V V COLLEGE OF ENGINEERING

V V Nagar, Tisaiyanvilai

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

SOLVED UNIVERSITY QUESTIONS BANK

[QUESTIONS & ANSWERS]

Subject Code/Title : **EC8652 / Wireless Communication**

Sem/Class : **III/ECE A&B**

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AP/ECE

V V COLLEGE OF ENGINEERING
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EC 8652 WIRELESS COMMUNICATION
2 MARK QUESTIONS AND ANSWERS

UNIT I WIRELESS CHANNELS

1.Compare Fast and Slow fading (Apr/May 2018)

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

2.Give the Differences Between Frequency flat and Frequency selective fading (Apr/May 2018, Nov/Dec 2017,Nov/Dec 2016)

Flat Fading	Frequency Selective Fading
Bandwidth of signal < Bandwidth of Channel	Bandwidth of signal > Bandwidth of Channel
Delay Spread < Symbol Period	Delay Spread >Symbol Period
Flat fading channels are also known as amplitude uarying channels and are sometimes referred to as narrowband channel	Frequency selective fading channels are a1so known as wideband channel
Flat fading channels cause deep fades	Channel induces intersymbol interference (ISI)

3.What is meant by Multipath propagation(Nov/Dec 2017)

The presence of reflecting objects and scatterers in the channel creates a constantly changing environment that dissipates the signal energy in amplitude, phase, and time. These effects result in multiple versions of the transmitted signal that arrive at the receiving antenna, displaced with respect to one another in time and spatial orientation.

4.What is the major advantages of wireless communication(Apr/May 2017)

(i)Flexibility is the major benefit of wireless communication

(ii) low cost

(iii) Support mobility

(iv) Easy to install

(v) More convenient

5.Define Coherence time . In what way does this parameter decide the behaviour of wireless channel(Apr/May 2017)

Coherence time Tc, is the time domain dual of Doppler spread and is used to characterize the time varying nature of the frequency dispersiveness of the channel in the time domain.

The Doppler spread and coherence time are inversely proportional to one another. That is,

$$T_c = 1/f_m$$

6.Give the equation for average large scale path loss between the transmitter and receiver as a function of distance (Nov/Dec 2016)

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

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

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

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

7. State the difference between small-scale fading and large scale fading. (May2015)(June 2013)

Small-scale fading :

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

Large scale fading:

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

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

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

9.**What is fading and Doppler spread. (May/June 2016)**
Fading takes place in mobile signal propagation due to multi path time delay spread .Doppler spread is denoted as BD and it is defined as a set of frequencies over which the Doppler spread at the receiver end is non zero value.For example if a pure sinusoidal tone of frequencyis transmitted and is denoted as fc and the received signal spectrum is called Doppler spectrum consisting of components in the range from fc-fd to fc+fd, in which fd refers to Doppler shift in frequency .

10.Find the far- field distance for an antenna with maximum dimension of 2m and operating frequency of 1GHz.(Nov/Dec 2015)
Solution:
Operating frequency f= 1GHz
Large dimension of antenna D= 2m
Operating wavelength λ=c/f= 0.3m
Far field distance df = $2D^2/\lambda$ =8/0.3= 26.67 m

11. What is Doppler Spread and Coherence Bandwidth(Nov/Dec 2015)
Doppler Spread:
Doppler spread B_D is a measure of the spectral broadening caused by the time rate of change of the mobile radio channel and is defined as the range of frequencies over which the received Doppler spectrum is essentially non-zero.
Coherence Bandwidth:
The coherence bandwidth,B_c is a defined relation derived from the rms delay spread. Coherence bandwidth is a statistical measure of the range of frequencies over which the channel can be considered "flat”.
If the coherence bandwidth is defined as the bandwidth over which the frequency correlation function is above 0.9, then the coherence bandwidth is approximately

12.Define EIRP
The effective isotropic radiated power (EIRP) is defined as

$$EIRP = P_t G_t$$

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

13.Define path loss
The path loss, which represents signal attenuation as a positive quantity measured in dB, is defined as the difference (in dB) between the effective transmitted power and the received power, and may or may not include the effect of the antenna gains.

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

14.What is Fraunhofer region

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

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

15.Define Brewster Angle The Brewster angle is the angle at which no reflection occurs in the medium of origin.

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

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

16.Define Mean excess delay, RMS delay spread, Excess delay Spread(Nov/Dec 2015)

Mean excess delay:

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

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

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

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

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

Excess delay Spread:

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

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

17.What are the effects of Small Scale fading

The three most important effects are:

• Rapid changes in signal strength over a small travel distance or time interval

• Random frequency modulation due to varying Doppler shifts on different multipath signals

• Time dispersion (echoes) caused by multipath propagation delays.

18.Define Doppler Shift

Due to the relative motion between the mobile and the base station, each multipath wave experiences an apparent shift in frequency. The shift in received signal frequency due to motion is called the Doppler shift, and is directly proportional to the velocity and direction of motion of the mobile with respect to the direction of arrival of the received multipath wave.

$$f_d = \frac{1}{2\pi} \cdot \frac{\Delta\phi}{\Delta t} = \frac{v}{\lambda} \cdot \cos\theta$$

19.What is the necessity of link budget?

The necessities of link budget are:

i. A link budget is the clearest and most intuitive way of computing the required Transmitter power. It tabulates all equations that connect the Transmitter power to the received SNR
ii. It is reliable for communications.
iii. It is used to ensure the sufficient receiver power is available.
iv. To meet the SNR requirement link budget is calculated.

20. What are the factors influencing small scale fading?

Factors influencing small scale fading are

1. Speed of surrounding objects
2. Multipath propagation
3. Speed of the mobile
4.Transmission bandwidth of the signal.

UNIT II CELLULAR ARCHITECTURE

1.Why is cellular concept used for mobile telephony?(April/May 2017)
Base stations provide the cell with the network coverage which can be used for transmission of voice, data and others. A cell typically uses a different set of frequencies from neighboring cells, to avoid interference and provide guaranteed service quality within each cell.

2.In a cellular network, among a handoff call and a new call, which one is given priority ?why?(April/May 2017)
If the call is terminated abruptly in the middle of conversation then it is more annoying than the new originating call being blocked. So in order to avoid this abrupt termination of ongoing call handoff request should be given priority to new call. This is called as handoff prioritization.

3.What do you mean by forward and reverse channel (Nov/Dec 2017)
Forward channel is a radio channel used for transmission of information from base station to mobile. reverse channel used for transmission from mobile to base station.

4.What is Hard and soft Handoff in mobile communication(May/June 2016)

When the person is Moving from one BS to other without interrupting connection is called **hand off**.

Hard handoff- when the user moves to a new cell, he will be assigned with a new set of channels.

Soft Handoff- when the user moves to a new cell, the channel itself will be switched to the new base station. CDMA uses soft Handoff.

5.What is Multiple access techniques(May/June 2016)

The available spectrum bandwidth for our wireless communication is limited. Multiple access techniques enable multiple signals to occupy a single communications channel.

Major Types,

- **Frequency division multiple access (FDMA)**
- **Time division multiple access (TDMA)**
- **Code division multiple access (CDMA)**

6. What are the effects of multi path propagation on CDMA? (April/May 2015,April/May 2016,Nov/Dec 2014)
Reflection - occurs when signal encounters a surface that is large relative to the wavelength of the signal.
Diffraction - occurs at the edge of an impenetrable body that is large compared to wavelength of radio wave.

7. State advantages of CDMA over FDMA? (Dec 2014,Nov/Dec 2016)

FDMA:

- Channel bandwidth is subdivided into a number of sub channels
- Narrow band system
- Commonly used for voice and data transmission
- Code word is not required
- Hard handoff

CDMA:

- The sharing of both bandwidth and time takes place
- wide band system
- Commonly used for digital voice signals and multimedia services
- Code words are required
- soft handoff

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

Cell splitting, Sectoring, Coverage Zone approaches are the techniques used to expand the capacity of cellular system.

Cell splitting – Cell-splitting is a technique which has the capability to add new smaller cells in specific areas of the system. i.e. divide large cell size into small size.

Sectoring – use of directional antennas to reduce Co-channel interference.

Coverage Zone approaches –large central BS is replaced by several low power transmitters on the edge of the cell.

9. Define Grade of service(GOS). (Nov/Dec 2015, Nov/Dec 2016)

It is a measure of the ability of a user to access a trunked system during the busiest hour. The busy hour is based upon customer demand at the busiest hour during a week, month, or year.

10. Define co-channel reuse ratio (Nov/Dec 2015).

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

$$Q=D/R \text{ co-channel reuse ratio(Q)}$$

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

11.What is meant by frequency reuse?(April/ May 2018, Nov/Dec 2017,May /June 2016)

If an area is served by a single Base Station, then the available spectrum can be divided into N frequency channels that can serve N users simultaneously. If more than N users are to be served, multiple BSs are required, and frequency channels have to be reused in different locations. Since spectrum is limited, the same spectrum has to be used for different wireless connections in different locations. This method of reusing the frequency is called as **frequency reuse.**

12. Write the nonlinear effects in FDMA

In a FDMA system, many channels share the same antenna at the base station. The power amplifiers or the power combiners, when operated at or near saturation for maximum power efficiency, are nonlinear. The nonlinearities cause signal spreading in the frequency domain and generate intermodulation (IM) frequencies. IM is undesired RF radiation which can interfere with other channels in the FDMA systems, Spreading of the spectrum results in adjacent-channel interference. Intermodulation is the generation of undesirable harmonics. Harmonics generated outside the mobile radio band cause interference to adjacent services, while those present inside the band cause interference to other users in the mobile system

13. What is channel assignment? What are the types?

channel assignment :

For efficient utilization of radio spectrum a frequency reuse scheme with increasing capacity and minimizing interference is required. For this channel assignment is used.

Types: Fixed channel assignment, dynamic channel assignment.

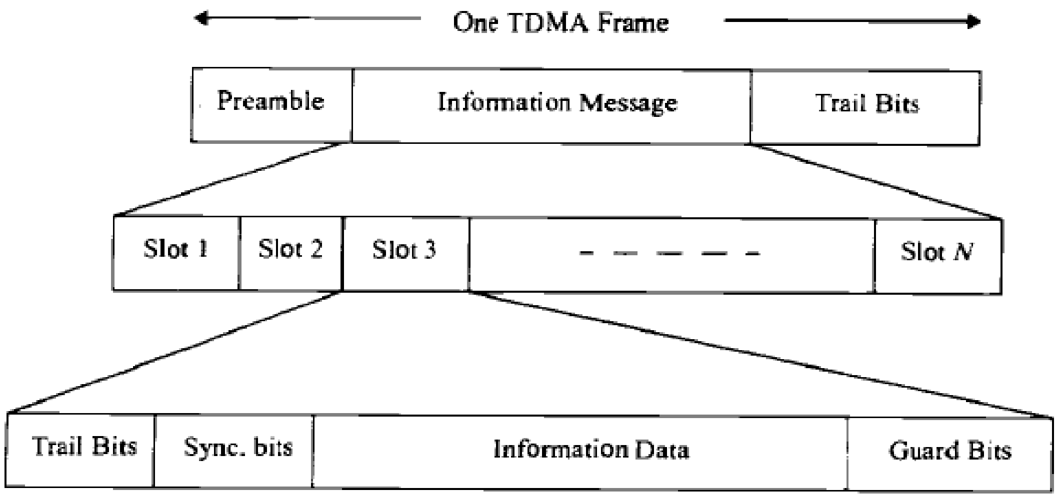
Fixed channel assignment

If the channels in each cell is allocated to the users within the cell, it will be called as fixed channel assignment. If all channels are occupied, the call Will be blocked.

Dynamic channel assignment

If the voice channels are not allocated permanently in a cell, it will be called as dynamic channel assignment. In this assignment, channels are dynamically allocated to users by the MSC.

14.Draw the TDMA frame Structure



15.Define Near far problem in CDMA

The near-far problem occurs when many mobile users share the same channel. In general, the strongest received mobile signal will capture the demodulator at a base station. In CDMA, stronger received signal levels raise the noise floor at the base station demodulators for the weaker signals, thereby decreasing the probability that weaker signals will be received. lb combat the near-far problem, power control is used in most CDMA implementations.

16.Compare Different Multiple Access Schemes(April/May 2018)

	station equipment, space and maintenance • No frequency guard band required. • Easy for mobile or base stations to initiate and execute handoffs.	filtering and correlation detection. • Demands high peak power on uplink in transient mode • Requires network wide timing synchronization.
CDMA	No absolute limit on the number of users • Easy addition of more users • Better signal quality • Impossible for the hackers to decipher the code sent. • Multipath fading may be substantially reduced because of large signal bandwidth.	Near-far problem arises Self-jamming. As the number of users increases, the overall quality of service decreases.

17. Define dwell time.

The time over which the call may be maintained within a cell without handoff is called as dwell time. This time is governed by factors such as propagation, interference, distance between subscribers and base station.

18.What is MAHO

Mobile assisted handoff (MAHO) allows subscribers to monitor the neighboring base stations, and the best base station choice may be made by each subscriber. MAHO allows the deployment of densely packed microcells, thus giving substantial capacity gains in a system.

19.Define Co channel Interference (Nov/Dec 2015)

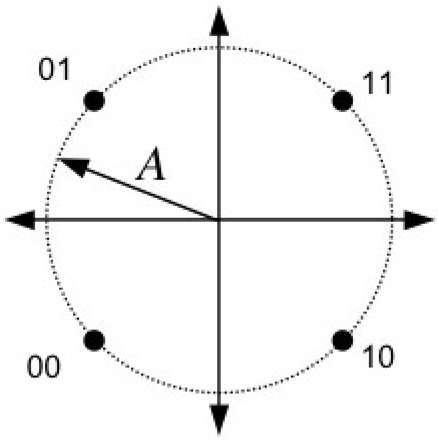
Frequency reuse implies that in a given coverage area there are several cells that use the same set of frequencies. These cells are called co-channel cells, and the interference between signals from these cells is called co-channel interference

UNIT III DIGITAL SIGNALING FOR FADING CHANNELS

1.What do you mean by cyclic prefix(April/ May 2018,Nov/Dec 2016)

In delay dispersive channel, inter carrier interference occur. To overcome the effect of inter carrier interference and ISI, cyclic prefix is introduced. It is a cyclically extended guard interval whereby each symbol sequence is preceded by a periodic extension of the sequence itself.

2.Draw the constellation diagram for offset QPSK modulation scheme(April/ May 2018)



3.Why is MSK referred to as fast FSK?(May/ June 2016)

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

4.What is windowing?(May/ June 2016)

Windowing is a technique proposed to help reduce sensitivity to frequency offsets in an OFDM system.This process involves cyclically extending the time domain signal with each symbol by ‘v’ samples.The resulting signal is then shaped with a window function.

5.Give the function of Gaussian filter in GMSK(Nov/Dec 2016)

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

6.What is the basic advantage of using Multicarrier schemes such as OFDM?(April/May 2017)

- Makes efficient of the spectrum by allowing overlap
- Eliminates ISI through use of cyclic prefix
- By dividing the channel into narrowband flat fading channel.OFDM is more resistant to frequency selective fading than single carrier system
- Channel equalization becomes simpler than by using adaptive equalization technique with single carrier systems

7.State any two advantage of MSK.(April/May 2017)

- Constant envelope
- Spectral efficiency high
- Good BER
- Self-synchronizing capability

8.Define PAPR.(Nov/Dec 2017)

The peak-to-average power ratio (PAPR) is the peak amplitude squared (giving the peak power) Divided by the RMS value squared (giving the average power).

9.Define Offset QPSK and π/4 QPSK(Nov/Dec 2017)

Offset QPSK

In QPSK phase shift of π radians will cause the amplitude to fluctuations. This will lead to generation of side lobes and spectral widening. To reduce this 180o phase shift we use O-QPSK. In this we have only 90o shifts. This is achieved by delaying one channel by Tb sec.

π/4 QPSK

π/4 shifted QPSK modulation is a quadrature phase shift keying technique which offers a compromise between conventional QPSK and OQPSK in terms of the maximum phase variations. The maximum phase change is limited to ±135°, as compared to 180° for QPSK and 90° for OQPSK. Hence, the bandlimited p/4 QPSK signal preserves the constant envelope property better than bandlimited QPSK, but is more susceptible to envelope variations than OQPSK

10.Find the 3-DB bandwidth for a Gaussian low pass filter used to produce 0.25 GMSK with a channel data rate of R_b = 270 kbps. What is the 90% power bandwidth in RF channel?(Nov/Dec 2015)

Solution

T= 1/Rb=1/270*10^3 =3.7us

BT=0.25

B=0.25/T= 67.567kHz

To determine the 90% power bandwidth, 0.57 Rb is the desired value

RF BW= 0.57 Rb = 0.57* 270*10^3 = 153.9 kHz

11.List the advantages of digital modulation techniques.

The advantages of digital modulation techniques are:

i. Immunity to channel noise and external interference.
ii. Flexibility operation of the system.
iii. Security of information.
iv. Reliable since digital circuits are used.
v. Multiplexing of various sources of information into a common format is possible.
vi. Error detection and correction is easy.

12. Why GMSK is preferred for multiuser, cellular communication?

It is a simple binary modulation scheme. Premodulation is done by Gaussian pulse shaping filter, so side lobe levels are much reduced. GMSK has excellent power efficiency and spectral efficiency than FSK. For the above reasons GMSK is preferred for multiuser, cellular communication.

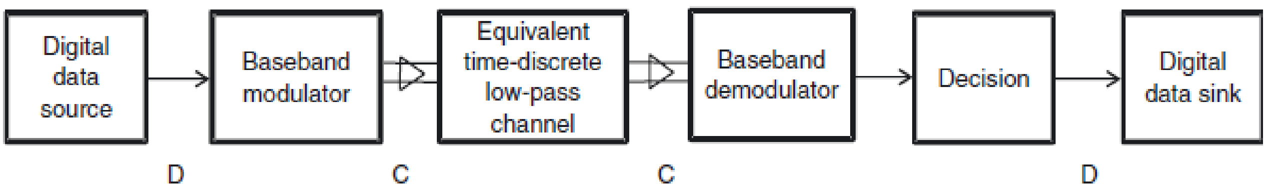
13. Give the comparison between MSK and QPSK

MSK	QPSK
Bandwidth is 1.5fb	Bandwidth is fb
MSK signals have continious phase in all cases	QPSK signals have abrupt amplitude variations
The Main lobe of MSK is wider than that of QPSK and contains around 99% of signal energy	QPSK Main lobe contains around 90% of signal energy
Side lobes of MSK are smaller compared to QPSK	Side lobes are bigger.So inter channel interference due to side lobes is significantly large
Band pass filtering is not required to avoid the ICI	To avoid ICI problems,QPSK needs band pass filtering

14.State the Principle of OFDM

OFDM splits a high-rate data stream into N parallel streams, which are then transmitted by modulating N distinct carriers (henceforth called subcarriers or tones). Symbol duration on each subcarrier thus becomes larger by a factor of N. In order for the receiver to be able to separate signals carried by different subcarriers, they have to be orthogonal.

15.Draw Mathematical link model for the analysis of modulation formats.



UNIT IV MULTIPATH MITIGATION TECHNIQUES

1.List different types of Diversity schemes(Apr/May 2018)

- **Space diversity**
  1.Selection diversity
  2. Feedback diversity
  3. Maximal ratio combining
  4. Equal gain diversity
- Polarization Diversity
- Frequency Diversity
- Time Diversity

2.Why is an adaptive equalizer required?(Apr/May 2017,May/June 2016)

To combat ISI, the equalizer coefficients should change according to the channel status so as to track the channel variations. Such an equalizer is called an adaptive equalizer since it adapts to the channel variations.

3.What is diversity?Why it is employed?(Apr/May 2017)

Diversity is used to compensate for 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.

4.What are linear equalizers and non linear equalizers(Nov/Dec 2016)

If the output is not used in the feedback path to adapt, then this type of equalizer is called linear equalizer. If the output is fed back to change the subsequent outputs of the equalizer, this type of equalizer is called nonlinear equalizers.

5.What is Macro Diversity(Nov/Dec 2016)

Large-scale fading is caused by shadowing due to variations in both the terrain profile and the nature of the surroundings. By selecting a base station which is not shadowed when others are, the mobile can improve substantially the average ratio on the forward link. This is called **macroscopic diversity.**

6.Define spatial diversity(Nov/Dec 2017)

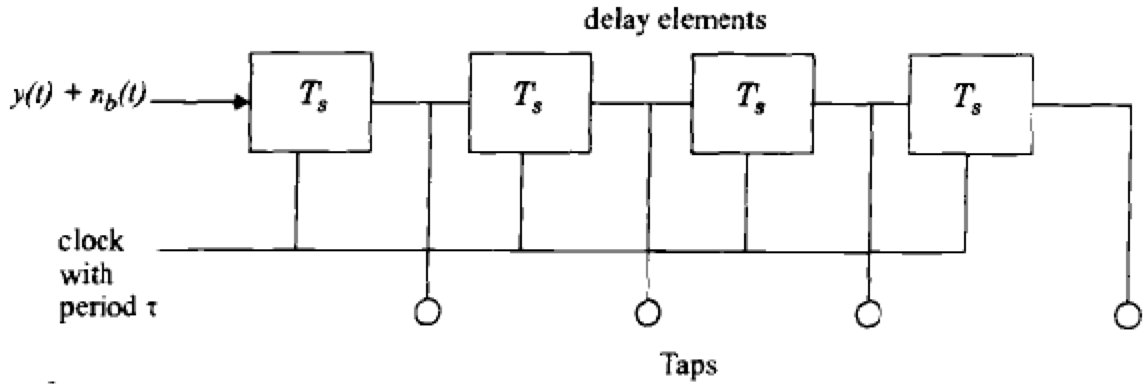
The most common diversity technique is called spatial diversity, whereby multiple antennas are strategically spaced and connected to a common receiving system. While one antenna sees a signal null, one of the other antennas may see a signal peak, and the receiver is able to select the antenna with the best signals at any time.

7.Define STCM(Nov/Dec 2017)

Channel coding can also be combined with diversity a technique called Space Time Coded Modulation, the space-time coding is a bandwidth and power efficient method of wireless.

8.What are the factors used in adaptive algorithms?(Nov/Dec 2015)
1. Rate of convergence
2. Mis adjustments
3. Computational complexity

9.Draw the structure of a linear transversal equalizer(Nov/Dec 2015)



10.What is the benefit of Rake receiver(May/June 2016)

- Rake receiver is designed to counter the effects of multipath fading.
- It is mainly used in reception of CDMA signals.
- CDMA codes are designed to provide very low correlation between successive chips, so they appear like uncorrelated noise at a CDMA receiver.
- The chip rate of a code is the number of pulses per second .

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

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

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

12. Write the basic principle of DFE

The basic idea behind decision feedback equalization is that once an information symbol has been detected and decided upon, the 1ST that it induces on future symbols can be estimated and subtracted out before detection of subsequent symbols

13. Why non linear equalizers are preferred?

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

14. Write the advantages of LMS algorithm.

1. The LMS equalizer maximizes the signal to distortion at its output within the constraints of the equalizer filter length.
2. Low computational complexity
3. Simple program

15. Define training mode in an adaptive equalizer?

First, a known fixed length training sequence is sent by the transmitter then the receivers equalizers may adapt to a proper setting of minimum bit error detection where the training sequence is a pseudo random binary signal or a fixed and prescribed bit pattern.

16. What is tracking mode in an adaptive equalizer?

Immediately following this training sequence the user data is sent and the adaptive equalizer at the receiver utilizes a recursive algorithm to evaluate the channel and estimate filter coefficients to compensate for the distortion created by multipath in the channel.

17.Differentiate micro and macro diversity.

Micro diversity	Macro diversity
Used to reduce small scale fading effects.	Used to reduce large scale fading effects.
Multiple reflection causes deep fading. This effect is reduced.	Deep shadow causes fading. This effect is reduced.
BS-MS are separated by small distance.	BS-MS are separated by large distance.

18. Define Frequency Diversity

In frequency diversity, the same signal is transmitted at two (or more) different frequencies. If these frequencies are spaced apart by more than the coherence bandwidth of the channel, then their fading is approximately independent, and the probability is low that the signal is in a deep fade at both frequencies simultaneously.

