DSP Questions

Compilation of previous year questions and internal papers

Module 1

---

1. Compare DFT and FFT Algorithms. How many complex additions and multiplications are needed to compute a 64 pt DFT when using DFT equations and FFT? Also comment on the computational advantage of the FFT algorithm over the direct method.
2. Define Discrete Fourier Transform (DFT) [*2025 sem*]
3. State and prove the time shifting property of DTFT, DFT
4. Find the output of an LTI system if the input $x(n) = \{2, 2, 1\}$ and the impulse response $h(n) = \{3, 2\}$ using DFT
5. Find the four point circular convolution of the sequences $x_1(n) = \{2, 1, 2, 1\}$ and $x_2(n) = \{1, 2, 3, 4\}$.
6. If $x(n) = \{1, 2, 3, 4\}$ find the DFT.
7. How will you obtain linear convolution from circular convolution? For $x(n) = \{1, 2, 3\}$ and $h(n) = \{-1, -2\}$, obtain linear convolution $x(n) * h(n)$ using circular convolution.
8. Find the DFT of the sequence $\{1, 2, 3, 4, 4, 3, 2, 1\}$ using DIT FFT algorithm.
9. Find the circular convolution of $x_1(n) = \{1, 2, 1, 3\}$ and $x_2(n) = \{1, -1, 1\}$.
10. State and prove any 3 properties of DFT.
11. Compute the 4 point DFT of $x[n] = \{1, 1, 0, 0\}$.
12. What is inplace computation?
13. Compare DIT and DIF FFT algorithms.
14. Find circular convolution of the sequences $x_1(n) = \{1, 2, 0, -1\}$ and $x_2(n) = \{2, 1\}$
15. An 8 point sequence is given by $x(n) = \{1, -1, -1, -1, 1, 1, 1, -1\}$, compute the 8 point DFT of $x(n)$ using radix-2 DIT-FFT.
16. Find the inverse DFT of $X(k) = \{7, -\sqrt{2} - j\sqrt{2}, -j, \sqrt{2} - j\sqrt{2}, 1, \sqrt{2} + j\sqrt{2}, j, -\sqrt{2} + j\sqrt{2}\}$ using FFT,
17. Using DIT FFT, compute the 8-point DFT of $x(n) = \{1, 0, 1, 0, 1, 0, 1, 0\}$. [*2025 sem*]
18. Find IDFT of sequence $X(k) = \{10, -2 + 2j, -2, -2 - 2j\}$ using DIT algorithm.
    $\rightarrow$ *This can be done by a 4 point IFFT, and you would get $x(n) = \{1, 2, 3, 4\}$*
19. Find the DTFT of the sequence $h(n) = \{4, 2, 3, 2, 4\}$ and plot the magnitude response.

20. The first 5 points of an 8 pt. DFT is given as $X(k) = \{20, -5.828 - j2.414, 0, -0.172 - j0.414, 0, \cdots\}$. Find the corresponding $x(n)$. Use DIT algorithm.

21. find DTFT of $x[n] = a^n u(n); |a| < 1$

22. Using the properties of DFT, compute the circular convolution of $x_1(n) = \{1, 2, 1, 2\}$ and $x_2(n) = \{1, 2, 3, 4\}$. [*2025 sem*]

23. Compute the 4-point DFT of $x(n) = \{1, 2, 2, 1\}$. [*2025 sem*]

24. ***Block convolution and DIF-FFT was not asked in any previous year questions. But do study it***

Module 2

1. Define phase delay and group delay of FIR filters.

2. Discuss the finite word length effects in FIR filters.

3. Discuss the effect of coefficient quantization in FIR filters.

4. What are Gibbs Oscillations (Gibb's phenomenon)? How can they be overcome them?

5. What are the disadvantages of Fourier series method? [*CUCEK 2025 internal*]

6. What are the desirable characteristics of window functions? [*CUCEK 2025 internal*]

7. Realize the system with impulse response $h(n) = \{4, 2, 3, 2, 4\}$ in Direct form and Linear Phase form. Calculate the phase delay and group delay.

8. Give the Hamming window function and plot its spectrum.

9. Compare the characteristics of Hamming and Blackman windows. [*2025 sem*]

10. Give the equations for the N point Hamming and Hanning, Rectangular, Bartlett window functions. Compare them in terms of main lobe width and side lobe level. [*2024 supplimentary exam*]

11. Explain the procedure for designing FIR filters using windows.

12. Explain the frequency sampling method used for the design of FIR filters. Discuss the principle of sampling the desired frequency response and how it determines the filter coefficients. [*2025 sem*]

13. Briefly explain the different types of windows used in FIR filter design.

14. Design an FIR low-pass filter of length $N = 7$ using a Hamming window with cutoff frequency, $\omega_c = 0.4\pi$. [*2025 sem*]

15. Design an FIR HPF filter using Bartlett window. The cut off frequency $\omega_c = 50\pi$. $\omega_s = 200\pi$. Assume $N = 9$

16. Realize an FIR filter with $h[n] = \{1, 0.5, 0.25, 0.5, 1\}$ using minimum number of multipliers.

17. Determine a direct form realization of the FIR filter with the following filter function using minimum number of multipliers. $h(n) = \{1, 2, 3, 4, 3, 2, 1\}$

18. Get the filter coefficients for the following FIR filter using Fourier series truncation method. Assume $N = 7$.

$$H_d(e^{j\omega}) = \begin{cases} 1 & , 0 \leq |\omega| < \frac{\pi}{2} \\ 0 & , \frac{\pi}{2} \leq \omega \leq \pi \end{cases}$$

