WIRELESS COMMUNICATION (ELECTIVE)

V C.S.

WIRELESS CHANNELS 1

IMPORTANT QUESTIONS

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.

Q.2 ***Consider an indoor wireless LAN with $f_c = 900MHz$, cells of radius 100 m, and non-directional antennas. Under the free-space path loss model, what transmit power is required at the access point such that all terminals within the cell receive a minimum power of $10\mu W$. How does this change if the system frequency is 5 GHz?***

Ans. We must find the transmit power such that the terminals at the cell boundary receive the minimum required power.

$$P_t = P_r\left[\frac{4\pi d}{\sqrt{G_l}\lambda}\right]^2$$

Substituting in $G_l = 1$ (non-directional antennas), $\lambda = c/f_c = 0.33$ m, d = 10 m, and $P_r = 10\mu W$ yields $P_t = 1.45W = 1.61$ dBW (recall that P Watts equals $10\log_{10}[P]$ dBW, dB relative to one Watt, and $10\log_{10}[P/.001]$ dBm, dB relative to one milliwatt). At 5 GHz only $\lambda = .06$ changes, so $P_t = 43.9$ KW = 16.42 dBW.

Q.3 ***Write down the types of Small-Scale Fading.***

Ans.

Small-Scale Fading (Based on multipath time delay spread)

- **Flat Fading**
  1. BW of signal < BW of channel
  2. Delay spread < Symbol period
- **Frequency Selective Fading**
  1. BW of signal > BW of channel
  2. Delay spread > Symbol period

Small-Scale Fading (Based on Doppler spread)

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

Q.4 ***Define coherence bandwidth.***

Ans. Coherence Bandwidth : The coherence bandwidth is related to the specific multipath structure of the channel. The coherence bandwidth is a measure of the maximum frequency difference for which signals are still strongly correlated in amplitude.

Q.5 ***Distinguish coherence time and coherence bandwidth.***

Ans. Corehence Bandwith: Refer to Q. 4.

Coherence bandwidth is inversely proportional to the rms value of time delay spread. The coherence timer is defined as the required time interval to obtain an envelope correlation of 0.9 or less.

denoted by B_d. By the Fourier transform relationship between $A_C(\Delta t)$ and $S_C(\rho)$, $B_d \approx 1/T_c$. If the transmitter and reflectors are all stationary and the receiver is moving with velocity v, then $B_d \le v/\lambda = f_d$. Recall that in the narrowband fading model samples became independent at time $\Delta t = .4/f_d$, so in general $B_d \approx k/T_c$ where k depends on the shape of $S_c(\rho)$. We illustrate the Doppler power spectrum $S_C(\rho)$ and its inverse Fourier transform $A_C(\Delta_t)$.

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.

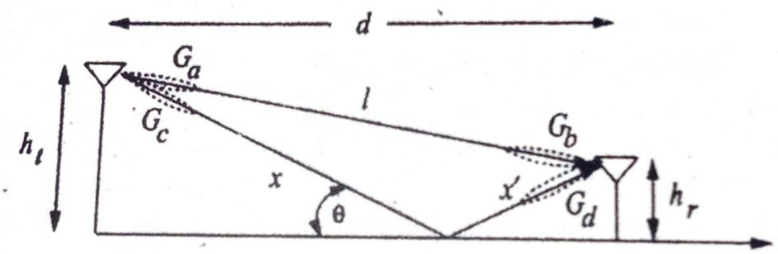


Fig. : Two-Ray Model

$d_c = 4h_t h_r/\lambda =$ 800 meters for the urban microcell and 160 meters 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.

Q.9 ***Write down the factor that influencing small-scale fading.***

Ans. Factors Influencing Small-Scale Fading : Many physical factors in the radio propagation channel influence small-scale fading. These include the following:

Multipath propagation : Th...

conservative design.

Using equation

$$T_C \approx \frac{9}{16\pi f_m} = \frac{9\lambda}{16\pi v}$$

$$= \frac{9c}{16\pi v f_c} = \frac{9\times 3\times 10^8}{16\times 3.14\times 50\times 1900\times 10^6}$$

$$T_C = 565\ \mu s$$

Taking time samples at less than half T_C, at 282.5 μs corresponds to a spatial sampling interval of

$$\Delta x = \frac{vT_C}{2}$$

$$= \frac{50\times 565\mu s}{2} = 0.014125\text{m}$$

$$= 1.41\text{ cm}$$

Therefore, the number of samples required over a 10 m travel distance is

$$N_x = \frac{10}{\Delta x} = \frac{10}{0.014125} = 708\text{ samples}$$

The time taken to make this measurement is equal to $\frac{10\text{m}}{50\text{ m/s}} = 0.2$ s. The Doppler spread is

$$B_d = f_m = \frac{vf_c}{c}$$

$$= \frac{50\times 1900\times 10^6}{3\times 10^8} = 316.66\text{ Hz}$$

PART-C

Q.1 *Due to Doppler Spread what are the Effects of Fading?*

Ans. Fading Effects Due to Doppler Spread

1. Fast Fading

Depending on how rapidly the transmitted baseband signal changes as compared to the rate of change of the channel, a channel may be classified either as a fast fading or slow fading channel. 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. This causes frequency dispersion (also called time selective fading) due to Doppler spreading, which leads to signal distortion. Viewed in the frequency domain, signal distortion due to fast fading increases with increasing Doppler spread relative to the bandwidth of the transmitted signal. Therefore, a signal undergoes fast fading if

$$T_s > T_c$$

And $B_s > B_d$

It should be noted that when a channel is specified as a fast or slow fading channel, it does not specify whether the channel is flat fading or frequency selective in nature. Fast fading only deals with the rate of change of the channel due to motion. In the case of the flat fading channel, we can approximate the impulse response to be simply a delta function (no time delay). Hence, a flat fading, fast fading channel is a channel in which the amplitude of the delta function varies faster than the rate of change of the transmitted baseband signal. In the case of a frequency selective, fast fading channel, the amplitudes, phases, and time delays of any one of the multipath components very faster than the rate of change of the transmitted signal. In practice, fast fading only occurs for very low data rates.

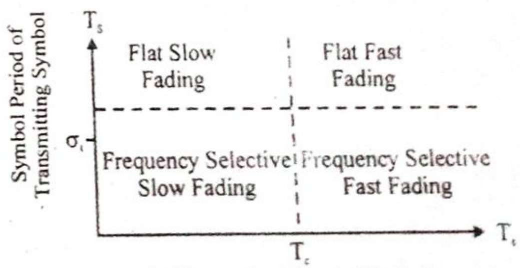
2. Slow Fading

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 signal. Therefore, a signal under-goes slow fading if

$$T_s \ll T_c$$

And $B_s \gg B_d$

It should be clear that the velocity of the mobile (or velocity of objects in the channel) and the baseband signaling determines whether a signal undergoes fast fading or slow fading. The relation between the various multipath parameters and the type of fading experienced by the signal are summarized in Fig. Over the years, some authors have confused the term fast and slow fading with the terms large-scale and small-scale fading. It should be emphasized that fast and slow fading deal with the relationship between the time rate of change in the channel and the transmitted signal, and not with propagation path loss models.



(a) Transmitted Symbol Period

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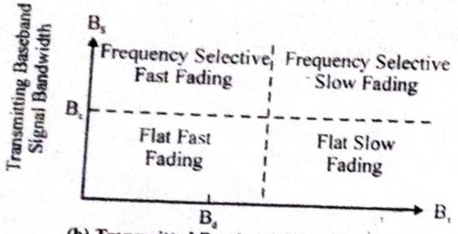

(b) Transmitted Baseband Signal Bandwidth

Fig. : Matrix illustrating type of fading experienced by a signal as a function of (a) symbol period (b) baseband signal bandwidth

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

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

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

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

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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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.

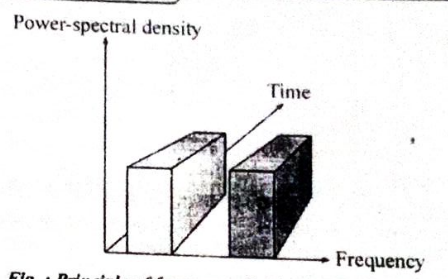

Fig. : Principle of frequency division multiple access

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.

Q.7 *What are the disadvantages face by FDMA at the time of it used for speech communications?*

Ans. Pure FDMA also has significant disadvantages, especially when used for speech communications. These problems arise from spectral efficiency considerations, as well as from sensitivity to multipath effects:

- **Frequency synchronization and stability are difficult:** For speech communications, each frequency sub band is quite narrow (typically between 5 and 30 kHz). Local oscillators thus must be very multiple access and the cellular principle accurate and stable; jitters in the carrier frequency result in adjacent channel interference. High spectral efficiency also requires the use of very steep filters to extract the desired signal. Both accurate oscillators and steep filters are expensive, and thus undesirable. If they are not admissible, guard bands can be used to mitigate filter requirements. This, however, reduces the spectral efficiency of the system.
- **Sensitivity to fading:** since each user is assigned a distinct frequency band, these bands are narrower than for other multi access methods (compare TDMA, CDMA) i.e., 5 – 30 kHz. For such narrow subbands, fading is flat in practically all environments. This has the advantage that no equalization is required; the drawback is that there is no frequency diversity. Remember that frequency diversity is mainly provided by signal components that are more than one channel coherence bandwidth apart .

include police departments, fire departments, taxis, and similar services. The closed user group allows implementation of several technical innovations that are not possible (or more difficult) in normal cellular systems:

1.Group calls: a communication can be sent to several users simultaneously, or several users can set up a conference call between multiple users of the system.

