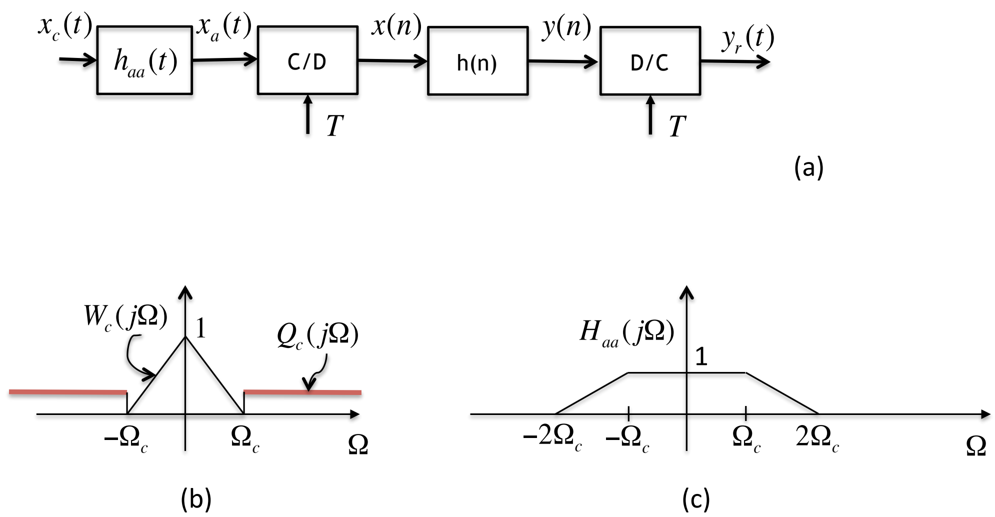
Homework 4 due 10/28 in class

1. Consider the following system. The input $x_c(t) = w_c(t) + q_c(t)$ is a sum of the desired signal $w_c(t)$ and noise $q_c(t)$, where $w_c(t)$ is band limited to $|\Omega| < \Omega_c$ as shown in (b) of the figure. The antialiasing filter $H_{aa}(j\Omega)$ is lowpass as shown in (c) of the figure and $\Omega_s = 4\Omega_c$.

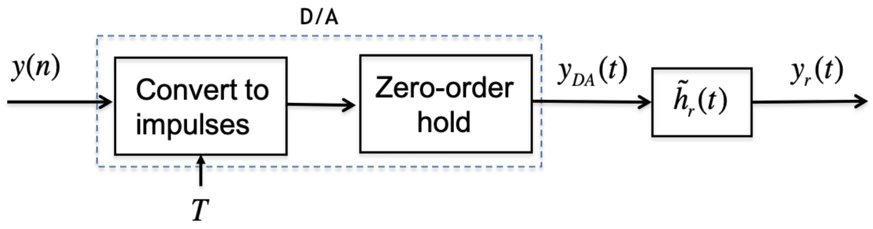


   (a) Plot $X_a(j\Omega)$ and $X(e^{j\omega})$.

   (b) For the given $H_{aa}(j\Omega)$, find the smallest sampling frequency so that $y_r(t) = w_c(t)$. What is $H(e^{j\omega})$ in this case?

   (c) Suppose the sampling frequency is increased to $\Omega_s = 12\Omega_c$. Design $H_{aa}(j\Omega)$ and $H(e^{j\omega})$ so that the transition band of $H_{aa}(j\Omega)$ is as wide as possible and $y_r(t) = w_c(t)$.

2. Consider the D/A converter followed by an LTI system with impulse response $\tilde{h}_r(t) = h_0(t)$, i.e., the zero-order-hold system that is inherently part of the D/A converter.



Suppose the input of the D/A converter is $y(n)$ and

$$y(n) = \begin{cases} n, & 0 \le n \le 4, \\ 0, & \text{otherwise.} \end{cases}$$

(a) Plot $y_{DA}(t)$.

(b) Plot $y_r(t)$. Comment on the difference between $y_{DA}(t)$ and $y_r(t)$ using time-domain and frequency-domain approaches.

3. Design a b-bit uniform quantizer Q for x that is uniformly distributed over the interval $(-1, 1)$. Let $\hat{x}$ be the quantizer output $\hat{x} = Q(x)$.

(a) Suppose $b = 2$. Find the quantization error $e = \hat{x} - x$ when $x = 0.2$ and $x = -0.5$.

(b) Determine the variance of e for $b = 2$.

(c) Choose b so that the signal to noise ratio σ_x^2/σ_e^2 is larger than 96 dB.

4. **MATLAB.** In this assignment, you are asked to record your own voice and perform experiments on the voice signal.

   Note: All Matlab assignments should be accompanied by observations, or comments on why the plots are reasonable. Unexplained plots are not given credits.

   (a) Record your own voice by saying a vowel sound steadily, e.g., 'A' or 'O'. What are the sampling period T and number of bits for each sample in your recording? Indicate the app that you have used to acquire the recording. Try different sampling frequencies to see what is the lowest sampling frequency for sampling your own voice without audible distortion.

   (b) Examine the signal in (a) and plot a segment of your recording that is approximately periodic and identify the period of the discrete sequence. Determine the frequency of the recoding. (You may want to use `plot` for this figure.)

   (c) Find your highest and lowest frequency. (Speak with a frequency that is as high as possible (as low as possible) and repeat (b).)

   (d) Make a recording of a whole sentence. (It is more fun if your recording is a sentence rather than a short vowel sound in this experiment.) Playback the recording using a sampling period that is larger then T. Repeat the experiment using a sampling period smaller than T. Make observations of your experiments for sampling period smaller or larger than T. Reason what you have observed. Justify the results you got using the results in Problem 2 of Homework 3.

Note: The voice can be recorded on MATLAB directly, for which you may find the following commands useful: `audiorecorder, recordblocking, getaudiodata, play`. You can also record the voice using many other options, for example, Smart Voice Recorder (android app, voice recorded in a .wav file), Voice Recorder (iOS, voice recorded in a .m4a file), and Audacity (Windows, Mac OS). When these other options are used, you can load the file using `audioread`. For saving .wav files, use `audiowrite`.