WIRELESS CHANNELS

Introduction:

- Propagation models that predict the mean signal strength for an arbitrary transmitter - Receiver separation distance are useful in estimating the radio coverage area of a transmitter and are called large-scale propagation models. Since they characterize signal strength over larg T-R separation distance (several 100 or 1000 of meters).

- The propagation models that characterize the rapid flutuations of the received signal strength over very short travel distances (a few wavelength) or short time durations are called Small Scale (or) fading models

Free space propagation Model:-

It is used to predict the received signal strength when transmitter and receiver have a clear, unobstructed LOS path b/w them.

Ex: Satellite communication systems, Microwave LOS radio links

As with most large scale radio wave propagation models, the free space model predicts that received power decays as a function of the T-R separation distance raised to some power

The free space power received by a receiver antenna which is separated from a radiating (transmitting) antenna by a distance d is given by the Friis free space equation

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

P_t - Transmitter power G_t - Transmitter antenna gain

G_r - Receiver antenna gain d - Transmitter - Receiver separation distance in meters

L - System loss factor λ - wavelength in meters

$P_r(d)$ - Received power which is a function of the T-R separation

The gain of an antenna is related to its effective aperture A_e by

$$G = \frac{4\pi A_e}{\lambda^2} \longrightarrow 2$$

The effective aperture A_e is related to the physical size of the antenna & λ is related to the carrier frequency f is given by

$$\lambda = c/f \qquad f_c = \frac{\omega_c}{2\pi} \qquad \omega_c = 2\pi f_c$$

$$\lambda = \frac{c}{\frac{\omega_c}{2\pi}} = \frac{2\pi c}{\omega_c}$$

$$\lambda = \frac{2\pi c}{\omega_c} \longrightarrow 3$$

ω_c = Carrier frequency in radians/second

c = speed of light in meters/s

A value of L = 1 indicates no loss in the system hardware

The friis space equation (1) shows that the received (2) power falls off as the square of the T-R separation distance. This implies that the received power decays with distance at a rate of 20 dB/decade.

An isotropic radiator is an ideal antenna which radiates power with unit gain uniformly in all directions and is often used to reference antenna gains in wireless systems. The effective isotropic radiated power (EIRP) is defined as

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

and represents the maximum radiated power available from a transmitter in the direction of maximum antenna gain as compared to an isotropic radiator.

Effective radiated power is used instead of EIRP to denote the maximum radiated power as compared to a half wave dipole antenna. The ERP will be smaller than the EIRP for the same transmission system.

In practice antenna gains are given in units of dBi (dB gain w.r.t isotropic antenna) or dBd (gain w.r.t a half wave dipole)

The path loss which represents signal alternation as a positive quantity measured in dB. is defined as the difference b/w transmitted power & the received power and may or may not include the effect of the antenna

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

$$P_L(dB) = 10\log\left(P_t/P_r\right) \longrightarrow (5)$$

$$P_L(dB) = -10\log\left(P_r/P_t\right) \longrightarrow (6)$$

Apply equation (1) in equation (6) we get

$$P_L(dB) = -10\log\left(\frac{P_t G_t G_r \lambda^2}{P_t (4\pi)^2 d^2 L}\right) \longrightarrow (7)$$

For no loss system $L=1$

$$P_L(dB) = -10\log\left(\frac{G_t G_r \lambda^2}{(4\pi)^2 d^2}\right) \longrightarrow (8)$$

when antenna gains are excluded, the antennas are assumed to have unity gain and path loss is given by

$$P_L(dB) = 10\log\left(P_t/P_r\right) = -10\log\left(\frac{\lambda^2}{(4\pi)^2 d^2}\right) \longrightarrow (9)$$

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

The Fraunhofer distance is given by

$$d_f = 2D^2/\lambda \longrightarrow (10)$$

... linear dimensions of the antenna

Additionally df must satisfy

$$d_f >> D \quad \& \quad \rightarrow (11)$$

$$d_f >> \lambda \quad \rightarrow (12)$$

The equation does not hold for d=0. For this reason only the large scale propagation models use a close in distance, do as a known received power reference point.

The received power Pr(d) at any distance d>do may be related to Pr at do. The value Pr(do) may be prüted from equation ① & do ≥ df. where do is chosed to be smaller than any praitical distance used in the mobile communication system.

Using eqn ① the received power in free space at a distance greater than do is given by

$$\boxed{P_r(d) = P_r(d_0)\left(\frac{d_0}{d}\right)^2} \quad d \geq d_0 \geq d_f \rightarrow (13)$$

Eqn (13) may be expressed in units of dBm or dBW by simply taking logarithm of both sides & multiplying by 10, than the received power is given by

$$P_r(d)\,dBm = 10\log\left[\frac{P_r(d_0)}{0.001W}\right] + 20\log\frac{(d_0)}{d} \quad d \geq d_0 \geq d_f \quad (14)$$

Where Pr(do) is in units of watts

The reference distance do = 1m in the indoor environments

$d_0 = 100\,m$ or $1\,km$ in Outdoor environments

Sum:

1) Find the far field distance for an antenna with maximum dimension of 1m & operating frequency of 800 MHZ

Given: Largest dimensions of antenna $D = 1m$

operating frequency $f = 800\,MHZ$

$$\lambda = c/f = \frac{3\times10^8}{800\times10^6} = 0.375$$

The far-field distance d_f is obtained as

$$d_f = 2D^2/\lambda = \frac{2\times1^2}{0.375} = 5.33m$$

$$\boxed{d_f = 5m}$$

2) A power of 100 W is supplied to an isotropic radiator what is the power density at a point 10 km away?

Given $P_t = 100W$ $d = 10\,km = 10\times10^3 m$

$$P_D = \frac{P_t}{4\pi d^2} = \frac{100}{4\pi\times(10\times10^3)^2}$$

$$\boxed{P_D = 79.6\ nW/m^2}$$

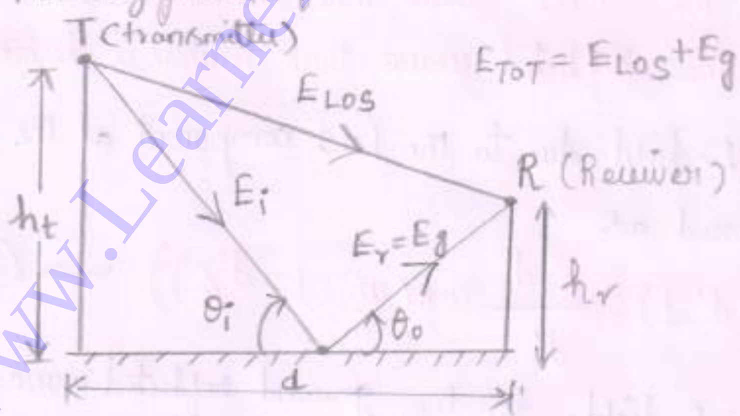
Ground Reflection (Two-Ray) Model :-

In a mobile radio channel a single direct path b/w the base station & a mobile is seldom the only physical means for propagation & hence the free space propagation model

$$\boxed{P_L(dB) = -10\log\left(\frac{G_t G_r \lambda^2}{(4\pi)^2 d^2}\right)}$$

The two-ray ground reflection model is a useful propagation that is based on geometric optics & considers both the direct path & a ground reflected propagation path b/w transmitter and receiver

Two-Ray ground reflection model



In most mobile communication systems, the maximum T-R separation distance is at most only a few tens of kilometers and the earth may be assumed to be flat

The total received field E_{TOT} is given by

$$\boxed{E_{TOT} = E_{LOS} + E_g} \rightarrow (1)$$

E_{LOS} → direct or LOS component

In figure h_t = height of the transmitter

h_r = height of the receiver

E_0 - Free space E field in units of V/m

d_0 - Reference distance from the transmitter

For $d > d_0$ the free space propagation E-field is given by

$$E(d,t) = \frac{E_0 d_0}{d} \cos(\omega_c (t - d/c)) \quad (d > d_0) \rightarrow (2)$$

where $|E(d,t)| = E_0 d_0 / d$ represents the envelope of the E-field at d meters from the transmitter

From the above figure two propagating waves arrives at the receiver

i) The direct wave that travels a distance d'

ii) The reflected wave that travels a distance d''

The E-field due to the LOS component at the receiver can be expressed as

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

and the E-field for the ground reflected wave which has a propagation distance of d'' can be expressed as

$$E_g(d'', t) = \Gamma \frac{E_0 d_0}{d''} \cos\left(\omega_c \left(t - \frac{d''}{c}\right)\right) \rightarrow (9)$$

According to laws of reflection in dielectrics

$$\theta_i = \theta_0 \rightarrow 10$$

$$E_g = \Gamma E_i \rightarrow 11$$

$$E_t = (1+\Gamma)E_i \rightarrow 12$$

Γ - reflection coefficient for ground

For small values of θ_i (grazing incidence) the reflected wave is equal in magnitude & 180° out of phase with the incident wave.

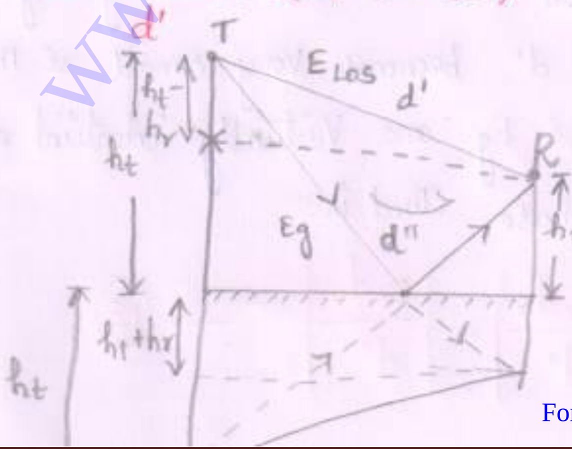
The resultant E-field assuming perfect horizontal E field polarization & ground reflection (i.e $\Gamma_1 = -1$ & $E_t = 0$) is the vector sum of E_{LOS} & E_g

And the resultant E field envelope is given by

$$|E_{TOT}| = |E_{LOS} + E_g| \rightarrow (13)$$

The electric field $E_{TOT}(d,t)$ can be expressed as

$$E_{TOT}(d,t) = \frac{E_0 d_0}{d'} \cos\left(\omega_c\left(t - d'/c\right)\right) + (-1)\frac{E_0 d_0}{d''}\cos\left(\omega_c\left(t - \frac{d''}{c}\right)\right) \rightarrow (14)$$



The method of images used to find the path differences

From the above figure the path difference Δ b/w the LOS and the ground reflected paths canbe expressed as

$$\boxed{\Delta = d'' - d' = \sqrt{(h_t+h_r)^2+d^2} - \sqrt{(h_t-h_r)^2+d^2}} \rightarrow (14)$$

d is very large compared to $h_t + h_r$. By using Taylor series expansion the equation can be simplified by

$$\Delta = d'' - d' \simeq \frac{2h_t h_r}{d}. \rightarrow (15)$$

The path difference θ_Δ is defined by

$$\theta_\Delta = \frac{2\pi\Delta}{\lambda} = \frac{\Delta\omega_c}{c}.$$

Time delay $T_d = \frac{\Delta}{c} = \frac{\theta_\Delta c}{\omega_c c} = \frac{\theta_\Delta}{\omega_c} \rightarrow (16)$

where $\omega_c = 2\pi f_c$.

$$T_d = \frac{\theta_\Delta}{2\pi f_c} \rightarrow (17)$$

It should be noted that d becomes very large the difference b/w d' & d'' becomes very small, & the amplitudes of E_{LOS} & E_g are virtually identical and differ only in phase. That is

$$\left|\frac{E_0 d_0}{d}\right| = \left|\frac{E_0 d_0}{d'}\right| = \left|\frac{E_0 d_0}{d''}\right| \rightarrow (18)$$

Substitute $t = d''/c$ in eqn (14) can be expressed as a phasor sum

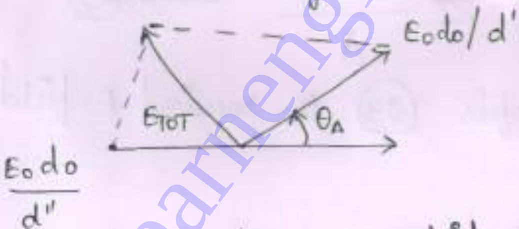
$$E_{TOT}\left(d, t = \frac{d''}{c}\right) = \frac{E_0 d_0}{d'} \cos\left(\omega_c\left(\frac{d''-d'}{c}\right)\right) - \frac{E_0 d_0}{d''}\cos 0^\circ$$

$$= \frac{E_0 d_0}{d'} \angle \theta_\Delta - \frac{E_0 d_0}{d''}\cos 0^\circ$$

$$= \frac{E_0 d_0}{d}\left[\angle\theta_\Delta - 1\right] \longrightarrow (19)$$

where d is the distance over a flat earth b/w the bases of the transmitter & receiver antennas

phasor diagram



Referring to the phasor diagram which shows how the direct and ground reflected wave combines the electric field (at the receiver) at a distance d from the transmitter can be written as

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

$$= \sqrt{\left(\frac{E_0 d_0}{d}\right)^2 \left(\cos^2\theta_\Delta + 1 - 2\cos\theta_\Delta + \sin^2\theta_\Delta\right)} \qquad (\because \cos^2\theta + \sin^2\theta = 1)$$

$$= \frac{E_0 d_0}{\;} \sqrt{2 - 2\cos\theta_\Delta} \qquad (20)$$

Using trigonometric equation, eqn (20) can be expressed as

$$|E_{TOT}(d)| = 2\frac{E_0 d_0}{d} \sin\left(\frac{\theta_\Delta}{2}\right) \rightarrow (21)$$

Eqn (21) may be simplified whenever $\sin(\theta_{\Delta/2}) \simeq \theta_{\Delta/2}$. This Occurs when $\theta_{\Delta/2}$ is less than 0.3 radian. Using eqn (15) & (16)

$$\frac{\theta_\Delta}{2} \simeq \frac{2\pi h_t h_r}{\lambda d} < 0.3 \text{ rad} \rightarrow (22)$$

the eqn (21) may be simplified

$$d > \frac{20\pi h_t h_r}{3\lambda} \approx \frac{20 h_t h_r}{\lambda} \rightarrow (23)$$

As long as d satisfies (23) the received E-field can be approximated as

$$E_{TOT}(d) = \frac{2E_0 d_0}{d} \frac{2\pi h_t h_r}{\lambda d} \simeq \frac{K}{d^2} \text{ V/m} \rightarrow (24)$$

K = constant related to E_0, the antenna heights & the wavelengths

Then the received power at a distance d from the transmitter for the 2 ray ground reflection model can be expressed as

$$P_r = P_t G_t G_r \frac{h_t^2 h_r^2}{d^4} \rightarrow (25)$$

The eqn (25) at large distances $(d >> \sqrt{h_t h_r})$ the received power falls off with distance raised to the fourth power

At large values of d the received power & path loss become independance of frequency

The path loss for the 2 ray model can be expressed in dB as

$$P_L(dB) = 40\log d - (10\log G_t + 10\log G_r + 20\log h_t + 20\log h_r) \quad (26)$$

when equation (26) is evaluated for $\theta_\Delta = \pi$ then $d = 4h_t h_r/\lambda$ is where the ground appears in the 1st Fresnel Zone b/w the transmitter & receiver

Advantages:-

This model has been accurate for predicting the large scale signal strength over distance of several kms for mobile radio system, that use tall towers as well as for LOS microcell channels in urban environments.

Sum:-

A mobile is located 5 km away from base station and uses of vertical $\lambda/4$ monopole antenna with a gain of 2.55 dB to receive cellular radio signals. The E field at 1 km from the transmitter is measured to be 10^{-3} V/m. The carrier frequency used for this system

Given

T-R separation distance = 5 km

E-field at a distance of 1 km = 10^{-3} V/m

Frequency of operation = f = 900 MHz

$$\lambda = c/f = \frac{3\times10^8}{900\times10^6} = 0.333\,m$$

a) Length of the antenna $L = \lambda/4$

$$= 0.333/4 = 0.0833\,m$$
$$= 8.33\,cm$$

Effective aperture $A_e = G\lambda^2/4\pi$

$$= 2.55\times(0.333)^2/4\pi = 0.016\,m^2$$

(b) Since $d >> \sqrt{h_t h_r}$

$$E_R(d) = \frac{2E_0 d_0 2\pi h_t h_r}{d\lambda d} = \frac{k}{d^2}\ V/m$$

$$= \frac{2\times10^{-3}\times1\times10^3}{5\times10^3}\left[\frac{2\pi\times50\times1.5}{0.33\times5\times10^3}\right]$$

$$= 1131.1\times10^{-6}\ V/m$$

The received power

$$P_r(d) = \frac{(1131.1\times10^{-6})^2}{377}\left[\frac{1.8\times(0.33)^2}{0.33\times5\times10^3}\right]$$

$$P_r(d) = -92.68\ dBm$$

Link Budget Design

Most propagation models are derived using a combination of analytical & empirical methods. The empirical approach is based on fitting curves or analytical expressions that recreate a set of measured data. This has the advantage of simplicity taking into account all propagation factors both known & unknown through actual field measurement. It consist of two models. The models are

(i) long distance path loss model
(ii) log-normal shadowing model

Long distance path loss model

The average large scale path loss for an arbitrary T-R separation is expressed as a function of distance by using a path loss exponent, n

$$\overline{P_L}(d) \propto \left(\frac{d}{d_0}\right)^n \longrightarrow ①$$

(or)

$$\overline{P_L}(dB) = \overline{P_L}(d_0) + 10n \log(d/d_0) \longrightarrow ②$$

n – path loss exponent which indicates the rate at which path loss ↑ with distance

d_0 – close in reference distance which is determined from measurements close to the transmitter

d – distance b/w T-R

Eqn 1 & 2 denoted ensemble average of all possible path loss value for a given value of d.

In large coverage cellular systems 1km reference distances are commonly used. Whereas in μcellular systems much smaller distance are used.

Log-Normal Shadowing

Measurements have shown that at any value of d pathloss $P_L(d)$ at a particular location is random & distributed log normally about the mean distance dependent value.

$$\boxed{P_L(d)[dB] = \overline{P_L}(d) + X_\sigma} \rightarrow 1$$

$$= \overline{P_L}(d_0) + 10n \log\left(\frac{d}{d_0}\right) + X_\sigma \rightarrow 2$$

&

$$P_r(d)[dBm] = P_t(dBm) - P_L(d)[dB] \rightarrow 3$$

X_σ - Zero mean gaussian distributed random variables with standard deviation

The log-normal distribution describes the random shadowing effects with occure over a large number of measurement location which have the same T-R separation but have different levels of clutter on the propagation path. This phenomenon is referred to as log-normal shadowing.

Since $P_L(d)$ is a random variable with a normal distribution in dB about the distance dependent mean so is $P_r(d)$ & the Q function or error function may be used to determine the probability that the received signal level will exceed a particular level. The Q

$$Q(z) = \frac{1}{2\pi} \int_{z}^{\alpha} \exp\left(\frac{-x^2}{2}\right) dx \rightarrow (4)$$

$$= \frac{1}{2}\left[1 - \text{erf}\left(\frac{z}{\sqrt{2}}\right)\right] \rightarrow (5)$$

where $$Q(z) = 1 - Q(-z) \rightarrow (6)$$

The probability that the received signal will exceed a certain value γ can be calulated from the cumulative density function as

$$P_r(P_r(d) > \gamma) = Q\left(\frac{\gamma - \overline{P_r(d)}}{\sigma}\right) \rightarrow (7)$$

$||^{ly}$ the probability that the received signal level will below γ is given by

$$P_r(P_r(d) < \gamma) = Q\left(\frac{\overline{P_r(d)} - \gamma}{\sigma}\right) \rightarrow (8)$$

Small scale fading

Small scale fading (or) simply fading is used to describe the rapid fluctuations of the amplitudes. phases or multipath delays of a radio signal over a short period of time or travel distance so that large scale path loss effect may be ignored.

Multipath propagation :-

There are many obstacles and reflectors in the wireless propagation channel so the transmitted signal arrives at the receiver from various directions over a multiplicity of paths. Such a phenomenon is called multipath. This multipath waves combine at the receiver antenna to give a resultant signal which can vary widely in amplitude & phase. This process is known as Fading.

Effects of fading

(i) Rapid changes in signal strength over a small travel distance or time interval.

(ii) Doppler shift in frequency

(iii) Time dispersion caused by multipath propagation delays.

Factors influencing small-scale fading

- Multipath propagation
- Speed of the mobile
- Speed of surrounding object
- coherence Bandwidth

Doppler shift

Due to the relative motion b/w the mobile & the base station each multipath wave experiences an apparent shift in frequency. The shift in received signal frequency due to motion is called Doppler shift

$$f_d = \frac{V}{\lambda} \cos\theta$$

Types of Small Scale fading :-

Small scale fading (Based on multipath time delay spread)

- Flat Fading
  1. BW of signal < BW of channel
  2. Delay spread < symbol period
- Frequency selective Fading
  1. BW of signal > BW of channel
  2. Delay spread > symbol period

Small scale fading (Based on Doppler spread)

- Slow fading
  1. Low Doppler spread
  2. coherence time > symbol period
  3. channel variation slower than base band signal variations
- Fast fading
  1. High Doppler spread
  2. coherence time < symbol period
  3. channel variation faster than base band signal variations

Parameters of mobile multipath channels:-

For a multipath channel the power delay profile is useful tool in making a lot of measurements about the multipath channel parameters.

Time dispersion Parameters

The mean excess delay, rms delay spread, & excess delay spread are multipath channel parameters that can be determined from a power delay profile

(i) Mean excess Delay

It's denoted by $\bar{\tau}$ is an indication of the time dispersive properties of a channel. The mean excess delay is given by

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

(ii) RMS Delay Spread

It's the square root of the second control moment of the Power delay profile and is given by $\sigma_\tau = \sqrt{\overline{\tau^2} - (\bar{\tau})^2}$

Where

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

These delays are measured relative to the first

that the ... arriving at the receiver at $\tau_0 = 0$

(iii) Excess Delay spread

The maximum excess delay of the power delay profile is defined to be the time delay during which multipath energy falls to x dB below the maximum.

The maximum excess delay is defined as $\tau_x - \tau_0$

τ_0 - first arriving signal

τ_x - maximum delay at which a multipath component is within x dB of the strongest arriving multipath signal

Coherence Bandwidth :-

The coherence B.W B_c is defined as a statistical measure of the ranges of frequencies over which the channel can be considered flat (i.e) a channel which passes all spectral components with approximately linear phase and equal gain.

The ranges of frequencies over which two frequency components have a strong potential for amplitude correlation is also called as a coherence Bandwidth.

If the coherence B.W is defined as the bandwidth over which the frequency correlation function is above 0.9, then the coherence B.W approximately

$$B_c = 1/50\sigma_\tau$$

If the frequency correlation function is above 0.5, then the B_c is

Doppler spread

Time dispersive nature of the channel in a local area is described by delay spread & coherence B.W

The time varying nature of the channel in a small-scale region is described by Doppler spread & coherence time.

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

The Doppler spectrum in the range from $f_c - f_d$ to $f_c + f_d$

$f_c \rightarrow$ Pure sinusoidal tone of frequency

$f_d \rightarrow$ Doppler shift

Coherence Time (T_c)

Time domain dual of doppler spread is called coherence Time. It is used to characterize the time varying nature of the frequency dispersiveness of the channel in the time domain.

The coherence time & Doppler spread are inversely proportional to one another

$$\boxed{T_c \simeq 1/f_m}$$

If the coherence time is define as the time over which the time correlation function is above 0.5

$T_c \simeq 9/16\pi f_m$ f_m - max. Doppler shift

Fading due to time Delay spread

It causes the transmitted signal to undergo either flat or frequeny selective fading

Flat Fading

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

In this type of fading the strength of the received signal changes with time due to flutuation in the gain of the channel caused by multipath

The condition for flat fading is

$$B_s \ll B_c$$

$$T_s \gg \sigma_\tau$$

where

B_s - Signal Band width

B_c - coherence Bandwidth

T_s - Symbal period (i.e) Reciprocal B.w

σ_τ - rms delay spread

Frequency Selective fading

If the mobile radio channel has a constant gain & linear phase response over bandwidth that is Smaller than Bandwidth of the Transmitted signal than the received signal will undergo frequency selective Fading.

The frequency selective fading is due to time dispersion of the transmitted symbols within the channel. Thus the channel includes Intersymbol interference (ISI)

The condition for frequency selective fading is

$$B_S \gg B_C$$

$$T_S \ll \sigma_\tau$$

Frequency selective fading channels are also known as wideband channels since the Bandwidth of the signal is wider than the bandwidth of the channel impulse response.

Fading effects due to Doppler spread

It is classified into 2 types

i) Slow fading ii) Fast fading

Slow fading

If the channel impulse response changes at a rate much slower than the transmitted baseband signal than the fading is

The condition for slow fading is

$T_S << T_C$ T_S - Symbol period

$B_S >> B_D$ T_C - coherence Time

B_S - Signal B.W

B_D - Doppler B.W

Fast fading

If the channel impulse response changes rapidly within the symbol duration then the type of fading is called fast fading.

