NOORUL ISLAM COLLEGE OF ENGG,

Kumaracoil

DEPARTMENT OF ECE

2 MARKS & QUESTION- ANSWERS

EC 1302- Digital Signal Processing

Class: S5 ECE (A&B)

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Electronics & Communication Engineering.

Digital Signal Processing
S5ECE A&B

1. What is a continuous and discrete time signal?

Ans:

Continuous time signal: A signal x(t) is said to be continuous if it is defined for all time t. Continuous time signal arise naturally when a physical waveform such as acoustics wave or light wave is converted into a electrical signal. This is effected by means of transducer.(e.g.) microphone, photocell.

Discrete time signal: A discrete time signal is defined only at discrete instants of time. The independent variable has discrete values only, which are uniformly spaced. A discrete time signal is often derived from the continuous time signal by sampling it at a uniform rate.

2. Give the classification of signals?

Ans:

Continuous-time and discrete time signals
Even and odd signals
Periodic signals and non-periodic signals
Deterministic signal and Random signal
Energy and Power signal

3. What are the types of systems?

Ans:

Continuous time and discrete time systems
Linear and Non-linear systems
Causal and Non-causal systems
Static and Dynamic systems
Time varying and time in-varying systems
Distributive parameters and Lumped parameters systems
Stable and Un-stable systems.

4. What are even and odd signals?

Ans:

Even signal: continuous time signal x(t) is said to be even if it satisfies the condition x(t)=x(-t) for all values of t.

Odd signal: he signal x(t) is said to be odd if it satisfies the condition x(-t)=-x(t) for all t. In other words even signal is symmetric about the time origin or the vertical axis, but odd signals are anti-symmetric about the vertical axis.

5. What are deterministic and random signals?

Ans:

Deterministic Signal: deterministic signal is a signal about which there is no certainty with respect to its value at any time. Accordingly we find that deterministic signals may be modeled as completely specified functions of time.

Random signal: random signal is a signal about which there is uncertainty before its actual occurrence. Such signal may be viewed as group of signals with each signal in the ensemble having different wave forms

.(e.g.) The noise developed in a television or radio amplifier is an example for random signal.

6. What are energy and power signal?

Ans:

Energy signal: signal is referred as an energy signal, if and only if the total energy of the signal satisfies the condition $0<E<\infty$. The total energy of the continuous time signal x(t) is given as

$E=\lim_{T\to\infty}\int x^2(t)dt$, integration limit from –T/2 to +T/2

Power signal: signal is said to be powered signal if it satisfies the condition $0<P<\infty$. The average power of a continuous time signal is given by

$P=\lim_{T\to\infty}1/T\int x2(t)dt$, integration limit is from-T/2 to +T/2.

7. What are the operations performed on a signal?

Ans:

Operations performed on dependent variables:

Amplitude scaling: y (t) =cx (t), where c is the scaling factor, x(t) is the continuous time signal.

Addition: y (t)=x_1(t)+x_2(t)

Multiplication y (t)=x1(t)x2(t)

Differentiation: y (t)=d/dt x(t)

Integration (t) =∫x(t)dt

Operations performed on independent variables

Time shifting

Amplitude scaling

Time reversal

8. What are elementary signals and name them?

Ans:

The elementary signals serve as a building block for the construction of more complex signals. They are also important in their own right, in that they may be used to model many physical signals that occur in nature.

There are five elementary signals. They are as follows

Unit step function
Unit impulse function
Ramp function
Exponential function
Sinusoidal function

9. What are the properties of a system?
Ans:
Stability: A system is said to be stable if the input x(t) satisfies the condition(t)│$\leq M_x < \infty$ and the out put satisfies the condition │y(t)│$\leq M_y < \infty$ for all t.
Memory: A system is said to be memory if the output signal depends on the present and the past inputs.
Invertibility: A system is said to be invertible if the input of the system con be recovered from the system output.
Time invariance: A system is said to be time invariant if a time delay or advance of the input signal leads to an identical time shift in the output signal.
Linearity: A system is said to be linear if it satisfies the super position principle

$$\text{i.e.) } R(ax_1(t)+bx_2(t))=ax_1(t)+bx_2(t)$$

10. What is memory system and memory less system?
Ans:
A system is said to be *memory system* if its output signal at any time depends on the past values of the input signal. circuit with inductors capacitors are examples of memory system..
A system is said to be *memory less system* if the output at any time depends on the present values of the input signal. An electronic circuit with resistors is an example for memory less system.

