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APJ ABDUL KALAM TECHNOLOGICAL UNIVERSITY Eighth Semester B. Tech Degree Supplementary Examination October 2023 (2019 Scheme)

Course Code: ECT402

Course Name: WIRELESS COMMUNICATION

Max. Marks: 100 **Duration:3 Hours**

PART A (3 marks.)

1. Compare and contrast the analog and digital cellular systems.

Ans:

FEATURE	ANALOG CELLULAR SYSTEM	DIGITAL CELLULAR SYSTEM
Signal Encoding	Use continuous waveforms to represent sound or data. Signals are modulated to carry voice or data information.	Encode voice and data into binary digits (0s and 1s). These digits are then transmitted as discrete pulses of electricity.
Voice Quality	Often suffer from noise and interference, resulting in poorer voice quality, especially over long distances.	Digital systems typically offer better voice quality with reduced noise and clearer transmission, as the digital signals are less susceptible to interference.
Capacity and Efficiency	This have lower capacity and are less efficient in terms of spectrum utilization compared to digital systems.	It can carry more calls per unit of bandwidth, making them more efficient in spectrum usage.

Security	Analog systems are generally less secure as they can be intercepted and eavesdropped more easily.	Digital systems offer better security through encryption algorithms, making it harder for unauthorized parties to intercept and decode communication.
Data services	Analog systems have limited support for data services and are primarily designed for voice communication.	Digital systems provide better support for data services such as text messaging, internet browsing, and multimedia transmission.
Battery life	Analog phones generally have better battery life compared to digital phones.	Digital phones may consume more power due to additional processing requirements, potentially leading to shorter battery life.
Global Standardization	Analog systems had multiple standards across different regions, leading to compatibility issues.	Digital systems, such as GSM and CDMA, are more standardized globally, enabling seamless roaming and interoperability between networks.

2. What are the methods adopted for hand-off procedures?

Ans : **<u>Hard handoff:</u>**

- Channel in source cell is released and cahnnel in target cell is engaged
- Connection to source is broken even before connection to target is made(break before make)
- This is perceived as an event during call by n/w engr.
- Earlier in analog system beep is heard, but in digital it is unnoticeable.

Advantage:

- Phone h/w does not need to receive 2 or more channels in parallel

Disadvantage:

- Temporary disruption of a call

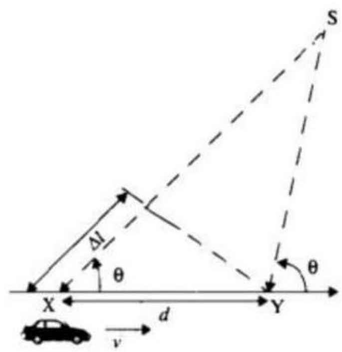
<u>Soft handoff:</u>

- Source channel is retained in parallel with channel in target (make before break)
- Reliability of connection is higher
- The soft handoff is perceived by network engineers as a state of the call, rather than a brief event
- A soft handoff may involve using connections to more than two cells, e.g. connections to three, four or more cells can be maintained by one phone at the same time. When a call is in a state of soft handoff the signal of the best of all used channels can be utilized for the call at a given moment or all the signals can be combined to produce a clearer copy of the signal.
- When such combining is performed both in the downlink(forward link) and the uplink(reverse link) the handoff is termed as softer
- This advantage comes at the cost of more complex hardware in the phone,which must be capable of processing several channels in parallel. Another price to pay for soft handoffs is use of several channels in the network to support just a single call. This reduces the number of remaining free channels and thus reduces the capacity of the network .

3. How does fading occur? Derive the expression for Doppler shift.

Ans: In wireless communication fading is a phenomenon in which the strength and quality of a radio signal fluctuate over time and distance. Fading is caused by a variety of factors, including multipath propagation, atmospheric conditions and the movement of objects in the transmission path. Fading can have a significant impact on the performance of wireless communication systems, particularly those that operate in high frequency band.

Doppler Shift



The phase change in the received signal due to the difference in path lengths is therefore

$$\Delta\phi = \frac{2\pi\Delta l}{\lambda} = \frac{2\pi v\Delta t}{\lambda}\cos\theta$$

and hence the apparent change in frequency, or Doppler shift, is given by fd where

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

4. Assume a receiver is located 10km away from a 50W transmitter. Given f = 900 MHz, Gt = 1 and Gr= 2. Find the power at receiver and RMS voltage at receiver antenna matched with 50 Ω resistor.

Ans:

using equation

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

Given,

Transmitter power, $P_t = 50W$

carrier frequency, $f_c = 900 MHz$

Transmitter anteena gain, $G_t = 1$

Receiver anteena gain, $G_r = 2$

Receiver anteena resistance $= 50\,\Omega$

$$P_r(d) = 10 \log \left[\frac{50 \times 1 \times 2 \times (1/3)^2}{(4\pi)^2\ 10000^2} \right]$$

$$= -91.5\ dBW = -61.5\ dBm$$

using equation

$$V_{ant} = \sqrt{P_r(d) \times 4 R_{ant}} = \sqrt{7 \times 10^{-10} \times 4 \times 50} = 0.374\ mV$$

5. How is the outage probability computed for a wireless channel?

Ans : In AWGN the probability of symbol error depends on the received SNR or, equivalently, on γs

• In a fading environment the received signal power varies randomly over distance or time as a result of shadowing and/or multipath fading

• Thus, in fading, γsis a random variable

• The performance metric when γs is random depends on the rate of change of the fading.

• The outage probability, Pout, defined as the probability that γs falls below a given value corresponding to the maximum allowable Ps

The outage probability relative to γ0 is defined as

$$P_{out} = p(\gamma_s < \gamma_0) = \int_0^{\gamma_0} p_{\gamma_s}(\gamma)\, d\gamma$$

• Where γ0 typically specifies the minimum SNR required for acceptable performance

• In Rayleigh fading the outage probability becomes

$$P_{out} = \int_0^{\gamma_0} \frac{1}{\bar{\gamma}_s} e^{-\gamma_s/\bar{\gamma}_s} \, d\gamma_s = 1 - e^{-\gamma_0/\bar{\gamma}_s}$$

• Inverting this formula shows that, for a given outage probability, the required average SNR γs is

$$\bar{\gamma}_s = \frac{\gamma_0}{-\ln(1 - P_{out})}$$

In decibels this means that 10 log γs must exceed the target 10 log γ0 by

$$F_d = -10 \log\left[-\ln(1 - P_{out})\right]$$

• in order to maintain acceptable performance more than 100 · (1− Pout) percent of the time.

• The quantity Fd is typically called the dB fade margin.

6. Explain the significance of using cyclic prefix in an OFDM system

Ans : This hybrid technique does not require contiguous spectrum between sub-bands, and it also allows more flexibility in spreading user signals over different-size sub- bands depending on their requirements. The cyclic prefix (CP) plays a crucial role in mitigating the effects of multipath fading and inter-symbol interference (ISI), thus ensuring reliable communication over wireless channels.

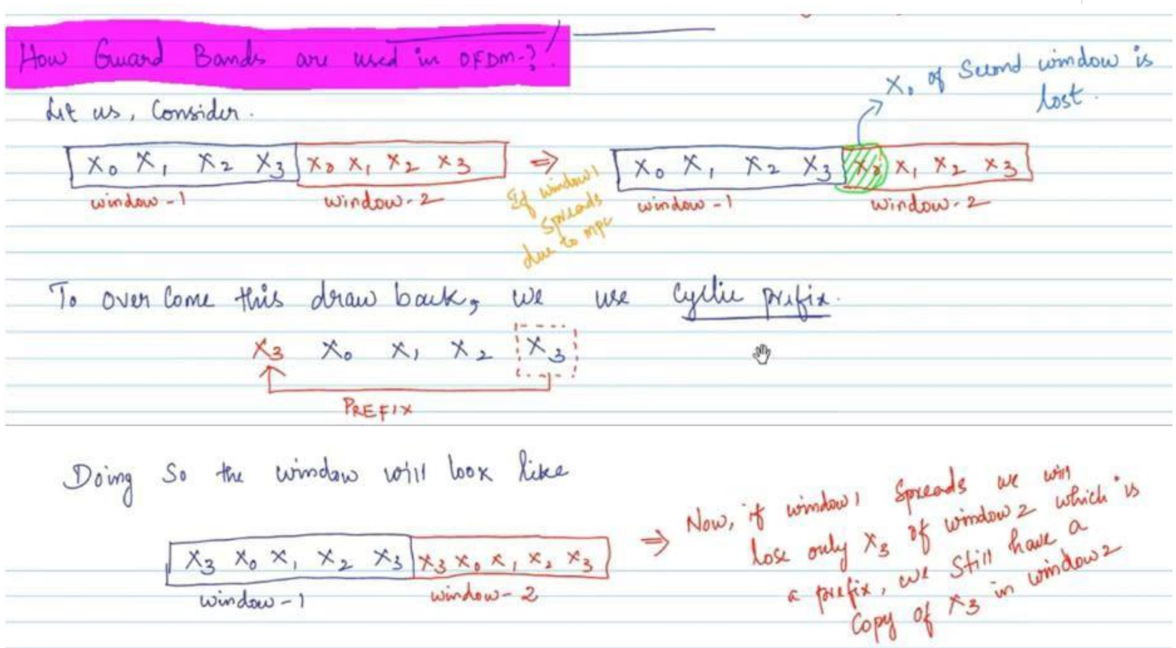
I. **Guard Interval**: The cyclic prefix is essentially a guard interval inserted at the beginning of each OFDM symbol.

