UNIT I
WIRELESS CHANNELS

Syllabus:

Large scale path loss – Path loss models: Free Space and Two-Ray models - Link Budget design – Small scale fading - Parameters of mobile multipath channels – Time dispersion parameters - Coherence bandwidth – Doppler spread & Coherence time, Fading due to Multipath time delay spread – flat fading – frequency selective fading – Fading due to Doppler spread – fast fading – slow fading.

INTRODUCTION TO WIRELESS COMMUNICATION

Wireless:

- ✓ Wireless is used to describe types of devices and technologies that use space as a signal propagating medium, and are not connected by a wire or cable.
- ✓ Wireless communication may be defined as the transmission of user information without the use of wires.
- ✓ The user information could be in the form human voice, digital data, e-mail messages, video and other services.
- ✓ The wireless communications covering radio, television,, radar, satellite, wireless and mobile, cellular and other wireless networks.

Advantages of wireless communications:

1. **Mobility:** The users have freedom to move about while remaining connected, as compared with the network with its coverage area.
2. **Increased reliability:** Use of wireless technology eliminate cable failures, so overall reliability
3. **Ease of installation:** Wireless communications and networks make it easier for any office to be modified with new cubicles or furniture, without worrying about providing network connectivity through cables.
4. **Rapid disaster recovery:** Accidents may happen due to fire, etc., the organization hot prepared to recover such natural disasters. So, disaster recovery plan is must for business.
5. **Low cost:** Eliminating the need to install cabling and using wireless communications results in significant cost savings.

Disadvantages of wireless communications:

1. **Radio signal interference**: Signals from other wireless devices can disturb its radio transmission.
2. **Security**: Data transmitted between the wireless device and the access point can also be encrypted in such a way that only the intended recipient can decode the message.
3. **Health hazards**: High powered levels of RF energy can produce biological damage.
4. Radio transmitters in wireless communications devices emit radio frequency (RF) energy cause adverse health effects.

Wireless network generations:

The cellular systems have been classified into three generations

i) First generation analog cellular systems
ii) Second generation digital cellular systems
iii) Third generation digital cellular systems

1G analog cellular systems:

- ✓ Amps (Advance Mobile Phone System) and ETACS (Enhanced Total Access Communication System)

2G digital cellular systems:

- ✓ IS-136, GSM (Global System for Mobile or Group Special for Mobile) and PDC (Personal Digital Cellular)

3G digital cellular systems:

- ✓ It aims to combine telephony, internet and multimedia into a single device.

Applications of wireless communications:

1. Office and household environments
2. Industry control
3. Education sector and Health service.

1.1 LARGE SCALE PATH LOSS

- ✓ The design of a communication system involves selection of values for several parameters.
- ✓ One of the important parameter is the transmit power.
- ✓ Higher transmit power ensures large allowable separation distance between the transmitter (Tx) and receiver (Rx).
- ✓ The loss in signal power per unit distance depends on the properties of the medium.
- ✓ In case of wireless communication on one hand it is desired to have a very large coverage (large allowable separation between Tx and Rx) on the other hand it is also desired that co-channel interference be as low as possible.
- ✓ In terrestrial mobile communication system, electro-magnetic wave propagation is affected by
    1. Reflection
    2. Diffraction
    3. Scattering
- ✓ These lead to dynamic variation of signal strength as a function of time, frequency, distance of separation, antenna height, antenna configuration, local scattering environment etc.
- ✓ **Propagation models** are necessary in order to predict the received signal strength for a given set of parameters as mentioned above.
- ✓ These models can be broadly classified in to two models
    1. Large scale Fading Model.
    2. Small Scale Fading Model.

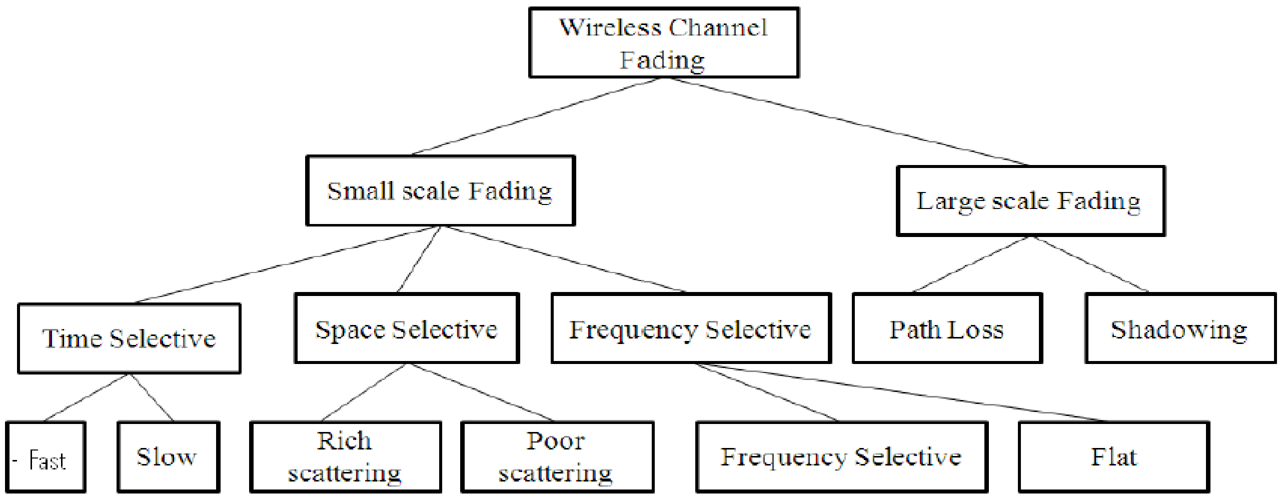

Figure 1.1 classification of fading

Large-scale propagation models

- ✓ Large-scale propagation models predict the mean signal strength for an arbitrary transmitter-receiver (T-R) separation distance.
- ✓ They are useful in estimating the radio coverage area of a transmitter.
- ✓ They characterize signal strength over large T-R separation distances.

<u>Small-scale propagation models</u>

- ✓ Small-scale (or) fading models characterize the rapid fluctuations of the received signal strength over very short travel distances or short time durations.

<u>Large scale and Small scale fading:</u>

1. **Large-scale fading**, due to path loss of signal as a function of distance and shadowing by large objects such as buildings and hills. This occurs as the mobile moves through a distance of the order of the cell size, and is typically frequency independent.
   Large Scale fading can be broadly classified as:-
   1. Path Loss.
   2. Shadowing.
2. **Small-scale fading**, due to the constructive and destructive interference of the multiple signal paths between the transmitter and receiver. This occurs at the spatial scale of the order of the carrier wavelength, and is frequency dependent.

<u>The Three Basic Propagation Mechanisms:</u>

1. **In free space propagation describe how the signals are affected by reflection, diffraction and scattering. [16m - May 2016]**
2. **Explain in brief about the three propagation mechanisms which have impact on propagation in mobile environment. [8m - May 2015, 8m -Nov 2013, 8m-Nov 2012]**
3. **Explain the different types of multipath propagation in wireless communication. [10m - Nov 2014].**
4. **In free space propagation describe how the signals are affected by reflection, diffraction and scattering. (16m) [May/June 2016]**

The three basic radio propagation mechanisms are,

1. Reflection
2. Diffraction
3. Scattering

<u>1. Reflection</u>

- ✓ Reflection occurs when a propagating electromagnetic wave impinges upon an object which has very large dimensions when compared to the wavelength of the propagating wave.
- ✓ Reflections occur from the surface of the earth and from buildings and walls.
- ✓ When the radio wave propagating in one medium impinges upon another medium having different electrical properties, the wave is partially reflected and partially transmitted.
- ✓ If the plane wave incident on a perfect dielectric, part of the energy is transmitted into the second medium, and part of the energy is reflected back into the first medium, and there is no loss of energy in absorption.
- ✓ If the second medium is a perfect conductor, then all incident energy is reflected back into the first medium without loss of energy.
- ✓ The electric field intensity of the reflected and transmitted waves may be related to the incident wave in the medium of origin through the Fresnel reflection coefficient (Γ)
- ✓ The reflection coefficient is a function of the
  - (*i*) Material properties
  - (*ii*) Wave polarization
  - (*iii*) Angle of incidence
  - (*iv*) Frequency of propagating wave

1. Reflection from Dielectrics

- ✓ When an electromagnetic wave incident at an angle θ_i with the two dielectric media, part of the energy is reflected back to the first media at an angle θ_r and part of the energy is refracted (transmitted) into the second media at an angle θ_t .
- ✓ The plane of incidence is defined as the plane containing the incident, reflected and transmitted rays.

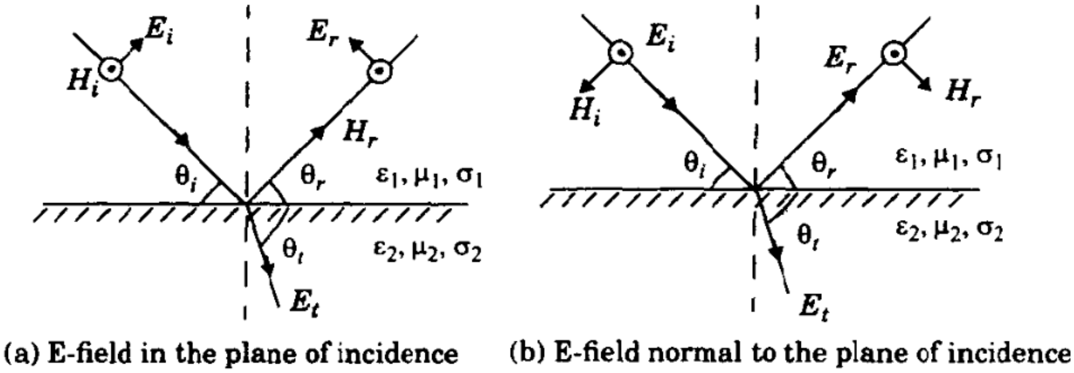

Figure 1.2: Geometry for calculating the reflection coefficients between two dielectrics

- ✓ In figure (a), the E-field polarization is parallel with the plane of incidence.

  (That is, the E-field has a vertical polarization, or normal component, with respect to the reflecting surface.)
- ✓ In figure (b), E field polarization is perpendicular to the plane of incidence.
  (That is, the incident E-field is pointing out of the page towards the reader, and is perpendicular to the page and parallel to the reflecting surface.)
- ✓ In figure, the subscripts ***i***, ***r***, ***t*** refer to the incident, reflected and transmitted fields, respectively.
- ✓ Parameters ε_1, μ_1, σ_1 and ε_2, μ_2, σ_2 represent the permitivity, permeability and conductance of the two media, respectively.

For a Perfect Lossless Dielectric Material

- ✓ The dielectric constant of a perfect (lossless) dielectric is related to a relative value of permitivity, ε_r , such that $\varepsilon = \varepsilon_0\varepsilon_r$, where ε_0 is a constant give by $8.85\times10^{-12}\ F/m$.

Permittivity $\varepsilon = \varepsilon_0\,\varepsilon_r$

$\varepsilon_0 \Rightarrow$ constant = 8.85×10^{-12} F/m

$\varepsilon_r \rightarrow$ relative permittivity

For a Lossy Dielectric Material

- ✓ If a dilelectric material is lossy, it will absorbpower and may be described by a complex dielectric constant given by
- ✓ The complex dielectric constant $\varepsilon = \varepsilon_0\,\varepsilon_r - j\,\varepsilon'$ because it absorbs power.

where,

$$\varepsilon' = \frac{\sigma}{2\pi f}$$

$\sigma \rightarrow$ Conductivity of the material

$f \rightarrow$ frequency

- ✓The terms ε_r and σ are insensitive to frequency for good conductor ε_0 and ε_r are constant with frequency but σ may be sensitive to frequency for lossy dielectrics.
- ✓A polarized wave is represented as a sum of vertical and horizontal components.

E-field in plane of incidence : $\Gamma_{\parallel} = \frac{E_r}{E_i} = \frac{\eta_2 \sin\theta_t - \eta_1 \sin\theta_i}{\eta_2 \sin\theta_t + \eta_1 \sin\theta_t}$

E-field not in plane of incidence : $\Gamma_{\perp} = \frac{E_r}{E_i} = \frac{\eta_2 \sin\theta_i - \eta_1 \sin\theta_t}{\eta_2 \sin\theta_i + \eta_1 \sin\theta_t}$

where η_i = intrinsic impedance of the i^{th} medium (i = 1, 2).

Intrinsic impedance, $\eta = \sqrt{\frac{\mu_i}{\epsilon_i}}$

Velocity $V = \frac{1}{\sqrt{\mu\epsilon}}$

The boundary conditions at the surface of incidence obey Snell's law is given by

$$\sqrt{\mu_1\epsilon_1}\,\sin(90-\theta_i) = \sqrt{\mu_2\epsilon_2}\,\sin(90-\theta_t)$$

$$\text{Incident angle } \theta_i = \theta_r \text{ reflected angle}$$

$$E_r = \Gamma E_i$$

$$E_t = E_i + \Gamma E_i = E_i(1+\Gamma)$$

Where Γ is either vertical or horizontal. Reflection coefficients for the vertical and horizontal polarization can be simplified to

$$\Gamma_{\parallel} = \frac{-\epsilon_r \sin\theta_i + \sqrt{\epsilon_r - \cos^2\theta_i}}{\epsilon_r \sin\theta_i + \sqrt{\epsilon_r - \cos^2\theta_i}}$$

and $\Gamma_{\perp} = \frac{\sin\theta_i - \sqrt{\epsilon_r - \cos^2\theta_i}}{\sin\theta_i + \sqrt{\epsilon_r - \cos^2\theta_i}}$

For $\theta_i = 0$,

$\Gamma = -1$ for both parallel and perpendicular polarization

Brewster angle

1. Define Brewster angle. [8m - May 2015, 8m - Nov 2013]

- ✓ The Brewster angle is the angle at which no reflection occurs in the medium of origin.
- ✓ It occurs when the incident angle is such that the reflection coefficient is equal to zero.
- ✓ The Brewster angle is given by

$$\sin(\theta_B) = \sqrt{\frac{\varepsilon_1}{\varepsilon_1 + \varepsilon_2}} \quad \rightarrow (1)$$

- ✓ When the first medium is free space and the second medium has a relative permittivity ε_r, equation (1) can be expressed as

$$\sin(\theta_B) = \frac{\sqrt{\varepsilon_r - 1}}{\sqrt{\varepsilon_r^2 - 1}} \quad \rightarrow (2)$$

- ✓ The Brewster angle occurs only for vertical (i.e. parallel) polarization.

b. <u>Reflection from perfect conductors</u>:

- ✓ Since electromagnetic energy pass through a perfect conductor a plane wave incident on a conductor has all of its energy reflected.
- ✓ As the electric field at the surface of the conductor must be equal to zero at all times in order to obey Maxwell's equations, the reflected wave must be equal in magnitude to the incident wave.
- ✓ For the case when E-field polarization is in the plane of incidence, the boundary conditions require that,

$$\theta_i = \theta_r \qquad \rightarrow (1)$$

and

$$E_i = E_r \quad \text{(E-field in plane of incidence)} \qquad \rightarrow (2)$$

- ✓ Similarly, for the case when the E-field is horizontally polarized, the boundary conditions require that

$$\theta_i = \theta_r \qquad \rightarrow (3)$$

and

$$E_i = -E_r \quad \text{(E-field plane of incidence)} \rightarrow (4)$$

- ✓ Referring to equations (1) to (4), we see that for a perfect conductor, $\Gamma_{\parallel} = 1$, and $\Gamma_{\perp} = 1$, regardless of incident angle.

<u>2. Diffraction</u>

- ✓ Diffraction occurs when the radio path between the transmitter and receiver is obstructed by a surface that has sharp irregularities.
- ✓ The secondary waves resulting from the obstructing surface are present throughout the space and even behind the obstacle
- ✓ It gives `rise to a bending of waves around the obstacle, even when a line-of-sight path does not exist between transmitter and receiver.
- ✓ At high frequencies depends on the geometry of the object, as well as the amplitude, phase, and polarization of the incident wave at the point of diffraction.
- ✓ Diffraction allows radio signals to propagate around the curved surface of the earth, beyond the horizon and behind obstructions.
- ✓ The received field strength decreases as the receiver moved deeper into the obstructed (shadowed) region.
- ✓ But, the diffraction field still has sufficient strength to produce useful signal.
- ✓ This phenomenon can be explained by the **Huygen's principle**, which states that all points on a wavefront acts as point sources for the production of secondary wavelets, and they combine to produce a new wavefront in the direction of propagation. The propagation of secondary wavelets in the shadowed region results diffraction. The field in the shadowed region is the vector sum of the electric field components of all the secondary wavelets that are received by the receiver.

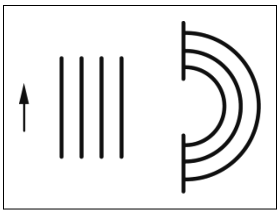

Figure 1.5: Diffraction

Fresnel Zone Geometry

- ✓ Consider a transmitter and receiver separated in free as shown in figure.
- ✓ Let an obstructing screen of height 'h' with infinite width be placed between them at a distance d_1 from the transmitter and d_2 from the receiver.
- ✓ Assuming $h << d_1, d_2$ and $h >> \lambda$, then the difference between the direct path and the diffracted path called excess path length (Δ).
- ✓ From the geometry,

$$\Delta \approx \frac{h^2}{2}\frac{(d_1 + d_2)}{d_1 d_2}$$

$$\text{Phase difference } \phi = \frac{2\pi\Delta}{\lambda} \approx \frac{2\pi}{\lambda}\; l\; \frac{h^2}{2}\;\frac{(d_1 + d_2)}{d_1 d_2}$$

- ➢ Corresponding phase difference $\alpha = \beta + \gamma$

$$\alpha \approx h\,\frac{(d_1 + d_2)}{d_1 d_2}$$

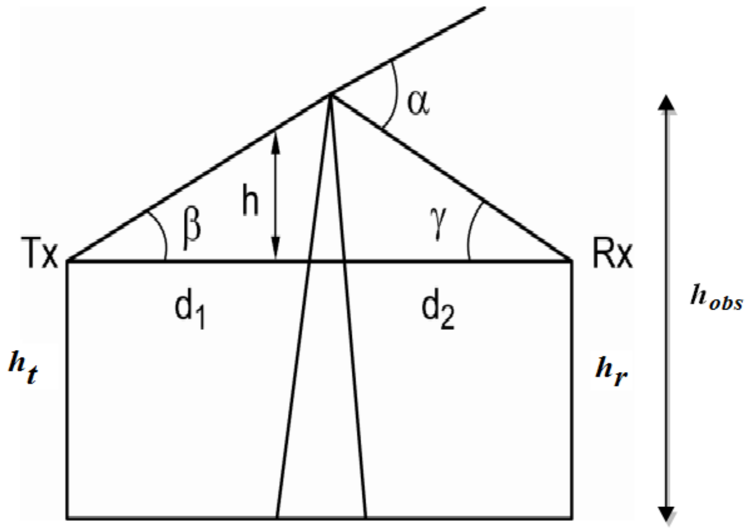

Figure 1.6: Knife edge model

The Fresnel Kirchhoff diffraction parameter is a dimensionless quantity.

- ✓ It characterizes the phase difference between two propagation paths.
- ✓ It is used to characterize diffraction losses in a general situation.
- ✓ As a general rule of thumb, we must keep the "first Fresnel zone" free of obstructions in order to obtain transmission under free space conditions.
- ✓ The concept of diffraction loss as a function of the path difference around an obstruction is explained by Fresnel zones.
- ✓ It represents successive regions, where secondary waves have a path length from the transmitter and receiver which are $\frac{n\lambda}{2}$ greater than the total path length of a line of sight path.
- ✓ The concentric circles on the plane represent the loci of the origins of the secondary wavelet. The circles are called Fresnel zones.

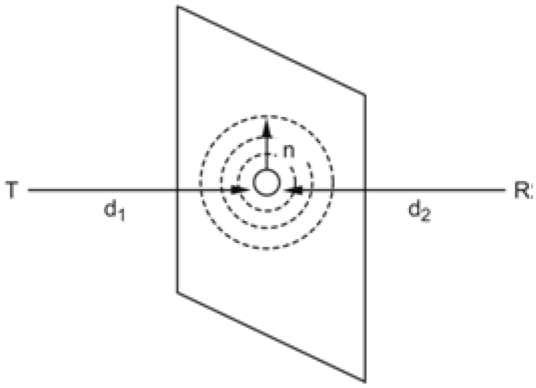

Figure 1.7: Fresnel zones

- ✓ The successive Fresnel zones have the effect of alternately providing constructive and destructive interference to the total received signal.
- ✓ The radius of the n[th] Fresnel Zone circle is denoted by r_n and can be expressed in terms of n, λ, d_1 and d_2.

$$r_n = \sqrt{\frac{n\, d_1\, d_2\, \lambda}{d_1 + d_2}}$$

- ✓ The excess total path length traversed by a ray of each circle is $\frac{n\lambda}{2}$.
- ✓ The received energy will be a vector sum of the energy contributions from all unobstructed Fresnel zones.

- ✓ The diffraction loss occurs from the blockage of secondary waves such that only a portion of energy is diffracted around an obstacle.
- ✓ That is, an obstruction causes a blockage of energy from some of the Fresnel zones, thus allowing only some of the transmitted energy to reach the receiver.
- ✓

Knife Edge Diffraction Model

- ✓ When shadowing is caused by a single object such as hill or mountain, the attenuation caused by diffraction can be estimated by treating the obstruction as a diffracting knife edge.

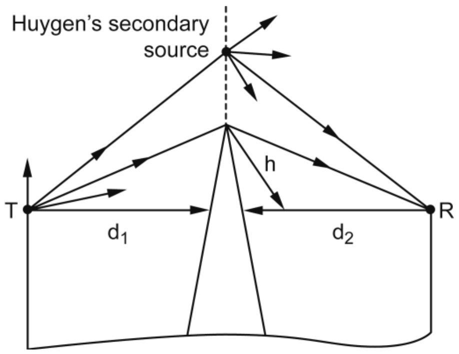

Figure 1.8: Illustration of knife-edge diffraction geometry.The rceiver R is locaated in the shadow.

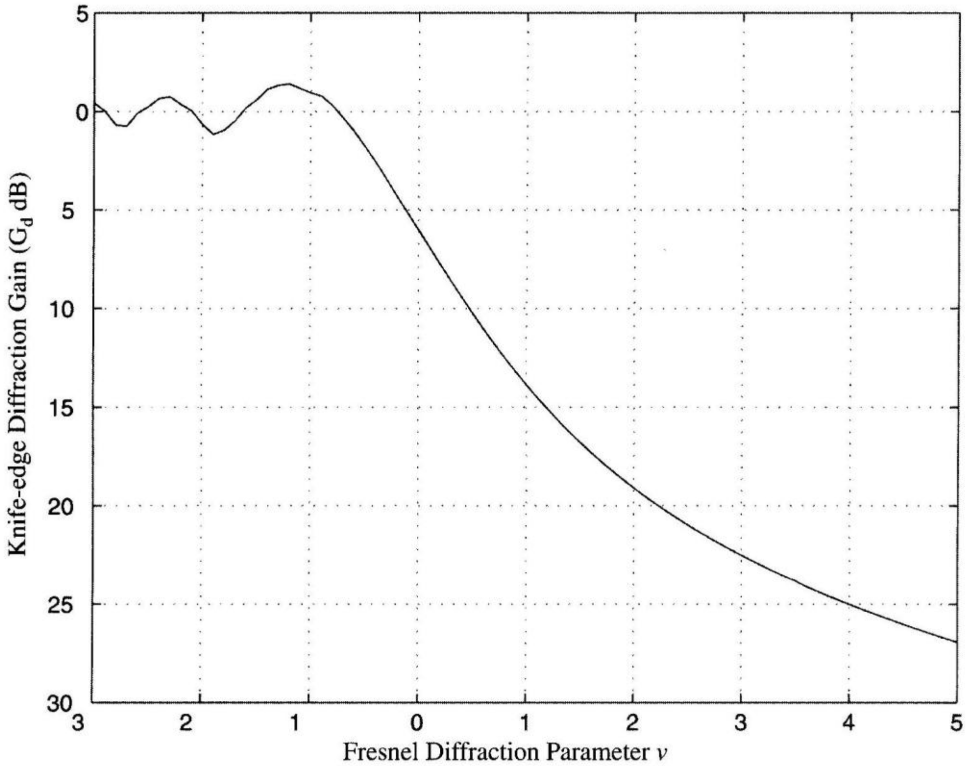

Figure 1.9: Knife edgediffraction gain as a function of Frenel diffraction parameter *v*.

- ✓ The field strength at point R is a vector sum of the fields due to all of the secondary Huygens sources in the plane above the knife edge.

Comparison of the Different Methods

1. The Bulling ton method:

This equivalent screen is derived in the following way:

- ✓ Put a tangential straight line from the TX to the real obstacles, and select the steepest one (i.e., the one with the largest elevation angle), so that all obstacles either touch this tangent, or lie below it.

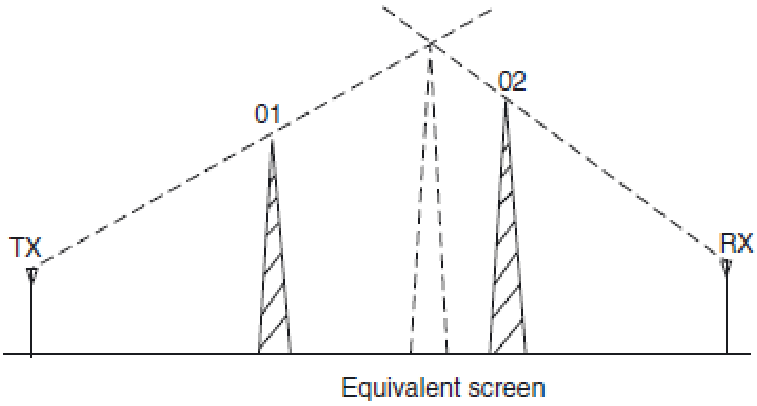

Figure 1.10: Bulling ton method

The major attraction of Bullington's method is its simplicity.

2.The Epstein–Petersen approach:

- ✓ This approach computes the diffraction losses for each screen separately.
- ✓ The attenuation of a specific screen is computed by putting a virtual "TX" and "RX" on the tips of the screens to the left and right of this considered screen

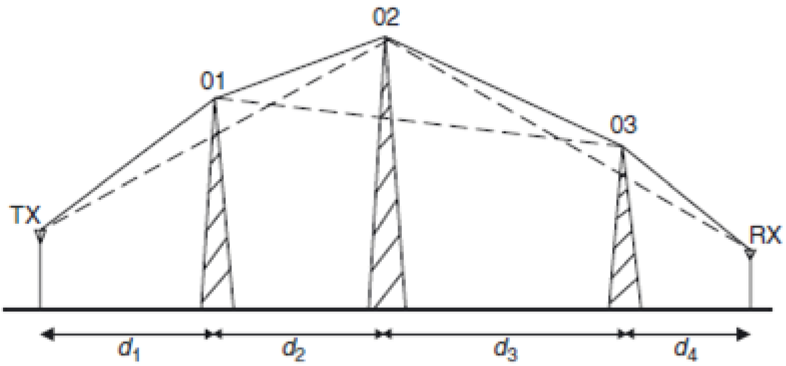

Figure 1.11: Epstein–Petersen approach

- ✓ **The Bullington method** is independent of the number of screens, and thus obviously gives a wrong functional dependence.
- ✓ **The Epstein–Petersen** method adds the attenuations *on a logarithmic scale* and thus leads to an exponential increase of the total attenuation on a linear scale.
- ✓ Similarly, the **Deygout method and the ITU-R method** predict an exponential increase of the total attenuation as the number of screens increases.
- ✓ The slope diffraction method (up to 15 screens) and the *modified* ITU method lead to a linear increase in total attenuation, and thus predict the trend correctly.

3. Scattering

- ✓ Scattering occurs when the medium through which the wave travels consists of objects with dimensions that are small compared to the wavelength, and where the number of obstacles per unit volume is large.
- ✓ Scattered waves are produced by rough surfaces, small objects, or by other irregularities in the channel.
- ✓ In practice, foliage, street signs, and lamp posts induce scattering in a mobile communications system.

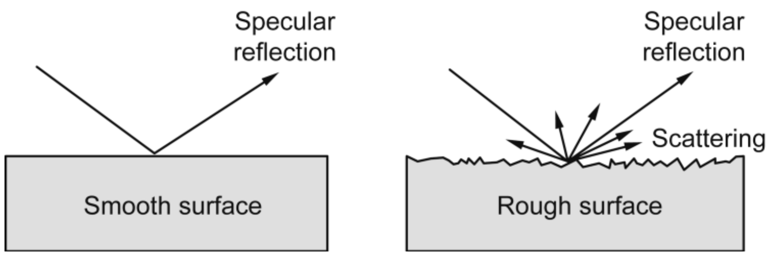

Figure 1.12: Scattering

- ✓ Scattering has two effects important to us:
  1. Reduces the power in the reflected wave.
  2. Causes additional multipath to be received in directions other than the specular direction (recall θr = θi).

Scattering is explained by two theories. They are

1. **Kirchhoff theory:**
   - ✓ The Kirchhoff theory is conceptually very simple
   - ✓ Requires the probability density function of surface amplitude (height). one point of the surface does not "cast a shadow" onto other points of the surface

$$\rho_{rough} = \rho_{smooth} \exp[-(k_0 \sigma_h \sin\psi)^2]$$

where, σ_h *is the standard deviation of the height distribution,*

k_0 *is the wavenumber* $2\pi/\lambda$*, and*

Ψ is the angle of incidence

$2k_0\sigma_h \sin\psi$ ------ *Rayleigh roughness.*

for grazing incidence ($\psi \approx 0$), the effect of the roughness vanishes.

2. **Perturbation theory**

- ✓ The actual received power at the receiver is somewhat stronger than claimed by the models of reflection and diffraction.
- ✓ The cause is that the trees, buildings and lampposts scatter energy in all directions. This provides extra energy at the receiver.
- ✓ Roughness is tested by a Rayleigh criterion, which defines a critical height (h_c) of surface protuberances for a given angle of incidence θ_i, given by,

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

- ✓ The reflected E-fields for $h>h_c$ can be solved for rough surfaces using a modified reflection coefficient given as

$$\Gamma_{rough} = \rho_s \Gamma$$

Radar Cross Section Model

- ✓ **In radio channels** where large, distant objects induce scattering, knowledge of the physical location of such objects can be used to accurately predict scattered signal strengths.
- ✓ The ***radar cross section (RCS)*** of a scattering object is defined as the ratio of the power density of the signal scattered in the direction of the receiver to the power density of the radio wave incident upon the scattering object, and has units of square meters.
- ✓ Analysis based on the geometric theory of diffraction and physical optics may be used to determine the scattered field strength.
- ✓ **For urban mobile radio systems**, models based on the ***bistatic radar equation*** may be used to compute the received power due to scattering in the far field.
- ✓ The bistatic radar equation describes the propagation of a wave traveling in free space which impinges on a distant scattering object, and is then reradiated in the direction of the receiver, given by

$$P_R(dBm) = P_T(dBm) + G_T(dBi) + 20\log(\lambda) + RCS[dBm^2] - 30\log(4\pi) - 20\log d_T - 20\log d_R$$

where d_T and d_R are the distance from the scattering object to the transmitter and receiver, respectively.

- ✓ It is useful for predicting receiver power which scatters off large objects, such as buildings, which are for both the transmitter and receiver.

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1.2 PATH LOSS MODELS

✓ Path loss models can be divided into two models.

1. Free Space model
2. Two-Ray model

1. **Free space propagation model:**

1. Explain how signal propagates against free space attenuation and reflection. [8m-May 2014]
2. How the received signal strength is predicated using the free space propagation model? Explain. [10m - Nov 2012]
3. Explain the free space path loss and derive the gain expression. [8m - May 2012]
4. Derive the path loss for large scale propagation in a multipath wireless environment. What is Doppler spread? [April 2010]
5. Describe briefly about free space propagation model.[April/May 2018]

- ✓ The free space propagation model is used to predict received signal strength.
- ✓ The transmitter and receiver should have a clear, unobstructed line-of-sight path between them.
- ✓ The free space power received by a receiver antenna is given by the **Friis free space equation**,

$$P_r(d) = \frac{P_t G_t G_r \lambda^2}{(4\pi)^2 d^2 L} \qquad \rightarrow (1)$$

where, $P_r(d)$ $\rightarrow$ Received power
P_t $\rightarrow$ Transmitted power
G_t $\rightarrow$ Transmitter antenna gain
G_r $\rightarrow$ Receiver antenna gain
d $\rightarrow$ T-R separation distance in meters
λ $\rightarrow$ Wavelength in meters
L $\rightarrow$ System loss factor

- ✓ The Friis free space equation shows that the received power decays with distance at a rate of 20dB/decade.
- ✓ The miscellaneous losses L are usually due to
  - ➢ Transmission line attenuation
  - ➢ Filter losses
  - ➢ Antenna losses in the communication system.
- ✓ The gain of an antenna is related to its effective aperture,

$$G = \frac{4\pi A_e}{\lambda^2} \qquad \rightarrow (2)$$

where, G $\rightarrow$ Gain of an antenna
A_e $\rightarrow$ Effective aperture

- ✓ Effective aperture, A_e is related to the physical size of the antenna.
- ✓ Wavelength λ , is related to the carrier frequency by

$$\lambda = \frac{c}{f} = \frac{2\pi c}{\omega_c} \qquad \rightarrow (3)$$

where f $\rightarrow$ carrier frequency in Hertz
ω_c $\rightarrow$ Carrier frequency in radians per second
c $\rightarrow$ Speed of light given in meters/s.

- ✓ An isotropic radiator is an ideal antenna which radiates power with unit gain uniformly in all directions.
- ✓ It is often used to reference antenna gains in wireless systems.
- ✓ The ***Effective Isotropic Radiated Power (EIRP)*** represents the maximum radiated power available from a transmitter in the direction of maximum antenna gain and is defined

$$EIRP = P_t G_t \qquad \rightarrow (4)$$

- ✓ In practice, ***effective radiated power (ERP)*** is used instead of EIRP to denote the maximum radiated power as compared to a half-wave dipole antenna (Instead of an Isotropic antenna).
- ✓ Antenna gains are given in units of
  - ➢ ***dBi*** (***dB*** gain with respect to an ***i***sotropic antenna) or
  - ➢ ***dBd*** (***dB*** gain with respect to a hail-wave ***d***ipole antenna)
- ✓ The path loss is defined as the difference between the effective transmitted power and the received power.
- ✓ The path loss represents signal attenuation as a positive quantity measured in dB.
- ✓ The path loss for the free space model when antenna gains are included is given by

$$PL(dB) = 10\log\frac{P_t}{P_r} = -10\log\left[\frac{G_t G_r \lambda^2}{(4\pi)^2 d^2}\right] \qquad \rightarrow (5)$$

- ✓ The path loss for the free space model when antenna gains are excluded is given by

$$PL(dB) = 10\log\frac{P_t}{P_r} = -10\log\left[\frac{\lambda^2}{(4\pi)^2 d^2}\right] \rightarrow (6)$$

- ✓ The Friis free space model is a valid for values of d which are in the far-field of the transmitting antenna.
- ✓ **Far-field** or **Fraunhofer region:**

  The far-field or **Fraunhofer region** of a transmitting antenna is defined as the region beyond the far-field distance d_f, which is related to the largest linear dimension of the transmitter antenna aperture and the carrier wavelength.

- ✓ The **Fraunhofer distance** is given by

$$d_f = \frac{2D^2}{\lambda} \rightarrow (7)$$

where, D → Largest physical linear dimension of the antenna.
d_f → Far-field distance

- ✓ To be in the far-field region, d_f must satisfy

$$d_f >> D \rightarrow (8)$$

$$d_f >> \lambda \rightarrow (9)$$

- ✓ Friis free space equation does not hold for d = 0. For this reason, close-in distance, d_0, known as received power reference point is used.
- ✓ The received power, $P_r(d)$ at any distance $d > d_0$, may be related to P_r at d_0.
- ✓ The value (d_0) may be predicted by taking the average received power at many points located at a close-in radial distance d_0 from the transmitter.
- ✓ The reference distance must be chosen such that it lies in the far-field region, that is, $d_0 \geq d_f$, and d_0 is chosen to be smaller than any practical distance used in the mobile communication system.
- ✓ The received power in free space at a distance greater than d_0 is given by

$$P_r(d) = P_r(d_0)\left(\frac{d_0}{d}\right)^2, \qquad d \geq d_0 \geq d_f \rightarrow (10)$$

- ✓ ***dBm*** or ***dBW*** units are used to express received power levels.
- ✓ The received power in dBm, is given by

$$P_r(d)\ dBm = 10\log\left[\frac{P_r(d_0)}{0.001W}\right] + 20\log\left(\frac{d_0}{d}\right), \qquad d \geq d_0 \geq d_f \rightarrow (11)$$

$P_r(d_0)$ is in units of Watts.

- ✓ The reference distance d_0 is chosen to be 1m in indoor environments and 100m or 1km in outdoor environments. So that the numerator in equations (10) and (11) is a multiple of 10.

******

Problem 1:
A communication system has the following parameters:
P_t **=5W,** G_t **(db)=13dB,** G_r **(dB)=17dB,** d **=80km, f=3GHz. Determine the value of the received power. [6m - May 2013]**

Given

Transmitted power, P_t =5W

Transmitter antenna gain, G_t =13dB

Receiver antenna gain, G_r =17dB

T-R separation distance in meters, d =80Km

Frequency=3GHz

W.K.T., System loss factor, =1

To Find

Received power, $P_r(d)$

Solution

$$\lambda = \frac{c}{f} = \frac{3\times10^8\,m/s}{3\times10^9\,Hz} = 0.1m$$

$$P_r(d) = \frac{P_t G_t G_r \lambda^2}{(4\pi)^2 d^2 L} = \frac{5\times13\times17\times0.1^2}{(4\pi)^2\left(80\times10^3\right)^2\times1} = 10.9446\times10^{-12}W$$

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2. TWO RAY (2-RAY) OR GROUND REFLECTION MODEL

1. **Explain the advantages and disadvantages of two-ray ground reflection model in the analysis of path loss. [Nov 2015]**
2. **Prove that in the two-ray ground reflected model, $\Delta = d'' - d' \approx \frac{2h_t h_r}{d}$. [16m - Nov 2015]**
3. **Explain the time variant two-path model of wireless Propagation channel. [16m-May 2016]**
4. **Derive the expression for the total electric field E_{TOT} *(d)* and received power at distance, P_r(d) using two ray ground reflection model. [Nov 2015]**
5. **Explain in detail two path model propagation mechanisms. [8m - May 2014, 8m-May 2012]**
6. **Explain the time variant two-path model of a wireless propagation channel. [8m - May 2013]**
7. **Derive the equation of the path loss for the two-ray model with antenna gains. [Nov 2009]**
8. **Explain briefly about Two ray Ground reflection model. [April/May 2018]**

✓ The free space propagation model is inaccurate when used alone.

$$PL(dB) = -10\log\left[\frac{G_t G_r \lambda^2}{(4\pi)^2 d^2}\right] \rightarrow (1)$$

PL → Path loss for the free space model

G_t → Transmitter antenna gain

G_r → Receiver antenna gain

λ → Wavelength in meters

d → T-R separation distance in meters

✓ The 2-ray ground reflection model is a useful propagation model that is based on geometric optics.

✓ It considers both the direct path and a ground reflected propagation path between transmitter and receiver.

✓ This model is accurate for predicting the large-scale signal strength over distances of several kilometers for mobile radio systems.

✓ The maximum T-R separation distance is at most only a few tens of kilometers.

✓ Earth is assumed to be flat.

- The total received E-field, E_{TOT}, is then a result of the direct line-of-sight component, E_{LOS}, and the ground reflected component, E_g.

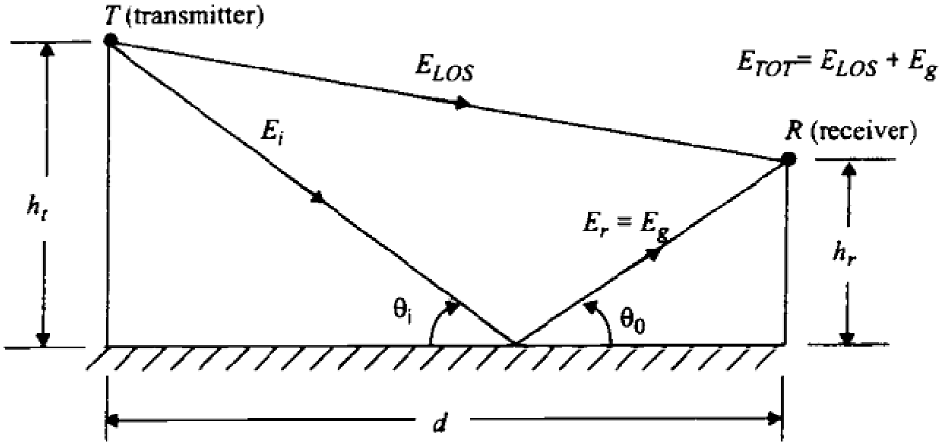

Figure 1.3: Two-ray ground reflection model.

- The free space propagating E-field is given by

$$E(d,t) = \frac{E_0 d_0}{d}\cos\left(\omega_c\left(t - \frac{d}{c}\right)\right) \qquad d \geq d_0 \qquad \rightarrow (2)$$

where E_0 → Free space E-field (in units of V/rn)
d_0 → Reference distance from the transmitter
$|\mathrm{E}(\mathrm{d},\mathrm{t})| = \frac{E_0 d_0}{d}$ → Envelope of the E-field.

- Two propagating waves arrive at the rece iver:
  - Direct wave that travels a distance *d'*
  - Reflected wave that travels a distance *d''*
- E-field due to the line-of-sight component at the receiver can be expressed as

$$E_{LOS}(d', t) = \frac{E_0 d_0}{d'}\cos\left(\omega_c\left(t - \frac{d'}{c}\right)\right) \qquad \rightarrow (3)$$

- E-field due to ground reflected wave component at the receiver can be expressed as

$$E_g(d'', t) = \Gamma\frac{E_0 d_0}{d''}\cos\left(\omega_c\left(t - \frac{d''}{c}\right)\right) \qquad \rightarrow (4)$$

- According to laws of reflection in dielectrics

$$\theta_i = \theta_0 \qquad \rightarrow (5)$$

$$E_g = \Gamma E_i \qquad \rightarrow (6)$$

$$E_t = (1+\Gamma)E_i \qquad \rightarrow (7)$$

where θ_i → Angle of incidence
θ_0 → Angle of reflection
E_i → Incident E-field
Γ → Reflection coefficient for ground

- For small values of θ_i, the reflected wave is equal in magnitude and 180° out of phase with the incident wave.
- The resultant total E-field is the vector sum of E_{LOS} and E_g is given by

$$|E_{TOT}| = |E_{LOS} + E_g| \qquad \rightarrow (8)$$

where E_{TOT} → Total received E-field

E_{LOS} → Direct line-of-sight component

E_g → Ground reflected component

- ✓ The electric field $E_{TOT}(d, t)$ is expressed as the sum of equations (3) and (4)

$$E_{TOT}(d,t) = \frac{E_0 d_0}{d'} \cos\left(\omega_c\left(t - \frac{d'}{c}\right)\right) + (-1)\frac{E_0 d_0}{d''}\cos\left(\omega_c\left(t - \frac{d''}{c}\right)\right) \quad \rightarrow (9)$$

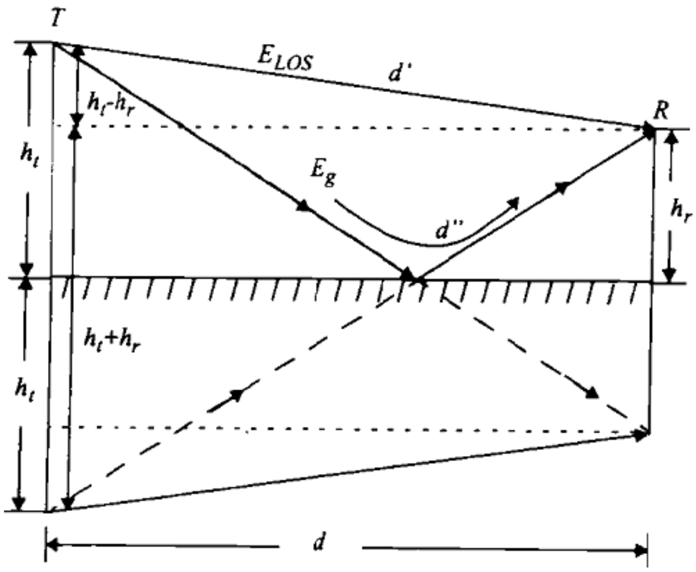

Figure 1.4: The method of images is used to find the path difference between the line-of-sight and the ground reflected paths.

- ✓ Using the method of images, the path difference, Δ, between the line-of-sight and the ground reflected paths can be expressed as

$$\Delta = d'' - d' = \sqrt{(h_t + h_r)^2 + d^2} - \sqrt{(h_t - h_r)^2 + d^2} \quad \rightarrow (10)$$

where h_t → Height of the transmitter

h_r → Height of the receiver.

- ✓ When the T-R separation distance d is very large compared to $h_t + h_r$, equation (10) can be simplified using a Taylor series approximation

$$\Delta = d'' - d' \approx \frac{2h_t h_r}{d} \quad \rightarrow (11)$$

- ✓ Phase difference, θ_Δ between the two E field components is given by

$$\theta_\Delta = \frac{2\pi\Delta}{\lambda} = \frac{\Delta\omega_C}{c} \quad \rightarrow (12)$$

- ✓ Substitute (11) in (12),

$$\theta_\Delta = \frac{2\pi}{\lambda}\frac{2h_t h_r}{d}$$

$$\frac{\theta_\Delta}{2} = \frac{2\pi}{\lambda}\frac{h_t h_r}{d} \quad \rightarrow (13)$$

- ✓ Time delay τ_d between the arrival of the two components is given by

$$\tau_d = \frac{\Delta}{c} = \frac{\theta_\Delta}{2\pi f_c} \quad \rightarrow (14)$$

- ✓ When d becomes large, the difference between the distances d' and d becomes very small, and the amplitudes of E_{LOS} and E_g are virtually identical and differ only in phase.

$$\left|\frac{E_0 d_0}{d}\right| \approx \left|\frac{E_0 d_0}{d'}\right| \approx \left|\frac{E_0 d_0}{d''}\right| \qquad \rightarrow (15)$$

- ✓ If the received E-field is evaluated when t = d"/c, equation (9) can be expressed as

$$E_{TOT}\left(d, t=\frac{d''}{c}\right) = \frac{E_0 d_0}{d'}\cos\left(\omega_c\left(\frac{d''-d'}{c}\right)\right) - \frac{E_0 d_0}{d''}\cos 0^0 \qquad \rightarrow (16)$$

$$= \frac{E_0 d_0}{d'}\angle\theta_\Delta - \frac{E_0 d_0}{d''}$$

$$\approx \frac{E_0 d_0}{d}\left[\angle\theta_\Delta - 1\right]$$

- ✓ Referring to the phasor diagram, the electric field is given by

$$\left|E_{TOT}(d)\right| = \sqrt{\left(\frac{E_0 d_0}{d}\right)^2 (\cos\theta_\Delta - 1)^2 + \left(\frac{E_0 d_0}{d}\right)^2 \sin^2\theta_\Delta} \qquad \rightarrow (17)$$

$$\left|E_{TOT}(d)\right| = \frac{E_0 d_0}{d}\sqrt{2 - 2\cos\theta_\Delta} \qquad \rightarrow (18)$$

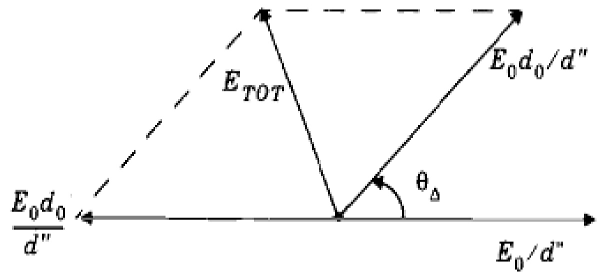

Figure: Phasor diagram showing the electric field components of the line-of-sight, ground reflected, and total received E-fields, derived from equation

- ✓ Using trigonometric identities, equation (17) can be expressed as

$$\left|E_{TOT}(d)\right| = 2\frac{E_0 d_0}{d}\sin\left(\frac{\theta_\Delta}{2}\right) \qquad \rightarrow (19)$$

- ✓ Note that equation (19) may be simplified whenever $\sin\left(\frac{\theta_\Delta}{2}\right) \approx \left(\frac{\theta_\Delta}{2}\right)$. This occurs when $\left(\frac{\theta_\Delta}{2}\right)$ is less than 0.3 radian. Using equation (13)

$$\sin\left(\frac{\theta_\Delta}{2}\right) \approx \frac{\theta_\Delta}{2} \approx \frac{2\pi h_t h_r}{\lambda d} < 0.3 \text{ rad} \qquad \rightarrow (20)$$

$$d > \frac{2\pi h_t h_r}{0.3\lambda} \approx \frac{20 h_t h_r}{\lambda} \qquad \rightarrow (21)$$

- ✓ As long as d satisfies (21), the received E-field is approximated as
  Put (20) in (19) we get

$$E_{TOT}(d) \approx \frac{2E_0 d_0}{d}\frac{2\pi h_t h_r}{\lambda d} \approx \frac{4\pi E_0 d_0 h_t h_r}{d^2 \lambda} \approx \frac{k}{d^2}\, V/m \qquad \rightarrow (22)$$

where k $\rightarrow$ Constant related to E_0, the antenna heights, and the wavelength.

- ✓ The power received at *d* is related to the square of the electric field through **equation (15).**
- ✓ The received power at a distance d from the transmitter can be expressed as

$$P_r = P_t G_t G_r \frac{h_t^2 h_r^2}{d^4} \qquad \rightarrow (23)$$

- ✓ The path loss for the 2-ray model (with antenna gains) can be expressed in dB as

$$PL(dB) = 40\log d - (10\log G_t + 10\log G_r + 20\log h_t + 20\log h_r) \quad \rightarrow (24)$$

- ✓ At small distances, equation (3.39) must be used to compute the total E field.
- ✓ The first Fresnel zone distance is a useful parameter in microcell path loss models

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1.3. LINK BUDGET DESIGN USING PATH LOSS MODELS

1. **What is the need for link calculation? Explain with suitable example. [8M-May 2012]**
2. **Explain on path loss estimation techniques using path loss models. Briefly explain the factors that influence small scale fading. [Nov/Dec 2013, Nov/Dec 2014]**
3. **(i) What do you mean by path loss model? Explain in detail about log-distance path loss model.**
   (ii) What is the need for link calculation? Explain with suitable example. [16m – Nov 2017]

- ✓ Radio propagation models are derived using analytical and empirical ***(observed)*** methods.
- ✓ Empirical approach:
  - ➢ It is based on fitting curves or analytical expressions that recreate a set of measured data.
  - ➢ It takes all propagation factors in account, both known and unknown, through actual field measurements.
- ✓ Some classical propagation models have emerged to predict large scale coverage for mobile communication systems design.
- ✓ In mobile communication systems, ***pathloss models*** are used ***to calculate received signal level*** as a function of distance, to predict the SNR.
- ✓ Some ***practical path loss estimation techniques*** are
  - ➢ Log-distance Path Loss Model
  - ➢ Log-normal Shadowing
  - ➢ Determination of percentage coverage area.

a. Log-distance path loss model

- ✓ Both analytical and measurement based models indicate that average received signal power decreases logarithmically with distance (whether in outdoor or indoor channels).
- ✓ The average large-scale path loss for an arbitrary T-R separation is expressed as a function of distance *d* using the pathloss component, '*n*'

$$\overline{PL}(d) \propto \left(\frac{d}{d_0}\right)^n \quad \rightarrow (1)$$

in dB,

$$\overline{PL}(dB) = \overline{PL}(d_0) + 10\ n\ \log\left(\frac{d}{d_0}\right) \quad \rightarrow (2)$$

where $\overline{PL}(d)$ → Average large-scale path loss
d → T-R separation distance
d_0 → Close-in reference distance
(determined from measurements close to the transmitter)
n → Path loss exponent (the rate at which the pathloss increases)

- ✓ Path loss exponent *n*, varies with environment.
  Eg: n=2 for free space

Table 1.1: Path loss exponents for different environments

Environment	Path Loss Exponent, n
Free space	2
Urban area cellular radio	2.7 to 3.5
Shadowed urban cellular radio	3 to 5
In building line-of-sight	1.6 to 1.8
Obstructed in building	4 to 6
Obstructed in factories	2 to 3

- ✓ Reference distance d_0,
  - ➢ For large cellular system → d_0=1Km
  - ➢ For Microcellular system → d_0=100m (or) 1m *[i.e., much smaller]*

b. <u>Log-normal Shadowing</u>

- ✓ The log-distance path loss model in equation (2) does not consider the fact that the environmental may be vastly different at two different locations having the same T-R separation.
- ✓ The path loss PL(d) at a particular location is random and distributed log-normally (in dB) about the mean distance-dependent value.

$$PL(d)\ [dB] = \overline{PL}(d) + X_\sigma \quad \rightarrow (3)$$

$$= \overline{PL}(d_0) + 10\ n\ \log\left(\frac{d}{d_0}\right) + X_\sigma \quad \rightarrow (4)$$

Received power = Transmitted power – Path loss → (5)

$$P_r(d)\ [dBm] = P_t\ [dBm] - PL(d)\ [dB] \quad \rightarrow (6)$$

where $PL(d)$ → Path loss at a particular location

$P_r(d)$ → Normal distribution about $PL(d)$ → Received power

X_σ → Zero-mean Gaussian distributed random variable (in dB)

σ → Standard deviation (in dB)

- ✓ d_0, n, σ determine the path loss.
- ✓ The log-normal distribution describes the random *shadowing* effects occur over a large number of locations with the same T-R separation, but with different levels of clutter on the propagation path. This phenomenon is referred to as log-normal shadowing.

Q function (or) error function (erf)

- ✓ Q function is used to determine the probability that received signal strength will exceed a particular level.

$$Q(z) = \frac{1}{\sqrt{2\pi}} \int_z^\infty \exp\left(-\frac{x^2}{2}\right) dx = \frac{1}{2}\left[1 - erf\left(\frac{z}{\sqrt{2}}\right)\right] \quad \rightarrow (7)$$

where $Q(z) = 1 - Q(-z)$ → (8)

- ✓ Probability that received signal level will exceed a certain value γ can be calculates as,

$$\Pr[P_r(d) > \gamma] = Q\left(\frac{\gamma - \overline{P_r(d)}}{\sigma}\right) \quad \rightarrow (9)$$

- ✓ Similarly, the probability that received signal level will be below γ is given by,

$$\Pr[P_r(d) < \gamma] = Q\left(\frac{\overline{P_r(d)} - \gamma}{\sigma}\right) \quad \rightarrow (10)$$

where, γ →certain value of received power

c. Determination of percentage coverage area

- ✓ Due to random effects of shadowing, some locations within a coverage area will be below a particular desired received signal threshold.
- ✓ It is useful to compute the percentage of coverage area in boundary coverage.
- ✓ For a circular coverage area of radius R from the base station and the desired signal threshold of γ, the percentage of useful threshold area $U(\gamma)$ can be found by

$$U(\gamma) = \frac{1}{\pi R^2}\int \Pr[P_r(r) > \gamma]dA = \frac{1}{\pi R^2}\int_0^{2\pi}\int_0^{R} \Pr[P_r(r) > \gamma]\ dr\, d\theta$$

where,

$\Pr[P_r(r) > \gamma]$ → Probability that the random received signal at $d = r$ exceeds the threshold – within an incremental area dA.

$d = r$ → It represents the radial distance from the transmitter.

- ✓ For large number of values of σ and n

$$U(\gamma) = \frac{1}{2}\left[1 + \exp\left(\frac{1}{b^2}\right)\left(1 - erf\left(\frac{1}{b}\right)\right)\right]$$

where,

$$b = (10\ n\ \log e)/\sigma\sqrt{2}$$

- ✓ $U(\gamma)$ is the percentage useful service area where received signal strength equals or greater than γ i.e. $P_R(d) > \gamma$.
- ✓ If n=2 and σ = 8 dB, a 75% boundary coverage provides 91% area coverage.
- ✓ If n=3 and σ = 9 dB, a 50% boundary coverage provides 71% area coverage

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Problem:

Consider a mobile radio system at 900 MHz carrier frequency, with 25 kHz bandwidth that is affected by thermal noise only. Antenna T_X gain is 8 dB and R_X gain is – 2dB. Losses at the T_X are 2 dB. The noise figure of the receiver is 7 dB, 3 dB bandwidth of the signal is 25 KHz. The required operating SNR is 18 dB; the desired range of coverage is 2 km. The break point is at 10 m distance, beyond that point, the path loss exponent is 3.8 and the fading margin is 10 dB. The minimum T_X power is

Noise spectral density = k_B T_e = – 174 dBm/Hz

Bandwidth = B = 44 dBHz

Thermal noise power at R_X = P_n = – 130 dBm

R_X excess noise = 7 dB

Required SNR = 18 dB

Required receiver power (P R_X) = – 105 dBm

Pathloss = 87 dB

$$\text{Pathloss from } T_X \text{ to breakpoint at 10 m} = \left(\frac{\lambda}{4\pi\, d}\right)^2 = 52 \text{ dB}$$

$$\text{Antenna gain at MS} = -(-2) \text{ dB}$$

$$\text{Fading margin} = 10 \text{ dB}$$

Required EIRP = Equivalent isotropic radiated power = 460 dBm

$$T_X \text{ antenna gain} = G\,T_x = -8 \text{ dB}$$

$$\text{Losses in cables} = L_f = 2 \text{ dB}$$

$$\text{Required } T_X \text{ power} = 40 \text{ dBm}$$

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1.4. SMALL SCALE FADING

1. Discuss in detail the constructive and destructive interference. Describe in detail about the effects of multipath propagation in wireless environment. [Nov/Dec 2011, May/June 2013 Nov/Dec 2014]
2. Explain about the factors that influence small-scale fading.
Nov/Dec 2012 Nov/Dec 2013, Nov/Dec 2014.

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

- ✓ Fading is caused by interference between two or more versions of the transmitted signal which arrive at the receiver at slightly different times. These waves are called ***multipath waves***.

Small-Scale Multipath Propagation:

- ➢ In principle, the following are the main multipath effects:
  1. Rapid changes in signal strength over a small travel distance or time interval.
  2. Random frequency modulation due to varying Doppler shifts on different multipath signals.
  3. Time dispersion or echoes caused by multipath propagation delays.

Factors Influencing Fading:

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

(1) Multipath propagation

- ➢ The reflecting objects and scatterers in the channel creates a constantly changing environment.
- ➢ This changing environment dissipates the signal energy in amplitude, phase and time.
- ➢ This effect results in multiple versions of the transmitted signal that arrive at the receiving antenna.
- ➢ These received signals displace with respect to one another in time and spatial orientation.
- ➢ The random phase and amplitudes of the different multipath components cause fluctuations in signal strength, includes small scale fading, signal distortion or both.
- ➢ Multipath causes signal smearing ***(spreading)*** due to ISI.

(2) Speed of the mobile

- ➢ The relative motion between the base station and the mobile results in random frequency modulation due to different Doppler shifts on each of the multipath components.
- ➢ Doppler shift will be positive *(i.e., apparent receiving frequency is increased)* or negative, it depends on the mobile moving toward or away from the base station respectively.

(3) Speed of surrounding objects

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

(4) Transmission Bandwidth of the signal

- If the transmitted radio signal bandwidth is greater than the "bandwidth" of the multipath channel (quantified by coherence bandwidth), the received signal will be distorted.
- But, the received signal strength will not fade much over a local area.

1.5. PARAMETERS OF MOBILE MULTIPATH CHANNELS

1. **Explain in detail about Small Scale Fading Parameters of Mobile Multipath Channels**
2. **Explain the time dispersion parameters of mobile multipath channels.**

- ✓ Power delay profiles are averaging instantaneous power delay profile over a local area to determine an average small-scale power delay profile.
- ✓ Parameters of mobile multipath channels includes
  - Time dispersion parameters
  - Coherence bandwidth
  - Doppler spread & Coherence time
- ✓ Parameters that describes the time dispersive nature of the channel in a local area are
  - Delay spread
  - Coherence bandwidth
- ✓ Parameters that describes the time varying nature of the channel in a small scale region are
  - Doppler spread
  - Coherence time

1.5.1. TIME DISPERSION PARAMETERS

- ✓ In multipath phenomenon, the time difference between the arrival moment of the first multipath component and the last one is called ***delay spread***.
- ✓ Some parameters are used to quantify the multipath channel.
- ✓ These parameters are used to compare different multipath channels and to develop design guidelines for wireless systems.
- ✓ The following multipath parameters are used to quantify the time dispersive properties of wide band multipath channels:

  a. Mean excess delay ($\bar{\tau}$)
  b. RMS delay spread (σ_τ) (frequently used properties)
  c. Maximum Excess Delay (X dB)

- ✓ The *rms* delay spread and mean excess delay are defined from a single power delay profile which is the temporal or spatial average of consecutive impulse response over a local area.

a. Mean excess delay

- ✓ The mean excess delay is the first moment of the power delay profile (PDP).
- ✓ It is expressed as

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

b. RMS delay spread

- ✓ This is the most important single measure for the delay times extent of a multipath delay channel.
- ✓ This parameter calculates the standard deviation value of the delay of reflections.

- ✓ The standard deviation value will be weighted proportional to the energy in the reflected waves.
- ✓ The *rms* delay spread is the square root of the second central moment of the power delay profile.

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

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

where,

σ_τ → *rms* delay spread

a_k → Amplitude

$P(\tau_k)$ → Relative power levels of the individual multipath components

τ_k → Excess delay

c. Maximum excess delay

- ✓ The maximum excess delay (X dB) of the power delay profile is defined to be the time delay during which multipath energy falls to X dB below the maximum.
- ✓ The maximum excess delay is defined $\tau_x - \tau_0$ as

  Where, τ_0 → First arriving signal

  τ_x → Maximum delay at which a multipath component is within X dB of the strongest multipath signal
- ✓ Maximum excess delay is sometimes called the ***excess delay spread***.
- ✓ The maximum excess delay defines the temporal extent of multipath that is above a particular threshold.
- ✓ In all cases, must be specified with a threshold that relates the multipath noise floor to maximum received multipath component.

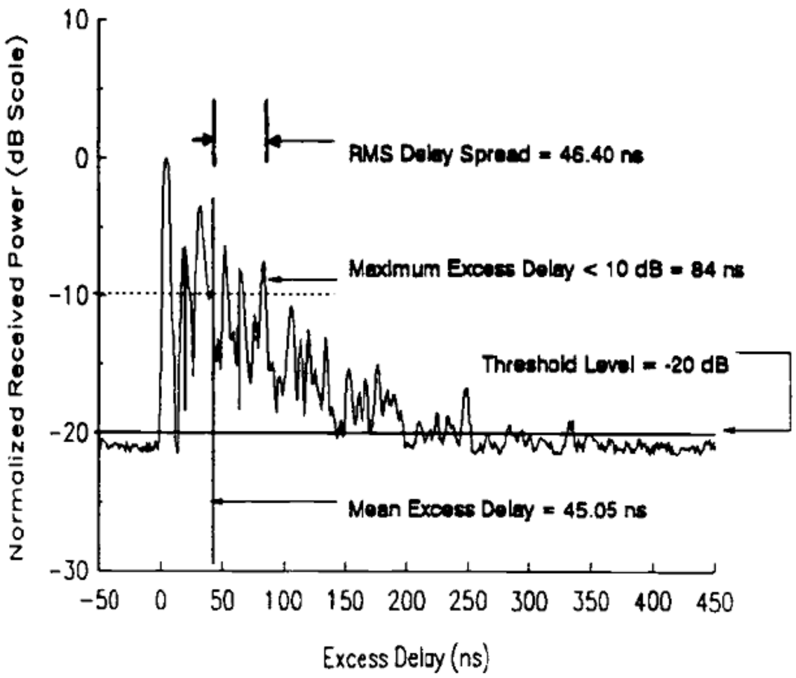

Figure 1.13: Measured multipath power delay profiles

1.5.2. COHERENCE BANDWIDTH

1.**Given that the coherence band width is approximately by equation** $B_c \approx \frac{1}{50\sigma_\tau}$.

Show that a flat fading channel occurs when $T_s >= 100\sigma_\tau$. [April/May 2018]

- ✓ The maximum frequency separation for which the signals are still strongly correlated is called coherence bandwidth (B_c).
- ✓ Coherence bandwidth, B_c, is derived from the rms delay spread.
- ✓ Coherence bandwidth is a statistical measure of the range of frequencies over which the channel is considered to be 'flat'.
- ✓ If frequency correlation function is above 0.9, then the coherence bandwidth is $B_c \approx \frac{1}{50\sigma_\tau}$
- ✓ If frequency correlation function is above 0.5. then the coherence bandwidth is $B_c \approx \frac{1}{5\sigma_\tau}$
- ✓ If the signal bandwidth larger than B_c is transmitted through the channel, it will subject to Frequency selective distortion.
  The channel will be referred as a frequency selective channel.
- ✓ If the signal bandwidth is larger than B_c, it will experience amplitude attenuation only with no distortion.
- ✓ This channel will be referred as a frequency non-selective fading channel.

1.5.3. DOPPLER SPREAD AND COHERENCE TIME

Derive the path loss for large scale propagation in a multipath wireless environment.–What is Doppler spread? [April 2010]

- ✓ Doppler spread and Coherence time are parameters which describe the time dispersive nature of the channel in a local area.
- ✓ Also, describes the time varying nature of the channel in a small-scale region.
- ✓ They do not give information about the time varying nature of the channel in large scale region.

Doppler spread (B_D)

- ✓ Doppler spread B_D is a measure of the spectral broadening caused by the time rate of change of the mobile radio channel
- ✓ Doppler spread B_D is defined as the range of frequencies over which the received Doppler spectrum is essentially non-zero.
- ✓ When pure dc signal is transmitted, then Doppler spectrum is $f_c, f_c + f_d, f_c - f_d$
  where $f_d \rightarrow$ Doppler frequency.
- ✓ It describes the time varying nature frequency dispersiveness of the channel
- ✓ If the baseband signal bandwidth is much greater than B_D, the effects of Doppler spread are negligible at the receiver. This is a slow fading channel.

Coherence time (T_C)

- ✓ Coherence time is the time over which two signals are having strong potential for amplitude correlation.

- ✓ Coherence time is the range of time over which similar fading occurs.
- ✓ The Doppler spread and coherence time are inversely proportional to one another.

$$Cohrence\ Time = \frac{1}{Doppler\ Spread}$$

- ✓ Thus, if the transmitter, receiver, or the intermediate objects move very fast, the Doppler spread is large and the coherence time is small, i.e., the channel changes fast.
- ✓ That is,

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

Where,

$f_m \rightarrow$ Maximum Doppler Shift, $f_m = \frac{v}{\lambda}$

- ✓ The time over which the time correlation > 0.5, then the coherence time is

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

- ✓ Equation is valid for Rayleigh fading channels.
- ➢ Coherence time is defined as the geometric mean

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

- ✓ If the reciprocal bandwidth of the baseband signal is greater than the coherence time of the channel, then the channel will change during the transmission of the baseband message, thus causing distortion at the receiver.

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1.6. TYPES OF SMALL SCALE FADING

1. **Explain in detail about types of Small Scale Fading. [May 2010]**
2. **Distinguish fast and slow fading in wireless channel and explain in detail. [16M-Nov 2017]**
3. **Explain the types of small scale fading based on multipath time delay spread.**
4. **Explain the types of small scale fading based on Doppler spread.**

- ✓ The type of fading in the signal propagating through a mobile radio channel depends on the nature of the ***transmitted signal*** with respect to the ***characteristics of the channel***.
- ✓ Depending on the relation between the signal parameters and channel parameters, different transmitted signals will undergo different types if fading.

  Eg: Signal parameters – Bandwidth, symbol period, etc

  Channel parameters – RMS delay and Doppler spread.
- ✓ The time dispersion and frequency dispersion mechanisms in a mobile radio channel lead to four possible effects.
- ✓ The four different types of fading are

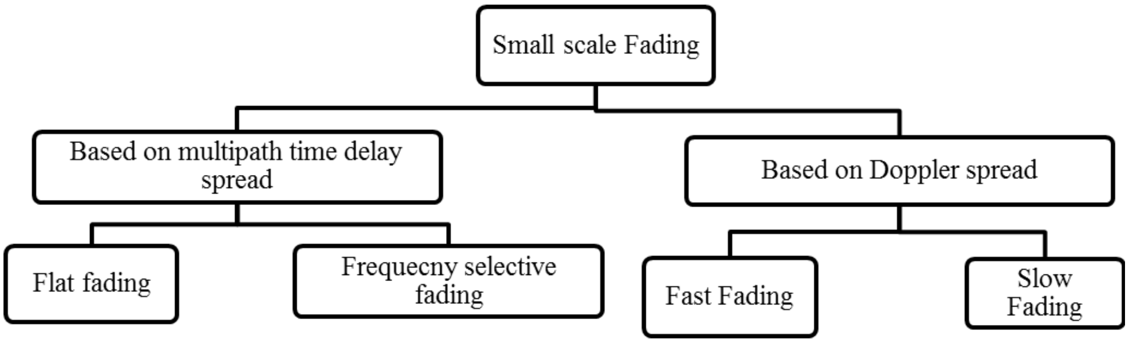


Figure 1.14: Types of small scale fading

1.6.1. FADING EFFECTS DUE TO MULTIPATH TIME DELAY SPREAD

- ✓ Time dispersion due to multipath causes the transmitted signal to undergo either ***flat*** or ***frequency selective fading***.

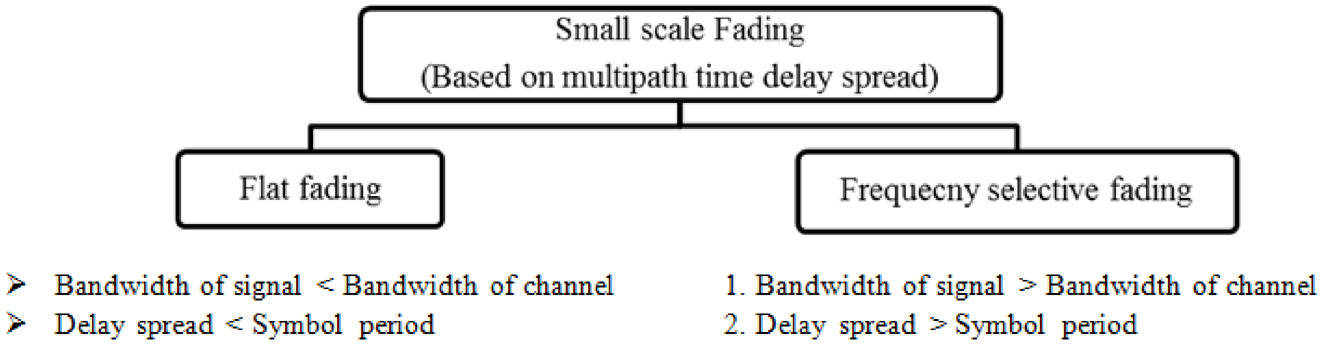


Figure 1.15: Types of small scale fading (Based on Multipath time delay spread)

a.<u>Flat Fading</u>:

- ✓ This form of multipath fading affects all the frequencies (almost equally) in the given channel.
- ✓ Flat multipath fading changes the amplitude and rising & falling time of the signal.
- ✓ If mobile radio channel has a constant gain and linear phase response over a bandwidth that is greater than the bandwidth of the transmitted signal, then the received signal will undergo flat fading.
- ✓ ***Channel impulse response***

$$\textit{Mutipath delay spread} \ll \frac{1}{\textit{Bandwidth of transmitted message waveform}}$$

- ✓ ***Time characteristics***
  - Received signal changes with time due to fluctuations in the gain of the channel caused by multipath.
- ✓ ***Spectral characteristics***
  - Multipath structure of the channel is such that the spectral characteristics of the transmitted signal are preserved at the receiver.

➢ ***Channel characteristics***

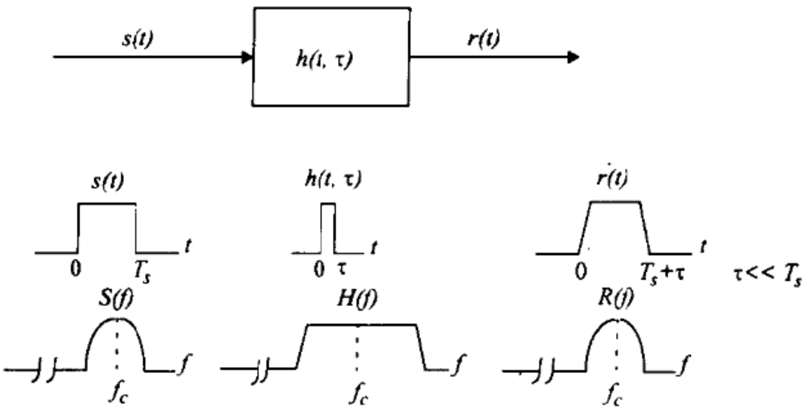


Figure 1.16: Flat fading channel characteristics

- ✓ The amplitude and channel gain varies with time, but the spectrum of the transmission is preserved.
- ✓ Flat fading channels are also known as amplitude varying channels (or) narrowband channels *(since, the bandwidth of the applied signal is narrow than the channel bandwidth).*
- ✓ Flat fading channels cause deep fades and requires 20 or 30 dB more transmitter power to achieve low bit error rates.
- ✓ Time varying statistics are like Rayleigh flat fading.
- ✓ Rayleigh distribution channels are used to measure variations of amplitude.
- ✓ A signal undergoes flat fading if

$$B_S << B_c \text{ and } T_S >> \sigma_\tau$$

where, $B_s \rightarrow$ Signal Bandwidth.
$B_c \rightarrow$ Coherence bandwidth
$T_s \rightarrow$ Reciprocal bandwidth (symbol period)
$\sigma_\tau \rightarrow$ rms delay spread

b.<u>Frequency Selective Fading</u>

- ✓ Assume the channel possesses a constant-gain and linear phase response over a bandwidth that is smaller than the bandwidth of transmitted signal, then the channel creates frequency selective fading on the received signal.
- ✓ ***Channel impulse response***
  - $\textit{Mutipath delay spread} \,\square\, \dfrac{1}{\textit{Bandwidth of transmitted message waveform}}$
- ✓ ***Time characteristics***
  - For this, the received signal includes multiple versions of the transmitted waveform which are attenuated (delayed) and delayed in time. Hence the received signal is distorted.
- ✓ ***Spectral characteristics***
  - Frequency selective fading is due to time dispersion of the transmitted symbols within the channel. Thus the channel induces intersymbol interference (ISI).
  - The received signal spectrum has greater gains than others.

