T-61.140 Signal Processing Systems

Summer exam 21st June 2004.

Equipment: pencil, (function) calculator, mathematical formulae. A Fourier transform table is delivered in the exam.

Write down some intermediate phases to your final answer.

1) (6p) Explain briefly with a couple of sentences and give some examples:

   a) What do you mean with a discrete-time LTI?

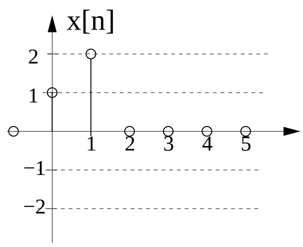
   b) What do you mean with a periodic signal?

2) (6p) Consider a discrete-time LTI-system with the impulse response $h[n]$ below. It consists of two components $h_1[n]$ and $h_2[n]$, which are connected as shown in (b). The impulse response of the subsystem h_1 is known to be $h_1[n] = \delta[n] - \delta[n-1]$. However, $h_2[n]$ is unknown. When an input $x[n] = \delta[n] + 2\delta[n-1]$ like seen in (a) is fed to the system, the output is $y[n] = 2\delta[n-2] + 3\delta[n-3] + 4\delta[n-5]$ like in (c).

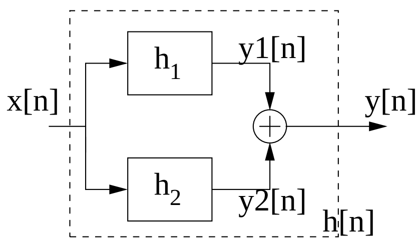
   a) Compute $y_1[n] = h_1[n] * x[n]$.

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

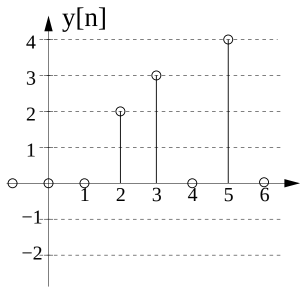
   c) Determine the impulse response $h_2[n]$ of the other subsystem.



(a)

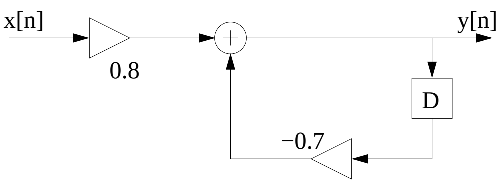


(b)



(c)

3) (6p) Consider a discrete-time system, whose flow (block) diagram is shown below.



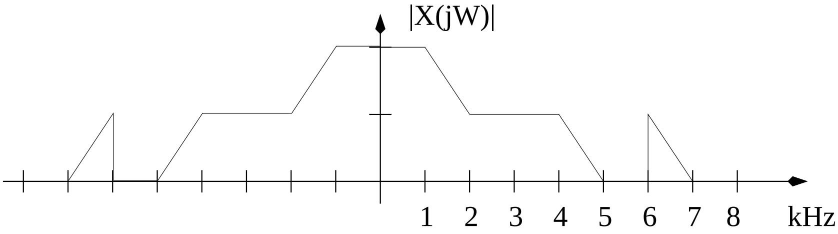
   a) What is the difference equation, i.e. what is $y[n] = \ldots$?

   b) Is the filter FIR or IIR? What is the order of the filter?

   c) Determine the frequency response $H(e^{j\omega}) = Y(e^{j\omega})/X(e^{j\omega})$ of the system.

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

4) (6p) Continuous real signal spectrum $|X(j\Omega)|$ is drawn in the figure below. The highest signal component is 7 kHz. What is the lowest sampling frequency, with which no aliasing occurs? Sketch the spectrum $|X(e^{j\omega})|$ of the discrete-time sequence, when the sampling frequency has been $f_s = 6$ kHz.



5) (6p) **Reply to either A or B.**

5A) The following program read from A/D-converter a sequence (`input_stream`), and manipulates it numerically, and returns it back to D/A-converter (`output_stream`). Disctete-time filter is shown with pseudo code, where read, write and computation use 16 bits per sample:

```
y1 := 0; y2 := 0; x1 := 0; x2 := 0; x3 := 0;   % init
while TRUE {
  x3 := x2; x2 := x1; y2 := y1;
  x1 := read_next_item(input_stream);
  y1 := x1  + 1.902 * x2 + x3 + 0.95 * y2;
  write_item(output_stream, y1);
}
```

a) Write down the difference equation for the filter and draw the flow (block) diagram using the signs from the course.

b) Determine the frequency response $H(e^{j\omega}) = Y(e^{j\omega})/X(e^{j\omega})$ of the filter.

c) What is the filter order?

d) What is the value for the impulse response $h[n]$ at the moment $n = 100$? (Write also some intermediate phases!)

5B) Consider a simple equalizer which can be found in several software programs, like WinAmp below. The frequency area is divided to channels (WinAmp: 60, 170, 310, 600, 1k, 3k, 6k, 12k, 14k, 16k) and it is assumed that the voice is sampled with 44100 Hz. How do you use equalizer and how does it affect voice? Explain, how an equalizer could be constructed using the knowledge from the course?