II. **Multipath Fading Mitigation**: Wireless channels introduce multipath propagation, where transmitted signals reach the receiver via multiple paths due to reflections, diffractions, and scattering.

III. **Simplified Equalization**: The cyclic prefix simplifies the equalization process at the receiver. Instead of employing complex equalization techniques to counter the effects of multipath fading and ISI, the cyclic prefix allows for a simple and efficient frequency-domain equalization.

IV. **Robustness to Channel Variations**: OFDM systems with cyclic prefix exhibit robustness against variations in channel conditions, such as time-varying fading channels.

V. **Spectral Efficiency**: Despite the overhead introduced by the cyclic prefix, OFDM systems remain spectrally efficient.



7. Differentiate between microdiversity and macrodiversity.

Ans: **Micro-diversity Technique**: These techniques are used in a small-scale fading environment--two antennas are separated by a fraction of a meter.

Macro-diversity Technique: These techniques are used in a large scale fading environment- antennas are quite far apart and not shadowed (variation in signal strength caused by obstructions or obstacles in the propagation environment, such as buildings, trees, terrain, and other physical objects.)

8. Compare pros and cons of linear equaliser over non-linear equaliser.

Ans:

Pros of Linear Equalizers over Non-linear Equalizers:

1. Simplicity: Linear equalizers are generally simpler to design and implement compared to their non-linear counterparts. They often involve straightforward filtering techniques like FIR or IIR filters, which are easier to understand and implement.

2. Stability: Linear equalizers tend to be more stable, particularly in systems with rapidly changing channel conditions. They are less prone to convergence issues and instability compared to non-linear equalizers, making them more reliable in certain scenarios.

3. Low Complexity: Linear equalizers typically have lower computational complexity, making them suitable for real-time applications or environments with limited computational resources.

4. Precise Frequency Response Control: They offer precise control over the frequency response, allowing for targeted adjustments to specific frequency bands. This makes them suitable for applications where fine-tuning of the frequency response is critical.

Cons of Linear Equalizers compared to Non-linear Equalizers:

1. Limited Correction Capability: Linear equalizers may struggle to correct nonlinear distortions effectively, particularly in channels with severe nonlinearities. They are not as adept at compensating for nonlinear effects such as amplitude and phase distortion.

2. Less Effective in Highly Nonlinear Channels: In channels with significant nonlinearities, such as high-power amplifiers, linear equalizers may not provide sufficient compensation, leading to degraded performance.

3. Inability to Handle Nonlinear Noise: Linear equalizers are not well-suited for handling nonlinear noise or other nonlinear channel impairments beyond basic linear distortions.

4. Potential Performance Trade-offs: While linear equalizers offer simplicity and stability, they may sacrifice performance compared to non-linear equalizers in scenarios where nonlinear effects dominate the channel behavior.

9. Deduce the expression for critical frequency of an ionised region in terms of its maximum ionization density.

Ans: Critical frequency (f) The highest frequency that would be returned to earth in a wave directed for normal incidence. Maximum usable frequency (MUF).

The highest frequency that will be returned to earth for a given angle of incidence. If the angle of incidence to the normal is θ, the

$$\mathbf{MUF = fc / \cos\theta = fc \sec\theta} .$$

Critical value of electronic density for reflecting the wave back to earth is

$$\mathbf{Nmax = f^2 \cos^2\theta / 81} .$$

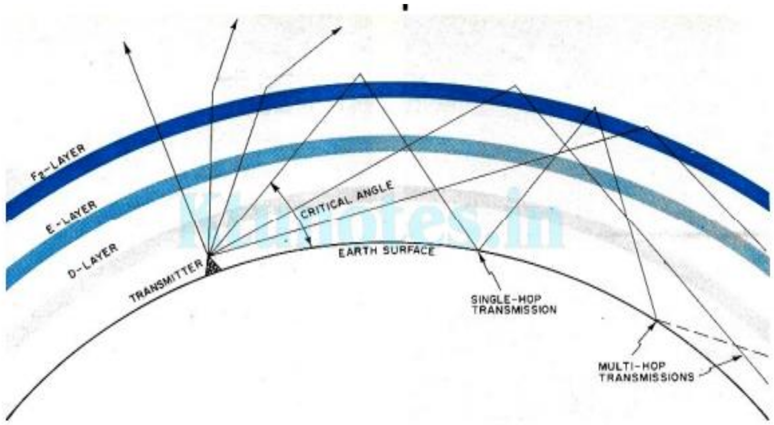
So critical frequency can be written as

$$\mathbf{fc = 9\sqrt{Nmax}}. \text{ And } \mathbf{MUF = 9\sqrt{Nmax} \sec\theta}$$

where θ is the angle of incidence MUF is usually 3 to 4 times of critical frequency.

10. Explain the mechanism of wave bending in ionosphere with suitable diagram.

Ans.



Upper part of atmosphere absorbs large quantity of radiant energy from sun, producing ionization. Ionized region consists of free electron, positive ions and negative ions.important ionizing agents are UV, α, β rays, cosmic rays and meteors.Different parts of ionosphere are called layers.Three principle layers during day time and are called E, F1 and F2. Beside theseregion below E, called D region.During night the F1 and F2 layers combine and form one layer called F layer and D region vanishes altogether.

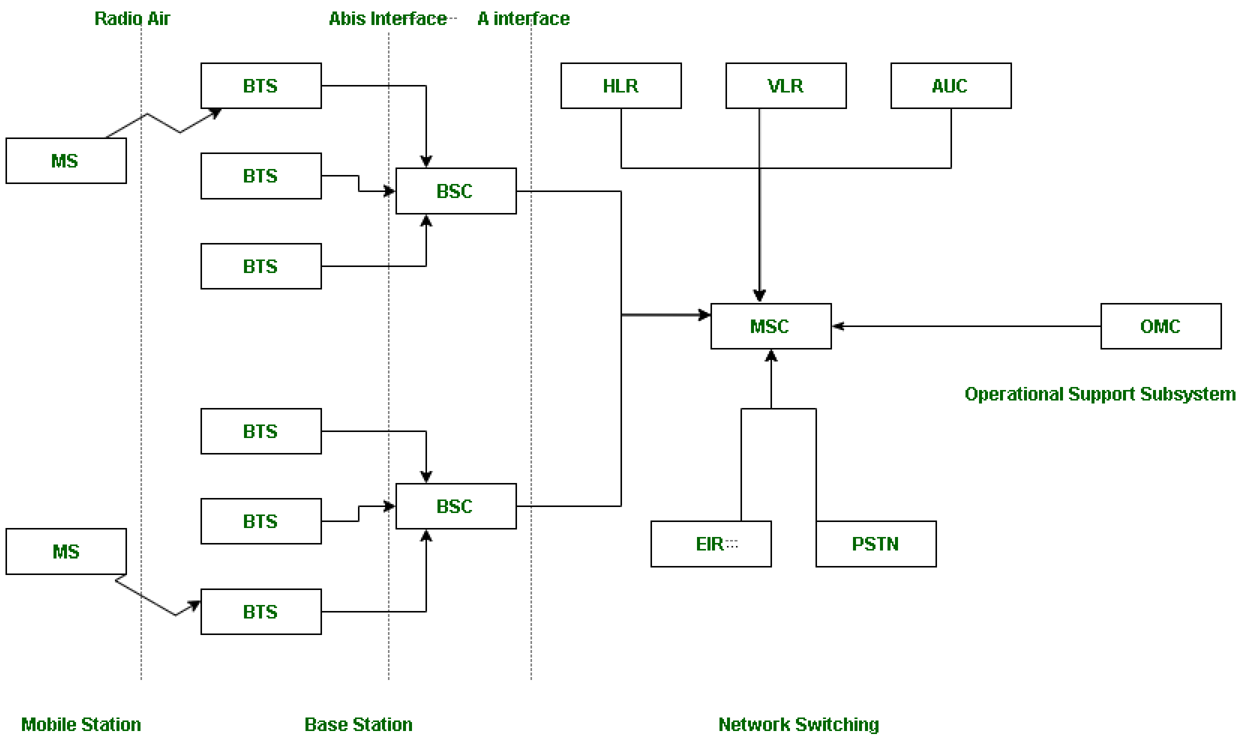
PART B

(14 marks)

