Audio and Video Signals (AVS): ***Module 1 – Audio Signals (AS)***	*Written test of February 12, 2024*
Name: **ID (Personal Code):**	

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

Q1. [9] We want to encode a speech signal using Linear Predictive Coding (LPC).

a) Describe the main core idea of LPC and how you can implement LPC solution as a Wiener filter. To do so, provide the data model, draw the block diagram, and briefly explain the overall idea. (Suggestion: call $\hat{s}(n)$ the estimated signal)
b) What is the whitening filter?
c) Provide the definition of the objective function to be minimized.
d) Define the Prediction Gain of LPC and comments its behavior.
e) Using Linear Predictive Coding, we want to estimate the coefficients of an auto regressive model of order 2 that approximates the audio signal $s(n)$ as

$$s(n) = \sum_{k=1}^{2} a_k s(n\text{-}k) + u(n)$$

The estimated values of the auto-correlation function of the signal $s(n)$ are $r(0)=0.5$, $r(1)=0.2$, $r(2)=0.1$. Estimate the optimum (in the MSE sense) coefficient set a_k.

f) Comment on the results obtained in the previous point.

Q2. [4] Consider the problem of implementing audio effects using delay lines.

a) Define what is a delay line, clearly explaining the relation between the input and the output.
b) Explain the main differences among: i) an integer delay line; ii) a fractional delay line; iii) a time-variant fractional delay line.
c) Which one (or ones) may need the use of an interpolator? Why?
d) Report the main effect of linear interpolation on a delayed signal.

Q3. [3] Consider the problem of time and pitch scaling.

a) Describe in your own words what does it mean to apply time scaling to an audio signal.
b) Describe in your own words what does it mean to apply pitch scaling to an audio signal.
c) Explain what the duality principle is in the time/pitch scaling context.

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 a system implementing audio filtering through overlap and add (OLA). It is possible to use triangular windows still honouring the constant overlap and add (COLA) condition.

b) [T] [F] In order to apply reverberation to an audio recording, a room impulse response (RIR) is always necessary.

c) [T] [F] Two tones with different frequency and the same Sound Pressure Level (SPL) provide the same impression of loudness.

d) [T] [F] A median filter of 5 samples can be used to recover up to 3 corrupted samples in a row.

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

f) [T] [F] Consider the problem of resolving sinusoidal peaks in the frequency domain. Given a window with factor L=2 and two peaks at 100 and 300 Hz, we need at least 160 window samples if we work at 8 kHz.

g) [T] [F] In a perfectly anechoic environment (i.e., without reverberation), it is not possible to measure a room impulse response (RIR).

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

Implement the following steps.

Define some signals:

a) [1] Consider a sampling frequency of 8 kHz, generate a 2-second-long cosine function with frequency 250 Hz and amplitude equal to 1. This is your signal `y_1`.
b) [1] Generate the signal `y_2` as a 1-second-long sine function with frequency 100 Hz and amplitude equal to 2. This is your signal `y_2`.
c) [0.5] Obtain the signal `y` as the sum of `y_1` and `y_2` taking care of the length difference.
d) [1] Plot `y` in the time domain correctly expressing the time axis in seconds.

Do some processing:

e) [0.5] Define a Hann window `w` of 200 ms.
f) [0.5] Apply the window `w` to the signal `y` to extract the first 200 ms of it.
g) [1] Compute the zero-crossing rate (ZCR) of the windowed signal.
h) [1] Filter the windowed signal using the filter `h = [-0.5, 0, 0.5]` directly in the time domain.
i) [1.5] Plot the magnitude and phase of the Discrete Fourier Transform of the filtered signal correctly expressing the frequency axis in Hz.