EC8652 – Wireless Communication **L T P C 3 0 0 3**

OBJECTIVES: •

- To study the characteristic of wireless channel •
- To understand the design of a cellular system •
- To study the various digital signaling techniques and multipath mitigation techniques •
- To understand the concepts of multiple antenna techniques

UNIT I WIRELESS CHANNELS

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

UNIT II CELLULAR ARCHITECTURE

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

UNIT III DIGITAL SIGNALING FOR FADING CHANNELS

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

UNIT IV MULTIPATH MITIGATION TECHNIQUES

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

UNIT V MULTIPLE ANTENNA TECHNIQUES

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

OUTCOMES:

The student should be able to:

• Characterize a wireless channel and evolve the system design specifications
• Design a cellular system based on resource availability and traffic demands
• Identify suitable signaling and multipath mitigation techniques for the wireless channel and system under consideration.

TEXT BOOKS:

1. Rappaport,T.S., —Wireless communications‖, Pearson Education, Second Edition, 2010.(UNIT I, II, IV)
2. Andreas.F. Molisch, —Wireless Communications‖, John Wiley – India, 2006. (UNIT III,V)

REFERENCES:

1. Wireless Communication –Andrea Goldsmith, Cambridge University Press, 2011
2. Van Nee, R. and Ramji Prasad, —OFDM for wireless multimedia communications, Artech House, 2000
3. David Tse and Pramod Viswanath, —Fundamentals of Wireless Communication, Cambridge University Press, 2005.
4. Upena Dalal, —Wireless Communication‖, Oxford University Press, 2009.

Unit I
Wireless Channels.

Large scale path loss – Path loss models – Free space and Two Ray models – Link Budget design – Small Scale fading – Parameters of mobile multipath Channels – Time Dispersion parameters – Coherence bandwidth – Doppler Spread coherence time – Fading due to multipath time Delay Spread – flat fading – frequency selective fading – Fading due to doppler spread – fast fading – Slow fading.

Introduction:-

* The Radio wave propagation is influenced by the mechanisms namely scattering, reflection and diffraction.
* Most Cellular radio systems operate in urban areas where there is no direct Line-of-sight path b/w the transmitter and the Receiver.
* Due to multiple reflections from various objects, the electromagnetic waves travel along different paths of Varying lengths.
* As the distance is very large the probability of level of signal fade would be more.
* Estimation of T-R separation helps in determining the radio coverage.

* The propagation models are useful in calculation of the mean signal strength in transmission & these models are known as large scale propagation models.

* The characterization of fluctuations of Rxed signal over shorter time periods or shorter distances like few wave lengths by the models are known as small scale propagation models.

Large scale Path Loss:-

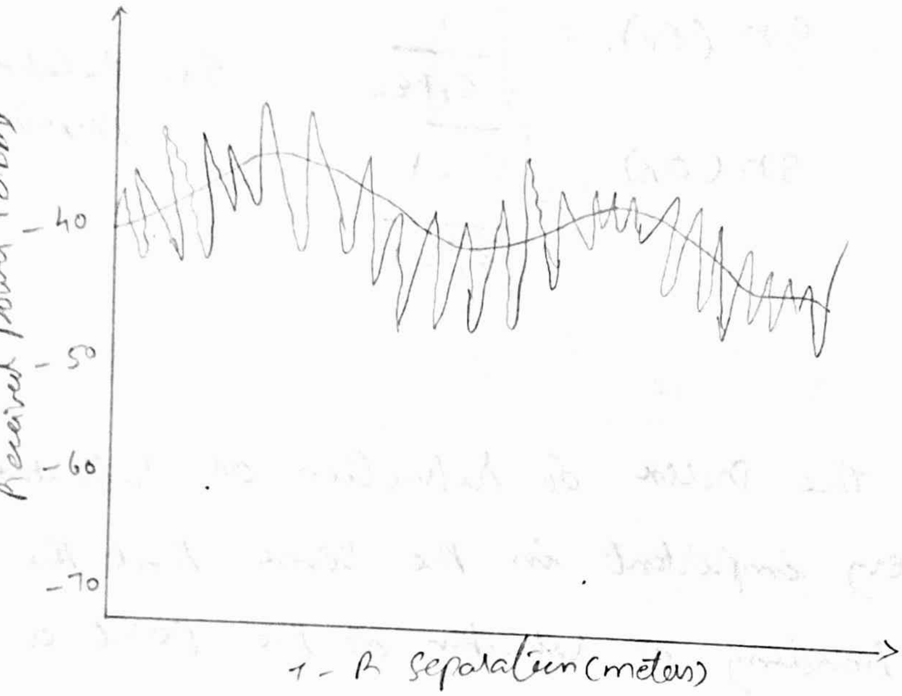
* The signal strength in wireless communication varies as it propagates, and it is closely related to the location through which it travels.

* The propagation models predicts mean signal strength (MSS) for a transmitter - Receiver sepelation (T-R) distance.

* The radio coverage area in this communication is also calculated and usually they are called as Large scale propagation models.

* This is due to signal strength calculations for Larger T-R separation distance.

* The propagation models predict rapidly fluctuating received signal strength.
* The T-R seperation is of shorter distances.
* This kind of models are said to be small scale fading models.



* From fig the attenuation gets varied as the distance gets changed.

The three basic propagation mechanism:-

Basic three propagation mechanisms are

* Reflection
* Diffraction
* Scattering.

Brewster angle:-

* The Brewster angle is the angle at which no reflection occurs in the medium of origin.

* It occurs when the incident angle θ_B is such that the reflection co-efficient $\Gamma_{\parallel}$ is equal to zero.

$$\sin(\theta_B) = \sqrt{\frac{\epsilon_1}{\epsilon_1+\epsilon_2}}$$

ϵ_r - Relative permittivity.

$$\sin(\theta_B) = \frac{\sqrt{\epsilon_r - 1}}{\sqrt{\epsilon_r^2 - 1}}$$

Refractive Index:-

The index of refraction or refractive index is very important in the sense that the amount of bending or refraction at the point of interface of two materials of different density depends on this refractive index.

Refractive index $n = c/v$

$c \rightarrow$ speed of light (3×10^8 m/s)

$V \rightarrow$ speed of light in a particular given material (m/sec).

Rayleigh Criterion:-

$$\cos \theta_i > \lambda / 8D.$$

Path loss model:-

Free space propagation model:-

Definition:-

The term "free space propagation model" refers to the measurement of Received signal strength when the transmitter and receiver of the communication system have clear line of sight in b/w the two i.e. there is no obstacle in b/w the Txer and Rxer.

Application areas:-

The free space propagation is useful in areas of (a) Microwave Radio Link (b)

Analysis:-

* The free space propagation model predicts the received power delay level and for this prediction method the distance/range b/w receiver antenna and transmitter antenna is measured.

* The power received by the receiver antenna is denoted as $P_r(d)$ and it is known as "Friis free space equation"

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

P_t → transmitted power
$P_r(d)$ → Received power
G_t → transmitter antenna gain
G_r → Receiver antenna gain
d → T-R distance
L → System loss
λ → wave length

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

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

* The effective aperture Ae is related to the physical size of the antenna, λ is related to the carrier frequency by

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

where, $f \rightarrow$ Carrier frequency in Hertz
$\omega_c \rightarrow$ Carrier frequency in radians per sec.
$c \rightarrow$ Speed of Light in meters/s.

* An "isotropic radiator" is an ideal antenna which radiates power with unit gain uniformly in all directions.

* The "effective isotropic radiated power [EIRP]" is defined as.

$$EIRP = P_t G_t.$$

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

* The "path loss" which represents signal attenuation as a positive quantity measured in dB, is defined as the difference (in dB)

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

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

* The antenna gains are assumed to have unity gain, the path loss is given by,

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

* The "far field (or) Fraunhofer region" of a txing antenna is defined as the region beyond the far-field distance d_f, which is related to the largest linear dimension of the txer antenna aperture and the carrier wave length.

* The fraunhofer distance is given by,

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

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

d_f must satisfy,

① $d_f >> D$

② $d_f >> \lambda$

* Let 'd_o' be the reference 'close-in' distance. Such that d_o is selected to be a smaller value than the practical distance under mobile communication.

* The Received power in free space at a distance greater than do is given by,

$$P_r(d) = P_r(d_o) \left(\frac{d_o}{d}\right)^2 \qquad d \geq d_o \geq d_f$$

* P_r is in units of dBm, the received power is given by,

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

$$d \geq d_o \geq d_f$$

where, $P_r(d_o)$ is in units of watts.

Ground Reflection (Two-Ray) model:-

* Two ray Ground reflection model is more accurate than free space propagation model.
* The two ray model considers the direct path, reflected path and geometric optics
* This method is used for predicting the large scale signal strength over distances of several kilometers and line of sight microcell channels in urban environments.

E_{TOT} → Total Received E-field
E_{LOS} → Direct Light of sight component.
E_g → Ground Reflected component
h_t, h_r → height of Txer & Rxer.

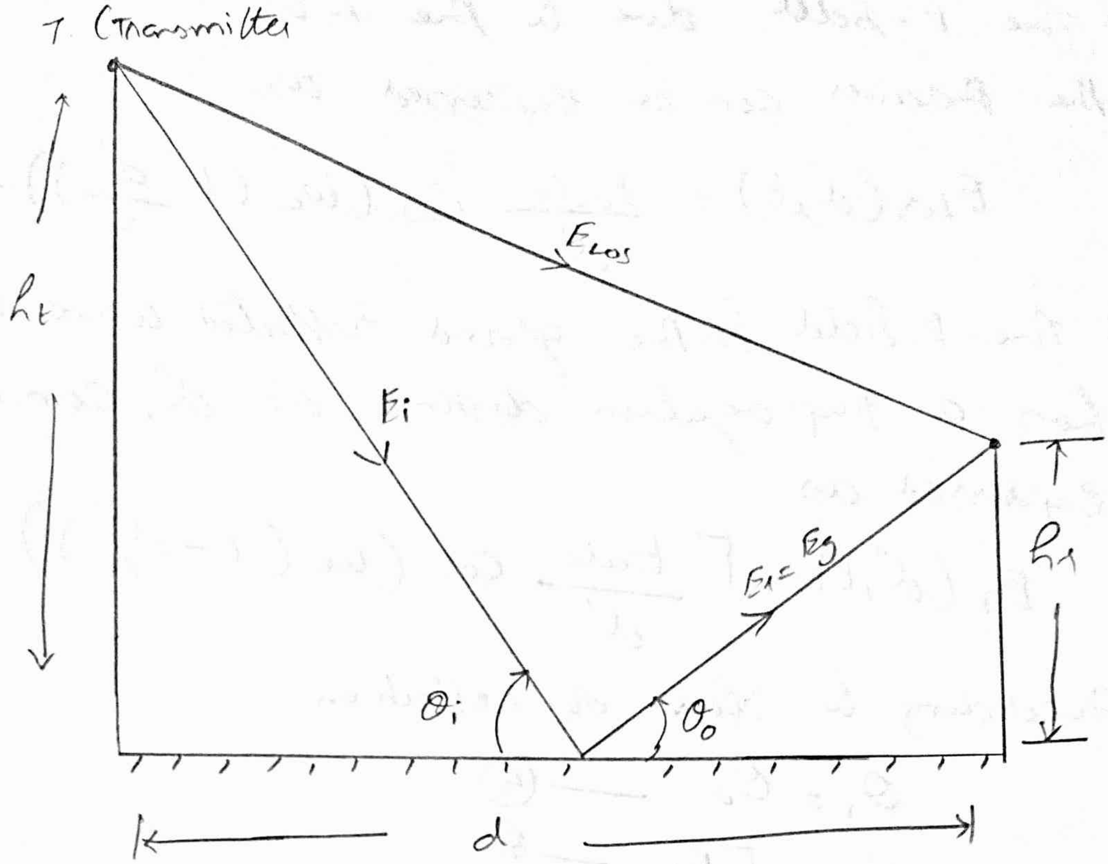


fig: Two-ray ground reflection model.

* If E_0 is the free space E-field at a reference distance d_0 from the transmitter, then for distance d_0, the free space propagating E field is given by

$$E(d,t) = \frac{E_0 d_0}{d} \cos\left(\omega_c\left(t - \frac{d}{c}\right)\right) \quad (d > d_0) \quad \text{—} ①$$

where $E(d,t) = \frac{E_0 d_0}{d}$ represents the envelop of the E field at d meters from the transmitter.

Two propagating waves arrive at the receiver.

① The direct wave that travels a distance d'.

② the reflected wave that travels a distance d''.

* The E-field due to the LOS component at the Receiver can be expressed as,

$$E_{LOS}(d',t) = \frac{E_0 d_0}{d'} \cos\left(\omega_c\left(t - \frac{d'}{c}\right)\right) \quad (2)$$

* The E field for the ground reflected wave which has a propagation distance of d'', can be expressed as

$$E_g(d'',t) = \Gamma \frac{E_0 d_0}{d''} \cos\left(\omega_c\left(t - d''/c\right)\right) \quad (3)$$

According to laws of reflection

$$\theta_i = \theta_0 \quad (4)$$

$$E_g = \Gamma E_i \quad (5)$$

$$E_t = (1+\Gamma) E_i \quad (6)$$

where $\Gamma \rightarrow$ reflection co-efficient.

The resultant total E field envelope is given by, $|E_{TOT}| = |E_{LOS} + E_g|$

* The electric field $E_{TOT}(d,t)$ can be expressed as the sum of eqn (2) & (3)

$$E_{TOT}(d,t) = \frac{E_0 d_0}{d'} \cos\left(\omega_c\left(t - \frac{d'}{c}\right)\right) + (-1) \frac{E_0 d_0}{d''} \cos\left(\omega_c\left(t - \frac{d''}{c}\right)\right) \quad (8)$$

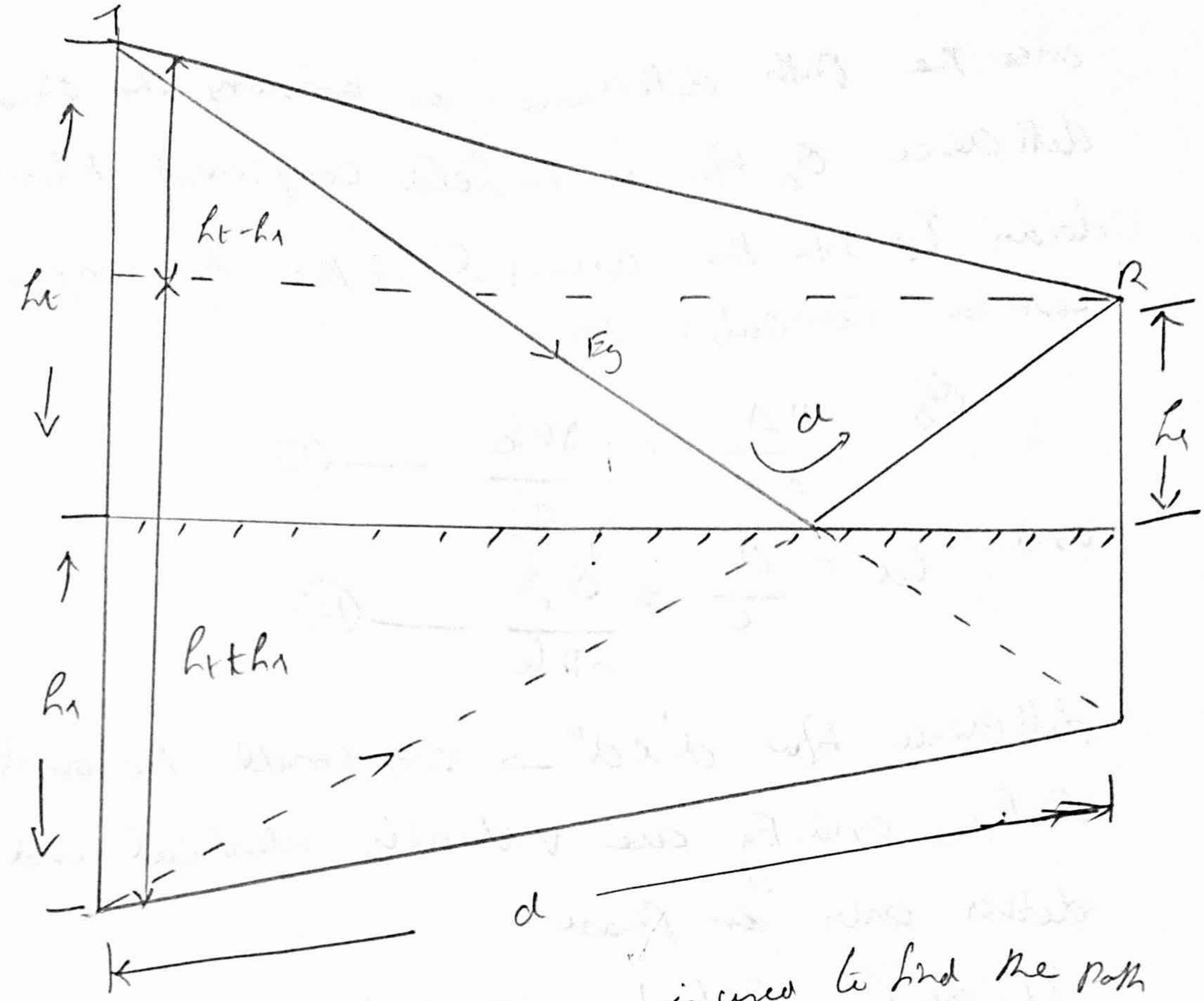


fig: The method of images is used to find the path difference b/w the LOS and the ground reflection paths.

using method of images, the path difference Δ, b/w the LOS and the ground reflected paths can be expressed as

$$\Delta = d'' - d' = \sqrt{(h_t + h_r)^2 + d^2} - \sqrt{(h_t - h_r)^2 + d^2} \quad \text{——} (9)$$

when the T-R separation distance d is very large compared to $h_t + h_r$, eqn (9) can be simplified using a Taylor series approximation

$$\Delta = d'' - d' \approx \frac{2 h_t h_r}{d} \quad \text{——} (10)$$

once the path difference is known, the phase difference θ_Δ b/w to E-field component & Time delay τ_d b/w the arrival of the two components can be computed by,

$$\theta_\Delta = \frac{2\pi\Delta}{\lambda} = \frac{\Delta w_c}{c} \quad \text{--- (11)}$$

and $$\tau_d = \frac{\Delta}{c} = \frac{\theta_\Delta}{2\pi f_c} \quad \text{--- (12)}$$

difference b/w d' & d'' → very small the amplitudes of E_{LOS} and E_g are virtually identical and differ only in phase

$$\left|\frac{E_0 d_0}{d}\right| \approx \left|\frac{E_0 d_0}{d'}\right| \approx \left|\frac{E_0 d_0}{d''}\right| \quad \text{--- (13)}$$

* If the received E-field is evaluated at $t = d''/c$, Eqn (8) can be expressed as a phasor.

$$E_{TOT}(d, t = d''/c) \approx \frac{E_0 d_0}{d}\left[\angle \theta_\Delta - 1\right] \quad \text{--- (14)}$$

Link Budget Design:-

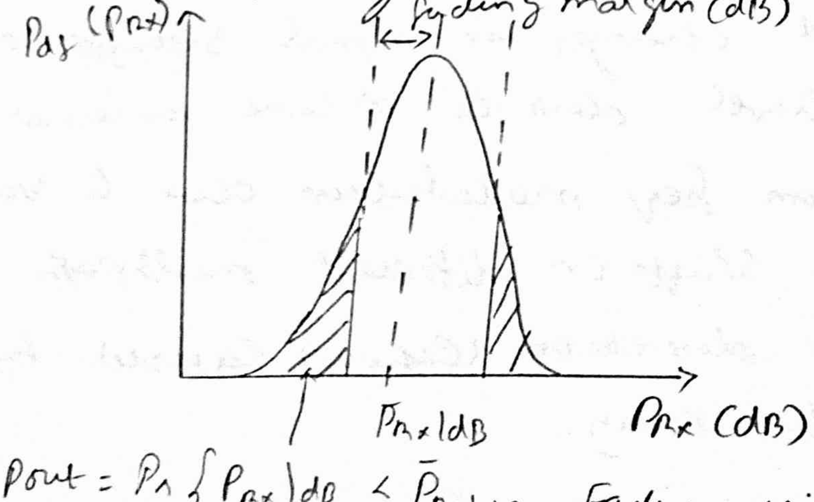
* A link budget is the clearest and most intuitive way of computing the required Tx power
* It tabulates all equations that connect the Tx power to the Received SNR.

* It is convenient to write all the equations in a logarithmic form - specifically in dB.
* The Link budget gives only an approximation for the total SNR, because some interactions b/w different effects are not taken into account.
* For distances $d < d_{break}$, the received power is proportional to d^{-n}, where $n = 3.5$ to 4.5

The received power is thus,

$$P_{Rx}(d) = P_{Rx}(d_{break}) \left(\frac{d}{d_{break}}\right)^{-n} \text{ for } d > d_{break}.$$

* wireless systems, especially mobile systems suffer from temporal and spatial variations of the transmission channel (fading)
* The ratio of the transmit power to this mean received power is also known as the path loss (inverse of the path gain)



$$P_{out} = P_r\{P_{Rx}|dB < \bar{P}_{Rx}|dB - \text{Fading margin}|dB\}$$

fig: fading margin to guarantee a certain outage probability.

* If the mean received power is used as the basis for the link budget, then the transmission quality will be above the threshold only in approximately 50% of the times and locations.

* We have to add a fading margin, the minimum received power is exceeded in atleast eg: 90% of all cases.

* The value of the fading margin depends on the amplitude statistics of the fading.

* uplink (MS to BS) and down link (BS to MS) are reciprocal, the voltage and currents at the antenna ports are reciprocal.

<u>Small Scale</u> multipath propagation:-

* multipath in the radio channel creates Small Scale fading effects.
* The three most important effects are
* Rapid changes in signal strength over a Small travel distance or time interval.
* Random freq. modulation due to varying Doppler shifts on different multipath signals.
* Time dispersion (echoes) caused by multipath propagation Delay.

* Fading occurs because the height of the mobile antennas, So there is no LOS path to the base Station.
* The incoming radio waves arrive from different directions with different propagation delays.
* Even when a mobile Rxer is Stationary the received Signal may fade due to movement of Surrounding objects in the radio channel
* The spatial variations of the resulting Signal are seen as temporal variations by the receiver as it moves through the multipath field.
* Due to Constructive and destructive effect of multipath waves Summing at various points in space, a receiver moving at high speed can pass through Several fades in a small period of time.

Factors influencing <u>Small</u> <u>Scale fading</u>:-

* multipath propagation
* Speed of mobile
* speed of surrounding objects
* The transmission bandwidth of the signal.

Doppler shift :-

* "Due to the relative motion b/w the mobile and the base station, each multipath wave experiences an apparent shift in freq. The shift in received signal freq. due to motion is called doppler shift".

* Doppler shift is directly proportional to the velocity and direction of motion of the mobile.

* Consider a mobile moving at a constant velocity 'v', having length 'd' b/w points x & y while it receives signal from a remote source 'S'.

* Path difference $\Delta L = d\cos\theta = V\Delta t\cos\theta$,

$\Delta t \rightarrow$ time required for the mobile to travel from x to y

* The phase angle change in received signal,

$$\Delta\phi = \frac{2\pi\Delta l}{\lambda} = \frac{2\pi V\Delta t}{\lambda}\cos\theta \quad \text{——— (1)}$$

* Doppler shift is given by f_d,

$$f_d = \frac{1}{2\pi}\cdot\frac{\Delta\phi}{\Delta t} = \frac{V}{\lambda}\cos\theta \quad \text{——— (2)}$$

* If the mobile is moving toward the direction of arrival of the wave the Doppler shift is positive

* If the mobile is moving away from the direction of arrival of the wave the Doppler shift is negative.

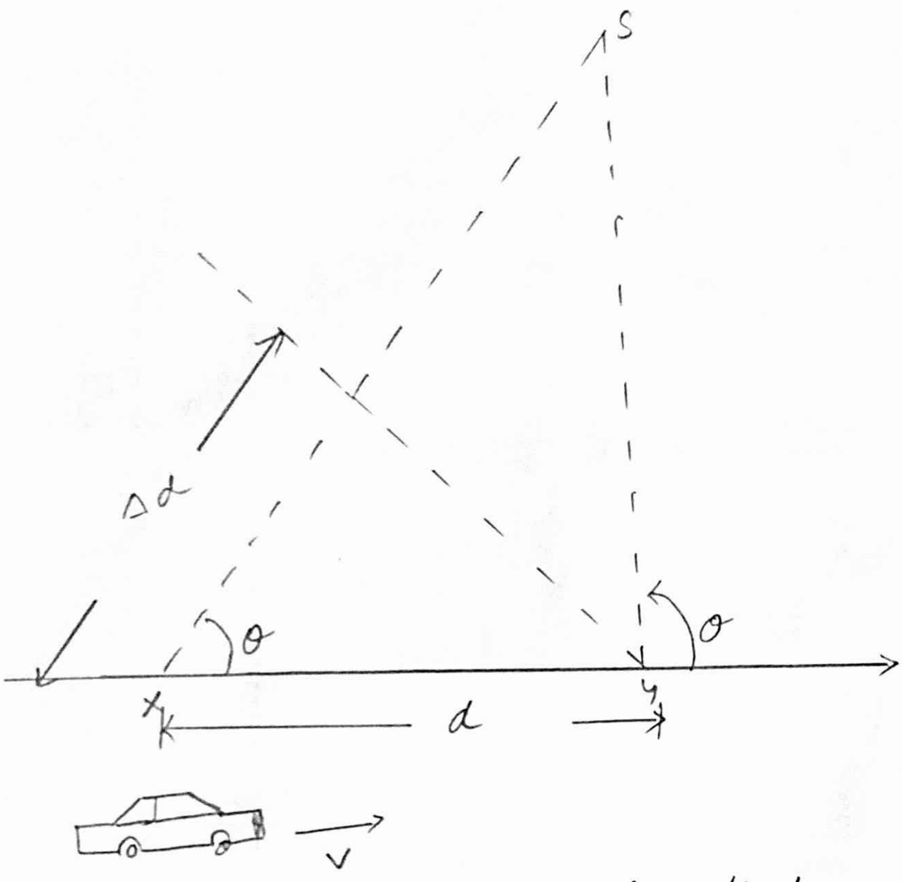
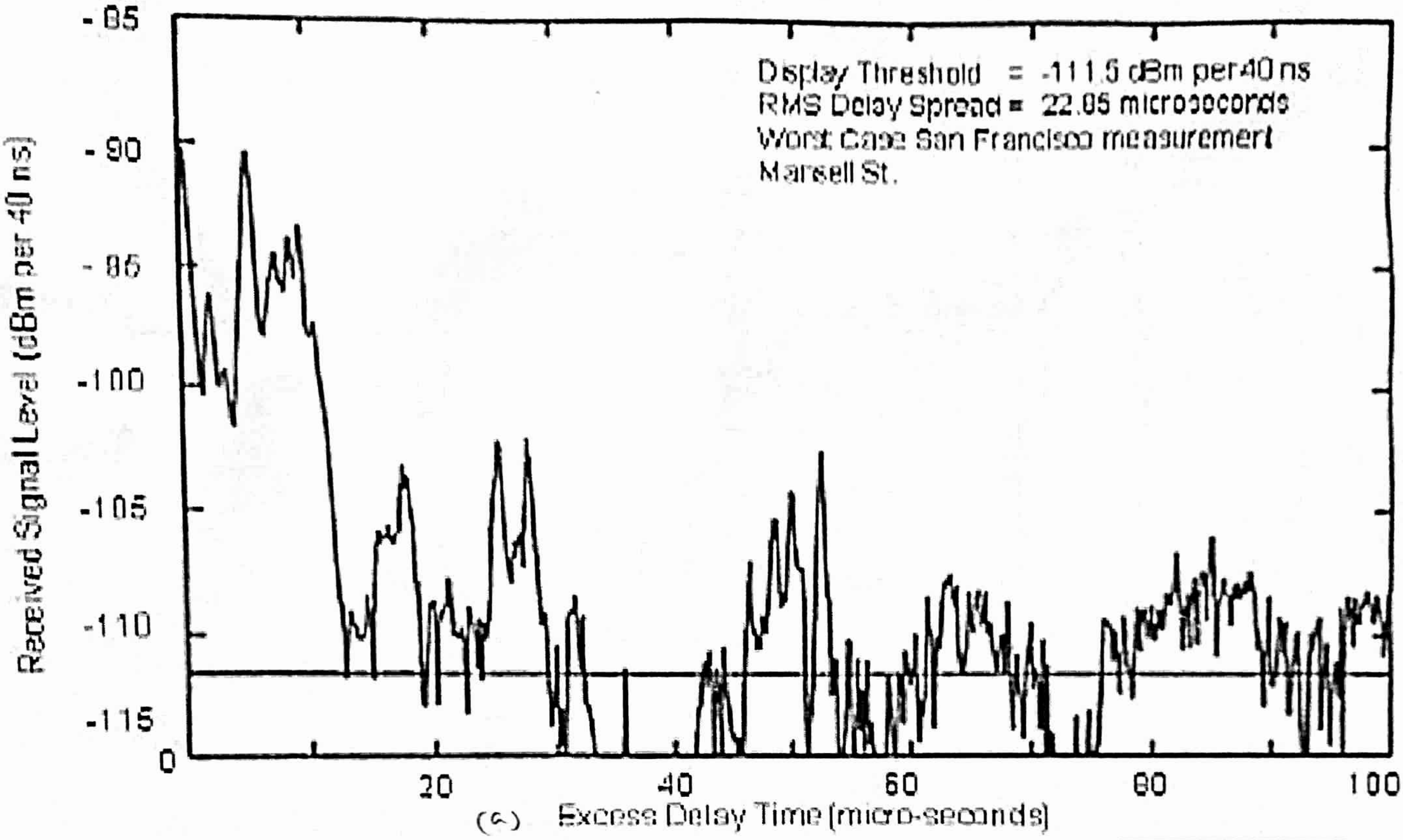


fig: Illustration of Doppler effect.

Parameters of mobile multipath channels:-

* Power delay profiles are found by averaging instantaneous power delay profile measurements over a local area in order to determine an average small scale power delay profile
    * Time Dispersion parameters
    * Coherence Bandwidth.
    * Doppler spread & Coherence Time.

* Small scale sampling avoids large scale averaging bias in the resulting small scale statistics.



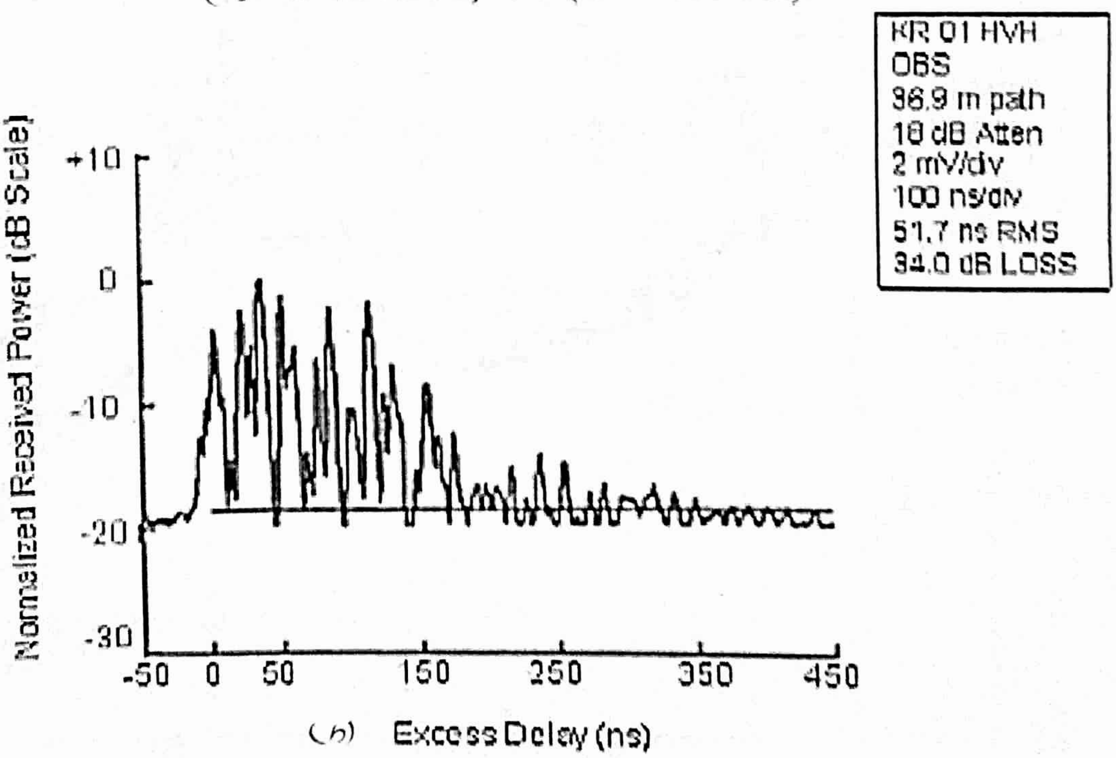


Fig: Measured multipath power delay profiles
(a) From a 900MHz cellular system in San Francisco
(b) Inside a grocery store at 4GHz.

Time Dispersion Parameters:-

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

* The mean excess delay ($\bar{\tau}$) is the first moment of the power delay profile and is defined to be

$$\bar{\tau} = \frac{\sum_K a_K^2 \tau_K}{\sum_K a_K^2} = \frac{\sum_K P(\tau_K)\, \tau_K}{\sum_K P(\tau_K)} \quad \text{— ①}$$

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

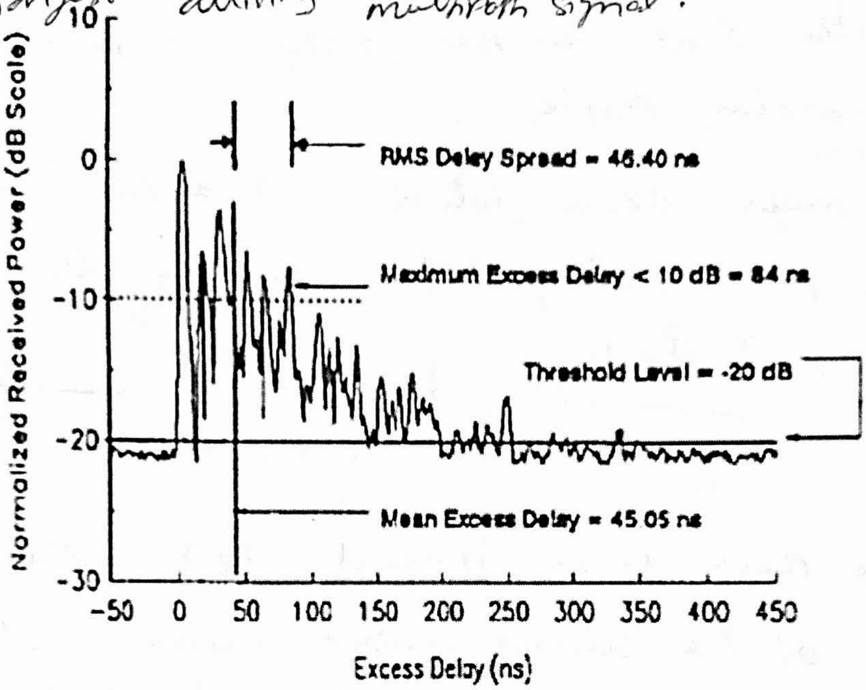
$$\sigma_\tau = \sqrt{\overline{\tau^2} - (\bar{\tau})^2} \quad \text{— ②}$$

where,
$$\overline{\tau^2} = \frac{\sum_K a_K^2 \tau_K^2}{\sum_K a_K^2} = \frac{\sum_K P(\tau_K)\, \tau_K^2}{\sum_K P(\tau_K)} \quad \text{— ③}$$

* Typical values of rms delay spread are on the order of micro seconds in outdoor mobile radio channels and on the order of nano seconds in indoor radio channels.

* The max. excess delay (X dB) of the power delay profile is defined to be the time delay during which multipath energy falls to X dB below the maximum.

* The max. excess delay is defined as, $\tau_x - \tau_0$, where τ_0 is the first arriving signal and τ_x is the max. delay at which a multipath component is within x dB of the strongest arriving multipath signal.



Environment	Frequency (MHz)	Rms delay spread (σ_τ)
urban	892	10 to 25 μs
Suburban	910	200 to 310 ns
Indoors	1500	10 to 50 ns.

* Analogy to the delay spread parameters in the time domain, Coherence Bandwidth is used to characterize the channel in frequency domain.

Coherence Bandwidth:-

* Coherence Bandwidth is a Statistical measure of the range of frequencies over which the channel can be considered "flat".
* Coherence Bandwidth is the range of frequencies over which two frequency components have a strong potential for amplitude correlation.
* If Coherence Bandwidth is defined as the bandwidth over which the frequency correlation fn. is 0.9, then the Coherence Bandwidth is,

$$B_C \approx \frac{1}{50\,\sigma_\tau} \quad \text{--- (1)}$$

* Freq. correlation fn. is above 0.5, then the Coherence Bandwidth is,

$$B_C \approx \frac{1}{5\,\sigma_\tau} \quad \text{--- (2)}$$

Eqn (1) & (2) are "Ball park estimates".

* Spectral analysis techniques and simulation are required to determine the exact impact that time varying multipath has on a particular transmitted signal.

Doppler Spread and Coherence Time:-

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

* when a pure sinusoidal tone of freq. f_c is transmitted, the received signal spectrum called the Doppler spectrum, will have components in the range $f_c - f_d$. where f_d is the Doppler shift.

* If the baseband signal bandwidth is much greater than B_D, the effects of Doppler spread are negligible at the receiver.

* Coherence Time T_c is the time domain dual of Doppler spread and is used to characterize the time varying nature of the freq. dispersiveness of the channel in the time domain.

* The Doppler spread and coherence time are inversely proportional to one another.

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

* Coherence time is the time duration over which two received signals have a strong potential for amplitude correlation.

* If the coherence time is defined as the time over which the time correlation function is above 0.5, then coherence time is,

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

where $f_m \rightarrow$ max. Doppler shift.

$$f_m = v/\lambda$$

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

* The definition of coherence time implies that two signals arriving with a time separation greater than T_c are affected differently by the channel.

Types of Small Scale Fading:-

While multipath delay spread leads to time dispersion and frequency selective fading Doppler spread leads to frequency dispersion and time selective 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)

- **Fast Fading**
  1. High Doppler spread
  2. Coherence time < Symbol period
  3. Channel variations faster than base-band signal variations
- **Slow Fading**
  1. Low Doppler spread
  2. Coherence time > Symbol period
  3. Channel variations slower than baseband signal variations

Fig: Types of small scale fading.

Fading effects due to multipath time delay spread:-

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

Flat fading:-

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

* Spectral Characteristics of the transmitted signal are preserved at the receiver.
* The strength of the received signal changes with time, due to fluctuations in the gain of the channel caused by multipath.
* Over time, the received signal r(t) varies in gain, but the spectrum of the transmission is preserved.

* The Reciprocal Bandwidth is much Larger than the multipath time delay spread of the Channel.

* Flat fading channels are also known as amplitude varying channels and are sometimes are referred to as narrow band channels, since the Band width of the applied signal is narrow as compared to the channel flat fading bandwidth

A signal undergoes flat fading if,

$$B_S << B_C \quad —①$$

and

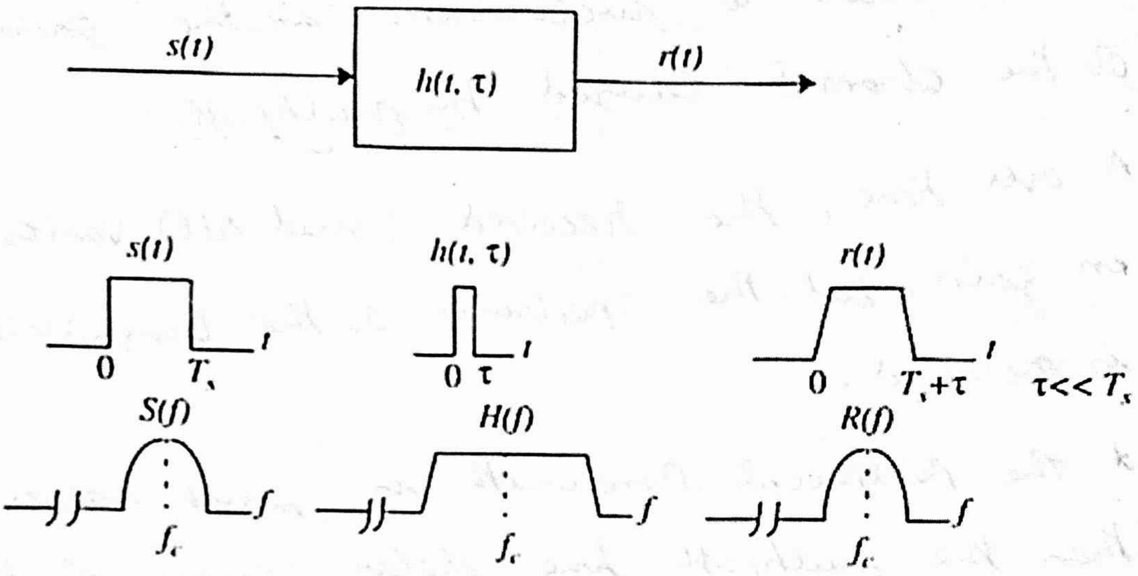
$$T_S >> \sigma_\tau \quad —②$$

where, $T_S \rightarrow$ Reciprocal Bandwidth

$B_S \rightarrow$ Band width

σ_τ & $B_C \rightarrow$ Rms delay spread & coherence Bandwidth.

fig: flat fading characteristics.



Frequency Selective fading:-

* "If the channel possess a constant gain and linear phase response over a bandwidth that is smaller than the bandwidth of transmitted signal, then the channel creates freq. selective fading on the received signal".

* Freq. selective fading is due to time dispersion of the transmitted symbols within the channel.

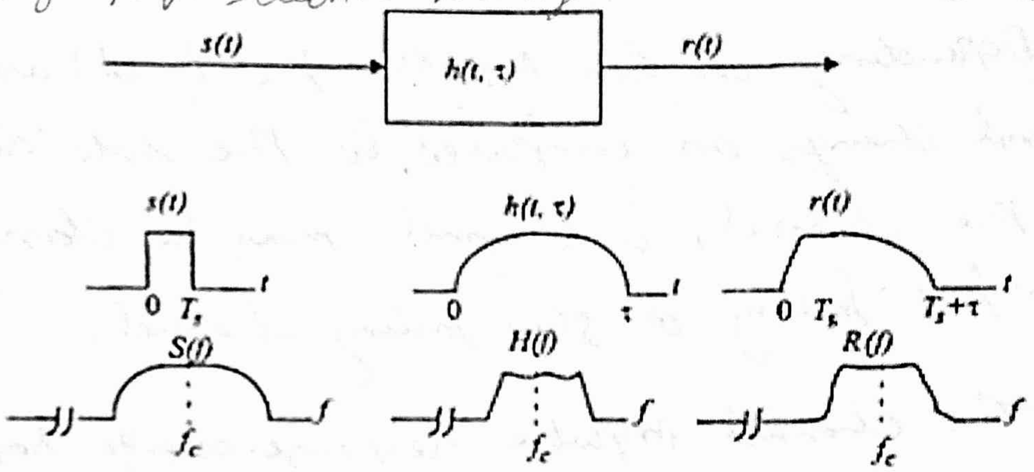
* thus the channel induces Intersymbol interference (ISI)

* Freq. selective fading channels are much more difficult to model than flat fading channels.

