UNIT-I
DIGITAL COMMUNICATION SYSTEM
Two marks

1. What are the advantages of digital transmission?

- 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 disadvantages of digital transmission?

- The transmission of digitally encoded analog signals requires significantly more bandwidth than simply transmitting the original analog signal.
- Analog signal must be converted to digital codes prior to transmission and converted back to analog form at the receiver, thus necessitating additional encoding and decoding circuitry.

3. State sampling theorem.

If a finite –energy signal g(t) contains no frequencies higher than W hertz ,it is completely determined by specifying its co=ordinates at a sequence of points spaced 1/2W seconds apart.

If a finite energy signal g(t) contains no frequencies higher than W hertz, it may be completely recovered from its co=ordinates at a sequence of points
spaced 1/2W seconds apart.

4. Define quadrature sampling.

Quadrature sampling is used for uniform sampling of band pass signals.The in-phase component gI(t) and the quadrature component gQ(t) may be respectively and then suppressing the sum-frequency components by means of appropriate low pass filter. Under the assumption that fc>W, we find that gI(t)&gQ(t) are both low-pass signals limited to -W<f<W. Accordingly each component may be sampled at the rate of 2W samples per second. This type of sampling is called quadrature sampling.

5. What is aliasing?

The phenomenon of a high-frequency in the spectrum of the original signal g(t) seemingly taking on the identity of a lower frequency in the spectrum of the sampled signal g(t) is called aliasing or fold over.

6. What is the Nyquist sampling rate?

Nyquist sampling rate states that, the minimum sampling rate is equal to twice the highest audio input frequency.

7. Define pulse code modulation.

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. This sample variable amplitude pulse is digitized by the analog to digital converter.

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. Define overload distortion.

If the magnitude of sample exceeds the highest quantization interval, overload distortion occurs.

10. Define quantization. (NOV/DEC 2003)

Quantization is a process of approximation or rounding off. Assigning PCM codes to absolute magnitudes is called quantizing.

11. What are the two fold effects of quantizing process?

- The peak-to-peak range of input sample values subdivided into a finite set of decision levels or decision thresholds
- The output is assigned a discrete value selected from a finite set of representation levels are reconstruction values that are aligned with the treads of the staircase.

12. Define dynamic range.

Dynamic range is the ratio of the largest possible magnitude to the smallest possible magnitude. Mathematically, dynamic range is DR= Vmax/Vmin

13. Define Quantization error?

The difference between the instantaneous values of the quantized signal and the input signal is called as quantization error.

PREDICTION ERROR:The difference between the actual sample of the process at the time of interest and the predictor output is called a prediction error.

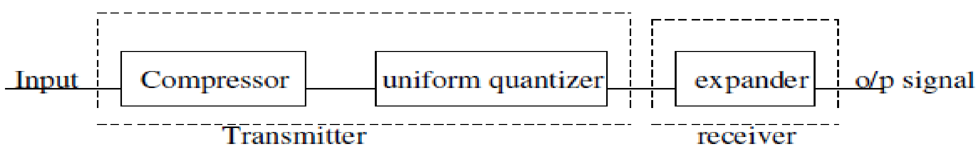
14. What are the three types of quantiser?

- Midtread quantiser
- Midriser quantiser
- Biased quantiser

15. Define companding. (APRIL/MAY 2004)

Companding is the process of compressing and expanding. With companded systems, the higher amplitude analog signals are compressed prior to transmission and then expanded at the receiver.

Block diagram:



Types of companding:

1. μ law companding
2. A law companding

16. Define slope overload. How it is reduced.

The slope of the analog signal is greater than the delta modulator can maintain, and is called slope overload.

Slope overload is reduced by increasing the clock frequency and by increasing the magnitude of the minimum step size.

17. Define granular noise. How it is reduced. (NOV/DEC 2004)

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.

18. 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.

19. Define delta modulation.

It transmits only one bit per sample. It compared the value with the previous sample value. It is used to reduce the signaling rate and transmission bandwidth.

20. What are two main disadvantages of delta modulation?

- Slope overload distortion
- Granular noise

21. What are two types of companding?

- E-law companding
- A-law companding

22. What is DPCM? (MAY/JUNE 2005)

DPCM is nothing but differential pulse code modulation where the differences in the amplitude of the two successive samples are transmitted rather than the actual sample.

23. Define TDM.

The signals to be multiplexed are transmitted sequentially one after the other. Each signal occupies a short time slot.

24. Define crosstalk and guard time.

Crosstalk means interference between the adjacent TDM channels. It is the unwanted coupling of information from one channel to the other. Guard time is the spacing introduced between the adjacent TDM channels.

25. Define Nyquist rate.

The sampling rate of 2W samples per second, for a signal of bandwidth of W Hertz, is called Nyquist rate.

26. What is Aliasing? (AU NOV/DEC 2004)

Aliasing is the phenomenon of a high frequency component in the spectrum of the signal taking on the identity of a lower frequency in the spectrum of its sampled version. This effect is due to the Sampling rate less than the Nyquist rate.

27. What are the measures to combat the effect of aliasing?

- Prior to sampling, a low pass filter is used to attenuate those high frequency Components of the signal that are not essential to the information being conveyed by the signal.
- The filtered signal is sampled at a rate slightly higher than the Nyquist rate.

28. What is Pulse Amplitude Modulation?

Pulse Amplitude Modulation is a process in which the amplitudes of regularly spaced pulses are varied in proportion to the corresponding sample values of a continuous message signal.

