QUIZ
March 19th, 2020
Documents allowed
Answers

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.~~ $\cancel{s(n) + s(n-1) + s(n-2)}$
   - ~~b.~~ $\cancel{s(n) + w(n)}$
   - c. $s(n) + s(n-1) + s(n-2) + w(n)$
   - ~~d.~~ $\cancel{s(n) + s(n-1) + s(n-2) + w(n) + w(n-1) + w(n-2)}$

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~~