UNIT V MULTIPLE ANTENNA TECHNIQUES

1.Distinguish between diversity Gain versus Array Gain(April/May 2018)

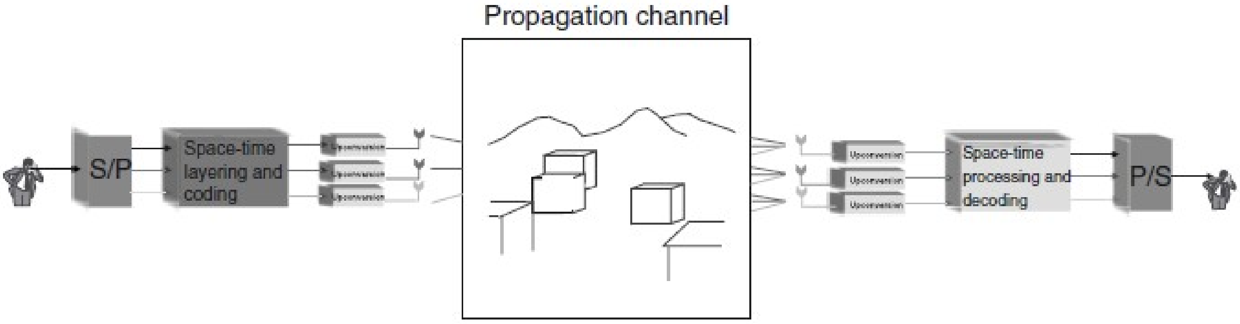
Array gain. In MIMO communication systems, **array gain** means a power **gain** of transmitted signals that is achieved by using multiple-antennas at transmitter and/or receiver, with respect to single-input single-output case. The **diversity gain** is dependent on spatial correlation coefficients between antenna signals.

2.What is spatial multiplexing(April/May 2017,Nov/Dec 2016,Nov/Dec 2017,Nov/Dec 2015)

spatial multiplexing -transmission of several data streams in parallel.It allows direct improvement of capacity by simultaneous transmission of multiple data streams.

How Does Spatial Multiplexing Work?

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



3.What is channel state information? What is its benefit(April/May 2017,Nov/Dec 2015)

Channel state information refers to the known channel properties of communication link.it describes how a signal propagates from the transmitter to the receiver and represents the combined effect of scattering ,fading and power delay with distance.

4.What is Ergodic capacity and outage capacity of a flat fading channel(Nov/Dec 2016)

Ergodic (Shannon) capacity**:** this is the expected value of the capacity, taken over all realizations of the channel. This quantity assumes an infinitely long code that extends over all the different channel realizations.

Outage capacity**:** this is the minimum transmission rate that is achieved over a certain fraction of the time – e.g., 90% or 95%.

5.Define Receiver Diversity(Nov/Dec 2017)

In receiver Diversity,one transmitting antenna and many receiving antennas are used.here the desired message is transmitted by using single transmitting antenna and received by multiple antennas.N_R different antennas appropriately seperated are deployed at the receiver to combine the uncorrelated fading signals.It is also called

Space Diversity.

6.What is MIMO Systems(May/June 2016)

MIMO systems are systems with *Multiple Element Antennas* (MEAs) at *both* link ends.The MEAs of a MIMO system can be used for four different purposes: (i) beamforming, (ii) diversity, (iii) interference suppression, and (iv) spatial multiplexing (transmission of several data streams in parallel).

7.What is transmit Diversity(May/June 2016, Nov/Dec 2015)

For many situations, multiple antennas can be installed at just one link end (usually the BS). For the uplink transmission from the MS to BS, multiple antennas can act as receive diversity branches. For the downlink, any possible diversity originates at the transmitter.

The two main types of transmit Diversity are,

- ***Transmitter Diversity with Channel State Information***
- ***Transmitter Diversity Without Channel State Information***

8.State the Shannon's Channel capacity theorem

Shannon's Channel capacity law defines,"the maximum rate at which error free data can be transmitted over a given bandwidth in the presence of noise".

$C=W \log_2(1+SNR)$

Where C-Channel Capacity(bps),

W-Bandwidth(Hz),

SNR-Signal to noise ratio

9.Mention the advantages of MIMO

Advantages of MIMO are,

i)Higher channel capacity

ii)Better Spectral efficiency

iii)Increased coverage

iv)Improved user position estimation

v)lower power consumption

vi)Minimize the errors

vii)Faster speeds

viii)Higher data rate

The main disadvantage is that it is complex.

10.List the applications of MIMO

The applications of MIMO are,

i)MIMO is currently being used within the telecommunications and networking industries,that is cellular,WMAN,WWAN and so forth.

ii)MIMO is used largely in cellular towers

iii)It is used in modern wireless standards,including in 3GPP LTE,and mobile Wimax systems

iv)MIMO-OFDM is considered a key technology in emerging high data rate systems such as 4G,IEEE 802.16 and IEEE 802.11n.

11.Define Precoding

Precoding is a technique which exploits transmit diversity by weighting information stream.The transmitter send the coded information to the receiver in order to the pre-knowledge of the channel. The receiver is a simple detector, such as a matched filter, and does not have to know the channel side information. This technique will reduce the corrupted effect of the communication channel.

12.What is joint Leakage Suppression

In joint leakage suppression, the precoding matrices are designed to maximize the ratio of the power of the desired signal received by the *k*-th MS and the sum of the noise and the total interference power (leakage) due to the transmit signal intended for the *k*-th user at all the other MSs (note that this is different from the SINR at the *k*-th MS). The key motivation for this criterion is that it allows an easier optimization (often in closed form) than the SINR at the RX.

13.Give the simple MIMO system model Equation

The received signal vector

$$\mathbf{r} = \mathbf{H}\mathbf{s} + \mathbf{n} = \mathbf{x} + \mathbf{n}$$

*contains the signals received by N*r antenna elements, where **s** is the transmit signal vector and **n** is the noise vector.

14.Differentiate Linear and Non linear Precoding

The capacity achieving algorithms are non linear,But Linear precoding approaches usually achieve a reasonable performance with much lower complexity.Linear precoding strategies include maximum Ratio Transmission,Zero Forcing Precoding and Transmit Wiener Precoding.

The nonlinear precoding is designed based on the concept of Dirty paper coding(DPC),Which shows that any known interference at the transmitter can usually be subtracted without the penalty of radio resources if the optimal precoding scheme can be applied on the transmit signal.

15.What is Beamforming

Beamforming techniques can be used with any antenna system - not just on MIMO systems. They are used to create a certain required antenna directive pattern to give the required performance under the given conditions.

16.Mention the types of Beamforming in MIMO system

Smart antennas are normally used - these are antennas that can be controlled automatically according the required performance and the prevailing conditions.

Smart antennas can be divided into two groups:

- ***Phased array systems:***Phased array systems are switched and have a number of pre-defined patterns - the required one being switched according to the direction required.

- ***Adaptive array systems (AAS):*** This type of antenna uses what is termed adaptive beamforming and it has an infinite number of patterns and can be adjusted to the requirements in real time.

17.list the Advantages of Beamforming

The main advantages of Beamforming are,

i)Increase signal-to-interference-plus-noise ratio (**SINR**)

ii)Support higher user Densities

18.Explain MRT

The optimum transmission scheme linearly weights signals transmitted 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*.

PART -B & C
UNIT I WIRELESS CHANNELS

1. (a)(i)If a transmitter produces 50 watts of power, express the transmit power in units of (a) dBm, and (b) dBW. If 50 watts is applied to a unity gain antenna with a 900 MHz carrier frequency, find the received power in dBm at a free space distance of 100m from the antenna. What is received power at a distance of 10km? Assume unity gain for the receiver antenna.(Apr/ May 2017)

Solution:

P_t (dbm) = 10log [P_t(W)/1mW]
=47dbm
P_t (dbW) = 10log [P_t(W)/1W]
=17dbW

$$P_r(d=100\,m)=\frac{P_t G_t G_r \lambda^2}{(4\pi)^2 d^2 L}=\frac{50\times1\times1\times\left(\frac{1}{3}\right)^2}{(4\pi)^2\times100^2\times1}=3.5\times(10)^{-6}W$$

$$P_r(dbm)=-24\,dbm$$

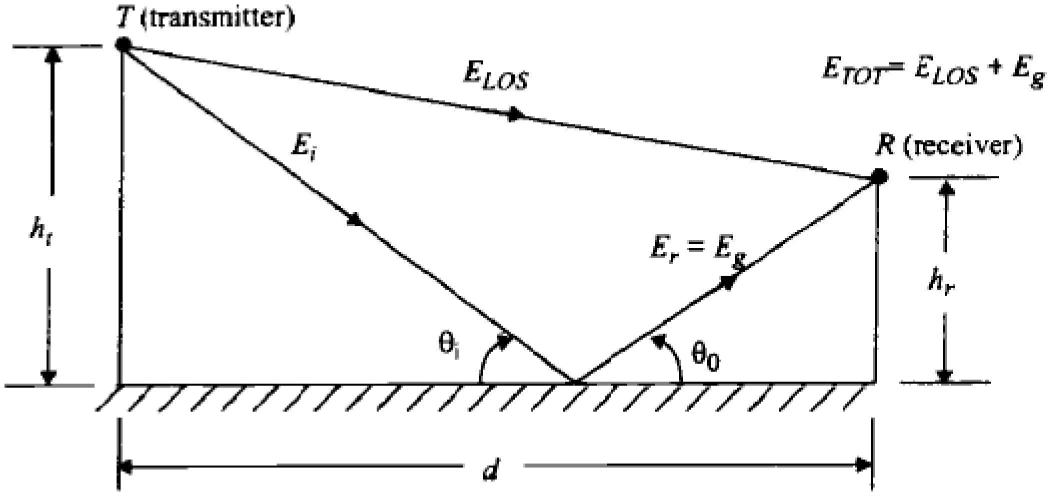
$$Here\ \ \lambda=\frac{C}{f}=\frac{3\times10^8}{900\times10^6}=\frac{1}{3}\,m$$

$$P_r(d=10\,m)=\frac{P_t G_t G_r \lambda^2}{(4\pi)^2 d^2 L}=\frac{50\times1\times1\times\left(\frac{1}{3}\right)^2}{(4\pi)^2\times10^2\times1}$$

$$P_r(dbm)=-64.52\,dbm$$

(ii)Derive the path loss model considering a two ray Model for the Propagation mechanism in a wireless channel. Is considering just two rays alone sufficient. Why?(Apr/ May 2017, Nov/Dec 2016, Apr/May 2018)

- Two Ray Ground Reflection model considers both the direct path and a ground reflected propagation path between transmitter and receiver.
- The total received E-field, ETOT, is then a result of the direct line-of-sight component, ELOS , and the ground reflected component, E_g.
- h_t is the height of the transmitter
- hr is the height of the receiver.



- Fig: Two Ray Ground Reflection modelIf E0 is the free space E-field (in units of V/rn) at a reference distance d_0 from the transmitter, then for d > d_0, the free space propagating E-field is given by

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

- Where |E (d, t)| = E0d0/d represents the envelope of the E-field at d meters from the transmitter.
- Two propagating waves arrive at the receiver: the direct wave that travels a distance d'; and the reflected wave that travels a distance d'.
- The E-field due to the line-of-sight component at the receiver can be expressed as

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

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

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

According to laws of reflection in dielectrics

$$\theta_i = \theta_0 \qquad \begin{aligned} E_g &= \Gamma E_i \\ E_t &= (1+\Gamma)E_i \end{aligned}$$

- Where Γ is the reflection coefficient for ground.

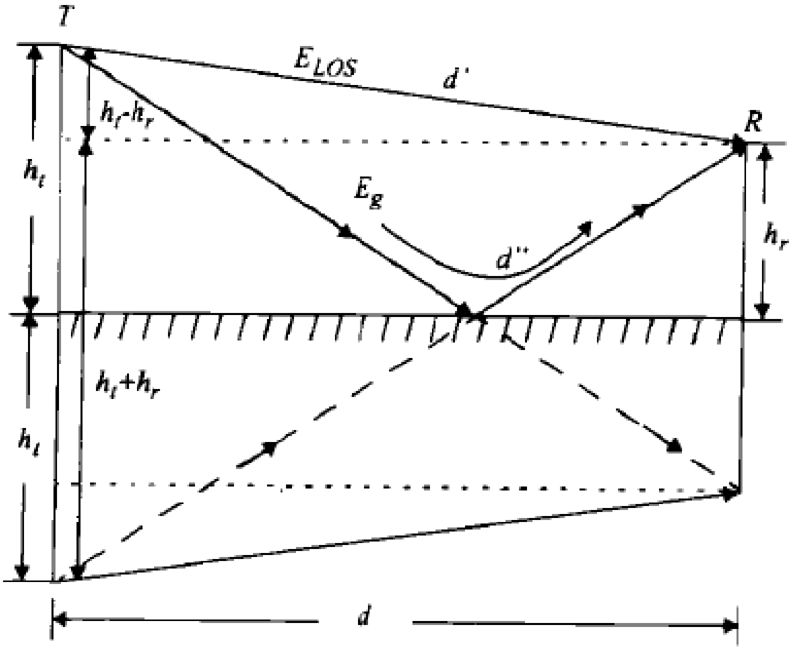


Fig: The method of images is used to find the path difference between the line-of-sight and the ground reflected paths.

- The resultant E-field, assuming perfect ground reflection (i.e., **Γ** = —1 and Et= 0) is the vector sum of ELOS and Eg and the resultant total E-field envelope is given by

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

- The electric field ETOT(d, t) can be expressed as the sum of equations

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

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

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

- When the T-R separation distance d is very large compared to ht+hr equation can be simplified using a Taylor series approximation

$$\Delta = d'' - d' \approx \frac{2h_t h_r}{d}$$

- Once the path difference is known, the phase difference Δθ between the two E field components and the time delay τ_d between the arrival of the two components can be easily computed using the following relations

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

$$\tau_d = \frac{\Delta}{c} = \frac{\theta_\Delta}{2\pi f_c}$$

- It should be noted that as d becomes large, the difference between the distances d' and d becomes very small, and the amplitudes of ELOS and E_g are virtually identical and differ only in phase.
- If the received E-field is evaluated at some time, say at t = d"/c,

$$E_{TOT}\left(d, t = \frac{d''}{c}\right) = \frac{E_0 d_0}{d'}\cos\left(\omega_c\left(\frac{d''-d'}{c}\right)\right) - \frac{E_0 d_0}{d''}\cos 0^o$$
$$= \frac{E_0 d_0}{d'}\cos\theta_\Delta - \frac{E_0 d_0}{d''}$$
$$\approx \frac{E_0 d_0}{d}[\cos\theta_\Delta - 1]$$

- Where d is the distance over a flat earth between the bases of the transmitter and receiver antennas. Referring to the phasor diagram of Figure which shows how the direct and ground reflected rays combine, the electric field (at the receiver) at a distance d from the transmitter can be written as

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

or

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

- Using Trigonometric identities,

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

- The received E field can be approximated as,

$$E_{TOT}(d) \approx \frac{2E_0 d_0}{d}\frac{2\pi h_t h_r}{\lambda d} \approx \frac{k}{d^2}\ \mathrm{V/m}$$

- Where k is a constant related to E0, the antenna heights, and the wavelength.
- The received power at a distance d from the transmitter can be expressed as

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

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

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

1.(b)(i)Determine the proper spatial sampling interval required to make a small-scale propagation measurements which assume that consecutive samples are highly correlated in time. How many samples will be required over 10 m travel distance if f_c= 1900 MHz and v= 50m/s. How long would it take to make these measurements, assuming they could be made in real time from a moving vehicle? What is the Doppler spread BD for the channel?(Apr/ May 2017)

Solution:

F_C= 1900 Mhz , V= 50 m/s , D= 10 m

F_m= V/λ

$T_C = 9\lambda / 16\pi v$

T_C = 565.7 µs

Taking Time samples at less half T_C

$\Delta x = VT_C /2$

$=50 x 565.7 x 10^{-6} / 2 = 14.14$ cm

No. of samples over a 10 m travel distance

Nx =10/Δx = 10/ 14.14 cm = 707 samples.

Time Taken for measurement = 10 m / 50m/s = 0.2 Seconds

(ii) Describe in detail, the parameters of mobile multipath channels with their significance.(Apr/ May 2017)

Parameters of Mobile Multipath Channels

- Time Dispersion Parameters
  - Mean excess delay(τ)
  - RMS delay spread (σ_τ)
  - Excess delay spread
- Coherence Bandwidth (Bc)
- Doppler spectrum
- Coherence Time (Tc)

Mean excess delay(τ):

- The mean excess delay (τ) is the first moment of the power delay profile, this is the expected value of the signal.

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

RMS delay spread (σ_τ):

- The RMS delay spread is the square root of the second central moment of the power delay profile

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

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

Excess delay spread or maximum *excess delay* (X dB):

- The maximum excess delay (X dB) of the power delay profile is defined to be the time delay during which multipath energy falls to X dB below the maximum.
- (i.e) The maximum excess delay is defined as Tx- T_0 , where t_0 is the first arriving signal and Tx is the maximum delay at which a multipath component is within X dB of the strongest arriving multipath signal.

Coherence Bandwidth (B_C)

- Coherence bandwidth is a statistical measure of the range of frequencies over which the channel can be considered "flat“.
- “Flat” refers to a channel which passes all spectral components with approximately equal gain and linear phase.
- In other words, coherence bandwidth is the range of frequencies over which two frequency components have a strong amplitude correlation.
- Mathematically coherence bandwidth is inversely proportional to RMS delay spread.

$$B_C \alpha \; 1 / \sigma_\tau$$

Doppler spread and Coherence Time

- When a pure sinusoidal signal of frequency fc is transmitted, the received signal spectrum will have components in the range f_c - f_d to f_c + f_d. This called the **Doppler spectrum.**
- wheref_d is the Doppler shift.
- f_d is a function of the relative velocity of the mobile, and the angle between the direction of motion of the mobile and direction of arrival of the scattered

waves.

Coherence Time (T_C):

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

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

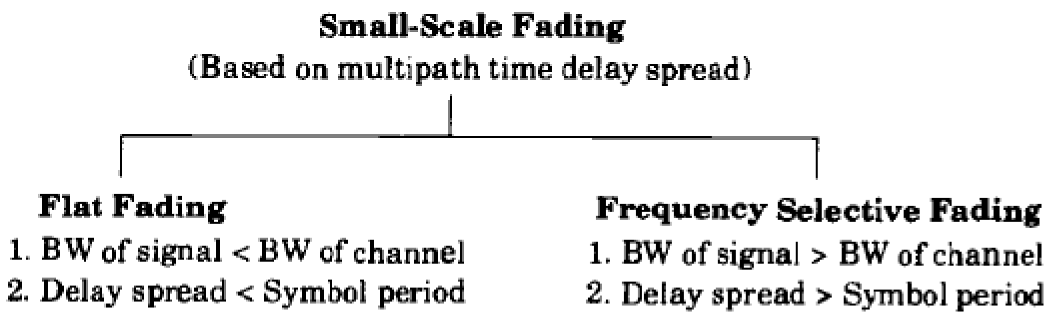
(iii) Compare and contrast fast fadi w fading.” In practice fast fading only occurs for very low data rate (:ations)”.Why?(Apr/ May 2017)

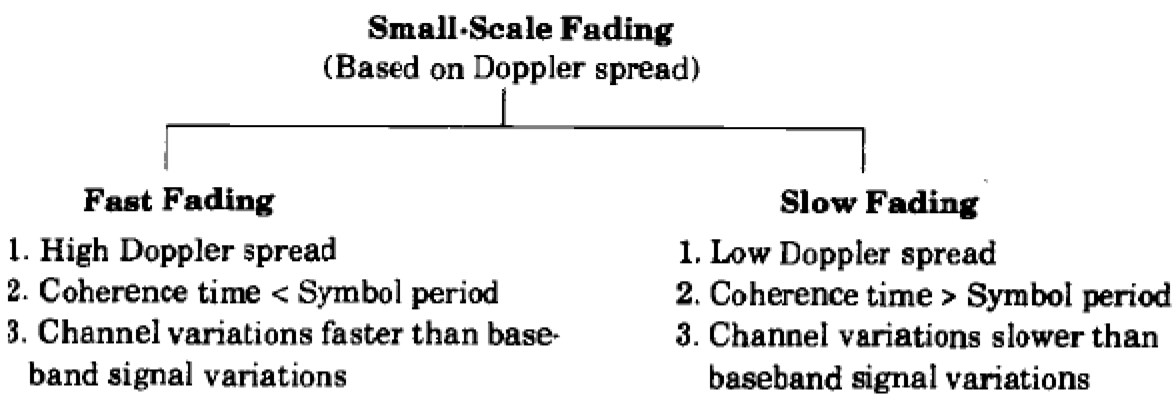
Fast Fading	Slow Fading
High Doppler spread	Low Doppler Spread
Coherence time is lesser than symbol period.	Coherence time is greater than symbol period.
Channel variations faster than base band signal variations	Channel variations slower than base band signal variations

In practice,fast fading only occurs for very low data rates, since it deal with rate of change of the channel.

2.(i)Explain fading effects due to multipath time delay spread and fading effects due to Doppler spread.(Nov/ Dec 2016)

Fading Effects Due to Multipath time Delay Spread

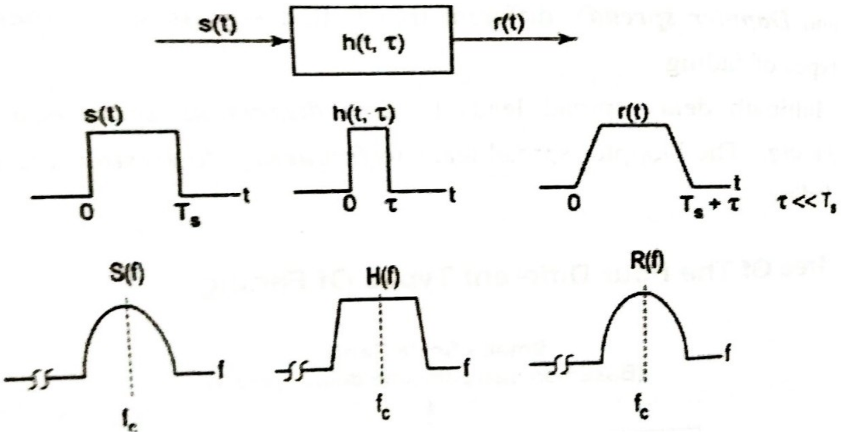




- Time dispersion due to multipath causes the transmitted signal to undergo either **flat or frequency selective fading.**

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.



Flat Fading Channel Charecteristics

- In a flat fading channel, the reciprocal bandwidth of the transmitted signal 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 referred to as narrowband channels.

Condition for flat fading

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

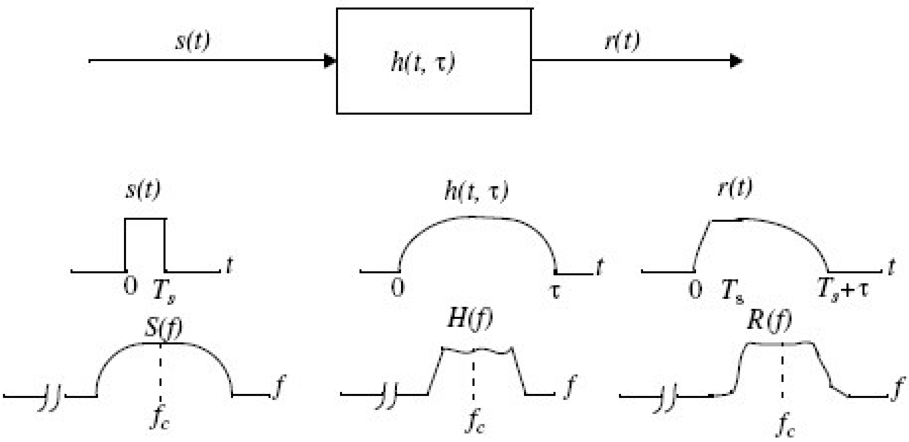
Where B_S = bandwidth of signal

$$B_S = 1 / T_S$$

B_C =Coherence bandwidth

σ_τ= rms delay spread

Frequency Selective Fading



Frequency selective Fading Channel Charecteristics

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

For frequency selective small-scale fading

$$\mathbf{B_S > B_C}$$

$$\mathbf{T_S \ll \sigma_\tau}$$

Fading Effects Due to Doppler Spread (NOV/DEC 2017)

Fast Fading

- In a fast fading channel, the channel impulse response changes rapidly within the symbol duration. That is, the coherence time of the channel is smaller than the symbol period of the transmitted signal.

$$T_S > T_C \text{ --- } (1)$$

And

$$B_S < B_D \text{ --- } (2)$$

Slow Fading

- In a slow fading channel, the channel impulse response changes at a rate much slower than the transmitted baseband signal s(t).

