Digital Electronics

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12EC123

M.Tech. Degree Examination, June/July 2013

Modern DSP

Time: 3 hrs. Max. Marks: 100

Note: ***Answer any FIVE full questions.***

Important Note : 1. On completing your answers, compulsorily draw diagonal cross lines on the remaining blank pages. 2. Any revealing of identification, appeal to evaluator and /or equations written eg, 42+8 = 50, will be treated as malpractice.

1 a. Explain the implications of analog and digital frequency relations due to periodic sampling of analog signal. **(08 Marks)**

b. Consider the analog signal $x_a(t) = 2\cos 100\pi t$:
 i) Determine the minimum sampling rate required to avoid aliasing.
 ii) If the signal is sampled at rate $F_S = 200$ Hz, what is the discrete time signal obtained after sampling.
 iii) If sampling rate $F_S = 75$ Hz used for sampling, then what is sampled signal obtained.
 iv) What is the effect of sampling done with $F_S = 75$ Hz? **(06 Marks)**

c. What are energy signals and power signals? **(06 Marks)**

2 a. If x(n) has N point DFT X(K), write the DFT if :
 i) x(n) is real and even
 ii) x(n) real and odd
 iii) x(n) is purely imaginary. **(08 Marks)**

b. If G(K) and H(K) are 6 point DFTs of length 6 sequences g(n) and h(n), if $h(n) = g((n-4))_6$, determine H(K) without computation of DFT, if
 $G(K) = [1+j,\ -2.1+j3.2,\ -1.2-j2.4,\ 0,\ 0.9+j3.1,\ -0.3+j1.1]$ **(06 Marks)**

c. If x(n), a finite duration sequence has N point DFT X(K), show that

$$\sum_{n=0}^{N-1} |x(n)|^2 = \frac{1}{N}\sum_{K=0}^{N-1} |X(K)|^2$$

from auto correlation property of DFT sequences. **(06 Marks)**

3 a. Explain the implications of DFT in frequency analysis of signal. **(07 Marks)**

b. A long sequence x(n) is filtered through a filter of impulse response h(n) to give output y(n). Given x(n) = [1 1 1 1 1 3 1 1 4 2 1 1 3 1 1 1], h(n) = [1, -1], compute y(n) using overlap add method. Use 5 point circular convolution. **(07 Marks)**

c. What is the condition for causality for physically realizable filter impulse response? Explain with Paley-Wiener relation. **(06 Marks)**

4 a. Calculate the filter coefficients for a 3-tap FIR LPF with cut-off frequency of 800 Hz and sampling rate of 8000 Hz. Determine transfer function of designed FIR system. Compute magnitude response. Use Hamming window. **(08 Marks)**

b. Write the design procedure for design of FIR filter by frequency sampling method. **(05 Marks)**

c. Design a linear phase LPFIR filter with 7 taps and a cut-off frequency $w_c = 0.3\pi$ radians by frequency sampling method. **(07 Marks)**

5 a. Design a digital lowpass Butterworth filter with the following specifications:
 i) 3 dB attenuation at the passband frequency of 1.5 Hz.
 ii) 10 dB stopband attenuation at the frequency of 3 kHz.
 iii) Sampling frequency of 8000 Hz. (Choose prototype $H_a(s)$ for $\epsilon = 1$). Use bilinear transformations. **(10 Marks)**

5 b. What are limitations of impulse invariant method of transformation? How is mapping improved with bilinear transformation? **(10 Marks)**

6 a. Write the efficient re-diagram of decimator and interpolator, justify your answer with FIR of the structure. **(10 Marks)**

b. A FIR filter based on Hamming window with following specification is to be implemented with (i) Single stage, (ii) Multistage (say 3 stage).

Passband is 0-450 Hz

Stopband > 500 Hz

Sampling frequency $F_S = 96$ kHz

Compare single stage and 3 stage implementation in terms of computation. **(10 Marks)**

7 a. Explain analysis and synthesis structure of UDFT filter bank, with efficient realization structure. **(10 Marks)**

b. With a block diagram, explain application of multirate DSP in subband coding of speech signals. **(10 Marks)**

8 Write technical note on:

a. Adaptive DF FIR filters and their applications **(10 Marks)**

b. Transmultiplexers for FDM to TDM and TDM to FDM. **(10 Marks)**

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