ROLL NO. ___________________

Code: AE77/AC77/AE121 **Subject: DIGITAL SIGNAL PROCESSING**

AMIETE – ET/CS (Current & New Scheme)

Time: 3 Hours **December 2016** **Max. Marks: 100**

PLEASE WRITE YOUR ROLL NO. AT THE SPACE PROVIDED ON EACH PAGE IMMEDIATELY AFTER RECEIVING THE QUESTION PAPER.

NOTE: There are 9 Questions in all.

- **Question 1 is compulsory and carries 20 marks. Answer to Q.1 must be written in the space provided for it in the answer book supplied and nowhere else.**
- **The answer sheet for the Q.1 will be collected by the invigilator after 45 minutes of the commencement of the examination.**
- **Out of the remaining EIGHT Questions answer any FIVE Questions. Each question carries 16 marks.**
- **Any required data not explicitly given, may be suitably assumed and stated.**

Q.1 Choose the correct or the best alternative in the following: **(2×10)**

a. Which of the following best defines Nyquist frequency?
(A) The frequency of resonance for the filtering circuit
(B) The second harmonic
(C) The lower frequency limit of sampling
(D) The highest frequency component of a given analog signal

b. If G(f) represents the Fourier Transform of a signal g(t) which is real and odd symmetric in time, then G(f) is
(A) complex **(B)** imaginary
(C) real **(D)** real and non-negative

c. A band pass signal extends from 1 kHz to 2 kHz. The minimum sampling frequency needed to retain all information in the sampled signal is
(A) 1 kHz **(B)** 2 kHz
(C) 3 kHz **(D)** 4 kHz

d. The step response of an LTI system when the impulse response h(n) is unit step u(n) is
(A) n+1 **(B)** n
(C) n-1 **(D)** n^2

e. Determine the convolution sum of two sequences x(n) = {1, 2, 1, 2} and h(n) = {1, 0, 1, 0}
(A) y(n) = {1,2,1,2,1,0,1} **(B)** y(n) = {1,2,2,4,2,1,0}
(C) y(n) = {0,2,1,4,2,2,1} **(D)** y(n) = {1,2,2,4,1,2,0}

f. What is the N-point Discrete Fourier Transform (DFT) of the following sequence: x[n]=δ[n]?
(A) 1 **(B)** 2
(C) π **(D)** 1/π

g. Calculate DFT of x (n) = {1, 0, 1, 0}
(A) X(k) ={2, 0, 2, 0} **(B)** X(k) = {1, 0, 1, 0}
(C) X(k) = {2, 0, 1, 0} **(D)** X(k) = {0, 0, 1, 1}

h. The FFT algorithms:
(A) Eliminate the redundant calculation and enable to analyze the spectral properties of a signal.
(B) Enable the redundant calculation and redundant to analyze the spectral properties of a signal.
(C) Both **(A)** and **(B)** are correct
(D) Both **(A)** and **(B)** are wrong

i. The bilinear transformation mapping between continuous time system function and discrete time system function is

(A) $s = \frac{2}{T}\left(\frac{1+z^{-1}}{1-z^{-1}}\right)$ **(B)** $s = \frac{T}{2}\left(\frac{1-z^{-1}}{1+z^{-1}}\right)$

(C) $s = \frac{2}{T}\left(\frac{1-z^{-1}}{1+z^{-1}}\right)$ **(D)** $s = \frac{T}{2}\left(\frac{1+z^{-1}}{1-z^{-1}}\right)$

j. The Hilbert transformer output signal is a ______ degree phase shift of the input signal.
(A) 0 **(B)**180
(C) 270 **(D)** 90

Answer any FIVE Questions out of EIGHT Questions.
Each question carries 16 marks.

Q.2 a. The continuous time signal $x_a(t) = \sin(20\pi t) + \cos(40\pi t)$ is sampled with a sampling period T to obtain the discrete time signal $x[n] = \sin(\frac{\pi n}{5}) + \cos(\frac{2\pi n}{5})$. (i) Determine a choice for T consistent with this information. (ii) Is your choice for T in part (i) unique? If so, explain why. If not, specify another choice of T consistent with the information given. **(8)**

b. Explain in detail about the following terms **(8)**
(i) Sampling (ii) Quantization
(iii) Aliasing (iv) Nyquist rate

Q.3 When the input to an LTI system is $x[n] = \left(\frac{1}{2}\right)^n u[n] + 2^n u[-n-1]$, the output is $y[n] = 6\left(\frac{1}{2}\right)^n u[n] - 6\left(\frac{3}{4}\right)^n u[n]$.

