1. Question

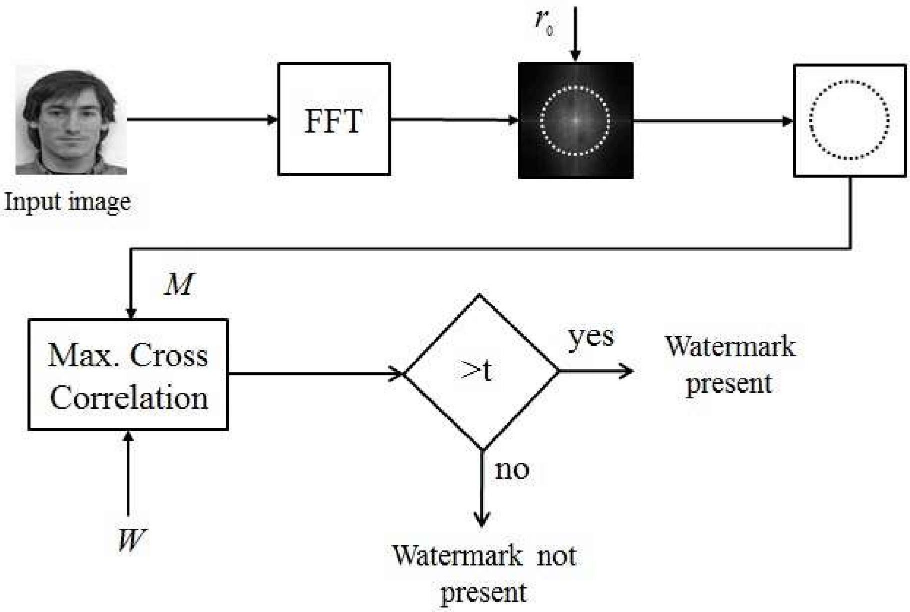


Figure: FFT Applied to Image Water Marking Application

The image filtering in the scenario is performed using a 4 x 1 filter. Using the filter matrix obtain the filtered intensity value of the given image intensity value.

Image intensity value, Q[n] = { 4, 3 2 , 2}.

Answer

Filter Matrix:

$$\begin{bmatrix} 1 & 1 & 1 & 1 \\ 1 & -i & -1 & i \\ 1 & -1 & 1 & -1 \\ 1 & i & -1 & -i \end{bmatrix}$$

(2 Marks)

Filtered Intensity Value

$Q_{filtered}[k]$ = {11, 2 - j , 1, 2 + j}

(2 Marks)

2. Question

A signal before and after processing is shown in the figure.1) Which of the property of DFT is represented in the figure given below ? 2) Write down the significance of the property.

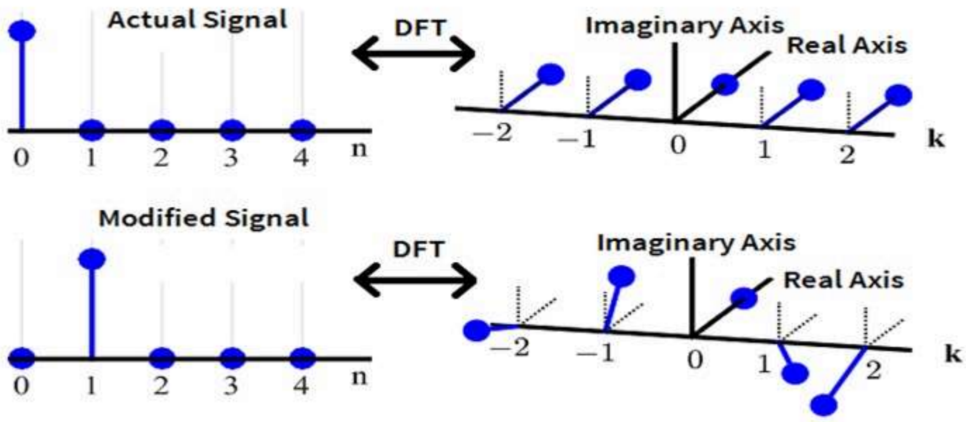


Figure: Signal and it's DFT

Answer

1) Circular Time Shift
(1 Mark)
2) Circular Time Shift property states that shifting a discrete-time signal circularly by k samples results in the corresponding frequency domain representation being multiplied by $e^{-j\frac{2\pi}{N}kn}$, where N is the signal length.
(1 Mark)

3. Question

In a DFT, what does a high magnitude at a specific frequency bin indicate?

a) The presence of that frequency component in the signal
b) The absence of that frequency component in the signal
c) The signal is noisy
d) The signal is periodic

Answer

a) The presence of that frequency component in the signal

4. Question

The Radix-2 FFT algorithm requires the sequence length to be:
a) Prime
b) Even
c) A power of 2
d) A power of 3

Answer

c) A power of 2
(1 Mark)

5. Question

In the context of FFT, what does the term "butterfly" refer to?
a) A mathematical function
b) A computational flow graph pattern
c) A method of signal sampling
d) A type of window function

Answer

b) A computational flow graph pattern
(1 Mark)

6. Question

The figure give below shows the application of Digital Signal Processing in speech enhancement.Here x[n] is the voice message in discrete time representation. The noisy signal (y[n] = x[n]+v[n]) is analysed in frequency domain with the help of DFT and noise is removed where v[n] represents the additive noise in the system.DFT of y[n] is represented as y[k]. After noise removal IDFT is perfomed to bring back the voice message y[n].

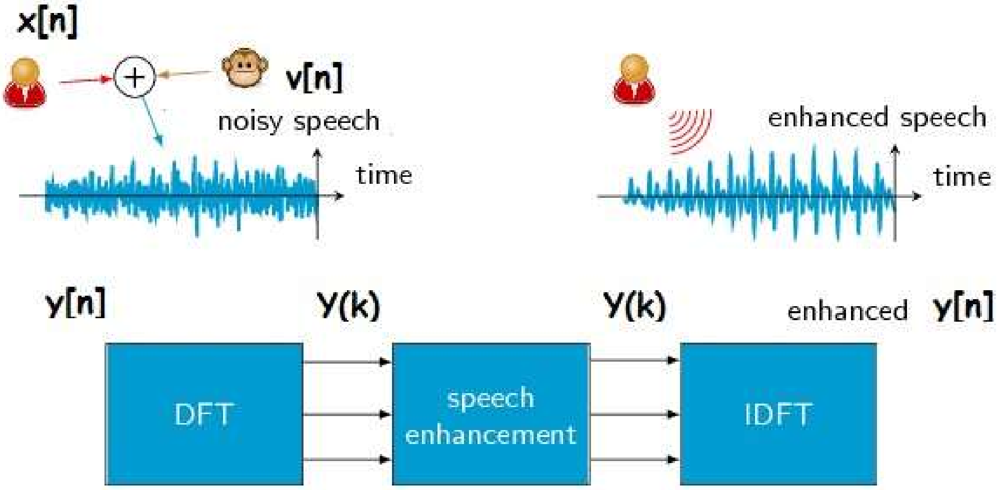


Figure: Speech Signal Processing

The following FFT method is used to get the DFT of x[n] speech signal.

