

T-61.3010 Digital Signal Processing and Filtering

2nd mid term exam / final exam, Tue 16th May 2006 at 16-19. Halls A-G (non-Finnish).

If you are doing the **2nd mid term exam**, reply to problems **1-3**. You are allowed to do 2nd MTE **only once either 9.5. or 16.5.**

If you are doing the **final exam**, reply to problems **2-5**. You are allowed to do final exam **only once either 9.5. or 16.5.**

Write down on first paper, if you are doing **mid term exam or final exam!**

You are not allowed to use any math table books of your own. A table of formulas is delivered as well as a form for multichoice problem (either 1 or 5). A (graphical) calculator is allowed.

Start a new task from a **new page**. Write all **intermediate steps**.

- 1) (**ONLY 2nd MTE**, 14 x 1p, max 12 p) Multichoice. There are 1-4 correct answers in the statements, but choose **one and only one**. Fill in your solutions in a specific **form**.

Correct answer +1 p, wrong answer -0.5 p, no answer 0 p. You do not need write why you chose your option. Reply to as many as you want. The maximum number of points is 12, and the minimum 0.

- 1.1 The filter in Figure 1(a):

- (A) is of allpass type
 (B) is of FIR type
 (C) the transfer function is $H(z) = \frac{0.8+1.2z^{-1}+z^{-2}}{1-1.2z^{-1}-0.8z^{-2}}$
 (D) is canonic with respect to delays

- 1.2 Zero-phase filter:

- (A) is not causal
 (B) is not stable
 (C) the group delay is the same as the order of the filter divided by two
 (D) does not delay any frequency components

- 1.3 The filter in Figure 2(a) where LTI systems $H_1(z) = 1 + 0.5z^{-1}$ and $H_2(z) = 0.5 + z^{-1}$. The output of the filter in Matlab can be computed for an input sequence x with the command $y = \text{filter}(B, 1, x)$; where

- (A) $B = [1 \ 0.5 \ 0.5 \ 1]$
 (B) $B = [1.5 \ 1.5]$
 (C) $B = [1 \ 1.5 \ 0.5]$
 (D) $B = \text{conv}([1 \ 0.5], [0.5 \ 1])$

- 1.4 Consider a block diagram in Figure 2(b), where three subsystems $S(z)$, $R(z)$ and $T(z)$ are LTI. The transfer function of the filter is

- (A) $H(z) = \frac{R(z)T(z)}{1-R(z)S(z)}$
 (B) $H(z) = \frac{T(z)R(z)}{1-S(z)}$
 (C) $H(z) = \frac{T(z)}{1+R(z)S(z)}$
 (D) $H(z) = \frac{T(z)R(z)+S(z)R(z)}{S(z)}$

- 1.5 Consider a block diagram in Figure 1(b), where coefficients a_1 , a_2 , k_1 and k_2 .

- (A) It is impossible to construct such a filter physically
 (B) The filter can be represented as a Direct Form I with feedback
 (C) The filter is of type FIR
 (D) The output sequence $y[n]$ is $y[n] = (a_2 - k_2 a_1)x[n-1] + a_1 x[n-2]$

- 1.6 In Matlab the command `ellip` computes the coefficients of an elliptic filter. What to type in Matlab in order to get more information?

- (A) `more ellip`
 (B) `man ellip`
 (C) `help ellip`
 (D) `info ellip`

- 1.7 The variables A , B , fT have been computed "correctly" in Matlab. If the phase response is wanted, what should be typed?

- (A) `tf2sos(A, B, fT)`
 (B) `freqz(B, A, 1024, fT)`
 (C) `[phase] = zplane(A, B)`
 (D) `impz(B, A, [], fT)`

- 1.8 In Matlab one wants to design an elliptic lowpass, whose passband ends at 3000 Hz and stopband starts from 4000 Hz. The sampling frequency is 20000 Hz. For the command `ellipord` one has to normalize frequencies for Matlab. The correct command is:

- (A) `[N, Wn] = ellipord(0.15, 0.2, 1, 40);`
 (B) `[N, Wn] = ellipord(3000, 4000, 20000);`
 (C) `[N, Wn] = ellipord(0.3, 0.4, 1, 40);`
 (D) `[N, Wn] = ellipord(2*3000*pi, 2*4000*pi, 1, 40, 'low');`

- 1.9 The value of Hamming window $w[n]$ at $n = 5$, when the length of the window is 11, is

