EC 6801- WIRELESS COMMUNICATION

UNIT I WIRELESS CHANNELS

PART-A

1. **What is meant by small and large scale fading? (June 2013)**

   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.

   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.

2. **What are the basic requirements for wireless services?(May 2014)**

   Wireless communication is a process of transmitting & receiving voice and data using electro-magnetic waves in free space. They are not physically connected. It is a Serial communication

3. **State the propagation effects in mobile radio. (May 2014)**

   When wavelength is less than the obstacle size, the in there is a chance of

   Blocking, Reflection, Refraction

4. **Interpret link budget equation. (May 2014)**

   In designing a system for reliable communications, it must to perform a link budge calculation to ensure that sufficient power is available at the receiver to close the link and to meet the SNR requirement. The basis for the link budget is the friis equation

5. **List the different types of propagation mechanisms.(Dec 2014)**

   Multipath propagation often lengthens the time required for the baseband portion of the signal to reach the receiver which can cause signal smearing due to inter-symbol interference.

6. **What are Rayleigh and Ricean fading? (June 2014)**

   Used to describe the statistical time varying nature of the received envelope of a flat fading signal OR the envelope of an individual multipath component Sum of 2 quadrature Gaussian noise signals obey Releigh distribution.

7. **What are the different modules of a basic cellular system? (Dec 2014)**

   It consist of Wireless wide-area networks (WWAN)

   Wireless local area networks (WLAN)

   Wireless personal area networks (WP AN

8. **What are the different fading effects due to Doppler spread?(Dec 2014)(Dec 2013)**

   Depends on QOW fast the baseband signal changes compared to the rate of change of the channel. Not due to propagation loss!!

9. **State the difference between small-scale fading and large scale fading. (May 2015)(June 2013)**

   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.

   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.

10. **Interpret snell's law. (May 2015) (June 2013)**

    It is a formula used to describe the relationship between the angles of incidence and refraction, when referring to light or other waves passing through a boundary between two different isotropic media.

11. **Define coherence time & coherence Bandwidth. (dec 2015)(May/June 2016)**

    Coherence time T_c is used to characterize the time varying nature of the frequency depressiveness of the channel in the domain

    $T_c = 1/f_m$ Doppler spread and coherence time are inversely proportional

    Coherence Bandwidth B_c is a statistical measure of the range of frequencies over which two frequency components have a strong potential for amplitude correlation.

12. **What is fading and Doppler spread. (May/June 2016)**

    Fading takes place in mobile signal propagation due to multi path time delay spread.Doppler spread is denoted as B_D and it is defined as a set of frequencies over which the Doppler spread at the receiver end is non zero value.For example if a pure sinusoidal tone of frequencyis transmitted and is denoted as f_c and the received signal spectrum is called Doppler spectrum consisting of components in the range from f_c-f_d to f_c+f_d, in which f_d refers to Doppler shift in frequency .

Part B

1. Discuss about technical challenges faced by the wireless communication. (June 2014)

Technical Challenges:

Multipath Propagation

For wireless communications, the transmission medium is the radio channel between transmitter TX and receiver RX. The signal can get from the TX to the RX via a number of different propagation paths. In some cases, a Line Of Sight (LOS) connection might exist between TX and RX. Furthermore, the signal can get from the TX to the RX by being reflected at or diffracted by different Interacting Objects (IOs) in the environment: houses, mountains (for outdoor environments), windows, walls, etc. The number of these possible propagation paths is very large. Each of the paths has a distinct amplitude, delay (runtime of the signal), direction of departure from the TX, and direction of arrival; most importantly, the components have different phase shifts with respect to each other. In the following, we discuss some implications of the multipath propagation for system design.

Fading:

A simple RX cannot distinguish between the different Multi Path Components (MPCs); it just adds them up, so that they interfere with each other. The interference between them can be constructive or destructive, depending on the phases of the MPCs. The phases, in turn, depend mostly on the run length of the MPC, and thus on the position of the Mobile Station (MS) and the IOs. For this reason, the interference, and thus the amplitude of the total signal, changes with time if TX, RX, or IOs is moving. This effect – namely, the changing of the total signal amplitude due to interference of the different MPCs – is called small-scale fading. Obstacles can lead to a shadowing of one or several MPCs. Imagine, e.g., the MS at first (at position A) has LOS to the Base Station (BS). As the MS moves behind the high-rise building (at position B), the amplitude of the component that propagates along the direct connection (LOS) between BS and MS greatly decreases. This is due to the fact that the MS is now in the radio shadow of the high-rise building, and any wave going through or around that building is greatly attenuated – an effect called shadowing. The MS has to move over large distances (from a few meters up to several hundreds of meters) to move from the light to the dark zone. For this reason, shadowing gives rise to large-scale fading.

Inter symbol Interference

The runtimes for different MPCs are different. We have already mentioned above that this can lead to different phases of MPCs, which lead to interference in narrowband systems. In a system with large bandwidth, and thus good resolution in the time domain,3 the major consequence is signal dispersion: in other words, the impulse response of the channel is not a single delta pulse but rather a sequence of pulses (corresponding to different MPCs), each of which has a distinct arrival time in addition to having a different amplitude and phase. This signal dispersion leads to InterSymbol Interference (ISI) at the RX.

Spectrum Limitations

The spectrum available for wireless communications services is limited, and regulated by international agreements. For this reason, the spectrum has to be used in a highly efficient manner. Two approaches are used: regulated spectrum usage, where a single network operator has control over the usage of the spectrum, and unregulated spectrum, where each user can transmit without additional control, as long as (s)he complies with certain restrictions on the emission power and bandwidth. In the following, we first review the frequency ranges assigned to different communications services. We then discuss the basic principle of frequency reuse for both regulated and unregulated access

Assigned Frequencies

The frequency assignment for different wireless services is regulated by the International Telecommunications Union (ITU), a suborganization of the United Nations. In its tri-annual conferences (World Radio Conferences), it establishes worldwide guidelines for the usage of spectrum in different regions and countries. Further regulations are issued by the frequency regulators of individual countries, including the Federal Communications

Commission (FCC) in the U.S.A., the Association of Radio Industries and Businesses (ARIB) in Japan, and the European Conference of Postal and Telecommunications Administrations (CEPT) in Europe.

While the exact frequency assignments differ, similar services tend to use the same frequency ranges all over the world:

• Below 100 MHz: at these frequencies, we find Citizens' Band (CB) radio, pagers, and analog cordless phones.

• 100–800 MHz: these frequencies are mainly used for broadcast (radio and TV) applications.

• 400–500 MHz: a number of cellular and trunking radio systems make use of this band. It is mostly systems that need good coverage, but show low user density.

• 800–1000 MHz: several cellular systems use this band (analog systems as well as secondgenerationcellular). Also some emergency communications systems (trunking radio) make use of this band.

• 1.8–2.1 GHz: this is the main frequency band for cellular communications. The current (secondgeneration) cellular systems operate in this band, as do most of the third-generation systems. Many cordless systems also operate in this band.

• 2.4–2.5 GHz: the Industrial, Scientific, and Medical (ISM) band. Cordless phones, Wireless Local Area Networks (WLANs) and wireless Personal Area Networks (PANs) operate in this band; they share it with many other devices, including microwave ovens.

• 3.3–3.8 GHz: is envisioned for fixed wireless access systems.

• 4.8–5.8 GHz: in this range, most WLANs can be found. Also, the frequency range between 5.7 and 5.8 GHz can be used for fixed wireless access, complementing the 3-GHz band. Also car-to-car communications are working in this band.

• 11–15 GHz: in this range we can find the most popular satellite TV services, which use 14.0–14.5 GHz for the uplink, and 11.7–12.2 GHz for the downlink.

Frequency Reuse in Regulated Spectrum

Since spectrum is limited, the same spectrum has to be used for different wireless connections in different locations. To simplify the discussion, let us consider in the following a cellular system where different connections (different users) are distinguished by the frequency channel (band around a certain carrier frequency) that they employ. If an area is served by a single BS, then the available spectrum can be divided into N frequency channels that can serve N users simultaneously. If more than N users are to be served, multiple BSs are required, and frequency channels have to be reused in different locations. For this purpose, we divide the area (a region, a country, or a whole continent) into a number of cells; we also divide the available frequency channels into several groups. The channel groups are now assigned to the cells. The important thing is that channel groups can be used in multiple cells. The only requirement is that cells that use the same frequency group do not interfere with each other significantly. It is fairly obvious that the same carrier frequency can be used for different connections in, say, Rome and Stockholm, at the same time. The large distance between the two cities makes sure that a signal from the MS in Stockholm does not reach the BS in Rome, and can therefore not cause any interference at all. But in order to achieve high efficiency, frequencies must actually be reused much more often – typically, several times within each city. Consequently, intercell interference.

Frequency Reuse in Unregulated Spectrum

In contrast to regulated spectrum, several services use frequency bands that are available to the general public. For example, some WLANs operate in the 2.45-GHz band, which has been assigned to "ISM" services. Anybody is allowed to transmit in these bands, as long as they (i) limit the emission power to a prescribed value, (ii) follow certain rules for the signal shape and bandwidth, and (iii) use the band according to the (rather broadly defined) purposes stipulated by the frequency regulators. As a consequence, a WLAN receiver can be faced with a large amount of interference. This interference can either stem from other WLAN transmitters or from microwave ovens, cordless phones, and other devices that operate in the ISM band. For this reason, a WLAN link must have the capability to deal with interference. That can be achieved by selecting a frequency band within the ISM band at which there is little interference, by using spread spectrum techniques, or some other appropriate technique.

Limited Energy

The requirement for small energy consumption results in several technical imperatives:

• The power amplifiers in the transmitter have to have high efficiency. As power amplifiers account for a considerable fraction of the power consumption in an MS, mainly amplifiers with an effi- ciency above 50% should be used in MSs. Such amplifiers – specifically, class-C or class-F amplifiers – are highly nonlinear. As a consequence, wireless communications tend to use modulation formats that are insensitive to nonlinear distortions. For example, constant envelope signals are preferred.

• Signal processing must be done in an energy-saving manner. This implies that the digital logic should be implemented using power-saving semiconductor technology like Complementary Metal Oxide Semiconductor (CMOS), while the faster but more power-hungry approaches like Emitter Coupled Logic (ECL) do not seem suitable for MSs. This restriction has important consequences for the algorithms that can be used for interference suppression, combating of ISI, etc.

• The RX (especially at the BS) needs to have high sensitivity. For example, Global System for Mobile Communications (GSM) is specified so that even a received signal power of −100 dBm leads to an acceptable transmission quality. Such an RX is several orders of magnitude more sensitive than a TV RX. If the GSM standard had defined −80 dBm instead, then the transmit power would have to be higher by a factor of 100 in order to achieve the same coverage. This in turn would mean that – for identical talktime – the battery would have to be 100 times as large – i.e., 20 kg instead of the current 200 g. But the high requirements on RX sensitivity have important consequences for the construction of the RX (low-noise amplifiers, sophisticated signal processing to fully exploit the received signal) as well as for network planning.

• Maximum transmit power should be used only when required. In other words, transmit power should be adapted to the channel state, which in turn depends on the distance between TX and RX (power control). If the MS is close to the BS, and thus the channel has only a small attenuation, transmit power should be kept low. Furthermore, for voice transmission, the MS should only transmit if the user at the MS actually talks, which is the case only about 50% of the time (Discontinuous Voice Transmission (DTX)).

• For cellular phones, and even more so for sensor networks, an energy-efficient "standby" or "sleep" mode has to be defined.

Several of the mentioned requirements are contradictory. For example, the requirement to build an RX with high sensitivity (and thus, sophisticated signal processing) is in contrast to the requirement of having energy-saving (and thus slow) signal processing. Engineering tradeoffs are thus called for.

2. (i) Explain how signal propagates against free space attenuation and reflection. (16) (June 2014)

Free Space Propagation

For propagation distances *d* much larger than the square of the antenna size divided by the wavelength, the far-field of the generated electromagnetic wave dominates all other components (in the far-field region the electric and magnetic fields vary inversely with distance). In free space, the power radiated by an *isotropic antenna* is spread uniformly and without loss over the surface of a sphere surrounding the antenna. An isotropic antenna is a hypothetical entity. Even the simplest antenna has some directivity. For example, a linear dipole has uniform power flow in any plane perpendicular to the axis of the dipole (omnidirectionality) and the maximum power flow is in the equatorial plane.

The surface area of a sphere of radius d is 4πd2, so that the power flow per unit area w(power flux in watts/meter2) at distance d from a transmitter antenna with input accepted power pT and antenna gain GT is

$$w = \frac{p_r G_r}{4\pi d^2}.$$

Transmitting antenna gain is defined as the ratio of the intensity (or power flux) radiated in some particular direction to the radiation intensity that would be obtained if the power accepted by the antenna were radiated isotropically. When the direction is not stated, the power gain is usually taken in the direction of maximum power flow. The product GT pT is called the effective radiated power (ERP) of the transmitter. The available power pR at the terminals of a receiving antenna with gain GR is

$$p_R = \frac{p_T G_T}{4\pi d^2} A = \left(\frac{\lambda}{4\pi d}\right)^2 G_R p_T G_T$$

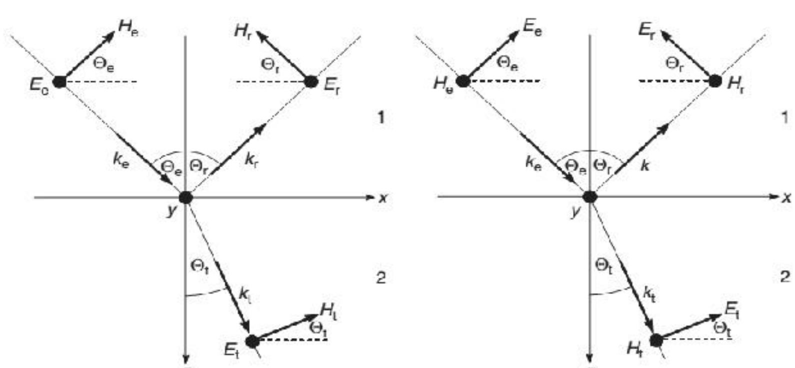
where A is the effective area or aperture of the antenna and $G_R = 4\pi A/\lambda^2$

The wavelength $\lambda = c / fc$ with c the velocity of light and fcthe carrier frequency.

While cellular telephone operator mostly calculate in received powers, in the planning of the coverage area of broadcast transmitters, the CCIR recommends the use of the electric field strength E at the location of the receiver. The conversion is $E = \sqrt{120\pi p_R}$

Reflection

Snell's Law



The reflection and transmission coefficients are different for TE and for TM waves. For TM polarization:

$$\rho_{TM} = \frac{\sqrt{\delta_2}\cos\Theta_e - \sqrt{\delta_1}\cos(\Theta_t)}{\sqrt{\delta_2}\cos\Theta_e + \sqrt{\delta_1}\cos(\Theta_t)}$$

$$T_{TM} = \frac{2\sqrt{\delta_1}\cos(\Theta_e)}{\sqrt{\delta_2}\cos\Theta_e + \sqrt{\delta_1}\cos(\Theta_t)}$$

and for TE polarization:

$$\rho_{TE} = \frac{\sqrt{\delta_1}\cos(\Theta_e) - \sqrt{\delta_2}\cos(\Theta_t)}{\sqrt{\delta_1}\cos(\Theta_e) + \sqrt{\delta_2}\cos(\Theta_t)}$$

$$T_{TE} = \frac{2\sqrt{\delta_1}\cos(\Theta_e)}{\sqrt{\delta_1}\cos(\Theta_e) + \sqrt{\delta_2}\cos(\Theta_t)}$$

The d^{-4} *Power Law*

$$P_{RX}(d) \approx P_{TX}G_{TX}G_{RX}\left(\frac{h_{TX}h_{RX}}{d^2}\right)^2$$

3. Explain in detail two path model propagation mechanisms. (June 2014)

Derive the expressions for the total Electric field, $E_{TOT}(d)$ and received power at distance , $P_r(d)$ using two –ray ground reflection model.(NOV/DEC 2015),(May/June 2014)

A single line-of-sight path between two mobile nodes is seldom the only means of propagation. The two-ray ground reflection model considers both the direct path and a ground reflection path. It is shown that this model gives more accurate prediction at a long distance than the free space model. The received power at distance d is predicted by

$$P_r(d) = \frac{P_t G_t G_r {h_t}^2 {h_r}^2}{d^4 L} \qquad (1)$$

The two ray Ground reflection model is a useful progation mode based on both the direct path and the ground refelected path between the transmitter and receiver.

In most of mobile systems,the maximum T-R separation is about few 10's of km and the earth is assumed to be flat.

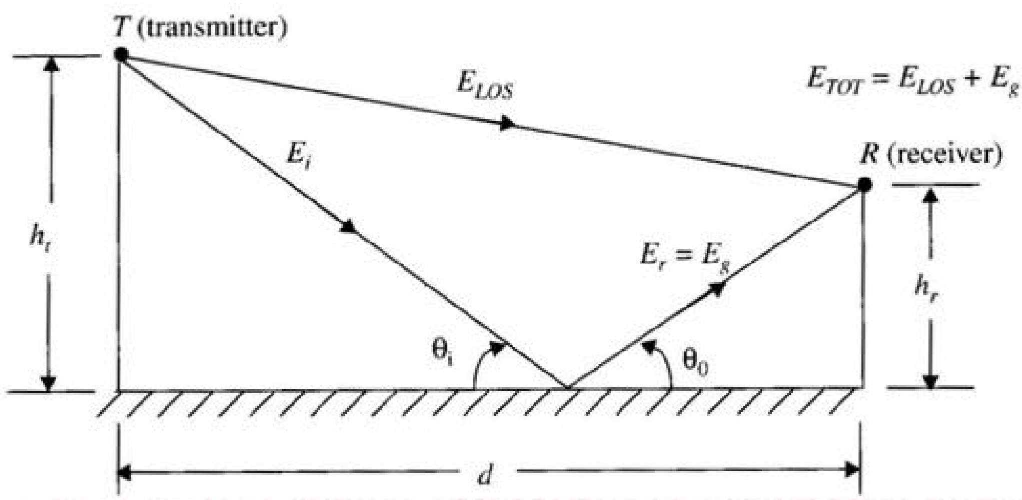
The total received E-field E_{TOT},is the result of direct Line-of-sight component E_{LOS} and the grond reflected components E_g

$E_{TOT} = E_{LOS} + E_g$

From figure

ht—height of Transmitter

hr—height of receiver.



1.1Two ray ground reflection model

If Eo is the free space electric field at a distance do from the transmitter, then at the distance $d>d_o$, the free space electric field is given by

$E(d,t)=\frac{E_o d_o}{d}\cos(w_c(t-\frac{d}{c}))$ for $d>d_o$

Where $\left|E(d,t)=\frac{E_o d_o}{d}\right|$ represent the envelope of the electric field at 'd' meter from the transmitter.

Two propagating waves arrive at the receiver.

1)The direct wave that travels at a distance d'

and

2)The reflected wave that travels a distance d’’

Finally,we can get the total electric field is the sum of above two components and is given by

$$E_{TOT}=\frac{2Eod\ o}{d}\frac{2\pi hthr}{\lambda_d}$$

Path loss:

For large values of d ,path loss is independent of frequency.It depends upon antenna heights h_t and h_r

PL(dB)=40 logd-[10log G_t+10log G_r+20 logh_t+20 log h_r]

4. i) Discuss in detail Two Ray Rayleigh Fading model (8)

ii) Describe on Rician distribution. (8) (Dec 2014)

Rayleigh fading model

$$E(t)=\sum_{i=1}^{N}|a_i|\cos[2\pi f_c t-2\pi\nu_{\max}\cos(\gamma_i)t+\varphi_i]$$

Rewriting this in terms of in-phase and quadrature-phase components in real passband notation,we obtain

$$E_{\mathrm{BP}}(t)=I(t)\cdot\cos(2\pi f_c t)-Q(t)\cdot\sin(2\pi f_c t)$$

$$I(t)=\sum_{i=1}^{N}|a_i|\cos[-2\pi\nu_{\max}\cos(\gamma_i)t+\varphi_i]$$

$$Q(t)=\sum_{i=1}^{N}|a_i|\sin[-2\pi\nu_{\max}\cos(\gamma_i)t+\varphi_i]$$

A zero-mean Gaussian random variable has the pdf:

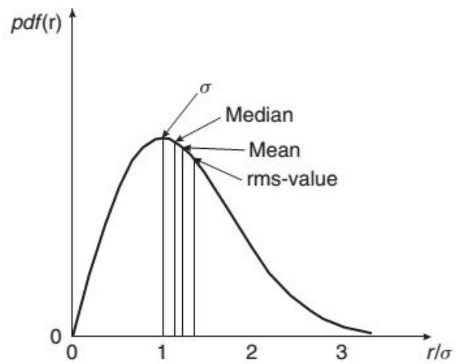
$$pdf_x(x)=\frac{1}{\sqrt{2\pi}\sigma}\exp\left(-\frac{x^2}{2\sigma^2}\right)$$

and a pdf for *r* – namely, a Rayleigh distribution:

$$pdf_r(r)=\frac{r}{\sigma^2}\cdot\exp\left[-\frac{r^2}{2\sigma^2}\right]\quad 0\le r<\infty$$

Properties of the Rayleigh Distribution

$$\left.\begin{array}{rl}
\text{Mean value} & \bar{r} = \sigma\sqrt{\frac{\pi}{2}} \\
\text{Mean square value} & \overline{r^2} = 2\sigma^2 \\
\text{Variance} & \overline{r^2} - (\bar{r})^2 = 2\sigma^2 - \sigma^2\frac{\pi}{2} = 0.429\sigma^2 \\
\text{Median value} & r_{50} = \sigma\sqrt{2 \cdot ln2} = 1.18\sigma \\
\text{Location of maximum} & \max\{pdf(r)\} \text{ occurs at } r = \sigma
\end{array}\right\}$$



Pdf of a Rayleigh distribution.

