Ref: ET/PC/B/T/313/2025

BETCE Third Year, First Semester - 2025

Subject: Digital Communication Systems

Time: 3 Hours **Full Marks: 100**

Follow the instructions

Answer must be written at one place for each attempted question

Module I- CO1 **Marks: 10**

Q.1 Justify the statements: Answer all

(a) Digital communication is less susceptible to cross talk, noise, waveform distortion than analog communication.

(b) Digital communication is bandwidth efficient, can accommodate large data within limited bandwidth.

(c) Digital communication caters flexible way of information transmission.

(d) Digital communication easily caters in interchanging SNR and BW.

(e) Digital signal transmission needs higher BW than analog for same information transmission.

Module II- CO2 **Marks: 36** **Answer any 2 questions**

Q.2 a) A signal is band limited to f_m Hz sampled by a periodic pulse train $\delta_{Ts}(t)$ made up of a rectangular pulse of width $f_m/8$ sec centered at the origin and repeating at the Nyquist rate $2f_m$ pulses/Sec. Show that the expression for the resulting sampled signal

$$g_s(t) = (1/4\ g(t) + \sum_{n=1}^{\infty} (2/n\pi)\ \sin\ (n\pi/4)\cos(2\pi f_s t)$$

Show also that the signal $g(t)$ can be reconstructed by passing $g_s(t)$ through an ideal low pass filter of band width f_m Hz and gain 4. **09**

b) A zero order hold circuit is often used to reconstruct a signal $g(t)$. You have applied in the laboratory for PAM signal reconstruction. Draw zero order hold circuit. Show that the unit impulse response of the circuit is

$$h(t) = \text{rect}\ (t - T/2)/T$$

Draw the magnitude spectrum $|H(w)|$. Show that when a sampled signal $g_s(t)$ is passed through zero order hold circuit, the output is a staircase approximation of $g(t)$. Consider the sampling interval is the T_s. **09**

Q.3 a) For quantization the following statements are made. Justify those statements.

i) In quantization each sample is rounded off to its nearest level **L**

ii) In quantization each sample is approximated to one of the **L** numbers

iii) The accuracy of the quantized signal can be increased by increasing **L** **05**

b) For voice signal of amplitude **2A** P-to-P is quantized with L=8 levels. If 1 is represented by +A and 0 is represented by −A. Then make a table for 0 to 7 digits by giving the content as **04**

Digit	**Binary Equivalent**	**Pulse code Waveform**

c) For analog signal to be transmitted by digital means the only error or uncertainty in the received signal is the quantization error as the probability of error is less than 10^{-6}. To sustain this condition over a random channel noise and distortion what should be the pulse amplitude w.r.t rms noise amplitude? **02**

d) For a uniform quantizer if x_k is the quantizer input samples and y_k is the quantizer output with total of 'L' levels symmetrically located with zero, then draw the characteristic curve for (x_k, y_k) and from that derive the quantization noise σ_Q^2.
If P denotes the average power of message signal m(t) as $P = \sigma_x^2$, then find $(SNR)_{q,o}$ and establish the relation SNR in dB = 6R + α, where R is the number of bits for PCM code. **07**

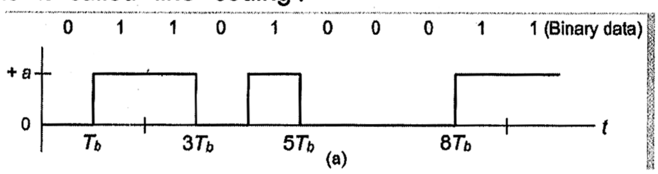
Q4. a) Let a baseband signal x(t) be modeled as the sample function of a zero mean Gaussian random process X(t). x(t) is the input to a quantizer with overload set at $\pm 4\sigma_x$, where σ_x is the variance of the input process. If quantization output is coded with R bits sequence, find $SNR_{q,o}$ for R=3, 4. Comment on the results. **06**

b) Human voice varies very low to high generating SNR 10 dB to 40 dB. So a uniform quantizer is not suitable for all types of speakers. How does this problem overcome? What should be the characteristic of compander function C(x)? Draw the block diagram of compander. **06**

c) If m(k) is the sampled signal of a message signal m(t) and m(k-1) is the previous value, what will happen for Transmission bandwidth and SNR if m(k) as a whole is sent and the difference signal d(k) =m(k) − m(k-1) is sent.
From the above establish the rationale behind DPCM technique. **06**

Module III- CO3 **Marks: 14**

Q. 4 a) What type of line coding or transmission coding represents the following figure? Why is it called line coding? **03**



(a)