2.Call priorities: a normal cellular system operates on a "first-come, first-serve" basis. Once a call is established, it cannot be interrupted.This is reasonable for cellphone systems, where the network operator cannot ascertain the importance or urgency of a call. However, for the trunk radio system of, e.g., a fire department, this is not an acceptable procedure. Notifications of emergencies have to go through to the affected parties, even if that means interrupting an existing, lower priority call. A trunking radio system thus has to enable the prioritization of calls and has to allow dropping a low-priority call in favor of a high-priority one.

3.Relay networks: the range of the network can be extended by using each Mobile Station(MS) as a relay station for other MSs. Thus, an MS that is out of the coverage region of the BS might send its information to another MS that is with in the coverage region, and that MS will forward the message to the BS; the system can even use multiple relays to finally reach the BS. Such an approach increases the effective coverage area and the reliability of the network. However, it can only be used in a trunking radio system and not in a cellular system – normal cellular users would not want to have to spend "their" battery power on relaying messages for other users.

Q.10 *Explain the features of CDMA.*

Ans. The features of CDMA including the following:

- 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. Increasing the Spread Spectrum Multiple Access number of users in a CDMA system raises the noise floor in a linear manner.Thus, there is no absolute limit on the number of users in CDMA. Rather,the system performance gradually degrades for all users as the number of users is increased, and improves as the number of users is decreased.
- Multipath fading may be substantially reduced because the signal is spreadover a large spectrum. If the spread spectrum bandwidth is greater than the coherence bandwidth of the channel, the inherent frequency diversity will mitigate the effects of small-scale fading.
- Channel data rates are very high in CDMA systems. Consequently, the symbol (chip) duration is very short and usually much less than the channel delay spread. Since PN sequences have low auto correlation, multipath which is delayed by more than a chip will appear as noise. A Rake receiver can be used to improve reception by collecting time delayed versions of the required signal.
- Since CDMA uses co-channel cells, it can use macroscopic spatial diversity to provide soft handoff. Soft handoff is performed by the MSC, which can simultaneously monitor a particular user from two or more base stations. The MSC may choose the best version of the signal at any time without switching frequencies.
- 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, hence in the despreading of a particular PN code, non-zero contributions to the receiver decision statistic for a desired user arise from the transmissions of other users in the system.
- The near-far problem occurs at a CDMA receiver if an undesired user has a high detected power as compared to the desired user.

Q.11 *What are the Channel Assignment Strategies in Cellular System?*

Ans. There are two channel assignment strategies in cellular system.

A. Fixed channel assignment:

1. In fixed channel assignment each cell is permanently allocated predetermined group of channels. Any call attempt within cell can only be served by unused channels in that particular cell.
2. If all channels are occupied, the call is blocked and subscriber does not receive service.
3. Borrowing technique where a cell is allowed to borrow channels from a neighbouring cell if all channels are already occupied is always used with this type of strategy. Mobile Base station (MSC) monitors the function of base station including borrowing ensuring that borrowing does not interfere with any call in progress in donor cell.

B. Dynamic channel assignment:

1. In dynamic channel assignment strategy, voice channels are not allocated permanently.
2. Entire pool of frequency channels lies with MSC and each time a call request is made, the serving base station requests a channel from the MSC. Switch then allocates a channel to the requested cell following a algorithm.
3. MSC allocates frequency channels on dynamic basis if that frequency channel 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.
4. It reduces chances of blocking which increases trunking capacity of system as all available channels are accessible to all cells.
5. In this MSC has to collect real time data on channel occupancy, traffic distribution, radio signal strength indication of all channels on continuous basis, thus increasing the computational load on MSC.

Q.12 *What is Interface Testing?*

Ans. Interface Testing is performed to evaluate whether systems or components pass data and control correctly to one another. It is to verify if all the interactions between these modules are working properly and errors are handled properly.

Interface Testing - Checklist

- Verify that communication between the systems are done correctly.
- Verify if all supported hardware/software has been tested.
- Verify if all linked documents be supported/opened on all platforms.
- Verify the security requirements or encryption while communication happens between systems.
- Check if a solution can handle network failures between website and application server.

Types of Interface Testing

During Interface Testing various types of testing done on the interface which may include :

- **Workflow:** It ensures that the interface engine handles your standard workflows as expected.
- **Edge cases -unexpected values :** This is considered when testing include date, month and day reversed.
- **Performance, load, and network testing :** A high-volume interface may require more load testing than a low-volume interface, depending on the interface engine and connectivity infrastructure.
- **Individual systems:** This includes testing each system individually. For example, billing system and inventory management system for the retail store should be able to operate separately.

Interface Testing Strategy

To test an interface with common tests regardless of implementation, we can use an abstract test case, and then create concrete instances of the test case for each implementation of the interface.

- The base or abstract test case performs the implementation-neutral tests.
- While the concrete tests take care of instantiating the object to test and perform any implementation-specific tests.

Interface Testing Vs Integration Testing

Interface Testing	Integration Testing
• An integration test type that is concerned with testing the interfaces between components or systems	• Testing performed to expose defects in the interfaces and in the interactions between integrated components or systems.

Q.13 *Why do we need to improve wireless bandwidth availability? Explain the ways to improve wireless coverage and capacity.*

Ans. Thinking about home environment, work environment and everywhere in between, you're likely to have mobile phones, iPads or other tablets, laptops, eReaders, smartwatches, games consoles, cameras and more as the Internet of Things expands. Each of these devices may need at least one connection throughout the day – with some continually connected.

As a result of all these connected devices we're seeing ever-increasing demand for wireless bandwidth. But this growth in data traffic is placing immense strain on operator's networks.

5 Ways To Improve Wireless Coverage and Capacity

(1) Adding Cell Sites

Adding cell sites is an effective but expensive approach to adding capacity. In general adding new real estate is time consuming and increasingly prohibitive.

(2) Adding Sectors

Adding sectors such as changing from 3 sectors to 6 sectors is a useful way to approximate the introduction of new cells. However, this does not quite double the capacity as the "petals" of 6 sector coverage do not interleave as well as 3 sector coverage.

(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.

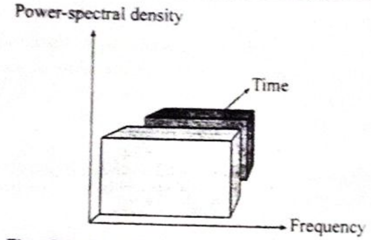

Fig. : Principle behind time division multiple access

- 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).

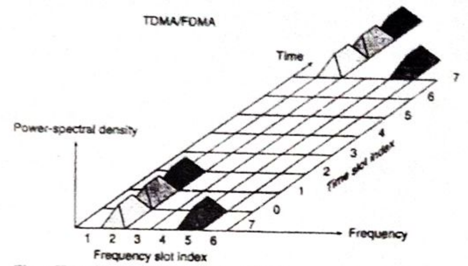

Fig. : How the Global System for Mobile communications (GSM) combines time division multiple access with frequency division multiple access.

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.

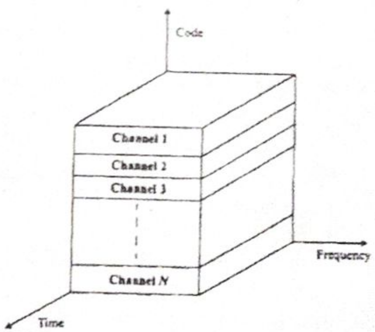

Fig. : CDMA in which each channel is assigned a unique PN code which is orthogonal to PN codes used by other users.

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.

In the receiver, we now have to invert the spreading operation. This can be achieved by correlating the received signal with the spreading sequence. This process reverses bandwidth spreading, so that after correlation, the desired signal again has a bandwidth of $1/T_S$. In addition to the desired-signal, the received signal also contains noise, other wideband interferers, and possibly narrow band interferers. Note that the effective bandwidth of noise and wideband interferers is not significantly affected by the despreading operation, while narrow band interferers are actually spread over a bandwidth W. As part of despreading, the signal passes through a low-pass filter of bandwidth $B = 1/T_S$. This leaves the desired signal essentially unchanged, but reduces the power of noise, wideband interferers, and narrow band interferers by a factor M_C. At the symbol demodulator, DSSS thus has the same Signal to Noise Ratio (SNR) as a narrow band system: for a narrow band system, the noise power at the demodulator is N_0/T_S. For a DSSS system, the noise power at the receiver input is $N_0/T_C = N_0 M_C/T_S$, which is reduced by narrow band filtering (by a factor of M_C); thus, at the detector input, it is N_0/T_S. A similar effect occurs for wideband interference.

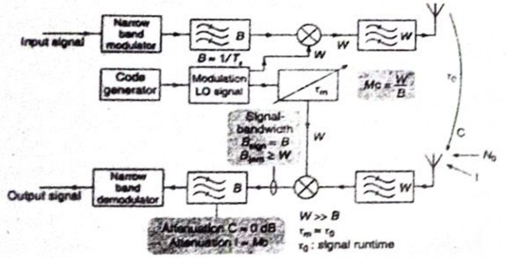

Fig. : Block diagram of a direct sequence-spread-spectrum transmitter and receiver

Q.16 *What are the main concerns for Wireless communications? Explain the system capacity.*

Ans. System Capacity : Wireless communications deal with at least two main concerns: coverage and capacity.

