80 Questions of Digital Signal Processing (DSP)

Q1. Plot the following functions:

a) x(n) = -2 u(-n-4) , b) x(n) = u(n+3) – u(n-5)

Q2. What is the purpose of a DSP processor?

Q3. What are the most related areas in DSP?

Q4. If y(n) = n x(n).

a) Is the system linear?
b) Is the system causal?
c) Is the system stable?

Q5. What are DSP applications?

Q6. Is it possible to digitize an infinite number of points in analog signal? Explain.

Q7. Define sampling rate.

Q8. Plot the block diagram of a DSP system and explain briefly about it.

Q9. What is the purpose of digital filtering block in a DSP system?

Q10. Define aliasing. When it occurs?

Q11. Define Shannon sampling theorem.

Q12. What are the cases that are used for recovery of the original signal spectrum X(f).

Q13. If x(n) = [1 2 3] and h(n) = [2 1 -1], find y(n).

Q14. What are the methods used to control aliasing noise?

Q15. Assuming that a DS processor with a sampling time interval of 0.01 second. Converts analog signals x(t) to the digital signal x (n), and determine its digital sequence

$$x(t) = 5\sin(20\pi t)u(t)$$

Q16. For the following system:

$$y(n) = 4x^3(n-1) - 2x(n)$$

a) Is the System linear?

b) Is the system causal?

c) Is the System linear time invariant?

Q17. Find the unit-impulse response of the following system and draw its block diagram $y(n) = 5x(n-10)$

Q18. Determine the stability of the following linear system:

$$y(n) = \sum_{k=0}^{\infty} 0.75^k x(n-k)$$

Q19. Find y(n) if

$$x(k) = \begin{cases} -2, & k = 0,1,2 \\ 1, & k = 3,4 \\ 0 & elsewhere \end{cases} \quad \text{and} \quad h(k) = \begin{cases} 2, & k = 0 \\ -1, & k = 1,2 \\ 0 & elsewhere, \end{cases}$$

a) using the graphical method;
b) using the table method;
c) using matrix by vector method

Q20. If h(n) = 0.8 h(n-1) + δ(n). Find the system frequency response. Plot the magnitude and phase.
Q21. Define analog frequency and digital frequency.
Q22. Prove that the output to a sinusoid is another sinusoid of the same frequency but with different phase and magnitude.
Q23. A discrete time system has a unit sample response h(n)

$$h(n) = \delta(n) + 2\,\delta(n-1) + \delta(n-2)$$

a) Find the system frequency response. Plot the magnitude and phase.
b) Find the steady-state response of the system to x(n) = 8 cos (π n /4).
c) Find the steady-state response of the system to x(n) = 8 cos (3 π n /4).

Q24. What are Properties of frequency response?
Q25. Use the residue theorem to find x(n), if X(Z) = 1/ [Z (Z – 1) (2Z – 1)]
Q26. What is the difference between Z-Transform and Fourier Transform?

Q27. Find the inverse z-transform for the following function using:

$$X(z) = 2 + \frac{4z}{z-1} - \frac{z}{z-0.5}$$

1. Using properties.
2. Partial fraction expansion method.

Q28. If x(n) = [1 2 -2 -1 0 3 -4 -3] for x(n) for 0 ≤n ≤ 8 . Find Evaluate its DFT X(k). Check your result using MATLAB function
Q29. Define frequency resolution.

Q30. Given a sequence x(n) for 0 ≤ n ≤ 3, where x(0) = 1, x(1) = 1, x(2) = -1, and x(3) = 0. Evaluate its DFT X(k). Check your result using MATLAB function

Q31. Consider a digital sequence sampled at the rate of 20,000 Hz. If we use the 8,000-point DFT to compute the spectrum, determine

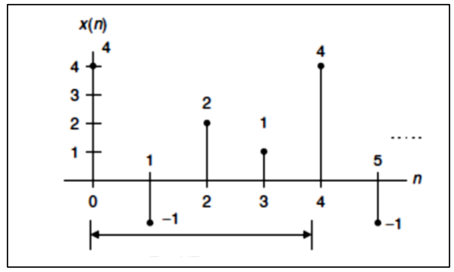
a. the frequency resolution

b. the folding frequency in the spectrum

Q32. Given a DFT X(K) for 0≤ k ≤ 3, where X(0) = 3, X(1) = -1+, X(2) = -1, and X(3) = -1-j1. Evaluate its x(n). Check your result using MATLAB function.

