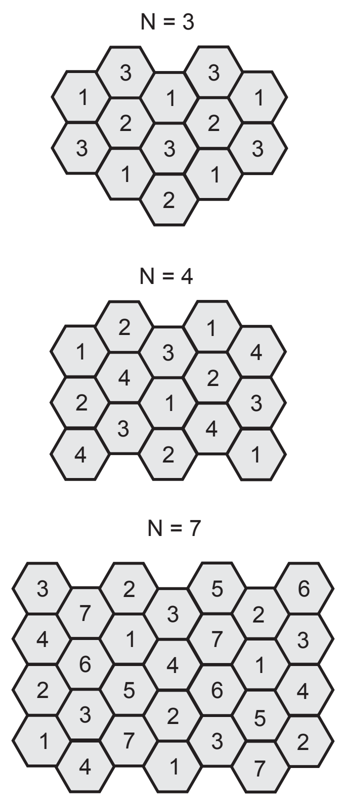


Fig. 3.2.3 Freuency reuse patterns for N = 3, N = 4 and N = 7

Solved Examples

Example 3.2.1 *For a mobile system of cluster size of 7, determine the frequency reuse distance if the cell radius is 5 km. Repeat the calculation for a cluster size of 4.*

Solution : Given : Cluster size N = 7

Cell radius R = 5 km

Frequency reuse distance is given by

$$D = \sqrt{3N}R$$

$$D = \sqrt{3*7} * 5 = 4.5823*5$$

$$D = \mathbf{22.913\ km} \qquad \text{... Ans.}$$

For cluster size

$$N = 4$$

$$D = \sqrt{3*4} *5 = 3.464*5$$

$$\mathbf{D = 17.32\ km} \qquad \text{... Ans.}$$

Example 3.2.2 *Determine frequency reuse distance for a cell radius of 2 km and cluster size of 8.*

Solution : Given : Cluster size N = 8

Cell radius R = 2 km

Frequency reuse distance is given by

$$D = \sqrt{3N} \cdot R$$

$$D = \sqrt{3*8} *2 = 4.8989*2$$

$$\mathbf{D = 9.7979\ km} \qquad \text{... Ans.}$$

Example 3.2.3 *For given path loss component n = 4 and frequency reuse factor of N = 7 calculate S/I ratio in a cellular system.* **SPPU : April-16, Marks 4**

Solution : Given : n = 4; Reuse pattern N = 7 cell

Frequency reuse factor = $\sqrt{3*N} = \sqrt{3*7} = 4.582$

Signal to noise ratio

$$S/I = \frac{\sqrt{(3*N)^n}}{i_0} = \frac{4.582^4}{6} = 75.3 = \mathbf{18.66\ dB}$$

Example 3.2.4 *A spectrum of 30 MHz is allocated to a wireless FDD cellular system which uses two 25 kHz simplex channels to provide full duplex voice and control channels, compute the number of channels available per cell if a system.*
i) Uses seven cell reuse and ii) 12 cell reuse **SPPU : April-16, Marks 4**

Solution : BW = 30 MHz

Channel BW = 25 kHz * 2 Simplex channel = 50 kHz per duplex channel.

Total No. of channels = 30 MHz / 50 kHz = 600 Channel

i) For N = 7

No. of channels available per cell = 600/7 = **85 Channels**

ii) For N = 12

No. of channels available per cell = 600/12 = **50 Channels**

Example 3.2.5 *A spectrum of 30 MHz is allocated to a wireless FDD cellular system which uses two 25 kHz simplex channels to provide full duplex voice and control channel, compute the number of channels available per cell if a system uses : i) 4 cell reuse ii) 7 cell reuse iii) 8 cell reuse*
Assume 1 MHz of spectrum is allocated to control channel. Give distribution of voice and control channels. **SPPU : April-17, Marks 5**

Solution : BW = 30 MHz

Channel BW = 25 kHz * 2 Simplex channel = 50 kHz per duplex channel.

Total No. of channels = 30 MHz / 50 kHz = 600 Channel

i) For N = 4

No. of channels available per cell = 600/4

= **150 Channels** **... Ans.**

ii) For N = 7

No. of channels available per cell = 600/7

= **85 Channels** **... Ans.**

ii) For N = 8

No. of channels available per cell = 600/8

= **75 Channels** **... Ans.**

3.2.2 Co-Channel Interference

- With frequency reuse, many cells at a distance will be using the same frequency bands within a given area. These cells are called as co-channels. There is possibility of interference between them since they are operating at same frequency, the interference between them is called as **co-channel interference.** For a 7 cell clusters there could be up to six immediate interferers as shown in Fig. 3.2.4.

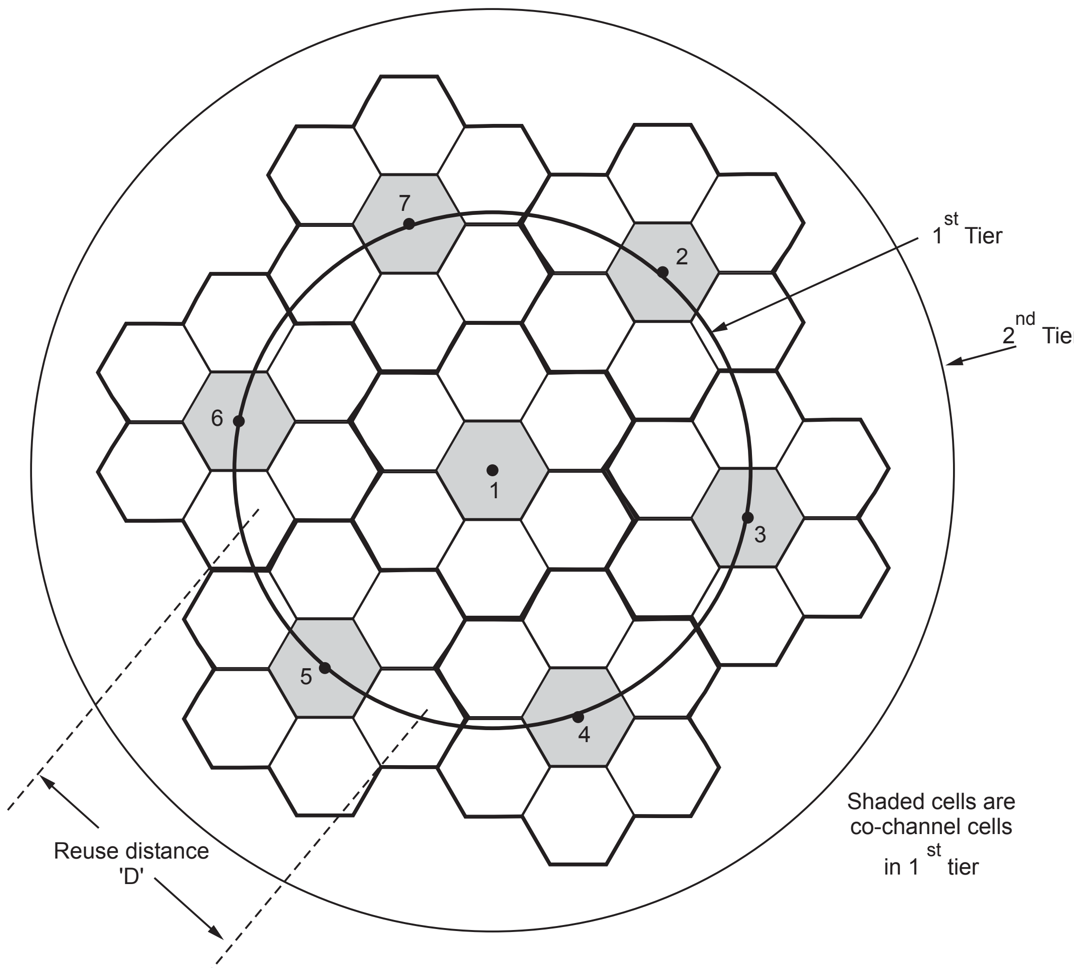


Fig. 3.2.4 Interfering cells in a 7 cell cluster pattern

- The co-channel interference exists even after the power levels of the interfering cells are low enough. The co-channel interference can be experienced by mobile unit and within cell site also.

- The co-channel interference is measured by the ratio of carrier to interference (C/I) at the cell site.
- For measuring carrier to interference ratio consider two cells using the same frequencies at a reuse distance ' D ' as shown in Fig. 3.2.5.

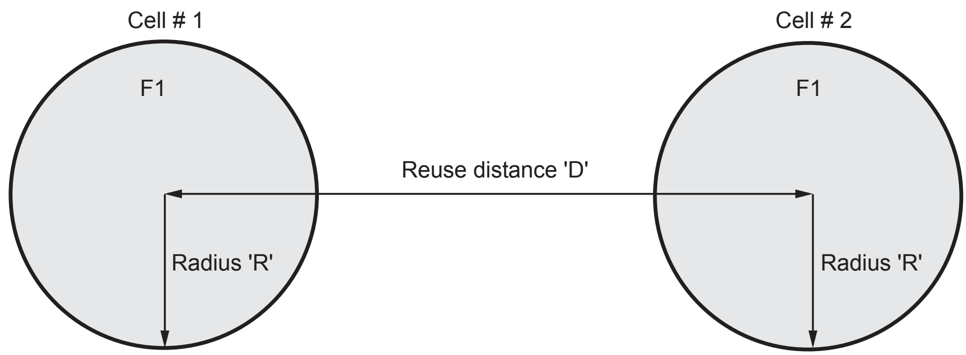


Fig. 3.2.5 Carrier to interference ratio

- Let the power transmitted by each base station is fixed. Then the received power by mobile at a distance from the base station is proportional to $r^{-\gamma}$ where γ is propagation constant.
- Typical value of γ for free space is 2 and for urban area it is 4.

Co-channel Interference Ratio (CIR) is given by

$$\frac{C}{I} = \frac{\text{Carrier}}{\text{Interference}} = \frac{C}{\sum_{k=1}^{m} I_k}$$

where I_k represents co-channel interference.

m represents number of interfering cells.

For 7-cluster size, m = 6

$$\therefore \quad \frac{C}{I} = \frac{C}{\sum 6\, I_k}$$

$$\frac{C}{I} = \frac{C}{6\left(\frac{D}{R}\right)^{\gamma}} = \frac{C}{6\left(\frac{D}{R}\right)^{4}}$$

- The co-channel interference is a function of a parameter, q

$$\boxed{q = \frac{D}{R}}$$

The parameter 'q' is called the **co-channel interference reduction factor**.

- Factor q is independent of actual power level P_0 which is assumed the same for all cells. Let all the cells contribute interference equally.

$$\frac{C}{I} = \left(\frac{R}{6D}\right)^{-4} = \frac{R^{-4}}{6D^{-4}} = \frac{1}{6q^4}$$

Now reuse factor

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

$$\therefore \quad \frac{R}{D} = \frac{1}{\sqrt{3N}} = (3N)^{-1/2}$$

Substituting in above equation

$$\frac{C}{I} = \frac{1}{6}(3N)^{\frac{-1}{2}\times -4} = \frac{1}{6}\left(3N^2\right)$$

$$\frac{C}{I} = 1.5\ N^2$$

i.e. The C/I ratio is a function of cluster size.

- In the cellular environment it is normal practice to specify that CIR should be greater than 18 dB for acceptable performance.

$$\because \quad 18\text{ dB} = 63.1$$

Also
$$\frac{C}{I} = \frac{q^4}{6}$$

$$63.1 = \frac{q^4}{6}$$

$$q^4 = 6 \times 63.1$$

Hence
$$q = 4.41$$

Again reuse factor

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

$$\therefore \quad q = \sqrt{3N}$$

Substituting value of q = 4.41

$$4.41 = \sqrt{3N}$$

$$N = 6.48$$

- Hence cluster size for this CIR is 7, note that this is approximate analysis.

University Questions

1. *Derive the approximate formula for S/I using co-channel reuse ratio Q.* **SPPU : May-16, Marks 6**
2. *Write a note on : i) Frequency reuse and its advantages*
   ii) Interference and system capacity **SPPU : April-17, Marks 5**
3. *Write short note on : Frequency reuse.* **SPPU : April-18, Marks 2**

3.3 Channel Assignment

SPPU : April-16, May-17, Dec.-16, 17

- **Channel assignment** is process of allocating specific channels to cell sites and mobile units. Careful channel assignment eliminates the interference in the system. The channel assignment is done on short term and long term basis. The long term channel assignment is also called as fixed channel assignment.
- Channel assignment is applied for setup channels and voice channels.

Channel assignment to cell sites

- The channel assignment to the cell sites is considered with respect to fixed channel assignment. Here the concept of fixed channel assignment is that channels are assigned to cell sites for a long period. The two different types of channels assigned are :

i) Voice channels

ii) Set-up channels

i) Voice channels

- The assignment of voice channels in every cell site is with idea of reducing the adjacent channel interference (ACI) and co-channel interference (CCI). As discussed earlier the voice channels is expressed as two groups named as A and B in the frequency management topic.

ii) Set-up channels

- In each cell in the system is allocated with 21 set-up channels. The corresponding cluster sites are N = 4, N = 7 or N = 12, reuse patterns.
- In case the antenna used is omnidirectional then one set-up channel is enough. It may lead to many unused set-up channels. But it is better not to use the set-up channel of neighbourhood of the block (from A to B or from B to A) so that it will avoid interference in the system.

Channel assignment to the mobile units in moving status :

- The channel assignment process to roaming mobile unit is high during peak hours, of morning and evening. But it is opposite during the night hours when traffic intensity reduces. In case the traffic is uniform in the system, there will be larger backward energy observed from the mobile unit also, the antenna pattern will not affect the system. But the case is reverse if the traffic is non-uniform.
- To have smooth call handling even for the cell sites away from city the transmit power should be low for voice and set-up channels for few cell-sites.
- For controlling the call acceptances and handoff calls three different methodologies are applied. They are given below,
  a) Frequency assignment
  b) Tilted assignment
  c) Underlay - overlay cell arrangement.

a) Frequency assignment

- The frequencies assigned to a cell is a part of one set or more than that of the total available 21 set-up channels. If necessary borrowing of channels is also permitted. In a sectored cell we can also assign required frequencies with no interferences with neighbouring sectors of the cell.

b) Tilted antennas

- The tilted directional antenna set is capable of eliminating interferences. It is a good design practice to tilt an antenna instead of reducing antenna height, particularly to handle foliage areas (with tall trees etc.)

i) Also if the tilting angle (θ) is 22° a notch will be created in the horizontal antenna pattern. Due to this, additional reduction of interference can be achieved.

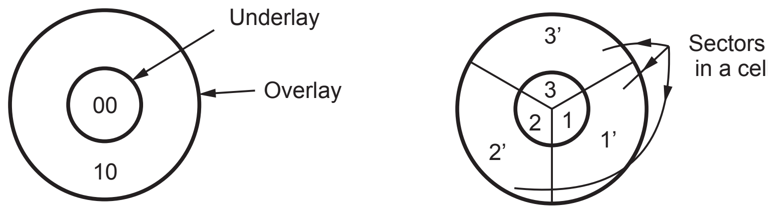
c) Underlay-overlay

- In the cellular arrangements shown below the inner circle represents underlay and the outer circle represents overlay structures. In these two areas the voice powers transmitted are slightly adjusted. To each area a unique voice frequency is assigned.
- For an omnidirectional cell, if N = 7 reuse pattern is used the radius is R then the reuse distance D is,

$$D = 4.6\ R$$

and cochannel interference factor reduction ratio q is,

$$q = \frac{D}{R} = 4.6$$



(a) In omni cells

(b) In sectored cells

Fig. 3.3.1 Overlay-underlay cellular arrangement

- From Fig. 3.3.1, for sectorization in a cell different algorithms are applied. For example in Fig. 3.3.1 (b) there are six regions in total for overlay and underlay areas available.
- Overlay regions/areas - 3
- Underlay regions/ areas - 3

  Total - 6 regions / areas.
- Thus the overlaid and underlaid cell arrangements are adopted in cell sites and its ultimate goal is to increase the traffic capacity in the system.

3.3.1 Fixed Channel Assignment

- Fixed channel assignment is the simplest strategy of system resources distribution.
- When distribution of carriers (channels) from the point of view of interchannel interference minimization. If this channel allocation is done once and kept constant, this type of assignment is called a **fixed channel assignment.**
- In fixed channel assignment each cell is permanently allocated predetermined group of channels. Any call attempt within cell can only be served by unused channels in that particular cell.
- In fixed channel assignment, setting a new connection in a given cell is possible only if there are unoccupied channels in that cell. In the case of a temporary lack of available channels, the user suffers from blocking of a connection.
- Fixed channel assignment can be an inefficient solution resulting in high probability of blocking during busy hours.
- There are more sophisticated channel assignment methods, which take into account a dynamically changing demand for channels such as channel sharing and channel borrowing.

3.3.1.1 Channel Sharing

- In a cell if the traffic is heavy it could share the channels of same cell. That is the channels of another face of the, same cell is shared. Such a channel sharing is done to handle short-term overload situations in the system as shown in the Fig. 3.3.2.

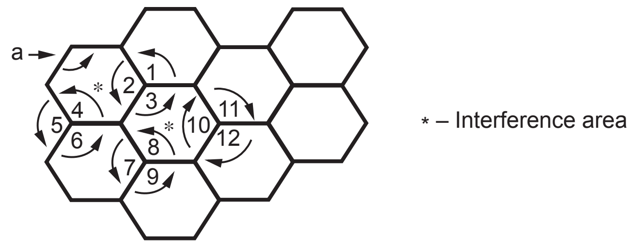


Fig. 3.3.2 Channel sharing concept

- It is considered that there are 21 channels sets, each set consisting of 16 channels. If one face of cell needs more channel it shares the channels of another face of same channel. In cell 'a' shown in Fig. 3.3.2, shares channel from one face to another and due to this channel sharing there will be interference developed in same channel marked with * symbol. In channel sharing concept it is flexible to use channel combiners and at one face of channel the combiner can handle upto 32 channels.

3.3.1.2 Channel Borrowing

- The concept of channel borrowing is done in long-term borrowing. The borrowing is done with other cells or borrowing channels from another face of same cell also can be done.
- This technique is suitable for slowly developing cellular systems. The cell-splitting technique is used in cells to handle different traffic densities. But implementation of cell-splitting technique is costly. Thus if it is badly required for the system, only then it should be used.
- The channel borrowing technique helps to avoid cell splitting in the system. It is a main advantage of channel borrowing technique.

Note : The central cell in the cluster can borrow channels from its neighbouring cells.

3.3.1.3 Sectorization

- In a cellular system cell sectoring would help to delay cell splitting. The available channels are divided into groups. There are different types of sector system is available.

For example i) 120° sector system

ii) 60° sector system

iii) 45° sector system.

A 60° (6 sector) and 120° (3 sectors) sector systems are shown in Fig. 3.3.3.

(a) 6 sectors of 60º each **(b) 3 sectors of 120º each**

Fig. 3.3.3 Cell sectorization concept

- If the cluster N size is 7 then 3 sectors of 120° each in one cell is used. The total channel sets will be 21.
- In special case the sector angle can be narrowed/reduced for the purpose of assigning more number of channels in one sector. But this has to done carefully that the neighbouring channel interference should not increase. In general the cochannels interference is avoided in sectored cells than in the case of cell spliting.
- Comparison of sectorized and non-sectorized cells in the cellular system

i) Sectorized cells :

E.g. Omni cells

- Considering N = 7 frequency reuse pattern : The frequency allotted is based on frequency management chart (as discussed in beginning of this chapter.)
- Terrain profile is assumed to be not flat.
- To reduce CCI cluster size should be N = 12 and cochannel reduction factor is q = 6.

ii) Sectorized cells :

- Three sectors in a cell each with 120° can be used for transmitting and receiving sectorization.
- Changing the sectors on call progress needs handoffs.

- Six sectors with 60° each, is also used for both transmission and receiving sectorization. But here in changing sectors for a call in progress needs more handoffs than 120° sectored cell.
- The 60° or 120° sector is used only for receiving sectorization. Thus an omni antenna should be used for transmitting.
- This method makes accurate decisions related to call handoffs but it does not reduce interference levels.

3.3.1.4 Advantages and Disadvantages of Fixed Channel Assignment

Advantages

1. Less load on Main Switching Center (MSC)
2. Simple to implement

Disadvantages

1. Blocking may happen

3.3.2 Dynamic Channel Assignment

- In the dynamic channel assignment strategy there are no channels permanently assigned to the cells. The channel is assigned to a particular call on a call-by-call basis.
- All channels are placed in a pool, and are assigned to new calls according to the reuse pattern. Signal is returned to the pool, when call is completed.
- MSC allocates frequency channels on dynamic basis if that frequency channel is not presently in use in the cell or any other cell which falls within the minimum restricted distance of frequency reuse to avoid co-channel interference.
- Decision upon the assigned channel can be made either by the mobile switching center or by the mobile station. It can be described as the distributed control of the channel assignment process.
- Dynamic channel assignment reduces chances of blocking which increases trunking capacity of system as all available channels are accessible to all cells.
- In dynamic channel assignment MSC has to collect real time data on channel occupancy, traffic distribution, radio signal strength indication of all channels on continuous basis, thus increasing the computational load on MSC.

3.3.2.1 Advantages and Disadvantages of Dynamic Channel Allocation

Advantages

1. Voice channels are not allocated permanently. That is shared on need-basis.

Disadvantages

1. Requires MSC for processing resulting in burden on MSC.
2. It is complicated system.
3. Issues related to channel allocation are still under research.

University Questions

1. *With a neat diagram, explain the terms : Cell sectoring.* **SPPU : April-16, Marks 3**
2. *Explain different channel assignment strategies.* **SPPU : Dec.-16, Marks 6**
3. *Explain different channel assignment strategies in mobile cellular system.* **SPPU : May-17, Marks 5**
4. *Why is handoff necessary in mobile cellular system ? Explain mobile assisted handoff ?* **SPPU : May-17, Marks 5**
5. *Exaplain with a neat diagram the concept of cell splitting and cell sectoring.* **SPPU : Dec.-17, Marks 5**

3.4 Handoff Strategies

SPPU : May-16, April-17

- The limited available power transmitted by the mobile subscriber determined the cell size. As the subscriber moves (roams) between cells during a journey, the communication with the base station of the departing cell ceases and communication with the base station of the entering cell commences. This process is known as **handoff or handover.**
- Each adjacent base station transmits a frequency that is different from its neighbour. The handoff is accomplished when the received signal from the base station is low enough to exceed a predetermined threshold. At the border between two cells the subscriber is under the influence of two or even three base stations, and the communication link can pass back and forth between base stations as the moving subscriber receiver experiences a fluctuating field strength depending upon the immediate environment such as being surrounded by tall buildings. Hence the Carrier-to-Interference ratio (C/I) on its allocated channel will vary, this is monitored by main switching center,the mobile can be instructed to hand over the strongest base station.

- Handoff mechanism is shown in Fig. 3.4.1 as the mobile unit passes through adjacent cell sites maintained by local base stations.

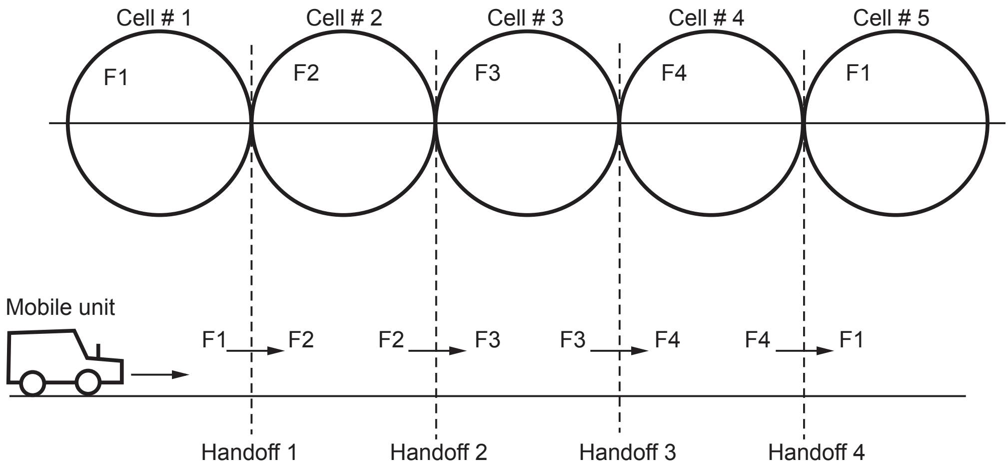


Fig. 3.4.1 Handoff mechanism

- Handoff assures the continuity of calls. Handoff of a call to a new connection implies transfer of security functions also.
- The handoff algorithm must be able to cope with-
  a) Whether the current loss in channel quality is due to short term fading.
  b) Whether a simple increase in power would be sufficient to restore the channel quality (it can produce an unacceptable co-channel interference in other cells using the same frequency).
  c) Whether the measurements from adjacent cells are valid.
  d) Whether the cell chosen for handover has spare channels available.
- In some systems the mobile unit continuously monitors the signals of surrounding base stations and initiates the process of hand off when required. Such systems are known as **Mobile Controlled Handoff (MCHO).**
- In analogue systems such as in AMPS the base stations performs the radio channel measurments and the mobile terminal is totally passive it is known as **Nnetwork-Controlled Handoff (NCHO)**
- In digital systems, both mobile terminals and base stations make measurements and report these to fixed networks for handover decisions, it is known as **Mmobile Assisted Handoff (MAHO)**. It is used in GSM and IS 95.

- The handoff without interruption is called a **soft handoff,** it takes normally 0.2 sec to switch over. A handoff which that is broken momentarily during call transfer is called **hard handoff.** During handoff, the information about the user stored in the earlier base station is transferred to the new base station.
- On receipt a command to 'handover' the mobile stores the new channel number, power level and signalling tone for ms before handing over. For AMPS/TACS the duration time for confirmation of handoff mechanism is 50 msec.
- Whenever handoff occurs communication is interrupted and voice channels are muted. This interruption is usually short and unnoticed during voice communications. After the completion of calls these channels are reallocated to other users.

3.4.1 Reasons of Handoff

- Handoff is required for any of reasons.
  1. The mobile unit moves out of range of a base station.
  2. Traffic in one cell is too high in order to balance the traffic in each cell hand off is initiated.
  3. In noise limited system when the signal strength goes below the threshold of – 100 dBm.
  4. When the signal strength is not at all reaching within the cell site. This happens because of geographical locations and the portion where the signal is not available is called as **holes or gaps.**
  5. When the capacity for connecting new calls of a given cell is used up.
  6. When there is interference in the channels due to the different phones using the same channel in different cells.

3.4.2 Types of Handoff

- 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.
- There exists two types of handoff -
  1. Handoff based on signal strength.
  2. Handoff based on carrier-to-interference ratio (C/I).

- While designing the mobile system the minimum acceptable level of signal strength is decided. For noise-limited system the signal strength is – 100 dBm and for interference limited system the threshold is – 95 dBm. When the signal level in any situation goes below the threshold level, the hand is initiated.
- Also for an acceptable quality of voice the value of carrier-to-interference ratio is decided within cell boundary. When the C/I ratio drops below 18 dB with cell area, handoff initiated.

Sr. No.	Based on SS	Based on C/I
1.	Threshold level i) – 100 dB M in noise limited systems ii) – 95 dB M in interference limited systems	At cell boundary C/I = 18 dB
2.	Easy implementation	Not easy to implenent

- Received Signal Strength (RSS) = C + I

where $C \rightarrow$ Carrier signal power

$I \rightarrow$ Interference level

- If C/I drops in a cell and if the occurance of handoffs is dependent on C/I then in this case as a response to drop C/I either the propagation distance or interference will increase.

3.4.3 Dropped Call

- When a mobile unit moves into a cell where all the channels are busy, there is possibility that the call may be dropped because of lack of free channels, it is called as **blocked call**. The reason of blocked call is non-availability of voice channels.
- The call may drop because of poor signal of the assigned voice channel the termination of call due to weak signal is referred to as **dropped call**. The signal becomes weak because of fading phenomena. The dropped call rate should be minimum to achieve the maximum efficiency of the system.

Dropped call rate is given by

$$p = \sum_{n=0}^{N} \alpha_n \cdot P_n$$

where P_n represents probability of dropped call after n handoff.

α_n represents weighted value for calls having n handoff.

N represents the maximum number of handoff.

3.4.4 Handoff Initiation

- In the cell site the signal strength is continously monitored using a reverse voice channel. Depending on the strength the decision for handoff is made.
- If the signal strength reaches a level that is higher than the threshold level set for minimum voice quality, cell site will request the switching office (MTSO) for handoff to continue the call. Occurance of handoff either earlier or later can be determined by intelligence within the call site also.
- Two points have to be considered and they should be avoided,
  1) An unneccessary handoff will be requested if the handoff decision is very early.
  2) A failure handoff would result if the handoff decision is very late.
- Thus the decision for a handoff on call should be perfect depending on accuracy of signal strength measurements. The threshold can be determined by two parameters namely velocity of vehicle 'V' and the pathloss slope γ in the pathloss curve.
- Assume the threshold level is – 100 dBm at cell boundary. To have a handoff here the signal strength level should be higher than – 100 dBm (Δ).
- If signal strength is = – 100 dBm + ΔdB then a request for handoff will be initiated. The value of Δ should not be too large or too small so that proper handoff initiation at right time will be made.