MODULE 1

11. a) Describe the features of the GSM system architecture with the help of a neat block diagram

Ans:



1. MS : MS stands for Mobile System. MS comprises user equipment and software needed for communication with a mobile network. Mobile Station (MS) = Mobile Equipment(ME) + Subscriber Identity Module (SIM). Now, these mobile stations are connected to tower and that tower connected with BTS through TRX. TRX is a transceiver which comprises transmitter and receiver. Transceiver has two performance of sending and receiving.

2. BTS : BTS stands for Base Transceiver Station which facilitates wireless communication between user equipment and a network. Every tower has BTS.

3. BSC : BSC stands for Base Station Controller. BSC has multiple BTS. You can consider the BSC as a local exchange of your area which has multiple towers and multiple towers have BTS.

4. MSC : MSC stands for Mobile Switching Center. MSC is associated with communication switching functions such as call setup, call release and routing. Call tracing, call forwarding all functions are performed at the MSC level. MSC is having further components like VLR, HLR, AUC, EIR and PSTN.

- **VLR :** VLR stands for Visitor Location Register. VLR is a database which contains the exact location of all mobile subscribers currently present in the service area of MSC. If you are going from one state to another state then your entry is marked into the database of VLR.
- **HLR :** HLR stands for Home Location Register. HLR is a database containing pertinent data regarding subscribers authorized to use a GSM network.. If you purchase SIM card from in the HLR. HLR is like a home which contains all data like your ID proof, which plan you are taking, which caller tune you are using etc.
- **AUC :** AUC stands for Authentication Center. AUC authenticates the mobile subscriber that wants to connect in the network.
- **EIR :** EIR stands for Equipment Identity Register. EIR is a database that keeps the record of all allowed or banned in the network. If you are banned in the network then you can't enter the network, and you can't make the calls.
- **PSTN :** PSTN stands for Public Switched Telephone Network. PSTN connects with MSC. PSTN originally a network of fixed line analog telephone systems. Now almost entirely digital in its core network and includes mobile and other networks as well as fixed telephones. The earlier landline phones which places at our home is nothing but PSTN.

5.OMC : OMC stands for Operation Maintenance Center. OMC monitor and maintain the performance of each MS, BSC and MSC within a GSM system.

Three subsystem BSS, NSS and OSS are connected with each other via some interfaces. Total three interfaces are there:

1. **Air Interface :** Air interface is also known as UM interface. Interface between MS and BTS is called as UM interface because it is mobile analog to the U interface of ISDN.
2. **Abis Interface :** It is a BSS internal interface linking with BTS and BSC.
3. **A interface :** It provides communication between BSS and MSC.

b) How does cell splitting and sectoring improve the capacity and coverage of the cellular system.

Ans : **Cell splitting :**

Cell splitting is the process of subdividing congested cell in to smaller cell with their own Base station Corresponding reduction in antenna height Corresponding reduction in transmitter power. Splitting of cells reduces the cell size and thus more number of cells has to be used. Cell splitting allows a system to grow by replacing large cells by small cells, without upsetting the channel allocation. By defining new cells which have a smaller radius than the original cells and by installing these smaller cells (called microcells) between the existing cells, capacity increases due to the additional number of channels per unit area. Cells are split to add channels with no new spectrum usage Depending on traffic patterns the smaller cells may be activated/deactivated in order to efficiently use cell resources.

In the figure shows that the original base station A has been surrounded by six new microcell base stations.

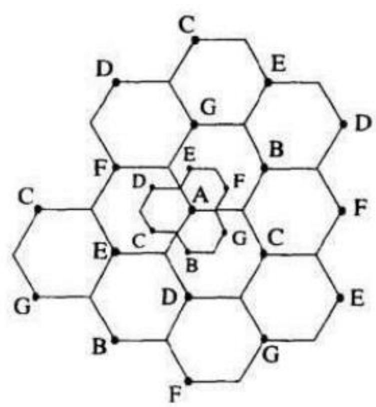
The smaller cells were added in such a way as to preserve the frequency reuse plan of the system.

Cell splitting scales the geometry of the cluster.

Suppose the cell radius of the new cell are reduced by half , reduce R to R/2.

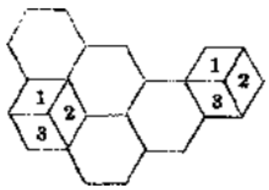
Transmission power reduction from to Pt1 to Pt2 .

Examining the receiving power at the new and old cell boundary.

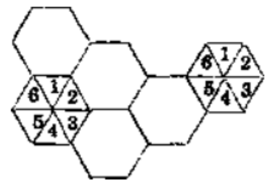


Cell sectoring :

The technique for decreasing co-channel interference and thus increasing system capacity by using directional antennas is called sectoring. The factor by which the co-channel interference is reduced depends on the amount of sectoring used. Cell Sectoring keeps R unchanged and reduces D/R. Capacity improvement is achieved by reducing the number of cells per cluster, thus increasing frequency reuse. It is necessary to reduce the relative interference without decreasing the transmitter power. The co-channel interference may be decreased by replacing the single omni-directional antenna by several directional antenna, each radiating within a specified sector. A directional antenna transmits to and receives from only a fraction of the total number of co-channel cells. Thus co-channel interference is reduced. A cell is normally partitioned into three 1200 sectors or six 60° sectors.



(a) 120^0 sectoring.



(b) 60^0 sectoring.

Radio transmitters called repeaters are used to provide coverage in these areas.

· Repeaters are bidirectional.

· Receive signals from the base station

· Amplify the signals

· Reradiates the signals.

· Received noise and interference is also reradiated.

OR

12. a) Explain the different channel assignment strategies used in cellular system.

Ans: It is a Frequency reuse scheme for increasing capacity and minimizing interference is required.

- For efficient utilization of the radio spectrum, a frequency reuse scheme is used.
- So that capacity is increased, interference is reduced.
- Channel assignment strategy improves the performance of the system.
- Used to manage calls when handoff is done.
- Minimize connection set-up time
- Adapt to changing load distribution Fault tolerance capability
- Low computation and communication overhead Minimize handoffs
- Maximize number of calls that can be accepted

Channel assignment strategies can be classified

1. Fixed Channel assignment

2. Dynamic Channel assignment

Fixed Channel assignment

- Channels are pre-allocated to the cells during planning phase.
- Each cell is allocated a predetermined set of voice channels.
- Any call attempt within the cell can only be served by the unused channels in that particular cell.
- If all the channels in that cell are occupied, the call is blocked and the subscriber does not receive service.
- Due to short term fluctuations in the traffic, FCA schemes are often not able to maintain high quality of service and capacity attainable with static traffic demands.
- One approach to address this problem is to borrow free channels from neighboring cells.
- Variation: Borrowing Strategy : Cell allowed to borrow channels from neighboring cells if it all of its own channels are already occupied
- Mobile Switching Center (MSC) supervises such borrowing procedures
- Ensure the borrowing of a channel does not disrupt or interfere with any of the call in progress in the donor cell.