* when analyzing mobile comm. systems, statistical impulse response models such as the two-ray Rayleigh fading model (or)

Computer generated or measured impulse responses are generally used for analyzing freq. selective small scale fading.

fig. freq. selective fading channel characteristics.



* For freq. selective fading, the spectrum S(f) of the transmitted signal has a bandwidth which is greater than the coherence bandwidth B_c of the channel.

* ~~A freq.~~ signal undergoes freq. selective fading

$$B_S > B_c \quad \text{—①}$$

and

$$T_s < \sigma_\tau \quad \text{—②.}$$

* A channel is flat fading if $T_s \geq 10\sigma_\tau$ and a channel is freq. selective if $T_s < 10\sigma_\tau$.

Fading Effects Due to Doppler Spread:-

* Fast fading
* Slow fading.

Fast fading:-

* Depending on how rapidly the txed base band signal changes as compared to the rate of change of the channel, a channel may be classified as fast fading or slow fading channel.

* The channel impulse response changes rapidly within symbol duration.
* Coherence time of the channel is smaller than the symbol period of the transmitted signal.
* This causes freq. dispersion due to Doppler spreading, which leads to signal distortion.
* A signal undergo fast fading if

$$T_S > T_C$$

and $B_S < B_D$.

* Fast fading only deals with the rate of change of the channel due to motion.
* The amplitude of the delta fn. varies faster than the rate of change of the txed baseband signal.
* Fast fading only occurs for very low data rate.

Slow fading:-

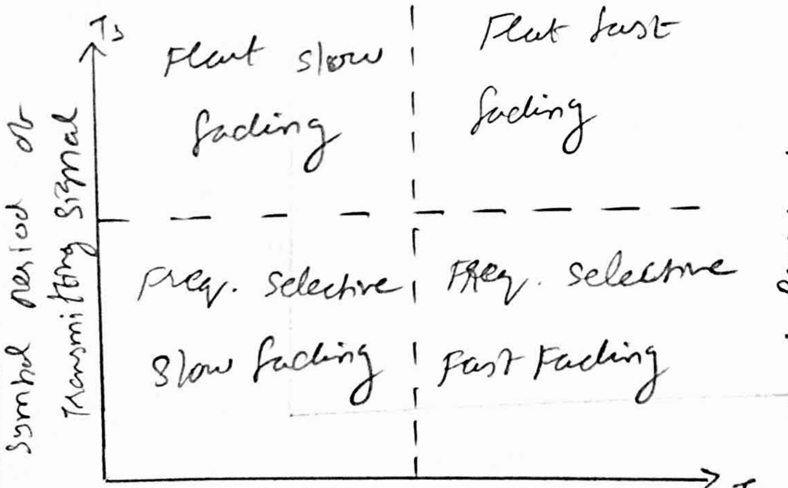
* In a slow fading channel, the channel impulse response changes at a rate much slower than the transmitted base band signal s(t).
* In freq. domain, the Doppler spread of the channel is much less than the Bandwidth of the base band signal
* A signal undergoes slow fading if

$$T_s \ll T_c$$

and $B_s \ll B_D$

* The velocity of the mobile and the base band signaling determines whether a signal undergoes fast fading or slow fading.

(a)



(b)

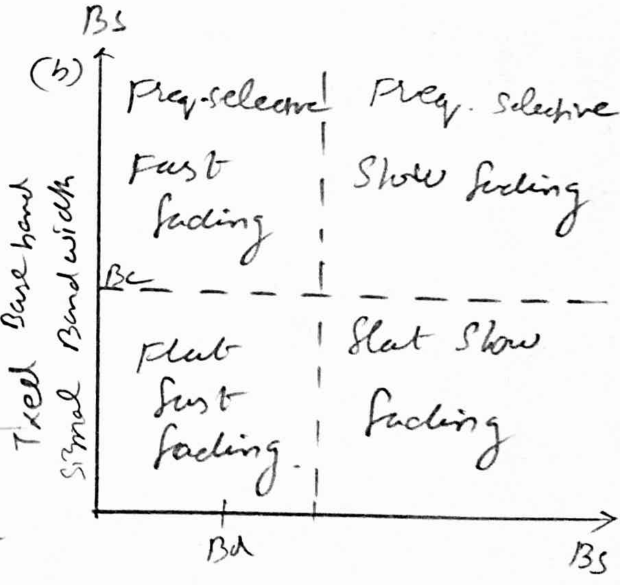


Fig: matrix illustrating type of fading experienced by a signal as a function of (a) symbol period (b) base band signal bandwidth.

Unit II

Cellular Architecture.

Multiple Access Techniques - FDMA, TDMA, CDMA - Capacity Calculations - Cellular concept - Frequency reuse - channel assignment - hand off - Interference & System capacity - Trunking & Grade of service - Coverage and capacity Improvement.

Multiple Access Techniques:-

"Multiple Access Schemes are used to allow many users to Share simultaneously a finite amount of radio Spectrum".

* The Sharing of spectrum is required to achieve high Capacity by simultaneously allocating the available Bandwidth to multiple users.
* "It is possible to talk and Listen simultaneously and this effect" is called duplexing.
* Duplexing may be done using freq. or time domain techniques.
* Freq. division duplexing provides two distinct bands of frequencies for every user.
* forward band - provides traffic from the BS to MS.
* reverse band - provides traffic from the MS to BS.

* Time division duplexing (TDD) uses time instead of freq. to provide both forward & Reverse Link.

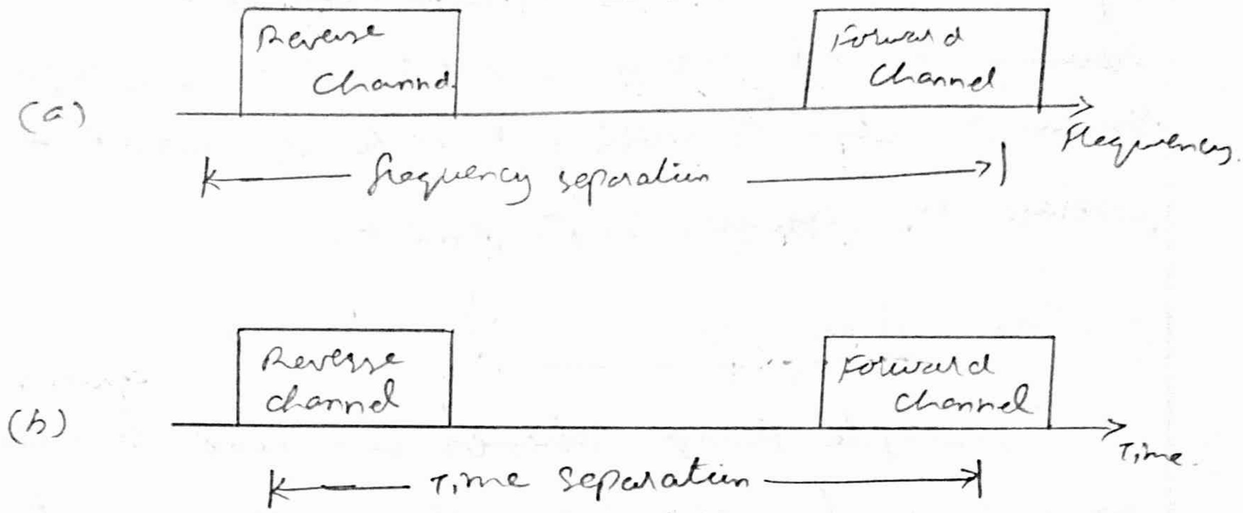


fig (a) FDD (b) TDD.

* There are 3 major access techniques used to share the available bandwidth in a wireless communication

1. Frequency division multiple Access (FDMA)
2. Time division multiple Access (TDMA)
3. Code division multiple Access (CDMA)

These techniques can be grouped as narrow band and wideband systems. depending upon how the available bandwidth is allocated to the users.

Frequency Division multiple Access (FDMA):-

Freq. division multiple Access (FDMA) assigns individual channels to individual users.

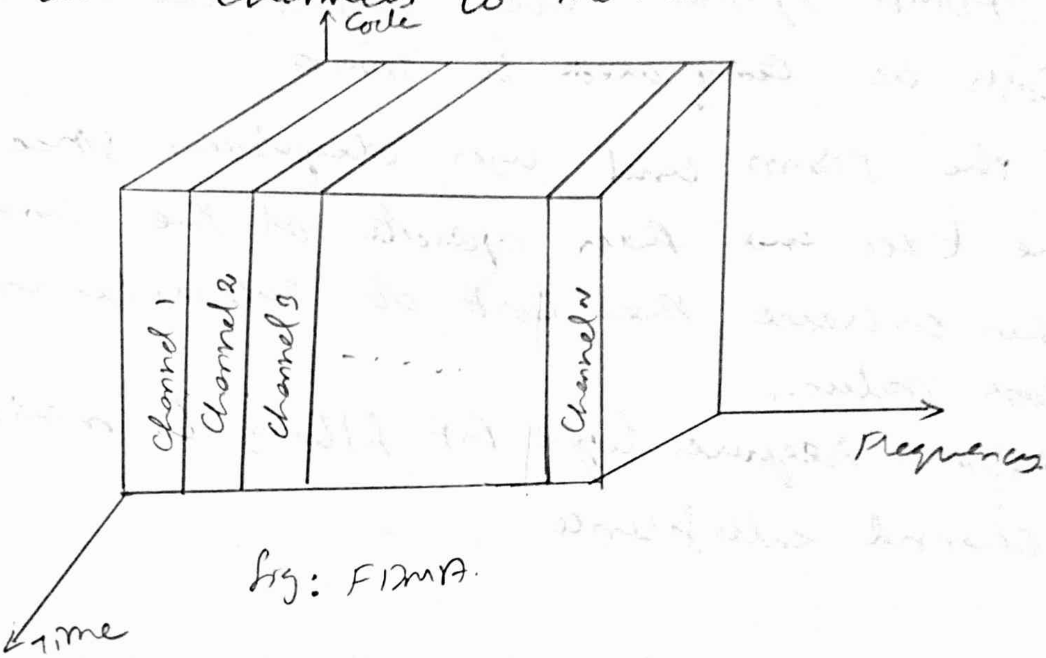


fig: FDMA.

* During the period of the call, no other user can share the same channel.
* In FDD systems, the users are assigned a channels as a pair of frequencies.
    - one freq. is used for forward channel
    - other freq. is used for Reverse channel.

Features of FDMA:-

* The FDMA channel carries only one phone circuit at a time.
* After the assignment of a voice channel, the BS and the mobile transmit simultaneously and continuously.
* The Bandwidth of FDMA channels are narrow (30KHZ in AMPS).

* The amount of ISI is Low.
* No Equalization is required in FDMA hallow systems.
* FDMA System have higher cell site system costs as compared to TDMA.
* The FDMA unit uses duplexers since both the Txer and Rxer operate at the same time. This increase the cost of subscriber unit and Base station.
* FDMA requires tight RF filtering to minimize channel interference.

Non Linear effects in FDMA:-

* Many channels share the same antenna of the BS.
* The Non-Linearities cause signal spreading in the freq. domain and generate Inter modulation freq.
* The first US analog cellular system, the AMPS is based on FDMA / FDD.
* The no. of channels that can be simultaneously supported in a FDMA System is given by.

$$N = \frac{B_t - 2B_{guard}}{B_c} \quad \text{---} \quad (1).$$

where $B_t \rightarrow$ Total spectrum allocation.

$B_{guard} \rightarrow$ Guard Band

$B_c \rightarrow$ Channel Bandwidth.

Time Division multiple Access (TDMA):-

* "Time Division multiple Access (TDMA) System divide the Radio spectrum into time slots, and in each slot only one user is allowed to either transmit or Receive".

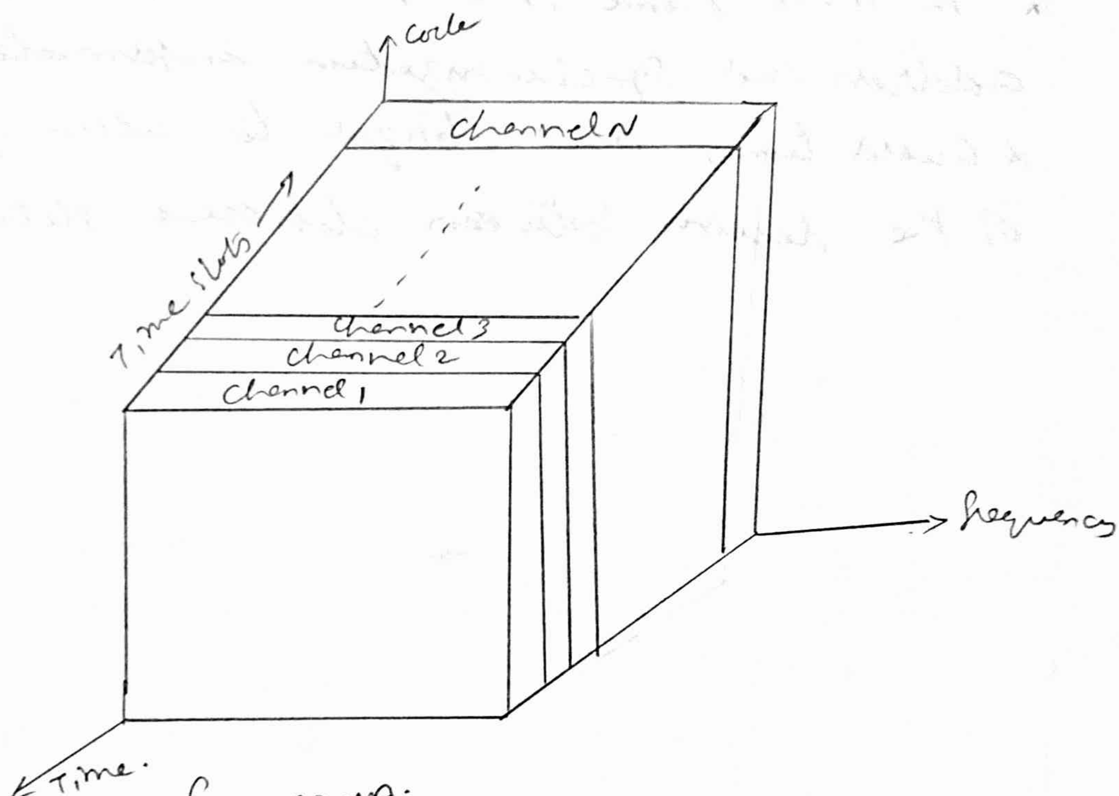


fig: TDMA.

* N time slots comprise a frame.
* TDMA systems transmit data in a buffer and burst method, thus the transmission for any user is non-continuous.
* A frame consist of a number of slots.
* Each frame is made up of a preamble an information, message and tail bits.

* In TDMA/TDD, half of the time slots in frame information message would be used for forward link channels and half would be used for reverse channels.
* In TDMA frame, the preamble contains the address and synchronization information.
* Guard times are utilized to allow synchronization of the receiver between different slots and frames.

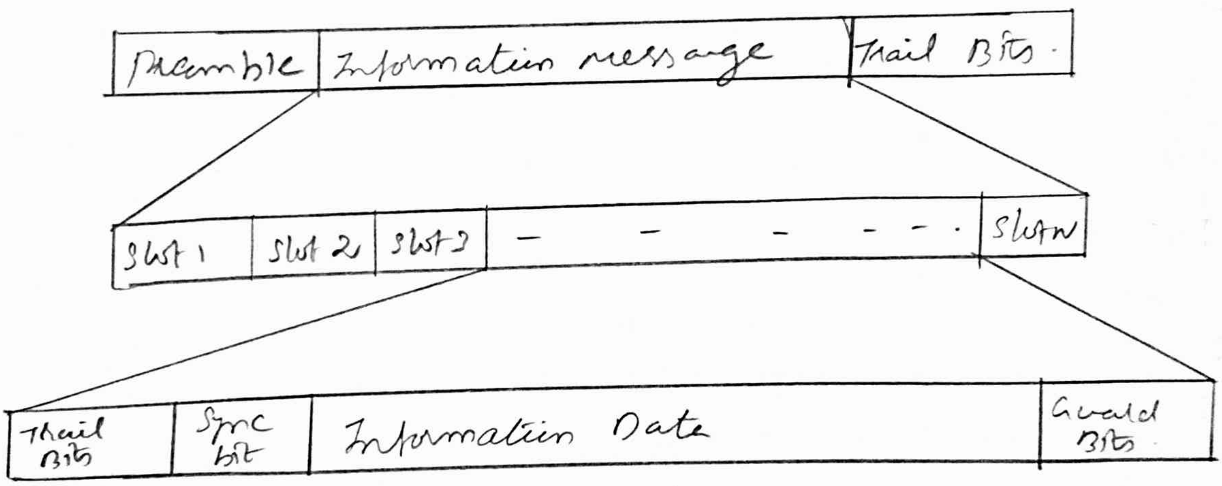


fig. TDMA frame structure.

Features of TDMA:-
* TDMA shares a single carrier freq. with several users, where each user makes use of non-overlapping time slots.
* Data transmission for users of a TDMA system is not continuous, but occurs in bursts.
* TDMA uses different time slots for trans & recep thus duplexers are not required.

* An enhanced link control, such as that provided by mobile assisted hand off (MAHO) can be carried out by a subscriber by listening on an idle slot in the TDMA frame.
* In TDMA, guard time should be minimized
* High synchronization overhead is required in TDMA systems because of burst transmission.
* TDMA has an advantage in that it is possible to allocate different numbers of time slots per frame to different users.

Efficiency of TDMA:-

* "The efficiency of a TDMA system is a measure of the percentage of trued data that contain information as opposed to providing overhead for the access scheme"
* The frame efficiency η_f, is the percentage of bits per frame which contain trued data".
* The number of overhead bits per frame

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

where,

$N_r \rightarrow$ Number of reference burst per frame.

$N_t \rightarrow$ Number of traffic bursts per frame.

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

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

$b_g \rightarrow$ No. of equivalent bits ~~per preample~~ in each guard time interval. ~~each slot~~

* The total number of bits per frame, b_T is

$$b_T = T_f R$$

where $T_f \rightarrow$ frame duration

$R \rightarrow$ channel bit rate.

* The frame efficiency η_f is,

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

Number of channels in TDMA System:-

The number of TDMA channel slots that can be provided in a TDMA is found by multiplying the number of TDMA slots per channel by the no. of channels available is given by,

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

where $m \rightarrow$ max. number of TDMA users supported on each radio channel.

Code division multiple Access (CDMA):-

* In code division multiple access (CDMA) systems, the narrow band message signal is multiplied by a large bandwidth signal called the spreading signal.
* The spreading signal is a pseudonoise code sequence that has a chip rate which is orders of magnitudes greater than the data rate of the message

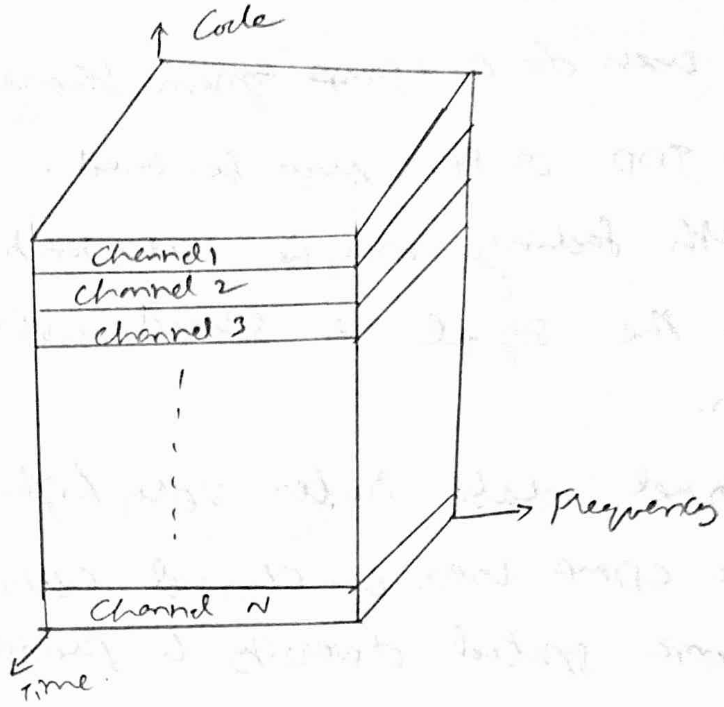


Fig: spread spectrum multiple access.

* Each user has its own pseudorandom code word which is approximately orthogonal to all other code words.
* The receiver performs a time correlation operation to detect only the specific desired code word.
* All other code words appear as noise due to decorrelation.
* To combat the near far problem, power control is used in most CDMA implementations.
* Power control is implemented at the base station by rapidly sampling the radio signal strength indicator (RSSI) levels of each mobile and then sending a power change command over the forward radio link.

Features of CDMA :-

* Many users of a CDMA System share the same freq.
* Either TDD or FDD may be used.
* multipath fading may be substantially reduced because the signal is spread over a large spectrum.
* Channel data rates very high in CDMA Systems.
* Since CDMA uses co-channel cells, it can use macroscopic spatial diversity to provide soft handoff.
* self jamming is a problem in CDMA System.
* the near far problem in CDMA System occurs CDMA receiver if an undesired user has a high detected power as compared to the desired user.

Capacity Calculations :-

* "Channel capacity for a radio system can be defined as the max. number of channels or users that can be provided in a fixed freq. band".
* "Radio capacity is a parameter which measure spectrum efficiency of a wireless system. This parameter is determined by the required C/I and the channel Bandwidth Bc.
* The interference at a BS Rxer will come from the subscriber units in surrounding cells - Reverse channel interference.

* The desired BS will provide the desired forward channel while the surrounding co-channel BS will provide the forward channel interference.

Let $D \rightarrow$ Distance b/w two co channel cells.

$R \rightarrow$ Cell Radius.

* The min. ratio of D/R that is required to provide a tolerable level of co-channel interference is called cochannel reuse ratio

$$Q = D/R \quad \text{——} (1)$$

* M closest co-channel cells may be considered as first order interference in which case C/I is given by,

$$\frac{C}{I} = \frac{D_0^{-n_0}}{\sum_{K=1}^{M} D_K^{-n_K}} \quad \text{——} (2)$$

where $n_0 \rightarrow$ path loss exponent in the desired cell.

$D_0 \rightarrow$ distance from the desired BS to mobile

$D_K \rightarrow$ Distance of the K^{th} cell from the mobile.

$n_K \rightarrow$ Path loss exponent to the K^{th} interfering BS.

* If only the six closest interfering cells are considered, then C/I is given by,

$$\frac{C}{I} = \frac{D_0^{-n}}{6D^{-n}} \quad \text{——} (3).$$

Carrier-to-Interference Ratio

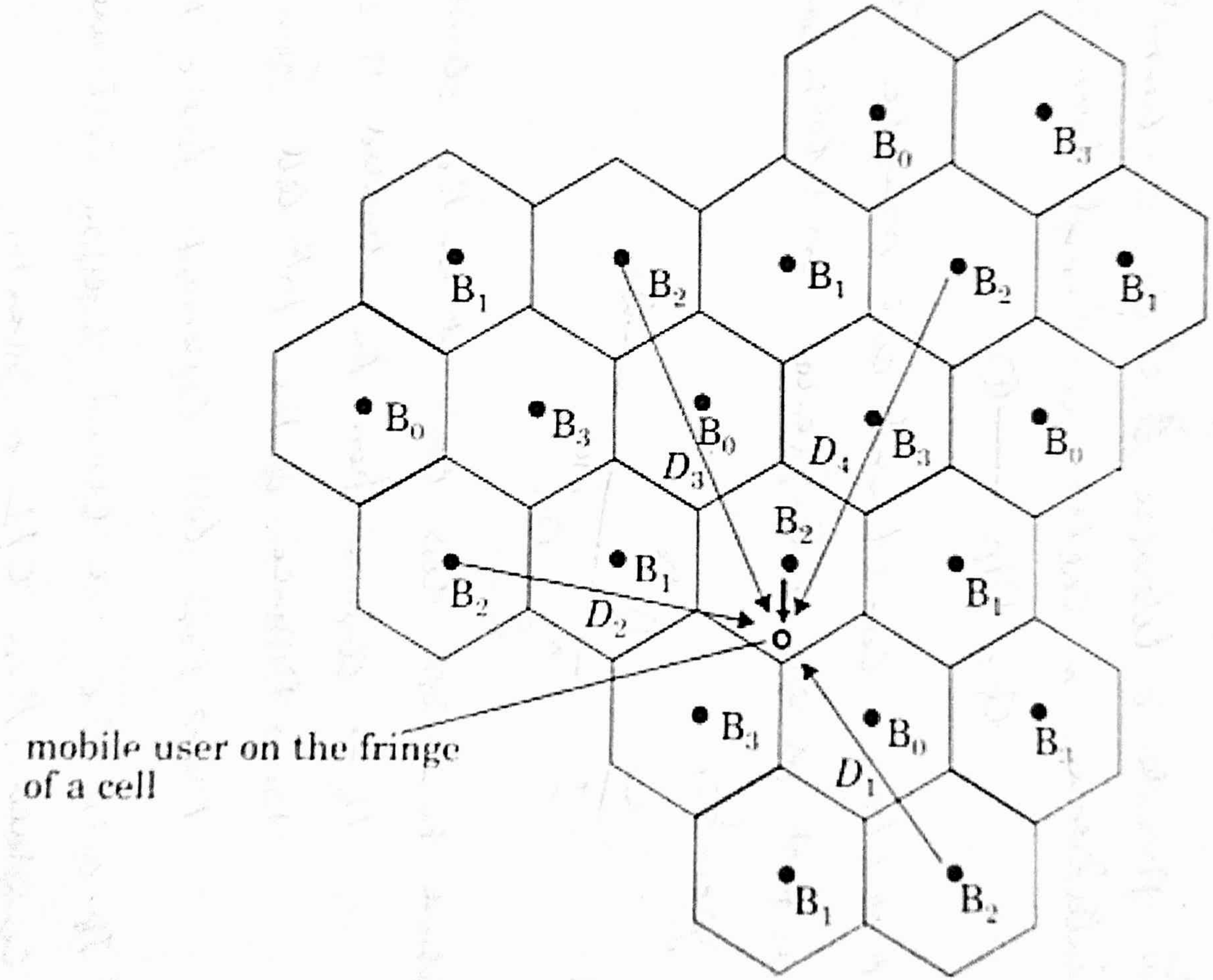


Figure 9.11 Illustration of forward channel interference for a cluster size of $N = 4$. Shown here are four co-channel base stations which interfere with the serving base station. The distance from the serving base station to the user is D_0, and interferers are a distance D_k from the user.

* The following eqn must hold for acceptable performance.

$$\frac{1}{6}\left(\frac{R}{D}\right)^{-n} \geq \left(\frac{C}{I}\right)_{min} \quad (4)$$

Cochannel reuse factor is,

$$Q = \left(6\left(\frac{C}{I}\right)_{min}\right)^{1/n} \quad (5)$$

The radio capacity of a cellular system is defined as,

$$m = \frac{B_t}{B_c N} \text{ radio channels / cell} \quad (6)$$

m → radio capacity metric,

B_t → total allocated spectrum for the system.

B_c → Channel Bandwidth

N → No. of cells in a freq. reuse pattern

The cochannel reuse factor Q by,

$$Q = \sqrt{3N} \quad (7)$$

from (5) (6) (7) the radio capacity is,

$$m = \frac{B_t}{B_c \frac{Q^2}{3}} = \frac{B_t}{B_c \left(\frac{6}{3^{n/2}}\left(\frac{C}{I}\right)_{min}\right)^{2/n}} \quad (8)$$

where n=4, the radio capacity is given by

$$m = \frac{B_t}{B_c \sqrt{\frac{2}{3}\left(\frac{C}{I}\right)_{min}}} \text{ radio channels/cells} \quad (8)$$

$$\left(\frac{C}{I}\right)_{eq} = \left(\frac{C}{I}\right)_{min} \left(\frac{B_c}{B_i}\right)^2 \quad \text{---} \quad (9)$$

Capacity of Digital cellular TDMA:-

* TDMA Systems improve capacity by a factor of three to six times as compared to analog cellular radio systems.
* TDMA also makes it possible to introduce Adaptive channel Allocation (ACA).
* ACA eliminates system planning since it is not required to plan frequencies.

Capacity of Cellular CDMA:-

* The capacity of CDMA systems is interference limited, while it is bandwidth limited in FDMA & TDMA.
* The link performance for each user increases as the no. of users decreases.
* Another way of increasing CDMA capacity is to operate in a discontinuous transmission mode (DTX).
* The average capacity of a CDMA system can be increased by a factor inversely proportional to the duty factor.
* CDMA can reuse the entire spectrum for all cells and this results in an increase of capacity by a large percentage over the normal freq. reuse factor.

the signal to noise ratio is

$$SNR = \frac{S}{(N-1)S} = \frac{1}{N-1} \quad \text{—— (10)}$$

R → base band information bit rate,

W → Total RF band width, w.

$$E_b/N_o = \frac{S/R}{(N-1)(S/W)} = \frac{W/R}{N-1} \quad \text{—— (11)}$$

η → spread bandwidth.

+ Eb/No can be represented as.

$$\frac{E_b}{N_o} = \frac{W/R}{(N-1)+(\eta/S)} \quad \text{—— (12)}$$

the no of users that can access the system is given as,

$$N = 1 + \frac{W/R}{E_b/N_o} + (\eta/S) \quad \text{—— (13)}$$

where, W/R is called the processing gain.

the back ground noise determines the cell radiation for a given transmitter power.

two techniques to reduce interference,

(1) Antenna sectorization.

(2) monitoring of voice activity (α)

N_s → No. of users per sector.

α the new average value of E_b/N_o' within a sector is given as,

$$E_b/N_o' = \frac{W/R}{(N_s-1)\alpha + (\eta/S)} \quad \text{—— (14)}$$

When the no. of users is large and the system is interference limited rather than noise limited, the no. of users can be shown to be,

$$N_s = 1 + \frac{1}{\alpha}\left[\frac{W/R}{E_b/N_0}\right] \quad \text{—— (15)}$$

Cellular Concept:-

Some of the important cellular concepts are,

(a) freq. reuse
(b) channel Assignment.
(c) Hand off
(d) Interference and System capacity.
(e) Trunking and Grade of service (GOS)
(f) Improving coverage and capacity.

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 and allocating channel groups for all of the cellular base stations within a system is called freq. reuse or freq. planning.
* The actual radio coverage of a cell is known as the foot print and is determined from field measurements or propagation prediction models.

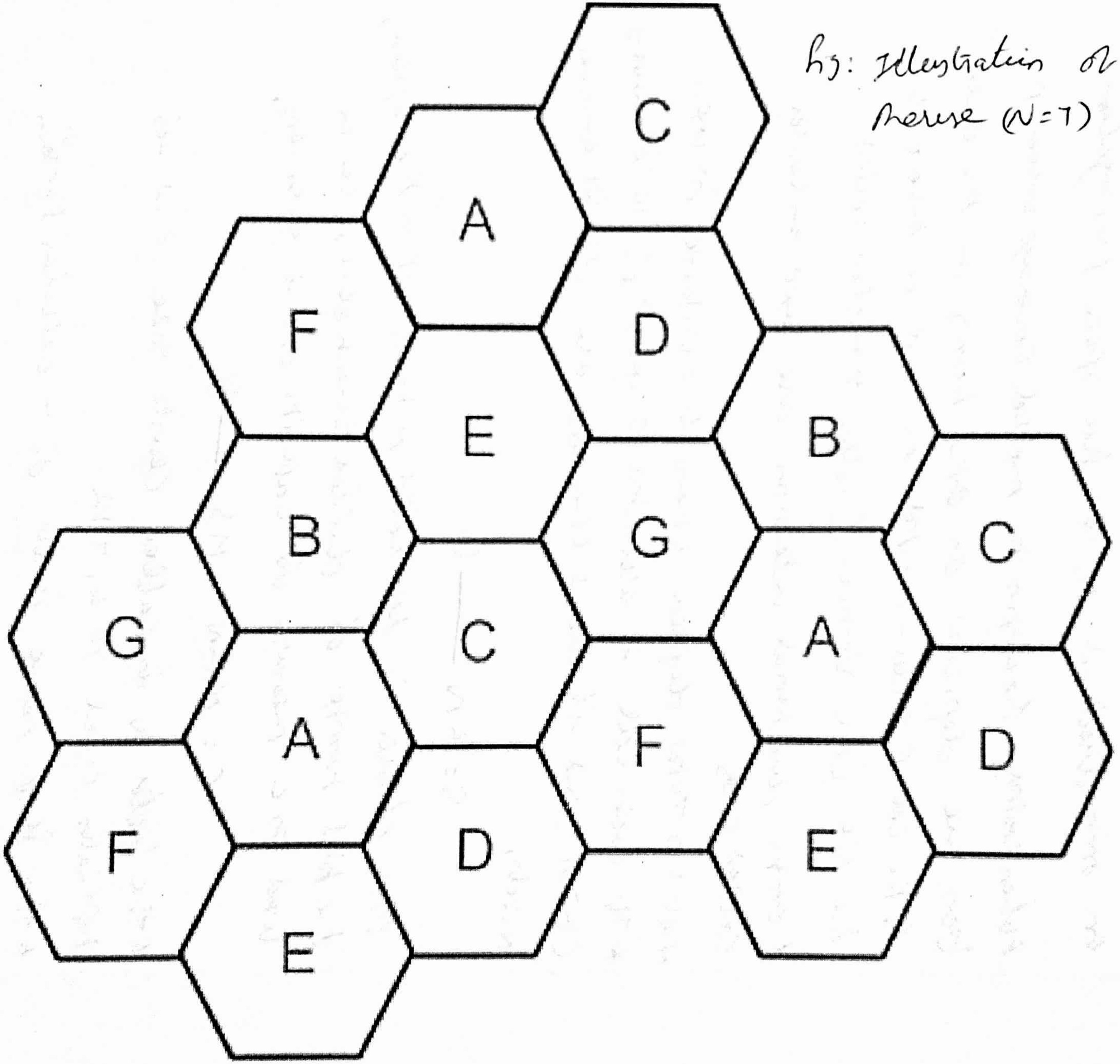


fig: Illustration of frequency reuse (N=7)

* The fewer no of cells can cover a geographic region and the hexagon closely approximates a circular radiation pattern would occur for an omni directional BS and free space propagation.

* When using hexagons to model coverage areas BSters are depicted as either being in the center of the cell (center-excited cells) or on three of the six cell vertices (Edge excited cells).

* omni directional antennas are used in center excited cells.

Let S -> Total duplex channels available for use.

* If each cell is allocated a group of K channels (K>S), and if the S channels are divided among N cells,

$$S = KN \quad \text{——} \quad (1)$$

If a cluster is replicated M times within the system, the total number of duplex channels C, can be used as a measure of capacity and is given by,

$$C = MKN = MS \quad \text{——} \quad (2)$$

* The factor N is called cluster size and is typically equal to 4, 7, 12.

* The freq. reuse factor of a cellular system is given by 1/N.

* The number of cells per cluster, N can only have values which satisfy the equation,

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

where i and j are non-negative integers.

To find the nearest co-channel neighbours of a particular cell.

① Move i cells along any chain of hexagons.

② Turn 60 degrees counter clockwise and move j cells.

Channel Assignments :-

For efficient utilization of the radio spectrum a frequency reuse scheme that is consistent with the objectives of increasing capacity and min. interference is required.

Channel Assignment.

- Fixed channel Assignment.
- Dynamic channel Assignment.

* The choice of channel assignment strategy impacts the performance of the system.
* Particularly as to how calls are managed when a mobile user is hand off from one cell to another.

Fixed channel Assignment :-

* Each cell is allocated a predetermined set of voice channels.

* Borrowing Strategy — a cell is allowed to borrow channels from a neighboring cell if all of its own channels are already occupied.

* The mobile switching center (MSC) supervises borrowing procedures.

Dynamic Channel Assignment:-

* voice channels are not allocated to different cells permanently.

* Each time a call request is made, the serving BS requests a channel from the MSC.

* Dynamic channel assignment reduce the likelihood of blocking, which increases the trunking capacity of the system.

* Dynamic channel assignment strategies require the MSC to collect real time-data on channel occupancy, traffic distribution and Radio signal strength indications (RSSI) of all channels on a continuous basis.

Hand off ~~stag~~ Strategies:-

* When a mobiles 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 basestation". This is called handoff.

* This handoff operation not only involves identifying a new BS, but also requires that the voice and control signal.

System designers must specify an optimum signal level at which to initiate a handoff.

Threshold level, $\Delta = P_{r\,handoff} - P_{r\,min.\,usable}$

* 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.
* The length of time needed to decide if a handoff is necessary depends on the speed at which the vehicle is moving.

Dwell Time:-

The time over which a call may be maintained within a cell, without handoff is called the dwell time.

The factors that influence Dwell time are

1. Propagation
2. Interference
3. Distance.

MAHO:-

In second generation system, handoff are mobile assisted. In mobile Assisted handoff (MAHO) every mobile station measures the received power from surrounding BS and continually reports the results of these measurement to the serving base station.

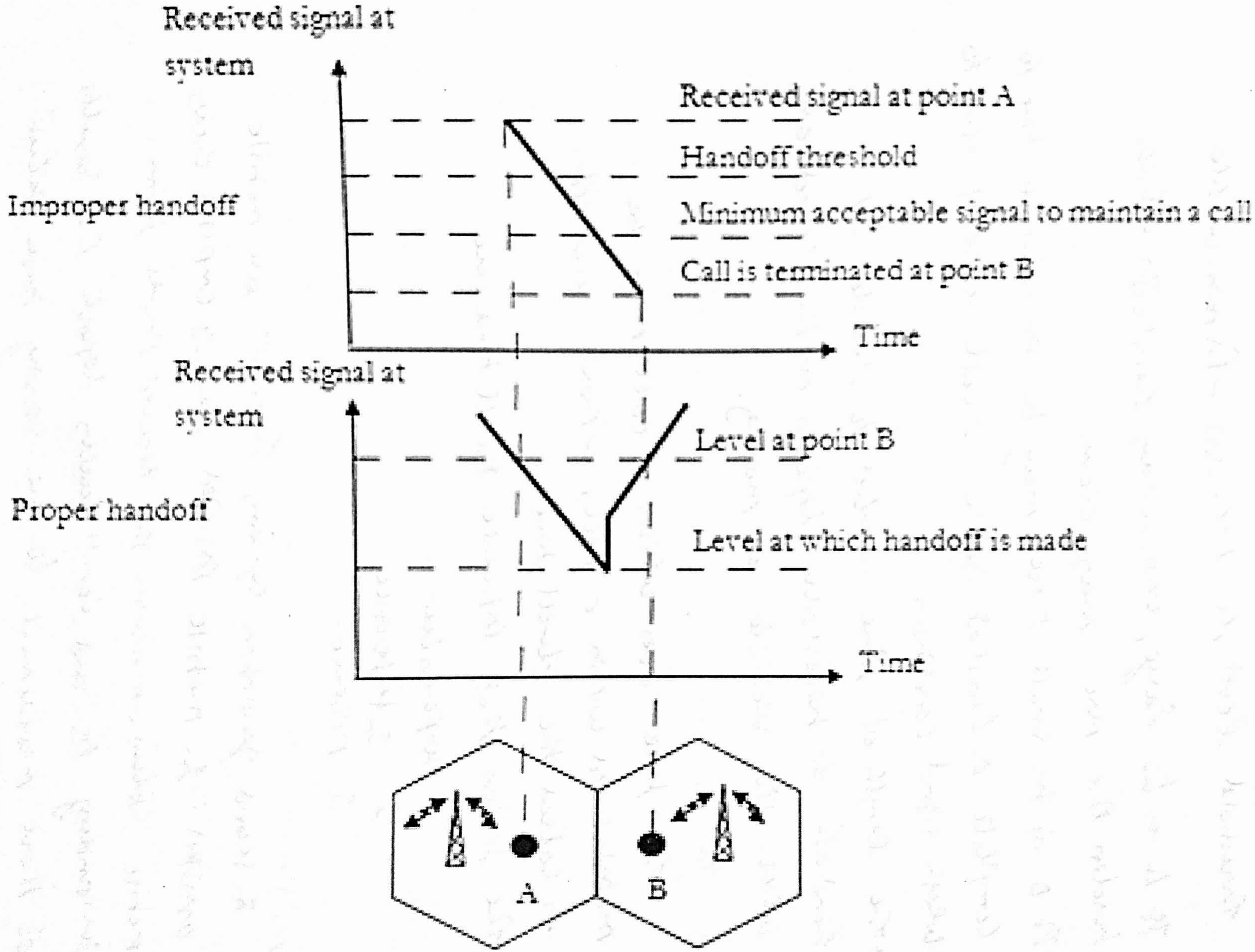


Fig: Proper and improper

Minimizing Hand offs :-

* Guard channel concept.

* Queuing of hand off request.

* Guard channel concept → A fraction of the total available channels in a cell is reserved exclusively for hand off requests from ongoing calls which may be handed off into the cell.

Practical Hand off Considerations :.

Umbrella cell approach:-

By using different antenna heights and different power levels, it is possible to provide "large" and "small" cells which are co-located at a single location. This technique is called umbrella cell approach.

* The number of handoffs is minimized for high speed users. and provides additional microcell channels for pedestrian users.

* The speed of each user may be estimated by the base station or MSC.

Cell dragging:-

* Cell dragging results from pedestrian users that provide a very strong signal to the base station.

* Cell dragging occurs when there is a LOS b/w the subscriber and the BS.