11. What is an invertible system?
Ans:
A system is said to be *invertible system* if the input of the system can be recovered from the system output. The set of operations needed to recover the input as the second system connected in cascade with the given system such that the output signal of the second system is equal to the input signal applied to the system.

$$H^{-1}\{y(t)\}=H^{-1}\{H\{x(t)\}\}.$$

12. What are time invariant systems?
Ans:
A system is said to be time invariant system if a time delay or advance of the input signal leads to an ideditical shift in the output signal. This implies that a time invariant system responds ideditically no matter when the input signal is applied. It also satisfies the condition

$$R\{x(n-k)\}=y(n-k).$$

13. Is a discrete time signal described by the input output relation $y[n]= r^n x[n]$ time invariant.

Ans:

A signal is said to be time invariant if $R\{x[n-k]\}= y[n-k]$

$R\{x[n-k]\}=R(x[n]) / x[n]\rightarrow x[n-k]$

$=r^n x [n-k]$ ---------------- (1)

$y[n-k]=y[n] / n\rightarrow n-k$

$=r^{n-k} x [n-k]$ -------------------(2)

Equations (1)≠Equation(2)

Hence the signal is time variant.

14. Show that the discrete time system described by the input-output relationship $y[n] =nx[n]$ is linear?

Ans:

For a sys to be linear $R\{a_1x_1[n]+b_1x_2[n]\}=a_1y_1[n]+b_1y_2[n]$

L.H.S:$R\{ a_1x_1[n]+b_1x_2[n] \}=R\{x[n]\} /x[n] \rightarrow a_1x_1[n]+b_1x_2[n]$

$= a_1 nx_1[n]+b_1 nx_2[n]$ ------------------(1)

R.H.S: $a_1y_1[n]+b_1y_2[n]= a_1 nx_1[n]+b_1 nx_2[n]$ --------------------(2)

Equation(1)=Equation(2)

Hence the system is linear

15. What is SISO system and MIMO system?

Ans:

A control system with single input and single output is referred to as single input single output system. When the number of plant inputs or the number of plant outputs is more than one the system is referred to as multiple input output system. In both the case, the controller may be in the form of a digital computer or microprocessor in which we can speak of the digital control systems.

16. What is the output of the system with system function H_1 and H_2 when connected in cascade and parallel?

Ans:

When the system with input x(t) is connected in cascade with the system H1 and H2 the output of the system is

$$y(t)=H_2\{H_1\{x(t)\}\}$$

When the system is connected in parallel the output of the system is given by

$$y(t)=H_1x_1(t)+H_2x_2(t).$$

17. What do you mean by periodic and non-periodic signals?

A signal is said to be periodic if

$x(n+N)=x(n)$

Where N is the time period.

A signal is said to be non-periodic if

$x(n+N)\neq x(n)$.

18.Determine the convolution sum of two sequences $x(n) = \{3, 2, 1, 2\}$ and

$h(n) = \{1, 2, 1, 2\}$

Ans: $y(n) = \{3,8,8,12,9,4,4\}$

19.Find the convolution of the signals

$x(n) = 1 \quad n=-2,0,1$

$\quad = 2 \quad n=-1$

$\quad = 0 \quad$ elsewhere.

Ans: $y(n) = \{1,1,0,1,-2,0,-1\}$

20.Detemine the solution of the difference equation

$y(n) = 5/6\ y(n-1) – 1/6\ y(n-2) + x(n)$ for $x(n) = 2^n\ u(n)$

Ans: $y(n) = -(1/2)^n\ u(n) + 2/3(1/3)^n\ u(n)+8/5\ 2^n u(n)$

21.Determine the response y(n), n>=0 of the system described by the second order difference equation

$y(n) – 4y(n-1) + 4y(n-2) = x(n) – x(n-1)$ when the input is $x(n) = (-1)^n\ u(n)$ and the initial condition are $y(-1) = y(-2)=1$.

Ans:$y(n) = (7/9-5/3n)2^n\ u(n) +2/9(-1)^n\ u(n)$

22. Differentiate DTFT and DFT

DTFT output is continuous in time where as DFT output is Discrete in time.

23.Differentiate between DIT and DIF algorithm

DIT – Time is decimated and input is bi reversed format output in natural order
DIF – Frequency is decimated and input is natural order output is bit reversed format.