Note : Handoff may be necessary but cannot be done at following cases

1) Mobile is at signal strength hole and not at cell boundary.
2) If the mobile is at cell boundary but no channel in the new cell is available to make handoff

- In these cases MTSO has to take step to make handoff faster before a dropped call occurance.

Number of handoffs

- If the call size is smaller the number of handoffs taking place will be high.

The number of handoffs for one call progress depends on the size of the cell.

e.g.

Cell area	Number of handoffs
16 to 24 km cell	0.2 handoff / call
3.2 to 8 km cell	1 - 2 handoffs / call

3.4.5 Delaying Handoff

- In simple case of handoff we have an efficient call communication without disturbance inspite of the moving mobile unit's status.
- There is also another handoff called as two-handoff-level that is applied to have successive handoff of a call.
- The use of this algorithm allows two request handoffs so as to provide more chances to have a successful handoff.

3.4.6 Delayed Handoff

- When a base station wants to handover the call to the base station of new cell where the subscriber enters, the new base station will accept it and takes call control. This smooth handoff is possible only if the new cell is free to take it. If there the cell not available (free) then the handoff will be delayed. This is known as delayed handoff scenario.
- A simple two-level handoff technique is shown below with a graph mentioning threshold level.

There are two cases 1 and 2 as shown.

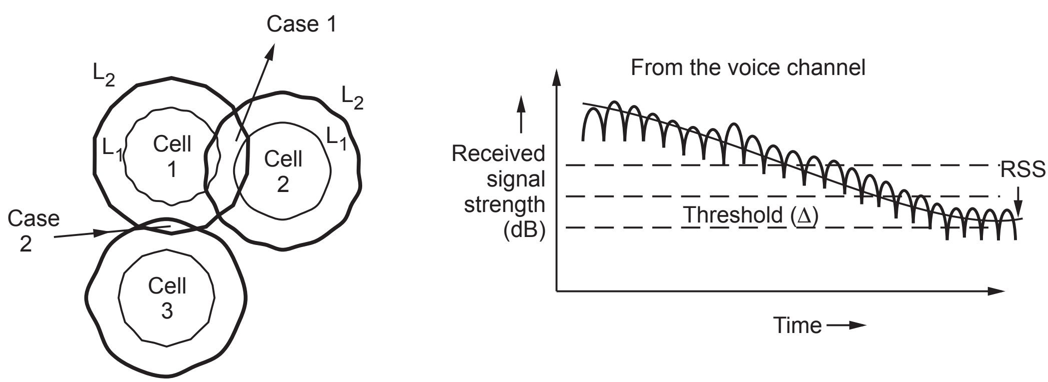


Fig. 3.4.2 Two level handoff technique

Assume that the mobile unit is in a hole or a weakspot or the neighbouring cell of it would be busy. Then handoff will be periodically requested say for every 5 sec.

- Consider the case 1. During first handoff level there will be successful handoff if the new signal is very strong. In case 2 the second handoff level is shown. Here the call in progress is handed it to new cell with no restricted conditions.
- When there are both handoff request and originating call request comes present at same time to MTSO, the MTSO will attend to the handoff call first. The originating new call will be given second priority. The call will be allowed to continue smoothly till the allowed threshold level is reached, as in the graph. Once the threshold level (Δ) is reached the call will be dropped.

Note : If the SAT tone is not sent from mobile unit within a time of 5 sec, then cell site will turn off the transmitter unit. Thus receiving of SAT tone from mobile unit is closely monitored.

Advantages of delayed handoff

1) If the neighbouring cells are busy delayed handoff helps to continue the call in progress smoothly till the new cell gets free.
2) In two-handoff-level alogorithm only after the second handoff the call will be dropped. Thus probability of call blocking is very less.
3) This algorithm also makes handoff to take place at correct location.
4) The algorithm avoids interference in the system .

3.4.7 Forced Handoffs

- A forced handoff technique can be explained with two different definitions
  1) Forced handoff is a technique that is defined as a handoff which should not occur but it is forced to occur.
  2) Forced handoff is a technique that is defined as a handoff that would occur normally but it is prevented to occur.
- In forced handoff two important aspects are :
  i) Controlling a handoff and ii) Creating a handoff

I) Controlling a handoff

- If handoff should occur earlier then handoff threshold level should be high. On the otherhand if handoff should occur later the handoff threshold level should be low.
- Depending upon these criteria a cellsite has to plan a low handoff threshold or high handoff threshold level in the cellsite.
- The Mobile Switching Office (MSC) can also control the handoff and make it to occur either earlier or later, after receiving a handoff request from the cell site in the system.

II) Creating a handoff

- The concept of creating a handoff is dependent on the cell congestion due to mobile traffic.
- If a cell is too congested then mobile switching office decides to create handoffs.
- It informs the cellsite those that are heavily congested due to mobile traffic to create early handoffs. If so some calls will be handed over to neighbour cells and the congested cell will be reaching a moderate mobile traffic.

- Thus handoff threshold level in cell site may be high or low according to the order of MTSO given to cellsites. Depending upon the instructions of MTSO either earlier or delayed handoff would take place in the cell.
- The advantage of this method is to have an efficient mobility management.

3.4.8 Handoff Prioritization

- Handoff fails for many reasons like, if no channel is available in the candidate cell. One of the ways to reduce the handoff failure rate is to prioritize handoff.
- Handoff algorithms try to minimize the number of handoffs which give poor performance in heavy traffic situations. In such situations, a significant handoff performance improvement can be obtained by prioritizing handoff.
- Two basic methods of handoff prioritization are guard channels and queuing of hand off.

1. Guard Channels

- Guard channels improve the probability of successful handoffs by reserving a fixed or dynamically adjustable number of channels exclusively for handoffs.
- An adaptive number of guard channels can help reduce this problem.

2. Queuing of Handoff

- Queuing is a way of delaying handoff. The MSC queues the handoff requests instead of denying access if the candidate BS is busy.
- The probability of a successful handoff can be improved by queuing handoff requests at the cost of increased new call blocking probability and a decrease in the ratio of carried-to-admitted traffic since new calls are not assigned a channel until all the handoff requests in the queue are served.

3.4.9 Handoff Failures

- The reason of handoff failures :
  1. No channel is available on selected BS.
  2. Handoff is denied by the network for reasons such as lack of resources. For example, no bridge or no suitable channel card; the MS has exceeded some limit on the number of handoffs that may be attempted in some period of time. It takes the network too long to set up the handoff after it has been initiated.
  3. The target link fails in some way during the execution of handoff.

3.4.10 Types of Protocols

- In cellular wireless networks, it is very important to deal with Mobile Station (MS) handoff between cells in order to maintain a continuous and QOS-guaranteed service.
- There are four basic types of handoff protocols which help in providing continuous and QOS-guaranteed service. Namely :
  1. Network-Controlled Handoff (NCHO)
  2. Mobile-Assisted Handoff (MAHO)
  3. Soft HandOff (SHO) and
  4. Mobile-Controlled HandOff (MCHO)

University Questions

1. *What is hand off ? Why is it necessary in mobile cellular system ? Explain Mobile Assisted Handoff ?* **SPPU : May-16, Marks 4**
2. *Explain the need of hand off and factors influencing hand off.* **SPPU : April-17, Marks 5**

3.5 Cell Splitting

SPPU : May-16, April-16, 17, 18, Dec.-17

- Cell splitting is a popular technique to expand the capacity of cellular system. Cells in area of high usage can be split into smaller cells.
- **Cell splitting** is the process of subdividing congested cells into smaller cells with their own Base Station (BS); corresponding reduction in antenna height and a corresponding reduction in transmitting power. Fig. 3.5.1 shows cell splitting technique.

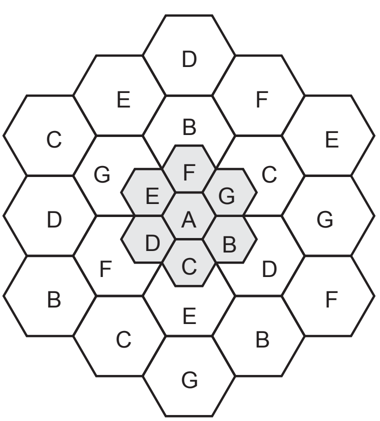


Fig. 3.5.1 Cell splitting

- Cell splitting is achieved through reducing cell radius and keeping the D/R ratio unchanged.
- These smaller cells within the large cell have very interesting characteristics. All of them have their own base stations but these base stations are not so high. The antennas are shorter and they also transmit less power.

- Splitting the cell reduces the cell size thus more number of cells is to be used. More number of cells implies more number of clusters, more number of clusters implies more number of channels because number of channels per cell is fixed and ultimately it leads to a higher capacity.
- Cell splitting allows a system to grow by replacing large cells by smaller cells without upsetting the channel allocation. Frequency planning requires a lot of effort and resource allocation.

Features of Cell Splitting

- Important features of cell splitting are as under :

1. Cell splitting enables more spatial use i.e. greater system capacity.
2. Cell splitting preserves original frequency reuse plan.
3. Cell splitting cause increased frequency handoffs.

3.5.1 Problems in Cell Splitting

- The problems of cell splitting can be categorised in to two categories : Channel assignment and handoff.

Channel Assignment

1. Not all cells are split at the same time.
2. It is often difficult to find real estate that is perfectly suitable for cell splitting.
3. Different cell sizes will exists simultaneously.
4. Special care needs to be taken to keep distance between co-channel cells at the required minimum, and hence channel assignments become more complicated.
5. High speed and low speed traffic needs to be accommodated simultaneously. For this , umbrella cell approach is commonly used.

University Questions

1. *With a neat diagram, explain the terms : Cell splitting.* **SPPU : April-16, 18, Marks 3**
2. *Explain with a neat diagram the concept of cell splitting and cell sectoring.* **SPPU : Dec.-17, Marks 5**

3.6 Short Answered Questions

Q.1 Find the frequency reuse factor if i=2 and j=3.

Ans : Cluster size $N = i^2 + ij + j^2$

$$N = 2^2 + 2 * 3 + 3^2 = 19$$

$$\text{Reuse factor } D/R = \sqrt{3N} = 7.54$$

Q.2 If the number of channels in a cell is 20, 7 cells per cluster and overall 100 clusters. Find the total capacity of the network.

Ans. : System capacity is given by number of clusters in a given area multiplied by number of channels in a cluster.

$$\text{No. Of clusters} = 00$$

$$\text{No of channels} = 20 \times 7 = 140$$

$$\therefore \quad \text{System capacity} = 100 * 140 = 14{,}000$$

Q.3 The minimum BW required for a PAM / TDM system is 300 kHz, and the number of channels transmitted is 20. Find the BW of each channel.

Ans. : BW required for each channel = 300kHz * 20 = 6000 kHz

Q.4 List advantages and disadvantages of cellular systems.

Ans. : Advantages of cellular systems :

1. Higher capacity
2. Less transmission power
3. Local interference only
4. Robustness

Disadvantages of cellular systems :

1. Infrastructure needed
2. Handover needed
3. Frequency planning

Q.5 What is an umbrella all pattern ?

Ans.. : In a biggest cellular region say 'macrocell', there may be many small cells called as microcells and this is known as umbrella cell pattern.

Q.6 What is the use of an umbrella cell ?

Ans. : In umbrella cell design, signal strength is more and coverage of radio signal is high. Macro cells are used for high speed traffic and micro cells are used for low speed traffic in the umbrella cellular pattern.

Q.7 What is microcell zone concept ? How is it used to improve capacity ?

Ans. : Microcells

- The microcells cover the interiors of streets and its antenna is located at the heights lower than the rooftop of the building.
- The microcell covers range of few hundreds of meter. It is used for personal communication systems.
- Large control base station is replaced by several lower power transmitters on the age of cell.
- The micro cell zone concept is associated with sharing the same radio equipment by different micro-cells. It results in decreasing of cluster size and therefore, increase in system capacity.
- The micro cell zone concept is used in practice to improve the capacity of cellular systems.
- Micro cell zone architecture minimizes the co-channel interference, improves system capacity, demands less handoffs, and the system is easy to implement.

Q.8 What is a mobile network ?

Ans. : Mobile Network -

- The mobile phone network is made of groups of cells, hexagonal in shape. Each cell site covers a specific geographic area of several kilometers in diameter.
- Each cell has its own base station linked to mobile telephone switching center.
- which connects the cell to public switched telephone network.

Q.9 Define cellular Technology.

Ans. : Cellular Technology

- Cellular technology involves the reuse of radio channels over and over again. This is done by using very low transmitter power.
- The radio waves can only travels a few kilometers and then be reused. One of the limitations of mobile phone is that you cannot make a call if the transmitter signals from the phone and carrier are not strong enough.

3.7 Multiple Choice Questions

Q.1 What are the main reasons for using cellular systems ?

a) To support many users, low power and localization.

b) Is profit maximization for service providers.

c) Are user localization and frequency reuse.

d) They are easy to use.

Q.2 Why are waves with a very low frequency not used for data transmission in computer networks ?

a They require large antennas, have lower bandwidth and are difficult to manage in cells and frequency reuse schemes

b They require small antennas and have higher bandwidth

c They do not penetrate material

d They can be easily shielded

Q.3 Which of the following is / are the main part(s) of basic cellular system ?

a A mobile unit

b A cell site

c A mobile telephone switching office

d All of the above

Q.4 State whether True or False.
i) The cells or subdivisions of a geographical area are always hexagonal.
ii) A land to Mobile call originates through the telephone exchange.

a True, False

b False, True

c False, False

d True, True

Answer Keys for Multiple Choice Questions :

Q.1	a	**Q.2**	a	**Q.3**	d	**Q.4**	b

❑❑❑

UNIT IV

4 Wireless System Planning

Syllabus

Link - Budget Analysis, Tele - traffic Theory, Tele - traffic System Model and Steady State Analysis.

Contents

4.1 Link Budget Analysis

- Path - loss models are commonly used to estimate link budgets, cell sizes and shapes, capacity, handoff criteria etc.
- The path - loss models used to estimate macroscopic or large scale variation of Received Signal Strength (RSS).

Path - loss = Loss in signal strength as a function of distance

- Path loss is :
  1. Terrain dependent (urban, rural, mountainous), ground reflection, diffraction, etc.
  2. Site dependent (antenna heights for example)
  3. Frequency dependent
  4. Line of sight or not
- By path - loss models, radio engineers calculate the coverage area of wireless base stations and Access Points (APs) also the maximum distance between two terminals in ad-hoc networks.
- Signal to Interference ratio (S/I) ratio should be computed and should be within acceptable level.
- Also, signal to noise (S/N) ratio should be computed and within acceptable level.
- C/N, or P_r/P_N ratio, where C stands for carrier power, P_r is received power, N stands for noise power. Noise power is due to thermal noise.
- A universal link budget for received power is :

$$P_r\,(\text{dBW}) = P_t\,(\text{dBW}) + \sum \text{dB Gains} - \sum \text{dB Losses}$$

- A universal link budget for S/N is :

$$\begin{aligned} S/N &= P_r\,(\text{dBW}) - P_N\,(\text{dBW}) \\ &= P_t\,(\text{dBW}) + \sum \text{dB Gains} - \sum \text{dB Losses} - P_N\,(\text{dBW}) \end{aligned}$$

4.2 Telecommunication Traffic **SPPU : April-16,18, Dec.-12, May-12,13,17,18**

- To design any industrial plant, a first decision must be made as to its size, in order to obtain the desired output. For example for an edible oil refinery this is the number of small barrels per day, for a machine shop, it is the number of piece parts per day.
- In case of telecommunication system it is the traffic to be handled, by using number of trunks to be provided in an exchange.

Trunk

- The term trunk is used to describe any entity that will carry one call.

Trunking

- The arrangement of trunks and switches within a telephone exchange is called trunking.

4.2.1 Unit of Traffic

- The traffic is defined as the average number of calls in progress.
- Traffic is a dimensionless quantity but a name given to unit of traffic is Erlang (abbreviation E) named after A. K. Erlang, a Danish pioneer of traffic theory.
- Traffic is sometimes expressed in terms of hundreds of call seconds per hour (CCS).

$$\boxed{1 \text{ erlang} = 36 \text{ CCS} = 3600 \text{ CS} = 60 \text{ CM}}$$

1. Busy hour

- The busy hour of an exchange is chosen 60 minute interval in which the telephone traffic is highest. The busy hour varies from exchange to exchange.

2. Busy Hour Calling Rate (BHCR)

- Busy Hour Calling Rate (BHCR) is defined as the average number of calls per subscriber during the busy hour.
- If the number of calls during busy hour be 6000 in an exchange serving 5000 subscribers, then BHCR is given as -

$$\boxed{\text{BHCR} = \frac{\text{Average busy hour calls}}{\text{Total number of subscribers}}}$$

$$\text{BHCR} = \frac{6000}{5000} = 1.2$$

$$\boxed{\text{Average Busy Hour Calls} = \text{BHCA} \times \text{CCR}}$$

3. Holding time

- The holding time of a call is the time from which the first selector is seized to set up to call up to the time when all the selectors are released.

or

- The duration of a call is called as holding time.

- From the definition of erlang, the traffic carried by a group of trunk is given by :

$$\boxed{A = \frac{Ch}{T}}$$

where,

A is traffic in erlangs.

C is average number of call arrivals during time 'T'.

h is average call holding time.

4. Occupancy

- A single trunk can carry only one call at a time i.e. $A \leq 1$. It means that the traffic is a fraction of an erlang equal to the average proportion of time for which the trunk is busy. This is referred as **occupancy** of the trunk.

5. Call Completion Rate (CCR)

- The Call Completion Rate (CCR) is defined as the ratio of the number of successfull calls to the number of call attempts.

$$\boxed{\text{CCR} = \frac{\text{Number of successfull calls}}{\text{Number of call attempts}}}$$

- CCR and BHCR are related by -

$$\boxed{\text{CCR} = \frac{\text{Average busy hour calls}}{\text{BHCA}}}$$

6. Busy Hour Call Attempts (BHCA) : The number of call attempts in the busy hour is called **Busy Hour Call Attempts (BHCA).**

Solved Examples

Example 4.2.1 *During the busy hour, a company makes 120 outgoing calls of average duration 2 minutes. It receives 200 incoming calls of average duration 3 minutes. Compute the following : 1) The outgoing traffic 2) The incoming traffic 3) The total traffic.*

Solution : Given data :

A company makes 120 outgoing calls in 2 minutes and company will receives 200 incoming calls in 3 minutes.

1) The outgoing traffic in erlang is

$$120 \times 2/60 = \mathbf{4\ E} \qquad \text{... Ans.}$$

2) The incoming traffic in erlang is

$$200 \times 3/60 = \mathbf{10\ E} \qquad \text{... Ans.}$$

3) The total traffic is

$$4 + 10 = \mathbf{14\ E} \quad \text{... Ans.}$$

Example 4.2.2 *During the busy hour, on average a customer with a single telephone line makes three calls and receives three calls.*

The average call duration is 2 minutes. What is the probability that a caller will find the line engaged ?

Solution : Given data :

A single telephone line makes 3 calls + 3 calls receives in time 2 minutes.

Probability of line engaged :

$$\text{Occupancy of line in erlang} = \frac{(3+3)\times 2}{60} = \mathbf{0.2\ E} \quad \text{... Ans.}$$

Example 4.2.3 *In a telephone system the average call duration is 2 minutes. A call has already lasted 4 minutes. What is the probability that*

1) The call will last atleast another 4 minutes ?

2) The call will end within the next 4 minutes ?

Solution : 1) Call will last atleast another 4 minutes call duration is 2 minutes.

$t = 4, \quad h = 2$

$$P(T \geq t) = e^{-t/h} = e^{-2} = \mathbf{0.135} \quad \text{... Ans.}$$

2)

$$P(T \leq t) = 1 - P(T \geq t)$$

$$= 1 - 0.135 = \mathbf{0.865} \quad \text{... Ans.}$$

Example 4.2.4 *An exchange serves 2000 subscribers. If the average BHCA is 10,000 and CCR is 60 %. Calculate the busy hour calling rate.*

Solution : Given :

$$N = 2000 \text{ subscriber}$$

$$BHCA = 10{,}000$$

$$CCR = 60\ \% = \frac{60}{100} = 0.6$$

But BHCR = ?

BHCR is given by :

$$BHCR = \frac{\text{Average busy hour calls}}{\text{Total number of subscriber}}$$

$$\text{Average busy hour calls} = \text{BHCA} \times \text{CCR}$$

$$= 10{,}000 \times 0.6 = 6000 \text{ calls}$$

$$\therefore \quad \text{BHCR} = \frac{6000}{2{,}000} = \mathbf{3}$$ **... Ans.**

Example 4.2.5 *In a group of 10 servers, each is occupied for 30 minutes in an observation interval of two hours. Calculate the traffic carried by the group.*

Solution : Given :

$$\text{Number of servers} = 10$$

$$\text{Traffic carried} = \text{A (To calculate)}$$

$$\text{Traffic carried per sources} = \frac{30}{120} = \frac{1}{4}$$

$$= \mathbf{0.25 \text{ erlang}}$$ **... Ans.**

$$\text{Traffic carried by the group} = 10 \times 0.25 = \mathbf{2.5 \text{ E}}$$ **... Ans.**

Example 4.2.6 *A group of 20 servers carry a traffic of 10 Erlangs. If the average duration of a call is three minutes, calculate the number of calls put through by a single server and the group as a whole is one hour period.* **SPPU : April-18, Marks 6**

Solution : Given :

$$\text{Number of server} = 20$$

$$A = 10 \text{ erlang}$$

$$t_n = 3 \text{ minutes}$$

$$\text{Number of calls/ server} = ?$$

$$\text{Number of calls/group} = ?$$

$$\text{Traffic per server} = \frac{10}{20} = 0.5 \text{ E}$$

i.e. a server is busy for 30 minute in 1 hour.

Number of calls through by one server $= \frac{30}{3} = 10$ calls.

Total number of calls put through by group $= 10 \times 20$

$$= \mathbf{200 \text{ calls.}}$$ **... Ans.**

Example 4.2.7 *A subscriber three phase calls of three minutes, four minutes and two minutes duration in a one hour period. Calculate the subscriber traffic in erlangs CCS and CM.*

Solution :

$$\text{Subscriber traffic in erlangs} = \frac{\text{Busy period}}{\text{Total period}}$$

$$= \frac{(3+4+2)\text{ min.}}{60\text{ min.}} = 0.15 \text{ E}$$

$$\text{Traffic is CCS} = 0.15 \times 36 = \mathbf{5.4\ CCS}$$

$$\text{CM} = 0.15 \times 60 = \mathbf{9\ CM}$$

Example 4.2.8 *Over a 20 minute observation interval, 40 subscriber initiate calls. Total duration of the calls is 4800 seconds. Calculate the load offered to the network by the subscriber and the average subscriber traffic.* **SPPU : April-18, Marks 4**

Solution : Number of subscriber = 40

$$\text{Mean arrival rate C} = \frac{\text{Number of subscribers}}{\text{Observation interval}}$$

$$= \frac{40}{20} = 2 \text{ calls/min}$$

$$\text{Mean holding time } t_h = \frac{\text{Total duration of calls}}{\text{Number of subscriber}}$$

$$= \frac{4800 / 60}{40} = 2 \text{ min/call}$$

$$\text{Offered traffic} = A \times C$$

$$= 2 \times 2 = \mathbf{4\ E} \qquad \text{... Ans.}$$

$$\text{Average subscriber traffic} = \frac{A}{N} = \frac{4}{40} = \mathbf{0.1\ E} \qquad \text{... Ans.}$$

Example 4.2.9 *During the busy hour a group of trunks are offered 120 calls having an message duration of 4 minutes. One of the calls fails to find a disengaged trunk. Find the traffic offered to the group and the traffic carried by the group.*

Solution : Given : C = 120

$$A = \frac{Ch}{T} = \frac{120 \times 4}{60} = \mathbf{8\ E} \qquad \text{... Ans.}$$

$$\text{120 calls} - \text{1 call} = \text{119 calls} = C$$

$$A = \frac{Ch}{T} = \frac{119 \times 4}{60} = \mathbf{7.93} \qquad \text{... Ans.}$$

Example 4.2.10 *A group of 20 trunks were found to have ten trunks engaged at 10 a.m., 15 at 10.10 a.m., 16 at 10.20 a.m. and 11 at 10.30 a.m. Calculate the average traffic intensity during this period.*

Solution : Given : 20 trunks.

At 10.00 a.m. - 10 trunked engaged

At 10.10 a.m. - 15 trunked engaged

At 10.20 a.m. - 16 trunked engaged

At 10.30 a.m. - 11 trunked engaged

$$\therefore \quad \text{Total} = \frac{10+15+16+11}{4} = 13 \text{ trunks}$$

$$A = \frac{13 \times 40}{60} = \mathbf{8.66\ E} \qquad \textbf{... Ans.}$$

Example 4.2.11 *During the busy hour, on average 30E is offered to a group of trunks. On average, the total period during which all trunks are busy is 12sec and 2 calls are lost. Find the average no. of calls carried by the group and the average call duration.*
[Show that the average no of calls offered to the group during a period equal to the average call duration is 30E.]

Solution : Given data : A = 30E, Total duration of period of congestion = 12 sec, Traffic lost = 2

$$\text{Time duration of period of congestion} = \text{Grade of service} \times 100$$

$$12 = B \times 3600$$

$$B = \frac{12}{3600}$$

$$B = 3.33 \times 10^{-3}$$

$$\boxed{B = 0.0033}$$

$$B = \frac{\text{Traffic lost}}{\text{Traffic offered}}$$

$$0.0033 = \frac{2}{\text{T.O.}}$$

$$\text{Traffic offered} = \frac{2}{0.0033} = 600$$

$$\text{Traffic carried} = \text{Traffic offered} - \text{Traffic lost}$$

$$= 600 - 2$$

$$\boxed{\text{Traffic carried} = 598}$$

$$A = \frac{Ch}{T}$$

$$30 = \frac{600 \times h}{60}$$

$$\boxed{h = 3 \text{ min}}$$

$$A = \frac{Ch}{T}$$

when $h = T$

$$\boxed{A = C = 30}$$ **... Ans.**

Consider same given data

$$\text{Busy period} = 12 \text{ sec}$$

$$\text{Total period} = 1 \text{ hours} = 60 \text{ sec}, \ A = 30 \text{ E}$$

$$B = \frac{\text{Busy period}}{\text{Total period}} = \frac{12}{3600} = \frac{1}{300}$$

$$\therefore \quad B = \frac{\text{Traffic lost}}{\text{Traffic offered}}$$

$$\frac{1}{300} = \frac{2}{x}$$

$$\therefore \quad x = 600$$

$$\text{Traffic carried} = \text{Traffic offered} - \text{Traffic lost}$$

$$= 600 - 2 = 598$$

$$A = \frac{Ch}{T}$$

$$30 = \frac{600 \text{ h}}{60}$$

$$h = 3 \text{ mins}$$

$$h = 180 \text{ seconds}$$

$$A = \frac{Ch}{T}$$

when $h = T$

$$\boxed{A = C = 30}$$ ... **Ans.**

Example 4.2.12 *During the busy hour, 1200 calls were offered to a group of trunks and six calls were lost. The average call duration was 3 minutes. Find*

1) The traffic offered *2) The traffic carried*

3) The traffic lost *4) The grade of service*

5) The total duration of percentage of congestion. **SPPU : May-17, 18, Marks 7**

Solution : $C = 1200$

$h = 3$ minutes

1) $$A = \frac{Ch}{T} = \frac{1200 \times 3}{60} = \mathbf{60\ E}$$ ... **Ans.**

2) Total calls – Lost calls

$1200 - 6 = 1194 = C$

$$A = Ch/T = \frac{1194 \times 3}{60} = \mathbf{59.7\ F}$$ ... **Ans.**

3) $C = 06$ lost calls

$$A = \frac{Ch}{T} = \frac{06 \times 03}{60} = \mathbf{0.3\ E}$$ ... **Ans.**

4) $$B = \frac{0.3}{60} = \mathbf{0.005}$$ ... **Ans.**

Or $$B = \frac{06}{1200} = \mathbf{0.005}$$ ... **Ans.**

5) $0.005 \times 3600 =$ **18 seconds** ... **Ans.**

Example 4.2.13 *During the busy hours, 1600 calls were offered to a group of trunks and 12 calls were lost. The average call duration was 6 minute. Find*

a) Traffic offered *b) Traffic carried*

c) Traffic lost *d) The grade of service*

e) The total duration of the periods of congestion.

Solution : Given : Busy hour = 60 min, 1600 calls, 12 calls lost, 6 min. is average call duration.

a) $$A = \frac{Ch}{T} = \frac{1600 \times 6}{60} = 160\ E$$

b) $$A = \frac{Ch}{T} = \frac{(1600 - 12) \times 6}{60} = 158.8\ E$$

c) $$A = \frac{Ch}{T} = \frac{12 \times 6}{60} = 1.2 \text{ E}$$

d) $$B = \frac{\text{Traffic lost}}{\text{Traffic carried}} = \frac{1.2}{158.8} = 0.00756$$

$\therefore$ Grade of service B = 0.00756

e) Total duration of periods of congestion is

$$= 0.00756 \times 3600$$

$$= \textbf{27.2 seconds}$$ **... Ans.**

Example 4.2.14 *A common control device in a telephone exchange is required to commence operation within a average period of 10 msecs after receiving a calling signal.*

a) If the device is held, on average for 50 msecs per call, how many calls can it handle per hour ?

b) If the device is required to handle 18,000 calls per hour, what is the maximum permissible average holding time ?