1. Channel Capacity : One fundamental concept of information theory is one of channel capacity, or how much information can be transmitted in a communication channel. In the 1940's Claude Shannon invented formal characterization of information theory and derived the well-known Shannon's capacity theorem. That theorem applies to wireless communications. The Shannon capacity equation gives an upper bound for the capacity in a non-faded channel with added white Gaussian noise:

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

where

C = capacity (bits/s),

W=bandwidth (Hz),

S/N= signal to noise (and interference) ratio.

That capacity equation assumes one transmitter and one receiver, though multiple antennas can be used in diversity scheme on the receiving side. The formula will be revisited for multi-antenna systems. The equation singles out two fundamentally important aspects: bandwidth and SNR.

2. Cellular Capacity

Practical capacity of many wireless systems are far from the Shannon's limit (although recent standards are coming close to it); and practical capacity is heavily dependent on implementation and standard choices.

Digital standards deal in their own way with how to deploy and optimize capacity. Most systems are limited by channel width, time slots, and voice coding characteristics. CDMA systems are interference limited, and have tradeoffs between capacity, coverage, and other performance metrics (such as dropped call rates or voice quality).

Cellular Analog Capacity : Fairly straight forward, every voice channel uses a 30 kHz frequency channel, these frequencies may be reused according to a reuse pattern, the system is FDMA. The overall capacity simply comes from the total amount of spectrum, the channel width and the reuse pattern.

TDMA/FDMA Capacity : In digital FDMA systems, capacity improvements mainly come from the voice coding and elaborate schemes (such as frequency hopping) to decrease reuse factor. The frequency reuse factor hides a lot of complexity; its value depends greatly on the signal to interference levels acceptable to a given cellular system. TDMA systems combine multiple time slots per channels.

CDMA Capacity : A usual capacity equation for CDMA systems may be fairly easily derived as follows (for the reverse link): first examine a base station with N mobiles, its noise and interference power spectral density dues to all mobiles in that same cell is $I_{SC} = (N - 1)S\alpha$, where S is the received power density for each mobile, and α is the voice activity factor. Other cell interferences I_{OC} are estimated by a reuse fraction β of the same cell interference level, such that $I_{OC} = \beta I_{SC}$; (usual values of β are around 1/2). The total noise and interference at the base is therefore $N_t = I_{SC}(1 + \beta)$. Next assume the mobile signal power density received at the base station is $S = RE_b/W$. Eliminating I_{SC}, we derive:

$$N = 1 + \frac{W}{R} \cdot \frac{1}{E_b/N_t} \cdot \frac{1}{\alpha} \cdot \frac{1}{1+\beta}$$

where

- W is the channel bandwidth (in Hz),

- R is the user data bit rate (symbol rate in symbol per second),
- E_b/N_t is the ratio of energy per bit by total noise (usually given in dB E_b/N_t H ≈ 7dB),
- α is the voice activity factor (for the reverse link), typically 0.5,
- β is the interference reuse fraction, typically around 0.5, and represents the ratio of interference level from the cell in consideration by interferences due to other cells. (The number $1 + \beta$ is sometimes called reuse factor, and $1/(1 + \beta)$ reuse efficiency)

This simple equation gives us a number of voice channels in a CDMA frequency channel.

We can already see some hints of CDMA optimization and investigate certain possible improvement for a 3G system. In particular: improving α can be achieved with dim and burst capabilities, β with interference mitigation and antenna downtilt considerations, R with vocoder rate, W with wider band CDMA, E_b/N_t with better coding and interference mitigation techniques.

Some aspects however are omitted in this equation and are required to quantify other capacity improvements mainly those due to power control, and softer/soft handoff algorithms.

Of course other limitations come into play for wireless systems, such as base station (and mobile) sensitivity, which may be incorporated into similar formulas; and further considerations come into play such as: forward power limitations, channel element blocking, backhaul capacity, mobility, and handoff.

□□□

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 π/4 QPSK Modulation.*

Ans. π/4 QPSK Modulation: In a π/4 QPSK modulation, signaling points of the modulated signal are selected from two QPSK constellations which are shifted by π/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

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

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

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

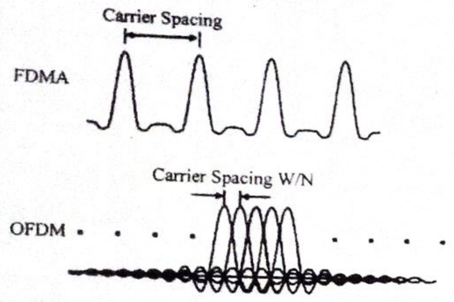

Fig. : Principle behind orthogonal frequency division multiplexing : N carriers within a bandwidth of W

Q.9 ***Create a Structure of an orthogonal frequency division multiplexing transmission chain with cyclic prefix and one-tap equalization.***

Ans. The block diagram of an OFDM system, including the cyclic prefix, is given in Figure. The original data stream is S/P converted. Each block of N data symbols is subjected to an IFFT, and then the last $N\,T_{cp}/T_S$ 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. The resulting samples (which can be interpreted as the samples in the frequency domain) are "equalized" by means of one-tap equalization – i.e., division by the complex channel attenuation – on each carrier.

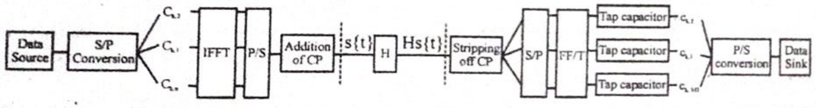

Fig. : Structure of an orthogonal frequency division multiplexing transmission chain with cyclic prefix and one-tap equalization

Q.10 ***Explain the principle of the cyclic prefix.***

Ans. Cyclic Prefix : Let us first define a new base function for transmission :

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

where again W/N is the carrier spacing, and $\hat{T}_S = N/W$. The symbol duration T_S is now $T_S = \hat{T}_S + T_{cp}$. This definition of the base function means that for duration $0 < t < \hat{T}_S$ the "normal" OFDM symbol is transmitted. It can be easily seen by substituting in Eq. that $g_n(t) = g_n(t + N/W)$. Therefore, during time $-T_{cp} < t < 0$, 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 $-T_{cp} < t < 0$ is a copy of s(t) during the last part, $\hat{T}_S - T_{cp} < t < \hat{T}_S$. This prepended part of the signal is called the "cyclic prefix."

Now that we know what a cyclic prefix is, let us investigate why it is beneficial in delay-dispersive channels. When transmitting any data stream over a delay-dispersive channel, the arriving signal is the linear convolution of the transmitted signal with the channel impulse response. The cyclic prefix converts this linear convolution into a cyclica convolution.

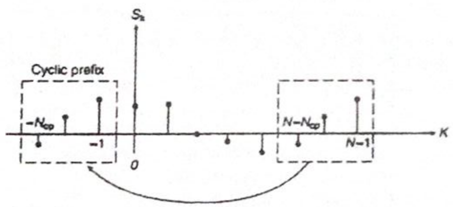

Fig. : Principle of the cyclic prefix. $N_{cp} = NT_{cp}/(N/W)$ is the number of samples in the cyclic prefix

During the time $-T_{cp} < t < -T_{cp} + \tau_{max}$, where τ_{max} i the maximum excess delay of the channel, the received signa suffers from "real" Inter Symbol Interference (ISI), a echoes of the last part of the preceding symbol interfere wit the desired symbol. This "regular" ISI is eliminated b discarding the received signal during this time interval. Durin the remainder of the symbol, we have cyclical ISI; especiall it is the last part of the current (not the preceding) symbc that interferes with the first part of the current symbol. In th

following, we show how can extremely simple mathematical operation can eliminate the effect of such a cyclical convolution.

Q.11 ***What is the Origin of the Peak-to-Average Ratio Problem.***

Ans. Origin of the Peak-to-Average Ratio Problem : One of the major problems of OFDM is that the peak amplitude of the emitted signal can be considerably higher than the average amplitude. 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. However, sometimes, the signals on the subcarriers add up constructively, so that the amplitude of the signal is proportional to N, and the power thus goes with N^2. We can thus anticipate the (worst case) power PAR to increase linearly with the number of subcarriers.

We can also look at this issue from a slightly different point of view: the contributions to the total signal from the different subcarriers can be viewed as random variables (they have quasi-random phases, depending on the sampling time as well as the values of the symbol with which they are modulated). If the number of subcarriers is large, we can invoke the central limit theorem to show that the distribution of the amplitudes of in-phase components is Gaussian, with a standard deviation $\sigma = 1/\sqrt{2}$ (and similarty for the quadrature components) such that mean power is unity. Since both in-phase and quadrature components are Gaussian, the absolute amplitude is Rayleigh distributed. Knowing the amplitude distribution, it is easy to compute the probability that the instantaneous amplitude will lie above a given threshold, and similarly for power. For example, there is a $\exp(-10^{6/10}) = 0.019$ probability that the peak power is 6dB above the average power. Note that the Rayleigh distribution can only be an approximation for the amplitude distribution of OFDM signals; an actual OFDM signal has a bounded amplitude (N• amplitude of signal on one subcarrier), while realizations of a Rayleigh distribution can take on arbitrarily large values.

Q.12 ***What are the methods to deal with the Peak-to-Average Power Ratio (PAPR).***

Ans.

