EC6501 DIGITAL COMMUNICATION

UNIT: I – SAMPLING AND QUANTIZATION (C301.1)

PART – A

1. Mention the advantages of digital communication system over analog communication. (May2013,May 2015)

i) It gives ruggedness to transmission noise and interference.
ii) Efficient regeneration of coded signal along transmission path.
iii) Possible to have a uniform format for different kinds of baseband signal.

2. What are the parameters used to describe the channels?

Data rate and attenuation

3. How do you improve BER in digital communication system?(Dec 2012)

Increasing transmitted signal power, Improving frequency filtering techniques, Modulation and demodulation, Coding and decoding technique

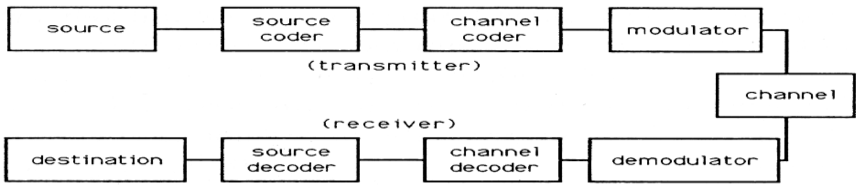
4. State the classification of channels.(May 2013)

Wired Channel -Telephone wire, TV cable or Ethernet cable
Wireless Channel-Underwater ocean channel carrying acoustic wave, Free space carrying EM wave

5. What is a Channel? Give examples.(Nov2013)

Communication Channel is the physical medium between transmitter and receiver. Examples are Wired Channel -Telephone wire, TV cable or Ethernet cable. Wireless Channel-Underwater ocean channel carrying acoustic wave, Free space carrying electromagnetic wave

6. Draw the typical digital communication system (Dec 2012)



7. Give the advantages of digital Communication.(Nov 2013)

Simpler and cheaper, Dynamic range is possible, Noise interference is less, Channel coding and decoding is easier, Accuracy.

8. Give the disadvantages of digital communication.(May2014, May2015)

Requires more transmission bandwidth & needs Synchronization for modulation.

9. What is the main difference between natural sampling and instantaneous sampling instantaneous sampling?(Dec2014)

Consider an arbitrary signal g(t) of finite energy which is specified for all time as shown in fig a .Suppose that we sample the signal g(t) instantaneously and at uniform rate once every T seconds .As a result of this sampling process we obtain an infinite sequence of numbers spaced T_s seconds apart and denoted by g(nTs) where n takes all possible integer values. T_s is the sampling period. Natural sampling: This is applied in PDM (pulse duration modulation). Samples of message signal are used to vary the duration of individual pulses.

10. State sampling Theorem. (May2015, Dec2015)

If a finite energy signal g(t) contains no frequency component higher than ω Hz, it is completely determined by specifying its ordinates at a separation of points spaced 1/ 2ω seconds apart. If a finite energy signal g(t) contains no frequency component higher than ω Hz, it is completely recovered from its ordinates at a separation of points spaced 1/ 2ω seconds apart.

11. List the methods of sampling.

Ideal Sampling (or) Instantaneous Sampling, Natural Sampling, Flat-top sampling

12. What is quantization?

It is the process in which the analog sample of the original signal is converted into a digital form.(In PCM the sampled signal is rounded off to the nearest value which is permitted for transmission by the system. This process of rounding off is termed as quantization.)

13. Classify Quantizers.

Uniform Qunatizer – Representation levels are uniformly spaced
Non-Uniform Qunatizer – Representation levels are non-uniformly spaced

14. What is Natural Sampling?(May2013,Nov2013)

It is the process in which the original analog signal is converted into a discrete time and continuous amplitude signal. The top of each pulse in the sampled sequence retains the shape of the original signal is called natural sampling.

15. What is Quantization Noise?

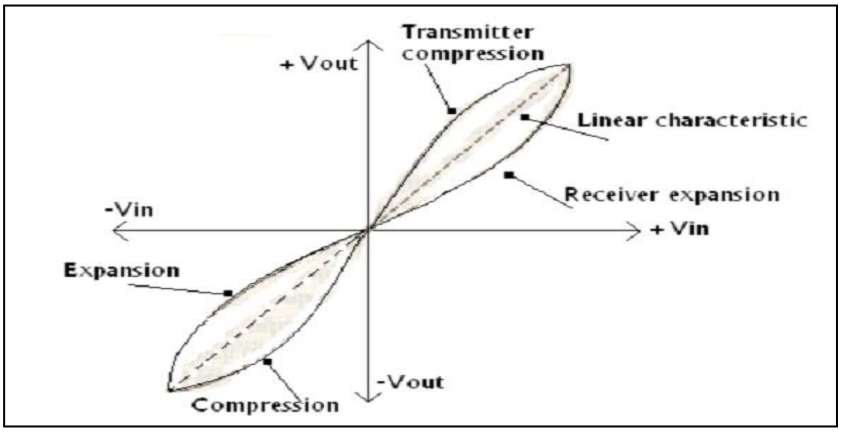
The difference between the output analog sample and the discrete output quantized signal gives raise to an error called Quantization Noise.