This causes frequency dispersion (i.e) time selective fading due to Doppler spreading which leads to signal distortion.

The condition for fast fading is

$T_S >> T_C$

$B_S << B_D$

This type of fading only deals with the rate of changes of the channel due to motion

This only occurs for very low data rates.

The velocity of the mobile & the baseband signaling determines whether a signal undergoes fast fading or slow fading

The fast & slow fading deal with the relationship b/w the time rate of change in the channel & the transmitted signal & nor with propagation path loss models.

UNIT II

CELLULAR ARCHITECTURE

Multiple Access Techniques :-

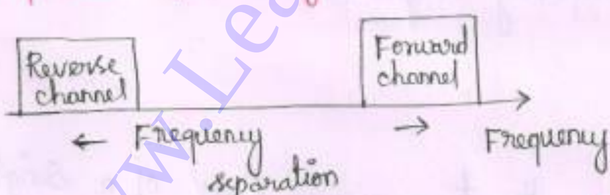
- In order to accomodate many mobile users within the finite amount of radio spectrum the multiple access techniques are used.

Three types of duplexing techniques are used. They are

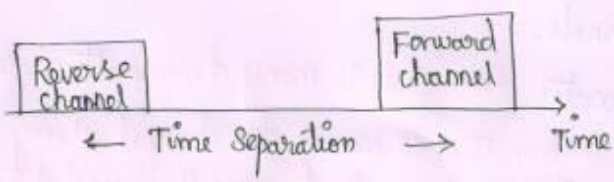
1. Simplex
2. Half duplex
3. Full duplex

The duplexing may be done by using frequency or time domain techniques.

FDD: The frequency division duplexing provides tub distinct bands of frequencies for every user.



TDD: The time division duplexing uses time instead of frequency to provide a forward and reverse link.



Types of Multiple access Techniques

3 major access techniques are used in wireless communication system. The types are

1. Frequency Division Multiple Access (FDMA)
2. Time Division Multiple Access (TDMA)
3. Code Division Multiple Access (CDMA)

- These technique can be grouped as narrowband & wideband systems, depending upon how the available bandwidth is allocated to the users.

Narrow band Systems :-

- In a narrowband multiple access system, the available radio spectrum is divided into a large no. of narrowband channels. The channels are operated by using FDD.

- The interference b/w fwd & reverse links on each channel is minimize by frequency separation is made within the frequency spectrum.

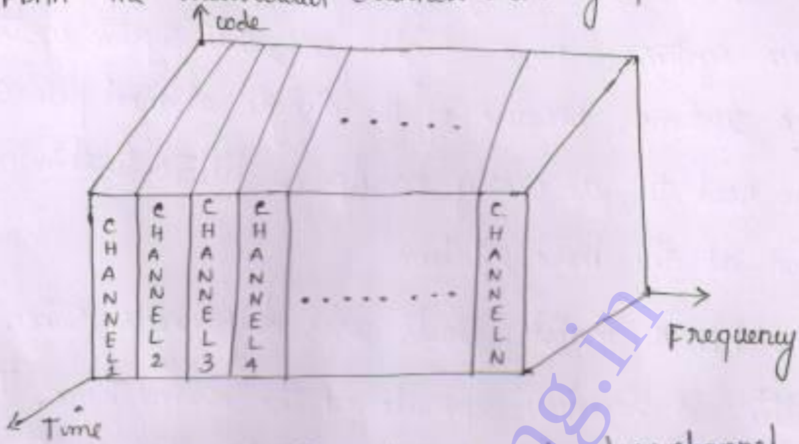
Wideband systems :-

- In this system the transmission B.W of a single channel is much larger than the coherence B.W of the channel. In this system a large number of transmitters are allowed to transmit on the same channels.

- TDMA allocates time slots to many transmitters on the same channel & allows only one transmitter to access the channel at any instant of time but in CDMA it allows all of the transmitter to access the channel at the same time. The

Frequency Division Multiple Access (FDMA)

In FDMA the individual channels are assigned to a individual users.



- Each user is allocated a unique frequency band or channel.

- These channel are assigned on demand to users who request the service. During the period of the call no other user can share the same channel.

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

Features of FDMA:

1) The FDMA channel carriers only one phone circuit at a time

2) The FDMA is usually implemented in narrowband systems.

3) If an FDMA channel is not in use, then it sits idle & cannot be used by other users to ↑ or share capacity. It is essentially a wasted resource.

4) After the assignment of a Voice channel. the base station & the mobile Transmit Simultaneously & continuously.

5) The complexity of FDMA mobile systems is lower when

6) FDMA is a continuous transmission scheme, fewer bits are needed for overhead proposes as compared to TDMA.

7) FDMA systems have higher cell site system costs as compared to TDMA systems, because of the single channel per carrier design and the need to use costly bandpass filters to eliminate spurious radiation at the base station

8) The FDMA mobile units uses duplexers since both the transmitter & receiver operate at the same time. This results in an ↑ in the cost of FDMA subscriber units & base stations

9) FDMA requires tight RF filtering to minimize adjacent channel interference.

Non linear effects of FDMA:-

- In a FDMA system, many channels share the same antenna at the base station. The power amplifiers or the power combiners when operated at or near saturation for maximum power efficiency are non-linear.

- The non-linearities causes signal spreading in the frequency domain & generate intermodulation frequencies. IM can interfere with other channels in the FDMA systems.

- IM is the generation of undesirable harmonics. The no. of channels than can be simultaneously supported in a FDMA system is given by

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

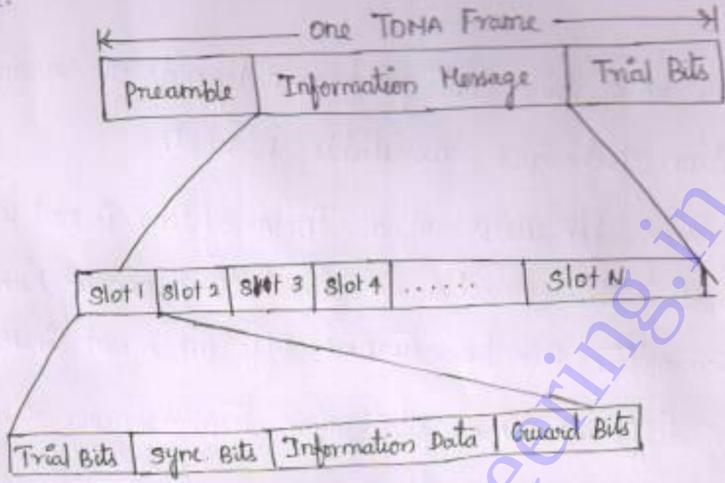
$B_t \rightarrow$ Total spectrum allocation

$B_{guard} \rightarrow$ Guard band allocated at the edge of the allocated spectrum band

$B_c \rightarrow$ channel B.W

Time Division Multiple Access

- TDMA systems divide the radio spectrum into time slots & in each slot only one user is allowed either to transmit or receive.



TDMA frame Structure

- TDMA systems transmit data in buffer & burst method. Thus the transmission for any user is noncontinuous.

- The frame consists of a no. of slots. Each frame is made up of a preamples, an information message & trial bits.

- In TDMA/TDD, half of the time slots in the frame information message would be used for the forward link channels & half used for reverse link channels.

- In TDMA/FDD, similar frame structure would be used for either forward or reverse transmission but the carrier frequency would be different for the forward & reverse links.

- In TDMA frame the preamble contains the address & synchronization information that both the base station & the subscribers used to identify each other.

- Guard times are utilized to allow synchronization of the receivers between different slots & frames. Different TDMA wireless standards have different TDMA frame.

Features of TDMA

1) In TDMA the no. of time slots per frame depends on several factors such as modulation technique, available B.W etc.

2) Data transmission for users of a TDMA system is not continuous but occurs in bursts. This results in low battery consumptions, since the subscriber transmitter can be turned off when not in use.

3) TDMA uses different time slots for transmission & reception thus duplexers are not required.

4) Adaptive equalization is usually necessary in TDMA systems, since the transmission rates are generally very high as compared to FDMA channels

5) In TDMA the guard time should be minimized.

6) High synchronization overhead is required in TDMA systems because of burst transmissions

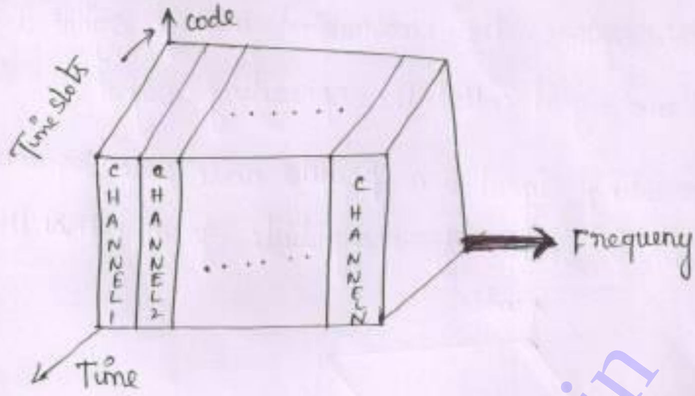
7) Advantage: It is possible to allocate different number of time slots / frame to different users. Thus B.W can be supplied on demand to different users by concatenating time slots based on priority

The no. of channel in TDMA system is given by

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

m → Maximum no. of TDMA users supported on each radio channel

TDMA scheme



Efficiency of TDMA:-

- The efficiency of TDMA system is a measure of the percentage of transmitted data that contains information as opposed to providing overhead for the access scheme.

The frame efficiency η_f is the percentage of bits / frame which contains transmitted data.

The no. of overhead bits / frame is

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

$N_r \rightarrow$ No. of reference bursts / frame

$N_t \rightarrow$ No. of traffic burst / frame

$b_r \rightarrow$ No. of overhead bits / reference burst

$b_p \rightarrow$ No. of overhead bits / preamble in each slot

$b_g \rightarrow$ No. of equivalent bits in each guard time interval

The total no. of bits / frame is

$$b_T = T_f R$$

T_f - frame duration

R - channel bit rate

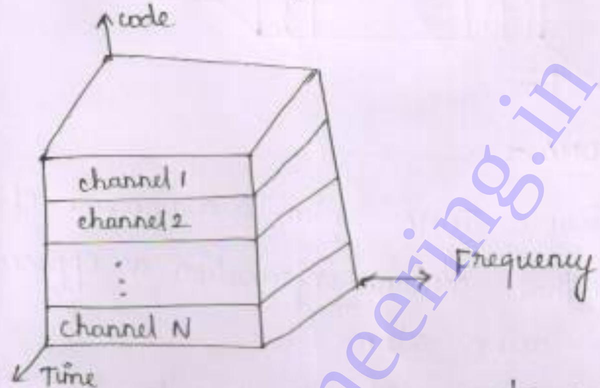
The frame efficiency η_f is given as

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

Code Division Multiple Access

- In this systems, the narrowband message signal is multiplied by a very large b.w signal called the spreading signal.

- The spreading signal is a pseudo noise code sequence that has a chip rate which is orders of magnitude greater than the data rate of the message.



- Uses the same carrier frequency & may transmit simultaneously

- Each user has its own pseudorandom codeword which is approximately orthogonal to all other codewords.

- Each user operates independently with no knowledge of the other users.

- The near far problem occurs when many mobile users share the same channel.

- In general the strongest received mobile signal will capture the demodulator at a base station. To combat the near-far problem power control is used in most CDMA implementation

- power control is implemented at the base station by rapidly sampling the radio signal strength indicator levels of each mobile & then sending a power change command over the forward radio link.

Features of CDMA

1) The CDMA system many users share the Same frequency either TDD or FDD may be used

2) Unlike TDMA or FDMA, CDMA has a soft capacity limit

3) Multipath fading may be substantially reduced because the Signal is spread over a large spectrum.

4) channel data rates are Very high in CDMA systems

5) CDMA uses co-channel cells it can use macroscopic Spatial diversity to provide Soft handoff. Soft handoff is performed by the MSC (i.e) The MSC Can simultaneously monitor a particular user from 2 or more stations. Then the MSC may chose the best Version of the signal at any time without Switching frequencies.

6) self jamming is a problem in CDMA system. It arises from the fact that the spreading sequences of different users are not exactly orthogonal hence in the de-spreading of a particulars PN code, non-Zero contributions to the receiver decision satistic for a desired user arise from the transmission of other users in the system.

7) The near far problem occurs at a CDMA receiver if an undesired user has a high detected power as compared to the desired user.

Cellular concept

In early mobile radio system large coverage area was achieved by using a single, high powered transmitter with an antenna mounted on a tall tower, but it was impossible to reuse those same frequencies throughout the system since any attempts to achieve frequency reuse would result in interference.

The cellular concept offered very high capacity in a limited spectrum allocation without any major technological changes. It is a system level idea which calls for replacing a single high powers transmitter (large cell) with many low power transmitters (small cells) each providing coverage to only a small portion of the service area.

Frequency reuse :-

Each cellular base station is allocated a group of radio channels to be used within a small geographic area called a cell.

The design process of selecting & allocating a channel groups for all of the cellular base stations within a system is called frequency reuse (or) frequency planning

- The concept of cellular frequency reuse where cells labelled with the same letter use the same group of channels

The actual radio coverage of a cell is known as the footprint. When using hexagons to model coverage areas, base stations transmitters are depicted as either being in the centre of the cell (center-excited cells) or on three of the six cell (Vertices edge excited cell)

For center excited cells Omnidirectional antennas & edge excited cells sectored directional antenna are used.

To understand the frequency reuse concept consider a cellular system which has a total of S duplex channels available for use. If each cell is allocated a group of K channels (K<S) & if the S channels are divided among N cells which each have the same number of channels, the total number of available radio channels can be expressed as

$$S = KN$$

cluster:- The N cells which collectively use the complete set of frequencies is called cluster. The above figure using 3 cluster

If a cluster is replicated M times within the system the total number of duplex channels C can be used as a measure eqn $S = kN$ the capacity of a cellular system is directly proportional to number of times a clustered is replicated in a fixed service area

N - cluster size may be equal 4, 7 or 12

Frequency reuse factor of a cellular system is given by $1/N$

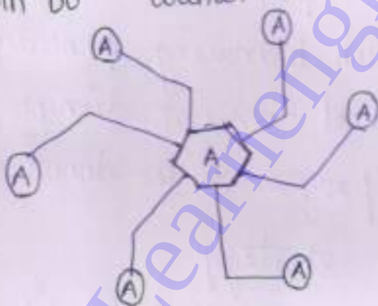
The number of cells / cluster is given by

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

where i & j are non-negative integers. To find nearest co-channel neighbours of a particular cell, one must do the following

(i) Move i cells along any chain of hexagons

(ii) Turn 60° counter clkwise & move j cells



channel Assignment strategies

For efficient utilization of the radio spectrum a frequency reuse is used in order to increase the capacity & minimize the interference.

The channel assignment strategies can be classified into 2 categories

1) Fixed channel assignment strategy

2) Dynamic channel assignment strategy

Fixed channel assignment strategy

- In this strategy each cell is allocated a predetermined set of voice channel.

- Any call attempt within the cell can only be served by the unused channels in that particular cell.

- If all the channel in that cell are occupied the call is blocked & the subscriber does not receive service.

- Several variations of the fixed assignment strategy exist.

- In one approach is called borrowing strategy, a cell is allowed to borrow channels from a neighbouring cell if all of its own channels are already occupied.

- The MSC supervises such borrowing procedures & ensures that the borrowing of a channel does not disrupt or interface with any of the calls in progress in the donar cell.

Dynamic channel assignment strategy

- Voice channels are not allocated to different cells permanently.

- Instead each time a call request is made the serving base station requests a channel from the MSC.

- Dynamic channel assignment reduce the call blocking which increases the trunking capacity of the system.

Handoff

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 is called Handoff.

This handoff operation not only involves identifying a new base station but also requires that the voice & control signals be allocated to channels associated with the new base station.

Handoff must be performed successfully & as infrequently as possible & be impercepitible to the users.

The margin level at which the hand off is made is given by

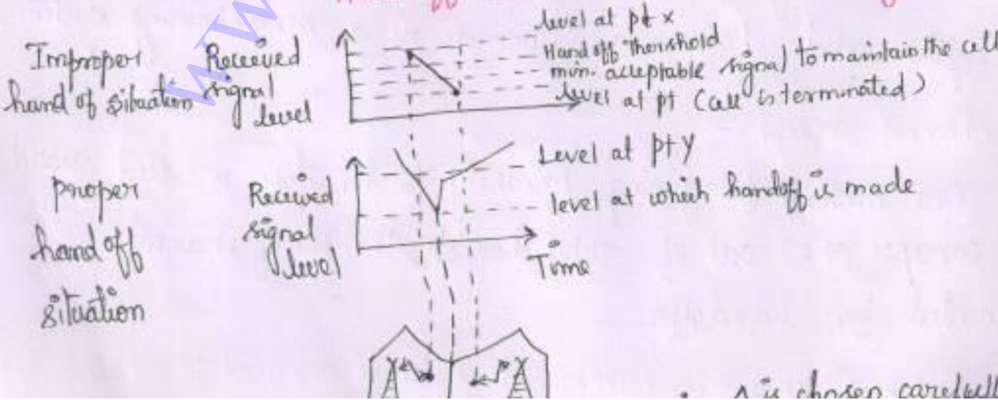
$$\boxed{\Delta = P_{r\,handoff} - P_{r\,minimum\ usable}}$$

$\Delta \rightarrow$ cannot be too large or too small

→ if Δ is too large, unnecessary handoffs which burden the MSC may occur

→ if Δ is too small, there may be insufficient time to complete a handoff before a call is lost due to weak signal conditions

Hand off Scenario at cell boundary

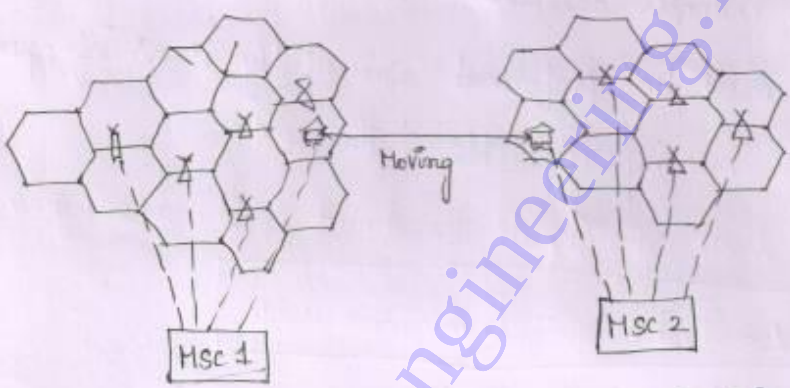


• Δ is chosen carefully to meet

Inter system handoff :-

If a mobile moves from one cellular system to a different cellular system controlled by a different MSC during the process of a call a handoff is made. This type of handoff is called "intersystem handoff"

Intersystem handoff



- There are so many problems occur in this type of handoff
- when a mobile moves from one cellular system to another the MSC cannot find another cell within its system to which it can transfer the call is in progress. and also a local call may become a long distance call as the mobile moves out of its home system and becomes a roamer in a neighbouring system.

Guard channel concept :-

One method for giving priority to handoff is called guard channel concept (i.e) out of total channels, the No. of channels are allocated for handoffs.

Disadvantage - Only fewer number of channels are used to Originating

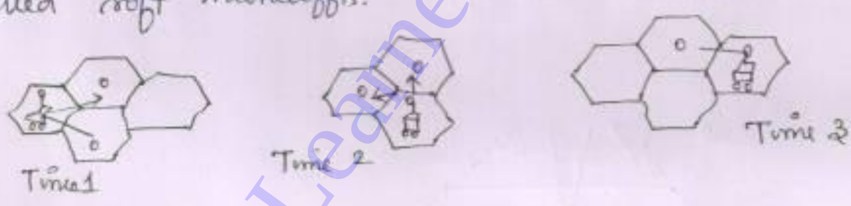
Advantage:- Efficient spectrum utilization when dynamic channel assignment strategies ⑦

Dwell Time: The time over which a call may be maintained within a cell without handoff is called dwell time.

Mobile Assisted Handoff (MAHO): In this type every mobile station measures the received power from surrounding base station & continuously reports the results of these measurements to the serving base station. MAHO is particularly suited for micro cellular environment.

There are 2 types of handoff's are used.

(1) Soft Handoff:- Mobile communicates with 2 or more cells at the same time & find which one is a strongest signal base station then it automatically transfers the cell to that base station is called soft handoffs.



Hard Handoff:-

If the MSC monitors the strongest signal base station & transfer the call to that base station than it is called hard handoff

- Another problem in the handoff is "cell dragging"

- It results from the pedestrain users that providing a strong signal to the base station. This type of situation occurs in the urban environment where there is a LOS radio path b/w the subscriber & the base station.

- When the user travels away from the base station at a very low speed, the average signal strength does not decay rapidly.

- Even when the user has travelled well beyond the designed range of the cell, the received signal at the base station may be above the handoff threshold. thus a handoff may not be made. This creates a potential interference & traffic management problem this is called "cell dragging"

To solve this problem handoff thersholds & radio coverage parameters must be adjusted carefully.

Interference and system capacity :-

Interference on voice channel causes cross talk where the subscribers hears interference in the background due to an undesired transmission.

There are 2 types of interference

1) Adjacent channel interference

2) co-channel interference

The co-channel reuse ratio is defined by

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

if Q is smaller then cluster size N also small that is providing larger capacity

Let i_0 be the number of co-channel interfering cells then the signal to interference ratio for a mobile receiver is

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

$S \rightarrow$ signal power for desired base station

$I_i \rightarrow$ Interference power caused by interfering co-channel cell BS.

The average received power P_r at a distance d from the transmitting antenna is given by

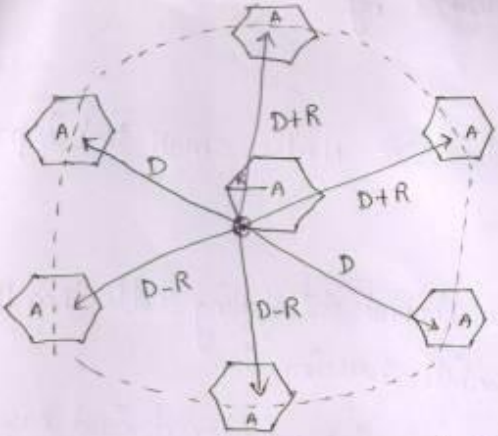
$$P_r = P_0 (d/d_0)^{-n} \quad \text{or} \quad P_{r(dBm)} = P_{0(dBm)} - 10n \log(d/d_0)$$

when the transmit power of each BS is equal & the path loss exponent is the same throughout the coverage area, S/I for a mobile can be approximated as

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

- considering only the first layer of interfering cells. if all the interfering base stations are equidistant from the desired base station & if this distance is equal to the distance D b/w cell centers then

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



- using an exact cell geometry layout it can be shown for a 7 cell cluster with the mobile unit at the cell boundary.

- The Mobile is a distance D-R from the 2 nearest co-channel interfering cells & is exactly D+R/2. D, D-R/2 & D+R from the other interfering cells.

If $n = 4$

$$S/I = \frac{R^{-n}}{\sum_{j=1}^{i_0} (D_i)^{-n}} = \frac{R^{-4}}{2(D-R)^{-4} + 2(D+R)^{-4} + 2D^{-4}}$$

rewritten in terms of Q

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

$$Q = \sqrt{3 \times N} = \sqrt{3 \times 7} = \sqrt{21} = 4.6$$

$N = 7$, $Q = 4.6$ & the worst case signal to interference ratio

$S/I \approx 49.56$

Trunking and Grade of Service :-

The 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 year.

The busy hours for cellular radio system typically occur during rush hours b/w 4 Pm & 6pm on a Thursday (or) Friday evening

Common terms used in trunking theory.

(i) Setup Time: The time required to allocated a trunked radio channel to a requesting user.

(ii) Blocked call: call which cannot be completed at a time of request, due to congestion. Also referred to as a lost call.

(iii) Traffic Intensity:- Measure of channel time utilization. which is the average channel occupancy measured is Erlangs denoted by A.

(iv) Request Rate:- The average number of call requests / unit time denoted by λ/sec. The traffic Intensity $A_u = \lambda H$

λ - average no. of call request / time for each user

H - average duration of a cell

U - No. of users

For a system containing U users & an unspecified no. of channels. The total offered traffic intensity A is given as

$$A = U A_u$$

This is determined by "Erlang formula" is given by

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

- If no channels are immediately available the call is delayed & the probability that the delayed call is forced to wait more than "t" s. The GoS of a trunked system where blocked calls are delayed is given by

$$P_r(\text{delay}>t) = P_r(\text{delay}>0)\, P_r(\text{delay}>t \mid \text{delay}>0)$$

$$= P_r(\text{delay}>0)\exp(-(c-A)t/H)$$

The average delay D for all calls in a queued system is given by

$$D = P_r(\text{delay}>0) = H/c-A$$

$H/c-A$ is the average delay for those calls which are queued.

