2.11 Statements

1. V0001] Are the following statements true or false? A right answer gives +0.5 points, no answer 0 points, and a wrong answer -0.5 points. The total point amount for this problem is, however, between 0–2 points.

   a) All causal and stable discrete-time systems are LTI systems.

   b) If the poles of a second-order filter $H(z)$ are at $z = \frac{1}{r}e^{\pm j\theta}$ and zeros at $z = re^{\pm j\theta}$, $r > 1$, the filter always has linear phase.

   c) If all the poles of a causal filter $H(z)$ are inside the unit circle, the filter is always also stable.

   d) An Nth order IIR filter always has N poles outside the origin.

   e) The length of the inpulse response of the filter in Figure 56 is infinite.

   f) A comb filter is an example of a FIR-type lowpass filter.

2. V0002] Väitteitä. Oikein +1p, väärin = -1p, ei vastausta 0p. Vastaa niin moneen kuin haluat. Tehtävän minimipistemäärä 0, maksimi 6.

   a) Kanoonisen suotimen viiverekisterien määrä on enintään suotimen asteluvun määrä.

   b) Impulssi-invarianttimenetelmässä digitaalisen suotimen impulssivaste $h[n]$ saadaan näytteistämällä analogisen suotimen impulssivastetta $h(t)$.

   c) Bilineaarimenetelmän heikkona puolena on laskostumisilmiö, jos analoginen suodin ei ole kaistarajoitettu.

   d) FIR-suotimien ikkunamenetelmässä suorakulmaisen (sk) ikkunan $(w_{sk}[n] = 1, \text{kun } -M \leq n \leq M,\ w_{sk}[n] = 0, \text{kun } n < -M, n > M)$ hyvänä puolena on sen hyvä taajuuserottelu (kapea pääkupu), mutta heikkona puolena suuri värähtely (vähän vaimenevat sivukuvut).

   e) Gibbsin ilmiön värähtely voidaan poistaa pidentämällä suorakulmaisen ikkunan $w_{sk}[n]$ pituutta, mutta tällöin taajuuserottelu huononee.

   f) FFT-algoritmin laskutoimitusten määrä (yleinen kompleksisuus) on $O(N\, log_2 N)$ ja DFT:ssä $O(N^2)$. Väite: Kun muunnettavan sekvenssin pituus on $N = 1024 = 2^{10}$, niin FFT on yli 1000 tehokkaampi kuin DFT yleisillä kompleksisuuksilla laskettuna.

   g) CD-tason 16 bitin näytetarkkuudella signaalissa on 44100 mahdollista kvantisointitasoa.

   h) Toimivassa suotimessa $H(z) = \frac{F(z)}{1-F(z)G(z)}$ takaisinkytketyn silmukan $G(z)$ voi olla viiveetön, kunhan $F(z)$:ssa on viiverekisterejä.

3. V0003] Are the following statements right or wrong? A right answer gives +1 point, no answer 0 points, and a wrong answer -0.5 points. However, the point minimum for this problem is still zero.

   a) An allpass filter is always linear phase.

   b) The order of the filter $H(z) = \frac{1-2z^{-1}+z^{-2}}{1+0.5z^{-1}}$ is three.

   c) Suppose f_s is the sampling frequency. All frequencies in the range $f_s \ldots 3f_s/2$ are aliased to the range $0 \ldots f_s/2$.

   d) Aliasing does not occur if the highest frequency component in a signal is less than twice the sampling frequency.

   e) The FFT (Fast Fourier Transform) algorithm approximates DFT (Discrete Fourier Transform) and gives equal results only with an infinite sampling frequency.

   f) Truncating a discrete signal (input sequence) e.g. using the windowing technique causes distortion in the spectrum of the signal.