29. What are the operations involved in the generation of the PAM signal?

- Instantaneous sampling of the message signal m (t) every Ts seconds, where the sampling rate fs = 1/ Ts is chosen in accordance with the sampling theorem.
- Lengthening the duration of each sample so obtained to some constant value T.

30. What is the channel bandwidth required for a PAM signal? (AU NOV/DEC-2007)

The channel bandwidth required for a PAM signal fc = NfM where N = fc/fM is the total number of signals which may be multiplexed. Multiplexing a number of signals by PAM time division requires no bandwidth that would be required to multiplex these signals by FDM using SSB transmission.

31. What is natural sampling?

In Natural sampling, the sampled signal consists of a sequence of pulses of varying amplitude whose tops are not flat but follow the waveform of the message signal m (t).

32. What is flat top sampling?

In flat top sampling, the duration of each sample is lengthened to T, to avoid the use of an excessive transmission bandwidth, since bandwidth is inversely proportional to pulse duration.

33. Define Aperture effect. (AU APR/MAY-2004)

In flat top sampling, due to the lengthening of the sample, amplitude distortion as well as a delay of T/2 was introduced. This distortion is referred to as Aperture effect.

34. How aperture effect can be corrected? (AU NOV/DEC-2008)

Aperture effect can be corrected by connecting an equalizer in cascade with the low pass reconstruction filter. This equalizer has the effect of decreasing the in-band loss of reconstruction filter as the frequency increases in such a manner as to compensate for the aperture effect. The difference between the actual sample of the process at the time of interest and the predictor output is called a prediction error.

35. Define delta modulation.

Delta modulation is the one-bit version of differential pulse code modulation.

36. Define adaptive delta modulation.

The performance of a delta modulator can be improved significantly by making the step size of the modulator assume a time- varying form. In particular, during a steep segment of the input signal the step size is increased. Conversely, when the input signal is varying slowly, the step is reduced, In this way, the step size is adapting to the level of the signal. The resulting method is called adaptive delta modulation (ADM).

37. How signal is recovered through holding? (AU APR/MAY-2006)

In signal recovery through holding, the sample pulses are extended; that is, the sample value of each individual baseband signal is held until the occurrence of the next sample of that same baseband signal. The output waveform consists of up and down staircase waveform with no blank intervals. These voltage transitions are rounded as the capacitor charges and discharges exponentially.

38. What is Pulse Width Modulation?

In Pulse Width Modulation, the width of regularly spaced pulses is varied in proportion to the corresponding sample values of a continuous message signal.

39. What is Pulse Position Modulation?

In Pulse Position Modulation, the positions of regularly spaced pulses are varied in proportion to the corresponding sample values of a continuous message signal.

40. How channel synchronization is done in PAM systems? (AU APR/MAY-2008)

In PAM systems, channel synchronization is done by transmitting a marker pulse in addition to the message bearing pulses. This marker pulse can be identified by making its amplitude exceed that of all possible message pulses.

16 MARKS

1. Explain the block Diagram and Quantization process of PCM system in detail. (APRIL/MAY 2007)
2. Derive the expression for sampling process and the reconstruction of original signals in time domain? (MAY 2008)
3. Explain about bandpass sampling in detail.
3. Explain the generation of PAM signals and its power spectrum? (MAY 2008)
4. Explain briefly about the effects of under sampling? (NOV 2005)
6. Explain DM and ADM with suitable waveforms? (APRIL//MAY 2008)
7. Derive the expression for quantization noise and signal to noise ratio in PCM?
8. Explain in detail about TDM? (NOV/DEC 2004)
9. Explain in detail about DPCM.

UNIT –II
BASEBAND FORMATTING TECHNIQUES

TWO MARKS

1. What is ISI? (AU APR/MAY 2004, 2005)

Inter Symbol Interference arises when the communication channel is dispersive. Intersymbol interference is caused by overlapping tails of the pulse with adjacent pulses. There is dual effect due to the occurrence of pulses before and after the sampling instant ti is called ISI. It is a major source of bit errors in the reconstructed data stream at the receiver.

2. Define Eye pattern. (AU-APR/MAY 2004)

Eye pattern is defined as the synchronized superposition of all possible realizations of the signal of interest viewed within a particular signaling interval. The interior region of the eye pattern is eye opening.

3. What is the information provided by the eye pattern? (AU-APR/MAY 2005)

The width of the eye opening defines the time interval over which the received signal can be sampled without error from ISI. The sensitivity of the system to timing error is determined by the rate of change of eye as the sampling time is varied.

4. How many openings are there in an M-ary system?

In the case of M-ary system, the eye pattern contains (M-1) eye openings stacked up vertically one on the other, where M is the number of discrete amplitude levels used to construct the transmitted signal.

5. What is delta sigma modulation?

A delta modulation scheme that incorporates integration at its input is called delta sigma modulation.

6. What is the effect of thermal noise on PCM system? (AU-APR/MAY 2005)

The effect of thermal noise on PCM system is to cause the matched filter detector to make an occasional error in determining whether a binary 1 or binary 0 was transmitted. If the thermal noise is white and Gaussian, the probability of error depends on the ratio Eb/J.