Trunking Efficiency :-

It is a measure of the number of users which can be offered a particular GoS with a particular configuration of fixed channels.

Load: Traffic Intensity across the entire trunked radio system, measured in Erlangs.

$\therefore$ P_r (at old cell boundary) $\propto P_{t_1} R^{-n}$ &

P_r (at new cell boundary) $\propto P_{t_2} (R/2)^{-n}$

P_{t_1} & P_{t_2} - Transmit powers of the larger & smaller cell base stations

$n \rightarrow$ path loss exponent

if $n=4$ $\quad P_{t_2} = P_{t_1}/16$

$\therefore$ the cells are splitted then the transmitter power will be reduced, so that handoff's occur less frequently.

cell Sectoring

- Sectoring is the another way to ↑ the capacity of the system. with sectorization S/R is improved due to usage of directional antennas.

- Directional antennas are used to radiate power in one particular direction. The area covered by each directional antenna is called as sector so that this method is called as cell sectoring

- By using directional antennas a given cell will receive interference & transmit with only a fraction of the available co-channel cells.

- The factor by which the co-channel interference is reduced depends on the amount of sectoring used. A cell is normally partitioned into three 120° sectors or six 60° sectors.

120° Sectoring

60° sectoring

- when sectoring is employed the channels used in particulars cell are broken down into sectored groups and are used only within a particular sector. From that interference also ↓ & capacity ↑.

how 120° sectoring reduces interference from co-channel cells

- consider the interference experienced by a mobile located in the right most sector in the centre cell labelled "5". It will experience interference on the forward link from only two sectors out of six.

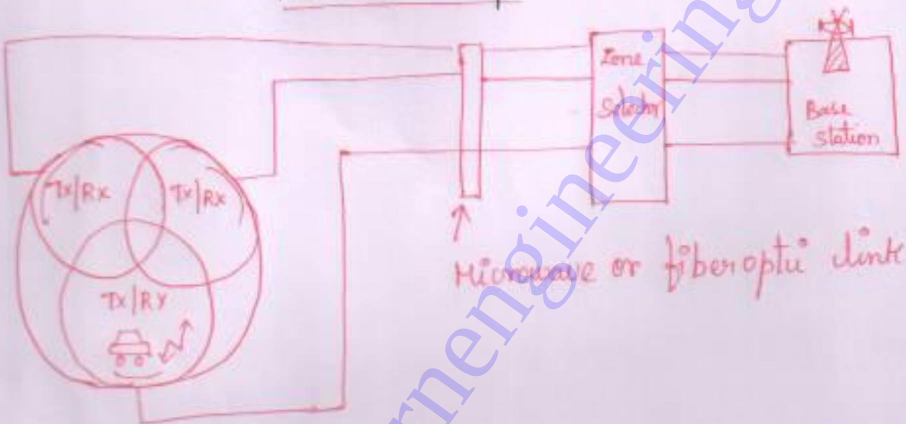
Micro cell Zone Approach

- The ↑ no. of handoffs required when sectoring is employed results in an ↑ load on the switching & control link elements of the mobile system.

- In order to avoid the above problem micro cell zone approach is used.

- In this method cells are divided into no. of zones & each no. of zones consists of individual Transceiver.
- so that interference also reduced and capacity is ↑ because ↓ co-channel interference improves the signal quality & also leads to an ↑ in capacity without degradation in trunking efficiency caused by sectoring

Microcell concept



- when the mobile station moves from one zone to next zone then the base station switches the channel to the new zone site. A given channel is active in only one zone. The channels are distributed in time & space are also reused in co-channel cells. This techniques particularly useful in highways or urban traffic corridors.

Advantages:-

- 1. The co-channel interference in the cellular system is reduced ... central BS is replaced by Several low power transmitters on the edge of the cell.

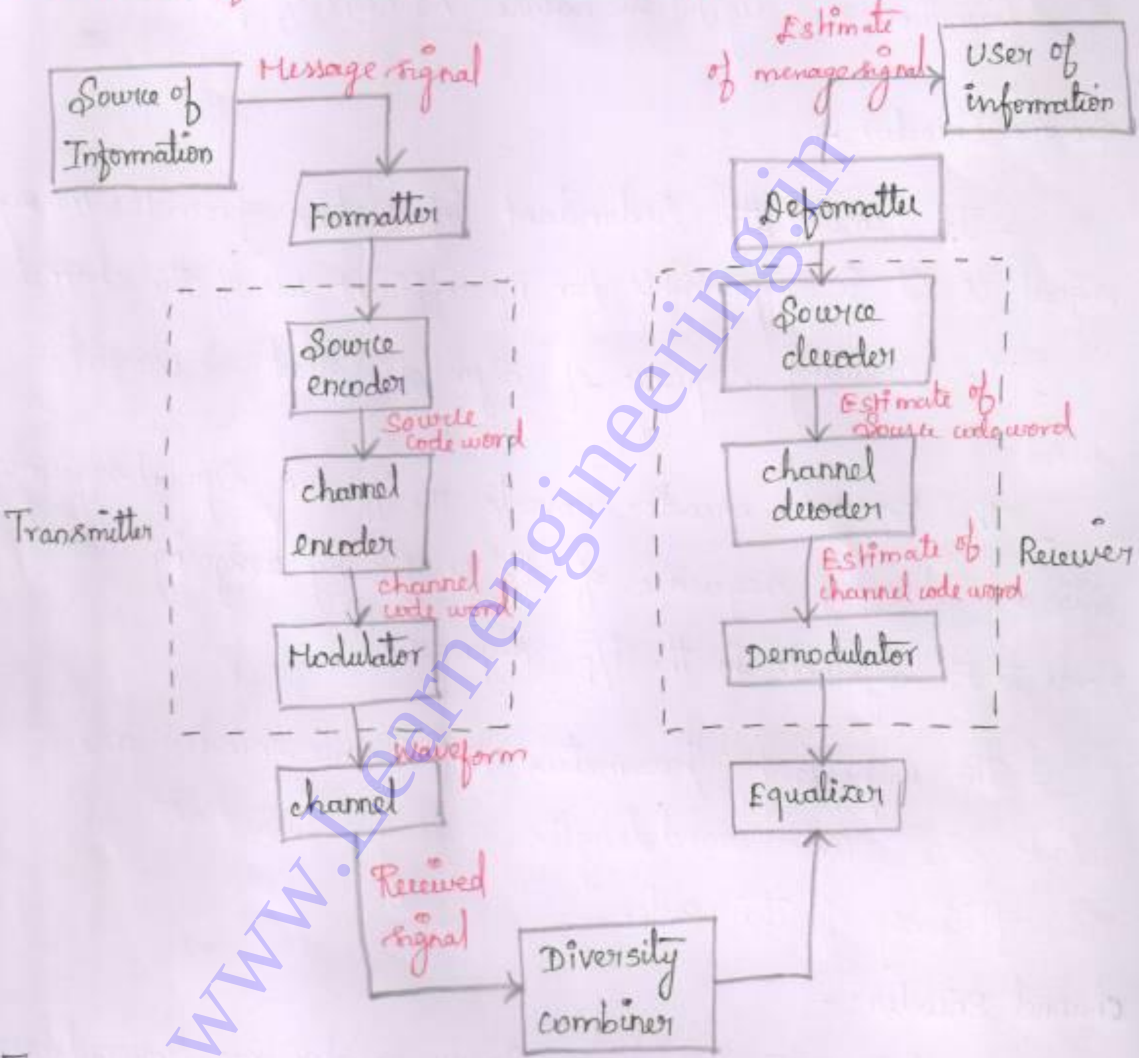
2. The signal quality is ↑ due to reduction in co-channel interference

3. channel capacity is ↑ without the degradation in trunking efficiency caused by sectoring.

Unit - III

Digital Signaling for fading channels

Structure of Wireless communication link



Formatter:-

- Practically source of information is in the form of analog signal Ex: Voice, electrical signal, Audio & video signal.
- Digital system requires only digital signal. so it's

- Some operation have to perform to convert the analog signal in to digital signal in discrete form.

- This operation is performed by Formatter. If the source of information is digital in nature, no need of formatter

Source Encoder:-

- It removes the redundant information from the message signal & it is responsible for the efficient use of the channel.

- The resulting sequence of symbol called Source code Word

- The source encoder converts the i/p i.e symbol sequence into a binary sequence of 0's & 1's by assigning code words to the symbols in the i/p sequence

- The important parameters of source encoder are block size, code word lengths, average data rate & the efficiency of the coder

Channel Encoder:-

- The data streams from source encoder are processed here.

- Link coding takes place here.

- It produces a new sequence of symbols called channel code word.

- Error control is accomplished by the channel coding operation that consists of systematically adding extra bits to the o/p of the Source coder.

- This extra bit do not convey any information but helps the receiver to detect &/or correct some of the errors in the information bearing bits.

There are 2 methods of channel coding

Block coding :-

The encoder takes a block of k information bits from the Source encoder & adds r error control bits where r is dependent on k & error control capabilities desired

convolution coding

The information bearing message stream is encoded in a continuous fashion by continuously interleaving information & error control ~~bits~~

Band-pass processor :-

To reduce noise over channel, pulse shaping operation must be employed in band-pass processor by using special filters.

Band-pass Modulator :-

- The modulator converts the i/p bit stream into an electrical waveform suitable for transmission over the communication channel.

- The main purpose of this band-pass modulator is to match the digital signal to high frequency analog signals.

- Modulator can be effectively used to minimize the effects of channel noise to match the frequency spectrum of transmitted signal with channel characteristic.

For ex: a band of 20 MHz width at a frequency of approximately 2.4 GHz

channel :

- The channel provides the electrical connection b/w the source & destination.

- The different channels are : pair of wires, coaxial cable, optical fibre. Radio channel, satellite channel or combination of any of these.

- The communication channels have only finite B.W, non-ideal frequency response the signal often suffers amplitude & phase distortion as it travels over the channels.

- The signal power decreases due to the attenuation of the

Diversity combiner

- It is used to combined received signal from one or more antennas

Equalizer: -

- It is used to reduce ISI & dispersion

Band pass Demodulator: -

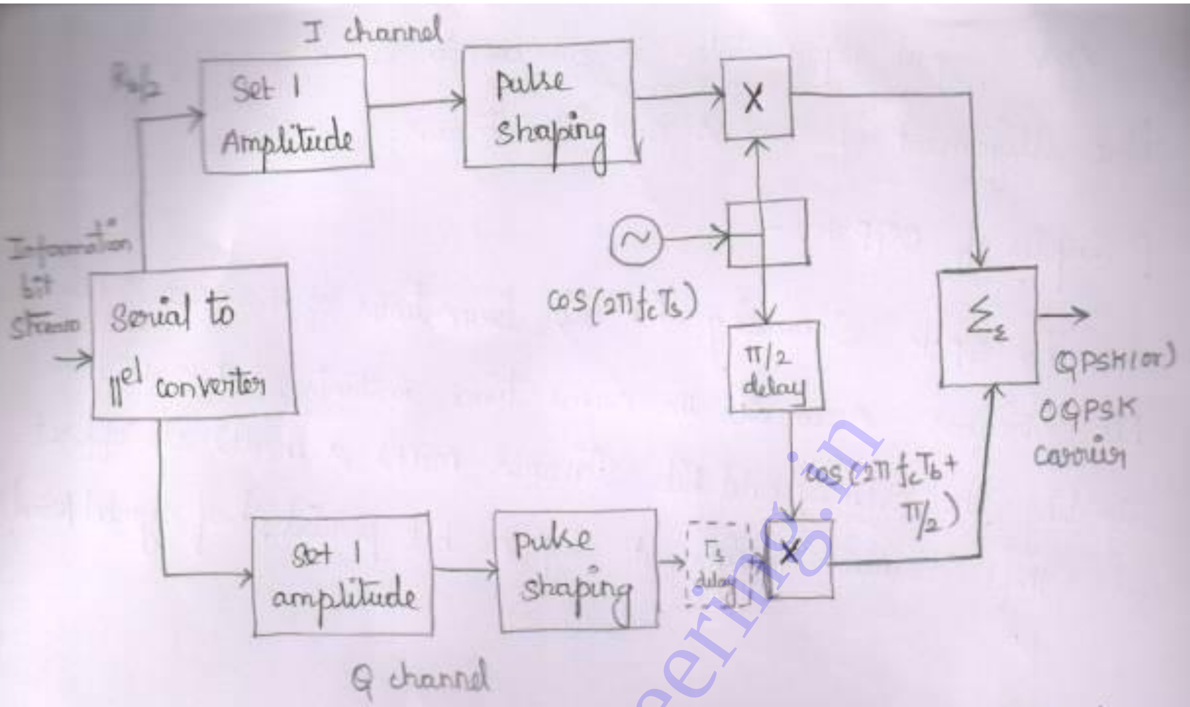
- The modulated Signal transmitted over channel, it may be affected by AWGN (Additive white Gaussian Noise)

- The extraction of the message from the information bearing waveform produced by the modulation is accomplished by the demodulator.

- The o/p of the demodulator is bit stream. It maps the extracted bit stream to estimate of Transmitted data sequence

Channel Decoder

Using channel encoding algorithm, channel decoder attempts to reconstruct the original waveform.



The result of this sample change is that phase shifts at any one time are limited & hence offset QPSK is more constant-envelope than straight QPSK.

Elimination of 180° phase transitions :-

→ Due to time alignment of $m_i(t)$ & $m_Q(t)$ in QPSK phase transitions occur only once every $T_S = 2T_b$ seconds.

→ In QPSK if there is a change in the value of both $m_i(t)$ & $m_Q(t)$ then there will be a maximum of phase shift of 180°

→ In OQPSK signalling bit transitions occur every T_b.

→ Since the transition instants of $m_i(t)$ & $m_Q(t)$ are offset at any given time only one of the 2 bit streams can change values. This implies that the maximum phase shift of the ... signal at any given time is limited to ±90°

→ Hence by switching phases more frequently OQPSK signaling eliminates 180° phase transitions.

Probability of Error

$$P_{e,\,QPSK} = \mathrm{erfc}\left(\sqrt{\frac{2E_b}{2N_0}}\right) = \mathrm{erfc}\left(\sqrt{\frac{E_b}{N_0}}\right)$$

Advantage of OQPSK:-

- OQPSK offers better performance in satellite applications
- It perform better than QPSK in presence of phase jitter

Application of OQPSK

1. In HP amplifiers & for certain satellite applications
2. Mobile communication systems

π/4 DIFFERENTIAL QPSK:-

The π/4 shifted QPSK modulation is a quadrature phase shift keying technique which offers a compromise b/w OQPSK & QPSK in terms of the allowed maximum phase transitions

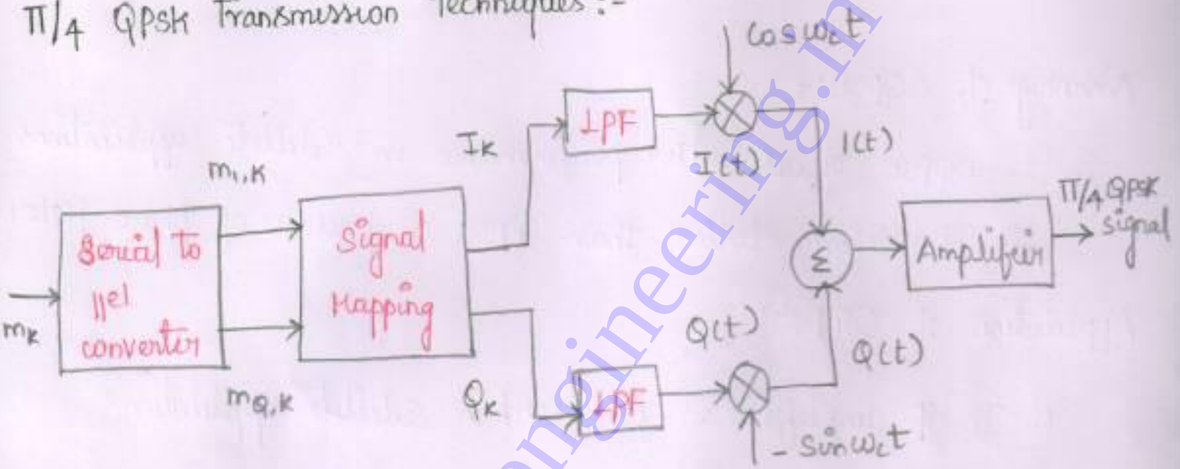
Principle:-

In π/4 QPSK the maximum phase change is limited to ±135° as compared to 180° for QPSK & 90° for OQPSK. Hence the band limited π/4 QPSK signal preserves the constant envelope ... than band limited QPSK but is more susceptible to envelope variations than OQPSK.

An extremely attractive feature of $\pi/4$ QPSK is that it can be non-coherently detected which greatly simplifies receiver design.

When differentially encoded $\pi/4$ QPSK is called $\pi/4$ DQPSK.

$\pi/4$ QPSK transmission techniques :-



Block Diagram

Serial to parallel converter

- The i/p bit stream is partitioned by a S/P converter
- It splits data streams into 2 ||el data streams as $m_{I,K}$ & $m_{Q,K}$
- Symbol rate equal to half that of the incoming bit rate

i.e $R_s = R_b/2$

Signal Mapping circuit

- The K^{th} inphase & quadrature pulses I_K & Q_K are produced at the O/P of the signal mapping circuit
- Time is in the following range $KT \le t \le (K+1)T$

- The K^{th} inphase & quadrature pulses can be determined from previous values I_{K-1} & Q_{K-1} as well as θ_K
- The θ_K is a function of ϕ_K which is a function of i/p symbols $m_{I}K$ & $m_{Q,K}$.
- I_K & Q_K represents rectangular pulses over one symbol duration having amplitudes given by

$$I_K = \cos\theta_K = I_{K-1}\cos\phi_K - Q_{K-1}\sin\phi_K$$

$$Q_K = \sin\theta_K = I_{K-1}\sin\phi_K - Q_{K-1}\sin\phi_K$$

where $\theta_K = \theta_{K-1} + \phi_K$ θ_K & θ_{K-1} are phases of the K^{th} & $(K-1)^{st}$ symbol.

Modulator :-

The inphase & quadrature bit streams I_K & Q_K are then separately modulated by 2 carrier which are in quadrature with one another to produce the $\pi/4$ QPSK waveform given by

$$S_{\pi/4\,QPSK}(t) = I(t)\cos(\omega_c t) - Q(t)\sin(\omega_c t)$$

$$I(t) = \sum_{K=0}^{N-1} I_K\, p\left(t - KT_s - \frac{T_s}{2}\right) = \sum_{K=0}^{N-1} \cos\theta_K\, p\left(t - KT_s - \frac{T_s}{2}\right)$$

$$Q(t) = \sum_{K=0}^{N-1} Q_K\, p\left(t - KT_s - \frac{T_s}{2}\right) = \sum_{K=0}^{N-1} \sin\theta_K\, p\left(t - KT_s - \frac{T_s}{2}\right)$$

where $p(t)$ is pulse shape & T_s - symbol period

- Both I_k & Q_k are passed through raised cosine roll off pulse shaping filters before modulation in order to reduce the B.W Occupancy.

- peak amplitude of the waveforms $I(t)$ & $Q(t)$ can take one of the 5 possible Values $0, +1, -1, +1/\sqrt{2}, -1/\sqrt{2}$

Summer:-

It adds 2 $I(t)$ & $Q(t)$ to give $\pi/4$ QPSK signal.

Amplifier:-

After summer the $\pi/4$ QPSK signal is then given into amplifier. The signal is amplified by non-linear amplifier with greater efficiency. The information in a $\pi/4$ QPSK signal is completely contained in the phase difference ϕ_k of the carrier b/w 2 adjacent symbols. Since the information is completely contained in the phase difference it is possible to use non-coherent differential detection even in the absence of differential encoding

$\pi/4$ QPSK Detection:-

Due to simple hardware implementations differential detection is often employed to demodulate $\pi/4$ QPSK signals. In an AWGN channel the BER performance of a differentially detected $\pi/4$ QPSK is about 3dB inferior to QPSK. while coherently detected $\pi/4$ QPSK has the same error performance as QPSK.

Types of Detection techniques

There are various types of detection techniques that are used for the detection of $\pi/4$ QPSK signals.

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

Minimum shift keying :-

MSK is a continuous phase modulation technique that offers advantages in performance & ease of implementation. In coherent detection of BFSK signal the phase information contained in the received signal is not fully exploited. It is used only in the receiver to the transmitter to provide synchronization of the receiver to the transmitter.

By the proper use of the phase it is possible to improve the noise performance the receiver significantly. This is done through the use of MSK.

Principle

It is possible to control the phase changes so that the rate of change of phase is uniform & that the correct I-Q location is reached just at the end of the data period. That system is known as Minimum shift keying since the minimum rate of

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A constant rate of change of phase corresponds to a constant frequency. MSK can thus also be seen as a special case of FSK where the change of frequency is such that a 90° phase change is obtained after one data period.

MSK signals must satisfy following 2 conditions

1. The modulating pulse must be symmetrical about $t + \frac{T_b}{2}$ & 0
2. The modulated carrier has to have a constant envelope.

CPFSK Signal Representation :-

consider a continuous phase frequency shift keying signal given by the equation

$$S(t) = \begin{cases} \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_1 t + \theta(0)) & \text{for symbol } 1 \\ \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_2 t + \theta(0)) & \text{for symbol } 0 \end{cases}$$

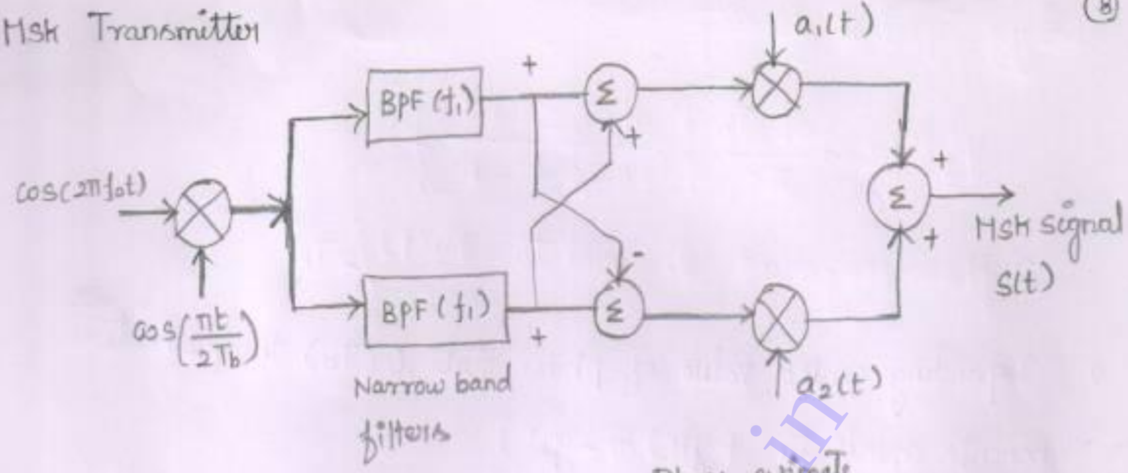
$s(t)$ – The region $0 \le t \le T_b$ E_b – Transmitted signal energy per bit

T_b – bit duration

Signal space characteristic

Transmitted symbol	Phase state $\theta(0)$	$\theta(T_b)$	coordination of message point S_1	S_2
Message point m_1 symbol 0	0	$-\pi/2$	$\sqrt{E_b}$	$\sqrt{E_b}$
Message point m_2 symbol 1	π	$-\pi/2$	$-\sqrt{E_b}$	$\sqrt{E_b}$
Message point m_3 symbol 0	π	$\pi/2$	$-\sqrt{E_b}$	$-\sqrt{E_b}$
Message point m_4 symbol 1	0	$\pi/2$	$\sqrt{E_b}$	$-\sqrt{E_b}$

Msk Transmitter



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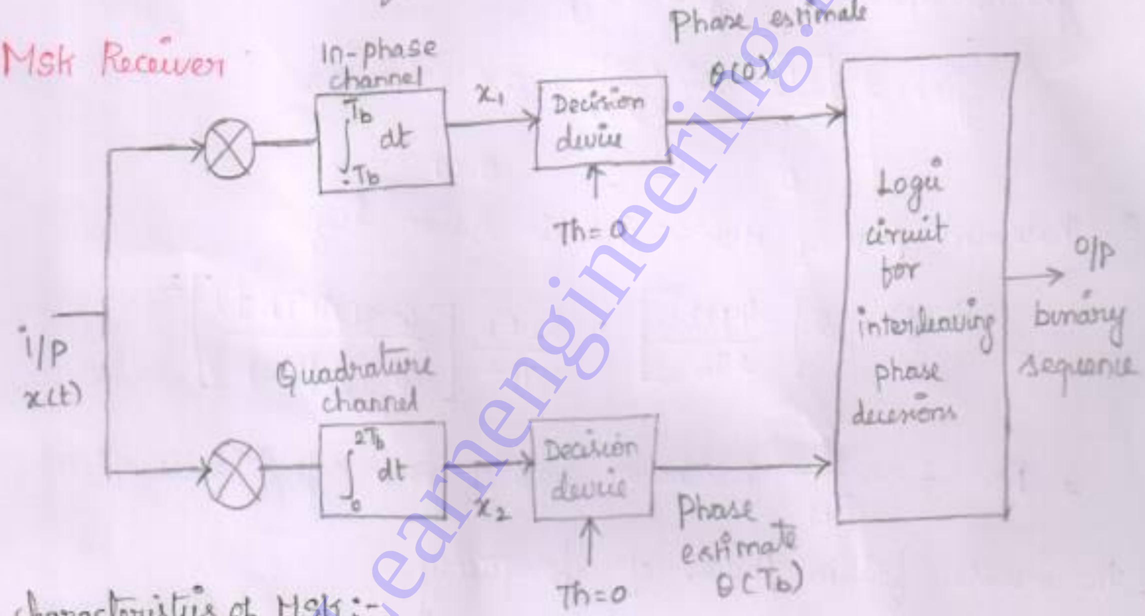

