Time allowed: 3 Hours   Max. Marks: 50

NOTE: *Attempt five questions in all, including Question No. I which is compulsory and selecting two questions from each Part.*

x-x-x

1. (a) What is Nyquist sampling theorem?   (2)

   (b) In an FM system, if m_f is doubled by halving the modulating frequency, what will be the effect on the maximum deviation?   (2)

   (c) What A-law companding?   (2)

   (d) What is the capture effect?   (2)

   (e) What is Inter Symbol Interference?   (2)

Part- A

2.(a) What is the need for modulation ? Give significance of modulation index.   (5)

(b) The output current of a 60% modulated AM generator is 1.5 A. To what value will this current rise if the generator is modulated additionally by another audio wave, whose modulation index is 0.5? What will be the percentage power saving if the carrier and one of the sidebands are now suppressed?   (5)

3.(a) What is Phase Modulation?   (5)

 (b) What is Pulse width modulation? How is it generated? How is it demodulated?   (5)

4.(a) What is Phase Locked Loop? Explain its linear, non linear models & second order PLL.(5)

(b) When the modulating frequency in an FM system is 400 Hz and modulating voltage is 2.4 V, the modulation index is 50. Calculate the maximum deviation. What is the modulation index when the modulating frequency is reduced to 250 Hz and the modulating voltage is simultaneously raised to 3.2 V?   (5)

P.T.O.

Part-B

5.(a) What is companding ? Why is it used? Explain μ-law companding. (5)

(b) What is Adaptive Delta Modulation? What is the effect of Noise specific to Delta modulation? (5)

6.(a) What is pre-emphasis? Why is it used? Sketch a typical pre-emphasis circuit & explain why de-emphasis must be used also. (5)

(b) Explain the effect of Noise in Amplitude Modulation and SSB receivers. (5)

7. (a) What is Inter Symbol Interference ? What is the criterion for zero ISI and Controlled ISI? (5)

(b) Compare the system bandwidth requirements for a terrestrial 3-kHz analog telephone voice channel with that of a digital one. For the digital channel, the voice is formatted as a PCM bit stream, where the sampling rate for the analog-to-digital (A/D) conversion is 8000 samples/s and each voice sample is quantized to one of 256 levels. The bit stream is then transmitted using a PCM waveform and received with zero ISI. (5)

x-x-x