



[1000 Digital Signal Processing MCQs](https://www.sanfoundry.com/1000-digital-signal-processing-questions-answers/)

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Our 1000+ Digital Signal Processing questions and answers focuses on all areas of DSP covering 100+ topics. These topics are chosen from a collection of most authoritative and best reference books on Digital Signal Processing. One should spend 1 hour daily for 2-3 months to learn and assimilate DSP comprehensively. This way of systematic learning will prepare anyone easily towards Digital Signal Processing interviews, online tests, examinations and certifications.

**Highlights**  
– 1000+ Multiple Choice Questions & Answers in Digital Signal Processing with explanations  
– Every MCQ set focuses on a specific topic in Digital Signal Processing Subject

**Who should Practice these Digital Signal Processing Questions?**  
– Anyone wishing to sharpen their knowledge of Digital Signal Processing Subject  
– Anyone preparing for aptitude test in Digital Signal Processing  
– Anyone preparing for interviews (campus/off-campus interviews, walk-in interview and company interviews)  
– Anyone preparing for entrance examinations and other competitive examinations  
– All – Experienced, Freshers and Students

**Here’s list of Questions & Answers on Digital Signal Processing Subject covering 100+ topics:**

**1. Questions & Answers on Discrete Time Signals and Systems**

The section contains questions and answers on discrete time systems, their analysis and implementation, discrete time signals, differential equations and the correlation of discrete time signals.

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| [Discrete Time System Implementation](https://www.sanfoundry.com/digital-signal-processing-questions-answers-implementation-discrete-time-systems/) [Discrete Time Systems Difference Equations](https://www.sanfoundry.com/digital-signal-processing-questions-answers-discrete-time-systems-described-difference-equations/) [Discrete Time System Analysis](https://www.sanfoundry.com/digital-signal-processing-questions-answers-analysis-discrete-time-lti-systems/) | [Discrete Time Signals](https://www.sanfoundry.com/digital-signal-processing-questions-answers-discrete-time-signals/) [Discrete Time Systems](https://www.sanfoundry.com/digital-signal-processing-questions-answers-discrete-time-systems/) [Discrete Time Signal Correlation](https://www.sanfoundry.com/digital-signal-processing-questions-answers-correlation-discrete-time-signals/) |

**2. Questions on DSP – Basic Signaling**

The section contains questions on A/D and D/A converters, signal classification and signal and system processing.

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| [A2D and D2A Converters](https://www.sanfoundry.com/digital-signal-processing-questions-answers-a2d-d2a-converters/) [Signal Classification](https://www.sanfoundry.com/digital-signal-processing-questions-answers-classification-signals/) | [Signal and System Processing](https://www.sanfoundry.com/digital-signal-processing-questions-answers-signals-systems-signal-processing/) |

**3. Questions & Answers on Z transform and its Application – Analysis of the LTI Systems**

The section contains questions and answers on Z transforms and its properties, types of Z transforms which include rational, inverse and one sided Z transform and their analysis.

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| [Z Transform](https://www.sanfoundry.com/digital-signal-processing-questions-answers-z-transform/) [Z Transform Properties-1](https://www.sanfoundry.com/digital-signal-processing-questions-answers-properties-z-transform-1/) [Z Transform Properties-2](https://www.sanfoundry.com/tough-digital-signal-processing-questions-answers/) [Rational Z Transform](https://www.sanfoundry.com/digital-signal-processing-questions-answers-rational-z-transform/) | [Z Transform Inversion](https://www.sanfoundry.com/digital-signal-processing-questions-answers-inverse-z-transform/) [One Sided Z Transform](https://www.sanfoundry.com/digital-signal-processing-questions-answers-one-sided-z-transform/) [Z Domain System Analysis](https://www.sanfoundry.com/digital-signal-processing-questions-answers-analysis-lti-system-z-domain/) |

**4. Questions on Frequency Analysis of Signals and Systems**

The section contains questions on frequency analysis of discrete and continuous time signals, fourier transform properties, convolution and de-convolution concepts, inverse systems, LTI systems and discrete time signals.

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| [Continuous Time Signal Analysis](https://www.sanfoundry.com/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal/) [Discrete Time Signal Analysis-1](https://www.sanfoundry.com/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1/) [Discrete Time Signal Analysis-2](https://www.sanfoundry.com/digital-signal-processing-interview-questions-answers/) [Fourier Transforms Properties](https://www.sanfoundry.com/digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals/) | [LTI System Frequency Characteristics](https://www.sanfoundry.com/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system/) [Frequency Selective Filters](https://www.sanfoundry.com/digital-signal-processing-questions-answers-lti-system-frequency-selective-filters/) [Inverse Systems and Deconvolution](https://www.sanfoundry.com/digital-signal-processing-questions-answers-inverse-systems-deconvolution/) |

**5. Questions & Answers on Discrete Fourier Transform – Properties and Applications**

The section contains questions and answers on discrete fourier transforms, their sampling and properties, linear filtering methods on DFT and their frequency analysis.

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| [Frequency Domain Sampling](https://www.sanfoundry.com/digital-signal-processing-questions-answers-frequency-domain-sampling-dft/) [DFT Properties](https://www.sanfoundry.com/digital-signal-processing-questions-answers-properties-dft/) | [DFT Signal Analysis](https://www.sanfoundry.com/digital-signal-processing-questions-answers-frequency-analysis-signals-using-dft/) |

**6. Questions on DFT Efficient Computation – Fast Fourier Transform Algorithms**

The section contains questions on computation of discrete fourier transforms and fast fourier transforms, various approaches to their computation which include filtering and quantization and applications of FFT algorithms.

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| [DFT Algorithm Computation 1](https://www.sanfoundry.com/digital-signal-processing-questions-answers-efficient-computation-dft-fft-algorithms-1/) [DFT Algorithm Computation 2](https://www.sanfoundry.com/digital-signal-processing-questions-answers-freshers/) [FFT Algorithms Applications](https://www.sanfoundry.com/digital-signal-processing-questions-answers-applications-fft-algorithms/) | [DFT Computation Filtering Approach](https://www.sanfoundry.com/digital-signal-processing-interview-questions-answers-freshers/) [Quantization Effects](https://www.sanfoundry.com/digital-signal-processing-questions-answers-quantization-effects-computation-dft/) |

**7. Questions & Answers on Discrete Time Systems Implementation**

The section contains questions and answers on realization structures for discrete time systems, FIR system structures, IIR system structures, number representation, state space system analysis, quantization error analysis and bilinear transformations.

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| [Structures for Realization](https://www.sanfoundry.com/digital-signal-processing-questions-answers-structures-realization-discrete-time-systems/) [FIR System Structures 1](https://www.sanfoundry.com/digital-signal-processing-questions-answers-structures-fir-systems-1/) [FIR System Structures 2](https://www.sanfoundry.com/digital-signal-processing-questions-answers-experienced/) [IIR System Structures](https://www.sanfoundry.com/digital-signal-processing-questions-answers-structures-iir-systems/) [State-Space System Analysis](https://www.sanfoundry.com/digital-signal-processing-questions-answers-state-space-system-analysis-structures/) | [Number Representation 1](https://www.sanfoundry.com/digital-signal-processing-questions-answers-representation-numbers-1/) [Number Representation 2](https://www.sanfoundry.com/digital-signal-processing-interview-questions-answers-experienced/) [Discrete-Time Signal Processing](https://www.sanfoundry.com/advanced-digital-signal-processing-questions-answers/) [Quantization Errors Analysis](https://www.sanfoundry.com/digital-signal-processing-questions-answers-analysis-quantization-errors/) [IIR Filter Design](https://www.sanfoundry.com/tricky-digital-signal-processing-questions-answers/) |

**8. Questions on Digital Filters Design**

The section contains questions on design of low pass butterworth filters and chebyshev filters, bilinear transformations, filter coefficient quantization, design considerations for filters, FIR filter design using windows, forward and backward difference methods, filter design using frequency sampling method, FIR differentiator design, Hilbert transformer design, IIR filter design, approximation of derivatives, impulse variance, analog filter characteristics, various approximation methods, sampling rate conversion and interpolation techniques.

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| [Butterworth Filters Design 1](https://www.sanfoundry.com/digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1/) [Butterworth Filters Design 2](https://www.sanfoundry.com/digital-signal-processing-test/) [Chebyshev Filters 1](https://www.sanfoundry.com/digital-signal-processing-questions-answers-chebyshev-filters-1/) [Chebyshev Filters 2](https://www.sanfoundry.com/digital-signal-processing-quiz/) [Backward Difference Method](https://www.sanfoundry.com/digital-signal-processing-questions-answers-backward-difference-method/) [Bilinear Transformations](https://www.sanfoundry.com/digital-signal-processing-questions-answers-bilinear-transformations/) [Filter Coefficients Quantization](https://www.sanfoundry.com/digital-signal-processing-questions-answers-quantization-filter-coefficients/) [Digital Filters Round Off Effects](https://www.sanfoundry.com/digital-signal-processing-questions-answers-round-off-effects-digital-filters/) [Digital Filters Design Consideration](https://www.sanfoundry.com/digital-signal-processing-questions-answers-general-considerations-design-digital-filters/) [FIR Filters Design](https://www.sanfoundry.com/digital-signal-processing-questions-answers-design-fir-filters/) [FIR Filters Windows Design 1](https://www.sanfoundry.com/digital-signal-processing-questions-answers-design-linear-phase-fir-filters-using-windows-1/) [FIR Filters Windows Design 2](https://www.sanfoundry.com/digital-signal-processing-mcqs/) [Frequency Sampling Method FIR Design](https://www.sanfoundry.com/digital-signal-processing-multiple-choice-questions-answers/) [Optimum Equi Ripple Filter Design 1](https://www.sanfoundry.com/digital-signal-processing-online-test/) [Optimum Equi Ripple Filter Design 2](https://www.sanfoundry.com/digital-signal-processing-quiz-online/) | [FIR Differentiator Design](https://www.sanfoundry.com/digital-signal-processing-questions-answers-design-fir-differentiators/) [Hilbert Transformers Design](https://www.sanfoundry.com/digital-signal-processing-questions-answers-design-hilbert-transformers/) [FIR Filters Design Comparison](https://www.sanfoundry.com/digital-signal-processing-question-bank/) [Analog Filters Design](https://www.sanfoundry.com/digital-signal-processing-questions-answers-design-iir-filters-analog-filters/) [Approximation of Derivatives design Method](https://www.sanfoundry.com/digital-signal-processing-questions-entrance-exams/) [Impulse Invariance Filter Design](https://www.sanfoundry.com/digital-signal-processing-questions-campus-interviews/) [Matched Z Transformation](https://www.sanfoundry.com/digital-signal-processing-questions-answers-matched-z-transformation/) [Analog Filter Characteristics](https://www.sanfoundry.com/digital-signal-processing-aptitude-test/) [Analog Domain Frequency Transformations](https://www.sanfoundry.com/digital-signal-processing-problems/) [Digital Domain Frequency Transformations](https://www.sanfoundry.com/basic-digital-signal-processing-questions-answers/) [Pade Approximation Method](https://www.sanfoundry.com/digital-signal-processing-questions-answers-pade-approximation-method/) [Least Squares Design Methods](https://www.sanfoundry.com/digital-signal-processing-questions-answers-least-squares-design-methods/) [FIR Least Squares Filters](https://www.sanfoundry.com/digital-signal-processing-questions-answers-fir-least-squares-inverse-filters/) [IIR Frequency Domain Filter Analysis](https://www.sanfoundry.com/digital-signal-processing-questions-answers-design-iir-filters-frequency-domain/) [Analog Filters Classification](https://www.sanfoundry.com/digital-signal-processing-questions-answers-specifications-classification-analog-filters/) [Butterworth Filters](https://www.sanfoundry.com/digital-signal-processing-questions-answers-butterworth-filters/) [Frequency Transformations](https://www.sanfoundry.com/digital-signal-processing-questions-answers-frequency-transformations/) [Factor I Interpolation](https://www.sanfoundry.com/digital-signal-processing-questions-answers-interpolation-factor-i/) [Sampling Rate Conversion](https://www.sanfoundry.com/tough-digital-signal-processing-questions-answers/) |

**9. Questions & Answers on Multirate Digital Signal Procesing**

The section contains questions and answers on factor decimation and multirate digital signal processing.

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| [Multirate Signal Processing](https://www.sanfoundry.com/digital-signal-processing-questions-answers-multirate-digital-signal-processing/) | [Factor D Decimation](https://www.sanfoundry.com/digital-signal-processing-questions-answers-decimation-factor-d/) |

**10. Questions on Sampling and Reconstruction of Signals**

This section contains questions on A/D Converters and their oversampling, band pass signal sampling and representation, sample and hold concepts and quantization and coding techniques.

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| [A/D Converter Oversampling](https://www.sanfoundry.com/digital-signal-processing-questions-answers-oversampling-ad-converters/) [Sample and Hold](https://www.sanfoundry.com/digital-signal-processing-questions-answers-sample-hold/) [Band Pass Signal Sampling](https://www.sanfoundry.com/digital-signal-processing-questions-answers-sampling-band-pass-signals/) | [Bandpass Signal Representation](https://www.sanfoundry.com/digital-signal-processing-questions-answers-representation-bandpass-signals/) [Quantization and Coding](https://www.sanfoundry.com/digital-signal-processing-questions-answers-quantization-coding/) [Digital to Analog Conversion](https://www.sanfoundry.com/digital-signal-processing-questions-answers-digital-analog-conversion-sample-hold/) |

**1. Questions & Answers on Discrete Time Signals and Systems**

The section contains questions and answers on discrete time systems, their analysis and implementation, discrete time signals, differential equations and the correlation of discrete time signals.

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| [Discrete Time System Implementation](https://www.sanfoundry.com/digital-signal-processing-questions-answers-implementation-discrete-time-systems/) [Discrete Time Systems Difference Equations](https://www.sanfoundry.com/digital-signal-processing-questions-answers-discrete-time-systems-described-difference-equations/) [Discrete Time System Analysis](https://www.sanfoundry.com/digital-signal-processing-questions-answers-analysis-discrete-time-lti-systems/) | [Discrete Time Signals](https://www.sanfoundry.com/digital-signal-processing-questions-answers-discrete-time-signals/) [Discrete Time Systems](https://www.sanfoundry.com/digital-signal-processing-questions-answers-discrete-time-systems/) [Discrete Time Signal Correlation](https://www.sanfoundry.com/digital-signal-processing-questions-answers-correlation-discrete-time-signals/) |

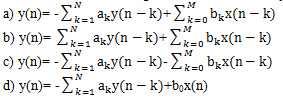
Questions & Answers (MCQs) focuses on “Implementation of Discrete Time Systems”.

1. The system described by the equation y(n)=ay(n-1)+b x(n) is a recursive system.  
a) True  
b) False  
View Answer

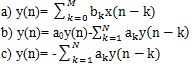
Answer: a  
Explanation: Since the present output depends on the value of the previous output, the system is called as Recursive system.

2. To implement the linear time invariant recursive system described by the difference equation [digital-signal-processing-questions-answers-implementation-discrete-time-systems-q2](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-implementation-discrete-time-systems-q2.png)in Direct form-I, how many number of delay elements and multipliers are required respectively?  
a) M+N+1,M+N  
b) M+N-1,M+N  
c) M+N,M+N+1  
d) None of the mentioned  
View Answer

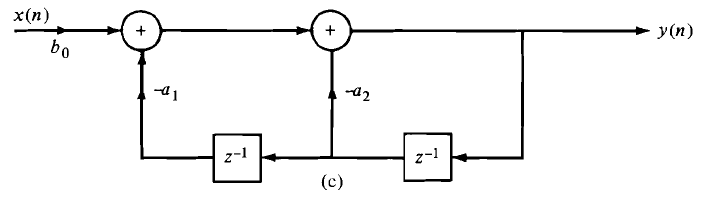
Answer: c  
Explanation: From the given equation, there are M+N delays, so it requires M+N number of delay elements and it has to perform M+N+1 multiplications, so it require that many number of multipliers.

3. Which of the following linear time invariant system is a purely recursive system?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-implementation-discrete-time-systems-q3.png)  
View Answer

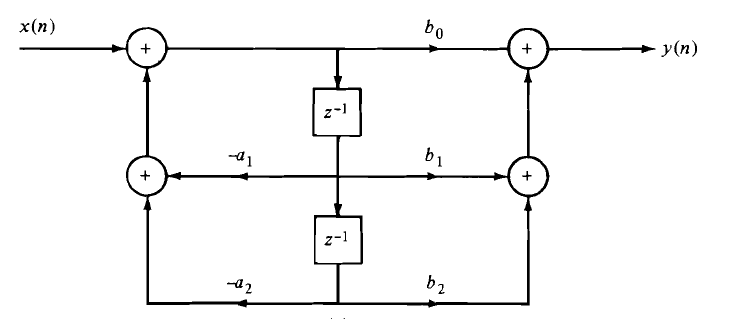
Answer: d  
Explanation: Since the output of the system depend only on the past values of output and the present value of the input, the system is called as “purely recursive” system.

4. Which of the following is the difference equation of a special case of FIR system?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-implementation-discrete-time-systems-q4.png)  
d) None of the mentioned  
View Answer

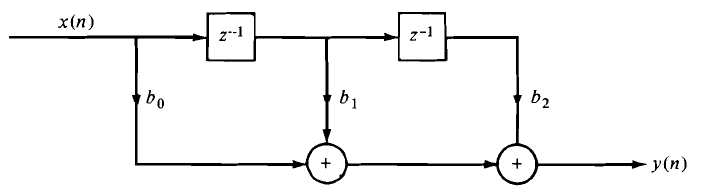
Answer: a  
Explanation: If the coefficients of the past values of the output in the difference equation of the system, then the system is said to be FIR system.

5. What is the system does the following direct form structure represents?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-implementation-discrete-time-systems-q5.png)  
a) FIR system  
b) Purely recursive system  
c) General second order system  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: Since the output of the system depends only on the present value of the input and the past values of the output, the system is a purely recursive system.

6. What is the output of the system represented by the following direct form?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-implementation-discrete-time-systems-q6.png)  
a) y(n)= -a1y(n-1)-a2y(n-2)- b0x(n)-b1x(n-1)-b2x(n-2)  
b) y(n)= -a1y(n-1)-a2y(n-2)+b0x(n)  
c) y(n)= -a1y(n-1)-a2y(n-2)+ b0x(n)+b1x(n-1)+b2x(n-2)  
d) y(n)= a1y(n-1)+a2y(n-2)+ b0x(n)+b1x(n-1)+b2x(n-2)  
View Answer

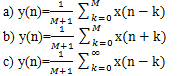
Answer: c  
Explanation: The equation of the difference equation of any system is defined as  
[digital-signal-processing-questions-answers-implementation-discrete-time-systems-q2](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-implementation-discrete-time-systems-q2.png)  
In the given diagram, N=M=2  
So, substitute the values of the N and M in the above equation.  
We get, y(n)= -a1y(n-1)-a2y(n-2)+ b0x(n)+b1x(n-1)+b2x(n-2)

7. The system represented by the following direct form structure is:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-implementation-discrete-time-systems-q7.png)  
a) General second order system  
b) Purely recursive system  
c) Partial recursive system  
d) FIR system  
View Answer

Answer: d  
Explanation: The output of the system according to the direct form given is  
y(n)= b0x(n)+b1x(n-1)+b2x(n-2)  
Since the output of the system is purely dependent on the present and past values of the input, the system is called as FIR system.

8. An FIR system is also called as “recursive system”.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: For a system to be recursive, the output of the system must be dependent only on the past values of the output. For an FIR system the output of the system must be depending only on the present and past values of the input. So, FIR system is not an recursive system.

9. What is the form of the FIR system to compute the moving average of the signal x(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-implementation-discrete-time-systems-q9.png)  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: A normal FIR non-recursive system with the impulse response h(n)= 1/(M+1) is the system which is used to compute the moving average of a signal x(n).

10. Which of the following is a recursive form of a non-recursive system described by the equation [digital-signal-processing-questions-answers-implementation-discrete-time-systems-q10](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-implementation-discrete-time-systems-q10.png)  
a) y(n)=y(n-1)+ 1/(M+1)[x(n)+x(n-1-M)].  
b) y(n)=y(n-1)+ 1/(M+1)[x(n)+x(n-1+M)].  
c) y(n)=y(n-1)+ 1/(M+1)[x(n)-x(n-1+M)].  
d) y(n)=y(n-1)+ 1/(M+1)[x(n)-x(n-1-M)].  
View Answer

Answer: d  
Explanation: The given system equation is [digital-signal-processing-questions-answers-implementation-discrete-time-systems-q10](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-implementation-discrete-time-systems-q10.png)  
It can be expressed as follows [digital-signal-processing-questions-answers-implementation-discrete-time-systems-q10a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-implementation-discrete-time-systems-q10a.png)

11. The system described by the equation y(n)=ay(n+1)+b x(n) is a recursive system.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: Since the present output depends on the value of the future output, the system is not called as Recursive system.

Questions & Answers (MCQs) focuses on “Discrete Time Systems Described by Difference Equations “.

1. If the system is initially relaxed at time n=0 and memory equals to zero, then the response of such state is called as:  
a) Zero-state response  
b) Zero-input response  
c) Zero-condition response  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: The memory of the system, describes, in some case, the ‘state’ of the system, the output of the system is called as ‘zero-state response’.

2. Zero-state response is also known as:  
a) Free response  
b) Forced response  
c) Natural response  
d) None of the mentioned  
View Answer

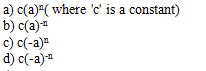
Answer: b  
Explanation: The zero-state response depends on the nature of the system and the input signal. Since this output is a response forced upon it by the input signal, it is also known as ‘Forced response’.

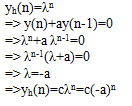
3. Zero-input response is also known as Natural or Free response.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: For a zero-input response, the input is zero and the output of the system is independent of the input of the system. So, the response if such system is also known as Natural or Free response.

4. The solution obtained by assuming the input x(n) of the system is zero is:  
a) General solution  
b) Particular solution  
c) Complete solution  
d) Homogenous solution  
View Answer

Answer: d  
Explanation: By making the input x(n)=0 we will get a homogenous difference equation and the solution of that difference equation is known as Homogenous or Complementary solution.

5. What is the homogenous solution of the system described by the first order difference equation y(n)+ay(n-1)=x(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-discrete-time-systems-described-difference-equations-q5.png)  
View Answer

Answer: c  
Explanation: The assumed solution obtained by assigning x(n)=0 is  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-discrete-time-systems-described-difference-equations-q5a.png)

6. What is the zero-input response of the system described by the homogenous second order equation y(n)-3y(n-1)-4y(n-2)=0 if the initial conditions are y(-1)=5 and y(-2)=0?  
a) (-1)n-1 + (4)n-2  
b) (-1)n+1 + (4)n+2  
c) (-1)n+1 + (4)n-2  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: Given difference equation is y(n)-3y(n-1)-4y(n-2)=0—-(1)  
Let y(n)=λn  
Substituting y(n) in the given equation  
=> λn – 3λn-1 – 4λn-2 = 0  
=> λn-2(λ2 – 3λ – 4) = 0  
the roots of the above equation are λ=-1,4  
Therefore, general form of the solution of the homogenous equation is  
[digital-signal-processing-questions-answers-discrete-time-systems-described-difference-equations-q6](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-discrete-time-systems-described-difference-equations-q6.png)  
The zero-input response of the system can be calculated from the homogenous solution by evaluating the constants in the above equation, given the initial conditions y(-1) and y(-2).  
From the given equation (1)  
y(0)=3y(-1)+4y(-2)  
y(1)=3y(0)+4y(-1)  
=3[3y(-1)+4y(-2)]+4y(-1)  
=13y(-1)+12y(-2)  
From the equation (2)  
y(0)=C1+C2 and  
y(1)=C1(-1)+C2(4)=-C1+4C2  
By equating these two set of relations, we have  
C1+C2=3y(-1)+4y(-2)=15  
-C1+4C2=13y(-1)+12y(-2)=65  
On solving the above two equations we get C1=-1 and C2=16  
Therefore the zero-input response is Yzi(n) = (-1)n+1 + (4)n+2.

7. What is the particular solution of the first order difference equation y(n)+ay(n-1)=x(n) where |a|<1, when the input of the system x(n)=u(n)?  
a) 1/(1+a) u(n)  
b) 1/(1-a) u(n)  
c) 1/(1+a)  
d) 1/(1-a)  
View Answer

Answer: a  
Explanation: The assumed solution of the difference equation to the forcing equation x(n), called the particular solution of the difference equation is  
yp(n)=Kx(n)=Ku(n) (where K is a scale factor)  
Substitute the above equation in the given equation  
=>Ku(n)+aKu(n-1)=u(n)  
To determine K we must evaluate the above equation for any n>=1, so that no term vanishes.  
=> K+aK=1  
=>K=1/(1+a)  
Therefore the particular solution is yp(n)= 1/(1+a) u(n).

8. What is the particular solution of the difference equation y(n)= 5/6y(n-1)- 1/6y(n-2)+x(n) when the forcing function x(n)=2n, n≥0 and zero elsewhere?  
a) (1/5) 2n  
b) (5/8) 2n  
c) (8/5) 2n  
d) (5/8) 2-n  
View Answer

Answer: c  
Explanation: The assumed solution of the difference equation to the forcing equation x(n), called the particular solution of the difference equation is  
yp(n)=Kx(n)=K2nu(n) (where K is a scale factor)  
Upon substituting yp(n) into the difference equation, we obtain  
K2nu(n)=5/6K2n-1u(n-1)-1/6 K2n-2u(n-2)+2nu(n)  
To determine K we must evaluate the above equation for any n>=2, so that no term vanishes.  
=> 4K= 5/6(2K)-1/6 (K)+4  
=> K= 8/5  
=> yp(n)= (8/5) 2n.

9. The total solution of the difference equation is given as:  
a) yp(n)-yh(n)  
b) yp(n)+yh(n)  
c) yh(n)-yp(n)  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The linearity property of the linear constant coefficient difference equation allows us to add the homogenous and particular solution in order to obtain the total solution.

10. What is the impulse response of the system described by the second order difference equation y(n)-3y(n-1)-4y(n-2)=x(n)+2x(n-1)?  
a)[-1/5 (-1)n-6/5 (4)n]u(n)  
b)[1/5 (-1)n – 6/5 (4)n]u(n)  
c)[ 1/5 (-1)n+ 6/5 (4)n]u(n)  
d)[- 1/5 (-1)n+ 6/5 (4)n]u(n)  
View Answer

Answer: d  
Explanation: The homogenous solution of the given equation is yh(n)=C1(-1)n+C2(4)n—-(1)  
To find the impulse response, x(n)=δ(n)  
now, for n=0 and n=1 we get  
y(0)=1 and  
y(1)=3+2=5  
From equation (1) we get  
y(0)=C1+C2 and  
y(1)=-C1+4C2  
On solving the above two set of equations we get  
C1=- 1/5 and C2= 6/5  
=>h(n)= [-1/5 (-1)n + 6/5 (4)n]u(n).

Questions & Answers (MCQs) focuses on “Analysis of Discrete time LTI Systems”.

1. Resolve the sequence [digital-signal-processing-questions-answers-analysis-discrete-time-lti-systems-q1](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-discrete-time-lti-systems-q1.png)into a sum of weighted impulse sequences  
a) 2δ(n)+4δ(n-1)+3δ(n-3)  
b) 2δ(n+1)+4δ(n)+3δ(n-2)  
c) 2δ(n)+4δ(n-1)+3δ(n-2)  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: We know that, x(n)δ(n-k)=x(k)δ(n-k)  
x(-1)=2=2δ(n+1)  
x(0)=4=4δ(n)  
x(2)=3=3δ(n-2)  
Therefore, x(n)= 2δ(n+1)+4δ(n)+3δ(n-2).

2. The formula [digital-signal-processing-questions-answers-analysis-discrete-time-lti-systems-q2](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-discrete-time-lti-systems-q2.png)that gives the response y(n) of the LTI system as the function of the input signal x(n) and the unit sample response h(n) is known as:  
a) Convolution sum  
b) Convolution product  
c) Convolution Difference  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: The input x(n) is convoluted with the impulse response h(n) to yield the output y(n).As we are summing the different values, we call it as Convolution sum.

3. What is the order of the four operations that are needed to be done on h(k) in order to convolute x(k) and h(k)?  
Step-1:Folding  
Step-2:Multiplicaton with x(k)  
Step-3:Shifting  
Step-4:Summation  
a) 1-2-3-4  
b) 1-2-4-3  
c) 2-1-3-4  
d) 1-3-2-4  
View Answer

Answer: d  
Explanation: First the signal h(k) is folded to get h(-k). Then it is shifted by n to get h(n-k). Then it is multiplied by x(k) and then summed over -∞ to ∞.

4. The impulse response of a LTI system is h(n)={1,1,1}. What is the response of the signal to the input x(n)={1,2,3}?  
a) {1,3,6,3,1}  
b) {1,2,3,2,1}  
c) {1,3,6,5,3}  
d) {1,1,1,0,0}  
View Answer

Answer: c  
Explanation: Let y(n)=x(n)\*h(n)(‘\*’ symbol indicates convolution symbol)  
From the formula of convolution we get,  
y(0)=x(0)h(0)=1.1=1  
y(1)=x(0)h(1)+x(1)h(0)=1.1+2.1=3  
y(2)=x(0)h(2)+x(1)h(1)+x(2)h(0)=1.1+2.1+3.1=6  
y(3)=x(1)h(2)+x(2)h(1)=2.1+3.1=5  
y(4)=x(2)h(2)=3.1=3  
Therefore, y(n)=x(n)\*h(n)={1,3,6,5,3}.

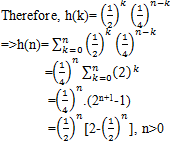
5. Determine the output y(n) of a LTI system with impulse response h(n)=anu(n),|a|<1with the input sequence x(n)=u(n).  
a) (1-a(n+1))/(1-a)  
b) (1-a(n-1))/(1-a)  
c) (1+a(n+1))/(1+a)  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: Now fold the signal x(n) and shift it by one unit at a time and sum as follows  
y(0)=x(0)h(0)=1  
y(1)=h(0)x(1)+h(1)x(0)=1.1+a.1=1+a  
y(2)=h(0)x(2)+h(1)x(1)+h(2)x(0)=1.1+a.1+a2.1=1+a+a2  
Similarly, y(n)=1+a+a2+….an= (1-a(n+1))/(1-a).

6. x(n)\*(h1(n)\*h2(n))=(x(n)\*h1(n))\*h2(n)  
a) True  
b) False  
View Answer

Answer: a  
Explanation: According to the properties of convolution, Convolution of three signals obeys Associative property.

7. Determine the impulse response for the cascade of two LTI systems having impulse responses h1(n)=(1/2)2 u(n) and h2(n)= (1/4)2 u(n).  
a) (1/2)n[2-(1/2)n], n<0  
b) (1/2)n[2-(1/2)n], n>0  
c) (1/2)n[2+(1/2)n], n<0  
d) (1/2)n[2+(1/2)n], n>0  
View Answer

Answer: b  
Explanation: Let h2(n) be shifted and folded.  
so, h(k)=h1(n)\*h2(n)=[digital-signal-processing-questions-answers-analysis-discrete-time-lti-systems-q7](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-discrete-time-lti-systems-q7.png)  
For k<0, h1(n)= h2(n)=0 since the unit step function is defined only on the right hand side. [](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-discrete-time-lti-systems-q7a.png)

8. x(n)\*[h1(n)+h2(n)]=x(n)\*h1(n)+x(n)\*h2(n)  
a) True  
b) False  
View Answer

Answer: a  
Explanation: According to the properties of the convolution, convolution exhibits distributive property.

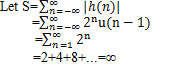
9. An LTI system is said to be causal if and only if  
a) Impulse response is non-zero for positive values of n  
b) Impulse response is zero for positive values of n  
c) Impulse response is non-zero for negative values of n  
d) Impulse response is zero for negative values of n  
View Answer

Answer: d  
Explanation: Let us consider a LTI system having an output at time n=n0given by the convolution formula  
[digital-signal-processing-questions-answers-analysis-discrete-time-lti-systems-q9](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-discrete-time-lti-systems-q9.png)  
=(h(0)x(n0)+h(1)x(n0-1)+h(2)x(n0-2)+….)+(h(-1)x(n0+1)+h(-2)x(n0+2)+…)  
As per the definition of the causality, the output should depend only on the present and past values of the input. So, the coefficients of the terms x(n0+1), x(n0+2)…. should be equal to zero.  
that is, h(n)=0 for n<0 .

10. x(n)\*δ(n-n0)=  
a) x(n+n0)  
b) x(n-n0)  
c) x(-n-n0)  
d) x(-n+n0)  
View Answer

Answer: b  
Explanation:  
[digital-signal-processing-questions-answers-analysis-discrete-time-lti-systems-q10](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-discrete-time-lti-systems-q10.png)

11. Is the system with impulse response h(n)=2nu(n-1) stable?  
a) True  
b) False  
View Answer

Answer: b  
Explanation:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-discrete-time-lti-systems-q11.png)  
So, the system is not stable.

Questions & Answers (MCQs) focuses on ” Discrete Time Signals”.

1. If x(n) is a discrete-time signal, then the value of x(n) at non integer value of ‘n’ is:  
a) Zero  
b) Positive  
c) Negative  
d) Not defined  
View Answer

Answer: d  
Explanation: For a discrete time signal, the value of x(n) exists only at integral values of n. So, for a non- integer value of ‘n’ the value of x(n) does not exist.

2. The discrete time function defined as u(n)=n for n≥0;=0 for n<0 is an:  
a) Unit sample signal  
b) Unit step signal  
c) Unit ramp signal  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: When we plot the graph for the given function, we get a straight line passing through origin with a unit positive slope. So, the function is called as unit ramp signal.

3.The phase function of a discrete time signal x(n)=an, where a=r.ejθ is:  
a) tan(nθ)  
b) nθ  
c) tan-1(nθ)  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: Given x(n)=an=(r.ejθ)n =rn.ejnθ  
=>x(n)=rn.(cosnθ+jsinnθ)  
Phase function is tan-1(cosnθ/sinnθ)=tan-1(tan nθ)=nθ.

4. The signal given by the equation[digital-signal-processing-questions-answers-discrete-time-signals-q4](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-discrete-time-signals-q4.png) is known as:  
a) Energy signal  
b) Power signal  
c) Work done signal  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: We have used the magnitude-squared values of x(n), so that our definition applies to complex-valued as well as real-valued signals. If the energy of the signal is finite i.e., 0<E<∞ then the given signal is known as Energy signal.

5. x(n)\*δ(n-k)=?  
a) x(n)  
b) x(k)  
c) x(k)\*δ(n-k)  
d) x(k)\*δ(k)  
View Answer

Answer: c  
Explanation: The given signal is defined only when n=k by the definition of delta function. So, x(n)\*δ(n-k)= x(k)\*δ(n-k).

6. A real valued signal x(n) is called as anti-symmetric if:  
a) x(n)=x(-n)  
b) x(n)=-x(-n)  
c) x(n)=-x(n)  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: According to the definition of anti-symmetric signal, the signal x(n) should be symmetric over origin. So, for the signal x(n) to be symmetric, it should satisfy the condition x(n)=-x(-n).

7. The odd part of a signal x(t) is:  
a) x(t)+x(-t)  
b) x(t)-x(-t)  
c) (1/2)\*(x(t)+x(-t))  
d) (1/2)\*(x(t)-x(-t))  
View Answer

Answer: d  
Explanation: Let x(t)=xe(t)+xo(t)  
=>x(-t)=xe(-t)-xo(-t)  
By subtracting the above two equations, we get  
xo(t)=(1/2)\*(x(t)-x(-t)).

8. Time scaling operation is also known as:  
a) Down-sampling  
b) Up-sampling  
c) Sampling  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: If the signal x(n) was originally obtained by sampling a signal xa(t), then x(n)=xa(nT). Now, y(n)=x(2n)(say)=xa(2nT). Hence the time scaling operation is equivalent to changing the sampling rate from 1/T to 1/2T, that is to decrease the rate by a factor of 2. So, time scaling is also called as down-sampling.

9. What is the condition for a signal x(n)=Brn where r=eαT to be called as an decaying exponential signal?  
a) 0<r<∞  
b) 0<r<1  
c) r>1  
d) r<0  
View Answer

Answer: b  
Explanation: When the value of ‘r’ lies between 0 and 1 then the value of x(n) goes on decreasing exponentially with increase in value of ‘n’. So, the signal is called as decaying exponential signal.

10. The function given by the equation x(n)=1, for n=0;=0, for n≠0 is a:  
a) Step function  
b) Ramp function  
c) Triangular function  
d) Impulse function  
View Answer

Answer: d  
Explanation: According to the definition of the impulse function, it is defined only at n=0 and is not defined elsewhere which is as per the signal given.

Questions & Answers (MCQs) focuses on “Discrete Time Systems”.

1. The output signal when a signal x(n)=(0,1,2,3) is processed through an ‘Identical’ system is:  
a) (3,2,1,0)  
b) (1,2,3,0)  
c) (0,1,2,3)  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: An identical system is a system whose output is same as the input, that is it does not perform any operation on the input and transmits it.

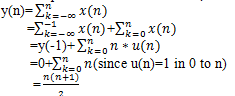
2. If a signal x(n) is passed through a system to get an output signal of y(n)=x(n+1), then the signal is said to be:  
a) Delayed  
b) Advanced  
c) No operation  
d) None of the mentioned  
View Answer

Answer: d  
Explanation: For example, the value of the output at the time n=0 is y(0)=x(1), that is the system is advanced by one unit.

3. If the output of the system is [digital-signal-processing-questions-answers-discrete-time-systems-q3](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-discrete-time-systems-q3.png)with an input of x(n) then the system will work as:  
a) Accumulator  
b) Adder  
c) Subtractor  
d) Multiplier  
View Answer

Answer: a  
Explanation: From the equation given, y(n)=x(n)+x(n-1)+x(n-2)+…. .This system calculates the running sum of all the past input values till the present time. So, it acts as an accumulator.

4. What is the output y(n) when a signal x(n)=n\*u(n)is passed through a accumulator system under the conditions that it is initially relaxed?  
a) (n2+n+1)/2  
b) (n(n+1))/2  
c) (n2+n+2)/2  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: Given that the system is initially relaxed, that is y(-1)=0  
According to the equation of the accumulator,  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-discrete-time-systems-q4.png)

5. The block denoted as follows is known as:  
[digital-signal-processing-questions-answers-discrete-time-systems-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-discrete-time-systems-q5.png)  
a) Delay block  
b) Advance block  
c) Multiplier block  
d) Adder block  
View Answer

Answer: a  
Explanation: If the function to this block is x(n) then the output from the block will be x(n-1). So, the block is called as delay block or delay element.

6. The output signal when a signal x(n)=(0,1,2,3) is processed through an ‘Delay’ system is:  
a) (3,2,1,0)  
b) (1,2,3,0)  
c) (0,1,2,3)  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: An delay system is a system whose output is same as the input, but after a delay.

7. The system described by the input-output equation y(n)=nx(n)+bx3(n) is a:  
a) Static system  
b) Dynamic system  
c) Identical system  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: Since the output of the system y(n) depends only on the present value of the input x(n) but not on the past or the future values of the input, the system is called as static or memory-less system.

8. Whether the system described by the input-output equations y(n)=x(n)-x(n-1) a Time-variant system?  
a) True  
b) False  
View Answer

Answer: b  
Explanation: If the input is delayed by k units then the output will be y(n,k)=x(n-k)-x(n-k-1)  
If the output is delayed by k units then y(n-k)=x(n-k)-x((n-k)-1)  
=>y(n,k)=y(n-k). Hence the system is time-invariant.

9. The system described by the input-output equations y(n)=x2(n) is a Non-linear system.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Given equation is y(n)=x2(n)  
Let y1(n)=x12(n) and y2(n)=x22(n)  
y3(n)=y1(n)+y2(n)= x12(n)+ x22(n)≠(x1(n)+x2(n))2  
So the system is non-linear.

10. If the output of the system of the system at any ‘n’ depends only the present or the past values of the inputs then the system is said to be:  
a) Linear  
b) Non-Linear  
c) Causal  
d) Non-causal  
View Answer

Answer: c  
Explanation: A system is said to be causal if the output of the system is defined as the function shown below  
y(n)=F[x(n),x(n-1),x(n-2),…] So, according to the conditions given in the question, the system is a causal system.

11. The system described by the input-output equations y(n)=x(-n) is a causal system.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: For n=-1, y(-1)=x(1)  
That is, the output of the system at n=-1 is depending on the future value of the input at n=1. So the system is a non-causal system.

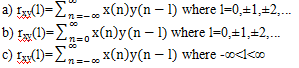
12. If a system do not have a bounded output for bounded input, then the system is said to be:  
a) Causal  
b) Non-causal  
c) Stable  
d) Non-stable  
View Answer

Answer: d  
Explanation: An arbitrary relaxed system is said to be BIBO stable if it has a bounded output for every value in the bounded input. So, the system given in the question is a Non-stable system.

Questions & Answers (MCQs) focuses on “Correlation of Discrete Time Signals”.

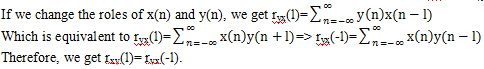
1. Which of the following parameters are required to calculate the correlation between the signals x(n) and y(n)?  
a) Time delay  
b) Attenuation factor  
c) Noise signal  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: Let us consider x(n) be the input reference signal and y(n) be the reflected signal.  
Now, the relation between the two signals is given as y(n)=αx(n-D)+w(n)  
Where α-attenuation factor representing the signal loss in the round-trip transmission of the signal x(n)  
D-time delay between the time of projection of signal and the reflected back signal  
w(n)-noise signal generated in the electronic parts in the front end of the receiver.

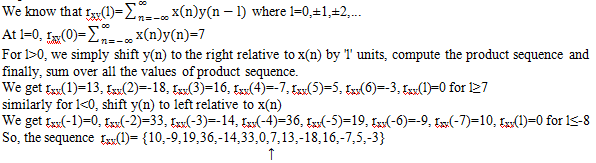
2. The cross correlation of two real finite energy sequences x(n) and y(n) is given as:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-correlation-discrete-time-signals-q2.png)d) None of the mentioned  
View Answer

Answer: a  
Explanation: If any two signals x(n) and y(n) are real and finite energy signals, then the correlation between the two signals is known as cross correlation and its equation is given as  
[digital-signal-processing-questions-answers-correlation-discrete-time-signals-q2a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-correlation-discrete-time-signals-q2a.png)

3. Which of the following relation is true?  
a) rxy(l)= rxy(-l)  
b) rxy(l)= ryx(l)  
c) rxy(l)= ryx(-l)  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: we know that, the correlation of two signals x(n) and y(n) is [digital-signal-processing-questions-answers-correlation-discrete-time-signals-q3](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-correlation-discrete-time-signals-q3.png)  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-correlation-discrete-time-signals-q3a.png)

4. What is the cross correlation sequence of the following sequences?  
x(n)={….0,0,2,-1,3,7,**1**,2,-3,0,0….}  
y(n)={….0,0,1,-1,2,-2,**4**,1,-2,5,0,0….}  
a) {10,9,19,36,-14,33,0,**7**,13,-18,16,7,5,-3}  
b) {10,-9,19,36,-14,33,0,**7**,13,-18,16,-7,5,-3}  
c) {10,9,19,36,14,33,0,**-7**,13,-18,16,-7,5,-3}  
d) {10,-9,19,36,-14,33,0,**-7**,13,18,16,7,5,-3}  
View Answer

Answer: b  
Explanation:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-correlation-discrete-time-signals-q4.png)

5. Which of the following is the auto correlation of x(n)?  
a) rxy(l)=x(l)\*x(-l)  
b) rxy(l)=x(l)\*x(l)  
c) rxy(l)=x(l)+x(-l)  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: We know that, the correlation of two signals x(n) and y(n) is [digital-signal-processing-questions-answers-correlation-discrete-time-signals-q3](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-correlation-discrete-time-signals-q3.png)  
Let x(n)=y(n)=> [digital-signal-processing-questions-answers-correlation-discrete-time-signals-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-correlation-discrete-time-signals-q5.png)

6. What is the energy sequence of the signal ax(n)+by(n-l) in terms of cross correlation and auto correlation sequences?  
a) a2rxx(0)+b2ryy(0)+2abrxy(0)  
b) a2rxx(0)+b2ryy(0)-2abrxy(0)  
c) a2rxx(0)+b2ryy(0)+2abrxy(1)  
d) a2rxx(0)+b2ryy(0)-2abrxy(1)  
View Answer

Answer: c  
Explanation:  
The energy signal of the signal ax(n)+by(n-l) is  
[digital-signal-processing-questions-answers-correlation-discrete-time-signals-q6](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-correlation-discrete-time-signals-q6.png)

7. What is the relation between cross correlation and auto correlation?  
a) |rxy(l)|=√(rxx(0).ryy(0))  
b) |rxy(l)|≥√(rxx(0).ryy(0))  
c) |rxy(l)|≠√(rxx(0).ryy(0))  
d) |rxy(l)|≤√(rxx(0).ryy(0))  
View Answer

Answer: d  
Explanation:  
We know that, a2rxx(0)+b2ryy(0)+2abrxy(l) ≥0  
=> (a/b)2rxx(0)+ryy(0)+2(a/b)rxy(l) ≥0  
Since the quadratic is nonnegative, it follows that the discriminate of this quadratic must be non positive, that is 4[r2xy(l)- rxx(0) ryy(0)] ≤0 =>|rxy(l)|≤√(rxx(0).ryy(0)).

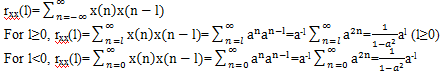
8. The normalized auto correlation ρxx(l) is defined as:  
a) (rxx (l))/(rxx (0))  
b) -(rxx (l))/(rxx (0))  
c) (rxx (l))/(rxy (0))  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: If the signal involved in auto correlation is scaled, the shape of auto correlation does not change, only the amplitudes of auto correlation sequence are scaled accordingly. Since scaling is unimportant, it is often desirable, in practice, to normalize the auto correlation sequence to the range from -1 to 1. In the case of auto correlation sequence, we can simply divide by rxx (0). Thus the normalized auto correlation sequence is defined as ρxx(l)= (rxx (l))/(rxx (0)).

9. Auto correlation sequence is an even function.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Let us consider a signal x(n) whose auto correlation is defined as rxx (l).  
We know that, for auto correlation sequence rxx (l)=rxx (-l).  
So, auto correlation sequence is an even sequence.

10. What is the auto correlation of the sequence x(n)=anu(n), 0<a<l?  
a) 1/(1-a2 )al (l≥0)  
b) 1/(1-a2 )a-l (l<0)  
c) 1/(1-a2 )a|l|(-∞<l<∞)  
d) All of the mentioned  
View Answer

Answer: d  
Explanation:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-correlation-discrete-time-signals-q10.png)

11. Which of the following relation is true?  
a) ryx(l)=h(l)\*ryy(l)  
b) rxy(l)=h(l)\*rxx(l)  
c) ryx(l)=h(l)\*rxx(l)  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: Let x(n) be the input signal and y(n) be the output signal with impulse response h(n).  
We know that [digital-signal-processing-questions-answers-correlation-discrete-time-signals-q11](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-correlation-discrete-time-signals-q11.png)  
The cross correlation between the input signal and output signal is  
ryx(l)=y(l)\*x(-l)=h(l)\*[x(l)\*x(-l)]= h(l)\*rxx(l).

12. If x(n) is the input signal of a system with impulse response h(n) and y(n) is the output signal, then the auto correlation of the signal y(n) is:  
a) rxx(l)\*rhh(l)  
b) rhh(l)\*rxx(l)  
c) rxy(l)\*rhh(l)  
d) ryx(l)\*rhh(l)  
View Answer

Answer: b  
Explanation: ryy(l)=y(l)\*y(-l)  
=[h(l)\*x(l)]\*[h(-l)\*x(-l)] =[h(l)\*h(-l)]\*[x(l)\*x(-l)] =rhh(l)\*rxx(l).

**2. Questions on DSP – Basic Signaling**

The section contains questions on A/D and D/A converters, signal classification and signal and system processing.

|  |  |
| --- | --- |
| [A2D and D2A Converters](https://www.sanfoundry.com/digital-signal-processing-questions-answers-a2d-d2a-converters/) [Signal Classification](https://www.sanfoundry.com/digital-signal-processing-questions-answers-classification-signals/) | [Signal and System Processing](https://www.sanfoundry.com/digital-signal-processing-questions-answers-signals-systems-signal-processing/) |

Questions & Answers (MCQs) focuses on “A2D and D2A Converters”.

1. Which of the following should be done in order to convert a continuous-time signal to a discrete-time signal?  
a) Sampling  
b) Differentiating  
c) Integrating  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: The process of converting a continuous-time signal into a discrete-time signal by taking samples of continuous time signal at discrete time instants is known as ‘sampling’.

2. The process of converting discrete-time continuous valued signal into discrete-time discrete valued(digital) signal is known as:  
a) Sampling  
b) Quantization  
c) Coding  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: In this process, the value of each signal sample is represented by a value selected from a finite set of possible values. Hence this process is known as ‘quantization’

3. The difference between the unquantized x(n) and quantized xq(n) is known as:  
a) Quantization coefficient  
b) Quantization ratio  
c) Quantization factor  
d) Quantization error  
View Answer

Answer: d  
Explanation: Quantization error is the difference in the signal obtained after sampling i.e., x(n) and the signal obtained after quantization i.e., xq(n) at any instant of time.

4. Which of the following is a digital-to-analog conversion process?  
a) Staircase approximation  
b) Linear interpolation  
c) Quadratic interpolation  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: The process of joining in terms of steps is known as staircase approximation, connecting two samples by a straight line is known as Linear interpolation, connecting three samples by fitting a quadratic curve is called as Quadratic interpolation.

5. The relation between analog frequency ‘F’ and digital frequency ‘f’ is:  
a) F=f\*T(where T is sampling period)  
b) f=F\*T  
c) No relation  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: Consider an analog signal of frequency ‘F’, which when sampled periodically at a rate Fs=1/T samples per second yields a frequency of f=F/Fs=>f=F\*T.

6. What is output signal when a signal x(t)=cos(2\*pi\*40\*t) is sampled with a sampling frequency of 20Hz?  
a) cos(pi\*n)  
b) cos(2\*pi\*n)  
c) cos(4\*pi\*n)  
d) cos(8\*pi\*n)  
View Answer

Answer: c  
Explanation: From the question F=40Hz, Fs=20Hz  
=>f=F/Fs  
=>f=40/20  
=>f=2Hz  
=>x(n)=cos(4\*pi\*n).

7. If ‘F’ is the frequency of the analog signal, then what is the minimum sampling rate required to avoid aliasing?  
a) F  
b) 2F  
c) 3F  
d) 4F  
View Answer

Answer: a  
Explanation: According to Nyquist rate, to avoid aliasing the sampling frequency should be equal to twice of the analog frequency.

8. What is the nyquist rate of the signal x(t)=3cos(50\*pi\*t)+10sin(300\*pi\*t)-cos(100\*pi\*t)?  
a) 50Hz  
b) 100Hz  
c) 200Hz  
d) 300Hz  
View Answer

Answer: d  
Explanation: The frequencies present in the given signal are F1=25Hz, F2=150Hz, F3=50Hz  
Thus Fmax=150Hz and from the sampling theorem,  
nyquist rate=2\*Fmax  
Therefore, Fs=2\*150=300Hz.

9. What is the discrete-time signal obtained after sampling the analog signal x(t)=cos(2000\*pi\*t)+sin(5000\*pi\*t) at a sampling rate of 5000samples/sec?  
a) cos(2.5\*pi\*n)+sin(pi\*n)  
b) cos(0.4\*pi\*n)+sin(pi\*n)  
c) cos(2000\*pi\*n)+sin(5000\*pi\*n)  
d) none of the mentioned  
View Answer

Answer: b  
Explanation: From the given analog signal, F1=1000Hz F2=2500Hz and Fs=5000Hz  
=>f1=F1/Fs and f2=F2/Fs  
=>f1=0.2 and f2=0.5  
=>x(n)= cos(0.4\*pi\*n)+sin(pi\*n).

10. If the sampling rate Fs satisfies the sampling theorem, then the relation between quantization errors of analog signal(eq(t)) and discrete-time signal(eq(n)) is:  
a) eq(t)=eq(n)  
b) eq(t)<eq(n)  
c) eq(t)>eq(n)  
d) not related  
View Answer

Answer: a  
Explanation: If it obeys sampling theorem, then the only error in A/D conversion is quantization error. So, the error is same for both analog and discrete-time signal.

11. The quality of output signal from a A/D converter is measured in terms of:  
a) Quantization error  
b) Quantization to signal noise ratio  
c) Signal to quantization noise ratio  
d) Conversion constant  
View Answer

Answer: c  
Explanation: The quality is measured by taking the ratio of noises of input signal and the quantized signal i.e., SQNR and is measured in terms of dB.

12. Which bit coder is required to code a signal with 16 levels?  
a) 8 bit  
b) 4 bit  
c) 2 bit  
d) 1 bit  
View Answer

Answer: b  
Explanation: To code a signal with L number of levels, we require a coder with (log L/log 2) number of bits. So, log16/log2=4 bit coder is required.

Questions & Answers (MCQs) focuses on “Classification of Signals”.

1. Which of the following is done to convert a continuous time signal into discrete time signal?  
a) Modulating  
b) Sampling  
c) Differentiating  
d) Integrating  
View Answer

Answer: b  
Explanation: A discrete time signal can be obtained from a continuous time signal by replacing t by nT, where T is the reciprocal of the sampling rate or time interval between the adjacent values. This procedure is known as sampling.

2. The deflection voltage of an oscilloscope is a ‘deterministic’  
signal. True or False?  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The behavior of the signal is known and can be represented by a saw tooth wave form. So, the signal is deterministic.

3. The even part of a signal x(t) is:  
a) x(t)+x(-t)  
b) x(t)-x(-t)  
c) (1/2)\*(x(t)+x(-t))  
d) (1/2)\*(x(t)-x(-t))  
View Answer

Answer: c  
Explanation: Let x(t)=xe(t)+xo(t)  
=>x(-t)=xe(-t)-xo(-t)  
By adding the above two equations, we get  
xe(t)=(1/2)\*(x(t)+x(-t)).

4. Which of the following is the odd component of the signal x(t)=e(jt)?  
a) cost  
b) j\*sint  
c) j\*cost  
d) sint  
View Answer

Answer: b  
Explanation: Let x(t)=e(jt)  
Now, xo(t)=(1/2)\*(x(t)-x(-t))  
=(1/2)\*(e(jt) – e(-jt))  
=(1/2)\*(cost+jsint-cost+jsint)  
=(1/2)\*(2jsint)  
=j\*sint.

5. For a continuous time signal x(t) to be periodic with a period T, then x(t+mT) should be equal to:  
a) x(-t)  
b) x(mT)  
c) x(mt)  
d) x(t)  
View Answer

Answer: d  
Explanation: If a signal x(t) is said to be periodic with period T, then x(t+mT)=x(t) for all t and any integer m.

6. Let x1(t) and x2(t) be periodic signals with fundamental periods T1 and T2 respectively. Which of the following must be a rational number for x(t)=x1(t)+x2(t) to be periodic?  
a) T1+T2  
b) T1-T2  
c) T1/T2  
d) T1\*T2  
View Answer

Answer: c  
Explanation: Let T be the period of the signal x(t)  
=>x(t+T)=x1(t+mT1)+x2(t+nT2)  
Thus, we must have  
mT1=nT2=T  
=>(T1/T2)=(k/m)= a rational number.

7. Let x1(t) and x2(t) be periodic signals with fundamental periods T1 and T2 respectively. Then the fundamental period of x(t)=x1(t)+x2(t) is:  
a) LCM of T1 and T2  
b) HCF of T1and T2  
c) Product of T1 and T2  
d) Ratio of T1 to T2  
View Answer

Answer: a  
Explanation: For the sum of x1(t) and x2(t) to be periodic the ratio of their periods should be a rational number, then the fundamental period is the LCM of T1 and T2.

8. All energy signals will have an average power of:  
a) Infinite  
b) Zero  
c) Positive  
d) Cannot be calculated  
View Answer

Answer: b  
Explanation: For any energy signal, the average power should be equal to 0 i.e., P=0.

9. x(t) or x(n) is defined to be an energy signal, if and only if the total energy content of the signal is a:  
a) Finite quantity  
b) Infinite  
c) Zero  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: The energy signal should have total energy value that lies between 0 and infinity.

10. What is the period of cos2t+sin3t?  
a) pi  
b) 2\*pi  
c) 3\*pi  
d) 4\*pi  
View Answer

Answer: b  
Explanation: Period of cos2t=(2\*pi)/2=pi  
Period of sin3t=(2\*pi)/3  
LCM of pi and (2\*pi)/3 is 2\*pi.

Questions & Answers (MCQs) focuses on “signals,Systems ans Signal Processing”.

1. Which of the following is common independent variable for speech signal, EEG and ECG?  
a) Time  
b) Spatial coordinates  
c) Pressure  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: Speech, EEG and ECG signals are the examples of information-bearing signals that evolve as functions of a single independent variable, namely, time.

2. Which of the following conditions made digital signal processing more advantageous over analog signal processing?  
a) Flexibility  
b) Accuracy  
c) Storage  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: Digital programmable system allows flexibility in reconfiguring the DSP operations by just changing the program, as the digital signal is in the form of 1 and 0’s it is more accurate and it can be stored in magnetic tapes.

