SYED AMMAL ENGINEERING COLLEGE

(Approved by the AICTE, New Delhi, Govt. of Tamilnadu and Affiliated to Anna University, Chen

Established in 1998 - An ISO 9001:2008 Certified Institution

Dr. E.M.Abdullah Campus, Lanthai, Ramanathapuram – 623 502.

Phone: 304000, 304222, 304101 (04567)

Web: www.syedengg.ac.in, E.mail:saec@syedengg.ac.in

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

EC6501 -DIGITAL COMMUNICATION

UNIT-I

SAMPLING & QUANTIZATION

1. Define Dirac comb or ideal sampling function. What is its Fourier Transform?

Dirac comb is nothing but a periodic impulse train in which the impulses are spaced by a time interval of Ts seconds. The equation for the function is given

by

$$\delta_{Ts}(t) = \sum_{n=-\infty}^{\infty} \delta(t - nTs)$$

The Fourier Transform of $\delta_{Ts}(t)$ is given by

$$F[\delta_{Ts}(t)] = f_s \sum_{m=-\infty}^{\infty} \delta(f - m f_s)$$

2. Give the interpolation formula for the reconstruction of the original signal.

The interpolation formula for the reconstruction of the original signal g(t) from the sequence of sample values {g(n/2W)}.

$$g(t) = \sum_{n=-\infty}^{\infty} g(n/2W)\ \mathrm{sinc}(2Wt - n)$$

where 2W is the bandwidth

n is the number of samples.

3. State sampling theorem. (Madras Univ., Nov-97,Oct-98,Dec-06,08,09,May-07,09,12)

- 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.
- A band limited signal of finite energy, which has no frequency components higher than W Hz, may be completely recovered from the knowledge of its samples taken at the rate of 2W samples per second.

4. Define quadrature sampling.

Quadrature sampling is used for uniform sampling of band pass signals.

Consider $g(t) = g_I(t)\cos(2\Pi f_c t) - g_Q(t)\sin(2\Pi f_c t)$.

5. Define Nyquist rate. (Madras Univ, April-97)

Let the signal be band limited to W Hz. Then Nyquist rate is given as,

Nyquist rate = 2W samples/sec

Aliasing will not take place if sampling rate is greater than Nyquist rate.

- A band limited signal of finite energy, which has no frequency components higher than W Hz, may be completely recovered from the knowledge of its samples taken at the rate of 2W samples per second.

7. What is meant by PCM?

Pulse code modulation (PCM) is a method of signal coding in which the message signal is sampled, the amplitude of each sample is rounded off to the nearest one of a finite set of discrete levels and encoded so that both time and amplitude are represented in discrete form.. This allows the message to be transmitted by means of a digital waveform.

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

1. The peak-to-peak range of input sample values subdivided into a finite set of decision levels or decision thresholds
2. 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.

9. What is meant by idle channel noise?

Idle channel noise is the coding noise measured at the receiver output with zero transmitter input.

10. What is meant by 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.

11. Define delta modulation.

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

12. 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).

13. Name the types of uniform quantizer?

1. Mid tread type quantizer.
2. mid riser type quantizer.

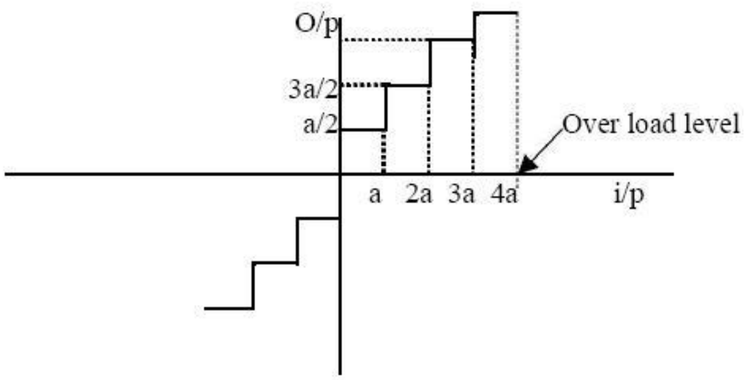
14. Define mid tread quantizer?

