Study of the inhibition of sound

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Abstract: Sound is a physics-related subject, studied in the two fields of mechanics and electronics. In the present work, we have managed to inhibit a sound using knowledge from both disciplines. Particularly, we have altered the normal functioning of sound waves by inverting its signal.

I. INTRODUCTION

Applied and engineering physics study how to use technology and how it affects our lives. Along our study, we have taken into account environmental noises that may interfere with daily routines. For example, those that we may find at work, or while sleeping, etc.

Sound is understood through the propagation of elastic waves, usually employing a fluid that generates vibrations. The audible spectrum ranges from frequencies around 20 Hz (bass sounds) to 20 KHz (treble sounds).

Our first experiment was conducted using a software called Cubase, usually employed in music production [1]. We therefore created an acoustic system [Fig. 1] based on an electric guitar (Guitar), a jack connector, an amplifier (Speaker) and two microphones (Mic1 and Mic2). The connection with the interface is made through a Canon cable. This interface contains two preamplifiers (Preamp1 and Preamp2) that convert the signals from analog to digital. This whole set is connected to a computer with a USB cable [1].

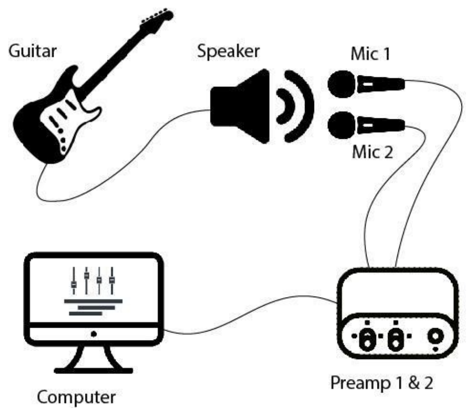

Fig. 1: Acoustic system for our first experiment.

We then generated a sound with the guitar while inverting the second phase of the microphone with Cubase. Thereby, if both signals come from the first and the second microphone sound, nothing should be heard. However, this does not work because the distance between the two microphones is excessively large.

With this study, we have successfully proved that sound inhibition is feasible. However, in experiment 1 [Fig. 1], we managed to use an analogical converter, but this would not be possible in a real case situation. Since in our study we were processing both the original and the inverted signals at the same time, we could apply the same sound delay to both of them. Nevertheless, in real life this would not be possible since the original sound could not be processed in advance. Therefore we would need to implement the same process in an analogic system.

II. EXPERIMENTAL STUDY

The analogic system we have designed is shown in Fig 2.

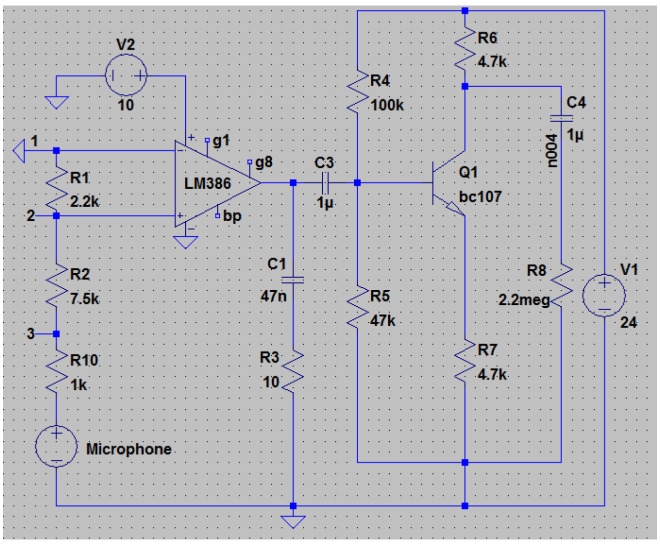

Fig. 2: Analog system we worked with on the experiment.