3. Which property does y(t)=x(1-t) exhibit?  
a) Time scaling  
b) Time shifting  
c) Reflecting  
d) Time shifting and reflecting  
View Answer

Answer: d  
Explanation: First the signal x(t) is shifted by 1 to get x(1+t) and it is reflected to get x(1-t). So, it exhibits both time shifting and reflecting properties.

4. If x(n)=(0,1,2,3,3,0,0,0) then x(2n) is:  
a) (0,2,4,6,6,0,0,0)  
b) (0,1,2,3,3,0,0,0)  
c) (0,2,3,0,0,0,0,0)  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: Substitute n=0,1,2… in x(2n) and obtain the values from the given x(n).

5. If x(n)=(0,0,1,2,3,4,0,0) then x(n-2) is:  
a) (0,0,2,4,6,8,0,0)  
b) (0,0,1,2,3,4,0,0)  
c) (1,2,3,4,0,0,0,0)  
d) (0,0,0,0,1,2,3,4)  
View Answer

Answer: d  
Explanation: The signal x(n) is shifted right by 2.

6. If x(n)=(0,0,1,1,1,1,1,0) then x(3n+1) is:  
a) (0,1,0,0,0,0,0,0)  
b) (0,0,1,1,1,1,0,0)  
c) (1,1,0,0,0,0,0,0)  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: First shift the given signal left by 1 and then time scale the obtained signal by 3.

7. If a signal x(t) is processed through a system to obtain the signal (x(t)2), then the system is said to be:  
a) Linear  
b) Non-linear  
c) Exponential  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: Let the input signal be ‘t’. Then the output signal after passing through the system is y=t2 which is the equation of a parabola. So, the system is non-linear.

8. What are the important block(s) required to process an input analog signal to get an output analog signal?  
a) A/D converter  
b) Digital signal processor  
c) D/A converter  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: The input analog signal is converted into digital using A/D converter and passed through DSP and then converted back to analog using D/A converter.

9. Which of the following block is not required in digital processing of an RADAR signal?  
a) A/D converter  
b) D/A converter  
c) DSP  
d) All of the mentioned  
View Answer

Answer: b  
Explanation: In the digital processing of the radar signal, the information extracted from the radar signal, such as the position of the aircraft and its speed, may simply be printed on a paper. So, there is no need of an D/A converter in this case.

10. Which of the following wave is known as “amplitude modulated wave” of x(t)?  
a) C.x(t) (where C is a constant)  
b) x(t)+y(t)  
c) x(t).y(t)  
d) dx(t)/dt  
View Answer

Answer: c  
Explanation: The multiplicative operation is often encountered in analog communication, where an audio frequency signal is multiplied by a high frequency sinusoid known as carrier. The resulting signal is known as “amplitude modulated wave”.

11. What is the physical device that performs an operation on the signal?  
a) Signal source  
b) System  
c) Medium  
d) None of the mentioned  
View Answer

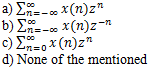
Answer: b  
Explanation: A system is a physical device which performs the operation on the signal and modifies the input signal.

#### 3. Questions & Answers on Z transform and its Application – Analysis of the LTI Systems

The section contains questions and answers on Z transforms and its properties, types of Z transforms which include rational, inverse and one sided Z transform and their analysis.

|  |  |
| --- | --- |
| [Z Transform](https://www.sanfoundry.com/digital-signal-processing-questions-answers-z-transform/) [Z Transform Properties-1](https://www.sanfoundry.com/digital-signal-processing-questions-answers-properties-z-transform-1/) [Z Transform Properties-2](https://www.sanfoundry.com/tough-digital-signal-processing-questions-answers/) [Rational Z Transform](https://www.sanfoundry.com/digital-signal-processing-questions-answers-rational-z-transform/) | [Z Transform Inversion](https://www.sanfoundry.com/digital-signal-processing-questions-answers-inverse-z-transform/) [One Sided Z Transform](https://www.sanfoundry.com/digital-signal-processing-questions-answers-one-sided-z-transform/) [Z Domain System Analysis](https://www.sanfoundry.com/digital-signal-processing-questions-answers-analysis-lti-system-z-domain/) |

Questions & Answers (MCQs) focuses on “Z Transform”.

1. The Z-Transform X(z) of a discrete time signal x(n) is defined as:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-z-transform-q1.png)  
View Answer

Answer: b  
Explanation: The z-transform of a real discrete time sequence x(n) is defined as a power of ‘z’ which is equal to [digital-signal-processing-questions-answers-z-transform-q1a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-z-transform-q1a.png), where ‘z’ is a complex variable.

2. What is the set of all values of z for which X(z) attains a finite value?  
a) Radius of convergence  
b) Radius of divergence  
c) Feasible solution  
d) None of the mentioned  
View Answer

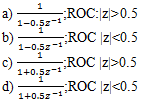
Answer: a  
Explanation: Since X(z) is a infinite power series, it is defined only at few values of z. The set of all values of z where X(z) converges to a finite value is called as Radius of Convergence(ROC).

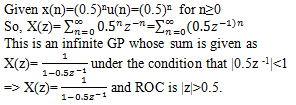
3. What is the z-transform of the finite duration signal  
[digital-signal-processing-questions-answers-z-transform-q3](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-z-transform-q3.png)  
a) 2 + 4z + 5z2 + 7z3 + z4  
b) 2 + 4z + 5z2 + 7z3 + z5  
c) 2 + 4z-1 + 5z-2 + 7z-3 + z-5  
d) 2z2 + 4z + 5 +7z-1 + z-3  
View Answer

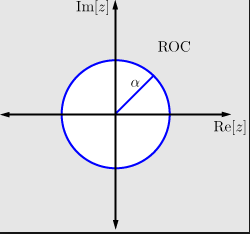
Answer: d  
Explanation: We know that, for a given signal x(n) the z-transform is defined as [digital-signal-processing-questions-answers-z-transform-q1a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-z-transform-q1a.png)  
Substitute the values of n from -2 to 3 and the corresponding signal values in the above formula  
We get, X(z) = 2z2 + 4z + 5 +7z-1 + z-3.

4. What is the ROC of the signal x(n)=δ(n-k),k>0?  
a) z=0  
b) z=∞  
c) Entire z-plane, except at z=0  
d) Entire z-plane, except at z=∞  
View Answer

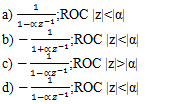
Answer: c  
Explanation: We know that, the z-transform of a signal x(n) is [digital-signal-processing-questions-answers-z-transform-q1a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-z-transform-q1a.png)  
Given x(n)= δ(n-k)=1 at n=k  
=> X(z)=z-k  
From the above equation, X(z) is defined at all values of z except at z=0 for k>0.  
So ROC is defined as Entire z-plane, except at z=0.

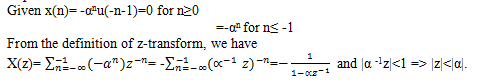
5. What is the z-transform of the signal x(n)=(0.5)nu(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-z-transform-q5.png)  
View Answer

Answer: a  
Explanation: For a given signal x(n), its z-transform [digital-signal-processing-questions-answers-z-transform-q1a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-z-transform-q1a.png)  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-z-transform-q5a.png)

6. Which of the following series has an ROC as mentioned below?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-z-transform-q6.png)  
a) α-nu(n)  
b) αnu(n)  
c) α-nu(-n)  
d) αnu(n)  
View Answer

Answer: b  
Explanation:  
Let x(n)= αnu(n)  
[digital-signal-processing-questions-answers-z-transform-q6a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-z-transform-q6a.png)

7. What is the z-transform of the signal x(n)= -αnu(-n-1)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-z-transform-q7.png)  
View Answer

Answer: d  
Explanation:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-z-transform-q7a.png)

8. What is the ROC of the z-transform of the signal x(n)= anu(n)+bnu(-n-1)?

a) |a|<|z|<|b|

b) |a|>|z|>|b|

c) |a|>|z|<|b|

d) |a|<|z|>|b|

View Answer

Answer: a  
Explanation: We know that,

ROC of z-transform of a<sup>n</sup>u(n) is |z|>|a|.

ROC of z-transform of b<sup>n</sup>u(-n-1) is |z|<|b|.

By combining both the ROC's we get the ROC of z-transform of the signal x(n) as |a|<|z|<|b|.

9. What is the ROC of z-transform of finite duration anti-causal sequence?  
a) z=0  
b) z=∞  
c) Entire z-plane, except at z=0  
d) Entire z-plane, except at z=∞  
View Answer

Answer: d  
Explanation: Let us an example of anti causal sequence whose z-transform will be in the form X(z)=1+z+z2 which has a finite value at all values of ‘z’ except at z=∞.So, ROC of an anti-causal sequence is entire z-plane except at z=∞.

10. What is the ROC of z-transform of an two sided infinite sequence?

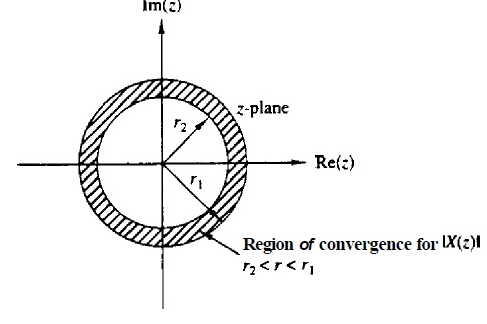
a) |z|>r1

b) |z|<r1

c) r2<|z|<r1

d) None of the mentioned

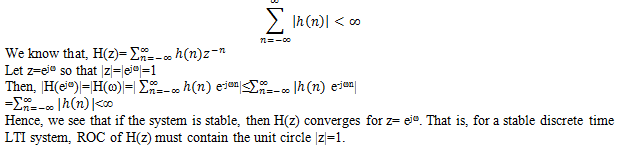
View Answer

Answer: c  
Explanation: Let us plot the graph of z-transform of any two sided sequence which looks as follows.  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-z-transform-q10.png)  
From the above graph, we can state that the ROC of a two sided sequence will be of the form r2 < |z| < r1.

11. The z-transform of a sequence x(n) which is given as [digital-signal-processing-questions-answers-z-transform-q1a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-z-transform-q1a.png)is known as:  
a) Uni-lateral Z-transform  
b) Bi-lateral Z-transform  
c) Tri-lateral Z-transform  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The entire timing sequence is divided into two parts n=0 to ∞ and n=-∞ to 0.  
Since the z-transform of the signal given in the questions contains both the parts, it is called as Bi-lateral z-transform.

12. What is the ROC of the system function H(z) if the discrete time LTI system is BIBO stable?  
a) Entire z-plane, except at z=0  
b) Entire z-plane, except at z=∞  
c) Contain unit circle  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: A discrete time LTI is BIBO stable, if and only if its impulse response h(n) is absolutely summable. That is,  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-z-transform-q12.png)

13. The ROC of z-transform of any signal cannot contain poles.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Since the value of z-transform tends to infinity, the ROC of the z-transform does not contain poles.

14. Is the discrete time LTI system with impulse response h(n)=an(n) (|a| < 1) BIBO stable?  
a) True  
b) False  
View Answer

Answer: a  
Explanation:

Given h(n)= a<sup>n</sup>(n) (|a|<1)

The z-transform of h(n) is H(z)=z/(z-a),ROC is |z|>|a|

If |a|<1, then the ROC contains the unit circle. So, the system is BIBO stable.

15. What is the ROC of a causal infinite length sequence?

a) |z|<r1

b) |z|>r1

c) r2<|z|<r1

d) None of the mentioned

View Answer

Answer: b  
Explanation: The ROC of causal infinite sequence is of form |z|>r1 where r1 is largest magnitude of poles.

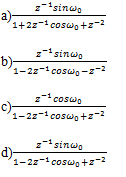
Questions & Answers (MCQs) focuses on “Properties of Z Transform-1”.

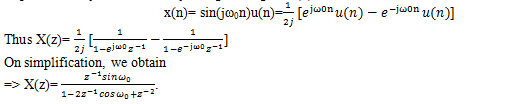
1. Which of the following justifies the linearity property of z-transform?[x(n)↔X(z)].  
a) x(n)+y(n) ↔X(z)Y(z)  
b) x(n)+y(n) ↔X(z)+Y(z)  
c) x(n)y(n) ↔X(z)+Y(z)  
d) x(n)y(n) ↔X(z)Y(z)  
View Answer

Answer: b  
Explanation: According to the linearity property of z-transform, if X(z) and Y(z) are the z-transforms of x(n) and y(n) respectively then, the z-transform of x(n)+y(n) is X(z)+Y(z).

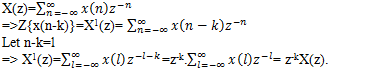
2. What is the z-transform of the signal x(n)=[3(2n)-4(3n)]u(n)?  
a) 3/(1-2z-1)-4/(1-3z-1)  
b) 3/(1+2z-1)-4/(1+3z-1)  
c) 3/(1-2z)-4/(1-3z)  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: Let us divide the given x(n) into x1(n)= 3(2n)u(n) and x2(n)= 4(3n)u(n)  
and x(n)=x1(n)-x2(n)  
From the definition of z-transform X1(z)= 3/(1-2z-1) and X2(z)= 4/(1-3z-1)  
So, from the linearity property of z-transform  
X(z)=X1(z)-X2(z)  
=> X(z)= 3/(1-2z-1)-4/(1-3z-1).

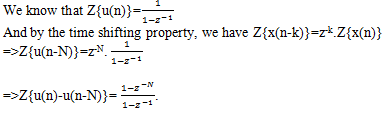
3. What is the z-transform of the signal x(n)=sin(jω0n)u(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-z-transform-1-q3.png)  
View Answer

Answer: d  
Explanation: By Euler’s identity, the given signal x(n) can be written as  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-z-transform-1-q3a.png)

4. According to Time shifting property of z-transform, if X(z) is the z-transform of x(n) then what is the z-transform of x(n-k)?  
a) zkX(z)  
b) z-kX(z)  
c) X(z-k)  
d) X(z+k)  
View Answer

Answer: b  
Explanation: According to the definition of Z-transform  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-z-transform-1-q4.png)

5. What is the z-transform of the signal defined as x(n)=u(n)-u(n-N)?  
[digital-signal-processing-questions-answers-properties-z-transform-1-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-z-transform-1-q5.png)  
View Answer

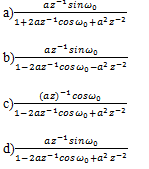
Answer: d  
Explanation:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-z-transform-1-q5a.png)

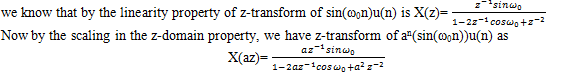
6. If X(z) is the z-transform of the signal x(n) then what is the z-transform of anx(n)?  
a) X(az)  
b) X(az-1)  
c) X(a-1z)  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: We know that from the definition of z-transform  
[digital-signal-processing-questions-answers-properties-z-transform-1-q6](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-z-transform-1-q6.png)

7. If the ROC of X(z) is r1<|z|<r2, then what is the ROC of X(a-1z)?  
a) |a|r1<|z|<|a|r2  
b) |a|r1>|z|>|a|r2  
c) |a|r1<|z|>|a|r2  
d) |a|r1>|z|<|a|r2  
View Answer

Answer: a  
Explanation: Given ROC of X(z) is r1<|z|<r2  
Then ROC of X(a-1z) will be given by r1<|a-1z |<r2=|a|r1<|z|<|a|r2

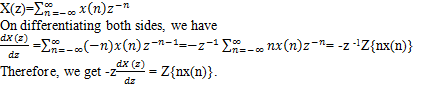
8. What is the z-transform of the signal x(n)=an(sinω0n)u(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-z-transform-1-q8.png)  
View Answer

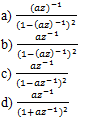
Answer: d  
Explanation:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-z-transform-1-q8a.png)

9. If X(z) is the z-transform of the signal x(n), then what is the z-transform of the signal x(-n)?  
a) X(-z)  
b) X(z-1)  
c) X-1(z)  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: From the definition of z-transform, we have  
[digital-signal-processing-questions-answers-properties-z-transform-1-q9](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-z-transform-1-q9.png)

10. X(z) is the z-transform of the signal x(n), then what is the z-transform of the signal nx(n)?  
a) -z(dX(z))/dz  
b) zdX(z)/dz  
c) -z-1dX(z)/dz  
d) z-1(dX(z))/dz  
View Answer

Answer: a  
Explanation:  
From the definition of z-transform, we have  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-z-transform-1-q10.png)

11. What is the z-transform of the signal x(n)=nanu(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-z-transform-1-q11.png)  
View Answer

Answer: c  
Explanation:  
[digital-signal-processing-questions-answers-properties-z-transform-1-q11a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-z-transform-1-q11a.png)

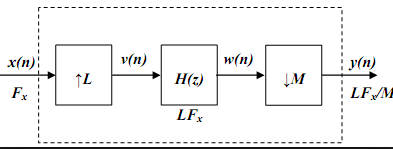
Questions & Answers focuses on “Sampling Rate Conversion by a Rational Factor I/D”.

1. Sampling rate conversion by the rational factor I/D is accomplished by what connection of interpolator and decimator?  
a) Parallel  
b) Cascade  
c) Convolution  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: A sampling rate conversion by the rational factor I/D is accomplished by cascading an interpolator with a decimator.

2. Which of the following has to be performed in sampling rate conversion by rational factor?  
a) Interpolation  
b) Decimation  
c) Either interpolation or decimation  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: We emphasize that the importance of performing the interpolation first and decimation second, is to preserve the desired spectral characteristics of x(n).

3. Which of the following operation is performed by the blocks given the figure below?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/tough-digital-signal-processing-questions-answers-q3.png)  
a) Sampling rate conversion by a factor I  
b) Sampling rate conversion by a factor D  
c) Sampling rate conversion by a factor D/I  
d) Sampling rate conversion by a factor I/D  
View Answer

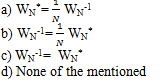
Answer: d  
Explanation: In the diagram given, a interpolator is in cascade with a decimator which together performs the action of sampling rate conversion by a factor I/D.

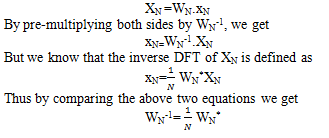
4. The Nth root of unity WN is given as:  
a) ej2πN  
b) e-j2πN  
c) e-j2π/N  
d) ej2π/N  
View Answer

Answer: c  
Explanation: We know that the Discrete Fourier transform of a signal x(n) is given as  
[tough-digital-signal-processing-questions-answers-q4](https://www.sanfoundry.com/wp-content/uploads/2015/10/tough-digital-signal-processing-questions-answers-q4.png)  
Thus we get Nth rot of unity WN= e-j2π/N

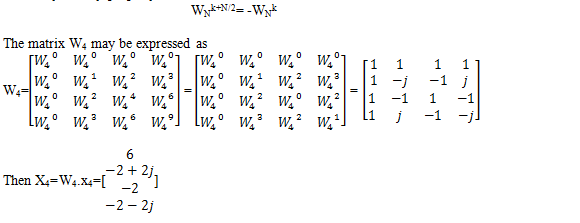
5. Which of the following is true regarding the number of computations requires to compute an N-point DFT?  
a) N2 complex multiplications and N(N-1) complex additions  
b) N2 complex additions and N(N-1) complex multiplications  
c) N2 complex multiplications and N(N+1) complex additions  
d) N2 complex additions and N(N+1) complex multiplications  
View Answer

Answer: a  
Explanation: The formula for calculating N point DFT is given as  
[tough-digital-signal-processing-questions-answers-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/tough-digital-signal-processing-questions-answers-q5.png)  
From the formula given at every step of computing we are performing N complex multiplications and N-1 complex additions. So, in a total to perform N-point DFT we perform N2 complex multiplications and N(N-1) complex additions.

6. Which of the following is true?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/tough-digital-signal-processing-questions-answers-q6.png)  
View Answer

Answer: b  
Explanation: If XN represents the N point DFT of the sequence xN in the matrix form, then we know that[](https://www.sanfoundry.com/wp-content/uploads/2015/10/tough-digital-signal-processing-questions-answers-q6a.png)

7. What is the DFT of the four point sequence x(n)={0,1,2,3}?  
a) {6,-2+2j-2,-2-2j}  
b) {6,-2-2j,2,-2+2j}  
c) {6,-2+2j,-2,-2-2j}  
d) {6,-2-2j,-2,-2+2j}  
View Answer

Answer: c  
Explanation: The first step is to determine the matrix W4. By exploiting the periodicity property of W4 and the symmetry property  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/tough-digital-signal-processing-questions-answers-q7.png)

8. If X(k) is the N point DFT of a sequence whose Fourier series coefficients is given by ck, then which of the following is true?  
a) X(k)=Nck  
b) X(k)=ck/N  
c) X(k)=N/ck  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: The Fourier series coefficients are given by the expression  
[tough-digital-signal-processing-questions-answers-q8](https://www.sanfoundry.com/wp-content/uploads/2015/10/tough-digital-signal-processing-questions-answers-q8.png)

9. What is the DFT of the four point sequence x(n)={0,1,2,3}?  
a) {6,-2+2j-2,-2-2j}  
b) {6,-2-2j,2,-2+2j}  
c) {6,-2-2j,-2,-2+2j}  
d) {6,-2+2j,-2,-2-2j}  
View Answer

Answer: d  
Answer: Given x(n)={0,1,2,3}  
We know that the 4-point DFT of the above given sequence is given by the expression  
[tough-digital-signal-processing-questions-answers-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/tough-digital-signal-processing-questions-answers-q5.png)  
In this case N=4  
=>X(0)=6,X(1)=-2+2j,X(2)=-2,X(3)=-2-2j.

10. If W4100=Wx200, then what is the value of x?  
a) 2  
b) 4  
c) 8  
d) 16  
View Answer

Answer: c  
Explanation: We know that according to the periodicity and symmetry property,  
100/4=200/x=>x=8.

Questions & Answers (MCQs) focuses on “Rational Z Transform”.

1. What are the values of z for which the value of X(z)=0?  
a) Poles  
b) Zeros  
c) Solutions  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: For a rational z-transform X(z) to be zero, the numerator of X(z) is zero and the solutions of the numerator are called as ‘zeros’ of X(z).

2. What are the values of z for which the value of X(z)=∞?  
a) Poles  
b) Zeros  
c) Solutions  
d) None of the mentioned  
View Answer

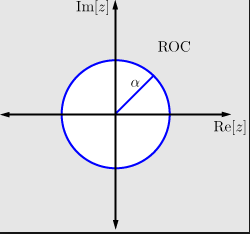
Answer: a  
Explanation: For a rational z-transform X(z) to be infinity, the denominator of X(z) is zero and the solutions of the denominator are called as ‘poles’ of X(z).

3. If X(z) has M finite zeros and N finite poles, then which of the following condition is true?  
a) |N-M| poles at origin(if N>M)  
b) |N+M| zeros at origin(if N>M)  
c) |N+M| poles at origin(if N>M)  
d) |N-M| zeros at origin(if N>M)  
View Answer

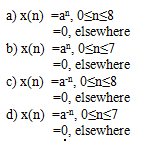
Answer: d  
Explanation: If X(z) has M finite zeros and N finite poles, then X(z) can be rewritten as X(z)=z -M+N.X'(z).  
So, if N>M then z has a positive power. So, it has |N-M| zeros at origin.

4. If X(z) has M finite zeros and N finite poles, then which of the following condition is true?  
a) |N-M| poles at origin(if N < M)  
b) |N+M| zeros at origin(if N < M)  
c) |N+M| poles at origin(if N < M)  
d) |N-M| zeros at origin(if N < M)  
View Answer

Answer: a  
Explanation: If X(z) has M finite zeros and N finite poles, then X(z) can be rewritten as X(z)=z-M+N.X'(z).  
So, if N < M then z has a negative power. So, it has |N-M| poles at origin.

5. Which of the following signals have a pole-zero plot as shown below?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-rational-z-transform-q5.png)  
a) a.u(n)  
b) u(an)  
c) anu(n)  
d) none of the mentioned  
View Answer

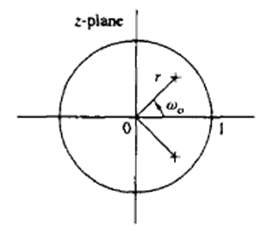
Answer: c  
Explanation: From the given pole-zero plot, the z-transform of the signal has one zero at z=0 and one pole at z=a.  
So, we obtain X(z)=z/(z-a)  
By applying inverse z-transform for X(z), we get  
x(n)= anu(n).

6. Which of the following signals have a pole-zero plot as shown below?(Let M=8 in the figure)  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-rational-z-transform-q6.png)  
View Answer

Answer: b  
Explanation: From the figure given, the z-transform of the signal has 8 zeros on circle of radius ‘a’ and 7 poles at origin.  
[digital-signal-processing-questions-answers-rational-z-transform-q6a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-rational-z-transform-q6a.png)

7. The z-transform X(z) of the signal x(n)=anu(n) has:  
a) One pole at z=0 and one zero at z=a  
b) One pole at z=0 and one zero at z=0  
c) One pole at z=a and one zero at z=a  
d) One pole at z=a and one zero at z=0  
View Answer

Answer: d  
Explanation: The z-transform of the given signal is X(z)= z/(z-a)  
So, it has one pole at z=a and one zero at z=0.

8. What is the nature of the signal whose pole-zero plot is as shown?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-rational-z-transform-q8.png)  
a) Rising signal  
b) Constant signal  
c) Decaying signal  
d) None of the mentioned  
View Answer

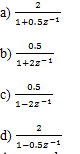
Answer: c  
Explanation: From the pole-zero plot, it is shown that r < 1, so the signal is a decaying signal.

9. What are the values of z for which the value of X(z)=0?  
a) Poles  
b) Zeros  
c) Solutions  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: For a rational z-transform X(z) to be zero, the numerator of X(z) is zero and the solutions of the numerator are called as ‘zeros’ of X(z).

10. If Y(z) is the z-transform of the output function, X(z) is the z-transform of the input function and H(z) is the z-transform of system function of the LTI system, then H(z)=?  
a) (Y(z))/(X(z))  
b) (X(z))/(Y(z))  
c) Y(z).X(z)  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: We know that for an LTI system, y(n)=h(n)\*x(n)  
On applying z-transform on both sides we get, Y(z)=H(z).X(z)=>H(z)= ( Y(z))/(X(z) ).

11. What is the system function of the system described by the difference equation y(n)=0.5y(n-1)+2x(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-rational-z-transform-q11.png)  
View Answer

Answer: d  
Explanation: Given difference equation of the system is y(n)=0.5y(n-1)+2x(n)  
On applying z-transform on both sides we get, Y(z)=0.5z-1Y(z)+2X(z)  
[digital-signal-processing-questions-answers-rational-z-transform-q11a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-rational-z-transform-q11a.png)

12. What is the unit sample response of the system described by the difference equation y(n)=0.5y(n-1)+2x(n)?  
a) 0.5(2)nu(n)  
b) 2(0.5)nu(n)  
c) 0.5(2)nu(-n)  
d) 2(0.5)nu(-n)  
View Answer

Answer: b  
Explanation: By applying the z-transform on both sides of the difference equation given in the question we obtain,  
By applying the inverse z-transform we get h(n)= 2(0.5)nu(n).

Questions & Answers (MCQs) focuses on “Inversion of Z Transform”.

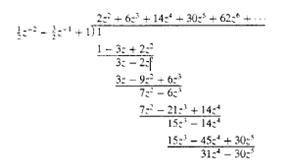
1. Which of the following method is used to find the inverse z-transform of a signal?  
a) Counter integration  
b) Expansion into a series of terms  
c) Partial fraction expansion  
d) All of the mentioned  
View Answer

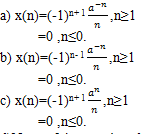
Answer: d  
Explanation: All the methods mentioned above can be used to calculate the inverse z-transform of the given signal.

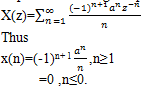
2. What is the inverse z-transform of X(z)=1/(1-1.5z-1+0.5z-2 ) if ROC is |z|>1?  
a) {1,3/2,7/4,15/8,31/16,….}  
a) {1,2/3,4/7,8/15,16/31,….}  
a) {1/2,3/4,7/8,15/16,31/32,….}  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: Since the ROC is the exterior circle, we expect x(n) to be a causal signal. Thus we seek a power series expansion in negative powers of ‘z’. By dividing the numerator of X(z) by its denominator, we obtain the power series  
[digital-signal-processing-questions-answers-inverse-z-transform-q2](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-inverse-z-transform-q2.png)  
So, we obtain x(n)= {1,3/2,7/4,15/8,31/16,….}.

3. What is the inverse z-transform of X(z)=1/(1-1.5z-1+0.5z-2 ) if ROC is |z| < 0.5?  
a) {….62,30,14,6,2}  
b) {…..62,30,14,6,2,0,0}  
c) {0,0,2,6,14,30,62…..}  
d) {2,6,14,30,62…..}  
View Answer

Answer: b  
Explanation: In this case the ROC is the interior of a circle. Consequently, the signal x(n) is anti causal. To obtain a power series expansion in positive powers of z, we perform the long division in the following way:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-inverse-z-transform-q3.png)  
Thus [digital-signal-processing-questions-answers-inverse-z-transform-q3a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-inverse-z-transform-q3a.png)  
In this case x(n)=0 for n≥0.Thus we obtain x(n)= {…..62,30,14,6,2,0,0}

4. What is the inverse z-transform of X(z)=log(1+az-1) |z|>|a|?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-inverse-z-transform-q4.png)  
d)None of the mentioned  
View Answer

Answer: c  
Explanation: Using the power series expansion for log(1+x), with |x|<1, we have  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-inverse-z-transform-q4a.png)

5. What is the proper fraction and polynomial form of the improper rational transform  
X(z)= (1+3z-1+11/6 z-2+1/3 z-3)/(1+5/6 z-1+1/6 z-2 )?  
a) 1+2z -1+(1/6 z-1)/(1+5/6 z-1+1/6 z-2 )  
b) 1-2z -1+(1/6 z-1)/(1+5/6 z-1+1/6 z-2 )  
c) 1+2z -1+(1/3 z-1)/(1+5/6 z-1+1/6 z-2)  
d) 1+2z -1-(1/6 z-1)/(1+5/6 z-1+1/6 z-2 )  
View Answer

Answer: a  
Explanation: First, we note that we should reduce the numerator so that the terms z-2 and z -3 are eliminated. Thus we should carry out the long division with these two polynomials written in the reverse order. We stop the division when the order of the remainder becomes z -1. Then we obtain  
X(z)= 1+2z -1+(1/6 z-1)/(1+5/6 z-1+1/6 z-2 ).

6. What is the partial fraction expansion of the proper function X(z)= 1/(1-1.5z-1+0.5z-2 )?  
a) 2z/(z-1)-z/(z+0.5)  
b) 2z/(z-1)+z/(z-0.5)  
c) 2z/(z-1)+z/(z+0.5)  
d) 2z/(z-1)-z/(z-0.5)  
View Answer

Answer: d  
Explanation: First we eliminate the negative powers of z by multiplying both numerator and denominator by z2.  
Thus we obtain X(z)= z2/(z2-1.5z+0.5)  
The poles of X(z) are p1=1 and p2=0.5. Consequently, the expansion will be  
(X(z))/z = z/((z-1)(z-0.5)) = 2/((z-1) ) – 1/((z-0.5) )( obtained by applying partial fractions)  
=>X(z)= 2z/(z-1)-z/(z-0.5).

7. What is the partial fraction expansion of X(z)= (1+z-1)/(1-z-1+0.5z-2 )?  
a) (z(0.5-1.5j))/(z-0.5-0.5j) – (z(0.5+1.5j))/(z-0.5+0.5j)  
b) (z(0.5-1.5j))/(z-0.5-0.5j) + (z(0.5+1.5j))/(z-0.5+0.5j)  
c) (z(0.5+1.5j))/(z-0.5-0.5j) – (z(0.5-1.5j))/(z-0.5+0.5j)  
d) (z(0.5+1.5j))/(z-0.5-0.5j) + (z(0.5-1.5j))/(z-0.5+0.5j)  
View Answer

Answer: b  
Explanation: To eliminate the negative powers of z, we multiply both numerator and denominator by z2. Thus,  
X(z)=(z(z+1))/(z-2-z+0.5)  
The poles of X(z) are complex conjugates p1=0.5+0.5j and p2=0.5-0.5j  
Consequently the expansion will be  
X(z)= (z(0.5-1.5j))/(z-0.5-0.5j) + (z(0.5+1.5j))/(z-0.5+0.5j).

8. What is the partial fraction expansion of X(z)=1/((1+z-1 )(1-z-1)2)?  
a) z/(4(z+1)) + 3z/(4(z-1)) + z/(2〖(z+1)〗2 )  
b) z/(4(z+1)) + 3z/(4(z-1)) – z/(2〖(z+1)〗2 )  
c) z/(4(z+1)) + 3z/(4(z-1)) + z/(2〖(z-1)〗2 )  
d) z/(4(z+1)) + z/(4(z-1)) + z/(2〖(z+1)〗2 )  
View Answer

Answer: c  
Explanation: First we express X(z) in terms of positive powers of z, in the form X(z)=z3/((z+1)〖(z-1)〗2 )  
X(z) has a simple pole at z=-1 and a double pole at z=1. In such a case the approximate partial fraction expansion is  
(X(z))/z = z2/((z+1)〖(z-1)〗2 ) =A/(z+1) + B/(z-1) + C/〖(z-1)〗2  
On simplifying, we get the values of A, B and C as 1/4, 3/4 and 1/2 respectively.  
Therefore, we get X(z)= z/(4(z+1)) + 3z/(4(z-1)) + z/(2〖(z-1)〗2 ) .

9. What is the inverse z-transform of X(z)= 1/(1-1.5z-1+0.5z2-2 ) if ROC is |z|>1?  
a) (2-0.5n)u(n)  
b) (2+0.5n)u(n)  
c) (2n-0.5n)u(n)  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: The partial fraction expansion for the given X(z) is  
X(z)= 2z/(z-1)-z/(z-0.5)  
In case when ROC is |z|>1, the signal x(n) is causal and both the terms in the above equation are causal terms. Thus, when we apply inverse z-transform to the above equation, we get  
x(n)=2(1)nu(n)-(0.5)nu(n)=(2-0.5n)u(n).

10. What is the inverse z-transform of X(z)= 1/(1-1.5z-1+0.5z-2 ) if ROC is |z|<0.5? a) [-2-0.5n]u(n)  
b) [-2+0.5n]u(n)  
c) [-2+0.5n]u(-n-1)  
d) [-2-0.5n]u(-n-1)  
View Answer

Answer: c  
Explanation: The partial fraction expansion for the given X(z) is  
X(z)= 2z/(z-1)-z/(z-0.5)  
In case when ROC is |z|<0.5,the signal is anti causal. Thus both the terms in the above equation are anti causal terms. So, if we apply inverse z-transform to the above equation we get x(n)= [-2+0.5n]u(-n-1).

11. What is the inverse z-transform of X(z)= 1/(1-1.5z-1+0.5z-2 ) if ROC is 0.5<|z|<1?  
a) -2u(-n-1)+(0.5)nu(n)  
b) -2u(-n-1)-(0.5)nu(n)  
c) -2u(-n-1)+(0.5)nu(-n-1)  
d) 2u(n)+(0.5)nu(-n-1)  
View Answer

Answer: b  
Explanation: The partial fraction expansion of the given X(z) is  
X(z)= 2z/(z-1)-z/(z-0.5)  
In this case ROC is 0.5<|z|<1 is a ring, which implies that the signal is two sided. Thus one of the signal corresponds to a causal signal and the other corresponds to an anti causal signal. Obviously, the ROC given is the overlapping of the regions |z|>0.5 and |z|<1. Hence the pole p2=0.5 provides the causal part and the pole p1=1 provides the anti causal part. SO, if we apply the inverse z-transform we get x(n)= -2u(-n-1)-(0.5)nu(n).

12.What is the causal signal x(n) having the z-transform X(z)= 1/((1+z-1 ) [(1-z-1)]2 )?  
a)[1/4(-1)n+3/4-n/2]u(n)  
b)[1/4(-1)n+3/4-n/2]u(-n-1)  
c)[1/4+3/4(-1)n-n/2]u(n)  
d)[1/4(-1)n+3/4+n/2]u(n)  
View Answer

Answer: d  
Explanation: The partial fraction expansion of X(z) is X(z)= z/(4(z+1)) + 3z/(4(z-1)) + z/(2[(z-1)]2)  
When we apply the inverse z-transform for the above equation, we get  
x(n)=[1/4(-1)n+3/4+n/2]u(n).

Questions & Answers (MCQs) focuses on “One Sided Z Transform”.

1. The z-transform of a signal x(n) whose definition is given by [digital-signal-processing-questions-answers-one-sided-z-transform-q1](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-one-sided-z-transform-q1.png)is known as:  
a) Unilateral z-transform  
b) Bilateral z-transform  
c) Rational z-transform  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: The z-transform of the x(n) whose definition exists in the range n=-∞ to +∞ is known as bilateral or two sided z-transform. But in the given question the value of n=0 to +∞. So, such a z-transform is known as Unilateral or one sided z-transform.

2. For what kind of signals one sided z-transform is unique?  
a) All signals  
b) Anti-causal signal  
c) Causal signal  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: One sided z-transform is unique only for causal signals, because only these signals are zero for n<0.

3. What is the one sided z-transform X+(z) of the signal x(n)={1,2,**5**,7,0,1}?  
a) z2+2z+5+7z-1+z-3  
b) 5+7z+z3  
c) z-2+2z-1+5+7z+z3  
d) 5+7z-1+z-3  
View Answer

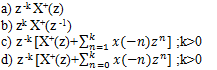
Answer: d  
Explanation: Since the one sided z-transform is valid only for n>=0, the z-transform of the given signal will be X+(z)= 5+7z-1+z-3.

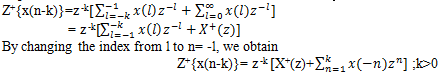
4. What is the one sided z-transform of x(n)=δ(n-k)?  
a) z-k  
b) zk  
c) 0  
d) 1  
View Answer

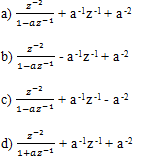
Answer: a  
Explanation: Since the signal x(n)= δ(n-k) is a causal signal i.e., it is defined for n>0 and x(n)=1 at z=k  
So, from the definition of one sided z-transform X+(z)=z-k.

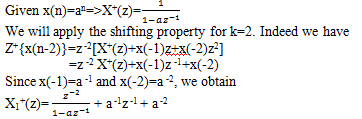
5. What is the one sided z-transform of x(n)=δ(n+k)?  
a) z-k  
b) 0  
c) zk  
d) 1  
View Answer

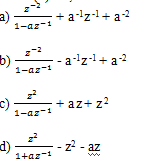
Answer: b  
Explanation: Since the signal x(n)= δ(n+k) is an anti causal signal i.e., it is defined for n<0 and x(n)=1 at z= -k. Since the one sided z-transform is defined only for causal signal, in this case X+(z)=0.

6. If X+(z) is the one sided z-transform of x(n), then what is the one sided z-transform of x(n-k)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-one-sided-z-transform-q6.png)  
View Answer

Answer: c  
Explanation: From the definition of one sided z-transform we have,  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-one-sided-z-transform-q6a.png)

7. If x(n)=an, then what is one sided z-transform of x(n-2)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-one-sided-z-transform-q7.png)  
View Answer

Answer: a  
Explanation:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-one-sided-z-transform-q7a.png)

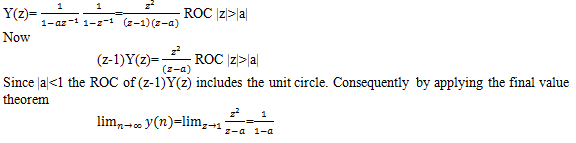
8. If x(n)=an, then what is one sided z-transform of x(n+2)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-one-sided-z-transform-q8.png)  
View Answer

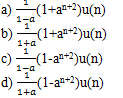
Answer: d  
Explanation: We will apply the time advance theorem with the value of k=2.We obtain,  
[digital-signal-processing-questions-answers-one-sided-z-transform-q8a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-one-sided-z-transform-q8a.png)

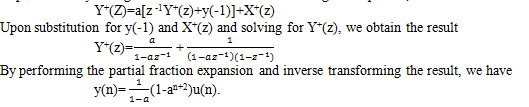
9. If X+(z) is the one sided z-transform of the signal x(n), then lim┬(n→∞)x(n)=lim┬(z→1)(z-1) X+ (z) is called Final value theorem.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: In the above theorem, we are calculating the value of x(n) at infinity, so it is called as final value theorem.

10. The impulse response of a relaxed LTI system is h(n)=anu(n),|a|<1. What is the value of the step response of the system as n→∞?  
a) 1/(1+a)  
b) 1/(1-a)  
c) a/(1+a)  
d) a/(1-a)  
View Answer

Answer: b  
Explanation: The step response of the system is y(n)=x(n)\*h(n) where x(n)=u(n)  
On applying z-transform on both sides, we get  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-one-sided-z-transform-q10.png)

11. What is the step response of the system y(n)=ay(n-1)+x(n) -1<a<1, when the initial condition is y(-1)=1?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-one-sided-z-transform-q11.png)  
View Answer

Answer: c  
Explanation: By taking the one sided z-transform of the given equation, we obtain  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-one-sided-z-transform-q11a.png)

Questions & Answers (MCQs) focuses on “Analysis of LTI System in Z Domain”.

1. What is the unit step response of the system described by the difference equation  
y(n)=0.9y(n-1)-0.81y(n-2)+x(n) under the initial conditions y(-1)=y(-2)=0?  
a) [1.099+1.088(0.9)n.cos(πn/3+5.2o)]u(n)  
b) [1.099+1.088(0.9)n.cos(πn/3-5.2o)]u(n)  
c) [1.099+1.088(0.9)n.cos(πn/3-5.2o)].  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The system function is H(z)=1/(1-0.9z-1+0.81z-2 )  
The system has two complex-conjugate poles at p1=0.9ejπ/3 and p2=0.9e -jπ/3  
The z-transform of the unit step sequence is  
X(z)=1/(1-z-1 )  
Therefore,  
Yzs(z) = 1/((1-0.9e^(jπ/3) z-1)(1-0.9e-jπ/3 z-1 )(1-z-1))  
= (0.542-j0.049)/((1-0.9ejπ/3 z-1) ) + (0.542-j0.049)/((1-0.9e^(jπ/3) z-1 ) ) + 1.099/(1-z-1 )  
and hence the zero state response is yzs(n)= [1.099+1.088(0.9)n.cos(πn/3-5.2o)]u(n)  
Since the initial conditions are zero in this case, we can conclude that y(n)= yzs(n).

2. If all the poles of H(z) are outside the unit circle, then the system is said to be:  
a) Only causal  
b) Only BIBO stable  
c) BIBO stable and causal  
d) None of the mentioned  
View Answer

Answer: d  
Explanation: If all the poles of H(z) are outside an unit circle, it means that the system is neither causal nor BIBO stable.

3. If pk, k=1,2,…N are the poles of the system and |pk| < 1 for all k, then the natural response of such a system is called as Transient response.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: If |pk| < 1 for all k, then ynr(n) decays to 0 as n approaches infinity. In such a case we refer to the natural response of the system as the transient response.

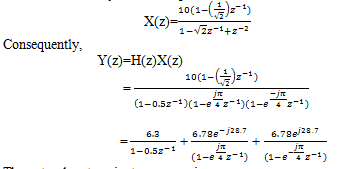
4. If all the poles have small magnitudes, then the rate of decay of signal is:  
a) Slow  
b) Constant  
c) Rapid  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: If the magnitudes of the poles of the response of any system is very small i.e., almost equal to zero, then the system decays very rapidly.

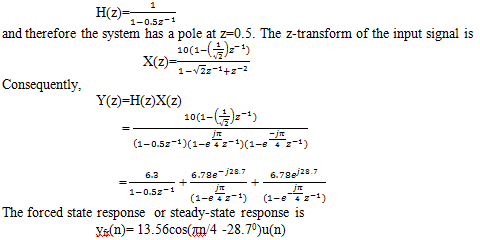
5. If one or more poles are located near the unit circle , then the rate of decay of signal is:  
a) Slow  
b) Constant  
c) Rapid  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: If the magnitudes of the poles of the response of any system is almost equal to one, then the system decays very slowly or the transient will persist for a relatively long time.

6. What is the transient response of the system described by the difference equation y(n)=0.5y(n-1)+x(n) when the input signal is x(n)= 10cos(πn/4)u(n) and the system is initially at rest?  
a) (0.5)nu(n)  
b) 0.5(6.3)nu(n)  
c) 6.3(0.5)n  
d) 6.3(0.5)nu(n)  
View Answer

Answer: d  
Explanation: The system function for the system is  
[digital-signal-processing-questions-answers-analysis-lti-system-z-domain-q6](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-lti-system-z-domain-q6.png)and therefore the system has a pole at z=0.5. The z-transform of the input signal is  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-lti-system-z-domain-q6a.png)  
The natural or transient response is  
ynr(n)= 6.3(0.5)nu(n)

7. What is the steady-state response of the system described by the difference equation y(n)=0.5y(n-1)+x(n) when the input signal is x(n)= 10cos(πn/4)u(n) and the system is initially at rest?  
a) 13.56cos(πn/4 -28.7o)  
b) 13.56cos(πn/4 +28.7o)u(n)  
c) 13.56cos(πn/4 -28.7o)u(n)  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The system function for the system is  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-lti-system-z-domain-q7.png)

8. If the ROC of the system function is the exterior of a circle of radius r < ∞, including the point z = ∞, then the system is said to be:  
a) Stable  
b) Causal  
c) Anti causal  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: A linear time invariant system is said to be causal if and only if the ROC of the system function is the exterior of a circle of radius r < ∞, including the point z = ∞.

9. A linear time invariant system is said to be BIBO stable if and only if the ROC of the system function:  
a) Includes unit circle  
b) Excludes unit circle  
c) Is an unit circle  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: For an LTI system, if the ROC of the system function includes the unit circle, then the systm is said to be BIBO stable.

10. If all the poles of H(z) are inside the unit circle, then the system is said to be:  
a) Only causal  
b) Only BIBO stable  
c) BIBO stable and causal  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: If all the poles of H(z) are inside an unit circle, then it follows the condition that |z|>r < 1, it means that the system is both causal and BIBO stable.

11. A linear time invariant system is characterized by the system function H(z)=1/(1-0.5z-1 )+2/(1-3z-1 ).What is the h(n) if the system is stable?  
a) (0.5)nu(n)-2(3)nu(n)  
b) (0.5)nu(-n-1)-2(3)nu(-n-1)  
c) (0.5)nu(-n-1)-2(3)nu(n)  
d) (0.5)nu(n)-2(3)nu(-n-1)  
View Answer

Answer: d  
Explanation: The system has poles at z=0.5 and at z=3.  
Since the system is stable, its ROC must include unit circle and hence it is 0.5<|z|<3 . Consequently, h(n) is non causal and is given as h(n)= (0.5)nu(n)-2(3)nu(-n-1).

12. A linear time invariant system is characterized by the system function H(z)=1/(1-0.5z-1 )+2/(1-3z-1 ).What is the ROC of H(z) if the system is causal?  
a) |z|<3  
b) |z|>3  
c) |z|<0.5  
d) |z|>0.5  
View Answer

Answer: b  
Explanation: The system has poles at z=0.5 and at z=3.  
Since the system is causal, its ROC is |z|>0.5 and |z|>3. The common region is |z|>3. So, ROC of given H(z) is |z|>3.

13. A linear time invariant system is characterized by the system function H(z)=1/(1-0.5z-1 )+2/(1-3z-1 ).What is the h(n) if the system is anti causal?  
a) (0.5)nu(n)+2(3)nu(n)  
b) (0.5)nu(-n-1)-2(3)nu(-n-1)  
c) -[(0.5)n+2(3)n]u(-n-1)  
d) (0.5)nu(n)-2(3)nu(-n-1)  
View Answer

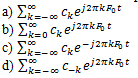
Answer: c  
Explanation: The system has poles at z=0.5 and at z=3.  
If the system is anti causal, then the ROC is |z|<0.5.Hence  
h(n)= -[(0.5)n+2(3)n]u(-n-1).

#### 4. Questions on Frequency Analysis of Signals and Systems

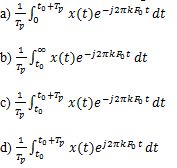
The section contains questions on frequency analysis of discrete and continuous time signals, fourier transform properties, convolution and de-convolution concepts, inverse systems, LTI systems and discrete time signals.

|  |  |
| --- | --- |
| [Continuous Time Signal Analysis](https://www.sanfoundry.com/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal/) [Discrete Time Signal Analysis-1](https://www.sanfoundry.com/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1/) [Discrete Time Signal Analysis-2](https://www.sanfoundry.com/digital-signal-processing-interview-questions-answers/) [Fourier Transforms Properties](https://www.sanfoundry.com/digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals/) | [LTI System Frequency Characteristics](https://www.sanfoundry.com/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system/) [Frequency Selective Filters](https://www.sanfoundry.com/digital-signal-processing-questions-answers-lti-system-frequency-selective-filters/) [Inverse Systems and Deconvolution](https://www.sanfoundry.com/digital-signal-processing-questions-answers-inverse-systems-deconvolution/) |

Questions & Answers (MCQs) focuses on “Frequency Analysis of Continuous Time Signals”.

1. The Fourier series representation of any signal x(t) is defined as:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q1.png)  
View Answer

Answer: a  
Explanation: If the given signal is x(t) and F0 is the reciprocal of the time period of the signal and ck is the Fourier coefficient then the Fourier series representation of x(t) is given as [digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q1a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q1a.png).

2. Which of the following is the equation for the Fourier series coefficient?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q2.png)  
View Answer

Answer: c  
Explanation: When we apply integration to the definition of Fourier series representation, we get  
[digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q2a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q2a.png)

3. Which of the following is a Dirichlet condition with respect to the signal x(t)?  
a) x(t) has a finite number of discontinuities in any period  
b) x(t) has finite number of maxima and minima during any period  
c) x(t) is absolutely integrable in any period  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: For any signal x(t) to be represented as Fourier series, it should satisfy the Dirichlet conditions which are x(t) has a finite number of discontinuities in any period, x(t) has finite number of maxima and minima during any period and x(t) is absolutely integrable in any period.

4. The equation [digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q4](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q4.png)is known as analysis equation.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: Since we are synthesizing the Fourier series of the signal x(t), we call it as synthesis equation, where as the equation giving the definition of Fourier series coefficients is known as analysis equation.

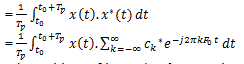
5. Which of the following is the Fourier series representation of the signal x(t)?  
[digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q5.png)d) None of the mentioned  
View Answer

Answer: b  
Explanation: In general, Fourier coefficients ck are complex valued. Moreover, it is easily shown that if the periodic signal is real, ck and c-k are complex conjugates. As a result  
ck=|ck|ejθkand ck=|ck|e-jθk  
Consequently, we obtain the Fourier series as [digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q5a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q5a.png)

6. The equation [digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q6](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q6.png)is the representation of Fourier series.  
a) True  
b) False  
View Answer

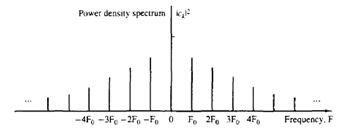
Answer: a  
Explanation: cos(2πkF0 t+θk)= cos2πkF0 t.cosθk-sin2πkF0 t.sinθk  
θk is a constant for a given signal.  
So, the other form of Fourier series representation of the signal x(t) is [digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q6a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q6a.png)

7. The equation of average power of a periodic signal x(t) is given as:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q7.png)  
View Answer

Answer: d  
Explanation: The average power of a periodic signal x(t) is given as  
[digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q7c](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q7c.png)  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q7a.png)

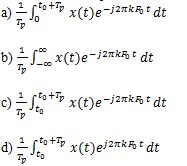
By interchanging the positions of integral and summation and by applying the integration, we get[digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q7b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q7b.png)

8. What is the spectrum that is obtained when we plot |ck |2 as a function of frequencies kF0, k=0,±1,±2..?  
a) Average power spectrum  
b) Energy spectrum  
c) Power density spectrum  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: When we plot a graph of |ck |2 as a function of frequencies kF0, k=0,±1,±2… the following spectrum is obtained which is known as Power density spectrum.  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q8.png)

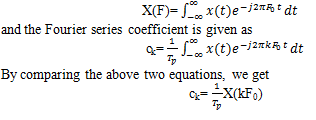
9. What is the spectrum that is obtained when we plot |ck| as a function of frequency?  
a) Magnitude voltage spectrum  
b) Phase spectrum  
c) Power spectrum  
d) None of the mentioned  
View Answer

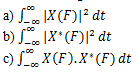
Answer: a  
Explanation: We know that, Fourier series coefficients are complex valued, so we can represent ck in the following way.  
ck=|ck|ejθk  
When we plot |ck| as a function of frequency, the spectrum thus obtained is known as Magnitude voltage spectrum.

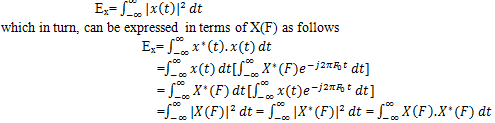
10. What is the equation of the Fourier series coefficient ck of an non-periodic signal?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q10.png)  
View Answer

Answer: b  
Explanation: We know that, for an periodic signal, the Fourier series coefficient is  
[digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q10a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q10a.png)  
If we consider a signal x(t) as non-periodic, it is true that x(t)=0 for |t|>Tp/2. Consequently, the limits on the integral in the above equation can be replaced by -∞ to ∞. Hence,  
[digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q10b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q10b.png)

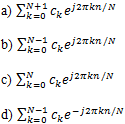
11. Which of the following relation is correct between Fourier transform X(F) and Fourier series coefficient ck?  
a) ck=X(F0/k)  
b) ck= 1/TP (X(F0/k))  
c) ck= 1/TP(X(kF0))  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: Let us consider a signal x(t) whose Fourier transform X(F) is given as  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q11.png)

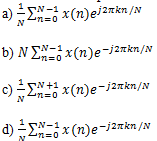
12. According to Parseval’s Theorem for non-periodic signal, [digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q12](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q12.png)  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q12a.png)d) All of the mentioned  
View Answer

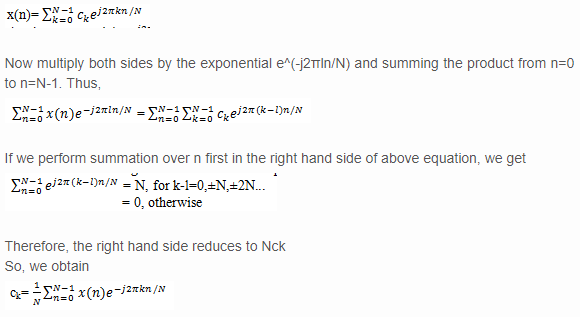
Answer: d  
Explanation: Let x(t) be any finite energy signal with Fourier transform X(F). Its energy is  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-continuous-time-signal-q12b.png)

Questions & Answers (MCQs) focuses on “Frequency Analysis of Discrete Time Signals-1”.

1. What is the Fourier series representation of a signal x(n) whose period is N?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q1.png)  
View Answer

Answer: b  
Explanation: Here, the frequency F0 of a continuous time signal is divided into 2π/N intervals.  
So, the Fourier series representation of a discrete time signal with period N is given as  
[digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q1a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q1a.png)  
where ck is the Fourier series coefficient

2. What is the expression for Fourier series coefficient ck in terms of the discrete signal x(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q2.png)  
View Answer

Answer: d  
Explanation: We know that, the Fourier series representation of a discrete signal x(n) is given as [](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q2a.png)

3. Which of the following represents the phase associated with the frequency component of discrete-time Fourier series(DTFS)?  
a) ej2πkn/N  
b) e-j2πkn/N  
c) ej2πknN  
d) None of the mentioned  
View Answer

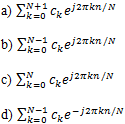
Answer: a  
Explanation: We know that, [digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q1a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q1a.png)  
In the above equation, ck represents the amplitude and ej2πkn/N represents the phase associated with the frequency component of DTFS.

4. The Fourier series for the signal x(n)=cos√2πn exists.  
a) True  
b) False  
View Answer

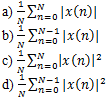
Answer: b  
Explanation: For ω0=√2π, we have f0=1/√2. Since f0 is not a rational number, the signal is not periodic. Consequently, this signal cannot be expanded in a Fourier series.

5. What are the Fourier series coefficients for the signal x(n)=cosπn/3?  
a) c1=c2=c3=c4=0,c1=c5=1/2  
b) c0=c1=c2=c3=c4=c5=0  
c) c0=c1=c2=c3=c4=c5=1/2  
d) None of the mentioned  
View Answer

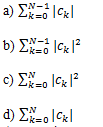
Answer: a  
Explanation: In this case, f0=1/6 and hence x(n) is periodic with fundamental period N=6.  
Given signal is x(n)= cosπn/3=cos2πn/6=1/2 e^(j2πn/6)+1/2 e^(-j2πn/6)  
We know that -2π/6=2π-2π/6=10π/6=5(2π/6)  
[digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q5.png)  
So, we get c1=c2=c3=c4=0 and c1=c5=1/2.

