Reg. No.									

MANIPAL INSTITUTE OF TECHNOLOGY
MANIPAL
(A constituent unit of MAHE, Manipal)

V SEMESTER B.TECH (ELECTRICAL & ELECTRONICS ENGINEERING)
END SEMESTER EXAMINATIONS, NOVEMBER 2019

DIGITAL SIGNAL PROCESSING [ELE 3102]
REVISED CREDIT SYSTEM

Time: 3 Hours **18^{th} November 2019** **Max. Marks: 50**

Instructions to Candidates:

- Answer **ALL** the questions.
- Missing data may be suitably assumed.
- DSP Quick reference table may be provided.

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

i. What value of the minimum sampling rate F_S will ensure y(t)=x(t)?
ii. Determine the discrete-time signal x(n) for Fs=300 Hz.
iii. What is the value of F_S to be chosen so that $\mathrm{y(t)} = \mathrm{D} + 20\cos\left(40\pi\mathrm{t} - \frac{\pi}{3}\right)$? where D is a constant. ***(03)***

1B. Show that M length FIR filter with impulse response satisfying the condition h(M-n-1) = h(n) has linear phase response. (Assume odd length for derivation). ***(03)***

1C. Illustrate the over-lap save method for filtering long data sequence for $x(n)=[\underset{\uparrow}{1}\quad 2\quad 3\quad 2\quad 1\quad -1\quad 2\quad 4\quad 5]$ and $h(n)=[\underset{\uparrow}{1}\quad -1\quad 1]$ using input sub frames of length 5. ***(04)***

2A. Explain the need for antialiasing filter in digital signal processing system. ***(02)***

2B. Determine the 8 point DFT of the sequence x(n) using radix 2 decimation in frequency FFT algorithm where $x(n) = \begin{cases} 1; & -3 \le n \le 1 \\ 0; & otherwise \end{cases}$

Show all the intermediate values in the signal flow graph. ***(05)***

2C. Compare the characteristics of Butterworth, Chebyshev Type-I with N even and N odd, where N is the order of the filter. Also plot magnitude response in each case. ***(03)***

3A. For the given 6 point sequence x(n) = [1, 1, 4, 3, 2, 2] for -2≤n≤3.

i. Find the 3 point DFT Z(k) of the sequence z(n)=x(2n) with 0≤k≤2.
ii. Find 3 point DFT Y(k) of a y(n)=z(n+1) ***(05)***

3B. A linear phase FIR digital filter is described by difference equation $y(n) = \sum_{k=0}^{2} b_k x(n-k)$. Determine the filter coefficient such that it rejects a frequency component at $\omega_o = \frac{\pi}{3}$ and its frequency response is normalised so that $\left|H(e^{j\omega})\right|_{\omega=0} = 1$ ***(03)***

3C. Write a technical note on all pass filter. Also mention the location of poles and zeros in z plane. ***(02)***

4A. Determine the lattice-ladder parameters for the following digital filter. Sketch lattice-ladder structure.

$$H(z) = \frac{1 - 8z^{-1} + 0.15z^{-2}}{2 + 0.2z^{-1} - 1.44z^{-2}}$$

Comment on the stability of the system. ***(04)***

4B. A causal linear phase symmetric FIR system $H(z)$ which has 4 zeros is given by $H(z) = H_1(z)(1 - 0.8z^{-1} + 0.64z^{-2})$. Find impulse response h(n) of the FIR system. Take h(n) =1 for n=0. ***(02)***

4C. From fundamental design an FIR linear phase digital filter approximating the ideal frequency response

$$\left|H_d(e^{j\omega})\right| = \begin{cases} 1; & \frac{\pi}{4} \le |\omega| \le \frac{3\pi}{4} \\ 0; & otherwise \end{cases}$$

Determine the coefficients of a seven tap filter using Blackman window. ***(04)***

5A. Linear phase digital low pass filter with the following specifications is required.

Pass band 0 to 5 kHz

Sampling frequency 18 kHz

Filter length 9

Design this filter using frequency sampling method. ***(05)***

5B. Design a digital Butterworth filter using Bilinear transformation to satisfy the following specification.

$$\begin{cases} 0.8 \le \left|H(e^{j\omega})\right| \le 1; & for \quad 0 \le \omega \le 0.4\pi \\ \left|H(e^{j\omega})\right| \le 0.2; & for \quad 0.6\pi \le \omega \le \pi \end{cases}$$

Take sampling frequency of 1 Hz. ***(05)***