Dynamic Channel assignment

- No pre-allocation: In a dynamic channel assignment strategy, voice channels are not allocated to different cells permanently.
- Each time a call request is made, the serving base station requests a channel from the MSC.
- MSC then allocates a channel to the requested cell using an algorithm that takes into account
- To ensure minimum quality of service, the MSC only allocates a given frequency if that frequency is not currently in use in the cell or any other cell which falls within the limiting reuse distance.
- Dynamic channel assignment reduces the likelihood of blocking increasing the capacity of the system.
- Dynamic channel assignment strategies require the MSC to collect real-time data on channel occupancy and traffic distribution on a continuous basis.

b) Enumerate the features of 4G wireless network

Ans:

- High usability: anytime, anywhere and with any technology.
- Support for multimedia services at low transmission cost.
- Higher bandwidth , tight network security.
- Much higher data rate up to 1Gbps

- Enhanced security and mobility
- Reduced latency for mission-critical applications
- High-definition video streaming and gaming
- Voice over LTE network VoLTE (use IP packets for voice)

Module II

13. a) Consider a wireless channel, where power falloff with distance follows the formula Pr(d)=Pt(d0/d)^3 for d0=50m. Assume the channel has bandwidth B =50KHz and AWGN with noise PSD N0/2 , Where N0=10-9 W/Hz. For a transmit power of 2W, find the capacity of this channel for a receive transmit distance of 200m and 1KM? What is your conclusion?

Ans: The received SNR is $\gamma = P_r(d)/N_0 B$

$$= 0.2/(10^{-9} \times 50 \times 10^3) = 160 = 22\,dB$$

for $d = 200m$.

$$\gamma = 0.02^3/10^{-9} \times 50 \times 10^3 = 0.16$$

$$10\log_{10}(0.16) = 7.9\,dB$$ for $d = 1000m$.

The corresponding capacities are,

$$C = B\log_2(1+\gamma)$$

$$= 50000\log_2(1+160) = 366.5\,kbps$$ for $d = 200m$

$$C = 50000\log_2(1+0.16) = 1.07\,kbps$$ for $d = 1000m$

b) Derive the expression for the impulse response model of a multipath channel

The channel is linear time-varying channel, where the channel characteristics changes with distance (hence time, t = d/v)

$$y(d,t) = x(t) \otimes h(d,t) = \int_{-\infty}^{\infty} x(\tau) h(d, t-\tau) d\tau$$

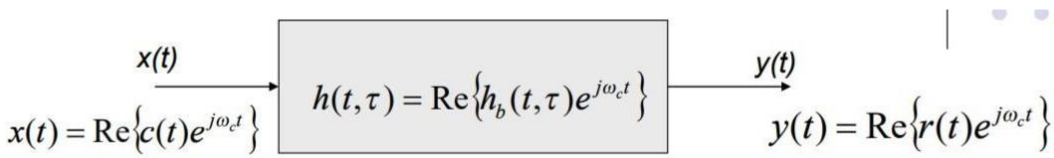
For a causal system, $h(d,t) = 0$ for $t < 0$; hence

$$y(d,t) = \int_{-\infty}^{t} x(\tau) h(d, t-\tau) d\tau$$

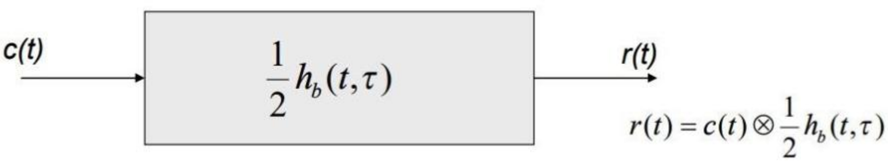
$$y(t) = \int_{-\infty}^{t} x(\tau) h(vt, t-\tau) d\tau = x(t) \otimes h(vt,t) = x(t) \otimes h(d,t)$$

We assume v is constant over short time.
x(t): transmitted waveform
y(t): received waveform
h(t,τ): impulse response of the channel. Depends on d (and therefore t=d/v)
and also to the <u>multiple delay</u> for the channel for a fixed value of t.
τ is the multipath delay of the channel for a fixed value of t.

$$y(t) = \int_{-\infty}^{\infty} x(\tau) h(t,\tau) d\tau = x(t) \otimes h(t,\tau)$$



Bandpass Channel Impulse Response Model $y(t) = x(t) \otimes h(t,\tau)$



Baseband Equivalent Channel Impulse Response Model

$$r(t) = c(t) \otimes \frac{1}{2}h_b(t,\tau)$$

$$x(t) = \mathrm{Re}\left\{c(t)e^{j2\pi f_c t}\right\} \quad \omega_c = 2\pi f_c$$

$$y(t) = \mathrm{Re}\left\{r(t)e^{j2\pi f_c t}\right\}$$

c(t) is the complex envelope representation of the transmitted signal
r(t) is the complex envelope representation of the received signal

$h_b(t,\tau)$ is the complex baseband impulse response

OR

14. a) What is the received power in dBm for a free space signal, whose transmit power is 1W and carrier frequency is 2.4GHz. If the receiver is at a distance of 1 mile (1.6 km) from the transmitter. What is the path loss in dB?

Ans:

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

$$\lambda = \frac{c}{f}$$

Given $P_t = 1W$

$f = 2.4 GHz$

$d = 1 mile = 1.6 km$

$$\text{Wave length } (\lambda) = \frac{c}{f} = \frac{3 \times 10^8}{2.4 \times 10^9} = \frac{3 \times 10^8}{2.4 \times 10^9} = \frac{1}{8} = 0.125m$$

$d = 1.6 \times 1000 = 16000m$

$$P_r = 1 \times \left(\frac{1 \times 1 \times (0.125)^2}{(4\pi)^2 \times 1600^2} \right)$$

$$= \frac{1}{(4\pi)^2 \times 1600^2} \times 0.015625$$

$$P_r (dBm) = 10 \times \log_{10} (P_r \times 1000)$$

$$\text{Pathloss}(dB) = 10 \log_{10} \left(\frac{P_t}{P_r} \right)$$

$$P_r = \frac{0.015625}{(4\pi)^2 \times 1600^2} \simeq 4.88 \times 10^{-13} W$$

$$P_r (dBm) = 10 \times \log_{10} (4.88 \times 10^{-13} \times 1000) = -129.95 dBm$$

$$\text{Pathloss } (dB) = 10 \log_{10} \left(\frac{1}{4.88 \times 10^{-13}} \right) = 154.79 dB$$

b) What is inferred by the channel capacity of AWGN channel?

Ans : Consider a discrete-time AWGN channel with channel input /output relationship

$$\mathbf{y[i] = x[i] + n[i],}$$

where

· x[i] is the channel input at time i,

· y[i] is the corresponding channel output,

· n[i] is a white Gaussian noise random process.

- ✓ Assume a channel bandwidth Band received signal power P.
- ✓ The received signal-to-noise ratio (SNR) – the power in x[i]divided by the power in n[i] – is constant and given by

γ = P/N0B,

· where N0/2 is the power spectral density (PSD) of the noise.

- ✓ The capacity of this channel is given by Shannon's well-known formula

$$C = B \log_2(1 + \gamma),$$

- ✓ where the capacity units are bits per second (bps).
- ✓ Shannon's coding theorem proves that a code exists that achieves data rates arbitrarily close to capacity with arbitrarily small probability of bit error.
- ✓ The converse theorem shows that any code with rate R > C has a probability of error bounded away from zero. The theorems are proved using the concept of mutual information between the channel input and output.
- ✓ For a discrete memoryless time-invariant channel with random input x and random output y, the channel's mutual information is defined as

$$I(X;Y) = \sum_{x\in\mathcal{X},\, y\in\mathcal{Y}} p(x,y)\log\left(\frac{p(x,y)}{p(x)p(y)}\right),$$

Shannon proved that channel capacity equals the mutual information of the channel maximized over all possible input distributions:

$$C = \max_{p(x)} I(X;Y) = \max_{p(x)} \sum_{x,y} p(x,y)\log\left(\frac{p(x,y)}{p(x)p(y)}\right).$$

Shannon capacity is generally used as an upper bound on the data rates that can be achieved under real system constraints.

- ✓ At the time that Shannon developed his theory of information, data rates over standard telephone lines were on the order of 100 bps.
- ✓ Thus, it was believed that Shannon capacity, which predicted speeds of roughly 30 kbps over the same telephone lines, was not a useful bound for real systems.
- ✓ Wireless channels typically exhibit flat or frequency-selective fading.

c) What is meant by time selective Fading?

