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

V SEMESTER B.TECH (ELECTRICAL & ELECTRONICS ENGINEERING)

MAKEUP EXAMINATIONS, MAY 2018

SUBJECT: DIGITAL SIGNAL PROCESSING [ELE 3102]

REVISED CREDIT SYSTEM

Time: 3 Hours **Date: 07 May 2018** **Max. Marks: 50**

Instructions to Candidates:

- Answer **ALL** the questions.
- Missing data may be suitably assumed.
- DSP–Quick Reference Table may be supplied.

1A. An analog input signal is composed of a weighted sum of three sinusoidal signals of different frequencies is given by

$$x_a(t) = 2\cos 120\pi t + 5\sin 180\pi t + 2\cos 350\pi t$$

If the sampling rate is $200\,\text{Hz}$. (i) Find the Nyquist rate for this signal. (ii) Determine the discrete time signal $x[n]$ obtained after sampling (iii) Illustrate aliasing if any. ***(05)***

1B. For the given real sequence $x[n] = \delta[n+2] + 2\delta[n] - \delta[\text{n}-3]$
Find 6 - point DFT of $x[n]$ making use of Twiddle factor. ***(05)***

2A. Given the two 4-point DT sequences $x[n]$ and $\text{h}[n]$ as below

$$x[n] = \{1,\ 1,\ 0,\ 2\};\ 0 \le n \le 3\ ;\ h[n] = \{1,\ 0,\ 2,\ 1\};\ 0 \le n \le 3$$

Determine 4-point circular convolution of $x[n]$ and $h[n]$ ***(05)***

2B. Determine the output $y[n]$ of a filter using Overlap save method whose impulse response is
$h[n] = \delta[\text{n}] - \delta[\text{n}-1] + 2\delta[\text{n}-2]$ and
input sequence $x[n] = \{1,\ -1,\ 2,\ -2,\ 3,\ -3,\ 4,\ -4,\ 1,\ 3,\ 2,\ 1\}$. Take sub-frame length of 4. ***(05)***

3A. Find the 8-point DFT using radix–2 DIT FFT algorithm and show all intermediate values on butterfly diagram for a time domain sequence $x[n] = \{1,\ 2,\ -2,\ 1,\ 0,\ 0,\ 3,\ -2\}$ ***(05)***

3B. Draw the direct form I, direct form II structures for digital IIR filter described by the system function

$$H(z) = \frac{5 - 4z^{-1} + 7z^{-2} - 2z^{-3}}{1 - \frac{3}{2}z^{-1} + \frac{4}{5}z^{-2} - \frac{1}{8}z^{-3}}$$

(05)

4A. Sketch the lattice filter implementation of an FIR filter for the system function

$$H(z) = 10 + 5z^{-1} + 2z^{-2} + z^{-3}$$

(05)

4B. Design a linear phase FIR low-pass filter using Hamming window by taking 7 samples of the window sequence with desired frequency response:

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

(05)

5A. Design an ideal low-pass FIR digital filter using frequency sampling method to satisfy the following conditions.

Length of filter: 7

Sampling frequency: $16\,kHz$

Pass-band: $0 \le F \le 4\,kHz$

(05)

5B. Design a digital low-pass Butterworth filter using Bilinear transformation to meet the following specifications:

$$0.8 \le \left|H\left(e^{j\omega}\right)\right| \le 1; \quad 0 \le \omega \le 0.2\pi$$

$$\left|H\left(e^{jw}\right)\right| \le 0.2; \qquad 0.6\pi \le \omega \le \pi$$

(05)