Characteristics of MSK :-

Power spectra of MSK

symbols transmitted during different times are satistically independent

1. Depending on the value of phase state $\theta(0)$ in phase components equals $+g(t)$ or $-g(t)$ where

$$g(t) = \begin{cases} \sqrt{\frac{2E_b}{T_b}} \cos\left(\frac{\pi t}{2T_b}\right) & -T_b \le t \le T_b \\ 0 & o.w \end{cases}$$

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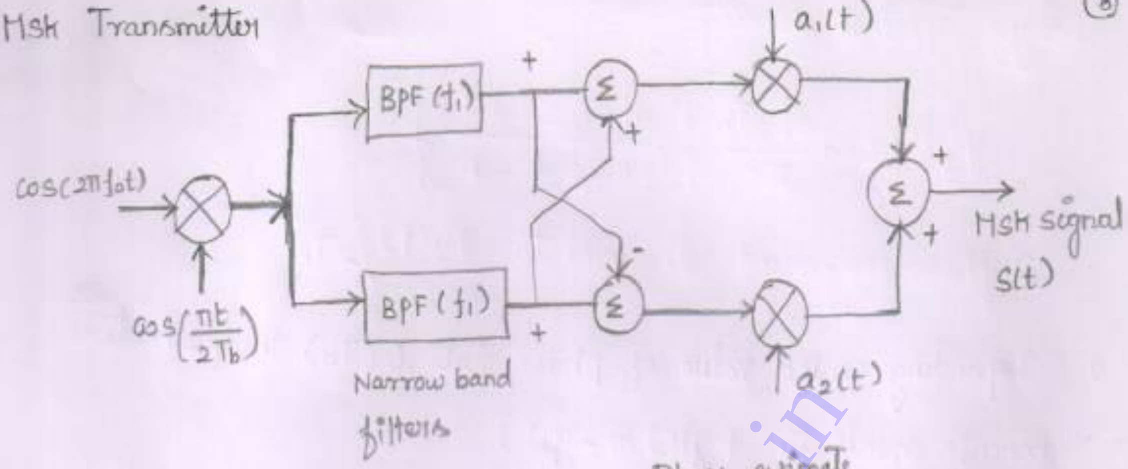

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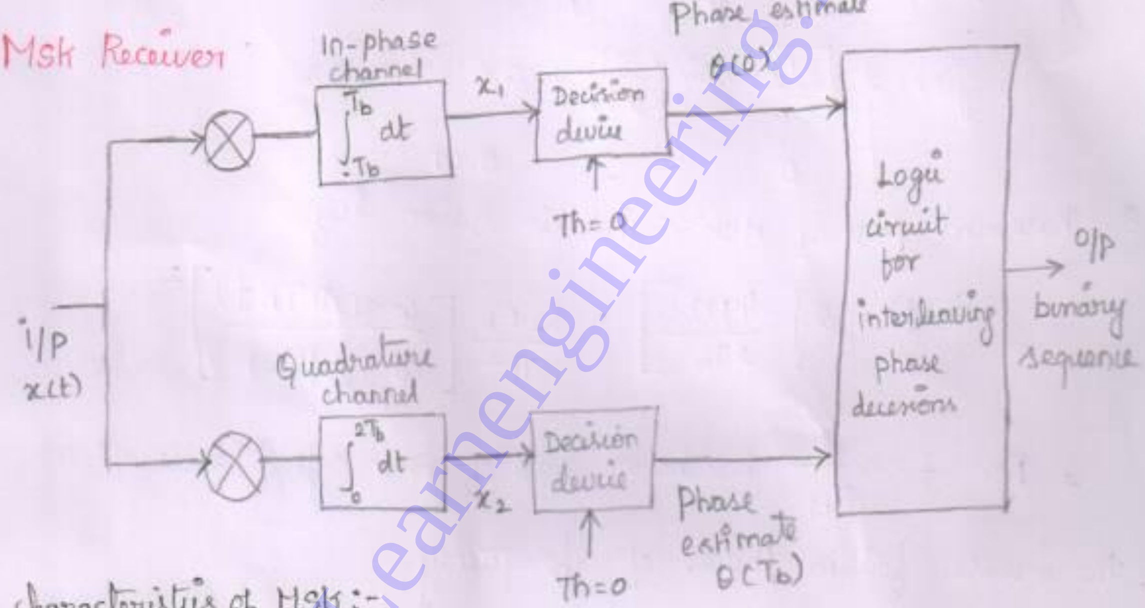

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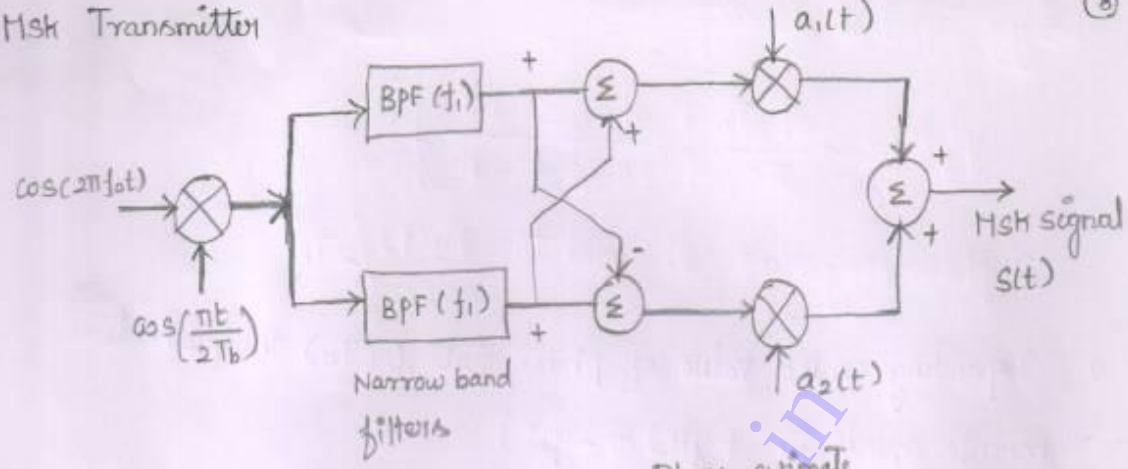

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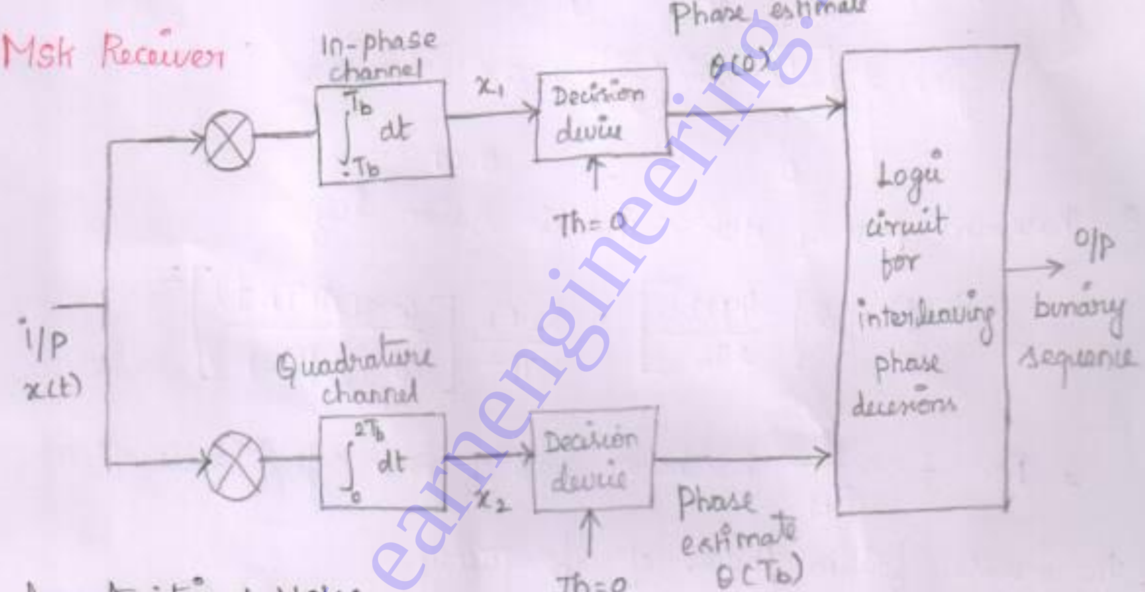

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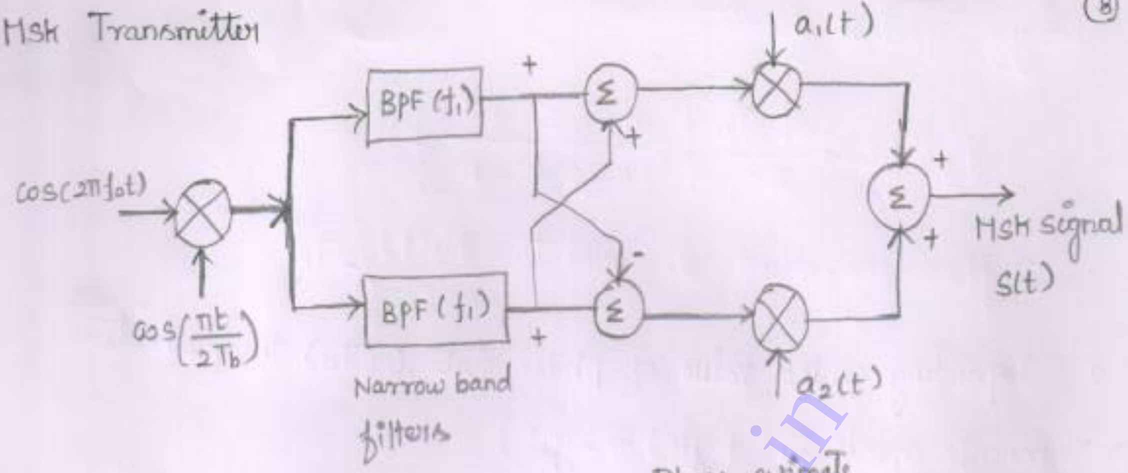

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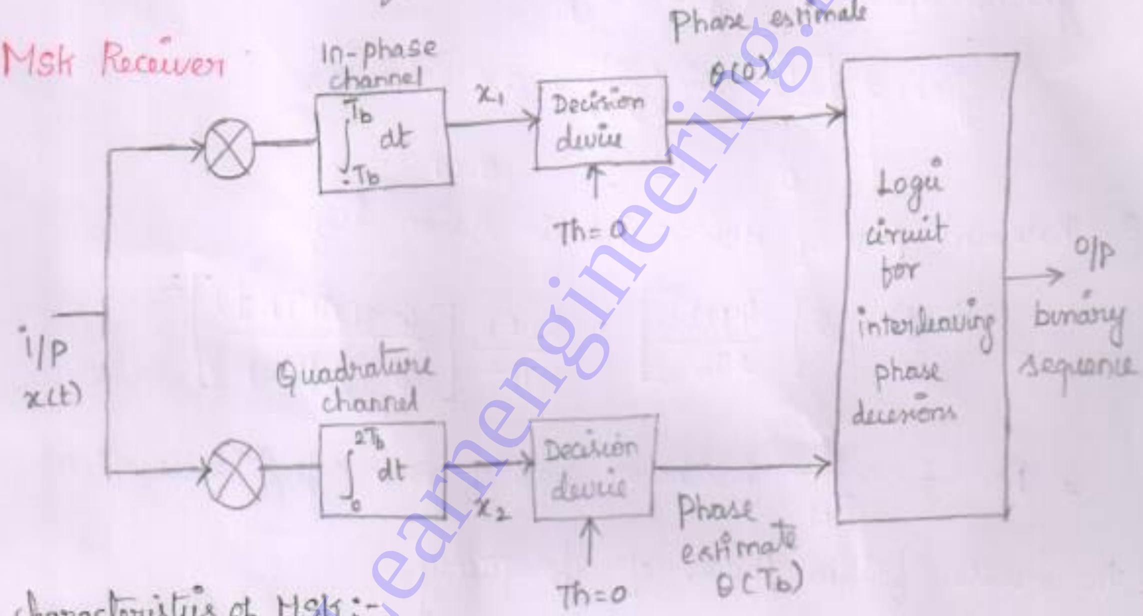

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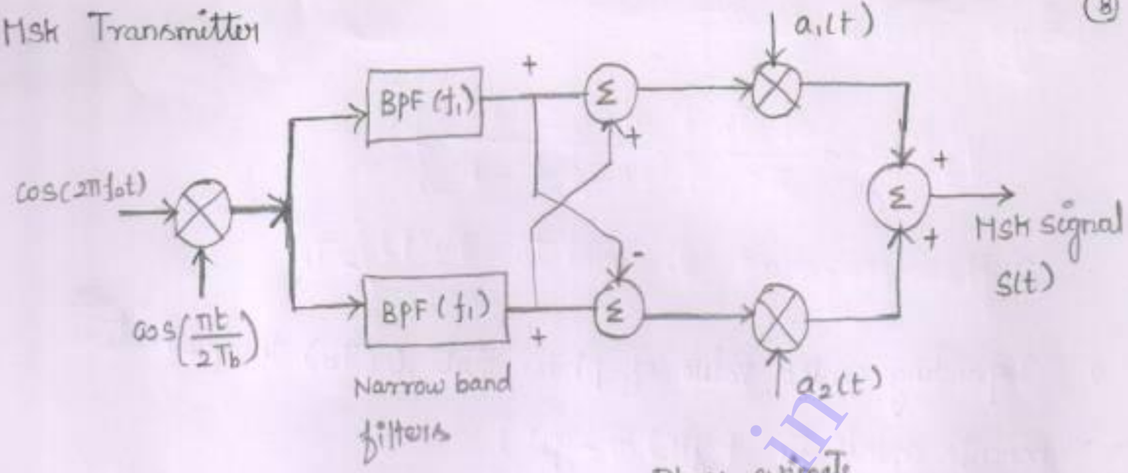

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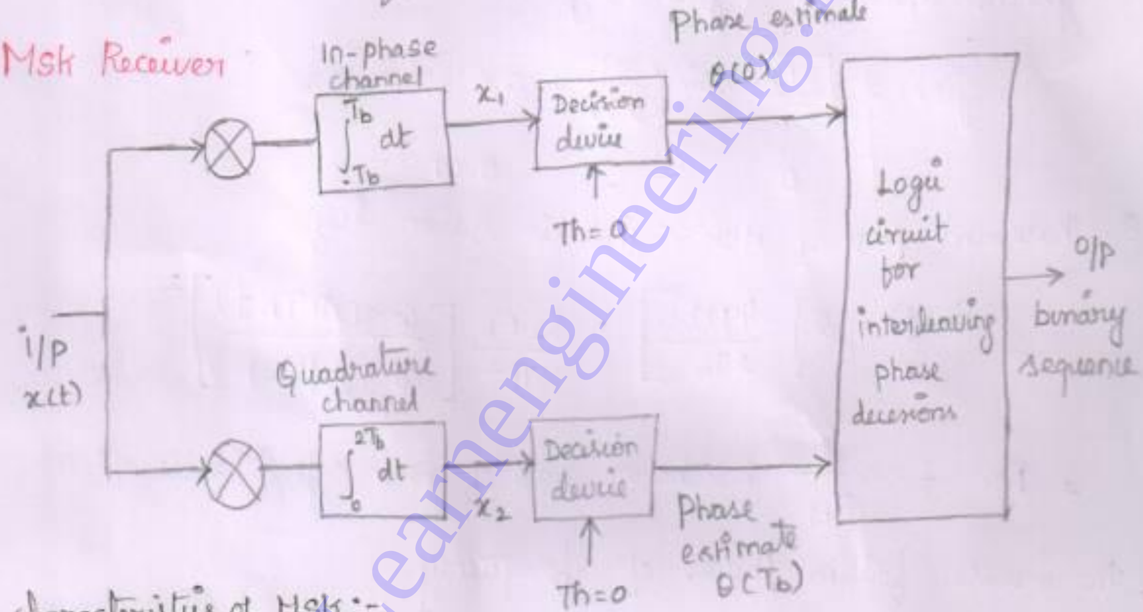

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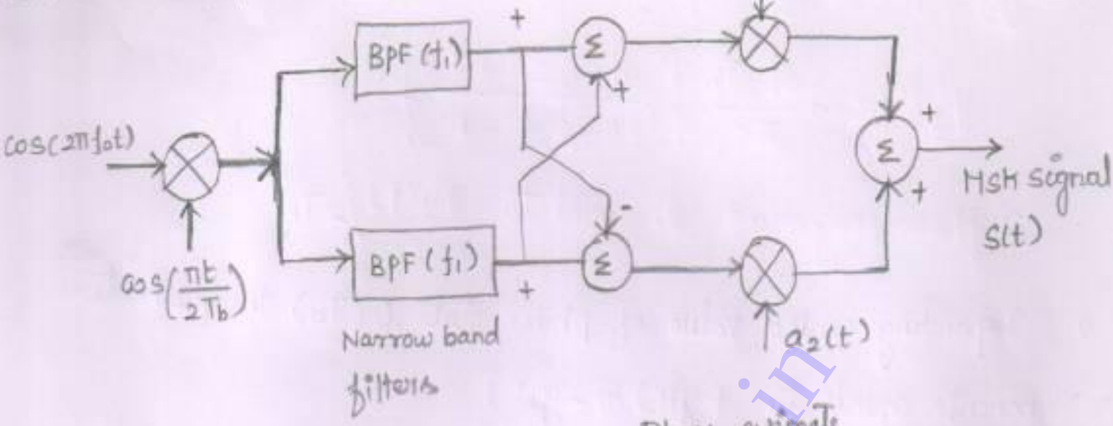

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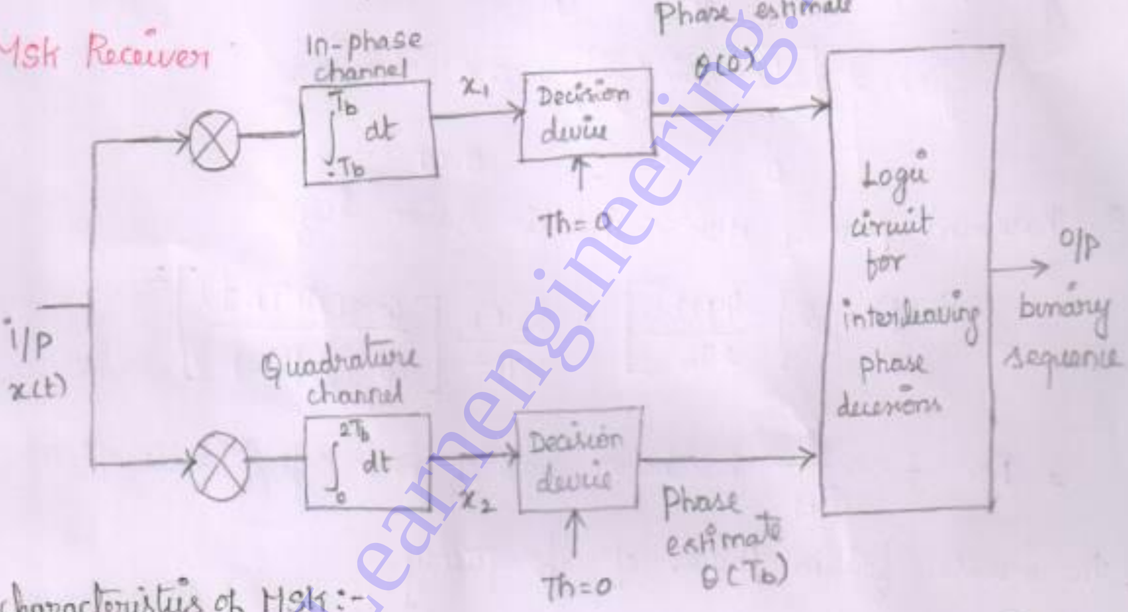

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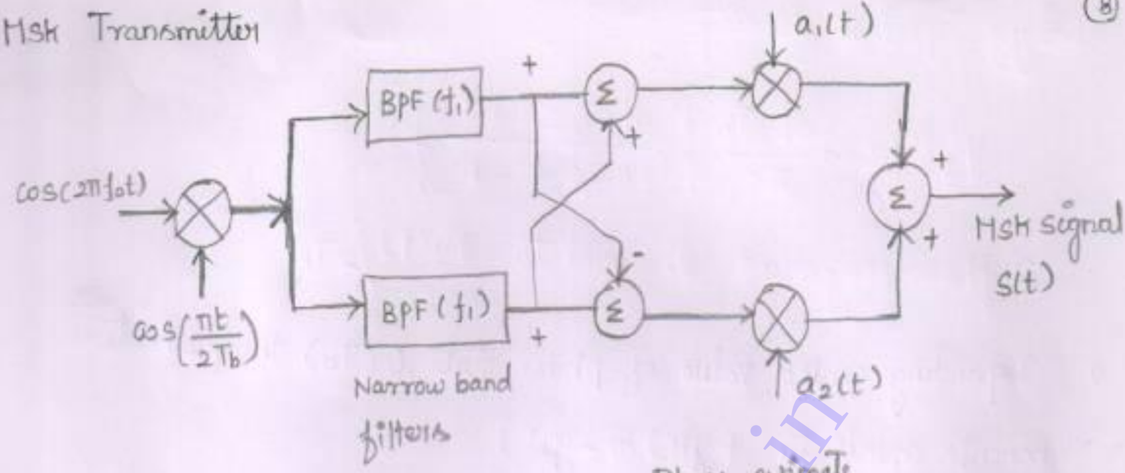

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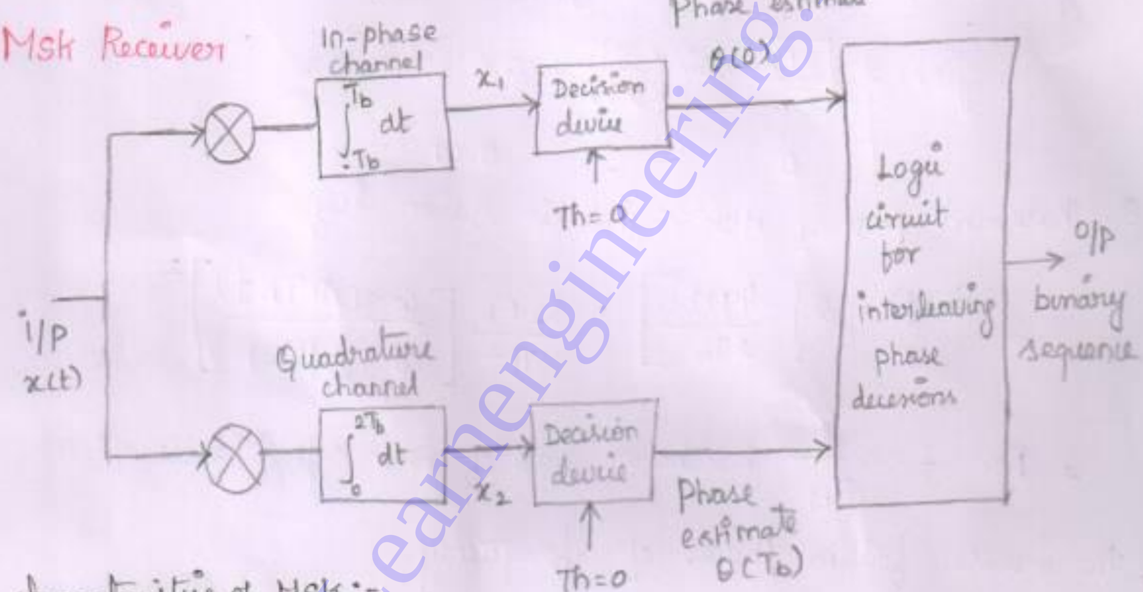

