Printed Pages – 5 **EC – 801**

(Following Paper ID and Roll No. to be filled in your Answer Book)

PAPER ID : 3048 **Roll No.** | | | | | | | | | | |

B.Tech.

EIGHTH SEMESTER EXAMINATION, 2005-2006

DIGITAL SIGNAL PROCESSING

Time : 3 Hours *Total Marks : 100*

Note : *(i)* *Attempt* ***ALL*** *questions.*

(ii) *All questions carry equal marks.*

(iii) *In case of numerical problems assume data wherever not provided.*

(iv) *Notations have their usual meaning unless otherwise stated.*

(v) *Be precise in your answer.*

1. Attempt *any four* parts of the following : **(5x4=20)**

(a) Prove that a sequence obtained by sampling a continuous time sinusoid at a rate $F_s = \frac{1}{T}$ is periodic if $\frac{T}{T_p} = \frac{K}{N}$. Where K and N are positive integers and T_p is the fundamental period of sinusoid.

(b) Show that the discrete-time signals $S_k(n) = exp\left(j\frac{2\pi}{N}Kn\right)$ are orthogonal over any interval of length N.

(c) Compute the fourier transform of

(i) $x(n) = \alpha^n \text{ Sin } \omega_o n,\ o<\alpha<1$

(ii) $x(n) = \left(\frac{1}{4}\right)^n u(n+4)$

(d) Determine z transform of

(i) $x(n) = -na^n\ u(-n-1)$

(ii) $x(n) = \text{Sin } (\omega_o n)\ u(n)$

(e) Show that an FIR filter can always be realized recursively.

(f) Show explicitly how one should interconnect 8 point FFT chips to compute a 24 point DFT.

2. Attempt *any four* parts of the following : **(5x4=20)**

(a) Define a type -II linear phase FIR filter and obtain its frequency response.

(b) Show that the impulse response of an ideal Band Pass filter is given by

$$h(n) = \begin{cases} \frac{\omega_2 - \omega_1}{\pi}, & n=0 \\ \frac{\text{Sin } n\omega_2}{\pi n} - \frac{\text{Sin } n\omega_1}{\pi n}, & n \neq 0 \end{cases}$$

Where ω_1 and ω_2 are lower and higher cut-off frequencies.

(c) A low pass causal FIR filter has following impulse response $\{h(n)\} = \{0.00126,\ -0.00523,\ -0.0197,\ -0.023,\ 0.023,\ 0.128,\ 0.247,\ 0.3\}$ Find the impulse response of a high-pass filter by suitable frequency transformation on the frequency characteristic of above low-pass filter.

(d) The impulse response of a causal FIR filter is $\{h(n)\} = \{-0.3, 0.4, 0.2, 0.5, 0.2, 0.4, -0.3\}$ Draw a realization structure so that a minimum number of multiplication is required for each output computation.

(e) What is the effect of finite wordlength with recursive frequency sampling FIR filters ? How can it be overcome in practice ?

(f) Kaiser window is to be used to design a linear phase FIR filter that meets following specification $|H(e^{j\omega})| \leq 0.01, \quad 0 \leq |\omega| \leq 0.25\pi$

$$0.95 \leq |H(e^{j\omega})| \leq 1.05, \; 0.35\pi \leq |\omega| \leq 0.6\pi$$

$$|H(e^{j\omega})| \leq 0.01, \; 0.65\pi \leq |\omega| \leq \pi$$

Determine the minimum length (M+1) of the impulse response and Kaiser Window parameter β.

3. Attempt *any two* parts of the following : **(10x2=20)**

(a) A bandpass digital filter is required to meet following specifications :

(i) Complete signal rejection at dc and at 250Hz.

(ii) Passband contered at 125 Hz.

(iii) 3 dB Bandwidth of 10 Hz.

(vi) Sampling frequency of 500 Hz.

Design the filter by the pole-zero placement method and obtain its transfer function and difference equation.

(b) Design a digital lowpass filter. Using bilinear transformation that is maximally flat in the passband to meet following specifications :

Passband	:	$0-4$ KHz
Stop band attenuation	:	>25dB at f >10KHz
Sampling frequency	:	32 KHz

Obtain the cascade form realization for the filter using second order sections.

(c) Compare the features of digital butterworth, chebyshev type I, chebyshev type II and elliptic filters in terms of :

(i) filter order (ii) transition width

4. Attempt *any two* parts of the following : **(10x2=20)**

(a) Show that the Bartlett estimate of the power spectral density is asymptotically unbiased and the spectrum estimates are consistent.

(b) A one stage decimator is characterized by the following :

Decimation factor = 3

anti aliasing filter coeff. are

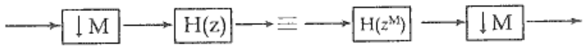
$h(0) = h(4) = -0.06$

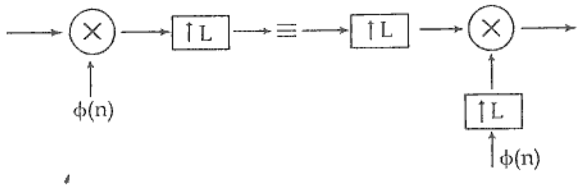
$h(1) = h(3) = 0.30$

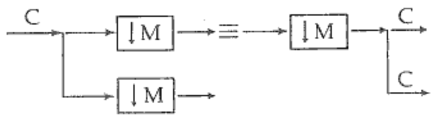
$h(2) = 0.62$

Given the input $\{x(n)\} = \{6, -2, -3, 8, 6, 4, -2\}$. Calculate and list the out put of the anti aliasing filter and output of the decimator.

(c) Prove that the following are pairs of equivalent systems :

(i) 

(ii) 

(iii) 

5. Attempt *any two* parts of the following : **(10x2=20)**

(a) What is channel vocoder ? Give its block diagram and explain its operation.

(b) (i) Define the following in speech signals

(1) Pitch period

(2) Unvoiced speech segment

(ii) Describe briefly an algorithm for estimating formant frequencies from sampled speech signal.

(c) (i) Give the block diagram of a radar system and explain its operation with associated signal processing. [7]

(ii) What is ambiguity function ? Find it for sinusoidal pulses. [3]

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