Ricean Distribution

The pdf of the amplitude is given by the *Rice distribution*

$$pdf_r(r) = \frac{r}{\sigma^2} \cdot \exp\left[-\frac{r^2 + A^2}{2\sigma^2}\right] \cdot I_0\left(\frac{rA}{\sigma^2}\right) 0 \le r < \infty \qquad \overline{r^2} = 2\sigma^2 + A^2$$

*I*0*(x)* is the modified Bessel function of the first kind, zero order.

5. Explain in Detail-Doppler spread and coherence time.

Doppler shift: A receiver is moving toward the source. Zero crossings of the signal appear faster therefore the received frequency is higher The opposite effect occurs if the receiver is moving away from the source.For example just as a train whistle or car horn appears to have a different pitch, depending on whether it is moving towards or away from one's location, radio waves demonstrate the same phenomenon.

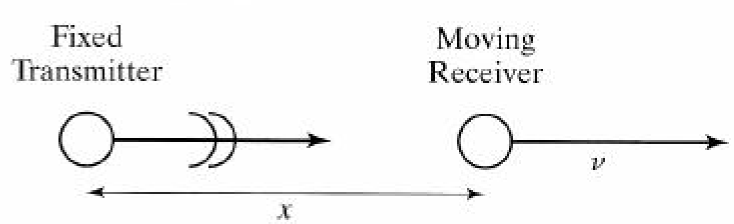


Illustration of Doppler effect

For complex envelope emitted by transmitter is Ae $^{j2\pi fot,}$ with the A(x) is the amplitude and c is the speed of light, then the signal at a points along the x-axis is given by

$$\tilde{r}(t,x) = A(x)e^{j2\pi f_0(t - x/c)} \qquad (2.39)$$

If x represents the position of the constant velocity receiver, then we may write

$$x = x_0 + vt \qquad (2.40)$$

Where x_o is the receiver' s initial position and *v* is its velocity. Substituting Eq. (2.40) into (2.39) the signal at the receiver is

$$\begin{aligned}\tilde{r}(t) &= A(x_0 + vt)e^{j2\pi f_0\left(t - \frac{x_0 + vt}{c}\right)} \\ &= A(x_0 + vt)e^{-j2\pi f_0 x_0/c}e^{j2\pi f_0(1 - v/c)t}\end{aligned} \qquad (2.41)$$

If we focus on the frequency term in the last exponent of equation the received frequency is given by

$$f_r = f_0\left(1 - \frac{v}{c}\right) \qquad (2.42)$$

The Doppler shift is given by

$$f_D = f_r - f_0 = -f_0\frac{v}{c} \qquad (2.43)$$

Relationship between Doppler frequency and velocity

$$\frac{v}{c} = -\frac{f_D}{f_0} \quad (2.44)$$

If the terminal motion and the direction of radiation are at an angle ψ, shift can be expressed as

$$f_D = -\frac{f_0}{c} v \cos\psi \quad (2.45)$$

For operating frequencies between 100MHz and 2GHz and for speeds up to 100Km/hr, the Doppler shift can be as large as 185 Hz

UNIT II CELLULAR ARCHITECTURE

PART-A

1. What are the effects of multi path propagation on CDMA? (May 2015,2016)(Dec 2014)

Reflection - occurs when signal encounters a surface that is large relative to the wavelength of the signal.

Diffraction - occurs at the edge of an impenetrable body that is large compared to wavelength of radio wave.

2. **State advantages of CDMA over FDMA? (Dec 2014)**

   This is the best & required wireless access method. Many wireless users are employed in the CDMA along with Various bandwidth needs, Switching methods, Technical characteristics.

3. **Mention a few techniques used to expand the capacity of a cellular system.(May 2015)**

   Increasing the amount of spectrum used, more efficient modulation format and coding, discontinuous transmission, multi user detection, reduction of cell radius, use of sector cells and multiple antennas.

4. **Define frequency reuse distance. (Dec 2012)**

   It is defined as the distance between two cells that can use the same frequency channels.

5. **What is meant by frequency reuse or frequency planning? (June 2013)(May /June 2016)**

   By limiting the coverage area to within the boundaries of a cell, the same group of channels may be used to cover different cells that are separated from one another by distances large enough to keep interference levels within tolerable limits. This design process of selecting and allocating channel groups for all of the cellular base stations within a system is called frequency reuse.

6. **What are the different methods available to increase the capacity of the system? (May 2012)**

   Increasing the amount of spectrum used, more efficient modulation format and coding, discontinuous transmission, multi user detection, reduction of cell radius, use of sector cells and multiple antennas.

7. **What is handoff? (Dec 13)**

   When the person is Moving from one BS to other without interrupting connection.

8. **What is Signal-to-Noise Ratio?**

   Wireless systems are required to provide a certain minimum transmission quality This transmission quality in turn requires a minimum Signal-to-Noise Ratio (SNR) at the receiver (RX).

9. Define signal to self-interference ratio.

The signal-to-interference ratio (S / I or SIR), also known as the carrier-to- interference ratio (CII, CIR), is the quotient between the average received modulated - carrier power S or C and the average received co-channel interference power.

10. What is channel assignment.

It is responsible for channel assignment, maintenance of link quality and handover, power control, coding, and encryption

11. Define Grade of service(GOS). (dec 2015)

It is a measure of the ability of a user to access a trunked system during the busiest hour

12. Define co-channel reuse ratio (Dec 2015).

co-channel reuse ratio(Q) is a function of the radius of the cell(R)and distance between the centres of the nearest co-channel cells(D)

Q=D/R co-channel reuse ratio(Q)

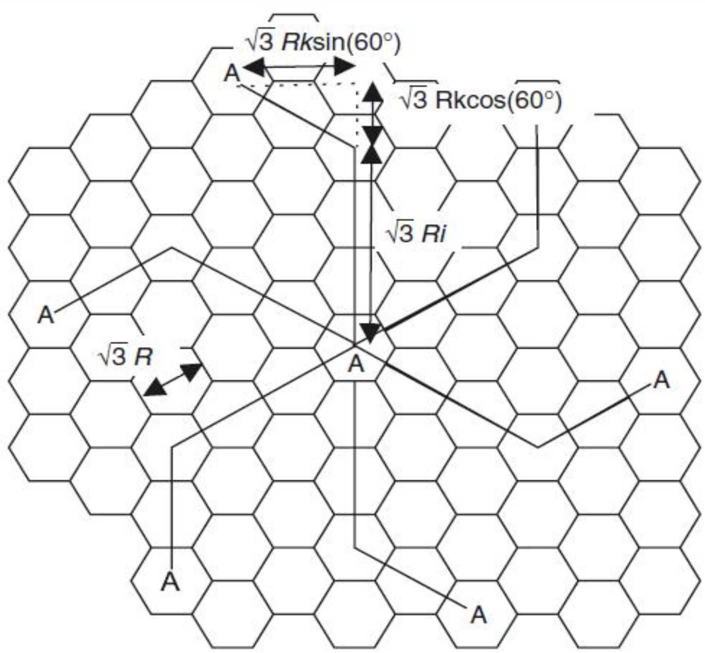
By increasing the ratio (Q),the spatial separation between co-channel cells relative to the coverage distance of a cell is increased.

Part - B

1. Explain the principle of cellular networks and various types of Handoff techniques. (Dec 13)

Cellular Systems:

Most commercial radio and television systems are designed to cover as much area as possible. These systems typically operate at maximum power and with the tallest antennas allowed by the Federal Communications Commission (FCC). The frequency used by the transmitter cannot be reused again until there is enough geographical separation so that one station does not interfere significantly with another station assigned to that frequency. There may even be a large region between two transmitters using the same frequency where neither signal is received. The cellular system takes the opposite approach. It seeks to make an efficient use of available channels by employing low-power transmitters to allow frequency reuse at much smaller distances. Maximizing the number of times each channel may be reused in a given geographic area is the key to an efficient cellular system design. Cellular systems are designed to operate with groups of low-power radios spread out over the geographical service area.



Each group of radios serve mobile stations located near them. The area served by each group of radios is called a cell. Each cell has an appropriate number of low-power radios to communicate within the cell itself. The power transmitted by the cell is chosen to be large enough to communicate with mobile stations located near the edge of the cell. The radius of each cell may be chosen to be perhaps 28 km (about 16 miles) in a start-up system with relatively few subscribers, down to less than 2 km (about 1 mile) for a mature system requiring considerable frequency reuse. As the traffic grows, new cells and channels are added to the system. If an irregular cell pattern is selected, it would lead to an inefficient use of the spectrum due to its inability to reuse frequencies because of cochannel interference. In addition, it would also result in an uneconomical deployment of equipment, requiring relocation from one cell site to another. Therefore, a great deal of engineering effort would be required to readjust the transmission, switching, and control resources every time the system goes through its development phase. The use of a regular cell pattern in a cellular system design eliminates all these difficulties. In reality, cell coverage is an irregularly shaped circle. The exact coverage of the cell depends on the terrain and many other factors. For design purposes and as a first-order approximation, we assume that the coverage areas are regular polygons. For example, for omnidirectional antennas with constant signal power, each cell site coverage area would be circular. To achieve full coverage without dead spots, a series of regular polygons are required for cell sites.

Any regular polygon such as an equilateral triangle, a square, or a hexagon can be used for cell design. The hexagon is used for two reasons: a hexagonal layout requires fewer cells and, therefore, fewer transmitter sites, and a hexagonal cell layout is less expensive compared to square and triangular cells. In practice, after the polygons are drawn on a map of the coverage area, radial lines are drawn and the signal-to-noise ratio (SNR) calculated for various directions using the propagation models, or using appropriate computer programs.we assume regular polygons for coverage areas even though in practice that is only an approximation.

Handoff Strategies

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. This 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. Another feature of newer cellular systems is the ability to make handoff decisions based on a wide range of metrics other than signal strength. The cochannel and adjacent channel interference levels may be measured at the base station or the mobile, and this information may be used with conventional signal strength data to provide a multi-dimensional algorithm for determining when a handoff is needed. The IS-95 code division multiple access (CDMA) spread spectrum cellular system described provides a unique handoff capability that cannot be provided with other wireless systems. Unlike channelized wireless systems that assign different radio channels during a handoff (called a hard handoff), spread spectrum mobiles share the same channel in every cell. Thus, the term handoff does not mean a physical change in the assigned channel, but rather that a different base station handles the radio communication task. By simultaneously evaluating the received signals from a single subscriber at several neighboring base stations, the MSC may actually decide which version of the user's signal is best at any moment in time. This technique exploits macroscopic space diversity provided by the different physical locations of the base stations and allows the MSC to make a "soft" decision as to which version of the user's signal to pass along to the PSTN at any instance EPad94]. The ability to select between the instantaneous received signals from a variety of base stations is called soft handoff. Since CDMA uses cochannel cells, it can use macroscopic spatial diversity to provide soft handoff. Soft handoff is performed by the MSC, which can simultaneously monitor a particular user from two or more base stations. The MSC may choose the best version of the signal at any time without switching frequencies.

Types of Hand – off:

1st generation handoff, MAHO (Mobile assisted handoff), Intersystem handoff, Guard channel concept, Umbrella approach, Soft and hard handoff, Cell dragging

2. What are the differences between TDMA, FDMA and CDMA? Explain in detail about each multiple access techniques. (16) (June 2014)

Frequency Division Multiple Access

The FDMA is the simplest scheme used to provide multiple access. It separates different users by assigning a different carrier frequency. Multiple users are isolated using bandpass filters. In FDMA, signals from various users are assigned different frequencies, just as in an analog system. Frequency guard bands are provided between adjacent signal spectra to minimize crosstalk between adjacent channels. The advantages and disadvantages of FDMA with respect to TDMA or CDMA are:

Advantages

1. Capacity can be increased by reducing the information bit rate and using an efficient digital speech coding scheme.

2. Technological advances required for implementation are simple. A system can be configured so that improvements in terms of a lower bit rate speech coding could be easily incorporated.

3. Hardware simplicity, because multiple users are isolated by employing simple bandpass filters.

Disadvantages

1. The system architecture based on FDMA was implemented in firstgeneration analog systems such as advanced mobile phone system (AMPS) or total access communication system (TACS). The improvement in capacity depends on operation at a reduced signal-to-interference (*S/I*) ratio. But the narrowband digital approach gives only limited advantages in this regard so that modest capacity improvements could be expected from the allocated spectrum.

2. The maximum bit-rate per channel is fixed and small, inhibiting the flexibility in bit-rate capability that may be a requirement for computer fi le transfer in some applications in the future.

3. Inefficient use of spectrum, in FDMA if a channel is not in use, it remains idle and cannot be used to enhance the system capacity.

4. Crosstalk arising from adjacent channel interference is produced by nonlinear effects.

Time Division Multiple Access

In a TDMA system, each user uses the whole channel bandwidth for a fraction of time compared to an FDMA system where a single user occupies the channel bandwidth for the entire duration. In a TDMA system, time is divided into equal time intervals, called *slots*. User data is transmitted in the slots. Several slots make up a frame. Guard times are used between each user's transmission to minimize crosstalk between channels. Each user is assigned a frequency and a time slot to transmit data. The data is transmitted via a radio-carrier from a base station to several active mobiles in the downlink. In the reverse direction (uplink), transmission from mobiles to base stations is time-sequenced and synchronized on a common frequency for TDMA. The preamble carries the address and synchronization information that both the base station and mobile stations use for identification. In a TDMA system, the user can use multiple slots to support a wide range of bit rates by selecting the lowest multiplexing rate or multiple of it. This enables supporting a variety of voice coding techniques at different bit rates with different voice qualities. Similarly, data communications customers could use the same kinds of schemes, choosing and paying for the digital data rate as required. This would allow customers to request and pay for a bandwidth on demand. Depending on the data rate used and the number of slots per frame, a DMA system can use the entire bandwidth of the system or can employ an FDD scheme. The resultant multiplexing is a mixture of frequency division and time division. The entire frequency band is divided into a number of duplex channels (about 350 to 400 kHz). These channels are deployed in a frequency-reuse pattern, in which radio-port frequencies are assigned using an autonomous adaptive frequency assignment algorithm. Each channel is configured in a TDM mode for the downlink direction and a TDMA mode for the uplink direction.

The advantages and disadvantages of TDMA are:

Advantages

1. TDMA permits a flexible bit rate, not only for multiples of the basic single channel rate but also submultiples for low bit rate broadcast-type traffic.

2. TDMA offers the opportunity for frame-by-frame monitoring of signal strength/bit error rates to enable either mobiles or base stations to initiate and execute handoffs.

3. TDMA, when used exclusively and not with FDMA, utilizes bandwidth more efficiently because no frequency guard band is required between channels.

4. TDMA transmits each signal with sufficient guard time between time slots to accommodate time inaccuracies because of clock instability, delay spread, transmission delay because of propagation distance, and the tails of signal pulse because of transient responses.

Disadvantages

1. For mobiles and particularly for hand-sets, TDMA on the uplink demands high peak power in transmit mode that shortens battery life.

2. TDMA requires a substantial amount of signal processing for matched filtering and correlation detection for synchronizing with a time slot.

3. TDMA requires synchronization. If the time slot synchronization is lost, the channels may collide with each other.

4. One complicating feature in a TDMA system is that the propagation time for a signal from a mobile station to a base station varies with its distance to the base station.

Code Division Multiple Access

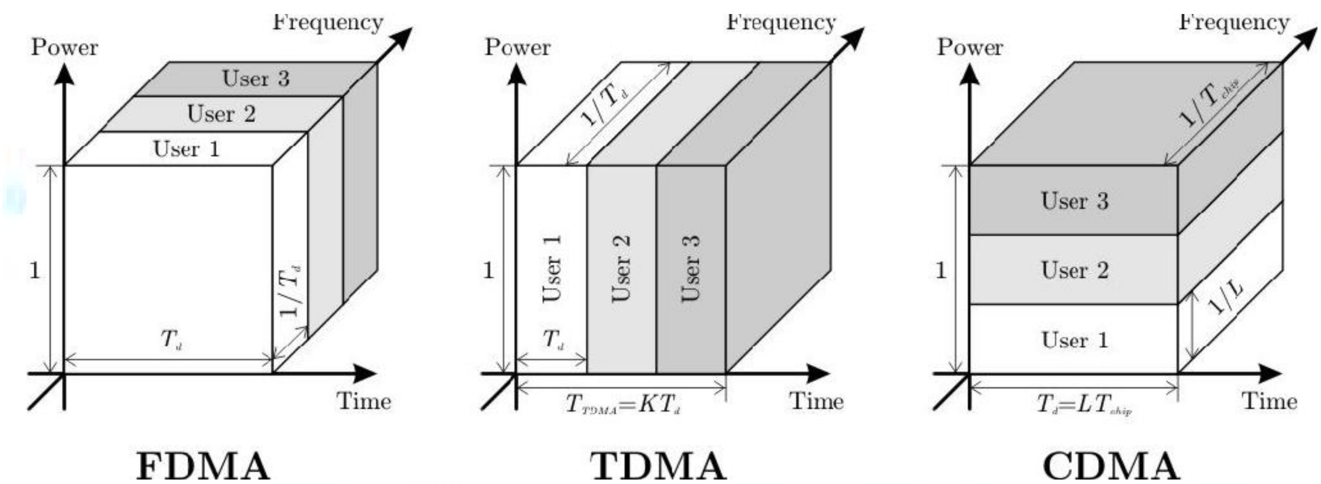
In CDMA, the same bandwidth is occupied by all the users, however they are all assigned separate codes, which differentiates them from each other. CDMA utilize a spread spectrum technique in which a spreading signal (which is uncorrelated to the signal and has a large bandwidth) is used to spread the narrow band message signal. Direct Sequence Spread Spectrum (DS-SS) This is the most commonly used technology for CDMA. In DS-SS, the message signal is multiplied by a Pseudo Random Noise Code. Each user is given his own codeword which is orthogonal to the codes of other users and in order to detect the user, the receiver must know the codeword used by the transmitter. There are, however, two problems in such systems which are discussed in the sequel. CDMA/FDD in IS-95 In this standard, the frequency range is: 869-894 MHz (for Rx) and 824-849 MHz (for Tx). In such a system, there are a total of 20 channels and 798 users per channel. For each channel, the bit rate is 1.2288 Mbps. For orthogonality, it usually combines 64 Walsh-Hadamard codes and a m-sequence.

CDMA and Self-interference Problem

In CDMA, self-interference arises from the presence of delayed replicas of signal due to multipath. The delays cause the spreading sequences of the different users to lose their orthogonality, as by design they are orthogonal only at zero phase offset. Hence in despreading a given user’s waveform, nonzero contributions to that user’s signal arise from the transmissions of the other users in the network. This is distinct from both TDMA and FDMA, wherein for reasonable time or frequency guardbands, respectively, orthogonality of the received signals can be preserved.

CDMA and Near-Far Problem

The near-far problem is a serious one in CDMA. This problem arises from the fact that signals closer to the receiver of interest are received with smaller attenuation than are signals located further away. Therefore the strong signal from the nearby transmitter will mask the weak signal from the remote transmitter. In TDMA and FDMA, this is not a problem since mutual interference can be filtered. In CDMA, however, the near-far effect combined with imperfect orthogonality between codes (e.g. due to different time sifts), leads to substantial interference. Accurate and fast power control appears essential to ensure reliable operation of multiuser DS-CDMA systems.



3. Explain the various methods that increase the system capacity. (May 13)

System capacity is the most important measure for a cellular network. Methods for increasing capacity are thus an essential area of research:

1. Increasing the amount of spectrum used: this is the “brute force” method. It turns out to be very expensive, as spectrum is a scarce resource, and usually auctioned off by governments at very high prices. Furthermore, the total amount of spectrum assigned to wireless systems can change only very slowly; changes in spectrum assignments have to be approved by worldwide regulatory conferences, which often takes ten years or more.

