Audio and Video Signals (AVS): ***Module 1 – Audio Signals (AS)***	*Written test of June 17, 2024*
Name: **ID (Personal Code):**	

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

Q1. [9] You are asked to design a compressed coding for speech signal based on linear prediction. At the current time instant, the speech signal can be expressed as

$$s(n) = 0.2s(n-1) + 0.3s(n-2).$$

a) Describe the block diagram of LPC problem providing the name of the filters and their role.

b) Provide the equation for computing the optimum (in the MSE sense) coefficients $a_k, k = 1, \dots, P$ and explain the relationship with the Wiener filter problem.

c) The autocorrelation of the signal has been estimated as

$$r(0) = 0.25, \quad r(1) = 0.15, \quad r(2) = 0.2.$$

Estimate the optimum coefficients of a 2nd order LPC model and write the "whitening filter" formula.

d) Explain why LPC can be considered as a power spectrum envelope matching.

Q2. [4] Consider the analysis of a digital audio signal consisting of a sum of sinusoids. Your goal is to estimate the number of sinusoids by means of a spectral analysis performed frame-by-frame.

a) Explain which are the main limitations and constraints in relation to the used window function, reporting any assumption on the frequency distance among the sinusoids.

b) Briefly explain the difference between the concept of frequency resolution and accuracy.

Q3. [3] Consider the problem of audio reverberation.

a) Explain what a room impulse response (RIR) is with the help of a sketch.

b) Report in which regions a RIR can be split.

c) How can you apply audio reverberation to a dry signal using a RIR?

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). Triangular windows honour the constant overlap and add (COLA) condition with any hop-size.

b) [T] [F] All-pass and linear interpolators introduce less attenuations on sub-sampled signals.

c) [T] [F] Consider an acoustic source in far field and a linear microphone array. The source direction of arrival (DOA) can be estimated from time differences of arrival (TDOAs) at different microphone pairs.

d) [T] [F] The median filter is a linear operator that can be used for audio restoration.

e) [T] [F] Consider an audio synthesis system based on granular synthesis. All sound grains must be recorded from real sounds.

f) [T] [F] MUSIC and Capon are parametric methods for DOA estimation.

g) [T] [F] Time-scaling can be obtained applying pitch-scaling and resampling.

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

Implement the following steps.

Define some signals:

a) [1] Consider a sampling frequency of 8 kHz, generate a 1-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 train of pulses. The amplitude of each pulse is 0.8. The distance between two pulses is equal to the period of `y_1`.

c) [0.5] Obtain the signal `y` as the sum of `y_1` and `y_2`.

d) [1] Plot the first 500 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 a rectangular window `w` of 250 ms.

g) [2] Compute the zero-crossing rate (ZCR) for each 250 ms frame of `y` (not considering any overlap).
   - [0.5] Compute how many frames you can extract with the defined window `w` and correctly define the needed for loop.
   - [0.5] Apply the window `w` to each frame of the signal `y` to extract 250 ms of it.
   - [1] Compute the zero-crossing rate (ZCR) of each frame.

h) [1] Plot the ZCR over time defining a correct time axis in seconds. Motivate your choice for the time axis.