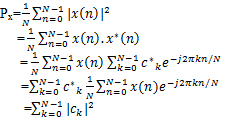
6. What is the Fourier series representation of a signal x(n) whose period is N?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q6.png)  
View Answer

Answer: b  
Explanation: Here, the frequency F0 of a continuous time signal is divided into 2π/N intervals.  
So, the Fourier series representation of a discrete time signal with period N is given as  
[digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q1a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q1a.png)  
where ck is the Fourier series coefficient

7. What is the average power of the discrete time periodic signal x(n) with period N ?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q7.png)  
View Answer

Answer: d  
Explanation: Let us consider a discrete time periodic signal x(n) with period N.  
The average power of that signal is given as  
[digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q7a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q7a.png)

8. What is the equation for average power of discrete time periodic signal x(n) with period N in terms of Fourier series coefficient ck?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q8.png)  
View Answer

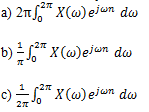
Answer: b  
Explanation: We know that[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q8a.png)

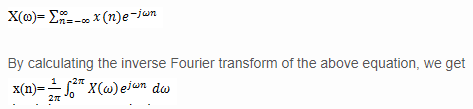
9. What is the Fourier transform X(ω) of a finite energy discrete time signal x(n)?  
[digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q9](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q9.png)  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: If we consider a signal x(n) which is discrete in nature and has finite energy, then the Fourier transform of that signal is given as

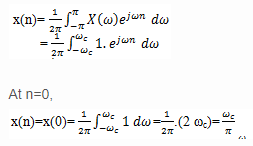
10. What is the period of the Fourier transform X(ω) of the signal x(n)?  
a) π  
b) 1  
c) Non-periodic  
d) 2π  
View Answer

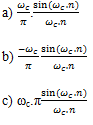
Answer: d  
Explanation: Let X(ω) be the Fourier transform of a discrete time signal x(n) which is given as  
[digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q9a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q9a.png)  
Now  
So, the Fourier transform of a discrete time finite energy signal is periodic with period 2π.

11. What is the synthesis equation of the discrete time signal x(n), whose Fourier transform is X(ω)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q11.png)d) None of the mentioned  
View Answer

Answer: c  
Explanation: We know that the Fourier transform of the discrete time signal x(n) is  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q11a.png)  
The above equation is known as synthesis equation or inverse transform equation.

12. What is the value of discrete time signal x(n) at n=0 whose Fourier transform is represented as below?  
a) ωc.π  
b) -ωc/π  
c) ωc/π  
d) None of the mentioned  
View Answer

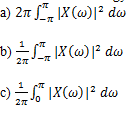
Answer: c  
Explanation: We know that, [](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q12.png)  
Therefore, the value of the signal x(n) at n=0 is ω\_c/π.

13. What is the value of discrete time signal x(n) at n≠0 whose Fourier transform is represented as below?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q13.png)d) None of the mentioned  
View Answer

Answer: a  
Explanation: We know that, [digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q13a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q13a.png)

14. The oscillatory behavior of the approximation of XN(ω) to the function X(ω) at a point of discontinuity of X(ω) is known as Gibbs phenomenon.  
a) True  
b) False  
View Answer

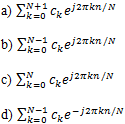
Answer: a  
Explanation: We note that there is a significant oscillatory overshoot at ω=ωc, independent of the value of N. As N increases, the oscillations become more rapid, but the size of the ripple remains the same. One can show that as N→∞, the oscillations converge to the point of the discontinuity at ω=ωc. The oscillatory behavior of the approximation of XN(ω) to the function X(ω) at a point of discontinuity of X(ω) is known as Gibbs phenomenon.

15. What is the energy of a discrete time signal in terms of X(ω)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q15.png)  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: We know that, [digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q15a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-discrete-time-signal-1-q15a.png)

Questions & Answers focuses on “Frequency Analysis of Discrete Time Signal “.

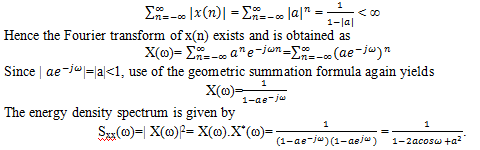
1. Which of the following relation is true if the signal x(n) is real?  
a) X\*(ω)=X(ω)  
b) X\*(ω)=X(-ω)  
c) X\*(ω)= -X(ω)  
d) None of the mentioned  
View Answer

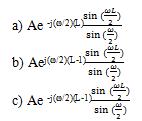
Answer: b  
Explanation: We know that,  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-interview-questions-answers-q1.png)

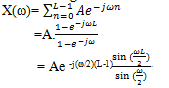
2. For a signal x(n) to exhibit even symmetry, it should satisfy the condition | X(-ω)|=| X(ω)|.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: We know that, if a signal x(n) is real, then  
X\*(ω)=X(-ω)  
If the signal is even symmetric, then the magnitude on both the sides should be equal.  
So, |X\*(ω)|=|X(-ω)| =>| X(-ω)|=| X(ω)|.

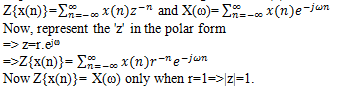
3. What is the energy density spectrum Sxx(ω) of the signal x(n)=anu(n), |a|<1? a) 1/(1+2acosω+a2 )  
b) 1/(1+2asinω+a2 )  
c) 1/(1-2asinω+a2 )  
d) 1/(1-2acosω+a2 )  
View Answer

Answer: d  
Explanation: Since |a|<1, the sequence x(n) is absolutely summable, as can be verified by applying the geometric summation formula. [](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-interview-questions-answers-q3.png)

4. What is the Fourier transform of the signal x(n) which is defined as shown in the graph below?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-interview-questions-answers-q4.png)d) None of the mentioned  
View Answer

Answer: c  
Explanation: The Fourier transform of this signal is  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-interview-questions-answers-q4a.png)

5. Which of the following condition is to be satisfied for the Fourier transform of a sequence to be equal as the Z-transform of the same sequence?  
a) |z|=1  
b) |z|<1  
c) |z|>1  
d) Can never be equal  
View Answer

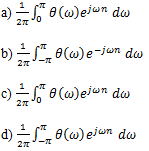
Answer: a  
Explanation: Let us consider the signal to be x(n)  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-interview-questions-answers-q5.png)

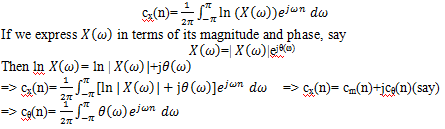
6. The sequence [digital-signal-processing-interview-questions-answers-q6](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-interview-questions-answers-q6.png)does not have both z-transform and Fourier transform.  
a) True  
b) False  
View Answer

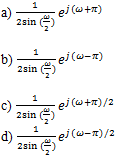
Answer: b  
Explanation: The given x(n) do not have Z-transform. But the sequence have finite energy. So, the given sequence x(n) has a Fourier transform.

7. If x(n) is a stable sequence so that X(z) converges on to a unit circle, then the complex cepstrum signal is defined as:  
a) X(ln X(z))  
b) ln X(z)  
c) X-1(ln X(z))  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: Let us consider a sequence x(n) having a z-transform X(z). We assume that x(n) is a stable sequence so that X(z) converges on to the unit circle. The complex cepstrum of the signal x(n) is defined as the sequence cx(n), which is the inverse z-transform of Cx(z), where Cx(z)=ln X(z)  
=> cx(z)= X-1(ln X(z))

8. If cx(n) is the complex cepstrum sequence obtained from the inverse Fourier transform of ln X(ω), then what is the expression for cθ(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-interview-questions-answers-q8.png)  
View Answer

Answer: d  
Explanation: We know that,  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-interview-questions-answers-q8a.png)

9. What is the Fourier transform of the signal x(n)=u(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-interview-questions-answers-q9.png)  
View Answer

Answer: d  
Explanation: Given x(n)=u(n)  
We know that the z-transform of the given signal is[digital-signal-processing-interview-questions-answers-q9a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-interview-questions-answers-q9a.png) ROC:|z|>1  
X(z) has a pole p=1 on the unit circle, but converges for |z|>1.  
If we evaluate X(z) on the unit circle except at z=1, we obtain  
[digital-signal-processing-interview-questions-answers-q9b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-interview-questions-answers-q9b.png)

10. If a power signal has its power density spectrum concentrated about zero frequency, the signal is known as:  
a) Low frequency signal  
b) Middle frequency signal  
c) High frequency signal  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: We know that, for a low frequency signal, the power signal has its power density spectrum concentrated about zero frequency.

11. What are the main characteristics of Anti aliasing filter?  
a) Ensures that bandwidth of signal to be sampled is limited to frequency range  
b) To limit the additive noise spectrum and other interference, which corrupts the signal  
c) All of the mentioned  
d) None  
View Answer

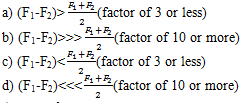
Answer: c  
Explanation: T he anti aliasing filter is an analog filter which has a twofold purpose. First, it ensures that the bandwidth of the signal to be sampled is limited to the desired frequency range. Using an anti aliasing filter is to limit the additive noise spectrum and other interference, which often corrupts the desired signal. Usually, additive noise is wide band and exceeds the bandwidth of the desired signal.

12. In general, a digital system designer has better control of tolerances in a digital signal processing system than an analog system designer who is designing an equivalent analog system.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Analog signal processing operations cannot be done very precisely either, since electronic components in analog systems have tolerances and they introduce noise during their operation. In general, a digital system designer has better control of tolerances in a digital signal processing system than an analog system designer who is designing an equivalent analog system.

13. The term ‘bandwidth’ represents the quantitative measure of a signal.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: In addition to the relatively broad frequency domain classification of signals, it is often desirable to express quantitatively the range of frequencies over which the power or energy density spectrum is concentrated. This quantitative measure is called the ‘bandwidth’ of a signal.

14. If F1 and F2 are the lower and upper cutoff frequencies of a band pass signal, then what is the condition to be satisfied to call such a band pass signal as narrow band signal?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-interview-questions-answers-q14.png)  
View Answer

Answer: d  
Explanation: If the difference in the cutoff frequencies is much less than the mean frequency, the such a band pass signal is known as narrow band signal.

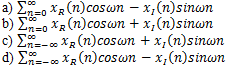
15. What is the frequency range(in Hz) of Electroencephalogram(EEG)?  
a) 10-40  
b) 1000-2000  
c) 0-100  
d) None of the mentioned  
View Answer

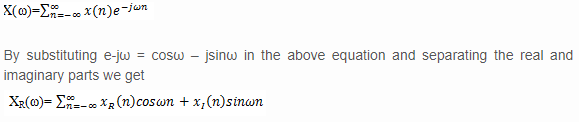
Answer: c  
Explanation: Electroencephalogram(EEG) signal has a frequency range of 0-100 Hz.

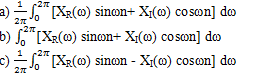
16. Which of the following electromagnetic signals has a frequency range of 30kHz-3MHz?  
a) Radio broadcast  
b) Shortwave radio signal  
c) RADAR  
d) Infrared signal  
View Answer

Answer: a  
Explanation: Radio broadcast signal is an electromagnetic signal which has a frequency range of 30kHz-3MHz.

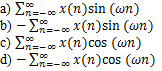
Questions & Answers (MCQs) focuses on “Properties of Fourier Transform for Discrete Time Signals”.

1. If x(n)=xR(n)+jxI(n) is a complex sequence whose Fourier transform is given as X(ω)=XR(ω)+jXI(ω), then what is the value of XR(ω)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q1.png)  
View Answer

Answer: c  
Explanation: We know that [](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q1a.png)

2. If x(n)=xR(n)+jxI(n) is a complex sequence whose Fourier transform is given as X(ω)=XR(ω)+jXI(ω), then what is the value of xI(n) ?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q2.png)d) None of the mentioned  
View Answer

Answer: a  
Explanation: We know that the inverse transform or the synthesis equation of a signal x(n) is given as  
[digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q2a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q2a.png)  
By substituting ejω = cosω + jsinω in the above equation and separating the real and imaginary parts we get  
[digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q2b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q2b.png)

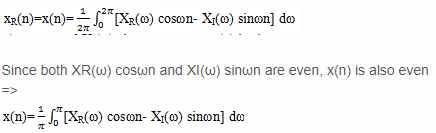
3. If x(n) is a real sequence, then what is the value of XI(ω)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q3.png)  
View Answer

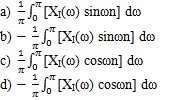
Answer: b  
Explanation: If the signal x(n) is real, then xI(n)=0  
We know that,  
[digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q3a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q3a.png)  
Now substitute xI(n)=0 in the above equation=>xR(n)=x(n)  
[digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q3b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q3b.png)

4. Which of the following relations are true if x(n) is real?  
a) X(ω)=X(-ω)  
b) X(ω)= -X(-ω)  
c) X\*(ω)=X(ω)  
d) X\*(ω)=X(-ω)  
View Answer

Answer: d  
Explanation: We know that, if x(n) is a real sequence  
[digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q4](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q4.png)  
If we combine the above two equations, we get  
X\*(ω)=X(-ω)

5. If x(n) is a real signal, then [digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q5.png)  
a) True  
b) False  
View Answer

Answer: a  
Explanation: We know that if x(n) is a real signal, then xI(n)=0 and xR(n)=x(n)  
We know that,  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q5a.png)

6. If x(n) is a real and odd sequence, then what is the expression for x(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q6.png)  
View Answer

Answer: b  
Explanation: If x(n) is real and odd then, x(n)cosωn is odd and x(n) sinωn is even. Consequently  
XR(ω)=0  
[digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q6a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q6a.png)

7. What is the value of XR(ω) given X(ω)=1/(1-ae-jω ) ,|a|<1?  
a) asinω/(1-2acosω+a2 )  
b) (1+acosω)/(1-2acosω+a2 )  
c) (1-acosω)/(1-2acosω+a2 )  
d) (-asinω)/(1-2acosω+a2 )  
View Answer

Answer: c  
Explanation: Given, X(ω)= 1/(1-ae-jω ) ,|a|<1  
By multiplying both the numerator and denominator of the above equation by the complex conjugate of the denominator, we obtain  
X(ω)= (1-aejω)/((1-ae(-jω) )(1-aejω)) = (1-acosω-jasinω)/(1-2acosω+a2 )  
This expression can be subdivided into real and imaginary parts, thus we obtain  
XR(ω)= (1-acosω)/(1-2acosω+a2 ).

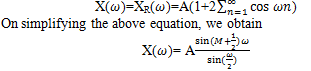
8. What is the value of XI(ω) given X(ω)=1/(1-ae-jω ) ,|a|<1?  
a) asinω/(1-2acosω+a2 )  
b) (1+acosω)/(1-2acosω+a2 )  
c) (1-acosω)/(1-2acosω+a2 )  
d) (-asinω)/(1-2acosω+a2 )  
View Answer

Answer: d  
Explanation: Given, X(ω)= 1/(1-ae-jω ) ,|a|<1  
By multiplying both the numerator and denominator of the above equation by the complex conjugate of the denominator, we obtain  
X(ω)= (1-aejω)/((1-ae(-jω) )(1-aejω)) = (1-acosω-jasinω)/(1-2acosω+a2 )  
This expression can be subdivided into real and imaginary parts, thus we obtain  
XI(ω)= (-asinω)/(1-2acosω+a2 ).

9. What is the value of |X(ω)| given X(ω)=1/(1-ae-jω ) ,|a|<1?  
a) 1/√(1-2acosω+a2 )  
b) 1/√(1+2acosω+a2)  
c) 1/(1-2acosω+a2 )  
d) 1/(1+2acosω+a2 )  
View Answer

Answer: a  
Explanation: For the given X(ω)=1/(1-ae-jω ) ,|a|<1 we obtain  
XI(ω)= (-asinω)/(1-2acosω+a2 ) and XR(ω)= (1-acosω)/(1-2acosω+a2 )  
We know that |X(ω)|=√(〖X\_R (ω)〗2+〖X\_I (ω)〗2 )  
Thus on calculating, we obtain  
|X(ω)|= 1/√(1-2acosω+a2 )

10. If x(n)=A, -M<n<M, then what is the Fourier transform of the signal?  
=0, elsewhere  
a) Asin[(M-1/2)ω]/sin(ω/2)  
b) A2 sin[(M+1/2)ω]/sin(ω/2)  
c) Asin[(M+1/2)ω]/sin[(ω/2)].  
d) sin[(M+1/2)ω]/sin(ω/2)  
View Answer

Answer: c  
Explanation: Clearly, x(n)=x(-n). Thus the signal x(n) is real and even signal. So, we know that  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q10.png)

11. What is the Fourier transform of the signal x(n)=a|n|, |a|<1?  
a) (1+a2)/(1-2acosω+a2)  
b) (1-a2)/(1-2acosω+a2)  
c) 2a/(1-2acosω+a2 )  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: First we observe x(n) can be expressed as  
x(n)=x1(n)+x2(n)  
where x1(n)= an, n>0  
=0, elsewhere

x2(n)=a-n, n<0 =0, elsewhere Now applying Fourier transform for the above two signals, we get X1(ω)= 1/(1-aejω)/((1-ae(-jω) )(1-aejω)) = (1-acosω-jasinω)/(1-2acosω+a2 )

Now, X(ω)= X1(ω)+ X2(ω)= 1/(1-ae^(-jω) )+(ae^jω)/(1-ae^jω ) = (1-a2)/(1-2acosω+a2).

12. If X(ω) is the Fourier transform of the signal x(n), then what is the Fourier transform of the signal x(n-k)?  
a) ejωk. X(-ω)  
b) ejωk. X(ω)  
c) e-jωk. X(-ω)  
d) e-jωk. X(ω)  
View Answer

Answer: d  
Explanation: Given  
[digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q12](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-fourier-transform-discrete-time-signals-q12.png)

13. What is the convolution of the sequences of x1(n)=x2(n)={1,**1**,1}?  
a) {1,2,**3**,2,1}  
b) {1,2,3,2,1}  
c) {1,1,1,1,1}  
d) {1,1,**1**,1,1}  
View Answer

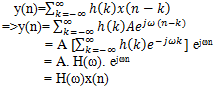
Answer: a  
Explanation: Given x1(n)=x2(n)={1,**1**,1}  
By calculating the Fourier transform of the above two signals, we get  
X1(ω)= X2(ω)=1+ ejω + e -jω = 1+2cosω  
From the convolution property of Fourier transform we have,  
X(ω)= X1(ω). X2(ω)=(1+2cosω)2=3+4cosω+2cos2ω  
By applying the inverse Fourier transform of the above signal, we get  
x1(n)\*x2(n)={1,2,**3**,2,1}

14. What is the energy density spectrum of the signal x(n)=anu(n), |a|<1?  
a) 1/(1+2acosω+a2 )  
b) 1/(1-2acosω+a2 )  
c) 1/(1-2acosω-a2 )  
d) 1/(1+2acosω-a2 )  
View Answer

Answer: b  
Explanation: Given x(n)= anu(n), |a|<1  
The auto correlation of the above signal is  
rxx(l)=1/(1-a2 ) a|l|, -∞< l <∞  
According to Wiener-Khintchine Theorem,  
Sxx(ω)=F{ rxx(l)}= [1/(1-a2)].F{a|l|} = 1/(1-2acosω+a2 )

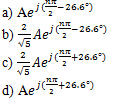
Questions & Answers (MCQs) focuses on “Frequency Domain Characteristics of LTI System”.

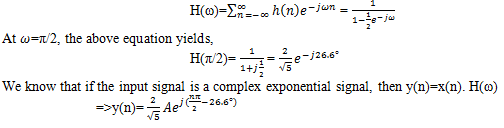
1. If x(n)=Aejωn is the input of an LTI system and h(n) is the response of the system, then what is the output y(n) of the system?  
a) H(-ω)x(n)  
b) -H(ω)x(n)  
c) H(ω)x(n)  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: If x(n)= Aejωn is the input and h(n) is the response o the system, then we know that  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q1.png)

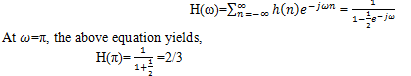
2. If the system gives an output y(n)=H(ω)x(n) with x(n)= Aejωnas input signal, then x(n) is said to be Eigen function of the system.  
a) True  
b) False  
View Answer

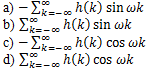
Answer: a  
Explanation: An Eigen function of a system is an input signal that produces an output that differs from the input by a constant multiplicative factor known as Eigen value of the system.

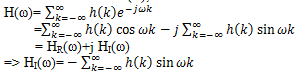
3. What is the output sequence of the system with impulse response h(n)=(1/2)nu(n) when the input of the system is the complex exponential sequence x(n)=Aejnπ/2?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q3.png)  
View Answer

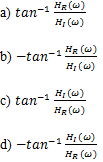
Answer: b  
Explanation: First we evaluate the Fourier transform of the impulse response of the system h(n)  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q3a.png)

4. If the Eigen function of an LTI system is x(n)= Aejnπ and the impulse response of the system is h(n)=(1/2)nu(n), then what is the Eigen value of the system?  
a) 3/2  
b) -3/2  
c) -2/3  
d) 2/3  
View Answer

Answer: d  
Explanation: First we evaluate the Fourier transform of the impulse response of the system h(n)  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q4.png)  
If the input signal is a complex exponential signal, then the input is known as Eigen function and H(ω) is called the Eigen value of the system. So, the Eigen value of the system mentioned above is 2/3.

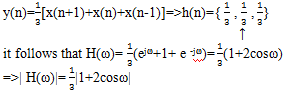
5. If h(n) is the real valued impulse response sequence of an LTI system, then what is the imaginary part of Fourier transform of the impulse response?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q5.png)  
View Answer

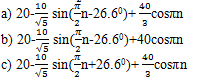
Answer: a  
Explanation: From the definition of H(ω), we have  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q5a.png)

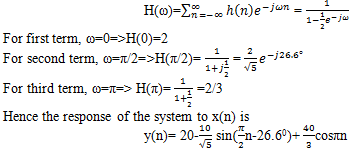
6. If h(n) is the real valued impulse response sequence of an LTI system, then what is the phase of H(ω) in terms of HR(ω) and HI(ω)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q6.png)  
View Answer

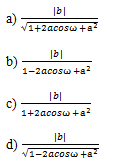
Answer: c  
Explanation: If h(n) is the real valued impulse response sequence of an LTI system, then H(ω) can be represented as HR(ω)+j HI(ω).  
=> [digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q6a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q6a.png)

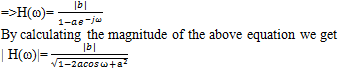
7. What is the magnitude of H(ω) for the three point moving average system whose output is given by y(n)=1/3[x(n+1)+x(n)+x(n-1)]?  
a) 1/[3|1-2cosω|].  
b) 1/[3|1+2cosω|].  
c) |1-2cosω|.  
d) |1+2cosω|.  
View Answer

Answer: b  
Explanation: For a three point moving average system, we can define the output of the system as  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q7.png)

8. What is the response of the system with impulse response h(n)=(1/2)nu(n) and the input signal x(n)=10-5sinπn/2+20cosπn?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q8.png)d) None of the mentioned  
View Answer

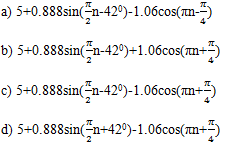
Answer: a  
Explanation: The frequency response of the system is  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q8a.png)

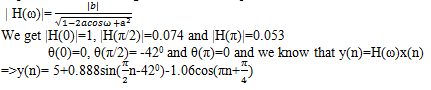
9. What is the magnitude of the frequency response of the system described by the difference equation y(n)=ay(n-1)+bx(n), 0<a<1?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q9.png)  
View Answer

Answer: d  
Explanation: Given y(n)=ay(n-1)+bx(n)  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q9a.png)

10. If an LTI system is described by the difference equation y(n)=ay(n-1)+bx(n), 0 < a < 1, then what is the parameter ‘b’ so that the maximum value of | H(ω)| is unity?  
a) a  
b) 1-a  
c) 1+a  
d) None of the mentioned  
View Answer

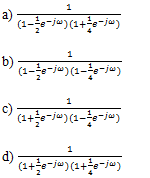
Answer: b  
Explanation: We know that,  
[digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q10](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q10.png)  
Since the parameter ‘a’ is positive, the denominator of | H(ω)| becomes minimum at ω=0. So, | H(ω)| attains its maximum value at ω=0. At this frequency we have,  
(|b|)/(1-a) =1 =>b=±(1-a).

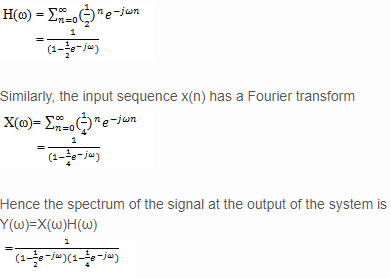
11. If an LTI system is described by the difference equation y(n)=ay(n-1)+bx(n), 0<a[[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q11.png)  
View Answer</a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q11.png)

[Answer: c  
Explanation: From the given difference equation, we obtain](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q11.png)[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q11a.png)

12. The output of the Linear time invariant system cannot contain the frequency components that are not contained in the input signal.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: If x(n) is the input of an LTI system, then we know that the output of the system y(n) is y(n)= H(ω)x(n) which means the frequency components are just amplified but no new frequency components are added.

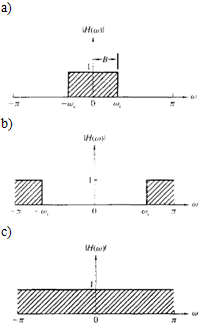
13. An LTI system is characterized by its impulse response h(n)=(1/2)nu(n). What is the spectrum of the output signal when the system is excited by the signal x(n)=(1/4)nu(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q13.png)  
View Answer

Answer: b  
Explanation: The frequency response function of the system is  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q13a.png)

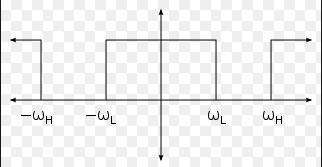
14. What is the frequency response of the system described by the system function H(z)=1/(1-0.8z-1 )?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-characteristics-lti-system-q14.png)d) None of the mentioned  
View Answer

Answer: a  
Explanation: Given H(z)=1/(1-0.8z-1)=z/(z-0.8)  
Clearly, H(z) has a zero at z=0 and a pole at p=0.8. hence the frequency response of the system is given as  
H(ω)= ejω/(ejω-0.8).

Questions & Answers (MCQs) focuses on “LTI System as Frequency Selective Filters”.

1. Which of the following is the magnitude frequency response of High pass filter?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-lti-system-frequency-selective-filters-q1.png)d) None of the mentioned  
View Answer

Answer: b  
Explanation: The property of a high pass filter is to pass the signals with high frequency and stop low frequency signal, which is as shown in the magnitude frequency response of ‘b’.

2. Which filter has a magnitude frequency response as shown in the plot given below?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-lti-system-frequency-selective-filters-q2.png)  
a) Low pass Filter  
b) High pass Filter  
c) Band pass Filter  
d) Band stop Filter  
View Answer

Answer: d  
Explanation: In the magnitude response shown in the question, the system is stopping a particular band of signals. Hence the filter is called as Band stop filter.

3. An ideal filter should have zero gain in their stop band.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: For an ideal filter, in the magnitude response plot at the stop band it should have a sudden fall which means an ideal filter should have a zero gain at stop band.

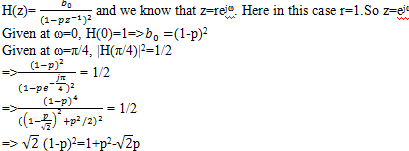
4. The ‘Envelope delay’ or ‘Group delay’ is the time delay that the signal component of frequency ω undergoes as it passes from the input to the output of the system.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The time delay taken to reach the output of the system from the input by a signal component is called as envelope delay or group delay.

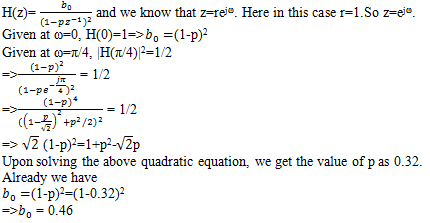
5. If the phase ϴ(ω) of the system is linear, then the group delay of the system:  
a) Increases with frequency of signal  
b) Constant  
c) Decreases with frequency of signal  
d) Independent of frequency of signal  
View Answer

Answer: b  
Explanation: We know that the group delay of the system with phase ϴ(ω) is defined as  
Tg(ω)=(dϴ(ω))/dω  
Given the phase is linear=> the group delay of the system is constant.

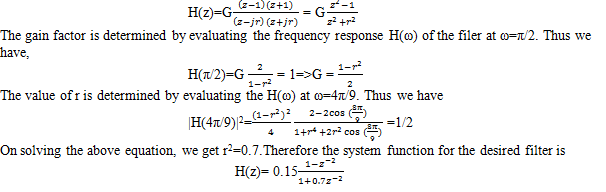
6. A two pole low pass filter has a system function [digital-signal-processing-questions-answers-lti-system-frequency-selective-filters-q6](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-lti-system-frequency-selective-filters-q6.png)What is the value of ‘p’ such that the frequency response H(ω) satisfies the condition |H(π/4)|2=1/2 and H(0)=1?  
a) 0.46  
b) 0.38  
c) 0.32  
d) 0.36  
View Answer

Answer: c  
Explanation: Given  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-lti-system-frequency-selective-filters-q6a.png)  
Upon solving the above quadratic equation, we get the value of p as 0.32.

7. A two pole low pass filter has a system function [digital-signal-processing-questions-answers-lti-system-frequency-selective-filters-q6](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-lti-system-frequency-selective-filters-q6.png)What is the value of ‘b0’ such that the frequency response H(ω) satisfies the condition |H(π/4)|2=1/2 and H(0)=1?  
a) 0.36  
b) 0.38  
c) 0.32  
d) 0.46  
View Answer

Answer: d  
Explanation: Given  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-lti-system-frequency-selective-filters-q7.png)

8. What is the system function for a two pole band pass filter that has the centre of its pass band at ω=π/2, zero its frequency response characteristic at ω=0 and at ω=π, and its magnitude response is 1/√2 at ω=4π/9?  
a) 0.15(1-z-2)/(1+0.7z-2 )  
b) 0.15(1+z-2)/(1-0.7z-2 )  
c) 0.15(1-z-2)/(1-0.7z-2 )  
d) 0.15(1+z-2)/(1+0.7z-2 )  
View Answer

Answer: a  
Explanation: Clearly, the filter must have poles at P1,2=re±jπ/2 and zeros at z=1 and z=-1. Consequently the system function is  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-lti-system-frequency-selective-filters-q8.png)

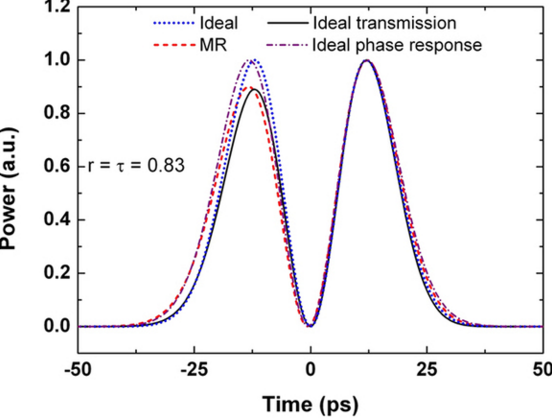
9. If hlp(n) denotes the impulse response of a low pass filter with frequency response Hlp(ω), then what is the frequency response of the high pass filter in terms of Hlp(ω)?  
a) Hlp(ω-π/2)  
b) Hlp(ω+π/2)  
c) Hlp(ω-π)  
d) Hlp(ω+π)  
View Answer

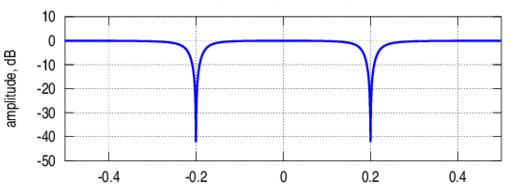
Answer: c  
Explanation: The impulse response of a high pass filter is simply obtained from the impulse response of the low pass filter by changing the signs of the odd numbered samples in hlp(n). Thus  
hhp(n)=(-1)n hlp(n)=(ejπ)n hlp(n)  
Thus the frequency response of the high pass filter is obtained as Hlp(ω-π).

10. If the low pass filter described by the difference equation y(n)=0.9y(n-1)+0.1x(n) is converted into a high pass filter, then what is the frequency response of the high pass filter?  
a) 0.1/(1+0.9ejω )  
b) 0.1/(1+0.9e-jω)  
c) 0.1/(1-0.9ejω )  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The difference equation for the high pass filter is  
y(n)=-0.9y(n-1)+0.1x(n)  
and its frequency response is given as  
H(ω)= 0.1/(1+0.9e-jω).

11. A digital resonator is a special two pole band pass filter with the pair of complex conjugate poles located near the unit circle.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The magnitude response of a band pass filter with two complex poles located near the unit circle is as shown below.  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-lti-system-frequency-selective-filters-q11.png)  
The filter gas a large magnitude response at the poles and hence it is called as digital resonator.

12. Which of the following filters have a frequency response as shown below?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-lti-system-frequency-selective-filters-q12.png)  
a) Band pass filter  
b) Band stop filter  
c) All pass filter  
d) Notch filter  
View Answer

Answer: d  
Explanation: The given figure represents the frequency response characteristic of a notch filter with nulls at frequencies at ω0 and ω1.

13. A comb filter is a special case of notch filter in which the nulls occur periodically across the frequency band.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: A comb filter can be viewed as a notch filter in which the nulls occur periodically across the frequency band, hence the analogy to an ordinary comb that has periodically spaced teeth.

14. The filter with the system function H(z)=z -k is a:  
a) Notch filter  
b) Band pass filter  
c) All pass filter  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The system with the system function given as H(z)=z -k is a pure delay system . It has a constant gain for all frequencies and hence called as All pass filter.

15. If the system has a impulse response as h(n)=Asin(n+1)ω0u(n), then the system is known as Digital frequency synthesizer.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The given impulse response is h(n)=Asin(n+1)ω0u(n).  
According to the above equation, the second order system with complex conjugate poles on the unit circle is a sinusoid and the system is called a digital sinusoidal oscillator or a Digital frequency synthesizer.

Questions & Answers (MCQs) focuses on “Inverse Systems and Deconvolution”.

1. If a system is said to be invertible, then:  
a) One-to-one correspondence between its input and output signals  
b) One-to-many correspondence between its input and output signals  
c) Many-to-one correspondence between its input and output signals  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: If we know the output of a system y(n) of a system and if we can determine the input x(n) of the system uniquely, then the system is said to be invertible. That is there should be one-to-one correspondence between the input and output signals.

2. If h(n) is the impulse response of an LTI system T and h1(n) is the impulse response of the inverse system T-1, then which of the following is true?  
a) [h(n)\*h1(n)].x(n)=x(n)  
b) [h(n).h1(n)].x(n)=x(n)  
c) [h(n)\*h1(n)]\*x(n)=x(n)  
d) [h(n).h1(n)]\*x(n)=x(n)  
View Answer

Answer: c  
Explanation: . If h(n) is the impulse response of an LTI system T and h1(n) is the impulse response of the inverse system T-1, then we know that h(n)\*h1(n)=δ(n)=> [h(n)\*h1(n)]\*x(n)=x(n).

3. What is the inverse of the system with impulse response h(n)=(1/2)nu(n)?  
a) δ(n)+1/2 δ(n-1)  
b) δ(n)-1/2 δ(n-1)  
c) δ(n)-1/2 δ(n+1)  
d) δ(n)+1/2 δ(n+1)  
View Answer

Answer: b  
Explanation: Given impulse response is h(n)=(1/2)nu(n)  
The system function corresponding to h(n) is  
H(z)=1/(1-1/2 z-1 ) ROC:|z|>1/2  
This system is both stable and causal. Since H(z) is all pole system, its inverse is FIR and is given by the system function  
HI(z)= 1- 1/2 z-1  
Hence its impulse response is δ(n)-1/2 δ(n-1).

4. What is the inverse of the system with impulse response h(n)= δ(n)-1/2 δ(n-1)?  
a) (1/2)nu(n)  
b) -(1/2)nu(-n-1)  
c) (1/2)nu(n) & -(1/2)nu(-n-1)  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The system function of given system is H(z)= 1- 1/2 z-1  
The inverse of the system has a system function as H(z)= 1/(1-1/2 z-1 )  
Thus it has a zero at origin and a pole at z=1/2.So, two possible cases are |z|>1/2 and |z|<1/2  
So, h(n)= (1/2)nu(n) for causal and stable(|z|>1/2)  
and h(n)= -(1/2)nu(-n-1) for anti causal and unstable for |z|<1/2.

5. What is the causal inverse of the FIR system with impulse response h(n)=δ(n)-aδ(n-1)?  
a) δ(n)-aδ(n-1)  
b) δ(n)+aδ(n-1)  
c) a -n  
d) an  
View Answer

Answer: d  
Explanation: Given h(n)= δ(n)-aδ(n-1)  
Since h(0)=1, h(1)=-a and h(n)=0 for n≥a, we have  
hI(0)=1/h(0)=1.  
and  
hI(n)=-ahI(n-1) for n≥1  
Consequently, hI(1)=a, hI(2)=a2,….hI(n)=an  
Which corresponds to a causal IIR system as expected.

6. If the frequency response of an FIR system is given as H(z)=6+z-1-z-2, then the system is:  
a) Minimum phase  
b) Maximum phase  
c) Mixed phase  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: Given H(z)=6+z-1-z-2  
By factoring the system function we find the zeros for the system.  
The zeros of the given system are at z=-1/2,1/3  
So, the system is minimum phase.

7. If the frequency response of an FIR system is given as H(z)=1-z-1-z-2, then the system is:  
a) Minimum phase  
b) Maximum phase  
c) Mixed phase  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: Given H(z)= 1-z-1-z-2  
By factoring the system function we find the zeros for the system.  
The zeros of the given system are at z=-2,3  
So, the system is maximum phase.

8. If the frequency response of an FIR system is given as H(z)=1-5/2z-1-3/2z-2, then the system is:  
a) Minimum phase  
b) Maximum phase  
c) Mixed phase  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: Given H(z)= 1-5/2z-1-3/2z-2  
By factoring the system function we find the zeros for the system.  
The zeros of the given system are at z=-1/2, 3  
So, the system is mixed phase.

9. An IIR system with system function H(z)=(B(z))/(A(z)) is called a minimum phase if:  
a) All poles and zeros are inside the unit circle  
b) All zeros are outside the unit circle  
c) All poles are outside the unit circle  
d) All poles and zeros are outside the unit circle  
View Answer

Answer: a  
Explanation: For an IIR filter whose system function is defined as H(z)=(B(z))/(A(z)) to be said a minimum phase,  
then both the poles and zeros of the system should fall inside the unit circle.

10. An IIR system with system function H(z)=(B(z))/(A(z)) is called a mixed phase if:  
a) All poles and zeros are inside the unit circle  
b) All zeros are outside the unit circle  
c) All poles are outside the unit circle  
d) Some, but not all of the zeros are outside the unit circle  
View Answer

Answer: d  
Explanation: For an IIR filter whose system function is defined as H(z)=(B(z))/(A(z)) to be said a mixed phase and if the system is stable and causal, then the poles are inside the unit circle and some, but not all of the zeros are outside the unit circle.

11. A causal system produces the output sequence y(n)={1,0.7} when excited by the input sequence x(n)={1,-0.7,0.1}, then what is the impulse response of the system function?  
a) [3(0.5)n+4(0.2)n]u(n)  
b) [4(0.5)n-3(0.2)n]u(n)  
c) [4(0.5)n+3(0.2)n]u(n)  
d) None of the mentioned  
View Answer

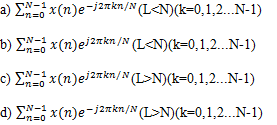
Answer: b  
Explanation: The system function is easily determined by taking the z-transforms of x(n) and y(n). Thus we have  
H(z)=(Y(z))/(X(z)) = (1+0.7z-1)/(1-0.7z-1+0.1z-2 ) = (1+0.7z-1)/((1-0.2z-1)(1-0.5z-1))  
Upon applying partial fractions and applying the inverse z-transform, we get  
[4(0.5)n-3(0.2)n]u(n).

#### 5. Questions & Answers on Discrete Fourier Transform – Properties and Applications

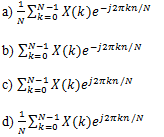
The section contains questions and answers on discrete fourier transforms, their sampling and properties, linear filtering methods on DFT and their frequency analysis.

|  |  |
| --- | --- |
| [Frequency Domain Sampling](https://www.sanfoundry.com/digital-signal-processing-questions-answers-frequency-domain-sampling-dft/) [DFT Properties](https://www.sanfoundry.com/digital-signal-processing-questions-answers-properties-dft/) | [DFT Signal Analysis](https://www.sanfoundry.com/digital-signal-processing-questions-answers-frequency-analysis-signals-using-dft/) |

Questions & Answers (MCQs) focuses on “Frequency Domain Sampling DFT”.

1. If x(n) is a finite duration sequence of length L, then the discrete Fourier transform X(k) of x(n) is given as:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q1.png)  
View Answer

Answer: a  
Explanation: If x(n) is a finite duration sequence of length L, then the Fourier transform of x(n) is given as  
[digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q1a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q1a.png)  
If we sample X(ω) at equally spaced frequencies ω=2πk/N, k=0,1,2…N-1 where N>L, the resultant samples are  
[digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q1b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q1b.png)

2. If X(k) discrete Fourier transform of x(n), then the inverse discrete Fourier transform of X(k) is:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q2.png)  
View Answer

Answer: d  
Explanation: If X(k) discrete Fourier transform of x(n), then the inverse discrete Fourier transform of X(k) is given as  
[digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q2a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q2a.png)

3. A finite duration sequence of length L is given as x(n) =1 for 0≤n≤L-1=0 otherwise , then what is the N point DFT of this sequence for N=L?  
a) X(k) =L for k=0, 1,2….L-1  
b) X(k) =L for k=0  
=0 for k=1,2….L-1  
c) X(k) =L for k=0  
=1 for k=1,2….L-1  
d) None of the mentioned  
View Answer

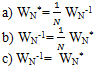
Answer: b  
Explanation: The Fourier transform of this sequence is  
[digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q3](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q3.png)  
If N=L, then X(k)= L for k=0  
=0 for k=1,2….L-1

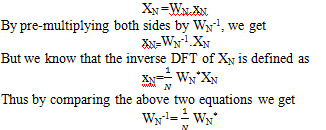
4. The Nth rot of unity WN is given as:  
a) ej2πN  
b) e -j2πN  
c) e-j2π/N  
d) ej2π/N  
View Answer

Answer: c  
Explanation: We know that the Discrete Fourier transform of a signal x(n) is given as[digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q4](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q4.png)  
Thus we get Nth rot of unity WN= e-j2π/N

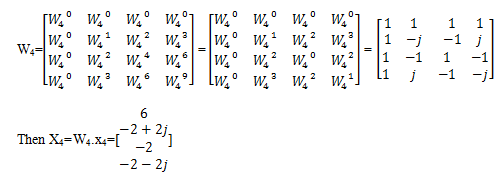
5. Which of the following is true regarding the number of computations requires to compute an N-point DFT?  
a) N2 complex multiplications and N(N-1) complex additions  
b) N2 complex additions and N(N-1) complex multiplications  
c) N2 complex multiplications and N(N+1) complex additions  
d) N2 complex additions and N(N+1) complex multiplications  
View Answer

Answer: a  
Explanation: The formula for calculating N point DFT is given as  
[digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q1b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q1b.png)  
From the formula given at every step of computing we are performing N complex multiplications and N-1 complex additions. So, in a total to perform N-point DFT we perform N2 complex multiplications and N(N-1) complex additions.

6. Which of the following is true?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q6.png)d) None of the mentioned  
View Answer

Answer: b  
Explanation: If XN represents the N point DFT of the sequence xN in the matrix form, then we know that  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q6a.png)

7. What is the DFT of the four point sequence x(n)={0,1,2,3}?  
a) {6,-2+2j-2,-2-2j}  
b) {6,-2-2j,2,-2+2j}  
c) {6,-2+2j,-2,-2-2j}  
d) {6,-2-2j,-2,-2+2j}  
View Answer

Answer: c  
Explanation: The first step is to determine the matrix W4. By exploiting the periodicity property of W4 and the symmetry property  
WNk+N/2= -WNk  
The matrix W4 may be expressed as  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q7.png)

8. If X(k) is the N point DFT of a sequence whose Fourier series coefficients is given by ck, then which of the following is true?  
a) X(k)=Nck  
b) X(k)=ck/N  
c) X(k)=N/ck  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: The Fourier series coefficients are given by the expression  
[digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q8](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q8.png)

9. What is the DFT of the four point sequence x(n)={0,1,2,3}?  
a) {6,-2+2j-2,-2-2j}  
b) {6,-2-2j,2,-2+2j}  
c) {6,-2-2j,-2,-2+2j}  
d) {6,-2+2j,-2,-2-2j}  
View Answer

Answer: d  
Answer: Given x(n)={0,1,2,3}  
We know that the 4-point DFT of the above given sequence is given by the expression  
[digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q1b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-domain-sampling-dft-q1b.png)  
In this case N=4  
=>X(0)=6,X(1)=-2+2j,X(2)=-2,X(3)=-2-2j.

10. If W4100=Wx200, then what is the value of x?  
a) 2  
b) 4  
c) 8  
d) 16  
View Answer

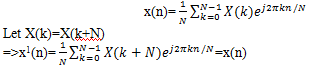
Answer: c  
Explanation: We know that according to the periodicity and symmetry property,  
100/4=200/x=>x=8.

Questions & Answers (MCQs) focuses on “Properties of DFT”.

1. If x(n) and X(k) are an N-point DFT pair, then x(n+N)=x(n).  
a) True  
b) False  
View Answer

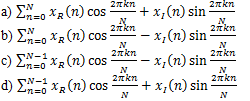
Answer: a  
Explanation: We know that the expression for an DFT is given as  
[digital-signal-processing-questions-answers-properties-dft-q1](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-dft-q1.png)  
Therefore, we got x(n)=x(n+N)

2. If x(n) and X(k) are an N-point DFT pair, then X(k+N)=?  
a) X(-k)  
b) -X(k)  
c) X(k)  
d) None of the mentioned  
View Answer

: c  
Explanation: We know that  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-dft-q2.png)  
Therefore, we have X(k)=X(k+N)

3. If X1(k) and X2(k) are the N-point DFTs of x1(n) and x2(n) respectively, then what is the N-point DFT of x(n)=ax1(n)+bx2(n)?  
a) X1(ak)+X2(bk)  
b) aX1(k)+bX2(k)  
c) eakX1(k)+ebkX2(k)  
d) None of the mentioned  
View Answer

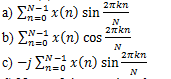
Answer: b  
Explanation: We know that, the DFT of a signal x(n) is given by the expression  
[digital-signal-processing-questions-answers-properties-dft-q3](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-dft-q3.png)  
=>X(k)= aX1(k)+bX2(k).

4. If x(n) is a complex valued sequence given by x(n)=xR(n)+jxI(n), then what is the DFT of xR(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-dft-q4.png)  
View Answer

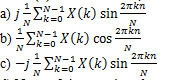
Answer: d  
Explanation: Given x(n)=xR(n)+jxI(n)=> xR(n)=1/2(x(n)+x\*(n))  
Substitute the above equation in the DFT expression  
Thus we get, [digital-signal-processing-questions-answers-properties-dft-q4a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-dft-q4a.png)

5. If x(n) is a real sequence and X(k) is its N-point DFT, then which of the following is true?  
a) X(N-k)=X(-k)  
b) X(N-k)=X\*(k)  
c) X(-k)=X\*(k)  
d) All of the mentioned  
View Answer

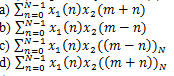
Answer: d  
Explanation: We know that  
[digital-signal-processing-questions-answers-properties-dft-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-dft-q5.png)  
Therefore,  
X(N-k)=X\*(k)=X(-k)

6. If x(n) is real and even, then what is the DFT of x(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-dft-q6.png)d) None of the mentioned  
View Answer

Answer: b  
Explanation: Given x(n) is real and even, that is x(n)=x(N-n)  
We know that XI(k)=0. Hence the DFT reduces to  
[digital-signal-processing-questions-answers-properties-dft-q6a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-dft-q6a.png)

7. If x(n) is real and odd, then what is the IDFT of the given sequence?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-dft-q7.png)d) None of the mentioned  
View Answer

Answer: a  
Explanation: If x(n) is real and odd, that is x(n)=-x(N-n), then XR(k)=0. Hence X(k) is purely imaginary and odd. Since XR(k) reduces to zero, the IDFT reduces to  
[digital-signal-processing-questions-answers-properties-dft-q7a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-dft-q7a.png)

8. If x1(n),x2(n) and x3(m) are three sequences each of length N whose DFTs are given as X1(k),X2(k) and X3(k) respectively and X3(k)=X1(k).X2(k), then what is the expression for x3(m)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-dft-q8.png)  
View Answer

Answer: c  
Explanation: If x1(n),x2(n) and x3(m) are three sequences each of length N whose DFTs are given as X1(k),X2(k) and X3(k) respectively and X3(k)=X1(k).X2(k), then according to the multiplication property of DFT we have x3(m) is the circular convolution of x1(n) and x2(n).

9. What is the circular convolution of the sequences x1(n)={2,1,2,1} and x2(n)={1,2,3,4}?  
a) {14,14,16,16}  
b) {16,16,14,14}  
c) {2,3,6,4}  
d) {14,16,14,16}  
View Answer

Answer: d  
Explanation: We know that the circular convolution of two sequences is given by the expression  
[digital-signal-processing-questions-answers-properties-dft-q9](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-properties-dft-q9.png)  
For m=0,x2((-n))4={1,4,3,2}  
For m=1,x2((1-n))4={2,1,4,3}  
For m=2,x2((2-n))4={3,2,1,4}  
For m=3,x2((3-n))4={4,3,2,1}  
Now we get x(m)={14,16,14,16}.

10. What is the circular convolution of the sequences x1(n)={2,1,2,1} and x2(n)={1,2,3,4}, find using the DFT and IDFT concepts?  
a) {16,16,14,14}  
b) {14,16,14,16}  
c) {14,14,16,16}  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: Given x1(n)={2,1,2,1}=>X1(k)=[6,0,2,0] Given x2(n)={1,2,3,4}=>X2(k)=[10,-2+j2,-2,-2-j2] when we multiply both DFTs we obtain the product  
X(k)=X1(k).X2(k)=[60,0,-4,0] By applying the IDFT to the above sequence, we get  
x(n)={14,16,14,16}.

11. If X(k) is the N-point DFT of a sequence x(n), then circular time shift property is that N-point DFT of x((n-l))N is X(k)e-j2πkl/N.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: According to the circular time shift property of a sequence, If X(k) is the N-point DFT of a sequence x(n), then the N-pint DFT of x((n-l))N is X(k)e-j2πkl/N.

12. If X(k) is the N-point DFT of a sequence x(n), then what is the DFT of x\*(n)?  
a) X(N-k)  
b) X\*(k)  
c) X\*(N-k)  
d) None of the mentioned  
View Answer

Answer: According to the complex conjugate property of DFT, we have if X(k) is the N-point DFT of a sequence x(n), then what is the DFT of x\*(n) is X\*(N-k).

Questions & Answers (MCQs) focuses on “Frequency Analysis of Signals Using DFT”.

1. If the signal to be analyzed is an analog signal, we would pass it through an anti-aliasing filter with B as the bandwidth of the filtered signal and then the signal is sampled at a rate:  
a) Fs ≤ 2B  
b) Fs ≤ B  
c) Fs ≥ 2B  
d) Fs = 2B  
View Answer

Answer: c  
Explanation: The filtered signal is sampled at a rate of Fs≥ 2B, where B is the bandwidth of the filtered signal to prevent aliasing.

2. What is the highest frequency that is contained in the sampled signal?  
a) 2Fs  
b) Fs/2  
c) Fs  
d) None of the mentioned  
View Answer

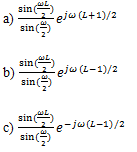
Answer: b  
Explanation: We know that, after passing the signal through anti-aliasing filter, the filtered signal is sampled at a rate of Fs≥ 2B=>B≤ Fs/2.Thus the maximum frequency of the sampled signal is Fs/2.

3. The finite observation interval for the signal places a limit on the frequency resolution.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: After sampling the signal, we limit the duration of the signal to the time interval T0=LT, where L is the number of samples and T is the sample interval. So, it limits our ability to distinguish two frequency components that are separated by less than 1/T0=1/LT in frequency. So, the finite observation interval for the signal places a limit on the frequency resolution.

4. If {x(n)} is the signal to be analyzed, limiting the duration of the sequence to L samples, in the interval 0≤ n≤ L-1, is equivalent to multiplying {x(n)} by:  
a) Kaiser window  
b) Hamming window  
c) Hanning window  
d) Rectangular window  
View Answer

Answer: d  
Explanation: The equation of the rectangular window w(n) is given as  
w(n)= 1, 0≤ n≤ L-1  
=0, otherwise  
Thus, we can limit the duration of the signal x(n) to L samples by multiplying it with a rectangular window of length L.

5. What is the Fourier transform of rectangular window of length L?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-signals-using-dft-q5.png)d) None of the mentioned  
View Answer

Answer: c  
Explanation: We know that the equation for the rectangular window w(n) is given as  
w(n)= 1, 0≤ n≤ L-1  
=0, otherwise  
We know that the Fourier transform of a signal x(n) is given as  
[digital-signal-processing-questions-answers-frequency-analysis-signals-using-dft-q5a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-analysis-signals-using-dft-q5a.png)

6. If x(n)=cosω0n and W(ω) is the Fourier transform of the rectangular signal w(n), then what is the Fourier transform of the signal x(n).w(n)?  
a) 1/2[W(ω-ω0)- W(ω+ω0)].  
b) 1/2[W(ω-ω0)+ W(ω+ω0)].  
c) [W(ω-ω0)+ W(ω+ω0)].  
d) [W(ω-ω0)- W(ω+ω0)].  
View Answer

Answer: b  
Explanation: According to the exponential properties of Fourier transform, we get  
Fourier transform of x(n).w(n)= 1/2[W(ω-ω0)+ W(ω+ω0)]

7. The characteristic of windowing the signal called “Leakage” is the power that is leaked out into the entire frequency range.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: We note that the windowed spectrum X ̂(ω) is not localized to a single frequency, but instead it is spread out over the whole frequency range. Thus the power of the original signal sequence x(n) that was concentrated at a single frequency has been spread by the window into the entire frequency range. We say that the power has been leaked out into the entire frequency range and this phenomenon is called as “Leakage”.

8. Which of the following is the advantage of Hanning window over rectangular window?  
a) More side lobes  
b) Less side lobes  
c) More width of main lobe  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The Hanning window has less side lobes and the leakage is less in this windowing technique.

9. Which of the following is the disadvantage of Hanning window over rectangular window?  
a) More side lobes  
b) Less side lobes  
c) More width of main lobe  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: In the magnitude response of the signal windowed using Hanning window, the width of the main lobe is more which is the disadvantage of this technique over rectangular windowing technique.

10. The condition with less number of samples L should be avoided.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: When the number of samples L is small, the window spectrum masks the signal spectrum and, consequently , the DFT of the data reflects the spectral characteristics of the window function. So, this situation should be avoided.

#### 6. Questions on DFT Efficient Computation – Fast Fourier Transform Algorithms

The section contains questions on computation of discrete fourier transforms and fast fourier transforms, various approaches to their computation which include filtering and quantization and applications of FFT algorithms.

|  |  |
| --- | --- |
| [DFT Algorithm Computation 1](https://www.sanfoundry.com/digital-signal-processing-questions-answers-efficient-computation-dft-fft-algorithms-1/) [DFT Algorithm Computation 2](https://www.sanfoundry.com/digital-signal-processing-questions-answers-freshers/) [FFT Algorithms Applications](https://www.sanfoundry.com/digital-signal-processing-questions-answers-applications-fft-algorithms/) | [DFT Computation Filtering Approach](https://www.sanfoundry.com/digital-signal-processing-interview-questions-answers-freshers/) [Quantization Effects](https://www.sanfoundry.com/digital-signal-processing-questions-answers-quantization-effects-computation-dft/) |

Questions & Answers (MCQs) focuses on “Efficient Computation of DFT FFT Algorithms-1′.

1. Which of the following is true regarding the number of computations required to compute an N-point DFT?  
a) N2 complex multiplications and N(N-1) complex additions  
b) N2 complex additions and N(N-1) complex multiplications  
c) N2 complex multiplications and N(N+1) complex additions  
d) N2 complex additions and N(N+1) complex multiplications  
View Answer

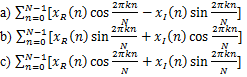
Answer: a  
Explanation: The formula for calculating N point DFT is given as  
[digital-signal-processing-questions-answers-efficient-computation-dft-fft-algorithms-1-q1](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-efficient-computation-dft-fft-algorithms-1-q1.png)  
From the formula given at every step of computing we are performing N complex multiplications and N-1 complex additions. So, in a total to perform N-point DFT we perform N2 complex multiplications and N(N-1) complex additions.

2. Which of the following is true regarding the number of computations required to compute DFT at any one value of ‘k’?  
a) 4N-2 real multiplications and 4N real additions  
b) 4N real multiplications and 4N-4 real additions  
c) 4N-2 real multiplications and 4N+2 real additions  
d) 4N real multiplications and 4N-2 real additions  
View Answer

Answer: d  
Explanation: The formula for calculating N point DFT is given as  
[digital-signal-processing-questions-answers-efficient-computation-dft-fft-algorithms-1-q1](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-efficient-computation-dft-fft-algorithms-1-q1.png)  
From the formula given at every step of computing we are performing N complex multiplications and N-1 complex additions. So, it requires 4N real multiplications and 4N-2 real additions for any value of ‘k’ to compute DFT of the sequence.

3. WNk+N/2=  
a) WNk  
b) -WNk  
c) WN-k  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: According to the symmetry property, we get WNk+N/2=-WNk.