2. More efficient modulation formats and coding: using modulation formats that require

less bandwidth (higher order modulation) and/or are more resistant to interference. The former allows an increase in data rate for each user (or an increase in the number of users in a cell while keeping the data rate per user constant). However, the possible benefits of higher order modulation are limited: they are more sensitive to noise and interference, so that the reuse distance might have to be increased. The use of interference-resistant modulation allows a reduction in reuse distance. The introduction of near-capacity-achieving codes – turbo codes and low-density parity check codes – is another way of achieving better immunity to interference, and thus increases system capacity.

3. Better source coding: depending on required speech quality, current speech coders need data rates between 32 kbit/s and 4 kbit/s. Better models for the properties of speech allow the data rate to be decreased without decreasing quality. Compression of data files and music/video compression also allows more users to be served.

4. Discontinuous Voice Transmission DTX: exploits the fact that during a phone conversation each participant talks only 50% of the time. A TDMA system can thus set up more calls than there are available timeslots. During the call, the users that are actively talking at the moment are multiplexed onto the available timeslots, while quiet users do not get assigned any radio resources.

5. Multiuser detection: this greatly reduces the effect of interference, and thus allows more users per cell for CDMA systems or smaller reuse distances for FDMA systems

6. Adaptive modulation and coding: employs the knowledge at the TX of the transmission channel, and chooses the modulation format and coding rate that are “just right” for the current link situation. This approach makes better use of available power, and, among other effects, reduces interference.

7. Reduction of cell radius: this is an effective, but very expensive, way of increasing capacity, as a new BS has to be built for each additional cell. For FDMA systems, it also means that the frequency planning for a large area has to be redone.Furthermore, smaller cells also require more handovers for moving users, which is complicated, and reduces spectral efficiency due to the large amount of signaling information that has to be sent during a

handover.

8. Use of sector cells: a hexagonal (or similarly shaped) cell can be divided into several (typically three) sectors. Each sector is served by one sector antenna. Thus, the number of cells has tripled, as has the number of BS antennas. However, the number of BS locations has remained the same, because the three antennas are at the same location.

9. Use of an overlay structure: an overlay structure combines cells with different size and different traffic density. Therefore, some locations may be served by several BSs simultaneously. An umbrella cell provides basic coverage for a large area. Within that coverage area, multiple microcells are placed in areas of high traffic density. Within the coverage area of the microcells, most users are served by the microcell BS, but fast-moving users are assigned to the umbrella cell, in order to reduce the number of handovers between cells.

4. What are the features of interference limited systems.

Interference Limited Systems

Noise-Limited Systems

Wireless systems are required to provide a certain minimum transmission quality .This transmission quality in turn requires a minimum *Signal-to-Noise Ratio* (SNR) at the receiver (RX). Consider now a situation where only a single BS transmits, and a Mobile Station (MS) receives; thus, the performance of the system is determined only by the strength of the (useful) signal and the noise. As the MS moves further away from the BS, the received signal power decreases, and at a certain distance, the SNR does not achieve the required threshold for reliable communications. Therefore, the range of the system is noise limited. Depending on the interpretation, it is too much noise or too little signal power that leads to bad link quality.

Let us assume for the moment that the received power decreases with d^2, the square of the distance

distance between BS and MS. More precisely, let the received power

$$P_{RX} = P_{TX} G_{RX} G_{TX} \left(\frac{\lambda}{4\pi d} \right)^2 \qquad (1.1)$$

where G_{RX} and G_{TX} are the gains of the receive and transmit antennas, respectively,1 /λ is the wavelength, and P_{TX} is the transmit power.

The noise that disturbs the signal can consist of several components, as follows:

1. *Thermal noise*: The power spectral density of thermal noise depends on the environmental temperature T_e that the antenna "sees." The temperature of the Earth is around 300 K, while the temperature of the (cold) sky is approximately $T_e \approx 4$K (the temperature in the direction of the Sun is of course much higher). As a first approximation, it is usually assumed that the environmental temperature is isotropically 300 K. Noise power spectral density is then

$$N_0 = k_B T_e \tag{1.2}$$

where k_B is Boltzmann's constant, $k_B = 1.38 \times 10^{-23}$ J/K, and the noise power is

$$P_n = N_0 B \tag{1.3}$$

where B is RX bandwidth (in units of Hz). It is common to write Eq. (1.2) using logarithmic units (power P expressed in units of dBm is $10 \log_{10} (P/1 \text{ mW})$):

$$N_0 = -174 \text{ dBm/Hz} \tag{1.4}$$

This means that the noise power contained in a 1-Hz bandwidth is −174 dBm. The noise power contained in bandwidth B is

$$-174 + 10 \log_{10}(B) \text{ dBm} \tag{1.5}$$

The logarithm of bandwidth B, specifically $10 \log_{10}(B)$, has the units dBHz.

2. *Man-made noise*: We can distinguish two types of man-made noise:

(a) *Spurious emissions*: Many electrical appliances as well as radio transmitters (TXs) designed for other frequency bands have spurious emissions over a large bandwidth that includes the frequency range in which wireless communications systems operate. For urban outdoor environments, car ignitions and other impulse sources are especially significant sources of noise. In contrast to thermal noise, the noise created by impulse sources decreases with frequency (see Figure 1.15). At 150 MHz, it can be 20 dB stronger than thermal noise; at 900 MHz, it is typically 10 dB stronger.

Note that frequency regulators in most countries impose limits on "spurious" or "out-of-band" emissions for *all* electrical devices. Furthermore, for communications operating in licensed bands, such spurious emissions are the only source of man-made noise. It lies in the nature of the license (for which the license holder usually has paid) that no other intentional emitters are allowed to operate in this band. In contrast to thermal noise, man-made noise is not necessarily Gaussian distributed. However, as a matter of convenience, most system-planning tools, as well as theoretical designs, assume *Gaussianity* anyway.

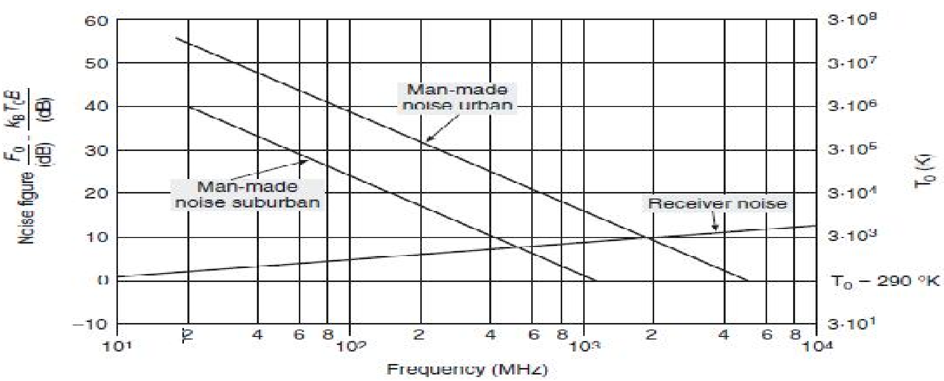


Figure 1.15 Noise as a function of frequency.

(b) *Other intentional emission sources*: Several wireless communications systems operate in unlicensed bands. In these bands, everybody is allowed to operate (emit electromagnetic radiation) as long as certain restrictions with respect to transmit power, etc. are fulfilled. The most important of these bands is the 2.45-GHz Industrial, Scientific, and Medical (ISM) band. The amount of interference in these bands can be considerable.

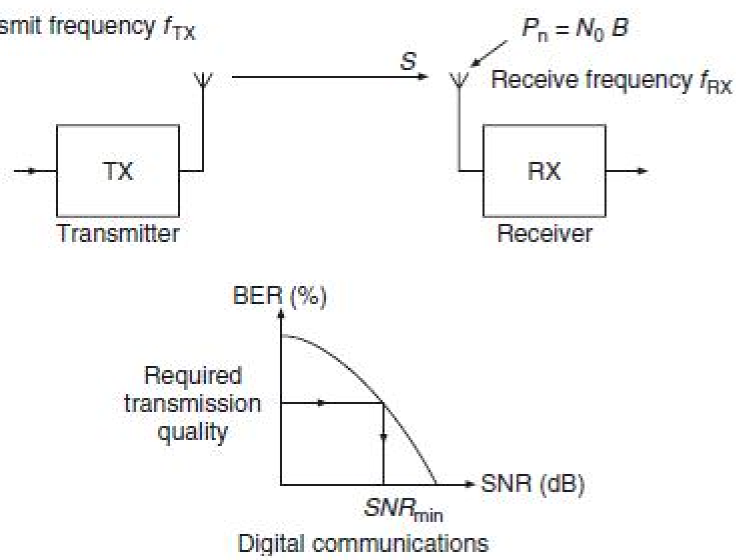
3. *Receiver noise*: The amplifiers and mixers in the RX are noisy, and thus increase the total noise power. This effect is described by the noise figure F, which is defined as the SNR at the RX input (typically after down conversion to baseband) divided by the SNR at the RX output. As the amplifiers have gain, noise added in the later stages does not have as much of an impact

as noise added in the first stage of the RX. Mathematically, the total noise figure F_{eq} of a cascade of components is

$$F_{eq} = F_1 + \frac{F_2 - 1}{G_1} + \frac{F_3 - 1}{G_1 G_2} + \cdots \quad (1.6)$$

where *Fi* and *Gi* are noise figures and noise gains of the individual stages in absolute units (not in decibels (dB)). Note that for this equation, passive components, like attenuators with gain *m* < 1, can be interpreted as *either* having a noise figure of *F* = 1/*m* and unit gain of
G =1,*or* unit noise figure *F* = 1, and gain *G* = *m*.

For a digital system, the transmission quality is often described in terms of the *Bit Error Rate* (BER) probability. Depending on the modulation scheme, coding, and a range of other factors, there is a relationship between SNR and BER for each digital communications systems. A minimum transmission quality can thus be linked to the minimum SNR, *SNR* min, by this mapping (see Figure 1.16). Thus, the planning methods of all analog and digital links in noise-limited environments are the same; the goal is to determine the minimum where all quantities are in dB.



Noise Limited Systems

5.Explain in brief about Trunking and grade of Service.

Trunking and Grade of Service

The concept of trunking allows a large number of users to share the relatively small number of channels in a cell by providing access to each user, on demand, from a pool of available channels. In a trunked radio system, each user is allocated a channel on a per call basis, and upon termination of the call, the previously occupied channel is immediately returned to the pool of available channels.

The telephone company uses trunking theory to determine to determine the number of telephone circuits that need to be allocated for office buildings with hundreds of telephones , and this same principle is used in designing cellular radio systems. In a trunked mobile radio system, when a particular user requests service and all of the radio channels are already in use, the user is blocked, or denied access to the system. To design trunked radio systems that can handle a specific capacity at a specific "grade of service", it is essential to understand trunking theory and queuing theory.

The grade of service (GOS) is a measure of the ability of a user to access a trunked system during the busiest hour. The busy hour is based upon customer demand at the busiest hour during a week, month or a year. The grade of service is a bench mark 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 a system. GOS is typically given as the likelihood that a call is blocked, or the likelihood of a call experiencing a delay greater than a certain queuing time.

The traffic intensity offered by each user is equal to the call request rate multiplied by the call holding time.

$$A_u = \lambda H$$

Where H is the average duration of a call and λ is the average number of call requests per unit time for each user.

For a system containing U users and an unspecified number of channel, the total offered traffic intensity A, is given as

$$A = U A_u$$

Furthermore, in a C channel trunked system, if the traffic is equally distributed among the channels, then the traffic intensity per channel, A_c, is given as

$$A_c = U\, A_u / C$$

There are two types of trunked systems which are commonly used.

(1) Blocked calls cleared
(2) Blocked calls delayed

The blocked calls cleared type offers no queuing for call requests. If no channels are available, the requesting user is blocked without access and is free to try again later. The blocked calls cleared formula or Erlang B formula determines the probability that a call is blocked and is a measure of the GOS for a trunked system which provides no queuing for blocked calls. The Erlang B formula is given by

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

Where C is the number of trunked channels offered by a trunked radio system and A is the total offered traffic.

The blocked calls delayed type is a kind of trunked system in which a queue is provided to hold calls which are blocked. If a channel is not available immediately, the call request may be delayed until a channel becomes available. The Erlang C formula determines the likelihood of a call not having immediate access to a channel. The Erlang C formula is given by

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

UNIT III DIGITAL SIGNALING FOR FADING CHANNELS

PART-A

1. **What are the main features of QPSK? (June 2014)**

   As with DSB -SC modulation, BPSSK requires a transmission bandwidth twice the message bandwidth. Now, Channel bandwidth is a primary resource that should be conserved, particularly in wireless communications.

2. **List the advantages of GMSK. (Dec 2014)**

   The properties of the MSK signal are

   The MSK signal should have a constant envelope and a relatively narrow bandwidth

3. **List the advantages of third generation (3G) networks. (Dec 2014)**

   Another3G system based on CDMA is a direct sequence (OS) spread spectrum system in which the entire bandwidth of the carrier channel is made available to each user simultaneously.

4. **Differentiate Cellular telephony and Cordless telephony.**

   A cordless telephone or portable telephone replaces the handset cord with a radio link. The handset communicates with a base station connected to a fixed telephone line.

5. **Compute the Rayleigh distance of a square Antenna with 20 dB gain.**

The minimum range length to avoid this error is the Rayleigh distance:

A few trial calculations will show that miles of range can be required for large dishes. Fortunately, the Rayleigh distance for the 25 inch dish which I wanted to measure is only 91 feet.

6. **List out any two properties of wideband channel.**

Radio channels are inherently barvl-limited, implying that they may be sampled at the Nyquist rate. This result simplifies the simulation and analysis of complicated wideband channels.

The wideband models treat the propagation channel as frequency -selective; different frequency sub-bands have different channel responses

7. **Draw the mathematical link model for analysis of modulation schemes.**

Modulation is the process of varying one or more properties of a periodic waveform, called the carrier signal, with a modulating signal which typically contains information to be transmitted

8. **Comment on the necessity of a Gaussian filter in GMSK.**

The properties of the MSK signal are The MSK signal should have a constant envelope and a relatively narrow bandwidth

9. **Discuss the principle of OFDM modulation scheme.**

Orthogonal frequency-division multiplexing (OFDM) is a method of encoding digital data on multiple carrier frequencies.

10. **Define cyclic prefix.(Dec 2012)**

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.

Part – B

1. Explain MSK transmitter and receiver with signal space diagram and give an expression for spectral efficiency. (June 13)

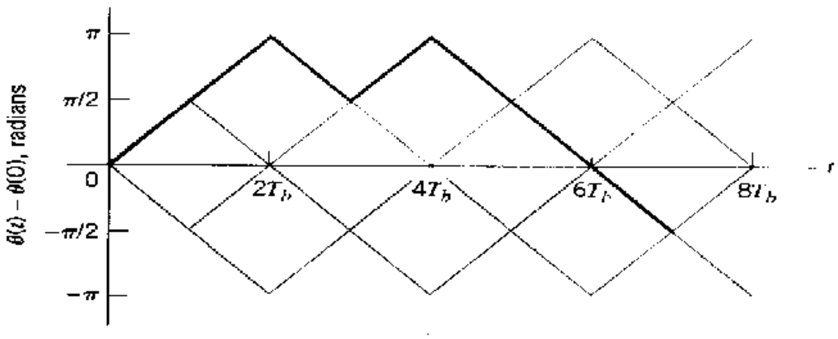
Minimum shift keying (MSK) is a special type of continuous phase.frequency shift keying (CPFSK) wherein the peak frequency deviation is equal to 1/4 the bit rate. In other words, MSK is continuous phase FSK with a modulation index of 0.5 .A modulation index of 0.5 corresponds to the minimum frequency spacing that allows two FSK signals to be coherently orthogonal, and the name minimum shift keying implies the minimum frequency separation (i,e. bandwidth) that allows orthogonal detection.

$$s(t) = \sqrt{\frac{2E_b}{T_b}} \cos[\theta(t)] \cos(2\pi f_c t) - \sqrt{\frac{2E_b}{T_b}} \sin[\theta(t)] \sin(2\pi f_c t)$$

Where

$$\theta(t) = \theta(0) \pm \frac{\pi}{2T_b} t, \qquad 0 \leq t \leq T_b$$

MSK is sometimes referred to as fast FSK, as the frequency spacing used is only half as much as that used in conventional noncoherent FSK.



Phase trellis; boldfaced path represents the sequence 1101000.

It contains orthogonal basis functions Φ1(t) and Φ2(t) they form a pair of modulated carriers.

$$\phi_1(t) = \sqrt{\frac{2}{T_b}} \cos\left(\frac{\pi}{2T_b} t\right) \cos(2\pi f_c t), \qquad 0 \leq t \leq T_b$$

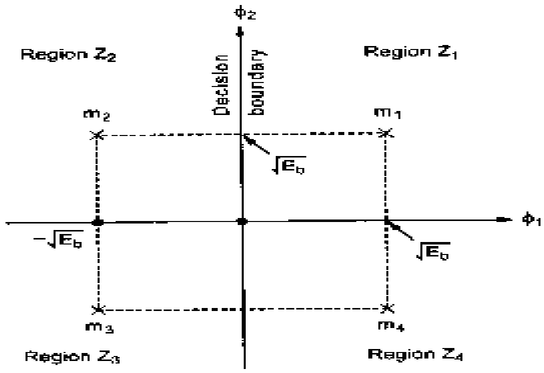
$$\phi_2(t) = \sqrt{\frac{2}{T_b}} \sin\left(\frac{\pi}{2T_b} t\right) \sin(2\pi f_c t), \qquad 0 \leq t \leq T_b$$

We may express the MSK signal in the expanded form

$$s(t) = s_1\phi_1(t) + s_2\phi_2(t), \qquad 0 \leq t \leq T_b$$

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

Constellation diagram



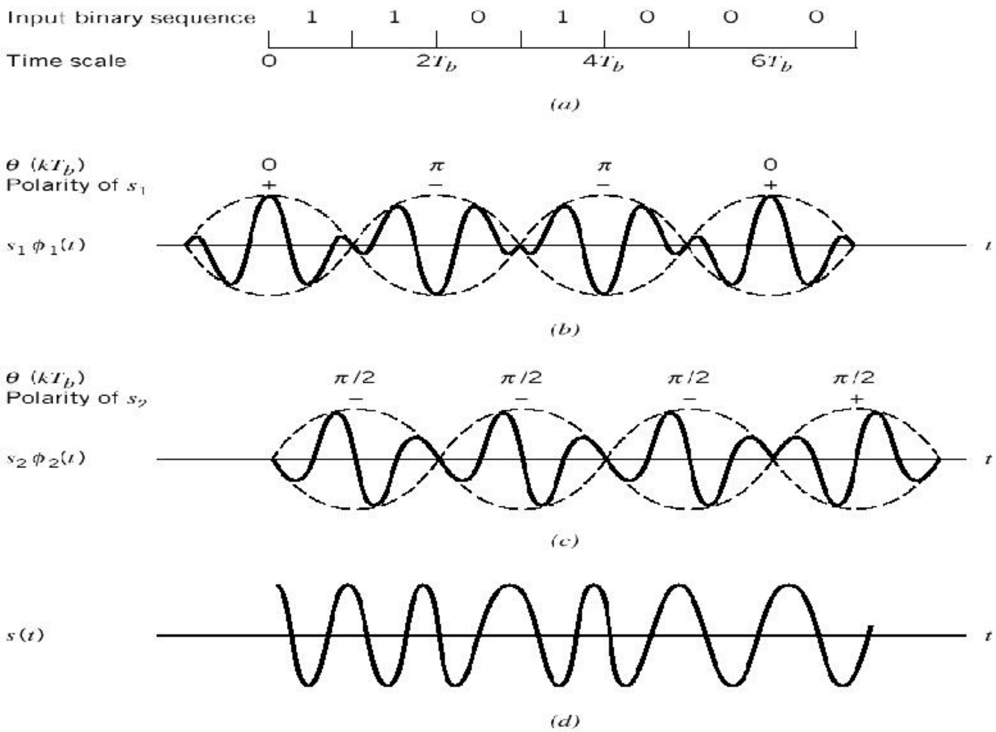
Constellation diagram of an MSK signal is two dimensional with four message points.