- In this case, the channel may be assumed to be static over one or several reciprocal bandwidth intervals. In the frequency domain, this implies that the Doppler spread of the channel is much less than the bandwidth of the baseband signals.

Therefore, a signal undergoes slow fading if

$$T_S \ll T_C \text{ -- } (3)$$

And

$$B_S \gg B_D \text{ -- } (4)$$

(ii)What are the factors influencing Small scale fading?(Nov/ Dec 2016)

Factors Influencing Small-Scale Fading:

- Many physical factors in the radio propagation channel influence small scale fading. These include the following:

Multipath propagation:

- The presence of reflecting objects and scatters in the channel creates a constantly changing environment that dissipates the signal energy in amplitude, phase, and time.
- These effects result in multiple versions of the transmitted signal that arrive at the receiving antenna, displaced with respect to one another in time and spatial orientation.

Speed of the mobile:

- The relative motion between the mobile results in random frequency modulation due to different Doppler shifts on each of the multipath components.
- Doppler shift will be positive or negative depending on whether the mobile receiver is moving toward or away from the base station.

Speed of surrounding objects:

- If objects in the radio channel are in motion, they induce a time varying Doppler shift on multipath components.

The transmission bandwidth of the signal:

- If the transmitted radio signal bandwidth is greater than the "bandwidth" of the multipath channel, the received signal will be distorted, but the received signal strength will not fade much over a local area
- The coherence bandwidth is a measure of the maximum frequency difference for which signals are still strongly correlated in amplitude.

3.a)(i)Describe briefly about Free space propagation model.(Apr/May 2018)

- The free space propagation model is used to predict received signal strength when the transmitter and receiver have a clear, unobstructed line-of-sight path between them.
- The free space power received by a receiver antenna which is separated from a radiating transmitter antenna by a distance d, is given by the Friis free space equation,

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

Where, P_t - transmitted power,

$P_r(d)$ - received power from distance "d"

L - system loss factor

λ - wavelength in meters

G_t - transmitter antenna gain,

G_r - receiver antenna gain

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

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

-

The validity of Friis' law is restricted to the far field of the antenna or Fraunhofer regionfar-field distance d_f is

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

- *D*- largest physical linear dimension of the antenna
- The far field requires d > λ and d >>D
- An **isotropic radiator** is an ideal antenna which radiates power with unit gain uniformly in all directions, and is often used to reference antenna gains in wireless systems.
- The **effective isotropic radiated power (EIRP)** is defined as

$$EIRP = P_t G_t$$

- Effective radiated power (ERP) is used instead of EIRP to denote the maximum radiated power as compared to a half-wave dipole antenna
- The **path loss** is defined as the difference (in dB) between the effective transmitted power and the received power.

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

- If the antennas are assumed to have unity gain

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

- Considering d_0 as a known received power reference point, the received power, $P_r(d)$, at any distance $d > d_0$ is

$$P_r(d) = P_r(d_0)\left(\frac{d_0}{d}\right)^2$$

- The reference distance d_0 for practical systems is typically chosen to be 1 min indoor environments and 100 m or 1 km in outdoor environments
- The P_r from the above expression will be a large value so for our convenience it can be represented in dBm(measured power referenced to one milliwatt (mW) …

$$P_r(d)\ \text{dBm} = 10\log\left[\frac{P_r(d_0)}{0.001\ \text{W}}\right] + 20\log\left(\frac{d_0}{d}\right)$$

(ii)Consider a transmitter which radiates a sinusoidal carrier frequency of 1850

Mhz. For a vehicle moving 60 mph, compute the received carrier frequency if the mobile is moving directly toward the transmitter.(Apr/May 2017,Apr/May 2018)
Solution:
Carrier frequency fc= 1850 Mhz
Vehicle speed v= 60 mph
Wavelength λ = C/fc = $3*10^8$ / $1850*10^6$ =0.162
Vehicle speed v= 60 mph (1 mile=1.609 km)
v=96.54 km/h
=$96.54*10^3$ m/3600 s
=26.79 m/s
The mobile is moving directly toward the transmitter.
In this case, the doppler shift is negative and the received frequency is given by
f=fc+fd
fc= 1850 Mhz
fd= v/ λ =26.79/0.162 =165.37 Hz
f =$1850*10^6$ +165.37 = 1850.000165 MHz
b(i)What do u mean by path loss model? Explain in detail about log distance pathloss model.(NOV/DEC 2017)
PATH LOSS MODEL:

- Most radio propogation models are derived using a combination of analytical and empirical methods.
- The empirical approach is based on fitting curves or analytical expressions that recreate a set of measured data.
- The advantage of implicity taking into account all propogation factors, both knowns and unknown, through actual field measurements.
- By using path loss models to estimate the received signal level as a function of distance, it becomes possible to predict the SNR for a mobile communication system.

Log-distance Path Loss Model:
- Both theoretical and measurement-based propagation models indicate that average received signal power decreases logarithmically with distance, whether in outdoor or indoor radio channels
- The average large-scale path loss for an arbitrary T-R separation is expressed as a function of distance by using a path loss exponent, n.

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

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

- where n is the path loss exponent which indicates the rate at which the path loss increases with distance

- d0 is the close-in reference distance which is determined from measurements close to the transmitter
- d is the T-R separation

(ii)What is the need for link Calculation? Explain with suitable example.(NOV/DEC 2017)

- 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 (often a worst case estimate) for the total SNR

Example:

*Consider the downlink of a GSM system . The carrier frequency is 950MHz and the RX sensitivity is (according to GSM specifications) −102 dBm. The output power of the TX amplifier is 30 W. The antenna gain of the TX antenna is 10 dB and the aggregate attenuation of connectors, combiners, etc. is 5 dB. The fading margin is 12 dB and the breakpoint d*break is at a distance of 100 m. What distance can be covered?

TX side:				
	TX power	P_{TX}	30 W	45 dBm
	Antenna gain	G_{TX}	10	10 dB
	Losses (combiner, connector, etc.)	L_f		−5 dB
	EIRP (Equivalent Isotropically Radiated Power)			50 dBm
RX side:				
	RX sensitivity	P_{min}		−102 dBm
	Fading margin			12 dB
	Minimum RX power (mean)			−90 dBm
	Admissible path loss (difference EIRP and min. RX power)			140 dB
	Path loss at $d_{break} = 100\,m$	$[\lambda/(4\pi d)]^2$		72 dB
	Path loss beyond breakpoint	$\propto d^{-n}$		68 dB

f strategies

Cha

Depending on the path loss exponent,

$$n = 1.5 \ldots 2.5 \text{ (line-of-sight)}^3$$

$$n = 3.5 \ldots 4.5 \text{ (non-line-of-sight)}$$

we obtain the coverage distance,

$$d_{cov} = 100 \cdot 10^{68}/^{(10n)}\,\mathrm{m} \tag{3.9}$$

If, e.g., $n = 3.5$, then the coverage distance is 8.8 km.

- For efficient utilization of the radio spectrum, a frequency reuse scheme that is consistent with the objectives of increasing capacity and minimizing interference is required.
- A variety of channel assignment strategies have been developed to achieve

these objectives. **Channel assignment strategies can be classified as either fixed or dynamic.**

- The choice of channel assignment strategy impacts the performance of the system, particularly as to how calls are managed when a mobile user is handed off from one cell to another

Fixed channel assignment strategy:

- Each cell is allocated a predetermined set of voice channels. Any call attempt within the cell can only be served by the unused channels in that particular cell.
- If all the channels in that cell are occupied, the call is blocked and the subscriber does not receive service.
- Several variations of the fixed assignment strategy exist.
- In one approach, called the 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 such borrowing procedures and ensures that the borrowing of a channel does not disrupt or interfere with any of the calls in progress in the donor cell.

Dynamic channel assignment strategy:

- Voice channels are not allocated to different cells permanently.
- Instead, each time a call request is made, the serving base station requests a channel from the MSC.
- The switch then allocates a channel to the requested cell following an algorithm.
- The MSC only allocates a given frequency if that frequency is not presently in use in the cell or any other cell which falls within the minimum restricted distance of frequency reuse to avoid co-channel interference.
- Dynamic channel assignment red**uces 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, arid radio signal strength indications (RSSI) of all channels on a continuous basis.
- This increases the storage and computational load on the system but provides the advantage of increased channel utilization and decreased probability of a blocked call.

Handoff Strategies

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

Handoff Considerations:

- Handoffs must be performed successfully and as infrequently as possible, and be imperceptible to the users.
- Failure to perform a handoff successfully results in termination of the ongoing call.

Selection of Handoff Signal Level

- System designers must specify an **optimum**signal level at which to initiate a handoff.
- Once a particular signal level is specified as the minimum usable signal for acceptable voice quality at the base station receiver, a slightly stronger signal level is used as a threshold at which a handoff is made.
- This margin is given by Δ

$$\Delta = Pr\ handoff - Pr\ min\ usable$$

- Δ too large Unnecessary Handoffs causing burden on MSC
- Δ too smallI unsufficient time to complete Handoff.
- Call terminated due to weak signal conditions.
- Δ is chosen carefully to balance these conflicting requirements.

Probable causes for excessive delays:

–Computational load on the MSC.

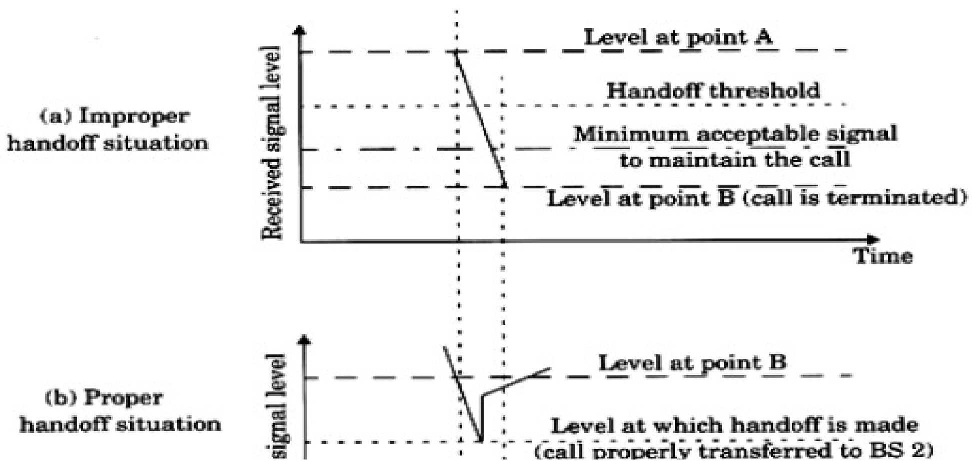
–Unavailability of channels on nearby base stations.

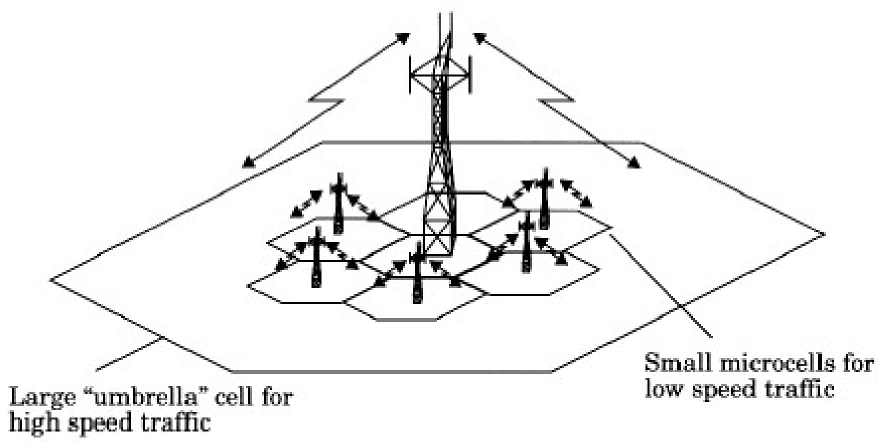
- The time over which a call may be maintained within a cell, without handoff, is called the *dwell time*. MAHO (Mobile Assisted HandOff)

MAHO (Mobile Assisted HandOff):

- In this every mobile station measures the received power from surrounding base stations and continually reports the results of these measurements to the serving base station.

Practical Handoff Considerations





The umbrella cell approach.

- Cell DraggingCell dragging occurs in an urban environment when there is a line-of-sight (LOS) radio path between the pedestrian subscriber and the base station.
- Even after the user has traveled well beyond the designed range of the cell, the received signal at the base station does not decay rapidly resulting in Cell Dragging

–Practical handoff problem in microcell systems

–For users at pedestrian speeds having LOS radio path with base stations

Handoff Types

Hard handoff- when the user moves to a new cell, he will be assigned with a new set of channels.

Soft Handoff- when the user moves to a new cell, the channel itself will be switched to the new base station. CDMA uses soft Handoff.

(ii)If a total of 33MHZ of bandwidth is allocated to a particular FFD cellular telephone system which uses two 25khz simplex channels to provide full duplex voice and control channels, compute the number of channels available per cell if a system uses (a) four cell reuse (b) seven cell reuse (c) Twelve cell reuse. If 1 MHz of the allocated spectrum is dedicated to control channels, determine the equitable distribution of control channels and voice channels in each cell of the each of the three systems. (Apr/May 2017)

Sol. Given total bandwidth= 33MHZ
Channel bandwidth = 25 khz x 2 simplex channels= 50 khz/duplex channel
Total available channels = 33,000/50 = 660 channels
(a) For N = 4
Total number of channels available per cell = 660/4=165 channels
(b) For N = 7
Total number of channels available per cell = 660/7 = 95 channels
(c) For N = 12
Total number of channels available per cell = 660/12 = 55 channels.

(b)(i)Derive the expression for cellular CDMA schemes for both noise limited and interference limited scenarios.(Apr/May 2017)

- CDMA uses CO-Channel Cells
- All the users use the same carrier frequency and may transmit simultaneously without any knowledge of others.
- The receiver performs a time correlation operation to detect only the specific desired codeword.
- All other code words appear as noise
- Multipath fading may be substantially reduced because the signal is spread over a large spectrum

- Channel data rates are very high in CDMA systems
- CDMA supports Soft handoff MSC can simultaneously monitor a particular user from two or more base stations. The MSC may chose the best version of the signal at any time without switching frequencies.
- In CDMA, the power of multiple users at a receiver determines the noise floor.
- In CDMA, stronger received signal levels raise the noise floor at the base station demodulators for the weaker signals, thereby decreasing the probability that weaker signals will be received. This is called Near- Far problem.
- To combat the Near- Far problem, power control is used in most CDMA

CDMA in Cellular:

- Frequency reuse in cdma is feasible as long as multiple access Interfernce(MAI) is kept below a given level.MAI ∞ channel loading
- Total interference = $(1+f)k/Q\ E_b$
- SINR = $E_b / N_o + I_o$
- Cellular CDMA often – Interference limited $I_o > N_o$
- Large cells – Noise limited $I_o / N_o = 0$
- Small cells – Interference limited.
- SINR = $1/ (1+f)\ K/Q\ (1+N_o / I_o)$

(ii)Consider GSM, which is a TDMA/FDD system that uses 25MHz for the forward link, which is broken into radio channels of 200kHz. If 8 speech channels are supported on a single radio channel, and if no guard band is assumed, find the number of simultaneously users that can be accommodated in GSM.(Apr/May 2017)

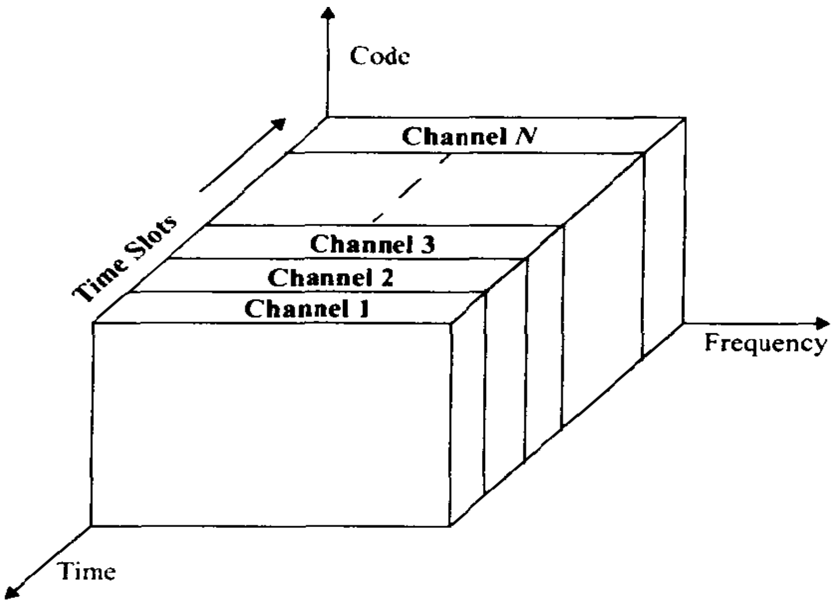
Number of simultaneously users that can be accommodated in GSM
N = (25 MHz/200kHz) x 8 = 1000 simultaneous user
(iii)If GSM uses a frame structure where each frame consists of eight time slots, and each time slot contains 156.25 bits, and data are transmitted at 270.833 kbps in the channel, find
(a) the time duration of a bit,
(b) the time duration of a slot,
(c) the time duration of a frame,
(d) how long must a user occupying a single time slot wait between two successive transmissions.(Apr/May 2017)

(a) the time duration of a bit Tb = 1/270.833 kbps = 3.692 μs
(b) the time duration of a slot Ts = 156.25 Tb = 0.577ms
(c) the time duration of a frame Tf = 8 Ts = 4.615 ms

(d) a user needs to wait one frame duration, i.e., 4.615 ms, between two successive transmissions.

2.a) Identify the channel capacity of TDMA in cell system(Nov/Dec 2017)
Time Division Multiple Access:

- Time division multiple access (TDMA) systems divide the radio spectrum into time slots.
- Each user occupies a cyclically repeating time slot A set of 'N' slots form a Frame.
- Each frame is made up of a preamble, an information message, and tail bits
- TDMA systems transmit data in a **buffer-and-burst method.**

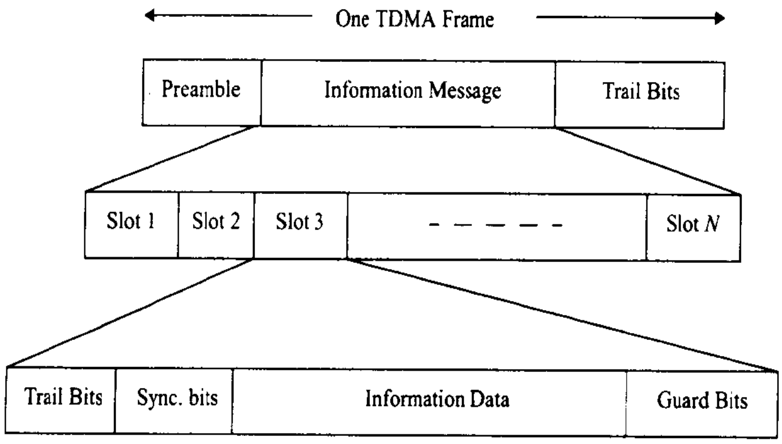


Features of TDMA:

- TDMA shares a single carrier frequency with several users
- Data transmission for users of a TDMA system is not continuous, but occurs in bursts.
- TDMA uses different time slots for transmission and reception, thus duplxers are not required
- Adaptive equalization is usually necessary in TDMA systems
- Guard time Should be minimized
- High synchronization overhead is required in TDMA systems because of burst transmissions.

Frame Structure:

- The preamble contains the address and synchronization information that both the base station and the subscribers use to identify each other.
- Trial bits specify the start of a data.
- Synchronization bits will intimate the receiver about the data transfer. Guard Bits are used for data isolation.



Efficiency of TDMA

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

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

where

b_{0H} – no over head bits per frame

b_r - no of overhead bits per

b_p - no overhead bits per preamble in each slot

b_g - no equivalent bits in each guard time interval

N_r - reference bursts per frame,

N_t - traffic bursts per frame

- The total number of bits per frame, b_T, is

$$b_T = T_f \; R$$

T_f is the frame duration, and R is the channel bit rate

- Then the frame efficiency is

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

Number of channels in TDMA Sytem:

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

m - Maximum number of TDMA users supported on each radio channel

(b)Write short notes on trunking and Grade of service of cell system(Nov/Dec 2017)

Trunking and Grade of System:

- Trunking to accommodate a large number of users in a limited radio spectrum.
- The concept of trunking allows a large number of users to share the relatively small number of channels in a cell by providing access to each user, on demand, from a pool of available channels
- **Trunking theory** to determine the number of telephone circuits that need to be allocated for office buildings with hundreds of telephones.
- The fundamentals of trunking theory were developed by Erlang
- **One Erlang** represents the amount of traffic intensity carried by a channel that is completely

occupied.

- The **grade of service (GOS)** is a measure of the ability of a user to access a trunked system during the busiest hour.
- The busy hour is based upon customer demand at the busiest hour during a week, month, or year.
- The busy hours for cellular radio systems typically occur during rush hours, between 4 p.m. And 6 p.m. on a Thursday or Friday evening.Each user generates a traffic intensity of Au Erlangs given by

$$A_u = \lambda H$$

- where H is the average duration of a call and λ is the average number of call requests per unit time.
- For a system containing U users and an unspecified number of channels, the total offered traffic intensity A, is given as

$$A = UA_u$$

-
- There are two types of trunked systems which are commonly used. The first type offers no queuing for call requests.
- If no channels are available, the requesting user is blocked without access and is free to try again later.
- This type of trunking is called **blocked calls cleared** and assumes that calls arrive as determined by' a Poisson distribution

3.(a) Explain about Co channel interference and Adjacent channel interference. Describe the techniques to avoid Interference (Nov/Dec 2016,Nov/Dec 2015)

- Interference is the major limiting factor in the performance of cellular radio systems
- The two major types of system-generated cellular interference are co-channel interference and adjacent channel interference

Co-channel Interference and System Capacity:

- Frequency reuse implies that in a given coverage area there are several cells that use the same set of frequencies.
- These cells are called **co-channel cells**, and the interference between signals from these cells is called **co-channel interference.**
- The co-channel interference ratio is independent of the transmitted power and becomes a function of the radius of the cell (B) and the distance between centers of the nearest co-channel cells (D).
- The parameter **Q, called the cochannel reuse ratio**, is related to the cluster size. For a hexagonal geometry

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

- Let i0 be the number of co-channel interfering cells. Then, the signal-tointerference ratio (S/I or SIR) for a mobile receiver which monitors a forward channel can be expressed as

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

- The average received power Pr at a distance d from the transmitting antenna is approximated by

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

- Considering only the first layer of interfering cells, if all the interfering base stations are equidistant from the desired base station and if this distance is equal to the distance D between cell centers,

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

- Adjacent Channel Interference:Interference resulting from signals which are adjacent in frequency to the desired signal is called adjacent channel interference

- This is referred to as the near-far effect, where a nearby transmitter (which may or may not be of the same type as that used by the cellular system) captures the receiver of the subscriber.
- Adjacent channel interference can be minimized through careful **filtering and channel assignments.**
- Since each cell is given only a fraction of the available channels, a cell need not be assigned channels which are all adjacent in frequency.
- If the frequency reuse factor is small, the separation between adjacent channels may not be sufficient to keep the adjacent channel interference level within tolerable limits.

(b)Explain in detail how frequency is efficiently allocated in an cellular systems(Nov/Dec 2016)

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 **frequency reuse or frequency planning**
- The actual radio coverage of a cell is known as the **footprint** and is determined from field measurements or propagation prediction models.

