KSRCE/QM/7.5.1/40/ECE

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING
K.S.R. COLLEGE OF ENGINEERING: TIRUCHENGODE – 637 215.
(Autonomous)

COURSE / LESSON PLAN SCHEDULE

NAME : R. Veeramani
CLASS : III- ECE (A & B)
SUBJECT : 16EC511/ DIGITAL COMMUNICATION SYSTEMS

A) TEXT BOOKS:

1. Amitabha Bhattacharya, "Digital Communications", Tata McGraw Hill, 2006, Re-print 2015.
2. Simon Haykin, "Digital Communications", John Wiley, 2001, 4th edition, Re-print 2012.
3. Bernard Sklar, "Digital Communication", Pearson Education, 2nd Edition,2006.

B) REFERENCES:

1. John.G. Proakis, "Fundamentals of Communication Systems", Pearson Education, 2nd edition, 2007.
2. Michael. B. Purrsley, "Introduction to Digital Communication", Pearson Education, 2006.
3. Herbert Taub & Donald L Schilling, "Principles of Communication Systems", Tata McGraw Hill, 3rd edition, 2008.
4. Leon W. Couch, "Digital and Analog Communication Systems", Pearson Education, 6th edition, 2001.

C) LEGEND:

L	Lecture	**Tx**	Text	**LCD**	LCD Projector
BB	Black Board	**OHP**	Over Head Projector	**Ex**	Extra
pp	Pages	**Rx**	Reference		

Sl. No	Lecture hour	Topic(s) to be covered	Teaching aid required	Book no./Page no.
UNIT-I		**DIGITAL COMMUNICATION SYSTEM**		
1	L1	Elements of digital communication systems	BB	Tx1/pp 1-7,Rx4/pp30-33
2	L2	Model of Digital Communication Systems	OHP	Tx1/pp7-9,Tx2/pp4-5,Rx4/pp48
3	L3	Communication Channel Classification	OHP	Tx1/pp9-10,Tx2/pp5-7
4	L4	Performance measure of Communication System	BB	Tx1/pp10-12, Tx2/pp 13-15
5	L5	Geometric representation of Signals	BB	Tx1/pp15-18, Rx2/pp5-10
6	L6	Problems in Geometric representation of Signals	BB	Tx1/18-20 , Tx3/7-9
7	L7	Gram-Schmidt Orthogonalization Procedure (GSOP)	OHP	Tx1/pp20-21, Rx4/pp9-11
8	L8	Bandwidth in Communication Channel	BB	Tx1/pp25-29, Tx3/pp 47
9	L9	Mathematical Models of Communication Channel, **Hilbert Transform**	BB	Tx1/pp29-31, Tx1/pp 86-88, Rx1pp/10-12
UNIT- II		**BASEBAND FORMATTING TECHNIQUES**		
10	L10	**Formatting Text,** Sampling & Impulse Sampling	OHP	Tx1/110-113, Rx4/114-120
11	L11	Natural Sampling, Sampler Implémentation	BB	Tx1/pp113-118, Rx4/pp151-155
12	L12	Uniform & Non-uniform Quantization	BB	Tx1/pp118-124, Tx3/pp81-85
13	L13	Encoding Techniques for analog sources, Temporal waveform encoding (PCM)	BB	Tx1/pp124-146, Tx2/pp172-180
14	L14,T4	Problems on Quantization & Encoding techniques	BB	Tx1/ pp118-167, Tx2/pp 130-143
15	L15	Adaptive Pulse code Modulation(APCM), Differential Pulse Code Modulation(DPCM)	LCD	Tx1/pp146-152, Tx2/pp200-203
16	L16	Adaptive Differential Pulse code Modulation(ADPCM), Delta Modulation	LCD	Tx1/pp157-166, Tx2/pp203-208 Tx2/pp211-218
17	L17	Adaptive Delta Modulation, Spectral waveform encoding	LCD	Tx1/pp166-169, Tx2/pp208-210
18	L18	Model-based encoding & Comparison of	BB	Tx1/pp169-174, Tx2/pp221-225

		speech encoding methods.		
UNIT- III		**BASEBAND CODING TECHNIQUES**		
19	L19	Entropy, Block Codes	BB	Tx1/pp 180-184,199-220, Tx2/pp 370-379, Rx1/pp 416-470, Tx3/pp328-342, Rx4/pp48-49
20	L20	Problems on Block Codes	BB	Tx1/pp 199-220, Tx2/373-392
21	L21	Convolutional Codes	BB	Tx1/pp220-235, Rx1/pp471-521 Tx3/pp408-421, Rx4/pp50-52 Tx2/pp393-403
22	L22	Problems on Convolutional Codes	BB	Tx1/pp220-235, TX3/pp 60-63
23	L23	Concept of Error Free Communication	OHP	Tx1/pp238-240, Rx4/pp53-56
24	L24	Classification of line codes	OHP	Tx1/pp265-267, Rx4/pp182-185
25	L25	Desirable characteristics and Power Spectra of line code(Unipolar Non Return to Zero - NRZ)	BB	Tx1/pp267-271, Rx4/pp185
26	L26	Bipolar NRZ, Unipolar RZ	BB	Tx1/pp271-274, Tx2/pp412-415
27	L27	Return to Zero Alternate Mark Inversion – RZ AMI , Manchester NRZ	BB	Tx1/pp274-277, Tx3/pp440-446
UNIT– IV		**BASE BAND CODING TECHNIQUES**		
28	L28	**Baseband Decoder**, Noise in communication systems	BB	Tx1/pp293-296, Tx3/pp30-33
29	L29	Correlator Type Receiving filter	BB	Tx1/pp296-302, Tx2/pp84-86
30	L30	Matched filter	BB	Tx1/pp302-309, Rx4/pp459-476 Tx3/pp122-125, Tx2/pp86-96
31	L31	Equalizing filter	BB	Tx1/pp310-328, Tx3/pp125-126
32	L32	Signal and system design for ISI elimination, Implementation of equalizing filter & Decision Feedback equalizer	LCD	Tx1/pp311-336, Rx4/pp460-463
33	L33	Eye Pattern & Synchronization	LCD	Tx1/pp336-341, Tx3/pp151-152 Rx4/pp192-193, Tx2/pp261-262
34	L34	Detector & Maximum likelihood detector	BB	Tx1/pp341-346, Tx2/pp73-77
35	L35	Error Probability, Binary baseband signal & M-ary orthogonal signal	OHP	Tx1/pp346-354, Rx2/pp210-213
36	L36	M-ary orthogonal signal & Figure of merit for digital detection	BB	Tx1/pp 354-360, Rx3/pp221-223
UNIT V		**BANDPASS SIGNAL TRANSMISSION AND RECEPTION**		
37	L37	Memory less modulation methods, Representation and spectral characteristics: – Amplitude Shift Keying (ASK)	BB	Tx1/pp373-376, Tx3/pp175-177
38	L38	PSK (Phase Shift Keying)	OHP	Tx1/pp376-378, Tx3/pp173-175
39	L39	QAM (Quadrature Amplitude Modulation)	OHP	Tx1/pp378-381, Rx1/pp276-280 Rx4/pp373-376
40	L40	QPSK (Quadrature Phase Shift Key)	OHP	Tx1/pp381-390, Rx4/pp370-373
41	L41	FSK (Frequency Shift Keying)	OHP	Tx1/pp390-393, Tx3/pp175
42	L42	Band pass receiving filter & error Performance of band pass system	BB	Tx1/pp435-437, Rx4/pp380-382
43	L43	Coherent detection system ASK ,Binary FSK, Binary PSK	BB	Tx1/pp437-441, Tx2/pp275-283 Tx3/pp183-194, Rx4/pp494-500
44	L44	M-ary PSK ,QAM ,QPSK, MSK	LCD	Tx1/pp442-454, Tx2/pp283-300
45	L45	Non-Coherent Detection System, **Performance evaluation of Communication systems**	BB	Tx1/pp454-484, Tx2/pp300-310 Tx3/pp194-204, Rx4/pp500-508

UNIT-I DIGITAL COMMUNICATION SYSTEM

1. What is PWM?(R) **(Nov – 97)**

Pulse Width Modulation (PWM), where, width of the pulse of the career signal changes in accordance to the amplitude of the message signal. It is also referred as Pulse Duration Modulation (PDM).

2. Briefly explain message recovery from PAM.(U) **(Apr – 98)**

The PAM signal is passed through the reconstruction filter. The filter integrates amplitude of PAM pulses. Amplitude smoothing of the reconstructed signal is done to remove the amplitude discontinuities due to pulses.

3. How is PDM wave converted into PPM system? (U) **(Apr – 97, Oct-98)**

The PDM signal is given as a clock signal to monostable multivibrator. The multivibrator 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 this 'width' of PDM pulse. The position of the PPM pulse depends upon input signal amplitude.

4. Give the advantages, dis-advantages and applications of PAM signal. (R) **(Dec-09)**

Advantage

1. PAM easily generated and detected.
2. PAM is the base for PCM, DM and ADM.

Dis-advantage

1. Bandwidth required for PAM transmission is large when compared to its maximum frequency content.
2. Noise interference is high in PAM signal and cannot be easily removed.
3. Peak power required to transmit PAM varies with modulating signal (message signal).

5. Give the applications of PAM signal. (R) **(Dec-07)**

1. Used for short distance and simple communication.
2. Used in instrumentation system.
3. Used in analog-to-digital converters.

6. What is pulse position modulation. (R) **(May-08)**

The amplitude and width of the pulses are kept constant, while the position of each pulse, with reference to the position of a reference pulse, is changed according to the instantaneous sampled value of the modulating signal.

7. Differentiate: Noise and Fading. (R) **(Dec-08)**

S.No	Noise	Fading
1	It is an unwanted signal that tends to interference with the required signal.	The signal is randomly attenuated due to random or semi-periodic variations in the channel. This is called fading.
2	The noise can have particular range of frequencies depending upon its source.	Fading is frequency dependent and different frequency components are affected unequally.
3	Effect of noise can be minimized by using appropriate filters.	Effects of fading can be reduced by employing Automatic Gain Control (AGC)

8. What is meant by distortion less transmission? (Applying) **(Dec-10)**

For distortion less transmission, the transfer function of the system is given as,

$H(\omega) = K\, e^{-j\omega t0 \text{<- linear phase shift}}$

K =constant magnitude response

Above transfer function imposes two requirements on the system:

1. The system response must have constant magnitude response.
2. The systems phase shift response must be linear with the frequency.

