# QUESTION BANK

**Name of the Department** : **Computer and Science Engineeering**

**Subject Code & Name** : **EC8395 / Communication Engineering**

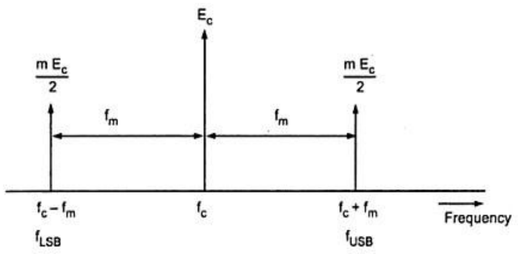
**Year & Semester** : **II / III**

## UNIT I ANALOG MODULATION

### PART-A

1. **Define amplitude Modulation and draw its spectrum. (Dec'13) (Dec'16)**

   Amplitude modulation is the process by which the amplitude of the carrier signal is varied in accordance with the instantaneous amplitude of modulating signal, but frequency and phase remains constant .It consists of carrier ($f_c$) and two sidebands at $f_c \pm f_m$.




**Spectrum of AM wave**

2. **Define image frequency.**

   Image frequency is defined as the signal frequency plus twice the intermediate frequency. This has the effect of two stations being received simultaneously and hence it is undesirable. fsi = fs + 2 fi fsi - image frequency It can be eliminated by providing adequate image signal selectivity between antenna and mixer input.

3. **Define Heterodyning**

   Heterodyning is the process of mixing two frequencies together in a nonlinear device or to translate one frequency to another using nonlinear mixing.

4. **What are the disadvantages of conventional (or) double side band full carrier system?**

   In conventional AM, carrier power constitutes two thirds or more of the total transmitted power. This is a major drawback because the carrier contains no information; the sidebands contain the same information. The conventional AM systems utilize twice as much bandwidth as needed with single sideband systems.

**5. Define Bandwidth efficiency. (Nov'17)**

Bandwidth/Spectrum efficiency is the optimized use of spectrum or bandwidth such that the maximum number of users can be provided while maintaining an acceptable quality of service (QoS).

**6. Define AM Vestigial Sideband**

AM vestigial sideband is a form of amplitude modulation in which the carrier and one complete sideband are transmitted, but only part of the second sideband is transmitted.

**7. What are the advantages of single sideband transmission?**

1. Power conservation: only one sideband is transmitted and the carrier is suppressed. So less power is required to produce essentially the same quality signal.

2. Bandwidth conservation: Single sideband transmission requires half as much bandwidth as conventional AM double side band transmission.

3. Noise reduction

**8. What are the disadvantages of single side band transmission?**

1. Complex receivers

2. Tuning Difficulties: receivers require more complex and precise

**9. Define direct frequency modulation & indirect frequency Modulation.**

In direct frequency modulation, frequency of a constant amplitude carrier signal is directly proportional to the amplitude of the modulating signal at a rate equal to the frequency of the modulating signal.

In indirect frequency modulation, the modulating signal is first used to produce a narrow band FM signal, and signal frequency multiplication is next used to increase the frequency deviation to the desired level.

**10. Define frequency deviation.**

The deviation Δf is defined as the amount by which the carrier frequency is varied from its un modulated value. The magnitude of the frequency and phase deviation is proportional to the amplitude of the modulated signal (Vm).

Maximum frequency deviation Δf can be written as $\Delta f = K_f V_m$ (HZ)

**11. State Carson's rule of FM bandwidth.**

Carson rule states that the bandwidth required to transmit an angle modulated wave as twice the sum of the peak frequency deviation and the highest modulating signal frequency. Carson's rule is $BW = 2(\Delta f + f_m (max))$ Hz.

Δf = maximum frequency deviation

fm (max)=maximum signal frequency



**12. What is the need for modulations? (Dec'13) (Dec'16)**

Modulation serves the following purposes:

1. Reduces the height of antenna.
2. Avoids mixing of signals
3. Increase range of communication
4. Allows multiplexing of signals
5. Allows adjustments in the bandwidth
6. Improves quality of reception.

**13. What are the advantages of super heterodyne receivers?**

1. The selectivity of this receiver is better since its IF amplifiers are narrowband and they operate only at IF.
2. The design of IF amplifiers is simple since they operate only at IF.
3. It eliminates the image frequency.

**14. State the advantages and disadvantages of FM over AM.**

Advantages,

i) The amplitude of FM is constant. It is independent of depth of modulation.

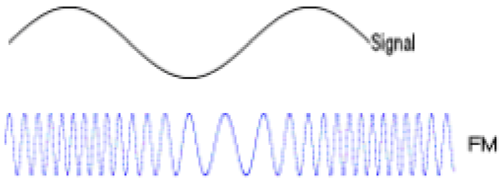
ii) Since amplitude of FM constant, the noise interference is minimum in FM. iii) the depth of modulation has limitation in AM.

Disadvantages,

1. Bandwidth requirement of FM is much higher than AM.
2. FM transmitting and receiving equipment is complex and costly.

**15. Define frequency modulation.**

Frequency modulation is defined as the process of changing the frequency of the carrier in accordance with the modulating signal. The amplitude of the modulated carrier remains constant. FM equation is given as, $FM(t) = E_c \sin[\omega_c t + m\sin\omega_m t]$




**16. For an AM DSBFC modulator with a carrier frequency of 100KHZ and maximum modulating signal frequency of 5KHZ determine upper and lower side band frequency and bandwidth.**

Upper sideband, fusb=fc+fm=100+5 = 105 kHz

Lower sideband, flsb=fc-fm=100-5 = 95 KHz

Bandwidth (B) = 2fm=10 KHz

**17. Why carrier frequencies are generally selected in HF range than low frequency range?**

The antenna size is very large at low frequencies. Such antenna is practically not possible to fabricate. High carrier frequencies require reasonable antenna size for transmission and reception.

**18. Calculate percent modulation in AM if carrier amplitude is 20 V and modulating signal is of 15 V.**

Em=15V; Ec=20V

Modulation Index: m=Em/Ec=15/20= 0.75

Percent modulation: =m*100=75%

**19. Why Armstrong method of FM is superior to reactance modulator?**

Reactance modulator is direct FM whereas Armstrong method is indirect FM. Armstrong method generates FM from PM. Hence crystal oscillators can be used in Armstrong method. Therefore frequency stability is better than reactance modulator.

**20. Differentiate between narrow band FM and wideband FM? (May '18)**

In narrow band FM the frequency deviation is very small. Hence the frequency spectrum consists of two major sidebands like AM. Other sidebands are negligible and hence they can be neglected. Therefore the bandwidth is limited only to twice of highest modulating frequency. If the deviation in carrier frequency is large enough so that other sidebands can't be neglected, then it is called as wideband FM. The bandwidth of wideband FM is calculated as per Carson's rule.

**21. Define PM. (Nov'17)**

In phase modulation the phase of the carrier varies according to amplitude variations of the modulating signal. The Pm signal can be expressed as,

$e_{PM}=E_c \sin(\omega_c t+m_p \sin \omega_m t)$

$m_p$ =modulation index for phase modulation It is given by, $m_p =\Phi_m$

$\Phi_m$ = maximum value of phase change.

**22. What is meant by indirect method of FM generation?**

In the indirect method FM is generated from PM. The phase modulated signal is represented as, $_{PM}=E_c\sin(\omega_c t+m_p\sin\omega_m t)$

The modulated frequency has deviation of $\Delta f$ with respect to $f_c$.

Here maximum deviation=$\Delta f=mf_m$

If $f_m$ remains constant then frequency deviation will be directly proportional to m. Thus as long as modulating frequency does not change phase modulation produces FM output?

**23. What is AGC?**

Automatic gain control (AGC) keeps the output signal level constant irrespective of the increase or decrease in the signal level at the input of the receiver. The AGC circuit takes part of the detected signal and derives a dc control voltage for RF, mixer and IF stages. This control voltage acts as negative feedback and controls the overall gain of these stages. The gain is varied such that output signal level is constant.

**24. Why FM signal is less susceptible to noise than an AM signal?**

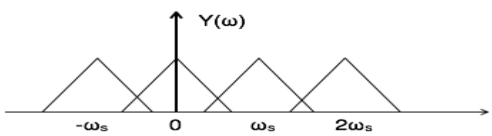
In FM the frequency of the carrier is varied as per the amplitude variations of the modulating signals. The amplitude of FM signal remains constant. The interference of external noise can be easily removed by amplitude limiter. Hence FM is less susceptible to noise.

