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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

HANDOUT

EC 8652-WIRELESS COMMUNICATION

UNIT-1

WIRELESS CHANNELS

Effects of fading.

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

Coherence bandwidth.

The coherence bandwidth is related to the specific multipath structure of the channel. The coherence bandwidth is a measure of the maximum frequency difference for which signals are still strongly correlated in amplitude. This bandwidth is inversely proportional to the rms value of time delay spread.

Multipath Propagation

The signal can get from the TX to the RX via a no. of different propagation paths. The signal gets reflected and diffracted by different objects. So each of the paths has a distinct amplitude, delay and direction of arrival. This effect is known as multipath propagation.

Factors influencing small scale fading.

1) Multipath propagation

2) Speed of the Mobile

3) Speed of surrounding objects

4) Transmission bandwidth of the signal

Far field distance for an antenna with maximum dimension of 2m and operating frequency 1 GHz

$$D_f = 2D^2/\lambda$$

$$= 2 * 2 * 2/0.3$$

$$D_f = 26.7 \text{ m}$$

Two-Ray Ground Reflection Model

- This model is based on geometric optics.
- It considers both direct path and ground reflected path between Transmitter and Receiver.
- It is accurate model for predicting the large scale signal strength over distance of several kilometres.
- Earth is assumed to be flat.

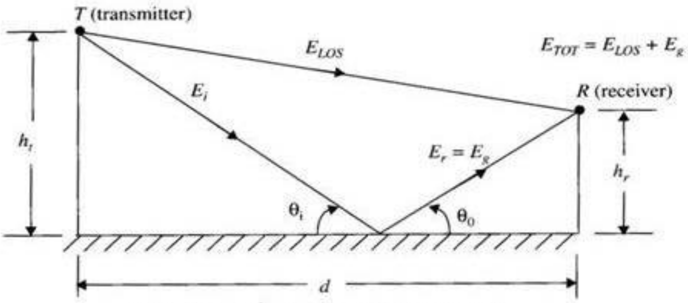


Figure: Two-ray ground reflection model.

where,

$E_0 \rightarrow$ Free space E-Field at d_0

$d_0 \rightarrow$ Reference distance

$d \rightarrow$ T-R separation distance

$c \rightarrow$ speed of light

$$|E(d,t)| = \frac{E_0 d_0}{d} \rightarrow \text{Envelope of E-Field}$$

E_{TOT} = Direct LoS Component + Ground reflected component

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

* For $d > d_0$, Free space propagation E-Field is,

$$E(d,t) = \frac{E_0 d_0}{d} \cos\left[\omega_c\left(t - \frac{d}{c}\right)\right]$$

* E-Field due to LoS component is,

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

* E-Field due to ground reflected component is,

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

* Two propagating waves are arrive at receiver as,

i) Direct wave that travels a distance d'

ii) Reflected wave that travels a distance d''

* According to the laws of reflection in dielectric,

$$\theta_i = \theta_o$$
$$E_g = \Gamma E_i$$
$$E_t = E_i + E_g = E_i + \Gamma E_i = (1+\Gamma)E_i$$

where,

$\Gamma \rightarrow$ Reflection coefficient for Ground $= -1$

Path difference (Using method of images)

$$\Delta = \text{Ground reflected path} - \text{LOS path}$$
$$= d'' - d'$$
$$\Delta = \sqrt{(h_t + h_r)^2 + d^2} - \sqrt{(h_t - h_r)^2 + d^2}$$

$$\boxed{\Delta = \frac{2 h_t h_r}{d}}$$

Phase difference

$$\theta_\Delta = \frac{2\pi\Delta}{\lambda} = \frac{2\pi f_c \Delta}{c} = \frac{\omega_c \Delta}{c} \rightarrow (6)$$

$$\left[\because \omega_c = 2\pi f_c, \quad \lambda = \frac{c}{f_c}\right]$$

Time delay

$$\tau_d = \frac{\Delta}{c}$$

From (6) $\Rightarrow$

$$\theta_\Delta = \frac{\omega_c \Delta}{c} \Rightarrow \Delta = \frac{\theta_\Delta c}{\omega_c}$$

$$\therefore \tau_d = \frac{\theta_\Delta c}{\omega_c \cdot c} = \frac{\theta_\Delta}{\omega_c}$$

If distance 'd' is large, the difference between d' and d'' becomes very small.

i.e. $d' \simeq d''$

. Amplitudes of E_{LOS} & E_g are identical

$$\left|\frac{E_0 d_0}{d}\right| = \left|\frac{E_0 d_0}{d'}\right| = \left|\frac{E_0 d_0}{d''}\right|$$

The electric Field at receiver at distance d' is,

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

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

If $d = d' = d''$, then

$$E_{TOT}\left(d, t = \frac{d''}{c}\right) = \frac{E_0 d_0}{d}\left[\cos\theta_\Delta - 1\right]$$

$$= \frac{E_0 d_0}{d}\sqrt{2\left(2\sin^2\frac{\theta_\Delta}{2}\right)}$$

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

If the distance d' increases,

$$\sin\frac{\theta_\Delta}{2} \simeq \frac{\theta_\Delta}{2}$$

From (6)

$$\theta_\Delta = \frac{2\pi\Delta}{\lambda}$$

Sub (5) =>

$$\theta_\Delta = \frac{2\pi\left(\frac{2h_t h_r}{d}\right)}{\lambda}$$

$$\frac{\theta_\Delta}{2} = \frac{\cancel{2}\pi(2h_t h_r)}{\cancel{2}\cdot\lambda d}$$

$$\boxed{\frac{\theta_\Delta}{2} = \frac{2\pi h_t h_r}{\lambda d}} \longrightarrow (8)$$

sub (8) in (7) =>

$$|E_{TOT}(d)| = \frac{2E_0 d_0}{d} \cdot \frac{2\pi h_t h_r}{\lambda d}$$

$$|E_{TOT}(d)| = \frac{k}{d^2} \text{ V/m}$$

where,

$$k = \frac{2E_0 d_0 \cdot 2\pi h_t h_r}{\lambda}$$ => constant related to E_0, Antenna heights & wavelength

* The received power at a distance 'd' from the transmitter for the two-ray ground reflection model is,

$$\boxed{P_r = P_t G_t G_r \frac{h_t^2 h_r^2}{d^4}}$$

Link budget design using path loss models. .

- It is the clearest and most sensitive way of computing the required transmitting or transmitter power
- Gives only an approximation for the total SNR

Consists of two models

i) Log distance path loss model

ii) Log-normal shadowing model

<u>Log-distance Path Loss Model</u>

- Average received power decreases logarithmically with distance
- According to this model the received power at distance d is given by,

$$P_L(d)(d/d_0)^n \Longrightarrow P_L(dB) = P_L(d_0)+10n\log(d/d_0)$$

d_0= close in reference distance, often determined emperically

d = transmitter - receiver separation

n = path loss exponent - indicates rate of path loss increase with d_0

- The value of n varies with propagation environments. The value of n is 2 for free space. The value of n varies from 4 to 6 for obstruction of building, and 3 to 5 for urban scenarios.
- The important factor is to select the correct reference distance d0. For large cell area it is 1 Km, while for micro-cell system it varies from 10m-1m.

Limitations:

- Surrounding environmental clutter may be different for two locations having the same transmitter to receiver separation.
- Moreover it does not account for the shadowing effects.

Log Normal Shadowing

- surrounding clutter isn't considered by log distance model
- averaged received power is inconsistent with measured data

The equation for the log normal shadowing is given by,

$$PL(dB) = PL(dB) + X\sigma = PL(d0) + 10n\log(d/d_0) + X_\sigma$$

Where X_σ is a zero mean Gaussian distributed random variable in dB with standard deviation σ also in dB. In practice n and σ values are computed from measured data.

Average received power:

The 'Q' function is given by,

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

And $Q(z) = 1 - Q(-z)$

So the probability that the received signal level (in dB) will exceed a certain value γ is

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

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

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

Types of small scale fading

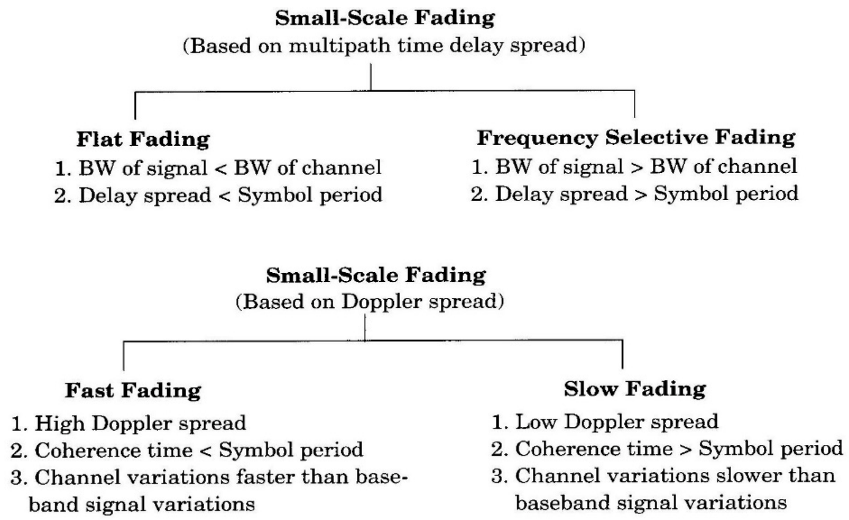


Figure 5.11 Types of small-scale fading.

1) Fading due to Multipath Delay

A） Flat Fading → $B_s << B_c$ or $T_s >>$

$$T_s \geq 10\sigma_\tau$$

- signal fits easily within the bandwidth of the channel
- channel BW >> signal BW

• spectral properties of Tx signal are preserved signal is called a *narrowband* channel, since the bandwidth of the signal is narrow with respect to the channel bandwidth signal is not distorted

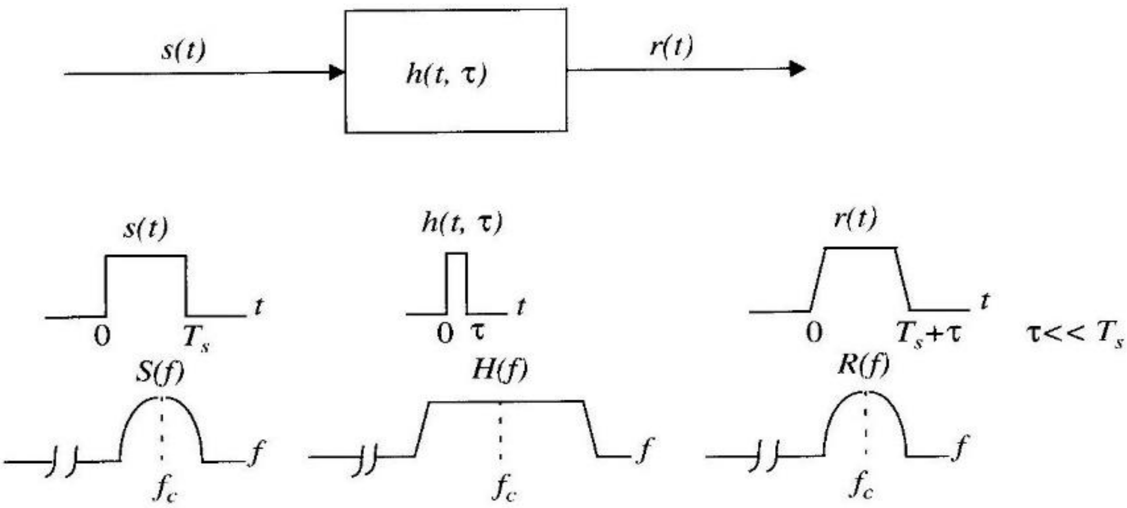


Figure Flat fading channel characteristics.