16. What are the two types of Quantization Noise?

1. Slope – overload distortion. 2. Granular Noise

17. What is Companding?Sketch the input-output characteristics of a compressor and an expander (May2016,Dec2016)

It is the combined process of compressing and Expanding used for improving the dynamic range of signal and also to increase the SNR of low level signals.



18. What is the need for non uniform quantization?(May2014)

In uniform quantization, the step size remains the same throughout the range of quantizer. For the low signal amplitude, the maximum quantization error is quite high. But for high amplitude, the maximum quantization error is small. This problem arises because of uniform quantization. We need stepsize to be a varying one according to the signal level to keep SNR at the required value. This is achieved by non uniform quantization.

19. What are the corrective measures taken to avoid aliasing effect?

A low pass anti-alias filter is used to attenuate high frequency of the signal that lie outside the frequency band of interest prior to sampling. The filtered signal is sampled at a slightly higher rate than Nyquist rate.

20. Write the A law Compression.(Nov 2013)

$$Z(m) = \frac{A|m|}{1+\ln A} \quad \text{for } 0 \leq m \leq 1/A$$

$$\frac{1+\ln(A|m|}{1+\ln A} \quad \text{for } 1/A \leq |m| \leq 1$$

21. What is aliasing?(May2016, Dec2016))

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

22. State any two non-uniform quantization rules.(May2013)

1. The step size is small at low signal levels. Hence quantization error is also small at these inputs. Therefore signal to quantization noise power ratio is improved at low signal levels. 2. The step size is higher at high input signal levels. Hence signal to noise power ratio remains almost same throughout the dynamic range of quantizer.

23. What are the two fold effects of quantizing process? (Dec2015)

(a) The peak to peak range of input sample values subdivided into a finite sot of decision levels or decision thersholds. (b) 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.

24.**Define non-uniform quantization. (May2015)**

The step size of the quantizer increases as the separation from the origin of the transfer characteristics increases. The SNR will be maintained constant in the wide range of input power levels.

25. Define TDM.

Time-division multiplexing (TDM) is a method of transmitting and receiving independent signals over a common signal path by means of synchronized switches at each end of the transmission line so that each signal appears on the line only a fraction of time in an alternating pattern.

26.A PCM system uses a uniform quantizer followed by a 7 bit binary encoder. The bit rate of system is 50 x10^6 bits/sec. What is the maximum message BW for which the system operates satisfactorily. What is the SNR for a full load sinusoidal signal?

$2f_mN = 50 \times 10^6$; $2f_m \times 7 = 50 \times 10^6$; $f_m = 50 \times 10^6/14 = 3.57$ MHz. ; SNR = 1.8 + 6N = 43.8 dB

27. 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.

28. How aperture effect can be corrected?

Aperture effect can be corrected by connecting an equalizer in cascade with the low pass reconstruction filter. This equalizer has the effect of decreasing the in-band loss of reconstruction filter as the frequency increases in such a manner as to compensate for the aperture effect.

PART-B

1. With neat block diagram explain the digital communication system. **(Nov2013,May2014)**

2. i)State and prove the sampling theorem for low pass signals and explain the reconstruction of the signal from its samples **(Dec2015, May 2016)**

ii) The signal x(t) =4cos400πt+12cos360 πt is ideally sampled at a frequency of 300 samples per second. The sampled signal is passed through a unit gain LPF with a cut off frequency of 220Hz. List the frequency components present at the output of the LPF.

3. A signal is sampled at Nyquist rate of 8 kHz and is quantized using 8 bit uniform quantizer. Assuming SNRq for a sinusoidal signal, calculate the bit rate,SNRq and BW.

4.Write short notes on aliasing and how to overcome it.

5. Derive an expression for Signal to Quantization noise ratio of Uniform quantizer. **(Nov 2013)**

6. i) Derive the expression for quantization noise of a PCM system with uniform quantizer. Explain how it can be improved. (10)**(May2013)**

ii) A sinusoidal signal is transmitted using PCM. An output SNR if 55.8 dB is required. Find the number of representation levels required? (6)

7. i) Draw the block diagram and explain the process of a PCM system in Detail. **(Dec2015)**

ii) Explain how PCM is influenced by noise sources.

8. i) Explain Uniform Quantization in detail with its types.(**Dec2016)**

ii) Discuss the necessity for Non-Uniform Quantization of speech signal. **(May2014)**

9. Explain the principle of companding in detail with its types. **(May2014)**

10. For the signal m(t) = 10cos2000πtcos8000πt, determine the minimum sampling rate based on

i) Low pass sampling theorem

ii) Band pass sampling theorem

11.i)Explain PCM system with neat block diagram(May 2016)

ii). What is TDM?Explain the difference between analog TDM and digital TDM

12. Draw and explain the TDM with its applications(Dec2016))

UNIT : II – WAVEFORM CODING (C301.2)

PART - A

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

- To Remove redundancies from the speech signal as far as possible,
- To Assign the available bits in a perceptually efficient manner.

2. What are the types of adaptive predictors?

- Adaptive prediction with forward estimation (APF),
- Adaptive prediction with backward estimation (APB)

3. Mention the use of Vocoders.

To remove redundancies from the speech signal and to constantly adapting to the speech statistics.

4. What is the advantage of DM over PCM?

DM use one bit to encode one sample.Hence bit rate of delta modulation is low than PCM.