19. Obtain the linear phase realization of

$$y(n) = x(n) + 2x(n-1) - 0.5x(n-2) + 3x(n-3) - 0.5x(n-4) + 2x(n-5) + x(n-6)$$

20. Obtain the cascade realization with minimum number of multipliers for the system function [*2025 sem*]

$$H(z) = \left(\frac{1}{2} + z^{-1} + \frac{1}{2}z^{-2}\right)\left(1 + \frac{1}{3}z +^{-1} + z^{-2}\right)$$

21. Design a filter with desired frequency response using a Hamming window for N=7.

$$H_d(e^{j\omega}) = \begin{cases} e^{-j3\omega} & , -\frac{\pi}{4} \leq \omega \leq \frac{\pi}{4} \\ 0 & , \frac{\pi}{4} < |\omega| \leq \pi \end{cases}$$

22. Design an ideal high pass filter with a frequency response

$$H_d(e^{j\omega}) = \begin{cases} 1 & , \text{for } \frac{\pi}{4} \leq |\omega| \leq \pi \\ 0 & , \text{for } 0 \leq |\omega| \leq \frac{\pi}{4} \end{cases}$$

using Hamming window. Find the values of $h(n)$ for $N = 11$. Find $H(z)$. Realize the filter using a suitable method.

23. Using frequency sampling method, designa bandpass filter with the following specifications:

    - sampling frequency = 8000Hz
    - cut-off frequencies = 1000Hz, 3000Hz

    Determine filter coefficient for $N = 7$ [*CUCEK 2025 internal*]

    Solution uploaded here

Module 3

---

1. Describe the characteristics of a Butterworth filter

2. All stable analog filters are transformed to stable digital filters using Impulse Invariance Technique - Prove. What are the drawbacks?

3. Write short notes on finite word length effects in IIR digital filters.
4. What is frequency warping? Discuss how it can be eliminated?
5. What is the fundamental principle of impulse invariant method? *[CUCEK 2025 internal]*
6. What are the properties of the impulse invariant transformation? Explain. *[2025 sem]*
7. Compare IIR and FIR filters
8. Distinguish between Butterworth and Chebyshev filters.
9. Explain limit cycle behaviors in signal processing, and their types.
10. Illustrate the bilinear transformation method of obtaining digital filter from analog filter.
11. Find the poles of an analog Chebyshev LP filter whose pass band ripple is 2 dB at 20 r/sec and the stop band attenuation is 15 dB at 40 r/s
12. Design a first order low pass digital Butterworth filter with a cut off frequency of 500 Hz. The sampling frequency is 10 KHz. Use bilinear transformation technique.
13. Design a digital Butterworth LP filter with the following specifications, use Impulse invariance transformation.

$$0.7071 \leq |H(e^{j\omega})| \leq 1, \qquad 0 \leq |\omega| \leq 0.2\pi$$
$$|H(e^{j\omega})| \leq 0.17783, \qquad 0.4\pi \leq |\omega| \leq \pi$$

[Analog filter design + transformation to digital (8+2 marks)]

14. Design a digital low-pass Butterworth filter using the Bilinear Transformation (BLT) method to satisfy the following specifications: *[2025 sem]*

$$0.9 \leq |H(e^{j\omega})| \leq 1, \qquad 0 \leq |\omega| \leq 0.4\pi$$
$$|H(e^{j\omega})| \leq 0.15, \qquad 0.6\pi \leq |\omega| \leq \pi$$

Take $T = 1$s. Realize the filter.

15. Obtain the canonical form, Cascade and parallel realizations of the IIR filter:

$$y(n) = -0.1y(n-1) + 0.72y(n-2) + 0.7x(n) - 0.252x(n-2)$$

16. Obtain the direct form I, direct form II and cascade realization for the system having the difference equation:

$$y(n) + 0.1y(n-1) - 0.2y(n-2) = 3x(n) + 3.6x(n-1) + 0.6x(n-2)$$

17. Draw the Direct Form I, Direct Form II, Parallel, and Cascade realizations of the digital filter with the following difference equation *[2025 sem]*

$$y[n] - \frac{3}{4}y[n-1] + \frac{1}{8}y[n-2] = x[n] + \frac{1}{2}x[n-1] + \frac{1}{4}x[n-2]$$

18. Realize the given IIR filter in Direct Form I, Direct Form II, cascade and parallel forms:

$$H(z) = \frac{1 - z^{-1}}{\left(1 - \frac{1}{2}z^{-1}\right)\left(1 - \frac{1}{4}z^{-1}\right)}$$

Module 4

1. What is pipelining? What are the different stages in pipelining?
2. Discuss the special instructions used in DSP processors. [*2023 sem exam*]
3. Describe Harvard architecture.
4. Explain the operation of a MAC unit in a DSP processor.
5. Describe how MAC operations are performed in a single instruction cycle in TMS320C54X. Explain its architecture with diagram. [*2025 sem*]
6. List the advantages of floating point processors.
7. Discuss how, DSP processors are more advantageous than microcontrollers?
8. Differentiate between fixed and floating point processors.
9. Discuss the different addressing modes of a TMS320C54X processor with examples.
10. Draw and explain the architecture of the TMS320C54x processor [*2022 special Supplementary exam*]
11. Draw and explain architecture of the TMS320C67X [*2022 special Supplementary exam*]
12. Explain the different addressing modes in TMS320C67X processors.