Simulation of acoustic echoes (May 2020)

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Abstract*- This document serves as our answer to the SESSION 2 → Simulation of acoustic echoes, for any further clarification on the subject debated in this document please send an email to *esther.martin.cuartero@estudiantat.upc.edu* *joel.delgado@estudiantat.upc.edu

The main objective of this lab session is to simulate and characterize several acoustic effects using MATLAB and some basic tools of digital signal processing. More concretely, the objective is to simulate the acoustic reverberation (i.e., the echoes) in a room

I. INTRODUCTION

To start with, we had to follow some different stages to achieve the objective of the session.

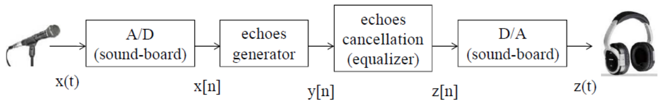

Figure 1. General scheme and block diagram.

As can be seen from "Figure 1", we have a total of four phases (four blocks), we do not count the record of the voice signal and the final listening, so for every block we will have a MATLAB function. In order to increase the understanding of this scheme, we explain a little bit what is the main target of every block:

- **A/D (Analogical to Digital):** digitalization of the analogical signal.
- **Echoes generator:** simulate the effect of the acoustic reverberation in a room.
- **Echoes cancellation:** remove the echoes from the "dirty signal".
- **D/A (Digital to Analogical):** produce the analogical signal from an input digital signal.

The parts of echoes generation and echoes cancellation will be divided in: **simple echo** (just one echo) and **multiple echoes** (more than one echo).

To finish, the idea is to play the voice signal after it has been processed by all the programmed systems.

II. PROCEDURE FOR PAPER SUBMISSION

Along this paper, you will find the codes that correspond on every of the required functions.

A. Analogical to Digital

This function has to convert a signal from the analogical domain to the digital domain. We need a microphone to do this part.

The inputs of this function are the duration of the signal "s" and the sampling frequency "fs". The output will be a vector containing the samples of the digital signal "x".

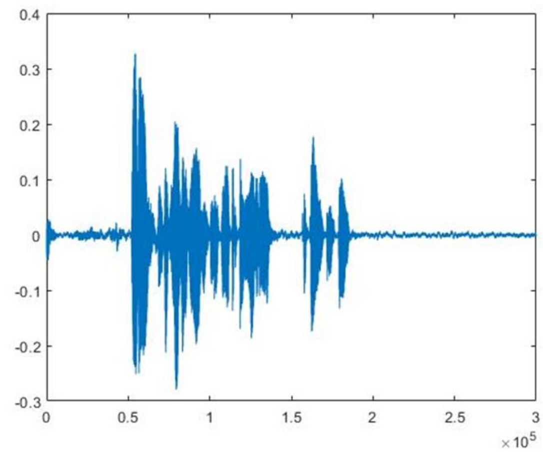

Figure 2. Signal x[n].

"Figure 2" shows the representation of our voice signal in the digital domain.

B. Echo Generation

Now we will simulate the acoustic reverberation. We need to create a function that creates echoes. The inputs of the function are the vector containing the signal samples "x", sampling frequency "fs", amplitude of the echo "α" and the delay of the echo. The output is the vector with the signal samples "y".

B1. Simple Echo

A simple echo will be created due the following equation:

$$y[n]=x[n]+\alpha \cdot x[n-N]$$

.

Where y[n] is the signal with echoes, x[n] is our voice signal, "α" is the amplitude of the echo and "N" is the delay of the echo. In the function, "N" will be the product of the delay and the sampling frequency. We will extract the length of the vector "L", and then get through the vector and introducing the echo.