Origin of the signal lies in the middle of a tread of the staircase.

3a

15. Define mid-riser quantizer?

Origin of the signal lies in the middle of a rise of the staircase



16. What is meant by quantization? (May-12)

While converting the signal value from analog to digital, quantization is performed. The analog value is assigned to nearest digital value. This is called quantization. The quantized value is then converted into equivalent binary value. The quantization levels are fixed depending upon the number of bits. Quantization is performed in every Analog to Digital Conversion.

17. The signal to quantization noise ratio in a PCM system depends on what criteria? (MAY-06)

The signal to quantization noise ratio in PCM is given as,

$$(S/N)\ db \leq (4.8+6v)\text{dB}$$

Here v is the number of bits used to represent samples in PCM. Hence signal to quantization noise ratio in PCM depends upon the number of bits or quantization levels.

18. Define quantization error? (May-07)

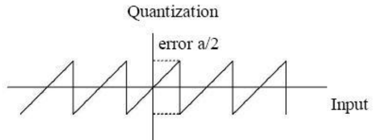
Quantization error is the difference between the output and input values of quantizer.

19. What you mean by non-uniform quantization? (May-08)

Step size is not uniform. Non-uniform quantizer is characterized by a step size that increases as the separation from the origin of the transfer characteristics is increased. Non-uniform quantization is otherwise called as robust quantization. .

20. Draw the quantization error for the mid tread and mid-rise type of quantizer?

For mid tread type:



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21. What is the disadvantage of uniform quantization over the non-uniform quantization?

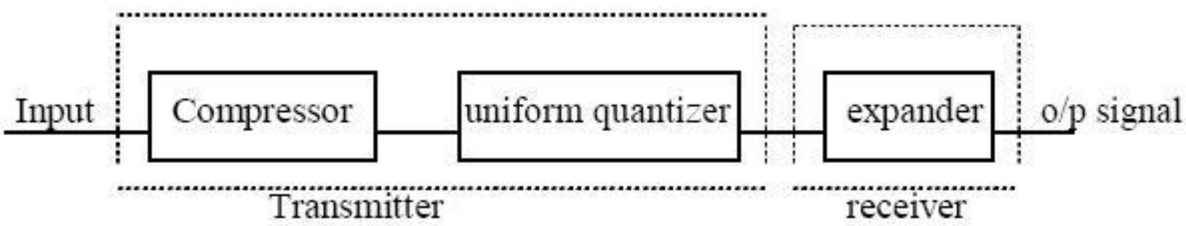
SNR decreases with decrease in input power level at the uniform quantizer but non-uniform quantization maintains a constant SNR for wide range of input power levels. This type of quantization is called as robust quantization.

22. What do you mean by companding?

The signal is compressed at the transmitter and expanded at the receiver. This is called as companding.
The combination of a compressor and expander is called a compander.

23. Draw the block diagram of compander? Mention the types of

companding? Block diagram:



Types of companding:

1. A-law companding.
2. µ-law companding.

UNIT-II

WAVEFORM CODING

1. 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.

2. 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.

3. 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.

4. 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 theprediction error are used to derive estimates of the predictor coefficients.

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

- Side information
- Buffering
- Delay

6. How are the predictor coefficients determined?

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

7. 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.

8. 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.

9. 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.

10. 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.

11. 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.

12. 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.

13. 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.

14. How the message can be recovered from PAM?

The message can be recovered from PAM by passing the PAM signal through reconstruction filter integrates amplitude of PAM pulses. Amplitude reconstruction signal is done to remove amplitude discontinuities due to pulses.

