Reg. No.										

MANIPAL INSTITUTE OF TECHNOLOGY

Manipal University

FOURTH SEMESTER B.TECH (E & C) DEGREE END SEMESTER EXAMINATION - APRIL / MAY 2017

SUBJECT: DIGITAL SIGNAL PROCESSING (ECE - 2203)

TIME: 3 HOURS **MAX. MARKS: 50**

Instructions to candidates

- Answer **ALL** questions.
- Missing data may be suitably assumed.

1A. Determine the total solution for $n \geq 0$ of the difference equation

$y[n] - \frac{1}{9}y[n-2] = x[n-1]$, with initial conditions $y[-1] = 1$ and $y[-2] = 0$ and $x[n] = 2u[n]$.

1B. An LTI system is used to filter data. Explain how the system output can be obtained through DFT/IDFT evaluation. Discuss overlap-add method of filtering long length input data.

1C. Consider an LTI system having 3 poles at -3, -0.5, 2 and a zero at 1. Determine the ROC of the system function H (z) for the system to be stable. Is it possible to have a causal and stable system?

(5+3+2)

2A. A receiver in a communication system has to detect 5 KHz component in a signal by computing 32 point DFT using Goertzel algorithm. Determine the impulse response and second order system function of the Goertzel filter to perform this computation. Assume sampling frequency as 20KHz.

2B. Certain analog signal $x_a(t)$ is band limited to 3.3KHZ and sampled. The spectrum is to be computed using $N=2^m$ point DFT with a resolution of 60Hz. Determine

i)Minimum sampling frequency

ii) Minimum number of required signal samples and corresponding length of analog signal record.

iii)Minimum DFT length.

2C. If X(k) is the N point DFT of x(n), determine N point DFT of

$$x_C(n) = x(n)\cos\left(\frac{2\pi k_0 n}{N}\right) \; and \; x_S(n) = x(n)\sin\left(\frac{2\pi k_0 n}{N}\right); 0 \leq n \leq N-1$$

(5+3+2)

3A. Compute the coefficients of Linear phase FIR LPF filter using window design to meet the following specifications.

Pass band edge frequency: 1Khz

Stop band edge frequency: 7.6Khz

Minimum stop band attenuation: 50dB. Assume sampling frequency of 22Khz.

3B. Express the system function H(z) of an M length FIR filter in terms of sampled frequency response H(k). Describe how coefficients of such filter are estimated using this concept.

3C. Why are linear phase FIR filters preferred? Determine the group delay of the filter designed in Q3a.

(5+3+2)

4A. Certain IIR Butterworth filter satisfies the following specifications

$$0.75 \le \left|H(e^{j\omega})\right| \le 1 \text{ , } 0 \le \omega \le 0.25\pi$$
$$\left|H(e^{j\omega})\right| \le 0.23 \text{ , } 0.63\pi \le \omega \le \pi$$

This filter has to be designed using bilinear transformation technique. Determine the corresponding edge frequencies, the minimum order and 3db cut-off frequency of the corresponding analog filter. Assume sampling frequency of 8Khz. Also determine the poles and transfer function H(s) of the analog filter

4B. Describe the digitisation of analog filter using impulse invariance technique.

4C. Determine the corresponding system function H(z) of the analog filter designed in Q4A using bilinear transformation technique with sampling frequency 8KHZ.

(5+3+2)

5A. A system is described by the difference equation

$$y(n) = \frac{-13}{24}y(n-1) - \frac{5}{8}y(n-2) - \frac{1}{3}y(n-3) + x(n) + 2x(n-1) + 2x(n-2) + x(n-3).$$

Determine the lattice and ladder parameters for this filter and draw the corresponding lattice ladder structure. Also comment on the stability of the system.

5B. Define PSD of a random signal. Explain the principle of Welch method of PSD estimation. Highlight the computational requirement of this method.

5C. List the main advantages of parametric methods of PSD estimation.

(5+3+2)