ROLL NO. ___________________

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

AMIETE – ET/CS (Current Scheme)

Time: 3 Hours **JUNE 2015** **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. z- Transform of u[n] is

(A) z/1–z **(B)** z^{-n}
(C) z^{-1} **(D)** $1/1-z^{-1}$

b. A quantizer operates at a sampling frequency of 16 KHz. What is the Nyquist limit

(A) 4 KHz **(B)** 8 KHz
(C) 16 KHz **(D)** 32 KHz

c. Basic process going on inside DSP chip is

(A) Quantization **(B)** MAC
(C) Logarithmic transformation **(D)** Vector calculation

d. Which window is used to alter FIR filter coefficients so that they smoothly approach zero at both ends.

(A) Blackman window **(B)** Rectangular window
(C) Laplace window **(D)** Hilbert transform

e. IIR filters

(A) use feedback **(B)** are sometimes recursive
(C) can oscillate **(D)** all of these

f. Inverse Fourier Transform converts:

(A) Frequency domain to time domain
(B) Time domain to frequency domain
(C) Continuous time to discrete time
(D) None of these

g. Coefficient symmetry in FIR provides.

(A) Smaller bandwidth **(B)** Less ripples in passband
(C) Less stopband ripples **(D)** Linear phase

h. A DSP convolves discrete samples with these coefficients: –0.75, –0.25, 1, 0.25, 0.75, this must be

(A) Low pass filter **(B)** High pass filter
(C) Band pass filter **(D)** Band stop filter

i. FFT is used to compute

(A) Laplace Transform **(B)** Fourier Transform
(C) DFT **(D)** Hilbert Transform

j. If x[n + N] = x[n] then X[k + N] is

(A) X [k] **(B)** 0
(C) 1 **(D)** X [N]

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

Q.2 a. Derive the frequency domain relation between input and output of an ideal continuous to discrete converter. **(8)**

b. With the help of complete calculations show that SNR increases by approximately 6 dB with each bit added to word length. **(8)**

Q.3 a. Find the system function & its ROC for an LTI system with input and output related as $y[n]-\frac{5}{2}y[n-1]+y[n-2]=x[n]$ when **(8)**

(i) system is neither stable nor causal
(ii) system is stable
(iii) system is causal and unstable

b. What is a 'minimum phase system'? Discuss its three properties. **(8)**

Q.4 a. Consider a LTI system

with $H(z)=\dfrac{1+2z^{-1}}{1-1.5z^{-1}+0.9z^{-2}}$

Implement it using Direct form I & Direct form II. **(8)**

b. Determine FIR linear phase and cascade realization of the system function which is expressed as

$$H(z)=\left(1+\frac{1}{2}z^{-1}+z^{-2}\right)\left(1+\frac{1}{4}z^{-1}+z^{-2}\right)$$ **(8)**

Q.5 a. Explain the mapping of s-plane to z-plane using Bilinear Transformation. Can we obtain a discrete time Low Pass filter with Linear phase characteristics by applying bilinear transformation to a continuous time low pass filter with linear phase characteristics? Justify. **(8)**

b. Explain the process of windowing using illustrations. Obtain frequency domain characteristics of rectangular window function. **(8)**

Q.6 a. Find DFT of the sequence which is expressed as: **(8)**

$$x[n] = \begin{cases} \frac{1}{5}, & -1 \le n \le 1 \\ 0, & \text{otherwise} \end{cases}$$

b. Sketch the linear and circular convolution of the two finite length sequences $x_1[n] = \{1,2,3,4,5,6\}$ and $x_2[n] = \{0,0,1\}$ **(8)**

Q.7 a. Determine 8-point DFT of

$x[n] = \left\{\frac{1}{2},\frac{1}{2},\frac{1}{2},\frac{1}{2},0,0,0,0\right\}$

Use Radix -2 decimation in time FFT algorithm. **(8)**

b. Write short note on The Goertzel Algorithm & Bit reversal. **(8)**

Q.8 a. Show that Fourier Transform of windowed signal consists of Fourier Transform of window replicated at frequencies $\pm\omega_0$ and $\pm\omega_1$ and scaled by complex amplitudes of individual complex exponentials that make up the signal. **(8)**

b. Prove that power density spectrum is Fourier Transform of auto correlation function. **(8)**

Q.9 a. Explain Hilbert Transform relationships. **(8)**

b. Show that when complex cepstrum of a sequence is causal, both poles & zeros of its z-transform lie inside the unit circle. **(8)**