Simulation of acoustic echoes

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I. Introduction

The objective of this lab session is to simulate and characterize some acoustic effects using matlab and basic tools of digital signal processing. Specifically, the objective is simulating the acoustic reverberation in a room.

During the session we had to create several functions, like simulate a given discrete system described by a time impulse response, simulate a discrete system able to eliminate the effects of the echoes in the signal, the capture of a voice signal using the soundboard of the pc and matlab, the processing of the voice signal using the previously programmed functions. and play the voice signal after it has been processed by the programmed systems.

With these functions we are capable of recreate and analyze a basic chain of signal processing and cleaning.

II. Procedure

A. Analog to digital

First, we had to record our voice, for that we had to create with matlab the function "analog to digital". In this function we must choose the parameters for the recording, like the duration of the audio and the frequency. And before creating the function we had to use "audiorecorder" for create the object that records the voice in the function.

When we finished, we had to copy the function in the command window and for 5 seconds our voice had been saved in the vector x.
This audio will be a variable that we will use in the future functions.

B. Echoes generation

This block simulates the effect of the acoustic reverberation by creating echoes in the signal. The input signal x[n] is the "clean signal" and the output signal y[n] the "dirty signal", the signal with echoes. And there are two different type of functions that we had used: the simple echo and the multiple echoes.

B1. Simple echo

This function has the objective of create a delay in the signal x[n], the number of delays is determined by "N", and with an amplitude alfa.

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

It is a stable (fiir) and causal system, the output is the same input (because we convolve whit a delta) and the same input delayed ant attenuated.

B2. Multiple echo

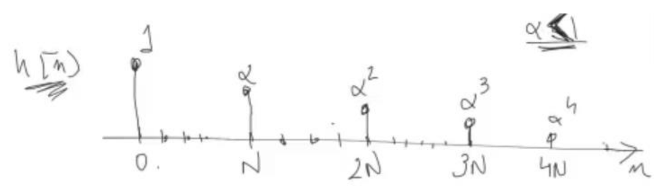
In this function we have an equation with recursion, because it depends on current and past samples of the output implies that it is an iir filter.

The equation of this signal is:

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

Where alfa is the amplitude and N is the delay, like in single echo.

In the matlab program we had to set alfa <1, because for being a stable system we need a decreasing system. If we set an alfa>1 the system would grow and would diverge, therefore it would not be a stable system.



C. Echoes cancellation (equalizer)

The objective of this functions is to remove the echoes from the dirty signal. After, the input signal y[n] will be the dirty signal", and the output signal should be the clean signal z[n]=x[n].

CI. Equalizer for the simple reverberation

The transfer function of the equalizer for the simple echo is the inverse of the simple echo transfer function.

$$H_{eq\,ES}(z) = \frac{1}{1+\alpha z^{-N}}$$

This response transformed in z[n] give us the follow equation:

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

The equalizer of a fiir system becomes an iir system, and we have that the output is equal to the input minus alpha multiplied the same delayed output value N.

CII. Equalizer for the multiple reverberation

This equalizer is very similar to the previous one, because the equalizer Heqme(z) transfer function is the inverse of the multiple echoes transfer function (Hme(z)).

$$H_{eq\,ME}(z) = 1 - \alpha z^{N}$$

Transforming this response, we obtain the following equation:

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

We can see that it is a fiir system because the samples do not depend on the output, and when we convolve the equalizer impulse response with the multiple echoes impulse response, we obtain a delta.

III. Units

The units used in this lab session are:

- alfa (no units only a number)
- delayseg (this is the variable that i used in matlab for the delay in seconds).
- fs is the variable for frecuency in hz.
- recObj is the variable for the object recorded in matlab.
- t is the time in seconds.
- te is the variable for the audiorecorder.

IV. Conclusion

With this practice, we have learned how transform the signal, adding echoes and cancelling echoes to a voice signal recorded by us.

In this lab session we started to get familiar with matlab, using functions and commands. We have also worked with the theoretical part that we have studied during this course, practically seeing, and playing with the signal process.

And this lab report has allowed me to understood better the most theoretical part of the session.