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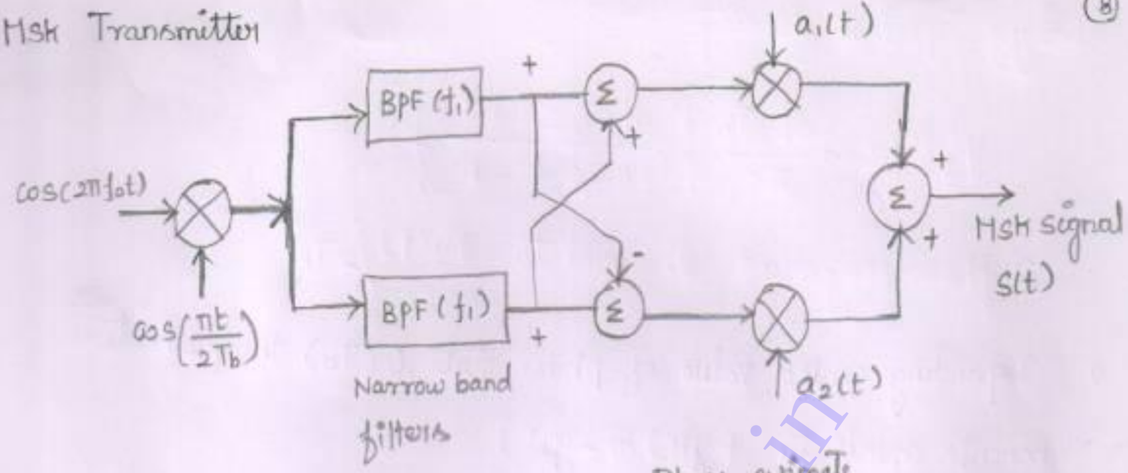

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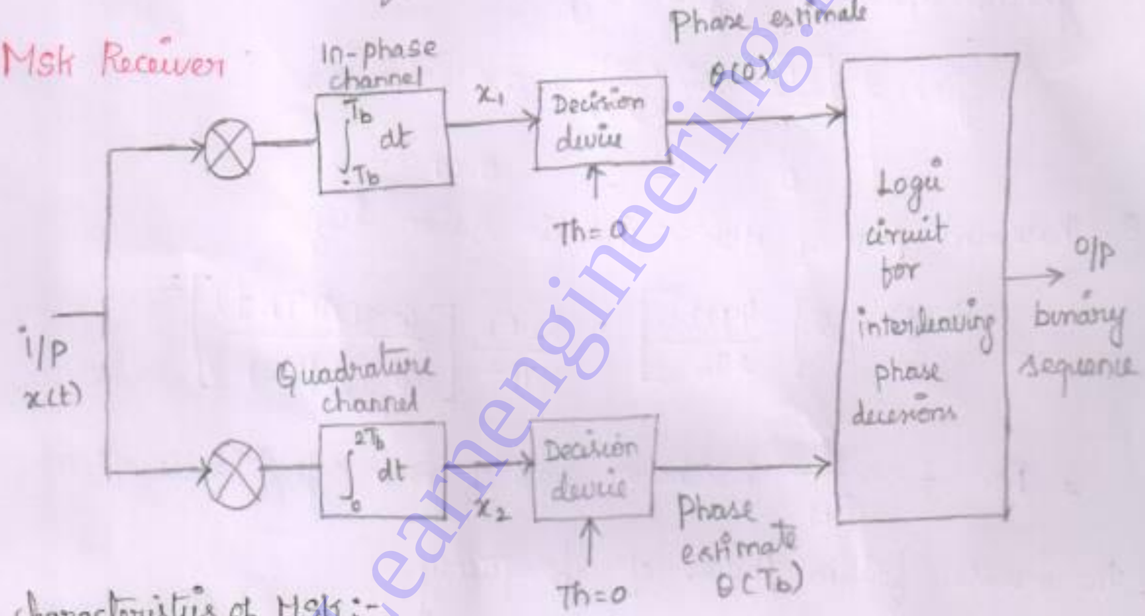

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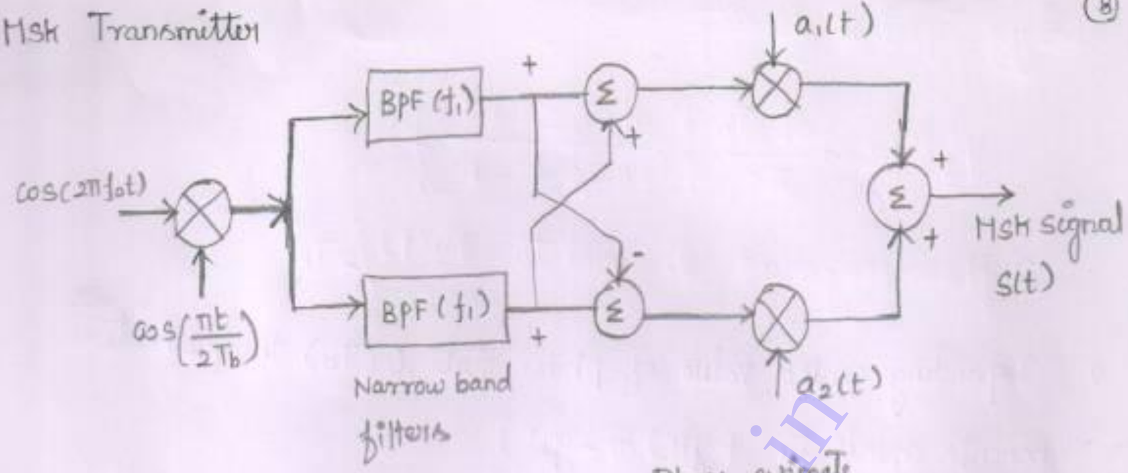

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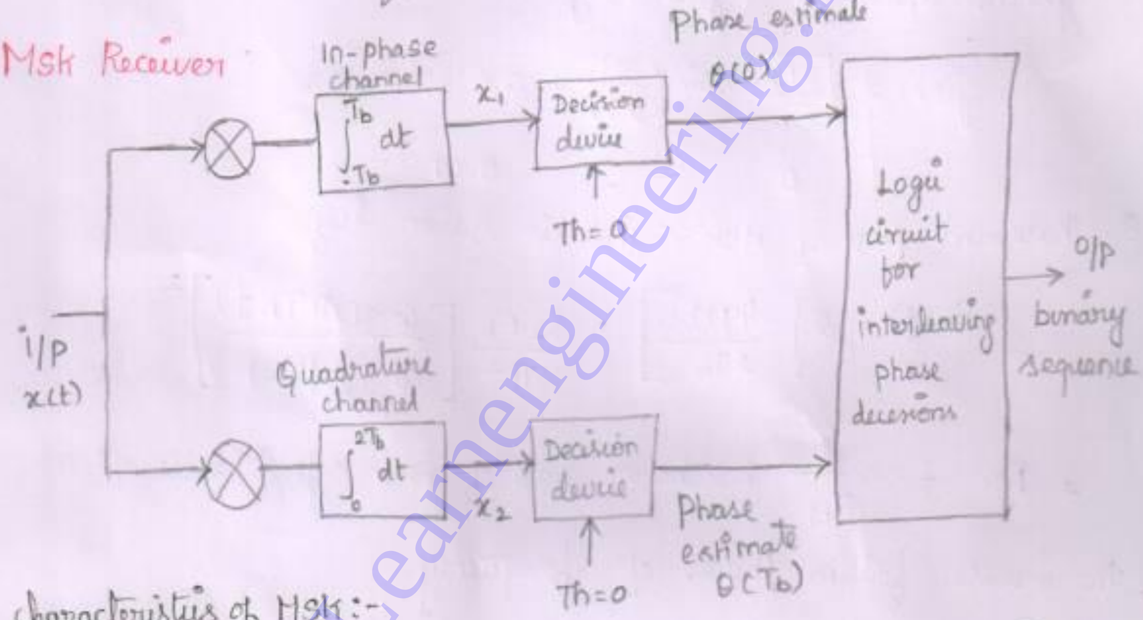

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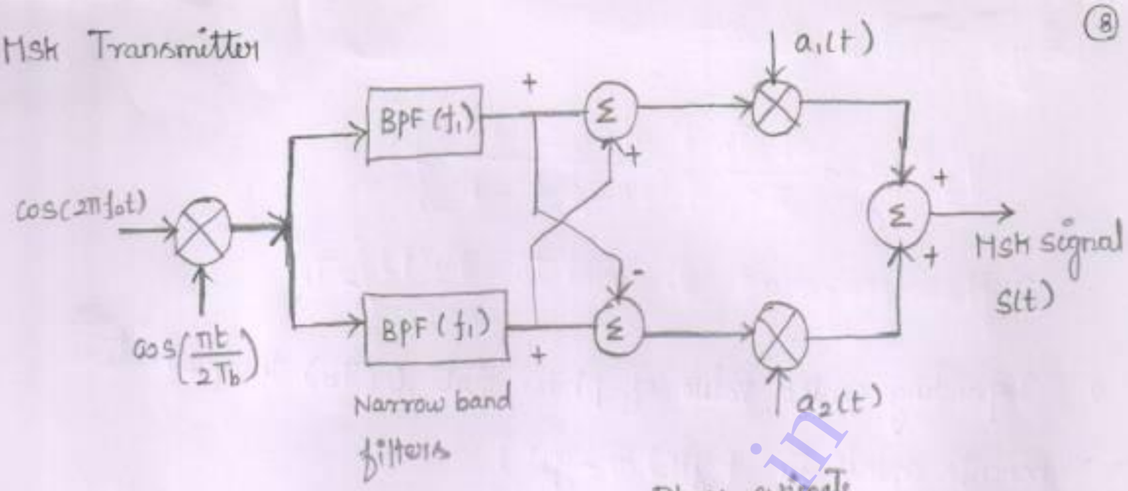

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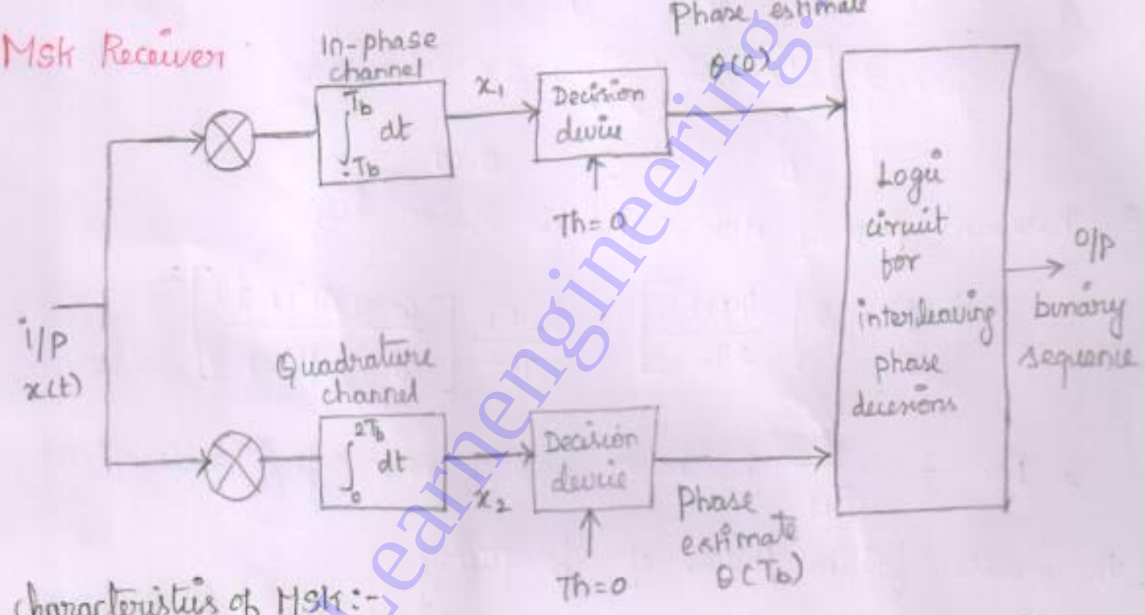

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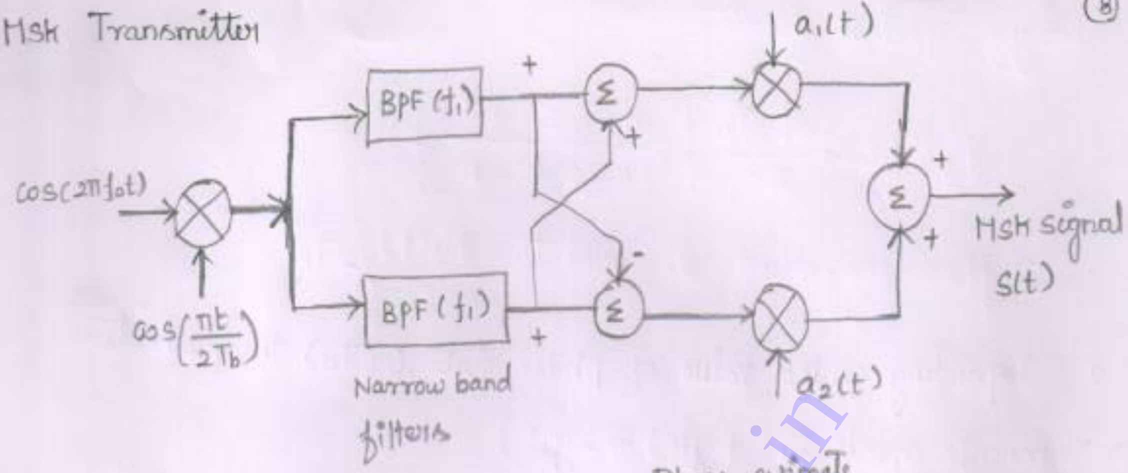

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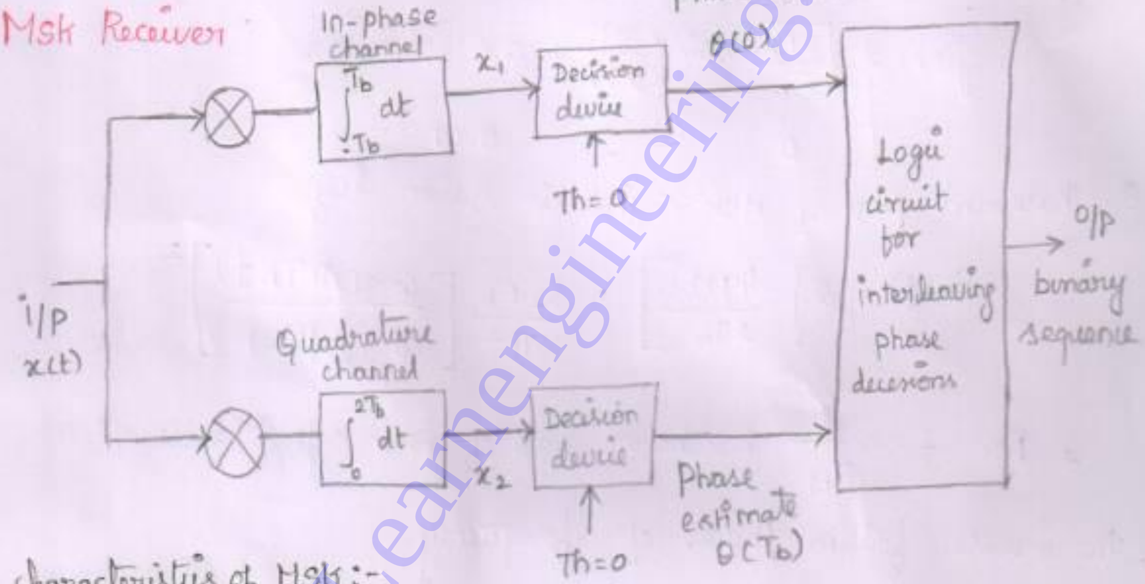

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Energy spectral density

$$\psi_g(t) = \frac{32 E_b T_b}{\pi^2}\left[\frac{\cos(2\pi T_b f)}{16 T_b^2 f^2 - 1}\right]^2$$

Inphase component is equal to $\psi_g(f)/2T_b$

2. Depending on the value of phase state $\theta(T_b)$ the quadrature components equals to $+g(t)$ or $-g(t)$

$$g(t) = \begin{cases} \sqrt{\frac{2E_b}{T_b}} \sin\left(\frac{\pi t}{2T_b}\right) & 0 \le t \le 2T_b \\ 0 & o.w \end{cases}$$

3. Baseband PSD of MSK signal is given by

$$S_B(f) = 2\left[\frac{\psi_g(f)}{2T_b}\right] = \frac{32 E_b}{\pi^2}\left[\frac{\cos(2\pi T_b f)}{16 T_b^2 f^2 - 1}\right]^2$$

* For $f \gg \frac{1}{T_b}$ Baseband PSD of MSK signal falls off as the inverse fourth power of frequency

* For QPSK baseband PSD fall off as inverse square of frequency

* MSK does not produce as much interference outside the signal band of interest of QPSK.

Advantages of MSK

1. No amplitude Variations
2. Continuous phase in all case
3. Less interchannel interference
4. Good spectral efficiency

Disadvantages of MSK

1. phase filter may exist due to incorrect synchronisation
2. B.W requirements is higher than QPSK
3. Not Suitable for multiuser communication due to slow decay of MSK
4. B-W required is higher than QPSK

Gaussian Minimum shift keying

It is a simple modulation scheme that may be taken as a derivation of MSK.

Term Gaussian refers to shaping. In GMSK the sidelobe levels of the spectrum are futhur reduced by passing a non-return to zero data waveforms through a pre modulation Gaussian Pulse shaping filters

GMSK must satisfy the following properties

1. Frequency response with narrow B.W & sharp cut off characteristic
2. Impulse response with relatively low overshoot.

Probability of Error

For AWGN the received signal is given by

$$x(t) = s(t) + w(t)$$

$s(t)$ - Tx signal $w(t)$ - sample function WGN with 0 mean & power spectral density $N_0/2$

For optimum detection of $\theta(0)$, received signal $x(t)$

$$x_1 = \int_{-T_b}^{T_b} x(t)\,\theta_1(t)\,dt$$

$$-T_b \le t \le T_b$$

$$x_1 = S_1 + W_1$$

$$S_1 = \int_{-T_b}^{T_b} s(t)\,\phi_1(t)\,dt$$

If $x_1 > \theta$; Receiver choose the estimate $\hat{\theta}_{(0)} = 0$

If " " " $\hat{\theta}_{(0)} = \pi$

$$P_e = \frac{1}{2}\,\mathrm{erfc}\left(\frac{\sqrt{E_b}}{N_0}\right) = Q\sqrt{\frac{2E_f}{\eta}}$$

$$= Q\sqrt{\frac{2d_{12}^2}{2\eta}}$$

E_b - bit Energy E_f - Rectangular pulse energy

d_{12} - Distance b/w signalling points

η = PSD

Features of GMSK

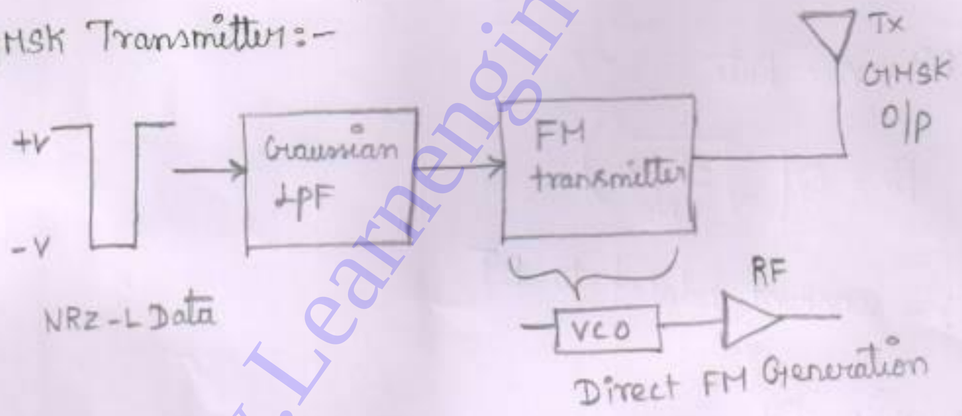
Premodulation Gaussian filtering introduces ISI in the transmitted signal but it can be shown that the degradation is not serve if the 3dB B.W bit duration product of filter is greater than 0.5.

It has good power efficiency & constant envelope properties

Spectrum of GMSK

↓ BT Product the out of band response ↓ but On the other hand irreducible error rate of the LPF for ISI ↑. ∴ a compromise b/w these 2 is required.

GMSK Transmitter:-



Impulse response can be expressed as follows

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

Transfer function of GMSK can be expressed as

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

$$\alpha = \frac{\sqrt{\ln 2}}{\sqrt{2}B} = 0.5887/B$$

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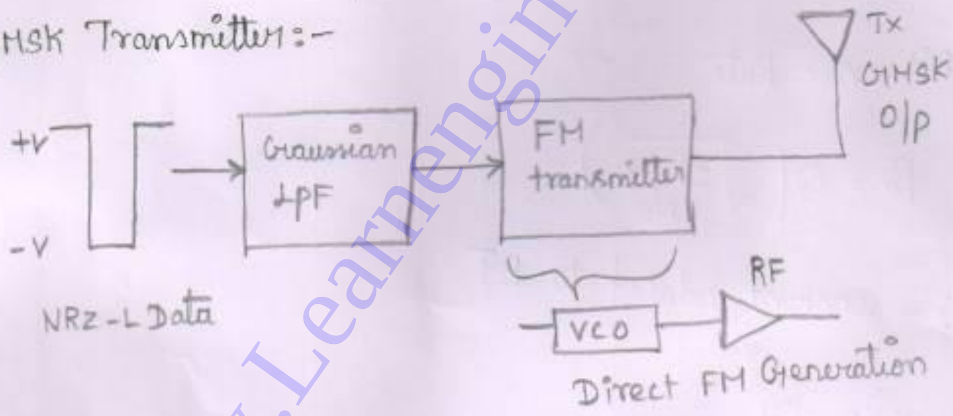

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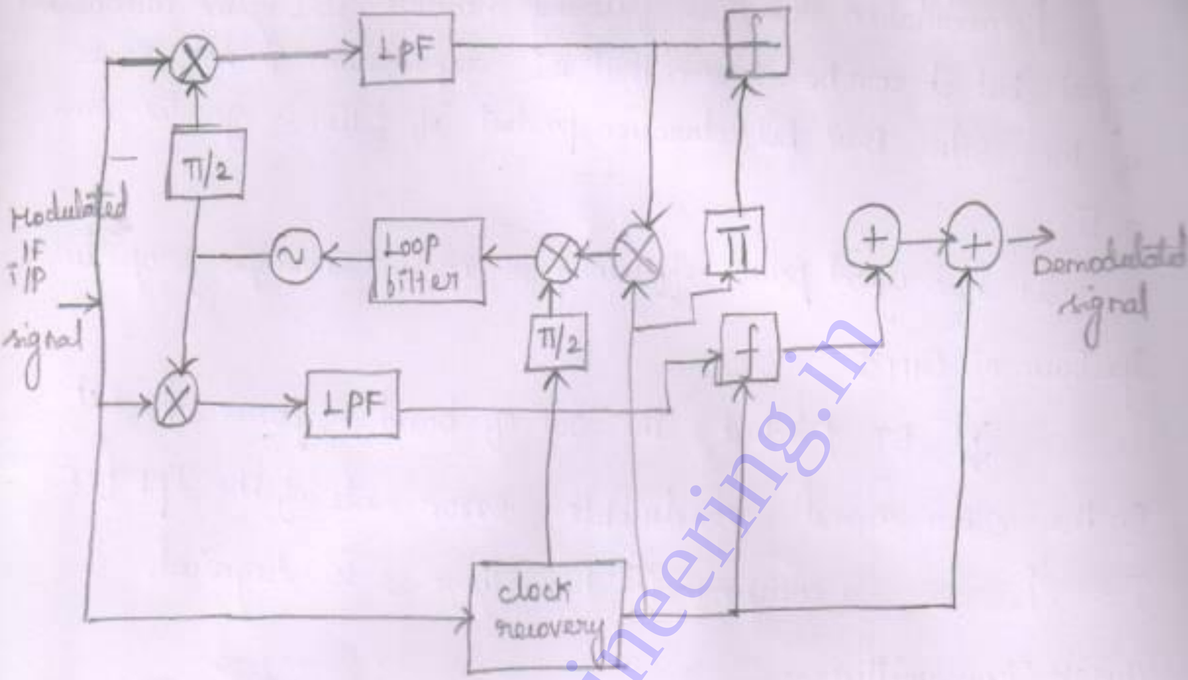


GMSK Bit error Rate

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

γ – constant related to BT

γ is given by

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

Error performance in fading channels:- ⑪

During propagation signal strength reduces is called as fading

Type of signal fading depends on

1. Nature of transmitted signal
2. characteristics of channel.

Slow, flat fading channels change much slower than the applied modulation. Flat fading channels cause a gain variation in the transmitted signal $s(t)$. Then the received signal $r(t)$ can be derived as

$$r(t) = \alpha(t)\exp(-j\theta(t))\,s(t) + n(t) \quad 0 \le t \le T$$

$g(t)$ - Gain of the channel $\theta(t)$ - phase shift of the channel

$n(t)$ - Additive white gaussian Noise

The probability of error

$$P_e = \int_0^\infty P_e(x)\,p(x)\,dx$$

$$p(x) = \frac{1}{\Gamma}\exp\left(-\frac{x}{\Gamma}\right) \quad x > 0$$

$\Gamma = \frac{E_b}{N_0}\alpha^2$ is the average value of the signal to noise ratio

$\alpha^2 = 1$ for unity gain fading channel

Digital Modulation in frequency selective mobile channels

Frequency selective fading caused by multipath time delay spread causes ISI which results in an irreducible BER floor for mobile systems.