N=2

M=4

In MSk,the transmitted symbol is represented by one of two message points at any one time depending on the value of $\theta(0)$

Transmitted symbol $0 \le t \le T_b$	Phase states		Coordinates of message points	
	$\theta(0)$	$\theta(T_b)$	S_1	S_2
0	0	$-\frac{\pi}{2}$	$+\sqrt{E_b}$	$+\sqrt{E_b}$
1	π	$-\frac{\pi}{2}$	$-\sqrt{E_b}$	$+\sqrt{E_b}$
0	π	$+\frac{\pi}{2}$	$-\sqrt{E_b}$	$-\sqrt{E_b}$
1	0	$+\frac{\pi}{2}$	$+\sqrt{E_b}$	$-\sqrt{E_b}$



Error performance of MSK

$$P_e = \frac{1}{2} erfc \sqrt{\frac{E_b}{N_0}} = Q\sqrt{\frac{2\,E_f}{\eta}} = Q\sqrt{\frac{d_{12}^2}{2\,\eta}}$$

Generation of MSK

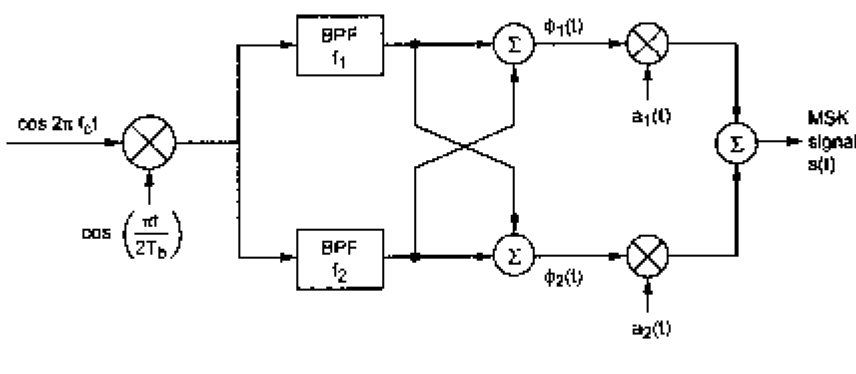


Figure 3.13 Block diagram of MSK modulator

Figure 3.13 shows a typical MSK modulator. Multiplying a carrier signal with cos [**π** t/2T] produces two phase-coherent signals at fc + 1 /4T and –fc + — I /4T. These 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. These carriers are multiplied with the odd and even bit streams, rnI(t) and mQ(t), to produce the MSK modulated signal SMSK(t).

Demodulation of MSK

The block diagram of an MSK receiver is shown in Figure 3.14. The received signal SMSK(t) (in the absence of noise and interference) is multiplied by the respective in-phase and quadrature carriers x(t) and 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 rnI(t) and mQ(t), which are offset combined to obtain the demodulated signal.

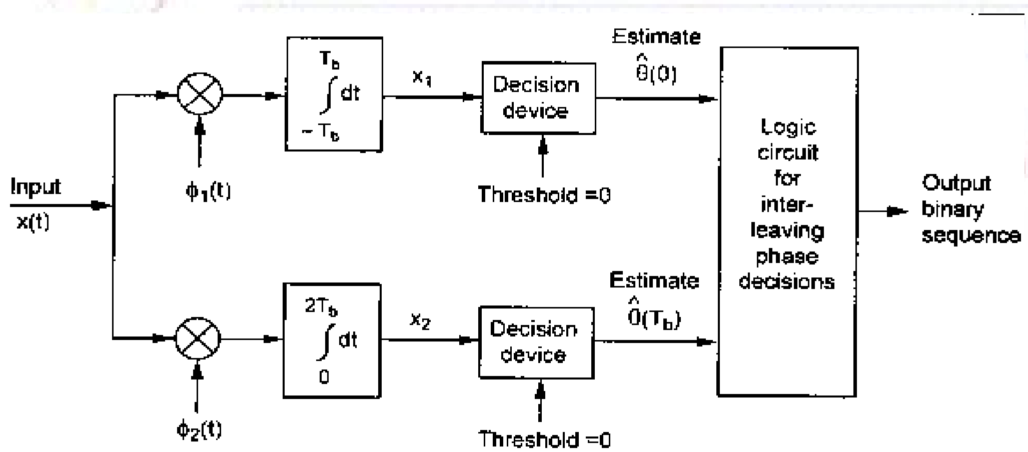


Figure 3.14 block diagram of an MSK receiver

Properties

- It has constant envelope, smoother waveforms are obtained
- Relatively narrow bandwidth
- Coherent detection suitable for satellite communication

- Side lobes are zero outside the frequency band, so it has resistance to co-channel interference

Advantages

1. Smoother waveform than QPSK
2. There is no amplitude variation, constant envelope.
3. Main lobe is wider, contains 99% of signal energy.
4. Less inter channel interference
5. Spectral efficiency is good; BER performance is suitable for mobile radio communication systems.

Disadvantages

1. Complex circuits are needed for generation and detection of MSK signal.
2. Main lobe of MSK is wide
3. Slow decay of MSK power spectral density creates adjacent channel interference.
4. Not suitable for multi user communications
5. Bandwidth required is higher than that of QPSK scheme.

2. Explain with neat diagram QPSK transmission and reception technique and their significance in wireless system. (16) (June 2014)

Quadrature-Phase Shift Keying

A Quadrature-Phase Shift Keying (QPSK)-modulated signal is a PAM where the signal carries bit per symbol interval on both the in-phase and quadrature-phase component. The original data stream is split into two streams, $b1i$ and $b2i$:

$$\left.\begin{aligned} b1_i &= b_{2i} \\ b2_i &= b_{2i+1} \end{aligned}\right\}$$

each of which has a data rate that is half that of the original data stream:

$$R_S = 1/T_S = R_B/2 = 1/(2T_B)$$

Let us first consider the situation where basis pulses are rectangular pulses, g(t) = gR(t, TS). Then we can give an interpretation of QPSK as either a phase modulation or as a PAM. We first define two sequences of pulses

$$\left.\begin{aligned} p1_{\mathrm{D}}(t) &= \sum_{i=-\infty}^{\infty} b1_i g(t - iT_{\mathrm{S}}) = b1_i * g(t) \\ p2_{\mathrm{D}}(t) &= \sum_{i=-\infty}^{\infty} b2_i g(t - iT_{\mathrm{S}}) = b2_i * g(t) \end{aligned}\right\}$$

When interpreting QPSK as a PAM, the bandpass signal reads

$$s_{\mathrm{BP}}(t) = \sqrt{E_{\mathrm{B}}/T_{\mathrm{B}}}[p1_{\mathrm{D}}(t)\cos(2\pi f_{\mathrm{c}}t) - p2_{\mathrm{D}}(t)\sin(2\pi f_{\mathrm{c}}t)]$$

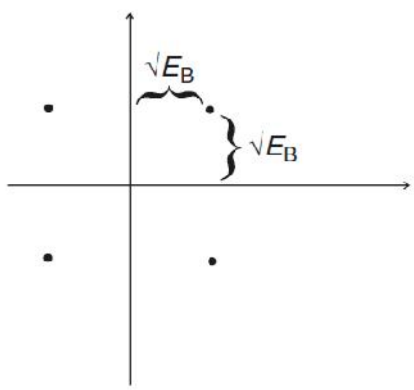
Normalization is done in such a way that the energy within one symbol interval is 2EB, where EB is the energy expended on transmission of a bit. The baseband signal is

$$s_{\mathrm{LP}}(t) = [p1_{\mathrm{D}}(t) + jp2_{\mathrm{D}}(t)]\sqrt{E_{\mathrm{B}}/T_{\mathrm{B}}}$$

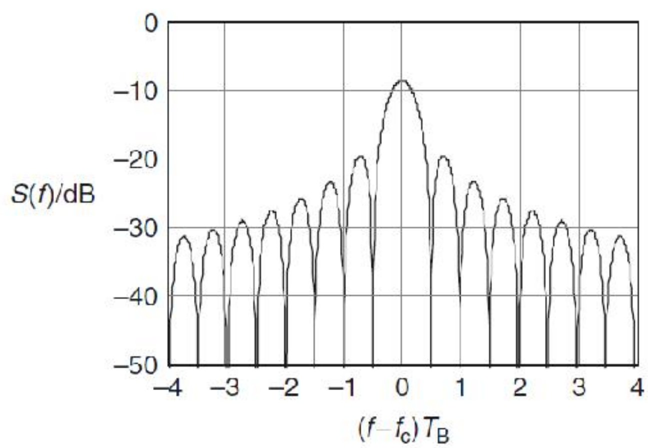
When interpreting QPSK as a phase modulation, the low-pass signal can be written as √2EB/TB exp(jΦS(t)) with:

$$\Phi_{\mathrm{S}}(t) = \pi \cdot \left[\frac{1}{2} \cdot p2_{\mathrm{D}}(t) - \frac{1}{4} \cdot p1_{\mathrm{D}}(t) \cdot p2_{\mathrm{D}}(t)\right]$$

It is obvious from this representation that the signal is constant envelope, except for the transition at t = iTS



Signal space diagram of quadrature-phase shift keying.

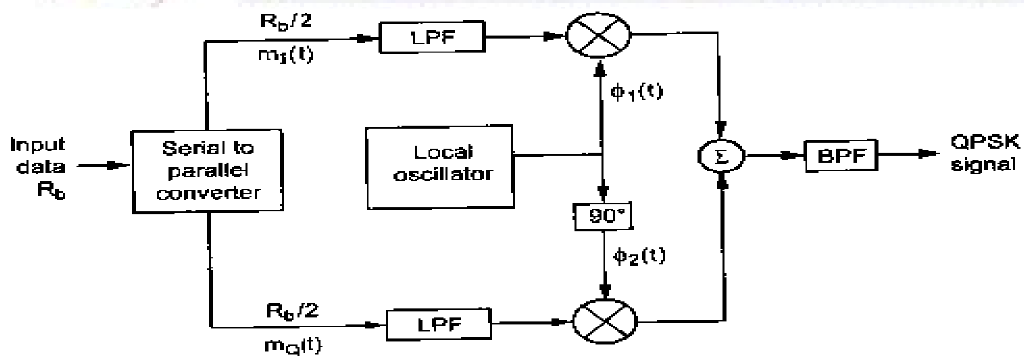


Normalized power-spectral density of quadrature-phase shift keying.

QPSK Transmission

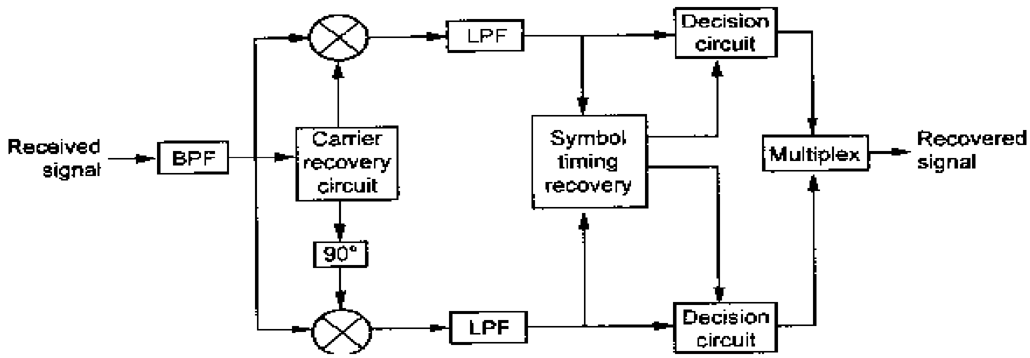
Figure 3.4 shows a block diagram of a typical QPSK transmitter. The unipolar binary message stream has bit rate Rb and is first converted into a bipolar non-return-to-zero (NRZ) sequence using a unipolar to bipolar convener. The bit stream m (t) is then split into two bit streams mI (t) and mQ (t) (in-phase and quadrature streams), each having a bit rate of Rs = Rb/2. The bit stream m1 (t) is called the "even" stream and mQ (t) is called the "odd" stream. The two binary sequences are separately modulated by two carriers Φ1(t) and Φ2(t) which are in quadrature. The two modulated signals, each of which can be considered to be a BPSK signal, are summed to produce a QPSK signal.

The filter at the output of the modulator confines the power spectrum of the QPSK signal within the allocated band. This prevents spill-over of signal energy into adjacent channels and also removes out-of-band spurious signals generated during the modulation process. In most implementations, pulse shaping is done at baseband to provide proper RF filtering at the transmitter output.



Block diagram of a QPSK transmitter.

Figure 3.5 shows a block diagram of a coherent QPSK receiver. The frontend bandpass filter removes the out-of-band noise and adjacent channel interference. The filtered output is split into two parts, and each part is coherently demodulated using the in-phase and quadrature carriers. The coherent carriers used for demodulation are recovered from the received signal using carrier recovery circuits. The outputs of the demodulators are passed through decision circuits which generate the in-phase and quadrature binary streams. The two components are then multiplexed to reproduce the original binary sequence.



Block diagram of a QPSK receiver.

Advantages:

1. Higher data rate.
2. Bandwidth conservation is achieved.

Disadvantages:

1. Signals are amplified using linear amplifiers, which are less efficient
2. Only suitable for rectangular data pulses
3. QPSK phase changes by 90° or 180°. This creates abrupt amplitude variations in the waveform

3. Explain with neat diagram, the principle of Gaussian Minimum shift keying receiver and mention how this is different from MSK. (16) (June 2014)

GMSK (Gaussian MSK) is CPFSK with modulation index *h*mod = 0.5 and *Gaussian* phase basis pulses:

$$\tilde{g}(t) = g_G(t, T_B, B_G T)$$

Thus the sequence of transmit phase pulses is

$$p_D(t) = \sum_{i=-\infty}^{\infty} b_i \tilde{g}(t - iT_B) = b_i * \tilde{g}(t)$$

We see that GMSK achieves better spectral efficiency than MSK because it uses the smoother Gaussian phase basis pulses as opposed to the rectangular ones of MSK

GMSK is a simple binary modulation scheme. The side lobe levels of the spectrum are much reduced by passing the modulating NRZ data waveform through a pre-modulation Gaussian pulse-shaping filter. Gaussian refers to the shape of filter. GMSK has excellent power efficiency and spectral efficiency than conventional FSK.

Properties of Gaussian filter

- Suppress the high frequency components of the transmitted signal
- Avoids excessive deviations in the instantaneous frequency of the FM signal
- Detect coherently or non-coherently the GMSK signal
- Gaussian filter is used before the modulator to reduce the transmitting bandwidth of the signal.

Features

- It has excellent power efficiency due to the constant envelope and it has excellent spectral efficiency.
- The pre-modulation Gaussian filtering introduces ISI in the transmitted signal. But the degradation is not severe if 3dB bandwidth bit duration product of the filter is greater than 0.5.

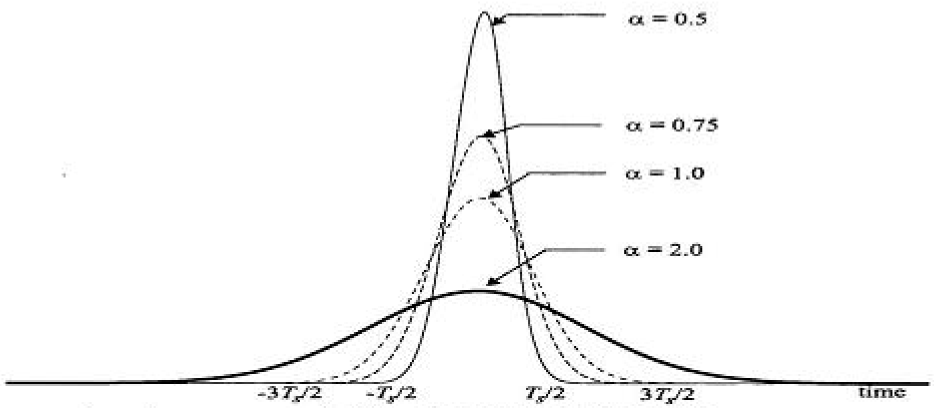
GMSK Bit Error Rate

The bit en-or rate for GMSK was first found in for AWON channels, and was shown to offer performance within 1 dB of optimum MSK when BT=0.25. The bit error probability is a function of BT, since the pulse shaping impacts ISI. The bit error probability for GMSK is given by

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

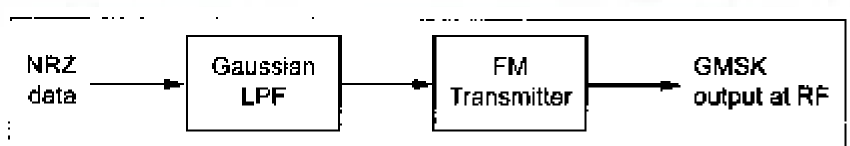
where γ is a constant related to BT by

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



Impulse response of a Gaussian pulse-shaping filter.

GMSK Transmitter

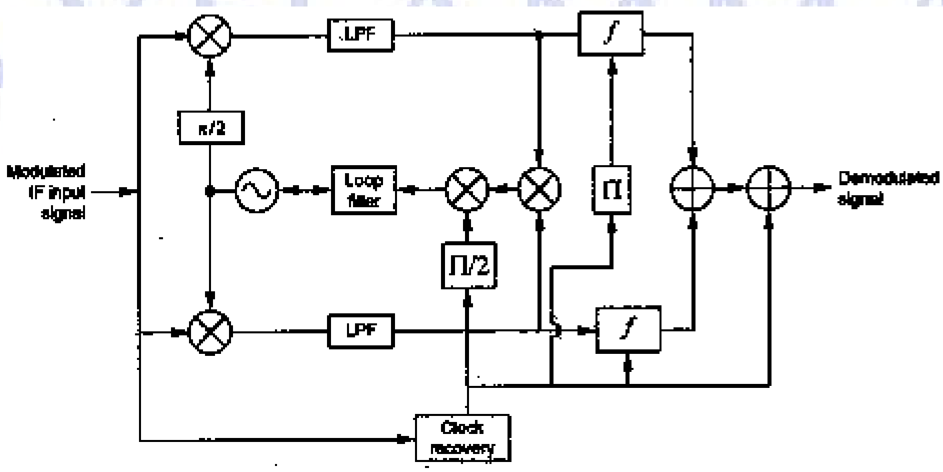


Block diagram of GMSK Transmitter

This modulation technique is shown in Figure 3.15 and is currently used in a variety of analog and digital implementations for the U.S. Cellular Digital Packet Data (CDPD) system as well as for the Global System for Mobile (GSM) system. Figure 3.15 may also be implemented digitally using a standard I/Q modulator.

GMSK Receiver

GMSK signals can be detected using orthogonal coherent detectors as shown in Figure 3.16, or with simple noncoherent detectors such as standard FM discriminators. Carrier recovery is sometimes performed using a method suggested by de Buda where the sum of the two discrete frequency components contained at the output of a frequency doubler is divided by four. DeBuda's method is similar to the Costas loop and is equivalent to that of a PLL with a frequency doubler. This type of receiver can be easily implemented using digital logic as shown in Figure 3.17. The two D flip-flops act as a quadrature product demodulator and the XOR gates act as baseband multipliers. The mutually orthogonal reference carriers are generated using two D flip-flops, and the VCO center frequency is set equal to four times the carrier center frequency. A non-optimum, but highly effective method of detecting GMSK signal is to simply sample the output of an FM demodulator.



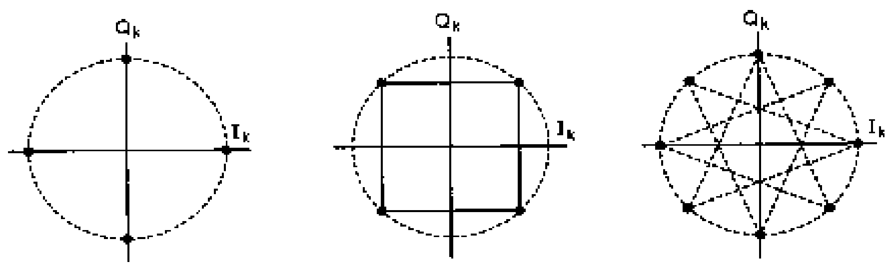
Block diagram of a GMSK receiver

4. **Describe with block diagram π/4 QPSK and its advantages. (Dec 2014)**

The π **/4** shifted QPSK modulation is a quadrature phase shift keying unique which 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. In π **/4** QPSK, the maximum phase change is limited to ± 135°. as compared to 180° for QPSK and 90° for OQPSK. Hence, the band limited π **/4** QPSK signal preserves the constant envelope property better than band limited QPSK, but is more susceptible to envelope variations than OQPSK.

An extremely attractive feature of π **/4** QPSK is that it can be noncoherently detected, which greatly simplifies receiver design. Further, it has been found that in the presence of multipath spread and fading, π **/4** QPSK performs better than OQPSK. When differentially encoded, π **/4** QPSK is called π **/4** DQPSK

Constellation diagram of π /4 QPSK



(a) *Possible states for θ_k when $\theta_{k-1} = n\pi/2$* (b) *Possible states when $\theta_{k-1} = n\pi/4$* (c) *All possible states*

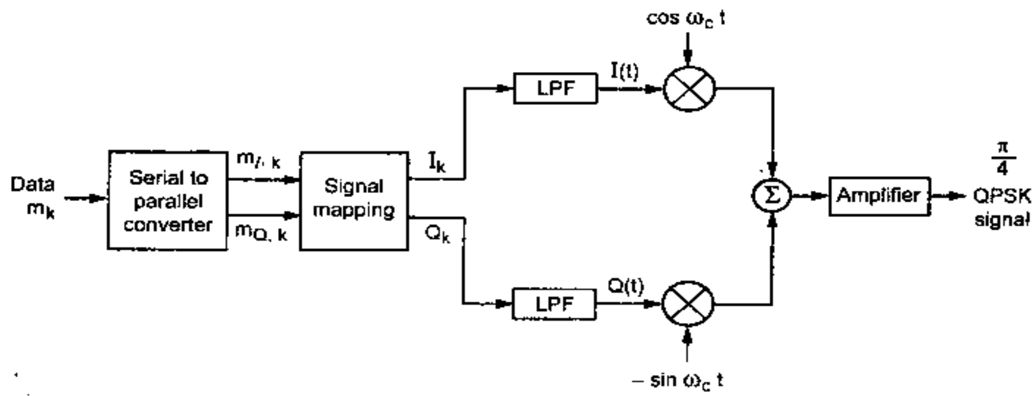
Features of π /4 QPSK

- Class C power efficient amplifiers are used
- Low out of band radiation of the order of -60 dB to -70 dB can be achieved
- Simple limiter, discriminator circuits are used.
- Receiver circuits provide high immunity
- Constant envelope modulation are power efficient.

