1. **Question 1**

**10 Points**

An analogue signal has a bandwidth of 20 kHz. It is passed through an anti-aliasing filter with a cutoff of 15 kHz and is then sampled at 20% above the Nyquist rate by an ADC using N bits per sample. If the ADC o/p is 468kbps what is the value of N?

1. **Question 2**

**10 Points**

An analogue signal with a bandwith of 6 kHz is sampled at 1kHz above the Nyquist rate. The ADC uses N=10 bits per sample. The ADC output is then transmitted at baseband across a copper wire using a line code that requires 750Hz bandwidth per 1000bps. What is the total bandwidth (in Hz) required to transmit the ADC output across the copper wire?

1. **Question 3**

**10 Points**

The SNR at the output of an ADC using N bits per sample is 40 dB. The zero mean analogue input signal has a bandwidth of 1 MHz. How many extra bits per sample are needed to increase the SNR by at least 20 dB?

1. **Question 4**

**10 Points**

Let d(n) be the unit impulse and let x(n)=u(n) be the unit step function. Let y(n)=x(-2+n)u(-n+3)-3d(n-2). What value is y(-4)+y(-3) + …..+ y(3)+y(4) ?

1. **Question 5**

**10 Points**

Calculate the period N ( x(n) = x(n+N) ) for x(n) = sin (0.6 pi n) +cos((8pi/50)n).

1. **Question 6**

**10 Points**

Calculate the period N ( x(n) = x(n+N) ) where x(n) = cos((pi/8)n) cos((2pi/15)n) + sin((pi/10)n) sin((pi/48)n) + sin((44pi/1920)n).

1. **Question 7**

**10 Points**

Consider a DSP system with a sampling rate of Fs = 2000 Hz. Assume that a low pass filter is implemented with a (real) cutoff frequency of 500 Hz. The normalized cutoff frequency in rad/s; normalized cutoff frequency in Hz; and the real cutoff frequency in rad/s, are respectively:

pi/2   1  1000pi

0.5  pi   2000pi

pi/2  1/4  1000pi

pi/4   0.25  4000pi

1. **Question 8**

**10 Points**

Consider a discrete-time linear system with input x(n) , impulse response h(n) and output y(n). If x(n) (0=< n <=4) = {1, 2, -4, -9, 6} and h(n) (0=< n <=4) = {2, -1, 4, 5} then the output y(n) is:

{2, 3, -6, -1, 15, -62, -21, 30}

{2, 3, 6, -1, 15, -62, 21, 30}

{2, 3, -6, -1, 15, -62, 30, 30}

{2, -3, -6, -1, 15, -62, 30}

1. **Question 9**

**10 Points**

Let a DSP chip implement an IIR digital filter where the input-output linear difference equation is:

y(n) = aox(n) + a1x(n-1) + … + aM-1x(n-(M-1)) – b1y(n-1) – b2y(n-2) - … - bN-1y(n-(N-1)).

Assume that we use a Texas Instrument TMS320C6713B DSP chip which can perform 600 million multiply-accumulate (MAC) operations per second. If M=N=6, first calculate the maximum possible ADC sampling frequency (in MHz). Then from this calculate the maximum bandwidth of the analogue input signal (to two decimal places).

So the maximum allowable bandwidth of the analogue input signal (in MHz) is given as:

1. **Question 10**

**10 Points**

A digital LPF has a cutoff (in normalised rad/sec) of pi/10. If the sampling frequency is 230 kHz, what is the LPF cutoff in ‘real kHz’?

1. **Question 1**

**10 Points**

Which of the following is correct regarding the ‘frequency response’ of a linear system (either discrete-time or continuous-time)?

It is the magnitude and phase of the DTFT of the impulse response of a linear system.

It tells you how both the magnitude and the phase of a sinusoidal input to the linear system are altered (at the output) for different frquencies (compared to the input sinusoid) when the system has fully reached steady-state

For any sinusoidal input to a linear system it is both the magnitude and phase of the DTFT (or the CTFT) of the steady-state response at different frequencies.

For any input to a linear system it is the DFT/CTFT of the output divided by the DFT/CTFT of the input.

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1. **Question 2**

**10 Points**

The one-sided frequency response of a linear system (or the one-sided spectrum of a signal) gives the same information as the two-sided equivalent, assuming only real signals?

True

False

1. **Question 3**

**10 Points**

Consider the following LDE is used to filter a signal: y(n)=x(n)+0.8y(n-1). If the (music industry) sampling frequency used is Fs=44,000 Hz, what will be the phase response in degrees at 4,400 Hz? (Give your answer correct to two decimal places.)