9. Which parameter is called figure of merit of a digital communication system and why? (U) (**Dec-10)**

The ratio E_b/N_0 or bit energy to noise power spectral density is called figure of merit for digital communication systems.

Reason

1. Energy is calculated for bit. Hence E_b/N_0 allows us to compare different systems at bit level.
2. E_b/N_0 is also unit less as does S/N.
3. Bit energy (E_b) can be calculated easily for digital data.

10. Give an example each for time limited and time unlimited signals. (R) **(May-11)**

i) Time limited signals – Rectangular pulse, triangular pulse.
ii) Time unlimited signals – Sinusoidal signal, exponential signal and step signal.

11. Give advantages and disadvantages of digital communication. (R) **(May-11, Dec-12)**

Advantage:

i) Speech, video and other data can be transmitted simultaneously.
ii) Wide dynamic range is possible since data is digital.
iii) Digital communication systems are simpler and cheaper.

Dis-advantage

i) Data rates are very high because of digital conversion. This requires high bandwidths.
ii) Digital communication requires synchronization.

12. Define half power bandwidth. (R) **(Dec-11)**

Half power bandwidth is the bandwidth where PSD of the signal drops to half (3 db) of its maximum value. It is also called 3-db bandwidth.

13. Define signal space and bandwidth. (R) **(May-11)**

Signal space: The digital symbols modulate orthogonal carriers. If these carriers are represented by vectors, they are perpendicular to each other and form a multidimensional space. It is called signal space. Each digitally modulated carrier is represented as a point in this signal space.

Bandwidth: The bandwidth can be of various types. They are half power bandwidth, Null-to-null bandwidth, Fractional power bandwidth, absolute bandwidth, noise equivalent bandwidth and bounded PSD

14. Define channels and its types. (R) **(June 17)(Dec 2016)(Dec-11)**

Its is a physical medium between transmitter and receiver. Any device linking transmitter with receiver can be called a channel.

1) Additive noise channel 2) Linear filter channel 3) Linear time variant filter channel 4) Gaussian channel

15. Define measure of information. (R) **(May-12)**

If the communication system transmits messages m1,m2,m3….. with probabilities of occurrence p1,p2,p3……. Then the amount of information transmitted through the message m_k with probability p_k is given as, $\mathbf{I_k = \log_2(1/ p_k) = - \log_2(p_k)}$. The unit of information is bit.

16. What is meant by symmetric channel. (R) **(May-12)**

The channel is symmetric if,

i) The channel is invariant under the group of symmetry operations.
ii) The channel is invariant to interchanging the input symbols and interchanging the output symbol.

The channel is said to be symmetric if the set of outputs can be partitioned into subsets in such a way that for each subset the matrix of transition probabilities has the property that each row is the permutation of each other row and each column is the permutation of each other column.

17. How can BER of a system be improved. (R) **(Dec-12) (Nov/Dec 2012)**

i) Increasing the transmitted signal power.
ii) Employing the modulation and demodulation techniques.
iii) Employing suitable coding and decoding methods.
iv) Reducing noise interference with the help of improved filtering.

18. What are the different modulation techniques used in analog pulse communication systems? (R)

The modulating signals can modulate amplitude, width (Duration) or Position of the pulse. Depending upon the modulation, three techniques are possible

- Pulse Amplitude Modulation (PAL)
- Pulse Width Modulation (PWM)
- Pulse Position Modulation (PPM) **(Dec-07, May-11)**

19. What are the applications of PWM? (R)

- PWM is used for asynchronous transmission over noisy channel.
- PWM is used to generate PPM It is also used in Motor control

20. Compare different modulation techniques used in digital communication system.(U)

S.No	PAM	PWM	PPM
1	Amplitude of the pulse is proportional to amplitude of the Modulating signal	Width of the pulse is proportional to amplitude of the Modulating signal	Relative position of the pulse is proportional to amplitude of the Modulating signal
2	Bandwidth of the transmission channel depends on width of the pulse	Bandwidth of the transmission channel depends on rise time of the pulse	Bandwidth of the transmission channel depends on rising time of the pulse
3	The instantaneous power of the transmitter varies	The instantaneous power of the transmitter varies	The instantaneous power of the transmitter remains constant
4	Noise interference is high	Noise interference is minimum	Noise interference is minimum

21. List out the applications of Dimensionality theorem(R)

- It is used to calculate storage space required to store digital signal
- It is used to estimate bandwidth of the signal

22. Why do we go for Gram-Schmidt Orthogonalization procedure?(U)(KSRCE June 15 & May 2018)

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

23. State the advantages and disadvantages of the digital communication system over analog communication systems. (U) (May/June 2013)

Advantages

1. Speech, video and other data can be transmitted simultaneously
2. Wide dynamic range is possible since data is digital.
3. Digital communication systems are simpler and cheaper.

Disadvantages

1. Data rates are very high because of digital conversion these requires high bandwidth
2. Digital communication requires synchronization.

24. State the classifications of the channels.(U) (May/June 2013)

The channel can be classified as follows

1. frequency bands and applications
2. propagation modes and transmission media

25. Draw the typical communication system. (R)

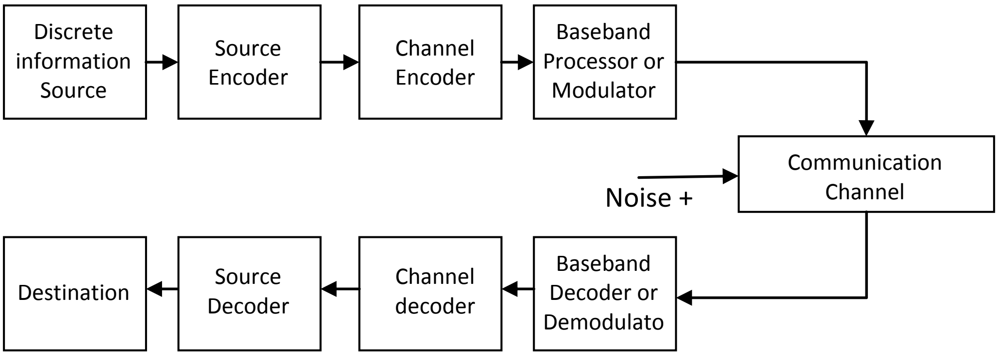


Figure: Typical Communication System

26. List out the mathematical model of channel. (U) (KSRCE June 2015 & May 2018)

1. Additive White noise
2. Linear Filter Channel
3. Linear Time variant filter channel

27. State central limit theorem. (R) (Dec 2016)

The central limit theorem shows that the distribution of the summed phenomenon in such cases is very close to Gaussian, even if the individual distributions are far from Gaussian.

Let X1,X2,…….X_N be independent random variables with means μ1, μ2….. μ_N and variances σ_1^2, σ_2^2…… σ_N^2 respectively. Then the pdf of, $Z=\sum_{i=1}^{N} X_i$

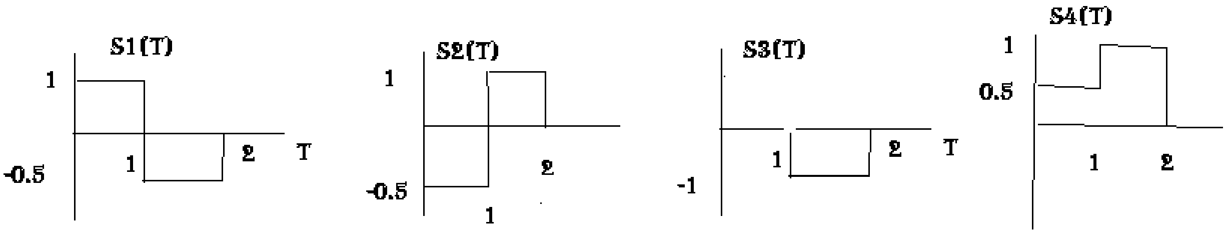
28.Why are mathematical models for communication channels necessary? (June 17)

A General communication model consists of transistor, receiver and channel. Both Transmitter and receiver are completely designed and analyzed by mathematical models. To analyze complete communication we need to fine the mathematical model for communication channel necessary.

16 Marks

1. Explain different channel models. (R) (12) (Dec-11,Dec-12)
2. Briefly Discuss the Digital Communication Design objectives and its constraints with example. **(June 17)**
3. Illustrate the geometric representation of signals when n=2 and m=3.(U) (KSRCE June 2015)
4. Explain binary symmetric channel and Gaussian channel with their mathematical models.(10) (U) (Dec-10)
5. Classify channels. Explain the mathematical model of any two communication channels. (U) (Dec-10,11,12. May-11,12).
6. Give the model of discrete-time memory less Gaussian channel and derive channel capacity for band limited additive white Gaussian channel.(10) (U) (Dec-10)
7. Draw the block diagram of digital communication systems and explain each block detail.(R)(KSRCE June 2015)(May-12) (Dec 2016)

8. Compare PAM, PPM and DPM. (8) (U) (May-11)
9. With Neat circuit diagram, explain the detection of PWM signals. (U) (8) (Dec-07)
10. Briefly explain the functional description of Digital communication systems. (R) (12) (Dec-10, Aug-06)
11. Draw a neat block diagram of a typical digital communication system and explain the function of the key signal processing blocks. (R) (16) (May-12, Dec-10)
12. Distinguish between baseband and Band pass signaling. (6) (R) (Dec-10)
13. With suitable equation explain the geometric representation of signals.(10) (R) (Aug-07, May-12)
14. Explain in detail the Gram-schmidt orthogonalisation procedure. (16) (Ana)(June 2015) (Dec 2016) **(Jun 17)**
15. Explain any analog pulse communication systems? (U) (May-13)
16. Discuss the characteristics of various discrete communication channels? (U) (May-13)
17. State the advantages and dis-advantages of a digital system. (U) **May 2018 (June 17)** (Dec-12)
18. Explain the vector space concept for signal space representation of digital passband signals. **May 2018**
19. Consider the signals $s_1(t)$, $s_2(t)$, $s_3(t)$ and $s_4(t)$ shown in the figure. Find the orthogonal basis function using Gram Schmidt orthogonalisation process. **(May 2108)**



UNIT-II BASEBAND FORMATTING TECHNIQUES

1. Define Nyquist rate. (R) (KSRCE June 2015) (Apr-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 Nyquist rate.

2. What is meant by aliasing effect? (R) (Apr-97,Dec-05)

Aliasing effect takes place when sampling frequency is less than Nyquisy rate. Under such condition, the spectrum of the sampled signal overlaps with itself. Hence, higher frequencies, takes the form of lower frequencies. This interference of the frequency components is called as aliasing effect.

