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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

SUBJECT: EC8501/ DIGITAL COMMUNICATION **SEM / YEAR: V / III**

2MARKS WITH ANSWERS

UNIT I INFORMATION THEORY

1. What is entropy?

Entropy is also called average information per message. It is the ratio of total information to number of messages. i.e.,

Entropy, H = Total information / Number of messages

2. What is channel redundancy?

Redundancy = 1 – code efficiency

Redundancy should be as low as possible.

3. Name the two source coding techniques.

The source coding techniques are,

a) prefix coding b) Shannon-fano coding c) Huffman coding

4. Write the expression for code efficiency in terms of entropy.

Code efficiency = Entropy(H) / Average code word length(N)

5. What is memory less source? Give an example.

The alphabets emitted by memory less source do not depend upon previous alphabets. Every alphabet is independent. For example a character generated by keyboard represents memory less source.

6. Explain the significance of the entropy H(X/Y) of a communication system where X is the transmitter and Y is the receiver.

a) H(X/Y) is called conditional entropy. It represents uncertainty of X, on average, when Y is known.

b) In other words H(X/Y) is an average measure of uncertainty in X after Y is received.

c) H(X/Y) represents the information lost in the noisy channel.

7. What is prefix code?

In prefix code, no codeword is the prefix of any other codeword. It is variable length code. The binary digits (codewords) are assigned to the messages as per their probabilities of occurrence.

8. Define information rate.

Information rate(R) is represented in average number of bits of information per second. It is calculated as,

R = r H Information bits / sec

9. Calculate the entropy of source with a symbol set containing 64 symbols each with a probability pi = 1/ 64 .

Here, there are M = 64 equally likely symbols. Hence entropy of such source is given as,H = log 2 M = log 2 64 = 6 bits / symbol

10. State the channel coding theorem for a discrete memory less channel.

Statement of the theorem:

Given a source of ‗M' equally likely messages, with M >>1, which is generating information at a rate. Given channel with capacity C. Then if,R ≤ C

There exits a coding technique such that the output of the source may be transmitted over the channel with a probability of error in the received message which may be made arbitrarily small.

Explanation: This theorem says that if $R \leq C$; it is possible to transmit information without any error even if noise is present. Coding techniques are used to detect and correct the errors.

11. What is information theory?

Information theory deals with the mathematical modeling and analysis of a communication system rather than with physical sources and physical channels

12. Explain Shannon-Fano coding.

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.

13. Define bandwidth efficiency.

The ratio of channel capacity to bandwidth is called bandwidth efficiency. i.e,

Bandwidth efficiency = Channel Capacity / Bandwidth (B)

14. Define channel capacity of the discrete memory less channel.

The channel capacity of the discrete memory less channel is given as maximum average mutual information. The maximization is taken with respect to input probabilities.

15. Give the relation between the different entropies.

$$H(X,Y) = H(X)+H(Y/X)$$
$$= H(Y)+H(X/Y)$$

H(X)- entropy of the source,H(Y/X),

H(X/Y)-conditional entropy

H(Y)-entropy of destination

H(X,Y)- Joint entropy of the source and destination

16. Difference between Huffman and shannon fano coding

s.no	Huffman coding	Shannon fano coding
1	Codewords are assigned as per probability of symbol	Codewords are assigned as per probability of symbol
2	Lower probability symbols (two) are combined for next stage	Symbols are partitioned into two groups and combined
3	Average number of bits per message nearly equal to entropy	Average number of bits per messege are little higher than that of Huffman coding

17. State the advantages of Lempel-Ziv coding.

i) It doesnot require prior probabilities of the data sequence.

ii) This algorithm is adaptive

iii) It gives higher coding efficiencies for longer data sequences.

18. Differentiate lossy source coding from lossless source coding

S.no	Lossy source coding	Lossless source coding
1	Some information of the source is lost during encoding.	No information is lost during encoding
2	PCM,DM,ADM,DPCM are lossy source coding technique.	Huffman coding, instantaneous coding ,Shannon Fano coding are lossless source coding technique.

PART B

1. Enumerate Shannon's Fano algorithm and Huffman coding with a suitable example.
2. Five symbols of the alphabet of discrete memory less source and their probabilities are given below, S={S0, S1,S2,S3,S4) P(S)={0.4,0.19,0.16,0.15,0.15}. Predict the symbols using Huffman coding and calculate the average codeword length and efficiency

3. Illustrate the following with equations

   (i) Uncertainity

   (ii) Information

   (iii) Entropy and it's properties

4. i) Infer Hamming codes. Analyse the conditions which hamming codes has to satisfy.

   (ii) Examine the following terms - Code efficiency, Channel data rate and code rate.

5. Five symbols of the alphabet of discrete memory less source and their probabilities are given below, S={S0, S1,S2,S3,S4) P(S)={0.4,0.19,0.16,0.15,0.15}. Point out the symbols using Shannon Fano coding and calculate the average codeword length and efficiency

6. (i) Summarize Source Coding with block diagram and mention it's functional requirements.

   (ii) Deduce the equations for average codeword length and coding efficiency using entropy.