The first element we find is the microphone. It is worth mentioning, however, that we have used a functions generator. When configuring the input signal we have considered that the internal resistance was 1 kΩ(R10). The point 1, 2 and 3 simulated the three terminals of a potentiometer. During the first laboratory trials we did use a potentiometer with a variable resistance of 0 to 10 kΩ, and determined that the optimal values for the circuit would be R1 = 2 kΩ y R2 = 8 kΩ. We continued replacing this potentiometer with two resistances with the objective of decreasing the noise. We chose the following values: R1 = 2.2 kΩ and R2 = 7.5 kΩ.

The main objective of the potentiometer is to add an offset to the input signal. Particularly, the input signal is centered at 0 V. The potentiometer increases the signal so it is not any more centered at 0. Thereby, when it increases the negative part does not get cut.

After, the signal is being treated and access to LM386 amplifier [2]. We have chosen this model for its unipolar properties. The negative pole is been plugged to the floor, so the input signal can be compared with a value of 0. We observed that the 10V value is sour inferior threshold for this amplifier to function. If the value is increased, we do not observe the expected changes. As can be appreciate in figure 2 [Fig. 2], the terminals g1, g8 and the bypass terminal bp rested unplugged. As we have previously said, we can observe the offset in the amplifier output signal [Fig. 3].

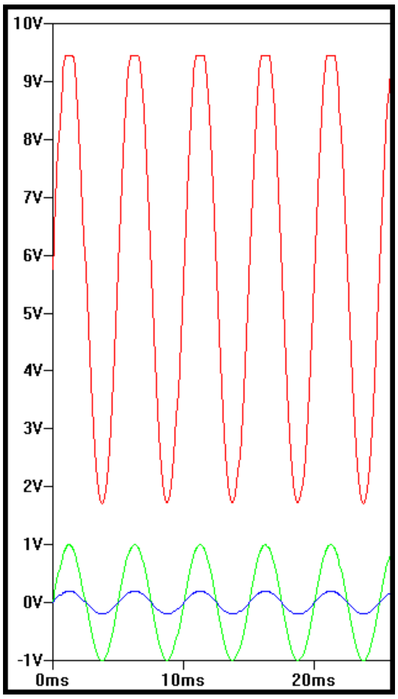

Fig. 3: V(t) signals. Green signal is microphone input signal. Blue signal is the input signal to the positive terminal of the amplifier. Red signal is the amplifier output signal.

We observe that at the exit of the amplifier the signal gets cut at 9,45V. This cut is controlled by the R1 and R2 values. We proved that R1 and R2 must have values in the proportion R2:4R1. R1 should ideally have 2 kΩ and R8 8 kΩ. However, our laboratory material have forced us to choose the values 2,2 kΩ and 7,5 kΩ, as we said before. In the Fig. 4 we can observe the ideal values for R1 and R2.

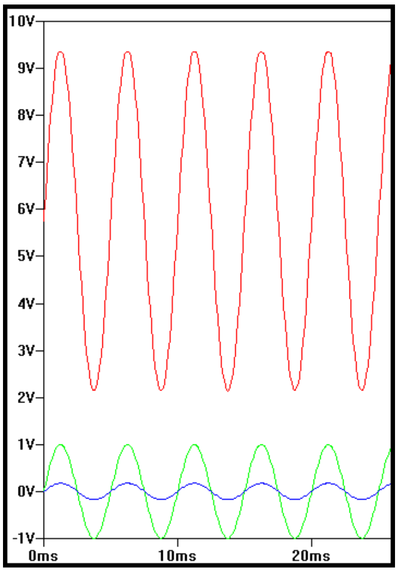

Fig. 4: V(t) signals. We considered the ideal values of R1 and R2 Green signal is microphone input signal. Blue signal is the input signal to the positive terminal of the amplifier. Red signal is the amplifier output signal.

We continued analyzing the block's circuit where the signal got inverted. As Fig. 2 shows, there are two differentiated paths, one following the C3 condenser and one following the C1. C3 is used to set a point in order to invert the signal and add-on the offset, as shown in Fig. 5.