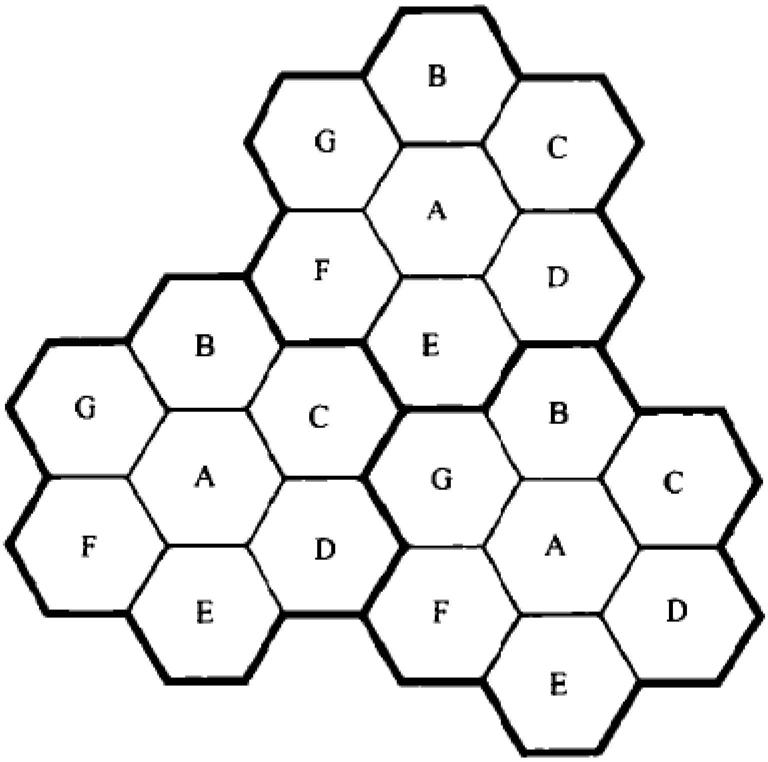


Fig: Frequency Reuse Concept

- There are three sensible choices: **a square; an equilateral triangle; and a hexagon.**
- For a given distance between the center of a polygon and its farthest perimeter points, the hexagon has the largest area of the three.
- Thus, by using the hexagon geometric the fewest number of cells can cover a geographic region, and the hexagon closely approximates a circular radiation pattern which would occur for an omni-directional base station antenna and free space propagation
- When using hexagons to model coverage areas, base station transmitters 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).**
- Normally, omni-directional antennas are used in center-excited cells and sectored directional antennas are used in corner-excited cells.
- The total number of available radio channels can be expressed as

$$S = kN$$

- The N cells which collectively use the complete set of available frequencies is called a **cluster**
- The total number of duplex channels, C, can be used as a measure of capacity and is given

$$C = MkN = MS$$

-
- The factor N is called the cluster size and is typically equal to 4, 7, or 12.The frequency reuse factor of a cellular system is given by I /N
- N, can only have values which satisfy

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

- Where i and j are non-negative integers.
- To find the nearest co-channel neighbors of a particular cell, one must do the following: (1) move i cells along any chain of hexagons and then (2) turn 60 degrees counter-clockwise and move / cells

4.(a) Summarise the features of various multiple access techniques used in wireless communication. State the Advantages and Disadvantages of Each Technique(May/June 2016)

Multiple Access Techniques:

- Multiple access schemes are used to allow many mobile users to share simultaneously a finite amount of radio spectrum. Major Types

- **Frequency division multiple access (FDMA)**
- **Time division multiple access (TDMA)**
- **Code division multiple access (CDMA)**

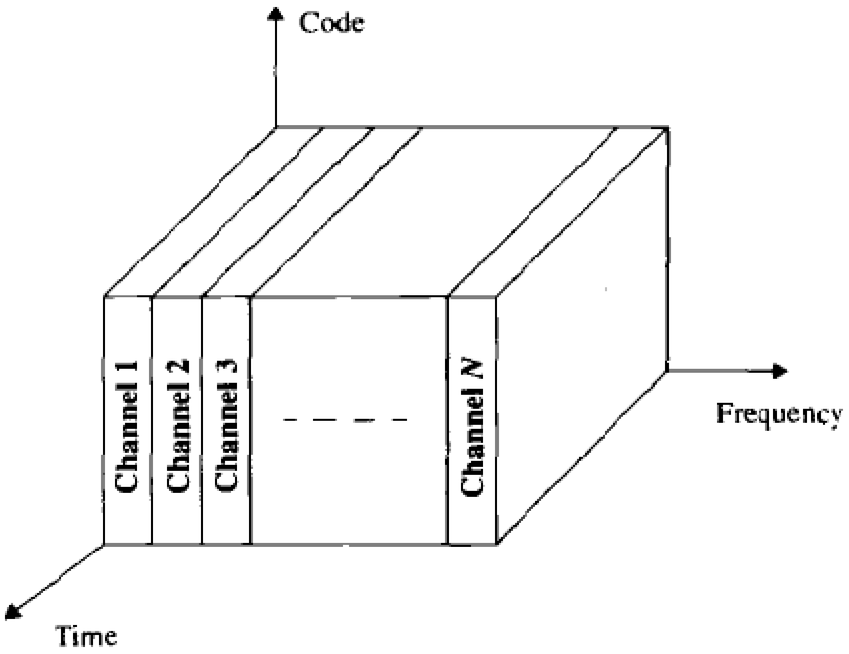
Frequency division multiple access (FDMA):

- Frequency division multiple access (FDMA) assigns individual channels to individual users

- During the period of the call, no other user can share the same frequency band.

Features of FDMA:

- The FDMA channel carries only one phone circuit at a time.
- If an FDMA channel is not in use, then it sits idle and cannot be used by other users to increase or share capacity.
- After the assignment of a voice channel, the base station and the mobile transmit simultaneously and continuously.
- The bandwidths of FDMA channels are relatively narrow (30 kHz)



- The symbol time is large as compared to the average delay spread.The complexity of FDMA mobile systems is lower when compared to TDMA system
- Since FDMA is a continuous transmission scheme, fewer bits are needed for overhead purposes (such as synchronization and framing bits) as compared to TDMA.
- FDMA systems have higher cell site system costs as compared to TDMA systems.
- FDMA requires tight RF filtering to minimize adjacent channel interference.

Nonlinear Effects in FDMA:

- In a FDMA system, many channels share the same antenna at the base station.
- The power amplifiers or the power combiners, when operated at or near saturation for maximum power efficiency, are nonlinear.
- The nonlinearities cause signal spreading in the frequency domain and generate intermodulation (IM) frequencies
- The number of channels that can be simultaneously supported in a FDMA system is given by

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

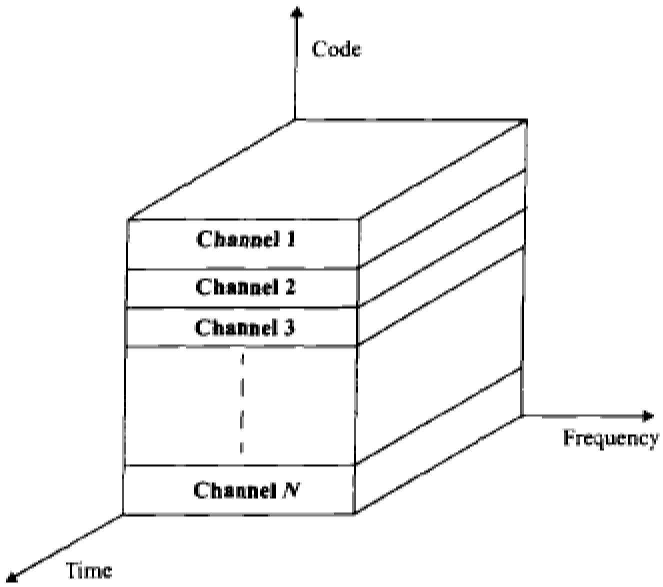
b_t -> total spectrum allocation, b_{guard} ->the guard band

b_c -> the channel bandwidth

Time Division Multiple Access(TDMA): Refer Q.No 2a

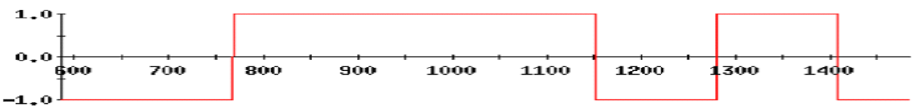
Code Division Multiple Access (CDMA):

- Direct sequence multiple access is also called code division multiple access (CDMA).
- In code division multiple access (CDMA) systems, the narrowband message signal is multiplied by a very large bandwidth signal called the spreading signal.
- The spreading signal is a pseudo-noise code sequence that has a chip rate which is orders of magnitudes greater than the data rate of the message.
- Each user has its own pseudorandom codeword which is approximately orthogonal to all other codewords.
- For detection of the message signal, the receiver needs to know the codeword used by the transmitter

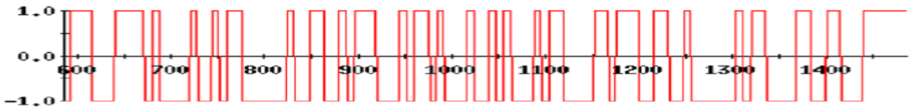


- If the power of each user within a cell is not controlled such that they do not appear equal at the base station receiver, then the near-far problem occurs.
- The near-far problem occurs when many mobile users share the same channel
- To combat the near-far problem, power control is used in most CDMA implementations.

Message



PN sequence



Features of CDMA:

- Many users of a CDMA system share the same frequency. Either TDD or FDD may be used.
- Unlike TDMA or FDMA, CDMA has a soft capacity limit.
- Multipath fading may be substantially reduced because the signal is spread over a large spectrum
- Channel data rates are 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. Self-jamming arises from the fact that the spreading sequences of different users are not exactly orthogonal
- The near-far problem occurs at a CDMA receiver if an undesired user has a high detected power as compared to the desired user.

(b)Explain in detail how to improve coverage and channel capacity in cellular systems(May/June 2016)

Improving Coverage and Capacity:

Need:

- The no of users are getting increased drastically.
- But the available channels are limited.
- So it is necessary to find some new channels for them.

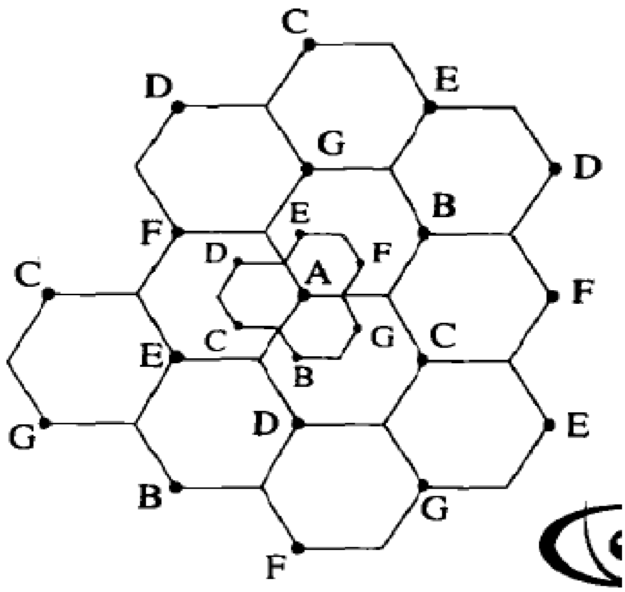
Methods:

- Cell Splitting
- Sectoring
- A Microcell Zone Concept
- Repeaters

Cell Splitting

- Cell splitting 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 transmitter power.

- Cell splitting increases the capacity of a cellular system since it increases the number of times that channels are reused.
- By defining new cells which have a smaller radius than the original cells and by installing these smaller cells (called microcells) between the existing cells, capacity increases due to the additional number of channels per unit area.
- In order to cover the entire service area with smaller cells, approximately four times as many cells would be required.
- This can be easily shown by considering a circle with radius R.
- The area covered by such a circle is four times as large as the area covered by a circle with radius R/2.



- This is necessary to ensure that the frequency reuse plan for the new microcells behaves exactly as for the original cel's.

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

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

- Antenna downtilting, which deliberately focuses radiated energy from the base station towards the ground (rather than towards the horizon), is often used to limit the radio coverage of newly formed microcells.

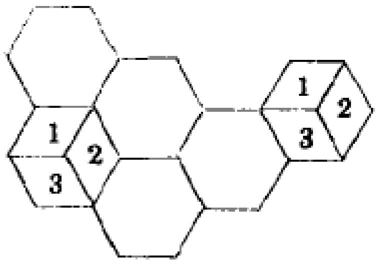
Sectoring

- In this the cells are **not divided** into microcells.
- Instead by using directional antennas we will create multiple sectors in a single cell.
- And the total allotted channels are divided among the sectors.

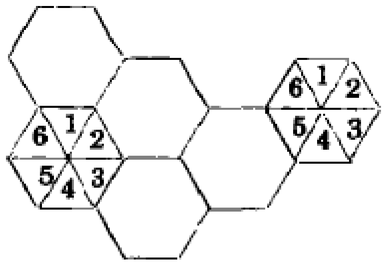
Types

•60^{o} sectoring

•120^{o} sectoring



(a)



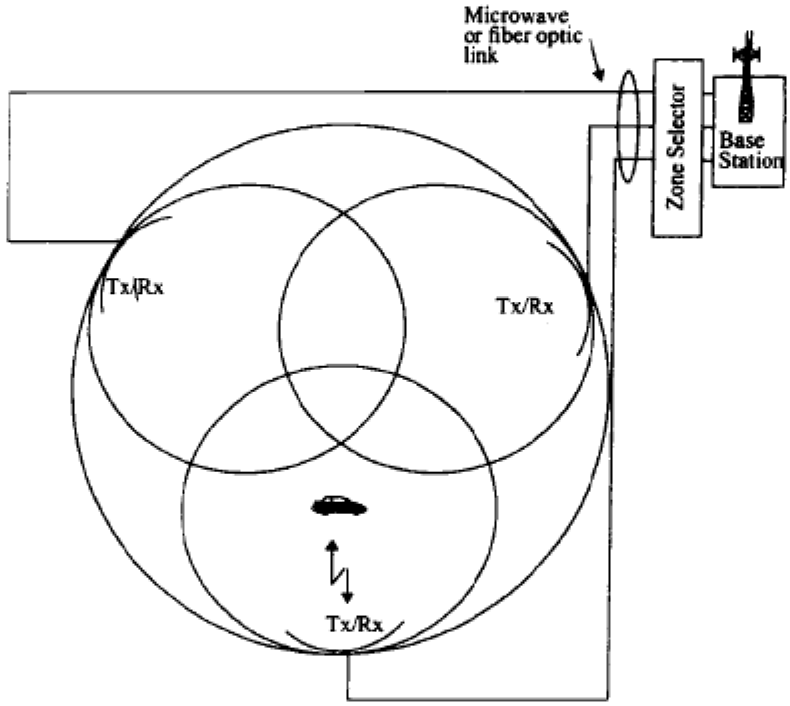
- Co-channel interference can be decreased by replacing a single Omni-directional antenna by several directional antennas, each radiating within a specified sector alone.
- Since a sector has only a fraction of the available co-channels.

Disadvantage

- The available channels must be subdivided and dedicated to a specific antenna.
- So at a given point of time, some sectors may be busy and others will be free.
- Since the channels are dedicated to a particular antenna we cannot transfer them for the congested sector.So the efficiency of the system is reduced.
- A handoff is required when user moves between Sectors.

A Microcell Zone Concept:

- A Microcell Zone is an advancement of Sectoring.
- In this channels are not dedicated to any antennas.
- When the user moves from zone to zone, base station switches the channel to a different zone.
- A handoff is not required when the mobile travels between zones within the cell.



Microwave
or fiber optic
link
Zone Selector
Base
Station
Tx/Rx
Tx/Rx
Tx/Rx

UNIT III DIGITAL SIGNALING FOR FADING CHANNELS

1.a) Prove that the OFDM system converts the delay spread channel into a set of parallel fading channels, using the concept of cyclic prefix (Apr/May 2018,Apr/ May 2017,Nov/Dec 2016)

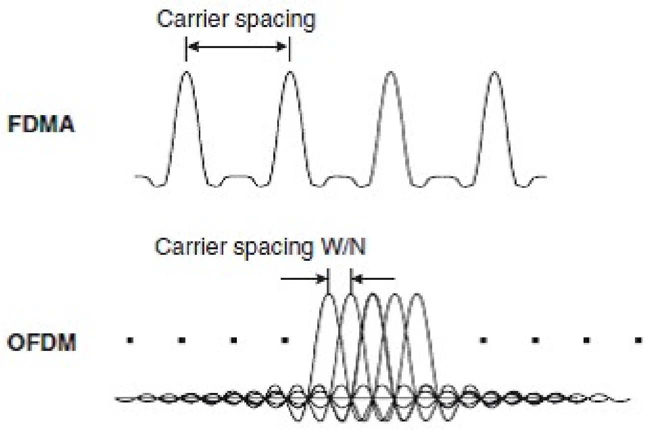
Orthogonal Frequency Division Multiplexing (OFDM)

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

Principle of Orthogonal Frequency Division Multiplexing

- OFDM splits a high-rate data stream into N parallel streams, which are then transmitted by modulating N distinct carriers (henceforth called subcarriers or tones).
- Symbol duration on each subcarrier thus becomes larger by a factor of N.
- In order for the receiver to be able to separate signals carried by different subcarriers, they have to be orthogonal.
- let subcarriers be at the frequencies fn = nW/N, where n is an integer, and W the total available bandwidth; in the most simple case, W = N/Ts.
- Subcarriers are mutually orthogonal, since the relationship

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



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

Fig Principle behind OFDM : N Carriers with a bandwidth of W

Implementation of Transceivers

OFDM can be interpreted in two ways: one is an “analog” interpretation.

- First split our original data stream into N parallel data streams, each of which has a lower data rate.
- Number of local oscillators (LOs) available, each of which oscillates at a frequency fn = nW/N, where n = 0, 1, . . .,N − 1.
- Each of the parallel data streams then modulates one of the carriers.

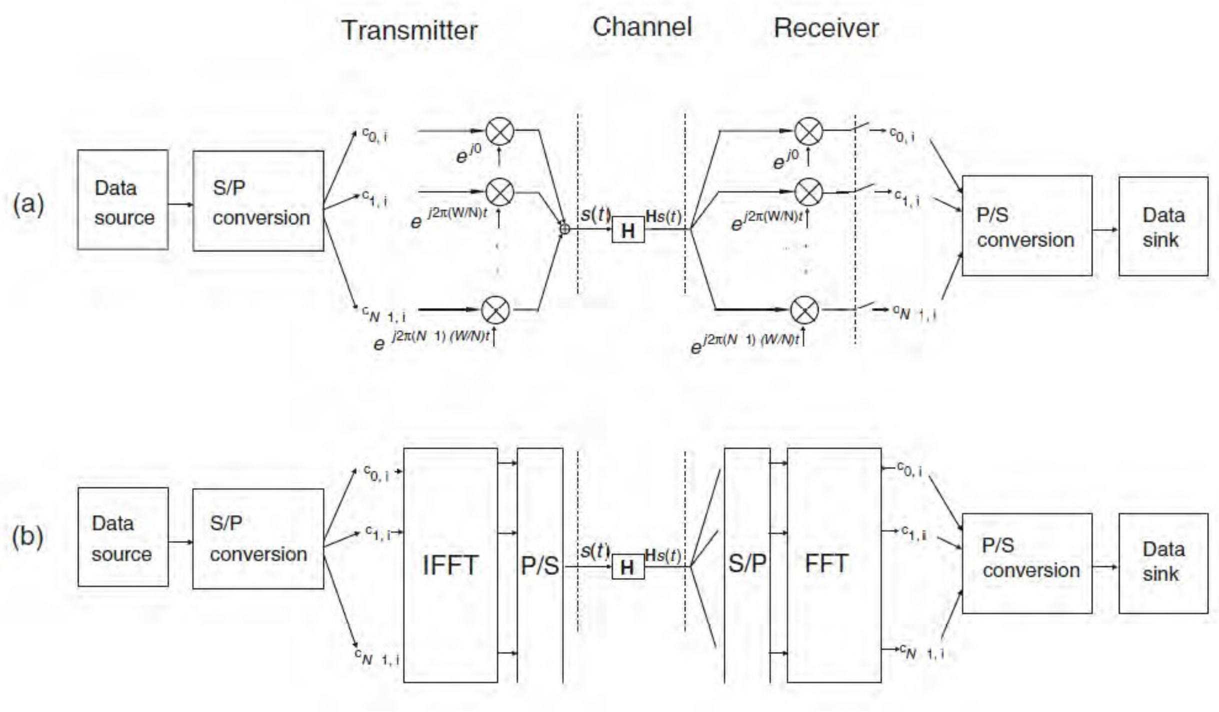


Fig. (a)Transceiver for OFDM (b) Using Inverse fourier Transform

- An alternative implementation is digital .
- 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 .
- This approach is much easier to implement with integrated circuits.
- Let us first consider the analog interpretation.
- Let the complex transmit symbol at time instant i on the nth carrier be cn,i . The transmit signal is then

$$s(t) = \sum_{i=-\infty}^{\infty} s_i(t) = \sum_{i=-\infty}^{\infty} \sum_{n=0}^{N-1} c_{n,i} g_n(t - iT_S) \qquad \text{---------------------------- (2)}$$

where the basis pulse gn(t) is a normalized, frequency-shifted rectangular pulse:

$$g_n(t) = \begin{cases} \frac{1}{\sqrt{T_S}} \exp\left(j2\pi n \frac{t}{T_S}\right) & \text{for } 0 < t < T_S \\ 0 & \text{otherwise} \end{cases} \qquad \text{----------------------(3)}$$

consider the signal only for i = 0, and sample it at instances tk = kTs/N

$$s_k = s(t_k) = \frac{1}{\sqrt{T_S}} \sum_{n=0}^{N-1} c_{n,0} \exp\left(j2\pi n \frac{k}{N}\right)$$ -- (4)

- Now, this is nothing but the inverse Discrete Fourier Transform (DFT) of the transmit symbols.
- Therefore, the transmitter can be realized by performing an Inverse Discrete Fourier Transform(IDFT) on the block of transmit symbols.

Cyclic Prefix:

- Let us first define a new base function for transmission:

$$g_n(t) = \exp\left[j2\pi n \frac{W}{N} t\right] \quad \text{for} - T_{cp} < t < \hat{T}_S$$

- where again W/N is the carrier spacing, and TS ˆ = N/W. The symbol duration TS is now TS =ˆ TS + Tcp
- Therefore, during time −Tcp < t < 0, a copy of the last part of the symbol is transmitted. From linearity, it also follows that the total signal s(t) transmitted during time −Tcp < t < 0 is a copy of s(t) during the last part, ˆ TS − Tcp < t < ˆ TS. This prepended part of the signal is called the **"cyclic prefix."**
- The cyclic prefix converts this linear convolution into a cyclical convolution

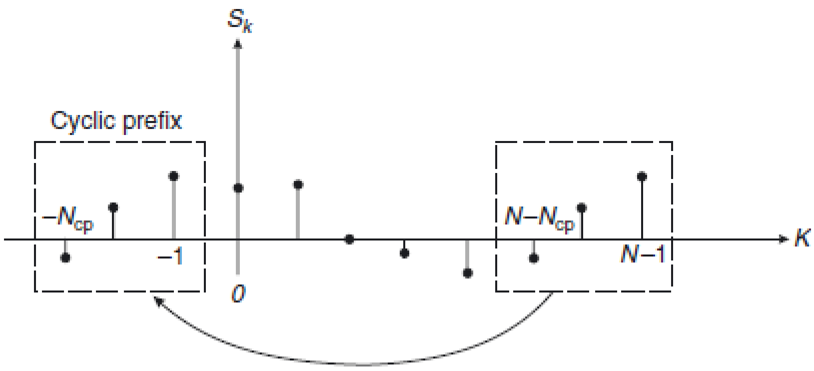
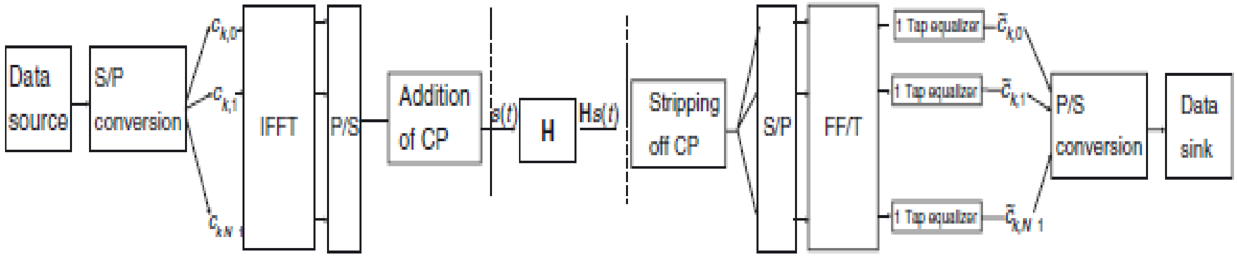


Fig:Principle of the cyclic prefix

- During the time −Tcp < t < −Tcp + τmax, where τmax is the maximum excess delay of the channel, the received signal suffers from "real" InterSymbol Interference (ISI), as echoes of the last part of the preceding symbol interfere with the desired symbol
- In the receiver, there is a bank of filters that are matched to the basis functions without the cyclic prefix:

$$\tilde{g}_n(t) = \begin{cases} g_n^*(\hat{T}_S - t) & \text{for } 0 < t < \hat{T}_S \\ 0 & \text{otherwise} \end{cases}$$

-



The signal at the output of the matched filter is thus convolution of the transmit signal with the channel impulse response and the receive filter:Fig:Structure of an orthogonal-frequency-division-multiplexing transmission chain with cyclic prefix and one-tap equalization.