Solution : a) Number of calls handled per hour without delay

$$= \frac{3600}{50 \times 10^{-3}} = 72000$$

The delay is 10 ms = 10×10^{-3} sec

Number of calls handled per hour with delay $= \dfrac{3600}{(50+10) \times 10^{-3}} = 60{,}000$

The actual no. of calls handled per hour = 72,000 – 60,000

$$\boxed{\text{Calls} = 12000}$$ **... Ans.**

b) We know that

$$\overline{T} = \frac{A\,h}{1-A}$$

Given $$\overline{T} = 10 \times 10^{-3}$$

$$10 \times 10^{-3} = \frac{A\,h}{1-A}$$

$$\therefore \quad A = \frac{10 \times 10^{-3}}{(h + 10 \times 10^{-3})}$$

We also know

$$A = \frac{Ch}{T}$$

$$T = 3600 \text{ secs}$$

$$C = 18000 \text{ calls}$$

$$\therefore \quad \frac{10 \times 10^{-3}}{h + 10 \times 10^{-3}} = \frac{18000\,h}{3600} = 5h$$

$$(10 \times 10^{-3}) = (5h)\,(h + 10 \times 10^{-3})$$

$$1 = (5h)\,(100h + 1)$$

$$1 = 500h^2 + 5h$$

$$\boxed{h_1 = 40 \text{ msec}} \qquad \text{... Ans.}$$

$$h_2 = -\text{ve (not possible)}$$

Example 4.2.15 *During busy hour, 1000 calls were offered to a group of trunks and 5 calls were lost. The average call duration was 2 minutes.*

i) Find traffic offered ii) Traffic carried iii) Traffic lost iv) Grade of service

v) Total duration of periods of congestion. **SPPU : April-16, Marks 5**

Solution : T = 1 Hour = 60 minutes; C = 1000 Calls; Lost calls = 5 ; Average call duration = 2 min

a) Traffic offered A = Ch/T = 1000*2/60 = 33.34 E

b) Traffic carried A_0 = Ch/T = (1000 – 5)*2/60 = 33.16 E

c) Traffic lost = 5*2/60 = 0.16 E

d) GOS = 5/1000 = 0.005

e) Total duration = 0.005*3600 =18 Seconds

Example 4.2.16 *On an average, one call arrives every 5 seconds. During a period of 10 seconds. What is probability that*

i) No call arrivals ii) More than 1 call arrives **SPPU : April-16, Marks 6**

Solution : Given : $\lambda = \frac{1}{5}$ call per second

$$t = 10 \text{ seconds}$$

$$\therefore \quad \lambda \cdot t = \frac{1}{5} \times 10 = 2$$

i) No call arrives : k = 0

Probability of call arrivals is gieven by -

$$\boxed{P_k(t) = \frac{(\lambda t)^k e^{-\lambda t}}{k!}}$$

$$P_0(10) = \frac{(2)^0 \cdot e^{-2}}{0!} = 0.13533. \qquad \textbf{... Ans.}$$

ii) More than two call arrives : k > 2

$$P_{(k>2)}(10) = 1 - P_0(10) - P_1(10) - P_2(10)$$

$$P_{(k>2)}(10) = 1 - 0.13533 - 0.27066 - 0.27067$$

$$P_{(k>2)}(10) = 0.32334. \qquad \textbf{...Ans.}$$

4.2.2 Grade of Service and Blocking Probability

1. Grade of Service (GOS)
2. Blocking probability

- It is theoretically possible for every subscriber to make a call simultaneously in an telephone exchange.
- The situation is that all the trunks in a group of trunks are busy and so it can accept no further calls. This state is known as "congestion". In a message switched system, calls that arrive during congestion wait in a queue until an outgoing trunks becomes free. Thus they are delayed but not lost. Such systems are therefore called queuing systems or delay systems.

Traffic carried = Traffic offered – Traffic lost.

Traffic lost = Traffic offered – Traffic carried

Traffic lost = $A - A_0$

Grade of Service (G.O.S.)

- The proportion of calls that is lost or delayed due to congestion is a measure of the service provided. It is called the grade of service.

$$\text{G.O.S.} = \frac{\text{Number of calls lost}}{\text{Number of calls offered}}$$

$$\boxed{\text{G.O.S.} = \frac{\text{Traffic lost}}{\text{Traffic offered}} = \frac{A - A_0}{A}}$$

Blocking Probability

- The blocking probability (P_B) is defined as the probability that all the servers in a system are busy. Under such condition, no further traffic can be carried by the system.
- The probability that all the servers are busy may well represent the fraction of calls lost, which is GOS. But fundamental difference is that GOS is a measure from subscriber point of view whereas the blocking probability is a measure from network or switching system.

4.2.3 Comparison of GOS and Blocking Probability

Sr. No.	GOS	Blocking probability
1.	GOS is index of quality of service offered by network.	It is the probability that all the servers in a system are busy.
2.	GOS can be zero.	It is always non-zero value.
3.	GOS is measured from subscriber point of view.	Blocking probability is measured from switching point of view.
4.	GOS is referred as call congestion or loss probability.	Blocking probability is referred as time congestions.

Useful Results

Sr. No.	FORMULA FOR	FORMULA
1.	Traffic carried by a group of trunk is given.	$A = Ch/T$ where A = Traffic in Erlangs, C = Average number of call arrivals during time 'T' h = Average holding time
2.	Average Busy Hour Calls (ABHC)	ABHC = Busy hour calling attempts × Call completion rate
3.	Busy Hour Calling Rate (BHCR)	$\text{BHCR} = \frac{\text{Average busy hour calls}}{\text{Total number of subscribers}}$
4.	Grade of Service (B)	$B = \frac{\text{Traffic lost}}{\text{Traffic offered}}$
5.	Traffic Carried (TC)	TC = Traffic offered – Traffic lost
6.	Traffic Offered (TO)	TO = Traffic carried + Traffic lost

Distribution of Traffic Over Trunks of a Group with Sequential Search

- In many switching systems, trunks in a group are selected by means of sequential search. A call is not connected to trunk numbered 2 unless 1 is busy. It is not connected to 3 unless both the previous trunks 1 and 2 are busy and so on. Calls finding the last choice trunks busy are lost. As a result, the 1^{st} trunk has a very high occupancy and traffic carried by subsequent trunks is less. The last choice trunk is very lightly loaded. This behaviour is illustrated in the following Fig. 4.2.1.

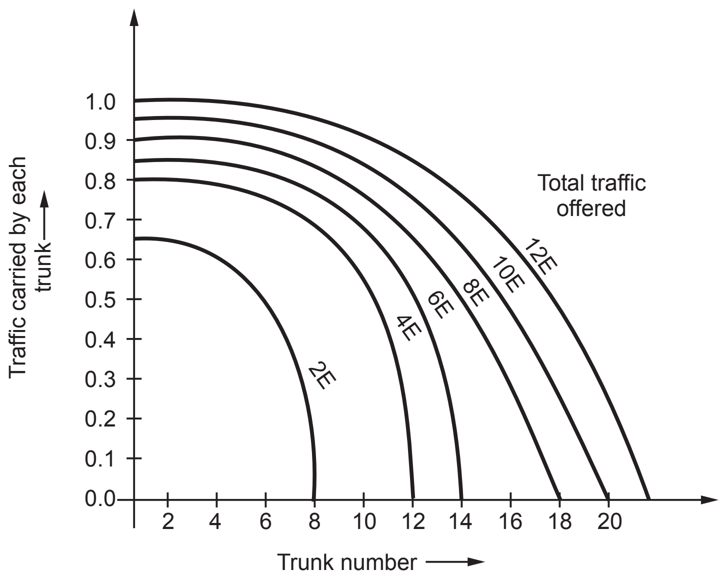


Fig. 4.2.1 Traffic distribution of a group with sequential search

- The performance of such an arrangement can be analyzed as follows :

Let traffic 'A' E be offered to a group of trunks. Consider Erlang's lost call formula.

$$B = E_{IN}(A) = \frac{\frac{A^N}{N!}}{\sum_{K=0}^{N} \frac{A^K}{K!}} \qquad \text{... (4.2.1)}$$

The grade of service for a single trunk group is put N = 1, in equation (4.2.1).

$$\Rightarrow \qquad E_{1,1}(A) = \frac{\frac{A^1}{1!}}{\sum_{K=0}^{1} \frac{A^K}{K!}} = \frac{A}{\frac{A^0}{0!}+\frac{A^1}{1!}} = \frac{A}{1+\frac{A^1}{1!}}$$

$$\Rightarrow \qquad E_{1,1}(A) = \frac{A}{1+A}$$

Traffic overflowing from 1st trunk to 2nd is ;

$$A\ E_{1,1}\ (A) = \frac{A^2}{1+A}$$

$\therefore$ Traffic carried by 1st trunk = Traffic offered – Traffic lost

$$= A - \frac{A^2}{1+A}$$

$$= \frac{A + A^2 - A}{1+A} = \frac{A}{1+A}$$

In general,

Traffic carried by K^{th} trunk = Traffic lost from group of 1st (K – 1) trunks – traffic lost from group of 1st K trunks.

$\therefore$ Traffic offered = $A\ \{E_{1,(K-1)}(A) - E_{1,K}(A)\}$

4.2.4 Mathematical Model

- A mathematical model is used to analyze the teletraffic problems.
- Two assumptions for mathematical model are :
  1. Pure - chance traffic
  2. Statistical equilibrium

1. Pure - chance traffic

- It means that call arrivals and call terminations are independent random events.
- The assumption of random call arrivals and terminations leads to the following results.

i) The number of call arrivals in a given time has a poisson-distribution :

$$\boxed{P(x) = \frac{\mu^x}{x!} \cdot e^{-\mu}}$$

where,

x represents the number of call arrivals in time T.

μ represents the mean number of call arrivals in time T.

ii) The intervals between call arrivals (T) have a negative exponential distribution

$$\boxed{P(T \geq t) = e^{-t/\overline{T}}}$$

where,

$\overline{T}$ is mean interval between call arrivals

iii) The call durations (T) have a negative exponential distribution.

$$\boxed{P(T \geq t) = e^{-t/h}}$$

where, h represents the mean call duration (holding time).

2. Statistical equilibrium

- It means the generation of traffic is a stationary random process i.e. probabilities 'do not change during the period being considered. The mean number of calls in a progress remains constant.

4.2.4.1 Markov Chain

- In a group of N trunks, the number of calls in progress varies randomly between 0 and N. Therefore, it has N + 1 states and its behavior depends on the probability of change from each state to one above and below it. This process is called simple **Markov chain.**

  where,

  P(j) is probability of state j

  P(k) is probability of next higher state

  P(0), P(1), ... P(N) are state probabilities

  $P_{j,\,k}$, $P_{k,\,j}$ .... are transition probabilities

- Under static equilibrium the transition probabilities do not change and this process is called **regular Markov chain.**
- The probability of call arriving, if there are x calls in progress is given by -

$$\boxed{P(x) = \frac{A^{x}}{x\,!} \cdot P(0)}$$

Here,

A is mean number of calls arriving during the average holding time h is C = A.

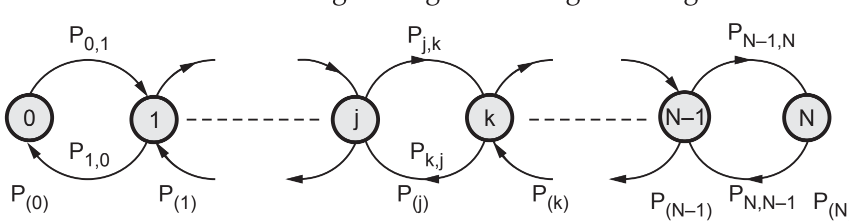


Fig. 4.2.2 Simple Markove chain with N trunks

Solved Examples

Example 4.2.17 *a) By using traffic capacity table for full availability groups find number of trunks required for to give grade of service of 0.001 for the following load offered.*

b) *Grade of service for 20 % overload 0.02.*

1E, 2E, 4E, 10E, 40E, 80E

in each case determine the occupancy and number of trunks required per Erlang.

Solution : a) For **0.001** from traffic capacity table for full availability of groups. We have

0.001	Number of trunks
1E	5
2E	7
4E	10
10E	18
40E	59

b) For **0.02** with 20 %

For lost call 0.02	Number of trunks from table	For 20 %	Total number of trunks
1E	3	0.6	$3 + 0.6 = 3.6 \cong 4$
2E	5	1	$5 + 1 = 6$
4E	8	1.6	$8 + 1.6 = 9.6 \cong 10$
10E	16	3.2	$16 + 3.2 = 19.2 \cong 19$
40E	49	9.8	$49 + 9.8 = 58.8 \cong 59$
80E	91	18.2	$91 + 18.2 = 109.2 \cong 109$

Example 4.2.18 *A telecommunication company dimensions is route by the following criteria.*

i) Grade of service for Normal load 0.005.

ii) Grade of service for 10 % overload 0.02.

iii) Grade of service for 30 % overload 0.001

iv) Grade of service for 20 % overload 0.01.

From the Traffic-capacity table for full availability groups find the number of trunks required for 15E, 55E, 65E.

Solution : i) For normal load with 0.005

From table for 0.005 lost call	Number of trunks
15E	25
55E	71
65E	82

ii) Grade of service for 10 % overload 0.02

From table for 0.02 lost call	Number of trunks	For 10 %	Total Number of trunks
15E	22	2.2	22 + 2.2 = 24.2 ≅ 24
55E	65	6.5	65 + 6.5 = 71.5 ≅ 72
65E	76	7.6	76 + 7.6 = 83.6 ≅ 84

iii) Grade of service for 30 % over load 0.001

From table for 0.001 lost call	Number of trunks	For 10 %	Total Number of trunks
15E	27	8.1	27 + 8.1 = 35.1 ≅ 35
55E	76	22.8	76 + 22.8 = 98.8 ≅ 99
65E	88	26.4	88 + 26.4 = 114.4 ≅ 115

iv) Grade of service for 20 % overload 0.01

From table for 0.01 lost call	Number of trunks	For 10 %	Total Number of trunks
15E	23	4.6	23 + 4.6 = 27.6 ≅ 28
55E	68	13.6	68 + 13.6 = 81.6 ≅ 82
65E	79	15.8	79 + 15.8 = 94.8 ≅ 95

Example 4.2.19 *Use traffic capacity table for full availability groups find the number of trunks required to give grade of service of 0.01 for the following loads of offered traffic*

1E, 2E, 4E, 10E , 40E, 80E

in each case determine the occupancy and the number of trunks required per Erlang.

Solution : Given 1E, 2E, 4E, 10E, 40E, 80E.

From traffic capacity table for full availability we have for 0.01.

For lost call in 0.01	Number of trunks from table
1E	4
2E	6
4E	9
10E	17
40E	52
80E	95

Example 4.2.20 *A telecommunications company dimensions routes by the following criteria. i) Grade of service for normal load 0.005 ii) Grade of service for 10 % overload 0.02 from traffic-capacity table for full availability groups. Find the number of trunks required for 10E, 40E, 50E, 60E, 79E. In each case state which is the determining criteria.*

Solution : Given 10E, 40E, 50E, 60E, 79E.

i) Grade of service for normal load 0.005.

For lost call in 0.005	Number of trunks required
10E	18
40E	54
50E	66
60E	77
79E	98

ii) Grade of service for 10 % overload 0.02.

For lost call in 0.02	Number of trunks	for 10 %	Total number of trunks required
10E	16	1.6	$16 + 1.6 = 17.6 \cong 18$
40E	49	4.9	$49 + 4.9 = 53.9 \cong 54$
50E	60	6.0	$60 + 6.0 = 66$
60E	71	7.1	$71 + 7.1 = 78.1 \cong 78$
79E	91	9.1	$91 + 9.1 = 100.1 \cong 100$

University Questions

1. *Define and explain*

   i) Grade of service. *ii) Average holding time.*

   iii) Call completion rate. *iv) Erlang and CCS.* **SPPU : May-12, Marks 10**

2. *During the busy hour 1350 calls were offered to a group of trunks and 10 calls were lost. The average call duration was 210 seconds.*
   Find i) The traffic offered. ii) The traffic carried.
   iii) The traffic lost. iv) The grade of service.
   v) The total duration of period of congestion. **SPPU : May-12, Marks 10**
3. *Define and explain*
   - Busy hour call attempts/rate
   - Average holding time
   - Call completion rate **SPPU : Dec.-12, Marks 4**
4. *Define grade of service and blocking probability.* **SPPU : Dec.-12, Marks 4**
5. *Define and explain following terms*
   i) BHCA ii) CCR iii) Erlang iv) Grade of Service **SPPU : May-13, Marks 8**
6. *Explain the assumptions in :*
   i) Pure chance traffic ii) Statistical equillibrium **SPPU : May-17, Marks 5**
7. *Define and explain*
   i) Grade of service ii) Blocking probability iii) Traffic intensity
   SPPU ; May-18, Marks 5

4.3 Lost Call System

SPPU : May-12, April-16,17, Dec.-12

- Erlang determine the grade of service of a lost call system having N trunks, when offered traffic A. His solution depends on the following assumptions.
  1. Pure chance traffic
  2. Statistical equilibrium
  3. Full availability
  4. Calls which encounter congestion are lost.
- The assumption of perchance traffic, that call arrival and call terminates are independent random events.

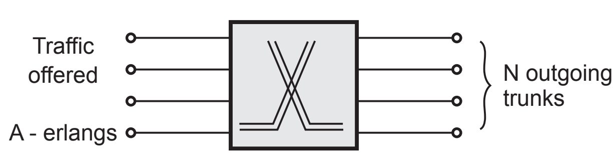


Fig. 4.3.1 Lost call system

- The statistical equilibrium implies that probabilities donot change.
- Full availability means that the every call that arrives can be connected to any outgoing trunk which is free.

- The lost call assumption implies that any attempted call which encounters congestion is immediately cleared from the system.

Here it will simply be assumed that the traffic offered is the total arising from all successful and unsuccessful calls.

If there are "x" calls in progress,

then equation

$$P(x) = \frac{A^x}{x!} P(0) \quad 0 \le x \le N \qquad \text{... (4.3.1)}$$

Taking summation on both side

$$\sum_{x=0}^{N} P(x) = 1 = \sum_{x=0}^{N} \frac{A^x}{x!} \cdot P(0)$$

Hence $$P(0) = \frac{1}{\sum_{x=0}^{N} \frac{A^x}{x!}}$$

Substituting in equation (4.3.1),

$$P(x) = \frac{A^x / x!}{\sum_{K=0}^{N} \frac{A^K}{K!}} \qquad \text{... (4.3.2)}$$

- This is the first Erlang distribution. The probability of a lost call, which is the grade of service B. This is given by the symbol $E_{1N}(A)$ which denotes the loss probability for a full availability group of N trunks offered traffic Erlang.

$$\boxed{B = E_{1N}(A) = \frac{A^N / N!}{\sum_{K=0}^{N} \frac{A^K}{K!}}} \qquad \text{... (4.3.3)}$$

Equation (4.3.3) is Erlangs lost call formula.

Replace $N = N - 1$

$$E_{1(N-1)}^{(A)} = \frac{A^{N-1} / (N-1)!}{\sum_{K=0}^{N-1} \frac{A^K}{K!}} \qquad \text{... (4.3.4)}$$

$$E_{1(N-1)}^{(A)} * \sum_{K=0}^{N-1} \frac{A^K}{K!} = \frac{A^{N-1}}{(N-1)!}$$

$$\sum_{K=0}^{N-1} \frac{A^K}{K!} = \frac{\frac{A^{N-1}}{(N-1)!}}{E_{1(N-1)}(A)} \qquad \text{... (4.3.4 (a))}$$

Consider

$$\sum_{K=0}^{N} \frac{A^K}{K!} = \underbrace{\frac{A^0}{0!} + \frac{A^1}{1!} + \frac{A^2}{2!} + \ldots \frac{A^{N-1}}{(N-1)!}} + \frac{A^N}{N!}$$

$$\sum_{K=0}^{N} \frac{A^K}{K!} = \sum_{K=0}^{N-1} \frac{A^K}{K!} + \frac{A^N}{N!}$$

$$\sum_{K=0}^{N} \frac{A^K}{K!} - \frac{A^N}{N!} = \sum_{K=0}^{N-1} \frac{A^K}{K!} \qquad \text{... (4.3.5)}$$

Put equation (4.3.5) in equation (4.3.4 (a)).

$$\sum_{K=0}^{N} \frac{A^K}{K!} - \frac{A^N}{N!} = \frac{\frac{A^{N-1}}{(N-1)!}}{E_{1(N-1)}(A)}$$

$$\sum_{K=0}^{N} \frac{A^K}{K!} = \frac{\frac{A^{N-1}}{(N-1)!}}{E_{1(N-1)}(A)} + \frac{A^N}{N!} \qquad \text{... (4.3.6)}$$

Put equation (4.3.6) in equation (4.3.3)

$$E_{1N}(A) = \frac{A^N / N!}{\frac{\frac{A^{N-1}}{(N-1)!}}{E_{1(N-1)}(A)} + \frac{A^N}{N!}}$$

Divide numerator and denominator by $A^N / N!$

$$= \frac{\left(A^N / N!\right) / \left(A^N / N!\right)}{\frac{\frac{A^{N-1} / (N-1)!}{E_{1(N-1)}(A)}}{A^N / N!} + \frac{A^N}{N!} / \left(A^N / N!\right)}$$

$$= \frac{1}{1 + \frac{A^{N-1}}{(N-1)!} \times \frac{1}{E_{1(N-1)}(A)} \times \frac{N!}{A^N}}$$

$$= \frac{1}{1+\frac{A^{N} A^{-1}}{(N-1)\ !} \times \frac{1}{E_{1(N-1)}(A)} \times \frac{N!}{A^{N}}}$$

$$= \frac{1}{1+\frac{N}{A\,E_{1(N-1)}(A)}}$$

$$E_{1N}(A) = \frac{1}{\frac{A\,E_{1(N-1)}(A)+N}{A\,E_{1,\,(N-1)}(A)}}$$

$$\boxed{E_{1,\,N}(A) = \frac{A\,E_{1,\,N-1}(A)}{N+AE_{1,\,N-1}(A)}}$$

Example 4.3.1 *A group of 20 trunks provides a grade of service of 0.015 when offered 12 E of traffic.*

i) How much is the grade of service improved if one extra trunk is added to the group ?

ii) How much does the grade of service deteriorates if one trunk is out of service ?

SPPU : May-12, Marks 8

Solution :

Traffic A = 12 E

Group of trunk N = 20

Grade of service $E_{1,\,20-1}(12) = 0.015$

$$\boxed{E_{1,\,N}(A) = \frac{A\,E_{1,\,N-1}(A)}{N+AE_{1,\,N-1}(A)}}$$

i) GOS if one extra trunk is added i.e. N = 21.

$$E_{1,\,21}(12) = \frac{12\cdot E_{1,\,20}(12)}{21+12\cdot E_{1,\,20}(12)}$$

$$= \frac{12\times 0.015}{21+12\times 0.015} = \frac{0.18}{21.18}$$

$$= \mathbf{0.00843}$$ **... Ans.**

ii) GOS if one trunk is out of order i.e. N = 19.

$$E_{1,\,20}(12) = \frac{12\,E_{1,\,19}(12)}{20 + 12\,E_{1,\,19}(12)}$$

$$0.015 = \frac{12E_{1,\,19}(12)}{20 + 12E_{1,\,19}(12)}$$

$$0.3 + 0.18\,E_{1,19}(12) = 12\,E_{1,19}(12)$$

$$E_{1,19}(12) = \frac{0.3}{11.82}$$

$$E_{1,19}(12) = \mathbf{0.0253}$$ **... Ans.**

Example 4.3.2 *A group of 5 trunks is offered 2 Erlang of traffic. Find*

i) Grade of service

ii) Probability that only one trunk is busy.

iii) Probability that only one trunk is free.

iv) Probability that at least one trunk is free. **SPPU : April-17, Marks 5**

Solution : For a lost-call system having N trunks, when offered traffic A, the first Erlang distribution is given by :

$$P(x) = \frac{\dfrac{A^x}{x!}}{\displaystyle\sum_{k=0}^{N} \frac{A^k}{k!}}$$

Where, x is the number of occupied trunks is the probability of, and P(x) occupied trunks

i) The grade of service

$$B = P(x = N) = \frac{\dfrac{A^x}{x!}}{\displaystyle\sum_{k=0}^{N} \frac{A^k}{k!}} = \frac{\dfrac{2^5}{5!}}{\displaystyle\sum_{k=0}^{5} \frac{2^k}{k}} = \frac{0.2667}{7.2667} = 0.037$$

ii) The probability that only one trunk is busy

$$P(1) = \frac{\dfrac{2^1}{1!}}{\displaystyle\sum_{k=0}^{N} \frac{A^k}{k!}} = \frac{2}{7.2667} = 0.275$$

iii) The probabiliy that only one trunk is free

$$P(4) = \frac{\frac{2^4}{4!}}{\sum_{k=0}^{N} \frac{A^k}{k!}} = \frac{16/24}{7.2667} = 0.0917$$

iv) The probability that atleast one trunk is free

$$P(x<5) = 1 - P(5) = 1 - B = 1 - 0.037 = 0.963$$

4.3.1 GoS and Blocking Probability in Lost Call System

Grade of Service and Blocking Probability in lost call system

- The grade of service (B) in a lost-call system is defined as :

$$B = \text{Number of lost calls / Number of offered calls}$$

$$= \text{Lost traffic / Offered traffic}$$

- Two formulae are used for calculating the blocking probability : the Erlang-B and Erlang-C. The choise of formula is dependent upon the method of handling of customers when all resources are busy.
  1. **Erlang - B :** Used for lost-call systems whereby calls are lost should all resources be busy.
  2. **Erlang - C :** Used for queueing systems whereby calls are queued should all resources be busy.
- The Erlang - B formula is expressed as :

$$P_B = \frac{\frac{A^N}{N!}}{\sum_{i=0}^{N} \frac{A^i}{i!}}$$

- The Erlang-C formula is expressed as :

$$P_c = \frac{\frac{A^N}{N!}\frac{N}{N-A}}{\sum_{i=0}^{N} \frac{A^i}{i!} + \frac{A^N}{N!}\frac{N}{N-A}}$$

Significance of Grade of Service and Blocking Probability

- The grade of service is the blocking probability. A higher grade of service implies high probability of loss during the busy hour.
- Blocking probability is the chance that a customer will be denied service due to lack of resources. A blocking probability of 0.01 means 1 % of customers will be denied service.

University Questions

1. *Explain the design procedure for 'N' by 'N' switch with two stages and number of links = N. What is the total number of cross-points required ?* **SPPU : Dec.-12, Marks 8**
2. *Derive the first Erlang Distribution for Lost call systems.* **SPPU : April-16, Marks 5**
3. *Define Grade of service and blocking probability for lost call system and explain its significance.* **SPPU : April-16, Marks 6**

4.4 Traffic Performance

SPPU : May-13, April-17

- Traffic A increases, the number of trunks N must be increased to provide a given grade of service.
- However for the same trunk occupancy the probability of finding all trunks busy is less for a large group of trunks than for a small group.
- Thus, for a given grade of service a large group of trunks can have a higher occupancy than a small one, i.e. the large group is more efficient.
- Fig. 4.4.1 shows for a grade of service of 0.002. For example 2E of traffic requires seven trunks and their occupancy is 0.275E. However 20E requires 32 trunks and their occupancy is 0.61E.