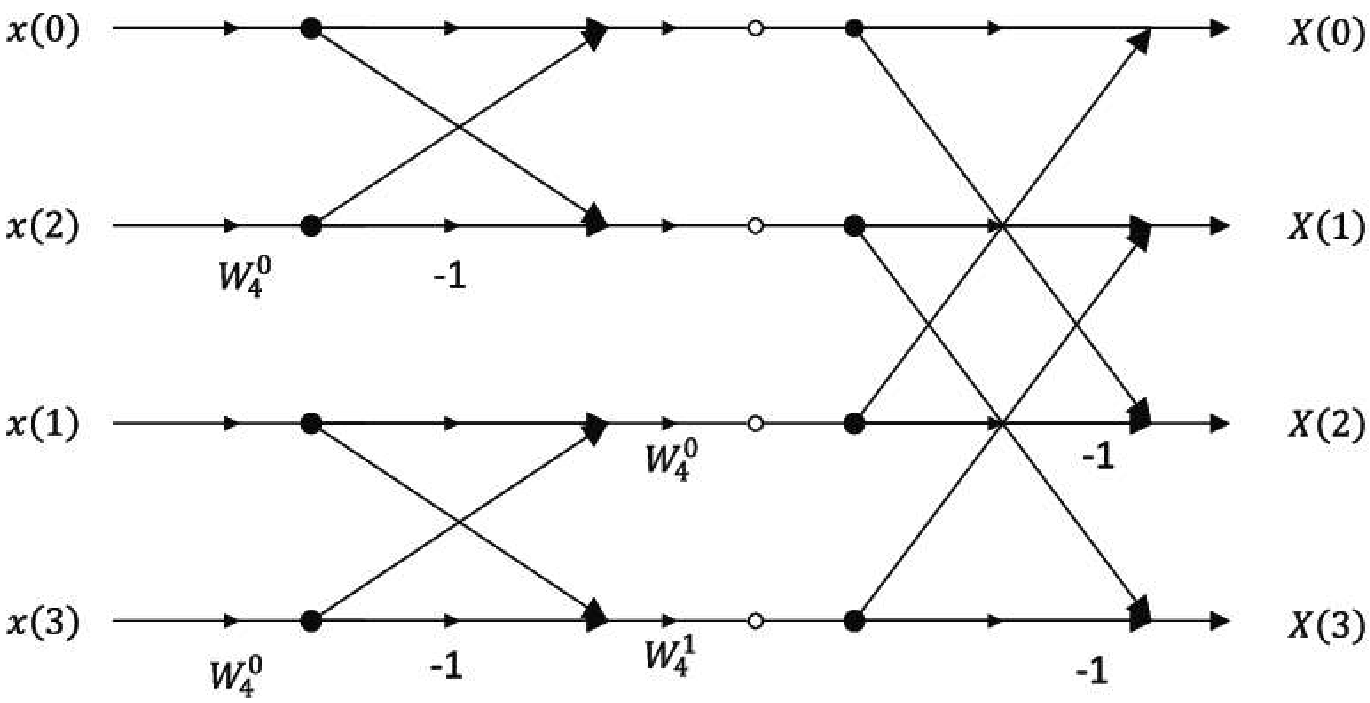


Figure: FFT Method Signal Flow Graph

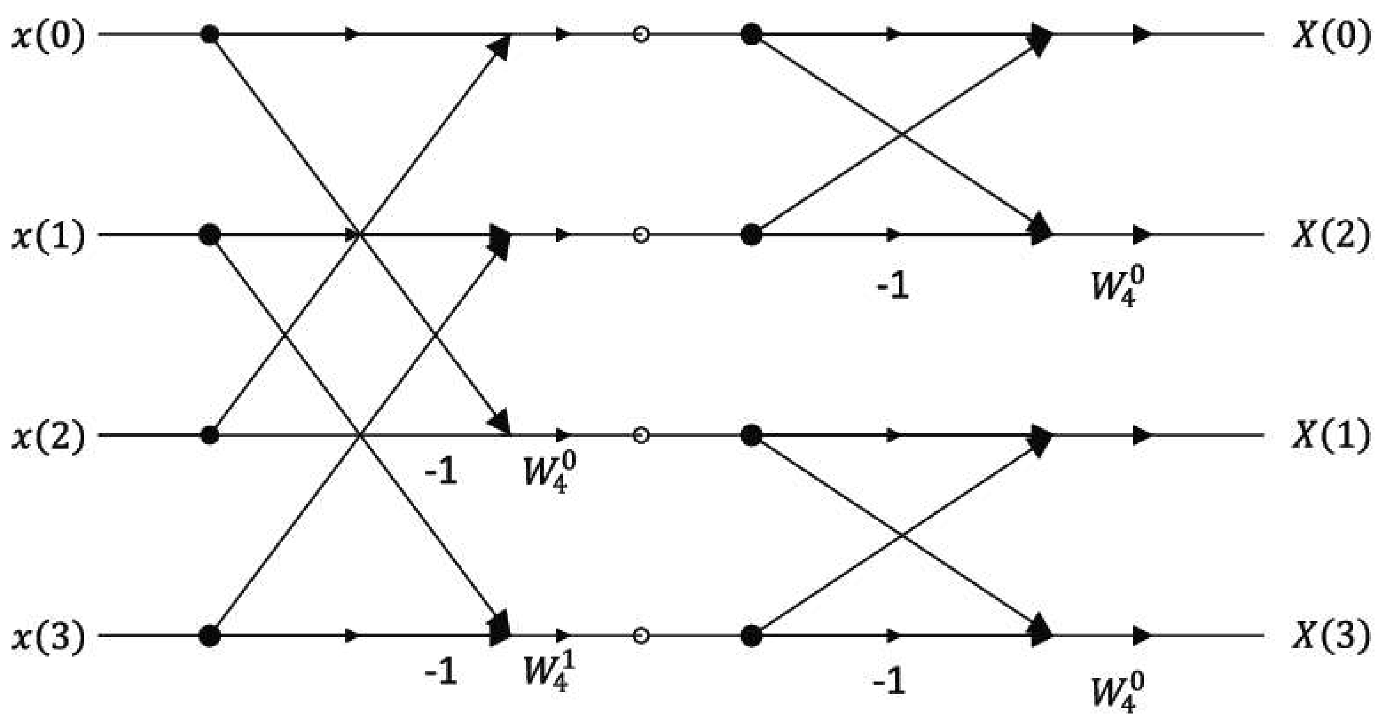
From the signal flow graph, analyse the number of additions and multiplications involved in FFT computation, compare with conventional DFT approach.

Visualise the FFT of the given noisy signal y[n] = {4,8,4,8} using the given FFT method.

Modify the FFT method in such a way that the input can be given in the correct sequence order.

Answer

The calculation of a four-point DFT from an FFT signal flow graph involves 4 additions and 8 multiplications. The conventional DFT equation approach requires 16 additions and 16 multiplications.
(1 Mark)
FFT of the given noisy signal y[k] = {24,0,-8,0}
(2 Marks)
Modified FFT method so that input can be given in correct sequence order.



(2 Marks)

7. Question

The output of an FIR filter is a:
a) Linear combination of past inputs
b) Linear combination of past outputs
c) Combination of past inputs and outputs
d) Non-linear function of inputs

Answer

a) Linear combination of past inputs

8. Question

A spectrum/ signal analyzer will use the DFT/FFT and show the individual sinewaves as peaks in its display. The simple periodic function is shown in Figure.

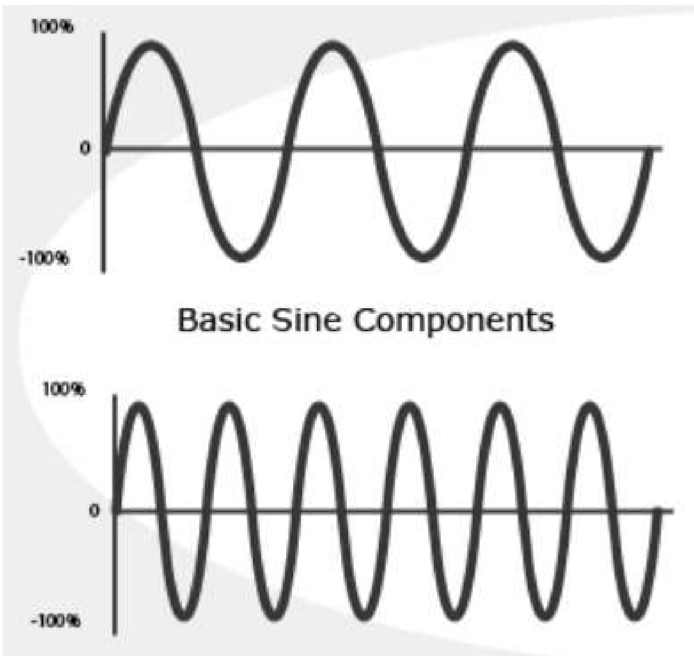


Figure: Simple periodic function

DFT is computed using direct method using the following formula:

$$X(k) = \sum_{n=0}^{N-1} x(n) e^{-j\frac{2\pi nk}{N}} \qquad 0 \le k \le N-1, \text{ Where, } W_N^{nk} = e^{-j\frac{2\pi nk}{N}};$$

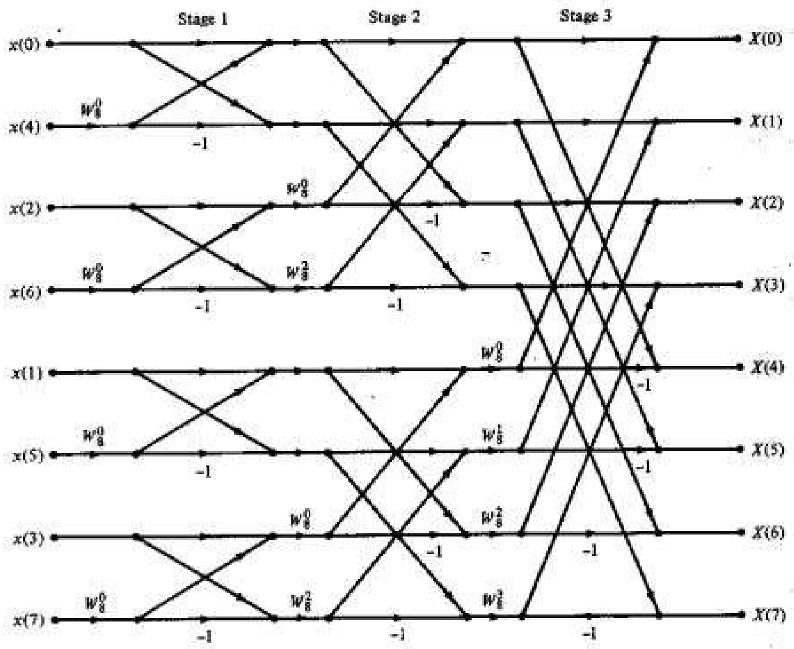


Figure: DIT - FFT butterfly structure for 8 point DFT.

1) If the first five samples of the 8 point DFT spectrum is X(k)={8,4+j2,-3,2-j,6,…}, then obtain the remaining samples of the spectrum without computing DFT.

2) Out of direct computation and DIT-FFT method, which method is easier and why?

Answer

1) X(5) = X*(3) = 2 + j

(1 Mark)

X(6) = X*(2) = -3

(1 Mark)

X(7) = X*(1) = 4 - 2j

(1 Mark)

2) Out of direct computation and DIT-FFT method, DIT-FFT method is easier.

(1 Mark)

9. Question

The inverse FFT (IFFT) is used primarily for:

a) Converting a time-domain signal to the frequency domain

b) Performing convolution in the frequency domain

c) Reconstructing a signal from its frequency components

d) Reducing computational complexity of DFT

Answer

a) Converting a time-domain signal to the frequency domain

(1 Mark)

10. Question

How does the Radix-2 FFT algorithm contribute to reducing noise in signals?

a) Filters out the noise

b) Enhances the noise

c) No effect on noise

d) Amplifies the noise

Answer

a) Filters out the noise

11. Question

The image filering operation uses linear convolution as one of the methods.

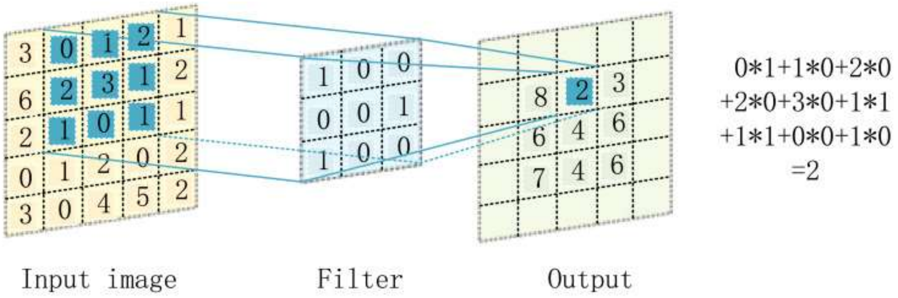


Figure: Image Filtering Operation

Where Linear convolution used for filtering is given by:

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

Using linear Convolution filter the image segment {x(n)} ={1, 2, 3, 1} of Length 'L' with the filter {h(n)} = {4, 3, 2, 1} of Length 'M'. Infer the length of the linear convolution process.

Answer

Filtered signal y(n) ={4, 11, 20, 18, 11, 5, 1}
(2 Marks)
The length of the linear convolution process = L + M - 1 = 4+4-1 = 7
(1 Mark)

12. Question

The Fast Fourier transform (FFT) computes the Discrete Fourier transform (DFT) of the given sequence.The following method is used to improve the computation speed in DSP processors.

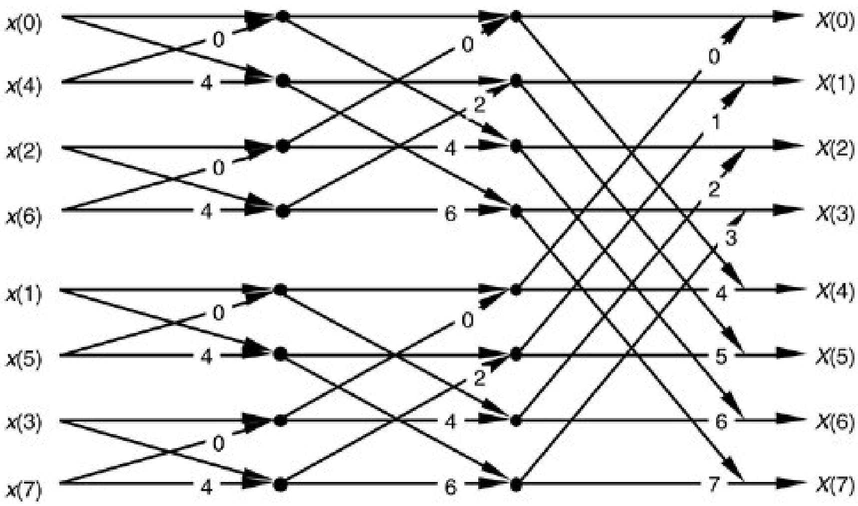


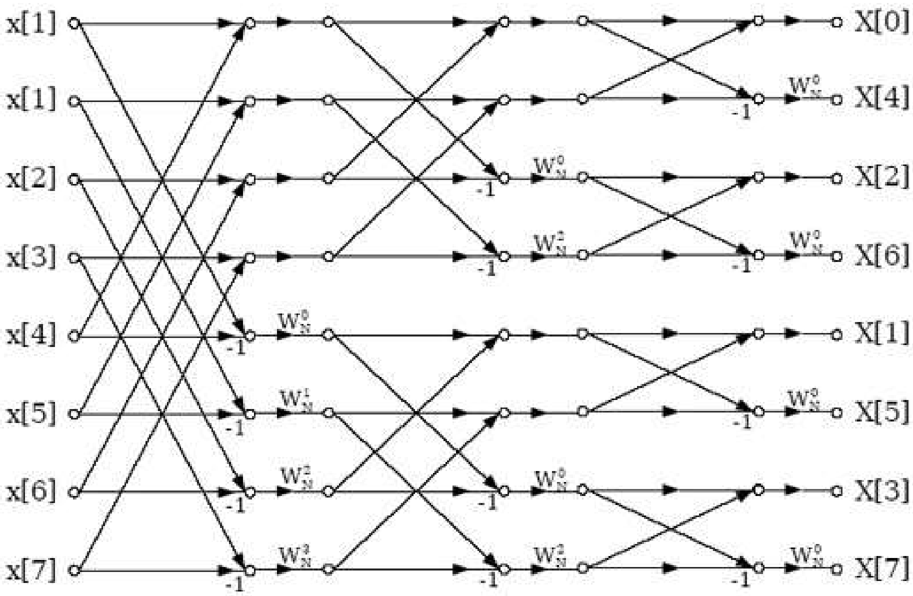
Figure: Fast Fourier Transform

Identify the type of FFT which speeds up the processing in DSP processors.
In which order this FFT takes the input? Write the significance of the order.
Modify the FFT diagram in such a way that the input time sequence can be given in order to the FFT computation unit.