Disadvantages:

1. They occupy a large bandwidth
2. So, poor bandwidth efficiency.

π /4 QPSK Transmission Techniques



Block diagram of π /4 QPSK Transmission

Just as in a QPSK modulator, the in-phase and quadrature bit streams Ik and Qk are then separately modulated by two carriers which are in quadrature with one another, to produce the **π /4** QPSK Both Ik and Qk are usually passed through raised cosine rolloff pulse shaping filters before modulation, in order to reduce the bandwidth occupancy. Pulse shaping also reduces the spectral restoration problem.

Carrier phase shifts corresponding to various input Bit pairs

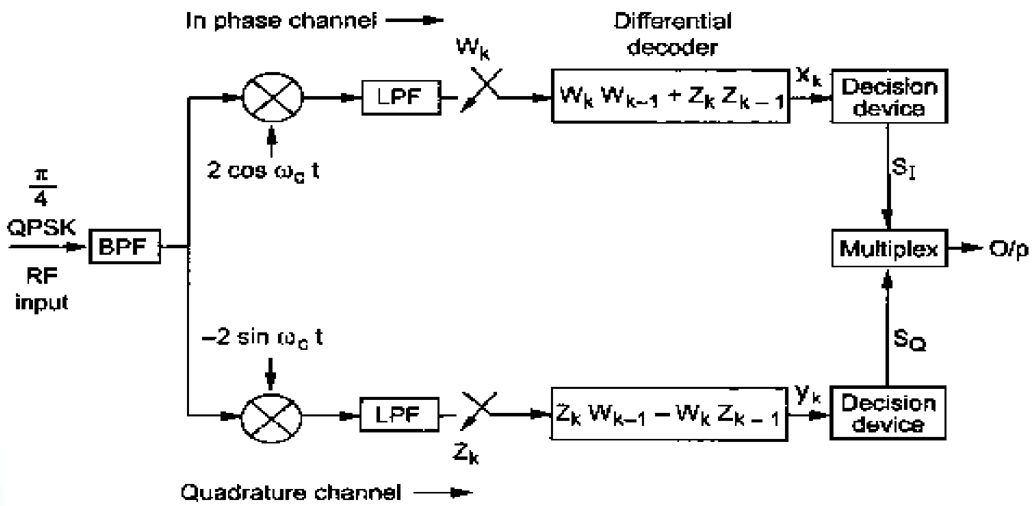
Information bits	Phase shifts
11	π/4
01	3π/4
00	-3π/4
10	-π/4

π /4 QPSK Detection Techniques

Due to ease of hardware implementation, differential detection is often employed to demodulate **π /4** QPSK signals. In an AWGN channel, the BER performance of a differentially detected **π /4** QPSK is about 3 dB inferior to QPSK, while coherently detected **π /4** QPSK has the same error performance as QPSK. There are various types of detection techniques that are used for the detection of **π /4** QPSK signals. They include

1. Baseband differential detection,
2. IF differential detection,and
3. FM discriminator detection.

Baseband Differential Detection



Block diagram of a baseband differential detector

Figure 3.7 shows a block diagram of a baseband differential detector. The incoming **π /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.In decision device SI and SQ are the detected bits in the in-phase and quadrature terms, respectively. These two bits are multiplexed to get an output.

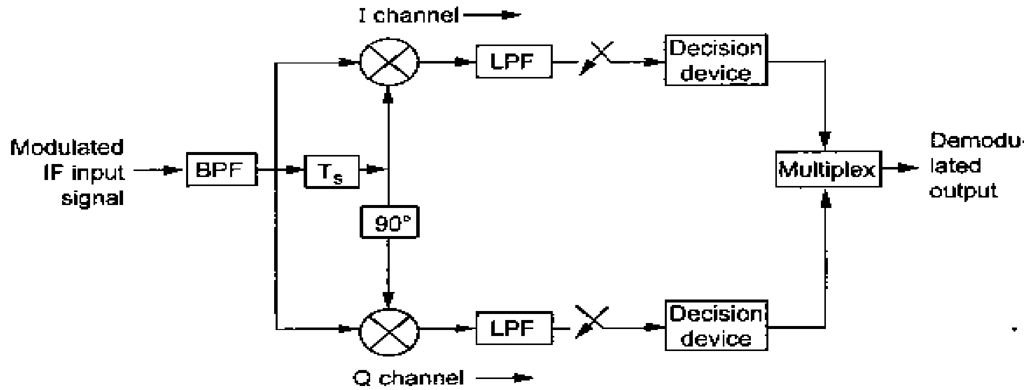
Advantages

1. Easy to implement
2. Simple hardware circuits

Disadvantages

1. Any drift in the carrier frequency will cause a drift in the output phase.
2. It leads to BER degradation

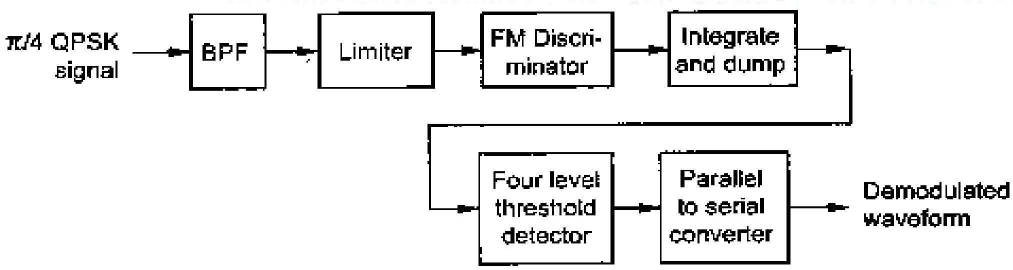
IF Differential Detector



IF differential detector

The IF differential detector shown in Figure 3.8 avoids the need for a local oscillator by using a delay line and two phase detectors. 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. The minimize the effect of IS! and noise, the bandwidth of the filters are chosen to be 0.57/Ts. 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

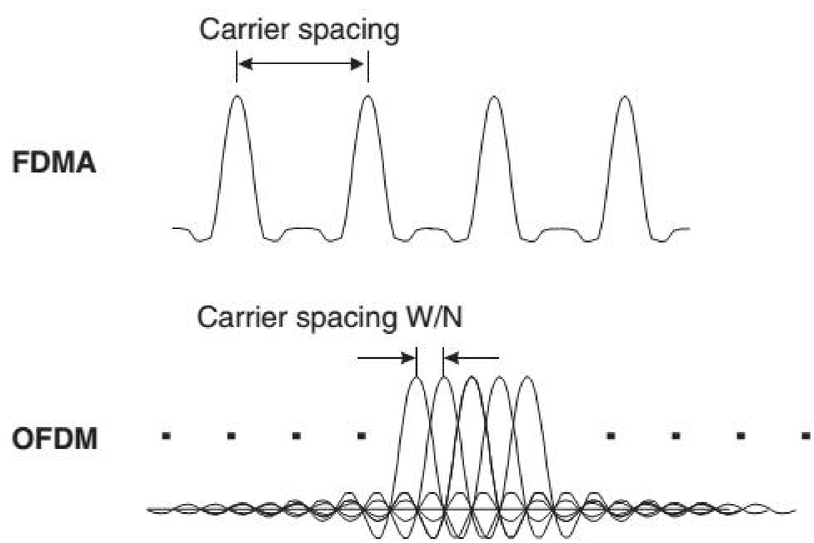


Block diagram of an FM discriminator detector

Figure 3.9 shows a block diagram of an FM discriminator detector for it **π /4** QPSK. 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-2 **π** phase detector. The modulo-2 **π** phase detector improves the BER performance and reduces the effect of click noise.

5. Describe with necessary diagram the operation of Orthogonal Frequency Division Multiplexing Transceiver. (June 2014)

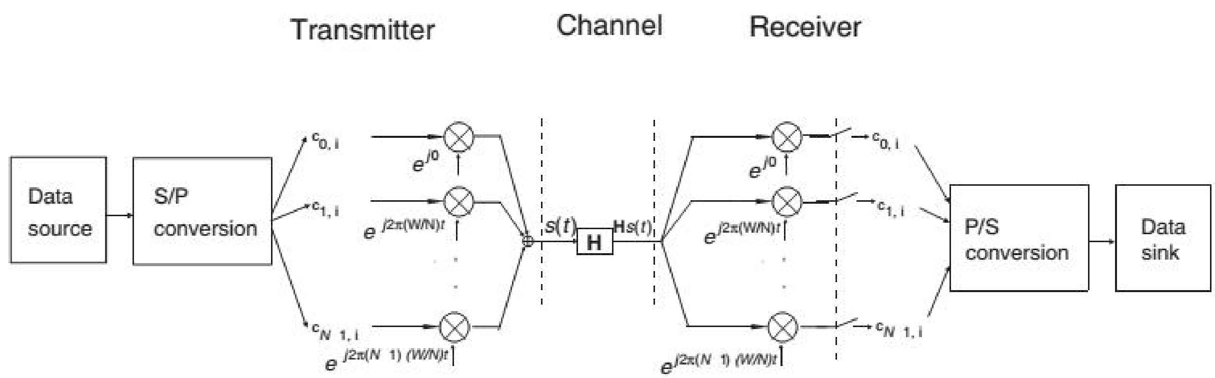
Orthogonal Frequency Division Multiplexing (OFDM) is a modulation scheme that is especially suited for high-data-rate transmission in delay-dispersive environments. It converts a high-rate data stream into a number of low-rate streams that are transmitted over parallel, narrowband channels that can be easily equalized.

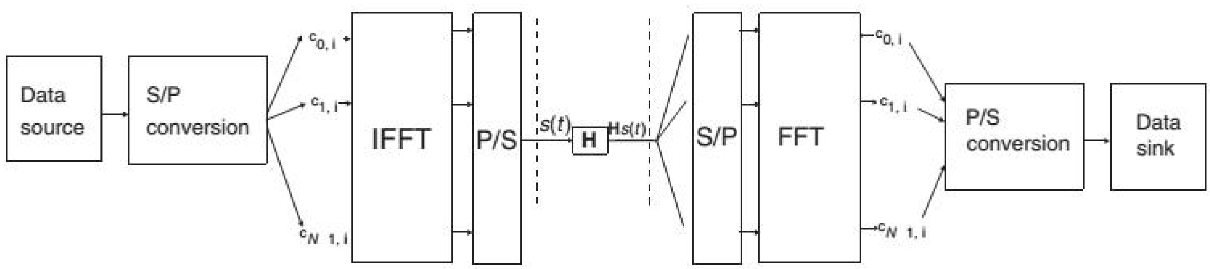


Principle behind orthogonal frequency division multiplexing: N carriers within a bandwidth of W

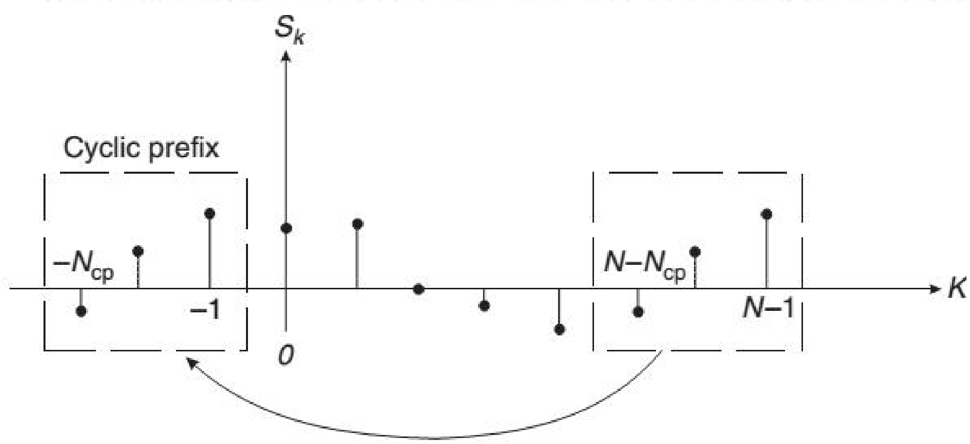
Implementation of Transceivers

Transceiver structures for orthogonal frequency division multiplexing in purely analog technology, and using inverse fast Fourier transformation

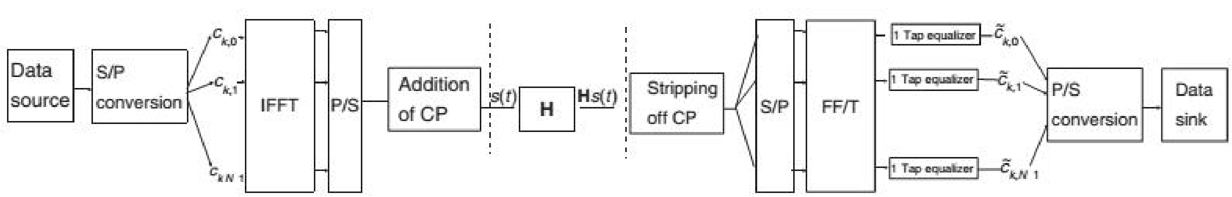




.*Cyclic Prefix*



Principle of the cyclic prefix.*N*cp = *NT*cp*/(N/W)* is the number of samples in the cyclic prefix



Structure of an orthogonal-frequency-division-multiplexing transmission chain with cyclic prefix and one-tap equalization.

UNIT IV MULTIPATH MITIGATION TECHNIQUES

PART-A

1. **State the principle of diversity.(June 2013) (Dec 2013)**

   There are many categories of diversity implementations. Diversity is a way to increase the bit error rate by changing the channel statistics. It assures that the signal to noise ratio is less when increasing the bit error rate.

2. **What is Equalization? (Dec 2013)**

   Generally equalization is a technique which is used to increase the received signal quality and link performance. This compensates for inter symbol interference created by multipath within time dispersive channels.

3. **Compare macro and micro diversity. (Dec 2014)**

There are many categories of diversity implementations. Diversity is a way to increase the bit error rate by changing the channel statistics. It assures that the signal to noise ratio is less when increasing the bit error rate.

4. What are the applications of non linear equalizers? (June 2014)

The fundamental concept of this type of equalizer is that once an information system has been detected and decided upon, the IS! which it induces on future symbols can be estimated and subtracted out before detection of subsequent symbols.

5. List the advantages of digital modulation techniques.

Greater noise immunity, robustness to channel impairments, easier multiplexing of various forms of information, Greater security

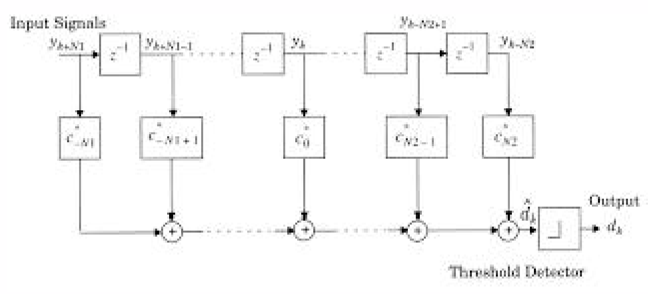
6. State the significance of linear and decision feedback equalizer.

If the analog signal is not used in the feedback path to adapt the equalizer then it is referred as linear equalization.

7. 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 8NR will drop below 10 dB. Compare this. with the case of a single receiver without diversity.

8. Draw the structure of linear transversal equalizer? (Dec 2015

Structure of linear Transversal equalizer:



9. Why nonlinear equalizers are preferred? List out the nonlinear equalization methods. (Dec 2012)

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

Decision feedback equalization (DFE), Maximum likelihood symbol detection and Maximum likelihood sequence estimation (MLSE) are the nonlinear equalization methods used.

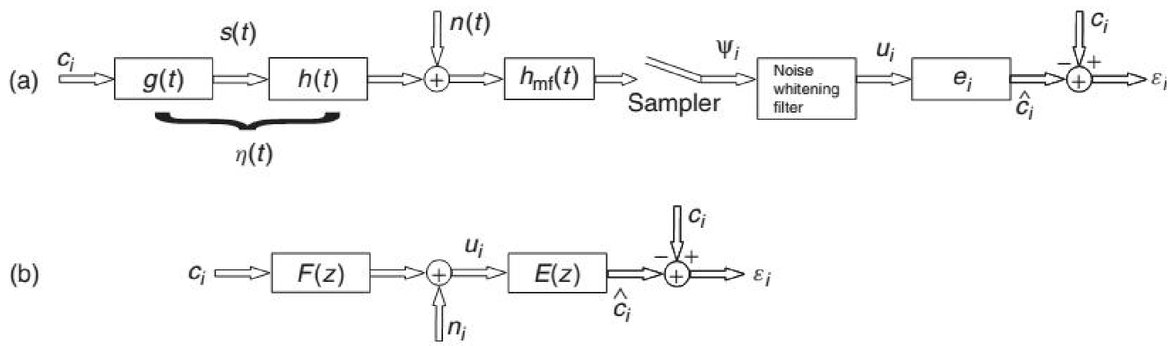
10. What is the need for diversity schemes?(May 2012)

To increase signal to noise ratio, For error free digital transmission, To degrade the bit error probability.

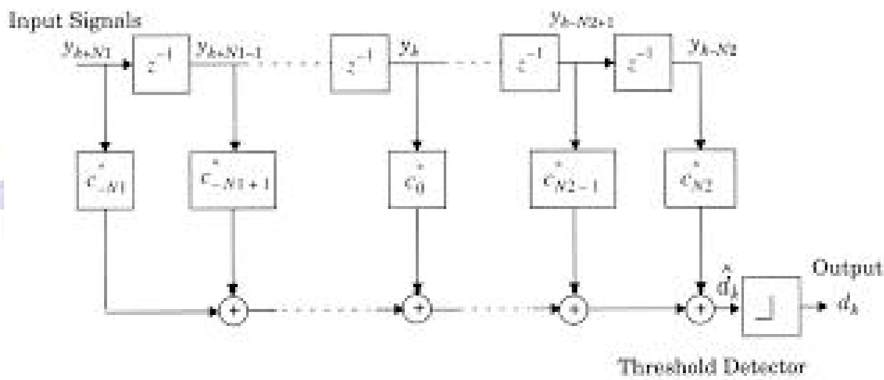
Part –B

1. Briefly explain about linear equalizers. (Dec 13)

Linear equalizer in the time domain (a) and time-discrete equivalent system in the z-transform domain (b)



Structure of a linear transversal filter. Remember that $z-1$ represents a delay by one sample



Zero-Forcing Equalizer

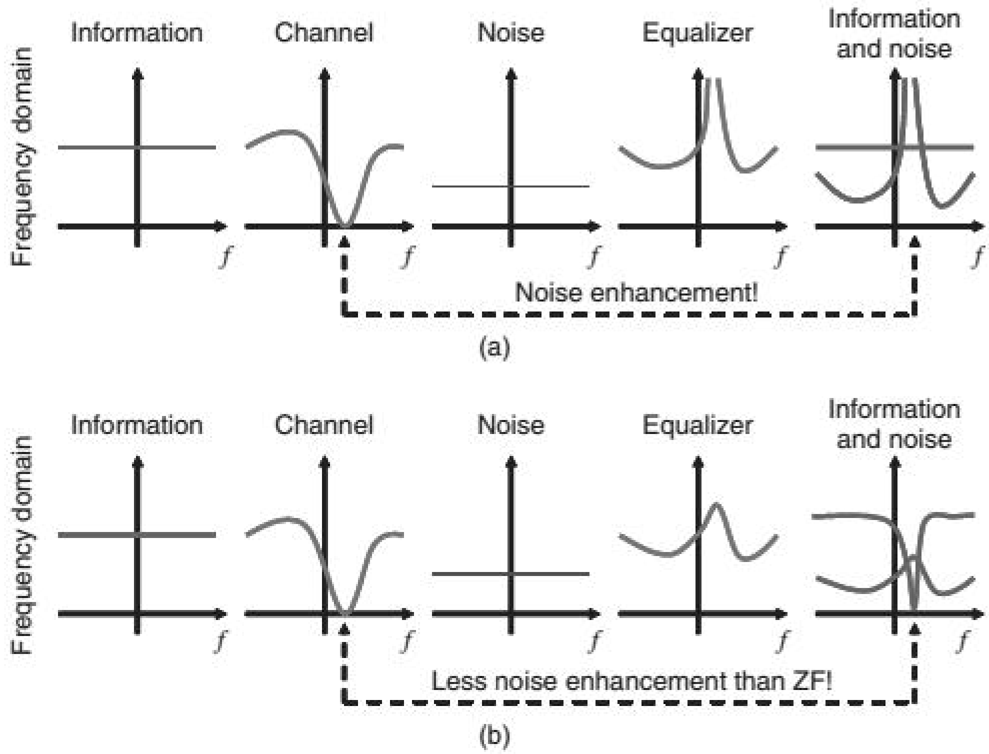


Illustration of noise enhancement in zero-forcing equalizer (a), which is mitigated in an MMSElinear equalizer (b)