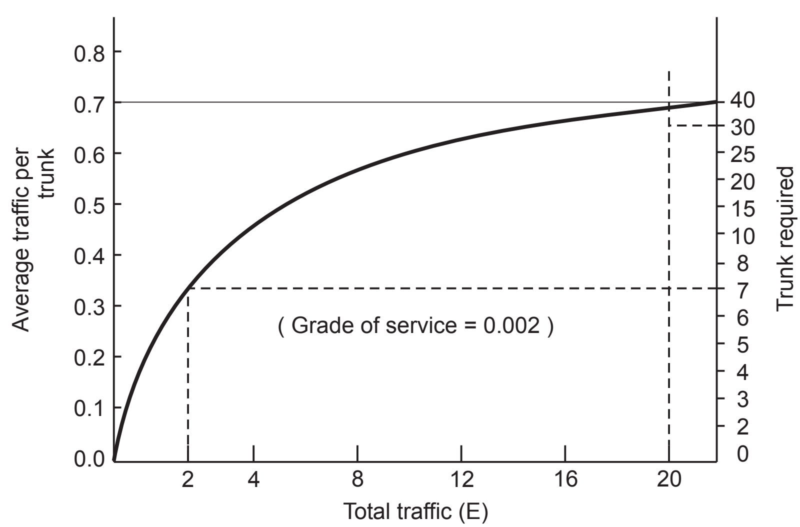


Fig. 4.4.1 Traffic performance

Important Useful Results

Parameter	When there is delay	Averaged overall time
Mean number of calls in the system.	$\bar{x}' = \frac{A}{N-A} + N$	$\bar{x}' = \frac{A}{N-A} E_{2N}(A) + A$
Mean length of queue.	$\bar{q}' = \bar{x}' = N = \frac{A}{N-A}$	$\bar{q}' P_D = \frac{A}{N-A} E_{2N}(A) = \bar{q}$
Mean delay time.	$\bar{T}' = h/(N-A)$	$\bar{T}' = \frac{E_{2N}(A)h}{N-A}$
Distribution of delay.	$P(T_D \geq t) = e^{\frac{-t}{\bar{T}'}}$	$P(T_D \geq t) = E_{2N}(A)\, e^{\frac{-t}{\bar{T}'}}$
$P_D = \frac{A}{1-A}$	$P_{(D)} = \frac{A^2}{1-A}$	$P(x) = \frac{A\,h}{1-A}$

4.4.1 Traffic Tables

Number of trunks	1 lost call in			
	50 (0.02)	100 (0.01)	200 (0.005)	1000 (0.001)
	E	E	E	E
1	0.020	0.010	0.005	0.001
2	0.22	0.15	0.105	0.046
3	0.60	0.45	0.35	0.19
4	1.1	0.9	0.7	0.44
5	1.7	1.4	1.1	0.8
6	2.3	1.9	1.6	1.1
7	2.9	2.5	2.2	1.6
8	3.6	3.2	2.7	2.1
9	4.3	3.8	3.3	2.6
10	5.1	4.5	4.0	3.1
11	5.8	5.2	4.6	3.6
12	6.6	5.9	5.3	4.2
13	7.4	6.6	6.0	4.8

14	8.2	7.4	6.6	5.4
15	9.0	8.1	7.4	6.1
16	9.8	8.9	8.1	6.7
17	10.7	9.6	8.8	7.4
18	11.5	10.4	9.6	8.0
19	12.3	11.2	10.3	8.7
20	13.2	12.0	11.1	9.4
21	14.0	12.8	11.9	10.1
22	14.9	13.7	12.6	10.8
23	15.7	14.5	13.4	11.5
24	16.6	15.3	14.2	12.2
25	17.5	16.1	15.0	13.0
26	18.4	16.9	15.8	13.7
27	19.3	17.7	16.6	14.4
28	20.2	18.6	17.4	15.2
29	21.1	19.5	18.2	15.9
30	22.0	20.4	19.0	16.7
31	22.9	21.2	19.8	17.4
32	23.8	22.1	20.6	18.2
33	24.7	23.0	21.4	18.9
34	25.6	23.8	22.3	19.7
35	26.5	24.6	23.1	20.5
36	27.4	25.5	23.9	21.3
37	28.3	26.4	24.8	22.1
38	29.3	27.3	25.6	22.9
39	30.1	28.2	26.5	23.7
40	31.0	29.0	27.3	24.5
41	32.0	29.9	28.2	25.3
42	32.9	30.8	29.0	26.1
43	33.8	31.7	29.9	26.9
44	34.7	32.6	30.8	27.7
45	35.6	33.4	31.6	28.5

46	36.6	34.3	32.5	29.3
47	37.5	35.2	33.3	30.1
48	38.4	36.1	34.2	30.9
49	39.4	37.0	35.1	31.7
50	40.3	37.9	35.9	32.5
51	41.2	38.8	36.8	33.4
52	42.1	39.7	37.6	34.2
53	43.1	40.6	38.5	35.0
54	44.0	41.5	39.4	35.8
55	45.0	42.4	40.3	36.7
56	45.9	43.3	41.2	37.5
57	46.9	44.2	42.1	38.3
58	47.8	45.1	43.0	39.1
59	48.7	46.0	43.9	40.0
60	49.7	46.9	44.7	40.8
61	50.6	47.9	45.6	41.6
62	51.6	48.8	46.5	42.5
63	52.5	49.7	47.4	43.4
64	53.4	50.6	48.3	44.1
65	54.4	51.5	49.2	45.0
66	55.3	52.4	50.1	45.8
67	56.3	53.3	51.0	46.6
68	57.2	54.2	51.9	47.5
69	58.2	55.1	52.8	48.3
70	59.1	56.0	53.7	49.2
71	60.1	57.0	54.6	50.1
72	61.0	58.0	55.5	50.9
73	62.0	58.9	56.4	51.8
74	62.9	59.8	57.3	52.6
75	63.9	60.7	58.2	53.5
76	64.8	61.7	59.1	54.3
77	65.8	62.6	60.0	55.2

78	66.7	63.6	60.9	56.1
79	67.7	64.5	61.8	56.1
80	68.6	65.4	62.7	58.7
81	69.6	66.3	63.6	58.7
82	70.5	67.2	64.5	59.5
83	71.5	68.1	65.4	60.4
84	72.4	69.1	66.3	61.3
85	73.4	70.1	67.2	62.1
86	74.4	71.0	68.1	63.0
87	75.4	71.9	69.0	63.9
88	76.3	72.8	69.9	64.8
89	77.2	73.7	70.8	65.6
90	78.2	74.7	71.8	66.6
91	79.2	75.6	72.7	67.4
92	80.1	76.6	73.6	68.3
93	81.0	77.5	74.3	69.1
94	81.9	78.4	75.4	70.0
95	82.9	79.3	76.3	70.9
96	83.9	80.3	77.2	71.9
97	84.8	81.2	78.2	72.6
98	85.7	82.2	79.1	73.5
99	86.7	83.2	80.0	74.4
100	87.6	84.0	80.9	75.3

Example 4.4.1 *The traffic statistics of a company using EPABX indicates that 200 outgoing calls are initiated every hour during work hours. Equal number of calls comes in. On averages each call lasts for 180 seconds. If GOS required is 0.01, determine number of lines required between EPABX and main exchange.* **SPPU : May-13, Marks 10**

Solution : GOS = 0.01

$$\text{Outgoing traffic A} = \frac{Ch}{T}$$

$$A = 200 \times \frac{1}{180} = 1.11 \text{ E}$$

$$\text{Incoming traffic A} = 200 \times \frac{1}{180} = 1.11 \text{ E}$$

From traffic capacity table for GOS = 0.01; 1.11 E of outgoing traffic needs 5 lines and 1.11 E of incoming traffic needs 5 lines.

1) If incoming and outgoing calls are handled on separate group of lines, then total number of lines required is **10**.
2) If a common group of lines is used for both incoming and outgoing calls, then total traffic is 1.11 + 1.11 = 2.22 E.

Total number of lines required is **7.**

University Question

1. *Write a note on : Traffic performance* **SPPU : April-17, Marks 2**

4.5 Loss System in Tandem

SPPU : April-17

- A complete connection consists of several links in tandem. If a connection consisting of two links with grades of service B_1 and B_2. The connection has offered traffic of A erlangs.

$$\text{Traffic offered to link-2} = A(1-B_1)$$

$$\text{Traffic reaching destination} = A(1-B_1)(1-B_2)$$

$$= A(1+B_1B_2-B_1-B_2)$$

- The overall grade of service (i.e. GoS for entire connection) is expressed as :

$$B = B_1 + B_2 - B_1B_2$$

- Usually B_1B_2 << 1; hence can be neglected, therefore overall grade of service is,

$$B = B_1 + B_2$$

- Generalized formula for grade of service for n-link connection is written as :

$$\boxed{B = \sum_{k=1}^{n} B_k}$$

University Question

1. *Write a note on : Loss system in Tandem* **SPPU : April-17, Marks 2**

4.6 Queuing Systems

SPPU : May-13,16, April-17, May-18, Dec.-17

4.6.1 Second Erlang Distribution

- A queueing system with traffic offered A and N trunks is shown in Fig. 4.6.1.

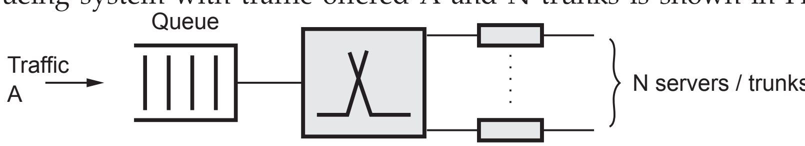


Fig. 4.6.1 Queueing system

- The probability of encountering delay is based on assumptions :
  1. Pure - chance traffic
  2. Statistical equilibrium
  3. Full availability

  } Common assumptions for lost call system

  4. Calls which encounter congestion enter a queue and are stored there until a server becomes free.
- The queueing system operation with all above assumptions is known as M|M|N system.
- Under static equilibrium condition $A \leq N$. But if $A \geq N$; i.e. traffic offered is higher than number of trunks then length of queue increases towards infinity.

Condition 1 : $x \leq N$

- If total number of calls (x) in system is less than number of trunks then calls will be served without delay i.e. there is no queue.
- The system behaves as lost call system without congestion. Then

$$\boxed{P(x) = \frac{A^x}{x!} P(0)}$$

Condition 2 : $x \geq N$

- The number of calls arriving is much greater than number of trunks. The incoming calls encounter delay as all the servers are busy. Under this condition,

$$\boxed{P(x) = \frac{A^x}{N^{x-N} N!} P(0)}$$ or

$$\boxed{P(x) = \frac{N^N}{N!}\left(\frac{A}{N}\right)^x P(0)}$$

- The state probability at trunk 0 is given by :

$$\boxed{P(0) = \left[\frac{N A^N}{N!\,(N-A)} + \sum_{x=0}^{N-1} \frac{A^x}{x!}\right]^{-1}}$$

This is second erlang distribution.

4.6.2 Probability of Delay

- When incoming calls x are much more than available trunks N i.e. $x \geq N$. The delay occurs.
- The probability that there are at least Z calls in system is expressed as :

$$P(x \geq Z) = \sum_{x=z}^{\infty} P(x)$$

$$= \frac{N^N}{N!} P(0) \sum_{x=z}^{\infty} \left(\frac{A}{N}\right)^x$$

$$= \frac{N^N}{N!} P(0) \left(\frac{A}{N}\right)^z \sum_{k=0}^{\infty} \left(\frac{A}{N}\right)^k \quad \because k = x - N$$

$$P(x \geq Z) = \frac{N^N}{N!} \left(\frac{A}{N}\right)^z P(0) \left[1 - \frac{A}{N}\right]^{-1}$$

$$P(x \geq Z) = \frac{N^N}{N!} \left(\frac{A}{N}\right)^z \left(\frac{N}{N-A}\right) P(0)$$

The probability of delay, $P_D = P(x \geq N)$

$$\boxed{P_D = \frac{A^N}{N!} \cdot \frac{N}{N-A} P(0)}$$

$$= E_{2,N}(A)$$

- The above expression is the probability of delay for a system with N servers and offered traffic A erlangs.
- The formula for $E_{2,N}(A)$ is also called as **Erlang delay formula.**

4.6.3 Finite Queue Capacity

- Practically it is not possible to hold infinite queue. After certain length the incoming calls gets lost.

- Let a queue can hold up to Q calls, then $x \leq G + N$ then,

$$\frac{1}{P(0)} = \sum_{x=0}^{N-1} \frac{A^x}{x!} + \frac{N}{N!}\left(\frac{A}{N}\right)^N \sum_{k=0}^{Q}\left(\frac{A}{N}\right)^k$$

- The loss probability can be estimated by assuming queue capacity is infinite i.e. $P(x \geq Q+N)$.

$$P(x \geq Q+N) = \frac{N^N}{N!}\left(\frac{A}{N}\right)^{Q+N} \frac{N}{N-A} P(0)$$

$$= \left(\frac{A}{N}\right)^Q \cdot P_D$$

4.6.4 System with Single Server

- When system with single server is used, then the probability of it being busy is its occupancy (A). This is also equal to probability of delay $E_{2,\,1}(A) = A$.
- The expressions can be simplified as :

$$\text{Probability of delay : } P_D = \frac{A}{(1-A)}$$

$$\text{Probability of } P(0) = \frac{A^2}{1-A}$$

$$\text{Probability of } P(x) = \frac{Ah}{(1-A)}$$

$$\text{Probability of } P(x \geq Z) = A^Z$$

4.6.5 Queues in Tandem

- When more than one queueing systems are connected in tandem, the delays are cumulative. The queues can be considered as independent for calculating their delays.
- The delay probability and mean delay for tandem queues are sums of these for the individual stages.
- The probability distribution of sum of several random variables is computed by convolution of the separate distributions.

4.6.6 Applications of Delay Formulae

- A message switch and packet switch are examples of queueing system. In these systems, if outgoing trunks are busy, the messages or packets are hold in queue until outgoing trunk becomes free.
- A system should be designed to meet a specified delay probability or a specified mean delay.
- A telephone exchange is a circuit switched network and its switching network is a lost call system. Common controls in an exchange makes queueing system.
- In a Stored Program Controlled (SPC) system, a central processor performs various tasks. These wait in queue untill processor completes previous tasks. Therefore, common controls are designed to meet delay criteria.

University Questions

1. *State and explain Erlang's delay formula.* **SPPU : May-13, Marks 6**
2. *Explain the assumptions used in second Erlang distribution for queuing systems.* **SPPU : May-16, April-17, Marks 8, May-18, Marks 5**
3. *Derive second Erlang distribution formula of a Queuing system.* **SPPU : Dec.-17, Marks 5**

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UNIT V

5 Wireless and Mobile Technologies and Protocols and their Performance Evaluation

Syllabus

Introduction, Wireless and mobile technologies, LTE- advanced, 5G - Architecture, wireless local area network and Simulations of wireless networks.

Contents

5.1 Long Term Evolution (LTE)

- LTE stands for Long Term Evolution and it was started as a project in 2004 by telecommunication body known as the Third Generation Partnership Project (3GPP). SAE (System Architecture Evolution) is the corresponding evolution of the GPRS/3G packet core network evolution. The term LTE is typically used to represent both LTE and SAE.
- Long Term Evolution (LTE) is the next step forward in cellular 3G services. LTE enhanced the Universal Mobile Telecommunication Services (UMTS) in a set of points on account of the future generation cellular technology needs and growing mobile communication services requirements.
- LTE evolved from an earlier 3GPP system known as the Universal Mobile Telecommunication System (UMTS), which in turn evolved from the Global System for Mobile Communications (GSM). Even related specifications were formally known as the evolved UMTS terrestrial radio access (E-UTRA) and evolved UMTS terrestrial radio access network (E-UTRAN). First version of LTE was documented in Release 8 of the 3GPP specifications.
- A rapid increase of mobile data usage and emergence of new applications such as MMOG (Multimedia Online Gaming), mobile TV, Web 2.0, streaming contents have motivated the 3rd Generation Partnership Project (3GPP) to work on the Long-Term Evolution (LTE) on the way towards fourth-generation mobile.
- LTE offers a reduced latency delay, which is achieved with a simplified flat radio infrastructure in which some of the functions have been moved from the Radio Network Controller (RNC) to the evolved NodeB (eNB).
- Another design goal is to increase spectrum efficiency and as such a better cost per bit ratio and better service provisioning.
- Increased data rates will be realized by including the support of multi antenna techniques and in combination with techniques such as Orthogonal Frequency Division Multiplexing (OFDM), it will offer the required flexibility in spectrum deployment and provide higher robustness against frequency selective fading for the system.
- The main goal of LTE is to provide a high data rate, low latency and packet optimized radio access technology supporting flexible bandwidth deployments.

5.1.1 Features of LTE

1. LTE is the successor technology not only of UMTS but also of CDMA 2000.
2. LTE is important because it will bring up to 50 times performance improvement and much better spectral efficiency to cellular networks.

3. LTE introduced to get higher data rates, 300 Mbps peak downlink and 75 Mbps peak uplink. In a 20 MHz carrier, data rates beyond 300 Mbps can be achieved under very good signal conditions.
4. LTE is an ideal technology to support high data rates for the services such as Voice Over IP (VOIP), streaming multimedia, videoconferencing or even a high-speed cellular modem.
5. LTE uses both Time Division Duplex (TDD) and Frequency Division Duplex (FDD) mode. In FDD uplink and downlink transmission used different frequency, while in TDD both uplink and downlink use the same carrier and are separated in time.
6. LTE supports flexible carrier bandwidths, from 1.4 MHz up to 20 MHz as well as both FDD and TDD. LTE designed with a scalable carrier bandwidth from 1.4 MHz up to 20 MHz which bandwidth is used depends on the frequency band and the amount of spectrum available with a network operator.
7. All LTE devices have to support (MIMO) Multiple Input Multiple Output transmissions, which allow the base station to transmit several data streams over the same carrier simultaneously.
8. All interfaces between network nodes in LTE are now IP based, including the backhaul connection to the radio base stations. This is great simplification compared to earlier technologies that were initially based on E1/T1, ATM and frame relay links, with most of them being narrowband and expensive.
9. Quality of Service (QoS) mechanism have been standardized on all interfaces to ensure that the requirement of voice calls for a constant delay and bandwidth, can still be met when capacity limits are reached.
10. Works with GSM/EDGE/UMTS systems utilizing existing 2G and 3G spectrum and new spectrum. Supports hand-over and roaming to existing mobile networks.

5.1.2 Advantages of LTE

1. **High throughput :** High data rates can be achieved in both downlink as well as uplink. This causes high throughput.
2. **Low latency :** Time required to connect to the network is in range of a few hundred milliseconds and power saving states can now be entered and exited very quickly.
3. **FDD and TDD in the same platform :** Frequency Division Duplex (FDD) and Time Division Duplex (FDD), both schemes can be used on same platform.

4. **Superior end-user experience :** Optimized signaling for connection establishment and other air interface and mobility management procedures have further improved the user experience. Reduced latency (to 10 ms) for better user experience.
5. **Seamless Connection :** LTE will also support seamless connection to existing networks such as GSM, CDMA and WCDMA.
6. **Plug and play :** The user does not have to manually install drivers for the device. Instead system automatically recognizes the device, loads new drivers for the hardware if needed, and begins to work with the newly connected device.
7. **Simple architecture :** Because of simple architecture low operating expenditure (OPEX).

5.1.3 LTE - QoS

- LTE architecture supports **hard QoS,** with end-to-end quality of service and Guaranteed Bit Rate (GBR) for radio bearers. Just as ethernet and the internet have different types of QoS, for example, various levels of QoS can be applied to LTE traffic for different applications. Because the LTE MAC is fully scheduled, QoS is a natural fit.
- Evolved Packet System (EPS) bearers provide one-to-one correspondence with RLC radio bearers and provide support for Traffic Flow Templates (TFT).
- There are four types of EPS bearers :
  1. **GBR Bearer** resources permanently allocated by admission control.
  2. **Non-GBR Bearer** no admission control.
  3. **Dedicated Bearer** associated with specific TFT (GBR or non - GBR).
  4. **Default Bearer** Non GBR, **catch-all** for unassigned traffic.

5.1.4 Frequency Bands of LTE

- Following is the table for E-UTRA operating bands taken from LTE Specification 36.101(v860).
- To support reliable global roaming, the best results will be achieved if the same frequency bands are used around the world.
- From the column "duplex mode" one can also see that frequency bands have been specified working either in FDD or TDD duplex mode. The frequency bands that are used by UMTS today are also part of the frequency bands specified for LTE.

E-UTRA operating band	Uplink (UL) operating band (MHz)	Downlink (DL) operating band (MHz)	Duplex mode
1	1920-1980	2110-2170	FDD
2	1850-1910	1930-1990	FDD
3	1710-1785	1805-1880	FDD
4	1710-1755	2110-2155	FDD
5	824-849	869-894	FDD
6	830-840	875-885	FDD
7	2500-2570	2620-2690	FDD
8	880-915	925-960	FDD
9	1849.9-1784.9	1844.9-1879.9	FDD
10	1710-1770	2110-2170	FDD
11	1427.9-1452.9	1475.9-1500-9	FDD
12	698-716	729-746	FDD
13	777-787	746-756	FDD
14	788-798	758-768	FDD
15	Reserved	Reserved	FDD
16	Reserved	Reserved	FDD
17	704-716	734-746	FDD
18	815-830	860-876	FDD
19	830-845	875-890	FDD
33	1900-1920	1900-1920	TDD
34	2010-2025	2010-2025	TDD
35	1850-1910	1850-1910	TDD
36	1930-1990	1930-1990	TDD
37	1910-1930	1910-1930	TDD

38	2570-1620	2570-1620	TDD
39	1880-1920	1880-1920	TDD
40	2300-2400	2300-2400	TDD

5.2 LTE Network Architecture

- The high-level network architecture of LTE is comprised of following three main components :
  1. The User Equipment (UE).
  2. The Evolved UMTS Terrestrial Radio Access Network (E-UTRAN).
  3. The Evolved Packet Core (EPC).
- The evolved packet core communicates with packet data networks in the outside world such as the internet, private corporate networks or the IP multimedia subsystem.
- The interfaces between the different parts of the system are denoted Uu, S1 and SGi as shown in Fig. 5.2.1.

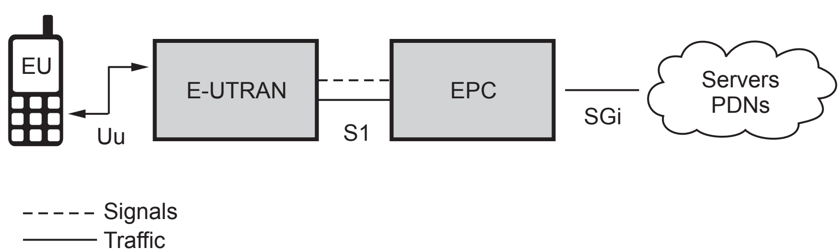

Fig. 5.2.1 LTE structure

5.2.1 User Equipment (UE)

- The internal architecture of the user equipment for LTE is identical to the one used by UMTS and GSM which is actually a Mobile Equipment (ME). The mobile equipment comprised of the following important modules :
  1. **Mobile Termination (MT) :** It handles all the communication functions.
  2. **Terminal Equipment (TE) :** It terminates the data streams.
  3. **Universal Integrated Circuit Card (UICC) :** It is also known as the SIM card for LTE equipments. It runs an application known as the Universal Subscriber Identity Module (USIM).
- A **USIM** stores user-specific data very similar to 3G SIM card. This keeps information about the user's phone number, home network identity and security keys etc.

5.2.2 The E-UTRAN / Access Network

- The architecture of evolved UMTS Terrestrial Radio Access Network (E-UTRAN) is shown in Fig. 5.2.2.

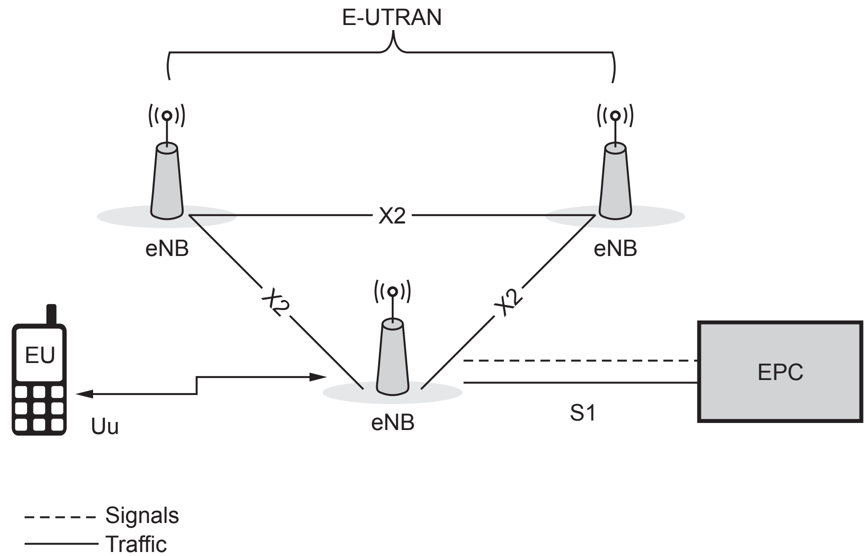

Fig. 5.2.2 E-UTRAN architecture

- For normal user traffic (as opposed to broadcast), there is no centralized controller in E-UTRAN ; hence the E-UTRAN architecture is said to be flat.
- The E-UTRAN handles the radio communications between the mobile and the evolved packet core and just has one component, the evolved base stations, called **eNodeB** or **eNB.**
- Each eNB is a base station that controls the mobiles in one or more cells. The base station that is communicating with a mobile is known as its serving eNB.
- LTE Mobile communicates with just one base station and one cell at a time.
- The protocols that run between the eNodeBs and the UE are known as the "AS protocols."

Functions of E-UTRAN

- The E-UTRAN is responsible for all radio-related functions, which can be summarized briefly as :
  1. Radio Resource Management (RRM) - This covers all functions related to the radio bearers, such as radio bearer control, radio admission control, radio mobility control, scheduling and dynamic allocation of resources to UEs in both uplink and downlink.

2. Header compression - This helps to ensure efficient use of the radio interface by compressing the IP packet headers that could otherwise represent a significant overhead, especially for small packets such as VoIP.
3. Security - All data sent over the radio interface is encrypted.
4. Connectivity to the EPC - This consists of the signaling toward MME and the bearer path toward the S-GW.

- On the network side, all of these functions reside in the eNodeBs, each of which can be responsible for managing multiple cells.
- Unlike some of the previous second and third-generation technologies, LTE integrates the radio controller function into the eNodeB. This allows tight interaction between the different protocol layers of the Radio Access Network (RAN), thus reducing latency and improving efficiency.
- Such distributed control eliminates the need for a high-availability, processing-intensive controller, which in turn has the potential to reduce costs and avoid "single points of failure." Furthermore, as LTE does not support soft handover there is no need for a centralized data-combining function in the network.

Functions of eNB

- There are following main functions supported by eNB :

1. The eBN sends and receives radio transmissions to all the mobiles using the analogue and digital signal processing functions of the LTE air interface.
2. The eNB controls the low-level operation of all its mobiles, by sending them signalling messages such as handover commands.
3. Functions for Radio Resource Management : Radio Bearer Control, Radio Admission Control, Connection Mobility Control, Dynamic allocation of resources to UEs in both uplink and downlink (scheduling).
4. IP header compression and encryption of user data stream.
5. Selection of an MME at UE attachment when no routing to an MME can be determined from the information provided by the UE.
6. Routing of user plane data towards Serving Gateway.
7. Scheduling and transmission of paging messages (originated from the MME).
8. Scheduling and transmission of broadcast information (originated from the MME or Operation and Maintenance [O&M]).
9. Measurement and measurement reporting configuration for mobility and scheduling.

- Each eBN connects with the EPC by means of the S1 interface and it can also be connected to nearby base stations by the X2 interface, which is mainly used for signalling and packet forwarding during handover.
- A home eNB (HeNB) is a base station that has been purchased by a user to provide femtocell coverage within the home.
- A home eNB belongs to a closed subscriber group (CSG) and can only be accessed by mobiles with a USIM that also belongs to the closed subscriber group.

5.2.3 Evolved Packet Core (EPC) / Core Network

- The simplified architecture of Evolved Packet Core (EPC) is shown in Fig. 5.2.3. It comprises of following components :
  1. Mobility Management Entity (MME)
  2. Serving gateway (S-GW)
  3. Packet-data Network Gateway (P-GW)
  4. Home Subscriber Server (HSS)

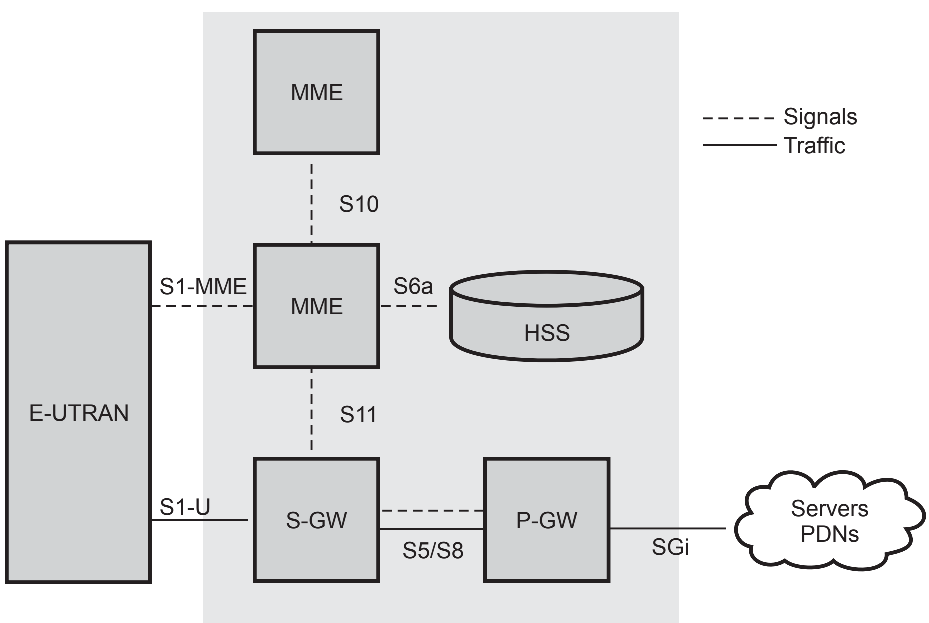

Fig. 5.2.3 EPC structure

5.2.3.1 Mobility Management Entity (MME)

- The Mobility Management Entity (MME) controls the high-level operation of the mobile by means of signalling messages and Home Subscriber Server (HSS).