- The original data stream is S/P converted.
- Each block of *N* data symbols is subjected to an IFFT, and then the last *NT*cp/ *T*S samples are prepended.
- The resulting signal is modulated onto a (single) carrier and transmitted over a channel, which distorts the signal and adds noise.
- At the receiver, the signal is partitioned into blocks. For each block, the cyclic prefix is stripped off, and the remainder is subjected to an FFT

Peak-to-Average Power Ratio:

Origin of the Peak-to-Average Ratio Problem:

- This Peak-to-Average Ratio (PAR) issue originates from the fact that an OFDM signal is the superposition of N sinusoidal signals on different subcarriers.
- On average the emitted power is linearly proportional to N the contributions to the total signal from the different subcarriers can be viewed as random variables

There are three main methods to deal with the Peak-to-Average Power Ratio (PAPR):

1. Put a power amplifier into the transmitter that can amplify linearly up to the possible peak
value of the transmit signal.
2. Use a nonlinear amplifier, and accept the fact that amplifier characteristics will lead to distortions in the output signal
3. Use PAR reduction techniques.

Peak-to-Average Ratio Reduction Techniques:

1. Coding for PAR reduction: under normal circumstances, each OFDM symbol can represent one
of 2^N codewords (assuming BPSK modulation). Now, of these codewords only a subset of size 2^K is acceptable in the sense that its PAR is lower than a given threshold.

2. Phase adjustments: this scheme first defines an ensemble of phase adjustment vectors φl, l =
1, . . . , L, that are known to both the transmitter and receiver; each vector has N

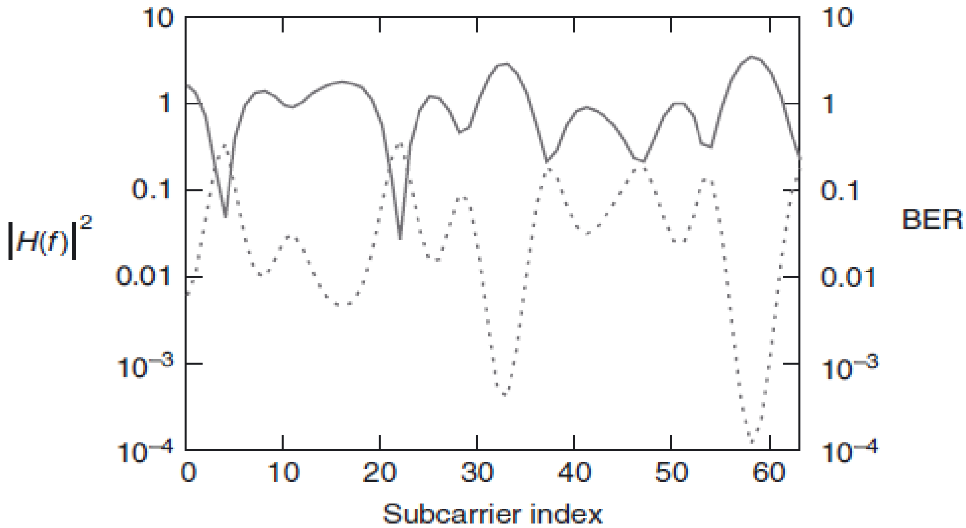
entries {φn}l .

3. Correction by multiplicative function: Another approach is to multiply the OFDM signal by a time-dependent function whenever the peak value is very high

4. Correction by additive function: In a similar spirit, we can choose an additive, instead of a multiplicative, correction function

b)Derive the bit error rate for binary phase shift keying modulation for frequency flat fading channels (Apr/May 2018)

- The cyclic prefix converts a frequency-selective channel into a number of parallel flat-fading channels
- An uncoded OFDM system does not show any frequency diversity
- If a subcarrier is in a fading dip, then error probability on that subcarrier is very high, and dominates the Bit Error Rate (BER) of the total system for high SNRs.
- The BER is highest in fading dips. The results are plotted on a logarithmic scale – while the BER on "good" subcarriers can be as low as $10^{-4,}$ the BER on subcarriers that are in fading dips are up to 0.5.



- Fig:Transfer function and the BER of a Binary-Phase Shift Keying (BPSK)The BER decreases only linearly as the SNR increases

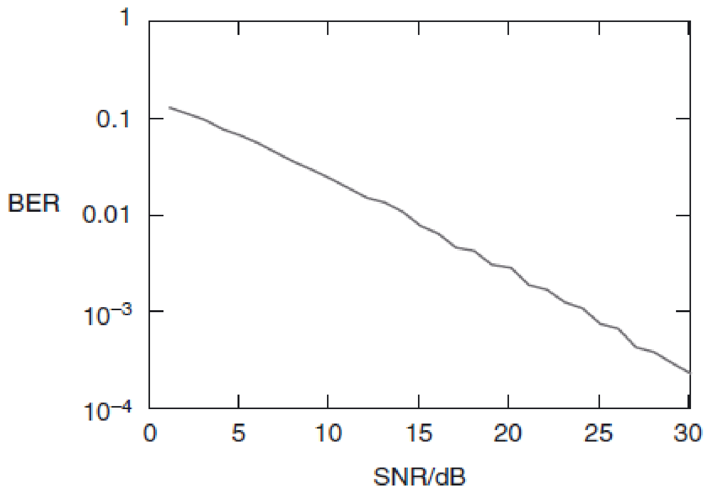


Fig:simulation of the average BER for a frequency-selective channel

- Frequency selectivity gives us different channel realizations on different subcarriers; time variations give us different channel realizations at different times.
- Main problem lies in the fact that carriers with poor SNR dominate the performance of the system.
- Any of the following approaches circumvents this problem:
    - Coding across the different tones:
    - *Spreading the signal over all tones*:
    - *Adaptive modulation*

2.(a)Describe with neat diagram, the modulation technique of QPSK(Nov/Dec 2017)

Quadrature Phase Shift Keying (QPSK):

- 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, and 3 π/2, 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

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

-

- where Ts is the symbol duration and is equal to twice the bit periodUsing trigonometric identities, the above equations can be rewritten for the interval 0 < t _<Ts as,

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

If basis functions $\phi_1(t) = \sqrt{2/T_s}\cos(2\pi f_c t)$, $\phi_2(t) = \sqrt{2/T_s}\sin(2\pi f_c t)$ are defined over the interval $0 \le t \le T_s$ for the QPSK signal set, then the 4 signals in the set can be expressed in terms of the basis signals as

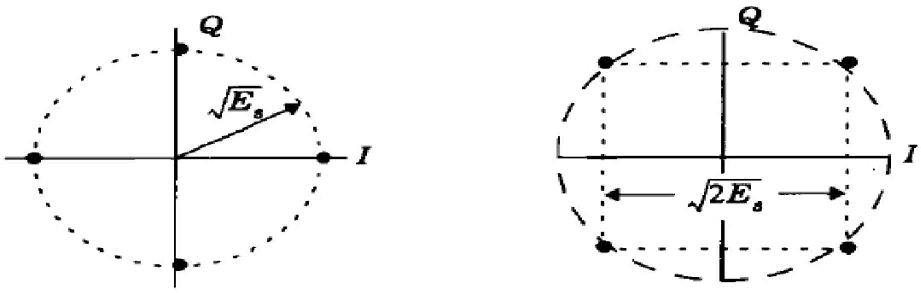


Fig: Constellation diagram for QPSK

- QPSK signal can be depicted using a two dimensional constellation diagram with four points.
- It should be noted that different QPSK signal sets can be derived by simply rotating the constellation.
- The average probability of bit error in the additive white Gaussian noise (AWGN) channel is obtained as

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

- Thus when compared to BPSK, QPSK provides twice the spectral efficiency with exactly the same energy efficiency.
- QPSK can also be differentially encoded to allow noncoherent detection.

QPSK Transmission and Detection Techniques

- The unipolar binary message stream has bit rate Rb and is first converted into a bipolar non-return-to-zero (NRZ) sequence using a unipolar to bipolar convener.
- The bit stream m (t) is then split into two bit streams in $m_I(t)$ and m_Q (t) (in-phase and quadrature streams), each having a bit rate of Rs = Rb/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 $\Phi_1(t)$ and $\Phi_2(t)$ 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.

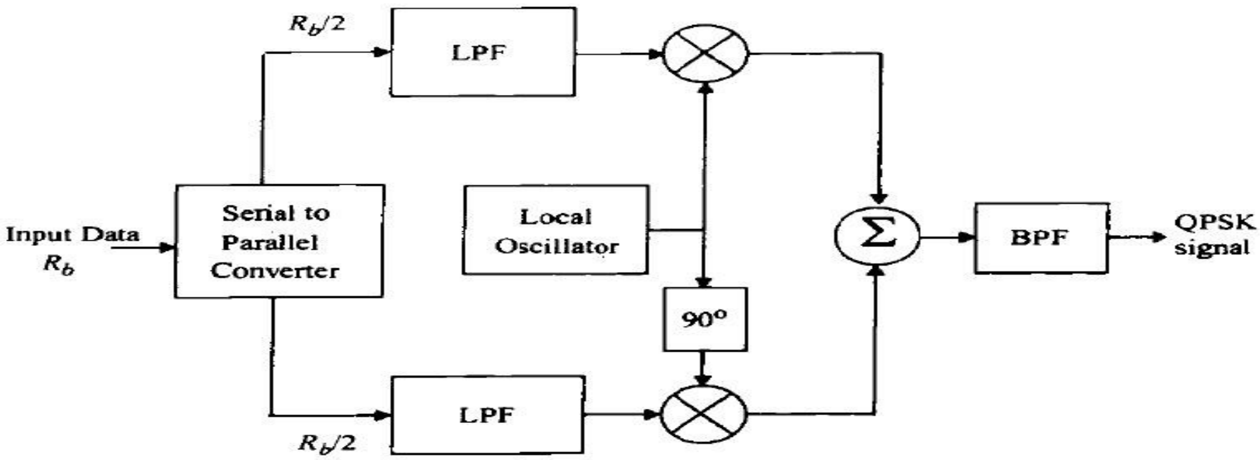


Fig 3.7 QPSK Transmitter

- Figure shows a block diagram of a coherent QPSK receiver.
- The frontend bandpass filter removes the out-of-band noise and adjacent channel interference.
- The filtered output is split into two parts, and each part is coherently demodulated using the in-phase and quadrature carriers.
- The 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.

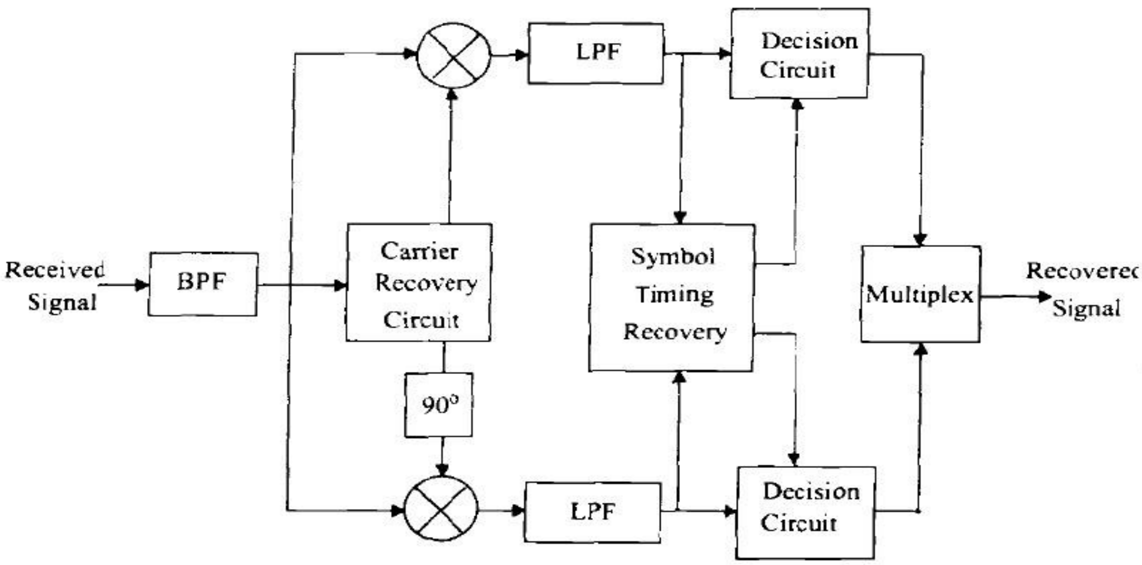


Fig 3.8 QPSK Receiver

(b)Explain the principle of MSK modulation and derive the expression for

power spectral density(Nov/Dec 2017,Nov/Dec 2016,Nov/Dec 2015)

- 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.
- MSK is continuous phase FSK with a modulation index of 0.5.
- The modulation index of an FSK signal is similar to the FM modulation index

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

-
- where ΔF is the peak RF frequency deviation and Rb is the bit rate.Two FSK signals vH(t) and VL(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 noncoherent FSK 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

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

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

- and where mI(t) and mQ(t) are the "odd" and "even" bits of the bipolar data stream which have values of ± 1 and which feed the in-phase and quadrature phase of the modulator at a rate of Rb/2

MSK Power Spectrum:

- The RF power spectrum is obtained by frequency shifting the magnitude squared of the Fourier transform of the baseband pulse shaping function.
- For MSK, the baseband pulse shaping function is given by

$$p(t) = \begin{cases} \cos\left(\frac{\pi t}{2T}\right) & |t| < T \\ 0 & \text{elsewhere} \end{cases} \qquad (5.106)$$

Thus the normalized power spectral density for MSK is given by [Pas79]

$$P_{MSK} = \frac{16}{\pi^2}\left(\frac{\cos 2\pi(f+f_c)T}{1.16 f^2 T^2}\right)^2 + \frac{16}{\pi^2}\left(\frac{\cos 2\pi(f-f_c)T}{1.16 f^2 T^2}\right)^2 \qquad (5.107)$$

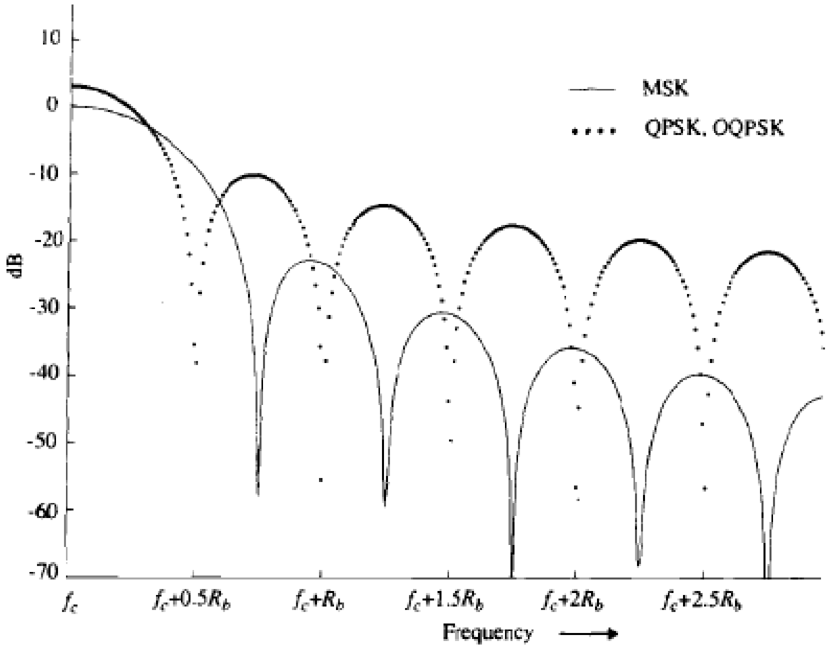


Fig:Power spectral density of MSK signals as compared to QPSK and OQPSK signals.MSK Transmitter and Receiver:

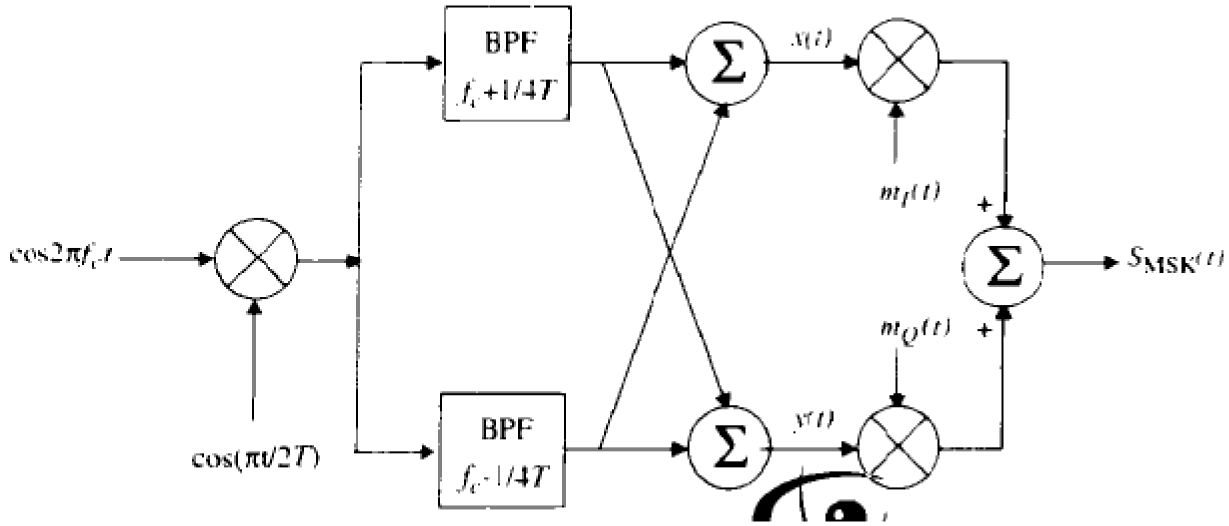
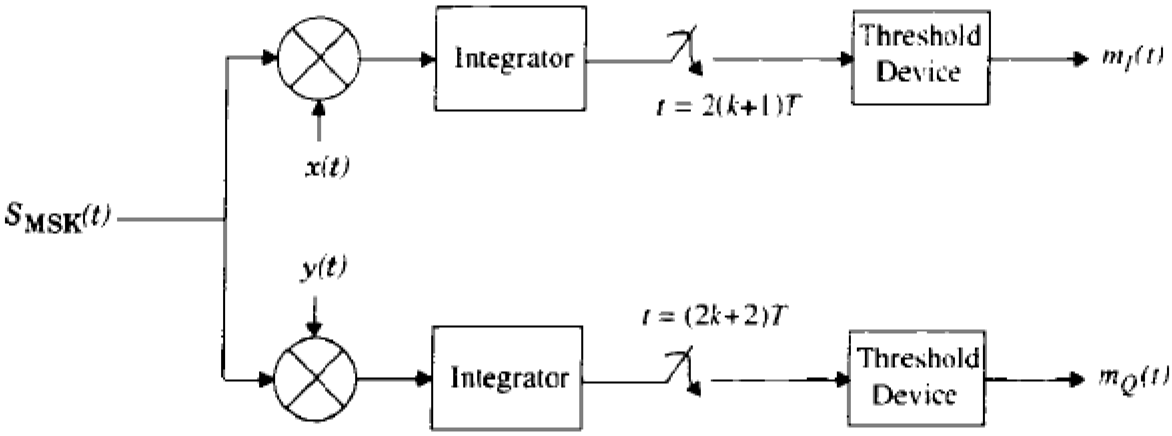


Fig: Block diagram of an MSK transmitter

- Multiplying a carrier signal with cos [πt/2T] produces two phase-coherent signals at fc+1\4T and fc-1\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).
- These carriers are multiplied with the odd and even bit streams, $m_I(t)$ and $m_Q(t)$, to produce the MSK modulated signal $S_{MSK}(t)$.



- Fig: Block diagram of an MSK ReceiverThe received signal $S_{MSK}(t)$ (in the absence of noise and interference) is multiplied by the respective in-phase and quadrature carriers x(t) and y(t).
- The output of the multipliers are integrated over two bit periods and dumped to a decision circuit at the end of each two bit periods.

3.(a)Discuss the error performance of different modulation schemes in fading channels (Apr/May 2017)

Comparison of Digital Modulation Schemes

Scheme	$S_1(t)$ and $S_2(t)$	BW	P_e	SNR	Complexity
Coherent ASK	$S_1(t) = A_c \cos 2\pi f_c t$ $S_2(t) = 0$	$2R_b$	high	low	high
Non-coherent ASK	$S_1(t) = A_c \cos \omega_c t$ $S_2(t) = 0$	$2R_b$	high	low	low
Coherent FSK	$S_1(t) = A_c \cos 2\pi f_1 t$ $S_2(t) = A_c \cos 2\pi f_2 t$ $(f_1 > f_2)$	$2R_b + (f_1 - f_2)$	moderate	high	high
Non-Coherent FSK	$S_1(t) = A_c \cos 2\pi f_1 t$ $S_2(t) = A_c \cos 2\pi f_2 t$	$> 2R_b$	moderate	high	low
Coherent PSK	$S_1(t) = A_c \cos \omega_c t$ $S_2(t) = -A_c \cos \omega_c t$	$2 R_b$	low	high	high
Non-coherent PSK	$S_1(t) = A_c \cos \omega_c t$ $S_2(t) = -A_c \cos \omega_c t$	$2R_b$	low	high	low

Digital Modulation Schemes	Probability of Error P_e
ASK (Coherent)	$P_e = Q\left[\sqrt{\frac{A_c^2 T_b}{4N_0}}\right]$
PSK (Coherent)	$P_e = Q\left[\sqrt{\frac{A_c^2 T_b}{N_0}}\right]$
FSK (Coherent)	$P_e = Q\left[\sqrt{\frac{A_c^2 T_b}{2N_0}}\right]$
ASK (Non-coherent)	$P_e = \frac{1}{2}\exp\left[-\frac{A_c^2}{8N_0}\right]$ if $A_c^2 >> N_0$
FSK (Non-coherent)	$P_e = \frac{1}{2}\exp\left[-\frac{A_c^2}{4N_0}\right]$
DPSK	$P_e = \frac{1}{2}\exp\left[-\frac{A_c^2 T_b}{2N_0}\right]$

b)(i)What is offset QPSK?What is its advantage? Describe the offset QPSK & π/ 4 DQPSK scheme (Apr/May 2017, May/June 2016)Offset- quadrature phase shift keying (Offset QPSK)

- A modified form of QPSK, called offset QPSK (OQPSK) or staggered QPSK is less susceptible to these deleterious effects arid supports more efficie t amplification.
- OQPSK signaling is similar to QPSK signaling, as represented by equation, except for the time alignment of the even and odd bit streams.
- In QPSK signaling, the bit transitions of the even and odd bit streams occur at the same time instants, but in OQPSK signaling, the even and odd bit streams, m_I (t) and are offset in their relative alignment by one bit period (half-symbol period).
- Due to the time alignment of m_I (t) and m_Q (t) in standard QPSK, phase transitions occur only once every Ts = 2Tb s, and will be a maximum of 1800 if there is a change in the value of both m_I (t) and m_Q (t)

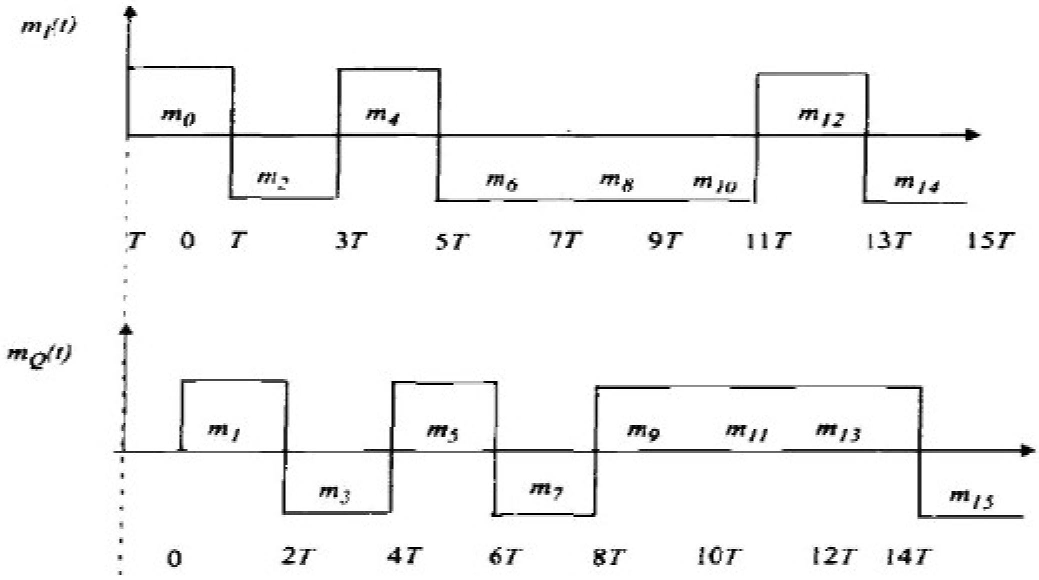


Fig Offset QPSK wave form

- In OQPSK Signaling, bit transitions (and hence phase transitions) occur every Ts.
- Since the transitions instants of m_I (t) and m_Q (t) are offset, at any given time only one of the two bit streams can change values.
- This implies that the maximum phase shift of the transmitted signal at any given time is limited to ±90° - Hence, by switching phases more frequently OQPSK signaling eliminates 180° phase transitions.
- Since 180° phase transitions have been eliminated, bandlimiting of (i.e., pulse shaping) OQPSK signals does not cause the signal envelope to go to zero.
- The spectrum of an OQPSK signal is identical to that of a QPSK signal, hence both signals occupy the same bandwidth.
- The staggered alignment of the even and odd bit streams does not change the nature of the spectrum.
- OQPSK signals also appear to perform better than QPSK in the presence of phase jitter due to noisy reference signals at the receiver.