4. What is the real part of the N point DFT XR(k) of a complex valued sequence x(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-efficient-computation-dft-fft-algorithms-1-q4.png)d) None of the mentioned  
View Answer

Answer: c  
Explanation: For a complex valued sequence x(n) of N points, the DFT may be expressed as  
[digital-signal-processing-questions-answers-efficient-computation-dft-fft-algorithms-1-q4a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-efficient-computation-dft-fft-algorithms-1-q4a.png)

5. The computation of XR(k) for a complex valued x(n) of N points requires:  
a) 2N2 evaluations of trigonometric functions  
b) 4N2 real multiplications  
c) 4N(N-1) real additions  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: The expression for XR(k) is given as [digital-signal-processing-questions-answers-efficient-computation-dft-fft-algorithms-1-q4a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-efficient-computation-dft-fft-algorithms-1-q4a.png)  
So, from the equation we can tell that the computation of XR(k) requires 2N2 evaluations of trigonometric functions, 4N2 real multiplications and 4N(N-1) real additions.

6. Divide-and-conquer approach is based on the decomposition of an N-point DFT into successively smaller DFTs. This basic approach leads to FFT algorithms.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: T he development of computationally efficient algorithms for the DFT is made possible if we adopt a divide-and-conquer approach. This approach is based on the decomposition of an N-point DFT into successively smaller DFTs. This basic approach leads to a family of computationally efficient algorithms known collectively as FFT algorithms.

7. If the arrangement is of the form in which the first row consists of the first M elements of x(n), the second row consists of the next M elements of x(n), and so on, then which of the following mapping represents the above arrangement?  
a) n=l+mL  
b) n=Ml+m  
c) n=ML+l  
d) none of the mentioned  
View Answer

Answer: b  
Explanation: If we consider the mapping n=Ml+m, then it leads to an arrangement in which the first row consists of the first M elements of x(n), the second row consists of the next M elements of x(n), and so on.

8. If N=LM, then what is the value of WNmqL?  
a) WMmq  
b) WLmq  
c) WNmq  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: We know that if N=LM, then WNmqL= WN/Lmq= WMmq.

9. How many complex multiplications are performed in computing the N-point DFT of a sequence using divide-and-conquer method if N=LM?  
a) N(L+M+2)  
b) N(L+M-2)  
c) N(L+M-1)  
d) N(L+M+1)  
View Answer

Answer: d  
Explanation: The expression for N point DFT is given as [digital-signal-processing-questions-answers-efficient-computation-dft-fft-algorithms-1-q9](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-efficient-computation-dft-fft-algorithms-1-q9.png)  
The first step involves L DFTs, each of M points. Hence this step requires LM2 complex multiplications, second require LM and finally third requires ML2. So, Total complex multiplications= N(L+M+1).

10. How many complex additions are performed in computing the N-point DFT of a sequence using divide-and-conquer method if N=LM?  
a) N(L+M+2)  
b) N(L+M-2)  
c) N(L+M-1)  
d) N(L+M+1)  
View Answer

Answer: b  
Explanation: The expression for N point DFT is given as [digital-signal-processing-questions-answers-efficient-computation-dft-fft-algorithms-1-q9](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-efficient-computation-dft-fft-algorithms-1-q9.png)  
The first step involves L DFTs, each of M points. Hence this step requires LM(M-1) complex additions, second step do not require any additions and finally third step requires ML(L-1) complex additions. So, Total number of complex additions= N(L+M-2).

11. Which is the correct order of the following steps to be done in one of the algorithm of divide and conquer method?  
1) Store the signal column wise  
2) Compute the M-point DFT of each row  
3) Multiply the resulting array by the phase factors WNlq.  
4) Compute the L-point DFT of each column.  
5) Read the result array row wise.  
a) 1-2-4-3-5  
b) 1-3-2-4-5  
c) 1-2-3-4-5  
d) 1-4-3-2-5  
View Answer

Answer: c  
Explanation: According to one of the algorithm describing the divide and conquer method, if we store the signal in column wise, then compute the M-point DFT of each row and multiply the resulting array by the phase factors WNlq and then compute the L-point DFT of each column and read the result row wise.

12. If we store the signal row wise then the result must be read column wise.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: According to the second algorithm of divide and conquer approach, if the input signal is stored in row wise, then the result must be read column wise.

13. If we store the signal row wise and compute the L point DFT at each column, the resulting array must be multiplied by which of the following factors?  
a) WNlq  
b) WNpq  
c) WNlq  
d) WNpm  
View Answer

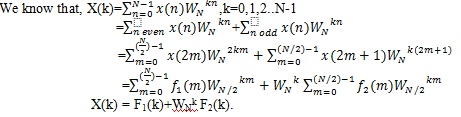
Answer: d  
Explanation: According to the second algorithm of divide and conquer approach, if the input signal is stored in row wise, then we calculate the L point DFT of each column and we multiply the resulting array by the factor WNpm.

Questions & Answers for freshers focuses on “Efficient Computation of DFT FFT Algorithms”.

1. If we split the N point data sequence into two N/2 point data sequences f1(n) and f2(n) corresponding to the even numbered and odd numbered samples of x(n), then such an FFT algorithm is known as decimation-in-time algorithm.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Let us consider the computation of the N=2v point DFT by the divide and conquer approach. We select M=N/2 and L=2. This selection results in a split of N point data sequence into two N/2 point data sequences f1(n) and f2(n) corresponding to the even numbered and odd numbered samples of x(n), respectively, that is  
f1(n)=x(2n)  
f2(n)=x(2n+1) ,n=0,1,2…N/2-1  
Thus f1(n) and f2(n) are obtained by decimating x(n) by a factor of  
2, and hence the resulting FFT algorithm is called a decimation-in-time algorithm.

2. If we split the N point data sequence into two N/2 point data sequences f1(n) and f2(n) corresponding to the even numbered and odd numbered samples of x(n) and F1(k) and F2(k) are the N/2 point DFTs of f1(k) and f2(k) respectively, then what is the N/2 point DFT X(k) of x(n)?  
a) F1(k)+F2(k)  
b) F1(k)- WNk F2(k)  
c) F1(k)+WNkNk F2(k)  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: From the question, it is given that  
f1(n)=x(2n)  
f2(n)=x(2n+1) ,n=0,1,2…N/2-1  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-freshers-q2.png)

3. If X(k) is the N/2 point DFT of the sequence x(n), then what is the value of X(k+N/2)?  
a) F1(k)+F2(k)  
b) F1(k)- WNk F2(k)  
c) F1(k)+WNk F2(k)  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: We know that, X(k) = F1(k)+WNk F2(k)  
We know that F1(k) and F2(k) are periodic, with period N/2, we have F1(k+N/2)= F1(k) and F2(k+N/2)= F2(k). In addition, the factor WNk+N/2= -WNk.  
Thus we get, X(k+N/2)= F1(k)- WNk F2(k).

4. How many complex multiplications are required to compute X(k)?  
a) N(N+1)  
b) N(N-1)/2  
c) N2/2  
d) N(N+1)/2  
View Answer

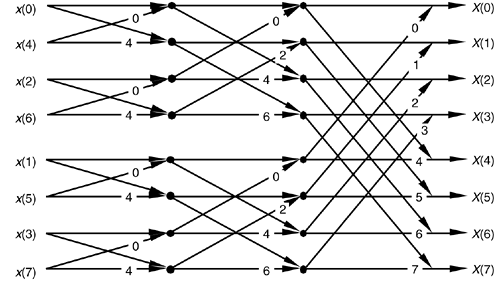
Answer: d  
Explanation: We observe that the direct computation of F1(k) requires (N/2)2 complex multiplications. The same applies to the computation of F2(k). Furthermore, there are N/2 additional complex multiplications required to compute WNk. Hence it requires N(N+1)/2 complex multiplications to compute X(k).

5. The total number of complex multiplications required to compute N point DFT by radix-2 FFT is:  
a) (N/2)log2N  
b) Nlog2N  
c) (N/2)logN  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: The decimation of the data sequence should be repeated again and again until the resulting sequences are reduced to one point sequences. For N=2v, this decimation can be performed v=log2N times. Thus the total number of complex multiplications is reduced to (N/2)log2N.

6. The total number of complex additions required to compute N point DFT by radix-2 FFT is:  
a) (N/2)log2N  
b) Nlog2N  
c) (N/2)logN  
d) None of the mentioned  
View Answer

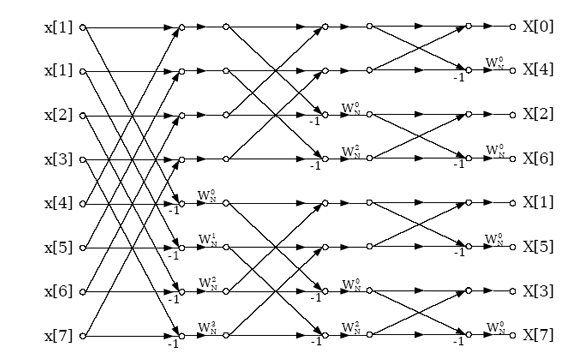
Answer: b  
Explanation: The decimation of the data sequence should be repeated again and again until the resulting sequences are reduced to one point sequences. For N=2v, this decimation can be performed v=log2N times. Thus the total number of complex additions is reduced to Nlog2N.

7. The following butterfly diagram is used in the computation of:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-freshers-q7.png)  
a) Decimation-in-time FFT  
b) Decimation-in-frequency FFT  
c) All of the mentioned  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: The above given diagram is the basic butterfly computation in the decimation-in-time FFT algorithm.

8. For a decimation-in-time FFT algorithm, which of the following is true?  
a) Both input and output are in order  
b) Both input and output are shuffled  
c) Input is shuffled and output is in order  
d) Input is in order and output is shuffled  
View Answer

Answer: c  
Explanation: In decimation-in-time FFT algorithm, the input is taken in bit reversal order and the output is obtained in the order.

9. The following butterfly diagram is used in the computation of:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-freshers-q9.png)  
a) Decimation-in-time FFT  
b) Decimation-in-frequency FFT  
c) All of the mentioned  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The above given diagram is the basic butterfly computation in the decimation-in-frequency FFT algorithm.

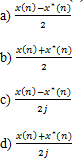
10. For a decimation-in-time FFT algorithm, which of the following is true?  
a) Both input and output are in order  
b) Both input and output are shuffled  
c) Input is shuffled and output is in order  
d) Input is in order and output is shuffled  
View Answer

Answer: d  
Explanation: In decimation-in-frequency FFT algorithm, the input is taken in order and the output is obtained in the bit reversal order.

Questions & Answers (MCQs) focuses on “Applications of FFT Algorithms”.

1. FFT algorithm is designed to perform complex operations.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The FFT algorithm is designed to perform complex multiplications and additions, even though the input data may be real valued. The basic reason for this is that the phase factors are complex and hence, after the first stage of the algorithm, all variables are basically complex valued.

2. If x1(n) and x2(n) are two real valued sequences of length N, and let x(n) be a complex valued sequence defined as x(n)=x1(n)+jx2(n), 0≤ n≤ N-1, then what is the value of x1(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-applications-fft-algorithms-q2.png)  
View Answer

Answer: b  
Explanation: Given x(n)=x1(n)+jx2(n)  
=>x\*(n)= x1(n)-jx2(n)  
Upon adding the above two equations, we get x1(n)= (x(n)+x\*(n))/2.

3. If x1(n) and x2(n) are two real valued sequences of length N, and let x(n) be a complex valued sequence defined as x(n)=x1(n)+jx2(n), 0≤ n≤ N-1, then what is the value of x2(n)?  
a) (x(n)-x\*(n))/2  
b) (x(n)+x\*(n))/2  
c) (x(n)+x\*(n))/2j  
d) (x(n)-x\*(n))/2j  
View Answer

Answer: d  
Explanation: Given x(n)=x1(n)+jx2(n)  
=>x\*(n)= x1(n)-jx2(n)  
Upon subtracting the above two equations, we get x2(n)= (x(n)-x\*(n))/2j.

4. If X(k) is the DFT of x(n) which is defined as x(n)=x1(n)+jx2(n), 0≤ n≤ N-1, then what is the DFT of x1(n)?  
a) 1/2 [X\*(k)+X\*(N-k)].  
b) 1/2 [X\*(k)-X\*(N-k)].  
c) 1/2j [X\*(k)-X\*(N-k)].  
d) 1/2j [X\*(k)+X\*(N-k)].  
View Answer

Answer: a  
Explanation: We know that if x(n)=x1(n)+jx2(n) then x1(n)= (x(n)+x\*(n))/2  
On applying DFT on both sides of the above equation, we get  
X1(k)= 1/2 {DFT[x(n)]+DFT[x\*(n)]}  
We know that if X(k) is the DFT of x(n), the DFT[x\*(n)]=X\*(N-k)  
=>X1(k)= 1/2 [X\*(k)+X\*(N-k)].

5. If X(k) is the DFT of x(n) which is defined as x(n)=x1(n)+jx2(n), 0≤ n≤ N-1, then what is the DFT of x1(n)?  
a) (1/2) [X\*(k)+X\*(N-k)].  
b) (1/2) [X\*(k)-X\*(N-k)].  
c) (1/2j) [X\*(k)-X\*(N-k)].  
d) (1/2j) [X\*(k)+X\*(N-k)].  
View Answer

Answer: c  
Explanation: We know that if x(n)=x1(n)+jx2(n) then x2(n)= (x(n)-x^\* (n))/2j.  
On applying DFT on both sides of the above equation, we get  
X2(k)= (1/2j) {DFT[x(n)]-DFT[x\*(n)]}  
We know that if X(k) is the DFT of x(n), the DFT[x\*(n)]=X\*(N-k)  
=>X2(k)= (1/2j) [X\*(k)-X\*(N-k)].

6. If g(n) is a real valued sequence of 2N points and x1(n)=g(2n) and x2(n)=g(2n+1), then what is the value of G(k), k=0,1,2…N-1?  
a) X1(k)-W2kNX2(k)  
b) X1(k)+W2kNX2(k)  
c) X1(k)+W2kX2(k)  
d) X1(k)-W2kX2(k)  
View Answer

Answer: b  
Explanation: Given g(n) is a real valued 2N point sequence. The 2N point sequence is divided into two N point sequences x1(n) and x2(n)  
Let x(n)= x1(n)+jx2(n)  
=> X1(k)= 1/2 [X\*(k)+X\*(N-k)] and X2(k)= 1/2j [X\*(k)-X\*(N-k)] We know that g(n)= x1(n)+x2(n)  
=>G(k)= X1(k)+W2kNX2(k), k=0,1,2…N-1.

7. If g(n) is a real valued sequence of 2N points and x1(n)=g(2n) and x2(n)=g(2n+1), then what is the value of G(k), k=N,N-1,…2N-1?  
a) X1(k)-W2kX2(k)  
b) X1(k)+W2kNX2(k)  
c) X1(k)+W2kX2(k)  
d) X1(k)-W2kNX2(k)  
View Answer

Answer: d  
Explanation: Given g(n) is a real valued 2N point sequence. The 2N point sequence is divided into two N point sequences x1(n) and x2(n)  
Let x(n)= x1(n)+jx2(n)  
=> X1(k)= 1/2 [X\*(k)+X\*(N-k)] and X2(k)= 1/2j [X\*(k)-X\*(N-k)] We know that g(n)= x1(n)+x2(n)  
=>G(k)= X1(k)-W2kNX2(k), k= N,N-1,…2N-1.

8. Decimation-in frequency FFT algorithm is used to compute H(k).  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The N-point DFT of h(n), which is padded by L-1 zeros, is denoted as H(k). This computation is performed once via the FFT and resulting N complex numbers are stored. To be specific we assume that the decimation-in frequency FFT algorithm is used to compute H(k). This yields H(k) in the bit-reversed order, which is the way it is stored in the memory.

9. How many complex multiplications are need to be performed for each FFT algorithm?  
a) (N/2)logN  
b) Nlog2N  
c) (N/2)log2N  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The decimation of the data sequence should be repeated again and again until the resulting sequences are reduced to one point sequences. For N=2v, this decimation can be performed v=log2N times. Thus the total number of complex multiplications is reduced to (N/2)log2N.

10. How many complex additions are required to be performed in linear filtering of a sequence using FFT algorithm?  
a) (N/2)logN  
b) 2Nlog2N  
c) (N/2)log2N  
d) Nlog2N  
View Answer

Answer: b  
Explanation: The number of additions to be performed in FFT are Nlog2N. But in linear filtering of a sequence, we calculate DFT which requires Nlog2N complex additions and IDFT requires Nlog2N complex additions. So, the total number of complex additions to be performed in linear filtering of a sequence using FFT algorithm is 2Nlog2N.

11. How many complex multiplication are required per output data point?  
a) [(N/2)logN]/L  
b) [Nlog22N]/L  
c) [(N/2)log2N]/L  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: In the overlap add method, the N-point data block consists of L new data points and additional M-1 zeros and the number of complex multiplications required in FFT algorithm are (N/2)log2N. So, the number of complex multiplications per output data point is [Nlog22N]/L.

Questions & Answers focuses on “Linear Filtering Approach to Computation of DFT”.

1. By means of the DFT and IDFT, determine the response of the FIR filter with impulse response h(n)={1,2,3} to the input sequence x(n)={1,2,2,1}?  
a) {1,4,11,9,8,3}  
b) {1,4,9,11,8,3}  
c) {1,4,9,11,3,8}  
d) {1,4,9,3,8,11}  
View Answer

Answer: b  
Explanation: The input sequence has a length N=4 and impulse response has a length M=3. So, the response must have a length of 6(4+3-1).  
We know that, Y(k)=X(k).H(k)  
Thus we obtain Y(k)={36,-14.07-j17.48,j4,0.07+j0.515,0,0.07-j0.515,-j4,-14.07+j17.48}  
By applying IDFT to the above sequence, we get y(n)={1,4,9,11,8,3,0,0}  
Thus the output of the system is {1,4,9,11,8,3}.

2. What is the sequence y(n) that results from the use of four point DFTs if the impulse response is h(n)={1,2,3} and the input sequence x(n)={1,2,2,1}?  
a) {9,9,7,11}  
b) {1,4,9,11,8,3}  
c) {7,9,7,11}  
d) {9,7,9,11}  
View Answer

Answer: d  
Explanation: The four point DFT of h(n) is H(k)=1+2e-jkπ/2+3 e-jkπ (k=0,1,2,3)  
Hence H(0)=6, H(1)=-2-j2, H(3)=2, H(4)=-2+j2  
The four point DFT of x(n) is X(k)= 1+2e-jkπ/2+2 e-jkπ+3e-3jkπ/2(k=0,1,2,3)  
Hence X(0)=6, X(1)=-1-j, X(2)=0, X(3)=-1+j  
The product of these two four point DFTs is  
Y ̂(0)=36, Y ̂(1)=j4, Y ̂(2)=0, Y ̂(3)=-j4  
The four point IDFT yields y ̂(n)={9,7,9,11}  
We can verify as follows  
We know that from the previous question y(n)={1,4,9,11,8,3}  
y ̂(0)=y(0)+y(4)=9  
y ̂(1)=y(1)+y(5)=7  
y ̂(2)=y(2)=9  
y ̂(3)=y(3)=11.

3. Overlap add and Overlap save are the two methods for linear FIR filtering a long sequence on a block-by-block basis using DFT.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: In these two methods, the input sequence is segmented into blocks and each block is processed via DFT and IDFT to produce a block of output data. The output blocks are fitted together to form an overall output sequence which is identical to the sequence obtained if the long block had been processed via time domain convolution. So, Overlap add and Overlap save are the two methods for linear FIR filtering a long sequence on a block-by-block basis using DFT.

4. In Overlap save method of long sequence filtering, what is the length of the input sequence block?  
a) L+M+1  
b) L+M  
c) L+M-1  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: In this method, each data block consists of the last M-1 data points of the previous data block followed by L new data points to form a data sequence of length N=L+M-1.

5. In Overlap save method of long sequence filtering, how many zeros are appended to the impulse response of the FIR filter?  
a) L+M  
b) L  
c) L+1  
d) L-1  
View Answer

Answer: d  
Explanation: The impulse of the FIR filter is increased in length by appending L-1 zeros and an N-point DFT of the sequence is computed once and stored.

6. The first M-1 values of the output sequence in every step of Overlap save method of filtering of long sequence are discarded.  
a) True  
b) False  
View Answer

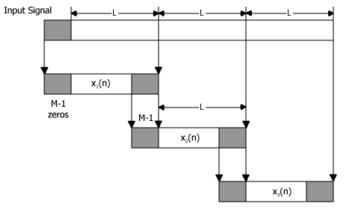
Answer: a  
Explanation: Since the data record of length N, the first M-1 points of ym(n) are corrupted by aliasing and must be discarded. The last L points of ym(n) are exactly as same as the result from linear convolution.

7. In Overlap add method, what is the length of the input data block?  
a) L-1  
b) L  
c) L+1  
d) None of the mentioned  
View Answer

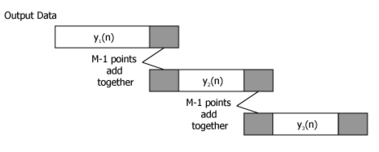
Answer: b  
Explanation: In this method the size of the input data block is L points and the size of the DFTs and IDFT is N=L+M-1.

8. Which of the following is true in case of Overlap add method?  
a) M zeros are appended at last of each data block  
b) M zeros are appended at first of each data block  
c) M-1 zeros are appended at last of each data block  
d) M-1 zeros are appended at first of each data block  
View Answer

Answer: c  
Explanation: In Overlap add method, to each data block we append M-1 zeros at last and compute N point DFT, so that the length of the input sequence is L+M-1=N.

9. In which of the following methods, the input sequence is considered as shown in the below diagram?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-interview-questions-answers-freshers-q9.png)  
a) Overlap save method  
b) Overlap add method  
c) Overlap add & save method  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: From the figure given, we can notice that each data block consists of the last M-1 data points of the previous data block followed by L new data points to form a data sequence of length N+L+M-1 which is same as in the case of Overlap save method.

10. In which of the following methods, the output sequence is considered as shown in the below diagram?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-interview-questions-answers-freshers-q10.png)  
a) Overlap save method  
b) Overlap add method  
c) Overlap add & save method  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: From the figure given, it is clear that the last M-1 points of the first sequence and the first M-1 points of the next sequence are added and nothing is discarded because there is no aliasing in the input sequence. This is same as in the case of Overlap add method.

11. What is the value of x(n)\*h(n), 0≤n≤11 for the sequences x(n)={1,2,0,-3,4,2,-1,1,-2,3,2,1,-3} and h(n)={1,1,1} if we perform using overlap add fast convolution technique?  
a) {1,3,3,1,1,3,5,2,2,2,3,6}  
b) {1,2,0,-3,4,2,-1,1,-2,3,2,1,-3}  
c) {1,2,0,3,4,2,1,1,2,3,2,1,3}  
d) {1,3,3,-1,1,3,5,2,-2,2,3,6}  
View Answer

Answer: d  
Explanation: Since M=3, we chose the transform length for DFT and IDFT computations as L=2M=23=8.  
Since L=M+N-1, we get N=6.  
According to Overlap add method, we get  
x1′(n)={1,2,0,-3,4,2,0,0} and h'(n)={1,1,1,0,0,0,0,0}  
y1(n)=x1′(n)\*N h'(n) (circular convolution)={1,3,3,-1,1,3,6,2}  
x2′(n)={-1,1,-2,3,2,1,0,0} and h'(n)={1,1,1,0,0,0,0,0}  
y2(n)= x2′(n)\*N h'(n)={-1,0,-2,2,3,6,3,1}  
Thus we get, y(n)= {1,3,3,-1,1,3,5,2,-2,2,3,6}.

12. What is the value of x(n)\*h(n), 0≤n≤11 for the sequences x(n)={1,2,0,-3,4,2,-1,1,-2,3,2,1,-3} and h(n)={1,1,1} if we perform using overlap save fast convolution technique?  
a) {1,3,3,-1,1,3,5,2,-2,2,3,6}  
b) {1,2,0,-3,4,2,-1,1,-2,3,2,1,-3}  
c) {1,2,0,3,4,2,1,1,2,3,2,1,3}  
d) {1,3,3,1,1,3,5,2,2,2,3,6}  
View Answer

Answer: a  
Explanation: Since M=3, we chose the transform length for DFT and IDFT computations as L=2M=23=8.  
Since L=M+N-1, we get N=6.  
According to Overlap save technique, we get  
x1′(n)={0,0,1,2,0,-3,4,2} and h'(n)={1,1,1,0,0,0,0,0}  
=>y1(n)={1,3,3,-1,1,3}  
x2′(n)={4,2,-1,1,-2,3,2,1} and h'(n)={1,1,1,0,0,0,0,0}  
=>y2(n)={5,2,-2,2,3,6}  
=>y(n)= {1,3,3,-1,1,3,5,2,-2,2,3,6}.

Questions & Answers (MCQs) focuses on “Quantization Effects in the Computation of DFT”.

1. The effect of round off errors due to the multiplications performed in the DFT with fixed point arithmetic is known as Quantization error.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Since DFT plays a very important role in many applications of DSP, it is very important for us to know the effect of quantization errors in its computation. In particular, we shall consider the effect of round off errors due to the multiplications performed in the DFT with fixed point arithmetic.

2. What is the model that has been adopt for characterizing round of errors in multiplication?  
a) Multiplicative white noise model  
b) Subtractive white noise model  
c) Additive white noise model  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: Additive white noise model is the model that we use in the statistical analysis of round off errors in IIR and FIR filters.

3. How many quantization errors are present in one complex valued multiplication?  
a) One  
b) Two  
c) Three  
d) Four  
View Answer

Answer: d  
Explanation: We assume that the real and imaginary components of {x(n)} and {WNkn} are represented by ‘b’ bits. Consequently, the computation of product x(n). WNkn requires four real multiplications. Each real multiplication is rounded from 2b bits to b bits and hence there are four quantization errors for each complex valued multiplication.

4. What is the total number of quantization errors in the computation of single point DFT of a sequence of length N?  
a) 2N  
b) 4N  
c) 8N  
d) 12N  
View Answer

Answer: b  
Explanation: Since the computation of single point DFT of a sequence of length N involves N number of complex multiplications, it contains 4N number of quantization errors.

5. What is the range in which the quantization errors due to rounding off are uniformly distributed as random variables if Δ=2-b?  
a) (0,Δ)  
b) (-Δ,0)  
c) (-Δ/2,Δ/2)  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The Quantization errors due to rounding off are uniformly distributed random variables in the range (-Δ/2,Δ/2) if Δ=2-b. This is one of the assumption that is made about the statistical properties of the quantization error.

6. The 4N quantization errors are mutually uncorrelated.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The 4N quantization errors are mutually uncorrelated. This is one of the assumption that is made about the statistical properties of the quantization error.

7. The 4N quantization errors are correlated with the sequence {x(n)}.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: According to one of the assumption that is made about the statistical properties of the quantization error, the 4N quantization errors are uncorrelated with the sequence {x(n)}.

8. How is the variance of the quantization error related to the size of the DFT?  
a) Equal  
b) Inversely proportional  
c) Square proportional  
d) Proportional  
View Answer

Answer: d  
Explanation: We know that each of the quantization has a variance of Δ2/12=2-2b/12.  
The variance of the quantization errors from the 4N multiplications is 4N. 2-2b/12=2-2b(N/3).  
Thus the variance of the quantization error is directly proportional to the size of the DFT.

9. Every fourfold increase in the size N of the DFT requires an additional bit in computational precision to offset the additional quantization errors.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: We know that, the variance of the quantization errors is directly proportional to the size N of the DFT. So, every fourfold increase in the size N of the DFT requires an additional bit in computational precision to offset the additional quantization errors.

10. What is the variance of the output DFT coefficients |X(k)|?  
a) 1/N  
b) 1/2N  
c) 1/3N  
d) 1/4N  
View Answer

Answer: c  
Explanation: We know that the variance of the signal sequence is (2/N)2/12=1/(3N2)  
Now the variance of the output DFT coefficients |X(k)|=N. 1/(3N^2 2) = 1/3N.

11. What is the signal-to-noise ratio?  
a) σX2. σq2  
b) σX2/ σq2  
c) σX2+ σq2  
d) σX2-σq2  
View Answer

Answer: b  
Explanation: The signal-to-noise ratio of a signal, SNR is given by the ratio of the variance of the output DFT coefficients to the variance of the quantization errors.

12. How many number of bits are required to compute the DFT of a 1024 point sequence with a SNR of 30db?  
a) 15  
b) 10  
c) 5  
d) 20  
View Answer

Answer: a  
Explanation: The size of the sequence is N=210. Hence the SNR is  
10log10(σX22/ σq2)=10 log1022b-20  
For an SNR of 30db, we have  
3(2b-20)=30=>b=15 bits.  
Note that 15 bits is the precision for both addition and multiplication.

13. How many number of butterflies are required per output point in FFT algorithm?  
a) N  
b) N+1  
c) 2N  
d) N-1  
View Answer

Answer: d  
Explanation: We find that, in general, there are N/2 in the first stage of FFT, N/4 in the second stage, N?8 in the third state, and so on, until the last stage where there is only one. Consequently, the number of butterflies per output point is N-1.

14. What is the value of the variance of quantization error in FFT algorithm, compared to that of direct computation?  
a) Greater  
b) Less  
c) Equal  
d) Cannot be compared  
View Answer

Answer: c  
Explanation: If we assume that the quantization errors in each butterfly are uncorrelated with the errors in the other butterflies, then there are 4(N-1) errors that affect the output of each point of the FFT. Consequently, the variance of the quantization error due to FFT algorithm is given by  
4(N-1)( Δ2/12)=N(Δ2/3)(approximately)  
Thus, the variance of quantization error due to FFT algorithm is equal to the variance of the quantization error due to direct computation.

15. How many number of bits are required to compute the FFT of a 1024 point sequence with a SNR of 30db?  
a) 11  
b) 10  
c) 5  
d) 20  
View Answer

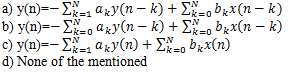
Answer: a  
Explanation: The size of the FFT is N=210. Hence the SNR is 10 log1022b-v-1=30  
=>3(2b-11)=30  
=>b=21/2=11 bits.

#### 7. Questions & Answers on Discrete Time Systems Implementation

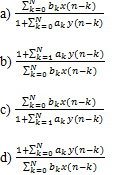
The section contains questions and answers on realization structures for discrete time systems, FIR system structures, IIR system structures, number representation, state space system analysis, quantization error analysis and bilinear transformations.

|  |  |
| --- | --- |
| [Structures for Realization](https://www.sanfoundry.com/digital-signal-processing-questions-answers-structures-realization-discrete-time-systems/) [FIR System Structures 1](https://www.sanfoundry.com/digital-signal-processing-questions-answers-structures-fir-systems-1/) [FIR System Structures 2](https://www.sanfoundry.com/digital-signal-processing-questions-answers-experienced/) [IIR System Structures](https://www.sanfoundry.com/digital-signal-processing-questions-answers-structures-iir-systems/) [State-Space System Analysis](https://www.sanfoundry.com/digital-signal-processing-questions-answers-state-space-system-analysis-structures/) | [Number Representation 1](https://www.sanfoundry.com/digital-signal-processing-questions-answers-representation-numbers-1/) [Number Representation 2](https://www.sanfoundry.com/digital-signal-processing-interview-questions-answers-experienced/) [Discrete-Time Signal Processing](https://www.sanfoundry.com/advanced-digital-signal-processing-questions-answers/) [Quantization Errors Analysis](https://www.sanfoundry.com/digital-signal-processing-questions-answers-analysis-quantization-errors/) [IIR Filter Design](https://www.sanfoundry.com/tricky-digital-signal-processing-questions-answers/) |

Questions & Answers (MCQs) focuses on “Structures for Realization of Discrete Time Systems”.

1. The general linear constant coefficient difference equation characterizing an LTI discrete time system is:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-realization-discrete-time-systems-q1.png)  
View Answer

Answer: a  
Explanation: We know that, the general linear constant coefficient difference equation characterizing an LTI discrete time system is given by the expression.[digital-signal-processing-questions-answers-structures-realization-discrete-time-systems-q1a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-realization-discrete-time-systems-q1a.png)

2. Which of the following is the rational system function of an LTI system characterized by the difference equation [digital-signal-processing-questions-answers-structures-realization-discrete-time-systems-q2](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-realization-discrete-time-systems-q2.png)  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-realization-discrete-time-systems-q2a.png)  
View Answer

Answer: c  
Explanation: The difference equation of the LTI system is given as [digital-signal-processing-questions-answers-structures-realization-discrete-time-systems-q2b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-realization-discrete-time-systems-q2b.png)  
By applying the z-transform on both sides of the above equation and by rearranging the obtained equation, we get the rational system function as  
[digital-signal-processing-questions-answers-structures-realization-discrete-time-systems-q2c](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-realization-discrete-time-systems-q2c.png)

3. We can view [digital-signal-processing-questions-answers-structures-realization-discrete-time-systems-q2b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-realization-discrete-time-systems-q2b.png)as the computational procedure (an algorithm) for determining the output sequence y(n) of the system from the input sequence x(n).  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The computations in the given equation can be arranged into equivalent sets of difference equations. Each set of equations defines a computational procedure or an algorithm for implementing the system.

4. Which of the following is used in the realization of a system?  
a) Delay elements  
b) Multipliers  
c) Adders  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: From each set of equations, we can construct a block diagram consisting of an interconnection of delay elements, multipliers and adders.

5. Computational complexity refers to the number of:  
a) Additions  
b) Arithmetic operations  
c) Multiplications  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: Computational complexity is one of the factor which is used in the implementation of the system. It refers to the numbers of Arithmetic operations (Additions, multiplications and divisions).

6. The number of times a fetch from memory is performed per output sample is one of the factor used in the implementation of the system.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: According to the recent developments in the design and fabrication of rather sophisticated programmable DSPs, other factors, such as the number of times a fetch from memory is performed or the number of times a comparison between two numbers is performed per output sample, have become important in assessing the computational complexity of a given realization of a system.

7. Which of the following refers the number of memory locations required to store the system parameters, past inputs, past outputs and any intermediate computed values?  
a) Computational complexity  
b) Finite world length effect  
c) Memory requirements  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: Memory requirements refers the number of memory locations required to store the system parameters, past inputs, past outputs and any intermediate computed values.

8. Finite word length effects refer to the quantization effects that are inherent in any digital implementation of the system, either in hardware or software.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The parameters of the system must necessarily be represented with finite precision. The computations that are performed in the process of computing an output from the system must be rounded off or truncated to fit within the limited precision constraints of the computer or hardware used in the implementation. Thus, Finite word length effects refer to the quantization effects that are inherent in any digital implementation of the system, either in hardware or software.

9. Which of the following are called as finite word length effects?  
a) Parameters of the system must be represented with finite precision  
b) Computations are truncated to fit in the limited precision constraints  
c) Whether the computations are performed in fixed point or floating point arithmetic  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: All the three of the considerations given above are called as finite word length effects.

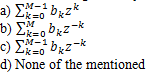
10. The factors Computational complexity, memory requirements and finite word length effects are the ONLY factors influencing our choice of the realization of the system.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: Apart from the three factors given in the question, other factors such as, whether the structure or the realization lends itself to parallel processing or whether the computations can be pipelined are also the factors which influence our choice of the realization of the system.

Questions & Answers (MCQs) focuses on “Structures for FIR Systems-1”.

1. In general, an FIR system is described by the difference equation [digital-signal-processing-questions-answers-structures-fir-systems-1-q1](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-fir-systems-1-q1.png)  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The difference equation a href=”https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-fir-systems-1-q1.png”>digital-signal-processing-questions-answers-structures-fir-systems-1-q1 describes the FIR system.

2. What is the general system function of an FIR system?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-fir-systems-1-q2.png)  
View Answer

Answer: c  
Explanation: We know that the difference equation of an FIR system is given by [digital-signal-processing-questions-answers-structures-fir-systems-1-q2a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-fir-systems-1-q2a.png)

3. Which of the following is an method for implementing an FIR system?  
a) Direct form  
b) Cascade form  
c) Lattice structure  
d) All of the mentioned  
View Answer

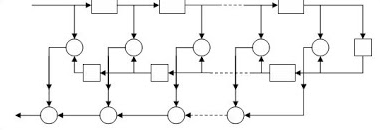
Answer: d  
Explanation: There are several structures for implementing an FIR system, beginning with the simplest structure, called the direct form. There are several other methods like cascade form realization, frequency sampling realization and lattice realization which are used for implementing and FIR system.

4. How many memory locations are used for storage of the output point of a sequence of length M in direct form realization?  
a) M+1  
b) M  
c) M-1  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The direct form realization follows immediately from the non-recursive difference equation given by [digital-signal-processing-questions-answers-structures-fir-systems-1-q1](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-fir-systems-1-q1.png)  
We observe that this structure requires M-1 memory locations for storing the M-1 previous inputs.

5. The direct form realization is often called a transversal or tapped-delay-line filter.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The structure of the direct form realization, resembles a tapped delay line or a transversal system.

6. What is the condition of M, if the structure according to the direct form is as follows?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-fir-systems-1-q6.png)  
a) M even  
b) M odd  
c) All values of M  
d) Doesn’t depend on value of M  
View Answer

Answer: b  
Explanation: When the FIR system has linear phase, the unit sample response of the system satisfies either the symmetry or asymmetry condition, h(n)= ±h(M-1-n)  
For such a system the number of multiplications is reduced from M to M/2 for M even and to (M-1)/2 for M odd. Thus for the structure given in the question, M is odd.

7. By combining two pairs of poles to form a fourth order filter section, by what factor we have reduced the number of multiplications?  
a) 25%  
b) 30%  
c) 40%  
d) 50%  
View Answer

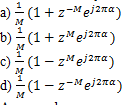
Answer: d  
Explanation: We have to do 3 multiplications for every second order equation. So, we have to do 6 multiplications if we combine two second order equations and we have to perform 3 multiplications by directly calculating the fourth order equation. Thus the number of multiplications are reduced by a factor of 50%.

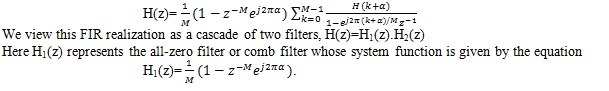
8. The desired frequency response is specified at a set of equally spaced frequencies defined by the equation:  
a) π/2M(k+α)  
b) π/M(k+α)  
c) 2π/M(k+α)  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: To derive the frequency sampling structure, we specify the desired frequency response at a set of equally spaced frequencies, namely ωk=2π/M(k+α) ,k=0,1…(M-1)/2 for M odd  
k=0,1….(M/2)-1 for M even  
α=0 or 0.5.

9. The realization of FIR filter by frequency sampling realization can be viewed as cascade of how many filters?  
a) Two  
b) Three  
c) Four  
d) None of the mentioned  
View Answer

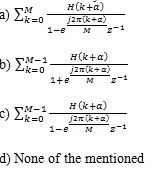
Answer: a  
Explanation: In frequency sampling realization, the system function H(z) is characterized by the set of frequency samples {H(k+ α)} instead of {h(n)}. We view this FIR filter realization as a cascade of two filters. One is an all-zero or a comb filter and the other consists of parallel bank of single pole filters with resonant frequencies.

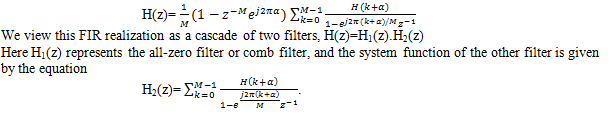
10. What is the system function of all-zero filter or comb filter?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-fir-systems-1-q10.png)  
View Answer

Answer: d  
Explanation: The system function H(z) which is characterized by the set of frequency samples is obtained as  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-fir-systems-1-q10a.png)

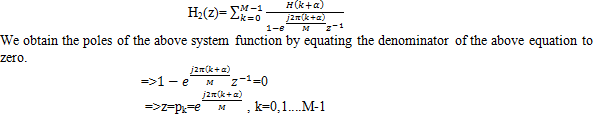
11. The zeros of the system function of comb filter are located:  
a) Inside unit circle  
b) On unit circle  
c) Outside unit circle  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The system function of the comb filter is given by the equation  
[digital-signal-processing-questions-answers-structures-fir-systems-1-q11](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-fir-systems-1-q11.png)

12. What is the system function of the second filter other than comb filter in the realization of FIR filter?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-fir-systems-1-q12.png)  
View Answer

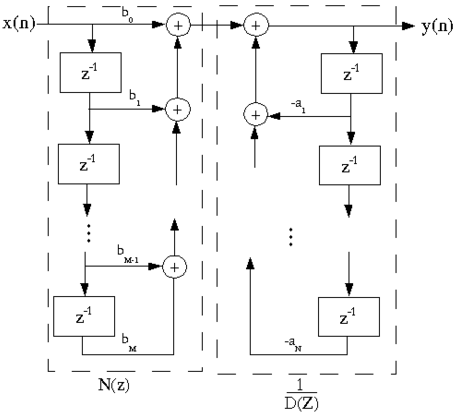
Answer: c  
Explanation: The system function H(z) which is characterized by the set of frequency samples is obtained as  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-fir-systems-1-q12a.png)

13. Where does the poles of the system function of the second filter locate?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-fir-systems-1-q13.png)  
View Answer

Answer: b  
Explanation: The system function of the second filter in the cascade of an FIR realization by frequency sampling method is given by  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-fir-systems-1-q13a.png)

14. When the desired frequency response characteristic of the FIR filter is narrowband, most of the gain parameters {H(k+α)} are zero.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: When the desired frequency response characteristic of the FIR filter is narrowband, most of the gain parameters {H(k+α)} are zero. Consequently, the corresponding resonant filters can be eliminated and only the filters with nonzero gains need be retained.

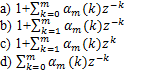
15. Which of the following filters have a cascade realization as shown below?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-fir-systems-1-q15.png)  
a) IIR filter  
b) Comb filter  
c) High pass filter  
d) FIR filter  
View Answer

Answer: d  
Explanation: The system function of the FIR filter according to the frequency sampling realization is given by the equation  
[digital-signal-processing-questions-answers-structures-fir-systems-1-q15a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-fir-systems-1-q15a.png)  
The above system function can be represented in the cascade form as shown in the above block diagram.

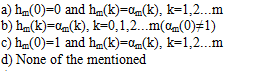
Questions & Answers for experienced focuses on “Structures for FIR Systems”.

1. Which of the following is the application of lattice filter?  
a) Digital speech processing  
b) Adaptive filter  
c) Electroencephalogram  
d) All of the mentioned  
View Answer

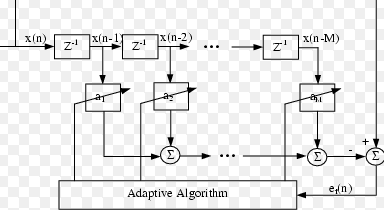
Answer: d  
Explanation: Lattice filters are used extensively in digital signal processing and in the implementation of adaptive filters.

2. If we consider a sequence of FIR filer with system function Hm(z)=Am(z), then what is the definition of the polynomial Am(z)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-experienced-q2.png)  
View Answer

Answer: b  
Explanation: Consider a sequence of FIR filer with system function Hm(z)=Am(z), m=0,1,2…M-1  
where, by definition, Am(z) is the polynomial  
[digital-signal-processing-questions-answers-experienced-q2a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-experienced-q2a.png)

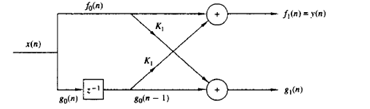
3. What is the unit sample response of the mth filter?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-experienced-q3.png)  
View Answer

Answer: c  
Explanation: We know that Hm(z)=Am(z) and Am(z) is a polynomial whose equation is given as [digital-signal-processing-questions-answers-experienced-q3a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-experienced-q3a.png)

4. The FIR filter whose direct form structure is as shown below is a prediction error filter.  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-experienced-q4.png)  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The FIR structure shown in the above figure is intimately related with the topic of linear prediction. Thus the top filter structure shown in the above figure is called a prediction error filter.

5. What is the output of the single stage lattice filter if x(n) is the input?  
a) x(n)+Kx(n+1)  
b) x(n)+Kx(n-1)  
c) x(n)+Kx(n-1)+Kx(n+1)  
d) Kx(n-1)  
View Answer

Answer: b  
Explanation: The single stage lattice filter is as shown below.  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-experienced-q5.png)  
Here both the inputs are excited and output is selected from the top branch.  
Thus the output of the single stage lattice filter is given by y(n)= x(n)+Kx(n-1).

6. What is the output from the second stage lattice filter when two single stage lattice filers are cascaded with an input of x(n)?  
a) K1K2x(n-1)+K2x(n-2)  
b) x(n)+K1x(n-1)  
c) x(n)+K1K2x(n-1)+K2x(n-2)  
d) x(n)+K1(1+K2)x(n-1)+K2x(n-2)  
View Answer

Answer: d  
Explanation: When two single stage lattice filters are cascaded, then the output from the first filter is given by the equation  
f1(n)= x(n)+K1x(n-1)  
g1(n)=K1x(n)+x(n-1)  
The output from the second filter is obtained as  
f2(n)=f1(n)+K2g1(n-1)  
=x(n)+K2[K1x(n-1)+x(n-2)]+ K1x(n-1)  
= x(n)+K1(1+K2)x(n-1)+K2x(n-2).

7. What is the value of the coefficient α2(1) in the case of FIR filter represented in direct form structure with m=2 in terms of K1 and K2?  
a) K1(K2)  
b) K1(1-K2)  
c) K1(1+K2)  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The equation for the output of an FIR filter represented in the direct form structure is given as  
y(n)=x(n)+ α2(1)x(n-1)+ α2(2)x(n-2)  
The output from the double stage lattice structure is given by the equation,  
f2(n)= x(n)+K2(1+K2)x(n-1)+K2x(n-2)  
By comparing the coefficients of both the equations, we get  
α2(1)= K1(1+K2).

8. The constants K1 and K2 of the lattice structure are called as reflection coefficients.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The equation of the output from the second stage lattice filter is given by  
f2(n)= x(n)+K1(1+K2)x(n-1)+K2x(n-2)  
In the above equation, the constants K1 and K2 are called as reflection coefficients.

9. If a three stage lattice filter with coefficients K1=1/4, K2=1/2 K3=1/3, then what are the FIR filter coefficients for the direct form structure?  
a) (1,8/24,5/8,1/3)  
b) (1,5/8,13/24,1/3)  
c) (1/4,13/24,5/8,1/3)  
d) (1,13/24,5/8,1/3)  
View Answer

Answer: d  
Explanation: We get the output from the third stage lattice filter as  
A3(z)=1+(13/24)z-1+(5/8)z-2+(1/3)z-3.  
Thus the FIR filter coefficients for the direct form structure are (1,13/24,5/8,1/3).

10. What are the lattice coefficients corresponding to the FIR filter with system function H(z)= 1+(13/24)z-1+(5/8)z-2+(1/3)z-3?  
a) (1/2,1/4,1/3)  
b) (1,1/2,1/3)  
c) (1/4,1/2,1/3)  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: Given the system function of the FIR filter is  
H(z)= 1+(13/24)z-1+(5/8)z-2+(1/3)z-3  
Thus the lattice coefficients corresponding to the given filter is (1/4,1/2,1/3).

Questions & Answers (MCQs) focuses on “Structures for IIR Systems”.

1. If M and N are the orders of numerator and denominator of rational system function respectively, then how many multiplications are required in direct form-I realization of that IIR filter?  
a) M+N-1  
b) M+N  
c) M+N+1  
d) M+N+2  
View Answer

Answer: c  
Explanation: From the direct form-I realization of the IIR filter, if M and N are the orders of numerator and denominator of rational system function respectively, then M+N+1 multiplications are required.

2. If M and N are the orders of numerator and denominator of rational system function respectively, then how many additions are required in direct form-I realization of that IIR filter?  
a) M+N-1  
b) M+N  
c) M+N+1  
d) M+N+2  
View Answer

Answer: b  
Explanation: From the direct form-I realization of the IIR filter, if M and N are the orders of numerator and denominator of rational system function respectively, then M+N additions are required.

3. If M and N are the orders of numerator and denominator of rational system function respectively, then how many memory locations are required in direct form-I realization of that IIR filter?  
a) M+N+1  
b) M+N  
c) M+N-1  
d) M+N-2  
View Answer

Answer: a  
Explanation: From the direct form-I realization of the IIR filter, if M and N are the orders of numerator and denominator of rational system function respectively, then M+N+1 memory locations are required.

4. In direct form-I realization, all-pole system is placed before the all-zero system.  
a) True  
b) False  
View Answer

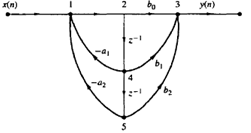
Answer: b  
Explanation: In direct form-I realization, all-zero system is placed before the all-pole system.

5. If M and N are the orders of numerator and denominator of rational system function respectively, then how many memory locations are required in direct form-II realization of that IIR filter?  
a) M+N+1  
b) M+N  
c) Min [M,N].  
d) Max [M,N].  
View Answer

Answer: d  
Explanation: From the direct form-II realization of the IIR filter, if M and N are the orders of numerator and denominator of rational system function respectively, then Max[M,N] memory locations are required.

6. The basic elements of a flow graph are branches and nodes.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: A signal flow graph provides an alternative, but an equivalent graphical representation to a block diagram structure that we have been using to illustrate various system realization. The basic elements of a flow graph are branches and nodes.

7. Which of the following is true for the given signal flow graph?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-iir-systems-q7.png)  
a) Two pole system  
b) Two zero system  
c) Two pole and two zero system  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The equivalent filter structure of the given signal flow graph in the direct form-II is given by as  
Thus from the above structure, the system has two zeros and two poles.

8. What are the nodes that replace the adders in the signal flow graphs?  
a) Source node  
b) Sink node  
c) Branch node  
d) Summing node  
View Answer

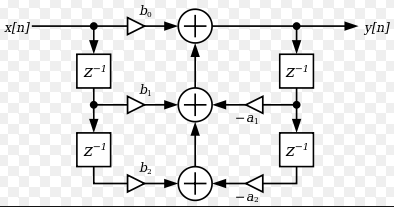
Answer: d  
Explanation: Summing node is the node which is used in the signal flow graph which replaces the adder in the structure of a filter.

9. The output signal of a system is extracted at a sink node.  
a) True  
b) False  
View Answer

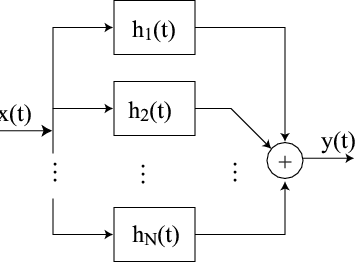
Answer: a  
Explanation: The input to a system originates at a source node and the output signal is extracted at a sink node.

10. If we reverse the directions of all branch transmittances and interchange the input and output in the flow graph, then the resulting structure is called as:  
a) Direct form-I  
b) Transposed form  
c) Direct form-II  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: According to the transposition or flow-graph reversal theorem, if we reverse the directions of all branch transmittances and interchange the input and output in the flow graph, then the system remains unchanged. The resulting structure is known as transposed structure or transposed form.

11. What does the structure given below represents?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-iir-systems-q11.png)  
a) Direct form-I  
b) Regular Direct form-II  
c) Transposed direct form-II  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The structure given in the question is the transposed direct form-II structure of a two pole and two zero IIR system.

12. The structure shown below is known as:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-structures-iir-systems-q12.png)  
a) Parallel form structure  
b) Cascade structure  
c) Direct form  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: From the given figure, it consists of a parallel bank of single pole filters and thus it is called as parallel form structure.

Questions & Answers (MCQs) focuses on “State Space System Analysis and Structures”.

1. The state space or the internal description of the system still involves a relationship between the input and output signals, what are the additional set of variables it also involves?  
a) System variables  
b) Location variables  
c) State variables  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: Although the state space or the internal description of the system still involves a relationship between the input and output signals, it also involves an additional set of variables, called State variables.

2. State variables provide information about all the internal signals in the system.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The state variables provide information about all the internal signals in the system. As a result, the state-space description provides a more detailed description of the system than the input-output description.

3. Which of the following gives the complete definition of the state of a system at time n0?  
a) Amount of information at n0 determines output signal for n≥n0  
b) Input signal x(n) for n≥n0 determines output signal for n≥n0  
c) Input signal x(n) for n≥0 determines output signal for n≥n0  
d) Amount of information at n0+input signal x(n) for n≥n0 determines output signal for n≥n0  
View Answer

Answer: d  
Explanation: We define the state of a system at time n0 as the amount of information that must be provided at time n0, which, together with the input signal x(n) for n≥n0 determines output signal for n≥n0.

4. From the definition of state of a system ,the system consists of only one component called memory less component.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: According to the definition of state of a system, the system consists of two components called memory component and memory less component.

5. If we interchange the rows and columns of the matrix F, then the system is called as:  
a) Identity system  
b) Diagonal system  
c) Transposed system  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The transpose of the matrix F is obtained by interchanging its rows and columns, and it is denoted by FT. The system thus obtained is known as Transposed system.

6. A single input-single output system and its transpose have identical impulse responses and hence the same input-output relationship.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: If h(n) is the impulse response of the single input-single output system, and h1(n) is the impulse response of the transposed system, then we know that h(n)=h1(n). Thus, a single input-single output system and its transpose have identical impulse responses and hence the same input-output relationship.

7. A closed form solution of the state space equations is easily obtained when the system matrix F is:  
a) Transpose  
b) Symmetric  
c) Identity  
d) Diagonal  
View Answer

Answer: d  
Explanation: A closed form solution of the state space equations is easily obtained when the system matrix F is diagonal. Hence, by finding a matrix P so that F1=PFP-1 is diagonal, the solution of the state equations is simplified considerably.

8. What is the condition to call a number λ is an Eigen value of F and a nonzero vector U is the associated Eigen vector?  
a) (F+ λI)U=0  
b) (F- λI)U=0  
c) F- λI=0  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: A number λ is an Eigen value of F and a nonzero vector U is the associated Eigen vector if  
FU= λU  
Thus, we obtain (F- λI)U=0.

9. The determinant |F- λI|=0 yields the characteristic polynomial of the matrix F.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: We know that (F- λI)U=0  
The above equation has a nonzero solution U if the matrix F- λI is singular, which is the case if the determinant of (F- λI) is zero. That is, |F- λI|=0.  
This determinant yields the characteristic polynomial of the matrix F.

10. The parallel form realization is also known as normal form representation.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The parallel form realization is also known as normal form representation, because the matrix F is diagonal, and hence the state variables are uncoupled.

ely to represent ‘5’ in binary floating point representation?  
a) 011,0.110000  
b) 0.110000,011  
c) 011,0.101000  
d) 0.101000,011  
View Answer

Answer: d  
Explanation: We can represent 5 as  
5=0.625\*8=0.625\*23  
The above number can be represented in binary float point representation as 0.101000\*2011  
Thus Mantissa=0.101000, Exponent=011.

6. If the two numbers are to be multiplied, the mantissas are multiplied and the exponents are added.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Let us consider two numbers X=M.2E and Y=N.2F  
If we multiply both X and Y, we get X.Y=(M.N).2E+F  
Thus if we multiply two numbers, the mantissas are multiplied and the exponents are added.

7. What is the smallest floating point number that can be represented using a 32-bit word?  
a) 3\*10-38  
b) 2\*10-38  
c) 0.2\*10-38  
d) 0.3\*10-38  
View Answer

Answer: d  
Explanation: Let the mantissa be represented by 23 bits plus a sign bit and let the exponent be represented by 7 bits plus a sign bit.  
Thus, the smallest floating point number that can be represented using the 32 bit number is  
(1/2)\*2-127=0.3\*10-38  
Thus, the smallest floating point number that can be represented using the 32 bit number is  
(1-2-23)\*2 127/sup>=1.7\*1038.

8. If 0<E<255, then which of the following statement is true about X?  
a) Fractional number  
b) Infinity  
c) Mixed number  
d) Zero  
View Answer

Answer: c  
Explanation: According to the IEEE 754 standard, for a 32-bit machine, single precision floating point number is represented as X=(-1)c(M).  
From the above equation we can interpret that,  
If 0<E<255, then X=(-1)s.2E-127(1.M)=>X is a mixed number.

9. For a twos complement representation, the truncation error is:  
a) Always positive  
b) Always negative  
c) Zero  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: For a two’s complement representation, the truncation error is always negative and falls in the range  
-(2-b-2-bm) ≤ Et ≤ 0.

10. Due to non-uniform resolution, the corresponding error in a floating point representation is proportional to the number being quantized.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: In floating point representation, the mantissa is either rounded or truncated. Due to non-uniform resolution, the corresponding error in a floating point representation is proportional to the number being quantized.

Questions & Answers focuses on “Representation of Numbers”.

1. What is the binary equivalent of (-3/8)?  
a) (10011)2  
b) (0011)2  
c) (1100)2  
d) (1101)2  
View Answer

Answer: d  
Explanation: The number (-3/8) is stored in the computer as the 2’s complement of (3/8)  
We know that the binary equivalent of (3/8)=0011  
Thus the twos complement of 0011=1101.

2. Which of the following is the correct representation of a floating point number X?  
a) 2E  
b) M.2E(1/2<M<1 )  
c) 2M.2E(1/2<M<1 )  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The binary floating point representation commonly used in practice, consists of a mantissa M, which is the fractional part of the number and falls in the range 1/2<M<1, multiplied by the exponential factor 2E, where the exponent E is either a negative or positive integer. Hence a number X is represented as X= M.2E(1/2<M<1).

3. What is the mantissa and exponent respectively obtained when we add 5 and 3/8 in binary float point representation?  
a) 0.101010,011  
b) 0.101000,011  
c) 0.101011,011  
d) 0.101011,101  
View Answer

Answer: c  
Explanation: We can represent the numbers in binary float point as  
5=0.101000(2011)  
3/8=0.110000(2101)=0.000011(2011)  
=>5+3/8=(0.101000+0.000011)(2011)=(0.101011)(2011)  
Therefore mantissa=0.101011 and exponent=011.

