

## T-61.3010 Digital Signal Processing and Filtering

Mid term exam 1, Sat 6.3.2010 at 10-13, main building.

**You are allowed to do MTE1 only once either 7.3. or 13.3.**

You are not allowed to use any calculators or math reference books. A list of formulas is delivered in the exam. A special form is delivered for Problem 1.

Return a special form and the other answer paper separately. Both ones have to have at least student number and name written on. Problem paper and the formulas you may keep.

Problem 3 is a course feedback which is open from Sat 6-March to Mon 22-March 2010.

- 1) (0-9 p) Multichoice statements. There are 1-4 correct answers, but choose **one and only one**. Fill in **into a separate form**, which will be read optically.

Correct answer +1 p, incorrect -0.5 p, no answer 0 p. You do not need to explain your choices. Reply to as many statements as you want. The maximum points of this problem is 9 and the minimum 0.

- 1.1 Consider a sequence  $x[n] = A_1 \cos(\omega_1 n + \theta_1) + A_2 \cos(\omega_2 n + \theta_2) + A_3 \cos(\omega_3 n + \theta_3)$ , where fundamental periods of each subsequence are  $N_1 = 5$ ,  $N_2 = 8$  and  $N_3 = 10$ , and  $A_i$  are non-zero. What can be said about periodicity of sequence  $x[n]$ ?

- (A) Fundamental period  $N_0$  exists if and only if all phases are zero:  $\theta_1 = 0$ ,  $\theta_2 = 0$ ,  $\theta_3 = 0$
- (B) Fundamental period  $N_0$  exists if and only if all coefficients  $A_i$  are equal
- (C) Fundamental period  $N_0 = 40$
- (D) Fundamental period  $N_0 = 400$

- 1.2 Compute linear convolution  $y[n] = h[n] \otimes x[n]$  of sequences  $x[n] = \delta[n] + 2\delta[n-1] + \delta[n-2] = \{\underline{1}, 2, 1\}$  and  $h[n] = \delta[n] - 2\delta[n-1] = \{\underline{1}, -2\}$ . where underline shows the origin.

- (A) Length of  $y[n]$  is 5
- (B)  $y[n] = 0$ , kun  $n \leq 0$
- (C)  $y[1] = -4$
- (D)  $y[1] = 0$

- 1.3 Two LTI systems  $h_1[n]$  and  $h_2[n]$  in cascade (serial) form the total impulse response  $h[n]$  of the system. We know that  $h_2[n] = \{1, \underline{2}, -1\}$  and  $h[n] = \{-2, -5, \underline{1}, 3, -1\}$ , where underline shows the origin. Hence, the unknown  $h_1[n]$  is of form

- (A)  $h_1[n] = a \cdot \delta[n+2] + b \cdot \delta[n+1] + c \cdot \delta[n] + d \cdot \delta[n-1] + e \cdot \delta[n-2]$
- (B)  $h_1[n] = b \cdot \delta[n+1] + c \cdot \delta[n] + d \cdot \delta[n-1]$
- (C)  $h_1[n] = d \cdot \delta[n-1] + e \cdot \delta[n-2] + f \cdot \delta[n-3]$
- (D)  $h_1[n]$  is a causal filter

where  $\{a, b, c, d, e, f\} \in \mathbb{R}$  and non-zero.

- 1.4 Impulse response of a LTI system is  $h[n] = \sum_{k=0}^{\infty} (-1)^k \delta[n-3k]$

- (A) The length of the impulse response is infinite
- (B) Filter is not stable
- (C) Filter is not causal
- (D) Corresponding difference equation is  $y[n] = x[n] - y[n-3]$

- 1.5 Based on properties of discrete systems, what can be said about the system in Figure 1?

- (A) It is a FIR filter
- (B) It is a linear and time-invariant system
- (C) It is a stable filter
- (D) It is a causal filter

- 1.6 We know a band-limited spectrum  $|X(j\Omega)|$  of an analog real-valued signal  $x(t)$ , see Figure 2(a). The signal is sampled with sampling frequency  $f_T = 10000$  Hz.