7. State the Nyquist criterion for distortion less baseband binary criterion.

The frequency function P (f) eliminates ISI for samples taken at intervals Tb provided that it satisfies the equation

$$p(nT) = \begin{cases} 1, & n = 0 \\ 0, & n \neq 0 \end{cases}$$

8. Define a sinc pulse.

Sinc pulse is a signal waveform that produces zero ISI and is defined by
P (t) = sin (2MWt)/2MWt = sinc (2Wt)

10. State an important property of the matched filter.

The peak pulse signal to noise ratio of a matched filter depends only on the Signal energy to the power spectral density of the white noise at the filter output.

11. Define raised cosine spectrum. (AU-APR/MAY 2007)

The raised cosine spectrum consists of a flat portion and a roll off portion that has a sinusoidal form as follows:

$$p(t) = \sin c\left(\frac{t}{T}\right) \propto \frac{1}{|t|}$$

-

The raised cosine frequency characteristic is given by

$$P(f)=\begin{cases}\frac{1}{2B_0} & 0\le|f|<(1-\alpha)B_0\\ \frac{1}{4B_0}\left[1+\cos\frac{\pi(|f|-(1-\alpha)B_0)}{2\alpha B_0}\right] & (1-\alpha)B_0\le|f|<(1+\alpha)B_0\\ 0 & |f|\ge(1+\alpha)B_0\end{cases}$$

where $\alpha\in[0,1]$ is called the rolloff factor and $B_0=\frac{R}{2}$ (i.e., $B_0=\frac{1}{2T}$) .

12. What are the practical difficulties in a sinc pulse?

The amplitude characteristics of the sinc pulse should be flat from –W to W And zero elsewhere. This is physically unrealizable because of the abrupt Transitions at the band edges ±W. The function decreases as 1/t for large t, resulting in a slow rate of decay. There is practically no margin of error in sampling times in the receiver.

13. Define roll off factor.

The roll off factor O = 1 – f1/W, it indicates the excess bandwidth over the Ideal solution.The amount of ISI resulting from timing errors decreases as the roll off factor is increased from zero to unity.

14. What is correlative coding? (AU-NOV/DEC 2007,2005)

By adding intersymbol interference to the transmitted signal in a controlled manner, it is possible to achieve a signaling rate equal to the Nyquist rate of 2W symbols per second in a channelbandwidth of W Hertz. Such schemes are called correlative level coding or partial response signaling schemes.

15. What is duobinary signaling? (AU-OCT 2006)

Duobinary signaling scheme is also called as class I partial response scheme. The term duo implies doubling the transmission capacity of a straight binary system.

16. What is the major drawback of duobinary signaling scheme? (OCT 98,)

The major drawback of duobinary signaling scheme is that once errors are made they tend to propagate through the output because a decision on the current input depends on the correctness of the decision made on the previous input.

17. What is precoding?

A practical means of avoiding the error propagation phenomenon is to use precoding before duobinary coding. The precoding operation performed on the binary data sequence {bk} converts into another binary sequence {dk} defined by dk = bk + dk-1, where the symbol + denotes modulotwo addition.

18. What is modified duobinary signaling? (AU-NOV/DEC 2008)

Modified duobinary signaling or Class IV partial response technique, which involves a correlation span of two binary digits. This special form of correlation is achieved by subtracting amplitude modulated pulses spaced 2Tb seconds apart.

19. Define data Signaling Rate

Data signaling rate is defined as the rate measured in terms bits per second(b/s) at which data are transmitted. Data signaling rate Rb=I/Tb, Where Tb=bit duration.

20. What is the width of the eye?

It defines the time interval over which the received waveform can be sampled without error from intersymbol interference.

21. What is sensitivity of an eye?

The sensitivity of the system to timing error is determined by the rate of closure of the eye as the sampling time is varied

22. What is margin over noise?

The height of the eye opening at a specified sampling time defines the margin over noise.

23. What are the types of adaptive equalization.

- Prechannel equalization.
- Post channel equalization.

24. Define prechannel equalization.

When an equalization is done at the transmitting side is called prechannel equalization.

25. Define post channel equalization

When an equalization is done at the receiving side is called post channel equalization.

26. Define duobinary base band PAM system.

Duo binary encoding reduces the maximum frequency of the base band signal. The word "duo" means to the double the transmission capacity of the binary system. Let the PAM signal ak represents Kth bit. Then encoder generates the new waveform as, Ck=ak+ak-1 .Thus two successive bits are added to get encoded value of kth bit. Hence Ck becomes a correlated signal even though ak is not correlated .This introduces inter symbol interference in the controlled manner to reduce the bandwidth.

27. What is meant by forward and backward estimation?

AQF: Adaptive quantization with forward estimation. Unquantized samples of the input signal are used to derive the forward estimates.
AQB: Adaptive quantization with backward estimation. Samples of the quantizer output are used to derive the backward estimates.
APF: Adaptive prediction with forward estimation, in which unquantized samples of the input signal are used to derive the forward estimates of the predictor coefficients.
APB: Adaptive prediction with backward estimation, in which Samples of the quantizer output and the prediction error are used to derive estimates of the predictor coefficients.

28. What are the limitations of forward estimation with backward estimation?

- Side information
- Buffering
- Delay

29. How are the predictor coefficients determined?

For the adaptation of the predictor coefficients the least mean square (LMS) algorithm is used.

30. Define adaptive sub band coding

It is a frequency domain coder, in which the speech signal is divided in to number of subbands and each one is coded separately. It uses non masking phenomenon in perception for a better speech quality. The noise shaping is done by the adaptive bit assignment.