4. What is the largest floating point number that can be represented using a 32-bit word?  
a) 3\*1038  
b) 1.7\*1038  
c) 0.2\*1038  
d) 0.3\*1038  
View Answer

Answer: b  
Explanation: Let the mantissa be represented by 23 bits plus a sign bit and let the exponent be represented by 7 bits plus a sign bit.

5. If E=0 and M=0, then which of the following statement is true about X?  
a) Not a number  
b) Infinity  
c) Defined  
d) Zero  
View Answer

Answer: d  
Explanation: According to the IEEE 754 standard, for a 32-bit machine, single precision floating point number is represented as X=(-1)s.2E-127(M).  
From the above equation we can interpret that,  
If E=0 and M=0, then the value of X is 0.

6. The truncation error for the sign magnitude representation is symmetric about zero.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The truncation error for the sign magnitude representation is symmetric about zero and falls in the range  
-(2-b-2-bm) ≤ Et ≤ (2-b-2-bm).

7. What is the range of round-off error for a foxed point representation?  
a) [-0.5(2-b+2-bm), 0.5(2-b+2-bm)].  
b) [0, (2-b+2-bm)].  
c) [0, (2-b-2-bm)].  
d) [-0.5(2-b-2-bm), 0.5(2-b-2-bm-bm)].  
View Answer

Answer: d  
Explanation: The round-off error is independent of the type of fixed point representation. The maximum error that can be introduced through rounding is 0.5(2-b-2-bm) and this can be either positive or negative, depending on the value of x. Therefore, the round-off error is symmetric about zero and falls in the range  
[-0.5(2-b-2-bm), 0.5(2-b-2-bm-bm)].

8. What is the 2’s complement of (1100)2?  
a) (0100)2  
b) (0011)2  
c) (0111)2  
d) (1100)2  
View Answer

Answer: a  
Explanation: a  
Explanation: The ones complement of (1100)2 is (0011)2. Thus the two complement of this number is obtained as (0011)2+(0001)2=(0100)2.

9. The binary digit b-A is called as:  
a) LSB  
b) Total value  
c) MSB  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: Since the binary digit b-A is the first bit in the representation of the real number, it is called as the most significant bit(MSB) of the number.

10. If E=255 and M≠0, then which of the following statement is true about X?  
a) Not a number  
b) Infinity  
c) Defined  
d) Zero  
View Answer

Answer: a  
Explanation: According to the IEEE 754 standard, for a 32-bit machine, single precision floating point number is represented as X=(-1)s.2E-127(M).  
From the above equation we can interpret that,  
If E=255 and M≠0, then X is not a number.

Questions & Answers focuses on “Discrete-Time Processing of Continuous Time Signals”.

1. When the frequency band is selected we can specify the sampling rate and the characteristics of the pre filter, which is also called as \_\_\_\_\_\_\_\_\_\_ filter.  
a) Analog filter  
b) Anti aliasing filter  
c) Analog & Anti aliasing filter  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: Once the desired frequency band is selected w e can specify the sampling rate and the characteristics of the pre filter, which is also called an anti aliasing filter. The anti aliasing filter is an analog filter which has a twofold purpose.

2. What are the main characteristics of Anti aliasing filter?  
a) Ensures that bandwidth of signal to be sampled is limited to frequency range  
b) To limit the additive noise spectrum and other interference, which corrupts the signal  
c) All of the mentioned  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: T he anti aliasing filter is an analog filter which has a twofold purpose. First, it ensures that the bandwidth of the signal to be sampled is limited to the desired frequency range. Using an antialiasing filter is to limit the additive noise spectrum and other interference, which often corrupts the desired signal. Usually, additive noise is wideband and exceeds the bandwidth of the desired signal.

3. In general, a digital system designer has better control of tolerances in a digital signal processing system than an analog system designer who is designing an equivalent analog system.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Analog signal processing operations cannot be done very precisely either, since electronic components in analog systems have tolerances and they introduce noise during their operation. In general, a digital system designer has better control of tolerances in a digital signal processing system than an analog system designer who is designing an equivalent analog system.

4. The selection o f the sampling rate Fs=1/T, where T is the sampling interval, not only determines the highest frequency (Fs/2) that is preserved in the analog signal, but also serves as a scale factor that influences the design specifications for digital filters  
a) True  
b) False  
View Answer

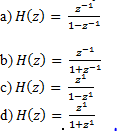
Answer: a  
Explanation: Once we have specified the pre filter requirements and have selected the desired  
sampling rate, w e can proceed with the design of the digital signal processing operations to be performed on the discrete-time signal. The selection of the sampling rate Fs=1/T , where T is the sampling interval, not only determines the highest frequency (Fs/2) that is preserved in the analog signal, but also serves as a scale factor that influences the design specifications for digital filters and any other  
discrete-time systems through which the signal is processed.

5. What is the configuration of system for digital processing of an analog signal?  
a) Analog signal|| Pre-filter ->D/A Converter -> Digital Processor -> A/D Converter -> Post-filter  
b) Analog signal|| Pre-filter ->A/D Converter -> Digital Processor -> D/A Converter -> Post-filter  
c) Analog signal|| Post-filter ->D/A Converter -> Digital Processor -> A/D Converter -> Pre-filter  
d) None of the mentioned  
View Answer

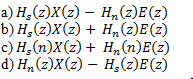
Answer: b  
Explanation: The anti-aliasing filter is an analog filter which has a twofold purpose.  
Analog signal|| Pre-filter ->A/D Converter -> Digital Processor -> D/A Converter -> Post-filter

6. In DM, further the two integrators at encode are replaced by one integrator placed before comparator, and then such system is called?  
a) System-delta modulation  
b) Sigma-delta modulation  
c) Source-delta modulation  
d) None of the mentioned  
View Answer

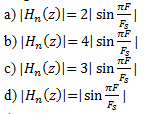
Answer: b  
Explanation: In DM, Furthermore, the two integrators at the encoder can be replaced by a single integrator placed before the comparator. This system is known as sigma-delta modulation (SDM ).

7. What is the system function of the integrator that is modeled by the discrete time system?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/advanced-digital-signal-processing-questions-answers-q7.png)  
View Answer

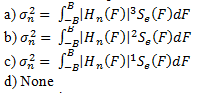
Answer: a  
Explanation: The integrator is modeled by the discrete time system with system function  
[advanced-digital-signal-processing-questions-answers-q7a](https://www.sanfoundry.com/wp-content/uploads/2015/10/advanced-digital-signal-processing-questions-answers-q7a.png)

8. What is the z-transform of sequence dq(n) i.e., Dq(z)= ?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/advanced-digital-signal-processing-questions-answers-q8.png)  
View Answer

Answer: b  
Explanation: [advanced-digital-signal-processing-questions-answers-q8a](https://www.sanfoundry.com/wp-content/uploads/2015/10/advanced-digital-signal-processing-questions-answers-q8a.png)

9.The performance of the SDM system is determined by the noise system function Hn(z), which has a magnitude of?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/advanced-digital-signal-processing-questions-answers-q9.png)  
View Answer

Answer: a  
Explanation: The performance o f the SDM system is therefore determined by the noise system function H\_( n) (z), which has a magnitude frequency response: [advanced-digital-signal-processing-questions-answers-q9a](https://www.sanfoundry.com/wp-content/uploads/2015/10/advanced-digital-signal-processing-questions-answers-q9a.png)

10. The in-band quantization noise variance is given as?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/advanced-digital-signal-processing-questions-answers-q10.png)  
View Answer

Answer: b  
Explanation:  
[advanced-digital-signal-processing-questions-answers-q10a](https://www.sanfoundry.com/wp-content/uploads/2015/10/advanced-digital-signal-processing-questions-answers-q10a.png)

Questions & Answers (MCQs) focuses on “Analysis of Quantization Errors”.

1. If the input analog signal is within the range of the quantizer, the quantization error e\_q (n) is bounded in magnitude i.e.,

| e\_q (n) | < ∆/2

and the resulting error is called?  
a) Granular noise  
b) Overload noise  
c) Particulate noise  
d) Heavy noise  
View Answer

Answer: a  
Explanation: In the statistical approach, we assume that the quantization error is random in nature. We model this error as noise that is added to the original (unquantized) signal. If the input analog signal is within the range of the quantizer, the quantization error e\_q (n) is bounded in magnitude  
i.e.,

| e\_q (n) | < ∆/2

and the resulting error is called Granular noise.

2. If the input analog signal falls outside the range of the quantizer (clipping), e\_q (n) becomes unbounded and results in \_\_\_\_\_\_\_\_\_\_\_\_\_  
a) Granular noise  
b) Overload noise  
c) Particulate noise  
d) Heavy noise  
View Answer

Answer: b  
Explanation: In the statistical approach, we assume that the quantization error is random in nature. We model this error as noise that is added to the original (unquantized) signal. If the input analog signal falls outside the range of the quantizer (clipping), e\_q (n) becomes unbounded and results in overload noise.

3. In the mathematical model for the quantization error e\_q (n), to carry out the analysis, what are the assumptions made about the statistical properties of e\_q (n)?  
 1. The error e\_q (n) is uniformly distributed over the range — ∆ /2 < e\_q (n) < ∆ /2.  
 2. The error sequence is a stationary white noise sequence. In other words, the error e\_q (m) and the error e\_q (n) for m≠n are uncorrelated.  
 3. The error sequence {e\_q (n)} is uncorrelated with the signal sequence x(n).  
 4. The signal sequence x(n) is zero mean and stationary.  
a) 1, 2 & 3  
b) 1,2,3,4  
c) 1, 3  
d) 2, 3, 4  
View Answer

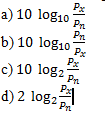
Answer: b  
Explanation: In the mathematical model for the quantization error e\_q (n). To carry out the analysis, the following are the assumptions made about the statistical properties of e\_q (n).  
1. The error e\_q (n) is uniformly distributed over the range — ∆ /2 < e\_q (n) < ∆ /2.  
2. The error sequence is a stationary white noise sequence. In other words, the error e\_q (m)and the error e\_q (n) for m≠n are uncorrelated.  
3. The error sequence {e\_q (n)} is uncorrelated with the signal sequence x(n).  
4. The signal sequence x(n) is zero mean and stationary.

4. What is the abbreviation of SQNR?  
a) Signal-to-Quantization Net Ratio  
b) Signal-to-Quantization Noise Ratio  
c) Signal-to-Quantization Noise Region  
d) Signal-to-Quantization Net Region  
View Answer

Answer: b  
Explanation: The effect of the additive noise e\_q (n) on the desired signal can be quantified by evaluating the signal-to-quantization noise (power) ratio (SQNR).

5. What is the scale used for the measurement of SQNR?  
a) DB  
b) db  
c) dB  
d) All of the mentioned  
View Answer

Answer: c  
Explanation: The effect of the additive noise e\_q (n) on the desired signal can be quantified by evaluating the signal-to-quantization noise (power) ratio (SQNR), which can be expressed on a logarithmic scale (in decibels or dB).

6. What is the expression for SQNR which can be expressed in a logarithmic scale?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-quantization-errors-q6.png)  
View Answer

Answer: a  
Explanation: The signal-to-quantization noise (power) ratio (SQNR), which can be expressed on a logarithmic scale (in decibels or dB) :  
SQNR =[digital-signal-processing-questions-answers-analysis-quantization-errors-q6a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-quantization-errors-q6a.png)

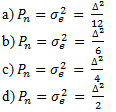
.

7. In the equation SQNR = [digital-signal-processing-questions-answers-analysis-quantization-errors-q6a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-quantization-errors-q6a.png)what are the terms P\_x and P\_n are called \_\_\_ respectively?  
a) Power of the Quantization noise and Signal power  
b) Signal power and power of the quantization noise  
c) None of the mentioned  
d) All of the mentioned  
View Answer

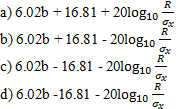
Answer: b  
Explanation: In the equation SQNR =[digital-signal-processing-questions-answers-analysis-quantization-errors-q6a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-quantization-errors-q6a.png) then the terms P\_x is the signal power and P\_n is the power of the quantization noise

8. In the equation SQNR = [digital-signal-processing-questions-answers-analysis-quantization-errors-q6a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-quantization-errors-q6a.png)what are the expressions of P\_x and P\_n ?  
[digital-signal-processing-questions-answers-analysis-quantization-errors-q8](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-quantization-errors-q8.png)  
d) None of the mentioned  
View Answer

Answer: a  
Explanation:[digital-signal-processing-questions-answers-analysis-quantization-errors-q8a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-quantization-errors-q8a.png)

9. If the quantization error is uniformly distributed in the range (-∆ /2, ∆ /2) ,the mean value of the error is zero then the variance P\_n is?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-quantization-errors-q9.png)  
View Answer

Answer: a  
Explanation: [digital-signal-processing-questions-answers-analysis-quantization-errors-q9a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-quantization-errors-q9a.png)

10. By combining [digital-signal-processing-questions-answers-analysis-quantization-errors-q10](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-quantization-errors-q10.png)with P\_n=σ\_(e )^2= ∆^2/12 and substituting the result into SQNR =[digital-signal-processing-questions-answers-analysis-quantization-errors-q6a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-quantization-errors-q6a.png) what is the final expression for SQNR = ?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-quantization-errors-q10a.png)  
View Answer

Answer: b  
Explanation: SQNR  
[digital-signal-processing-questions-answers-analysis-quantization-errors-q10b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-quantization-errors-q10b.png)

11. In the equation SQNR = [digital-signal-processing-questions-answers-analysis-quantization-errors-q11](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-analysis-quantization-errors-q11.png)for R = 6σ\_x the equation becomes?  
a) SQNR = 6.02b-1.25 dB  
b) SQNR = 6.87b-1.55 dB  
c) SQNR = 6.02b+1.25 dB  
d) SQNR = 6.87b+1.25 dB  
View Answer

Answer: c  
Explanation: For example, if we assume that x(n) is Gaussian distributed and the range o f the quantizer extends from -3σ\_x to 3σ\_x (i.e., R = 6σ\_x ), then less than 3 out o f every 1000 input signal amplitudes would result in an overload on the average. For R = 6σ\_x , then the equation becomes  
SQNR = 6.02b+1.25 dB.

Questions & Answers focuses on “IIR Filter Design by the Bilinear Transformation”.

1. In IIR Filter design by the Bilinear Transformation, the Bilinear Transformation is a mapping from  
a) Z-plane to S-plane  
b) S-plane to Z-plane  
c) S-plane to J-plane  
d) J-plane to Z-plane  
View Answer

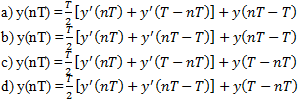
Answer: b  
Explanation: From the equation,  
[tricky-digital-signal-processing-questions-answers-q1](https://www.sanfoundry.com/wp-content/uploads/2015/10/tricky-digital-signal-processing-questions-answers-q1.png)it is clear that transformation occurs from s-plane to z-plane

2. In Bilinear Transformation, aliasing of frequency components is been avoided.  
a) True  
b) False  
View Answer

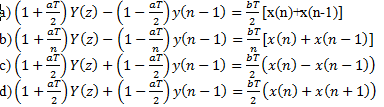
Answer: a  
Explanation: The bilinear transformation is a conformal mapping that transforms the jΩ-axis into the unit circle in the z-plane only once, thus avoiding the aliasing.

3. Is IIR Filter design by Bilinear Transformation is the advanced technique when compared to other design techniques?  
a) True  
b) False  
View Answer

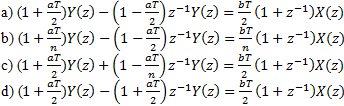
Answer: True  
Explanation: Because in other techniques, only lowpass filters and limited class of bandpass filters are been supported. But this technique overcomes the limitations of other techniques and supports more.

4. The approximation of the integral in [tricky-digital-signal-processing-questions-answers-q4](https://www.sanfoundry.com/wp-content/uploads/2015/10/tricky-digital-signal-processing-questions-answers-q4.png)by the Trapezoidal formula at t = nT and t0=nT-T yields equation?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/tricky-digital-signal-processing-questions-answers-q4a.png)  
View Answer

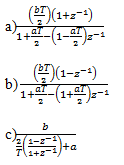
Answer: b  
Explanation: By integrating the equation,  
[tricky-digital-signal-processing-questions-answers-q4b](https://www.sanfoundry.com/wp-content/uploads/2015/10/tricky-digital-signal-processing-questions-answers-q4b.png)

5. We use y'(nT)=-ay(nT)+bx(nT) to substitute for the derivative in[tricky-digital-signal-processing-questions-answers-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/tricky-digital-signal-processing-questions-answers-q5.png)  
and thus obtain a difference equation for the equivalent discrete-time system. With y(n) = y(nT ) and x(n) = x( nT), we obtain the result as \_ of the following?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/tricky-digital-signal-processing-questions-answers-q5a.png)  
View Answer

Answer: a  
Explanation: When we substitute the given equation in the derivative of other we get the resultant required equation.

6. The z-transform of below difference equation is?  
[tricky-digital-signal-processing-questions-answers-q6](https://www.sanfoundry.com/wp-content/uploads/2015/10/tricky-digital-signal-processing-questions-answers-q6.png)  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/tricky-digital-signal-processing-questions-answers-q6a.png)  
View Answer

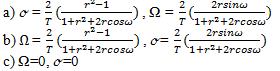
Answer: a  
Explanation: By performing the z-transform of the given equation, we get the required output/equation.

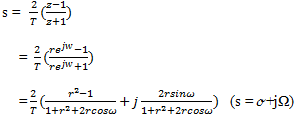
7. What is the system function of the equivalent digital filter? H(z) = Y(z)/X(z ) = ?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/tricky-digital-signal-processing-questions-answers-q7.png)d) Both a and c  
View Answer

Answer: d  
Explanation: As we considered analog linear filter with system function H(s) = b/s+a  
Hence, we got an equivalent system function  
where, [tricky-digital-signal-processing-questions-answers-q1](https://www.sanfoundry.com/wp-content/uploads/2015/10/tricky-digital-signal-processing-questions-answers-q1.png).

8. In the Bilinear Transformation mapping, which of the following are correct?  
a) All points in the LHP of s are mapped inside the unit circle in the z-plane  
b) All points in the RHP of s are mapped outside the unit circle in the z-plane  
c) Both a & b  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The bilinear transformation is a conformal mapping that transforms the jΩ-axis into the unit circle in the z-plane and all the points are linked as mentioned above.

9. In Nth order differential equation, the characteristics of bilinear transformation, let z= rejw,s=o+jΩ Then for[tricky-digital-signal-processing-questions-answers-q1](https://www.sanfoundry.com/wp-content/uploads/2015/10/tricky-digital-signal-processing-questions-answers-q1.png) the values of Ω, ℴ are [](https://www.sanfoundry.com/wp-content/uploads/2015/10/tricky-digital-signal-processing-questions-answers-q9.png)d) None  
View Answer

Answer: a  
Explanation: [](https://www.sanfoundry.com/wp-content/uploads/2015/10/tricky-digital-signal-processing-questions-answers-q9a.png)

10. In equation [tricky-digital-signal-processing-questions-answers-q10](https://www.sanfoundry.com/wp-content/uploads/2015/10/tricky-digital-signal-processing-questions-answers-q10.png)if r < 1 then ℴ < 0 and then mapping from s-plane to z-plane occurs in which of the following order?  
a) LHP in s-plane maps into the inside of the unit circle in the z-plane  
b) RHP in s-plane maps into the outside of the unit circle in the z-plane  
c) All of the mentioned  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: In the above equation, if we substitute the values of r, ℴ then we get mapping in the required way

11. In equation [tricky-digital-signal-processing-questions-answers-q10](https://www.sanfoundry.com/wp-content/uploads/2015/10/tricky-digital-signal-processing-questions-answers-q10.png)if r < 1 then ℴ > 0 and then mapping from s-plane to z-plane occurs in which of the following order?  
a) LHP in s-plane maps into the inside of the unit circle in the z-plane  
b) RHP in s-plane maps into the outside of the unit circle in the z-plane  
c) All of the mentioned  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: In the above equation, if we substitute the values of r, ℴ then we get mapping in the required way

#### 8. Questions on Digital Filters Design

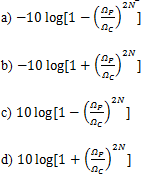
The section contains questions on design of low pass butterworth filters and chebyshev filters, bilinear transformations, filter coefficient quantization, design considerations for filters, FIR filter design using windows, forward and backward difference methods, filter design using frequency sampling method, FIR differentiator design, Hilbert transformer design, IIR filter design, approximation of derivatives, impulse variance, analog filter characteristics, various approximation methods, sampling rate conversion and interpolation techniques.

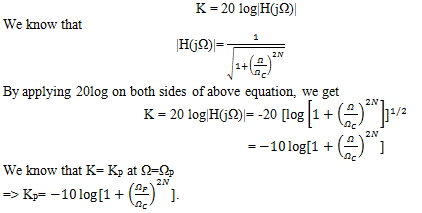
|  |  |
| --- | --- |
| [Butterworth Filters Design 1](https://www.sanfoundry.com/digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1/) [Butterworth Filters Design 2](https://www.sanfoundry.com/digital-signal-processing-test/) [Chebyshev Filters 1](https://www.sanfoundry.com/digital-signal-processing-questions-answers-chebyshev-filters-1/) [Chebyshev Filters 2](https://www.sanfoundry.com/digital-signal-processing-quiz/) [Backward Difference Method](https://www.sanfoundry.com/digital-signal-processing-questions-answers-backward-difference-method/) [Bilinear Transformations](https://www.sanfoundry.com/digital-signal-processing-questions-answers-bilinear-transformations/) [Filter Coefficients Quantization](https://www.sanfoundry.com/digital-signal-processing-questions-answers-quantization-filter-coefficients/) [Digital Filters Round Off Effects](https://www.sanfoundry.com/digital-signal-processing-questions-answers-round-off-effects-digital-filters/) [Digital Filters Design Consideration](https://www.sanfoundry.com/digital-signal-processing-questions-answers-general-considerations-design-digital-filters/) [FIR Filters Design](https://www.sanfoundry.com/digital-signal-processing-questions-answers-design-fir-filters/) [FIR Filters Windows Design 1](https://www.sanfoundry.com/digital-signal-processing-questions-answers-design-linear-phase-fir-filters-using-windows-1/) [FIR Filters Windows Design 2](https://www.sanfoundry.com/digital-signal-processing-mcqs/) [Frequency Sampling Method FIR Design](https://www.sanfoundry.com/digital-signal-processing-multiple-choice-questions-answers/) [Optimum Equi Ripple Filter Design 1](https://www.sanfoundry.com/digital-signal-processing-online-test/) [Optimum Equi Ripple Filter Design 2](https://www.sanfoundry.com/digital-signal-processing-quiz-online/) | [FIR Differentiator Design](https://www.sanfoundry.com/digital-signal-processing-questions-answers-design-fir-differentiators/) [Hilbert Transformers Design](https://www.sanfoundry.com/digital-signal-processing-questions-answers-design-hilbert-transformers/) [FIR Filters Design Comparison](https://www.sanfoundry.com/digital-signal-processing-question-bank/) [Analog Filters Design](https://www.sanfoundry.com/digital-signal-processing-questions-answers-design-iir-filters-analog-filters/) [Approximation of Derivatives design Method](https://www.sanfoundry.com/digital-signal-processing-questions-entrance-exams/) [Impulse Invariance Filter Design](https://www.sanfoundry.com/digital-signal-processing-questions-campus-interviews/) [Matched Z Transformation](https://www.sanfoundry.com/digital-signal-processing-questions-answers-matched-z-transformation/) [Analog Filter Characteristics](https://www.sanfoundry.com/digital-signal-processing-aptitude-test/) [Analog Domain Frequency Transformations](https://www.sanfoundry.com/digital-signal-processing-problems/) [Digital Domain Frequency Transformations](https://www.sanfoundry.com/basic-digital-signal-processing-questions-answers/) [Pade Approximation Method](https://www.sanfoundry.com/digital-signal-processing-questions-answers-pade-approximation-method/) [Least Squares Design Methods](https://www.sanfoundry.com/digital-signal-processing-questions-answers-least-squares-design-methods/) [FIR Least Squares Filters](https://www.sanfoundry.com/digital-signal-processing-questions-answers-fir-least-squares-inverse-filters/) [IIR Frequency Domain Filter Analysis](https://www.sanfoundry.com/digital-signal-processing-questions-answers-design-iir-filters-frequency-domain/) [Analog Filters Classification](https://www.sanfoundry.com/digital-signal-processing-questions-answers-specifications-classification-analog-filters/) [Butterworth Filters](https://www.sanfoundry.com/digital-signal-processing-questions-answers-butterworth-filters/) [Frequency Transformations](https://www.sanfoundry.com/digital-signal-processing-questions-answers-frequency-transformations/) [Factor I Interpolation](https://www.sanfoundry.com/digital-signal-processing-questions-answers-interpolation-factor-i/) [Sampling Rate Conversion](https://www.sanfoundry.com/tough-digital-signal-processing-questions-answers/) |

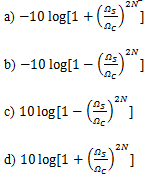
Questions & Answers (MCQs) focuses on “Design of Low Pass Butterworth Filters-1”.

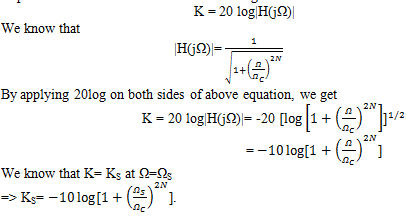
1. Which of the following is a frequency domain specification?  
a) 0≥ 20 log|H(jΩ)|  
b) 20 log|H(jΩ)| ≥ KP  
c) 20 log|H(jΩ)| ≤ KS  
d) All of the mentioned  
View Answer

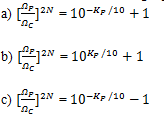
Answer: d  
Explanation: We are required to design a low pass Butterworth filter to meet the following frequency domain specifications.  
KP ≤ 20 log|H(jΩ)| ≤ 0  
and 20 log|H(jΩ)| ≤ KS.

2. What is the value of gain at the pass band frequency, i.e., what is the value of KP?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q2.png)  
View Answer

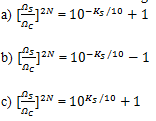
Answer: b  
Explanation: We know that the formula for gain is[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q2a.png)

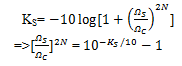
3. What is the value of gain at the stop band frequency, i.e., what is the value of KS?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q3.png)  
View Answer

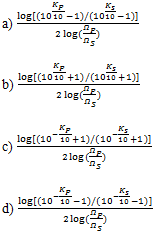
Answer: a  
Explanation: We know that the formula for gain is  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q3a.png)

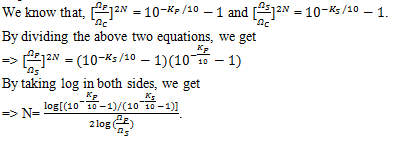
4. Which of the following equation is True?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q4.png)d) None of the mentioned  
View Answer

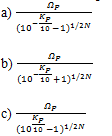
Answer: c  
Explanation: We know that,  
[digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q4a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q4a.png)

5. Which of the following equation is True?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q5.png)d) None of the mentioned  
View Answer

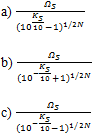
Answer: b  
Explanation: We know that,  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q5a.png)

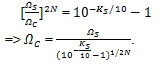
6. What is the order N of the low pass Butterworth filter in terms of KP and KS?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q6.png)  
View Answer

Answer: d  
Explanation:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q6a.png)

7. What is the expression for cutoff frequency in terms of pass band gain?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q7.png)d) None of the mentioned  
View Answer

Answer: a  
Explanation: We know that,  
[digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q7a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q7a.png)

8. What is the expression for cutoff frequency in terms of stop band gain?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q8.png)d) None of the mentioned  
View Answer

Answer: c  
Explanation: We know that,  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q8a.png)

9. The cutoff frequency of the low pass Butterworth filter is the arithmetic mean of the two cutoff frequencies as found above.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The arithmetic mean of the two cutoff frequencies as found above is the final cutoff frequency of the low pass Butterworth filter.

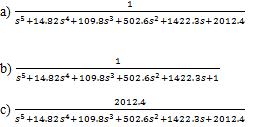
10. What is the lowest order of the Butterworth filter with a pass band gain KP= -1 dB at ΩP= 4 rad/sec and stop band attenuation greater than or equal to 20dB at ΩS= 8 rad/sec?  
a) 4  
b) 5  
c) 6  
d) 3  
View Answer

Answer: b  
Explanation: We know that the equation for the order of the Butterworth filter is given as  
[digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q10](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-low-pass-butterworth-filters-1-q10.png)  
From the given question,  
KP= -1 dB, ΩP= 4 rad/sec, KS= -20 dB and ΩS= 8 rad/sec  
Upon substituting the values in the above equation, we get  
N=4.289  
Rounding off to the next largest integer, we get N=5.

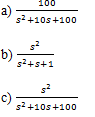
“Design of Low Pass Butterworth Filters”.

1. What is the cutoff frequency of the Butterworth filter with a pass band gain KP= -1 dB at ΩP= 4 rad/sec and stop band attenuation greater than or equal to 20dB at ΩS= 8 rad/sec?  
a) 3.5787 rad/sec  
b) 1.069 rad/sec  
c) 6 rad/sec  
d) 4.5787 rad/sec  
View Answer

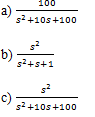
Answer: d  
Explanation: We know that the equation for the cutoff frequency of a Butterworth filter is given as  
[digital-signal-processing-test-q1](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q1.png)  
We know that KP= -1 dB, ΩP= 4 rad/sec and N=5  
Upon substituting the values in the above equation, we get  
Ω\_C = 4.5787 rad/sec.

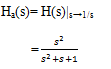
2. What is the system function of the Butterworth filter with specifications as pass band gain KP= -1 dB at ΩP= 4 rad/sec and stop band attenuation greater than or equal to 20dB at ΩS= 8 rad/sec?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q2.png)d) None of the mentioned  
View Answer

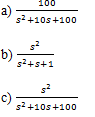
Answer: c  
Explanation: From the given question,  
KP= -1 dB, ΩP= 4 rad/sec, KS= -20 dB and ΩS= 8 rad/sec  
We find out order as N=5 and Ω\_C = 4.5787 rad/sec  
We know that for a 5th order normalized low pass Butterworth filter, system equation is given as  
[digital-signal-processing-test-q2a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q2a.png)  
The specified low pass filter is obtained by applying low pass-to-low pass transformation on the normalized low pass filter.  
That is, Ha(s)=H5(s)|s→s/Ωc  
= H5(s)|s→s/4.5787  
upon calculating, we get option c.

3. If H(s)= 1/(s2+s+1) represent the transfer function of a low pass filter(not Butterworth) with a pass band of 1 rad/sec, then what is the system function of a low pass filter with a pass band 10 rad/sec?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q3.png)d) None of the mentioned  
View Answer

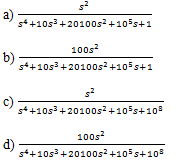
Answer: a  
Explanation: The low pass-to-low pass transformation is  
s→s/Ωu  
Hence the required low pass filter is  
Ha(s)= H(s)|s→s/10  
= 100/(s2+10s+100).

4. If H(s)= 1/(s2+s+1) represent the transfer function of a low pass filter(not Butterworth) with a pass band of 1 rad/sec, then what is the system function of a high pass filter with a cutoff frequency of 1rad/sec?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q3.png)d) None of the mentioned  
View Answer

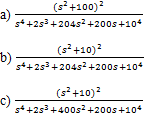
Answer: b  
Explanation: The low pass-to-high pass transformation is  
s→Ωu/s  
Hence the required high pass filter is  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q4.png)

5. If H(s)= 1/(s2+s+1) represent the transfer function of a low pass filter(not Butterworth) with a pass band of 1 rad/sec, then what is the system function of a high pass filter with a cutoff frequency of 10 rad/sec?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q3.png)d) None of the mentioned  
View Answer

Answer: c  
Explanation: The low pass-to-high pass transformation is  
s→Ωu/s  
Hence the required low pass filter is  
[digital-signal-processing-test-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q5.png)

6. If H(s)= 1/(s2+s+1) represent the transfer function of a low pass filter(not Butterworth) with a pass band of 1 rad/sec, then what is the system function of a band pass filter with a pass band of 10 rad/sec and a center frequency of 100 rad/sec?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q6.png)  
View Answer

Answer: d  
Explanation: The low pass-to-band pass transformation is  
[digital-signal-processing-test-q6a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q6a.png)  
Thus the required band pass filter has a transform function as  
[digital-signal-processing-test-q6b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q6b.png)

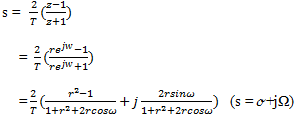
7. If H(s)= 1/(s2+s+1) represent the transfer function of a low pass filter(not Butterworth) with a pass band of 1 rad/sec, then what is the system function of a stop band filter with a stop band of 2 rad/sec and a center frequency of 10 rad/sec?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q7.png)d) None of the mentioned  
View Answer

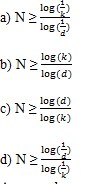
Answer: a  
Explanation: The low pass-to- band stop transformation is  
[digital-signal-processing-test-q7b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q7b.png)  
Hence the required band stop filter is  
[digital-signal-processing-test-q7a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q7a.png)

8. What is the stop band frequency of the normalized low pass Butterworth filter used to design a analog band pass filter with -3.0103dB upper and lower cutoff frequency of 50Hz and 20KHz and a stop band attenuation 20dB at 20Hz and 45KHz?  
a) 2 rad/sec  
b) 2.25 Hz  
c) 2.25 rad/sec  
d) 2 Hz  
View Answer

Answer: c  
Explanation: Given information is  
Ω1=2π\*20=125.663 rad/sec  
Ω2=2π\*45\*103=2.827\*105 rad/sec  
Ωu=2π\*20\*103=1.257\*105 rad/sec  
Ωl=2π\*50=314.159 rad/sec  
We know that  
[digital-signal-processing-test-q8](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q8.png)  
=> A= 2.51 and B=2.25  
Hence ΩS= Min{|A|,|B|}=> ΩS=2.25 rad/sec.

9. What is the order of the normalized low pass Butterworth filter used to design a analog band pass filter with -3.0103dB upper and lower cutoff frequency of 50Hz and 20KHz and a stop band attenuation 20dB at 20Hz and 45KHz?  
a) 2  
b) 3  
c) 4  
d) 5  
View Answer

Answer: b  
Explanation: Given information is  
Ω1=2π\*20=125.663 rad/sec  
Ω2=2π\*45\*103=2.827\*105 rad/sec  
Ωu=2π\*20\*103=1.257\*105 rad/sec  
Ωl=2π\*50=314.159 rad/sec  
We know that  
=> A= 2.51 and B=2.25  
Hence ΩS= Min{|A|,|B|}=> ΩS=2.25 rad/sec.[digital-signal-processing-test-q8](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q8.png)  
The order N of the normalized low pass Butterworth filter is computed as follows  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/tricky-digital-signal-processing-questions-answers-q9a1.png)= 2.83﻿  
Rounding off to the next large integer, we get, N=3.

10. Which of the following condition is true?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q10.png)  
View Answer

Answer: d  
Explanation: If ‘d’ is the discrimination factor and ‘K’ is the selectivity factor, then  
[digital-signal-processing-test-q10a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-test-q10a.png)

Questions & Answers (MCQs) focuses on “Chebyshev Filters-1”.

1. Which of the following defines a chebyshev polynomial of order N, TN(x)?  
a) cos(Ncos-1x) for all x  
b) cosh(Ncosh-1x) for all x  
c) cos(Ncos-1x), |x|≤1  
cosh(Ncosh-1x), |x|>1  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: In order to understand the frequency-domain behavior of chebyshev filters, it is utmost important to define a chebyshev polynomial and then its properties. A chebyshev polynomial of degree N is defined as  
TN(x) = cos(Ncos-1x), |x|≤1  
cosh(Ncosh-1x), |x|>1.

2. What is the formula for chebyshev polynomial TN(x) in recursive form?  
a) 2TN-1(x)- TN-2(x)  
b) 2TN-1(x)+ TN-2(x)  
c) 2xTN-1(x)+ TN-2(x)  
d) 2xTN-1(x)- TN-2(x)  
View Answer

Answer: d  
Explanation: We know that a chebyshev polynomial of degree N is defined as  
TN(x) = cos(Ncos-1x), |x|≤1  
cosh(Ncosh-1x), |x|>1  
From the above formula, it is possible to generate chebyshev polynomial using the following recursive formula  
TN(x)= 2xTN-1(x)- TN-2(x), N ≥ 2.

3. What is the value of chebyshev polynomial of degree 0?  
a) 1  
b) 0  
c) -1  
d) 2  
View Answer

Answer: a  
Explanation: We know that a chebyshev polynomial of degree N is defined as  
TN(x) = cos(Ncos-1x), |x|≤1  
cosh(Ncosh-1x), |x|>1  
For a degree 0 chebyshev filter, the polynomial is obtained as  
T0(x)=cos(0)=1.

4. What is the value of chebyshev polynomial of degree 1?  
a) 1  
b) x  
c) -1  
d) -x  
View Answer

Answer: b  
Explanation: We know that a chebyshev polynomial of degree N is defined as  
TN(x) = cos(Ncos-1x), |x|≤1  
cosh(Ncosh-1x), |x|>1  
For a degree 1 chebyshev filter, the polynomial is obtained as  
T0(x)=cos(cos-1x)=x.

5. What is the value of chebyshev polynomial of degree 3?  
a) 3x3+4x  
b) 3x3-4x  
c) 4x3+3x  
d) 4x3-3x  
View Answer

Answer: d  
Explanation: We know that a chebyshev polynomial of degree N is defined as  
TN(x) = cos(Ncos-1x), |x|≤1 cosh(Ncosh-1x), |x|>1  
And the recursive formula for the chebyshev polynomial of order N is given as  
TN(x)= 2xTN-1(x)- TN-2(x)  
Thus for a chebyshev filter of order 3, we obtain  
T3(x)=2xT2(x)-T1(x)=2x(2x2-1)-x= 4x3-3x.

6. What is the value of chebyshev polynomial of degree 5?  
a) 16x5+20x3-5x  
b) 16x5+20x3+5x  
c) 16x5-20x3+5x  
d) 16x5-20x3-5x  
View Answer

Answer: c  
Explanation: We know that a chebyshev polynomial of degree N is defined as  
TN(x) = cos(Ncos-1x), |x|≤1  
cosh(Ncosh-1x), |x|>1  
And the recursive formula for the chebyshev polynomial of order N is given as  
TN(x)= 2xTN-1(x)- TN-2(x)  
Thus for a chebyshev filter of order 5, we obtain  
T5(x)=2xT4(x)-T3(x)=2x(8x4-8x2+1)-( 4x3-3x )= 16x5-20x3+5x.

7. For |x|≤1, |TN(x)|≤1, and it oscillates between -1 and +1 a number of times proportional to N.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: For |x|≤1, |TN(x)|≤1, and it oscillates between -1 and +1 a number of times proportional to N.  
The above is evident from the equation,  
TN(x) = cos(Ncos-1x), |x|≤1.

8. Chebyshev polynomials of odd orders are:  
a) Even functions  
b) Odd functions  
c) Exponential functions  
d) Logarithmic functions  
View Answer

Answer: b  
Explanation: Chebyshev polynomials of odd orders are odd functions because they contain only odd powers of x.

9. What is the value of TN(0) for even degree N?  
a) -1  
b) +1  
c) 0  
d) ±1  
View Answer

Answer: d  
Explanation: We know that a chebyshev polynomial of degree N is defined as  
TN(x) = cos(Ncos-1x), |x|≤1  
cosh(Ncosh-1x), |x|>1  
For x=0, we have TN(0)=cos(Ncos-10)=cos(N.π/2)=±1 for N even.

10. TN(-x)=(-1)NTN(x)  
a) True  
b) False  
View Answer

Answer: a  
Explanation: We know that a chebyshev polynomial of degree N is defined as  
TN(x) = cos(Ncos-1x), |x|≤1  
cosh(Ncosh-1x), |x|>1  
=> TN(-x)= cos(Ncos-1(-x))= cos(N(π-cos-1x))= cos(Nπ-Ncos-1x)= (-1)N cos(Ncos-1x)= (-1)NTN(x)  
Thus we get, TN(-x)=(-1)NTN(x).

“Chebyshev Filters”.

1. What is the value of |TN(±1)|?  
a) 0  
b) -1  
c) 1  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: We know that a chebyshev polynomial of degree N is defined as  
TN(x) = cos(Ncos-1x), |x|≤1  
cosh(Ncosh-1x), |x|>1  
Thus |TN(±1)|=1.

2. The chebyshev polynomial is oscillatory in the range |x|<∞.  
a) True  
b) False  
View Answer

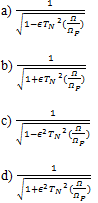
Answer: b  
Explanation: The chebyshev polynomial is oscillatory in the range |x|≤1 and monotonic outside it.

3. If NB and NC are the orders of the Butterworth and Chebyshev filters respectively to meet the same frequency specifications, then which of the following relation is true?  
a) NC=NB  
b) NC<NB  
c) NC>NB  
d) Cannot be determined  
View Answer

Answer: b  
Explanation: The equi-ripple property of the chebyshev filter yields a narrower transition band compared with that obtained when the magnitude response is monotone. As a consequence of this, the order of a chebyshev filter needed to achieve the given frequency domain specifications is usually lower than that of a Butterworth filter.

4. The chebyshev-I filter is equi-ripple in pass band and monotonic in the stop band.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: There are two types of chebyshev filters. The Chebyshev-I filter is equi-ripple in the pass band and monotonic in the stop band and the chebyshev-II filter is quite opposite.

5. What is the equation for magnitude frequency response |H(jΩ)| of a low pass chebyshev-I filter?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-quiz-q5.png)  
View Answer

Answer: d  
Explanation: The magnitude frequency response of a low pass chebyshev-I filter is given by  
[digital-signal-processing-quiz-q5a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-quiz-q5a.png)  
where ϵ is a parameter of the filter related to the ripple in the pass band and TN(x) is the Nth order chebyshev polynomial.

6. What is the number of minima’s present in the pass band of magnitude frequency response of a low pass chebyshev-I filter of order 4?  
a) 1  
b) 2  
c) 3  
d) 4  
View Answer

Answer: b  
Explanation: In the magnitude frequency response of a low pass chebyshev-I filter, the pass band has 2 maxima and 2 minima(order 4=2 maxima+2 minima).

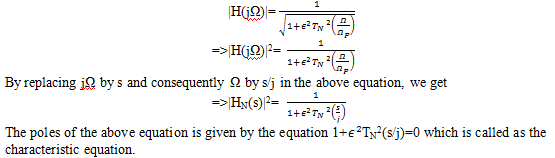
7. What is the number of maxima present in the pass band of magnitude frequency response of a low pass chebyshev-I filter of order 5?  
a) 1  
b) 2  
c) 3  
d) 4  
View Answer

Answer: c  
Explanation: In the magnitude frequency response of a low pass chebyshev-I filter, the pass band has 3 maxima and 2 minima(order 5=3 maxima+2 minima).

8. The sum of number of maxima and minima in the pass band equals the order of the filter.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: In the pass band of the frequency response of the low pass chebyshev-I filter, the sum of number of maxima and minima is equal to the order of the filter.

9. Which of the following is the characteristic equation of a Chebyshev filter?  
[digital-signal-processing-quiz-q9](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-quiz-q9.png)d) None of the mentioned  
View Answer

Answer: a  
Explanation: We know that for a chebyshev filter, we have  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-quiz-q9a.png)

10. The poles of HN(s).HN(-s) are found to lie on:  
a) Circle  
b) Parabola  
c) Hyperbola  
d) Ellipse  
View Answer

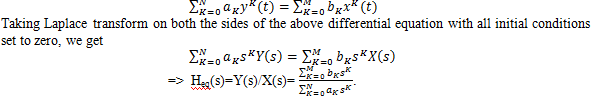
Answer: d  
Explanation: The poles of HN(s).HN(-s) is given by the characteristic equation 1+ϵ2TN2(s/j)=0.  
The roots of the above characteristic equation lies on ellipse, thus the poles of HN(s).HN(-s) are found to lie on ellipse.

11. If the discrimination factor ‘d’ and the selectivity factor ‘k’ of a chebyshev I filter are 0.077 and 0.769 respectively, then what is the order of the filter?  
a) 2  
b) 5  
c) 4  
d) 3  
View Answer

Answer: b  
Explanation: We know that the order of a chebyshev-I filter is given by the equation,  
N=cosh-1(1/d)/cosh-1(1/k)=4.3  
Rounding off to the next large integer, we get N=5.

Questions & Answers (MCQs) focuses on “Backward Difference Method”.

1. The equation for Heq(s) is [digital-signal-processing-questions-answers-backward-difference-method-q1](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-backward-difference-method-q1.png)  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The analog filter in the time domain is governed by the following difference equation,  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-backward-difference-method-q1a.png)

2. What is the first backward difference of y(n)?  
a) [y(n)+y(n-1)]/T  
b) [y(n)+y(n+1)]/T  
c) [y(n)-y(n+1)]/T  
d) [y(n)-y(n-1)]/T  
View Answer

Answer: d  
Explanation: A simple approximation to the first order derivative is given by the first backward difference. The first backward difference is defined by  
[y(n)-y(n-1)]/T.

3. Which of the following is the correct relation between ‘s’ and ‘z’?  
a) z=1/(1+sT)  
b) s=1/(1+zT)  
c) z=1/(1-sT)  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: We know that s=(1-z-1)/T=> z=1/(1-sT).

4. What is the center of the circle represented by the image of jΩ axis of the s-domain?  
a) z=0  
b) z=0.5  
c) z=1  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: Letting s=σ+jΩ in the equation z=1/(1-sT) and by letting σ=0, we get  
|z-0.5|=0.5  
Thus the image of the jΩ axis of the s-domain is a circle with centre at z=0.5 in z-domain.

5. What is the radius of the circle represented by the image of jΩ axis of the s-domain?  
a) 0.75  
b) 0.25  
c) 1  
d) 0.5  
View Answer

Answer: d  
Explanation: Letting s=σ+jΩ in the equation z=1/(1-sT) and by letting σ=0, we get  
|z-0.5|=0.5  
Thus the image of the jΩ axis of the s-domain is a circle of radius 0.5 centered at z=0.5 in z-domain.

6. The frequency response H(ω) will be considerably distorted with respect to H(jΩ).  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Since jΩ axis is not mapped to the circle |z|=1, we can expect that the frequency response H(ω) will be considerably distorted with respect to H(jΩ).

7. The left half of the s-plane is mapped to which of the following in the z-domain?  
a) Outside the circle |z-0.5|=0.5  
b) Outside the circle |z+0.5|=0.5  
c) Inside the circle |z-0.5|=0.5  
d) Inside the circle |z+0.5|=0.5  
View Answer

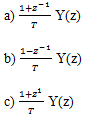
Answer: c  
Explanation: The left half of the s-plane is mapped inside the circle of |z-0.5|=0.5 in the z-plane, which completely lies in the right half z-plane.

8. An analog high pass filter can be mapped to a digital high pass filter.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: An analog high pass filter cannot be mapped to a digital high pass filter because the poles of the digital filter cannot lie in the correct region, which is the left-half of the z-plane(z < 0) in this case.

9. Which of the following is the correct relation between ‘s’ and ‘z’?  
a) s=(1-z-1)/T  
b) s=1/(1+zT)  
c) s=(1+z-1)/T  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: We know that z=1/(1-sT)=> s=(1-z-1)/T.

10. What is the z-transform of the first backward difference equation of y(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-backward-difference-method-q10.png)d) None of the mentioned  
View Answer

Answer: b  
Explanation: The first backward difference of y(n) is given by the equation  
[y(n)-y(n-1)]/T  
Thus the z-transform of the first backward difference of y(n) is given as  
[digital-signal-processing-questions-answers-backward-difference-method-q10a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-backward-difference-method-q10a.png)

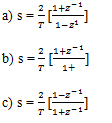
Questions & Answers (MCQs) focuses on “Bilinear Transformations”.

1. Bilinear Transformation is used for transforming an analog filter to a digital filter.  
a) True  
b) False  
View Answer

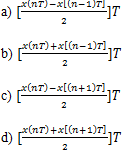
Answer: a  
Explanation: The bilinear transformation can be regarded as a correction of the backward difference method. The bilinear transformation is used for transforming an analog filter to a digital filter.

2. Which of the following rule is used in the bilinear transformation?  
a) Simpson’s rule  
b) Backward difference  
c) Forward difference  
d) Trapezoidal rule  
View Answer

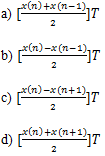
Answer: d  
Explanation: Bilinear transformation uses trapezoidal rule for integrating a continuous time function.

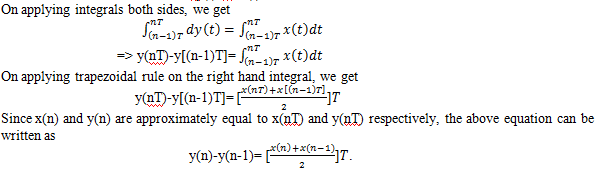
3. Which of the following substitution is done in Bilinear transformations?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-bilinear-transformations-q3.png)d) None of the mentioned  
View Answer

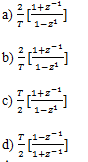
Answer: c  
Explanation: In bilinear transformation of an analog filter to digital filter, using the trapezoidal rule, the substitution for ‘s’ is given as  
[digital-signal-processing-questions-answers-bilinear-transformations-q3a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-bilinear-transformations-q3a.png).

4. What is the value of[digital-signal-processing-questions-answers-bilinear-transformations-q4](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-bilinear-transformations-q4.png)according to trapezoidal rule?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-bilinear-transformations-q4a.png)  
View Answer

Answer: b  
Explanation: The given integral is approximated by the trapezoidal rule. This rule states that if T is small, the area (integral) can be approximated by the mean height of x(t) between the two limits and then multiplying by the width. That is  
[digital-signal-processing-questions-answers-bilinear-transformations-q4b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-bilinear-transformations-q4b.png).

5. What is the value of y(n)-y(n-1) in terms of input x(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-bilinear-transformations-q5.png)  
View Answer

Answer: a  
Explanation: We know that the derivate equation is  
dy(t)/dt=x(t)  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-bilinear-transformations-q5a.png)

6. What is the expression for system function in z-domain?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-bilinear-transformations-q6.png)  
View Answer

Answer: c  
Explanation: We know that  
y(n)-y(n-1)= [(x(n)+x(n-1))/2]T  
Taking z-transform of the above equation gives  
=>Y(z)[1-z-1]=([1+z-1]/2).TX(z)  
=>H(z)=Y(z)/X(z)= T/2[(1+z-1)/(1-z1 )].

7. In bilinear transformation, the left-half s-plane is mapped to which of the following in the z-domain?  
a) Entirely outside the unit circle |z|=1  
b) Partially outside the unit circle |z|=1  
c) Partially inside the unit circle |z|=1  
d) Entirely inside the unit circle |z|=1  
View Answer

Answer: d  
Explanation: In bilinear transformation, the z to s transformation is given by the expression  
z=[1+(T/2)s]/[1-(T/2)s].  
Thus unlike the backward difference method, the left-half s-plane is now mapped entirely inside the unit circle, |z|=1, rather than to a part of it.

8. The equation [digital-signal-processing-questions-answers-bilinear-transformations-q3a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-bilinear-transformations-q3a.png)is a true frequency-to-frequency transformation.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Unlike the backward difference method, the left-half s-plane is now mapped entirely inside the unit circle, |z|=1, rather than to a part of it. Also, the imaginary axis is mapped to the unit circle. Therefore, equation [digital-signal-processing-questions-answers-bilinear-transformations-q3a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-bilinear-transformations-q3a.png)is a true frequency-to-frequency transformation.

9. If s=σ+jΩ and z=rejω, then what is the condition on σ if r<1?  
a) σ > 0  
b) σ < 0  
c) σ > 1  
d) σ < 1  
View Answer

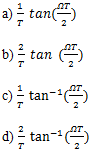
Answer: b  
Explanation: We know that if =σ+jΩ and z=rejω, then by substituting the values in the below expression  
[digital-signal-processing-questions-answers-bilinear-transformations-q3a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-bilinear-transformations-q3a.png)  
[digital-signal-processing-questions-answers-bilinear-transformations-q9](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-bilinear-transformations-q9.png)  
When r<1 => σ < 0.

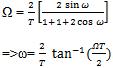
10. If s=σ+jΩ and z=rejω and r=1, then which of the following inference is correct?  
a) LHS of the s-plane is mapped inside the circle, |z|=1  
b) RHS of the s-plane is mapped outside the circle, |z|=1  
c) Imaginary axis in the s-plane is mapped to the circle, |z|=1  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: We know that if =σ+jΩ and z=rejω, then by substituting the values in the below expression  
[digital-signal-processing-questions-answers-bilinear-transformations-q9](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-bilinear-transformations-q9.png)  
=>σ = 2/T[(r2-1)/(r2+1+2rcosω)] When r=1 => σ = 0.  
This shows that the imaginary axis in the s-domain is mapped to the circle of unit radius centered at z=0 in the z-domain.

11. If s=σ+jΩ and z=rejω, then what is the condition on σ if r>1?  
a) σ > 0  
b) σ < 0  
c) σ > 1  
d) σ < 1  
View Answer

Answer: a  
Explanation: We know that if =σ+jΩ and z=rejω, then by substituting the values in the below expression  
s = 2/T[(1-z-1)/(1+z-1)] =>σ = 2/T[(r2-1)/(r2+1+2rcosω)] When r>1 => σ > 0.

12. What is the expression for the digital frequency when r=1?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-bilinear-transformations-q12.png)  
View Answer

Answer: d  
Explanation: When r=1, we get σ=0 and  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-bilinear-transformations-q12a.png)

13. What is the kind of relationship between Ω and ω?  
a) Many-to-one  
b) One-to-many  
c) One-to-one  
d) Many-to-many  
View Answer

Answer: c  
Explanation: The analog frequencies Ω=±∞ are mapped to digital frequencies ω=±π. The frequency mapping is not aliased; that is, the relationship between Ω and ω is one-to-one. As a consequence of this, there are no major restrictions on the use of bilinear transformation.

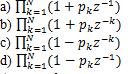
Questions & Answers (MCQs) focuses on “Quantization of Filter Coefficients”.

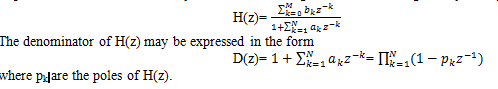
1. The system function of a general IIR filter is given as [digital-signal-processing-questions-answers-quantization-filter-coefficients-q1](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-quantization-filter-coefficients-q1.png)  
a) True  
b) False  
View Answer

Answer: a  
Explanation: If ak and bk are the filter coefficients, then the transfer function of a general IIR filter is given by the expression  
[digital-signal-processing-questions-answers-quantization-filter-coefficients-q1](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-quantization-filter-coefficients-q1.png)

2. If ak is the filter coefficient and a ̅k represents the quantized coefficient with Δak as the quantization error, then which of the following equation is true?  
a) a ̅k = ak. Δak  
b) a ̅k = ak/Δak  
c) a ̅k = ak + Δak  
d) None of the mentioned  
View Answer

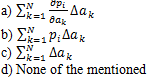
Answer: c  
Explanation: The quantized coefficient a ̅k can be related to the un-quantized coefficient ak by the relation  
a ̅k = ak + Δak  
where Δak represents the quantization error.

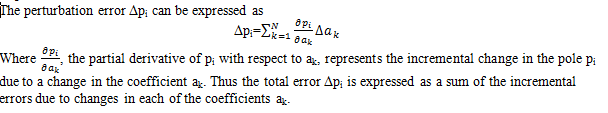
3. Which of the following is the equivalent representation of the denominator of the system function of a general IIR filter?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-quantization-filter-coefficients-q3.png)  
View Answer

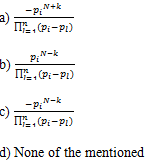
Answer: d  
Explanation: We know that the system function of a general IIR filter is given by the equation  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-quantization-filter-coefficients-q3a.png)

4. If pk is the set of poles of H(z), then what is Δpk that is the error resulting from the quantization of filter coefficients?  
a) Pre-turbation  
b) Perturbation  
c) Turbation  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: We know that p ̅k = pk + Δpk, k=1,2…N and Δpk that is the error resulting from the quantization of filter coefficients, which is called as perturbation error.

5. What is the expression for the perturbation error Δpi?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-quantization-filter-coefficients-q5.png)  
View Answer

Answer: a  
Explanation:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-quantization-filter-coefficients-q5a.png)

6. Which of the following is the expression for  
[digital-signal-processing-questions-answers-quantization-filter-coefficients-q6](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-quantization-filter-coefficients-q6.png)  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-quantization-filter-coefficients-q6a.png)  
View Answer

Answer: c  
Explanation:  
[digital-signal-processing-questions-answers-quantization-filter-coefficients-q6b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-quantization-filter-coefficients-q6b.png)

7. If the poles are tightly clustered as they are in a narrow band filter, the lengths of |pi-pl| are large for the poles in the vicinity of pi.  
a) True  
b) False  
View Answer

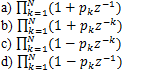
Answer: b  
Explanation: If the poles are tightly clustered as they are in a narrow band filter, the lengths of |pi-pl| are small for the poles in the vicinity of pi. These small lengths will contribute to large errors and hence a large perturbation error results.