Answer

The FFT type is correctly identified as DIT-FFT.
(1 Mark)
The order in which the FFT takes input is the bit-reversed order.
(1 Mark)
We give inputs in bit-reversed order for computing the FFT using the DIT-FFT algorithm to <u>eliminate the need for additional data reordering</u> during the computation and <u>reduce computation complexity.</u>

(1 Mark)
Modified FFT diagram in such a way that the input time sequence can be given in order.



(3 Marks)

13. Question

The response given below is the spectrum of time domain signal. 1) i) Identify the actual time domain signal from the given options. 1) ii) Also, when the sharp discontinuities in the signals increases, what happens to the width of the spectrum?

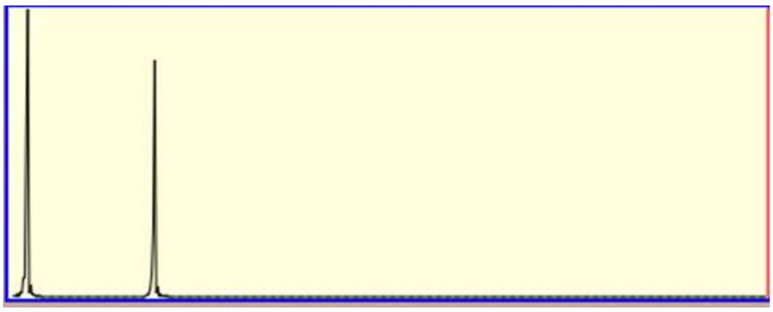
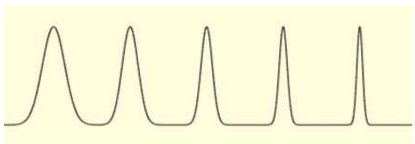
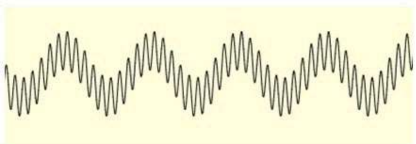


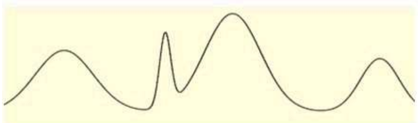
Figure: Frequency Response of time domain signal



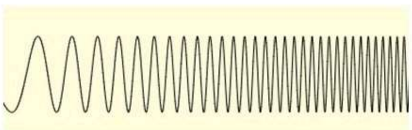
Option 1



Option 2



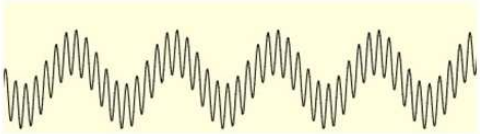
Option 3



Option 4

Answer

1) i)



Option 2

(1 Mark)
1) ii) When the sharp discontinuities in the Signals increases, the Width of the Spectrum will increase
(1 Mark)

14. Question

Which filter is preferred for applications requiring a maximally flat passband?
a) Chebyshev Filter
b) Elliptic Filter
c) Butterworth Filter
d) FIR Filter

Answer

c) Butterworth Filter

15. Question

Which of the following is true for a Butterworth filter compared to a Chebyshev filter?
a) Butterworth has a faster roll-off.
b) Chebyshev has a linear phase response.
c) Butterworth has a flatter passband.
d) Chebyshev is always of lower order.

Answer

c) Butterworth has a flatter passband.

16. Question

A low pass filter has to be designed using impulse invariance method for audio filtering application in such a way that the pass band cutoff frequency is ω_p = 0.15 pi and stop band cutoff frequency is ω_s = 0.35 pi. The relation between frequency in the continuous-time and discrete-time domains is ω = Ω / pi.

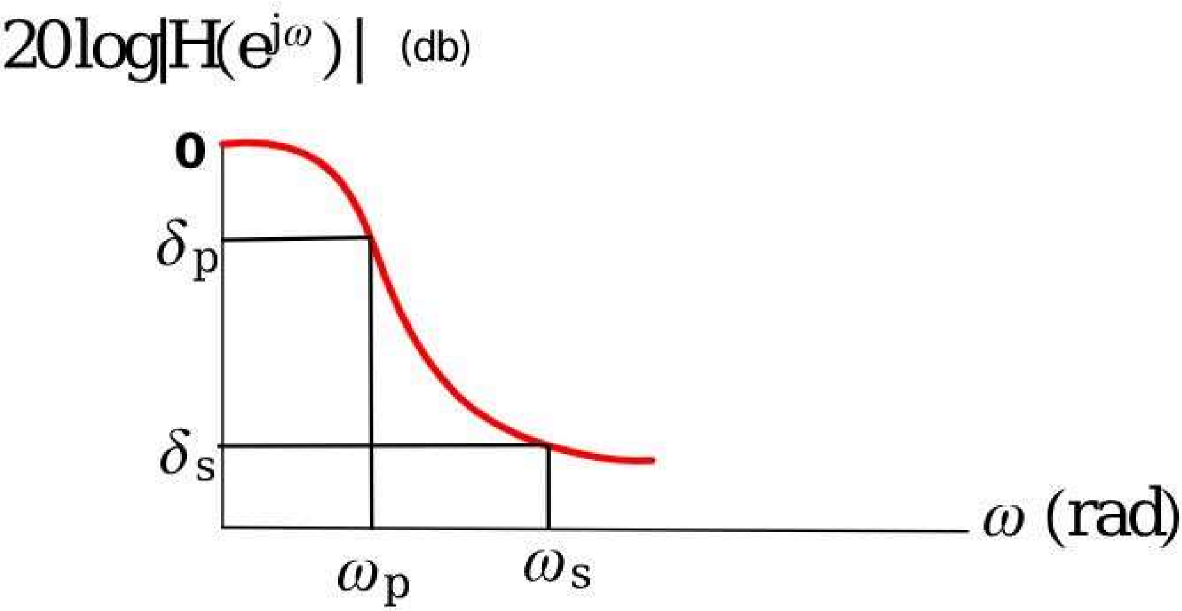


Figure: Low Pass Filter

For the above given filter estimate the pass band frequency ω_p and stop band frequency ω_S.
If $?_p$ (also termed as $1-?_1$) and $?_s$ (also termed as $?_2$) is 0.7079 and 0.1 respectively, analyze the scenario and predict the order of the filter.

$$\text{Order of the filter is given by: } N = \frac{1}{2}\frac{\log\left(\frac{k2}{k1}\right)}{\log\left(\frac{\Omega s}{\Omega p}\right)}$$

Parameter $k_1 = 1 / (1 - \delta_1)^2$ and $k_2 = ((1/ (\delta_2^2)) -1)$

Answer

The estimated pass band frequency Ω_p= 0.15
(1 Mark)
Estimated stop band frequency Ω_s = 0.35
(1 Mark)
Order of the filter:

$$N \approx \frac{0.8499}{0.3686} \approx 2.31$$

(2 Marks)
Filter order must be an integer, we round up to the next whole number. N=3
(1 Mark)

17. Question

An Analog Butterworth filter is used to separate audio signals while applying to a loud speaker is observed to have a -2dB pass band attenuation at a frequency Ωp of 20 rad/sec and atleast -10 db stop band attenuation at at a frequency Ωs = 30 rad / sec.The pass band and stop band attenuation freqiuencies are pre-warp analog frequencies.

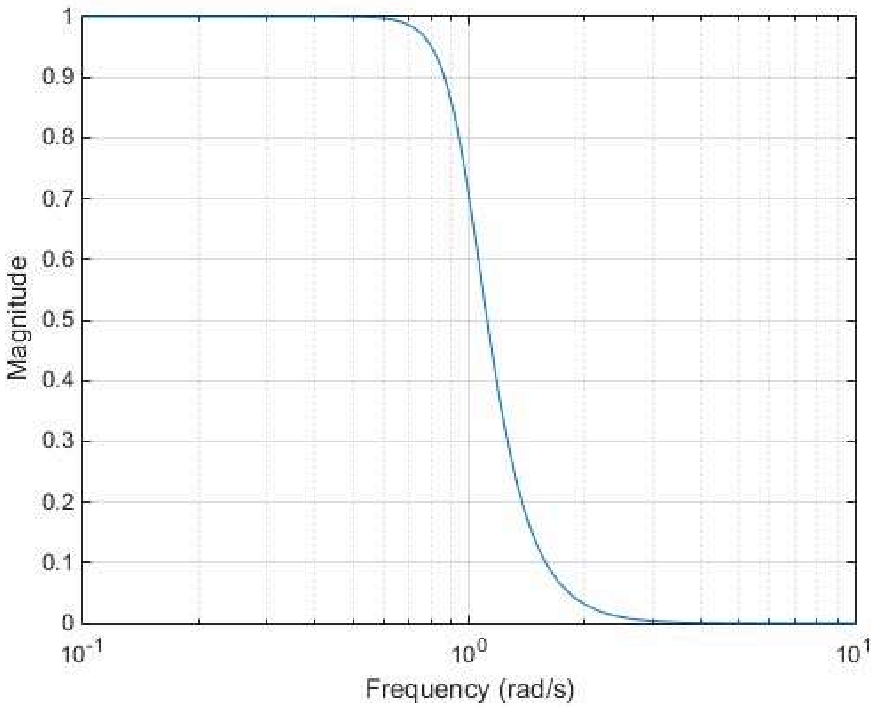


Figure : Butterworth Filter Magnitude Response

Order of the filter is given by:

$$N = \frac{\log\sqrt{\left(\frac{10^{\alpha_s/10}-1}{10^{\alpha_p/10}-1}\right)}}{\log\left(\frac{\Omega_s}{\Omega_p}\right)}$$

For the above given filter, obtain the gain at the pass band edge.
How the gain at the stop band edge would differ from gain at pass band edge?
Also predict the order of the filter.

