MODEL TEST PAPER

Model Test Paper

B.Tech. V Sem. Examination Paper
Computer Science and Engineering
5CS5-11 Wireless Communication

Time : 2 Hours Maximum Marks : 80

Instructions to Candidates :
Attempt all five questions from Part A, four questions out of six questions from Part B and two questions out of three questions from Part C. (Schematic diagrams must be shown wherever necessary). Any data you feel missing suitably be assumed and stated clearly. Units of quantities used/calculated must be stated clearly.

Part - A

(Answer should be given up to 25 words only). All questions are Compulsory.

(5 × 2 = 10)

1. For a channel with Doppler spread $B_d = 80$Hz, what time separation is required in samples of the received signal such that the samples are approximately independent.
2. Mention the types of multiple access techniques.
3. Why GMSK is preferred for multiuser in cellular communication?
4. What is Macro diversity? Depending on the spatial degree of freedom (DoF) of the system what the user may transmit or receive?
5. Distinguish ergodic capacity and outage capacity of a flat fading channel?

Part - B

(Analytical/Problem solving questions). Attempt any four questions.

(4 × 10 = 40)

1. What is Two-Ray Model? Determine the critical distance for the two-ray model in an urban microcell ($h_t = 10$ m, $h_r = 3$m) and an indoor microcell ($h_t = 3$m, $h_r = 2$m) for $f_c = 2$GHz.
2. Explain Multiple Access via Frequency Division Multiple Access.
3. Compare the spectral efficiency of MSK and QPSK with rectangular constituent pulses. Consider systems with equal bit duration. Compute the out-of-band energy at $1/T_B$, $2/T_B$, and $3/T_B$.
4. Explain the mathematical implementation of linear equalizers.
5. What is the existence of Zero-Forcing Equalizer in Liner Equalizer?
6. Define MIMO (multiple input, multiple output).

Part - C

(Descriptive/Analytical/Problem Solving/Design question). Attempt any two questions.

(2 × 15 = 30)

1. What is parameters of Mobile Multipath Channels. Explain its type.
2. Explain the term Time Division Multiple Access(TDMA) and how it is different from FDMA.
3. Write down the difference between QPSK and π/4-Differential Quadrature-Phase Shift Keying.

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PART-A

Q.1 ***For a channel with Doppler spread $B_d = 80Hz$, what time separation is required in samples of the received signal such that the samples are approximately independent.***

Ans. The coherence time of the channel is $T_c \approx 1/B_d = 1/80$, so samples spaced 12.5 ms apart are approximately uncorrelated and thus, given the Gaussian properties of the underlying random process, these samples are approximately independent.

CELLULAR ARCHITECTURE 2

IMPORTANT QUESTIONS

PART-A

Q.1 *What is multiple access?*

Ans. Multiple Access: Multiple access is a signal transmission situation in which two or more users wish to simultaneously communicate with each other using the same propagation channel.

Q.2 *Write the applications of multiple access methods.*

Ans. The multiple access methods are used in
- (i) Satellite networks
- (ii) Cellular and mobile communication networks
- (iii) Military communication
- (iv) Underwater acoustic networks

Q.3 *Mention the types of multiple access techniques.*

Ans. Types of Multiple Access Techniques:
- (i) Frequency division multiple access (FDMA)
- (ii) Time division multiple access (TDMA)
- (iii) Code division multiple access (CDMA)
- (iv) Space division multiple access (SDMA)

Q.4 Define FDMA.

Ans. FDMA: In FDMA, the total bandwidth is divided into non-overlapping frequency sub bands. Each user is allocated a unique frequency sub band (channels) for the duration of the connection, whether the connection is in an active or idle state.

Q.5 *What are the application of FDMA?*

Ans. FDMA is mostly used for the following applications:
- **Analog communications systems:** FDMA is the only practicable multiple access method.
- **Combination of FDMA with other multiple access methods:** The spectrum allocated for a service (or a network operator) is divided into larger subbands, each of which is used for serving a group of users. Within this group, multiple access is done by means of another multiple access method – e.g., TDMA or CDMA. Most current wireless systems use FDMA in that way.
- **High-data-rate systems:** The disadvantages of FDMA are mostly relevant if each user requires only a small bandwidth – e.g., 20 kHz. The situation can be different for wireless Local Area Networks (LANs), where a single user requires a bandwidth on the order of 20 MHz, and only a few frequency channels are available.

PART-B

Q.6 *Explain Multiple Access via Frequency Division Multiple Access.*

Ans. FDMA is the oldest, and conceptually most simple, multi access method. Each user is assigned a frequency (sub)band – i.e., a (usually contiguous) part of the available spectrum. The assignment of frequency bands is usually done during call setup, and retained during the whole call. FDMA is usually combined with the Frequency Domain Duplexing (FDD), so that two frequency bands (with a fixed duplex distance) are assigned to each user: one for downlink (BS-to-MS) and one for uplink (MS-to-BS) communication.

DIGITAL SIGNALING FOR FADING CHANNELS 3

IMPORTANT QUESTIONS

PART-A

Q.1 *List out the advantages of QPSK.*

Ans. Advantages of QPSK:

(i) Low error probability

(ii) Very good noise immunity

(iii) Carrier power remains constant

Q.2 *Define $\pi/4$ QPSK Modulation.*

Ans. $\pi/4$ QPSK Modulation: In a $\pi/4$ QPSK modulation, signaling points of the modulated signal are selected from two QPSK constellations which are shifted by $\pi/4$ with respect to each other.

Q.3 *Define PAPR in OFDM?*

Ans. PAPR: PAPR can be defined as the relation between the maximum power of a sample in a transmit OFDM symbol and its average power.

Q.4 *Why GMSK is preferred for multiuser in cellular communication?*

Ans. 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.

Q.5 *What is windowing?*

Ans. Windowing: In communication window function is a mathematical function that is zero valued outside of some chosen interval and is the process of taking a small subset of a larger dataset for processing and analysis.

PART-B

Q.6 *Compare the spectral efficiency of MSK and QPSK with rectangular constituent pulses. Consider systems with equal bit duration. Compute the out-of-band energy at $1/T_B$, $2/T_B$ and $3/T_B$*

Ans. The power-spectral density of MSK is given by

$$S_{MSK}(f) = \frac{16T_B}{\pi^2}\left(\frac{\cos(2\pi f T_B)}{1-16f^2T_B^2}\right)^2$$

whereas, the power-spectral density for QPSK with rectangular pulses is the same as for ordinary QAM given by (note that we normalize such that the integral over the power-spectral density becomes unity):

$$S_{OQPSK}(f) = (1/T_S)(T_S \operatorname{sinc}(\pi f T_S))^2$$

where it must be noted that $T_S = 2T_B$ for QPSK. The out-of-band power is, for MSK and $T_B = 1$, given by

$$P_{out}(f_0) = 2\int_{f=f_0}^{\infty} S(f)\,df$$

$$= 2\int_{1}^{\infty}\frac{16}{\pi^2}\left(\frac{\cos(2\pi f)}{1-16f^2}\right)^2 df$$

$$= \frac{32}{\pi^2}\int_{1}^{\infty}\frac{\cos^2 2\pi f}{256f^4-32f^2+1}\,df$$

PART-A

Q.1 ***What is Equalizer and Adaptive equalization?***

Ans. Equalizer : Equalizers are RX structures that work both ways: they reduce or eliminate ISI, and at the same time exploit the delay diversity inherent in the channel. The operational principle of an equalizer can be visualized either in the time domain or the frequency domain.