B) Frequency Selective Fading → $B_s > B_c$

_ $T_s \leq 10\sigma_\tau$

– $B_s > B_c \rightarrow$ certain frequency components of the signal are attenuated much more than others

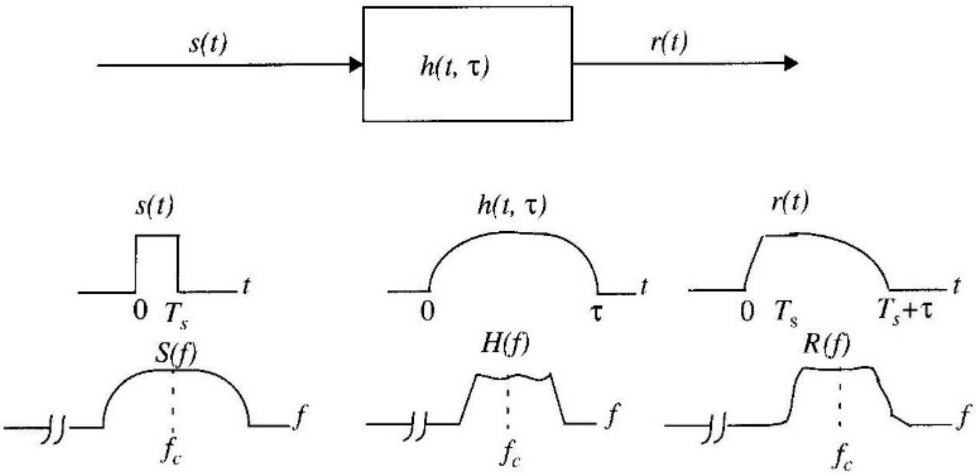


Figure Frequency selective fading channel characteristics.

- $Ts < \sigma_\tau \rightarrow$ delayed versions of Tx signal arrive during different symbol periods
  - e.g. receiving an LOS → "1" & multipath "0" (from prior symbol!)
  - This results in intersymbol interference (ISI)
  - Undesirable
  - it is very difficult to predict mobile Rx performance with frequency selective channels

2) <u>Fading due to Doppler Spread</u>

– Caused by motion of Tx and Rx and reflection sources.

A) Fast Fading → $B_s < B_D$ or $T_s > T_c$

– $B_s < B_D$

- Doppler shifts significantly alter spectral BW of TX signal
- signal "spreading"

– $Ts > Tc$

- MRC changes within 1 symbol period
- rapid amplitude fluctuations

– uncommon in most digital communication systems

B) Slow Fading → $T_s << T_c$ or $B_s >> B_D$

– MRC constant over many symbol periods

– slow amplitude fluctuations

– for v = 60 mph @ f_c= 2 GHz → B_D= 178 Hz

$\therefore B_s \approx$ 2 kHz $>> B_D$

- B_s almost always $>> B_D$ for most applications

Parameters involved in mobile multipath channels

1) Time Dispersion Parameters

i) Mean excess delay

- Average delay measured w.r.t the first moment of the power delay profile and is defined as

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

ii)RMS delay spread

- It is the square root of the second central moment of the power delay profile.

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

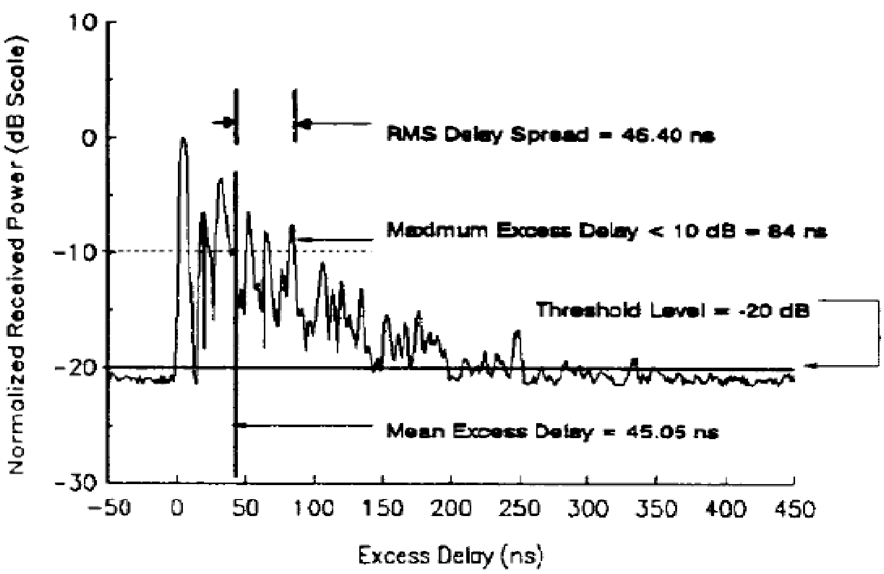
where

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

iii)Maximum Excess Delay (X dB)

- The time delay during which multipath energy falls to X dB below the maximum level.
- It is also called excess delay spread

Maximum Excess Delay= $\tau_X - \tau_0$



Coherence bandwidth(Bc)

- Coherence bandwidth is the range of frequencies over which two frequency components have a strong potential for amplitude correlation.

If the frequency correlation function is above 0.9

$$B_c \approx \frac{1}{50\sigma_\tau}$$

If the frequency correlation function is above 0.5

$$B_c \approx \frac{1}{5\sigma_\tau}$$

Doppler spread

Doppler spread, B_D, is defined as the maximum Doppler shift f_m

$$f_m = v/\lambda$$

Coherence Time (Tc)

- Doppler spread and coherence time (Tc) are inversely proportional.

Coherence time (T_C) is defined as:

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

- time over which the time corrleation function is above 0.5, then

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

Coherence time is also defined as: $$T_C \approx \sqrt{\frac{9}{16\pi f_m^2}} = \frac{0.423}{f_m}$$

UNIT-II

CELLULAR ARCHITECTURE

Near-Far problem.

- near–far problem or near ability problem is a situation that is common in wireless communication systems, in particular CDMA..
- The near–far problem is a condition in which a receiver captures a strong signal and thereby makes it impossible for the receiver to detect a weaker signal.

FCA-Fixed Channel assignment

- Each cell is allocated a predetermined set of channels.

- Any call attempt within the cell can only be served by the unused channels in that particular cell.
- A cell is allowed to borrow a channel from neighboring cell if all of its channels are occupied.
- Mobile switching center (MSC) supervises borrowing strategy procedures.

Forward and reverse channel

Forward channel is a radio channel used for transmission of information from base station to mobile. Reverse channel is a radio channel used for transmission from mobile to base station.

Frequency reuse.

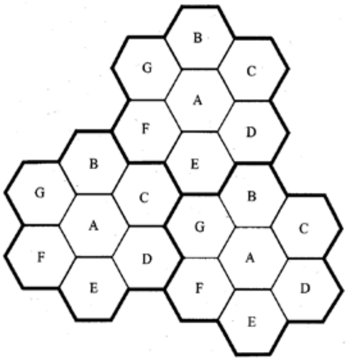
Frequency reuse is the process of using the same radio frequencies on radio transmitter sites within a geographic area, which are separated by sufficient distance to cause minimal interference with each other. Frequency reuse allows for a dramatic increase in the number of customers that can be served (capacity) within a geographic area on a limited amount of radio spectrum.

Cell splitting.

Cell splitting is the process of subdividing congested cells into smaller cells each with its own base stations and a corresponding reduction in antenna height and transmitter power. It increases the capacity of cellular system.

frequency reuse concept in cellular systems

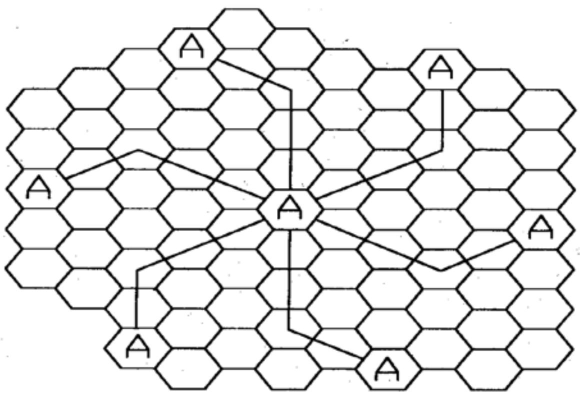
- Each cellular base station is allocated a group of radio channels within a small geographic area called a *cell*.
- Neighboring cells are assigned different channel groups.
- By limiting the coverage area to within the boundary of the cell, the channel groups may be reused to cover different cells.
- Keep interference levels within tolerable limits



- Frequency reuse or frequency planning
- Hexagonal geometry has
- exactly six equidistance neighbors
- each of its neighbors are separated by multiples of 60 degrees.
- Consider a cellular system which has a total of *S* duplex channels.
- Each cell is allocated a group of *k* channels.
- The *S* channels are divided among *N* cells.
- The total number of available radio channels $S = kN$
- The *N* cells which use the complete set of channels is called *cluster*.
- The cluster can be repeated *M* times within the system. The total number of channels, *C*, is used as a measure of capacity $C = MkN = MS$
- The capacity is directly proportional to the number of replication *M*.
- The cluster size, *N*, is typically equal to 4, 7, or 12.

Rules for determining the nearest co-channel neighbours

- Step 1: Move i cells along any chain of hexagons.
- Step 2: Turn 60 degrees counter clockwise and more j cells
- Only certain cluster sizes and cell layout are possible.



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

- Co-channel neighbors of a
- particular cell,
- ex, *i=3* and *j=2*.

Various multiple access technique :

To accommodate a number of users, many traffic channels need to be made available.

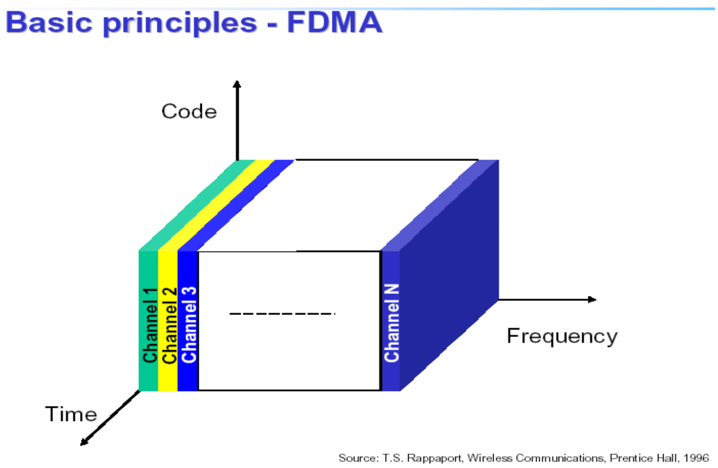
There are three basic ways to have many channels within an allocated bandwidth:

a. Frequency Division Multiple Access (FDMA)

b. Time Division Multiple Access (TDMA)

c. Code Division Multiple Access (CDMA)

<u>Frequency Division Multiple Access (FDMA):</u>



This was the initial multiple-access technique for cellular systems in which each individual user is assigned a pair of frequencies while making or receiving a call as shown in Fig .1. One frequency is used for downlink and one pair for uplink. This is called frequency division duplexing (FDD).

