Reg. No.									

MANIPAL INSTITUTE OF TECHNOLOGY
MANIPAL

A Constituent Institution of Manipal University

V SEMESTER B.TECH (ELECTRICAL & ELECTRONICS ENGINEERING)

END SEMESTER EXAMINATIONS, NOV/DEC 2016

SUBJECT: DIGITAL SIGNAL PROCESSING [ELE 3102]

REVISED CREDIT SYSTEM

Time: 3 Hours **Date: 26 November 2016** **MAX. MARKS: 50**

Instructions to Candidates:

- Answer **ALL** the questions.
- Missing data may be suitable assumed.
- DSP quick – reference table may be provided.

1A. Find 4 – point DFT for a discrete – time signal obtained by sampling a continuous – time signal $x(t) = \sin(4\pi Ft)$ with $F = 60\,Hz$ and sampling frequency $F_s = 960\,cycles/\sec$ using radix – 2, DIT – FFT algorithm. Show all intermittent values on the butterfly diagram. ***(04)***

1B. Consider an analog signal represented as $x(t) = 20\cos\left(40\pi t - \frac{\pi}{3}\right) - 10\cos(100\pi t)$ applied to a sampling and reconstruction system.

(i) What value of minimum sampling rate F_s will ensure $y(t) = x(t)$?

(ii) How should F_s be chosen so that $y(t) = B + 20\cos\left(40\pi t - \frac{\pi}{3}\right)$?

(iii) What is the value of constant B? ***(03)***

1C. The 8-point DFT of sequence $v[n] = x[n] + jh[n]$ is

$V[k] = \{-2 + j3, 1 + j5, -4 + j7, 2 + j6, -1 - j3, -4 - j2, 3 + j8, j6\}$

Without computing IDFT, determine the 8-point DFTs of $x[n]$ and $h[n]$. ***(03)***

2A. Determine the output of a filter whose impulse response is

$h[n] = \delta[n] - 2\delta[n-1] + \delta[n-2]$ and the input signal $x[n] = \begin{cases} \cos[2\pi n]; & 0 \le n \le 11 \\ 0 & ; otherwise \end{cases}$

Use Overlap-save method. Take sub-frame length of 4. ***(04)***

2B. Realize the given IIR filter using

i) Direct form – II

ii) Parallel form structure

$$H(z) = \frac{1 + z^{-1}}{(1 - 0.5z^{-1})(1 - 3z^{-1} + 2z^{-2})}$$

(03)

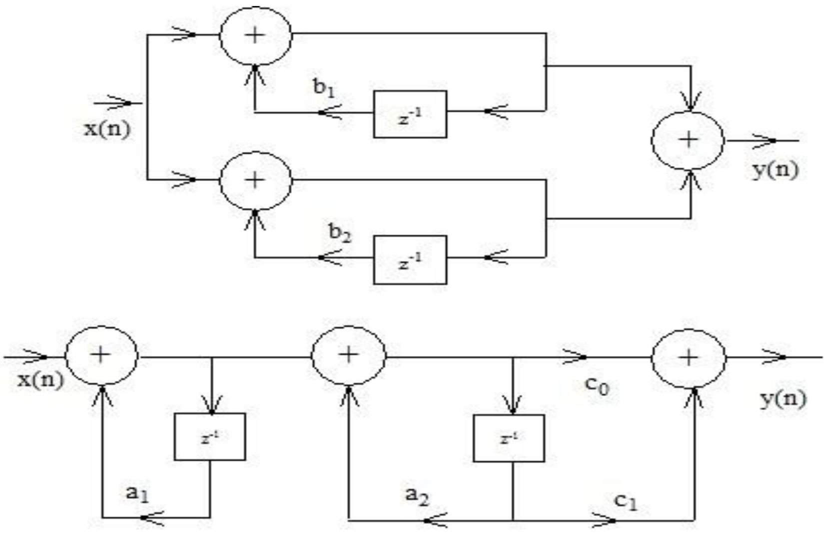
2C. The reflection coefficients of a 3 – stage FIR lattice structure are: $K_1 = 0.1$, $K_2 = 0.2$, and $K_3 = 0.3$. Obtain the system function. ***(03)***

3A. Determine the coefficients h(n) of a highpass linear phase FIR filter of length M = 4 which has an antisymmetric unit sample response h(n) = – h(M–1–n) and a frequency response that satisfies the condition

$$\left|H\left(\frac{\pi}{4}\right)\right| = \frac{1}{2} \quad and \quad \left|H\left(\frac{3\pi}{4}\right)\right| = 1$$ ***(03)***

3B. Design a FIR digital that will reject a very strong 50 Hz sinusoidal interference contaminating a 200 Hz useful sinusoidal signal. Determine the gain of the filter such that the filter does not change the amplitude of the useful signal. Assume sampling frequency F_s = 500 Hz. Draw the pole zero plot. Suggest the scheme to improve the performance of such filtering. ***(04)***

3C. Determine a_1, a_2, c_1 and c_0 in terms of b_1 and b_2 so that the two systems shown in figure below are equivalent.



(03)

Figure (3C)

4A. List the differences between FIR and IIR filters. ***(03)***

4B. Why ideal filters are non – realizable filters? Give reasons. ***(02)***

4C. Design a symmetric finite impulse response filter with desired frequency response

$$|H_d[e^{j\omega}]| = \begin{cases} 1; & for\ |\omega| \le \frac{\pi}{4} \\ 0; & otherwise \end{cases}$$

Determine the coefficients of 4th order filter using Hanning window. Draw relevant filter structure. Obtain its frequency response equation. ***(05)***

5A. The transfer function of an analog filter is given by

$$H(s) = \frac{2}{s+2}$$

Design a digital filter $H[z]$ from $H(s)$ using impulse invariance method. Use a sampling frequency of 1 Hz. Also, draw the pole – zero plot. ***(04)***

5B. Design a Butterworth IIR filter that satisfies the following specifications:

$$0.75 \le \left|H\left[e^{j\omega}\right]\right| \le 1; \quad for\ 0 \le |\omega| \le 0.25\pi$$

$$\left|H\left[e^{j\omega}\right]\right| \le 0.23; \quad for\ 0.63\pi \le |\omega| \le \pi$$

Use bi-linear transformation technique. Take sampling frequency of 8 KHz. ***(06)***