Audio and Video Signals (AVS): *Module 1 – Audio Signals (AS)*	*Written test of June 23, 2023*
Name: **ID (Personal Code):**	

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

Q1. [8] We want to design a noise cancellation system to improve hands free interaction in a van. The main noise is generated by the engine, and you are required to implement a Wiener filter to attenuate the noise.

a) [1] Sketch the block diagram of a noise cancellation system based on Wiener filter and briefly comment it.

b) [3] Compute the optimum filter coefficients assuming the following:

- The signal acquired by the microphone is

$$v(n) = s(n) + 2.3u(n) + 1.6u(n-1)$$

- $s \sim \mathcal{N}(0, \sigma_s^2 = 0.1) \in \mathbb{R}$ represents the speech signal.
- $u \sim \mathcal{N}(0, \sigma_u^2 = 0.2) \in \mathbb{R}$ is the engine noise and $s \perp u$.

In addition, to implement video calls directly using the infotainment, a source localization system is implemented to automatically point a camera to the speakers using a uniform linear array (ULA) composed by 24 microphones.

c) [1.5] Which is the distance between two consecutive microphones to avoid aliasing at 4000 Hz?

d) [1] How long is the entire array? Is it feasible for a van deployment?

e) [1.5] Which is the angular resolution of the array at 500 Hz?

Q2. [4] Consider the problem of time-scaling.

a) Describe the meaning of applying time-scaling to an audio signal.

b) Highlight the difference with respect to pure time-stretching (or resampling).

c) Consider an audio signal composed by the sum of two 3-second-long sinusoids with frequency 400 Hz and 800 Hz. Which signal do you obtain if you apply a time-scaling operation with factor 2?

Q3. [4] Consider the problem of implementing audio effects using the Dattoro industry standard scheme.

a) Explain the concept of a time-variant fractional delay line (what are they? What do they do? How they work? etc.).

b) Explain the role of the interpolator in such a delay line.

c) Draw and comment the Dattoro industry standard scheme, highlighting the delay line.

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 orthogonality principle states that the optimum error of an infinite length Wiener filter is orthogonal with the desired response.

b) [T] [F] Background noise removal can be achieved by means of a filter defined as the ratio between the target signal power spectrum and the corrupted power spectrum.

c) [T] [F] The forward filter in LPC is also known as whitening filter.

d) [T] [F] Pitch-scaling can only be applied to sinusoidal signals.

e) [T] [F] Median filtering can be implemented by means of a convolution.

f) [T] [F] Considering a source emitting in far field with respect to a uniform linear array, the DOA can be estimated starting from the knowledge of the TDOAs at different microphones.

g) [T] [F] In granular synthesis, grains can only be obtained by recording real sounds.

Q5. MATLAB [8] Considering a sampling frequency of 2 kHz, implement the following steps.

Define some signals

- [0.5] Generate a sinusoidal signal `y_1` with the following characteristics: frequency 500 Hz, amplitude 1, phase π/2, duration 1 second.
- [0.5] Generate a sinusoidal signal `y_2` with the following characteristics: frequency 600 Hz, amplitude 0.5, phase 0, duration 0.5 second.
- [0.5] Generate the signal `y` as the sum of `y_1` and `y_2`. Pay attention to the signal's length.

Plot

- [1] Plot `y` in the time domain expressing the time axis in seconds and not in samples.
- [1] Plot the spectrum of `y` expressing the frequency axis in Hz and not in samples.

Windowing

- [0.5] Define a 20 ms rectangular window.
- [0.5] Extract the first frame of 20 ms from `y`, and window it.
- [1] Considering no overlap, compute how many frames you can extract from `y`.

Feature computation

- [1] Compute the zero-crossing rate (ZCR) of `y`.

Filtering

- [0.5] define the three-sample filter [-0.5, 0, 0.5].
- [1] Apply the linear filter to `y` in the frequency domain rather than using convolution. Explain each step.