24. How many stages are there for 8 point DFT

8

25 How many multiplication terms are required for doing DFT by expressional method and FFT method

expression –n2 FFT N /2 log N

26. Distinguish IIR and FIR filters

FIR	IIR
Impulse response is finite They have perfect linear phase	Impulse Response is infinite They do not have perfect linear phase
Non recursive	Recursive
Greater flexibility to control the shape of magnitude response	Less flexibility

27. Distinguish analog and digital filters

Analog	digital
Constructed using active or passive components and it is described by a differential equation	Consists of elements like adder, subtractor and delay units and it is described by a difference equation
Frequency response can be changed by changing the components	Frequency response can be changed by changing the filter coefficients
It processes and generates analog output	Processes and generates digital output
Output varies due to external conditions	Not influenced by external conditions

28. Write the expression for order of Butterworth filter?

The expression is $N=\log(\lambda/€)^{1/2}/\log(1/k)^{½}$

29. Write the expression for the order of chebyshev filter?

$N=\cosh^{-1}(\lambda/e)/\cosh^{-1}(1/k)$

30. Write the various frequency transformations in analog domain?

LPF to LPF:$s=s/\Omega c$
LPF to HPF:$s=\Omega c/s$
LPF to BPF:$s=s2xlxu/s(xu-xl)$
LPF to BSF:$s=s(xu-xl)?s2=xlxu$. $X=\Omega$

31. Write the steps in designing chebyshev filter?

1. Find the order of the filter.
2. Find the value of major and minor axis. λ
3. Calculate the poles.

4. Find the denominator function using the above poles.
5. The numerator polynomial value depends on the value of n.
If n is odd: put s=0 in the denominator polynomial.
If n is even put s=0 and divide it by $(1+e2)^{1/2}$

32. Write down the steps for designing a Butterworth filter?

1. From the given specifications find the order of the filter
2 find the transfer function from the value of N
3. Find Ωc
4 find the transfer function ha(s) for the above value of Ωc by su s by that value.

33. State the equation for finding the poles in chebyshev filter

sk=acos¢k+jbsin¢k,where ¢k=∏/2+(2k-1)/2n)∏

34. State the steps to design digital IIR filter using bilinear method

Substitute s by 2/T (z-1/z+1), where T=2/Ω (tan (w/2) in h(s) to get h (z)

35. What is warping effect?

For smaller values of w there exist linear relationship between w and .but for larger values of w the relationship is nonlinear. This introduces distortion in the frequency axis. This effect compresses the magnitude and phase response. This effect is called warping effect

36. Write a note on pre warping.

The effect of the non linear compression at high frequencies can be compensated. When the desired magnitude response is piecewise constant over frequency, this compression can be compensated by introducing a suitable rescaling or prewar ping the critical frequencies.

37. Give the bilinear transform equation between s plane and z plane

s=2/T (z-1/z+1)

38. Why impulse invariant method is not preferred in the design of IIR filters other than low pass filter?

In this method the mapping from s plane to z plane is many to one. Thus there ire an infinite number of poles that map to the same location in the z plane, producing an aliasing effect. It is inappropriate in designing high pass filters. Therefore this method is not much preferred.

39. By impulse invariant method obtain the digital filter transfer function and the differential equation of the analog filter h(s) =1/s+1

H (z) =$1/1-e^{-T}z^{-1}$

Y/x(s) =1/s+1
Cross multiplying and taking inverse lap lace we get,
D/dt(y(t)+y(t)=x(t)

40. What is meant by impulse invariant method?

In this method of digitizing an analog filter, the impulse response of the resulting digital filter is a sampled version of the impulse response of the analog filter. For e.g. if the transfer function is of the form, 1/s-p, then
H (z) =$1/1-e^{-pT}z^{-1}$

41. What do you understand by backward difference?

One of the simplest methods of converting analog to digital filter is to approximate the differential equation by an equivalent difference equation.
d/dt(y(t)/t=nT=(y(nT)-y(nT-T))/T

42. What are the properties of chebyshev filter?

1. The magnitude response of the chebyshev filter exhibits ripple either in the stop band or the pass band.
2. The poles of this filter lies on the ellipse

43. Give the Butterworth filter transfer function and its magnitude characteristics for different orders of filter.

The transfer function of the Butterworth filter is given by
H (jΩ) =1/1+j (Ω/Ωc) N

44. Give the magnitude function of Butterworth filter.

The magnitude function of Butterworth filter is
$|h(j\Omega)=1/[1+(\Omega/\Omega c)^{2N}]^{1/2}$,N=1,2,3,4,….