The features of FDMA are as follows:

1. The FDMA channel carries only one phone circuit at a time.

2. If an FDMA channel is not in use, then it sits idle and it cannot be used by other users to increase share capacity.

3. After the assignment of the voice channel the BS and the MS transmit simultaneously and continuously.

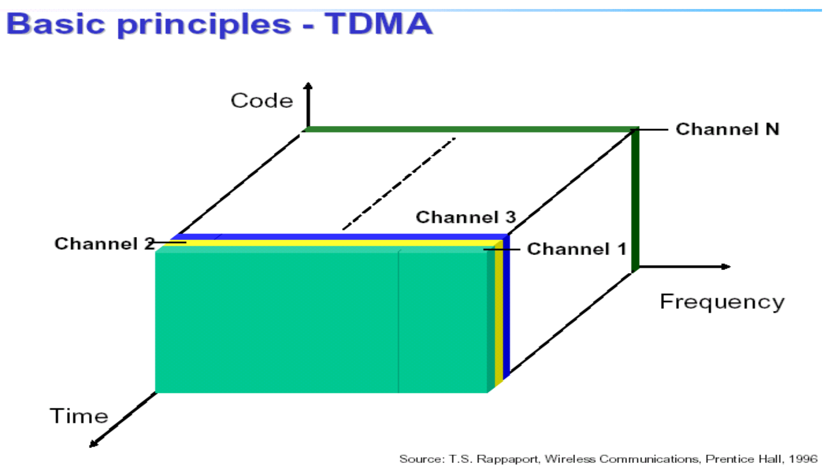
4. The bandwidths of FDMA systems are generally narrow i.e. FDMA is usually 159 implemented in a narrow band system.

6. The symbol time is large compared to the average delay spread.

7. The complexity of the FDMA mobile systems is lower than that of TDMA mobile systems.

Time Division Multiple Access (TDMA)

In digital systems, continuous transmission is not required because users do not use the allotted bandwidth all the time. In such cases, TDMA is a complimentary access technique to FDMA. Global Systems for Mobile communications (GSM) uses the TDMA technique.



The features of TDMA includes the following:

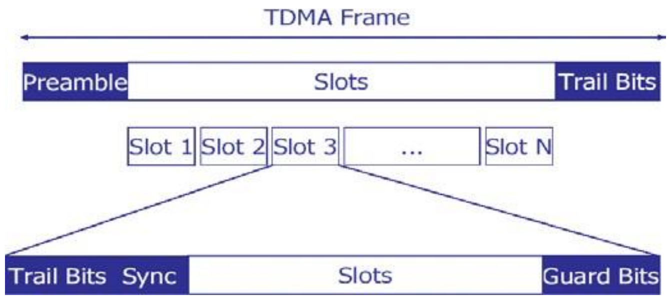
1. TDMA shares a single carrier frequency with several users where each users makes use of non overlapping time slots.

2. The number of time slots per frame depends on several factors such as modulation technique, available bandwidth etc.

3. Data transmission in TDMA is not continuous but occurs in bursts. This results in low battery consumption since the subscriber transmitter can be turned OFF when not in use.

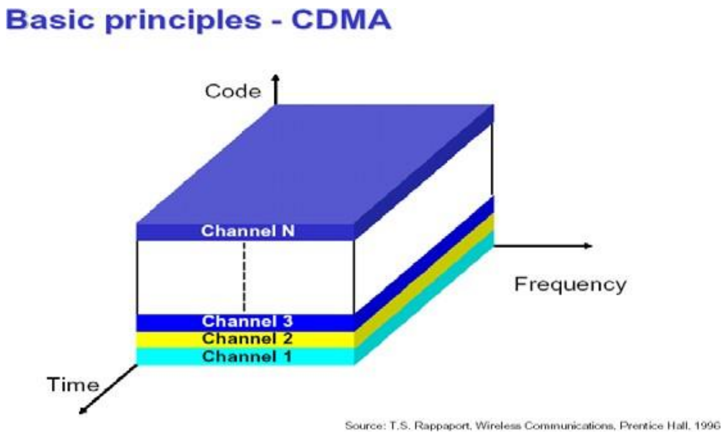
4. Because of a discontinuous transmission in TDMA the handoff process is much simpler for a subscriber unit, since it is able to listen to other base stations during idle time slots.

5. TDMA uses different time slots for transmission and reception thus duplexers are not required.



Code Division Multiple Access (CDMA):

In CDMA, the same bandwidth is occupied by all the users, however they are all assigned separate codes, which differentiates them from each other as shown in Figure.4. CDMA utilize a spread spectrum technique in which a spreading signal (which is uncorrelated to the signal and has a large bandwidth) is used to spread the narrow band message signal. Direct Sequence Spread Spectrum (DS-SS). This is the most commonly used technology for CDMA. In DS-SS, the message signal is multiplied by a Pseudo Random Noise Code. Each user is given his own codeword which is orthogonal to the codes of other users and in order to detect the user, the receiver must know the codeword used by the transmitter.

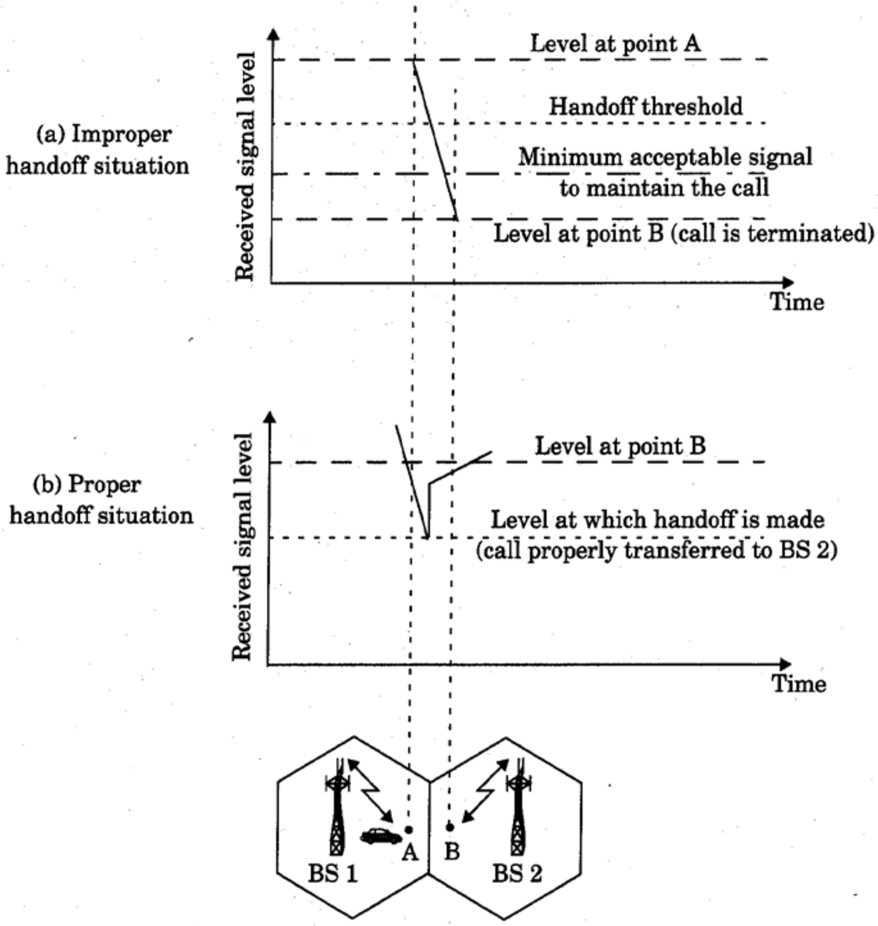


There are two basic types of implementation methodologies:

- Direct Sequence (DS)
- Frequency Hoping (FH)

Handoff scenario with neat diagram.

When a mobile moves into a different cell while a conversation is in progress, the MSC automatically transfers the call to a new channel belonging to the new base station.



Handoff operation

- identifying a new base station
- re-allocating the voice and control channels with the new base station.

Handoff Threshold

- Minimum usable signal for acceptable voice quality (-90dBm to -100dBm)
- Handoff margin cannot be too large or too small.
- If it is too large, unnecessary handoffs burden the MSC
- If it is too small, there may be insufficient time to complete handoff before a call is lost.
- Handoff must ensure that the drop in the measured signal is not due to momentary fading and that the mobile is actually moving away from the serving base station.
- Running average measurement of signal strength should be optimized so that unnecessary handoffs are avoided.
  - Depends on the speed at which the vehicle is moving.
  - Steep short term average -> the hand off should be made quickly

- The speed can be estimated from the statistics of the received short-term fading signal at the base station

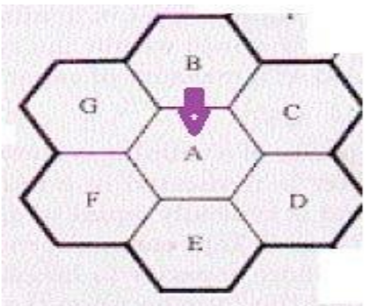
- Dwell time: the time over which a call may be maintained within a cell without handoff.
- Dwell time depends on
  - propagation
  - interference
  - distance
  - speed

Channel assignment strategies.

For efficient utilization of radio spectrum

- Frequency reuse scheme
  - increases capacity
  - minimize interference
- Two Channel assignment approach
  1) fixed channel assignment
  2) dynamic channel assignment

Fixed channel assignment(FCA)



- each cell is allocated a predetermined set of voice channel
- any new call attempt can only be served by the unused channels in that particular cell
- the call will be *blocked* if all channels in that cell are occupied

- To improve cell spectrum utilization, a borrowing option is considered
- A cell is allowed to borrow channels from neighbouring cell if all of its own channels are already occupied

Dynamic channel assignment(DCA)strategy

- channels are not allocated to cells permanently.
- allocate channels based on request.

- reduce the likelihood of blocking, increase capacity.
- If traffic in the network is uniform , number of active users in each cell are same, results in optimum channel allocation strategy

UNIT-III

DIGITAL SIGNALLING FOR FADING CHANNELS

QPSK.

QPSK is defined as the multilevel modulation scheme in which four phase shifts are used for representing four different symbols.

Merits of MSK.

- Constant envelope
- Spectral efficiency
- Good BER performance
- Self-synchronizing capability
- MSK is a spectrally efficient modulation scheme and is particularly attractive for use in mobile radio communication systems.

Non linear modulation.

In the non linear modulation the amplitude of the carrier is constant, regardless of the variation in the modulating signals. Non-linear modulations may have either linear or constant envelopes depending on whether or not the baseband waveform is pulse shaped.

What is the need of Gaussian filter?

- Gaussian filter is used before the modulator to reduce the transmitted bandwidth of the signal.
- It uses less bandwidth than conventional FSK.

Advantages of MSK over QPSK.

- In QPSK the phase changes by 90degree or 180 degree .This creates abrupt amplitude variations in the waveform, Therefore bandwidth requirement of QPSK is more filters of other methods overcome these problems , but they have other side effects.
- MSK overcomes those problems. In MSK the output waveform is continuous in phase hence there are no abrupt changes in amplitude.

Quadrature Phase Shift Keying with neat diagram.