- (A) The spectrum  $|X(e^{j\omega})|$  of the sampled sequence in range  $[0, f_T/2]$  is in Figure 3(a). (y-axis values propotional.)

- (A) The spectrum  $|X(e^{j\omega})|$  of the sampled sequence in range  $[0, f_T/2]$  is in Figure 3(b). (y-axis values propotional.)

- (C) The obtained sequence  $x[n]$  is a sinusoidal of form  $x[n] = \cos(\omega_0 n + \theta)$ , where  $\omega_0 = 2\pi(f_0/f_T)$  is normalized fundamental angular frequency

- (D) All those frequency components, whose period  $T_i$  is shorter than  $2/f_T$  seconds, fold (alias) to lower frequency in range  $[0, f_T/2]$  Hz of the digital spectrum  $|X(e^{j\omega})|$

1.7 The transfer function of the filter is

$$H(z) = \frac{1 + (0.2 - 0.4j)z^{-1}}{1 - 0.8z^{-1}} \cdot \frac{1 + (0.2 + 0.4j)z^{-1}}{1 + 0.9z^{-1}} \cdot \frac{1}{1 - 0.7z^{-1}}, \quad |z| > 0.9$$

- (A) The pole-zero diagram is in Figure 4(a)
- (B) The magnitude response is in Figure 4(b)
- (C) The order of the filter is 5
- (D) The filter has a linear phase response

1.8 Discrete Fourier transform (DFT) of a sequence  $x_1[n] = \{\underline{1}, 2, 2, 1\}$  is

$$X_1[k] = \sum_{n=0}^3 x_1[n]W_N^{nk} = \{\underline{6}, -1 - j, 0, -1 + j\}$$

and correspondingly for  $x_2[n]$  there are  $x_2[n] = \{\underline{1}, 1, 0, 0\}$  and  $X_2[k] = \{\underline{2}, 1 - j, 0, 1 + j\}$ . Compute DFT  $X_3[k]$  of a sequence  $x_3[n] = x_1[n] + 2x_2[n]$ . (DFT can be found in formula table.)

	$k =$	0	1	2	3
(A)	$X_3[k] =$	10	$1 - 3j$	$2j$	$2 + 2j$
(B)	$X_3[k] =$	10	$2 - 2j$	3	$2 + 2j$
(C)	$X_3[k] =$	10	$1 - 3j$	0	$1 + 3j$
(D)	$X_3[k] =$	10	$3 - j$	1	$-1 + 3j$

1.9 Consider a LTI filter whose transfer function is

$$H(z) = 1 + z^{-8}$$

- (A) It is a comb filter
- (B) Zeros of the filter are at  $d_1 = +j$  and  $d_2 = -j$
- (C) The length of impulse response  $h[n]$  is 8
- (D) The group delay is  $\tau(\omega) = 8$

1.10 Read a file into Matlab with command `[x, fT] = wavread('kiisseli.wav');`. There is an audio signal  $x[n]$  with sampling frequency  $f_T = 22050$  Hz. It is fed into a LTI system with impulse response

$$h[n] = \sum_{k=0}^9 \frac{10 - k}{50} \cdot \delta[n - k]$$

and an output  $y[n]$  is obtained with command `y = conv(h, x);`

- (A) The filter can filter out noise at 50 Hz
- (B) The filter produces audible “echo effect”
- (C) It is a linear-phase filter, and therefore there is no phase distortion
- (D) None of above holds

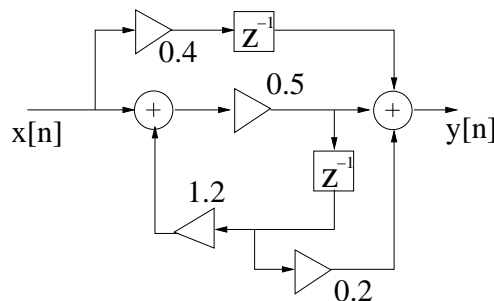


Figure 1: Statement 1.5: Flow diagram of the filter.