7. (i) Give the main idea of discrete memoryless channel and it's matrix form involving transition probabilities.

   (ii) Relate the concept of Binary symmetric channel with Binary communication channel & Binary erasure channel

8. Interpret the following

   (i) Mutual information and its properties.

   (ii) Channel capacity and it's equation.

8. Five symbols of the alphabet of discrete memory less source and their probabilities are given below, S={S0, S1,S2,S3,S4) P(S)={0.4,0.2,0.2,0.1,0.1}. Construct the symbols using Huffman coding and calculate the average codeword length and efficiency.

9. (i) Brief the properties of entropy.

   (ii) Describe the BSC and BEC with their channel diagram and transition matrix.

10. A telephone channel has a bandwidth of 3 kHz .

    (i) Predict channel capacity of the telephone channel for a SNR of 20 dB

    (ii) Estimate minimum SNR required to support a rate of 5 kbps.

11. Reproduceshannon's 3 laws that govern the Information theory

12. A telephone channel has a bandwidth of 3 kHz and output SNR of 20 dB. The source has a total of 512 symbols and the occurrence of all symbols are equiprobable. Point out the following

(i) Channel capacity

(ii) Information content per symbol.

(iii) Maximum symbol rate for which error free transmission is possible.

PART C

1. The source of information A generates the symbols {A0, A1, A2, A3& A4} with the corresponding probabilities {0.4, 0.3, 0.15, 0.1 and 0.05}. Evaluate the code for source symbols using Huffman and Shannon-Fano encoder and compare its efficiency.
2. Draw the block diagram of Digital Communication system and Construct each of it's components.
3. Propose the following with suitable diagrams and equations.

(i) Discrete memoryless source

(ii) Discrete memoryless channel

4. The source of information A generates the symbols {A0, A1, A2, A3 , A4 , A5} with the corresponding probabilities {0.45, 0.41, 0.4, 0.3 , 0.29 and 0.05}. Evaluate the code for source symbols using Huffman and Shannon-Fano encoder and compare its efficiency.

UNIT II WAVEFORM CODING & REPRESENTATION

1.What is meant by temporal waveform coding

The signal which varying with time can be digitzed by periodic time sampling and amplitude quantization.this process is called temporal waveform coding .DM,ADM,DPCM are example of temporal waveform coding.

2. Differentiate the principle of temporal waveform coding and model based coding.

Temporal Waveform Coding

The signal which varying with time can be digitized by periodic time sampling and amplitude quantization. This process is called temporal waveform coding .DM,ADM,DPCM are example of temporal waveform coding.

Model Based Coding

The signal is characterized in various parameter. This parameter represent the model of the signal.LPC is an example model based coding.

3. What is meant by DPSK?

In DPSK, the input sequence is modified. Let input sequence be d(t) and output sequence be b(t). Sequence b(t) changes level at the beginning of each interval in which d(t)=1 and it does not changes level when d(t)=0.

When b(t) changes level, phase of the carrier is changed. And as stated above, b(t) changes t=its level only when d(t) =1. This means phase of the carrier is changed only if d(t)=1. Hence the technique is called Differential PSK.

4. Mention the merits of DPCM.

1 .Bandwidth requirement of DPCM is less compared to PCM.

2. .Quantization error is reduced because of prediction filter

3. .Numbers .of bits used to represent .one sample .value are also reduced compared to PCM.

5. What is the main difference in DPCM and DM?

DM encodes the input sample by one bit. It sends the information about + δ or -δ, ie step rise or fall. DPCM can have more than one bit of encoding the sample. It sends the information about difference .between .actual sample value and the predicted sample value.

6. **What is the advantage of delta modulation over pulse modulation schemes?**

Delta modulation encodes one bit per samples. Hence signalling rate is reduced in DM.

7. What is meant by adaptive delta modulation?

In adaptive delta modulation, the step size is adjusted as per the slope of the input signal. Step size is made high if slope of the input signal is high. This avoids slope overload distortion.

8. What are the two limitations of delta modulation?

1 Slope of overload distortion.

2. Granular noise.

9. How does Granular noise occurs?

It occurs due to large step size and very small amplitude variation in the input signal.

10. What are the advantages of the Delta modulation?

- Delta modulation transmits only one bit for one sample. Thus the signalling rate and transmission channel bandwidth is quite small for delta modulation.
- The transmitter .and .receiver .implementation is very much simple for delta modulation. There is no analog to digital converter involved in delta modulation.

11. What do you understand from adaptive coding?

In adaptive coding, the quantization step size and prediction filter coefficients are changed as per properties of input signal. This reduces the quantization error and number of bits to represent the sample value. Adaptive coding is used for speech coding at low bits rates.

12. Mention the use of adaptive quantizer in adaptive digital waveform cod ing schemes.

Adaptive quantizer changes its .step .size .according variance of .the .input signal. Hence quantization error is significantly reduced due to the adaptive quantization. ADPCM uses adaptive quantization. The bit rate of such schemes is reduced due to adaptive quantization.