45. Give the equation for the order N, major, minor axis of an ellipse in case of chebyshev filter?

The order is given by $N=\cosh^{-1}(((10^{.1\alpha p})-1/10^{.1\alpha s}-1)1/2))/\cosh^{-1}\Omega s/\Omega p$

$A= (\mu^{1/N}-\mu^{-1/N})/2\Omega p$
$B=\Omega p\ (\mu^{1/N}+ \mu^{-1/N})/2$

46. Give the expression for poles and zeroes of a chebyshev type 2 filters

The zeroes of chebyshev type 2 filter SK=jΩs/sinkΦk, k=1….N

The poles of this filter xk+jyk
$xk = \Omega' s\sigma k / \Omega' s^2+\sigma k^2$
$yk = \Omega' s\Omega' k / \Omega' s^2+\sigma k^2$ $\sigma k = a\cos\Phi k$

47. How can you design a digital filter from analog filter?

Digital filter can de designed from analog filter using the following methods

1. Approximation of derivatives
2. Impulse invariant method
3. Bilinear transformation

48. write down bilinear transformation.

s=2/T (z-1/z+1)

49. List the Butterworth polynomial for various orders.

N	Denominator polynomial
1	S+1
2	$S^2+.707s+1$
3	$(s+1)(s^2+s+1)$
4	$(s^2+.7653s+1)(s^2+1.84s+1)$
5	$(s+1)(s^2+.6183s+1)(s^2+1.618s+1)$
6	$(s^2+1.93s+1)(s^2+.707s+1)(s^2+.5s+1)$
7	$(s+1)(s^2+1.809s+1)(s^2+1.24s+1)(s^2+.48s+1)$

50. Differentiate Butterworth and Chebyshev filter.

Butterworth dampimg factor 1.44 chebyshev 1.06
Butterworth flat response damped response.

51. What is filter?

Filter is a frequency selective device ,which amplify particular range of frequencies and attenuate particular range of frequencies.

52. What are the types of digital filter according to their impulse response?

IIR(Infinite impulse response)filter
FIR(Finite Impulse Response)filter.

53. How phase distortion and delay distortion are introduced?

The phase distortion is introduced when the phase characteristics of a filter is nonlinear with in the desired frequency band.
The delay distortion is introduced when the delay is not constant with in the desired frequency band.

54. what is mean by FIR filter?

The filter designed by selecting finite number of samples of impulse response (h(n) obtained from inverse fourier transform of desired frequency response H(w)) are called FIR filters

55. Write the steps involved in FIR filter design

Choose the desired frequency response $H_d(w)$
Take the inverse fourier transform and obtain $H_d(n)$
Convert the infinite duration sequence $H_d(n)$ to h(n)
Take Z transform of h(n) to get H(Z)

56. What are advantages of FIR filter?

Linear phase FIR filter can be easily designed .
Efficient realization of FIR filter exists as both recursive and non-recursive structures.
FIR filter realized non-recursively stable.
The round off noise can be made small in non recursive realization of FIR filter.

57. What are the disadvantages of FIR FILTER

The duration of impulse response should be large to realize sharp cutoff filters.
The non integral delay can lead to problems in some signal processing applications.

58. What is the necessary and sufficient condition for the linear phase characteristic of a FIR filter?

The phase function should be a linear function of w, which inturn requires constant group delay and phase delay.

59. List the well known design technique for linear phase FIR filter design?

Fourier series method and window method
Frequency sampling method.
Optimal filter design method.

60. Define IIR filter?

The filter designed by considering all the infinite samples of impulse response are called IIR filter.

61. For what kind of application , the antisymmetrical impulse response can be used?

The ant symmetrical impulse response can be used to design Hilbert transforms and differentiators.

62. For what kind of application , the symmetrical impulse response can be used?

The impulse response ,which is symmetric having odd number of samples can be used to design all types of filters ,i.e , lowpass,highpass,bandpass and band reject.
The symmetric impulse response having even number of samples can be used to design lowpass and bandpass filter.

63.What is the reason that FIR filter is always stable?

FIR filter is always stable because all its poles are at the origin.

64.What condition on the FIR sequence h(n) are to be imposed n order that this filter can be called a liner phase filter?

The conditions are

(i) Symmetric condition h(n)=h(N-1-n)

(ii) Antisymmetric condition h(n)=-h(N-1-n)

65. Under what conditions a finite duration sequence h(n) will yield constant group delay in its frequency response characteristics and not the phase delay?

If the impulse response is anti symmetrical ,satisfying the condition

H(n)=-h(N-1-n)

The frequency response of FIR filter will have constant group delay and not the phase delay .

66. State the condition for a digital filter to be causal and stable?

A digital filter is causal if its impluse response h(n)=0 for n<0.