3. State sampling theorem? (R) (June 17) (Dec 06, 08, 09, May 07, 09, 12)

A continuous time signal can be completely represented in its samples and recovered back if the sampling frequency fs>=2w. Here fs, is the sampling frequency and w is the maximum frequency present in the signal.

4. What is meant by quantization? (R) (May 12)

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

5. State band pass sampling theorem. (R) (May 04, 06)

The band pass signal x(t) whose maximum bandwidth is 2W can be completely represented into and recovered from its samples, if it is sampled at the minimum rate of twice the bandwidth.

6. What should be the pass band for antialiasing and smoothing filters used with pulse modulation / demodulation systems?(U) (Dec 04)

i) Antialiasing filter is used before sampling. The filter should band limit the signal to a maximum signal frequency of 'W' Hz. Hence its pass band should be 'W' Hz.

ii) Smoothing filter is used after reconstruction or interpolation. It should successfully pass all the frequencies of O to 'W' Hz and block frequencies greater than 'W' Hz.

7. What is the interpolatory property for sinc function? (Applying) (May 05)

The sinc function is used for interpolation of signal from its samples.

- The signal is given as,
  $x(t) = \sum_{n=-\infty}^{\infty} x(nTs)\mathrm{sinc}(2Wt-n)$, Here Fs = 2W, Hence Ts = 1/2W,
  $x(t) = \sum_{n=-\infty}^{\infty} x(nTs)\mathrm{sinc}(t/Ts-n)$
- Above equation shows that sinc function will assume zero value when t = +/- Ts, +/- 2Ts………

- Thus the main lobe of sinc function contributes for interpolation. It is weighed by x(n Ts).. i.e. sample value.

8. What do you understand by the term aliasing? (R) (Dec 05)

Aliasing effect takes place when sampling frequency is less than Nyquist rate . Under such condition, the spectrum of the sampled signal overlaps with itself. Hence, higher frequencies, takes the form of lower frequencies. This interference of the frequency components is called as aliasing effect.

9. A band pass signal has the spectral range that extends from 20 kHz to 82 kHz. Find the acceptance range of sampling frequency fs. (Dec 05)(Applying)

Here BW = 82 kHz-20 kHz = 62 kHz

fs = 2XBW = 2X 62 kHz = 124 kHz

10. What is the SNR of PCM system if number of quantization levels is 2^8? (May 06) (Applying)

Quantization levels = $2^v = 2^8$, v =8.

(S/N)dB = 4.8 + 6v dB = 4.8 + 6 x 8 = 52.8 dB

11. Define quantization error. (May 07)(U)

Quantization error : Because of quantization, inherent errors are introduced in the signal. This error is called quantization error. It is expressed mathematically as,

E= Xq(nTs)-X(nTs), Here,Xq(nTs) is quantized value of the signal., X(nTs) is the value of the sample, before quantization.

12. A message has zero mean value and a peak value of 10 V. It is to be quantized using a step size of 0.1 V with one level coinciding to 0 V. Find number of bits required for encoding the quantized signal. (Applying) (May 07)

Step size (∂) = 2Xmax/q

Here Xmax is peak amplitude of the signal. , 0.1= 2x10/q, Therefore, q = 200 levels.

Number of bits (v) = $\log_2 q = \log_{10} q / \log_{10} 2$ = log 200/ log 2 = 7.643 = 8 approx.

13. For a uniform quantizer, discuss the way in which the number of quantization levels (L) influence the bandwidth and the quantization noise. (Applying) (May 08)

Number of levels L = 2^v , v = $\log_2 L$

Bandwidth = vW = W $\log_2 L$

Quantization noise ($\partial^2/12$) = $(2Xmax/q)^2/12$, since ∂ = (2Xmax/q)/12

= $X^2max/3q^2 = X^2max/3L^2$, Since levels q = L

14. Discuss the need for non-uniform quantization of speech signal. (U)(May 08)

In non-uniform quantization, the step size is not fixed. It varies according to certain law or as per input signal amplitude. The transfer characteristic and error in non-uniform quantization. For low input signal levels the step size is small. Hence quantization error is also small at these inputs. Therefore signal to quantization noise power ratio is improved at low signal levels. Step size is higher at high input levels. Hence signal to noise power ratio remains almost same throughout the dynamic range of quantizer.

15. Outline the principles of compander used for speech signal. (U) (May 08)

Normally we don't know how the signal level will vary in advance. Therefore the non-uniform quantization becomes difficult to implement. Therefore the signal is amplified at low signal levels and attenuated at high signal levels. After this process, uniform quantization is used. This is equivalent to more step size at low signal levels and small step size at high signal levels. At the receiver a reverse process is done. That is signal is attenuated at low signal levels and amplified at high signal levels to get original signal. Thus the compression of signal at transmitter and expansion at receiver is called combinely as companding.

16. Define quantization noise. (U) (Dec 08)

When the signal is converted from analog to digital form, the analog sample amplitude is assigned the nearest available quantization amplitude level. The difference between quantized value and actual value of the sample introduces permanent distortion in the signal. It is called quantization error or quantization noise. Thus,

Quantization error or noise, E= Xq(nTs)-X(nTs)

Hence, Xq(nTs) is quantized value of sample and X(nTs) is actual value of the sample.

17. Mention two merits of DPCM. (U) (Oct 98, Nov 97)

i) Bandwidth requirement of DPCM is less compared to PCM.

ii)Quantization error is reduced because if prediction filter.

ii) Number of bits used to represent one sample values are also reduced compared to PCM.

18. What is the main difference in DPCM and DM ? (U) (Oct 98)

DM encodes the input one sample by only bit. It sends the information about $+\partial$ or $-\partial$, i.e. step rise or fall. DPCM can have more than one bit for encoding the sample. It sends the information about difference between actual sample value and predicted sample value.

19. Write an expression for bandwidth of binary PCM with N messages each with a maximum frequency of fm Hz. (U) (Apr 98)

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

BT >= N.v.fm , Here v.fm is the bandwidth required by one message.

20. Mention the use of adaptive quantizer in adaptive digital waveform coding schemes. (R)(Apr 97)

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

21. What do you understand from adaptive coding?(U) (Nov 97)

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 used to represent the sample value. Adaptive coding is used for speech coding at low bit rates.

22. The signal to quantization noise ratio in a PCM System depends on what quantity? (Applying)

The signal to quantization noise ratio in PCM is given by,(S/N)dB<=(4.8+6v)dB

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

23. For the transmission of normal speech signal in the PCM channel, give the BW requirement. (Applying) (Dec 2016)

Bt = vW, Bt =8x3.4Hz i.e. 27.2 kHz

24. What is meant by adaptive delta modulation? (U) (APR 98)

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. Delta modulation encodes one bit per sample. Hence signaling rate is reduced in DM.

25. What are the two types of quantization errors that occur in delta modulation((U) (KSRCE June 2015) (May 10)

1) Slope overload error: The step size of quantization is not enough to follow the large changes input signal. Hence there is difference between approximated signal and input signal. It is called slope overload error.
2) Granular noise: The step size is to large, hence approximated signal cannot follow the small variations in input signal, there is large change in approximated signal. It is called hunting or granular noise.

26. State the advantage of adaptive delta modulation over delta modulation?(U) (May 10)

1) ADM eliminates slope overload error and granular noise.
2) ADM has wide dynamic range.
3) Bandwidth utilization is better.

27. Why is pre-filtering done before sampling? (Applying) (Dec 10)

- The signal must be limited to some highest frequency 'w' Hz before sampling. Then the signal is sampled at the frequency of fs = 2W or higher.
- Hence the signal should be pre-filtered (low pass filtered) to eliminate any frequency components higher than 'W' Hz.
- If the signal is not pre-filtered, then frequency components higher than 'W' Hz will generate aliasing in the sampled signal spectrum.

28. Define quantization noise power.(U) (Dec 10)

Quantization noise power is the noise power due to quantization noise. Let the quantization noise have the pdf of Fe(€). Then quantization noise power is given as, $E(€^2)=\int_{-\infty}^{\infty} €^2 F_€(€)d€$

29. Compare uniform and non-uniform quantization. (U)(Dec 11)

S.No.	Uniform Quantization	Non-uniform Quantization
1.	The quantization step size remains same throughout the dynamic range of the signal.	The quantization step size varies according to specific law depending upon amplitude of the input signal.
2.	The signal to noise ratio varies with input signal amplitude	The signal to noise ratio can be maintained constant with non-uniform quantization.

30. What is meant by temporal waveform coding? (U) (Dec 11)

- The signal which is varying with time can be digitized by periodic time sampling and amplitude quantization. This process is called temporal waveform coding.
- The signal such as speed, video etc. can be coded using temporal waveform coding.
- PCM, DM, ADM, DPCM are some of the temporal waveform coding techniques.

31. What is natural sampling?(R) (May 11) (May/June 2013)

The sampled pulse of natural sampling has a finite width T. The top of the sampled pulse follows the same amplitude variations of original signal over its width of T. The naturally sampled pulse appears to be chopped off from the original signal.

32. State the principle of model based encoding.(R) (Dec 11)

- The signal is characterized in various parameters. These parameters represent the model of the signal.
- The parameters are encoded and transmitted to the receiver. The receiver synthesizes the signal from encoded parameters. This is called model based encoding.
- Linear predictive coding (LPC) is an example of model based coding

33. What is meant by PCM?(R)

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.

34. What are the two fold effects of quantizing process.(R)

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.

35. What is the width of the eye?(R) (May/June 2013)

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

36. State any two non-uniform Quantization rules? (Applying) (May/June 2013)

- µ law companding for speed signal. They are normally used for speech and music signals

$$Z(x) = S_{gn}(x)\frac{ln\,(1+\mu\,1*1)}{\ln(1+\mu)}\;1*1\;\leq 1$$

- **A law of compounding**

$$Z(x) = \begin{cases} \dfrac{A\,|x|}{1+\ln A} & for\; 0 \leq |x| \leq \dfrac{1}{A} \\ \dfrac{1+\ln(A\,|x|\,)}{1+\ln A} & for\; \dfrac{1}{A} \leq |x| \leq 1 \end{cases}$$

if A=1, then we get uniform Quantization, A law and µ law companding is used for PCM telephone systems.