4. V0004] Väitteitä. Oikein +1p, väärin = -1p, ei vastausta 0p. Vastaa niin moneen kuin haluat. Tehtävän minimipistemäärä 0, maksimi 6.

   a) Jos LTI-järjestelmän impulssivasteelle pätee $\sum_{n=-\infty}^{+\infty} |h[n]| < \infty$, on suodin silloin kausaalinen.

   b) CD-tason näytteenottotaajuudella 44100 Hz ei enää tapahdu lainkaan signaalin laskostumista.

   c) Amplitudimodulaatio voidaan toteuttaa LTI-järjestelmällä.

   d) Jos kaksi sarjaan kytkettyä diskreettiaikaista järjestelmää, joiden impulssivasteet ovat $h_1[n]$ ja $h_2[n]$, ovat molemmat lineaarisia ja kausaalisia, niiden keskinäinen järjestys voidaan vaihtaa niin, että lopputulos pysyy samana.

   e) Diskreeteille lukujonoille $g_1[n] = cos(0.8\pi n)$ ja $g_2[n] = cos(3.2\pi n)$ pätee $g_1[n] = g_2[n]$ kaikilla arvoilla n.

   f) Siirfofunktion $H(z) = Y(z)/X(z)$ osoittajapolynomin kertoimia kutsutaan nolliksi ja nimittäjäpolynomin kertoimia navoiksi.

   g) Jos suotimen $H(e^{j\omega})$ vaihevasteen kulmakerroin on vakio, niin myös suotimen ryhmäviive on vakio.

   h) $H(z) = 0.5(1+z^{-1})$ on esimerkki yksinkertaisesta alipäästösuotimesta.

5. V0005] Are the following statements true or false? A right answer gives +0.5 points, no answer 0 points, and a wrong answer -0.5 points. The total point amount for this problem is, however, between 0–3 points.

a) The feedback loop $G(z)$ in filter $H(z) = \frac{F(z)}{1-F(z)G(z)}$ may be delay-free if $F(z)$ contains delay registers. HUONO!

b) In the impulse-invariant method, the impulse response $h[n]$ is obtained by sampling the impulse response $h(t)$ of the corresponding analog filter.

c) A weakness of the bilinear transform is that aliasing occurs if the original analog filter is not band-limited.

d) The Gibbs phenomenon in the FIR window method can be removed by increasing the length of the rectangular window $w_{rec}[n]$, but at the same time frequency resolution is reduced.

e) The numbers of computation steps (general complexities) in the FFT and DFT algorithms are $O(N\,log_2 N)$ and $O(N^2)$, respectively. Statement: When the length of the sequence to be transformed is $N = 1024 = 2^{10}$, the FFT is over 1000 times more effective than DFT (when computed with the above general complexities).

f) With CD (compact disc) level sampling rate and 16-bit word length, the signal may contain 44100 different quantization levels.

g) Truncating a discrete signal (input sequence) using e.g. the window $w_{rec}[n]$ causes distortion to the spectrum of the signal.

h) By rounding a value to the nearest quantization level, the expected value of the quantization error is zero.

6. V0006] Are the following statements true or false? A right answer gives +0.5 points, no answer 0 points, and a wrong answer -0.5 points. The total point amount for this problem is, however, between 0–3 points.

a) Structures in which the coefficient values (or their opposite numbers) can be directly found in the flow diagram are called direct form structures.

b) In a linear-phase filter, we can utilize cofficient symmetry and thus reduce the number of multiplications needed in the implementation.

c) In the impulse-invariant method, the impulse response $h[n]$ of the digital IIR filter can be obtained from the coefficients of the corresponding analog filter $H_a(s)$.

d) A nonlinear mapping (frequency distortion) of the frequency axis happens in bilinear transform. Statement: If the desired cutoff frequency of a digital lowpass filter is very close to one half of the sampling frequency, the significance of the frequency distortion is reduced and, in practical implementations, compensating it by prewarping can be neglected.

e) The numbers of computation steps (general complexities) in the FFT and DFT algorithms are $O(N\, log_2 N)$ and $O(N^2)$, respectively. Statement: When the length of the sequence to be transformed is $N = 128 = 2^7$, the FFT is over 10 times more effective than DFT (when computed with the above general complexities).

f) Truncating a discrete signal (input sequence) with a Hann window $w_{Hn}[n]$ has no effect in the spectrum of the signal.

g) Using CD (compact disc) level 16-bit word lengths, the signal may contain 44100 different quantization levels.

h) In a FIR structure, quantizing signals immediately after multiplication blocks reduces the variance of the total error when compared to only one quantization block in the end of the filter.

