Dr Amit Kaul

5/5/2023

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National Institute of Technology, Hamirpur (HP)

Name of the Examination: B.Tech.

Branch	: Electrical Engineering	Semester	: 6th
Course Name	: Digital Signal Processing	Course Code	: EE-323
Time: 3 Hours		Maximum Marks: 50	

Note :

1. All Questions are compulsory
2. Draw the relevant diagrams/figures
3. Assume data wherever required

Q1. Evaluate the inverse Z- transforms of the following functions using Residue method

(i) $X(z)=\dfrac{1}{1-1.5z^{-1}+0.5z^{-2}},\ |z|>1$ (ii) $X(z)=\dfrac{1}{z^2+a^2}$ (5)

Q2. Determine the 4- point circular convolution of the two length- 4 sequences $g[n]$ and $h[n]$ given by

$$g[n]=[\underset{\uparrow}{1},\ 2,\ 0,\ 1] \qquad h[n]=[\underset{\uparrow}{2},\ 2,\ 1,\ 1]$$

Verify the result by computing circular convolution via DFT based approach. (5)

Q3. (a) Consider a 4-point sequence $x[n]=[\underset{\uparrow}{1},2,3,4]$. Use Goertzel algorithm to compute DFT coefficient $X(k)$ at frequency bin k=0.

(b)The decimation-in-time and decimation-in-frequency FFT algorithms evaluate the DFT of a complex-valued sequence. Show how a N-point FFT program may be used to evaluate N-point DFT of two *real-valued* sequences. (5)

Q4. (a)A length 10-sequence $x[n]$ has a real valued 10-point DFT $X[k]$. The first six samples of $x[n]$ are given by $x[0]$=2.5; $x[1]$= 7-j3; $x[2]$= -3.2+j1.3; $x[3]$ = -2+j5; $x[4]$ = 7+j ; and $x[5]$=5. Find the remaining four samples of $x[n]$.

(b)Assume that a complex multiply takes 1μs and that the amount of time to compute a DFT is determined by the amount of time it takes to perform all the multiplications. How much time does it take to compute a 4096-point DFT directly? How much time is required if FFT algorithm is used? (5)

Q5. (a) IIR digital filter can be designed, from an analog filter, by converting an analog filter $H(s)$ into a digital filter $H(z)$ using suitable transformation. However, the transformation should be such that stability is preserved in the Z-domain. One of the methods of this category is approximation of derivatives. Derive the expression for this method and show that the implication of this transformation is that the $j\ \Omega$-axis is mapped into a circle of radius 0.5, centered at z=0.5, in Z-plane.

(b) What are the conditions to be satisfied by the impulse response of FIR filter in order have a linear phase? Why IIR filters do not have a linear phase? (5)

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Q6. Design a linear-phase FIR low pass filter with following desired frequency response

$$H_d(\omega) = \begin{cases} e^{-j2\omega} & 0 \le |\omega| \le \frac{\pi}{4} \\ 0 & \frac{\pi}{4} < |\omega| \le \pi \end{cases}$$

Use a Hamming window. (5)

Q7. For the analog transfer function, $H(s) = \frac{2}{s^2 + 3s + 2}$, determine $H(z)$ using impulse invariant transformation if (a)T=1 second and (b)T=0.1 second. (5)

Q8. An FIR filter is given by the following difference equation.

$y(n) = 2x(n) + \frac{4}{5}x(n-1) + \frac{3}{2}x(n-2) + \frac{2}{3}x(n-3)$, Determine its lattice form. (5)

Q9. Write a short note on **any two** of the following

(a) DIF radix-4 FFT algorithm
(b) Overlap-Add Method
(c) Bilinear Transformation
(d) Transposed structures. (5+5)