T-61.140 Signal Processing Systems

2nd mid term exam / final exam, Tue 4.5.2004 13-16 M.
You may use a mathematical handbook and graphical calculator. There are formulae on accompanying papers - use them!
2nd mid term exam: Write to the top of the concept “MID TERM EXAM” and **reply to problems 3, 4, 5 and 6.**
Final exam: Write to the top of the concept “FINAL EXAM” and **reply to problems 1, 2, 4, 5, and 6.**

1) (Final exam, 3 x 2p = 6p) Compute or explain clearly.

   a) Discrete-time system is defined by $y[n] = \sqrt{x[n] + x[n-1]}$. Is the system LTI?

   b) LTI-system is defined by its impulse response $h[n] = \left(\frac{1}{3}\right)^{-n+1} u[n-1]$. Is the system stable? Is the system time-invariant?

   c) Consider a discrete-time sequence $x[n] = \cos(\frac{\pi}{3}n) + 2\sin(\frac{\pi}{4}n + \frac{\pi}{2})$. Is $x[n]$ periodic? If it is, what is the fundamendal period N_0?

2) (Final exam, 6p) Consider a discrete-time system depicted in Figure 1. It consists of two components which are connected as shown in (b). The impulse response of the subsystem h_1 is $h_1[n] = \delta[n] - \delta[n-1]$. $h_2[n]$ is unknown. When there is an input $x[n]$ shown in (a), there will be output $y[n]$ shown in (c).

   a) Compute $y_1[n] = h_1[n] * x[n]$.

   b) Determine the values $h[0]$ and $h[1]$.

   c) Determine the impulse response $h_2[n]$ of the other subsystem.

   d) If the input is $x_m[n] = -x[1-n]$, what is the output sequence $y_m[n]$?

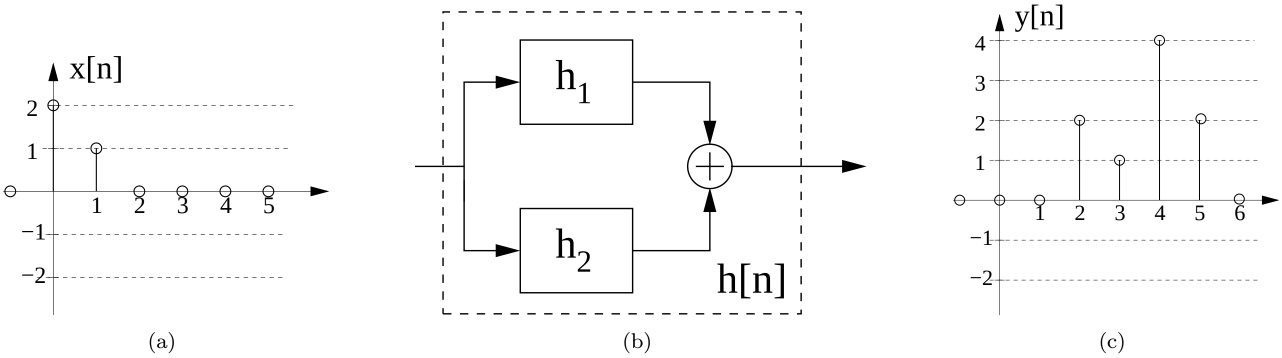


Figure 1: Figures for Problem 2: (a) Input $x[n]$, $x[n] = 0$, when $n < 0,\ n > 1$, (b) LTI-system, (c) Output $y[n]$, $y[n] = 0$, when $n < 2,\ n > 5,\ n \in \mathbf{Z}$.

3) (Mid term exam, 3 x 2p = 6p) Reply to **at most three** statements, if it is true (T) or false (F). Explain briefly.

   a) Any sequence $y[n]$ can be recovered from its amplitude spectrum $|Y(e^{j\omega})|$, if there are enough computational power.

   b) Consider a continuous signal $x(t) = \cos(2\pi f t)$, whose frequency $f = 1$ MHz. Signal is sampled with the sampling frequency of $f_s = 44100$ Hz. Statement: Signal does not alias in the frequency range $0 \ldots f_s/2$ at all.

   c) LTI filter defined by its impulse response $h[n] = \delta[n+2] - \delta[n+1] - \delta[n-1] - \delta[n-2]$ has linear phase response.

   d) Second order LTI system, whose difference equation is $y[n] = 0.3y[n-1] + 0.4y[n-2] + 13x[n]$, can be represented with a parallel connection of two first-order LTI-system: $H(e^{j\omega}) = 8/(1 - 0.8e^{-j\omega}) + 5/(1 + 0.5e^{-j\omega})$.