Answer

Convert -2dB to a gain ratio:
20*log10(G) = -2
log10(G) = -0.1
G = 10^(-0.1)
G ≈ 0.7943
gain at the passband edge (Ωp) = 0.7943.
(2 Marks)
Convert -10dB to a gain ratio:
20*log10(G) = -10
log10(G) = -0.5
G = 10^(-0.5)
G ≈ 0.3162
gain at the passband edge (Ωs) = 0.3162
(2 Marks)
N >= 2.45
The filter order is predicted as 3.
(2 Marks)

---

18. Question

Which of the following is a characteristic of an analog Chebyshev filter?
a) Maximally flat in the passband
b) Exhibits equiripple behavior in the passband
c) No ripple in the passband or stopband
d) Linear phase response

Answer

b) Exhibits equiripple behavior in the passband

---

19. Question

The Chebyshev filter is known for its:
a) Flat passband
b) Ripple in the stopband
c) Ripple in the passband
d) Linear phase response

Answer

c) Ripple in the passband

---

20. Question

The linear phase characteristic in FIR filters is achieved by:
a) Using complex coefficients
b) Symmetric impulse response
c) Asymmetric impulse response
d) Using only zeros

Answer

b) Symmetric impulse response

21. Question

Which of the following filters is typically used when a sharp transition between passband and stopband is required?

a) FIR filter

b) IIR filter

c) Both FIR and IIR filters

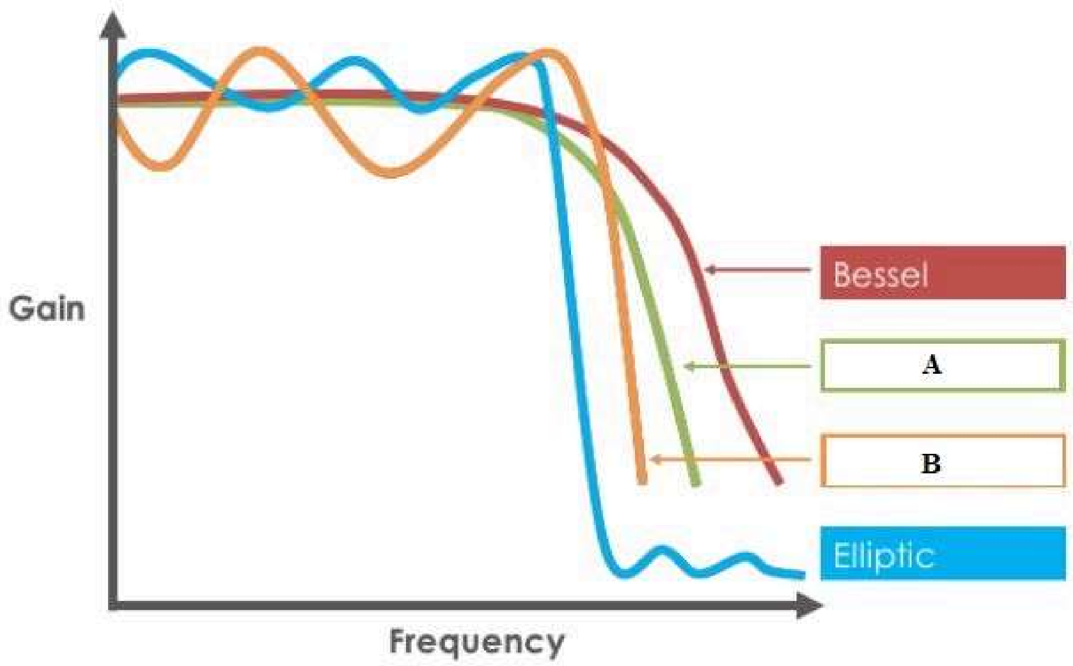
d) Neither FIR nor IIR filters

Answer

b) IIR filter

22. Question

From the Gain Vs. Frequency reponse of various filters identify which filter is Butterworth and which filter is Chebyshev. Among the below given four filters , find the filter with the sharpest roll-off.



Answer

Filter A is Butterworth and Filter B is Chebyshev

(1 Mark)

The filter that has the sharpest roll-off of in this group is elliptic filter.

(1 Mark)

23. Question

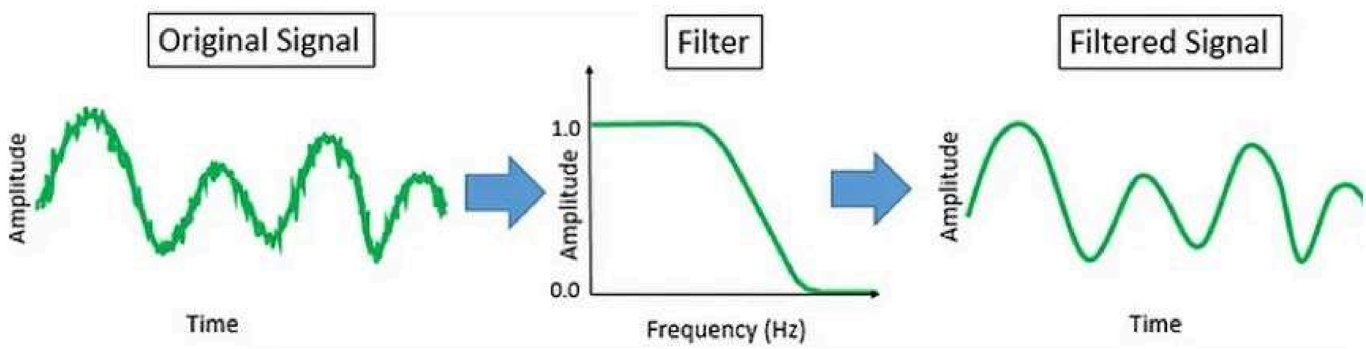


Figure: Audio FIltering Process

For audio filtering application we need the transfer function in H(z) using the impulse invariant method at Fs = 2 Hz. Given, the analog filter transfer function H(s) is given by:

$$H(s) = \frac{4s+7}{s^2+5s+4}$$

Follow the steps and obtain H(z)
1) Obtain partial fraction.
2) Apply mapping between the laplace domain and z domain using the relation

$$\frac{1}{s+p_k} \leftrightarrow \frac{1}{1-e^{-p_k T} z^{-1}}$$

3) Apply 'T' value and obtain the final H(z)

Answer

Partial Fraction

$$\frac{3}{s+4} + \frac{1}{s+1}$$

(1 Mark)
Apply mapping between the laplace domain and z domain using the relation

$$\frac{3}{1-e^{-4T} z^{-1}} + \frac{1}{1-e^{-T} z^{-1}}$$

(1 Mark)
Apply 'T' value and obtain the final H(z)

$$\frac{3}{1-e^{-2} z^{-1}} + \frac{1}{1-e^{-0..5} z^{-1}}$$

$$H(z) = \frac{4-1.9549z^{-1}}{1-0.7419z^{-1}+0.0821z^{-2}}$$

(1 Mark)

24. Question

Magnitude response of IIR low pass filter is given below. Specifications in the design of the filter is given as,
Pass band attenuation αp<= 3 dB
Stop band attenuation αs>= 30 dB
Pass band edge frequency Ωp=200 rad/sec
Stop band edge frequency Ωs=600 rad/sec.

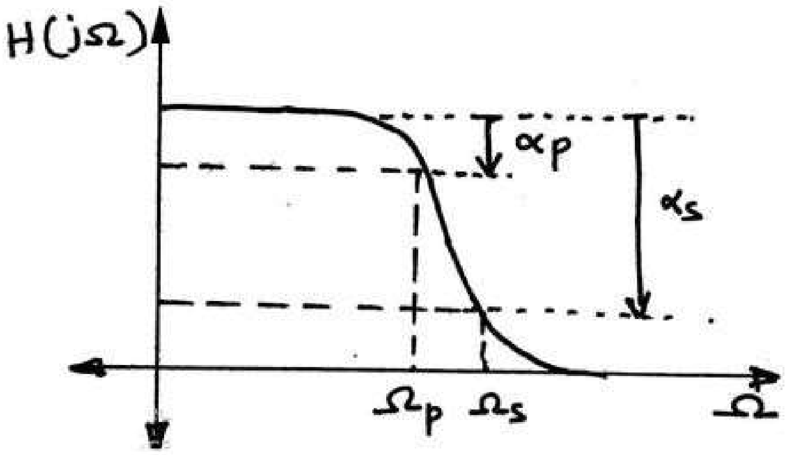


Figure: Magnitude Response

1) Compute the average cutoff frquency of the IIR low pass filter
2) For the given magnitude response, what is the significance of passband attenuation and stopband attenuation in this filter design?
3) What will be the transition band in the given case ?

Answer

1) Average cutoff frquency of the IIR low pass filter is = (Ωp + Ωs) / 2
(1 Mark)
Solution = 200 rad/sec
(1 Mark)
2) The passband edge frequency defines the frequency range where the signal experiences minimal attenuation, typically less than 3 dB.
(1 Mark)
The stopband edge frequency defines the point where attenuation is significant, typically at least 30 dB.
(1 Mark)
3) Transition Band = Ωs - Ωp
(1 Mark)
Solution = 400 rad/sec
(1 Mark)

25. Question

A digital signal processing engineer is tasked with designing an FIR filter for a communication system. The system requires that the filter have a linear phase response to prevent distortion in transmitted signals. The filter must also approximate the derivative of the input signal to enhance high-frequency components in the signal.
What type of FIR filter would the engineer use to ensure a linear phase response? Also specify the type of FIR filter should be designed to approximate the derivative of the input signal.

Answer

FIR filter would the engineer use to ensure a linear phase response is a linear phase FIR filter.
(1 mark)
The type of FIR filter should be designed to approximate the derivative of the input signal is an FIR differentiator.
(1 mark)

26. Question

FIR filters can be designed using:
a) Impulse invariance method
b) Fourier series method
c) Bilinear transformation
d) Chebyshev approximation

Answer

b) Fourier series method

27. Question

The primary advantage of FIR filters over IIR filters is:
a) Infinite response

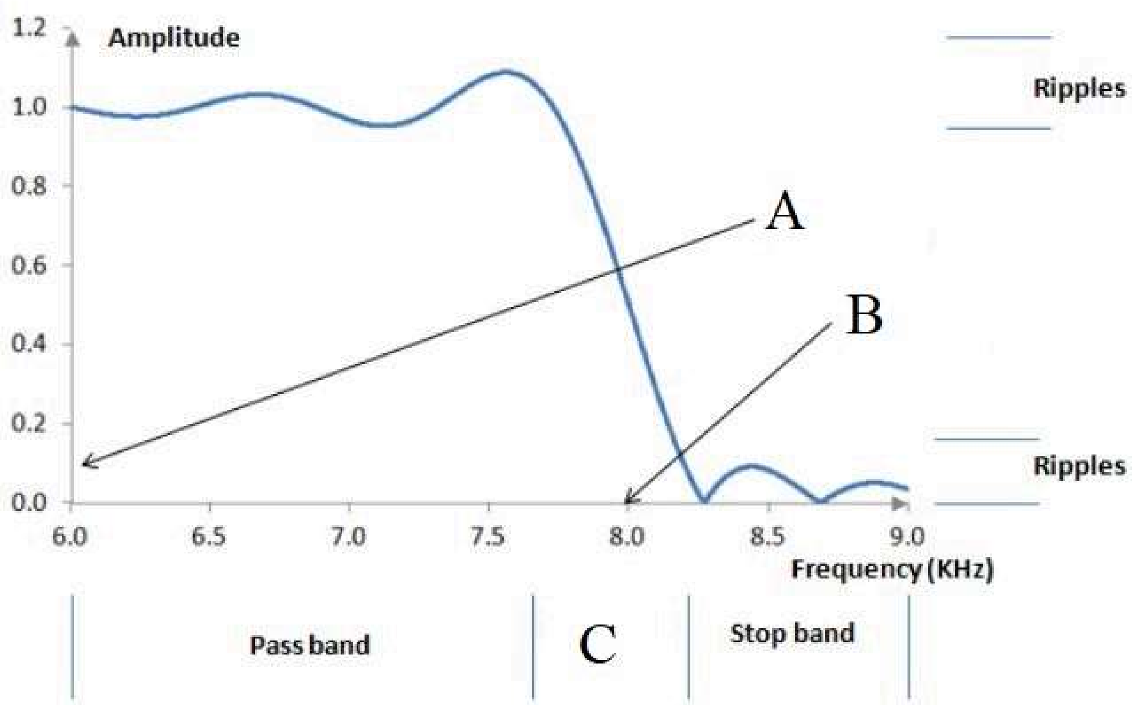
b) Linear phase property
c) Lower computational cost
d) Non-causal nature

Answer

b) Linear phase property

28. Question

Magnitude response of a real digital filter is given below.



