Audio and Video Signals (AVS): ***Module 1 – Audio Signals (AS)***	*Written test of July 11, 2023*
Name: **ID (Personal Code):**	

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

Q1. [8] You are requested to design and implement an acoustic source localization system satisfying the following requirements:

- At least M=4 microphones
- Possibility to localize sources with minimum angular distance of 16°
- Work in the speech frequency range (300Hz-8kHz)

Assume that the speed of sound in air is approximately 340 m/s, and the microphones are connected to an audio card with a sampling frequency of 96 kHz.

a) [1] Assuming the adoption of a Delay and Sum (DAS) beamformer and a uniform linear array (ULA), compute the distance between two consecutive sensors
b) [0.5] What does it happen if the distance between microphones is shorter than that?
c) [0.5] What does it happen if the distance between microphones is larger than that?
d) [1] Determine the length of the ULA
e) [0.5] What does it happen if the array is shorter than that?
f) [0.5] What does it happen if the array is longer than that?
g) [1] Provide the number of microphones required for the actual implementation.
h) [1] Which is the maximum TDOA (in samples and seconds) that you can measure with this array?
i) [1] Which is a possible location of the source (with respect to the array) that gives you this TDOA?
j) [1] Consider that the last microphone of the array suddenly breaks. What does it change in terms of aliasing and angular resolution (explain the idea without re-computing any parameter)?

Q2. [4] Consider the problem of implementing a delay line.

a) Describe the general concept of delay line applied to an audio signal.
b) Describe the differences among: i) an integer delay line; ii) a fractional delay line; iii) a time-variant fractional delay line.
c) Which kinds of delay lines need an interpolator? Why?

Q3. [4] Consider the problem of synthesizing a reverberated audio signal.

a) Define the concept of Room Impulse Response (RIR).
b) Define the concept of reverberation time T60.
c) Consider a rectangular room with one microphone and one source. Considering only first-order reflections (i.e., after one reflection, the signal does not “bounce” on walls anymore), sketch a possible RIR. *(hint: ignore the floor and the ceiling)*

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 to the input signal.

b) [T] [F] The forward filter in LPC is also known as shaping filter.

c) [T] [F] By applying time-scaling after pitch-scaling to an audio signal, it is possible to obtain a time-stretching (or resampling) effect.

d) [T] [F] A RIR always contains one peak related to the direct path from the source to the receiver.

e) [T] [F] All-pass interpolators attenuate high frequency components.

f) [T] [F] Consider a harmonic signal with fundamental frequency 100 Hz acquired with a sampling frequency of 50 kHz. To resolve spectral peaks using a rectangular window, at least 1000 samples are needed.

g) [T] [F] Consider a uniform linear microphone array and a source that emits a narrow-band signal in far field. To estimate the DOA from the TDOAs, at least 3 microphones are needed.

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

Define some signals

- [0.5] Generate a sinusoidal signal `x` with the following characteristics: frequency 500 Hz, amplitude 2, phase $\pi/2$, duration 0.5 second.
- [0.5] Generate a sinusoidal signal `y` with the following characteristics: frequency 500 Hz, amplitude 0.5, phase 0, duration 0.25 second.
- [0.5] Generate the signal `z` as the sum of `x` and `y`. Pay attention to the signal's length.

Plot

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

Windowing

- [0.5] Define a 0.1 second Hann window.
- [0.5] Plot the window in the time domain expressing the time axis in seconds and NOT in samples.
- [0.5] Extract the first frame of 0.1 second from `z`, and window it.

Feature computation

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

Filtering

- [0.5] define the two-sample filter [0.5, 0.5].
- [0.5] Apply the linear filter to `y` in the time domain.
- [1] Describe how would you tell if the used filter is a low-pass or high-pass filter.