

A 1220

B.E./B.Tech. DEGREE EXAMINATION, MAY/JUNE 2007.

Sixth Semester

Electrical and Electronics Engineering

EE 337 — DIGITAL SIGNAL PROCESSING

Time : Three hours

Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. Differentiate between digital and analog signals.
2. Name any four elementary time-domain operations for discrete-time signals.
3. Compute the energy of the signal $x[n] = -(0.5)^n u[n]$.
4. Determine the Z-transform of $x[n] = -(0.5)^n u[-n-1]$ and the region of convergence.
5. State the time-shifting and frequency-shifting properties of DTFT.
6. Determine the 3-point circular convolution of the two sequences $x[n] = \{1, 2, 3\}$ and $h[n] = \{3, 2, 1\}$.
7. For the signal $f(t) = \cos^2(4000\pi t) + 2\sin(6000\pi t)$, determine the minimum sampling rate for recovery without aliasing.
8. Define the acquisition time and aperture time of a sample-and-hold circuit.
9. Find the digital transfer function $H(z)$ by using impulse invariance method for the analog transfer function $H(s) = 1/(s+2)$. Assume $T = 0.5$ sec.
10. What is frequency warping in bilinear transformation?

PART B — (5 × 16 = 80 marks)

11. (a) Discuss the important characteristics and processing of the following signals:
 - (i) ECG signals
 - (ii) EEG signals
 - (iii) Seismic signals
 - (iv) Speech signals.

(16)

Or

(b) (i) Describe how filters and equalizers modify the frequency response of recording or monitoring channels. (8)

(ii) Explain the need for echo cancellation in telephone networks. With suitable diagrams, explain how echo cancellation can be achieved. (8)

12. (a) (i) What is the input $x[n]$ that will generate the output sequence

$$y[n] = \{ 1, 5, 10, 11, 8, 4, 1 \} \text{ for a system with impulse response } h[n] = \{ 1, 2, 1 \} ? \quad (8)$$

(ii) Check whether the system defined by

$$h[n] = \left[5\left(\frac{1}{2}\right)^n + 4\left(\frac{1}{3}\right)^n \right] u(n) \text{ is stable?} \quad (3)$$

(iii) Determine the Z-transform $x[n] = \{ r^n \cos \omega_0 n \} u(n)$ and the region of convergence. (5)

Or

(b) (i) The unit sample response $h[n]$ of a system is represented by $h[n] = nu[n-1] - n^2 u[n+2] + 3nu[n-3]$ for $-5 \leq n \leq 5$. Plot the unit sample response. (8)

(ii) Determine the inverse Z-transform of $X(z) = z(z+1)/(z-0.5)^3, |z| > 0.5$. (8)

13. (a) (i) Determine DTFT of $x[n] = (0.2)^n u(n)$. Obtain the expressions for the magnitude and phase spectra. (8)

(ii) Determine the DFT of the sequence

$$x[n] = \frac{1}{3} \delta[n] - \frac{1}{3} \delta[n-1] + \frac{1}{3} \delta[n-2]. \quad (8)$$

Or

(b) (i) From first principles obtain the signal flow graph for computing 8-point DFT using radix-2 decimation-in-frequency FFT algorithm. (8)

(ii) Using the above signal flow graph compute DFT of $x[n] = \cos(n\pi/4), 0 \leq n \leq 7$. (8)

14. (a) (i)

(ii)

(b)

15. (a)

(b)

Frequency response (8)

Networks. With
be achieved. (8)

Sequence

Impulse response (8)

(3)

in the region (5)

represented by
Plot the (8)

of (8)

expressions for the (8)

(8)

Computing
algorithm. (8)

DFT of (8)

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14. (a) (i) From first principles establish the condition on the sampling rate when a band limited signal is to be sampled in such a way that the original signal can be recovered through ideal low pass filtering. Assume ideal sampling. (8)

(ii) What are the merits of discrete-time processing of continuous-time signals? Draw the block diagram of a system for discrete-time processing of continuous-time signals. Explain the functions of each block. (8)

Or

(b) (i) Design an analog low pass Butterworth filter having a low pass characteristic with a -3 dB cutoff frequency at 800 rad/sec and a minimum attenuation of -10dB at 1800 rad/sec. (10)

(ii) Explain briefly the various errors encountered in the implementation of a practical A/D converter. (6)

15. (a) (i) With a suitable example, illustrate how an FIR filter can be realized based on the polyphase decomposition of its transfer function. (8)

(ii) Convert the following analog transfer function $H(s)$ into equivalent digital transfer function $H(z)$ using impulse invariance method $H(s) = (s + 0.1) / ((s + 0.1)^2 + 9)$. Assume $T = 1$ sec. (8)

Or

(b) (i) With suitable examples, describe the realization of linear-phase FIR filter structures. (8)

(ii) Convert the analog filter with transfer function $H(s) = 2 / ((s + 1)(s + 3))$ into a digital filter using bilinear transformation with $T = 0.1$ sec. Draw direct form II realization for the digital filter. (8)