31. What are formant frequencies?

In the context of speech production the formant frequencies are the resonant frequencies of the vocal tract tube. The formants depend on the shape and dimensions of the vocal tract.

32. What is the bit rate in ASBC?

Nfs= (MN) (fs/M)

PART-B(16 MARKS)

1. Describe the conditions for zero ISI based on sampling theorem.
2. Elaborate on the concept and characteristics of correlative coding.
3. Explain the following with necessary figures and expression
   - Raised cosine channels.
   - Adaptive equalization for data transmission.
4. Draw the Eye pattern and explain its parts.

UNIT – III
-BASEBAND CODING TECHNIQUES
<u>TWO MARKS</u>

1.What are the properties of matched filter? (NOV'2007)
The signal to noise ratio of the matched filter depends only upon the ratio of the signal energy to the psd of white noise at the filter input .The output signal of a matched filter is proportional to a shifted version of the autocorrelation function of the input signal to which the filter is matched.

2. Why do we go for Gram-Schmidt Orthogonalization procedure?
Consider a message signal m. The task of transforming an incoming message mi=1,2,…..M, into a modulated wave si(t) may be divided into separate discrete time & continuous time operations.
The justification for this separation lies in the Gram-Schmidt orthogonalization procedure which
permits the representation of any set of M energy signals, {si(t)}, as linear combinations of N orthonormal basis functions

3.What is matched filter receiver? (MAY'2009)
A filter whose impulse response is a time reversed & delayed version of some signal Zj (t),
then it is said to be matched to j(t) correspondingly, the optimum receiver based on the detector is referred to as the matched filter receiver.

4.What are the three broad types of synchronization ?
1. Carrier synchronization
2. Symbol & Bit synchronization
3. Frame synchronization.

5.What is carrier synchronization ?
The carrier synchronization is required in coherent detection methods to generate a coherent reference at the receiver. In this method the data bearing signal is modulated on the carrier in such a way that the power spectrum of the modulated carrier signal contains a discrete component at the carrier frequency.

6. What are the two methods for carrier synchronization.
1. Carrier synchronization using Mth Power loop
2. Costas loop for carrier synchronization

7. What is called symbol or bit synchronization ?
In a matched filter or correlation receiver, the incoming signal is sampled at the end of one bit or symbol duration. Therefore the receiver has to know the instants of time at which a symbol or bit is transmitted. That is the instants at which a particular bit or symbol status and when it is ended. The estimation of these times of bit or symbol is called symbol or bit synchronization.

8. What are the two methods of bit and symbol synchronization.
1) Closed loop bit synchronization
2) Early late gate synchronizer

9. What are the disadvantages of closed loop bit synchronization.
1) If there is a long string of 1's and o's then y(t) has no zero crossings and synchronization may be lost.
2) If zero crossing of y(t) are not placed at integer multiples of Tb, the synchronization suffers from timing Jitter.

10. What is called frame synchronization ?
Depending on bits used for encoding, the word length is defined. Thus each word container some fixed number of bits. The receiver has to know when a particular frame status and when its individual message bits status. This type of synchronization is called frame synchronization.

11. Why synchronization is required ?
The signals from various sources are transmitted on the single channel by multiplexing. This requires synchronization between transmitter and receiver. Special synchronization bits are added in the transmitted signal for the purpose. Synchronization is also required for detectors to recover the digital data properly from the modulated signal.

12. What is maximum likelihood detector?
Maximum likelihood detector computes the metric for each transmitted message compares them and then decides in favor of maximum. The device for implementing the decision rule
i.e; set ^m = mi if
In [fx(x/mk)] is maximum for k=i is called maximum –likelihood detector and the decision rule is called maximum likelihood.

13. Define antipodal signals.
A pair of sinusoidal signals that differ only in a phase shift of 180 degrees are referred to as antipodal signals.

14. Explain how QPSK differs from PSK in term of transmission bandwidth and bit information it carries?
For a given bit rate 1/Tb, a QPSK wave requires half the transmission bandwidth of the corresponding binary PSK wave. Equivalently for a given transmission bandwidth, a QPSK wave carries twice as many bits of information as the corresponding binary PSK wave.

15. Give the equation for average probability of symbol error for coherent binary
(MAY'2009)
PSK. Average probability of signal error, Pe = 1 / 2 erfc (ZEb / No)

16. Define QPSK.
QPSK is Quadriphase –shift keying. In QPSK the phase of the carrier takes on one of the four equally spaced values Such as $\pi/4$, $3\pi/4$, $5\pi/4$ and $7\pi/4$.

17. Define Dibit.
A unique pair of bits is called a dibit. Gray encoded set of dibits 10, 00, 01 & 11.
18. Give the two basic operation of DPSK transmitter.
1. Differential encoding of the input binary wave
2. Phase –shift keying hence, the name differential phase shift keying.

19. Define deviation ratio in MSK.
The parameter h is defined by h= Tb(f1-f2)
h is deviation ratio , measured with respect to bit rate 1/Tb.

20. What is nominal carrier frequency in MSK?
Nominal carrier frequency is the arithmetic mean of the two frequencies f1 and f2 and it is given as fc = . (f1 + f2), Where f1 is the frequency for symbol –1, f2 is the frequency for symbol – 0

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29. Hierarchy of digital modulation techniques.