- Quadrature Phase Shift Keying (QPSK) has twice the bandwidth efficiency of BPSK, since 2 bits are transmitted in a single modulation symbol.
- The phase of the carrier takes on 1 of 4 equally spaced values, such as 0, π/2, π, 3π/2 [or] π/4, 3π/4, 5π/4, 7π/4. where each value of phase corresponds to a unique pair of message bits.

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

$$S_{QPSK}(t) = \sqrt{\frac{2E_S}{T_S}} \cos\left[2\pi f_c t + (i-1)\frac{\pi}{2}\right] \qquad 0 \le t \le T_S,\quad i = 1, 2, 3, 4$$

where Ts → symbol duration and is equal to twice the bit period

using $\cos(A+B) = \cos(A)\cos(B) - \sin(A)\sin(B)$

S $_{QPSK}$ can be rewritten as,

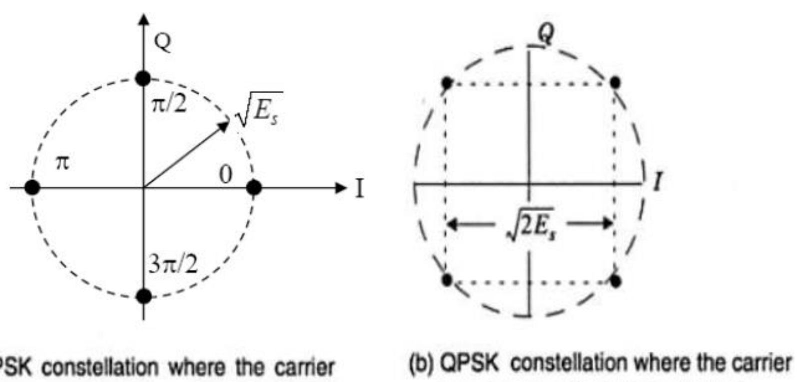
$$S_{QPSK} = \sqrt{\frac{2E_s}{T_S}} \cos(2\pi f_c t)\cos[(i-1)\frac{\pi}{2}] - \sqrt{\frac{2E_s}{T_S}} \sin(2\pi f_c t)\sin[(i-1)\frac{\pi}{2}]$$

$$\text{If } \phi_1(t) = \sqrt{\frac{2}{T_S}} \cos(2\pi f_c t) \text{ and } \phi_2(t) = \sqrt{\frac{2}{T_S}} \sin(2\pi f_c t)$$

Then, QPSK signal can be expressed as

$$S_{QPSK} = \sqrt{E_s}\cos[(i-1)\frac{\pi}{2}]\phi_1(t) - \sqrt{E_s}\sin[(i-1)]\frac{\pi}{2}\phi_2(t) \qquad \text{for } i = 1, 2, 3, 4$$

<u>Constellation Diagram</u>



(a) QPSK constellation where the carrier phases are 0, π/2, π, 3π/2;

(b) QPSK constellation where the carrier phases are π/4, 3π/4, 5π/4, 7π/4.

From the constellation diagram of a QPSK signal, it can be seen that the distance between adjacent points in the constellation is $\sqrt{2E_s}$ .Since each symbol corresponds to two bits, then Es = 2E_b, thus the distance between two neighboring points in the QPSK constellation is equal to $2\sqrt{E_b}$.

The average probability of bit error in the AWGN channel is obtained as

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

Spectrum and Bandwidth of QPSK

The PSD of a QPSK signal using rectangular pulses can be expressed as

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

Sub $E_s = 2E_b$

$T_s = 2T_b$

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

QPSK Transmission and Detection Techniques

QPSK Transmitter

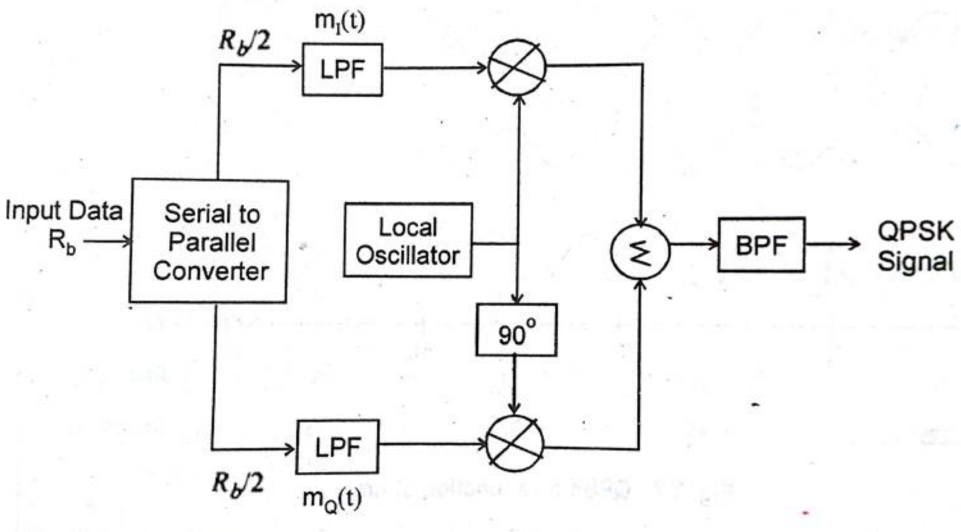


Fig. QPSK Transmitter

➢ The unipolar binary message stream has bit rate R_b and it is first converted into a bipolar NRZ sequence.

- The bit stream m(t) is then split into two bit streams in $m_I(t)$ and m_Q (t) (in-phase and quadrature streams), each having a bit rate of Rs = $R_b/2$.
- The bit stream $m_I(t)$ is called the "even" stream and m_Q (t) is called the "odd" stream.
- The two binary sequences are separately modulated by two carriers $\varphi_1(t)$ and $\varphi_2(t)$, which are in quadrature. The two modulated signals, each of which can be considered to be a BPSK signal, are summed to produce a QPSK signal.
- The filter at the output of the modulator confines the power spectrum of the QPSK signal within the allocated band.

QPSK Receiver

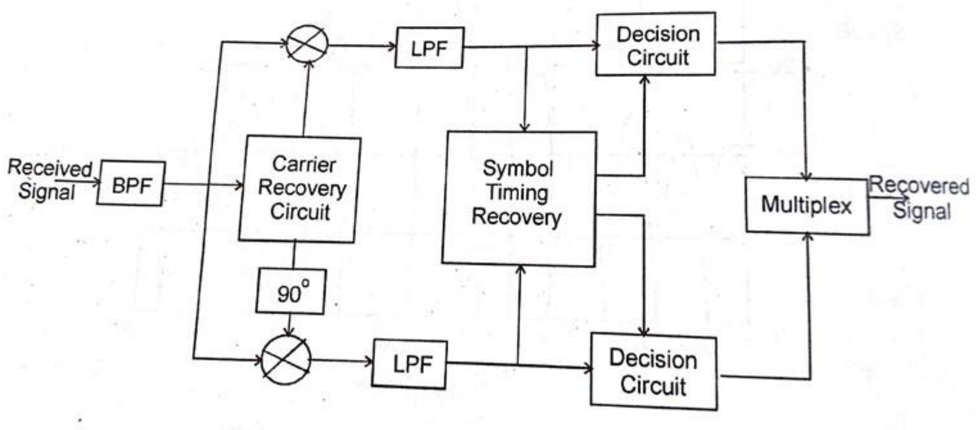


Fig. QPSK Receiver

- The frontend bandpass filter removes the out-of-band noise and adjacent channel interference.
- The filtered output is split into two parts, and each part is coherently demodulated using the in-phase and quadrature carriers.
- The outputs of the demodulators are passed through decision circuits which generate the in-phase and quadrature binary streams.
- The two components are then multiplexed to reproduce the original binary sequence.

Advantages of QPSK:

- Low error probability
- Very good noise immunity
- Reduced bandwidth
- High bit rate

Disadvantages of QPSK:

- Inter Channel Interference(ICI) is large due to side lobes.
- Only suitable for rectangular pulses.
- QPSK phase changes by 90^0 and 180^0.This creates abrupt amplitude variations in the waveform.

Offset QPSK with neat diagram.

Drwabacks of QPSK

The amplitude of QPSK signal is ideally constant. When QPSK signals are pulse shaped, they lose constant envelope property. It causes,

i) Regeneration of side lobes
ii) Spectral widening

Why OQPSK?

To overcome the above drawbacks, A modified form of QPSK called offset QPSK (OQPSK) or staggered QPSK is preferred. It has more advantages than QPSK as follows:

i) Used for Non-rectangular pulses
ii) supports more efficient RF amplification

OQPSK signaling is similar to QPSK signaling, as represented by equation

$$S(t) = \sqrt{\frac{2E_s}{T_S}} \cos(2\pi f_c t) \cos[(i-1)\frac{\pi}{2}] - \sqrt{\frac{2E_s}{T_S}} \sin(2\pi f_c t) \sin[(i-1)\frac{\pi}{2}]$$

- In QPSK , first input bit stream is split into two streams referred as even and odd bit streams . These can be applied simultaneously to the mixers.

- But in OQPSK, after splitting the even and odd bit streams, $m_I(t)$ and $m_Q(t)$, one bit stream is made offset by one bit period with each other. After this, the direct and shifted bit streams are fed to the mixers.

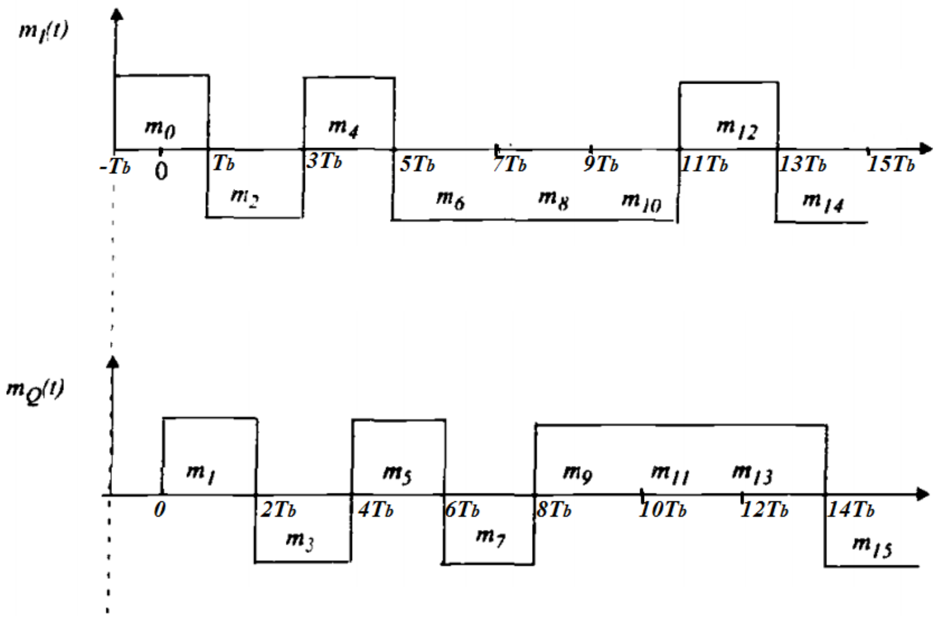


Figure: Offset waveforms of in-phase and quadrature arms of an OQPSK modulator

- In QPSK, phase transitions occur only once every Ts = $2T_b$ sec. maximum phase shift = 180^0 .
- However, in OQPSK signaling, phase transitions occur every T_b sec. maximum phase shift = ±90°
- Due to 90° phase transitions, OQPSK signals does not cause the signal envelope to go to zero.
- OQPSK allows nonlinear amplification, so regeneration of side lobes are eliminated.
- OQPSK signal is identical to that of a QPSK signal. Hence, both signals occupy the same bandwidth.