**25. Why is VSB preferred for TV video transmission? [Nov '14]**

VSB is mainly used for TV transmission, since low frequencies near fc represent significant picture details. They are unaffected due to VSB.

**26. What is aliasing? (May '18)**

When the continuous time signal g(t) is sampled at the rate less than Nyquist rate, frequencies higher than W takes on the identity of the low frequencies in sampled signal spectrum . This is called aliasing.



**27. What are the applications of FM?**

i) Radio broadcasting
ii) Sound broadcasting in T.V.
iii) Satellite Communication
iv) Police wireless
v) Point to point Communication

**28. What will be the power in each sideband in amplitude modulated signal if power of carrier wave is 176 W and there is 60% modulation? (April '19)**

$$P_{SB} = \frac{\mu^2 P_C^2}{8R_L} = \frac{0.6^2 x 176^2}{8R_L} = 1393.92W \text{ (if } R_L = 1)$$

**29. What is Pre-emphasis and De-emphasis circuit? Where these circuits are used? (April'19)**

Pre-emphasis refers to boosting the relative amplitudes of the modulating voltage for higher audio frequencies from 2 to approximately 15 KHz. De-emphasis means attenuating those frequencies by the amount by which they are boosted. The purpose is to improve the signal-to-noise ratio for FM reception. Pre-emphasis is done at the transmitter and the de-emphasis is done in the receiver.

**30. For an AM-DSBFC wave with a peak un modulated carrier voltage Vc=10Vp, a load resistance $R_L$=1 ohm and a modulation coefficient m=1. Determine power of the carrier and the USB, LSB, total sideband power. (Nov '19)**

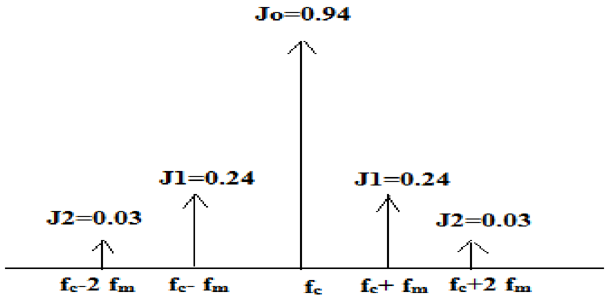
$$P_C = \frac{V_C^2}{2R_L} = \frac{10^2}{2} = 50W$$

$$P_{SB} = \frac{\mu^2 V_C^2}{8R_L} = \frac{1^2 x 10^2}{8} = 12.5W$$

$$P_T = P_C\left(1 + \frac{\mu^2}{2}\right) = 50\left(1 + \frac{1^2}{2}\right) = 75 W$$

**31. A carrier wave of amplitude 10V and frequency 100MHz is frequency modulated by a sinusoidal voltage. The modulating voltage has a amplitude of 5V and frequency fm=20KHz, frequency deviation constant is 2KHz/V. Given Jo=0.94, J1=0.24,J2=0.03. Draw the FM spectrum. (Nov'19)**

$$f_c = 100MHz$$
$$f_m = 20KHz$$
$$f_c + f_m = 100020KHz$$
$$f_c - f_m = 99980KHz$$
$$f_c + 2f_m = 100040KHz$$
$$f_c - 2f_m = 99960KHz$$



## PART B

1. Obtain AM wave equation and explain each term with the help of frequency spectrum and also obtain an expression for its power? **(May'16)**

2. i)What is the need for modulation? ii) Explain with necessary diagram any one method for generation of AM waves. **(June' 13).**
3. Draw the block diagram and explain AM super hetrodyne receiver. **(Dec'13)**

4. (i)Derive the expression for the instantaneous voltage of AM wave. **(Nov'17)**
   (ii)For an AM DSBFC transmitter with an un modulated carrier power $P_c = 100W$ that is modulated simultaneously by three modulating signals with coefficients of modulation m1 = O.2, m2= 0.4 and mB = 0.b, determine 1) Total coefficient of modulation 2) Upper and lower sideband power 3) Total transmitted power (6)

5. Compare AM, FM and PM systems and also Compare the performance of AM, DSB-SC, and SSB-SC systems in terms of BW power, frequency spectrum, phasor diagram and efficiency. **(May'15) (May '18)**

6. (i)Derive the expression for the instantaneous voltage of SSB wave. **(May'16)**

7. (i) Draw the block diagram of Armstrong indirect FM transmitter and describe its operation.
   (ii) Discuss the advantages and disadvantages of angle modulation. **(Nov'17)**

8. Explain the nature of SSB spectrum if the modulating signal is m (t) =cos2π.100t+ cos2π.2000t and carrier is given by c (t) = cos2π.10000t. **(May'15)**
   (ii) A 25 MHz carrier is modulated by a 400 Hz audio sine wave. If the carrier voltage is 4V and the maximum frequency deviation is 10 KHz and phase deviation is 25 radians .write the equation of this modulated wave for(1)FM (2)PM. If the modulating frequency is now changed to 2 KHz, all else remaining constant. Write a new equation for FM and PM. **(Nov'16)**

9. A baseband signal x (t) =5cos (2π15X 103 t) angle modulates a carrier signal A cos ϖct. Determine the modulation index and bandwidth for a FM system and PM system.**(Nov'16)**

10. (i) The efficiency η of ordinary AM wave is defined as the percentage of the total power carried by the sidebands, that is,

$$\eta = \frac{P_s}{P_t} x100\%$$

Where $P_s$ is the power carried by the sideband and $P_t$ is the total power of the AM signal.

(1) Find $\eta$ for $\mu$=0.5

(2) Show that for a single tone AM, $\eta_{max}$ is 33.3 percent at $\mu$=1. Explain the working of FM super heterodyne receiver with neat diagram. **(April '19) (Nov '19)**

**11.** (i)Discuss the method for the generation of FM using direct method
(ii) Explain the detection of FM using PLL detector. **(April '19)**

**12.** Outline the principle of slope detector and explain the operation of balanced slope detector method of FM demodulation technique. **(Nov '19)**

## UNIT II PULSE MODULATION

## PART A

**1. What is sampling process?**

The sampling process is a process of converting a continuous time signal into an equivalent discrete time signal. In sampling process an analog signal is converted into a corresponding sequence of samples that are usually spaced uniformly in time.

**2. State sampling theorem (Nov '19)**

A continuous time signal can be completely represented in its sample and received back. If the sampling frequency is twice of the highest frequency content of the signal.i.e. $f_S \geq 2f_m$

Here fs= sampling frequency

Fm=maximum frequency of the continuous signal

**3. Write the types of sampling method.**

There are three sampling methods that can be employed

(i) Ideal or instantaneous sampling

(ii)Natural sampling and (iii) Flat top sampling

**4. Define quantization error. (May '18)**

Quantization error is defined as the difference between amplitudes of quantized sample and original sample amplitude. i.e. Quantization error ($\varepsilon$)=$x_q(nT_S)$-$x(nT_S)$

where Xq ($nT_S$) = quantized value of the sample. x ($nT_S$)=original sample amplitude.

**5. What is Nyquist rate, Nyquist interval and aliasing?**

**Nyquist rate "**When the sampling rate becomes exactly equal to 2W samples per second, for a signal bandwidth of W Hertz, then it is called Nyquist Rate."

**Aliasing “**When the signals are sampled at the rate less than Nyquist (i.e. fs < 2W), then aliasing takes place. Frequencies higher than 𝕎 takes of lower frequencies in sampled spectrum. This is called aliasing. Aliasing can be reduced by sampling at a rate higher than Nyquist rate”.

**Nyquist interval “**It is the time interval between any two adjacent samples when sampling rate is Nyquist rate.”

**6. Define pulse modulation.**

In pulse modulation some parameter of a carrier pulse train is varied in accordance with the message signal, such as amplitude, time and position.

**7. What are the most four common methods of pulse modulation? (May’16)**

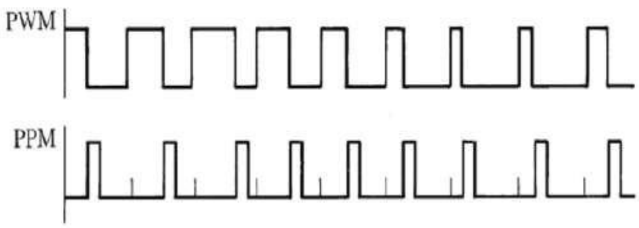
1. Pulse code modulation
2. Delta modulation
3. Adaptive delta modulation
4. Differential pulse code modulation

**8. What is the purpose of the sample and hold circuit?**

The sample and hold circuit periodically samples the analog input signal and converts those samples to a multilevel PAM signal.