15. Write an expression for bandwidth of binary PCM with N messages each with a maximum frequency of *fm* Hz.

If „v‟ number of bits are used to code each input sample, then bandwidth of PCM is given as,

$BT \geq N.v.fm$ .Here v. *fm* is the bandwidth required by one message.

16. How is PDM wave converted into PPM message?

The PDM is signal is clock signal to monostable multivibraor. The multivibraor triggers on falling edge. Hence a PPM pulse of fixed width is produced after falling edge of PDM pulse. PDM represents the input signal amplitude in the form of width of the pulse. A PPM pulse is produced after the width of PDM pulse. In other words,the position of the PPM pulse depends upon input signal amplitude.

17. Mention the use of adaptive quantizer in adaptive digital waveform coding schemes.

Adaptive quantizers change 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.

18. 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.

19. 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.

20. What is the advantage of delta modulation over pulse modulation schemes? Delta modulation encodes one bit per samples. Hence signaling rate is reduced in DM.

21. What is the advantage of delta modulation over PCM?

Delta modulation uses one bit to encode on sample. Hence bit rate of delta modulation is low compared to PCM.

22. What are the two limitations of delta modulation?

1. Slope of overload distortion.
2. Granular noise.

23. How does Granular noise occurs?

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

24. What are the advantages of the Delta modulation?

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

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UNIT III

BASEBAND TRANSMISSION

1. What is inter symbol 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.

2. What are eye pattern?

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

1. Width of eye opening defines the interval over which the received wave can be sampled without error from ISI.

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

3. Height of the eye opening at sampling time is called margin over noise.

3. 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 $1/T_b$ Hz. This makes one sweep of beam equal to T_b seconds.

4. What is correlative coding?

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

5. Define Duo binary baseband PAM system.

Duo binary 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 k^{th} bit. Then the encoder the new waveform as

$$C_k = a_k + a_{k-1}$$

Thus two successive bits are added to get encoded value of the kth bit. Hence Ck becomes a correlated signal even though a_k is not correlated. This introduces intersymbol interference in the controlled manner to reduce the bandwidth.

6. What are the three broad types of synchronization?

1. Carrier synchronization
2. Symbol & Bit synchronization
3. Frame synchronization.

7. 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.

8. What are the two methods for carrier synchronization?

1. Carrier synchronization using Mth Power loop
2. Costas loop for carrier synchronization

9. What it 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.

10. What are the two methods used in bit and symbol synchronization?

1) Closed loop bit synchronization

2) Early late gate synchronizer

11. What are the disadvantages of closed loop bit synchronization?

1) If there is a long string of 1's and 0'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.

12. What it 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.

13. 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.

14. 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.

15. Define the principle of adaptive equalization. **(Dec-08)**

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

16. Define duo binary encoding.

Duo binary encoding reduces the maximum frequency of the base band signal the “word duo” means to the double transmission capacity of the binary system

17. Write a note on correlative level coding.

Correlative level coding allows the signal scaling rate of 2Bo in the channel of bandwidth Bo. This is made physically possible by allowing ISI in the transmitted signal in controlled manner this ISI is known to the receiver.

18. Define the term ISI. **(Dec-10)**

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

19. 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 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

20. 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 becomes difficult this distraction can be compensated with the help of equalizer.

21. What is matched filter? **(May-09)**

The matched filter is a baseband signal receiver, which works in presence of white Gaussian noise. The impulse response of the matched filter is matched to the shape of the input signal.

22. Why do we need equalization in base band transmission? (May-07, Dec-08)

When the signal is passed through the channel, distortion is introduced in terms
of i) Amplitude. Ii) Delay.

This distortion creates of ISI.The detection of the signal also becomes difficult. This distortion can be compensated with the help of equalizers. Equalizers are basically filters which connect the channel distortion.

23. List the primary causes for the noise in communication system. (DEC-11)

i) Band limited nature of the channel.

ii) Environmental effects such as lighting, humidity, temperature etc.

iii) EMI and RFI

iv) Thermal noise due to electronic components.

