Audio and Video Signals (AVS): ***Module 1 – Audio Signals (AS)***	*Written test of January 22, 2024*
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) [1] Determine the length of the ULA.

c) [1] How many microphones do you actually need?

d) [1] Which is the location of the source (with respect to the array) that gives you the maximum measurable TDOA with this array?

e) [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)?

Consider that now one source emits a pulse in the same environment where the ULA is placed.

f) [1] Sketch which is a possible signal received by two microphones.

g) [1] Given the two signals that you sketched, sketch their cross correlation.

h) [1] Explain how you would measure the TDOA (in seconds and in samples) between the two microphones starting from the sketched cross correlation.

Q2. [4] Consider the problem of synthesizing a reverberated audio signal.

a) Define the concept of Room Impulse Response (RIR) for a source and a receiver within a room.

b) Sketch one example of RIR, defining the regions in which it can be split.

c) Formally define the reverberation time T60.

Q3. [4] Consider the problem of synthesizing audio signals by means of different approaches.

a) Describe how Wavetable Synthesis method works.

b) Describe how Granular Synthesis works.

c) Describe how Additive Synthesis works.

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] Consider the problem of background noise removal by means of Wiener filtering. The Wiener filter tends to attenuate spectral bands in which the power spectrum of the target signal is significantly higher than that of the noise.

b) [T] [F] It is possible to convolve a room impulse response (RIR) with a dry audio signal to introduce reverberation effects.

c) [T] [F] Consider a ULA whose length is 50 cm. It is possible that you measure a TDOS of 100 ms.

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

e) [T] [F] Time and pitch-scaling can be applied independently one from the other.

f) [T] [F] In Linear Predictive Coding, the shaping filter is an all-pole filter.

g) [T] [F] On average, if a sound is emitted with a given sound pressure level (SPL), a person perceives the same loudness independently from the sound frequency.

Q5. <u>MATLAB</u> [8]

Implement the following steps.

Define some signals:

- [1] Consider a sampling frequency of 10 kHz, generate a 2-second-long cosine with frequency 500 Hz and amplitude equal to 1. This is your signal `y_1`.
- [1.5] Generate the signal `y_2` as a 1-second-long train of pulses spaced by 200 samples and with amplitude 0.7.
- [0.5] Obtain the signal `y` as the sum of `y_1` and `y_2`. The two signals have different length, make a choice, and explain it.
- [1] Plot the first 200 ms of `y` in the time domain correctly expressing the time axis in seconds.

Do some processing:

- [1] Apply a Hamming window to the extracted 200 ms of `y`.
- [2] Filter the windowed signal using the filter `h = [0.3, 0.3, 0.3]`. Apply the filtering operation in the frequency domain using a number of samples that is a power of 2.
- [1] Plot the magnitude and phase of the Discrete Fourier Transform of the filtered signal.