Applications

- Very attractive for mobile communication systems
- OQPSK signals perform better than QPSK in the presence of phase jitter.

Minimum Shift Keying with neat diagram.

- Minimum shift keying (MSK) is a special type of Continuous Phase Frequency Shift Keying (CPFSK) wherein the peak frequency deviation is equal to 1/4 the bit rate.
- In other words, MSK is continuous phase FSK with a modulation index of 0.5.
- The modulation index is defined as,

$$k_{FSK} = \frac{(2\Delta F)}{R_b}$$

Where,

$\Delta F \rightarrow$ peak RF frequency deviation

Rb $\rightarrow$ bit rate.

- Two FSK signals to be coherently orthogonal, and the name minimum shift keying implies the minimum frequency separation (i,e. bandwidth) that allows orthogonal detection.
- Two FSK signals $v_H(t)$ and $V_L(t)$ are said to be orthogonal, if

$$\int_0^T V_H(t)\, V_L(t)\, dt = 0$$

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

Properties of MSK:

1)Constant Envelope

2)Spectral efficiency

3)Good BER performance

4)Self-Synchronization

- MSK signal can be defined as,

$$S_{MSK}(t)=\sum_{i=0}^{N-1} m_I(t)P(t-2iT_b)\cos 2\pi f_c t+\sum_{i=0}^{N-1} m_Q(t)P(t-2iT_b-T_b)\sin 2\pi f_c t$$

$$\text{where}\quad P(t)=\begin{cases}\sin\left(\dfrac{\pi t}{2T_b}\right) & 0\le t\le 2T_b\\ 0 & \text{elsewhere}\end{cases}$$

where $m_I(t)$ and $m_Q(t)$ are the "odd" and "even" bits of the bipolar data streams.

- MSK waveform can be seen as a special type of a continuous phase FSK and the above equation is rewritten as,

$$S_{MSK}(t)=\sqrt{\frac{2E_b}{T_b}}\cos\left[2\pi f_c t-m_I(t)m_Q(t)\frac{\pi t}{2T_b}+\phi_k\right]$$

where φ_k is 0 or π depending on whether $m_I(t)$ is 1 or -1 and MSK has a constant amplitude.

MSK Power Spectrum

For MSK, the baseband pulse shaping function is given by

$$P(t)=\begin{cases}\cos\left(\dfrac{\pi t}{2T}\right) & |t|<T\\ 0 & \text{elsewhere}\end{cases}$$

Thus the normalized power spectral density for MSK is given by

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

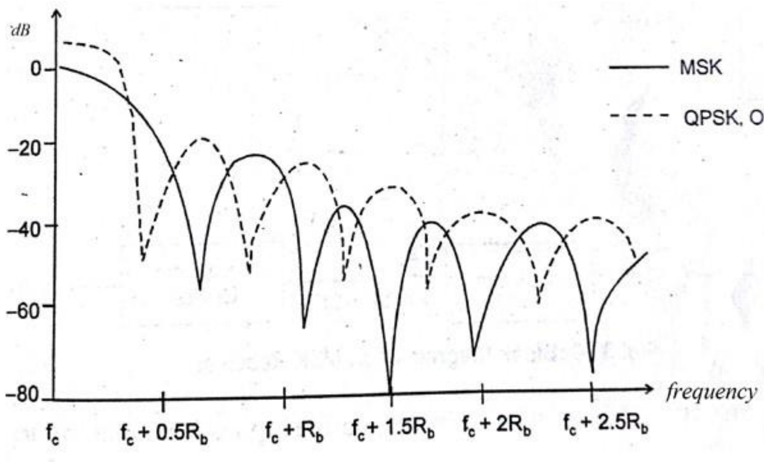


Fig:Power spectral density of MSK signals as compared to QPSK and OQPSK signals.

- From above Figure, it is seen that the MSK spectrum has lower sidelobes than QPSK and OQPSK.
- Main lobe of MSK is wider than that of QPSK and OQPSK.

- MSK is less spectrally efficient than the phase-shift keying techniques.

MSK Transmitter and Receiver

MSK Transmitter

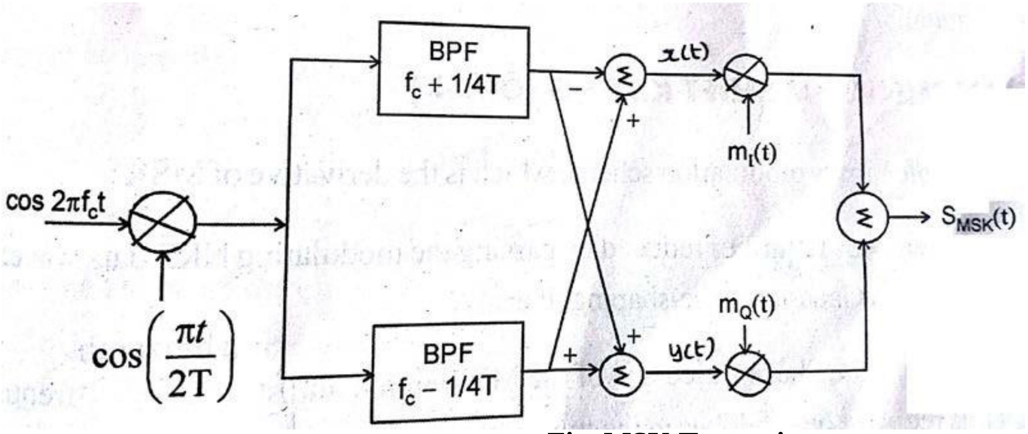


Fig: MSK Transmitter

- Multiplying a carrier signal with cos (πt/2T) produces two phase-coherent signals at fc+ 1/4T and fc- I/4T.
- These two FSK signals are separated using two narrow bandpass filters and appropriately combined to form the in-phase and quadrature carrier components *x(t) and y(t)*, respectively.
- These carriers are multiplied with the odd and even bit streams, mI(t) and mQ(t), to produce the MSK modulated signal S_{MSK}(t).

MSK Receiver

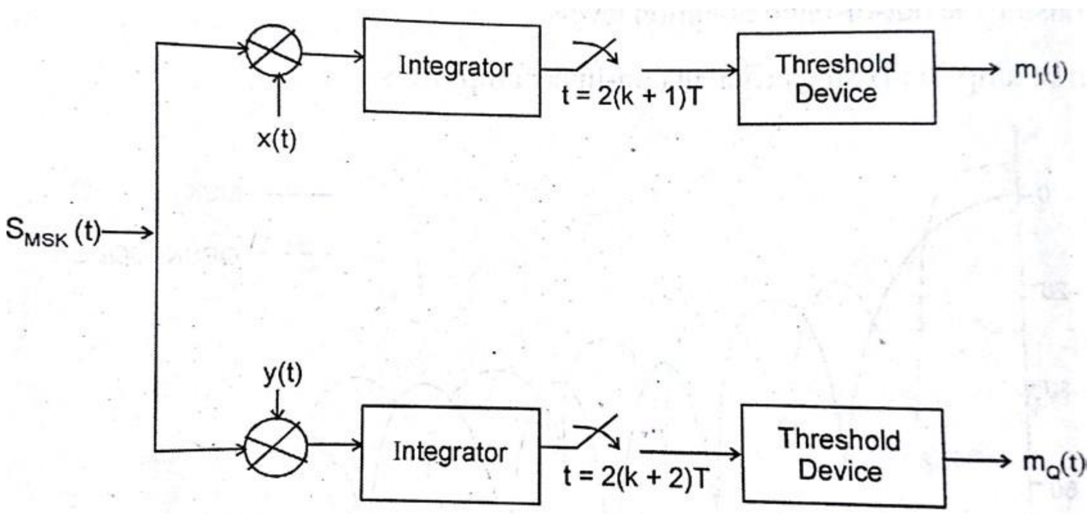


Fig: MSK Receiver

- The received signal S_{MSK}(t) is multiplied by the respective in-phase and quadrature carriers *x(t)* and *y(t)*.

- The output of the multipliers are integrated over two bit periods. Then, it is given to a decision circuit which has a threshold detector.
- Based on the level of the signal above or below the threshold, the threshold detector decides whether the signal is 0 or 1.
- The output data streams corresponds to $m_I(t)$ and $m_Q(t)$, which are offset combined to obtain the demodulated signal.

Advantages of MSK as compared to QPSK

- MSK baseband waveforms are smoother than QPSK.
- MSK signal have continuous phase in all cases, whereas QPSK signals have abrupt amplitude variations.
- The main lobe of MSK is wider than that of QPSK and contains around 99% of signal energy. Whereas QPSK main lobe contains around 90% of signal energy.
- Side lobes of MSK are smaller than QPSK. Whereas side lobes are bigger in QPSK. So, Inter channel Interference(ICI) occurs due to large side lobes in QPSK.
- BPF is not required to avoid the ICI in MSK. But QPSK needs BPF to avoid ICI problems.

GMSK transmission and reception with necessary diagrams.

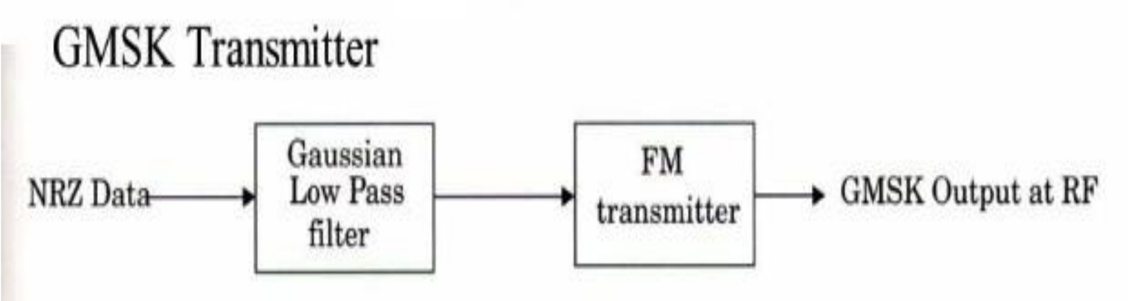
- Main lobe of MSK is wide. It is unsuitable for narrow bandwidth applications. This can be overcome by GMSK.
- derivative of MSK.
- In GMSK, the side lobe levels of the spectrum are further reduced by passing the modulating NRZ data waveform through a pre-modulation Gaussian pulse-shaping filter.

