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10EC123

M.Tech. Degree Examination, June 2012

Modern DSP

Time: 3 hrs. Max. Marks:100

Note: ***1. Answer any FIVE full questions.***
2. Missing data, if any, may be suitably assumed.

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. With the help of block diagrams, explain the prediction based sampling methods. **(08 Marks)**

b. A continuous time signal x(t) has a bandwidth F_B = 10 kHz and it is sampled at F_S = 22 kHz, using 8 bits/sample. The signal is properly scaled so that $|x[n]| < 128$ for all n.

i) Determine a best estimate of the variance of the quantization error σ_e^2.

ii) If the sampling rate was increased by 16 times, how many bits per samples would you use in order to maintain the same level of quantization error? **(06 Marks)**

c. Find the DTFT of $x[n] = (0.8)^n u[n]$. Using the properties of DTFT, compute the DTFT of

i) $x[n] = (0.8)^n u[n-2]$ ii) $x[n] = n(0.8)^n u[n]$ **(06 Marks)**

2 a. Perform the i) circular convolution ii) linear convolution, using circular convolution of the sequences $x[n] = \{ \underset{\uparrow}{2}, 1, 2, 1 \}$ and $h[n] = \{ 1, 2, 3, 4 \}$ **(06 Marks)**

b. Consider a FIR filter with impulse response h(n) = { 3, 2, 1, 1 }. If the input is x[n] = { 1, 2, 3, 3, 2, 1, –1, –2, –3, 5, 6, –1 }, find the output using overlap.Add method. **(08 Marks)**

c. Compare IIR and FIR filters. **(06 Marks)**

3 a. Consider an IIR filter with transfer function

$$H(z) = \frac{2z^2 - z + 1.5}{z^2 - 1.6z + 0.8}$$

Draw the type I block diagram realization of H(z) and obtain the state space equations of the output equations in matrix form. **(08 Marks)**

b. Find the reflection coefficients and the lattice structure of an IIR filter given by H(z) = 1/A(z) where

$A(z) = 1 - 0.09z^{-1} - 1.4299z^{-2} + 0.2789z^{-3} + 0.4048z^{-4} - 0.1823z^{-5} + 0.0316z^{-6}$ **(12 Marks)**

4 a. Consider a sinusoidal signal $x[n] = 5\cos(0.1\pi n)$. Determine the frequency spectrum of the sampled signal $v(n) = \delta_3[n]\, x[n]$. **(10 Marks)**

b. Explain the frequency domain characterization of down sampling & up sampling. **(10 Marks)**

5 a. Explain the polyphase decomposition of a linear filter for down sampling and up sampling. **(10 Marks)**

b. Given that $H(z) = 1 + z^{-1} + 2z^{-2} - z^{-3} + z^{-4} - z^{-5} + z^{-6}$. Show a more efficient realization in terms of polyphase decomposition for systems shown in Fig.Q5(b). **(10 Marks)**

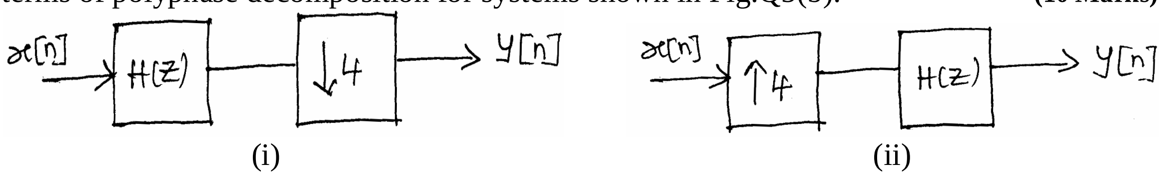

Fig.Q5(b)

6 a. Explain the analysis and synthesis of maximally decimated DFT filter banks. **(10 Marks)**

b. Explain the TDMA and FDMA techniques in the context of multirate DSP theory. **(10 Marks)**

7 a. Explain the design of perfect reconstruction filter banks with real coefficients. **(10 Marks)**

b. With the help of an example, explain the multiresolution decomposition by maximally decimated filter banks as applied to an audio signal. **(10 Marks)**

8 a. Define short time Fourier transform (STFT). Explain how STFT overcomes the limitations of the Fourier transform. **(07 Marks)**

b. Explain the Gabor transform. **(07 Marks)**

c. Write short notes on:
i) The Haar wavelet   ii) The Daulechies wavelets. **(06 Marks)**

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