Echoes Generation and Cancellation (November 2023)

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***Abstract*— This document presents a report on the generation of echoes, both multiple and simple, and their elimination by means of an Equalizer System. In addition, the conversion of the signal from Analog to Digital and from Digital to Analog is addressed. We have tried to capture our knowledge in the most rigorous, accurate and clear way possible. We hope it will be of your liking. For any questions or clarifications, we are at your disposal through our emails.**

Keywords— Analogue, Digital, Single Echo, Multiple Echo, Equalizer, Sampling Frequency, Delay.

Throughout the study, we will be using a damping coefficient $\alpha = 0'6$ a sampling frequency of $f_s = 40$ kHz, and a delay of 1s with respect to the original signal.

I. INTRODUCTION

FREQUENTLY, we choose to convert the signal from analog to digital. This process is carried out since digital signals have greater robustness, flexibility and reliability. They are easier to reconfigure, transport, and have a greater capacity for storage and carrying out operations. In addition, they offer a significant reduction in errors, providing some of their advantages compared to the analog signal. [1]

In this report we are going to consider the creation of a signal that we wish to transmit through a communications channel to be heard anywhere. The procedure we will follow is as follows: initially, we will convert the sound emitted by our mouth, which manifests itself as pressure waves, into an electrical signal. This process will be carried out by the microphone. Later, with the analog sound signal we will transform it into a digital signal using an A/D converter that will operate at a certain sampling frequency. Once we have the digital signal, we will send it to its final destination where we will finally convert it from Digital to Analogue to be heard through a speaker.

However, along this route, distortions, modifications to the signal, repetitions, attenuations, among other changes, could have been added. For example, echoes of the same signal could have been added, so we will proceed to eliminate them with the aim of being able to hear the original signal without unwanted interference.

II. CONVERSION FROM ANALOG TO DIGITAL

To convert the signal from analog to digital, we will sample it to be able to work with it later.

The bandwidth of the human voice goes from 50 Hz to 4 kHz, so if we wanted to hear what we recorded we would have enough sampling at 8 kHz. However, the bandwidth of the ear goes from 20 Hz to 20 kHz, so if we want our ear to perceive the audio we have captured with the greatest possible fidelity, with all the frequencies that make up our signal original we will sample at a frequency of about 40 kHz. If we sampled at a frequency lower than 8 kHz, we would start to lose a lot of information.

In general, in telecommunications, a sampling frequency of 44.100 Hz is used to guarantee a sufficient margin, avoiding aliasing and ensuring that we capture the maximum possible frequencies that make up the original signal and avoid information loss.

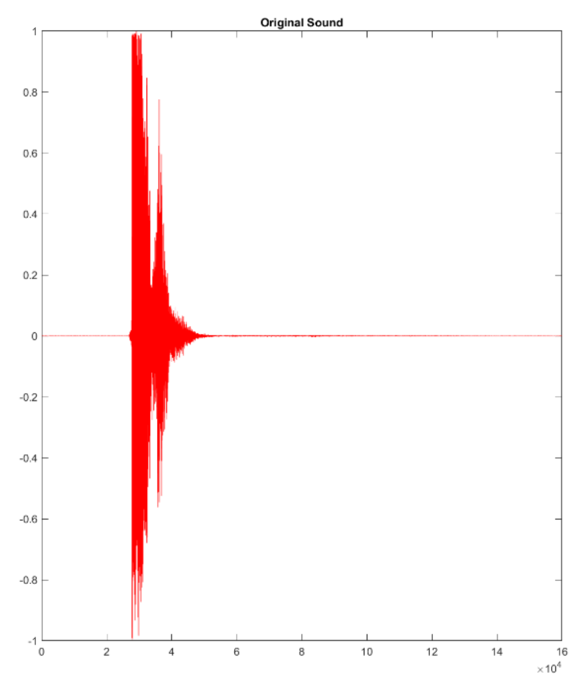

Figure I. Original Digital Signal.

In this image we can see what the original signal $x[n]$ looks like.

III. Echoes Generation

An echo in telecommunications is defined as the repetition of received signals produced by the reflection of radio waves. The distance between the original source of the sound and the reflecting system will determine how long it will take for the echo to return.

For this reason, we can think that our signal has been delayed and damped/attenuated by a factor α, and subsequently, it has been added to the original signal.

