Reg. No.										

MANIPAL INSTITUTE OF TECHNOLOGY
Manipal University

FIFTH SEMESTER B.TECH (E & C) DEGREE END SEMESTER EXAMINATION
NOV/DEC 2015
SUBJECT: DIGITAL SIGNAL PROCESSING [ECE 303]

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

Instructions to candidates

- Answer **ANY FIVE** full questions.
- Missing data may be suitably assumed.

1A. Distinguish between z-transform and unilateral z-transform. Using unilateral z-transform, compute y(n) for $n \geq 0$ for

$$y(n) + 0.5y(n-1) - 0.25y(n-2) = 0; \quad y(-1) = y(-2) = 1.$$

1B. It is required to have a stable digital pole-zero system function to produce peak at 2KHz with unity gain operating at sampling frequency 8KHz. The response should decay on either side of the peak with 3-dB bandwidth 100Hz and reach zero at 0 and 4KHz. Identify the poles and zeroes of the system in z-domain and deduce the system function.

1C. Draw the magnitude response, phase response and z-domain pole-zero plot for the system $h(n) = \delta(n+1) + \delta(n) + \delta(n-1)$

(5+3+2)

2A. With relevant mathematical analysis, describe filtering of long length data through DFT-IDFT calculations.

2B. Compute 6-point DFT of the signal x(n)={3,2,1,0,1,2}

2C. Let x(n) be a N-length signal and v(n) is one period ($0 \leq n \leq N-1$) of a periodic signal obtained by periodically extending x(n). Write the relation between Fourier transform X(w) of x(n) and N- point DFT V(k) of v(n).

(5+3+2)

3A. Develop N point radix-2 DIT FFT algorithm. Illustrate with signal flow diagram and mention the computational advantage of this algorithm.

3B. Deduce second order Goertzel filter for the computation of N-point DFT. Draw the Direct form-2 implementation structure of this filter.

3C. Let $x_1(n)$ and $x_2(n)$ be two real valued sequences with N-point DFTs $X_1(k)$ and $X_2(k)$. Let X(k) be N-point DFT of $x(n) = x_1(n) + jx_2(n)$. Express $X_1(k)$ and $X_2(k)$ in terms of X(k).

(5+3+2)

4A. It is required to have a 9-length digital FIR low pass filter with pass band 0 to 5KHz at a sampling frequency of 18 KHz. Obtain and implement the system function in recursive form using frequency sampling design. Only real coefficients are allowed for the implementation.

4B. Briefly describe the design of liner phase FIR digital filters using window functions.

4C. Mention the time domain and frequency domain conditions for N-length digital FIR filter to have linear phase response. What is the group delay for such filters?

(5+3+2)

5A. Design a digital Chebyshev type-1 filter with following specifications using bilinear transformation.

Pass band:0-500Hz; Stop band:2-4KHz; Pass band ripple=3dB; Stop band attenuation=30dB; Sampling frequency=8KHz

5B. It is required to design a digital filter to approximate the following normalized analog transfer function using impulse invariant transformation. The 3dB cut-off frequency is 100Hz and sampling frequency is 1Khz.

$$H(s) = \frac{1}{\left(s^2 + \sqrt{2}s + 1\right)}$$

5C. Determine the poles and zeroes of digital transfer function corresponding to following analog transfer function obtained by using matched z-transform.

$$H(s) = \frac{(s+1)}{\left(s^2 + 2s + 5626\right)}$$

(5+3+2)

6A. Develop and draw the lattice ladder structure for the filter described by the difference equation, $y[n] + \frac{3}{4}y[n-1] + \frac{1}{4}y[n-2] = x[n] + 2x[n-1]$.

6B. Discuss the salient features of Welch method of PSD estimation.

6C. Define Periodogram. List limitations of periodogram method of PSD estimation.

(5+3+2)