(i) Find the system function H(z) of the system. Plot the poles and zeros of H(z), and indicate the region of convergence.
(ii) Find the impulse response h[n] of the system for all values of n.
(iii) Write the difference equation that characterizes the system.
(iv) Is the system stable? Is it causal? **(16)**

Q.4 a. For the system function $H(z)=\dfrac{2-\frac{8}{3}z^{-1}-2z^{-2}}{\left(1-\frac{1}{3}z^{-1}\right)\left(1+\frac{2}{3}z^{-1}\right)}$, draw a signal flow graph that implements this system as a parallel combination of first order transposed direct form II sections. **(6)**

b. The signal flow graph for a causal discrete time LTI system is shown in Figure 1. Branches without gains explicitly indicated have a gain of unity.
(i) Determine h[1], the impulse response at n = 1.
(ii) Determine the difference equation relating x[n] and y[n]. **(10)**

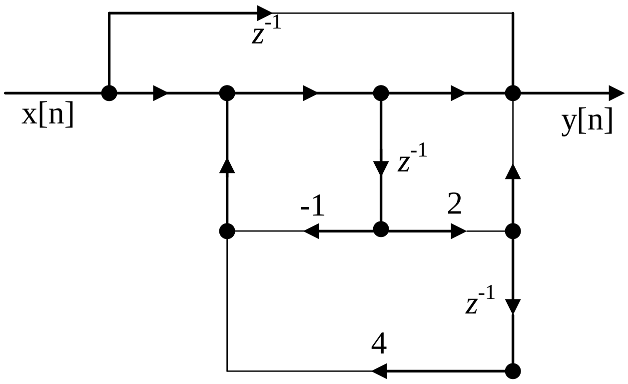

Figure 1.

Q.5 a. Use Kaiser window method to design a discrete time filter with generalized linear phase that meets specifications of the following form:

$$\begin{array}{rl} |H(e^{j\omega})| \le 0.01 & 0 \le |\omega| \le 0.25\pi \\ 0.95 \le |H(e^{j\omega})| \le 1.05 & 0.35\pi \le |\omega| \le 0.6\pi \\ |H(e^{j\omega})| \le 0.01 & 0.65\pi \le |\omega| \le \pi \end{array}$$

(i) Determine the minimum length of the impulse response and the value of the Kaiser window parameter β for a filter that meets the proceeding specifications.
(ii) What is the delay of the filter?
(iii) Determine the ideal response $h_d[n]$ to which the Kaiser window should be applied. **(8)**

b. Design a type I lowpass Chebyshev filter that has a 1-dB ripple in the pass band, a cutoff frequency $\Omega_p = 1000\pi$, a stopband frequency of 2000π, and an attenuation of 40 dB or more for $\Omega \ge \Omega_s$. Also determine the order and poles of the filter. **(8)**

Q.6 a. Determine the Discrete Fourier transform of the following signals.
(i) $x_1(n)=u(n)$, (ii) $x_1(n)=(\cos\omega_0 n)u(n)$. **(8)**

b. Prove the following DFT properties: **(8)**
(i) circular shift of a sequence
(ii) Duality

Q.7 Compute the eight point DFT of the sequence $x(n)=\left\{\frac{1}{2},\frac{1}{2},\frac{1}{2},\frac{1}{2},0,0,0,0\right\}$ using the in-place radix-2 decimation in time and radix-2 decimation in frequency algorithms. **(16)**

Q.8 a. Explain in detail about time dependent Fourier analysis of speech signals and radar signals. **(8)**

b. Derive the expression for estimation of the autocorrelation and power spectrum of random signals. **(8)**

Q.9 a. Consider a sequence $x[n]$ with discrete time Fourier transform $X(e^{j\omega})$. The sequence $x[n]$ is real valued and casua, and $\mathrm{Re}\{X(e^{j\omega})\}=2-2a\cos\omega$. Determine, $\mathrm{Im}\{X(e^{j\omega})\}$. **(4)**

b. Find the Hilbert transforms $x_i[n]=\mathrm{H}\{x_r[n]\}$ of the following sequences: (i) $x_r[n]=\cos\omega_0 n$. **(6)**

c. The imaginary part of $X(e^{j\omega})$ for a causal, real sequence $x[n]$ is $X_I(e^{j\omega})=2\sin\omega-3\sin 4\omega$. Additionally, it is known that $X(e^{j\omega})\big|_{\omega=0}=6$. Find $x[n]$. **(6)**