1. Put a power amplifier into the transmitter that can amplify linearly up to the possible peak value of the transmit signal. This is usually not practical, as it requires expensive and power-consuming class-A amplifiers. The larger the number of subcarriers N, the more difficult this solution becomes.
2. Use a nonlinear amplifier, and accept the fact that amplifier characteristics will lead to distortions in the output signal. Those nonlinear distortions destroy orthogonally between subcarriers, and also lead to increased out-of-band emissions (spectral regrowth – similar to third-order inter-modulation products – such that the power emitted outside the nominal band is increased). The first effect increases the BER of the desired signal, while the latter effect causes interference to other users and thus decreases the cellular capacity of an OFDM system. This means that in order to have constant adjacent channel interference we can trade off power amplifier performance against spectral efficiency (note that increased carrier separation decreases spectral efficiency).
3. Use PAR reduction techniques.

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 :

$$p1_D(t) = \sum_{i=-\infty}^{\infty} b1_i\,g(t - iT_S) = b1_i * g(t)$$
$$p2_D(t) = \sum_{i=-\infty}^{\infty} b2_i\,g(t - iT_S) = b2_i * g(t)$$

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.

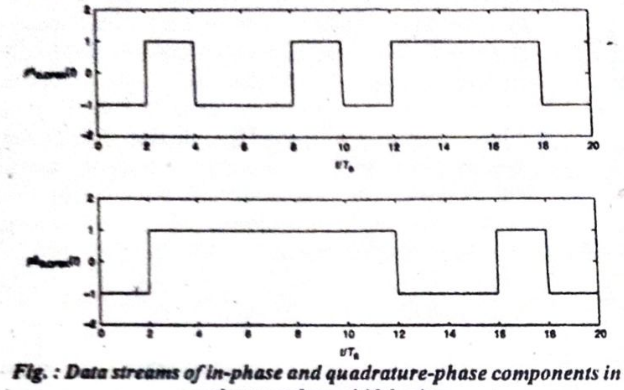

Fig. : Data streams of in-phase and quadrature-phase components in quadrature-phase shift keying.

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 $\sqrt{2\,E_B / T_B}\,\exp(j\Phi_S(t))$ 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 $t = iT_S$

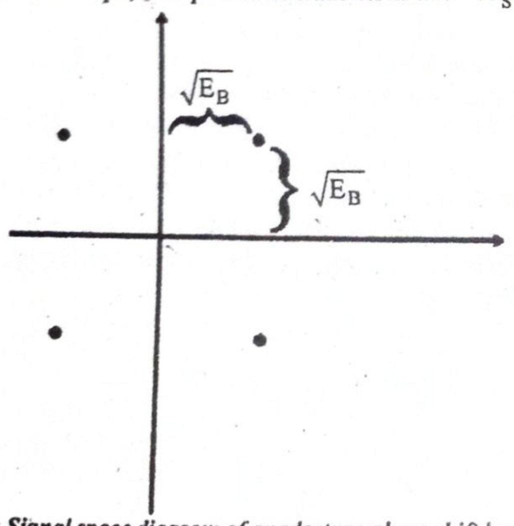

Fig. : Signal space diagram of quadrature-phase shift keying.

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.

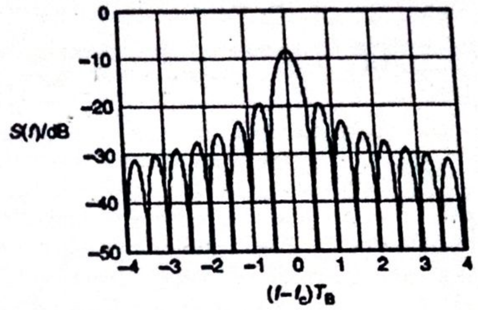

Fig. : Normalized power-spectral density of quadrature-phase shift keying

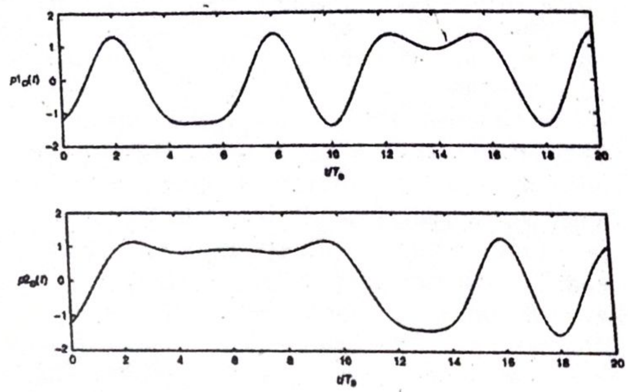

Fig. : Quadrature amplitude modulation pulse sequence

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 1-bit duration.

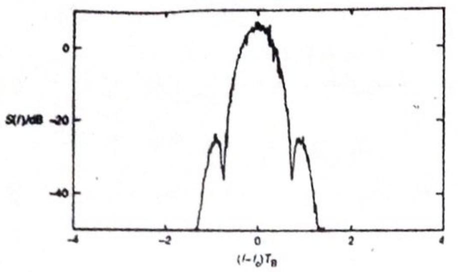

Fig. : Gaussian minimum shift keying power-spectral density (from simulations)

Q.16 *Explain the Structure of Wireless communication link.*

Ans. Structure of Wireless Communication Link : The structure of wireless communication link in wireless operations permit services like long-range communications which are impossible or impractical to implement with the wires usage in communication. This term is commonly used in the telecommunication industry in reference to telecommunications systems like (e.g. are radio transmitters and receivers and remote controls etc.) which use some form of the energy like (e.g. are radio waves and the acoustic energy, etc.) which is used to transfer information without the usage of wires.The information is then transferred in this manner over both short and long distances.

Transceiver Block Diagram Structure : Each of the gigabit transceiver block has a clock multiplier unit CMU which provides clocking flexibility and supports a range of incoming data streams. In each CMU two transmitter phase-locked loops that is PLLs which generates the required clock frequencies that is based upon the synthesis of an input reference clock.In each transmitter PLL supports all multiplication factors to allow the use of various input clock frequencies during transmission. Both of the transmitter's PLLs are identical for which they support data ranges from 600 Mbps to 6.375 Gbps data transfer. But however each PLL is configured to support different data rates where each transmitter PLL drives four channels. During PIPE x8 mode the transmitter PLL of the master transceiver block drives upto eight channels where CMU block is active both in single- and double-width modes and is powered off when not in use.

The Simplified models of wireless communication links

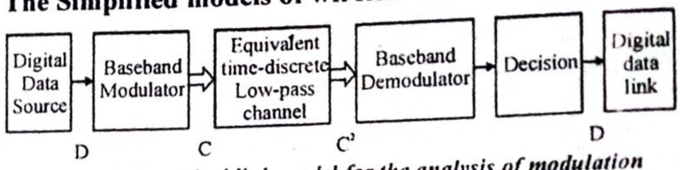

Fig. : Mathematical link model for the analysis of modulation formats

This is often preferable to have simplified models for wireless communication links. Moreover the analog radio channels with the downconverters, upconverters, RF elements and noise interfere the signals and it is then added to time discrete low pass channel during transmission.The other simplified models use a digital representation of the channel suitable for the analysis of the coding scheme.

The Modulation Formats : The most simple modulation is binary modulation where +1 bit value is mapped to one specific wave form while a –1 bit value is mapped to a different wave form. During choosing of a modulation wave format in wireless system the ultimate goal is to transmit with certain energy as much as information can transmit over a channel.

$$G_{N_a}(f,\alpha,T)=\begin{cases}1 & 0\le|2\pi f|\le(1-\alpha)\frac{\pi}{T}\\ \frac{1}{2}\left(1-\sin\left(\frac{T}{2\alpha}\left(|2\pi f|-\frac{\pi}{T}\right)\right)\right) & (1-\alpha)\frac{\pi}{T}\le|2\pi f|\le(1+\alpha)\frac{\pi}{T}\\ 0 & (1+\alpha)\frac{\pi}{T}\le|2\pi f|\end{cases}$$

the spectrum of raised cosine pulse is :

$$G_N(f,\alpha,T)=\frac{1}{\sqrt{1-\frac{\alpha}{4}}}.T.G_N0(f,\alpha,T)\exp(-j\pi f\,T_s)$$

Structure of a Demodulator

The single chip QAM demodulator with low Implemenation loss is a :

- Double Loop AGC for optimum usage of the A/D Converter.
- The delay in half Nyquist filter and equaliser require double carrier recovery loop structure to achieve high performance on phase noise and microphonics.
- The adaptive equaliser LE/MSE or * LE/ZF preferred for QAM with M£64* DFE/MSE required for QAM with M>64.

❑❑❑

MULTIPATH MITIGATION TECHNIQUES 4

IMPORTANT QUESTIONS

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?*

Ans. Signal copies may undergo different attenuation, distortions, delays and phase shifts in the signal.

The aim of microdiversity is to calculate the system capacity.

Q.4 *What are the factors used in adaptive algorithms?*

Ans.

- Rate of convergence
- Maladjustments
- Computational complexity

Q.5 *List the benefit of RAKE receiver?*

Ans. Reduces the multipath interference by combining direct and reflected signals in the receiver.

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.

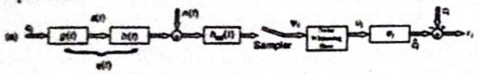

Fig. 1 : Linear equalizer in the time domain

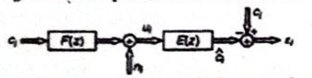

Fig. 2 : Time-discrete equivalent system in the z-transform domain

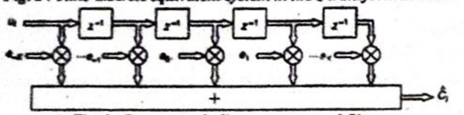

Fig. 3 : Structure of a linear transversal filter.