Q33. For the figure shown:

Assuming that fs = 100 Hz, compute and plot the amplitude spectrum, phase spectrum, and power spectrum. Repeat for one sided spectrums too.



Q34. Consider the sequence x(n) = [-1 2 -1 1] for 0≤ n ≤ 3. Assuming that fs = 50 Hz, Compute the amplitude spectrum, phase spectrum, and power spectrum.

Q35. For the sequence [2 -1 -1 2 -3 -4 0 2]. Find: A) Reduced DIF FFT, B)Reduced DIT FFT.

Q36. Considering the sequence x(0) = 1, x(1) = 0, x(2) = -1, and x(3) = 3, and given f_s = 50 Hz. compute the amplitude spectrum, phase spectrum, and power spectrum

a. Using the rectangular window function.

b. Using the Hanning window function.

Q37. Define spectral leakage.

Q38. Derive the algorithm of reduced DIF FFT.

Q39. For the sequence [1 -1 -1 -1 1 1 1 -1]. Find: A) Reduced DIF FFT , B)Reduced DIT FFT.

Q40. Explain the relation between DFT and Fourier transform

Q41. Considering the sequence x(0) = 0, x(1) = 3, x(2) = -3, and x(3) = 5, and given f_s = 75 Hz. compute the amplitude spectrum, phase spectrum, and power spectrum

a. Using the triangular window function.

b. Using the Hamming window function.

Q42. Find x(n) for $X_R(K)$ = [2 0 5 0 1 0 5 0] and $X_I(K)$ =[0 2 -3 0 0 0 3 -2] , for 0≤ k ≤ 7 then find xa(t) if T = 0.01 sec.

Q43. For the sequence [1 2 3 4 -4 -3 -2 -1]. Find: A) Reduced DIF FFT, B)Reduced DIT FFT. C) DFT. Check your result using MATLAB function.

Q44. Given the following filter:

$$y(n) = 0.2\,x(n) + 0.4\,x(n-1) + 0.5\,y($$

A) what is the type of this filter? why?

B) Determine the transfer function, nonzero coefficients, and impulse response.

Q45. If H(S) = 1/ [S+1) (S+2)], use the numerical solutions of differential equations to obtain H(Z) for, a) T = 1 sec., and b) fs = 100 Hz.

Q46. A Butterworth LPF is required with cutoff frequency 0.2 π. Its response should be at least 30 dB down at Ω = 0.4 π. Calculate the order of the filter.

Q47. A transfer function of a first order LPF is defined by: H(S)= 1/ (1+sτ) . Where τ is called time constant.

a)Sketch the frequency response magnitude |H(W) | and impulse response h(n) assuming τ = 1 sec.

b) Find the transfer function and difference equation of the impulse invariant digital filter having a sampling interval of 0.05 sec.

c) Repeat (b) if the sampling interval is changed to 0.5 sec

Q48. Design and realize a digital high-pass filter using bilinear transformation method to satisfy the following characteristics (c/cs):

1. − 3.01 dB cutoff frequency of 0.75 π rad

2. Magnitude down at least 15 dB at 0.5 π rad.

Q49. Use Digital-to digital transformation method. Find H(Z) for HP digital filter that satisfies the following requirements:

1- A − 3.0102 dB cutoff digital frequency of 0.75 π rad.

2- Attenuation of 0.5 π rad is at least 15 dB

Q50. Design a HP digital filter to be used in A/D- H(Z) – D/A structure that will have a − 3 dB cutoff of 45 π rad / sec. and an attenuation of 50 dB at 30 π rad/sec. The filter is required to have linear phase. The system will use a sampling rate of 100 samples/sec.

Q51. Explain the usefulness of Z-Transform

Q52. Find the Z.T of :

1. A δ(n-m)
2. x(n) = 3n. u(n-2)

Q53. Find the frequency response for the following difference equation:

y(n) = - (1/2) y(n-1) + x(n)

Q54. Is the system $y(n) = x(n^2)$ linear or not?

Q55. Check the stability of the system $y(n)= x(n) + e^a y(n-1)$.

Q56. For a DSP system if the sampling rate of 4,000 Hz is used and the anti-aliasing filter is a third-order Butterworth lowpass filter with a cutoff frequency of 2.5 kHz.

a. Determine the percentage of aliasing level at the cutoff frequency.

b. Determine the percentage of aliasing level at the frequency of 500 Hz.