- The Mobility Management Entity (MME) is the control node that processes the signaling between the UE and the CN. The protocols running between the UE and the CN are known as the Non Access Stratum (NAS) protocols.

Functions of MME

- The MME hosts the following functions :
  1. Inter CN node signaling for mobility between 3GPP access networks.
  2. Idle mode UE reachability (including control and execution of paging retransmission).
  3. Tracking area list management (for UE in idle and active mode).
  4. PDN GW and serving GW selection.
  5. MME selection for handovers with MME change.
  6. SGSN selection for handovers to 2G or 3G 3GPP access networks.
  7. Roaming.
  8. Authentication.
  9. Bearer management functions including dedicated bearer establishment.

5.2.3.2 Serving Gateway (S-GW)

- The serving gateway (S-GW) acts as a router, and forwards data between the base station and the PDN gateway.
- All user IP packets are transferred through the serving gateway, which serves as the local mobility anchor for the data bearers when the UE moves between eNodeBs.
- It also retains the information about the bearers when the UE is in the idle state (known as "EPS Connection Management - IDLE" [ECM-IDLE]) and temporarily buffers downlink data while the MME initiates paging of the UE to re-establish the bearers.
- In addition, the S-GW performs some administrative functions in the visited network such as collecting information for charging (for example, the volume of data sent to or received from the user) and lawful interception.
- It also serves as the mobility anchor for interworking with other 3GPP technologies such as general packet radio service (GPRS) and UMTS.

Functions of Serving Gateway(S-GW)

- The Serving Gateway (S-GW) hosts the following functions :
  1. The local mobility anchor point for inter-eNB handover.
  2. Mobility anchoring for inter-3GPP mobility.

3. E-UTRAN idle mode downlink packet buffering and initiation of network triggered service request procedure.
4. Lawful interception.
5. Packet routing and forwarding.
6. Transport level packet marking in the uplink and the downlink.
7. Accounting on user and QCI granularity for inter-operator charging.
8. UL and DL charging per UE, PDN and QCI.

5.2.3.3 Packet-data Network Gateway (P-GW)

- The PDN gateway is responsible for IP address allocation for the UE, as well as QoS enforcement and flow-based charging according to rules from the PCRF.
- It is responsible for the filtering of downlink user IP packets into the different QoS-based bearers. This is performed based on Traffic Flow Templates (TFTs).
- The P-GW performs QoS enforcement for guaranteed bit rate (GBR) bearers. It also serves as the mobility anchor for interworking with non-3GPP technologies such as CDMA2000 and WiMAX® networks.

5.2.3.4 Home Subscriber Server (HSS)

- The Home Subscriber Server (HSS) component has been carried forward from UMTS and GSM and is a central database that contains information about all the network operator's subscribers.
- The Home Subscriber Server contains users' SAE subscription data such as the EPS-subscribed QoS profile and any access restrictions for roaming. It also holds information about the PDNs to which the user can connect. This could be in the form of an Access Point Name (APN) (which is a label according to DNS naming conventions describing the access point to the PDN) or a PDN address (indicating subscribed IP addresses).
- In addition the HSS holds dynamic information such as the identity of the MME to which the user is currently attached or registered.
- The HSS may also integrate the authentication center (AUC), which generates the vectors for authentication and security keys.

5.2.3.5 Packet Data Network (PDN)

- The Packet Data Network (PDN) Gateway (P-GW) communicates with the outside world i.e. Packet Data Networks PDN, using SGi interface. Each packet data network is identified by an Access Point Name (APN).

- The PDN gateway has the same role as the GPRS support node (GGSN) and the serving GPRS support node (SGSN) with UMTS and GSM.
- The interface between the serving and PDN gateways is known as S5/S8. This has two slightly different implementations, namely S5 if the two devices are in the same network, and S8 if they are in different networks.

Functions of PDN

- The PDN Gateway (P-GW) hosts the following functions :
  1. Per-user based packet filtering (by e.g. deep packet inspection)
  2. Lawful interception
  3. UE IP address allocation
  4. Transport level packet marking in the downlink
  5. UL and DL service level charging, gating and rate enforcement
  6. DL rate enforcement based on APN-AMBR.

5.2.4 Functional Split between E-UTRAN and EPC

- The functional split between the E-UTRAN and the EPC for an LTE network is shown in Fig. 5.2.4.

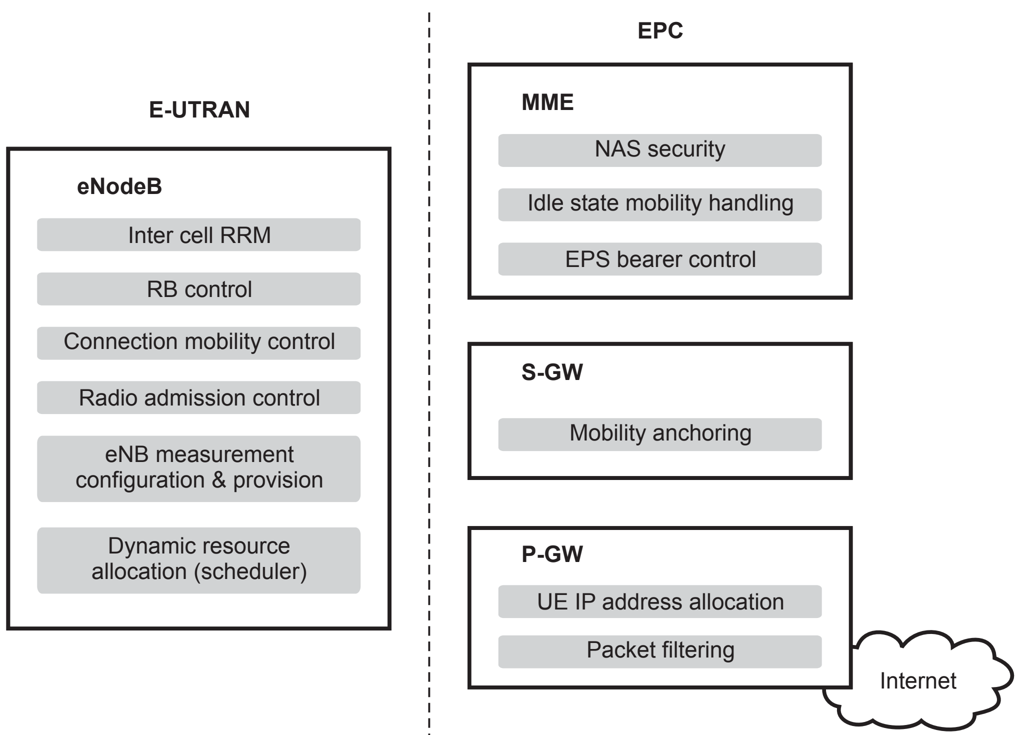

Fig. 5.2.4

5.2.5 2G/3G Versus LTE

- Following Table 5.2.1 compares various important Network Elements and Signaling protocols used in 2G/3G abd LTE.

2G/3G	LTE
GERAN and UTRAN	E-UTRAN
SGSN/PDSN-FA	S-GW
GGSN/PDSN-HA	PDN-GW
HLR/AAA	HSS
VLR	MME
SS7-MAP/ANSI-41/RADIUS	Diameter
DiameterGTPc-v0 and v1	GTPc-v2
MIP	PMIP

Table 5.2.1

5.3 LTE Roaming Architecture

- A network run by one operator in one country is known as a Public Land Mobile Network (PLMN) and when a subscribed user uses his operator's PLMN then it is said Home-PLMN but roaming allows users to move outside their home network and using the resources from other operator's network. This other network is called **Visited-PLMN.**
- A roaming user is connected to the E-UTRAN, MME and S-GW of the visited LTE network. However, LTE/SAE allows the P-GW of either the visited or the home network to be used, as shown in below :
- Using the home network's P-GW allows the user to access the home operator's services even while in a visited network.
- A P-GW in the visited network allows a "local breakout" to the Internet in the visited network.
- The interface between the serving and PDN gateways is known as S5/S8. This has two slightly different implementations, namely S5 if the two devices are in the same network, and S8 if they are in different networks. For mobiles that are not roaming, the serving and PDN gateways can be integrated into a single device, so that the S5/S8 interface vanishes altogether.

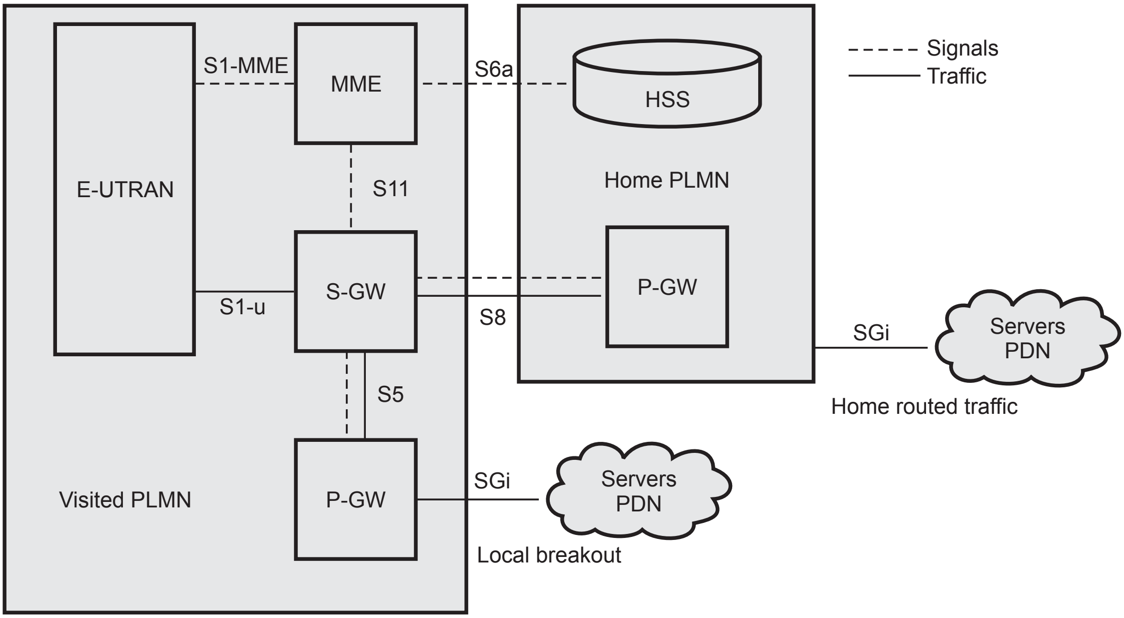

Fig. 5.3.1 LTE roaming architecture

5.3.1 LTE Roaming Charging

- The complexities of the new charging mechanisms required to support 4G roaming are much more abundant than in a 3G environment.
- Few words about both pre-paid and post-paid charging for LTE roaming are explained below :

1. Pre-paid Charging

- The CAMEL standard, which enables prepaid services in 3G, is not supported in LTE ; therefore, prepaid customer information must be routed back to the home network as opposed to being handled by the local visited network. As a result, operators must rely on new accounting flows to access prepaid customer data, such as through their P-Gateways in both IMS and non-IMS environments or via their CSCF in an IMS environment.

2. Post-paid Charging

- Post-paid data-usage charging works the same in LTE as it does in 3G, using versions TAP 3.11 or 3.12. With local breakout of IMS services, TAP 3.12 is required.
- Operators do not have the same amount of visibility into subscriber activities as they do in home-routing scenarios in case of local breakout scenarios because subscriber-data sessions are kept within the visited network ; therefore, in order for the home operator to capture real-time information on both pre- and post-paid

customers, it must establish a diameter interface between charging systems and the visited network's P-Gateway.

- In case of local breakout of IMS services scenario, the visited network creates call detail records (CDRs) from the S-Gateway(s), however, these CDRs do not contain all of the information required to create a TAP 3.12 mobile session or messaging event record for the service usage. As a result, operators must correlate the core data network CDRs with the IMS CDRs to create TAP records.

5.4 LTE Numbering and Addressing

- An LTE network area is divided into three different types of geographical areas :

1. MME pool areas

- This is an area through which the mobile can move without a change of serving MME.
- Every MME pool area is controlled by one or more MMEs on the network.

2. S-GW service areas

- This is an area served by one or more serving gateways S-GW, through which the mobile can move without a change of serving gateway.

3. Tracking areas

- The MME pool areas and the S-GW service areas are both made from smaller, non-overlapping units known as Tracking Areas (TAs). They are similar to the location and routing areas from UMTS and GSM and will be used to track the locations of mobiles that are on standby mode.
- Thus an LTE network will comprise of many MME pool areas, many S-GW service areas and lots of tracking areas.

5.4.1 Network IDs

- The network itself will be identified using Public Land Mobile Network Identity (PLMN-ID) which will have a three digit Mobile Country Code (MCC) and a two or three digit Mobile Network Code (MNC).
- For example, the Mobile Country Code for the UK is 234, while Vodafone's UK network uses a Mobile Network Code of 15.

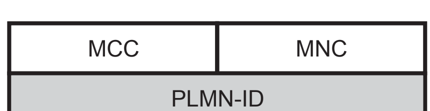

Fig. 5.4.1

5.4.2 MME IDs

- Each MME has three main identities. An MME code (MMEC) uniquely identifies the MME within all the pool areas. A group of MMEs is assigned an MME Group Identity (MMEGI) which works along with MMEC to make MME identifier (MMEI).
- A MMEI uniquely identifies the MME within a particular network.
- If we combine PLMN-ID with the MMEI then we arrive at a Globally Unique MME Identifier (GUMMEI), which identifies an MME anywhere in the world :

MMEC	MMEGI
MMEI	

Fig. 5.4.2

PLMN-ID	MMEI
GUMMEI	

Fig. 5.4.3

5.4.3 Tracking Area IDs

- Each tracking area has two main identities. The tracking area code (TAC) identifies a tracking area within a particular network and if we combining this with the PLMN-ID then we arrive at a Globally Unique Tracking Area Identity (TAI).

PLMN-ID	TAC
TAI	

Fig. 5.4.4

5.4.4 Cell IDs

- Each cell in the network has three types of identity. The E-UTRAN Cell Identity (ECI) identifies a cell within a particular network, while the E-UTRAN Cell Global Identifier (ECGI) identifies a cell anywhere in the world.
- The physical cell identity, which is a number from 0 to 503 and it distinguishes a cell from its immediate neighbours.

5.4.5 Mobile Equipment ID

- The International Mobile Equipment Identity (IMEI) is a unique identity for the mobile equipment and the International Mobile Subscriber Identity (IMSI) is a unique identity for the UICC and the USIM.
- The M temporary mobile subscriber identity (M-TMSI) identifies a mobile to its serving MME. Adding the MME code in M-TMSI results in a S Temporary Mobile Subscriber Identity (S-TMSI), which identifies the mobile within an MME pool area.
- Finally adding the MME group identity and the PLMN identity with S-TMSI results in the Globally Unique Temporary Identity (GUTI).

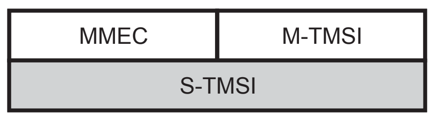

Fig. 5.4.5

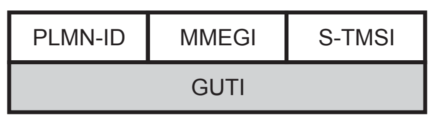

Fig. 5.4.6

5.5 LTE Radio Protocol Architecture

- The radio protocol architecture for LTE can be separated into **control plane** architecture and **user plane** architecture as shown in Fig. 5.5.1.

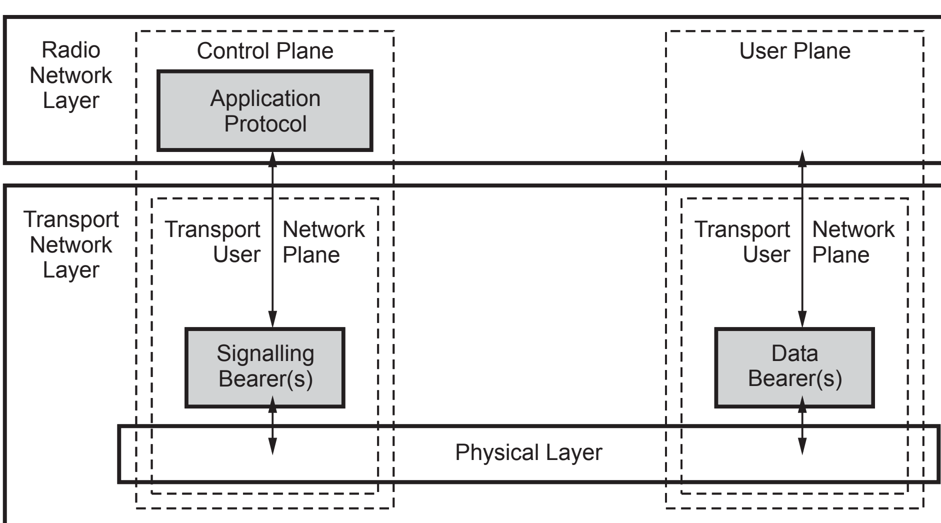

Fig. 5.5.1

- At user plane side, the application creates data packets that are processed by protocols such as TCP, UDP and IP, while in the control plane ; the Radio Resource Control (RRC) protocol writes the signalling messages that are exchanged between the base station and the mobile.
- In both cases, the information is processed by the Packet Data Convergence Protocol (PDCP), the Radio Link Control (RLC) protocol and the Medium Access Control (MAC) protocol, before being passed to the physical layer for transmission.

5.5.1 User Plane

- The user plane protocol stack between the e-Node B and UE consists of the following sub-layers :
  1. PDCP (Packet Data Convergence Protocol)
  2. RLC (Radio Link Control)
  3. Medium Access Control (MAC)
- On the user plane, packets in the core network (EPC) are encapsulated in a specific EPC protocol and tunneled between the P-GW and the eNodeB.
- Different tunneling protocols are used depending on the interface. GPRS Tunneling Protocol (GTP) is used on the S1 interface between the eNodeB and S-GW and on the S5/S8 interface between the S-GW and P-GW.
- The E-UTRAN user plane protocol stack is shown in shading in Fig. 5.5.2 consisting of
  1. Packet Data Convergence Protocol (PDCP),
  2. Radio Link Control (RLC)
  3. Medium Access Control (MAC) sublayers that are terminated in the eNodeB on the network side.
- Packets received by a layer are called Service Data Unit (SDU) while the packet output of a layer is referred to by Protocol Data Unit (PDU) and IP packets at user plane flow from top to bottom layers.

Data handling during handover

- In the absence of any centralized controller node, data buffering during handover due to user mobility in the E-UTRAN must be performed in the eNodeB itself.
- Data protection during handover is a responsibility of the PDCP layer. The RLC and MAC layers both start afresh in a new cell after handover.

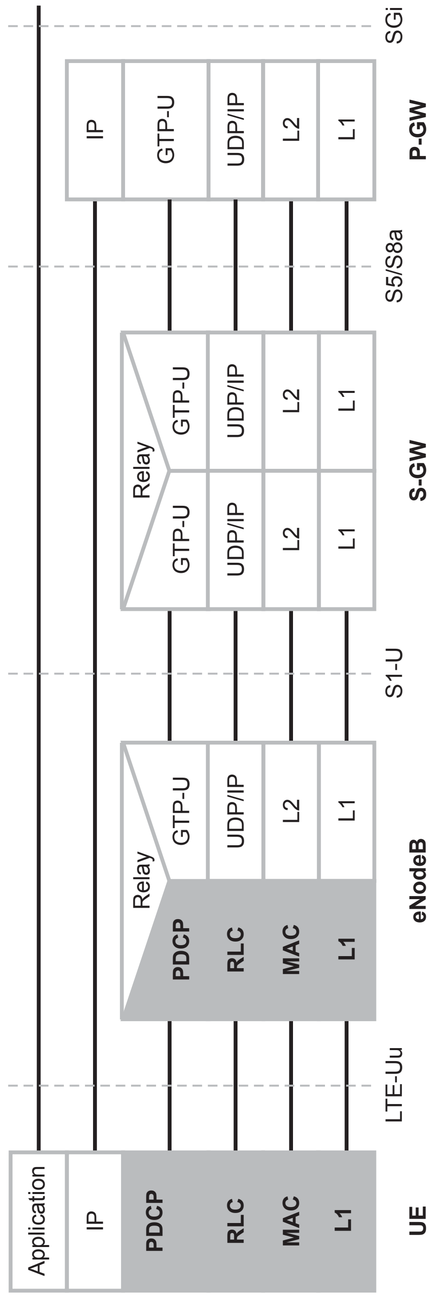

Fig. 5.5.2 E-UTRAN user plane protocol stack

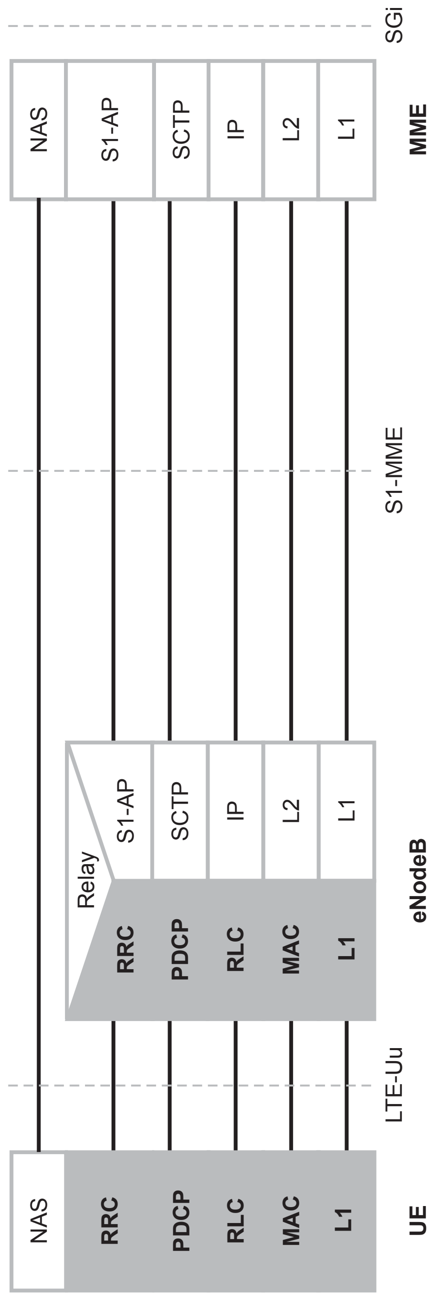

Fig. 5.5.3 Control plane protocol stack

5.5.2 Control Plane

- The control plane includes additionally the Radio Resource Control layer (RRC) which is responsible for configuring the lower layers.

- The Control plane handles radio-specific functionality which depends on the state of the user equipment which includes two states : idle or connected.

Mode	Description
Idle	The user equipment camps on a cell after a cell selection or reselection process where factors like radio link quality, cell status and radio access technology are considered. The UE also monitors a paging channel to detect incoming calls and acquire system information. In this mode, control plane protocols include cell selection and reselection procedures.
Connected	The UE supplies the E-UTRAN with downlink channel quality and neighbour cell information to enable the E-UTRAN to select the most suitable cell for the UE. In this case, control plane protocol includes the Radio Link Control (RLC) protocol.

- The protocol stack for the control plane between the UE and MME is shown in Fig. 5.5.3. The shaded region of the stack indicates the AS protocols. The lower layers perform the same functions as for the user plane with the exception that there is no header compression function for the control plane.
- The Radio Resource Control (RRC) protocol is known as "layer 3" in the AS protocol stack. It is the main controlling function in the AS, being responsible for establishing the radio bearers and configuring all the lower layers using RRC signaling between the eNodeB and the UE.

5.6 LTE Protocol Stack Layers

- The elaborated diagram of E-UTRAN Protocol Stack is shown in Fig. 5.6.1.

Physical Layer (Layer 1)

- Physical layer carries all information from the MAC transport channels over the air interface. Takes care of the link adaptation (AMC), power control, cell search (for initial synchronization and handover purposes) and other measurements (inside the LTE system and between systems) for the RRC layer.

Medium Access Layer (MAC)

- MAC layer is responsible for mapping between logical channels and transport channels, Multiplexing of MAC SDUs from one or different logical channels onto Transport Blocks (TB) to be delivered to the physical layer on transport channels, de multiplexing of MAC SDUs from one or different logical channels from Transport Blocks (TB) delivered from the physical layer on transport channels, Scheduling information reporting, Error correction through HARQ, Priority handling between UEs by means of dynamic scheduling, Priority handling between logical channels of one UE, Logical channel prioritization.

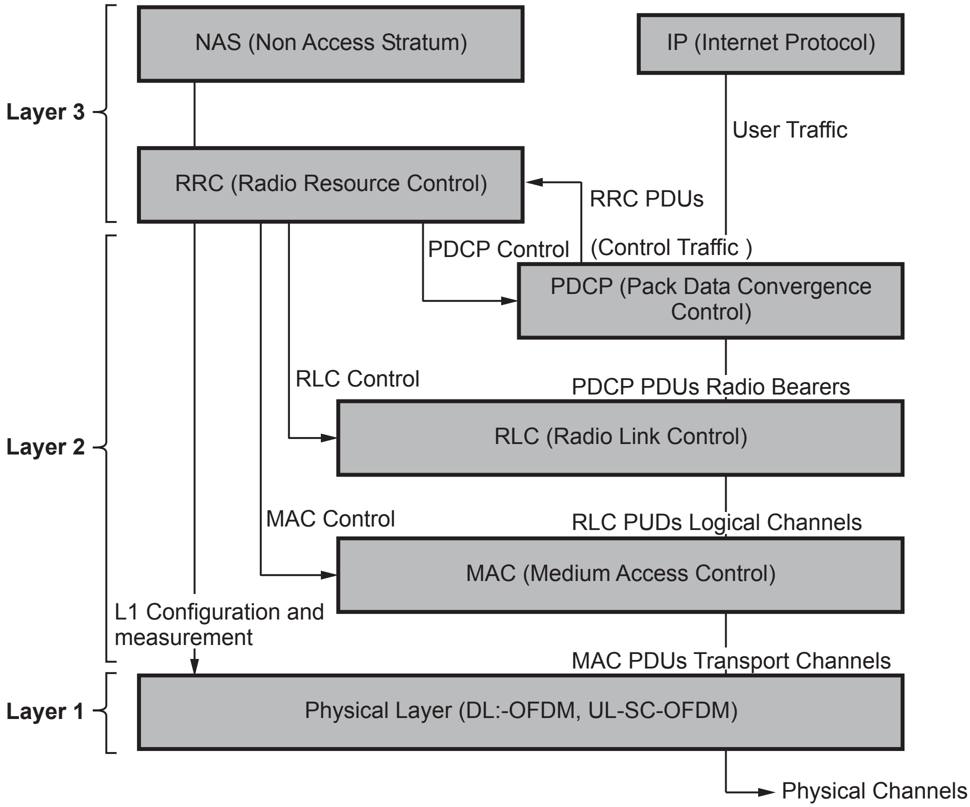

Fig. 5.6.1 LTE protocol stack

Radio Link Control (RLC)

- RLC operates in 3 modes of operation : Transparent Mode (TM), Unacknowledged Mode (UM) and Acknowledged Mode (AM).
- RLC layer is responsible for transfer of upper layer PDUs, error correction through ARQ (Only for AM data transfer), Concatenation, segmentation and reassembly of RLC SDUs (Only for UM and AM data transfer).
- RLC is also responsible for re-segmentation of RLC data PDUs (Only for AM data transfer), reordering of RLC data PDUs (Only for UM and AM data transfer), duplicate detection (Only for UM and AM data transfer), RLC SDU discard (Only for UM and AM data transfer), RLC re-establishment, and protocol error detection (Only for AM data transfer).

Radio Resource Control (RRC)

- The main services and functions of the RRC sublayer include broadcast of System Information related to the Non-Access Stratum (NAS), broadcast of System Information related to the Access Stratum (AS), Paging, establishment, maintenance and release of an RRC connection between the UE and E-UTRAN, Security functions including key management, establishment, configuration, maintenance and release of point to point radio bearers.

Packet Data Convergence Control (PDCP)

- PDCP layer is responsible for header compression and decompression of IP data, Transfer of data (user plane or control plane), Maintenance of PDCP Sequence Numbers (SNs), In-sequence delivery of upper layer PDUs at re-establishment of lower layers, Duplicate elimination of lower layer SDUs at re-establishment of lower layers for radio bearers mapped on RLC AM, Ciphering and deciphering of user plane data and control plane data, Integrity protection and integrity verification of control plane data, Timer based discard, duplicate discarding, PDCP is used for SRBs and DRBs mapped on DCCH and DTCH type of logical channels.

Non Access Stratum (NAS) Protocols

- The Non-Access Stratum (NAS) protocols form the highest stratum of the control plane between the User Equipment (UE) and MME.
- NAS protocols support the mobility of the UE and the session management procedures to establish and maintain IP connectivity between the UE and a PDN GW.

5.6.1 Frame Structure

- One element shared by the LTE DL and UL is the generic frame structure. The LTE specifications define both FDD and TDD modes of operation.
- The generic frame structure applies to both the DL and UL for FDD operation.
- LTE transmissions are segmented into frames, which are 10 msec in duration. Frame consists of 20 slot periods of 0.5 msec. Sub-frames contain two slot periods and are 1.0 msec in duration.