The irreducible error floor in a frequency selective channel is primarily caused by the errors due to the ISI which interferes with the signal component at the receiver sampling instants. This occurs when

a) The main signal component is removed through multipath cancellations

b) A non-zero value of C_i causes ISI

c) The sampling time of a receiver is shifted as a result of delay spread

Normalized RMS delay spread $d = \frac{\sigma_\tau}{T_s}$

Based on the results of simulations it is known that

* For small delay spreads, flat fading is the dominant causes of error bursts.

* For large delay spread, timing errors & ISI are the dominant error mechanism.

Error performance of π/4 DPSK in fading & Interference (12)

The performance of π/4 DQPSK in the mobile radio environment. They modeled the channel as a frequency selective 2 ray channel with AGIN & cochannel interference.

In a slow flat fading channel the multipath time dispersion & Doppler spread are negligible & errors are caused mainly by fades & co channel interference.

- If $C/I > 20dB$ the errors are primarily due to fading & interference has little effect
- If $C/I < 20dB$ interference dominates the link performance

comparison b/w P_e for smaller & larger of E_b/N_0

P_e for Smaller Value of $\frac{E_b}{N_0}$	P_e for larger Value of E_b/N_0
$P_{psk} = \frac{1}{2}\left(1-\left(\frac{\Gamma}{1+\Gamma}\right)^{0.5}\right)$ for (coherent binary psk)	$P_{psk} = \frac{1}{4\Gamma}$ for (coherent binary psk)
$P_{psk} = \frac{1}{2}\left(1-\left(\frac{\Gamma}{2+\Gamma}\right)^{0.5}\right)$ for (coherent binary FSK)	$P_{psk} = \frac{1}{2\Gamma}$ for (coherent binary FSK)
$P_{Dpsk} = \frac{1}{2(1+\Gamma)}$ for (DBPSK)	$P_{Dpsk} = \frac{1}{2\Gamma}$ for (DPSK)
$P_{NCSK} = \frac{1}{2+\Gamma}$ for (NC orthogonal Bpsk)	$P_{NCFSK} = \frac{1}{\Gamma}$ for (NC orthogonal BFSK)

ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING (OFDM)

OFDM is a modulation scheme suitable for high data rate transmission in delay dispersive environments. It converts a high rate data stream into a no. of low rate streams that are transmitted over ||el. narrowband channels that can be easily equalized.

Principle of OFDM

- OFDM splits a high data stream into N ||el streams, which are then transmitted by modulating N distict carriers.
- Symbol duration on each subcarrier thus becomes larger by a factor of N.
- In order for the receiver to be able to separate signals carried by different subcarriers they have to be orthogonal
- In OFDM a much narrower spacing of subcarriers can be achieved.

$$f_n = \frac{nW}{N}$$

Total available B.W

$$W = N/T_s$$

The subcarriers are mutually orthogonal

$$\int_{iT_s}^{(i+1)T_s} \exp(j2\pi f_k t)\exp(-2\pi i f_n t)\,dt = \delta_{nk}$$

Features of OFDM

1. No carrier guard bands
2. Controlled overlapping of bands
3. Maximum spectral efficiency
4. Easy implementation using FFT
5. Very sensitive time frequency synchronization

Cycle prefix

Necessity :-

Delay dispersion also leads to a loss of orthogonality b/w the subcarriers & thus to ISI. The negative effects can be eliminated by a special type of guard interval called the cyclic prefix

- When transmitting any data stream over a delay dispersive channel the arriving signal is the linear convolution of the transmitted signal with the channel impulse response.
- The cyclic prefix converts this linear convolution into a cyclical convolution.

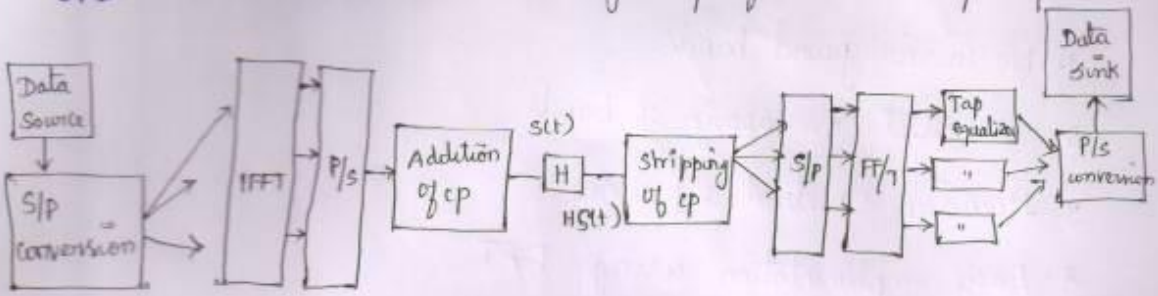
condition for cycle prefix is

$$s[-k] = s[N-k]$$

Transmission function

$$g_n(t) = \exp\left(j 2\pi n \frac{W}{N} t\right) \quad \text{for } -T_{cp} < t < \hat{T}_s$$

OFDM transmission chain with cyclic prefix & one-tap equalization



Advantages of cyclic prefix :-

1. It is beneficial in delay-dispersive channels
2. It is convert this linear convolution into a cyclical convolution
3. Equalization of the system thus becomes exceedingly simple
4. It is recovered the orthogonality of the sub carriers

Disadvantages of CP

1. SNR loss occur
2. Spectral efficiency is reduced.

Performance of OFDM:-

1. coding across the different tones
2. spreading the signal over all tones
3. Adaptive modulation

Advantage of OFDM

1. Flat fading per carrier
2. N long pulses
3. ISI is comparatively low
4. Easy to exploit frequency diversity
5. Use of dynamic signaling & adaptive coding

Drawbacks of OFDM

1. perfect synchronization is required which is difficult to achieve

2. peak to average power ratio
   - coding for PAR reduction
   - phase adjustments
   - correction by multiplicative function
   - correcting by additive function.

3. Inter carrier Interference

4. ISI

Parameters for designing an OFDM systems

1. Number of carriers
2. Guard time
3. Symbol duration
4. Subcarrier spacing
5. Modulation type per subcarrier
6. Type of forward error correction coding

Application

- Used in HS transmission in mobile communication
- Used in WLAN
- Digital audio Broadcasting
- Digital Video Broadcasting
- Wireless local area Networks

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Peak to average power Ratio

The peak amplitude of the emitted signal can be considerably higher than the average amplitude.

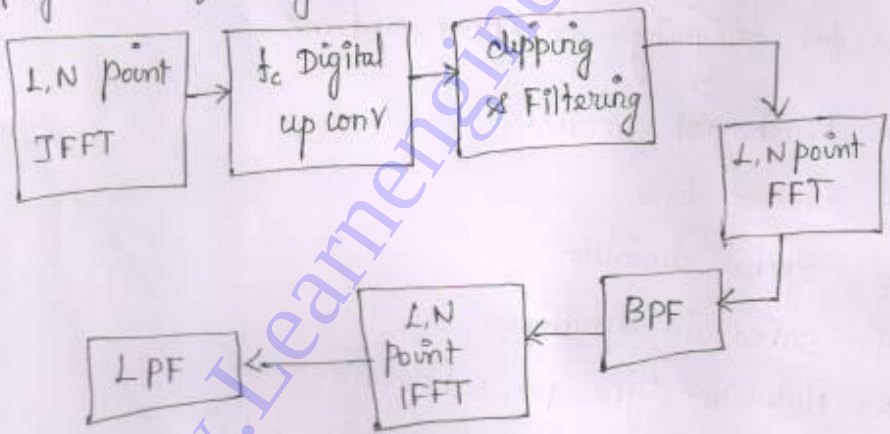
$$PAPR(x(T)) = \frac{\max_{0 \le t \le T_s} \left[|x(t)|^2 \right]}{P_{av}}$$

PAPR Reduction Techniques

a) Distortion Based Techniques

b) Non-distortion Techniques

Clipping and filtering



Companding :-

It is a composite word formed by combining compressing & expanding. It is used to reduce peak to average ratio.

Multipath Mitigation Techniques

EQUALISATION:-

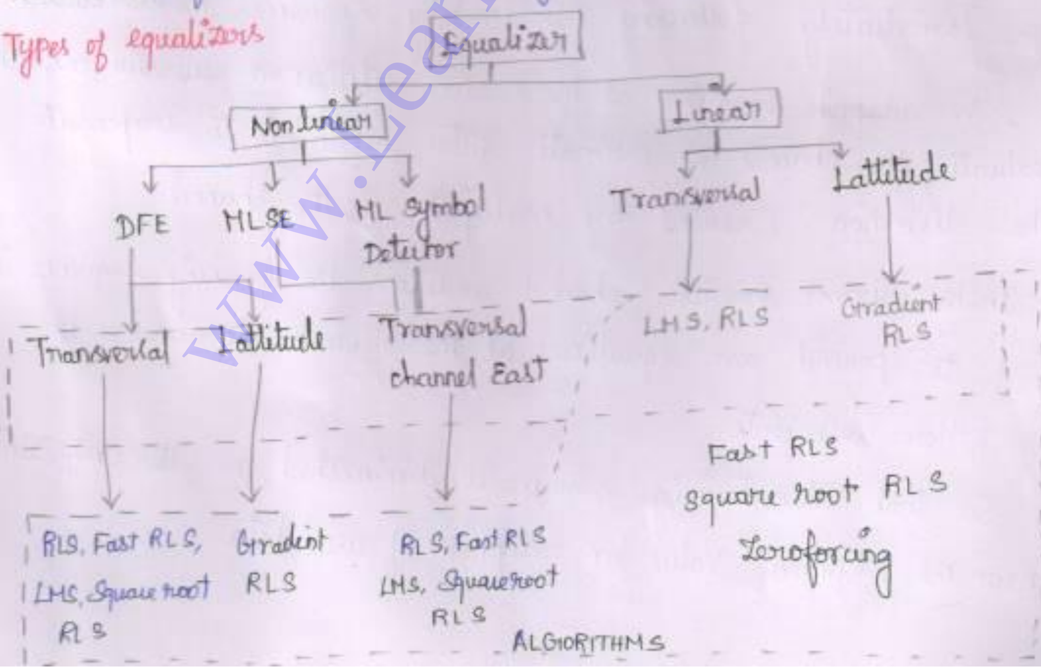
Three techniques which can be used independently or in tandem to improve received signal quality & link performance over small scale times & distances.

The techniques are

1) Equalization
2) Diversity
3) channel coding

Equalization can be used for any signal processing operation that minimizes ISI. In radio channels a variety of adaptive equalizers can be used to cancel interference while providing diversity.

Types of equalizers



Adaptive equalizers:-

- Equalizer are used to track the time varying characteristic of the mobile channel & it is called adaptive equalizers Since the mobile fading channel is random & time varying modes of an adaptive equalizer

The operation modes of an adaptive equalizer is

(i) training (ii) tracking

- First a fixed length training sequence is sent by the transmitter then the receivers equalizer may adapt to a proper setting for minimum bit error rate detection.

- The training sequence is a pseudorandom binary signal or a fixed prescribed bit pattern

- Immediately following this training sequences the user data is sent

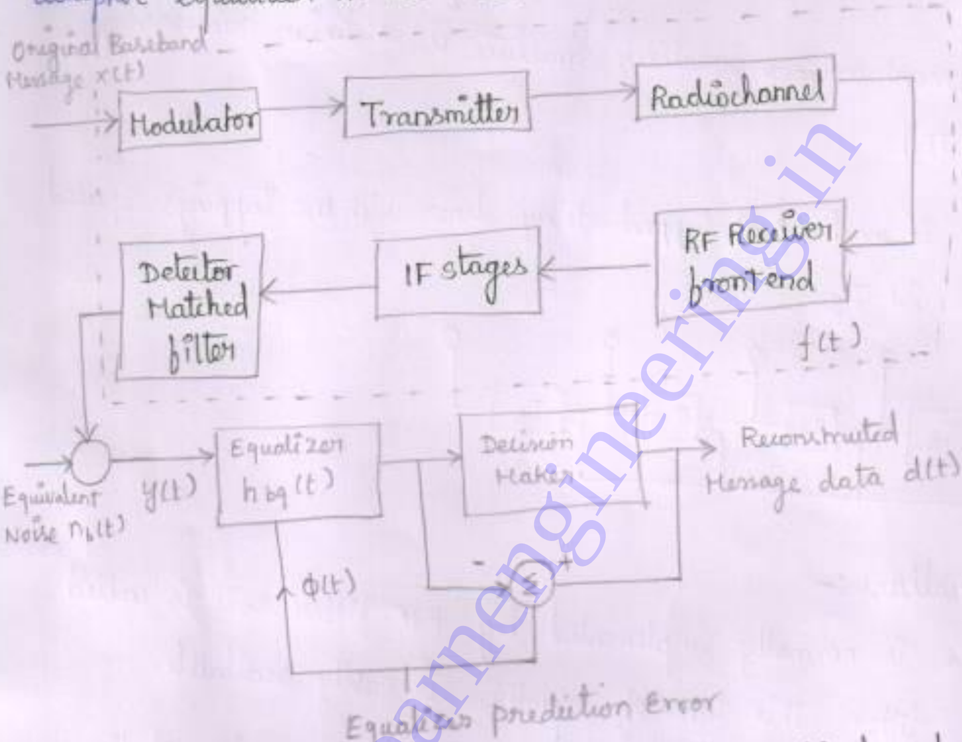
- The adaptive equalizer at the receiver utilizes a recursive algorithm to evaluate the channel & estimate filter coefficients to compensate for the distortion created by multipath in the channel.

- In the worst possible channel conditions the training sequence is designed to permit an equalizer at the receiver to acquire the proper filter coefficients

- So that when training sequence is diminished the filter coefficients are near the optimal value for reception of user data

- As user data are received the adaptive algorithm of the equalizer tracks the changing channel.

- The block diagram of a communication system with an adaptive equalizer in the receiver



f(t) = combined IR of transmitter multipath radio channel & receiver RF/IF

The equalization techniques can be broadly classified into 2 types based upon the o/p of an adaptive equalizer.

1) Linear Equalizer
2) Non linear Equalizer

Linear Equalization :-

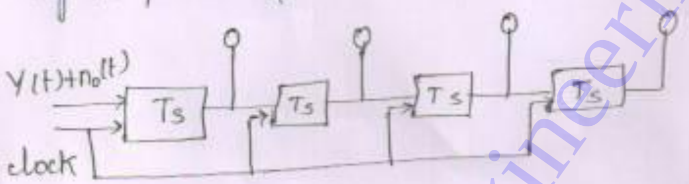
If the output d(t) is not used in the feedback path to adapt the equalizer then the type equalization is called linear equalization

Non-linear equalization :-

If d(t) is feedback to change the subsequent o/ps of the equalization then the type of equalization is called non-linear equalization

- The most common equalizer structure is a linear transverse equalizer (LTE).

A LTE is made up of tapped delay lines with the tappings spaced a symbol period T_s

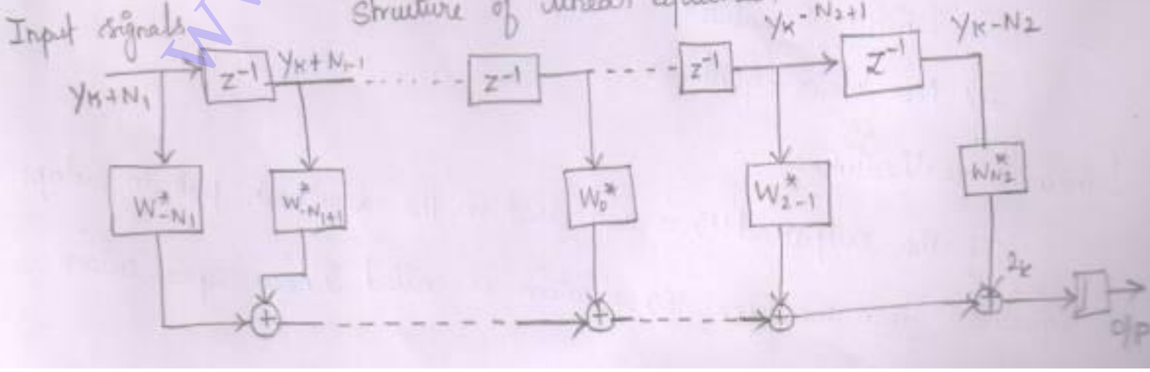


Linear Equalizers :-

It is normally implemented by FIR filter so it is called transversal filter. This type of equalizer is easily available

In this type of equalizer the present & past values of the received signals are linearly weighted by the filter coefficients & summed to produce the o/p d_k

Structure of linear equalizer



From the o/p of this linear equalizer before a threshold direction is

$$\hat{d}_k = \sum_{n=-N_1}^{N_2} (W_n^*)\, y_{k-n}$$

Where

W_n^* - Complex filter coefficients (or) tap weights

$\hat{d}_k$ - o/p at time index k

y_i - The input received signal at time t_0+iT

t_0 - Equalizer starting time

$N = N_1+N_2+1$ is no. of taps

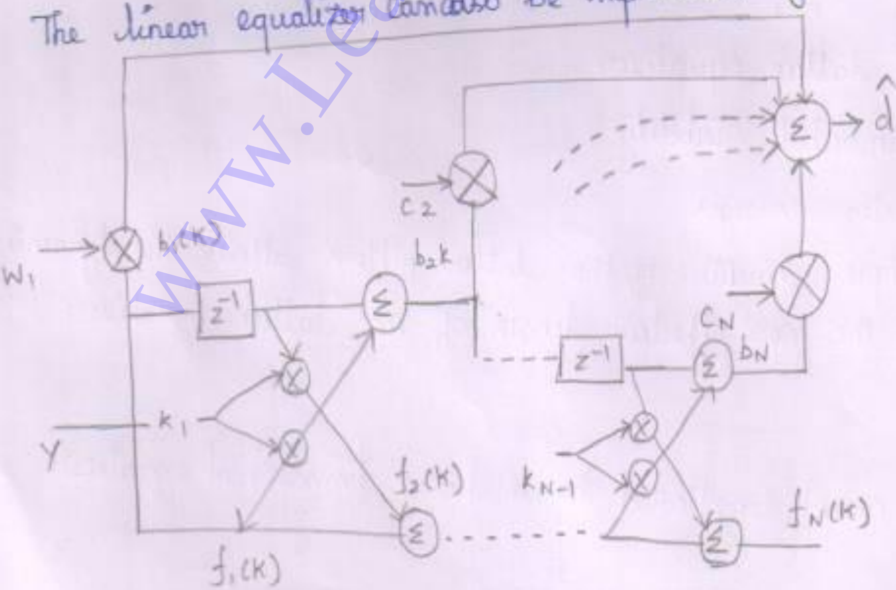
The minimum mean squared error $E(|e(n)|^2)$ that a linear equalizer can achieve is

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

$X(e^{j\omega T})$ - frequency response of the channel

N_0 - Noise power spectral density

The linear equalizer can also be implemented by lattice filter



In this type of filter the i/p signal y_k is transformed into set of n intermediate forward & backward error signals $f_n(k)$ & $b_n(k)$ respectively

The each stage of the lattice is then characterized by the following recursive equation

$$f_1(k) = b_1(k) = y(k)$$

$$f_n(k) = y(k) - \sum_{i=1}^{n} k_i\, y(k-i) = f_{n-1}(k) + k_{n-1}(k)\, b_{n-1}(k-1)$$

$$b_n(k) = y(k-n) - \sum_{i=1}^{n} k_i\, y(k-n+i) = b_{n-1}(k-1) + k_{n-1}(k)\, f_{n-1}(k)$$

$k_n(k)$ - reflection coefficient for the n^{th} stage of the lattice

The b_n backward error signals are than used as i/ps to the tap weights & the o/p of the equalizer is given by

$$\hat{d}_k = \sum_{n=1}^{N} w_n(k)\, b_n(k)$$

Advantage of lattice equalizer

- Numerical stability
- Faster coverages
- Unique structure of the lattice filters allows the dynamic ~~assg~~ assignment of the most effective length of the lattice equalizer.

Disadvantage

It is more complicated than a linear transversal equalizer

Non-linear Equalization :-

If the channel distortion is too serve in linear equalizer to handle then the non-linear equalizers are used to compensate the distortion

There are 3 non-linear methods are used in most 2G & 3G systems.

These are

1) Decision Feedback Equalization
2) Maximum likelihood Symbol detection
3) Maximum likelihood Sequence Estimation

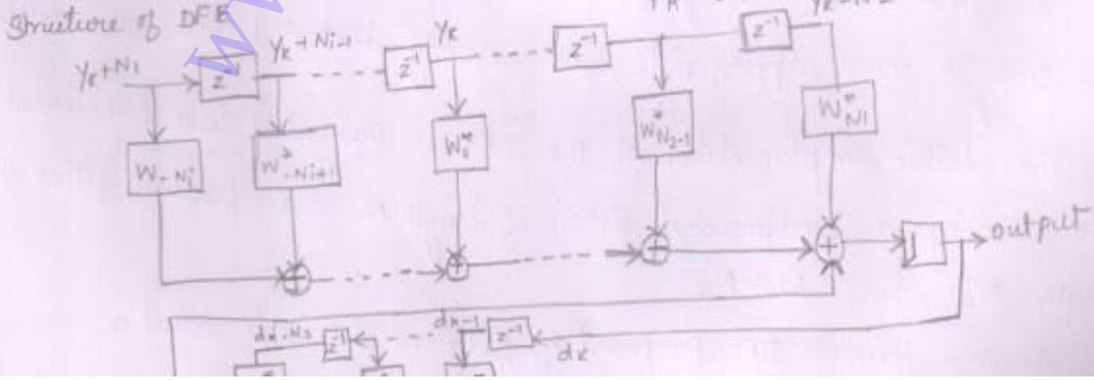
Decision Feedback equalization

The basic idea used in the DFE is if the further symbols induce ISI that can be estimated and subtracted out before detection of subsequent symbols.

The DFE can be realized by direct transversal form or as a lattice filter

It consist of a feed forward filter & a feedback filter

Structure of DFE



The FBF is driven by decisions on the o/p of the detector & its coefficients can be adjusted to cancel the ISI on the current symbol from past detected symbols.

The equlizer has N_1+N_2+1 taps the feedforward filter & N_3 taps in the FBF & its o/p can be expressed as

$$\hat{d}_k = \sum_{n=-N_1}^{N_2} W_n^* y_{k-n} + \sum_{i=1}^{N_3} F_i d_{k-i}$$

W_n^* & y_n → Tap gains & the i/p to the feed filter

F_i^* – Tap gains for the FBF

$d_i (i<k)$ – previous decision made on the detected signal

Once $\hat{d}_k$ is obtained than d_k is decide from it. Then d_{k-1}, d_{k-2} are feedback into the equalizer & $\hat{d}_{k+1}$ is obtained by using the above equation d_k.

The minimum mean squared error a DFE can achieve is

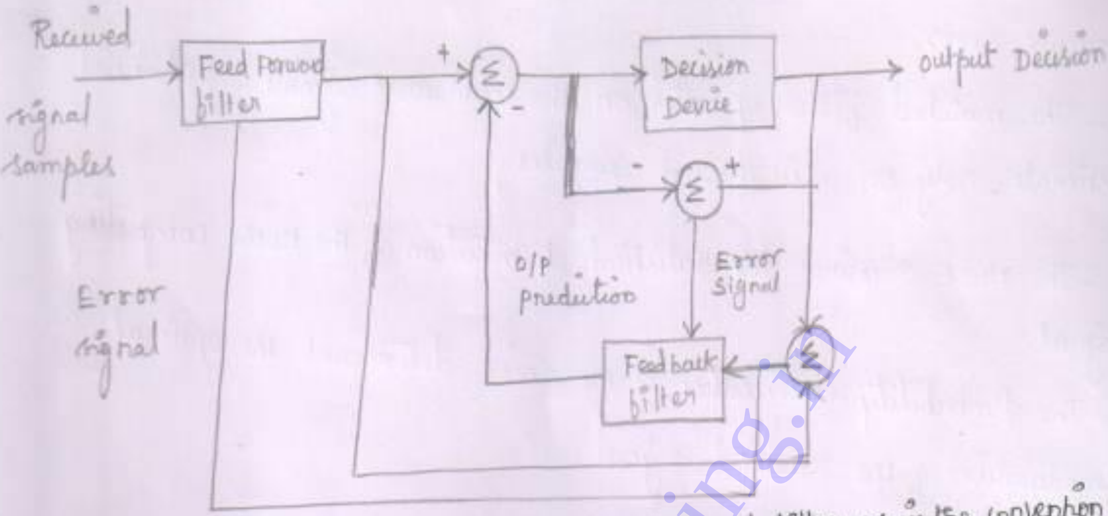
$$E(|e(n)|^2)_{min} = \exp\left\{\frac{T}{2\pi}\int_{-\pi/T}^{\pi/T} \ln\left[\frac{N_0}{|X(e^{j\omega T})|^2+N_0}\right]\right\}$$

A DFE has significantly smaller minimum MSE than an LTE. The DFE is more appropriate for severely distorted wireless channels.