13. What is PAM?

PAM is the pulse amplitude modulation. In pulse amplitude modulation, the amplitude of a carrier consisting of a periodic train of rectangular pulses is varied in proportion to sample values of a message signal.

14. What is the need for speech coding at low bit rates?

The use of PCM at the standard rate of 64 Kbps demands a high channel bandwidth for its transmission ,so for certain applications, bandwidth is at premium, in which case there is a definite need for speech coding at low bit rates, while maintaining acceptable fidelity or quality of reproduction.

15.Define ADPCM.(Oct-98)

It means adaptive differential pulse code modulation, a combination of adaptive quantization and adaptive prediction. Adaptive quantization refers to a quantizer that operates with a time varying step size. The autocorrelation function and power spectral density of speech signals are time varying functions of the respective variables. Predictors for such input should be time varying. So adaptive predictors are used.

16.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.

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

- Side information
- Buffering
- Delay

18. How are the predictor coefficients determined?

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

19. Define adaptive sub band coding? (Nov-97)

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.

20.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.

21.What is the bit rate in ASBC?

Nfs= (MN) (fs/M) Nfs->bit rate

Where , M-number of sub bands of equal bandwidths

N-average number of bits

Fs/M-sampling rate for each sub band.

22. Define Adaptive filter.

It is a nonlinear estimator that provides an estimate of some desired response without requiring knowledge of correlation functions, where the filter coefficients are data dependent. A popular filtering algorithm is the LMS algorithm.

23. 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.

Part A

1. Define prediction error.
2. State the 2 properties of linear prediction.
3. Differentiate PCM and DPCM.
4. What is prediction gain? State its significance.
5. What is delta modulation?
6. Mention the drawbacks of DM.
7. A speech signal with maximum frequency of 3.4 KHz and maximum amplitude of 1 V. this speech signal is applied to a delta modulator whose bit rate is set at 20 Kbps. Discuss the choice of an appropriate step size for the delta modulator.
8. The idle channel noise in a delta modulator is negligibly small. Justify the validity of this statement.
9. What is slope overload and granular noise?
10. State the principle of ADM.
11. Compare DM and ADM.
12. What are the objectives of speech coding?
13. Define ADPCM.
14. What are the different types of adaptive quantization?
15. Define APF.
16. Define APB.
17. What is the principle of linear predictive coder (LPC)?
18. Draw the model of LPC.
19. Mention the applications of LPC.
20. What is adaptive sub-band coding?

Part B

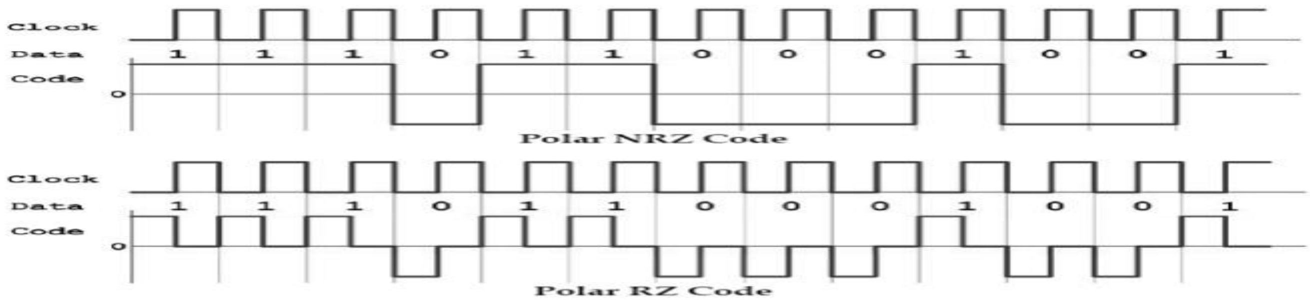
1) What is DM? Explain the transmitter and receiver of DM system. (16)
2) Explain a DPCM system with the expressions and block diagram. Show that SNR of DPCM is better than that of PCM. (16)
3) Explain the noises in delta modulation systems. How to overcome this effect in Delta modulation? (16)
4) Describe temporal and spectral waveform encoding methods. (16)
5) With necessary diagrams, explain ADPCM system. (16)
6) Write short notes on
   (i) Adaptive quantization schemes
   (ii) Adaptive prediction schemes.
7) What is low bit rate speech coding? Draw the block diagram of adaptive sub-band coding scheme for speech signal and explain. (16)
8) Discuss about the structure of linear predictor. Also explain the process of prediction error. (16)
9) Explain the principle of LPC model with diagrams. (16)
10) Compare the various types of speech encoding techniques. (16)

UNIT III BASE BAND TRANSMISSION & RECEPTION

1. What is meant by transparency with respect to line codes

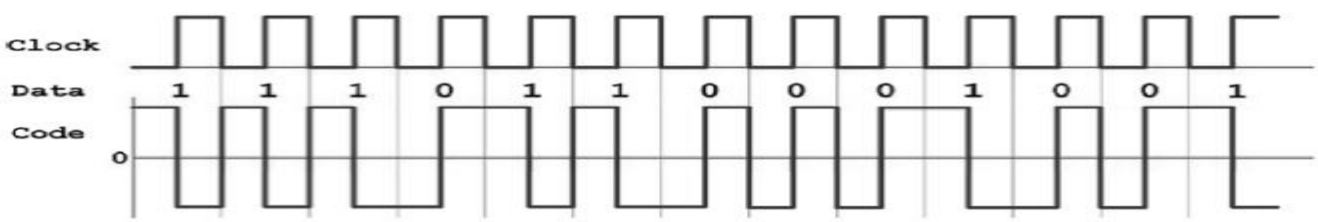
The line code is said to be transparent if the synchronization between the transmitter and receiver is maintained for any type of input data sequence.