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

Q.7 What is Least Mean Square Algorithm?

Ans. Least Mean Square Algorithm : The LMS algorithm, also known as the stochastic gradient method, consists of the following steps:

1. Initialize the weights with values e_0.
2. With this value, compute an approximation for the gradient of the MSE. The true gradient cannot be computed, because it is an expected value. Rather, we are using an estimate for R and P – namely, their instantaneous realizations:

$$\hat{R}_n = u_n^* u_n^T$$

$$\hat{P}_n = u_n^* c_n$$

where subscript n indexes the iterations. The gradient is estimated as

$$\hat{\nabla}_n = -2\hat{P}_n + 2\hat{R}_n e_n$$

3. We next compute an updated estimate of the weight vector e by adjusting weights in the direction of the negative gradient:

$$e_{n+1} = e_n - \mu\hat{\nabla}_n$$

where μ is a user-defined parameter that determines convergence and residual error.

4. If the stop criterion is fulfilled e.g., the relative change in weight vector falls below a predefined threshold – the algorithm has converged. Otherwise, we return to step 2.

It can be shown that the LMS algorithm converges if

$$0 < \mu < \frac{2}{\lambda_{max}}$$

Here λ_{max} is the largest eigen value of the correlation matrix R. The problem is that we do not known this eigen value (computing it requires larger computational effort than inverting the correlation matrix). We thus have to guess values for μ. If μ is too large, we obtain faster convergence, but the algorithm might sometimes diverge. If we choose μ too small, then convergence is very probable, but slow. Generally, convergence speed depends on the condition number of the correlation matrix (i.e., the ratio of largest to smallest eigen value): the larger the condition number, the slower the convergence f the LMS algorithm.

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.

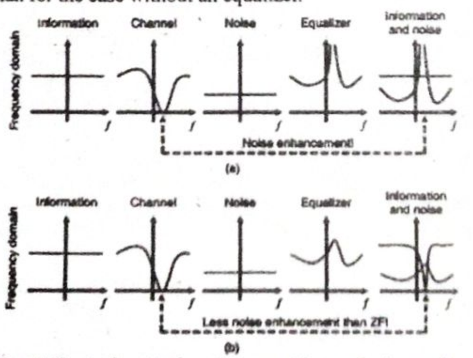

Fig. (a) Illustration of noise enhancement in zero-forcing equlizer (b) which is mitigated in an MMSE linear equilizer

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

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

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

Q.9 How to calculate error probability in fading channels?

Ans. Error Probability in Fading Channels : For AWGN channels, the advantages of the alternative representation of the Q-function are rather limited. They allow a simpler formulation for higher order modulation formats, but do not exhibit significant advantages for the modulation formats that are mostly used in practice. The real advantage emerges when we apply this description method as the basis for computations of the BER in fading channels. We find that we have to average over the pdf of the SNR $pdf_\gamma(\gamma)$, as described in Eq.

$$\overline{BER} = \int Pdf_{\gamma B}(\gamma B) BER(\gamma B) d_{\gamma B}$$

We have now seen that the alternative representation of the Q-function allows us to write the SER (for a given SNR) in the generic form:

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

Thus, the average SER becomes

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

$$= \int_0^\infty pd\, f_\gamma(\gamma)\int_{\theta_1}^{\theta_2} f_1(\theta)\exp(-\gamma f_2(\theta))d\theta\, d\gamma$$

$$= \int_{\theta_1}^{\theta_2} f_1(\theta)\int_0^\infty pd\, f_\gamma(\gamma)\exp(-\gamma f_2(\theta))d\gamma\, d\theta$$

Let us now have a closer look at the inner integral:

$$= \int_0^\infty pd\, f_\gamma(\gamma)\exp(-\gamma f_2(\theta))d\gamma$$

We find that it is the moment-generating function of $pdf_\gamma(\gamma)$, evaluated at the point $-f_2(\theta)$. Remember (see also, e.g., Papoulis [1991]) that the moment-generating function is defined as the Laplace transform of the pdf of γ :

$$M_\gamma(s) = \int_0^\infty pd\, f_\gamma(\gamma)\exp(\gamma s)\, d\gamma$$

and the mean SNR is the first derivative, evaluated at $s = 0$:

$$\bar{\gamma} = \left.\frac{dM_\gamma(s)}{ds}\right|_{s=0}$$

Summarizing, the average SER can be computed as

$$\overline{SER} = \int_{\theta_1}^{\theta_2} f_1(\theta) M_\gamma(-f_2(\theta))d\theta$$

The next step is then finding the moment-generating function of the distribution of the SNR. Without going into the details of the derivations, we find that for a Rayleigh distribution of the signal amplitude, the moment generating function of the SNR distribution is

$$M_\gamma(s) = \frac{1}{1 - s\bar{\gamma}}$$

for a Rice distribution it is

$$M_\gamma(s) = \frac{1+K_r}{1+K_r - s\bar{\gamma}}\exp\left[\frac{K_r s\bar{\gamma}}{1+K_r - s\bar{\gamma}}\right]$$

and for a Nakagami distribution with parameter m:

$$M_\gamma(s) = \left(1 - \frac{s\bar{\gamma}}{m}\right)^{-m}$$

Q.10 Draw a block diagram of Rake Receiver

Ans.

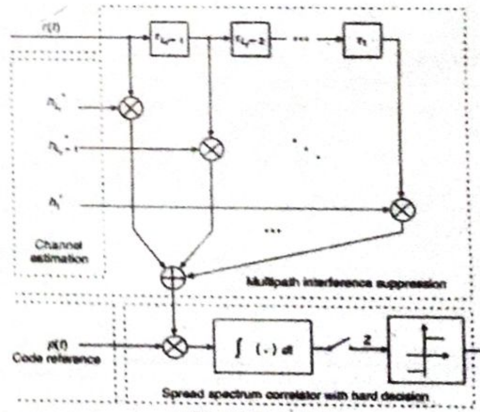

Fig. : Rake receiver

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When combining the signals as done in the Rake receiver we have

$$\gamma_{Rake} = \gamma_1 + \ldots + \gamma_6$$

Since only four MPCs carry energy, only four Rake fingers are effectively used. If the $\gamma_1, \ldots, \gamma_4$ are independent, the joint pdf of $f_{\gamma_1 \ldots \gamma_4}(\gamma_1, \ldots, \gamma_4) = f_{\gamma_1}(\gamma_1) \ldots f_{\gamma_4}(\gamma_4)$

$$\overline{BER} = \int d\gamma_1 pd\, f_{\gamma_1} \int d\gamma_2 pd\, f_{\gamma_2}(\gamma_2) \ldots \int d\gamma_4 pd\, f_{\gamma_4}(\gamma_4)$$

$$\int_0^{\pi/2} d\theta f_1(\theta) \prod_{k=1}^{N_r} \exp(-\gamma k f_2(\theta))$$

$$= \int_0^{\pi/2} \frac{1}{\pi} \prod_{k=1}^{4} \int_{\gamma_k} f_{\gamma_k}(\gamma_k) e^{\left(-\frac{\gamma_k}{\sin^2\theta}\right)} d_{\gamma_k} d\theta$$

$$= \int_0^{\pi/2} \frac{1}{\pi} \prod_{k=1}^{4} M_{\gamma_k}\left(-\frac{1}{\sin^2\theta}\right) d\theta$$

Thus, $$\overline{SER} = \int_0^{\pi/2} \frac{1}{\pi} \prod_{k=1}^{4} \left[\frac{\sin^2(\theta)}{\sin^2(\theta) + \bar{\gamma}_k}\right] d\theta$$

For the same transmit SNR $\bar{\gamma}_{TX} = 28.75$, we then get:

$$\overline{BER} = \int_0^{\pi/2} \frac{1}{\pi} \frac{\sin^2\theta}{\sin^2\theta + \bar{\gamma}_{TX}} \cdot \frac{\sin^2\theta}{\sin^2\theta + 0.33^2\bar{\gamma}_{TX}} \cdot \frac{\sin^2\theta}{\sin^2\theta + 0.1^2\bar{\gamma}_{TX}} \cdot \frac{\sin^2\theta}{\sin^2\theta + 0.07^2\bar{\gamma}_{TX}} d\theta$$

$$= 9.9 \times 10^{-4}$$

Q.12 ***What do you understand by diversity? Explain the Macro and Micro Diversity.***

Ans. Diversity : In the field of wireless communication channel ,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. The distance between the transmitters is much longer than the signal's wavelength.

In a cellular network macro-diversity implies that the antennas are typically situated in different BS sites or AP. Receiver macro-diversity is a form of combining antenna and requires an infrastructure that mediates the signals from the local antennas or receivers to a central receiver or central decoder. Transmitter macro-diversity is a form of simulcasting and where the same signal is sent from several nodes to the destined nodes. If the signals are sent over the same physical channel i.e.the channel frequency and spreading sequence, the transmitters are said to form a network with single frequency- a term which is used especially in the broadcasting world.

The aim is for combating fading and to increase the received signal strength and signal quality in exposed positions in between the BS or AP. Macro diversity also facilitates efficient broadcasing and multicasting services and where the same frequency channel can be used for all transmitters sending the same information to the destination nodes. The diversity scheme can be based on transmitter (downlink) macro-diversity and/or receiver (uplink) macro-diversity.