1. **Question 4**

**10 Points**

Consider the LDE in Q3: y(n)=x(n)+0.8y(n-1). From either the pole-zero plot for H(z) or the DTFT, it operates as a crude:

LPF

HPF

Notch Filter

BPF

1. **Question 5**

**10 Points**

If we truncate a discrete-time sequence x(n) in the time-domain, the effect in the DTFT frequency domain will be:

To make the new sequence non-causal

To change the phase of the DTFT of x(n) by a linear phase term

To change both the magnitude and the phase of the DTFT of x(n).

To change both the magnitude and phase of the DTFT of x(n) and make the new sequence causal.

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1. **Question 6**

**10 Points**

Is this statement true or false: The expression for the DFT of an N-length sequence x(n) (i.e., X(k)) contains exactly the same ‘information’ as the expression for the DTFT (i.e., X(e^jw)).

True

False

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1. **Question 7**

**10 Points**

Is this statement true or false: The magnitude of the DFT of an N-length sequence x(n) (i.e., |X(k)|) contains exactly the same ‘information’ as the magnitude of the DTFT (i.e., |X(e^jw)|).

True

False

1. **Question 8**

**10 Points**

Let x1(n) = [1 4 -7 9] and x2(n) = [3 8 7 -1]. Let y(n) be the circular convolution of x1(n) and x2(n). Calculate y(n) by hand and sum all the values of y(n) to get:

116

117

118

119

1. **Question 9**

**10 Points**

Let x1(n) = [1 4 -7 9] and x2(n) = [3 8 7 -1]. Let y(n) be the linear convolution of x1(n) and x2(n). Let w1(n) = [1 4 -7 9 0 0 0] and w2(n) = [3 8 7 -1 0 0 0]. Let z(n) be the circular convolution of w1(n) and w2(n). What does the comparison of y(n) and z(n) tell you?

Both have the same DFTs

Both have the same mean value

Both have the same DFT magnitudes

Both y(n) and z(n) are identical

1. **Question 10**

**10 Points**

Let x(n) = [3 -4 8 6 4] and let the DFT be X(k) = [X(0) X(1) X(2) X(3) X(4)]. Calculate (i) E1 = x(0)^2 + x(1)^2+ x(2)^2+ x(3)^2+ x(4)^2 and (ii) E2 = [ |X(0)|^2 + |X(1)|^2 + |X(2)|^2 + |X(3)|^2 + |X(4)|^2 ]/5. Which of the following are correct (may be more than one)?

E2 is the energy in x(n)

E1 is the energy in x(n)

For x(n), E1 is the energy and E2 is the power

Both E1 and E2 are identical and so we can estimate the energy in a signal via the FFT/DFT

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Question:

An analog signal has a bandwidth of 30 kHz. It is passed through an anti-aliasing filter with a cutoff frequency of 25 kHz, and then sampled at 30% above the Nyquist rate. If the ADC output is 702kbps, what is the value of N bits per sample?

Answer:

First, let's find out the Nyquist rate using formula: 2(Bandwidth) = 230 kHz = 60 kHz. Then add 30% to this rate to get the actual sampling rate: 60 kHz \* 1.30 = 78 kHz. The rate at which bits are being transferred is 702 kbps. Each sample consists of N bits so the formula to find N is:

(ADC output rate) / (Sampling rate) = N

(702 kbps) / 78 k samples per second = 9

So, N = 9 bits per sample.

Note: All calculations assume ideal Nyquist conditions and no errors during signal processing. Actual N could be different due to factors such as noise, signal attenuation, filter design, etc.

Sure, here are ten more questions based on the previous ones:

Question: An analog signal with a bandwidth of 10 kHz is sampled at 2 kHz above the Nyquist rate. The ADC uses N=8 bits per sample. The ADC output is then transmitted at baseband across a copper wire using a line code that requires 500Hz bandwidth per 1000bps. What is the total bandwidth (in Hz) required to transmit the ADC output across the copper wire?

Answer: Firstly, the Nyquist rate for the 10kHz analogue signal is 20kHz. Sampling at 2kHz above gives us a sampling rate of 22kHz. If the ADC uses 8 bits per sample, the ADC output bitrate will be 22kHz \* 8 bits = 176kbps. Using a line code that requires 500 Hz bandwidth per 1000bps, we get Total bandwidth = 176kbps \* 500 Hz/1000bps = 88kHz.

Question: The SNR at the output of an ADC using N bits per sample is 60 dB. The zero mean analog input signal has a bandwidth of 2 MHz. How many extra bits per sample are needed to increase the SNR by at least 30 dB?