2. Draw the NRZ and RZ code for the digital data 10110001 [OR] Draw the RZ bipolar line code format for the information {10110}



3. What is Manchester code? Draw the Manchester format for the data stream 10110?

In Manchester code each bit of data is signified by at least one transition. Manchester encoding is therefore considered to be self-clocking, which means that accurate clock recovery from a data stream is possible. In addition, the DC component of the encoded signal is zero. Although transitions allow the signal to be self-clocking, it carries significant overhead as there is a need for essentially twice the bandwidth of a simple NRZ or NRZI encoding



4. State any four desirable properties of line code

- The PAM signal should have adequate timing content,
- The PAM signal should immune to channel noise and interference
- The PAM signal should allow error detection and error correction
- The PAM signal should be transparent to digital data being transmitted.

5. What is intersymbol interference in baseband binary PAM systems?

In baseband binary PAM, symbols are transmitted one after another. These symbols are separated by sufficient time durations. The transmitter, channel and receiver acts as a filter to this baseband data. Because of the filtering characteristics, transmitted PAM pulses are spread in time.

6. What is correlative coding?

Correlative level coding is used to transmit a baseband signal with the signalling 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 .signalling .and modified duobinary signalling.

7. Define Duobinary baseband PAM system K

Duobinary encoding reduces the maximum frequency of the baseband signal. The word „duo‟ means to double the transmission capacity of the binary system. Let the PAM signal ak represents kth bit. Then the encoder the new waveform as C*k* =a*k* + ak-1

8. What are eye pattern?

Eye pattern is used to study the effect of ISI in baseband transmission.

- Width of .eye .opening defines the .interval over .which the .received wave can be sampled without error from ISI.
- The sensitivity of the system to timing error is determined by the rate of closure of the eye as the sampling time is varied.
- Height of the eye opening at sampling time is called margin over noise.

9. How is eye pattern obtained on the CRO?

Eye pattern can be obtained on CRO by applying the signal to one of the input channels and given an external trigger of 1/Tb Hz. This makes one sweep of beam equal to Tb seconds

.

10. Why do you need adaptive equalization in a switched telephone network.

In switched telephone network the distortion depends upon

1) Transmission characteristics of individual links.

2) Number of links in connection.

Hence fixed pair of transmit and receive filters will not serve the equalization problem. The transmission characteristics keep on changing. Therefore adaptive equalization is used.

11. What are the necessity of adaptive equalization?

Most of the channels are made up of individual links in switched telephone network,the distortion induced depends upon 1) Transmission characteristics of individual links 2) number of links in connection

12. Define the principle of adaptive equalization?

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

13. Define the term ISI?

The presence of outputs due to other bits interference with the output of required bit . This effect is called inter symbol interference (ISI).

14. Write the performance of data transmission system using eye pattern technique?

The width of the eye opening defines .the interval over which the received wave can can be .sampled without error from inter symbol interference .

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

15. What is the necessity of equalization?

When the signal is passed through the channel distortion is introduced in terms of 1) amplitude 2) delay this distortion creates problem of ISI. The detection of the signal also become difficult this distraction can be compensated with the help of equalizer.

16. What is raised cosine spectrum?

In the raised cosine spectrum, the frequency response P(f) decreases towards zero gradually That is there is no abrupt transition.

17. What is nyquist Bandwidth?

The B.is called nyquist bandwidth. .The nyquist bandwidth is the minimum transmission bandwidth for zero ISI.

18. Give two applications for Eye pattern.

- To determine an interval over which the received wave van be sampled without error due ot ISI.·
- To determine the sensitivity of the system to timing error

19.What are the information that can be obtained from eye pattern regarding the signal quality?

- To determine an interval over which the received wave van be sampled without error due to ISI.
- To determine the sensitivity of the system to timing error
- The margin over the noise is determined from eye pattern

20. A 64 kbps binary PCM polar NRZ signal is passed through a communication system with a raised-cosine filter with roll-off factor 0.25. Find the bandwidth of a filtered PCM signal.

Fb=64lpbs

B0=FB/2=32kpbs

α=0.25

B=B0(1+α)=30*103(1+0.25)=40kHz

<u>Part – A</u>

1. What is line coding?
2. Define transparency of a line code. Give two examples of line codes which are not transparent.
3. State any 4 properties of a line code.
4. What is Manchester code? Draw its format for the data 10011.

