EC51-DIGITAL COMMUNICATION
UNIT-1
DIGITAL COMMUNICATION SYSTEM

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. Define data Signalling Rate.

Data signalling 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.

4. Define modulation rate.

It is defined as the rate at which signal level is changed depending On the nature of the format used to represent the digital data.It is measured in Bauds or symbols per second.

5.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

6. What is called processing gain ?

Processing Gain (PG) is defined as the ratio of the bandwidth of spread message signal to the bandwidth of unspreaded data signal ie).

$$\text{Processing Gain} = \frac{\text{BW (spreaded signal)}}{\text{BW (Unspreaded signal)}}$$

7. 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.

8. 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.

9.How are the predictor coefficients determined?

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

10. Define adaptive subband 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.

11.Define a sinc pulse.

Sinc pulse is a signal waveform that produces zero ISI and is defined by
$P(t) = \sin(2\Pi Wt)/2\Pi Wt = \mathrm{sinc}(2Wt)$

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.

14. Define Bandwidth

Bandwidth is simply a measure of frequency range. The range of frequencies contained in a composite signal is its bandwidth.

15. What are the two characteristics of a channel?

Bandwidth
Power

16. Name the channels considered in digital communication

Telephone channels, coaxial cables, optical fibers, microwave radio, and satellite channels.

UNIT-2
BASEBAND FORMATTING TECHNIQUES

1.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.

2. 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.

3. What is the Nyquist sampling rate?

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

4.Define overload distortion.

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

5. Define quantization.

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

6. 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

7. Define Quantization error?

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

8.What are the three types of quantiser?

Midtread quantiser

Midriser quantiser

Biased quantiser

9. Define companding.

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.

10.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.

11.Define granular noise. How it is reduced.

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.

12. 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.

13. 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.

14.What are two main disadvantages of delta modulation?

Slope overload distortion

Granular noise

15.What are two types of companding?

μ -law companding

A-law companding

16. What is DPCM?

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.

17. Define TDM.

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

18. 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.

19.Define Nyquist rate.

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

20.What is Aliasing?

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.

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

(i). 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.

(ii). the filtered signal is sampled at a rate slightly higher than the Nyquist rate. 24.What is Pulse Amplitude Modulation?

22.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).

23.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.

24.Define Aperture effect.

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.

25.How signal is recovered through holding?

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.

26.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 continous message signal.

27.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 continous message signal.

28.How channel synchronization is done in PAM systems?

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.

UNIT-3
BASEBAND CODING TECHNIQUES

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?

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.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.

8. 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.

9. 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.

10. 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.

11What are the advantages of cyclic codes?

Encoders and decoders for cyclic codes are simple
Cyclic codes also detect error burst that span many successive bits.

12. Define free distance and coding gain.

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,

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

13. What is RS code?

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

14. 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.

15. What is the difference between block codes and convolutional codes?

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.

16 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.

17. 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]

18. What is convolutional code?

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.

19.Mention any two methods used for error control coding.

Forward acting error correction
Error detection with retransmission

20.Mention the two types of errors introduced during transmission on the data.

Random errors
Burst errors

21.What are the properties of cyclic code?

Linear property
Cyclic property

22.What are the needs for error control coding?

The needs for error control coding are :To change the data quality from problematic to acceptable one , To reduce the required Eb/No for a fixed bit error rate.

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.

23.What are the types of error correcting codes?

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.

24.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.

25.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.

26. 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.

27. 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:

Linearity property: The sum of any two code words in the code is also a code word.

Cyclic property: Any cyclic shift of a code word in the code is also a code word.

28. 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.

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

convolutional code with free distance dfree can correct t errors if and only if dfree is greater than 2t.

30.State Channel coding theorem.

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 ofsymbolerror.

31.Define Hamming weight.

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

32. 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.

33. Define constraint length of a convolutional code.

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.

UNIT-4
BASEBAND RECEPTION TECHNIQUES

1.What is an eye pattern?

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

2. 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.

3. 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.

4. What is margin over noise?

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

5. What is Inter symbol interference?

The transmitted signal will undergo dispersion and gets broadened during its transmission through the channel. So they happen to collide or overlap with the adjacent symbols in the transmission. This overlapping is called Inter Symbol Interference.

6. How eye pattern is obtained?

The eye pattern is obtained by applying the received wave to the vertical deflection plates of an oscilloscope and to apply a saw tooth wave at the transmitted symbol rate to the horizontal deflection plate.

7. Properties of matched filter.

The signal to noise ratio of the matched filter depends only upon the 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 auto_correlation function of the input signal to which the filter is matched.

8. What is matched filter receiver ?

A filter whose impulse response is a time reversed & delayed version of some signal j (t) then it is said to be matched to I (t) correspondingly, the optimum receiver based on the detector is referred to as the matched filter receiver.

9. 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 = m_i if

In [f_x(x/mk)] is maximum for k=i is called maximum ±likelihood detector and the decision rule is called maximum likelihood.

10. Define antipodal signals.

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

11.What are the three broad types of synchronization ?

Carrier synchronization

Symbol & Bit synchronization

Frame synchronization.

12. 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.

13.What are the two methods for carrier synchronization.

Carrier synchronization using M^{th} Power loop, Costas loop for carrier synchronization

14.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.

15.What are the two methods of bit and symbol synchronization.

Closed loop bit synchronization

Early late gate synchronizer

16.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.

17. 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.

18. 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.

19.What is ISI?