- ✓ ***Channel characteristics***

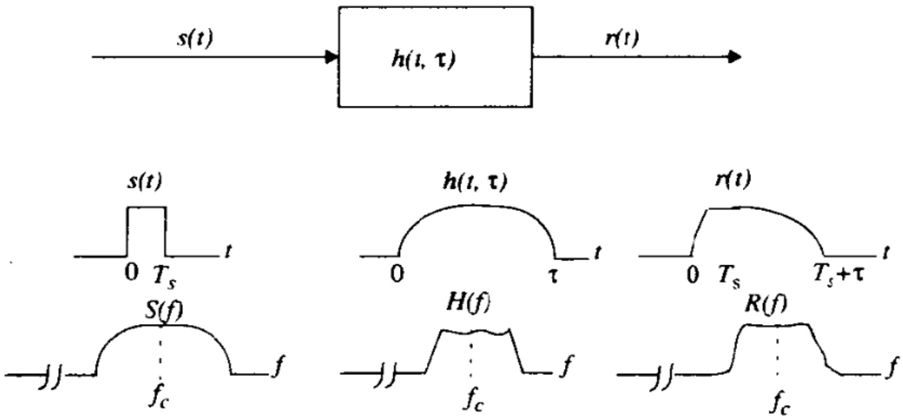


Figure 1.17: Frequency selective fading channel characteristics

- ✓ The gain is different for different frequency components.
- ✓ Frequency selective fading channels are also known as wideband channels since the bandwidth of the signal s(t) is wider than the bandwidth of the channel impulse response
- ✓ Selective fading lowers less fading effects [Time varying distortions]
- ✓ Fading models are used to measure frequency.
- ✓ For Frequency selective fading, the spectrum S(f) of the transmitted signal bandwidth (B_S) which is greater than the coherence bandwidth (B_C)
- ✓ That is, a signal undergoes frequency selective fading if

$$B_S > B_C \quad and \quad T_S > \sigma_C,$$

where
B_s → Signal Bandwidth.
B_c → Coherence bandwidth
T_s → Reciprocal bandwidth
σ_τ → rms delay spread

*****

1.6.2. FADING EFFECTS DUE TO DOPPLER SPREAD

Distinguish fast fading and slow fading in wireless channel and explain in detail.[Nov/Dec2017]

- ✓ Depending on how rapidly the transmitted baseband signal changes as compared to the rate of change of the channel, a channel may be classified as follows.

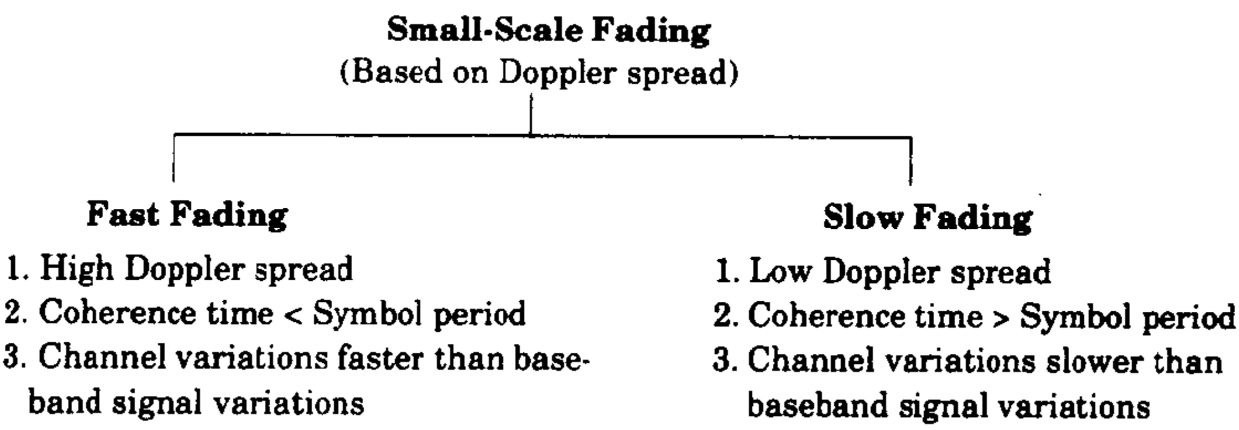


Figure 1.18: Types of small scale fading (Based on Doppler spread)

a. Fast Fading / Time Selective Fading

- ✓ In a fast fading channel, the channel impulse response changes rapidly within the symbol duration.
- ✓ That is, the coherence time of the channel is smaller than the symbol period of the transmitted signal.
- ✓ This causes Frequency dispersion (also called time selective fading) due to Doppler spreading, which leads to signal distortion.
- ✓ In the frequency domain, signal distortion due to fast fading increases with increasing Doppler spread relative to the bandwidth of the transmitted signal.
- ✓ Therefore, a signal undergoes fast fading if

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

- ✓ It should be noted that when a channel is specified as a fast or slow fading channel, it does not specify whether the channel is flat fading or frequency selective in nature.
- ✓ Fast fading only deals with the rate of change of the channel due to motion.
- ✓ In flat fading channel, the impulse response of the flat fading channel is a delta function (no time delay).
- ✓ Hence, a *flat fading*, *fast fading channel* is a channel in which amplitude of the delta function varying faster than the rate of change of the transmitted baseband signal.
- ✓ In case of a *frequency selective,* fast fading channel, the amplitudes, phases, and time delays of any one of the multipath components vary faster than the rate of change of the transmitted signal.
- ✓ In practice, fast fading only occurs for very low data rates.

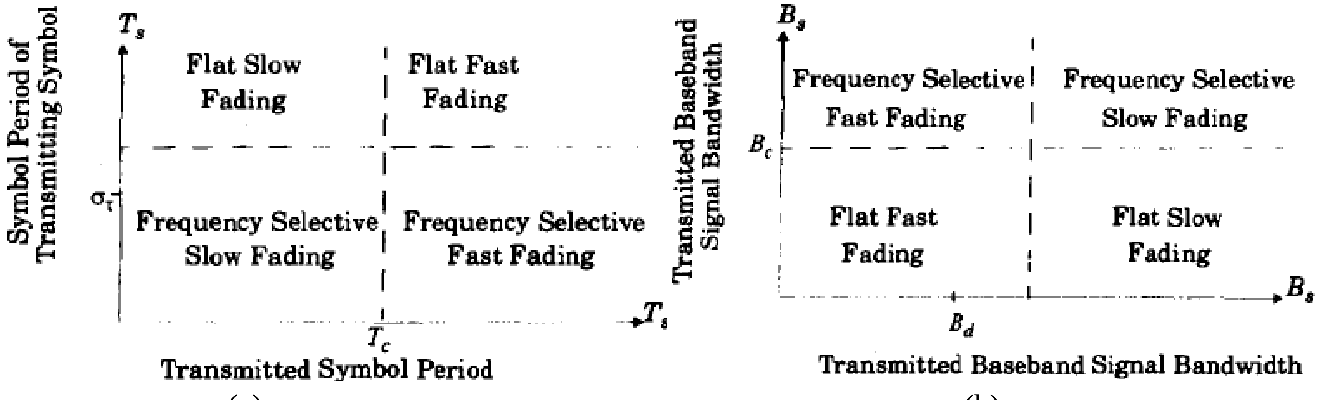
b. Slow Fading

- ✓ In a slow fading channel, the channel impulse response changes at a rate much slower than the transmitted baseband signal *s(t)*.
- ✓ In this case, the channel is assumed to be static over one or several reciprocal bandwidth intervals.
- ✓ In the frequency domain, the Doppler spread of the channel is much less than the bandwidth of the baseband signals.
- ✓ Therefore, a signal undergoes slow fading if

$$T_S << T_c$$

$$B_S >> B_D$$

- ✓ The relation between the various multipath parameters and the type of fading experienced by the signal are summarized in Figure.



(a) (b)

Figure 1.19: Matrix illustrating type of fading experienced by a signal as a function of

(a) symbol period

(b) baseband signal bandwidth

State the difference between small-scale and large-scale propagation.

LARGE-SCALE PROPAGATION	SMALL-SCALE PROPAGATION
redicts mean signal strength for an arbitrary transmitter-receiver (T-R) separation distance are useful in estimating the radio coverage area of a transmitter is called large-scale propagation	Rapid fluctuations of the received signal strength over very short travel distance/short duration are called Small-scale propagation.
As the mobile moves away from transmitter over large distances, the local average received signal will gradually decrease	As the mobile moves away from transmitter over small distances, , the received signal may fluctuate, giving rise to small scale fading
The local average signal is computed by large scale propagation models typically (Computed by averaging signal measurements over measurement track) cellular=> 1 GHz -2 GHz band→ power movement from 1m to 10m.	The received power may vary from [30/40 dB] when the receiver is moved by fraction of wavelength.

Solved Examples

Example 1.1 A mobile is located 10 Km away from a base station and uses a vertical λ/4 monopole antenna with a gain of 3 dB to receive cellular radio signals. The E-field at 1 km from the transmitter is measured to be 10^{-3} V/m. The carrier frequency used for this system is 1000 MHz.

(a) Find the length and effective aperture of the receiving antenna

(b) Find the received power at the mobile using the two-ray ground reflection model assuming h_t is 50 m and h_r is 1.5 m above ground.

Given data

Transmitter-receiver distance = 10 Km

E-filed at 1 Km = 10^{-3} V/m

Frequency of operation = f = 1000 MHz

Solution:

(*a*) Length of antenna $L = \frac{\lambda}{4}$

$$\text{Wavelength } \lambda = \frac{c}{f}$$

$$= \frac{3 \times 10^8}{1000 \times 10^6} = 0.3\ m$$

$$\text{Length } L = \frac{\lambda}{4}$$

$$= \frac{0.3}{4} = 0.075\ m$$

$$\text{Effective aperture of antenna } A_e = \frac{G\,\lambda^2}{4\,\pi}$$

$$= \frac{(0.3)^2}{4\,\pi} = 0.015\ m^2$$

(b) Received Power $P_r(d) = \dfrac{|E|^2 G_r \lambda^2}{377 (4\pi)}$ watts

$$\text{Electric field } |E| = \frac{2 E_o \ d_o}{d}\left(\frac{2\pi\, h_t\, h_r}{\lambda\ d}\right)$$

$$= \frac{2 \times 10^{-3} \times 10^3}{10 \times 10^3}\left(\frac{2\pi(50)(1.5)}{0.3 \times 10 \times 10^3}\right)$$

$$\text{Received Power } P_r = 3.14 \times 10^{11} \text{ watts}$$

Example 1.2 *Consider a transmitter which radiates a sinusoidal carrier frequency of 2000 MHz. For a vehicle moving 100 mph, compute the received carrier frequency if the mobile is moving*

(a) directly toward the transmitter

(b) away from the transmitter

(c) direction which is perpendicular to the direction of arrival of the transmitted signal.

Given Data:

Carrier frequency f_c = *2000MHz*

Vehicle speed V = *100 mph*

Solution:

(a) Doppler shift is positive for vehicle is moving towards the transmitter

$$\text{Received frequency } f = f_c + f_d$$

$$= 2000 \times 10^6 + \frac{V}{\lambda}$$

$$= 2000 \times 10^6 + \frac{100 \times 60 \times 60}{0.15}$$

$$= 2002 \text{ MHZ}$$

$$f = \frac{c}{\lambda}$$

$$\lambda = \frac{c}{f_c}$$

$$= \frac{3 \times 10^8}{2000 \times 10^6} = 0.15\text{m}$$

(b) Doppler shift is negative for vehicle is moving away from the transmitter

$$f = f_c - f_d$$

$$= 2000 \times 10^6 - \frac{V}{\lambda}$$

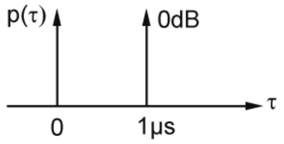
$$= 1997 \text{ MHZ}$$

(*c*) Vehicle is moving perpendicular to the angle of arrival of the transmitted signal

$\theta = 90°$ $\cos\theta = \cos 90° = 0$ there is no Doppler shift received frequency = transmitted frequency.

Example 1.3 *Compute the RMS delay spread for power delay profile*

(*a*)



(b) If BPSK modulation is used, what is the maximum bit rate that can be sent through the channel?

Solution:

(*a*) RMS delay spread $\sigma_\tau = \sqrt{\tilde{\tau}^2 - (\tilde{\tau})^2}$

$$\text{Mean excess delay } \bar{\tau} = \frac{\sum_K P(\tau_K)\tau_K}{\sum_K P(\tau_K)}$$

$$= \frac{1(0) + 1(1)}{1+1} = \frac{1}{2}$$

$$= 0.5\ \mu s$$

$$\bar{\tau}^2 = \frac{\sum_K P(\tau_K)\tau_K^2}{\sum_K P(\tau_K)}$$

$$= \frac{1(0)^2 + 1(1)^2}{1+1}$$

$$= \frac{1}{2} = 0.5\ \mu\text{sec}$$

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

$$= \sqrt{0.5 - (0.5)^2} = 0.5\ \mu\text{sec}$$

(*b*) Bit rate R_b = symbol rate R_S

$$R_S = \frac{1}{T_s}$$

$$\frac{\sigma_\tau}{T_s} \le 0.1$$

$$T_s \ge \frac{\sigma_\tau}{0.1} = \frac{0.5\ \mu s}{0.1} = 5\ \mu s$$

$$R_S = \frac{1}{T_s}$$

$$= \frac{1}{5 \times 10^6} = 200 \text{ ksps}$$

$$\text{Bit rate } R_b = 200 \text{ kbps}$$

(*c*) Coherence bandwidth $B_C = \frac{1}{5\ \sigma_\tau}$

$$= \frac{1}{5(0.5 \times 10^{-6})} = 4 \text{ MHZ}$$

Example 1.4 *Calculate coherence time, Doppler spread for carrier frequency f_c = 1900 MHz and V = 50 m/s of moving vehicle distance of 10 m.*

Solution:

(*a*) Coherence time $T_c = \frac{9}{16\,\pi f_m} = \frac{9\,\lambda}{16\,\pi\,V} = \frac{9\,c}{16\,\pi\,V f_c}$

$$\text{Wavelength } \lambda = \frac{c}{f_c}$$

$$V = f\lambda$$

$$\text{Velocity } V = f_m \lambda$$

$$f_m = \frac{V}{\lambda}$$

$$T_c = \frac{9 \times 3 \times 10^8}{16\,\pi(50 \times 1900 \times 10^6)} = 565\ \mu s$$

(*b*) Doppler spread $B_D = f_m = \frac{V}{\lambda} = \frac{V f_c}{c} = \frac{50 \times 1900 \times 10^6}{3 \times 10^8}$

$$B_D = 316.66 \text{ Hz}$$

Example 1.5 ***Find the Fraunhofer (far-fied)distance for an antenna with maximum dimension of 1m and operating frequency of 9000 MHz. If antennas have unity gain, calculate the path loss.***
[Nov 2013, April 2017, Nov 2012, Nov 2009]
Solution:
Operating frequency, f = 900 MHz

$$\lambda = c/f = \frac{3\times10^8 m/s}{900M} = 0.33m$$

Fraunhofer distance, $df = \frac{2d2}{\lambda} = \frac{2(1)}{0.33} = 6m$

Path loss $PL(dB) = -10log\left[\frac{(\lambda^2)}{(4\pi)^2 d^2}\right] = -10\log\left[\frac{(0.33^2)}{(4\times3.14)^2 \bullet 6^2}\right] = 47dB$

Example 1.6 **If a transmitter produces 50W of power, express the transmit power in units of (a) dBm, and (b) dBW. If 50W is applied to a unity gain antenna with a 900 MHz carrier frequency, find the received power in dBm at a free space distance of 1000m from the antenna. What is P_r(10km)? Assume unity gain for the receiver antenna.**

Solution:

Given:

Transmitter power, $P_t = 50W$

Carrier frequency, $f_C = 900\ MHz$

Using equation $P_r(d)dBm = 10\log\left[\dfrac{P_r(d_o)}{0.001\ W}\right] + 20\log\left(\dfrac{d_o}{d}\right)$

(a) Transmitter power,

$$P_t(d)dBm = 10\log\left[\frac{P_t(mW)}{1\ mW}\right] = 10\log\left[50\times10^3\right] = 47.0\ dBm$$

(b) Transmitter power,

$$P_t(dBW) = 10\log\left[\frac{P_t(W)}{1\ W}\right] = 10\log\left[50\right] = 17.0\ dBW$$

The received power can be determined using equation

$$P_t = \frac{P_t\, G_t G_r \lambda^2}{(4\pi)^2 d^2 L} = \frac{50(1)(1)(1/3)^2}{(4\pi)^2(100^2)(1)} = (.5\times10^{-6})W = 3.5\times10^{-3} mW$$

$$P_r(\text{dBm}) = 10\log P_r(\text{mW}) = 10\log(3.5\times10^{-3}\ \text{mW}) = -24.5\,\text{dBm}$$

The received power at 10m can be expressed in terms of dBm using equation $P_r(d)dBm = 10\log\left[\dfrac{P_r(d_o)}{0.001\ W}\right]$, where d_o = 100mand d = 10km.

$$P_r(10k\ \text{m}) = P_r(100) + 20\log\left[\frac{100}{10000}\right]$$

$$= -24.5\ dBm - 40\ dB$$

$$= -64.5\ dBm$$

Example 1.6 **Determine the proper spatial sampling interval required to make small scale propagation measurement which assume that consecutive samples are highly correlated in time. How many samples will be required over *10m* travel distance if f_c=*1900 MHz* and v=50 m/s. How long would it take to make these measurements, assuming they could be made in real time from a moving vehicle? What is the Doppler spread B_D for the channel? *[April 2017]***

Solution:

For correlation, ensure that the time between samples is equal to $T_C/2$, and use the smallest value of T_C for conservative design.

Using equation $T_C \approx \dfrac{9}{16\pi f_m}$

$$T_C \approx \frac{9}{16\ \pi\ f_m} = \frac{9\lambda}{16\ \pi\ v} = \frac{9c}{16\ \pi\ v\ f_c}$$

$$= \frac{9\times3\times10^8}{16\times3.14\times50\times1900\times10^6} = 565\ \mu s$$

Taking time samples at less than half , at 282.5 T_C corresponds to a spatial sampling interval of

$$\Delta x = \frac{vT_C}{2} = \frac{50\times565\mu s}{2} = 0.014125\ m = 1.41\ cm$$

Therefore, the number of samples required over a 10m travel distance is

$$N_x = \frac{10}{\Delta x} = \frac{100}{0.014125} = 701\ samples$$

The time taken to make this measurement is equal to $\dfrac{10}{50ms} = 0.2s$

The Doppler spread is $B_D = f_m = \dfrac{vf_c}{c} = \dfrac{50\times1900\times10^6}{3\times10^8} = 316.66\ Hz$

1.7. Consider a transmitter which radiates a sinusoidal carrier frequency of 1850 MHz. For a vehicle moving 60 mph, compute the received carrier frequency if the mobile is moving directly toward the transmitter. [April/May 2018]

Given Data:

$$\text{Carrier frequency } f_c = 1850\text{MHz}$$

$$\text{Vehicle speed } V = 60 \text{ mph}$$

Solution:

Doppler shift is positive for vehicle is moving towards the transmitter

$$\text{Received frequency } f = f_c + f_d$$

$$= 1850 \times 10^6 + \frac{V}{\lambda}$$

$$= 1850 \times 10^6 + \frac{100\times60\times60}{0.15}$$

$$= 2002 \text{ MHZ}$$

$$f = \frac{c}{\lambda}$$

$$\lambda = \frac{c}{f_c}$$

$$= 0.15\text{m}$$

*****

TWO MARKS

1. **What is meant by multipath propagation? [Nov/Dec 2017]**

   Multipath propagation:

   Multipath means the transmitted signal may arrive at the receiver over many paths. The signal gets reflected and diffracted by different objects. So, each of the paths have a distinct amplitude, delay and direction of arrival. This effect is known as multipath propagation

2. **What is the major advantage of wireless communication?**

   Advantages of wireless communications:

   - ✓ **Mobility:** The users have freedom to move about while remaining connected, as compared with the network with its coverage area.
   - ✓ **Increased reliability:** Use of wireless technology eliminate cable failures, so overall reliability
   - ✓ **Ease of installation:** Wireless communications and networks make it easier for any office to be modified with new cubicles or furniture, without worrying about providing network connectivity through cables.
   - ✓ **Rapid disaster recovery:** Accidents may happen due to fire, etc., the organization hot prepared to recover such natural disasters. So, disaster recovery plan is must for business.

     Low cost: Eliminating the need to install cabling and using wireless communications results in significant cost savings.

3. **What is the significance of propagation model?**

   The major significance of propagation model is:

   i. Propagation model predicts the parameter of receiver.

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

4. **What are the types of propagation models?**

   The two types of propagation models are

Large Signal Propagation Models	Small Scale Fading Models
They characterize signal strength over large transmitter-receiver separation distances. e.g., several hundred or 1000s of meters.	They characterize signal strength over short travel distance. e.g., mobile moves over small distance, for cellular and PCS frequencies in the 1 GHz to 2 GHz band, coverage area from 1 m to 10 m.

5. **Define large scale propagation. [Nov 2010]**

   Large-scale propagation models predict the mean signal strength for an arbitrary transmitter-receiver (T-R) separation distance, which are useful in estimating the radio coverage area of a transmitter and they characterize signal strength over large T-R separation distances.

6. **Define path loss. [Nov 2012]**

   Path loss: 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.

7. **What is free space propagation model?**

   Free space propagation model: The free space propagation model is used to predict received signal strength, when the transmitter-receiver has a clear, line of sight path between them.

8. **What is free space propagation model? Write the expression for free space path loss. [June 2013]**

   Free space propagation model: The free space propagation model is used to predict received signal strength, when the transmitter-receiver has a clear, line of sight path between them.

   The path loss for the free space model when antenna gains are included is given by

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

The path loss for the free space model when antenna gains are excluded is given by

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

9. Write an expression for free space propagation model.

The received power is given by the Friis free space equation as

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

where,
- P_{tx} → transmitted power
- P_{RX} → the received power
- G_{TX} → the transmitter antenna gain
- G_{RX} → the receiver antenna gain
- d → the transmitter-receiver separation distance in meter
- λ → Wavelength in meters

10. List the different types of propagation mechanisms. [Nov 2014]

The different types of propagation mechanisms include

- Reflection
- Diffraction
- Scattering

11. What is reflection?

Reflection:

Reflection occurs when a propagating electromagnetic wave impinges upon an object, which has very large dimension when compared to the wavelength of propagating wave.

12. What is diffraction?

Diffraction:

Diffraction occurs when the radio path between the transmitter and receiver is obstructed by a surface that has sharp irregularities

13. What is scattering?

Scattering:

Scattering occurs when the medium through which the wave travels consists of objects with dimensions that are small compared to the wavelength and where the number of obstacles per unit volume is large.

14. What is small-scale fading?

Small-scale fading, due to the constructive and destructive interference of the multiple signal paths between the transmitter and receiver. This occurs at the spatial scale of the order of the carrier wavelength, and is frequency dependent.

15. What is large scale fading?

Large-scale fading, due to path loss of signal as a function of distance and shadowing by large objects such as buildings and hills. This occurs as the mobile moves through a distance of the order of the cell size, and is typically frequency independent

State the difference between small-scale fading and large-scale fading. [May 2015, May 2013]

Large-scale fading	Small-scale fading
The rapid fluctuations of the amplitudes, phases, or multipath delays of a radio signal over a long period of time or travel distance is known as large scale fading.	The rapid fluctuations of the amplitudes, phases; or multipath delays of a radio signal over a short period of time or travel distance is known as small scale fading.

16. Find the far-field distance for an antenna with maximum dimension of 2 m and operating frequency of 1GHz. [Nov 2015, Nov 2016]

Given:

Largest dimension of antenna, D = 2 meter

Operating frequency, f = 1GHz

To Find:

Far field distance, d_f

Solution:

$$\lambda = \frac{c}{f} = \frac{3\times10^8 m/s}{1\times10^9 Hz} m = 0.3\text{m}$$

Far field distance, $d_f = \frac{2D^2}{\lambda}$

$$= \frac{2(2)^2}{0.3} = 26.27\ m$$

17. Calculate the Brewster angle for wave impinging on ground having a permittivity ε_r = 5. [May 2016, Dec 2009]

Given:

Permittivity, ε_r =5

To find:

Brewster angle for wave

Solution:

$$\sin(\theta_i) = \frac{\sqrt{\varepsilon_r - 1}}{\sqrt{\varepsilon_r^2 - 1}} = \frac{\sqrt{(5)-1}}{\sqrt{(5)^2 - 1}} = \sqrt{\frac{4}{24}} = 0.4082$$

$$\theta_i = \sin^{-1}(0.4082) = 24.09^0$$

The Brewster angle for ε_r =5 is equal to 24.09^0

35. Calculate the Brewster angle for wave impinging on ground having a permittivity ε_r =4. (8m - May 2015, 8m - Nov 2013]

Given: Permittivity, ε_r =4

To find:

Brewster angle for wave

Solution:

$$\sin(\theta_i) = \frac{\sqrt{\varepsilon_r - 1}}{\sqrt{\varepsilon_r^2 - 1}} = \frac{\sqrt{(4)-1}}{\sqrt{(4)^2 - 1}} = \sqrt{\frac{3}{15}} = 0.577$$

$$\theta_i = \sin^{-1}(0.577) = 24.090$$

The Brewster angle for ε_r =4 is equal to 24.090

18. Interpret Snell's law. [May 2015, May 2013]

Snell's law state that

$$\sqrt{\mu_1\varepsilon_1}\sin(90-\theta_i) = \sqrt{\mu_2\varepsilon_2}\sin(90-\theta_t)$$

μ_1, μ_2 → Permittivity of two media

ε_1,ε_2 → Permeability of two media

θ_i → Incident angle

θ_t → Transmitted angle

19. List the advantages and disadvantages of 2 ray ground reflection model in the analysis of model in the analysis of path loss. [Dec 2012]

Advantages of 2 ray model:

- The 2 Ray model gives more accurate prediction at a long distance than the free space model.
- models predicts the mean received power at distance
- The 2 Ray model is used for mobile radio channels

Disadvantages of **2 ray model**:

- The formula is not applicable for short distances like 10 meters. not accurate for a distance less than approximately 4.7 Km for GSM 1800.
- The two-ray model does not give a good result for a short distance due to the oscillation caused by the constructive and destructive combination of the two rays.
- Generally, the transmitter antenna height will be at least 10 meters to clear trees and buildings.

20. What are the three most important effects of small-scale multipath propagation?
State the propagation Effects in mobile radio. [May 2014]

The three most important effects of small-scale multipath propagation are

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

21. What is Doppler shift?

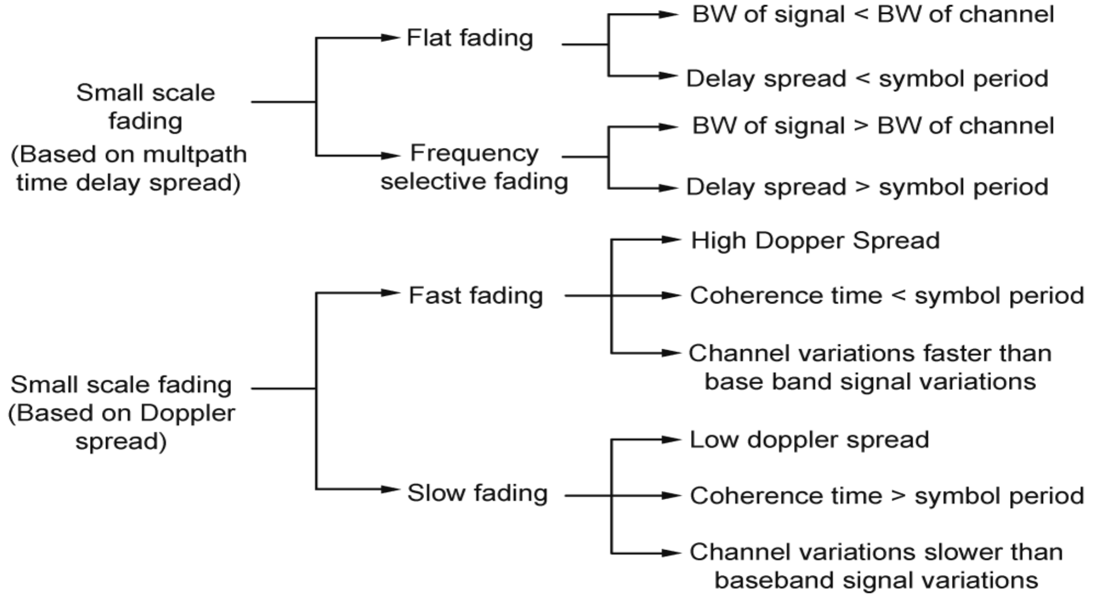
Doppler shift:

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).

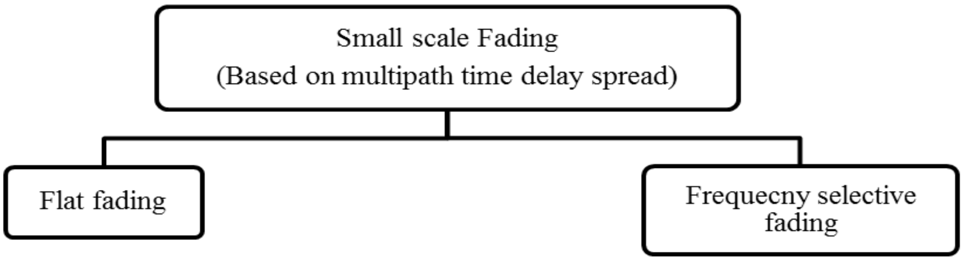
22. Differentiate the propagation effects with mobile radio. (or)
Compare fast and slow fading. [April/May 2018]

Slow Fading	Fast Fading
Slow variations in the signal strength	Rapid variations in the signal strength.
Mobile station (MS) moves slowly	Local objects reflect the signal causes fast fading.
It occurs when the large reflectors and diffracting objects along the transmission paths are distant from the terminal.	It occurs when the user terminal (MS) move for short distances.

23. What are the types of small scale fading? [May 013]



24. What are the different fading effects due multipath time delay spread?



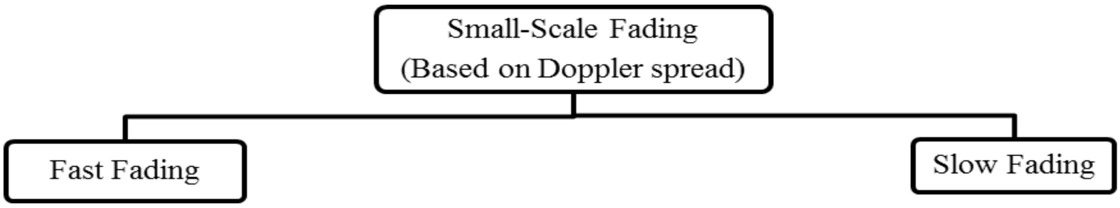
Flat fading:

1. Bandwidth of signal < Bandwidth of channel
2. Delay spread < Symbol period

Frequency selective fading:

1. Bandwidth of signal > Bandwidth of channel
2. Delay spread > Symbol period

25. What are the different fading effects due to Doppler spread? [Nov 2014]



Fast Fading:

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

Slow Fading:

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

26. Define coherence time and coherence bandwidth. [May 2016, Nov 2015, Nov 2016]

Coherence time is the maximum duration for which the channel can be assumed to be approximately constant. It is the time separation over which two received signals have strong potential for amplitude correlation.

Coherence bandwidth is the maximum frequency difference for which signals are strongly correlated in amplitude.

27. Give the Friis free space equation.

The ***Friis free space equation*** is given by

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

where,
$P_r(d)$ $\rightarrow$ Received power
P_t $\rightarrow$ Transmitted power
G_t $\rightarrow$ Transmitter antenna gain
G_r $\rightarrow$ Receiver antenna gain
d $\rightarrow$ T-R separation distance in meters
λ $\rightarrow$ Wavelength in meters
L $\rightarrow$ System loss factor

28. Define EIRP.

EIRP:

EIRP (Equivalent Isotropic 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

29. Give the formula to calculate Fraunhofer distance.

Fraunhofer distance is given by

$$d_f = \frac{2D^2}{\lambda}$$

where, D $\rightarrow$ Largest physical linear dimension of the antenna.

d_f $\rightarrow$ Far-field distance

λ $\rightarrow$ Wavelength in meters

30. When miscellaneous loss occurs?

The miscellaneous losses L are usually due to

- Transmission line attenuation
- Filter losses
- Antenna losses in the communication system.

31. Define path loss.

Path loss:

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.

32. Give the path loss for the free space model.

The path loss for the free space model when antenna gains are included is given by

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

The path loss for the free space model when antenna gains are excluded is given by

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

33. Give the path loss for the 2-ray model.

The path loss for the 2-ray model (with antenna gains) can be expressed in dB as

$$PL(dB) = 40\log d - (10\log G_t + 10\log G_r + 20\log h_t + 20\log h_r)$$

where,

h_t $\rightarrow$ Height of the transmitter

h_r $\rightarrow$ Height of the receiver.

d $\rightarrow$ T-R separation distance in meters

G_t $\rightarrow$ Transmitter antenna gain

G_r $\rightarrow$ Receiver antenna gain

34. What is the necessity of link budget?

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 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.

35. Express Log-distance Path Loss Model mathematically.

Average received signal power decreases logarithmically with distance.

$$\overline{PL}(d) \propto \left(\frac{d}{d_0}\right)^n$$

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

where,

$\overline{PL}(d)$ → Average large-scale path loss

d → T-R separation

d_0 → Close-in reference distance

n → Path loss exponent

36. Differentiate ISI, fading, attenuation, shadowing and small scale, large scale fading? Nov/dec 2012

ISI: Signal dispersion leads to Inter Symbol Interference (ISI) at the Receiver.

Fading: Variations in signal strength are known as fading. It describes how the received signal amplitude changes with time.

Small Scale Fading is used to describe the rapid fluctuations of the amplitudes, phases or multipath delays of a radio signal over a short period of time. Wavelength $\lambda \simeq 1\ m$

Attenuation: It is the drop in the signal power when transmitting from one point to another.

It can be caused by the transmission path length, obstructions in the signal path and multipath effects.

Shadowing of the signal can occur whenever there is an obstruction between the transmitter and receiver.

Eg., buildings and hills

Large-scale fading causes signal power attenuation due to motion over large area.

Eg., large terrain (ex. hills, forest, billboard…) between the transmitter and the receiver

37. Define shadowing. [Nov 2012]

Shadowing of the signal can occur whenever there is an obstruction between the transmitter and receiver.

38. Give various small scale fading parameters of Mobile Multipath Channels? [Dec 2012]

Small scale fading parameters of Mobile Multipath Channels

(i) Mean excess delays

(ii) RMS delay spread

(iii) Excess delay spread

39. Express Log-normal shadowing mathematically.

The path loss PL(d) at a particular location is random and distributed log-normally (normal in dB) about the mean distance dependent value .That is

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

$$P_r(d)[dBm] = P_t[dBm] - PL(d)[dB]$$

where X_σ → Zero-mean Gaussian distributed random variable (in dB)

σ → Standard deviation (in dB)

40. What is flat fading? [Nov 2012, Nov /Dec 2017]

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

41. What is frequency selective fading? How to avoid fading problem? [May 2012]

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

42. What are the factors influencing small scale fading?

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

Narrow band signal with bandwidth $B > B_C$, then the channel behaves like frequency selective fading.

It occurs when

$$T_s \approx \frac{1}{B} \square \frac{1}{B_c} \approx \sigma T_m \quad \text{causes performance degradation}$$

where,

$\sigma Tm \rightarrow$ rms delay spread

$Ts \rightarrow$ symbol duration

$B \rightarrow$ Bandwidth

$Bc \rightarrow$ Coherence bandwidth

43. Define mean excess delay and rms delay spread. [Nov 2015]

The mean excess delay is the first moment of the power delay profile and is defined to be

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

The rms delay spread is the square root of the second central moment of the power delay profile and is defined to be

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

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

44. When a signal undergoes flat fading?

A signal undergoes flat fading if

$$B_S << B_c$$

$$T_S >> \sigma_\tau$$

where Bs $\rightarrow$ Signal Bandwidth.

Bc $\rightarrow$ Coherence bandwidth

Ts $\rightarrow$ Reciprocal bandwidth

$\sigma\tau$ $\rightarrow$ rms delay spread

45. When a signal will undergoe frequency selective fading?

A signal undergoes frequency selective fading if

$$B_S > B_c$$

$$T_S < \sigma_\tau$$

where $B_s \rightarrow$ Signal Bandwidth.

$B_c \rightarrow$ Coherence bandwidth

$T_s \rightarrow$ Reciprocal bandwidth

$\sigma_\tau \rightarrow$ *rms* delay spread

46. When a signal undergoes fast fading?

A signal undergoes fast fading if

$$T_S > T_c$$

$$B_S < B_D$$

where B_S → Bandwidth of the transmitted modulation

T_S → Reciprocal bandwidth of the transmitted modulation

47. When a signal will undergo slow fading?

A signal undergoes fast fading if

$$T_S << T_c$$

$$B_S >> B_D$$

where B_S → Bandwidth of the transmitted modulation

T_S → Reciprocal bandwidth of the transmitted modulation

48. State the difference between small-scale and large-scale propagation.

LARGE-SCALE PROPAGATION	SMALL-SCALE PROPAGATION
Predicts the mean signal strength for an arbitrary transmitter-receiver (T-R) separation distance are useful in estimating the radio coverage area of a transmitter is called large-scale propagation	Rapid fluctuations of the received signal strength over very short travel distance/short duration are called Small-scale propagation.
As the mobile moves away from transmitter over large distances, the local average received signal will gradually decrease	As the mobile moves away from transmitter over small distances, , the received signal may fluctuate, giving rise to small scale fading

49. What is fading and Doppler spread? [Nov 2013, Nov 2016]

- **Fading:** The term small-scale fading or simply ***fading***, means rapid fluctuations of the amplitudes, phases, or multipath delays of a radio signal over a short period of time or short travel distance, so that the large scale path loss effects may be ignored
- ✓ **Doppler spread:** Doppler spread B_D is a measure of the spectral broadening caused by the time rate of change of the mobile radio channel.

50. What is Doppler spread? [May 2016]

- ✓ Doppler spread B_D is a measure of the spectral broadening caused by the time rate of change of the mobile radio channel.
- ✓ Doppler spread B_D is defined as the range of frequencies over which the received Doppler spectrum is essentially non-zero.

51. Distinguish between Narrowband and Wideband systems. [DEC 2012, DEC 2013]

Sl. No.	Narrow band system	Wide band system
1.	In narrow band system, the available radio spectrum is divided into a large number of narrowband channels.	In wideband systems, the transmission bandwidth of a single channel is much larger than the coherence bandwidth of the channel.
2.	Small delay spread	Large delay spread
3.	High coherence bandwidth	Small coherence bandwidth

52. What is coherence bandwidth?

- ✓ Coherence bandwidth is defined as the bandwidth over which the frequency correlation function is above 0.9

$$B_C = \frac{1}{50\,\sigma_\tau}$$

If the frequency correlation function is above 0.5, then

$$B_C = \frac{1}{5\,\sigma_\tau}$$

$$\sigma_\tau \rightarrow \textit{rms} \text{ delay spread}$$

Coherence bandwidth and *rms* delay spread is a function of specific channel impulse responses and applied signals.

53. What is meant by Doppler spread?

Doppler Spread is defined as the range of frequencies over which the received Doppler spectrum is essentially non-zero. Doppler spread is a measure of the spectral broadening caused by the time rate of change of the mobile radio channel.

$f_c \rightarrow$ Pure sinusoidal tone of frequency

$f_d \rightarrow$ Doppler shift

If f_c is transmitted then received Doppler spectrum will have components

spectrum = $f_c + f_d$ and $f_c - f_d$

54. Define coherence time. In what way does this parameter decide the behavior of wireless channel? [April 2017, Dec 2015]

- ➢ Coherence time is the time over which two signals are having strong potential for amplitude correlation.
- ✓ Coherence time is the range of time over which similar fading occurs.
- ✓ The Doppler spread and coherence time are inversely proportional to one another.

$$\textit{Cohrence Time} = \frac{1}{\textit{Doppler Spread}}$$

55. Define mean excess delay. [Dec 2015]

Mean excess delay

- o The mean excess delay is the first moment of the power delay profile (PDP).
- o It is expressed as

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

56. Define RMS delay spread. [Dec 2015]

RMS delay spread

- ✓ This is the most important single measure for the delay times extent of a multipath delay channel.
- ✓ This parameter calculates the standard deviation value of the delay of reflections.
- ✓ The standard deviation value will be weighted proportional to the energy in the reflected waves.
- ✓ The *rms* delay spread is the square root of the second central moment of the power delay profile.

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

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

where,

σ_τ $\rightarrow$ *rms* delay spread

a_k $\rightarrow$ Amplitude

$P(\tau_k)$ $\rightarrow$ Relative power levels of the individual multipath components

τ_k $\rightarrow$ Excess delay

58. Define maximum excess delay. [Dec 2015]

Maximum excess delay

- ✓ The maximum excess delay (X dB) of the power delay profile is defined to be the time delay during which multipath energy falls to X dB below the maximum.
- ✓ The maximum excess delay is defined $\tau_x - \tau_0$ as

  Where, τ_0 $\rightarrow$ First arriving signal

  τ_x $\rightarrow$ Maximum delay at which a multipath component is within X dB of the strongest multipath signal
- ✓ Maximum excess delay is sometimes called the ***excess delay spread***.

59. Give the difference between frequency flat and frequency selective fading. [April/May 2018]

Sl. No.	**Flat fading**	**Frequency selective fading**
1.	Bandwidth of signal < Bandwidth of channel	Bandwidth of signal > Bandwidth of channel
2.	Delay spread < Symbol period	Delay spread > Symbol period

ANNA UNIVERSITY, CHENNAI
AFFILIATED INSTITUTIONS
B.E. ELECTRONICS AND COMMUNICATION ENGINEERING
SEMESTER IV (R-2013)
SYLLABUS

EC6801 WIRELESS COMMUNICATION

OBJECTIVES: The student should be made to:

- Know the characteristic of wireless channel
- Learn the various cellular architectures
- Understand the concepts behind various digital signaling schemes for fading channels
- Be familiar the various multipath mitigation techniques
- Understand the various multiple antenna systems

UNIT I WIRELESS CHANNELS

Large scale path loss – Path loss models: Free Space and Two-Ray models -Link Budget design – Small scale fading- Parameters of mobile multipath channels – Time dispersion parameters-Coherence bandwidth – Doppler spread & Coherence time, Fading due to Multipath time delay spread – flat fading – frequency selective fading – Fading due to Doppler spread – fast fading – slow fading.

UNIT II CELLULAR ARCHITECTURE

Multiple Access techniques - FDMA, TDMA, CDMA – Capacity calculations–Cellular concept- Frequency reuse - channel assignment- hand off- interference & system capacity- trunking & grade of service – Coverage and capacity improvement.

UNIT III DIGITAL SIGNALING FOR FADING CHANNELS

Structure of a wireless communication link, Principles of Offset-QPSK, p/4-DQPSK, Minimum Shift Keying, Gaussian Minimum Shift Keying, Error performance in fading channels, OFDM principle – Cyclic prefix, Windowing, PAPR.

UNIT IV MULTIPATH MITIGATION TECHNIQUES

Equalisation – Adaptive equalization, Linear and Non-Linear equalization, Zero forcing and LMS Algorithms. Diversity – Micro and Macrodiversity, Diversity combining techniques, Error probability in fading channels with diversity reception, Rake receiver,

UNIT V MULTIPLE ANTENNA TECHNIQUES

MIMO systems – spatial multiplexing -System model -Pre-coding - Beam forming - transmitter diversity, receiver diversity- Channel state information-capacity in fading and non-fading channels.

TOTAL: 45 PERIODS

OUTCOMES: At the end of the course, the student should be able to:

- Characterize wireless channels
- Design and implement various signaling schemes for fading channels
- Design a cellular system
- Compare multipath mitigation techniques and analyze their performance
- Design and implement systems with transmit/receive diversity and MIMO systems and analyze their performance

TEXTBOOKS:

1. Rappaport,T.S., "Wireless communications", Second Edition, Pearson Education, 2010.
2. Andreas.F. Molisch, "Wireless Communications", John Wiley – India, 2006.

REFERENCES:

1. David Tse and Pramod Viswanath, "Fundamentals of Wireless Communication", Cambridge University Press, 2005.
2. Upena Dalal, " Wireless Communication", Oxford University Press, 2009.
3. Van Nee, R. and Ramji Prasad, "OFDM for wireless multimedia communications", Artech House, 2000.

UNIT II
CELLULAR ARCHITECTURE

Syllabus

Multiple Access techniques - FDMA, TDMA, CDMA – Capacity calculations–Cellular Concept Frequency reuse - channel assignment- hand off- interference & system capacity- trunking & grade of service – Coverage and capacity improvement.

INTRODUCTION

Cellular concept:

If a given set of frequencies or radio channels can be reused without increasing the interference, then the large geographical area covered by a single high power transmitter can be divided into a number of small areas, each allocated power transmitters with lower antennas can be used.

Wireless communication systems

Simplex

- ✓ Simplex is a one-way communication.
- ✓ Single transmitter (Satellite) sends information to multiple receivers.

  Eg:
  - Radio
  - TV Broadcast systems

Duplexing

- ✓ Duplexing is two-way communication.
- ✓ Both terminals can transmit simultaneously.
- ✓ Any duplex channel actually consists of two simplex channels. (Forward and Reverse).
- ✓ Duplexing may be done using frequency or time domain techniques.

  Eg:
  - Wireless telephone systems
- ✓ For cellular systems it is possible to talk or send data in both directions simultaneously.
- ✓ When the subscriber gets information from the base station, it transmits the information to base station at the same time.
- ✓ It is represented as duplexing.
- ✓ There is a need of duplexing in wireless telephone systems.
- ✓ It is based on
  - o Time domain
  - o Frequency domain

Frequency division duplexing (FDD)

- ✓ FDD provides two distinct bands of frequencies for every user.
  - Forward band provides traffic from the base station to the mobile.
  - Reverse band provides traffic from the mobile to the base.

Time division duplexing (TDD)

- ➢ TDD uses time to provide both a forward and reverse link.
- ➢ Individual users are allowed to access the channel in assigned time slot.
- ➢ Each duplex channel has both a forward time slot and a reverse time slot to facilitate bidirectional communication.

2.1. MULTIPLE ACCESS TECHNIQUES

1. Summarize the features of various multiple access techniques used in wireless mobile communication. State the advantages and disadvantages of each technique. (16m-May 2016)

- ✓ Multiple access techniques are used to allow a more number of mobile users to share the allocated spectrum in the most efficient manner.
- ✓ As the spectrum is limited, the sharing is required to increase the capacity of cell or over a geographical area by allowing the available bandwidth to be used at the same time by different users.
- ✓ Multiple access or channel access method is based on a multiplexing method that allows several data streams or signals to share the same communication channel or physical medium.
- ✓ Multiplexing is in this context provided by the physical layer.

Types of multiple access techniques:

- ✓ There are several methods in which the multiple users can send information through the communication channel to the receiver.
  - (i) **FDMA (**Frequency division multiple access): The total bandwidth is divided into non-overlapping frequency sub bands.
  - (ii) **TDMA (**Time division multiple access): The total bandwidth is divided into time slots, different timeslots is assigned to different users.
  - (iii) **CDMA (**Code division multiple access): Many users share the same frequency with same time using different coding.

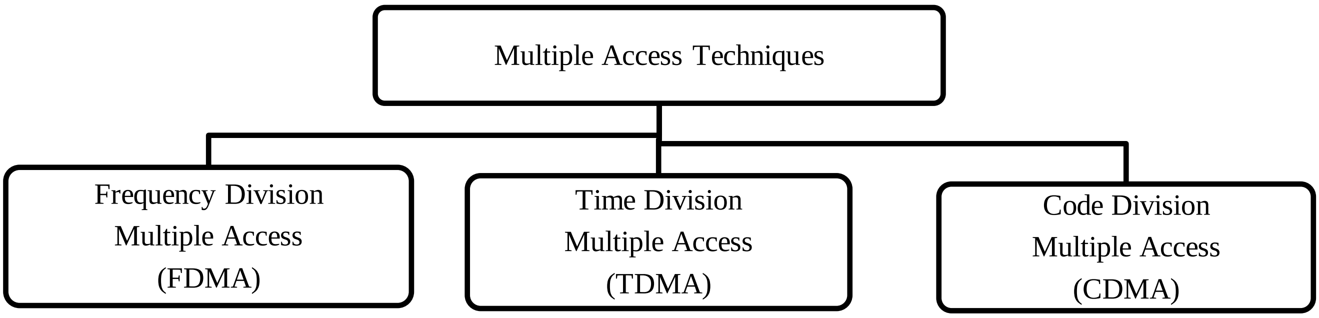


Figure: Types of multiple access techniques

Applications of multiple access techniques:

- ✓ The first generation wireless systems used FDMA
- ✓ In second generation TDMA and CDMA are used.
- ✓ CDMA is the targeted multiple access technology for the third generation wireless communication systems.
- ✓ This Multiple access techniques can be grouped as narrowband and wideband systems, depending upon how the available bandwidth is allotted to the user.

Narrowband Systems:

- ✓ In a narrowband multiple access system, the available radio spectrum is divided into a large number of narrowband channels.
- ✓ The channels are usually operated using FDD.
- ✓ FDMA and TDMA are fixed capacity allocation schemes
- ✓ An individual user is assigned a frequency band (In FDMA) or a time slot (TDMA) for the duration of the connection.

Wideband systems:

- ✓ In a wideband multiple access system, large number of transmitters are allowed to transmit on the same channel.
- ✓ CDMA is spread spectrum technique. Based on the orthogonal property, an individual user can transmit using the entire system bandwidth.

1. FREQUENCY DIVISION MULTIPLE ACCESS (FDMA)

1. Explain any one type of multiple access schemes. (6M-May 2012)

Concept of FDMA

- ✓ FDMS (frequency division multiplexing) is the division of frequency band allocated for the wireless telephone communication.
- ✓ FDMA is a basic technology in the Analog Mobile Phone Service (AMPS).
- ✓ Frequency division multiple access (FDMA) assigns individual channels to individual users.
- ✓ Each user is allocated a unique frequency band or channel.
- ✓ These channels are assigned on demand to users who request service.
- ✓ During the period of the call, no other user can share the same frequency band.

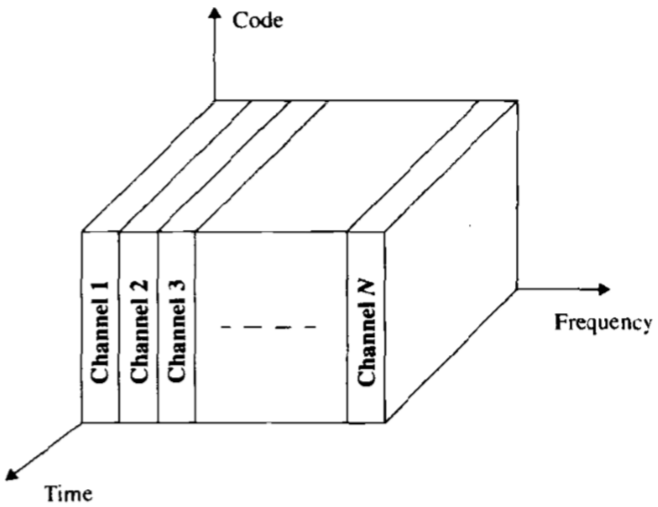


Figure: FDMA where different channels are assigned different frequency bands.

- ✓ In FDD systems, the users are assigned a channel as a pair of frequencies.
  - One frequency is used for the forward channel, while the other frequency is used for the reverse channel.

Features of FDMA

The features of FDMA are as follows:

(Advantages)

- ✓ The FDMA channel carries ***only one phone circuit at a time***.
- ✓ After the assignment of a voice channel, the base station and the mobile ***transmit simultaneously and continuously***.
- ✓ The bandwidths of FDMA channels are relatively narrow (30 kHz), because each channel supports only ***one circuit per carrier***. That is, FDMA is usually ***implemented in narrowband systems***.
- ✓ The complexity of FDMA mobile systems is lower when compared to TDMA systems.
- ✓ FDMA is a ***continuous transmission scheme***, so, fewer bits are ***needed for overhead purposes*** (such as synchronization and framing bits) as compared to TDMA.
- ✓ ***Inter-symbol interference is low*** and no equalization is required.

(Disadvantages)

- ✓ If an FDMA channel is not in use, then it sits idle and cannot be used by other users to increase or share capacity. It is essentially a ***wasted resource***.
- ✓ FDMA need to use ***costly bandpass filters*** to eliminate spurious radiation at the base station.
- ✓ FDMA systems have higher cell site ***system costs as compared to TDMA systems***, because of the single channel per carrier design.
- ✓ The FDMA mobile unit uses duplexers. This increases in the cost of FDMA.

Nonlinear Effects in FDMA

- ✓ In FDMA system, many channels share same antenna at the base station.
- ✓ The power amplifiers or the power combiners, when operated at or near saturation for maximum power efficiency, are non- linear.
- ✓ The nonlinearities cause signal spreading in the frequency domain and generate intermodulation ***(IM)*** frequencies.
- ✓ IM is ***undesired RF radiation*** which can interfere with other channels in the FDMA systems.
- ✓ To minimize inter modulation distortion, RF filters are used.

Number of channels

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

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

where B_t - Total spectrum allocation (or) Bandwidth

B_{guard} - Guard band allocated at the edge of the allocated spectrum

B_C - Channel bandwidth

2. TIME DIVISION MULTIPLE ACCESS (TDMA)

1. Brief about the principle of time division multiple access (TDMA). (6M-May 2013)
2. Explain TDMA and discuss the time division multiple access frame structure.
3. Identify the channel capacity of TDMA in cell system. [Nov/Dec 2017]

Concept of TDMA

- ✓ Each user occupies a cyclically repeating time slot,
  - A channel may be thought of as particular time slot that reoccurs every frame, where N time slots comprise a frame.

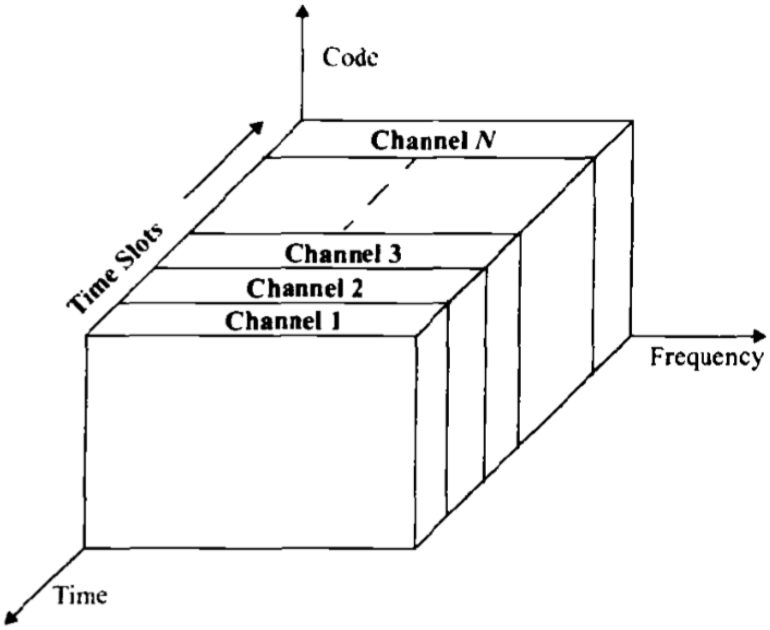


Figure: TDMA scheme where each channel occupies a cyclically repeating time slot.

- ✓ TDMA systems transmit data in a buffer-and-burst method, thus the transmission for any user is non continuous.
  - Digital data and digital modulation must be used with TDMA.

Frame structure

- ✓ The transmission from various users is interlaced into a repeating frame structure as shown in Figure.
- ✓ Frame consists of a number of slots. Each frame is made up of a preamble, an information message, and tail bits.
- ✓ Preamble contains the address and synchronization information that both the base station and the subscribers use to identify each other.
- ✓ Guard times allow synchronization of the receivers between different slots and frames.

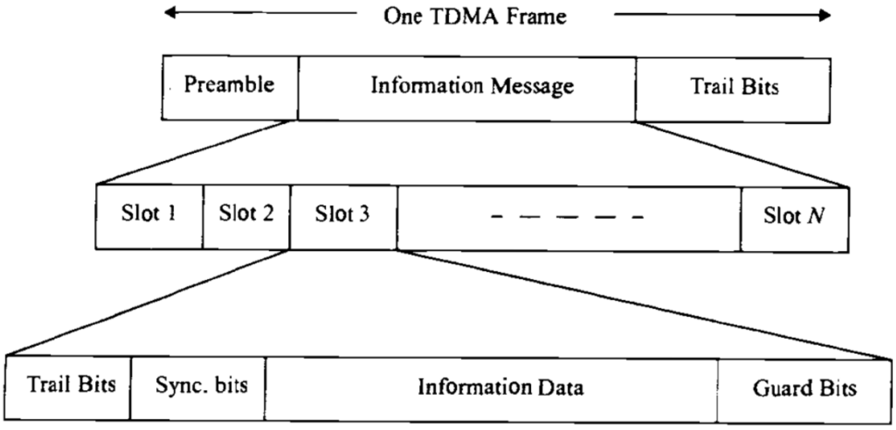


Figure: TDMA frame structure. The frame is cyclically repeated over time

Features of TDMA

The features of TDMA include the following:

(Advantages)

- ✓ TDMA shares a single carrier frequency with several users, where each user makes use of non overlapping time slots.
- ✓ Data transmission for users of a TDMA system is not continuous, but occurs in bursts. This results in low battery consumption, since the subscriber transmitter can be turned off when not in use.
- ✓ Because of discontinuous transmissions in TDMA, the handoff process is much simpler for a subscriber unit.
- ✓ TDMA uses different time slots for transmission and reception, thus duplexers are not required.

- ✓ Adaptive equalization is necessary in TDMA systems, since the transmission rates are generally very high as compared to FDMA channels.
- ✓ It is possible to allocate different numbers of time slots per frame to different users.

(Disadvantages)

- ✓ High synchronization overhead is required in TDMA systems because of burst transmissions.
- ✓ In TDMA, the guard time should be minimized.

Efficiency of TDMA

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

$$b_{OH} = N_r b_r + N_t b_p + N_t b_g + N_r b_g$$

where, N_r → Number of reference bursts per frame

N_t → Number of traffic bursts per frame

b_r → Number of overhead bits per reference burst,

b_p → Number of overhead bits per preamble in each slot, and

b_g → Number of equivalent bits in each guard time interval.

- ✓ The total number of bits per frame is

$$b_T = T_f R$$

where, T_f → Frame duration,

R → Channel bit rate.

- ✓ The frame efficiency is

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

Number of channels

- ✓ The number of channel in TDMA is given by

$$N = \frac{m\left(B_{tot} - 2B_{guard}\right)}{B_C}$$

where m → Maximum number of TDMA users supported on each radio channel.

3. **CODE DIVISION MULTIPLE ACCESS (CDMA)**

4. With neat illustration, explain CDMA. (6M-Nov 2014)

Concept of CDMA:

- ✓ In code division multiple access (CDMA) systems, the narrowband message signal is multiplied by a very large bandwidth signal called the spreading signal.
- ✓ The spreading signal is a pseudo-noise code sequence that has a chip rate which is orders of magnitudes greater than the data rate of the message.
- ✓ Each user has its own pseudorandom codeword which is approximately orthogonal to all other codewords.
- ✓ The receiver performs a time correlation operation to detect only the specific desired codeword.
- ✓ All other codewords appear as noise due to decorrelation. For detection of the message signal, the receiver needs to know the codeword used by the transmitter.
- ✓ Each user operates independently with knowledge of the other users.
- ✓ In CDMA, the power of multiple users at a receiver determines the noise floor after decorrelation.
- ✓ The symbol (chip) duration is very short and usually much less than the channel delay spread.

- ✓ A RAKE receiver can be used to improve reception by collecting time delayed versions of the required signal.

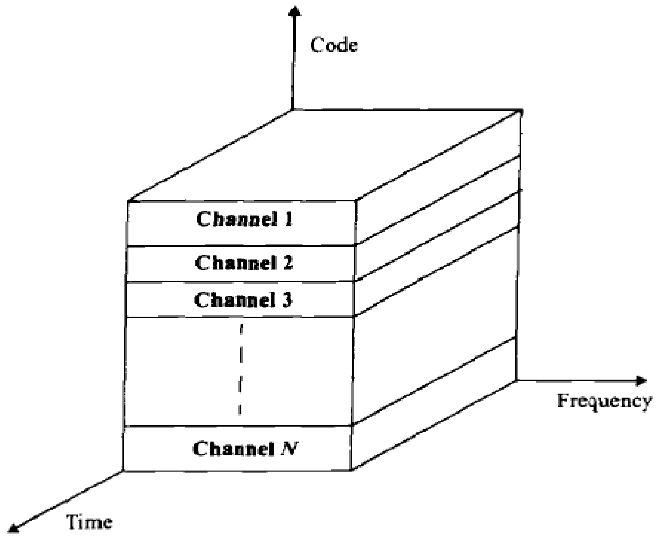


Figure: CDMA -Each channel is assigned a unique PN code

Advantages

- ✓ Frequency reuse
  - All users in a CDMA system use the same carrier frequency and may transmit simultaneously.
- ✓ Soft capacity
  - During peak hours, if the user can tolerate a lower QoS to a certain stage, the stem can accommodate more user to satisfy the high service demands in that period.
- ✓ Reduction in multipath fading
  - .Multipath fading may be reduced because the signal is spread over the large spectrum.
- ✓ Data rates
  - Channel data rates are very high in CDMA systems.
- ✓ Handoff performance
  - When mobile user is in the cell boundary, it can establish a connection with the new base station before terminating the connection with old base station. This will improve handoff performance.
- ✓ Flexibility
  - CDMA are more flexible than TDMA systems in supporting multimedia services.

Disadvantages

- ✓ Near-far problem
  - Some of the mobile units are close to the base station while others are far from it.
  - A strong signal received at the base from a near –in mobile unit and the weak signal from a far –end mobile unit. This phenomenon is called the near-far problem.
- ✓ Self-jamming
  - Self-jamming arises by the spreading sequences of different users are not exactly orthogonal.
  - In the despreading of a particular PN code, non-zero contributions to the receiver decision statistic for a desired user arise from the transmissions of other users in the system.

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COMPARISON OF TDMA, FDMA AND CDMA

1. **What are the major difference between TDMA, FDMA and CDMA? Explain in detail about each multiple access. (16m-May 2014)**

S.N	FDMA	TDMA	CDMA
1	Channel bandwidth is subdivided into number of sub channels	The radio spectrum is divided into time slots and each slot is allotted for only one user who can either transmit or receive.	Sharing of bandwidth and time takes place.
2	FDMA uses Narrow band Systems.	TDMA uses Narrow band Systems or wide band Systems	CDMA uses Wide band Systems.
3	FDMA is First generation wireless standard (1G).	TDMA is Second generation wireless standard (2G).	CDMA is third generation wireless standard (3G).
4	FDMA is use for the voice and data transmission	TDMA is used for data and digital voice signals	CDMA is use for digital voice signals and multimedia services.
5	Due to non-linearity of power amplifiers, inter-modulation products are generated due to interference between adjacent channels.	Due to incorrect synchronization there can be interference between the adjacent time slots.	Both type of interference will be present.
6	Synchronization is not necessary	Synchronization is necessary	Synchronization is not necessary
7	Code word is not required	Code word is not required	Code words are required
8	Guard bands between adjacent channels are necessary.	Guard times between adjacent time slots are necessary.	Guard bands and guard times are necessary.

****

2.2. THE CELLULAR CONCEPT

2. Explain the concept of cellular topology and cell fundamentals with examples. (Nov - 2015)

- ✓ If a given set of frequencies or radio channels can be reused without increasing the interference, then the large geographical area covered by a single high power transmitter can be divided into a number of small areas, each allocated power transmitters with lower antennas can be used.
- ✓ The ***Hexagon shape*** was chosen for cell because it provides the most effective transmission by approximating a circular pattern while eliminating gaps present between adjacent circles.
- ✓ Each cellular base station is allocated a group of radio channels to be used with a small geographic area called a cell.
- ✓ A group of cells that use a different set of frequencies in each cell is called a ***cell cluster***.

Types of cell

- ✓ The physical size of a cell varies, depending on user density and calling patterns.
  - Macro cells are large cells typically have a radius between 1 mile and 15 miles with base station transmit powers between 1W and 6W.
  - Microcells are the smallest cells typically have a radius between of 1500 feet or less with base station transmit powers between 0.1W and 1W.
- ✓ Microcells are used in high-density areas such as in large cities and inside the buildings.
- ✓ Cellular radio signals are too weak to provide reliable communication at indoor, especially in well-shielded areas or areas with high levels of interference. To overcome this, very small cells, called picocells are used in same frequencies as regular cells in the same areas.

Location of base station

- ✓ When designing a system using hexagonal-shaped cells, main consideration is the location of the base station transmitters.

- ***Center-excited cell- Base station transmitters*** can be located in the center of the cell and uses Omni directional antennas which radiate and receive signals equally well in all directions.
- ***Edge- excited cell- Base station transmitters*** can be located in the edge of the cell and uses sectored antennas which radiate for a particular direction.
- ***Corner- excited cell- Base station transmitters*** can be located in the corner of the cell and uses sectored directional antennas.

Cellular system

- ✓ Figure shows a basic cellular system which consists of mobile stations, base stations and a mobile switching center (MSC).
- ✓ The Mobile Switching Center is sometimes called a mobile telephone switching office (MTSO), since it is responsible for connecting all mobiles to the PSTN in a cellular system.
- ✓ Each mobile communicates via radio with one of the base stations and may be handed off to any number of base stations throughout the duration of a call.
- ✓ The mobile station contains a transceiver, an antenna, and control circuitry, and may be mounted in a vehicle or used as a portable hand-held unit.

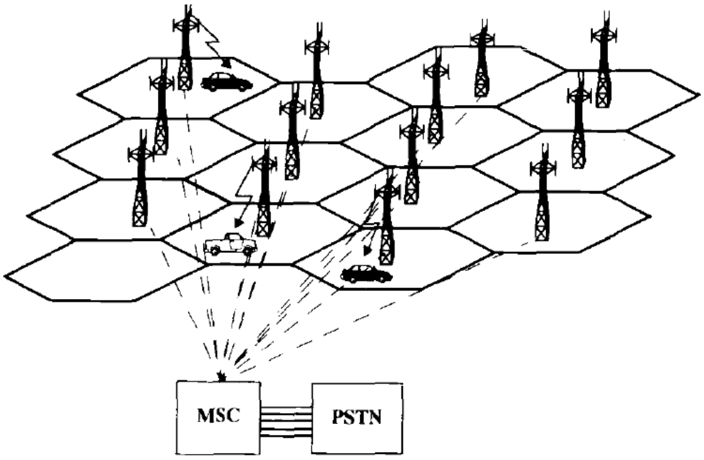

Figure: Cellular system

Base station:

- ✓ The base stations consist of several transmitters and receivers which simultaneously handle full duplex communications.
- ✓ The base stations generally have towers which support several transmitting and receiving antennas.
- ✓ The base station serves as a bridge between all mobile users in the cell and connects the simultaneous mobile calls via telephone lines or microwave links to the MSC.

Mobile Switching Center:

- ✓ The MSC coordinates the activities of all of the base stations and connects the entire cellular system to the PSTN.
- ✓ A typical MSC handles 100,000 cellular subscribers' and 5,000 simultaneous conversations at a time, and accommodates all billing and system maintenance functions.
- ➢ The channels used for voice transmission from the base station to mobiles are called ***forward voice channels*** (FVC).
- ✓ The channels used for voice transmission from mobiles to the base station are called ***reverse voice channels*** (RVC).
- ✓ The two channels responsible for initiating mobile calls are the ***forward control channels*** (FCC) and ***reverse control channels*** (RCC).
- ✓ Control channels are often called setup channels because they are only involved in setting up a call and moving it to an unused voice channel.

2.2.1. FREQUENCY REUSE

1. **Discuss in detail about cellular concept and frequency reuse. [8m] [Nov 2014]**
2. **Give the concept of cellular and explain with an example. [8m] [April 2010]**

- ✓ Each cellular base station is allocated a group of radio channels to be used with a small geographic area called cell. Cells are grouped into clusters. Each cluster utilizes the entire available radio spectrum.

Frequency reuse or Frequency planning

- ✓ Frequency reuse is the process in which the same set of frequencies can be allocated to more than one cell and the cells are separated by sufficient distance.
- ✓ The ability to reuse the frequencies offers a means to expand the total system capacity without the need to employ high power transmitters.
- ✓ Figure shows a geographic cellular radio coverage area containing three groups of cell called clusters. Each cluster has seven cells in it and all cells are assigned the same number of full duplex cellular telephone channels.
- ✓ Spatially reusing the available spectrum so that the same spectrum can support multiple users separated by a distance is the primary approach for efficiently using the spectrum.
- ✓ Cells with the same letter use the same set of frequencies.
- ✓ The letters A, B, C, D, E, F and G denote the seven sets of frequencies.
- ✓ A cell cluster is outlined in bold and replicate over the coverage area.
- ✓ The actual radio coverage of a cell is known as the foot print. It is determined from field measurement or propagation prediction models.

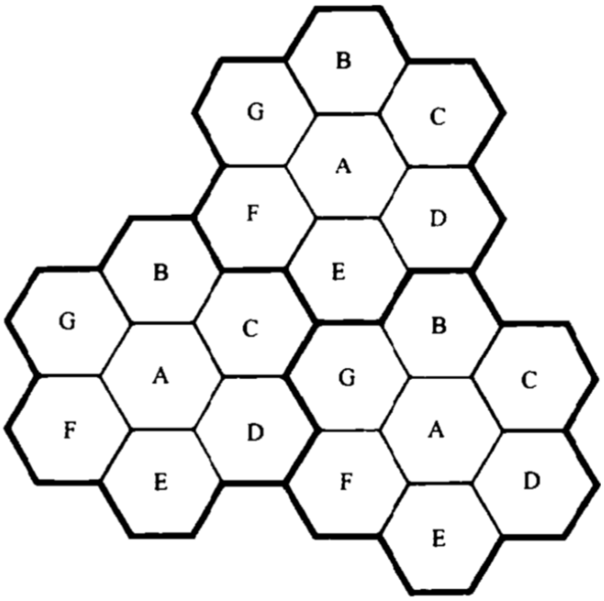

Figure: Illustration of the cellular frequency reuse concept.

Capacity expansion by frequency reuse

- ✓ Consider a cellular system which has a total of S duplex channels available for use.
- ✓ Let N be the cluster size in terms of the number of cells within it and each cells is allocated a group of K channels (K<S).
- ✓ The N cells which collectively use the complete set of available frequencies is called cluster. The cluster can be replicated many times to form the entire cellular communication systems.
- ✓ The N cells in the cluster would utilize all K available channels.
- ✓ For the total number of Channels C, available in the cluster is given

$$S = KN$$

where S → Number of full duplex cellular channels available in the cluster.

K → Number of channels in a cell
N → Number of cells in the cluster

- ✓ Let M be the number of times the cluster is replicated and C be the total number of channels used in the entire cellular system with frequency reuse. C is then the system capacity and is given by

$$C=MKN$$
$$C=MS$$

where C → Total channel capacity in a given area
M → Number of clusters in a given area

- ✓ The capacity of a cellular system is directly proportional to the number of times a cluster is replicated in a fixed service area.
- ✓ The cluster size factor N= 4, 7, or 12.If the cluster size N is reduced while the cell size is kept constant, more clusters are required to cover a given area and hence more capacity is achieved.

- ✓ The number of subscribers who can use the same set of frequencies in non-adjacent cells at the same area is dependent on the total number of cells in the area.
- ✓ The number of users use the same set of frequencies is called the frequency reuse factor (FRF) and is defined as

$$FRF = \frac{N}{C}$$

where N → Total number of full duplex in an area
C → Total number of full duplex in an a cell.

Rules for determining the nearest co-channel neighbors

- ✓ To find the nearest co-channel neighbors of a particular cell, one must do the following:
  - Step 1: Move I cells along any chain of hexagons;
  - Step 2: Turn 60 degrees counter clockwise and more j cells.

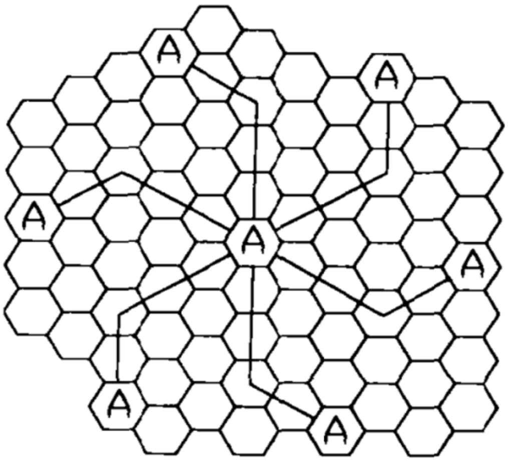

Figure: Method of locating co-channel cells in a cellular system.
In this example, N = 19 (i.e., i = 3, j = 2).

- ✓ The method of locating co-channel cells in a cellular system using the preceding rule is shown in figure for i = 3 and j= 2.
- ✓ The parameters i and j measure the number of nearest neighbor between co-channel cells, N is related to I and j by the equation

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

- ✓ Frequency reuse needs to be structured so that co-channel interference is kept at an acceptable level.
- ✓ As a distance between co-channel cell increases, co-channel interference will decrease.

- ✓ If cell size is fixed, the average signal-to-co-channel interference ratio will be independent of the transmitted power of each cell.
- ✓ Co-channel reuse ratio,

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

where Q → Co channel reuse ratio
D → Distance to the nearest co-channel cells
R → Radius of the cell
N → Number of cells in the cluster

- ✓ The advantages of Cellular Systems are
  - The use of low power transmitter and
  - It allows frequency reuse for capacity improvement.

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Problem:

If a total of 33 MHz of bandwidth is allocated to a particular FDD cellular telephone 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 uses (a) 4-cell reuse, (b) 7-cell reuse (c) 12-cell reuse. If 1 MHz of the allocated spectrum is dedicated to control channels, determine an equitable distribution of control channels and voice channels in each cell for each of the three systems.