**9. How is PPM obtained from PWM?**

The width of the carrier pulses varies in proportion with the amplitude of modulating signal. The amplitude and width of the pulses are kept constant but the position of each pulse is varied in accordance with the amplitude of the sampled values of the modulating signal The PWM signal is given as a triggering signal to nonstable multivibrator. The multivibrator triggers on falling edge of PWM. The monostable multivibrator generates its output pulse of fixed duration after being triggered by falling edge of PWM.




**10. Define coding efficiency. (Nov’17)**

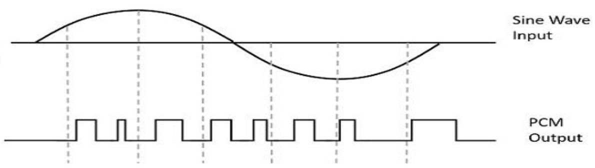
Coding efficiency is the ratio of the minimum number of bits required to achieve a certain dynamic range to the actual number of PCM bits used. Mathematically, coding efficiency is **Coding efficiency= Minimum number of bits (including sign bit) X 100 Actual number of bits (including sign bit)**

**11. Compare PAM, PWM and PPM.**

| S.No | PAM | PWM / PDM | PPM |
|---|---|---|---|
| 1. | Amplitude of the pulse is proportional to the amplitude of the modulating signal | Width of the pulse is proportional to the amplitude of the modulating signal | The relative position of the pulse is proportional to the amplitude of the modulating signal |
| 2. | The bandwidth of the transmission channel depends on width of | The bandwidth of the transmission channel depends on rise | The bandwidth of the transmission channel depends on rising time of the pulse. |
| 3. | The instantaneous Power of the transmitter varies | The instantaneous power of the transmitter varies. | The instantaneous power of the transmitter remains constant |
| 4. | Noise is interference is | Noise is interference is minimum. | Noise is interference is |
| 5. | Similar to amplitude | Similar to | Similar to phase modulation. |

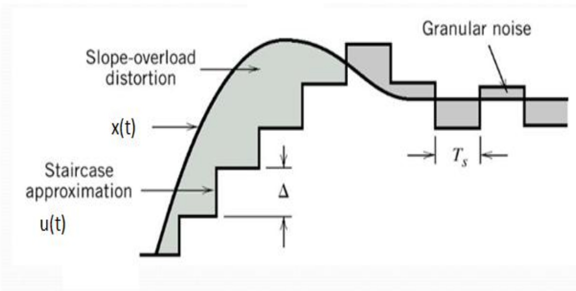
**12. Define pulse code modulation. (Nov'16)**

In pulse code modulation, analog signal is sampled and converted to fixed length, serial binary number for transmission. The binary number varies according to the amplitude of the analog signal.




**13. Define slope overload distortion.**

In delta modulation, the rate of rise/fall of input signal is very high at some time instants. This rapid change of input signal cannot be achieved by staircase signal generated by the predictor. The step size ‗δ ‗is too small for the predictor to follow the rapid changes in input signal. Hence there is large difference between the actual signal and predicted signal. The difference introduces the distortion. It is called as slope overload distortion.

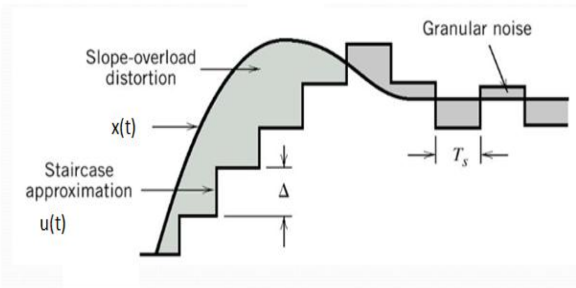



**14. How can we reduce the slope overload distortion?**

Slope overload error in DM system can be eliminated by (i) Filtering the signal to limit the maximum rate of change. (ii) Increasing step size. (iii) Increasing sample rate.(fs)

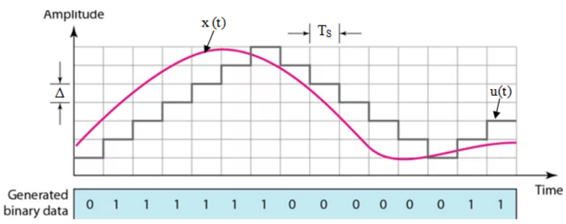
**15. Define granular noise. How it is reduced?**

When the step size is too large compared to small variations in the input signal, then granular noise occurs. When the original input signal has relatively constant amplitude, the reconstructed signal has variations that were not present in the original signal. This is called granular noise. Granular noise can be reduced by decreasing the step size.



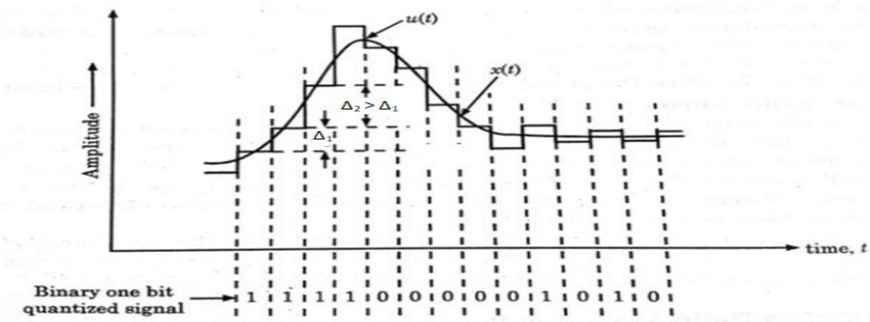

**16. What is the principle of delta modulation?**

Delta modulation transmits only one bit per sample. That is the present sample value is compared with the previous sample value and the indication, whether the amplitude is increased or decreased is sent. Input signal x(t) is approximated to step signal by the delta modulator. This step size is fixed. The difference between the input signal x(t) and staircase approximated signal confined to two levels, i.e. +δ and—δ. If the difference is positive, then approximated signal is increased by one step i.e. δ. If the difference is negative, then approximated signal is reduced by ' δ'



**17. Define adaptive delta modulation?**

Adaptive delta modulation is a delta modulation system where the step size of the AC is automatically varied depending on the amplitude characteristics of the analog input signal. The performance of delta modulator can be improved significantly by making the step size of the modulator as a time-varying form. The system which uses the above technique for reducing the quantization error is called adaptive delta modulation (ADM) system.




**18. Define companding with respect to PCM.(April '19)**

The signal is amplified at low voltage levels and attenuated at high voltages level. This is called as compression. Uniform quantization is used after compression. This is equivalent to more step size at low voltage levels and small step size at high voltage levels. That is signal is attenuated at low voltage levels and amplified at high voltage levels to get original signal. Thus the compression of signal at transmitter and expansion at receiver is called companding.

**19. List any two advantages and limitations of DPCM.**

**Advantages:**

1. DPCM requires less bandwidth compared to PCM.

2. Its signal to noise ratio is better than DM and ADM.

**Limitations:**

1. Implementation of DPCM is complex compared to PCM.

2. Slope overload distortion and quantization noise is present in DPCM. DPCM requires high sampling frequency.

**20. State the principle of working of differential pulse code modulated systems.**

The differential pulse code modulation works on the principle of prediction. The value of the present sample is predicted from the past samples. The prediction may not be exact but it is very close to the actual sample value. The difference between quantized input sample and its predicted value is obtained. This error signal is encoded and transmitted.

**21. Define SNR. (May'16)**

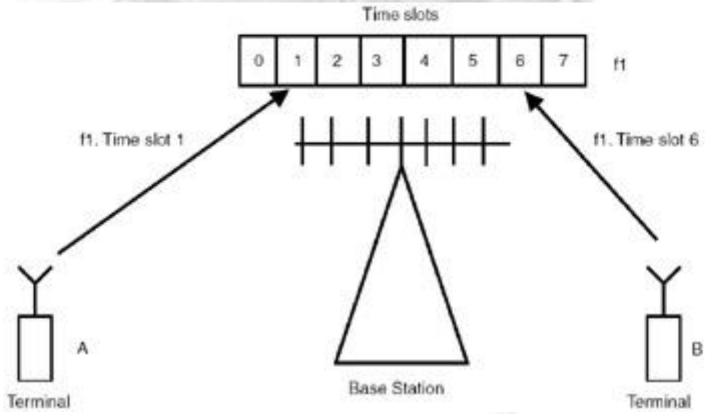
SNR, is a measure of signal strength relative to background noise. The ratio is usually measured in decibels (dB). Additionally it is found that when using AM the level of modulation has an effect. The greater the level of modulation, the higher the audio output from the receiver.