Inter Symbol Interference arises when the communication channel is dispersive. ISI is caused by overlapping tails of the pulse with adjacent pulses. The residual effect due to the occurrence of pulses before and after the sampling instant is called ISI.

20. What are the types of adaptive equalization.

1. Prechannel equalization.
2. Post channel equalization.

21.Define roll off factor.

The roll off factor $\alpha = 1 - f_1/W$, 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.

UNIT-5
BANDPASS SIGNAL TRANSMISSION AND RECEPTION

1.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.

2. 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 $7\Pi/4$.

3.Define Dibit.

A unique pair of bits is called a dibit. Gray encoded set of dibits 10, 00, 01 & 11.

4.Give the two basic operation of DPSK transmitter.

Differential encoding of the input binary wave
Phase –shift keying hence, the name differential phase shift keying

5.Define deviation ratio in MSK.

The parameter h is defined by $h= T_b(f_1-f_2)$
h is deviation ratio , measured with respect to bit rate 1/Tb.

6. 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 $f_c = \frac{1}{2}(f_1 + f_2)$, Where f1 is the frequency for symbol –1, f2 is the frequency for symbol **–** 0

7. 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

8. What is coherent binary PSK?

In coherent binary PSK system, the pair of signals $s_1(t)$ and $s_2(t)$ used to represent binary symbols 1 and 0 respectively

9. What is coherent binary FSK?

In coherent binary PSK system, the pair of signals $s_1(t)$ and $s_2(t)$ used to represent binary symbols 1 and 0 respectively

10. 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 $\Pi/4$, $3\Pi/4$, $5\Pi/4$, $7\Pi/4$.

11. What is noncoherent modulation?

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 $g_1(t)$ and $g_2(t)$ denote the phase shifted versions of $s_1(t)$ and $s_2(t)$, respectively. It is assumed that the signals $g_1(t)$ and $g_2(t)$ remain orthogonal and have the same energy E, regardless of the unknown carrier phase. Such a signaling scheme is referred as noncoherent modulation.

12. What is noncoherent binary FSK?

In noncoherent binary FSK, the transmitted signal is defined by Eq: (6.180)[413] where the carrier frequency f_i equals one of two possible values f_1 and f_2, to ensure that the signals representing these two frequencies are orthogonal, $f_i = n_i/T_b$, n_i is an integer.

13. What is DPSK?

In DPSK, two basic operations are performed at the transmitter:

Differential encoding of the input binary wave

Phase shift keying.

Let $s_1(t)$ denote the transmitted DPSK signal for $0 \le t \le 2T_b$.When there is a symbol 1 at the transmitter input for the second part of the interval $T_b \le t \le 2T_b$, 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. Eq: (6.182&6.183)[414]

14. What is M-ary modulation?

In M-ary modulation, one of M possible signals $s_1(t)$, $s_2(t)$,… $s_M(t)$ 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.

15.When M-ary signaling schemes are preferred over binary signaling schemes and why?

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 T_b, whereas in M-ary, the bandwidth required is inversely proportional to nT_b, reduction in bandwidth by the factor $n = \log_2 M$ over binary PSK.

16. What is M-ary PSK?

In m-ary PSK, the phase of the carrier takes on one of M possible values namely $\theta_i = 2(i- 1)\Pi/M$, where i = 1,2,3,..,M

17. What are the needs for data compression?

The data compression is employed in two types of situations:

In source coding where the permitted coding alphabet cannot exactly represent the information source.

Information transmission at a rate greater than the channel capacity.

16 MARK QUESTIONS

1.Explain in detail about Binary Phase Shift Keying and obtain an expression for its probability of error.
2.Explain in detail about Quadrature Phase Shift Keying and obtain an expression for its probability of error.
3.Explain in detail about Minimum Shift Keying and obtain an expression for its probability of error.
4.Explain in detail about the operation of Non Coherent Receivers in the presence of Random Phase Channel and implement the receiver.
5. Explain in detail about In phase and Quadrature Modulation systems. With necessary diagrams explain the operation of Quadrature Amplitude Modulation systems.
6.Draw the code tree of a Convolutional code of code rate r=1/2 and Constraint length of K=3 starting from the state table and state diagram for an encoder which is commonly used.
7. Draw the trellis diagram of a Convolutional code of code rate r=1/2 and Constraint length of K=3 starting from the state table and state diagram for an encoder which is commonly used.
8. Decode the given sequence 11 01 01 10 01 of a convolutional code with a code rate of r=1/2 and constraint length K=3, using viterbi decoding algorithm.
9. Explain in detail about Continuous Phase Frequency Shift Keying and obtain an expression for its probability of error.
10. With neat block diagram, explain briefly how symbol synchronization is achieved?
11. State and prove the properties of syndrome decoding
12. Consider a rate ., 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.
13. 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).
14. Briefly explain the viterbi decoding algorithm.
15. 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.
16. Explain any four characteristics of the following block codes (i) BCH codes (ii) CRC codes (iii) maximum length codes.
17. Explain the syndrome 'S' for all five probable single error patterns in (5,1) repetition code.
18. Generate the code words for (7, 4) Hamming code.
19. Describe the design procedure for linear block code.
20.Explain PCM with suitable waveforms
21. Explain DM and ADM with suitable waveforms?
22. Derive the expression for quantization noise and signal to noise ratio in PCM?

23.Derive the expression for sampling process in time domain?
24. Discuss channel classification and digital communication systems.
25. Discuss Gram-Schmidt Orthogonalization procedure.