A digital filter is stable if its impulse response is absolutely summable ,i.e,

$$\sum_{n=-\infty}^{\infty} |h(n)| < \infty$$

67. What are the properties of FIR filter?

1.FIR filter is always stable.

2.A realizable filter can always be obtained.

3.FIR filter has a linear phase response.

68. When cascade from realization is preferred in FIR filters?

The cascade from realization is preferred when complex zeros with absolute magnitude less than one.

69. What are the disadvantage of Fourier series method ?

In designing FIR filter using Fourier series method the infinite duration impulse response is truncated at $n=\pm(N-1/2)$.Direct truncation of the series will lead to fixed percentage overshoots and undershoots before and after an approximated discontinuity in the frequency response .

70. What is Gibbs phenomenon?

OR

What are Gibbs oscillations?

One possible way of finding an FIR filter that approximates H(ejω)would be to truncate the infinite Fourier series at $n=\pm(N-1/2)$.Abrupt truncation of the series will lead to oscillation both in pass band and is stop band .This phenomenon is known as Gibbs phenomenon.

71. What are the desirable characteristics of the windows?

The desirable charaterstics of the window are

1.The central lobe of the frequency response of the window should contain most of the energy and should be narrow.
2.The highest side lobe level of the frequency response should be small.
3.The sides lobes of the frequency response should decrease in energy rapidly as ω tends to π .

72. Compare Hamming window with Kaiser window.

Hamming window	Kaiser window
1.The main lobe width is equal to$8\pi/N$ and the peak side lobe level is –41dB. 2.The low pass FIR filter designed will have first side lobe peak of –53 dB	The main lobe width ,the peak side lobe level can be varied by varying the parameter α and N. The side lobe peak can be varied by varying the parameter α.

73.What is the necessary and sufficient condition for linear phase characteristics in FIR filter?

The necessary and sufficient condition for linear phase characteristics in FIR filter is the impulse response h(n) of the system should have the symmetry property,i.e,

$$H(n) = h(N-1-n)$$

Where N is the duration of the sequence .

74.What are the advantage of Kaiser widow?

1.It provides flexibility for the designer to select the side lobe level and N .
2. It has the attractive property that the side lobe level can be varied continuously from the low value in the Blackman window to the high value in the rectangle window .

75.What is the principle of designing FIR filter using frequency sampling method?

In frequency sampling method the desired magnitude response is sampled and a linear phase response is specified .The samples of desired frequency response are defined as DFT coefficients. The filter coefficients are then determined as the IDFT of this set of samples.

76.For what type of filters frequency sampling method is suitable?

Frequency sampling method is attractive for narrow band frequency selective filters where only a few of the samples of the frequency response are non-zero.

77.What is meant by autocorrelation?

The autocorrelation of a sequence is the correlation of a sequence with its shifted version, and this indicates how fast the signal changes.

78.Define white noise?

A stationary random process is said to be white noise if its power density spectrum is constant. Hence the white noise has flat frequency response spectrum.

$S_X(w) = \sigma_x^2$, $-\pi \leq w\pi$

79.what do you understand by a fixed-point number?

In fixed point arithmetic the position of the binary point is fixed. The bit to the right represent the fractional part of the number & those to the left represent the integer part. For example, the binary number 01.1100 has the value 1.75 in decimal.

80.What is the objective of spectrum estimation?

The main objective of spectrum estimation is the determination of the power spectral density of a random process. The estimated PSD provides information about the structure of the random process which can be used for modeling, prediction or filtering of the deserved process.

81.List out the addressing modes supported by C5X processors?

1. Direct addressing
2. Indirect addressing
3. Immediate addressing
4. Dedicated-register addressing
5. Memory-mapped register addressing
6. Circular addressing

82.what is meant by block floating point representation? What are its advantages?

In block point arithmetic the set of signals to be handled is divided into blocks. Each block have the same value for the exponent. The arithmetic operations with in the block uses fixed point arithmetic & only one exponent per block is stored thus saving memory. This representation of numbers is more suitable in certain FFT flow graph & in digital audio applications.

83.what are the advantages of floating point arithmetic?

1. Large dynamic range
2. Over flow in floating point representation is unlike.

84.what are the three-quantization errors to finite word length registers in digital filters?

1. Input quantization error 2. Coefficient quantization error 3. Product quantization error

85.How the multiplication & addition are carried out in floating point arithmetic?

In floating point arithmetic, multiplication are carried out as follows,

Let f1 = M1*2c1 and f2 = M2*2c2. Then f3 = f1*f2 = (M1*M2) 2(c1+c2)

That is, mantissa is multiplied using fixed-point arithmetic and the exponents are added.