The Forms of Macrodiversity

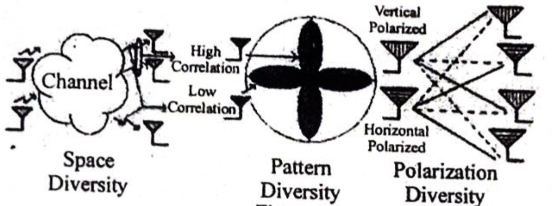


Fig.

The baseline form of macrodiversity is called as single-user macrodiversity. In this form a single user which may have multiple antennas can communicate with several BS. Therefore, 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 of the signal while communication.

Single-user macrodiversity can be :

1. Uplink macrodiversity
2. Downlink macrodiversity

The next more advanced form of macrodiversity in which multiple distributed users communicate with multiple distributed base stations in the same time and frequency resource while communication. This type of configuration has been shown to utilize available spatial DoF optimally that increasing the cellular system capacity and user capacity considerably.

Multi-user macrodiversity can be :

1. Macrodiversity multiple access channel (MAC)
2. Macrodiversity broadcast channel

Mathematical Description of Macro Diversity Representation

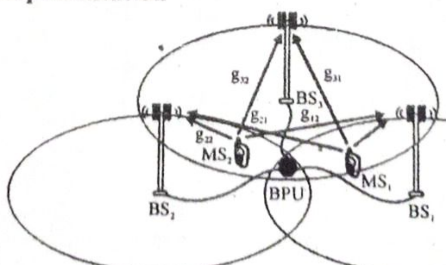


Fig.

The macrodiversity multi-user MIMO uplink communication system considered here includes N distributed single antenna users with n_R distributed single antenna BS.

Following is the well established narrow band flat fading MIMO system model whose input-output relationship can be given as :

$$y = Hx + n$$

Diversity Gain : In wireless communication system, **diversity gain** is the increase in signal-to-interference ratio due to some diverse schemes i.e. or how much the transmission power can be reduced when a diversity scheme is introduced without a performance loss factor. Diversity gain is usually expressed in dB and sometimes as a power ratio. An example is soft handoff gain in a system. For combining selection N signals are received and the strongest signal is selected. When the N signals are independent and distributed, the expected diversity gain has been shown to be as $\sum_{k=1}^{N} \frac{1}{k}$, expressed as a power ratio.

Microdiversity : Here we focus on a MIMO system that uses nT transmitting and nR receiving antenna elements, in a micro-diversity scenario. The aim here is to calculate the system capacity. This computation is based on the propagation model, and on the signal and noise variances as given by the RFFE model. Under the classical assumptions made by Foschini , the SNR is the same on all receiver branches, and is expressed as: SNR =P/N, where P is the average received power on each branch. This is the "standard" model. Here, the extension of the single antenna RFFE model described in section 3 to the MIMO case puts into question these assumptions, and leads to a modified relation between the SNR at the output of the RFFE block and the received power on a given receiver antenna. This new relationship also depends on the way the multi-antenna front-end is designed, in terms of number of analog RFFE blocks and in terms of AGC strategies. In any case, the resulting SNR versus received power relation affects capacity.

Fading Signal : In a typical wireless communication environment multiple propagation paths often exist from a transmitter to a receiver because of scattering by different objects in the communication. The signal copies following different paths may undergo different attenuation, distortions, delays and phase shifts in the signal. Constructive and destructive interference can occur at the receiver end of the destination. When the destructive interference occurs the signal power can be significantly lowered. This phenomenon is called as fading. The performance of a system (in terms of probability of error) can be severely degraded by fading phenomenon.Very often especially in mobile communication systems not only do multiple propagation paths exist and they are also time-varying. The result is a system with time-varying fading channel. Communication through these channels may be difficult. Special techniques can be required to achieve satisfactory performance.

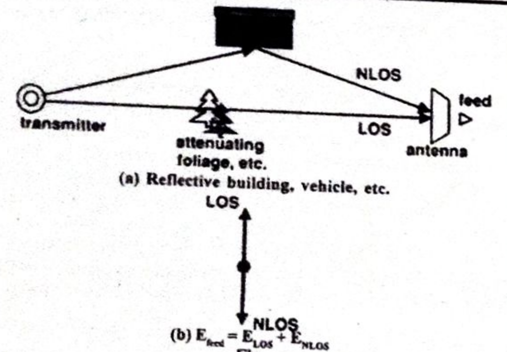


Fig.

Parameters of Fading Channels : The general time varying fading channel model is too complex for the understanding and performance analysis of wireless communication channels. Fortunately many practical wireless channels can be adequately approximated by the wide-sense stationary uncorrelated scattering (WSSUS) model i.e. model [2, 3]. In the WSSUS model the time-varying fading process is assumed to be a wide-sense stationary random process and the signal copies from the scattering by different objects are assumed to be independent on their own.

Q.13 ***What is the reason of choosing Non-linear Equalizer in place of Linear Equalizer? Explain Non-Linear Equalizer.***

Ans. Non-linear equalizers are used in applications where the channel distortion is to severe for a linear equalizer to mitigate the effect of channel impairments. The reason for choosing non-linear equalizers over linear equalizer is that the latter's performance in channel that exhibit nulls is not effective. Noise enhancement in these regions and long impulse response are a problem. The basic reason for this problem is that in linear filtering and noise are processed together, causing noise enhancement problem.

Based upon the importance, the noise-linear are classified as:

1. Decision Feedback Equalizer (DFE).
2. Maximum Likelihood Sequence Estimation (MLSE).

1. Decision Feedback Equalizer : A decision feedback equalizer is a simple non-linear equalizer, particularly useful for channel with severe amplitude distortion. DFE consist of a feed forward filter (FFF) and feedback filter (FBF). The fig. shows the block diagram of decision feedback equalizer. The feed forward section is nothing but a linear equalizer whose output is given to the decision device. The feedback section is driven by the output of the decision device.

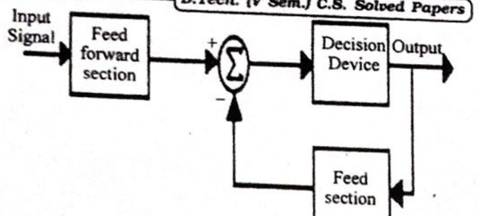


Fig. : Block diagram of decision feedback equilizer

The basic idea behind DFE is that once we have detected information symbol a decided upon, the ISI that induces on the future symbols can be estimated an subtracted out before detection of subsequent symbols. The minimum mean squared error that a decision feedback equalizer can achieve is:

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

The minimum mean squared error of decision feedback equalizer is smaller than that of a linear equalizer.

2. Maximum Likelihood Sequence Estimation : A DFE is not an optimum equalizer because it just outmatches the linear equalizer. MLSE gives optimum performance as it tests all the possible data sequences and choose that data as output which has the maximum probability. MLSE as an equalizer was first proposed by Forney [For78] in which he setup a basic estimator structure and implement it with Viterbi algorithm. However, the computational complexity of an MLSE increases with large delay spread and signal constellation size. The number of states of the Viterbi decoder is expressed as LM, where M is the number of symbols in constellation, and L is the channel-speed length. The block diagram of MLSE receiver based on DFE is shown in fig. The MLSE is optimal in the sense that it minimized the probability of a sequence error.

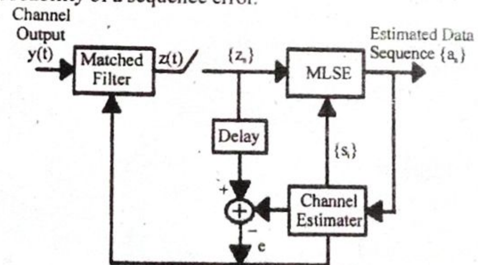


Fig. : Block diagram of MLSE

The MLSE requires the knowledge of :

1. The channel characteristics in order to compute the metrics for making decisions.
2. The statistical distribution of the noise corrupting the signal.

□□□

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"?***

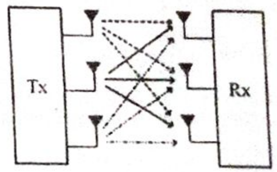
Ans. It is a preprocessing technique that performs transmit diversity and it is similar to equalization, but the main difference is that you have to optimize the precoder with a decoder. Channel equalization aims to minimize channel errors, but the precoder aims to minimize the error in the receiver 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.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.9 *What is the relation between Shannon's Law and MIMO spatial multiplexing*

Ans. As with many areas of science, there are theoretical boundaries, beyond which it is not possible to proceed. This is true for the amount of data that can be passed along a specific channel in the presence of noise. The law that governs this is called Shannon's Law, named after the man who formulated it. This is particularly important because MIMO wireless technology provides a method not of breaking the law, but increasing data rates beyond those possible on a single channel without its use.

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

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

Where C is the channel capacity in bits per second, W is the bandwidth in Hertz, and S/N is the SNR (Signal to Noise Ratio).

From this it can be seen that there is an ultimate limit on the capacity of a channel with a given bandwidth. However before this point is reached, the capacity is also limited by the signal to noise ratio of the received signal.

In view of these limits many decisions need to be made about the way in which a transmission is made. The modulation scheme can play a major part in this. The channel capacity can be increased by using higher order modulation schemes, but these require a better signal to noise ratio than the lower order modulation schemes. Thus a balance exists between the data rate and the allowable error rate, signal to noise ratio and power that can be transmitted.