A. Simple Echo Generation

The present scenario only considers the addition of a single echo since the signal does not pass through the communications channel again.

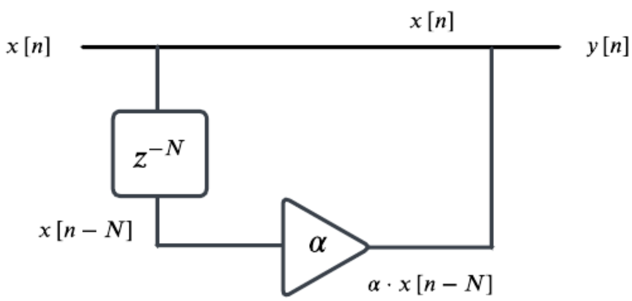

Figure II. Simple Echo Generation Block Diagram.

Well, the equation in finite differences of a system that generates a Simple Echo would look like:

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

Equation I. Simple Echo Generation Finite Difference Equation.

Where, capital N represents the number of delayed samples that we could think of as the integer product (since the coefficient that must go between brackets must be an integer because we are in the digital domain) of the number of seconds that the system delays you by the number of samples you take every second, by the sampling frequency, this impulsive displacement will not be affected by the type of system we are in.

B. Multiple Echo Generation

When we think of a system that generates multiple echoes, we think of a recursive system, that is, a system that, cyclically, passes the previous signal through the communications channel, dampening it and delaying it a certain number of samples each time it passes through the communications system again.

So, the finite difference equation of this system would look like:

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

Equation II. Multiple Echo Generation Finite Difference Equation.

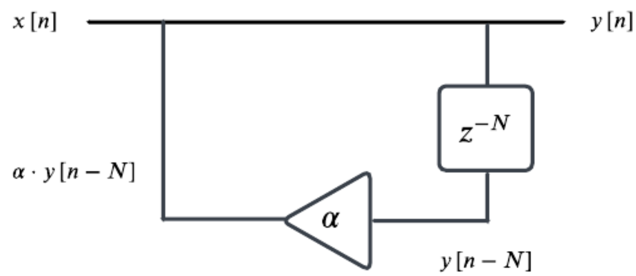

Figure III. Multiple Echo Generation Block Diagram.

The graph of the signal resulting from passing through a Multiple Echo Generator System is quite interesting.

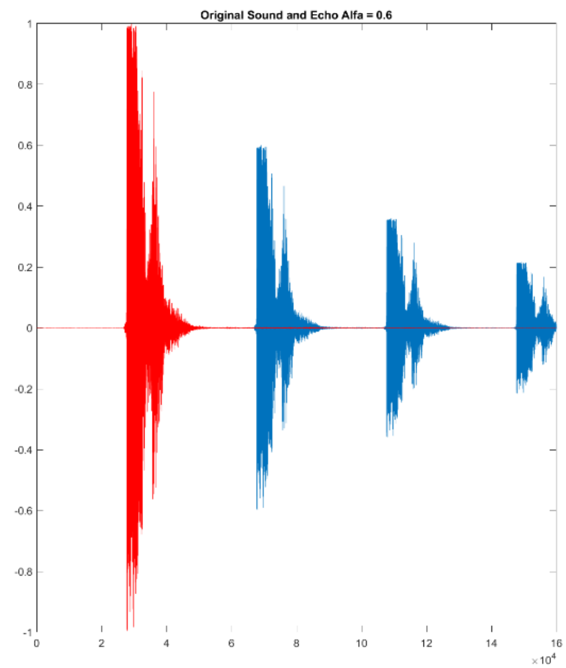

Figure IV. The Original Digital Signal and it's Echoes.

You can see the original signal (in red) and its attenuated and delayed repeats.

You can see how, the next repetition is attenuated by a factor α and delayed with respect to the previous N samples, and so on. From the graph we can see that the first samples that are not worth 0, are around $n = 25000$ and the samples of the first reverberation are around $n = 65000$, if we make the difference between these two and divide it between the sampling rate will give us a 1 s delay, which is the delay we applied.