The GMSK premodulation filter has an impulse response given by

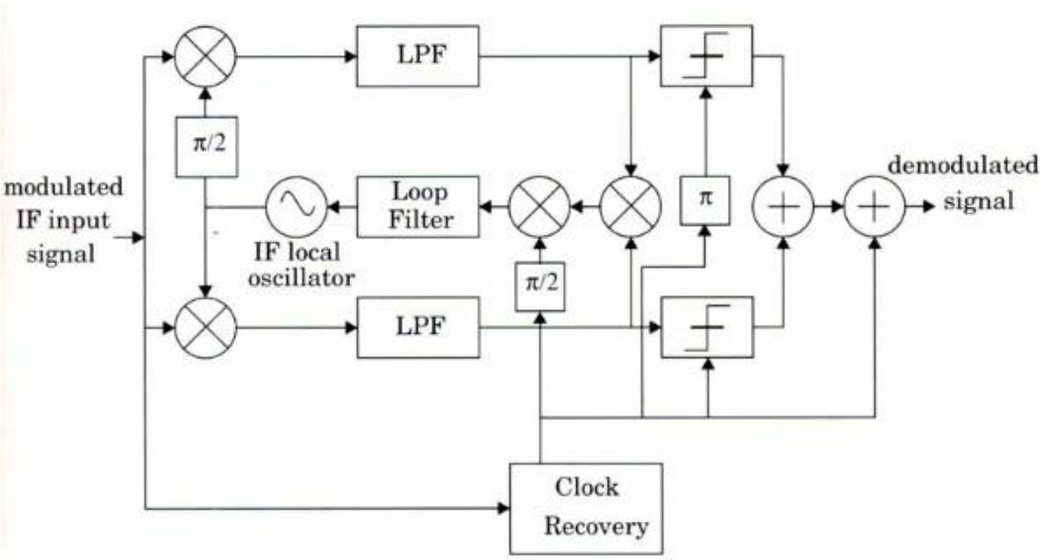
$$h_G(t) = \frac{\sqrt{\pi}}{\alpha}\exp\left(-\frac{\pi^2}{\alpha^2}t^2\right)$$

transfer function given by

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

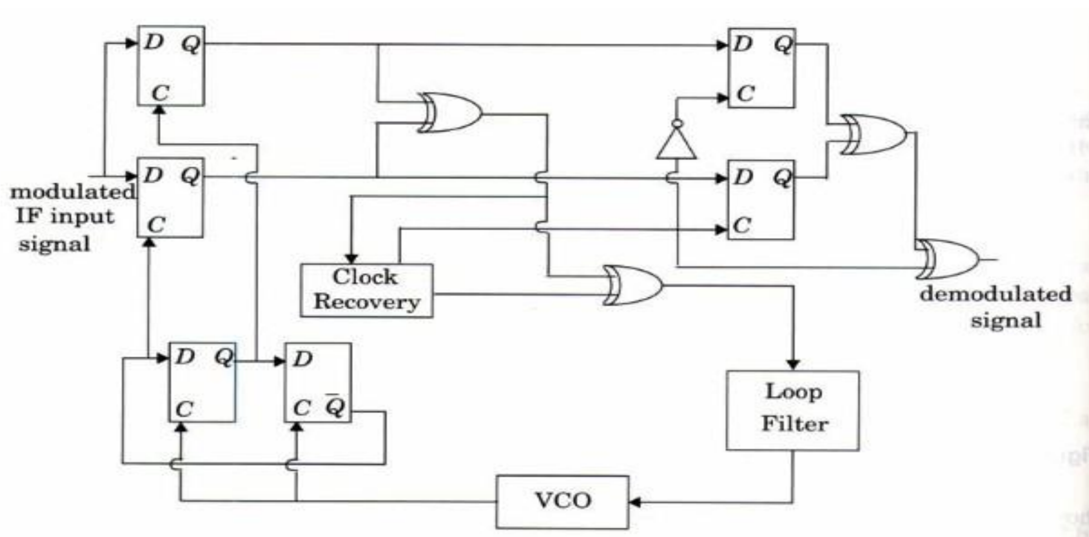


GMSK Receiver



GMSK receiver.

Digital logic circuit



Digital logic circuit for GMSK demodulation

GMSK bit error rate

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

where γ is a constant related to BT by

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

UNIT-IV

MULTIPATH MITIGATION TECHNIQUES

link performance

Link performance can be improved by various techniques such as

- Equalization
- Diversity
- Channel coding

Diversity

Diversity is used to compensate the fading channel impairments and is usually implemented by using two or more receiving antennas. Diversity improves transmission performance by making use of more than one independently faded version of the transmitted signal.

Nonlinear equalization methods:

Commonly used non linear equalization methods are:

1) Decision feedback equalization
2) Maximum likelihood symbol detection
3) Maximum likelihood sequence estimation

Frequency diversity

Correlation is increased by transmitting information on more than one carrier frequency. Frequencies are separated by more than one coherence bandwidth of the channel. So the signals will not experience same fades.

linear and non-linear equalizer

Linear equalizer: the current and past values of the received signal are linearly weighted by the filter coefficients and summed to produce the output. No feedback path is used.

Simple and easy to implement. Not suitable for severely distorted channel. Noise power signal is enhanced.

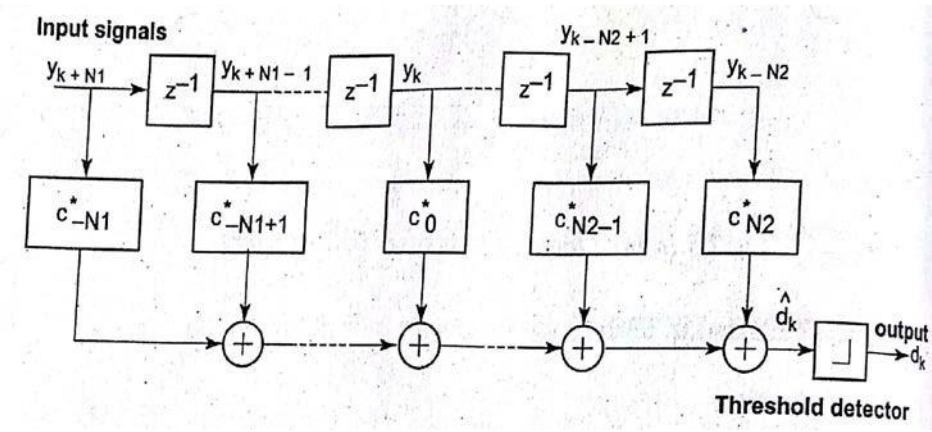
Nonlinear equalizer: If the past decisions are correct, then the ISI contributed by present symbol can be cancelled exactly, feedback path is used. Suitable for severely distorted channel. Noise power signal is not enhanced. Complex in structure. channels with low SNR. Suffers from error propagation.

Linear equalizers.

- A linear equalizer can be implemented as an FIR filter, otherwise known as the transversal filter. This type of equalizer is the simplest type available.
- In such an equalizer, the current and past values of the received signal are linearly weighted by the filter coefficient and summed to produce the output.
- If the delays and the tap gains are analog, the continuous output of the equalizer is sampled at the symbol rate and the samples are applied to the decision device.

1) <u>Linear Transversal Equalizer</u>

- Implemented like FIR filter.



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

Error = Desired output – Actual output

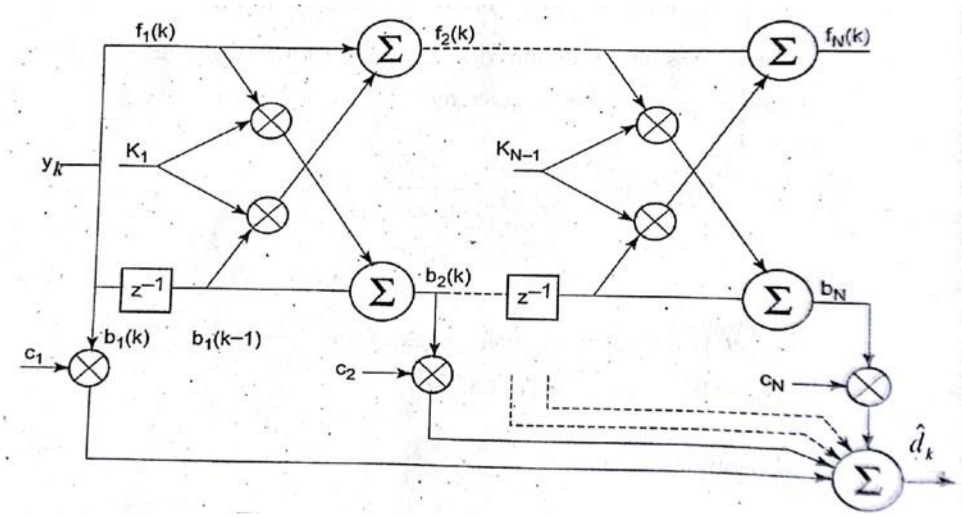
$$\varepsilon_k = d_k - \hat{d}_k$$

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

Lattice Equalizer:

The each stage of the lattice is characterized by the following *recursive* equations as,

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



output of the equalizer is given by,

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

Advantages

- stability
- Faster convergence

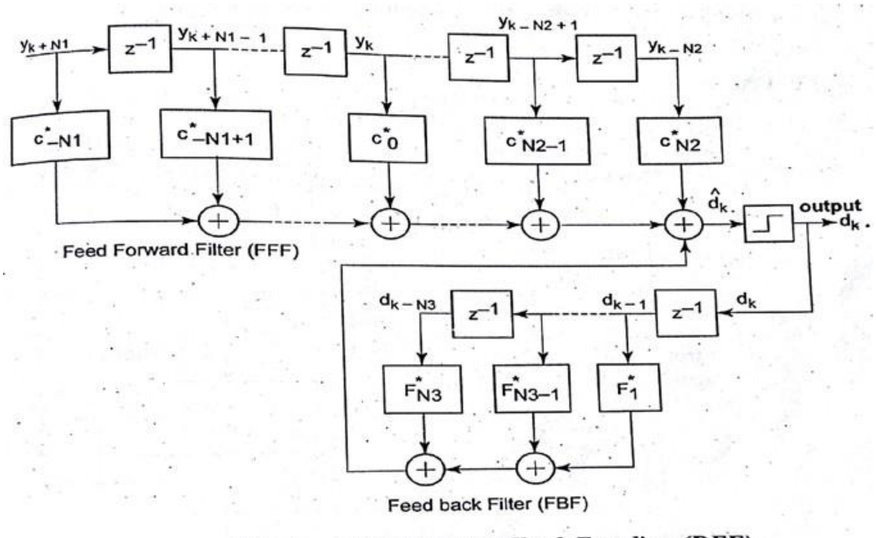
Disadvantages

- The structure is more complicated

Non-linear equalizers.