**22. Write any four primary applications of FDM.**

- It is used to public telephones and in cable TV systems.
- It is used in broad casting.
- It is used in AM and FM broadcasting.
- Communications satellites to transmit multiple channels of data on uplink and downlink radio beams

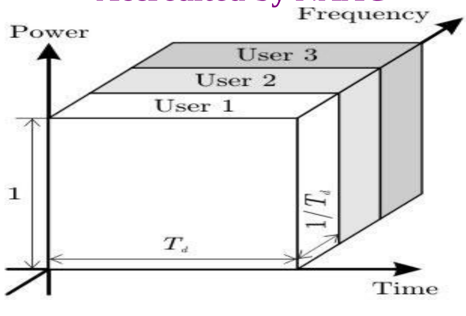
**23. What is called TDM?**

Time division multiplexing (TDM) is a communications process that transmits two or more streaming digital signals over a common channel. In TDM, incoming signals are divided into equal fixed-length time slots.




**24. What is called FDM?**

Frequency-division multiplexing (FDM) is a scheme in which numerous signals are combined for transmission on a single communications line or channel. Each signal is assigned a different frequency (subchannel) within the main channel.



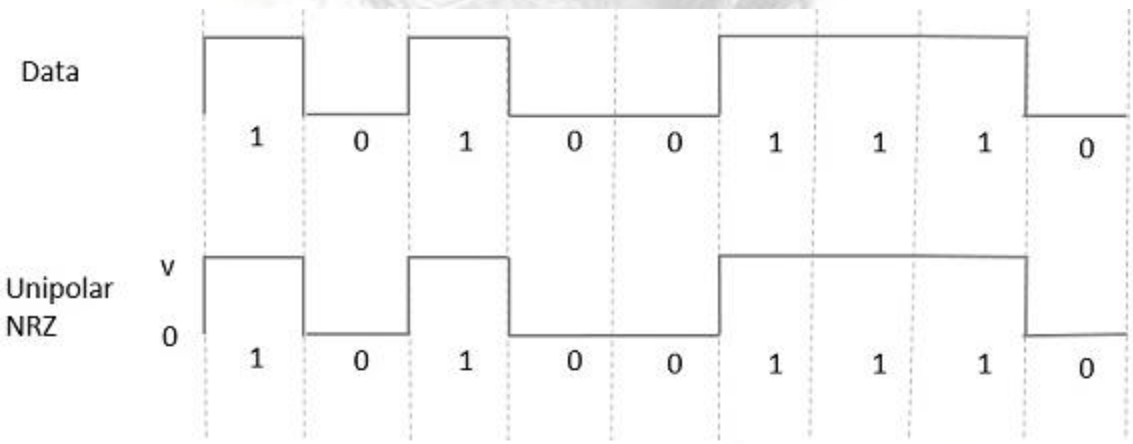

FDMA

**25. What is called vocoder?**

A vocoder is a category of voice codec that analyzes and synthesizes the human voice signal for audio data compression, multiplexing, voice encryption, voice transformation, etc., Speech coding is an application of data compression of digital audio signals containing speech. The two most important applications of speech coding are mobile telephony and voice over IP.

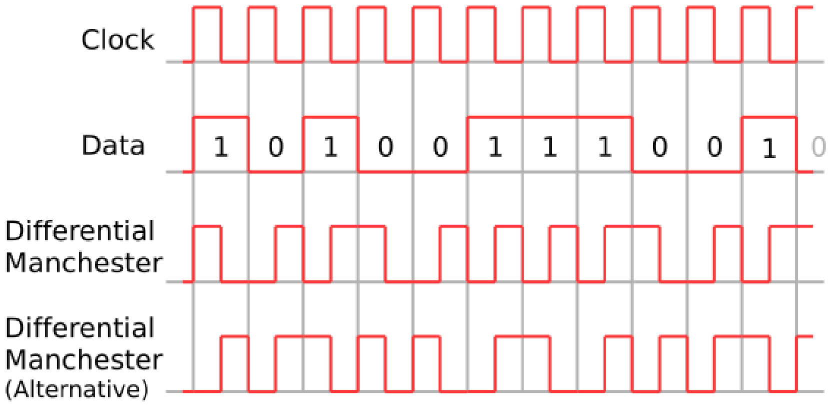
**26. What is NRZ polar format & RZ polar format?**

NRZ: Symbol 0 is represented by negative pulse and symbol 1 is represented by a positive pulse. For NRZ format, the pulse will occupy the entire symbol duration. RZ: Symbol 0 is represented by negative pulse and symbol 1 is represented by a positive pulse. For RZ format, the pulse will occupy the half the symbol duration.




**27. What is Manchester coding and write its advantages? (Dec'14)**

It is a multilevel binary code. Binary 1 is represented by +A,-A and Binary 0 is represented by –A,+A. Advantages are i) Null at dc. So, this code is more efficient than other code, ii) Due to alternate +A,-A single error can be easily detected, iii) the code is transparent.




**28. What is bit depth in PCM? (April '19)**

In digital audio using pulse-code modulation, bit depth is the number of bits of information in each sample, and it directly corresponds to the resolution of each sample. The number of possible values that can be represented by an integer bit depth can be calculated by using $2^n$, where n is the bit depth. Thus, a 16-bit system has a resolution of 65,536 ($2^{16}$) possible values.

## PART- B

1. Explain in detail about PPM modulation and demodulation. (Or) Explain how you will convert PWM to PPM with diagrams? **(Nov'17)**

2. Explain in detail about PWM modulation and demodulation.

3. Explain the procedure of PCM generation and detection with its block diagram.**(June'13),(Nov'14) (Nov'16) (Nov'17) (Nov'19)**

4. For a PCM system with the following parameters, maximum analog input frequency =4kHz maximum decoded voltage at the receiver = +- 2.25V and maximum dynamic range =46db. Determine **(May'16)** (1) Minimum sampling rate (2)minimum number of bits used in PCM codes.(3)resolution. (4)Quantization error**.**

5. Explain in detail about sampling theorem? (Or) State and prove sampling theorem.

6. With a neat sketch explain the generation of delta modulated signal. Describe quantization error in delta modulation. **(April '19) (Nov' 19)**

7. State the drawbacks of DM and suggest a method to overcome it. (Or) What is the need for Adaptive Delta Modulation and explain its features in detail. (Or) How does ADM differ from DM? Support your answer with block diagram and waveforms.

**8.** Explain the principle, generation and reconstruction of DPCM System in detail.

**9.** (i)What are the advantages, Limitations and Modifications of PCM? ii) Compare various digital Pulse modulation Methods.

**10.** Explain in detail about PAM and derive its equation for transmission bandwidth.

**11.** Explain about TDM/FDM with neat sketch. Explain its applications**(April '19)**

**12.** Explain Line coding Techniques with neat waveforms.

**13.** An analog signal having 4KHz bandwidth is sampled at 1.25 times the Nyquist rate and each sample is quantized into one of 256 equally likely levels. Assume that the successive samples are statistically independent.

(i) What is the information rate of this source?

(ii) Can the output of this source be transmitted without error over an AWGN channel with a bandwidth of 10KHz and an S/N ratio of 20 dB?

(iii) Find the S/N ratio required for error-free transmission for part(ii).

(iv) Find the bandwidth required for an AWGN channel for error-free transmission of the output of this source if the S/N ratio is 20 dB. **(April '19)**

## UNIT III DIGITAL COMMUNICATION

### PART A

**1. What are the advantages of digital transmission? (Nov'16)**

The advantage of digital transmission over analog transmission is noise immunity. Digital pulses are less susceptible than analog signals to variations caused by noise. Digital signals are better suited to processing and multiplexing than analog signals. Digital transmission systems are more noise resistant than the analog transmission systems. Digital systems are better suited to evaluate error performance.

**2. What are the elements of digital communication system?**

The elements of digital communication systems are source, source encoder, channel encoder, digital modulator, channel, digital demodulator, source decoder.

**3. What are the disadvantages of digital transmission?**

1. The required bandwidth is increased due to digital technology.
2. System Complexity is increased

3. In order to convert the analog signal to digital prior to transmission and then from digital to analog receiver, we need to use the additional encoder and decoder circuit.
4. Synchronization is necessary for digital systems (between transmission and receiver clocks)
5. Digital transmission systems are not compatible to older analog transmission systems.

**4. Mention the advantages of digital communication over analog communication**.

The effect of distortion, noise, and interference is much less in digital signals as they are less affected. Digital signal can be easily stored and manipulated. Digital circuits are easy to design and cheaper than analog circuits. The hardware implementation in digital circuits, is more flexible than analog.