While some improvements can be made in terms of optimizing the modulation scheme and improving the signal to noise ratio, these improvements are not always easy or cheap and they are invariably a compromise, balancing the various factors involved. It is therefore necessary to look at other ways of improving the data throughput for individual channels. MIMO is one way in which wireless communications can be improved and as a result it is receiving a considerable degree of interest.

Q.10 *Show the Mathematical MIMO spatial multiplexing.*

Ans. MIMO Spatial Multiplexing : To take advantage of the additional throughput capability, MIMO utilises several sets of antennas. In many MIMO systems, just two are used, but there is no reason why further antennas cannot be employed and this increases the throughput. In any case for MIMO spatial multiplexing the number of receive antennas must be equal to or greater than the number of transmit antennas.

To take advantage of the additional throughput offered, MIMO wireless systems utilise a matrix mathematical approach. Data streams $t_1, t_2, \ldots tn$ can be transmitted from antennas 1, 2, . . . n. Then there are a variety of paths that can be used with each path having different channel properties. To enable the receiver to be able to differentiate between the different data streams it is necessary to use. These can be represented by the properties h_{12}, travelling from transmit antenna one to receive antenna 2 and so forth. In this way for a three transmit, three receive antenna system a matrix can be set up:

$$r_1 = h_{11} t_1 + h_{21} t_2 + h_{31} t_3$$
$$r_2 = h_{12} t_1 + h_{22} t_2 + h_{32} t_3$$
$$r_3 = h_{13} t_1 + h_{23} t_2 + h_{33} t_3$$

Where r_1 = signal received at antenna 1, r_2 is the signal received at antenna 2 and so forth.

In matrix format this can be represented as:

$$[R] = [H] \times [T]$$

To recover the transmitted data-stream at the receiver it is necessary to perform a considerable amount of signal processing. First the MIMO system decoder must estimate the individual channel transfer characteristic h_{ij} to determine the channel transfer matrix. Once all of this has been estimated, then the matrix [H] has been produced and the transmitted data streams can be reconstructed by multiplying the received vector with the inverse of the transfer matrix.

$$[T] = [H]^{-1} \times [R]$$

This process can be likened to the solving of a set of N linear simultaneous equations to reveal the values of N variables.

In reality the situation is a little more difficult than this as propagation is never quite this straightforward, and in addition to this each variable consists of an ongoing data stream, this nevertheless demonstrates the basic principle behind MIMO wireless systems.

Q.11 *What is the Objective of Beamforming?*

Ans. Beamforming : The objective of beamforming is to use multiple antennas to form beams, increasing the SINR, and thereby the throughput, to a receiver. Wireless InSite currently supports two methods for beamforming:

- Maximum Ratio Transmission (MRT): maximizes the beam (adaptively) between T_x and R_x points
- Precoding Tables: Allows a user to define tabulated beams, supporting a number of approaches (codebooks, etc.) that allow for selection from predefined beams.

MRT uses information about the channel between the transmitter and receiver antennas to form an optimum beam to the receiver. In practice, this technique would typically be used for a time-division-duplexing (TDD) system, in which the uplink and downlink share the same band, allowing the receiver to send a pilot signal that can be used by the base stations to adaptively form this optimum beam.

Wireless InSite's precoding tables are more general purpose in nature. A user can define multiple sets of predefined beamforming weights, and Wireless InSite will evaluate the different weightings and choose the strongest beam to each receiver point. This simulates a MIMO base station that has predefined beams (e.g., codebooks), and uses one of a variety of methods to determine the best to use for a given channel.

Q.12 *What is the difference between Transmitter Diversity with Channel State Information and Transmitter Diversity without Channel State Information?*

Ans. Transmitter Diversity with Channel State Information : The first situation we analyze is the case where the T_x knows the channel perfectly. This knowledge might be obtained from feedback from the R_x, or from reciprocity principles. There is a complete equivalence between transmit diversity and receive diversity. In other words, 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.

Transmitter Diversity Without Channel State Information : In many cases, Channel State Information (CSI) is not available at the T_x. We then cannot simply transmit weighted copies of the same signal from different transmit antennas, because we cannot know how they would add up at the R_x. It is equally likely for the addition of different components to be constructive or destructive; in other words, we would just be adding up MPCs with random phases, which results in Rayleigh fading. We thus cannot gain any diversity (or beamforming). In order to give benefits, transmission of the signals from different antenna elements has to be done is such a way that it allows the R_x to distinguish different transmitted signal components. One way is delay diversity. In this scheme, signals transmitted from different antenna elements are delayed copies of the same signal. This makes sure that the effective impulse response is delay dispersive, even if the channel itself is flat fading. So, in a flat-fading channel, we transmit data streams with a delay of 1 symbol duration (relative to preceding antennas) from each of the transmit antennas. The effective impulse response of the channel then becomes

$$h(\tau) = \frac{1}{\sqrt{N_t}} \sum_{n=1}^{N_t} h_n \delta(\tau - nT_s)$$

where the h_n are gains from the n^{th} transmit antenna to the receive antenna, and the impulse response has been normalized so that total transmit power is independent of the number of antenna elements. The signals from different transmit antennas to the R_x act effectively as delayed MPCs. If antenna elements are spaced sufficiently far apart, these coefficients fade independently.

If the channel from a single transmit antenna to the R_x is already delay dispersive, then the scheme still works, but care has to be taken in the choice of delays for different 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.

Beam

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(γ) 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).

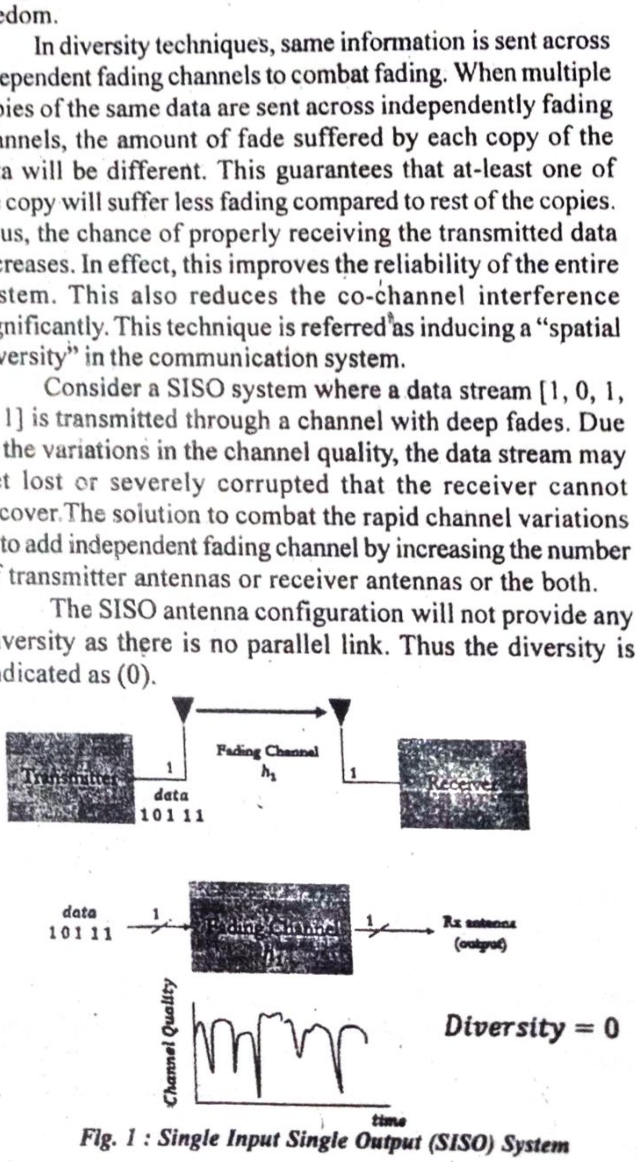


Fig. 1 : Single Input Single Output (SISO) System

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 $1 \times 2 = 2$.

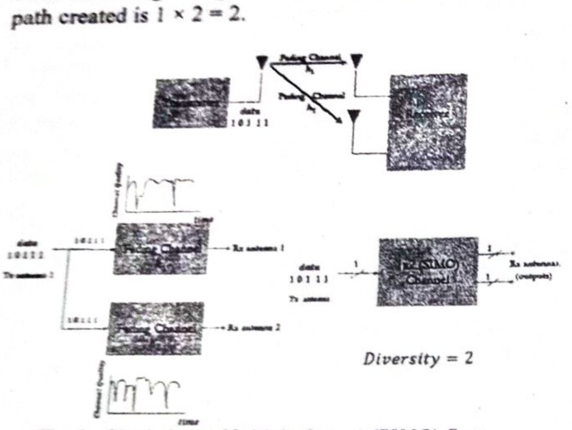


Fig. 2 : Single Input Multiple Output (SIMO) System

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 $2 \times 2 = 4$

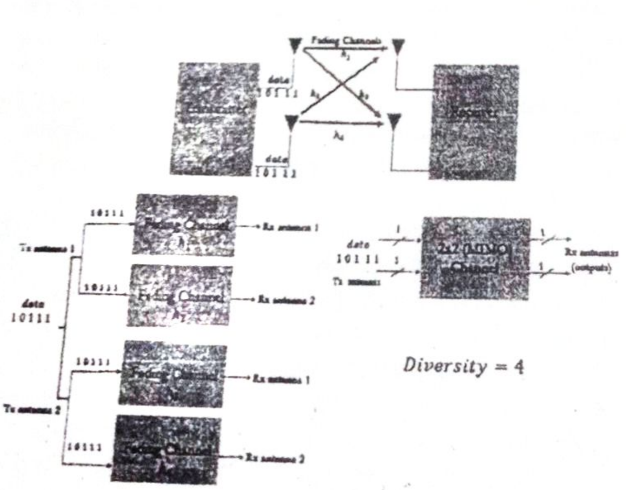


Fig. 3 : Multiple Input Multiple Output (MIMO) System

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.

users with favorable channel fading conditions to improve the system throughput. In order to achieve multiuser diversity and apply zero-forcing precoding, the CSI of all users are required at the base station. However, the amount of overall feedback information increases with the number of users. Therefore, it is important to perform a user selection at the receiver to determine the users which feedback their quantized CSI to the transmitter based on a pre-defined threshold.