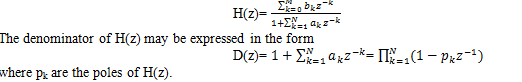
8. Which of the following operation has to be done on the lengths of |pi-pl| in order to reduce the perturbation errors?  
a) Maximize  
b) Equalize  
c) Minimize  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: The perturbation error can be minimized by maximizing the lengths of |pi-pl|. This can be accomplished by realizing the high order filter with either single pole or double pole filter sections.

9. The sensitivity analysis made on the poles of a system results on which of the following of the IIR filters?  
a) Poles  
b) Zeros  
c) Both of the mentioned  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The sensitivity analysis made on the poles of a system results on the zeros of the IIR filters.

10. Which of the following is the equivalent representation of the denominator of the system function of a general IIR filter?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-quantization-filter-coefficients-q10.png)  
View Answer

Answer: d  
Explanation: We know that the system function of a general IIR filter is given by the equation  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-quantization-filter-coefficients-q10a.png)

Questions & Answers (MCQs) focuses on “Round Off Effects in Digital Filters”.

1. The quantization inherent in the finite precision arithmetic operations render the system linear.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: In the realization of a digital filter, either in digital hardware or in software on a digital computer, the quantization inherent in the finite precision arithmetic operations render the system linear.

2. In recursive systems, which of the following is caused because of the nonlinearities due to the finite-precision arithmetic operations?  
a) Periodic oscillations in the input  
b) Non-Periodic oscillations in the input  
c) Non-Periodic oscillations in the output  
d) Periodic oscillations in the output  
View Answer

Answer: d  
Explanation: In the recursive systems, the nonlinearities due to the finite-precision arithmetic operations often cause periodic oscillations to occur in the output even when the input sequence is zero or some non zero constant value.

3. The oscillations in the output of the recursive system are called as ‘limit cycles’.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: In the recursive systems, the nonlinearities due to the finite-precision arithmetic operations often cause periodic oscillations to occur in the output even when the input sequence is zero or some non zero constant value. The oscillations thus produced in the output are known as ‘limit cycles’.

4. Limit cycles in the recursive are directly attributable to which of the following?  
a) Round-off errors in multiplication  
b) Overflow errors in addition  
c) Both of the mentioned  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The oscillations in the output of the recursive system are called as limit cycles and are directly attributable to round-off errors in multiplication and overflow errors in addition.

5. What is the range of values called as to which the amplitudes of the output during a limit cycle ae confined to?  
a) Stop band  
b) Pass band  
c) Live band  
d) Dead band  
View Answer

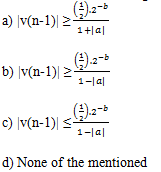
Answer: d  
Explanation: The amplitudes of the output during a limit circle are confined to a range of values that is called the ‘dead band’ of the filter.

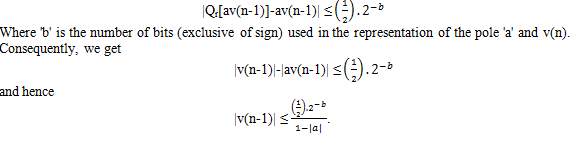
6. Zero input limit cycles occur from non-zero initial conditions with the input x(n)=0.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: When the input sequence x(n) to the filter becomes zero, the output of the filter then, after a number of iterations, enters into the limit cycle. The output remains in the limit cycle until another input of sufficient size is applied that drives the system out of the limit cycle. Similarly, zero input limit cycles occur from non-zero initial conditions with the input x(n)=0.

7. Which of the following is true when the response of the single pole filter is in the limit cycle?  
a) Actual non-linear system acts as an equivalent non-linear system  
b) Actual non-linear system acts as an equivalent linear system  
c) Actual linear system acts as an equivalent non-linear system  
d) Actual linear system acts as an equivalent linear system  
View Answer

Answer: b  
Explanation: We note that when the response of the single pole filter is in the limit cycle, the actual non-linear system acts as an equivalent linear system with a pole at z=1 when the pole is positive and z=-1 when the poles is negative.

8. Which of the following expressions define the dead band for a single-pole filter?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-round-off-effects-digital-filters-q8.png)  
View Answer

Answer: c  
Explanation: Since the quantization product av(n-1) is obtained by rounding, it follows that the quantization error is bounded as  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-round-off-effects-digital-filters-q8a.png)

9. What is the dead band of a single pole filter with a pole at 1/2 and represented by 4 bits?  
a) (-1/2,1/2)  
b) (-1/4,1/4)  
c) (-1/8,1/8)  
d) (-1/16,1/16)  
View Answer

Answer: d  
Explanation: We know that  
[digital-signal-processing-questions-answers-round-off-effects-digital-filters-q9](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-round-off-effects-digital-filters-q9.png)  
Given |a|=1/2 and b=4 => |v(n-1)| ≤ 1/16=> The dead band is (-1/16,1/16).

10. The limit cycle mode with zero input, which occurs as a result of rounding the multiplications, corresponds to an equivalent second order system with poles at z=±1.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: There is an possible limit cycle mode with zero input, which occurs as a result of rounding the multiplications, corresponds to an equivalent second order system with poles at z=±1. In this case the two pole filter exhibits oscillations with an amplitude that falls in the dead band bounded by 2-b/(1-|a1|-a2).

11. What is the necessary and sufficient condition for a second order filter that no zero-input overflow limit cycles occur?  
a) |a1|+|a2|=1  
b) |a1|+|a2|>1  
c) |a1|+|a2|<1  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: It can be easily shown that a necessary and sufficient condition for ensuring that no zero-input overflow limit cycles occur is |a1|+|a2|<1  
which is extremely restrictive and hence an unreasonable constraint to impose on any second order section.

12. An effective remedy for curing the problem of overflow oscillations is to modify the adder characteristic.  
a) True  
b) False  
View Answer

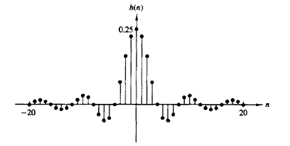
13. What is the dead band of a single pole filter with a pole at 3/4 and represented by 4 bits?  
a) (-1/2,1/2)  
b) (-1/8,1/8)  
c) (-1/4,1/4)  
d) (-1/16,1/16)  
View Answer

Answer: b  
Explanation: We know that  
[digital-signal-processing-questions-answers-round-off-effects-digital-filters-q9](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-round-off-effects-digital-filters-q9.png)  
Given |a|=3/4 and b=4 => |v(n-1)| ≤ 1/8=> The dead band is (-1/8,1/8).

Questions & Answers (MCQs) focuses on “General Consideration for Design of Digital Filters”.

1. The ideal low pass filter cannot be realized in practice.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: We know that the ideal low pass filter is non-causal. Hence, a ideal low pass filter cannot be realized in practice.

2. The following diagram represents the unit sample response of which of the following filters?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-general-considerations-design-digital-filters-q2.png)  
a) Ideal high pass filter  
b) Ideal low pass filter  
c) Ideal high pass filter at ω=π/4  
d) Ideal low pass filter at ω=π/4  
View Answer

Answer: d  
Explanation: At n=0, the equation for ideal low pass filter is given as h(n)=ω/π.  
From the given figure, h(0)=0.25=>ω=π/4.  
Thus the given figure represents the unit sample response of an ideal low pass filter at ω=π/4.

3. If h(n) has finite energy and h(n)=0 for n<0, then which of the following are true? [digital-signal-processing-questions-answers-general-considerations-design-digital-filters-q3](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-general-considerations-design-digital-filters-q3.png)d) None of the mentioned  
View Answer

Answer: b  
Explanation: If h(n) has finite energy and h(n)=0 for n<0, then according to the Paley-Wiener theorem, we have [digital-signal-processing-questions-answers-general-considerations-design-digital-filters-q3a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-general-considerations-design-digital-filters-q3a.png)

4. If |H(ω)| is square integrable and if the integral [digital-signal-processing-questions-answers-general-considerations-design-digital-filters-q4](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-general-considerations-design-digital-filters-q4.png)is finite, then the filter with the frequency response H(ω)=|H(ω)|ejθ(ω) is?  
a) Anti-causal  
b) Constant  
c) Causal  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: If |H(ω)| is square integrable and if the integral [digital-signal-processing-questions-answers-general-considerations-design-digital-filters-q4](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-general-considerations-design-digital-filters-q4.png)is finite, then we can associate with |H(ω)| and a phase response θ(ω), so that the resulting filter with the frequency response H(ω)=|H(ω)|ejθ(ω) is causal.

5. The magnitude function |H(ω)| can be zero at some frequencies, but it cannot be zero over any finite band of frequencies.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: One important conclusion that we made from the Paley-Wiener theorem is that the magnitude function |H(ω)| can be zero at some frequencies, but it cannot be zero over any finite band of frequencies, since the integral then becomes infinite. Consequently, any ideal filter is non-causal.

6. If h(n) is causal and h(n)=he(n)+ho(n),then what is the expression for h(n) in terms of only he(n)?  
a) h(n)=2he(n)u(n)+he(0)δ(n) ,n ≥ 0  
b) h(n)=2he(n)u(n)+he(0)δ(n) ,n ≥ 1  
c) h(n)=2he(n)u(n)-he(0)δ(n) ,n ≥ 1  
d) h(n)=2he(n)u(n)-he(0)δ(n) ,n ≥ 0  
View Answer

Answer: d  
Explanation: Given h(n) is causal and h(n)= he(n)+ho(n)  
=>he(n)=1/2[h(n)+h(-n)] Now, if h(n) is causal, it is possible to recover h(n) from its even part he(n) for 0≤n≤∞ or from its odd component ho(n) for 1≤n≤∞.  
=>h(n)= 2he(n)u(n)-he(0)δ(n) ,n ≥ 0.

7. If h(n) is causal and h(n)=he(n)+ho(n),then what is the expression for h(n) in terms of only ho(n)?  
a) h(n)=2ho(n)u(n)+h(0)δ(n) ,n ≥ 0  
b) h(n)=2ho(n)u(n)+h(0)δ(n) ,n ≥ 1  
c) h(n)=2ho(n)u(n)-h(0)δ(n) ,n ≥ 1  
d) h(n)=2ho(n)u(n)-h(0)δ(n) ,n ≥ 0  
View Answer

Answer: b  
Explanation: Given h(n) is causal and h(n)= he(n)+ho(n)  
=>he(n)=1/2[h(n)+h(-n)] Now, if h(n) is causal, it is possible to recover h(n) from its even part he(n) for 0≤n≤∞ or from its odd component ho(n) for 1≤n≤∞.  
=>h(n)= 2ho(n)u(n)+h(0)δ(n) ,n ≥ 1  
since ho(n)=0 for n=0, we cannot recover h(0) from ho(n) and hence we must also know h(0).

8. If h(n) is absolutely summable, i.e., BIBO stable, then the equation for the frequency response H(ω) is given as?  
a) HI(ω)-j HR(ω)  
b) HR(ω)-j HI(ω)  
c) HR(ω)+j HI(ω)  
d) HI(ω)+j HR(ω)  
View Answer

Answer: c  
Explanation: . If h(n) is absolutely summable, i.e., BIBO stable, then the frequency response H(ω) exists and  
H(ω)= HR(ω)+j HI(ω)  
where HR(ω) and HI(ω) are the Fourier transforms of he(n) and ho(n) respectively.

9. HR(ω) and HI(ω) are interdependent and cannot be specified independently when the system is causal.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Since h(n) is completely specified by he(n), it follows that H(ω) is completely determined if we know HR(ω). Alternatively, H(ω) is completely determined from HI(ω) and h(0). In short, HR(ω) and HI(ω) are interdependent and cannot be specified independently when the system is causal.

10. What is the Fourier transform of the unit step function U(ω)?  
a) πδ(ω)-0.5-j0.5cot(ω/2)  
b) πδ(ω)-0.5+j0.5cot(ω/2)  
c) πδ(ω)+0.5+j0.5cot(ω/2)  
d) πδ(ω)+0.5-j0.5cot(ω/2)  
View Answer

Answer: d  
Explanation: Since the unit step function is not absolutely summable, it has a Fourier transform which is given by the equation  
U(ω)= πδ(ω)+0.5-j0.5cot(ω/2).

11. The HI(ω) is uniquely determined from HR(ω) through the integral relationship. This integral is called as Continuous Hilbert transform.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: If the HI(ω) is uniquely determined from HR(ω) through the integral relationship. This integral is called as discrete Hilbert transform.

12. The magnitude |H(ω)| cannot be constant in any finite range of frequencies and the transition from pass-band to stop-band cannot be infinitely sharp.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Causality has very important implications in the design of frequency-selective filters. One among them is the magnitude |H(ω)| cannot be constant in any finite range of frequencies and the transition from pass-band to stop-band cannot be infinitely sharp. This is a consequence of Gibbs phenomenon, which results from the truncation of h(n) to achieve causality.

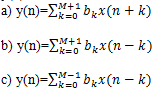
13. The frequency ωP is called as:  
a) Pass band ripple  
b) Stop band ripple  
c) Pass band edge ripple  
d) Stop band edge ripple  
View Answer

Answer: c  
Explanation: Pass band edge ripple is the frequency at which the pass band starts transiting to the stop band.

14. Which of the following represents the bandwidth of the filter?  
a) ωP+ ωS  
b) -ωP+ ωS  
c) ωP-ωS  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: If ωP and ωS represents the pass band edge ripple and stop band edge ripple, then the transition width -ωP+ ωSS gives the bandwidth of the filter.

Questions & Answers (MCQs) focuses on “Design of FIR Filters”.

1. Which of the following is the difference equation of the FIR filter of length M, input x(n) and output y(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-fir-filters-q1.png)d) None of the mentioned  
View Answer

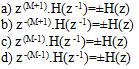
Answer: c  
Explanation: An FIR filter of length M with input x(n) and output y(n) is described by the difference equation  
[digital-signal-processing-questions-answers-design-fir-filters-q1a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-fir-filters-q1a.png)  
where {bk} is the set of filter coefficients.

2. The lower and upper limits on the convolution sum reflect the causality and finite duration characteristics of the filter.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: We can express the output sequence as the convolution of the unit sample response h(n) of the system with the input signal. The lower and upper limits on the convolution sum reflect the causality and finite duration characteristics of the filter.

3. Which of the following condition should the unit sample response of a FIR filter satisfy to have a linear phase?  
a) h(M-1-n) n=0,1,2…M-1  
b) ±h(M-1-n) n=0,1,2…M-1  
c) -h(M-1-n) n=0,1,2…M-1  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: An FIR filter has an linear phase if its unit sample response satisfies the condition  
h(n)= ±h(M-1-n) n=0,1,2…M-1.

4. If H(z) is the z-transform of the impulse response of an FIR filter, then which of the following relation is true?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-fir-filters-q4.png)  
View Answer

Answer: d  
Explanation: We know that [digital-signal-processing-questions-answers-design-fir-filters-q4a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-fir-filters-q4a.png)and h(n)= ±h(M-1-n) n=0,1,2…M-1  
When we incorporate the symmetric and anti-symmetric conditions of the second equation into the first equation and by substituting z -1 for z, and multiply both sides of the resulting equation by z -(M-1) we get  
[digital-signal-processing-questions-answers-design-fir-filters-q4b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-fir-filters-q4b.png)

5. The roots of the polynomial H(z) are identical to the roots of the polynomial H(z -1).  
a) True  
b) False  
View Answer

Answer: a  
Explanation: We know that [digital-signal-processing-questions-answers-design-fir-filters-q4b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-fir-filters-q4b.png). This result implies that the roots of the polynomial H(z) are identical to the roots of the polynomial H(z -1).

6. The roots of the equation H(z) must occur in:  
a) Identical  
b) Zero  
c) Reciprocal pairs  
d) Conjugate pairs  
View Answer

Answer: c  
Explanation: We know that the roots of the polynomial H(z) are identical to the roots of the polynomial H(z -1). Consequently, the roots of H(z) must occur in reciprocal pairs.

7. If the unit sample response h(n) of the filter is real, complex valued roots need not occur in complex conjugate pairs.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: We know that the roots of the polynomial H(z) are identical to the roots of the polynomial H(z -1). This implies that if the unit sample response h(n) of the filter is real, complex valued roots must occur in complex conjugate pairs.

8. What is the value of h(M-1/2) if the unit sample response is anti-symmetric?  
a) 0  
b) 1  
c) -1  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: When h(n)=-h(M-1-n), the unit sample response is anti-symmetric. For M odd, the center point of the anti-symmetric is n=M-1/2. Consequently, h(M-1/2)=0.

9. What is the number of filter coefficients that specify the frequency response for h(n) symmetric?  
a) (M-1)/2 when M is odd and M/2 when M is even  
b) (M-1)/2 when M is even and M/2 when M is odd  
c) (M+1)/2 when M is even and M/2 when M is odd  
d) (M+1)/2 when M is odd and M/2 when M is even  
View Answer

Answer: d  
Explanation: We know that, for a symmetric h(n), the number of filter coefficients that specify the frequency response is (M+1)/2 when M is odd and M/2 when M is even.

10. What is the number of filter coefficients that specify the frequency response for h(n) anti-symmetric?  
a) (M-1)/2 when M is even and M/2 when M is odd  
b) (M-1)/2 when M is odd and M/2 when M is even  
c) (M+1)/2 when M is even and M/2 when M is odd  
d) (M+1)/2 when M is odd and M/2 when M is even  
View Answer

Answer: b  
Explanation: We know that, for a anti-symmetric h(n) h(M-1/2)=0 and thus the number of filter coefficients that specify the frequency response is (M-1)/2 when M is odd and M/2 when M is even.

11. Which of the following is not suitable either as low pass or a high pass filter?  
a) h(n) symmetric and M odd  
b) h(n) symmetric and M even  
c) h(n) anti-symmetric and M odd  
d) h(n) anti-symmetric and M even  
View Answer

Answer: c  
Explanation: If h(n)=-h(M-1-n) and M is odd, we get H(0)=0 and H(π)=0. Consequently, this is not suitable as either a low pass filter or a high pass filter.

12. The anti-symmetric condition with M even is not used in the design of which of the following linear-phase FIR filter?  
a) Low pass  
b) High pass  
c) Band pass  
d) Bans stop  
View Answer

Answer: a  
Explanation: When h(n)=-h(M-1-n) and M is even, we know that H(0)=0. Thus it is not used in the design of a low pass linear phase FIR filter.

13. The anti-symmetric condition is not used in the design of low pass linear phase FIR filter.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: We know that if h(n)=-h(M-1-n) and M is odd, we get H(0)=0 and H(π)=0. Consequently, this is not suitable as either a low pass filter or a high pass filter and when h(n)=-h(M-1-n) and M is even, we know that H(0)=0. Thus it is not used in the design of a low pass linear phase FIR filter. Thus the anti-symmetric condition is not used in the design of low pass linear phase FIR filter.

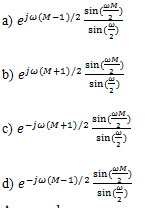
Questions & Answers (MCQs) focuses on “Design of Linear Phase FIR Filters Using Windows-1”.

1. Which of the following defines the rectangular window function of length M-1?  
a) w(n)= 1, n=0,1,2…M-1  
=0, else where  
b) w(n)= 1, n=0,1,2…M-1  
=-1, else where  
c) w(n)= 0, n=0,1,2…M-1  
=1, else where  
d) None of the mentioned  
View Answer

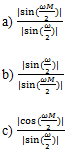
Answer: a  
Explanation: We know that the rectangular window of length M-1 is defined as  
w(n)= 1, n=0,1,2…M-1  
=0, else where.

2. The multiplication of the window function w(n) with h(n) is equivalent to the multiplication of H(w) and W(w).  
a) True  
b) False  
View Answer

Answer: b  
Explanation: According to the basic formula of convolution, the multiplication of two signals w(n) and h(n) in time domain is equivalent to the convolution of their respective Fourier transforms W(w) and H(w).

3. What is the Fourier transform of the rectangular window of length M-1?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-linear-phase-fir-filters-windows-1-q3.png)  
View Answer

Answer: d  
Explanation: We know that the Fourier transform of a function w(n) is defined as  
[digital-signal-processing-questions-answers-design-linear-phase-fir-filters-windows-1-q3a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-linear-phase-fir-filters-windows-1-q3a.png)  
For a rectangular window, w(n)=1 for n=0,1,2….M-1  
Thus we get  
[digital-signal-processing-questions-answers-design-linear-phase-fir-filters-windows-1-q3b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-linear-phase-fir-filters-windows-1-q3b.png)

4. What is the magnitude response |W(ω)| of a rectangular window function?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-linear-phase-fir-filters-windows-1-q4.png)d) None of the mentioned  
View Answer

Answer: a  
Explanation: We know that for a rectangular window  
[digital-signal-processing-questions-answers-design-linear-phase-fir-filters-windows-1-q3b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-linear-phase-fir-filters-windows-1-q3b.png)  
Thus the window function has a magnitude response  
[digital-signal-processing-questions-answers-design-linear-phase-fir-filters-windows-1-q4a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-linear-phase-fir-filters-windows-1-q4a.png)

5. What is the width of the main lobe of the frequency response of a rectangular window of length M-1?  
a) π/M  
b) 2π/M  
c) 4π/M  
d) 8π/M  
View Answer

Answer: c  
Explanation: The width of the main lobe width is measured to the first zero of W(ω)) is 4π/M.

6. The width of each side lobes decreases with an increase in M.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Since the width of the main lobe is inversely proportional to the value of M, if the value of M increases then the main lobe becomes narrower. In fact, the width of each side lobes decreases with an increase in M.

7. With an increase in the value of M, the height of each side lobe:  
a) Do not vary  
b) Does not depend on value of M  
c) Decreases  
d) Increases  
View Answer

Answer: d  
Explanation: The height of each side lobes increase with an increase in M such a manner that the area under each side lobe remains invariant to changes in M.

8. As M is increased, W(ω) becomes wider and the smoothening produced by the W(ω) is increased.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: As M is increased, W(ω) becomes narrower and the smoothening produced by the W(ω) is reduced.

9. Which of the following windows has a time domain sequence [digital-signal-processing-questions-answers-design-linear-phase-fir-filters-windows-1-q9](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-linear-phase-fir-filters-windows-1-q9.png)  
a) Bartlett window  
b) Blackman window  
c) Hanning window  
d) Hamming window  
View Answer

Answer: a  
Explanation: The Bartlett window which is also called as triangular window has a time domain sequence as  
[digital-signal-processing-questions-answers-design-linear-phase-fir-filters-windows-1-q9](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-linear-phase-fir-filters-windows-1-q9.png), 0≤n≤M-1.

10. The width of each side lobes decreases with an decrease in M.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: Since the width of the main lobe is inversely proportional to the value of M, if the value of M increases then the main lobe becomes narrower. In fact, the width of each side lobes decreases with an increase in M.

11. What is the approximate transition width of main lobe of a Hamming window?  
a) 4π/M  
b) 8π/M  
c) 12π/M  
d) 2π/M  
View Answer

Answer: b  
Explanation: The transition width of the main lobe in the case of Hamming window is equal to 8π/M where M is the length of the window.

“Design of Linear Phase FIR Filters Using Windows”.

1. What is the peak side lobe(in dB) for a rectangular window?  
a) -13  
b) -27  
c) -32  
d) -58  
View Answer

Answer: a  
Explanation: The peak side lobe in the case of rectangular window has a value of -13dB.

2. What is the peak side lobe(in dB) for a Hanning window?  
a) -13  
b) -27  
c) -32  
d) -58  
View Answer

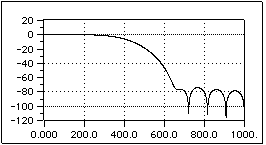
Answer: c  
Explanation: The peak side lobe in the case of Hanning window has a value of -32dB.

3. How does the frequency of oscillations in the pass band of a low pass filter varies with the value of M?  
a) Decrease with increase in M  
b) Increase with increase in M  
c) Remains constant with increase in M  
d) None of the mentioned  
View Answer

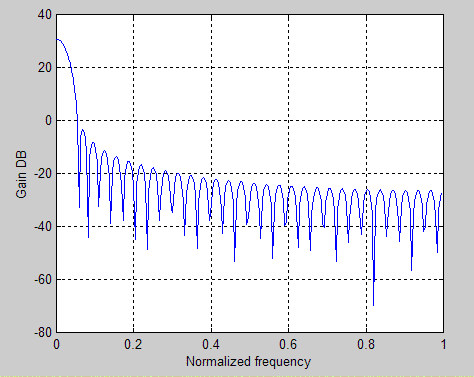
Answer: b  
Explanation: The frequency of oscillations in the pass band of a low pass filter increases with an increase in the value of M, but they do not diminish in amplitude.

4. The oscillatory behavior near the band edge of the low pass filter is known as Gibbs phenomenon.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The multiplication of hd(n) with a rectangular window is identical to truncating the Fourier series representation of the desired filter characteristic Hd(ω). The truncation of Fourier series is known to introduce ripples in the frequency response characteristic H(ω) due to the non-uniform convergence of the Fourier series at a discontinuity. The oscillatory behavior near the band edge of the low pass filter is known as Gibbs phenomenon.

5. Which of the following window is used in the design of a low pass filter to have a frequency response as shown in the figure?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-mcqs-q5.png)  
a) Hamming window  
b) Hanning window  
c) Kaiser window  
d) Blackman window  
View Answer

Answer: d  
Explanation: The frequency response shown in the figure is the frequency response of a low pass filter designed using a Blackman window of length M=61.

6. Which of the following window is used in the design of a low pass filter to have a frequency response as shown in the figure?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-mcqs-q6.png)  
a) Hamming window  
b) Hanning window  
c) Kaiser window  
d) Blackman window  
View Answer

Answer: c  
Explanation: The frequency response shown in the figure is the frequency response of a low pass filter designed using a Kaiser window of length M=61 and α=4.

7. What is the approximate transition width of main lobe of a Blackman window?  
a) 4π/M  
b) 8π/M  
c) 12π/M  
d) 2π/M  
View Answer

Answer: c  
Explanation: The transition width of the main lobe in the case of Blackman window is equal to 12π/M where M is the length of the window.

8. Which of the following windows has a time domain sequence [digital-signal-processing-mcqs-q8](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-mcqs-q8.png)  
a) Bartlett window  
b) Blackman window  
c) Hamming window  
d) Hanning window  
View Answer

Answer: d  
Explanation: The Hanning window has a time domain sequence as  
[digital-signal-processing-mcqs-q8](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-mcqs-q8.png), 0≤n≤M-1.

9. If the value of M increases then the main lobe in the frequency response of the rectangular window becomes broader.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: Since the width of the main lobe is inversely proportional to the value of M, if the value of M increases then the main lobe becomes narrower.

10. The large side lobes of W(ω) results in which of the following undesirable effects?  
a) Circling effects  
b) Broadening effects  
c) Ringing effects  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The larger side lobes of W(ω) results in the undesirable ringing effects in the FIR filter frequency response H(ω), and also in relatively large side lobes in H(ω).

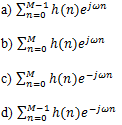
Questions & Answers focuses on “Design of Linear Phase FIR Filters by Frequency Sampling Method”.

1. In the frequency sampling method for FIR filter design, we specify the desired frequency response Hd(ω) at a set of equally spaced frequencies.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: In the frequency sampling method, we specify the frequency response Hd(ω) at a set of equally spaced frequencies, namely  
[digital-signal-processing-multiple-choice-questions-answers-q1](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-multiple-choice-questions-answers-q1.png)

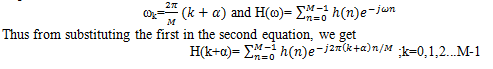
2. To reduce side lobes, in which region of the filter the frequency specifications has to be optimized?  
a) Stop band  
b) Pass band  
c) Transition band  
d) None of the mentioned  
View Answer

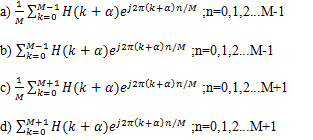
Answer: c  
Explanation: To reduce the side lobes, it is desirable to optimize the frequency specification in the transition band of the filter. This optimization can be accomplished numerically on a digital computer by means of linear programming techniques.

3. What is the frequency response of a system with input h(n) and window length of M?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-multiple-choice-questions-answers-q3.png)  
View Answer

Answer: d  
Explanation: The desired output of an FIR filter with an input h(n) and using a window of length M is given as  
[digital-signal-processing-multiple-choice-questions-answers-q3a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-multiple-choice-questions-answers-q3a.png)

4. What is the relation between H(k+α) and h(n)?  
[digital-signal-processing-multiple-choice-questions-answers-q4](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-multiple-choice-questions-answers-q4.png)d) None of the mentioned  
View Answer

Answer: b  
Explanation: We know that  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-multiple-choice-questions-answers-q4a.png)

5. Which of the following is the correct expression for h(n) in terms of H(k+α)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-multiple-choice-questions-answers-q5.png)  
View Answer

Answer: a  
Explanation: We know that  
[digital-signal-processing-multiple-choice-questions-answers-q5a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-multiple-choice-questions-answers-q5a.png)  
If we multiply the above equation on both sides by the exponential exp(j2πkm/M), m=0,1,2….M-1 and sum over k=0,1,….M-1, we get the equation  
[digital-signal-processing-multiple-choice-questions-answers-q5b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-multiple-choice-questions-answers-q5b.png)

6. Which of the following is equal to the value of H(k+α)?  
a) H\*(M-k+α)  
b) H\*(M+k+α)  
c) H\*(M+k-α)  
d) H\*(M-k-α)  
View Answer

Answer: d  
Explanation: Since {h(n)} is real, we can easily show that the frequency samples {H(k+α)} satisfy the symmetry condition  
H(k+α)= H\*(M-k-α).

7. The linear equations for determining {h(n)} from {H(k+α)} are not simplified.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: The symmetry condition, along with the symmetry conditions for {h(n)}, can be used to reduce the frequency specifications from M points to (M+1)/2 points for M odd and M/2 for M even. Thus the linear equations for determining {h(n)} from {H(k+α)} are considerably simplified.

8. The major advantage of designing linear phase FIR filter using frequency sampling method lies in the efficient frequency sampling structure.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Although the frequency sampling method provides us with another means for designing linear phase FIR filters, its major advantage lies in the efficient frequency sampling structure, which is obtained when most of the frequency samples are zero.

9. Which of the following is introduced in the frequency sampling realization of the FIR filter?  
a) Poles are more in number on unit circle  
b) Zeros are more in number on the unit circle  
c) Poles and zeros at equally spaced points on the unit circle  
d) None of the mentioned  
View Answer

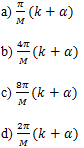
Answer: c  
Explanation: There is a potential problem for frequency sampling realization of the FIR linear phase filter. The frequency sampling realization of the FIR filter introduces poles and zeros at equally spaced points on the unit circle.

10. In a practical implementation of the frequency sampling realization, quantization effects preclude a perfect cancellation of the poles and zeros.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: In the ideal situation, the zeros cancel the poles and, consequently, the actual zeros of the H(z) are determined by the selection of the frequency samples H(k+α). In a practical implementation of the frequency sampling realization, however, quantization effects preclude a perfect cancellation of the poles and zeros.

11. In the frequency sampling method for FIR filter design, we specify the desired frequency response Hd(ω) at a set of equally spaced frequencies.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: According to the frequency sampling method for FIR filter design, the desired frequency response is specified at a set of equally spaced frequencies.

12. What is the equation for the frequency ωk in the frequency response of an FIR filter?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-multiple-choice-questions-answers-q12.png)  
View Answer

Answer: d  
Explanation: In the frequency sampling method for FIR filter design, we specify the desired frequency response Hd(ω) at a set of equally spaced frequencies, namely  
[digital-signal-processing-multiple-choice-questions-answers-q12a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-multiple-choice-questions-answers-q12a.png)  
where k=0,1,2…M-1/2 and α=0 0r 1/2.

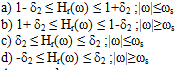
13. Why is it desirable to optimize frequency response in the transition band of the filter?  
a) Increase side lobe  
b) Reduce side lobe  
c) Increase main lobe  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: To reduce side lobes, it is desirable to optimize the frequency specification in the transition band of the filter.

Design of Optimum Equi Ripple Linear Phase FIR Filters”.

1. Which of the following filter design is used in the formulation of design of optimum equi ripple linear phase FIR filter?  
a) Butterworth approximation  
b) Chebyshev approximation  
c) Hamming approximation  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The filter design method described in the design of optimum equi ripple linear phase FIR filters is formulated as a chebyshev approximation problem.

2. If δ2 represents the ripple in the stop band for a chebyshev filter, then which of the following conditions is true?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-online-test-q2.png)  
View Answer

Answer: d  
Explanation: Let us consider the design of a low pass filter with the stop band edge frequency ωs and the ripple in the stop band is δ2, then from the general specifications of the chebyshev filter, in the stop band the filter frequency response should satisfy the condition  
[digital-signal-processing-online-test-q2a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-online-test-q2a.png)

3. If the filter has anti-symmetric unit sample response with M even, then what is the value of Q(ω)?  
a) cos(ω/2)  
b) sin(ω/2)  
c) 1  
d) sinω  
View Answer

Answer: b  
Explanation: If the filter has a anti-symmetric unit sample response, then we know that  
h(n)= -h(M-1-n)  
and for M even in this case, Q(ω)=sin(ω/2).

4. It is convenient to normalize W(ω) to unity in the stop band and set W(ω)=δ2/ δ1 in the pass band.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The weighting function on the approximation error allows to choose the relative size of the errors in the different frequency bands. In particular, it is convenient to normalize W(ω) to unity in the stop band and set W(ω)=δ2/ δ1 in the pass band.

5. Which of the following defines the weighted approximation error?  
a) W(ω)[Hdr(ω)+ Hr(ω)].  
b) W(ω)[Hdr(ω)- Hr(ω)].  
c) W(ω)[Hr(ω)- Hdr(ω)].  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The weighted approximation error is defined as E(ω) which is given as  
E(ω)= W(ω)[Hdr(ω)- Hr(ω)].

6. The error function E(ω) does not alternate in sign between two successive extremal frequencies.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: The error function E(ω) alternates in sign between two successive extremal frequency, Hence the theorem is called as Alternative theorem.

7. At most how many extremal frequencies can be there in the error function of ideal low pass filter?  
a) L+1  
b) L+2  
c) L+3  
d) L  
View Answer

Answer: c  
Explanation: We know that we can have at most L-1 local maxima and minima in the open interval 0<ω<π. In addition, ω=0 and π are also usually extrema. It is also maximum at ω for pass band and stop band frequencies. Thus the error function of a low pass filter has at most L+3 extremal frequencies.

8. The filter designs that contain more than L+2 alternations are called as:  
a) Extra ripple filters  
b) Maximal ripple filters  
c) Equi ripple filters  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: In general, the filter designs that contain more than L+2 alternations or ripples are called as Extra ripple filters.

9. If M is the length of the filter, then at how many number of points, the error function is computed?  
a) 2M  
b) 4M  
c) 8M  
d) 16M  
View Answer

Answer: d  
Explanation: Having the solution for P(ω), we can now compute the error function E(ω) from  
E(ω)= W(ω)[Hdr(ω)- Hr(ω)] on a dense set of frequency points. Usually, a number of points equal to 16M, where M is the length of the filter.

10. If |E(ω)|<δ for some frequencies on the dense set, then a new set of frequencies corresponding to the L+2 largest peaks of |E(ω)| are selected and computation is repeated.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: If |E(ω)|≥δ for some frequencies on the dense set, then a new set of frequencies corresponding to the L+2 largest peaks of |E(ω)| are selected and computation is repeated.

11. What is the value of JTYPE in the Parks-McClellan program for a Hilbert transformer?  
a) 1  
b) 2  
c) 3  
d) 4  
View Answer

Answer: c  
Explanation: The value of JTYPE=3 in the Parks-McClellan program to select a filter that performs Hilbert transformer.

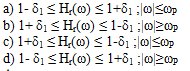
12. In Parks-McClellan program, the grid density for interpolating the error function is denoted by which of the following functions?  
a) NFILT  
b) NBANDS  
c) EDGE  
d) LGRID  
View Answer

Answer: d  
Explanation: In Parks-McClellan program, LGRID represents the grid density for interpolating the error function. The default value is 16 if left unspecified.

13. In Parks-McClellan program, an array of maximum size 10 that specifies the desired frequency response in each band is denoted by?  
a) WTX  
b) FX  
c) EDGE  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: FX denotes an array of maximum size 10 that specifies the desired frequency response in each band.

Questions & Answers (MCQs) focuses on “Design of Optimum Equi Ripple Linear Phase FIR Filters-2”.

1. If δ1 represents the ripple in the pass band for a chebyshev filter, then which of the following conditions is true?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-quiz-online-q1.png)  
View Answer

Answer: a  
Explanation: Let us consider the design of a low pass filter with the pass band edge frequency ωP and the ripple in the pass band is δ1, then from the general specifications of the chebyshev filter, in the pass band the filter frequency response should satisfy the condition  
[digital-signal-processing-quiz-online-q1a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-quiz-online-q1a.png)

2. If the filter has symmetric unit sample response with M odd, then what is the value of Q(ω)?  
a) cos(ω/2)  
b) sin(ω/2)  
c) 1  
d) sinω  
View Answer

Answer: c  
Explanation: If the filter has a symmetric unit sample response, then we know that  
h(n)=h(M-1-n)  
and for M odd in this case, Q(ω)=1.

3. If the filter has anti-symmetric unit sample response with M odd, then what is the value of Q(ω)?  
a) cos(ω/2)  
b) sin(ω/2)  
c) 1  
d) sinω  
View Answer

Answer: d  
Explanation: If the filter has a anti-symmetric unit sample response, then we know that  
h(n)= -h(M-1-n)  
and for M odd in this case, Q(ω)=sin(ω).

4. In which of the following way the real valued desired frequency response is defined?  
a) Unity in stop band and zero in pass band  
b) Unity in both pass and stop bands  
c) Unity in pass band and zero in stop band  
d) Zero in both stop and pass band  
View Answer

Answer: c  
Explanation: The real valued desired frequency response Hdr(ω) is simply defined to be unity in the pass band and zero in the stop band.

5. The error function E(ω) should exhibit at least how many extremal frequencies in S?  
a) L  
b) L-1  
c) L+1  
d) L+2  
View Answer

Answer: d  
Explanation: According to Alternation theorem, a necessary and sufficient condition for Pω) to be unique, best weighted chebyshev approximation, is that the error function E(ω) must exhibit at least L+2 extremal frequencies in S.

6. The filter designs that contain maximum number of alternations are called as:  
a) Extra ripple filters  
b) Maximal ripple filters  
c) Equi ripple filters  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: In general, the filter designs that contain maximum number of alternations or ripples are called as maximal ripple filters.

7. Remez exchange algorithm is an iterative algorithm used in error approximation.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Initially, we neither know the set of external frequencies nor the parameters. To solve for the parameters, we use an iterative algorithm called the Remez exchange algorithm, in which we begin by guessing at the set of extremal frequencies.

8. When |E(ω)|≤δ for all frequencies on the dense set, the optimal solution has been found in terms of the polynomial H(ω).  
a) True  
b) False  
View Answer

Answer: a  
Explanation: |E(ω)|≥δ for some frequencies on the dense set, then a new set of frequencies corresponding to the L+2 largest peaks of |E(ω)| are selected and computation is repeated. Since the new set of L+2 extremal frequencies are selected to increase in each iteration until it converges to the upper bound, this implies that when |E(ω)|≤δ for all frequencies on the dense set, the optimal solution has been found in terms of the polynomial H(ω).

9. In Parks-McClellan program, an array of maximum size 10 that specifies the weight function in each band is denoted by?  
a) WTX  
b) FX  
c) EDGE  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: FX denotes an array of maximum size 10 that specifies the weight function in each band.

10. The filter designs which are formulated using chebyshev approximating problem have ripples in?  
a) Pass band  
b) Stop band  
c) Both of the mentioned  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The chebyshev approximation problem is viewed as an optimum design criterion on the sense that the weighted approximation error between the desired frequency response and the actual frequency response is spread evenly across the pass band and evenly across the stop band of the filter minimizing the maximum error. The resulting filter designs have ripples in both pass band and stop band.

11. If the filter has symmetric unit sample response with M even, then what is the value of Q(ω)?  
a) cos(ω/2)  
b) sin(ω/2)  
c) 1  
d) sinω  
View Answer

Answer: a  
Explanation: If the filter has a symmetric unit sample response, then we know that  
h(n)=h(M-1-n)  
and for M even in this case, Q(ω)= cos(ω/2).

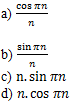
Questions & Answers (MCQs) focuses on “Design of FIR Differentiators”.

1. How is the frequency response of an ideal differentiator related to the frequency?  
a) Inversely proportional  
b) Linearly proportional  
c) Quadratic  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: An ideal differentiator has a frequency response that is linearly proportional to the frequency.

2. Which of the following is the frequency response of an ideal differentiator, Hd(ω)?  
a) -jω ; -π ≤ ω ≤ π  
b) -jω ; 0 ≤ ω ≤ π  
c) jω ; 0 ≤ ω ≤ π  
d) jω ; -π ≤ ω ≤ π  
View Answer

Answer: d  
Explanation: An ideal differentiator is defined as one that has the frequency response  
Hd(ω)= jω ; -π ≤ ω ≤ π.

3. What is the unit sample response corresponding to Hd(ω)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-fir-differentiators-q3.png)  
View Answer

Answer: a  
Explanation: We know that, for an ideal differentiator, the frequency response is given as  
Hd(ω)= jω ; -π ≤ ω ≤ π  
Thus, we get the unit sample response corresponding to the ideal differentiator is given as  
h(n)=cosπn/n.

4. The ideal differentiator ahs which of the following unit sample response?  
a) Symmetric  
b) Anti-symmetric  
c) Cannot be explained  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: We know that the unit sample response of an ideal differentiator is given as  
h(n)=cosπn/n  
So, we can state that the unit sample response of an ideal differentiator is anti-symmetric because cos⁡πn is also an anti-symmetric function.

5. If hd(n) is the unit sample response of an ideal differentiator, then what is the value of hd(0)?  
a) 1  
b) -1  
c) 0  
d) 0.5  
View Answer

Answer: c  
Explanation: Since we know that the unit sample response of an ideal differentiator is anti-symmetric,  
=>hd(0)=0.

6. In this section, we confine our attention to FIR designs in which h(n)=-h(M-1-n).  
a) True  
b) False  
View Answer

Answer: a  
Explanation: In view of the fact that the ideal differentiator has an anti-symmetric unit sample response, we shall confine our attention to FIR designs in which h(n)=-h(M-1-n).

7. Which of the following is the condition that an differentiator should satisfy?  
a) Infinite response at zero frequency  
b) Finite response at zero frequency  
c) Negative response at zero frequency  
d) Zero response at zero frequency  
View Answer

Answer: d  
Explanation: For an FIR filter, when M is odd, the real valued frequency response of the FIR filter Hr(ω) has the characteristic that Hr(0)=0. A zero response at zero frequency is just the condition that the differentiator should satisfy.

8. Full band differentiators can be achieved with an FIR filters having odd number of coefficients.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: Full band differentiators cannot be achieved with an FIR filters having odd number of coefficients, since Hr(π)=0 for M odd.

9. If fp is the bandwidth of the differentiator, then the desired frequency characteristic should be linear in the range:  
a) 0 ≤ ω ≤ 2π  
b) 0 ≤ ω ≤ 2fp  
c) 0 ≤ ω ≤ 2πfp  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: In most cases of practical interest, the desired frequency response characteristic need only be linear over the limited frequency range 0 ≤ ω ≤ 2πfp , where fp is the bandwidth of the differentiator.

10. What is the desired response of the differentiator in the frequency range 2πfp ≤ ω ≤ π?  
a) Left unconstrained  
b) Constrained to be zero  
c) Either of the mentioned  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: In the frequency range 2πfp ≤ ω ≤ π, the desired response may be either left unconstrained or constrained to be zero.

11. What is the weighting function used in the design of FIR differentiators based on the chebyshev approximation criterion?  
a) 1/ω  
b) ω  
c) 1+ω  
d) 1-ω  
View Answer

Answer: a  
Explanation: In the design of FIR differentiators based on the chebyshev approximation criterion, the weighting function W(ω) is specified in the program as  
W(ω)=1/ω  
in order that the relative ripple in the pass band be a constant.

12. The absolute error between the desired response ω and the approximation Hr(ω) decreases as ω varies from 0 to 2πfp.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: We know that the weighting function is  
W(ω)=1/ω  
in order that the relative ripple in the pass band be a constant. Thus, the absolute error between the desired response ω and the approximation Hr(ω) increases as ω varies from 0 to 2πfp.

13. Which of the following is the important parameter in a differentiator?  
a) Length  
b) Bandwidth  
c) Peak relative error  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: The important parameters in a differentiator are its length, its bandwidth and the peak relative error of the approximation. The inter relationship among these three parameters can be easily displayed parametrically.

14. In this section, we confine our attention to FIR designs in which h(n)=h(M-1-n).  
a) True  
b) False  
View Answer

Answer: b  
Explanation: In view of the fact that the ideal differentiator has an anti-symmetric unit sample response, we shall confine our attention to FIR designs in which h(n)=-h(M-1-n).

15. What is the maximum value of fp with which good designs are obtained for M odd?  
a) 0.25  
b) 0.45  
c) 0.5  
d) 0.75  
View Answer

Answer: b  
Explanation: Designs based on M odd are particularly poor if the bandwidth exceeds 0.45. The problem is basically the zero in the frequency response at ω=π(f=1/2). When fp <0.45, good designs are obtained for M odd.

Questions & Answers (MCQs) focuses on “Design of Hilbert Transforms”.

1. What kind of filter is an ideal Hilbert transformer?  
a) Low pass  
b) High pass  
c) Band pass  
d) All pass  
View Answer

Answer: d  
Explanation: An ideal Hilbert transformer is a all pass filter.

2. How much phase shift does an Hilbert transformer impart on the input?  
a) 45o  
b) 90o  
c) 135o  
d) 180o  
View Answer

Answer: b  
Explanation: An ideal Hilbert transformer is a all pass filter that imparts a 90o phase shift on the signal at its input.

3. Which of the following is the frequency response of the ideal Hilbert transform?

a) -j ;0 < ω < π

j ;-π < ω < 0

b) j ;0 < ω < π

-j ;-π < ω < 0

c) -j ;-π < ω < π

d) None of the mentioned

View Answer

Answer: a  
Explanation: The frequency response of an ideal Hilbert transform is given as

H(ω)= -j ;0 < ω < π

j ;-π < ω < 0

4. In which of the following fields, Hilbert transformers are frequently used?  
a) Generation of SSB signals  
b) Radar signal processing  
c) Speech signal processing  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: Hilbert transforms are frequently used in communication systems and signal processing, as, for example, in the generation of SSB modulated signals, radar signal processing and speech signal processing.

5. The unit sample response of an ideal Hilbert transform is  
[digital-signal-processing-questions-answers-design-hilbert-transformers-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-hilbert-transformers-q5.png)  
a) True  
b) False  
View Answer

Answer: a  
Explanation: We know that the frequency response of an ideal Hilbert transformer is given as  
H(ω)= -j ;0 < ω < π  
j ;-π < ω < 0  
Thus the unit sample response of an ideal Hilbert transform is obtained as  
[digital-signal-processing-questions-answers-design-hilbert-transformers-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-hilbert-transformers-q5.png)

6. The unit sample response of Hilbert transform is infinite in duration and causal.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: We know that the unit sample response of the Hilbert transform is given as  
[digital-signal-processing-questions-answers-design-hilbert-transformers-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-hilbert-transformers-q5.png)  
it sample response of an ideal Hilbert transform is infinite in duration and non-causal.

7. The unit sample response of Hilbert transform is:  
a) Zero  
b) Symmetric  
c) Anti-symmetric  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: We know that the unit sample response of the Hilbert transform is given as  
[digital-signal-processing-questions-answers-design-hilbert-transformers-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-hilbert-transformers-q5.png)  
Thus from the above equation, we can tell that h(n)=-h(-n). Thus the unit sample response of Hilbert transform is anti-symmetric in nature.

8. In this section, we confine our attention on the design of FIR Hilbert transformers with h(n)=-h(M-1-n).  
a) True  
b) False  
View Answer

Answer: a  
Explanation: In view of the fact that the ideal Hilbert transformer has an anti-symmetric unit sample response, we shall confine our attention to FIR designs in which h(n)=-h(M-1-n).

9. Which of the following is true regarding the frequency response of Hilbert transform?  
a) Complex  
b) Purely imaginary  
c) Purely real  
d) Zero  
View Answer

Answer: b  
Explanation: Our choice of an anti-symmetric unit sample response is consistent with having a purely imaginary frequency response characteristic.

10. It is impossible to design an all-pass digital Hilbert transformer.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: We know that when h(n) is anti-symmetric, the real valued frequency response characteristic is zero at ω=0 for both M odd and even and at ω=π when M is odd. Clearly, then, it is impossible to design an all-pass digital Hilbert transformer.

11. If fl and fu are the cutoff frequencies, then what is the desired real valued frequency response of a Hilbert transform filter in the frequency range 2π fl < ω < 2πfu?  
a) -1  
b) -0.5  
c) 0  
d) 1  
View Answer

Answer: d  
Explanation: The bandwidth of Hilbert transformer need only cover the bandwidth of the signal to be phase shifted. Consequently, we specify the desired real valued frequency response of a Hilbert transformer filter is  
H(ω)=1; 2π fl < ω < 2πfu  
where fl and fu are the cutoff frequencies.

12. What is the value of unit sample response of an ideal Hilbert transform for ‘n’ even?  
a) -1  
b) 1  
c) 0  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The unit sample response of the Hilbert transformer is given as  
[digital-signal-processing-questions-answers-design-hilbert-transformers-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-hilbert-transformers-q5.png)  
From the above equation, it is clear that h(n) becomes zero for even values of ‘n’.

“Comparision of Design Methods for Linear Phase FIR Filters”.

1. Which of the following is the first method proposed for design of FIR filters?  
a) Chebyshev approximation  
b) Frequency sampling method  
c) Windowing technique  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The design method based on the use of windows to truncate the impulse response h(n) and obtaining the desired spectral shaping, was the first method proposed for designing linear phase FIR filters.

2. The lack of precise control of cutoff frequencies is a disadvantage of which of the following designs?  
a) Window design  
b) Chebyshev approximation  
c) Frequency sampling  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: The major disadvantage of the window design method is the lack of precise control of the critical frequencies.

3. The values of cutoff frequencies in general depend on which of the following?  
a) Type of the window  
b) Length of the window  
c) None of the mentioned  
d) Both of the mentioned  
View Answer

Answer: d  
Explanation: The values of the cutoff frequencies of a filter in general by windowing technique depend on the type of the filter and the length of the filter.

4. In frequency sampling method, transition band is a multiple of which of the following?  
a) π/M  
b) 2π/M  
c) π/2M  
d) 2πM  
View Answer

Answer: b  
Explanation: In the frequency sampling technique, the transition band is a multiple of 2π/M.

5. The frequency sampling design method is attractive when the FIR filter is realized in the frequency domain by means of the DFT.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Frequency sampling design method is particularly attractive when the FIR is realized either in the frequency domain by means of the DFT or in any of the frequency sampling realizations.

6. Which of the following values can a frequency response take in frequency sampling technique?  
a) Zero  
b) One  
c) Either of them  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The attractive feature of the frequency sampling design is that the frequency response can take either zero or one at all frequencies, except in the transition band.

7. Which of the following technique is more preferable for design of linear phase FIR filter?  
a) Window design  
b) Chebyshev approximation  
c) Frequency sampling  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The chebyshev approximation method provides total control of the filter specifications, and as a consequence, it is usually preferable over the other two methods.

8. By optimal filter design, the maximum side lobe level is minimized.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: By spreading the approximation error over the pass band and stop band of the filter, this method results in an optimal filter design and using this the maximum side lobe level is minimized.

9. Which of the following is the correct expression for transition band Δf?  
a) (ωp- ωs)/2π  
b) (ωp+ωs)/2π  
c) (ωp.ωs)/2π  
d) (ωs- ωp)/2π  
View Answer

Answer: d  
Explanation: The expression for Δf i.e., for the transition band is given as  
Δf=(ωs- ωp)/2π.

10. If the resulting δ exceeds the specified δ2, then the length can be increased until we obtain a side lobe level that meets the specification.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The estimate is used to carry out the design and if the resulting δ exceeds the specified δ2, then the length can be increased until we obtain a side lobe level that meets the specification.

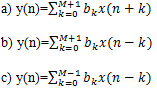
Questions & Answers (MCQs) focuses on “Design of IIR Filters from Analog Filters”.

1. What is the duration of the unit sample response of a digital filter?  
a) Finite  
b) Infinite  
c) Impulse(very small)  
d) Zero  
View Answer

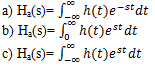
Answer: b  
Explanation: Digital filters are the filters which can be designed from analog filters which have infinite duration unit sample response.

2. Which of the following methods are used to convert analog filter into digital filter?  
a) Approximation of Derivatives  
b) Bilinear transformation  
c) Impulse invariance  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: There are many techniques which are used to convert analog filter into digital filter of which some of them are Approximation of derivatives, bilinear transformation, impulse invariance and many other methods.

3. Which of the following is the difference equation of the FIR filter of length M, input x(n) and output y(n)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-iir-filters-analog-filters-q3.png)d) None of the mentioned  
View Answer

Answer: c  
Explanation: An FIR filter of length M with input x(n) and output y(n) is described by the difference equation  
[digital-signal-processing-questions-answers-design-iir-filters-analog-filters-q3a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-iir-filters-analog-filters-q3a.png)  
where {bk} is the set of filter coefficients.

4. What is the relation between h(t) and Ha(s)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-iir-filters-analog-filters-q4.png)d) None of the mentioned  
View Answer

Answer: a  
Explanation: We know that the impulse response h(t) and the Laplace transform Ha(s) are related by the equation  
[digital-signal-processing-questions-answers-design-iir-filters-analog-filters-q4a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-iir-filters-analog-filters-q4a.png)

5. Which of the following is a representation of system function?  
a) Normal system function  
b) Laplace transform  
c) Rational system function  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: There are many ways how we represent a system function of which one is normal representation i.e., output/input and other ways like Laplace transform and rational system function.

6. For an analog LTI system to be stable, where should the poles of system function H(s) lie?  
a) Right half of s-plane  
b) Left half of s-plane  
c) On the imaginary axis  
d) At origin  
View Answer

Answer: b  
Explanation: An analog linear time invariant system with system function H(s) is stable if all its poles lie on the left half of the s-plane.

7. If the conversion technique is to be effective, the jΩ axis in the s-plane should map into the unit circle in the z-plane.  
a) True  
b) False  
View Answer

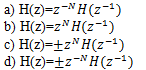
Answer: a  
Explanation: If the conversion technique is to be effective, the jΩ axis in the s-plane should map into the unit circle in the z-plane. Thus there will be a direct relationship between the two frequency variables in the two domains.

8. If the conversion technique is to be effective, then the LHP of s-plane should be mapped into:  
a) Outside of unit circle  
b) Unit circle  
c) Inside unit circle  
d) Does not matter  
View Answer

Answer: c  
Explanation: If the conversion technique is to be effective, then the LHP of s-plane should be mapped into the inside of the unit circle in the z-plane. Thus a stable analog filter will be converted to a stable digital filter.

9. Physically realizable and stable IIR filters cannot have linear phase.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: If an IIR filter is stable and if it can be physically realizable, then the filter cannot have linear phase.

10. What is the condition on the system function of a linear phase filter?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-iir-filters-analog-filters-q10.png)  
View Answer

Answer: d  
Explanation: A linear phase filter must have a system function that satisfies the condition  
[digital-signal-processing-questions-answers-design-iir-filters-analog-filters-q10a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-iir-filters-analog-filters-q10a.png)  
where z(-N) represents a delay of N units of time.

11. If the filter is in linear phase, then filter would have a mirror-image pole outside the unit circle for every pole inside the unit circle.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: For a linear phase filter, we know that  
[digital-signal-processing-questions-answers-design-iir-filters-analog-filters-q10a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-design-iir-filters-analog-filters-q10a.png)  
where z(-N) represents a delay of N units of time. But if this were the case, the filter would have a mirror image pole outside the unit circle for every pole inside the unit circle. Hence the filter would be unstable.

12. What is the order of operations to be performed in order to realize linear phase IIR filter?  
(i) Passing x(-n) through a digital filter H(z)  
(ii) Time reversing the output of H(z)  
(iii) Time reversal of the input signal x(n)  
(iv) Passing the result through H(z)  
a) (i),(ii),(iii),(iv)  
b) (iii),(i),(ii),(iv)  
c) (ii),(iii),(iv),(i)  
d) (i),(iii),(iv),(ii)  
View Answer

Answer: b  
Explanation: If the restriction on physical reliability is removed, it is possible to obtain a linear phase IIR filter, at least in principle. This approach involves performing a time reversal of the input signal x(n), passing x(-n) through a digital filter H(z), time reversing the output of H(z), and finally, passing the result through H(z) again.

13. When an application requires a linear phase filter, it should be an FIR filter.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The signal processing is computationally cumbersome and appear to offer no advantages over linear phase FIR filters. Consequently, when an application requires a linear phase, it should be an FIR filter.

“IIR Filter Design by Approximation of Derivatives”.

1. An analog filter can be converted into digital filter by approximating the differential equation by an equivalent difference equation.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: One of the simplest methods for converting an analog filter into digital filter is to approximate the differential equation by an equivalent difference equation.

2. Which of the following is the backward difference for the derivative of y(t) with respect to ‘t’ for t=nT?  
a) [y(n)+y(n+1)]/T  
b) [y(n)+y(n-1)]/T  
c) [y(n)-y(n+1)]/T  
d) [y(n)-y(n-1)]/T  
View Answer

Answer: d  
Explanation: For the derivate dy(t)/dt at time t=nT, we substitute the backward difference [y(nT)-y(nT-T)]/T. Thus  
dy(t)/dt =[y(nT)-y(nT-T)]/T  
=[y(n)-y(n-1)]/T  
where T represents the sampling interval and y(n)=y(nT).