1) Identify the regions marked as 'A', 'B', and 'C'.
2) Which filter has this magnitude response ?

Answer

1) Regions are:
A - Stop band attenuation
B - Cutoff Frequency
C - Transition Band
2) Filter with this magnitude response is a low pass filter (Chebychev Filter Type II)

29. Question

A subwoofer system has to be designed as shown in the figure.

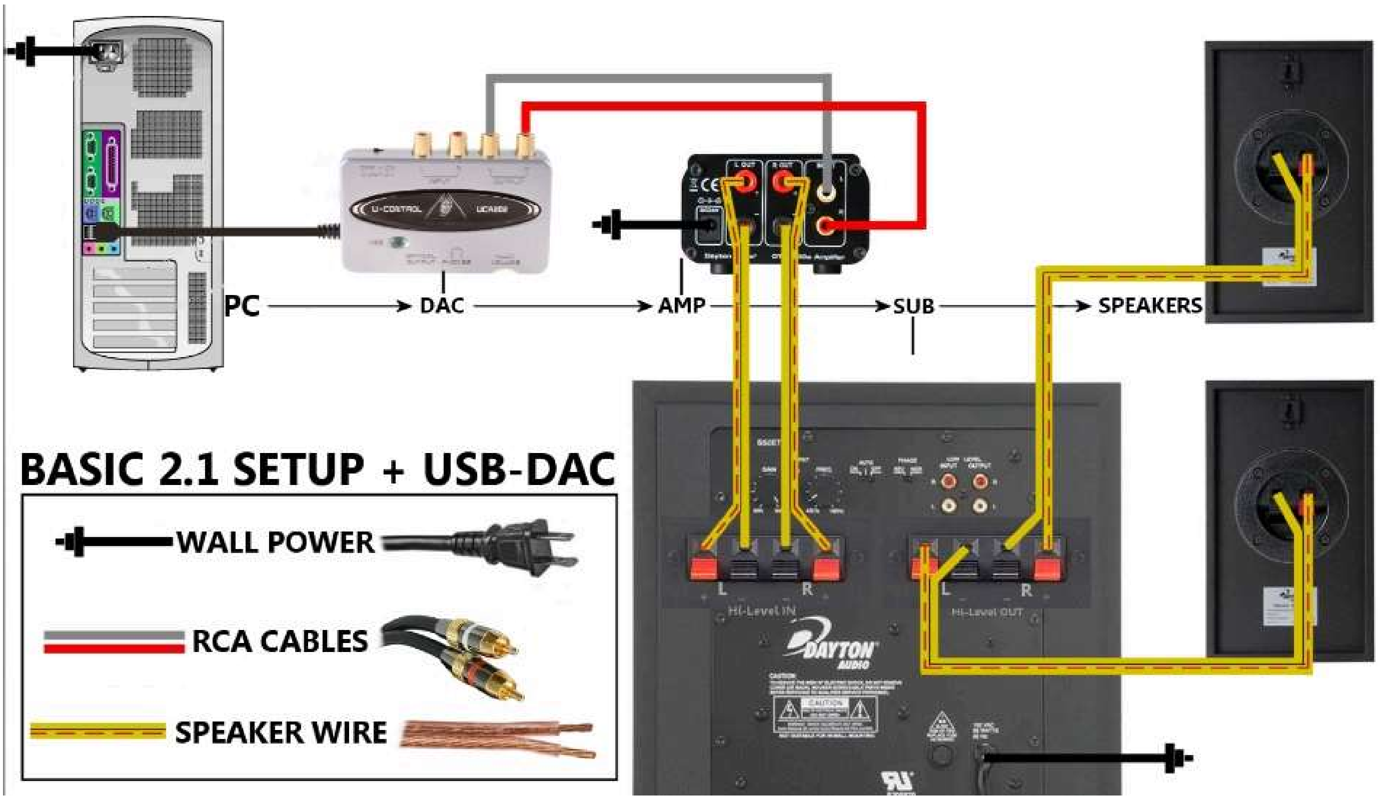


Figure: Subwoofer Circuit

The filter coefficients/ impulse response of the digital filter in the subwoofer system is obtained by performing various operations. These operations are given in jumbled order. Arrange them in a correct sequence based on the application given in the scenario.

i. Find the frequency response of the filter
ii. Obtain the impulse response of the filter for infinite duration
iii. Find the finite impulse response of the filter
iv. Apply z-transform
v. Apply inverse Fourier transform
vi. Find the impulse response of the realizable FIR filter
vii. Find the realizable filter transfer function
viii. Apply window function

Answer

The correct order is
i. Find the frequency response of the filter
(1 Mark)
v. Apply inverse Fourier transform
(1 Mark)
ii. Obtain the impulse response of the filter for infinite duration
viii. Apply window function
(1 Mark)
iii. Find the finite impulse response of the filter
vi. Find the impulse response of the realizable FIR filter
(1 Mark)
iv. Apply z-transform
(1 Mark)
vii. Find the realizable filter transfer function
(1 Mark)

30. Question

The number of multiplications required for an FIR filter depends on:
a) Input signal length
b) Filter length
c) Sampling frequency
d) Transition bandwidth

Answer

b) Filter length

31. Question

An engineer is tasked with designing an FIR filter for an audio processing system. The system requires a linear phase response to prevent signal distortion, and the filter must act as a differentiator to emphasize high-frequency components. The engineer decides to implement the FIR filter using the direct form realization structure.

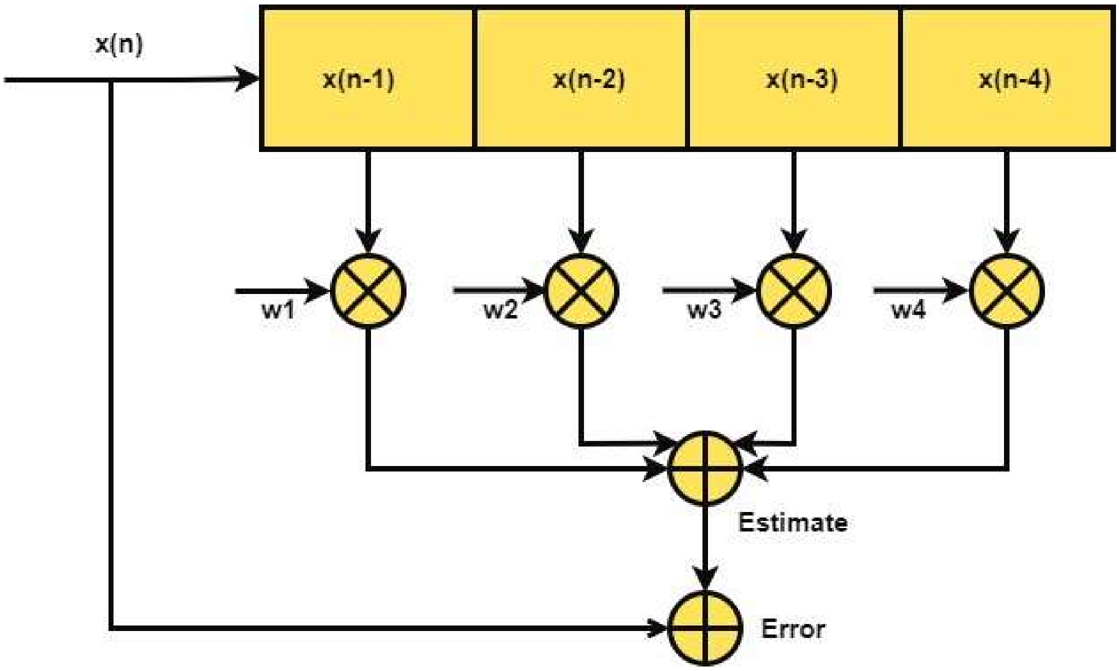


Figure: 4-tap Filter

Why is a linear phase response important and What does the FIR differentiator do to the input signal in this audio processing system in this scenario ?
Also explain the purpose of the delays and multipliers in the direct form realization structure in the given diagram.

Answer

A linear phase response is important to prevent phase distortion, ensuring that all frequency components of the audio signal are delayed by the same amount, preserving the signal's shape.
(1 mark)
The FIR differentiator enhances high-frequency components in the input signal, which helps in emphasizing sharp changes or transient features in the audio.
(1 mark)
The delays, represented by [z^(-1)], store previous input values, while the multipliers (h[0], h[1], h[2]) scale the current and delayed inputs by the filter coefficients. The outputs of the multipliers are summed to produce the final output y[n].
(1 mark)

32. Question

In the design of FIR differentiators, the ideal frequency response is:
a) Linearly increasing with frequency
b) Constant
c) Linearly decreasing with frequency
d) Exponentially increasing with frequency

Answer

a) Linearly increasing with frequency

33. Question

A signal processing engineer is tasked with designing low pass, high pass and bandpass filter for an edge detection system used in image processing. The filter needs to have a linear phase response to avoid distorting the sharp transitions in the image. The engineer plans to use a direct form FIR realization structure for the design, as shown in the diagram below, where the filter has 'n' taps

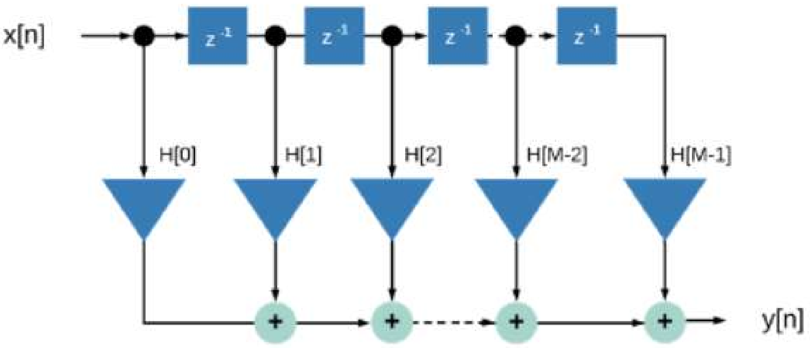
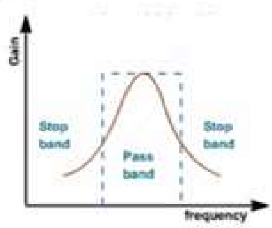
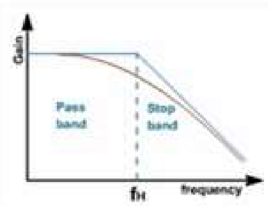
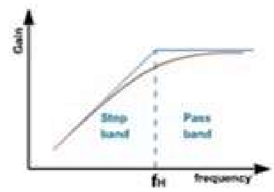


Figure: 4 Tap Filter

1) In this scenario, why FIR filters are used instead of an IIR filter. Support your answer with two reasons.
2) Also, match the following Question Based on the 4-Tap Filter Characteristics.