Adaptive Equalization : The problem of time variance is solved by repeating the transmission of the training sequence at "sufficiently short" time intervals, so that the equalizer can be adapted to the channel state at regular intervals. The concept is thus known as "adaptive equalization."

Q.2 ***What is Macro diversity? Depending on the spatial degree of freedom (DoF) of the system what the user may transmit or receive?***

Ans. Macro Diversity : Macro diversity is a kind of diversity in space scheme using several receiver antennas and/or transmitter antennas for transferring the same signal to distances where the distance between is much longer than the signal's wavelength.

Depending on the spatial degree of freedom (DoF) of the system user may transmit or receive multiple independent data streams to/from BS in the same time and frequency resource.

Q.3 ***During coping if the signal what are the distortions? What is the main aim of microdiversity?***

Multiple Antenna Techniques 5

Important Questions

Part-A

Q.1 *What does Spatial Multiplexing (SM) mean?*

Ans. Spatial Multiplexing (SM) : Spatial multiplexing is a MIMO wireless protocol that sends separate data signals or streams between antenna to enhance wireless signal performance or functionality. It is a type of "spatial diversity" and an engineering trick that helps to increase the possibilities for various types of end-to-end transmission.

In spatial multiplexing, multipath propagation involves multiple-input/multiple-output or MIMO wireless technology setups – the transmit stations use multiple transmit and receive antennas to produce sophisticated signal results. A wireless access point uses multiple radios to enable more than one unique data stream to go between the transmitter and receiver. This increases throughput and is a common technique in order to innovate with wireless setups.

Q.2 *What do you understand by Precoding?*

Ans. Precoding is a technique which exploits transmit diversity by weighting the information stream, i.e. the transmitter sends the coded information to the receiver to achieve pre-knowledge of the channel. The receiver is a simple detector, such as a matched filter, and does not have to know the channel state information. This technique will reduce the corrupted effect of the communication channel. To prevent a potential misunderstanding here, precoding does not cancel out the impact of the channel, but it aligns the vector containing the transmit symbols (i.e. transmit vector) with the eigen vector(s) of the channel. In simple terms, it transforms the transmit symbols' vector in such a way that the vector reaches the receiver in the strongest form that is possible in the given channel.

Q.3 *Why preprocessing called "coding"?*

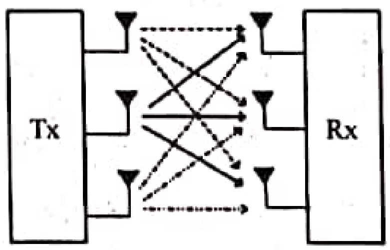
Ans. It is a preprocessing technique that performs transmit diversity and it is similar to equalization, but the main difference is that you have to optimize the precoder with a decoder. Channel equalization aims to minimize channel errors, but the precoder aims to minimize the error in the receiver output.

Q.4 *Define antenna diversity.*

Ans. Antenna diversity is a transmission method using more than one antenna to receiver or transmits signals along different propagation paths to compensate for multipath interferences.

Q.5 *Draw the structure of MIMO system model.*

Ans.



Q.6 *Distinguish ergodic capacity and outage capacity of a flat fading channel?*

Ans. Ergodic capacity is the expected value of the capacity taken over all realization of the channel. Outage capacity is the minimum transmission rate that is achieved over a certain fraction of time.

Q.8 *What is Two-Ray Model? Determine the critical distance for the two-ray model in an urban microcell (h_t=10m,h_r=3m)and an indoor microcell (h_t = 3m, h_r = 2m) for f_c = 2GHz.*

Ans. The two-ray model is used when a single ground reflection dominates the multipath effect. The received signal consists of two components: the LOS component or ray, which is just the transmitted signal propagating through free space, and a reflected component or ray, which is the transmitted signal reflected off the ground.

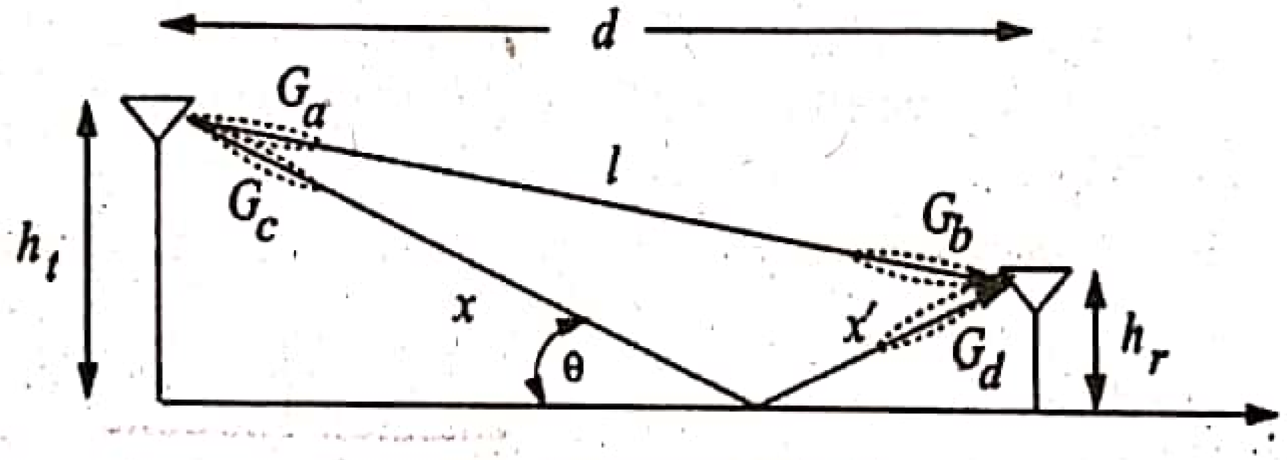


Fig. : Two-Ray Model

$d_c = 4h_th_r/\lambda$= 800 meters for the urban microcell and160meters for the indoor system. A cell radius of 800 m in an urban microcell system is a bit large: urban microcells today are on the order of 100 m to maintain large capacity. However, if we used a cell size of 800 m under these system parameters, signal power would fall off inside the cell, and interference from neighboring cells would fall, and thus would be greatly reduced. Similarly, 160 m is quite large for the cell radius of an indoor system, as there would typically be many walls the signal would have to go through for an indoor cell radius of that size. So an indoor system would typically have a smaller cell radius, on the order of 10-20 m.

PART-B

Q.6 Explain Multiple Access via Frequency Division Multiple Access.

Ans. FDMA is the oldest, and conceptually most simple, multi access method. Each user is assigned a frequency (sub)band – i.e., a (usually contiguous) part of the available spectrum. The assignment of frequency bands is usually done during call setup, and retained during the whole call. FDMA is usually combined with the Frequency Domain Duplexing (FDD), so that two frequency bands (with a fixed duplex distance) are assigned to each user: one for downlink (BS-to-MS) and one for uplink (MS-to-BS) communication.