IV. Echoes Cancellation

For the elimination of the generated echoes, we must think of a system that allows us to recover the original signal, that is to say, that allows us to neutralize the echo introduced and thus clean the signal.
In terms of signal processing, we need a system whose total impulse response is a $\delta[n]$.
Mathematically it is expressed as follows:

$$y[n] = x[n] * \delta[n] = x[n]$$

Equation III. Relation between input and output signal.

If we take the Kronecker Delta and separate it into the two systems that can be in the communications channel, the echo generator systems and the equalizer systems, for each case it will be different depending on whether we have an Echo Generator System Single or Multiple, but they share the same goal.

$$\delta[n] = h_{echo}[n] * h_{eq}[n]$$

Equation IV. Relation between the two impulse responses of the Systems of the Communication Channel.

And if we pass this expression to the transformed domain

$$1 = H_{echo}(z) \cdot H_{eq}(z)$$

Equation VI. Relation between the Transfers Functions of an Echo Generator System and its Equalizer.

And if we isolate the Transfer Function of the Equalizer System we are left with:

$$H_{eq}(z) = \frac{1}{H_{echo}(z)}$$

Equation VII.

A. Simple Echo Equalizer

From the previous expression, if we isolate and antitransform , we can deduce the equation in finite differences of the Eco Simple Equalizer System (Following the same terminology as in the practice script) :

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

Equation VIII. *Simple Echo Equalizer System Finite Difference Equation.*

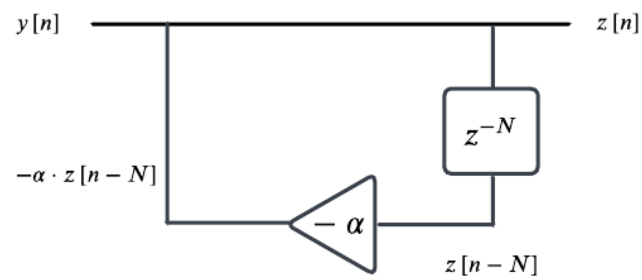

Figure V. Simple Echo Equalizer System Block Diagram.

From the block diagram we can see that the signal $z[n]$ will be delayed by a value N and attenuated by a factor $-\alpha$ and added to the signal $y[n]$ making the final signal $z[n]$ equal to a $x[n]$.

B. Multiple Echo Equalizer

If from Equation VII we repeat the same procedure as for the Eco Simple Equalizer System but for the Eco Multiple Equalizer System, we are left with the equation in finite differences of this one is:

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

Equation IX. Multiple Echo Equalizer System Finite Difference Equation.

If we substitute the finite difference equation of the Multiple Echo Generator System into Equation IX.

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

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

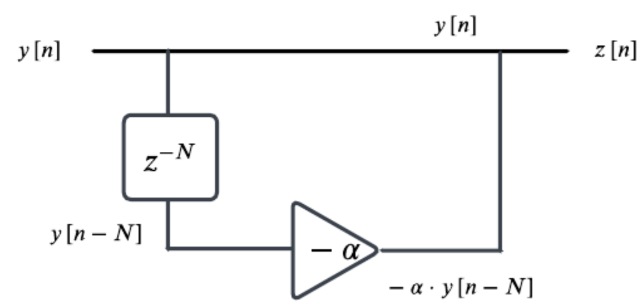

Figure VI. Multiple Echo Equalizer System Block Diagram.

From the block diagram you can see that the System Equalizer that adds the previous value $y[n-N]$ damped by a factor $-\alpha$ to the current value of $y[n]$, making the output $z[n] = x[n]$.

Seen from the frequency domain we can see the gain and the phase shift of the transfer function of the Multiple Echo Generator System and the Multiple Echo Equalizer System in the following graphs.

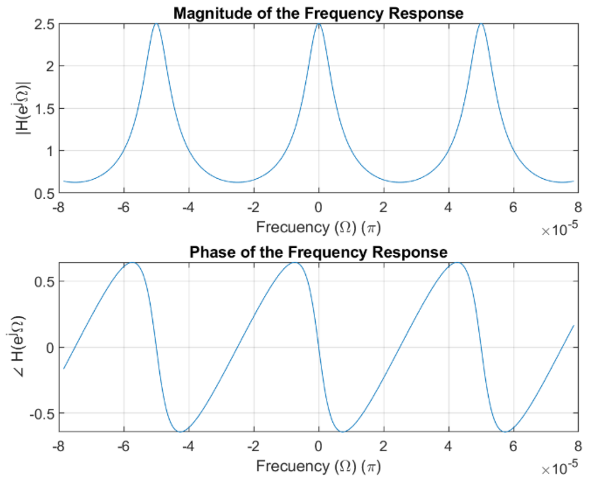

Figure VII. Magnitude and Phase of the Frequency Response of the Multiple Echo Generation System.