**5. Define QAM.**

**QAM** (quadrature amplitude modulation) is a method of combining two amplitude-modulated (AM) signals into a single channel, thereby doubling the effective bandwidth. It is a combination of AM and PSK.

**6. What is Phase Shift Keying (PSK)? Mention its applications.**

PSK is the digital modulation technique in which the phase of the carrier signal is changed by varying the sine and cosine inputs at a particular time. PSK technique is widely used for wireless LANs, bio-metric, contactless operations, along with RFID and Bluetooth communications.

**7. What is Binary Phase Shift Keying (BPSK)?**

This is also called as 2-phase PSK or Phase Reversal Keying. In this technique, the sine wave carrier takes two phase reversals such as 0° and 180°.

**8. What is QPSK?**

QPSK stands for Quadrature Phase Shift Keying (QPSK).This is the phase shift keying technique, in which the sine wave carrier takes four phase reversals such as 0°, 90°, 180°, and 270°.If this kind of techniques are further extended, PSK can be done by eight or sixteen values also, depending upon the requirement.

**9. What is DPSK?**

In Differential Phase Shift Keying (**DPSK**) the phase of the modulated signal is shifted relative to the previous signal element. No reference signal is considered.

**10. What is called M-ary Signaling?**

An M-ary transmission is a type of digital modulation where instead of transmitting one bit at a time, two or more bits are transmitted simultaneously. This type of transmission results in reduced channel bandwidth. However, sometimes, two or more quadrature carriers are used for modulation.

**11. What is M-ary PSK?**

This is called as M-ary Phase Shift Keying (M-ary PSK).The **phase** of the carrier signal, takes on **M** different levels.

**12. What are the prominent features of M-ary PSK?**

(i)The envelope is constant with more phase possibilities.(ii)This method was used during the early days of space communication.(iii) Better performance than ASK and FSK.(iv) Minimal phase estimation error at the receiver.(v) The bandwidth efficiency of M-ary PSK decreases and the power efficiency increases with the increase in **M**.

**13. What is Inter symbol Interference (ISI) ? (Nov 2018)**

The presence of outputs due to other bits interference with the output of required bit. This effect is called inter symbol interference (ISI).Thus when a sequence of short pulses are transmitted through the system, one pulse every $T_b$ seconds, the dispersed responses originating from different symbol intervals will interfere with each other, there by resulting in ISI.

**14. What are the Causes of ISI?**

The main causes of ISI are Multi-path Propagation and Non-linear frequency in channels. ISI is usually caused by multipath propagation or the inherent linear or non-linear frequency response of a communication channel causing successive symbols to "blur" together. The presence of ISI in the system introduces errors in the decision device at the receiver output.

**15. ISI cannot be avoided. Justify the statement.(May'13)**

A communication Channel is always band limited, hence it always disperses or spreads a pulse waveform passing through it. ISI means the spreading of signal pulses and overlap with another pulses. Equalization techniques are used to combat ISI. So, signal quality is affected by noise as well as by ISI. Even if noise is absent, ISI may be present in a high speed digital communication system.

**16.What is eye pattern? State any two applications of eye pattern?(Nov'13) (Dec'12)**

**(May '15)**

When the sequence is transmitted over a baseband binary data transmission system, the output is a continuous time signal. If this signal is out at each interval (Tb) and all such pieces are placed over one another, then we obtain eye pattern. It looks like eye. Eye pattern is particularly useful in studying ISI problem. Applications: To study the intersymbol interference, to measure the additive noise, to measure the timing synchronization and jitter, non –Linearities in the channel.

**17. What is called Raised-cosine filter?**

The raised-**cosine filter** is a **filter** frequently used for pulse-shaping in digital modulation due to its ability to minimize inter symbol interference (ISI).

**18. Define duo binary encoding. Why pre coding is used? (April 2019)**

Duo binary encoding reduces the maximum frequency of the base band signal the "word duo" means to the double transmission capacity of the binary system. A precoder is used to eliminate error propagation in the receiver.

**19. Define correlative level coding.**

Correlative level coding is used to transmit a baseband signal with the signaling rate of 2Bo over the channel of bandwidth Bo. This is made physically possible by allowing ISI in the transmitted in controlled manner. This ISI is known to receiver. The correlative coding is implemented by duobinary signaling and modified duobinary signaling.

**20. Why bit reduction is needed while coding of the speech signal?**

Bit reduction is needed while coding because the channel bandwidth required for a standard PCM is 64kbps.But in certain applications the channel bandwidth is at premium, then the speech signal must be coded at low bit rates without affecting the quality of reproduction.

**21. How does pulse shaping reduce inter symbol interference?**

The shape of the pulse is selected such that the instant of detection, the interference due to all other symbol is zero. The effect of ISI totally eliminates if the signal is sampled at Tb,2Tb,3Tb… and so on.

**22. What is the necessity of equalization? (Nov '19)**

1. To reduce higher order harmonics and 2. To reduce channel distortion

**23. Define the principle of adaptive equalization.**

The filters adapt themselves to the dispersive effects of the channel that is .the coefficients of the filters are changed continuously according to the received data. The filter coefficients are changed in such a way that the distortion in the data is reduced.

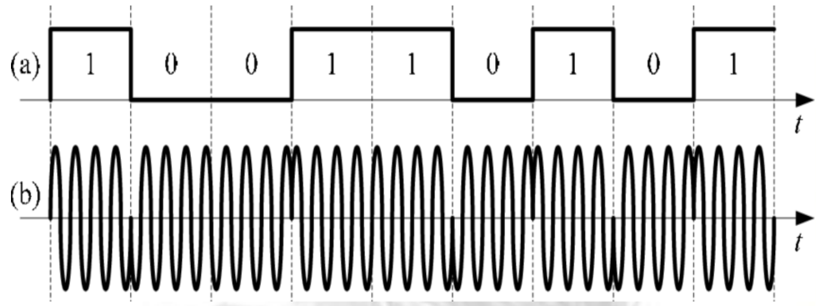
**24. What is the function of equalizing filter?(Dec'14)**

Equalising filters are used in the receiver, it cancels any residual ISI present in the received signal.

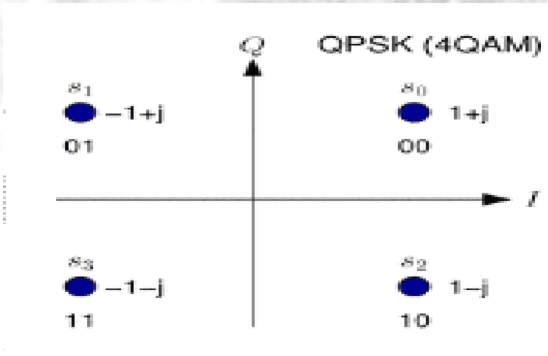
**25. What is called Pulse Shaping?**

Pulse Shaping is the process of changing the waveform of transmitted pulses. Its purpose is to make the transmitted signal better suited to its purpose or the communication channel, typically by limiting the effective bandwidth of the transmission.

**26. Consider the data bit sequence 100110101. Sketch the wave of BPSK transmitter.(Nov '19)**

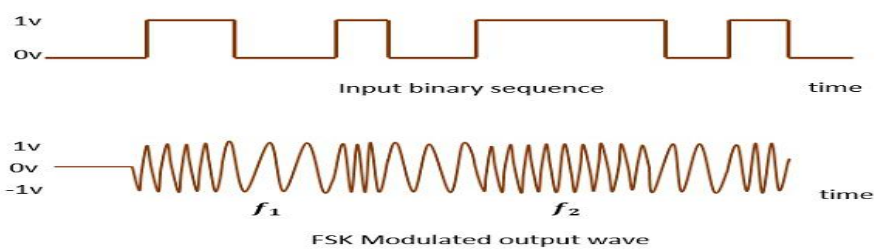



**27. Draw the constellation diagram of QPSK. (April '19)**

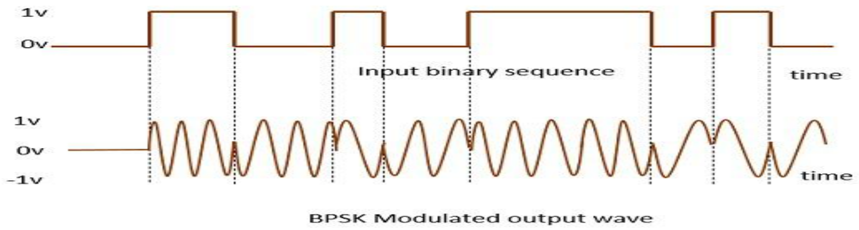



**26. Draw the modulated waveform representing PSK and FSK.(Nov '18)**

**Modulated waveform of FSK:**




**Modulated waveform of PSK:**




## PART B

1.Explain the concept of BPSK ,QAM and QPSK techniques in data communication.**(May'17)**