37. An analog waveform with maximum frequency content of 3KHz is to be transmitted over an M ary PCM system, Where M=16, What is the minimum number of bits/sample that should be used in digitizing the analog waveform? (Nov/Dec 2012) (Applying)

$l > \log_2 (1/0.02) = 5.6.$

38. Differentiate the principles of temporal waveform coding and model based coding?(U) (Nov/Dec 2012) (Dec 2016)

Temporal waveform coding	Model based coding
Temporal waveform encoding works under the principle of PCM, DM, ADM and DPCM.	The signal is characterized in various parameters. These parameters are encoded. They work under the principle of linear predictive coding.
Here the time domain waveform is encoded	The decoder at the receiver synthesized the signal from these parameters. the parameter represents the modes of the signal/
Bit rate is HIgh	Bit rate is Low
Bit allocation depends on their domain features	Bit allocation depends on the parameters of the signal

39. List two drawbacks of delta-PCM technique. (May 2018)

1. Granular noise
2. Slope overload error

16 marks

1. Show that a bandlimited signal of finite energy Which has no frequency component higher than W Hz is completely described by specifying values of the signals at instants of time separated by 1/2W seconds and also show that if the instantaneous values of the signal are separated by intervals larger than 1/2W seconds, they fail to describe the signal. A bandpass signal has spectral range that extends from 20 to 82 KHz. Find the acceptable range of sampling frequency fs. (8) (Applying) (May 05,06)

2. State the Nyquist sampling theorem. Demonstrate for an analog signal s(t) having a Fourier transform X(f) which is zero outside the interval [-fm<f<fm]. (16) (Applying) (Dec 10)
3. Write in detail about Pulse Amplitude Modulation (PAM). (U) (12)(May 05)
4. Distinguish between instantaneous sampling, natural sampling and flat top sampling with functional block diagram, explain the working of a circuit that provide flat top sampling. (8) (Applying) (May-05)
5. Explain the following sampling techniques with necessary waveforms. a)Impulse Sampling b. Natural Sampling (U). (Dec 11)
6. Explain what is natural sampling and flat-top sampling. (U) (6) (May 12)
7. A television signal has a bandwidth of 4.5 MHz. This signal is sampled, quantized and binary code to obtain a PCM signal.(12) (Dec 11) (Applying)
8. Determine the sampling rate if the signal is to sampled at a rate 20% above Nyquist rate. (Applying)
9. If the samples are quantized into 1024 levels, determine the number of binary pulses required to encode each sample. Determine the binary pulse rate of the binary coded signal and the minimum bandwith required to transmit this signal. (Applying)
10. With neat block diagram explain pulse code modulation and demodulation system.(10) (U) (May 12)
11. With the help of neat diagrams, explain the transmitter and receiver of pulse code modulation.(10)(U) (May 10)
12. Explain in detail Quantization Noise and Signal to Noise Ratio. (12) (U) (May 06,Dec 05,08) **(May 2108)**

13. The information in an analog signal voltage waveform is to be transmitted over a PCM system with an accuracy of +/-0.1% (full scale). The analog voltage waveform has a bandwidth of 100 Hz and an amplitude range of -10 to +10 volts.(16) (Dec 08) (Applying)
    a) Determine the maximum sampling rate required.
    b) Determine the number of bits in each PCM word.
    c) Determine minimum bit rate required in the PCM signal.
    d) Determine the minimum absolute channel bandwidth required for the transmission of the PCM signal.
14. A sinusoidal signal is transmitted using PCM. An output SNR of 55.8 dB is required. Find the number of representation levels required to achieve this performance. (6) (Dec 05) (Applying)
15. Derive expression for the quantization noise and signal to noise ratio in a PCM system using uniform quantizer.(10) (Dec 05) (KSRCE June 2015) (Dec 2016) (Applying)
16. What is meant by compander? What are the two type of compression?(8) (May 04) (U)
17. Explain the non-uniform quantizer.(8) (May 10,11)(U)
18. Explain temporal waveform encoding-delta modulation.(16) (U) (May 04, 05, 10, 12)
19. Explain temporal waveform encoding-adaptive delta modulation.(16)(U) (Dec 04, May 11)
20. Explain the operation of delta modulation systems. (10) (KSRCE June 2015) (May 10)(U)
21. Explain the noises in delta modulation systems. How to overcome this effect in Delta modulation?(U) (Dec 16)
22. Write notes on temporal waveform coding.(8) (U) (May 11)
23. Explain temporal waveform encoding-differential delta modulation.(16) (U) (Dec 04, May 10)
24. Explain the working of Differential PCM and hence drive the expression for signal to noise ratio.(12) (May 10) (Dec 12, May/June 2013) (Applying)
25. Explain DPCM system. Derive the expression for slope overload noise of the system. Show that SNR of DPCM is better than that of PCM.(16) (Applying) (Dec 12)
26. Explain in detail spectral waveform encoding.(12)(U) (May 11, Dec 12)
27. Explain a spectral waveform encoding process. (8)(U) (May 11) (May/June 2013)
28. Explain sub band coding.(8) (U) (Dec 12)
29. Compare various speech encoding methods. (8) (U)(May 11, Dec 10, 11, 12, 13)
30. Explain in detail the various source coding techniques for speech signal and compare their performance.(16) (Dec 10, 11, 12, May 11) (Applying)
31. Explain model based encoding.(U) (May/June 2013)
32. Compare all Digital Modulation Techniques.(U) (KSRCE June 2015)
33. Explain the principle of LPC coder with diagram. (U) (Dec 2016)
34. Find out the number of quantization levels for sending 8 bits per sample if more bits are used for the encoding purpose. What will be the effect on the bandwidth? If fewer bits are used, comment on the reconstruction of the signal. **(June 17)**
35. A signal in the frequency range 300 to 3300Hz is limited to peak to peak swing of 10V. it is sampled to 8000 samples/s and the samples are quantized to 64 evenly spaced levels. Calculate the bandwidth and ratio of peak signal power to rms quantization noise if the quantized samples are transmitted either as binary pulses of as four level pulses. **(June 17)**

36. In a single integration delta modulation scheme. The voice signal sampled at a rate of 64KHz. The maximum signal amplitude is 1 volts and voice signal BW is 4KHz. Determine (MAY 2108)

1. The axiom value of step size to avoid slope overload.
2. Grannular Noise
3. Signal Power
4. Signal to noise ratio
5. Assuming that noise signal amplitude is uniformly distributed in the range (-1, 1). Determine the signal to noise ratio.

37. What is meant by ideal nyquit channel. **(May 2108)**

UNIT-III BASEBAND FORMATTING TECHNIQUES

1. Define code efficiency. (R) (June 17)

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 = Message bits/Transmitted bits = k/n

2. What is meant by systematic and nonsystematic codes? (U)

In a systematic block code, message bits appear first and then check bits. In the nonsystematic code , message and check bits cannot be identified in the code vector.

3. 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.

4. What are the error detection and correction capabilities of Hamming codes? (U) (May-09)

The minimum distance (d_{min}) 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 d_{min}=3.

5. What is meant by cyclic code?(R)

Cyclic codes are the subclass of linear block codes. They have the property that a cyclic shift of one codeword produces another codeword. For example consider the codeword.

$X = (x_{n-1}, x_{n-2}, \dots x_1, x_{n-1})$

Let us shift above code vector to left cyclically,

$X' = (x_{n-2}, x_{n-3}, \dots x_0, x_1, x_{n-1})$, Above code vector is also a valid code vector.

6. How syndrome is calculated in Hamming codes and cyclic codes? (R)(Dec-04)

In Hamming codes the syndrome is calculated as,

$$S = YH^T$$

Here Y is the received and H^T is the transpose of parity check matrix. In cyclic code, the syndrome vector polynomial is given as,

$$S(p) = rem\left[\frac{Y(p)}{G(p)}\right]$$

Here $Y(p)$ is received vector polynomial and $G(p)$ is generator polynomial.

7. What is BCH code? (R)(May-06, Dec-09)

BCH codes are most extensive and powerful error correcting cyclic codes. The decoding of BCH codes is comparatively simpler. For any positive integer 'm' and 't' (where $t<2^{m-1}$) there exists a BCH code with following parameters:

Block length: $n=2^{m-1}$, Number of parity check bits: $n-k \leq mt$

Minimum distance: $d_{min} \geq 2t+1$

8. What is RS code? (R)(May-05)

These are nonbinary BCH codes. The encoder for RS codes operate on multiple bits simultaneously. The (n,k) RS code takes the groups of m-bit symbols of the incomming binary data stream. It takes such 'k' number of symbols in one block. Then the encoder adds (n-k) redundant symbols to form the codeword of 'n' symbols.

RS code has:

Block length: $n = 2^m-1$ symbols , Message size: k symbols

Parity check size: n-k = 2t symbols , Minimum distance: d_{min} = 2t+1 symbols.

9. What is the difference between block codes and convolutional codes?(U) (KSRCE June 2015) (Dec-05, May-09,12)

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

10. Define constraint length in convolutional codes.(R)

Constraint length is the number of shifts over which the single message bit can influence the encoder output. It is expressed interms of message bits.

11. Define free distance and coding gain. (R)(Dec-05)

Free distance is the minimum distance between code vectors. It is also equal to minimum weight of the code vectors.

Coding gain is used as a basis of comparison for different coding methods. To achieve the same bit error rate the coding gain is defined as,

$$A = \frac{\left(\frac{E_b}{N_o}\right) encoded}{\left(\frac{E_b}{N_o}\right) coded}$$

For convolution coding, the coding gain is given as,

$$A = \frac{rd_f}{2}$$

Here 'r' is the code rate,And 'd_f' is the free distance.

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

The non-zero output of the product YH^T is called syndrome and it is used to detect the errors in y. Syndrome is detected by 'S' and it is given as $S = YH^T$

13. What is convolution code? (R)(May-05,06,12)

Fixed number of inputs are stored in the shift register and they are combined with the help of mod-2 adders. This operation is equivalent to binary convolution and hence it is called convolution coding.