Digital modulation techniques can be classified in to coherent and non coherent techniques.
Each of these two classes can be subdivided in to binary and M-ary techniques .
Coherent binary modulation techniques: Amplitude shift keying,Phase shift keying,Frequency shift keying .
Coherent M-ary modulation techniques: M-ary ASK,M-ary FSK,M-ary PSK.Noncoherent binary modulation techniques: Noncoherent ASK,DPSK
Noncoherent FSK
Noncoherent M-ary modulation techniques: M-ary ASK,M-ary FSK,M-ary DPSK

30. What is coherent binary PSK?

In coherent binary PSK system, the pair of signals s1(t) and s2(t) used to represent binary symbols 1 and 0 respectively, are defined by

$$PSK(t) = \begin{cases} \sin(2\pi f t) & for\, bit\, 1 \\ \sin(2\pi f t + \pi) & for\, bit\, 0 \end{cases}$$

31. What is coherent binary FSK?

In coherent binary PSK system, the pair of signals s1(t) and s2(t) used to represent binary symbols 1 and 0 respectively, are defined by

$$FSK(t) = \begin{cases} \sin(2\pi f_1 t) & \mathit{for\ bit}\ 1 \\ \sin(2\pi f_2 t) & \mathit{for\ bit}\ 0 \end{cases}$$

32. What is coherent QPSK?

In QPSK, information carried by the transmitted signal is contained in the phase. The phase of the carrier takes on one of four equally spaced values, such as M/4, 3M/4, 5M/4, 7M/4. The transmitted signal is defined as

$$s_i(t) = A_c ps(t) \cos\left(2\pi f_c t + \frac{2\pi i}{M}\right)$$

33. What is noncoherent modulation? (NOV'2009)

Whenever it is impractical to have knowledge of the carrier phase at the receiver, noncoherent modulation is employed. One of these two signals is sent over an imperfect channel that shifts the carrier phase by an unknown amount. Let g1(t) and g2(t) denote the phase shifted versions of
s1(t) and s2(t), respectively. It is assumed that the signals g1(t) and g2(t) remain orthogonal and have the same energy E, regardless of the unknown carrier phase. Such a signaling scheme is referred as noncoherent modulation.

34. What is noncoherent binary FSK?

In noncoherent binary FSK, the transmitted signal is defined by Eq: (6.180)[413] where the carrier frequency fi equals one of two possible values f1 and f2, to ensure that the signals representing these two frequencies are orthogonal, fi = ni/Tb, ni is an integer.

35. What is DPSK?

In DPSK, two basic operations are performed at the transmitter:
(i) Differential encoding of the input binary wave
(ii) Phase shift keying.
Let s1(t) denote the transmitted DPSK signal for 0 _ t _ 2Tb.When there is a symbol 1 at the transmitter input for the second part of the interval Tb _ t _ 2Tb, the transmission of the symbol 1
leaves the carrier phase unchanged. When there is a symbol 0 in the second part of the interval, the transmission of symbol 0 advances the carrier phase by 180 degrees.

36. What is M-ary modulation?

In M-ary modulation, one of M possible signals s1(t), s2(t),… sM(t) are send during each signaling interval of duration T. These signals are generated by changing the amplitude, phase or frequency of a carrier in M discrete steps.

37. When M-ary signaling schemes are preferred over binary signaling schemes and why? (NOV'2006)

M-ary signaling schemes are preferred over binary signaling schemes when the requirement is to conserve bandwidth at the expense of increased power. In binary PSK, bandwidth required in inversely proportional to Tb, whereas in M-ary, the bandwidth required is inversely proportional to nTb, reduction in bandwidth by the factor n = log2M over binary PSK.

38. What is M-ary PSK?

In m-ary PSK, the phase of the carrier takes on one of M possible values namely _i = 2(i– 1)_/M, where i = 1,2,3,..,M

$$s_i(t) = \underbrace{\sqrt{\frac{2E_s}{T}}}_{Constant} \cos\left(\underbrace{2\pi f_c t}_{\substack{Changing \\ with\ time}} + \underbrace{\frac{2\pi i}{M}}_{\substack{Changing \\ with\ info}}\right) \qquad i = 0,1,....M$$

39. What is M-ary FSK?
In M-ary FSK, the transmitted signals are defined by,

$$FSK(t) = \begin{cases} \sin(2\pi f_1 t) & \textit{for bit } 1 \\ \sin(2\pi f_2 t) & \textit{for bit } 0 \end{cases}$$

Where i = 1, 2, 3, M and carrier frequency fc = nc/2T for some fixed integer nc.

40. What are the needs for data compression? (NOV'2007)
The data compression is employed in two types of situations:
a. In source coding where the permitted coding alphabet cannot exactly represent the information source.
b. Information transmission at a rate greater than the channel capacity.

<u>16 MARKS</u>

1. With necessary equations and signal space diagram, obtain the probability of error for coherent binary FSK systems. (MAY'2005)
2. Draw the block diagram of QPSK and obtain the probability of error. (MAY'2007)
3. With necessary equations and signal space diagram, obtain the probability of error for coherent binary MSK systems.
4. Explain BPSK signal transmission and coherent BPSK reception with suitable diagrams. Derive an expression for the probability of symbol error for the scheme.(MAY'2006,NOV'2008)
5. With neat block diagram, explain briefly how symbol synchronization is achieved? (MAY'2009)
6.Explain about Correlation receivers and matched filters in detail.
7.Compare the Signal Space diagram for various modulation techniques.