Following C3 path, we find a NPN transistor [3] [4]. As is shown in Fig. 5, both the transistor input signal and the circuit's are displayed, but the inversion has not yet been produced. We can also appreciate that the offset and amplitude are 7V, which is a perfect value for the transistor.

The resistances R4, R5, R6 y R7 suppress the signal emitted by the fonts DC V1 and V2. V1 add power to the transistor through R4, R5, R6 y R7, so we would have enough to scan the signal coming from LM386.

The circuit branch corresponding to C1 and R3 only acts as a noise reduction filter for the high frequencies [5].

Finally, the C4 condenser suppresses the offset and the continuous inputs added by the power meter, V1 and V2. The inversion of the signal takes place at the R6 notch, so the output is taken after C4.

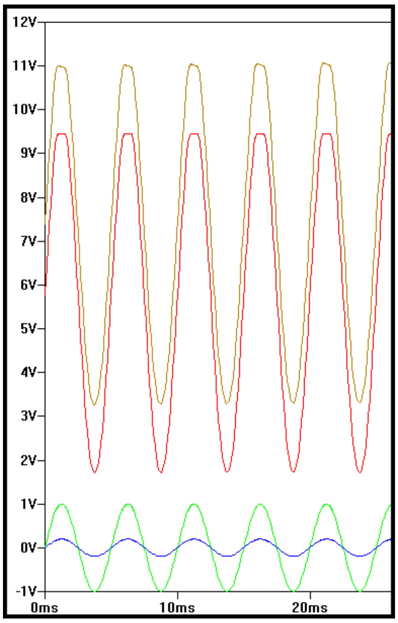

Fig. 5: V(t) signals. Green signal is microphone input signal. Blue signal is the input signal to the positive terminal of the amplifier. Red signal is the amplifier output signal. Brown signal is after C3, the same as input signal transistor.

Theoretically, we considered adding a 1,6 kΩ linear resistance with the C4 and a 50 kΩ with the C8. However, we observed in the laboratory that a 2,2 kΩ resistance was needed in R8, as shown in the images below.

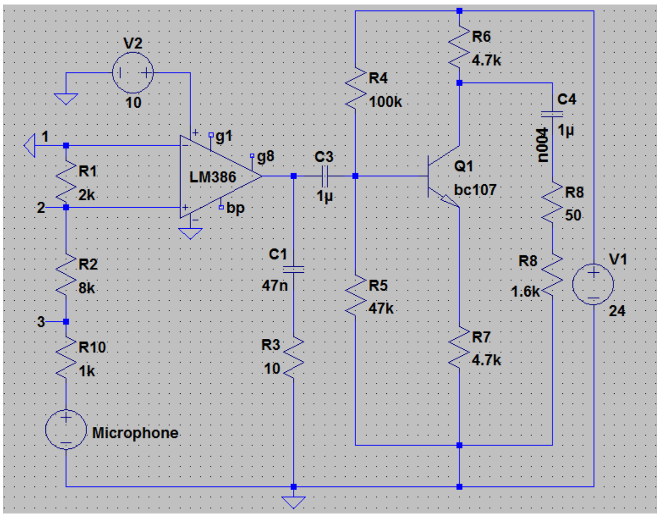

Fig. 6: Ideal circuit.

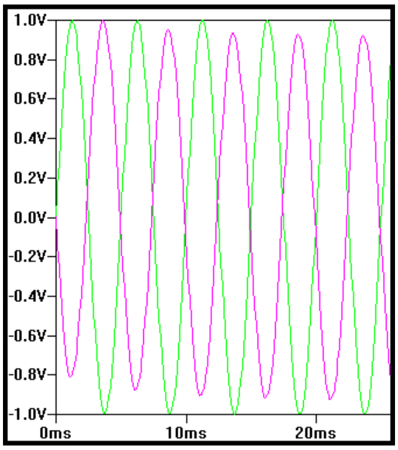

Fig. 7: V(t) signals. Simulation input signal (green) and output signal (pink) of the ideal circuit.