The sum of two floating-point number is carried out by shifting the bits of the mantissa of the smaller number to the right until the exponents of the two numbers are equal and then adding the mantissas.

86.What do you understand by input quantization error?

In digital signal processing, the continuous time input signals are converted into digital using a b-bit ACD. The representation of continuous signal amplitude by a fixed digit produce an error, which is known as input quantization error.

87.List the on-chip peripherals in 5X.

The C5X DSP on-chip peripherals available are as follows:

1. Clock Generator
2. Hardware Timer
3. Software-Programmable Wait-State Generators
4. Parallel I/O Ports
5. Host Port Interface (HPI)
6. Serial Port
7. Buffered Serial Port (BSP)
8. Time-Division Multiplexed (TDM) Serial Port
9. User-Maskable Interrupts

88.what is the relationship between truncation error e and the bits b for representing a decimal into binary?

For a 2's complement representation, the error due to truncation for both positive and negative values of x is $0 >= xt-x > -2-b$

Where b is the number of bits and xt is the truncated value of x.

The equation holds good for both sign magnitude, 1's complement if $x>0$

If $x<0$, then for sign magnitude and for 1's complement the truncation error satisfies.

89.what is meant rounding? Discuss its effect on all types of number representation?

Rounding a number to b bits is accomplished by choosing the rounded result as the b bit number closest to the original number unrounded.

For fixed point arithmetic, the error made by rounding a number to b bits satisfy the inequality

$$\frac{-2-b}{2} <= xt-x <= \frac{2-b}{2}$$

for all three types of number systems, i.e., 2's complement, 1's complement & sign magnitude.

For floating point number the error made by rounding a number to b bits satisfy the inequality

$$-2-b <= E <= 2-b \qquad \text{where } E = \frac{xt-x}{x}$$

90.what is meant by A/D conversion noise?

A DSP contains a device, A/D converter that operates on the analog input x(t) to produce xq(t) which is binary sequence of 0s and 1s.

At first the signal x(t) is sampled at regular intervals to produce a sequence x(n) is of infinite precision. Each sample x(n) is expressed in terms of a finite number of bits given the sequence xq(n). The difference signal e(n)=xq(n)-x(n) is called A/D conversion noise.

91.what is the effect of quantization on pole location?

Quantization of coefficients in digital filters lead to slight changes in their value. This change in value of filter coefficients modify the pole-zero locations. Some times the pole locations will be changed in such a way that the system may drive into instability.

92.which realization is less sensitive to the process of quantization?

Cascade form.

93.what is meant by quantization step size?

Let us assume a sinusoidal signal varying between +1 and -1 having a dynamic range 2. If the ADC used to convert the sinusoidal signal employs b+1 bits including sign bit, the number of levels available for quantizing x(n) is 2b+1. Thus the interval between successive levels

$$q = \frac{2}{2b+1} = 2-b$$

Where q is known as quantization step size.

94.How would you relate the steady-state noise power due to quantization and the b bits representing the binary sequence?

Steady state noise power

Where b is the number of bits excluding sign bit.

95.what is overflow oscillation?

The addition of two fixed-point arithmetic numbers cause over flow the sum exceeds the word size available to store the sum. This overflow caused by adder make the filter output to oscillate between maximum amplitude limits. Such limit cycles have been referred to as over flow oscillations.

96.what are the methods used to prevent overflow?

There are two methods used to prevent overflow

1. Saturation arithmetic 2. Scaling

97.what are the two kinds of limit cycle behavior in DSP?

1.zero input limit cycle oscillations

2.Overflow limit cycle oscillations

98.Determine "dead band" of the filter

The limit cycle occur as a result of quantization effect in multiplication. The amplitudes of the output during a limit cycle are confined to a range of values called the dead band of the filter.

***99.Explain briefly the need for scaling in the digital filter implementation*.**

To prevent overflow, the signal level at certain points in the digital filter must be scaled so that no overflow occurs in the adder.

100.What are the different buses of TMS320C5X and their functions?

The C5X architecture has four buses and their functions are as follows:

Program bus (PB):

It carries the instruction code and immediate operands from program memory space to the CPU.

Program address bus (PAB):

It provides addresses to program memory space for both reads and writes.

Data read bus (DB):

It interconnects various elements of the CPU to data memory space.

Data read address bus (DAB):

It provides the address to access the data memory space.