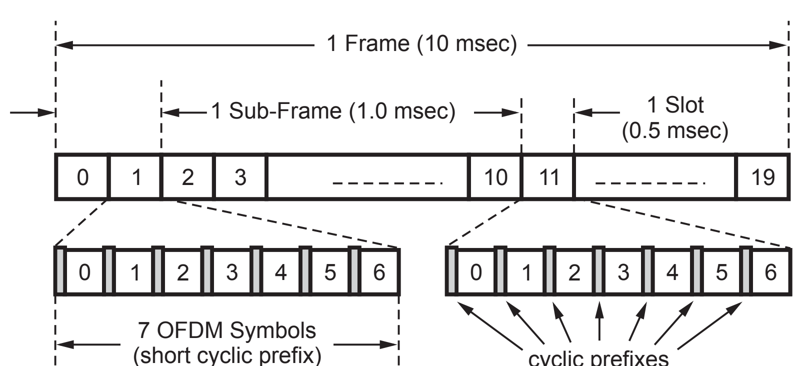

Fig. 5.6.2 Frame structure

5.7 LTE Layers Data Flow

- The logical diagram of E-UTRAN protocol layers with a depiction of data flow through various layers is shown in Fig. 5.7.1.

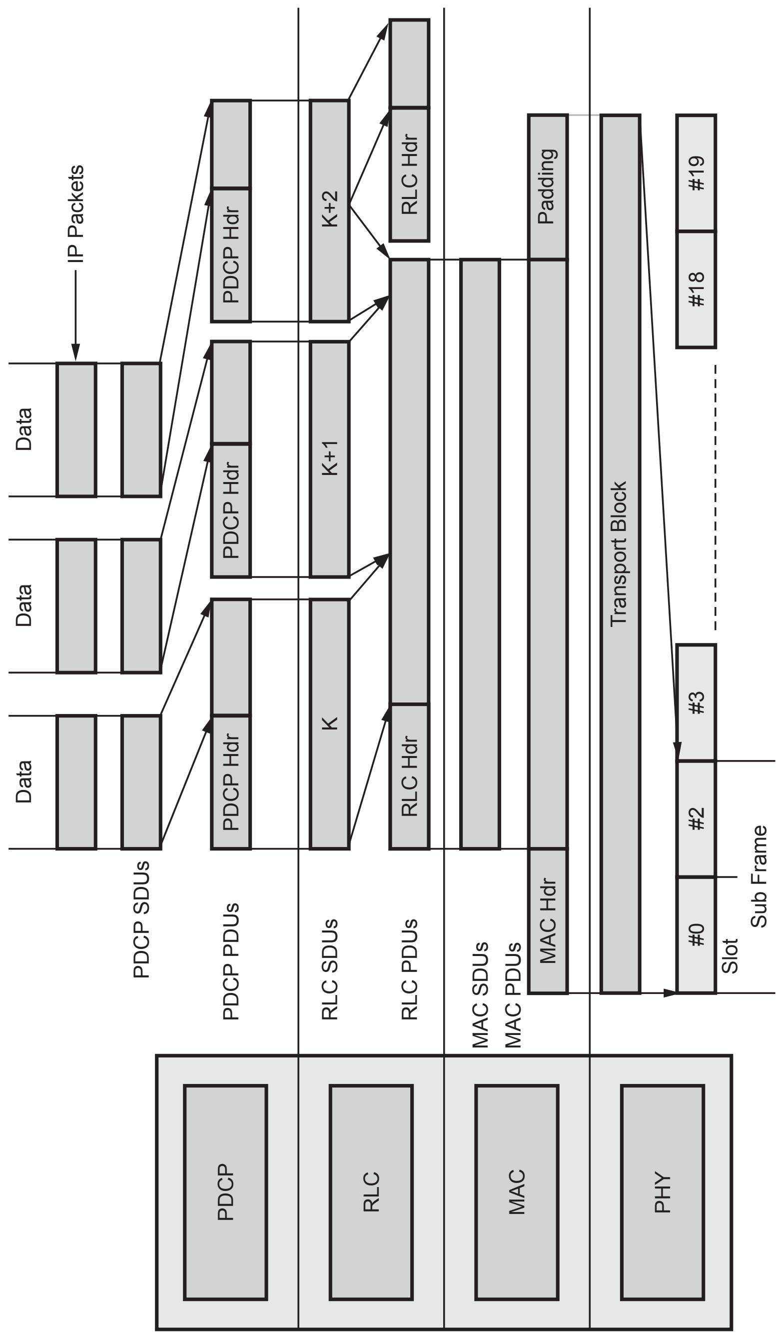

Fig. 5.7.1 E-UTRAN protocol layer

- Packets received by a layer are called **Service Data Unit (SDU)** while the packet output of a layer is referred to by **Protocol Data Unit (PDU)**.

Process of data flow from top to bottom

- IP Layer submits PDCP SDUs (IP Packets) to the PDCP layer. PDCP layer does header compression and adds PDCP header to these PDCP SDUs. PDCP layer submits PDCP PDUs (RLC SDUs) to RLC layer.

PDCP Header Compression

- PDCP removes IP header (Minimum 20 bytes) from PDU, and adds token of 1-4 bytes. This provides a tremendous savings in the amount of header that would otherwise have to go over the air.
- RLC layer does segmentation of these SDUS to make the RLC PDUs. RLC adds header based on RLC mode of operation. RLC submits these RLC PDUs (MAC SDUs) to the MAC layer.

RLC Segmentation

- If an RLC SDU is large, or the available radio data rate is low (resulting in small transport blocks), the RLC SDU may be split among several RLC PDUs. If the RLC SDU is small, or the available radio data rate is high, several RLC SDUs may be packed into a single PDU.
- MAC layer adds header and does padding to fit this MAC SDU in TTI. MAC layer submits MAC PDU to physical layer for transmitting it onto physical channels.
- Physical channel transmits this data into slots of sub frame.

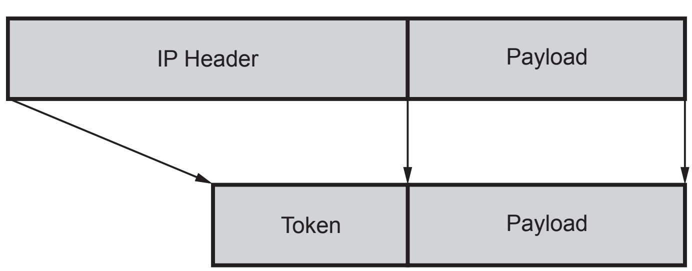

Fig. 5.7.2

5.8 LTE Communication Channels

- The information flows between the different protocols are known as channels and signals. LTE uses several different types of logical, transport and physical channel, which are distinguished by the kind of information they carry and by the way in which the information is processed.

1. **Logical Channels :** Define **whattype** of information is transmitted over the air, e.g. traffic channels, control channels, system broadcast, etc. Data and signalling messages are carried on logical channels between the RLC and MAC protocols.
2. **Transport Channels :** Define **howis** something transmitted over the air, e.g. what are encoding, interleaving options used to transmit data. Data and signalling messages are carried on transport channels between the MAC and the physical layer.
3. **Physical Channels :** Define **whereis** something transmitted over the air, e.g. first N symbols in the DL frame. Data and signalling messages are carried on physical channels between the different levels of the physical layer.

5.8.1 Logical Channels

- Logical channels define what type of data is transferred. These channels define the data-transfer services offered by the MAC layer.
- Data and signalling messages are carried on logical channels between the RLC and MAC protocols.
- Logical channels can be divided into control channels and traffic channels.
- Control Channel can be either common channel or dedicated channel. A common channel means common to all users in a cell (Point to multipoint) while dedicated channels means channels can be used only by one user (Point to Point).
- Logical channels are distinguished by the information they carry and can be classified in two ways. Firstly, logical traffic channels carry data in the user plane, while logical control channels carry signalling messages in the control plane. Following table lists the logical channels that are used by LTE :

Channel Name	Acronym	Control channel	Traffic channel
Broadcast control channel	BCCH	X	
Paging control channel	PCCH	X	
Common control channel	CCCH	X	
Dedicated control channel	DCCH	X	
Multicast control channel	MCCH	X	

Dedicated traffic channel	DTCH		X
Multicast traffic channel	MTCH		X

5.8.2 Transport Channels

- Transport channels define how and with what type of characteristics the data is transferred by the physical layer. Data and signalling messages are carried on transport channels between the MAC and the physical layer.
- Downlink data-transporting channels are specified to transport the downlink transport channels.
- Transport channels are distinguished by the ways in which the transport channel processor manipulates them. Following table lists the transport channels that are used by LTE :

Channel Name	Acronym	Downlink	Uplink
Broadcast channel	BCH	X	
Downlink shared channel	DL-SCH	X	
Paging channel	PCH	X	
Multicast channel	MCH	X	
Uplink shared channel	UL-SCH		X
Random access channel	RACH		X

- The **transport channel** processor composes several types of control information, to support the low-level operation of the physical layer. These are listed in the below table :

Field Name	Acronym	Downlink	Uplink
Downlink control information	DCI	X	
Control format indicator	CFI	X	
Hybrid ARQ indicator	HI	X	
Uplink control information	UCI		X

5.8.3 Physical Channels

- The physical or PHY layer is responsible for coding, modulation, multi antenna processing, and mapping the signal to the appropriate resources.
- It is also responsible for mapping the transport channels to the corresponding physical channel. The PHY layer offers the physical channels in form of services to the MAC layer.
- Data and signalling messages are carried on physical channels between the different levels of the physical layer and accordingly they are divided into two parts :
  1. Physical Data Channels
  2. Physical Control Channels
- Physical data channels are distinguished by the ways in which the physical channel processor manipulates them, and by the ways in which they are mapped onto the symbols and sub-carriers used by Orthogonal Frequency-Division Multiplexing (OFDM).
- Following table lists the **physical data channels** that are used by LTE :

Channel Name	Acronym	Downlink	Uplink
Physical downlink shared channel	PDSCH	X	
Physical broadcast channel	PBCH	X	
Physical multicast channel	PMCH	X	
Physical uplink shared channel	PUSCH		X
Physical random access channel	PRACH		X

Physical Control Channels

- The transport channel processor also creates control information that supports the low-level operation of the physical layer and sends this information to the physical channel processor in the form of physical control channels.
- The information travels as far as the transport channel processor in the receiver, but is completely invisible to higher layers. Similarly, the physical channel processor creates physical signals, which support the lowest-level aspects of the system.

- Physical control channels are listed in the below table

Channel Name	Acronym	Downlink	Uplink
Physical control format indicator channel	PCFICH	X	
Physical hybrid ARQ indicator channel	PHICH	X	
Physical downlink control channel	PDCCH	X	
Relay physical downlink control channel	R-PDCCH	X	
Physical uplink control channel	PUCCH		X

- The base station also transmits two other physical signals, which help the mobile acquire the base station after it first switches on. These are known as the Primary Synchronization Signal (PSS) and the Secondary Synchronization Signal (SSS).

5.8.3.1 Physical Broadcast Channel (PBCH)

- The PBCH is used for the transport of BCH information, which is necessary for the operation of a mobile system. It contains information, which is used for the configuration of UEs and transports basic system information.
- For UEs that are not connected to the network the PBCH has to be detectable without prior knowledge of the channel and its bandwidth. Therefore the PBCH is mapped to the central 72 subcarriers.
- The synchronization signals are followed by the PBCH, which are also located in the center of the bandwidth.
- Other requirements concerning the PBCH are low system overhead, reliable reception even at the cell edge and that it is decodable with low latency and low impact on battery life. Therefore information that is carried on the PBCH is reduced to a minimum and through the use of time diversity, forward error correction and antenna diversity good reception is realized.
- Because the information on the PBCH is repeated every 40 ms, low latency with little processing afford can be realized through the use of a low coding rate in combination with soft combining. This allows the reception after a short time under good channel conditions, while it is still possible to receive the PBCH under sub-optimal channel conditions by combining several partially received signals.

5.8.3.2 Physical Downlink Shared Channel (PDSCH)

- The PDSCH is used for the main downlink data that may consist of any kind of user data as well as for the broadcast of system information that is not transmitted over the PBCH. Because there is no dedicated physical layer paging channel in LTE the transport of paging messages is also done on the PDSCH.
- Data is transported in the form of transport blocks on the PDSCH, they correspond to MAC-layer Protocol Data Units (PDUs).
- The PBCH offers a variety of modulation schemes that can be applied according to the present channel conditions, ranging from QPSK up to 64QAM.

5.8.3.3 Physical Multicast Channel (PMCH)

- The purpose of the PMCH is to broadcast the same modulated symbols from multiple cells. The structure of the PMCH is still very similar to the one of PDSCH.
- In both channels the synchronization of the signals is critical for the operation of the system. In the ideal case the same signals from different cells should arrive at the UE in the time span of the cyclic prefix. Therefore the PMCH makes use of an extended cyclic prefix. This method is called **MBSFN (MBMS Single Frequency Network).**

5.8.3.4 Physical Control Format Indicator Channel (PCFICH)

- PCFICH is used to indicate the number of OFDM symbols that are used to transmit control channel information in each subframe. This information is included in the Control Format Indicator (CFI) and allows a reduction of process load for the UE. Otherwise the UE may need multiple attempts to decode the CFI value, which would result in a significant additional processing load.
- The 16 resource elements that transport the PCFICH are distributed across the frequency domain with a predefined pattern to achieve frequency diversity.

5.8.3.5 Physical Downlink Control Channel (PDCCH)

- PDCCH is used to carry Downlink Control Information (DCI), which includes resource assignments and other control information for a UE or a group of UEs.
- LTE offers the possibility to transport several PDCCH in a subframe, where the information is grouped in so-called Control Channel Elements (CCEs). The number of CCE that are transmitted to a UE is determined by the eNB. This is dependent on what channel conditions are present for the connection of the UE.
- Under bad channel conditions the UE might be scheduled more CCEs to acquire the needed robustness against interference.

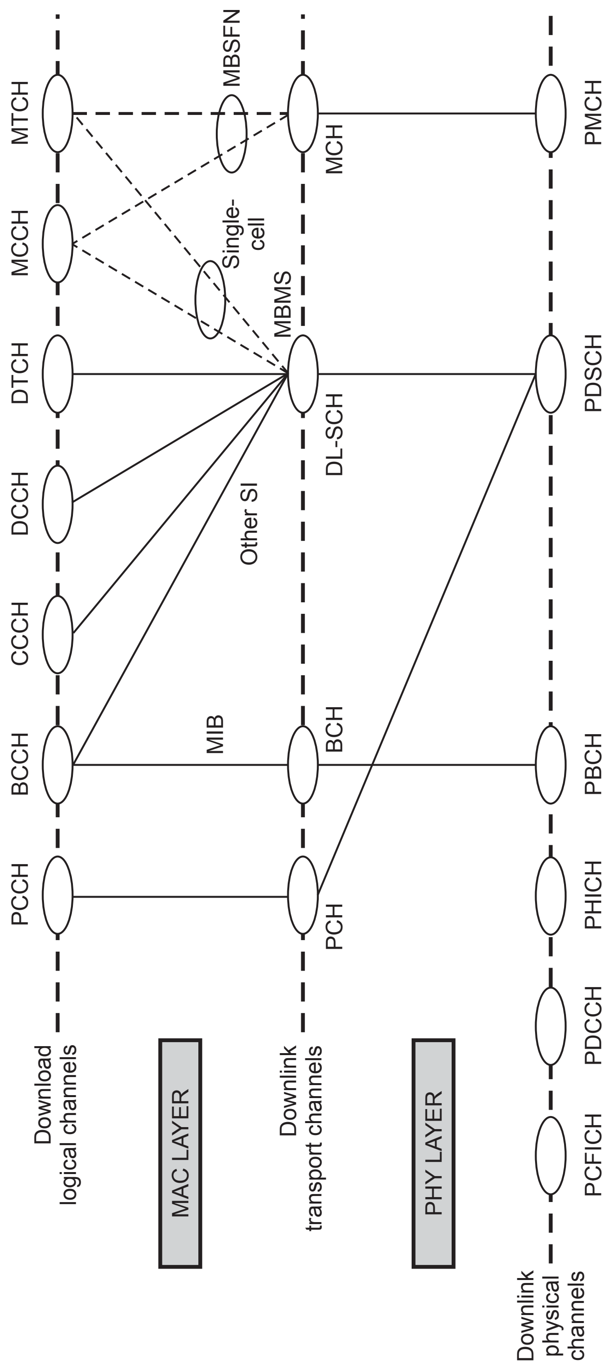

Fig. 5.8.1 Summary of downlink physical channels and mapping to higher layers

5.8.3.6 Physical Hybrid ARQ Indicator Channel (PHICH)

- PHICH is used to carry the HARQ ACK and NACK for the PUSCH. Negative acknowledgements are represented by a 1 while 0 stands for a positive acknowledgement. This information is repeated three times and modulated with BPSK, to add robustness against interference.
- Multiple PHICHs are mapped to the same resource element, which is referred to as a PHICH group. The different PHICHs within the PHICH group are separated by complex orthogonal Walsh sequences.

5.8.3.7 Physical Uplink Control Channel (PUCCH)

- PUCCH is used for uplink data and supports modulation with QPSK, 16QAM, 64QAM, where 64QAM is only supported by the highest UE category.

5.8.3.8 Physical Random Access Channel (PRACH)

- The PRACH is used to transport the RACH only. It is used when no bearer between the UE and the eNB is established in uplink direction, not been active for a while or as a response to the paging from the network.

5.8.3.9 Physical Uplink Shared Channel (PUSCH)

- PUSCH is used for transmitting control signaling information in uplink direction. It takes advantage of the use of frequency diversity to minimize resources needed for transmission.
- The Resource Blocks (RBs) are therefore located alternating at the edges of the system bandwidth. This also reduces the out-of-band emissions, because the UE only transmits a single RB per frame.
- Other control signaling that is transported on the PUCCH for instance Scheduling Requests (SR), HARQ ACK/NACK in response to downlink data packages. Furthermore the PUCCH is used for the transmission of the Channel Quality Indicator (CQI), which provides the eNB with information about channel condition, so the eNB can determine which modulation and coding to use for the data downlink.

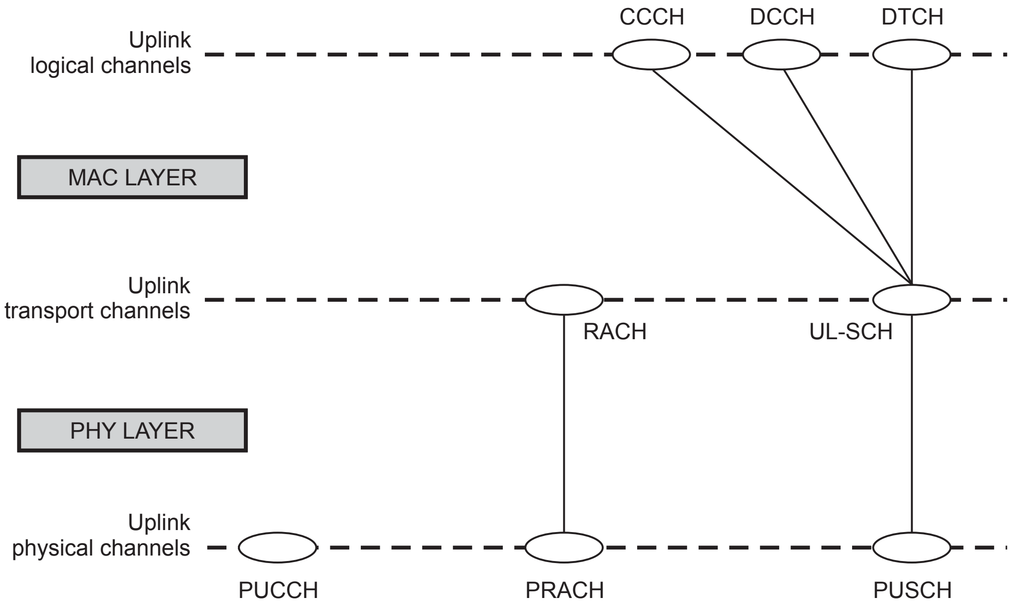

Fig. 5.8.2 Summary of uplink physical channels and mapping to higher layers

- Table 5.8.1 gives a resumption of all physical channels with their corresponding modulation scheme.

Downlink	
Downlink channels	**Modulation scheme**
PBCH	QPSK
PDCCH	QPSK
PDSCH	QPSK, 16QAM, 64QAM
PMCH	QPSK, 16QAM, 64QAM
PCFICH	QPSK
PHICH	BPSK modulated on I and Q with the spreading factor 2 or 4 Walsh codes
Physical signals	**Modulation scheme**
RS	Complex I+Q pseudo random sequence (length 31 Gold sequence) derived from cell ID
Primary synchronization	One of three Zadoff-Chu sequences
Secondary synchronization	Two 31-bit BPSK M-sequence
Uplink	

Physical channels	**Modulation scheme**
PUCCH	BPSK, QPSK
PUSCH	QPSK. 16QAM, 64QAM
PRACH	u^{th} root Zadoff-Chu
Physical signals	**Modulation scheme**
Demodulation RS	Zadoff-Chu
Sounding RS	Based on Zadoff-Chu

Table 5.8.1

5.9 LTE Transmitter and Receiver

- In transmitter side the digital random data set is generated uniformly. These blocks of digital data set have been paralleled and mapped into complex data blocks using M-QAM modulation technique.
- Every complex data block referred to a symbol of data is attached to an individual sub-carrier.
- The Inverse Fast Fourier Transform (IFFT) is used in order to generate the time version of transmitted signal.
- The time domain signals corresponding to all subcarriers are orthogonal to each other. However, the frequency spectrum overlaps.
- To remove ISI on the transmitted signal inserted cyclic prefix in front of every transmitted symbol in the block diagram.

1. **Constellation mapper :** It converts incoming bit stream to single carrier symbols (BPSK, QPSK, or 16QAM depending on channel conditions)

   1. **Serial/parallel converter :** It formats time domain SC symbols into blocks for input to FFT engine.
   2. **M-point DFT :** It converts time domain SC symbol block into M discrete tones.
   3. **Subcarrier mapping :** Maps DFT output tones to specified subcarriers for transmission. SC-FDMA systems either use contiguous tones (localized) or uniformly spaced tones (distributed).
      - The current working assumption in LTE is that localized subcarrier mapping will be used. The trades between localized and distributed subcarrier mapping are discussed further below.

5. **N-point IDFT :** Converts mapped subcarriers back into time domain for transmission

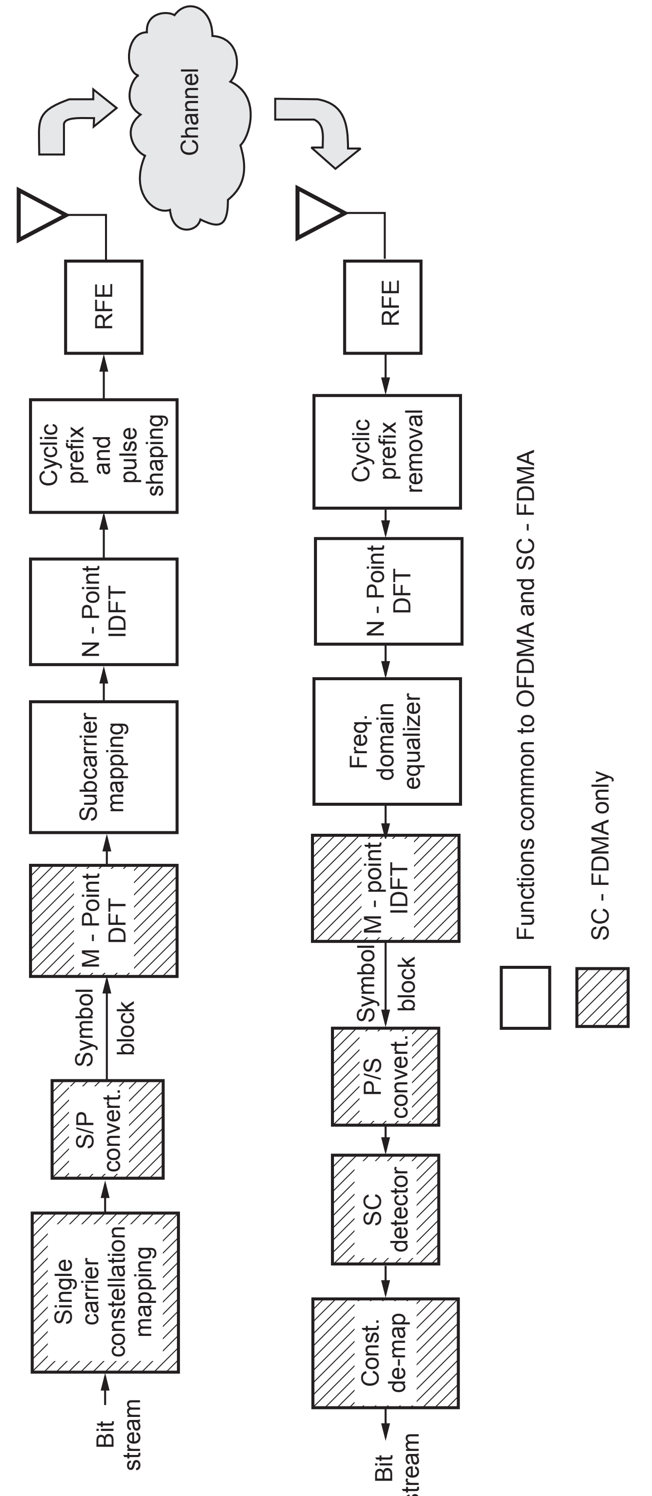

Fig. 5.9.1 LTE transmitter and receiver

6. **Cyclic prefix and pulse shaping :** Cyclic prefix is pre-pended to the composite SC-FDMA symbol to provide multipath immunity in the same manner as described for OFDM. As in the case of OFDM, pulse shaping is employed to prevent spectral regrowth.
7. **RFE :** Converts digital signal to analog and upconvert to RF for transmission

5.9.1 LTE for Uplink and Downlink

LTE uses OFDMA for DL

- OFDMA requires more power for transmission whereas SC-FDMA requires lesser power and since higher power can't be processed using UE (mostly mobile phones or CPEs) due to battery size limitations therefore SC-FDMA is used for uplink by UEs.
- Other reasons also include higher data rates achievable using OFDMA than the other one.

LTE uses SC-FDMA for UL

- In order to reduce the PAPR **(Peak-to-Average Power Ratio)** a variant of OFDMA is used. It is called **SC-FDMA** (Single Carrier Frequency Division Multiple Access).
- SC-FDMA is the method of choice for EUTRAN in the uplink direction.
- OFDM has a disadvantage that it has a high PAPR Peak to Average Power Ratio, which makes the receiver design complex and is prone to ISI in sub carriers in case of multi path propagation. This makes it impractical to be used for the UE radio design.
- In OFDM the data symbols modulate the sub carriers independently whereas in SC FDMA the modulated signal is a linear combination of all data symbols transmitted one after the other within the time instant.

5.9.2 Comparison of LTE, LTE-Advanced and IMT-Advanced

Parameter		LTE	LTE-Advanced	IMT-Advanced
Version/ Release		3GPP Release 8	3GPP Release 10	True-4G
Peak data rate	DL	300 Mbps	1 Gbps	High mobility-100Mbps
	UL	75 Mbps	500Mbps	Low Mobility-1Gbps
Peak spectrum efficiency(bps/Hz)	DL	15	30	15

	UL	3.75	15	6.75
Tx bandwidth UL & DL		Up to 200 MHz	Up to 100 MHz	Up to 40 MHz

5.10 Introduction to 4G

- This is the term used to refer to the fourth generation of mobile wireless services that has been defined by the ITU and its Radio communication sector (ITU-R) and established as an agreed upon and globally accepted definition in IMT-Advanced.
- 4G used broadly to include several types of broadband wireless access communication systems, not only cellular telephone systems.
- 4G is a network that operates on Internet technology, combines it with other applications and technologies such as Wi-Fi, and runs at speeds ranging from 100 Mbps (in cell-phone networks) to 1 Gbps (in local WI-FI networks).
- The 4G systems are about seamlessly integrating terminals, networks, and applications to satisfy increasing user demands.
- The 4G system encompass all systems from various networks, public to private, operator-driven broadband networks to personal areas, and ad hoc networks.
- 4G is a set of standards of mobile technology that entail increased data transfer speeds, enhanced security, and reduced blips in transmission when a device moves between areas covered by different networks.
- The 4G systems has to be interoperable with 2G and 3G systems, as well as with digital (broadband) broadcasting systems.
- The 4G intends to integrate from satellite broadband to high altitude platform to cellular 2G and 3G systems to Wireless Local Loop (WLL) and broadband wireless access (BWA) to WLAN and wireless personal area networks (WPANs), all with IP as the integrating.

5.10.1 Limitations of 3G

1. The maximum bit rates were still a factor of 10 and more behind the simultaneous state of systems like IEEE 802.11n and 802.16e/m.
2. The latency of user plane traffic (UMTS : >30 ms) and of resource assignment procedures (UMTS : >100 ms) is too big to handle traffic with high bit rate variance efficiently.
3. The UE terminal complexity for WCDMA or CDMA systems is quite high, making terminals expensive, resulting in poor performing implementations of receivers and inhibiting the implementation of other performance enhancements.