24. 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.

25. State NRZ Unipolar format.

In this format binary 0 is represent by no pulse and binary 1 is represented by the positive pulse.

26. State NRZ polar format.

Binary 1 is represented by a positive pulse and binary 0 is represented by a Negative pulse.

27. State NRZ bipolar format.

Binary 0 is represented by no pulse and binary one is repre ented by the alternative p sitive and negative pulse.

28. State Manchester format.

Binary 0 Æ The first half bit duration negative pulse and the second half Bit duration positive pulse.

Binary 1Æ first half bit duration positive pulse and the second half Bit duration negative pulse.

UNIT –IV

DIGITAL MODULATION SCHEME

1. 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:

1) Optimum filter integrates the signal during the bit interval and checks the output at the time instant where signal to noise ratio is maximum

2) Transfer function of the optimum filter is selected so as to maximize signal to noise ratio.

3) Optimum filter minimizes the probability of error.

2. 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.

3. What is meant by DPSK? **(Madras Univ, April-98)**

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. Explain coherent detection? **(Madras Univ,Nov-97,May-04)**

In coherent detection, the local carrier generated at the receiver is phase locked with the carrier at the transmitter. The detection is done by correlating received noisy signal and locally generated carrier. The coherent detection is a synchronous detection.

5. What is the difference between PSK and FSK? **(Madras Univ, April-97)**

In PSK, phase of the carrier is switched according to input bit sequence. In FSK frequency of the carrier is switched according to input bit sequence. FSK needs double of the bandwidth of PSK.

6. 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.

7. What is the advantage of coherent PSK over coherent ASK? **(Madras Univ, Oct-98)**

ASK is on-off signalling, where as the modulated carrier is continuously transmitted in PSK. Hence peak power requirement is more ASK, whereas it is reduced in case of PSK.

8. Explain the model of band pass digital data transmission system?

The band pass digital data transmission system consists of source, encoder and modulator in the transmitter. Similarly receiver, decoder and destination form the transmitter.

9. What is baseband signal receiver?

A baseband signal receiver increases the signal to noise ratio at the instant of sampling. This reduces the probability of error. The baseband signal receiver is also called optimum receiver.

10. What is matched filter?

The matched filter is a baseband signal receiver, which works in presence of white Gaussian noise. The impulse response of the matched response of the matched filter is matched to the shape pf the input signal.

11. What is the value of maximum signal to noise ratio of the matched filter? When it becomes maximum?

Maximum signal to noise ratio is the ratio of energy to PSD of white noise.

$$\text{i.e., } \rho max = E/ (N0/2)$$

This maximum value occurs at the end of bit duration i.e. Tb.

12. What is correlator?

Correlator is the coherent receiver. It correlates the received noisy signal f(t) with the locally generated replica of the unknown signal x(t). Its output is denoted as r(t).

13. On what factor, the error probability of matched filter depends.

Error probability is given as

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

This equation shows that error probability depends only on energy but not on shape of the signal.

14. Bring out the difference between coherent & non coherent binary modulation scheme. (May-05,07,09,Dec-08,09)

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.

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

Bit error rate for coherent binary FSK is given as,

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

16. Highlight the major difference between a QPSK & MSK signal. (Dec-05)

MSK signal have continuous phase in all the cases, where as QPSK has phase shift of $\pi/2$ or π.

17. What is the error probability of MSK & DPSK? (May-06)

Error probability of MSK: $Pe = 1/2 erfc\sqrt{E/No}$

Error probability of DPSK: $Pe = 1/2 e^{-Eb/No}$

18. In minimum shift keying what is the relation between the signal frequencies & bit rate.

Let the bit rate be fb and the frequency of carrier be f0. The higher and lower MSK signal frequencies are given as,

$$fH = f0 + fb/4$$

$$fL = f0 - fb/4$$

19. List the advantages of Pass band transmission.a. Long distance.

b. Analog channels can be used for transmission.

c. Multiplexing techniques can be used for bandwidth conservation. d. Transmission can be done by using wireless channel also.