Part B

1. Determine the DFT of the sequence

x(n) =1/4, for 0<=n <=2

0, otherwise

Ans: The N point DFT of the sequence x(n) is defined as

$$x(k)= \sum_{n=0}^{N-1} x(n)e^{-j2\pi nk/N} \qquad K=0,1,2,3,\dots N-1$$

$$x(n) = (1/4,1/4,1/4)$$

$$X(k) = ¼\, e^{-j2\pi k/3}[1+2\cos(2\pi k/3)] \quad \text{where } k= 0,1,\dots\dots\dots,N-1$$

2. Derive the DFT of the sample data sequence x(n) = {1,1,2,2,3,3}and compute the corresponding amplitude and phase spectrum.

Ans: The N point DFT of the sequence x(n) is defined as

$$X(k)= \sum^{N-1} x(n)e^{-j2\pi nk/N} \qquad K=0,1,2,3,\dots N-1$$

n=0

$X(0) = 12$

$X(1) = -1.5 + j2.598$

$X(2) = -1.5 + j0.866$

$X(3) = 0$

$X(4) = -1.5 - j0.866$

$X(5) = -1.5-j2.598$

$X(k) = \{12, -1.5 + j2.598, -1.5 + j0.866, 0, -1.5 - j0.866, -1.5-j2.598\}$

$|X(k)|=\{12, 2.999, 1.732, 0, 1.732, 2.999\}$

$\angle X(k)=\{0, -\pi/3, -\pi/6, 0, \pi/6, \pi/3\}$

3.Given x(n) = {0,1,2,3,4,5,6,7} find X(k) using DIT FFT algorithm.

Ans: Given N = 8

$W_N^k = e^{-j(2\pi/N)k}$

$W_8^0 = 1$

$W_8^1 = 0.707-j0.707$

$W_8^2 = -j$

$W_8^3 = -0.707-j0.707$

Using butterfly diagram

$X(k) = \{28, -4+j9.656, -4+j4, -4+j1.656, -4, -4-j1.656, -4-j4, -4-j9.656\}$

4.Given X(k) = {28,-4+j9.656,-4+j4,-4+j1.656,-4,-4-j1.656,-4-j4,-4-j9.656} ,find x(n) using inverse DIT FFT algorithm.

$W_N^k = e^{j(2\pi/N)k}$

$W_8^0 = 1$

$W_8^1 = 0.707+j0.707$

$W_8^2 = j$

$W_8^3 = -0.707+j0.707$

$x(n) = \{0,1,2,3,4,5,6,7\}$

5.Find the inverse DFT of X(k) = {1,2,3,4}

Ans: The inverse DFT is defined as

$$x(n)=(1/N)\sum_{k=0}^{N-1} x(k)e^{j2\pi nk/N} \qquad n=0,1,2,3,\ldots N-1$$

$x(0) = 5/2$

$x(1) = -1/2-j1/2$

$x(2) = -1/2$

$x(3) = -1/2+j1/2$

$x(n) = \{5/2, -1/2-j1/2, -1/2, -1/2+j1/2\}$

6. Design an ideal low pass filter with a frequency response $H_d(e^{jw})$ =1 for –∏/2<=w<=∏/2

0 otherwise

find the value of h(n) for N=11 find H(Z) plot magnitude response

a. Find h(n) by IDTFT
b. Convert h(n) in to a fine length by truncation
c. H(0)=1/2,
   h(1)=h(-1)=0.3183
   h(2)=h(-2)=0
   h(3)=h(-3)= -0.106
   h(4)=h(-4)=0
   h(5)=h(-5)=0.06366
d. Find the transfer function H(Z) which is not realizable conver in to realizable by multiplying by $z^{-(N-1/2)}$
e. H'(Z) obtained is $0.06366-0.106z^{-2}+.3183Z^{-4}+.5Z^{-5}+.3183Z^{-6}-.106Z^{-8}+0.06366Z^{-10}$
f. Find H (e^{jw}) and plot amplitude response curve.

7. Design an ideal low pass filter with a frequency response $H_d(e^{jw})$ =1 for –∏/4<=|w|<=∏

0 otherwise

find the value of h(n) for N=11 find H(Z) plot magnitude response

g. Find h(n) by IDTFT
h. Convert h(n) in to a fine length by truncation
i. H(0)=0.75
   h(1)=h(-1)=-.22
   h(2)=h(-2)=-.159
   h(3)=h(-3)= -0.075
   h(4)=h(-4)=0
   h(5)=h(-5)=0.045
j. Find the transfer function H(Z) which is not realizable conver in to realizable by multiplying by $z^{-(N-1/2)}$
k. H'(Z) obtained is $0.045-0.075z^{-2}$ -.159 Z-3$-0.22Z^{-4}+0.75Z^{-5}-.22Z^{-6}$ $-0.159Z^{-7}$ $-.075Z^{-8}+0.045Z^{-10}$
l. Find H (e^{jw}) and plot amplitude response curve.