- Output d(t) is fed back to change the subsequent outputs

(1) Direct transversal form of DFE:



Structure of Decision Feedback Equalizer (DFE)

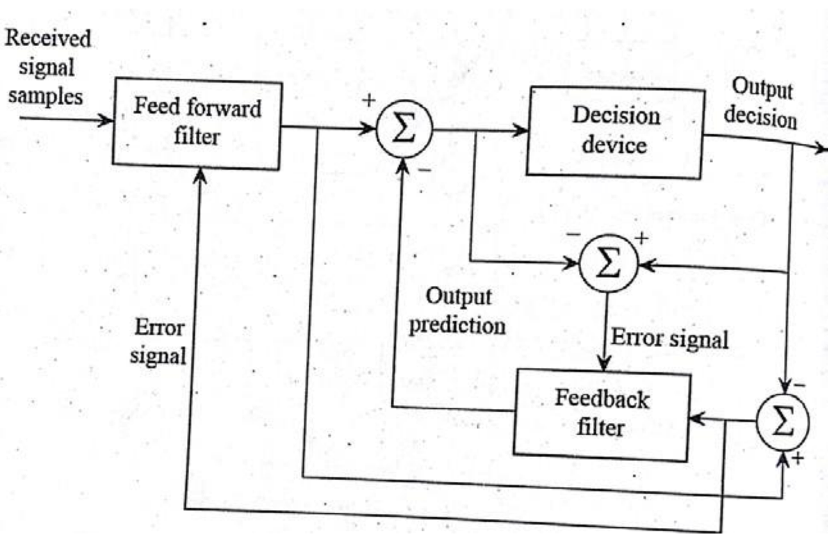
The equalizer has $N_1 + N_2 + 1$ taps in the feed forward filter

N_3 taps in the feedback filter

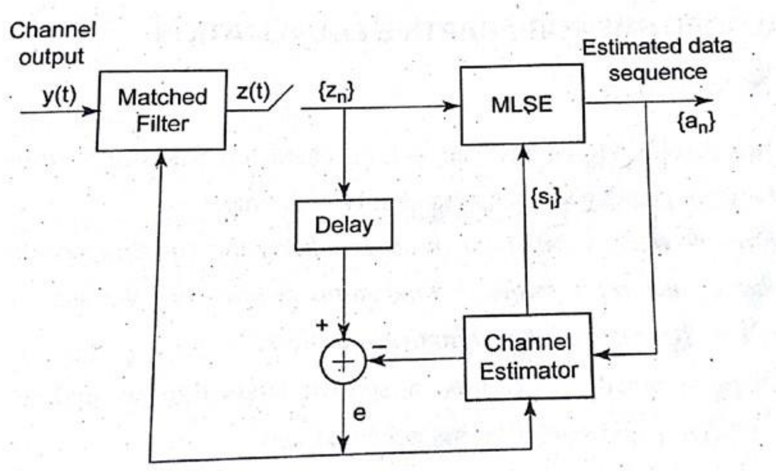
output can be expressed as,

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

(2) Predictive DFE:



3) Maximum Likelihood Sequence Estimation (MLSE) Equalizer



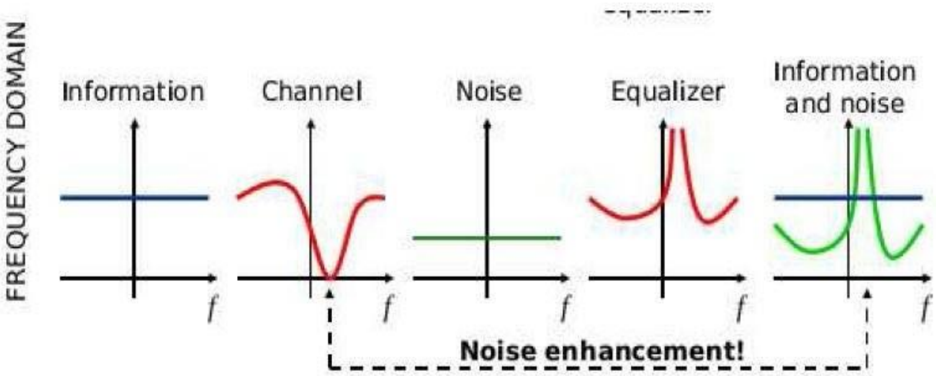
- Computational complexity of an MLSE increases with large delay spread
- MLSE is optimal and to minimizes the probability of a sequence error.
- The MLSE requires the knowledge of the channel characteristics in order to compute the metrics for making decisions.
- The statistical distribution of the noise corrupting the signal.

zero forcing and LMS algorithm for adaptive equalization.

<u>Zero Forcing(ZF) algorithm</u>

- zero-forcing (ZF) equalizer, the equalizer coefficients Cn are chosen to force the samples of the combined channel and equalizer impulse response to zero.

- When each of the delay elements provides a time delay equal to the symbol duration T, the frequency response Heq(f) of the equalizer is periodic with a period equal to the symbol rate 1/T.



$$H_{ZF}(Z) = \frac{1}{F(Z)}$$

Advantage:

performs well for static channels with high SNR→local wired telephone lines

Disadvantage:

Noise Enhancement makes not suitable for Wireless.

LMS Algorithm

- LMS algorithm is the simplest algorithm based on minimization of the MSE between the desired equalizer output and the actual equalizer output.Here the system error, the MSE and the optimal Wiener solution remain the same as given the adaptive equalization framework.
  minimization of the MSE is carried out recursively, and may be performed by use of the stochastic gradient algorithm. It is the simplest equalization algorithm and requires only 2N+1 operation per iteration.

- The filter weights are updated by the update equation. Letting the variable n denote the sequence of iteration, LMS is computed iteratively by

$$w_k(n + 1) = w_k(n) + \mu e_k(n)\ x(n - k)$$

- Where the subscript k denotes the k^{th} delay stage in the equalizer and μ is the step size which controls the convergence rate and stability of the algorithm.
- The LMS equalizer maximizes the signal to distortion ratio at its output within the constraints of the equalizer filter length.
- If an input signal has a time dispersion characteristics that is greater than the propagation delay through the equalizer, then the equalizer will be unable to reduce distortion.
- The convergence rate of the LMS algorithm is slow due to the fact that there is only one parameter, the step size, that controls the adaptation rate.
- To prevent the adaptation from becoming unstable, the value off is chosen from

$$0 < \mu < 2/ \left(\sum_{i=1}^{N} \lambda_i \right)$$

Where λ_i is the i^{th}eigen value of the covariance matrix R.

Let us denote the received sequence vector at the receiver and the input to the equalizer as

$$x_k = [x_k; x_{k-1} ; :::::; x_{k-N}]^T;$$

and the tap coefficient vector as

$$w_k= [w^0_k ;w^1_k ; :::::;W^n_k]T$$

Now, the output sequence of the equalizer y_k is the inner product of $x_k$and w_k, i.e.,

$$y_k = x_k;w_k = x^T_k w_k = w^T{}_k x^k$$

The error signal is defined as

$$e_k = d_k-y_k = d_k-x^T{}_k w_k$$

Assuming d_k and x_k to be jointly stationary, the Mean Square Error (MSE) is given as

$$MSE = E[e^2{}_k]$$

Various diversity techniques used in wireless communication.

Five common methods are

(1) spatial diversity: Several antenna elements separated in space

(2) Temporal diversity : Repetition of the transmit signal at different times

(3) frequency diversity : Transmission of the signal on different frequencies

(4) Angular diversity: Multiple antennas with different antennas with different antenna patterns

(5) polarization diversity: Multiple antennas receiving different polarizations

Spatial Diversity

- Space Diversity or Antenna diversity
- Use more than one antenna to receive the signal.

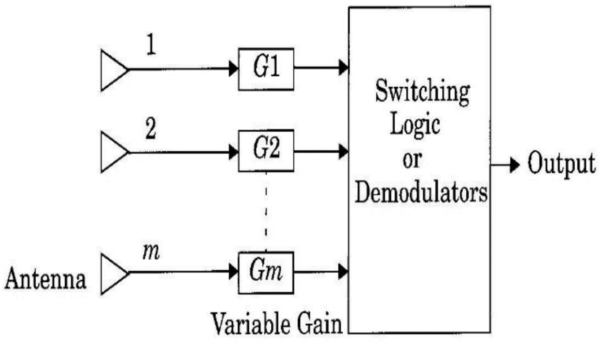


Figure 7.12 Generalized block diagram for space diversity.

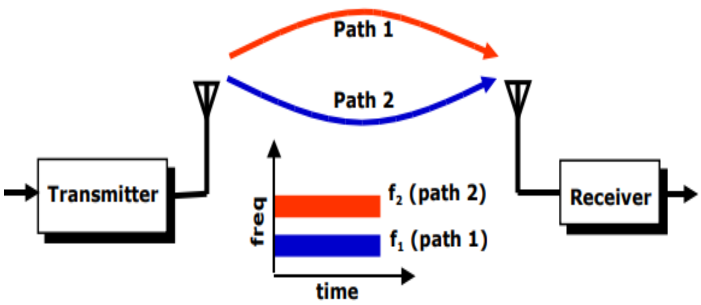
- The space separation between adjacent antennas should be large enough so as to ensure that the signals from different antennas are independently faded

Temporal diversity

- As the wireless propagation channels are time variant, signals that are received at different times are uncorrelated.
- Temporal distance is $D = \frac{1}{2Vmax}$
- Vmax→ Maximum Doppler frequency
- Temporal diversity can be realized in 3 different ways.
- Repetition coding
- Automatic repeat request
- Combination of interleaving and coding

Frequency Diversity

- Frequency diversity is implemented by transmitting same signal at two or more different carrier frequencies.
- Our aim is to make these carrier frequencies uncorrelated to each other, so that they will not experience the same fades.
- To make them least correlated, these carrierfrequencies are separated by more than the coherence bandwidth of the channel.

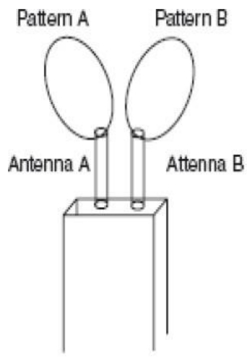


- Frequency diversity is often employed in microwave line –of sight links.
- These links uses Frequency division multiplexing mode(FDM).

Angular or Pattern diversity

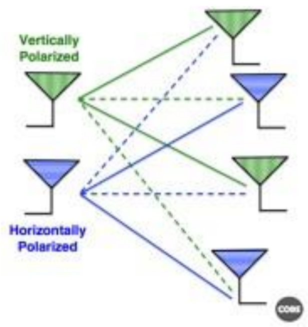
- It enhances the decorrelation of signals at closely spaced antennas by using Different antenna patterns can be achieved very easily.
- Even identical antennas can have different patterns when they are mounted close to each other.
- This effect is due to Mutual Coupling.

- Place 2 identical antennas close to each other
- Here antenna B acts as a reflector for antenna A.
- Antenna A acts as a reflector for Antenna B.



Angle diversity for closely spaced antennas.

Polarization diversity



- Multiple versions of a signal are transmitted and received via antennas with different polarization. i.e. horizontal and vertical.
- A diversity combining technique is applied on the receiver side.

UNIT-V

MULTIPLE ANTENNA TECHNIQUES

PART -A

Beam forming

The multiple antennas at the transmitter and receiver can be used to obtain array and diversity gain instead of capacity gain. In this setting the same symbol weighted by a complex scale factor is sent over each transmit antenna, so that the input covariance matrix has unit rank. This scheme is also referred to as MIMO beam forming.

MIMO systems

Systems with multiple antennas at the transmitter and receiver, which are commonly referred to as multiple-input multiple-output (MIMO) systems. The multiple antennas can be used to increase data rates through multiplexing or to improve performance through diversity.

Precoding.

Pre-coding is generalized to allow multi-layer transmission in MIMO systems. As conventional beamforming considers as linear single layer pre-coding, increasing the signal power at the output of the receiver by emitting the same signal from each of the transmit antennas with suitable weighting.

Spatial multiplexing working

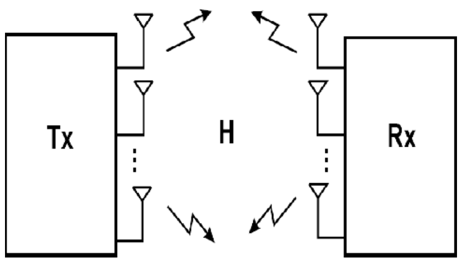
Spatial multiplexing uses MEA's (Multiple element antennas) at the transmitter for transmission of data streams. An original high-rate data stream is multiplexed into several parallel streams, each of which is sent from one transmit antenna element. The channel mixes up these data streams so that each of the receive antenna elements sees a combination of them.