Given:

- Total bandwidth =33 MHz
- Channel bandwidth = 25 kHz x 2 simplex channels = 50 kHz/duplex channel
- Total available channels = 33,000/50 = 660 channels

(a) For N= 4, total number of channels available per cell = 660/4 =165 channels.
(b) For N=7, total number of channels available per cell = 660/7 =95 channels.
(c) For N = 12, total number of channels available per cell = 660/12 = 55 channels.

- ✓ A 1 MHz spectrum for control channels implies that there are 1000/50 = 20 control channels out of the 660 channels available.
- ✓ To evenly distribute the control and voice channels, simply allocate the same number of channels in each cell wherever possible.
- ✓ Here, the 660 channels must be evenly distributed to each cell within the cluster. In practice, only the 640 voice channels would be allocated, since the control channels are allocated separately as 1 per cell.

(a) For N = 4, we can have 5 control channels and 160 voice channels per cell. In practice, however, each cell only needs a single control channel (the control channels have a greater reuse distance than the voice channels). Thus, one control channel and 160 voice channels would be assigned to each cell.

(b) For N = 7, 4 cells with 3 control channels and 92 voice channels, 2 cells with 3 control channels and 90 voice channels, and 1 cell with 2 control channels and 92 voice channels could be allocated. In practice, however, each cell would have one control channel, four cells would have 91 voice channels, and three cells would have 92 voice channels.

(c) For N = 12, we can have 8 cells with 2 control channels and 53 voice channels, and 4 cells with 1 control channel and 54 voice channels each. In an actual system, each cell would have 1 control channel, 8 cells would have 53 voice channels, and 4 cells would have 54 voice channels.

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2.2.2. CHANNEL ASSIGNMENT (OR) ALLOCATION TECHNIQUES

1. Briefly discuss the process of channel assignment in cellular networks.
2. Explain channel assignment in detail. [April/May 2018]

✓ A scheme for increasing capacity and minimizing interference is required.
✓ For efficient utilization of the radio spectrum, a frequency reuse scheme is used. So that capacity is increased, interference is reduced.
✓ Channel assignment strategy improves the performance of the system.

- Used to manage calls when handoff is done.
- Minimize connection set-up time
- Adapt to changing load distribution
- Fault tolerance
- Scalability
- Low computation and communication overhead
- Minimize handoffs
- Maximize number of calls that can be accepted concurrently

Call Admission Control

✓ The function of call admission control is to determine whether or not to grant radio resources to a new incoming/handoff call based on information such as the current channel occupation, the bandwidth and
✓ QoS requirements of calls in service, and the characteristics of the call that requests admission

– Call rejection (reject the admission of new call)

– Call dropping (forcing an ongoing call to premature termination)

✓ Channel assignment strategies can be classified
(a) Fixed Channel assignment
(b) Dynamic Channel assignment
(c) Hybrid Channel Allocation schemes (HCA schemes: combining both FCA and DCA techniques)
✓ The choice of the channel assignment strategy impacts the performance of the system, particularly how a call is managed when a mobile user is handoff from one cell to another.

(a) .Fixed Channel assignment

✓ Channels are pre-allocated to the cells during planning phase.
✓ Each cell is allocated a predetermined set of voice channels.
✓ Any call attempt within the cell can only be served by the unused channels in that particular cell.
✓ If all the channels in that cell are occupied, the call is blocked and the subscriber does not receive service.
✓ Due to short term fluctuations in the traffic, FCA schemes are often not able to maintain high quality of service and capacity attainable with static traffic demands.
✓ One approach to address this problem is to borrow free channels from neighboring cells.

(b). Dynamic Channel assignment

✓ No pre-allocation:
  - o In a dynamic channel assignment strategy, voice channels are not allocated to different cells permanently.

✓ Each time a call request is made, the serving base station requests a channel from the MSC.
✓ MSC then allocates a channel to the requested cell using an algorithm that takes into account
  - The likelihood of future blocking within the cell
  - The frequency of use of the candidate channel
  - The reuse distance of the channel

- ✓ To ensure minimum quality of service, the MSC only allocates a given frequency if that frequency is not currently in use in the cell or any other cell which falls within the limiting reuse distance.
- ✓ Dynamic channel assignment reduces the likelihood of blocking increasing the capacity of the system.
- ✓ Dynamic channel assignment strategies require the MSC to collect real-time data on channel occupancy and traffic distribution on a continuous basis.
- ✓ Advantage of dynamic Channel assignment are
  - Increased channel utilization
  - Decreased probability of a blocked call.
- ✓ Disadvantage of dynamic Channel assignment are
  - Increases the storage
  - Increases computational load on the system

(c). Hybrid Channel Allocation (HCA)

- ✓ HCA schemes are the combination of both FCA and DCA techniques.
- ✓ In HCA schemes, the total number of channels available for service is divided into fixed and dynamic sets.
- ✓ The fixed set contains a number of nominal channels that are assigned to cells as in the FCA schemes and, in all cases, are to be preferred for use in their respective cells.
- ✓ The dynamic set is shared by all users in the system to increase flexibility.
- ✓ **Example:** When a call requires service from a cell and all of its nominal channels are busy, a channel from the dynamic set is assigned to the call.
- ✓ Request for a channel from the dynamic set is initiated only when the cell has exhausted using all its channels from the fixed set.
- ✓ Optimal ratio: ratio of number of fixed and dynamic channels.
- ✓ 3:1 (fixed to dynamic), provides better service than fixed scheme for 50% traffic.
- ✓ Beyond 50% fixed scheme perform better.
- ✓ For dynamic, with traffic load of 15% to 32%, better results are found with HCA.

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COMPARISON FCA AND DCA

Compare FCA and DCA.

Attribute	Fixed Channel Allocation	Dynamic Channel Allocation
Traffic load	Fixed Channel Allocation is better under heavy traffic load	Dynamic Channel Allocation is better under light/moderate traffic load
Flexibility of channel allocation	Fixed Channel Allocation is less flexible	Dynamic Channel Allocation is more flexible
Reusability of channels	Fixed Channel Allocation has a maximum possibility.	Dynamic Channel Allocation has a limited possibility.
Temporal and spatial changes	Fixed Channel Allocation are very sensitive	Dynamic Channel Allocation are very insensitive

Grade of service	Fixed Channel Allocation is fluctuating	Dynamic Channel Allocation is stable.
Forced call termination	Large probability in Fixed Channel Allocation	Low/ Moderate probability in Dynamic Channel Allocation
Suitability of cell size	Fixed Channel Allocation uses macro cellular system	Dynamic Channel Allocation uses micro cellular system
Radio Equipment	Fixed Channel Allocation covers only the channels allotted to the cell.	Dynamic Channel Allocation has to cover all possible channel that could be assigned to the cell
Computational effort	In Fixed Channel Allocation, Computational effort is low.	In Dynamic Channel Allocation, Computational effort id high
Call setup delay	Low in Fixed Channel Allocation	Moderate/High in Dynamic Channel Allocation
Implementation complexity	Low in Fixed Channel Allocation	Moderate/High in Dynamic Channel Allocation
Frequency planning	Laborious and complex in Fixed Channel Allocation	None in Dynamic Channel Allocation
Signaling load	Low in Fixed Channel Allocation	Moderate/High in Dynamic Channel Allocation
Control	Centralized in Fixed Channel Allocation	Centralized, decentralized or distributed in Dynamic Channel Allocation

2.2.3 HAND OFF STRATEGIES

1. **Explain the principle of cellular networks and various types of handoff techniques. [16M-May 2016, 16M-May 2013]**
2. **Explain in detail a handoff scenario at cell boundary. [6M-Nov 2014]**
3. **Explain hand off strategies in detail. [April/May 2018]**

- ✓ When a mobile moves into a different cell while a conversation is in progress, the MSC automatically transfers the call to a new channel belonging to the new base station.
- ✓ The handoff operation not only involves a new base station, but also requires that the voice and control signals be allocated to channels associated with the new base station.
- ✓ Handoff calls can be admitted at a higher priority than new calls.
- ✓ To manage the admission of requests based on priority, it is necessary to reserve capacity for admitting handoff requests.
- ✓ A particular signal level is specified as the minimum usable signal for acceptable voice quality at the base station receiver.
- ✓ A sligthy stronger signal level is used as threshold at which a handoff is made.
- ✓ The time over which a call may be maintained within a cell, without handoff, is called the dwell time.

- ✓ Dwell time depends on
  - Propagation
  - Interference
  - Distance between the subscriber
  - Speed
- ✓ Handoff Margin Δ
  - Margin $\Delta = P_{handoff\ threshold} - P_{minimum\ usable\ signal}$ dB
  - Δ is carefully selected
  - Δ too large → unnecessary handoff → MSC loaded down
  - Δ too small → not enough time to transfer → call dropped.

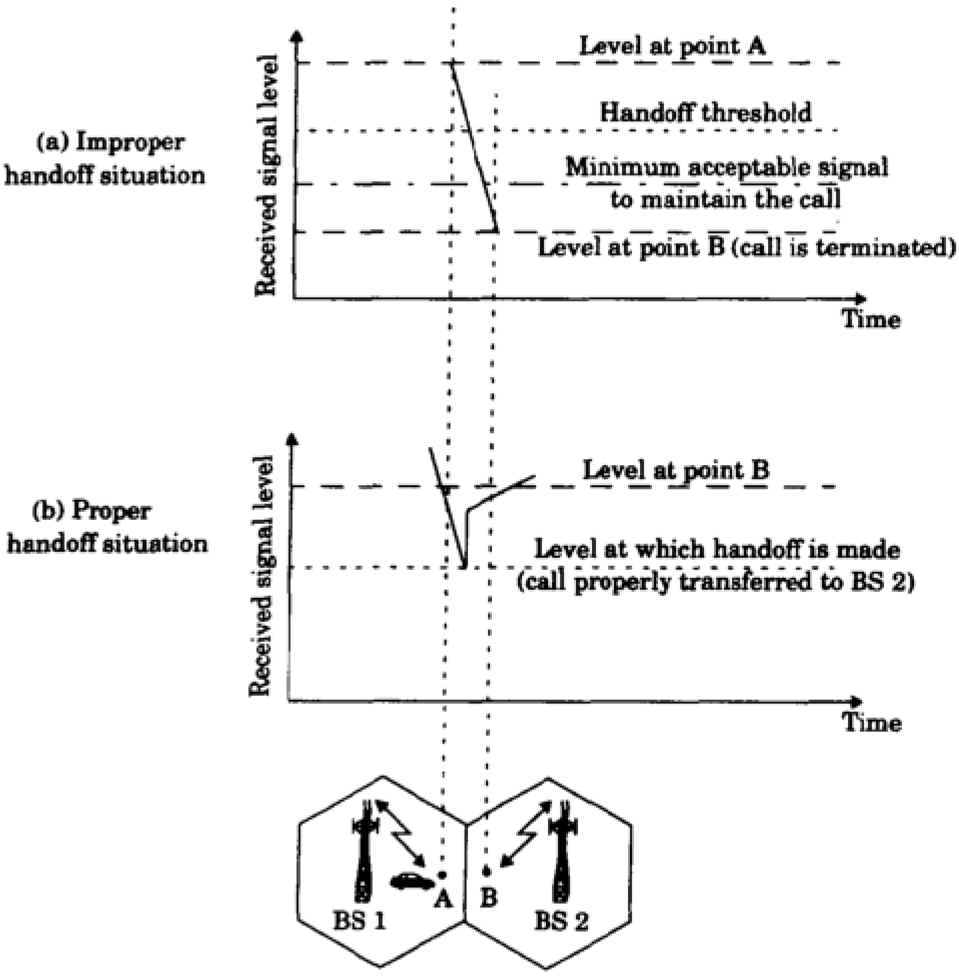

Figure: Illustration of a handoff scenario at cell boundary.

<u>Hand off methods</u>

- ✓ Depending on the information used and the action taken to initiate the handoff, the methods for handoff can be
  - Mobile Controlled Hand off (MCHO)
  - Network Controlled Hand off (NCHO) and
  - Mobile Assisted Hand off (MAHO)
- ✓ **MCHO**
  - MCHO is a desirable method because it reduces the burden on the network.
  - However it increases the complexity of the mobile terminal.
- ✓ **NCHO**
  - In NCHO, the BSs or Access Points (APs) monitor the signal quality from the mobile and report the measurements to the MSC.
  - The MSC is responsible for choosing the candidate AP and initiating the handoff.

  - The mobile plays a passive role in the handoff process.
- ✓ **MAHO**
  - In MAHO, the mobile measures the signal levels from the various APs using periodic beacon generated by the APs.
  - The mobile collects a set of power levels from different APs and feeds it back to the MSC via the serving AP, for handoff decision making.

Prioritizing Handoffs

- ✓ Method for giving priority to handoffs are
  - Guard channel concept
  - Queuing of handoff requests

Guard channel concept

- ✓ Guard channel concept is a fraction of the total available channels in a cell is reserved exclusively for handoff requests from ongoing calls which may be handed off into the cell.
- ✓ Disadvantage of guard channel concept is reducing the total carried traffic as fewer channels are allocated to originating calls.
- ✓ Advantage of guard channel is efficient spectrum utilization during dynamic channel assignment strategies.

Queuing of handoff requests

- ✓ Queuing of handoff requests decreases the probability of forced termination of a call due to lack of available channels.
- ✓ Queuing of handoffs is possible due to the fact that there is a finite time interval between the time the received signal level drops below the handoff threshold and the time the call is terminated due to insufficient signal level.
- ✓ The delay time and size of the queue is determined from the traffic pattern of the particular service area.

Practical Handoff Considerations

- ✓ In practical cellular systems, several problems arise when attempting to design for a wide range of mobile velocities.
- ✓ High speed vehicles pass through the coverage region of a cell within a matter of seconds, whereas pedestrian users may never need a handoff during a call.
- ✓ The MSC can quickly become burdened if high speed users are constantly being passed between very small cells.

Umbrella cell approach

- ✓ To solve the above problem, Umbrella cell approach is used.
- ✓ By using different antenna heights (same building or tower) and different power levels, it is possible to provide “large” and” small” cells which are co-located at a single location.
- ✓ The umbrella cell approach is used to provide large area coverage to high speed users while providing small area coverage to users travelling at low speeds.

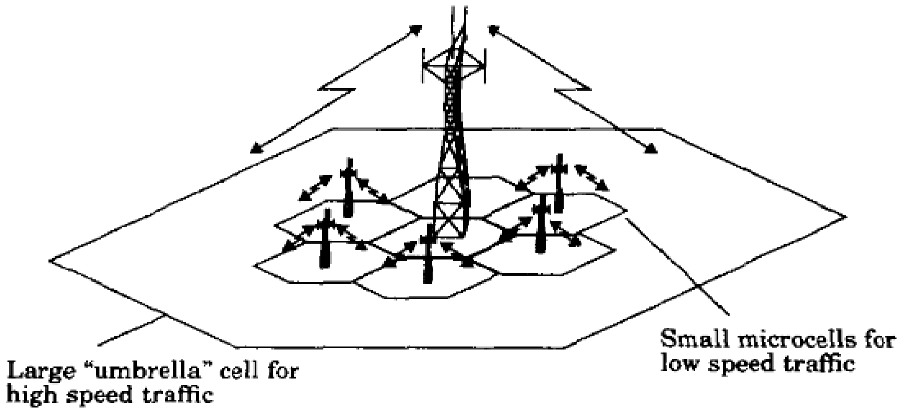

Figure: The umbrella cell approach.

Cell Dragging

- ✓ Another practical handoff problem in microcell systems is known as cell dragging.
- ✓ Cell dragging results from pedestrian users that provide a very strong signal to the base station.
- ✓ Such a situation occurs in an urban environment when there is a line-of-sight (LOS) radio path between the subscriber and the base station.
- ✓ As the user travels away from the base station at a very slow speed, the average signal strength does not decay rapidly.
- ✓ Even when the user has traveled well beyond the designed range of the cell, the received signal at the base station may be above the handoff threshold, thus a handoff may not be made.

Intersystem handoff:

During a call if a mobile moves from one cellular system to a different cellular system controlled by a different MSC, type of handoff is called intersystem handoff.

Types of Handoff

- ✓ **Hard handover** (hard handoff)- If the MSC monitors the strongest signal base station and transfer the call to that base station then it is called hard handoff.
- ✓ **Soft handoff**: Mobile communicates with two or more cells at the same time and find which one is the strongest signal base station then it automatically transfers the call to that base station is called soft handoffs.
- ✓ **Softer hand over:** In this instance a new signal is either added to or deleted from the active set of signals.

1. **Hard handover**

- ✓ The definition of a hard handover or handoff is one where an existing connection must be broken before the new one is established.
- ✓ Intersystem handoff: During a call if a mobile moves from one cellular system to a different cellular system controlled by a different MSC. This type of handoff is called intersystem handoff.
- ✓ The connection must be broken before it can move to the new channel where the connection is re-established.
- ✓ **Intra-frequency hard handovers** where the frequency channel remains the same.
- ✓ Although there is generally a short break in transmission, this is normally short enough not to be noticed by the user.
- ✓ In UMTS- Universal Mobile Telecommunication System most of the handovers that are performed are intra-frequency soft handovers.

2. **Soft handover**

- ✓ The new 3G technologies use CDMA, it is not necessary to break the connection. This is called soft handover.
- ✓ Soft handoff is defined as a handover where a new connection is established before the old one is released.

3. **Softer handover**

- ✓ The third type of hand over is termed a softer handover, or handoff.
- ✓ In this instance a new signal is either added to or deleted from the active set of signals.

- ✓ It may also occur when a signal is replaced by a stronger signal from a different sector under the same base station.
- ✓ This type of handover or handoff is available within UMTS as well as CDMA2000.
- ✓ Cellular handover or cellular handoff is performed by all cellular telecommunications networks, and they are a core element of the whole concept of cellular telecommunications.
- ✓ Soft handover is also less efficient than hard handover, but again more reliable as the connection is never lost.
- ✓ It is therefore necessary for the cellular telecommunications network provider to arrange the network to operate in the most efficient manner, while still providing the most reliable service.

Features of Handoff:

- ✓ Fast and lossless
- ✓ Minimal number of control signal exchanges.
- ✓ Scalable with network size.
- ✓ Capable of recovering from link failures.
- ✓ Efficient use of resources.

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2.2.4 INTERFERENCE AND SYSTEM CAPACITY

1. Describe various interferences and increasing the system capacity of wireless cellular networks

- ✓ Interference is the major limiting factor in the performance of cellular radio.
- ✓ It limits capacity and increases the number of dropped calls.
- ✓ Sources of interference include
  - Another mobile in the same cell
  - Call in progress in a neighboring cell,
  - other base stations operating in the same frequency band,
  - Any non cellular system which leaks energy into the cellular frequency band.
- ✓ Interference is more severe in urban areas due to
  - Greater RF noise floor
  - large number of base stations and mobiles
- ✓ The two major types of interferences:
  - Co-channel interference (CCI)
  - Adjacent channel interference. (ACI)
- ✓ Adjacent channel interference is caused due to the signals that are adjacent in frequency.

Co-channel Interference and System Capacity

- ✓ Co-channel interference is caused due to the cells that reuse the same frequency set.
- ✓ The cells using the same frequency set are called co-channel cells.
- ✓ The interference between signals from the co-channel cells is called co-channel interference.
- ✓ Unlike thermal noise, co-channel interference cannot be overcome by increasing the carrier power of a transmitter
- ✓ This is because an increase in transmitter power increases the interference to neighboring co-channel cells.
- ✓ For similar sized cells, the co-channel interference is independent of the transmitted power and depends on the radius of the cell and the distance to the nearest co-channel cells.

- ✓ To reduce co-channel interference, co-channel cells must be physically separated.

- ✓ Co channel reuse ratio, Q=D/R

  where Q → Co channel reuse ratio
  D → Distance to the nearest co-channel cells
  R → Radius of the cell
- ✓ It determines the spatial separation relative to the coverage distance of the cell.
- ✓ For a hexagonal geometry

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

- ✓ Thus, a small value of Q provides larger capacity but higher co-channel interference.
- ✓ Hence there is a trade-off between capacity and interference.

Calculation of signal-to-interference ratio(S/I or SIR)

- ✓ The signal-to-interference ratio for a mobile is

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

where S → Desired signal power
I_i → Interference power caused by the i^{th} co-channel cell
i_0 → Number of co-channel interfering cells

- ✓ The average received power at a distance d is

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

$$P_r\left(dBm\right)=P_0\left(dBm\right)-10n\log\left(\frac{d}{d_0}\right)$$

where P_0 → Power received at a close-in reference point in the far field region of the antenna
d_0 → Small distance from the transmitting antenna
n → Path loss exponent.

- ✓ If D_i is the distance of the i^{th} interferer, the received power is proportional to $(D_i)^{-i}{}_0$.
- ✓ The path loss exponent, n ranges between 2 and 4.
- ✓ Thus the S/I for a mobile can be written as

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

- ✓ For only the first layer of equidistant interferers

$$\frac{S}{I}=\frac{\left(D/R\right)^n}{i_0}=\frac{\left(\sqrt{3N}\right)^n}{i_0} \qquad \because Q=\frac{D}{R}=\sqrt{3N}$$

- ✓ S/I is usually the worst case when a mobile is at the cell edge

$$\frac{S}{I}=\frac{R^{-4}}{2\left(D-R\right)^{-4}+2\left(D+R\right)^{-4}+2D^{-4}}$$

$$\frac{S}{I}=\frac{1}{2\left(Q-1\right)^{-4}+2\left(Q+1\right)^{-4}+2Q^{-4}}$$

- ✓ For a hexagonal cluster of cells

$$\frac{S}{I} = \frac{1}{6}\left(\frac{D}{R}\right)^{i_0} = \frac{1}{6}\left(\sqrt{3N}\right)^{i_0}$$

- ✓ Hence, S/I is independent of the cell radius.

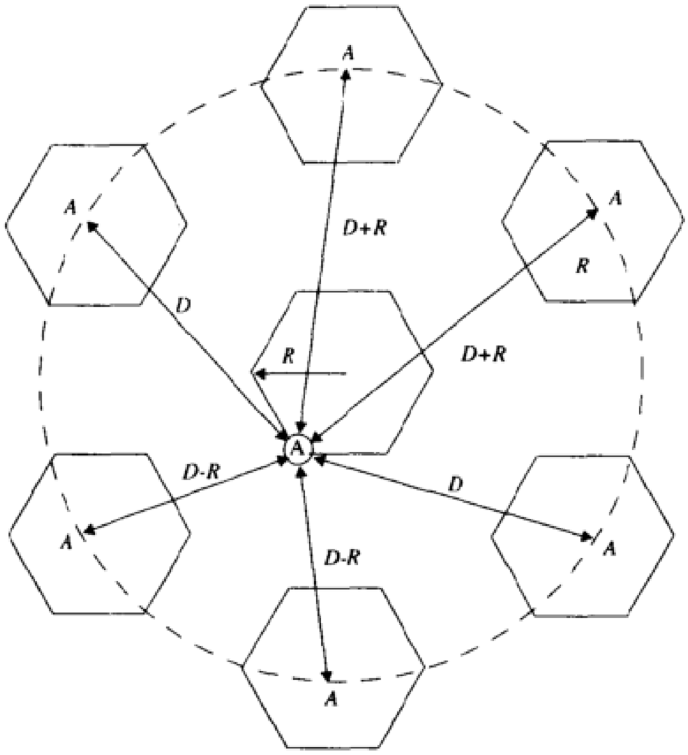


Figure: Illustration of the first tier of co-channel cells for a cluster size of N=7. When the mobile is at the cell boundary (point A), it experiences worst case co-channel interference on the forward channel. The marked distances between the mobile and different co-channel cells are based on approximations made for easy analysis.

Adjacent Channel Interference

- ✓ Interference resulting from signals which are adjacent in frequency to the desired signal is called adjacent channel interference.
- ✓ Adjacent channel interference results from imperfect receiver filters that allow nearby frequencies to leak into the passband.
- ✓ The problem can be severe if an adjacent channel user is transmitting in very close range to a subscriber's receiver.
- ✓ The near-far effect occurs when a mobile close to a base station radiates in the adjacent channel, while the subscriber is far away from the base station.
- ✓ Adjacent channel interference can be reduced by
  - Careful filtering
  - Careful channel assignments.
- ✓ The frequency separation between each channel in a cell should be made as large as possible.
- ✓ If the subscriber is at a distance d_1 and the interferer is at d_2, then signal-to-interference ratio is

$$\frac{S}{I} = \left(\frac{d_1}{d_2}\right)^{-n}$$

- ✓ The frequency separation between each channel in a cell should be made as large as possible while assigning them.

Power Control to Reduce Interference

- ✓ In practical systems, the power levels of every subscriber are under constant control by the serving base stations.

- ✓ Power control
  - Reduces interference levels
  - Prolongs battery life
- ✓ In CDMA spread spectrum systems, power control is a key feature to ensure maximal utilization of the system capacity.
- ✓ Reduced interference leads to higher capacity.

****

2.2.5. TRUNKING AND GRADE OF SERVICE

1.Describe the various terms involved in trunking and grade of service
2. Write short notes on i) Trunking ii) Grade of service of cell system. [Nov/Dec 2017]

- ✓ Cellular radio systems rely on trunking to accommodate a large number of users in a limited radio spectrum.
- ✓ In a trunked radio system, each user is allocated a channel on a per call basis.
- ✓ Upon termination of the call, the previously occupied channel is immediately returned to the pool of available channels.
- ✓ The time required to allocate a trunked radio channel to a requesting user is called Set-up Time.
- ✓ Call which cannot be completed at time of request due to congestion is called Blocked Call or lost call.
- ✓ Average duration of a typical call is called Holding Time, H.
- ✓ Request Rate is the average number of call requests per unit time. It is denoted by λ $seconds^{-1}$.

- ✓ Traffic Intensity is the measure of channel time utilization, which is the average channel occupancy measured in Erlangs.
- ✓ Load is the Traffic intensity across the entire trunked radio system, measured in Erlangs.
- ✓ A channel kept busy for one hour is defined as having a load of one Erlang.

- ✓ Grade of Service (GOS) is measure of congestion which is specified as the probability
  - Probability of a call being blocked (Erlang B)
  - Probability of a call being delayed beyond a certain amount of time (Erlang C)
- ✓ The grade of service (GOS) is a measure of the ability of a user to access a trunked system during the busiest hour.
- ✓ The grade of service is used to define the desired performance of a particular trunked system by specifying a desired likelihood of a user obtaining channel access given a specific number of channels available in the system.
- ✓ In order to obtain proper GOS, it is the necessary to estimate
  - Maximum required capacity
  - To allocate the proper number of channels.
- ✓ Each user generates a traffic intensity of A_u Erlangs given by

$$A_u = \lambda H$$

where A_u → Traffic intensity
λ → Average number of call requests per unit time
H → Average duration of a call

- ✓ Total offered traffic intensity A, is given as

$$A = UA_u$$

where U → Number of users in the system.
A → Total offered traffic.

- ✓ Traffic intensity per channel is given as

$$A_c = UA_u / C$$

where C → Number of trunked channels offered by a trunked radio system

- ✓ The Erlang B formula is given by

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

- ✓ The likelihood of a call not having immediate access to a channel is determined by the Erlang C formula

$$P_r[delay > 0] = \frac{A^c}{A^c + c!\left(1 - \frac{A}{C}\right)\sum_{k=0}^{C-1} \frac{A^k}{k!}}$$

****

2.2.6 IMPROVING CAPACITY IN CELLULAR SYSTEMS

1. **Explain some techniques intended to improve the coverage area and capacity of cellular system. (8M-Nov 2015) (or)**
2. **Explain in detail how to improve coverage and channel capacity in cellular systems. (16M-May 2016) (or)**
3. **Define the methods of increasing the capacity of wireless cellular networks. (10M-May 2013)**
4. **Explain the capacity improvement techniques used in cellular system. (10M – May 2010]**

- ✓ As demand for service increases, system designers have to provide more channels per unit coverage area.
- ✓ Common Techniques used to expand the capacity of cellular systems are
  - Cell splitting
  - Sectoring
  - Microcell Zoning
- ✓ Cell splitting increases the number of base station deployed and allows an orderly growth of the cellular system.
- ✓ Sectoring uses directional antennas to further control the interference and frequency reuse of channels.
- ✓ Microcell Zoning distributes the coverage of a cell and extends the cell boundary to hard-to-reach places.

Cell Splitting

- ✓ Cell splitting is the process of subdividing a congested cell into smaller cells with
  - its own base station
  - Corresponding reduction in antenna height
  - Corresponding reduction in transmitter power.
- ✓ Splitting of cells reduces the cell size and thus more number of cells has to be used.
- ✓ More number of cells => More number of clusters =>More Channels => Higher capacity
- ✓ By defining new cells which have a smaller radius than the original cells and by installing these smaller cells (called microcells) between the existing cells, capacity increases due to the additional number of channels per unit area.
- ✓ Cell splitting allows a system to grow by replacing large cells by small cells, without upsetting the channel allocation.
- ✓ Cells are split to add channels with no new spectrum usage

- ✓ Depending on traffic patters the smaller cells may be activated /deactivated in order to efficiently use cell resources.
- ✓ In the figure that the original base station A has been surrounded by six new microcell base stations.
- ✓ The smaller cells were added in such a way as to preserve the frequency reuse plan of the system.
- ✓ Cell splitting scales the geometry of the cluster.

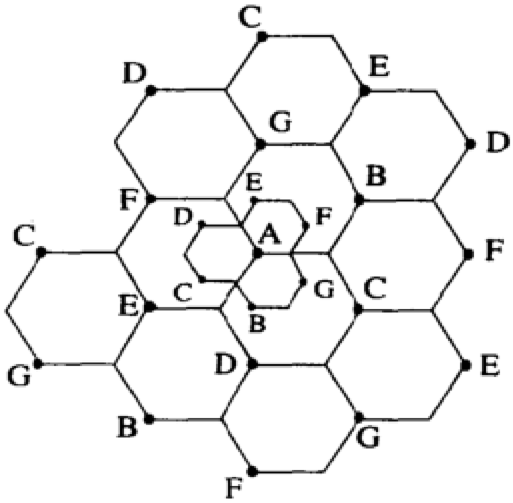


Figure: Illustration of cell splitting

- ✓ When new cell radius is half the original cell radius,

$$P_r \text{ [at old cell boundary] } \propto P_{t1} R^{-n}$$

$$P_r \text{ [at new cell boundary] } \propto P_{t2} (R/2)^{-n}$$

Where P_r → Received power

P_{t1} → Transmit power of larger cell base station

P_{t2} → Transmit power of smaller cell base station

N → Path loss exponent

- ✓ Transmit power must be reduced by 12db in order to fill in the original coverage area with microcell while maintaining the S/I requirement.

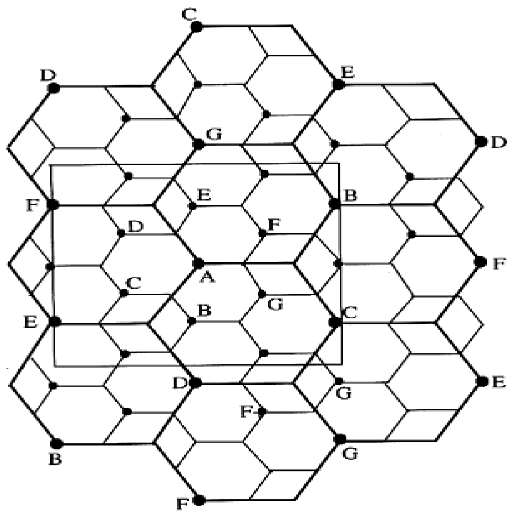
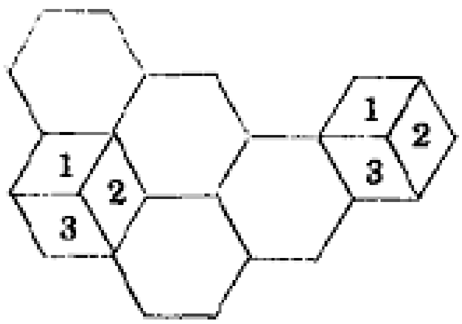


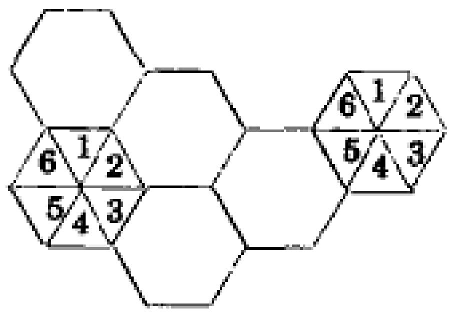
Figure: Illustration of cell splitting within
a 3 km square centered base station A.

Sectoring

- ✓ The technique for decreasing co-channel interference and thus increasing system capacity by using directional antennas is called sectoring.
- ✓ The factor by which the co-channel interference is reduced depends on the amount of sectoring used.
- ✓ Cell Sectoring keeps R untouched and reduces D/R.
- ✓ Capacity improvement is achieved by reducing the number of cells per cluster, thus increasing frequency reuse.
- ✓ It is necessary to reduce the relative interference without decreasing the transmitter power.
- ✓ The co-channel interference may be decreased by replacing the single omni-directional antenna by several directional antenna, each radiating within a specified sector.
- ✓ A directional antenna transmits to and receives from only a fraction of the total number of co-channel cells. Thus co-channel interference is reduced.
- ✓ A cell is normally partitioned into three 120^0 sectors or six 60° sectors.



(a) 120^0 sectoring.



(b) 60^0 sectoring.

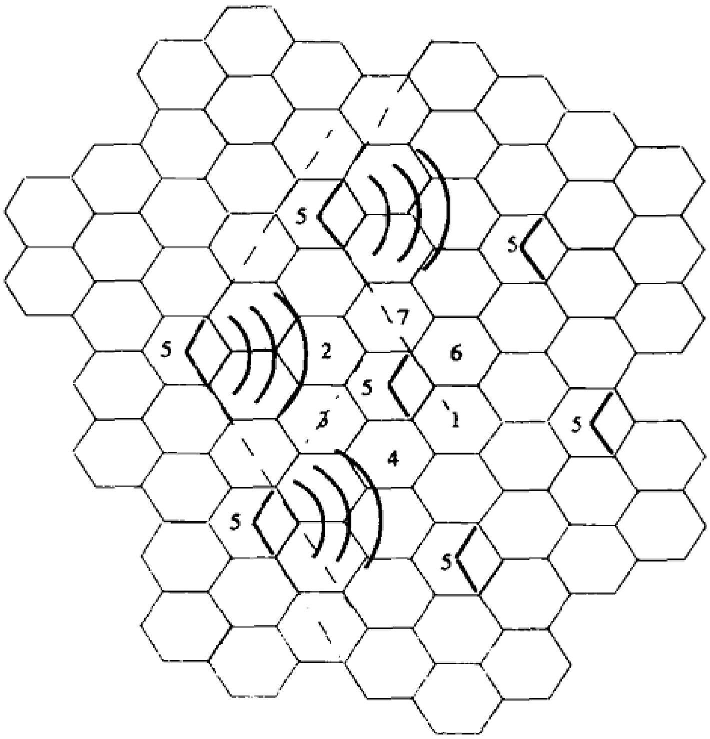


Figure: Illustration of how 120° sectoring reduces interference from co-channel cells. Out of the 6 co-channel cells in the first tier, only 2 of them, interfere with the center cell. If omni-directional antennas were used at each base station, all 6 co-channel cells would interfere with the center cell.

Advantages

- ✓ it improves Signal-to-interference ratio.

Disadvantages

- ✓ Disadvantages of cell sectoring includes

- Increased number of antennas at each base station.
- Decrease in trunking efficiency
- Increased number of handoffs.

Repeaters for Range Extension

✓ Repeaters are useful for hard to reach areas
  - Within buildings and basement
  - Tunnels
  - Valleys

✓ Radio transmitters called repeaters are used to provide coverage in these areas.

✓ Repeaters are bidirectional.
  - Receive signals from the base station
  - Amplify the signals
  - Reradiates the signals.

✓ Received noise and interference is also reradiated.

A Novel Microcell Zone Concept

✓ Zone Concept
  - A cell is divided into microcell or zones.
  - Each microcell (Zone) is connected to the same base station by coaxial cable, fiber optic cable, or microwave link.
  - Each Zone uses a directional antenna
  - As mobile travels from one zone to another, it retains the same channel. i.e. Without handoff.
  - The base station simply switches the channel to the next zone site.
  - Mobile is served by the zone with the strongest signal.

✓ While the cell maintains a particular coverage area, the co-channel interference is reduced because:
  - The large central base station is replaced by several low power transmitters.
  - Directional Antennas are used.

✓ Decreased co-channel interference improves
  - Signal Quality
  - Capacity

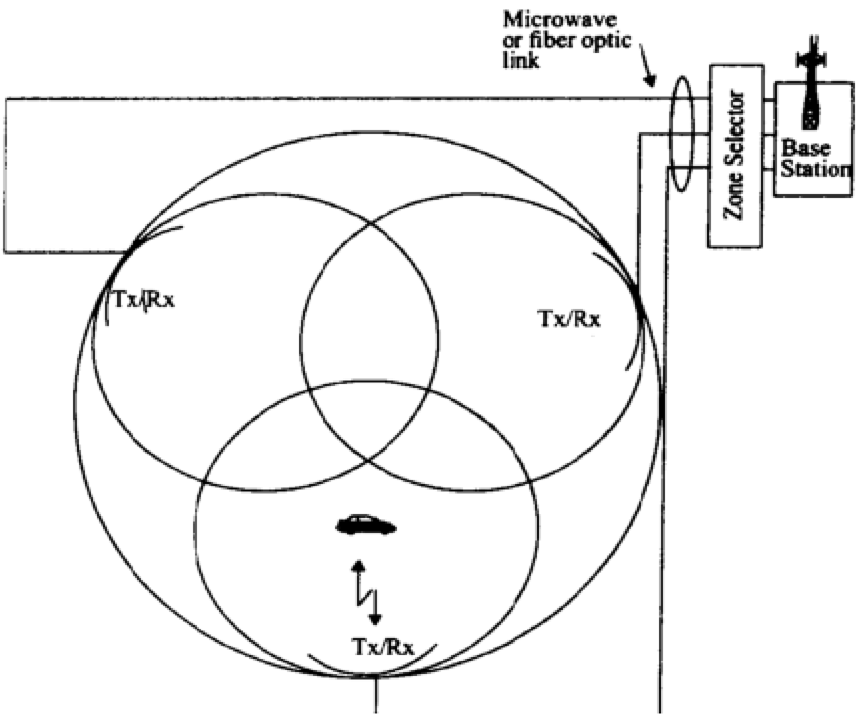


Figure: The microcell concept

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PROBLEMS:

1. If a total of 33 MHz of bandwidth is allocated to a particular FDD cellular telephone 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 uses (a) 4-cell reuse, (b) 7-cell reuse (c) 12-cell reuse. If 1 MHz of the allocated spectrum is dedicated to control channels, determine an equitable distribution of control channels and voice channels in each cell for each of the three systems.

(**Apr/may 2010**, **Apr/may 2017)**

Solution:

Given:

Total bandwidth =33 MHz

Channel bandwidth = 25 k}{z x 2 simplex channels = 50 kHz/duplex channel

Total available channels = 33,000/50 = 660 channels

(a) For N= 4,

Total number of channels available per cell = 660/4 = 165 channels.

(b) ForN=7,

Total number of channels available per cell = 660/7= 95 channels.

(c) For N = 12,

Total number of channels available per cell = 660/12 = 55 channels.

A 1 MHz spectrum for control channels implies that there are 1000/50 = 20 control channels out of the 660 channels available. It evenly distribute the control and voice channels, simply allocate the same number of channels in each cell wherever possible. Here, the 660 channels must be evenly distributed to each cell within the cluster. In practice, only the 640 voice channels would be allocated, since the control channels are allocated separately as 1 per cell.

(a) For N = 4, we can have 5 control channels and 160 voice channels per cell.

In practice, however, each cell only needs a single control channel (the control channels have a greater reuse distance than the voice channels).

Thus, one control channel and 160 voice channels would be assigned to each cell.

(b) For N = 7, 4 cells with 3 control channels and 92 voice channels, 2 cells with 3 control channels and 90 voice channels, and 1 cell with 2 control channels and 92 voice channels could be allocated.

In practice, however, each cell would have one control channel, four cells-4vould have 91 voice channels, and three cells would have 92 voice channels.

(c) For N = 12, we can have 8 cells with 2 control channels and 53 voice channels, and 4 cells with 1 control channel and 54 voice channels each.

In an actual system, each cell would have 1 control channel, 8 cells would have 53 voice channels, and 4 cells would have 54 voice channels.

2.If a signal to interference ratio of 15 dB is required for satisfactory forward channel performance of a cellular system, what is the frequency reuse factor and cluster size that should be used for maximum capacity if the path loss exponent is (a) n = 4 , (b) nt = 3? Assume that there are 6 co-channels cells in the first tier, and all of them are at the same distance from the mobile. Use suitable approximations.

Solution:

(a) n = 4

First, let us consider a 7-cell reuse pattern.co-channel reuse ratio D/R = 4.583.

The sigal-to-noise interference ratio is given by S/I = (I/6)x(4.583) = 75.3 = 18.66 dB.

Since this is greater than the minimum required S/I, N = 7 can be used.

b) n=3

First, let us consider a 7-cell reuse pattern.

S/I = (l,6)x(4.583) = 16.04 = 12.05 dB.

Since this is less than the minimum required S/I, we need to use a larger N.

The next possible value of N is 12, (i = j = 2).

The corresponding co-channel ratio is given as D/R = 6.0.

The signal-to-interference ratio is given by

S/I = (1/6) x = 36 = 15.56 dB.

Since this is greater than the minimum required S/I, N = 12 can be used

3. A certain city has an area of 1,300 square miles and is covered by a cellular system using a 7-cell reuse pattern. Each cell has a radius of 4 miles and the city is allocated 40 MHz of spectrum with a full duplex channe] bandwidth of 60 kHz. Assume a GOS of 2% for an Erlang B system is specified. If the offered traffic per user is 0.03 Erlangs, compute (a) the number of cells in the service area, (b) the number of channels per cell, (c) traffic intensity of each cell, (d) the maximum carried traffic; (e) the total number of users that can be served for 2% GOS, (f') the number of mobiles per channel, and (g) the theoretical maximum number of users that could be served at one time by the system.

Solution:

(a) Given:

Total coverage area = 1300 miles

Cell radius = 4 miles

The area of a cell (hexagon) can be shown to be 2.598 / R2, thus each cell covers 2.5981 x (4)2 = 41.57 sqm..

Hence, the total number of cells are = 1300/41.57 = 31 cells.

(b) The total number of channels per cell (C)

= allocated spectrum I (channel width x frequency reuse factor)

= 40, 000,000/ (60,000 x 7) = 95 channels/cell

(c) Given: C = 95, and GOS = 0.02

From the Erlang B chart, we have traffic intensity per cell A = 84 Erlangs/cell

(d) Maximum carried traffic = number of cells x traffic intensity per cell

= 31 x 84 = 2604 Erlangs.

(e) Given traffic per user = 0.03 Erlangs

Total number of users = Total traffic / traffic per user

= 2604 / 0.02 = 86,800 users.

(f) Number of mobiles per channel = number of users/number of channels

= 86,800 / 666 = 130 mobiles/ehannel.

(g) The theoretical maximum number of served mobiles is the number of availablechannels in the system (all channels occupied)

= Cx Nc = 95 x 31 = 2945 users, which is 3.4% of the customer base.

4.A hexagonal cell within a 4-cell system has a radius of 1.387 km. A total of 60 channels are used within the entire system. If the load per user is 0.029 Erlangs, and λ = 1 calls/hour, compute the following for an Erlang C system that has a 5% probability of a delayed call:

(a) How many users per square kilometer will this system support?

(b) What is the probability that a delayed call will have to wait for more than 10 sec?

(c) What is the probability that a call will be delayed for more than 10 seconds?

Solution :

Given, Cell radius, R = 1.387 km

Area covered per cell is 2.598 x (1.387)2 = 5 sq kin

Number of cells per cluster = 4

Total number of channels = 60

Therefore, number of channels per cell = 60 / 4 = 15 channels.

(a) From Erlang C chart, for 5% probability of delay with C = 15, traffic intensity = 9.0 Erlangs.

Therefore, number of users = total traffic intensity/ traffic per user =9.0/0.029 = 310 users

= 310 users/S sq km = 62 users/sq km

(b) Given λ=1 = 1 , holding time

H = Au/λ = 0.029 hour = 104.4 seconds.

The probability that a delayed call will have to wait for more than 10 s is

Pr[delay > t ׀ dleay ׀= exp(.—(C.-A)t / H) = exp(—(I5—9.0)I0/l04.4) = 56.29 %

(c) Given Pr delay >0] = 5% = 0.05

Probability that a call is delayed more than 10 seconds

Pr{delay> 10] = = 0.05 x 0.5629 = 2.81 %

5. A digital mobile communication system has a forward channel frequency band ranging between 810 MHz to 826 MHz and a reverse channel band between 940 MHz to 956 MHz. Assume that 90 per cent of the band width is used by traffic channels. It is required to support at least 1150 simultaneous calls using FDMA The modulation scheme employed has a spectral efficiency of 1.68 bps / Hz. Assuming that the channel

impairments necessitate the use of rate ½ FEC codes, find the upper bound on the transmission bit rate that a speech coder used in this system should provide?

Solution :

Tbtal Bandwidth available for traffic channels = 0.9 x (810—826) = 14.4 MHz.

Number of simultaneous users = 1150.

Therefore, maximum channel bandwidth = 14.4 /1150 MHz = 12.5 kHz.

Spectral Efficiency = 1.68 bps/Hz.

Therefore, maximum channel data rate = 1.68 x 12500 bps = 21kbps.

FEC coder rate = 0.5.

Therefore, maximum net data rate = 21 x 0.5 kbps = 10.5 kbps.

Therefore, we need to design a speech coder with a data rate less than or equal to 10.5 kbps.

6. Apr/may 2017

Consider the design of the U.S. digital cellular equalizer [Pro91]. If f = 900 MHz and the mobile velocity υ = 80 km/hr, determine the following:

(a) the maximum Doppler shift

(b) the coherence time of the channel

(c) the maximum number of symbols that could be transmitted without updating the equalizer, assuming that the symbol rate is 24.3 ksymbols/sec

Solution to Example 6.3

(a) From equation (4.2), the maximum Doppler shift is given by

$$f_d = \frac{\upsilon}{\lambda} = \frac{(80,000/3600)\,\mathrm{m/s}}{(1/3)\,\mathrm{m}} = 66.67\ \mathrm{Hz}$$

(b) From equation (4.40.c), the coherence time is approximately

$$T_c = \sqrt{\frac{9}{16\pi f_d^2}} = \frac{0.423}{66.67} = 6.34\ \mathrm{msec}$$

Note that if (4.40.a) or (4.40.b) were used, T_C would increase or decrease by a factor of 2 – 3.

(c) To ensure coherence over a TDMA time slot, data must be sent during a 6.34 ms interval. For R_s=24.3 ksymbols/sec, the number of bits that can be sent is

$$N_b = R_s T_C = 24,300 \times 0.00634 = 154\ \text{symbols}$$

As shown in Chapter 10, each time slot in the U.S. digital cellular standard has a 6.67 ms duration and 162 symbols per time slot, which are very close to values in this example

7. The output of a speech coder has bits which contribute to signal quality with varying degree of importance. Encoding is done on blocks of samples of 20 ms duration (260 bits of coder output). The first 50 of the encoded speech bits (say type 1) in each block are considered to be the most significant and hence to protect them from channel errors are appended with 10 CRC bits and convolutionally encoded with a rate 1/ 2 FEC coder. The next 132 bits (say type 2) are appended with 5 CRC bits and the tast 78 bits (say type 3) are not error protected. Compute the gross channel data rate achievable.

Solution :

Number of type 1 channel bits to be transmitted every 20 ms

(5+l0x2) = 120 bits

Number of type 2 channel bits to be transmitted every 20 ms

132 +5 = 137 bits

Number of type 3 channel bits to be encoded = 78 bits

Total number of channel bits to be transmitted every 20 ms

120 + 137 +78 bits = 335 bits

Therefore, gross channel bit rate = 335/ (20 x I0-3) = 16.75 kbps.

8. Nov/Dec 2010

Example 8.2

If B_t is 12.5 MHz, B_{guard} is 10 kHz, and B_c is 30 kHz, find the number of channels available in an FDMA system.

Solution to Example 8.2

The number of channels available in the FDMA system is given as

$$N = \frac{12.5 \times 10^6 - 2(10 \times 10^3)}{30 \times 10^3} = 416$$

In the U.S., each cellular carrier is allocated 416 channels.

9. If GSM uses a frame structure where each frame consists of 8 time slots, and each time slot contains 156.25 bits, and data is transmitted at 270.833 kbps in the channel, find (a) the time duration of a bit, (b) the time duration of a slot, (c) the time duration of a frame, and (d) how long must a user occupying a single time slot must wait between two simultaneous transmissions

Solution

(a) The time duration of a bit, $T_b = \frac{1}{270.833 \text{ kbps}} = 3.692\ \mu s.$

(b) The time duration of a slot, $T_{slot} = 156.25 \times T_b = 0.577$ ms.

(c) The time duration of a frame, $T_f = 8 \times T_{slot} = 4.615$ ms.

(d) A user has to wait 4.615 ms, the arrival time of a new frame, for its next transmission.

Example 8.5

If a normal GSM time slot consists of 6 trailing bits, 8.25 guard bits, 26 training bits, and 2 traffic bursts of 58 bits of data, find the frame efficiency.

Solution to Example 8.5

A time slot has $6 + 8.25 + 26 + 2(58) = 156.25$ bits.
A frame has $8 \times 156.25 = 1250$ bits/frame.
The number of overhead bits per frame is given by

$$b_{OH} = 8(6) + 8(8.25) + 8(26) = 322 \text{ bits}$$

Thus, the frame efficiency

$$\eta_f = \left[1 - \frac{322}{1250}\right] \times 100 = 74.24\ \%$$

11**. In a cellular system with total of 917 radio channels available for handling traffic. The area of a cell is 4 km2 and the total area is 1400 km2 with cluster of 7.**

(a) Calculate the system capacity.

(b) How many times signal can be replicated?

(c) Calculate the system capacity for N = 4.

(d) Compare the performance.

Solution:

(a) System capacity C = MKN

M = Number of times the cluster has to be replicated

$$M = \frac{A_{sys}}{A_{cluster}} = ?$$

$$A_{cluster} = N \times A_{singlecell} = 7 \times 4 = 28 \text{ km}^2$$

$$M = \frac{1400}{28} = 50$$

$$K = \text{Number of channels per cell}$$

$$K = \frac{S}{N} = \frac{917}{7} = 131 \text{ channels/cell}$$

$$\text{System capacity } C = MKN = 50 \times 131 \times 7$$

$$\boxed{C = 45850 \text{ channels}}$$

$$(b) \qquad M = \frac{A_{sys}}{A_{cluster}} = \frac{1400}{28} = 50$$

$$(c)\quad \text{For } N = 4, \quad A_{cluster} = N \times A_{cell} = 4 \times 4 = 16 \text{ km}^2$$

$$M = \frac{A_{sys}}{A_{cluster}} = \frac{1400}{16} = 87.5 \approx 87$$

$$C = MKN$$

$$K = \frac{S}{N} = \frac{917}{4} = 229.25 \approx 229 \text{ channels/cell}$$

$$C = MKN$$

$$C = 87 \times 229 \times 4 = 79692 \text{ channels}$$

$$(d)\quad \text{For } N = 7, \quad C = 45850 \text{ channels}$$

$$\text{For } N = 4, \quad C = 79692 \text{ channels}$$

$\therefore$ The cluster size decreases, the capacity of the system increases.

---

11. For a cellular system with a total bandwidth of 15 MHz uses 10 KHz simplex channels to provide full duplex voice and control channels. For 12 cell reuse pattern and 1 MHz of the total bandwidth is allocated for control channels.

(a) Calculate the total available channels.

(b) Determine the number of control channels.

(c) Calculate the number of voice channels per cell.

$$(a)\quad \text{Total channels} = \frac{\text{Total bandwidth}}{\text{Channel bandwidth}}$$

$$\text{Channel bandwidth for duplex} = 10 \text{ KHz} \times 2$$

$$= 20 \text{ KHz/duplex channel}$$

$$\text{Total channels} = \frac{15{,}000 \text{ KHz}}{20 \text{ KHz}} = 750$$

$$(b)\quad \text{Number of control channels} = \frac{\text{Bandwidth of control channel}}{\text{Channel bandwidth}}$$

$$= \frac{1000 \text{ KHz}}{20 \text{ KHz}} = 50$$

$$(c)\qquad \text{The number of voice channels per cell} = \frac{\begin{matrix}\text{Total}\\ \text{channel}\end{matrix} - \begin{matrix}\text{Number of}\\ \text{control channel}\end{matrix}}{\text{Cluster size}}$$

$$= \frac{750 - 50}{12} = \frac{700}{12} = 58.3 \approx 58$$

12. In the FDMA system, the total spectrum bandwidth is 12.5 MHz, each channel is 30 KHz. The edge guard spacing is 10 KHz. Find the total number of channels available in the system.

$$\text{Total number of available channels} = \frac{\begin{Bmatrix}\text{Spectrum}\\ \text{bandwidth}\end{Bmatrix} - 2 \times \begin{Bmatrix}\text{Guard}\\ \text{spacing}\end{Bmatrix}}{\text{Channel bandwidth}} \quad \text{(in KHz)}$$

$$= \frac{12500 - 2 \times 10}{30}$$

$$= \frac{12480}{30}$$

$$\boxed{\text{Total channels} = 416 \text{ channels}}$$

13. Consider Global System for Mobile, which is a TDMA/FDD system that uses 25 MHz for the forward link, which is broken into radio channels of 200 kHz. If speech channels are supported on a single radio channel, and if no guard band is assumed, find the number of simultaneous users that can be accommodated in GSM.

Datas: 25 MHz for the forward link radio channels of 200 kHz

$$N = \frac{25\text{ MHz}}{(200\text{ kHz})/8} = 1000$$

Thus, GSM can accommodate 1000 simultaneous users.

TWO MARKS

MULTIPLE ACCESS TECHNIQUE

1. What is multiple access technique? [May 2016, Nov 2013]

Multiple access or channel access method is based on a multiplexing method that allows several data streams or signals to share the same communication channel or physical medium.

2. Write the applications of multiple access methods.

- The multiple access methods are used in
  - Satellite networks
  - Cellular and mobile communication networks
  - Military communication and
  - Underwater acoustic networks.

3. What are the different types of multiple access schemes? [May 2012]

The different types of multiple access schemes are

- Frequency Division Multiple Access (FDMA)
- Time Division Multiple Access (TDMA)
- Code Division Multiple Access (CDMA)

4. State the difference between Narrowband and wideband systems. [Nov 2013, Nov 2012]

NARROWBAND SYSTEMS	WIDEBAND SYSTEMS
In a narrowband system, the available radio spectrum is divided into a large number of narrowband channels.	In wideband system, a large number of transmitters are allowed to transmit on the same channels.

5. Define FDMA.

In FDMA, the total bandwidth is divided into non-overlapping frequency sub bands. Each user is allocated a unique frequency sub band (channels) for the duration of the connection, whether the connection is in an active or idle state.

6. What is the need of guard bands in FDMA?

The adjacent frequency bands in the FDMA spectrum are likely to interference with each other. Therefore it is necessary to include the guard bands between the adjacent frequency bands.

7. Mention some features of FDMA.

- ✓ FDMA is relatively simple to implement.
- ✓ To provide interference-free transmissions between the uplink and the downlink channels, the frequency allocations have to be separated by a sufficient amount (guard bands).

8. Write the nonlinear effects in FDMA.

- ✓ In FDMA system, many channels share same antenna at the base station. The power amplifiers and the power combiners used are nonlinear, and tend to generate inter modulation frequencies resulting in inter modulation distortion.

9. Write the expression for number of channels used in FDMA system.

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

$$N_S = \frac{B_s - 2B_g}{B_c}$$

Where, B_s-Total spectrum allocation (or) system bandwidth

B_g-Guard band allocated at the edge of the allocated spectrum band and

B_g-Channel bandwidth

10. Write the formula for spectral efficiency of FDMA.

- ✓ The spectral efficiency of FDMA is given by

$$\eta_{FDMA} = \frac{bandwidth\ available\ for\ data\ transmission}{system\ bandwidth}$$

$$\eta_{FDMA} = \frac{N_{data}B_c}{B_s} < 1$$

Where N_{data}=Number of data channels in the system.

$N_{data} = N_S - N_{ctl}$

N_{ctl} = Number of allocated control channels

11. Mention the disadvantages of FDMA.

- ✓ This type of multiple access support is narrow band, and is not suitable for multimedia communications with various transmission rates.
- ✓ If a FDMA channel is not in use, then it is idle and cannot be used by other users to increase or share capacity. It is essentially a wasted resource.
- ✓ FDMA is an old and is used for the analog signal.

12. Define TDMA.

- ✓ Time Division Multiple Access (TDMA) systems divide the ratio spectrum into time slots, and in each slot only one user is allowed to either transmit or receive.

13. What is W- TDMA?

- ✓ In wideband TDMA, transmission in each slot uses the entire frequency band.

14. Define N- TDMA.

- ✓ In narrow band TDMA, the whole frequency band is divided into sub band, transmission in each slot only uses the frequency width of one sub band.

15. Write the features of TDMA.

- ✓ TDMA shares a single carrier frequency with several users, where each user makes use of non-overlapping time slots.
- ✓ Data transmission for users of a TDMA system is not continuous, but occurs in bursts. This results in low battery consumption, since the subscriber transmitter can be turned off when not in use.
- ✓ Because of discontinues transmissions in TDMA, the handoff process is much simpler for a subscriber unit, since it is able to listen for other base stations during idle time slots.

16. What is frame efficiency in TDMA?

- ✓ The frame efficiency is the ***percentage*** of bits per frame which contain transmitted data.

  The frame efficiency is given by

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

$$b_{OH} = \text{Number of overhead bits per frame and}$$

$$b_T = \text{Number of total bits per frame}$$

17. How does near/far problem influence TDMA systems? [Nov 2015]

The near-far problem is one of detecting or filtering out a weaker signal amongst stronger signals. The near-far problem is particularly difficult in CDMA systems where transmitters share transmission frequencies and transmission time. In contrast, FDMA and TDMA systems are less vulnerable

18. State advantages of CDMA over FDMA. [Nov 2014]

CDMA sends digital signals spread out over a larger bandwidth constantly with each signal having a unique sequence code so that each call can be separated at the receiver. In theory, CDMA can carry 8-10 times the number of calls as FDMA, although probably not nearly that many times in the real world.

19. Define near-far problem in CDMA.

- ✓ Some of the mobile units are close to the base station while others are far from it. A strong signal received at the base from a near –in mobile unit and the weak signal from a far –end mobile unit. This phenomenon is called the near-far problem.

20. Write some features of CDMA.

- ✓ Many user of CDMA system share the same frequency.
- ✓ Channel data rates are very high in CDMA system.
- ✓ CDMA has more flexibility than TDMA in supporting multimedia service.

CELLULAR CONCEPT

21. Write the cellular concept. (or)
Why is cellular concept used for mobile telephony? [May 2017]

If a given set of frequencies or radio channels can be reused without increasing the interference, then the large geographical area covered by a single high power transmitter can be divided into a number of small areas, each allocated power transmitters with lower antennas can be used.

22. Why hexagon shape was selected for cell?

The Hexagon shape was chosen for cell because it provides the most effective transmission by approximating a circular pattern while eliminating gaps present between adjacent circles.

23. Differentiate between macro cells and microcells.

The physical size of a cell varies, depending on user density and calling patterns.

- ✓ Macro cells are large cells typically have a radius between 1 mile and 15 miles with base station transmit powers between 1W and 6W.
- ✓ Microcells are the smallest cells typically have a radius between of 1500 feet or less with base station transmit powers between 0.1W and 1W.

24. Mention the need of Pico cells.

- ✓ Cellular radio signal are to week to provide reliable communication at indoor, especially in well-shielded areas or areas with high levels of interference.
- ✓ To overcome this, very small cells called Pico cells are used in same frequencies as regular cells in the same areas.

25. Define cell & cell cluster.

- ✓ Each cellular base station is allocated a group of radio channels to be used with a small geographic area called a cell.
- ✓ A group of cells that use a different set of frequencies in each cell is called a cell cluster.

26. Based on the location of BS, how cells are classified?

- ✓ When designing a system using hexagonal-shaped cells, main consideration is the location of the base station transmitters.
  - ➢ Center-excited cell- Base station transmitters can be located in the center of the cell and uses Omni directional antennas which radiate and receive signals equally well in all directions.
  - ➢ Edge- excited cell- Base station transmitters can be located in the edge of the cell and uses sectored antennas which radiate for a particular direction.
  - ➢ Corner- excited cell- Base station transmitters can be located in the corner of the cell and uses sectored directional antennas.

FREQUENCY REUSE

27. Define Frequency reuse. [May 2016, May 2013, Nov 2016, Nov/Dec 2017, April/May 2018]

- ✓ The design process of selecting and allocating channel groups for all of the cellular base stations within a system is called frequency reuse or frequency planning.
- ✓ Physical separation of two cells is sufficiently wide; the same subset of frequencies can be used in both cells.
  - ▪ This is the concept of frequency reuse.
  - ▪ The same spectrum can support multiple users and available spectrum is efficiently utilized.

28. Define foot print.

- ✓ The actual radio coverage of a cell is known as the foot print. It is determined from field measurement or propagation prediction models.

29. Express the total number of channels available in cluster.

For total number of cellular channels available in a cluster can be expressed mathematically as $S=Kn$

Where, S-Number of full-duplex cellular channels available in cluster.
K-Number of channels in a cell and
n-Number of cells in a cluster.

30. What are the rules used to determine the nearest co channel neighbors?

The following two-step rules can be used to determine the location of the nearest co channel cell:

Step 1: Move I cells along any chain of hexagons;

Step 2: Turn 60 degrees counter clockwise and more j cells.

31. Write the expression for cellular system capacity.

Let M be the number of times the cluster is replicated and C be the total number of channels used in the entire cellular system with frequency reuse. C is then the system capacity and is given by **C=MKn; C=MS**

Where C- Total channel capacity in a given area
M-Number of clusters in a given area

32. Define FRF.

The number of user use the same set of frequencies is called the frequency reuse factor (FRF) and is defined mathematically as

$$\mathbf{FRF} = \frac{\boldsymbol{N}}{\boldsymbol{C}}$$

Where N-Total number of full-duplex channels in an area
C-Total number of full-duplex channels in a cell.

<u>HAND OFF</u>

33. Write the advantages of cellular systems?

- ✓ The advantages of Cellular Systems:
  - The use of low power transmitter and
  - It allows frequency reuse for capacity improvement.

34. Define Dwell time.

The time over which a call may be maintained within a cell, without handoff, is called the dwell time.

35. What are the methods used for handoffs?

- ✓ Depending on the information used and the action taken to initiate the handoff, the methods for handoff can be
  - Mobile Controlled Hand off (MCHO)
  - Network Controlled Hand off (NCHO) and
  - Mobile Assisted Hand off (MAHO)

36. Write about umbrella cell approach and its usage.

- ✓ By using different antenna heights (same building or tower) and different power levels, it is possible to provide "large" and" small" cells which are co-located at a single location.
- ✓ The umbrella cell approach is used to provide large area coverage to high speed users while providing small area coverage to users travelling at low speeds.

37. Write a short note on hard handoff and Soft handoff.
What is soft handoff in mobile communication? [May 2016]

- ✓ **Hard Handoff:** If the MSC monitors the strongest signal base station and transfer the call to that base station then it is called hard handoff.
- ✓ **Soft handoff:** Mobile communicates with two or more cells at the same time and find which one is a strongest signal base station then it automatically transfers the call to that base station is called soft handoffs.

38. In a cellular network, among a handoff call and new call, which one is given priority? Why? [April 2017]

- ✓ Different systems have different methods for handling and managing handoff request.
- ✓ Some systems handle handoff in same way as they handle new originating call.
- ✓ In such system the probability that the handoff will not be served is equal to blocking probability of new originating call.

- ✓ But if the call is terminated abruptly in the middle of conversation then it is more annoying than the new originating call being blocked.
- ✓ So in order to avoid this abrupt termination of ongoing call handoff request should be given priority to new call this is called as handoff prioritization.

39. What are the techniques used to prioritize the handoff call and new call?

There are two techniques for this:

Guard Channel Concept

In this technique, a fraction of the total available channel in a cell is reserved exclusively for handoff request from ongoing calls which may be handed off into the cell.

Queuing

Queuing of handoffs is possible because there is a finite time interval between the time the received signal level drops below handoff threshold and the time the call is terminated due to insufficient signal level. The delay size is determined from the traffic pattern of a particular service area.

40. Mention the limitations of cellular communication systems? [June 2013]

Limitations of cellular communication systems

i. fixed network needed for the base stations
ii. handover (changing from one cell to another) necessary
iii. interference with other cells

41. What are the reasons for handover? [Nov 2013]

There are different reasons for handover:

i. When the phone is moving away from the area covered by one cell and entering the area covered by another cell, the call is transferred to the second cell, in order to avoid call termination.
ii. When the capacity for connecting new calls of a given cell is used up and an existing or new call from a phone is transferred to that cell in order to free-up some capacity in the first cell.

42. Write the features of handoff.

- ✓ Fast and lossless
- ✓ Minimal number of control signal exchanges.
- ✓ Scalable with network size.
- ✓ Capable of recovering from link failures and
- ✓ Efficient use of resources.

CHANNEL ASSIGNMENT

43. Name the two channels assignments.

- ✓ There are essentially two channels assignment approaches
  - ▪ Fixed channel assignment and
  - ▪ Dynamic channel assignment

44. What is FCA?

- ✓ In FCA, each cell is allocated a predetermined (permanently) set of voice channels. Any call attempt within the cell can only be served by the unused channels in that particular cell.

45. Define borrowing strategy.

- ✓ To improve utilization, a borrowing option may be considered borrowing strategy; a cell is allowed to borrow channels from a neighboring cell if all of its own channels are already occupied.

46. What do you meant by DCA? Give its advantages.

- ✓ In DCA, voice channels are not allocated to different cells permanently. Each time a cell request is made, the serving base station request a channel from the MSC.
- ✓ Dynamic channel assignment reduces the call blocking, which increases the trucking capacity of the system, since all available channel under the control of the MSC are accessible to the entire cell.

47. Define co-channel reuse ratio. [Nov 2015]

The co-channel reuse ratio Q is defined as

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

Where, D-Distance between centers of the nearest co-channel cells

R-Radius of the cell

48. Mention a few techniques used to expand the capacity of a cellular system. [May 2015]

Cell splitting, Sectoring, Coverage Zone approaches are the techniques used to expand the capacity of cellular system.

- Cell splitting
  - Cell-splitting is a technique which has the capability to add new smaller cells in specific areas of the system. i.e. divide large cell size into small size.
- Sectoring
  - use of directional antennas to reduce Co-channel interference.
- Coverage Zone approaches
  - Large central BS is replaced by several low power transmitters on the edge of the cell.

49. Define co-channel Interference. [Nov 2015, May 2016]

- ✓ Co-channel interference is caused due to the cells that reuse the same frequency set.
- ✓ The cells using the same frequency set are called co-channel cells.
- ✓ The interference between signals from the co-channel cells is called co-channel interference.

50. Define adjacent channel Interference.

- ✓ Interference resulting from signals which are adjacent in frequency to the desired signal is called adjacent channel interference.
- ✓ Adjacent channel interference results from imperfect receiver filters that allow nearby frequencies to leak into the passband.

51. What do you mean by forward and reverse channel? [Nov/Dec 2017]

- ✓ The channels used for transmission from the base station to mobiles are called ***forward channels***
- ✓ The channels used for transmission from mobiles to the base station are called ***reverse channels***.

52. Differentiate between FDMA, TDMA and CDMA technologies.[April/May 2018]

S.N	FDMA	TDMA	CDMA
1	Channel bandwidth is subdivided into number of sub channels	The radio spectrum is divided into time slots and each slot is allotted for only one user who can either transmit or receive.	Sharing of bandwidth and time takes place.
2	FDMA uses Narrow band Systems.	TDMA uses Narrow band Systems or wide band Systems	CDMA uses Wide band Systems.
3	FDMA is First generation wireless standard (1G).	TDMA is Second generation wireless standard (2G).	CDMA is third generation wireless standard (3G).
4	FDMA is use for the voice and data transmission	TDMA is used for data and digital voice signals	CDMA is use for digital voice signals and multimedia services.
5	Due to non-linearity of power amplifiers, inter-modulation products are generated due to interference between adjacent channels.	Due to incorrect synchronization there can be interference between the adjacent time slots.	Both type of interference will be present.
6	Synchronization is not necessary	Synchronization is necessary	Synchronization is not necessary
7	Code word is not required	Code word is not required	Code words are required
8	Guard bands between adjacent channels are necessary.	Guard times between adjacent time slots are necessary.	Guard bands and guard times are necessary.

******

1. Consider a time invariant frequency selective block fading channel consisting of a 3 sub-channels of $B = 1$ MHZ. The frequency response associated with each channels is $H_1 = 1, H_2 = 2, H_3 = 3$. The transmit power constraint is $P = 10mW$ and noise power spectral density is $N_0 = 10^{-9} W/HZ$. Find the Shannon capacity of the channel and optimal power allocation that achieves this capacity. [April / May 2018]

γ_i = SNR, j^{th} channel $= \frac{|H_j|^2 P}{N_0 B}$ for each sub channel.

$\gamma_1 = 10,$

$\gamma_2 = 40,$

$\gamma_3 = 90.$

cut off γ_0 must satisfy $\sum_j \left(\frac{1}{\gamma_0} - \frac{1}{\gamma_j}\right) = 1.$

this yields,

$\frac{3}{\gamma_0} = 1 + \sum_j \frac{1}{\gamma_j} = 1.14, \quad \gamma_0 = 2.64 < \gamma_j$

$\gamma_0 < \gamma_j$ for all j

$$C = \sum_{j=1}^{3} B \log_2 \left(\frac{\gamma_i}{\gamma_0}\right) = 10{,}000{,}000 \left\{ \log_2\left(\frac{10}{2.64}\right) + \log_2\left(\frac{40}{2.64}\right) + \log_2\left(\frac{90}{2.64}\right) \right\}$$

$= 10.93$ Mbps.

UNIT III
DIGITAL SIGNALING FOR FADING CHANNELS

Syllabus:

Structure of a wireless communication link, Principles of Offset-QPSK, p/4-DQPSK, Minimum Shift Keying, Gaussian Minimum Shift Keying, Error performance in fading channels, OFDM principle – Cyclic prefix, Windowing, PAPR.

3.1 STRUCTURE OF A WIRELESS COMMUNICATION LINK

Briefly explain the structure of a wireless communication link. (6M-May 2016) (6M-May 2015) (6M-Nov 2102)

Transmitter

- ✓ The transmitter adds redundancy in the form of a **F**orward **E**rror **C**orrection (FEC) code
- ✓ Addition of FEC code makes it more resistant to errors introduced by the channel.
- ✓ The encoded data are used as input to a modulator, which maps the data to output waveforms that can be transmitted.
- ✓ By transmitting these symbols on specific frequencies or at specific times, different users can be distinguished.
- ✓ The signal is then sent through the propagation channel, which attenuates and distorts it, and adds noise.

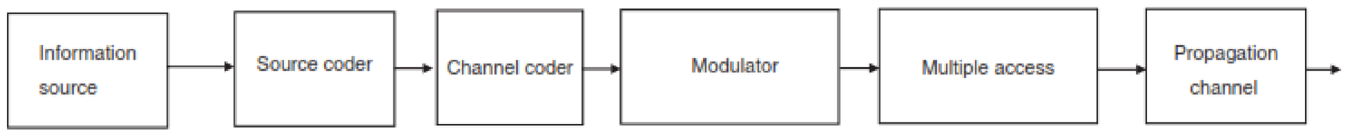

Figure: Block diagram of a Transmitter

Receiver

- ✓ The signal is received by one or more antennas.
- ✓ The different users are separated.
- ✓ If the channel is delay dispersive, then an equalizer can be used to reverse that dispersion, and eliminate intersymbol interference.
- ✓ The signal is demodulated.
- ✓ A channel decoder eliminates the errors that are present in the resulting bitstream.
- ✓ A source decoder finally maps this bit stream to an analog information stream that goes to the information sink; in the case when the information was originally digital, this last stage is omitted.

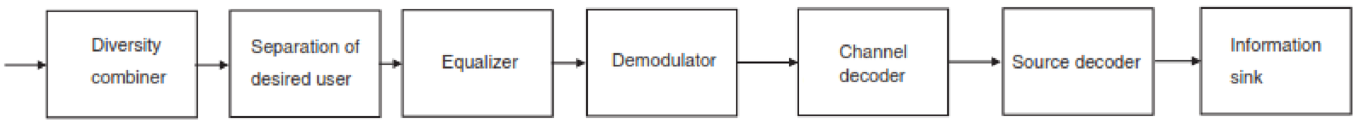

Figure: Block diagram of a Receiver

Radio link with digital transmitter

Information source

- ✓ The information source provides an analog source signal.
- ✓ It feeds it into the source Analog to Digital Converter.