8. Design band pass filter with a frequency response $H_d(e^{jw})$ =1 for –∏/3<=|w|<=2∏/3

0 otherwise

find the value of h(n) for N=11 find H(Z) plot magnitude response

m. Find h(n) by IDTFT
n. Convert h(n) in to a fine length by truncation
o. Find the transfer function H(Z) which is not realizable conver in to realizable by multiplying by $z^{-(N-1/2)}$
p. H'(Z) obtained Find H (e^{jw}) and plot amplitude response curve.

9. Design band reject filter with a frequency response $H_d(e^{jw}) = 1$ for ∏/4<=|w|<=3∏/4
0 otherwise

find the value of h(n) for N=11 find H(Z) plot magnitude response

q. Find h(n) by IDTFT
r. Convert h(n) in to a fine length by truncation
s. Find the transfer function H(Z) which is not realizable conver in to realizable by multiplying by $z^{-(N-1/2)}$
t. H'(Z) obtained Find H (e^{jw}) and plot amplitude response curve.

10. Derive the condition of FIR filter to be linear in phase.

Conditions are
Group delay and Phase delay should be constant
And
And show the condition is satisfied

11Derive the expression for steady state I/P Noise Power and Steady state O/P Noise Power.

Write the derivation.

12 Draw the product quantatization model for first order and second order filter

Write the difference equation and draw the noise model.

13 For the second order filter Draw the direct form II realization and find the scaling factor S0 to avoid over flow

Find the scaling factor from the formula

$$I = \frac{1+r2}{(1-r2)(1-2r2\cos 2ø = r4)}$$

14 Explain Briefly about various number representation in digital computer.

1 Fixed point
2 Floating point
3 Block floating point

Signed magnitude represenation
1's Complement
2's Complement
etc

15 Consider the transfer function H(Z)=H1(Z)H2(Z) where H1(Z) =1/1-a1Z-1
H2(z) =1/ 1-a2Z-1
Find the o/p Round of noise power Assume a1=0.5 and a2= 0.6 and find o.p round off noise power.

Draw the round of Noise Model.
By using residue method find σ01
By using residue method find σ 02
$= \sigma_{01}\ 2+ \sigma\ 02\ \ 2$

$$\text{Ans:}\ \frac{2^{-2b}\,(5.43)}{12}$$

16.Explain the architecture of DSP processor .

Diagram. & explanation.

17.Describe briefly the different methods of power spectral estimation?

1. Bartlett method
2. Welch method
3. Blackman-Tukey method

and its derivation.

18.what is meant by A/D conversion noise. Explain in detail?

A DSP contains a device, A/D converter that operates on the analog input x(t) to produce xq(t) which is binary sequence of 0s and 1s.

At first the signal x(t) is sampled at regular intervals to produce a sequence x(n) is of infinite precision. Each sample x(n) is expressed in terms of a finite number of bits given the sequence xq(n). The difference signal e(n)=xq(n)-x(n) is called A/D conversion noise.

+ derivation.

19 onsider the transfer function H(Z)=H1(Z)H2(Z) where H1(Z) =1/1-a1Z-1
H2(z) =1/ 1-a2Z-1
Find the o/p Round of noise power Assume a1=0.7 and a2= 0.8and find o.p round

off noise power.

Draw the round of Noise Model.
By using residue method find $\sigma 01$
By using residue method find σ 02
$= \sigma_{01}\, 2 + \sigma\, 02\ \ 2$

20.Given X(k) = {1,1,1,1,1,1,1,1,} ,find x(n) using inverse DIT FFT algorithm.

$$W_N^{\ k} = e^{j(2\pi/N)k}$$

Find x(n)

21.Find the inverse DFT of X(k) = {3,4,5,6}

Ans: The inverse DFT is defined as

$$x(n)=(1/N)\sum_{k=0}^{N-1} x(k)e^{j2\pi nk/N} \qquad n=0,1,2,3,\ldots N-1$$

22. Explain various addressing modes of TMS processor.

Immediate.
Register
Register indirect
Indexed

& its detail explanation.

23 Derive the expression for steady state I/P Noise Variance and Steady state O/P Noise Variance

Write the derivation.

24. Explain briefly the periodogram method of power spectral estimation?

Write the derivation with explanation.

25. Explain various arithmetic instruction of TMS processor.

All arithmetic instruction with explanation.