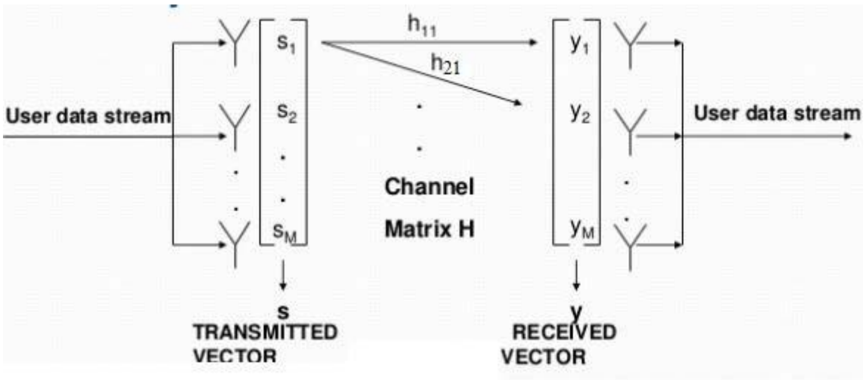
Advantages of MIMO systems.

- Multiple-input multiple-output systems can significantly enhance performance of wireless systems through multiplexing or diversity gain.
- For a given transmit energy per bit, multiplexing gain provides a higher data rate where as diversity gain provides a lower BER in fading.
- Support a higher data rate for a given energy per bit, so it transmits the bits more quickly and can then shut down to save energy.

System model of MIMO.

MIMO systems are systems with *Multiple Element Antennas* (MEAs) at *both* transmitter and receiver. MIMO system offers high data rates and lower error rates.





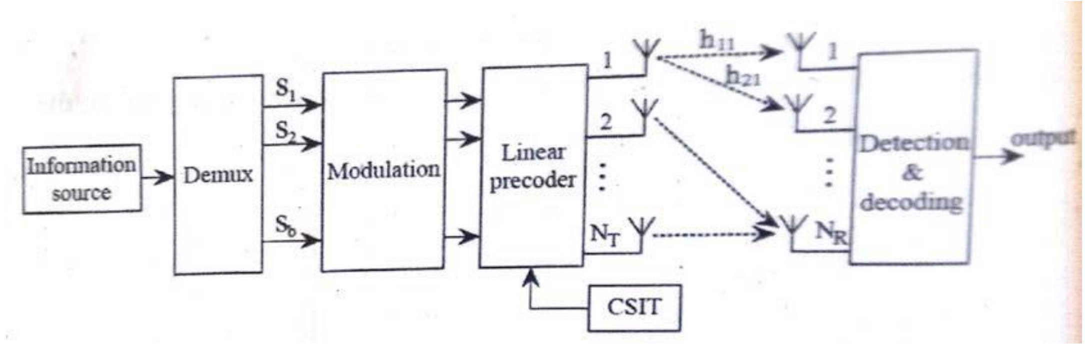
$$\mathbf{y} = \mathbf{H}\mathbf{S} + \mathbf{n}$$

$$\mathbf{H} = \begin{array}{c c} & \overleftrightarrow{\quad M_T \quad} \\ M_R \updownarrow & \begin{bmatrix} h_{11} & h_{12} & \dots\dots & h_{1M} \\ h_{21} & h_{22} & \dots\dots & h_{2M} \\ . & . & \dots\dots & . \\ h_{M1} & h_{M2} & \dots\dots & h_{MM} \end{bmatrix} \end{array}$$

- At the transmitter the data stream enters an encoder whose outputs are forwarded to N_T transmit antennas
- From this antenna, the signal is sent through the wireless propagation channel, which is assumed to be quasi static and frequency flat
- Quasi means coherent time of the channel is so long that is "a large number" of bits can be transmitted within this time.
- MIMO wireless system utilize a matrix mathematical approach to analyse the system.
- Consider the data streams t1,t2,t3 that can be transmitted from antennas 1,2 and 3.
- The channel transfer function h21 represents that the data travelling from transmit antenna to the second receiving antenna.

Precoding

Tx sends coded information to Rx to get prior knowledge about channel.



Vertical Bell Labs layered space-time Architecture

- V-BLAST is a transmitter –receiver which is mainly used to implement multiplexing MIMO
- Achieve diversity of the order of Nr
- Each coded symbol is transmitted from one antenna and received by Nr antennas.
- BLAST can average over the randomness of the individual sub-channels and get better outage performance.

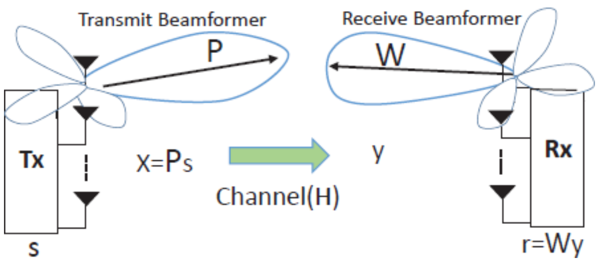
Working Principle

- Eg., you are sending information ‘s’ and it will pass through the channel, ‘h’ and Gaussian noise, ‘n’.
- Then the received signal at the receiver front-end will be,
  $r=sh+n$
- Receiver will have to know the information about ‘h’ and ‘n’.
- It will suppress the effect of ‘n’ by increasing SNR.
- It needs information about the channel ‘h’ and this will increase the complexity.
- The transmitter, base station, will predict the channel.
- The predicted channel ,hest and a system with precoder, information will be coded:(s/hest)
- The received signal will be r=(h/hest)s+n.
- If the prediction is perfect, hest =h and r = s+n
- It turns out to be the detection problem in Gaussian channels

Beamforming.

- Used in any antenna system, particularly in MIMO systems in order to create certain required antenna directive pattern to give required performance.

- Combination of radio signals from a set of small non-directional antennas to simulate a large directional antenna
- Smart antenna are normally used because it can be controlled automatically according to the required performance



- Smart antennas are divided into two types

(i) <u>Phased array systems</u>(PAS)

- ✓ Have number of predefined patterns
- ✓ The required one being switched according to the direction required

(ii) <u>Adaptive array systems</u>(AAS)

- ✓ Uses infinite number of patterns
- ✓ Adjusted based on the requirements in real time

- MIMO beam forming using phased array systems requires the overall system to determine the direction of arrival of the incoming signalswitch in the most appropriate beam
- Adaptive array systems able to direct the beam in the exact direction neededto move the beam in real time

Working principle

- Sends the same signal over each transmit antenna with different scale factor.
- At the receiver , all the received signals are coherently combined using different scale factor.
- Produces a transmit/receiver diversity system whose SNR can be maximized by optimizing the scale factors(MRC).

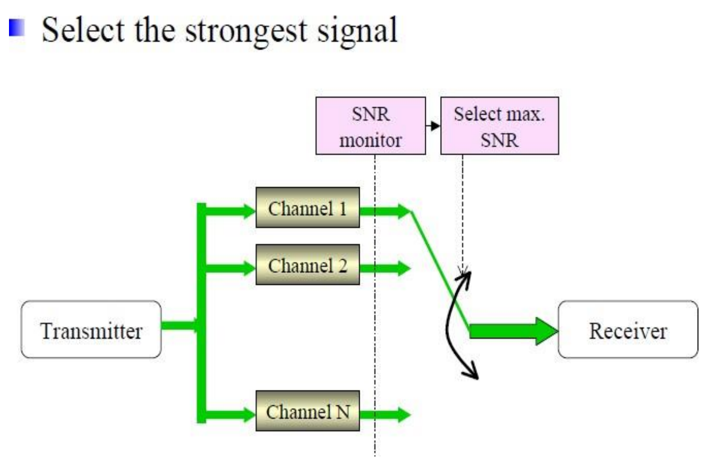
Classification of space Diversity

Classification of Space Diversity

1) Selection diversity

2) Feedback diversity
3) Maximal radio combining
4) Equal gain diversity

Selection Diversity



- The receiver monitors the SNR value of each diversity channel and chooses the one with the maximum SNR value.
- The receiver with M demodulators are used to provide M diversity branches, whose gains are adjusted to provide the same average SNR for each branch
- Receiver branch having the highest instantaneous SNR is connected to the demodulator.

Feedback/scanning Diversity

- Similar to selection diversity except that instead of always using the best of ‘n’ signals, the ‘n’ signals are scanned in a fixed sequence until one is found to be above a predetermined threshold.

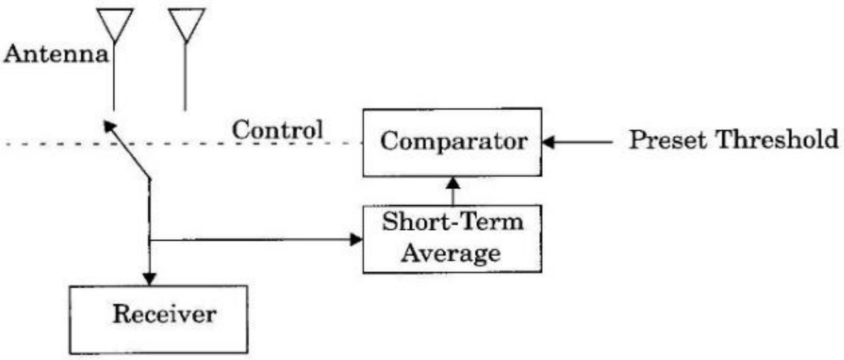


Figure Basic form of scanning diversity.

Maximal Ratio Combining(MRC)

- The signals from all of the M branches are weighted according to their individual signal voltage to noise power ratios and then summed up
- Individual signals must be co-phased before being summed which generally requires an individual receiver and the phasing circuit for each antenna element
- MRC produces an output SNR which is equal to the sum of the individual SNRs.

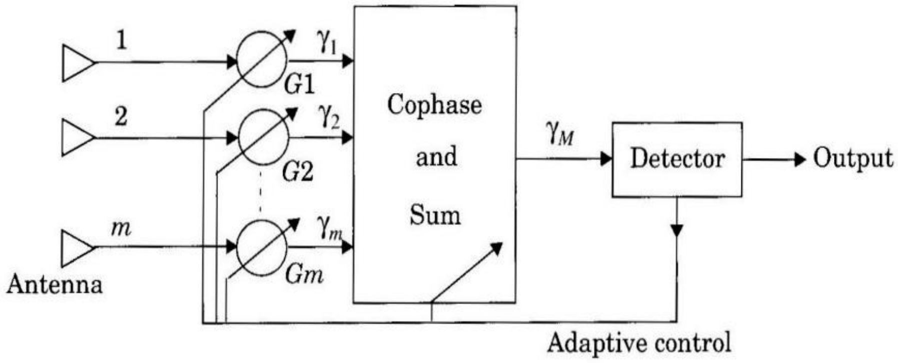


Figure 7.14 Maximal ratio combiner.

- Similar to MRC except that there will be an omission of the weighted circuits
- The branch weights are all set to unity but the signals from each branch are co-phased to provide an equal gain combining diversity
- Allows the receiver to exploit the signals that are simultaneously received on each branch.

Equal Gain Diversity

- Similar to MRC except that there will be an omission of the weighted circuits
- The branch weights are all set to unity but the signals from each branch are co-phased to provide an equal gain combining diversity
- Allows the receiver to exploit the signals that are simultaneously received on each branch.