Filter type	Magnitude Response graph	Magnitude Response
(A1) Low pass filter	**(B1)** 	(C1) $\left\|H(e^{j\omega})\right\|=\begin{cases}0 & \omega=0\\ 1 & \omega=\pi\end{cases}$
(A2) High pass filter	**(B2)** 	**(C2)** $\left\|H(e^{j\omega})\right\|=\begin{cases}0 & \omega=0,\pi\\ 1 & \omega=\frac{\pi}{2}\end{cases}$
(A3) Band pass filter	**(B3)** 	(C3) $\left\|H(e^{j\omega})\right\|=\begin{cases}1 & \omega=0\\ 0 & \omega=\pi\end{cases}$

Answer

1) An FIR differentiator can be designed to have a perfect linear phase response, ensuring no phase distortion in the signal, whereas an IIR filter often introduces phase distortion due to its recursive structure.
(1 mark)
FIR filters are always stable, which is important for applications like signal differentiation where stability is critical, while IIR filters may become unstable depending on their pole locations.
(1 mark)
2) A1-B2-C3 ; A2-B3-C1; A3-B1-C2
(One mark for each correct answer - Total 3 Marks)

34. Question

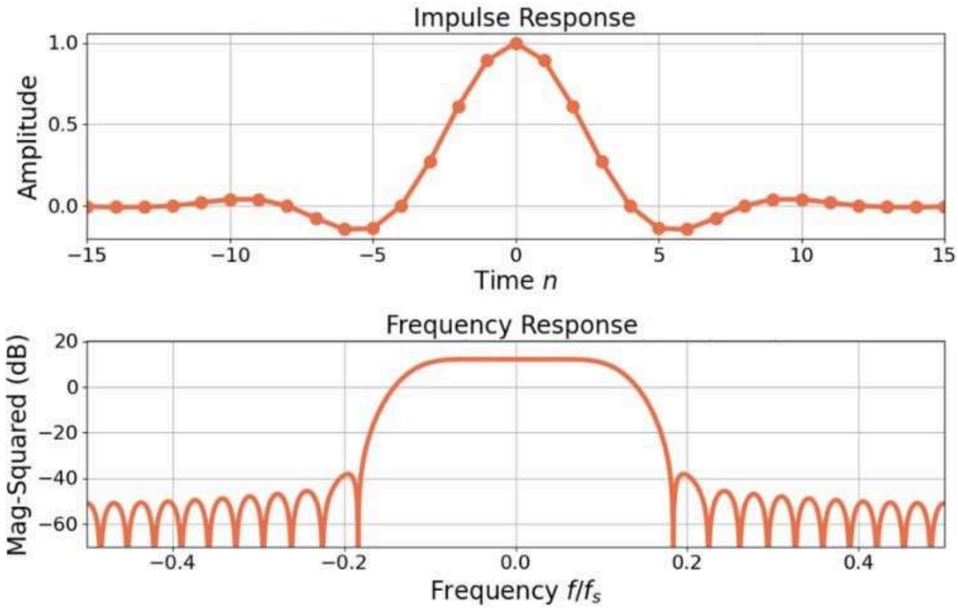
Which of the following is a property of the Hamming window?
a) Low sidelobe levels
b) Wide main lobe
c) Very high main lobe peak
d) Non-zero end values

Answer

a) Low sidelobe levels

35. Question

The impluse response and frequency reponse of a filter after applying window is given below:



1) i) What is the significance of impulse response? ii) How it is related to the overall characteristics of the system?
2) i) Based on the frequency response plot shown in the figure, at what normalized frequency does the magnitude start to significantly attenuate, and ii) what is the approximate level of magnitude at a normalized frequency of 0.4?

Answer

1) i) The impulse response represents how a system responds to a single unit impulse input.
(1 Mark)
ii) It reflects the overall characteristics of the system that is exhibited for any input signals.
(1 Mark)
2) i) The magnitude starts to significantly attenuate at a normalized frequency of approximately ±0.2.
(1 Mark)
ii) At a normalized frequency of 0.3, the approximate level of attenuation is around -50 dB.
(1 Mark)

36. Question

The order of an FIR filter is determined by:
a) The number of delays
b) The sampling rate
c) The transition bandwidth
d) The frequency response

Answer

a) The number of delays

37. Question

The rounding of multiplication results in digital filters leads to which of the following errors?
a) Coefficient quantization error
b) Input quantization error
c) Product quantization error
d) Filter overflow error

Answer

c) Product quantization error

38. Question

In a real-time signal processing system, a digital filter is implemented using 8-bit fixed-point arithmetic as shown in the figure. The input range is from -1 to 1, also, the coefficient of a filter is quantized from 0.7832 to 0.7812 due to finite word length.

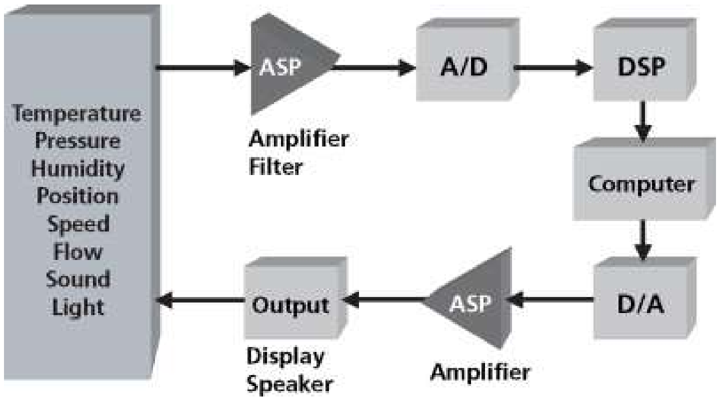


Figure: Digital Signal Processing

1) what is the quantization step size?
2) Estimate quantization error in the scenario.
3) In this DSP system, quantization noise becomes more significant when the signal is weaker. Why does this happen?

Answer

1) The quantization step size is calculated as 2 / (2^8) = 0.0078 for 8-bit representation.
(1 Mark)
2) Quantization error = |0.7832 - 0.7812| = 0.002.
(1 Mark)
3) Quantization noise is a fixed error due to precision limitations, so its relative effect becomes more prominent when the signal is weak.
(1 Mark)

39. Question

What is the typical cause of coefficient quantization errors in digital filters?
a) Approximating irrational numbers in filter design
b) Incorrect filter order
c) Excessive filter stability
d) Using floating-point arithmetic

Answer

a) Approximating irrational numbers in filter design

40. Question

Rearrange the sentences to match the correct order to achieve a fully functional digital filter implementation.
1.Apply quantization to filter coefficients to fit the system's finite word length.
2.Choose between fixed-point or floating-point representation for the filter implementation.
3. Determine the desired filter specifications such as cutoff frequency and filter order.
4. Implement the filter using the chosen digital signal processing algorithm.
5. Introduce input quantization to match the system's bit resolution (e.g., 16-bit or 24-bit audio).
6. Convert the audio signal from analog to digital using an ADC.
7. Quantize the filter's output to prevent overflow and maintain the precision within system limits.
8. Test the filter for quantization noise effects, adjusting parameters if needed.
9. Process the digitized signal through the filter to remove unwanted frequencies.
10. Output the filtered digital signal, followed by converting it back to analog using a DAC.

Answer

Correct Order:
3. Determine the desired filter specifications such as cutoff frequency and filter order.
2. Choose between fixed-point or floating-point representation for the filter implementation.
(1 Mark)
1. Apply quantization to filter coefficients to fit the system's finite word length.
6. Convert the audio signal from analog to digital using an ADC.
(1 Mark)
5. Introduce input quantization to match the system's bit resolution (e.g., 16-bit or 24-bit audio).
4. Implement the filter using the chosen digital signal processing algorithm.
1 Mark)

9. Process the digitized signal through the filter to remove unwanted frequencies.
7. Quantize the filter's output to prevent overflow and maintain the precision within system limits.
(1 Mark)
8. Test the filter for quantization noise effects, adjusting parameters if needed.
10. Output the filtered digital signal, followed by converting it back to analog using a DAC.
(1 Mark)
(Marks are awarded only for two consecutive steps in correct order)

41. Question

Which type of quantization noise is associated with multiplying two numbers in digital filters?
a) Input quantization
b) Coefficient quantization
c) Product quantization
d) Overflow quantization

Answer

c) Product quantization

42. Question

Finite word length effects in digital filters lead to:
a) Infinite precision in calculations
b) Reduced filter stability and accuracy
c) Increased filter robustness
d) Elimination of noise in the system

Answer

b) Reduced filter stability and accuracy

43. Question

Assertion (A): Finite word length effects can lead to limit cycle oscillations in digital filters.
Reason (R): Quantization of filter coefficients introduces non-linear behavior in the filter response, causing instability.
a) Both A and R are true, and R is the correct explanation of A
b) Both A and R are true, but R is not the correct explanation of A
c) A is true, but R is false
d) A is false, but R is true

Answer

b) Both A and R are true, but R is not the correct explanation of A

44. Question

The figure shows the process of sampling and quantization of a continuous signal x(t). Based on the plots, answer the following questions:

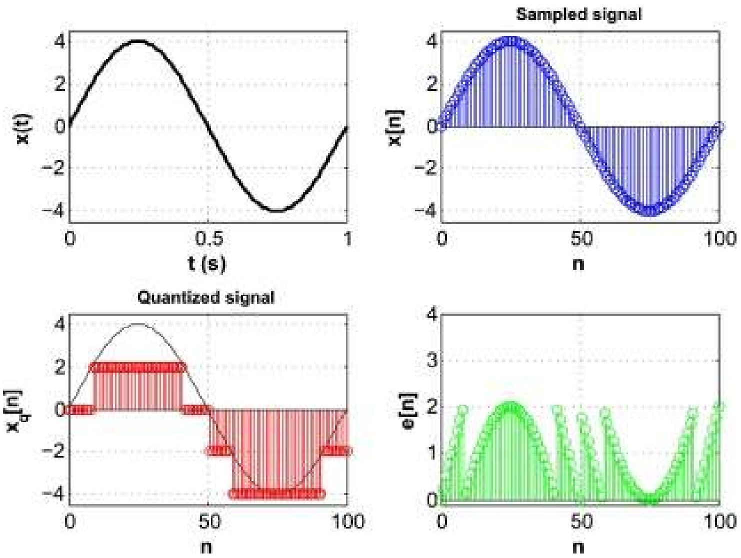


Figure: process of sampling and quantization

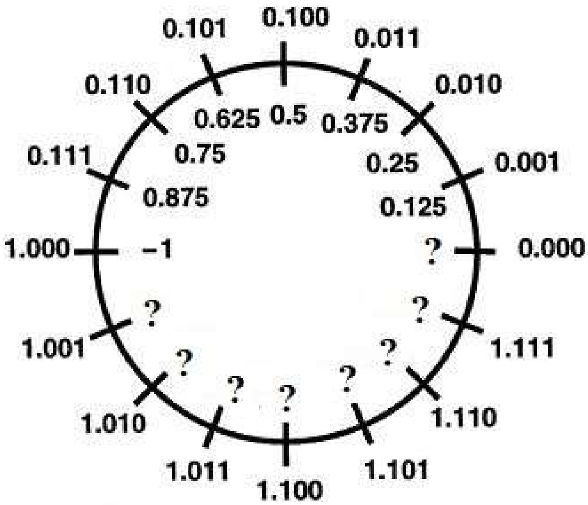
1) From the sampled signal plot, i) Determine the sampling rate if the original signal x(t) is a sinusoidal wave with a frequency of 1 Hz. ii) How it is calculated.
2) Based on the Quantized Signal plot, assume the signal is quantized using 3 bits. i) How many quantization levels are there, and ii) what is the step size if the signal amplitude ranges from -4 to 4?
3) From the Error Signal plot, i) describe the effect of quantization error on the signal. ii) What is the maximum possible error for the given quantization scheme?