3. Which of the following is true relation among s-domain and z-domain?  
a) s=(1+z-1)/T  
b) s=(1+z )/T  
c) s=(1-z-1)/T  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The analog differentiator with output dy(t)/dt has the system function H(s)=s, while the digital system that produces the output [y(n)-y(n-1)]/T has the system function H(z) =(1-z-1)/T. Thus the relation between s-domain and z-domain is given as  
s=(1-z-1)/T.

4. What is the second difference that is used to replace the second order derivate of y(t)?  
a) [y(n)-2y(n-1)+y(n-2)]/T  
b) [y(n)-2y(n-1)+y(n-2)]/T2  
c) [y(n)+2y(n-1)+y(n-2)]/T  
d) [y(n)+2y(n-1)+y(n-2)]/T2  
View Answer

Answer: b  
Explanation: We know that dy(t)/dt =[ y(n)-y(n-1)]/T  
Second order derivative of y(t) is d(dy(t)/dt)/dt=[y(n)-2y(n-1)+y(n-2)]/T2.

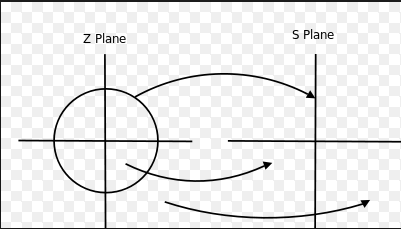
5. Which of the following in z-domain is equal to s-domain of second order derivate?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-entrance-exams-q5.png)d) None of the mentioned  
View Answer

Answer: a  
Explanation: We know that for a second order derivative  
[digital-signal-processing-questions-entrance-exams-q5a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-entrance-exams-q5a.png)

6. If s=jΩ and if Ω varies from -∞ to ∞, then what is the corresponding locus of points in z-plane?  
a) Circle of radius 1 with centre at z=0  
b) Circle of radius 1 with centre at z=1  
c) Circle of radius 1/2 with centre at z=1/2  
d) Circle of radius 1 with centre at z=1/2  
View Answer

Answer: c  
Explanation: We know that  
s=(1-z-1)/T  
=> z=1/(1-sT)  
Given s= jΩ => z = 1/(1- jΩT)  
Thus from the above equation if Ω varies from -∞ to ∞, then the corresponding locus of points in z-plane is a circle of radius 1/2 with centre at z=1/2.

7. Which of the following mapping is true between s-plane and z-domain?  
a) Points in LHP of the s-plane into points inside the circle in z-domain  
b) Points in RHP of the s-plane into points outside the circle in z-domain  
c) Points on imaginary axis of the s-plane into points onto the circle in z-domain  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: The below diagram explains the given question  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-entrance-exams-q7.png)

8. This mapping is restricted to the design of low pass filters and band pass filters having relatively small resonant frequencies.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The possible location of poles of the digital filter are confined to relatively small frequencies and as a consequence, the mapping is restricted to the design of low pass filters and band pass filters having relatively small resonant frequencies.

9. Which of the following filter transformation is not possible?  
a) High pass analog filter to low pass digital filter  
b) High pass analog filter to high pass digital filter  
c) Low pass analog filter to low pass digital filter  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: We know that only low pass and band pass filters with low resonant frequencies in the digital can be designed. So, it is not possible to transform a high pass analog filter into corresponding high pass digital filter.

10. It is possible to map the jΩ-axis into the unit circle.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: By proper choice of the coefficients of {αk}, it is possible to map the jΩ-axis into the unit circle.

“IIR Filter Design by Impulse Invariance”.

1. By impulse invariance method, the IIR filter will have a unit sample response h(n) that is the sampled version of the analog filter.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: In the impulse invariance method, our objective is to design an IIR filter having a unit sample response h(n) that is the sampled version of the impulse response of the analog filter. That is  
h(n)=h(nT) ; n=0,1,2…  
where T is the sampling interval.

2. If a continuous time signal x(t) with spectrum X(F) is sampled at a rate Fs=1/T samples per second, the spectrum of the sampled signal is:  
a) Non periodic repetition  
b) Non periodic non-repetition  
c) Periodic repetition  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: When a continuous time signal x(t) with spectrum X(F) is sampled at a rate Fs=1/T samples per second, the spectrum of the sampled signal is periodic repetition.

3. If a continuous time signal x(t) with spectrum X(F) is sampled at a rate Fs=1/T samples per second, then what is the scaled spectrum?  
a) X(F)  
b) Fs.X(F)  
c) X(F)/ Fs  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: When a continuous time signal x(t) with spectrum X(F) is sampled at a rate Fs=1/T samples per second, the spectrum of the sampled signal is periodic repetition of the scaled spectrum Fs.X(F).

4. When σ=0, then what is the condition on ‘r’?  
a) 0<r<1  
b) r=1  
c) r>1  
d) None of the mentioned  
View Answer

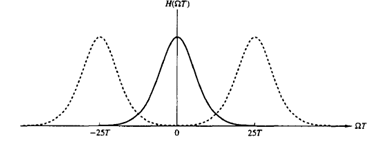
Answer: b  
Explanation: We know that z=esT  
Now substitute s=σ+jΩ and z=r.ejω, that is represent ‘z’ in the polar form  
On equating both sides, we get  
r=eσT  
Thus when σ=0, the value of ‘r’ varies from r=1.

5. What is the equation for normalized frequency?  
a) F/Fs  
b) F.Fs  
c) Fs/F  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: In the impulse invariance method, the normalized frequency f is given by  
f= F/Fs.

6. Aliasing occurs if the sampling rate Fs is more than twice the highest frequency contained in X(F).  
a) True  
b) False  
View Answer

Answer: b  
Explanation: Aliasing occurs if the sampling rate Fs is less than twice the highest frequency contained in X(F).

7. Which of the filters have a frequency response as shown in the figure below?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-campus-interviews-q7.png)  
a) Analog filter  
b) Digital filter without aliasing  
c) Digital filter with aliasing  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: In the given diagram, the continuous line is the frequency response of analog filter and dotted line is the frequency response of the corresponding digital filter with aliasing.

8. The frequency response given in the above question is for a low pass digital filter.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The above given frequency response depicts the frequency response of a low pass analog filter and the frequency response of the corresponding digital filter.

9. Sampling interval T is selected sufficiently large to completely avoid or at least minimize the effects of aliasing.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: The digital filter with frequency response H(ω) has the frequency response characteristics of the corresponding analog filter if the sampling interval T is selected sufficiently small to completely avoid or at least minimize the effects of aliasing.

10. Which of the following filters cannot be designed using impulse invariance method?  
a) Low pass  
b) Band pass  
c) Low and band pass  
d) High pass  
View Answer

Answer: d  
Explanation: It is clear that the impulse invariance method is in -appropriate for designing high pass filter due to the spectrum aliasing that results from the sampling process.

11. Which of the following is the correct relation between ω and Ω?  
a) Ω=ωT  
b) T=Ωω  
c) ω=ΩT  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: We know that z=esT  
Now substitute s=σ+jΩ and z=r.ejω, that is represent ‘z’ in the polar form  
On equating both sides, we get  
ω=ΩT.

12. When σ<0, then what is the condition on ‘r’?  
a) 0<r<1  
b) r=1  
c) r>1  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: We know that z=esT  
Now substitute s=σ+jΩ and z=r.ejω, that is represent ‘z’ in the polar form  
On equating both sides, we get  
r=eσT  
Thus when σ<0, the value of ‘r’ varies from 0<r<1.

13. When σ>0, then what is the condition on ‘r’?  
a) 0<r<1  
b) r=1  
c) r>1  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: We know that z=esT  
Now substitute s=σ+jΩ and z=r.ejω, that is represent ‘z’ in the polar form  
On equating both sides, we get  
r=eσT  
Thus when σ>0, the value of ‘r’ varies from r>1.

14. What is the period of the scaled spectrum Fs.X(F)?  
a) 2Fs  
b) Fs/2  
c) 4Fs  
d) Fs  
View Answer

Answer: d  
Explanation: When a continuous time signal x(t) with spectrum X(F) is sampled at a rate Fs=1/T samples per second, the spectrum of the sampled signal is periodic repetition of the scaled spectrum Fs.X(F) with period Fs.

Questions & Answers (MCQs) focuses on “Matched Z Transformation”.

1. In which of the following transformations, poles and zeros of H(s) are mapped directly into poles and zeros in the z-plane?  
a) Impulse invariance  
b) Bilinear transformation  
c) Approximation of derivatives  
d) Matched Z-transform  
View Answer

Answer: d  
Explanation: In this method of transforming analog filter into an equivalent digital filter is to map the poles and zeros of H(s) directly into poles and zeros in the z-plane.

2. Which of the following is true in matched z-transform?  
a) Poles of H(s) are directly mapped to poles in z-plane  
b) Zeros of H(s) are directly mapped to poles in z-plane  
c) Both of the mentioned  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: In the transformation of analog filter into digital filter by matched z-transform method, the poles and zeros of H(s) directly into poles and zeros in the z-plane.

3. In matched z-transform, the poles and zeros of H(s) are directly mapped into poles and zeros in z-plane.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: In this method of transforming analog filter into an equivalent digital filter is to map the poles and zeros of H(s) directly into poles and zeros in the z-plane.

4. The factor of the form (s-a) in H(s) is mapped into which of the following factors in z-domain?  
a) 1-eaTz  
b) 1-eaTz-1  
c) 1-e-aTz-1  
d) 1+eaTz-1  
View Answer

Answer: b  
Explanation: If T is the sampling interval, then each factor of the form (s-a) in H(s) is mapped into the factor (1-eaTz-1) in the z-domain.

5. The factor of the form (s+a) in H(s) is mapped into which of the following factors in z-domain?  
a) 1-eaTz  
b) 1-eaTz-1  
c) 1-e-aTz-1  
d) 1+eaTz-1  
View Answer

Answer: c  
Explanation: If T is the sampling interval, then each factor of the form (s+a) in H(s) is mapped into the factor (1-e-aTz-1) in the z-domain.

6. If the factor of the form (s-a) in H(s) is mapped into 1-eaTz-1 in the z-domain, the that kind of transformation is called as:  
a) Impulse invariance  
b) Bilinear transformation  
c) Approximation of derivatives  
d) Matched Z-transform  
View Answer

Answer: d  
Explanation: If T is the sampling interval, then each factor of the form (s-a) in H(s) is mapped into the factor (1-eaTz-1) in the z-domain. This mapping is called the matched z-transform.

7. The poles obtained from matched z-transform are identical to poles obtained from which of the following transformations?  
a) Bilinear transformation  
b) Impulse invariance  
c) Approximation of derivatives  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: We observe that the poles obtained from the matched z-transform are identical to the poles obtained with the impulse invariance method.

8. The zero positions obtained from matched z-transform and impulse invariance method are not same.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: We observe that the poles obtained from the matched z-transform are identical to the poles obtained with the impulse invariance method. However, the two techniques result in different zero positions.

9. The sampling interval in the matched z-transform must be properly selected to yield the pole and zero locations at the equivalent position in the z-plane.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: To preserve the frequency response characteristic of the analog filter, the sampling interval in the matched z-transformation must be properly selected to yield the pole and zero locations at the equivalent position in the z-plane.

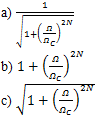
10. What should be value of sampling interval T, to avoid aliasing?  
a) Zero  
b) Sufficiently large  
c) Sufficiently small  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: Aliasing in this matched z-transformation can be avoided by selecting the sampling interval T sufficiently small.

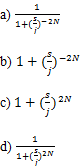
“Characteristics of Commonly Used Analog Filters”.

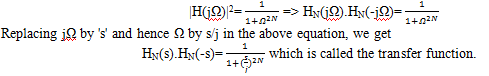
1. Low pass Butterworth filters are also called as:  
a) All-zero filter  
b) All-pole filter  
c) Pole-zero filter  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: Low pass Butterworth filters are also called as all-pole filters because it has only non-zero poles.

2. What is the equation for magnitude square response of a low pass Butterworth filter?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-aptitude-test-q2.png)d) None of the mentioned  
View Answer

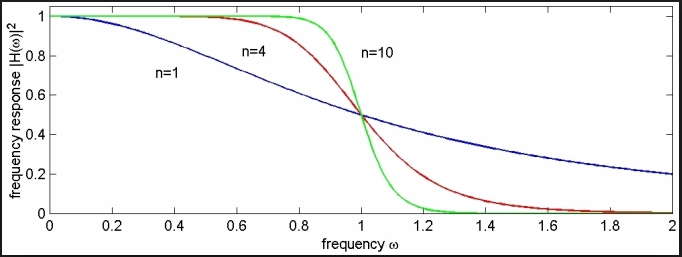
Answer: a  
Explanation: A Butterworth is characterized by the magnitude frequency response  
[digital-signal-processing-aptitude-test-q2a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-aptitude-test-q2a.png)  
where N is the order of the filter and ΩC is defined as the cutoff frequency.

3. What is the transfer function of magnitude squared frequency response of the normalized low pass Butterworth filter?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-aptitude-test-q3.png)  
View Answer

Answer: d  
Explanation: We know that the magnitude squared frequency response of a normalized low pass Butterworth filter is given as  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-aptitude-test-q3a.png)

4. Which of the following is the band edge value of |H(Ω)|2?  
a) (1+ε2)  
b) (1-ε2)  
c) 1/(1+ε2)  
d) 1/(1-ε2)  
View Answer

Answer: c  
Explanation: 1/(1+ε2) gives the band edge value of the magnitude square response |H(Ω)|2.

5. The magnitude square response shown in the below figure is for which of the following given filters?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-aptitude-test-q5.png)  
a) Butterworth  
b) Chebyshev  
c) Elliptical  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: The magnitude square response shown in the given figure is for Butterworth filter.

6. What is the order of a low pass Butterworth filter that has a -3dB bandwidth of 500Hz and an attenuation of 40dB at 1000Hz?  
a) 4  
b) 5  
c) 6  
d) 7  
View Answer

Answer: d  
Explanation: Given Ωc=1000π and Ωs=2000π  
For an attenuation of 40dB, δ2=0.01. We know that  
[digital-signal-processing-aptitude-test-q6](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-aptitude-test-q6.png)  
Thus by substituting the corresponding values in the above equation, we get N=6.64  
To meet the desired specifications, we select N=7.

7. Which of the following is true about type-1 chebyshev filter?  
a) Equi-ripple behavior in pass band  
b) Monotonic characteristic in stop band  
c) Both of the mentioned  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: Type-1 chebyshev filters are all-pole filters that exhibit equi-ripple behavior in pass band and a monotonic characteristic in the stop band.

8. Type-2 chebyshev filters consists of:  
a) Only poles  
b) Both poles and zeros  
c) Only zeros  
d) Cannot be determined  
View Answer

Answer: b  
Explanation: Type-1 chebyshev filters are all-pole filters where as the family of type-2 chebyshev filters contains both poles and zeros.

9. Which of the following is false about the type-2 chebyshev filters?  
a) Monotonic behavior in the pass band  
b) Equi-ripple behavior in the stop band  
c) Both of the mentioned  
d) Monotonic behavior in the stop band.  
View Answer

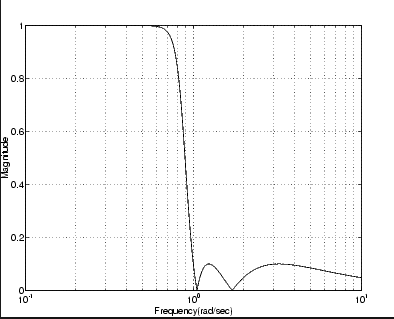
Answer: d  
Explanation: Type-2 chebyshev filters exhibit equi-ripple behavior in stop band and a monotonic characteristic in the pass band.

10. The zeros of type-2 class of chebyshev filters lies on:  
a) Imaginary axis  
b) Real axis  
c) Zero  
d) Cannot be determined  
View Answer

Answer: a  
Explanation: The zeros of this class of filters lie on the imaginary axis in the s-plane.

11. Which of the following defines a chebyshev polynomial of order N, TN(x)?  
a) cos(Ncos-1x) for all x  
b) cosh(Ncosh-1x) for all x  
c) cos(Ncos-1x), |x|≤1  
cosh(Ncosh-1x), |x|>1  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: In order to understand the frequency-domain behavior of chebyshev filters, it is utmost important to define a chebyshev polynomial and then its properties. A chebyshev polynomial of degree N is defined as  
TN(x) = cos(Ncos-1x), |x|≤1  
cosh(Ncosh-1x), |x|>1

12. The frequency response shown in the figure below belongs to which of the following filters?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-aptitude-test-q12.png)  
a) Type-1 chebyshev  
b) Type-2 chebyshev  
c) Butterworth  
d) Elliptical  
View Answer

Answer: b  
Explanation: Since the pass band is monotonic in behavior and the stop band exhibit equi-ripple behavior, it is the magnitude square response of a type-2 chebyshev filter.

13. What is the order of the type-2 chebyshev filter whose magnitude square response is as shown in the above figure?  
a) 2  
b) 4  
c) 6  
d) 3  
View Answer

Answer: d  
Explanation: Since from the magnitude square response of the type-2 chebyshev filter, it has odd number of maxima and minima in the stop band, the order of the filter is odd i.e., 3.

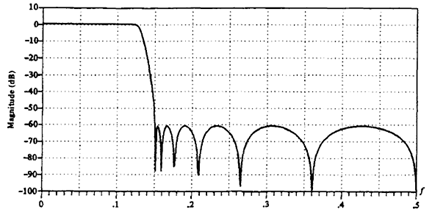
14. Which of the following is true about the magnitude square response of an elliptical filter?  
a) Equi-ripple in pass band  
b) Equi-ripple in stop band  
c) Equi-ripple in pass band and stop band  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: An elliptical filter is a filter which exhibit equi-ripple behavior in both pass band and stop band of the magnitude square response.

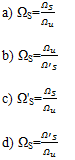
15. Bessel filters exhibit a linear phase response over the pass band of the filter.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: An important characteristic of the Bessel filter is the linear phase response over the pass band of the filter. As a consequence, Bessel filters has a larger transition bandwidth, but its phase is linear within the pass band.

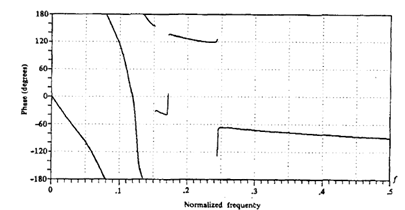
“Frequency Transformations in the Analog Domain”.

1. The following frequency characteristic is for which of the following filter?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-problems-q1.png)  
a) Type-2 Chebyshev filter  
b) Type-1 Chebyshev filter  
c) Butterworth filter  
d) Bessel filter  
View Answer

Answer: a  
Explanation: The frequency characteristic given in the figure is the magnitude response of a 13-order type-2 chebyshev filter.

2. Which of the following is the backward design equation for a low pass-to-high pass transformation?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-problems-q2.png)  
View Answer

Answer: b  
Explanation: If Ωu is the desired pass band edge frequency of new high pass filter, then the transfer function of this new high pass filter is obtained by using the transformation s→ Ωu /s. If ΩS and Ω’S are the stop band frequencies of prototype and transformed filters respectively, then the backward design equation is given by  
[digital-signal-processing-problems-q2a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-problems-q2a.png).

3. Which of the following filter has a phase spectrum as shown in figure?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-problems-q3.png)  
a) Chebyshev filter  
b) Butterworth filter  
c) Bessel filter  
d) Elliptical filter  
View Answer

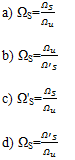
Answer: d  
Explanation: The phase response given in the figure belongs to the frequency characteristic of a 7-order elliptic filter.

4. What is the pass band edge frequency of an analog low pass normalized filter?  
a) 0 rad/sec  
b) 0.5 rad/sec  
c) 1 rad/sec  
d) 1.5 rad/sec  
View Answer

Answer: c  
Explanation: Let H(s) denote the transfer function of a low pass analog filter with a pass band edge frequency ΩP equal to 1 rad/sec. This filter is known as analog low pass normalized prototype.

5. Which of the following is a low pass-to-high pass transformation?  
a) s→ s / Ωu  
b) s→ Ωu / s  
c) s→ Ωu.s  
d) None of the mentioned  
View Answer

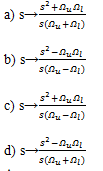
Answer: b  
Explanation: The low pass-to-high pass transformation is simply achieved by replacing s by 1/s. If the desired high pass filter has the pass band edge frequency Ωu, then the transformation is  
s→ Ωu / s.

6. Which of the following is the backward design equation for a low pass-to-low pass transformation?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-problems-q2.png)  
View Answer

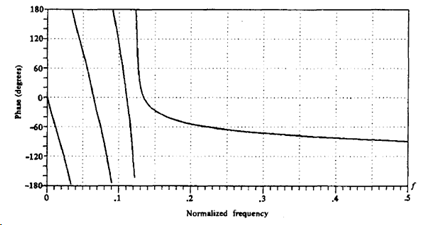
Answer: d  
Explanation: If Ωu is the desired pass band edge frequency of new low pass filter, then the transfer function of this new low pass filter is obtained by using the transformation s→ s / Ωu. If ΩS and Ω’S are the stop band frequencies of prototype and transformed filters respectively, then the backward design equation is given by  
[digital-signal-processing-problems-q6](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-problems-q6.png).

7. If H(s) is the transfer function of a analog low pass normalized filter and Ωu is the desired pass band edge frequency of new low pass filter, then which of the following transformation has to be performed?  
a) s→ s / Ωu  
b) s→ s .Ωu  
c) s→ Ωu/s  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: If Ωu is the desired pass band edge frequency of new low pass filter, then the transfer function of this new low pass filter is obtained by using the transformation s→ s / Ωu.

8. Which of the following is a low pass-to-band pass transformation?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-problems-q8.png)  
View Answer

Answer: c  
Explanation: If Ωu and Ωl are the upper and lower cutoff pass band frequencies of the desired band pass filter, then the transformation to be performed on the normalized low pass filter is  
[digital-signal-processing-problems-q8a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-problems-q8a.png)

9. Which of the following filter has a phase spectrum as shown in figure?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-problems-q9.png)  
a) Chebyshev filter  
b) Butterworth filter  
c) Bessel filter  
d) Elliptical filter  
View Answer

Answer: a  
Explanation: The phase response given in the figure belongs to the frequency characteristic of a 13-order type-1 Chebyshev filter.

10. If [digital-signal-processing-problems-q10](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-problems-q10.png), then which of the following is the backward design equation for a low pass-to-band pass transformation?  
a) ΩS= |B|  
b) ΩS= |A|  
c) ΩS= Max{|A|,|B|}  
d) ΩS= Min{|A|,|B|}  
View Answer

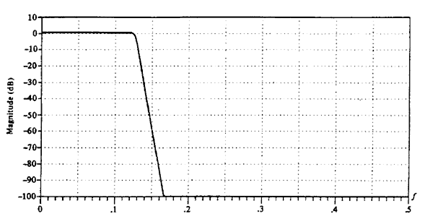
Answer: d  
Explanation: If Ωu and Ωl are the upper and lower cutoff pass band frequencies of the desired band pass filter and Ω1 and Ω2 are the lower and upper cutoff stop band frequencies of the desired band pass filter, then the backward design equation is  
ΩS= Min{|A|,|B|}  
where, [digital-signal-processing-problems-q10](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-problems-q10.png).

11. If[digital-signal-processing-problems-q11](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-problems-q11.png), then which of the following is the backward design equation for a low pass-to-band stop transformation?  
a) ΩS= Max{|A|,|B|}  
b) ΩS= Min{|A|,|B|}  
c) ΩS= |B|  
d) ΩS= |A|  
View Answer

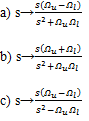
Answer: b  
Explanation: If Ωu and Ωl are the upper and lower cutoff pass band frequencies of the desired band stop filter and Ω1 and Ω2 are the lower and upper cutoff stop band frequencies of the desired band stop filter, then the backward design equation is  
ΩS= Min{|A|,|B|}  
where, [digital-signal-processing-problems-q11](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-problems-q11.png).

12. Which of the following is a low pass-to-high pass transformation?  
a) s→ s / Ωu  
b) s→ Ωu / s  
c) s→ Ωu.s  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The low pass-to-high pass transformation is simply achieved by replacing s by 1/s. If the desired high pass filter has the pass band edge frequency Ωu, then the transformation is  
s→ Ωu / s

13. The following frequency characteristic is for which of the following filter?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-problems-q13.png)  
a) Type-2 Chebyshev filter  
b) Type-1 Chebyshev filter  
c) Butterworth filter  
d) Bessel filter  
View Answer

Answer: c  
Explanation: The frequency characteristic given in the figure is the magnitude response of a 37-order Butterworth filter.

14. Which of the following is a low pass-to-band stop transformation?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-problems-q14.png)d) None of the mentioned  
View Answer

Answer: c  
Explanation: If Ωu and Ωl are the upper and lower cutoff pass band frequencies of the desired band stop filter, then the transformation to be performed on the normalized low pass filter is  
[digital-signal-processing-problems-q14a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-problems-q14a.png)

Questions & Answers focuses on “Frequency Transformations in the Digital Domain”.

1. The frequency transformation in the digital domain involves replacing the variable z-1 by a rational function g(z-1).  
a) True  
b) False  
View Answer

Answer: a  
Explanation: As in the analog domain, frequency transformations can be performed on a digital low pass filter to convert it to either a band pass, band stop or high pass filter. The transformation involves the replacing of the variable z-1 by a rational function g(z-1).

2. The mapping z-1→ g(z-1) must map inside the unit circle in the z-plane into:  
a) Outside the unit circle  
b) On the unit circle  
c) Inside the unit circle  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The map z-1→ g(z-1) must map inside the unit circle in the z-plane into itself to apply digital frequency transformation.

3. The unit circle must be mapped outside the unit circle.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: For the map z-1→ g(z-1) to be a valid digital frequency transformation, then the unit circle also must be mapped inside the unit circle.

4. The mapping z-1→ g(z-1) must be:  
a) Low pass  
b) High pass  
c) Band pass  
d) All-pass  
View Answer

Answer: d  
Explanation: We know that the unit circle must be mapped inside the unit circle.  
Thus it implies that for r=1, e-jω = g(e-jω)=|g(ω)|.ej arg [ g(ω) ]  
It is clear that we must have |g(ω)|=1 for all ω. That is, the mapping is all-pass.

5. What should be the value of |ak| to ensure that a stable filter is transformed into another stable filter?  
a) < 1  
b) =1  
c) > 1  
d) 0  
View Answer

Answer: a  
Explanation: The value of |ak| < 1 to ensure that a stable filter is transformed into another stable filter to satisfy the condition to satisfy the condition 1.

6. Which of the following methods are inappropriate to design high pass and many band pass filters?  
a) Impulse invariance  
b) Mapping of derivatives  
c) Impulse invariance & Mapping of derivatives  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: We know that the impulse invariance method and mapping of derivatives are inappropriate to use in the designing of high pass and band pass filters.

7. The impulse invariance method and mapping of derivatives are inappropriate to use in the designing of high pass and band pass filters due to aliasing problem.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: We know that the impulse invariance method and mapping of derivatives are inappropriate to use in the designing of high pass and band pass filters due to aliasing problem.

8. We can employ the analog frequency transformation followed by conversion of the result into the digital domain by use of impulse invariance and mapping the derivatives.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: Since there is a problem of aliasing in designing high pass and many band pass filters using impulse invariance and mapping of derivatives, we cannot employ the analog frequency transformation followed by conversion of the result into digital domain by use of these two mappings.

9. It is better to perform the mapping from an analog low pass filter into a digital low pass filter by either of these mappings and then perform the frequency transformation in the digital domain.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: It is better to perform the mapping from an analog low pass filter into a digital low pass filter by either of these mappings and then perform the frequency transformation in the digital domain because by this kind of frequency transformation, problem of aliasing is avoided.

10. In which of the following transformations, it doesn’t matter whether the frequency transformation is performed in the analog domain or in frequency domain?  
a) Impulse invariance  
b) Mapping of derivatives  
c) Bilinear transformation  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: In the case of bilinear transformation, where aliasing is not a problem, it does not matter whether the frequency transformation is performed in the analog domain or in frequency domain.

Questions & Answers (MCQs) focuses on “Pade Approximation Method”.

1. Which of the following techniques of designing IIR filters do not involve the conversion of an analog filter into digital filter?  
a) Bilinear transformation  
b) Impulse invariance  
c) Approximation of derivatives  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: Except for the impulse invariance method, the design techniques for IIR filters involve the conversion of an analog filter into a digital filter by some mapping from the s-plane to the z-plane.

2. Using which of the following methods, a digital IIR filter can be directly designed?  
a) Pade approximation  
b) Least square design in time domain  
c) Least square design in frequency domain  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: There are several methods for designing digital filters directly. The three techniques are Pade approximation and least square method, the specifications are given in the time domain and the design is carried in time domain. The other one is least squares technique in which the design is carried out in frequency domain.

3. What is the number of parameters that a filter consists of?  
a) M+N+1  
b) M+N  
c) M+N-1  
d) M+N-2  
View Answer

Answer: a  
Explanation: The filter has L=M+N+1 parameters, namely, the coefficients {ak} and {bk}, which can be selected to minimize some error criterion.

4. The minimization of ε involves the solution of a set of non-linear equations.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: In general, h(n) is a non-linear function of the filter parameters and hence the minimization of ε involves the solution of a set of non-linear equations.

5. What should be the upper limit of the solution to match h(n) perfectly to the desired response hd(n)?  
a) L  
b) L+1  
c) L-1  
d) L+2  
View Answer

Answer: c  
Explanation: If we select the upper limit as U=L-1, it is possible to match h(n) perfectly to the desired response hd(n) for 0 < n < M+N.

6. For how many values of the impulse response, a perfect match is present between h(n) and hd(n)?  
a) L  
b) M+N+1  
c) 2L-M-N-1  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: We obtain a perfect match between h(n) and the desired response hd(n) for the first L values of the impulse response and we also know that L=M+N+1.

7. The degree to which the design technique produces acceptable filter designs depends in part on the number of filter coefficients selected.  
a) True  
b) False  
View Answer

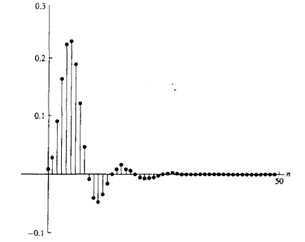
Answer: a  
Explanation: The degree to which the design technique produces acceptable filter designs depends in part on the number of filter coefficients selected. Since the design method matches hd(n) only up to the number of filter parameters, the more complex the filter, the better the approximation to hd(n).

8. According to this method of designing, the filter should have which of the following in large number?  
a) Only poles  
b) Both poles and zeros  
c) Only zeros  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The major limitation of Pade approximation method, namely, the resulting filter must contain a large number of poles and zeros.

9. Which of the following conditions are in the favor of Pade approximation method?  
a) Desired system function is rational  
b) Prior knowledge of the number of poles and zeros  
c) Both of the mentioned  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The Pade approximation method results in a perfect match to Hd(z) when the desired system function is rational and we have prior knowledge of the number of poles and zeros in the system.

10. Which of the following filters will have an impulse response as shown in the below figure?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-pade-approximation-method-q10.png)  
a) Butterworth filters  
b) Type-I chebyshev filter  
c) Type-II chebyshev filter  
d) None of the mentioned  
View Answer

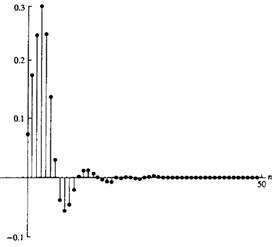
Answer: a  
Explanation: The diagram that is given in the question is the impulse response of Butterworth filter.

11. For what number of zeros, the approximation is poor?  
a) 3  
b) 4  
c) 5  
d) 6  
View Answer

Answer: a  
Explanation: We observe that when the number of zeros in minimum, that is when M=3, the resulting frequency response is a relatively poor approximation to the desired response.

12. Which of the following pairs of M and N will give a perfect match?  
a) 3,6  
b) 3,4  
c) 3,5  
d) 4,5  
View Answer

Answer: d  
Explanation: When M is increased from three to four, we obtain a perfect match with the desired Butterworth filter not only for N=4 but for N=5, and in fact, for larger values of N.

13. Which of the following filters will have an impulse response as shown in the below figure?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-pade-approximation-method-q13.png)  
a) Butterworth filters  
b) Type-I chebyshev filter  
c) Type-II chebyshev filter  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The diagram that is given in the question is the impulse response of type-II chebyshev filter.

Questions & Answers (MCQs) focuses on “Least Squares Design Methods”.

1. Which of the following filter we use in least square design methods?  
a) All zero  
b) All pole  
c) Pole-zero  
d) Any of the mentioned  
View Answer

Answer: b  
Explanation: Let us assume that hd(n) is specified for n > 0, and the digital filter is an all-pole filter.

2. Which of the following are cascaded in this method?  
a) Hd(z), H(z)  
b) 1/Hd(z), 1/H(z)  
c) 1/Hd(z), H(z)  
d) Hd(z), 1/H(z)  
View Answer

Answer: d  
Explanation: In this method, we consider the cascade connection of the desired filter Hd(z) with the reciprocal, all zero filter 1/H(z).

3. If δ(n) is the input, then what is the ideal output of yd(n)?  
a) δ(n)  
b) 0  
c) u(n)  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: We excite the cascade configuration by the unit sample sequence δ(n). Thus the input to the inverse system 1/H(z) is hd(n) and the output is y(n). Ideally, yd(n)= δ(n).

4. What should be the value of y(n) at n=0?  
a) 0  
b) -1  
c) 1  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The condition that yd(0)= y(0)=1 is satisfied by selecting b0=hd(0).

5. The error between the desired output and actual output is represented by y(n).  
a) True  
b) False  
View Answer

Answer: a  
Explanation: For n > 0, y(n) represents the error between the desired output yd(n)=0 and the actual output.

6. Which of the following parameters are selected to minimize the sum of squares of the error sequence?  
a) {bk}  
b) {ak}  
c) Both of the mentioned  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The parameters {ak} are selected to minimize the sum of squares of the error sequence.

7. By integrating the error equation with respect to the parameters {ak}, we obtain set of linear equations.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: By differentiating the square of the error sequence with respect to the parameters {ak}, it is easily established that we obtain the set of linear equations.

8. Which of the following operation is done on the sequence in least square design method?  
a) Convolution  
b) DFT  
c) Circular convolution  
d) Correlation  
View Answer

Answer: d  
Explanation: In a practical design problem, the desired impulse response hd(n) is specified for a finite set of points, say 0 < n <L where L>> N. In such a case, the correlation sequence can be computed from the finite sequence hd(n).

9. The least squares method can also be used in a pole-zero approximation for Hd(z).  
a) True  
b) False  
View Answer

Answer: a  
Explanation: We know that we can perform pole-zero approximation for Hd(z) by using the least squares method.

10. In which of the following condition we can use the desired response hd(n)?  
a) n < M  
b) n=M  
c) n > M  
d) none of the mentioned  
View Answer

Answer: c  
Explanation: Nevertheless, we can use the desired response hd(n) for n < M to construct an estimate of hd(n).

11. Which of the following parameters are used to determine zeros of the filter?  
a) {bk}  
b) {ak}  
c) Both of the mentioned  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: The parameters {bk} are selected to determine the zeros of the filter that can be obtained where h(n)=hd(n).

12. The foregoing approach for determining the poles and zeros of H(z) is sometimes called Prony’s method.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: We find the coefficients {bk} by pade approximation and find the coefficients {ak} by least squares method. Thus the foregoing approach for determining the poles and zeros of H(z) is sometimes called as Prony’s method.

Questions & Answers (MCQs) focuses on “FIR Least Squares Inverse Filters”.

1. Wiener filter is an FIR least-squares inverse filter.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: FIR least square filters are also called as Wiener filters.

2. If h(n) is the impulse response of an LTI system and hI(n) is the impulse response of the inverse LTI system, then which of the following is true?  
a) h(n).hI(n)=1  
b) h(n).hI(n)=δ(n)  
c) h(n)\*hI(n)=1  
d) h(n)\*hI(n)=δ(n)  
View Answer

Answer: d  
Explanation: The inverse to a linear time invariant system with impulse response h(n) is defined as the system whose impulse response is hI(n), satisfy the following condition h(n)\*hI(n)=δ(n).

3. If H(z) is the system function of an LTI system and HI(z) is the system function of the inverse LTI system, then which of the following is true?  
a) H(z)\*HI(z)=1  
b) H(z)\*HI(z)=δ(n)  
c) H(z).HI(z)=1  
d) H(z).HI(z)=δ(n)  
View Answer

Answer: c  
Explanation: The inverse to a linear time invariant system with impulse response h(n) and system function H(z) is defined as the system whose impulse response is hI(n) and system function HI(z), satisfy the following condition  
H(z).HI(z)=1.

4. It is not desirable to restrict the inverse filter to be FIR.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: In most of the practical applications, it is desirable to restrict the inverse filter to be an FIR filter.

5. Which of the following method is used to restrict the inverse filter to be FIR?  
a) Truncating hI(n)  
b) Expanding hI(n)  
c) Truncating HI(z)  
d) None of the mentioned  
View Answer

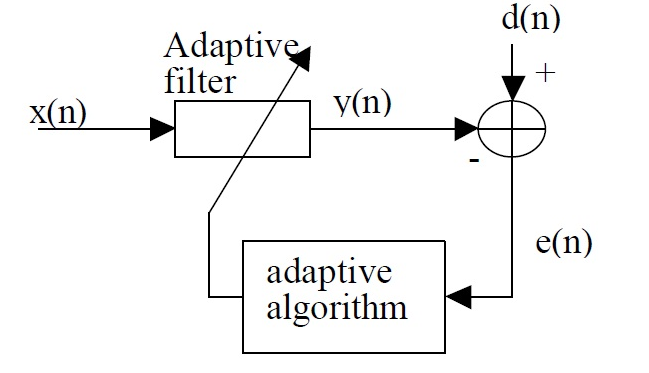
Answer: a  
Explanation: In many practical applications, it is desirable to restrict the inverse filter to be FIR. One of the simple method to get this requirement is to truncate hI(n).

6. What should be the length of the truncated filter?  
a) M  
b) M-1  
c) M+1  
d) Infinite  
View Answer

Answer: c  
Explanation: In the process of truncating, we incur a total squared approximation error where M+1 is the length of the truncated filter.

7. Which of the following criterion can be used to optimize the M+1 filter coefficients?  
a) Pade approximation method  
b) Least squares error criterion  
c) Both of the mentioned  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: We can use the least squares error criterion to optimize the M+1 coefficients of the FIR filter.

8. Which of the following filters have a block diagram as shown in the figure?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-fir-least-squares-inverse-filters-q8.png)  
a) Pade wiener filter  
b) Pade FIR filter  
c) Least squares FIR filter  
d) Least squares wiener filter  
View Answer

Answer: d  
Explanation: Since from the block diagram, the coefficients of the FIR filter coefficients are optimized by the least squares error criterion, it belongs to the least squares FIR inverse filter or wiener filter.

9. The auto correlation of the sequence is required to minimize ε.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: When ε is minimized with respect to the filter coefficients, we obtain the set of linear equations which are dependent on the auto correlation sequence of the signal h(n).

10. Which of the following are required to minimize the value of ε?  
a) rhh(l)  
b) rdh(l)  
c) d(n)  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: When ε is minimized with respect to the filter coefficients, we obtain the set of linear equations  
[digital-signal-processing-questions-answers-fir-least-squares-inverse-filters-q10](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-fir-least-squares-inverse-filters-q10.png)  
and we know that rdh(l) depends on the desired output d(n).

11. FIR filter that satisfies [digital-signal-processing-questions-answers-fir-least-squares-inverse-filters-q10](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-fir-least-squares-inverse-filters-q10.png)is known as wiener filter.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The optimum, in the least square sense, FIR filter that satisfies the linear equations in [digital-signal-processing-questions-answers-fir-least-squares-inverse-filters-q10](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-fir-least-squares-inverse-filters-q10.png)is called the wiener filter.

12. What should be the desired response for an optimum wiener filter to be an approximate inverse filter?  
a) u(n)  
b) δ(n)  
c) u(-n)  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: If the optimum least squares FIR filter is to be an approximate inverse filter, the desired response is  
d(n)=δ(n).

13. If the set of linear equations from the equation[digital-signal-processing-questions-answers-fir-least-squares-inverse-filters-q10](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-fir-least-squares-inverse-filters-q10.png)are expressed in matrix form, then what is the type of matrix obtained?  
a) Symmetric matrix  
b) Skew symmetric matrix  
c) Toeplitz matrix  
d) Triangular matrix  
View Answer

Answer: c  
Explanation: We observe that the matrix is not only symmetric but it also has the special property that all the elements along any diagonal are equal. Such a matrix is called a Toeplitz matrix and lends itself to efficient inversion by means of an algorithm.

14. What is the number of computations proportional to, in Levinson-Durbin algorithm?  
a) M  
b) M2  
c) M3  
d) M1/2  
View Answer

Answer: b  
Explanation: The Levinson-Durbin algorithm is the algorithm which is used for the efficient inversion of Toeplitz matrix which requires a number of computations proportional to M2 instead of the usual M3.

Questions & Answers (MCQs) focuses on “Design of IIR Filters in Frequency Domain”.

1. Filter parameter optimization technique is used for designing of which of the following?  
a) FIR in time domain  
b) FIR in frequency domain  
c) IIR in time domain  
d) IIR in frequency domain  
View Answer

Answer: d  
Explanation: We describe a filter parameter optimization technique carried out in the frequency domain that is representative of frequency domain design methods.

2. In this type of designing, the system function of IIR filter is expressed in which form?  
a) Parallel form  
b) Cascade form  
c) Mixed form  
d) Any of the mentioned  
View Answer

Answer: b  
Explanation: The design is most easily carried out with the system function for the IIR filter expressed in the cascade form as  
H(z)=G.A(z).

3. It is more convenient to deal with the envelope delay as a function of frequency.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Instead of dealing with the phase response ϴ(ω), it is more convenient to deal with the envelope delay as a function of frequency.

4. Which of the following gives the equation for envelope delay?  
a) dϴ(ω)/dω  
b) ϴ(ω)  
c) -dϴ(ω)/dω  
d) -ϴ(ω)  
View Answer

Answer: c  
Explanation: Instead of dealing with the phase response ϴ(ω), it is more convenient to deal with the envelope delay as a function of frequency, which is  
Tg(ω)= -dϴ(ω)/dω.

5. What is the error in magnitude at the frequency ωk?  
a) G.A(ωk) + Ad(ωk)  
b) G.A(ωk) – Ad(ωk)  
c) G.A(ωk) – A(ωk)  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The error in magnitude at the frequency ωk is G.A(ωk) – Ad(ωk) for 0 ≤ |ω| ≤ π, where Ad(ωk) is the desired magnitude response at ωk.

6. What is the error in delay at the frequency ωk?  
a) Tg(ωk)- Td(ωk)  
b) Tg(ωk)+ Td(ωk)  
c) Td(ωk)  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: Similarly as in the previous question, the error in delay at ωk is defined as Tg(ωk)- Td(ωk), where Td(ωk) is the desired delay response.

7. The choice of Td(ωk) for error in delay is complicated.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: We know that the error in delay is defined as Tg(ωk)- Td(ωk). However, the choice of Td(ωk) for error in delay is complicated by the difficulty in assigning a nominal delay of the filter.

8. If the error in delay is defined as Tg(ωk)- Tg(ω0)- Td(ωkk), then what is Tg(ω0)?  
a) Filter delay at nominal frequency in stop band  
b) Filter delay at nominal frequency in transition band  
c) Filter delay at nominal frequency  
d) Filter delay at nominal frequency in pass band  
View Answer

Answer: d  
Explanation: We are led to define the error in delay as Tg(ωk)- Tg(ω0)- Td(ωk), where Tg(ω0) is the filter delay at some nominal centre frequency in the pass band of the filter.

9. We cannot choose any arbitrary function for the errors in magnitude and delay.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: As a performance index for determining the filter parameters, one can choose any arbitrary function of the errors in magnitude and delay.

10. What does ‘p’ represents in the arbitrary function of error?  
a) 2K- dimension vector  
b) 3K- dimension vector  
c) 4K- dimension vector  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: In the error function ‘p’ denotes the 4K dimension vector of the filter coefficients.

11. What should be the value of λ for the error to be placed entirely on delay?  
a) 1  
b) 1/2  
c) 0  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: The emphasis on the errors affecting the design may be placed entirely on the delay by taking the value of λ as 1.

12. What should be the value of λ for the error to be placed equally on magnitude and delay?  
a) 1  
b) 1/2  
c) 0  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The emphasis on the errors affecting the design may be equally weighted between magnitude and delay by taking the value of λ as 1/2.

13. Which of the following is true about the squared-error function E(p,G)?  
a) Linear function of 4K parameters  
b) Linear function of 4K+1 parameters  
c) Non-Linear function of 4K parameters  
d) Non-Linear function of 4K+1 parameters  
View Answer

Answer: d  
Explanation: The squared error function E(p,G) is a non-linear function of 4K+1 parameters.

14. Minimization of the error function over the remaining 4K parameters is performed by an iterative method.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Due to the non-linear nature of E(p,G), its minimization over the remaining 4K parameters is performed by an iterative numerical optimization method.

15. The iterative process may converge to a global minimum.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: The major difficulty with any iterative procedure that searches for the parameter values that minimize a non-linear function is that the process may converge to a local minimum instead of a global minimum.

Questions & Answers (MCQs) focuses on “Specifications and Classification of Analog Filters”.

1. What is the region between origin and the pass band frequency in the magnitude frequency response of a low pass filter?  
a) Stop band  
b) Pass band  
c) Transition band  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: From the magnitude frequency response of a low pass filter, we can state that the region before pass band frequency is known as ‘pass band’ where the signal is passed without huge losses.

2. What is the region between stop band and the pass band frequencies in the magnitude frequency response of a low pass filter?  
a) Stop band  
b) Pass band  
c) Transition band  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: From the magnitude frequency response of a low pass filter, we can state that the region between pass band and stop band frequencies is known as ‘transition band’ where no specifications are provided.

3. What is the region after the stop band frequency in the magnitude frequency response of a low pass filter?  
a) Stop band  
b) Pass band  
c) Transition band  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: From the magnitude frequency response of a low pass filter, we can state that the region after stop band frequency is known as ‘stop band’ where the signal is stopped or restricted.

4. If δP is the forbidden magnitude value in the pass band and δS is the forbidden magnitude value in th stop band, then which of the following is true in the pass band region?  
a) 1- δS≤|H(jΩ)|≤1  
b) δP≤|H(jΩ)|≤1  
c) 0≤|H(jΩ)|≤ δS  
d) 1- δP≤|H(jΩ)|≤1  
View Answer

Answer: d  
Explanation: From the magnitude frequency response of the low pass filter, the hatched region in the pass band indicate forbidden magnitude value whose value is given as  
1- δP≤|H(jΩ)|≤1.

5. If δPis the forbidden magnitude value in the pass band and δS is the forbidden magnitude value in th stop band, then which of the following is true in the stop band region?  
a) 1- δP≤|H(jΩ)|≤1  
b) δP≤|H(jΩ)|≤1  
c) 0≤|H(jΩ)|≤ δS  
d) 1- δP≤|H(jΩ)|≤1  
View Answer

Answer: c  
Explanation: From the magnitude frequency response of the low pass filter, the hatched region in the stop band indicate forbidden magnitude value whose value is given as  
0≤|H(jΩ)|≤ δS.

6. What is the value of pass band ripple in dB?  
a) -20log(1- δP)  
b) -20log(δP)  
c) 20log(1- δP)  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: 1- δP is known as the pass band ripple or the pass band attenuation, and its value in dB is given as -20log(1- δP).

7. What is the value of stop band ripple in dB?  
a) -20log(1- δS)  
b) -20log(δS)  
c) 20log(1- δS)  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: δS is known as the stop band attenuation, and its value in dB is given as -20log(δS).

8. What is the pass band gain of a low pass filter with 1- δP as the pass band attenuation?  
a) -20log(1- δP)  
b) -20log(δP)  
c) 20log(δP)  
d) 20log(1- δP)  
View Answer

Answer: d  
Explanation: If 1- δP is the pass band attenuation, then the pass band gain is given by the formula 20log(1- δP).

9. What is the stop band gain of a low pass filter with δS as the pass band attenuation?  
a) -20log(1- δS)  
b) -20log(δS)  
c) 20log(δS)  
d) 20log(1- δS)  
View Answer

Answer: c  
Explanation: If δS is the stop band attenuation, then the stop band gain is given by the formula 20log(δS).

10. What is the cutoff frequency of a normalized filter?  
a) 2 rad/sec  
b) 1 rad/sec  
c) 0.5 rad/sec  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: A filter is said to be normalized if the cutoff frequency of the filter, Ωc is 1 rad/sec.

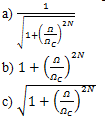
11. The low pass, high pass, band pass and band stop filters can be designed by applying a specific transformation to a normalized low pass filter.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: It is known that the low pass, high pass, band pass and band stop filters can be designed by applying a specific transformation to a normalized low pass filter. Therefore, a lot of importance is given to the design of normalized low pass analog filter.

Questions & Answers (MCQs) focuses on “Butterworth Filters”.

1. Which of the following is true in the case of Butterworth filters?  
a) Smooth pass band  
b) Wide transition band  
c) Not so smooth stop band  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: Butterworth filters have a very smooth pass band, which we pay for with a relatively wide transmission region.

2. What is the magnitude frequency response of a Butterworth filter of order N and cutoff frequency ΩC?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-butterworth-filters-q2.png)d) None of the mentioned  
View Answer

Answer: a  
Explanation: A Butterworth is characterized by the magnitude frequency response  
[digital-signal-processing-questions-answers-butterworth-filters-q2a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-butterworth-filters-q2a.png)  
where N is the order of the filter and ΩC is defined as the cutoff frequency.

3. What is the factor to be multiplied to the dc gain of the filter to obtain filter magnitude at cutoff frequency?  
a) 1  
b) √2  
c) 1/√2  
d) 1/2  
View Answer

Answer: c  
Explanation: The dc gain of the filter is the filter magnitude at Ω=0.  
We know that the filter magnitude is given by the equation  
[digital-signal-processing-questions-answers-butterworth-filters-q2a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-butterworth-filters-q2a.png)  
[digital-signal-processing-questions-answers-butterworth-filters-q3](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-butterworth-filters-q3.png)

4. What is the value of magnitude frequency response of a Butterworth low pass filter at Ω=0?  
a) 0  
b) 1  
c) 1/√2  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The magnitude frequency response of a Butterworth low pass filter is given as  
[digital-signal-processing-questions-answers-butterworth-filters-q2a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-butterworth-filters-q2a.png)At Ω=0 => |H(jΩ)|=1 for all N.

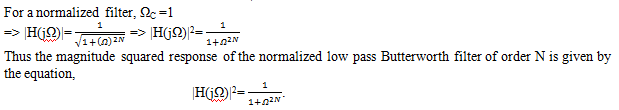
5. As the value of the frequency Ω tends to ∞, then |H(jΩ)| tends to:  
a) 0  
b) 1  
c) ∞  
d) None of the mentioned  
View Answer

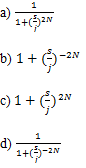
Answer: a  
Explanation: We know that the magnitude frequency response of a Butterworth filter of order N is given by the expression  
[digital-signal-processing-questions-answers-butterworth-filters-q2a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-butterworth-filters-q2a.png)  
In the above equation, if Ω→∞ then |H(jΩ)|→0.

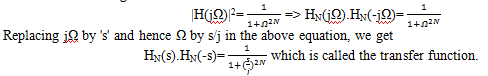
6. |H(jΩ)| is a monotonically increasing function of frequency.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: |H(jΩ)| is a monotonically decreasing function of frequency, i.e., |H(jΩ2)| < |H(jΩ1)| for any values of Ω1 and Ω2 such that 0 ≤ Ω1 < Ω2.

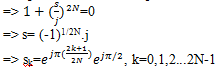
7. What is the magnitude squared response of the normalized low pass Butterworth filter?  
a) 1/(1+Ω-2N)  
b) 1+Ω-2N  
c) 1+Ω2N  
d) 1/(1+Ω^2N)  
View Answer

Answer: d  
Explanation: We know that the magnitude response of a low pass Butterworth filter of order N is given as  
[digital-signal-processing-questions-answers-butterworth-filters-q2a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-butterworth-filters-q2a.png)  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-butterworth-filters-q7.png)

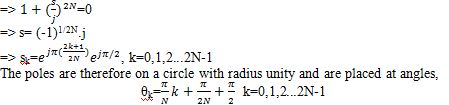
8. What is the transfer function of magnitude squared frequency response of the normalized low pass Butterworth filter?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-butterworth-filters-q8.png)  
View Answer

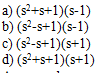
Answer: a  
Explanation: We know that the magnitude squared frequency response of a normalized low pass Butterworth filter is given as  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-butterworth-filters-q8a.png)

9. Where does the poles of the transfer function of normalized low pass Butterworth filter exists?  
a) Inside unit circle  
b) Outside unit circle  
c) On unit circle  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The transfer function of normalized low pass Butterworth filter is given as  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-butterworth-filters-q9.png)  
The poles are therefore on a circle with radius unity.

10. What is the general formula that represent the phase of the poles of transfer function of normalized low pass Butterworth filter of order N?  
a) π/N k+π/2N k=0,1,2…N-1  
b) π/N k+π/2N+π/2 k=0,1,2…2N-1  
c) π/N k+π/2N+π/2 k=0,1,2…N-1  
d) π/N k+π/2N k=0,1,2…2N-1  
View Answer

Answer: d  
Explanation: The transfer function of normalized low pass Butterworth filter is given as  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-butterworth-filters-q10.png)

11. What is the Butterworth polynomial of order 3?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-butterworth-filters-q11.png)  
View Answer

Answer: d  
Explanation: Given that the order of the Butterworth low pass filter is 3.  
Therefore, for N=3 Butterworth polynomial is given as B3(s)=(s-s0) (s-s1) (s-s2)  
[digital-signal-processing-questions-answers-butterworth-filters-q11a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-butterworth-filters-q11a.png)

12. What is the Butterworth polynomial of order 1?  
a) s-1  
b) s+1  
c) s  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: Given that the order of the Butterworth low pass filter is 1.  
Therefore, for N=1 Butterworth polynomial is given as B3(s)=(s-s0).  
We know that, sk=e(jπ((2k+1)/2N)) e(jπ/2)  
=>s0= -1  
=> B1(s)=s-(-1)=s+1.

13. What is the transfer function of Butterworth low pass filter of order 2?  
a) 1/(s2+√2 s+1)  
b) 1/(s2-√2 s+1)  
c) s2-√2 s+1  
d) s2+√2 s+1  
View Answer

Answer: a  
Explanation: We know that the Butterworth polynomial of a 2nd order low pass filter is  
B2(s)= s2+√2 s+1  
Thus the transfer function is given as 1/(s2+√2 s+1).

Questions & Answers (MCQs) focuses on “Frequency Transformations”.

1. What is the pass band edge frequency of an analog low pass normalized filter?  
a) 0 rad/sec  
b) 0.5 rad/sec  
c) 1 rad/sec  
d) 1.5 rad/sec  
View Answer

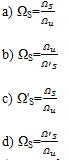
Answer: c  
Explanation: Let H(s) denote the transfer function of a low pass analog filter with a pass band edge frequency ΩP equal to 1 rad/sec. This filter is known as analog low pass normalized prototype.

2. If H(s) is the transfer function of a analog low pass normalized filter and Ωu is the desired pass band edge frequency of new low pass filter, then which of the following transformation has to be performed?  
a) s→ s / Ωu  
b) s→ s .Ωu  
c) s→ Ωu/s  
d) None of the mentioned  
View Answer

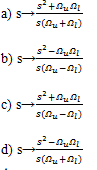
Answer: a  
Explanation: If Ωu is the desired pass band edge frequency of new low pass filter, then the transfer function of this new low pass filter is obtained by using the transformation s→ s / Ωu.

3. Which of the following is a low pass-to-high pass transformation?  
a) s→ s / Ωu  
b) s→ Ωu / s  
c) s→ Ωu.s  
d) none of the mentioned  
View Answer

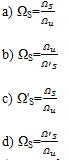
Answer: b  
Explanation: The low pass-to-high pass transformation is simply achieved by replacing s by 1/s. If the desired high pass filter has the pass band edge frequency Ωu, then the transformation is  
s→ Ωu / s.

4. Which of the following is the backward design equation for a low pass-to-low pass transformation?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-transformations-q4.png)  
View Answer

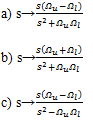
Answer: d  
Explanation: If Ωu is the desired pass band edge frequency of new low pass filter, then the transfer function of this new low pass filter is obtained by using the transformation s→ s / Ωu. If ΩS and Ω’S are the stop band frequencies of prototype and transformed filters respectively, then the backward design equation is given by  
[digital-signal-processing-questions-answers-frequency-transformations-q4a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-transformations-q4a.png).

5. Which of the following is a low pass-to-band pass transformation?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-transformations-q5.png)  
View Answer

Answer: c  
Explanation: If Ωu and Ωl are the upper and lower cutoff pass band frequencies of the desired band pass filter, then the transformation to be performed on the normalized low pass filter is  
[digital-signal-processing-questions-answers-frequency-transformations-q5a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-transformations-q5a.png)

6. Which of the following is the backward design equation for a low pass-to-high pass transformation?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-transformations-q4.png)  
View Answer

Answer: b  
Explanation: If Ωu is the desired pass band edge frequency of new high pass filter, then the transfer function of this new high pass filter is obtained by using the transformation s→ Ωu /s. If ΩS and Ω’S are the stop band frequencies of prototype and transformed filters respectively, then the backward design equation is given by  
[digital-signal-processing-questions-answers-frequency-transformations-q6](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-transformations-q6.png).