Analog to Digital Converter

- ✓ ADC band limits the signal from the analog information source
- ✓ It converts the signal into a stream of digital data at a certain sampling rate and resolution

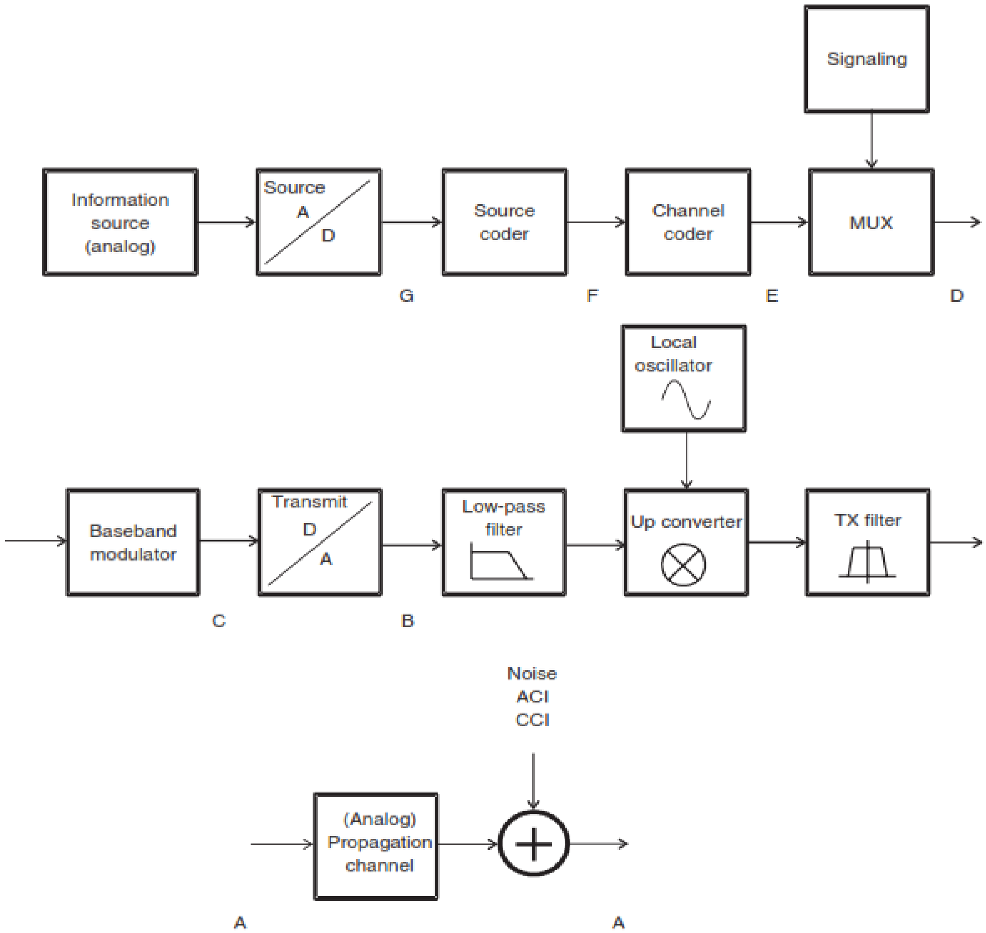

Figure: Block diagram of a radio link with digital transmitter and analog propagation channel.

Source coder

- ✓ The source coder uses a priori information on the properties of the source data in order to reduce redundancy in the source signal.
- ✓ This reduces the amount of source data to be transmitted
- ✓ This also reduces the required transmission time and bandwidth.
- ✓ The source coder increases the entropy of the data.
- ✓ For some applications, source data are encrypted in order to prevent unauthorized listening in.

Channel coder

- ✓ The channel coder adds redundancy in order to protect data against transmission errors.
- ✓ This increases the data rate.
- ✓ Channel coders use information about the error sources in the channel to design codes that are well suited for certain types of channels.

- ✓ Data can be sorted according to importance; more important bits get stronger protection.
- ✓ It is possible to use interleaving to break up error bursts.

Signaling

- ✓ Signaling adds control information for the establishing and ending of connections, for associating information with the correct users, synchronization, etc. Signaling information is usually strongly protected by error correction codes.

Multiplexer

- ✓ The multiplexer combines user data and signaling information, and combines the data from multiple users.

Baseband modulator

- ✓ The baseband modulator assigns the gross data bits to complex transmit symbols in the baseband.
- ✓ Spectral properties, intersymbol interference, peak- to-average ratio, and other properties of the transmit signal are determined by baseband modulator.
- ✓ The output from the baseband modulator provides the transmit symbols in oversampled form, discrete in time and amplitude.
- ✓ Oversampling and quantization determine the aliasing and quantization noise.
- ✓ High resolution is desirable, and the data rate at the output of the baseband modulator should be much higher than at the input.

Digital to Analog Converter (DAC)

- ✓ The TX Digital to Analog Converter generates a pair of analog, discrete amplitude voltages corresponding to the real and imaginary part of the transmit symbols, respectively.

Analog low-pass filter

- ✓ The analog low-pass filter in the TX eliminates the spectral components outside the desired transmission bandwidth.
- ✓ These components are created by the out-of-band emission of an baseband modulator, which stem from the properties of the chosen modulation format.
- ✓ Furthermore, imperfections of the baseband modulator and imperfections of the DAC lead to additional spurious emissions that have to be suppressed by the TX filter.

Local Oscillator

- ✓ The TX Local Oscillator (LO) provides an unmodulated sinusoidal signal, corresponding to one of the admissible center frequencies of the considered system.
- ✓ The requirements for frequency stability, phase noise, and switching speed between different frequencies depend on the modulation and multiaccess method.

Up converter

- ✓ The up converter converts the analog, filtered baseband signal to a passband signal by mixing it with the LO signal.
- ✓ Upconversion can occur in a single step, or in several steps. Finally, amplification in the Radio Frequency (RF) domain is required.

RF TX filter

- ✓ The RF TX filter eliminates out-of-band emissions in the RF domain.
- ✓ Even if the low-pass filter succeeded in eliminating all out-of-band emissions, upconversion can lead to the creation of additional out-of-band components.
- ✓ Nonlinearities of mixers and amplifiers lead to intermodulation products and spectral regrowth
- ✓ Spectral regrowth is creation of additional out-of-band emissions.

Propagation channel

- ✓ The (analog) propagation channel attenuates the signal, and leads to delay and frequency dispersion. Furthermore, the environment adds noise (Additive White Gaussian Noise – AWGN) and co-channel interference.

Radio link with digital receiver

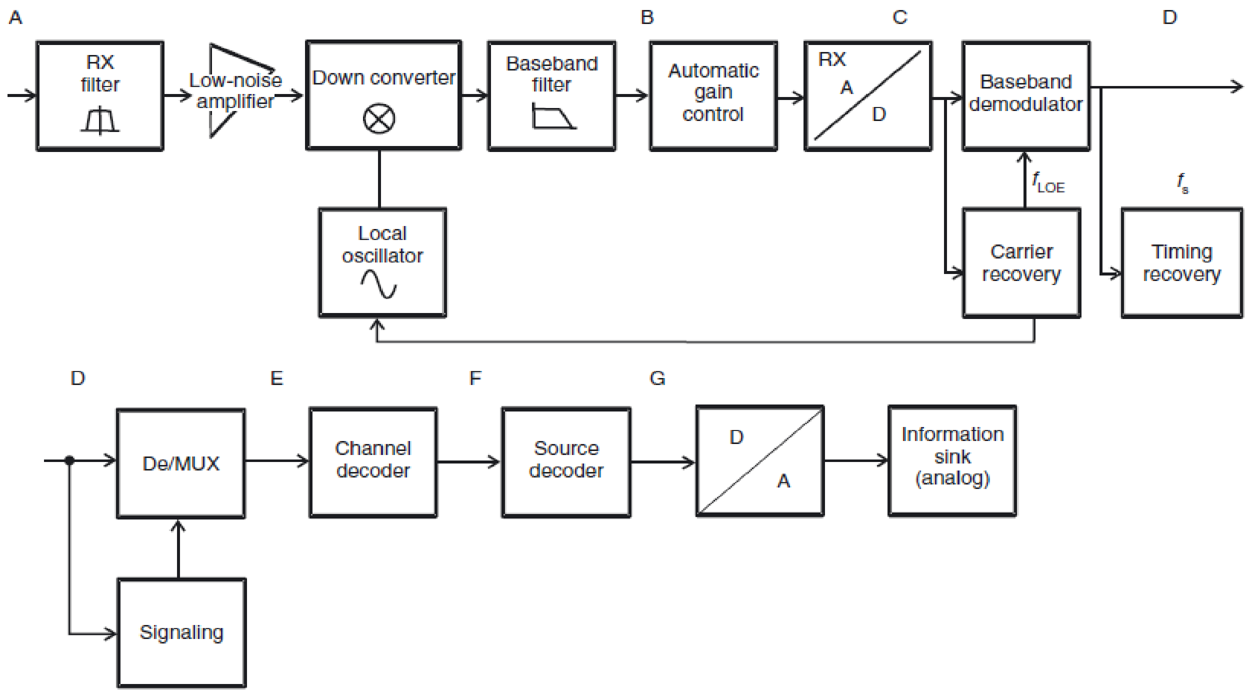

Figure: Block diagram of a digital receiver chain for mobile communications

RX filter

- ✓ The RX filter performs a rough selection of the received band.
- ✓ The bandwidth of the filter corresponds to the total bandwidth assigned to a specific service, and can thus cover multiple communications channels belonging to the same service.

Low-noise amplifier

- ✓ The low-noise amplifier amplifies the signal, so that the noise added by later components of the RX chain has less effect on the Signal-to-Noise Ratio (SNR).
- ✓ Further amplification occurs in the subsequent steps of down conversion.

RX Local Oscillator

- ✓ The RX LO provides sinusoidal signals corresponding to possible signals at the TX LO.
- ✓ The frequency of the LO can be fine-tuned by a carrier recovery algorithm to make sure that the LOs at the TX and the RX produce oscillations with the same frequency and phase.

Down converter

- ✓ The RX down converter converts the received signal into baseband.
- ✓ In baseband, the signal is thus available as a complex analog signal.

RX low-pass filter

- ✓ The RX low-pass filter provides a selection of desired frequency bands for one specific user.
- ✓ It eliminates adjacent channel interference as well as noise. The filter should influence the desired signal as little as possible.

Automatic Gain Control

- ✓ The Automatic Gain Control (AGC) amplifies the signal such that its level is well adjusted to the quantization at the subsequent ADC.

RX ADC

- ✓ The RX ADC converts the analog signal into values that are discrete in time and amplitude.
- ✓ The required resolution of the ADC is determined essentially by the dynamics of the subsequent signal processing.
- ✓ The sampling rate is of limited importance as long as the conditions of the sampling theorem are fulfilled.
- ✓ Oversampling increases the requirements for the ADC, but simplifies subsequent signal processing.

Carrier recovery

- ✓ Carrier recovery determines the frequency and phase of the carrier of the received signal, and uses it to adjust the RX LO.

Baseband demodulator

- ✓ The baseband demodulator obtains soft-decision data from digitized baseband data, and hands them over to the decoder.
- ✓ The baseband demodulator can be an optimum, coherent demodulator, or a simpler differential or incoherent demodulator.
- ✓ This stage can also include further signal processing like equalization.
- ✓ If there are multiple antennas, then the RX either selects the signal from one of them for further processing or the signals from all of the antennas have to be processed (filtering, amplification, down conversion).
- ✓ In the latter case, those baseband signals are then either combined before being fed into a conventional baseband demodulator or they are fed directly into a "joint" demodulator that can make use of information from the different antenna elements.

Symbol-timing recovery

- ✓ Symbol-timing recovery uses demodulated data to determine an estimate of the duration of symbols, and uses it to fine-tune sampling intervals.
- ✓ The decoder uses soft estimates from the demodulator to find the original (digital) source data.
- ✓ In the simplest case of an uncoded system, the decoder is just a hard-decision (threshold) device.
- ✓ For convolutional codes, Maximum Likelihood Sequence Estimators (MLSEs, such as the Viterbi decoder) are used.
- ✓ Recently, iterative RXs that perform joint demodulation and decoding have been proposed.

- ✓ Remaining errors are either taken care of by repetition of a data packet (Automatic Repeat reQuest – ARQ) or are ignored.
- ✓ The latter solution is usually resorted to for speech communications, where the delay entailed by retransmission is unacceptable.

Signaling

- ✓ Signaling recovery identifies the parts of the data that represent signaling information and controls the subsequent demultiplexer.

Demultiplexer

- ✓ The demultiplexer separates the user data and signaling information and reverses possible time compression of the TX multiplexer. Note that the demultiplexer can also be placed earlier in the transmission scheme; its optimum placement depends on the specific multiplexing and multi access scheme.

Source decoder

- ✓ The source decoder reconstructs the source signal from the rules of source coding.
- ✓ If the source data are digital, the output signal is transferred to the data sink.
- ✓ The data are from source decoder are transferred to the DAC.

DAC

- ✓ DAC converts the transmitted information into an analog source decoder

Information sink

- ✓ DAC hands over the converted signal to the information sink.

<u>Simplified Models</u>

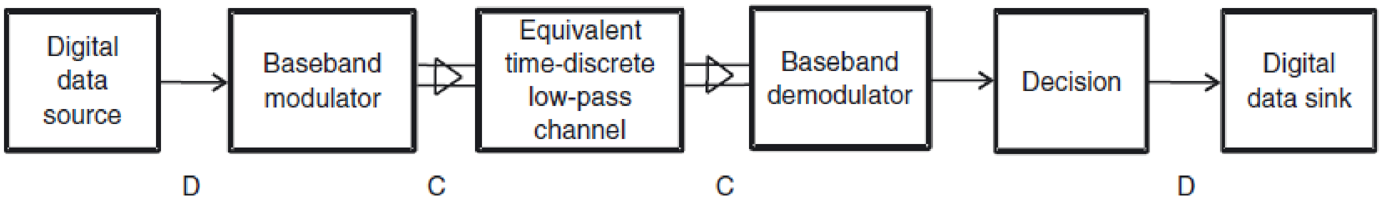

Figure: Mathematical link model for the analysis of modulation formats

- ✓ Figure shows a model that is suitable for the analysis of modulation methods.
- ✓ The parts of the TX between the information source and the output of the TX multiplexer are subsumed into a black box digital data source.
- ✓ The analog radio channel, together with the up converters, down converters, RF elements (filters, amplifiers), and all noise and interference signals, is subsumed into an equivalent time-discrete low-pass channel, characterized by a time-variant impulse response and the statistics of additive disturbances.
- ✓ The quality of the modulation is determined by the bit error probability.
- ✓ Other simplified models use a digital representation of the channel (e.g., binary symmetric channel), and are mainly suitable for the analysis of coding schemes.

****

Digital Modulation

- ✓ In modern mobile communication digital modulation is used.
- ✓ Advancements in VLSI and DSP have made digital modulation techniques more cost efficient and more useful than analog transmission systems.
- ✓ Advantages of digital modulation includes
  - ➢ Greater noise immunity
  - ➢ Easier multiplexing of information, voice, data and video
  - ➢ Accommodates digital transmission errors, source coding, encryption and equalization techniques
  - ➢ Digital signal processors can implement digital modulators, demodulators completely in software
- ✓ Factors that influence the choice of digital modulation are
  - ➢ Low bit error at low signal to noise ratio
  - ➢ Perform well in multipath fading environments
  - ➢ Occupies a minimum bandwidth
  - ➢ Easy and cost effective implementation
  - ➢ According to our requirement tradeoffs are made for selecting a digital modulation technique

Linear	Non Linear	Spread spectrum
Bandwidth efficient. Useful to accommodate more users in a limited spectrum.	Higher Bandwidth but high immunity against random FM noise.	Inefficient for single user but efficient for multi users.
Eg: QPSK OQPSK $\pi/4$ QPSK	Eg: FSK GMSK MFSK	
Amplitude of transmitted signal (t) varies with message signal	Amplitude of the carrier is constant	Transmission bandwidth >> Minimum required signal bandwidth.

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3.2 PRINCIPLES OF OFFSET-QPSK

1. **Describe with neat diagram, the modulation technique of OQPSK and its advantage? (Apr/May 2017, Nov/Dec 2017)**
2. **Explain in detail Offset QPSK linear digital modulation techniques employed in wireless communication. (May 2012, May/June 2016)**

- ✓ The amplitude of a QPSK signal is ideally constant.
- ✓ When QPSK signal is pulse shaped, they lose the constant envelope property.
- ✓ The occasional phase shift of π radians can cause the signal envelope to pass through zero for just an instant.
- ✓ Any nonlinear amplification of the zero-crossings brings back the filtered sidelobes.
- ✓ This is because the fidelity of the signal is lost due to small voltage levels during transmission.

- ✓ To prevent the generation of side lobes and spectral widening offset QPSK was introduced.
- ✓ The QPSK signals are amplified only using linear amplifiers, which are less efficient.
- ✓ A modified form of QPSK, called ***offset QPSK (OQPSK)*** or ***staggered QPSK*** is less susceptible to these deleterious effects arid support more efficient amplification.
- ✓ OQPSK signaling is represented by equation,

$$S_{OQPSK}(t) = \sqrt{\frac{2E_S}{T_S}}\cos\left[(i-1)\frac{\pi}{2}\right]\cos(2\pi f_c t) - \sqrt{\frac{2E_S}{T_S}}\sin\left[(i-1)\frac{\pi}{2}\right]\sin(2\pi f_c t)$$

- ✓ In QPSK signaling, the bit transitions of the even and odd bit streams occur at the same time instants.
- ✓ But in OQPSK signaling, the even and odd bit streams, $m_I(t)$ and $m_Q(t)$, are offset in their relative alignment by one bit period (half-symbol period). This is shown in the waveforms of Figure.

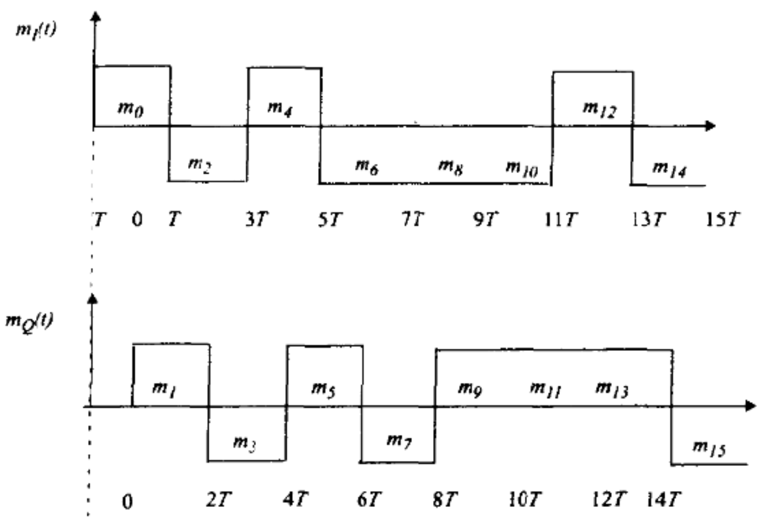

Figure: The time offset waveforms that are applied to the in-phase and quadrature arms of an OQPSK modulator. Notice that a half-symbol offset is used

- ✓ Due to the time alignment of $m_I(t)$ and $m_Q(t)$ in standard QPSK, phase transitions occur only once every $T_s = 2T_b s$, and will be a maximum of 180^0 if there is a change in the value of both $m_I(t)$ and $m_Q(t)$
- ✓ However, in OQPSK signaling, bit transitions occur every $T_b s$.
- ✓ Since the transitions instants of $m_I(t)$ and $m_Q(t)$ are offset, at any given time only one of the two bit streams can change values.
- ✓ This implies that the maximum phase shift of the transmitted signal at any given time is limited to $\pm 90^o$.
- ✓ By switching phases more frequently (i.e., every $T_b\ s$ instead of $2T_b\ s$) OQPSK signaling eliminates 180° phase transitions.
- ✓ Nonlinear amplification of OQPSK signals does not regenerate the high frequency sidelobes as much as in QPSK.
- ✓ Thus, spectral occupancy is significantly reduced, while permitting more efficient RF amplification.
- ✓ The spectrum of an OQPSK signal is identical to that of a QPSK signal; hence both signals occupy the same bandwidth.

- ✓ OQPSK retains its bandlimited nature even after nonlinear amplification.
  - Therefore this is very attractive for mobile communication systems where bandwidth efficiency and efficient nonlinear amplifiers are critical for 1.0W power drain.
- ✓ OQPSK signals also appear to perform better than QPSK in the presence of phase jitter due to noisy reference signals at the receiver.

Difference between OQPSK and QPSK

OQPSK	QPSK
In OQPSK signaling, the even and odd bit streams, $m_I(t)$ and $m_Q(t)$ are offset in their relative alignment by one bit period	In QPSK signaling, the bit transitions of the even and odd bit streams occur at the same time instants.
OQPSK signals does not regenerate the high frequency side lobes	QPSK signals regenerate the high frequency side lobes.

Similarities of OQPSK and QPSK

- ✓ The signals are identical and hence both signals occupy the same bandwidth.
- ✓ The staggered alignment of even and odd bits does not change the nature of the alignment.

Advantages of OQPSK

The advantages of offset-QPSK includes

- ✓ Lower amplitude fluctuations.
- ✓ Suppress out-of-band interference.
- ✓ Limits the phase-shift to maximum of 90° at a time.
- ✓ Spectral occupancy is significantly reduced.
- ✓ More efficient RF amplification.
- ✓ Better performance in the presence of phase jitter due to noisy reference signals at the receiver

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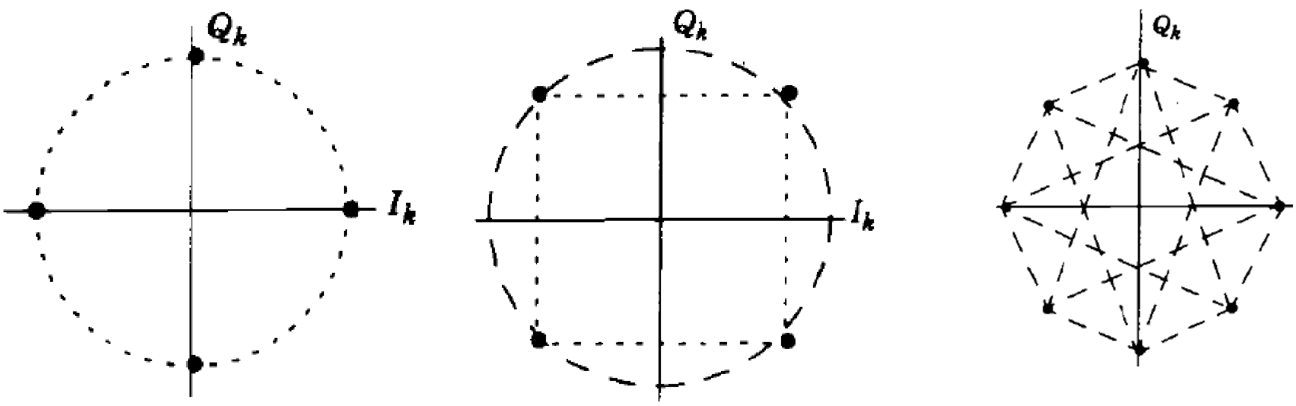
3.3 $\frac{\pi}{4}$ - DQPSK

1. **Explain in detail $\pi/4$ DQPSK linear digital modulation techniques employed in wireless communication. (8M-May 2016)**
2. **Describe with a block diagram $\pi/4$ quadrature phase shift keying and its advantages.**
3. **(8M-Nov 2014)**
4. **Explain the principle of $\pi/4$ Differential quadrature phase shift keying from a signal space diagram.(8M-May 2013)**
5. **Draw the constellation of QPSK. (4M-April 2010]**
6. **With neat diagram, explain the modulation and demodulation of $\pi/4$ DQPSK modulation techniques. [April/May2018]**

Description of $\frac{\pi}{4}$ - DQPSK

- ✓ The **π/4** shifted QPSK modulation is a quadrature phase shift keying technique.
- ✓ It offers a compromise between OQPSK and QPSK in terms of the allowed ***maximum phase transitions***.
- ✓ It may be demodulated in a coherent or noncoherent fashion.
- ✓ The ***maximum phase change*** is limited to $\pm 135^o$ as compared to 180^o for QPSK and 90^o for ***Offset QPSK***.
- ✓ Hence, the band limited ***π/4QPSK*** signal preserves the constant envelope property better than bandlimited QPSK.
- ✓ But ***π/4QPSK*** is more susceptible to envelope variations than OQPSK.
- ✓ ***Attractive feature of π/4 QPSK:*** This can be non-coherently detected fashion, which greatly simplifies the receiver design.
- ✓ In the presence of multipath spread and fading, $\pi/4$ QPSK is found to perform better.
- ✓ Very often, ***π/4QPSK*** signals are differentially encoded to offer easier differential detection or coherent demodulation.
- ✓ When differentially encoded *π/4 QPSK* is called differential QPSK (DQPSK).



(a) Possible States for θ_k when $\theta_{k-1} = \frac{n\pi}{4}$

(b) Possible States for θ_k when $\theta_{k-1} = \frac{n\pi}{2}$

(c) All Possible States

Figure: Constellation diagram of a $\frac{\pi}{4}$ QPSK signal

$\pi/4$ QPSK Transmission Techniques

- ✓ A block diagram of a generic $\pi/4$ QPSK transmitter is shown in Figure.

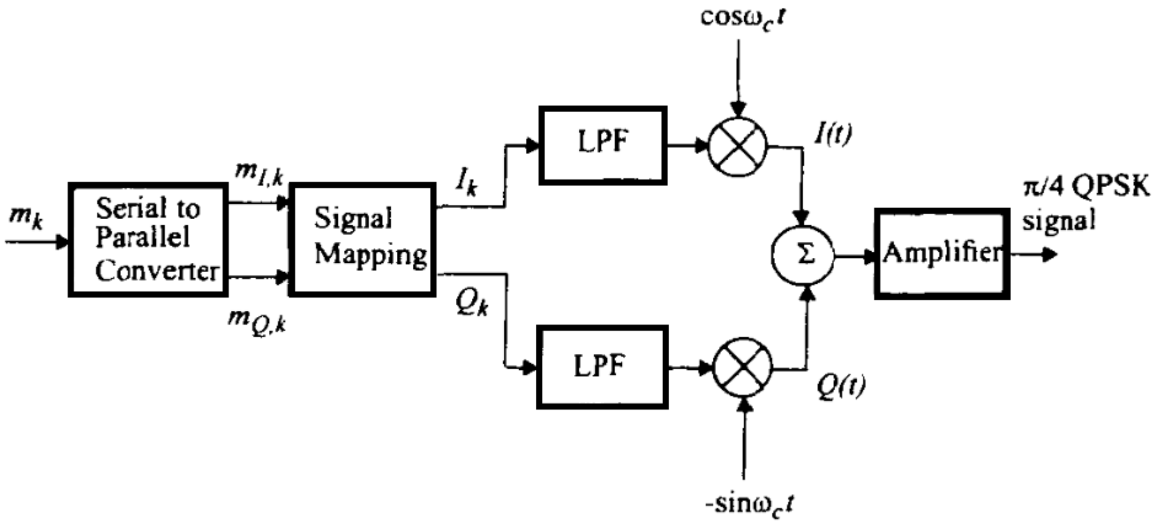


Figure: Generic $\frac{\pi}{4}$ QPSK transmitter

- ✓ The input bit stream is partitioned by a serial-to-parallel (S/P) converter into two parallel data streams $m_{I,k}$ and $m_{Q,k}$ each with a symbol rate equal to half that of the incoming bit rate.
- ✓ The k^{th} in-phase and quadrature pulses, I_k and Q_k are produced at the output of the signal mapping circuit over time $kT \le t \le (k+1)T$ and are determined by their previous values, I_{k-1} and Q_{k-1} and as well as θ_k which itself is a function of ϕ_k which is a function of the current input symbols m_{Ik} and m_{Qk}.
- ✓ I_k and Q_k represent rectangular pulses over one symbol duration having amplitudes given by

$$I_k = \cos\theta_k = I_{k-1}\cos\phi_k - Q_{k-1}\sin\phi_k$$

$$Q_k = \sin\theta_k = I_{k-1}\sin\phi_k - Q_{k-1}\cos\phi_k$$

where $$\theta_k = \theta_{k-1} + \phi_k$$

and θ_k and θ_{k-1} are phases of the k^{th} and $k\text{-}1^{st}$ symbols.

- ✓ The phase shift ϕ_k is related to the input symbols m_{Ik} and m_{Qk} according to Table.

Table: Carrier phase shifts corresponding to various input bit pairs

Information bits m_{Ik}, m_{Qk}	**Phase shift** ϕ_k
11	$\pi/4$
01	$3\pi/4$
00	$-3\pi/4$
10	$-\pi/4$

- ✓ As in a QPSK modulator, the in-phase and quadrature bit streams I_k and Q_k are then separately modulated by two carriers which are in quadrature with one another, to produce the $\pi/4$ QPSK waveform given by

$$S_{\pi/4QPSK}(t) = I(t)\cos\omega_c t - Q(t)\sin\omega_c t$$

- ✓ Both I_k and Q_k are usually passed through raised cosine rolloff pulse shaping filters before modulation, in order to reduce the bandwidth occupancy.

$$I(t) = \sum_{k=0}^{N-1} I_k \; p(t - kT_s - T_s/2) = \sum_{k=0}^{N-1} \cos\theta_k \; p(t - kT_s - T_s/2)$$

$$Q(t) = \sum_{k=0}^{N-1} Q_k \; p(t - kT_s - T_s/2) = \sum_{k=0}^{N-1} \sin\theta_k \; p(t - kT_s - T_s/2)$$

- ✓ The function $p(t)$ in equations and corresponds to the pulse shape, and T_s is the symbol period.
- ✓ Pulse shaping also reduces the spectral restoration problem which may be significant in fully saturated, nonlinear amplified systems.
- ✓ It should be noted that the values of I_k and Q_k and the peak amplitude of the waveforms I(t) and Q (t) can take one of the five possible values, 0,+1,-1, $+1/\sqrt{2}$ and $-1/\sqrt{2}$.

$\pi/4$ QPSK Detection Techniques:

- ✓ Due to ease of hardware implementation, differential detection is employed to demodulate $\pi/4$ QPSK signals.
- ✓ In an AWGN channel, the BER performance of
  - ➢ Differentially detected $\pi/4$ QPSK is about 3 dB inferior to QPSK,
  - ➢ Coherently detected $\pi/4$ QPSK has the same error performance as QPSK.
- ✓ In low bit rate, fast Rayleigh fading channels, differential detection offers a ***lower error floor*** since it does not depends on phase synchronization.
- ✓ There are various types of detection techniques that are used for the detection of $\pi/4$ QPSK signals that includes
  - ***Baseband differential detection*** determines the cosine and sine functions of the phase difference, and then decides on the phase difference
  - ***IF differential detection*** determines the cosine and sine functions of the phase difference, and then decides on the phase difference
  - ***FM discriminator detection*** detects the phase difference directly in a noncoherent manner.

<u>Baseband Differential Detection</u>

- ✓ Figure shows a block diagram of a baseband differential detector.

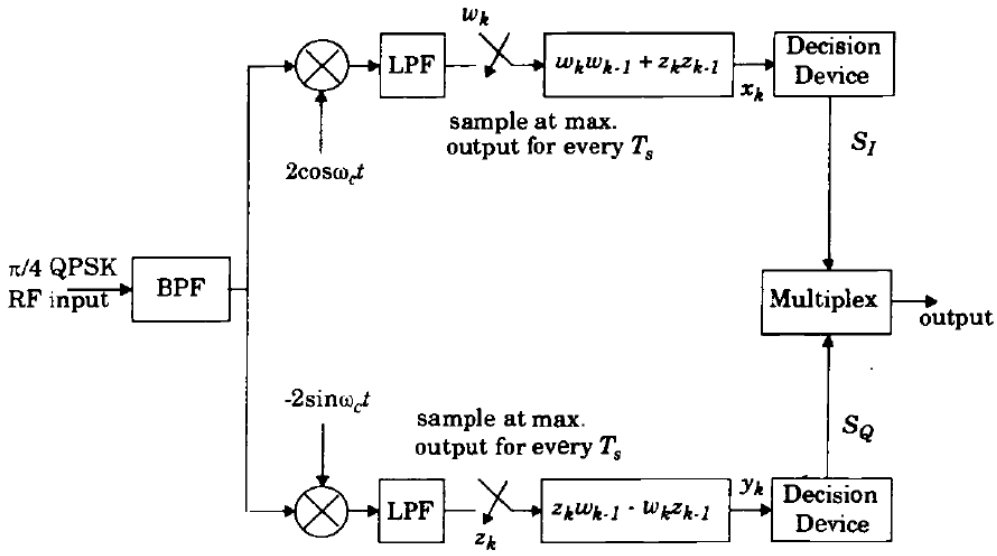


Figure: Block diagram of a baseband differential detector

- ✓ The incoming $\pi/4$ QPSK signal is quadrature demodulated using two local oscillator signals that have the same frequency as the unmodulated carrier at the transmitter, but not necessarily the same phase.
- ✓ If $\phi_k = \tan^{-1}\left(Q_k / I_k\right)$ is the phase of the carrier due to the k^{th} data bit, the output and w_k and z_k from the two low pass filters in the in-phase and quadrature arms of the demodulator can be expressed as

$$w_k = \cos\left(\phi_k - \gamma\right)$$

$$z_k = \sin\left(\phi_k - \gamma\right)$$

where,

γ is a phase shift due to noise, propagation, and interference.

- ➢ The phase γ is assumed to change much slower than ϕ_k so it is essentially a constant.
- ➢ The two sequences w_k and z_k are passed through a differential decoder which operates on the following rule.

$$x_k = w_k w_{k-1} + z_k z_{k-1}$$

$$y_k = z_k w_{k-1} - w_k z_{k-1}$$

- ✓ The output of the differential decoder can be expressed as

$$x_k = \cos\left(\phi_k - \gamma\right)\cos\left(\phi_{k-1} - \gamma\right) + \sin\left(\phi_k - \gamma\right)\sin\left(\phi_{k-1} - \gamma\right)$$

$$= \cos\left(\phi_k - \phi_{k-1}\right)$$

$$y_k = \sin\left(\phi_k - \gamma\right)\cos\left(\phi_{k-1} - \gamma\right) + \cos\left(\phi_k - \gamma\right)\sin\left(\phi_{k-1} - \gamma\right)$$

$$= \sin\left(\phi_k - \phi_{k-1}\right)$$

Table: Carrier phase shifts corresponding to various input bit pairs

Information bits m_{Ik}, m_{Qk}	**Phase shift** ϕ_k
11	$\pi/4$
01	$3\pi/4$
00	$-3\pi/4$
10	$-\pi/4$

- ✓ The output of the differential decoder is applied to the decision circuit, which uses the above table to determine

$$S_I = 1,\ \text{if}\ x_k > 0 \quad \text{or} \quad S_I = 0,\ \text{if}\ x_k < 0$$

$$S_Q = 1,\ \text{if}\ y_k > 0 \quad \text{or} \quad S_Q = 0,\ \text{if}\ y_k < 0$$

where S_I and S_Q are the detected bits in the in-phase and quadrature arms, respectively.

- ✓ It is important to ensure the local receiver oscillator frequency is the same as the transmitter carrier frequency, and that it does not drift.
- ✓ Any drift in the carrier frequency will cause a drift in the output phase which will lead to BER degradation.

IF Differential Detector

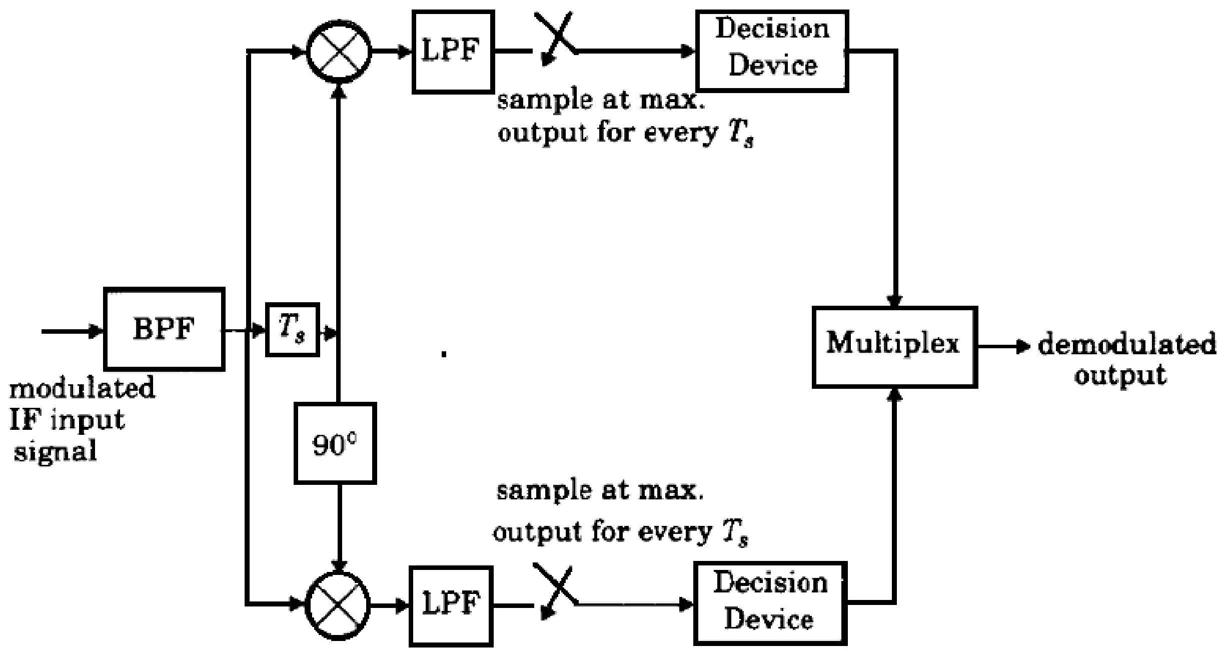


Figure: Block diagram of an IF differential detector for $\frac{\pi}{4}$ QPSK

- ✓ The IF differential detector avoids the need for a local oscillator by using a delay line and two phase detectors is shown in Figure.
- ✓ The received signal is converted to IF and is bandpass filtered.

- ✓ The bandpass filter is designed to match the transmitted pulse shape, so that the carrier phase is preserved and noise power is minimized.
- ✓ To minimize the effect of ISI and noise, the bandwidth of the filters is chosen to be $0.57/T_S$.
- ✓ The received IF signal is differentially decoded using a delay line and two mixers.
- ✓ The bandwidth of the signal at the output of the differential detector is twice that of the baseband signal at the transmitter end.

FM Discriminator

- ✓ A block diagram of an FM discriminator detector for $\frac{\pi}{4}$ QPSK is shown in figure.

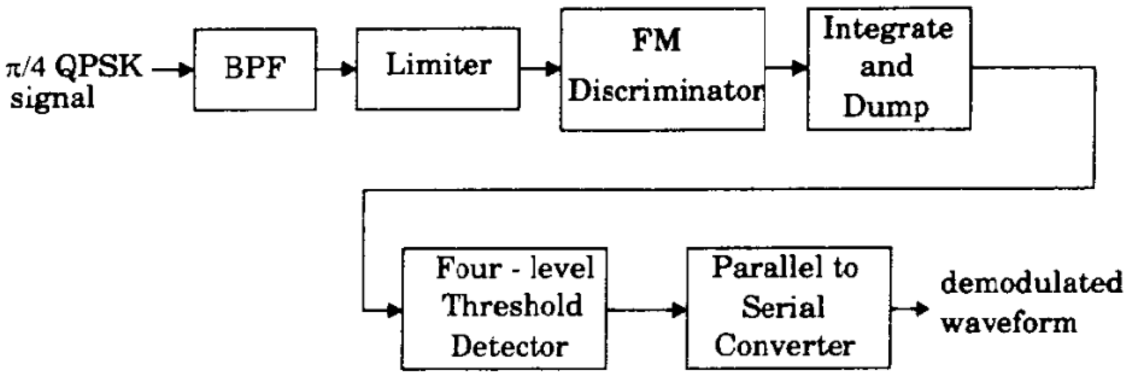


Figure: FM discriminator detector for $\frac{\pi}{4}$ QPSK demodulation

- ✓ The input signal is first filtered using a bandpass filter that is matched to the transmitted signal.
- ✓ The filtered signal is then hard limited to remove any envelope fluctuations.
- ✓ Hard limiting preserves the phase changes in the input signal and hence no information is lost.
- ✓ The FM discriminator extracts the instantaneous frequency deviation of the received signal which, when integrated over each symbol period gives the phase difference between two sampling instants.
- ✓ The phase difference is then detected by a four level threshold comparator to obtain the original signal.
- ✓ The phase difference can also be detected using a modulo-2π phase detector.
- ✓ The modulo-2π phase detector improves the BER performance and reduces the effect of click noise.

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3.4 MINIMUM SHIFT KEYING (MSK) / CONTINUOUS PHASE FREQUENCY SHIFT KEYING (CPFSK) / FAST FSK

1. **What is MSK? Also derive the expression of MSK signal as a special type of FSK signal and explain its spectral density. (16M-Nov 2016)**
2. **Derive the expression for MSK as a special type of FSK signal. (16M-Nov 2015)**
3. **With block diagram, explain the MSK transmitter and receiver. Derive an expression for MSK and its power spectrum. (10M-May 2016) (10M-Nov 2012)**
4. **Derive the expression for MSK signal as a special type of continuous phase FSK signal.**

(16M-Nov 2015)

5. **How MSK signals are generated. Explain in detail. (8M-May 2015)**
6. **Discuss in detail the demodulation techniques for Minimum shift keying. (8M-May 2015)**
7. **What is MSK? Explain its power spectral density. (8M-Nov 2014)**
8. **Explain the principle of Minimum shift keying (MSK) modulation and derive the expression for power spectral density. (8M-May 2013)[Nov/Dec 2017]**
9. **Discuss in detail any two demodulation techniques of minimum shift keying method. (8M-Nov 2011)**
10. **Explain the concept of minimum shift keying and Gaussian MSK. (16m-April 2010]**

Concept of MSK:

- ✓ Minimum shift keying (MSK) is a special type of continuous phase frequency shift keying (CPFSK).
- ✓ In MSK, the peak frequency deviation is equals to 1/4 the bit rate.
- ✓ Modulation index of MSK is given by

$$k_{FSK} = \left(2\Delta f\right) / R_b = 0.5$$

where $\Delta f \rightarrow$ Peak RF frequency deviation

$R_b \rightarrow$ Bit rate

- ✓ A modulation index of 0.5 corresponds to the minimum frequency spacing that allows two FSK signals to be coherently orthogonal,
- ✓ Two FSK signal $v_H(t)$ and $v_L(t)$ are said to be orthogonal if

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

- ✓ MSK is called as fast FSK since the frequency spacing used is only half as much as that used in conventional non-coherent FSK.
- ✓ MSK is a spectrally efficient modulation scheme.
- ✓ It possesses properties such as
- ✓ Constant envelope
  - Spectral efficiency
  - Good BER performance
  - Self-synchronizing capability
- ✓ An MSK signal is a special form of OQPSK where the baseband rectangular pulses are replaced with half-sinusoidal pulses.
- ✓ If half-sinusoidal pulses are used instead of rectangular pulses in OQPSK signal, the modified signal is defined as MSK
- ✓ N-bit stream of MSK signal is given by

$$s_{MSK} = \sum_{i=0}^{N-1} m_I(t)\ p(t-2iT_b)\ \cos 2\pi f_c t\ +\ \sum_{i=0}^{N-1} m_Q(t)\ p(t-2iT_b-T_b)\ \sin 2\pi f_c t$$

where $p(t) = \begin{cases} \sin\left(\dfrac{\pi t}{2T_b}\right) & 0 \le t \le 2T_b \\ 0 & elsewhere \end{cases}$

$m_I(t)$→ Odd bits of the bipolar data stream with $+1$ value and feed the in-phase arms of the modulator at a rate of $R_b/2$.

$m_Q(t)$→ Even bits of the bipolar data stream with-1 value and which feed the quadrature arms of the modulator at a rate of $R_b/2$.

- ✓ The MSK waveform is rewritten using trigonometric identities as

$$s_{MSK} = \sqrt{\frac{2E_b}{T_b}} \cos\left[2\pi f_c t - m_I(t) m_Q(t) \frac{\pi t}{2T_b} + \Phi_k\right]$$

where Φ_k is 0 or 1 depending on wheher $m_I(t)$ is 1or -1.

- ✓ MSK has constant amplitude.
- ✓ Phase continuity at the bit transition periods is ensured by choosing the carrier frequency to be an integral multiple of one fourth the bit rate, 1/4T.
- ✓ MSK signal is an FSK signal with binary signaling frequencies of $f_c + 1/4T$ and $f_c - 1/4T$.
- ✓ Phase of the MSK signal varies linearly during each bit period.

MSK Power Spectrum

- ✓ The RF power spectrum is obtained by frequency shifting the magnitude squared of the Fourier transform of the baseband ***pulse shaping*** function.
- ✓ For MSK, the baseband pulse shaping function is given by

$$p(t) = \begin{cases} \sin\left(\dfrac{\pi t}{2T}\right) & |t| < T \\ 0 & elsewhere \end{cases}$$

- ✓ Thus the normalized power spectral density for MSK is given by

$$P_{MSK}(f) = \frac{16}{\pi^2}\left(\frac{\cos 2\pi(f + f_c)T}{1.16 f^2 T^2}\right)^2 + \frac{16}{\pi^2}\left(\frac{\cos 2\pi(f - f_c)T}{1.16 f^2 T^2}\right)^2$$

- ✓ Figure shows the power spectral density of an MSK signal.

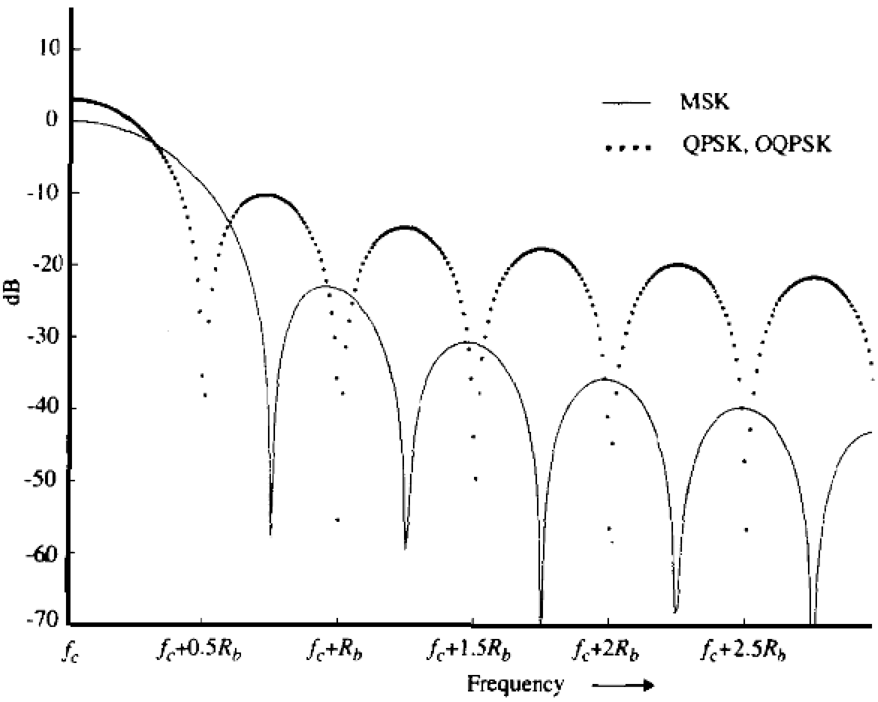

Figure: Power spectral density of MSK signals

- ✓ 99% of the MSK power is contained within a bandwidth $B = 1.2/T$.
- ✓ The faster rolloff of the MSK spectrum is due to the fact that smoother pulse functions are used.
- ✓ MSK spectrum has lower side lobes than QPSK and OQPSK.
- ✓ The main lobe of MSK is wider than that of QPSK and OQPSK.
- ✓ In terms of null bandwidth, MSK is less spectrally efficient than the PSK techniques.
- ✓ Since there is no abrupt change in phase at bit transition periods, band limiting the MSK signal to meet required out-of-band specifications does not cause the envelope to go through zero.
- ✓ The envelope is kept more or less constant even after band limiting.
- ✓ Any small variations in the envelope level can be removed by band limiting at the receiver without raising the out-of-band radiation levels. Since the amplitude is kept constant.
- ✓ MSK signals can be amplified using efficient nonlinear amplifiers.
- ✓ The continuous phase property makes it highly desirable for highly reactive loads.
- ✓ MSK has simple demodulation and synchronization Circuits.

MSK Transmitter

- ✓ A typical MSK modulator is shown.

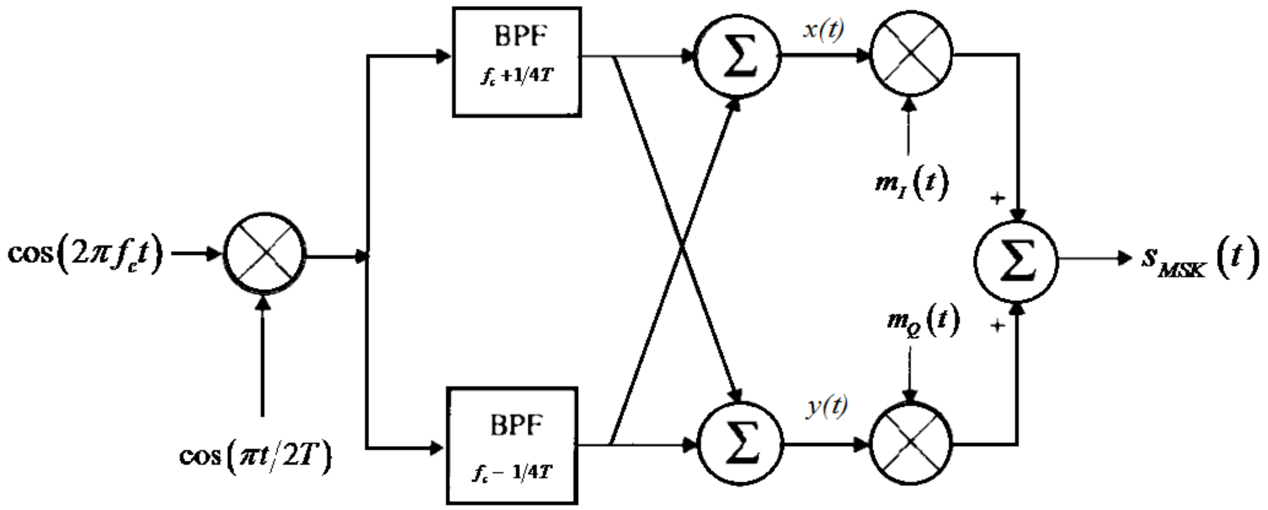

Figure: Block diagram of an MSK transmitter

- ✓ Multiplying a carrier signal with $\cos(\pi t/2T)$ produces two phase-coherent signals at $f_c + 1/4T$ and $f_c - 1/4T$.
- ✓ The two FSK signals are separated using two narrow bandpass filters and appropriately combined to form the in-phase and quadrature carrier components *x(t)* and *y(t)*, respectively.
- ✓ Carriers are multiplied with the odd and even bit streams, $m_I(t)$ and $m_Q(t)$, to produce the MSK modulated signal $s_{MSK}(t)$.

MSK Receiver

- ✓ The block diagram of an MSK receiver is shown in Figure.

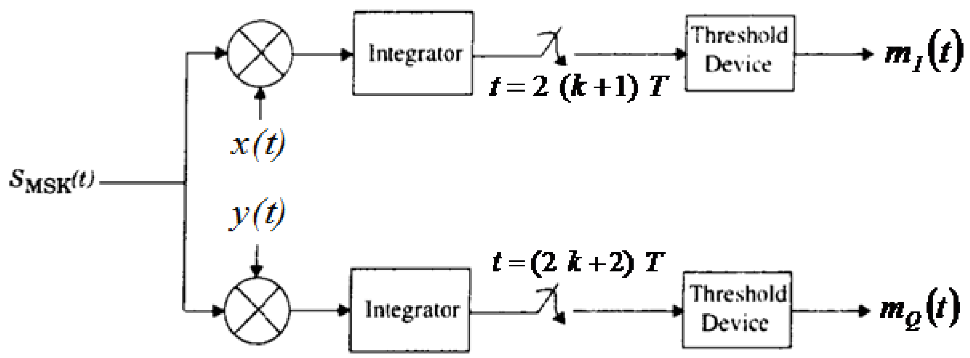

Figure: Block diagram of an MSK receiver

- ✓ The received signal $s_{MSK}(t)$ is multiplied by the respective in-phase and quadrature carriers *x(t)* and *y(t)*.
- ✓ The output of the multipliers are integrated over two bit periods and dumped to a decision circuit at the end of each two bit periods.
- ✓ Based on the level of the signal at the output of the integrator, the threshold detector decides whether the signal is a *0* or a *1*.

- ✓ The output data streams correspond to $m_I(t)$ and $m_Q(t)$, which are offset combined to obtain the demodulated signal.

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3.5 GAUSSIAN MINIMUM SHIFT KEYING (GMSK)

1. **Explain in detail Gaussian Minimum shift Keying (GMSK) transmission and reception with necessary diagrams. (16M-May 2016) (10M-Nov 2015)**
2. **Explain with neat diagram, the principle of Gaussian Minimum shift Keying (GMSK) receiver and mention how it is different from MSK. (16M-May 2014)**
3. **Mention the advantages of GMSK and mention its advantages. (8M-May 2012)**
4. **Explain the concept of minimum shift keying and Gaussian MSK. (16m-April 2010]**
5. **With block diagram, explain the MSK transmitter and receiver. Derive an expression for MSK and its power spectrum.(Nov /Dec 2012)**

Concept of Gaussian Minimum Shift Keying

- ✓ GMSK is a simple binary modulation scheme.
- ✓ GMSK is a derivative of MSK.
- ✓ The sidelobe levels of the spectrum are further reduced by passing the modulating NRZ data waveform through a premodulation Gaussian pulse-shaping filter.
- ✓ Gaussian pulse shaping to MSK
  - smoothens phase trajectory of MSK signal → over time, stabilizes instantaneous frequency variations
  - results in significant additional **reduction** of sidelobe levels
- ✓ GMSK detection can be coherent (like MSK) or noncoherent (like FSK)
- ✓ Premodulation pulse shaping filter used to filter NRZ data
  - converts full response message signal into partial response scheme
    Full response → baseband symbols occupy T_b
    Partial response → transmitted symbols span several T_b
  - Pulse shaping doesn't cause pattern's averaged phase trajectory to deviate from simple MSK trajectory

Gaussian MSK

- ✓ Impulse response of pre-modulation Gaussian filter is given by

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

- ✓ Transfer function of pre-modulation Gaussian Filter is given by

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

- ✓ The parameter α is related to B_{3dB} of $H_G(f)$, by

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

Power spectral density of GMSK Signals

- ✓ GMSK filter is defined from baseband bandwidth, B and the baseband symbol duration, T.
- ✓ Increasing BT_b
  - reduces signal spectrum
  - results in temporal spreading and distortion
- ✓ Figure shows the simulated RF power spectrum of the GMSK signal for various values of BT

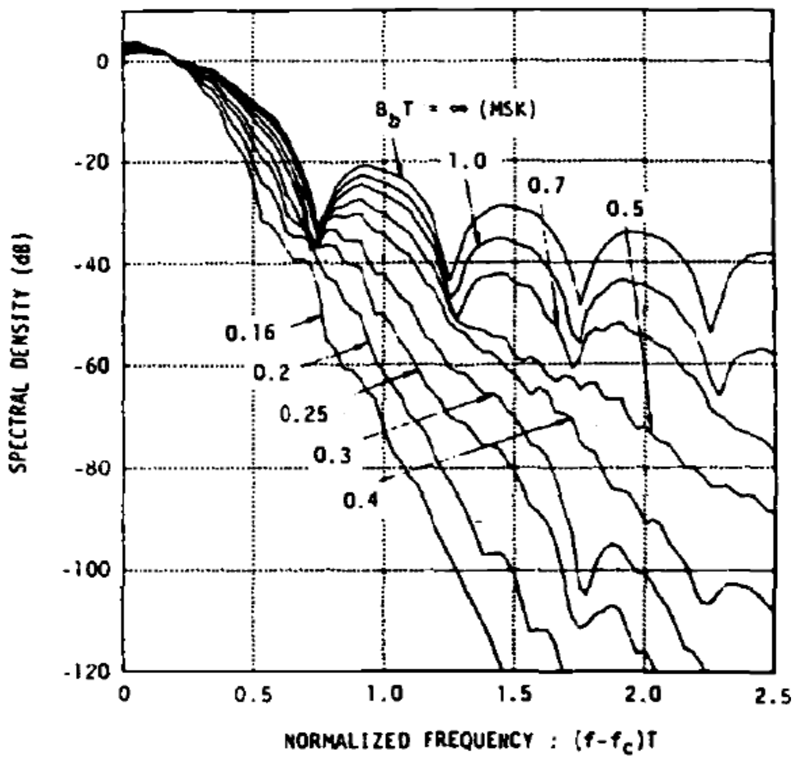

Figure: Power spectral density of a GMSK signal

Impact of $B_{3dB}{\cdot}T_b$

- ✓ $B_{3dB}{\cdot}T_b$ product decreases, spectrum becomes more compact (spectral efficiency)
  - causes sidelobes of GMSK to fall off rapidly

    $B_{3dB}{\cdot}T_b = 0.5$ → GMSK → 2nd lobe peak is 30dB below main lobe

    MSK → 2nd peak lobe is 20dB below main lobe
  - MSK ≈ GMSK with $B_{3dB}{\cdot}T_b = \infty$
- ✓ Reducing $B_{3dB}{\cdot}T_b$, increases irreducible error rate (IER) due to ISI
  - ISI degradation caused by pulse shaping increases
  - Mobile channels induce IER due to mobile's velocity
  - If GMSK IER < mobile channel IER → no penalty for using GMSK
- ✓ Table shows occupied bandwidth containing a given percentage of power in a GMSK signal as a function of the BT product.

BT	90%	99%	99.9%	99.99%
0.2 GMSK	0.52	0.79	0.99	1.22
0.25 GMSK	0.57	0.86	1.09	1.37
0.5 GMSK	0.69	1.04	1.33	2.08
MSK	0.78	1.20	2.76	6.00

Occupied RF Bandwidth containing a Given Percentage of Power

- ✓ e.g. for BT = 0.2 → 99% of the power is in the bandwidth of $1.22R_b$
- ✓ While the GMSK spectrum becomes more and more compact with decreasing BT value, the degradation due to ISI increases.
- ✓ BER degradation due to ISI caused by filtering is minimum for a BT value of 0.5887, where the degradation in the required E_b/N_0 is only 0.14 dB from the case of no ISI.

Advantages of GMSK

- ✓ GMSKs main advantages are
  - Power efficiency - from constant envelope (non-linear amplifiers)
  - Excellent spectral efficiency

Advantages of GMSK over MSK:

- ✓ GMSK is a derivative of MSK where the bandwidth required is further reduced by passing the modulating waveform through a Gaussian filter.
- ✓ The Gaussian filter minimizes the instantaneous frequency variations over time.
- ✓ GMSK is a spectrally efficient modulation scheme and it's particularly useful in mobile radio systems.
- ✓ It has a constant envelop spectrally efficient, good BER performance and self-synchronizing Baseband

GMSK Bit Error Rate (BER) for AWGN channel

- ✓ GMSK performs within 1dB of optimal MSK with $B_{3dB}T_b$ = 0.25
- ✓ The pulse shaping impacts ISI.
- ✓ The bit error probability, P_e is a function of $B_{3dB}T_b$.

$$P_e = Q\left(\sqrt{\frac{2\lambda E_b}{N_0}}\right)$$

Where P_e → Bit error probability

λ → Constant related to $B_{3dB}T_b$

GMSK Transmitter

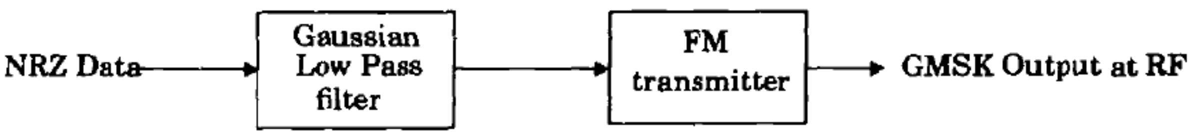

Figure: Block diagram of a GMSK transmitter using direct FM generation

- ✓ The simplest way to generate a GMSK signal is to pass a NRZ message bit stream through a Gaussian baseband filter followed by an FM modulator.

GMSK Receiver

- ✓ GMSK signals can be detected using orthogonal coherent detectors as shown in Figure, or with simple non coherent detectors such as standard FM discriminators.

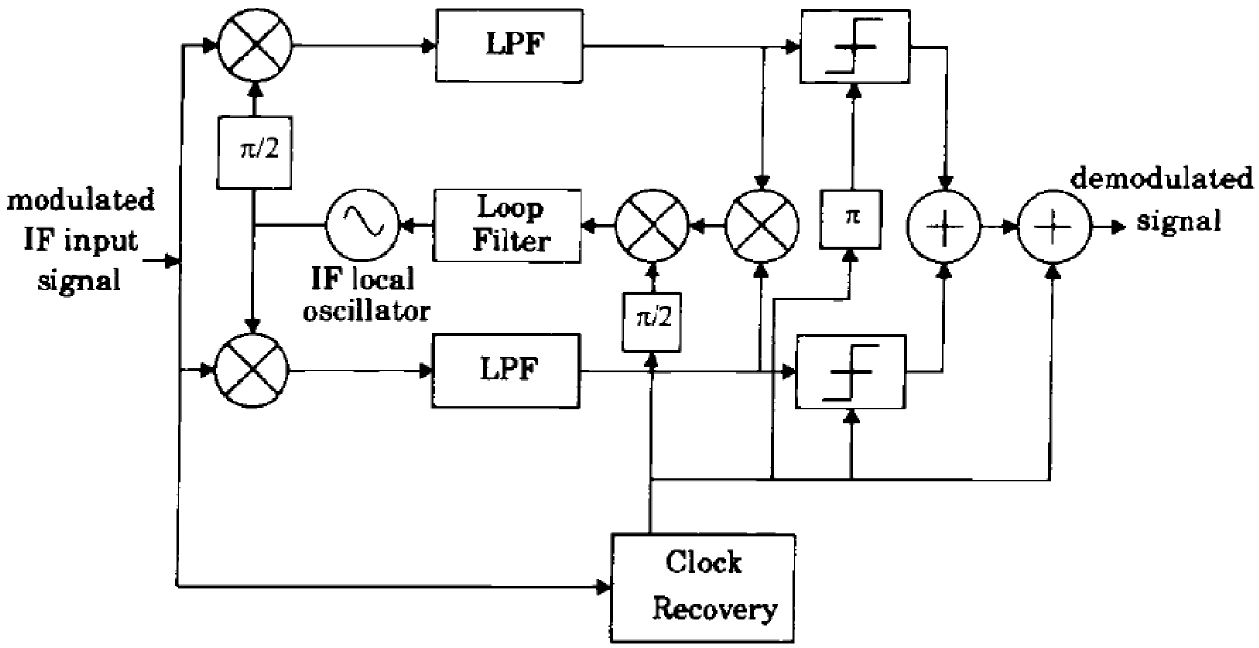

Figure: Block diagram of a GMSK receiver

- ✓ Carrier recovery is sometimes performed using a method where the sum of the two discrete frequency components contained at the output of frequency doublers is divided by four.
- ✓ De Buda's method receiver can be easily implemented using digital logic as shown in Figure.

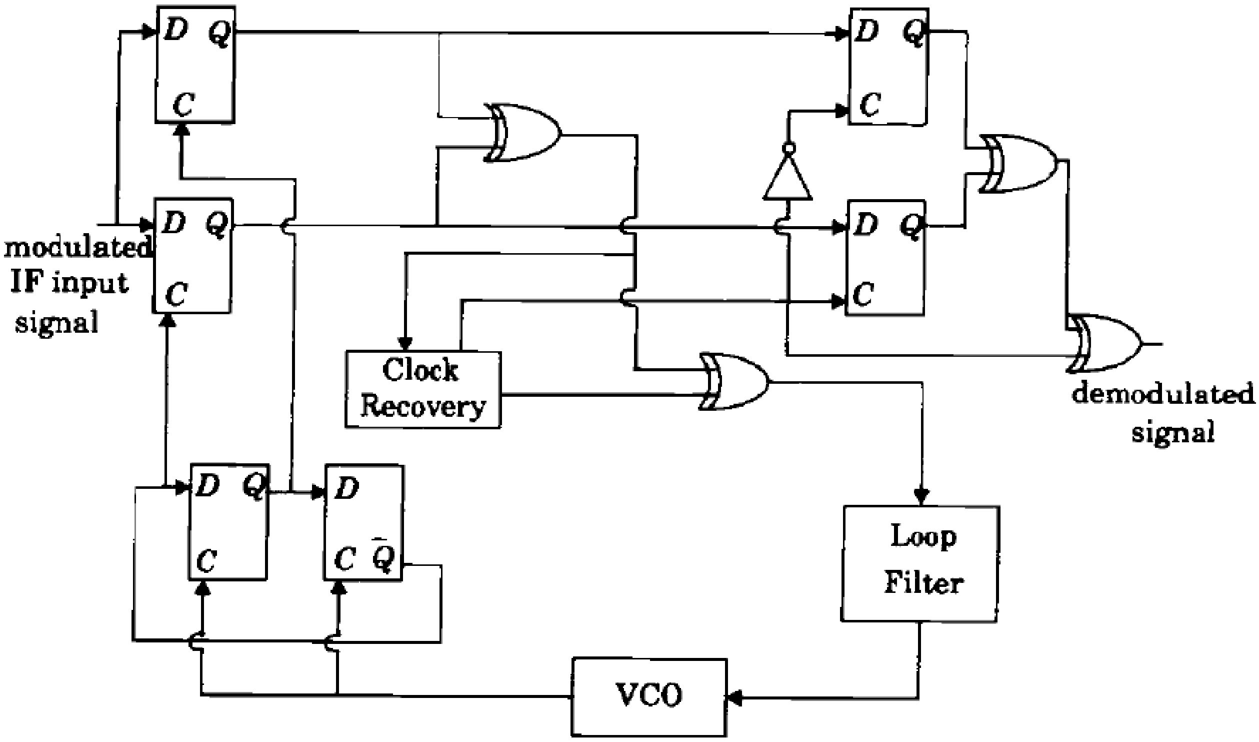

Figure: Digital logic circuit for GMSK demodulation

- ✓ 2 D flip flops (DFF) act as quadrature product demodulator
- ✓ XORs act as based band multipliers
- ✓ mutually orthogonal reference carriers are generated using two D flip-flops
- ✓ VCO center frequency is set to $4 \times f_c$ (f_c = carrier center frequency)
- ✓ Detecting GMSK signal by sampling output of FM demodulator is a non-optimal, effective method

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3.6 ERROR PERFORMANCE IN FADING CHANNELS

1. **Explain the performance of digital modulation in slow flat-fading channels. (8M-Nov 2013)**
2. **Derive the expression for the probability of error in flat-fading channels. (8M-May 2013)**
3. **Derive the expression for the probability of error in Frequency-dispersive fading channels. (8M-May 2013)**
4. **Discuss the error performance of different modulation schemes in fading channels. (May/June 2016)**
5. **Derive the bit error rate for binary phase shift keying modulation for frequency flat fading channels. [April/May 2018]**

Introduction

- ✓ The mobile radio channel is affected by many factors such as
  - Fading
  - multipath
  - Interference.

Performance of the modulation schemes

- ✓ BER gives a good indication of the performance of a particular modulation schemes. But it does not provide enough information.

- ✓ Probability of error is another way to judge the effectiveness of the signaling scheme in a mobile radio channel.
- ✓ BER and probability of error for many modulation schemes under various types of channel impairments can be evaluated either through analytical techniques or through simulation.

<u>Performance of the Digital modulation in slow, flat fading channels</u>

- ✓ Flat fading channels cause a multiplicative variation in the transmitted signal $s(t)$.
- ✓ Slow, flat fading channels change much slower than the applied modulation.
- ✓ It can be assumed that the attenuation and phase shift of the signal is constant over at least one symbol interval.
- ✓ The received signal r(t) may be expressed as

$$r(t) = \alpha(t)e^{-j\theta(t)}s(t) + n(t) \qquad 0 \le t \le T$$

where $\alpha(t)$ → Gain of the channel

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

$n(t)$ → Additive Gaussian noise

- ✓ To evaluate the probability of error of any digital modulation scheme in a slow, flat fading channel, it has to average the probability of error of the particular modulation in AWGN (Additive white Gaussian Noise Channel) channels.
- ✓ The probability of error in AWGN channels is viewed as a conditional error probability, where the condition is α is fixed.
- ✓ AWGN is the simplest practical case of the mobile radio.
- ✓ The received signal is the sum of the transmitted signal and Gaussian noise.

<u>Probability of error</u>

- ✓ The probability of error in slow, flat fading channels can be obtained by averaging the error in AWGN channels over the fading probability density function.
- ✓ The probability of error in a slow, Flat fading channel can be evaluated as

$$P_e = \int_0^\infty P_e(X)p(X)dX$$

$$X = \alpha^2 \frac{E_b}{N_0}$$

where $P_e(X)$ → probability of error for an arbitrary modulation in AWGN

$p(X)$ → probability density function of X due to the fading channel

α → Amplitude values of the fading channel

<u>Probability of error for some modulation scheme in an AWGN channel</u>

- ✓ Digital modulation
- ✓ Coherent

$$P_{e,BPSK} = Q\left(\sqrt{2\frac{E_b}{N_0}}\right)$$

$$P_{e,FSK} = Q\left(\sqrt{\frac{E_b}{N_0}}\right)$$

$$P_{e,DPSK} = \frac{1}{2}e^{\left(\frac{-E_b}{N_0}\right)}$$

- Non-Coherent $$P_{e,NCFSK} = \frac{1}{2} e^{\left(\frac{-E_b}{2N_0}\right)}$$

Probability density function (PDF) of X in Rayleigh Fading Channel

- Rayleigh Probability density function (PDF) is given by

$$P(t) = \frac{r}{\sigma^2} e^{\left(\frac{-r^2}{2\sigma^2}\right)} \qquad 0 \le r \le \infty$$

- When input signal at a receiver is Rayleigh

$$X = \frac{Signal\ Power}{Noise\ Power} - \frac{A^2 r^2}{2.P_n}$$

- Let α be a complex Gaussian random variable

$$\alpha = x + yi$$

$$|\alpha| - r$$

σ → Standard Deviation

- The average SNR for the channel Γ is the mean of X

$$= \frac{A^2 E(r^2)}{2.P_n} - \frac{A^2 \sigma^2}{P_n}$$

- To find the distribution of r, we use the identity

$$P(X) = P(r)\frac{dr}{dx}$$

$$= \frac{r}{\sigma^2} e^{\left(\frac{-r^2}{2\sigma^2}\right)} \frac{P_n}{(A^2 r)}$$

where $\Gamma = \frac{A^2\sigma^2}{P_n}$, the average value of the SNR.

- The Final result for the PDF of X

$$P(X) = \frac{1}{\Gamma} e^{\left(\frac{-X}{\Gamma}\right)} \qquad X \ge 0$$

Performance of the Digital modulation in slow, flat fading channels

- Probability of error for BPSK (Coherent) in the AWGN channel.

$$P(X) = \frac{1}{\Gamma} e^{\left(\frac{-X}{\Gamma}\right)}$$

$$P_{e,PSK} = Q\left(\sqrt{\frac{2E_b}{N_0}}\right) = Q\left(\sqrt{2X}\right)$$

- Substituting two equations above in the following equation

$$P_e = \int_0^\infty P_e(X) p(X) dX - \int_0^\infty \frac{1}{\Gamma} e^{\left(\frac{-X}{\Gamma}\right)} Q\sqrt{2X}\, dX$$

$$P_{e,\mathrm{P}SK} = \frac{1}{2}\left[1 - \sqrt{\frac{\frac{E_b}{N_0}\overline{\alpha^2}}{1+\frac{E_b}{N_0}\overline{\alpha^2}}}\right]$$

$$= \frac{1}{2}\left[1 - \sqrt{\frac{\Gamma}{1+\Gamma}}\right] \qquad \Gamma = \frac{E_b}{N_0}\overline{\alpha^2}$$

Probability of error for other modulation techniques

- ✓ $P_{e,\mathrm{P}SK} = \frac{1}{2}\left[1 - \sqrt{\frac{\frac{E_b}{N_0}\overline{\alpha^2}}{1+\frac{E_b}{N_0}\overline{\alpha^2}}}\right] = \frac{1}{2}\left[1 - \sqrt{\frac{\Gamma}{1+\Gamma}}\right]$ (Coherent binary PSK)

- ✓ $P_{e,\mathrm{F}SK} = \frac{1}{2}\left[1 - \sqrt{\frac{\frac{E_b}{N_0}\overline{\alpha^2}}{2+\frac{E_b}{N_0}\overline{\alpha^2}}}\right] = \frac{1}{2}\left[1 - \sqrt{\frac{\Gamma}{2+\Gamma}}\right]$ (Coherent binary FSK)

- ✓ $P_{e,\mathrm{DP}SK} = \frac{1}{2\left(1+\frac{E_b}{N_0}\overline{\alpha^2}\right)} = \frac{1}{2(1+\Gamma)}$ (Differential binary PSK)

- ✓ $P_{e,\mathrm{NCF}SK} = \frac{1}{2+\frac{E_b}{N_0}\overline{\alpha^2}} = \frac{1}{2+\Gamma}$ (noncoherent orthogonal binary FSK)

- ✓ $P_{e,GMSK} = \frac{1}{2}\left(1 - \sqrt{\frac{\delta\Gamma}{\delta\Gamma+1}}\right)$ (Coherent GMSK)

  where $\delta \cong \begin{cases} 0.68 & BT = 0.25 \\ 0.85 & BT = \infty \end{cases}$

Performance of the Digital modulation in slow, flat fading channels

- ✓ For large values of signal-to-noise ratio, the error probability equations may be simplified as

$$P_{e,\mathrm{P}SK} == \frac{1}{2}\left[1 - \sqrt{\frac{\Gamma}{1+\Gamma}}\right] \qquad \text{(Coherent binary PSK)}$$

Approximated $P_{e,\mathrm{P}SK} = \frac{1}{4\Gamma}$ (Coherent binary PSK)

- ✓ For large values of signal-to-noise ratio the error probability equations can be simplified as

Original equation		**Approximated Equation**
✓ $P_{e,\mathrm{P}SK} = \frac{1}{2}\left[1 - \sqrt{\frac{\Gamma}{1+\Gamma}}\right]$	→	$P_{e,\mathrm{PSK}} = \frac{1}{4\Gamma}$

- ✓ $P_{e,FSK} = \frac{1}{2}\left[1 - \sqrt{\frac{\Gamma}{2+\Gamma}}\right]$ → $P_{e,\mathrm{FSK}} = \frac{1}{2\Gamma}$
- ✓ $P_{e,DPSK} = \frac{1}{2(1+\Gamma)}$ → $P_{e,\mathrm{DPSK}} = \frac{1}{2\Gamma}$
- ✓ $P_{e,NCFSK} = \frac{1}{2+\Gamma}$ → $P_{e,\mathrm{NCFSK}} = \frac{1}{\Gamma}$
- ✓ $P_{e,GMSK} = \frac{1}{2}\left(1 - \sqrt{\frac{\delta\Gamma}{\delta\Gamma+1}}\right)$ → $P_{e,GMSK,approx} = \frac{1}{4\delta\Gamma}$

$$\text{where} \quad \delta \cong \begin{cases} 0.68 & BT = 0.25 \\ 0.85 & BT = \infty \end{cases}$$

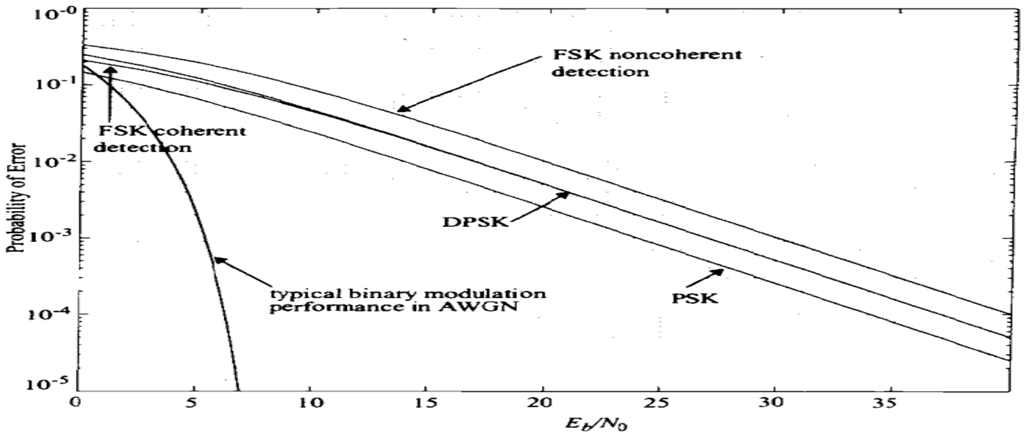


Figure: Bit error rate performance of binary modulation schemes in a Rayleigh flat fading channel

- ✓ For fading Channels, Bit error Rate has simply an inversely relation with average SNR.
- ✓ For AWGN Gaussian Channels, Bit error Rate falls exponentially

Digital Modulation in Frequency Selective Mobile Channels

- ✓ Frequency selective fading caused by multipath time delay spread.
- ✓ Multipath time delay spread causes intersymbol interference, which results in an irreducible BER floor.
- ✓ The irreducible error occurs when
  1. The main (undelayed) signal component is removed through multipath cancellation
  2. non-zero value of d $\left(= \sigma_\tau / T_s\right)$ causes ISI
  3. Sampling time of a receiver is shifted as a result of delay spread.
- ✓ QPSK, OQPSK and MSK are more resistant to delay spread than BPSK.