5.10.2 Vision for 4G

- The new generation of wireless is intended to complement and replace the 3G systems.
- The future 4G infrastructures are aimed to consist of a set of various networks using IP (Internet protocol) as a common protocol so that users are in control because they will be able to choose every application and environment.
- New 4G networks are designed to provide a wide variety of new services, such as high-quality voice to high definition video to high-data-rate wireless channels, several types of broadband wireless access communication systems.
- Based on the developing trends of mobile communication, 4G will have broader bandwidth, higher data rate, and smoother and quicker handoff and will focus on ensuring seamless service across a multitude of wireless systems and networks.
- The 4G technology is designed for seamlessly integrating terminals, networks, and applications to meet ever increasing subscriber expectations.
- The key concept is integrating the 4G capabilities with all of the existing mobile technologies through advanced technologies. Application adaptability and being highly dynamic are the main features of 4G services of interest to users.
- The fourth generation will encompass all systems from various networks, public to private ; operator-driven broadband networks to personal areas ; and ad hoc networks.
- The 4G systems will interoperate with 2G and 3G systems, as well as with digital (broadband) broadcasting systems.
- In addition, 4G systems will be fully IP-based wireless Internet. This all encompassing integrated perspective shows the broad range of systems that the fourth generation intends to integrate, from satellite broadband to high altitude platform to cellular 3G and 3G systems to WLL (Wireless Local Loop) and FWA (Fixed Wireless Access) to WLAN and PAN (Personal Area Network), all with IP as the integrating mechanism.
- With 4G, a range of new services and models will be available. These services and models need to be further examined for their interface with the design of 4G systems.

5.10.3 Comparison of 3G and 4G

Parameters	3G	4G
Data Throughput	Up to 3.1 Mbps with an average speed range between 0.5 to 1.5 Mbps	Practically, 2 to 12 Mbps
Peak upload rate	5 Mbps	500 Mbps
Peak download rate	100 Mbps	1 Gbps
Internet protocol	Various air link protocol including IPv5.0	IPv6.0
Switching technique	Circuit and Packet switching	Packet switching
Network architecture	Wide area cell based	Integration of wireless LAN and WAN
Frequency band	1.8 to 2.5 GHz	2 to 8 GHz
Forward error correction (FEC)	Turbo codes	Concatenated codes
Services and applications	CDMA 2000, UMTS, EDGE	Wimax2 and LTE-Advance

5.11 5G

- 5G isn't just an incremental improvement over 4G - it's the next major evolution of mobile communication technology with performance improvements of several orders of magnitude over today's networks.
- 5G does not replace 4G, it simply enables a huge diversity of tasks that 4G cannot perform.
- 4G will continue to advance in parallel with 5G, as the network to support more routine tasks.
- 5G will enable services yet to be imagined, in a world where national economies are driven by sophisticated communications networks.

5.11.1 5G Challenges

1. Increased Capacity and Connectivity

- With hundreds of potential new use cases and the spectacular growth of the Internet of Things (IoT), 5G networks must be able to handle massive increases in capacity and connectivity.
- From 2018 to 2024, total mobile data traffic is expected to increase by a factor of five and, in the same time frame, video traffic will increase over 6 times.
- With the use of middle-and-high-spectrum frequencies, more base stations will be necessary due to the shorter distances that traffic can be transmitted over.
- Because of the many new locations to manage, service providers will also face new environmental considerations, new types of access points and new interface types.
- This densification, of which a significant amount will come from small cells, is expected to increase the connectivity in the network by 3 to 5 times.

2. Enhanced Capabilities

- New services and applications such as autonomous cars, remote healthcare and virtual reality are expected to create the need for new network capabilities.
- These applications are expected to require extremely low latency, which can only be delivered by having compute power at the edge.
- Meeting timing and synchronization requirements in 5G is a key requirement that is gaining increasing attention among operators. While network-based timing and synchronization are standard procedure for mobile networks in many parts of the world, some countries, to date, relies mainly on GPS clocks.
- However, GPS in dense, urban areas populated by small cells (many indoors) may not be viable economically, or in many cases, technically, due to interference.
- Additionally, in the case of losing a GPS-based clock source, it is hard to hold the required synchronization accuracy for more than one hour.
- To meet reliability demands, there is an emerging need for backup timing via the network, even when GPS is present.
- Quality of service (QoS) will need to improve, too. Network slicing is expected to add additional QoS granularity and will enable service providers to prioritize network packets more effectively, gain greater control of traffic flow and support new and enhanced 5G use cases across the same physical network.
- Finally, the new network, increased antennas and the proliferation of IoT will bring new security challenges that can only be addressed with end-to-end, device-to-core security solutions.

3. Increased Complexity and Low Cost of Ownership

- The new capabilities combined with the sheer magnitude of new connections driven by the increase in radio and core sites, interfaces and technologies will lead to greater complexity in the network.
- Coping with this complexity will require high levels of automation, seamless management and control, and effective cross-domain orchestration between the radio, core and transport elements of the network.

4. New Operating Model

- Market expectations of 5G are high. Personal users - and commercial and public sector organizations - expect to connect seamlessly to the new network and benefit from new connectivity, services and applications.
- But the fact is they don't expect to pay more. Therefore, service providers must also find a way to manage costs. Doing so will require greater radio efficiencies, as well as increased automation, higher productivity and more cost effectiveness built into every element of their investment.

5.11.2 5G Requirements

- 5G use cases and their requirements have already been studied by 5G research organizations, academic institutes and telecommunications firms, and published as their white papers. All of these dedicated efforts have converged to a common framework of use cases and network requirements.

1) High Speed, High Capacity

- Since LTE has been widely used worldwide already and LTE-Advanced is being developed in various nations or regions as well, certain level of demands for high speeds and high capacity communications caused by growing communication traffic as the consequence of increasing number of smartphones with enriched applications would be satisfied for the time being.

2) Massive connected devices

- Up to now, communications among people or communications between people and their targeting objects to utilize variety of service contents on servers have been main scenarios supported by communication systems. However, as has been represented by emerging Internet of things (IoT) or Machine to Machine (M2M) communications, massive number of objects will be starting to communicate each other sooner or later.

3) Ultra-low latency and ultra-high reliability

- LTE or LTE advanced has achieved short transmission latency in the order of 10 milliseconds, it is said that more drastic reduction of the latency would be required for certain use cases, for instance, tactile communications.

4) Energy Saving, Cost Saving

- Energy saving is the top priority in every industry or society in recent years, and the ICT industry is not an exemption.
- The ICT industry's energy consumption share to the whole industry increases as its growth and it cannot be regarded as marginal. Energy saving would result in cost saving as well.

5.11.3 Features of 5G

- Immersive 5G services : Virtual reality/augmented reality (VR/AR), massive contents streaming.
- Intelligent 5G services : User-centric computing, crowded area services.
- Omnipresent 5G services : Internet of things.
- Autonomous 5G services : Smart transportation, drones, robots.
- Public 5G services : Disaster monitoring, private security/public safety, emergency services.

5.11.4 Comparison of 4G and 5G

- Fig. 5.11.1 shows comparison of 4G and 5G and Table 5.11.1 show comparison.

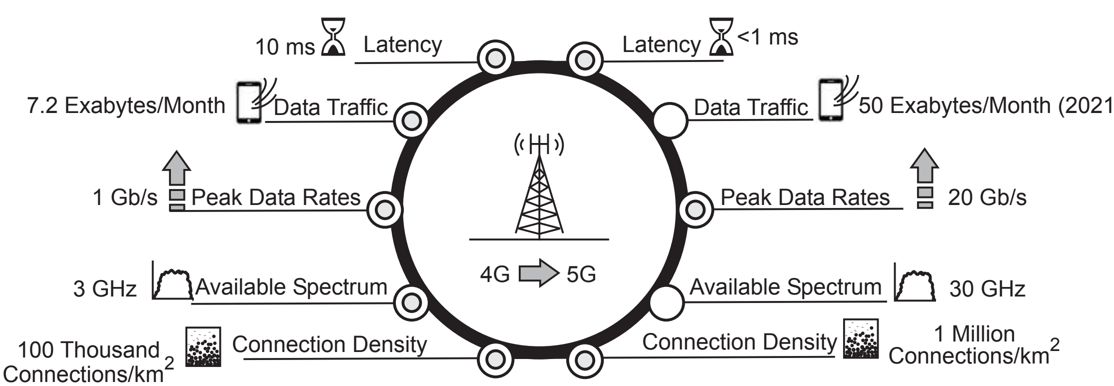

Fig. 5.11.1 Comparing 4G and 5G

5.11.5 Opportunities and Technical Requirements of 5G

1. **Enhanced Mobile Broadband (eMBB)** requiring hundreds of megahertz (MHz) of channel bandwidth using new frequencies for mobile wireless - from 2.5 gigahertz (GHz) for 4G LTE Pro and 3.5 GHz for 5G, to tens of gigahertz and beyond into the millimeter wave (mmWave) spectrum.

Parameters	4G	5G
Latency	10 ms	Less than 1 ms
Peak data rates	1 Gbps	20 Gbps
Number of mobile connections	8 billion (2016)	11 billion (2021)
Channel bandwidth	20 MHz	100 MHz below 6 GHz
	200 kHz (for Cat-NB 1 loT	400 MHz above 6 GHz.
Frequency band	600 MHz to 5.925 GHz	600 MHz-mmWave (for example 28 GHz 39 GHz and onword to 80 GHz.
Uplink waveform	Single-camer frequency division multiple access (SC-FDMA)	Option for cyclic prefix orthogonal frequency division multiplexing (CP-OFDM)
User Equipment (UE) transmitted power	+23 decibel-miliwatts (dBm) except 2.5 GHz time-division duplexing (TDD) Band 41 where + 26 dBm, HPUE is allowed loT haa a lower power-class at + 20 dBm.	+26 dBm for less than 6 GHz 5G bands at and above 2.5 GHz.

Table 5.11.1

2. **Ultra efficient** for streaming data, taking full advantage of **Carrier Aggregation (CA)** and massive multiple input/multiple output (MIMO).
3. **Fixed wireless**, giving more choices to get 20 gigabit per second (Gbps) connections to your home and business.
4. **Wireless infrastructure**, using beam steering and **high-power Gallium Nitride (GaN)**, ideally suited to adaptive-array steerable antennas.
5. **Low latency** for real-time connections enabling **autonomous vehicles** and augmented reality/virtual reality (AR/VR).
6. **Internet of Things (IoT)** connecting more than a trillion devices to the Internet in the next ten years with extremely low data rates, battery life greater than ten years, and the longest possible communication range.

Technical requirements for 5G core networks

- The core network requirements are described in three aspects to support various 5G services.
  1. Functional requirements (F),
  2. Architectural requirements (A)
  3. Operational requirements (O).
- Table 5.11.2 shows technical requirements of 5G network and its brief description.

	Technical Requirement	Brief Description
F1	Seamless mobility	Shall support seamless mobility regardless of the cell types and RATs where the macro-cell BSs, small-cell BSs, WLANAPs. and relay stations are mixed and overlapped.
F2	Wired/wireless terminal switching	Shall support terminal and/or session mobility to provide fast handover between wireless and wired terminals
F3	Context-aware best connection	Shall utilize the various context information (device, user, environment, network) to provide always best connection / service.
F4	Single ID for multiple access	Shall recognize a mobile terminal as a single entity regardless of its access network.
A1	Distributed architecture	Shall support the distributed network architecture to accommodate anticipated 1000 times of traffic.
A2	Lightweight signaling	Shall have lightweight signaling to support a variety of terminals such as massive MTC terminal.
A3	Multiple RAT inter working	Shall have architecture to support 'Flow over Multi-RAT' to provide the high volume service with low cost and guarantee the service continuity in spite of the bandwidth deficiency in a wireless access.
A4	Fine-grained location tracking	Shall have function to trace the mobile terminal location in a fine granularity in order to provide advanced location based service.
01	Flexible reconfiguration and upgrade	Shall provide virtualization environment and support to reconfigure and upgrade the core network at low cost without changing the physical network infrastructure.
02	Network on-demand	Shall be able to build the network based on the QoS/QoE, charging and service characteristics.

Table 5.11.2

RATs (radio access technologies)

5.11.6 5G Core Network Architecture

- Fig. 5.11.2 shows a software-centric 5G core and access network architecture designed to support various 5G mobile services.

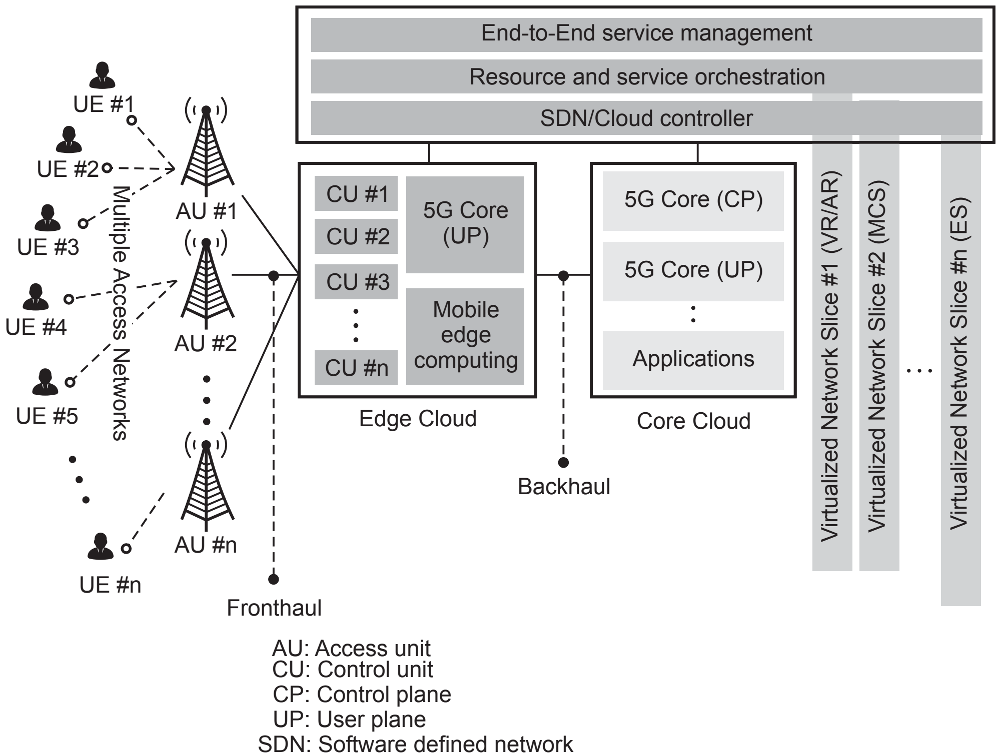

Fig. 5.11.2 Software-centric 5G core and access network architecture

- The fronthaul and the backhaul represent the interface between access units (AUs) and the edge cloud and the interface between the edge cloud and the core cloud, respectively.
- The AU-cloud unit (CU) configuration is similar to the conventional radio unit-digital unit (RU-DU) configuration of a cloud radio access network (C-RAN) in the 3GPP-LTE system.
- The legacy RUs are remote RF units located at cell-sites, and centralized DUs are connected with these RUs and have relatively heavy functionalities regarding the Medium Access Control (MAC) layer, Radio Link Control (RLC) layer and the Packet Data Convergence Protocol (PDCP) layer, compared with RUs.
- Because the fronthaul data overhead between RUs and DUs is predicted to increase explosively, several functionalities of DUs will be moved into RUs at cell-sites.

- Therefore, this modified cell-site unit in the 5G core network the AU, and CUs can be considered as the lightweight DUs.
- The Mobile Edge Computing (MEC) entity in the edge cloud provides application developers and content providers with cloud computing capabilities and a 5G service environment at the edge of the 5G mobile networks.
- The MEC technology is leveraged to support ultra-low latency and high-bandwidth services.
- In the 5G core network architecture, the separation of the Control Plane (CP) and the User Plane (UP) is one of the most important features in order to increase operational efficiency, as well as to improve network simplicity and flexibility.

Relationship between 5G services and 5G core network requirements

- Table 5.11.3 shows the relation between 5G services and 5G core network requirement.

Service Category		F1	F2	F3	F4	A1	A2	A3	A4	01	02
Innersive 5G Service	Virtual Reality and Augmented Reality	✓	✓	✓						✓	✓
	Massive Contents Streaming	✓	✓			✓				✓	✓
Intelligent 5G Service	User Centric Computing		✓	✓						✓	✓
	Growded Area service	✓		✓		✓				✓	✓
Ommipresent 5G Service	Internet of Things.		✓	✓		✓	✓			✓	✓
Autonomous 5G Service	Smart Transportation	✓	✓	✓	✓				✓	✓	✓
	Smart Drone	✓		✓					✓	✓	✓
	Smart Robot			✓			✓	✓		✓	✓
Public 5G Service	Disaster Monitoring			✓				✓	✓	✓	✓
	Private Safety and Public Security			✓				✓	✓	✓	✓
	Emergency Service	✓		✓				✓	✓	✓	✓

Table 5.11.3

5.11.7 Disruptive Technologies for 5G

- Five technologies for 5G that could lead to both architectural and component disruptions :
  1) Device-centric architectures,
  2) Millimeter wave ;
  3) Massive MIMO ;
  4) Smarter devices ; and
  5) Native support for machine - to - machine communications.
- The key ideas for each technology are described, along with their potential impact on 5G and the research challenges that remain.

1) **Device-centric architectures :** The base - station - centric architecture of cellular systems may change in 5G. It may be time to reconsider such concepts as uplink and downlink, as well as control and data channels. 5G systems will use nodes on an ad hoc basis.

2) **Millimeter wave (mmWave) :** While spectrum has become scarce at microwave frequencies, it is plentiful in the mmWave region. Such a spectrum "el Dorado" has led to an mmWave "gold rush". Although far from being fully understood, mmWave technologies have already been standardized for short-range services (IEEE 802.11ad) and deployed for niche applications such as small-cell backhaul.

3) **Massive MIMO :** Massive multiple-input multiple-output (MIMO) proposes using a very large number of antennas to spatially multiplex data. Massive MIMO may require major architectural changes, particularly in the design of macro base stations, and it may also lead to new types of deployments.

4) **Smarter devices :** 2G-3G-4G cellular networks were built under the design premise of having complete control at the infrastructure side. The proposal is for 5G systems to drop this design assumption and exploit intelligence at the device side within different layers of the software protocol stack. For example, one could allow device-to-device (D2D) connectivity or exploit smart caching at the mobile smart-phone side. While this design philosophy mainly requires a change at the node level (component change), it also has implications at the architectural level.

5) **Native support for machine-to-machine (M2M) communication :** A native inclusion of M2M communication in 5G has three main requirements : Support of a massive number of low-data-rate devices, sustaining a minimal data rate in virtually all circumstances, and very-low-latency data transfer. Addressing these requirements in 5G requires new methods and ideas at both the component and architectural levels.

5.12 WLAN

- A wireless local-area network (WLAN) provides the features and benefits of traditional LAN technologies such as Ethernet and Token ring without the limitations of wires or cables.
- A WLAN system is different from a traditional wired LAN in many ways : The destination address is not equivalent to a physical location, WLANs deal with fixed, portable and mobile stations and of course, the physical layers used here are fundamentally different from wired media.

Advantages of WLAN

1. Light of sight is not required for propagation.
2. Installation is quick and easy, and can eliminate the need to pull cable through walls / ceiling and wall / ceiling.
3. The signal is not blocked by any objects like buildings, trees etc.
4. Easier to add or move workstations.
5. High data rate, as coverage area is small.
6. Reliable type of communication
7. Economical for a small area access.
8. Fast setup : If your computer has a wireless adapter, locating a wireless network can be as simple as clicking "Connect to a Network" - in some cases ; you will connect automatically to networks within range.
9. Wireless transmission provides great flexibility.
10. Expandability : Adding new computers to a wireless network is as easy as turning the computer on (as long as you do not exceed the maximum number of devices).

5.12.1 IEEE 802.11

- For the wireless LAN a standard defined by IEEE is referred to as IEEE 802.11.
- IEEE 802.11 has variants like 802.11b, 802.11a, 802.11e, and 802.11g. The IEEE 802.11 specifications are wireless standards that define an "over-the-air" interface between a wireless client and a base station or access point, as well as among wireless clients.
- In WLANs, the **Medium Access Control (MAC)** protocol is the main element that determines the efficiency of sharing the limited communication bandwidth of the wireless channel. The fraction of channel bandwidth used by successfully transmitted messages gives a good indication of the protocol efficiency, and its maximum value is referred to as protocol capacity.

5.12.2 Overview of IEEE 802.11

- IEEE 802.11 specifically devoted to wireless LANs. In 1997 the first IEEE802.11 standards (1 and 2Mb/s) was completed. After that a number of its variants are evolving gradually.
- The aim of the 802.11 standard was to develop a MAC and PHY layer for wireless connectivity for fixed, portable and moving stations within a local area. The higher OSI-layers are the same as in any other 802.X standard ; this means that at this level there is no difference perceptible between wired and wireless media.
- The standard defines the MAC procedures to support the asynchronous MAC service data unit (MSDU) delivery services ; several PHY signaling techniques and interface functions that are controlled by the IEEE802.11 MAC.
- The 802.11 standard provide MAC and PHY functionalities for wireless connectivity of fixed, portable, and moving stations moving at pedestrian and vehicular speeds within a local area.
- Specific features of the IEEE 802.11 standard include the following :
  1. Support of asynchronous and time-bounded delivery service
  2. Continuity of service within extended areas via a distribution system.

5.12.3 Requirements of IEEE 802.11

- To be IEEE 802.11 standard compatible a device should fulfill the following requirements :
  1. Single MAC supporting multiple PHYs
  2. Mechanisms to allow multiple overlapping networks in the same area
  3. Provisions to handle the interface from other ISM band radios and microwave ovens.
  4. Mechanisms to handle "hidden" terminal.
  5. Options to support time-bounded services.
  6. Provisions to handle privacy and access security.

5.12.4 Reference Architecture

- There are two operation modes defined in IEEE 802.11 : **Infrastructure Mode** and **Ad Hoc Mode**.

1. Infrastructure mode

- In infrastructure mode, the wireless network consists of at least one access point (AP) connected to the wired network infrastructure and a set of wireless end stations. An access point controls encryption on the network and may bridge or route the wireless traffic to a wired Ethernet network (or the Internet).

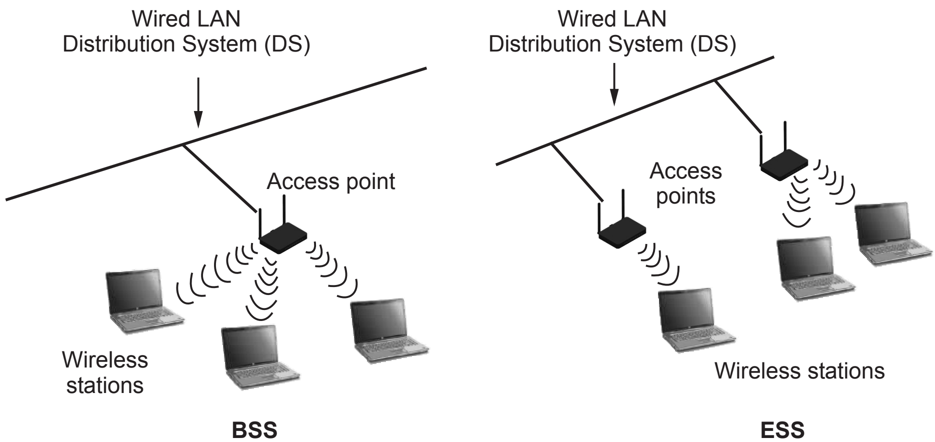

Fig. 5.12.1 Infracture mode

2. Ad-Hoc Mode

- Ad-Hoc mode is a set of 802.11 wireless stations that communicate directly with each other without using an access point or any connection to a wired network. Ad-Hoc Mode is also called peer-to-peer mode or an Independent Basic Service Set (IBSS) as mentioned earlier.

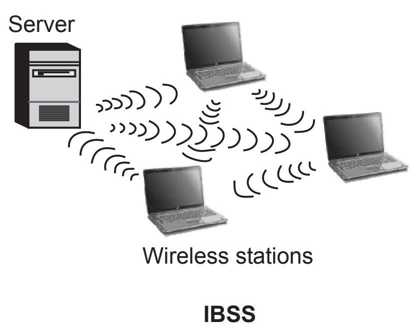

Fig. 5.12.2 Ad - Hoc Mode

5.12.5 Layered Protocol Architecture

- Fig. 5.12.3 shows the protocol stack of IEEE 802.11 standard. The traditional MAC sublayer is divided into MAC sublayer and MAC management sublayer entities.
- The MAC sublayer performs : Access mechanism, fragmentation and reassembly of packets.

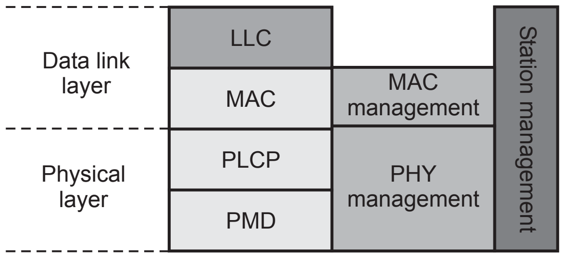

Fig. 5.12.3 Protocol entities for the IEEE 802.11

- The MAC layer management performs : Roaming in ESS, power management, association, disassociation and reassociation.
- The PHY layer is divided into three sublayer :
  1. **PHY layer convergence protocol (PLCP) :** Performs carrier sensing assessment and packet forming for different PHY layers.
  2. **PHY medium dependent (PMD) protocol :** Specifies the modulation and coding techniques for signalling with the medium.
  3. **PHY layer management sublayer :** Decides on channel tuning for PHY layers.

5.12.6 IEEE 802.11 Services

- The IEEE 802.11 defines a set of services that provides the functionality needed to let the LLC layer send and receive MSDUs (MAC Service Data Units).
- The services includes the following : Authentication, deauthentication, privacy, MSDU delivery, association, disassociation, distribution, integration and reassociation.
- Following table shows IEEE802.11 services, provider and the function for which services used to support.

Service	Provider	Used to support
Association	Distribution system	MSDU delivery
Authentication	Station	LAN access and security
De-authentication	Station	LAN access and security
Disassociation	Distribution system	MSDU delivery
Distribution	Distribution system	MSDU delivery

Integration	Distribution system	MSDU delivery
MSDU delivery	Station	MSDU delivery
Privacy	Station	LAN access and security
Reassociation	Station	MSDU delivery

- Authentication and deauthentication deals with the station authenticating itself to the network, when joining and leaving the network. Authentication is necessary because the network is wireless which means that it is very easy to eavesdrop.
- The IEEE 802.11 standard defines two kinds of authentication : *Open system authentication* where the station wishing to join the network sends an authentication management frame containing its identification, if the station is known it is accepted otherwise it is not and shared key authentication where a shared secret key has to be known by the station trying to join the network. To use shared Key Authentication, implementation of the Wired Equivalent Privacy (WEP) algorithm is required.
- Privacy protects the network from eavesdropping. This is necessary because wireless networks are very easy to listen in on, the eavesdropper does not even have to be in the building. Privacy is obtained through the use of the Wired Equivalent Privacy (WEP) algorithm.
- WEP is designed to provide at least the same level of security as wired networks. WEP allows different ciphers to be used as long as a consensus on the network is reached of course. If the wireless network is an ESS, which means that a distribution system is present, additional steps has to be taken. In this case a station has to associate with the access point it is currently using to let the distribution system properly map its location. When the station leaves the access point it will disassociate from the access point.
- The distribution service concerns itself with moving packets from one BSS to another. The distribution has to be aware of the proper destination BSS for the packets that it receives.
- Because the characteristics of the distribution system is unspecified by the IEEE 802.11 standard, integration is needed. This is acquired through the use of a portal that converts IEEE 802.11 packets to the format of the distribution system and back.
- When a station moves from one BSS to another it needs to reassociate in order to receive packets properly. The reassociation service allows a station to change from one access point to another. Reassociation is always initiated by the mobile station.

Comparison of Re - association and Dissociation

Sr. No	Re-association	Dissocaition
1.	The re-association service is used when a MS moves from one BSS to another within the same ESS.	The dissociation service is used to terminate an association.
2.	It is always initiated by the MS.	It may be invoked by either party to an association (the AP or the MS)
3.	It enables the distribution system to recognize the fact that MS has moved its association from one AP to another.	It is a notification and not a request. It cannot be refused. MSs leaving a BSS will send dissociation message to the AP which need not be always received.

5.12.7 IEEE 802.11 Medium Access Control

- The IEEE 802.11 MAC layer covers three functional areas :
  1. Reliable data delivery,
  2. Access control, and
  3. Security

5.12.7.1 Reliable Data Delivery

- A wireless LAN using the IEEE 802.11 physical & MAC layers is subject to unreliable.
- Noise, interference and other propagation effects result in loss of significant no. of frames. This situation can be dealt with by reliability mechanisms at a higher layer, such as TCP.
- For this purpose, IEEE 802.11 includes a frame exchange protocol.
- Frame exchange protocol -
  1. Source station transmits data
  2. Destination responds with acknowledgment (ACK)
  3. If source doesn't receive ACK, it retransmits frame

5.12.7.2 Access Control

- IEEE 802.11 considered 2 types of MAC algorithm :
  1. Distributed access protocols
  2. Centralized access protocols.
- End result for 802.11 is a MAC algorithm called DFWMAC (Distributed Foundation Wireless MAC).