Umbrella Cells

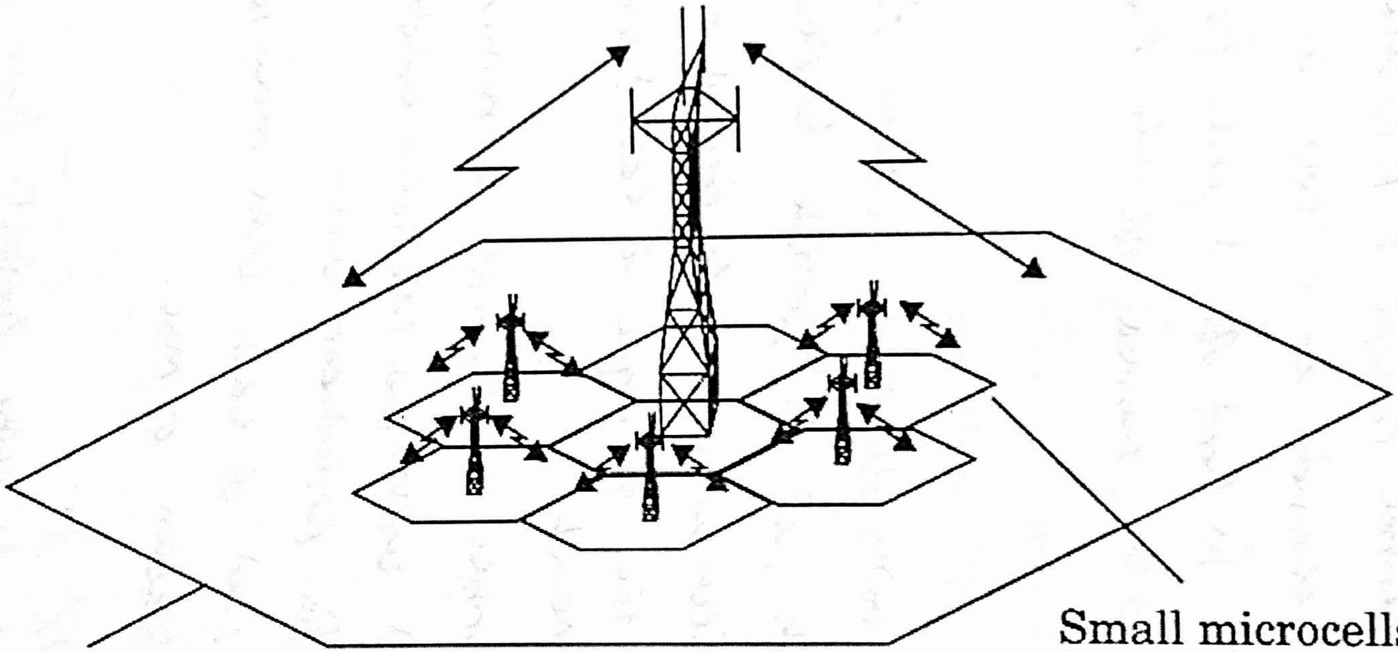


Figure 3.4 The umbrella cell approach.

* to solve the cell dragging problem, handoff thresholds and radio coverage parameters must be adjusted carefully.

Hard and Soft handoff:-

* Assign different radio channels during a hand off is called hard handoff.
* The ability to select b/w the instantaneous received signals from a variety of base stations is called soft handoff.

Interference & System Capacity:-

Interference is the major limiting factor in the performance of cellular radio systems.

Sources of interference:-

* Signal generated from other mobile.
* Call that is in progress in the neighbouring cell.
* BS operating in the same freq. range.
* Cross talk.

Types of interference:-

① Co-channel interference.
② Adjacent channel interference.

Co-channel interference and System capacity:-

* Freq. reuse implies that in a given coverage area there are several cells that use the same set of frequencies. these cells are called cochannel cells, and the interference signals from these cells is called co-channel interference.

* To reduce co-channel interference cochannels must be physically separated by a min. distance to provide sufficient isolation due to propagation.

R - Radius of the cell

D → Distance b/w centres of the nearest cochannel cells (D)

Q → Cochannel reuse ratio is related to cluster size.

* For a hexagonal geometry

$$Q = \frac{D}{R} = \sqrt{3N} \quad \text{——} \quad (1)$$

* A small value of Q provides larger capacity since the cluster size N is small.

* Let i be the no of cochannel interfering cells.

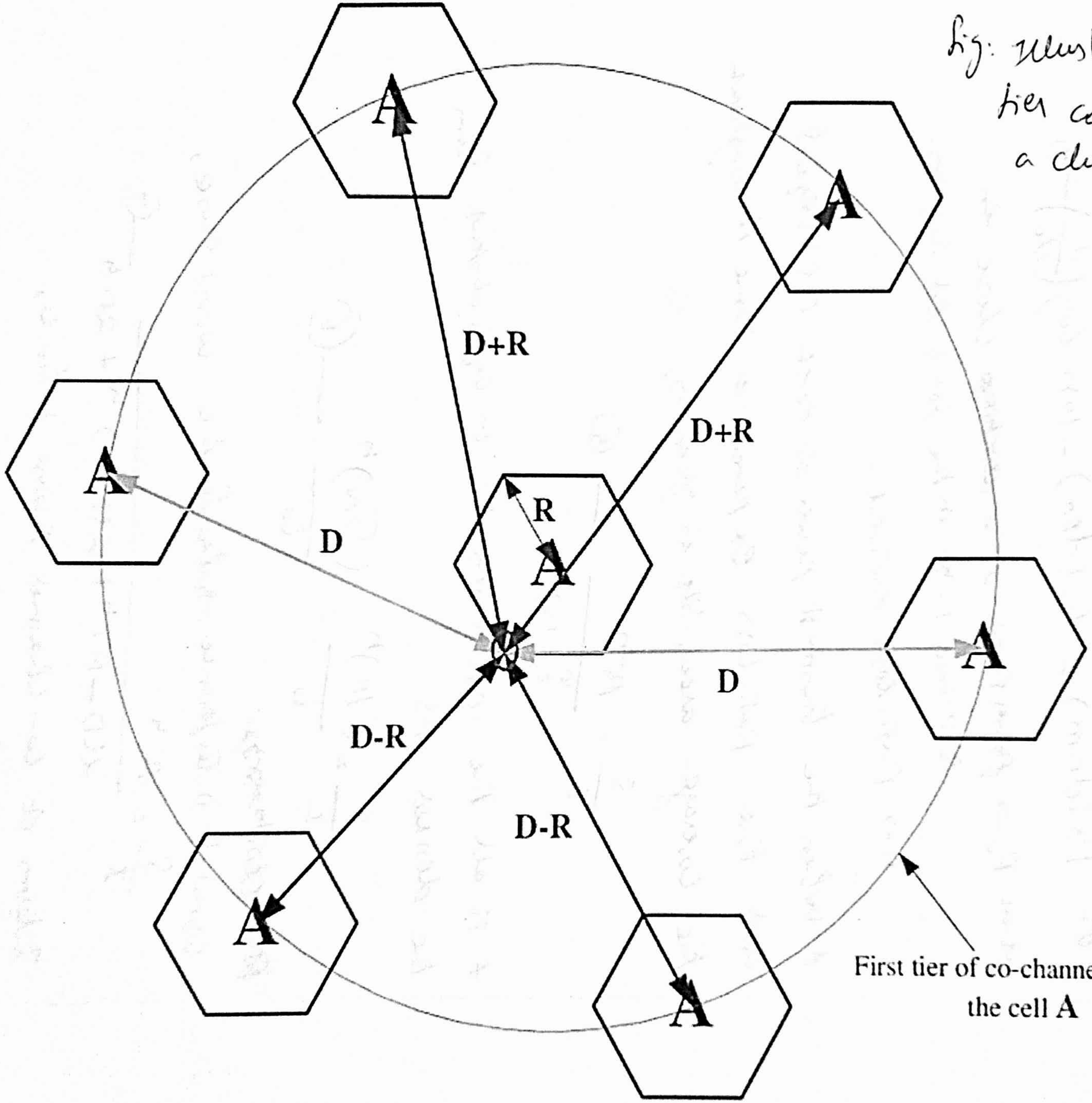
* Then the S/N ratio is given by.

$$\frac{S}{I} = \frac{S}{\sum_{i=1}^{i_0} I_i} \quad \text{——} \quad (2).$$

where S → Desired signal power from desired BS.

I_i → Interference power caused by the ith interfering cochannel cell BS.

fig: Illustration of the first tier co-channel cell for a cluster size of N=7



* The average received power P_r at a distance d from the transmitting antenna is approximated by,

$$P_r = P_0 \left(\frac{d}{d_0}\right)^{-n} \quad \text{--- (3)}$$

or $$P_r(dBm) = P_0(dBm) - 10n \log\left(\frac{d}{d_0}\right) \quad \text{--- (4)}$$

where $P_0 \rightarrow$ power rxed at a ~~distance~~ close-in reference point in the far field region.

$n \rightarrow$ path loss exponent.

* When the transmit power of each BS is equal and the path loss exponent is same throughout the coverage area, S/I is given by,

$$\frac{S}{I} = \frac{R^{-n}}{\sum_{i=1}^{10} (D_i)^{-n}} \quad \text{--- (5)}$$

* If all the interfering BS are equidistant from the desired BS

$$\frac{S}{I} = \frac{(D/R)^n}{i_0} = \frac{(\sqrt{3N})^n}{i_0} \quad \text{--- (6)}$$

~~[illegible]~~

Signal to interference ratio for the worst case,

$$\frac{S}{I} = \frac{R^{-4}}{2(D-R)^{-4} + 2(D+R)^{-4} + 2D^{-4}} \quad \text{--- (7)}$$

In terms of co-channel reuse ratio Q,

$$\frac{S}{I} = \frac{1}{2(Q-1)^{-4} + 2(Q+1)^{-4} + 2Q^{-4}} \quad \text{--- (8)}$$

Adjacent channel interference:-

* Interference resulting from signals which are adjacent in freq. to the desired signal is called Adjacent channel interference.
* Rxer filters which allow near by frequencies to leak into the pass band.

Near far effect:-

If an user is transmitting in a range that is very close to another subscribers receiver then the Rxer will make an attempt to Receive the BS signal which will result in serious interference problem. Such an effect of interference due to distance/range is called "near-far" effect.

* Adjacent channel interference minimized by,
    * filtering
    * Channel Assignments.
    * Channel allocation Schemes.
* If the freq. Reuse factor is large, the separation b/w adjacent channels at the BS may not be sufficient to keep the adjacent channel.
* For eg: If a close-in mobile is 20 times as close to the BS as another mobile, the signal to interference ratio at the BS,

$$S/I = (20)^{-n}$$

* For a path loss exponent $n=4$, this is equal to $-52dB$.

Trunking and Grade of service:-

Trunking:-

* to accomodate a large number of users in a limited radio spectrum.
* Trunking theory - to determine the number of telephone circuits that need to be allocated for office buildings with hundereds of telephone.
* when a particular user request service and all of the radio channels are already in use, the user is blocked or access denied.

Grade of service (GOS):-

* GOS is a measure of the ability of a user to access a trunked system during the busiest hour.
* The busy hours for cellular radio systems are b/w 4pm and 6pm on thursday or friday.
* The grade of service is a benchmark used to define the desired performance of a particular trunked system by specifying a desired likelihood of a user obtaining channel access given a specific number of channels available in the system.

Definitions of common terms used in Trunking theory:-

Setup time :-

The time required to allocate a trunked radio channel to a requesting user.

Blocked Call or Lost Call :-
Call which cannot be completed at time of request due to congestion

Holding time :-
Average duration of a typical call denoted by H (in seconds).

Load :-
Traffic intensity across the entire trunkcated radio system, measured in Erlangs.

Grade of service (GOS) :-
A measure of congestion which is specified as the probability of call being blocked or the probability of a call being delayed beyond a certain amount of time (for Erlang C)

Request rate :-
The average no of call request per time. Denoted by λ $seconds^{-1}$

Coverage and Capacity Improvement :-
* As the demand for wireless service increases the no of channels assigned to a cell eventually becomes insufficient to support the required no of users.
* For enhancing the cellular coverage & capacity the following techniques are used.

1. Cell spliting
2. Sectoring
3. Coverage zone approaches.
4. Repeaters.

* Cell spliting allows an orderly growth of the cellular system.

* Sectoring uses directional antennas to further control the interference and freq. reuse of channels.

* The micro cell concept distributes the coverage of a cell and extends the cell boundary to hard-to-reach places.

Cell spliting:-

* Cell spliting is the process of subdividing a congested cell into smaller cells, each with its own base station and ~~a~~ corresponding reduction in antenna height and transmitted power.

* Cell spliting increases the capacity of a cellular system since it increases the no. of times that channels are reused.

* BS are placed at the corners of the cells.

* The area served by base station A is assumed to be saturated with traffic.

* The original BS A has been surrounded by six new microcells.

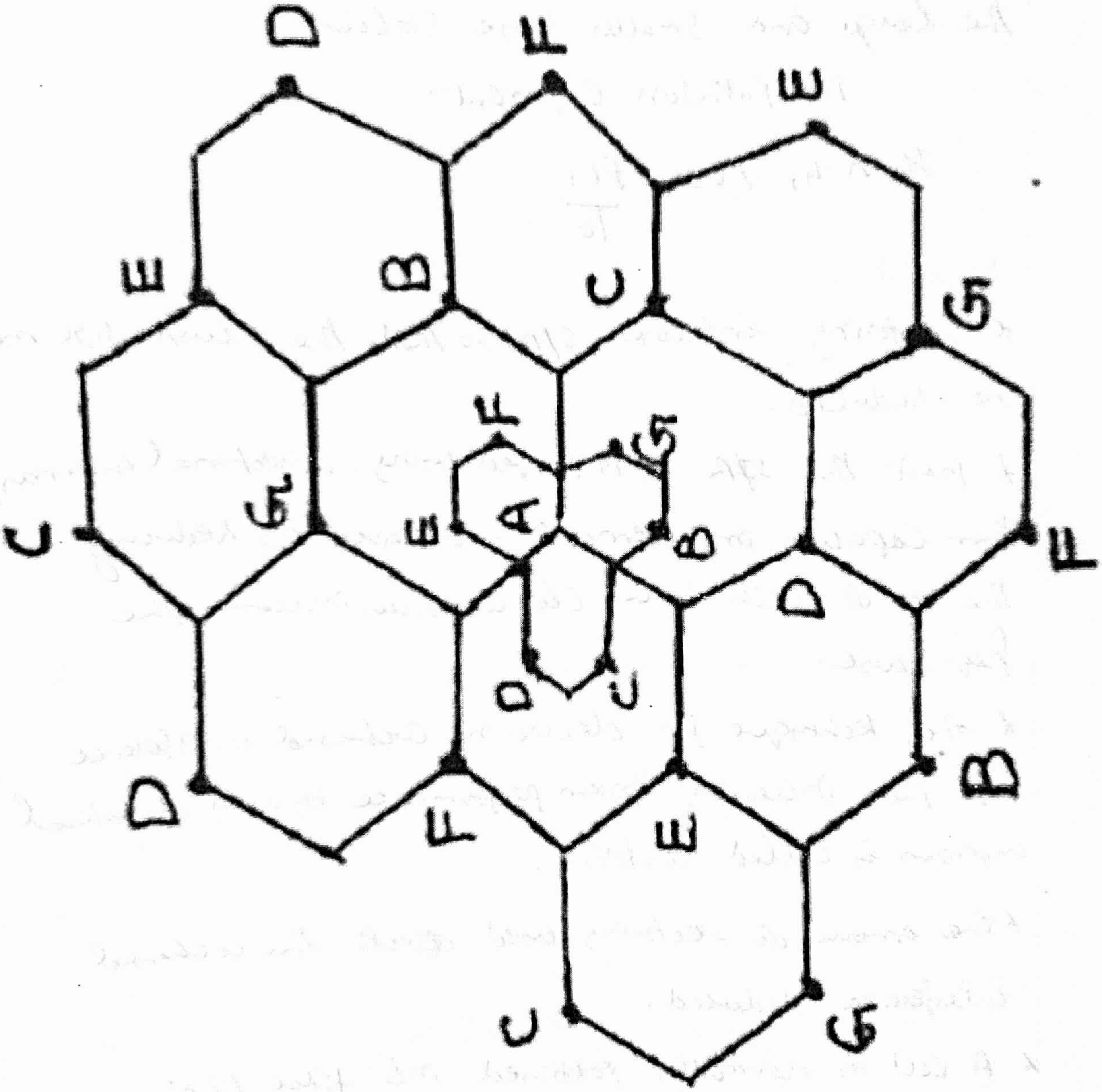


fig.. Illustration of cell splitting.

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

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

where, P_{t1} & P_{t2} are the transmit powers of the larger and smaller base stations.

$n \rightarrow$ path loss exponent.

If $n=4$, $P_{t2} = \frac{P_{t1}}{16}$

Sectoring:-

* Sectoring increases S/R so that the cluster size may be reduced.

* First the S/R is improved using directional antennas, then capacity improvement is achieved by reducing the no of cells in a cluster, thus increasing the freq. reuse.

* The technique for decreasing cochannel interference and thus increasing system performance by using directional antennas is called sectoring.

* The amount of sectoring used affects the cochannel interference reduced.

* A cell is normally partitioned into three 120° sectors or six 60° sectors.

* In 120° sectors the no. of interferences in the first tier is reduced from six to two.

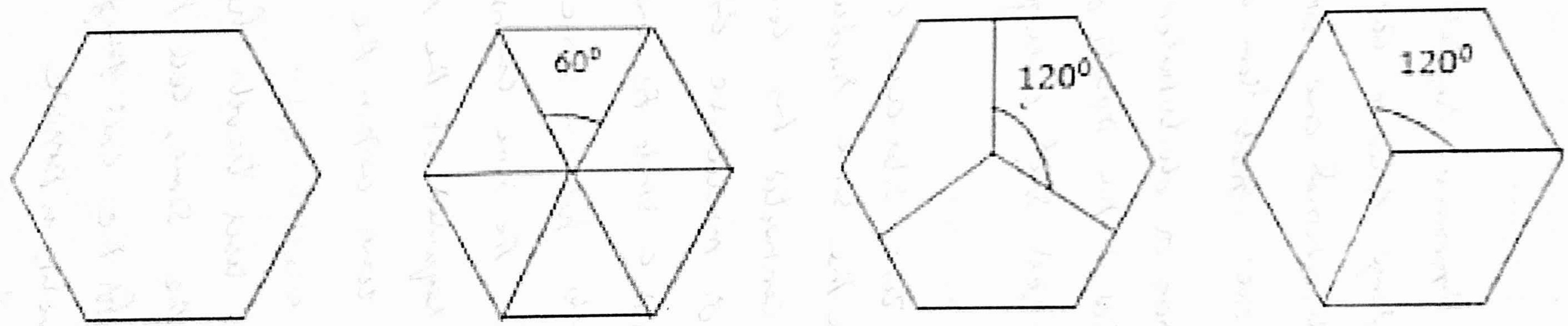


Fig: omni-directional 60° sectoring 120° sectoring

Repeaters for Range Extension :-

* Radio transmitters known as repeaters are often used to provide such range extension capabilities.

* Repeaters are Bidirectional and simultaneously send signals to and receive signals from a serving BS.

* Directional antennas or distributed Antenna Systems (DAS) are connected to the inputs or outputs of repeaters for localized spot coverage.

A Microcell Zone Concept :-

* Each of the 3 zone sites are connected to a single BS and share the same radio equipment.

* The zones are connected by coaxial cable, fiber optic cable or microwave link to the BS.

* multiple zones and a single BS make up a cell.

* As a mobile travels from one zone to another within the cell it retains the same channel.

* Hand off is not required at the MSC when the mobile travels b/w zones within the cell.

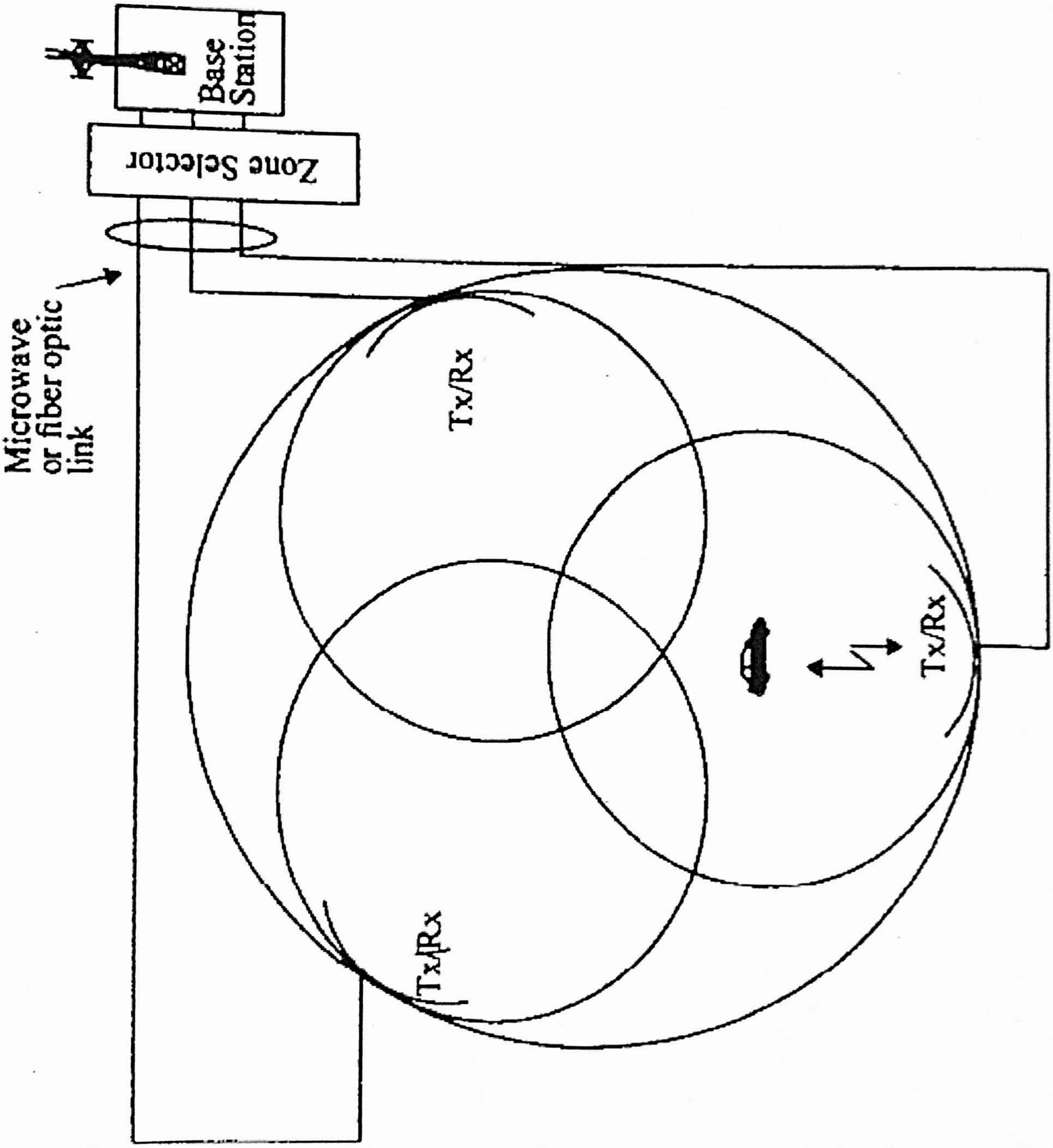
Advantages of micro cell concept :-

* when the mobile user travels from one zone to another zone within the same, cell the same channel is still maintained for the call progress.

* Improved signal quality is possible

* Reduced no of handoffs when a call is in progress.

fig: The micro cell Concept:-



Application:-

It is used in urban traffic (or) in high ways traffic conditions.

unit III

Digital Signaling for fading channels.

Structure of a wireless communication link, principles of offset — QPSK, π/4 - DQPSK, minimum Shift Keying, Gaussian minimum Shift Keying, Error performance in fading channels, OFDM principle — cyclic prefix, windowing, PAPR.

Structure of a wireless communication Link

Transceiver Block Structure :-

The goal of a wireless link to transmit information from analog source through an analog wireless propagation channel to the destination Link.

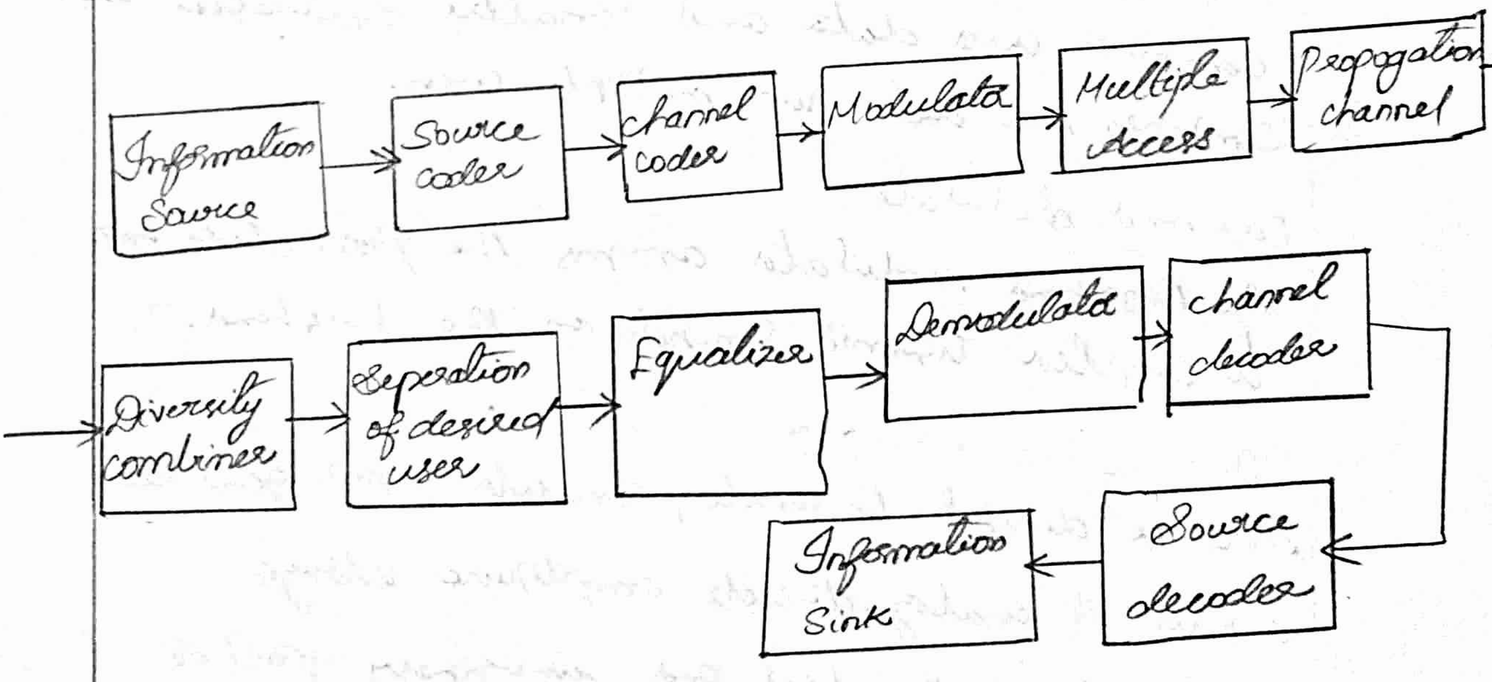


fig:- Block diagram of a transmitter and Receiver

Information Source:-

* The information source provides an analog source signal and feed it into the source ADC.
* This ADC first band limits the signal from the analog information source and then convert the signal into a stream of digital data.

Source Coder:-

The source coder uses a priori information on the properties of the source data in order to reduce redundancy in the source signal.

Channel coder:-

The channel coder adds redundancy in order to protect data against transmission errors.

multiplexer:-

Combines user data and signalling information and combines the data from multiple users.

Base band modulator:-

The Baseband modulator assigns the gross data bits to complex transmit symbols in the baseband.

DAC:-

* The TX digital to analog converter (DAC) generates a pair of analog, discrete amplitude voltages corresponding to the real and imaginary part of the symbols.

Analog Low pass filter:-

* The analog LPF in the TX eliminates the spectral components outside the desired transmission bandwidth.

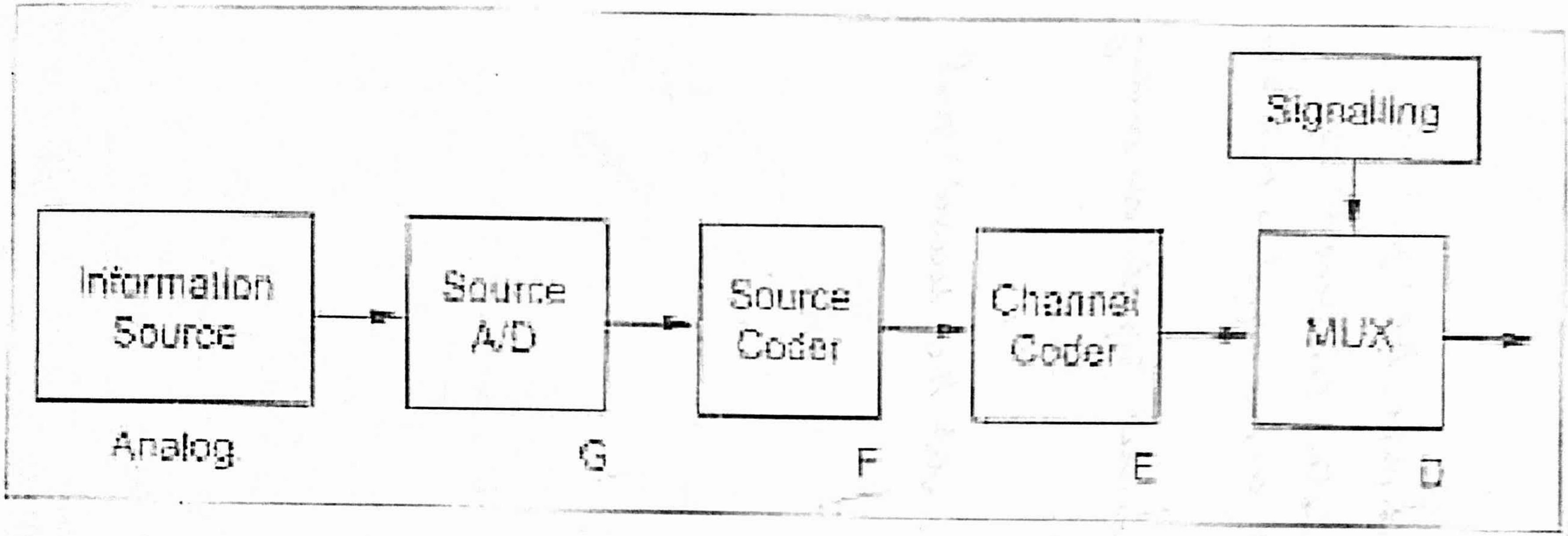


Fig. 3.2. (a) Block diagram of a radio link with digital transmitter

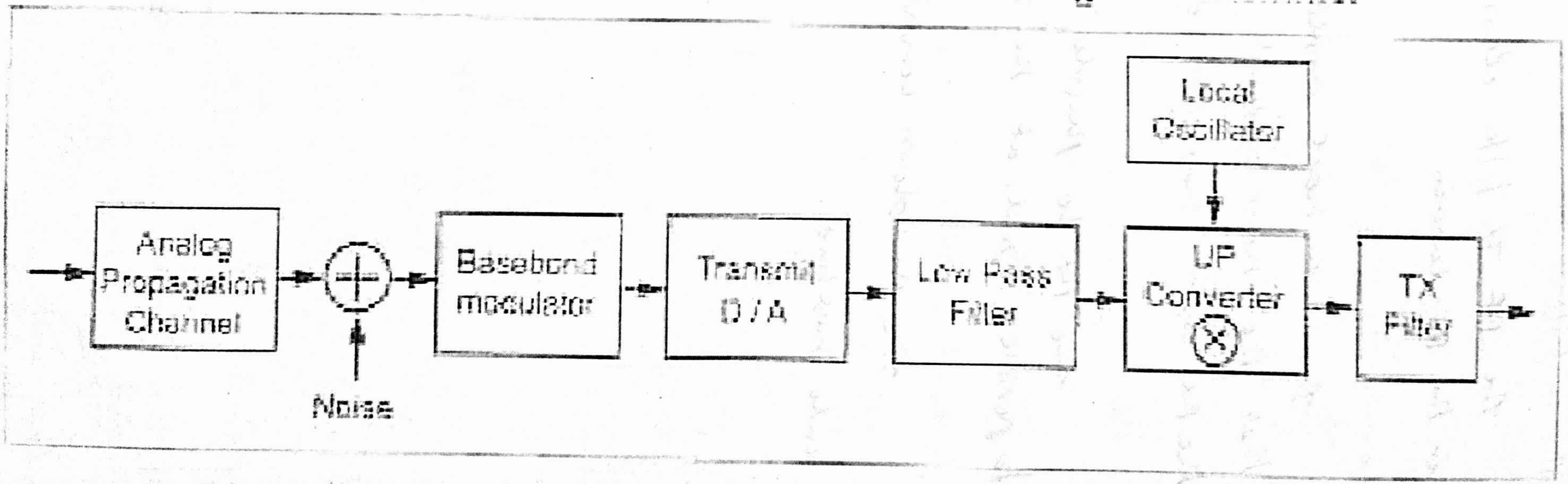


Fig. 3.2. (b) Block diagram of a radio link with digital receiver and analog propagation channel

RF TX filter:-

The RF TX filter eliminates out of band emissions in the RF domain.

Low noise Amplifier:-

The Low noise amplifier amplifies the signal, so that the noise added by later component of the Rx chain has less effect on the signal to noise ratio.

Rx LO:-

The Rx Lo provides sinusoidal signals corresponding to possible signals at the Tx Lo.

Rx down converter:-

The Rx down converter converts the received signal into baseband.

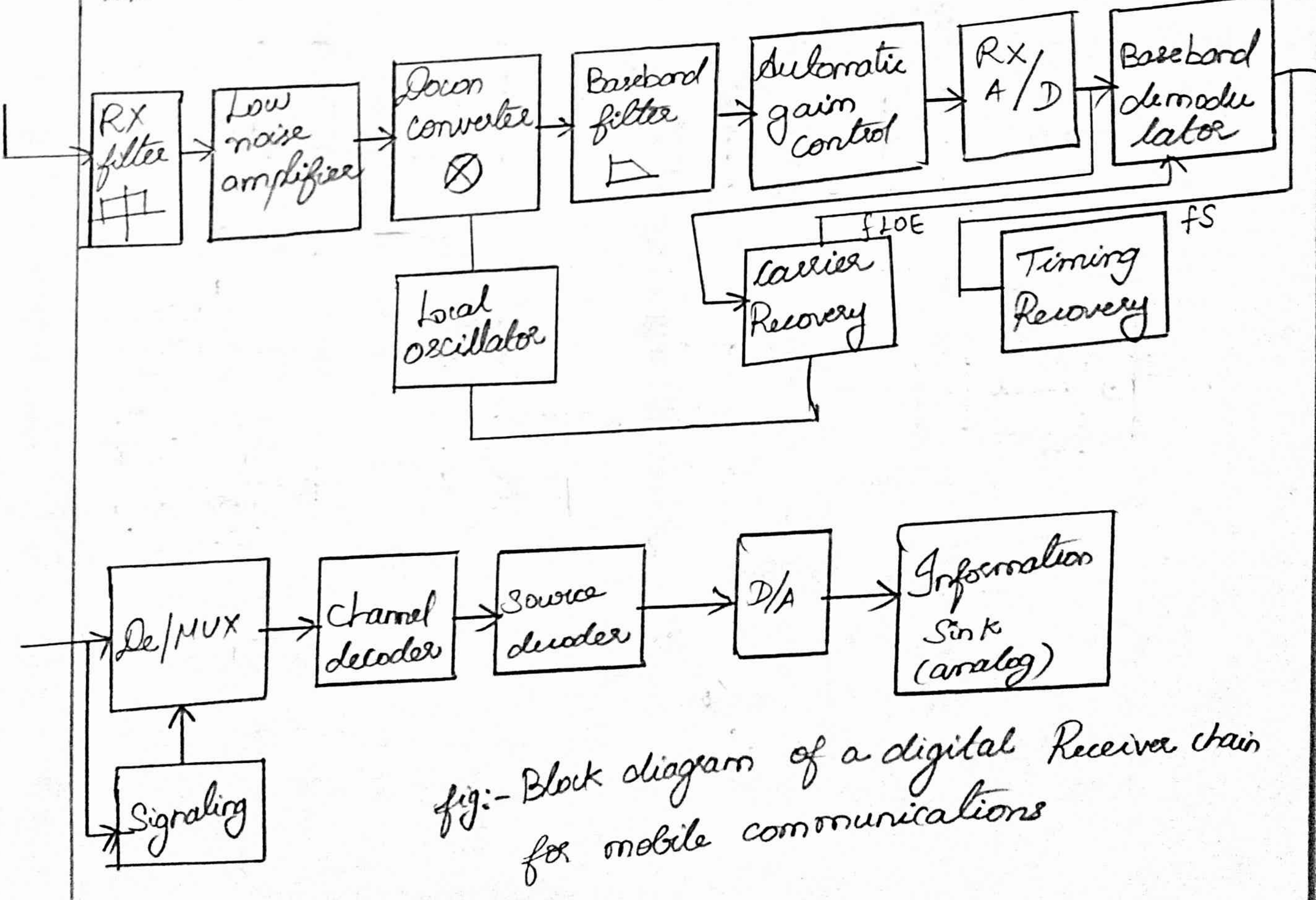


fig:- Block diagram of a digital Receiver chain for mobile communications

Rx Low pass filter:-

* The Rx LPF provides a selection of desired freq. bands for one specific user.

Automatic gain control:-

The AGC amplifies the signal such that its level is well adjusted to the quantization at the subsequent ADC.

Carrier recovery:-

Determines the freq. and phase of the carrier of the received signal.

Symbol Timing recovery:-

* It uses demodulated data to determine an estimate of the duration of symbols.

Decoder:-

The decoder uses soft estimates from the demodulator to find the original source data.

Signaling recovery:-

Identifies the parts of the data that represent signaling information and controls the subsequent demultiplexer.

Demultiplexer:-

Separates the user data and signaling information and reverse possible time compression of the Tx multiplexer.

Source decoder:-

Reconstructs the source signal from the rules of Source coding.

Simplified models:-

* The parts of the Tx b/w the information Source and the o/p of the Tx multiplexer are subsumed into a "black box" digital data Source.

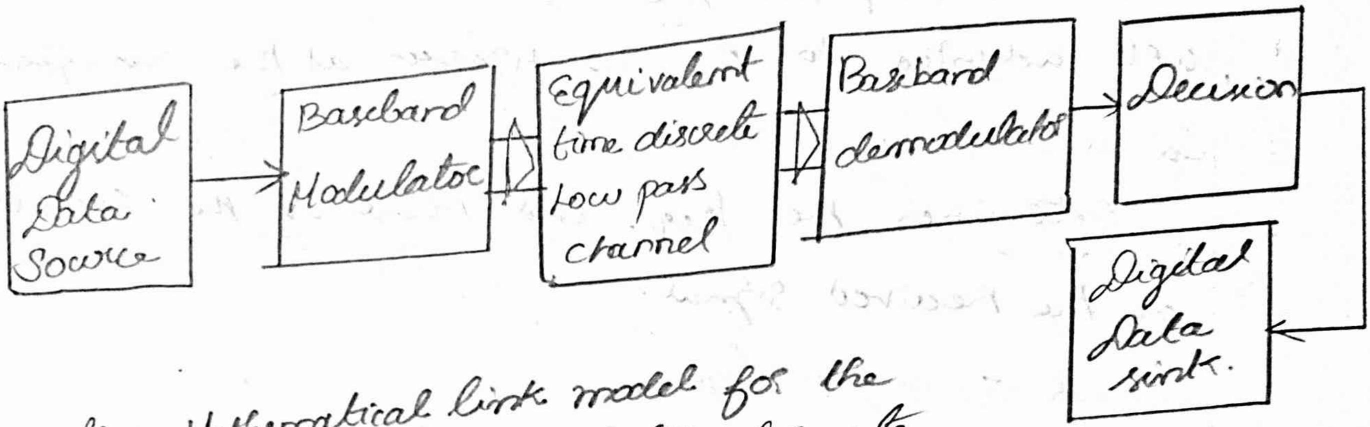


fig:- Mathematical link model for the analysis of modulation formats

QPSK (Quadrature Phase Shift Keying):-

* QPSK has twice the bandwidth efficiency of BPSK, since two bits are transmitted in a single modulation symbol.

* The phase of the carrier takes on one of four equally spaced values, such as 0, $\pi/2$, π & $3\pi/2$

* The QPSK signal for this set of symbol rates may be defined as,

$$S_{QPSK}(t) = \sqrt{\frac{2E_s}{T_s}} \cos[2\pi f_c t + (i-1)\pi/2] \quad 0 \le t \le T_s$$
$$i = 1,2,3,4 \quad \text{—— (1)}$$

where $T_s \to$ Symbol duration and is equal to twice the bit period.

using Trigonometric identities, $0 \le t \le T_s$,

$$S_{QPSK}(t) = \sqrt{\frac{2E_s}{T_s}} \cos[(i-1)\pi/2] \cos(2\pi f_c t) \quad \text{—— (2)}$$
$$- \sqrt{\frac{2E_s}{T_s}} \sin[(i-1)\pi/2] \sin(2\pi f_c t)$$

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

$$\phi_2(t) = \sqrt{\frac{2}{T_2}} \sin(2\pi f_c t)$$

$$S_{QPSK}(t) = \left\{ \sqrt{E_s} \cos[(i-1)\pi/2]\, \phi_1(t) - \sqrt{E_s} \sin[(i-1)\pi/2]\, \phi_2(t) \right.$$

$i = 1, 2, 3, 4 \ldots$ (3)

* A QPSK signal can be depicted using a 2 dimensional constellation diagram with four points.

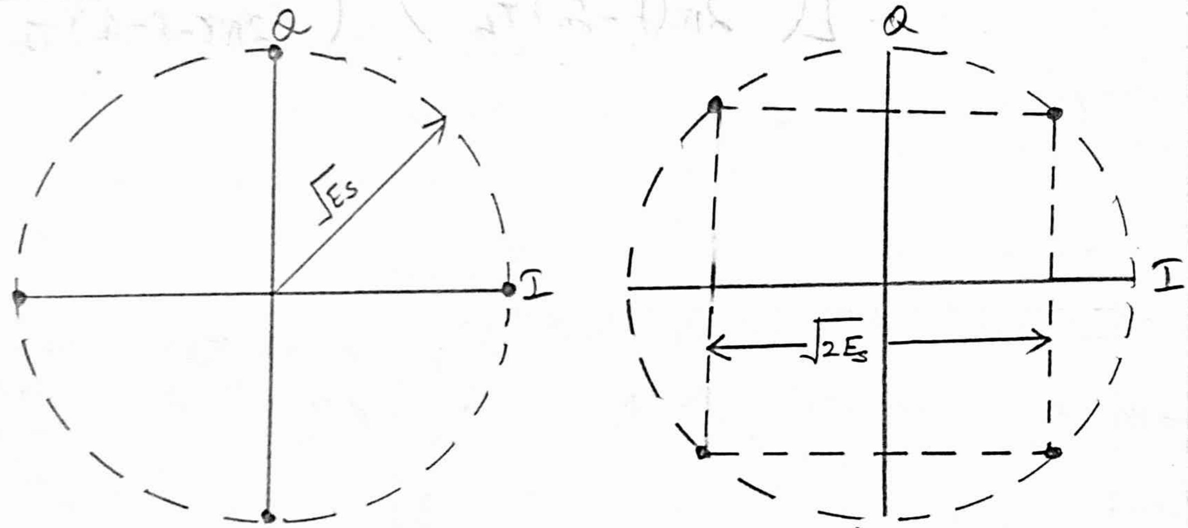


fig:- (a) QPSK constellation where the carrier phases are $0, \pi/2, \pi, 3\pi/2$
(b) QPSK constellation where the carrier phases are $\pi/4, 3\pi/4, 5\pi/4, 7\pi/4$

* Distance b/w adjacent points in the constellation is $\sqrt{2E_s}$

$E_s = 2E_b$

* The average probability of bit error in the additive white Gaussian noise (AWGN) channel is obtained as.

$$P_{e,QPSK} = Q\left(\sqrt{\frac{2E_b}{N_0}}\right) \quad \text{——— (4)}$$

* Compared to BPSK, QPSK provides twice the spectral efficiency.