(ii) Describe with neat diagram, the modulation technique of GMSK(Apr/May 2017,Nov/Dec 2015,May/June 2016)

Gaussian Minimum Shift Keying (GMSK)

- GMSK is a simple binary modulation scheme which may be viewed as a derivative of MSK.
- In GMSK, the sidelobe levels of the spectrum are further reduced by passing the modulating NRZ data waveform through a premodulation Gaussian pulse-shaping filter
- 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 detected just as an MSK signal, or noncoherently detected as simple FSK.
- In practice, GMSK is most attractive for its excellent power efficiency
- The GMSK premodulation filter has an impulse response given by

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

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

- and the transfer function given byThe parameter α is related to B, the S dB baseband bandwidth of $H_G(f)$ by

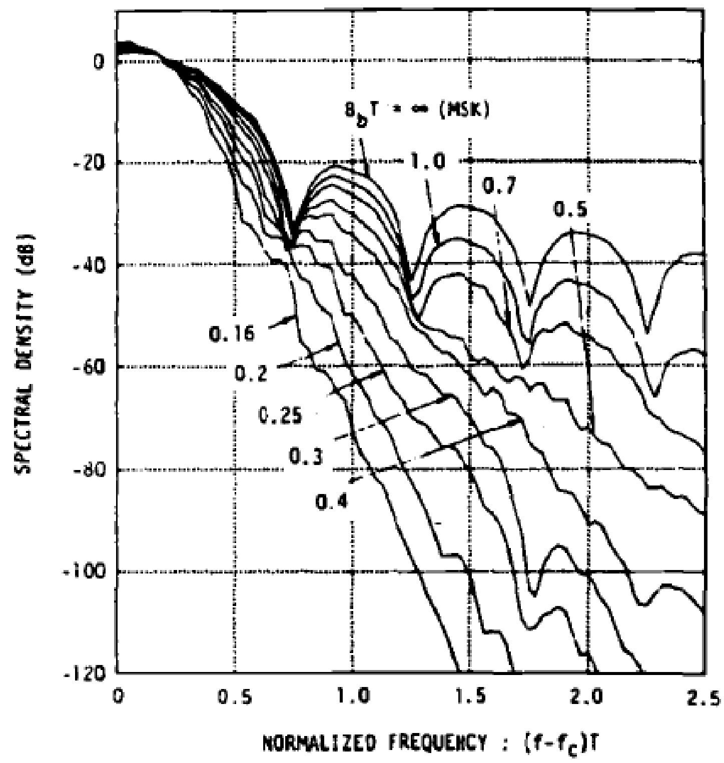


Fig:Power spectral density of a GMSK signal

GMSK Bit Error Rate:

- The bit error probability is a function of BT, since the pulse shaping impacts ISI.
- The bit error probability for GMSK is given by

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

where γ is a constant related to BT by

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

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 an FM modulator

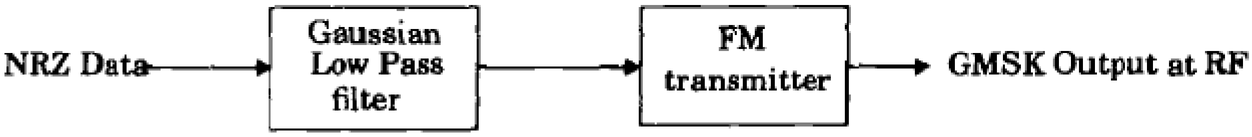


Fig:Block diagram of a GMSK transmitter using direct FM generation

- GMSK signals can be detected using orthogonal coherent detectors
- Carrier recovery is sometimes performed using a method suggested by de Buda where the sum of the two discrete frequency components contained at the output of a frequency doubler is divided by four
- The two D flip-flops act as a quadrature product demodulator and the XOR gates act as baseband multipliers.
- The mutually orthogonal reference carriers are generated using two D flip-flops, and the VCO center frequency is set equal to four times the carrier center frequency.

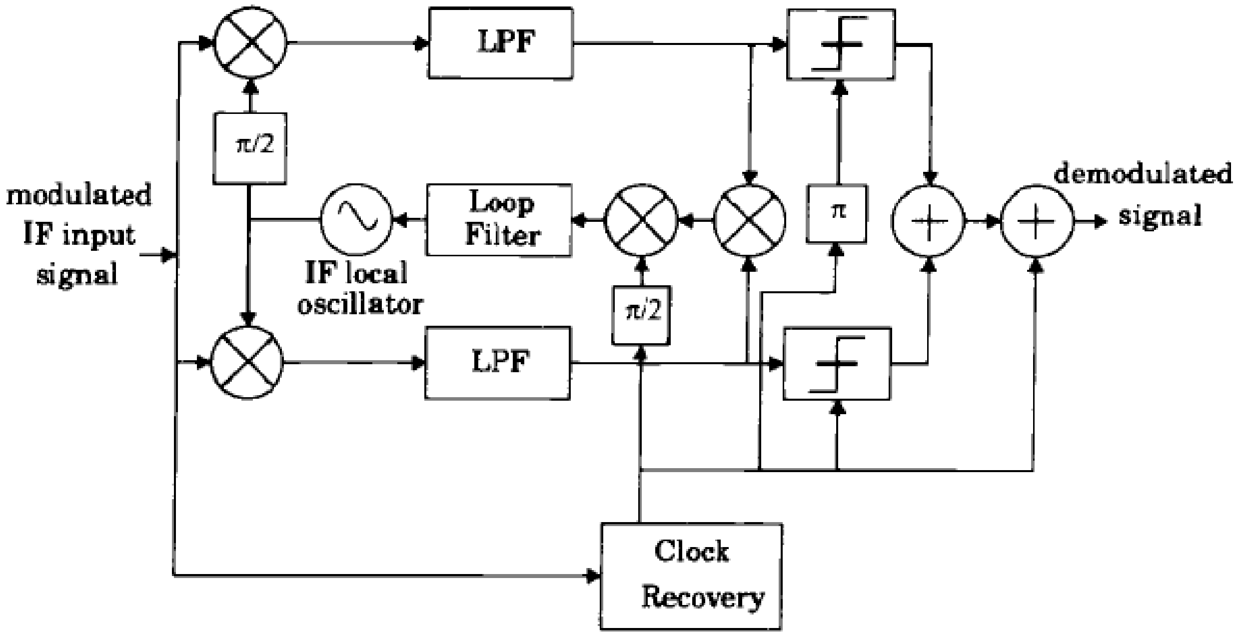


Fig:Block diagram of a GMSK receiver

4.Determine the error probability for fading channels with diversity Reception (April/May 2018, Nov/Dec 2017)

- We determine the Symbol Error Rate (SER) in fading channels when diversity is used at the RX.

Error Probability in Flat-Fading Channels:

Classical Computation Method

- The error probability of diversity systems by averaging the conditional error probability (conditioned on a certain SNR) over the distribution of the SNR:

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

- The SER of BPSK in AWGN is

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

- Computation via the Moment-Generating Function:The SER conditioned on a given SNR in the form

$$SER(\gamma) = \int_{\theta_1}^{\theta_2} f_1(\theta) \exp(-\gamma_{\mathrm{MRC}} f_2(\theta))\, d\theta$$

$$\gamma_{\mathrm{MRC}} = \sum_{n=1}^{N_r} \gamma_n$$

- The error probability for BPSK in Rayleigh fading as

$$\overline{SER} = \frac{1}{\pi} \int_0^{\pi/2} \left[\frac{\sin^2(\theta)}{\sin^2(\theta) + \overline{\gamma}} \right]^{N_r} d\theta$$

Symbol Error Rate in Frequency-Selective Fading Channels:

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

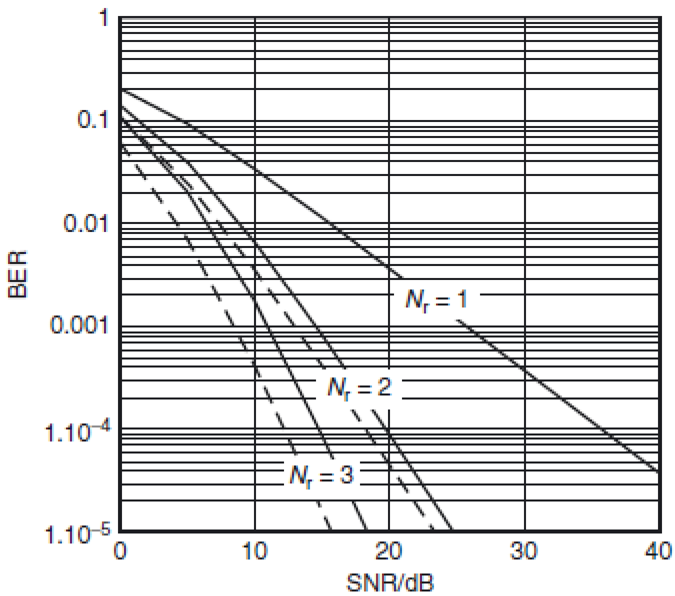


Fig:Bit error rate of minimum shift keying (MSK) with received-signal strength-indication-driven selection diversity (solid) and maximum ratio combining (dashed) as a function of the signal-to-noise ratio with Nr diversity antennas.

- We assume here FSK with differential phase detection.
- For binary FSK with selection diversity:

$$\overline{SER} = \frac{(2N_r - 1)!!}{2} \left(\frac{1 - |\rho_{XY}|^2}{2(\mathrm{Im}\{\rho_{XY}\})^2} \right)^{N_r}$$

- For binary FSK with MRC:

$$\overline{SER} = \frac{(2N_r - 1)!!}{2(N_r!)} \left(\frac{1 - |\rho_{XY}|^2}{2(\mathrm{Im}\{\rho_{XY}\})^2} \right)^{N_r}$$

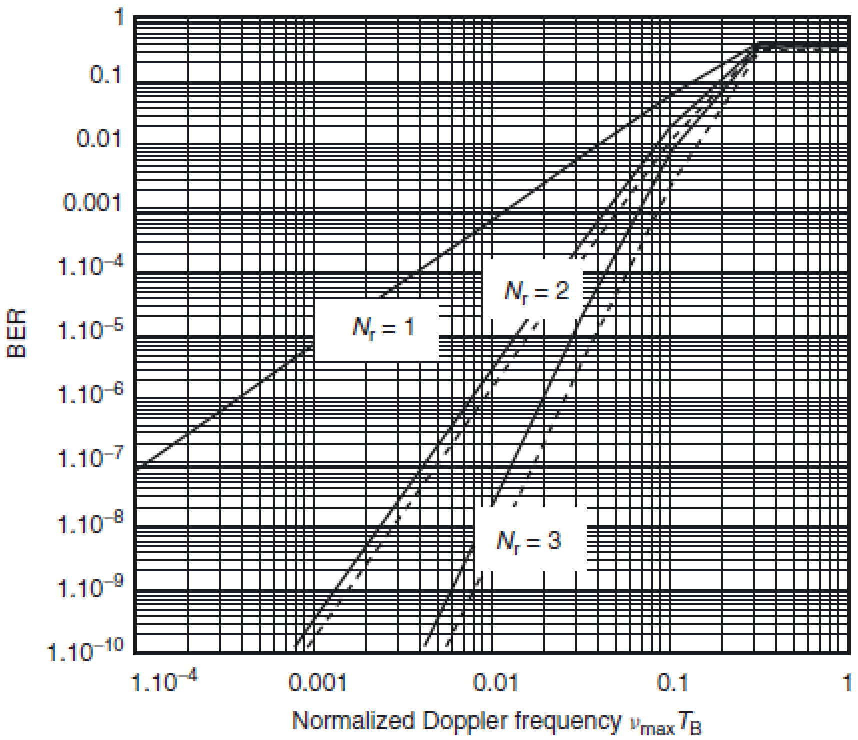


Fig: Bit error rate of MSK with received-signal-strength-indication-driven selection diversity (solid) and maximum ratio combining (dashed) as a function of the normalized Doppler frequency with *N*r diversity antennas.

UNIT IV MULTIPATH MITIGATION TECHNIQUES

1.(a)(i)Explain the principles of RAKE receiver in detail (April/May 2018,April/ May 2017, Nov/Dec 2016)

RAKE Receiver:

- In CDMA spread spectrum systems , the chip rate is typically much greater than the flat fading bandwidth of the channel.
- CDMA spreading codes are designed to provide very low correlation between successive chips
- If these multipath components are delayed in time by more than a chip duration, they appear like uncorrelated noise at a CDMA receiver, and equalization is not required.
- CDMA receivers may combine the time delayed versions of the original signal transmission in order to improve the signal to noise ratio at the receiver
- It attempts to collect the time-shifted versions of the original signal by providing a separate correlation receiver for each of the multipath signals.

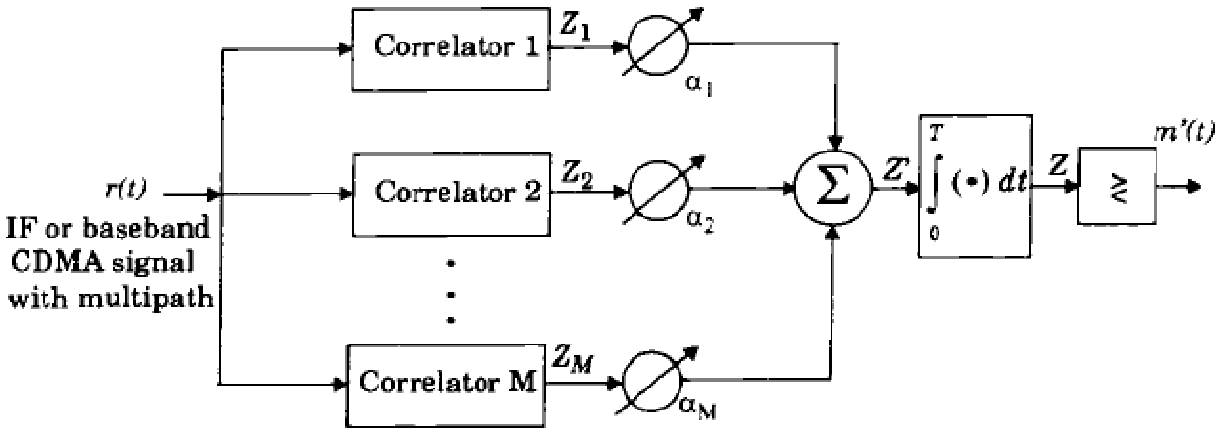


Fig: Rake Receiver

- A RAKE receiver utilizes multiple correlators to separately detect the M strongest multipath components
- The outputs of each correlator are weighted to provide a better estimate of the transmitted signal than is provided by a single component.
- Demodulation and bit decisions are then based on the weighted outputs of the M correlators
- Correlator 1 is synchronized to the strongest multipath m1.
- Multipath component m2 arrives τ1 later than component m1.
- Once the output of the single correlator is corrupted by fading, the receiver cannot correct the value
- The outputs of the M correlators are denoted as Z1, Z2 ,... and ZM. They are weighted by α1, α2,... αm respectively.
- The weighting coefficients are based on the power or the SNR from each correlator output.
- The overall signal Z' is given by

$$Z' = \sum_{m=1}^{M} \alpha_m Z_m$$

$$\alpha_m = \frac{Z_m^2}{\sum_{m=1}^{M} Z_m^2}$$

- Choosing weighting coefficients based on the actual outputs of the correlators yields better RAKE performance.

(ii)Assuming four branch diversity is used, where each branch receives an independent Rayleigh fading signal. If the *average SNR*is 20 dB, determine the probability that the *SNR will drop below* 10 dB. Compare this with the case of a single receiver without diversity. April/May 2017

$\gamma = 20\ dB$
$\tau = 10\ dB$
$\gamma / \tau = 0.1$

With Selection Diversity
$P_4(10\ dB) = [1 - e^{-0.1}]^4 = 0.000082$

Without Diversity
$P_1(10\ dB) = [1 - e^{-0.1}]^1 = 0.095$

(iii)Consider the design of the US Digital cellular equalizer, where f=900MHz and the mobile velocity v=80km/hr, determine the maximum Doppler shift, the coherence time of the channel and the maximum number of symbols that could be transmitted without updating the equalizer assuming that the symbol rate is 24.3 k symbols/sec. April/May 2017

Maximum Doppler Shift

$V = f_d\ \lambda$

$f_d = V/\lambda$
V= 80km/hr
$\lambda = C/f = 3*10^8$ m/s / $900*10^6$ m/s
f_d=66.67 Hz

Coherence Time

Tc= $\sqrt{\frac{9}{16\Pi\ f2d}}$ =0.423/66.67=6.34 msec

The Number of bits that can be sent without updating the equalizer
Nb=Rs * Tc=24300*0.00634=154 symbols

1.(b)Derive an expression for performance improvement due to Maximal Ratio combining April/May 2017

Maximum Ratio Combining

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

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

- Where αn is the (instantaneous) gain of diversity branch n.
- These signals at the different branches are multiplied with weights w∗n and added up, so that the SNR becomes

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

- where Pn is the noise power per branch (assumed to be the same in each branch).
- According to the Cauchy–Schwartz inequality, $\left|\sum_{n=1}^{N} w_n^* \alpha_n\right|^2 \le \sum_{n=1}^{N} |w_n^*|^2 \sum_{n=1}^{N} |\alpha_n|^2,$ where equality holds if and only if wn = αn. Thus, the SNR is maximized by choosing the weights as

$$w_{MRC} = \alpha_n$$

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

$$\gamma_{\mathrm{MRC}} = \sum_{n=1}^{N_r} \gamma_n$$

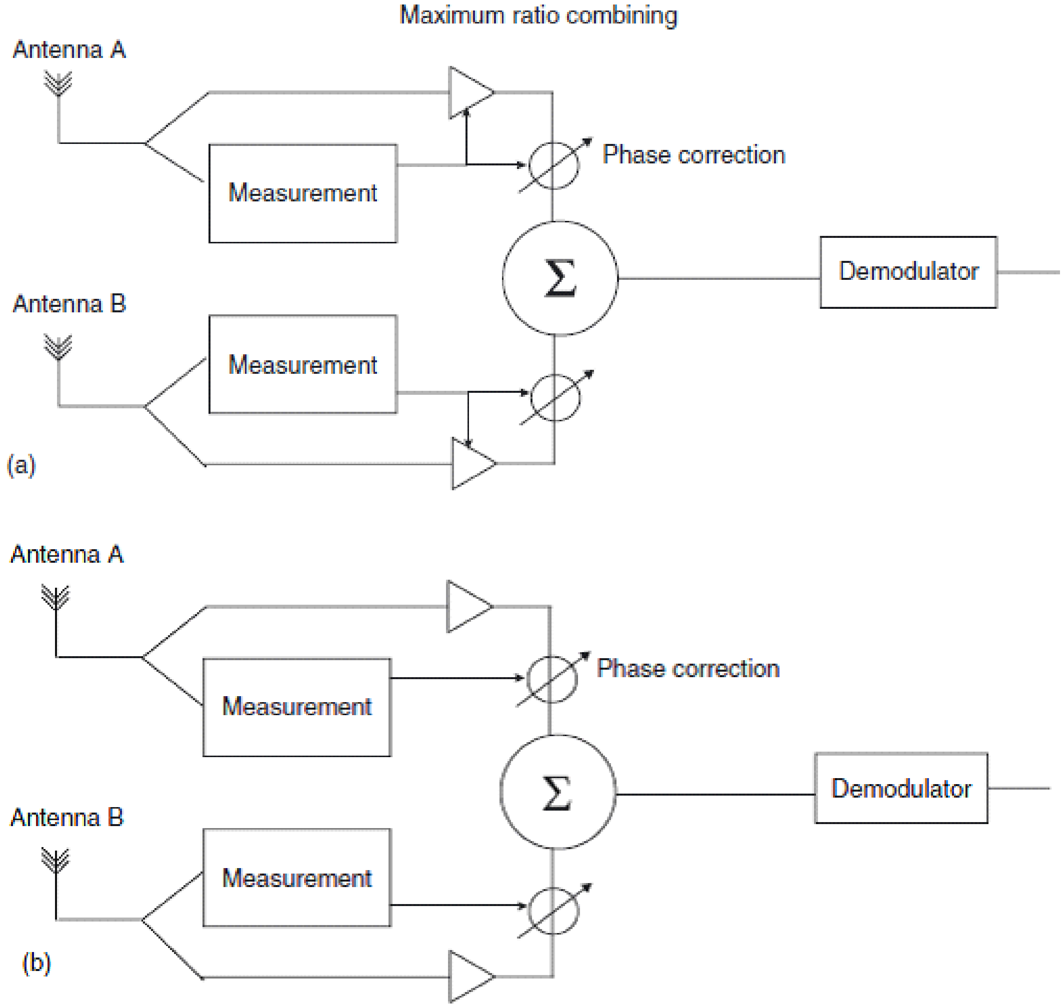


Fig: Combining diversity principle: (a) maximum ratio combining, (b) equal gain combining.

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

- If, the SNR distribution in each branch is exponential (corresponding to Rayleigh fading), and all branches have the same mean SNR γ n= γ , we find after some manipulations that

$$pdf_\gamma(\gamma) = \frac{1}{(N_r - 1)!} \frac{\gamma^{N_r - 1}}{\overline{\gamma}^{N_r}} \exp\left(-\frac{\gamma}{\overline{\gamma}}\right)$$

- and the mean SNR of the combined output is just the mean branch SNR, multiplied by the number of diversity branches:

$$\overline{\gamma}_{\mathrm{MRC}} = N_{\mathrm{r}}\overline{\gamma}$$

- Equal Gain Combining For EGC, we find that the SNR of the combined output is

$$\gamma_{\mathrm{ECG}} = \frac{\left(\sum_{n=1}^{N_{\mathrm{r}}} \sqrt{\gamma_n}\right)^2}{N_{\mathrm{r}}}$$

- Where we have assumed that noise levels are the same on all diversity branches. The mean SNR of the combined output can be found to be

$$\overline{\gamma}_{\mathrm{EGC}} = \overline{\gamma}\left(1 + (N_{\mathrm{r}} - 1)\frac{\pi}{4}\right)$$

- If all branches suffer from Rayleigh fading with the same mean SNR γ .
- The performance difference between EGC and MRC becomes bigger when mean branch SNRs are also different.), but become unwieldy very quickly.

2.Explain in detail the various factors to determine the algorithm for adaptive equalizer.Also derive the Least Mean square Algorithm for adaptive equalizer(Nov/Dec 2016)

The various factors to determine the algorithm for adaptive equalizer:

- **Rate of convergence —** This is defined as the number of iterations required for the algorithm, in response to stationary inputs, to converge close enough to the optimum solution.

- A fast rate of convergence allows the algorithm to adapt rapidly to a stationary environment of unknown statistics.

- **Misadjustment -** this parameter provides a quantitative measure of the amount by which the final value of the mean square error, averaged over an ensemble of adaptive filters, deviates from the optimal minimum mean square error.