2.Explain QPSK modulation schemes with its constellation diagram and phasor diagram.**(Nov '18)**

3.Briefly describe the concept of QAM and draw the constellation diagram of 16 QAM. **(May'16) (Apr'2019)**

4. i) Determine the baud, minimum bandwidth and bandwidth efficiency for an 8 PSK system operating with an information bit rate of 24 kbps.
ii) Draw the block diagram of 8-QAM transmitter and explain its working.**(Nov'17)**

5.Discuss on signal design for ISI elimination. **(May'14)**

6.Explain modified duo-binary signaling scheme without & with precoder. **(Dec'12,Dec'15)**

7.Describe how eye pattern can be obtained and can be used for observing the characteristics of a communication channel. **(Dec'14, Dec'15)**

8.Illustrate the modes of operation of an adaptive equalizer with a neat block diagram. **(Dec'15)**

9.Explain the duobinary signalling technique in detail. Draw the frequency response and impulse of duobinary scheme. How modified duobinary scheme overcomes the basic method? **(Nov'2019)**

10.Write short notes on (1) Pulse shaping (2) Correlative coding.

11.Explain about Cosine filters.

12.Compare various digital modulation schemes.

13,Define DPSK**.** Draw the waveform representing DPSK. With neat diagram, explain the generation and detection of DPSK with neat block diagram.(**Nov'18)**

**14.** Explain coherent detection of BFSK signal and derive the expression for probability of error. **(Apr '19)**

**15.** Explain the expression of error probability of error in BPSK. **(Apr '19)**



## UNIT IV INFORMATION THEORY & CODING

### PART -A

**1. Give the factors which influence reliable transmission?**

Transmitted signal power, channel bandwidth.

**2. What is the aim of error control coding? List the error control mechanism.(Nov 2018)**

Reduces the required transmitted power, Reduces the size of antennas, Reduces the hardware cost. The different error control codes are block codes , cyclic codes, convolutional codes.

**3. What are the disadvantages of error control coding?**

Increases the transmission bandwidth, Increases the complexity of decoder.

**4. Give the types of error control codes. (June'13)**

Block codes , Convolutional codes.

**5. List the types of block codes.**

Linear block codes, Cyclic codes.

**6. Define block codes.**

The codes which consists of (n-k) parity bits for every k bit message block are known as block codes. E.g cyclic codes, linear block codes.

**7. Define linear block codes .(May'16)**

Block code is the code in which every 'k-bit' message block (n-k) parity bits are appended to produce 'n' bit code word. If the parity bits are the linear combination of 'k' message bits then the code is referred as linear block codes.

**8. What are the systematic codes?**

Block codes in which the message bits are transmitted in unaltered form are called systematic codes. An (n, k) block code in which every codeword can be separated into k information symbols and (n – k) check symbols. The information symbols are identical with those of the source message before encoding. Thus the process of encoding a systematic code involves the insertion of (n – k) check symbols into (i.e. among, before, or, most usually, after) the information symbols.

**9. Define generator matrix.**

Generator matrix Gkxn is used in the encoding operation and its k rows are linearly independent the encoding operation and its k rows are linearly independent

$G_{kxn}=[P_{kx(n-k)} \mid I\ kxk]$ ; Where, P-parity matrix, I-identity matrix

**10. Explain Shannon-Fano coding. (June'13)**

An efficient code can be obtained by the following simple procedure, known as Shannon-Fano algorithm. List the source symbols in order of decreasing probability. Partition the set into two sets that are as close to equiprobable as possible, and sign 0 to the upper set and 1 to the lower set. Continue this process, each time partitioning the sets with as nearly equal probabilities as possible until further partitioning is not possible.

**11. Define information rate.**

If the time rate at which source X emits symbols is r symbols per second. The information rate R of the source is given by R=r H(X) bits/second, H(X)- entropy of the source.

**12. What is data compaction?**

For efficient signal transmission the redundant information must be removed from the signal prior to transmission .This information with no loss of information is ordinarily performed on a signal in digital form and is referred to as data compaction or lossless data compression.

**13. Define mutual information and channel capacity.**

Mutual information I(X,Y) of a channel is defined by I(X,Y)=H(X)-H(X/Y) bits/symbol H(X)- entropy of the source, H(X/Y)- conditional entropy of Y.

The **channel capacity**, C, is **defined** to be the maximum rate at which information can be transmitted through a channel.

**14. Define entropy.(Nov 2018)**

Entropy is the measure of the average information content per second.

$$\text{Entropy } H(S) = \sum_{k=1}^{k} p_k \log_2\left(\frac{1}{p_k}\right)$$

**15. Define syndrome.**

Syndrome contains information about the error pattern 'e ' and may therefore be used for error detection . S is a x (n-k0 vector and is used to decode the vector C from the received vector 'r '

$S = r\,H^T$ where $r = C + e$ .

**16. Give the properties of syndrome.**

The syndrome depends only on the error pattern and not on the transmitted code word. All error patterns that differ by a codeword have the same syndrome.

**17. Define: Cyclic codes and its properties.(May'16)**

A code is said to be cyclic if every cyclic shift of the codeword produces some other valid codeword. its properties are

1. The nonzero code polynomial of minimum degree in a linear block code is unique
2. A binary polynomial of degree n − 1 or less is a code polynomial if and only if it is a multiple of g(X).
3. The degree of the generator polynomial of an (n, k) binary cyclic code is n − k.
4. The generator polynomial of an (n, k) binary cyclic code is a factor of Xn + 1.
5. If g(X) is a polynomial of degree n − k and is a factor of Xn + 1, then g(X) generates an (n, k) cyclic code.

**18.Give the graphical representation of convolutional encoder?**

Code tree, Trellis, State diagram

**19. An event has six possible outcomes with probabilities {1/2, 1/4,1/8,1/16,1/32,1/32}. Find the entropy of the system. (Nov' 17)**

$$H = -\sum_{k=1}^{6} P_k . \log_2 P_k$$

$= (-3.32[1/2.\log_{10}(1/2)+1/4.\log_{10}(1/4)+1/8.\log_{10}(1/8)+1/16.\log_{10}(1/16)$
$+1/32.\log_{10}(1/32)+1/32.\log_{10}(1/32)])$
H=1.936 bits/symbol

**20. What is the need for convolution coding?**

**Convolutional codes** are widely used as channel codes in practical communication systems for error correction. The encoded bits depend on the current k input bits and a few past input bits. The main decoding strategy for convolutional codes is based on the widely used Viterbi algorithm.

**21. What is trellis?**

Trellis is a tree like structure with remerging branches. The code branch with an input 'o' is drawn by a solid line and a branch by an input '1' is drawn as a dashed line. each input sequence corresponds to a specific path through the trellis. trellis contains (l+k) levels where l - length the message and k-constraint length level, j- is the depth of the trellis

**22. What is code tree ?**

Each branch of the tree responds an input symbol with the corresponding pair of input binary symbols indicated on the branch the input 'o' specifies the upper branch of the tree and the input '1' specifies the lower branch of the tree. a specific path is traced from left to right in accordance with the input sequence .the corresponding coded symbols on the branches of that path constitute the output sequence.

**23.What is the advantage of Sequential decoding?(Nov'17)**

**Sequential decoding** is a limited memory technique for decoding tree codes. Sequential decoding is mainly used as an approximate decoding algorithm for long constraint-length convolutional codes. This approach can save a substantial amount of computer memory.

**24. Distinguish block codes and convolution codes?**

| Block codes | Convolution codes |
|---|---|
| 1) The information bits are followed by the parity bits. | 1 )The information bits are spread along the sequence. |
| 2) Needs the buffer to store msg block. | 2) does not need the buffer since the bits are arriving in serial fashion. |

**25.What is prefix code?Give example.(Nov 2018)**

A prefix code is a type of code system distinguished by its possession of the "prefix property", which requires that there is no whole code word in the system that is a prefix (initial segment) of any other code word in the system. Prefix codes are also known as prefix-free codes, prefix condition codes and instantaneous codes.

For example, a code with code words {9, 55} has the prefix property; a code consisting of {9, 5, 59, 55} does not, because "5" is a prefix of "59" and also of "55".

**26.What is Viterbi decoding?(Nov 2018)**

A Viterbi decoder uses the Viterbi algorithm for decoding a bit stream that has been encoded using convolutional code or trellis code.
The Viterbi algorithm is the most resource-consuming, but it does the maximum likelihood decoding. It is most often used for decoding convolutional codes with constraint lengths $k \le 3$, but values up to $k=15$ are used in practice.