5. Draw the RZ-Bipolar line code for the data 110100.
6. Draw the NRZ and RZ code for the digital data 10110001.
7. What are the requirements of a line code?
8. What is ISI?
9. ‘ISI can-not be avoided’. Justify the statement.
10. How does pulse shaping reduce ISI?
11. Draw the ideal and basic amplitude response of pulse waveforms.
12. Define roll-off factor.
13. State the Nyquist criterion for zero ISI.
14. What is Eye pattern? State any 2 applications of eye pattern..
15. What is equalization?
16. What is correlative coding?
17. Define duo binary system. What are the drawbacks of it?
18. Draw the block diagram of adaptive equalizer.
19. What are the methods used to implement adaptive equalizer?
20. Why do we need equalization filter?

Part B

1) Derive the power spectral density of polar RZ code and explain. (16)
2) Derive the expression for power spectral density of unipolar NRZ line code. Hence discuss its characteristics. (16)
3) (i) List and explain the properties of line codes. (8)

   (ii) Derive the power spectral density of Manchester code. (8)
4) Explain modified duo-binary signaling scheme with & without procedure. (16)
5) Explain how ISI occurs in base-band binary data transmission system. (16)
6) Describe the Nyquist criterion method for distortion less transmission. (16)
7) The Fourier transform P(f) of the basis pulse p(t) employed in a certain binary communication system is given by

$$P(f) = \begin{cases} 10^{-6}\left(1 - \dfrac{|f|}{10^{6}}\right) & if \quad 10^{-6} \leq f(Hz) \leq 10^{6} \\ 0 & Otherwise \end{cases}$$

(a) From the shape of P(f), explain whether this pulse satisfies the Nyquist criterion for ISI free transmission.
(b) Determine p(t) and verify your result in part a.
(c) If the pulse does satisfy the Nyquist criterion. What is the transmission rate (in bits/sec.) and what is the roll-off factor?

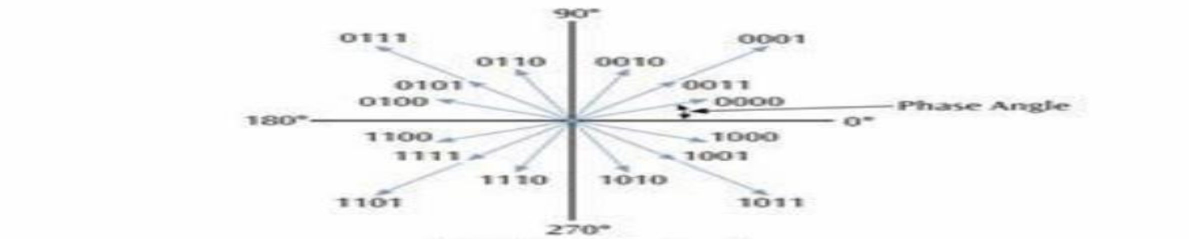
8) Explain the pulse shaping method to minimize ISI. (16)

9) Draw and explain the block diagram of duo-binary signaling scheme for controlled ISI. (16)

10) Briefly discuss about
(i) Eye pattern. (8)
(ii) Adaptive equalization. (8)

UNIT IV DIGITAL MODULATION SCHEME

1.Define QAM and draw its constellation diagram. ?



(a) Twelve Phase Angles

(b) A Phase Change with Two Amplitudes

2. A binary frequency shift keying system employs two signaling frequencies 1f and 2f. The lower frequency 1f is 1200 Hz and signaling rate is 500 Baud. Calculate 2f. ?