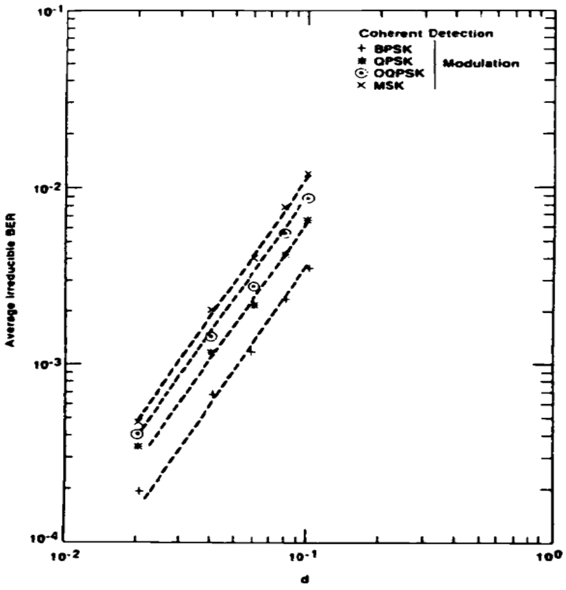
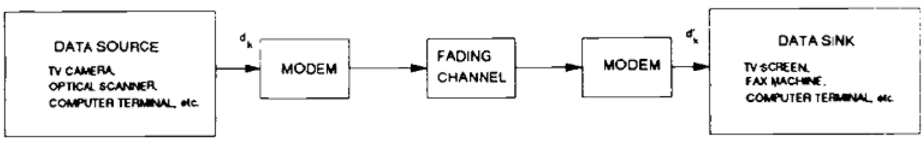


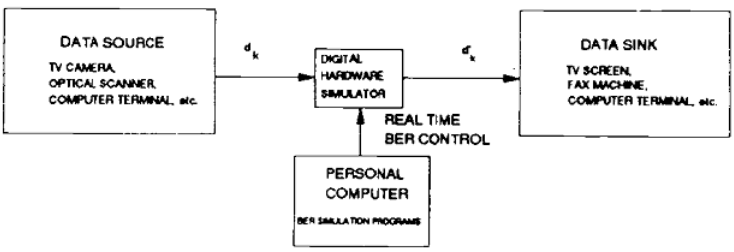
Figure: The irreducible BER performance for different modulations Schemes

Performance of π/4 DQPSK in Fading and Interference

- ✓ BER performance of $\pi / 4$ DQPSK can be studied by using computer simulations.
- ✓ BER can be calculated and analyzed as a function of the following parameters:
  - ➢ The Doppler spread normalized to the symbol rate $B_D T_S$ or B_D / R_S .
  - ➢ Delay of the second multipath τ, normalized to the symbol duration: τ/T
  - ➢ Energy to noise Ratio: Eb I N 0 dB
  - ➢ Average carrier to interference power ratio in decibels: C/I dB
  - ➢ Average main-path to delayed-path power ratio : C/D dB



(a)



(b)

Figure: BERSIM (Bit Error Rate SIMulator)

P(e) vs. C/N in slowly flat fading channels

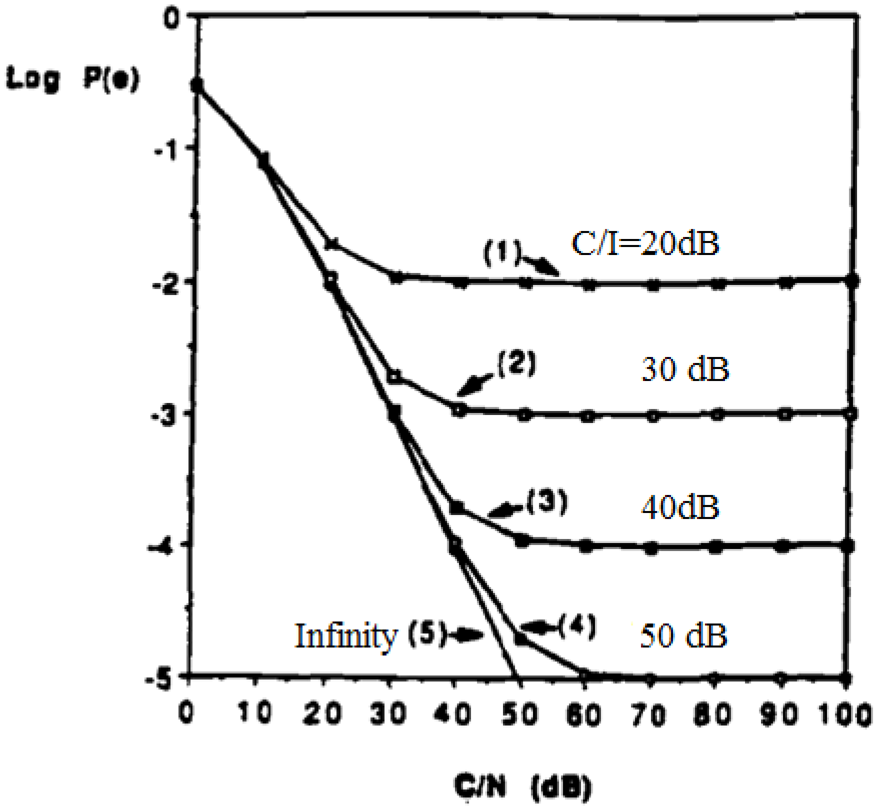


Figure: BER performance of π/4 DQPSK Vs C/N (dB)

- ✓ For C/I >20 dB, the errors are primarily due to fading
- ✓ When C/I drop to below 20 dB, interference affects the link performance.

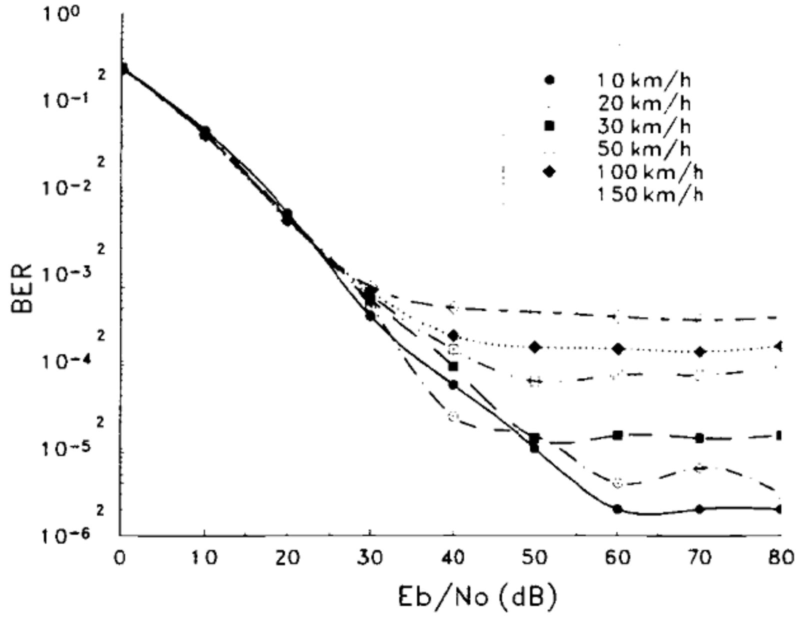


Figure: BER performance of π/4 DQPSK Vs SNR (dB)

- ✓ When velocity increases, the irreducible error floor increases.

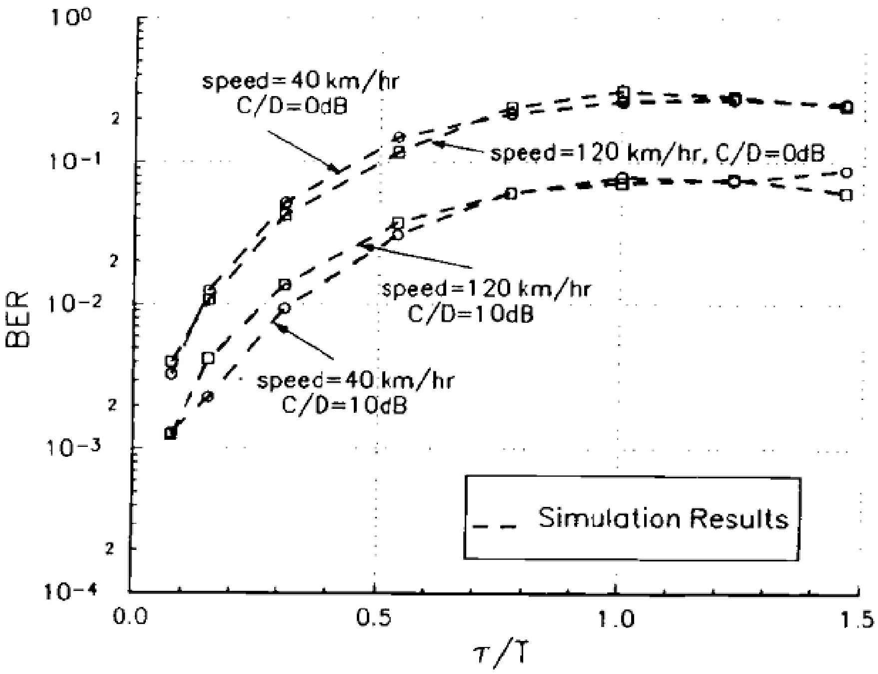


Figure: BER performance of π/4 DQPSK Vs τ/T

- ✓ The delay and amplitude of the second ray have strong impact on the average.

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3.7 OFDM PRINCIPLE

1. **Draw the basic arrangements of orthogonal frequency division Multiplexing transceivers and discuss its overall operation. (16M-Nov 2016)**
2. **Explain about OFDM principle and operation. (Nov/Dec 2015)**
3. **Describe OFDM scheme and state the reason behind using cyclic prefix in OFDM scheme.**

Concept of OFDM

- ✓ Orthogonal Frequency Division Multiplexing (OFDM) is a modulation scheme that is especially suited for high-data-rate transmission in delay-dispersive environments.
- ✓ OFDM converts a high-rate data stream into a number of low-rate streams that are transmitted over parallel, narrowband channels that can be easily equalized.
- ✓ OFDM is used for
  - ✓ Digital Audio Broadcasting (DAB)
  - ✓ Digital Video Broadcasting (DVB)
  - ✓ Wireless Local Area Networks (LANs) (IEEE 802.11a, IEEE 802.11g)
- ✓ OFDM splits a high-rate data stream into N parallel streams, which are then transmitted by modulating N distinct carriers called subcarriers or tones.
- ✓ Symbol duration on each subcarrier thus becomes larger by a factor of N.

- ✓ In order for the receiver to be able to separate signals carried by different subcarriers, they have to be orthogonal.
- ✓ Conventional Frequency Division Multiple Access (FDMA) wastes precious spectrum.
- ✓ Narrower spacing of subcarriers can be achieved.
- ✓ Frequencies of the subcarriers $f_n = nW/N = N/T_S$.

  where n → integer,

  W → Total available bandwidth
- ✓ Assume Pulse Amplitude Modulation (PAM) with rectangular basis pulses on each of the subcarriers.
- ✓ Subcarriers are mutually orthogonal, since the relationship holds

$$\int_{iT_S}^{(i+1)T_S} \exp(j2\pi f_k t)\exp(-j2\pi f_n t)dt = \delta_{nk}$$

- ✓ Figure shows this principle in the frequency domain.

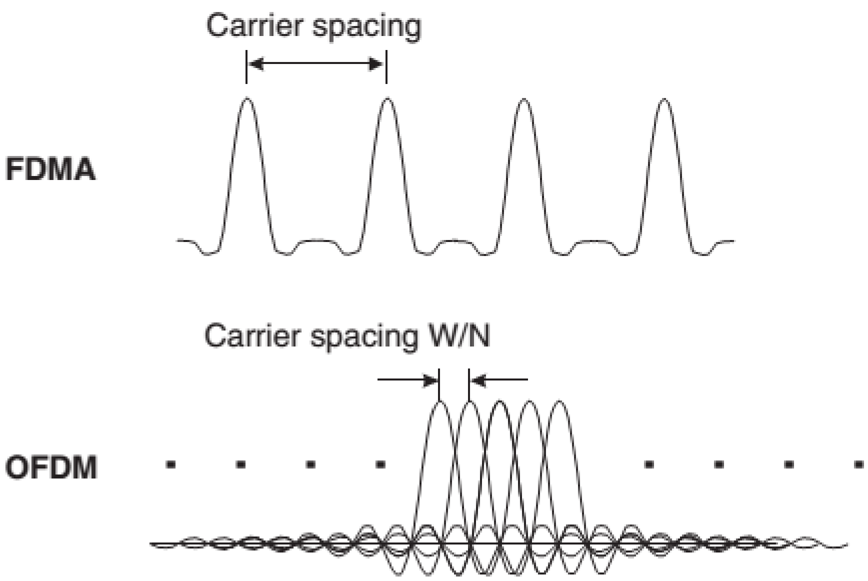

Figure: Principle in the frequency domain

- ✓ Due to the rectangular shape of pulses in the time domain, the spectrum of each modulated carrier has a sin*(x)/x* shape. The spectra of different modulated carriers overlap, but each carrier is in the spectral nulls of all other carriers.
- ✓ Therefore, as long as the receiver does the appropriate demodulation the data streams of any two subcarriers will not interfere.

Implementation of Transceivers

- ✓ OFDM can be interpreted in two ways:
  - Analog implementation
  - Digital implementation

Analog implementation

- ✓ Data stream are split into N parallel data streams, each with lower data rate.

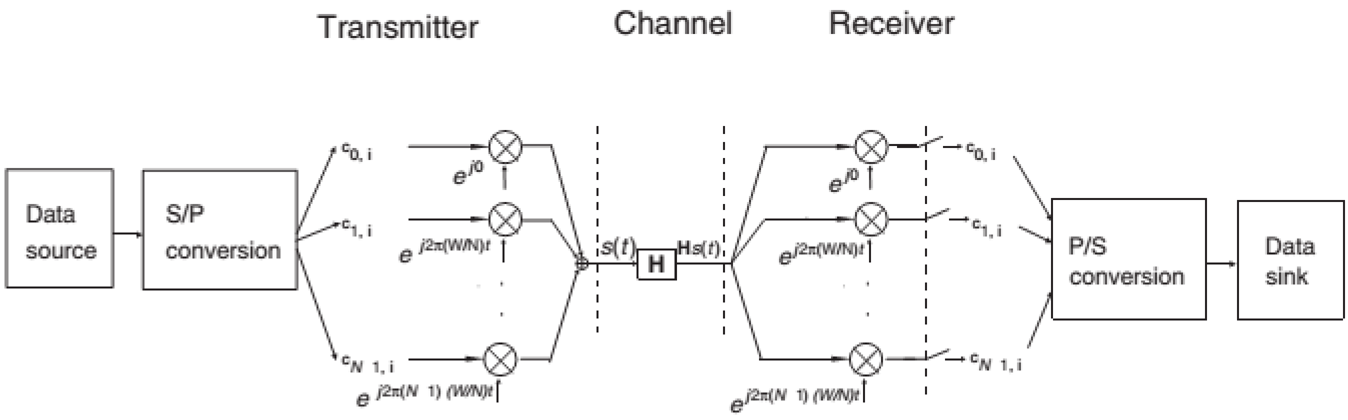

Figure: Transceiver structures for orthogonal frequency division multiplexing in purely analog technology

- ✓ Number of local oscillators (Los) are available,
- ✓ Each of local oscillators oscillates at a frequency $f_n = nW/N$, where n = 0, 1. . . N – 1.
- ✓ Each of the parallel data streams then modulates one of the carriers.
- ✓ Analog implementation of OFDM would require multiple Los, each of which has to operate with little phase noise and drift, in order to retain orthogonality between the different subcarriers.
- ✓ Disadvantage
  - ▪ Hardware implementation of multiple local oscillators is too high.

Digital implementation

- ✓ An alternative implementation is *digital*.
- ✓ Digital implementation divides the transmit data into blocks of N symbols.
- ✓ Each block of data is subjected to an Inverse Fast Fourier Transformation (IFFT), and then transmitted

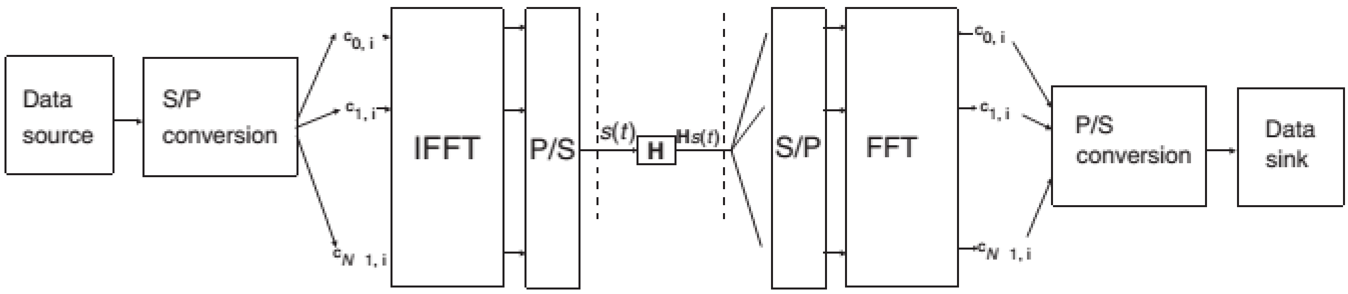

Figure: Transceiver structures for orthogonal frequency division multiplexing using inverse fast Fourier transformation.

- ✓ Input to this IFFT is made up of *N* samples, and therefore the output from the IFFT also consists of *N* values.
- ✓ These *N* values have to be transmitted, one after the other. Thus P/S (Parallel to Serial) conversion is directly after the IFFT.
- ✓ At the receiver, the process is reversed.
- ✓ Sample the received signal, write a block of *N* samples into a vector – i.e., an S/P (Serial to Parallel) conversion – and perform an FFT on this vector. The result is an estimate $\hat{c}_n$ of the original data c_n.

- Advantages
  - Easier to implement with integrated circuits.
  - Implementation of the transceivers is simpler and cheaper.
- OFDM can also be interpreted in the time–frequency plane.
  - Each index *i* corresponds to a (temporal) pulse;
  - Each index *n* to a carrier frequency.
  - This ensemble of functions spans a grid in the time–frequency plane.
- Let us first consider the analog interpretation.
- Let the complex transmit symbol at time instant *i* on the n^{th} carrier be $c_{n,i}$.
- The transmit signal is then:

$$s(t) = \sum_{i=-\infty}^{\infty} s_i(t) = \sum_{i=-\infty}^{\infty}\sum_{n=0}^{N-1} c_{n,i}\, g_n(t - iT_S)$$

- Basis pulse $g_n(t)$ is a normalized, frequency-shifted rectangular pulse given by

$$g_n(t) = \begin{cases} \frac{1}{\sqrt{T_S}} \exp\left(j2\pi \frac{1}{T_S}\right) & 0 < t < T_S \\ 0 & otherwise \end{cases}$$

- consider the signal only for $i = 0$, and sample it at instances $tk = kTs/N$:

$$s_k = S(t_k) = \frac{1}{\sqrt{T_S}} \sum_{n=0}^{N-1} c_{n,0} \exp\left(j2\pi n \frac{k}{N}\right)$$

- Now, this is nothing but the inverse Discrete Fourier Transform (DFT) of the transmit symbols.
- Therefore, the transmitter can be realized by performing an *Inverse Discrete Fourier Transform* (IDFT) on the block of transmit symbols (the blocksize must equal the number of subcarriers).
  - In almost all practical cases, the number of samples N is chosen to be a power of 2, and the IDFT is realized as an IFFT.

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3.8 CYCLIC PREFIX

1. Explain cyclic prefix in detail with necessary diagrams.

2. Prove that the OFDM system converts the delay spread channel into a set of parallel fading channels, using the concept of cyclix prefix. [Nov/Dec 2017]

- Basic function for transmission is given by

$$g_n(t) = \exp\left[j2\pi n \frac{W}{N} t \right] \qquad -T_{cp} < t < \hat{T}_S$$

where W/N is the carrier spacing

$$\hat{T}_S = \frac{N}{W}$$

- Symbol duration T_s

$$T_S = \hat{T}_S + T_{cp}$$

- ✓ For the duration $0 < t < \hat{T}_S$, Normal OFDM symbol is transmitted
- ✓ For the duration $-T_{cp} < t < 0$, a copy of the last part of the symbol is transmitted.
- ✓ This implies that $g_n(t) = g_n(t + N/W)$.
- ✓ This appended part of the signal is called the cyclic prefix.

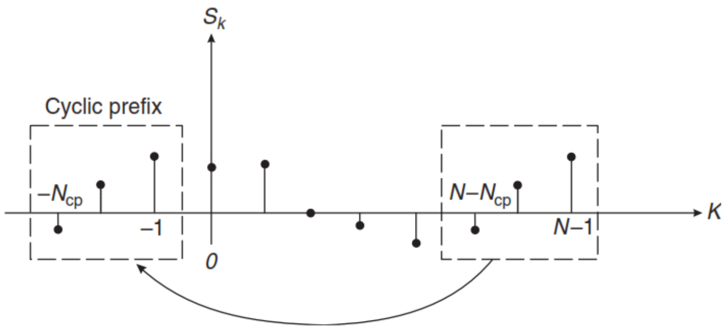

Figure. Principle of the cyclic prefix. $N_{cp} = NT_{cp}/(N/W)$ is the number of samples in the cyclic prefix.

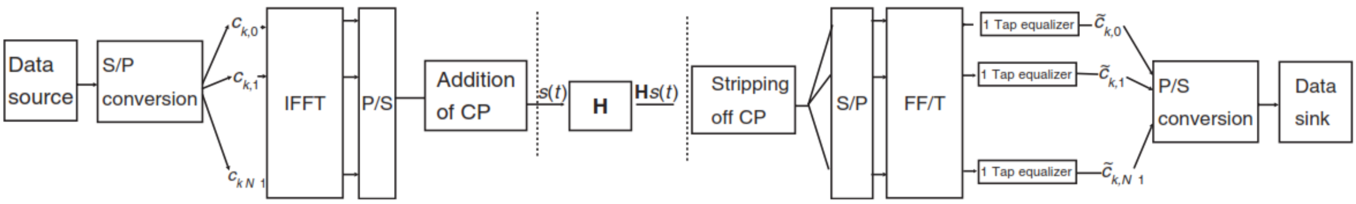

Figure. Structure of an orthogonal-frequency-division-multiplexing transmission chain with cyclic prefix and one-tap equalization.

- ✓ Therefore, during time $-T_{cp} < t < 0$, a copy of the last part of the symbol is transmitted.
- ✓ From linearity, it also follows that the total signal s(t) transmitted during time $-T_{cp} < t < 0$ is a copy of s(t) during the last part, $\hat{T}_S - T_{cp} < \breve{t} < \hat{T}_S$
- ✓ This prepended part of the signal is called the "cyclic prefix."
- ✓ Now that we know what a cyclic prefix is, let us investigate why it is beneficial in delay-dispersive channels.
- ✓ When transmitting any data stream over a delay-dispersive channel, the arriving signal is the linear convolution of the transmitted signal with the channel impulse response.
- ✓ The cyclic prefix converts this *linear* convolution into a *cyclical* convolution.
- ✓ During the time $-T_{cp} < t < -T_{cp} + \tau_{max}$, where τ_{max} is the maximum excess delay of the channel, the received signal suffers from "real" InterSymbol Interference (ISI), as echoes of the last part of the preceding symbol interfere with the desired symbol.
- ✓ This "regular" ISI is eliminated by discarding the received signal during this time interval. During the remainder of the symbol, we have cyclical ISI; especially, it is the last part of the current (not the preceding) symbol that interferes with the first part of the current symbol.

- ✓ In the following, we show how an extremely simple mathematical operation can eliminate the effect of such a cyclical convolution.
  - ✓ For the following mathematical derivation, we assume that the duration of the impulse response is exactly equal to the duration of the prefix; furthermore, in order to simplify the notation, we assume (without restriction of generality) $i = 0$.
  - ✓ In the receiver, there is a bank of filters that are matched to the basis functions without the cyclic prefix:

$$r_{n,0} = \int_0^{\hat{T}_s} \left[\int_0^{T_{cp}} h(t,\tau) \left(\sum_{k=0}^{N-1} c_{k,0} g_k(t-\tau) \right) d\tau \right] g_n^*(t)\, dt + n_n$$

  - ✓ This operation removes the first part of the received signal (of duration T_{cp})from the detection process; as discussed above, the matched filtering of the remainder can be realized as an FFT operation.
  - ✓ The signal at the output of the matched filter is thus convolution of the transmit signal with the channel impulse response and the receive filter:

$$r_{n,0} = \int_0^{\hat{T}_s} \left[\int_0^{T_{cp}} h(t,\tau) \left(\sum_{k=0}^{N-1} c_{k,0} g_k(t-\tau) \right) d\tau \right] g_n^*(t)\, dt + n_n$$

    where n_n is the noise at the output of the matched filter.
  - ✓ If the channel can be considered as constant during the time T_S , then $h(t, \tau) = h(\tau)$, and we obtain:

$$r_{n,0} = \sum_{k=0}^{N-1} c_{k,0} \int_0^{\hat{T}_s} \left[\int_0^{T_{cp}} h(\tau)(g_k(t-\tau))\, d\tau \right] g_n^*(t)\, dt + n_n$$

  - ✓ The inner integral can be written as

$$\exp\left[j2\pi t k \frac{W}{N} \right] \int_0^{T_{cp}} h(\tau) \exp\left(-j2\pi \tau k \frac{W}{N} \right) d\tau = g_k(t) H\left(k \frac{W}{N} \right)$$

  - ✓ The OFDM system is thus represented by a number of parallel *nondispersive*, fading channels, each with its own complex attenuation $H\left(n\frac{W}{N}\right)$.
  - ✓ Equalization of the system thus becomes exceedingly simple: it just required division by the transfer function at the subcarrier frequency, independently for each subcarrier. In other words, the cyclic prefix has recovered the orthogonality of the subcarriers.
  - ✓ Two warnings have to be noted:

    (i) we assumed in the derivation that the channel is static for the duration of the OFDM symbol. If this assumption is not fulfilled, interference between the subcarriers can still occur;

(ii) (ii) discarding part of the received signal decreases the Signal-to-Noise Ratio (SNR), as well as spectral efficiency. For usual operating parameters (cyclic prefix about 10% of symbol duration), this loss is tolerable.

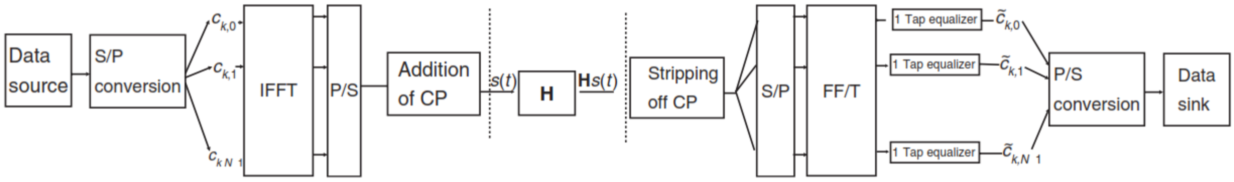

Figure. Structure of an orthogonal-frequency-division-multiplexing transmission chain with cyclic prefix and one-tap equalization.

- ✓ The block diagram of an OFDM system, including the cyclic prefix, is given in Figure.
- ✓ The original data stream is S/P converted.
- ✓ Each block of N data symbols is subjected to an IFFT and then the last NT_{cp}/N_S samples are prepended.
- ✓ The resulting signal is modulated onto a (single) carrier and transmitted over a channel, which distorts the signal and adds noise.
- ✓ At the receiver, the signal is partitioned into blocks. For each block, the cyclic prefix is stripped off, and the remainder is subjected to an FFT.
- ✓ The resulting samples (which can be interpreted as the samples in the frequency domain) are "equalized" by means of one-tap equalization – i.e., division by the complex channel attenuation – on each carrier.

Drawbacks of CP:

1. Spectral efficiency is reduced.
2. SNR loss occurs.

Advantages of OFDM:

- ✓ It offers high spectral efficiency.
- ✓ It has multi path delay tolerance.
- ✓ Robustness to channel fading.
- ✓ Immunity to impulse interference.

Disadvantages of OFDM:

- ✓ Very sensitive to frequency errors.
- ✓ Inter carrier interference (ICI) due to Doppler spreading.
- ✓ Poor frequency- timing synchronization causes ICI.
- ✓ Transmitter suffers from high signal peak-to-average power ratio
- ✓ Receiver is sensitive to frequency and phase offset in carriers

Applications of OFDM

Used in many wireless communication standards such as

- ✓ Digital Audio Broadcasting (DAB)
- ✓ Digital Video Broadcasting (DVB)
- ✓ Wireless Local Area Networks (LANs)
- ✓ Asymmetrical Digital subscriber Line (ADSL)

Performance of OFDM can be improved by any of the following approaches

(1) Coding across the different tones: such coding helps to compensate for fading dips on one subcarrier by a good SNR in another subcarrier.

(2) Spreading the signal over all tones: In this approach, each symbol is spread across all carriers, so that average SNR also spreads.

(3) Adaptive modulation: If the transmitter knows the SNR on each of the subcarriers, it can choose its modulation and coding rate adaptively.

Thus, on carriers with low SNR, the transmitter will send symbols using stronger encoding and a smaller modulation alphabet. Also the power allocated to each subcarrier can be varied.

****

3.9 WINDOWING

Give a brief Note on Windowing.

Features of OFDM:

It is one of multi-carrier modulation techniques offers a considerable

(i) High spectral efficiency,
(ii) Multipath delay spread tolerance,
(iii) Immunity to the frequency selective fading channels
(iv) Power efficiency.

Applications of OFDM:

(1) Digital Audio Broadcasting (DAB)
(2) Digital Video Broadcasting (DVB)
(3) Wireless Local Area Networks (LANs)

Challenges of OFDM:

i. Due to the large number of sub carriers, OFDM systems have a large dynamic signal range with a very high PAPR- peak to average power ratio.
ii. As a result, the OFDM signal will be clipped when passed through a non linear power amplifier at the transmitter end.
iii. Clipping degrades the bit-error-rate (BER) performance and causes spectral spreading.

Solution to improve BER:

(i) To force the amplifier to work in its linear region. Unfortunately, such a solution is not power efficient.
(ii) Operation of the power amplifier with low back-off values and try to prevent the occurrence of signal clipping.
(iii) The BER performance improvement can be achieved by the **peak windowing method.**

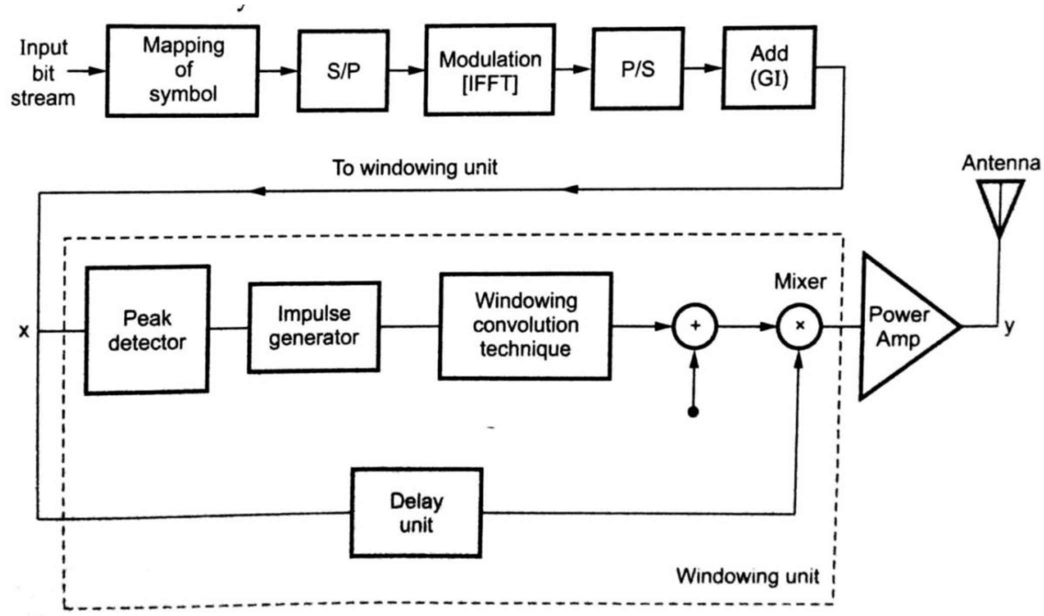

Figure 3.42 OFDM Transmitter with Peak Windowing

- ✓ Passing through a PAPR reduction block such as peak windowing, the signals undergoes a digital-to analog conversion and are transmitted after high power amplifier.
- ✓ At the receiver, the received signals can be demodulated by the reverse process of the transmitter.
- ✓ If we assume the input complex-valued data symbol of N sub carriers as Xk for k = 0, L, N −1, the output signal of the IFFT block is given by

$$x(t) = \sum_{K=0}^{N-1} X_K e^{j2\pi f_k t}, 0 \le t \le NT,$$

where f_k is the frequency of the *k*-th sub carrier defined as *f* = *k f (f* = *1/NT)* and *T* is the sample interval.

- ✓ In order to maintain the out-of-band radiation within a certain level, it is benefit to increase the window length.
- ✓ Examples of suitable window functions are the Cosine, Kaiser, Hamming, and Hamming window.
- ✓ The Kaiser Window function with window length *M* +1 and shape parameter ß is given by

$$w(n) = 0.54 - 0.46\cos\left(2\pi \frac{n}{N}\right), 0 \le n \le N$$

- ✓ Where the window length is L=N+1 , a is defined as α = *M*/2,

Table 3.3: Comparison of Hamming and Kaiser Window

Window	Leakage factor (%)	Side Lobe attenuation Factor(db)	Main Lobe Width(-3db)
Hamming	0.04	-41.8	0.078125
Kaiser	1.32	-21.1	0.0097656

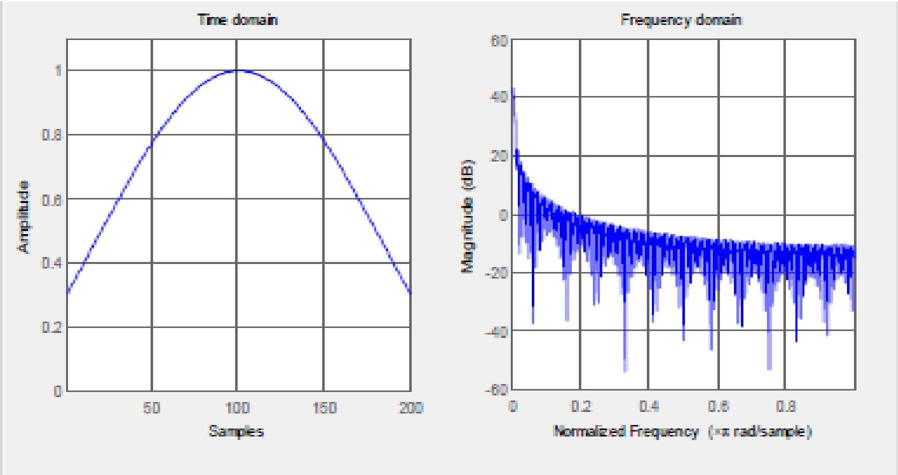

Figure: 3.43 Frequency response of Kaiser Window

- ✓ In general, Kaiser Window is used because it is easy to shape spectrum by changing window length and shape parameter.

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3.10 PEAK AVERAGE POWER RATIO (PAPR)

What is PAPR? Why is it normally larger in an OFDM scheme? (Apr/May 2016)

Major problems of OFDM

(1) Peak-to-average power ratio (PAR)

(2) Inter carrier interference (ICI)

(3) ISI – Inter symbol interference

- ✓ Peak-to-Average Power Ratio (PAR): It is defined as the ratio between the maximum powers occurring in OFDM symbol to the average power of the same OFDM symbol
- ✓ Continuous-time PAPR- In general, the PAPR of OFDM signal x (t) is defined as the ratio between the maximum instantaneous power and its average power

$$PAPR[x(t)] = \frac{\max\limits_{0 \le t \le NT}[x(t)^2]}{P_{av}}$$

where P_{av} is the average power of $x(t)$ and it can be computed in the frequency domain because Inverse Fast Fourier Transform (IFFT) is a (scaled) unitary transformation.

- ✓ This Peak-to-Average Ratio (PAR) issue originates from the fact that an OFDM signal is the superposition of N sinusoidal signals on different subcarriers.
- ✓ On average the emitted power is linearly proportional to N. However, sometimes, the signals on the subcarriers add up constructively, so that the amplitude of the signal is proportional to N, and the power thus goes with N2. So, for worst case the power PAR increases linearly with the number of subcarriers.
- ✓ The low PAR allows the transmit power signal to operate efficiently, while high PAR force the amplifier to operate in linear region.

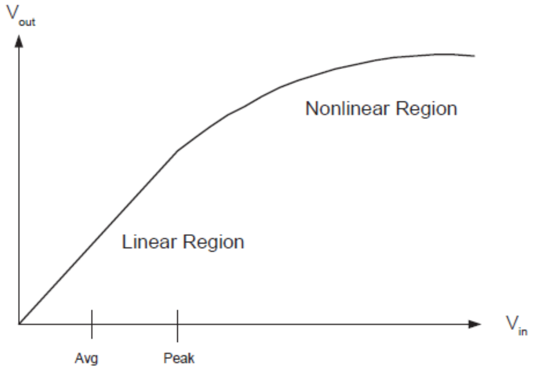


Figure: 3.44 Power amplifier response

High PAR requires high resolution A/D converter.

Methods to deal with the Peak-to-Average Power Ratio

There are three main methods to deal with the Peak-to-Average Power Ratio (PAPR):

1. Use a power amplifier
2. Use a nonlinear amplifier
3. PAR reduction technique

I. Use a power amplifier

- ✓ Put a power amplifier into the transmitter that can amplify linearly up to the possible peak value of the transmit signal.
- ✓ This is usually not practical, as it requires expensive and power-consuming class-A amplifiers.
- ✓ The larger the number of subcarriers *N*, the more difficult this solution becomes.

Any constant amplitude signal, e.g. a square wave, has PAR $= 0$ dB. A sine wave has PAR $= 3$ dB since $\max[\sin^2(t/T)] = 1$ and

$$E[\sin^2(t/T)] = \int_0^T \sin^2(t/T)dt = .5,$$

PAR = 1/0.5 = 2

- ✓ Consider, time domain samples output from IFFT

$$x[n] = \frac{1}{\sqrt{N}} \sum_{i=0}^{N-1} X[i] e^{j\frac{2\pi in}{N}}, \ 0 \le n \le N-1.$$

- ✓ If *N* is large, the Central Limit Theorem is applicable, and x[n] are Zero mean complex Gaussian random variables.
- ✓ The probability that the PAR of discrete time signal exceeds a threshold $P_0 = \frac{\sigma_0^2}{\sigma_n^2}$ is given as

$$p(\text{PAR} \ge P_0) = 1 - (1 - e^{-P_0})^N$$

- ✓ So, we go for peak-to-average ratio reduction technique.

II. Use a nonlinear amplifier:

- ✓ Accept the fact that amplifier characteristics will lead to distortions in the output signal.
- ✓ Those nonlinear distortions destroy orthogonality between subcarriers, and also lead to increased out-of-band emissions (spectral regrowth – similar to third-order intermodulation products – such that the power emitted outside the nominal band is increased).

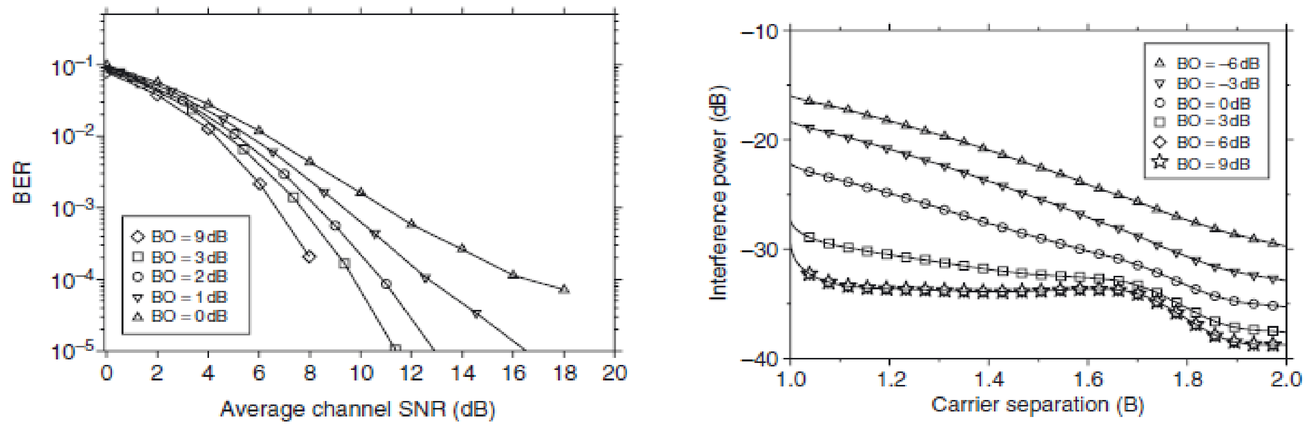


Figure: 3.45 BER as a function of the SNR, for different back off levels of the transmit amplifier.

- ✓ The first effect increases the BER of the desired signal,
- ✓ While the latter effect causes interference to other users and thus decreases the cellular capacity of OFDM system.
- ✓ This means that in order to have constant adjacent channel interference we can trade off pow amplifier performance against spectral efficiency.
- ✓ Increased carrier separation decreases spectral efficiency also.

III. Peak-to-Average Ratio Reduction Techniques

The methods used are:

(1) ***Coding for PAR reduction:*** Coding scheme can guarantee a certain value for the PAR. But it require considerable overhead and thus reduced throughput.

(2) ***Phase adjustments:*** This method has the advantage that the overhead is rather small on the downlink. But cannot give a guaranteed performance.

(3) ***Correction by multiplicative function:*** Multiply the OFDM signal by a time-dependent function whenever the peak value is very high. This reduces intercarrier interference.

(4) ***Correction by additive function:*** Choose additive function instead of multiplicative function. This increases the BER of the system.

APR reduction was required for radar and speech synthesis applications

Inter carrier interference (ICI)

- ✓ The CP – cyclic prefix provides an excellent way of ensuring orthogonality of the carriers in a delay-dispersive environment.
- ✓ ICI reduces the bit error rate (BER) performance of the system.

ISI – Inter Symbol Interference - Spreading of one symbol time with another symbol causes ISI. It has been reduced by the use of equalizer.

TWO MARKS

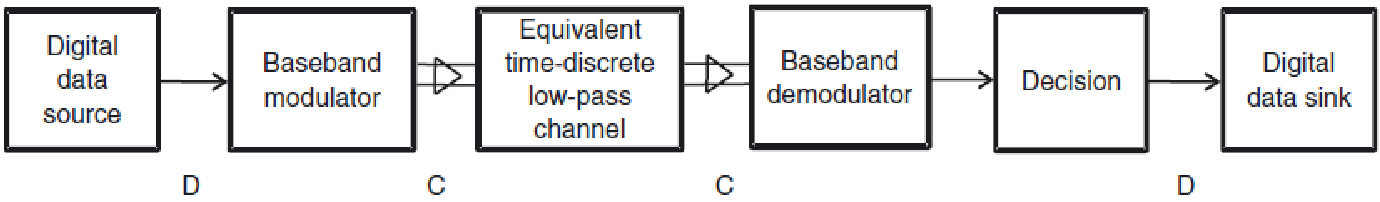
1. **What are the steps involved in the wireless communication link?**

   The steps involved in the wireless communication link are

   - ✓ Source coding
   - ✓ Channel coding
   - ✓ Modulation
   - ✓ Multiple accessing
   - ✓ Transmission through radio channel

2. **Draw the mathematical link for analysis of modulation scheme. (Nov 2011)**

   The mathematical link for analysis of modulation scheme is given below



3. **What is linear modulation?**

   - ✓ In linear modulation technique, the amplitude of the transmitted signal varies linearly with the modulating digital signal.
   - ✓ Linear modulation does not have a constant envelope.
   - ✓ Ex. Pulse shaped QPSK, OQPSK, $\pi/4$QPSK

4. **Define non linear modulation.**

   - ✓ In the non linear modulation the amplitude of the carrier is constant, regardless of the variation in the modulating signals.
   - ✓ Non-linear modulations may have either linear or constant envelopes depending on whether or not the baseband waveform is pulse shaped.
   - ✓ Eg: BFSK, MSK, GMSK

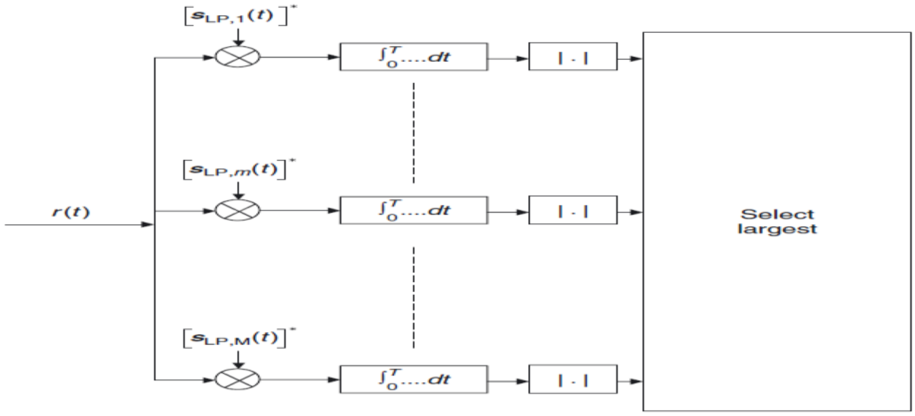
5. **List the advantages of digital modulation techniques. (May 2015)**

   The advantages of digital modulation techniques includes

   - ✓ Greater noise immunity
   - ✓ Robustness to channel impairments
   - ✓ Easier multiplexing of various forms of information.
   - ✓ Greater security

6. **Draw the structure of generic optimum receiver. (May 2013)**

   The structure of generic optimum receiver is given below



7. Mention any two criteria for choosing a modulation technique for a specific wireless application. (May 2013)

Criteria for choosing a modulation technique for a specific wireless application are

- ✓ Low BER at low received SNR.
  - i.e Bit-error rate performance
- ✓ Minimum bandwidth requirement.
  - i.e.Bandwidth efficiency
- ✓ Power efficiency
  - Adjacent channel interference must be small
  - Power spectrum of the signal should show a strong roll-off outside the desired band.
- ✓ Spectral efficiency to be high.
- ✓ Better performance in multipath and fading conditions.
- ✓ Transmission of many data bits with each symbol
- ✓ Ease of implementation and low cost.

8. What is OQPSK? (Nov 2011)

- ✓ To prevent the regeneration of sidelobes and spectral widening, it is imperative that QPSK signals be amplified only using linear amplifiers, which are less efficient.
- ✓ A modified form of QPSK, called offset QPSK (OQPSK) or staggered QPSK is less susceptible to these deleterious effects arid supports more efficient amplification.
- ✓ OQPSK signaling is represented by equation,

$$S_{OQPSK}(t)=\sqrt{\frac{2E_S}{T_S}}\cos\left[(i-1)\frac{\pi}{2}\right]\cos(2\pi f_c t)-\sqrt{\frac{2E_S}{T_S}}\sin\left[(i-1)\frac{\pi}{2}\right]\sin(2\pi f_c t)$$

9. Differentiate between OQPSK and QPSK.

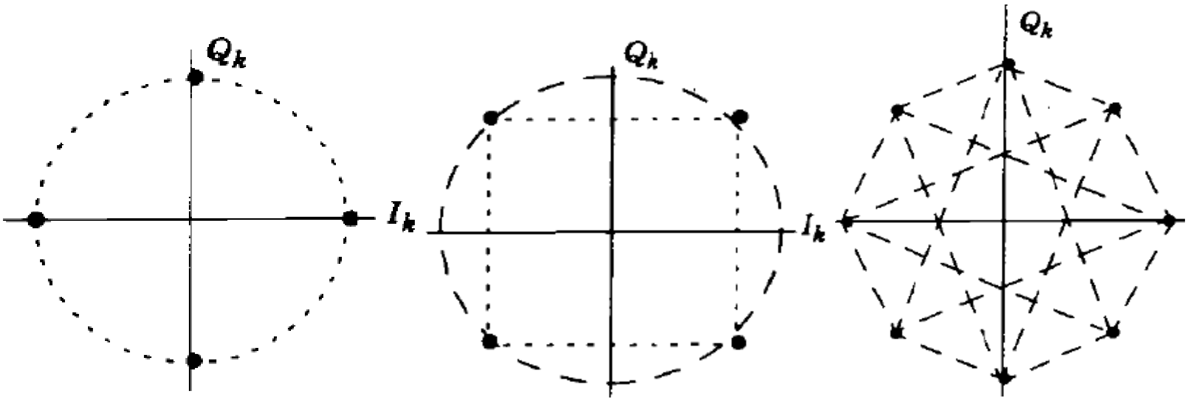
OQPSK	QPSK
In OQPSK signaling, the even and odd bit streams, $m_I(t)$ and $m_Q(t)$ are offset in their relative alignment by one bit period	In QPSK signaling, the bit transitions of the even and odd bit streams occur at the same time instants.
OQPSK signals does not regenerate the high frequency side lobes	QPSK signals regenerate the high frequency side lobes.

10. State the advantages of offset-QPSK. (Nov 2014)

The advantages of offset-QPSK includes

- ✓ Lower amplitude fluctuations.
- ✓ Suppress out-of-band interference.
- ✓ Limits the phase-shift to maximum of 90° at a time.
- ✓ Spectral occupancy is significantly reduced.
- ✓ More efficient RF amplification.
- ✓ Better performance in the presence of phase jitter due to noisy reference signals at the receiver

11. Draw the signal constellation and phase transition of $\pi/4$ QPSK signal.



(a) (b) (c)

(a) Possible States for θ_k when $\theta_{k-1} = \frac{n\pi}{4}$

(b) Possible States for θ_k when $\theta_{k-1} = \frac{n\pi}{2}$

(c) All Possible States

Figure: Constellation diagram of a $\frac{\pi}{4}$ QPSK signal

12. List the various types of detection techniques that are used for the detection of $\pi/4$ QPSK signals.

- ✓ There are various types of detection techniques that are used for the detection of $\pi/4$ QPSK signals that includes
  - Baseband differential detection
  - IF differential detection
  - FM discriminator detection

13. Differentiate offset QPSK and π/4 differential QPSK.

Offset QPSK	$\pi/4$ **DQPSK**
The amplitude of data pulses are kept constant. The time alignment of the even and odd bit streams are offset by one bit period in offset QPSK.	Signaling points of the modulated signal are selected from two QPSK constellations which are shifted by π/4 with respect to each other. It is differentially encoded and detected so called π/4 differential QPSK.

14. Define offset QPSK and π/4 differential QPSK. [Nov/Dec 2017]

The amplitude of data pulses are kept constant. The time alignment of the even and odd bit streams are offset by one bit period in offset QPSK.

Signaling points of the modulated signal are selected from two QPSK constellations which are shifted by π/4 with respect to each other. It is differentially encoded and detected so called π/4 differential QPSK.

15. What is meant by MSK?

- ✓ A continuous phase FSK signal with a deviation ratio of one half is referred to as MSK.
- ✓ MSK is a spectrally efficient modulation scheme.

16. Why is MSK referred to as fast FSK? (May 2016)

MSK is called as fast FSK since the frequency spacing used is only half as much as that used in conventional non-coherent FSK.

17. List the salient features of MSK scheme.

Salient features of MSK are:

- ✓ MSK has constant envelope.
- ✓ Relatively narrow bandwidth.
- ✓ Coherent detection suitable for satellite communications.
- ✓ Side lobes are zero outside the frequency band.
- ✓ Resist to co channel interference.

18. Mention the advantages of MSK over QPSK.

The advantages of MSK over QPSK are

- ✓ Output waveform is continuous in phase
- ✓ No abrupt changes in amplitude.
- ✓ Bandwidth requirement is less

19. Why MSK cannot be directly used in multi user communications?

- ✓ The main lobe of MSK is wide. This makes MSK unsuitable for the applications where extremely narrow bandwidths and sharp cut-offs are required.
- ✓ Slow decay of MSK power spectral density curve creates adjacent channel interference. Hence MSK cannot be used for multiuser communications.

20. Give the function of Gaussian filter in GMSK. (Nov 2016)

Comment on the necessity of a Gaussian filter in GMSK. (May 2015)

- ✓ Gaussian filters are used before the modulator to reduce the transmitted bandwidth of the signal.
- ✓ Gaussian filters use less bandwidth than conventional FSK.
- ✓ Gaussian filtering converts the full response message signal into a partial response scheme where each transmitted symbol spans several bit periods.

21. List the advantages of GMSK.

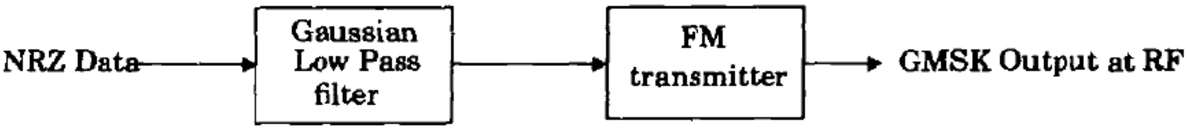
The advantages of GMSK are

- ✓ Improved spectral efficiency when compared to other phase shift keyed modes.
- ✓ Amplified by a non-linear amplifier and remain undistorted.

- ✓ Immune to amplitude variations
- ✓ More resilient to noise.
- ✓ Excellent power efficiency

22. How is GMSK generated?

The simplest way to generate a GMSK signal is to pass a NRZ message bitstream through a Gaussian baseband filter followed by an FM modulator.



23. What do you mean by Non-Coherent detection? (Nov 2015)

When no phase information is required for detection, the type of detection is called non coherent detection.

24. Find the 3-dB bandwidth for a Gaussian low pass filter used to produce 0.25 GMSK with a channel data rate of R_b=270 kbps. What is the 90% power bandwidth in the RF channel? (Nov 2012)

Given

Channel data rate of R_b=270 kbps

Bandwidth-bit duration =0.25

Solution:

$$T = \frac{1}{R_b} = \frac{1}{270 \times 10^3} = 3.7\mu s$$

BT=0.25

$$B = \frac{BT}{T} = \frac{0.25}{3.7 \times 10^{-6}} = 67.567kHz$$

For the 3-dB bandwidth = $67.567kHz$. For 90% power bandwidth, 0.57 is the reserved value. The occupied RF spectrum for a 90% power bandwidth is given by

RF BW=0.57 R_b= $0.57 \times 270 \times 10^3$ =153.9 kHz.

25. Find the 3-dB bandwidth for a Gaussian low pass filter used to produce 0.25 GMSK with a channel data rate of R_b=300 kbps. (Nov 2015)

Given

Channel data rate of R_b=300 kbps

Bandwidth-bit duration =0.25

Solution:

$$T = \frac{1}{R_b} = \frac{1}{300 \times 10^3} = 3.3\mu s$$

BT=0.25

$$B = \frac{BT}{T} = \frac{0.25}{3.3 \times 10^{-6}} = 75.757kHz$$

26. Differentiate MSK and GMSK. (May 2012)

Minimum shift keying (MSK)	Gaussian Minimum shift Keying (GMSK)
Minimum shift keying (MSK) is a special type of continuous phase frequency shift keying (CPFSK).	GMSK is the derivative of Minimum shift keying
The sidelobe levels of the spectrum are high	The sidelobe levels of the spectrum are reduced by passing the modulating NRZ data waveform through a premodulation Gaussian pulse-shaping filter
Pulse shaping is not efficient	Gaussian pulse shaping is done.
MSK has instantaneous frequency variations over time	GMSK stabilizes the instantaneous frequency variations over time

27. Give the expression for bit error probability of Gaussian Minimum shift Keying modulation. (May 2016) (Nov 2013)

- ✓ The expression for bit error probability of Gaussian Minimum shift Keying modulation is

$$P_{e,GMSK} = \frac{1}{2}\left(1-\sqrt{\frac{\delta\Gamma}{\delta\Gamma+1}}\right) \rightarrow P_{e,GMSK,approx} = \frac{1}{4\delta\Gamma} \qquad \text{where} \quad \delta \cong \begin{cases} 0.68 & BT = 0.25 \\ 0.85 & BT = \infty \end{cases}$$

28. What is OFDM?

- ✓ Orthogonal frequency division multiplexing splits the information into N parallel streams, which are then transmitted by modulating N distinct carriers. In order to separate the subcarriers by the receiver, they have to be orthogonal.
- ✓ Orthogonal Frequency Division Multiplexing (OFDM) is a modulation scheme that is especially suited for high-data-rate transmission in delay-dispersive environments.
- ✓ OFDM converts a high-rate data stream into a number of low-rate streams that are transmitted over parallel, narrowband channels that can be easily equalized.

29. What is the purpose of OFDM?

The purpose of applying OFDM is to reduce frequency selective fading and burst errors generated by wide band fading channels in wireless communications.

30. List the applications of OFDM.

- ✓ OFDM is used for
  - ✓ Digital Audio Broadcasting (DAB)
  - ✓ Digital Video Broadcasting (DVB)
  - ✓ Wireless Local Area Networks (LANs) (IEEE 802.11a, IEEE 802.11g)

31. How is OFDM Interpreted?

- ✓ OFDM can be interpreted in two ways:
  - ▪ Analog implementation and Digital implementation

32. Define cyclic prefix. (Nov 2016) (or) What do you mean by cyclic prefix? [April/May 2018]

In OFDM, delay dispersion leads to a loss of orthogonality between the subcarriers and thus leads to Inter Carrier Interference (ICI). These negative effects can be eliminated by a special type of guard interval called the cyclic prefix.

33. Give the mathematical expression for cyclic prefix.

- ✓ Symbol duration T_s of basic function for transmission is given by

$$T_S = \hat{T}_S + T_{cp}$$

- ✓ For the duration $0 < t < \hat{T}_S$, Normal OFDM symbol is transmitted
- ✓ For the duration $-T_{cp} < t < 0$, a copy of the last part of the symbol is transmitted.
- ✓ This implies that $g_n(t) = g_n(t + N/W)$.
- ✓ This prepended part of the signal is called the cyclic prefix

34. What is windowing? (May 2016)

- ✓ In the OFDM technique, in real systems PAPR would affect the transmission efficiency and to improve the bit error rate (BER) windowing technique is applied.
- ✓ When successive peaks emerge at less than half of window size in use, then the windows would overlap and it may lead to degradation of BER.
- ✓ Hence the advanced windowing technique can be used to improve BER.

35. Define PAPR. [Nov/Dec 2017]

The peak-to-average power ratio (PAPR) is the peak amplitude squared (giving the peak power) divided by the RMS value squared (giving the average power).

36. What are the PAPR reduction techniques?

- ➢ The approaches for reducing Peak-to-Average Ratio problem are
  1. Coding for PAR reduction
  2. Phase adjustments
  3. Correction by multiplicative function
  4. Correction by additive function

37. Write down the expression for probability of error for PSK modulation techniques, with coherent detection for the following cases. (a) AWGN (b) Rayleigh fading (May 2015)

Probability of error for BPSK (Coherent) in the AWGN channel.

$$P(X) = \frac{1}{\Gamma} e^{\left(\frac{-X}{\Gamma}\right)}$$

$$P_{e,PSK} = Q\left(\sqrt{\frac{2E_b}{N_0}}\right) = Q\left(\sqrt{2X}\right)$$

Substituting two equations above in the following equation

$$P_e = \int_0^\infty P_e(X) p(X) dX - \int_0^\infty \frac{1}{\Gamma} e^{\left(\frac{-X}{\Gamma}\right)} Q\sqrt{2X}\, dX$$

$$P_{e,PSK} = \frac{1}{2}\left[1 - \sqrt{\frac{\frac{E_b}{N_0}\overline{\alpha^2}}{1+\frac{E_b}{N_0}\overline{\alpha^2}}}\right]$$

$$= \frac{1}{2}\left[1 - \sqrt{\frac{\Gamma}{1+\Gamma}}\right] \qquad \Gamma = \frac{E_b}{N_0}\overline{\alpha^2}$$

- ✓ Rayleigh Probability density function (PDF) is given by

$$P(t) = \frac{r}{\sigma^2} e^{\left(\frac{-r^2}{2\sigma^2}\right)} \qquad 0 \le r \le \infty$$

- ✓ When input signal at a receiver is Rayleigh

$$X = \frac{Signal\ Power}{Noise\ Power} - \frac{A^2 r^2}{2.P_n}$$

- ✓ Let α be a complex Gaussian random variable

$$\alpha = x + yi$$

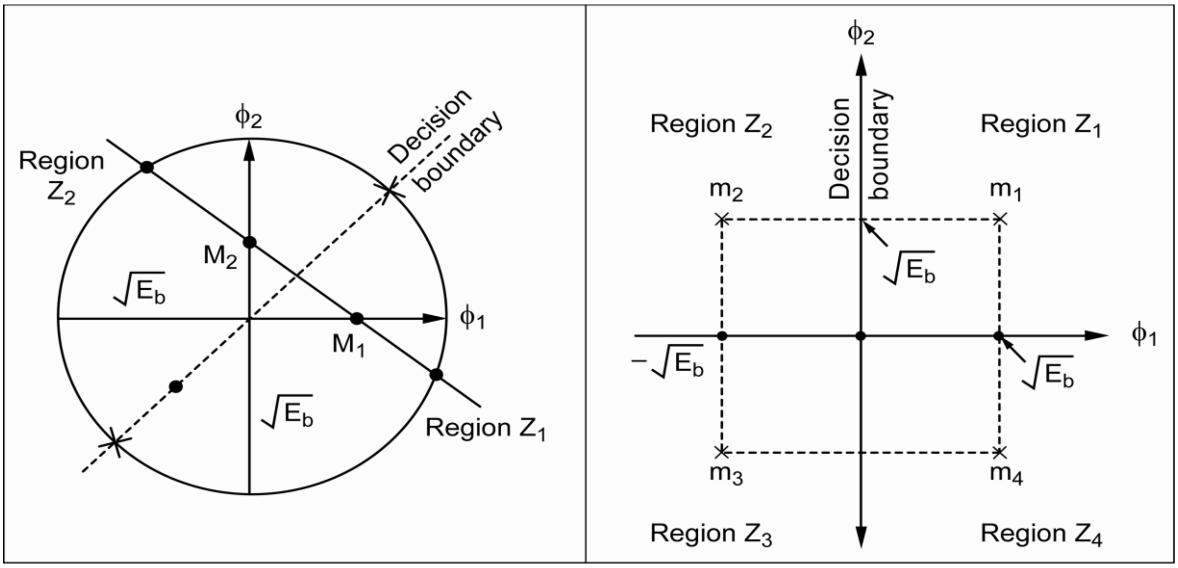
$$|\alpha| - r$$

σ → Standard Deviation

- ✓ The average SNR for the channel Γ is the mean of X

$$= \frac{A^2 E(r^2)}{2.P_n} - \frac{A^2\sigma^2}{P_n}$$

37.Draw the constellation diagram of BFSK, MSK.(Nov 2015,Aprril/May 2018)



UNIT IV
MULTIPATH MITIGATION TECHNIQUES

Syllabus:

Equalization – Adaptive equalization, linear and Non-Linear equalization, Zero forcing and LMS Algorithms. Diversity – Micro and Macro diversity, Diversity combining techniques, Error probability in fading channels with diversity reception, Rake receiver.

INTRODUCTION

✓ The following three techniques which can be used independently or in tandem to **improve received signal quality** and link performance over small scale times and distances.

1. Diversity
2. Equalization
3. Channel coding

Diversity

✓ It is a technique used to ***compensate for fading*** channel impairments and is usually implemented by using two or more receiving antennas.

✓ Diversity improves the quality of a wireless communication link without altering the common air interface, without increasing the transmitting power or band width.

Equalization

✓ A filter which equalizes the dispersive effect of a channel is referred to as an equalizer.

✓ Equalization can be used to ***compensate the inter symbol interference*** (ISI) created by multipath within time dispersion channel.

Channel coding

✓ It is a technique to overcome transmission errors over a noisy channel.

✓ Channel coding and decoding is used to detect and correct bit errors introduced by the channel.

✓ Channel coding improves the small-scale link performance.

e.g. Space time coding.

✓ Channel coding + diversity $\Rightarrow$ Space time coded modulation.

✓ It is a bandwidth and power efficient method for wireless communication.

4.1. EQUALIZATION

Q.Explain about equalization

✓ A filter which equalizes the dispersive effect of a channel is referred to as an equalizer.

✓ Equalization can be used to ***compensate the inter symbol interference*** (ISI) created by multipath within time dispersion channel.

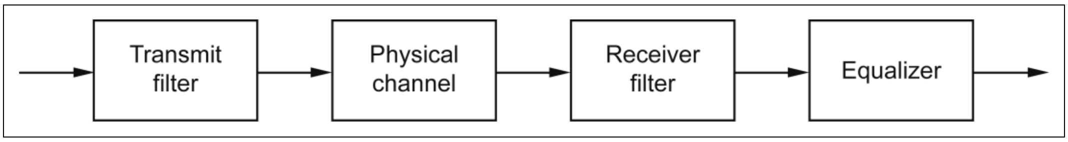


Fig. 4.1. The equalized system

✓ Inter Symbol Interference (ISI) caused by multipath in band limited time dispersive channels.

✓ It distorts the transmitted signal, causing bit errors at the receiver.

- ✓ ISI has been recognized as the major obstacle to high speed data transmission over wireless channels.
- ✓ Equalization is a technique used to ***combat inters symbol interference.*** The device which equalizes the dispersive effects of a channel is referred to as an ***equalizer.***
- ✓ In a broad sense, the term equalization can be used to describe any signal processing operation that minimizes ISI.
- ✓ In radio channels, a variety of adaptive equalizers can be used to cancel interference while providing diversity. Since the mobile fading channel is (*i*) random, (*ii*) time varying.
- ✓ Equalizers must track the time varying characteristics of the mobile channel and these are called adaptive equalizers.

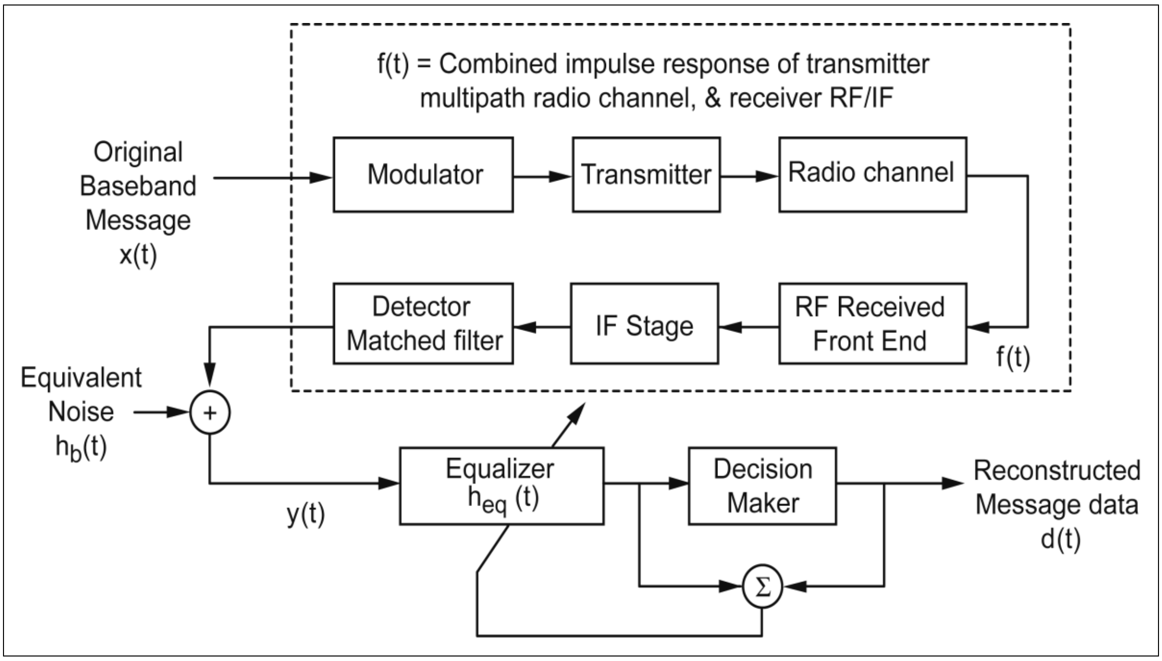


Fig. 4.2. General Block diagram of a communications system using an adaptive equalizer at the receiver

- ✓ The above Fig.4.2 shows a block diagram of a communication system with an adaptive equalizer in the receiver.
- ✓ If $x(t)$ is the original information signal and $f(t)$ is the combined complex baseband impulse response of the transmitter, channel and the RF/IF sections of the receiver, the signal received by the equalizer may be expressed as

$$y(t) = x(t) \otimes f^*(t) + n_b(t) \qquad \ldots (4.1)$$

where

$f^*(t)$ $\rightarrow$ the complex conjugate of $f(t)$,

$n_b(t)$ $\rightarrow$ the baseband noise at the input of the equalizer

$\otimes$ $\rightarrow$ the convolution operation.

- ✓ If the impulse response of the equalizer is $h_{eq}(t)$, then the output of the equalizer is

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

$$= x(t) \otimes g(t) + n_b(t) \otimes h_{eq}(t) \qquad \ldots (4.2)$$

- Where $g(t)$ is the combined impulse response of the transmitter, channel, RF/IF sections of the receiver, and the equalizer at the receiver. The complex baseband impulse response of the transversal filter equalizer is given by

$$h_{eq}(t) = \sum_{n} C_n \delta(t - nT) \qquad \ldots (4.3)$$

Where c_n are the complex filter coefficients of the equalizers. The descried output of the equalizers is $x(t)$, the original source data. Assume that $n_b(t) = 0$. Then in order to force $\hat{d}(t) = x(t)$ in equation (4.2), $g(t)$ must be equal to

$$g(t) = f*(t) \otimes h_{eq}(t) = \delta(t) \qquad \ldots (4.4)$$

- The goal of equalization is to satisfy the equation (4.4) so that the combination of the transmitter, channel and receiver appear to be an all-pass channel. In the frequency domain equation (4.4) can be expressed as

$$H_{eq}(f) F*(-f) = 1 \qquad \ldots (4.5)$$

where $H_{eq}(f) \leftarrow F(f)$ are Fourier transforms of $h_{eq}(t) \leftarrow f(t)$ respectively.

- Training sequence is transmitted before the information data sequence to compute the initial optimum tap coefficients of the adaptive equalizer.

4.2. ADAPTIVE EQUALIZATION

1. **How does a generic adaptive equalizer work during training? Nov/Dec 2012**
2. **Explain about adaptive equalizers. [Nov/Dec 2017]**

- An adaptive equalizer is a time-varying filter which must constantly be returned.

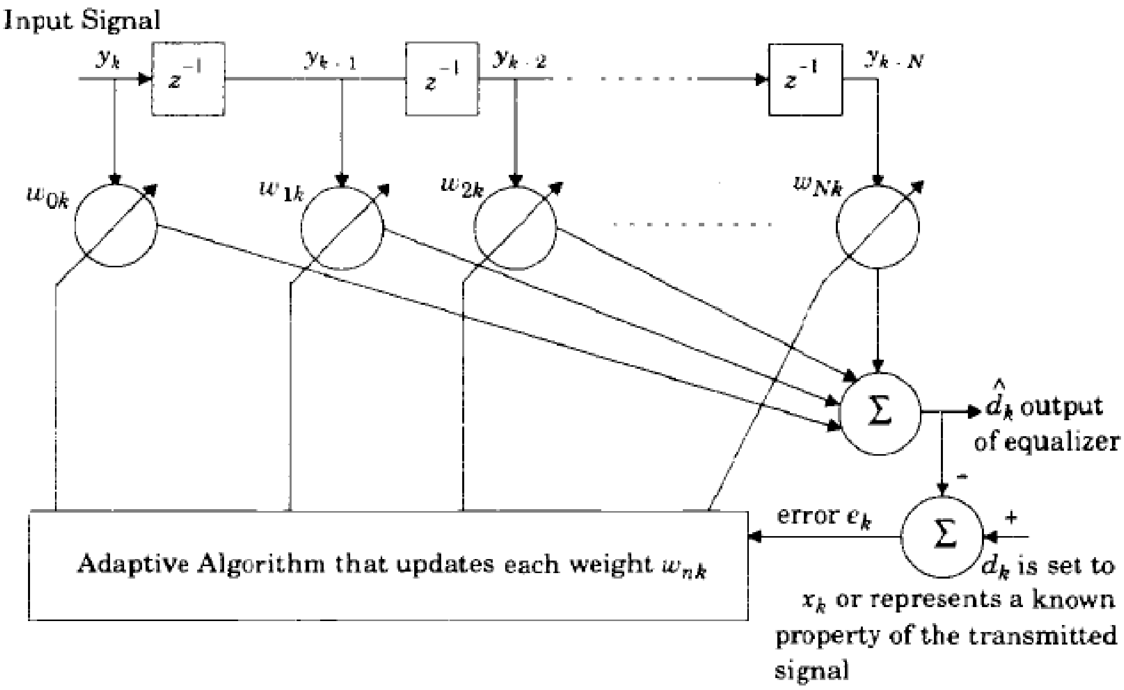


Figure 4.2: A basic adaptive equalizer during training.

