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Reg. No. : | | | | | | | | | | | | |

Question Paper Code : 27200

B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2015.

Fifth Semester

Medical Electronics

EC 6502 — PRINCIPLES OF DIGITAL SIGNAL PROCESSING

(Common to Electronics and Communication Engineering)

(Regulations 2013)

Time : Three hours   Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. Define DT system.

2. How do you obtain a digital signal for DT signal?

3. Define pass band.

4. Use the backward difference for the derivative to convert analog LPF with system function $H(S) = \frac{1}{S+2}$.

5. List the disadvantages of FIR filters.

6. List the desirable window characteristics.

7. What does the truncation of data result in?

8. List the representations for which truncation error is analyzed.

9. List the areas in which multirate processing is used.

10. State sampling theorem for a band limited signal.

PART B — (5 × 16 = 80 marks)

11. (a) (i) Illustrate the construction of an 8 – point DFT from two 4 – point DFTs. (8)

(ii) Illustrate the reduction of an 8 – point DFT to two 4 – point DFTs by decimation in frequency. (8)

Or

(b) (i) Check whether the following systems are linear :

(1) $y_{(n)} = \frac{1}{N} \sum_{M=0}^{N-1} x(n-m)$. (4)

(2) $y_{(n)} = [x(n)]^2$. (4)

(ii) Find the impulse response of the causal system $y(n) - y(n-1) = x(n) + x(n-1)$. (8)

12. (a) (i) An analog filter has the following system function. Convert this filter into a digital filter using backward difference for the derivative $H(s) = \frac{1}{(S+0.1)^2 + 9}$. (8)

(ii) Convert the analog filter into a digital filter whose system function is $H(s) = \frac{s+0.2}{(s+0.2)^2 + 9}$. Use impulse invariance technique. Assume $T = 1$ sec. (8)

Or

(b) (i) Convert the analog filter with system function $H(s) = \frac{s+0.1}{(s+0.1)^2 + 9}$ into a digital IIR filter using bilinear transformation. The digital filter should have a resonant frequency of $w_r = \frac{\pi}{4}$. (8)

(ii) A digital filter with a 3 dB bandwidth of 0.25 π is to be designed from the analog filter whose system response is $H(s) = \frac{\Omega_c}{s + \Omega_c}$. Use bilinear transformation and obtain $H(Z)$. (8)

13. (a) (i) List the steps involved by the general process of designing a digital filter. (8)

(ii) List the advantages of FIR filters. (8)

Or

(b) (i) The transfer function $H(z)=\sum_{N=0}^{M-1} h(n)z^{-n}$ characterizes a FIR filter $(M=11)$. Find the magnitude response. (8)

(ii) Use Fourier series method to design a low pass digital filter to approximate the ideal specifications given by

$$H(e^{jw})=\begin{cases}1, & |f|\le f_p\\ 0, & f_p<|f|\le F/2\end{cases}$$

Where f_p = pass band frequency

F = sampling frequency (8)

14. (a) (i) The output of an ADC is applied to a digital filter with system function $H(z)=\frac{0.5z}{z-0.5}$. Find the output noise power from digital filter when input signal is quantized to have 8 bits. (8)

(ii) Prove that $\sum_{n=0}^{\infty} x^2(n)=\frac{1}{2\pi j}\oint_c x(z)x(z^{-1})z^{-1}dz$. (8)

Or

(b) A digital system is characterized by the difference equation $y(n)=0.9y(n-1)+x(n)$ with $x(0)=0$ and initial condition $y(-1)=12$. Find the dead band of the system. Verify with formula for largest integer. (16)

15. (a) (i) Obtain the decimated signal $y(n)$ by a factor 3 from the input signal $x(n)$. (8)

(ii) Implement a 2 – stage decimator for the following specifications :

Sampling rate of the input signal = 20 kHz, M= 100

Pass band = 0 to 40 Hz

Transition band = 40 to 50 Hz

Pass band ripple = 0.01

Stop band ripple = 0.002. (8)

Or

(b) (i) Draw the signal flow graph for IIR structure M–to–1 decimator. (8)

(ii) Draw the signal flow graph for 1 – to –L interpolator. (8)

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