In a fast fading channel, the channel impulse response changes rapidly within the symbol duration. That is, the coherence time of the channel is smaller than the symbol period of the transmitted signal. This causes Frequency dispersion (also called time selective fading) due to Doppler spreading, which leads to signal distortion. 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

$$T_S > T_C \text{ and } B_S < B_D$$

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 flat fading channel, the impulse response of the flat fading channel is a delta function (no time delay). Hence, a flat fading, fast fading channel is a channel in which amplitude of the delta function varying faster than the rate of change of the transmitted baseband signal. In case of a frequency selective, fast fading channel, the amplitudes, phases, and time delays of any one of the multipath components vary faster than the rate of change of the transmitted signal. In practice, fast fading only occurs for very low data rates.

Module III

15 a) With the help of mathematical equations show how linear convolution is converted to circular convolution in OFDM using Cyclic prefix.

Ans:

Let we have N number of samples $x[n] - x[0] \ldots x[N-1]$

$h[n]$ has a length of M and Impulse response

$$h(n) = h[0] \ldots h(M)$$

$$\boxed{X_{N-M}} \ldots \boxed{X_{N-1}} \; \boxed{X_0} \; X_1 \ldots \underbrace{\boxed{X_{N-M}} \ldots \boxed{X_{N-1}}}$$

$$\underbrace{x_{-M} \quad x_{-1} \quad x_0 \ldots \ldots x_{N-1}}_{(N+M)}$$

it have a range from $-M \leq n \leq N-1$

The cyclic prefix $x[n]$ is defined as $\{x[N-M] \ldots x[N-1]\}$;
it consists of the last M values of the $x[n]$ sequence
For each input sequence of length N, the last M samples are apparded to the begining of the sequence. Having a new sequence $\tilde{x}(n)$, $-M \leq n \leq N$ of length $N+M$
$\tilde{x}[n]$ is the i/p with impulse response $h[n]$ The o/p

$$y(n) = \tilde{x}(n) * h(n)$$

$$= \sum_{K=0}^{M} h(k)\, \tilde{x}[n-k]$$

$$= \sum_{K=0}^{M} h[k]\, x[n-k]_N$$

$$\boxed{y[n] = x[n] \circledast h[n]}$$

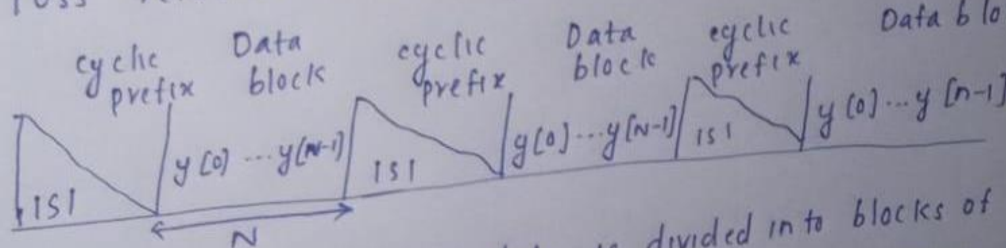
$$Y[K] = DFT\, y[n] = X[K]\, H[K]$$

$$X[K] = \frac{Y[K]}{H[K]}$$

$$x[n] = IDFT\left\{\frac{Y[K]}{H[K]}\right\}$$

$$\underbrace{y_{-N} \cdots y_{-2}}_{\text{cyclic prefix does not require back to reverse}} \quad \underbrace{y_{-1}\ y_0\ y_1 \cdots y_{N-1}}_{\text{Require to reover back}} \text{ N samples}$$

The cyclic prefix is removed the ISI is totally eliminated the cyclic prefix serves to eliminate ISI between the data blocks, because the first μ samples of the channel output affected by this ISI discarded without any loss relative to the original information sequence.



In OFDM the i/p data is divided into blocks of size N, where each block is refered to as OFDM symbol, The cyclic prefix is added to each OFDM symbol to induce circular convolution of the i/p and channel impulse response. At the receiver the o/p simples affected by ISI b/w OFDM symbols are removed. The DFT of the remaining sample are used to recover the original i/p sequence.

How Guard Bands are used in OFDM-?

Let us, Consider.

$\boxed{X_0\ X_1\ X_2\ X_3}\boxed{X_0\ X_1\ X_2\ X_3}$ $\Rightarrow$ $\boxed{X_0\ X_1\ X_2\ X_3}\boxed{X_0\ X_1\ X_2\ X_3}$

window-1 window-2 window-1 window-2

If window1 spreads due to mpc

X_0 of Second window is lost.

To over come this draw back, we use cyclic prefix.

$X_3\ X_0\ X_1\ X_2\ X_3$

PREFIX

Doing So the window will look like

$\boxed{X_3\ X_0\ X_1\ X_2\ X_3}\boxed{X_3\ X_0\ X_1\ X_2\ X_3}$

window-1 window-2

$\Rightarrow$ Now, if window1 Spreads we will lose only X_3 of window 2 which is a prefix, we still have a copy of X_3 in window2

b) Determine the average SNR per bit of BPSK modulation in Rayleigh slow fading channel in such that 90% of the times , the average probability of bit error is less than 10^-4 .

Ans:

$P_b(\gamma_b) < 10^{-4}$

$P_b(\gamma_b) = Q\sqrt{2\gamma_b} < 10^{-4}$

ie; maximum possible error is 10^{-4}

$Q(\sqrt{2\gamma_b}) \simeq 10^{-4} = 0.0001$

From Q fn table; $Q(z) = 0.0001$

ie; $Q(3.7) = 0.001$

$\Rightarrow \sqrt{2\gamma_b} = 3.7$

$2\gamma_b = 3.7^2 \quad \gamma_b = \frac{3.7^2}{2} = 6.845$

In dB ; $\gamma_b = 10\log_{10} 6.845 = 8.35$ dB

ie; minimum SNR, γ_0 should be $= 8.35$ dB

Thus, outage probability can be

$P_{out} = P(\gamma_b < \gamma_0) = 1 - 0.95 = \underline{\underline{0.05}}$

because 95% of time $P_b(\gamma_b) < 10^{-4}$

ie; no outage for 95% time

outage only for 5% of time.

Hence $P_{out} = 0.05$

$\therefore \bar{\gamma}_b = \frac{\gamma_0}{-\ln(1-P_{out})}$

Average SNR

$= \frac{6.845}{-\ln(1-0.05)} = \underline{\underline{21.4\ \text{dB}}}$

OR

16. a) How can the subcarrier fading be mitigated in multicarrier modulation system?

Ans. • Multicarrier modulation is relatively narrowband, which mitigates the effect of delay spread.

• However, each sub-channel experiences flat fading, which can cause large bit error rates on some of the sub-channels.

• The received signal-to-noise power ratio is

$$\gamma_i = \alpha_i^2 P_i / N_0 B_N$$

Where

- BN is the bandwidth of each sub-channel
- transmit power on subcarrier i is Pi
- fading on that subcarrier is αi

• If αi is small then the received SNR on the ith sub-channel is low, which can lead to a high BER on that sub-channel

• In wireless channels αi will vary over time according to a given fading distribution, resulting in performance degradation

• It is important to compensate for flat fading in the sub-channels

b) Explain the techniques employed to reduce PAPR in OFDM .

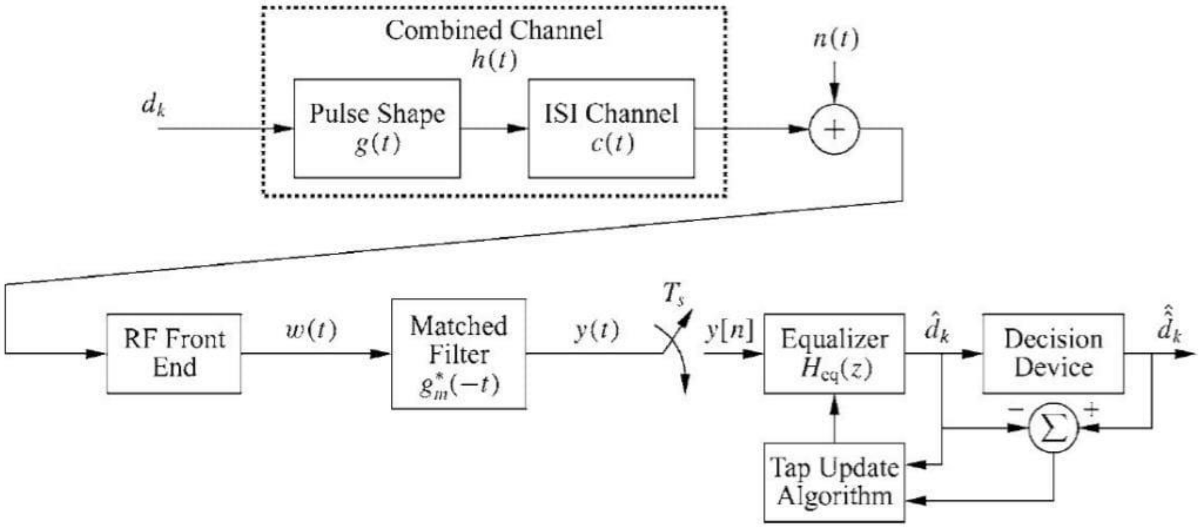
Ans: There are a number of ways to reduce or tolerate the PAR of OFDM signals

– clipping the OFDM signal above some threshold
– peak cancellation with a complementary signal
– allowing nonlinear distortion from the power amplifier (and correction for it)
– special coding techniques

Module IV

17. a) Describe the working principle of a Zero Forcing Equaliser with the help of a neat diagram.