14. What are the fundamental properties exhibited by cyclic codes? (R)(Dec-06)

i) Linearily Property: It states that sum of any two codewords is also a valid codeword. For example if X_1 and X_2 are two codewords,

$$X_3 = X_1 \oplus X_2$$

ii) Cyclic Property: Every cyclic shift of a valid code vector produces another valid codevector.
For example, $X = \{x_{n-1}, x_{n-2}, \dots x_1, x_0\}$
Shifting the bits of above code vector to left cyclically by one bit,

$$X' = \{x_{n-2}, x_{n-3}, \dots x_0, x_1, x_{n-1}\}$$

Here X' is also a valid code vector.

15. Define minimum distance.(R) (May-07)

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

16. What is meant by transparency with respect to line codes? (R)

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.

17. What are the classifications of line codes?(R)

Line codes are classified as
i) Polar ii) Unipolar and iii) Bipolar
Further above categories are classified sa,
i) Return to zero (RZ) ii) Not return to zero (NRZ)
There are additional classifications,
i) Alternate mark inversion ii) Split phase Manchester iii) Polar quaternary iv) Gray coding
v) Phase encoding and vi) M-ary encoding.

18. What are the conditions to satisfy the Hamming code?(R)

Four conditions are

i) Number of check bits, $q \geq 3$
ii) Block length, $n = 2^q-1$
iii) Number of message bits, $k = n-q$
iv) Minimum distance, $d_{min}=3$.

19. What are the components of a convolutional coder?(R)

It consists of i)Shift registers ii) Multiplier iii) Adders iv) Sampler or selector.

20. What is Manchester code?(U)(May-12)

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

21. Define linear block code?(R)(Dec-07)

Consider the block of 'k' message bits, (n-k) parity bits or check bits are added. Hence the total bits at the output of channel encoder are 'n'. Such codes are called (n,k) block codes.

22. What is meant by "pseudoternary signaling"? (U) (Dec-06)

Pseudoternary signaling is nothing but successive 1s are coded with alternate positive and negative pulses. There are no pulses for zeros. Thus there are three voltage levels, +1, -1 and 0. It can be NRZ as well as RZ type.

23. What is hamming distance?(U)(Dec 09)

The hamming distance between the 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,Y is two.

24. State NRZ unipolar format(R)

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

25. State NRZ polar format.(R)

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

26. State NRZ bipolar format.(R)

Binary 0 is represented by no pulse and binary one is represented by the alternative positive and negative pulse.

27. State Manchester format.(R)

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

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

28. Compare Baseband binary PAM and M-ary PAM. (U)

S. No	Baseband binary PAM	M-ary PAM
1.	There are only two level of representation. Thus there are only two symbols.	There are M-levels or amplitude of the waveform. Thus there are 'M' symbols
2.	Symbol duration is T_b	Symbol duration is $T_b \log_2 M$.
3.	For every symbol transmitted power is less and remains constant.	For every symbol transmitted power is more and it varies from symbol to symbol.
4.	Effect of noise is reduced.	Effect of noise is more compared to binary PAM.

29. What are the desirable properties of PAM signals?(U)

- The PAM signal should have adequate timing content, so that clock information can be extracted from the waveform.
- The PAM signal should be immune to channel noise and interference.
- The PAM signal should allow error detection and correction.
- The waveform of the PAM signal should be transparent to the digital data being transmitted.

30. What are the advantages and disadvantages of convolutional codes?(U)

Advantages:

- The decoding delay is small in convolutional codes since they operate on smaller blocks of data.
- Synchronization problem does not affect the performance of convolutional codes.

Disadvantages:

- Convolutional codes are difficult to analyze since their analysis is complex.
- Convolutional codes are not developed much as compare to block codes.

31. Define code efficiency.(R)

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

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

32. State the significance of minimum distance of a Block code?(U)(May/June 2013)

Minimum distance (d_{min}) it is the smallest hamming distance between the rated code vectors. the error detection is possible if the received. Vector is not equal to some other code vectors. This source that the transmission errors in the received code vector should be less than the minimum distance.

33. Define transparency of a line code. Give two examples of line codes which are not transparent?(R)

Line code is set to be transparent if the synchronization between the transmitter and receiver is maintained for any type of input data sequence

Two examples of line codes which are not transparent are: unipolar NRZ, polar NRZ, and polar Quaternary. **(May/June 2013)**

34. Find the hamming distance between 101010 and 010101. If the minimum hamming distance of a (n.k0) linear block code is 3. What is the minimum hamming weight? (Applying)(nov/dec 2013, May 2018)

The two codes are 101010 010101. The hamming distance between the two codes are 6. Given the minimum hamming distance of (a, k) linear block code is 3. Then the minimize hamming weight also 3. Hence the Minimum hamming distance Equal to minimum hamming weight.

35. State any four desirable properties of a line code? (R)(nov/dec 2013)

The analog waveform are converted to digital signal by PCM, DM, ADM & DPCM techniques. This digital data can be represented by different forms of waveforms. These waveforms are commonly known as digital data formats or their representation called as line coding.

Properties of line coding

1. The signal should have adequate tuning content, so that the clock information can be extracted from the waveform.
2. The signal should allow error correction and error detection.
3. The signal should be immune to channel noise and interference.
4. The waveform of the signal should be transparent to the digital dated being transmitted.

36. How syndrome is calculated in linear block code and cyclic code. (R)(KSRCE June 2015)

Linear Block Code [S] =[r] [H ^{T}] **Cyclic Code = s(x) = rem [r(x) / g(x)]**

37. What is viterbi decoding scheme? (R) (Dec 2016)

A **Viterbi decoder** uses the Viterbi algorithm for decoding a bitstream that has been encoded using convolutional code or trellis code.

A Viterbi algorithm applies maximum likelihood principles to limit the comparison to 2^{Kk} surveying paths.

38. Differentiate convolutional code and block code. (Jun17)

LINEAR BLOCK CODES:

In coding theory, the linear block code generally referred as a error correcting code for which the obtained resultant codeword is the linear combination of any two codewords. In simple words, the linear block code possesss the linearity property that is the sum of any two codewords is also a codeword. These linear block codes are divided in to the block odes and convolutional codes, eventhough turbo codes is the combination of these two types. More efficient encoding and decodig algorithms are provided by the linear codes when compared with the other codes.

CONVOLUTIONAL CODES

The main principle involved in the convolutional code is the weighted sum of the various input message symbols is the resultant codeword symbol. This resemblance the convolution used in the LTI systems where we find the output of a syatem by knowing the impulse response and the respective input. So hence the output of a convolutional encoder can be obtained by the convolution of the input bits with the states of the convolution encoder registers.(http://en.wikipedia.org/wiki/convolutional codes).

16 Marks

1. Compare the power spectra of different binary formats.(6) (U)(Dec-06)
2. Derive the power spectral of polar codes and on-off codes. Discuss their characteristics.(16)(U) (May-11)
3. List and explain the properties of line codes.(8)(U)(Dec-11)
4. Derive the expression for power spectral density of unipolar NRZ line code. Hence discuss its characteristics.(16)(Applying)(Dec-12)
5. Derive the power spectral density of polar signaling and explain.(10)(Applying)(May-12)
6. What is meant by free distance of the convolutional code? How does it affect the number of errors that can be corrected and coding again?(6)(Analyzing) (Dec-04)
7. Explain the error detecting and correcting capabilities of linear block code.(6)(U)(May-12)
8. With neat block diagram explain the role of operation of linear block codes in baseband coding techniques.(12)(U)(May-04,05,06,07,09,12, Dec-06,07,09,10,12)
9. Write the generator matrix and parity check matrix of (7,4) hamming code.(6)(Applying)(May-04)
10. For a linear block code, prove with example that:(8)(E)(May-05,06)
    i) The syndrome depends only on error pattern and not on transmitted codeword
    ii) All error patterns that differ by a codeword have the same syndrome.
11. Consider a (7,4) linear block code with the parity-check matrix H given by

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

i) Constuct code words for this (7,4) code
ii) Show that this code is hamming code.
ii) Illustrate the relation between minimum distance and the structure of the parity-check matrix H by considering the code word 0101100.(16)(Applying)(May-07) (Dec 2016)

12. For (6,3) systematic linear block code, the code word comprises I_1, I_2, I_3, P_1, P_2, P_3 where the three parity check bits P_1, P_2, P_3 are formed from the information bits as follows: **(Jun17) (May 2108)**

$P_1=I_1 \oplus I_2$, $P_2=I_1 \oplus I_3$, $P_3=I_2 \oplus I3$.
Find:
i) The parity check matrix ii) The generator matrix iii) All possible code words.
iv) Minimun weight and minimum distance
v) The error detecting and correcting capability of the code.
vi) If the received sequence is 100010 calculate the syndrome and decode the received sequence.(16)(Applying)(Dec-10)

13. Explain in detail coding and decoding of linear block codes.(16)(R)(May-09, Dec-09)
14. What are all the message vectors that can be extracted from the code vector that was generated by $1 + X + X^3$?. (KSRCE June 2015)
15. Explain the encoding and decoding methods for cyclic codes giving proper block diagrams.(6) (May-08)
16. Explain any four characteristics of the following block codes i) BCH codes ii) CRC codes iii) Maximum length codes.(12)(R)(Dec-04)
17. Why are called cyclic codes? Explain with merits and demerits.(8)(U)(May-09)
18. Define convolutional coding. With neat diagram explain its operation in detail.(U) (May-04,05,06,08,09,10,11,12, Dec-04, 05, 07, 08, 09, 10, 12)
19. A convolutional code is described by the following generator sequences: **(Jun17) (May 2108)**
$g^{(1)} = (1\ 0\ 1)$, $g^{(2)} = (100)$, $g^{(3)} = (111)$
i) Draw the encoder to this code ii) Draw the state diagram
iii) If the message sequence is 10110, determine the codeword.(8)(Applying)(Dec-07)
20. A rate 1/3 convolutional coder with constraint length of '3' uses the generating vectors
$g_1 = (100)$, $g_2 = (101)$, $g_3 = (111)$
i) Sketch encoder configuration and prepare the logic table.
ii) Draw the state diagram for the coder.
iii) Determine the d_{free} distance of the coder.(10)(Applying)(Dec-08)
21. What are code tree, code trellis and state diagrams for convolutional encoders?(6)(R)(KSRCE June 2015) (Dec-10) (Dec 2016)
22. Explain the Viterbi algorithm and sequential decoding of convolutional codes.(16)(U)(May/June 2013) (May-08, 05, 06, Dec-08, 09)
23. Draw the block diagram of a convolutional encoder of constraint length 3 and code rate ½.(6)(Applying)(May-08)
24. Explain the transform domain approach analysis of convolutional code.(6)(U)(May-12)
25. Briefly explain the concept of error-free communication.(6)(U)(Dec-11)
26. Explain the classification of line codes with suitable signal formats.(6)(U)(Dec-11)
27. Compare a baseband binary PAM systems with that of M-ary PAM system.(10)(U)(May-08)
28. With various PAM signals discuss the operation of Power spectra of line codes.(12)(Evaluating) (Dec-11,12, May-11,12)
29. Derive and draw the power spectra of a NRZ. a). polar coded waveform b). Bipolar coded waveform. (Applying)(May 13)
30. Explain syndrome circuit for linear block code.(U)(KSRCE June 2015)
31. Design a block code for a message block of size 8 that can correct for single errors.(Applying)(Dec 12)

UNIT-IV BASEBAND RECEPTION TECHNIQUES

1. What is an Intersymbol Interference (ISI) in baseband binary PAM systems?(U)(May-06, Dec-09, 11)

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. Let the transmitted waveform be represented as,

$$X(t) = \sum_{k=-\infty}^{\infty} A_k g(t - kT_b)$$

Here $A_k$is the amplitude of the K^{th} pulse.

