Audio and Video Signals (AVS): *Module 1 – Audio Signals (AS)*	*Written test of July 08, 2024*
Name: **ID (Personal Code):**	

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

Q1. [8] A sound localization system is designed based on a Uniform Linear Array (ULA) of M=8 microphones. Consider the situation in which the array is used to record signals coming from two sources approximately on the broadside of the array. The sources emit in the band 2-4kHz. Considering the speed of sound equal to c=340[m/s] and a Delay-and-Sum (DAS) beamformer:

a) Determine the length of the array.

b) Provide the minimum angular separation to localize the two sources.

c) What is the difference between spatial filtering and parametric methods?

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) 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.

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] By applying time-scaling after pitch-scaling to an audio signal, it is possible to obtain a time-stretching (or resampling) effect.
b) [T] [F] Given a fixed SPL, all tones with that SPL have the same loudness for a listener.
c) [T] [F] The orthogonality principle states that the optimum error of an infinite length Wiener filter is orthogonal to the input signal.
d) [T] [F] The median filter is a linear operator that can be implemented by means of convolution.
e) [T] [F] Consider the problem of resolving sinusoidal peaks in the frequency domain. Given a rectangular window and two peaks at 100 and 200 Hz, we need at least 160 window samples if we work at 8 kHz.
f) [T] [F] The just noticeable difference (JND) describes the ability of distinguishing different tones played together.
g) [T] [F] Consider a harmonic sound associated with a pitch. By removing the fundamental, the sensation of pitch is completely lost.

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

Implement the following steps.

Define some signals:

a) [1] Consider a sampling frequency of 16 kHz, generate a 2-second-long cosine function with frequency 1000 Hz and amplitude equal to 1. This is your signal `y_1`.
b) [1] Generate the signal `y_2` as a 1-second-long train of pulses with amplitude 1 and spaced 100 samples one from another.
c) [0.5] Obtain the signal `y` as the sum of `y_1` and `y_2`.
d) [1] Plot the first 200 ms of `y` in the time domain correctly expressing the time axis in seconds.
e) [1] Plot the magnitude of the Discrete Fourier Transform of `y` correctly expressing the frequency axis in Hz.

Do some processing:

f) [0.5] Define the FIR filter `h` composed by the three samples [0.3, 0.3, 0.3].
g) [2] Obtain the signal `y_filt` that consists of `y` filtered with `h`
   - The filtering operation must be implemented in the frequency domain.
   - Work with a number of samples that is a power of two in the frequency domain.
h) [1] Plot the first 200 ms of `y_filt` in the time domain, correctly expressing the time axis in seconds.