**27.What is the need of channel coding? (May/June 19)**

Channel coding is often used in digital communication systems to protect the digital information from noise and interference and reduce the number of bit errors. Channel coding is mostly accomplished by selectively introducing redundant bits into the transmitted information stream.

**28.List the properties of cyclic code.(May/June 19)**

The nonzero code polynomial of minimum degree in a linear block code is unique.
A binary polynomial of degree $n - 1$ or less is a code polynomial if and only if it is a multiple of $g(X)$.

The degree of the generator polynomial of an (n, k) binary cyclic code is n − k.



## PART –B

1. Five symbols of the alphabet of DMS and their probabilities are given below. S={So,S1,S2,S3,S4} P(S) = {0.1, 0.1, 0.2, 0.2, 0.4}.Code the symbols using Huffman coding. Find the efficiency of the code. **(Nov'14) .(May'16) (Nov'16)**

2. Find the Shannon –fanno code for the following seven messages with probabilities indicated. S={S1,S2,S3,S4,S5, S6,S7},P(S)= {0.05,0.15,0.2,0.05,0.15,0.3,0.1}.**(Nov'15)**

3. Construct a convolution encoder whose constraint length is 3 and has 3 modulo- 2 adders and an output multiplexer. The generator sequences of the encoder are $g^{(1)}$=(1,0,1),$g^{(2)}$=(1,1,0),$g^{(3)}$=(1,1,1). Draw the block diagram of the encoder. Find the encoder output produced by the message sequence 10111.

4.(i)Derive the expression for mutual information and channel capacity. (7) **(Nov'17)**

(ii) What are the types of error control coding ? Describe the working of viterbi decoding algorithm.

5.Write in detail the procedure of Shannon-fano coding scheme **.(May'15)**

6.Define entropy. Explain the properties of entropy.

7.Five source messages are probable to appear as mL =0.4, m2= 0.15, m3 = 0.15,m4 = 0.15, and m5 = 0.15. Determine the coding efficiency for 1) Shannon-Fano coding 2) Huffman coding **(Nov'17)**

8.Write short notes on (1)Linear Block Codes(2)Viterbi Algorithm **(Nov'14)**

9.The generator polynomial of a (7, 4) cyclic code is given by G(D) =1 + D + $D^2$.compute all the non-systematic codewords. **(May'16)**

10.Consider a systematic block code whose parity check equation
are P1 = M1 + M2 + M4

P2 = M1 + M3 + M4

P3 = M1 + M2 + M3

P4 = M2 + M3 + M4

(a)Find the generator matrix and the parity check matrix. (b)How many errors can be detected and corrected ? if the received code word is 10101010,find the syndrome **(May'15)**

11.Explain the concept of code generation and decoding of correlation codes. **(Nov'16)**

12.A source produces three symbols A,B and C with probabilities P(A) = ½, P(B) = ¼, P(C) =1/4. Determine the source entropy.

13.Find the entropy of a binary memory less source and find when it is maximum.**(Nov 2018)**

14.(i) Explain Shannon's Channel Capacity theorem.**(Nov/Dec 2019)**

(ii)A transmission channel has a bandwidth of 4 KHz and signal to noise power ratio of 31. a) How much should the bandwidth be in order to have the same channel capacity, if S/N ratio is reduced to 15? b)What will be the signal to noise power ratio required if the bandwidth is reduced to 3KHz.

15. (i) Consider a binary memory less source N with two symbols x, and x; . Show that If(X) is maximum when both zl and .v; are equiprobable.

(ii) A discrete memory less source X has four symbols xl , z 2 , x', and x4 with P( z1 ) = 0.5 , P(x 2 ) = 0.25 and P(x3 ) = P (z 4 ) = 0.125 . Construct a Shanon- Fanno code for X; show that this code has the optimum property that ni — /(i;) and that the code efficiency is 100 percent. **(May/June 19)**

## UNIT V SPREAD SPECTRUM & MULTIPLE ACCESS

## PART A

**1. Define spread spectrum**

Spread-spectrum techniques are methods by which a signal (e.g. an electrical, electromagnetic, or acoustic signal) generated in a particular bandwidth is deliberately spread in the frequency domain, resulting in a signal with a wider bandwidth. These techniques are used for establishment of secure communications, increasing resistance to natural interference and jamming, to prevent detection, and to limit power flux density (e.g. in satellite downlinks).

**2. Mention the Spread Spectrum System Criteria.**

The following two criteria must be satisfied :( i)The transmitted signal must occupy a bandwidth much greater than the bandwidth of the modulating signal (i.e. the input signal to the system).

(ii) The bandwidth occupied by the transmitted signal must be determined by a prescribed waveform and not by the modulating frequency (i.e. carrier frequency)

**3. Give the reasons for use of spread spectrum systems.**

There are three major reasons for the use of spread spectrum techniques in communication systems today.(i) They aid privacy of the transmission, since the spectral density of the spread spectrum may be less than the noise spectral density of the receiver.

(ii)The despreading process in the receiver will spread the spectra of unwanted narrowband signals, thus improving interference rejection.

(iii)The effect on a spread spectrum receiver, that receives a spread spectrum from a different spread spectrum system using the same frequency bands but implementing a different spreading pattern, approximates to noise in the receiver.

**4. Give the necessary bandwidth equation for error-free transmission information at very low SNR.**

$$B_w \approx \frac{NC}{S}$$

where C is the capacity of a communication channel in bits per hertz, $B_w$ is the bandwidth in hertz, S is the signal power, and N is the noise power.

**5. What are the three ways to spread the bandwidth of the signal?**

(i)Direct sequence. The digital data is directly coded at a much higher frequency. The code is generated pseudo-randomly, the receiver knows how to generate the same code, and correlates the received signal with that code to extract the data.

(ii)Frequency hopping. The signal is rapidly switched between different frequencies within the hopping bandwidth pseudo-randomly, and the receiver knows beforehand where to find the signal at any given time.

(iii)Time hopping. The signal is transmitted in short bursts pseudo-randomly, and the receiver knows beforehand when to expect the burst.

**6. Define DSSS.**

Direct-sequence spread spectrum (DSSS) is a spread spectrum modulation technique in which the transmitted signal takes up more bandwidth than the information signal that is being modulated. The name 'spread spectrum' comes from the fact that the carrier signals occur over the full bandwidth (spectrum) of a device's transmitting frequency.

**7. How signals are transmitted in DSSS?**

Direct-sequence spread-spectrum transmissions multiply the data being transmitted by a "noise" signal. This noise signal is a pseudorandom sequence of 1 and −1 values, at a frequency much higher than that of the original signal, thereby spreading the energy of the original signal into a much wider band.

If an undesired transmitter transmits on the same channel but with a different PN sequence (or no sequence at all), the de-spreading process results in no processing gain for that signal. This effect is the basis for the code division multiple access (CDMA) property of DSSS, which allows multiple transmitters to share the same channel within the limits of the cross-correlation properties of their PN sequences.

**8. Give the feature of DSSS.**

(i)DSSS phase-modulates a sine wave pseudo-randomly with a continuous string of pseudo-noise (PN) code symbols called " chips", each of which has a much shorter duration than an information bit. That is, each information bit is modulated by a sequence of much faster chips. Therefore, the chip rate is much higher than the information signal bit rate.

(ii)DSSS uses a signal structure in which the sequence of chips produced by the transmitter is known *a priori* by the receiver. The receiver can then use the same PN sequence to counteract the effect of the PN sequence on the received signal in order to reconstruct the information signal.

**9. What are the benefits of DSSS?**

Resistance to intended or unintended jamming, Sharing of a single channel among multiple user, Reduced signal/background-noise level hampers interception (stealth) & Determination of relative timing between transmitter and receiver.

**10. Define the term pseudo random noise.**

A Pseudo-random Noise (PN) sequence is a sequence of binary numbers, e.g. ±1, which appears to be random; but is in fact perfectly deterministic. The sequence appears to be random in the sense that the binary values and groups or runs of the same binary value occur in the sequence.

Linear feedback shift register (LFSR). LFSRs are one of the simplest ways to generate pseudorandom sequences. In an LFSR, any bit is determined by a linear combination of the previous n bits, for a suitable choice of n. In particular, we may have an LFSR in which

$$B_n = A_0B_0 \oplus A_1B_1 \oplus A_2B_2 \oplus \ldots \oplus A_{n-1}B_{n-1}$$

**11. Define FHSS.**

Frequency-hopping spread spectrum (FHSS) is a method of transmitting radio signals by rapidly switching a carrier among many frequency channels, using a pseudorandom sequence known to both transmitter and receiver. It is utilized as a multiple access method in the frequency-hopping code division multiple access (FH-CDMA) scheme.