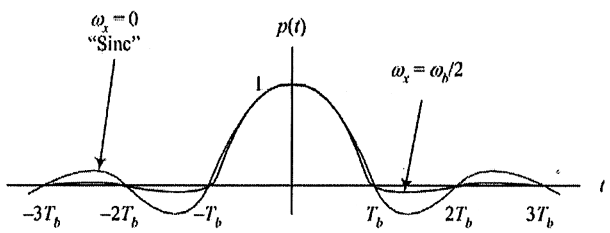
b) Convert the above bits stream in (a) into polar and bipolar representation? Write two important advantages for bipolar format for which it is used in PCM telephony. **05**

c) A sequence of symbols are transmitted using some basic pulse p(t). Let the resulting waveform process be denoted by X(t), given by

$$X(t) = \sum_{k=-\infty}^{\infty} A_k\, p(t - kT - T_d)$$

What is its power spectrum density if autocorrelation $R_A(n) = E(A_K . A_{K\pm n})$? **02**

d) For the pulse p(t) as shown in the figure, draw P(ω) and find the roll off factor "r". **04**



Module IV- CO4 **Marks: 10** **Answer any one question**

Q.5 a) Define signal space and signal constellation diagrams. **03**

b) GSO is used to convert non-orthogonal signals into orthonormal signals. For the signal sets

$S_1(t) = 1,\ 0 \le t \le 3$ $\quad S_2(t) = 1,\ 0 \le t \le 2$ $\quad S_3(t) = 1,\ 1 \le t \le 3$

Find orthonormal basis functions using Gram Schmidt Orthogonal procedure, and draw the signal space diagram. **07**

Q. 6 a) The Euclidian distance between the observation point **r** and message point $\mathbf{s}_k$ is represented by $\|\mathbf{r} - \mathbf{s}_k\|$. Establish the maximum likelihood decision rule and the condition for optimum receiver design. **05**

b) Draw the components of Optimum receiver circuit based on the above design consisting two parts (i) M-Correlators or product integrator to produce **r** and (ii) the transmission decoder based on Maximum Likelihood Estimation for estimating message signal. 05

Module V- CO5 **Marks: 15** **Answer any one question**

Q.7 a) For Binary BFSK signal if f_1 frequency and f_2 frequency difference is $1/2T_b$ and carrier frequency is the mean of f_1 and f_2. Then write the symbol expressions for 1 and 0 i.e $s_1(t)$ and $s_2(t)$. Then form the s(t) for BFSK signal and draw. Express s(t) in In-phase and Quadrature components (I and Q). **05**

[Turn over

b) Find the auto-correlation function of I and Q components and then find the expression for finding power spectral density of BFSK. **10**

Q.8 a) In general the angle modulated wave for MSK (minimum Frequency Shift Keying) is represented by

$S(t)$ = Sqrt($2E_b/T_b$) cos[$2\pi f_c t + \theta(t)$, where carrier frequency $f_c = (f_1+f_2)/2$ and phase of $s(t) = \theta(t)$ is a continuous function of time linearly increases or decreases over 0 to T_b, bit duration.

Define and explain significance of phase trellis for MSK modulation and draw the phase trellis for 11010001. **05**

b) For MSK signal show that I – phase component is $S_I(t) = \pm$ Sqrt($2E_b/T_b$)cos($\pi t/2T_b$) and Q-phase component $S_Q(t) = \pm$ Sqrt($2E_b/T_b$) sin($\pi t/2T_b$)

Justify the statement that for MSK pulse shaping is obtained to a half sinusoid. **05**

c) Draw the QPSK and MSK signal constellation diagrams and point out the basic difference between the two. **05**

Module VI- CO6 **Marks: 15** **Answer any one question**

Q.9 a) A source is transmitting six(6)messages with probabilities 0.3, 0.25, 0.15, 0.12, 0.10 and 0.08 respectively. Find the binary Huffman code.

Then compute average code length and entropy of the source.

What is the efficiency of the code you have found in %? What is the average information per message interval if messages are equally likely? **4+3+2+1=10**

b) Describe the binary Symmetric channel with channel matrix P(Y/X). Then find the joint probability matrix P(X, Y), entropy of source H(X). Draw the probability P vs. H(X) curve for Binary Symmetric Channel. **05**

Q.10 a) If H(X) is the average information transmitted over channel/ symbol under noise free condition, and then what would be the amount of received information?

If there is channel noise, then what will be the amount of received information/symbol? Write the expression and significance of mutual information. **05**

b) Calculate the expression for channel capacity for BSC in term of its error probability P_e. **05**

d) The Shannon's capacity law is defined as **C= B log2 (1 + S/N)** bits /sec. What would be the maximum channel capacity C_{smax} when the channel bandwidth tends to ∝. **05**

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