- (A) $w[5] = 0.08$
 (B) $w[5] = 0.54$
 (C) $w[5] = 5/11$
 (D) $w[5] = \omega_c/\pi$, where ω_c is the cut-off of the ideal filter

- 1.10 "Decimation in time" (DIT) algorithm

- (A) computes DFT according to its definition
 (B) uses memory efficiently ("in-place computation")
 (C) uses bit-reversed ordering
 (D) is implemented as a multirate system with decimation filter

- 1.11 Effects of finite wordlength can be caused by

- (A) quantizing filter coefficients
 (B) rounding multiplication results
 (C) truncating multiplication results
 (D) quantizing analog input

- 1.12 Poles of a second-order IIR filter are at $p = 0.99995 \cdot e^{\pm j0.25\pi}$ in order to get a tight bandpass at $\omega = 0.25\pi$, see Figure 2(c), which is plotted with Matlab.

- (A) There would be no problems, if one could operate with infinite accuracy (wordlength)
 (B) In practice the filter becomes easily unstable
 (C) The impulse response attenuates towards zero very slowly
 (D) The position of poles remains the same even if the magnitude response changes with finite accuracy

- 1.13 Periodic sequence ($N_0 = 6$) $x[n] = \{\dots, 3, 8, 4, 2, 9, -1, \dots\}$ is fed into digital system $x[n] \rightarrow \downarrow 2 \rightarrow \uparrow 2 \rightarrow y[n]$. What comes out?

- (A) the same sequence $y[n] = x[n]$
 (B) sequence $y[n]$ but with half length
 (C) $y[n] = \{\dots, 3, 0, 4, 0, 9, 0, \dots\}$
 (D) $y[n] = 2x[n]$

- 1.14 The sampling frequency of a digital sequence is to be increased to $(4/3)$ of the original. When having proper "anti-alias" and "anti-imaging" filters $H_i(z)$, what is the correct way to do this?

- (A) $x[n] \rightarrow \uparrow 4 \rightarrow H(z) \rightarrow \downarrow 3 \rightarrow y[n]$
 (B) $x[n] \rightarrow \uparrow 3 \rightarrow H(z) \rightarrow \downarrow 4 \rightarrow y[n]$
 (C) $x[n] \rightarrow \downarrow 4 \rightarrow H(z) \rightarrow \uparrow 3 \rightarrow y[n]$
 (D) $x[n] \rightarrow \downarrow 3 \rightarrow H(z) \rightarrow \uparrow 4 \rightarrow y[n]$

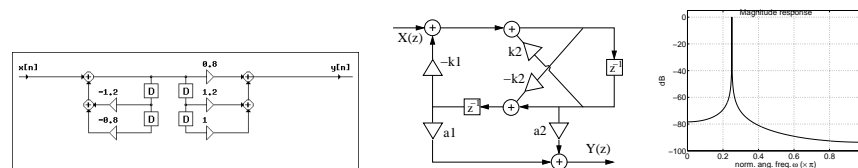


Figure 1: (a), (b) and (c): Figures for multichoice statements 1.1, 1.5 and 1.12.

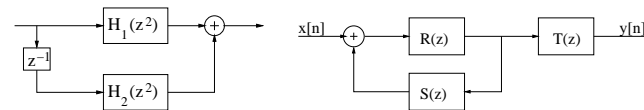


Figure 2: (a) and (b): Figures for multichoice statements 1.3 and 1.4.

- 2) (**MTE and FINAL EXAM**, 6 p) Design a digital IIR filter based on analog first-order Butterworth highpass filter

$$H_{HP}(s) = \frac{s}{s + \Omega_c}$$

and using bilinear transform with prewarping.

The filter has to be highpass, whose cut-off frequency is $f_c = 6000$ Hz, when the sampling frequency is $f_T = 16000$ Hz. Write down the transfer function $H(z)$ so that the maximum of the filter is scaled to one. Draw the pole-zero diagram of the filter.

- 3) (**MTE and FINAL EXAM**, 6 p) Consider a filter $H(z)$, whose magnitude response is plotted in Figure 3 and whose poles are $p = -0.7 \pm 0.2j$ and zeros $z = 0.9 \pm 0.3j$.

- (A) Write down the transfer function $H(z)$ with help of zeros and poles. You can omit the scaling factor K .
 (B) Draw the block (flow) diagram.
 (C) Replace each delay register with three times delay (each z^{-1} is replaced by z^{-3} , i.e. for impulse response $h[n] \rightarrow \uparrow 3 \rightarrow h_{L=3}[n]$). Sketch the magnitude response of the new filter $H(z^3)$ in range $0 \dots \pi$.