Answer

1) i) Sampling rate = 100 Hz
(1 Mark)
ii) It is calculated with the assumption that 100 samples are taken over 1 second for a 1 Hz signal.
(1 Mark)
2) i) Quantization levels = 2^3 = 8
(1 Mark)
ii) The step size = 4 - (-4)/ 8 = 1
(1 Mark)
3) Quantization introduces a stair-step pattern, and the error signal shows the difference between the original and quantized signals.
(1 Mark)
The maximum error is half of the step size, i.e., 0.5
(1 Mark)

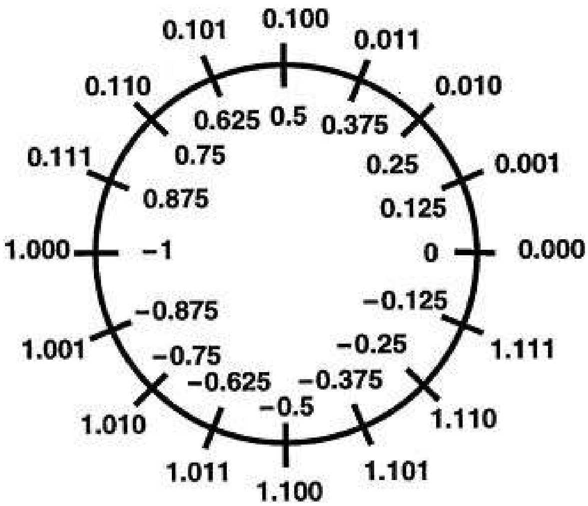
45. Question

Apply two's complement arithmetic to fill the circle.



Answer

Correct Solution:



One mark for each two correct answers, Total = 4 Marks.

46. Question

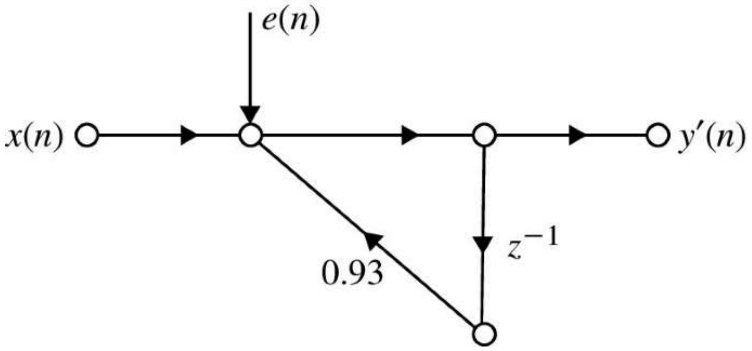
Quantization noise occurs because of:
a) Rounding numbers to fit within the available word length
b) Using higher bit-depth representations
c) Incorrect filter design
d) Sampling signals at high rates

Answer

a) Rounding numbers to fit within the available word length

47. Question

The figure shows a signal flow graph of a discrete-time system affected by quantization error e(n), with a feedback loop containing a delay element z^-1 and a multiplier of 0.93.



Signal Flow Graph of a discrete-time system affected by quantization error

1) Write the difference equation for the system based on the signal flow graph.
2) Assuming finite word length effects cause quantization noise e(n) to be added at each step, express how this affects the system's output y'(n).
3) If the system is implemented using a 4-bit word length for coefficients, calculate the approximate value of the coefficient 0.93 (fraction: 15/16) after quantization (round to the nearest representable 4-bit fraction).
4) Discuss how the finite word length (4 bits) for the coefficient might affect the system's stability, given that the original coefficient is 0.93.

Answer

1) The difference equation from the signal flow graph is:
$y'(n) = x(n) + 0.93 \cdot y\,(n-1) + e(n)$
(1 Mark)
2) Quantization noise e(n) introduces an error at each step, which accumulates over time due to the feedback loop, potentially amplifying the error in the output.
(1 Mark)
3) With 4 bits, the coefficient 0.93 is approximated as 15/16=0.9375 after quantization.
(1 Mark)
4) The quantized coefficient 0.9375 is close to 0.93, but any reduction in accuracy may affect the system's stability, particularly if the actual feedback coefficient drifts toward 1, which could cause oscillations or instability.
(1 Mark)

48. Question

Which of the following is a key drawback of fixed-point representation in digital filters?
a) Limited dynamic range
b) Increased processing complexity
c) Higher memory requirements
d) Overflow protection

Answer

a) Limited dynamic range

49. Question

The Architecture of TMS320C50 processor shown in the figure below:

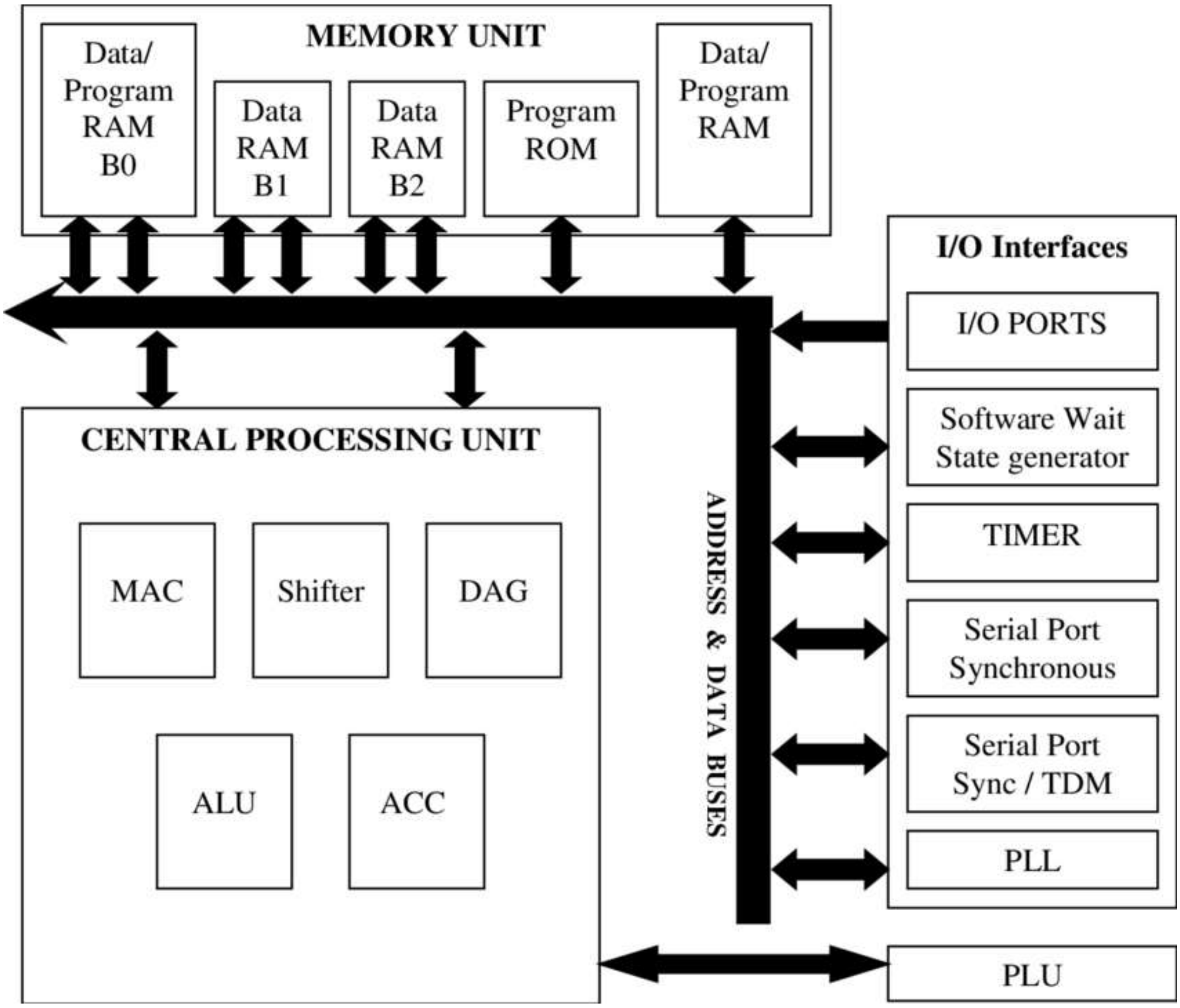


Figure: DSP Processor TMS320C50 Architecture

1) State True or False. The TMS320C50 DSP chip uses a single RAM unit for both data and program storage. Also justify your answer in one line.
2) Using the architecture of the TMS320C50 DSP chip, describe how the Barrel Shifter unit can be used to shift the following 8-bit binary number by 3 positions to the left.
3) Using the architecture of the TMS320C50 DSP chip, describe how the MAC (Multiply-Accumulate) unit can be used to compute the following expression:

$$y = \sum_{i=1}^{3} x_i \cdot w_i$$

,

$$x = [2, 3, 4] \textit{ and } w = [1, 0, -1]$$

Answer

1) False.
(1 Mark)
The diagram shows separate RAM blocks for data and program storage, as well as shared Data/Program RAM blocks.
(1 Mark)
2)
Shift the binary number 11010011 by 3 positions to the left.
(1 Mark)
Perform the left shift on the binary number: 11010011 becomes 10011000. The result is 152 in decimal.
(1 Mark)
3) $y=(2\cdot1)+(3\cdot0)+(4\cdot-1)$
(1 Mark)
$y=2+0+(-4)=-2$
(1 Mark)

50. Question

How many functional units does the TMS320C6713 processor have for executing operations?
a) 4
b) 6

c) 8
d) 10

Answer
c) 8

51. Question
A pressure monitoring system implemented using digital signal processing is given in the figure.

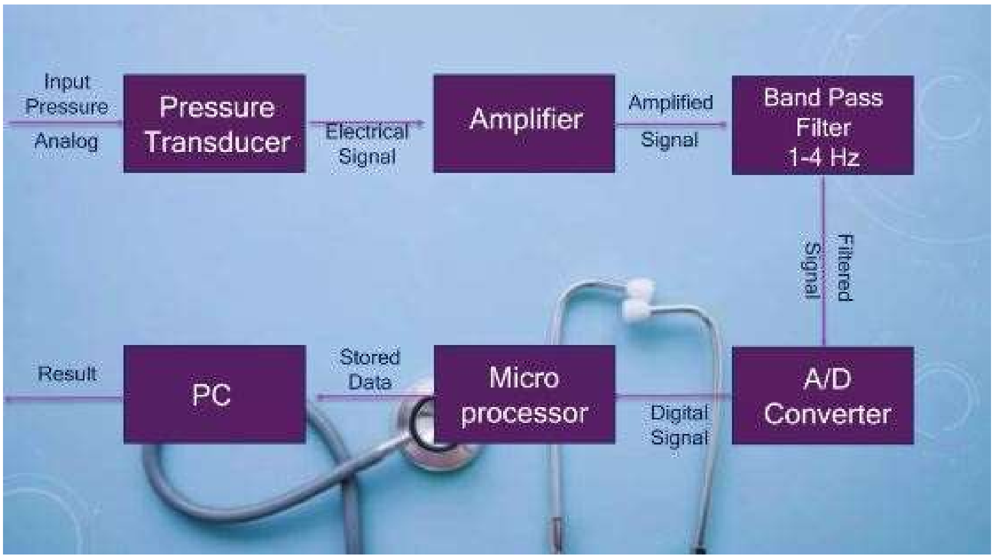


Figure: Pressure monitoring system using digital signal processing

1) Explain the role of the Band Pass Filter (1-4 Hz) in this system. Why is it necessary to filter the signal after amplification, and what does the frequency range indicate?
2) If the A/D Converter samples the filtered signal at 10 kHz and each sample is represented with 16 bits, calculate the data rate generated for the microprocessor to store.