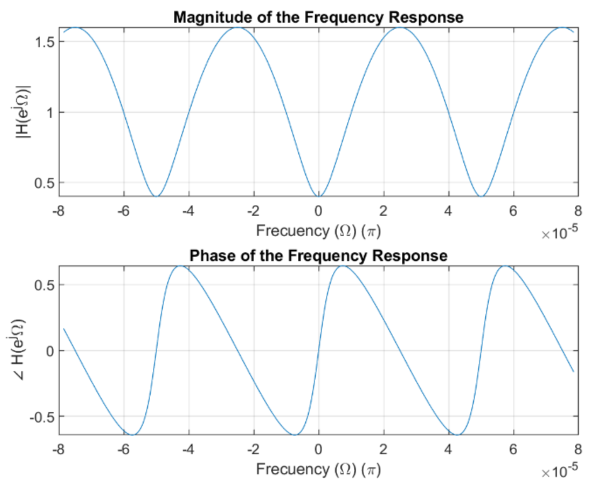

***Figure VIII.* Magnitude and *Phase* of the Frequency Response of the Multiple Echo Equalizer.**

From the above graphs it can be seen that for a certain signal frequency, the Multiple Echo Generator and the Multiple Echo Equalizer will have an inverse gain and an equal but opposite sign phase shift. So what the Echo Generator distorts and out of phase will be fixed by the Equalizer by inverting the gain and out of phase by the same value but in the opposite direction, making the output and $Z(z) = X(z)$ in the impulse domain $z[n] = x[n]$.

Continuing with the practical case from the beginning, once we have the Equalizer System, we pass the distorted signal through the Equalizer filter, and as we can see in the image below the resulting signal is equal to the original signal x[n] . In this case we have passed the signal of Image 4, a signal with multiple echoes, through a Multiple Echo Equalizer System.

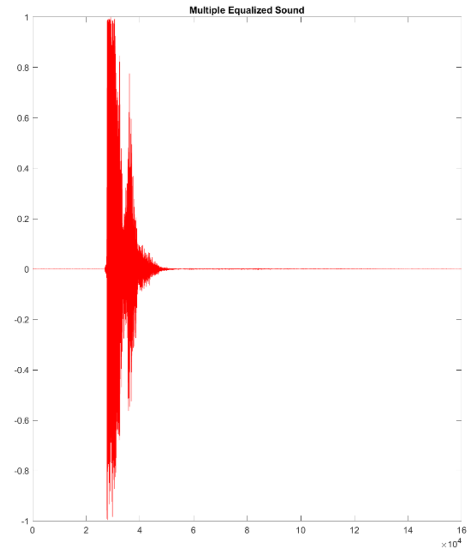

Figure IX. Equalized Signal.

V. Conversion from Analog to Digital

Once we have the clean digital signal, we will proceed with its conversion to analog format to facilitate listening. To this end, we will use a set of functions in MATLAB, just as we did during the A/D conversion. These functions will allow us to enjoy listening to the already equalized signal.

VI. Conclusions

In this experimental study carried out in the laboratory, different audio signal manipulation processes were explored and analyzed using MATLAB software.

The introduction of single and multiple echoes allowed us to identify how these phenomena affect the quality and integrity of the original signal.
Through the implementation of functions designed for echo simulation, it was observed how time delay and attenuation affect the auditory perception of the signal significantly.

In addition, the application of equalizers proved to be crucial in the restoration of the original signal. These functions, aimed at cleaning the echo present in the signal, were able to effectively mitigate the unwanted effects of the echo.

This study highlights how deliberate manipulation of a signal can introduce distortions that compromise its original integrity. And for this, the application of equalization techniques emerges as an essential resource to counteract these unwanted effects.

VII. References

[1] E. Bertran, “Señales y sistemas de tiempo discreto”, Edicions UPC, Edition 1, 2003, pages 28-30.