- 4) (**ONLY FINAL EXAM**, 6 p) A 4-length input $x[n]$ was fed into a linear, time-invariant, stable and causal discrete-time filter, and a finite-length output $y[n]$ was received:

n	-1	0	1	2	3	4	5	6
$x[n]$	0	1	0	-2	1	0	0	0
$y[n]$	0	0	?	?	?	5	-1	0

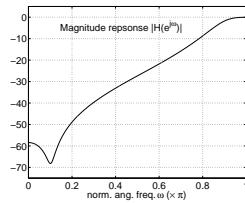


Figure 3: Magnitude response of the filter in range of $0 \dots \pi$ in Problem 3.

- a) Determine the impulse response $h[n]$ of the system using input $x[n]$ and output $y[n]$, when it is known that initial values are zeros and $h[n]$ is of form $(a, b, c, d, \text{ and } e \text{ constants})$:

$$h[n] = \begin{cases} a, & \text{when } n < 0 \\ b, & \text{when } n = 0 \\ c, & \text{when } n = 1 \\ d, & \text{when } n = 2 \\ e, & \text{when } n > 2 \end{cases}$$

- b) Compute the missing values of $y[n]$.

- 5) **(ONLY FINAL EXAM, 14 x 1p, max 12 p)** Multichoice. There are 1-4 correct answers in the statements, but choose **one and only one**. Fill in your solutions in a specific **form**.

Correct answer +1 p, wrong answer -0.5 p, no answer 0 p. You do not need write why you chose your option. Reply to as many as you want. The maximum number of points is 12, and the minimum 0.

- 5.1 Spectrogram can be used when analyzing speech signal.

- (A) In x-axis there is frequency and in y-axis time
 (B) Spectrogram visualizes signal waveform as a function of time
 (C) A spectrogram can be plotted by computing Short-Time Fourier Transform (STFT) and drawing the spectrum of each window as a function of time
 (D) Color (or gray scale) of spectrogram represents number of zeros in the signal

- 5.2 The fundamental period N_0 of sequence $x[n] = \cos(0.5\pi n + \pi/21) + 2\sin(0.5\pi n)$

- (A) $N_0 = 2$
 (B) $N_0 = 4$
 (C) $N_0 = 8$
 (D) $N_0 = 42$

- 5.3 Sequence $x[n] = \sin(0.5n + \pi/4)$

- (A) is not periodic
 (B) has the fundamental period $N_0 = 4$
 (C) has the fundamental period $N_0 = 8$
 (D) has the fundamental period $N_0 = 4\pi$

- 5.4 LTI filter whose impulse response is $h[n] = K \cdot (-1)^{n+2} \mu[n-2]$, where K is constant

- (A) is causal
 (B) is not causal when $K < 0$
 (C) is stable, if $0 < K < \infty$
 (D) is not stable

- 5.5 Discrete-time filter whose input-output relation is $y[n] = (n+2) \cdot x[n]$

- (A) is causal

- (B) is linear

- (C) is time-invariant

- (D) can be implemented using LTI system blocks (sum, delay, multiplication by constant)

- 5.6 In order to avoid aliasing in the sampling process, the sampling period T_s has to be

- (A) at least ten times larger than the highest frequency component in the signal
 (B) at least two times as long as the fundamental period T_0 of the highest frequency component
 (C) at most half of the fundamental period T_0 of the highest frequency component
 (D) the same as the fundamental period T_0 of that component which has the longest fundamental period

- 5.7 Discrete-time Fourier transform $X(e^{j\omega})$ of sequence $x[n] = 0.3^{n-3} \mu[n-3]$ is

- (A) $e^{j3\omega} (1/(1+0.3e^{-j\omega}))$
 (B) $e^{j3\omega} (1/(1+0.3e^{j\omega}))$
 (C) $e^{-j3\omega} (1/(1-0.3e^{-j\omega}))$
 (D) $e^{-j3\omega} (1/(1-0.3e^{j\omega}))$

- 5.8 The pole-zero diagram of a LTI system $H(z)$ in Figure 4(a) corresponds the magnitude response

- (A) in Figure 5(a)
 (B) in Figure 5(b)
 (C) in Figure 5(c)
 (D) in Figure 5(d)

- 5.9 Magnitude response of a LTI filter $H(z)$ in Figure 4(b) corresponds the pole-zero diagram

- (A) in Figure 6(a)
 (B) in Figure 6(b)
 (C) in Figure 6(c)
 (D) in Figure 6(d)

- 5.10 The frequency response of a filter is $H(e^{j\omega}) = 1 + 0.1e^{-8j\omega}$

- (A) Impulse response is 8 samples long

- (B) Phase response is linear

- (C) Filter is lowpass filter even if the curve of magnitude response does not decrease monotonically

- (D) Zeros of the filter are on the same circle with equal intervals

- 5.11 Examine a transfer function $H(z) = (1 + 0.5z^{-1})/(1 - 1.7z^{-1} + 0.72z^{-2})$.