And g(t) is shaping pulse.

The output pulse at $t=iT_b$ can be expressed as,

$$y(t_i) = \mu A_i + \mu \sum_{k=-\infty}^{\infty} A_k p[(i-k)T_b]$$

Here T_b is the bit duration and 't_i' indicates instant of the i^{th} pulse. $\mu A_i$is the contribution of i^{th} transmitted bit. The second term in above equation occurs due to filtering nature of the transmitter receiver and channel. The second term represents the residual effect (time spread) of all other bits transmitted before and after t_i. this presence of outputs(second term) due to other bits (symbols) interfere with the output of required bit (symbol). This effect is called Intersymbol Interference.

2. **What are eye patterns? (Nov-97) (U)(KSRCE June 2015)**

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

- Width of eye opening defines the interval overwhich 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.

3. **What is baseband signal receiver?(R)**

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.

4. **What is matched filter?(R)**

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

5. **What is the impulse response of matched filter?(R)**

Impulse response is given as,

$h(t) = 2k/N_0\{x1(T-t)-x2(T-t)\}$

Here T is the period of sampling x1(t) and x2(t) are the two signals used for transmission.

6. **What is the value of maximum signal to noise ratio of the matched filter? When it becomes maximum?(Nov-97)(R)**

Maximum signal to noise ratio of the matched filter is the ratio of energy of the signal to PSD of white noise. i.e.,

$$\rho_{max} = \frac{E}{N_o/2}$$

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

7. **On what factor, the error probability of matched filter depends?(U)**

Error probability of matched filter is given as,

$$P_e = \frac{1}{2} erfc\sqrt{E}/N_o$$

This equation shows that error probability depends only on energy of the signal. It doesnot depend upon shape (waveform) of the signal.

8. **What is correlator?(R)**

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

$$r(t) = \int_0^T f(t)x(t)\,dt$$

Matched filter and correlator are functionally same.

9. **How is eye pattern obtained on the CRO? (R)(May-04,09)**

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

10. **What is the condition for zero inter symbol interference?(R)(May-04, Dec-07)**

Zero ISI can be obtained if the transmitted pulse satisfies the following condition:

Time domain:

$$p[(i-k)T_b] = \begin{cases} 1 \; for \; i = k \\ 0 \; for \; i \neq k \end{cases}$$

Frequency domain:

$$p[(i-k)T_b] = \sum_{n=-\infty}^{\infty} p(f - nf_b) = T_b$$

11. How is the transfer function of the matched filter related to the spectrum of the input signal?(R) (May-04)

It is given as,

$$H(f) = \frac{2k}{N_o} X^*(f) e^{-j2\pi fT}$$

Here X(f) is the spectrum of input signal.

12. A TDM signal with bit time of 0.5 µs is to be transmitted using a channel with raised cosine roll off factor of 0.5. what is the bandwidth required?(Applying)(Dec-04)

T_b=0.5 µs, α=0.5

$B_o = \frac{f_b}{2} = \frac{1}{2T_b} = \frac{1}{2 \times 0.5 \times 10^{-6}} = 1 \times 10^6$, $B = B_o(1+\alpha) = 1\times10^6(1+0.5) = 1.5\times10^6$

13. From the eye pattern, how is the best time for sampling determined?(U)**(Dec-04)**

It is preferable to sample the instant at which eye is open widest. At this instant, the chances of error are minimum.

14. What is the purpose of using an eye pattern?(R)**(Dec-05,12, May-12)**

Eye pattern can be used for:

- To determine an interval over which the received wave can 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.

15. Why do you need adaptive equalization in a switched telephone network?(U)(Dec-05)

In switched telephone network the distortion depends upon

- Transmission characteristics of individual links.
- 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.

16. What is an ideal Nyquist channel?(R)

The ideal Nyquist channel uses sinc pulse for transmission. i.e,

$$p(t) = \frac{\sin(2\pi B_o t)}{2\pi B_o t}$$

Such pulse have the spectrum of,

$$p(f) = \begin{cases} \frac{1}{2B_o} & for -B_o < f \le B_o \\ 0 & elsewhere \end{cases}$$

17. Bring out the difference between carrier recovery and clock recovery.(U)(Dec-06)

S.no.	Carrier Recovery	Clock Recovery
1.	Carrier is required for coherent detection at the receiver.	Clock is required to estimate correct bit timing at the receiver.
2.	M^{th} power loop, Costas loop are used for carrier recovery.	Closed loop bit synchronizer, early-late synchronizer are used for clock recovery.

18. Why do we need equalization in base band pulse transmission?(U)(May-07 & May 2018)

When the signal is passed through the channel, distortion is introduced in terms of i) Amplitude and ii) Delay. This distortion creates the problems 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 correct the channel distortion.

19. What is meant by a matched filter?(R)(May-09)

The shape of the impulse response of the matched filter is similar (or matched) to the shape of the input signal x(t). Hence it is called matched filter.

20. Define error probability.(R)(May-10)

i) Error probability is defined as the number of bits or symbols that are detected wrongly in a given number of total bits or symbols.

ii) For example error probability of 10^{-4} indicates that 1 bit will be detected wrongly in 10000 bits. Here $\frac{1}{10^{-4}} = 10000$.

iii) Error probability is the important measure to evaluate performance of receivers.

21. What is the need for a demodulator in case of baseband signaling when the received waveforms are already in pulse like form? (U) (Dec -10)

i) When the pulsed waveform is transmitted across the channel, noise interferes the signal. This distorts the pulses. Sometimes the amplitude of noise is so high that it is wrongly detected as a signal pulse.
ii) The transmission channel exhibits lowpass characteristic. Because of this portion of each pulse is dispersed over infinite time. Thus in the time frame of each pulse, the small signal portion due to all the other pulses is present. This is called inter symbol interference. It also creates errors in detection.
iii) Hence a detector is required that eliminates interference and detects the pulse correctly.

22. How does pulse shaping reduce ISI? (R) (Dec- 10)

i) The shape of the pulse is selected such that at the instant of detection, the interference due to all other symbols is zero.
ii) The effect of ISI is totally eliminates if signal is sampled at T_b, $2T_b$, $3T_b$, .. and so on.

23. Bipolar pulse waveforms g_i (t) (i=1,2) of amplitude $\pm$1V are received in the presence of AWGN that has a variance of 0.1V^2. Find the optimum detection threshold γ of MAP detector, if the a priori probability is p(g$_1$) = 0.5.(Applying) (Dec-11)

Here $a_1 = 1V$ and $a_2 = -1V$

$\sigma_0^2 = 0.1V^2, \rho(g_1) = 0.5$, hence $\rho(g_2) = 1 - \rho(g_1)$ = 1-0.5 =0.5

$$\gamma = \frac{\sigma_0^2}{a_1 - a_2}\left\{\ln\left[\frac{\rho(g_1)}{\rho(g_2)}\right] + \frac{a_1^2 - a_2^2}{2\sigma_0^2}\right\}$$

$$= \frac{0.1}{1-(-1)}\left\{\ln\left[\frac{0.5}{0.5}\right] + \frac{1^2-(-1)^2}{2\times 0.1}\right\}$$

$$= 0.05\ln[1] = 0$$

24. List the primary causes for the noise in communication system.(R) (Dec-11)

Noise interferes communication system because of

i) Bandlimited nature of the channel.
ii) Environmental effects such as lighting, humidity, temperature etc.,
iii) EMI and RFI
iv) Thermal noise due to electronic components.

25. Define figure of merit in digital communication.(R) (Dec-11)(Dec 2016)

The figure of merit for digital detection is defined as,

$$\gamma = \frac{E_b}{N_0}$$

Here E_b is the bit energy and

$N_0 = \frac{N}{W}$ is noise power within signal bandwidth W.

26. ISI cannot be avoided justify the statement.(R)(May/June 2013)

$$Y\,(t_i) = yA_{i5}\,y\sum_{k=-\infty}^{\infty} A_k P\,[(i-k)]T_b\,, i = 0, \pm1, \pm2, \pm3.,$$

27. State the principle of maximum likelihood detectors. (U) (May/June 2013)(Dec 2016)

The vector x lies in region R_i if ln [f_x (x/m_k)] is maximum, for k=i.

Here f x (x/m_k) is the likely hood function which result when symbol Mk is transmitted. This value is called maximum likelihood and the corresponding detectors which uses this rule is called maximum likelihood detector.

28. A 64 Kbps binary PCM polar NRZ signal is passed through a communication system with raised cosine filter with roll off factor 0.25. Find the bandwidth of the filtered PCM signal.(Applying)(Nov/Dec 2012)

D=64 kbps, W= D(1+r)/2 = 40 kHz. Therefore, compared to 32 kHz PCM Bandwidth, the pulse shaping filter requires 25% more bandwidth to combat ISI

29. State any two applications of eye pattern.(R)(Nov/Dec 2012)

Eye pattern can be used for

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

30. How do the raised cosine spectrums reduces ISI? (U)(KSRCE June 2015)

3. The shape of the pulse is selected such that at the instant of detection, the interference due to all other symbols is zero.
4. The effect of ISI is totally eliminates if signal is sampled at T_b, $2T_b$, $3T_b$, .. and so on.