5. Define delta modulation.

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

6. Discuss the noise effect in delta modulation?

In delta modulation we observe quantization noise .there are two major sources of quantizing error in DM systems. They are slope over load distortion, Granular noise

7.Define APB. (Dec2015)

Adaptive prediction with backward estimation (APB), in which samples of the quantizer output and the prediction error are used to derive estimates of the prediction error are used to derive estimates of the predictor coefficients.

8. Define granular noise. How it can be 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.

9. Determine $(S/N)_q$ of a delta modulation system at a bit rate of 64 kbps and BW of 4 kHz.

f_c = 4kHz; f_b = 64kbps; $(S/N)_q = (0.04)\ (f_b/f_c)^3 = 163.84 = 22\text{dB}$

10. What are the drawbacks of DM?

DM requires a large transmission bandwidth than PCM to achieve the same SNR..Speech signal requires large dynamic range,but to avoid slope overload DM has small dynamic range. So DM is not suitable for high dynamic range speech

11. Define Adaptive delta modulation.

In adaptive delta modulation, 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 size is decreased.

12. What is meant by Prediction error?

The difference between the actual samples of the process at the time of interest and the predictor output is called prediction error.

13. Mention two merits of DPCM.

BW requirement of DPCM is less compared to PCM, Quantization error is reduced because of prediction filter, Number of bits used to represent one sample value is reduced as compared to PCM.

14. State the differences between DPCM and DM.

i)DM uses only one bit information for transmission ii)Replacement of the prediction filter in DPCM by a single delay element constitutes DM system.

15. What are the advantages of adaptive delta modulation?

1. Slope overload noise is reduced; 2. Granular noise is reduced 3.less number of bits are used

16. Define ADPCM.

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.

17. Mention the use of adaptive quantizer in adaptive digital wave form coding scheme.

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

18. What do you understand from adaptive coding?

In adaptive coding quantization step size and prediction filter co-efficient are changed as per properties of input signals. This quantization error and number of bits used to represent the sample value. Adaptive coding is used at low bit rates.

19. What are the two limitation of delta modulation? (Dec2015)

1. Slope overload distortion. It occurs due to limited step size and fast variation in the signal.
2. Granular noise occurs due to large step size and very small amplitude variations in the input signal.

20. How distortions are overcome in ADM?

i)The slope overload and granular noise occur mainly because of fixed step size in delta modulator.
ii)Step size is more for fast amplitude changes and step size is less for slowly varying amplitude.
iii)The step size is varied according to amplitude variations of input signal

21. Define APF. (Dec2015)

Adaptive prediction with forward estimation (APF), in which unquantized samples of the input signal are used to derive estimates of the predictor coefficients.

22. What are the advantages of delta modulator? (May 2016)

High SNR, Low bandwidth consumption and cost effective systems.

23. **What is a linear predictor?On what basis are predictor coefficients determined?(May 2016)**

Linear predictor is a filter that uses linear combination of finite set of present and past samples of a stationary process to predict a sample of the process in the future.The predictor coefficients are determined in such a way that it minimizes the mean squre value of the prediction error.

24. What is the need of prediction filtering?(Dec2016)

To reduce the error we would have by encoding the actual sample directly.

25. How to overcome the slope overload?(Dec2016)

To reduce slope overload distortion, the step size must be increased when the slope of the input signal is high.

PART-B

1. A voice signal is prefiltered to 4kHz and its rms bandwidth is 1.3kHz. Find out the SNR of this system for a bandwidth expansion factor of 8.Compare this bandwidth to a PCM System. (Assume that there is an optimum slope overloading factor)

2. Explain noise in PCM system. Compare DPCM with PCM and DM. **(Dec 2012)**

3. Explain Delta Modulation system in detail.What is slope overload noise and granular noise and how it is overcome in ADM. **(Dec2015, Dec2016)**

4. Explain adaptive quantization and prediction in ADPCM with neat block diagram.**(May2013)**

5. Explain adaptive delta modulation with necessary diagram and compare PCM with DM. **(May 2016, Dec2016)**

6. Compare DM with ADM and explain linear prediction filter.

7 .i) Compare and contrast DPCM and ADPCM (8)
ii) Explain noise in DM systems. (8)

8. Compare PCM, APCM, DPCM, ADPCM, DM, ADM.(**Dec2016)**

9. With transmitter and receiver block diagram , explain DPCM in detail.

10. Illustrate how the adaptive time domain coders codes the speech at low bit rate and compare it with the frequency domain coders. **(Dec2015)**

11.i)Draw the block diagram of ADPCM system and explain its function. (10)

ii)A DM with a fixed step size of 0.75v, is given a sinusoidal message signal. If the sampling frequency is 30 times the Nyquist rate, determine the maximum permissible amplitude of the message signal if slope overload is to be avoided.(6) **(May 2016)**

UNIT – III- BASEBAND TRANSMISSION (C301.3)

PART - A

1. What is NRZ polar format?

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

2. What is RZ polar format?

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

3. Give the properties of line codes. (Dec 2012)

1. Self-synchronisation,2.Error detection 3.Bandwidth compression 4. Differential encoding 5. Noise immunity 6. Spectral compatibility with channels 7. Transparency

4. What is meant by transparency with respect to line codes?(May2013)

A line code should be so designed such that receiver does not go out of synchronization with the any sequence of data symbols. A clock is must for this synchronization.