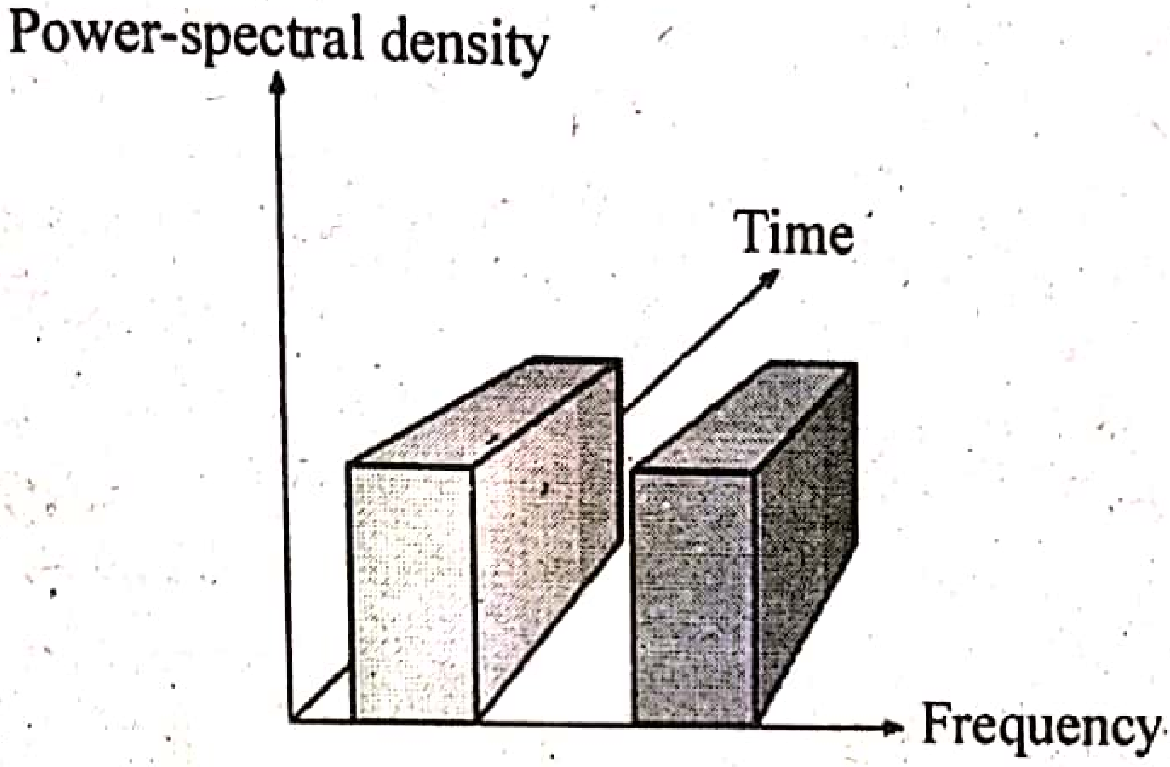


Fig. : Principle of frequency division multiple access

Pure FDMA is conceptually very simple, and has some advantages for implementation:

- The transmitter (T_X) and receiver (R_X) require little digital signal processing. However, this is not so important in practice anymore, as the costs for digital processing are continuously decreasing.
- (Temporal) synchronization is simple. Once synchronization has been established during the call setup, it is easy to maintain it by means of a simple tracking algorithm, as transmission occurs continuously.

PART-B

Q.6 ***Compare the spectral efficiency of MSK and QPSK with rectangular constituent pulses. Consider systems with equal bit duration. Compute the out-of-band energy at $1/T_B$, $2/T_B$ and $3/T_B$.***

Ans. The power-spectral density of MSK is given by

$$S_{MSK}(f) = \frac{16T_B}{\pi^2}\left(\frac{\cos(2\pi f T_B)}{1-16f^2T_B^2}\right)^2$$

whereas, the power-spectral density for QPSK with rectangular pulses is the same as for ordinary QAM given by (note that we normalize such that the integral over the power-spectral density becomes unity):

$$S_{OQPSK}(f) = (1/T_S)(T_S\,\mathrm{sinc}(\pi f T_S))^2$$

where it must be noted that $T_S = 2T_B$ for QPSK. The out-of-band power is, for MSK and $T_B = 1$, given by

$$P_{out}(f_0) = 2\int_{f=f_0}^{\infty} S(f)\,df$$

$$= 2\int_{1}^{\infty}\frac{16}{\pi^2}\left(\frac{\cos(2\pi f)}{1-16f^2}\right)^2 df$$

$$= \frac{32}{\pi^2}\int_{1}^{\infty}\frac{\cos^2 2\pi f}{256f^4-32f^2+1}\,df$$

$$= \frac{32}{\pi^2}\int_1^{\infty}\frac{\frac{1}{2}\cos 4\pi f+\frac{1}{2}}{256f^4-32f^2+1}\,df$$

and for QPSK with $T_B = 1$ given by

$$P_{out}(f_0) = \int_1^{\infty}\left(2\frac{\sin(2\pi f)}{(2\pi f)}\right)^2 df$$

Solving these integrals numerically gives the following table:

	$1/T_B$	$2/T_B$	$3/T_B$
QPSK	0.050	0.025	0.017
MSK	0.0024	$2.8*10^{-4}$	$7.7*10^{-5}$

PART-B

Q.6 Explain the mathematical implementation of linear equalizers.

Ans. Mathematical Implementation of Linear Equalizers : Linear equalizers are simple linear filter structures that try to invert the channel in the sense that the product of the transfer functions of channel and equalizer fulfills a certain criterion. This criterion can either be achieving a completely flat transfer function of the channel – filter concatenation, or minimizing the mean-squared error at the filter output.

$$\hat{c}_i = \sum_{n=-K}^{K} e_n u_{i-n}$$

that should be "as close as possible" to the sequence $\{c_i\}$. Defining the deviation ε_i as

$$\varepsilon_i = c_i - \hat{c}_i$$

we aim to find a filter so that

$$\varepsilon_i = 0 \text{ for } N_0 = 0$$

which gives the ZF equalizer, or that

$$E\{|\varepsilon_i|^2\} \rightarrow \min \text{ for } N_0 \text{ having a finite value}$$

which gives the Minimum Mean Square Error (MMSE) equalizer.