Answer: The increase in SNR by 30dB corresponds to doubling (since 3dB is equivalent to a doubling). From the Shannon's quantization theorem, we know the relation between SNR and bits per sample is SNR\_dB = 6.02N + 1.76. So, if we want to increase SNR\_dB by 30, we simply divide by 6.02 to get N ≈ 30/6.02 ≈ 4.98 or approximately 5 extra bits are required.

Question: Let x(n) be the unit step function and let d(n) be the unit impulse. Let y(n)=x(-2+n)u(-n+4)3d(n-3). What value is y(-6)+y(-5) + …..+ y(4)+y(5) ?

Answer: Zero, due to the impulse function d(n-3), y(n) is only non-zero for n=3, so y(-6)+y(-5) + …..+ y(4)+y(5) = y(3) = 0, since there are no values of 3 in that sum.

Question: With x(n) = cos(0.75 pi n) + sin((8pi/10)n), calculate the period N where x(n) = x(n+N).

Answer: The period of each component is the reciprocal of their frequency. So, for cos(0.75 pi n) the period is 2/0.75 = 8/3 ≈ 2.67 (approximated to 3 when considering integer values). For sin((8pi/10)n), the period is 2/(8/10) = 2/0.8 = 2.5 (approximated to 3). The overall period of x(n) = cos(0.75 pi n) + sin((8pi/10)n) will be the least common multiple (LCM) of 3 and 3, which is 3. So, N = 3.

Question: Assume a DSP system with a sampling rate of Fs = 1500 Hz. Assume that a low pass filter is implemented with a (real) cutoff frequency of 400 Hz. What is normalized cutoff frequency in rad/s, normalized cutoff frequency in Hz, and the real cutoff frequency in rad/s?

Answer: Given that normalized frequency is the actual frequency divided by the sampling frequency, the normalized cutoff in Hz is 400 Hz / 1500 Hz = 0.267 Hz. The normalized cutoff in rad/s is 2pi0.267 rad/s = 1.677 rad/s. The real cutoff frequency in rad/s is 2pi400 rad/s = 2513.27 rad/s.

Question: Compute the period N, where x(n) = x(n+N) for x(n) = cos((pi/7)n) cos((3pi/20)n) + sin((pi/15)n) sin((pi/60)n) + sin((47pi/2500)n).

Answer: Each term within the cosine and sine functions gives us a frequency associated with the signal. In order to compute the period of x(n), we need to compute the period of each signal component and take their least common multiplier (LCM). The periods for the cosine and sine terms are respectively 7, 20, 15, 60 and 2500. Hence their LCM would be 42000. So, N=42000.

Question: Let g(n) be the unit impulse, and let b(n) = u(n) be the unit step function. Let z(n) = b(-2 + n) u(-n + 3) 3g(n - 1). What value is z(-4) + z(-3) + …..+ z(2) + z(3)?

Answer: Zero. This function is only non-zero when n = 1, due to its impulse component. Since 1 is not included within the given sum, the sum of z from -4 to 3 is zero.

Question: An analog signal possesses a bandwidth of 12 kHz, is passed through an anti-aliasing filter with a cut-off of 9 kHz, and then sampled at 50% above the Nyquist rate by an ADC using N bits per sample. If the ADC o/p is 280kbps, what is N?

Answer: First, we calculate the Nyquist rate, which is 2 \* 12 kHz = 24 kHz. Then add 50% to this rate to get the actual sampling rate: 24 kHz \* 1.50 = 36 kHz. The rate at which bits are being transferred is 280 kbps. Therefore, N = (280 kbps) / (36 k samples per second) = 7.78 ≈ 8 bits per sample.

Question: Calculate the period N (i.e., x(n) = x(n + N)) for x(n) = sin(0.8 pi n) + cos((9pi/45)n).

Answer: The period of each component is the reciprocal of their frequency. The period of the first term, sin(0.8 pi n), is 2/0.8 = 2.5 RDD (rounded to 3 due to use of integers). The second term, cos((9pi/45)n), has a period of 2/(9/45) = 10. So, the minimum period of the signal, N, will be the least common multiple (LCM) of these two periods, N = LCM(3, 10) = 30.

Question: MLK:An analog signal possessing a bandwidth of 7 kHz, is passed through an anti-aliasing filter with a cut-off of 6 kHz, then sampled at 20% above the Nyquist rate by an ADC using N bits per sample. If the ADC o/p is 378kbps, what is N?

Answer: First, we calculate the Nyquist rate, which is 2 \* 7 kHz = 14 kHz. Then add 20% to this rate to get the actual sampling rate: 14 kHz \* 1.20 = 16.8 kHz. The rate at which bits are being transferred is 378 kbps. Therefore, N = (378 kbps) / (16.8 k samples per second) = 22.5 ≈ 23 bits per sample.

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