20. List the requirements of Pass band transmission.

a. Maximum data transmission rate.

b. Minimum probability of symbol error.

c. Minimum transmitted power.

21. What is signal constellation diagram? (Dec-08)

The signal constellation diagram is similar to the phasor diagram but the entire phasor is not drawn. The signal constellation diagram shows only relative positions of the peaks of the phasors.The signal constellation diagram is also called state space diagram.

22. Define QPSK. (Dec-09)

- QPSK is Quadriphase –shift keying. In QPSK the phase of the carrier takes on one of the four equally spaced values Such as /4 , 3 /4, 5 /4 and 7 /4.
- In QPSK two successive bits in the data sequence are grouped together. This combination of two bits forms four distinct symbols. When symbols are changed to next symbol the phase of the carrier is changed by 45^0.

23. What is meant by memory less modulation?

When the digital symbol modulates amplitude, Phase or frequency of the carrier without any reference to provide symbol, it is called memory less modulation.ASK, FSK, PSK, QPSK etc.are memory less modulation techniques.

UNIT-V

ERROR CONTROL CODING

1. What is linear code? (Dec-07)

A code is linear if the sum of any two code vectors produces another code vector. 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.

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? (Dec-09)

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.

5. What is meant by systematic and non-systematic codes?

In a Systematic block code, message bits 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? (Dec-04)

In Hamming codes the syndrome is calculated as ,

$$S = YH^T$$

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 are the conditions to satisfy the hamming code?

1) No. of Check bits q ³ 3
2) Block length n = 2q –1
3) No of message bits K = n-q
4) Minimum distance dmin =3.

8. What are the error detection and correction capabilities of hamming codes? (May-09)

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.

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. What is difference between block codes and convolutional codes? (Dec-05)

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.

11. What is convolutional code? (May-05)

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.

12. What is meant by syndrome of linear block code? (May-04)

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

13. What are the advantages of convolutional codes?

Advantages:

1 .The decoding delay is small in convolutional codes since they operate of smaller blocks of data.

2. .The storage hardware required by convolutional decoder is less since the block sizes are smaller. **Disadvantages:**

1. Convolutional codes are difficult to analyze since their analysis is complex.

2. Convolutional codes are not developed much as compared to block codes.

14. Define sates of encoder?

The constraint length of the given convolutional encoder is K=2. Its rate is ½ means for single message bit .input, two bits .x1 and x2 .are encoded at the output.S1 represents the input message bit and S.stores the previous message 2 bit. .Since only one previous message bit is stored, .this encoder can have states depending upon this stored message bit.

Let s represent,

$S_2 = 0$ state „a"
$S_2 = 1$ state „b"

15. Define constraint length in convolutional codes?

Constraint length is the number of shifts over which the single message bit can influence the encoder output. .This expressed in terms of message bits.

16. Define minimum distance. (MAY-07)

It is the smallest hamming distance between the valid code vectors. The error detecting correcting capabilities of the codes depend upon the minimum distance.

17. Define hamming weight. (MAY-09)

The number of 1's in the code word of the hamming code is called as hamming distance.

18. What are the classifications if line codes?

Line code is classified as

1. Polar
2. Unipolar.
3. Bipolar

19. 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 pul e.

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

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

21. 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.

22. 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.

23. 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.

24. What are the advantages and disadvantages of cyclic codes? Advantages:

- The error detection and decoding methods of cyclic codes are simpler and easy to implement.
- The encoders and decoders are simpler than non cyclic codes.
- Cyclic codes have well defined mathematical structure. Hence, it is an efficient and powerful code to detect burst errors.

Disadvantages:

- The error detection is simple but error correction is little complicated
- The decoders used are complex circuit.

25. 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.

26. 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