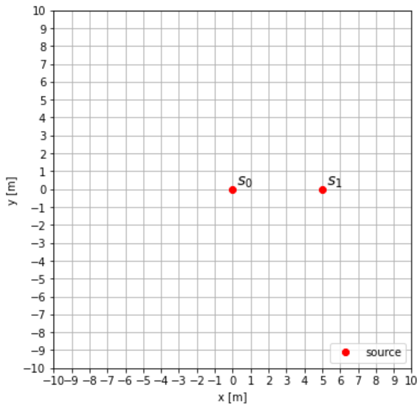
Audio and Video Signals (AVS): *Module 1 – Audio Signals (AS)*	*Written test of September 11, 2023*
Name: **ID (Personal Code):**	

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

Q1. [8] Consider a room containing two sources s_0 and s_1 as shown in the following figure.



In the room, there is a uniform linear microphone array (ULA) composed by three microphones m_0, m_1 and m_2. The recording system connected to the array has a working sampling frequency of 24 kHz.

Consider that s_0 emits a signal with maximum frequency 2 kHz, and minimum frequency 1.5 kHz. Conversely, s_1 emits a signal with maximum frequency 1.8 kHz, and minimum frequency 1 kHz.

a) [1] Which is the length of the microphone array that ensures avoiding spatial aliasing?

b) [1] Knowing that you estimate with the array a direction of arrival (DOA) of 0 degrees for one source, and 45 degrees for the other source, sketch a possible position of the array in the room.

c) [1] Is it possible to distinguish the two sources in terms of direction of arrival if they emit concurrently with the given array?

Consider that the source s_0 emits an impulse at time $t=0$, while s_1 is silent.

d) [1] Sketch the signal obtained at m_0 and m_1, considering the time axis in seconds.

e) [1] Which is the expected TDOA between microphones m_0 and m_1 in seconds?

f) [1] Sketch the GCC measured between microphones m_0 and m_1.

Consider that you can change the array length and / or the number of microphones.

g) [1] What happens to the angular resolution?

h) [1] What happens in terms of spatial aliasing?

Q2. [4] Consider the problem of time and pitch scaling.

a) Define the concept of time scaling and pitch scaling.

b) Consider a two-second sinusoid with frequency 440Hz. What happens if you apply time-scaling with factor 2?

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) How can you apply a RIR to a dry audio recording to obtain a reverberated audio signal?

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 just noticeable difference (JND) describes the ability of distinguishing different tones played together.

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

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

d) [T] [F] Different tones with the same SPL may have different loudness to a human listener.

e) [T] [F] Consider the sequence of samples [1, 1, 1, 5, 1]. If we apply a median filter of size 3, we obtain the sequence [1, 1, 1, 1, 1] (considering repeating the first and last sample as boundary condition).

f) [T] [F] Consider a harmonic signal with fundamental frequency 100 Hz acquired at 50 kHz. To resolve spectral peaks using a rectangular window (L=2), at least 1000 samples are needed.

g) [T] [F] In a perfectly anechoic environment (i.e., without reverberation), the theoretical room impulse response (RIR) is composed of a single impulse.

Q5. MATLAB [8]

Implement the following steps.

Define some signals:

- [1] Consider a sampling frequency of 8 kHz, generate a 1-second-long cosine with frequency 1000 Hz and amplitude equal to 0.5. This is your signal `y_1`.
- Generate the signal `y_2` by
  - [1] Generating a 1-second-long train of pulses spaced by 100 samples and with amplitude 0.7.
  - [1] Filtering the train of pulses with a filter whose transfer function is

$$H(z) = 1 - 1.5z^{-1} + 0.4z^{-2} + 0.2z^{-3} + 0.2z^{-4}$$

- [1] Obtain the signal `y` as the sum of `y_1` and `y_2`.
- [1] Plot the first 100 ms of `y` in the time domain correctly expressing the time axis in seconds.

Apply a filtering operation in the frequency domain on the whole `y` signal:

- [0.5] Define the FIR filter `h` composed by the three samples [0.3, 0.3, 0.3].
- [2] Obtain the signal `y_filt` that consists of `y` filtered with `h`. Implement the filtering operation with a technique of your choice.
- [0.5] Plot the first 50 ms of `y_filt` in the time domain, correctly expressing the time axis in seconds.