7. Which of the following is a low pass-to-band stop transformation?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-transformations-q7.png)d) None of the mentioned  
View Answer

Answer: c  
Explanation: If Ωu and Ωl are the upper and lower cutoff pass band frequencies of the desired band stop filter, then the transformation to be performed on the normalized low pass filter is  
[digital-signal-processing-questions-answers-frequency-transformations-q7a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-transformations-q7a.png)

8. If [digital-signal-processing-questions-answers-frequency-transformations-q8](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-transformations-q8.png)then which of the following is the backward design equation for a low pass-to-band pass transformation?  
a) ΩS= |B|  
b) ΩS= |A|  
c) ΩS= Max{|A|,|B|}  
d) ΩS= Min{|A|,|B|}  
View Answer

Answer: d  
Explanation: If Ωu and Ωl are the upper and lower cutoff pass band frequencies of the desired band pass filter and Ω1 and Ω2 are the lower and upper cutoff stop band frequencies of the desired band pass filter, then the backward design equation is  
ΩS= Min{|A|,|B|}  
where,[digital-signal-processing-questions-answers-frequency-transformations-q8](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-transformations-q8.png).

9. If [digital-signal-processing-questions-answers-frequency-transformations-q9](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-transformations-q9.png)then which of the following is the backward design equation for a low pass-to-band stop transformation?  
a) ΩS= Max{|A|,|B|}  
b) ΩS= Min{|A|,|B|}  
c) ΩS= |B|  
d) ΩS= |A|  
View Answer

Answer: b  
Explanation: If Ωu and Ωl are the upper and lower cutoff pass band frequencies of the desired band stop filter and Ω1 and Ω2 are the lower and upper cutoff stop band frequencies of the desired band stop filter, then the backward design equation is  
ΩS= Min{|A|,|B|}  
where, [digital-signal-processing-questions-answers-frequency-transformations-q9](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-frequency-transformations-q9.png).

10. Which of the following is a low pass-to-high pass transformation?  
a) s→ s / Ωu  
b) s→ Ωu / s  
c) s→ Ωu.s  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The low pass-to-high pass transformation is simply achieved by replacing s by 1/s. If the desired high pass filter has the pass band edge frequency Ωu, then the transformation is  
s→ Ωu / s.

Questions & Answers (MCQs) focuses on “Interpolation by a Factor I”.

1. Which of the following operation has to be performed to increase the sampling rate by an integer factor I?  
a) Interpolating I+1 new samples  
b) Interpolating I-1 new samples  
c) Extrapolating I+1 new samples  
d) Extrapolating I-1 new samples  
View Answer

Answer: b  
Explanation: An increase in the sampling rate by an integer factor of I can be accomplished by interpolating I-1 new samples between successive values of the signal.

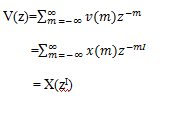
2. In one of the interpolation process, we can preserve the spectral shape of the signal sequence x(n).  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The interpolation process can be accomplished in a variety of ways. Among them there is a process that preserves the spectral shape of the signal sequence x(n).

3. If v(m) denote a sequence with a rate Fy=I.Fx which is obtained from x(n), then which of the following is the correct definition for v(m)?  
a) x(mI), m=0,±I,±2I….  
0, otherwise  
a) x(mI), m=0,±I,±2I….  
x(m/I), otherwise  
c) x(m/I), m=0,±I,±2I….  
0, otherwise  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: If v(m) denote a sequence with a rate Fy=I.Fx which is obtained from x(n) by adding I-1 zeros between successive values of x(n). Thus  
v(m)= x(m/I), m=0,±I,±2I….  
0, otherwise.

4. If X(z) is the z-transform of x(n), then what is the z-transform of interpolated signal v(m)?  
a) X(zI)  
b) X(z+I)  
c) X(z/I)  
d) X(zI)  
View Answer

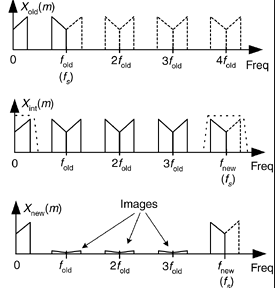
Answer: d  
Explanation: By taking the z-transform of the signal v(m), we get  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-interpolation-factor-i-q4.png)

5. If x(m) and v(m) are the original and interpolated signals and ωy denotes the frequency variable relative to the new sampling rate, then V(ωy)= X(ωyI).  
a) True  
b) False  
View Answer

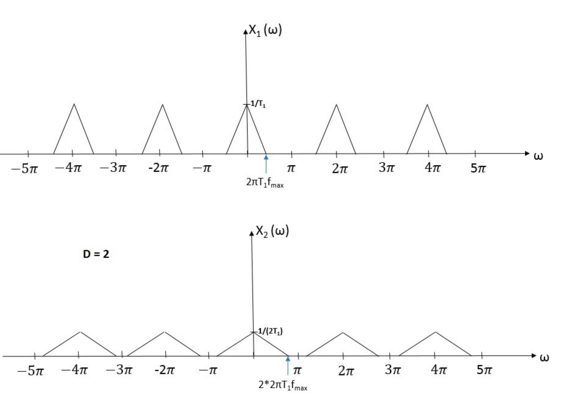
Answer: a  
Explanation: The spectrum of v(m) is obtained by evaluating V(z)= X(zI) on the unit circle. Thus V(ωy)= X(ωyI), where ωy denotes the frequency variable relative to the new sampling rate.

6. What is the relationship between ωx and ωy?  
a) ωy= ωx.I  
b) ωy= ωx/I  
c) ωy= ωx+I  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: We know that the relationship between sampling rates is Fy=IFx and hence the frequency variables ωx and ωy are related according to the formula  
ωy= ωx/I.

7. The following sampling rate conversion technique is interpolation by a factor I.  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-interpolation-factor-i-q7.png)  
a) True  
b) False  
View Answer

Answer: a  
Explanation: From the diagram, the values are interpolated between two successive values of x(n), thus it is called as sampling rate conversion using interpolation by a factor I.

8. The following sampling rate conversion technique is interpolation by a factor I.  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-interpolation-factor-i-q8.png)  
a) True  
b) False  
View Answer

Answer: b  
Explanation: The sampling rate conversion technique given in the diagram is decimation by a factor D.

9. Which of the following is true about the interpolated signal whose spectrum is V(ωy)?  
a) (I-1)-fold non-periodic  
b) (I-1)-fold periodic repetition  
c) I-fold non periodic  
d) I-fold periodic repetition  
View Answer

Answer: d  
Explanation: We observe that the sampling rate increase, obtained by the addition of I-1 zero samples between successive values of x(n), results in a signal whose spectrum is a I-fold periodic repetition of the input signal spectrum.

10. C=I is the desired normalization factor.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The amplitude of the sampling rate converted signal should be multiplied by a factor C, whose value when equal to I is called as desired normalization factor.

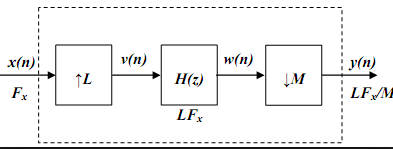
Questions & Answers focuses on “Sampling Rate Conversion by a Rational Factor I/D”.

1. Sampling rate conversion by the rational factor I/D is accomplished by what connection of interpolator and decimator?  
a) Parallel  
b) Cascade  
c) Convolution  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: A sampling rate conversion by the rational factor I/D is accomplished by cascading an interpolator with a decimator.

2. Which of the following has to be performed in sampling rate conversion by rational factor?  
a) Interpolation  
b) Decimation  
c) Either interpolation or decimation  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: We emphasize that the importance of performing the interpolation first and decimation second, is to preserve the desired spectral characteristics of x(n).

3. Which of the following operation is performed by the blocks given the figure below?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/tough-digital-signal-processing-questions-answers-q3.png)  
a) Sampling rate conversion by a factor I  
b) Sampling rate conversion by a factor D  
c) Sampling rate conversion by a factor D/I  
d) Sampling rate conversion by a factor I/D  
View Answer

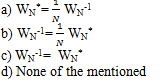
Answer: d  
Explanation: In the diagram given, a interpolator is in cascade with a decimator which together performs the action of sampling rate conversion by a factor I/D.

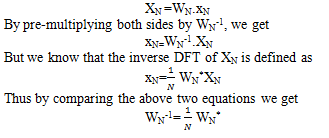
4. The Nth root of unity WN is given as:  
a) ej2πN  
b) e-j2πN  
c) e-j2π/N  
d) ej2π/N  
View Answer

Answer: c  
Explanation: We know that the Discrete Fourier transform of a signal x(n) is given as  
[tough-digital-signal-processing-questions-answers-q4](https://www.sanfoundry.com/wp-content/uploads/2015/10/tough-digital-signal-processing-questions-answers-q4.png)  
Thus we get Nth rot of unity WN= e-j2π/N

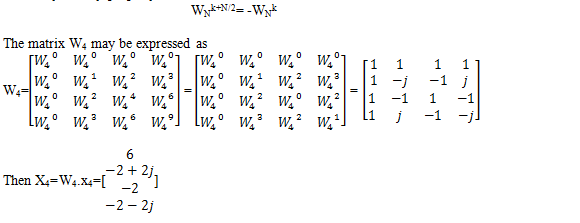
5. Which of the following is true regarding the number of computations requires to compute an N-point DFT?  
a) N2 complex multiplications and N(N-1) complex additions  
b) N2 complex additions and N(N-1) complex multiplications  
c) N2 complex multiplications and N(N+1) complex additions  
d) N2 complex additions and N(N+1) complex multiplications  
View Answer

Answer: a  
Explanation: The formula for calculating N point DFT is given as  
[tough-digital-signal-processing-questions-answers-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/tough-digital-signal-processing-questions-answers-q5.png)  
From the formula given at every step of computing we are performing N complex multiplications and N-1 complex additions. So, in a total to perform N-point DFT we perform N2 complex multiplications and N(N-1) complex additions.

6. Which of the following is true?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/tough-digital-signal-processing-questions-answers-q6.png)  
View Answer

Answer: b  
Explanation: If XN represents the N point DFT of the sequence xN in the matrix form, then we know that[](https://www.sanfoundry.com/wp-content/uploads/2015/10/tough-digital-signal-processing-questions-answers-q6a.png)

7. What is the DFT of the four point sequence x(n)={0,1,2,3}?  
a) {6,-2+2j-2,-2-2j}  
b) {6,-2-2j,2,-2+2j}  
c) {6,-2+2j,-2,-2-2j}  
d) {6,-2-2j,-2,-2+2j}  
View Answer

Answer: c  
Explanation: The first step is to determine the matrix W4. By exploiting the periodicity property of W4 and the symmetry property  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/tough-digital-signal-processing-questions-answers-q7.png)

8. If X(k) is the N point DFT of a sequence whose Fourier series coefficients is given by ck, then which of the following is true?  
a) X(k)=Nck  
b) X(k)=ck/N  
c) X(k)=N/ck  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: The Fourier series coefficients are given by the expression  
[tough-digital-signal-processing-questions-answers-q8](https://www.sanfoundry.com/wp-content/uploads/2015/10/tough-digital-signal-processing-questions-answers-q8.png)

9. What is the DFT of the four point sequence x(n)={0,1,2,3}?  
a) {6,-2+2j-2,-2-2j}  
b) {6,-2-2j,2,-2+2j}  
c) {6,-2-2j,-2,-2+2j}  
d) {6,-2+2j,-2,-2-2j}  
View Answer

Answer: d  
Answer: Given x(n)={0,1,2,3}  
We know that the 4-point DFT of the above given sequence is given by the expression  
[tough-digital-signal-processing-questions-answers-q5](https://www.sanfoundry.com/wp-content/uploads/2015/10/tough-digital-signal-processing-questions-answers-q5.png)  
In this case N=4  
=>X(0)=6,X(1)=-2+2j,X(2)=-2,X(3)=-2-2j.

10. If W4100=Wx200, then what is the value of x?  
a) 2  
b) 4  
c) 8  
d) 16  
View Answer

Answer: c  
Explanation: We know that according to the periodicity and symmetry property,  
100/4=200/x=>x=8.

#### 9. Questions & Answers on Multirate Digital Signal Procesing

The section contains questions and answers on factor decimation and multirate digital signal processing.

|  |  |
| --- | --- |
| [Multirate Signal Processing](https://www.sanfoundry.com/digital-signal-processing-questions-answers-multirate-digital-signal-processing/) | [Factor D Decimation](https://www.sanfoundry.com/digital-signal-processing-questions-answers-decimation-factor-d/) |

Questions & Answers (MCQs) focuses on “Multirate Digital Signal Processing”.

1. There is no requirement to process the various signals at different rates commensurate with the corresponding bandwidths of the signals.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: In telecommunication systems that transmit and receive different types of signals, there is a requirement to process the various signals at different rates commensurate with the corresponding bandwidths of the signals.

2. What is the process of converting a signal from a given rate to a different rate?  
a) Sampling  
b) Normalizing  
c) Sampling rate conversion  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The process of converting a signal from a given rate to a different rate is known as sampling rate conversion.

3. The systems that employ multiple sampling rates are called multi-rate DSP systems.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Systems that employ multiple sampling rates in the processing of digital signals are called multi rate digital signal processing systems.

4. Which of the following methods are used in sampling rate conversion of a digital signal?  
a) D/A convertor and A/D convertor  
b) Performing entirely in digital domain  
c) None of the mentioned  
d) Both of the mentioned  
View Answer

Answer: d  
Explanation: Sampling rate conversion of a digital signal can be accomplished in one of the two general methods. One method is to pass the signal through D/A converter, filter it if necessary, and then to resample the resulting analog signal at the desired rate. The second method is to perform the sampling rate conversion entirely in the digital domain.

5. Which of the following is the advantage of sampling rate conversion by converting the signal into analog signal?  
a) Less signal distortion  
b) Quantization effects  
c) New sampling rate can be arbitrarily selected  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: One apparent advantage of the given method is that the new sampling rate can be arbitrarily selected and need not have any special relationship with the old sampling rate.

6. Which of the following is the disadvantage of sampling rate conversion by converting the signal into analog signal?  
a) Signal distortion  
b) Quantization effects  
c) New sampling rate can be arbitrarily selected  
d) Signal distortion & Quantization effects  
View Answer

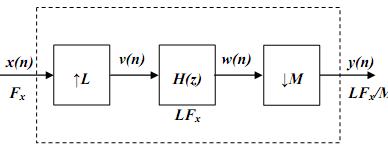
Answer: d  
Explanation: The major disadvantage by the given type of conversion is the signal distortion introduced by the D/A converter in the signal reconstruction and by the quantization effects in the A/D conversion.

7. In which of the following, sampling rate conversion are used?  
a) Narrow band filters  
b) Digital filter banks  
c) Quadrature mirror filters  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: There are several applications of sampling rate conversion in multi rate digital signal processing systems, which include the implementation of narrow band filters, quadrature mirror filters and digital filter banks.

8. Which of the following use quadrature mirror filters?  
a) Sub band coding  
b) Trans-multiplexer  
c) Both of the mentioned  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: There are many applications where quadrature mirror filters can be used. Some of these applications are sub-band coding, trans-multiplexers and many other applications.

9. The sampling rate conversion can be as shown in the figure below.  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-multirate-digital-signal-processing-q9.png)  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The process of sampling rate conversion in the digital domain can be viewed as a linear filtering operation as illustrated in the given figure.

10. If Fx and Fy are the sampling rates of the input and output signals respectively, then what is the value of Fy/Fx?  
a) D/I  
b) I/D  
c) I.D  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The input signal x(n) is characterized by the sampling rate Fx and he output signal y(m) is characterized by the sampling rate Fy, then  
Fy/Fx= I/D  
where I and D are relatively prime integers.

11. What is the process of reducing the sampling rate by a factor D?  
a) Sampling rate conversion  
b) Interpolation  
c) Decimation  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The process of reducing the sampling rate by a factor D, i.e., down-sampling by D is called as decimation.

12. What is the process of increasing the sampling rate by a factor I?  
a) Sampling rate conversion  
b) Interpolation  
c) Decimation  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The process of increasing the sampling rate by a integer factor I, i.e., up-sampling by I is called as interpolation.

Questions & Answers (MCQs) focuses on “Decimation by a Factor D”.

1. If we down-sample a signal x(n), then the resulting signal will be an aliased version of x(n).  
a) True  
b) False  
View Answer

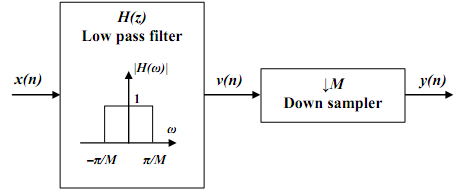
Answer: a  
Explanation: We know that if we reduce the sampling rate simply by selecting every Dth value of x(n), the signal will be an aliased version of x(n).

2. What is the folding frequency for the aliased version of x(n) with sampling rate F?  
a) F/D  
b) F/4D  
c) F/2  
d) F/2D  
View Answer

Answer: d  
Explanation: We know that if we reduce the sampling rate simply by selecting every Dth value of x(n), the signal will be an aliased version of x(n), with a folding frequency of F/2D.

3. To what value should the bandwidth of x(n) has to be reduced in order to avoid aliasing?  
a) F/D  
b) F/2D  
c) F/4D  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: To avoid aliasing, we must reduce the bandwidth of x(n) to Fmax=F/2D. Then we may down-sample by D and thus avoid sampling.

4. Which process has a block diagram as shown in the figure below?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-decimation-factor-d-q4.png)  
a) Sampling rate conversion  
b) Interpolation  
c) Decimation  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The block diagram shown in the figure is of sampling rate conversion by decimation by a factor D technique.

5. Which of the following is true about the filtering operation on x(n)?  
a) Linear  
b) Time variant  
c) None of the mentioned  
d) Linear and time invariant  
View Answer

Answer: d  
Explanation: Although the filtering operation on x(n) is linear and time invariant, the down-sampling operation in combination with the filtering results in a time variant system.

6.If x(n) produces y(m), then x(n-n0)) does imply y(n-n0) for any value of n0).  
a) True  
b) False  
View Answer

Answer: b  
Explanation: Given the fact that x(n) produces y(m), we note that x(n-n0)) does not imply y(n-n0)) unless n0) is a multiple of D.

7. The linear filtering operation followed by down sampling on x(n) is not time invariant.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: Given the fact that x(n) produces y(m), we note that x(n-n0)) does not imply y(n-n0)) unless n0) is a multiple of D. Consequently, the overall linear operation, that is linear filtering followed by down sampling on x(n) is not time invariant.

8. Which of the following gives the equation for y(m)?  
a) v(mD)  
b) p(mD)  
c) v(mD).p(mD)  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: We know that the equation for y(m) is given as  
y(m)= v ̅(mD)= v(mD).p(mD).

9. We need not relate the spectrum of y(m) to spectrum of x(n) to obtain frequency response characteristic of y(m).  
a) True  
b) False  
View Answer

Answer: b  
Explanation: The frequency domain characteristic of the output sequence y(m) can be obtained by relating the spectrum of y(m) to the spectrum of the input sequence x(n).

10. The sequence v ̅(n) can be obtained by multiplying v(n) with a signal p(n) of period D.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: v ̅(n) can be viewed as a sequence obtained by multiplying v(n) with a periodic train of impulses p(n) with period D.

11. If HD(ω) is the frequency response of the low pass filter, then for what value of ω, HD(ω)=1?  
a) |ω| = π/D  
b) |ω| ≥ π/D  
c) |ω| ≤ π/D  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The input sequence x(n) is passed through a low pass filter, characterized by the impulse response h(n) and a frequency response HD(ω), which ideally satisfies the condition,  
HD(ω)=1, |ω| ≤ π/D  
=0, otherwise.

12. The filter eliminates the spectrum of X(ω) in the range π/D < ω < π.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The input sequence x(n) is passed through a low pass filter, characterized by the impulse response h(n) and a frequency response HD(ω), which ideally satisfies the condition,  
HD(ω)=1, |ω| ≤ π/D  
=0, otherwise  
Thus the filter eliminates the spectrum of X(ω) in the range π/D < ω < π

#### 10. Questions on Sampling and Reconstruction of Signals

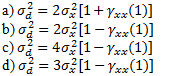
This section contains questions on A/D Converters and their oversampling, band pass signal sampling and representation, sample and hold concepts and quantization and coding techniques.

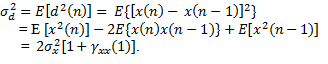
|  |  |
| --- | --- |
| [A/D Converter Oversampling](https://www.sanfoundry.com/digital-signal-processing-questions-answers-oversampling-ad-converters/) [Sample and Hold](https://www.sanfoundry.com/digital-signal-processing-questions-answers-sample-hold/) [Band Pass Signal Sampling](https://www.sanfoundry.com/digital-signal-processing-questions-answers-sampling-band-pass-signals/) | [Bandpass Signal Representation](https://www.sanfoundry.com/digital-signal-processing-questions-answers-representation-bandpass-signals/) [Quantization and Coding](https://www.sanfoundry.com/digital-signal-processing-questions-answers-quantization-coding/) [Digital to Analog Conversion](https://www.sanfoundry.com/digital-signal-processing-questions-answers-digital-analog-conversion-sample-hold/) |

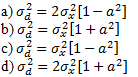
Questions & Answers (MCQs) focuses on “Oversampling A/D Converters”.

1. For a given number of bits, the power of quantization noise is proportional to the variance of the signal to be quantized.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The dynamic range of the signal, which is proportional to its standard deviation σx , should match the range R of the quantizer, it follows that ∆ is proportional to σx. Hence for a given number o f bits, the power o f the quantization noise is proportional to the variance of the signal to be quantized.

2. What is the variance of the difference between two successive signal samples, d(n) = x(n) – x(n-1) ?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-oversampling-ad-converters-q2.png)  
View Answer

Answer: b  
Explanation: [](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-oversampling-ad-converters-q2a.png)

3.What is the variance of the difference between two successive signal samples, d(n) = x(n) –ax(n-1) ?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-oversampling-ad-converters-q3.png)  
View Answer

Answer: c  
Explanation: An even better approach is to quantize the difference, d(n) = x(n) –ax(n-1), w here a is a parameter selected to minimize the variance in d(n). Therefore [digital-signal-processing-questions-answers-oversampling-ad-converters-q3a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-oversampling-ad-converters-q3a.png)

4. If the difference d(n) = x(n) –ax(n-1), then what is the optimum choice for a = ?  
[digital-signal-processing-questions-answers-oversampling-ad-converters-q4](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-oversampling-ad-converters-q4.png)  
View Answer

Answer: a  
Explanation: An even better approach is to quantize the difference, d(n) = x(n) –ax(n-1), w here a is a parameter selected to minimize the variance in d(n). This leads to the result that the optimum choice of a is [digital-signal-processing-questions-answers-oversampling-ad-converters-q4a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-oversampling-ad-converters-q4a.png)

5. What is the quantity ax(n-1) is called?  
a) Second-order predictor of x(n)  
b) Zero-order predictor of x(n)  
c) First-order predictor of x(n)  
d) Third-order predictor of x(n)  
View Answer

Answer: c  
Explanation: In the equation d(n) = x(n) –ax(n-1), the quantity ax(n-1) is called a First-order predictor of x(n).

6. The differential predictive signal quantizer system is known as?  
a) DCPM  
b) DMPC  
c) DPCM  
d) None of the mentioned  
View Answer

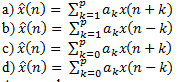
Answer: c  
Explanation: A differential predictive signal quantizer system. This system is used in speech encoding and transmission over telephone channels and is known as differential pulse code modulation (DPCM ).

7. What is the expansion of DPCM?  
a) Differential Pulse Code Modulation  
b) Differential Plus Code Modulation  
c) Different Pulse Code Modulation  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: A differential predictive signal quantizer system. This system is used in speech encoding and transmission over telephone channels and is known as differential pulse code modulation (DPCM ).

8. What are the main uses of DPCM?  
a) Speech Decoding and Transmission over mobiles  
b) Speech Encoding and Transmission over mobiles  
c) Speech Decoding and Transmission over telephone channels  
d) Speech Encoding and Transmission over telephone channels  
View Answer

Answer: d  
Explanation: A differential predictive signal quantizer system. This system is used in speech encoding and transmission over telephone channels and is known as differential pulse code modulation (DPCM ).

9. To reduce the dynamic range of the difference signal d(n) = x(n) – x ̂(n), thus a predictor of order p has the form?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-oversampling-ad-converters-q9.png)  
View Answer

Answer: b  
Explanation: T he goal of the predictor is to provide an estimate x ̂(n) of x(n) from a linear combination of past values of x(n), so as to reduce the dynamic range of the difference signal d(n) = x(n) – x ̂(n). Thus a predictor of order p has the form [digital-signal-processing-questions-answers-oversampling-ad-converters-q9a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-oversampling-ad-converters-q9a.png)

10. The simplest form of differential predictive quantization is called?  
a) AM  
b) BM  
c) DM  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The simplest form of differential predictive quantization is called delta modulation (DM ).

11. What is the abbreviation of DM?  
a) Diameter Modulation  
b) Distance Modulation  
c) Delta Modulation  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The simplest form of differential predictive quantization is called delta modulation (DM ).

12. In DM, the quantizer is a simple \_\_\_\_\_\_\_\_ bit and \_\_\_\_\_\_ level quantizer?  
a) 2-bit, one-level  
b) 1-bit, two-level  
c) 2-bit, two level  
d) 1-bit, one level  
View Answer

Answer: b  
Explanation: T he simplest form o f differential predictive quantization is called delta modulation (DM ). In DM, the quantizer is a simple 1-bit (two -level) quantizer.

13. In DM, What is the order of predictor is used?  
a) Zero-order predictor  
b) Second-order predictor  
c) First-order predictor  
d) Third-order predictor  
View Answer

Answer: c  
Explanation: In DM, the quantizer is a simple 1-bit (two -level) quantizer and the predictor is a first-order predictor.

14. In the equation xq(n)=axq(n-1)+ dq(n), if a = 1 then integrator is called?  
a) Leaky integrator  
b) Ideal integrator  
c) Ideal accumulator  
d) Both Ideal integrator & accumulator  
View Answer

Answer: d  
Explanation: In the equation xq(n)=axq(n-1)+ dq(n), if a = 1, we have an ideal accumulator (integrator).

15. In the equation xq(n)=axq(n-1)+ dq(n), if a < 1 then integrator is called?  
a) Leaky integrator  
b) Ideal integrator  
c) Ideal accumulator  
d) Both Ideal integrator & accumulator  
View Answer

Answer: a  
Explanation: In the equation xq(n)=axq(n-1)+ dq(n), a < 1 results in a ”leaky integrator”.

Questions & Answers (MCQs) focuses on “Sample and Hold”.

1. What is the main function of (A/D) or ADC converter?  
a) Converts Digital to Analog Signal  
b) Converts Analog to Digital signal  
c) All of the mentioned  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The electronic device that performs this conversion from an analog signal to a digital sequence is called an analog-to-digital (A /D ) converter (ADC ).

2. What is the main function of (D/A) or DAC converter?  
a) Converts Digital to Analog Signal  
b) Converts Analog to Digital signal  
c) All of the mentioned  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: A digital-to-analog ( D /A ) converter (DAC ) takes a digital sequence and produces at its output a voltage or current proportional to the size o f the digital word applied to its input.

3. T he S/H is a digitally controlled analog circuit that tracks the analog input signal during the sample mode, and then holds it fixed during the hold mode to the instantaneous value of the signal at the time the system is switched from the sample to the hold mode.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The sampling of an analog signal is performed by a sample-and-hold (S/H ) circuit. The sampled signal is then quantized and converted to digital form. Usually, the S/H is integrated into the (A/D) converter. T he S/H is a digitally controlled analog circuit that tracks the analog input signal during the sample mode, and then holds it fixed during the hold mode to the instantaneous value o f the signal at the time the system is switched from the sample mode to the hold mode.

4. The time required to complete the conversion of Analog to Digital is \_\_\_\_\_\_\_\_ the duration of the hold mode of S/H.  
a) Greater than  
b) Equals to  
c) Less than  
d) Greater than or Equals to  
View Answer

Answer: c  
Explanation: The A /D converter begins the conversion after it receives a convert command. The time required to complete the conversion should be less than the duration of the hold mode of S/H.

5. In A/D converter, what is the time relation between sampling period T and the duration of the sample mode and the hold mode?  
a) Should be larger than the duration of sample mode and hold mode  
b) Should be smaller than the duration of sample mode and hold mode  
c) Should be equal to the duration of sample mode and hold mode  
d) Should be larger than or equals to the duration of sample mode and hold mode  
View Answer

Answer: a  
Explanation: The A /D converter begins the conversion after it receives a convert command. The sampling period T should be larger than the duration of the sample mode and the hold mode.

6. In the practical A/D converters, what are the distortions and time- related degradations occur during the conversion process?  
a) Jitter errors  
b) Droops  
c) Nonlinear variations in the duration of the sampling aperture  
d) All of the mentioned  
View Answer

Answer: d  
Explanation: An ideal S/H introduces no distortion in the conversion process and is accurately modeled as an ideal sampler. However, time-related degradations such as errors in the periodicity of the sampling process ( “jitter”), nonlinear variations in the duration o f the sampling aperture, and changes in the voltage held during conversion ( “droop”) do occur in practical devices.

7. In the absence of an S/H, the input signal must change by more than one-half of the quantization step during the conversion, which may be an impractical constraint.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: The use of an S/H allows the A /D converter to operate more slowly compared to the time actually used to acquire the sample. In the absence of an S/H, the input signal must not change by more than one-half of the quantization step during the conversion, which may be an impractical constraint.

8. The noise power σn2 can be reduced by increasing the sampling rate to spread the quantization noise power over a larger frequency band (-Fs/2,Fs/2).  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The noise power σn2 can be reduced by increasing the sampling rate to spread the quantization noise power over a larger frequency band (-Fs/2,Fs/2), and then shaping the noise power spectral density by means o f an appropriate filter.

9. What is the process of down sampling called?  
a) Decimation  
b) Fornication  
c) Both Decimation & Fornication  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: To avoid aliasing, w e first filter out the out-of-band (fl, F2) noise by processing the wideband signal. The signal is then passed through the low pass filter and re-sampled (down sampled) at the lower rate. The down sampling process is called decimation.

10. If the interpolation factor is I = 256, the A /D converter output can be obtained by averaging successive non-overlapping blocks of 128 bits.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: If the interpolation factor is I = 256, the A /D converter output can be obtained by averaging successive non-overlapping blocks o f 128 bits. This averaging would result in a digital signal with a range of values from zero to 256(b as 8 bits) at the Nyquist rate. The averaging process also provides the required anti-aliasing filtering.

11. The crosshatched areas gives two types of Quantization error in DM ,They are ?  
a) Slope-overload distortion  
b) Granular noise  
c) Slope-overload distortion & Granular noise  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: The crosshatched areas illustrate two types of quantization error in DM , slope-overload distortion and granular noise.

12. The slope-overload distortion is avoided, if which of the following conditions satisfy?  
a) Min| dx(t)/d(t) | ≤ ∆/T  
b) Max| dx(t)/d(t) | ≤ ∆/T  
c) |dx(t)/d(t) | ≤∆/T  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The crosshatched areas illustrate two types of quantization error in DM , slope-overload distortion and granular noise. types of quantization error in DM , slope-overload distortion and granular noise. Since the maximum slope A (T in x ( n ) is limited by the step size, slope-overload distortion can be avoided if max| dx(t)/d(t) | ≤∆/T .

13. In DM, By increasing∆, reduces the overload distortion but increases the granular noise, and vice versa  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The granular noise occurs w hen the DM tracks a relatively flat (slowly changing) input signal. We note that increasing ∆ reduces overload distortion but increases the granular noise, and vice versa.

14. Which of the following is the right way to reduce distortion in the DM?  
a) By setting up an integrator in front of DM  
b) By setting up an integrator behind the DM  
c) By setting up an integrator in the middle of DM  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: We note that increasing ∆ reduces overload distortion but increases the granular noise, and vice versa. One way to reduce these two types of distortion is to use an integrator in front of the DM.

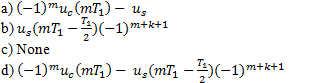
15. What are the effects produced by Dm by setting up an integrator at the front of DM?  
a) Simplifies the DM decoder  
b) Increases correlation of the signal into the DM input  
c) Emphasizes the low frequencies of x(t)  
d) All of the mentioned  
View Answer

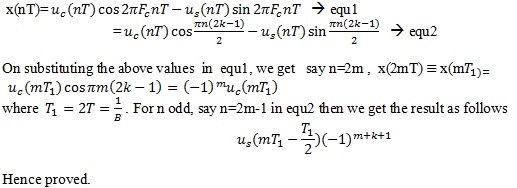
Answer: d  
Explanation: One way to reduce these two types of distortion is to use an integrator in front of the DM. This has two effects. First, it emphasizes the low frequencies of x (t) and increases the correlation of the signal into the DM input. Second, it simplifies the DM decoder because the differentiator (inverse system) required at the decoder is canceled by the DM integrator.

Questions & Answers (MCQs) focuses on “Sampling of Band Pass Signals”.

1. The frequency shift can be achieved by multiplying the band pass signal as given in equation  
[digital-signal-processing-questions-answers-sampling-band-pass-signals-q1](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q1.png)by the quadrature carriers cos[2πFct] and sin[2πFct] and lowpass filtering the products to eliminate the signal components of 2Fc.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: It is certainly be advantageous to perform a frequency shift of the band pass signal by and sampling the equivalent low pass signal. Such a frequency shift can be achieved by multiplying the band pass signal as given in the above equation by the quadrature carriers cos[2πFct] and sin[2πFct] and low pass filtering the products to eliminate the signal components at 2Fc. Clearly, the multiplication and the subsequent filtering are first performed in the analog domain and then the outputs o f the filters are sampled.

2. What is the final result obtained by substituting Fc=kB-B/2 , T= 1/2B and say n = 2m i.e., for even and n=2m-1 for odd in equation  
[digital-signal-processing-questions-answers-sampling-band-pass-signals-q2](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q2.png)  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q2a.png)  
View Answer

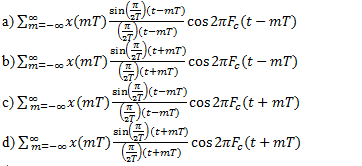
Answer: d  
Explanation:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q2b.png)

3. Which low pass signal component occurs at the rate of B samples per second with even numbered samples of x(t)?  
a) uc– lowpass signal component  
b) us– lowpass signal component  
c) uc & us – lowpass signal component  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: With the even-numbered samples o f x(t), which occur at the rate o f B samples per second, produce samples of the low pass signal component uc.

4. Which low pass signal component occurs at the rate of B samples per second with odd numbered samples of x(t)?  
a) uc– lowpass signal component  
b) us– lowpass signal component  
c) uc & us – lowpass signal component  
d) None of the mentioned  
View Answer

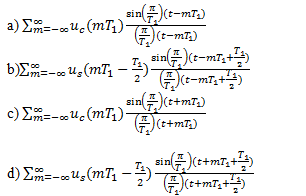
Answer: b  
Explanation: : With the odd-numbered samples o f x(t), which occur at the rate o f B samples per second, produce samples of the low pass signal component us.

5. What is the reconstruction formula for the bandpass signal x(t) with samples taken at the rate of 2B samples per second?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q5.png)  
View Answer

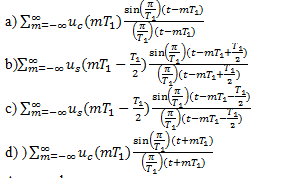
Answer: a  
Explanation:[digital-signal-processing-questions-answers-sampling-band-pass-signals-q5a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q5a.png) , where T=1/2B

6. What is the new centre frequency for the increased bandwidth signal ?  
a) Fc‘= Fc+B/2+B’/2  
b) Fc‘= Fc+B/2-B’/2  
c) Fc‘= Fc-B/2-B’/2  
d) None of the mentioned  
View Answer

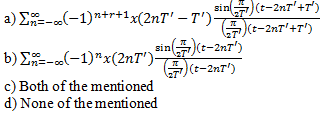
Answer: b  
Explanation: A new centre frequency for the increased bandwidth signal is Fc‘= Fc+B/2-B’/2

7. According to the sampling theorem for low pass signals with T1=1/B, then what is the expression for uc(t) = ?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q7.png)  
View Answer

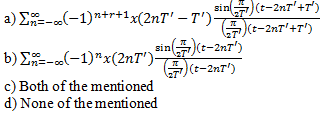
Answer: a  
Explanation: To reconstruct the equivalent low pass signals. Thus, according to the sampling  
theorem for low pass signals with T1= 1 / B .  
[digital-signal-processing-questions-answers-sampling-band-pass-signals-q7a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q7a.png)

8. According to the sampling theorem for low pass signals with T1=1/B, then what is the expression for us(t) = ?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q8.png)  
View Answer

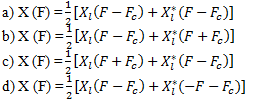
Answer: b  
Explanation: To reconstruct the equivalent low pass signals. Thus, according to the sampling  
theorem for low pass signals with T1= 1 / B .  
[digital-signal-processing-questions-answers-sampling-band-pass-signals-q8a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q8a.png)

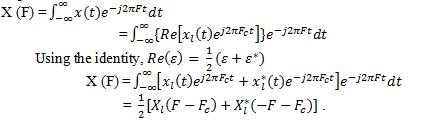
9. What is the expression for low pass signal component uc(t) that can be expressed in terms of samples of the bandpass signal ?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q9.png)  
View Answer

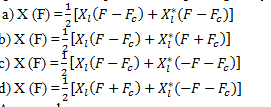
Answer: b  
Explanation: The low pass signal components uc(t) can be expressed in terms of samples of the  
band pass signal as follows:  
[digital-signal-processing-questions-answers-sampling-band-pass-signals-q9a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q9a.png)

10. What is the expression for low pass signal component us(t) that can be expressed in terms of samples of the bandpass signal ?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q9.png)  
View Answer

Answer: a  
Explanation: The low pass signal components us(t) can be expressed in terms of samples of the  
band pass signal as follows:  
[digital-signal-processing-questions-answers-sampling-band-pass-signals-q10a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q10a.png)

11. What is the Fourier transform of x(t) ?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q11.png)  
View Answer

Answer: d  
Explanation:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q11a.png)

12. What is the basic relationship between the spectrum o f the real band pass signal x( t ) and the spectrum of the equivalent low pass signal xl(t) ?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q12.png)  
View Answer

Answer: d  
Explanation:[digital-signal-processing-questions-answers-sampling-band-pass-signals-q12a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-sampling-band-pass-signals-q12a.png) , where Xl(F) is the Fourier transform of xl(t). This is the basic relationship between the spectrum o f the real band pass signal x ( t ) and the spectrum of the equivalent low pass signal xl(t).

Questions & Answers (MCQs) focuses on “The Representation of Bandpass Signals”.

1. Which of the following is the right way of representation of equation that contains only the positive frequencies in a given x(t) signal?  
a) X+(F)=4V(F)X(F)  
b) X+(F)=V(F)X(F)  
c) X+(F)=2V(F)X(F)  
d) X+(F)=8V(F)X(F)  
View Answer

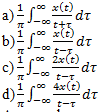
Answer: c  
Explanation: In a real valued signal x(t), has a frequency content concentrated in a narrow band of frequencies in the vicinity of a frequency Fc. Such a signal which has only positive frequencies can be expressed as X+(F)=2V(F)X(F)  
Where X+(F) is a Fourier transform of x(t) and V(F) is unit step function.

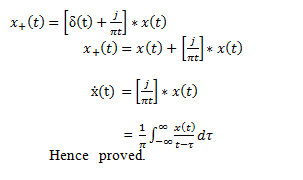
2. What is the equivalent time –domain expression of X+(F)=2V(F)X(F)?  
a) F(+1)2V(F)\*F(+1)X(F)  
b) F(-1)4V(F)\*F(-1)X(F)  
c) F(-1)V(F)\*F(-1)X(F)  
d) F(-1)2V(F)\*F(-1)X(F)  
View Answer

Answer: d  
Explanation: Given Expression , X+(F)=2V(F)X(F).It can be calculated as follows  
[digital-signal-processing-questions-answers-representation-bandpass-signals-q2](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q2.png)

3. In time-domain expression, [digital-signal-processing-questions-answers-representation-bandpass-signals-q3](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q3.png)The signal x+(t) is known as  
a) Systematic signal  
b) Analytic signal  
c) Pre-envelope of x(t)  
d) Both Analytic signal & Pre-envelope of x(t)  
View Answer

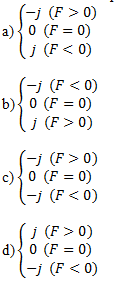
Answer: d  
Explanation: From the given expression, [digital-signal-processing-questions-answers-representation-bandpass-signals-q3](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q3.png)

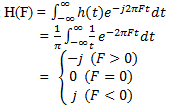
4. In equation [digital-signal-processing-questions-answers-representation-bandpass-signals-q4](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q4.png)  
Then the value of ẋ(t) is ?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q4a.png)  
View Answer

Answer: b  
Explanation:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q4b.png)

5. If the signal ẋ(t) can be viewed as the output of the filter with impulse response h(t) = 1/πt , -∞ < t < ∞  
when excited by the input signal x(t) then such a filter is called as\_\_\_  
a) Analytic transformer  
b) Hilbert transformer  
c) Both Analytic & Hilbert transformer  
d) None of the mentioned  
View Answer

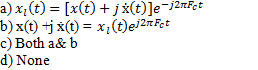
Answer: B  
Explanation: The signal ẋ(t) can be viewed as the output of the filter with impulse response h(t) = 1/πt ,  
-∞ < t < ∞ when excited by the input signal x(t) then such a filter is called as Hilbert transformer.

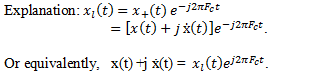
6. What is the frequency response of a Hilbert transform H(F)=?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q6.png)  
View Answer

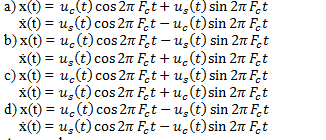
Answer: a  
Explanation: [](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q6a.png)  
We Observe that │H (F) │=1 and the phase response ʘ(F) = -1/2π for F > 0 and ʘ(F) = 1/2π for F < 0.

7. What is the equivalent lowpass representation obtained by performing a frequency translation of X+(F) to Xl(F)= ?  
a) X+(F+Fc)  
b) X+(F-Fc)  
c) X+(F\*Fc)  
d) X+(Fc-F)  
View Answer

Answer: a  
Explanation: The analytic signal x+(t) is a bandpass signal. We obtain an equivalent lowpass representation by performing a frequency translation of X+(F).

8. What is the equivalent time domain relation of xl(t) i.e., lowpass signal?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q8.png)  
View Answer

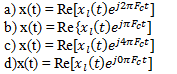
Answer: c  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q8a.png)

9. If we substitute the equation [digital-signal-processing-questions-answers-representation-bandpass-signals-q9](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q9.png)in equation x (t) +j ẋ (t) = xl(t) e(j2πFct) and equate real and imaginary parts on side, then what are the relations that we obtain?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q9a.png)  
View Answer

Answer: b  
Explanation: If we substitute the given equation in other, then we get the required result

10. In the relation, [digital-signal-processing-questions-answers-representation-bandpass-signals-q10](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q10.png)the low frequency components uc and us are called \_\_ of the bandpass signal x(t).  
a) Quadratic components  
b) Quadrature components  
c) Triplet components  
d) None of the mentioned  
View Answer

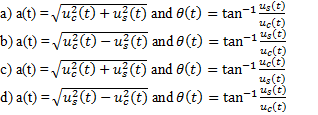
Answer: b  
Explanation: The low -frequency signal components uc(t) and us(t) can be viewed as amplitude modulations impressed on the carrier components cos2πFct and sin2πFct , respectively. Since these carrier components are in phase quadrature, uc(t) and us(t) are called the Quadrature components of the bandpass signal x (t).

11. What is the other way of representation of bandpass signal x(t)?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q11.png)  
View Answer

Answer: a  
Explanation: The above signal is formed from quadrature components, [digital-signal-processing-questions-answers-representation-bandpass-signals-q11a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q11a.png)where Re denotes the real part of complex valued quantity.

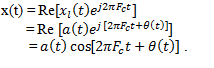
12. In the equation [digital-signal-processing-questions-answers-representation-bandpass-signals-q12](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q12.png)What is the lowpass signal xl (t) is usually called the \_\_\_ of the real signal x(t) ?  
a) Mediature envelope  
b) Complex envelope  
c) Equivalent envelope  
D) All of the mentioned  
View Answer

Answer: b  
Explanation: In the equation x(t) = Re[xl(t)e(j2πFct)],Re denotes the real part of the complex valued quantity in the brackets following. The lowpass signal x\_l (t) is usually called the Complex envelope of the real signal x(t) , and is basically the equivalent low pass signal.

13. If a possible representation of a band pass signal is obtained by expressing xl (t) as [digital-signal-processing-questions-answers-representation-bandpass-signals-q13](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q13.png)then what are the equations of a(t) and θ(t) ?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q13a.png)  
View Answer

Answer: a  
Explanation: A third possible representation of a band pass signal is obtained by expressing [digital-signal-processing-questions-answers-representation-bandpass-signals-q13b](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q13b.png)

14. What is the possible representation of x(t) if xl(t)=a(t)e(jθ(t)) ?  
a) x(t) = a(t) cos(2pFct – ?(t))  
b) x(t) = a(t) cos(2pFct + ?(t))  
c) x(t) = a(t) sin(2pFct + ?(t))  
d) x(t) = a(t) sin(2pFct – ?(t))  
View Answer

Answer: b  
Explanation: [](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-representation-bandpass-signals-q14.png)  
Hence proved.

15. In the equation x(t) = a(t)cos[2πFct+θ(t) ], Which of the following relations between a(t) and x(t), θ(t) and x(t) are true?  
a) a(t), θ(t) are called the Phases of x(t)  
b) a(t) is the Phase of x(t), θ(t) is called the Envelope of x(t)  
c) a(t) is the Envelope of x(t), θ(t) is called the Phase of x(t)  
d) None of the mentioned  
View Answer

Answer: c  
Explanation: In the equation x(t) = a(t) cos[2πFct+θ(t) ], the signal a(t) is called the Envelope of x(t), and θ(t) is called the phase of x(t)

Questions & Answers (MCQs) focuses on “Quantization and Coding”.

1. The basic task of the A/D converter is to convert a discrete set of digital code words into a continuous range of input amplitudes.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: The basic task of the A /D converter is to convert a continuous range of input amplitude into a discrete set of digital code words. This conversion involves the processes of Quantization and Coding.

2. What is the type of quantizer, if a Zero is assigned a quantization level?  
a) Midrise type  
b) Mid tread type  
c) Mistreat type  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: If a zero is assigned a quantization level, the quantizer is of the mid treat type.

3. What is the type of quantizer, if a Zero is assigned a decision level?  
a) Midrise type  
b) Mid tread type  
c) Mistreat type  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: If a zero is assigned a decision level, the quantizer is of the midrise type.

4. What is the term used to describe the range of an A/D converter for bipolar signals?  
a) Full scale  
b) FSR  
c) Full-scale region  
d) FS  
View Answer

Answer: b  
Explanation: The term Full-scale range (FSR) is used to describe the range of an A /D converter for bipolar signals (i.e., signals with both positive and negative amplitudes).

5. What is the term used to describe the range of an A/D converter for uni-polar signals?  
a) Full scale  
b) FSR  
c) Full-scale region  
d) FSS  
View Answer

Answer: a  
Explanation: The term Full scale (FS) is used for uni-polar signals

6. What is the fixed range of the quantization error eq(n)?  
a) – ∆/6 < eq(n) ≤ ∆/6  
b) – ∆/4 < eq(n) ≤ ∆/4  
c) – ∆/2 <eq(n) ≤ ∆/2  
d) – ∆/16 <eq(n) ≤ ∆/16  
View Answer

Answer: c  
Explanation: The quantization error eq(n) is always in the range – ∆/2 < eq(n) ≤ ∆/2 , where ∆ is quantizer step size.

7. If the dynamic range of the signal is smaller than the range of quantizer, the samples that exceed the quantizer are clipped, resulting in large quantization error.  
a) True  
b) False  
View Answer

Answer: b  
Explanation: If the dynamic range o f the signal, defined as xmax-xmin, is larger than the range of the quantizer, the samples that exceed the quantizer range are clipped, resulting in a large (greater than ∆/2) quantization error.

8. What is the relation defined by the operation of quantizer?  
a) xq(n) ≡ Q[x(n)]= xk  
b) xq(n) = Q[x(n)]= xk , if x(n) ∈ I\_k  
c) xq(k) ≡ Q[x(k)]= xk  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The possible outputs of the quantizer (i.e., the quantization levels) are denoted as x1,x 2,…xL . The operation of the quantizer is defined by the relation, xq(n) ≡ Q[x(n) ]= xk,if x(n) ∈ Ik.

9. What is the step size or the resolution of an A/D converter?  
a) ∆ = (R)/2(b+1)  
b) ∆ = (R )/2(b-1)  
c) ∆ = (R )/3(b+1)  
d) ∆ = (R )/2  
View Answer

Answer: a  
Explanation: The coding process in an A /D converter assigns a unique binary number to each quantization level. If we have L levels, we need at least L different binary numbers. With a word length of b + 1 bits we can represent 2^(b+1) distinct binary numbers. Hence we should have 2^(b+1) > L or, equivalently, b + 1 > log2 L. Then the step size or the resolution of the A /D converter is given by  
∆ = (R )/2(b+1), where R is the range of the quantizer.

10. In the practical A/D converters, if the first transition may not occur at exactly + 1/2 LSB ,then such kind of error is known as \_\_\_\_\_\_\_\_\_\_\_\_  
a) Scale-factor error  
b) Offset error  
c) Linearity error  
d) All of the mentioned  
View Answer

Answer: b  
Explanation: We note that practical A /D converters may have offset error (the first transition may not occur at exactly +1/2 LSB).

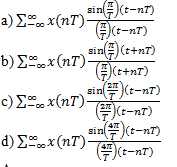
11. In the practical A/D converters, if the difference between the values at which the first transition and the last transition occur is not equal to FS – 2LSB, then such error is known as \_\_\_\_\_\_\_\_\_  
a) Scale-factor error  
b) Offset error  
c) Linearity error  
d) All of the mentioned  
View Answer

Answer: a  
Explanation: We note that practical A /D converters scale-factor (or gain) error (the difference between the values at which the first transition and the last transition occur is not equal to FS — 2LSB ).

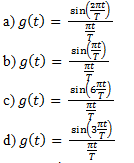
12. In the practical A/D converters, if the differences between transition values are not all equal or uniformly changing, then such error is known as ?  
a) Scale-factor error  
b) Offset error  
c) Linearity error  
d) All of the mentioned  
View Answer

Answer: c  
Explanation: We note that practical A /D converters, linearity error (the differences between transition values are not all equal or uniformly changing).

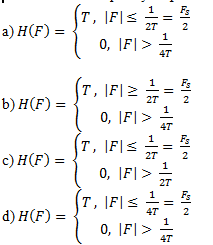
Questions & Answers (MCQs) focuses on “Digital to Analog Conversion Sample and Hold”.

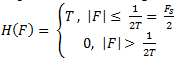
1. What is the ideal reconstruction formula or ideal interpolation formula for x(t) = \_\_\_\_\_\_\_\_\_  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-digital-analog-conversion-sample-hold-q1.png)  
View Answer

Answer: a  
Explanation: [digital-signal-processing-questions-answers-digital-analog-conversion-sample-hold-q1a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-digital-analog-conversion-sample-hold-q1a.png)where the sampling interval T = 1/Fs=1/2B,Fs is the sampling frequency and B is the bandwidth of the analog signal.

2. What is the new ideal interpolation formula described after few problems with previous one?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-digital-analog-conversion-sample-hold-q2.png)  
View Answer

Answer: b  
Explanation: The reconstruction of the signal x ( t) from its samples as an interpolation problem and have described the function:[digital-signal-processing-questions-answers-digital-analog-conversion-sample-hold-q2a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-digital-analog-conversion-sample-hold-q2a.png)

3. What is the frequency response of the analog filter corresponding to the ideal interpolator?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-digital-analog-conversion-sample-hold-q3.png)  
View Answer

Answer: c  
Explanation: The analog filter corresponding to the ideal interpolator has a frequency response:  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-digital-analog-conversion-sample-hold-q3a.png), H(F) is the Fourier transform of the interpolation function g(t).

4. The reconstruction o f the signal from its samples as a linear filtering process in which a discrete-time sequence of short pulses (ideally impulses) with amplitudes equal to the signal samples, excites an analog filter.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The reconstruction o f the signal from its samples as a linear filtering process in which a discrete-time sequence of short pulses (ideally impulses) with amplitudes equal to the signal samples, excites an analog filter.

5. The ideal reconstruction filter is an ideal low pass filter and its impulse response extends for all time.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: The ideal reconstruction filter is an ideal low pass filter and its impulse response extends for all time. Hence the filter is noncausal and physically nonrealizable. Although the interpolation filter with impulse response given can be approximated closely with some delay, the resulting function is still impractical for most applications where D /A conversion are required.

6. D /A conversion is usually performed by combining a D /A converter with a sample-and-hold (S/H ) and followed by a low pass (smoothing) filter.  
a) True  
b) False  
View Answer

Answer: a  
Explanation: D /A conversion is usually performed by combining a D /A converter with a sample-and hold (S/H) and followed by a low pass (smoothing) filter. T he D /A converter accepts at its input, electrical signals that correspond to a binary word, and produces an output voltage or current that is proportional to the value o f the binary word.

7. The time required for the output o f the D /A converter to reach and remain within a given fraction of the final value, after application of the input code word is called?  
a) Converting time  
b) Setting time  
c) Both Converting & Setting time  
d) None of the mentioned  
View Answer

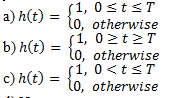
Answer: b  
Explanation: An important parameter o f a D /A converter is its settling time, which is defined as the time required for the output o f the D /A converter to reach and remain within a given fraction (usually,±1/2 LSB) of the final value, after application of the input code word.

8. In D/A converter, the application of the input code word results in a high-amplitude transient, called?  
a) Glitch  
b) Deglitch  
c) Glitter  
d) None of the mentioned  
View Answer

Answer: a  
Explanation: The application o f the input code word results in a high-amplitude transient, called a “glitch.” This is especially the case when two consecutive code words to the A /D differ by several bits.

9. In a D/A converter, the usual way to solve the glitch is to use deglitcher. How is the Deglitcher designed?  
a) By using a low pass filter  
b) By using a S/H circuit  
c) By using a low pass filter & S/H circuit  
d) None of the mentioned  
View Answer

Answer: b  
Explanation: The usual way to remedy this problem is to use an S/H circuit designed to serve as a “deglitcher”. Hence the basic task of the S/H is to hold the output of the D /A converter constant at the previous output value until the new sample at the output o f the D /A reaches steady state, and then it samples and holds the new value in the next sampling interval. Thus the S/H approximates the analog signal by a series of rectangular pulses whose height is equal to the corresponding value of the signal pulse.

10. What is the impulse response of an S/H, when viewed as a linear filter?  
[](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-digital-analog-conversion-sample-hold-q10.png)d) None  
View Answer

Answer: a  
Explanation: W hen viewed as a linear filter, the S/H has an impulse response:  
[digital-signal-processing-questions-answers-digital-analog-conversion-sample-hold-q10a](https://www.sanfoundry.com/wp-content/uploads/2015/10/digital-signal-processing-questions-answers-digital-analog-conversion-sample-hold-q10a.png)

# Best Reference Books – Digital Signal Processing

We have compiled a list of Best Reference Books on Digital Signal Processing Subject. These books are used by students of top universities, institutes and colleges.

Here is the full list of best reference books on Digital Signal Processing.

|  |  |
| --- | --- |
| 1. “Digital Signal Processing” by Proakis and Manoloki | |
|  |  |
| 2. “Digital Signal Processing” by S K Mitra | |
|  |  |
| 3. “Theory and Application of Digital Signal Processing” by Rabinar L R and Gold B | |
|  |  |
| 4. “Introduction to Digital Signal Processing” by Johnson | |
|  |  |
| 5. “Digital Signal Processing” by Alan V Oppennheim | |
|  |  |
| 6. “Understanding Digital Signal Processing” by Lyons | |
|  |  |
| 7. “Digital Signal Processing” by A Nagoor Kani | |
|  |  |
| 8. “Digital Signal Processing” by A Anand Kumar | |
|  |  |
| 9. “Digital Signal Processing” by Salivahanan | |
|  |  |
| 10. “Digital Signal Processing” by Ronald W Schafer | |
|  |  |
| 11. “Schaum’s Outline of Theory and Problems of Digital Signal Processing” by Monson H Hayes | |
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[Manish Bhojasia](https://www.sanfoundry.com/about/), a technology veteran with 20+ years @ Cisco & Wipro, is Founder and CTO at Sanfoundry. He is Linux Kernel Developer and SAN Architect and is passionate about competency developments in these areas. He lives in Bangalore and delivers focused training sessions to IT professionals in Linux Kernel, Linux Debugging, Linux Device Drivers, Linux Networking, Linux Storage & Cluster Administration, Advanced C Programming, SAN Storage Technologies, SCSI Internals and Storage Protocols such as iSCSI & Fiber Channel. Stay connected with him below:  
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