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B TECH
(SEM VI) THEORY EXAMINATION 2017-18
FUNDAMENTALS OF DIGITAL SIGNAL PROCESSING

Time: 3 Hours. Max. Marks: 100

Note: Be precise in your answer. In case of numerical problem assume data wherever not provided.

SECTION-A

1. Attempt all of the following questions: **(2×10=20)**

(a) What is difference between IIR and FIR filter?
(b) What is prewarping?
(c) What is the equation for order of Butterworth filter?
(d) What are the advantage and disadvantage of digital filter?
(e) Why windowing is necessary?
(f) What is difference between DFT & FT?
(g) Write Gibbs phenomena.
(h) What is difference between circular convolution and linear convolution?
(i) What is an energy signal?
(j) What is twiddle factor in DFT?

SECTION-B

2. Attempt any three of the following questions: **(3×10=30)**

(a) **(i)** Find out DFT of a sequence : $x(n) = \delta(n) + 2\delta(n-2) + \delta(n-3)$

(ii) Find linear convolution using circular convolution of the following sequence:

$x(n) = \{1, 2, 3\}$

$h(n) = \{1, 2\}$

(b) Determine the 8-point DFT of the following sequence using DIF FFT algorithm: x (n) = {2, 1. 2. 1}

(c) Proof the following:

(i) $X(N-k) = X^*(k) = X(-k)$

(ii) Why discrete time processing of continuous time signals, continuous time processing of discrete time signal is required? Explain with suitable example.

(d) Use bilinear transformation to convert low pass filter, $H(s) = 1/ s^2 + \sqrt{2s} + 1$ into a high pass filter with pass band edge at 100 Hz and F_s = 1 kHz.

(e) Design a low pass digital FIR filter having following specifications:

$$0.99 \leq H(e^{j\omega}) \leq 1.01, \qquad 0 \leq \omega \leq 0.19\pi$$

$$H(e^{j\omega}) \leq 0.01 \qquad 0.21\pi \leq \omega \leq \pi$$

Use Hanning window, Assume $\omega_c = 0.2\pi$, express the impulse response $h_d(n)$.

SECTION – C

3. Attempt any one of following questions: **(1×10=10)**

(a) What are the various sampling techniques used in DSP? How the reconstruction takes place? Explain them briefly.

(b) Design a single pole lowpass digital filter with a 3dB bandwidth of 0.2π, using the bilinear transformation applied to the analog filter,

$$H(s) = \frac{\Omega c}{s+\Omega c}$$, here Ωc is 3-dB bandwidth of the analog filter.

4. Attempt any one of following questions: **(1×10=10)**

(a) Write short notes on multi rate signal processing.

(b) If input to a linear shift –invariant system is

$X(n) = (1/2)^n u(n) - 2^n u(-n-1)$

The output is-

$Y(n) = 6(1/2)^n u(n) - 6(3/4)^n u(n)$

Find the system function and determine whether or not the system is stable and or causal. Also comment on the realizabilty of system.

5. Attempt any one of following questions: **(1×10=10)**

(a) Derive the signal to noise ratio of the Analog to Digital converters. Compare the truncation and rounding errors using fixed point & floating point representation.

(b) Explain the different properties of DFT. Compute the DFT of the sequence $x(n) = 2^n$, where N= 4 using DIF-FFT algorithm.

6. Attempt any one of following questions: **(1×10=10)**

(a) Determine the order and the pole of a Butterworth filter that has 3 dB bandwidth of 1000 Hz and an attenuation of 20 dB at 2000 Hz. Find the system function H(z) by bilinear transformation using T = 1/ 10000

(b) Design a normalized linear phase FIR filter having phase delay of τ = 4 and at least 40 dB attenuation in the stop band. Also obtain the magnitude/ frequency response of the filter.

7. Attempt any one of following questions: **(1×10=10)**

(a) Obtain the direct form I, direct form II, cascade and parallel form realization for the following system:

$y(n) = 0.75\, y(n-1) - 0.125\, y(n-2) + 6\, x(n) + 7\, x(n-1) + x(n-2)$

(b) Derive and draw the flow graph for DIT-FFT algorithm for N=8.