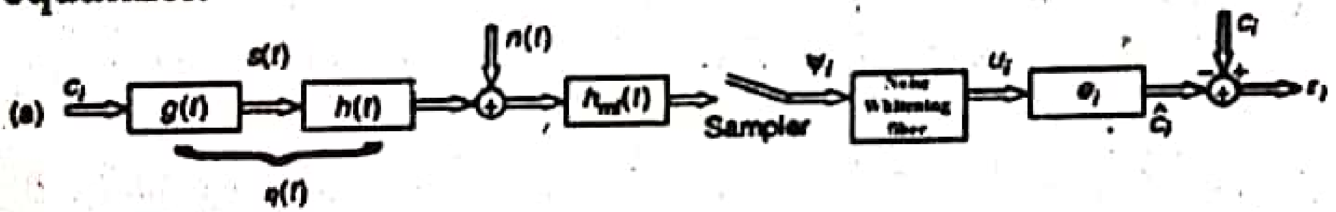


Fig. 1 : Linear equalizer in the time domain

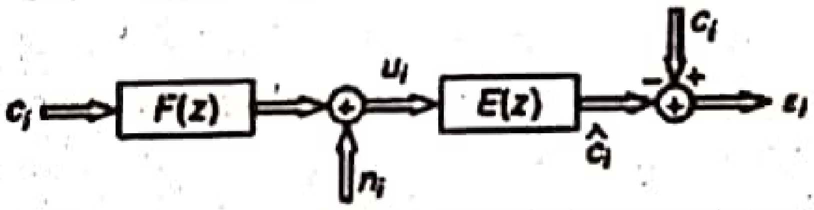


Fig. 2 : Time-discrete equivalent system in the z-transform domain

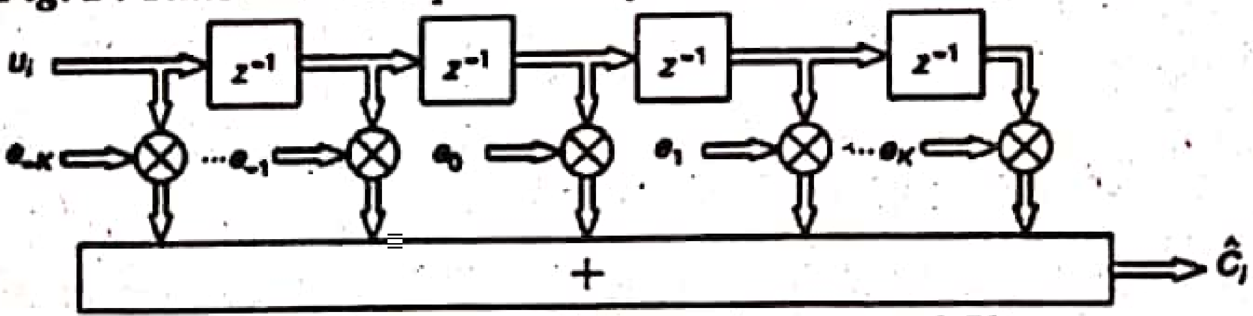


Fig. 3 : Structure of a linear transversal filter.

Remember that z^{-1} represents a delay by one sample.

Q.8 *What is the existence of Zero-Forcing Equalizer in Liner Equalizer?*

Ans. Zero-Forcing Equalizer : The ZF equalizer can be interpreted in the frequency domain as enforcing a completely flat (constant) transfer function of the combination of channel and equalizer by choosing the equalizer transfer function as E(z)=1/F(z). In the time domain, this can be interpreted as minimizing the maximum ISI (peak distortion criterion).

The ZF equalizer is optimum for elimination of ISI. However, channels also add noise, which is amplified by the equalizer. At frequencies where the transfer function of the channel attains small values, the equalizer has a strong amplification, and thus also amplifies the noise. As a consequence, the noise power at the detector input is larger than for the case without an equalizer.

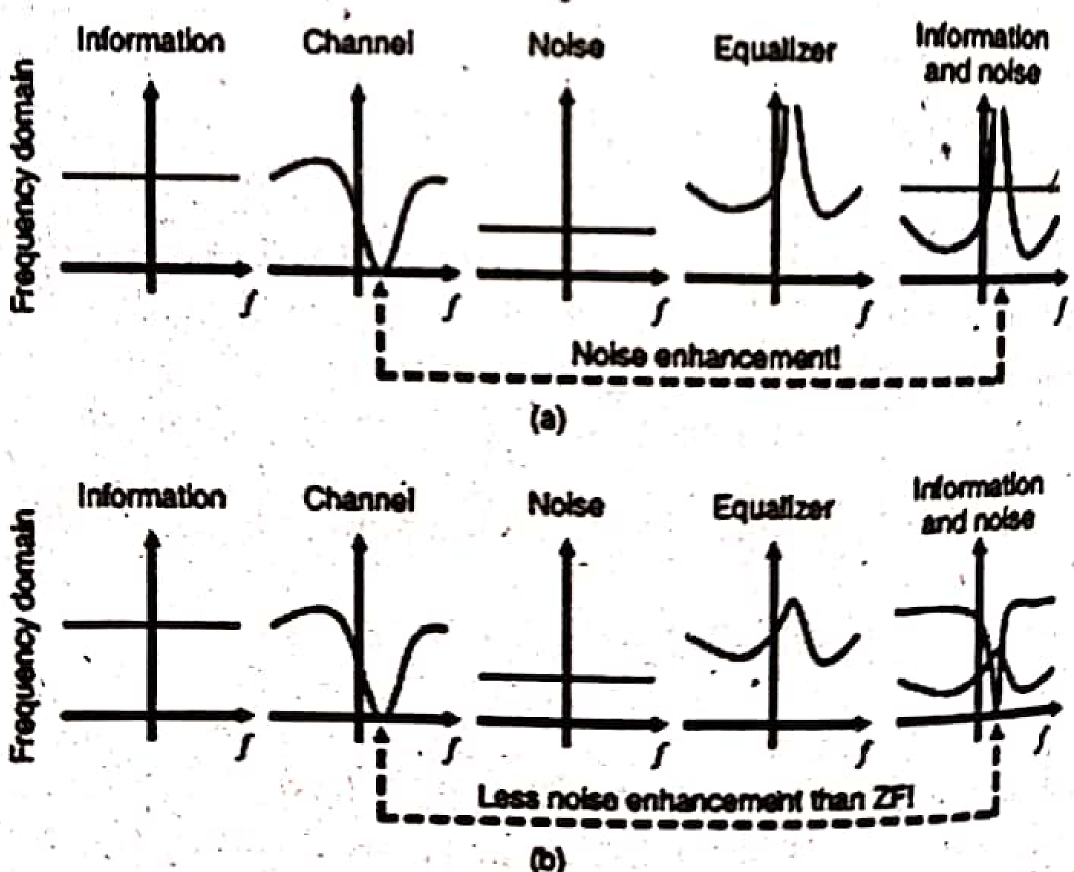


Fig. (a) Illustration of noise enhancement in zero-forcing equlizer (b) which is mitigated in an MMSE linear equilizer

The Fourier transform $\hat{\Xi}(e^{j\omega T_s})$ of the sample ACF ξ_i is related to $\hat{\Xi}(e^{j\omega T})$, the Fourier transform of $\eta(t)$, as

$$\Xi(e^{j\omega T_s}) = \frac{1}{T_S}\sum_{n=-\infty}^{\infty}\left|\hat{\Xi}\left(\omega+\frac{2\pi n}{T_S}\right)\right|^2, |\omega| \le \frac{\pi}{T_S}$$

The noise power at the detector is

$$\sigma^2_{n-LE-ZF} = N_0\frac{T_S}{2\pi}\int_{-\pi/T_S}^{\pi/T_S}\frac{1}{\Xi(e^{j\omega T_S})}d\omega$$

It is finite only if the spectral density Ξ has no (or only integrable) singularities.

Q.7 *Define capacity of a fading channel.*

Ans. Channel capacity repents the fundamental limitation for information transmission over any communication channel.

PART-B

Q.8 *Define MIMO (multiple input, multiple output).*

Ans. MIMO : MIMO (multiple input, multiple output) is an antenna technology for wireless communications in which multiple antennas are used at both the source (transmitter) and the destination (receiver). The antennas at each end of the communications circuit are combined to minimize errors and optimize data speed. MIMO is one of several forms of smart antenna technology, the others being MISO (multiple input, single output) and SIMO (single input, multiple output).

