MENG

1. You have access to a cheap analog low-pass filter which has a fairly flat unit gain response from $\Omega=0$ to $\Omega=60$ *KHz*, and a stopband attenuation of *100 dB* for $\Omega>80$ *KHz*. With this analog anti-aliasing filter you are asked to design a 16-bit A/D converter system with a baseband sampling rate at *60 KHz*.

a) If you were to use the above analog filter for the design of an oversampling A/D converter system, what is the **minimum** sampling frequency that you should choose?

b) With the sampling rate that you picked in part (a), draw a block diagram of the A/D converter system from the analog input signal to the final sampled output sequence at *60 KHz*. This block diagram needs to include the specification (passband bandwidth, stopband attenuation, gain, etc.) of any filter that would be needed in the system.

c) Is your answer in part (b) the most hardware-efficient solution? If not, use another sampling rate to design this A/D converter system with less amount of computation. Draw a block diagram to describe this system.

a): The minimum sampling rate is 30Khz + 80Khz = 110 Khz

b): The block diagram consists of the above anti-aliasing filter, a sampler at 110Khz, followed by an up-converter by a factor of 6, a digital filter, and a down-converter by a factor of 11. The digital filter specs are: passband, $\pi/11$; stopband attenuation: 100dB, gain: 6.

c): 120Khz.

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2. Indicate whether the following statements are true or false. If the statement is true, give a brief justification. If the statement is false, give a simple counter example or a clear reason.

   a) If a real-coefficient digital filter has a zero-phase frequency response, then it must be a non-causal filter.

   True.

   b) All periodic continuous-time signals will remain periodic after sampling.

   False. The sampled signal is periodic only if the ratio of the period of the original signal and the sampling period is a rational number.

   c) If a z-transform doesn't have a region of convergence on the Z-plane, then its time-domain sequence doesn't exist.

   False. Many time-domain sequences don't have a ROC in their Z-transform.

   d) The sum of the impulse responses of two minimum-phase filters is always the impulse response of another minimum-phase filter.

   False. The sum of the impulses responses will have a transfer function that is the sum of the transfer functions of the two original impulse responses. The poles of this transfer function will remain the same. The zeros of this transfer function, however, will not be the same and may move out of the unit circle.

   e) If N samples of the discrete-time Fourier transform of a discrete-time sequence $h(n)$ are taken at $2\pi k/N$, where $k=0,\ldots,N-1$, then this set of samples represents the N-point discrete-Fourier transform of $h(n)$.

   False. The N samples of the DTFT of $h(n)$ are the DFT of the "aliased" $h(n)$.

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1) What is the relationship between the Laplace transform and the Fourier transform of a continuous-time signal?

2) What is the relationship between the Z transform and the discrete-time Fourier transform (DTFT) of a discrete-time signal?

3) What is the relationship between the discrete Fourier transform (DFT) and DTFT of a finite-length discrete-time signal?

4) If an analog sinusoidal signal is input to a digital spectrum analyzer, which samples the analog signal and performs a DFT of the sampled sequence to display its frequency content, what do you expect to see at the display of the spectrum analyzer?

5) If an analog signal $x(t)$ has been sampled according to the Nyquist criterion to generate $x(n)$ where the samples are taken at $t = nT$, can you use a digital filter to generate $y(n)$ from $x(n)$, y(n) being the sequence as if it had been generated by sampling $x(t)$ at $t = nT+ 0.67T$?

1. For each of the following systems, determine whether the system is stable, causal, linear, and time invariant:

(a) $T(x(n)) = (\cos(\pi n))x(n)$

(b) $T(x(n)) = x(n^2)$

(c) $T(x(n)) = x(n)\sum_{k=0}^{\infty} \delta(n-k)$

(d) $T(x(n)) = \sum_{k=n-1}^{\infty} x(k)$

(e) $T(x(n)) = n^3 x(n)$

a	Stable	Causal	Linear	Time-Invariant
b				
c				
d				
d				

2. If the Nyquist rate for $x_a(t)$ is W_s, what is the Nyquist rate for each of the following signals that are derived from $x_a(t)$?

(a) $\dfrac{dx_a(t)}{dt}$

(b) $x_a(2t)$

(c) $x_a^2(t)$

(d) $x_a(t)\cos(W_0 t)$

3. Consider the system shown in Figure 1, where

$$H(e^{j\omega}) = \begin{cases} 1, & |\omega| < \pi / L, \\ 0, & \pi / L < |\omega| \le \pi. \end{cases}$$

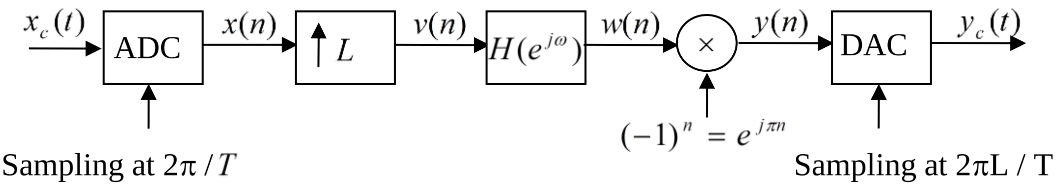

Figure 1

Sketch $Y_c(j\Omega)$ if $X_c(j\Omega)$ is as shown in Figure 2.

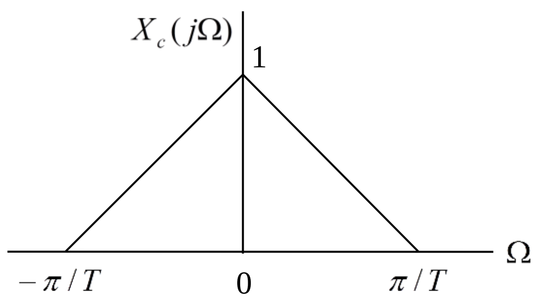

Figure 2