Rajiv Gandhi Proudyogiki Vishwavidyalaya, Bhopal

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MEVD-104

M.E./M.Tech. (First Semester) EXAMINATION, Dec 2011 (Grading/Non-Grading) DIGITAL SIGNAL PROCESSING

> Time: Three Hours Maximum Marks: GS:70

Note: Attempt any five questions.

- 1. (a) State sampling theorem. For any analog signal how the sampling rate is selected? Also discuss that if the sampling is performed at minimum sampling rate how reconstruction is done?
 - (b) Consider the analog signal:

$$X_a(t) = 3 \cos 100 \pi t$$

- i. Determine the minimum sampling rate required to avoid aliasing.
- ii. Suppose that the signal is sampled at the rate Fs = 75 Hz. What is the discrete time signal obtained after sampling?
- What is the frequency 0 < F < Fs/2 of a sinusoid that yields samples identical to these iii. obtained in part (ii).
- 2. (a) State and prove the following properties of Z-transform:
 - i. Time Shifting
 - Differentiation in Z-domain
 - (b) Determine the Z-transform and the ROC of the signal:

$$X(n) = [3(2^n) - 4(3^n)] 4(n)$$

3. (a) Determine the inverse Z-transform of:

$$\frac{1}{1 - 1.5z^{-1} + 0.5z^{-2}}$$

When:

İ. ROC: |z| > 1ROC: |z| < 0.5

(b) Determine the system function and the unit sample response of the system described by the difference equation:

$$Y(n) = \frac{1}{2}y(n-1) + 2x(n)$$

- 4. (a) State and prove the following properties of DFT:
 - Circular Convolution i.
 - ii. Time Reversal of a sequence

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(b) A finite duration sequence of length L is given as:

$$x(n) = 1$$
 $0 \le n \le L-1$

$$x(n) = 0$$
 otherwise

Determine the N point DFT of this sequence for $N \ge L$.

- 5. (a) Compare in detail the computational complexity for the direct computation of the DFT versus the FFT algorithm.
 - (b) Draw and explain the flow graph of eight point decimation in time FFT algorithm.
- 6. (a) Discuss design of FIR digital filters using window method. Explain different types of windows used in the window design method.
 - (b) Determine the order and the poles of a type I low pass Chebyshev filter that has a 1-dB ripple in the pass band a cutoff frequency $\Omega_P = 1000\pi$, a stop band frequency of 2000π and an attenuation of 40 dB or more for $\Omega \ge \Omega_s$.
- 7. (a) How pipelining results in increased throughput of the DSP's? Explain in detail.
 - (b) Discuss in detail designing of programmable DSPs.
- 8. (a) Explain the effect of finite register length in FIR filter design.
 - (b) Discuss design of IIR digital filters using Butterworth approximation. Draw and explain its frequency response characteristics.