In conventional wireless communications, a single antenna is used at the source, and another single antenna is used at the destination. In some cases, this gives rise to problems with multipath effects. When an electromagnetic field (EM field) is met with obstructions such as hills, canyons, buildings, and utility wires, the wave fronts are scattered, and thus they take many paths to reach the destination. The late arrival of scattered portions of the signal causes problems such as fading, cut-out (cliff effect), and intermittent reception (picket fencing). In digital communications systems such as wireless Internet, it can cause a reduction in data speed and an increase in the number of errors. The use of two or more antennas, along with the transmission of multiple signals (one for each antenna) at the source and the destination, eliminates the trouble caused by multipath wave propagation, and can even take advantage of this effect.

MIMO technology has aroused interest because of its possible applications in digital television (DTV), wireless local area networks (WLANs), metropolitan area networks (MANs), and mobile communications.

Q.12 *What is parameters of Mobile Multipath Channels. Explain its type.*

Ans. Parameters of Mobile Multipath Channels : Many multipath channel parameters are derived from the power delay profile. Power delay profiles are found by averaging instantaneous power delay profile measurements over a local area in order to determine an average small-scale power delay profile. Depending on the time resolution of the probing pulse and the type of multipath channels studied, researchers often choose to sample at spatial separations of a quarter of a wavelength and over receiver movements no greater than 6 m in outdoor channels and no greater than 2 m in indoor channels in the 450 MHz - 6 GHz range. This small-scale sampling avoids large-scale averaging bias in the resulting small-scale statistics.

(1) Time Dispersion Parameters : In order to compare different multipath channels and to develop some general design guidelines for wireless systems, parameters which grossly quantify the multipath channel are used. The mean excess delay, rms delay spread, and excess delay spread (X dB) are multipath channel parameters that can be determined from a power delay profile. The time dispersive properties of wide band multipath channels are most commonly quantified by their mean excess delay (τ) and rms delay spread (σ_τ). The mean excess delay is the first moment of the power delay profile and is defined to be

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

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

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

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

(2) Coherence Bandwidth : While the delay spread is a natural phenomenon caused by reflected and scattered propagation paths in the radio channel, 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" (i.e., 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 potential for amplitude correlation. Two sinusoids with frequency separation greater than B_c are affected quite differently by the channel. If the coherence bandwidth is designed as the bandwidth over which the frequency correlation function is above 0.9, then the coherence bandwidth is approximately

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

If the definition is relaxed so that the frequency correlation function is above 0.5. then the coherence bandwidth is approximately

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

(3) Doppler Spread and Coherence Time : Delay spread and coherence bandwidth are parameters which describe the time dispersive nature of the channel in a local area. However, they do not offer information about the time varying nature of the channel caused by either relative motion between the mobile and base station, or by movement of objects in the channel. Doppler spread and coherence time are parameters which describe the time varying nature of the channel in a small-scale region.

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(3) Adding Carriers

Adding carriers (or more accurately, bandwidth) directly adds to capacity. The LTE (Long Term Evolution) standard is particularly adapt at utilizing increased bandwidth without increasing control channel overheads.

(4) Improved Air Interface Capabilities

Improved air interface capabilities such as in evolving from UMTS (Universal Mobile Telecommunications System) to HSDPA (High Speed Downlink Packet Access) that provided well over four times the aggregate downlink capacity.

(5) Smart Antennas

Smart antennas provide the next substantial increase in throughput. By "smart antennas" we refer to adaptive antennas such as those with electrical tilt, beam width and azimuth control which can follow relatively slow-varying traffic patterns. As well as so called intelligent antennas that can form beams aimed at particular users or steer nulls to reduce interference. And finally Multiple-Input Multiple Output (MIMO) antenna schemes.

PART-C

Q.14 *Explain the term Time Division Multiple Access(TDMA) and how it is different from FDMA.*

Ans. For TDMA, different users transmit not at different frequencies but rather at different times. A time unit is subdivided into N time slots of fixed duration, and each user is assigned one such time slot. During the assigned timeslot, the user can transmit with a high data rate (as it can use the whole system bandwidth); subsequently, it remains silent for the next N-1 time slots, when other users take their turn. This process is then repeated periodically. At first glance, this approach has the same performance as FDMA: a user transmits only during 1/N of the available time, but then occupies N times the bandwidth. However, there are some important practical differences:

- Users occupy a larger bandwidth. This allows them to exploit the frequency diversity available within the bandwidth allocated to the system; furthermore, the sensitivity to random FM is reduced.
  On the flipside; equalizers are required to combat Inter Symbol Interference (ISI) for most operating environments; this increases the effort needed for digital signal processing.

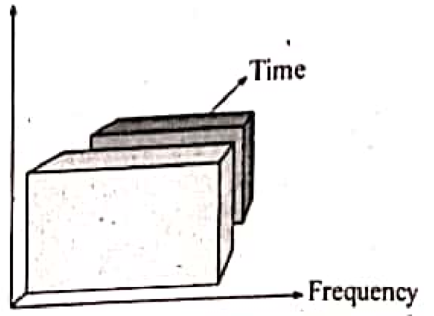

Fig. : Principle behind time division multiple access

- Temporal guard intervals are required. A TX needs a finite amount of time to ramp up from 0-W output power to "full power" (typically between 100 mW and 100 W). Furthermore, there has to be sufficient guard time to compensate for the runtime of the signal between the MS and BS. It is possible that one MS is far away from the BS, while the one that transmits in the subsequent time slot is very close to the BS and thus has negligible runtime. As the signals from the two users must not overlap at the BS, the second MS must not transmit during the time it takes the first signal to propagate to the BS. Note, however, that there is no need for frequency guard bands, as each user completely fills up the assigned band.
- Each time slot might require a new synchronization and channel estimation, as transmission is not continuous. Optimization of time slot duration is a challenging task. If it is too short, then a large percentage of the time is used for synchronization and channel estimates (in GSM, 17% of a time slot are used for this purpose). If the time slot is too long, transmission delays become too long (which users find annoying especially for speech communications), and the channel starts to change during one time slot. In that case, the equalizer has to track the channel during transmission of a time slot, which increase simple mentation effort (this was required, e.g., in the – now defunct – Interim Standard (IS)-136 cellular standard). If the time between two time slots assigned to one user is larger than coherence time, the channel has changed between these two time slots, and a new channel estimate is required.
- For interference-limited systems, TDMA has a major advantage: during its period of inactivity, the MS can "listen" to transmission on other time slots. This is especially useful for the preparation of handovers from one BS to another, when the MS has to find out whether a neighboring BS would offer better quality, and has communications channels available.
  TDMA is used in the worldwide cellular standard GSM as well as the cord-less standard DECT (Digital Enhanced Cordless Telecommunications.

LFSR.