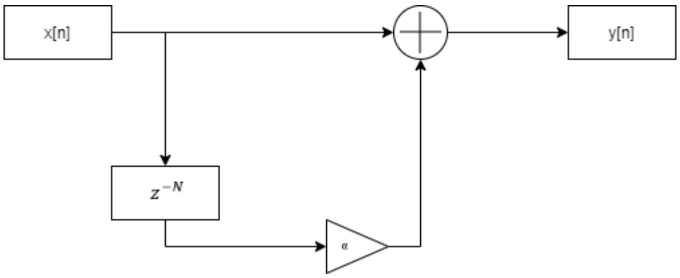

Figure 3. Programming Diagram.

"Figure 3" is the programming diagram for the system.

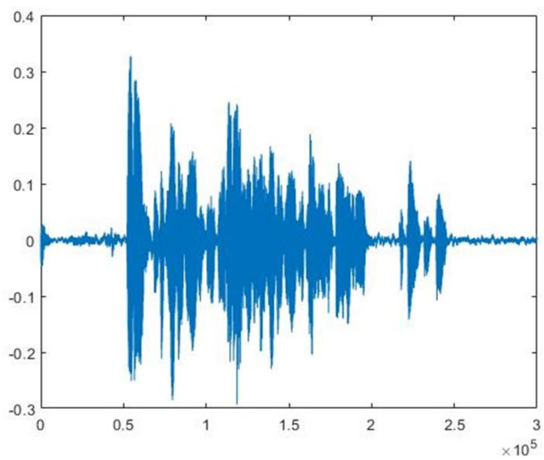

Figure 4. Signal with echo y[n].

This is the representation of the signal after putting it into the B1 function.

B2. Multiple Echoes

In this case, we want to create multiple echoes. We will be able to do it due the following equation:

$$y[n]=x[n]+ \alpha \cdot y[n-N]$$

Where y[n] is the signal with echoes, x[n] is our voice signal, "α" is the amplitude of the echo and "N" is the delay of the echo.

In the function, "N" will be the product of the delay and the sampling frequency. We will extract the length of the vector "L", and then get through the vector and introducing the echo.

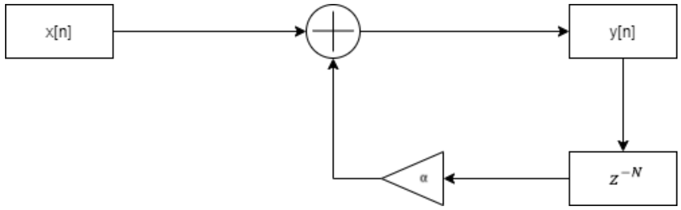

Figure 5. Programming Diagram.

"Figure 5" is the programming diagram for the system.

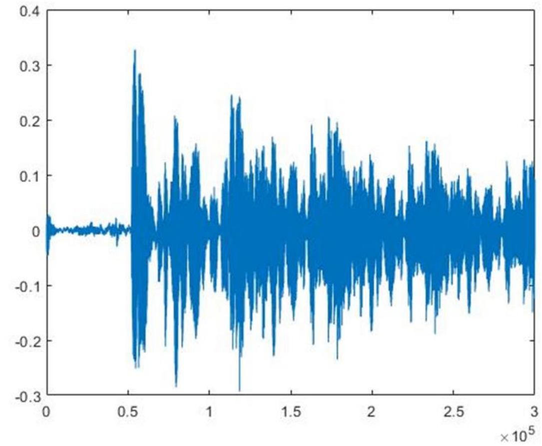

Figure 6. Signal with multiple echoes y[n].

C. Equalizer

At this part of the simulation we need to be able to implement a function that allows us to remove the echoes from the main signal. The inputs of the function are the vector containing the signal samples "y", sampling frequency "fs", amplitude of the echo "α" and the delay of the echo. The output is the vector with the signal samples "z".

C1. Simple Echo Equalizer

The simple echo will be removed due the following equation:

$$z[n]=y[n]- \alpha \cdot z[n-N]$$

Where z[n] is the signal with echoes, y[n] is our voice signal, "α" is the amplitude of the echo and "N" is the delay of the echo.