5. Here the BW is extended from B0 = rb/2 to an adjustable value between B0 and 2B0.

16Marks

1. Explain the various types of synchronization required in digital communication systems. Discuss in detail the closed loop bit synchronization technique used in binary receiver. State clearly the advantages and disadvantages of this systems.(U)(12)
2. Discuss the necessity of synchronization in digital communication. Explain different methods of synchronization in detail.(U)(10)
3. Explain frame synchronization required in time division multiplexing.(8) (U)
4. Discuss the method of bit synchronization.(6)(U)(May-04)
5. With neat block diagram, explain briefly how symbol synchronization is achieved?(8)(U)(May-05)
6. Enumerate on carrier and symbol synchronization.(10)(R)(May-07, 10)(KSRCE June 2015)
7. Explain the operation of Detection-Maximum Likelihood Detector using signal constellation diagram.(8)(U)(Dec-11, May-12)
8. Derive an expression of error probability of matched filter(6)(Applying)(Dec-07, May-08,11)(May/June 2013)
9. Explain the various types of noise present in communication systems.(Jun 17)
10. Obtain the expression for the average probability of symbol error assuming NRZ signaling, if the noise is modeled as AWGN (Additive with Gaussian Noise).(10)(Applying)(May-08)
11. Derive an expression for impulse response of the matched filter.(7)(Applying)(May-08, 10, 11) (Dec 2016)
12. A baseband binary digital communication system transmits data at 1 kbps. The PSD of noise is 10^{-7} W/Hz and the received signal amplitude is 20mV.(16)(Applying)(Dec-08)
    i) Find the error probability for bipolar rectangular signaling.
    ii) If the bit rate is 10kbps to what value must A be adjusted in order to attain the same error probability as in part (i)?
    iii) What is the required channel bandwidth in case (ii)?
    iv) If not more than a 5KHz channel is available, what should be the value of A so that the data rate is maximized and the error probability is same as in part (i)?
13. Derive the expression for error probability of on-off and polar signaling.(16)(Applying) (Dec-11)
14. Explain how a matched filter can maximize SNR for a given transmitted symbol.(6)(Analyzing) (Dec-10)
15. Discuss in detail about the receiving filter-correlation receiver.(10) (R)(Dec-06,10,12)
16. Define a matched filter and compare its functioning with a correlator. (10) (U)(KSRCE June 2015) (Dec-10)
17. Describe the baseband transmission system with a neat block diagram.(16)(U)(Dec-09)
18. Explain in detail M-ary baseband system.(10)(R)(May-10)
19. Sketch the time response and frequency response of signal with raised cosine pulse spectrum.(8)(U)(May-04)
20. What is meant by the ideal Nyquist Channel? What are its merits and limitations?(8)(U)(Dec-04)(May/June 2013) **(May 2108)**
21. Draw the eye pattern and explain the analysis of eye pattern.
22. Obtain an expression for Nyquist criterion for distortion less baseband transmission for zero inter symbol interference.(10)(Applying)(May-05,12,09,10, Dec -05)
23. Draw the block diagram of duo-binary signaling scheme for controlled ISI. Explain the scheme with and without precoder.(10)(U)(May, Dec -12)
24. Summarize inter symbol interference.(R)(KSRCE June 2015)
25. Model the eye patters and explain the analysis of eye pattern.(Analysing)(KSRCE June 2015)(Dec 2016) **(May 2108)**
26. Derive an expression for the probability of error in a matched filter**. (May 2108)**

UNIT –V BANDPASS SIGNAL TRANSMISSION AND RECEPTION

1. **Mention the need of optimum transmitting and receiving filter in baseband data transmission.** (U)(**April -97, Nov-97)**

   When binary data is transmitted over the baseband channel, noise interferes with it. Because of this noise interference, errors are introduced in signal detection. Optimum filter performs two functions while receiving the noise 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.
2. **Define ASK**.(R)(**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?(R)(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 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?(R)(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?(U)(April-97)**

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

6. **What is meant by coherent ASK?(R)(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 synchronous ASK.

7. **What is the major advantage of coherent PSK over coherent ASK?**(U)(**Oct-98)**

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

8. **Explain the model of bandpass digital data transmission system?(R)**

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

9. **Which digital modulation technique gives better error probability?(U)**

Binary PSK gives reduced error probability compared to ASK and FSK. It is given as

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

10. **In minimum shift keying what is the relation between the signal frequencies and bit rate?(U)(May-04)**

Let the bit rate be f_b and frequency of carrier be f_o. Then higher and lower MSK signal frequencies are given as,

$F_H = F_0 + F_b/4$ and $F_L = F_0 - F_b/4$

11. **What do you understand by coherent detection?**(U)**(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.

12. **Write the expression for bit error rate for coherent binary FSK.(R)(Dec-04)**

The bit error rate of coherent binary FSK is given as,

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

13. **Bring out the difference between coherent and noncoherent binary modulation schemes.**(R) (**May-05, 07, 09, Dec-08, 09)**

Coherent Detection:

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.

Noncoherent Detection:

In this method, the receiver carrier need not be phase locked with transmitter carrier. Hence it is also called envelope detection. Non-coherent detection is simple but it has higher probability of error.

14. **What is the error probability of MSK and DPSK?** (R)(**May-05)**

Error probability of MSK: $Pe = \frac{1}{2} erfc \sqrt{\frac{E_b}{No}}$, **Error probability of DPSK:** $Pe = \frac{1}{2} e^{\frac{-E_b}{N_0}}$

15. **Highlight the major difference between a QPSK signal and a MSK signal.(R)(Dec-05)**

MSK signal have continuous phase in all the cases, whereas QPSK signal has abrupt phase shift of π/2 or π.

16. Compare the probability of error of PSK with that of FSK.(R)(May-06)

BPSK: $Pe = \frac{1}{2} erfc\sqrt{\frac{E_b}{No}}$, **BFSK:** $Pe = \frac{1}{2} erfc\sqrt{\frac{0.6E_b}{4No}}$

- For the fixed value of E_b/N_0, error probability of BPSK is less than BFSK.
- For the given probability of error, the E_b/N_0 of BPSK is 3 dB less compared to that of BFSK.

17. What do you understand by continuous phase frequency shift keying?(R)(**May-07)**

In FSK, when the phase change is gradual at the bit transition times, the signal appears to be continuous in phase. This is called continuous phase FSK or CPFSK. To have phase continuity, the two FSK frequencies f_H and f_L must differ by a bit rate of f_b or $1/T_b$.

18. State the difference between coherent and non-coherent binary modulation techniques. (U)(May-06) (May/June 2013)

Coherent Detection:

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.

Non-coherent Detection:

In this method, the receiver carrier need not be phase locked with transmitter carrier. Hence it is also called envelope detection. Non-coherent detection is simple but it has higher probability of error.

19. Compare the probability of error of PSK with that of FSK.(U)(**Dec-06)**

S.No	**Error probability of PSK**	**Error probability of FSK**
1.	$Pe = \frac{1}{2} erfc\sqrt{\frac{E}{No}}$	$Pe = \frac{1}{2} erfc\sqrt{\frac{0.6E}{4No}}$
2.	For same E/N_0, error probability is less.	For same E/N_0, error probability is high.

20. What are the advantages and disadvantages of differential phase shift keying?(U)(**Dec-07)**

Advantages:

- DPSK does not need carrier at its receiver. Hence the complicated circuitry for generation of local carrier is avoided.
- The bandwidth requirement of DPSK is reduced compare to that of BPSK.

Disadvantages:

- The probability of error or bit error rate of DPSK is higher than that of BPSK.
- Since DPSK uses two successive bits for its reception, error in the first bit creates error in the second bit. Hence error propagation in DPSK is more. Whereas in PSK single bit can go in error since detection of each bit is independent.
- Noise interference in DPSK is more.

21. What is signal constellation diagram?(R)(Dec-08) (Dec 2016)

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.(R)(Dec-09) (May/June 2013)

- In QPSK two successive bits in the data sequence grouped together. This combination of two bits forms four distinct symbols. When the symbol is changed to the next symbol the phase of the carrier is changed by 45^0 (0r π/4).
- Became of combination of two bits ther will be four symbols. Hence the phase shift will be π/4, 3π/4, 5π/4 or 7π/4.
- QPSK reduces amplitude variations and required transmission bandwidth.

23. Differentiate baseband transmission from passband transmission.(U)(**May-10)**

S.No	**Baseband Transmission**	**Passband Transmission**
1.	Signal is transmitted without any modulation of high frequency carrier.	The signal modulates high frequency carrier.
2.	Used for short distance transmission.	Used for long distance transmission.
3.	Used for LANs, printers, short	Used for transmission of digital video,

	distance links	data and speech.

24. Define MSK.(R)(**May-10)**

- MSK uses quadrature carriers which are orthogonal and difference between them is minimum.
- There are no abrupt in phase of MSK signal and it appears to be continuous.

25. Define QAM and draw its constellation diagram.(R)(**Dec-10) (May/June 2013)**

QAM: The phase as well as amplitude of the quadrature carriers is modulated. Hence it is called Quadrature Amplitude Phase Shift Keying or simply QAM.

26. A binary frequency shift keying system employs two signalling frequencies f_1 and f_2. The lower frequency f_1 is 1200 Hz and signalling rate is 500 band. Calculate f_2.(Applying)

For binary FSK,

Baud=f_b

F_b=500Hz

Considering the FM modulation index (h) of '1' in FSK,

$$\frac{|f_m - f_s|}{f_b} = h = 1 \text{ (Here h=1)}$$

$f_m - f_s = f_b$, since $f_s = f_1$=1200HZ , f_m-1200 Hz=500 Hz , f_m=1700 Hz ,f2=f_m=1700 Hz

27. Define BER.(R)

BER is bit error rate. It is the number of bits that go in error in specific number transmitted bits. For example 10^{-3} indicates 1/1000, this means one bit goes in every 1000 transmitted bits.

28. What is meant by memoryless modulation?(R)(Nov/Dec 2012) (Dec 2016)

When the digital symbol modulates amplitude, phase or frequency of the carrier without any reference to previous symbol, it is called memoryless modulation. ASK, FSK, PSK, QPSK etc. are memoryless modulation techniques.