* QPSK can also be differentially encoded to allow non coherent detection.

Spectrum and Bandwidth of QPSK Signals:-

* PSD of a QPSK Signal using Rectangular pulses can be expressed as,

$$P_{QPSK}(f) = \frac{E_s}{2}\left[\left(\frac{\sin\pi(f-f_c)T_s}{\pi(f-f_c)T_s}\right)^2 + \left(\frac{\sin\pi(-f-f_c)T_s}{\pi(-f-f_c)T_s}\right)^2\right]$$

$$= E_b\left[\left(\frac{\sin 2\pi(f-f_c)T_b}{2\pi(f-f_c)T_b}\right)^2 + \left(\frac{\sin 2\pi(-f-f_c)T_b}{2\pi(-f-f_c)T_b}\right)^2\right] \quad \text{--- (5)}$$

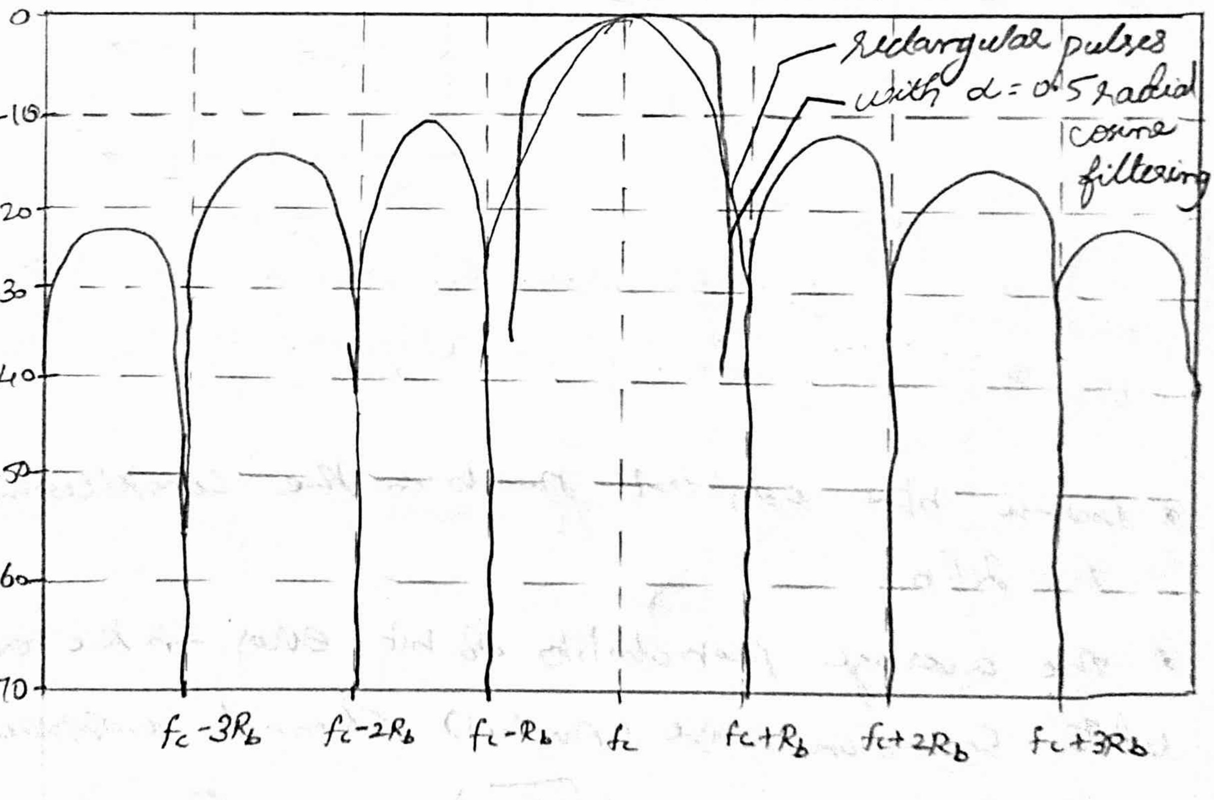


fig:- Power spectral density of a QPSK signal

QPSK Transmission and Detection Techniques:-

* The unipolar binary message stream has bit rate R_b and is first converted into a bipolar non-return-to-zero (NRZ) sequence using a unipolar to bipolar converter.

* The bit stream m(t) is then split into two bit streams $m_I(t)$ and $m_Q(t)$

  * Bit rate $R_s = R_b/2$

  * $m_I(t)$ is called even stream & $m_Q(t)$ is called odd stream.

* The 2 binary sequences are separately modulated by 2 carriers $\phi_1(t)$ and $\phi_2(t)$, which are in quadrature

* Two BPSK signals are summed to produce a QPSK signal.

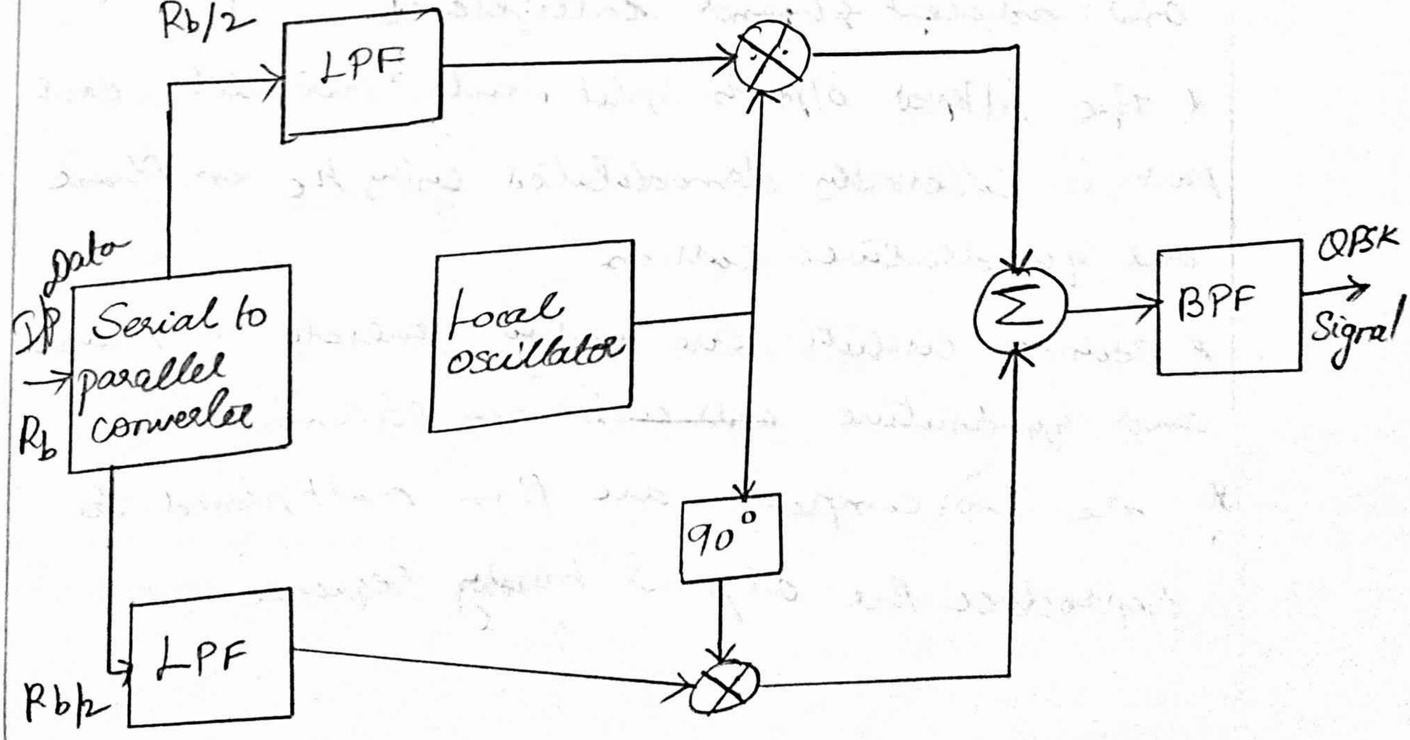


fig:- Block diagram of a QPSK transmitter

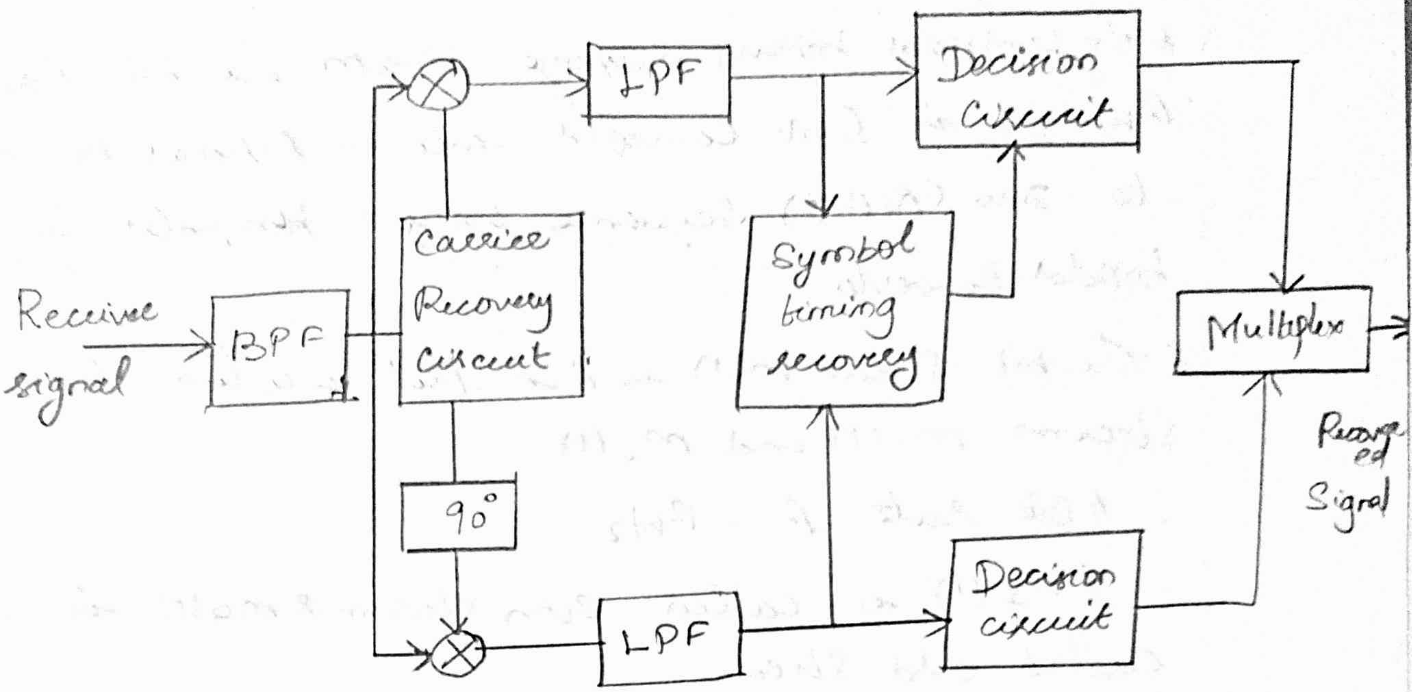


fig:- Block diagram of a QPSK Receiver

* the front end BPF removes the out of band noise and adjacent channel interference.
* The filtered O/P is split into two parts, each part is coherently demodulated using the in Phase and quadrature carriers.
* Decision circuits are used to generate in-Phase and quadrature ~~carriers~~ binary streams.
* The two components are then multiplexed to reproduce the original binary sequence.

Off set QPSK:-

* The amplitude of a QPSK signal is ideally constant.

* A modified form of QPSK called offset QPSK (OQPSK) or staggered QPSK is less susceptible to these deleterious effects.

* RF amp → which helps eliminate the spectrum regrowth after amplification.

* In QPSK signaling the bit transitions of the even and odd bit streams occur at the same instants, but in OQPSK signaling, the even and odd bit streams $m_I(t)$ and $m_Q(t)$ offset in their relative alignment by one bit period.

* Due to the time alignment of $m_I(t)$ and $m_Q(t)$ in standard QPSK, phase transitions occur only once every $T_S = 2T_b$ s.

* Bit transitions occur every T_b s.

* Max- phase shift of the transmitted signal at any given time is limited $\pm 90°$

* since $180°$ phase transitions have been eliminated, band limiting of OQPSK signals does not cause the signal envelope to go to zero.

* OQPSK signals does not regenerate the high freq. sidelobes as much as in QPSK.

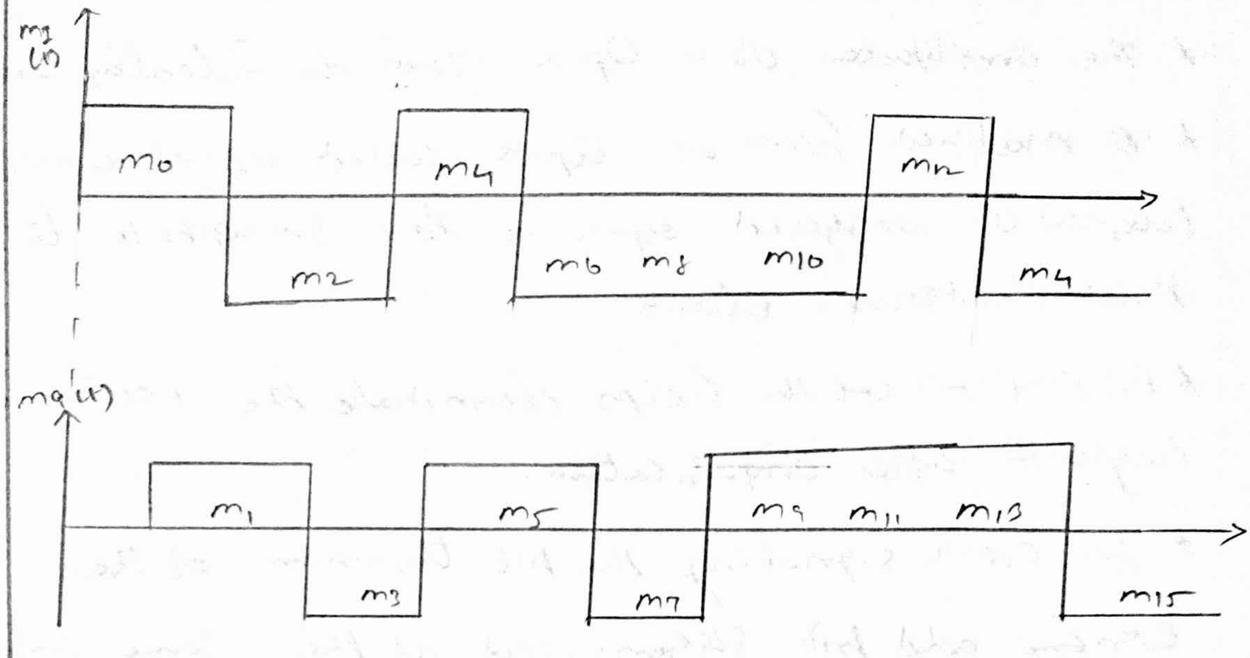


fig: the time offset wave forms of an OQPSK modulator.

* The staggered alignment of the even and odd bit streams does not change the nature of the spectrum.
* OQPSK retains its band-limited nature even after non linear amplification.

π/4 QPSK:-

* The π/4 shifted QPSK modulation is a quadrature phase shift keying technique which offers a compromise b/w OQPSK and QPSK in terms of the allowed max. phase transitions.
* In π/4 QPSK the max. phase change is limited to $\pm 135°$ as compared to $180°$ for QPSK & $90°$ for OQPSK.
* π/4 QPSK is non coherently detected, which simplifies receiver design.
* when differentially encoded, π/4 QPSK is called π/4 DQPSK.

* In a $\pi/4$ QPSK modulator, signalling points of the modulated signal are selected from 2 QPSK constellations which are shifted by $\pi/4$ with respect to each other.

* Every successive bit ensures that there is atleast a phase shift which is an integer multiple of $\pi/4$ radians b/w successive symbols.

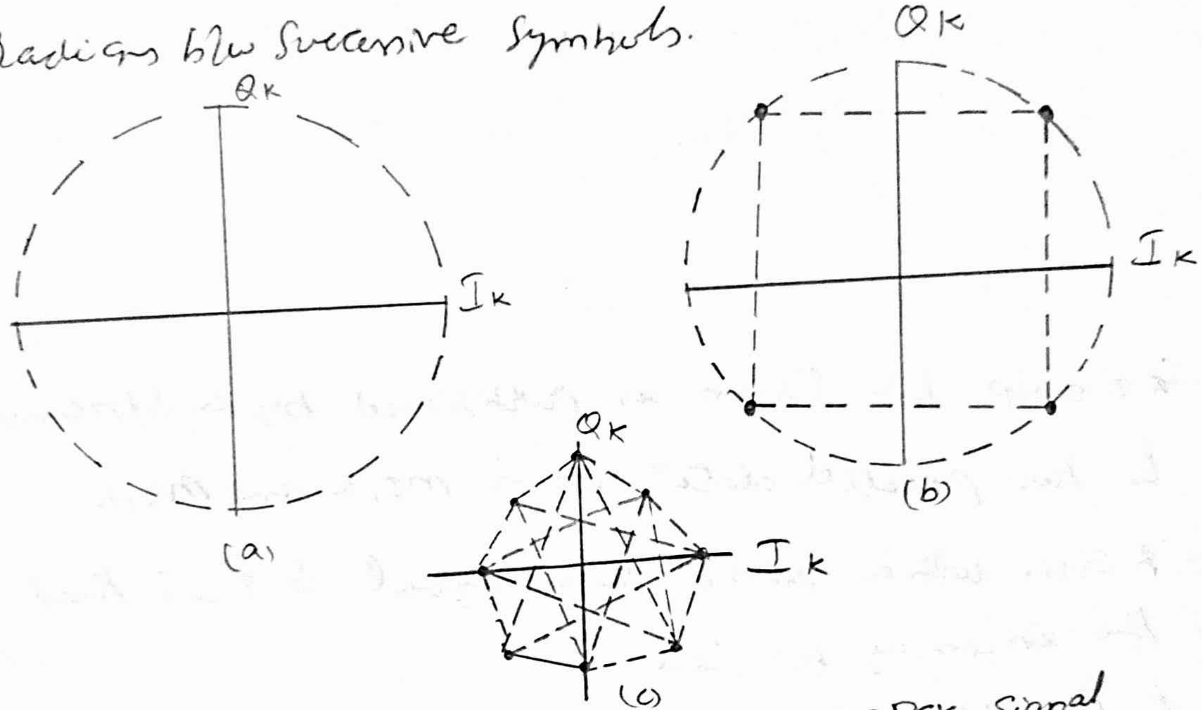


fig:- Constellation diagram of a $\pi/4$ QPSK signal
(a) Possible states for θ_K when $\theta_{K-1} = n\pi/4$
(b) Possible states when $\theta_{K-1} = n\pi/2$
(c) all possible states

Information bits m_{IK}, m_{QK}	Phase shift ϕ_K
11	$\pi/4$
01	$3\pi/4$
00	$-3\pi/4$
10	$-\pi/4$

π/4 QPSK Transmission Techniques:-

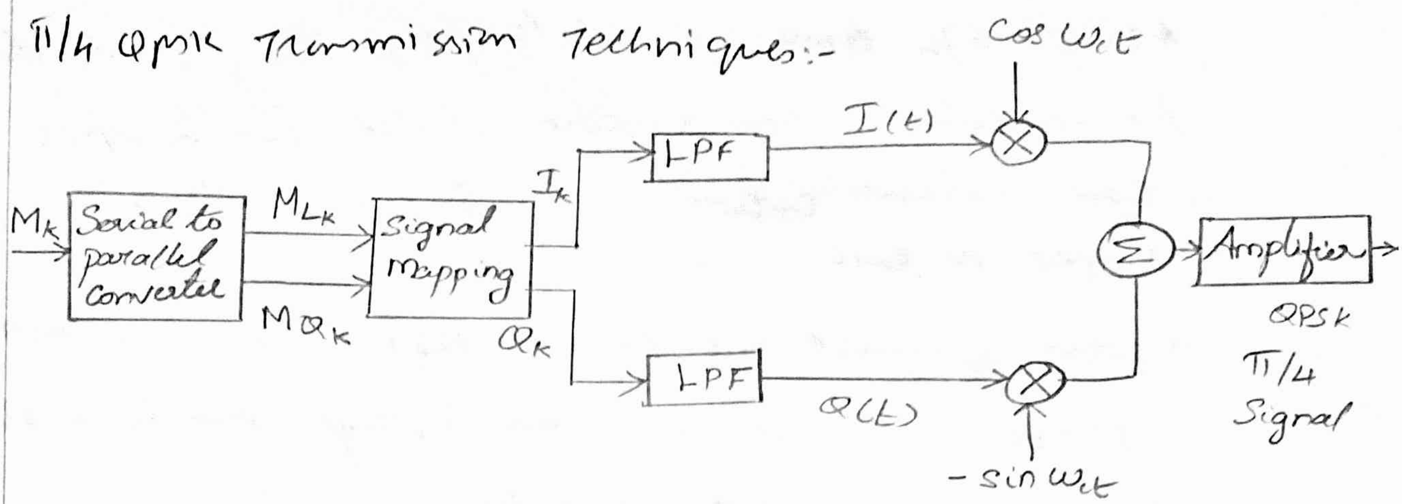


fig:- π/4 QPSK Transmitter

* The i/p bit stream is partitioned by a S/P converter to two parallel data streams $m_{I,K}$ and $m_{Q,K}$
* Each with a symbol rate equal to half that of the incoming bit rate.
* The K^{th} inphase and Quadrature pulses, I_K and Q_K, over time $KT \le t \le (K+1)T$ and are determined by their previous values I_{K-1} and Q_{K-1} & Q_K.

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

$$Q_K = \sin\theta_K = I_{K-1}\sin\phi_K + Q_{K-1}\cos\phi_K \quad \text{—②}$$

where $\theta_K = \theta_{K-1} + \phi_K$ —③

* θ_K and θ_{K-1} are phases of the K^{th} and $K-1$ symbols

* The inphase and quadrature bit streams I_k and Q_k are then separately modulated by two carriers which are in quadrature with one another, to produce the $\pi/4$ QPSK wave form given by,

$$S_{\pi/4\,QPSK}(t) = I(t)\cos w_c t - Q(t)\sin w_c t \quad \text{—— (4)}$$

where,

$$I(t) = \sum_{k=0}^{N-1} I_k\, P(t - kT_s - T_s/2) = \sum_{k=0}^{N-1} \cos\theta_k\, P(t - kT_s - T_s/2) \quad \text{—— (5)}$$

$$Q(t) = \sum_{k=0}^{N-1} Q_k\, P(t - kT_s - T_s/2) = \sum_{k=0}^{N-1} \sin\theta_k\, P(t - kT_s - T_s/2) \quad \text{—— (6)}$$

* Both I_k and Q_k are usually passed through raised cosine roll off pulse shaping filters before modulation in order to reduce the bandwidth occupancy.

* I_k & Q_k and the peak amplitude of the waveforms $I(t)$ and $Q(t)$ can take one of the five possible values $0, +1, -1, +1/\sqrt{2}, -1/\sqrt{2}$

$\pi/4$ QPSK Detection Techniques:-

In an AWGN channel, the BER performance of a differentially detected $\pi/4$ QPSK is about 3dB inferior to QPSK.

* $\pi/4$ QPSK Detection techniques are
    - ↳ Baseband differential Detection
    - ↳ IF differential detection
    - ↳ FM discriminator detection

* Base band and IF differential detector determine the Cosine & sine fn's of the phase difference.
* FM discriminator detects the phase difference directly in a non coherent manner.

Base band Differential Detection:-

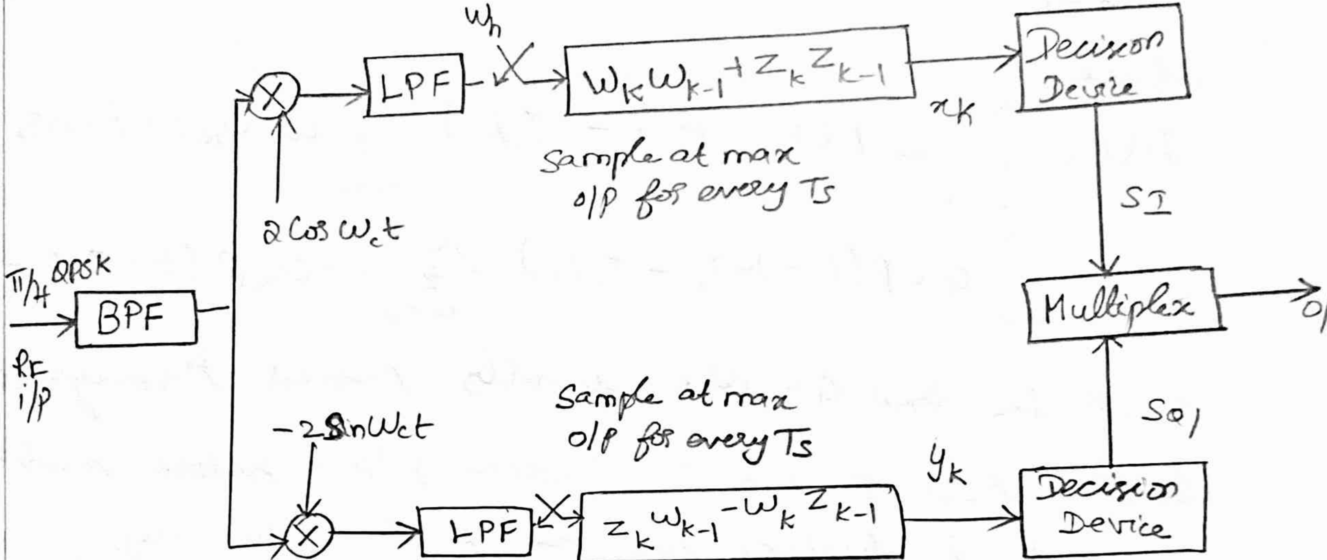


fig:- Block diagram of a baseband differential detector

* The incoming $\pi/4$ QPSK signal is quadrature demodulated using two local oscillator signals.
* If $\phi_k = \tan^{-1}(Q_k/I_k)$ is the phase of the carrier due to the k^{th} data bit, the o/p w_k and z_k can be expressed as

$$w_k = \cos(\phi_k - \gamma) \quad \text{--- (7)}$$

$$z_k = \sin(\phi_k - \gamma) \quad \text{--- (8)}$$

* where γ is a phase shift due to noise, propagation and interference

* The 2 sequences w_k and z_k are passed through a differential decoder which operates on the foll. Rule

$$x_k = w_k w_{k-1} + z_k z_{k-1} \quad \text{—(9)}$$

$$y_k = z_k w_{k-1} - w_k z_{k-1} \quad \text{—(10)}$$

* The output of the differential decoder can be expressed as

$$x_k = \cos(\phi_k - \phi_{k-1}) \quad \text{—(11)}$$

$$y_k = \sin(\phi_k - \phi_{k-1}) \quad \text{—(12)}$$

* The output of the differential decoder is applied to the decision circuit.

$$S_I = 1, \text{ if } x_k > 0 \text{ or } S_I = 0 \text{ if, } x_k < 0 \quad \text{—(13)}$$

$$S_Q = 1, \text{ if } y_k > 0 \text{ or } S_Q = 0 \text{ if, } y_k < 0 \quad \text{—(14)}$$

where S_I & S_Q are the detected bits in phase and quadrature arms.

IF Differential Dectector :-

* It avoids the need for a local oscillator by using a delay line and two phase detectors.

* The Received signal is converted to IF and is band pass filtered

* To minimized the effect of ISI and noise the bandwidth of the filters are chosen to be $0.57/T_s$.

* The Received IF signal is differentially decoded using a delay line and two mixers.
* The Band width of the signal at the o/p of the differential detector is twice that of the base band signal at the transmitter end.

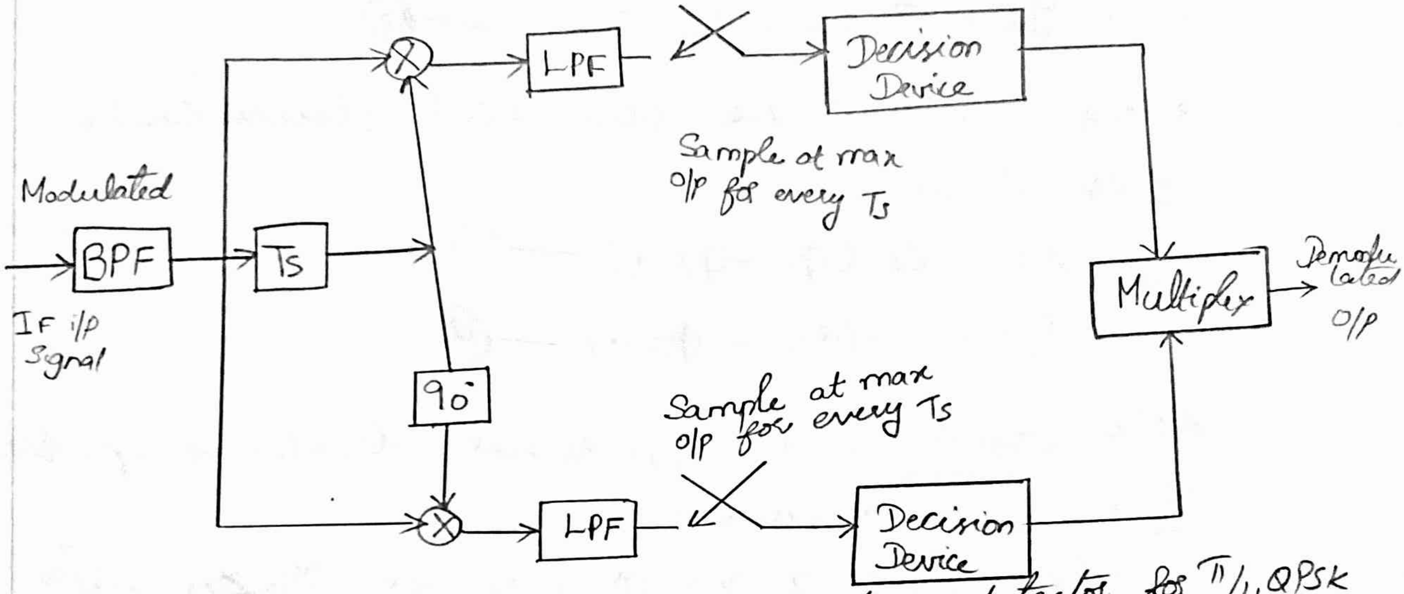


fig:- Block diagram of an IF differential detector for $\pi/4$ QPSK

FM Discriminator:

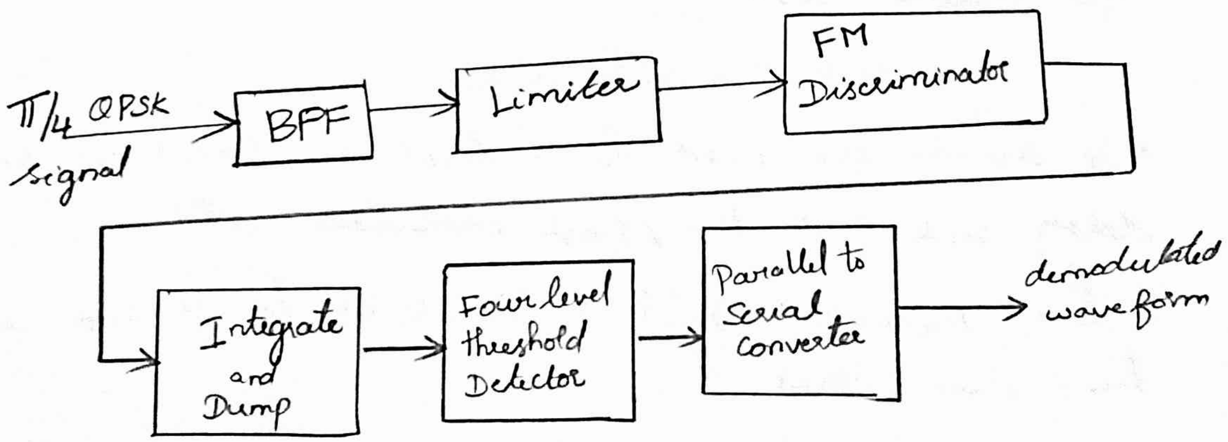


fig:- FM Discriminator detector for $\pi/4$ DQPSK Demodulation

* The i/p signal is first filtered using a band pass filter that is matched to the transmitted signal.
* The FM discriminator extracts the instantaneous freq. deviation of the received signal.
* when integrated over each symbol period gives the phase difference b/w two sampling instants.
* The modulo-2π phase detector improves the BER performance and reduces the effect of click noise

Minimum Shift Keying (MSK):-

* MSK is a special type of continuous phase freq. shift keying (CPFSK).
* The peak freq. deviation is equal to 1/4 the bit rate.
* The modulation index of an FSK is similar to the FM modulation index.

$$K_{FSK} = (2\Delta F)/R_b;$$

where $\Delta F \rightarrow$ peak RF freq. deviation.

$R_b \rightarrow$ bit rate.

* The name MSK implies the min. freq. seperation that allows orthogonal detection.
* Two FSK signals $V_H(t)$ and $V_L(t)$ are said to be orthogonal if,

$$\int_0^T V_H(t)\, V_L(t)\, dt = 0 \quad \text{—— } ①$$

* MSK sometimes referred to as fast FSK, as the freq. spacing used in only half as much as that used in conventional non coherent FSK.

MSK properties:-

* Constant envelope
* spectral efficiency.
* good BER performance.
* self-synchronizing capability.

* If half sinusoidal pulses are used instead of rectangular pulses, the modified signal can be defined as MSK and for an N bit stream is given by,

$$S_{MSK}(t) = \sum_{i=0}^{N-1} m_I(t)\, p[t - 2iT_b] \cos 2\pi f_c t + \sum_{i=0}^{N-1} m_Q(t)\, p(t - 2iT_b - T_b) \sin 2\pi f_c t \quad \text{—(2)}$$

where
$$p(t) = \begin{cases} \sin\left(\dfrac{\pi t}{2T_b}\right) & 0 \le t \le 2T_b \\ 0 & \text{o.w} \end{cases} \quad \text{—(3)}$$

* The MSK waveform can be seen as a special type of a continuous phase FSK.

$$S_{MSK}(t) = \sqrt{\frac{2E_b}{T_b}} \cos\left(2\pi f_c t - m_I(t)\, m_Q(t) \frac{\pi t}{2T_b} + \phi_k\right) \quad \text{—(5)}$$

where ϕ_k is 0 or π depending on whether $m_I(t)$ is 1 or -1

* MSK signal is an FSK signal with binary signaling frequencies of $f_c + 1/4T$ and $f_c - 1/4T$.

MSK power spectrum:-

* RF power spectrum is obtained by freq. shifting the magnitude squared of the fourier transform of the base band pulse shaping function.

* For MSK the base band pulse shaping fn. is given by

$$p(t) = \begin{cases} \cos\left(\frac{\pi t}{2T}\right) & |t| < T \\ 0 & o.w \end{cases} \quad \text{—— (6)}$$

* Thus the normalized power spectral density for MSK is given by.

$$P_{MSK}(f) = \frac{16}{\pi^2}\left(\frac{\cos 2\pi(f+f_c)T}{1.16f^2T^2}\right)^2 + \frac{16}{\pi^2}\left(\frac{\cos 2\pi(f-f_c)^2 T}{1.16f^2T^2}\right) \quad \text{—— (7)}$$

* MSK spectrum has lower sidelobes than QPSK & OQPSK.

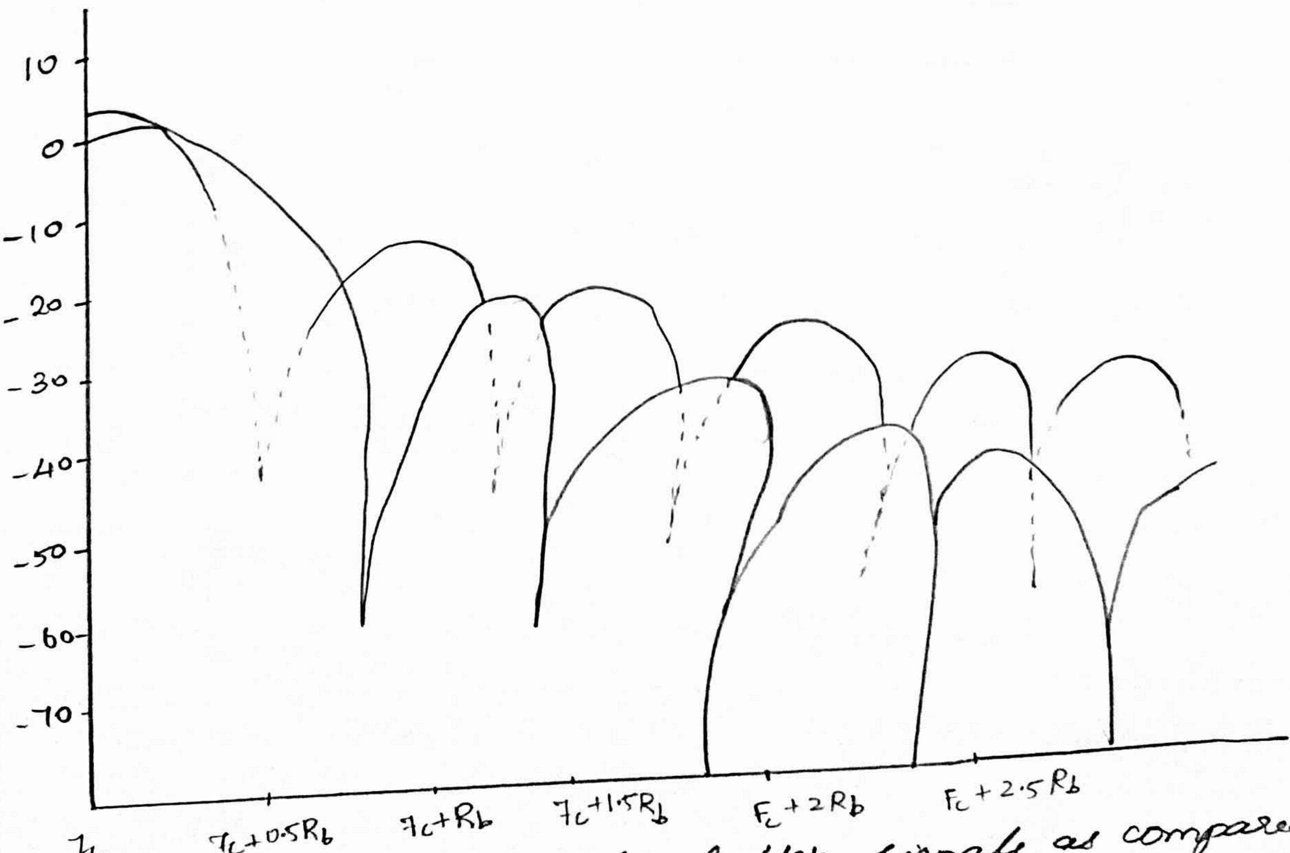


fig:- Power spectral density of MSK signals as compared to QPSK and OQPSK signals

* MSK is popular modulation scheme for mobile radio communication : Reasons are
* The envelope is kept more or less constant even after band limiting
* Any small variations in the envelope level can be removed by hard limiting at the receiver without raising the out of band radiation levels.
* The continuous phase property makes it highly desirable for highly reactive loads.
* MSK has simple demodulation and syn. circuits.

MSK Transmitter and Receiver:-

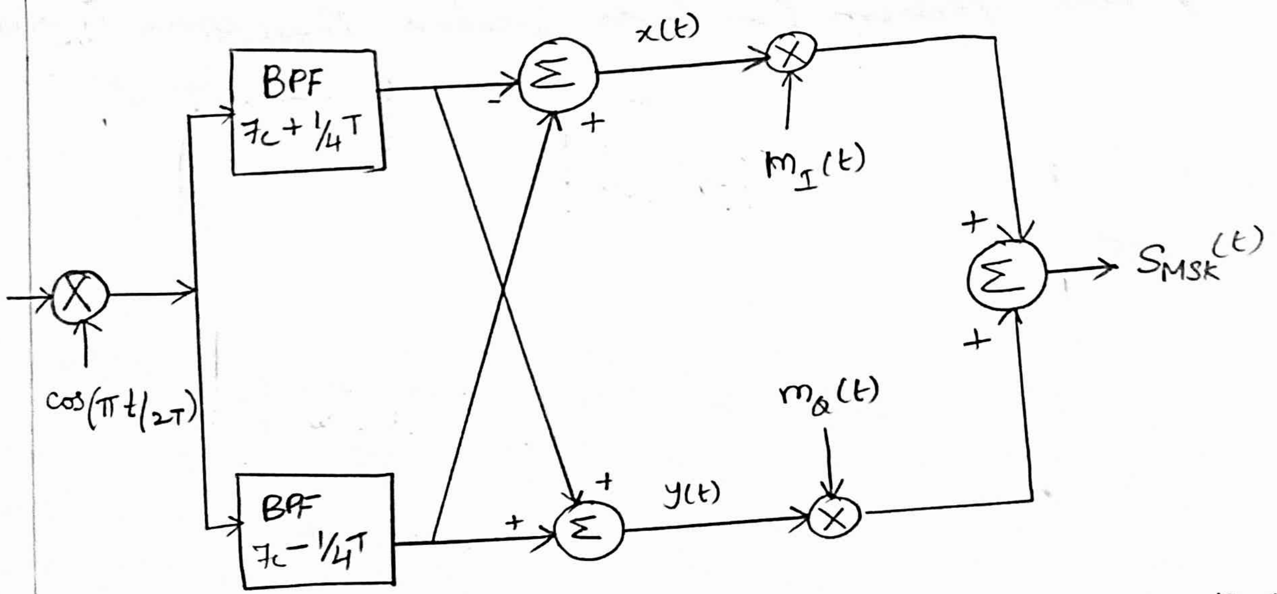


fig:- Block diagram of an MSK Transmitter. Note that $m_I(t)$ and $m_Q(t)$ are offset by T_b

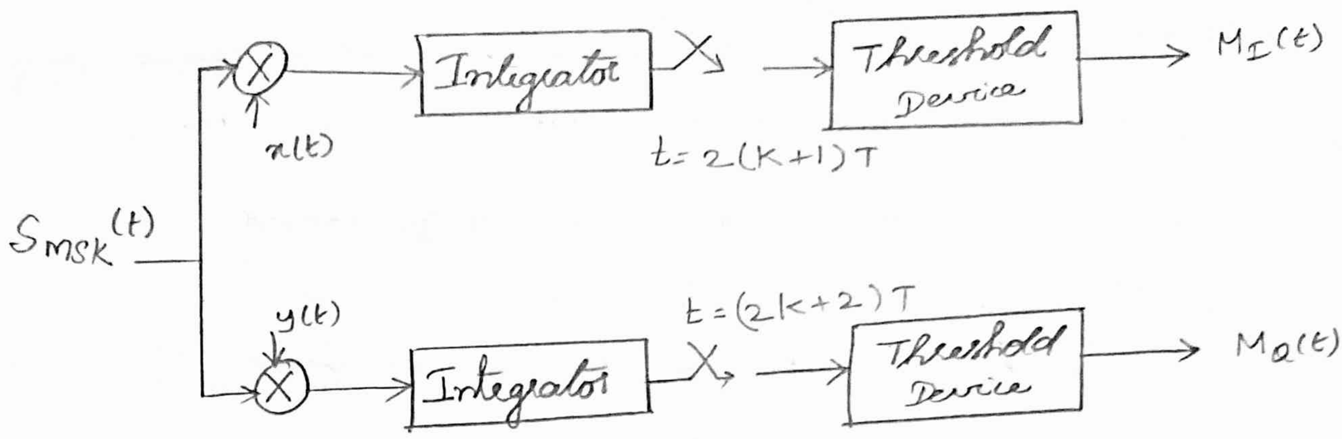


fig:- Block Diagram of MSK Receiver

* multiplying a carrier signal with $\cos(\pi t/2T)$ produces two phase coherent signals at $f_c + 1/4T$ and $f_c - 1/4T$
* two FSK signals are separated using two narrow BPF & combined to form inphase and Quadrature phase carrier component $x(t)$ and $y(t)$.
* These carriers are multiplied with odd & even bits streams $m_I(t)$ & $m_Q(t)$ to produce $S_{msk}(t)$.
* The o/p of the multipliers are integrated over 2 bit periods and dumped to a decision ckt.
* The threshold detector decides whether the signal is a 0 or a 1.
* The o/p data streams correspond to $m_I(t)$ and $m_Q(t)$ which are offset combined to obtain the demodulated signal.