UNIT – IV
BASEBAND RECEPTION TECHNIQUES
<u>TWO MARKS</u>

1. What is linear code?
A code is linear if the sum of any two code vectors produces another code Vector.

2. What is code rate?
Code rate is the ratio of message bits (k) and the encoder output bits (n).It is defined by r (i.e.)
r= k/N

3. Define code efficiency.
It is the ratio of message bits in a block to the transmitted bits for that block by the encoder i.e.

$$\text{Code efficiency} = \frac{\text{Message bits in a block}}{\text{Transmitted bits for the block}}$$

4. What is hamming distance?
The hamming distance between two code vectors is equal to the number of elements in which they differ. For example let the two code vectors be X=(101) and Y= (110). These two code vectors differ in second and third bits. Therefore the hamming distance between x and Y is two.

5. What is meant by systematic & non-systematic code?
In a systematic block code, message bit appear first and then check bits. In the non-systematic code, message and check bits cannot be identified in the code vector.

6. How syndrome is calculated in Hamming codes and cyclic codes? (NOV'2004)
In hamming codes the syndrome is calculated as, S = YHT
Here Y is the received and HT is the transpose of parity check matrix. In cyclic code, the syndrome vector polynomial is given as, S (P) = remainder (y (p)/ G (P))
Y (P) is received vector polynomial and G (p) is generator polynomial.

7. What is BCH Code? (MAY'2006)
BCH codes are most extensive and powerful error correcting cyclic code. The decoding of BCH coder is comparatively simpler. For any positive integer 'm' and 't' , there exists a BCH code with following parameters :
Block length n = 2m-1
No. of parity check bits: q= n-k
Minimum distance: dmin = 2t +1

8. What are the conditions to satisfy the hamming code?
1) No. of Check bits q >=3
2) Block length n = 2q –1
3) No of message bits K = n-q
4) Minimum distance dmin =3

9. Define code word & block length.
The encoded block of 'n' bits is called code word. The no. of bits 'n' after coding is called block length.

10. Give the parameters of RS codes.
Reed Solomon codes.
These are non binary BCH codes.
Block length = n =2m -1 symbols
Message size: k symbols
Parity check size: n-k= 2t symbols
Minimum distance, dmin =2t +1 symbols.

11. Why RS codes are called maximum distance separable codes?
(n,k) Linear block code for which the minimum distance equals n – k + 1 is called maximum
distance separable codes. For RS code minimum distance equals n – k + 1 so it is called as maximum distance separable codes.

12. What are Golay codes?
Golay code is the (23, 12) cyclic code whose generating polynomial is,
G(p) =P11+P9+p7+P6+p5+p+1
This code has a minimum distance of dmin=7. This code can correct upto 3 errors. It is perfect code.

13. What are the advantages of cyclic codes?
1. Encoders and decoders for cyclic codes are simple
2. Cyclic codes also detect error burst that span many successive bits.

14. Define free distance and coding gain. (NOV'2005)
Free distance is the minimum distance between code vectors. It is also equal to minimum weight of the code vectors. Coding gain is used as a basis of comparison for different coding methods. To achieve the same bit error rate the coding gain is defined as,

$$A = \frac{(Eb/No)\ \text{encoded}}{(Eb/No)\ \text{coded}}$$

15. What is RS code? (MAY'2005)
These are nonlinear BCH codes. The encoder for RS codes operate on multiple bits simultaneously. The (n,k) RS code takes the groups of m-bit symbols of the incoming binary data stream. It takes such 'k' number of symbols in one block. Then the encoder adds (n-k) redundant symbols to form the codeword of 'n' symbols.
RS code has:
Block length: n=2m-1 symbols
Message size: k symbols
Parity check size: n-k=2t symbols
Minimum distance: dmin=2t+1 symbols

16. Define constraint length in convolutional codes.
Constraint length is the number of shifts over which the single message bit can influence the encoder output. It is expressed in terms of message bits.

17. What is the difference between block codes and convolutional codes? (NOV'2005)
Block codes take 'k' number of message bit simultaneously and form 'n'-bit code vector. This code vector is called block. Convolutional code takes one message bit at a time and generates two or more encoded bits. Thus convolutional codes generate a string of encoded bits for input message string.

18. What are the error detection and correction capabilities of Hamming codes?
The minimum distance (dmin) of Hamming codes is '3'. Hence it can be used to detect double errors or correct single errors. Hamming codes are basically linear block codes with dmin=3.

19. Define channel data rate.
It is the bit rate at the output of encoder. If the bit rate at the input of encoder is Rs, then channel data rate will be, Channel data rate (Ro) = (n/k) Rs

20. What is convolutional code? (JUNE'2006)
Fixed number of input bits is stored in the shift register and they are combined with the help of mod-2 adders. This operation is equivalent to binary convolution and hence it is called convolution coding.

21. Mention any two methods used for error control coding.
1. Forward acting error correction
2. Error detection with retransmission

22. Mention the two types of errors introduced during transmission on the data.
1. Random errors
2. Burst errors

23. What are the properties of cyclic code?
1. Linear property

2. Cyclic property

24. What are the needs for error control coding?

The needs for error control coding are

(i) To change the data quality from problematic to acceptable one

(ii) To reduce the required Eb/No for a fixed bit error rate.

(iii) This reduction in Eb/No may be exploited to reduce the required transmitted power or reduce the hardware costs by requiring a smaller antenna size in the case of radio communications.