The Mean Square Error Equalizer

The ultimate goal of an equalizer is minimization, not of the ISI, but of the bit error probability. Noise enhancement makes the ZF equalizer ill-suited for this purpose. A better criterion is minimization of the Mean Square Error (MSE) between the transmit signal and the output of the equalizer. We are thus searching for a filter that minimizes:

$MSE = E\ |\varepsilon i|2 = E\ \varepsilon i \varepsilon i*$

This can be achieved with a filter whose coefficients **e** opt are given by

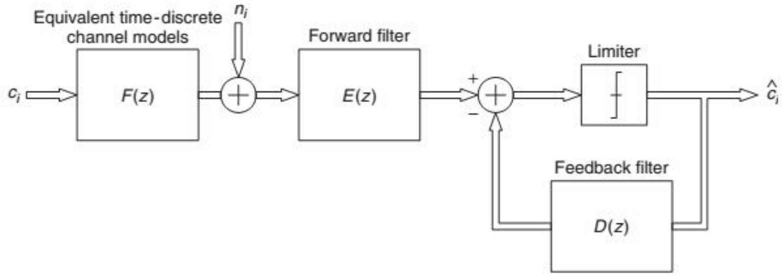
$\mathbf{e}_{\text{opt}} = \mathbf{R}{-1}\mathbf{p}$

where $\mathbf{R} = E\{\mathbf{u}*i\ \mathbf{u}T\ i\}$ is the correlation matrix of the received signal, and $\mathbf{p} = E\{\mathbf{u}*i\ ci\}$ the cross correlation between the received signal and the transmit signal. Considering the frequency domain, concatenation of the noise-whitening filter with the equalizer $E(z)$ has the transfer function: $E(z)^{\sim} = 1/E(z) + N0/\sigma^2_s$ which is the transfer function of the Wiener filter. Comparison shows that the noise power of an MMSE equalizer is smaller than that of a ZF equalizer

2. Explain about DFE and MLSE equalizer. (Dec 13)

A *decision feedback equalizer (DFE)* has a simple underlying premise: once we have detected a bit correctly, we can use this knowledge in conjunction with knowledge of the channel impulse response to compute the ISI caused by this bit. In other words, we determine the effect this bit will have on subsequent samples of the receive signal. The ISI caused by each bit can then be

subtracted from these later samples. The block diagram of a DFE is shown. The DFE consists of a *forward filter* with transfer function $E(z)$, which is a conventional linear equalizer, as well as a *feedbackfilter* with transfer function $D(z)$. As soon as the RX has decided on a received symbol, its impact on all *future* samples (*postcursorISI*) can be computed, and (via the feedback) subtracted from the received signal. A key point is the fact that the ISI is computed based on the signal *after* the hard decision; this eliminates additive noise from the feedback signal. Therefore, a DFE results in a smaller error probability than a linear equalizer. One possible source of problems is *error propagation*. If the RX decides incorrectly for one bit, then the computed postcursor ISI is also erroneous, so that later\ signal samples arriving at the decision device are even more afflicted by ISI than the unequalized samples. This leads to a vicious cycle of wrong decisions and wrong subtraction of postcursors. Error propagation does not usually play a role when the BER is small. Note, however, that small error rates are often achieved via coding. It may therefore be necessary to decode the bits, re-encode them (such that the signal becomes a noise-free version of the received signal), and use this new signal in the feedback from the DFE.



Structure of a decision feedback equalizer.

MMSE Decision Feedback Equalizer

The goal of the MMSE DFE is again minimization of the MSE, by striking a balance between noise enhancement and residual ISI. As noise enhancement is different in the DFE case from that of linear equalizers, the coefficients for the forward filter are different: as postcursor ISI does not contribute to noise enhancement, we now aim to minimize the sum of noise and (average) *precursor*

ISI. The coefficients of the feedforward filter can be computed from the following equation:

$$\sum_{n=-K_{\text{ff}}}^{0} e_n \left(\sum_{m=0}^{-l} f_m^* f_{m+l-n} + N_0 \delta_{nl} \right) = -f_{-l}^* \quad \text{for } l, n = -K_{\text{ff}}, \ldots, 0$$

where$K$ff is the number of taps in the feedforward filter. The coefficients of the feedback filter are then

$$d_n = -\sum_{m=-K_{\text{ff}}}^{0} e_m f_{n-m} \quad \text{for } n = 1, \ldots, K_{\text{fb}}$$

where$K$fb is the number of taps in the feedback filter

Assuming some idealizations (the feedback filter must be at least as long as the postcursor ISI;it must have as many taps as required to fulfill Eq. (16.30); there is no error propagation), the MSE

at the equalizer output is

$$\sigma_n^2(DFE-MMSE) = N_0 \exp\left(\frac{T_S}{2\pi}\int_{-\pi/T_S}^{\pi/T_S} \ln\left[\frac{1}{\Xi(e^{j\omega T})+N_0}\right]d\omega\right)$$

3. Explain diversity techniques to combat small scale fading (June 2013)

Methods that can be used to combat small-scale fading, which are therefore called "microdiversity." The five most common methods are as follows:

Spatial diversity: several antenna elements separated in space.

MS in cellular and cordless systems: it is a standard assumption that waves are incident from all directions at the MS. Thus, points of constructive and destructive interference of Multi Path Components (MPCs) – i.e., points where we have high and low received power, respectively – are spaced approximately $\lambda/4$ apart. This is therefore the distance that is required for decorrelation of received signals. This intuitive insight agrees very well with the results from the exact mathematical derivation (Eq. (13.4), with $f2 - f1 = 0$), decorrelation, defined as $\rho = 0.5$, occurs at an antenna separation of $\lambda/4$. The above considerations imply that the minimum distance for antenna elements in GSM (at 900 MHz) is about 8 cm, and for various cordless and cellular systems at the 1,800-MHz band it is about 4 cm. For Wireless Local Area Networks (WLANs) (at 2.4 and 5 GHz), the distances are even smaller. It is thus clearly possible to place two antennas on an MS of a cellular system.

BS in cordless systems and WLANs: in a first approximation, 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.

BSs in cellular systems: for a cellular BS, the assumption of uniform directions of incidence is no longer valid. Interacting Objects (IOs) are typically concentrated around the MS. Since all waves are incident essentially from one direction, the correlation

Temporal diversity: transmission of the transmit signal at different times.

Temporal diversity can be realized in different ways: 1. *Repetition coding*: this is the simplest form. 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.

2. *Automatic Repeat reQuest(ARQ)*: here, the RX sends a message to the TX to indicate whether it received the data with sufficient quality. If this is not the case, then the transmission is repeated (after a wait period that achieves decorrelation). 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, retransmissions occur always. On the downside, ARQ requires a feedback channel.

3. *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. The transmitted codeword can then be reconstructed.

Frequency diversity: transmission of the signal on different frequencies.

This spreading can be done by different methods: • *Compressing the information in time*: – i.e., sending short bursts that each occupy a large bandwidth – TDMA

• *Code Division Multiple Access* (*CDMA*)

• Multicarrier CDMA and coded orthogonal frequency division multiplexing

• *Frequency hopping in conjunction with coding*: different parts of a codeword are transmitted on different carrier frequencies

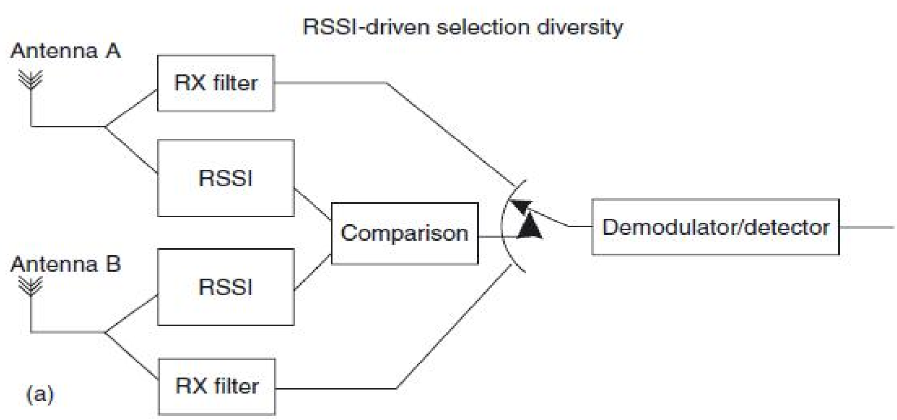
Angular diversity: multiple antennas (with or without spatial separation) with different antenna patterns.

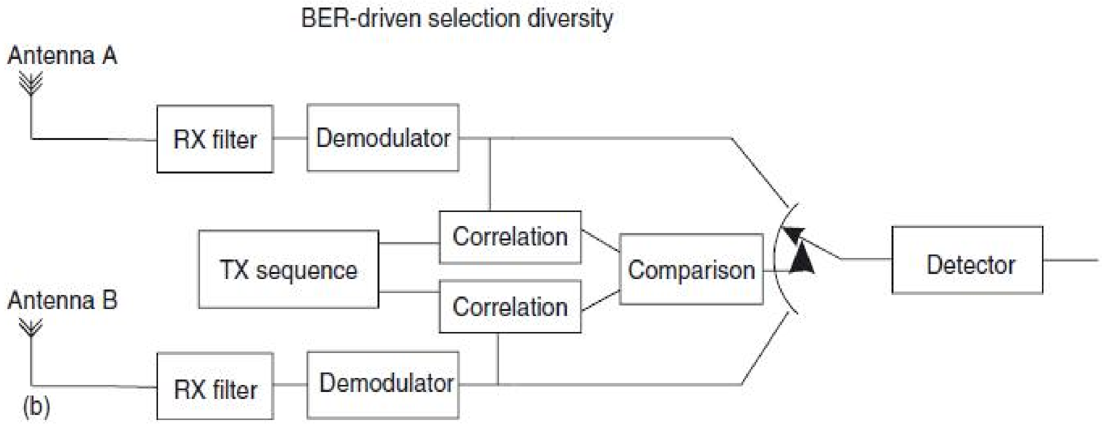
Polarization diversity: multiple antennas with different polarizations (e.g., vertical and horizontal).

Horizontally and vertically polarized MPCs propagate differently in a wireless channel, as the reflection and diffraction processes depend on polarization. Even if the transmit antenna only sends signals with a single polarization, the propagation effects in the channel lead to depolarization so that both polarizations arrive at the RX. The fading of signals with different polarizations is statistically independent. Thus, receiving both polarizations using a dualpolarized antenna, and processing the signals separately, offers diversity. This diversity can be obtained without any requirement for a minimum distance between antenna elements.

4. Explain with diagram the different techniques available for signal combining. (June 2014)

Selection diversity





Switched Diversity

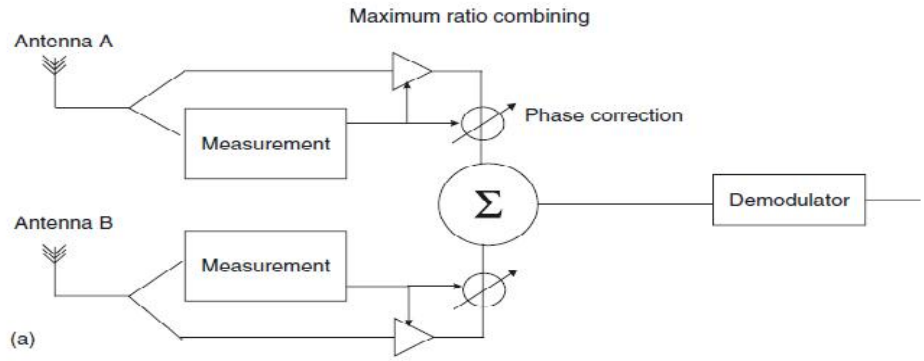
The main drawback of selection diversity is that the selection criteria (power, BER, etc.) of *all* diversity branches have to be monitored in order to know when to select a different antenna. As we have shown above, this leads to either increased hardware effort or reduced spectral efficiency. An alternative solution, which avoids these drawbacks, is *switched diversity*. In this method, the
selection criterion of just the active diversity branch is monitored.

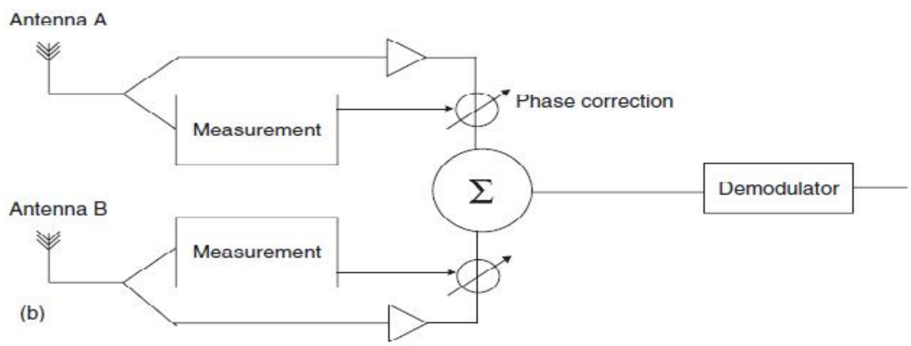
If it falls below a certain threshold, then the RX switches to a different antenna. Switching only depends on the quality of the active diversity branch; it does not matter whether the other branch actually provides a better signal quality or not. Switched diversity runs into problems when both branches 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 a 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: switching threshold and hysteresis time. These parameters have to be selected very carefully: 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. 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.

Summarizing, the performance of switched diversity is worse than that of selection diversity

Combining Diversity





5. Explain with diagram how Rake receiver provides diversity to improve the performance of CDMA receiver. (16) (June 2014)

The Rake receiver is a tapped delay line, whose outputs are weighted and added up. The tap delays, as well as the tap weights, are adjustable, and matched to the channel.

Note that the taps are usually spaced at least one chip duration apart, but there is no requirement for the taps to be spaced at regular intervals. The combination of the receiver filter and the Rake receiver constitutes a filter that is matched to the receive signal.

The receive filter is matched to the transmit signal, while the Rake receiver is matched to the channel. Independent of this interpretation, the receiver adds up the (weighted) signal from the different Rake fingers in a coherent way.

As these signals correspond to different MPCs, their fading is (approximately) statistically independent – in other words, they provide delay diversity (frequency diversity).

A Rake receiver is thus a diversity receiver, and all mathematical methods for the treatment of diversity remain valid.

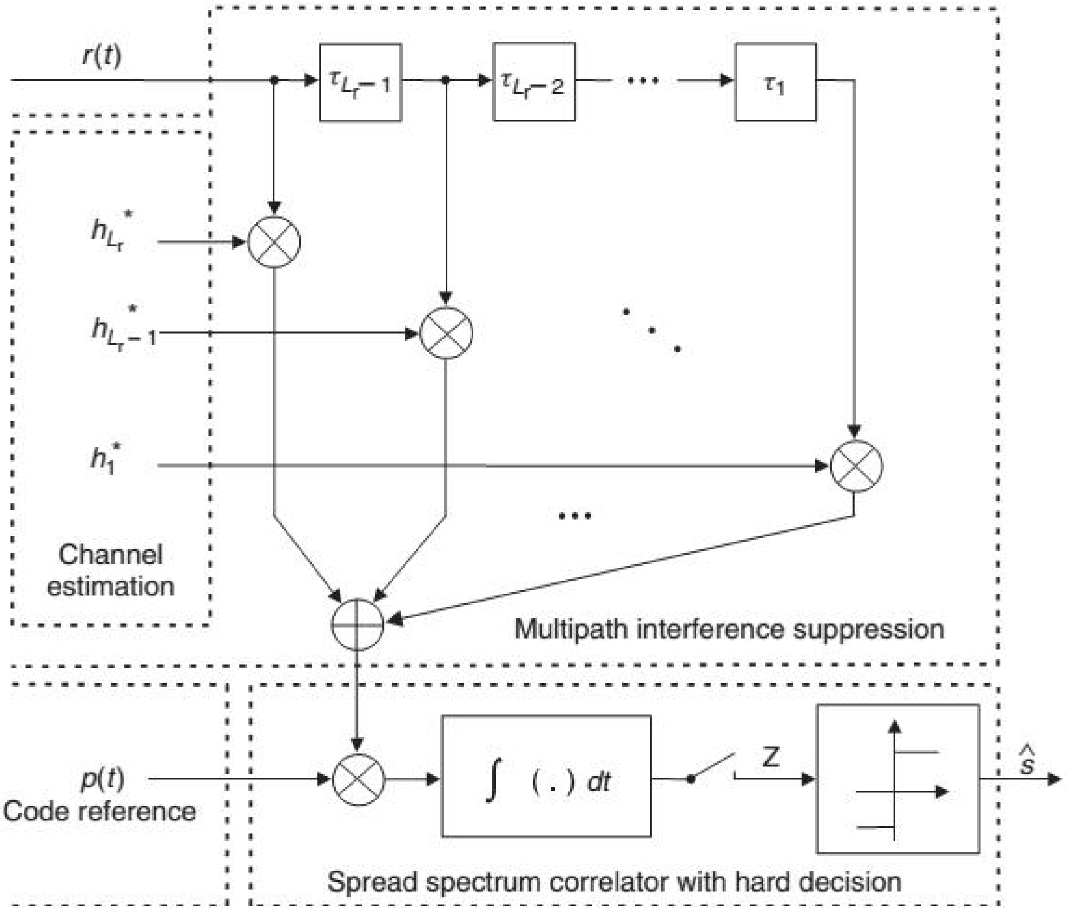


Diagram of Rake receiver.

A Rake receiver is thus a diversity receiver, can be implemented in a practical Rake combiner is limited by power consumption, design complexity, and channel estimation. A Rake receiver that processes only a subset of the available L_r resolved MPCs achieves lower complexity, while still providing a performance that is better than that of a single-path receiver. The Selective Rake (SRake) receiver selects the L_b best paths (a subset of the L_r available resolved MPCs) and then combines the selected subset using maximum-ratio combining.

Another possibility is the Partial Rake (PRake), which uses the first Lf MPCs. Although the performance it provides is not as good, it only needs to estimate L_f MPCs. Another generally important problem for Rake receivers is interpath interference. An alternative to this combination of Rake receiver and symbol-spaced equalizer is the chip-based equalizer, where an equalizer works directly on the output of the despreader sampled at the chip rate. This method is optimum, but very complex.

The basic nature of a CDMA system is to spread the signal over a large bandwidth; thus, it can be anticipated that the transfer function of the channel exhibits variations over this bandwidth. If the channel is slowly time variant, the effective impulse response can be written as

$$h_{\text{eff}}(t_i, \tau) = \tilde{\mathrm{p}}(\tau) * h(t_i, \tau) \tag{5.1}$$

where the effective system impulse response $\tilde{p}(\tau)$ is the convolution of the transmit and receive
spreading sequence:

$$\tilde{\mathrm{p}}(\tau) = p_{\text{TX}}(\tau) * p_{\text{RX}}(\tau) = ACF(\tau) \tag{5.2}$$

UNIT V MULTIPLE ANTENNA TECHNIQUES

PART-A

1. **List the different types of channel coding techniques. (Dec 2014)**

   Repetition coding, Automatic repeat request.

2. **What is frequency hopped multiple access?(June 2014)**

   In this type of method, at the same time, all the people can send the signals. These users are differ in the frequency of operation.

3. **Write about MMSE decision feedback equalizer. (May 2015)**

   The fundamental concept of this type of equalizer is that once an information system has been detected and decided upon, the ISI which it induces on future symbols can be estimated and subtracted out before detection of subsequent symbols.

4. **Characterize the effects of multipath propagation on code division multiple access.(May 2015)**

   Reflection - occurs when signal encounters a surface that is large relative to the wavelength of the signal. Diffraction - occurs at the edge of an impenetrable body that is large compared to
   wavelength of radio wave.

5. **What do you mean by transmit diversity? (May 2015)**

   There are many transmit antennas in the transmit diversity. The transmit power is divided among these antennas. The systems need transmit diversity when it holds more.

6. **Define coding gain. (MAY/JUNE 2016)**

   Coding is the embedding of signal constellation points in a higher dimensional signaling space. The theory of coding says that it holds for temporally constant and temporally varying channels.

7. **Define Spatial multiplexing.**

   Capacity gain at no additional power or bandwidth consumption obtained through the use of multiple antennas at both sides of a wireless radio link.

