FUNDAMENTALS OF DIGITAL SIGNAL PROCESSING (AEIE 3104)

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) Aliasing is the phenomenon of
(a) high frequency component getting the identity of low frequency component during sample
(b) low frequency component getting the identity of high frequency component during sample
(c) high frequency noise
(d) both b and c

(ii) For energy signal
(a) Energy will be finite and average power will be infinite
(b) Energy will be finite and average power will be zero
(c) Energy will be infinite and average power will be zero
(d) Energy will be finite and average power will be finite

(iii) For analog signal with maximum frequency F_m , the sampling frequency should be
(a) greater than $2F_m$ (b) less than $2F_m$
(c) greater than $F_m/2$ (d) less than $F_m/2$

(iv) The process of conversion of continuous time signal into discrete time signal is known as
(a) aliasing (b) sampling
(c) convolution (d) none of these

(v) The region of convergence (ROC) of $X(z)$ is the set of all values for which $X(z)$ attains
(a) unity (b) zero
(c) infinite (d) finite

(vi) The ripple in FIR filter arises both in pass band and stop band due to
(a) symmetrical coefficients (b) non-symmetrical coefficients
(c) Gibb's phenomenon (d) few coefficients

(vii) The factor that influence the choice of realization of digital filter structure is,
(a) memory requirements (b) computational complexity
(c) parallel processing and pipelining (d) all the above.

(viii) The quantization error increases, when the order of the system $'N'$ increases in case of,
(a) cascade or parallel form realization (b) direct form realization
(c) all IIR systems (d) all FIR systems

(ix) For short-time, low-energy transients, the change in the spectrum is easily detected by
(a) Fourier Transform (b) Wavelet Transform
(c) both (a) and (b) (d) none of (a) and (b)

(x) If a linear phase filter has a phase response of 40 degree at 200 Hz, what will its phase response be at a frequency of 400 Hz? (Assuming that both frequencies are in the passband of the filter)
(a) 20 degree (b) 40 degree
(c) 60 degree (d) 80 degree

Group – B

2. (a) Define Accumulator and express it as a recursive system. Determine the impulse and step response of an up-sampler.
(b) Express the sequence, $x(n) = \left\{\underset{\uparrow}{0}, 1,2,3\right\}$ as a function of unit step signal and unit impulse signal.

(2+2+3)+5= 12

3. (a) What are the differences between linear and circular convolution? How do you represent $U(n) + U(-n)$?
(b) Find the z-transform of $x(n) = e^{-a} \quad U(n)$.
(c) Determine the response of the LTI system whose impulse response $h(n)$ and input $x(n)$ are given by, $h(n) = \left\{\underset{\uparrow}{1}, 2,3,1\right\}$ and $x(n) = \left\{1, \underset{\uparrow}{2}, 1, -1\right\}$.

(3+2)+ (3+4) = 12

Group – C

4. (a) Define DFT and Inverse DFT of a discrete time system. State and prove the shifting property of DFT.
(b) Why FFT is needed? Compare the DIT and DIF radix-2 FFT.

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

5. (a) Find the DFT of the sequence $x(n) = \{1,1,0,0\}$. Also find magnitude and phase of the sequence.
   (b) Find the inverse DFT of $X(k) = \{1,0,1,0\}$

(4+2+2)+4=12

Group – D

6. (a) State the condition for a digital filter to be causal and stable.
   (b) A low pass digital Butterworth filter meeting the following specification is required:

Pass band	0-800 Hz
Stop band	2-4 kHz
Pass band ripple	2 dB
Stop band attenuation	20 dB
Sampling Frequency	8 kHz

Determine the following:
   (i) Pass and stop band edge frequencies for a suitable analog low pass prototype filter.
   (ii) Order N of the prototype low pass filter.
   (iii) Coefficients and hence the transfer function of the discrete time filter using bilinear Z-transform.

2+(3+2+5)=12

7. (a) Why impulse invariant method is not preferred in the design of IIR filters other than low pass filter?
   (b) Transform the analog filter $H_a(s) = \frac{s+1}{s^2+5s+6}$ into a digital filter $H(z)$ using the impulse invariance technique in which $T = 0.1$.
   (c) Obtain the direct form-II realization of the LTI system governed by the equation $y(n) = -\frac{3}{8}y(n-1) + \frac{3}{3}y(n-2) + \frac{1}{6}y(n-3) + x(n) + 3x(n-1) + 2x(n-2)$

2+4+6=12

Group – E

8. (a) What is multi rate DSP? What is up-sampling method? Explain with block diagram and its input-output relationship in time domain
   (b) Given an input sequence $x(n) = \{\ldots,2,3,1,\underline{4},-2,5,3,\ldots\}$, where the underlined number represents $x(0)$, to an up-sampler $y(n) = [\uparrow 2]x(n)$, find out the output sequence $y(n), X(z)$ and$Y(z)$.

(2+5)+5=12

9. (a) What are the limitations of FFT and how it is overcome by STFT? What is spectrogram?
   (b) What is wavelet transform? What are the applications of wavelets? Write down the expression of forward and inverse continuous wavelet transform and explain each term.

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

Department & Section	Submission Link
AEIE	https://classroom.google.com/c/MTM4NDgzODgxMDQy/a/MjcwOTcxOTM4NzUx/details