M.TECH/AEIE/1ST SEM/AEIE 5102/2016

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 alternative for the following: **10 × 1 = 10**

(i) The process of conversion of continuous time signal into discrete time signal is known as
(a) aliasing (b) sampling
(c) convolution (d) none of the above.

(ii) In a signal $x(n)$, if 'n' is replaced by $2n$, then it is called
(a) upsampling (b) folded version
(c) downsampling (d) shifted version.

(iii) The discrete time system, $y(n)=0.7x(n-4)-1.2x(n-5)$ is a,
(a) dynamic system (b) memoryless system
(c) time varying system (d) none of the above.

(iv) If F_s is sampling frequency then the relation between analog frequency F and digital frequency f signal is,
(a) $f=\frac{F}{2F_s}$ (b) $f=\frac{F_s}{F}$ (c) $f=\frac{F}{F_s}$ (d) $f=\frac{2F}{F_s}$

(v) The complex valued phase factor/twiddle factor, W_N can be expressed as
(a) $e^{-j2\pi n}$ (b) $e^{-\frac{j2\pi}{N}}$
(c) $e^{-j2\pi}$ (d) $e^{-j2\pi kn}$

(vi) In an N-point sequence, if $N=16$, the total number of complex additions and multiplications using Radix-2 FFT are
(a) 64 and 80 (b) 64 and 32
(c) 80 and 64 (d) 24 and 12

(vii) The inverse DFT of $x(n)$ is given by
(a) $x(n)=\frac{1}{N}\sum_{k=0}^{N}X(k)e^{-\frac{j2\pi kn}{N}}$ (b) $x(n)=\frac{1}{N}\sum_{k=0}^{N-1}X(k)e^{\frac{j2\pi kn}{N}}$
(c) $x(n)=\frac{1}{N}\sum_{k=0}^{N}X(n)e^{-\frac{j2\pi kn}{N}}$ (d) $x(n)=\frac{1}{N}\sum_{k=0}^{N-1}X(k)e^{-\frac{j2\pi kn}{N}}$

(viii) The linear phase realization structure is used to represent
(a) FIR systems (b) IIR systems
(c) both FIR and IIR systems (d) all discrete time systems.

(ix) In practice, the zero-valued samples inserted by the up-sampler are replaced with appropriate nonzero values using some type of filtering process, the process is called
(a) interpolation (b) decimation
(c) both (a) and (b) (d) none of above.

(x) If $\text{var}\{x\}$ and $\text{var}\{y\}$ are variances of two random uncorrelated signals x and y, respectively, then variance $\text{var}\{x+y\}$ of sum of the two signals is given by
(a) $\text{var}\{x+y\}=\text{var}\{x\}+\text{var}\{y\}$ (b) $\text{var}\{x+y\}=\text{var}\{x\}-\text{var}\{y\}$
(c) $\text{var}\{x+y\}=\text{var}\{x\}\times\text{var}\{y\}$ (d) $\text{var}\{x+y\}=\text{var}\{x\}/\text{var}\{y\}$

Group – B

2. (a) What are causal and non-causal signals?

(b) Determine the even and odd part of the signal $x(n)=a^n$.

(c) An analog signal is given by $x_a(t)=10\cos 100\pi t$. If the sampling frequency is 75Hz, find the discrete time signal $x(n)$. Also find an alias frequency corresponding to $F_s=75Hz$.

2 + 3 + 7 = 12

3. (a) What do you mean by a linear and nonlinear system? Test the linearity of the system $y(n) = x^2(n)$.

(b) Perform circular convolution of the two sequences $x_1(n) = \{2,1,2,1\}$ and $x_2(n) = \{1,2,3,4\}$.

(2 + 5) + 5 = 12

Group – C

4. (a) If $DFT\ \{x(n)\} = X(k)$, then prove that

$$DFT\ \{x_1(n)x_2(n)\} = \frac{1}{N}\left[X_1(k) \otimes X_2(k)\right].$$

(b) Compute the DFT of the sequence $x(n) = (-1)^n$ for the period N=16.

6 + 6 = 12

5. (a) Compute the ... $(n) = \{\underset{\uparrow}{1}, 0.5\}$

and $h(n) = \{($ ...

(b) What is the d ... et transform is different fi ... significance of continuou ...

... 2 + 2) = 12

Group – D

6. (a) What are the properties that are maintained same in the transfer process of analog filter into a digital filter?

(b) Prove that physically realizable and stable IIR filters cannot have linear phase.

(c) An analog filter has the following system function:

$$H(s) = \frac{s+0.1}{(s+0.1)^2+9}.$$

Convert this analog filter into a digital IIR filter using bilinear transformation method. The digital filter must have a resonant frequency of $\omega_r = \dfrac{\pi}{4}$.

3 + 3 + 6 = 12

7. (a) Design a lowpass Butterworth IIR digital filter using bilinear transformation technique with $T=1$, from its analog filter that satisfies following specifications:

Passband cutoff: $\Omega_p = 0.2\pi$; Passband ripple: $R_p = 7dB$

Stopband cutoff: $\Omega_s = 0.3\pi$; Stopband ripple: $A_s = 16\ dB$.

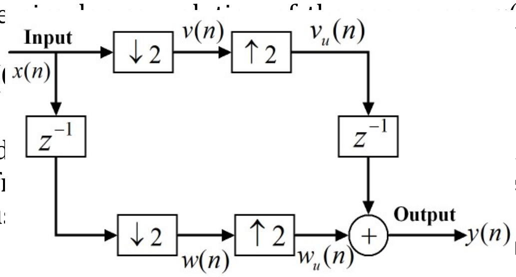
(b) Transform the analog filter $H_a(s) = \dfrac{s+1}{s^2+5s+6}$ into a digital filter $H(z)$ using the impulse invariance technique in which $T = 0.1$.

8 + 4 = 12

Group – E

8. (a) What are the applications of Multirate signal processing?

(b) A multirate sampling system is shown in figure below. Determine $y(n)$ as a function of $x(n)$.



4 + 8 = 12

9. Let a process $x(n)$ with zero-mean and its autocorrelation is given by $r_x(x) = 10\left(\frac{1}{2}\right)^{|k|} + 3\left(\frac{1}{2}\right)^{|k-1|} + 3\left(\frac{1}{2}\right)^{|k+1|}$

(i) Find a filter, which when driven by unit variance white noise, will yield a random process with this autocorrelation.

(ii) Find a stable and causal filter which, when excited by $x(n)$, will produce zero mean, unit variance, white noise.

6 + 6 = 12