8. **Define Diversity gain.**

Improvement in link reliability obtained by transmitting the same data on independently fading branches.

9. **Define Transmit diversity. (Dec 2015)**

Transmit diversity is radio communication using signals that originate from two or more independent sources that have been modulated with identical information-bearing signals and that may vary in their **transmission** characteristics at any given instant.

10. **Define Receiver diversity.**

Receive diversity improves the bit error rate (BER) performance. In this post, let us try to understand the BER improvement with **receive diversity.**

11. **What is antenna diversity(space diversity). ? (Dec 2015)**

Conventional wireless systems consist of an elevated base station antenna and a mobile antenna close to the ground .The existence of direct path between the transmitter & receiver is not guaranteed and the possibility of a number of scatterers is large from this JAKES deduced that the signal received from spatially separated antennas on the mobile would have essentially uncorrelated envelopes for antenna separations of one half wavelength or more.

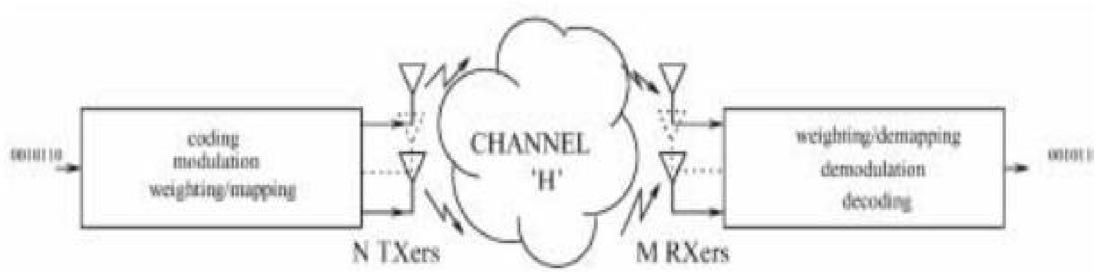
PART-B

1. Discuss about the MIMO systems in detail.

• A MIMO system consists of several antenna elements, plus adaptive signal processing, at both transmitter and receiver

• First introduced at Stanford University (1994) and Lucent (1996)

• Exploit multipath instead of mitigating it

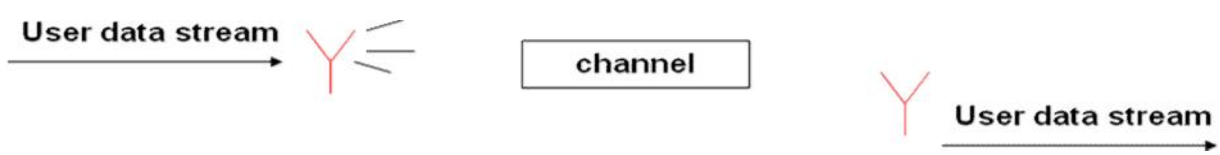


<u>MIMO:</u>

High data rate wireless communications links with transmission rates nearing 1 Gigabit/second (will quantify a "bit" shortly)

Provide high speed links that still offer good Quality of Service (QoS) (will be quantified mathematically)

- Single-Input-Single-Output (SISO) antenna system:

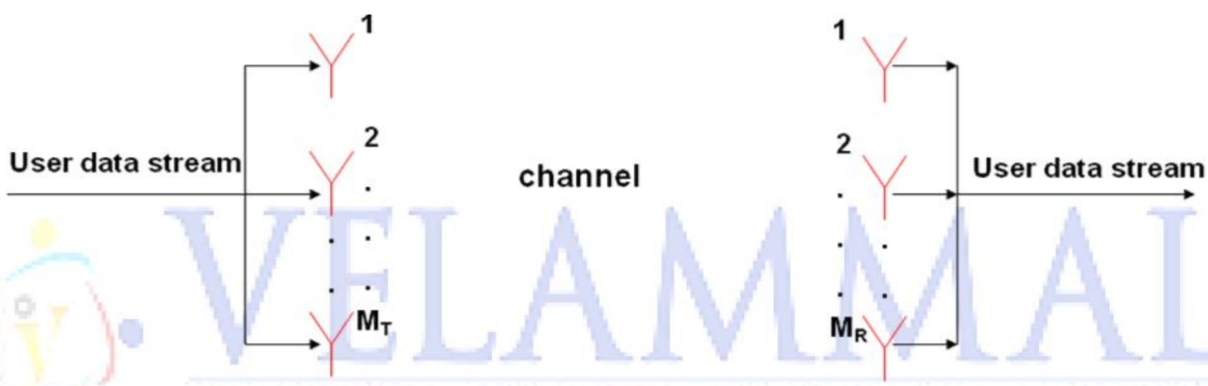


- Theoretically, the 1Gbps barrier can be achieved using this configuration if you are allowed to use much power and as much BW as you so please!

- Extensive research has been done on SISO under power and BW constraints. A combination a smart *modulation*, *coding* and *multiplexing* techniques have yielded good results but far from the 1Gbps barrier.

<u>MIMO Antenna Configuration:</u>

Use multiple transmit and multiple receive antennas for a single user



Now this system promises enormous data rates!

MIMO Design Criterion:

MIMO Systems can provide two types of gain

Spatial Multiplexing Gain:

- Maximize transmission rate   (optimistic approach)
- Use rich scattering/fading to   your advantage

Diversity Gain:

- Minimize P_e (conservative approach)
- Go for Reliability / QoS etc
- Combat fading

Diversity:

- Each pair of transmit-receive antennas provides a signal path from transmitter to receiver. By sending the SAME information through different paths, multiple independently-faded replicas of the data symbol can be obtained at the receiver end. Hence, more reliable reception is achieved

- A diversity gain d implies that in the high SNR region, my P_e decays at a rate of $1/SNR^d$ as opposed to 1/SNR for a SISO system
- The maximal diversity gain d_{max} is the total number of independent signal paths that exist between the transmitter and receiver
- For an (M_R,M_T) system, the total number of signal paths is M_RM_T
- $1 \leq d \leq d_{max} = M_RM_T$
- The higher my diversity gain, the lower my P_e

MIMO Benefits :

• higher capacity (bits/s/Hz)

(spectrum is expensive; number of base stations limited)

• better transmission quality (BER, outage)

• Increased coverage

• Improved user position estimation

2. Explain in detail how inherent delay in a multiuser system is overcome by beamforming.(MAY/JUNE 2016)

- Beamforming or spatial filtering is a signal processing technique used in sensor arrays for directional signal transmission or reception.
- This is achieved by combining elements in a phased array in such a way that signals at particular angles experience constructive interference while others experience destructive interference.
- Beamforming can be used at both the transmitting and receiving ends in order to achieve spatial selectivity. The improvement compared with omni directional reception/transmission is known as the receive/transmit gain (or loss).

- Beamforming can be used for radio or sound waves. It has found numerous applications in radar, sonar, seismology, wireless communications, radio astronomy, acoustics, and biomedicine.
- Adaptive beamforming is used to detect and estimate the signal-of-interest at the output of a sensor array by means of optimal (e.g., least-squares) spatial filtering and interference rejection.

    - Desired signal maximization mode
    - Interference signal minimization or cancellation mode

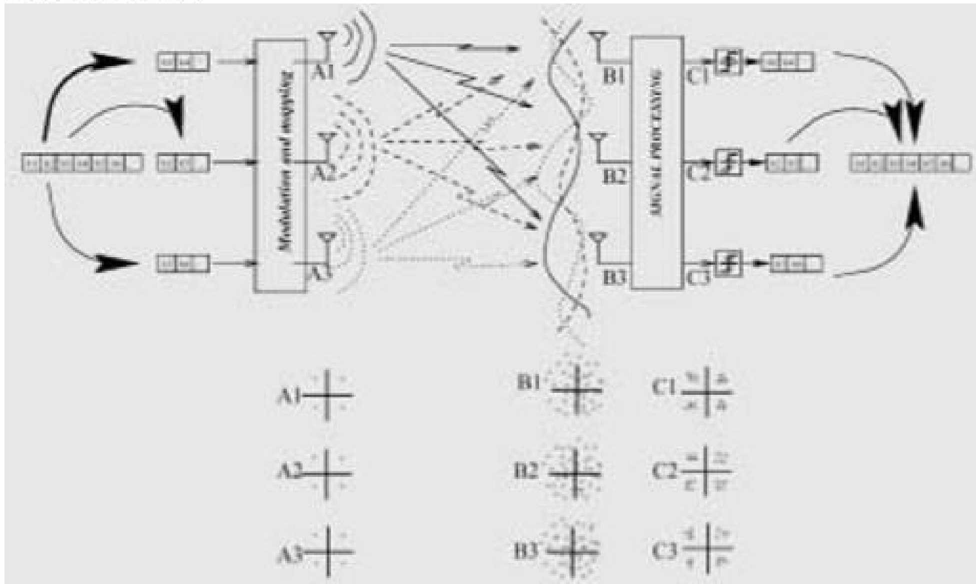
Conventional beamformers use a fixed set of weightings and time-delays (or phasings) to combine the signals from the sensors in the array, primarily using only information about the location of the sensors in space and the wave directions of interest. In contrast, adaptive beamforming techniques generally combine this information with properties of the signals actually received by the array, typically to improve rejection of unwanted signals from other directions. This process may be carried out in either the time or the frequency domain.

As the name indicates, an adaptive beamformer is able to automatically adapt its response to different situations. Some criterion has to be set up to allow the adaption to proceed such as minimizing the total noise output. Because of the variation of noise with frequency, in wide band systems it may be desirable to carry out the process in the frequency domain.

Beamforming can be computationally intensive. Sonar phased array has a data rate low enough that it can be processed in real-time in software, which is flexible enough to transmit and/or receive in several directions at once. In contrast, radar phased array has a data rate so high that it usually requires dedicated hardware processing, which is hard-wired to transmit and/or receive in only one direction at a time. However, newer field programmable gate arrays are fast enough to handle radar data in real-time, and can be quickly re-programmed like software, blurring the hardware/software distinction.

3. Discuss about Spatial multiplexing of a MIMO system in detail.

We send multiple signals, the receiver learns the channel matrix and inverts it to separate the data.



- MIMO spatial multiplexing in Line-of-sight
- The system is near rank one (non invertible)
- Spatial multiplexing requires multipath to work

Spatial Multiplexing:

$y = Hs + n \rightarrow y' = Ds' + n'$ (through SVD on H)

where D is a diagonal matrix that contains the eigenvalues of HH^H Viewing the MIMO received vector in a different but equivalent way,

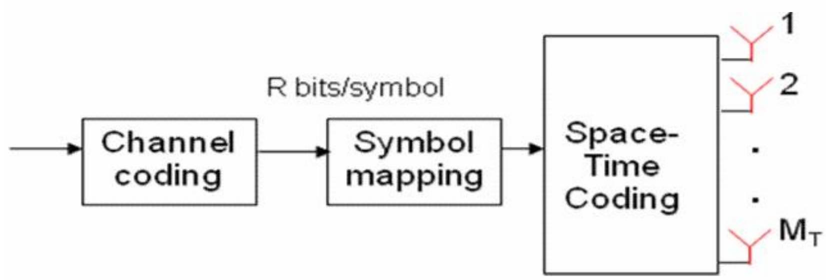
$C_{EP} = \log_2 [I_M + (P/M_T)DD^H] = \quad \log_2 [1 + (P/M_T)\lambda_i]$ b/s/Hz

Equivalent form tells us that an (M_T,M_R) MIMO channel opens up

$m = \min (M_T,M_R)$ independent SISO channels between the transmitter and the receiver

So, intuitively, I can send a maximum of m different information symbols over the channel at any given time.

Practical System:



r_s : number of different symbols N transmitted in T symbol periods

$r_s = N/T$

Spectral efficiency = ($R*r_c$ info bits/symbol)(r_s)(R_s symbols/sec)/w

= $R r_c r_s$ bits/s/Hz assuming $R_s = w$

r_s is the parameter that we are concerned about: $0 \le r_s \le M_T$

** If $r_s = M_T$, we are in spatial multiplexing mode (max transmission rate)

**If $r_s \le 1$, we are in diversity mode

- MIMO Systems are getting us closer to the 1Gbps landmark (aspiration 1)
- At the same time, they provide reliable communications (aspiration 2)

4. Discuss about transmitter diversity, receiver diversity in detail.

By combining signals from different at the RX, the total quality of the signal is improved. Signals selected from the multiple diversity branches by

1. Selection diversity, where the "best" signal copy is selected and processed (demodulated and decoded), while all other copies are discarded. There are different criteria for what constitutes the "best" signal.

2. Combining diversity, where all copies of the signal are combined (before or after the demodulator), and the combined signal is decoded. Again, there are different algorithms for combination of the signals.

Combining diversity leads to better performance, as all available information is exploited.

We also have to keep in mind that the gain of multiple antennas is due to two effects:

- diversity gain
- beamforming gain.

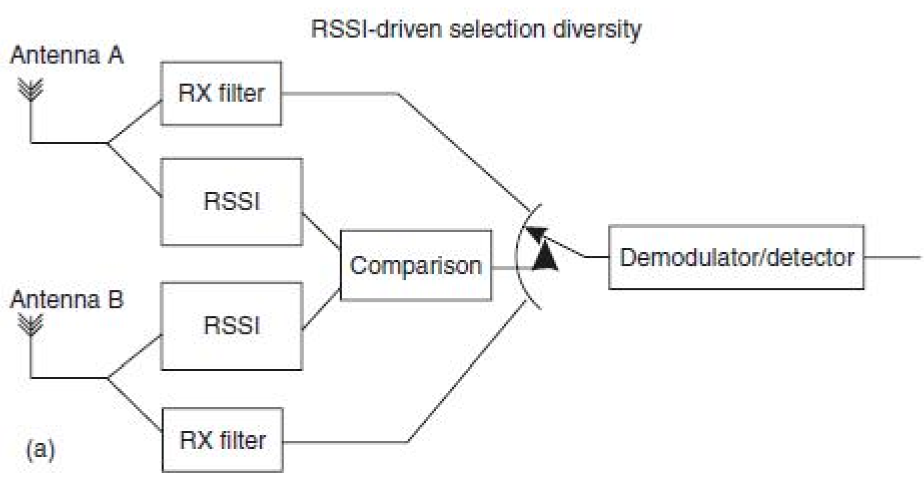
Diversity gain reflects the fact that it is improbable that several antenna elements are in a fading dip simultaneously; the probability for very low signal levels is thus decreased by the use of multiple antenna elements.

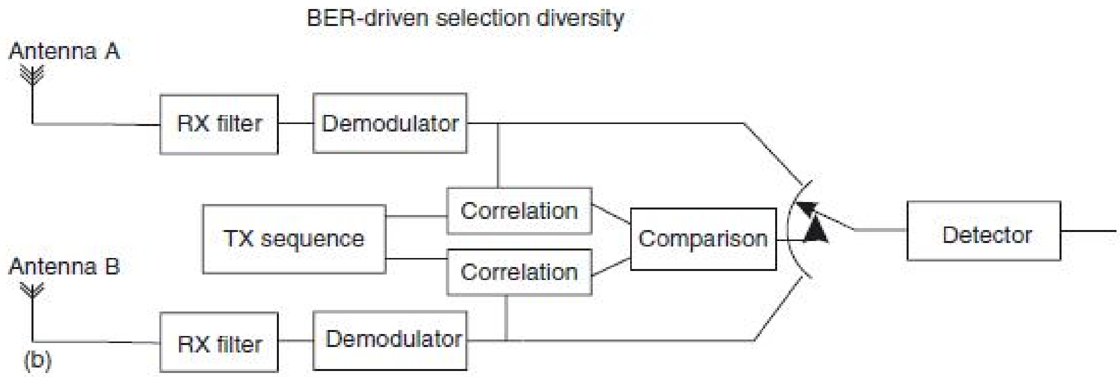
Beamforming gain reflects the fact that (for combining diversity) the combiner performs an averaging over the noise at different antennas. Thus, even if the signal levels at all antenna elements are identical; the combiner output SNR is larger than the SNR at a single-antenna element.

Selection Diversity

Received-Signal-Strength-Indication-Driven Diversity

In this method, the RX selects the signal with the largest instantaneous power (or Received Signal Strength Indication – RSSI), and processes it further. This method requires Nr antenna elements, Nr RSSI sensors, and a Nr-to-1 multiplexer (switch), but only one RF chain (see Figure 4.7). The method allows simple tracking of the selection criterion even in fast-fading channels. Thus, we can switch to a better antenna as soon as the RSSI becomes higher there.





Selection diversity principle: (a) Received-signal-strength-indication-controlled diversity. (b) Bit error- rate-controlled diversity.

1. If the BER is determined by noise, then RSSI-driven diversity is the best of all the selection diversity methods, as maximization of the RSSI also maximizes the SNR.

2. If the BER is determined by co-channel interference, then RSSI is no longer a good selection criterion. High receive power can be caused by a high level of interference, such that the RSSI criterion makes the system select branches with a low signal-to-interference ratio.

3. Similarly, RSSI-driven diversity is suboptimum if the errors are caused by the frequency selectivity of the channel. RSSI-driven diversity can still be a reasonable approximation, because that errors caused by signal distortion occur mainly in the fading

The cdf is, by definition, the probability that the instantaneous SNR lies below a given level. As the RX selects the branch with the largest SNR, the probability that the chosen signal lies below the threshold is the product of the probabilities that the SNR at each branch is below the threshold. In other words, the cdf of the selected signal is the product of the cdfs of each branch:

Advantages of RSSI:

1. Only one RF chain is required. It is processed with only a single received signal at a time.

2. Easy to implement.

Disadvantage of RSSI:

1. It wastes signal energy by discarding (Nr -1) copies of received signal.
2. It is not an optimum method.

Bit-Error-Rate-Driven Diversity

For BER-driven diversity, we first transmit a training sequence – i.e., a bit sequence that is known at the RX. The RX then demodulates the signal from each receive antenna element and compares it with the transmit signal. The antenna whose associated signal results in the smallest BER is judged to be the "best," and used for the subsequent reception of data signals. A similar approach is the use of the mean square error of the "soft-decision" demodulated signal, or the correlation between transmit and receive signal.

BER-driven diversity has several drawbacks:

1. The RX needs either Nr RF chains or demodulators (which makes the RX more complex), or the training sequence has to be repeated Nr times (which decreases spectral efficiency), so that the signal at all antenna elements can be evaluated.

2. If the RX has only one demodulator, then it is not possible to continuously monitor the selection criterion (i.e., the BER) of all diversity branches. This is especially critical if the channel changes quickly.

3. Since the duration of the training sequence is finite, the selection criterion – i.e., bit error probability – cannot be determined exactly. The variance of the BER around its true mean decreases as the duration of the training sequence increases.

Disadvantage of BER

1. More number of RXs are needed, which makes the RX more complex.

2. The training sequence has to be repeated Nr times, which decreases spectral efficiency.
3. If the channel changes quickly, more than one demodulators are required.
4. Duration of training sequence increases, BER decreases. So trade off between duration of training sequence and BER is maintained.
5. Diversity branches are monitored all the times, so hardware effort increases, spectral efficiency is reduced.

Switched Diversity

The main drawback of selection diversity is that the selection criteria (power, BER, etc.) of all diversity branches have to be monitored in order to know when to select a different antenna. As we have shown above, this leads to either increased hardware effort or reduced spectral efficiency. An alternative solution, which avoids these drawbacks, is switched diversity. In this method, the selection criterion of just the active diversity branch is monitored. If it falls below a certain threshold, then the RX switches to a different antenna.

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. This problem can be avoided by introducing a 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: switching threshold and hysteresis time. These parameters have to be selected very carefully: 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. 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.

Disadvantage:

- Performance is worst than that of selection diversity.

Combining Diversity

Basic Principle

Selection diversity wastes signal energy by discarding (Nr − 1) copies of the received signal. This drawback is avoided by combining diversity, which exploits all available signal copies. Each signal copy is multiplied by a (complex) weight and then added up.

complex weight = phase correction + real weight of the amplitude

• Phase correction causes the signal amplitudes to add up, while, on the other hand, noise is added incoherently, so that noise powers add up.

• For amplitude weighting, two methods are widely used:

- Maximum Ratio Combining (MRC) weighs all signal copies by their amplitude.
- Equal Gain Combining (EGC), where all amplitude weights are the same (in other words, there is no weighting, but just a phase correction). The two/ methods are outlined in Figure 4.8.