Gaussian minimum shift keying (GMSK):.

* GMSK is a simple binary modulation scheme which may be viewed as a derivative of MSK.

* Premodulation Gaussian filtering converts the full response message signal into a partial response scheme where each transmitted symbol spans several bit periods.

* GMSK can be coherently or non coherently detected.

* GMSK is most attractive for its power efficiency and spectral efficiency.

* The premodulation Gaussian filtering introduces ISI in the transmitted signal.

GMSK properties:-

* Irreducible error rate.
* Partial response signaling.
* Good spectral efficiency
* Constant envelope properties

* The GMSK premodulation filter has an impulse response given by.

$$h_G(t) = \frac{\sqrt{\pi}}{\alpha} \exp\left(\frac{-\pi^2}{\alpha^2} t^2\right) \text{ ——— } (8)$$

and the transfer fn is given by,

$$H_G(f) = \exp(-\alpha^2 f^2) \text{ ——— } (9)$$

* The parameter α is related to B, the 3dB base band Bandwidth of $H_G(f)$ by

$$\alpha = \frac{\sqrt{\ln 2}}{\sqrt{2}\,B} = \frac{0.5887}{B} \quad \text{— (19).}$$

and the GMSK filter may be completely defined from B and the base band symbol duration T.

Power spectrum of GMSK:-

* The power spectrum of MSK, which is equivalent to GMSK with a BT product of infinity.
* BT product decreases, the sidelobe levels fall off very rapidly.
* GMSK irreducible error rate is less than that produced by the mobile channel, there is no penalty in using GMSK.

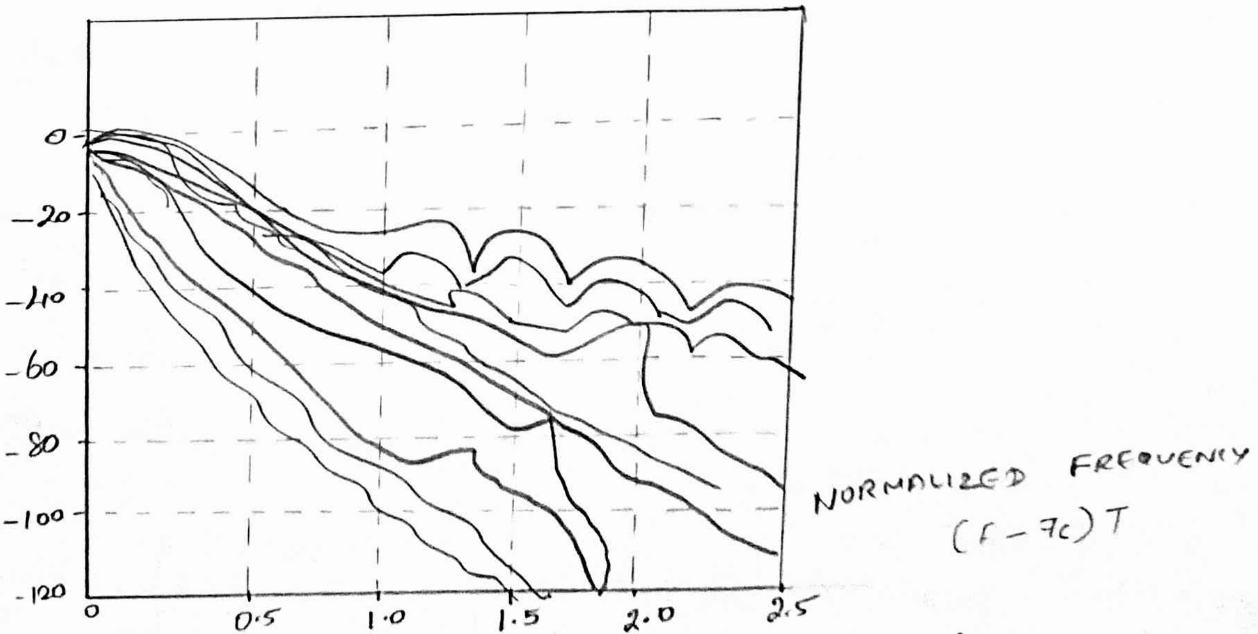


fig:- Power spectral density of a GMSK signal

GMSK Bit Error Rate:-

* The bit error probability is a fn of BT, since the pulse shaping impacts ISI
* The bit error probability for GMSK is given by,

$$P_e = Q\left\{\sqrt{\frac{2\gamma E_b}{N_0}}\right\} \quad \text{(11a)}$$

where γ is a constant related to BT by

$$\gamma = \begin{cases} 0.68 & \text{for GMSK with } BT = 0.25 \\ 0.85 & \text{for simple MSK } (BT = \infty) \end{cases} \quad \text{(11b)}$$

GMSK Transmitter and Receiver:-

* The simplest way to generate a GMSK signal is to pass a NRZ message bit stream through a Gaussian baseband filter, followed by FM modulator.
* It is used in US cellular digital packet data (CDPD) & Global System for mobile (GSM) system.
* Fig. below may also be implemented digitally using a standard I/Q modulator.

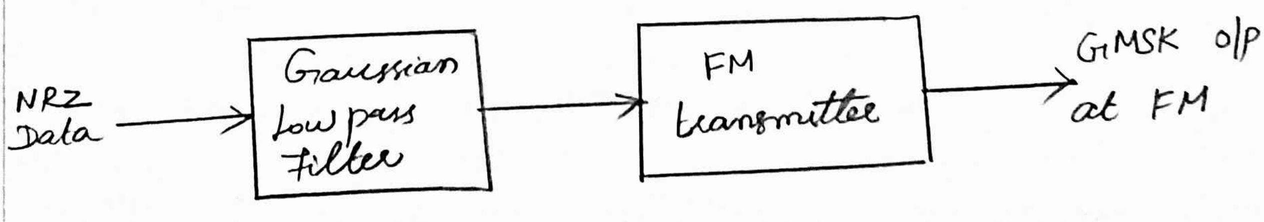


fig:- Block Diagram of a GMSK transmitter using Direct FM generation

* GMSK signals can be detected using orthogonal coherent detectors or non coherent detectors such as standard FM discriminators.

* The sum of two discrete freq. component centered at the o/p of a freq. doubler is divided by four.

* De Buda's method is similar to the Costas loops and is equivalent to that of a PLL with a frequency doubler.

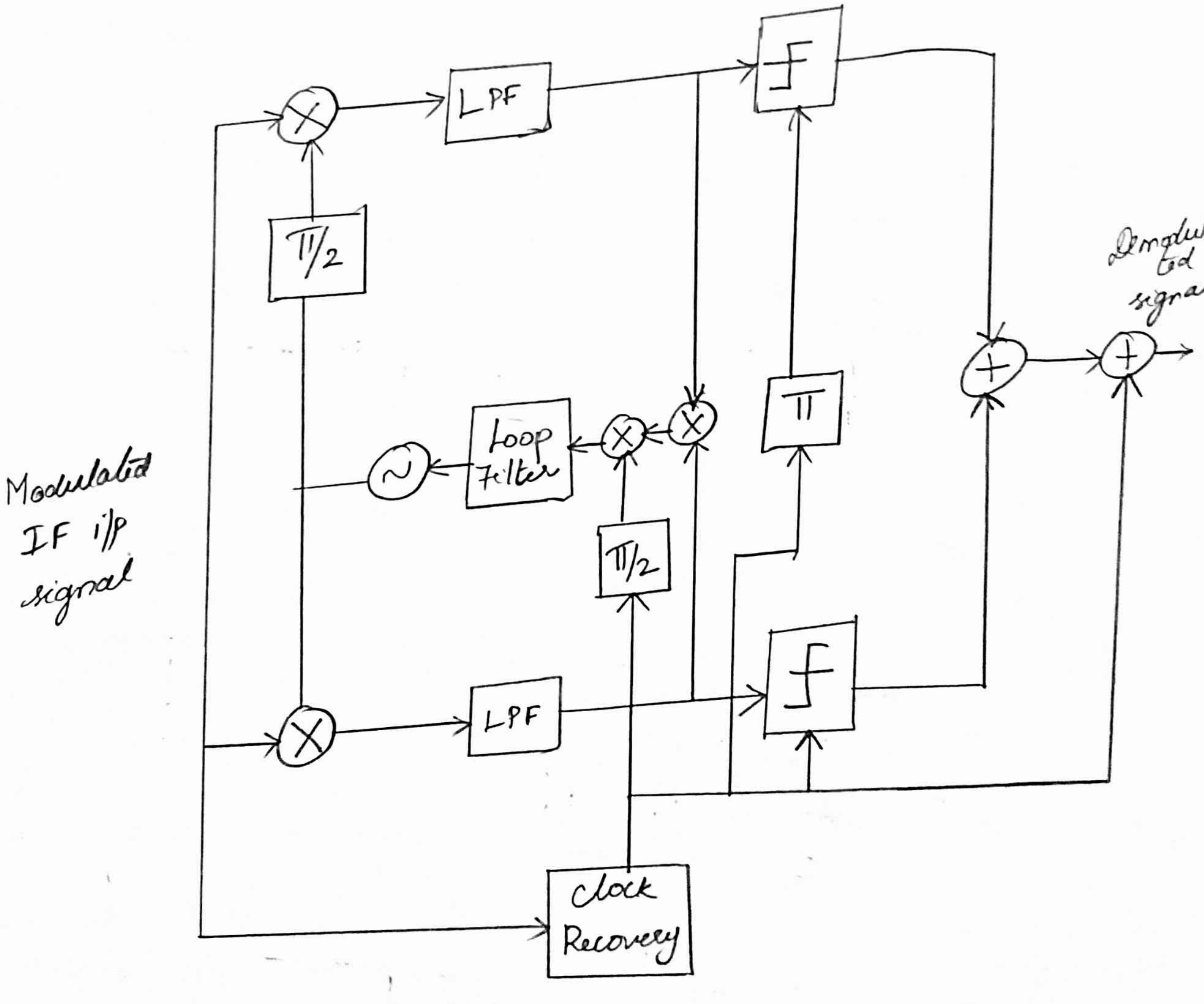


fig:- Block Diagram of a GMSK Receiver

* The two D-flip flops act as a quadrature product demodulator and the XOR gates act as baseband multipliers.

* The mutually orthogonal reference carriers are generated using two D flip flops.

* VCO center freq. is set equal to four times the carrier center freq.

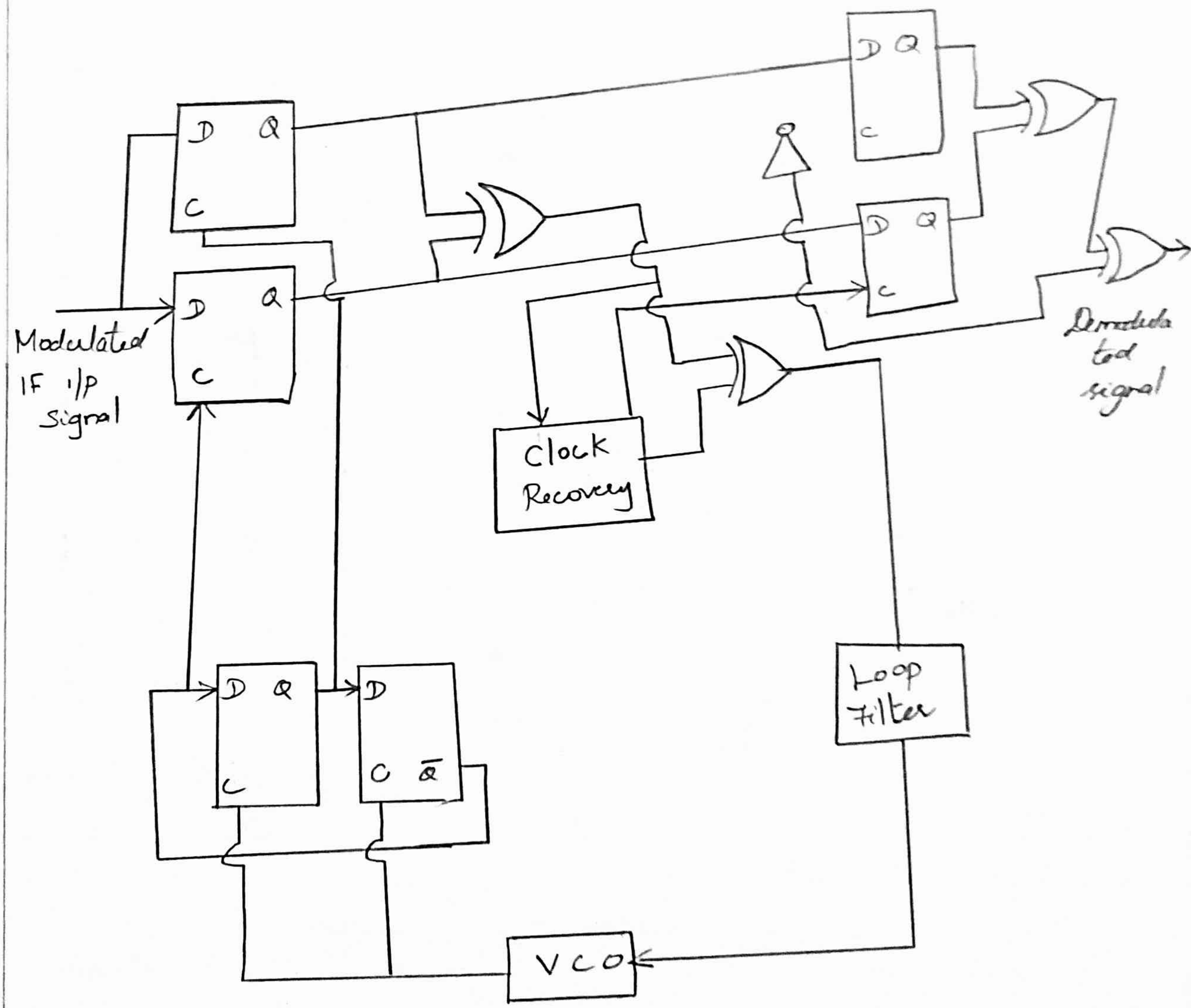


Fig:- Digital logic circuit for GMSK Demodulation

Error performance in fading channels:-

* The average BER has to be computed.
* It changes when fading of channel changes.
* In fading channel BER will decrease linearly with SNR.

* In AWGN Channels it will decrease exponentially.
* The Q function is given as

$$Q(x) = \frac{1}{\sqrt{2\pi}} \int_{x}^{\infty} \exp\left(-\frac{t^2}{2}\right) dt \quad \text{—} (1).$$

* This value is applied in computation of BER Q(x) can also be found as,

$$Q(x) = \frac{1}{\pi} \int_{0}^{\pi/2} \exp\left(\frac{-x^2}{2\sin^2\theta}\right) d\theta \quad \text{—} (2).$$

Reasons for errors:-

* In propagation channels the transmission errors are caused by signal distortions.
* They are initiated by delay dispersions and freq. dispersions.
* If the data rate is higher the delay dispersion is dominant
* These errors are called irreducible errors.

Modulation performance in fading and multipath channels:-

* The mobile radio channel is characterized by various impairments such as fading, multipath and Doppler spread.

* The Bit error rate (BER) evaluation gives a good indication of the performance of a particular modulation scheme it does not provide information about the type of errors.

* Evaluating the probability of outage is another means to judge the effectiveness of the signaling scheme in a mobile radio channel.

Performance of Digital modulation in slow flat fading channels:-

* Flat fading channels cause a multiplicative variation in the transmitted signal $s(t)$.

* Since slow flat fading channels change much slower than the applied modulation, it can be assumed that the attenuation and phase shift of the signal is constant over at least one symbol interval.

$r(t)$ is expressed as.

$$r(t) = \alpha(t)\, e^{-j\theta(t)}\, s(t) + n(t) \quad 0 \le t \le T.$$

where, $\alpha(t)$ – gain of the channel

$\theta(t)$ – phase shift of the channel

$n(t)$ – Additive Gaussian noise.

* The probability of error in AWGN channels is viewed as a conditional error probability, where the condition is that α is fixed
* The probability of error can be evaluated as,

$$P_e = \int_0^\infty P_e(X)\, p(X)\, dX.$$

where, $P_e(X)$ - probability of error for an arbitrary modulation at a specific value of signal to noise ratio X, $X = \alpha^2 E_b/N_0$.

* $p(X)$ - probability density function of α due to fading channels.
* E_b & N_0 are constants that represents the average energy per bit and noise power density in a non fading AWGN channel.
* α^2 is used to represent instantaneous power values of the fading channel, with respect to the non-fading E_b/N_0

$$p(X) = \frac{1}{\Gamma} e^{(-X/\Gamma)} \qquad X \geq 0$$

where $\Gamma = E_b/N_0\, \overline{\alpha^2}$.

* for $\overline{\alpha^2} = 1$, Γ corresponds to avg E_b/N_0 for fading channel.
* The probability of error in slow fading channel can be evaluated.
* For

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

Digital modulation in frequency selective mobile channels:-

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

* Simulation is the major tool used for analyzing freq. selective fading effects.

For coherent binary PSK

$$P_{e,PSK} = \frac{1}{2}\left[1 - \frac{\sqrt{\Gamma}}{1+\Gamma}\right]$$

For coherent binary FSK.

$$P_{e\,FSK} = \frac{1}{2}\left\{1 - \sqrt{\Gamma/2+\Gamma}\right\}$$

$$P_{e\,DPSK} = \frac{1}{2(1+\Gamma)}$$

For non coherent orthogonal binary FSK,

$$P_{e\,NCFSK} = \frac{1}{2+\Gamma}$$

For large values of E_b/N_0

$$P_{e,PSK} = \frac{1}{4\Gamma} \quad \text{(Coherent Binary PSK)}$$

$$P_{e\,FSK} = \frac{1}{2\Gamma} \quad \text{(Coherent FSK)}$$

$$P_{e,DPSK} = \frac{1}{2\Gamma} \quad \text{(Differential PSK)}$$

$$P_{e,NCFSK} = \frac{1}{\Gamma} \quad \text{(non coherent orthogonal binary FSK)}$$

$$P_{e,GMSK} = \frac{1}{2}\left(1 - \sqrt{\frac{\delta\Gamma}{\delta\Gamma+1}}\right)$$

$$= \frac{1}{4\delta\Gamma} \quad \text{(Coherent GMSK)}$$

Performance of π/4 DQPSK in fading and interference:-

* BER was calculated and analyzed as a fn of following parameters.
* The Doppler spread normalized to the symbol rate $B_D T_s$ or B_D/R_s.
* Delay of the second multipath τ, normalized to the symbol duration τ/T.
* Ratio of average carrier energy to noise power spectral density in decibels : E_b/N_0 dB.
* Average main path to delayed-path power ratio C/D dB.
* The irreducible error floor in a freq. selective channel is primarily caused by the errors due to ISI
* This occurs when.

(a) The main sgl component is removed through multipath cancellation.

(b) A non-zero values of 'd' causes ISI

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

* For large delay spreads, timing errors & ISI are dominant error mechanisms.

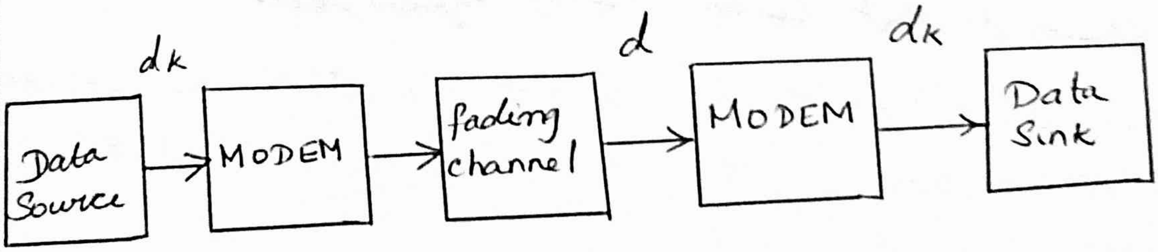


fig:- Block diagram of actual digital communication system

* Fung, Thoma & Rappaport developed a computer simulator called BERSIM (Bit error rate simulator).

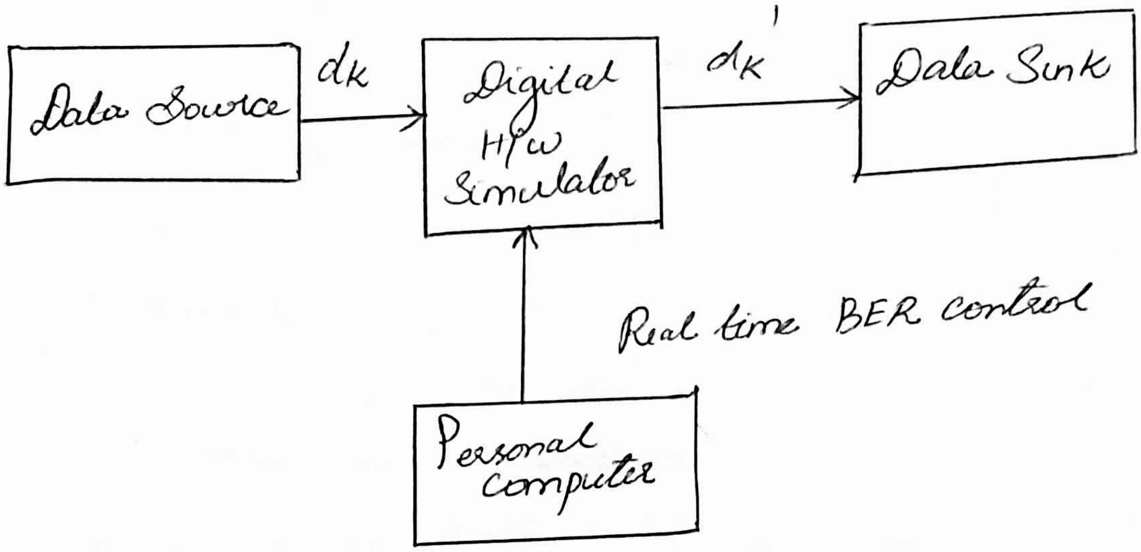


fig:- Block diagram of BERSIM

Orthogonal frequency division multiplexing (OFDM) :-

* OFDM is a modulation scheme that is especially suited for high-data rate transmission in delay dispersive environments.
* It converts a high rate data stream into a no. of low rate streams that are transmitted over ||el, narrow band channels that can be easily equalized.
* If symbol duration becomes very small, then the impulse response becomes very long in terms of symbol duration.

Applications of OFDM:-

* WLAN
* Evolution (3GPP - LTE)
* wimax. (WIMAX).

principle of orthogonal frequency division multiplexing:-

* OFDM splits a high rate data stream into N parallel streams, which are then tried by modulating N distinct carriers. Hence forth called sub carrier or tones.
* The sub carrier to be orthogonal.
* Narrow spacing of sub carriers can be achieved.
* Let sub carriers be at the frequencies.

$$f_n = nW/N,$$

where n is an integer

$W \rightarrow$ total available bandwidth

$W = N/T_s$

* sub carriers are mutually orthogonal, since the relationship

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

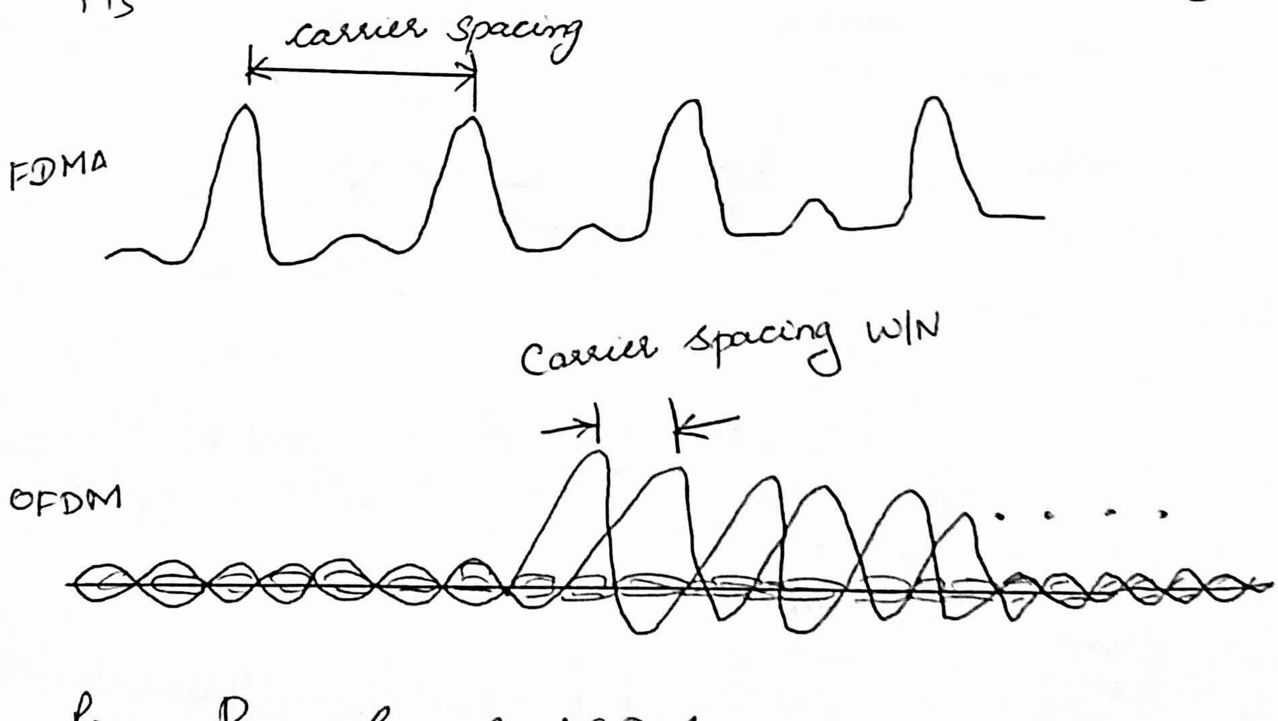


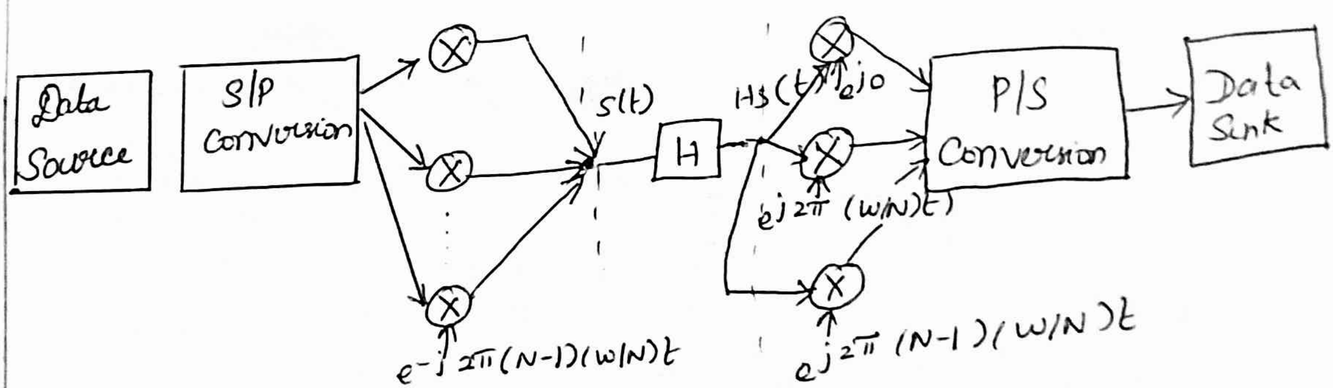
fig:- Principle of OFDM

* Due to the rectangular shape of pulses in the time domain, the spectrum of each modulated carrier has a sin (x) / x shape.

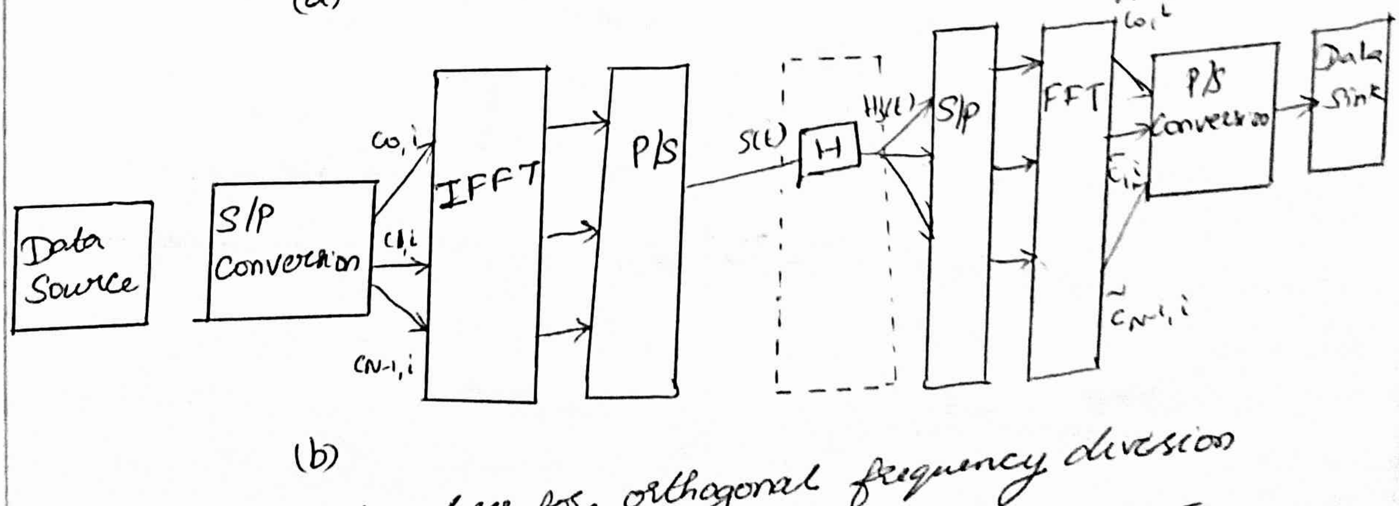
* The spectra of different modulated carriers overlap, but each carrier is in the spectral nulls of all other carriers.

Implementation of Transceivers:-

* OFDM can be interpreted in two ways
  ① Analog ② Digital.
* Local oscillators oscillates at a freq. $f_n = nW/N$, $n=0,1,\ldots N-1$.
* Each of the parallel data streams then modulates one of the carriers.



(a)



(b)

fig:- Transceiver structures for orthogonal frequency division multiplexing (a) Analog technology (b) Digital using IFFT

Digital implementation:-

* It first divides the transmit data into blocks of N symbols
* Each block of data is subjected to an inverse fast fourier Transformation (IFFT) and then transmitted.

Analog interpretation:-

* Let the complex transmit symbol at time instant i on the nth carrier be $C_{n,i}$ the transmit signal is then,

$$S(t) = \sum_{i=-\infty}^{\infty} S_i(t) = \sum_{i=-\infty}^{\infty} \sum_{n=0}^{N-1} C_{n,i}\, g_n(t - iT_s) \quad \text{--- (2)}$$

* where the basis pulse $g_n(t)$ is a normalized freq. shifted Rectangular pulse.

$$g_n(t) = \left\{ \frac{1}{\sqrt{T_s}} \exp\left(j 2\pi n \frac{t}{T_s}\right) \text{ for } 0 < t < T_s \quad \text{--- (3)}\right.$$

* Consider the signal only for $i=0$, and sample it at instances $t_k = k T_s / N$

$$S_k = S(t_k) = \frac{1}{\sqrt{T_s}} \sum_{n=0}^{N-1} C_{n,0} \exp\left(j 2\pi n \frac{k}{N}\right) \quad \text{--- (4)}$$

* This is inverse discrete fourier transform (DFT) of the transmit symbols.
* Input and output of IFFT is made up of N samples
* Analog implementation of OFDM would require multiple LOS, each of which has to operate with little phase noise and drift

Cyclic prefix:-

* Let us first define a new base fn for transmission.

$$g_n(t) = \exp[j 2\pi n \tfrac{\omega}{N} t] \text{ for } -T_{cp} < t < \hat{T}_s \quad \text{— (5)}$$

where, $\omega/N \to$ Carrier spacing

$\hat{T}_s = N/\omega$.

Symbol duration $T_s = \hat{T}_s + T_{cp}$

* In $0 < t < \hat{T}_s \to$ Normal OFDM symbol is Txed

$$g_n(t) = g_n(t + N/\omega)$$

* The total signal $s(t)$ Txed during time $-T_{cp} < t < 0$ is a copy of $s(t)$ during the last part, $\hat{T}_s - T_{cp} < t < \hat{T}_s$. This prepended part of the signal is called "cyclic prefix"

* cyclic prefix converts this linear convolution into circular convolution.

* τ_{max} — max. excess delay of the channel

without cyclic prefix.

$$\bar{g}_n(t) = \begin{cases} g_n^*(\hat{T}_s - t) & \text{for } 0 < t < \hat{T}_s \\ 0 & \text{o.w} \end{cases} \quad \text{— (6)}$$

* This operation removes the first part of the Rxed signal.

* The signal at the o/p of matched filter is thus convolution of the transmit signal with the channel impulse response and the receive filter.

$$r_{n,0} = \int_0^{\hat{T}_s} \left[\int_0^{T_{cp}} h(t,\tau) \left(\sum_{k=0}^{N-1} c_{k,0}\, g_k(t-\tau) \right) d\tau \right] g_n^*(t)\, dt + n_n \quad \text{— (7)}$$

where n_n is the noise at the o/p of matched filter.

(20)

* The basis for $g_n(t)$ are orthogonal during the time $0 \le t \le \hat{T}_s$

$$\int_0^{\hat{T}_s} g_k(t)\, g_n^*(t)\, dt = \delta_{kn}(0).$$

* The received signal samples r can be written as

$$r_{n,0} = H\left(n \frac{\omega}{N}\right) c_{n,0} + n_n \quad \text{—— (9)}$$

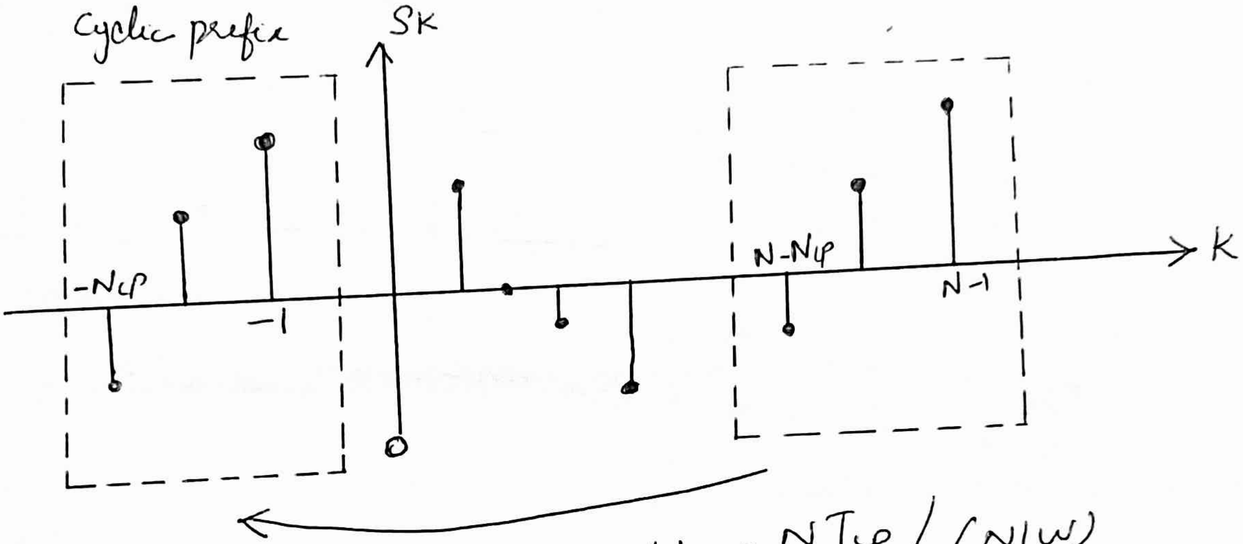


fig:- Principles of cyclic prefix $N_{cp} = N T_{cp} / (N/W)$

* Cyclic prefix has recovered the orthogonality of the subcarriers.

* Two caveats to be noted.

↳ we assumed in the derivation that the channel is static for the duration of the OFDM symbol.

↳ Discarding part of the received signal decreases the SNR as well as spectral efficiency.

* At the Rxer the signal is partitioned into blocks.

* For each block, the cyclic prefix is stripped off, and the remainder is subjected to an FFT.

* The resulting samples are equalized by means of one tap equalization - ie division by the complex channel attenuation on each carrier.

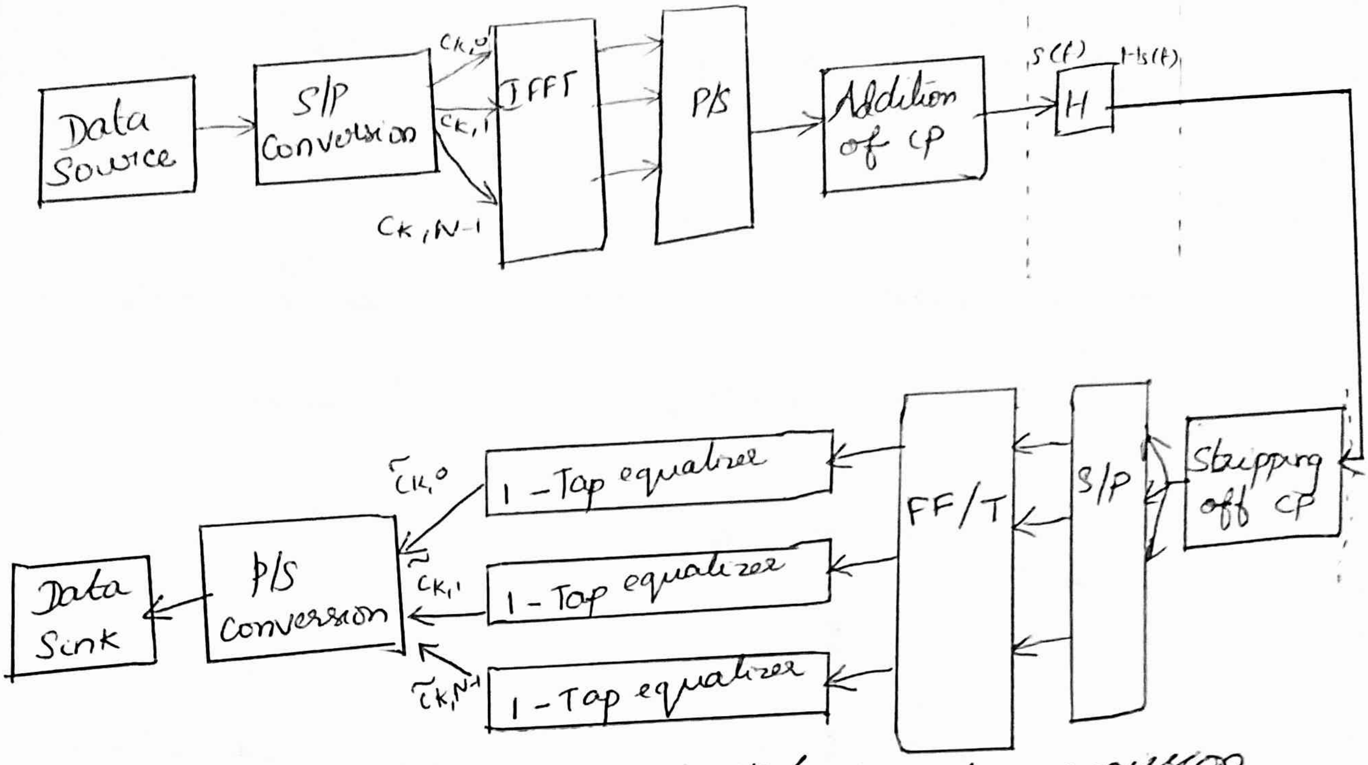


fig:- Structure of an OFDM multiplexing transmission chain with cyclic prefix and one-tap equalization

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Peak to average power ratio :-

* The PAPR issue originates from the fact that an OFDM signal is the superposition of N sinusoidal signals on different sub carriers.
* The amplitude of the signal is proportional to N, power goes with N^2.
* There are 3 main methods to deal with PAPR.

(1) Put a power amplifier into the transmitter that can amplify linearly upto the possible peak value of the transmit signal.

(2) Use a non linear amplifier and accept that amplifier characteristics will lead to distortions in the o/p signal

(3) Use PAPR reduction techniques.