- **Computational complexity** — This is the number of operations required to make one complete iteration of the algorithm.

- **Numerical properties** — When an algorithm is implemented numerically, inaccuracies are produced due to round-oft' noise and representation errors in the computer.

Least Mean Square Algorithm

- A more robust equalizer is the LMS equalizer where the criterion used is the minimization of the mean square error (MSE) between the desired equalizer output and the actual equalizer output.
- The prediction error is given by

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

$$e_k = x_k - y_k^T w_k = x_k - w_k^T y_k$$

- andTo compute the mean square error $|ek^2|$ at time instant k ,

$$\xi = E[e_k^* e_k]$$

- The LMS algorithm seeks to minimize the mean square error .
- For a specific channel condition, the prediction error ek is dependent on the tap gain vector W_N, so the MSE of an equalizer is a function of W_N.
- Let the cost function J (W_N) denote the mean squared error as a function of tap gain vector W_N.
- In order to minimize the MSE, it is required to set the derivative of equation to zero.

$$\frac{\partial}{\partial w_N} J(w_N) = -2p_N + 2R_{NN} w_N = 0$$

-

$$R_{NN} \hat{w}_N = p_N$$

-
- Simplifying equationEquation is called the normal equation, since the error is minimized and is made orthogonal (normal) to the projection related to the desired signal Xk. The MMSE of the equalizer is

$$J_{opt} = J(\hat{w}_N) = E[x_k x_k^*] - p_N^T \hat{w}_N$$

- To obtain the optimal tap gain vector then, the normal equation must be solved iteratively as the equalizer converges to an acceptably small value of Jopt .
- One obvious technique is to calculate

$$\hat{w} = R_{NN}^{-1} p_N$$

-
- The LMS algorithm is the simplest equalization algorithm and requires only 2N +1 operations per iteration. The filter weights are updated by the update equations given below. Letting the variable n denote the sequence of iterations, LMS is computed iteratively by

$$\hat{d}_k(n) = w_N^T(n) y_N(n)$$

$$e_k(n) = x_k(n) - \hat{d}_k(n)$$

$$w_N(n+1) = w_N(n) - \alpha e_k^*(n) y_N(n)$$

-
- where the subscript N denotes the number of delay stages in the equalizer, and α is the step size which controls the convergefice rate and stability of the algorithm. To prevent the adaptation from becoming unstable, the value of a is chosen from

$$0 < \alpha < 2 / \sum_{i=1}^{N} \lambda_i$$

- whereλ_i is the i th eigenvalue of the covariance matrix R_{NN}.
- Since

$$\sum_{i=1}^{N} \lambda_i = y_N^T(n) y_N(n)$$

- The step size a can be controlled by the total input power in order to avoid instability in the equalizer

3.(a) Describe in detail about linear and Non linear equalizers (Nov/ Dec 2017,April/May 2017)

Linear Equalizers:

- **Equalization** is a technique used to combat intersymbol interference.
- 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.**
- The general operating modes of an adaptive equalizer include training and tracking.
- A linear equalizer can be implemented as an **FIR filter, otherwise known as the transversal filter.**

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

- The current and past values of the received signal are linearly weighted by the filter coefficient and summed to produce the output

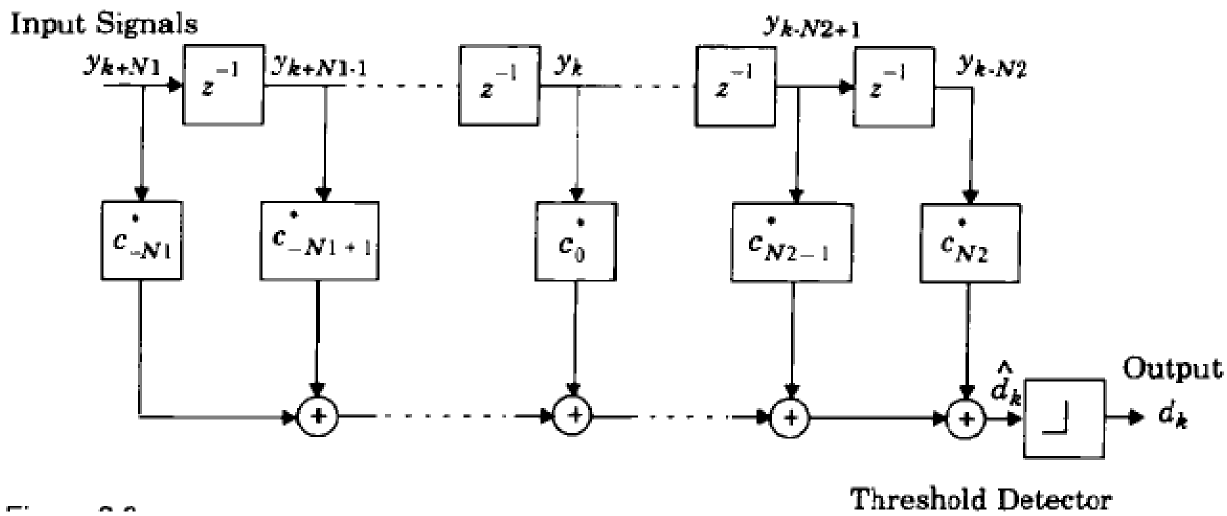


Figure Structure of a linear transversal equalizer.

- The output of this transversal filter before decision making (threshold detection) is

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

- **where Cn* represents** the complex filter coefficients or tap weights, dk^ is the output **at time index k, yi is the input received signal at time to + iT, t0 is the equalizer starting time, and N = N1 +N2 + I is the number of taps.**
- The values N1 and N2 denote the number of taps used in the forward and reverse portionsof the equalizer, respectively.
- The minimum mean squared error E [|e (n) |2] that a linear transversal equalizer can achieve is

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

- where F(ejwt) is the frequency response of the charmel, and N0 is the noise spectral density.
- The linear equalizer can also be implemented as a lattice filter.
- The input signal Yk is transformed into a set of N intermediate forward and backward error signals, fn(k) and bn(k) respectively, which are used as inputs to the tap multipliers and are used to calculate the updated coefficients.
- where Kn(k) is the reflection coefficient for the it th stage of the lattice.
- The backward error signals, bn are then used as inputs to the tap weights, and the output of the equalizer is given by

$$N$$

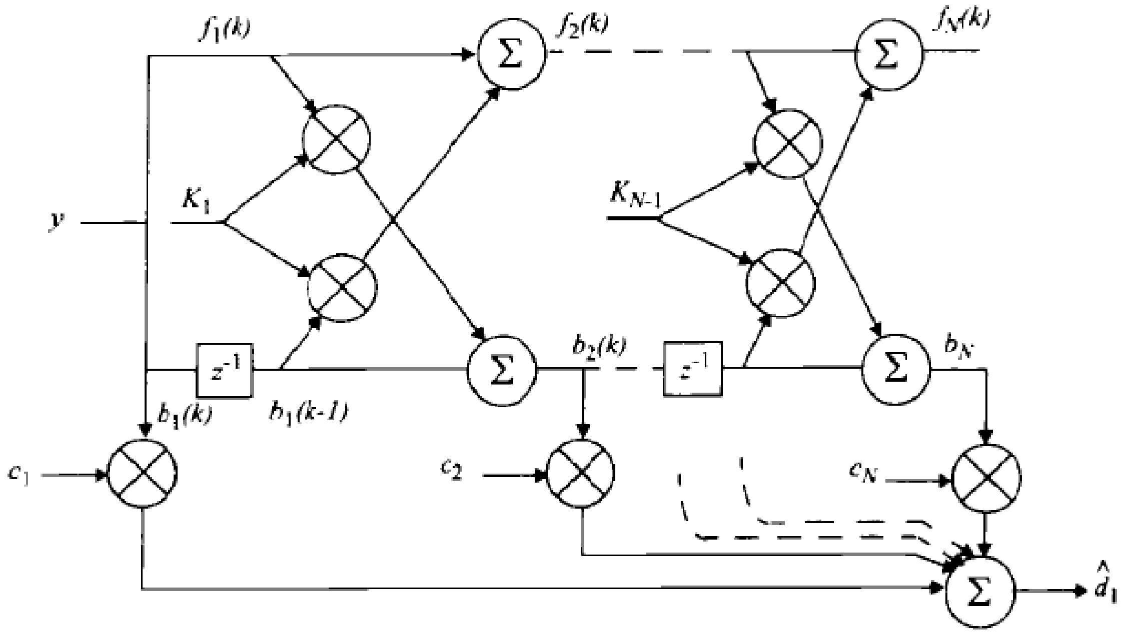
$$f_1(k) = b_1(k) = y(k)$$

$$f_n(k) = y(k) - \sum_{i=1}^{n} K_i y(k-i) = f_{n-1}(k) + K_{n-1}(k) b_{n-1}(k-1)$$

$$b_n(k) = y(k-n) - \sum_{i=1}^{n} K_i y(k-n+i)$$

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

-



Two main advantages of the lattice equalizer is its numerical stability and faster convergence.

Figure The structure of a lattice equalizer

Nonlinear Equalization:

- Nonlinear equalizers are used in applications where the channel distortion is too severe for a linear equalizer to handle.
- Linear equalizers do not perform well on channels which have deep spectral nulls in the passband
- Non linear methods 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 information symbol has been detected and decided upon, the 1ST that it induces on future symbols can be estimated and subtracted out before detection of subsequent symbols
- The DFE can be realized in either the direct transversal form or as a lattice filter.
- It consists of a feedforward filter (FFF) and a feedback filter (FBF).
- The equalizer has N1 + N2 + I taps in the feed forward filter and N3 taps in the feedback filter
- Its output can be expressed as:

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

-

- where cn*, and yn, are tap gains and the inputs, respectively, to the forward filter, Fi* are tap gains for the feedback filter,The minimum mean squared error a DFE can achieve is

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

- A DFE has significantly smaller minimum MSE than an LTE.

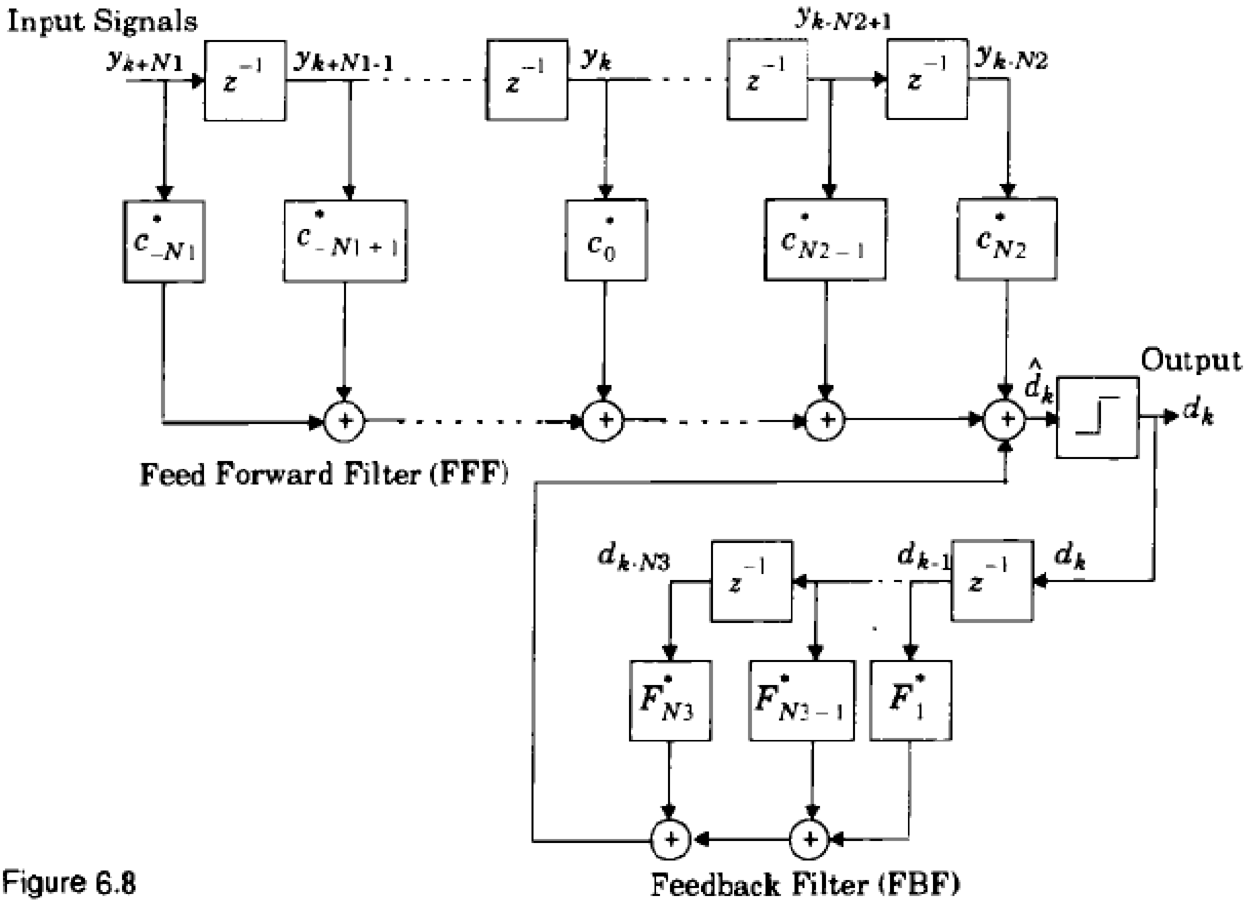


Figure 6.8
Decision feedback equalizer (DFE).

- The lattice implementation of the DFE is equivalent to a transversal DFE having a feed forward filter of length N1 and a feedback filter of length N2, where N1 > N2.

- Another form of DFE proposed by Belfiore and Park is called a predictive DFE,.

- It also consists **of** a feed forward filter (FFF) as in the conventional DFE.

- The feedback filter (FBF) is driven by an input sequence formed by the difference of the output of the detector and the output of the feed forward filter.

- Hence, the FBF here is called a noise predictor because it predicts the noise and the residual ISI contained in the signal at the FFF output and subtracts from it the detector output after some feedback delay.

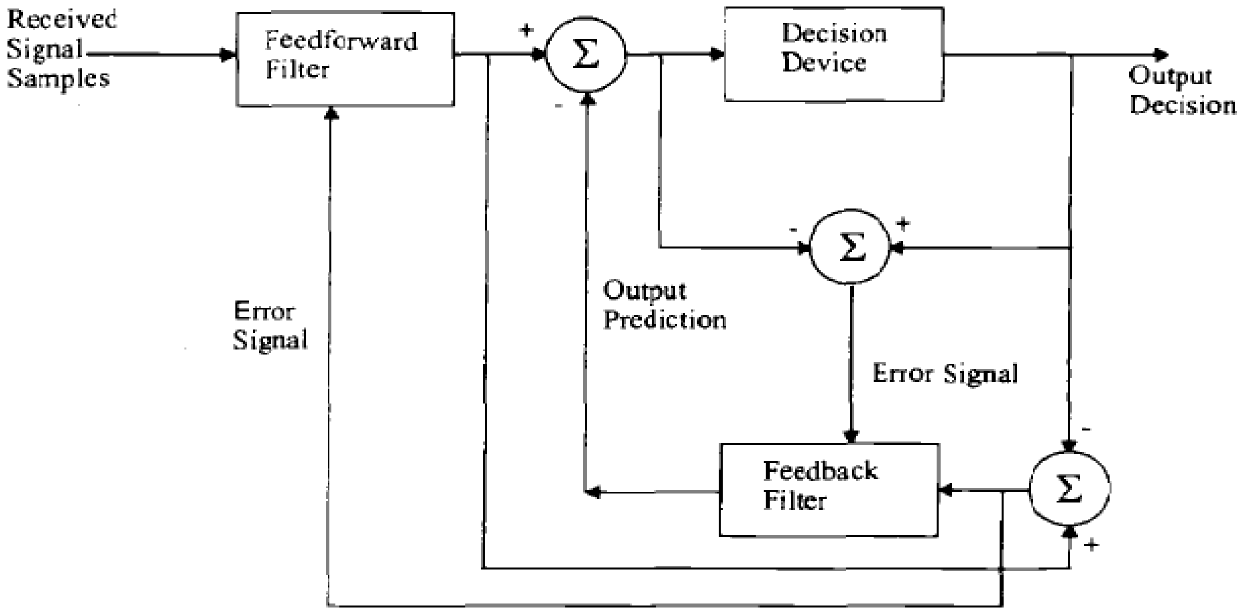


Fig:Predictive decision feedback equalizer.

Maximum Likelihood Sequence Estimation (MLSE) Equalizer:

- These equalizers use various forms of the classical maximum likelihood receiver structure.

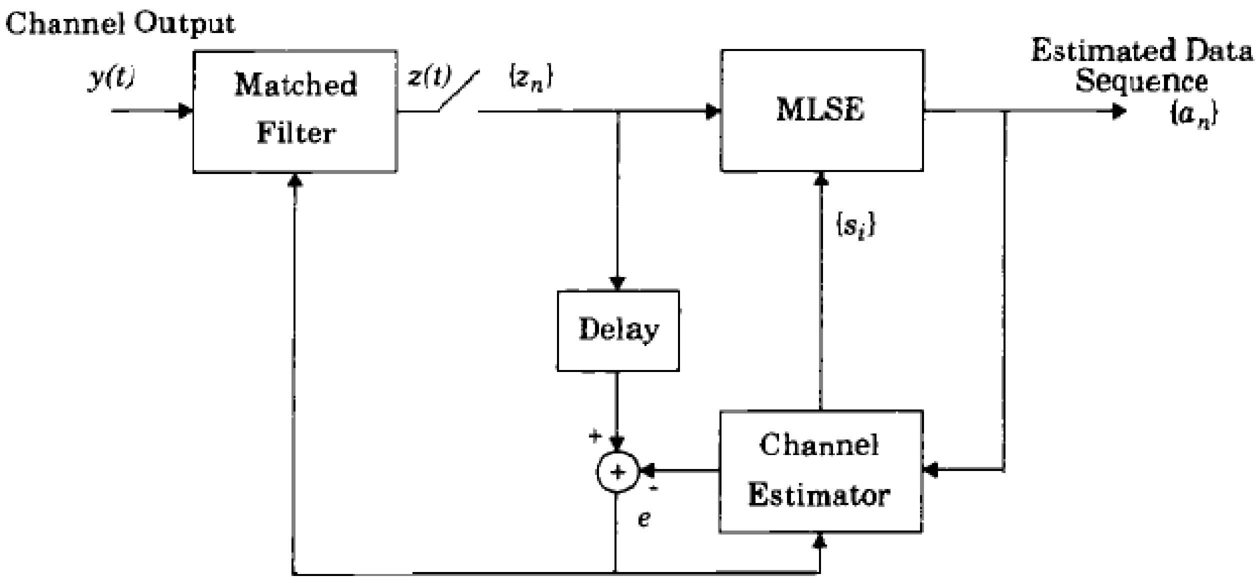


Fig: Structure of an MLSE estimator

- Using a channel impulse response simulator within the algorithm, the MLSE tests all possible data sequences, and chooses the data sequence with the maximum probability as the output.

- An MLSE usually has a large computational requirement, especially when the delay spread of the channel is large.
- MLSE estimator structure and implemented it with the Viterbi algorithm

- The channel has M^L states, where M is the size of the symbol alphabet of the modulation.
- **That is, an M^L trellis is used by the receiver to model the channel over time.**

(b)Analyze various diversity techniques used in wireless communication (Nov/ Dec 2017,April/May 2017)

Diversity Techniques:

- Diversity is a powerful communication receiver technique that provides wireless link improvement at relatively low cost
- Diversity requires no training overhead since a training sequence is not required by the transmitter.
- By selecting the best signal at all times, a receiver can mitigate small-scale fading effects (this is called antenna diversity or space diversity).
- Large-scale fading is caused by shadowing due to variations in both the terrain profile and the nature of the surroundings.
- By selecting a base station which is not shadowed when others are, the mobile can improve substantially the average Signal to noise ratio on the forward link.
- This is called macroscopic diversity.

Space 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 site, multiple base station receiving antennas are used to provide diversity reception
- Separations on the order of several tens of wavelengths are required at the base station

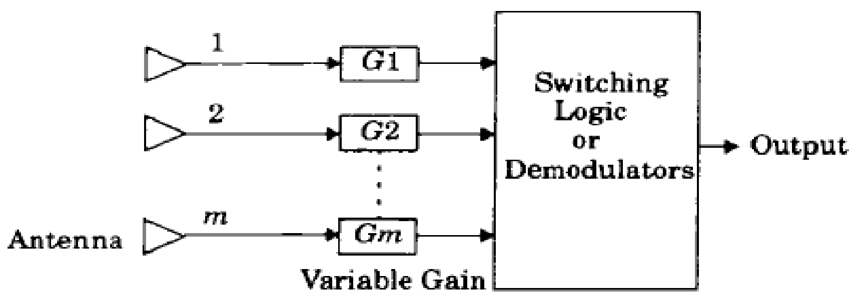


Fig:Generalized block diagram for space diversity.

- 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 antenna signals themselves could be sampled and the best one sent to a single demodulator

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.
- This signal is then received until it falls below threshold and the scanning process is again initiated.

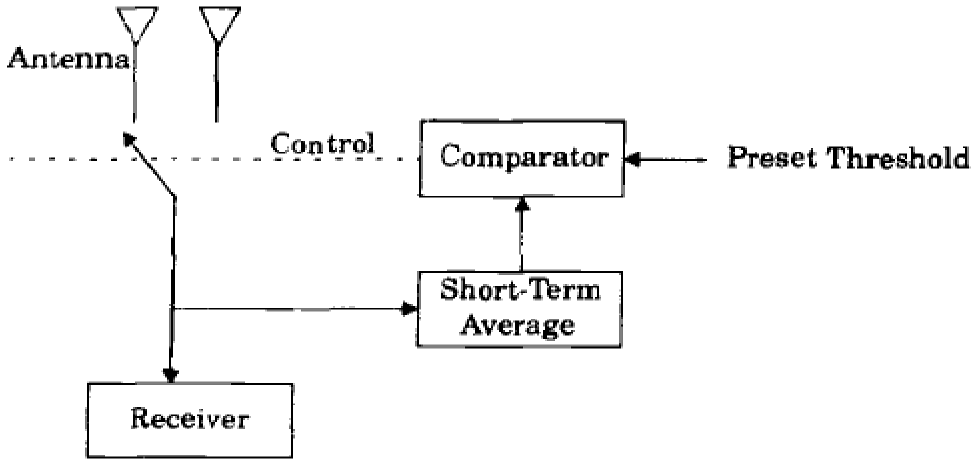


Fig:Basic form of scanning diversity

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.

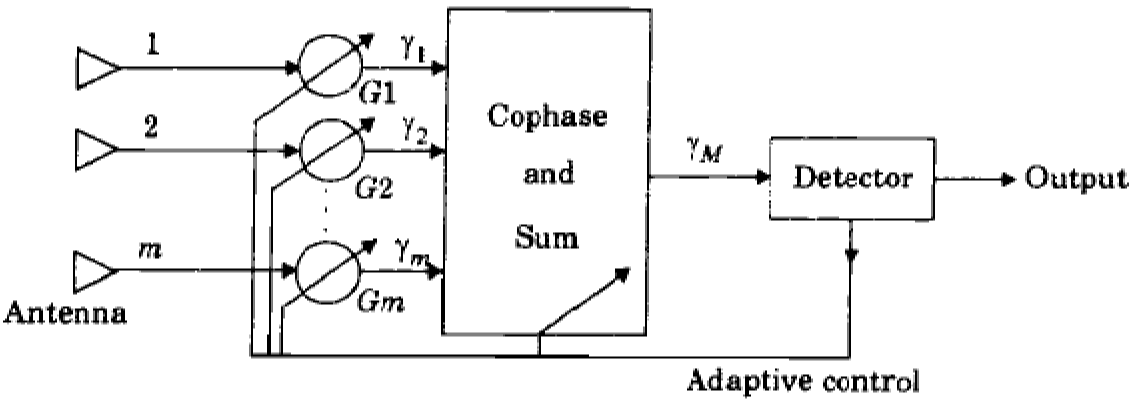


Fig:Maximal ratio combiner

- The individual signals must be co-phased before being summed which generally requires an individual receiver and phasing circuit for each antenna element.

- Maximal ratio combining produces an output SNR equal to the sum of the individual SNRs.

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.