Maximum Ratio Combining

MRC compensates for the phases, and weights the signals from the different antenna branches according to their SNR. This is the optimum way of combining different diversity branches – if several assumptions are fulfilled. Let us assume a propagation channel that is slow fading and flat fading. The only disturbance is AWGN. Under these assumptions, each channel realization can be written as a time-invariant filter with impulse response:

$$h_n(\tau) = \alpha_n \delta(\tau) \qquad (4.6)$$

where α_n is the (instantaneous) gain of diversity branch n. This signals at the different branches are multiplied with weights w^*_n and added up, so that the SNR becomes

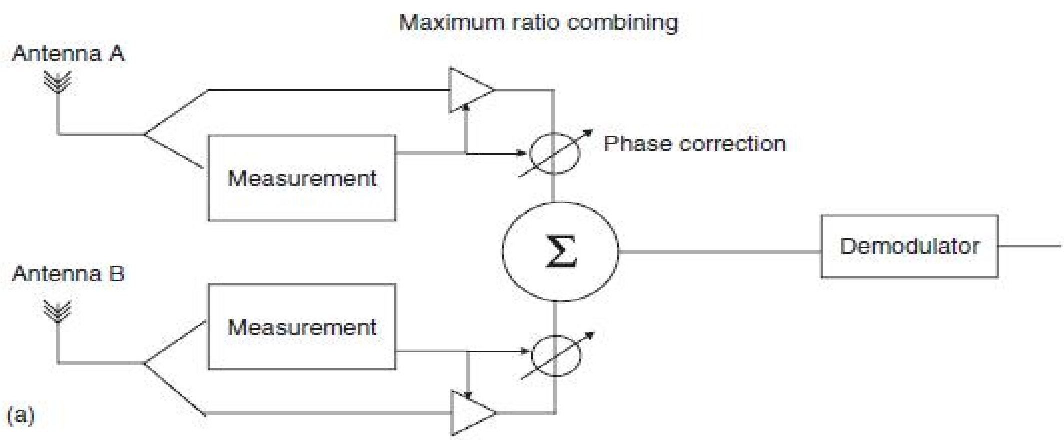
$$\frac{\left|\sum_{n=1}^{N} w_n^* \alpha_n\right|^2}{P_n \sum_{n=1}^{N} |w_n|^2} \qquad (4.7)$$

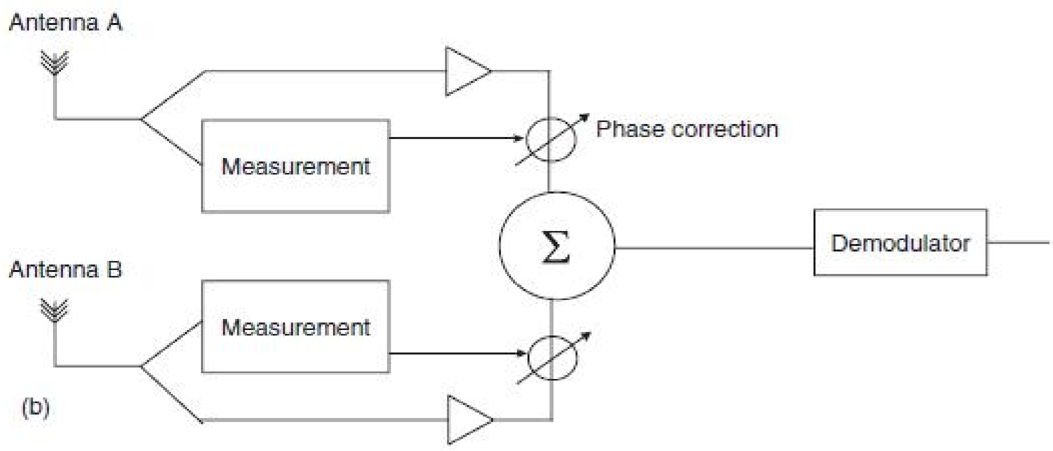
where P_n is the noise power per branch. The SNR is maximized by choosing the weights as

$$w_{MRC} = \alpha_n \qquad (4.8)$$

i.e., the signals are phase-corrected (remember that the received signals are multiplied with w*) and weighted by the amplitude. We can then easily see that in that case the output SNR of the diversity combiner is the sum of the branch SNRs:

$$\gamma_{MRC} = \sum_{n=1}^{N_r} \gamma_n \qquad (4.9)$$





Combining diversity principle: (a) maximum ratio combining, (b) equal gain combining.

If the branches are statistically independent, then the moment-generating function of the total SNR can be computed as the product of the characteristic functions of the branch SNRs.

5. Explain with relevant diagrams the layered space time structure with respect to MIMO systems.(MAy/JUNE 2016)

Layered space time architecture allow us to break up the demodulation process into several separate pieces.When this technique is combined with capacity achieving codes,it can closely approximate the capacity of a MIMO system..These structures are also widely known under the name of BLAST (Bell labs Layered Space Time) architectures.

Horizontal BLAST:

Horizontal BLAST (H-BLAST) is the simplest possible layered space time structures.The transmitter first demultiplexes the datastream into into N_t parallel streams ,each of which is encoded separately.Each encoded data stream is then transmitted from a different transmit antenna.The channel mixes up the different data streams the RX seperates them out by nulling the interference subtraction.In other words, the RX proceeds in the following steps:

- It considers the first datastream as the useful one, and regards the other datatastream as interference.It can be then use optimum combining for suppression of interfering streams.The Receiver has Nr≥Nt antenna elements available.If Nr=Nt.it can suppress all Nt-1 interfering datastreams,and receive the desired data stream with diversity order .if the RX has more antennas ,it can receive the first datastream with better quality.But at any case interference from the other streams can be eliminated.

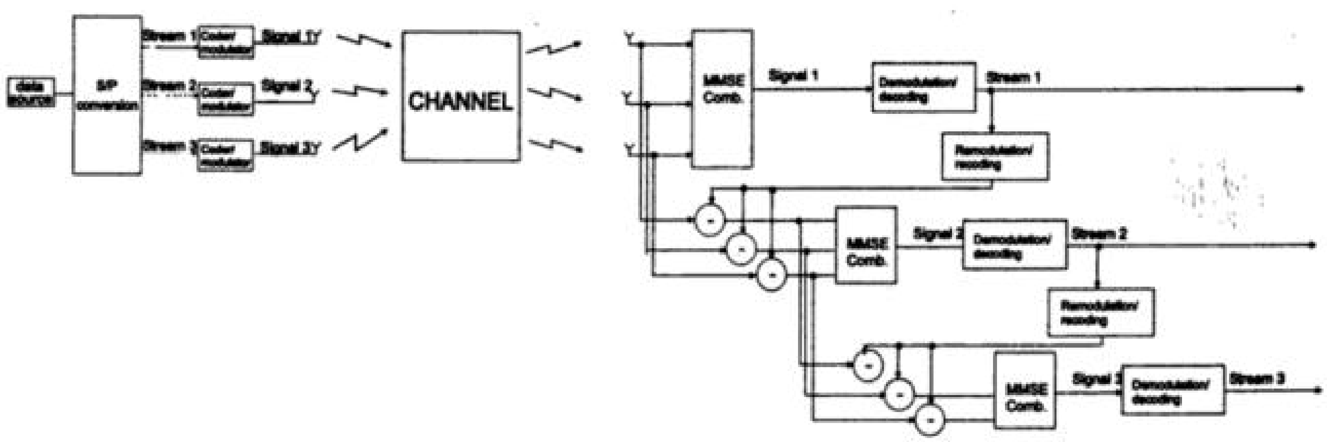


Figure: Block diagram of a horizontal BLAST transceiver

- The desired stream can now be demodulated and decoded.Outputs from that process are firm decisions on the bits of stream 1.Since we have separate encoding for different datasteams,we need only knowledge of the first data stream to complete the decoding process..
- The bits that have been decoded are now re-encoded and remodulated.Multiplying the symbol stream by the transfer function of the channel,we obtain the contribution that stream 1 has made to the total received signal at the different antenna elements.
- We sustract these contributions from the signals at the different antenna elements.
- Now we consider the cleaned up signal and try to detect the second data stream.We again have Nt received signals,but only Nt-2 interferers.Using optimum combining again,we can now receive the desired data stream with diversity order 2.
- The next step is again decoding,recoding and remodulating the considered datastream and subtraction of the associated signal from the total 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 datastream is decoded.

- Even with stream ordering,HBlast does not achieve channel capacity.However its simplity still makes the scheme an attractive one

Question paper Code:51416

B.E/B.Tech DEGREE EXAMINATION, MAY/JUNE 2014

Seventh Semester

Electronics & Communication Engineering

EC 2401-WIRELESS COMMUNICATION

(Regulation 2008/2010)

Time: Three hours Maximum: 100 marks

PART A(10x2=20 marks)

1. What are the basic requirements for wireless services?
2. What is frequency hopped multiple access?
3. State the propagation effects in mobile radio.
4. Interpret link budget equation.
5. What are the main features of QPSK?
6. What are Rayleigh and Ricean fading?
7. Compare macro and micro diversity.
8. What are the applications of non linear equalizers?
9. Why QPSK is preferred for wireless communications?
10. List the advantages of third generation (3G) networks.

Part B-(5*16=80 marks)

11. (a) (i) Discuss about the technical challenges face by the wireless communication. (10)

(ii)What are the features of interference limited systems. (6)

Or

(b) What are the major differences between TDMA, FDMA and CDMA?(16)

12. (a) (i)Explain how signal propagates against free space attenuation and reflection. (8)

(ii)Discuss about the temporal channel variations in Fixed Wireless Systems. (8)

Or

(b) (i) Explain in detail two path model propagation mechanism. (8)

(ii)Explain different models for characterizing wide band channels. (8)

13. (a) Explain with neat diagram the QPSK based transmission and reception technique and their significance in the wireless system. (16)

Or

(b)Explain with neat diagram, the principle of Gaussian Minimum shift Keying receiver and mention how this is different from MSK. (16)

14. (a) (i)What is the need for diversity ?List different types of diversity. (6)

(ii)Explain with diagram, the different techniques available for signal combining.(10)

Or

(b)With neat block diagram explain how RAKE receiver provides diversity to improve the performance of CDMA receiver (16)

15. (a)Describe the principle involved in Cellular Code-Division-Multiple-Access Systems. (16)

Or

(b)Explain with necessary diagram, the operation of Orthoganal FrequencyDivide Multiplexing transceiver. (16)

Question paper Code: 91419

B.E/B.Tech DEGREE EXAMINATION, NOVEMBER/DECEMBER 2014

Seventh Semester

Electronics & Communication Engineering

EC 2401-WIRELESS COMMUNICATION

(Regulation 2008/2010)

Time: Three hours Maximum: 100 marks

PART A(10x2=20 marks)

1. What are the different modules of a basic cellular system?
2. State advantages of CDMA over FDMA?
3. List the different types of propagation machanisms.
4. What are the different fading effects due to Doppler Spread?
5. State advantages of Offset-QPSK.
6. List the advantages of GMSK.
7. List the different types of channel coding techniques.
8. Differentiate between Macrodiversity and Microdiversity.
9. What are the effects of Multipath propagation on CDMA?
10. List some important features of 3G networks.

PART B(5x16=80 Marks)

11. (a) (i) With a block diagram of a basic cellular system, explain its various

functional modules and the method by which a call is routed. (10)

(ii) Explain in detail a handoff scenario at cell boundary (6)

Or

(b) (i) Explain the different types of multipath propagation in wireless communication. (10)

(ii) With neat illustration, explain CDMA. (6)

12. (a) (i) Explain briefly on outdoor propagation models. (8)

(ii) Describe in detail Two Ray Rayleigh Fading Model. (8)

Or

(b) (i) Explain on path loss estimation techniques using path loss models.(8)

(ii) Describe on Ricean distribution.

13. (a) (i) Explain with neat constellation diagram the modulation technique of QPSK. (8)

(ii) List the advantages and application of BPSK. (8)

Or

(b) (i) Describe with a block diagram π/4 Quadrature Phase Shift Keying and its advantages (8)

(ii) What is MSK? Explain its power spectral density. (8)

14. (a) (i) With a neat block diagram, explain the principle of Macrodiversity. (8)

(ii) Explain the operation an adaptive equalizer at the receiver side. (8)

Or

(b) (i) Explain with a block diagram Maximal ratio combiner. (8)

(ii) Describe on Polarization and Space Diversity. (8)

15. (a) Write short notes on the following :

(i) Frequency Hopping and its advantages. (8)

(ii)Orthogonal FDM (OFDM) (8)

Or

(b) Discuss in detail the 2G and 3G wireless network standards. Compare the relative merits and demerits of both the standards (16)

Question Paper Code: 71467

B.E/B.TECH DEGREE OF EXAMINATION, APRIL/MAY 2015

Seventh Semester

Electronics and Communication Engineering

EC2401/EC71/10144 EC701-Wireless communication

(Regulation 2008/2010)

Time: Three hours Maximum:100marks

Answer ALL Questions

PART A – (10 *2=20 marks)

1. State the difference between small-scale fading and large scale fading?
2. Mention a few techniques used to expand the capacity of a cellular system?
3. Interpret Snell's law?
4. List the properties of wideband channels?
5. Comment on the necessity of a Gaussian filter in GMSK?
6. List the advantages of digital modulation techniques?
7. What do you meant by transmit diversity?
8. Write about MMSK decision feedback equalizer?
9. Characteristics the effects of multipath propagation on code division multiple access?
10. What are the basic channels available in GSM?

PART B- (5*16=80 marks)

11. (a) (1) Elaborate about the factors which influence small scale fading. (8)

(2) Comment on the different types of services in detail. (8)

Or

(b) (1) Discuss about the construction and destructive interference in detail. (8)

(2) Give the details about the causes for Inter symbol Interference? How can ISI be eliminated. (8)

12. (a) (1) Explain in brief the three propagation mechanisms which have impact On Propagation in a mobile environment (8)

(2) Define Brewster angle .Calculate Brewster angle for a wave impinging On a ground permittivity=4 (8)

Or

(b) In detail explain about channel classification. (16)

13 (a) (1) Explain the structure of Wireless Communication link in detail (6)

(2) Demonstrate the generation and detection and bit error probability of QPSK scheme. (10)

Or

(b)(1)How MSK signals are generated? Explain in detail (8)

(2)Discuss in detail the demodulation techniques for Minimum Shift keying. (8)

14 (a) Write the short notes on

(1) Spatial diversity (4)
(2) Temporal diversity (4)
(3) Polarization diversity (4)
(4) Macrodiversity (4)

Or

(b) (1)Explain in detail about linear equalizers (8)

(2)With suitable diagrams, explain channel coding and speech codingTechniques. (8)

15.(a) Examine about the effects of multipath propagation on CDMA (16)

Or

(b) (1) Illustrate the block diagram of IS-95 Transmitter (8)

(2) Give a detailed description off OFDM Transceiver. (8)

Question paper Code: 51416

B.E/B.Tech DEGREE EXAMINATION, NOVEMBER/DECEMBER 2015

Seventh Semester

Electronics & Communication Engineering

EC 2401-WIRELESS COMMUNICATION

(Regulation 2008/2010)

Time: Three hours Maximum: 100 marks

PART A(10x2=20 marks)

1. Mention the operating frequency ranges for AMPS and ETACS systems.
2. Define mean excess delay and rms delay spread.
3. Define Co-channel Interference.
4. Define Coherence time.
5. What do you mean by Non-coherent Detection?
6. Draw the Constellation diagram of Binary Frequency Shift Keying system.
7. If a digital signal processing chip can perform one million multiplications per second, determine the time required between each iteration for the following adaptive equalizer algorithm LMS.
8. What is Transmit Diversity?

9. Draw the block diagram of a Direct Sequence Spread Spectrum Transmitter.

10. What is IS-95 Standard?

PART-B—(5*16=80marks)

11 (a) (1) With diagram explain Personal Access Communication system. (8)

(2) Briefly explain ETACS System. (8)

(or)

(b) (1) Explain some techniques intended to improve the coverage area and Capacity of cellular system. (8)

(2) Analyze co-channel interference and adjacent channel interference and suggest some measures to reduce them. (8)

12. (a) Derive the expressions for the total Electric field, Eror(d) and received power at distance, Pr(d) using two – ray ground reflection model. (16)

(or)

(b) The fading characteristics of a CW carrier in an urban area to be measured. The following assumptions are made :

I. The mobile receiver uses a simple vertical monopole.

II. Large-scale fading due to path loss is ignored.

III. The mobile has no line-of-sight path to the base station

IV. The pdf of the received signal follows a Rayleigh distribution

(1) Derive the ratio of the desired signal level to the rms signal Level that maximizes the level crossing rate. Express your Answer in dB. (5)

(2) Assuming the maximum velocity of the mobile is 50 km/hr, and the carrier frequency is 900MHz, determine the maximum number of times the signal envelope will fade below the level found in (1) during a one minute test. (6)

(3) How long, on average, will each fade in (2) last? (5)

13. (a) Derive the expression for MSK signal as a special type of continuous phase FSK signal. (16)

(or)

(b) Explain in detail about the Gaussian Minimum Shift keying (GMSK) Transmission and Reception with necessary diagrams. (16)

14. (a) Explain in detail about Space diversity with necessary diagrams (16)

(or)

(b) Derive the LMS Algorithm for an Adaptive Equalizer. (16)

15. (a) Explain in detail about various spread spectrum multiple access techniques with neat block diagrams. (16)

(or)

(b) Draw the basic arrangement of multitone OFDM transceiver and discuss its overall operation. (16)

Question paper Code:91419

B.E/B.Tech DEGREE EXAMINATION, MAY/JUNE 2016

Fifth Semester

Information Technology

EC 6801-WIRELESS COMMUNICATION

(Regulations 2013)

Time: Three hours Maximum: 100 marks

PART A(10x2=20 marks

1. Calculate the Brewster Angle for wave impinging on ground having a permittivity ε_r=5.
2. Define coherence bandwidth.
3. What is soft hand off in mobile communication?
4. What is multiple access technique?
5. Why is MSK referred to as fast FSK?
6. What is windowing?

7. Define adaptive equalization?
8. What are the benefits of RAKE receiver?
9. What is MIMO system?
10. What is transmit diversity?

PART B(5x16=80 Marks)

11. (a) In free space pronagation describe how the signals are affected by reflection diffraction and scattering. (16)

Or

(b) Explain in detail the various parameters involved in mobile multipath channels. (16)

12. (a) Summarise the features of various multiple access technique used in wireless mobile communication. State the advantages and disadvantages of each technique (16)

Or

(b) Explain in detail how to improve coverage and channel capacity in cellular systems (16)

13. (a) Explain in detail Offset QPSK and π/4-DQPSK linear digital modulation techniques employed in wireless communication. (16)

Or

(b) Explain in detail Gaussian Minimum Shift Keying(GMSK) transmission and reception with necessary diagrams. (16)

14. (a) Explain in detail about linear and non linear equalizer. (16)

Or

(b) Write short notes on : (16)

(i) Spatial Diversity
(ii) Frequency Diversity
(iii)Polarization Diversity
(iv)Time Diversity

15. (a) (i) Explain in detail how inherent delay in a multiuser system is overcome by beam forming. (8)

(ii) Explain in detail spatial multiplexing of a MIMO system. (8)

Or

(b) Explain with relevant diagrams the layered space time structure with respect to MIMO systems. (16)

Question Paper code:51467

B.E/B.Tech.DEGREE EXAMINATION,MAY/JUNE 2016.

Seventh Semester

Electronics and Communication Engineerimg

EC2401/EC71/10144EC701-WIRELESS COMMUNIATION

(Regulations 2008/2010)

(Common to PTEC 2401-Wireless Communication for B.E (Part Time)

Sixth Semester-ECE-Regulations 2009)

Time:Three hours Maximum:100 marks

Answer ALL questions

PARTA-(10X2=20 marks)

1.Define frequency reuse.

2.State the operating principle of adhocnetworks.

3.Define co-channel Interference.

4.Define Coherence time.

5.Give the expression for bit error probability of Gaussian Minimum shift keying modulation.

6.What is fading and Doppler spread?

7.Assume for 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 10dB.Compare this with the case of a single receiver without diversity.

8.Definecoding gain.

9.Characterize the effects of multipath propagation on Code Division Multiple Access.

10.What are the basic channels available in GSM?

PART B-(5X16=80 marks)

11.(a)Discuss the types of services,requirements,spectrum limitations and noise considerations of wireless communications. (16)

Or

(b)Expain the principle of Cellular Networks and various types of Handoff techniques. (16)

12. (a) (i) Explain the time invariant two-path model of a wireless propagation channel. (8)

(ii)Brief about the properties of Rayleigh distribution. (8)

Or

(b) (i)Explain the narrow band modeling methods for Short scale fading and large scale fading. (10)

(ii)Brief about the properties of Nakagam distribution. (6)

13.(a) (i) Briefly explain the structure of a wireless Communication Link.(6)

(ii)With block diagram,explain MSK transmitter andreceiver.Derive an expression for MSK and power spectrum. (10)

Or

(b)Derive an expression for :

(i) M-ary phase shift keying and (8)

(ii)Mary quadrature amplitude modulation.

Also derive an expression for their bit error probability (8)

14. (a)Explain in detail about Space diversity with necessary diagrams.

Or

(b)Derive the LMS algorithmfor an Adaptive Equaliser.

15.(a) Examine about the effects of multipath propagation on CDMA. (16)

Or

(b) (i)Illustrate the block diagram of IS-95 transmitter. (8)

(ii)Give a detaileddescription of OFDM transceiver (8)