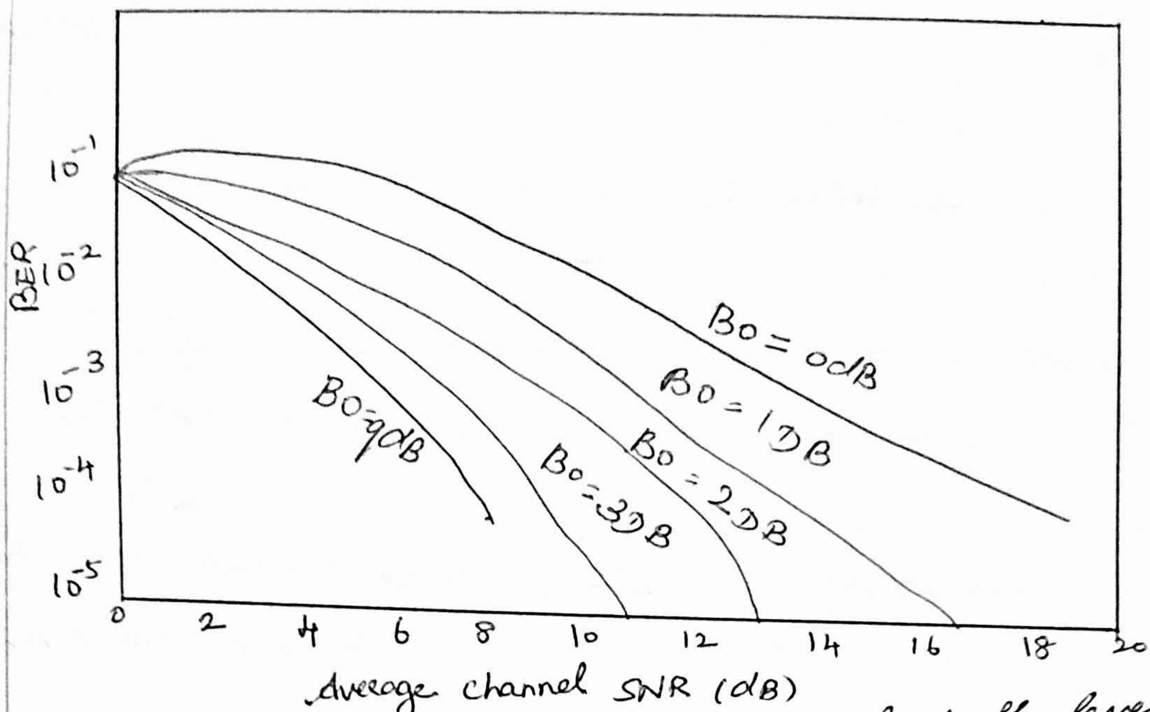


fig:- BER as a fn of SNR, for different back off levels of the transmit amplifier

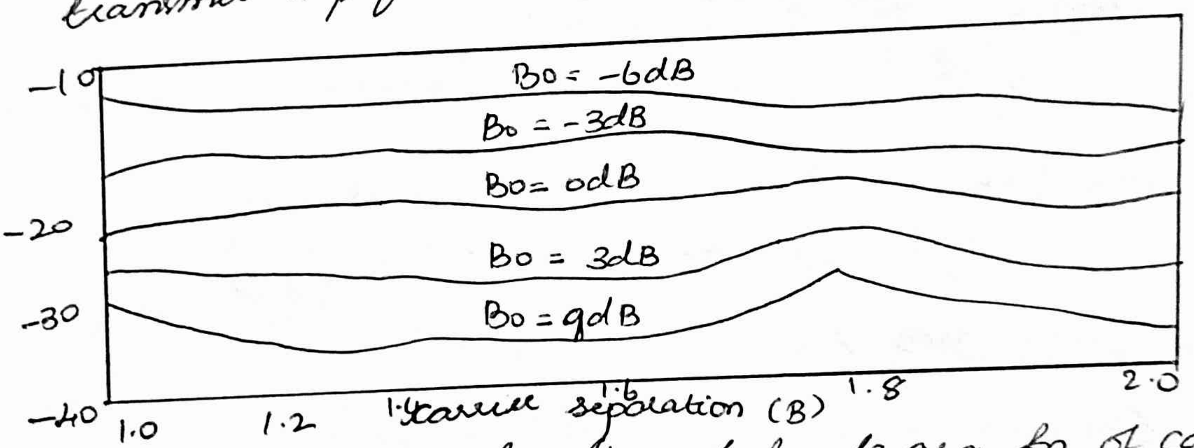


fig. - Interference power to adjacent bands, as a fn of carrier separation, for different values of backoff of the transmit amplifier

PAPR Reduction techniques:-

* Coding for PAPR reduction.
* Phase adjustments
* Correction by multiplicative fn.
* Correction by additive fn.

Coding for PAR reduction:-

* Each OFDM Symbol can represent one of 2^{N} Code word.
* PAR is lower than a given threshold
* The transmission scheme is thus the following.

(i) parse the incoming bit-stream into blocks of Length K.

(ii) Select the associated Code word of Length N.

(iii) Transmit this Code word via the OFDM modulator.

Phase adjustments:-

* First defines an ensemble of phase adjustment vectors $\phi_L, l = 1, \dots L$ that are known to both the txer & rxer.
* The txer then multiplies the OFDM Symbol to be txer C_n by each of these phase vector to get,

$$\{\hat{C}_n\}_l = C_n \exp[j(\phi_n)_l]$$

and then selects,

$$\hat{l} = \arg\min(PAPR(\{\hat{C}_n^{a}\}_l))$$

which gives the Lowest PAR.

Correction by multiplicative function:-

* To multiply the OFDM signal by a time dependent function whenever the peak value is very high.
* If the signal attains a level $S_x > A_0$, it is multiplied by a factor A_0/S_n. The transmit signal becomes,

$$\hat{S}(t) = S(t)\left[1 - \sum_{k} \max\left(0, \frac{|S_x| - A_0}{|S_x|}\right)\right]$$

$$\hat{S}(t) = S(t)\left[1 - \sum_{n} \max\left(0, \frac{|S_x| - A_0}{|S_x|}\right) \exp\left(-\frac{t^2}{2\sigma_t^2}\right)\right]$$

Collection by Additive function:-

* The collection function should be smooth enough not to introduce significant out of band interference.
* Correction fn. acts as additional pseudo noise and thus increases the BER of the system.

Windowing:-

Introduction:-

* Large signals are also difficult to analyze statistically, because statistical calculations require all points to be available for analysis.
* Window fn is a mathematical fn. that is zero valued outside of some chosen interval and is the process of taking a small subset of a larger data set for processing and analysis.

* The non linear distortion of the OFDM signal significantly increases the level of the out of band radiation.
* An OFDM signal consists of a number of unfiltered sub carriers.
* The out of band spectrum decreases rather slowly, with the speed depending on the number of subcarriers.

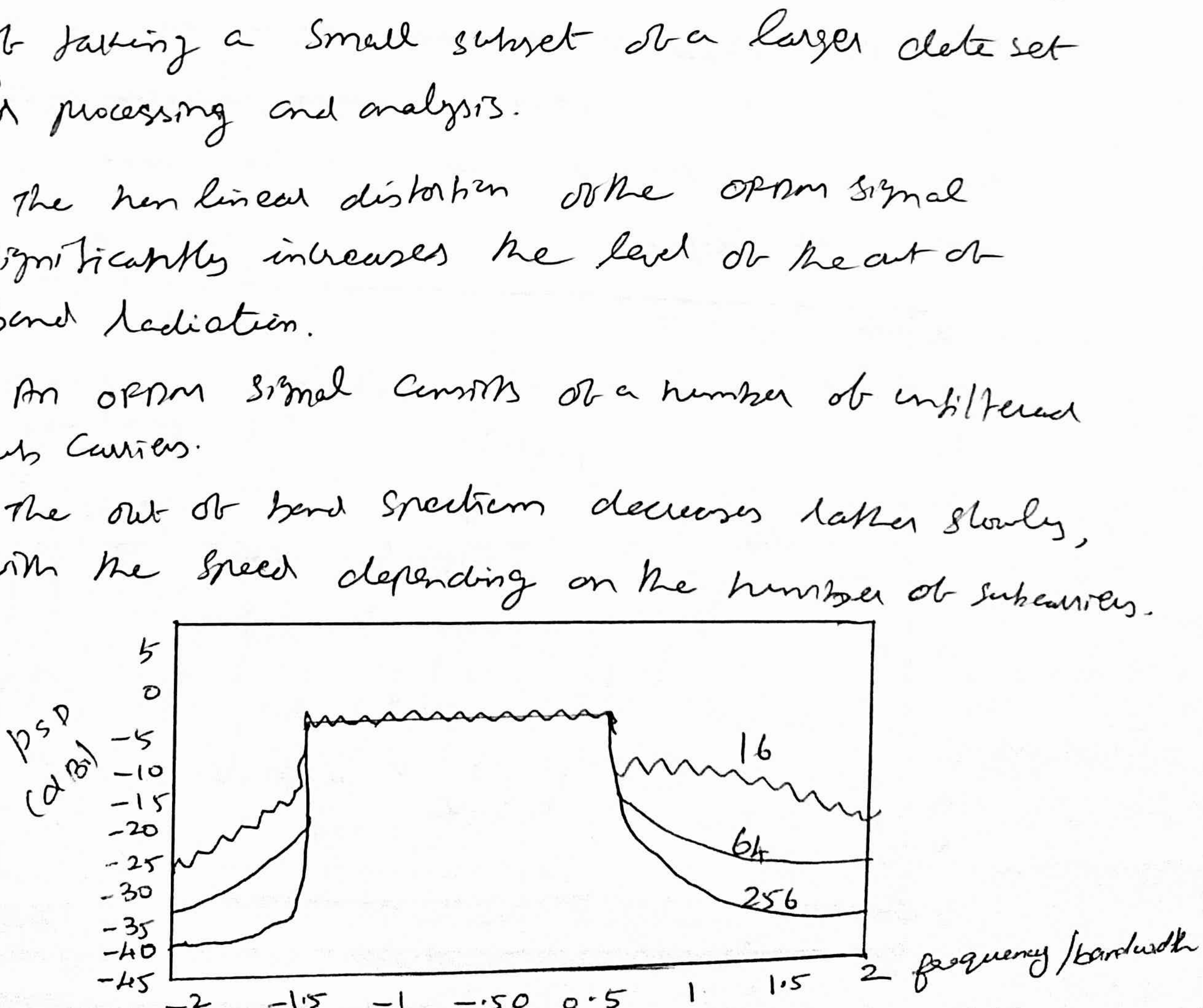


fig:- PSD without windowing for 16, 64, 256 subcarriers.

Raised Cosine window:-

* windowing can be applied to individual OFDM symbols. The mostly used windowing is the raised cosine window which is defined as,

$$w(t) = \begin{cases} 0.5 + 0.5\cos(\pi + t\pi / (\beta T_s)), & 0 \le t \le \beta T_s \\ 1.0 & \beta T_s \le t \le T_s \\ 0.5 + 0.5\cos((t - T_s)\pi / (\beta T_s)), & T_s \le t \le (1+\beta) T_s \end{cases}$$

where $\beta \rightarrow$ roll off factor.

Clipping and windowing:-

* The clipping operation is expressed as a multiplication of the OFDM signal by a rectangular window fn that equals one if the OFDM amplitude is below a threshold level and less than one when the amplitude needs to be clipped.

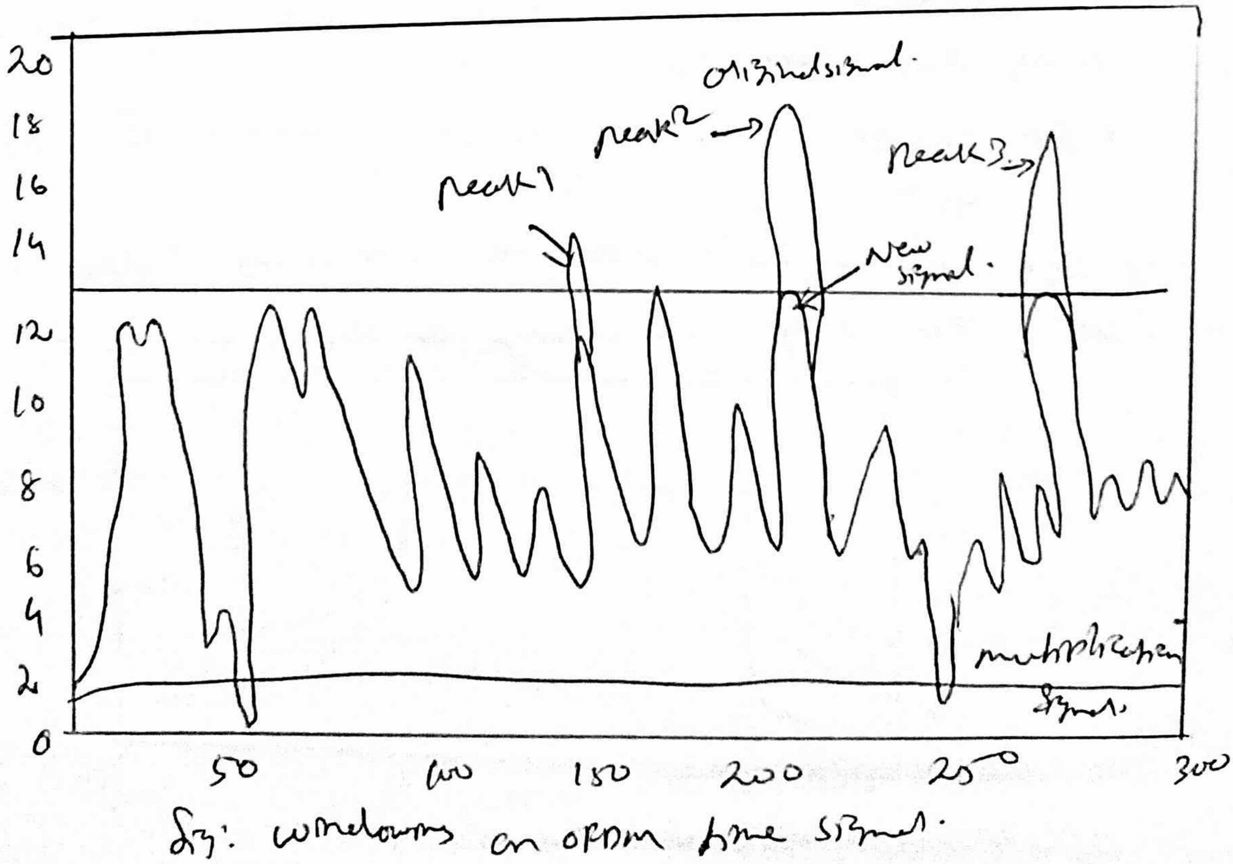


fig: windowing on OFDM time signal.

* The spectrum of the clipped OFDM signal is found as the i/p OFDM signal is found as the i/p OFDM spectrum combined with the spectrum of the window fn.
* The out of band spectral properties are mainly determined by the spectrum of the rectangular window fn.
* This spectrum has a very slow roll off that is inversely proportional to the freq.

Non-rectangular window:-

* To remedy the out of band problem of clipping, a different approach is to multiply the large signal peaks with a certain non rectangular window.
* The window should not be too long in terms of the time duration (T_p) because it implies that many signal samples may get affected, which increases the BER.

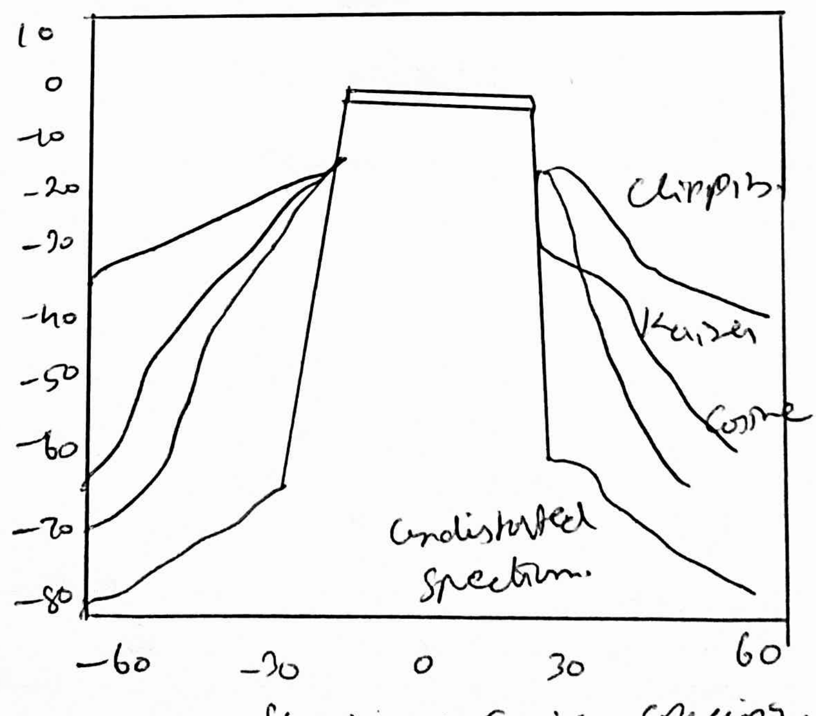


fig: freq. spectrum of an OFDM signal

Applications:-

* OFDM has been popularly standardized in many wireless applications such as,

(i) Digital video Broad Casting (DVB)

(ii) Digital Audio Broad Casting (DAB)

(iii) HIPERLAN

(iv) IEEE 802.11 (wifi) and.

(v) IEEE 802.16 (WIMAX).

Unit - IV

Multipath mitigation Techniques.

Equalisation - Adaptive Equalization linear and Non-linear Equalisation, Zero forcing and LMS Algorithms. Diversity - micro and macro diversity, Diversity combining techniques, Error probability in fading channels with diversity reception, Rake Receiver.

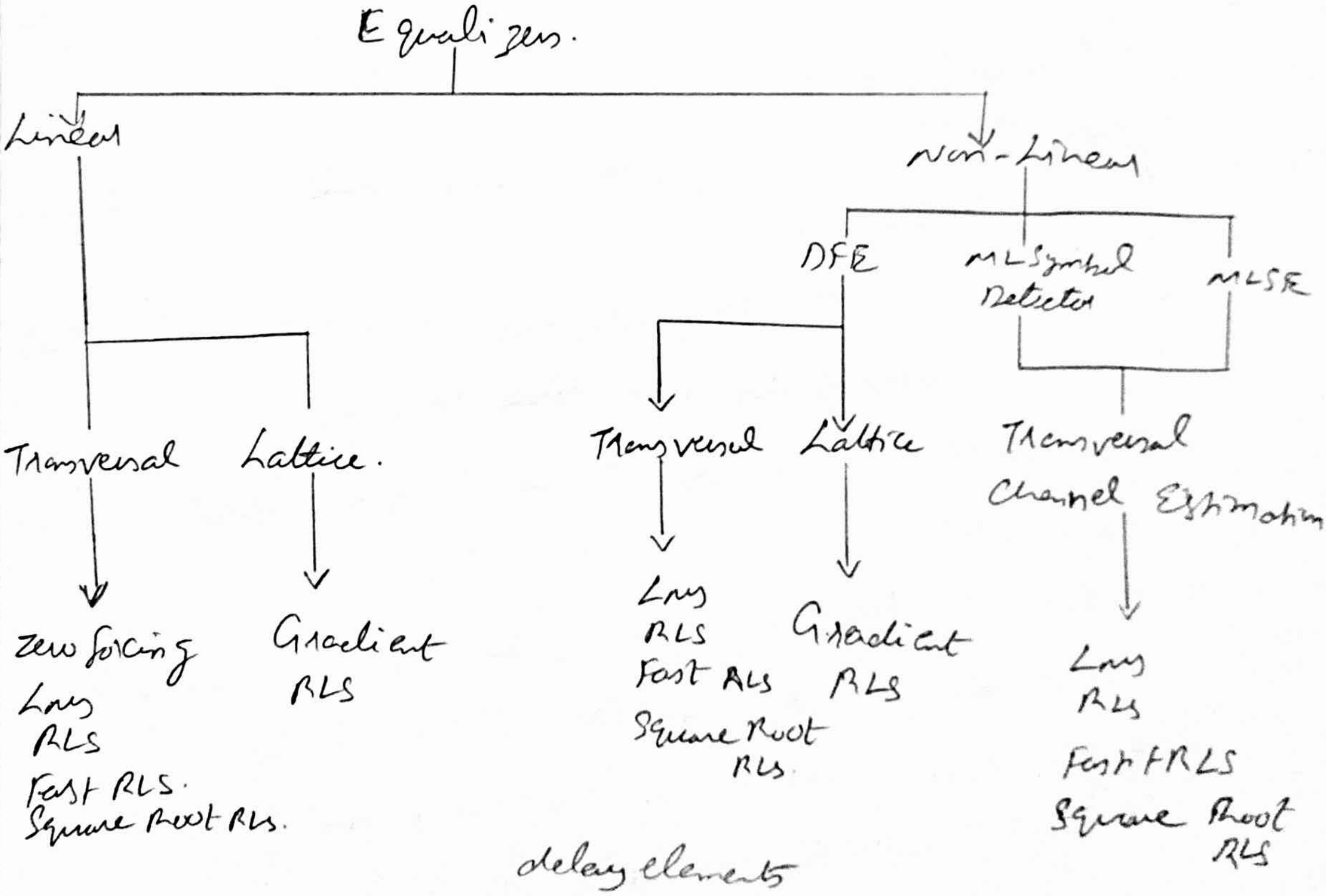
Equalization:-

* Equalization compensates for Intersymbol Interference (ISI) created by multipath within time dispersive channels.
* An equalizer within a receiver compensates for the average range of expected channel amplitude and delay characteristics.
* Equalizers must be adaptive since the channel is generally unknown and time varying.
* Equalisation is used to counter the effects of time dispersion ISI.
* Diversity techniques are often employed at both base station and mobile receivers.
* The most common diversity technique is called spatial diversity, whereby multiple antennas are strategically spaced and connected to a common receiving system

Other diversity techniques are

1. Antenna polarization diversity,
2. frequency diversity
3. Time diversity.

* Equalisation can be used to describe any signal processing operation that minimizes ISI.
* The mobile fading channel is random and time varying, Equalizers must track the time varying characteristics of the mobile channel, and thus are called "Adaptive Equalizers".



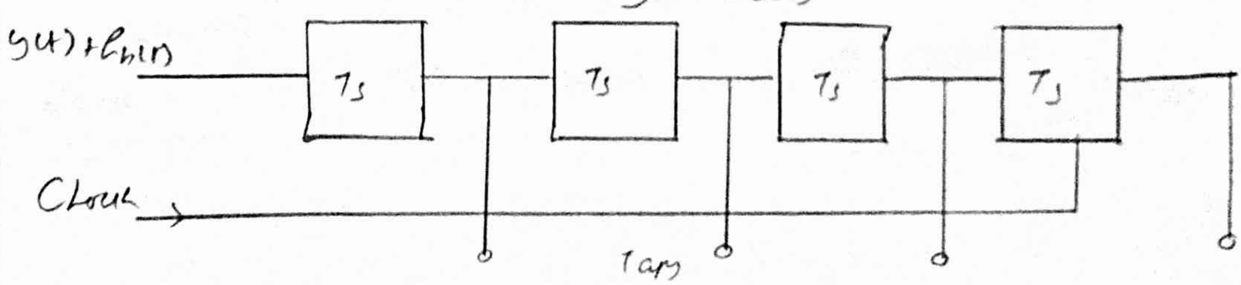


Fig: Basic Linear Transversal Equalizer Structure.

* The most common equalizer structure is linear transversal equalizer.
* A linear transversal filter is made up of tapped delay lines, with the tappings ~~operated~~ spaced a symbol period (T_s) apart.
* The transfer fn. of a linear transversal equalizer can be written as.

$$\exp(-j\omega T_s) \text{ or } z^{-1}$$

* This filter has many zeros but poles only at $z=0$ and is called a finite impulse response filter (FIR)
* If the equalizer has both feed forward and feedback fn of z^{-1} and is called an Infinite Impulse Response (IIR) filter with poles and zeros.

Adaptive equalization :-

* The mobile fading channel is random and time varying equalizers must track the time varying characteristics of the mobile channel and thus are called "Adaptive equalizers".
* If $x(t)$ is the original information signal and $f(t)$ is the combined complex baseband impulse response of the txer, channel and the RF/IF sections of the receiver.

$f(t)$ = Combined impulse response of transmitter multipath radio channel, & receiver RF/IF

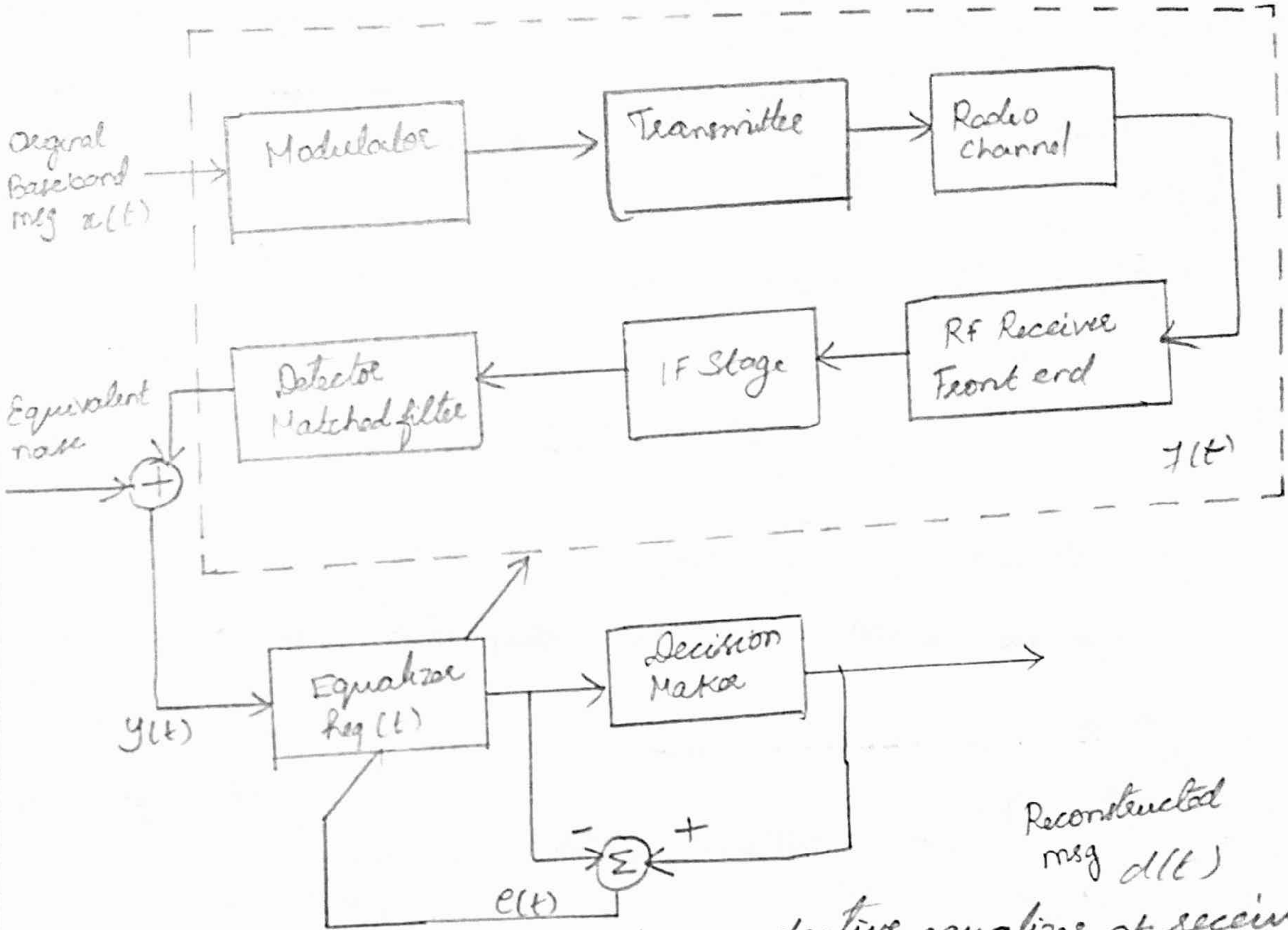


fig:- Simplified communication sym using an adaptive equalizer at receiver

* The signal received by the equalizer may be expressed as

$$y(t) = x(t) \otimes f^*(t) + n_b(t) \quad \text{—— (1)}$$

where $f^*(t) \rightarrow$ the complex conjugate of $f(t)$

$n_b(t) \rightarrow$ base band noise at the i/p of the equalizer.

$\otimes \rightarrow$ Convolution operation.

If the Impulse response of the equalizer is $h_{eq}(t)$, then the o/p of the equalizer is

$$\hat{d}(t) = x(t) \otimes f^*(t) \otimes h_{eq}(t) + n_b(t)\, h_{eq}(t) \quad \text{—— (2)}$$

$$= x(t) \otimes g(t) + n_b(t)\, h_{eq}(t).$$

where $g(t) \Rightarrow$ Combined Impulse Response.

* The complex baseband impulse response of a transversal filter equalizer is given by,

$$h_{eq}(t) = \sum_n c_n\, \delta(t - nT) \quad \text{— (3)}$$

* Assume $n_b(t) = 0$, to force $\hat{d}(t) = x(t)$ in eqn (2), $g(t)$ must be equal to

$$g(t) = f^*(t) \otimes h_{eq}(t) = \delta(t) \quad \text{— (4)}$$

In freq domain

$$H_{eq}(f)\, F^*(-f) = 1 \quad \text{— (5)}$$

where $H_{eq}(f)$ and $F(f)$ are fourier transforms of $h_{eq}(t)$ and $f(t)$ respectively.

Linear Equalizers:-

* A linear equalizer can be implemented as an FIR filter, otherwise known as the transversal filter.

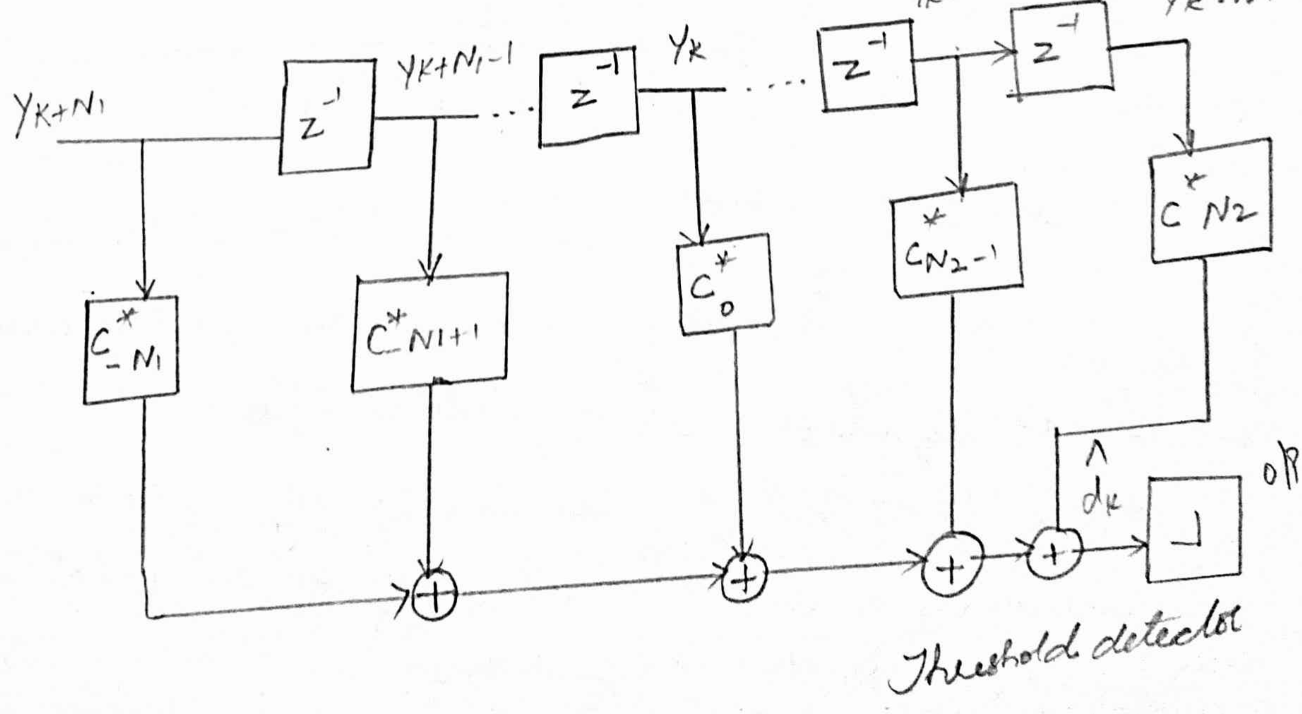


fig:- Structure of a linear transversal equalizer.

* If the delays and the tap gains are analog, the continuous output of the equalizer is sampled at the symbol rate and the sample are applied to the decision device.
* Samples of the received signal are stored in a shift register.
* The output of this transversal filter before a decision is made is,

$$\hat{d}_k = \sum_{n=-N_1}^{N_2} (c_n^*)\, y_{k-n} \quad \text{—} \ (7)$$

where $c_n^* \rightarrow$ complex filter co-efficients or tap weights.

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

$y_i \rightarrow$ I/p received signal at time $t_0 + iT$,

* The min. mean squared error $E[|e(n)|^2]$ that a linear transversal equalizer can achieve is,

$$E[|e(n)|^2] = \frac{T}{2\pi}\int_{-\pi/T}^{\pi/T} \frac{N_o}{|F(e^{j\omega T})|^2 + N_o}\, d\omega \quad \text{—} \ (2)$$

where $F(e^{j\omega T}) \rightarrow$ freq. response of the channel

$N_o \rightarrow$ Noise power spectral density.

Linear Equalizer as a lattice filter:-

* The i/p signal y_k is transformed into a set of N intermediate forward and backward error signals $f_n(k)$ and $b_n(k)$ respectively.

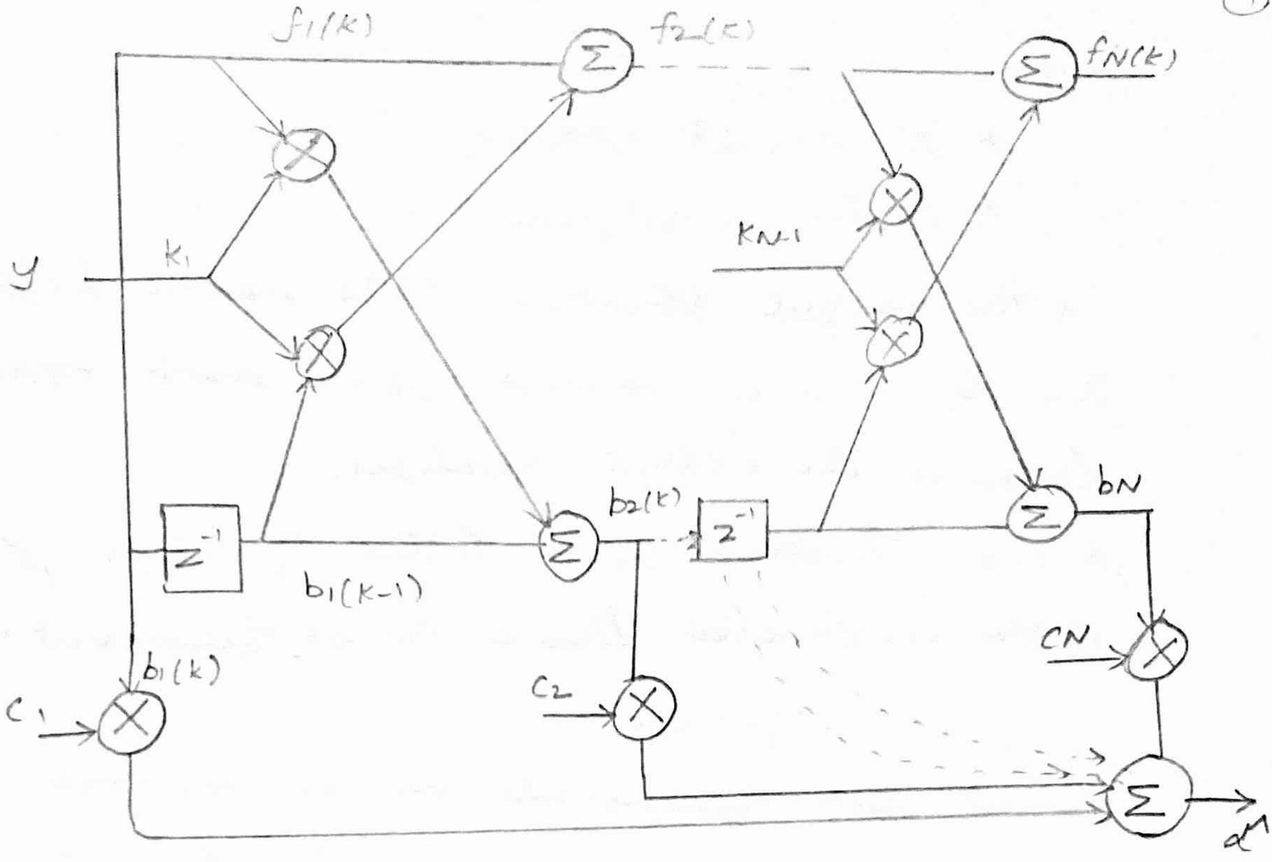


fig:- Structure of a Lattice equalizer

* which are used as i/ps to the top multiplexers and are used to calculate the updated co-efficients.

* Each stage of the Lattice is then characterising by the following recursive equations.

$$f_1(k) = b_1(k) = y(k) \quad \text{—} \ (2)$$

$$f_n(k) = y(k) - \sum_{i=1}^{n} k_i y(k-i) = f_{n-1}(k) + k_{n-1}(k) b_{n-1}(k+1)) \quad \text{—} \ (3)$$

$$b_n(k) = y(k-n) - \sum_{i=1}^{n} k_i y(k-n+i) \quad \text{—} \ (4)$$

$$= b_{n-1}(k-1) + k_{n-1}(k) f_{n-1}(k).$$

when the o/p of the equalizer is given by,

$$\hat{d}_k = \sum_{n=1}^{N} c_n(k) b_n(k) \quad \text{—} \ (5).$$

Advantages of Linear Equalizer:-

* Numerical Stability.
* faster convergence.
* The unique structure of the Lattice filter allows the dynamic assignment of the most effective length of the Lattice equalizer.
* The structure of a lattice equalizer, however is more complicated than a linear transversal equalizer.

Non linear Equalization:-

Non linear equalizers are used in applications where the channel distortion is too severe for a linear equalizer to handle.

Disadvantages of Linear Equalizer:-

Linear equalizers do not perform well on channels which have deep spectral null in the Passband.

Linear equalizer places too much gain.

Three effective non linear methods have been developed, they are

1. Decision Feedback Equalization (DFE)
2. Maximum Likelihood Symbol Detection.
3. Maximum Likelihood Sequence Estimation (MLSE).

Decision Feedback equalization (DFE):-

The basic idea behind decision feedback equalization is that once an decided upon, the ISI includes on future symbols can be estimated and subtracted out before the detection of subsequent symbols.

The DFE can be realized in either the direct transversal form or as a lattice filter.

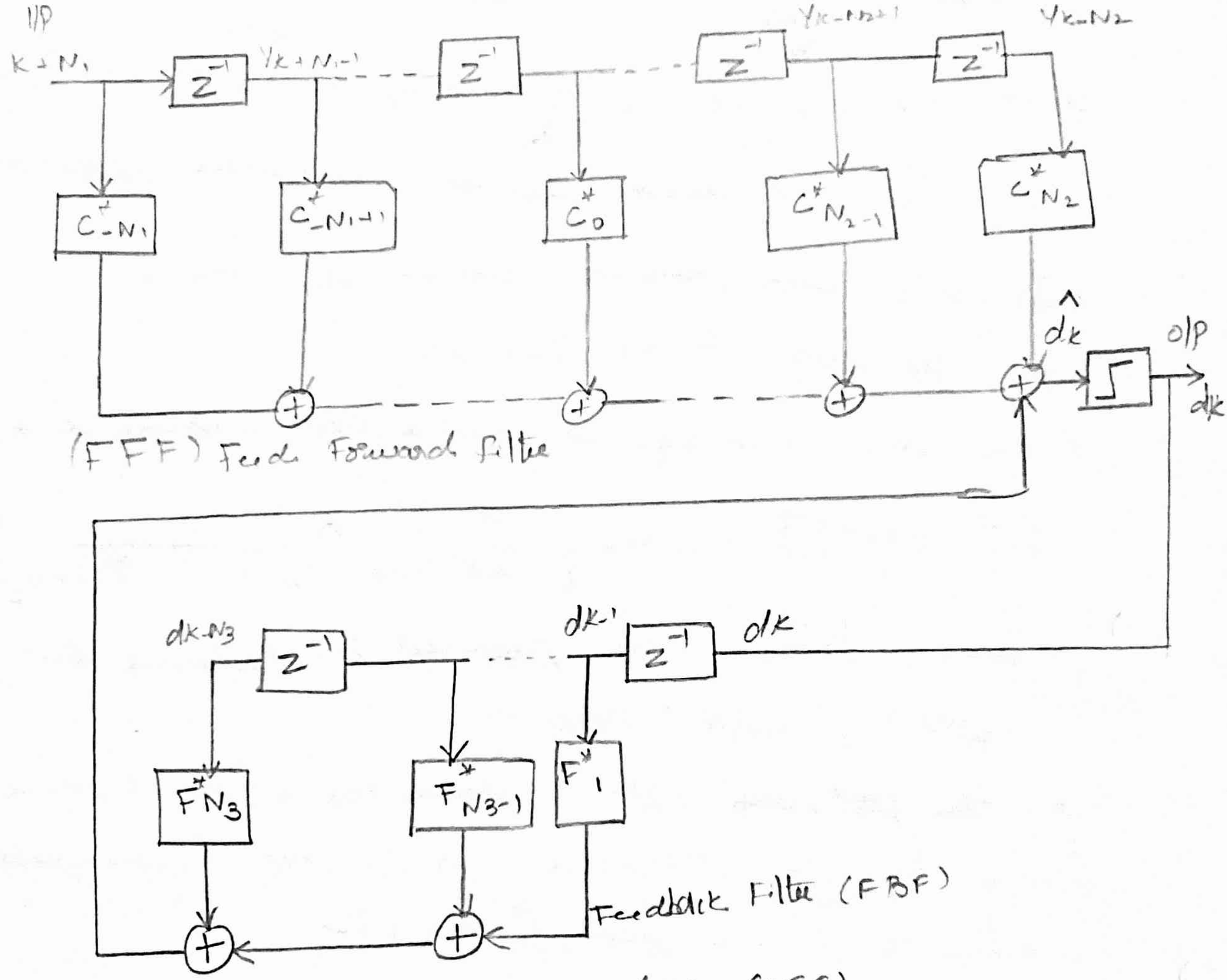


fig:- Decision Feedback Equalizer (DFE)

* It consist of a feed forward filter (FFF) and a feedback filter (FBF).

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

* The equalizer has N_1+N_2+1 taps in the feed forward filter and N_3 taps on the feedback filter.