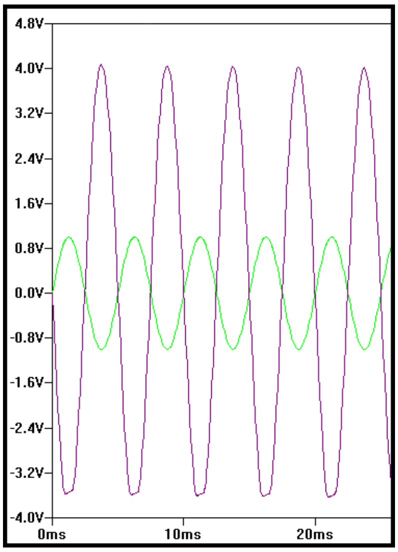

Fig. 8: V(t) signals. Simulation breadboard. Green, input signal. Purple, output signal.

III. CONCLUSIONS

In theory, the LTSpice simulations revealed that an analogic system should be capable of inverting the phase without any delay and therefore inhibit the sound.

Nevertheless, the finding show a different result. As appreciated in the following image [Fig. 10], the laboratory experiment conducted shows that there is a clear white noise in the output signal of the amplifier. That is observed for values of 3,638KHz in an interval of 0,5ms of 9,76V in amplitude.

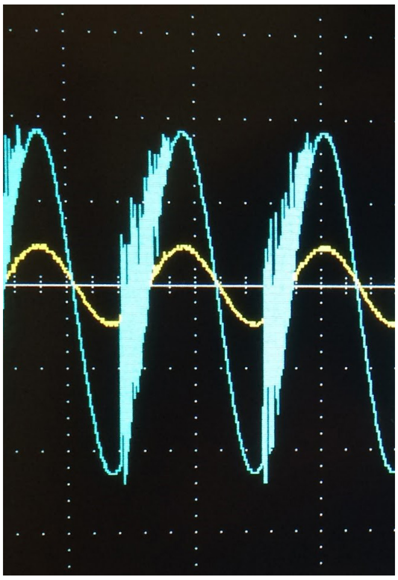

Fig. 9: Oscilloscope signals. Blue signal is the output amplifier signal. Yellow signal is the input signal of the system.

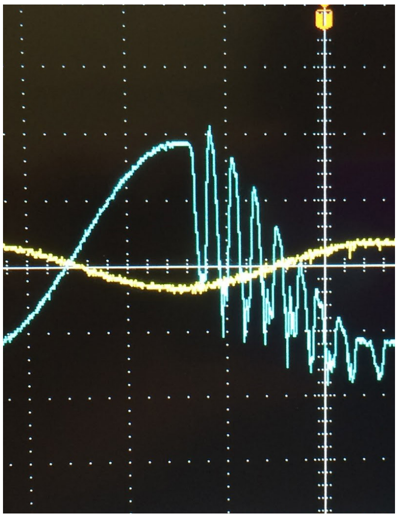

Fig. 10: Oscilloscope signals. Blue signal is the output signal of the system. Yellow signal is the input signal of the system.

This noise is generated by the amplifier and is the responsible of not being able to inhibit the sound. As can be appreciated in Fig. 10, there is a clear noise [Fig. 10].

On the other hand, we observed that the output signal arrives quite amplified, which matches our previous theoretical reasoning.

This study aims to be a first step toward the design of an inhibition sound system. The long term objective of this study is to conduct additional tests with real life sounds, adding a microphone at the end of the circuit. It would also be interesting to replicate the circuit on a phenolic board, where the different components were soldered, so that the noise was inferior.

Acknowledgments

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[1] Geoffrey Francis. *Home Recording for beginners*. Cengage Learning PTR (2009)
[2] Datasheet LM386. `http://www.alldatasheet.com/datasheet-pdf/pdf/8887/NSC/LM386.html`
[3] Datasheet BC107. `http://www.alldatasheet.com/datasheet-pdf/pdf/16088/PHILIPS/BC107.html`
[4] Notes of Physical Electronics. (2016)
[5] Notes of Applied Electronics. (2015)