DPC or DPC-like Nonlinear Precoding : Dirty paper coding is a coding technique that pre-cancels known interference without power penalty. Only the transmitter needs to know this interference, but full channel state information is required everywhere to achieve the weighted sum capacity. This category includes Costa precoding, Tomlinson-Harashima precoding and the vector perturbation technique.

Q.17 *Describe Capacity of a MIMO system over Fading Channels.*

Ans. A MIMO system is used to increase the capacity dramatically and also to improve the quality of a communication link. Increased capacity is obtained by spatial multiplexing and increased quality is obtained by diversity techniques (Space time coding). Capacity equations of a MIMO system over a variety of channels (AWGN, fading channels) is of primary importance.

- **Entropy :** The average amount of information per symbol (measured in bits/symbol) is called Entropy. Given a set of N discrete information symbols – represented as random variable $X \varepsilon \{x_1, x_2, \ldots, x_N\}$ having probabilities denoted by a Probability Mass Function $p(x) = \{p_1, p_2, \ldots, p_N\}$, the entropy of X is given by

$$h(X) = \sum_{i=1}^{N} p_i \log_2\left[\frac{1}{p_i}\right]$$

$$= \sum_{x \in X} p(x) \log_2\left[\frac{1}{p(x)}\right]$$

$$= -\sum_{x \in X} p(x) \log_2 p(x) \qquad \ldots (1)$$

Entropy is a measure of uncertainty of a random variable X, therefore reflects the amount of information required on an average to describe the random variable. in general, it has the following bounds

$$0 < h(X) < \log_2(N) \qquad \ldots (2)$$

Entropy hits the lower bound of zero (no uncertainty, therefore no information) for a completely deterministic system (probability of correct transmission $p_i = 1$). It reaches the upper bound when the input symbols x_i are equi-probable.

Capacity and Mutual Information : Following figure represents a discrete memoryless (noise term corrupts the input symbols independently) channel, where the input and output are represented as random variables X and Y respectively. Statistically, such a channel can be expressed by transition or conditional probabilities. That is, given a set of inputs to the channel, the probability of observing the output of the channel is expressed as conditional probability p(Y/X).

Fig.

For such a channel, the mutual information I(X; Y) denotes the amount of information that one random variable contains about the other random variable

$$I(X; Y) = h(X) - h(Y|X) \qquad \ldots (3)$$

h(X) is the amount of information in X before observing Y and thus the above quantity can be seen as the reduction of uncertainty of X from the observation of Y.

The information capacity C is obtained by maximizing this mutual information taken over all possible input distributions p(x).

$$C = \max_{p(x)} I(X; Y) \qquad \ldots (4)$$

MIMO Flat Fading Channel Model :

A $N_t \times N_r$ MIMO system over a flat fading channel can be represented in the complex baseband notation.

$$y = Hx + n \qquad \ldots (5)$$

where,

y = Received response from the channel – dimension $(Nr \times 1)$

H = The complex channel matrix of dimension $(N_r \times N_t)$

x = Vector representing transmitted signal – dimension $(N_t \times 1)$. Assuming Gaussian signals i.e., $x \sim N(0, K_x)$, where $K_x \triangleq E[xx^H]$ is the covariance matrix of the transmit vector x

N_t = The number of transmit antennas

N_r = The number of receive antennas

n = Complex baseband additive white Gaussian noise vector of dimension $(N_r \times 1)$. It is assumed that the noise is spatially white $n \sim (0, K_n)$ where K_n is the covariance matrix of noise.

Signal Covariance Matrices : It was assumed that the input signal vector x and the noise vector n are uncorrelated. Therefore, the covariance matrix of the received signal vector is given by

$$E[YY^H] = E[(HX + N)(HX + N)^H]$$

$$= HK_xH^H + K_n \qquad \ldots (6)$$

In the above equation, the H operator on the matrices denote Hermitian transpose operation. Thus, there are three covariance matrix involved here

$K_x = E[XX^H]$ – Covariance matrix of input signal vector

$K_y = E[YY^H]$ – Covariance matrix of channel response vector

$K_n = E[NN^H]$ – Covariance matrix of noise vector

Channel State Information

The knowledge of the channel matrix H, at the transmitter is called Channel State Information at the Transmitter (CSIT). If the receiver knows about the present state of the channel matrix that knowledge is called Channel State Information at the Receiver (CSIR).

Capacity with Transmit Power Constraint : Now, we would like to evaluate capacity for the most practical scenario, where the average power, given by $P = Tr(K_x) = E[||x||^2]$, that can be expensed at the transmitter is limited to P_t. Thus, the channel capacity is now constrained by this average transmit power, given as

$$C = \max_{p(x), P \le P_t} I(X; Y) \quad \dots (7)$$

For the further derivations, we assume that the receiver possesses perfect knowledge about the channel. Furthermore, we assume that the input random variable X is independent of the noise N and the noise vector is zero mean Gaussian distributed with covariance matrix K_n i.e., $N \sim N(0, K_n)$.

Note that both the input symbols in the vector x and the output symbols in the vector y take continuous upon transmission and reception and the values are discrete in time (Continuous input Continuous output discrete Memoryless Channel CCMC). For such continuous random variable, differential entropy – $h_d(.)$ is considered. Expressing the mutual information in terms of differential entropy.

$$I(X; Y) = h_d(Y) - h_d(Y|X)$$
$$= h_d(Y) - h_d(HX + N|X) \quad \dots (8)$$

Since it is assumed that the channel is perfectly known at the receiver, the uncertainty of the channel h conditioned on X is zero, i.e., $h_d(HX) = 0$.

Furthermore, it is assumed that the noise N is independent of the input X, i.e., $h_d(N|X) = h_d(N)$. Thus, the mutual information is

$$I(X; Y) = h_d(Y) - h_d(N) \quad \dots (9)$$

Following the procedure laid out here, the differential entropy $h_d(N)$ is calculated as

$$h_d(N) = \log_2(\det[\pi e K_n]) \quad \dots (10)$$

Using (6) and the similar procedure for calculating $h_d(N)$ above. The differential entropy $h_d(Y)$ is given by

$$h_d(Y) = \log_2(\det[\pi e(HK_xH^H + K_n + K_n)]) \quad \dots (11)$$

Substituting equations (10) and (11) in (9), the capacity is given by

$$C = h_d(Y) - h_d(N)$$
$$= \log_2(\det[\pi e(HK_xH^H + K_n)]) - \log_2(\det[\pi e K_n])$$
$$= \log_2(\det[HK_xH^H + K_n]) - \log_2(\det[K_n])$$
$$= \log_2(\det[(HK_xH^H + K_n)(K_n)^{-1}])$$
$$= \log_2(\det[(HK_xH^H)(K_n)^{-1} + I_{Nr}])$$
$$= \log_2(\det[I_{N_r} + (K_n)^{-1}(HK_xH^H)]) \quad \dots (12)$$

For the case, where the noise is uncorrelated (spatially white) between the antenna branches, $K_n = \frac{1}{\sigma_n^2} I_{N_r}$, where I_{N_r} is the identity marix of dimension $N_r \times N_r$.

Thus the capacity for MIMO flat fading channel can be written as

$$\boxed{C = \log_2\left[\det\left(I_{N_r} + \frac{1}{\sigma_n^2} HK_xH^H\right)\right]} \quad \dots (13)$$

The capacity equation (13) contains random variables, and therefore the capacity will also be random. For obtaining meaningful result, for fading channels two different capacities can be defined.

If the CSIT is **UNKNOWN** at the transmitter, it is optimal to evenly distribute the available transmit power at the transmit antennas. That is, $K_x = \frac{P_t}{N_t} I_{N_t}$, where I_{N_t} is the identity matrix of dimension $N_t \times N_t$.

Ergodic Capacity : Ergodic capacity is defined as the statistical average of the mutual information, where the expectation is taken over H

$$\boxed{C = E\left[\log_2\left[\det\left(I_{N_r} + \frac{1}{\sigma_n^2} HK_xH^H\right)\right]\right]} \quad \dots (14)$$

Outage Capacity : Defined as the information rate below which the instantaneous mutual information falls below a prescribed value of probability expressed as percentage – q.

$$\boxed{P\left(E\left(\left[\log_2\left[\det\left(I_{N_r} + \frac{1}{\sigma_n^2} HK_xH^H\right)\right]\right]\right)\right) < C_{out,q\%} = q\%} \quad \dots (15)$$

A word on capacity of a MIMO system over AWGN Channels : The capacity of MIMO system over AWGN channel can be derived in a very similar manner. The only difference will be the channel matrix. For the AWGN channel, the channel matrix will be a constant. The final equation for capacity will be very similar and will follow the lines of capacity of SISO over AWGN channel.

□□□