* output can be expressed as.

$$\hat{d}_K = \sum_{n=-N_1}^{N_2} \hat{C}_n\, y_{K-n} + \sum_{i=1}^{N_3} F_i d_{K-i}$$

where $\hat{C}_n$ & y_n → tap gains & i/p to the forward filter.

d_i → previous decision made on the detector.

* d_K along with previous decisions d_{K-1}, d_{K-2} ---
are fed back into the equalizer.

* The min. mean squared error a DFE can achieve is,

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

* Another form of DFE proposed by Belfiore and Park is called predictive DFE.

* The feedback filter is driven by an i/p sequence formed by the difference of the o/p of the detector and the o/p of the feed forward filter.

* Hence the FBF here is called a noise predictor.

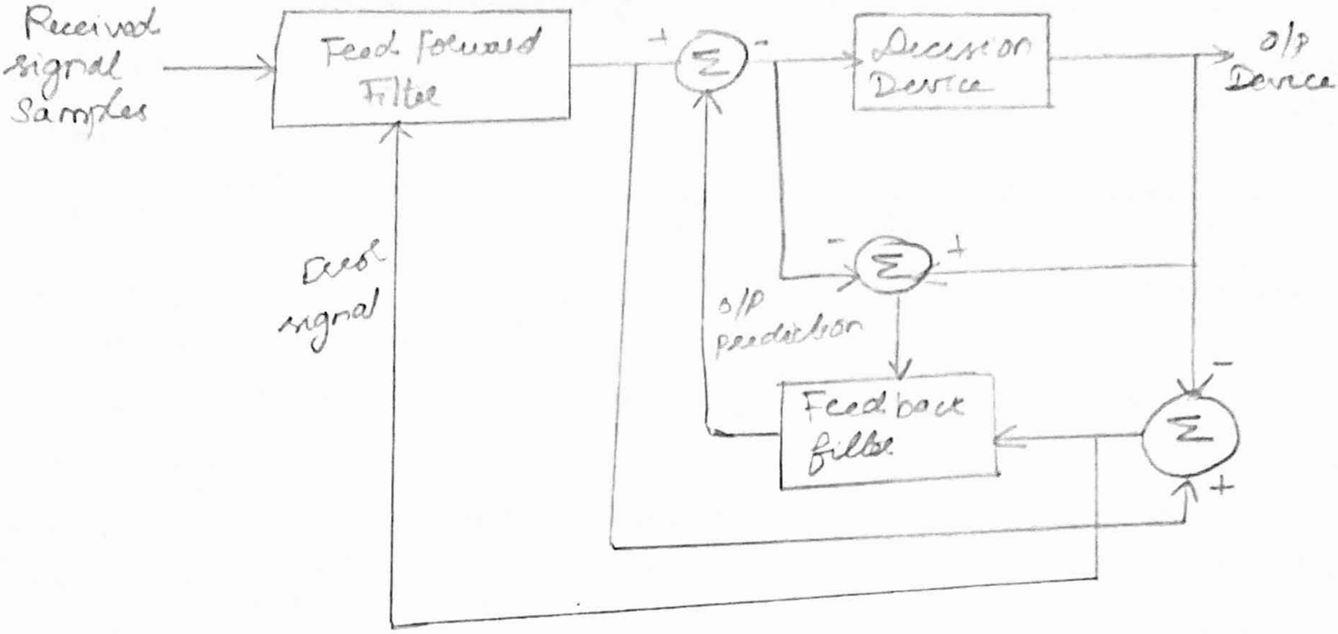


fig:- Predictive decision feedback equalizer

maximum Likelihood sequence Estimation (MLSE) equaliser:-

* using a channel Impulse Response Simulator within the algorithm, the MLSE tests all possible data sequences, and chooses the data sequence with the max. probability as the o/p.

* Basic MLSE estimator structure and implemented with it the Viterbi Algorithm.

* The radio channel with co efficients h_K and with a channel state which at any instant of time is estimated by the Receiver based L most recent i/p samples.

* The channel has M^L states, where M is the size of the symbol alphabet of the modulation.

* The Viterbi algorithm then tracks the state of the channel by the paths through the trellis.

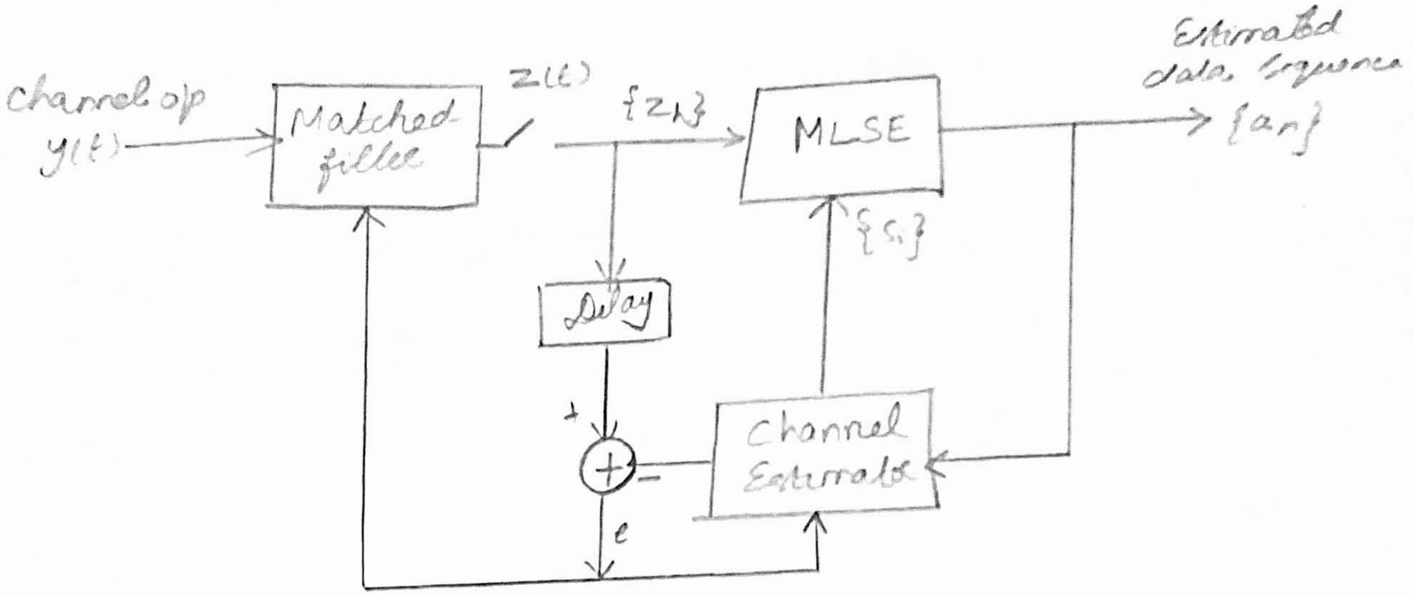


fig:- Structure of a maximum likelihood sequence

* The MLSE minimizes the probability of a sequence error.
* The MLSE requires knowledge of the channel characteristics in order to compute the metrics for making decisions.
* The MLSE also requires knowledge of the statistical distribution of the noise corrupting the signal.
* Matched filter operates on the continuous time signal, MLSE and channel estimator rely on discretized samples.

Algorithms for Adaptive Equalisation:-

* Since an adaptive equalizer compensates for an unknown and time varying channels, it require a specific algorithm to update the equalizer coefficient and track the channel variation.

* A wide range of algorithms exist to adapt the filter coefficients.

* performance of an algorithm is determined by various factors.

① Rate of convergence:-

It is defined as the no. of iterations required for the algorithm in response to stationary i/ps to converge close enough to the optimum solution.

② misadjustment:-

This parameter provides a quantitative measure of the amount by which the final value of mean square error, averaged ensemble of adaptive filters, deviates from the optimal min. mean square error.

③ Computational complexity:-

The no. of operations required to make one complete iteration of the algorithm.

④ numerical properties:-

when an algorithm is implemented numerical. inaccuracies are produced due to round off.

noise and representation errors in the computer

Three classic Equalizer algorithm:-

(a) zero forcing algorithm

(b) Least mean square algorithm

(c) Recursive Least square algorithm.

Zero forcing Algorithm:-

* In a zero forcing Equalizer, the equalizer coefficients C_n are chosen to force the samples of the combined channel and Equalizer impulse response to zero.

* Let the no. of coefficients increase without bound an infinite length equalizer with zero ISI at the o/p can be obtained.

* Delay elements provide a time delay equal to the symbol duration T.

* The freq. response $H_{eq}(f)$ of the equalizer is periodic with a period equal to the symbol rate $1/T$.

* The combined response of the channel with the equalizer must satisfy Nyquist's first criterion.

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

where $H_{ch}(f) \rightarrow$ folded freq. response of the channel.

$H_{eq}(f) \rightarrow$ freq. response of the equalizer.

* Thus an infinite length zero ISI equalizer is simply an inverse filter which inverts the folded freq. response of the channel.

Advantages:-

performs well for static channels with high SNR.

Disadvantages:-

* Inverse filter excessively amplifies noise, at certain freq.
* Neglects noise together, hence ZF equalizer is not often used for wireless link.

Least mean square Algorithm (LMSE):-

* A more equalizer is the LMS equalizer where the criterion used is the minimization of the mean square error [MSE] b/w the desired equalizer o/p and the equalizer o/p.

* The prediction error is given by

$$e_k = d_k - \hat{d}_k$$

$$= x_k - \hat{d}_k \quad \text{--- (1)}$$

where, $e_k = x_k - y_k^T w_k = x_k - w_k^T y_k$ --- (2).

$$\hat{d}_k = y_k^T w_k = w_k^T y_k.$$

To compute the mean square error $|e_k|^2$ at the time instant k.

$$\xi = E\left[e_k^* e_k\right] \quad \text{--- (3)}.$$

* For a specific channel condition, the e_k is dependent on the tap gain vector w_N.
* Let the cost fn $J(w_N)$ denote the mean squared error as a fn. of tap gain w_N.

$$\partial / \partial w_N \; J(w_N) = -2P_N + 2R_{NN} w_N = 0 \quad \text{—— (4)}$$

simplifying eqn (4).

$$R_{NN} \hat{w_N} = P_N \quad \text{—— (5)}$$

This equation is called normal equation.

The MMSE of the equalizer is

$$J_{opt} = J(\hat{w_N}) = E[x_k x_k^*] - P_N^T \hat{w_N} \quad \text{—— (6)}$$

* To obtain the optimal tap gain vector $\hat{w_N}$, the equalizer converges to an acceptably small value of J_{opt}.

$$\hat{w} = R_{NN}^{-1} P_N \quad \text{—— (7)}$$

* Gaussian elimination and Cholesky factorization require $O(N^2)$ operations per iteration.
* LMS Algorithm requires only $2N+1$ operations per iteration.
* LMS is computed iteratively by,

$$\hat{d_k}(n) = w_N^T(n)\, y_N(n) \quad \text{—— (8)a}$$

$$e_k(n) = x_k(n) - \hat{d_k}(n) \quad \text{—— (8)b}$$

$$w_N(n+1) = w_N(n) - \alpha\, e_k^*(n)\, y_N(n) \quad \text{—— (8)c}$$

N → No. of delay stages in the equalizer.

α → Step size.

To prevent the adaptation from becoming unstable the value of α is chosen from

$$0 < \alpha < 2 / \sum_{i=1}^{N} \lambda_i \quad \text{--- (9)}$$

where λ_i → ith eigen value of the covariance matrix R_{NN}.

Advantages:-

* Simple & offers low computational complex.
* Maximizes signal to distortion ratio.

Disadvantages:-

* Slower convergence and takes more no of iterations.

Diversity:-

"The principle of diversity is to ensure that the same information reaches the receiver (RX) on statistically independent channels".

"The correlation co-efficient characterizes the correlation b/w signals on different diversity branches".

* Diversity requires no training overhead since a training sequence is not required by the transmitter.

* In order to prevent deep fades from occurring, microscopic diversity techniques can exploit the rapidly changing signal.

* By selecting the best signal at all times a receiver can mitigate small scale fading effects. (This is called Antenna diversity or Space diversity).

* By selecting a BS which is not shadowed when others are the mobile can improve substantially the average SNR ratio on the forward link. This is called macroscopic diversity.

Micro diversity:.

* methods that can be used to combat small scale fading which are therefore called "micro diversity".

The five most common methods are

1. Spatial diversity
2. Temporal diversity.
3. Freq. diversity
4. Angular diversity
5. Polarization diversity.

Space diversity or spatial diversity or Antenna diversity:-

* space diversity, also known as antenna diversity is one of the most popular forms of diversity used in wireless systems.
* Jakes deduced that the signals received from spatially separated antennas on the mobile would have essentially uncorrelated envelopes for antenna separations of one half wavelength or more.
* At each cell size, multiple BS receiving antennas are used to provide diversity reception.
* separations on the order of several tens of wavelengths are required at the base station.

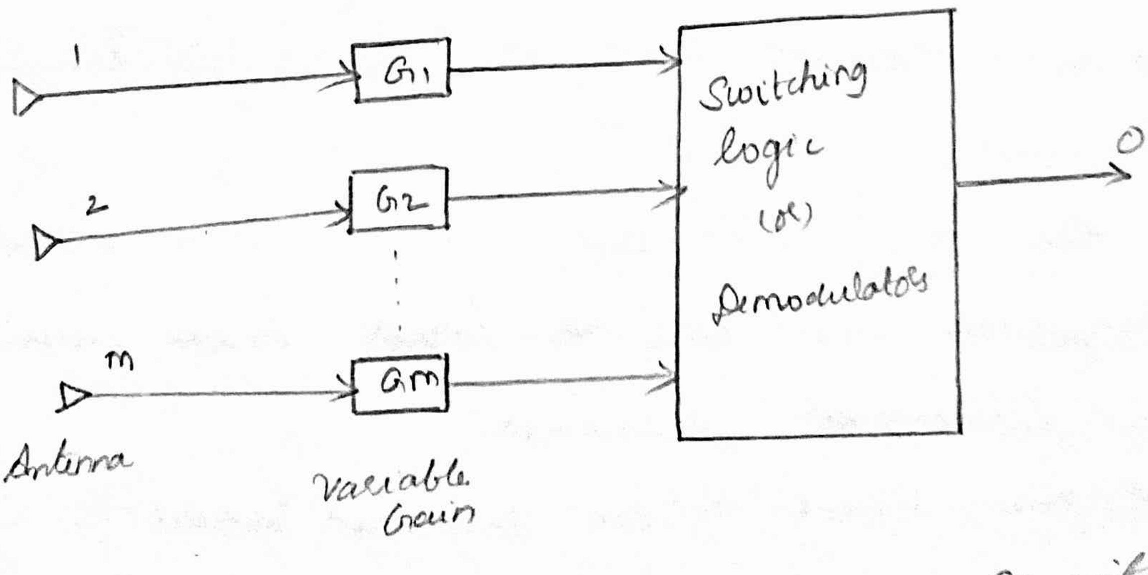


fig:- Generalized block diagram for space Diversity

Space diversity reception methods can be classified into four categories.

1. Selection diversity.
2. Feedback diversity.
3. Maximal ratio combining.
4. Equal gain Diversity.

Selection Diversity:-

* where m demodulators are used to provide m diversity branches whose gains are adjusted to provide the same average SNR for each branch.
* the branch with the Largest (S+N)/N is used since it is difficult to measure SNR alone.

Feedback or Scanning diversity.

* Scanning diversity is very similar to selection diversity except that instead of always using the best of M signals.
* The M signals are scanned in a fixed sequence until one is found to be above a predetermined threshold.
* The signal is the received until it falls below threshold and the scanning ~~[illegible]~~ process.

Advantages:-

* very simple to implement.
* only one receiver is required.

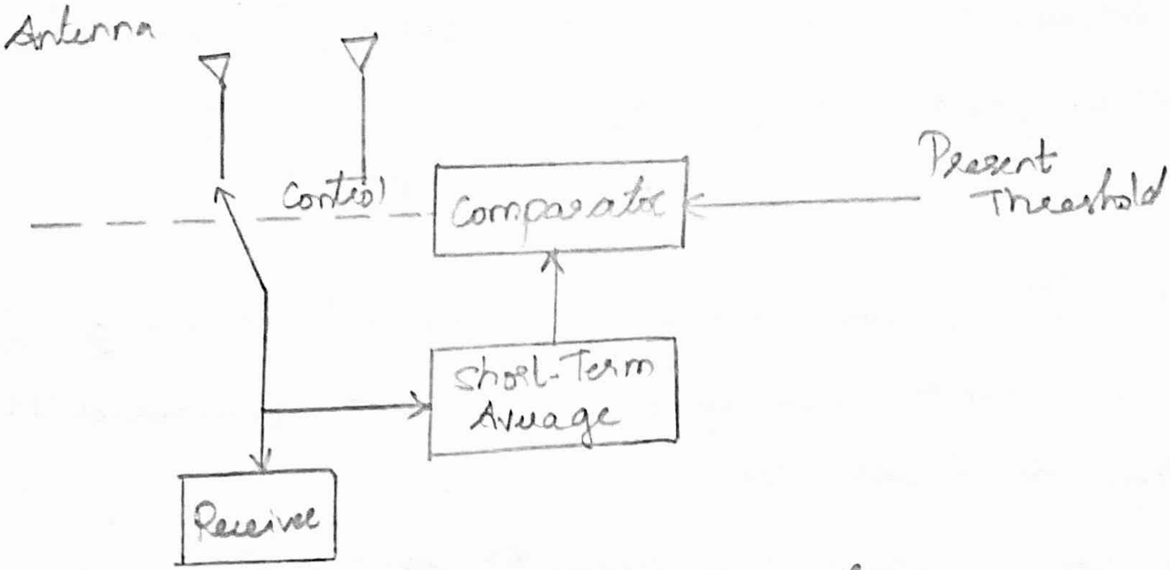


fig:- Basic form of scanning Diversity

Maximal Ratio Combining:-

* The signals from all of the M branches are weighted according to their individual signal voltage to noise power ratios and then summed.

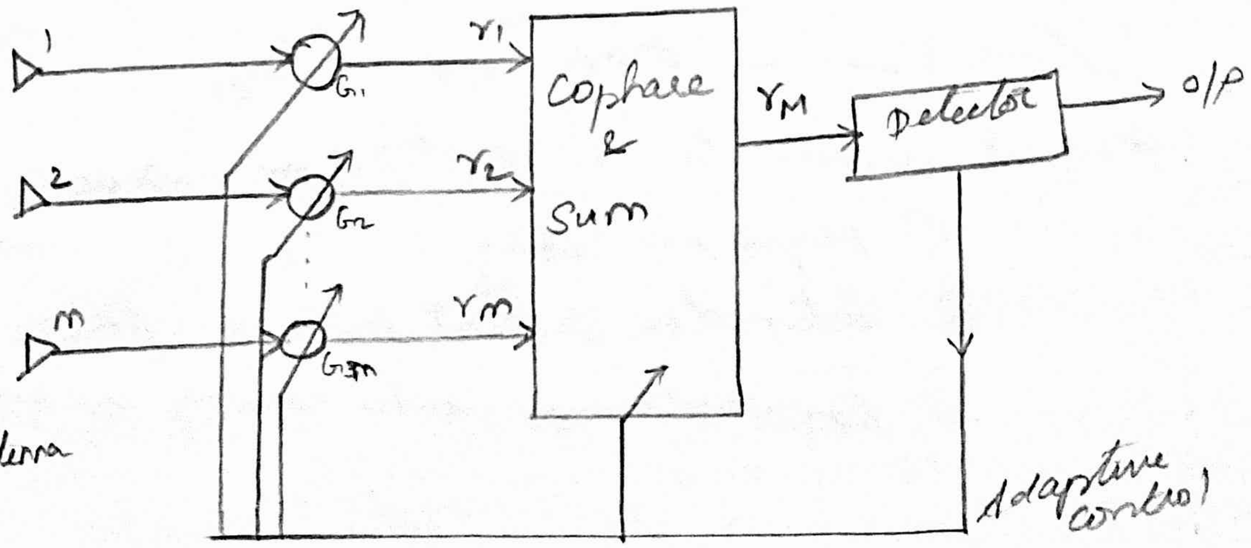


fig:- Maximal Ratio Combiner

* The individual signals must be co-phased before being summed.

* produce an o/p with an acceptable SNR even when none of the individual signals are themselves acceptables.

Equal gain combining:-

* The branch weights are all set to unity, but the signals from each branch are co-phased to provide Equal gain combining diversity.

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

Temporal diversity:-

* As the wireless propagation channel is time variant signals that are received at different times are un correlated.

* For sufficient decorrelation, the temporal distance must be atleast $1/2 V_{max}$

where V_{max} → max. doppler freq.

Temporal diversity can be realized in different ways.

① Repetition coding

② Automatic Repeat Request (ARQ)

③ Combination of interleaving and coding.

Frequency diversity:-

* In freq. diversity, the same signal is txed at two or more different frequencies.

$$\rho = \frac{1}{1+(2\pi)^2 s_\tau^2 (f_2 - f_1)^2}$$

* This gain confirms that two signals have to be atleast one coherence bandwidth apart from each other.

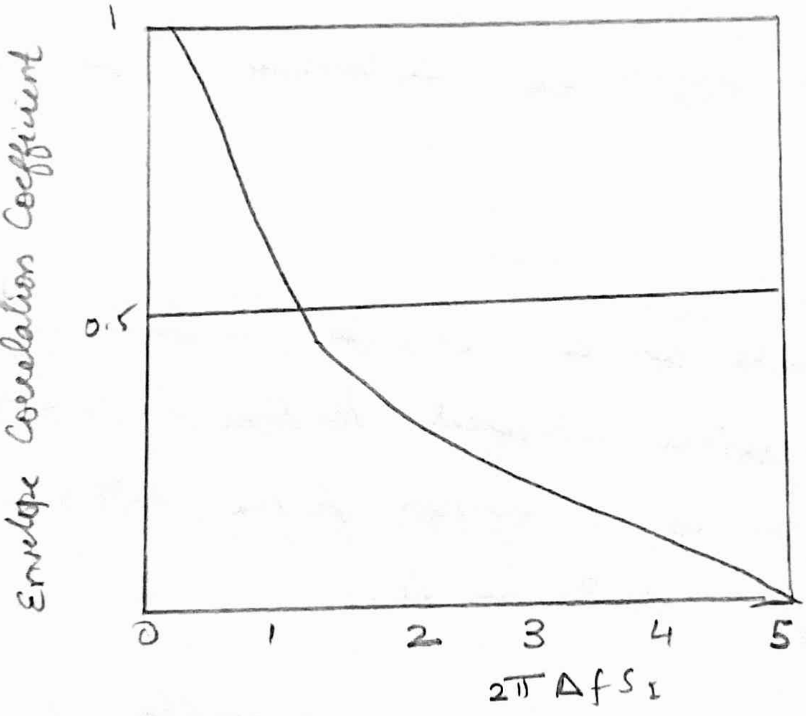


fig:- Correlation coefficient of the envelope as a function of normalized frequency spacing.

* Figure shows ρ as a fn. of the spacing b/w the two frequencies.

* Information is spread over a large bandwidth so that small parts of the information are conveyed by different frequency components.

Angle diversity or pattern diversity:-

* A fading dip is created when MPCs, which usually come from different directions, interfere destructively.

* MPCs interfere differently for the two antennas.

* This is the principle of Angle diversity.

* Angular diversity is usually used in conjunction with spatial diversity.

* Different types of antennas have different patterns.

Mutual Coupling:-

* Antenna B acts as a reflector for Antenna A, whose pattern is therefore skewed to the left. The pattern of antenna B is skewed to the right due to reflections from Antenna A.

* While dipole antennas are usually restrict to the top of the casing, patch antennas and inverted F antennas can be placed on all parts of the casing.

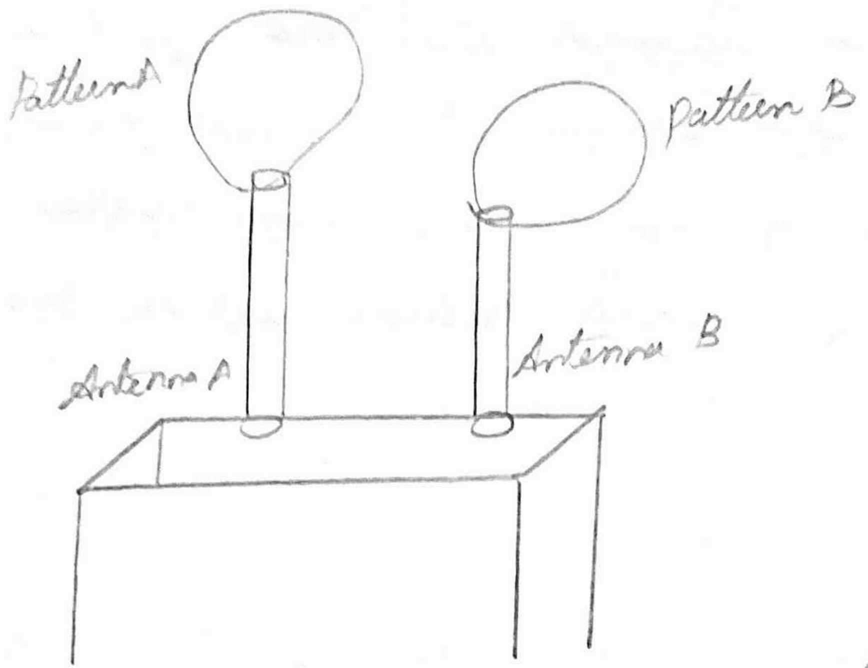


fig - Angle Diversity for closely spaced antennas

· Polarization diversity:-

* space diversity is considerably less practical than at the mobile because the narrow angle of incident fields requires large antenna spacings.

* the decorrelation for the signals in each polarization is caused by multiple reflections in the channel between the mobile and BS antennas.

* polarization diversity is used to reduce the multipath delay spread.

* polarization diversity support two simultaneous users on the same radio channel.

Theoretical model for polarization Diversity:-

* It is assumed that the signal is Txed from a mobile with vertical or horizontal polarization.

* It is received at the base station by a polarization diversity antenna with two branches.

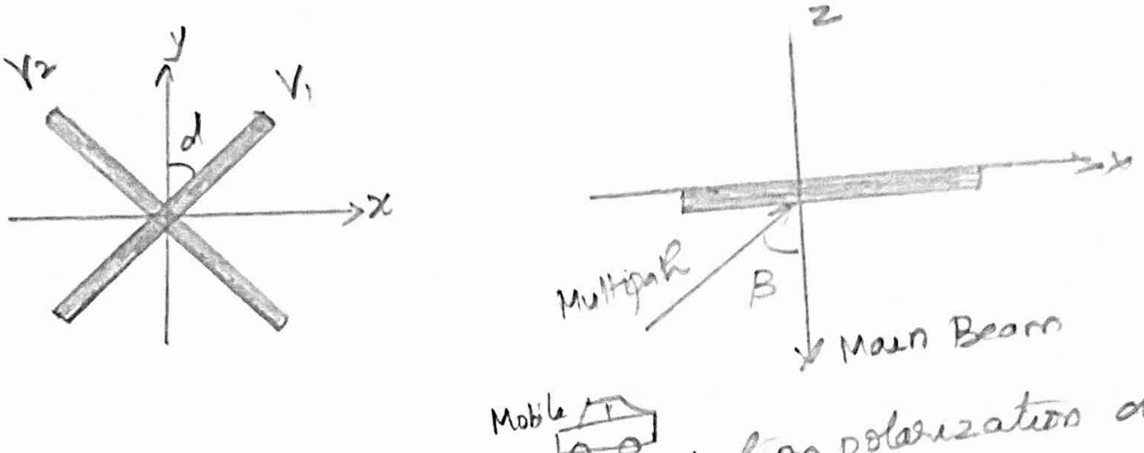


fig:- Theoretical model for base station polarization diversity based on (a) x-y plane (b) x-z plane

* A polarization diversity antenna is composed of two antenna elements V_1 & V_2.

* $\pm\alpha$ angle with y axis.

* Some of the vertically polarized signals Txed are converted to the horizontal polarized signal because of the multipath propagation.

The signal arriving at the base station.

$$x = r_1 \cos(\omega t + \phi_1) \quad \text{——} \quad (1a)$$

$$y = r_2 \cos(\omega t + \phi_2) \quad \text{——} \quad (1b)$$

* The Rxed signal values of elements V_1 & V_2 can be written as.

$$V_1 = (a\lambda_1 \cos\phi_1 + \lambda_2 b \cos\phi_2)\cos\omega t - (a\lambda_1 \sin\phi_1 + \lambda_2 b \sin\phi_2)\sin\omega t \quad \text{--- (2)}$$

$$V_2 = (-a\lambda_1 \cos\phi_1 + \lambda_2 b \cos\phi_2)\cos\omega t - (-a\lambda_1 \sin\phi_1 + \lambda_2 b \sin\phi_2)\sin\omega t \quad \text{--- (3)}$$

where $a = \sin\alpha \cos\beta$ and $b = \cos\alpha$.

The correlation coefficient ρ can be written as

$$\rho = \left(\frac{\tan^2(\alpha)\cos^2(\beta) - \Gamma}{\tan^2(\alpha)\cos^2(\beta) + \Gamma} \right)^2 \quad \text{--- (4)}$$

where $$X = \frac{\langle R_2^2 \rangle}{\langle R_1^2 \rangle} \quad \text{--- (5)}$$

and $$R_1 = \sqrt{\lambda_1^2 a^2 + \lambda_2^2 b^2 + 2\lambda_1\lambda_2 ab\cos(\phi_1 + \phi_2)} \quad \text{--- (6)}$$

$$R_2 = \sqrt{\lambda_1 a^2 + \lambda_2^2 b^2 - 2\lambda_1\lambda_2 ab(\phi_1 + \phi_2)} \quad \text{--- (7)}$$

The correlation co-efficient is determined by three factors

① polarization angle.

② offset angle from the main beam direction

③ cross polarization discrimination.

* the average value of signal loss L, relative to that received using vertical polarization is given by.

$$L = a^2/x + b^2 \quad \text{—— (8)}$$

Macro diversity and simul cast:-

* we should use a separate BS (BS2) that is placed in such a way that the hill is not in the connection line b/w the MS and BS2.

* the simplest method for macrodiversity is the use of freq. repeaters that receive the signal and retransmit an amplified version of it.

* simul cast — the same signal is txed simultaneously from different BSs.

* It is desirable that the synchronization error is no larger than the delay dispersion that the Rx can handle.

* simulcast is also widely used for broadcast applications, especially in digital TV.

Combination of signals:-

* Two ways of exploiting signals from the multiple diversity branches.

1. selection diversity – "the "best" signal copy is selected and processed, while all other copies are discarded.

2. Combining diversity – All copies of the signal are combined and the combined signal is decoded.

Diversity gain:

* Diversity gain reflects the fact that it is improbable that several antenna elements are in a fading dip simultaneously.

Beam forming gain.

Beam forming gain reflects the fact that the combiner performs an averaging over the noise at different antennas.

Received – signal – strength – indication – Driven diversity

The RX selects the signal with the largest instantaneous power.

This method requires N_r antenna elements N_r RSSI sensors and a N_r to 1 multiplexer but only one RF chain.

* Instaneous Signal amplitude is Rayleigh distributed

SNR of the nth diversity branch, γ_n

$$\text{Pdf}_{\gamma_n}(\gamma_n) = \frac{1}{\bar{\gamma}} \exp\left(-\frac{\gamma_n}{\bar{\gamma}}\right)$$

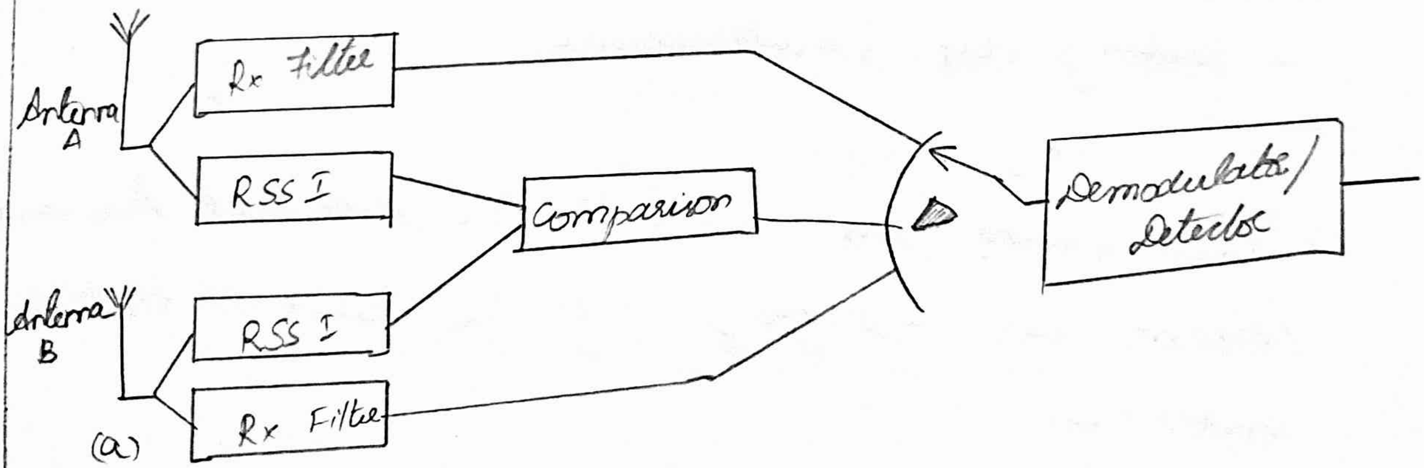
$\bar{\gamma}$ means mean SNR.

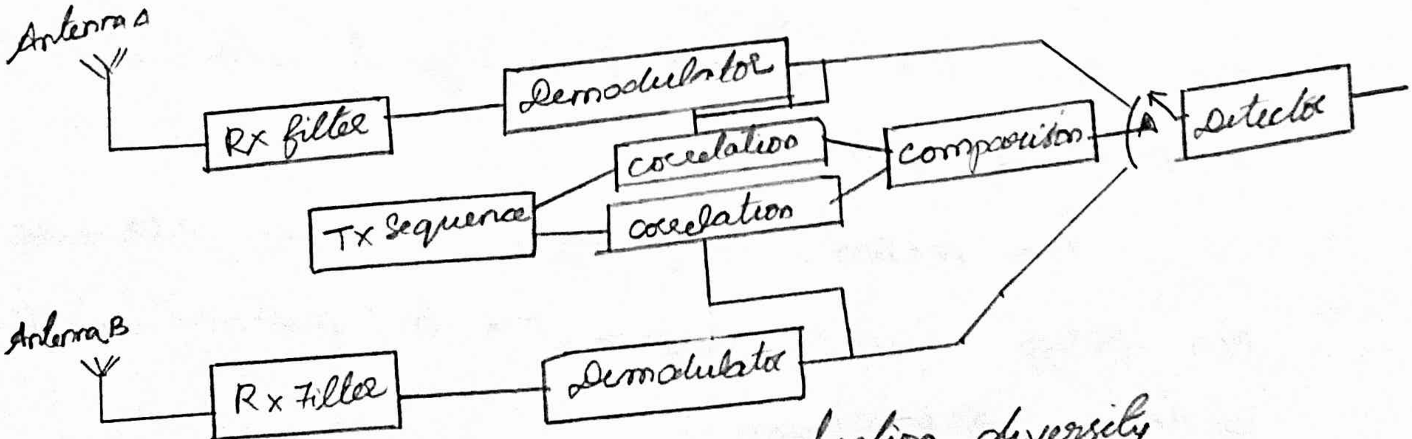
$$\text{cdf}_{\gamma_n} = 1 - \exp\left(1 - \frac{\gamma_n}{\bar{\gamma}}\right)$$

Bit error rate driven diversity:-

* For BER driven diversity, we first transmit a training sequence - bit sequence that is known at the Rx.

RSSI-driven Selection Diversity





BER-driven Selection diversity

Diversity combining Techniques:-

* Selection diversity wastes signal energy by discarding $(N_r - 1)$ copies of the Rxed signal.

* Combining diversity Exploits all available signal copies.

For amplitude weighting two methods are used.

① Maximum Ratio Combining (MRC)

② Equal gain combining (EGC).

MRC — weights all signal copies by their amplitude

EGC — All amplitude weights are the same.

Maximum ratio combining:-

* MRC compensates for the phases, and weights the signals from the different antenna branches according to their SNR.

* Let us assume a propagation channel that is slow fading & flat fading.

* Time variant filter with impulse response

$$h_n(\tau) = \alpha_n \delta(\tau) \quad — ①.$$

where $\alpha_n \rightarrow$ gain of diversity branch n.

The SNR becomes:

$$\frac{\left|\sum_{n=1}^{N} w_n^* \alpha_n\right|^2}{P_n \sum_{n=1}^{N} |w_n|^2} \quad — ②$$

* According to the Cauchy – Schwartz inequality.

$$\left|\sum_{n=1}^{N} w_n^{*} r_n\right|^2 \leq \sum_{n=1}^{N} |w_n^{*}|^2 \sum_{n=1}^{N} |r_n^{*}|^2 \quad \text{—— (3)}.$$

$$W_{MRC} = r_n \quad \text{—— (4)}.$$

* The o/p SNR of the diversity combiner is the sum of the branch SNRs.

$$\gamma_{MRC} = \sum_{n=1}^{N} \gamma_n \quad \text{—— (5)}.$$

$$pdf_{\gamma}(\gamma) = \frac{1}{(N_r-1)!} \frac{\gamma^{N_r-1}}{\bar{\gamma}^{N_r}} \exp\left(-\frac{\gamma}{\bar{\gamma}}\right) \quad \text{—— (6)}$$

* The mean SNR of the o/p just the mean branch SNR, multiplied by the number of diversity branches.

$$\bar{\gamma}_{MRC} = N_r \bar{\gamma} \quad \text{—— (7)}.$$

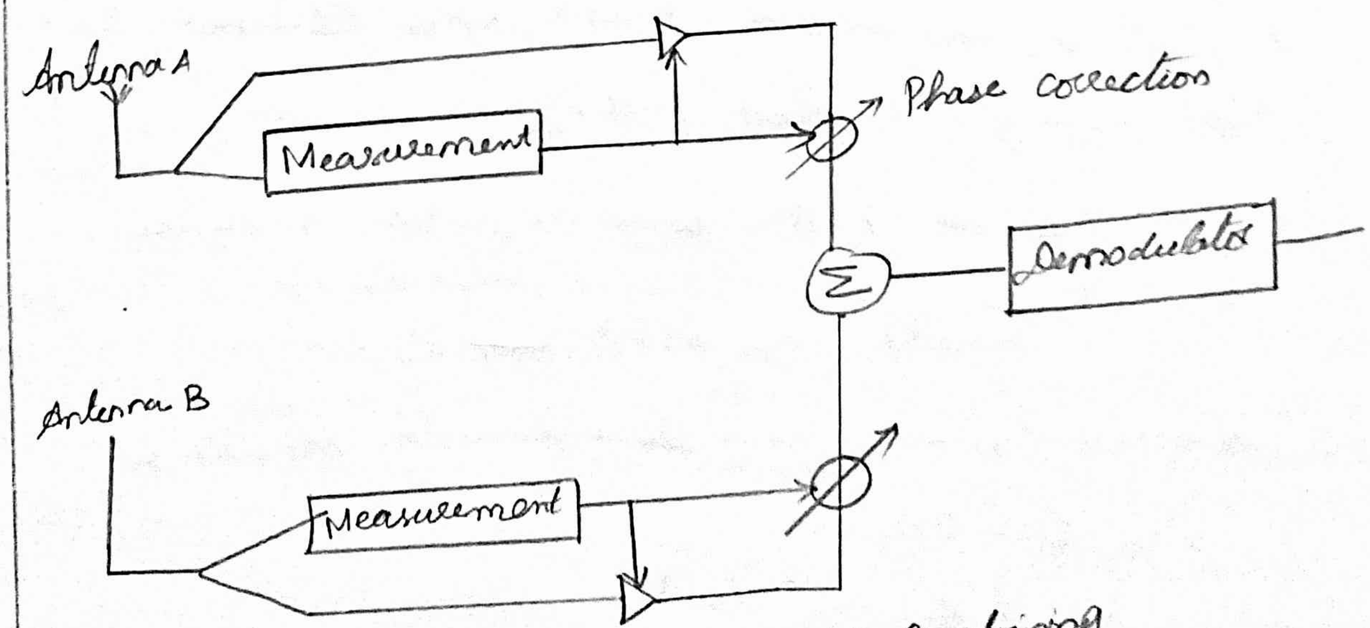


fig – Maximum Ratio Combining

Equal Gain Combining:-

For EGC, the SNR of the combiner o/p is

$$\gamma_{EGC} = \frac{\left(\sum_{n=1}^{N} \sqrt{\gamma_n}\right)^2}{N_r} \quad \text{—— (8).}$$

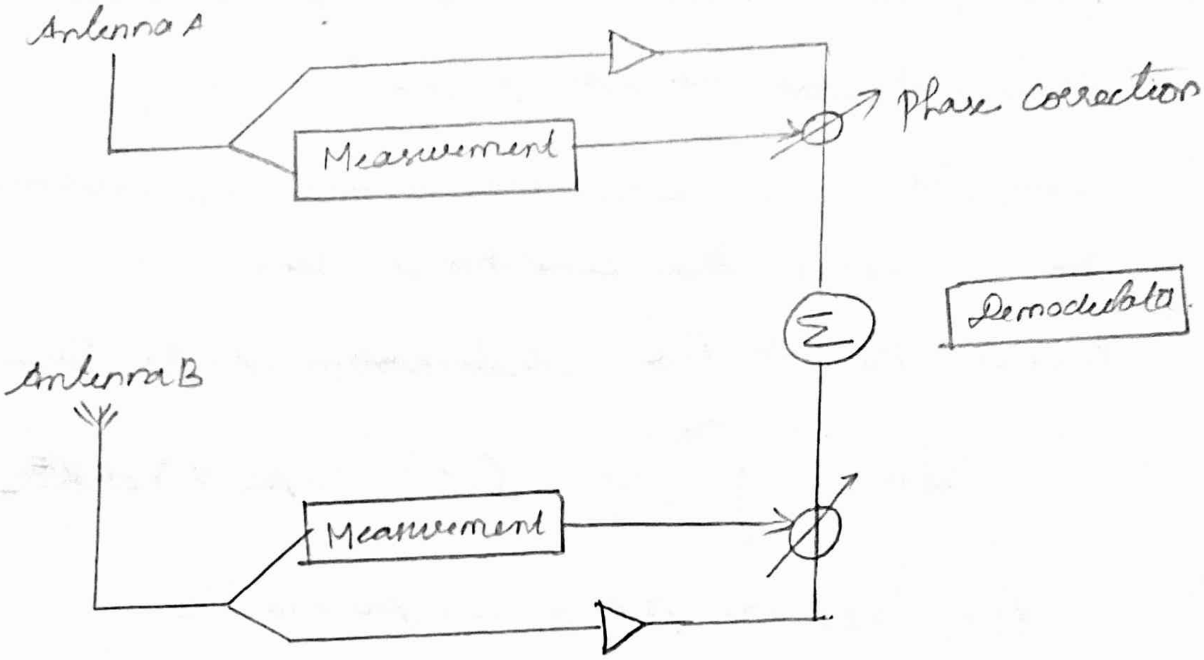


fig:- Equal gain combining

* The mean SNR of the combiner o/p can be found to be.

$$\bar{\gamma}_{EGC} = \bar{\gamma}\,(1 + (N_r - 1)\,\pi/4) \quad \text{—— (9).}$$

* The mean SNR is same in all branches.
* EGC performs worse than MRC by only a factor $\pi/4$.