Answer
1) The Band Pass Filter (1-4 Hz) allows only signals within this frequency range to pass through while attenuating frequencies outside this range, ensuring that only relevant low-frequency components are amplified.
(1 Mark)
Filtering after amplification is necessary to reduce noise and interference, while the 1-4 Hz range typically indicates the frequency of specific biological signals, such as certain brain or heart activity rhythms.
(1 Mark)
2) The data rate can be calculated using the formula, Data Rate = Data Rate=Sampling Rate x Bits per Sample
(1 Mark)
Substituting the values, we get,
Data Rate=10,000samples/secondx16bits/sample=160,000bits/second or 160kbps.
(1 Mark)

52. Question
Which functional unit in the TMS320C6713 DSP is responsible for multiplication operations?
a) L Unit
b) M Unit
c) S Unit
d) D Unit

Answer
b) M Unit

53. Question
Which instruction in TMS320C6713 assembly language would be used to place a value from a register into memory?
a) MOV
b) LDW
c) ADD
d) STW

Answer
d) STW

54. Question
The figure below shows the various spectrum outputs occurred during decimation process.

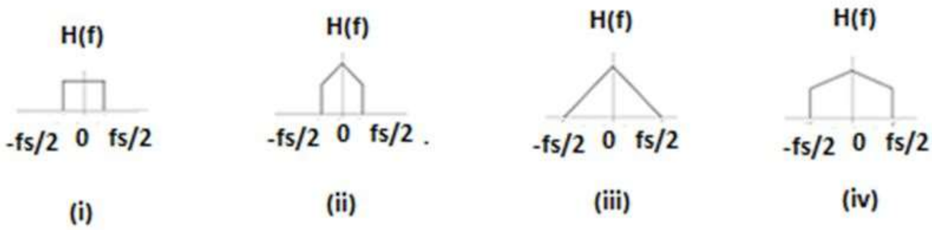


Figure: Spectrum of Decimation Process

Arrange them in order based on the step by step decimation process.

Answer
Correct Order:
(i), (iv), (ii), (iii)
(2 Marks)

55. Question
The application for digital signal processor in processing signal in earphone is given in the figure.

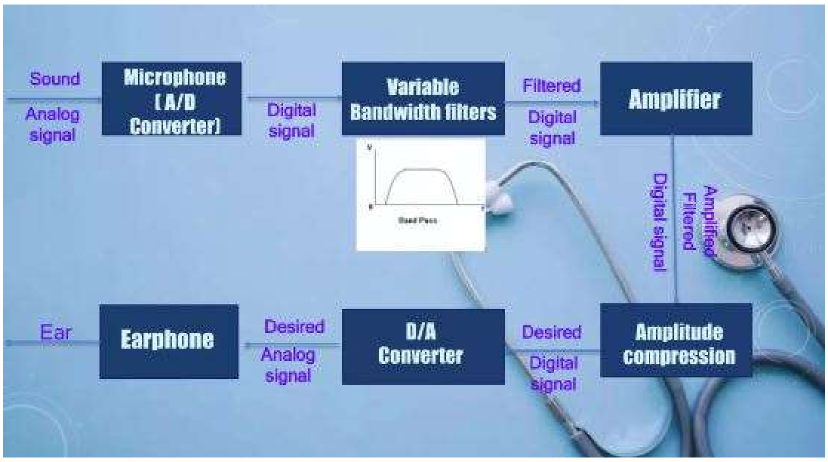


Figure: Digital Signal Processing performed in Earphone

1) In what types of medical applications would a system like the one shown (with microphones, filters, and DAC) be commonly used? Mention one specific real-life application.

2) If the D/A Converter operates at a sampling rate of 40 kHz and the output signal has a maximum frequency of 18 kHz, determine whether the sampling rate is sufficient according to the Nyquist criterion. What would be the required minimum sampling rate?

Answer
1) A system with microphones, filters, and a digital-to-analog converter (DAC) is commonly used in medical applications such as hearing aids.
(1 Mark)
A specific real-life application is the enhancement of speech signals for patients with hearing loss, allowing them to better understand conversations in noisy environments.
(1 Mark)
2) Apply Nyquist Criterion: According to the Nyquist criterion, the sampling rate must be at least twice the maximum frequency.
Calculate Required Minimum Sampling Rate: Required minimum sampling rate = $2 \times 18 kHz = 36 kHz$.
(1 Mark)
Compare with Actual Sampling Rate: The D/A converter operates at 40 kHz, which is greater than 36 kHz, thus the sampling rate is sufficient.
(1 Mark)

56. Question
The TMS320C6713 uses which type of pipeline for instruction processing?
a) Three-stage pipeline

b) Five-stage pipeline
c) Six-stage pipeline
d) Ten-stage pipeline

Answer

b) Five-stage pipeline

57. Question

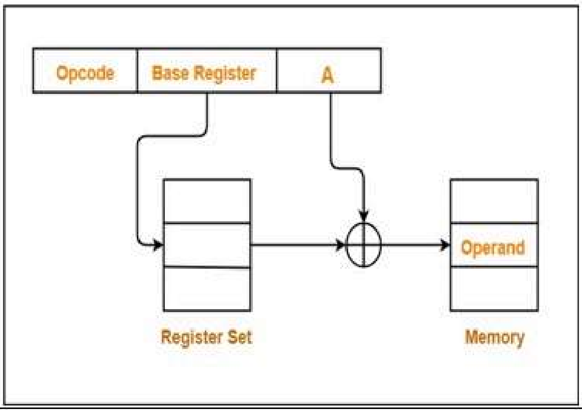
In the TMS320C6713 instruction pipeline, what is the role of the "fetch stage"?
a) Decoding the instruction
b) Executing the instruction
c) Fetching the instruction from memory
d) Writing the result back to memory

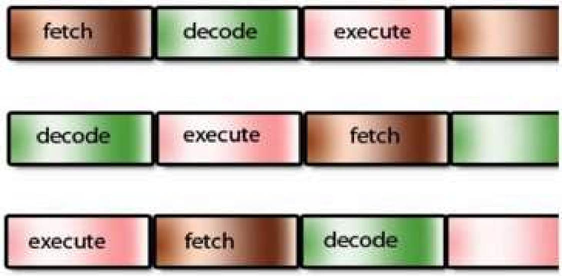
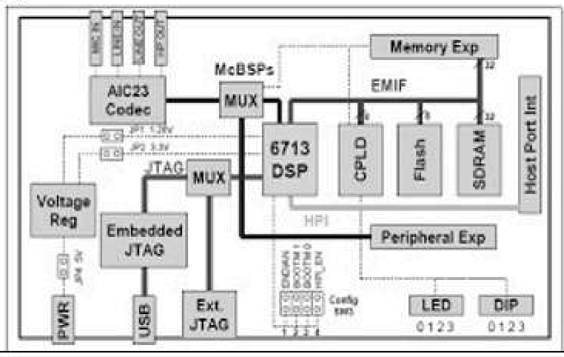
Answer

c) Fetching the instruction from memory

58. Question

Match the following

<table>
<tr><th>Concepts</th><th>Image</th><th>Descriptions</th><th>Examples</th></tr>
<tr><td>A1
TMS320C6713 Functional Unit</td><td>B1

</td><td>C1
Set of operations that the processor can execute, including data manipulation and control flow instructions.</td><td>D1
Immediate, Register, Direct</td></tr>
<tr><td>A2
Addressing Modes</td><td>B2

</td><td>C2
Contains execution units such as ALUs and multipliers used for performing DSP tasks.</td><td>D2
Fetch, Decode, Execute stages</td></tr>
<tr><td>A3
Instruction Set</td><td>B3

</td><td>C3
A technique that allows overlapping of instruction execution to improve throughput.</td><td>D3
Multiply-Accumulate (MAC) operation</td></tr>
<tr><td>A4
Pipelining</td><td>B4

</td><td>C4
Methods of specifying operands in assembly language instructions.</td><td>D4
ADD, SUB, MPY (Multiply), B (Branch)</td></tr>
</table>

Answer

A1-B4-C2-D3
A2-B1-C4-D1
A3-B2-C1-D4
A4-B3-C3-D2
(1 Mark for each correct answer = 4 Marks)

59. Question

Two commonly used processing architectures are shown in the figure. The TMS320C6713 processor uses the Harvard Architecture.

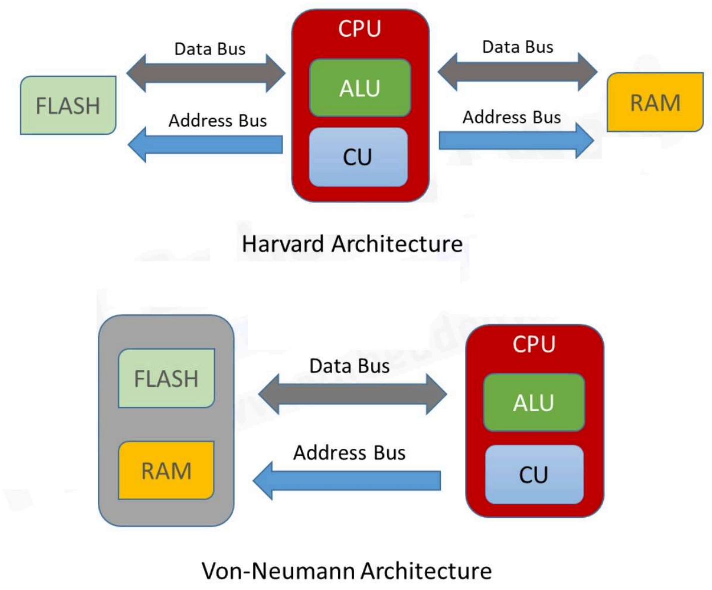


Figure: Processor Architectures

1) How do addressing modes in the TMS320C6713 influence performance in Harvard architecture compared to Von Neumann?
2) If the memory bandwidth of the instruction bus is 32 bits wide and runs at 100 MHz, what is the bandwidth in MBps?

Answer

1) In Harvard architecture, separate address spaces for instructions and data allow for more flexible and faster memory access.
(1 Mark)
But, in Von Neumann, the same address space could create bottlenecks.
(1 mark)
2) Bandwidth = 32 bits × 100 MHz ÷ 8 bits = 400 MBps.
(2 Marks)

60. Question

In the TMS320C6713, which functional unit performs logical operations?
a) M Unit
b) L Unit
c) S Unit
d) D Unit

Answer

b) L Unit