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) The odd part of a signal $x(t)$ is given by
(a) $x(t)+x(-t)$ (b) $x(t)-x(-t)$
(c) $\frac{1}{2}[x(t)+x(-t)]$ (d) $\frac{1}{2}[x(t)-x(-t)]$

(ii) If a system does not have a bounded output for bounded input, then the system is said to be
(a) causal (b) non-causal (c) stable (d) non-stable

(iii) DIT algorithm of FFT divides the sequence into
(a) positive and negative values (b) even and odd samples
(c) two halves (d) small and large values

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

(v) 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

(vi) The poles obtained from matched z-transform are identical to poles obtained from which of the following transformations?
(a) Approximation of derivatives (b) Bilinear transformation
(c) Impulse invariance (d) None of the above

(vii) 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) $M+N+2$

(viii) 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

(ix) Which of the following is the correct relation between ω and Ω?
(a) $\Omega = \omega T$ (b) $T = \omega\Omega$ (c) $\omega = \Omega T$ (d) none of the above

(x) Power spectral density function is a ________
(a) real and even function (b) non negative function
(c) periodic (d) all of the above mentioned.

Group – B

2. (a) Determine whether the following signal is energy or power signal:
$x(n) = u(n) - u(n-4)$.

(b) Determine the linearity of the system $y(n) = e^{x(n)}$.

(c) Perform the convolution of the following two signals:
$x(n) = \{0,1,-2,3,4\}$, $h(n) = \left\{\frac{1}{2}, \frac{1}{2}, 1, \frac{1}{2}\right\}$.

4 + 4 + 4 = 12

3. (a) A 4-point sequence is given by $x(n) = \{2,1,2,1\}$. Compute 4-point DFT of $x(n)$ by radix-2 DIT-FFT algorithm.

(b) Compute the IDFT of the sequence $Y(k) = \{4,2,0,2\}$.

6 + 6 = 12

Group – C

4. (a) Convert the analog filter transfer function $H(s) = \frac{1}{(s+2)^2(s+1)}$ into digital filter transfer function using bilinear transform technique.

(b) Design an FIR filter (low pass) using rectangular window with pass band gain of 0 dB, cut-off frequency of 200 Hz and sampling frequency of 1 kHz. Assume the length of the impulse response as 7.

4 + 8 = 12

5. (a) Realize the direct form-II structure of the following transfer function of a digital filter: $H(z) = \frac{8z^3-4z^2+11z-2}{\left(z-\frac{1}{4}\right)\left(z^2-z+\frac{1}{3}\right)}$

(b) For the given FIR filter obtain a realization structure with minimum number of multipliers: $H(z) = \frac{1}{4} + \frac{1}{2}z^{-1} + \frac{3}{4}z^{-2} + \frac{1}{2}z^{-3} + \frac{1}{4}z^{-4}$

6 + 6 = 12

Group – D

6. (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 8 kHz. Suppose that we wish to isolate the frequency components below 80 Hz with a filter that has a pass band $0 \leq F \leq 75$ and a transition band $75 \leq F \leq 80$. 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 pass band ripple $\delta_1 = 10^{-2}$ and a stop band ripple of $\delta_2 = 10^{-4}$.

3 + 9 = 12

7. (a) What are the characteristics of Adaptive filters? Define the adaptive filtering problem.

(b) Describe with block diagram how adaptive filters can be used for echo cancellation.

(c) Describe in brief the RLS algorithm of adaptive filter.

4 + 4 + 4 = 12

Group – E

8. (a) State the steps of parametric method of power spectrum estimation. What are AR and ARMA processes? Why AR model is widely used for power spectrum estimation?

(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 $|f| > 5\ kHz$. Only 10 seconds of the signal has been recorded and is available for the processing. We would like to estimate the power spectrum of $x_a(t)$ using the available data in a radix-2 FFT algorithm and is required that the estimate has a resolution of at least *10 Hz*. Suppose that we use Bartlett's method of periodogram averaging.

(i) If the data is sampled at Nyquist rate, what is the minimum section length that you may use to get the desired resolution?

(ii) With the minimum section length determined in part (i), with 10 seconds of data, how many sections are available for averaging?

(2 + 3 + 2) + 5 = 12

9. (a) State the linear prediction problem.

(b) Find an optimum linear predictor for an AR(1) process $x(n)$ that has autocorrelation sequence given by $r_x(k) = a^{|k|}$, with a first order predictor. Also deduce the mean-square error of the designed linear predictor.

3 + 9 = 12

Department & Section	Submission Link
AEIE	https://classroom.google.com/c/MjE4NzM2NjA1NTcx/a/MjcwOTcwOTkwOTE3/details