25.What are the types of error correcting codes? (NOV'2008)

The codes are classified in to block codes and convolution codes. The distinguishing feature for the classification is the presence or absence of memory in the encoders for the two codes.

26.What is discrete memory-less channel?

The waveform channel is said to be memory-less, if the detector output in a given time interval depends only on the signal transmitted in that interval, and not on any previous transmission.

27.What are systematic codes?

Block codes in which the message bits are transmitted in unaltered form are called systematic codes. For application requiring both error detection and error correction, the use of systematic codes simplifies implementation of the decoder.

28.Give the structure of the code word. (JUNE'2006)

Consider a (n,k) code word, in which k bits of the n code bits are always identical to the message sequence to be transmitted. The remaining n-k bits are computed from the message bits in accordance with a prescribed encoding rule that determines the mathematical structure of the code and these bits are referred as generalized parity check bits or simply parity bits.

b0, b1, . . . , bn-k-1 m0, m1, . . . , mk-1

29.What are repetition codes? (NOV'2008)

Repetition codes represent the simplest type of linear block codes. A single message bit is encoded into a block of n identical bits, producing a (n,1) block code. Such a code allows provision for a variable amount of redundancy. There are only two code words in the code: all zero code word and all one code word.

30. State 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 code word have the same syndrome.

31. What are cyclic codes?

Cyclic codes form a subclass of linear block codes. A binary code is said to be cyclic code if it exhibits two fundamental properties:

1. Linearity property: The sum of any two code words in the code is also a code word.
2. Cyclic property: Any cyclic shift of a code word in the code is also a code word.

32. State Channel coding theorem. (MAY 2007)

The channel coding theorem states that if a discrete memory-less channel has capacity C and a source generates information at a rate less than C, then there exists a coding technique such that the output of the source may be transmitted over the channel with an arbitrarily low probability of symbol error.

33. Define Hamming weight.

The Hamming weight w(c) of a code vector is defined as the number of nonzero elements in the code vector.

34. Define minimum distance dmin.

The minimum distance dmin of a linear block code is defined as the smallest Hamming distance between any pair of code vectors in the code word.

35. Define constraint length of a convolutional code. (NOV'2007)

The constraint length of a convolutional code , expressed in terms of message bits, is defined as the number of shifts over which a single message bit can influence the encoder output.

36. Define maximum likelihood decoding of convolutional codes.

Let p(r|c) denote the conditional probability of receiving r, given that c was sent and the log-likelihood function equals log p(r|c). Then the maximum likelihood decoder or decision rule is described as follows: Choose the estimate c for which the log likelihood function log p(r|c) is maximum.

37. Define free distance of a convolutional code.

The free distance of a convolutional code is defined as the minimum Hamming distance between any two code words in the code.

38. How may errors can be corrected by a convolutional code?

A convolutional code with free distance ‘d’ can correct t errors if and only if ‘d’ is greater than 2t.

16 MARKS

1. State and prove the properties of syndrome decoding (MAY’2006)
2. Consider a rate 1/3, non-systematic convolutional code with, g(1)(p)={1,0,1}, g(2)={1,1,1}. Determine the encoder output corresponding to the data sequence {1,0,1,0,1}. If the first and the fourth bits of the encoded sequence are affected during transmission, demonstrate the error correcting capability of the viterbi algorithm. (NOV’2005)
3. A (15,5) linear cyclic code has a generator polynomial,
g(D)=1+D+D2+D4+D5+D8+D10
Draw block diagrams of an encoder and syndrome calculator for this code. Find the code polynomial in systematic form, for the message polynomial m(D)=1+D2+D4. Is y(D)=1+D4+D6+D8+D14, a code polynomial? If not, find the syndrome of y(D). (NOV’2008)
4. Briefly explain the viterbi decoding algorithm.
5. Draw the diagram of the . rate convolutional encoder with generator polynomials g(1)(D)=1+D and g(2)(D)=1+D+D2. And compute the encoder output for input sequence 101101. (NOV’2009)
6. Explain any four characteristics of the following block codes (i) BCH codes (ii) CRC codes (iii) maximum length codes.
7. Explain the syndrome ‘S’ for all five probable single error patterns in (5,1) repetition code. (JUNE’2006)
8. Generate the code words for (7, 4) Hamming code.
9. Describe the design procedure for linear block code.

UNIT –V
BANDPASS SIGNAL TRANSMISSION AND RECEPTION
TWO MARKS

1. What is frequency hop spread spectrum?

In frequency hop spread spectrum, the spectrum of a data modulated carrier is widened by changing the carrier frequency in a pseudo-random manner and this technique rely on the availability of a noise-like spreading code called pseudo-noise sequence.

2. What is pseudo-noise (PN) sequence?

A PN sequence is a periodic binary sequence with a noise-like waveform that is usually generated by means of a feedback shift register, consists of an ordinary shift register made up of m flip-flops and a logic circuit that are interconnected to form a multi-loop feedback circuit.

3. What is synchronization?

For proper operation, a spread spectrum communication system requires that the locally generated PN sequence used in the receiver to despread the received signal be synchronized to the PN sequence used to spread the transmitted signal in the transmitter. It consists of acquisition and tracking.

4. What is acquisition? (MAY 2007)
In acquisition or coarse synchronization, the two PN codes are aligned to within a fraction of a chip in as short time as possible. PN acquisition takes place in two steps. First the receive signal is multiplied by a locally generated PN code to produce a measure of correlation between it and the PN code used in the transmitter. Next, an appropriate decision rule and search strategy is used to process the measure the correlation so obtained to determine whether the two codes are in synchronism and what to do if they are not.