Ans :



- The samples $\{y_n\}$ input to the equalizer can be represented based on the combined impulse response $f(t) = h(t) * g^*(-t)$ as

$$Y(z) = D(z)F(z) + N_g(z)$$

where $N_g(z)$ is the z-transform of the noise samples at the output of the matched filter $G^*_m(1/z^*)$ and

$$F(z) = H(z)G_m^*\left(\frac{1}{z^*}\right) = \sum_n f(nT_s)z^{-n}$$

- The zero-forcing equalizer removes all ISI introduced in the composite response f(t).

- To accomplish this, the equalizer impulse response should be

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

- The power spectrum N(z) of the noise samples at the equalizer output is given by

$$N(z) = N_0|G_m^*(1/z^*)|^2|H_{\mathrm{ZF}}(z)|^2 = \frac{N_0|G_m^*(1/z^*)|^2}{|F(z)|^2}$$
$$= \frac{N_0|G_m^*(1/z^*)|^2}{|H(z)|^2|G_m^*(1/z^*)|^2} = \frac{N_0}{|H(z)|^2}.$$

- If the channel H(z) is sharply attenuated at any frequency within the signal bandwidth of interest – as is common on frequency-selective fading channels – the noise power will be significantly increased.
- This requires an equalizer design that better **optimizes between ISI mitigation and noise enhancement**. One such equalizer is the **MMSE equalizer**.
- Since linear equalizers are implemented as transversal tap filters, we can find a set of coefficients $\{w_i\}$ that best approximates the zero-forcing equalizer.

b) Derive the expression for received SNR of transmitter diversity with 2 X 2 Alamouti scheme Ans:

2x2 Alamouti MIMO

2) 2x2 Alamouti STBC

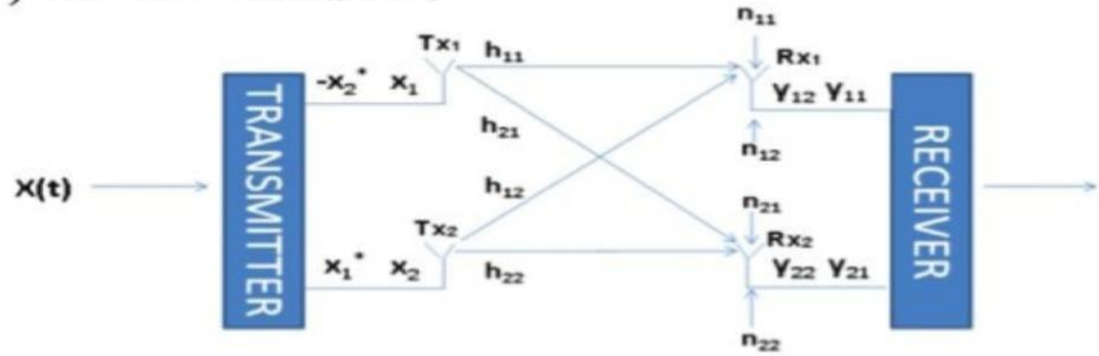


Figure 5:2x2 Alamouti STBC

The 2x2 Alamouti STBC System is shown in figure 5 which is similar to 2x1 Alamouti but we have one more receiver antenna. This system is consists of 2 Transmitter and 2 Receiver. Working principle is also similar to 2x1 Alamouti. In 2x1 Alamouti we send block codes to just one receiver but in this system we send block codes to 2 receiver.

- We assume that the channel parameters remain constant during the two time slots. Encoding is done in the same manner as discussed in Alamouti 2X1 code.

	Transmitter 1	Transmitter 2
Time t	x_1	x_2
Time t + T	$-x_2^*$	x_1^*

y_{11}=X_1h_{11}+X_2h_{12}+n_{11}(first timeslot received data in Rx1) (3.9)

y_{12}=$-X_2^*$ h_{11}+$X_1^*h_{12}$+n_{12}(second time slot received data in Rx1) (3.10)

y_{21}=X_1h_{21}+X_2h_{22}+n_{21}(first time slot received data in Rx2) (3.11)

y_{22}=$-X_2^*$ h_{21}+$X_1^*h_{22}$+n_{22}(second time slot received data in Rx2) (3.12)

Where;

X_1 and X_2transmitted data

y_{11} and y_{21} are received data (for first time slot)

y_{12} and y_{22} are received data in (for second time slot)

h_{11}, h_{12} , h_{21}, h_{22} are Rayleigh channel

n_{11} , n_{12} , n_{21}, $n_{22}n_{22}$ are AWGN noise

Now we take conjugates of y_{12} , y_{22} to get rid of conjugates of transmitted data at the receiver;

$$y_{12}^* = -x_2h_{11}^* + x_1h_{12}^* + n_{12}^* \qquad (3.13)$$

$$y_{22}^* = -x_2h_{21}^* + x_1h_{22}^* + n_{22}^* \qquad (3.14)$$

We write (3.9) ,(3.11), (3.13), (3.14) together to getting matrix form;

$$y_{11} = x_1h_{11} + x_2h_{12} + n_{11} \qquad (3.9)$$

$$y_{12}^* = -x_2h_{11}^* + x_1h_{12}^* + n_{12}^* \qquad (3.13)$$

$$y_{21} = x_1h_{21} + x_2h_{22} + n_{21} \qquad (3.11)$$

$$y_{22}^* = -x_2h_{21}^* + x_1h_{22}^* + n_{22}^* \qquad (3.14)$$

In matrix form of equations;

$$\begin{bmatrix} Y_{11} \\ Y_{12}^* \\ Y_{21} \\ Y_{22} \end{bmatrix} = \begin{bmatrix} h_{11} & h_{12} \\ h_{12}^* & -h_{11}^* \\ h_{21} & h_{22} \\ h_{22}^* & -h_{12}^* \end{bmatrix} \begin{bmatrix} x_1 \\ x_2 \end{bmatrix} + \begin{bmatrix} n_{11} \\ n_{12}^* \\ n_{21} \\ n_{22}^* \end{bmatrix} \qquad (3.15)$$

As can be seen from the (3.29) we get rid of conjugates of transmitted data.

We define X,Y,H,N matrices ;

$$X = \begin{bmatrix} x_1 \\ x_2 \end{bmatrix} \quad ,Y = \begin{bmatrix} Y_{11} \\ Y_{12}^* \\ Y_{21} \\ Y_{22} \end{bmatrix} \quad ,H = \begin{bmatrix} h_{11} & h_{12} \\ h_{12}^* & -h_{11}^* \\ h_{21} & h_{22} \\ h_{22}^* & -h_{12}^* \end{bmatrix}$$

$$,N = \begin{bmatrix} n_{11} \\ n_{12}^* \\ n_{21} \\ n_{22}^* \end{bmatrix}$$

We get;

$$Y = XH + N \qquad (3.16)$$

In order to get transmitted data at the receiver, we eliminate "H " matrix as 2x1 Alamouti STBC, however this is not easy as before. Since, 2x1 Alamouti STBCs "H" matrix is the square matrix and we eliminate multiply with its inverse easily. In 2x2 Alamouti STBCs H matrix is not square matrix so its inverse is not exists. Because of this reason we use Hermitian transposition, indicated with the symbol H. It is equivalent to transpose and do a complex conjugation of the matrix.