5. How the impulse response of the optimum filter is related to the input signal.

The impulse response is equal to the input signal displaced to a new origin at t=to and folded about this point so as to run backward. $H_{opt}(t) = K\ x(to-t)$

6. Define ISI(Dec2014)

There are effects of imperfection in the frequency response of the channel ie. Dispersion of the false shape by the channel. The residual effect of all other transmitted bits on the received bit is called as Inter symbol interference.

7. A 64kbps 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 the filtered PCM signal.(Dec2012)

D = 64kbps; γ= Roll-off factor=0.25; Bandwidth= 40kHz

8. What is minimum bandwidth required to transmit data at the rate of R_b bits per sec?

$$B = \frac{R_b}{2}\ \textbf{Hz.}$$

9. What is eye pattern?(Nov2013)

When the sequence is transmitted over a baseband binary data transmission system, the output is a continuous time signal. If this signal is out at each interval (Tb) and all such pieces are placed over one another, then we obtain eye pattern. It looks like eye. Eye pattern is particularly useful in studying ISI problem.

10. State Nyquist criterion for zero ISI.

The spectra of the transmitted pulse should satisfy following equation $\sum_{n=-\infty}^{\infty} P(f - nR_b) = T_b$, where P(f) is the spectrum of the transmitted pulse p(t) and $R_b = 1/ T_b$ is the rate at which pulses are transmitted.

11. What is the function of equalizing filter?(Dec2014)

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

12. What is correlative coding?

Adding ISI in a controlled manner to achieve a bit rate of $2B_0$ bits per second in a channel of bandwidth B_0 Hz

13. How does pulse shaping reduce inter symbol interference.

Pulse shaping compresses the bandwidth of the data impulse to a smaller bandwidth greater than the Nyquist minimum, so that they would not spread in time. System performance is not degraded.

14. State any two applications of eye pattern. (Dec 2012, May2015)

To study the intersymbol interference, to measure the additive noise, to measure the timing synchronization and jitter, Non –Linearities in the channel.

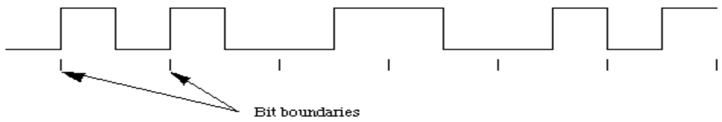
15. What are the information that can be obtained from eye pattern regarding the signal quality? (May2014)

From the eye pattern-(i) WIDTH defines the time interval over which the received signal can be sampled without error from ISI.(ii) SLOPE determines the sensitivity of the system to timing error.(iii) HEIGHT defines the margin over noise. For zero ISI, the eye is widely opened. For more ISI,the eye will be closed completely.

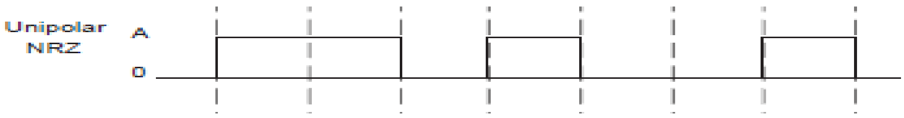
16. ISI cannot be avoided. Justify the statement.(May2013)

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

17.Draw the Manchester coding format for the data sequence 110100.



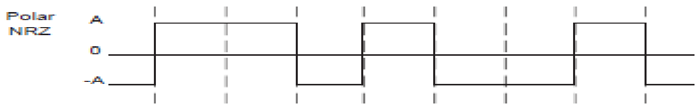
18. Draw the Unipolar NRZ format for the data sequence 1101001



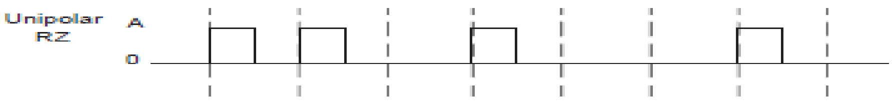
19. What is Manchester coding and write its advantages? (Dec2014)

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

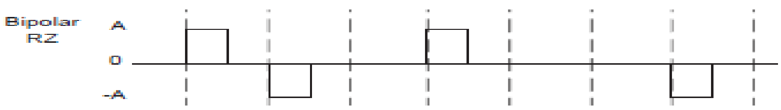
20. Draw the Polar NRZ format for the data sequence 1101001



21. Draw the Unipolar RZ format for the data sequence 1101001



22. Draw the Bipolar RZ format for the data sequence 1101001



23. What are line code? Name some popular line codes. (May 2016)

A line code is the code used for data transmission of a digital signal over a transmission line. This process of coding is chosen so as to avoid overlap and distortion of signal such as inter-symbol interference. Manchester code, gray code, Unipolar, polar and bipolar line codes.