- ✓ The basic structure of an adaptive equalizer is shown in Figure.
- ✓ The subscript k is used to denote a discrete time index.
- ✓ There is a single input y_k at any time instant. The value of y_k depends upon the instantaneous state of the radio channel and the y_k is a random process.
- ✓ The adaptive equalizer structure shown above is called a transversal filter, and in this case has N delay elements, $N+1$ taps, and $N+1$ tunable complex multipliers, called weights.
- ✓ The weights of the filter are described by their physical location in the delay line structure, and have a second subscript, k, to explicitly show they vary with time.
- ✓ These weights are updated continuously by the adaptive algorithm, either on a sample by sample basis (i.e., whenever k is incremented by 1) or on a block by block basis (i.e., whenever a specified number of samples have been clocked into the equalizer).
- ✓ The adaptive algorithm is controlled by the error signal e_k.
- ✓ This error signal is derived by comparing the output of the equalizer, d_k with some signal which is either an exact scaled replica of the transmitted signal x_k or which represents a known property of the transmitted signal.
- ✓ The adaptive algorithm uses C_k to minimize a cost function and updates the equalizer weights in a manner that iteratively reduces the cost function.
- ✓ For example, the least mean squares (LMS) algorithm searches for the optimum or near-optimum filter weights by performing the following iterative operation:

$$New\ weights = Previous\ weights\ +\ (constant) \times (Previous\ error) \times (Current\ input\ vector)$$

where,

$$Previous\ error = Previous\ desired\ output\ -\ Previous\ actual\ output$$

- ✓ The *constant* may be adjusted by the algorithm to control the variation between filter weights on successive iterations.
- ✓ This process is repeated rapidly in a programming loop while the equalizer attempts to converge, and many techniques (such as *gradient* or *steepest decent algorithms*) may be used to minimize the error.
- ✓ Upon reaching convergence, the adaptive algorithm freezes the filter weights until the error signal exceeds an acceptable level or until a new training sequence is sent.
- ✓ The most common cost function is the mean square error (MSE) between the desired signal and the output of the equalizer.
- ✓ The MSE is denoted by $E[e(k)e^*(k)]$ and a known training sequence must be periodically transmitted signal and do not required when a replica of the transmitted signal is required at the output of the equalizer (i.e., when d_k is set equal to X_k and is known *a priori*).
- ✓ By detecting the training sequence, the adaptive algorithm in the receiver is able to compute and minimize the cost function by driving the tap weights until the next training sequence is sent.
- ✓ More recent adaptive algorithms are able to exploit the characteristics of the transmitted signal and do not require training sequences.

- ✓ These modern algorithms acquire equalization through property restoral techniques of the transmitted signal, which are called as *blind algorithms*.
- ✓ These techniques include algorithms such as Constant Modulus Algorithm (CMA) and the Spectral COherence Restoral Algorithm (SCORE).
- ✓ Constant Modulus Algorithm (CMA) is used for *constant envelope modulation*.

Algorithm:

- ✓ To study the adaptive equalizer of figure, it is helpful to use vector and matrix algebra.
- ✓ Define the input signal to the equalizer as a vector Y_k where

$$y_k = \begin{bmatrix} y_k & y_{k-1} & y_{k-2} & \cdots & y_{k-N} \end{bmatrix}^T$$

✓ It should be clear that the output of the adaptive equalizer is a scalar given by

$$\hat{d}_k = \sum_{n=0}^{N} w_{nk} y_{k-n}$$

and following equation ,

a *weight vecto*r can be written as

$$w_k = \begin{bmatrix} w_{0k} & w_{1k} & w_{2k} & \cdots & w_{Nk} \end{bmatrix}^T$$

✓ It may be written in vector notation as

$$\hat{d}_k = y_k^T w_k = w_k^T y_k$$

✓ It follows that when the desired equalizer output is known (i.e., $d_k = x_k$), the error signal e_k is given by

$$e_k = d_k - \hat{d}_k = x_k - \hat{d}_k$$

$$e_k = x_k - y_k^T w_k = x_k - w_k^T y_k$$

✓ To compute the mean square error $|e_k|^2$ at time instant *k*, equation is squared to obtain

$$|e_k|^2 = x_k^2 + w_k^T y_k \ y_k^T w_k - 2x_k \ y_k^T w_k$$

✓ Taking the expected value of $|e_k|^2$ over k (which in practice amounts to computing a time average)

✓ Yields

$$E\left[\ |e_k|^2\right] = E\left[x_k^2\right] + w_k^T E\left[y_k \ y_k^T w_k\right] - 2E\left[x_k \ y_k^T\right] w_k$$

✓ Notice that the filter weights w_k are not included in the time average since, for convenience, it is assumed that they have converged to the optimum value and are not varying with time.

✓ Now, the cross correlation vector p between the desired response and the input signal is defined as

$$p = E\left[x_k y_k\right] = E\begin{bmatrix} x_k y_k & x_k y_{k-1} & x_k y_{k-2} & \cdots & x_k y_{k-N} \end{bmatrix}^T$$

$$R = E\left[y_k y_k{}^*\right] = \begin{bmatrix} y_k^2 & y_k y_{k-1} & y_k y_{k-2} & y_k y_{k-N} \\ y_{k-1} y_k & y_{k-1}^2 & y_{k-1} y_{k-2} & y_{k-1} y_{k-N} \\ \cdots & \cdots & \cdots & \cdots \\ y_{k-N} y_k & y_{k-N} y_{k-1} & y_{k-N} y_{k-2} & y_{k-N}^2 \end{bmatrix}$$

- ✓ The matrix R is sometimes called the input covariance matrix.
- ✓ The major diagonal of R contains the mean square values of each input sample, and the cross terms specify the autocorrelation terms resulting from delayed samples of the input signal.
- ✓ If x_k and y_k are stationary, then the elements in R and p are second order statistics which do not vary with time.
- ✓ Mean Square Error, $\xi = E\left[x_k^2\right] + w^T R w - 2p^T w$
- ✓ By minimizing this equation in terms of the weight vector w_k, it becomes possible to adaptively tune the equalizer to provide a flat spectral response (minimal ISI) in the received signal.
- ✓ This is due to the fact that when the input signal y_k and the desired response x_k are the mean square error (MSE) is quadratic on w_k, and minimizing the MSE leads to optimal solutions for w_k.

Equalizers in a Communications Receiver:

- ✓ Adaptive equalizers are implemented using digital logic; it is most convenient to represent all time signals in discrete form.
- ✓ Let T represent some increment of time between successive observations of signal states.
- ✓ Let $t = t_n$ where n is an integer that represents $time = nT$, time waveforms may be equivalently expressed as a sequence on n in the discrete domain.
- ✓ Then the output of the equalizer may be expressed as,
  - ✓ $\hat{d}(n) = x(n) \otimes g(n) + n_b(n) \otimes h_{eq}(n)$
- ✓ The prediction error is

$$e(n) = d(n) - \hat{d}(n) = d(n) - \left[x(n) \otimes g(n) + n_b(n) \otimes h_{eq}(n)\right]$$

- ✓ The mean squared error $E\left[\,|e_n|^2\right]$ is one of the most important measures of how well an equalizer works.
- ✓ $E\left[\,|e_n|^2\right]$ is the expected value *(ensemble average)* of the squared prediction error $|e_n|^2$, but time averaging can be used if $e(n)$ is ergodic.
- ✓ In practice, ergodicity is impossible to prove, and algorithms are developed and implemented using time averages instead of ensemble averages.
- ✓ This proves to be highly effective, better equalizers provide smaller values of ***mean squared error*** $E\left[\,|e_n|^2\right]$.
- ✓ Minimizing the mean square error tends to reduce the bite rate.
- ✓ Suppose $e(n)$ is guassian distributed with zero mean.
- ✓ Then $E\left[\,|e_n|^2\right]$ is the variance (or the power) of the error signal.
- ✓ If the variance is minimized, then there is less chance of perturbing the output signal $d(n)$.
- ✓ Thus, the decision device is likely to detect $d(n)$ as the transmitted signal $x(n)$.
- ✓ Consequently, there is a smaller probability of error when $E\left[\,|e_n|^2\right]$ is minimized.
- ✓ For wireless communication links, it would be best to minimize the instantaneous probability of error P_e instead of the mean squared error, but minimizing P_e generally results in nonlinear equations, which are much more difficult to solve in real-time.

Equalization Techniques

- The major classification of equalization techniques is linear and nonlinear equalization.

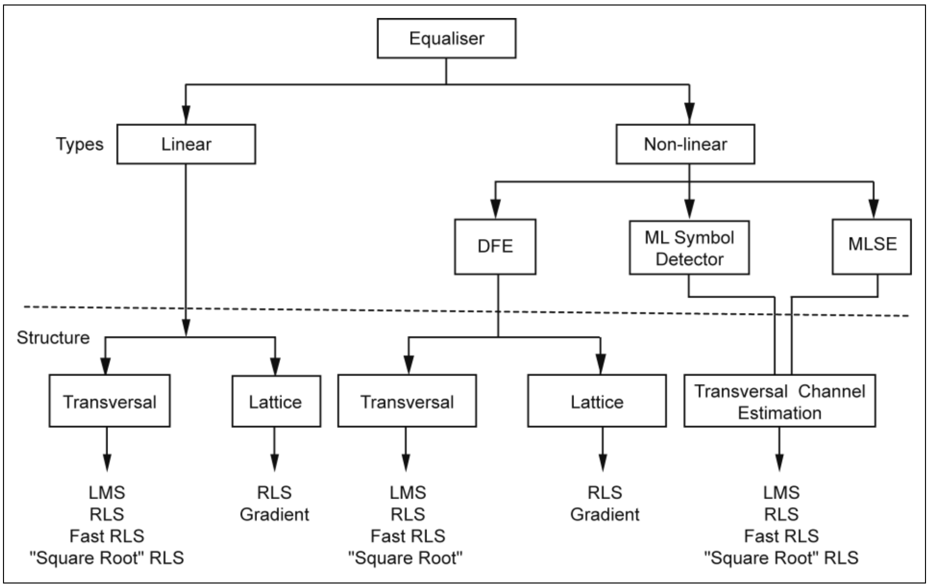


Figure 4.3: Classification of equalizers.

(i) Linear equalizers

If the output is not used in the feedback path to adapt the equalizer is called linear equalizer.

(ii) Non-linear equalizers

If the output is fed back to change the subsequent outputs of the equalizer is called non-linear equalizers.

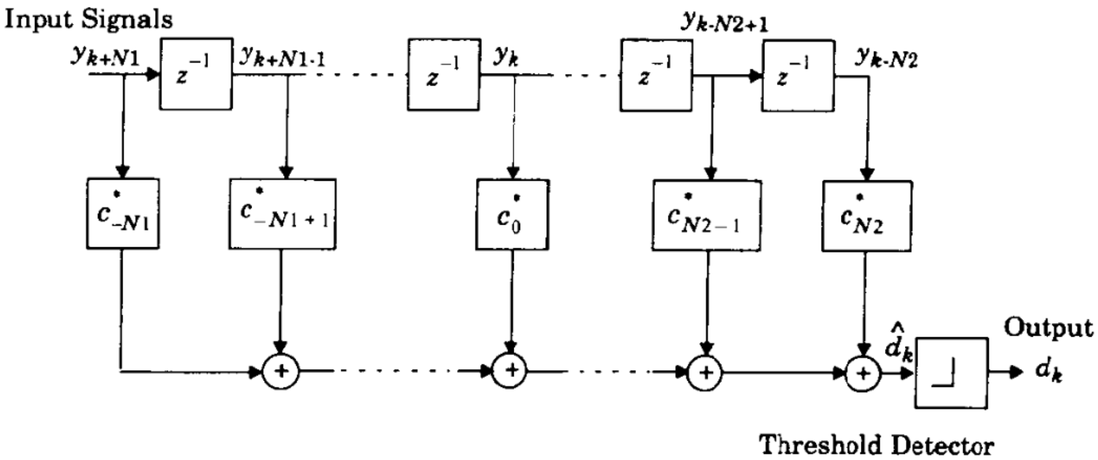


Figure: Structure of linear transversal equalizer

- ✓ The most common equalizer structure is a *linear transversal equalizer (LTE)*. A linear transversal filter is made up of tapped delay lines, with the tapping spaced a symbol period (T_S) apart, shown in figure.
- ✓ Assuming that the delay elements have unity gain and delay Ts, the transfer function of a linear transversal equalizer can be written as a function of the delay operator exp (—jwTs)
- ✓ The simplest LTE uses only feed forward taps is finite impulse response (FIR) filter.
- ✓ If the equalizer has both feed forward and feedback taps, it is called an infinite impulse response (IIR) filter.
- ✓ IIR filters are unstable when used in channels where the strongest pulse arrives after an echo pulse (i.e., leading echoes), they are rarely used.

4.3. LINEAR EQUALIZER

1. **Explain in detail about linear and non linear equalizer. (Nov/Dec 2017, May 2016, Nov/Dec 2012)**
2. **Describe the role played by equalization and diversity as multipath mitigation technique. Compare and contrast these two techniques. (Apr/May 2017)**

- ✓ Linear equalizers are simple linear filter structures, can be implemented as an FIR filter, otherwise known as the transversal filter.
- ✓ In linear equalizer, the current and past values of the received signal are linearly weighted by the filter coefficients and summed to produce the output as shown in figure 4.4.
- ✓ If the delays and the tap gains are analog, the continuous output of the equalizer is sampled at the symbol rate and the samples are applied to the decision device.

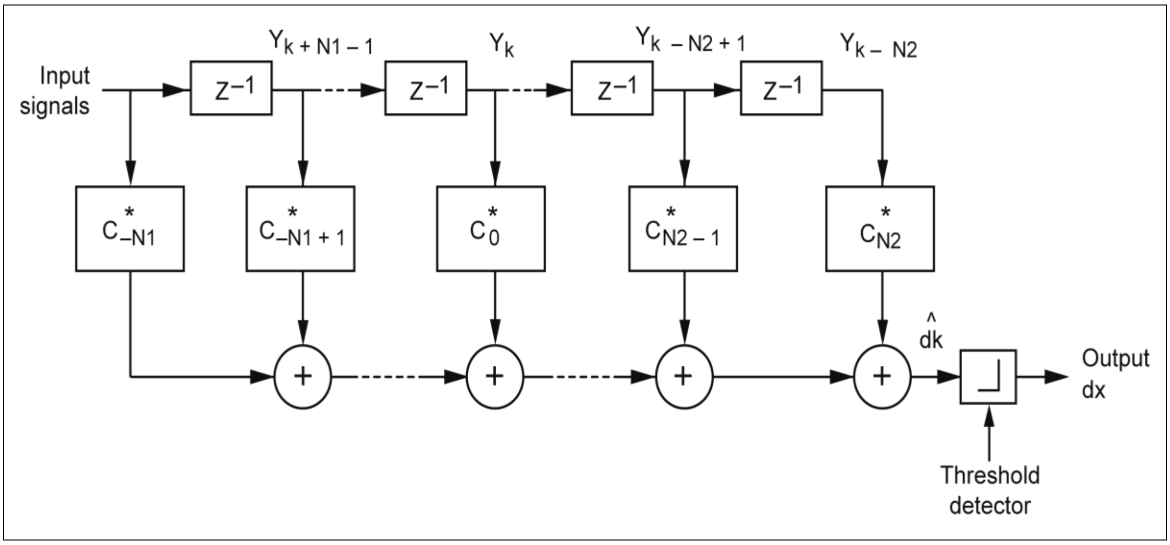


Fig. 4.4. Structure of a linear transversal equalizer

- ✓ It is implemented is carried out in ***digital domain***.
- ✓ The samples of the received signal are stored in a shift register.
- ✓ The output of this transversal filter before a decision is made (threshold decision) is

$$\hat{d}_k = \sum_{n=-N_1}^{N_2} (c_n^*)\, y_{k-n}$$

Where,

	c_n^*	$\rightarrow$	complex filter coefficients or tap weights
	$\hat{d}_k$	$\rightarrow$	output at time index k
	y_i	$\rightarrow$	input received signal at time $t_0 + i\,T$
	t_0	$\rightarrow$	equalizer starting time
	N	$\rightarrow$	$N_1 + N_2 + 1$ is the number of taps
	N_1, N_2	$\rightarrow$	forward, reverse portions of the equalizer using the taps

- ✓ If the delays z^{-1} and the tap gains C_N are analog, the continuous output of the equalizer is sampled at the symbol rate and the samples are applied to the decision device.
- ✓ The minimum mean square error is $E[|e(n)|^2] = \frac{T}{2\pi} \int_{-\pi/T}^{\pi/T} \frac{N_0}{|F(e^{j\omega T})|^2 + N_0} \, d\omega$

where, $F(e^{j\omega T}) \rightarrow$ Frequency response of the channel

$N_0 \rightarrow$ Noise power spectral density

$$\text{Equalizer transfer function} \propto \frac{1}{\text{Channel transfer function}} \Rightarrow \left\{ \begin{array}{c} \text{Minimize} \\ \text{mean} \\ \text{square} \\ \text{error} \end{array} \right\}$$

The structure of a lattice equalizer:

- ✓ The linear equalizer can also be implemented as a lattice filter
- ✓ In a lattice filter, the input signal y_k is transformed into a set of N intermediate forward and backward error signals $f_n(k)$ and $b_n(k)$ respectively, which are used as inputs to the tap multipliers and are used to calculate the updated coefficients.
- ✓ Each stage of the lattice is then characterized by the following recursive equations.

$$f_1(k) = b_1(k) = y(k)$$

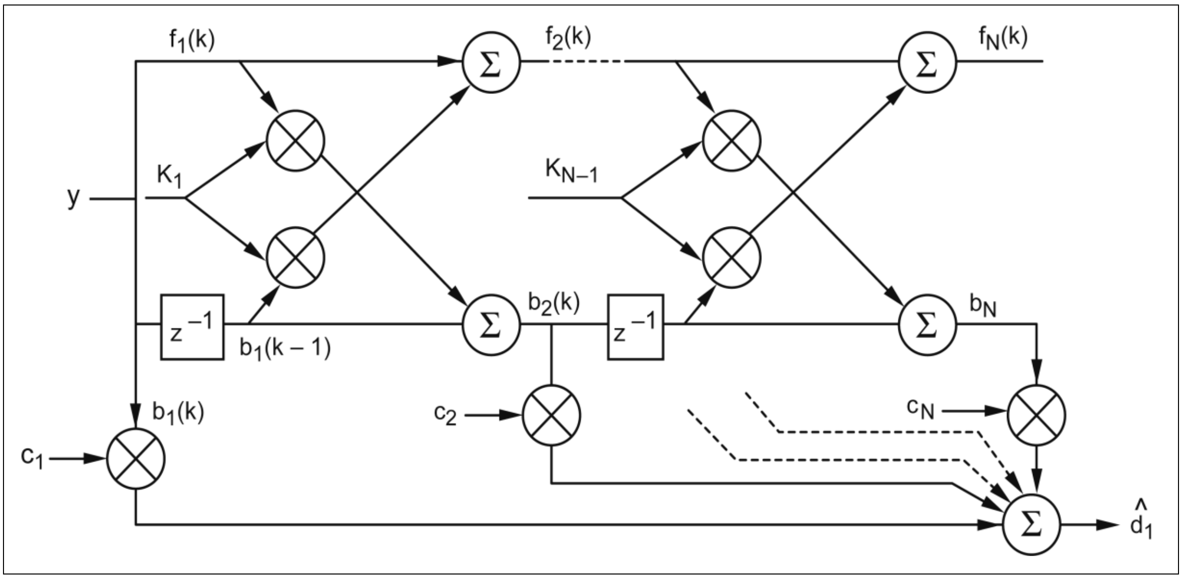


Figure 4.5. The structure of lattice equalizer

$$\begin{aligned} f_n(k) &= y(k) - \sum_{i=1}^{n} K_i \, y(k-i) \\ &= f_{n-1}(k) + K_{n-1}(k) \, b_{n-1}(k-1) \end{aligned}$$

$$\begin{aligned} b_n(k) &= y(k-n) - \sum_{i=1}^{n} K_i \, y(k-n+i) \\ &= b_{n-1}(k-1) + K_{n-1}(k) \, f_{n-1}(k) \end{aligned}$$

where,

$K_n(k)$ $\rightarrow$ Reflection coefficient for the n-th stage of the lattice.

The backward error signals b_n are used as inputs to the tap weights.

The output of the equalizer is

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

This, mean square error is minimized using lattice filter; desired response $\hat{d}_k$ is obtained as output.

Advantages

1. It is simple and easy to implement.
2. It has ***numerical stability*** and ***faster convergence***.
3. Unique structure of the lattice filter ***allows the dynamic assignment*** of the most effective length of the lattice equalizer. If the channel not more time dispersive, only a fraction of stages are used.

Disadvantages of lattice equaliser

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

2. The structure of lattice equalizer is more complicated than a linear transversal equalizer.

3. Not suitable for severely distorted channel.

4.4. NONLINEAR EQUALIZATION

1. Describe in detail about non-linear equalizers. (Nov/Dec 2017, Nov/Dec 2014, Nov/Dec 2012, May 2012, Nov 2011, Nov 2010) [April/May 2018]
2. With valid statements, analytically prove that the adaptive equalizers exhibit superior performance over the conventional equalizers. (Nov/Dec 2017)
3. Describe the role played by equalization and diversity as multipath mitigation technique. Compare and contrast these two techniques. (Apr/May 2017)

- ✓ Nonlinear equalizers are used in applications where the ***channel distortion is too severe***.
- ✓ Nonlinear equalizers perform well on channels which have deep spectral nulls in the pass band.
- ✓ In such channels, the linear equalizers are not performing well and places too much gain in the vicinity of the spectral null. Thereby, enhancing the noise present in those frequencies.
- ✓ Three very effective nonlinear methods have been developed which offer improvements over linear equalization techniques and are used in most 2G and 3G systems.
- ✓ They are

1. Decision Feedback Equalization (DFE)
2. Maximum likelihood symbol detection
3. Maximum Likelihood Sequence Estimation (MLSE)

Decision Feedback Equalization (DFE)

- ✓ The Decision Feedback Equalizer (DFE) is particularly useful for channels with ***severe*** amplitude ***distortions*** and has been widely used in wireless communications.
- ✓ Symbol by symbol detection method.
- ✓ ***Basic idea in DFE:*** Once an information symbol has been detected and decided upon, the ISI contributed by these symbols can be estimated and subtracted out before detection of subsequent symbols.

- ✓ DFE can be realized by

  (*i*) Direct transversal form

  (*ii*) Lattice filter form

Direct transversal form of DFE

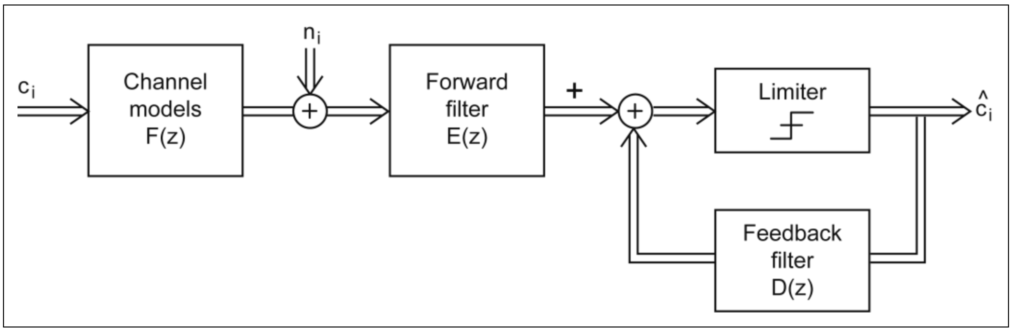


Figure 4.6. Structure of a direct transversal form- decision feedback equalizer (DFE)

- ✓ The direct form is shown in figure.
- ✓ It consists of a ***Feed Forward Filter (FFF)*** and ***a Feed Back Filter (FBF)***.
- ✓ FBF is driven by decisions on the output of the detector, and its coefficients can be adjusted to cancel the ISI on the current symbol from past detected symbols.
- ✓ The equalizer has $N_1 + N_2 + 1$ taps in the ***feed forward filter*** and N_3 taps in the *feedback filter*.

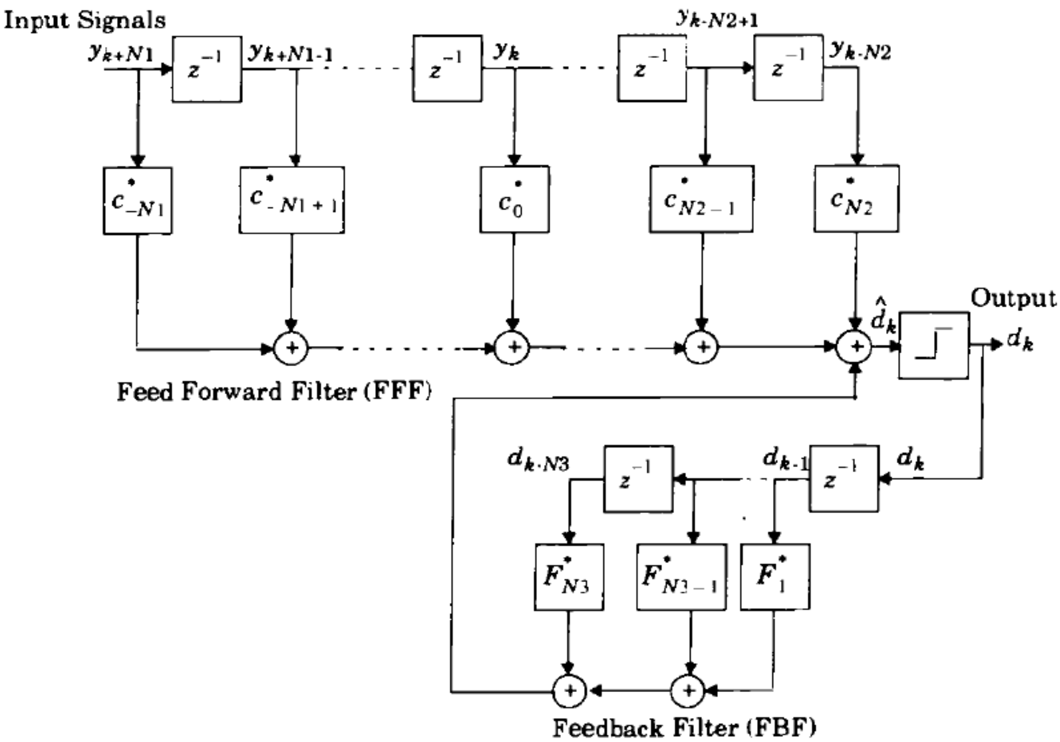


Figure 4.7: Decision Feedback Equalizer

- ✓ The output of the equalizer can be expressed as

$$\hat{d}_k = \sum_{n=-N_1}^{N_2} C_n^* \, y_{k-n} + \sum_{i=1}^{N_2} F_i^* \, d_{k-i} \qquad \ldots (1)$$

Where C_n^* → tap gains of the FFF filter

y_n → input to FFF

F_i^* → tap gains for the FBF

$d_i (i < k)$ → previous decision made on the detected signal

$\hat{d}_k$ is obtained from the above equation, d_k is decided from it.

d_k along with previous decisions $d_{k-1}, d_{k-2}, \ldots$ are fed back into the equalizer.

$\hat{d}_{k+1}$ is obtained from the above equation.

- ✓ ISI is computed based on the signal after the hard decision. So, additive noise is eliminated from the feedback signal.
- ✓ Minimize mean square error by striking a balance between noise enhancement and residual ISI.
- ✓ The Minimum Mean Square Error (MMSE) DFE is given by

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

- ➢ DFE has significantly smaller minimum MSE than an LTE unless $\left|F(e^{j\omega T})\right|$ is a constant (i.e., when adaptive equalization is not needed).
- ➢ If there are nulls in $\left|F(e^{j\omega T})\right|$, a DFE has significantly smaller minimum MSE than an LTE.
- ➢ Therefore, an LTE is well behaved when the channel spectrum is flat, but if the channel is severely distorted or exhibit null in the spectrum the performance of an LTE deteriorates and the mean squared error of a DFE is much better than a LTE.
- ➢ Thus, DFE is more appropriate for severely distorted wireless channels.
- ➢ The lattice implementation of the DFE is equivalent to a transversal DFE having a feed forward filter of length N1 and a feedback filter of length N2, where N1 > N2.

Maximum likelihood symbol detection

- ➢ Another form of DFE proposed by Belfiore and Park is called a ***predictive filter.***
- ➢ It also consists of a feed forward filter (FFF) as in the conventional DFE.

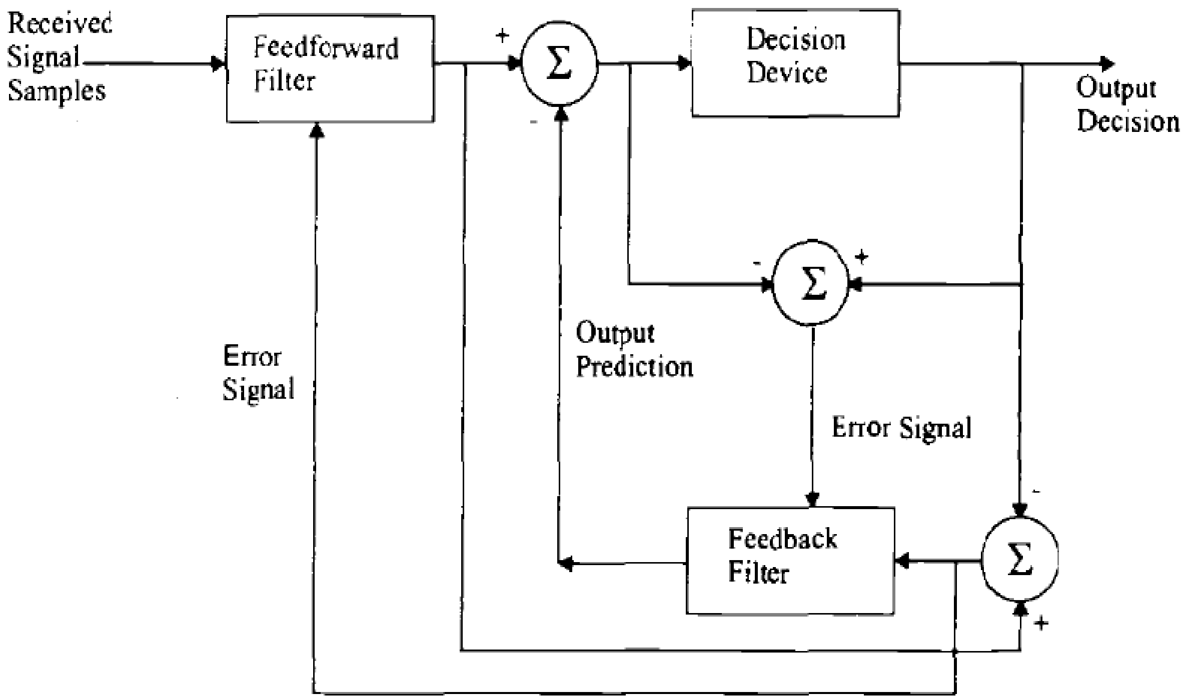


Figure 4.8: Predictive decision feedback equalizer.

- ✓ However, the feedback filter (FBF) is driven by an input sequence formed by the difference of the output of the detector and the output of the feed forward filter.
- ✓ Hence, the FBF here is called a *noise predictor* because it predicts the noise and the residual ISI contained in the signal at the FFF output and subtracts from it the detector output after some feedback delay.
- ✓ The predictive DFE performs as well as the conventional DFE as the limit in the number of taps in the FFF and the FBF approach infinity.
- ✓ The FBF in the predictive DFE can also be realized as a lattice structure.
- ✓ The RLS lattice algorithm can be used in this case to yield fast convergence.
- ✓ The MSE-based linear equalizers are optimum with respect to the criterion of minimum probability of symbol error when the channel does not introduce any amplitude distortion.

Maximum Likelihood Sequence Estimation (MLSE) Equalizer

- ✓ Using the MLSE as an equalizer was first proposed by Forney in which a basic MLSE estimator structure was implemented with the Viterbi algorithm.

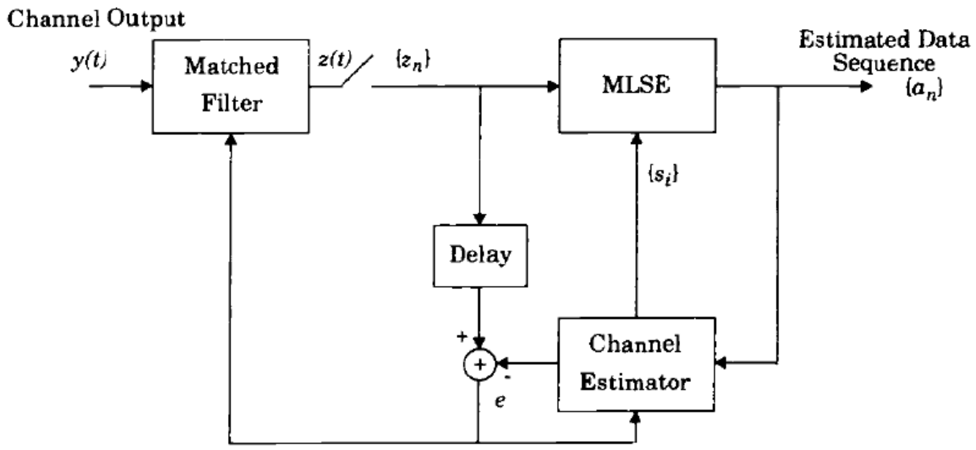


Figure 4.9: The structure of a maximum likelihood sequence estimator (MLSE) with an adaptive matched filter.

- ✓ The MLSE is optimal in the sense that it minimizes the probability of a sequence error.
- ✓ Using a channel impulse response simulator within the algorithm, the MLSE tests all possible data sequences (rather than decoding each received symbol by itself), and chooses the data sequence with the maximum probability as the output.
- ✓ An MLSE usually has a large computational requirement, especially when the delay spread of the channel is large.
- ✓ Thus the channel has ML states, where M is the size of the symbol alphabet of the modulation.
- ✓ That is, a trellis is used by the receiver to model the channel over time.
- ✓ The Viterbi algorithm then tracks the state of the channel by the paths through the trellis and gives at stage k a rank ordering of the M' most probable sequences terminating in the most recent L symbols.

- ✓ The MLSE requires knowledge of the channel characteristics in order to compute the metrics for making decisions.
- ✓ The MLSE also requires knowledge of the statistical distribution of the noise corrupting the signal. Thus, the probability distribution of the noise determines the form of the metric for optimum demodulation of the received signal.
- ✓ The matched filter operates on the continuous time signal, whereas the MLSE and channel estimator rely on discretized (nonlinear) samples.

4.5. ALGORITHMS FOR ADAPTIVE EQUALIZATION -ZERO FORCING AND LMS ALOGORITHMS

1. **Derive the LMS algorithm for an adaptive equalizer. (Nov/Dec 2016, Nov/Dec 2015)**
2. **Describe any two adaptation algorithms for mean square error equalizers. (May/June 2013)**

- ✓ An adaptive equalizer compensates for an unknown and time-varying channel, it requires a specific algorithm to update the equalizer coefficients and track the channel variations.
- ✓ In order to find the optimum equalizer weights, iterative algorithms have been developed.
- ✓ The quality of an iterative algorithm is described by

1. ***Convergence rate:*** This is the number of iterations required for the algorithm, in response to stationary inputs, to coverage enough to the optimum solution. A fast rate of convergence allows the algorithm to adapt rapidly to a stationary environment of unknown statistics. Furthermore, it enables the algorithm to track statistical variations when operating in a non-stationary environment.
2. ***Misadjustment:*** This is the quantitative measure of deviation of the *final value of the mean square error* (averaged over an ensemble of adaptive filters) and the *optimal minimum mean square error solution*.
3. ***Computational complexity:*** This is the number of operations required to make one complete iteration of the algorithm.
4. ***Numerical properties:*** When an algorithm is implemented numerically, inaccuracies are produced due to round-off noise and representation errors in the computer. These kinds of errors influence the stability of the algorithm.

- ✓ The choice of algorithm, and its corresponding rate of convergence, depends on the channel data rate and coherence time.
- ✓ An equalizer can only equalize over delay intervals less than or equal to the maximum delay within the filter structure.
- ✓ For example, if each delay element in an equalizer offers a 10 microsecond delay and 4 delay elements are used to provide a 5 tap equalizer.
- ✓ The maximum delay spread = 4 x 10 µs = 40 µs.

Three classic equalizer algorithms are

1. The zero forcing (ZF) algorithm,
2. The least mean squares (LMS) algorithm, and
3. The recursive least squares (RLS) algorithm.

1. Zero Forcing Algorithm

- ✓ In a zero forcing equalizer, the equalizer coefficients C_n are chosen to force the samples of the combined channel and equalizer impulse response to zero at all but one of the NT spaced sample points in the tapped delay line filter.
- ✓ By letting the number of coefficients increase without bound, an infinite length equalizer with zero ISI at the output can be obtained.
- ✓ When each of the delay elements provide a time delay equal to the symbol duration T, the frequency response $H_{eq}(f)$ of the equalizer is periodic with a period equal to the symbol rate $\frac{1}{T}$.
- ✓ The combined response of the channel with the equalizer must satisfy Nyquist's first criterion.

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

$H_{ch}(f) \rightarrow$ Folded frequency response of the channel.

- ✓ Thus, an infinite length, zero, ISI equalizer is simply an inverse filter which inverts the folded frequency response of the channel.
- ✓ This infinite length equalizer is usually implemented by a truncated length version.
- ✓ Zero forcing DFE is a simple filter.
- ✓ It performs well for static channels with high SNR. *e.g.* Local wired telephone lines.
- ✓ But noise power is enhanced after equalization.
- ✓ The ZF equalizer thus neglects the effect of noise altogether, and is not often used for wireless links.

2. Least Mean Square Algorithm

- ✓ LMS algorithm is used to minimize the mean square error (MSE) between the desired equalizer output and the actual equalizer output.

$$e_k = d_k - \hat{d}_k = x_k - \hat{d}_k$$

$$e_k = x_k - y_k^T w_k = x_k w_k^T y_k$$

- ✓ To compute the mean square error $E\left[|e_n|^2\right]$ at time instant k, is squared to obtain

$$\xi = E\left[e_k^* e_k\right]$$

- ✓ To minimize MSE the derivative is set to zero

$$\frac{\partial}{\partial w_N} J(w_N) = -2p_N + 2R_{NN} w_N = 0$$

where w_n is the tap gain vector, $J(w_n)$ is the cost function, R_{NN} is the deterministic correlation matrix of input data of the equalizer

$$R_{NN}\hat{w}_N = p_N$$

- ✓ This equation is a classic result, and is called the normal equation, since the error is minimized and is made orthogonal (normal) to the projection related to the desired signal x_k.

- ✓ When this equation is satisfied, the MMSE of the equalizer is

$$J_{opt} = J(\hat{w}_N) = E\left[x_k^* x_k\right] - p_N^T \hat{w}_N$$

- ✓ To obtain the optimal tap gain vector $\hat{w}_N$, the normal equation $R_{NN}\hat{w}_N = p_N$ must be solved iteratively as the equalizer converges to an acceptably small value of $J_{opt.}$
- ✓ There are several ways to do this, and many variants of the LMS algorithm have been built upon the solution of equation $J_{opt} = J(\hat{w}_N) = E\left[x_k^* x_k\right] - p_N^T \hat{w}_N$.
- ✓ One obvious technique is to calculate

$$\hat{w}_N = R_{NN}^{-1}\ p_N$$

- ✓ In practice, the minimization of the MSE is carried out recursively by use of the stochastic gradient algorithm commonly called the Least Mean Square (LMS) algorithm.
- ✓ The LMS algorithm is the simplest equalization algorithm and requires only 2N + 1I operations per iteration. The filter weights are updated by the update equations given below

$$\hat{d}_k(n) = w_N^T\ y_N(n)$$

$$e_k(n) = x_k(n) - \hat{d}_k(n)$$

$$w_N(n+1) = w_N(n) - \alpha\ e_k^*(n)\ y_N(n)$$

where the subscript N denotes the number of delay stages in the equalizer, and α is the step size which controls the convergence rate and stability of the algorithm.

- ✓ The LMS equalizer maximizes the signal to distortion ratio at its output within the constraints of the equalizer filter length.
- ✓ If an input signal has a time dispersion characteristic that is greater than the propagation delay through the equalizer, then the equalizer will be unable to reduce distortion.
- ✓ The convergence rate of the LMS algorithm is slow due to the fact that there is only one parameter, the step size α, that controls the adaptation rate.
- ✓ To prevent the adaptation from becoming unstable, the value of α is chosen from

$$0 < \alpha > 2 / \sum_{i=1}^{N} \lambda_i$$

where λ_i is the ith eigenvalue of the covariance matrix R_{NN}.

$$\sum_{i=1}^{N} \lambda_i = y_N^T(n)\ y_N(n)$$

The step size α can be controlled by the total input power in order to avoid instability in the equalizer.

4.6 DIVERSITY

1. Write short notes on spatial, temporal, polarization and Macro diversity. (Apr/May 2015)
2. Explain in detail about space diversity with necessary diagrams. (Nov/Dec 2016, Nov/Dec 2015)
3. Describe the role played by equalization and diversity as multipath mitigation technique. Compare and contrast these two techniques. (Apr/May 2017)
4. Analyze various diversity techniques used in wireless communications. (Nov/Dec 2017, Nov/Dec 2012)

Introduction of diversity

- ✓ Diversity is a powerful communication receiver technique that provides wireless link improvement at relatively low cost.
- ✓ Unlike equalization, diversity requires no training overhead since a training sequence is not required by the transmitter.
- ✓ Diversity implementations provides significant link improvement with little added cost.
- ✓ Diversity exploits the random nature of radio propagation by finding independent signal paths for communication.
- ✓ In virtually all applications, diversity decisions are made by the receiver, and the decisions are unknown to the transmitter.

Diversity concept

Signal is transmitted by more than one antenna via channel. If one radio path undergoes a deep fade, another independent path may have a strong signal. So the receiver has multiple copies of the transmit signal. By having more than one path to select from, both the instantaneous and average SNRs at the receiver may be improved, often by as much as 20 dB to 30 dB.

Uses of Diversity Technique

1. It is a technique used to compensate for fading channel impairments.
2. It improves the quality of a wireless communication link without increasing the transmitting power and bandwidth.

Need of diversity technique

- ✓ For AWGN – Additive White Gaussian Noise channels, BER decreases exponentially as the SNR increases. 10 dB SNR leads to BERs on the order of 10^{-4}.
- ✓ For Rayleigh fading channel, BER decreases linearly with the SNR. To achieve a 10^{-4} BER, SNR on the order of 40 dB is required, which is clearly unpractical. So, we go for diversity.

Fading means rapid fluctuations of the amplitudes, phase or multipath delays of a radio signal over a short period of time or travel distance.

Types of fading

1. Small scale fading.
2. Large scale fading.

1. Small scale fadings

Small scale fading are characterized by deep and rapid amplitude fluctuations which occur as the mobile moves over distances of just a few wavelengths.

- ✓ Fading is caused by multiple reflections from the surroundings in the vicinity of the mobile.

  e.g. Narrow band signal undergoes Rayleigh fading.

- ✓ To prevent deep fading, ***microscopic diversity*** techniques are used.

For example,

- ✓ In antenna diversity or space diversity, two antennas are separated by a fraction of a meter, one may receive weak signal while the other receives a strong signal.

By selecting the best signal at all times, a receiver can mitigate small scale fading effects

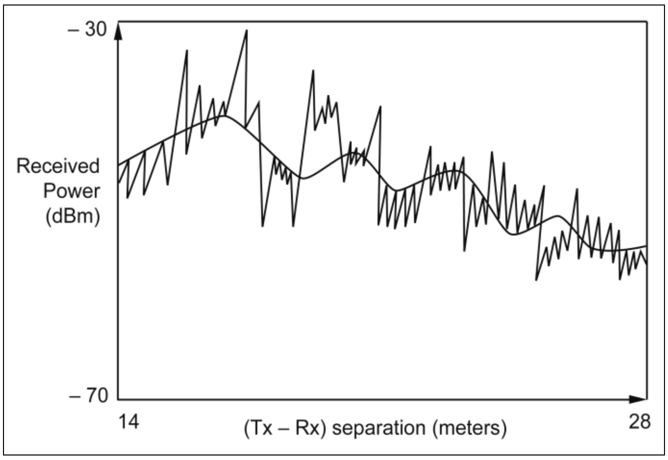


Figure 4.11. Small scale fading and Large scale fading

II. Large scale fadings

- ✓ It is caused by shadowing due to variations in both the terrain profile and the nature of the surroundings.
- ✓ At deep shadow, the received signal strength at a mobile can drop well below that of free space.
- ✓ To prevent this drop, ***macroscopic diversity*** technique is used.

Definition of correlation coefficient

- ✓ Diversity is most efficient when the different transmission channels carry independently fading copies of the same signal. This means that the joint probability density function of field strength $pdf_{r_1, r_2, \ldots}(r_1, r_2, \ldots)$ is equal to the product of the marginal pdfs for the channels, $pdf\ (r_1)$, $pdf\ (r_2)$.
- ✓ The correlation coefficient characterizes the correlation between signals on different diversity branches.
- ✓ The correlation coefficient of signal envelopes x and y.

$$\rho_{xy} = \frac{E\{x \cdot y\} - E\{x\} \cdot E\{y\}}{\sqrt{(E\{x^2\} - E\{x\}^2) \cdot (E\{y^2\} - E\{y\}^2)}}$$

- ✓ For two statistically independent signals,

$$E\{x\,y\} = E\{x\}\,E\{y\}.$$

∴ Correlation coefficient becomes zero.

- ✓ If $\rho < 0.5$ or 0.7, signals are effectively decorrelated.

4.6.1.MICRO DIVERSITY

1. **Analyze various diversity techniques used in wireless communication. (Nov/Dec 2017, Nov/Dec 2014)**
2. **Write short notes on: (i) Spatial Diversity(May 2016) (ii)Frequency Diversity (May 2016) (iii) Polarization Diversity(May 2016) (iv) Time Diversity(May 2016, Nov 2013, May 2013, May 2012, May 2010)**
3. **What is the need for diversity? List different types of diversity. (May/june 2014)**

- ✓ The ***principle of diversity*** is that the RX has multiple copies of the transmit signal, where each of the copies goes through a statistically independent channel.
- ✓ This section describes different ways of obtaining these statistically independent copies.
- ✓ The receiver has multiple copies of the transmit signal, where each of the copies goes through a statistically independent channel.
- ✓ By selecting the best signal at all times a receiver can reduce small scale fading effects.
- ✓ The five most common methods are

1. Spatial diversity → Several antenna elements separated in space.
2. Temporal diversity → Repetition of the transmit signal at different times.
3. Frequency diversity → Transmission of the signal on different frequencies.
4. Angular diversity → Multiple antennas (with or without spatial separation) with different antenna patterns.
5. Polarization → Multiple antennas receiving different polarizations (e.g., vertical and horizontal).

A. SPATIAL DIVERSITY

- ✓ Spatial diversity is the oldest and simplest form of diversity.
- ✓ So, it is the most widely used technique.
- ✓ ***Principle:*** The transmit signal is received at several antenna elements, and the signals from these antennas are then further processed.
- ✓ But, irrespective of the processing method, performance is influenced by ***correlation of the signals between the antenna elements***.
- ✓ A large correlation between signals at antenna elements is undesirable, as it decreases the effectiveness of diversity.
- ✓ A first important step in designing diversity antennas is to establish a relationship between ***antenna spacing and the correlation coefficient***.
- ✓ This relationship is different for BS antennas and MS antennas, and thus will be treated separately.

This relationship is different for BS antennas and MS antennas those are as follows:

(a) Mobile station in cellular and cordless systems

- ✓ It is a standard assumption that waves are incident from all directions at the MS.
- ✓ So, points of constructive and destructive interference of Multi Path Components (MPCs) [i.e., points of high and low received power, respectively] are spaced approximately $\lambda/4$ apart.
- ✓ Therefore, this is the distance that is required for decorrelation of received signals.
- ✓ This awareness agrees very well with the results from the exact mathematical derivation.

(b) BS in cordless systems and WLANs:

- ✓ The angular distribution of incident radiation at indoor BSs is also uniform [i.e., radiation is incident with equal strength from all directions.
- ✓ Therefore, the same rules apply as for MSs.

(c) BSs in cellular systems:

- ✓ For a cellular BS, the assumption of uniform directions of incidence is not valid.
- ✓ Interacting Objects (IOs) are typically concentrated around the MS (Figure 4.12).
- ✓ All waves are incident essentially from one direction`
- ✓ So, the correlation coefficient (for a given distance between antenna elements d_a) is much higher.
- ✓ That is, the antenna spacing required to obtain sufficient decorrelation increases.

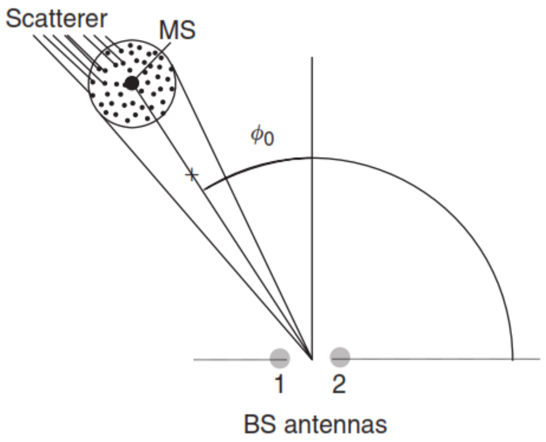


Figure 4.12: Scatterers concentrated around the mobile station.

- ➢ They are classified as

I. Selection diagram → Switched diversity
→ Feedback diversity

II. Combining diversity → Maximal ratio combining
→ Equal gain diversity

B.TEMPORAL DIVERSITY

Principle:

- ✓ As the wireless propagation channel is time variant, signals that are received at different times are uncorrelated. For sufficient decorrelation, the temporal distance must be at least $\frac{1}{(2\,\nu_{max})}$, where ν_{max} is the maximum Doppler frequency.
- ✓ Temporal diversity can be realized in different ways.
  - (*i*) Repetition coding
  - (*ii*) Automatic repeat request
  - (*iii*) Combination of interleaving and coding

(i) Repetition coding

- ✓ This is the simplest form of linear block codes.
- ✓ The signal is repeated several times, where the repetition intervals are long enough to achieve decorrelation.
- ✓ This obviously achieves diversity, but is also highly bandwidth inefficient.
- ✓ Spectral efficiency decreases by a factor that is equal to the number of repetitions.

(ii) Automatic Repeat Request (ARQ)

- ✓ The RX sends a message to the transmitter to indicate whether it received the data with sufficient quality. If this is not the case, then the transmission is repeated.
- ✓ The spectral efficiency of ARQ is better than that of repetition coding, since it requires multiple transmissions only when the first transmission occurs in a bad fading state, while for repetition coding, retransmission occur always.
- ✓ On the downside, ARQ requires a feedback channel.

(iii) Combination of Interleaving and Coding

- ✓ A more advanced version of repetition coding is forward error correction coding with interleaving.
- ✓ The different symbols of a codeword are transmitted at different times, which increase the probability that at least some of them arrive with a good SNR.
- ✓ When only the MS is moving, while the interacting objects (IOs) and the BS are fixed, temporal correlation can be converted into spatial correlation the correlation coefficient is $\rho = 1$ for all time intervals, and temporal diversity is useless.

C.FREQUENCY DIVERSITY

- ✓ ***Principle:*** In frequency diversity, the same signal is transmitted at two (or more) different frequencies.
- ✓ Frequency diversity is implemented by transmitting information on more than one carrier frequency.
- ✓ If these frequencies separated by more than the coherence bandwidth of the channel, then their fading is approximately independent, and the probability is low that the signal is in a deep fade at both frequencies simultaneously.
- ✓ For an exponential PDP, the correlation between two frequencies can be from the following equation by setting the numerator to unity as the signals at the two frequencies occur at the same time.

$$\rho_{xy} = \frac{J_0^2(k_0 \mathrm{v} \tau)}{1 + (2\pi)^2 S_\tau^2 (f_2 - f_1)^2}$$

Thus,

$$\rho = \frac{1}{1 + (2\pi)^2 S_\tau^2 (f_2 - f_1)^2}$$

- ✓ This again confirms that the two signals have to be at least one coherence bandwidth apart from each other.
- ✓ Figure shows ρ as a function of the spacing between the two frequencies.

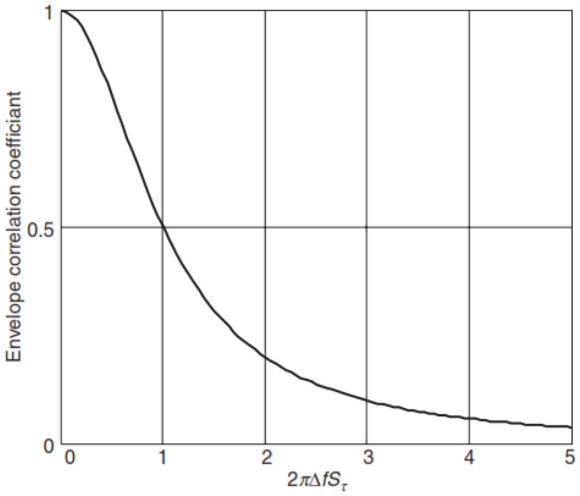


Fig. 4.13. Correlation coefficient of the envelope as a function of normalized frequency spacing.

- ✓ It is not common to actually repeat the same information at two different frequencies, as this would greatly decrease spectral efficiency.
- ✓ Rather information is spread over a large bandwidth, so that small parts of the information are conveyed by different frequency components.
- ✓ The RX can then average over the different frequencies to recover the original information.

This spreading can be done by different methods.

(i) Compressing the information in time that is, sending short burst that each occupy a large bandwidth - *e.g.* TDMA

(ii) Code Division Multiple Access – CDMA.

(iii) Multi carrier CDMA and coded orthogonal frequency division multiplexing.

(iv) Frequency hopping in conjunction with coding.

These methods allow the transmission of information without wasting bandwidth.

The use of frequency diversity requires the channel to be ***frequency selective***.

Advantages of frequency diversity

(*i*) By using redundant signal transmission, this diversity improve link transmission quality.

(*ii*) New OFDM modulation uses frequency diversity.

Disadvantages

(*i*) It requires large bandwidth.

(*ii*) More number of receivers are required.

(*iii*) High cost.

D.<u>ANGULAR DIVERSITY</u>

Principle: Two co-located antennas with different patterns “see” differently weighted MPCs (Multi Path Components), so that the MPCs interfere differently for the two antennas. This is the principle of angle diversity (also known as ***pattern diversity***).

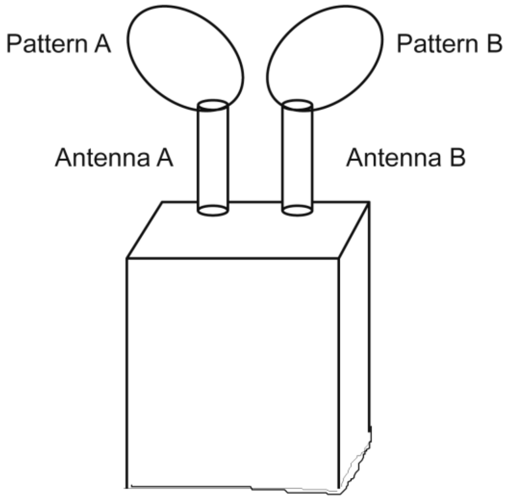


Figure 4.34. Angle diversity for closely spaced antennas

- ✓ Angular diversity is usually used in conjunction with spatial diversity. It enhances the decorrelation of signals at closely spaced antennas.
- ✓ Different antenna patterns can be achieved very easily.
- ✓ But even identical antennas can have different patterns when mounted close to each other.
- ✓ This effect is due to mutual coupling: antenna B acts as a reflector for antenna A, whose pattern is therefore skewed to the left.
- ✓ Analogously, the pattern of antenna B is skewed to the right due to reflections from antenna A. Thus the two patterns are different.

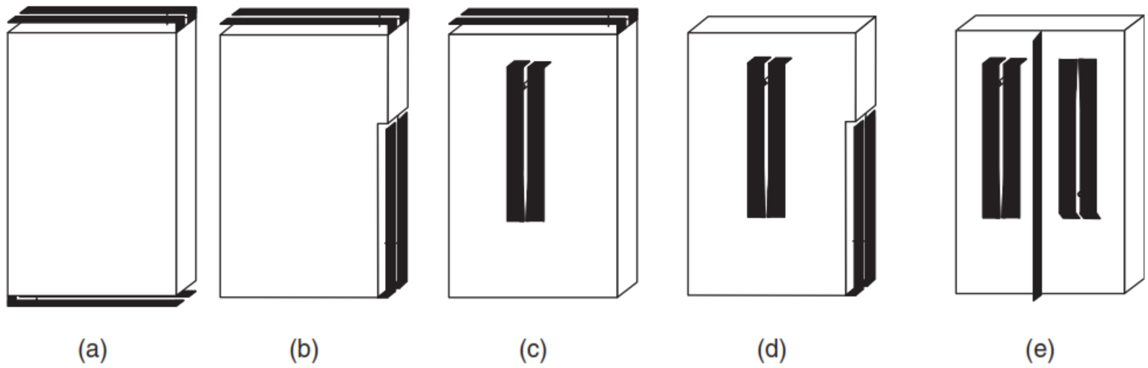


Figure 4.15. Configurations of diversity antennas at a mobile station.

- ✓ Example:
  - ✓ The different patterns are even more visible when the antennas are located on different parts of the casing.
  - ✓ While dipole antennas are usually restricted to the top of the casing, patch antennas and inverted–F antennas can be placed on all parts of the casing (above Figure 4.15).
  - ✓ In all of these cases, decorrelation is good even if the antennas are placed very closely to each other.

E. POLARIZATION DIVERSITY

Principle:

In Transmitter side, two diversity branches are used. Signals are passed through two orthogonally polarized propagation path.

In Receiver side, antenna with two elements receives the vertical or horizontally polarized signal.

- ✓ At the base station, space diversity is considerably less practical than at the mobile because the narrow angle of incident fields requires large antenna spacings.
- ✓ The comparatively high cost of using space diversity at the base station prompts the consideration of using orthogonal polarization to exploit polarization diversity.
- ✓ Measured horizontal and vertical polarization paths between a mobile and a base station are reported to be uncorrelated.
- ✓ The decorrelation for the signals in each polarization is caused by multiple reflections in the channel between the mobile and base stations antennas.
- ✓ The reflection coefficient for each polarization is different which results in different amplitudes and phases for each or at least some of the reflection.
- ✓ After sufficient random reflections, the polarization state of the signal will be independent of the transmitted polarization.
- ✓ Circular and linear polarized antennas have been used to characterize multipath inside buildings.
- ✓ When the path was obstructed, polarization diversity was found to dramatically reduce the multipath delay spread without significantly decreasing the received power.

Theoretical model for polarization diversity

- ✓ It is assumed that the signal is transmitted from a mobile with vertical polarization.
- ✓ It is received at the base station by a polarization diversity antenna with 2 branches.
- ✓ Figure shows the theoretical model and the system coordinates.
- ✓ As seen in the figure, a polarization diversity antenna is composed of two antenna elements V_1 and V_2, which make $\pm\alpha$ angle (polarization angle) with the Y axis.
- ✓ A mobile station is located in the direction of offset angle β from the main beam direction of the diversity antenna as seen in Figure 4.16 (b).

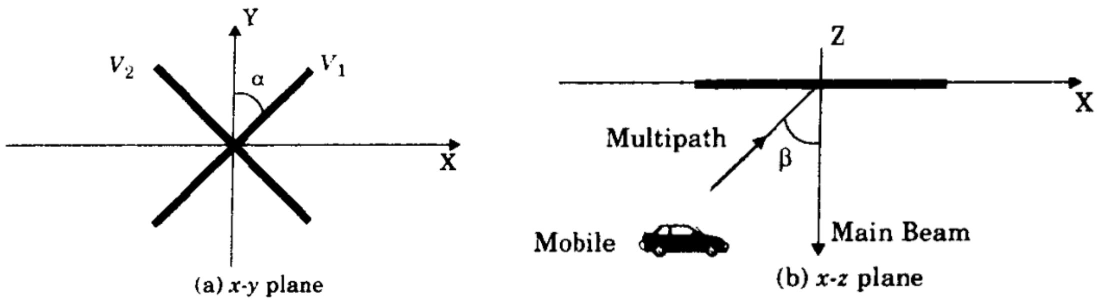


Figure 4.16. Theoretical model for base station polarization diversity

- ✓ Some of the vertically polarized signals transmitted are converted to the horizontal polarized signal because of multipath propagation.
- ✓ The signal arriving at the base station can be expressed as

$$x = r_1 \cos(\omega t + \phi_1)$$

$$y = r_2 \cos(\omega t + \phi_1)$$

Where x and y are signal levels which are received when $\beta = 0$.

- ✓ Assume r_1 and r_2 have independent Rayleigh distributions and ϕ_1 and ϕ_2 have uniform distributions.

The received signal values at V_1 and V_2 are

$$V_1 = (a\, r_1 \cos \phi_1 + r_2\, b \cos \phi_2) \cos \omega t - (a\, r_1 \sin \phi_1 + r_2 \sin \phi_2) \sin \omega t$$

$$V_2 = (-a\, r_1 \cos \phi_1 + r_2\, b \cos \phi_2) \cos \omega t - (-a\, r_1 \sin \phi_1 + r_2\, b \sin \phi_2) \sin \omega t$$

where, $a = \sin \alpha \cos \beta$

$b = \cos \alpha$

✓ Correlation coefficient $\rho = \left(\dfrac{\tan^2 (\alpha) \cos^2 (\beta) - \Gamma}{\tan^2 (\alpha) \cos^2 (\beta) + \Gamma} \right)^2$

where

$$\Gamma = \frac{\langle R_2^2 \rangle}{\langle R_1^2 \rangle}$$

where,

$$R_1 = \sqrt{r_1^2 a^2 + r_2^2 b^2 + 2 r_1 r_2 ab \cos(\phi_1 + \phi_2)}$$

$$R_2 = \sqrt{r_1^2 a^2 + r_2^2 b^2 - 2 r_1 r_2 ab \cos(\phi_1 + \phi_2)}$$

Here, Γ - the cross polarization discrimination of the propagation path between a mobile and a base station.

ρ is determined by three factors.

(*i*) Polarization angle

(*ii*) Offset angle from the main beam direction of the diversity antenna.

(*iii*) The cross polarization discrimination (Γ).

✓ When polarization angle α increases ρ becomes lower, then horizontal polarization component increases.

✓ Signal loss $L = \dfrac{a^2}{\Gamma + b^2}$ relative to vertical polarization.

Advantage

Multipath delay spread is reduced.

4.6.2. MACRO DIVERSITY

Significance of macro diversity

✓ The Micro diversity methods that combat small scale fading which is created by interference effects.

✓ These diversity methods are not suitable for combating large scale fading, which is created by shadowing effects.

✓ The correlation distances for large scale fading are on the order of tens or hundreds of meters, so that temporal diversity or spatial diversity cannot be used.

✓ So, to reduce large scale fading, macro diversity is used.

✓ In macro diversity, a large distance between BS_1 and BS_2 is maintained.

✓ ***On-frequency repeaters*** that receive a signal and retransmit an amplified version of it is used.

✓ The same signal is transmitted simultaneously from different BSs called ***simulcast.***

✓ Two BSs should be synchronized in cellular applications. *e.g.* digital TV.

Advantage

1. To compensate for large scale fading effects macro diversity technique is used.
2. Distance between each BS is increased.
3. On-frequency repeaters (or) simulcast methods are used.

Drawbacks

1. Simulcast requires large amount of signaling information that has to be carried on landlines, so it requires large bandwidth.
2. On-frequency repeater causes delay dispersion.

4.7.DIVERSITY COMBINING TECHNIQUES (or) COMBINATION OF SIGNALS

1. **Explain with the block diagram, maximal ratio combiner. (Nov/Dec 2014)**
2. **Explain with diagram, the different techniques available for signal combining. (May/June 2014)**

✓ By combining signals from different antenna at the RX, the total quality of the signal is improved.

✓ Signals selected from the multiple diversity branches by

a) Selection diversity: The "best" signal copy is selected and processed (demodulated and decoded) while all other copies are discarded.

b) Combining diversity: All copies of the signal are combined before or after the demodulation and the combined signals are decoded.

✓The gain of multiple antennas is due to two effects.

1. Diversity gain
2. Beam forming gain

✓ ***Diversity gain*** reflects the fact that it is improbable that several antenna elements are in a fading dip simultaneously. Thus probability of error is very low.

✓ ***Beam forming gain*** reflects the fact that the combiner performs an averaging over the noise at different antennas. Thus, even if the signal levels at all antenna elements are identical, the combiner output SNR is larger than the SNR at a single antenna element.

1. SELECTION DIVERSITY

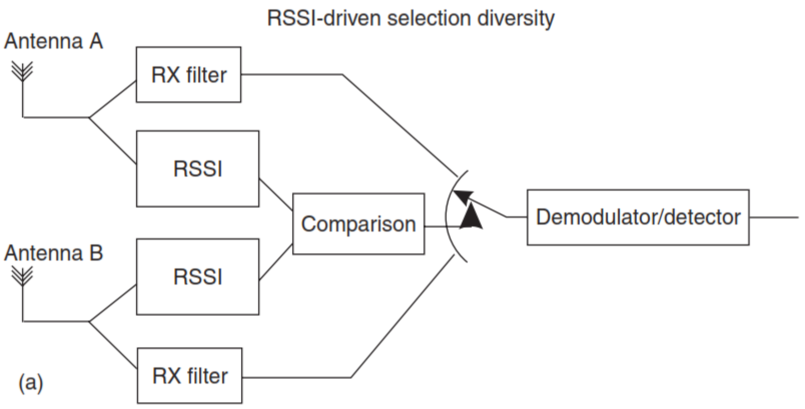
(i) Received – signal – strength – indication – driven diversity

✓***Principle:*** The RX selects the signal with the largest instantaneous power or RSSI – Received Signal Strength Indication.

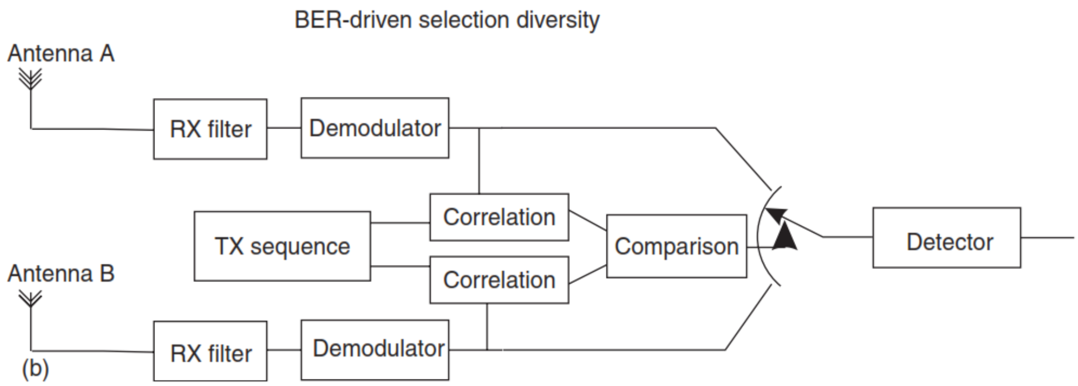
✓ The required components are

- ❖ N_r antenna elements
- ❖ N_r RSSI sensors and
- ❖ N_r to 1 multiplexer

- only one RF chain are used.



(a) Receiver – signal – strength – indication controlled diversity



(b) Bit error rate controlled diversity

Fig. 4.37. Selection diversity principle

- ✓ If the BER is determined by noise, then RSSI – driven diversity is the best of all the diversity methods. Maximization of RSSI also maximizes the SNR.
- ✓ If BER is determined by co-channel interference, then RSSI is no longer a good criterion.

  e.g. In FDMA and CDMA system, interference is caused by one dominant interferer.
- ✓ SSI driven diversity is suboptimum if the errors are caused by the frequency selectivity of the channel.
- ✓ For Rayleigh distributed channel,

SNR of the n^{th} diversity branch γ_n is

$$\mathrm{pdf}_{\gamma_n}(\gamma_n) = \frac{1}{\overline{\gamma}} \exp\left(-\frac{\gamma_n}{\overline{\gamma}}\right)$$

where $\overline{\gamma}$ is the mean branch SNR.

The cumulative distribution function (cdf) is

$$\mathrm{cdf}_{\gamma_n}(\gamma_n) = 1 - \exp\left(-\frac{\gamma_n}{\overline{\gamma}}\right)$$

RX selects the branch with the largest SNR.

The *cdf* of the selected signal is the product of the *cdf*s of each branch.

$$\mathrm{cdf}_{\gamma}(\gamma) = \left[1 - \exp\left(-\frac{\gamma}{\bar{\gamma}}\right)\right]^{N_r}$$

Advantages

1. Only one RF chain is required. It is processed with only a single received signal at a time.
2. Easy to implement.

Drawbacks

1. It wastes signal energy by discarding (N_r – 1) copies of received signal.
2. It is not an optimum method.

(ii) Bit-error-rate driven diversity

- ✓ For BER-driven diversity, at first a training sequence is transmitted *i.e.* a bit sequence that is known at the RX.
- ✓ The RX demodulates the signal from each receive antenna elements and compares it with the transmit signal.
- ✓ The signal with the smallest BER is judged to be the "best" signal and used for the subsequent reception of data signals.
- ✓ If the channel is time-variant, the training sequence has to be repeated at regular intervals and selection of the best antenna has to be done now.

Drawbacks of BER driven diversity

- ✓ More number of RXs are needed, which makes the RX more complex.
- ✓ The training sequence has to be repeated N_r times, which decreases spectral efficiency.
- ✓ If the channel changes quickly, more than one demodulators are required.
- ✓ Duration of training sequence increases, BER decreases. So, tradeoff between duration of training sequence and BER is maintained.
- ✓ Diversity branches are monitored all the times, so hardware effort increases, spectral efficiency is reduced.

2. SWITCHED DIVERSITY

Principle: Active branch is monitored. If it falls below a certain threshold, then the RX switches to a different antenna. Switching depends only on the quality of the active diversity branch.

- ✓ If two branches have signal quality below the threshold, then the RX switches back and forth between the branches.
- ✓ Switching only depends on the quality of the active diversity branch.
- ✓ It does not matter whether the other branch actually provides a better signal quality or not.
- ✓ Switched diversity runs into problems when both branches have signal quality below the threshold.
- ✓ In that case, the RX just switches back and forth between the branches.
- ✓ This problem can be avoided by introducing hysteresis or hold time.
- ✓ So that the new diversity branch is used for a certain amount of time, independent of the actual signal quality.
- ✓ We thus have two free parameters:
    - o Switching threshold

- o Hysteresis time.
- ✓ These parameters have to be selected very carefully.
- ✓ Switching threshold:
  - ➢ If the threshold is chosen too low, then a diversity branch is used even when the other antenna might offer better quality.
  - ➢ If it is chosen too high, then it becomes probable that the branch the RX switches to actually offers lower signal quality than the currently active one.
- ✓ Hysteresis time:
  - ➢ If hysteresis time is chosen too long, then a "bad" diversity branch can be used for a long time.
  - ➢ If it is chosen too short, then the RX spends all the time switching between two antennas.

Drawback

- ✓ Performance of switched diversity is worse than that of selection diversity.

3. FEEDBACK DIVERSITY (OR) SCANNING DIVERSITY

- ✓ ***Feedback diversity*** is also called scanning diversity which is very similar to selection diversity except that instead of always using the best of M signals, the M signals are scanned in a fixed sequence until one is found to be above a ***predetermined threshold***. This signal is then received until it falls below a threshold and the scanning process is again initiated.

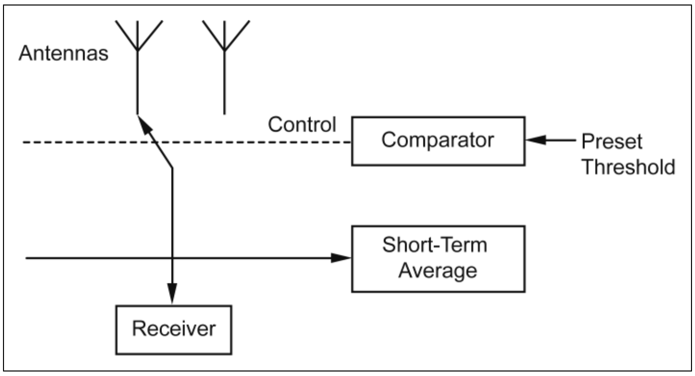


Fig. 4.48. Block diagram of feedback diversity

Advantages

(*i*) It is very simple to implement – only one receiver is required.

(*ii*) Low cost.

Drawback

The resulting fading statistics are somewhat inferior to those obtained by other methods.