The lattice implementation of the DFE is equivalent to a transversal DFE having a feedforward filter of length N_1 & a feedback filter of length N_2 where $N_1 > N_2$.

Another form of DFE proposed by Belfiore & park is called a

Predictive Decision feedback equalizer

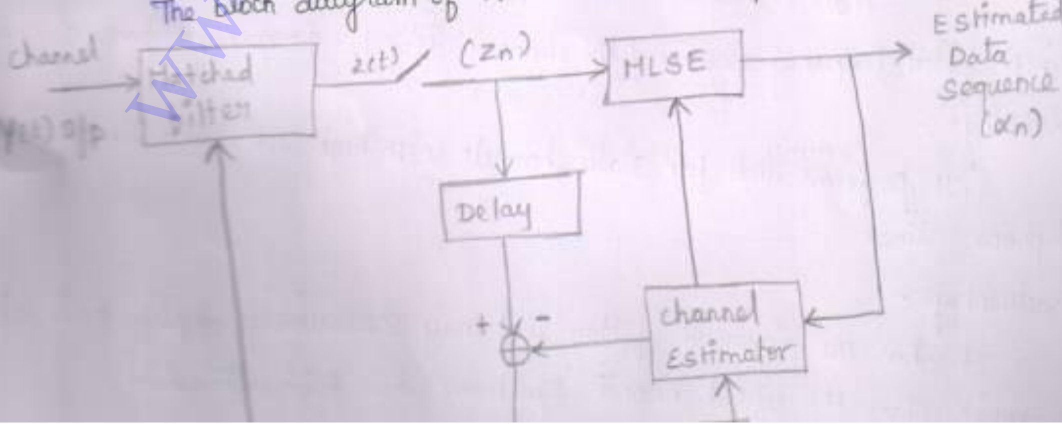


- The equalizer consists of a feedforward filter as in the convention DFE. The FBF is called a Noise predictor because it predicts the Noise & the residue ISI contained in the signal at the FFF O/P & Subtracts from it the detector O/P after some feedback delay.

Maximum likelihood Sequence Estimation (MLSE) Equalizer

- In this Type of equalizer we are using a channel impulse response Simulator within the algorithm.
- The MLSE Test all possible data sequences & chooses the data sequence with the maximum probability as the O/P.

The block diagram of MLSE with an adaptive matched filter



- The MLSE is optimal in the Sense that it minimizes the probability of a sequence error.

- The matched filter operates on the continuous signal & channel estimate rely on discretized samples.

- The MLSE requires the satistical distribution of the noise computing signal.

- The Probability distribution of the noise determines the optimal demodulation of the received signal.

Zero forcing algorithm [ZFA]

- Zero forcing means set equalizer impulse response to zero & the equalizer coefficients Wn are chosen to force the samples of the combined channel.

- In this algorithm the delay elements provide a time delay equal to the symbol duration T & the frequency response Heq(f) of the equalizer is periodic with a period combined equal to the symbol rate 1/T.

The combined response of the channel with the equalizer is

$$H_{ch}(f)\, H_{eq}(f) = 1, \quad |f| < 1/2T$$

Hch(f) - folded frequency response of the channel Heq(f) - frequency response of equalizer

Advantage:-

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

Disadvantage:-

It has the inversive filter that may excessively amplify noise at frequencies where the folded channel spectrum has high attenuation.

Least Mean Square algorithm (LMSA)

In order to minimize the mean square error b/w the desired equalizer o/p & the actual equalizer o/p the LMS equalizer is used.

The predictor error is given by

$$e_K = d_K - \hat{d}_K = x_K - \hat{d}_K$$

One more equation for e_K is

$$e_K = x_K - Y_K^T W_K = x_K - W_K^T Y_K$$

The mean square error $|e_K|^2$ at time instant K is obtained by

$$\mathcal{E} = E(e_K^* e_K)$$

For a specific channel condition the prediction error e_K is dependent on the tap gain vector W_N so that mean square error of an equalizer is a function of W_N.

Let the cost function $J(W_N)$ denote the mean squared error as a function of tap gain vector W_N

In order to minimize the MSE

$$\frac{d}{dW_N} J(W_N) = -2P_N + 2R_{NN} W_N = 0$$

$$R_{NN} \hat{W}_N = P_N$$

Then the MMSE of an equalizer is

$$J_{opt} = J(\hat{W}_N) = E(x_K x_K^*) - P_N^T \hat{W}_N$$

$$\hat{W} = R_{NN}^{-1} P_N$$

The LMSA is the simplest equalization algorithm and requires only 2N+1 operation per iteration

Advantages:

1. Low computation of complexity
2. Simple program

Disadvantage

1. Poor tracking
2. Slow convergence.

Diversity Techniques

It is a powerful technique which is used in the receiver to improve the efficiency of the wireless link with relatively low cost.

- In this type of reception the received signal is selected from many paths among those the strongest signal is predicted

- By placing many antennas in different directions, diversity reception can be achieved. This is termed as antenna diversity or space diversity.

For diversity reception technique

$$P(\gamma_1, \gamma_2 \ldots \gamma_m \le \gamma) = (1 - e^{-\gamma/\Gamma})^M = P_M(\gamma)$$

M - no. of paths / diversity branch

Γ - Mean SNR

$\gamma_1, \gamma_2 \ldots \gamma_M$ - SNR of each branch

γ - Threshold SNR

$P(\gamma_1, \gamma_2 \ldots \gamma_M)$ - Probability of SNRs falling below a threshold γ.

Micro diversity :-

The diversity methods that combat Small-Scale fading i.e the fading created by interference of MPCs is called as Micro diversity

TYPES

* Temporal diversity
* spatial diversity
* Frequency diversity
* Angular diversity
* polarization diversity

TEMPORAL DIVERSITY :-

As the wireless propagation channel is time variant signals that are received at different times are uncorrelated. For sufficient de correlation

The temporal distance must be atleast $1/2\nu_{max}$

ν_{max} - Maximum Doppler frequency.

Temporal diversity can be realized by following ways

1. Repetition coding
2. Automatic ~~Request~~ Repeat Request
3. Combination of interleaving & coding

Repetition coding

The signal is repeated several times where the repetition intervals are long enough to achieve de-correlation

To achieves diversity but is also highly b.w inefficient. spectral efficiency ↓ by a factor that is equal to the number of repetitions.

Disadvantage:- Retransmission Occur always

Automatic Repeat Request:

The Rx Sends a message to the Tx to ensure whether it received the data with Sufficient quality. If the data is not transmitted Successfully then the transmission is repeated

Advantage :-

1. The spectral efficiency of ARQ is better than that of repetition coding

2. It requires multiple transmission only when the 1st transmission Occurs in a bad fading state.

Disadvantage:- It requires feedback channel

Combination of interleaving & coding

A more advanced version of repetition coding is forward error correction coding with interleaving. The different symbols of a code word are transmitted at different times which ↑ the probability that at least some of them arrive with a good SNR.

Space Diversity (or) Antenna Diversity

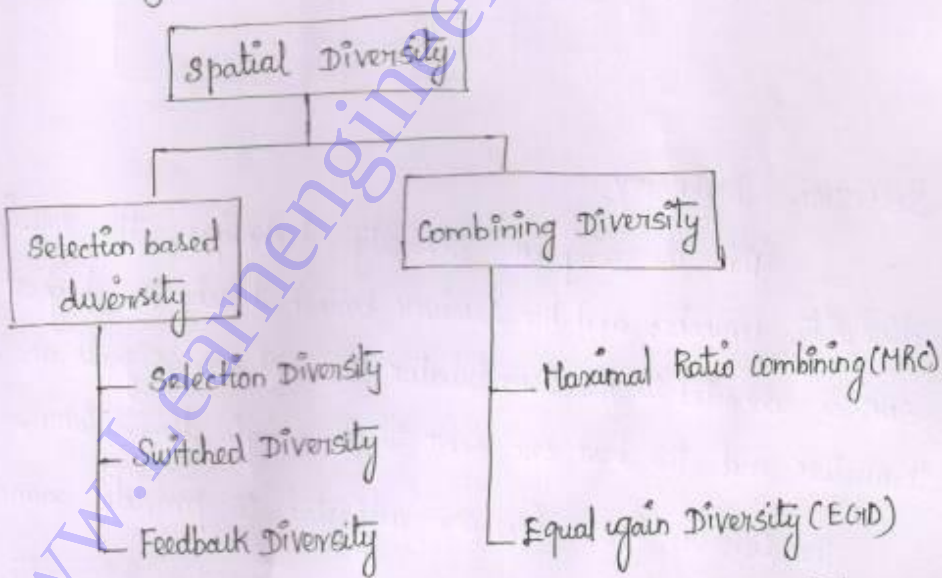
Principle

The basic principle of this type of diversity is selecting the best signal among all the signals received from different branches at the receiving end.

Space diversity reception methods can be classified as

a) Selection Based Diversity

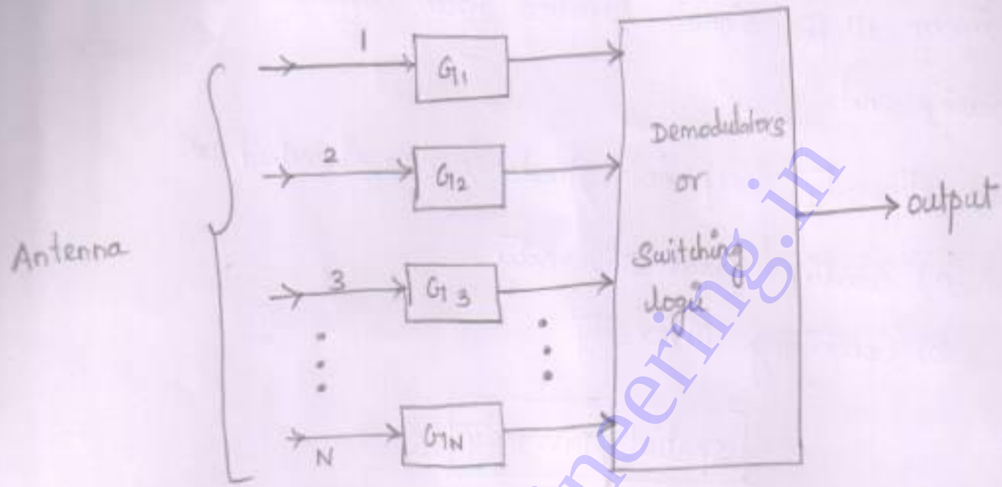
b) Combining Diversity



- One of the most popular forms of diversity used in the wireless system is Space diversity also known as antenna diversity.
- This diversity technique is used in base station design
- At each cell site, multiple base station receiving antennas are used to provide diversity reception

- The base station antennas are spaced considerably far apart to achieve decorrelation

Block diagrams for space diversity

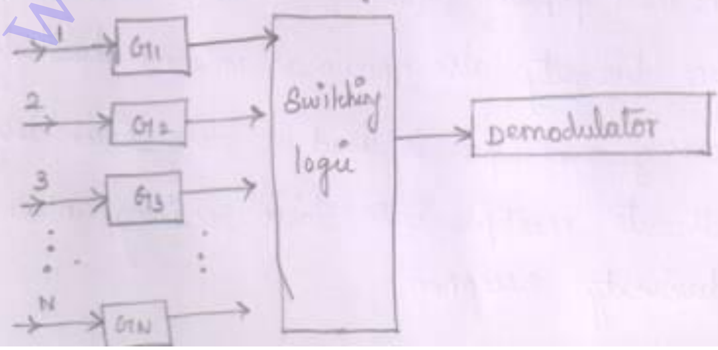


SELECTION DIVERSITY:

This the simplest Diversity technique. The receiver has M diversity branches and the receiver branch having the highest instaneous SNR is connected to the demodulator. The antenna signals are sampled themselves and the best one sent to a single demodulator.

The G_1, G_2, G_3 ... G_N are adjusted to provide same average SNR to all branches

Block diagram of selection Diversity



Switched Diversity

Principle

Selection Criterion of active diversity branch is monitored. If it falls below a certain thershold, 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.

Needs:-

selection diversity has selection criteria that all diversity branches have to be monitored in order to know when to select a different antenna. This leads to either ↑ hardware effort or reduced spectral efficiency. To overcome this selection diversity drawbacks switched diversity is used.

Parameters in Switched diversity

1. Switching thershold   2. Hysteresis time

FEEDBACK or SCANNING DIVERSITY

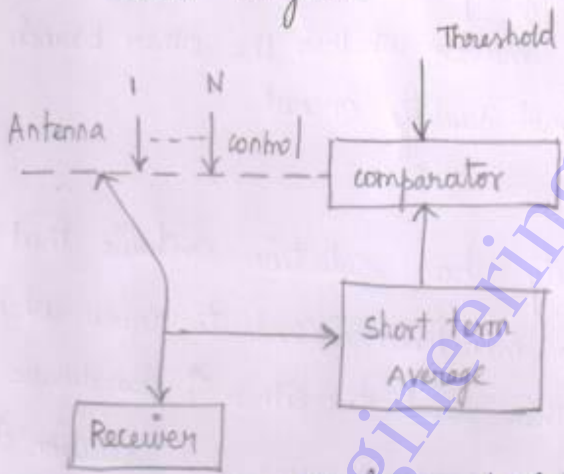
Principle

Scanning diversity is similar to selection diversity except that instead of always using the best of M signals. The M signals are scanned in a fixed Sequence until one is found to be above a predetermined threshold. This signal is received until it falls below threshold & the scanning process is again initiated

- Instead of selecting best among N signals the N i/ps are scanned one by one until an i/p above a thershold SNR is found.

- This i/p is used until the SNR falls below the thershold & the scanning is again initiated

Block diagram



Advantage: It is very simple to implement & only one receiver is required. Disadvantage: The resulting fading statistics are somewhat inferior to those obtained by other methods.

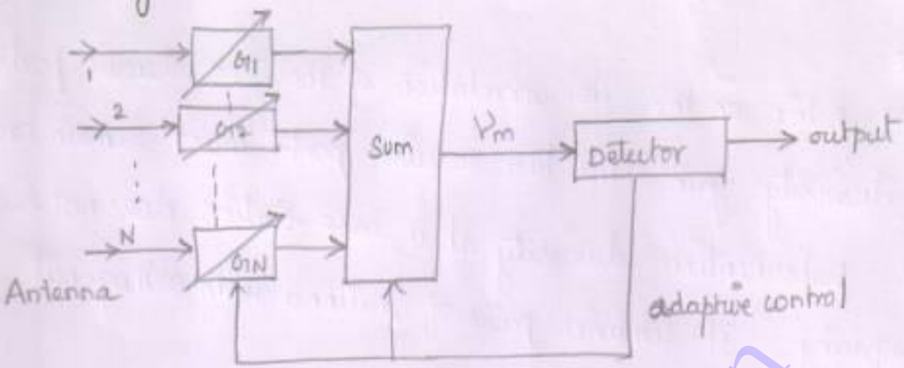
Maximal Ratio Combining:-

In this method the signals from all of the N branches are weighted according to their individual signal voltage to noise power ratios and then summed.

- An individual receiver and phasing circuit for each antenna element & the individual signals must be co-phased before being summed.

- It produces an o/p SNR equal to the sum of individual SNR's

Block diagram



Advantage:-

It gives the best satistical reduction of fading of any known linear diversity combines

Disadvantage:-

1. It requires individual receiver & phasing circuits for each antenna elements.

2. Needs high implementation cost.

Equal gain diversity

- In this method the branch weights are all set to unity but the signals from each branch are cophased to provide equal gain combining diversity

- This allows the receiver to exploit the signals that are simultaneously received on each branch.

Advantage:- Performance is only margenally inferior to maximal ratio combining & superior to selection diversity.

Polarization Diversity:-

Principle

It relies on the de-correlation of the two receive ports to achieve diversity gain. The two receiver ports must remain cross polarized. polarization diversity at a base station does not require antenna spacing. It combines pairs of antenna with orthogonal polarizations

- In this diversity the transmitted signal with horizontal or vertical polarization is received by antenna with two elements.

- One element is used for horizontal polarization & the other is used for vertical polarization. Wireless communication systems usually use vertical polarization because this is more convinent for use with portable & mobile antennas.

- A vertically polarized signal may be transformed into horizontal polarization due to multipath propagation the signal received in any polarization will be intercepted by the 2 element antennas.

The 2 elements are polarized $\pm\alpha$ to the vertical assuming vertical polarization employed in the transmitter

Advantage:

It reduces the multipath delay spread.

The mutual coupling which depicts that antenna B acts as reflector for antenna A, whose pattern is Skewed to left. By the by Antenna B pattern is skewed to right due to reflections from antenna A.

TIME Diversity

- The signal representing the Same information are sent over the same channel at different times. Time diversity repeatedly transmits information at time spacings that exceeds the coherence time of the channel. Multiple repetition of the signal will be received with independent fading conditions thereby providing for diversity.

- The information is transmitted at time spacings that exceeds the coherence time of the channel.

Disadvantage:- It requires spare B.W also as many receivers as there are channels used for the frequency diversity

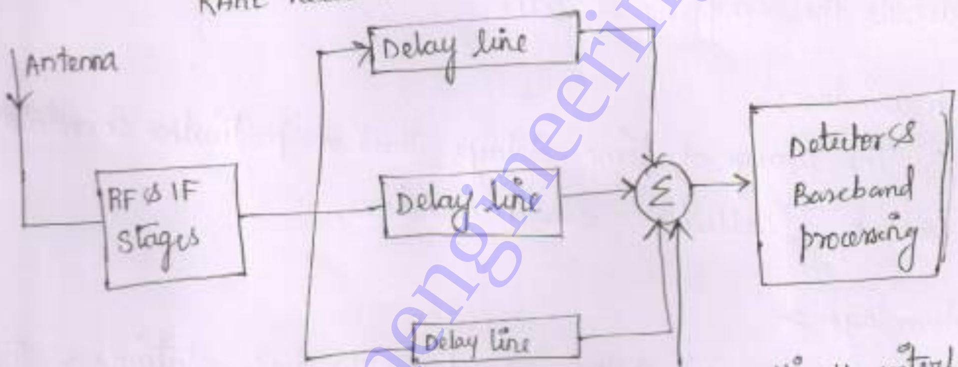
One modern implementation of time diversity involves the use of the RAKE receiver for spread spectrum CDMA

RAKE RECEIVER:-

- The CDMA receivers may combine the time delayed versions of the original signal transmission in order to improve the SNR at the receiver. This is done by RAKE receiver.

- It attempt to collect the time shifted versions of the original signal by providing a separate correlation receiver for each of the multipath signals. It provides link improvement through time diversity

RAKE Receiver



One of the advantage of CDMA system is the multipath interference can be reduced by combining direct & reflected signal in the receiver. The receiver used are called RAKE receivers

The mobile unit can combine three Rf signals delaying & of them to match the 3rd. One of these signals can be assumed to be the base station in the current cell.

The other 2 may be reflection is neighbouring base stations. Normally the base station receiver can combine 4 signals. The signals are

1. The direct signal from the mobile
2. Three reflected signal

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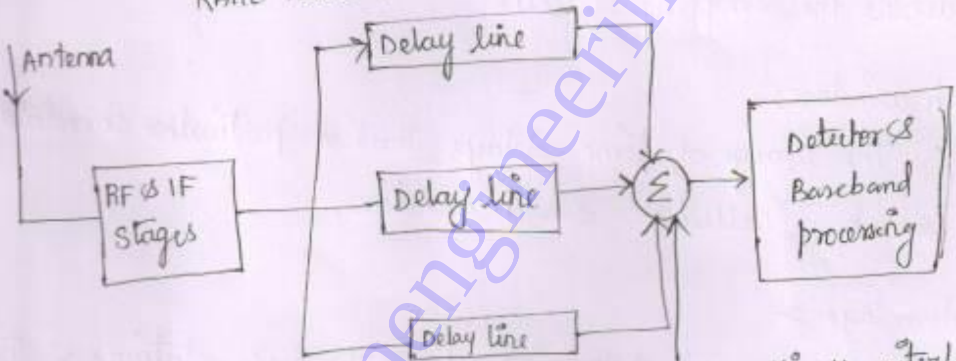

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2. Three reflected signal

- In addition two base station's may receive a signal from the same mobile.
- Then the BS send their to the MSC which use the higher quality signal. Decision about quality are made on a frame by frame basis every 20ms.

It is possible to have 2 base stations communicating with the mobile indefinitely in what is referred to as soft handoff. This avoids the dropping of calls.

Search window :-

The range of time delays that a particular correlator can search is called a search window.

Disadvantage :-

It considerably ↑ the load on the base stations & the Switching network.

Macrodiversity

- The diversity methods the combat large scale fading i.e the fading created by shadowing effects is called as Macro diversity
- shadowing is almost independent of transmit frequency & polarization so that frequency diversity or polarization diversity are not effective

If correlation distance for long scale fading are on the order of tens or hundreds of meters, then the spatial diversity can be used.

The simplest method for macrodiversity is the use of on frequency repeaters that receive the signal & retransmit an amplified version of it.

Advantage

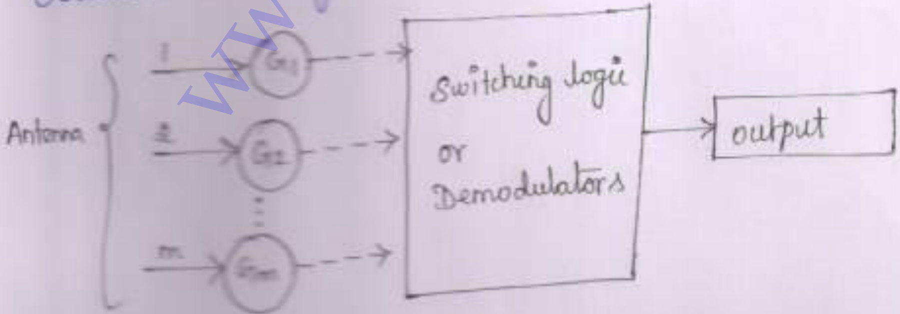
1. Use of on-frequency repeaters that receive signal & retransmit an amplified version of it

2. In cellular applications the two BS should be synchronized

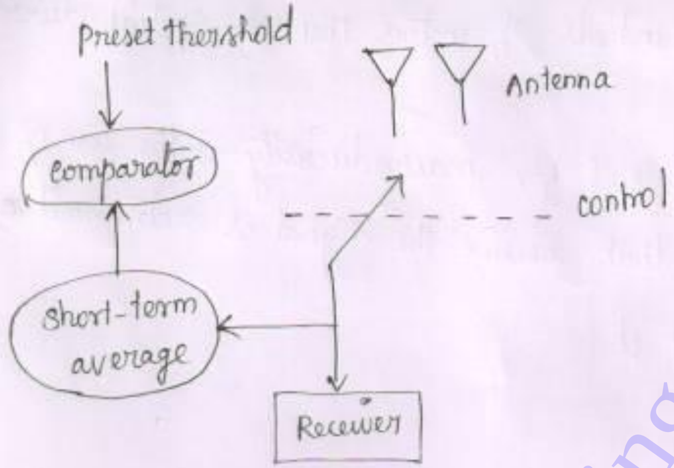
3. Compensating large signal fading Macro diversity is used.

Diversity Combining Techniques

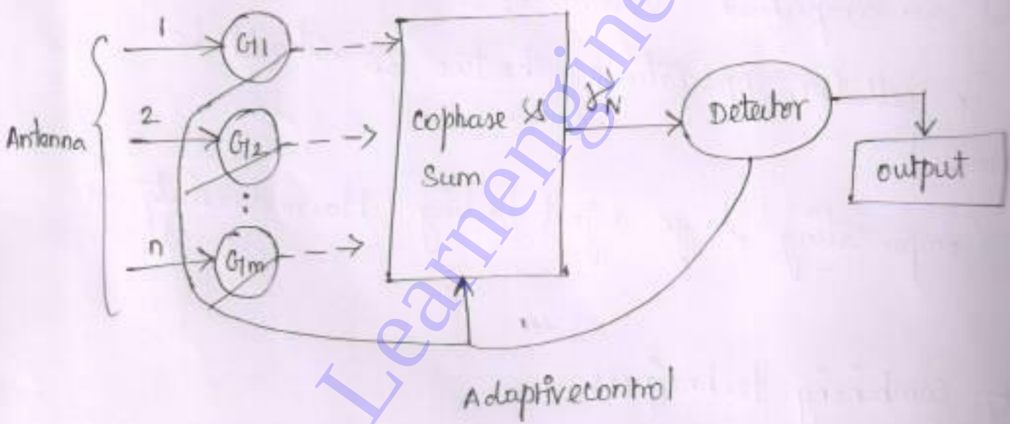
Selection diversity



Feedback or Scanning diversity



Maximal ratio combining



UNIT V

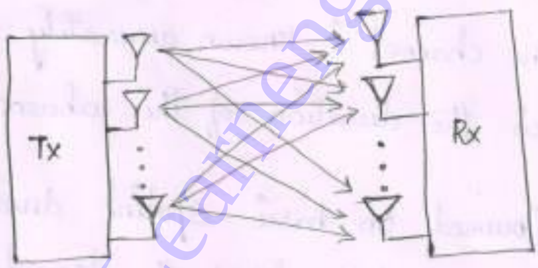
Multiple Antenna Techniques

MIMO systems :-

- Multiple input multiple output or MIMO is a radio communication technology or RF technology that is being mentioned and used in many new technologies these days.