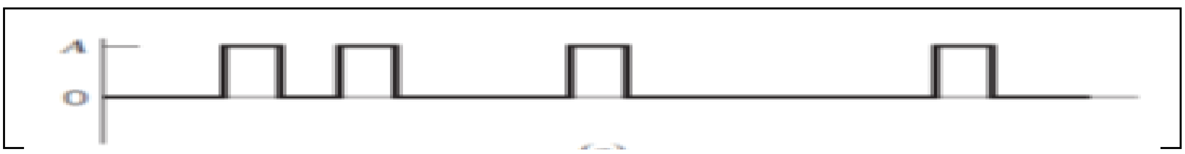
24. What is ISI and what are the causes of ISI? (May 2016)

ISI is a form of distortion of a signal in which one symbol interferes with subsequent symbols. This is an unwanted phenomenon as the previous symbols have similar effect as noise, thus making the communication less reliable. Spreading of the pulse beyond its allotted time interval causes it to interfere with neighboring pulses. ISI arises due to imperfections in the overall response of the system.

25. Define correlative level coding.(Dec2016)

Practical means of achieving the theoretical maximum signalling rate of $2B_o$ bits per sec in a banwidth of B_o Hz by adding ISI in a controlled manner.

26. For the binary data 01101001 draw the unipolar RZ signal(Dec2016)



PART - B

1. i) Derive the expression for the PSD of unipolar NRZ format.(5)**(May2015,Dec2015, May 2016)**

ii)Derive an expression for the PSD of Bipolar NRZ format.(5)**(May2013,Dec2015, May 2016)**

2. i)Derive an expression for the PSD of Polar RZ format.(8)**(May2013, May2015)**

ii)Derive an expression for the PSD of Manchester NRZ format.(8)

3.Discuss on signal design for ISI elimination. **(May2014)**

4. Explain modified duo-binary signaling scheme without & with precoder. **(Dec2015, May 2016)**

5. Obtain an expression for Nyquist criterion for distortion - less baseband transmission for zero symbol interference. **(May2013, May2015)**

6. Describe how eye pattern can be obtained and can be used for observing the characteristics of a communication channel. **(Dec2014, Dec2015)**

7. Illustrate the modes of operation of an adaptive equalizer with a neat block diagram. **(Dec2015, May 2016)**

8. Explain the duobinary signalling technique in detail.

9. (i)Draw the different code formats for the given sequence 1110 0100 (8)

(ii) list the properties of line codes (8)

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

11. i) Sketch the power spectra of a)Polar NRZand b) Bipolar RZ signals(10)

ii) Compare various line coding techniques and list their merits and demerits (6)(May 2016)

12.Explain how Nyquist Criterion eliminates interference in the absence of noise for distortion-less baseband binary transmission.(Dec 2016)

13.Describe how eye pattern is helpful to obtain the performance of the system in detail with a neat sketch. (Dec 2016)

UNIT : IV – DIGITAL MODULATION SCHEME (C301.4)

PART - A

1.What is Signal Constellation diagram?

The diagram which defines the collection of M message points in N dimensional Euclidean space is called signal constellation diagram. It helps to find the probability of error.

2. What is ASK and mention the drawbacks? (Nov2013)

The symbols 0 and 1 are differentiated by amplitude of the carrier. The drawbacks are i)Very sensitive to noise, ii) amplitude fluctuations occur in the channel, iii) Not suitable for pass band, wireless communication

3.What are the applications of digital modulation technique?

1.Voice grade modems uses 8 phase DPSK technique,2 Digital Radio uses 16-ary QAM,3.Satellite communication uses BPSK,QPSK technique,4.Voice grade telephone channel uses FSK,5.4 phase DPSK is used as international standard for modems operating at 2400 bits/sec.

4. Define PSK.

PSK is a modulation technique achieved by keying the phase of the carrier between either of two possible values corresponding to the binary symbols 0,1 with fixed limits set by the channel.

5. Compare M-ary modulators.

M-ary FSK requires a considerably increased bandwidth in comparison with M-ary PSK. The probability of error for M-ary FSK decreases as M increases. for M-ary PSK probability of error increases with M.

6.What is meant by Binary phase shift keying?

If the transmitted signal is sinusoid of fixed amplitude then it is called as Binary – phase- shift keying. It has one fixed phase when the data is at one level & when the data is at another level the 'phase' is different by 180°

7. How the BPSK signal is generated?

The BPSK signal is generated by applying the waveform Cos(u_o t) as a carrier to a 'balanced modulator' and applying the base band signal b(t) as the modulating waveform,in this sense BPSK can be thought of as an AM signal.

8. What is the probability of error of BPSK?(Dec 2012)

$P_e = \frac{1}{2}\mathrm{erfc}\left(\sqrt{\frac{E_b}{N_0}}\right)$ E_b = Transmitted signal energy, N_o = Noise PSD, erfc= error function

9. Define FSK.

PSK is a modulation technique achieved by keying the phase of the carrier between either of two possible values corresponding to the binary symbols 0,1 with fixed limits set by the channel.

10. What are the types of synchronization?

1. Carrier Synchronization 2. Symbol and bit synchronization 3. Frame synchronization

11. Why synchronization is needed?

Signals from various sources are transmitted on single channel by multiplexing. So, synchronization is needed, it is also required for detectors to recover the digital data properly from the modulated signal.

12. What is the bandwidth efficiency of M-ary PSK?

$$\rho = \frac{\log_2 M}{2}$$

13. What is the bandwidth efficiency of M-ary FSK?

$$\rho = \frac{2\log_2 M}{M}$$

14. Define bandwidth efficiency.

Ratio of data rate to channel bandwidth, measured in units of bits per second per hertz.