Error probability in fading channels with Diversity Reception:-

* Error probability in flat fading channels.
* Symbol Error rate in freq. selective fading channels.

Error probability in flat fading channels.

Classical computation method:-

compute the error probability of diversity systems by averaging the conditional error.

probability over the distribution of the SNR.

$$\overline{SER} = \int_0^\infty pdf_\gamma(\gamma)\, SER(\gamma)\, d\gamma \quad \text{—} ①$$

The SER of BPSK is AWGN is

$$SER(\gamma) = Q(\sqrt{2\gamma}) \quad \text{—} ②$$

we obtain an eqn that can be evaluated analytically.

$$\overline{SER} = \left(\frac{1-b}{2}\right)^{N_r} \sum_{n=0}^{N_r-1} \binom{N_r-1+n}{n} \left(1+b/2\right)^n \quad \text{—} ③$$

where b is defined as.

$$b = \sqrt{\overline{\gamma}/1+\overline{\gamma}} \quad \text{—} ④$$

For large values of $\overline{\gamma}$

$$\overline{SER} = \left(\frac{1}{4\overline{\gamma}}\right)^{N_r} \binom{2N_r-1}{N_r} \quad \text{—} ⑤$$

The BER decreases with the N_rth power of the SNR.

Error rate in frequency selective fading channels:-

Determine the SER in channels that suffer from time dispersion and freq. dispersion.

* For BPSK with selection diversity

$$\overline{SER} = \frac{1}{2} - \frac{1}{2}\sum_{n=1}^{N_r}\binom{N_r}{n}(-1)^{n+1}\frac{b_0\, \mathrm{Im}\{\rho_{x,y}\}}{\left(\mathrm{Im}\{\rho_{xy}\})^2 + n(1-|\rho_{xy}|^2\right)}$$

where b_0 is the transmitted bit. — (6).

This can be approximated as,

$$\overline{SER} = \left(\frac{2N_r-1}{2}\right)!!\left(\frac{1-|\rho_{xy}|^2}{2(\mathrm{Im}\{\rho_{xy}\}^2)}\right)^{N_r} \quad \text{— (7).}$$

where $(2N_r-1)!! = 1.3.5 \ldots (2N_r-1)$

For BPSK with MRC.

$$\overline{SER} = \frac{(2N_r-1)!!}{2(N_r!)}\left(\frac{1-|\rho_{xy}|^2}{2(\mathrm{Im}\,\rho_{xy})^2}\right)^{N_r} \quad \text{— (8)}$$

BER of MSK with differential detection becomes.

$$\overline{BER} \approx (\pi/2)^4 (S_\tau/T_B)^4 \quad \text{— (9).}$$

Compared with these RSSI driven result.

$$\overline{BER} \approx 3\,(\pi/4)^4\left(\frac{S_\tau}{T_B}\right)^4 \quad \text{— (10)}$$

RAKE Receiver:-

* A RAKE Receiver attempts to collect the time shifted versions of the original signal by providing a separate correlation receiver for each of the multipath signals.
* A microprocessor controller can cause different correlation receivers to search in different time windows for significant multipath.
* The range of time delays that a particular correlator can search is called a search window.
* A RAKE receiver utilizes multiple correlators to separately detect the M strongest multipath components.
* The o/p's of each correlator are then weighted to provide a better estimate of the txed signal than is provided by a single component.

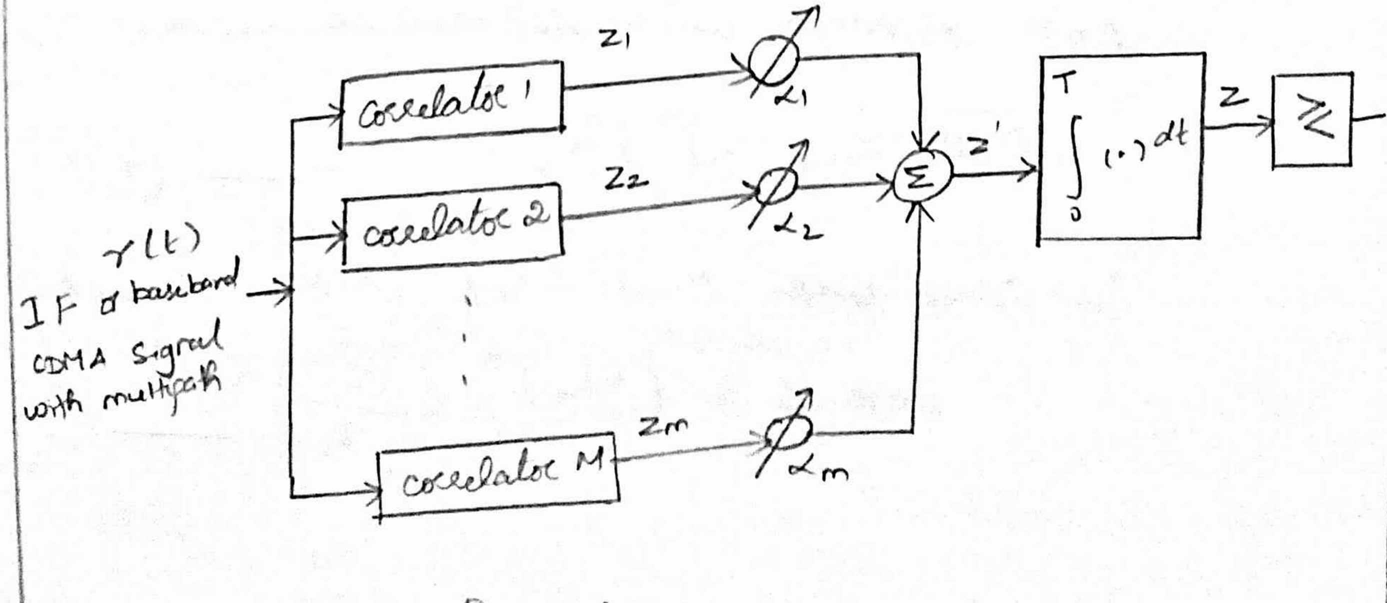


fig:- Rake Receiver

* M correlators are used in CDMA receiver. to capture the M strongest multipath components.

* correlator 1 is synchronized to the strongest multipath m_1

* second correlator is synchronized to m_2.

* Diversity which can overcome fading and hereby improve CDMA reception.

* The o/p's of the M correlators are denoted as $z_1, z_2 \ldots$ and z_m

* They are weighted by $\alpha_1, \alpha_2 \ldots$ and α_m.

The overall signal z' is given by-

$$z' = \sum_{m=1}^{M} \alpha_m z_m \quad \text{—— (1)}.$$

The weighting coefficients α_m are normalized to the o/p signal power of the correlator in such a way that the co-efficients sum to the unity.

$$\alpha_m = z_m^2 / \sum_{m=1}^{M} z_m^2 \quad \text{—— (2)}.$$

* choosing weighting coefficients based on the actual o/p's of the correlators yield better RAKE performance.

UNIT – V

MULTIPLE ANTENNA TECHNIQUES.

MIMO Systems – Spatial multiplexing – System model – pre coding – Beam forming – transmitter Diversity, Receiver Diversity – channel state Information – Capacity in fading and Non-fading channels.

MIMO Systems:-

"MIMO Systems are systems with multiple element antennas (MEA s) at both link ends".

The MEAS of a MIMO System Can be used for four different purposes

(i) Beam forming
(ii) Diversity
(iii) Interference suppression
(iv) spatial multiplexing.

Spatial multiplexing allows direct improvement of capacity by simultaneous transmission of multiple data streams.

Capacity for a single link increases linearly with the number of antenna elements.

MIMO of antenna was included in 4th generation cellular systems as well as high throughput wireless local area networks.

Spatial multiplexing:-

Spatial multiplexing uses MEAs at the Tx for transmission of parallel data streams.

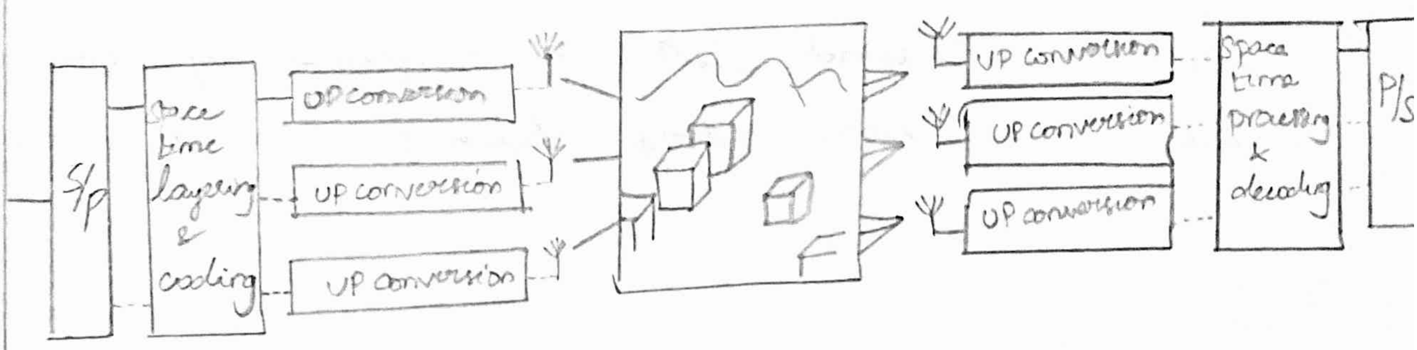


fig:- Principle behind spatial Multiplexing

An original high data rate stream is multiplexed into several parallel streams each of which is sent from one transmit antenna element.

The channel "mixes up" these data streams.

If the channel is well behaved, the Rxed signals represent linearly independent combinations.

Appropriate signal processing at the RX can seperate the data streams.

A basic condition is that the no. of Receiver antenna elements is at least as large as the no. of transmit data streams.

this approach allows the data rate to be drastically increased by a factor of min (N_t, N_r).

with N_t transmit antennas, we can form N_t different beams.

At the RX, we can use N_r antenna elements, to form N_r beams and also point them at different IOs.

If all the beams can be kept orthogonal to each other, there is no interference b/w the data streams

The IOS play the same role as wires in the transmission of multiple data streams on multiple wires.

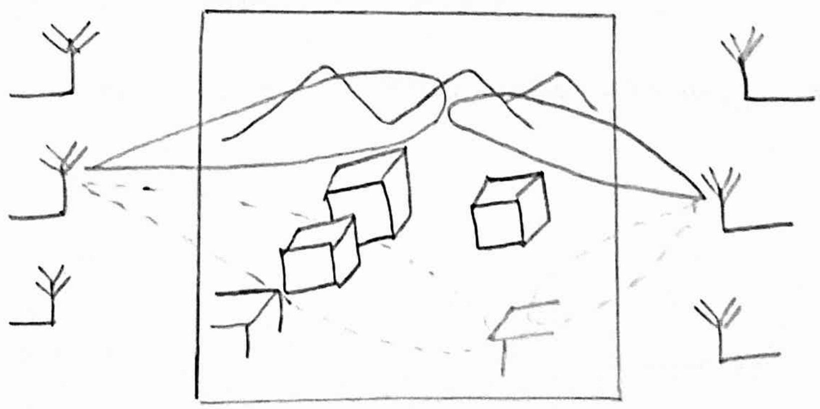


fig:- Transmission of different data streams

The number of possible data streams limited by min (N_t, N_r, N_s).

where $N_s \rightarrow$ No of IOs.

The no. of data streams cannot be larger than the no of transmit antenna elements.

If two data streams are fed to the same IO, then the Rx has no possibility of sorting them out by forming different beams.

System model:-

At the Tx, the data stream enters an encoder, whose o/p's are forwarded to Nt transmit antennas.

From the antennas the signal is sent through the wireless propagation channel, which is assumed to be quasi-static and frequency flat.

By quasi static, the coherence time of the channel is so long that "a large number" of bits can be transmitted within this time.

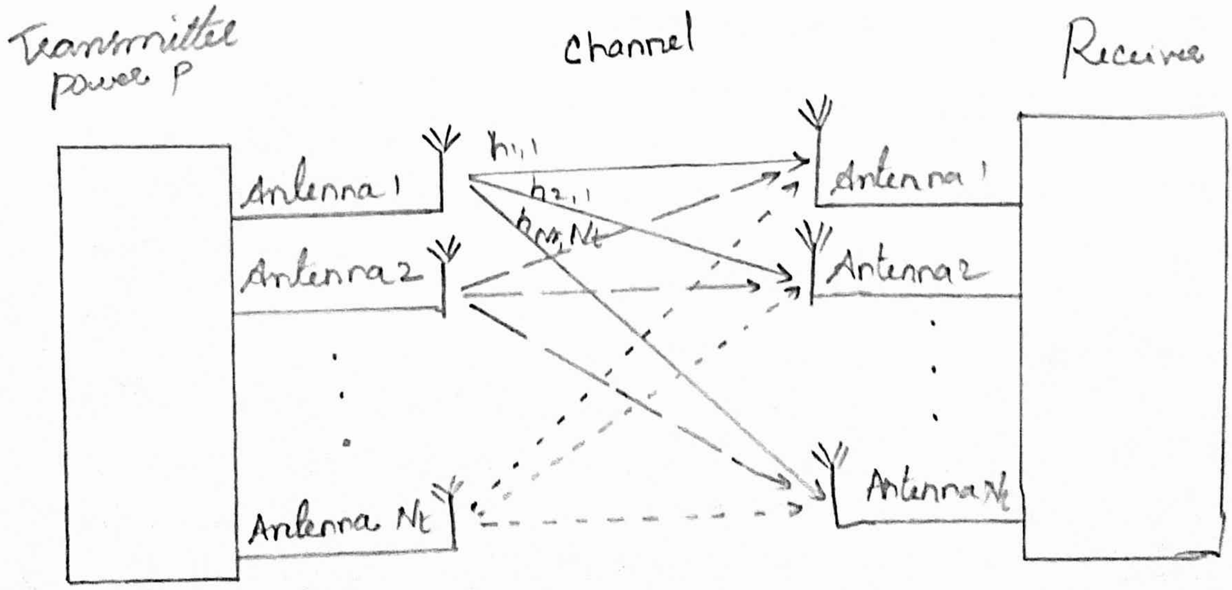


fig:- Block diagram of a multiple i/p multiple o/p system we denote the $N_r \times N_t$ matrix of the channel as

$$H = \begin{pmatrix} h_{11} & h_{12} & \cdots & h_{1N_t} \\ h_{21} & h_{22} & \cdots & h_{2N_t} \\ \vdots & \vdots & & \vdots \\ h_{N_r1} & h_{N_r2} & \cdots & h_{N_rN_t} \end{pmatrix} \quad \text{—— (1)}$$

whose entries h_{ij} are complex channel gains from the jth transmit to the ith receive antenna

The received signal vector,

$$r = Hs + n = x + n \quad \text{—— (2)}$$

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

Pre coding:-

The main difficulty in MIMO channels is the seperation of the data streams which are sent in parallel.

To decrease the multiuser interference and increase the data rate in MIMO system we are using the precoding technique.

The precoding is a technique which exploits transmit diversity by weighting information stream.

The txer sends the coded information to the Rxer in order to get the rich knowledge of the channel.

The Receiver is a simple detector such as matched filter and does not have to know the channel state information.

Precoding or pre-equalization of the txer signals for MIMO Systems is the type of processing at the txer required the channel State information (CSI).

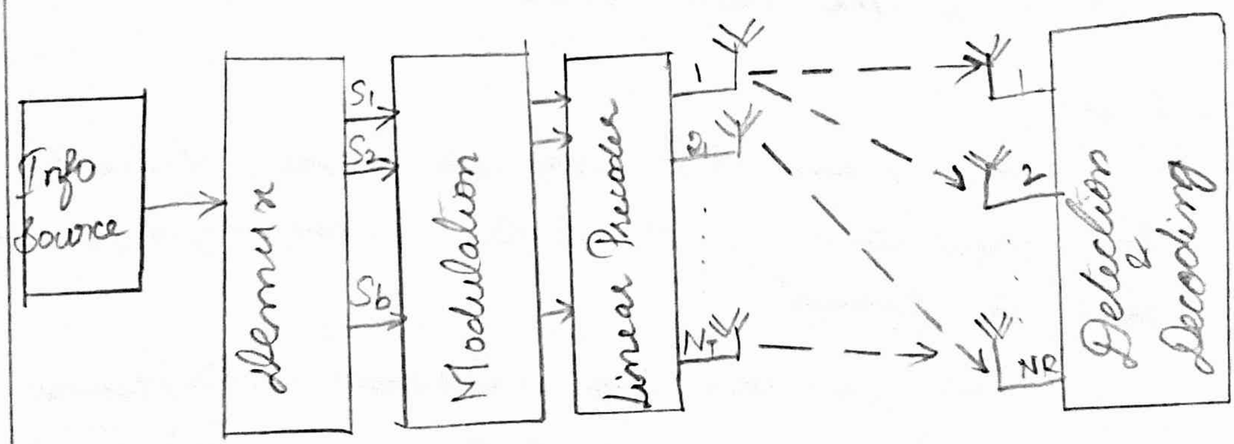


fig:- Block diagram of MIMO precoding

In order to obtain CSI at the txer, the channel should be fixed or, approximate constant over a reasonably large time period.

If CSI is available at the transmitter and then the txed symbols either for a single-user or for multiple users can be partially separated by means of the equalization at the txer.

Vertical Bell Labs Layered Space-Time Architecture (V-BLAST):-

V-BLAST is a txer Rxer architecture which is mainly used to implement multiplexing in MIMO

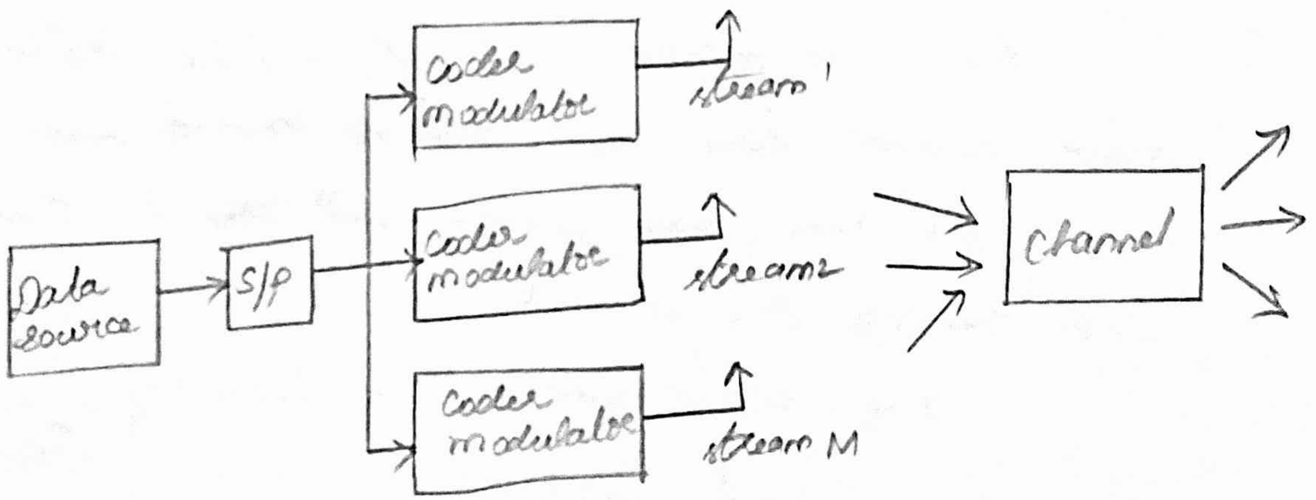


fig:- BLAST structure

This architecture can then achieve almost a diversity order of N_R, since each coded symbol is txed from one antenna and Rxed by N_R antennas.

By coding across the sub channels, BLAST can average over the randomness of the individual sub channels and better outage performance.

Working principle:-

For example, you are sending information 's' and it will pass through the channel 'h' and Gaussian Noise 'n' then the Rxed signal at the Rxer front end will be,

$$r = sh + n.$$

The Rxer will have to know the information about 'h' and 'n'. It will suppress the effect of 'n' by increasing SNR.

The Rxer mobile units have to be simple for many reasons like cost, size of mobile unit.

The txer, base station will do the hardwork and predicts the channel.

The information will be coded : $\left(\frac{s}{h_{est}}\right)$

The Rxed signal will be,

$$r = \left(\frac{h}{h_{est}}\right) s + n$$

If your prediction is perfect,

$h_{est} = h$ & $r = s + n$ and it turns out to be the detection problem in Gaussian channels which is simple.

Types of Pre coding:-

Pre Coding can be realized without requiring channel state CSI at the transmitter, while such information is essential to handle the interuser interference in multi user systems.

This is known as Space Division multiple Access (SDMA)

From an implementation perspective, precoding algorithms for SDMA systems can be subdivided into.

(i) Linear Precoding.
(ii) Non Linear Precoding.

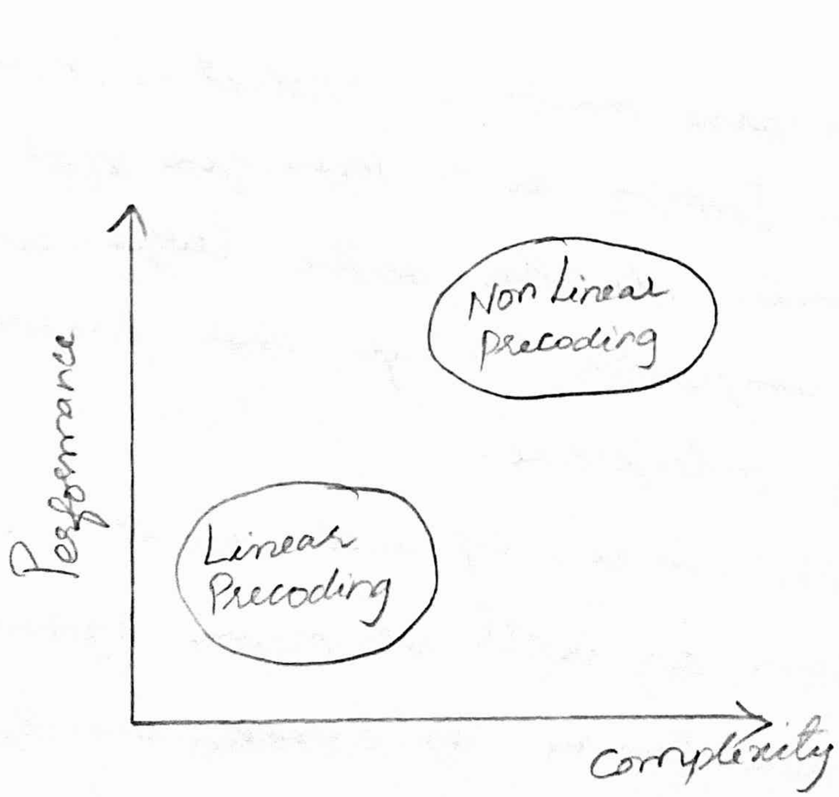


fig:- Linear Vs Non Linear Precoding

Linear precoding strategies include maximum Ratio Transmission (MRT), zero forcing precoding and Transmit wiener precoding.

The non linear precoding is designed based on the concept of Dirty paper coding (DPC) which shows that ~~for~~ any known interference at the txer can usually be subtracted without the penalty of Radio resources if the optimal precoding scheme can be applied on the transmit signal.

Beam forming:-

It is also known as spatial filtering.

Beam forming is a technique that focuses radio signals directly on the target antenna. there by improving range and performance by limiting interference.

It can be applied in all antenna array systems as well as MIMO systems.

Beam forming is exactly analogous to frequency domain analysis of time signals.

Since the decoding complexity is exponential in λ, we can keep the complexity low by keeping λ small i.e $\lambda=1$.

A transmit strategy where the i/p covariance matrix has unit rank is called Beam forming.

This corresponds to the precoding matrix being just a column vector $M=c$, the beam forming vector is shown in figure.

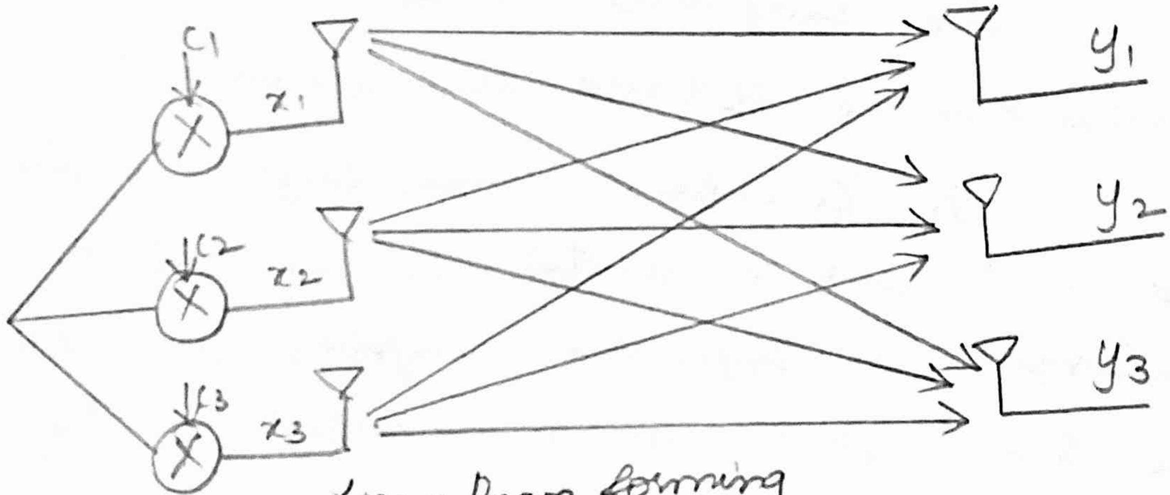


fig:- Beam forming

Spatial matched filtering yields a single SISO AWGN channel as follows

$$\bar{y} = \frac{\bar{c}H}{\|\bar{c}H\|} y$$

$$= \frac{\bar{c}H}{\|\bar{c}H\|} H\bar{c}x + \frac{\bar{c}H}{\|\bar{c}H\|} \bar{w}$$

where $\bar{w}$ is zero mean, unit variance AWGN

SNR becomes

$$SNR = \bar{C} \alpha H \alpha \bar{C} \; E[x x^{*}]$$

Types of beam forming:

Beam forming is divided into two groups

Phased Array systems with a finite number of fixed pre defined pattern.

Adaptive array systems with an infinite number of patterns adjusted to the scenario in real time

The complexity and the cost of such higher than the switched beam formers.

In conventional single layer beam forming the same signal is emitted from each of the transmit antennas with appropriate weighting such that the signal power is maximized at the receiver signal. output.

when the Rxer has multiple antennas single layer beamforming cannot simultaneously maximized at the signal level at all of the receive antennas.

Thus inorder to maximize the throughput multiple receive & antenna systems, multilayer beam forming is required.

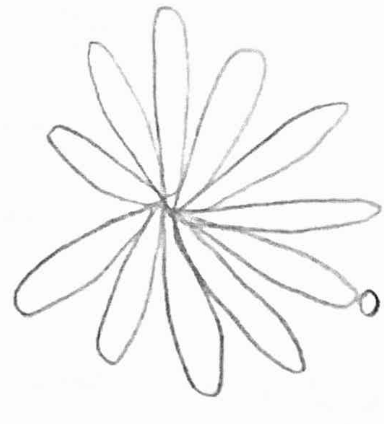


fig:- Switched Beamformer

fig:- Adaptive Beam former

Advantages of beam forming:-

It increase the Rxed signal gain by making signals emitted from different antennas add up constructively and to reduce the multipath fading effect.

Transmitter diversity:-

"Signals are transmitted from several Tx antennas i.e, diversity effect is achieved by multiple antennas are known as transmit diversity.

For the uplink txion from the ms to BS multiple antennas can act as receive diversity branches.

For the down link, any possible diversity originates at the txer.

* Transmitter Diversity with channel state Information
* Transmitter Diversity without channel state Information.

Transmitter Diversity with channel State Information:-

The Tx knows the channel perfectly there is a complete equivalence b/w transmit diversity and receive diversity.

The optimum transmission scheme linearly weights signals txed from different antenna elements with the complex conjugates of the channel transfer functions from the transmit antenna elements to the single receive antenna. This approach is known as "maximum Ratio Transmission".

Transmitter Diversity with channel state information:

CSI is not available at the Tx

The addition of different components to be constructive or destructive.

Adding up MPCs with random phases, which results in Rayleigh fading.

Transmission of the signals from different antenna elements has to be done is such a way that it allows the Rx to distinguish different txed signal components.

One way is delay diversity.

Signals txed from different antenna elements are delayed copies of the same signal.

Effective impulse response is delay dispersive even if the channel itself is flat fading.

In a flat fading channel, we transmit data streams with a delay of, symbol duration from each of the transmit antennas.

The effective impulse response of the channel then becomes

$$h(\tau) = \frac{1}{\sqrt{N_t}} \sum_{n=1}^{N_t} h_n \delta(\tau - nT_s).$$

where, h_i → gains from the ith transmit antenna to the receive antenna.

The signals from different transmit antennas to the Rx act effectively as delayed MPCs.

The delay b/w signals txed from different antenna elements should be at least as large as the maximum excess delay of the channel.

If there are only two antenna elements, the same signal is txed from both antenna elements.

One of the antenna signals undergoes a time varying phase shift.

Even if the Tx, Rx, and the IO, are stationary - the signal does not remain stuck in a fading dip.

Another possibility for achieving transmit diversity is space time coding.

Receiver diversity:-

One transmitting antenna and many receiving antenna are used.

The desired message is transmitted by using single transmitting antenna & received by multiple antennas.

NR different antennas appropriately separated are deployed at the receiver combine the uncorrelated fading signals.

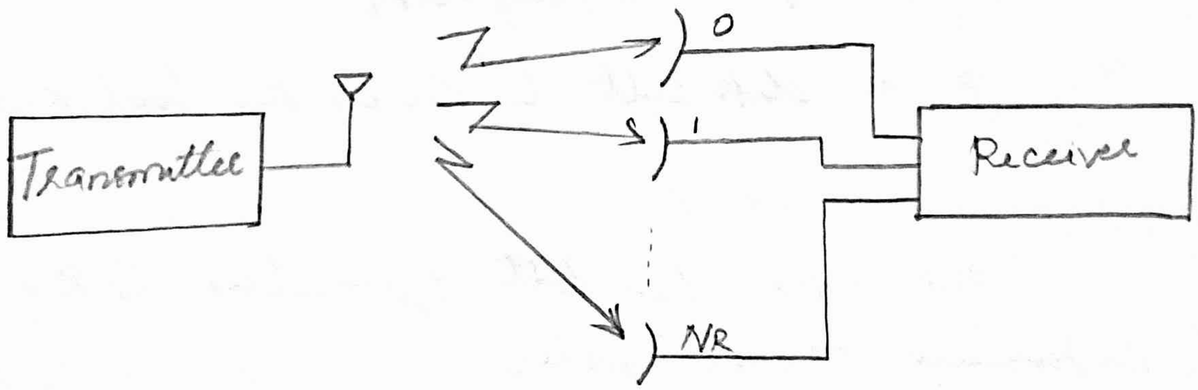


fig:- Receiver Diversity

[For receiver diversity refer unit 4, topics are, selection diversity, diversity combining techniques - maximal ratio combining, Equal gain combining].

Channel state information:-

Algorithms for MIMO transmission can be categorized by the amount of CSI that, they require.

Distinguish the following cases:

(1) Full CSI at the TX (CSIT) and full CSI at the RX (CSIR)

Both the Tx and the Rx have full and perfect knowledge of the channel.

Highest possible capacity.

It is difficult to obtain the full CSIT.

(2) Average CSIT and full CSIR:-

The Rx has full information of the instantaneous channel state,

but the TX knows only the average CSI - eg: the correlation matrix of H or the angular power spectrum.

Doesnot require reciprocity or fast f/B.

It does require calibration or slow feedback.

③ No CSIT and full CSIR:-

This is the case that can be achieved most easily, without any f/b or calibration.

The TX simply does not use any CSI, while the RX learns the instantaneous channel state from a training sequence or using blind estimation.

④ Noisy CSI :

Any received training sequence will be affected by additive noise as well as quantization noise.

Assume a "mismatched RX" where the RX process the signal based on the observed channel Hobs. while in reality the signals pass through channel Htrue

$$H_{true} = H_{obs} + \Delta.$$

⑤ No CSIT and No CSIR:-

channel capacity is also high when neither the TX nor the RX have CSI.

For high SNR, capacity no longer increases linearly with $m = \min(N_t, N_r)$ but increases as $\tilde{m}(1 - \tilde{m}/T_{coh})$, where $\tilde{m} = \min[N_t, N_r[T_{coh}/2]]$.

Capacity in Fading and non-Fading channels:-

Capacity in non-fading channels:-

The first key step in understanding MIMO Systems is the derivation of the capacity eqn for MIMO Systems in non-fading channels often known as "Foschini's equation".

The information-theoretic capacity of such a channel is,

$$C_{shannon} = \log_2 (1 + \gamma \cdot |H|^2) \quad \text{—— ①}$$

where $\gamma \rightarrow$ SNR at the Rx

$H \rightarrow$ Normalized transfer fn from the Tx to the Rx

Capacity increases only Logarithmically with the SNR.

Let us consider a singular value decomposition of the channel

$$H = W \Sigma U^* \quad \text{—— ②}$$

where $\Sigma \rightarrow$ diagonal matrix containing singular values.

W & U^* $\rightarrow$ unitary matrices composed of the left and right vectors.

The received signal is then.

$$r = Hs + n \quad \text{—— ③}$$

$$= W \Sigma U^* s + n \quad \text{—— ④}$$

Then, multiplication of the transmit data by matrix U, and the received signal vector by W^H diagonalizes the channel.

$$W^H r = W^H W \Sigma U^H U \tilde{s} + W^H n$$

$$\tilde{r} = \Sigma \tilde{s} + \tilde{n} \quad \text{—} \quad (5)$$

where U & W are unitary matrices.

The capacity of channel H is thus given by the sum of the capacities of the eigen modes of the channel.

$$C = \sum_{k=1}^{R_H} \log_2 \left[1 + \frac{P_k}{\sigma_n^2} \sigma_k^2 \right] \quad \text{—} \quad (6).$$

where $\sigma_n^2 \rightarrow$ noise variance.

$P_k \rightarrow$ power allocated to the kth eigen mode.

The capacity expression can be shown to be equivalent to

$$C = \log_2 \left[\det \left(I_{N_R} + \frac{\bar{\gamma}}{N_t} H R_{ss} H^H \right) \right] \quad \text{—} \quad (7).$$

where,

I_{N_R} is the $N_R \times N_R$ identity matrix.

$\bar{\gamma}$ – mean SNR per RX branch.

R_{ss} – correlation matrix of the transmit data.

No channel state information at the transmitter and Full CSI at the receiver:-

Capacity thus takes on the form:

$$C = \log_2 \left[\det \left(I_{N_r} + \frac{\bar{P}}{N_t} H H^* \right) \right] \text{ --- } (8)$$

The capacity of a MIMO system increases linearly with min (N_t, N_r).

Assume $N_t = N_r = N$.

(1) All transfer fn's are identical-ie;

$h_{1,1} = h_{1,2} = \cdots = h_{N,N}$ capacity is

$$C_{MIMO} = \log_2 (1 + N \bar{\gamma}) \text{ --- } (9)$$

(2) All transfer fn's are different such that the channel matrix is full rank and has N eigen values of equal magnitude capacity is,

$$C_{MIMO} = N \log_2 (1 + \bar{\gamma}) \text{ --- } (10)$$

(3) Parallel transmission channels - eg: ||el cables, Total capacity is

$$C_{MIMO} = N \log_2 \left(1 + \frac{\bar{\gamma}}{N} \right) \text{ --- } (11)$$

Full channel state information at the txer and full CSI at the Rxer:-

Both the Rx and Tx know the channel perfectly.

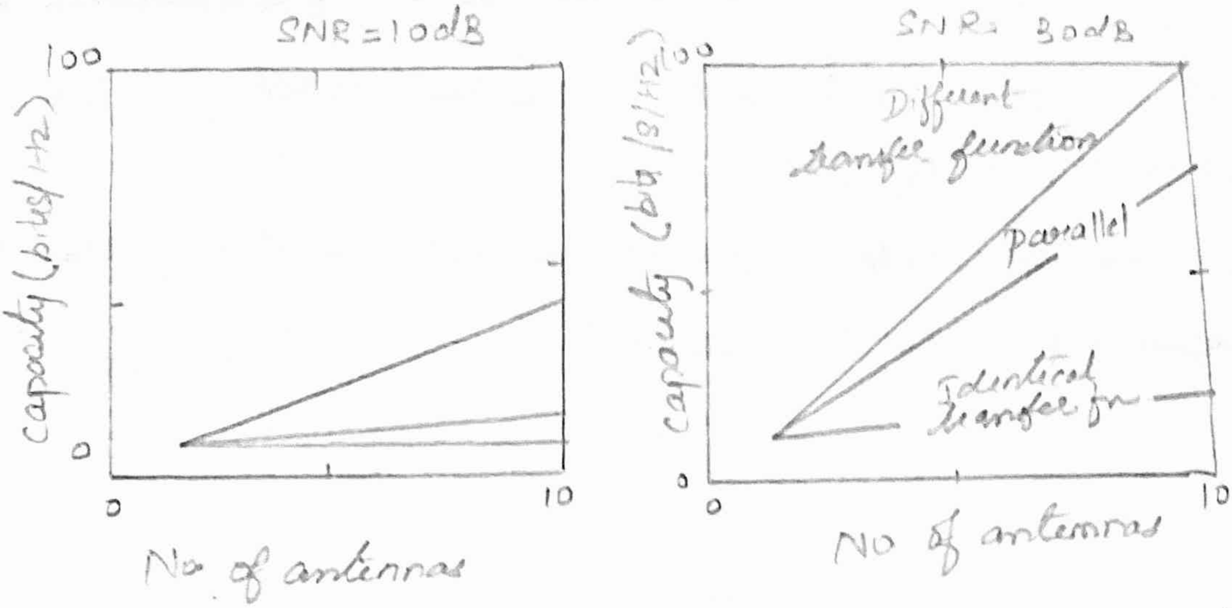


fig:- Capacity of multiple Input multiple Output

water filling — Eg of how the same mathematics can be applied to different communications problems.

Capacity in Flat fading channels::

General Concepts::

If the channel is Rayleigh fading and fading is independent at different antenna elements.

h_{ij} are zero mean circularly symmetric complex Gaussian random variables with unit variance.

The power carried by each h_{ij} is chi-square distributed with 2 degrees of freedom.

Ergodic (Shannon capacity) this is the expected value of the capacity, taken over all realization of the channel.

outage capacity :- this is the min. transmission rate that is achieved over a certain fraction of the time eg:, 90% or 95%.

No channel state information at the transmitter and perfect CSI at the receiver.

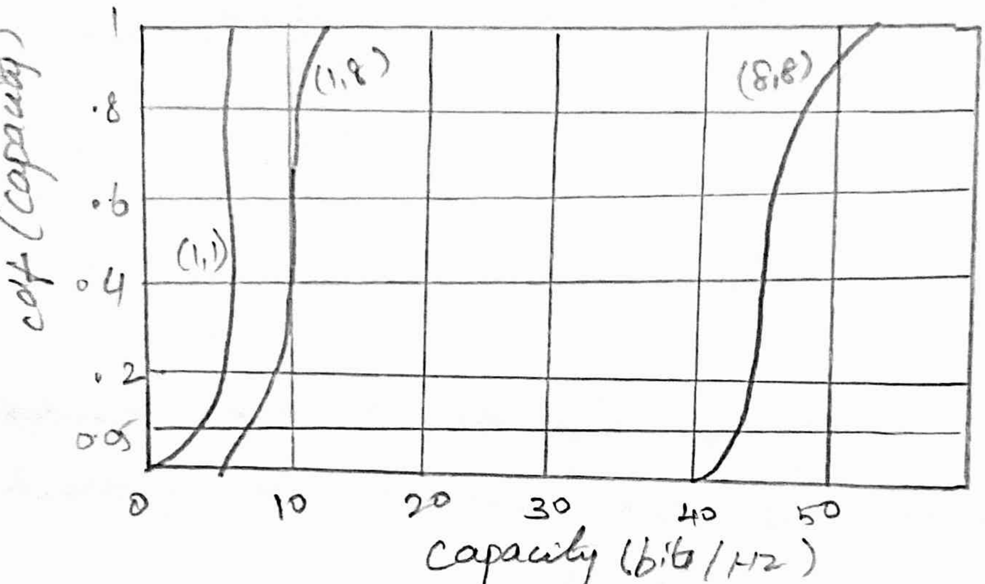


fig:- Cumulative Distribution function of capacity

. The (1,1) curve describes a single I/O system (SISO) median capacity is on the order of 6 bits/s/Hz, but the 5% outage capacity is considerably lower.

when using a (1,8) system - i.e 1 transmit antenna and 8 receive antennas.

mean capacity - from 6 to 10 bits/s/Hz.

outage capacity - 5%.

(8,8) system - i.e 8 transmit antenna & 8 receive antenna - the mean capacity is 46 bits/s/Hz.

outage mobility - 5%.

The expression for Ergodic capacity was,

$$E\{C\} = \int_0^{\infty} \log_2\left[1+\frac{\bar{\gamma}}{N_t}\lambda\right] \sum_{k=0}^{m-1} \frac{k!}{(k+n-m)!} \left[L_k^{n-m}(\lambda)\right]^2 \lambda^{n-m} \exp(-\lambda)\, d\lambda \quad \text{--- (12)}$$

where, $m = \min(N_t, N_r)$

$n = \max(N_t, N_r)$

$L_k^{n-m}(\lambda)$ – Laguerre polynomials.

Two approximations :-

(1) Capacity can be well approximated by a Gaussian distribution, such that only the mean.

(2) The following upper and lower bounds for capacity distribution for $N_t \geq N_r$.

$$\sum_{k=N_t-N_r+1}^{N_t} \log_2\left[1+\frac{\bar{\gamma}}{N_t}\chi^2_{2k}\right] < C < \sum_{k=1}^{N_t} \log_2\left[1+\frac{\bar{\gamma}}{N_t}\chi^2_{2N}\right] \quad \text{--- (13)}$$

The Lower bound corresponds to capacity that can be achieved with a Bell Labs Layered Space time (BLAST) system.

Upper bound corresponds to an idealized situation.

Lower bound is fairly tight.