5. What is spread spectrum?
Spread spectrum is a means of transmission in which the data sequence occupies a bandwidth in excess of the minimum bandwidth necessary to send it. The spread spectrum is accomplished before transmission through the use of a code that is independent of the data sequence. The same code is used in the receiver (operating in synchronism with the transmitter) to despread the received signal so that the original data sequence may be recovered.

6. What is the advantage of the spread spectrum Communication?
The primary advantage of the spread spectrum communication is its ability to reject interference whether it is the unintentional interference by another user simultaneously attempting to transmit through the channel, or the intentional interference by a hostile transmitter attempting to jam the transmission.

7. What is direct sequence spread spectrum? (APR/MAY 2004,2005)
In direct sequence spread spectrum, two stages of modulation are used. First the incoming data sequence is used to modulate a wide band code. This code transforms the narrow band data sequence into a noise-like wide-band signal. The resulting wide-band signal undergoes a second modulation using a phase-shift keying technique.

8. What is tracking? (NOV 2008)
Once the incoming PN code has been acquired, tracking or fine synchronization takes place. It is accomplished using phase lock techniques, which is similar to those used for the local generation of coherent carrier references. The principle difference between them lies in the way in which phase discrimination is implemented.

9. Give the expression for the (SNR)0/P. (NOV 2006)
$(SNR)_o = (2T_b/T_c)(SNR)_I$ or
$10 \log_{10}(SNR)_o = 10 \log_{10}(SNR)_I + 3 + 10 \log_{10}(PG)$ dB
The 3 dB term on the right hand side of the above equation accounts for the gain in SNR that is obtained through the use of coherent detection. The last term $10\log_{10}(PG)$ accounts for gain in SNR obtained by the use of spread spectrum.

10. Define processing gain.
The processing gain PG is defined as the ratio of bit duration T_b to the chip duration T_c.
$PG = T_b/T_c$

11. Define jamming margin.
The jamming margin is defined as the ratio J/P which is given by
Jamming margin)dB = (Processing gain)dB – $10 \log_{10} (E_b/N_o)_{min}$
Where $10 \log_{10} (E_b/N_o)_{min}$ = The minimum value needed to support a prescribed average probability of error.

12. What is slow frequency hoping?
The slow frequency hoping is a type of frequency hoping in which the symbol rate R_S, of the MFSK signal is an integer multiple of the hop rate R_h, that is several symbols are transmitted on each frequency hop.

13. What is fast frequency hoping? (MAY 2008)
The fast frequency hoping is a type of frequency hoping in which the hop rate R_h, of the MFSK

signal is an integer multiple of symbol rate RS, that is, the carrier frequency will change or hop several times during the transmission of one symbol.

14 .What is the advantage of CDMA system? (JUNE 2009)
The advantage of the CDMA system is that collisions are not destructive; each of the signals involved in a collision would be received with only a slight increase in error rate.

15. What is near-far problem?
When an unwanted user's received power is much larger than the received power presented by the desired user, errors can occur. This problem is referred to as near-far problem and limits the utility of DS systems to applications where each user's received power is same.

16 .What are the disadvantages of spread spectrum?
The disadvantages of spread spectrum are:
The capabilities of physical devices used to generate the PN spread spectrum signal impose a practical limit on the attainable processing gain which is not enough to overcome the effects of some jammers of concern. DS spread spectrum is affected by near-far problem.

17. Why FH spread spectrum is not affected by near-far problem?
In a FH system, if the transmitter and receiver are tuned to frequency fi and the interfering signal fj`fi, then the demodulator does not even know the presence of the interferer since its spectrum is not in the signal bandwidth. Thus, there is no near-far problem.

18. Mention about the run property.
Among the runs of 1s and 0s in each period of a maximum length sequence, one half the runs of each kind are of length one, one fourth are of length two, one eighth are of length three, and so or as long as these function represent meaningful numbers of runs. This property is called the run property.

19. What are the two function of fast frequency hopping?
1. Spread Jammer over the entire measure of the spectrum of Txed signal.
2. Retuning the Jamming signal over the frequency band of Txed signal.

20. What are the three codes used for the anti jamming application ?
1. Golay code (24, 12)
2. Expurgated Golay (24, 11)
3. Maximum length shift register code.

21. What is called frequency hop spread spectrum?
In frequency hop spread spectrum, the frequency of the carrier hops randomly from one frequency to another frequency.

22. What does the term catastrophic cyclic code represent ?
'000' is not a state of the shift register sequence in PN sequence generator, since this results in a catastrophic cyclic code i.e once the 000 state is entered, the shift register sequence cannot leave this state.

23. What is called jamming effect?
In the frequency band of the interest, somebody else transmits the signals intentionally since these signals the in the frequency band of transmission, they interface the required signal. Hence it becomes difficult to detect the required signals. This is called jamming effect.

24. What is Anti jamming?
With the help of spread spectrum method, the transmitted signals are spread over the mid Frequency band. Hence these signals appear as noise. Then it becomes difficult for the jammers to send jamming Signals. This is called antijamming.

16 Marks.

1.Explain about Spread spectrum techniques with its block diagram.
2.Explain about PN Sequence generation and its properties.
3.Explain about Gold Codes and ML Sequence codes.
4.Explain about Frequency hopping with its types.
5.Explain about processing gain in BPSK.