15.What is DPSK?

Differential phase shift keying uses differential encoding. Phase shift keying is modulated at the transmitter side. Receiver performs detection by comparing the phase of received symbol with that of previous symbol. Non coherent receiver is used.

16. What is memory less modulation?(Dec 2012, May2015)

If the symbol to waveform mapping is fixed from one interval to the next, i.e., m → $s_m(t)$, then the modulation is memoryless. If the mapping from symbol to waveform in the n-th symbol interval depends on previously transmitted symbols (or waveforms) then the modulation is said to have memory.

17. What is coherent detection/receiver?(Nov2013, May2015)

When the receiver exploits the knowledge of the carrier's phase to detect the signal, then the detection is coherent.

18. What is non-coherent detection/receiver? (May2015)

When the receiver does not utilize the phase reference information, then the detection is non coherent.

19. A binary shift keying system employs two signal frequencies f_1 and f_2, the lower frequency is 1200Hz and the signaling rate is 500 baud .Calculate f_2.

f_2=1200+500=1700Hz

20. What is QAM?(May2013)

In quadrature amplitude modulation, the information is contained in both amplitude and phase of the transmitted carrier. Signals from two separate information sources modulate the same carrier frequency at the same time. It conserves the bandwidth.

21. **Differentiate coherent and non-coherent detection methods. (May2013, May 2016, Dec2016))**

In coherent method, carrier is regenerated at the receiver.

In non-coherent method, carrier need not be regenerated at the receiver side.

22.Compare M-ary PSK and M-ary QAM. (Dec2015)

Sl. No	**M-ary PSK**	**M-ary QAM**
1.	Carrier experiences phase modulation	Carrier experiences amplitude and phase modulation
2.	Signal constellation is circular	Signal constellation is square

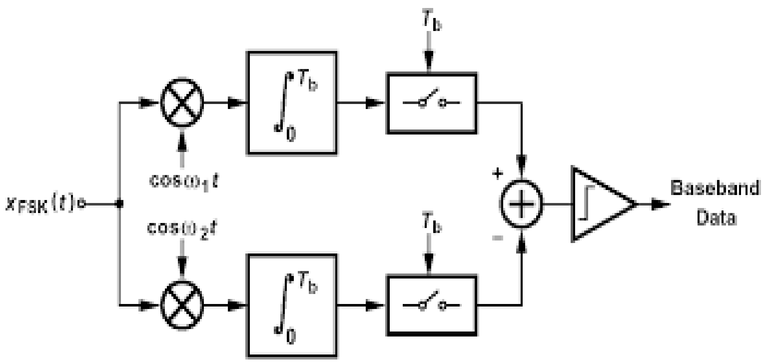
23. Mention the advantages of PSK systems (Dec2014).

i) Generation and Detection of PSK signals require simple circuit ii) Information transmission rate is higher because of educed bandwidth (iii) Carrier power remains constant

24. Define false alarm errors. (May2015)

Let hypothesis H_0 represents the presence of only noise and hypothesis H_1 represents presence of signal in addition to noise. False alarm represents the selection of H_1 when H_0 is the correct answer.

25.Draw the block diagram of coherent BFSK receiver. (Dec2015, Dec2016))



26.Define false dismissal errors. (May2015)

Let hypothesis H_0 represents the presence of only noise and hypothesis H_1 represents presence of signal in addition to noise. False dismissal represents the selection of H0 when H_1 is the correct answer.

27. Distinguish BPSK and QPSK techniques. (Dec2015)

Sl. No	BPSK	QPSK
1.	Two phases are used	Four different phases are used
2.	Lower data rate	Higher data rate

28.Mention two properties of matched filter?(Nov2013)

Property1: The peak pulse SNR of a matched filter depends only on the ratio of the signal energy to the power spectral density of noise. Max.SNR = 2E/η

Property: 2 The integral of the squared magnitude spectrum of a pulse signal with respect to frequency is equal to the signal energy,Error! Reference source not found.2df = E

29.What is QPSK? Write the expression for the signal set of QPSK.(May 2016)

In QPSK, the phase of the carrier wave takes on one of four equally spaced values, namely π/4, 3π/4, 5π/4 and 7π/4. The expression for QPSK is given by

$$s_i(t) = \begin{cases} \sqrt{\frac{2E}{T}} \cos\left[2\pi f_c t + (2i-1)\frac{\pi}{4}\right] & 0 \le t \le T \\ 0 & \text{elsewhere} \end{cases}$$

, where I=1,2,3,4 and E be the transmitted signal energy per symbol, T be the symbol duration and f_c be the carrier frequency.

30. What is matched filter?(Nov2013,May2014)

It is a linear filter designed to provide maximum SNR at its output for a given transmitted signal. A matched filter is obtained by correlating a known signal with the unknown signal to detect the presence of the known signal. In communication the matched filter is used to detect the transmitted pulse in the presence of noise.