As a result we get transmitted data at the receiver with noise and channel effect;

$$\widetilde{X}_1 = \frac{1}{d}(h_{11}^* y_{11} + h_{12} y_{12} + h_{12}^* y_{21} + h_{22} y_{22})$$

$$\widetilde{X}_2 = \frac{1}{d}(h_{12}^* y_{11} - h_{12} y_{12} + h_{22}^* y_{21} + h_{22} y_{22})$$

18. a)Describe the steps to compute tap weights iteratively in LMS algorithm?

Steps for LMS algorithm

- The LMS algorithm also known as the stochastic gradient method, consists of the following steps:

1. Initialize the weights with values e.. These weights represent the system's parameters that will be updated iteratively to minimize the error.
2. With this value, compute an approximation for the gradient of the MSE. We are using an estimate for R and p:

$$\hat{\mathbf{R}}_n = \mathbf{u}_n^* \mathbf{u}_n^T$$

$$\hat{\mathbf{p}}_n = \mathbf{u}_n^* c_n$$

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

$$\hat{\nabla}_n = -2\hat{\mathbf{p}}_n + 2\hat{\mathbf{R}}_n \mathbf{e}_n$$

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

$$\mathbf{e}_{n+1} = \mathbf{e}_n - \mu\hat{\nabla}_n$$

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, 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 eigenvalue of the correlation matrix R

- The LMS algorithm usually converges too slowly.
- The LMS algorithm requires fewer (complex) operations

Steps of the LMS algorithm:

- **Initialization**: Initialize the filter coefficients or weights $\mathbf{w}(0)$ to some initial values. These weights represent the system's parameters that will be updated iteratively to minimize the error.
- **Input Signal and Desired Response**: At each iteration n, receive an input signal vector $\mathbf{x}(n)$ and the corresponding desired response or target output $d(n)$.
- **Compute Output**: Compute the output of the system by taking the dot product of the input signal vector and the filter coefficients:

  $y(n)=\mathbf{w}T(n)\mathbf{x}(n)$
- Where $\mathbf{w}T(n)$ denotes the transpose of the weight vector.
- **Compute Error**: Calculate the error between the desired response and the actual output: $e(n)=d(n)-y(n)$
- **Update Weights**: Adjust the filter coefficients using the LMS update rule:

- $\mathbf{w}(n+1)=\mathbf{w}(n)+\mu e(n)\mathbf{x}(n)$
- where:
  - μ is the step size or learning rate, controlling the rate of adaptation.
  - $e(n)$ is the error at iteration n.
  - $\mathbf{x}(n)$ is the input signal vector at iteration n.
- **Convergence Criteria**: Repeat steps 3-5 until convergence criteria are met. Convergence criteria can be based on reaching a predetermined number of iterations, achieving a desired level of error, or observing stability in the filter coefficients.
- **Final Output**: Once convergence is achieved, the final filter coefficients represent the adapted parameters of the system, providing an estimate that minimizes the mean squared error between the desired and actual outputs.
- These steps constitute the basic operation of the LMS algorithm. It's important to choose an appropriate step size (μ) to balance convergence speed and stability. Additionally, careful initialization and monitoring of convergence are essential for the algorithm's effectiveness in real-world applications.

b)Compare and contrast any three types of multiple access methods adopted in wireless communication system.

Ans:

Features	FDMA	TDMA	CDMA
Full Forms	FDMA is an abbreviation for Frequency Division Multiple Access.	GPRS is an abbreviation for Time Division Multiple Access.	CDMA is an abbreviation for Code Division Multiple Access.
Mode of Operation	It distributes a single bandwidth among multiple stations by dividing it into sub-channels	It only shares the time of transmission through the satellite, not the channel.	It shares both time and bandwidth among multiple stations by allocating a unique code to each slot.
Flexibility	It has a little flexible.	It has moderate flexibility.	It has high flexibility.
Codeword	It doesn't require a codeword.	It also doesn't require a codeword.	It needs a codeword.
Rate of Data	It has a low data rate.	It has a medium data rate.	It has a high data rate.
Mode of Data transfer	It uses continuous signals for data transmission.	It uses signals in bursts for data transmission.	It uses digital signals for data transmission.
Advantages	It is a very reliable, well-established, and straightforward protocol.	It is highly flexible, entirely digital, and well-established.	It is more flexible, needs less frequency planning, and offers a softer signal handover.
Disadvantages	It is very flexible, and the frequencies are limited.	It requires guard space.	It works with extremely complicated receivers, and senders/transmitters need a more complex power control method.

Synchronization	It doesn't need any synchronization.	It requires synchronization.	It also doesn't require any synchronization.
Terminals	Every terminal has its own constant frequency.	Every terminal on the same frequency is active for just a short period of time.	Every terminal may remain operational at the same time and in the same location without interruption.
Cells Capacity	It has a limited cell capacity.	It also has a limited cell capacity.	It has no capacity restriction for a channel, although it is interference-limited.
Cost	It has a high cost.	It has a low cost.	Its installation cost is high, but the operational cost is low.
Guard times and Bands	It needed guard bands.	It needed guard times.	It needed both guard times and guard bands.
Fading Mitigation	It doesn't require an equalizer.	It needed an equalizer	RAKE receiver may be possible in CDMA.

Module V

19 a)A television transmitter antenna mounted at a height of 200 meters and the receiving antenna has a height of 20 meters. What is the maximum spacing between the transmitter and receiver through tropospheric propagation? Also compute the radio horizon in this case.

Ans:

Ans) $d = 4.12\left(\sqrt{h_T} + \sqrt{h_R}\right)$ Km

$= 4.12\left(\sqrt{200} + \sqrt{20}\right)$ Km

$= 76.7$ Km

The radio horizon is given by

$d_1 = 4.12\sqrt{h_T}$ Km

$= 58.27$ Km

b)Derive expression for critical frequency, maximum usable frequency and skip distance (assume flat earth's surface) for skywave propagation.

Ans:

Critical Frequency

11.7.3. *Critical Frequency.*

The critical frequency of an ionized layer of the ionosphere is defined as *the highest frequency which can be reflected by a particular layer at vertical incidence. This highest frequency is called critical frequency for that particular layer and it is different for different layers. It is usually denoted by f_0 or f_c.* Critical frequency for the particular regular layer is proportional to the square root of the maximum electron density in the layer as shown below. From eqn. 11.40 and 11.41 we can write

$$\mu = \frac{\sin i}{\sin r} = \sqrt{1 - \frac{81 N}{f^2}} \qquad \ldots (11.43)$$

By definition, at vertical incidence

Angle of incidence $\angle i = 0$; $N = N_{max}$ and $f = f_c$.

As the angle of incidence go on decreasing and reaches to zero, (*i.e.* vertical incidence) the electron density go on increasing and reaches to maximum electron density (N_m). Then the highest frequency that can be reflected back by the ionosphere is one for which refractive index μ becomes zero.

$$\therefore \quad \mu = \frac{\sin \theta}{\sin r} = \sqrt{1 - \frac{81 N_m}{f_c^2}} = 0$$

$$\text{or} \quad 1 = \frac{81 N_m}{f_c^2} \quad \text{or } f_c = \sqrt{81 (N_m)} \qquad \ldots 11.44 \text{ (a)}$$

$$\text{or} \quad \boxed{f_c = 9\sqrt{N_m}}$$

where f_c is expressed in MHz and N_m in per cubic metre. Thus if the maximum electron density N_m is known, the critical frequency can be calculated by eqn. 11.44. Of course critical frequency is the highest frequency which can be reflected by a particular layer *at vertical incidence but it is, not the highest frequency which will get reflected for any other angle of incidence.* The frequency that can be reflected from a layer is a function of angle of incidence (i) and is called maximum usable frequency MUF (to be seen next).

Thus critical frequency gives an idea that radio waves of frequency equal to or less than the critical frequency will certainly be reflected back by the ionospheric layer irrespective of the angle of incidence. Radio waves of frequency greater than critical frequency will also be returned to earth only when the angle of incidence (i) is sufficiently glancing so that eqn. 11.42 is satisfied at the frequency involved, otherwise the wave will penetrate the layer concerned. However, it may be reflected back by a still higher layer. Thus for a wave of frequency greater than critical frequency to be reflected, the condition is

$$\sin i > \mu_m \qquad \text{from eqn. } 11.42$$

$$> \sqrt{1 - \frac{81 N_m}{f^2}} \qquad \ldots (11.45)$$

$$\sin i > \sqrt{1 - \frac{f_c^2}{f^2}}$$

$$\text{where} \quad f_c = \frac{81 N_m}{f^2} = \frac{N_m e^2}{m 4\pi^2 f^2 k_0} \qquad \ldots (11.46)$$

Maximum Usable Frequency

11.12.3. *Calculation of MUF.*

Case **I.** ***Thin Layer (or Flat Earth).*** The ionized layer may be assumed to be thin layer with sharp ionization density gradient, which gives mirror like reflection of radio waves as shown in Fig. 11.24. For the shorter distance of communication (Say upto 500 km) the earth can be assumed to be flat.