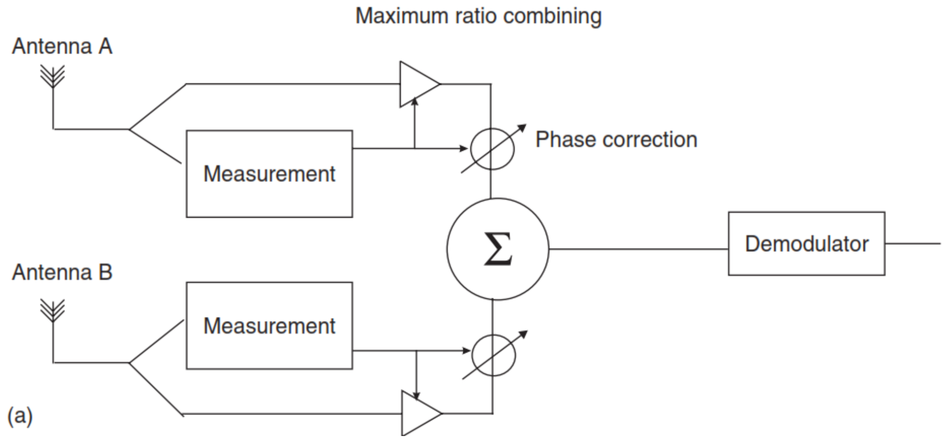
4 .COMBINING DIVERSITY

Basic Principle

- ✓ Selection diversity wastes signal energy by discarding (N- 1) copies of the received signal. This drawback is avoided by combining diversity, which exploits *all* available signal copies.
- ✓ Each signal copy is mulplied by a complex weight w_n^*, consisting of a phase correction, plus a (real) weight for the amplitude:, and then added up.
- ✓ ***Phase correction*** causes the signal amplitudes to add up, otherwise noise is added incoherently, so that noise powers add up.

- ✓ For amplitude weighting two methods are used:
  - ***(a) Maximal Ratio Combining (MRC)*** – weighs all signal copies by their amplitude. This is an optimum combination strategy.
  - ***(b) Equal gain combining (EGC)*** – signals are not weighted, but undergo a phase correction.

A.MAXIMUM RATIO COMBINING (MRC)



(a) Maximum ratio combining

Figure 4.19 Combining diversity principle

- ✓ MRC compensates for the phases, and weights of the signals from the different antenna branches according to their SNR. It is an optimum method.
- ✓ Channel is assumed as slow fading and fast fading. The channel is realized as a time-invariant filter with impulse response

$$h_n(\tau) = \alpha_n \delta(\tau)$$

Where, $\alpha_n \rightarrow$ attenuation diversity

$n \rightarrow$ branch

Antenna weight $= W_{MRC} = \alpha_n^*$

- ✓ The signals are phase-corrected and weighted by the amplitude.
- ✓ Output SNR of diversity combiner =Sum of the branch SNRs

$$\gamma_{MRC} = \sum_{n=1}^{N_r} \gamma_n$$

- ✓ If the branches are statistically independent, the SNR distribution in each branch is exponential.

$$pdf_\gamma(\gamma) = \frac{1}{(N_r - 1)!} \frac{\gamma^{N_r - 1}}{\bar{\gamma}^{N_r}} \exp\left(-\frac{\gamma}{\bar{\gamma}}\right)$$

$$\bar{\gamma}_{MRC} = N_r \bar{\gamma}$$

For $N_r = 1$ there is no diversity. For $N_r = 3$, diversity is applied.

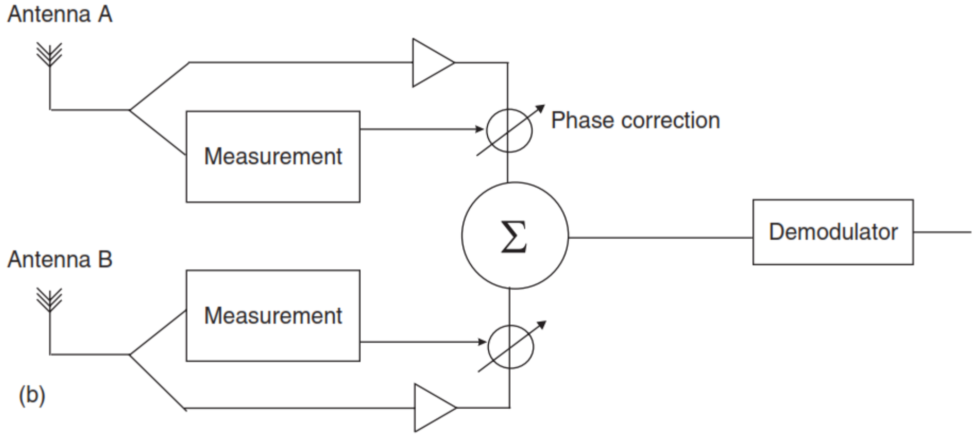
Advantages

- ✓ Output is produced with an acceptable SNR even when none of the individual signals are themselves acceptable.
- ✓ This technique gives the best statistical reduction of fading of any known linear diversity combiner.

Drawbacks

1. It requires individual receiver and phasing circuits for each antenna elements.
2. High cost.

B.EQUAL GAIN COMBINING (EGC)



(b) Equal gain combining

Figure 4.59. Combining diversity principle

- ✓ It is similar to maximal ratio combining except that the weighting circuits are omitted.
- ✓ The branch weights are all set of unity but the signals from each branch are co-phased to provide equal gain combining diversity.
- ✓ This allows the receiver to exploit the signals that are simultaneously received on each branch.

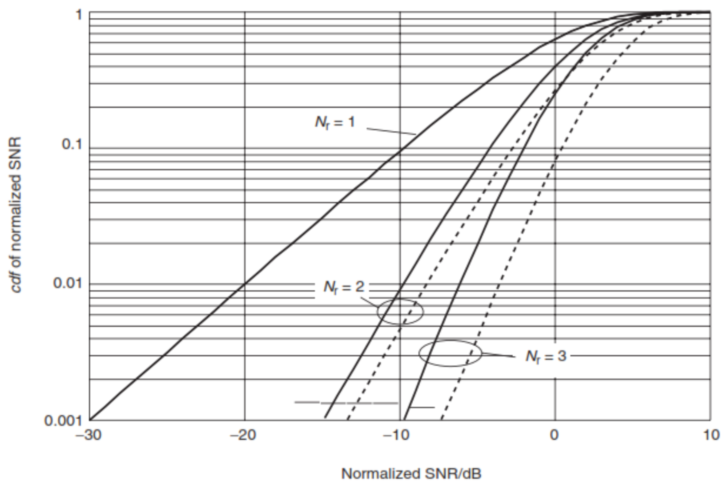


Figure 4.20. *Cumulative distribution function of the normalized instantaneous signal-to-noise ratio* $\gamma/\bar{\gamma}$ *for received-signal-strength-indication-driven selection diversity (solid), and maximum ratio combining (dashed) for Nr = 1, 2, 3. Note that for Nr =1, there is no difference between diversity types*

.

✓For EGC, the SNR of the combiner output is

$$\gamma_{\text{ECG}} = \frac{\left(\sum_{n=1}^{N_\text{r}} \sqrt{\gamma_n}\right)^2}{N_\text{r}}$$

✓The mean SNR is

$$\bar{\gamma}_{\text{EGC}} = \bar{\gamma}\left[1 + (N_r - 1)\frac{\pi}{4}\right]$$

for all branches suffer from ***Rayleigh fading***.

✓For N_r = 1, EGC performance = MRC performance.

✓For N_r = 2, 3 MRC is better than EGC.

Advantages

Equal gain combiner is superior to selection diversity.

Drawbacks

1. EGC is inferior to that of maximal ratio combiner, since interference and noise corrupted signals may be combined with high quality signals.
2. EGC performs worse than MRC by only a factor $\frac{\pi}{4}$.

4.8. ERROR PROBABILITY IN FADING CHANNELS WITH DIVERSITY RECEPTION

1. Analyze and compare the error performance in fading channels with and without diversity reception techniques. (Nov/Dec 2017, Apr/May 2017)
2. Determine the error probability for different fading channels with diversity reception. [April/May 2018].

- ✓ In this section we determine the Symbol Error Rate (SER) in fading channels when diversity is used at the RX.
- ✓ Starting with the case of flat-fading channels, computing the statistics of the received power and the BER.
- ✓ Then proceed to dispersive channels, where, how diversity can mitigate the detrimental effects of dispersive channels on simple RXs is analyzed.

Error Probability in Flat-Fading Channels

Classical Computation Method

- ✓ The error probability of diversity systems can be computed by averaging the conditional error probability (conditioned on a certain SNR) over the distribution of the SNR:

$$\overline{SER} = \int_0^\infty pdf_\gamma(\gamma) SER(\gamma)\, d\gamma \qquad \rightarrow \quad (1)$$

As an example, let us compute the performance of BPSK with N_r diversity branches with MRC.

The SER of BPSK in AWGN is

$$SER(\gamma) = Q(\sqrt{2\gamma}) \qquad \rightarrow \quad (2)$$

Let us apply this principle to the case of MRC.
When inserting

$$pdf_\gamma(\gamma) = \frac{1}{(N_r - 1)!}\frac{\gamma^{N_r-1}}{\overline{\gamma}^{N_r}} \exp\left(-\frac{\gamma}{\overline{\gamma}}\right)$$

and

$$SER(\gamma) = Q(\sqrt{2\gamma})$$

into (1) , we obtain an equation that can be evaluated analytically:

$$\overline{SER} = \left(\frac{1-b}{2}\right)^{N_r} \sum_{n=0}^{N_r-1} \binom{N_r - 1 + n}{n} \left(\frac{1+b}{2}\right)^n$$

where b is defined as

$$b = \sqrt{\frac{\overline{\gamma}}{1+\overline{\gamma}}}$$

For large values of γ, this can be approximated as

$$\overline{SER} = \left(\frac{1}{4\overline{\gamma}}\right)^{N_r} \binom{2N_r - 1}{N_r}$$

From this, we can see that (with Nr diversity antennas) the BER decreases with the N_r-th power of the SNR.

Computation via the Moment-Generating Function

- ✓ In the previous section, we averaged the BER over the distribution of SNRs, using the "classical" representation of the Q-function.
- ✓ As we have already seen (Chapter 12), there is an alternative definition of the Q-function, which can easily be combined with the moment-generating function $M\gamma(s)$ of the SNR.
- ✓ SER conditioned on a given SNR can be written in the form

$$SER(\gamma) = \int_{\theta_1}^{\theta_2} f_1(\theta) \exp(-\gamma_{\mathrm{MRC}} f_2(\theta))\, d\theta$$

Since

$$\gamma_{\mathrm{MRC}} = \sum_{n=1}^{N_r} \gamma_n$$

this can be rewritten as

$$SER(\gamma) = \int_{\theta_1}^{\theta_2} f_1(\theta) \prod_{n=1}^{N_r} \exp(-\gamma_n f_2(\theta))\, d\theta$$

Averaging over the SNRs in different branches then becomes

$$\begin{aligned}\overline{SER} &= \int d\gamma_1 pdf_{\gamma_1}(\gamma_1) \int d\gamma_2 pdf_{\gamma_2}(\gamma_2) \cdots \int d\gamma_{N_r} pdf_{\gamma_{N_r}}(\gamma_{N_r}) \int_{\theta_1}^{\theta_2} d\theta f_1(\theta) \prod_{n=1}^{N_r} \exp(-\gamma_n f_2(\theta)) \\ &= \int_{\theta_1}^{\theta_2} d\theta f_1(\theta) \prod_{n=1}^{N_r} \int d\gamma_n pdf_{\gamma_n}(\gamma_n) \exp(-\gamma_n f_2(\theta)) \\ &= \int_{\theta_1}^{\theta_2} d\theta f_1(\theta) \prod_{n=1}^{N_r} M_\gamma(-f_2(\theta)) \\ &= \int_{\theta_1}^{\theta_2} d\theta f_1(\theta) [M_\gamma(-f_2(\theta))]^{N_r}\end{aligned}$$

With that, we can write the error probability for BPSK in Rayleigh fading as

$$\overline{SER} = \frac{1}{\pi} \int_0^{\pi/2} \left[\frac{\sin^2(\theta)}{\sin^2(\theta) + \overline{\gamma}}\right]^{N_r} d\theta$$

Symbol Error Rate in Frequency-Selective Fading Channels

- ✓ We now determine the SER in channels that suffer from time dispersion and frequency dispersion.
- ✓ We assume here FSK with differential phase detection.
- ✓ The analysis uses the correlation coefficient ρ_{XY} between signals at two sampling times .
- ✓ For binary FSK with selection diversity:

$$\overline{SER} = \frac{1}{2} - \frac{1}{2} \sum_{n=1}^{N_r} \binom{N_r}{n} (-1)^{n+1} \frac{b_0 \operatorname{Im}\{\rho_{XY}\}}{\sqrt{(\operatorname{Im}\{\rho_{XY}\})^2 + n(1 - |\rho_{XY}|^2)}}$$

where b_0 is the transmitted bit.

This can be approximated as

$$\overline{SER} = \frac{(2N_r - 1)!!}{2}\left(\frac{1 - |\rho_{XY}|^2}{2(\mathrm{Im}\{\rho_{XY}\})^2}\right)^{N_r}$$

where $(2N_r - 1)!! = 1 \cdot 3 \cdot 5 \ldots (2N_r - 1)$.

For binary FSK with MRC:

$$\overline{SER} = \frac{1}{2} - \frac{1}{2}\frac{b_0 \mathrm{Im}\{\rho_{XY}\}}{\sqrt{1 - (\mathrm{Re}\{\rho_{XY}\})^2}} \sum_{n=0}^{N_r-1} \frac{(2n-1)!!}{(2n)!!}\left(1 - \frac{(\mathrm{Im}\{\rho_{XY}\})^2}{1 - (\mathrm{Re}\{\rho_{XY}\})^2}\right)^n$$

which can be approximated as

$$\overline{SER} = \frac{(2N_r - 1)!!}{2(N_r!)}\left(\frac{1 - |\rho_{XY}|^2}{2(\mathrm{Im}\{\rho_{XY}\})^2}\right)^{N_r}$$

- ✓ This formulation shows that MRC improves the SER by a factor $N_r!$ compared with selection diversity.
- ✓ A further important consequence is that the errors due to delay dispersion and random Frequency Modulation (FM) are decreased in the same way as errors due to noise.
- ✓ This is shown by the expressions in parentheses that are taken to the N_r-th power.
- ✓ These terms subsume the errors due to all different effects. The SER with diversity is approximately the N_r-th power of the SER without diversity (see following Figures 4.21 – 4.23).

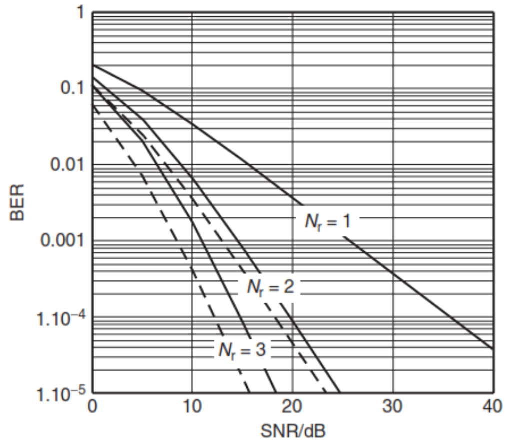


Figure 4.21 Bit error rate of minimum shift keying (MSK) with received-signal-strength-indication-driven selection diversity (solid) and maximum ratio combining (dashed) as a function of the signal-to-noise ratio with N_r diversity antennas.

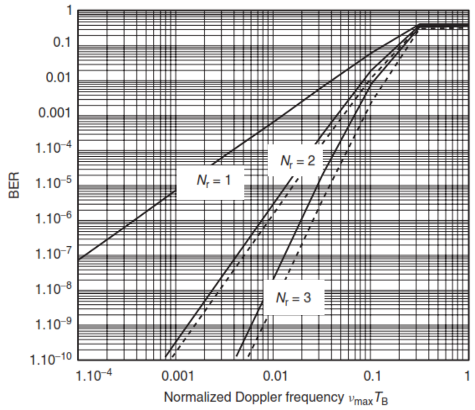


Figure 4.22 Bit error rate of MSK with received-signal-strength-indication-driven selection diversity (solid) and maximum ratio combining (dashed) as a function of the normalized Doppler frequency with N_r diversity antennas.

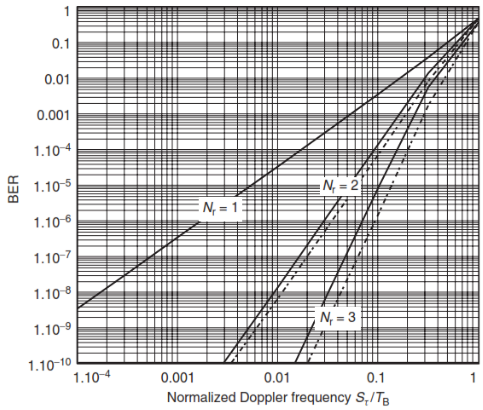


Figure 4.23 Bit error rate of MSK with received-signal-strength-indication-driven selection diversity (solid) and maximum ratio combining (dashed) as a function of the normalized rms delay spread with N_r diversity antennas.

✓For Differential Quadrature-Phase Shift Keying (DQPSK) with selection diversity, the average BER is

$$\overline{BER} = \frac{1}{2} - \frac{1}{4}\sum_{n=1}^{N_r}\binom{N_r}{n}(-1)^{n+1}\left[\frac{b_0\mathrm{Re}\{\rho_{XY}\}}{\sqrt{(\mathrm{Re}\{\rho_{XY}\})^2 + n(1-|\rho_{XY}|^2)}} + \frac{b_0'\mathrm{Im}\{\rho_{XY}\}}{\sqrt{(\mathrm{Im}\{\rho_{XY}\})^2 + n(1-|\rho_{XY}|^2)}}\right]$$

where $(b_0, b_0^{'})$ is the transmitted symbol.

✓For DQPSK with MRC:

$$\overline{BER} = \frac{1}{2} - \frac{1}{4}\sum_{n=0}^{N_r-1}\frac{(2n-1)!!}{(2n)!!}\left[\frac{b_0\mathrm{Re}\{\rho_{XY}\}}{\sqrt{1-(\mathrm{Im}\{\rho_{XY}\})^2}}\left(\frac{1-|\rho_{XY}|^2}{1-(\mathrm{Im}\{\rho_{XY}\})^2}\right)^n + \frac{b_0'\mathrm{Im}\{\rho_{XY}\}}{\sqrt{1-(\mathrm{Re}\{\rho_{XY}\})^2}}\left(\frac{1-|\rho_{XY}|^2}{1-(\mathrm{Re}\{\rho_{XY}\})^2}\right)^n\right]$$

✓More general cases can be treated with the Quadratic Form Gaussian Variable (QFGV) method, where the expression of error probability is given by

$$\begin{aligned} P(D<0) &= Q_M(p_1, p_2) - I_0(p_1 p_2)\exp\left[-\frac{1}{2}(p_1^2 + p_2^2)\right] \\ &+ \frac{I_0(p_1 p_2)\exp\left[-\frac{1}{2}(p_1^2+p_2^2)\right]}{(1+v_2/v_1)^{2N_r-1}}\sum_{n=0}^{N_r-1}\binom{2N_r-1}{n}\left(\frac{v_2}{v_1}\right)^n \\ &+ \frac{\exp\left[-\frac{1}{2}(p_1^2+p_2^2)\right]}{(1+v_2/v_1)^{2N_r-1}}\cdot\sum_{n=1}^{N_r-1} I_n(p_1 p_2)\sum_{k=0}^{N_r-1-n}\binom{2N_r-1}{k} \\ &\times\left[\left(\frac{p_2}{p_1}\right)^n\left(\frac{v_2}{v_1}\right)^k - \left(\frac{p_1}{p_2}\right)^n\left(\frac{v_2}{v_1}\right)^{2N_r-1-k}\right] \end{aligned}$$

✓ RSSI-driven diversity is not the best selection strategy when errors are mostly caused by frequency selectivity and time selectivity. It puts emphasis on signals that have a large amplitude, and not on those with the smallest distortion.

✓ In these cases, BER-driven selection diversity is preferable. For N_r = 2, the BER of Minimum Shift Keying (MSK) with differential detection becomes

$$\overline{BER} \approx \left(\frac{\pi}{4}\right)^4\left(\frac{S_\tau}{T_B}\right)^4$$

compared with the RSSI-driven result:

$$\overline{BER} \approx 3\left(\frac{\pi}{4}\right)^4 \left(\frac{S_\tau}{T_B}\right)^4$$

****

4.9. RAKE RECEIVER

1. **With block diagram, explain the operation of a RAKE receiver. (Apr/May 2017, Nov 2013, Nov/Dec 2012)**
2. **With neat diagram, explain how RAKE receiver provides diversity to improve the performance. (May/June 2014, May 2010)**
3. **Explain the principles of RAKE receiver in detail. [April/May 2018]**

- ✓ In CDMA spread spectrum systems, the chip rate is typically much greater than the flat fading bandwidth of the channel.
- ✓ Whereas conventional modulation techniques require an equalizer to undo the inter symbol interference between adjacent symbols.
- ✓ CDMA spreading codes are designed to provide very low correlation between successive chips. Thus, propagation delay spread in the radio channel merely provides multiple versions of the transmitted signal at the receiver.
- ✓ If these multipath components are delayed in time by more than chip duration, they appear like uncorrelated noise at a CDMA receiver, and equalization is not required.
- ✓ However, since there is useful information in the multipath components, CDMA receivers may combine the time delayed versions of the original signal transmission in order to improve the signal to noise ratio at the receiver.
- ✓ A RAKE receiver is to collect the time-shifted versions of the original signal by providing a separate correlation receiver for each of the multipath signals.

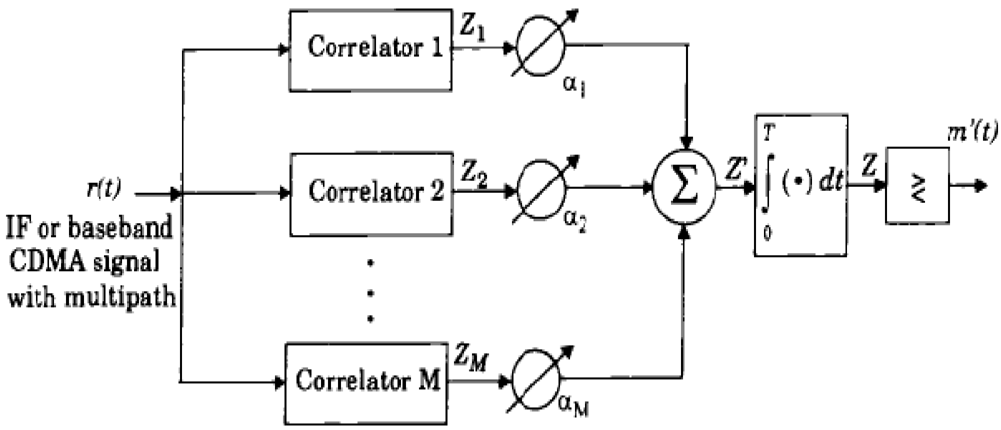


Fig: An M-branch (M-finger) RAKE receiver implementation.

- ✓ The outputs of the matched filters are passed through a filter matched to the channel response h and then sampled at time n+L is called **the Rake receiver.**
- ✓ It is taking inner products of the received signal with shifted versions at the candidate transmitted sequences. Each output is then weighted by the channel tap gains at the appropriate delays and summed.
- ✓ The signal path associated with a particular delay is sometimes called a finger of the Rake receiver

- ✓ Each correlator detects a time shifted version of the original CDMA transmission, and each finger of the RAKE correlates to a portion of the signal which is delayed by at least one chip in time from the other fingers.
- ✓ A RAKE receiver utilizes multiple correlators to separately detect the M strongest multipath components.
- ✓ The outputs of each correlator are weighted to provide a better estimate of the transmitted signal than is provided by a single component. Demodulation and bit decisions are then based on the weighted outputs of the M correlators.
- ✓ The basic idea of a RAKE receiver was first proposed by Price and Green.
- ✓ In outdoor environments, the delay between multipath components is usually large and, if the chip rate is properly selected, the low autocorrelation properties of a CDMA spreading sequence can assure that multipath components will appear nearly uncorrelated with each other.
- ✓ Assume M correlators are used in a CDMA receiver to capture the M strongest multipath components. A weighting network is used to provide a linear combination of the correlator output for bit detection.
- ✓ Correlator 1 is synchronized to the strongest multipath in1. Multipath component m2 arrives t1 later than component1.
- ✓ The second correlator is synchronized to m2. It correlates strongly with m2 but has low correlation with m1 . Note that if only a single correlator is used in the receiver, once the output of the single correlator is corrupted by fading, the receiver cannot correct the value.
- ✓ Bit decisions based on only a single correlation may produce a large bit error rate.
- ✓ In a RAKE receiver, if the output from one correlator is corrupted by fading, the others may not be, and the corrupted signal may be discounted through the weighting process.
- ✓ Decisions based on the combination of the M separate decision statistics offered by the RAKE provide a form of diversity which can overcome fading and thereby improve CDMA reception.
- ✓ The M decision statistics are weighted to form an overall decision statistic as shown in Figure The outputs of the 1W correlators are denoted as Z1, Z2 ,... and Z_M.
- ✓ They are weighted by α_1, α_2,, ... α_m respectively.
- ✓ The weighting coefficients are based on the power or the SNR from each correlator output. If the power or SNR is small out of a particular correlator, it will be assigned a small weighting factor.
- ✓ Just as in the case of a maximal ratio combining diversity scheme, the overall signal Z' is given by

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

- ✓ The weighting coefficients, α_m, are normalized to the output signal power of the correlator in such a way that the coefficients sum to unity, as shown in equation

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

- ✓ As in the case of adaptive equalizers and diversity combining, there are many ways to generate the weighting coefficients.
- ✓ However, due to multiple access interference, RAKE fingers with strong multipath amplitudes will not necessarily provide strong output after correlation.

Example 4.1

Assume four branch diversity is used, where each branch receives an independent Rayleigh fading signal. If the average SNR is 20 dB, determine the probability that the SNR will drop below 10 dB. Compare this with the case of a single receiver without diversity. (Apr/May 2017)

For this example the specified threshold γ = 10 dB, Γ = 20 dB, and there are four branches. Thus $\gamma/\Gamma = 0.1$ and using equation (6.58),

$$P_4(10\text{ dB}) = (1 - e^{-0.1})^4 = 0.000082$$

When diversity is not used, equation (6.58) may be evaluated using $M = 1$.

$$P_1(10\text{ dB}) = (1 - e^{-0.1})^1 = 0.095$$

Notice that without diversity the *SNR* drops below the specified threshold with a probability that is three orders of magnitude greater than if four branch diversity is used!

Example 4.2

Consider the design of the U.S. digital cellular equalizer where f = 900MHz and the mobile velocity u = 80 kmlhr, determine the following: (a) the maximum Doppler shift (b) the coherence time of the channel (c) the maximum number of symbols that could be transmitted without updating the equalizer, assuming that the symbol rate is 24.3 k symbols/sec.(Apr/May 2017)

Doppler shift

$$f_d = \frac{1}{2\pi} \cdot \frac{\Delta\phi}{\Delta t} = \frac{v}{\lambda} \cdot \cos\theta$$

Coherence time

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

$$T_c = \sqrt{\frac{9}{16\pi f_d^2}} = \frac{0.423}{66.67} = 6.34\text{ msec}$$

fm is the maximum Doppler shift given by fm = v/λ.

To ensure coherence over a TDMA time slot, data must be sent during a 6.34 ms interval. For R_s=24.3 ksymbols/sec, the number of bits that can be sent is

$$N_b = R_s T_C = 24,300 \times 0.00634 = 154\text{ symbols}$$

As shown in Chapter 10, each time slot in the U.S. digital cellular standard has a 6.67 ms duration and 162 symbols per time slot, which are very close to values in this example.

Example 4.3 **Consider the design of US DC digital cellular equalizer, where f=900 MHz and the velocity of mobile = 80k m/hour. Determine the maximum Doppler shift, the coherence time of the channel and maximum number of symbols that could be transmitted without updating the equalizer assuming that symbol rate is 24.3 k symbols/sec.**

Coherence time = $(9C/\ 16\Pi v f_c)$

TWO MARKS

1. How can we improve link performance?

1. Diversity
2. Equalization
3. Channel coding

2. What is diversity, equalization technique?

- ✓ To reduce ISI, equalization technique is used.
- ✓ To reduce fading effects, diversity technique is used.

3. What is equalization, an equalizer?(Nov 2013)

- ✓ The process of extracting the symbols from the received signal is called equalization.
- ✓ The goal of equalization is the combination of the transmitter; channel and receiver appear to be an all-pass channel. In the frequency domain equation

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

Where $H_{eq}(f) \leftarrow F(f)$ are Fourier transforms of $h_{eq}(t) \leftarrow f(t)$ respectively.

$$\text{Equalizer transfer function} \propto \frac{1}{\text{Channel transfer function}}$$

Equalizer is a linear pulse shaping filter, used to reduce the dispersive effects of a channel like ISI- inter symbol interference is referred to as an ***equalizer***.

4. Write the major classifications of equalizers. State the significance of each. (May 2012, May 2013)

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

Linear equalizer:

1. In linear equalizer, the current and past values of the received signal are linearly weighted by the filter coefficients and summed to produce the output. No feedback path is used.
2. Simple, easy to implement.
3. Not suitable for severely distorted channel, noise power signal is enhanced.

Non-linear equalizer:

1. If the past decisions are correct, then the ISI contributed by present symbol can be cancelled exactly, feedback path is used.
2. Suitable for severely distorted channel, also noise power is not enhanced.
3. Complex in structure, channels with low SNR, the DFE suffers from error propagation.

5. What are the types of non-linear equalizer?

It has three types

1. Decision feed back
2. Maximum likelihood symbol detector
3. Maximum likelihood sequence estimator

6. Write the advantages of lattice equalizer.

(i) It is simplest and easily available.

(ii) Numerical stability.

(iii) Faster convergence.

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

(v) Unique structure of the lattice filter allows the dynamic assignment.

7. Define adaptive equalization. Write the significance of it. (May 2016)

Adaptive equalizers assume channel is time varying channel and try to design equalizer filter whose filter coefficients are varying in time according to the change of channel, and try to eliminate ISI and additive noise at each time.

8. What are the applications of non linear equalizer? (May/June 2014)

Non linear equalizer is used for

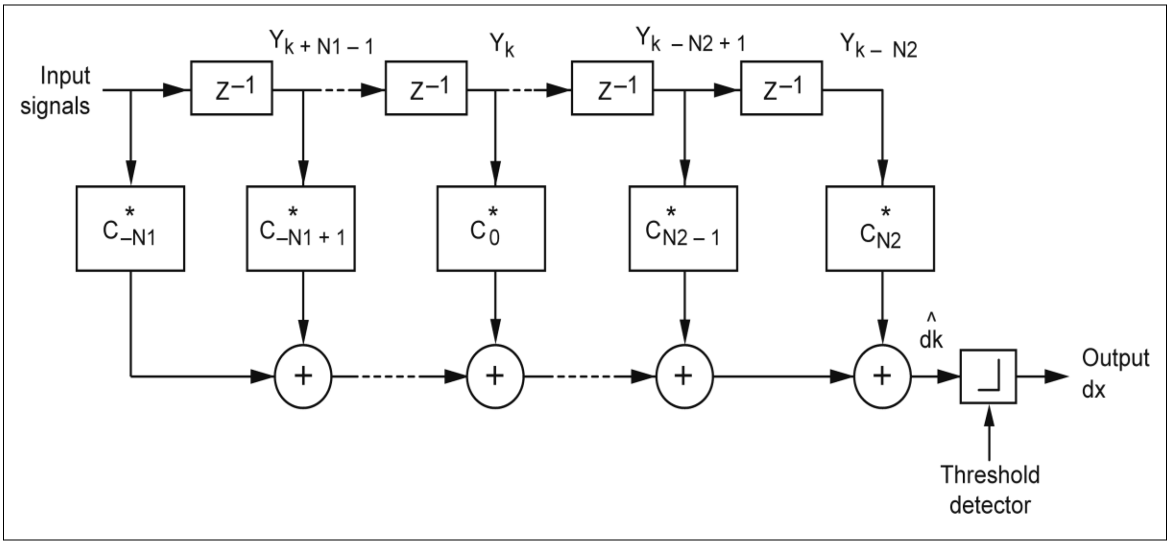
- Microwave communications
- Satellite communications
- Mobile communications

9. Why is an adaptive equalizer required? (APRIL/MAY 2017)

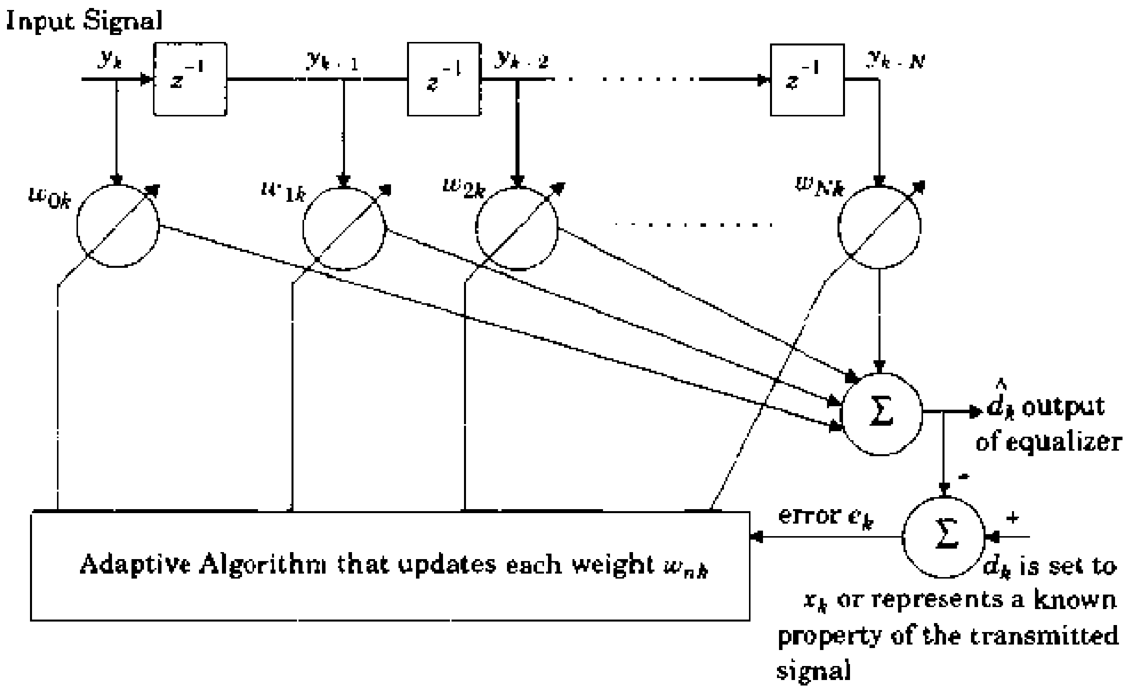
In practice, the channel response is unknown.

Hence the optimum matched filter must be adaptively estimated to reduce error.

10. Draw the structure of a linear transversal equalizer. (Nov/Dec 2015)



8. Draw the structure of generic optimum receiver. (May/June 2013)



9. What are the types of non-linear equalizer?

It has three types

1. Decision feed back
2. Maximum likelihood symbol detector
3. Maximum likelihood sequence estimator

10. What are the applications of non linear equalizer? (May/June 2014)

Non linear equalizer is used for i) Microwave communications ii) Satellite iii) Mobile communications

11. Write about MLSE decision feedback equalizer.(May 2015)

✓The MLSE is optimal in the sense that it minimizes the probability of a sequence error.

✓The MLSE requires knowledge of the channel characteristics, noise in order to compute the metrics for making decisions.

✓An MLSE usually has a large computational requirement, especially when the delay spread of the channel is large.

12. Write the advantages of MMSE equalizer.

Advantages of MMSE are

✓ The noise power of an MMSE equalizer is smaller than that of a ZF equalizer.

✓ Suitable for wireless link.

13. List the advantages and disadvantages of DFE equalizer.

Advantages of DFE are

✓ FBF can be realized as a lattice structure.

✓ RLS lattice algorithm can be used to yield fast convergence.

✓ DFE has a smaller error probability than a linear equalizer.

14. What are the factors used in adaptive algorithms?

(i) Rate of convergence

(ii) Misadjustment

(iii) Computational complexity

(iv) Numerical properties.

15. Write the basic algorithms used for adaptive equalizations.

(i) Zero forcing (ZF) algorithm.

(ii) Least mean squares (LMS) algorithm.

(iii). Recursive least square (RLS) algorithm.

16. Write the advantages and drawbacks of zero forcing algorithms.

Advantage: For low distortion channel, it is used to reduce mean square error.

Drawbacks: Noise is enhanced.

17. Write the advantages and drawbacks of LMS algorithm.

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

(ii) Low computational complexity and

(iii) Simple program.

18. Write the advantages drawbacks of RLS algorithm.

(i) Fast convergence

(ii) Good tracking ability. If smaller value of weighting coefficient L, the

Equalizer has better tracking ability.

19. What are the differences between zero-forcing and mean squared error equalizer?

Zero Forcing Equalizer (ZF)	Mean Squared Error Equalizer (MSE)
1. Simple filters easy to implement as a transversal structure.	1. Optimum filter, implemented as a lattice structure.
2. It performs well for static channels with low SNR.	2. Suitable for static channels with high SNR.
3. Noise enhancement makes ZF equalizer not suitable for wireless link.	3. suitable for wireless link.

20. What is the principle of diversity technique? (May/June 2013, Nov/Dec 2013, Apr/May 2017)

- ✓ Signal is transmitted by more than one antenna via channel.
- ✓ It ensures that the same information reaches the receiver on statistically independent channels.

21. Why diversity technique is needed? Why it is employed? (Apr/May2017, Nov/Dec 2010)

In AWGN – channel, BER decreases exponentially as the SNR increases.

In Rayleigh fading channel – BER decreases linearly with SNR.

So, to achieve 10^{-4} BER, diversity is used.

22. Define SNR.

SNR = average received signal energy per (complex) symbol time/ noise energy per (complex) symbol time

23. What is small scale fading, large scale fading?

Small scale fading	Large scale fading
• Due to multiple reflections from the surroundings, small scale fading occur over a short period of time or travel distance. • It causes deep and rapid amplitude fluctuations in the signal.	• Due to shadowing of large terrain profile large scale fading occurs. • It causes variations in the signal strength.

24. Define 5 common methods of micro diversity.(Nov 2011)[Nov/Dec 2017]

The five most common methods are

1. Spatial diversity → several antenna elements separated in space.
2. Temporal diversity → Repetition of the transmit signal at different times.
3. Frequency diversity → Transmission of the signal on different frequencies.
4. Angular diversity → Multiple antennas with different antenna patterns.
5. Polarization → Multiple antennas receiving different polarizations

25. Define space diversity/ antenna diversity and its types. (May 2010, Nov/Dec 2015)

Space diversity is also known as antenna diversity. The concept is at each cell site, multiple base station receiving antennas are used to provide diversity reception.

Space diversity reception methods can be classified into two categories

1. Selection diversity---- a) switched diversity b) Feedback diversity
2. Combining diversity ----a) Maximal ratio combining b) Equal gain diversity

26. Differentiate selection and combining diversity.

S.No.	Selection diversity	Combining diversity
1.	The "best signal" copy is selected and processed while all other copies are discarded. *e.g.* large RSSI – received signal strength indication is selected.	Combining diversity:- All copies of the signal are combined before (demodulation) processing and the combined signals are decoded.
2.	Simple circuits are needed.	Individual receiver phasing circuits are needed.
3.	None of the signal is not in acceptable SNR means, it is degraded.	It works better than selection diversity.

27. Write the classification in signal combining techniques.

1. Selection diversity---- Fading path with highest gain used to select best signal
   a) switched diversity----active branch is monitored continuously
   b) Feedback diversity ---- scanned in a fixed sequence
2. Combining diversity ---- All copies of the signal are combined before processing
   a) Maximal ratio combining --- The signals from all of the M branches are weighted according to their SNR and then summed
   b) Equal gain combining ------ The branch weights are all set to unity but the signals from each are co-phased to provide equal gain combining diversity

28. Compare MRC and EGC techniques.

MRC- Maximal ratio combining	EGC- Equal gain combining
The signals from all of the M branches are weighted according to their SNR and then summed	The branch weights are all set to unity but the signals from each are co-phased to provide equal gain combining diversity.

29. Define switched diversity.

- ✓ Active branch is monitored by the receiver.
- ✓ If the signal level falls below the threshold, then the receiver switches to a new branch (antenna).

30. Define feedback (or) scanning diversity.

- ✓ All the signals are scanned in a fixed sequence, until one is found to be above a predetermined threshold.
- ✓ Then, that signal is monitored until it falls below the threshold.

31. Define temporal diversity.

Wireless propagation channel is time variant, so for sufficient decorrelation, the temporal distance between antennas must be at least $\frac{1}{2}$ V_{max}.

$V_{max} \rightarrow$ maximum Doppler frequency.

32. Define frequency diversity.

Correlation is increased by

1. Transmitting information on more than one carrier frequency.
2. Frequencies are separated by more than one coherence bandwidth of the channel.

So, the signals will not experience the same fades.

33. What is angular diversity?

- ✓ Decorrelation of signals is enhanced for closely spaced antennas.
- ✓ For that, different antenna patterns are used.

34. Define Polarization diversity.

In TX side, two diversity branches are used. Signals are transmitted through two orthogonally polarized propagation path.

In RX side, antennas with two elements receive the vertical or horizontally polarized signal.

In the channel, if the signal is obstructed, polarization diversity will reduce the multipath spread without decreasing the receiver power.

35. Differentiate between Micro, Macro diversity. (May 2014, Nov 2014)

S.No.	Micro diversity	Macro diversity
1.	Used to reduce small scale fading effects.	Used to reduce large scale fading effects.
2.	Multiple reflections cause deep fading. This effect is reduced.	Deep shadow causes fading. This effect is reduced.
3.	BS – MS are separated by a small distance.	BS – MS are separated by a large distance.

36. What is transmit diversity? (Nov/Dec 2017, Apr/May 2015, Nov/Dec 2015)

Diversity effect is achieved by transmitting signals from several transmit antenna is known as transmit diversity.

37. What is receiving diversity (or) diversity reception technique? (Nov/Dec 2017, Apr/May 2015, Nov/Dec 2015, Nov/Dec 2012)

Diversity effect is achieved by receiving signals from several receive antenna is known as receive diversity.

38. What is the basic idea of Rake receiver.(Nov 2012)

- ✓It consists of a bank of correlators; each sampled at a different time with delay τ and thus collects energy from the MPC.
- ✓The sample values from the correlators are then weighted and combined to achieve improved communications reliability and performance.
- ✓The tap delays as well as the tap weights are adjustable and matched to the channel.

39. What are the benefits of RAKE receiver? (May 2016)

The main advantage of Rake Receiver is that it improves the SNR (or) E_b / N_o. Naturally, this improvement is observed in larger environments with many multipaths than in environments without obstruction.

40. What do you mean by coding gain? (Nov/Dec 16)

Coding gain is what allows a channel error rate of 10^{-2} to support decoded data rates which are 10^{-5} or better.

41. Distinguish between Diversity gain and array/ Beam forming gain.[April/May 2018]

Diversity gain reflects the fact that it is improbable that several antenna elements are in a fading dip simultaneously. Thus probability of error is very low.

Beam forming gain reflects the fact that the combiner performs an averaging over the noise at different antennas.

Thus, even if the signal levels at all antenna elements are identical, the combined output SNR is larger than the SNR at a single antenna element.

UNIT – V

MULTIPLE ANTENNA TECHNIQUES

Syllabus:

MIMO systems – spatial multiplexing -System model -Pre-coding - Beam forming - transmitter diversity, receiver diversity- Channel state information-capacity in fading and non-fading channels.

5.1 MIMO-MULTIPLE INPUT MULTIPLE OUTPUT SYSTEMS

1. **What is meant by MIMO systems? Explain the system model with necessary diagrams. [Nov/Dec 2017]**
2. **Describe MIMO systems with emphasis on their requirement in a wireless communication environment. [May 2017] [April/May 2018]**

- ✓ MIMO systems are modern wireless systems with ***Multiple Element Antennas (MEAs)*** at ***both*** ends of the link.

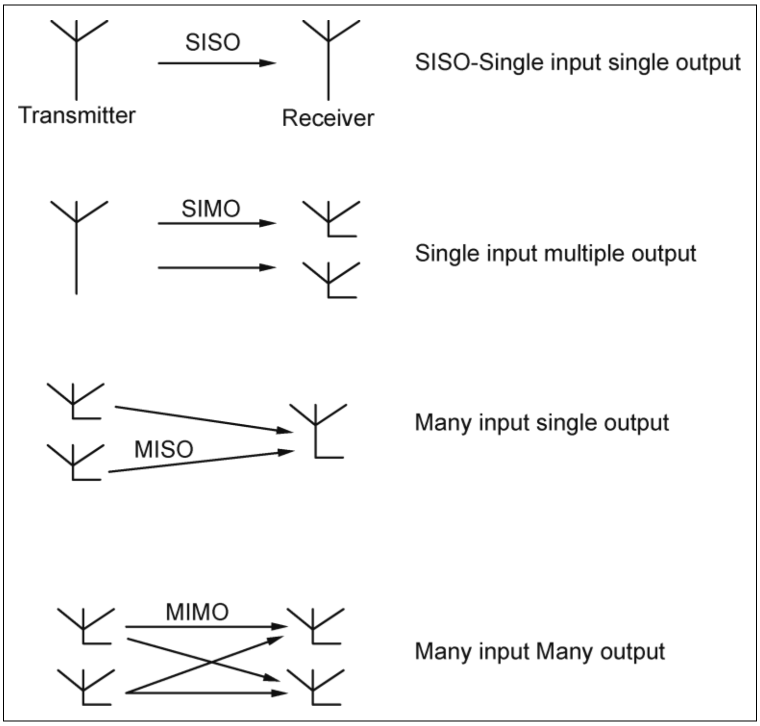


Figure 5.1. Principle of MIMO Techniques

- ✓ The MEAs of a MIMO system can be used for four different purposes:
  - (i) Beamforming
  - (ii) Diversity
  - (iii) Interference suppression, and
  - (iv) Spatial multiplexing (transmission of several data streams in parallel).
- ✓ **Beam Forming** (or) directional antennas are used for interference reduction.
- ✓ **Spatial multiplexing** allows direct improvement of capacity by simultaneous transmission of multiple data streams on multiple antennas in parallel.
  - ✓ It increases the capacity of the system.

- ✓ It effectively exploits multipath.
- ✓ Spectral efficiency is as high as 20-40 bps/Hz.

Examples:

- ➢ MIMO – OFDM are the key techniques for next generation wireless LAN and 4 G mobile communications.
- ➢ MIMO – OFDM is used in IEEE 802.11 n, IEEE 802.16 m and LTE.
- ➢ OFDM – Orthogonal frequency division multiplexing.
- ➢ LTE – Long term evolution

Features of MIMO System

- ✓ The transmit power and bandwidth is not increased
- ✓ Spatial resource is utilized properly.
- ✓ Wireless transmission capacity is increased.
- ✓ Spectral efficiency is increased.
- ✓ Good quality is achieved because of minimum probability of error.

5.2 SPATIAL MULTIPLEXING

Explain in detail spatial multiplexing of a MIMO system. [May 2016]

- ✓ Spatial multiplexing uses MEAs at the TX for transmission of parallel data streams.
- ✓ 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 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.
- ✓ So, data rate drastically increased by a factor of ***min*** (N_t, N_r).

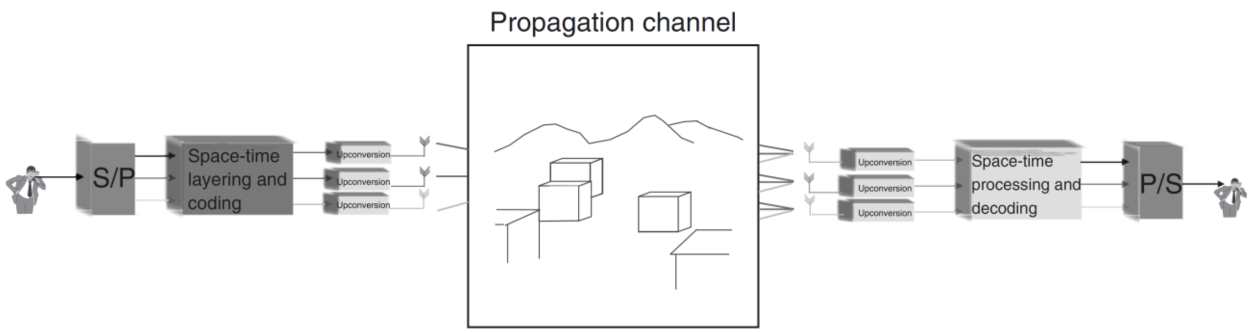


Figure 5.2 (a). Principle behind Spatial Multiplexing

- ✓ For the case when the TX knows the channel, can also develop another intuition (see Figure 5.3 (a)).
- ✓ With N_t transmit antennas, N_t different beams are formed.
- ✓ All these beams are pointed at different ***Interacting Objects*** (IOs), and transmit different data streams over them.
- ✓ At the RX, N_r antenna elements are used to form N_r beams, and also point them at different IOs.
- ✓ If all the beams can be kept orthogonal to each other, there is no interference between the data streams (it means *established parallel channels*).
- ✓ The IOs (along with the beams pointing in their direction) play the same role as wires in the transmission of multiple data streams on multiple wires.

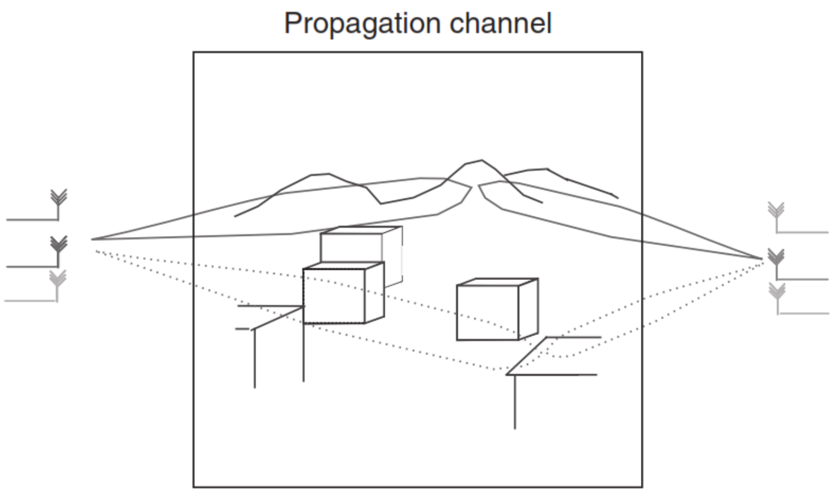


Figure 5.4 (b) Transmission of different data streams via different interacting objects.

- ✓ The number of possible data streams is limited by $min(N_t, N_r, N_s)$, where N_s is the number of (significant) IOs.
- ✓ The number of data streams cannot be larger than the number of transmit antenna elements.
- ✓ A sufficient number of receive antenna elements (at least as many as data streams) are needed to form the receive beams.
- ✓ Thus, it will be able to separate the data streams.
- ✓ But it is also very important to notice that the number of IOs poses an upper limit: if two data streams are transmitted to the same IO, then the RX has no possibility of sorting them out by forming different beams.
- ✓ The above intuitive pictures are somewhat simplified.

The Narrowband Multiple Antenna System Model

- ✓ A narrow band point-to-point communication system has M_t transmit and M_r receive antennas. This system can be represented by the discrete time model

$$\begin{bmatrix} y_1 \\ \vdots \\ y\mathrm{M_r} \end{bmatrix} = \begin{bmatrix} h_{11} & \cdots & h_1\mathrm{M}_t \\ \vdots & \ddots & \vdots \\ h\mathrm{M_r}1 & \cdots & h\mathrm{M_r}\mathrm{M_t} \end{bmatrix} \begin{bmatrix} x_1 \\ \vdots \\ x\,\mathrm{M}_r \end{bmatrix} + \begin{bmatrix} n_1 \\ \vdots \\ n\,\mathrm{M}_r \end{bmatrix}$$

y - Hx + n

x - M_t dimensional transmitted symbol

n - M_r dimensional noise vector

H - $\mathrm{M}_r \times \mathrm{M}_t$ matrix of channel

h_{ij} - the gain from transmit antenna j to receive antenna i,

h_{ij} represents the channel transfer function.

✓H is modeled as a random matrix characterized by an uncorrected (or) correlated Rayleigh fading channel. (or) uncorrelated or correlated Ricean fading channel.

B - Bandwidth of the channel

✓Noise presents in the channel has zero mean and covariance matrix = σ2 I Mr

whereσ2 = E [n_i^2]

σ2 = $\frac{\text{No}}{2}$ = power spectral density of the channel noise

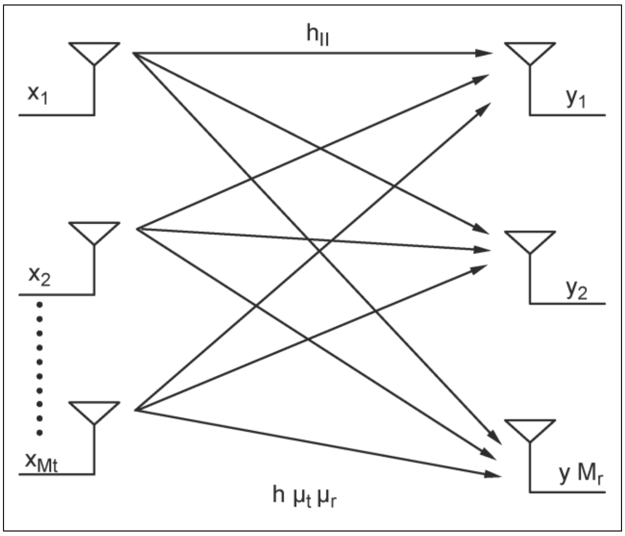


Figure 5.5. MIMO systems

For simplicity,

σ^2 = unity

transmit power = $\frac{P}{\sigma^2}$ = ρ

ρ = average SNR per receive antenna under unity channel gain

Input power constraint $T_r (R_x) = \rho$

Input covariance matrix $R_x = E [x\, x^H]$

CSIT - channel side information at the transmitter

CSIR - channel side information at the receiver

For a Static Channel,

CSIR is typically assumed. If the feedback path is available then CSIR from the receiver can be sent back to the transmitter to provide CSIT.

ZMSW – Zero Mean Spatially White Model

When the channel is not known to either the transmitter or receiver then some distribution on the channel gain matrix must be assumed.

For the channel matrix *H* the entries are assumed as ***independent and identically distributed (i. i. d)*** zero mean, unit variance, complex circularly symmetric Gaussian random variables.

In general,

- ✓ Different assumptions about CSI and distribution of the *H* entries lead to different channel capacities.
- ✓ Optimal decoding of received signal requires maximum likelihood demodulation.
- ✓ Decoding complexity is reduced if the channel is known to the transmitter.

5.3 SYSTEM MODEL

Explain the system model of a multiple-input multiple-output system.

Concept of system model:

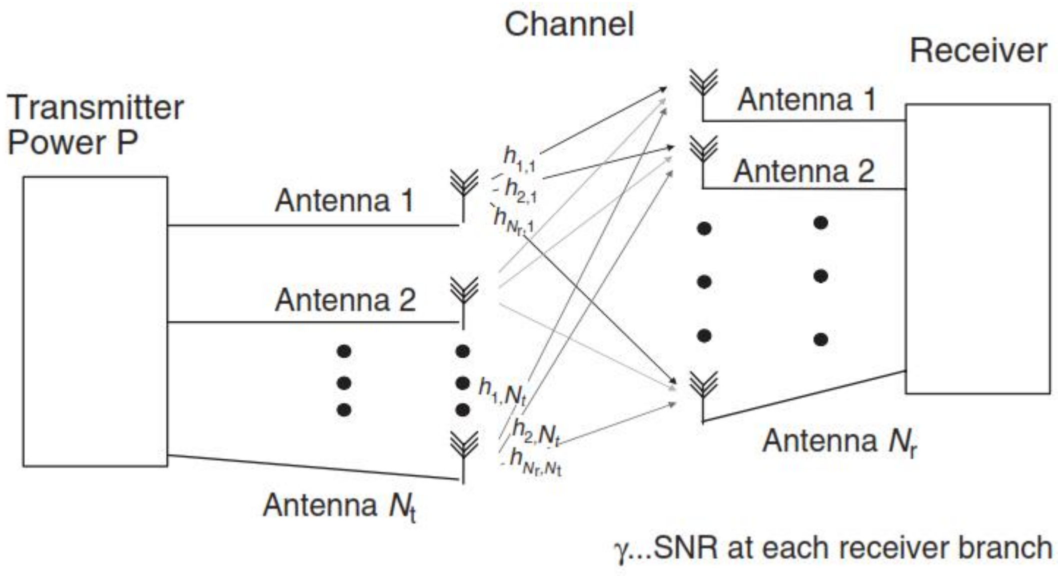


Figure 5.4 Block diagram of a multiple-input multiple-output system.

- ✓ The figure 5.4 shows the generic system that will be considered for capacity computations.
- ✓ At the TX, the data stream enters an encoder.
- ✓ The encoders outputs are forwarded to N transmit antennas.

- ✓ From the antennas, the signal is sent through the wireless propagation channel.
- ✓ The wireless channel is assumed to be quasi-static and frequency-flat.
- ✓ Quasi-static means that the coherence time of the channel is so long t that "a large number" of bits can be transmitted within this time.
- ✓ We denote the Nr × Nt matrix of the channel as

$$\mathbf{H} = \begin{pmatrix} h_{11} & h_{12} & \cdots & h_{1N_t} \\ h_{21} & h_{22} & \cdots & h_{2N_t} \\ \vdots & \vdots & \cdots & \vdots \\ h_{N_r1} & h_{N_r2} & \cdots & h_{N_rN_t} \end{pmatrix}$$

where,

h_{ij} - Complex channel gains (transfer functions) from the j^{th} transmit to the i^{th} receive antenna.

- ✓ The received signal vector

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

where,

s - the transmit signal vector

n - the noise vector.

- ✓ The received signal vector **r** contains the signals received by N_r antenna elements

Operation

- ✓ Spatial multiplexing techniques simultaneously transmit independent information sequences, often called layers, over multiple antennas.
- ✓ High-rate signal is split into multiple lower-rate streams and each stream is transmitted from a different transmit antenna in the same frequency channel.
- ✓ Received signal vector r = HS + n = x + n is received by N_r antenna elements , where S is the transmit signal vector and n is the noise vector.
- ✓ Spatial multiplexing is implemented with spatial rate $r_S = N_t$ and diversity order N_r .

In general,

$$N_r \geq N_t$$

- ✓ Spatial multiplexing provides high bandwidth efficiency but at low SNR, error rate increases.

Layered Space–Time Structure

Explain with relevant diagrams the layered space time structure with respect to MIMO systems. [May 2016]

- ✓ This is to realize these capacities of MIMO systems in practice (earlier, discussed with information-theoretic limits).
- ✓ One 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 (almost) 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 QPSK), such a scheme can actually be gainfully employed.
- ✓ However, for most practical cases, the complexity of a joint MLSE (Maximum Likelihood Sequence Estimators (or Estimation)) is high.
- ✓ For this reason, so-called layered space–time structures have been proposed.
- ✓ This allow to break up the demodulation process into several separate pieces, each of which has lower complexity.
- ✓ These structures are also widely known under the name of ***BLAST architectures***

  *(**B**ell Labs **LA**yered **S**pace **T**ime).*

1.Horizontal BLAST

- ✓ H-BLAST is the simplest possible layered space–time structure.
- ✓ The TX first demultiplexes the data stream into N_t parallel streams.
- ✓ Each of the streams is encoded *separately*.
- ✓ Each encoded data stream is then transmitted from a different transmit antenna.
- ✓ The channel mixes up the different data streams.
- ✓ The RX separates them out by successive nulling and interference subtraction.
- ✓ In other words, the RX proceeds in the following steps (Figure 5.5):

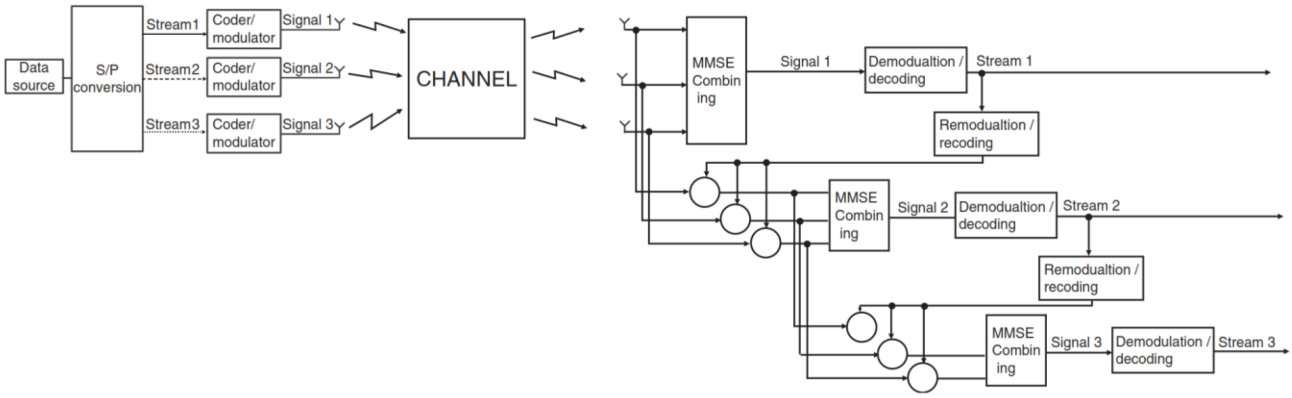


Figure 5.5 Block diagram of a horizontal BLAST transceiver.

- ✓ It considers the first data stream as the useful one, and the other data streams as interference.
- ✓ It can then use *optimum combining* for suppression of interfering streams.
- ✓ The RX has $N_r \geq N_t$ antenna elements available.
- ✓ If $N_r = N_t$, it can suppress all N_t - *1* interfering data streams and receive the desired data stream with diversity order 1.

- ✓ If the RX has more antennas, it can receive the first data stream with better quality.
- ✓ But in any case, interference from the other streams can be eliminated.
- ✓ The desired stream can now be demodulated and decoded.
- ✓ Outputs from that process are firm decisions on the bits of stream 1.
- ✓ Note that since separate encoding of different data streams are there, the knowledge of the first stream only needed to complete the decoding process.
- ✓ The bits that have thus been decoded are now re-encoded, and remodulated.
- ✓ Multiplying this symbol stream by the transfer function of the channel.
- ✓ The contribution of *stream 1* to the total received signal at the different antenna elements is obtained.
- ✓ These contributions are subtracted from the signals at the different antenna elements.
- ✓ Now the "cleaned-up" signal is considered and tried to detect the second data stream.
- ✓ There are N_r received signals, but only N_t - *2* interferers.
- ✓ Using optimum combining again, the desired data stream with diversity order 2 can be received.
- ✓ The next step is again decoding, recoding, and remodulating the considered data stream (stream 2 now).
- ✓ Then subtraction of the associated signal from the total signal at the receive antenna elements obtained in the previous step.
- ✓ This cleans up the received signal even more.
- ✓ The process is repeated until the last data stream is decoded.
- ✓ This scheme is actually very similar to multiuser detection: if different transmit streams were to come from different users, then H-BLAST would be normal serial interference cancellation.
- ✓ Note also that the encoding scheme does not require "cooperation" between different antenna elements (or users).
- ✓ Similar to serial interference cancellation, H-BLAST also faces the problem of error propagation, especially since the first decoded data stream has the worst quality.
- ✓ In other words: if data stream 1 is decoded incorrectly, then "wrong" signal is subtracted from the remaining signals at the antenna elements.
- ✓ Thus, instead of "cleaning up" the receive signal, even more interference is introduced.
- ✓ This in turn increases the likelihood that the second data stream is decoded incorrectly, and so on.
- ✓ In order to mitigate this problem, stream ordering should be used: the RX should first decode the stream that has the best SINR, then the one with the next best, and so on.

2 .Diagonal BLAST

- ✓ The main problem with 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 approach, streams are cycled through the different transmit antennas, such that each stream sees all possible antenna elements.
- ✓ In other words, 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 (compare Figure 5.6).
- ✓ Decoding can be done stream by stream; again, each decoded block can be subtracted from signals at the other antenna elements and, thus, enhances the quality of the residual signal.

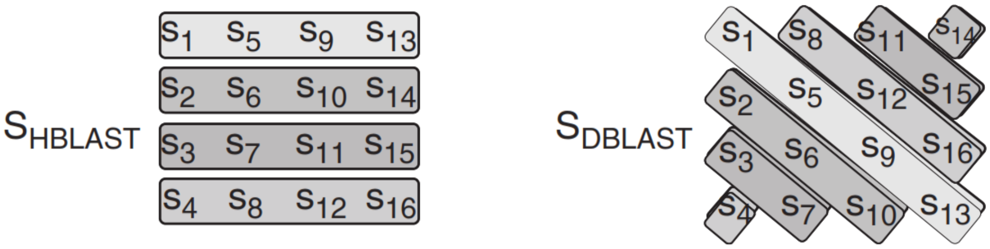


Figure 5.6 Assignment of bit streams to different antennas for horizontal BLAST and diversity BLAST.

- ✓ 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, while sometimes it is in a bad position, in the sense that it suffers full interference.
- ✓ Thus, each stream experiences full diversity.
- ✓ More precisely, each data stream alternatingly sees a channel with diversity order 1 (whose SNR has a pdf that is a chi-square distribution with 2 degrees of freedom χ_2^2), diversity order 2 (chi-square with 4 degrees of freedom χ_4^2), and so on.
- ✓ Therefore, total capacity in the case when $N_t = N_r$:

$$C_{\text{D-BLAST}} = \sum_{k=1}^{N_t} \log_2 \left[1 + \frac{\bar{\gamma}}{N_t} \chi_{2k}^2 \right] < C$$

3 .Structures for Channel State Information at the Transmitter

- ✓ When full CSI is available at the TX, transceiver schemes become much simpler, at least conceptually.
- ✓ By multiplying transmitted and received signal vectors by the right and left singular vectors of the channel matrix, respectively, diagonalization of the channel is achieved.
- ✓ Therefore, the different data streams do not interfere with each other; rather, there are a number of parallel channels, each of which can be separately encoded and decoded.
- ✓ The difficulties lie instead in the practical aspects of obtaining and using CSIT.

Advantages

1. Detection of V-BLAST is simpler than diagonal BLAST
2. An array gain of N_r can be achieved.

Spatial multiplexing with BLAST structure is used as a capacity-achieving technique

Drawbacks

1. Complex decoding scheme
2. Not practical in cellular environments

****

Comparison of schemes

Scheme	Spectral efficiency	Probability of error	Implementation complexity
V-Blast	High	High	Low
D-Blast	Moderate	Moderate	High
ALAMOUTI	Low	Low	Low

5.4 PRECODING

Describe the concepts of precoding. [Apr 2017]

Concept of precoding

- ✓ Precoding uses the same idea as frequency equalization.
- ✓ But here the fading is inverted at the transmitter instead of the receiver.
- ✓ This technique requires that the transmitter must have knowledge of the subchannel fading α_i .
- ✓ In this case, if the ***desired received signal power in the ith subchannel is P$_i$***, and the channel introduces ***fading of*** α_i in that subchannel, then ***the transmitter sends the signal in the ith subchannel with power*** P_i / α_i^2 .
- ✓ This signal is multiplied by the channel gain α_i , so the received signal power is $P_i\alpha_i^2 / \alpha_i^2 = P_i$, as desired.
- ✓ Note that the channel inversion takes place at the transmitter instead of the receiver, so the noise power remains as N_0B.
- ✓ Precoding is quite common on wireline multicarrier systems like HDSL (**H**igh-speed **D**igital **S**ubscriber **L**ines).
- ✓ There are two main problems with precoding in a wireless setting.
  - ***Precoding is basically channel inversion:*** The inversion is not power-efficient in fading channels. An infinite amount of power is needed to do channel inversion on a Rayleigh channel.
  - ***Precoding is the need for accurate channel estimates at the transmitter***, which are difficult to obtain in a rapidly fading channel.

Transmit Coding

- ✓ The parallel decomposition of the channel is obtained by defining a transformation on the channel input and output x and y through **transmit precoding** and **receiver shaping**.
- ✓ In ***transmit precoding*** the input to the antennas **x** is generated through a linear transformation on input vector $\tilde{\mathbf{x}}$ as $\mathbf{x} = \mathbf{V}^{\mathbf{H}}\tilde{\mathbf{x}}$.
- ✓ ***Receiver shaping*** performs a similar operation at the receiver by multiplying the channel output **y** with U^H , as shown in Figure 5.7.

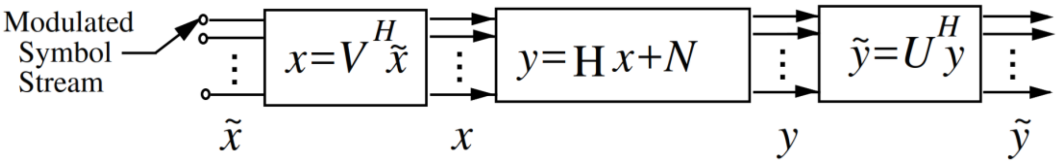


Figure 5.7: Transmit Precoding and Receiver Shaping.

- ✓ The transmit precoding and receiver shaping transform the MIMO channel into R_H parallel single-input single-output (SISO) channels with input $\tilde{\mathbf{x}}$ and output $\tilde{\mathbf{y}}$ since from the SVD, we have that

$$\begin{aligned}
\tilde{\mathbf{y}} &= \mathbf{U}^H(\mathbf{H}\mathbf{x}+\mathbf{n}) \\
&= \mathbf{U}^H(\mathbf{U}\mathbf{\Sigma}\mathbf{V}\mathbf{x}+\mathbf{n}) \\
&= \mathbf{U}^H(\mathbf{U}\mathbf{\Sigma}\mathbf{V}\mathbf{V}^H\tilde{\mathbf{x}}+\mathbf{n}) \\
&= \mathbf{U}^H\mathbf{U}\mathbf{\Sigma}\mathbf{V}\mathbf{V}^H\tilde{\mathbf{x}}+\mathbf{U}^H\mathbf{n} \\
&= \Sigma\tilde{\mathbf{x}}+\tilde{\mathbf{n}},
\end{aligned}$$

where $\tilde{\mathbf{n}} = \mathbf{U}^H\mathbf{n}$ and Σ is the diagonal matrix of singular values of **H** with σ_i the i^{th} diagonal.

U – the matrix M_rxM_r and M_r is the receive antenna

H – General channel matrix

- ✓ ***Note that multiplication by a unitary matrix*** does not change the distribution of the noise, i.e. **n** and $\tilde{n}$ are identically distributed.
- ✓ Thus, the transmit precoding and receiver shaping transform the MIMO channel into R_H parallel independent channels.
- ✓ Here the i^{th} channel has input $\tilde{x}_i$, output $\tilde{y}_i$, noise $\tilde{n}_i$ and channel gain σ_i.
- ✓ Note that the σ_i s are related since they are all functions of **H.**
- ✓ But since the resulting parallel channels do not interfere with each other, the channels with these gains are independent, linked only through the total power constraint.

3 .Parallel decomposition

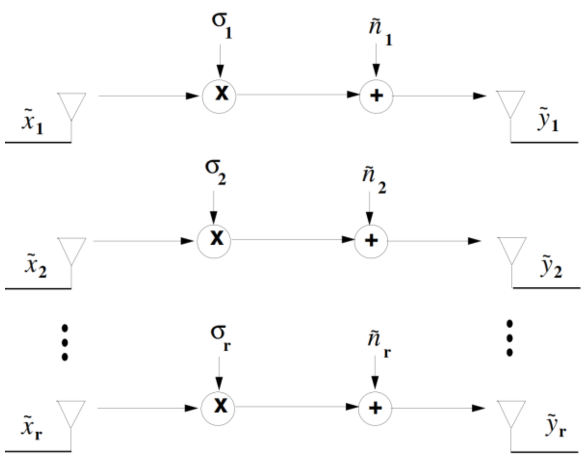


Figure 5.8: Parallel Decomposition of the MIMO Channel.

- ✓ This parallel decomposition is shown in ***Figure 5.8***.
- ✓ Since the parallel channels do not interfere with each other, the optimal ML demodulation complexity is linear in R_H, the number of independent paths that need to be decoded.
- ✓ Moreover, by sending independent data across each of the parallel channels, the MIMO channel can support R_H times the data rate of a system with just one transmit and receive antenna, leading to a multiplexing gain of R_H .

- ✓ Note, however, that the performance on each of the channels will depend on its gain σ_i.
- ✓ The next section will more precisely characterize the multiplexing gain associated with the Shannon capacity of the MIMO channel.
- ✓ Note: U and V unitary imply $\mathbf{UU^H} = \mathbf{I}_{\mathbf{M_r}}$ and $\mathbf{V^H V} = \mathbf{I}_{\mathbf{M_t}}$

Features

1. SVD precoding doesnot introduce noise enhancement.
2. If $N_r \geq N_t$ the complexity of the channel H is at $O(N_r\ N_t^2)$.
3. Closed loop spatial multiplexing can achieve high performance.

Linear Precoding

- ✓ Linear precoding is based on ZF beamforming
- ✓ It achieves high capacity
- ✓ As the number of user goes to infinity

ZF beamforming capacity = dirty paper coding capacity

The linear precoder decouples the input signal into orthogonal signal modes in the form of eigen-beams.

(*i*) In the case of perfect CSI [channel state information] the precoded orthogonal spatial modes match the channel eigen-directions. There is no interference between these signal streams.

(*ii*) With spatial CSI, precoder design must reduce the interference among signals

(*iii*) For perfect CSI at the transmitter, a diversity gain can also be delivered.

For example, precoding has been incorporated in IEEE 802.16 e as a closed loop MIMO scheme. 3GPP uses a closed-loop beamforming technique.

5.5.MIMO-BEAM FORMING

1. **Explain in detail how inherent delay in a multiuser system is overcome by beam forming. [May 2016]**
2. **Describe the concepts of beam forming. [May 2017]**
3. **Distinguish between different beam forming techniques. [Nov /De2017]**

Concept of Beamforming

- ✓ The multiple antennas at the transmitter and receiver can be used to obtain diversity gain instead of capacity gain.
- ✓ In this setting, the same symbol, weighted by a complex scale factor, is sent over each transmit antenna, so that the input covariance matrix has unit rank.
- ✓ This scheme is also referred to as **MIMO beamforming**.