31. State the principle of maximum likelihood detector.(May2013)

For equally likely symbols, the detector decides in favour of a particular transmitted symbol whose likelihood function is greater than the other. This strategy is called maximum likelihood detection

PART-B

1. Describe with diagrams, the generation and detection of coherent binary PSK. Derive the error probability of Binary PSK.Illustrate power spectra of BPSK**(Nov2013,May2014, May2015,Dec 2016).**

2. Describe with diagrams, the generation and detection of coherent binary FSK. Derive the error probability of Binary FSK.**(Nov2013, May2015)**

3. Describe with diagrams, the generation and detection of coherent binary QPSK. Derive the error probability of Binary QPSK.**(May2013, May2015,Dec2015, Dec 2016)**

4. Explain the techniques for carrier synchronization.

5. Compare the various coherent and non coherent digital detection systems.**(Dec 2012.May2014)**

6. Describe with diagrams, the generation and detection of coherent DPSK. Derive the error probability of DPSK.

7. Discuss the spectral & representation characteristics of PSK, QAM, QPSK & FSK.**(Dec 2012)**

8. Describe with signal space diagram QAM and its differences with respect to QPSK.**(Nov2013,May2014)**

9. Describe with diagrams, the generation and detection of coherent binary ASK. Derive the error probability of Binary ASK.**(Nov2013, May2015).**

10. Derive the expression for bit error probability of a non- coherent FSK system.**(May2013, May 16)**

11. Illustrate the transmitter, receiver and generation of non-coherent version of PSK with neat diagram and derive the PSD of BPSK and plot it .**(Dec2015, May 2016)**

UNIT : V – ERROR CONTROL CODING (C301.5)

PART - A

1. What is Channel Encoding?

The channel encoder systematically adds digits to the message. These redundant bits carry no information. But used to detect and correct errors in the receiver side.

2. List the four objectives of a Channel code or error control code.(Dec2014)

1) To have the capability to detect and correct errors.
2) To be able to keep the process of error detection and correction as more practicable.
3) To be able to encode the symbol in a fast and efficient way.
4) To be able to decode the symbol in a fast and efficient way.

3. What are the types of error control methods?

1) Error detection and retransmission (ARQ – Automatic Repeat Request Method)
2) Error detection and correction (FEC – Forward Error Correction method)

4. What is Convolutional Code?

Fixed number of input bits are stored in the fixed length shift register and they are combined with the help of modulo-2- adders. This operation is equivalent to binary

5. What are the structural properties of convolutional code?

(a) State diagram,(b) Code tree,(c)Trellis

It shows the transition between various states

6. What is constraint length of convolutional code? What is code rate?

The number of shifts over which a single message bit can influence the encoder output is called constraint length. Code Rate = L/n(L+M), L – Length of message sequence, n(L+M) – Code word length

7. Differentiate block code and convolutional code

In block code the encoder accepts a k-bit message block and generates an n-bit codeword. Thus, codewords are produced on a block by block basis.

In Convolutional code the encoder accepts the message bits come in serially rather than in large blocks and generates n- bit codeword. The resultant bits are generated using modulo-2 additions.

8. Explain the term syndrome.

The syndrome S of the received code word R is defined as S=G H^T. If S is not zero then there are one or more errors. If the syndrome is zero then either there are no error or the errors are so many that a transmitted code word has been changed to a different code word.

9. What is meant by cyclic codes?

A cyclic code has property that a cyclic shift on one code word forms another code word and are important because they are algebraic properties, which allow them to be easily encoded or decoded.

10. What is meant by line codes?(Nov2013)

The channel coded data is mapped to a particular pulse waveform before transmission. This waveform is called Line Coding

11. Prove that in linear block codes syndrome depends on error pattern not message bits.

$S = (x + e) H^T = x H^T + e H^T = e H^T$; Thus the syndrome pattern S depends on error pattern and not on message bits

12. Write the properties of syndrome in linear block codes. (Dec2015)

1. The syndrome depends only on the error pattern, and not on the transmitted codeword.
2. All error pattern that differ at most by a codeword have the same syndrome
3. The syndrome S is the sum of those columns of the matrix H corresponding to the error locations.

13. Write the advantages of cyclic codes over block codes.

1). They are easy to encode, 2). Cyclic codes possess a well-defined mathematical structure, which has led to the development of very efficient decoding schemes for them.

14. What is systematic code?

A code in which the message bits are transmitted in an unaltered form.

15. What is hamming code?

Hamming codes are of (n,k) linear block codes that have the parametes.1) Block Length = $2^m - 1$, Number of message bits k = $2^m - m - 1$,, Number of parity bits, n-k = m where $m \geq 3$ so called Hamming code

16. Determine the Hamming weight of the codeword 0110100.

Hamming weight = 3

17. Define the code rate of (n,k) code.(Nov 2013)

Code rate = k/n, where k is the length of message bits and n is length of the code word.

18. Define Constraint length. (May2015, May 2016)

Number of shifts over which a message bit can influence the encoder output.

19. What is a perfect code?

A code in which the hamming bound satisfies with equality sign

20.Find the hamming distance between 101010 and 010101.If the minimum hamming distance of a (n,k) linear block is 3.What is the minimum hamming weight?(Dec 2012)

Hamming Distance = 6. d_{min} (3) $\leq$ n-k+1. Hamming weight is no. of ones.

21. Define hamming distance.(May2014, May2015)

The hamming distance between two codes is equal to the number of elements in which they differ.