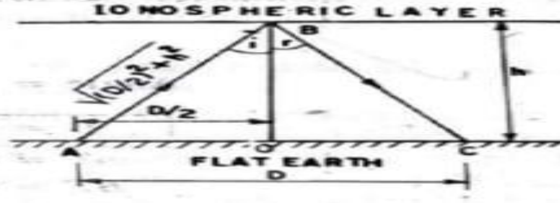


Fig. 11.24. Reflection from a thin layer on flat earth

Case II. ***Thin Layer (Curved Earth)***

(AMIETE, May 1975, Nov. 1969)

If the curvature of earth is taken into account, the reflecting region is considered to be concentric with earth as illustrated in Fig. 11.25 where transmitted wave leaves the transmitter tangentially to the earth. Let 2θ be the angle substended by the transmission distance D at the centre of the earth O.

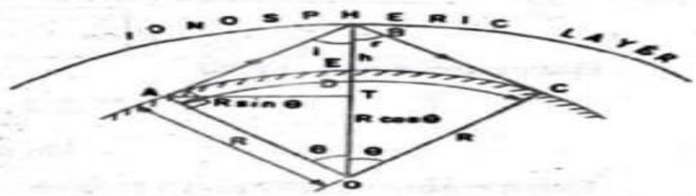


Fig. 11.25. Reflection from a thin ionospheric layer but on curved earth accouting its curvature.

$$\boxed{(f_{muf})_{max} = f_c \frac{\sqrt{\frac{D^2}{4} + \left(h + \frac{D^2}{8R}\right)^2}}{\sqrt{\left(h + \frac{D^2}{8R}\right)^2}}}$$

$$\boxed{(D_{skip})_{max} = \sqrt{8hR}}$$

As D is nothing but skip distance

$$D = 2\left(h + \frac{D^2}{8R}\right)\sqrt{\left(\frac{f_m}{f_c}\right)^2 - 1}$$

Skip Distance

11.14. SKIP DISTANCE

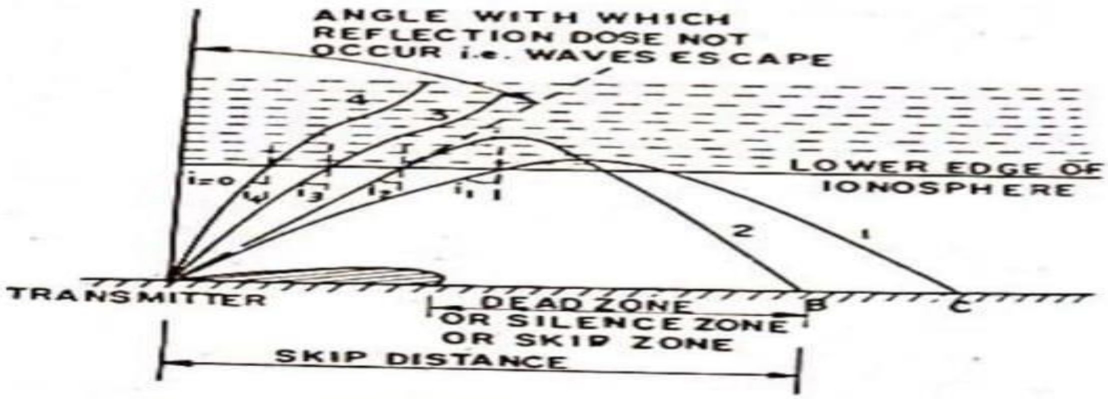


Fig. 11.27. Skip distance explantation.

11.14. SKIP DISTANCE

The higher the frequency, the higher the skip distance and for a frequency less than critical frequency of a layer skip distance is zero. As the frequency of a wave exceeds the critical frequency, the effect of the ionosphere depends upon the angle of incidence at the ionosphere as shown in Fig. 11.27 in which waves of different angle of incidence is shown.

As the angle of incidence at the ionosphere decreases, the distance from the transmitter, at which the ray returns to ground first decreases. This behaviour continues until eventually an angle of incidence is reached at which the distance becomes minimum. The minimum distance is called skip distance D (as with wave no. 2). With further decrease in angle of incidence, the wave penetrates the layer (as wave nos. 3 and 4) and does not return to earth. *Infact, skip distance is the distance skipped over by the sky wave.*

This happens because

(1) As the angle of incidence i is large (say for wave no.1), the eqn.

$$\mu = \sin i = \sqrt{1 - \frac{81N}{f^2}}$$

is satisfied with small electron density. This means μ is slightly less than unity and hence wave returns after slight penetration into the layer.

As the angle of incidence is further decreased (As in wave no. 2) sin i decrease still more and so also the μ, as N becomes comparatively more. Hence the wave penetrates still more before it reaches to earth.

Lastly when angle of incidence is small enough so that $\mu = \sin i$ can not be satisfied even by maximum electron density of the layer, then the wave penetrates (as the wave nos. 3 and 4).

The frequency which makes a given distance corresponds to the skip distance is the maximum usable frequency for those two points. If a receiver is placed with the skip distance no signals would be heard unless of course ground wave is strong enough as at A.

For a given frequency of propagation $f = f_{muf}$ the skip distance can be calculated from Eqn. 11.90 (b) in which D is the skip distance. Thus,

or $$\frac{f_{muf}}{f_c} = \sqrt{1 + \left(\frac{D}{2h}\right)^2} \quad \text{or} \quad \left(\frac{f_{muf}^2}{f_c}\right) - 1 = \left(\frac{D_{skip}}{2h}\right)^2$$

$$\boxed{D_{skip} = 2h\sqrt{\left(\frac{f_{muf}}{f_c}\right)^2 - 1}} \qquad \text{... (11.102)}$$

20 a)List out the features of the various modes of radio wave propagation.

Ans:

1. Ground Wave Propagation:

   - Occurs at lower frequencies (typically up to 2 MHz).
   - Follows the curvature of the Earth.
   - Ideal for short-range communication, typically up to a few hundred kilometers.
   - Affected by terrain, conductivity of the ground, and the Earth's curvature.

2. Sky Wave Propagation (Ionospheric Propagation):

- Utilizes the ionosphere to reflect radio waves back to Earth.
- Effective for medium and long-range communication, up to thousands of kilometers.
- Frequencies typically range from 3 MHz to 30 MHz (HF band).
- Affected by ionospheric conditions, such as solar activity and time of day.
- Subject to fading and interference due to changes in the ionosphere.

3. Line-of-Sight (LOS) Propagation:

- Requires a direct unobstructed path between transmitter and receiver.
- Used for short-range communication (typically up to 40-50 kilometers).
- Frequencies range from VHF (Very High Frequency) to microwave bands.
- Suitable for microwave links and terrestrial television broadcasting.

4. Tropospheric Scatter Propagation:

- Utilizes scattering of radio waves by irregularities in the troposphere.
- Effective for long-range communication, typically up to 800 kilometers.
- Frequencies range from UHF (Ultra High Frequency) to microwave bands.
- Less affected by terrain compared to LOS propagation but still requires clear paths.

5. Space Wave Propagation:

- Involves direct propagation of radio waves between transmitter and receiver without reflection.
- Used for communication with satellites in orbit around the Earth.
- Frequencies range from UHF to microwave bands.
- Signal quality depends on the satellite's position, antenna gain, and atmospheric conditions.

b)What is the critical frequency for reflection at vertical incidence if the maximum value of electron density is 1.24x 108 electrons/cc?

Ans:

$$\mu = \frac{\sin\theta}{\sin r} = \sqrt{1 - \frac{81 N_m}{f_c^2}} = 0$$

$$1 = \frac{81 N_m}{f_c^2} \quad \text{or} \quad f_c = \sqrt{81 N_m}$$

$$f_c = 9\sqrt{N_m}$$

Here $N_m = 1.24 \times 10^8$ electrons/cc

$$\therefore f_c = 9\sqrt{1.24 \times 10^8}$$

$$f_c = 100219.7585 \text{ Hz}$$