For binary FSK baud=fb

Fb=500Hz

Consider the FN modulation index(h) of 1 in FSK

Fm-fs/Fb =h=1

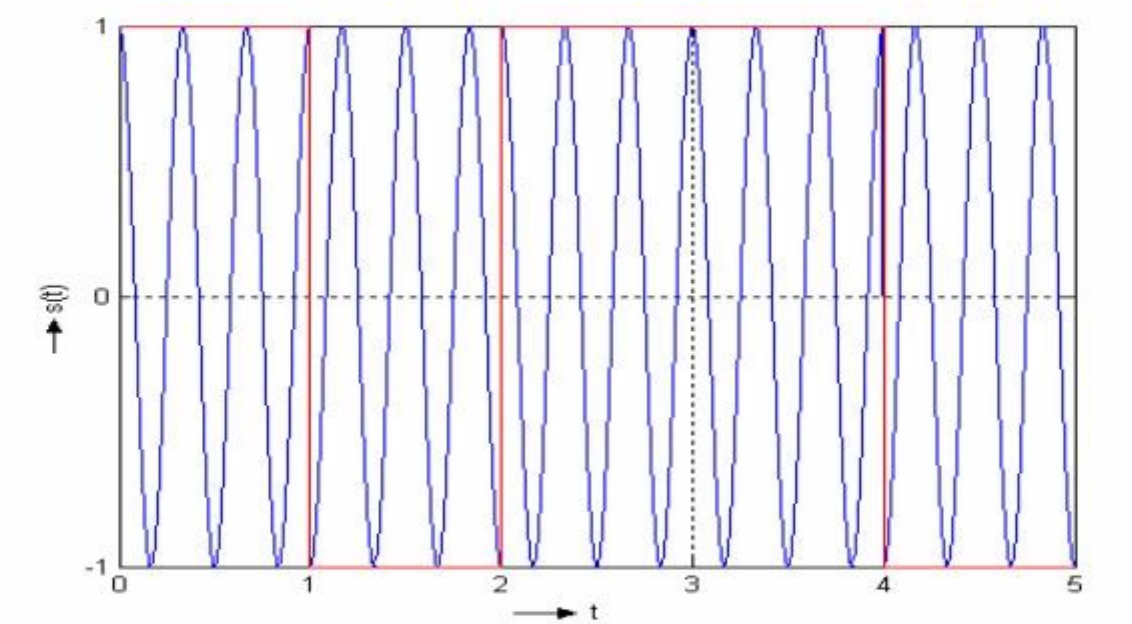
Fm-fs=fb

Fs=f1=1200HZ

Fm-1200Hz=500Hz

Fm=1700Hz,f2=fm=1700Hz.

3. Draw the PSK waveform for 011011.



4. What is meant by coherent detection system?

In coherent ASK, correlation receiver is used to detect the signal. Locally generated carrier is correlated with incoming Ask signal. The locally generated carrier is in exact phase either transmitted carrier. Coherent Ask is also called synchronous ASK.

5. Why is PSK always preferable over ASK in coherent detection?

ASK is on-off signaling where as the modulated carrier is continuously transmitted in PSK. Hence peak power requirement is more in ASK, where it is reduces in PSK.

6. Differentiate between coherent and non-coherent detection

In coherent detection the local carrier generated at the receiver is phase locked with the carrier at the transmitter. Hence it is also called synchronous detection. In non coherent detection the local carrier generated at the receiver not be phase locked with the carrier at the transmitter. It is simple, but it has higher probability of error.

7. What are the drawbacks of binary PSK system?

It is difficult to detect +b(t) and –b(t) because of squaring in the receiver Problem, of ISI and inter channel interference are present.

8. A BPSK system makes errors at the average rate of 1000 errors per delay. Data rate is 1 kbps . The single-sided noise power spectral density is 10-20 W/Hz. Assuming the system to be wide sense stationary,

what is the average bit error probability?

24*60*60=86400sec $86.4*10^{6}$

Bir error probability Pe=100/$86.4*10^{6}$ =$1.1157*10^{-6}$

9. What is meant by DPSK?

In DPSK, the input sequence is modified. Let input sequence be d(t) and output Sequence be b(t). Sequence b(t) changes level at the beginning of each interval in which d(t)=1 and it does not changes level when d(t)=0.

When b(t) changes level, phase of the carrier is changed. And as stated above, b(t) changes t=its level only when d(t) =1. This means phase of the carrier is changed only if d(t)=1. Hence the technique is called Differential PSK.

1**1. Bring out the difference between coherent & non coherent binary modulation scheme.**

a. Coherent detection:

In this method the local carrier generated at the receiver is phase locked with the carrier At the transmitter. Hence it is called synchronous detection

b. Non coherent detection:

In this method, the receiver carrier need not be phase locked with transmitter carrier. Hence it is called envelope detection.

12. Write the expression for bit error rate for coherent binary FSK.

Bit error rate for coherent binary FSK is given as,

$$Pe = 1/2\text{erfc}\sqrt{0.6E/No}$$

13. What is Signal constellation diagram?

Suppose that in each time slot of duration T seconds, one $s_2(t)$, . . $s_M(t)$ is transmitted with equal probability, 1/M For geometric representation, the signal s_i (t), = 1, 2, ..., M, is applied to a bank of correlators. The correlator outputs define the signal vector si. The set of message points corresponding to the set of transmitted signals $\{s_i(t))\}$ i=1..M is called a signal constellation.

14. What is meant by memory less modulations?

When the digital symbol modulates amplitude, phase or frequency of the carrier without any reference to previous symbol, it is called memory less modulations. Eg.:ASK,PSK,FSK,QPSK etc.

15.Define QPSK.

In QPSK two successive bits in the data sequence are grouped together.This combination of two bits forms four distinct symbols.When the symbol is changed to next symbol the phase of the carrier is changed by 45^0(or $\pi/4$). Because of combination of two bits there will be four symbols.Hence the phase shift will be $\pi/4$, $3\pi/4$, $5\pi/4$ or $7\pi/4$.

16.**List the advantages of Pass band transmission**.

- Long distance.
- Analog channels can be used for transmission.
- Multiplexing techniques can be used for bandwidth conservation.
- Transmission can be done by using wireless channel also.

17. List the requirements of Pass band transmission.

- Maximum data transmission rate.
- Minimum probability of symbol error.
- Minimum transmitted power.

18.Mention the need of optimum transmitting and receiving filter in baseband data transmission.(Madras Univ, April97, Nov-97)

When binary data is transmitted over the baseband channel, noise interfaces with it. Because of this noise interference, errors are introduced in signal detection. Optimum filter performs two functions while receiving the noisy signal:

- Optimum filter integrates the signal during the bit interval and checks the output at the time instant where signal to noise ratio is maximum
- Transfer function of the optimum filter is selected so as to maximize signal to noise ratio.
- Optimum filter minimizes the probability of error.

