Roll No. ____________

2E9201

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M. Tech. II Sem. (Main & Back) Exam., July 2013
Digital Communication
2MDC1 Digital Signal Processing
Common for 2 MLV1, 2MDC1 & 2MCI1

Time: 3 Hours

Maximum Marks: 100

Min. Passing Marks: 33

Instructions to Candidates:

Attempt any **five questions.** Marks of questions are indicated against each question. Draw neat and comprehensive sketches wherever necessary to clearly illustrate your answer. Assume missing data suitably if any & specify the same.

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Use of following supporting material is permitted during examination. (Mentioned in form No.205)

1.____________ 2.____________

Q.1. (a) Discuss FFT algorithm using decimation in frequency technique and explain quantization errors in FFT algorithm. [12]

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

$$x(n) = \begin{cases} 1, 0 \le n \le L-1 \\ 0, \quad \text{otherwise} \end{cases}$$

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Determine N – point DFT of this sequence for $N \ge L$. [8]

Q.2. (a) Write short note on discrete cosine transform. [8]

(b) Design a digital filter using bilinear transformation for the following analog transfer function. $H(S) = \dfrac{1}{S^2 + \sqrt{2}S + 1}$

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Obtain transfer function H(S) of digital filter assuming 3db cut off frequency of 150 Hz and sampling frequency of 1.28 KHz. [12]

3. Describe poly phase decomposition methods with their suitable representation, also explain decimation and interpolation technique w.r.t multi-rate DSP. [20]

4. (a) Given $x(n) = 2^n$ and N =8, find X(k) using DIF FFT algorithm. [10]

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(b) Discuss the various steps for the design of a linear phase FIR filter using window method. [10]

(a) Discuss IIR filter design for Butterworth filter. [12]

(b) Use the backward difference for the derivative to convert the analog low pass filter with system function $H(s) = \dfrac{1}{S+2}$ [8]

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With the help of flow graph explain the implementation of decimation filter after applying the down sampling to identify the polyphase decomposition. [10]

The output of an A/D converter is applied to a digital filter with the system function $H(Z) = \dfrac{0.5Z}{Z-0.5}$

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Find the output noise power from the digital filter, when the input signal is quantized to have 8 bits. [10]

Make short notes on the following :

Rounding and truncation errors, [10]

Rectangular window function. [10]