Polarization Diversity:

- The comparatively high cost of using space diversity at the base station prompts the consideration of using orthogonal polarization to exploit polarization diversity
- Measured horizontal and vertical polarization paths between a mobile and a base station are reported to be uncorrelated by Lee and Yeh
- Circular and linear polarized antennas have been used to characterize multipath inside buildings

Frequency Diversity:

- Frequency diversity transmits information on more than one carrier frequency.
- Frequency diversity is often employed in microwave line-of-sight links which carry several channels in a frequency division multiplex mode
- This technique has the disadvantage that it not only requires spare bandwidth but also requires that there be as many receivers as there are channels used for the frequency diversity

Time Diversity:

- Time diversity repeatedly transmits information at time spacings that exceed the coherence time of the channel, so that multiple repetitions of the signal will be received with independent fading conditions, thereby providing for diversity.
- One modem implementation of time diversity involves the use of the RAKE receiver for spread spectrum CDMA, where the multipath channel provides redundancy in the transmitted message.

UNIT V MULTIPLE ANTENNA TECHNIQUES

1.(a)Describe in detail the capacity in fading and non fading channels (April/ May 2017,(Nov/Dec 2016,Nov/Dec 2015)

Capacity in Nonfading Channels:

- MIMO systems in nonfading channels, often known as “Foschini’s equation”
- The information-theoretic (ergodic) capacity of such a channel is

$$C_{\text{shannon}} = \log_2\left(1 + \gamma \cdot |H|^2\right)$$

- where γ is the SNR at the RX, and H is the normalized transfer function from the TX to the RX
- Let us then consider a singular value decomposition of the channel:

$$\mathbf{H} = \mathbf{W\Sigma U}^{\dagger}$$

- where Σ is a diagonal matrix containing singular values, and W and U† are unitary matrices composed of the left and right singular vectors, respectively.
- The received signal is then

$$\begin{aligned}\mathbf{r} &= \mathbf{Hs} + \mathbf{n}\\ &= \mathbf{W\Sigma U}^{\dagger}\mathbf{s} + \mathbf{n}\end{aligned}$$

- The capacity of channel H is thus given by the sum of the capacities of the eigenmodes of the channel:

$$C = \sum_{k=1}^{R_{\mathrm{H}}} \log_2\left[1 + \frac{P_k}{\sigma_{\mathrm{n}}^2}\sigma_k^2\right]$$

- Where σn2 is noise variance, and Pk is the power allocated to the kth eigenmode

No Channel State Information at the Transmitter and Full CSI at the Receiver:

- When the RX knows the channel perfectly, but no CSI is available at the TX
- Capacity thus takes on the form:

$$C = \log_2\left[\det\left(\mathbf{I}_{N_r} + \frac{\overline{\gamma}}{N_t}\mathbf{HH}^{\dagger}\right)\right]$$

- Let us now look at a few special cases. To make the discussion easier, we assume that Nt = Nr = N:

  1. All transfer functions are identical – i.e., h1,1 = h1,2 = . . . = hN,N.

$$C_{\text{MIMO}} = \log_2(1 + N\overline{\gamma})$$

2. All transfer functions are different such that the channel matrix is full rank, and has N eigenvalues of equal magnitude

$$C_{\text{MIMO}} = N\log_2(1 + \overline{\gamma})$$

3. Parallel transmission channels – e.g., parallel cables

$$C_{\text{MIMO}} = N \log_2\left(1 + \frac{\overline{\gamma}}{N}\right)$$

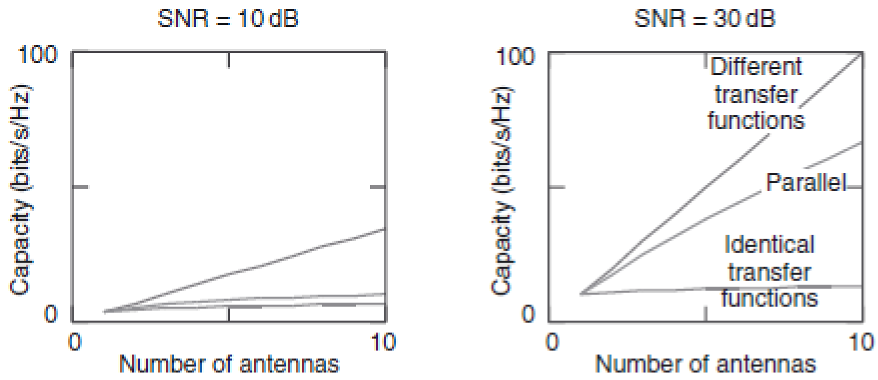


Fig: Capacity of multiple-input-multiple-output systems in additive white Gaussian noise channels.

Full Channel State Information at the Transmitter and Full CSI at the Receiver:

- Let us next consider the case where both the RX and TX know the channel perfectly.
- waterfilling. This is another nice example of how the same mathematics can be applied to different communications problems: replace the word "subchannel" or "subcarrier in an Orthogonal Frequency Division Multiplexing (OFDM) system" by "eigenmode in a MIMO system,"

Capacity in Flat-Fading Channels

General Concepts

- If the channel is Rayleigh fading, and fading is independent at different antenna elements, the hij are iid zero-mean, circularly symmetric complex Gaussian random variables with unit variance – i.e., the real and imaginary part each has variance 1/2.
- The power carried by each hij is chi-square-distributed with 2 degrees of freedom.
- Ergodic (Shannon) capacity: this is the expected value of the capacity, taken over all realizations of the channel. This quantity assumes an infinitely long code that extends over all the different channel realizations.
- Outage capacity: this is the minimum transmission rate that is achieved over a certain fraction of the time – e.g., 90% or 95%. Assume that data are encoded with a near-Shannon-limitachieving code that extends over a period that is much shorter than the channel coherence time.

No Channel State Information at the Transmitter and Perfect CSI at the Receiver:

- The exact expression for the ergodic capacity as

$$E\{C\} = \int_0^{\infty} \log_2\left[1 + \frac{\overline{\gamma}}{N_t}\lambda\right] \sum_{k=0}^{m-1} \frac{k!}{(k+n-m)!} \left[L_k^{n-m}(\lambda)\right]^2 \lambda^{n-m} \exp(-\lambda)\, d\lambda$$

- The (1, 1) curve describes a Single Input Single Output (SISO) system. the median capacity is on the order of 6 bit/s/Hz,but the 5% outage capacity.
- When using a (1, 8) system – i.e., 1 transmit antenna and 8 receive antennas – the mean capacity– from 6 to 10 bit/s/Hz, 5% outage capacity
- when going to a (8, 8) system – i.e., a system with 8 transmit and 8 receive antennas – both capacities increase dramatically: the mean capacity is on the order of 46 bit/s/Hz,and the 5% outage probability

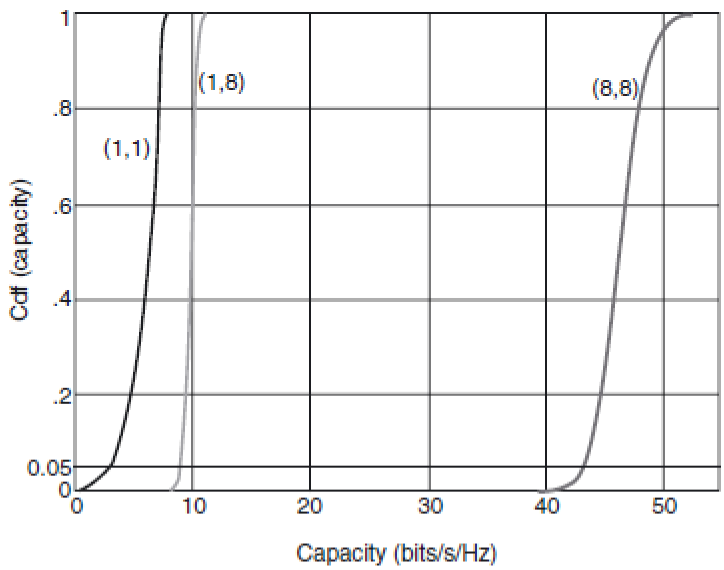


Fig: Cumulative distribution function of capacity for 1 × 1, 1 × 8, and the 8 × 8 optimum scheme

1.(b)(i)Describe the concepts of precoding and Beamforming (April/ May 2017, Nov/Dec 2017)

Linear Precoding – Uplink

- The simplest practical implementation of multiuser MIMO processing is based on linear processing at TX and RX.
- The RX (the BS) has all signals available, and can thus perform optimum (linear) processing for interference suppression.
- The details depend on the optimization criterion; in the following we consider the minimization of the overall Mean Square Error (MSE).
- The MSE can be written as

$$\mathrm{MSE} = tr\left\{ \sum_{k=1}^{K} \left\{ \sum_{j=1}^{K} (\mathbf{T}^{(j)})^{\dagger} (\mathbf{H}^{(j)})^{\dagger} \mathbf{W}^{(k)} (\mathbf{W}^{(k)})^{\dagger} \mathbf{H}^{(j)} \mathbf{T}^{(j)} - (\mathbf{T}^{(k)})^{\dagger} (\mathbf{H}^{(k)})^{\dagger} \mathbf{W}^{(k)} \right.\right.$$
$$\left.\left. - (\mathbf{W}^{(k)})^{\dagger} \mathbf{H}^{(k)} \mathbf{T}^{(k)} + \mathbf{I} + \sigma_n^2 (\mathbf{W}^{(k)})^{\dagger} \mathbf{W}^{(k)} \right\}\right\}$$

-

$$tr\left\{(\mathbf{T}^{(k)})^{\dagger}\mathbf{T}^{(k)}\right\} \leq P_k^{\max}$$

-
- where (W(k))† is the receive matrix for the k-th user. The goal is then to minimize this MSE under the power constraintsThe TX weights are functions of the receive weights of all users, while the optimum RX weights depend on the transmit weights of all users. Thus, the RX weights and transmit weights can be computed with the following iteration:

1. Update for all users (k = 1, . . .K)

$$(\mathbf{W}^{(k)})^{\dagger} = (\mathbf{T}^{(k)})^{\dagger}(\mathbf{H}^{(k)})^{\dagger}\left[\sigma_n^2\mathbf{I}+\sum_{j=1}^{K}\mathbf{H}^{(j)}\mathbf{T}^{(j)}(\mathbf{T}^{(j)})^{\dagger}(\mathbf{H}^{(j)})^{\dagger}\right]^{-1}$$

$$\mathbf{X}^{(k)}(\mu_k') = \left[\mu_k'\mathbf{I}+\sum_{j=1}^{K}(\mathbf{H}^{(k)})^{\dagger}\mathbf{W}^{(j)}(\mathbf{W}^{(j)})^{\dagger}\mathbf{H}^{(k)}\right]^{-1}(\mathbf{H}^{(k)})^{\dagger}\mathbf{W}^{(k)}$$

$$\mu_k = \max\left[\arg_{\mu_k'}\left(tr\left\{\mathbf{X}^{(k)}(\mu_k')\left(\mathbf{X}^{(k)}(\mu_k')\right)^{\dagger}\right\} = P_k^{\max}\right), 0\right]$$

$$\mathbf{T}^{(k)} = \left[\mu_k\mathbf{I}+\sum_{j=1}^{K}(\mathbf{H}^{(k)})^{\dagger}\mathbf{W}^{(j)}(\mathbf{W}^{(j)})^{\dagger}\mathbf{H}^{(k)}\right]^{-1}(\mathbf{H}^{(k)})^{\dagger}\mathbf{W}^{(k)}$$

- 2. Update for all users (k = 1, . . .K)Typically, 10 to 20 iterations are sufficient for convergence

Linear Precoding – Downlink

- Also for the downlink, the linear precoding is the method that can be implemented most easily.
- In the case of beamforming (linear precoding) at the TX, the total transmit signal intended for the k-th user is the source signal, multiplied with a beamforming matrix T(k)

$$\tilde{\mathbf{s}}^{(k)} = \mathbf{T}^{(k)}\mathbf{s}^{(k)}$$

-
- where the T(k) is the precoding matrix (beamformer) for the k-th user. The correlation matrix of the k-th user's signal is

$$\mathbf{R}_{\tilde{s}\tilde{s}}^{(k)} = E\{\tilde{\mathbf{s}}^{(k)}\tilde{\mathbf{s}}^{(k)\dagger}\},$$

-

$$\sum_k P_k \leq P_{\max}.$$

- and the power allocated to the k-th user is Pk = tr{R(k)} =Trace{T(k)T(k)†}, since we assume that the modulation symbols have unit energy Rss (k)= I. Usually the BS has a constraint on the total transmit powerBlock

DiagonalizationConsider the case where each MS has multiple antenna elements (N(k)r ≥ 1) and the BS transmitsN(k)r data streams to each user.

- A simple solution can be achieved by imposing the dimensionality constraint

$$\sum_k N_r^{(k)} = N_t.$$

- This constraint is very much along the lines of interpreting the multiple MSs as a “distributed array,” and the dimensionality constraint ensures that the number of data streams does not exceed the number of RX chains that can demodulate the signal.

$$\sum_k N_r^{(k)} = N_t.$$

- Keeping the data streams of different MSs apart is achieved by a block diagonalization technique. Define for each user an interference channel matrix

$$\widetilde{\mathbf{H}}^{(\mathbf{k})} = [(\mathbf{H}^{(1)})^T, \ldots (\mathbf{H}^{(k-1)})^T (\mathbf{H}^{(k+1)})^T, \ldots (\mathbf{H}^{(K)})^T]^T$$

-
- Any precoding matrix T(k) that lies in the nullspace of streams intended for the other MSs do not influence the k-th MS. Consequently, there is now a new dimensionality constraint.Let Jk be the rank of achieved if

$$N_t > \max_k (J_1, J_2, \ldots J_K)$$

- **If the propagation channels are all full rank, this means that the number of transmit antennas must be no smaller than the number of data streams.**

$$\widetilde{\mathbf{H}}^{(\mathbf{k})} = \widetilde{\mathbf{U}}^{(\mathbf{k})} \widetilde{\Sigma}^{(k)} [\widetilde{\mathbf{V}}_{\mathrm{fc}}^{(k)} \quad \widetilde{\mathbf{V}}_{\mathrm{lc}}^{(k)}]^\dagger$$

Define then the SVD of H(k)

- where Vfc (k) contains the first Jk and lc the last Nt − Jk right singular vectors; fc thus forms an orthonormal basis of the nullspace of H(k)
- For the k-th block,

$$\widehat{\mathbf{H}}^{(k)} = \widehat{\mathbf{U}}^{(\mathbf{k})} \begin{bmatrix} \widehat{\Sigma}^{(k)} & \mathbf{0} \\ \mathbf{0} & \mathbf{0} \end{bmatrix} [\widehat{\mathbf{V}}_{\mathrm{fc}}^{(k)} \quad \widehat{\mathbf{V}}_{\mathrm{lc}}^{(k)}]^\dagger \qquad (20.99)$$

- **The total beamforming/power allocation matrix T** is then

$$\mathbf{T} = \left[\widetilde{\mathbf{V}}_{\mathrm{lc}}^{(1)} \widehat{\mathbf{V}}_{\mathrm{fc}}^{(1)} \quad \widetilde{\mathbf{V}}_{\mathrm{lc}}^{(2)} \widehat{\mathbf{V}}_{\mathrm{fc}}^{(2)} \quad \ldots\ldots \quad \widetilde{\mathbf{V}}_{\mathrm{lc}}^{(K)} \widehat{\mathbf{V}}_{\mathrm{fc}}^{(K)} \right] \Lambda^{1/2} \qquad (20.100)$$

- where ^is a diagonal matrix that performs waterfilling on the elements of

$$\widehat{\mathbf{H}} = \begin{bmatrix} \mathbf{H}^{(1)} \widetilde{\mathbf{V}}_{\mathrm{lc}}^{(1)} & \mathbf{0} & \mathbf{0} \\ \mathbf{0} & .. & \mathbf{0} \\ \mathbf{0} & \mathbf{0} & \mathbf{H}^{(K)} \widetilde{\mathbf{V}}_{\mathrm{lc}}^{(K)} \end{bmatrix}$$

Coordinated Beamforming

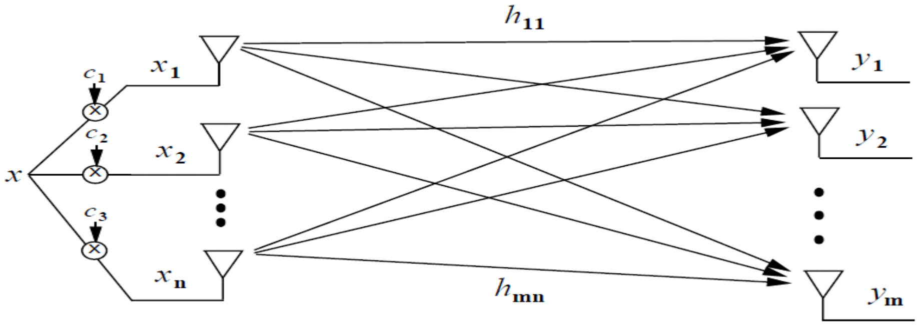
- Note that if the RX has a linear filter, then **H**(j) should be replaced by the concatenation of filter and channel matrix. Consequently, the rank of **H(k)** is influenced by all the RX filters.
- Let signals at the k-th MS be multiplied with a matrix **W**(k)†. The overall received signal is then

$$\mathbf{r}^{(k)} = \mathbf{W}^{(k)\dagger}\mathbf{H}^{(k)}\mathbf{T}^{(k)}\mathbf{s}^{(k)} + \mathbf{W}^{(k)\dagger}\mathbf{H}^{(k)}\sum_{l\neq k}\mathbf{T}^{(l)}\mathbf{s}^{(l)} + \mathbf{W}^{(k)\dagger}\mathbf{n}$$

- The second term on the r.h.s. now describes the interference.

Beamforming:

- Beamforming or spatial filtering is a signal processing technique used in sensor arrays for directional signal transmission or reception.
- This is achieved by combining elements in a phased array in such a way that signals at particular angles experience constructive interference while others experience destructive interference Adaptive beamforming is used to detect and estimate the signal-of-interest at the output of a sensor array by means of optimal spatial filtering and interference rejection.



MIMO Channel with Beamforming.

- Transmitter does not know the instantaneous channel
- It is no longer possible to transform the MIMO channel into non-interfering SISO channels.
- Since the decoding complexity is exponential in r, we can keep the complexity low by keeping r small.

$$\mathbf{r}^{(k)} = \mathbf{W}^{(k)\dagger}\mathbf{H}^{(k)}\mathbf{T}^{(k)}\mathbf{s}^{(k)} + \mathbf{W}^{(k)\dagger}\mathbf{H}^{(k)}\sum_{l\neq k}\mathbf{T}^{(l)}\mathbf{s}^{(l)} + \mathbf{W}^{(k)\dagger}\mathbf{n}$$

-

When r = 1, the input covariance matrix has unit rank and the process is called beamforming

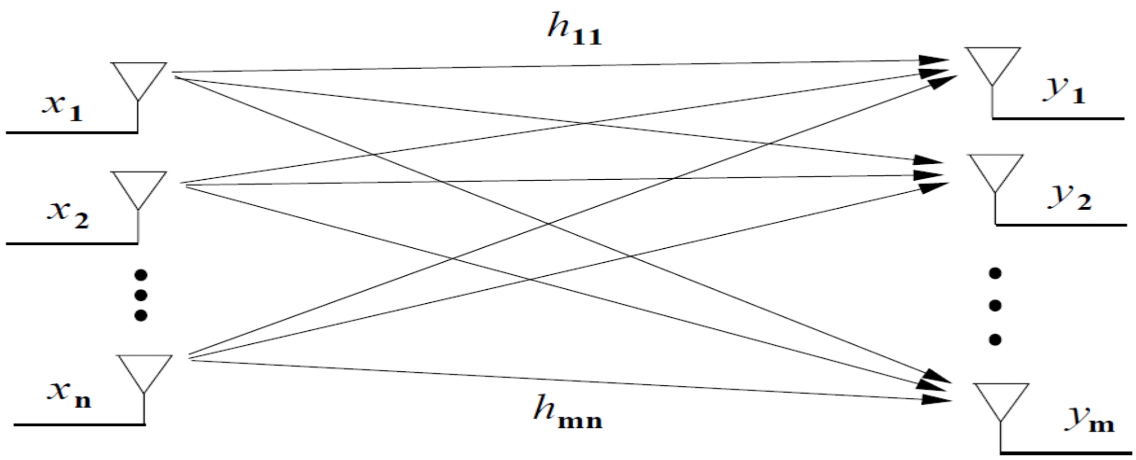
(ii)Describe MIMO systems with emphasis on their requirement in a wireless communication environment (April/ May 2017)

- MIMO systems are defined as point-to-point communication links with multiple antennas at both the transmitter and receiver.

- The use of multiple antennas at both transmitter and receiver clearly provide enhanced performance

- The cost of deploying multiple antennas, the space requirements of these extra antennas (especially on small handheld units), and the added complexity required for multi-dimensional signal processing are the challenges faced by MIMO Systems

- Spatial multiplexing is a transmission technique in MIMO wireless communication to transmit independent and separately encoded data signals, from each of the multiple transmit antennas. Therefore, the space dimension is reused, or multiplexed, more than one time.



MIMO Systems.

2.(a) With a neat diagram explain the system model for multiple input multiple output system(Nov/Dec 2016, Nov/Dec 2017)

System Model

- At the TX, the data stream enters an encoder, whose outputs 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 if not stated otherwise.
- By quasi-static we mean that the coherence time of the channel is so long that “a large number” of bits can be transmitted within this time.

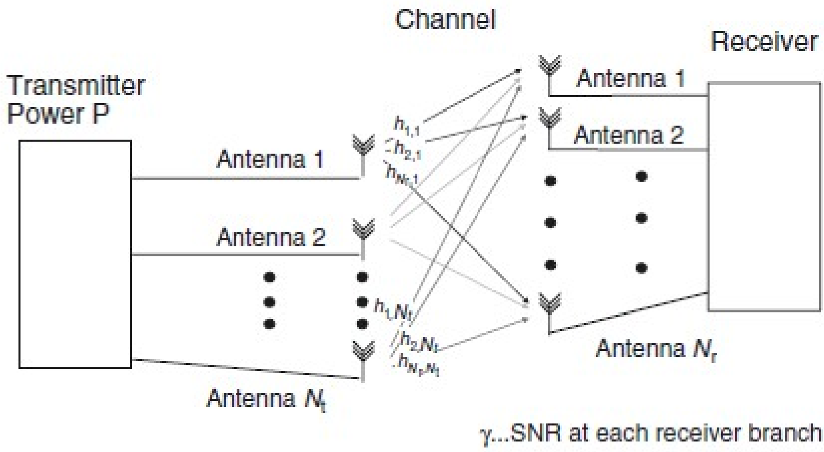


Fig: Block diagram of a multiple-input multiple-output system.

- We denote the Nr × Nt matrix of the channel as

$$\mathbf{H} = \begin{pmatrix} h_{11} & h_{12} & \cdots & h_{1N_t} \\ h_{21} & h_{22} & \cdots & h_{2N_t} \\ \vdots & \vdots & \cdots & \vdots \\ h_{N_r1} & h_{N_r2} & \cdots & h_{N_rN_t} \end{pmatrix}$$

- whose entries hij are complex channel gains (transfer functions) from the j th transmit to the ith receive antenna.
- The received signal vector

$$\mathbf{r} = \mathbf{H}\mathbf{s} + \mathbf{n} = \mathbf{x} + \mathbf{n}$$

- contains the signals received by Nr antenna elements, where **s** is the transmit signal vector and **n** is the noise vector.

3.(b) (i)What is known as channel state information? Explain in detail(Nov/Dec 2015)

Channel state information:

- Channel state information refers to the known channel properties of communication link.it describes how a signal propagates from the transmitter to the receiver and represents the combined effect of scattering ,fading and power delay with distance.

1.Full CSI at the TX (CSIT) and full CSI at the RX (CSIR)

- This ideal case, both the TX and the RX have full and perfect knowledge of the channel.
- This case results in the highest possible capacity.
- However, 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

***3.No CSIT and full CSIR*:**

- this is the case that can be achieved most easily, without any feedback or calibration.
- The TX simply does not use any CSI, while the RX learns the instantaneous channel state from a training sequence or using blind estimation.

4. *Noisy CSI* :

- when we assume “full CSI” at the RX, this implies that the RX has learned the channel state perfectly.
- However, any received training sequence will be affected by additive noise as well as quantization noise.

5.No CSIT and no CSIR: Neither has any knowledge about channel