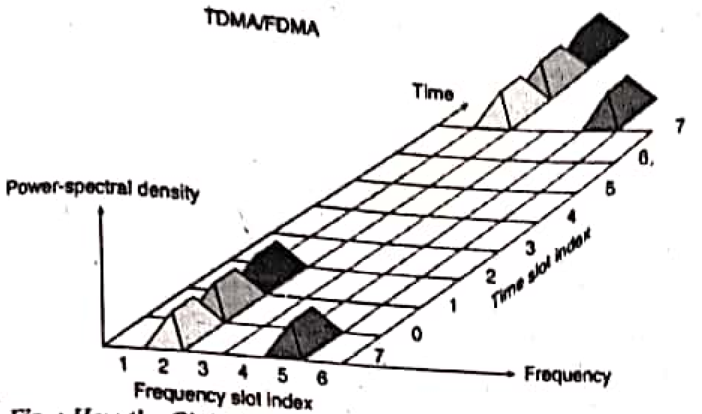

Fig. : How the Global System for Mobile communications (GSM) combines time division multiple access with frequency division multiple access.

In contrast, pure FDMA is used mainly in analog cellular and cordless systems.

Q.15 ***What is Code Division Multiple Access? What is its utilization in spread spectrum?***

Ans. In code division multiple access (CDMA) systems, the narrow band 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. All users in a CDMA system, use the same carrier frequency and may transmit simultaneously. Each user has its own pseudo random code word which is approximately orthogonal to all other code words. The receiver performs a time correlation operation to detect only the specific desired codeword. All other code words appear as noise due to correlation. For detection of the message signal, the receiver needs to know the code word used by the transmitter. Each user operates independently with no knowledge of the other users.

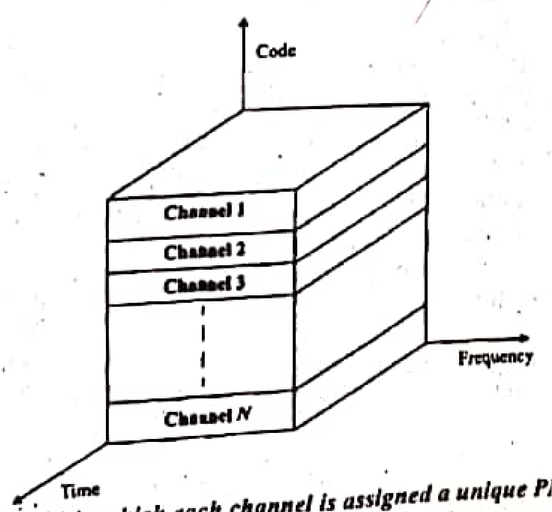

Fig. : CDMA in which each channel is assigned a unique PN code which is orthogonal to PN codes used by other users.

In CDMA, the power of multiple users at a receiver determines the noise floor after decorrelation. 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. 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. To overcome the near-far problem, power control is used in most CDMA implementations. Power control is provided by each base station in a cellular system and assures that each mobile within the base station coverage area provides the same signal level to the base station receiver. This solves the problem of a nearby subscriber over-powering the base station receiver and drowning out the signals of faraway subscribers. Power control is implemented at the base station by rapidly sampling the radio signal strength indicator (RSSI) levels of each mobile and then sending a power change command over the forward radio link. Despite the use of power control within each cell, out-of-cell mobiles provide interference which is not under the control of the receiving base station.

Basic Principle behind the Direct Sequence-Spread Spectrum

The DSSS spreads the signal by multiplying the transmit signal by a second signal that has a very large bandwidth. The bandwidth of this total signal is approximately the same as the bandwidth of the wideband spreading signal. The ratio of the bandwidth of the new signal to that of the original signal is again known as the spreading factor. As the bandwidth of the spread signal is large, and the transmit power stays constant, the power-spectral density of the transmitted signal is very small – depending on the spreading factor and the BS – MS distance, it can lie below the noise power-spectral density. This is important in military applications, because unauthorized listeners cannot determine whether a signal is being transmitted. Authorized listeners, on the other hand, can invert the spreading operation and thus recover the narrow band signal (whose power-spectral density lies considerably above the noise power) shows the block diagram of a DSSS transmitter. The information sequence (possibly coded) is multiplied by a broadband signal that was created by modulating a sinusoidal carrier signal with a spreading sequence. This can be interpreted alternatively as multiplying each information symbol of duration T_S by a spreading sequence(t) before modulation. We assume that the spreading sequence is M_C chips long, where each chip has the duration $T_C = T_S/M_C$. As the bandwidth is the inverse of the chip duration, the bandwidth of the total signal is now also $W=1/T_C = M_C/T_S$ i.e., larger than the bandwidth of a narrow band-modulated signal by a factor M_C. As we assume that the spreading operation does not change the total transmit power, it also implies that the power-spectral density decreases by a factor M_C.

PART-C

Q.13 ***Write down the difference between QPSK and π/4-Differential Quadrature-Phase Shift Keying.***

Ans. Quadrature-Phase Shift Keying : A Quadrature-Phase Shift Keying (QPSK)-modulated signal is a PAM where the signal carries 1 bit per symbol interval on both the in-phase and quadrature-phase component. The original data stream is split into two streams, b_{1i} and b_{2i} :

$$b_{1i} = b_{2i}$$
$$b_{2i} = b_{2i+1}$$

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

$$RS = 1/TS = RB/2 = 1/(2TB)$$

Let us first consider the situation where basis pulses are rectangular pulses,g(t)=gR(t, TS).

Then we can give an interpretation of QPSK as either a phase modulation or as a PAM. We first define two sequences of pulses :

$$\left.\begin{aligned} p1_D(t) &= \sum_{i=-\infty}^{\infty} b1_i g(t - iT_S) = b1_i * g(t) \\ p2_D(t) &= \sum_{i=-\infty}^{\infty} b2_i g(t - iT_S) = b2_i * g(t) \end{aligned}\right\}$$

When interpreting QPSK as a PAM, the bandpass signal reads

$$s_{BP}(t) = \sqrt{E_B/T_B}\,[p1_D(t)\cos[2\pi f_c t] - p2_D(t)\sin(2\pi f_c t)]$$

Normalization is done in such a way that the energy within one symbol interval is $\int_0^{T_s} s_{BP}(t)^2\,dt = 2E_B$, where E_B is the energy expended on transmission of a bit.

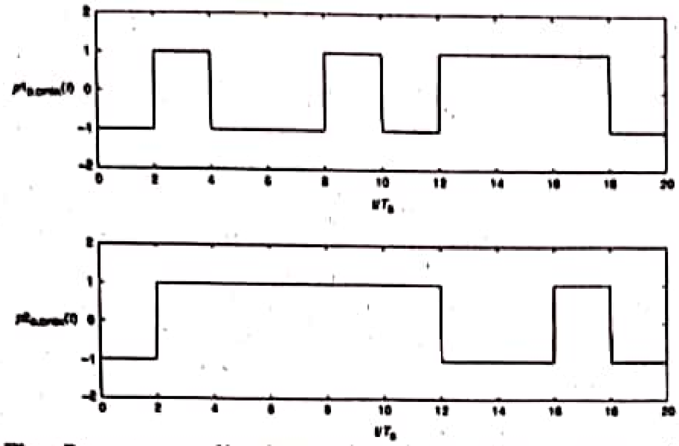

Fig. : Data streams of in-phase and quadrature-phase components in quadrature-phase shift keying.