**12.Give the advantages of FHSS**

A spread-spectrum transmission offers three main advantages over a fixed-frequency transmission:

1.Spread-spectrum signals are highly resistant to narrowband interference. The process of recollecting a spread signal spreads out the interfering signal, causing it to recede into the background.

2.Spread-spectrum signals are difficult to intercept. An FHSS signal simply appears as an increase in the background noise to a narrowband receiver. An eavesdropper would only be able to intercept the transmission if the pseudorandom sequence was known.

3.Spread-spectrum transmissions can share a frequency band with many types of conventional transmissions with minimal interference. The spread-spectrum signals add minimal noise to the narrow-frequency communications, and vice versa. As a result, bandwidth can be utilized more efficiently.

**13.Give the applications of FHSS**

Spread-spectrum signals are highly resistant to deliberate jamming, unless the adversary has knowledge of the spreading characteristics. Military radios use cryptographic techniques to generate the channel sequence under the control of a secret Transmission Security Key (TRANSEC) that the sender and receiver share.

By itself, frequency hopping provides only limited protection against eavesdropping and jamming. To get around this weakness most modern military frequency hopping radios often employ separate encryption devices such as the KY-57. U.S. military radios that use frequency hopping include HAVE QUICK and SINCGARS.

**14.Give the advantages of Spread Spectrum Techniques.**

1. Spread spectrum signals are highly resistant to the jamming.
2. Many users can share a signal band with no interference.
3. Unauthorized listening is prevented.
4. Unintentional interference occupying the same band is greatly minimized and in most cases virtually eliminated.
5. Resistant to fading.
6. Superior method for radar.

**15.Give the disadvantages of Spread Spectrum Techniques.**

1.Complex circuitry

2.expensive to develop

3.Very large bandwidth

4.Easily jammed and hence is not generally used in its true form.

**16. What are the advantages of Spread spectrum over a fixed-frequency transmission?**

(i)Spread-spectrum signals are highly resistant to narrowband interference. The process of recollecting a spread signal spreads out the interfering signal, causing it to recede into the background.

(ii)Spread-spectrum signals are difficult to intercept. A spread-spectrum signal may simply appear as an increase in the background noise to a narrowband receiver. An eavesdropper may have difficulty intercepting a transmission in real time if the pseudorandom sequence is not known.

(iii)Spread-spectrum transmissions can share a frequency band with many types of conventional transmissions with minimal interference.

**17. Mention the advantages of CDMA system. (Nov 2010).**

- Each user employs a unique spread spectrum signaling code. It provides communication privacy between users with different spreading signals.
- One of the main advantages of CDMA is that dropouts occur only when the phone is at least twice as far from the base station. Thus, it is used in the rural areas where GSM cannot cover.
- Another advantage is its capacity; it has a very high spectral capacity that it can accommodate more users per MHz of bandwidth.

**18. What are the benefits of multiple access techniques in Communication Engineering?**

Multiple access techniques are used to allow a large number of mobile users to share the allocated spectrum in the most efficient manner. As the spectrum is limited, so the sharing is required increase the capacity of cell or over a geographical area by allowing the available bandwidth to be used at the same time by different users. And this must be done in a way such that the quality of service doesn't degrade within the existing users.

**19. What is the difference between Multiple Access & Multiplexing?**

Multiplexing is the process of transmitting several messages simultaneously on the same circuit or channel. On the other hand Multiple Access is techniques that have been developed in the satellite industry which allow satellite spectrum and power to be shared efficiently among multiple users. In multiple accesses More than one simple signal can thus be transmitted as part of a single complex signal and separated out at the receiving end.

**20. Briefly explain the multiple access techniques used in Communication Engineering.(May 2017)**

Frequency Division Multiple Access (FDMA):FDMA channel-access scheme is based on the frequency-division multiplexing(FDM) scheme, which provides different frequency bands to different data-streams. Time division multiple access (TDMA):TDMA channel access scheme is based on the time-division multiplexing (TDM) scheme, which provides different time-slots to different data-streams (in the TDMA case to different transmitters) in a cyclically repetitive frame structure. Code division multiple access (CDMA) / Spread spectrum multiple access (SSMA) CDMA scheme is based on spread spectrum, meaning that a wider radio spectrum in Hertz is used than the data rate of each of the transferred bit streams, and several message signals are transferred simultaneously over the same carrier frequency, utilizing different spreading codes. Space division multiple access (SDMA):SDMA transmits different information in different physical areas.

**21.What are the features of TDMA?**

In TDMA a single carrier frequency with a wide bandwidth is shared among multiple users. Each user is assigned non-overlapping time slot.

**22.What is near far problem in CDMA.(May'16)**

If received signals from mobile units do not have equal power at the base station, in such a situation the strongest received signal from a mobile user captures the demodulation process at the base station to the detriment of other users.

**23.What is SDMA? (June '14)**

By using narrow beam antennas area on the earth covered by satellite can be divided into smaller segments. Earth station in each segment may actually use the same frequency but because of very narrow beam widths of antennas, there is no interference between adjacent segments. This is spatial division multiple access.

**24.What is CDMA? (May'17)**

CDMA scheme is based on spread spectrum, meaning that a wider radio spectrum in Hertz is used than the data rate of each of the transferred bit streams, and several message signals are transferred simultaneously over the same carrier frequency, utilizing different spreading codes.

**25.What is FDMA? (Nov'12)**

**FDMA** (frequency division multiple access) is the division of the frequency band allocated for wireless cellular telephone communication into 30 channels, each of which can carry a voice conversation.

**26.What are the benefits of Multiple Access Techniques.(May/June 19)**

The basic advantage of this type of multiple access is that it allows all users to coexist and use the entire bandwidth at the same time. Since each user has different code, there won't be any interference. In this technique, a number of stations can have number of channels unlike FDMA and TDMA.

**27.Define Near Far Problem in CDMA(May/June 19)**

The near–far problem or hearability problem is the effect of a strong signal from a near signal source in making it hard for a receiver to hear a weaker signal from a further source due to adjacent-channel interference, co-channel interference, distortion, capture effect, dynamic range limitation

**28.What do you mean by Jamming Margin (NOV/DEC 19)**

The level of interference (jamming) that a system is able to accept and still maintain a specified level of performance, such as maintain a specified bit-error ratio even though the signal-to-noise ratio is decreasing.

**29.List the advantages of Spread spectrum Modulation (NOV/DEC 19)**

- **Cross**-talk elimination.
- Better output with data integrity.
- Reduced effect of multipath fading.
- Better **security**.
- Reduction in noise.
- Co-existence with other systems.
- Longer operative distances.
- Hard to detect

## PART –B

1. Discuss the various multiple access techniques used in wireless communication with their merits and demerits.**(May'17)(Nov 2018)**

2. Explain with a neat block diagram the SDMA technique and Discuss its applications in wired and wireless communication. **(May'17)**

3. (i)Explain the principle of FDMA with diagram. **(May'17, 18)** (ii) Discuss the TDMA technique in detail and compare it with FDMA. **(Nov'16)**

4. Discuss in detail the concept of TDMA and SDMA and their applications in wire and wireless communications. **(Nov'16)**

5. Explain CDMA system and its application in wireless (or) Draw the block diagram of CDMA encoder and decoder and briefly explain its working. **(May'16)(Nov'17)**

6. What is the maximum bandwidth allocated to each user? What is the bit rate employed by each user? How long does it take to transmit a packet? What is CDMA? Mention its merits and demerits.

7. Draw the typical TDMA system. Explain the operation with the time pattern. **(Nov'14)**

8. Explain the operation of FH-SS. Compare slow and fast FH-SS. **(Nov'16)**

9. Compare in detail about TDMA,FDMA,SDMA and CDMA and mention its uses.

10. How Pseudo noise is generated? Explain in detail about PN sequence Generator with neat sketch.

11.(i)What are PN sequences? What are the properties of PN sequences?**(Nov 2018)**
(ii)What are the differences between FHSS and DSSS?

(iii) What are the advantages of spread spectrum?

15.Explain the operation of FH-SS. Compare slow and fast FH-SS.(May/June 19)

16 Discuss the FDMA and TDMA techniques used in wireless communication with their merits and demerits.( May/June 19)

17.Explain the generation of PN sequence and prove its properties. (NOV / DEC 19)

18.With a neat Sketch explain CDMA technique. (NOV/DEC 19)