Audio and Video Signals (AVS): *Module 1 – Audio Signals (AS)*	*Written test of November 4, 2023*
Name: **ID (Personal Code):**	

Please answer **thoughtfully**, **pertinently** and **briefly.**

Q1. [8] Let us consider a teleconferencing system, where the received signal v(n) is reproduced by a stereo loudspeaker system. We want to design an echo cancellation system based on Wiener filtering to reduce the echo components captured by the microphone and obtain an estimate of the talker signal u(n) and other noise sources s(n) present in the room.

a) Sketch the block diagram of the echo cancellation system and briefly comment it.

b) Explain the equation of orthogonality principle and derive the Wiener-Hopf equations formula.

c) The room impulse responses of the loudspeakers have been estimated as follows:

$$H_1(z) = 0.2 + 0.5z^{-1},$$
$$H_2(z) = 0.8 + 0.3z^{-1},$$

and for simplicity we assume the signals $\mathrm{v(n), s(n), u(n)}$ as real-valued uncorrelated zero mean white noises whose variances are $\sigma_s^2 = 0.2, \sigma_v^2 = 0.4, \sigma_u^2 = 0.5$. Compute the optimum filter $W_o(z)$ with $M = 2$ taps.

d) Write the filter output signal.

Q2. [4] Consider the analysis of a digital audio signal consisting of a sum of sinusoids. Your goal is to estimate the number of sinusoids by means of a spectral analysis performed frame-by-frame.

a) Explain which are the main limitations and constraints in relation to the used window function given that you know which are the possible frequencies of the sinusoids.

b) Briefly explain the difference between the concept of frequency resolution and accuracy.

Q3. [4] Consider the problem of applying some effects to a digital audio signal.

a) Define the concept of time-variant fractional delay line.

b) Explain why an interpolation step may be needed to implement such delay line.

c) Report the block diagram of the Dattoro industry standard scheme, explaining the role of each block.

Q4. [7] Each answer can be either TRUE or FALSE. Incorrect answers give you -1 points, unanswered 0 points, correct answers +1 point. Provide a brief optional comment when in doubt.

a) [T] [F] The just noticeable difference (JND) describes the ability of distinguishing different tones played together.

b) [T] [F] By whitening a signal through LPC, we always obtain a white residual independently from the LPC filter order.

c) [T] [F] By applying pitch-scaling it is possible to change the pitch of a tone without affecting its length.

d) [T] [F] Given a fixed SPL, all tones with that SPL have the same loudness for a listener.

e) [T] [F] Considering a 2D environment, different source-receiver configurations are characterized by different RIRs.

f) [T] [F] Consider a harmonic signal with fundamental frequency 100 Hz acquired at 50 kHz. To resolve spectral peaks using a rectangular window (L=2), at least 2000 samples are needed.

g) [T] [F] The median filter is a linear operator that can be used to restore local sound degradations.

Q5. MATLAB [8]

Implement the following steps.

Define some signals:

- [1] Load the audio file `tone.wav` from the working directory. From this file infer the sampling frequency. Assume this file contains a pure tone that lasts for exactly 2 seconds. This is your signal `y_1`.
- [1] Generate a 1-second-long sinusoid with frequency 2000 Hz and amplitude equal to 0.5. This is your signal `y_2`.
- [1] Obtain the signal `y` as the sum of `y_1` and `y_2` (hint: decide any policy to take the signals length into account).
- [1] Plot the first 100 ms of `y` in the time domain correctly expressing the time axis in seconds.

Apply some processing:

- [0.5] Define a 100 ms window of your choice.
- [0.5] Extract one frame from `y` and apply the window thus obtaining `y_w`.
- [1] Compute the zero-crossing rate of `y_w`.
- [1] Compute the spectrum of `y_w`.
- [1] Plot the spectrum in the frequency domain expressing the axis in Hz.