The baseband signal is

$$s_{LP}(t) = [p1_D(t) + jp2_D(t)]\sqrt{E_B/T_B}$$

When interpreting QPSK as a phase modulation, the low-pass signal can be written as $\sqrt{2E_B/T_B}\exp(j\Phi_S(t))$ with:

$$\Phi_S(t) = \pi.\left[\frac{1}{2}.p2_D(t) - \frac{1}{4}.p1_D(t).p2_D(t)\right]$$

It is obvious from this representation that the signal is constant envelope, except for the transitions at $t = iT_S$

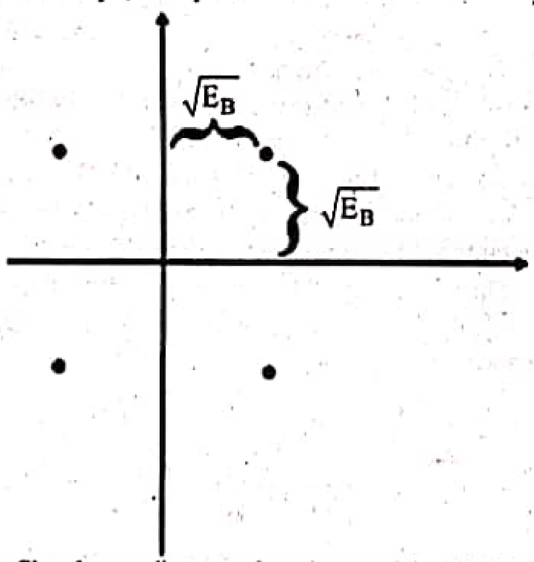

Fig. : Signal space diagram of quadrature-phase shift keying.

π/4-Differential Quadrature-Phase Shift Keying

Even though QPSK is nominally a constant envelope format, it has amplitude dips at bit transitions; this can also be seen by the fact that the trajectories in the I – Q diagram pass through the origin for some of the bit transitions.

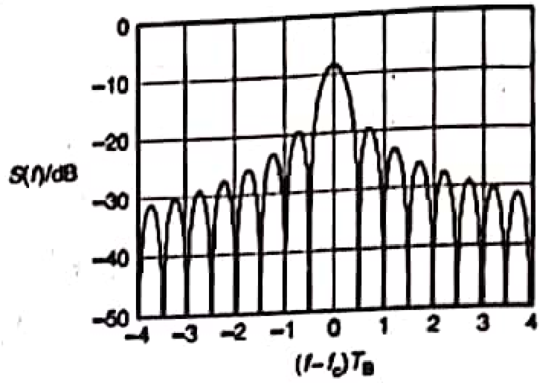

Fig. : Normalized power-spectral density of quadrature-phase shift keying

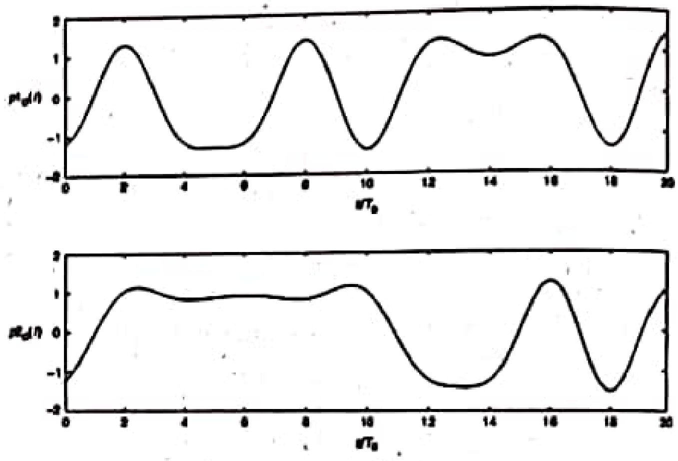

Fig. : Quadrature amplitude modulation pulse sequence

The duration of the dips is longer when non-rectangular basis pulses are used. Such variations of the signal envelope are undesirable, because they make the design of suitable amplifiers more difficult. One possibility for reducing these problems lies in the use of π/4-DQPSK (π/4 differential quadrature-phase shift keying). This modulation format had great importance for second-generation cellphones – it was used in several American standards (IS-54, IS-136, PWT), as well as the Japanese cellphone (JDC) and cordless (PHS) standards, and the European trunk radio standard (TETRA).

Q.14 ***Explain the working and need of Minimum Shift Keying.***

Ans. Minimum Shift Keying(MSK) is one of the most important modulation formats for wireless communications. However, it can be interpreted in different ways, which leads to considerable confusion:

1. The first interpretation is as CPFSK with a modulation index:

   $h_{mod} = 0.5$, $f_{mod} = 1/4T$

   This implies that the phase changes by $\pm\pi/2$ during a 1-bit duration.

antenna elements. The delay between signals transmitted from different antenna elements should be at least as large as the maximum excess delay of the channel.

Q.13 *What is Beamforming? How Precoding use in Multi Antenna Communication.*

Ans. Precoding is a generalization of beamforming to support multi-stream (or multi-layer) transmission in multi-antenna wireless communications. In conventional single-stream beamforming, the same signal is emitted from each of the transmit antennas with appropriate weighting (phase and gain) such that the signal power is maximized at the receiver output. When the receiver has multiple antennas, single-stream beamforming cannot simultaneously maximize the signal level at all of the receive antennas. In order to maximize the throughput in multiple receive antenna systems, multi-stream transmission is generally required.

In point-to-point systems, precoding means that multiple data streams are emitted from the transmit antennas with independent and appropriate weightings such that the link throughput is maximized at the receiver output. In multi-user MIMO, the data streams are intended for different users (known as SDMA) and some measure of the total throughput (e.g., the sum performance or max-min fairness) is maximized. In point-to-point systems, some of the benefits of precoding can be realized without requiring channel state information at the transmitter, while such information is essential to handle the inter-user interference in multi-user systems.Precoding in the downlink of cellular networks, known as network MIMO or coordinated multipoint (CoMP), is a generalized form of multi-user MIMO that can be analyzed by the same mathematical techniques.

Q.14 *What is the Capacity with Receiver Diversity?*

Ans. Receiver Diversity : Receiver diversity is a well known technique to improve the performance of wireless communications in fading channels. The main advantage of receiver diversity is that it mitigates the fluctuations due to fading so that the channel appears more like an AWGN channel. Since receiver diversity mitigates the impact of fading, an interesting question is whether it also increases the capacity of a fading channel. The capacity calculation under diversity combining first requires that the distribution of the received SNR $p(\gamma)$under the given diversity combining technique be obtained. Once this distribution is known it can be substituted into any of the capacity formulas above to obtain the capacity under diversity combining. The specific capacity formula used depends on the assumptions about channel side information.It was found that, as expected, the capacity with perfect transmitter and receiver CSI is bigger than with receiver CSI only, which in turn is bigger than with channel inversion. The performance gap of these different formulas decreases as the number of antenna branches increases. This trend is expected, since a large number of antenna branches make the channel look like AWGN, for which all of the different capacity formulas have roughly the same performance.

PART-C

Q.15 *Explain Transmitter Diversity in SISO, SIMO and MIMO with block diagram?*

Ans. Diversity : As indicated, two fundamental resources available for a MIMO system are diversity and degrees of freedom.

In diversity techniques, same information is sent across independent fading channels to combat fading. When multiple copies of the same data are sent across independently fading channels, the amount of fade suffered by each copy of the data will be different. This guarantees that at-least one of the copy will suffer less fading compared to rest of the copies. Thus, the chance of properly receiving the transmitted data increases. In effect, this improves the reliability of the entire system. This also reduces the co-channel interference significantly. This technique is referred as inducing a "spatial diversity" in the communication system.