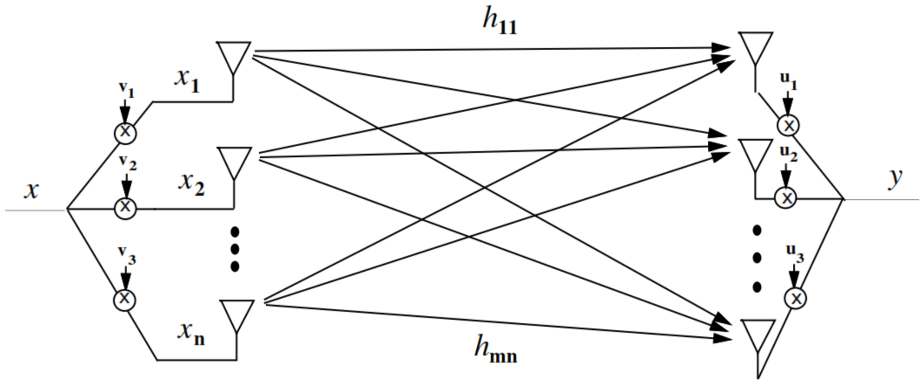


Figure 5.9: MIMO Channel with Beamforming.

- ✓ In the receiver side, symbol is weighted by a complex scale factor.
- ✓ Full diversity gain can be achieved by transmit beamforming and receive combining.

- ✓ The beamforming startegy corresponds to transmitter precoding and receiver shaping with

  V = v and

  U = u
- ✓ As indicated in the figure, the transmit symbol x is sent over the i^{th} antenna with weight v_i.
- ✓ On the receive side, the signal received on the ith antenna is weighted by u_i.

- ✓ Both transmit and receive weight vectors u and v are normalized so that,

$$\| u\| = \|v\| = 1$$
- ✓ The received received signal is

$$y = \mathbf{u}^*\mathbf{H}\mathbf{v}x + \mathbf{u}^*\mathbf{n}$$

  where if $\mathbf{n} = (n_1,...,n_{Mr})$ has i.i.d. elements, the statistics of $\mathbf{u^*n}$ are the same as the statistics for each of these elements.

- ✓ Beamforming provides diversity gain by coherent combining of the multiple signal paths.
- ✓ Channel knowledge at the receiver is typically assumed because this is required for coherent combining.
- ✓ The diversity gain then depends on whether or not the channel is known at the transmitter.
- ✓ When the channel matrix H is known, the received SNR is optimized by choosing u and v as the principal left and right singular vectors of the channel matrix H.

- ✓ The capacity of a scalar channel is

$$C = B \log_2(1 + \sigma_{max}^2 P / \sigma_n^2)$$

$$\sigma_{max}^2 \rightarrow \text{power gain}$$
- ✓ Implementation of transmit beamforming requires CSI at the transmitter.
- ✓ One solution is quantized beamforming.
- ✓ The receiver quantizes the beamforming vector using a fixed codebook.
- ✓ Codebook is availble at both the transmitter and the receiver.

  Eg: WCDMA - Wideband code division multiple access when CSI is not available at the transmitter, the Almouti or STBC scheme can be used to obtain full diversity gain and array gain.

Various codes:

- ✓ ***Space Time Block Codes:***
  - In space–time coding, redundancy is introduced by sending from each transmit antenna a differently encoded (and fully redundant) version of the same signal.
    Ex., Alamouti code, the most popular form of *space–time block codes*.
- ✓ ***Space–Time Trellis Codes:***
  - STBCs provide full diversity order, but no coding gain.
  - Such coding gain can be obtained from Space Time Trellis Codes (STTCs).
- ✓ **Code Books:**
  - The advantages of using codebooks are compounded if the number of spatial streams is smaller than the number of antenna elements.
  - Good performance can, however, only be achieved with a good design of the entries of the codebook.
  - The basic idea of codebook design is to exploit the intrinsic geometry of the underlying vector spaces.

Beam forming:

- ✓ In a wireless communication scenario, transmitted signals often propagate via just a few distinct paths, for example via a line-of-sight path between transmitter and receiver and/or via paths that are associated with significant reflectors and diffractors in the environment (such as large buildings or mountains).
- ✓ If the directions of these dominant propagation paths are known at the receiver side, beam forming techniques can be applied, in order to adjust the receiver beam pattern such that it has a high directivity towards the dominant angles of reception.
- ✓ Multiple-antenna techniques can also be utilized to improve the signal-to-noise ratio (SNR) at the receiver and to suppress co channel interference in a multiuser communications.
- ✓ The SNR gains achieved by means of beam forming are often called antenna gains or array gains.
- ✓ This suppresses the ***Co-Channel Interference*** (CCI) in a multiuser scenario, Thus improving the SNR at the receiver(s).
- ✓ Both goals can be achieved by means of beam forming techniques.
- ✓ This is achieved by means of adaptive antenna arrays, also called smart antennas or software antennas.

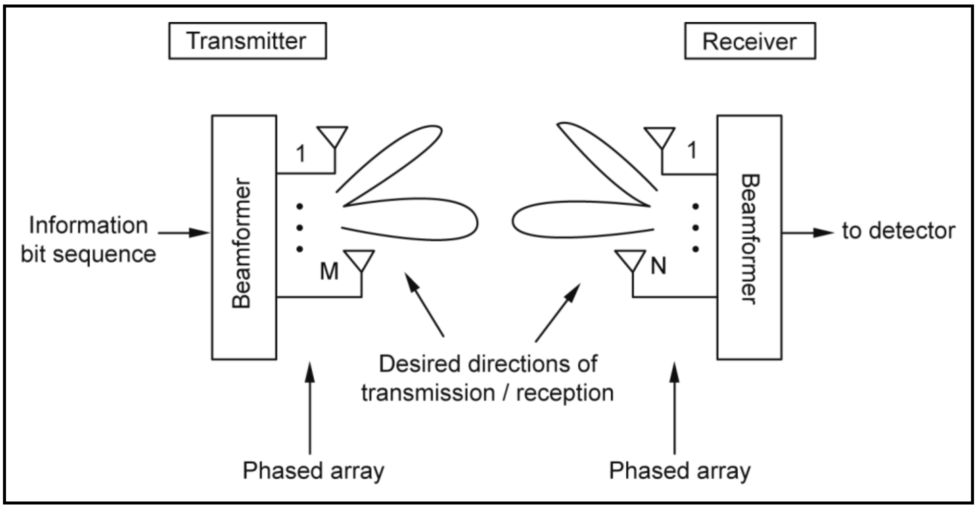


Figure 5.9 Principle of Beamforming

Smart antenna:

- ✓ The basic principle of beam forming is illustrated in Fig. 5.9.
- ✓ In the considered example, a beamformer is employed both at the transmitter and at the receiver side.
- ✓ In a practical system, the directions of dominant propagation paths must be estimated.
- ✓ This can, for example, be done by means of the well-known MUSIC algorithm.
- ✓ Moreover, when transmitter or receivers are moving, the antenna patterns must be updated on a regular basis.
- ✓ Such adaptive antenna arrays are often called smart antennas or software antennas in the literature.
- ✓ Due to the required equipment and processing power, however, the use of smart antenna technologies is currently limited to fixed stations, such as base stations, or mobile stations that are fixed on vehicles.
- ✓ Yet, for future wireless communication systems it is anticipated that smart antennas will also be feasible for hand-held devices employing small phased arrays fabricated by microstrip technology.
- ✓ In beam forming, both the amplitude and phase of each antenna element are controlled.
- ✓ Combined amplitude and phase control can be used to adjust side lobe levels and steer nulls better than can be achieved by phase control alone.
- ✓ The combined relative amplitude a_k and phase shift q_k for each antenna is called a "complex weight" and is represented by a complex constant w_k (for the k^{th} antenna).
- ✓ A beamformer for a radio transmitter applies the complex weight to the transmit signal (shifts the phase and sets the amplitude) for each element of the antenna.
- ✓ Smart antenna systems uses multiple antennas at the BS to Increase the ***coverage*** of the cell.
- ✓ Beam forming can be interpreted as ***linear filtering in the spatial domain***.
- ✓ Using beam forming techniques, the beam patterns of the transmit and receive antenna array can be steered in certain desired directions.
  The undesired directions (e.g.: directions of significant interference) can be suppressed ('nulled').
- ✓ Transmit and receive beam forming have been considered for cellular systems to improve the signal-to-Noise Ratio (SNR) or extend the coverage.

Beam forming techniques

Q. Distinguish between different beam forming techniques. [Nov/Dec 2107]

There are two types of beam forming.

1. **Digital Beam Forming**

- ✓ In digital beam forming, the operations of phase shift and amplitude scaling for each antenna element, and summation for receiving, are done digitally.
- ✓ Either general-purpose DSP's or dedicated beam forming chips are used.
- ✓ Digital processing requires that the signal from each antenna element is digitized using an A/D converter.
- ✓ Since radio signals above shortwave frequencies (>30 MHz) are too high to be directly digitized at a reasonable cost, digital beam forming receivers use analog "RF translators" to shift the signal frequency down before the A/D converters.
- ✓ Once the antenna signals have been digitized, they are passed to "digital down-converters" that shift the radio channel's center frequency down to 0 Hz and pass only the bandwidth required for one channel.
- ✓ The downconverters produce a "quadrature" baseband output at a low sample rate.

2.Adaptive Beam Forming

Figure 5.10 Adaptive beam forming

- ✓ The complex weights w_k for the antenna elements are carefully chosen to give the desired peaks and nulls in the radiation pattern of the antenna array.
- ✓ In a simple case, the weights may be chosen to give one central beam in some direction, as in a direction-finding application.
- ✓ The weights could then be slowly changed to steer the beam until maximum signal strength occurs and the direction to the signal source is found.
- ✓ In beam forming for communications, the weights are chosen to give a radiation pattern that maximizes the quality of the received signal.
- ✓ Usually, a peak in the pattern is pointed to the signal source and nulls are created in the directions of interfering sources and signal reflections
- ✓ Adaptive beam forming is the process of altering the complex weights on-the-fly to maximize the quality of the communication channel.
- ✓ Here are some commonly used methods:

Minimum Mean-Square Error

The shape of the desired received signal waveform is known by the receiver. Complex weights are adjusted to minimize the mean-square error between the beamformer output and the expected signal waveform.

Maximum Signal-to-Interference Ratio

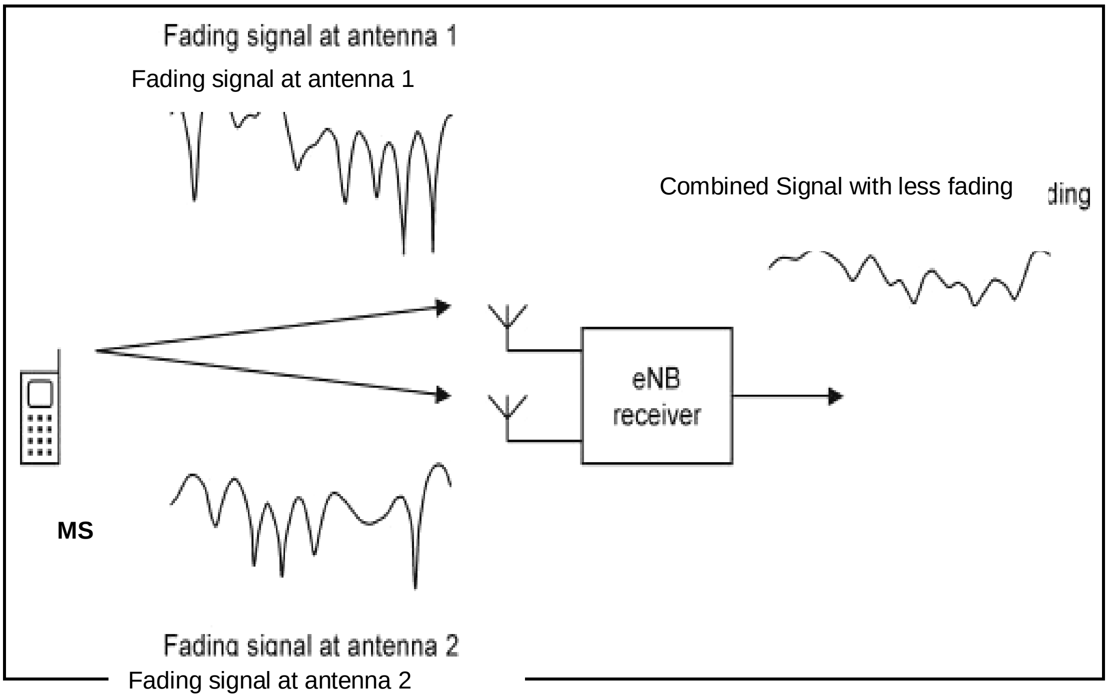
Where the receiver can estimate the strengths of the desired signal and of an interfering signal, weights are adjusted to maximize the ratio.

Minimum Variance

- ✓ When the signal shape and source direction are both known, chose the Weights to minimize the noise on the beamformer output antenna array.
- ✓ Adaptive beam forming systems for communications are sometimes referred to as "smart antenna" systems.
- ✓ For cellular telephone, one base station with a smart antenna system can support more than one user on the same frequency, as long as they are in different directions, by steering individual antenna beams at each user. This is sometimes called "spatial domain multiple access" (SDMA).
- ✓ It's estimated that the capacity of cellular telephone systems can be doubled by using smart antennas.
- ✓ Smart Adaptive Array Antenna can track the unknown interference signal in real time automatically.
- ✓ It can direct antenna pattern with nulls towards the interference and offer gain to the required signal to ensure the required signal reception, so output SINR (signal to Interference and Noise Ratio) is improved.

<u>Four techniques used to overcome delay problem in beam forming</u>

- ✓ In beam forming four techniques are used to overcome delay problem.



First method: Uplink with multiple receive antennas

- ✓ Space division multiple access (SDMA) is capacity-achieving:
- ✓ All users simultaneously transmit and are jointly decoded by the base-station.
- ✓ Total spatial degrees of freedom limited by number of users and number of receive antennas.
- ✓ Rule of thumb is to have a group of nr users in SDMA mode and different groups in orthogonal access mode.
- ✓ Each of the nr user transmissions in a group obtains the full receive diversity gain equal to n_r.

Orthogonal multiple access:

- ✓ **Uplink with multiple receive antennas is achieved by Orthogonal multiple access:**
- ✓ The array of multiple antennas is used to boost the received signal strength from the user within the cell via receive beam forming.
- ✓ One immediate benefit is that each user can lower its transmit power by a factor equal to the beam forming gain (proportional to n_r) to maintain the same signal quality at the base-station.
- ✓ This reduction in transmit power also helps to reduce inter-cell interference, so the effective SINR with the power reduction is in fact more than the SINR achieved in the original setting.
- ✓ Suppose all the K users are at the edge of the cell, W is the total bandwidth allotted to the cellular system.
- ✓ With SDMA used in each cell, K users simultaneously transmit over the entire bandwidth KW.

Second method: Uplink with multiple transmit and receive antennas

- ✓ The overall spatial degrees of freedom are still restricted by the number of receive antennas, but the diversity gain is enhanced.

Third method: Downlink with multiple transmit antennas

- ✓ Uplink-downlink duality identifies a correspondence between the downlink and the reciprocal uplink.
- ✓ Precoding is the analogous operation to successive cancelation in the uplink.
- ✓ A precoding scheme that perfectly cancels the intra-cell interference caused to a user was described.
- ✓ Precoding operation requires full CSI; hard to justify in an FDD system.

- ✓ With only partial CSI at the base-station, an opportunistic beam forming scheme with multiple orthogonal beams utilizes the full spatial degrees of freedom.

Fourth method: Downlink with multiple receive antennas

- ✓ Each user's link is enhanced by receive beam forming:
- ✓ Both a power gain and a diversity gain equal to the number of receive antennas are obtained.
- ✓ In downlink with multiple receive antennas, each user gets receive beam forming gain but reduced multiuser diversity gain.

1. No CSI-channel state information at the base-station: single spatial degree of freedom.
2. Full CSI: the uplink-downlink duality principle makes this situation analogous to the uplink with multiple receive antennas and now there are up to n_t spatial degrees of freedom.
3. Partial CSI at the base-station: the same spatial degrees of freedom as the full CSI.

This can be achieved by a modification of the opportunistic beam forming scheme: multiple spatially orthogonal beams are sent out and multiple users are simultaneously scheduled on these beams.

<u>Tradeoffs between Diversity, Beamforming Gain, and Spatial Multiplexing</u>:

- ✓ MIMO systems can be used to achieve spatial multiplexing, diversity, and/or beamforming.
- ✓ But, it is not possible to attain all the goals simultaneously at their full extent.
- ✓ First, the tradeoff between beamforming and diversity gain is found; this tradeoff also depends on the operating environment.
- ✓ Consider an LOS scenario.
- ✓ In this case, it is clear that the achievable beamforming gain is $N_t N_r$: The 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.
- ✓ In a heavily scattering environment, the slope of SNR distribution changes drastically due to the use of multiantenna elements.
- ✓ If the channel is known at the TX, the antenna weights are choose in such a way that it is unlikely that the receive signals are in a fading dip; furthermore, even if this happens, it is unlikely that *all* the receive signals are in a fading dip.
- ✓ In other words, it is easy to make sure that *at least one* receive signal has good quality.
- ✓ The diversity order is the slope of the BER versus SNR curve for very high SNRs:

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

- ✓ It can also be related to the slope of the SNR distribution at very low values of the SNR, and is thus a measure for the likelihood that all signals are in a bad fading dip simultaneously.
- ✓ The maximum diversity order in a heavily scattering environment can be shown to be $N_t N_r$.
- ✓ On the other hand, it also turns out that the maximum beamforming gain in such a heavily scattering environment is upper limited by $(\sqrt{N_t} + \sqrt{N_r})^2$.

- ✓ The reason is that it is not possible to form the transmit beam pattern in such a way that MPCs overlap constructively *at all receive antenna elements simultaneously*.
- ✓ Note the key tradeoff here between achieving full diversity order and beamforming gain.
- ✓ There is also a fundamental tradeoff between spatial multiplexing and diversity.
- ✓ Zheng and Tse [2003] showed that the optimum tradeoff curve between diversity order and rate r is piecewise linear, connecting the points:

$$d_{\text{div}}(r) = (N_{\text{t}} - r)(N_{\text{t}} - r), \quad r = 0, \ldots, \min(N_{\text{t}}, N_{\text{r}})$$

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

5.6 TRANSMITTER DIVERSITY AND RECEIVER DIVERSITY

Transmit diversity:

3. Write short notes on transmit diversity.

- ✓ In transmit diversity N_t transmit antennas and one receiver antenna are used.The transmit power is divided among these antennas.
- ✓ It is desirable in systems where more power, space and processing capability is available.

  eg. Cellular systems

  (*i*) Channel State Information (CSI) is known, the system is quite similar to receiver diversity.

  (*ii*) If CSI is not known, transmit diversity gain requires a combination of space and time diversity via the Alamouti scheme.
- ✓ The transmitter CSI can be obtained by feedback from the receiver.

Types of Transmit Diversity

1. Open-loop – CSI unknown at transmitter
2. Closed-loop – CSI is known

Channel Known at Transmitter (*Closed-Loop Transmit Diversity)*

- ✓ Consider a transmit diversity system with *M* transmit antennas and one receive antenna.
- ✓ It is assumed that, the path gain associated with the ith antenna given by $r_i e^{j\theta_i}$ is known at the transmitter.
- ✓ This is refered to as having channel side information (**CSI**) at the **T**ransmitter, or **CSIT**.
- ✓ Let *s(t)* denote the transmitted signal with total energy per symbol E_s .
- ✓ This signal is multiplied by complex gain $\alpha_i = a_i e^{-j\theta_i}, 0 \leq a_i \leq 1$ and sent through the i^{th} antenna.
- ✓ This complex multiplication performs both co-phasing and weighting relative to the channel gains.
- ✓ Due to the average total energy constraint E_s , the weights must satisfy $\sum_{i=1}^{M} a_i^2 = 1$.
- ✓ The weighted signals transmitted over all antennas are added "in the air", which leads to a received signal given by

$$r(t) = \sum_{i=1}^{M} a_i r_i s(t)$$

Let N_0 denote the noise PSD in the receiver.

- ✓ Suppose it is needed to set the branch weights to maximize received SNR.

- ✓ Using a similar analysis as in receiver MRC diversity, the weights a_i that achieve the maximum SNR are given by

$$a_i = \frac{r_i}{\sqrt{\sum_{i=1}^{M} r_i^2}},$$

and the resulting SNR is

$$\gamma_\Sigma = \frac{E_s}{N_0} \sum_{i=1}^{M} r_i^2 = \sum_{i=1}^{M} \gamma_i,$$

for $\gamma_i = r_i^2 E_s/N_0$ equal to the branch SNR between the i^{th} transmit antenna and the receive antenna.

- ✓ When the channel gains are known at the transmitter (transmitter diverity) is very similar to receiver diversity with MRC: the received SNR is the sum of SNRs on each of the individual branches.
- ✓ In particular, if all antennas have the same gain $r_i = r$, $\gamma_\Sigma = Mr^2E_s/N_0$, and *M*-fold increase over just a single antenna transmitting with full power.
- ✓ Using the Chernoff bound, we see that for static gains

$$P_s = \alpha_M Q(\sqrt{\beta_M \gamma_\Sigma}) \leq \alpha_M e^{-\beta_M \gamma_\Sigma/2} = \alpha_M e^{-\beta_M(\gamma_1+\ldots+\gamma_M)/2}.$$

- ✓ Integrating over the chi-squared distribution for γ_Σ yields

$$\overline{P}_s \leq \alpha_M \prod_{i=1}^{M} \frac{1}{1+\beta_M \overline{\gamma}_i/2}$$

In the limit of high SNR and assuming that the γ_i are identically distributed with $\overline{\gamma}_i = \overline{\gamma}$ this yields

$$\overline{P}_s \approx \alpha_M \left(\frac{\beta_M \overline{\gamma}}{2}\right)^{-M}.$$

- ✓ Thus, at high SNR, the diversity order of transmit diversity with MRC is *M*, so MRC achieves full diversity order.
- ✓ However, the performance of transmit diversity is worse than receive diversity.
- ✓ This is due to the extra factor of *M* in the denominator of the last equation,
- ✓ This results from dividing the transmit power among all the transmit antennas.
- ✓ Receiver diversity collects energy from all receive antennas, so it does not have this penalty.
- ✓ The analysis for EGC and SC assuming transmitter channel knowledge is the same as under receiver diversity, except that the transmit power must be divided among all transmit antennas.
- ✓ The complication of transmit diversity is to obtain the channel phase and, for SC and MRC, the channel gain, at the transmitter.
- ✓ These channel values can be measured at the receiver using a pilot technique and then fed back to the transmitter.

Channel Unknown at Transmitter - The Alamouti Scheme *(Open-Loop Transmit Diversity)*

- ✓ The transmitter has no CSIT. It is a 2 × 1 transmit diversity
- ✓ Alamouti's scheme is for two-antenna transmit diversity
- ✓ The transmit energy is divided equally between the two antennas.

- ✓ Symbol energy $= \frac{E_S}{2}$
- ✓ During first symbol period, two different symbols S_1 and S_2 are transmitted.
- ✓ During second symbol period, symbols $-S_2^*$ and S_1^* are transmitted.
- ✓ The received signal is give by

$$r(t) = \sqrt{0.5}\,(h_1 + h_2)S(t)$$

$S(t) \rightarrow$ Transmitted signal

$h_1, h_2 \rightarrow$ complex Gaussian random variables

- ✓ It has zero mean and variance of 1 .

For the two-symbol period, the received symbol is given by

$$y = H_A S + n$$

where

the output, $y = (y_1, y_2^*)^+$

transmitted symbol, $S = (S_1 \cdot S_2)^+$

AWG Noise $n = (n_1\ n_2)^+$ noise has mean = zero and power = N

The channel matrix is given as,

$$H_A = \begin{bmatrix} h_1 & h_2 \\ h_2^* & -h_1^* \end{bmatrix} \begin{bmatrix} S_1 \\ S_2 \end{bmatrix} + \begin{bmatrix} n_1 \\ n_2^* \end{bmatrix}$$

$$y = H_A S + n$$

for zero forcing receiver, $Z = (Z_1 \cdot Z_2)^+ = H_A^H\, y$

We can estimate S from Z

$$Z = (Z_1\ Z_2)^+ = (\,|h_1^2| + |h_2^2|\,)\,S + \tilde{n}$$

where $\tilde{n} = H_A^H n$ is a complex Gaussian noise vector with mean = zero

- ✓ Alamouti's scheme is a 2 × 1 transmit diversity. It can be generalized to multiple antenna.
- ✓ Alamouti's transmit diversity is known as transmit beamforming
- ✓ With MRC-Maximal Ratio Combiner, the received SNR linearly increases with the diversity order N_r.
- ✓ But 2 × 1 transmit diversity data rate = 1 × 2 receive diversity with a 3 dB power penalty.
- ✓ Due to this penalty, the received SNR doesnot always increase.
- ✓ The Alamouti scheme achieves a diversity order of 2 with array gain = 1
- ✓ For MRC, array gain = 2
- ✓ In space–time coding, redundancy is introduced by sending from each transmit antenna a differently encoded (and fully redundant) version of the same signal.

Ex., *Alamouti* code, the most popular form of *space–time block codes*, when the transmitter employs orthogonal flat-fading channel, where the complex channel gain from transmit antenna - 1 is given by $h1$ and the gain from transmit antenna - 2 to the receiver by $h2$.

- ✓ Now, transmit the two symbols, $c1$ and $c2$, from the two transmit antennas at time instant 1:

$$\mathbf{s}_1 = \frac{1}{\sqrt{2}} \begin{pmatrix} c_1 \\ c_2 \end{pmatrix}$$

- ✓ At the second time instant, transmit the vector

$$\mathbf{s}_2 = \frac{1}{\sqrt{2}} \begin{pmatrix} -c_2^* \\ c_1^* \end{pmatrix}$$

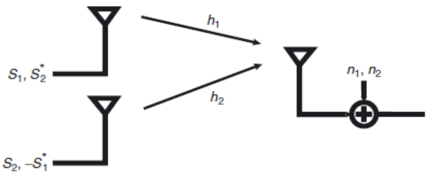


Figure: 5.12.a. Principle of Alamouti code

- ✓ The **Alamouti code** has been extended to the orthogonal space time block code (OSTBC) for $Nt > 2$.

For single receive antenna, SNR is given by

$$\text{SNR } \gamma_{td} = \frac{\gamma}{N_t} \sum_{i=1}^{N_t} |h_1|^2$$

$$\gamma = \gamma \text{ E } [\ |h_1|^2]$$

- ✓ As the number of transmit antenna N_t increases, received SNR approaches the average SNR.

$$\text{Received SNR} = \frac{\text{sum of SNRs in each branch}}{2}$$

<u>RECEIVER DIVERSITY</u>

Write short notes on receiver diversity.

System Model

- ✓ The 1 × 2 receive antennas are used, one transmit antenna and many receive antennas are used.
- ✓ In receiver diversity, the independent fading paths associated with multiple receive antennas are combined.
- ✓ Combining techniques are linear mostly.
- ✓ The output of the combiner is just a weighted sum of the different fading paths (or) branches.
- ✓ If weights of all the paths α_i are zero except one path means, only one path is passed.
- ✓ If more than one path α_i are non-zero means then the combiner adds together multiple paths.

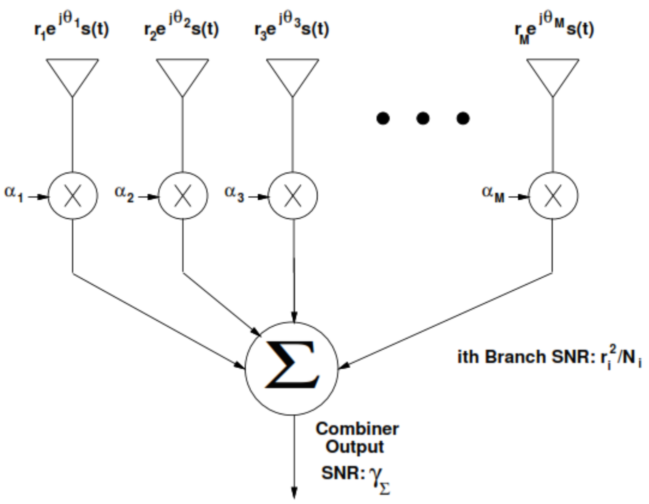


Figure 5.11. Linear Combiner

- ✓ The each path is weighted by a different value so it requires co-phasing. Co-phasing means the phase θ_i of the i^{th} branch is removed through multiplication by $\alpha_i = a_i\, e^{-j\ \theta_i}$.
- ✓ This phase removal requires coherent detection.
- ✓ Without co-phasing the resultant output exhibits fading.

- ✓ The main purpose of diversity is to coherently combine the independent fading paths so that the effects of fading are mitigated.

$$\alpha\sum = \sum_i a_i r_i$$

Combined output = Original transmitted signal S(t) × Random complex amplitude term

$$r_i = \sqrt{E_s}\ \text{, where } E_s \text{ is the energy per symbol}$$

PSD: Power Spectral Density of noise = $\frac{N_o}{2}$ on each branch

$$B\,T_s = 1$$

$B \rightarrow$ Bandwidth

$T_s \rightarrow$ symbol duration

The each branch has the same SNR

$$\text{SNR } \gamma_i = \frac{E_s}{N_o}$$

$$a_i = \frac{r_i}{\sqrt{N_o}} = \frac{\text{symbol energy}}{\text{Noise}}$$

The received SNR is

$$\gamma_\Sigma = \frac{\left(\sum_{i=1}^{M} a_i\, r_i\right)^2}{N_o \sum_{i=1}^{M} a_i^2} = \frac{\left(\sum_{i=1}^{M} \frac{E_s}{\sqrt{N_o}}\right)^2}{N_o \sum_{i=1}^{M} \frac{E_s}{N_o}} = \frac{M\,E_s}{N_o}$$

- ✓ In the absence of fading, SNR increases by M-fold.

Array Gain

- ✓ Increase in SNR in the absence of fading is referred as the array gain.
- ✓ Array Gain (A_g) is defined as the increase in the average combined SNR $\bar{\gamma}_\Sigma$ over the average branch SNR $\bar{\gamma}$.

$$A_g = \frac{\bar{\gamma}_\Sigma}{\bar{\gamma}}$$

The performance of the diversity system is defined interms of P_s and P_{out}

$P_s \rightarrow$ symbol power

$P_{out} \rightarrow$ output power

$$\bar{P}_s = \int_0^\infty P_s(\gamma)\, P_{\gamma\Sigma}(\gamma)\, d_\gamma$$

where $P_s(\gamma) \rightarrow$ probability of symbol error

$$P_{out} = \int_0^{\gamma_0} P_{\gamma\Sigma}(\gamma)\, d_\gamma$$

most favourite distribution for $\gamma\Sigma$ leads to decrease in P_s and P_{out} .

Diversity Gain

The average probability of error is given as

$$P_s = C\gamma^{-M}$$

where $M \rightarrow$ diversity order of the system.

$C \rightarrow$ constant that depends on modulation and coding.

Types of Combining Techniques are

(*i*) Selection combining
(*ii*) Threshold combining
(*iii*) Maximal-ratio combining
(*iv*) Equal gain combining

Selection Combining

- ✓ In selection combining (SC), the combiner outputs the signal on the branch with the highest SNR r_i^2 / N_i .
- ✓ This is equivalent to choosing the branch with the highest r_i^2 / N_i if the noise $N_i = N$ is the same on all branches.

- ✓ Since only one branch is used at a time, SC often requires just one receiver that is switched into the active antenna branch.
- ✓ However, a dedicated receiver on each antenna branch may be needed for systems that transmit continuously to simultaneously and continuously monitor SNR on each branch.
- ✓ With SC the path output from the combiner has an SNR equal to the maximum SNR of all the branches.
- ✓ Moreover, since only one branch output is used, co-phasing of multiple branches is not required, so this technique can be used with either coherent or differential modulation.

- ✓ For M-branch diversity, the cumulative distribution function (CDF) of γ_E is given by

$$P_{\gamma_E}(\gamma) = p(\gamma_E < \gamma) = p(\max[\gamma_1, \gamma_2, \ldots, \gamma_M] < \gamma)$$
$$= p(\gamma_i < \gamma)$$

- ✓ The pdf of γ_E is obtained by differentiating $P_{\gamma_E}(\gamma)$ relative to γ, and the outage probability by evaluating $P_{\gamma_E}(\gamma)$ at $\gamma = \gamma_0$.

- ✓ Assume, for M branches with uncorrelated Rayleigh fading amplitude r_i.
- ✓ Instantaneous SNR on the ith branch as $\bar{\gamma}_i = E[\gamma_i]$, the SNR distribution will be exponential:

$$p(\gamma_i) = \frac{1}{\gamma_i} e^{-\gamma_1/\gamma_i}$$

Outage probability $P_{out}(\gamma_o) = 1 - e^{-\gamma_o/\gamma_i}$

The outage probability of the selection combiner for the target γ_o is then

$$P_{out}(\gamma_o) = \prod_{i=1}^{M} p(\gamma_i < \gamma_o)$$

$$P_{out}(\gamma_o) = \prod_{i=1}^{M} (1 - e^{-\gamma_o/\gamma_i})$$

If the average SNR for all of the branches are the same, then this reduces to

$$P_{out}(\gamma_0) = p(\gamma_\Sigma < \gamma_0) = \left[1 - e^{-\gamma_0/\bar{\gamma}}\right]^M \quad \ldots(1)$$

Differentiating (1) with respect to γ_o yields the distribution for γ_Σ .

$$p_{\gamma_\Sigma}(\gamma) = \frac{M}{\bar{\gamma}}\left[1 - e^{-\gamma/\bar{\gamma}}\right]^{M-1} e^{-\gamma/\bar{\gamma}}$$

The average SNR of the combiner output in independent and identically distributed Rayleigh fading is

$$\begin{aligned}\bar{\gamma}_\Sigma &= \int_0^\infty \gamma p_{\gamma_\Sigma}(\gamma)d\gamma \\ &= \int_0^\infty \frac{\gamma M}{\bar{\gamma}}\left[1 - e^{-\gamma/\bar{\gamma}}\right]^{M-1} e^{-\gamma/\bar{\gamma}} d\gamma\end{aligned}$$

$$= \bar{\gamma}\sum_{i=1}^{M}\frac{1}{i}.$$

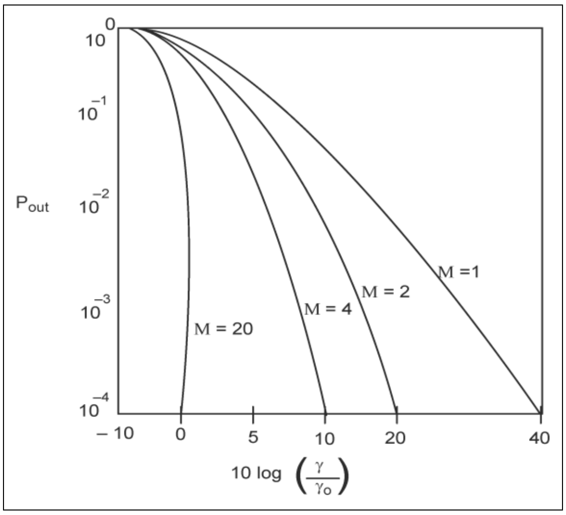


Figure 5.12. Outage Probability of selection combining in Rayleigh fading

- ✓ Thus, the average SNR gain and array gain increase with M but not linearly.
- ✓ The biggest gain is obtained by going from no diversity to two-branch diversity.
- ✓ Increasing the number of diversity branches from two to three will give much less gain than going from one to two.
- ✓ The number of branches (M) increases means array gain decreases.
- ✓ This is illustrated in Figure 5.12, which shows P_{out} versus $\bar{\gamma}/\gamma_0$ for different M in i.i.d. Rayleigh fading.
- ✓ For M = 1 to 2 at 1% outage probability yields 12 dB reduction in required SNR.
- ✓ Going from two-branch to three-branch diversity results in an additional reduction of approximately 7dB.
- ✓ For M = 3 to 4 branch, reduction in SNR of about 4 dB occur.
- ✓ It should be noted also that even with Rayleigh fading on all branches, the distribution of the combiner output SNR is no longer Rayleigh.

Drawbacks

- ✓ A dedicated receiver on each branch is required to continuously monitor branch SNR.
- ✓ Co-phasing is required to reduce interference.
- ✓ Number of branch (M) increases, array gain decreases.

Threshold Combining

- ✓ Selection Combining (SC) for systems that transmit continuously may require a dedicated receiver on each branch to continuously monitor branch SNR.
- ✓ A simpler type of combining,
- ✓ ***Threshold Combining***
  - ➢ This avoids the need for a dedicated receiver on each branch.
  - ➢ It scans each of the branches in sequential order.
  - ➢ Then it outputs the first signal with the SNR above a given threshold γ_T.

- This is the simplest combining technique.

- ✓ As in SC, since only one branch output is used at a time, co-phasing is not required.
- ✓ Thus, this technique can be used with either coherent or differential modulation.
- ✓ Once a branch is chosen, as long as the SNR on that branch remains above the desired threshold, the combiner outputs that signal.
- ✓ If the SNR on the selected branch falls below the threshold, the combiner switches to another branch.
- ✓ There are several criteria to decide ***the branch to switch*** by the combiner.
- ✓ The simplest criterion is to ***switch randomly*** to another branch.
- ✓ With only two-branch diversity this is equivalent to switching to the other branch when the SNR on the active branch falls below γ_T. This method is called ***switch-and-stay combining (SSC)***.

$\gamma_i \rightarrow$ SNR of the i^{th} branch.

- ✓ For two branch diversity with independent and identically distributed branch statistics, the cumulative distribution function of the combiner output is

$$P_{\gamma_\Sigma}(\gamma) = \begin{cases} P_{\gamma_1}(\gamma_T)P_{\gamma_2}(\gamma) & \gamma < \gamma_T \\ p(\gamma_T \le \gamma_1 \le \gamma) + P_{\gamma_1}(\gamma_T)P_{\gamma_2}(\gamma) & \gamma \ge \gamma_T \end{cases}$$

The outage probability P_{out} associated with a given γ_0 is obtained by evaluating $P_{\gamma_\Sigma}(\gamma)$ at $\gamma = \gamma_0$:

$$P_{out}(\gamma_0) = P_{\gamma_\Sigma}(\gamma_0) = \begin{cases} 1 - e^{-\gamma_T/\overline{\gamma}} - e^{-\gamma_0/\overline{\gamma}} + e^{-(\gamma_T+\gamma_0)/\overline{\gamma}} & \gamma_0 < \gamma_T \\ 1 - 2e^{-\gamma_0/\overline{\gamma}} + e^{-(\gamma_T+\gamma_0)/\overline{\gamma}} & \gamma_0 \ge \gamma_T. \end{cases}$$

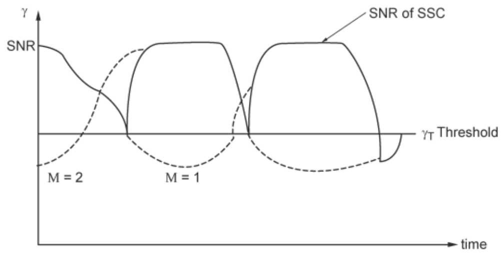


Figure 5.63. SNR of the SSC technique

Maximal-Ratio Combining

- ✓ In SC and SSC, the output of the combiner equals the signal on one of the branches.
- ✓ In maximal-ratio combining (MRC) the output is a weighted sum of all branches.
- ✓ The signals are co-phased and so complex amplitude $\alpha_i = a_i\, e^{-j\,\theta}$

$\theta_i \rightarrow$ phase of the incoming signal on the i^{th} branch

Envelope of combiner output

$$r = \sum_{i=1}^{M} a_i\, r_i$$

Assuming the same noise PSD N_0 in each branch

It yields, total noise power spectral density (PSD) = $\frac{N_{tot}}{2}$

At combiner output

$$N_{tot} = \sum_{i=1}^{M} a_i^2 N_0$$

Thus, the output SNR of combiner is

$$\gamma_\Sigma = \frac{r^2}{N_{tot}} = \frac{1}{N_0} \frac{\left(\sum_{i=1}^{M} a_i r_i\right)^2}{\sum_{i=1}^{M} a_i^2}$$

The goal is to choose a_i to maximize SNR, Optimal weight is obtained by using Cauchy-Schwartz inequality,

$$a_i^2 = r_i^2 / N_0$$

Thus, output SNR of combiner $= \gamma_\Sigma$

$$= \sum_{i=1}^{M} r_i^2 / N_0$$

$$= \sum_{i=1}^{M} \gamma_i$$

Thus the SNR of the combiner output is the sum on each branch.

The corresponding outage probability for a given threshold γ_0 is given by

$$P_{out} = p(\gamma_\Sigma < \gamma_0) = \int_0^{\gamma_0} p_{\gamma_\Sigma}(\gamma) d\gamma = 1 - e^{-\gamma_0/\overline{\gamma}} \sum_{k=1}^{M} \frac{(\gamma_0/\overline{\gamma})^{k-1}}{(k-1)!}$$

Advantages

- ✓ Number of branch (M) increases corresponding array gain increases linearly.
- ✓ MRC has better perfromance than selection combining.

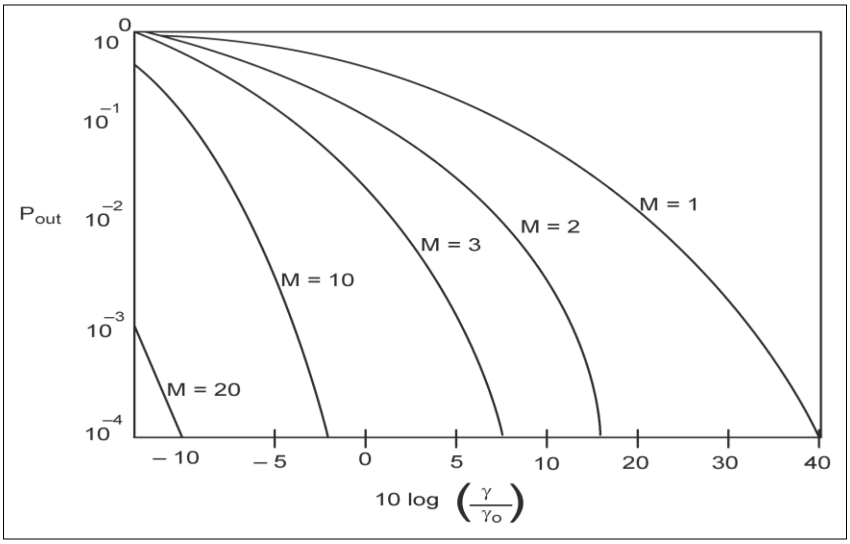


Fig. 5.74. P_{out} for MRC with i.i.d. Rayleigh fading.

At high SNR, the MRC achieves full diversity order.

Drawbacks

1. It requires the knowledge of the time varying SNR on each branch.
2. Complex technique.

Equal Gain Combining (EGC)

✓ Equal gain combining is a simple technique. It co-phases the signals on each branch and then combines them with equal weighting.

$$\alpha_i = e^{-\theta_i}$$

Combiner output SNR

$$\gamma_\Sigma = \frac{1}{N_0 M}\left(\sum_{i=1}^{M} r_i\right)^2$$

The pdf and CDF of γ_Σ do not exist in closed-form.

For i.i.d. Rayleigh fading and two-branch diversity and average branch SNR $\bar{\gamma}$, an expression for the CDF in terms of the Q function can be derived as

$$P_{\gamma_\Sigma}(\gamma) = 1 - e^{-2\gamma/\overline{\gamma}}\sqrt{\frac{\pi\gamma}{\overline{\gamma}}}e^{-\gamma/\overline{\gamma}}\left(1-2Q\left(\sqrt{2\gamma/\overline{\gamma}}\right)\right).$$

The resulting outage probability is given by

$$P_{out}(\gamma_0) = 1 - e^{-2\gamma_R} - \sqrt{\pi\gamma_R}e^{-\gamma_R}\left(1-2Q\left(\sqrt{2\gamma_R}\right)\right)$$

where

$$\gamma_R = \gamma_0/\overline{\gamma}$$

Differentiating $P_{\gamma_\Sigma}(\gamma)$ relative to γ yields the pdf

$$p_{\gamma_\Sigma}(\gamma) = \frac{1}{\overline{\gamma}}e^{-2\gamma/\overline{\gamma}} + \sqrt{\pi}e^{-\gamma/\overline{\gamma}}\left(\frac{1}{\sqrt{4\gamma\overline{\gamma}}} - \frac{1}{\overline{\gamma}}\sqrt{\frac{\gamma}{\overline{\gamma}}}\right)\left(1-2Q(\sqrt{2\gamma/\overline{\gamma}})\right).$$

for BPSK yields the average probability of bit error

$$\overline{P}_b = \int_0^\infty Q(\sqrt{2\gamma})p_{\gamma_\Sigma}(\gamma)d\gamma = .5\left(1-\sqrt{1-\left(\frac{1}{1+\overline{\gamma}}\right)^2}\right).$$

Advantage of EGC

✓ EGC is superior to selection combiner

✓ Simpler to implement

Drawbacks

✓ EGC is simpler than MRC but with 1 dB of power penalty.

5.7.CHANNEL STATE INFORMATION

1. Discuss in detail about the algorithms for categorizing the MIMO transmission by the amount of CSI that they require.

✓ Algorithms for MIMO transmission can be categorized by the amount of CSI that they require.

The following cases are discussed:

1. *Full CSI at the TX (CSIT) and full CSI at the RX (CSIR)*: in this ideal case, both the TX and the RX have full and perfect knowledge of the channel. This case obviously results in the highest possible capacity. However, it is difficult to obtain the full CSIT.

2. *Average CSIT and full CSIR*: in this case, the RX has full information of the instantaneous channel state, but the TX knows only the average CSI – e.g., the correlation matrix of **H** or the angular power spectrum. this is easier to achieve and does not require reciprocity or fast feedback; however, it does require calibration (to eliminate the nonreciprocity of transmit and receive chains) or slow feedback.

3. *No CSIT and full CSIR*: this is the case that can be achieved most easily, without any feedback or calibration. The TX simply does not use any CSI, while the RX learns the instantaneous channel state from a training sequence or using blind estimation.

4. *Noisy CSI* : when "full CSI" is assumed at the RX, this implies that the RX has learned the channel state perfectly. However, any received training sequence will be affected by additive noise as well as quantization noise. It is thus more realistic to assume a "mismatched RX," where the RX processes the signal based on the *observed* channel $\mathbf{H}_{obs}$, while in reality the signals pass through channel $\mathbf{H}_{true}$

$$\mathbf{H}_{true} = \mathbf{H}_{obs} + \Delta$$

Some papers have taken this into account by ad hoc modification of noise variance (replacing σ_n^2 by $\sigma_n^2 + \sigma_e^2$, where σ_e^2 is the variance of the entries of Δ).

5. *No CSIT and no CSIR*: it is remarkable that channel capacity is also high when neither the TX nor the RX have CSI. A generalization of differential modulation is used. For high SNR, capacity no longer increases linearly with $m=min(N_t,N_r)$, but rather increases as $\tilde{\mathbf{m}}(1-\tilde{\mathbf{m}}/T_{coh})$ where $\tilde{\mathbf{m}} = \min(N_t, N_r\lfloor T_{coh}/2\rfloor)$ and T_{coh} is the coherence time of the channel in units of symbol duration.

5.8 CAPACITY IN FADING AND NON-FADING CHANNELS

1. **Determine the capacity of frequency selective fading channel and explain the concept of water filling/ water pouring. (Nov/Dec 2015).**
2. **Discuss in detail, the capacity in fading and non-fading channels. (Apr/May 2017)**
3. **Derive the expression for the capacity of fading and non fading channels. [April/May 2018]**

1. Capacity in Nonfading Channels

- ✓ The first key step in understanding MIMO systems is ***the derivation of the capacity equation for MIMO systems in nonfading channels***.
- ✓ It is often known as "***Foschini's equation***" *[Foschini and Gans 1998]*.
- ✓ Let us start with the capacity equation for "normal" (single-antenna) Additive White Gaussian Noise (AWGN) channels.
- ✓ As Shannon showed, the information-theoretic (ergodic) capacity of such a channel is,

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

where

γ - the SNR at the RX,

H - the normalized transfer function from the TX to the RX

(now dealing with the frequency-flat case, the *transfer function is just a scalar number*).

- ✓ The key statement of this equation is that capacity increases logarithmically with the SNR. So that, boosting the transmit power is a highly ineffective way of increasing capacity.

- ✓ Consider now the MIMO case, where the channel is represented by matrix,

$$\mathbf{H} = \begin{pmatrix} h_{11} & h_{12} & \cdots & h_{1N_t} \\ h_{21} & h_{22} & \cdots & h_{2N_t} \\ \vdots & \vdots & \ldots & \vdots \\ h_{N_r1} & h_{N_r2} & \cdots & h_{N_rN_t} \end{pmatrix}$$

- ✓ Let us then consider a *singular value decomposition* of the channel:

$$\mathbf{H} = \mathbf{W}\mathbf{\Sigma}\mathbf{U}^{\dagger} \quad \rightarrow (1)$$

Where,

$\mathbf{\Sigma}$ **-** a diagonal matrix containing singular values

W and $\mathbf{U}^{\dagger}$ **-** unitary matrices composed of the left and right singular vectors, respectively.

- ✓ The received signal is then

$$\mathbf{r} = \mathbf{H}\mathbf{s} + \mathbf{n} \quad \rightarrow (2)$$
$$= \mathbf{W}\mathbf{\Sigma}\mathbf{U}^{\dagger}\mathbf{s} + \mathbf{n} \quad \rightarrow (3)$$

- ✓ Then, multiplication of the transmit data vector by matrix **U** and the received signal vector by $\mathbf{W}^{\dagger}$ diagonalizes the channel:

$$\mathbf{W}^{\dagger}\mathbf{r} = \mathbf{W}^{\dagger}\mathbf{W}\mathbf{\Sigma}\mathbf{U}^{\dagger}\mathbf{U}\widetilde{\mathbf{s}} + \mathbf{W}^{\dagger}\mathbf{n}$$
$$\widetilde{\mathbf{r}} = \mathbf{\Sigma}\widetilde{\mathbf{s}} + \widetilde{\mathbf{n}} \quad \rightarrow (4)$$

- ✓ Note that – because **U** and **W** are unitary matrices - $\widetilde{\mathbf{n}}$ has the same statistical properties as **n** – i.e., it is ***independent identically distributed (iid)*** white Gaussian noise.
- ✓ Now, computation of the capacity of nonzero entries Eq. (4) is rather straightforward.
- ✓ The matrix $\mathbf{\Sigma}$ is a diagonal matrix with R_H nonzero entries σ_k, where R_H is the rank of **H** (and thus defined as the number of nonzero singular values), and σ_k is the k^{th} singular value of **H**.
- ✓ So, there are R_H *parallel* channels (eigenmodes of the channel), and it is clear that the capacity of parallel channels just adds up.

- ✓ The capacity of channel **H** is thus given by the sum of the capacities of the eigenmodes of the channel:

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

where σ_n^2 is noise variance, and P_k is the power allocated to the k^{th} eigenmode; it is assumed that $\sum P_k = P$ is independent of the number of antennas.

This capacity expression can be shown to be equivalent to

$$C = \log_2\left[\det\left(\mathbf{I}_{N_r} + \frac{\overline{\gamma}}{N_t}\mathbf{H}\mathbf{R}_{ss}\mathbf{H}^{\dagger}\right)\right]$$

where

I$_{Nr}$ **-** the $N_r \times N_r$ identity matrix,

$\bar{\gamma}$ **-** the mean SNR per RX branch

***R**$_{SS}$* **-** the correlation matrix of the transmit data

- ✓ The distribution of power among the different eigenmodes (or antennas) depends on the amount of CSIT.
- ✓ The equations above confirm that capacity increases linearly with *min*(N_t,N_r,N_s), as the number of nonzero singular values R_H is upper-limited by *min*(N_t,N_r,N_s).

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:

$$C = \log_2\left[\det\left(\mathbf{I}_{N_r} + \frac{\overline{\gamma}}{N_t}\mathbf{H}\mathbf{H}^\dagger\right)\right]$$

- ✓ It is worth noting that (for sufficiently large N_s) the capacity of a MIMO system increases linearly with *min(Nt, Nr)*, irrespective of whether the channel is known at the TX or not.

Let us now look at a few special cases. To make the discussion easier, assume that *Nt, Nr = N*:

1. All transfer functions are identical – i.e., $h_{1,1} = h_{1,2} = \ldots\ldots = h_{N,N}$. This case occurs when all antenna elements are spaced very closely together, and all waves are coming from similar directions. In such a case, the rank of the channel matrix is unity. Then, capacity is

$$C_{\mathrm{MIMO}} = \log_2(1 + N\overline{\gamma})$$

In this case the SNR is increased by a factor of N compared with the single antenna case, due to beamforming gain at the RX. However, this only leads to a logarithmic increase in capacity with the number of antennas.

2. All transfer functions are different such that the channel matrix is full rank, and has N eigenvalues of equal magnitude. This case can occur when the antenna elements are spaced far apart and are arranged in a special way. In this case, capacity is

$$C_{\mathrm{MIMO}} = N\log_2(1 + \overline{\gamma})$$

and, thus, increases linearly with the number of antenna elements.

3. Parallel transmission channels – e.g., parallel cables. In this case, capacity also increases linearly with the number of antenna elements. However, the SNR per channel decreases with *N*, so that total capacity is

$$C_{\mathrm{MIMO}} = N\log_2\left(1 + \frac{\overline{\gamma}}{N}\right)$$

Figure shows capacity as a function of N for different values of SNR.

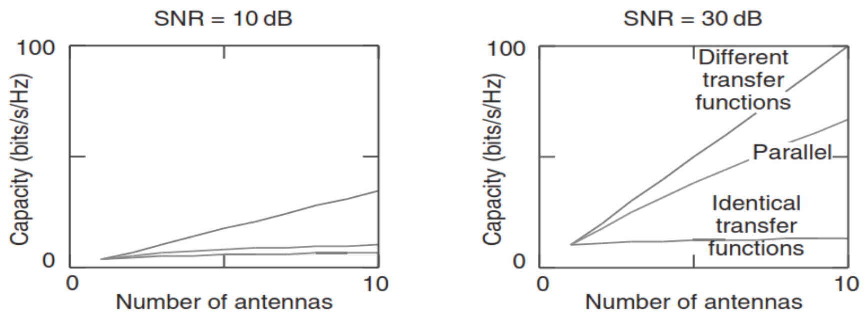


Figure 5.15. Capacity of multiple-input-multiple-output systems in additive white Gaussian noise channels.

2.Capacity in ading Channels

a.Capacity in Flat-Fading Channels:

General Concepts

- ✓ In the previous section, capacity for one given channel realization is considered – i.e., for one channel matrix **H**.
- ✓ Wireless systems are faced with channel fading.
- ✓ If the channel is Rayleigh fading, and fading is independent at different antenna elements, the h_{ij} are ***iid*** zero-mean, circularly symmetric complex Gaussian random variables with unit variance – i.e., the real and imaginary part each has variance 1/2.
- ✓ The power carried by each h_{ij} is chi-square-distributed with 2 degrees of freedom.
- ✓ This is the simplest possible channel model
- ✓ It requires the existence of "heavy multipath" – i.e., many MPCs of approximately equal strength as well as a sufficient distance between the antenna elements.
- ✓ The capacity increases linearly with the number of antenna elements.
- ✓ Thus, the existence of heavy multipath, which is usually considered a drawback, becomes a major advantage in MIMO systems.
- ✓ Because the entries of the channel matrix are random variables, the concept of information-theoretic capacity is considered.
- ✓ As a matter of fact, two different definitions of capacity exist for MIMO systems:
  - ➢ *Ergodic (Shannon) capacity*: this is the expected value of the capacity, taken over all realizations of the channel. This quantity assumes an infinitely long code that extends over all the different channel realizations.
  - ➢ *Outage capacity*: this is the minimum transmission rate that is achieved over a certain fraction of the time – e.g., 90% or 95%. We assume that data are encoded with a near-Shannon-limit achieving code that extends over a period that is much shorter than the channel coherence time.
- ✓ Thus, each channel realization can be associated with a (Shannon) capacity value. Capacity thus becomes a random variable (*rv*) with an associated cumulative distribution function (cdf); see also the discussion in Section 14.9.1.

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

- ✓ This is the capacity that can be achieved in a fading channel without CSI
- ✓ Figure 5.16 shows the result for some interesting systems at a 21-dB SNR.
- ✓ The (1, 1) curve describes a Single Input Single Output (SISO) system.
  - ➢ The median capacity is on the order of 6 bit/s/Hz
  - ➢ But the 5% outage capacity is considerably lower (on the order of 3 bit/s/Hz).
- ✓ When using a (1, 8) system
  - ➢ 1 transmit antenna and 8 receive antennas
  - ➢ The mean capacity does not increase that significantly – from 6 to 10 bit/s/Hz.
  - ➢ But, the 5% outage capacity increases significantly from 3 to 9 bit/s/Hz.
  - ➢ The reason for this is the much higher resistance to fading that such a diversity system has.
- ✓ When going to a (8, 8) system
  - ➢ A system with 8 transmit and 8 receive antennas
  - ➢ Both capacities increase dramatically.
  - ➢ The capacity is on the order of 46 bit/s/Hz
  - ➢ The 5% outage probability is more than 40 bit/s/Hz.

✓ The exact expression for the *ergodic capacity* was derived as

$$E\{C\} = \int_0^{\infty} \log_2\left[1 + \frac{\overline{\gamma}}{N_t}\lambda\right] \sum_{k=0}^{m-1} \frac{k!}{(k+n-m)!} \left[L_k^{n-m}(\lambda)\right]^2 \lambda^{n-m} \exp(-\lambda)\, d\lambda$$

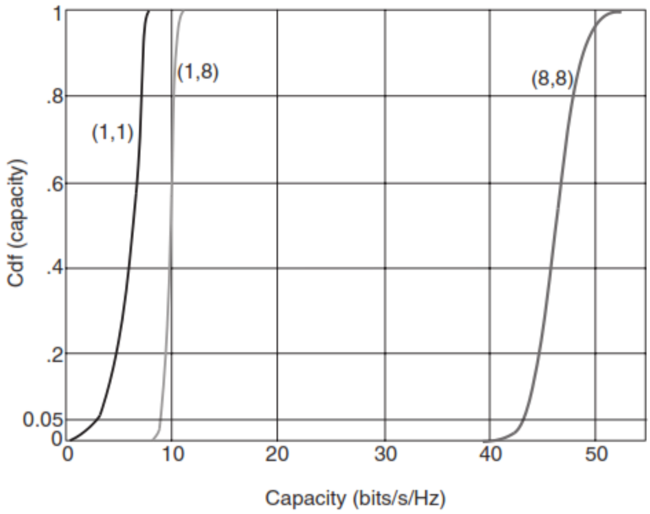


Figure 5.16. Cumulative distribution function of capacity for 1 × 1, 1 × 8, and the 8 × 8 optimum scheme.

b.Capacity of frequency-selective fading channels

1. **Determine the capacity of frequency selective fading channel and explain the concept of water filling/ water pouring. (Nov/Dec 2015).**
2. **Show that the optimal power allocation to maximize the capacity of a time-invariant frequency selective fading channel is given by the water filling formula.**

✓Let us consider the Shannon capacity of frequency-selective fading channels.

✓First consider the capacity of a time-invariant frequency-selective fading channel.

✓This capacity analysis is similar to that of a flatfading channel with the time axis replaced by the frequency axis.

Time-Invariant Channels

✓Consider a time-invariant channel with frequency response *H(f)*, as shown in Figure 5.17.

✓Assume a total transmit power constraint *P*.

✓When the channel is time-invariant it is typically assumed that *H(f)* is known at both the transmitter and receiver.

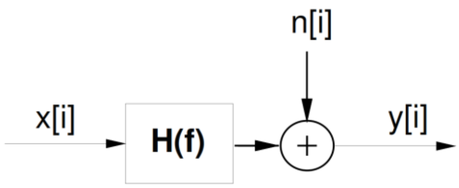


Figure 5.17: Time-Invariant Frequency-Selective Fading Channel.

Let us first assume that *H(f)* is block-fading.

✓ So that frequency is divided into subchannels of bandwidth B.

✓ Here $H(f)=H_j$ is constant over each block, as shown in Figure 5.18.

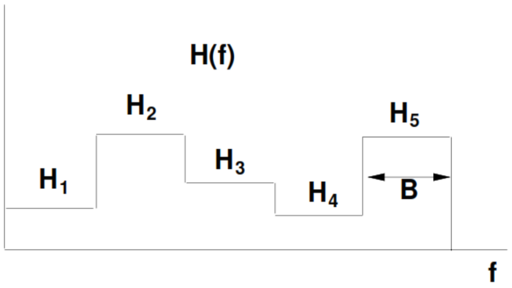


Figure 518: Block Frequency-Selective Fading

- The frequency-selective fading channel thus consists of a set of AWGN channels in parallel with SNR $|H_j|^2 P_j/(N_0 B)$ on the j^{th} channel, where P_j is the power allocated to the j^{th} channel in this parallel set, subject to the power constraint $\sum_j P_j \le P$.
- The capacity of this parallel set of channels is the sum of rates associated with each channel with power optimally allocated over all channels

$$C = \sum_{\max P_j : \sum_j P_j \le P} B \log_2 \left(1 + \frac{|H_j|^2 P_j}{N_0 B}\right)$$

- The optimal power allocation is found via the same ***Lagrangian technique*** used in the flat-fading case, which leads to the water-filling power allocation

$$\frac{P_j}{P} = \begin{cases} \frac{1}{\gamma_0} - \frac{1}{\gamma_j} & \gamma_j \ge \gamma_0 \\ 0 & \gamma_j < \gamma_0 \end{cases}$$

- For some cutoff value γ_0, where $|H_j|^2 P/(N_0 B)$ is the SNR associated with the j^{th} channel assuming it is allocated the entire power budget.
- This optimal power allocation is illustrated in Figure 5.19.

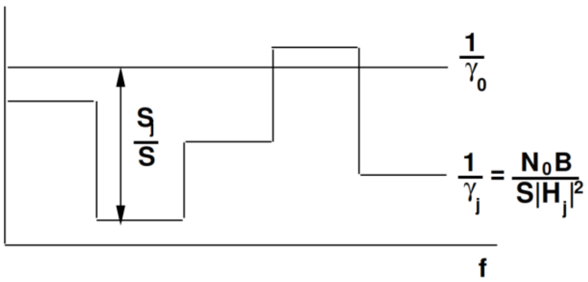


Figure 5.19: Water-Filling in Block Frequency-Selective Fading

- The cutoff value is obtained by substituting the power adaptation formula into the power constraint, so γ_0 must satisfy

$$\sum_j \left(\frac{1}{\gamma_0} - \frac{1}{\gamma_j}\right) = 1$$

- The capacity then becomes

$$C = \sum_{j:\gamma_j \ge \gamma_0} B \log_2(\gamma_j/\gamma_0)$$

- This capacity is achieved by sending at different rates and powers over each subchannel.
- The channel capacity is given by

$$C = \max_{P(f):\int P(f)df \le P} \int \log_2 \left(1 + \frac{|H(f)|^2 P(f)}{N_0}\right) df$$

where *H(f)* is continuous the capacity under power constraint *P* is similar to the case of the block-fading channel, with some mathematical complexities needed to show that

- ✓ The equation inside the integral is the ***incremental capacity*** associated with a given frequency *f* over the bandwidth *df* with power allocation *P(f)* and channel gain $|H(f)|^2$.
- ✓ The power allocation over frequency, P(f), that maximizes (4.27) is found via the Lagrangian technique.
- ✓ The resulting optimal power allocation is water-filling over frequency:

$$\frac{S(f)}{S} = \begin{cases} \frac{1}{\gamma_0} - \frac{1}{\gamma(f)} & \gamma(f) \geq \gamma_0 \\ 0 & \gamma(f) < \gamma_0 \end{cases}$$

- ✓ This results in channel capacity

$$C = \int_{f:\gamma(f)\geq\gamma_0} \log_2(\gamma(f)/\gamma_0)df.$$

*****

TWO MARKS

1. What is Antenna Diversity?(Nov/Dec 2015)

Signal is transmitted by more than one antenna via channel. The "best signal" copy is selected and processed**.**

2. What is transmit diversity?

Transmit diversity is radio communication using signals that originate from more than one antenna or path.

3.Define degrees of freedom.

- ✓ The symbol x(m) is the m^{th} sample of the transmitted signal: there are W samples per second.
- ✓ Each symbol is a complex number: we say that it represents one (complex) dimension or degrees of freedom.
- ✓ The received signal y(t) is also band limited to W.
- ✓ Thus the degree of freedom of the channel to be the dimension of the received signal space.

$$\text{Degrees of Freedom} = \min(N_T, N_R)$$

$$\text{Diversity} = N_T \times N_R$$

N_T is the number of propagation paths for the transmitted signal at time t,

N_R is the number of propagation paths for the received signal at time t.

4. Define MIMO.

Multi-input-multi-output (MIMO) systems are modern wireless systems with multiple antenna elements at both ends of the link.

5. List the applications of MIMO.

Multi-input-multi-output (MIMO) systems are used in

- ✓ Military wireless networks
- ✓ Soldier radios
- ✓ autonomous sensors or robotics

6. List the characteristics of Wireless Channels.

1. The signal transmitted over wireless channels may suffer dramatic attenuation, large delays, and severe signal distortion.
2. These non-ideal effects are often characterized by three factors, i.e., path loss, shadowing, and multi-path fading.

7. Define fading.

Fading is the variations of the signal strength over time and over frequency.

8. What are the types of fading?

Types of fading are

1. **Large-scale fading**, due to path loss of signal as a function of distance and shadowing by **large objects** such as buildings and hills.

   This occurs as the mobile moves through a distance of the order of the cell size, and is typically frequency independent.

2. **Small-scale fading**, due to the constructive and destructive interference of the multiple signal paths between the transmitter and receiver.

This occurs at the spatial scale of the order of the carrier wavelength, and is frequency dependent.

9. Write the features of MIMO.

- ✓ Mitigate problems inherent in ground-to-ground links (Non-LOS), which are the most common links used by wireless devices, including cell phones and WiFi.

10. Define Multipath Fading.

- ✓The signals eventually arriving at the receiver are often super positions of signals from multiple propagation paths.
- ✓The signals arriving from different paths may add up either constructively or destructively at the receiver.
- ✓The signal strength may fluctuate rapidly over time, space, and frequency. This is the so-called multipath fading effect.

11. What are the effects of Multipath Fading?

Multipath Fading results in fast and small-scale (or micro scale) amplitude and phase distortion.

12. List the channel parameters.

Two important channel parameters have often been used to describe the temporal and spectral characteristics of the fading channel, namely, coherence time and coherence bandwidth.

13. Define Doppler shift.

- ✓ The Doppler shift of the i-th path is defined as the amount of frequency-shift that occurs due to this time variation.
- ✓ The time-variation of time delay $\tau(t)$ causes the phase $\varphi(t)$ to change, resulting in a frequency-shift which we refer to as the Doppler effect.

14. What is the condition for flat fading, frequency selective fading?

- ✓Based on the channel variation characteristics in the frequency domain, we say that the channel is **flat fading**.
- ✓If the coherence bandwidth is much greater than the signal bandwidth, i.e., Wc>>W, and say that it is **frequency selective**, Otherwise the channel taps are independent and identically distributed (i.i.d.).

15. What is communication outage? How to overcome it?

- ✓ In wireless systems, transmission failures occur mostly when the channel is in **deep fade**, resulting in the so called **communication outage**.
- ✓ To overcome this effect, one can exploit different kinds of diversity techniques in space, time, and frequency.

16. Define SIMO, MISO and MIMO technique. (May/June 2016)

SIMO Techniques that only utilize multiple receive antennas (such as diversity reception schemes) are referred to as single-input multiple output (SIMO) techniques.

MISO techniques that utilize multiple transmit antennas only are called multiple-input single-output (MISO) techniques.

MIMO techniques that require multiple antennas at both ends of the wireless link (e.g., spatial multiplexing techniques such as the BLAST scheme are called multiple-input multiple-output (MIMO) techniques.

17. Define Narrowband and broadband techniques.

Narrowband techniques

-Transmission techniques that are designed for frequency flat fading channels are called narrowband techniques.

Wideband or broadband techniques

-**frequency-selective fading channels** (e.g., multiple antenna techniques that are based on OFDM) are referred to as **wideband or broadband techniques**.

18. Define Open-loop, closed-loop, and non-coherent techniques.

✓ Transmission techniques for multiple-antenna systems that require no channel knowledge at the transmitter side are referred to as **open-loop techniques**, because no feedback of channel state information from the receiver to the transmitter is required.

For example, space-time coding techniques and spatial multiplexing techniques such as the BLAST scheme.

✓ Transmission techniques that require full or partial channel knowledge at the transmitter (such as the transmitter-sided beam forming techniques or the limited feedback schemes are called **closed-loop techniques**.

✓ Finally, transmission techniques that require neither channel knowledge at the transmitter nor at the receiver side are called **non-coherent techniques**.

19. Compare various MIMO techniques.

MIMO Technique	Benefits	Best Conditions
SpatialMultiplexing	Increase throughput	Near to the base station (strong signal), best performance at low velocity
Transmit Diversity	Usually at base station: increases range by countering fading (less errors)	Good when beam forming is not appropriate
Receive Diversity	At mobile station: increases range by countering fading	Advantage over single antenna under all conditions
Beam forming	Base station: increasing range	Works best when distance is great (cell edge) and relatively low velocity

20. List the benefits of multiple antennas.

Multiple antenna benefits are

- ✓ Better overall signal quality
- ✓ Improved spectral efficiency
- ✓ Higher capacity
- ✓ Reduced power consumption
- ✓ Lower interference
- ✓ Improved coverage

21. List the applications of multiple antennas.

Applications of Multiple antennas (MIMO) are

✓ MIMO can reliably connect devices in home, such as computer networking devices, cabled video devices, phone lines, music, storage devices etc.

✓The IEEE 802.16e standard and the IEEE 802.11n standard also use MIMO system.

✓MIMO is used in mobile radio telephone standard such as 3GPP and 3GPP2 standard. 3GPP High Speed Packet Access Plus (HSPA+) and Long Term Evolution (LTE) standard use MIMO.

22. Define water filling method.

At low SNR-signal to noise ratios, all power will be allocated to the parallel channel with the largest SNR. This is called **Water-Filling**.

23. Define diversity gain.

Diversity gain is defined as the reduction in the probability of error due to multiple independent paths produced between the transmitter and receiver.

In other words if there are M transmits, N receive antennas, the order of diversity is MN. There is no diversity gain if the medium is line of sight channel.

24. Define Multiplexing gain.

- ✓Multiplexing gain is defined as the increase in the data rate.
- ✓Since independent data streams are sending through independent paths between multiple transmitters and multiple receivers.
- ✓In other words if there are M transmit antennas and N receive antennas, the increase in the data rate is min (M,N)-fold.

25. Define beam forming.

Beam forming is a signal processing technique used in sensor arrays for the transmission or reception of signal in the desired direction while nulling the interfering signal.

It has applications in wireless communications systems, radar, biomedicine, speech.

26. What is spatial multiplexing? [Nov/Dec 2017]

- ✓ **At the transmitter**, the information bit sequence is split into M sub-sequences, that are modulated and transmitted simultaneously over the transmit antennas using the same frequency band.
- ✓ **At the receiver**, the transmitted sequences are separated by employing an interference cancellation type of algorithm.

27. What are the uses of spatial multiplexing?

The uses of MIMO- spatial multiplexing is

- ✓ the increase of the system capacity
- ✓ to reduce the Bit Error Rate (BER).

28. What is receiving diversity (or) diversity reception technique? (Nov/Dec 2017, Apr/May 2015, Nov/Dec 2015, Nov/Dec 2012)

Diversity effect is achieved by receiving signals from several receive antenna is known as receive diversity.

29. List the different types of diversity. [April/May 2018]

1. Spatial diversity → several antenna elements separated in space.
2. Temporal diversity → Repetition of the transmit signal at different times.
3. Frequency diversity → Transmission of the signal on different frequencies.
4. Angular diversity → Multiple antennas with different antenna patterns.
5. Polarization → Multiple antennas receiving different polarizations

30. State true or false: Justify your answer. [April/May 2018]

Channel knowledge at the transmitter is not required in MIMO channels to extract multiplexing gain.

False.CSI (Channel State Information) is required

Channel knowledge at the transmitter is required in MIMO channels to extract diversity gain.

True.CSI (Channel State Information) is required

EC6801- WIRELESS COMMUNICATION
PART B-QUESTIONS

UNIT-I

1. Explain the **path loss model- free space model and two ray model.**
2. Explain in detail about **types of small scale fading**. **-Effects due to multipath time delay spread and Doppler spread**.
3. What do you mean by **path loss model**? Explain in detail about **log-distance path loss model**. (or)
   What is the need for **link calculation**? Explain with suitable example.
4. Explain the three **basic propagation mechanisms** in a mobile communication system. (or)
   In free space propagation describe how the signals are affected by **reflection, diffraction and scattering**.
5. Explain in detail about **Small Scale Fading Parameters** of Mobile Multipath Channels with their significance.

UNIT-II

1. Summarize the features of various **multiple access techniques (TDMA, CDMA & FDMA)** used in wireless mobile communication. State the advantages and disadvantages of each technique.
2. Explain the concept of **cellular topology (Cellular concept)** and frequency reuse fundamentals with examples.
3. Write short notes on **trunking and grade of service and channel assignment** of cell system.
4. Explain the principle of cellular networks and various types of **handoff techniques**.
5. Explain in detail how to **improve coverage and channel capacity** and various **interferences and increasing the system capacity** in cellular systems.

UNIT-III

1. Describe with neat diagram, the modulation technique of **OQPSK and π/4 DQPSK** employed in wireless communication.
2. What is **MSK**? Also derive the expression of MSK signal as a special type of FSK signal and explain its spectral density.
3. Explain in detail **Gaussian Minimum shift Keying (GMSK)** transmission and reception with necessary diagrams.
4. Discuss the **error performance** of different modulation schemes in fading channels
5. Draw the basic arrangements of **orthogonal frequency division Multiplexing (OFDM)** transceivers and discuss its overall operation.**-PAPR**

UNIT-IV

1. Analyze and compare the **error performance in fading channels** with and without diversity reception techniques and explain the operation of a **RAKE receiver** with neat diagram.
2. Explain in detail about adaptive equalizer and an **algorithm for an adaptive equalizer.**
3. Discuss in detailed about **linear non linear equalizer**.
4. Analyze various **diversity techniques** used in wireless communication.
5. Explain with diagram, the different techniques available for **signal combining**.

UNIT-V

1. Discuss in detail, the **capacity in fading and non-fading channels.**
2. Explain in detail **spatial multiplexing and system model** of a MIMO system.
3. Describe the concepts of **beam forming** in wireless communication.
4. Describe the concepts of **precoding**.
5. Write short notes on **transmit and receive diversity**.

Note: Problems in all units.