19. Define ASK.(Madras Univ, April-97,98)

In ASK, carrier is switched on when binary 1 is to be transmitted and it is switched off when binary D is to be transmitted ASK is also called on-off keying.

20. What is meant by coherent ASK? (Madras Univ, Oct-98)

In coherent ASK, correlation receiver is used to detect the signal. Locally generated carrier is correlated with incoming ASK signal. The locally generated carrier is in exact phase with the transmitted carrier. Coherent ASK is also called as synchronous ASK.

Part A

1. What is the need for geometric representation of signals?
2. Why we go for Gram- Schmidt orthogonalization procedure?
3. Define BPSK and DPSK.
4. Why is PSK always preferable over ASK in Coherent detection?
5. What are the drawbacks of binary PSK system?

6. A BFSK system employs two signaling frequencies f_1 and f_2. The lower frequency f_1 is 1200 Hz and signaling rate is 500 Baud. Calculate f_2.
7. What are the advantages of QPSK over PSK?
8. What is constellation diagram?
9. A BPSK system makes errors at the average rate of 100 errors per day. Data rate is 1 Kbps. The single-sided noise power spectral density is 10 W/Hz. Assume the system to be wide sense stationary, what is the average bit error probability?
10. Define QAM and draw its constellation diagram for M=8.
11. Write the special features of QAM.
12. Differentiate coherent and non-coherent detection.
13. Define spectral efficiency.
14. What is meant by symbol synchronization?
15. List out the difference between carrier recovery and clock recovery.
16. Compare the error probability for BPSK and QPSK.
17. What is the error probability of DPSK?
18. Write the features of DPSK.
19. What is meant by memoryless modulation?
20. Compare BER and SER

<u>Part B</u>

1. Describe with diagrams the generation and detection of coherent BFSK. Explain the probability of error for this scheme. (16)

2. Explain non coherent detection methods of binary frequency shift keying scheme. (16)

3. Explain the generation and detection of binary PSK. Also derive the probability of error for PSK. (16)

4. Compare the performance of various coherent and non-coherent digital detection systems. (16)

5. Discuss about coherent detection of QPSK and derive its power spectral density. (16)

6. With constellation diagram, explain the QAM transmitter. Also derive its power spectral density. (16)

7. A set of binary data is sent at the rate of R_b = 100 Kbps over a channel with 60 dB transmission loss and power spectral density $\eta\eta = 10^{-12}$ W/Hz at the receiver. Determine the transmitted power for a bit error probability $P_e = 10^{-3}$ for the following modulation schemes.
   (i) FSK (iii) PSK
   (ii) DPSK (iv) 16 QAM

8. Explain the carrier synchronization methods with block diagrams. (16)
9. Briefly discuss about the Non-coherent detection of PSK and QPSK. (16)
10. Briefly discuss about the principle of DPSK system. (16)

UNIT V ERROR CONTROL CODING

1. Mention is the properties of cyclic codes

Linearity property

The sum of any two code word is also a valid code word

Cyclic property

Every cyclic shift of a valid code vector produces another valid code vector.

2. Define 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 words be,

X = (101) and Y= (110) These two code words differ in second and third bits. Therefore the hamming distance between X and Y is two.

3. What is meant by transparency with respect to line codes

The line code is said to be transparent if the synchronization between the transmitter and receiver is maintained for any type of input data sequence.

4. Define hamming distance and calculate its value for two code words 11100 and 11011

The hamming distance between two code vectors is equal to the number of elements in which they differ. For example, let the two code words be,

X = (11100) and Y= (11011)

D= 2 These two code words differ in second and third bits. Therefore the hamming distance between X and Y is two.

5. What is convolution code? How is it different from block codes?

Fixed number of input bits is stored in the shift register & they are combined with the help of mod 2 adders. This operation is equivalent to binary convolution coding.

6. State any four desirable properties of line code

- The PAM signal should have adequate timing content,
- The PAM signal should immune to channel noise and interference
- The PAM signal should allow error detection and error correction
- The PAM signal should be transparent to digital data being transmitted

7. Find the hamming distance 101010 and 010101.If the minimum hamming distance of a (n,k) linear block code is 3, what is its minimum hamming weight?

$d(x_1,x_2)=x_1$ exor x_2 =111111

$d(x_1,x_2)=6$ Dmin=3 then Wmin=dmin=3

8. What is meant by syndrome of linear block code?

The non zero output of the produce YHT is called syndrome & it is used to detect errors in y. Syndrome is denoted by S & given as,

S=YHT

9. What is convolutional code? Explain the fundamental difference between block codes and convolutional codes.

Block codes takes"k" number of bits simultaneously form „n"-bit code vector. This code vector is also called block. Convolutional code takes one message bits at a time and generates two or more encoded bits. Thus convolutional codes generate a string of encoded bits for input message string.

10. 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 words be,

X = (101) and Y= (110)

These .two code words differ in second and third bits.Therefore .the .hamming distance between X and Y is two.

