M.TECH/AEIE/1ST SEM /AEIE 5102/2015
2015

Digital Signals and Systems
(AEIE 5102)

Time Allotted : 3 hrs **Full Marks : 70**

Figures out of the right margin indicate full marks.

Candidates are required to answer Group A and any 5 (five) from Group B to E, taking at least one from each group.

Candidates are required to give answer in their own words as far as practicable.

Group – A
(Multiple Choice Type Questions)

1. Choose the correct alternatives for the following: **10 x 1=10**

(i) A quantizer operates at a sampling frequency of 16 kHz. What is its Nyquist limit?
(a) 4 kHz (b) 8 kHz (c) 16 kHz (d) 32 kHz.

(ii) If F_s is sampling frequency then the highest analog frequency that can be uniquely represented in its sampled version of discrete time signal is,
(a) $\frac{F_s}{2}$ (b) $2F_s$ (c) F_s (d) $\frac{1}{F_s}$.

(iii) The system $y(n)=\sin[x(n)]$ is,
(a) stable (b) BIBO stable (c) unstable (d) marginally stable.

(iv) For energy signals, the energy will be finite and the average power will be
(a) infinite (b) finite (c) zero (d) cannot be defined.

(v) Twiddle factor is given by
(a) $W=e^{-i\left(\frac{2\pi}{N}\right)}$ (b) $W=e^{i\left(\frac{2\pi}{N}\right)}$ (c) $W=e^{-i\left(\frac{N}{2\pi}\right)}$ (d) $W=e^{i\left(\frac{N}{2\pi}\right)}$

(vi) DFT of $x(n)=\delta(n)$ is
(a) 1 (b) 0 (c) $W=e^{-i(2/N)}$ (d) $W=e^{i(2/N)}$

(vii) In N-point DFT of L-point sequence, the value of N to avoid aliasing in frequency spectrum is
(a) $N\neq L$ (b) $N\geq L$ (c) $N\leq L$ (d) $N=L$

(viii) The condition for symmetry of impulse response of FIR system is
(a) $h(n)=h(N-1)$ (b) $h(n)=h(N+1)$ (c) $h(n)=h(N-n)$ (d) $h(n)=h(N-1-n)$

(ix) Which one of the statements is correct with reference to wavelet domain signal analysis?

(a) Scaling functions extract the details and wavelet functions perform approximation.
(b) Scaling functions perform approximations and wavelet functions extract the details.
(c) Wavelet functions perform approximations and also extraction of details.
(d) None of the above.

(x) Down-sampling, the signal $v[n]$ should be bandlimited to $|\omega| \le \frac{\pi}{M}$ by means of a lowpass filter, called

(a) interpolation filter
(b) decimation filter
(c) both (a) and (b)
(d) none of above.

Group – B

2.(a) Determine the response of the LTI system whose input $x(n)$ and impulse response $h(n)$ are given by $x(n) = \{\underset{\uparrow}{1}, 2, 3, 1\}$ and $h(n) = \{1, \underset{\uparrow}{2}, 1, -1\}$.

(b) What do you mean by causal and non causal systems? Test the causality of the system $y(n) = x(n^2)$.

7 + (2+3) = 12

3.(a) Determine whether the system $y(n) = nx^2(n)$ is time variant or time invariant.

(b) Define periodic and aperiodic signals. Determine whether the signal $x(n) = \sin\left(\frac{6\pi}{7}n + 1\right)$ is periodic or not.

5 + (2+5) = 12

Group – C

4.(a) List the difference between linear convolution and circular convolution. Perform the circular convolution of the two sequences $x_1(n) = \{1, 2, 3\}$ and $x_2(n) = \{4, 5, 6\}$.

(b) Find the DFT of the sequence $x(n) = \{1, 1, 0, 0\}$. Also find magnitude and phase sequence.

(2+5) + 5 = 12

5.(a) Find the inverse DFT of $Y(k) = \{1, 0, 1, 0\}$.

(c) What is wavelet transform? What is wavelet packet transform? What are the applications of wavelets?

6 + (2+2+2) = 12

Group – D

6.(a) What are the factors that influence the choice of the structure for realization of an LTI system?

(b) Obtain the direct form-I and direct form-II realizations of a discrete time system represented by the transfer function $H(z)=\dfrac{8z^3-4z^2+11z-2}{\left(z-\frac{1}{4}\right)\left(z^2-z+\frac{1}{2}\right)}$.

2 + 10 = 12

7. Design an analog low pass Butterworth filter to satisfy

Passband cut-off: $\Omega_p=0.2\pi$ Passband ripple $R_p=7$ dB

Stopband cut-off: $\Omega_s=0.3\pi$ Stopband ripple $A_s=16$ dB

Convert the above designed analog filter to a digital filter by using impulse invariant technique. Assume $T=1$ sec.

(10 + 2) = 12

Group – E

8.(a) What is the need for Multirate signal processing? Give some examples of Multirate digital system.

(b) Consider an audio-band signal with a nominal bandwidth of 4 kHz that has been sampled at a rate of 8kHz. Suppose that we wish to isolate the frequency components below 80Hz with a filter that has a pass band $0\le F\le 75$ and a transition band $75\le F\le 80$. Design an one stage and a two stage decimator filter to achieve this. Calculate the factor by which the filter length reduced in two stage decimator than that of one stage decimator. Consider that the filter has a passband ripple $\delta_1=10^{-2}$ and a stopband ripple of $\delta_2=10^{-4}$.

3 + 9 = 12

9. A linear shift-invariant system having system function $H(z)=\dfrac{1-\frac{1}{2}z^{-1}}{1-\frac{1}{3}z^{-1}}$, is excited by a zero mean exponentially correlated noise $x(n)$ with an autocorrelation sequence $r_x(k)=\left(\frac{1}{2}\right)^{|k|}$.

Let $y(n)$ be the output process, $y(n)=x(n)*h(n)$. Calculate:

(i) the power spectrum, $P_y(z)$ of $y(n)$.

(ii) the autocorrelation sequence, $r_y(k)$, of $y(n)$.

(iii) the cross-correlation, $r_{xy}(k)$, between $x(n)$ and $y(n)$.

(4 + 4 + 4) = 12