Q57. What is the difference between analog signal and sampled signal? Is the sampled signal considered as digital signal? Explain

Q58. Explain the time invariant of a linear DSP system.

Q59. What is the difference between causal and non-causal DSP system?

Q60. Sketch the unit response and impulse response of a LTI system. Determine the difference between the two.

Q61. Given a LTI system: y(n) = x(n) + 0.5 x(n-1)

a. Determine the unit-impulse response h(n).
b. Draw the system block diagram.
c. Write the output using the obtained impulse response

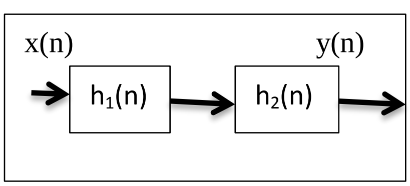
Q62. Find y(n) using table lookup method if x(n) = [5 3 0 1] and h(n) = [-2 1 2 0 4]
Q63. Repeat Q62 Using matrix by vector method.
Q64. If x(n) = [1 -1 1 -1], and h(n) = [2 0 2]. Find y(n) such that linear and circular convolution are the same.
Q65. Use graphical method to find circular convolution between $x_1(n)$= [3 2 1 4] and $x_2(n)$= [0 1 2 3].
Q66. Use graphical method to find x(n) if y(n) = [10 -3 7 12] and h(n) = [8 6].
Q67. Find and plot the frequency response if : $h(n) = \begin{bmatrix} 2 & 0 \le n \le N-1 \\ 0 & elesewhere \end{bmatrix}$

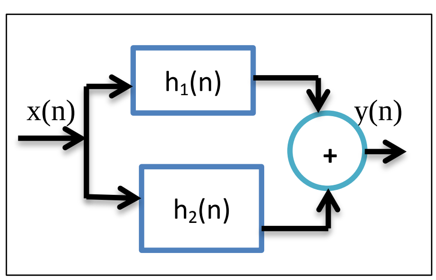
Q68. What is the response of a LTI system if the inputs are complex exponential and sinusoidal signals?.
Q69. What is the difference between Fourier series and Fourier transform?.
Q70. Find Z { n 3^n sin(wn) u(n) }
Q71. For the following difference equation**:** y(n) – (3/4) y(n-1) +(1/8) y(n-2) = x(n)
a) Find H(Z), b) h(n) , c) the step response of the system.
Q72. Solve following difference equation using the residue theorem:
y(n) – 4 y(n-1) + 4 y(n-2) = x(n) –x(n-1)
Q73. If $H(Z) = \frac{1}{\left(1+\frac{1}{3}z^{-1}\right)\left(1-\frac{1}{6}z^{-1}\right)}$

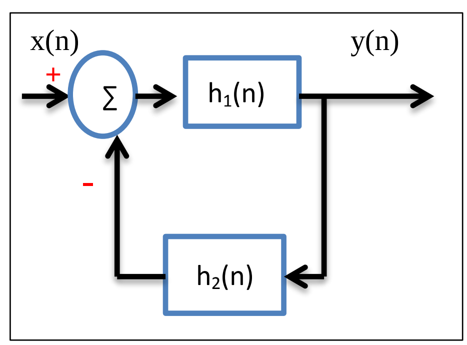
Perform the filter realizations and write the difference equations using the following realizations:

1. Direct form I and direct form II
2. Cascade form via the first-order sections
3. Parallel form via the first-order sections

Q74. Find h(n) and H(Z) for the systems shown below







Q75. What is the difference between DFT and FFT?.
Q76. What is folding frequency? Is it different from Nyquist frequency?.
Q77. The periodic signal x(t) = cos {(3 /4) π t} is sampled using the rate f_s = 4 Hz.

a. Compute the spectrum c_k using the samples in one period.
b. Plot the two-sided amplitude spectrum $|c_k|$ over the range from -2 to 2 Hz.

Q78. What is the twiddle factor?

Q79. Use Digital-to digital transformation method. Find H(Z) for LP digital filter that satisfies the following requirements:

1- A − 3.0102 dB cutoff digital frequency of 0.25 π rad.

2- Attenuation at and past 0.6 π rad is at least 18 dB

Q80. Explain about digital audio equalizer.