In the function, "N" will be the product of the delay and the sampling frequency. We will extract the length of the vector "L", and then get through the vector and introducing the echo.

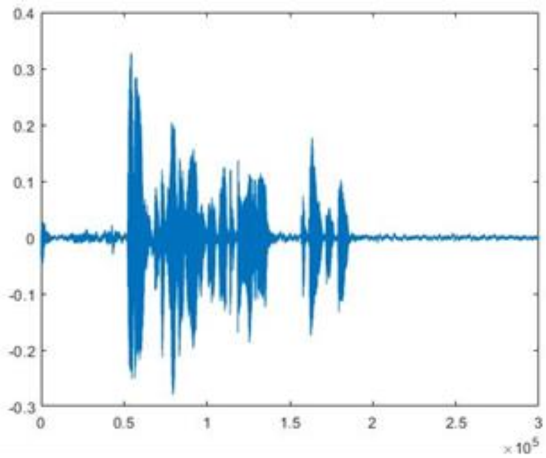

Figure 7 Signal without echo z[n].

This is the representation of the signal after putting it into the C1 function. As we can expect, the signal is exactly the same as x[n].

C2. Multiple Echoes Equalizer

The multiple echoes will be removed due the following equation:

$$z[n]=y[n]-\alpha\cdot y[n-N]$$

Where z[n] is the signal with echoes, x[n] is our voice signal, "α" is the amplitude of the echo and "N" is the delay of the echo.

In the function, "N" will be the product of the delay and the sampling frequency. We will extract the length of the vector "L", and then get through the vector and introducing the echo.

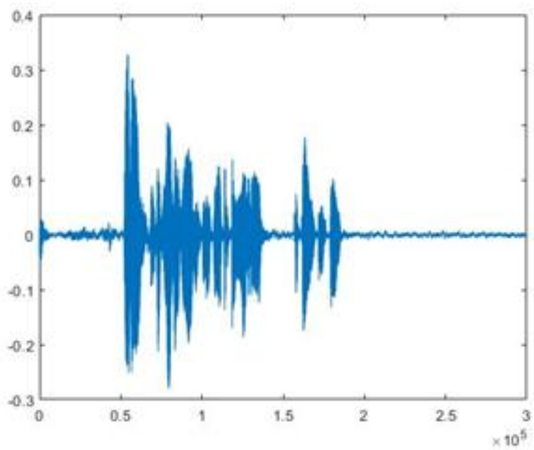

Figure 8. Signal without echo z[n].

This is the representation of the signal after putting it into the C2 function. As we can expect, the signal is exactly as x[n].

III. Units

These are the units we use during the entire lab session:

- "s" in seconds.
- "fs" in Hz.
- Delay in seconds

IV. Some Common Mistakes

Here we have some examples of mistakes we had during the lab session:

- Saving the functions in a folder where the program is not capable to find them.
- To put the delay bigger than the "s", we will not notice the repetition.
- If we repeat words when we are recording the voice, we will not know if it is echo or the original voice.

V. Conclusion

Due to this practice about simulating of acoustic echoes, we have learned how, thanks to transform an analogical signal to digital we are able to change it: adding or cancelling echoes. Before doing the lab session, we had to do a background study for a better understanding of the concepts.

In our opinion, it was interesting to carry out all the knowledge we learned doing the background study and the concept of the first session of MATLAB. Also, in the develop of this practice we enjoyed to play with the voice signal, so it was a practical mode to learn about the subject.

VI. References

[1] Introduction to MATLAB: *https://atenea.upc.edu/pluginfile.php/3123187/mod_resource/content/3/Background_Study_Introduction_to_MATLAB.pdf*

[2] Simulation of acoustic echoes: *https://atenea.upc.edu/pluginfile.php/3123189/mod_resource/content/4/simulation_acoustic_echoes.pdf*

[3] Official web page of MATLAB: *https://es.mathworks.com/help/matlab/getting-started-with-matlab.html*