ADVANCED DIGITAL SIGNALS AND SYSTEMS
(AEIE 5101)

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

(ii) The system described by the input-output equation $y(n) = nx(n) + bx^3(n)$ is a ____________
(a) static system (b) dynamic system
(c) identical system (d) none of the above.

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

(iv) If W is the twidal factor, then the value of DFT $F = W^N$, when, $N = 3$ is
(a) 0 (b) 1 (c) -1 (d) j.

(v) Coefficient symmetry is important in FIR filters because it provides
(a) a smaller transition bandwidth (b) less passband ripple
(c) less stopband ripple (d) a linear phase response.

(vi) A real valued signal $x(n)$ is called as anti-symmetric if __________
(a) $x(n) = x(-n)$ (b) $x(n) = -x(-n)$
(c) $x(n) = -x(n)$ (d) none of the mentioned.

(vii) Which of the following substitution is done in Bilinear transformations?
(a) $s = \frac{2}{T}\left[\frac{1+z^{-1}}{1-z^{-1}}\right]$ (b) $s = \frac{2}{T}\left[\frac{z^{-1}+1}{z^{-1}-1}\right]$
(c) $s = \frac{2}{T}\left[\frac{1-z^{-1}}{1+z^{-1}}\right]$ (d) None of the above.

(viii) Issue connected with finite word length effects
(a) quantization effects in A/D conversion
(b) product quantization and coefficient quantization errors in digital filters
(c) both (a) and (b)
(d) none of the above.

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

(x) In bilinear transformation, the left-half s-plane is mapped to which of the following in the z-domain?
(a) Entirely outside the unit circle |z|=1 (b) Partially outside the unit circle |z|=1
(c) Partially inside the unit circle |z|=1 (d) Entirely inside the unit circle |z|=1.

Group – B

2. (a) What are energy and power signals? [(CO1)(Remember/LOCQ)]
(b) Determine whether the signal $x(n) = \left(\frac{3}{8}\right)^n u(n)$ is energy or power signal. [(CO1)(Analyze/IOCQ)]
(c) Consider the analog signals $x_1(t) = 2\cos(2\pi 10t)$ and $x_2(t) = 1.5\cos(2\pi 40t)$. Derive a sampling frequency, so that the 40 Hz signal is an alias of the 10 Hz signal. [(CO1)(Evaluate/HOCQ)]

4 + 4 + 4 = 12

3. (a) The impulse response of a system is given by h(n)= {1,1,1,1,1}. Determine the output of the system for an input $x(n) = \{1,2,4\}$. [(CO2)(Apply/IOCQ)]
(b) What are the drawbacks of DFT? How it is overcome by FFT? [(CO2)(Understand/LOCQ)]
(c) Compute the DFT of the sequence $x(n) = \{1,2,0,2\}$ and sketch the magnitude and phase spectrum. [(CO2)(Apply/IOCQ)]

4 + 2 + 6 = 12

Group – C

4. (a) Convert the analog filter transfer function $H(s) = \frac{s+2}{(s+1)(s+3)}$ into digital filter transfer functions using impulse invariance technique. Consider $T = 1$ sec. [(CO3)(Apply/IOCQ)]
(b) Design a digital IIR filter from the analog filter with transfer function $H(s) = \frac{s+0.1}{(s+0.1)^2+16/3}$ by using bilinear transform technique. The digital filter is to have a resonant frequency of $\omega_r = \frac{\pi}{3}$. [(CO3)(Evaluate/HOCQ)]
(c) What is warping effect? [(CO3)(Understand/LOCQ)]

4 + 6 + 2 = 12

5. (a) What is the basis for Fourier series method of FIR digital filter design? Why truncation of Fourier series is necessary? What is the effect of this truncation in magnitude response of the filter? [(CO3)(Understand/LOCQ)]
(b) Construct a direct form-I realization of general 3rd order IIR filter. [(CO4)(Analyze/LOCQ)]
(c) Realize the linear phase structure of the following FIR system:
$H(z) = \left(1 + \frac{1}{2}z^{-1} + z^{-2}\right)\left(1 + \frac{1}{4}z^{-1} + z^{-2}\right)$. [(CO4)(Evaluate/HOCQ)]

(2 + 1 + 1) + 3 + 5 = 12

Group – D

6. (a) State the applications of multirate digital signal processing. Explain decimation and interpolation method of sampling rate conversion. [(CO5)(Remember/LOCQ)]
   (b) What are the disadvantages of increasing sampling rate by an up-sampler? Give an efficient implementation method to overcome the above draw backs of up-sampler. [(CO5)(Analyze/IOCQ)]
   (c) Derive the poly phase decomposition of the IIR digital filter given by $(z) = \frac{1-4z^{-1}}{1+5z^{-1}}$. [(CO5)(Analyze/IOCQ)]

(1 + 2) + (1 + 4) + 4 = 12

7. (a) What are the applications of Adaptive filters? How it is used for noise cancellation? [(CO5)(Remember/LOCQ)]
   (b) Explain how adaptive filters are used in system identifications. [(CO5)(Analyze/IOCQ)]
   (c) Describe in brief the LMS algorithm of adaptive filter. [(CO5)(Understand/LOCQ)]

4 + 4 + 4 = 12

Group – E

8. (a) Describe how parametric methods of power spectrum estimation is different from non-parametric method of power spectrum estimation? State the steps of parametric method of power spectrum estimation. [(CO6)(Remember/LOCQ)]
   (b) Differentiate between AR and ARMA processes. [(CO6) (Analyze/IOCQ)]
   (c) Suppose we have $N = 500$ samples from a sample sequence of a random process. Compute the frequency resolution Δf of the following: (i) Bartlett Method (ii) Welch Method (iii) Blackman-Tukey Method. Consider overlapping is 50% and quality factor 12. [(CO6)(Evaluate/IOCQ)]

(2 + 2) + 4 + 4 = 12

9. (a) What are stationary, non-stationary and wide sense stationary process? [(CO6)(Remember/LOCQ)]
   (b) A continuous-time signal is band limited to 5 kHz, i.e., $x_a(t)$ has a spectrum $x_a(f)$ that is zero for $|f| \geq 5$ kHz. Only 10 seconds of the signal has been recorded and is available for processing. We would like to estimate the power spectrum of $x_a(t)$ using the available data in a radix-2 FFT algorithm, and it is required that the estimate have a resolution of at least 10 Hz. Suppose that we use Bartlett's method of periodogram averaging.
      (i) If the data is sampled at the Nyquist rate, what is the minimum section length that you may use to get the desired resolution?
      (ii) Using the minimum section length determined in part (a), with 10 seconds of data, how many sections are available for averaging?
      (iii) How does your choice of sampling rate affect the resolution and variance of your estimate? Are there any benefits to sampling above the Nyquist rate? [(CO6)(Analyze/IOCQ)]

3 + 9 = 12

Cognition Level	LOCQ	IOCQ	HOCQ
Percentage distribution	34.38	50	15.62

Course Outcome (CO):

After the completion of the course students will be able to:

1. Characterize and analyze the properties of discrete time signals and systems.
2. Perform DFT, FFT and IDFT of a given discrete signal and learn STFT and DWT of discrete signal.
3. Design digital FIR and IIR filters according to the given specification.
4. Realize a digital filter structure from it's transfer function.
5. Understand theory of multirate DSP and adaptive filtering techniques, solve numerical problems.
6. Understand theory of prediction, solution of normal equations and methods of spectral estimation.

*LOCQ: Lower Order Cognitive Question; IOCQ: Intermediate Order Cognitive Question; HOCQ: Higher Order Cognitive Question.