7. V0007] Are the following statements right or wrong? Correct answer: +0.5 p, no answer: 0 p, wrong answer: -0.5 p; the point total is still between 0 and 2 points.

a) An allpass filter always has linear phase.

b) The order of the filter $H(z) = \frac{p_0+p_1z^{-1}+p_2z^{-2}}{1+d_1z^{-1}+d_2z^{-2}}$ is 4.

c) Gibbs phenomenon refers to the oscillatory behavior of the magnitude responses of FIR filters which can be reduced by selecting a suitable window function.

d) Quantizing the coefficients of an IIR filter causes error which can affect the stability of the filter.

d e) Quantizing the coefficients of a FIR filter causes error which can affect the stability of the filter.

8. V0008] Are the following statements right or wrong? (A right answer gives +1 points, a wrong answer −1 points, no answer 0 points. The minimum is still 0 points and the maximum 3 points.)

a) When converting an analog IIR type low pass filter to a digital filter using the bilinear transform, the order of the filter almost always increases.

b) With the impulse-invariant method, there is a one-to one mapping from the frequency axis of the s-plane to the unit circle in the z-plane.

c) If we use a Hamming window with length 25 to design a linear phase FIR filter, the group delay is always 12.

d) By scaling a filter $H(z)$ with $\max\{|KH(z)|\} = 1$, we can prevent overflowing and improve the signal-to-noise ratio.

9. V0009] Are the following statements false or true? (A correct answer: +1p, a wrong answer: -1p, no answer: 0p. The minimum points of the problem is 0p and maximum 3p.)

   a) In the inverse bilinear transform the unit circle in the z-domain maps one-to-one into the frequency axis of s-domain.

   b) The discrete Fourier-transform calculated with FFT-algoritm is not so accurate as calculated by definition (DFT), but with N big enough the difference is not significant.

   c) The phase response of a FIR filter can be non-linear, which implies that the group delay $\tau(\omega) = -d\theta(\omega)/d\omega$ is not constant.

   d) In the FIR filter design, the length of the windowing function defines always also the length of the filter.

10. V0010] Are the following statements true or false? (Correct answer: +1p, no answer 0p, wrong answer: –1p.)

    (a) multiplication in the frequency domain corresponds to convolution in the time domain

    (b) a causal filter is stable only if its zeroes lie inside the unit circle

    (c) stable filters are always causal

    (d) if f_s is the sampling frequency, the frequencies on the interval $f_s/2 \dots f_s$ are aliased on the interval $0 \dots f_s/2$

    (e) discrete Fourier transform corresponds to z-transform on the imaginary axis of the z-plane

    (f) The implementation of FIR filter cannot have feedback loops (no recursion). Otherwise the impulse response would'nt be finite.

11. V0011] Are the following statements true or false? (Correct answer: +1p, no answer 0p, wrong answer: –1p.)

    (a) FIR filters have always linear phase responses.

    (b) Elliptic IIR filter has sharper transition bands compared with filters of same order designed with other IIR filter design algorithms.

    (c) The scaling of signal in order to suppress the overflows in a filter also gives a better signal-to-noise ratio in the filter.

    (d) Bilinear transform causes folding of magnitude response in the digitalization of analog filters.

    (e) Combination of cascaded FIR and IIR filter is never stable.

    (f) Hanning-window (in spectrum analysis) has better suppression of the sidelobes and narrower mainlobe than the rectangular window.