ADVANCED DIGITAL SIGNALS AND SYSTEMS
(AEIE 5101)

Time Allotted : 2½ hrs **Full Marks : 60**

Figures out of the right margin indicate full marks.

Candidates are required to answer Group A and <u>any 4 (four)</u> from Group B to E, taking <u>one</u> from each group.

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

Group – A

1. Answer any twelve: **12 × 1 = 12**

Choose the correct alternative for the following

(i) The sampling is the process of conversion of continuous time signal into
(a) digital signal (b) unit impulse signal
(c) unit step signal (d) discrete time signal.

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

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

(iv) Condition for aliasing problem:
(a) $f_s < f_m$ (b) $f_s < 2f_m$ (c) $f_s = f_m$ (d) All of these.

(v) DTFT is the representation of
(a) periodic discrete time signals (b) aperiodic discrete time signals
(c) periodic continuous time signals (d) aperiodic continuous time signals.

(vi) Which operation is implemented by keeping every M-th sample of $x(n)$ and removing $M - 1$ in between samples to generate $y(n)$?
(a) Up-sampling (b) Down-sampling
(c) Both (a) and (b) (d) None of the above.

(vii) What is the process of increasing the sampling rate by a factor I?
(a) Sampling rate conversion (b) Interpolation
(c) Decimation (d) None of these.

(viii) IIR filters
(a) use feedback
(b) are sometimes called recursive filters
(c) can oscillate if not properly designed
(d) all of these.

(ix) The FFT algorithms
- (a) eliminate the redundant calculation and enable to analyze the spectral properties of a signal
- (b) eliminate the redundant calculation and redundant to analyze the spectral properties of a signal
- (c) both (a) & (b)
- (d) None.

(x) Drawbacks FIR filters are
- (a) more computation than an IIR with similar effect
- (b) prevent phase distortion
- (c) less computation
- (d) all of above.

Fill in the blanks with the correct word

(xi) The discrete time function defined as $u(n) = n$ for $n \geq 0$; $u(n) = 0$ for $n \leq 0$ is an __________ signal.

(xii) The total number of complex multiplications required to compute N point DFT by radix-2 FFT algorithm is __________.

(xiii) Causal systems are the systems in which the output of the system depends on the __________.

(xiv) The _________ of $X(z)$ is set of all values of z, for which $X(z)$ attains a finite value.

(xv) An LTI system is causal if and only if its impulse response is _________ for negative values of n.

Group - B

2. (a) Determine whether the signal $x(n) = 3\sin(5n + \frac{\pi}{6})$ is periodic or not.
[(CO1)(Analyse/IOCQ)]

(b) Perform the convolution of the two signals: $x(n) = \{\underline{0}, 0,1,1,1,1\}$ and $h(n) = \{1, \underline{-2}, 3\}$. *[(CO1)(Apply/IOCQ)]*

(c) Determine whether the following system is time variant or time invariant: $y(n) = nx^2(n)$. *[(CO1)(Analyze/IOCQ)]*

4 + 4 + 4 = 12

3. (a) Determine the response of the LTI system whose input $x(n)$ and response $h(n)$ are given by $x(n) = \{\underline{1}, 2,3,1\}$ and $h(n) = \{1, \underline{2}, 1, -1\}$. *[(CO1)(Apply/IOCQ)]*

(b) Compute the DFT of the following sequences using the radix-2 DIT FFT algorithm $x(n) = \{1,3,1,2\}$. *[(CO2)(Evaluate/HOCQ)]*

6 + 6 = 12

Group - C

4. (a) Design an analog Butterworth filter that has a $-2\ dB$ passband attenuation at a frequency of 20 rad/sec and at least *-10 dB* stopband attenuation at 30 rad/sec.
[(CO3)(Apply/IOCQ)]

(b) Transform the analog filter $H_a(s) = \frac{2}{s^2+3s+2}$ into a digital filter using the bilinear transformation in which $T = 1$ sec. *[(CO3)(Apply/IOCQ)]*

8 + 4 = 12

5. (a) Prove that, $\Omega_c = \frac{\Omega_p}{(10^{0.1\alpha_p}-1)^{\frac{1}{2N}}} = \frac{\Omega_s}{(10^{0.1\alpha_s}-1)^{\frac{1}{2N}}}$. *[(CO3)(Analyze/IOCQ)]*

(b) What are the possible types of impulse response for linear phase FIR filters? *[(CO3)(Remember/LOCQ)]*

(c) Explain the characteristics of rectangular window with typical sketches. *[(CO3)(Analyze/IOCQ)]*

5 + 2 + 5 = 12

Group - D

6. (a) What is adaptive filtering? Show a block diagram of basic adaptive filer structure. *[(CO5)(Understand/LOCQ)]*

(b) Illustrate, with a practical example, how adaptive filter may be used for noise cancellation. *[(CO5)(Understand/LOCQ)]*

(c) Describe in brief the LMS algorithm of adaptive filter. *[(CO5)(Remember/LOCQ)]*

4 + 4 + 4 = 12

7. (a) How adaptive filters differs from FIR and IIR filters? *[(CO5)(Remember/LOCQ)]*

(b) Illustrate how adaptive filters can be employed for linear prediction. *[(CO5)(Analyze/IOCQ)]*

(c) Given the DSP system for the noise cancellation application using an adaptive filter with two coefficients given by $y(n) = w_0(n)x(n) + w_1(n)x(n-1)$.
Perform adaptive filtering to obtain outputs $e(n)$ for $n = 0, 1, 2$ given the following inputs and outputs:
$x(0) = 1, x(1) = 1, x(2) = -1, d(0) = 2, d(1) = 1, d(2) = -2$ and initial weights: $w_0 = w_1 = 0$ and convergence point is set to be $\mu = 0.1$. *[(CO5)(Evaluate/HOCQ)]*

2 + 4 + 6 = 12

Group - E

8. (a) Consider a moving average MA(q) process that is generated by the difference equation $y(n) = \sum_{k=0}^{q} b(k)\omega(n-k)$
Where $\omega(n)$ is zero mean white noise with variance σ_ω^2.
(i) Find the unit sample response of the filter that generates $y(n)$ from $\omega(n)$.
(ii) Find the autocorrelation and power spectrum of $y(n)$. *[(CO6)(Apply/LOCQ)]*

(b) The power spectrum of a wide sense stationary process $x(n)$ is

$$P_x(e^{j\omega}) = \frac{25 - 24\cos\omega}{26 - 10\cos\omega}$$

Derive the expression of the whitening filter $H(z)$ that produces unit variance white noise when the input is $x(n)$. *[(CO6)(Analyze/IOCQ)]*

(4 + 4) + 4 = 12

9. (a) Derive the power spectral density of a stationary process x[n] with zero mean value and autocorrelation sequence $r_{xx}[l] = a^{|l|}, -1 < a < 1$. *[(CO6)(Evaluate/HOCQ)]*

(b) Explain Wiener filtering problem with a block diagram. Describe briefly what we need to design an FIR Wiener filter. *[(CO6)(Understand/LOCQ)]*

5 + (4 + 3) = 12

Cognition Level	LOCQ	IOCQ	HOCQ
Percentage distribution	32.29	50	17.71

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.*