- (A) Choosing region of convergence $|z| < 0.8$ filter is stable and causal

- (B) The order of the filter is 3

- (C) All poles of the filter are inside the unit circle

- (D) Filter is of highpass type

- 5.12 Signal quantization $x[n] \rightarrow \boxed{Q} \rightarrow Q\{x[n]\}$.

- (A) Quantization error $\epsilon[n]$ increases, when more and more bits are used in number representation

- (B) Quantization of a bit sequence can be done by rounding or truncation

- (C) In quantization the length of bit sequence grows to double

- (D) Does not cause any changes into signal spectrum

- 5.13 Consider a filter, whose magnitude response in Figure 1(c) has a peak at $\omega = 0.25\pi$. The sampling frequency f_T is decreased to half: $h[n] \rightarrow \boxed{\downarrow 2} \rightarrow h_{M=2}[n]$

- (A) If the original $f_T = 2$ kHz, then the peak moves to the frequency $f = 0.5$ kHz

- (B) If the original $f_T = 2$ kHz, then the peak moves to the frequency $f = 0.125$ kHz

- (C) Peak moves to the normalized angular frequency $\omega = 0.5\pi$

- (D) Peak moves to the normalized angular frequency $\omega = 0.125\pi$

- 5.14 Sampling frequency of a digital sequence is wanted to be increased to $(4/3)$ of the original. If proper anti-alias and anti-imaging filters $H_i(z)$ are available, what is the correct order?

- (A) $x[n] \rightarrow \boxed{\uparrow 4} \rightarrow \boxed{H(z)} \rightarrow \boxed{\downarrow 3} \rightarrow y[n]$

- (B) $x[n] \rightarrow \boxed{\uparrow 3} \rightarrow \boxed{H(z)} \rightarrow \boxed{\downarrow 4} \rightarrow y[n]$

- (C) $x[n] \rightarrow \boxed{\downarrow 4} \rightarrow \boxed{H(z)} \rightarrow \boxed{\uparrow 3} \rightarrow y[n]$

- (D) $x[n] \rightarrow \boxed{\downarrow 3} \rightarrow \boxed{H(z)} \rightarrow \boxed{\uparrow 4} \rightarrow y[n]$

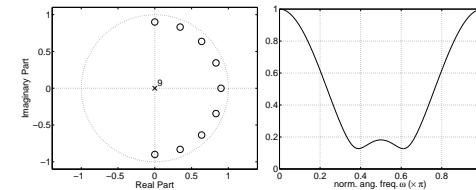


Figure 4: (a) and (b): Figures for multichoice statements 5.8 and 5.9.

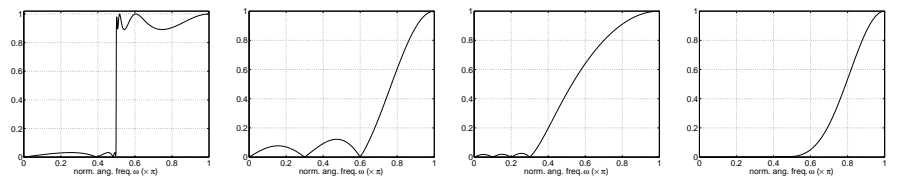


Figure 5: (a), (b), (c) and (d): Figures for multichoice statements 5.8.

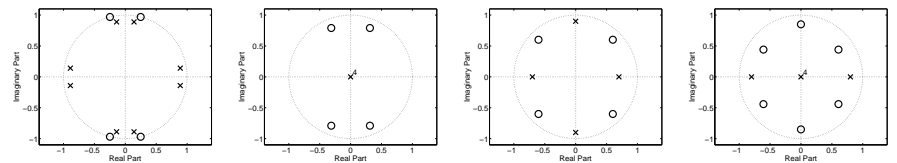


Figure 6: (a), (b), (c) and (d): Figures for multichoice statements 5.9.