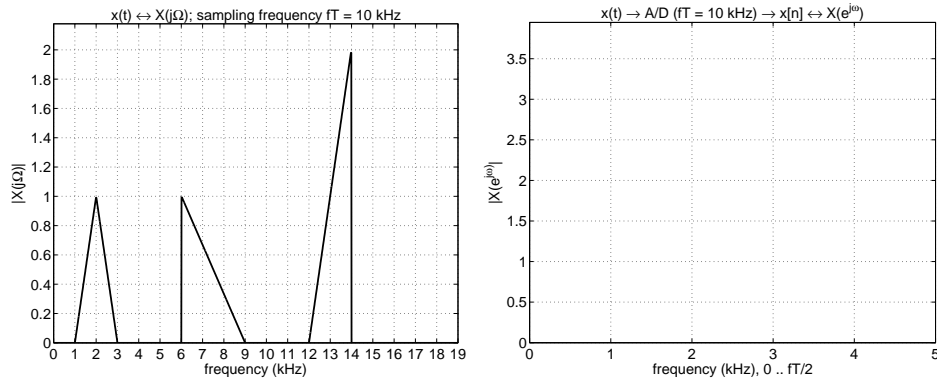


Figure 2: Statement 1.6: (a) Spectrum  $|X(j\Omega)|$  of analog signal  $x(t)$ , (b) empty axis  $f \in [0, f_T/2]$  for sketching.

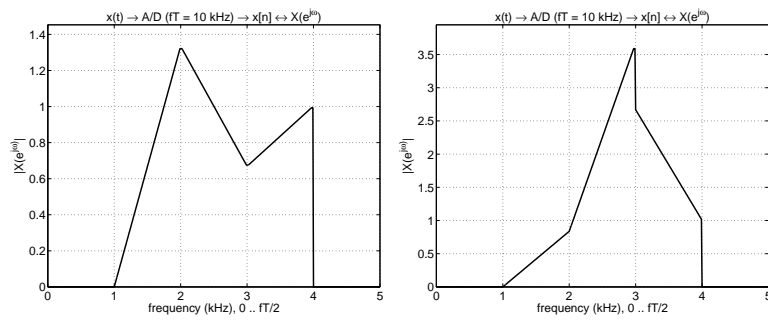


Figure 3: Statement 1.6: (a) option (A) , (b) option (B) .

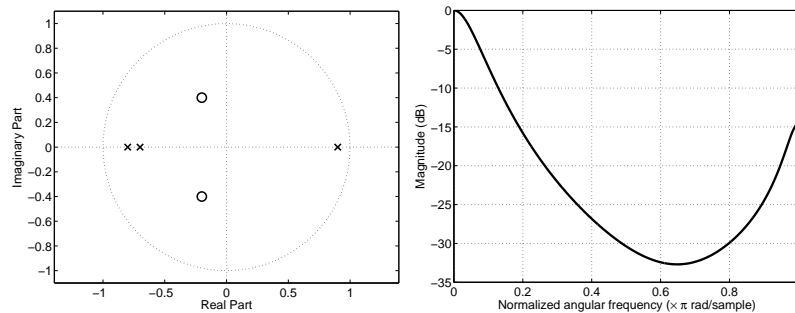


Figure 4: Statement 1.7: (a) option (A) , (b) option (B) .

- 2) (6 p) Consider a discrete-time linear and time-invariant system, whose impulse response is

$$h[n] = 4 \cdot (-0.8)^n \mu[n] - 3 \cdot (-0.6)^n \mu[n]$$

Examine the filter and its behavior with tools given in the course. Write down the facts as clearly as possible.

- 3) (1 p) Course feedback. Questionnaire [http://www.cis.hut.fi/Opinnot/T-61.3010/VK1\\_K2010/kyselyVK1\\_en.shtml](http://www.cis.hut.fi/Opinnot/T-61.3010/VK1_K2010/kyselyVK1_en.shtml) is open till 22-Mar 2010.