11. Define code efficiency.

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

Code efficiency= (k/n)

k=message bits

n=transmitted bits.

12. 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.

13. What is meant by linear code?

A code is linear if modulo-2 sum of any two code vectors produces another code vector. This means any code vector can be expressed as linear combination of other code vectors.

14. What is meant by cyclic codes?

Cyclic codes are the subclasses of linear block codes. They have the property that a cyclic shift of one codeword produces another code word.

15. How syndrome is calculated in Hamming codes and cyclic codes?

In hamming codes the syndrome is calculated as,

$$S=YH^T$$

Here Y is the received and H.is the transpose of parity check matrix

16. What is difference between block codes and convolutional codes?

Block codes takes‟k‟ .number of bits simultaneously form „n‟-bit .code vector. This code vector is also called block. Convolutional code takes one message bits at a time and generates two or more encoded bits. Thus convolutional codes generate a string of encoded bits for input message string.

17. What are the classifications if line codes?

Line code is classified as

a) Polar b) Unipolar. C) Bipolar

17.What is Manchester code? (May-12)

In the Manchester code binary ’1’is encoded by positive half pulse followed by negative pulse. And binary ‘0’
is encoded by negative half pulse followed by positive pulse.

18. State any desirable properties of a line code. (Dec-12)

- The PAM signal should have adequate timing content, so that clock information can be executed from the waveform.
- The PAM signal should be immune to channel noise and interference.
- The PAM signal should allow error detection and correction.

19. What 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.

20. 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= (Eb/No) encoded

(Eb/No) coded

For Convolutional coding, the coding gain is given as,

A = rdf /2

Here „r" is the code rate

And „df is the free distance.

21. What is Vitterbi decoding scheme?

It performs maximum likelihood decoding and it reduces the computational load by taking advantages in code trellis. Decoding is done with algorithm.

22. What are the limitations of Vitterbi decoding?

It can correct up to 2 errors. A triple error pattern is un correctable by the Vitterbi algorithm. Constraint length increases complexity also increases exponentially.

i) The error probability decreases easily

ii) Decoding procedure is ¾ of constraint length.

<u>Part A</u>

1. State the channel coding theorem.
2. What are the objectives of channel coding?
3. Define coding efficiency.
4. Define Hamming distance and calculate its value for two code words 11100 and 11011.
5. Define Hamming weight and minimum distance.
6. State the significance of minimum distance of a block code.
7. State the principle of error free communication.
8. Define linear block codes.
9. Write syndrome properties of linear block codes.
10. What is Hamming codes?
11. Write the advantages and disadvantages of Hamming codes.
12. Define syndrome vector.
13. Mention the properties of cyclic code.
14. State any 2 properties of generator polynomial.
15. What are the advantages and disadvantages of cyclic code?
16. What is convolutional code? How is it different from block codes?
17. Mention the structural properties of a convolutional encoder.
18. What is meant by BCH code?
19. Define CRC codes.
20. What is Viterbi decoding scheme?

Part B

1. Construct a single error correcting (7, 4) linear block code and the corresponding decoding table. (16)
2. (i) Explain the generation of (n, k) blocks codes and how block codes can be used for error control. (10)
   (ii) Explain the syndrome decoder for cyclic codes. (6)
3. Consider a (7, 4) linear block code whose parity check matrix is given by

$$H = \begin{matrix} 1 & 1 & 1 & 0 & 1 & 0 & 0 \\ 1 & 1 & 0 & 1 & 0 & 1 & 0 \\ 1 & 0 & 1 & 1 & 0 & 0 & 1 \end{matrix}$$

   (i) Find the generator matrix.
   (ii) How many errors this code can detect?
   (iii) How many errors can this code be corrected?
   (iv) Draw circuit for encoder and syndrome computation. (16)
4. The generator polynomial of a (7, 4) Hamming code is defined by

$$g(D) = 1 + D^2 + D^3$$

   Develop the encoder and syndrome calculator for this code. (16)
5. (i) Find a generator polynomial for a (7, 4) cyclic code and hence find the code word for [1 1 0 0]. (8)
   (ii) Construct the encoder for (7, 4) cyclic codes. (8)
6. Explain how encoding is done by convolutional codes with an example. (16)
7. For (6, 3) systematic linear block code, the code word comprises I_1, I_2, I_3 and P_1, P_2, P_3 where the three parity check bits are formed from the information bits as follows:

$$P_1 = I_1 \quad I_2$$
$$P_2 = I_1 \quad I_3$$
$$P_3 = I_2 \quad I_3$$

   Find: (i) Parity check matrix and generator matrix (3)
   (ii) All possible code words. (3)
   (iii) Minimum weight and minimum distance. (3)
   (iv) Error detecting and correcting capability of the code. (3)
   (v) If the received sequence is 101010, calculate the syndrome and decode the received sequence. (4)
8. Describe the steps involved in the generation of linear block codes. Define and explain properties of syndrome. (16)
9. Explain Viterbi algorithm to decode a convolutionally coded message.(16)

Design a convolutional coder of constraint length 6 and rate efficiency 1/2. Draw its tree diagram and trellis diagram.