QUIZ
March 19th, 2020
Documents allowed
First name: **Full name:**

For each question, there's only one good answer,
- ***1 good answer = 2 pt***
- ***1 wrong answer = -1 pt***
- ***No answer = 0 pt***

1. An audio signal, which spectrum is limited to $20\ kHz$ was digitalized at a sampling frequency $F_s = 24\ kHz$. The reconstruction of the original analog signal from the DAC output is:
   a. Possible after applying an ideal rectangular filter over $[-20\ kHz, 20\ kHz]$
   b. Possible after applying an ideal rectangular filter over $[-12\ kHz\ to\ 12\ kHz]$
   c. Possible after applying an ideal rectangular filter over $[-24\ kHz\ to\ 24\ kHz]$
   d. Impossible, as we don't respect the Shannon sampling theorem

2. We digitalize a real valued analog signal, which spectrum belongs to the frequency band $[140\ MHz\ 160\ MHz]$. Which sampling frequency allows the down sampling of the signal while respecting the Shannon sampling theorem?
   a. $30\ MHz$
   b. $50\ MHz$
   c. $70\ MHz$
   d. $40\ MHz$

3. We consider the digitalized signal $x(n)$ defined as: $x(n) = 1$ for $n \in \{0,1\}$ and $x(n) = 0$ for $n \in \{2,3,4,5\}$. The k^{th} sample of the Discrete Fourier Transform of this signal is equal to:
   a. $2cos\left(\frac{\pi k}{6}\right)e^{-\frac{j\pi k}{6}}$
   b. $cos\left(\frac{\pi k}{6}\right)e^{-\frac{j\pi k}{6}}$
   c. $2cos\left(\frac{\pi k}{2}\right)e^{-\frac{j\pi k}{2}}$
   d. $cos\left(\frac{\pi k}{2}\right)e^{-\frac{j\pi k}{2}}$

4. We consider the digitalized signal $x(n)$ defined as: $x(n) = (-1)^{n+1}$ for $n \in \{0,1,2,3\}$. The Z-transform of this signal is equal to:

   a. $1 + z^{-1} + z^{-2} + z^{-3}$
   b. $1 - z^{-1} + z^{-2} - z^{-3}$
   c. $-1 + z^{-1} - z^{-2} + z^{-3}$
   d. $1 - z + z - z^3$

5. We consider a digital signal $x(nT_s)$ with a sampling frequency $F_s = 512\ Hz$. We calculate the Discrete Fourier Transform (DFT) on $N = 256$ samples. The last sample of the DFT corresponds to the frequency:
   a. f = 256 Hz
   b. f = 250 Hz
   c. f = 512 Hz
   d. f = 510 Hz

6. We consider the following digital filter: $H(z) = \frac{1-z^{-2}}{1+z^{-2}}$. This filter has:
    a. Two poles $P_0 = e^{j\pi/2}$ et $P_0^* = e^{-j\pi/2}$ and two zeros $Z_1 = 1$ et $Z_2 = -1$
    b. Two poles $P_0 = e^{j\pi/4}$ et $P_0^* = e^{-j\pi/4}$ and two zeros $Z_0 = e^{j\pi/2}$ et $Z_0^* = e^{-j\pi/2}$
    c. Two poles $P_0 = e^{j\pi/2}$ et $P_0^* = e^{-j\pi/2}$ and two zeros at the origin
    d. Two zeros $Z_0 = e^{j\pi/2}$ et $Z_0^* = e^{-j\pi/2}$ and no poles

7. The impulse response of a FIR filter is given by $h(0) = 1,\ h(1) = 0\ and\ h(2) = -1$ and $h(n) = 0\ if\ n > 3$. Over the frequency band $[0, F_e/2]$, this filter is:
    a. An all pass
    b. A band-pass
    c. A low-pass
    d. A high-pass

8. The transfer function of an IIR filter is given by $H(z) = \frac{z^2+1}{z^2+1.96}$. This filter is
    a. Non-causal
    b. Stable
    c. Unstable
    d. A FIR filter

9. A white, zero mean, Gaussian noise $s(n)$ having a normalized variance is filtered with a FIR filter $H(z) = 1 + z^{-1} + z^{-2}$. The signal at the output of this filter is augmented with another white zero mean, Gaussian noise $w(n)$ having a variance equal to σ^2. The observed signal $y(n)$ is equal to:
    a. $s(n) + s(n-1) + s(n-2)$
    b. $s(n) + w(n)$
    c. $s(n) + s(n-1) + s(n-2) + w(n)$
    d. $s(n) + s(n-1) + s(n-2) + w(n) + w(n-1) + w(n-2)$

10. A white, zero mean, Gaussian noise having a normalized variance is filtered by a FIR filter $H(z) = 1 + z^{-1}$. The signal at the output of this filter:
    a. Is white
    b. Have a non-zero mean value
    c. Have a variance equal to 2
    d. Have a variance equal to 1