29. What are the drawbacks of binary PSK system?(U)(May-12)

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

30. Differentiate QPSK from PSK in terms of transmission bandwidth and bit information it carries.(U)(KSRCE June 2015)

- Bandwidth of PSK is $2f_b$ and bandwidth required is f_b.

  Where, f_b – maximum frequency in baseband signal.
- Bit information carried by BPSK is one bit (0 or 1) at a time.
- Bit information carried by QPSK is dibit (00, 01, 10 & 11) at a time.

31. When M-Ary signalling schemes are preferred over binary signalling schemes and why.(Analyzing)(KSRCE June 2015)

If We use M ary digital signalling the required bandwidth is reduced when compared with binary signalling scheme.

Example . in binary PSK bandwidth required is directional proportional to 1/Tb. In M Ary signalling M = 2^n , T = n Tb. Here the bandwidth required reduced by a factor of ($\log_2 M$)

32. A BPSK system makes errors at the average rate of 100 errors per day. Date rate is 1Kbps. The single sided noise power spectral density is 10-10* W/Hz. Assuming the system to be wide sense stationary, what is the average bit error probability?(Applying) (Nov/Dec 2012)

Bit error Probability, $P_e = \frac{1}{2} erfc \sqrt{\frac{100}{10^{-10}}} = \frac{1}{2}(27.63102)$

33. What is the transmission bandwidth reduction and average signal energy of 256 – QAM compared to 64 QAM. **(May 2018)**

64 QAM – fb / 2. 256 QAM – fb/8 - Bandwidth reductions by 4 times

64 QAM – 0.356 256 QAM – 0.42

16 Marks

1. Explain the error performance of M-ary systems. (R)(8)
2. Compare error performance of M-ary FSK and PSK.(U)(6)
3. Derive the expression for probability of error of BPSK and QAM system.(6)(Applying)(May-10)
4. Derive the bit error probability due to coherent ASK, PSK, QPSK and FSK systems. Compare the performance of these systems.(8)**(**Applying)(KSRCE June 2015)
5. Derive the bit error probability due to QPSK receiver. Compare the performance of QPSK receiver with that of PSK receiver.(8)(Applying)
6. Derive the expression for error-probability of QAM system.(8)(Applying)(May-12)

7. Compare the BER of coherent PSK, coherent QPSK and coherent FSK.(6)(U)(May-04)
8. With necessary equations and signal space diagram, obtain the error probability for MSK systems.(12)(U)(Dec-06)
9. Compare the performance of BPSK with that of BFSK.(4)(U)(May-08)
10. Compare the digital modulation techniques in terms of bit error rate and bandwidth efficiency.(U)(May-09)
11. Compare the performance of various coherent non-coherent digital detection systems.(16)(U)(Dec-12)
12. Explain the principle of BPSK.(6)(R)(Dec-05, 06, 07, 10 11)
13. Explain BPSK system with the help of transmitter and receiver, and state its advantages/disadvantages over other system.(8)(U)(Dec-05, 06, 10, 11)
14. Derive an expression for spectrum of BPSK system and hence calculate the bandwidth required.(6)(Applying)
15. With the help of block diagram, waveforms and expressions explain the operation of DPSK transmitter and receiver.(10) (U) (Dec 2016)
16. What are the advantages and disadvantages of DPSK? What is the bandwidth requirement of DPSK?(8)(U)
17. Write a note on differential phase shift keying.(6) (May-08)(R)
18. Explain the operation of QPSK with neat diagram.(10)(R)(May-04,05,09,11,12. Dec-07,09,10)
19. With the help of block diagram, waveforms and expressions explain the operation of DPSK transmitter and receiver.(10)(Analyzing)(May-04, 05 08, 06, 09, 12)
20. Compare psd and bandwidth requirements of QPSK with that of BPSK.(8)(U)(May-09)
21. Represent QPSK signal in the signal space and find distance between them. What is the significant of this distance?(10)(U)(Dec-10)
22. Explain carrier synchronization in QPSK signal.(6)(R)(Dec-07)
23. Discuss $\pi/4$ shifted QPSK and offset QPSK scheme in detail.(16)(U)(Dec-09)
24. Discuss the operation of QASK or QAM and M-Ary QAM.(10)(U)(May-12)(Jun 17)
25. Explain the difference between QASK and QPSK systems giving corresponding expressions and signal space representations.(10)(U)
26. Explain QASK system with its transmitter, receiver and signal representation.(8)(U)(May-12)
27. What is the bandwidth of the transmitter in terms of input bit duration i.e., input signal bandwidth? Explain the mechanism by which the bandwidth reduction is made possible in QASK system?(12)(U)
28. With neat diagram explain the operation of BFSK.(10)(May-06, 08 Dec-07, 10)(U) (Dec 2016)
29. Compare BFSK and BPSK.(12)(U)
30. Explain non-coherent detection methods of binary frequency shift keying scheme.(12)(Dec 10)(U)
31. Explain the detection of binary FSK signal with block diagram.(7)(U)(Dec-07) (Dec 2016)
32. With relevant expression and block diagrams explain the operation of M-ary FSK transmitter and receiver.(10)(Applying)
33. Determine the bandwidth required for M-ary FSK system. Draw the geometrical representation of M-ary FSK signals and findout distance between the signal points. What is the bandwidth of this system.(12)(Applying)
34. Explain the operation of MSK with neat waveform.(12)(U)(Dec-04,07,08,09 May-05,06,07,08)
35. Binary data are transmitted over a microwave link at the rate of 1 Mbps and PSD of the noise at the receiver input is 10^{-10} W/Hz. For each of the following pairs, determine which one requires more power than the other. Determine the extra average signal power required by the more power consuming scheme so that an average probability of error of 10^{-4} is always maintained.(16)(Applying)(Dec-08)
    - Coherent PSK and DPSK.
    - Coherent FSK and Coherent MSK
    - Coherent PSK and QPSK.
    - Coherent FSK and non-coherent FSK.
36. Discuss the representation and spectral characteristics of ASK , PSK, QAM, QPSK and FSK signals.(R)
37. What is QAM? Describe in detail about the generation of 8 QAM. Model the constellation diagram for 8 QAM and 16 QAm systems. (Applying) (KSRCE June 2015). (Dec 2016) **(May 2108)**
38. Determine the Baud and Minimum Bandwidth necessary to pass a 10 Kbps binary signal using ASK. (Applying)(KSRCE June 2015**)**
39. Explain with block diagram and mathematical equations the generation and coherent demodulation of ASK signal along with its signal space diagram?
40. With necessary equations and signal space diagram, obtain the probability of error for coherent binary FSK systems.
41. What are the advantages and disadvantages of MSK as compared to QPSK. **(May 2108)**

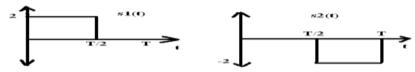
KSRCE/QM/7.5.1/43/ECE

K.S.R. College of Engineering, (Autonomous) - Tiruchengode – 637215

Digital Communication Systems, Cycle Test – 1

PART A (5x2=10)

1. Test the orthogonality of the given two functions. (Analysis)(CO1)



2. Distinguish between PAM and PWM?(Comprehension) (CO1)
3. Draw the block diagram for generation of PPM. (Knowledge) (CO1)
4. Interpret the instantaneous power in different analog pulse modulation techniques?(Evaluation)(CO1)
5. List the advantages and disadvantages of digital communication systems. (Analysis)(CO1)

PART B (2x10 = 20)

6. Derive the geometrical representation of signals.(Application)(CO2)
7. Describe the general block diagram of Digital Communication Systems.(Comprehension) (CO1)

Digital Communication Systems, Cycle Test – 2

PART A (5x2=10)

1. A Signal band limited to 1 MHz is sampled at a rate of 50% higher than the nyquist rate and quantized in to 256 levels. Determine sampling frequency and SNR.(Evaluating)(Co2)
2. M (t) = 6 Cos 2 π (10) t + 4 Cos 2 π (20) t. Examine the minimum sampling frequency to prevent aliasing effect. Analyze)(Co2)
3. Explain the reason why sampled signal has periodic spectrum.(Understanding)(Co2)
4. Define sampling and Nyquist rate for sampling. (Remembering)(Co2)
5. Justify q value should be minimum for lesser quantization noise.(Evaluating)(Co2)

PART B (2x10 = 20)

6. Explain in detail about Natural and impulse sampling concepts.(Remembering)(Co2)
7. Explain in detail about delta modulation. (Analyzing)(Co2)

12EC3602 Digital Communication Cycle Test – III

Part – A (Answer All Questions) 5 x2= 10

1. Define linear block code?(R) (Evaluating) (CO3)
2. What is meant by syndrome of linear block code? (Applying) (CO3)
3. State any four desirable properties of a line code? (Evaluating) (CO3)
4. What are the conditions to satisfy the Hamming code? (R) (CO3)
5. Find the hamming distance between 101010 and 010101. If the minimum hamming distance of a (n.k) linear block code is 3. What is the minimum hamming weight? (CO3)

Part – B (Answer All Questions)

6. Justify DPCM reduce the quantization noise and also develop the necessary equations. (Evaluating& Creating) (CO3) (8)
7. Explain the process of pulse code modulation and derive its signal to Noise Ratio (Understanding) (CO3) (7)

Digital Communication Systems - Assignment –I

PART –A (5 x 2= 10)

1. State any two applications of eye pattern.(R) (CO4)
2. Why do we need equalization in base band pulse transmission?(U) (CO4)
3. Define error probability. (R) (CO4)
4. ISI cannot be avoided justify the statement.(R) (CO4)
5. State the principle of maximum likelihood detectors. (U) (CO4)

PART-B

6. Explain the operation of Detection-Maximum Likelihood Detector using signal constellation diagram.(U) (CO4)
7. Model the eye patters and explain the analysis of eye pattern.(Analysing) (CO4)

Digital Communication Systems - Assignment –II

PART –A (5 x 2= 10)

1. What is meant by memory less modulation?(R) (CO5)
2. Define QAM and draw its constellation diagram.(R) (CO5)
3. What is signal constellation diagram?(R) (CO5)
4. State the difference between coherent and non-coherent binary modulation techniques (U) (CO5)
5. Write the expression for bit error rate for coherent binary FSK.(R) (CO5)

PART-B

6. Derive the bit error probability due to QPSK receiver. Compare the performance of QPSK receiver with that of PSK receiver. (U) (CO5)
7. Explain the detection of binary FSK signal with block diagram. (R) (CO5)