22. Define code efficiency or code rate.

It is the ratio of message bits in a block to the transmitted bits for that block by the encoder.

23. State the significance of minimum distance of a block code.(May2013)

The minimum distance d_{min} of a linear block code is the smallest hamming distance between any pair of code vectors in the code. Minimum distance is an important parameter of the code. It determines the error correcting capability of the code.

24. What is Viterbi decoding scheme?

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

Metrics: It is the discrepancy between the received signal and the decoded signal at particular node. **Survivor path**: This is the path of decoded signal with minimum metric.

25. What are the limitations of Viterbi decoding?

Viterbi decoding can correct upto2 errors. A triple error pattern is uncorrectable by the Viterbi algorithm. Constraint length increases, complexity also increases exponentially. Remedy: Sequential decoding procedure is used. In which the error probability decreases easily, Decoding procedure is independent of constraint length.

26. Define Hamming weight? (May2015)

Number of non-zero elements in the code vector

27. Define channel coding theorem(Dec2016)

Let a discrete memoryless source with an alphabet S and an entropy H(S), produce symbols once every T_s seconds. Let a discrete memoryless channel have capacity C be used once every T_c seconds.

Then if, $\frac{H(s)}{T_S} \le \frac{C}{T_C}$, there exists a coding scheme for which the source output can be transmitted over the channel and be reconstructed with an arbitrarily small probability of error. The parameter $\frac{C}{T_C}$ is called critical rate.

28. List the properties of cyclic codes. (Dec2015)

Linearity property: sum of any two code word is also a codeword in the given code.

Cyclic property: any cyclic shift of a code word is also a codeword in the given code.

29. What is a linear code? (May 2016)

In linear code, the parity bits are generated as a linear combination of message bits.

PART-B

1. Describe the steps involved in the generation of linear block codes. Define Syndrome and explain the properties of syndrome.**(Nov2013)**

Find a) Generator matrix b) all possible code words

2. Design a block code for a message block of size 8 that can correct for single errors.**(Dec 2012)**

3. Design a convolution coder of constraint length 6 and rate efficiency ½. Draw its tree and Trellis diagram. **(Dec 2012),(Nov2013)**

4. Design a syndrome calculator for (7, 4) cyclic Hamming code generated by the polynomial G (P)= p^3+p+1.Calculate the syndrome for Y=(1001101).

5. The generator polynomial of a (7, 4) cyclic code is a G(x)=x^7+1.Find all the code vectors in systematic form. Find also the generator matrix and parity check matrix.

6. Draw the diagram of a ½ rate convolution encoder with constraint length 3. What is the generator polynomial of the encoder? Find the encoded sequence corresponding to the message (10011).**(Nov2013)**

7. Explain Viterbi algorithm to decode a convolutionally coded message. **(May2013, May2014, May2015)**

8. All zero sequence is transmitted and error has occurred in two locations, Using Viterbi algorithm find the correct code from the received sequence (01, 00, 01, 00, 00)

9. Brief about any one decoding procedure of linear block codes. **(May2014)**

10. Find generator polynomial for a (7,4) cyclic code and hence find the code word for [1 0 0 0] **(May2014)**

11. For a systematic linear block code, the three parity check digits P_1, P_2, P_3 are given by

$$P_{k,n-k} = \begin{bmatrix} 1 & 0 & 1 \\ 1 & 1 & 1 \\ 1 & 1 & 0 \\ 0 & 1 & 1 \end{bmatrix}.$$

i)Construct generator matrix. ii) Construct codewords iii)Determine the error correcting capacity. iv)Decode received words with an example. **(Dec2015)**

12. A convolution code is described by g^1= [1 0 0]; g^2= [1 0 1]; g^3= [1 1 1].

(i) Draw the encoder corresponding to this code

(ii) Draw the state transition diagram for this code

(iii) Draw the Trellis diagram

(iv) Find the transfer function. **(Dec2015)**

13. Consider a linear block code with generator matrix (May 2016)

$$G = \begin{bmatrix} 1 & 1 & 0 & 1 & 0 & 0 & 0 \\ 0 & 1 & 1 & 0 & 1 & 0 & 0 \\ 1 & 1 & 1 & 0 & 0 & 1 & 0 \\ 1 & 0 & 1 & 0 & 0 & 0 & 1 \end{bmatrix}$$

i) Determine the parity check matrix (3)
ii) Determine the error detecting and capability of the code (3)
iii) Draw the encoder and syndrome calculation circuits. (6)
iv) Calculate the syndrome for the received vector r=[1 1 0 1 0 1 0] (4)

14) i) The generator polynomial of a (7,4) cyclic code is 1+X+X^3. Develop encoder and syndrome calculator for this code. (8) (May 2016)

ii) Explain Viterbi decoding algorithm for convolutional code. (8) (May 2016)

15. i)Describe the cyclic codes with the linear and cyclic property.Also represent the cyclic property of a codeword in polynomial notation(12)(**Dec 2016)**

ii)List the different types of errors detected by CRC Code(4)

16. i) Describe how the errors are corrected using Hamming code with an example.(12)

ii) The code vector [1110010] is sent, the received vector is[1100010]. Calculate the syndrome. **(4) (Dec 2016)**