- Fig. 5.12.4 shows IEEE 802.11 protocol architecture.

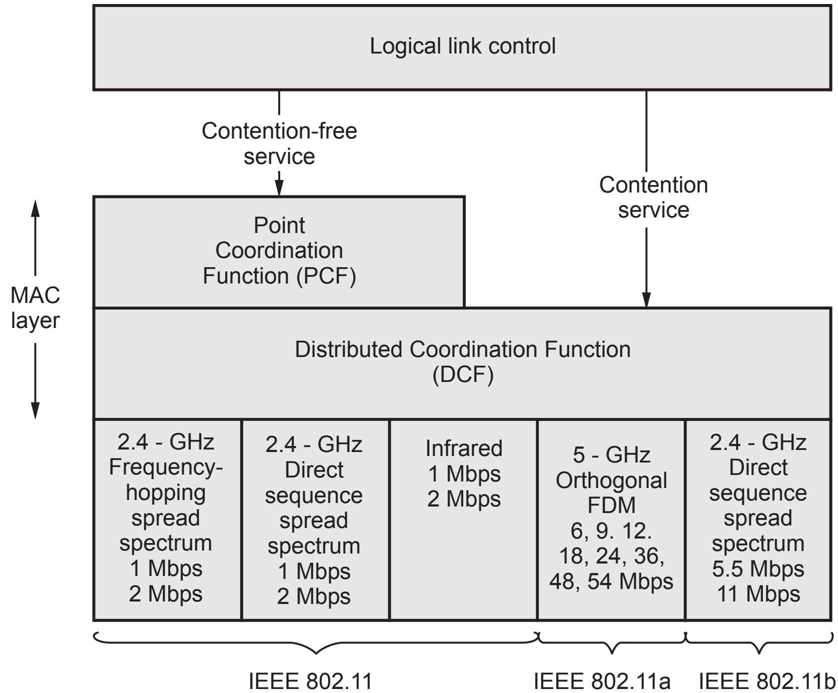

Fig. 5.12.4 IEEE 802.11 architecture

Distributed Coordination Function

- DCF makes use of simple CSMA algorithm.
  a) If a station has MAC frame to transmit, it listens to the medium.
  b) If the medium is idle, station may transmit.
  c) Otherwise it must wait until current transmission is complete.
- DCF does not include a collision detection function.
- To ensure smooth and fair functioning of this algorithm, DCF includes a set of delays that amounts to a priority scheme.
- Let us consider a single delay known as an Inter Frame Space (IFS). (Refer Fig. 5.12.5 on next page)

3. Different Interframe Space (IFS) values

1. SIFS (Short IFS) : The shortest IFS, Used for immediate response actions.
2. PIFS (Point Coordination Function IFS) : A mid-length IFS, Used by centralized controller in the PCF scheme.
3. DIFS (Distributed Coordination Function IFS) : The longest IFS, used as a minimum delay for asynchronous frames.

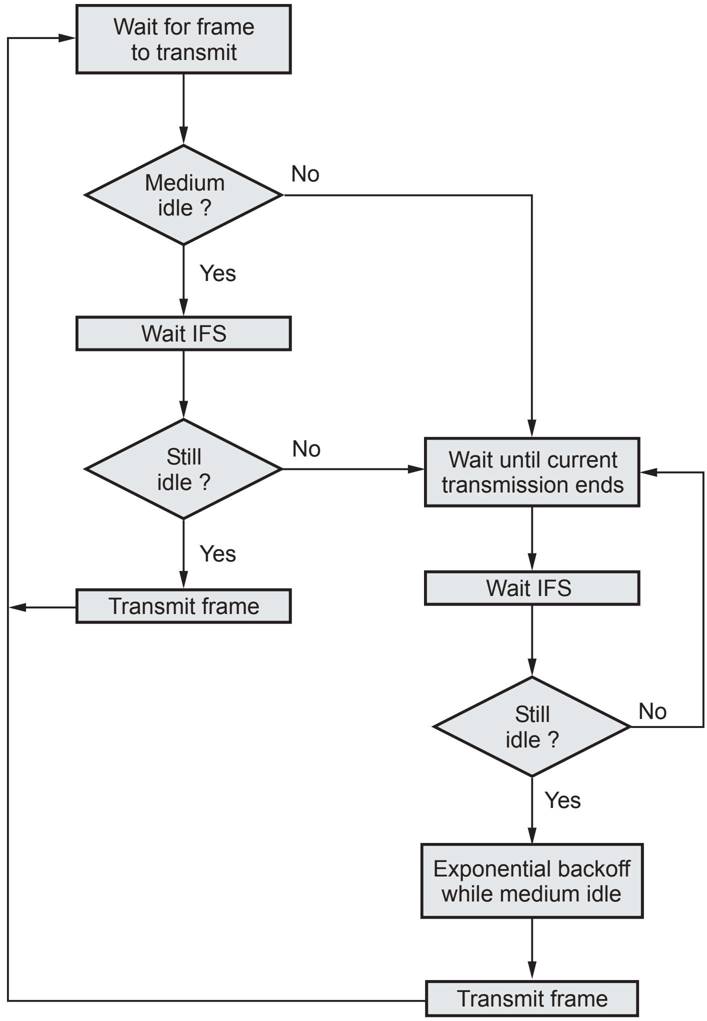

Fig. 5.12.5 IEEE 802.11 Medium access control logic

Interframe Space (IFS) Usage

1. **SIFS (Short IFS) :**
   a) Acknowledgment (ACK)
   b) Clear to send (CTS)
   c) Poll response
2. **PIFS (Point Coordination Function IFS)** :
   a) Used by centralized controller in issuing polls
   b) Takes precedence over normal contention traffic

3. **DIFS (Distributed Coordination Function IFS) :**
   a) Used for all ordinary asynchronous traffic

5.12.8 Hidden and Exposed Station Problems in WLAN

- Hidden terminal problems are refers to the collision of a packet at a receiving node due to the simultaneous transmission of those nodes that are not in the direct transmission range of the sender, but are within the transmission range of the receiver.
- Collision occurs when both nodes transmit packets at the same time without knowing about the transmission of each other.
- Fig. 5.12.6 shows the hidden and exposed terminal problems in WLAN

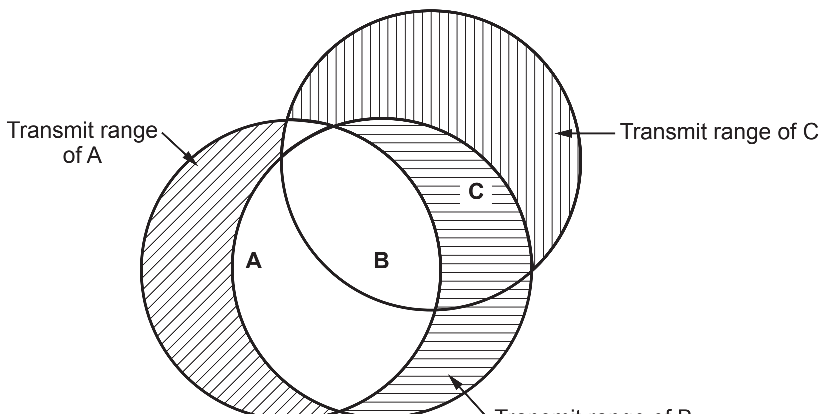

Fig. 5.12.6 Hidden and exposed terminal in WLAN

Hidden Terminal Problem

- Node B is within the range of nodes A and C, but A and C are not in each other's range. Let us consider the case where A is transmitting to B. Node C, being out of A's range, cannot detect carrier signal and may therefore send packet to B, thus causing a collision at B. This is referred to as the **hidden-terminal problem,** as nodes A and C are hidden from each other.

Exposed Terminal Problem

- The exposed terminal problem refers to the inability of a node, which is blocked due to transmission by a nearby transmitting node, to transmit to another node. Let us now consider another case where B is transmitting to A. Since C is within B's range, it senses carrier signal and decides to defer its own transmission.

- However, this is unnecessary because there is no way C's transmission can cause any collision at receiver A. This is referred to as the **exposed-terminal problem**, since B being exposed to C caused the latter to needlessly defer its transmission.

5.13 Short Answered Questions

Q.1 Define hidden terminal.

Ans. :
- The transmission range of A reaches B but not C. The transmission range of C reaches B but not A. B reaches A and C.
- A cannot detect C and vice versa. A starts sending to B, but C does not receive this transmission. C also wants to send something to B and senses the medium.
- The medium appears to be free, the carrier sense fails. C also starts sending, causing a collision at B. But A can't detect this collision at B and continues with its transmission. A is hidden for C and vice versa.

Q.2 What is near far problem ?

Ans. : At the receiver, the signals may come from various multiple sources.
- The strongest signal usually captures the modulator. The other signals are considered as noise.
- Each source may have different distances to the base station.

5.14 Multiple Choice Questions

Q.1 Hidden terminals in Aloha is _______.

a) Do not exist, as the scheme prevents their existence.

b) Wait for a network signal to start transmission.

c) Do not care about other terminals and may cause collisions.

d) DO NOT WAIT FOR A NETWORK SIGNAL TO START transmission.

Q.2 In TDMA, interference happens if _______.

a) Senders transmit data at the same time.

b) Senders do not transmit data at the same time.

c) Senders transmit data at the same frequency.

d) Senders do not transmit data at the same frequency.

Q.3 According to the IEEE Project 802.11, there are two types of wireless LAN. In an infrastructured-based network, what is a BSA (Basic Service Area) ?

a A BSA is a wireless station.

b A BSA is a gateway which connects a wireless station to a network.

c A BSA is simply a cell.

d A BSA is another word for server.

Q.4 What is the access point (AP) in wireless LAN ?

a Device that allows wireless devices to connect to a wired network

b Wireless devices itself

c Both (a) and (b)

d None of the mentioned

Q.5 Mostly ________ is used in wireless LAN.

a time division multiplexing.

b orthogonal frequency division multiplexing.

c space division multiplexing.

d none of the mentioned.

Answer Keys for Multiple Choice Questions :

Q.1	c	**Q.2**	a	**Q.3**	c	**Q.4**	a	**Q.5**	b

❑❑❑

UNIT VI

6 Performance Analysis Issues

Syllabus

Introduction to Network coding, basic hamming code and significance of Information Theory. Interference suppression and Power control. MAC layer scheduling and connection admission in mobile communication.

Contents

6.1 Network Coding

- Coding techniques are used to increase the transmitted signal immunity to radio channel noise and other impairments including frequency fading and multipath spread.
- In digital environment coding techniques helps reducing bit errors and frame errors.
- Specific error correction and detection codes are employed to reduce required number of requests for retransmission by the system.
- Error coding is a method of detecting and correcting these errors to ensure information is transferred intact from its source to its destination.
- Error coding is used for fault tolerant computing in computer memory, magnetic and optical data storage media, satellite and deep space communications, network communications, cellular telephone networks, and almost any other form of digital data communication.
- Error coding uses mathematical formulas to encode data bits at the source into longer bit words for transmission. The "code word" can then be decoded at the destination to retrieve the information.
- The extra bits in the code word provide redundancy that, according to the coding scheme used, will allow the destination to use the decoding process to determine if the communication medium introduced errors and in some cases correct them so that the data need not be retransmitted.
- Different error coding schemes are chosen depending on the types of errors expected, the communication medium's expected error rate and whether or not data retransmission is.
- There are two major types of coding schemes : linear block codes and convolutional codes.
- Linear block codes are characterized by segmenting a message into separate blocks of a fixed length and encoding each block one at a time for transmission. Convolutional codes encode the entire data stream into one long code word and transmit it in pieces.

6.1.1 Error Detection and Correction Coding

- Error Control Coding (ECC) is the term used to denote a technique that codes the transmitted bits in a way that attempts to control the overall bit rate.

- Different error detection and correction codes used in wireless system are -
  1. Block codes 2. Convolutional codes 3. Turbo codes

1. Block Codes

- The basic purpose of a block code is to determine whether or not an error has occurred during transmission.
- Block codes may be used to determine whether an error has occurred during data transmission. Schemes that use block codes to correct errors that might have occurred during data transmission are known as forward error correction.

2. Convolutional and Turbo Encoders

- Convolutional codes are generally more complicated than linear block codes, more difficult to implement, and have lower code rates (usually below 0.90), but have powerful error correcting capabilities.
- They are popular in satellite and deep space communications, where bandwidth is essentially unlimited, but the BER is much higher and retransmissions are infeasible.
- Convolutional codes are more difficult to decode because they are encoded using finite state machines that have branching paths for encoding each bit in the data sequence.
- A well-known process for decoding convolutional codes quickly is the Viterbi Algorithm. The Viterbi algorithm is a maximum likelihood decoder, meaning that the output code word from decoding a transmission is always the one with the highest probability of being the correct word transmitted from the source. possible.
- A continuous stream of bits is mapped into output stream that possesses redundancy. The redundancy introduced depends upon incoming bits and several of preceding bits.
- The number of preceding bits in encoding process is called constraint length (K). The ratio of input bits to output bits is called as code rate (R) of encoder.
- Convolutional encoder (with K = 9 and R = ½) for cdma2000 is shown in Fig. 6.1.1 (See Fig. 6.1.1 on next page.)

3. Turbo Encoder

- Turbo encoders are modified form of combined Convolutional encoders. Turbo encoders are a new enhanced error correction codes.

- A typical turbo codes consists of two symmetric, recursive Convolutional encoders connected in parallel with an interleaver.

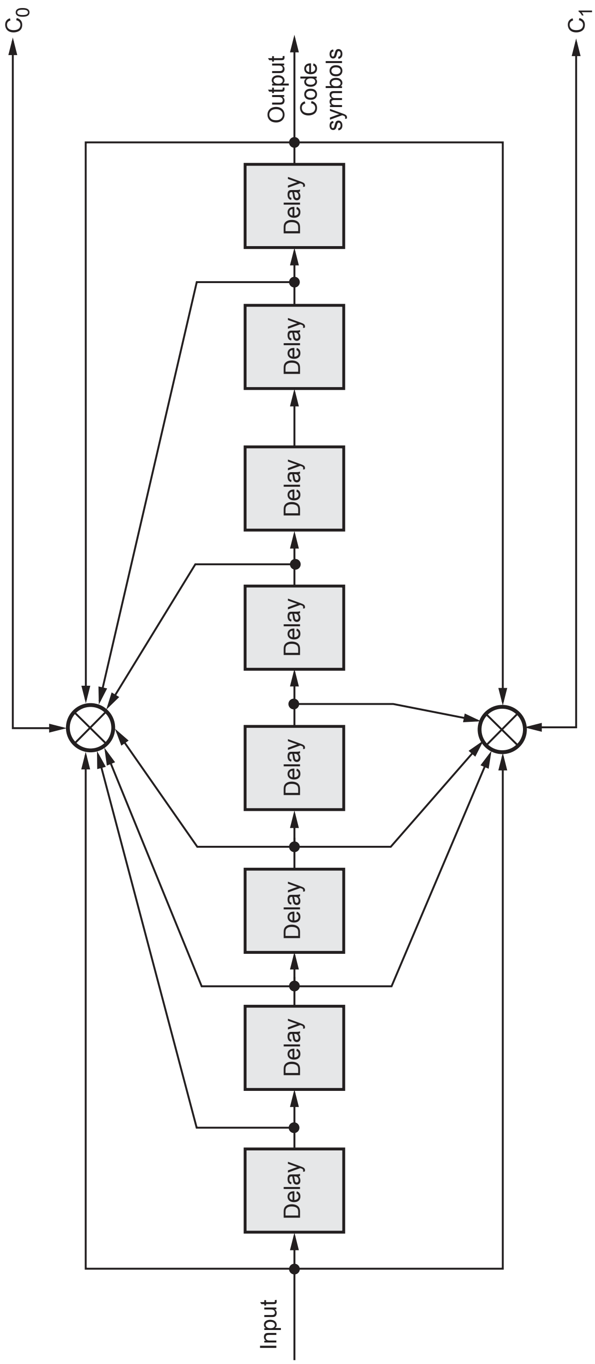

Fig. 6.1.1 Convolutional encoder

6.1.2 Block Interleaving

- It is a technique used by mobile wireless systems to combat the effect of bit errors introduced during transmission of errors.
- The basic process involved in the block interleaving of data bits before transmission is as follows : the bits of one data block are interleaved with the bits of another data block.
- This process will provide an important advantage in the transmission of digital data. If a noise burst causes several bit errors to occur, the bits effected will not all belong to the same data block. Hence the errors are spread out and thus easier to correct.

6.2 Interference Suppression and Power Control

- Signals from MT operating in the coverage area of a BS cause interference to MTs operating in the coverage area of another BS.
- Power control enhances battery life of the MTs.
- In order to reduce interference and maximize the battery life, there is a need for the wireless network to keep track of radio resources, signal strength, and other associated information related to communication between an MT and neighboring BSs.

6.2.1 Power Control

- Power control means the algorithms, protocols and techniques employed in wireless networks to adjust the transmit power of MS or BS for reducing cochannel interference and near-far interference.
- Controlling the transmit power of MSs and BSs is important. By reducing the transmit power, the mobile terminal will save on its battery life.
- When power control is applied properly, the quality of communication is improved by increasing Signal to Interference Ratio (SIR). Also, system capacity will be improved.
- If Signal to Interference Ratio (SIR) is increased, it means that lower frequency reuse can be employed.

Power Saving Mechanism

- Power saving mechanisms is employed to save the battery life of mobile terminal by making MS in suspended or semi-suspended mode of operation with limited capabilities.

Types of Power Control

1. Open-loop power control 2. Closed-loop power control

6.2.1.1 Open-loop Power Control

- The Open-loop power control depends solely on mobile unit.
- There is no feedback from BS.
- The open-loop control is not as accurate as close-loop control, but can react quickly to fluctuations in signal strength.
- The transmit powers of MS and BS dynamicaly changing because of fading,velocity of mobile and distance from BS.
- Open loop power control is usually implemented on the uplink or reverse link.
- An MS measures the quality of a reference channel from BS; Received Signal Strength (RSS) or BER frames may be used.
- If the RSS or BER above certain thresholds , the mobile will reduce its transmit power and vice versa.

Disadvantages of open loop power control

1. The decision is based on the measurement of the downlink
2. These channels are not usually correlated, which means a good signal reception on the uplink channel does not necessarily mean the same at the downlink channel.
3. There may be significant delay before implementing the power control
4. In TDMA, the MT reception and transmission times are different, and there will be a lag time in implementing time open loop control.

6.2.1.2 Closed Loop Power Control

- The closed-loop power control adjusts the signal in reverse channel based on metric of performance.
- BS makes power adjustment decision and communicates to on control channel.
- Closed loop power control eliminates the disadvantage of open loop power control by implementing a feedback mechanism between the BS and MS.
- The BS measures the quality of the signal received from the MS and indicates what actions the MS should take via control signalinh on the downlink channel.
- Closed loop power control can be used to control the transmit signal of the BS.

6.2.1.3 Centralized and Distributed Power Control

- In Centralized Power Control (CPC), a central controller in the BSC or MSC has full information of the radio links in the system.
- The scheme is extremely hard to implement because the centralized controller has to keep track of all the links in the system and compute the power for each MS.
- In Distributed Power Control (DPC), the mobile adjust their transmit power in discrete step.
- Power adjustment made by MSs result in the transmit powers iteratively converging to the optimum power control solution.

6.2.1.4 Power Saving Mechanism in Wireless Networks

- The most significant amount of power consumed in the transmission process.
- The second most consumed power in the reception process.
- The least consumed power in the standby process.
- The operation of wireless network is designed to ensure that the mobile station spends most of the time either in standby or sleep mode in order to conserve power.

Discontinuous Transmission and Repetition at Lower Transmit Powers

- Discontinuous transmission is mostly employed in cellular telephone networks with hardware and algorithms are used to detect the presence or absence of voice.
- Use of Voice Activity Detector (VAD) makes it possible for an MS to behave differently when no voice activity is present. While a VAD's performance does not affect clarity directly.

Problems of VAD

- If it is not operating correctly, it can certainly decrease the intelligibility of voice signals and overall conservation quality.

Sleep Modes

- Sleep mode is the common approach for saving battery power of MS.
- MS enters into a sleep mode during periods of inactivity.
- Activate the MS when needed. Shutting off the MS when not needed.

6.3 MAC Sublayer

- In WLANs, the Medium Access Control (MAC) protocol is the main element that determines the efficiency of sharing the limited communication bandwidth of the wireless channel.

- The fraction of channel bandwidth used by successfully transmitted messages gives a good indication of the protocol efficiency, and its maximum value is referred to as protocol capacity.
- The MAC layer is divided into **MAC sub layer** and **MAC management sub layer** entities. In addition IEEE 802.11 specifies a station management sub layer that is responsible for coordination of the interactions between MAC and PHY layers.

Major responsibilities of MAC sub layer

1. Define access scheme
2. Define packet formats

Major responsibilities of management sub layer

1. Support ESS
2. Power management
3. Security

- To access the medium, IEEE 802.11 provides different access schemes :
  1. CSMA/CA - Contention data
  2. RTS/CTS - Contention-free
  3. PCF - Contention-free, intended for time-bounded traffic
- First two modes are used by DCF function for channel accessing.

6.3.1 IFS (Inter Frame Spacing)

- To allow coordination of a number of options for the MAC operations, IEEE 802.11 recommends three Inter-Frame Spacings (IFSs) between the transmissions of the packets.
- These IFS's periods provide a-mechanism-for assigning priority, which is used for implementation of QOS support for time-bounded or other applications.
- After completion of each transmission, all terminals having information packets wait for one of the three IFS periods according to the level of priority of their information packet.

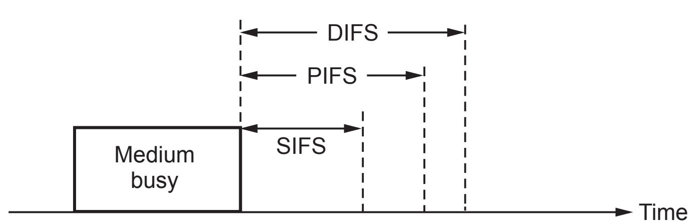

Fig. 6.3.1 Different IFS in MAC access protocol

DIFS : DCF-IFS (DIFS) is used for contention data spacing that has the lowest priority and longest duration.

SIFS : Short IFS (SIFS), used for highest priority packets such as ACK and CTS (clear to send), that has the lowest duration of time.

PIFS : The PCF IFS (PIFS), designed for PCF operation, has the second priority rate with duration between DIFS and SIFS.

- The carrier sensing mechanism in IEEE802.11 is more complicated than the IEEE802.3.Here it is performed in two ways :
  1. Physically (in the PHY layer)
  2. Virtually (in the MAC layer)
- In CSMA/CA, as the MAC has packet to transmit, it senses the channel for availability both physically and virtually.
- If the channel is virtually busy because a NAV signal is turned on, the operation is delayed until the NAV signal has disappeared. When the channel is virtually available, the MAC layer senses the PHY condition of the channel.
- If the channel is detected idle, the DIFS interval (i.e., 50 μs for 802.11b networks) starts, after waiting for the DIFS time the station starts packet transmission.
- Otherwise, the station continues to monitor the channel for busy or idle status. After finding the channel idle for a DIFS interval, the station:
  a) Starts to treat time in units of slot time.
  b) Generates a random back-off interval in units of slot time depending on the contention window.
  c) Continues to monitor whether the channel is busy or idle.
- In the latter step, for each slot time where the channel remains idle, the back-off interval is decremented by one.
- Here the slot time duration is at least the time required for a station to detect an idle channel plus the time required switching from listening mode to transmission mode. When the interval value reaches zero, the station starts packet transmission.
- During this back-off period, if the channel is sensed busy in a slot time, the decrement of the back-off interval stops (i.e. frozen) and resumes only after the channel is detected idle continuously for the DIFS interval and the following one slot time.
- Again packet transmission is started when the back-off interval reaches zero. The back-off mechanism helps avoid collision since the channel has been detected to be busy recently.

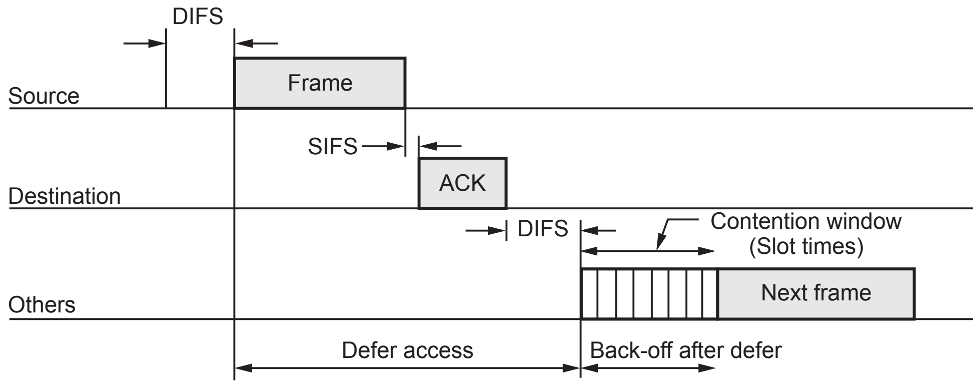

Fig. 6.3.2 CSMA/CA back-off algorithm

- To avoid channel capture, a station must wait for a back-off interval between two consecutive new packet transmissions, even if the channel is sensed idle in the DIFS interval.

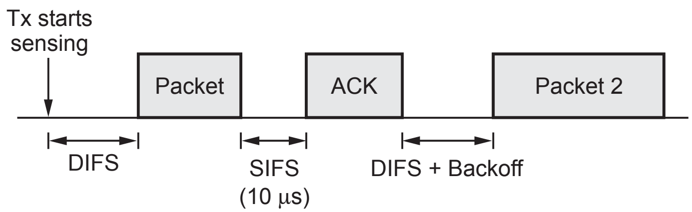

Fig. 6.3.3 DCF with channel capture avoidance

- The choice of the random number for the back-off timer is based on the binary exponential back-off algorithm, where a station chooses any of the numbers between 0 and CW-1 randomly with equal probability.
- The Contention Window (CW) is set to be CW_{min} for every new data frame transmission. CW is doubled each time when the transmission is unsuccessful, until it reaches CW_{max}, then it remains at CW_{max}.
- To determine whether a data frame transmission is successful, a positive Acknowledgement (ACK) is transmitted by the receiver. ACK is transmitted after a short inter-frame space (SIFS) period when successfully receiving the entire data frame.
- If ACK is not detected within a SIFS period after the completion of the data frame transmission, the transmission is assumed to be unsuccessful, and a retransmission is required.

6.4 MAC Layer Scheduling and Connection Admission in Mobile Communication

- Scheduling, Connection Admission Control (CAC) and traffic policing are the major issues to ensure QoS. In standard, scheduling and admission control are kept as open issues.
- Scheduling is one of the popular methods to distribute resources in wireless access networks. By scheduling, each user is able to access a specific radio resource in a given period of time.
- The scheduling strategies in wireless networks can be divided into channel-unaware, channel-aware, and energy-aware types.

Classification of scheduling problems

- Scheduling problems can be classified as -
  1. Channel unaware schemes
  2. Channel aware schemes :

     A) Opportunistic scheduling

     B) Deterministic scheduling
  3. Energy aware schemes
- Scheduling can be exploited to enhance energy efficiency in the wireless networks. In LTE networks, enhancing energy efficiency solutions can be deployed at both eNodeB and User Equipments (UEs).
- The energy efficiency can be improved when the resource allocation is implemented in time domain.

Admission Control

- Admission control is the ability of a network to control admission of new traffic based on the availability of resources.
- As per the specification the CAC considers minimum reserved rate of a connection as an admission criterion, in which the system can admit more connections, but packets of admitted connection may encounter large delays.
- Average data rate (avg-rate CAC) and maximum sustained rate (max-rate CAC) of the connections are admission criteria in CAC, along with minimum reserved rate (min-rate CAC).

6.5 Short Answered Questions

Q.1 Define network coding.

Ans. : Network coding :

- Network coding is a networking technique where operations, which in practice tend to be algebraic algorithms, are performed on data as it passes through the nodes within a network.
- NC is used to mitigate this by merging relevant messages at the relay node, using a given encoding, then forwarding this accumulated result to the destination for decoding.

Q.2 Define energy management in wireless network.

Ans. : Energy management :

- Energy management deals with the process of managing energy resources by means of controlling the battery discharge, adjusting the transmission power, and scheduling of power sources so as to increase the lifetime of the nodes of an ad hoc wireless network.
- Efficient battery management, transmission power management, and system power management are the three major means of increasing the life of a node.
- Battery management is concerned with problems that lie in the selection of battery technologies, finding the optimal capacity of the battery, and scheduling of batteries, that increase the battery capacity.
- Transmission power management techniques attempt to find an optimum power level for the nodes in the ad hoc wireless network.

6.6 Multiple Choice Question

Q.1 802.11 wireless networking uses what method as the media access method ?

a CSMA / CD

b CTS / RTS

c CSMA / CA

d CSCD / CA

Answer Key for Multiple Choice Question :

Q.1	c		

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