Consider a SISO system where a data stream [1, 0, 1, 1, 1] is transmitted through a channel with deep fades. Due to the variations in the channel quality, the data stream may get lost or severely corrupted that the receiver cannot recover.The solution to combat the rapid channel variations is to add independent fading channel by increasing the number of transmitter antennas or receiver antennas or the both.

The SISO antenna configuration will not provide any diversity as there is no parallel link. Thus the diversity is indicated as (0).

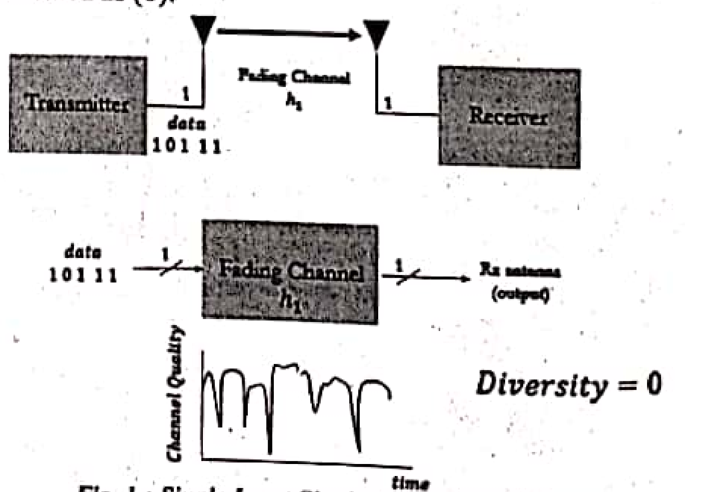

Fig. 1 : Single Input Single Output (SISO) System

Instead of transmitting with single antenna and receiving with single antenna (as in SISO), let's increase the number of receiving antennas by one more count. In this Single Input Multiple Output (SIMO) antenna system, two copies of the same data are put on two different channels having independent fading characteristics. Even if one of the link fails to deliver the data, the chances of proper delivery of the data across the other link is very high. Thus, additional fading channels increase the reliability of the overall transmission – this improvement in reliability translates into performance improvement – measured as *diversity gain*. For a system with NT transmitter antennas and N_r receiver antennas, the maximum number of diversity paths is $N_t \times N_r$. In the following configuration, the total number of diversity path created is $1 \times 2 = 2$.

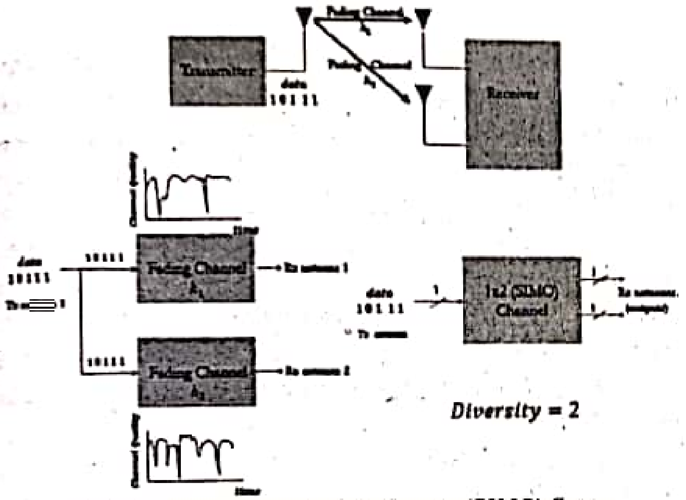

Fig. 2 : Single Input Multiple Output (SIMO) System

In this way, more diversity paths can be created by adding multiple antennas at transmitter or receiver or both. Figure 3 illustrates a 2 × 2.

MIMO system with number of diversity paths equal to $2 \times 2 = 4$

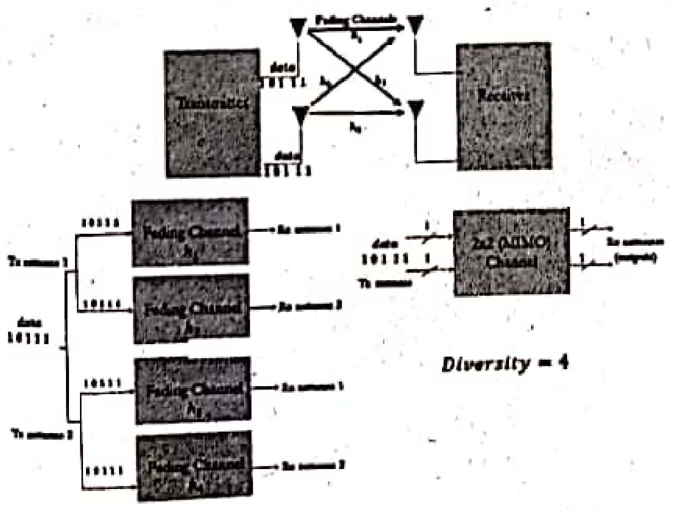

Fig. 3 : Multiple Input Multiple Output (MIMO) System

Q.16 *Write short note on:*

(a) *Precoding for Point-to-Point MIMO Systems*

(b) *Precoding for Multi-user MIMO Systems*

Ans.(a) Precoding for Point-to-Point MIMO Systems

In point-to-point multiple-input multiple-output (MIMO) systems, a transmitter equipped with multiple antennas communicates with a receiver that has multiple antennas. Most classic precoding results assume narrowband, slowly fading channels, meaning that the channel for a certain period of time can be described by a single channel matrix which does not change faster. In practice, such channels can be achieved, for example, through OFDM. The precoding strategy that maximizes the throughput, called channel capacity, depends on the channel state information available in the system.

Statistical Channel State Information : If the receiver knows the channel matrix and the transmitter has statistical information, eigen beamforming is known to achieve the MIMO channel capacity.In this approach, the transmitter emits multiple streams in eigen directions of the channel covariance matrix.

Full Channel State Information : If the channel matrix is completely known, singular value decomposition (SVD) precoding is known to achieve the MIMO channel capacity. In this approach, the channel matrix is diagonalized by taking an SVD and removing the two unitary matrices through pre- and post-multiplication at the transmitter and receiver, respectively. Then, one data stream per singular value can be transmitted (with appropriate power loading) without creating any interference whatsoever.

Ans.(b) Precoding for Multi-user MIMO Systems

In multi-user MIMO, a multi-antenna transmitter communicates simultaneously with multiple receivers (each having one or multiple antennas). This is known as space-division multiple access (SDMA). From an implementation perspective, precoding algorithms for SDMA systems can be sub-divided into linear and nonlinear precoding types. The capacity achieving algorithms are nonlinear, but linear precoding approaches usually achieve reasonable performance with much lower complexity. Linear precoding strategies include maximum ratio transmission (MRT), zero-forcing (ZF) precoding, and transmit Wiener precoding. There are also precoding strategies tailored for low-rate feedback of channel state information, for example random beamforming. Nonlinear precoding is designed based on the concept of dirty paper coding (DPC), which shows that any known interference at the transmitter can be subtracted without the penalty of radio resources if the optimal precoding scheme can be applied on the transmit signal.