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 any 4 (four) from Group B to E, taking one 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) 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) $f = \frac{2F_s}{F}$ (c) $f = \frac{F}{F_s}$ (d) $f = \frac{2F}{F_s}$

(ii) 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*

(iii) 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.

(iv) 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.

(v) If M and N are the orders of numerator and denominator of rational system function respectively, then how many additions are required in direct form-I realization of that IIR filter?
(a) M+N-1 (b) M+N (c) M+N+1 (d) M+N+2.

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

(vii) LMS algorithm stands for
(a) Largest Mean Square algorithm (b) Least Median Square algorithm
(c) Least Mean Square algorithm (d) none of the above.

(viii) According to Parseval's theorem the energy spectral density curve is equal to
(a) area under magnitude of the signal
(b) area under square of the magnitude of the signal
(c) area under square root of magnitude of the signal
(d) none of the mentioned.

(ix) Which theorem states that the total average power of a periodic signal is equal to the sum of average powers of the individual Fourier coefficients?
(a) Parseval's Theorm (b) Rayleigh's Theorm
(c) Both (a) and (b) (d) None of the above.

(x) If $s(f)$ is the power spectral density of a real, wide sense stationary random process, then which of the following is always true?
(a) $s(0) \geq s(f)$ (b) $s(f) \geq 0$
(c) $s(-f) = -s(f)$ (d) $\int_{-\infty}^{\infty} s(f)df = 0$.

Fill in the blanks with the correct word

(xi) Appending zeros to a sequence in order to increase its length is called _________.

(xii) The tolerance in the pass band and stop band are called _________.

(xiii) To change the sampling rate for better efficiency in two or multiple stages, the decimation and interpolation factors must be ________ unity.

(xiv) Power spectrum describes distribution of ________ under frequency domain.

(xv) If f_m is the maximum frequency in a signal, then the minimum sampling rate to avoid aliasing effect is _____.

Group - B

2. (a) Consider the analog signals, $x_a(t) = 6cos50\pi t + 3sin200\pi t - 3cos100\pi t$. Determine the minimum sampled frequency and the sampled version of analog signal at this frequency. *[(CO1)(Analyse/IOCQ)]*

(b) Test the stability of the system whose impulse response is $h(n) = \left(\frac{1}{2}\right)^n u(n)$. *[(CO1)(Analyze/IOCQ)]*

(c) Compute the IDFT of the sequence $Y(K) = \{4,2,0,2\}$. *[(CO2)(Apply/IOCQ)]*

(2 + 2) + 4 + 4 = 12

3. (a) Determine the response of the LTI system whose input x(n) and response h(n) are given by $x(n) = \{\underline{2}, 1,0,2\}$ and $h(n) = \{-1, \underline{1}, 2,1\}$. *[(CO1)(Apply/IOCQ)]*

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

6 + 6 = 12

Group - C

4. (a) Determine the order and the poles of lowpass Butterworth filter that has a 3 dB attenuation at 500 Hz and an attenuation of 40 dB at 1000 Hz. *[(CO3)(Apply/IOCQ)]*

(b) Realize the following system with minimum number of multipliers: $(z) = (1+z^{-1})\left(1+\frac{1}{2}z^{-1}+\frac{1}{2}z^{-2}+z^{-3}\right)$. *[(CO4)(Design/HOCQ)]*

(c) Distinguish between FIR and IIR filters. *[(CO3)(Remember/LOCQ)]*

4 + 6 + 2 = 12

5. (a) Convert the analog filter transfer function $\mathrm{H(s)} = \frac{\mathrm{s+2}}{\mathrm{(s+1)(s+3)}}$ into digital filter transfer functions using impulse invariance technique. Consider $\mathrm{T} = 1$ sec. *[(CO3)(Apply/IOCQ)]*

(b) Design a digital IIR filter from the analog filter with transfer function $\mathrm{H(s)} = \frac{\mathrm{s+0.1}}{\mathrm{(s+0.1)^2+16/3}}$ 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

Group - D

6. (a) What is multirate DSP? State the applications of it? *[(CO5)(Remember/LOCQ)]*

(b) Given the sequence $\mathrm{x(n)} = \{\ldots\ldots, 3, \underline{5}, 2, 9, 6, \ldots\ldots\}$, find out the output y(n) and Y(z) of a up-sampler with up-sampling factor 2? *[(CO5)(Analyze/IOCQ)]*

(c) Consider an audio-band signal with a nominal bandwidth of 4 kHz that has been sampled at a rate of 8 kHz. Suppose that we wish to isolate the frequency components below 80 Hz with a filter that has a pass band $0 \le \mathrm{F} \le 75$ and a transition band $75 \le \mathrm{F} \le 80$. Design a 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 $\delta_2 = 10^{-4}$. *[(CO5)(Evaluate/HOCQ)]*

(1 + 1) + 4 + 6 = 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)(Analyse/IOCQ)]*

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

2 + 4 + 6 = 12

Group - E

8. (a) What is periodogram? What are the non-parametric methods of power spectrum estimation? State the limitations of non-parametric methods of power spectrum estimation. *[(CO6)(Remember/LOCQ)]*

(b) Determine the frequency resolution of the Bartlett, Welch and Blackman-Tukey methods of power spectrum estimation for quality factor Q = 10. Assume that overlap in Welch's method is 50%. Given the length of the sample sequence is 1000. *[(CO6)(Analyze/IOCQ)]*

(2 + 4) + 6 = 12

9. (a) What is the need for spectral estimation? How can the energy density spectrum be determined? *[(CO6)(Understand/LOCQ)]*

(b) A discrete-time signal x(n) is given as under: $x(n) = \cos(2\pi f_1 n) + \cos(2\pi f_1 n)$, $n = 0,1,2,\ldots,7$.

Determine the power spectrum for the data sequence length $L = 8,16$; for different values of f_1 and f_2, where $f_2 = f_1 + \Delta\text{f}$ and Δf is small deviation from f_1 or simply frequency separation. *[(CO6)(Evaluate/HOCQ)]*

(2 + 2) + 8 = 12

Cognition Level	LOCQ	IOCQ	HOCQ
Percentage distribution	18.75	41.67	39.58