- The wireless MIMO channel is assumed to consist of a system with multiple transmit antennas and multiple receive antennas which are connected by means of fading channels.

MIMO Model



- The antennas are placed in order to ensure that the channels across antennas are independent.

- placement Separation in units of $\lambda/2$ is one means to ensure independences. Conventionally for a MIMO system with M transmit antennas & N receive antennas the number of channels counting each link from a transmitter to a receiver separately is MN much like in conventional Wireless Communication.

- These channels must be known at least the receiver or transmitter or both in order to communicate information successfully during the coherence interval.

- This is done by means of estimation of the channel coefficients using an appropriate techniques

- An M transmitter N receiver MIMO system represented as an NxM matrix denoted by H.

- A flat fading model for simplicity of analysis since techniques such as OFDM can be used to convert transform frequency selective channels into flat fading ones.

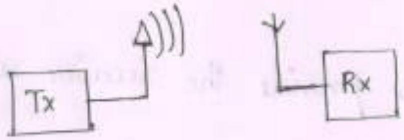
- Assume that the channel is known accurately to the receiver & does not change in the duration of this coherence interval.

- MIMO system focused on basic spatial diversity here the MIMO system was used to limit the degradation caused by multipath propagation.

- The first step as system then started to utilise the multipath propagation to advantage turning the additional signal paths into what might effectively be considered as additional channels to carry additional data.

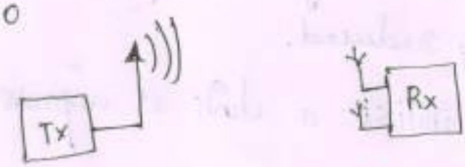
Multi antenna types

SISO:-



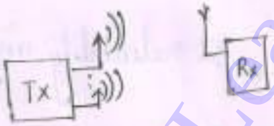
Single i/p single o/p means that the transmitter and receiver of the radio system have only one antenna.

SIMO



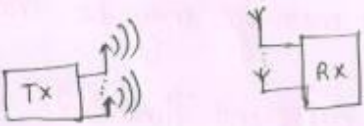
Single input multiple output means that the receiver has multiple antennas while the transmitter has one antenna.

MISO



Multiple input single output means that the transmitter has multiple antennas while the receiver has one antenna.

MIMO



Multiple i/p multiple o/p means that the both the transmitter & receiver have multiple antennas.

- The channel may be affected by fading & this will impact the signal to noise ratio.

- The principle of diversity is to provide the receiver with multiple version of the same signal.

- If these can be made to be affected in different ways by the signal path, the probability that they will be affected at the same time is considerably reduced.

- The diversity helps to stabilise a link & improve proformance reducing error rate.

- Multiple data streams transmitted in a single channel at the same time

- Multiple radios collect multipath signals

- Delivers simultaneous speed, covererage, & reliability improvements

Several different diversity modes are available & provide a number of

Time diversity :-

Using time diversity a message may be transmitted at different times e.g. Using different time slots & channel coding

FREQUENCY Diversity:-

This form of diversity uses different frequencies. It may be in the form of using different channels or technologies such as spread spectrum / OFDM

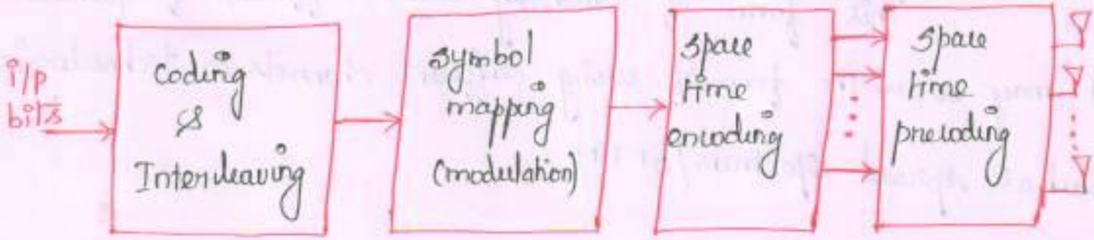
Space diversity:

Space diversity used in the broadest sense of the definition is used as the basis for MIMO. It uses antennas located in different positions to take advantage of the different radio paths that exist in a typical terrestrial environment.

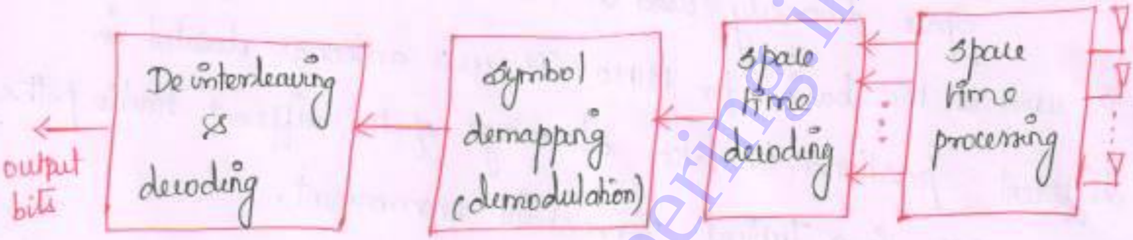
- MIMO uses multiple antennas on both the transmitter and receiver. They have dual capability of combining the SIMO & MISO technologies.
- They can also ↑ capacity by using spatial multiplexing.
- The MIMO method has some clear advantages over SISO methods
- The fading is greatly eliminated by spatial diversity low power is required compared to other techniques in MIMO.
- The number of antenna element ↑ the channel capacity is ↑
- The improving of MIMO from MISO channel capacity as antenna ↑.

Building Blocks of MIMO

Transmitter



Receiver



- The space dimensions to improve wireless systems capacity range & reliability

- It offers significant ↑ in data throughput & link range without additional bandwidth or increased transmit power.

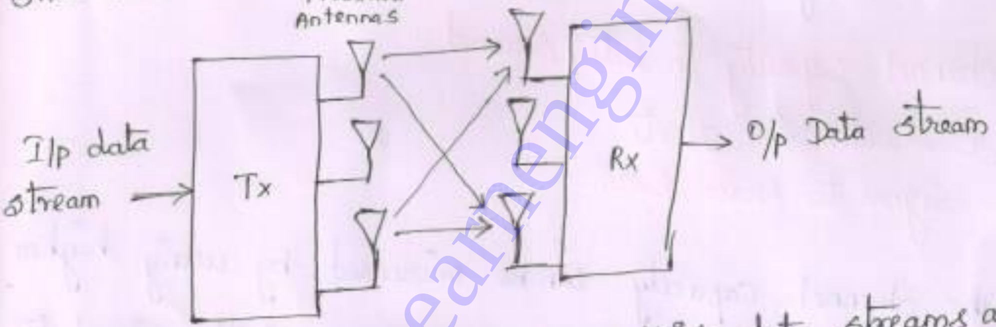
- MIMO achieves this goal by spreading the same total transmit power over the antenna to achieve an array gain that improves the spectral efficiency or to achieve a diversity gain that improves the link reliability.

Spatial Multiplexing

One of the key advantages of MIMO spatial multiplexing is the fact that it is able to provide additional data capacity.

- MIMO spatial multiplexing achieves this by utilising the multiple paths is effectively using them as additional channels to carry data.

- The maximum amount of data that can be carried by a radio channel is limited by the physical boundaries defined order Shannon's law



- The spatial multiplexing, multiple data streams are transmitted at the same time.

- They are transmitted On the same channel but by different antenna. They are recombined at the receiver using MIMO signal Processing.

Shannon's law :-

- The amount of data that can be passed along a specific channel in the presence of noise.

- The law that governs that is called shannon's law named after the formulated it.

Shannon's law defines the maximum rate at which error free data can be transmitted over a given bandwidth in the presence of noise. It is usually expressed in the form

$$C = W \log_2 (1 + S/N)$$

C - channel capacity in bits/second

W - Bandwidth in Hertz

S/N - Signal to Noise Ratio

- The channel capacity can be increased by using higher order modulation scheme but these require a better signal to noise ratio than the lower order modulation schemes. but these require a better signal to noise ratio than the lower order modulation schemes

- Thus a balance exists b/w the data rate & the allowable error rate signal to noise ratio & power that can be transmitted.

- spatial Multiplexing is a transmission technique in MIMO spatial multiplexing is a wireless communication to transmit independent & separately encoded data signals so called streams for each of the multiple transmit antennas.

- If the transmitter is equipped with N_t antennas & the receiver has N_R antennas the maximum spatial multiplexing order is $N_s = MIN(N_T, N_r)$

N_s - streams can be transmitted in parallel, ideally leading to an N_s ↑ of the spectral efficiency.

- The multiplexing gain can be limited by spatial correlation which means some of the parallel stream may have very weak channel gains.

BEAM FORMING:-

- Antenna technologies are the key in increasing network capacity. It started with sectorized antennas.

- These antenna illuminate 60° or 120° operate as one cell.

- Adaptive antenna arrays intensify spatial multiplexing

- Smart antennas belong to adaptive antenna arrays but differ in their smart direction of arrival estimation

- Smart antennas can form a user specific beam. optional feed back can reduce complexity of the array system

Beam forming is the method used to create the radiation pattern of an antenna array. It can be applied in all antenna array systems as well as MIMO systems.

Smart antennas are divided into 2 groups

* phased array system
* Adaptive array system with an infinite number of patterns adjusted to the scenario in real time

Switched Beam former

Adaptive Beam former

- Switched beam formers electrically calculate the DOA & switch on the fixed beam. The user only has the optimum signal strength along the center of the beam.

- The adaptive beam former deals with that problem & adjusts the beam in real time to the moving UE. The complexity & the cost of such a system is higher than

4. Lineal Type

Pre coding

- It is a generalization of beam forming to support multilayer transmission in multi antenna wireless communications.
- In conventional single layer beam forming the same signal is emitted from each of the transmit antennas with appropriate weighting such that the single power is maximized at the receiver o/p.

Pre coding can be separated by two classifications

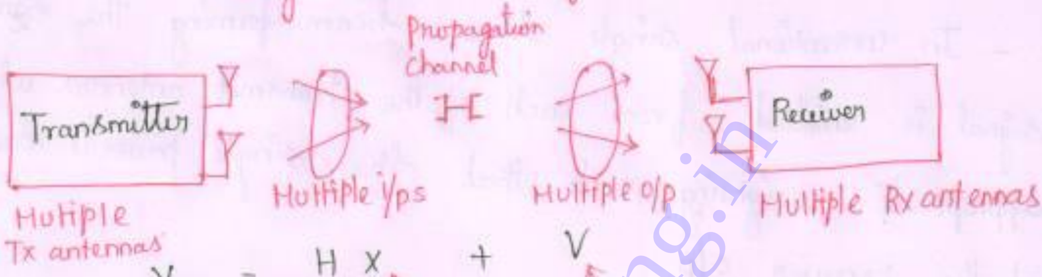
* pre coding for single user MIMO
* precoding for Multi user MIMO

Pre coding for single user MIMO

- In single user MIMO systems a transmitter equipped with multiple antennas communicate with a receiver that has multiple antennas.
- Most classic precoding assume narrowBand, slowly fading channels meaning that the channel for a certain period of time can be described by a single channel matrix which does not change faster.
- The precoding strategy that maximize the through put called channel capacity depends on the channel state information available in

- Single user MIMO communication systems exploit multiple transmit and receive antennas to improve capacity, reliability & resistance to interference.

Single User MIMO system



$$Y = H\,X + V$$

Y – Receiver vector; H – Matrix channel; X – Transmit vector; V – noise vector

Pre coding for Multi User MIMO

- In multi User MIMO a multi antenna transmitter communicates simultaneously with multiple receivers. This is known as space division multiple access.

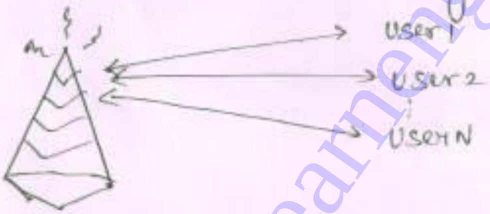
- precoding algorithm for SDMA System can be sub-divided into linear & non linear pre coding types.

- The capacity achieving algorithms are nonlinear but linear pre coding approaches usually achieve resonable performance with much lower complexity.

- Linear precoding strategies include MMSE precoding & the simplified zero forcing precoding. (7)

- There are also precoding strategies tailored for low-rate feedback of channel state information for ex random beam forming.

- Non linear precoding is designed based on the cocept of dirty paper coding which shows that any known interference at the transmitter can be subtracted without the penalty of radio resources if the optimal precoding scheme can be applied on the transmit signal.



Diversity coding:-

- It is used when there is no channel knowledge at the transmitter.

- In diversity methods a single stream is transmitted but the signal is coded using techniques called space time coding.

- The signal is emitted from each of the transmit antennas with full or near orthogonal coding.

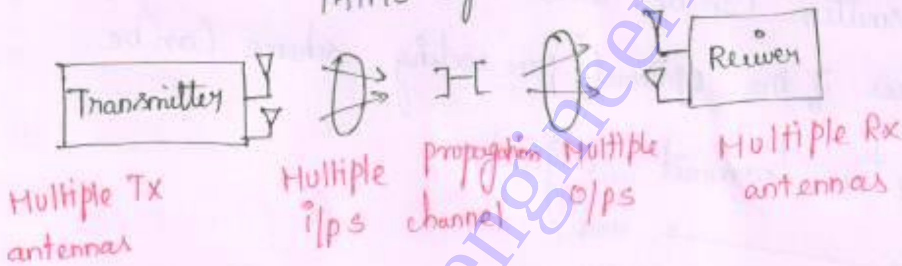
CHANNEL MODEL (SYSTEM MODEL)

- The transmitter & receiver are equipped with multiple antenna elements.

- The transmit stream go through a matrix channel which consists of multiple receive antennas at the receiver.

- Then the receiver gets the received signal vector by the multiple receive antennas & decodes the received signal vectors into the Original information

MIMO System Model



$$r = H s + n$$

Receive vector; matrix channel; transmit vector; Noise vector

- r is the $M\times 1$ received signal vector as there are M antennas in receiver

- H represented channel Matrix

- s is the $N\times 1$ Transmitted Signal Vector as there are N antennas in Transmitter

- n is an $M\times 1$ Vector of additive noise term

Let Q denote the covariance matrix of x then the capacity of the system described by information (8)

$$C = \log_2 \left[\det \left(I_M + HQH^*\right)\right] \; b/s/Hz$$

- This is optimal when is known at the transmitter & the i/p distribution maximizing the mutual information is the Gaussian distribution

- channel feedback may be known at the transmitter & the optimal is not proportional to the identity matrix but is constructed from a water filling argument.

- The effect of $Q = (P/N)$. Q based on perfect channel estimation & feedback then we can evaluate a maximum capacity gain due to feedback.

- The effect of H matrix has wide range of channel models including for ex. correlated fading & specular components,

$$C_{EP} = \sum_{L=1}^{M} \log_2\left(1 + \frac{P}{N}\lambda_i\right) \; b/s/Hz$$

where $\lambda_1, \lambda_2 \ldots \lambda$ the non zero eigen values of W

$m = \min(M, N)$

$$W = \begin{cases} HH^*, & M \le N \\ H^*H, & N \le M \end{cases}$$

The Singular Value Decomposition given by

$$H = UDV^*$$

U & V are Unitary

$$D = \mathrm{diag}(\sqrt{\lambda_1}, \sqrt{\lambda_2}, \sqrt{\lambda_3}, \ldots \sqrt{\lambda_m}, 0, \ldots 0)$$

The MIMO signal model

$$\hat{\gamma} = D\tilde{s} + \tilde{n}$$

where $\tilde{\gamma} = u + r$, $\tilde{s} = V*s$ & $\tilde{n} = U*n$

When the channel is known at the transmitter then H is known at above eqn

Capacity over Q subject to the power constraint $tr(Q) \leq P$

The optimal Q in this case is well known & is called a water filling solutions.

Capacity

$$C_{wp} = \sum_{i=1}^{M} \log_2 (\mu \lambda_i)^* \ b/s/Hz$$

where μ is chosen to satisfy

$$P = \sum_{i=1}^{M} (\mu - \lambda_i^{-1})^*$$

MIMO Diversity Techniques

- Diversity can be implemented at the transmit end at the receive end or at both ends of the wireless link.

- Generally MIMO diversity techniques can provide higher SNR and improve transmission reliability.

Transmit diversity :-

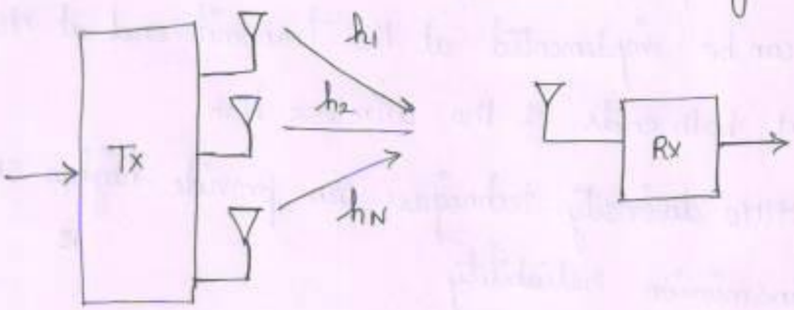
- It improves the signal quality and achieves a higher SNR ratio at the receiver side; it involves transmitting data stream through multiple antennas & receiving by single antenna or more.

- Transmit diversity can effectively mitigate multipath fading effects as multiple antennas afford a receiver several observations of the same data stream.

- Each antenna will experience a different interference environment & if one antenna experienced a deep fade, than it is likely that another has a sufficient signal.

- Ex: The transmit diversity techniques include Alamouti code & orthogonal j codes proposed by whole system Nt transmit antenna system.

Transmit Diversity



Receive Diversity

- It is widely Used in wireless communication systems it can be achieved by receiving redundant copies of the same signal.

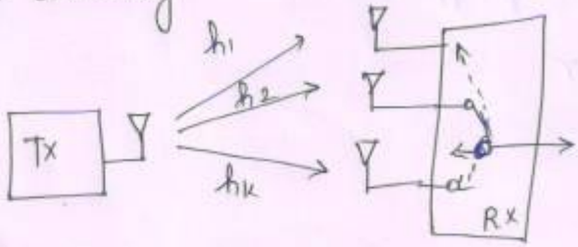
- The idea behind receive diversity is that each antenna at the receive end can observe an independent copy of the same signal.

- The probability that all signals are in deep fade simultaneously is significantly reduced.

- This type of diversity hasn't particular settings or requirements on the transmit end but requires a receiver that could simultaneously process all received Signals & Combines them by a proper combining method.

- There are several classical methods for combining the different diversity branches at the receiver most important of which & most widely used are slection combining, Maximal Ratio Combining & equal gain Combining

Selection combining



- The selection diversity improve SNR.

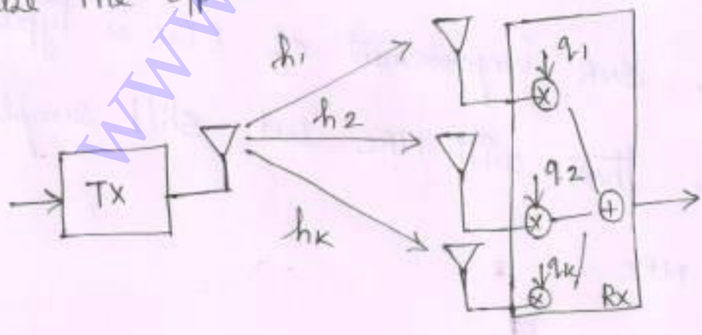
Let $\tilde{\gamma}$ is new average signal SNR. Γ_{avg}. SNR in each branch

$$\tilde{\gamma} = \gamma \sum_{K=1}^{M} \frac{1}{K} = \gamma \left(1 + \frac{1}{2} + \frac{1}{3} + \frac{1}{4} + \cdots \frac{1}{M}\right) > \Gamma$$

Maximal Ratio Combining

- The selection combining technique ignores information from all diversity branches except the particular branch that has highest SNR. This drawback is mitigated by using Maximal Ratio combining.

- The information from all branch is combined in order to maximize the o/p SNR.



The resulting signal envelop applied to detector

$$r_m = \sum_{i=1}^{n} G_i r$$

Total Noise power

$$N_t = N \sum_{i=1}^{M} G_i^2$$

SNR applied to detector

$$\gamma_m = r_m^2 / 2N_t$$

Equal gain Combining:

- Equal gain Combining is similar to Maximal Ratio Combining without weighting the signals before summation.

- In EGC co-phasing is needed to avoid signal cancellation.

- The average SNR improvement of EGC is typically about 1dB worse than with MRC but still simpler to implement than MRC.

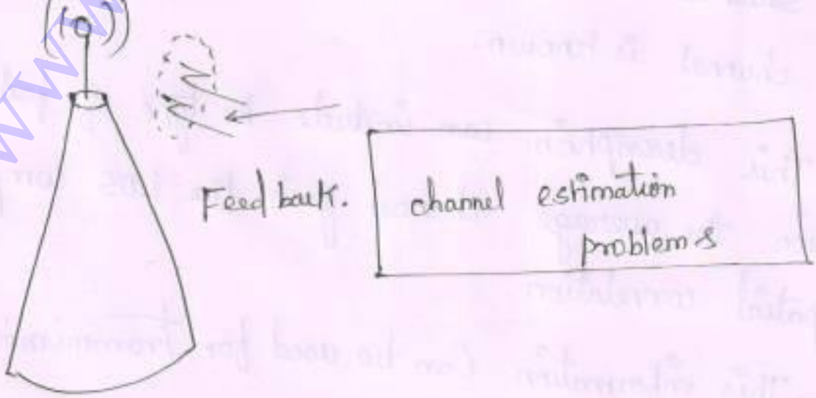
channel state information

- In wireless Communication channel state information simply represents the properties of a Communication link b/w the transmitter and receiver.

- The CSI describes how a signal propagates from the transmitter to the receiver & represents the combined effect of for ex scattering, fading & power delay with distance.

- The CSI makes it possible to adapt transmission to current channel conditions which is crucible for achieving reliable communication with high data rates in multiantenna systems.

- The CSI at the transmitter is vital in MIMO systems in Order to increase the transmission rate, to enhance coverage, to improve spectral efficiency and to reduce receiver complexity.



- The CSI is usually estimated at the receiving end & then quantized & fed back to transmitting side.

Instaneous CSI

- It is also known as short term CSI.

- CSI means that the current conditions of the channel are known which can be viewed as knowing the impulse response of a digital filter.

- This gives an opportunity to adapt the transmitted signal to the impulse response & thereby optimize the received signal for spatial multiplexing or to achieve low bit error rates.

Statistical CSI

- It is known as long-term CSI.

- Satistical CSI means that a statistical characterization of the channel is known.

- This description can include the type of fading distribution the average channel gain the LOS component & the spatial correlation.

- This information can be used for transmission

- The capacity of a MIMO channel is influenced by the degree of CSI available of both transmitter & receiver. (12)

- In most instances of multi antenna communication the receiver can accurately track the instantaneous state of the channel from pilot signals that are typically embedded within the transmission.

Capacity in fading & Non fading channel

MIMO Capacity: channel Unknown at the transmitter

- The generalized capacity equation for time space architecture.

- The transmitter only knows the channel statistics such as distributions of the channel distribution parameters.

$$C = \log_2 [I_N + (P/M) H^H H(+)] \text{ b/s/Hz}$$

where $(+)$, H, I_N & P represents transpose conjugate, $N \times M$ channel Matrix, $N \times N$ identity matrix & SNR.

The capacity of a MIMO system improves linearly with m fold where $m = \min(M, N)$

MIMO capacity channel known at the transmitter

- The additional performance gain can be achieved in MIMO systems with the CSI at the transmitter

- This scenario considers that the transmitter knows the random channel outcomes & adjust the transmit signal.

$$C = \log_2 [I_N + HQH(+)] \text{ b/s/Hz}$$

If Q denotes the covariance matrix of the transmitted M-D vector Gaussian signal of total radiated power P then the Shannon's capacity for a fading MIMO channel with AWGN is given as.

where (+), H & I_N represents the determinant transpose conjugate, N×M channel matrix & N×N identity matrix

$$Q = (P/M) I_N$$