4) (Final exam/Mid term exam, 6p) The impulse response of a LTI system is

$$h[n] = (-0.8)^n u[n] + (-0.8)^{n-1} u[n-1]$$

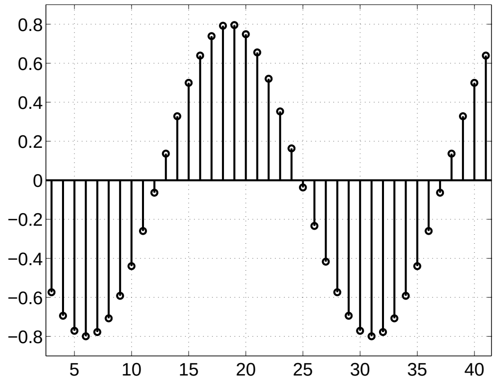
   a) What is the frequency response $H(e^{j\omega}) = Y(e^{j\omega})/X(e^{j\omega})$.

   b) Is the filter FIR or IIR? Is the algorithm recursive or not? What is the order of the filter?

   c) Determine the difference equation for the filter and draw the block (flow) diagram of the filter.

   d) Sketch the amplitude response $|H(e^{j\omega})|$. Is the filter of type lowpass, highpass, bandpass, bandstop or an allpass filter?

5) (Final exam/Mid term exam, 6p) Consider a sequence $x[n]$ in the figure below. It is a sampled version of a continuous signal $x(t)$, which is sampled using the sampling frequency of $f_s = 20$ kHz. X-axis contains the index number of n (not seconds). Sequence $x[n]$ is of form $x[n] = A_1 \cos(2\pi f_1 (n/f_s) + \theta_1)$.



a) What is the period T_s in seconds, in other words, the time between samples?

b) Determine the frequency of the sinusoidal and sketch the spectrum $X(e^{j\omega})$ of a discrete-time sequence $x[n]$ in range $0 \ldots f_s$ Hz.

c) Continuous-time signal $\hat{x}(t)$ is reconstructed from the sequence $x[n]$. Sketch the spectrum $|\hat{X}(j\Omega)|$ in range $0 \ldots f_s$ Hz.

d) What can be said about the original signal $x(t)$, from which the sequence $x[n]$ is a sampled version?

6) (Final exam/Mid term exam, 6p) **Reply to either A or B.**

6A) Write about speech signal and the analysis of that in the context of this course. How can you preprocess the speech signal so that it is easier to use in speech recognition.

6B) Examine the following code which is written in Matlab (and row numbers are inserted in the beginning of each line). It is applied to 2D-signal, which is represented with graylevel values (range 0-255) in the figure (a) below. Part of the pixels are totally white (255) and some are totally black (0).

```
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
01: figure, imshow(A,[0 255]);  % plot the original figure (lily / lilja)
02: number_of_rows = size(A, 1);
03: number_of_cols = size(A, 2);
04: B = zeros(number_of_rows, number_of_cols);  % initialize
05: C = zeros(number_of_rows, number_of_cols);  % initialize
06: for m = 1 : number_of_rows
07:   for n = 1 : number_of_cols-4
08:     temp = A(m, n:n+4);
09:     B(m,n) = mean(temp);
10:     C(m,n) = median(temp);
11:   end;
12: end;
13: figure, imshow(B,[0 255])
14: figure, imshow(C,[0 255])
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
```

Below there are four images. First one (a) is the input signal. Two images of (b-d) are outputs from the code above while one of them is from another operation. Explain what the program does, how does is relate to two images out of three (b-d), and why these two images look like they do. How does this problem relate to LTI filters in this course?