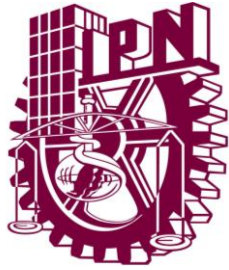




# ACTIVE NOISE CANCELLING SYSTEMS.

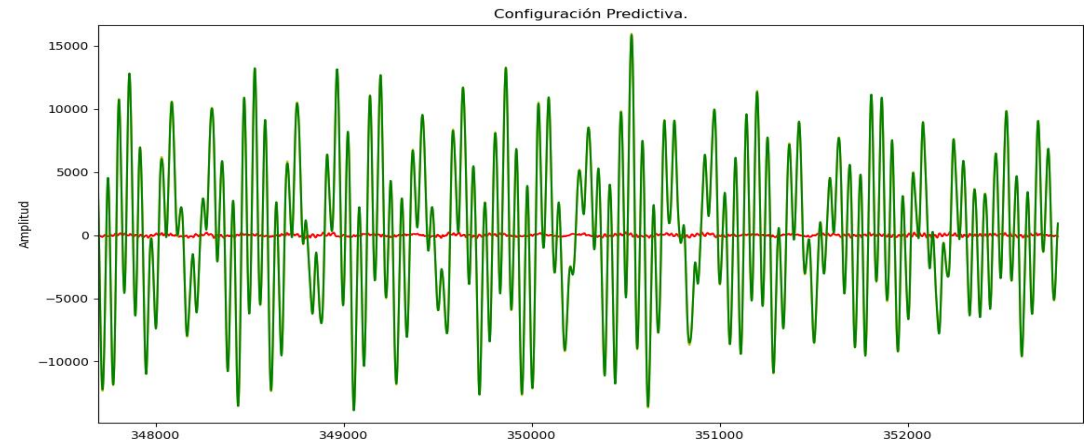
■ Presenter: Nigel Arias Prado



**INSTITUTO POLITÉCNICO NACIONAL**



# “ACTIVE NOISE CANCELLING SYSTEMS”



**PRESENTER:**  
**NIGEL ARIAS PRADO**

**ADVISERS:**  
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**Dr. Enrique Escamilla Hernández**

Mexico City, Mexico, March 25th, 2022

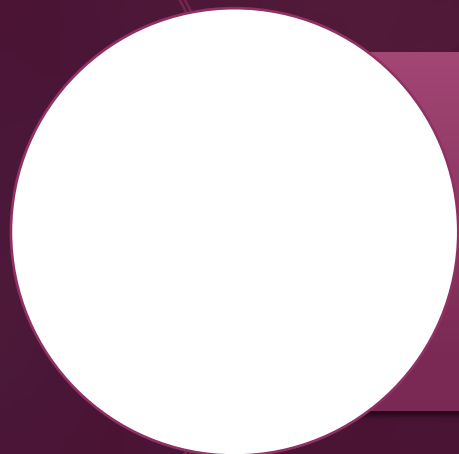


# PRESENTATION CONTENT





SECTION



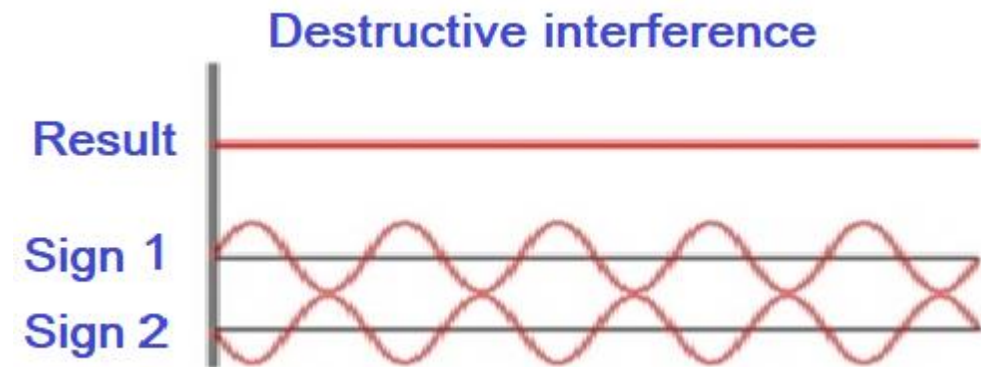
# Introduction

# WHAT IS NOISE CANCELLATION?

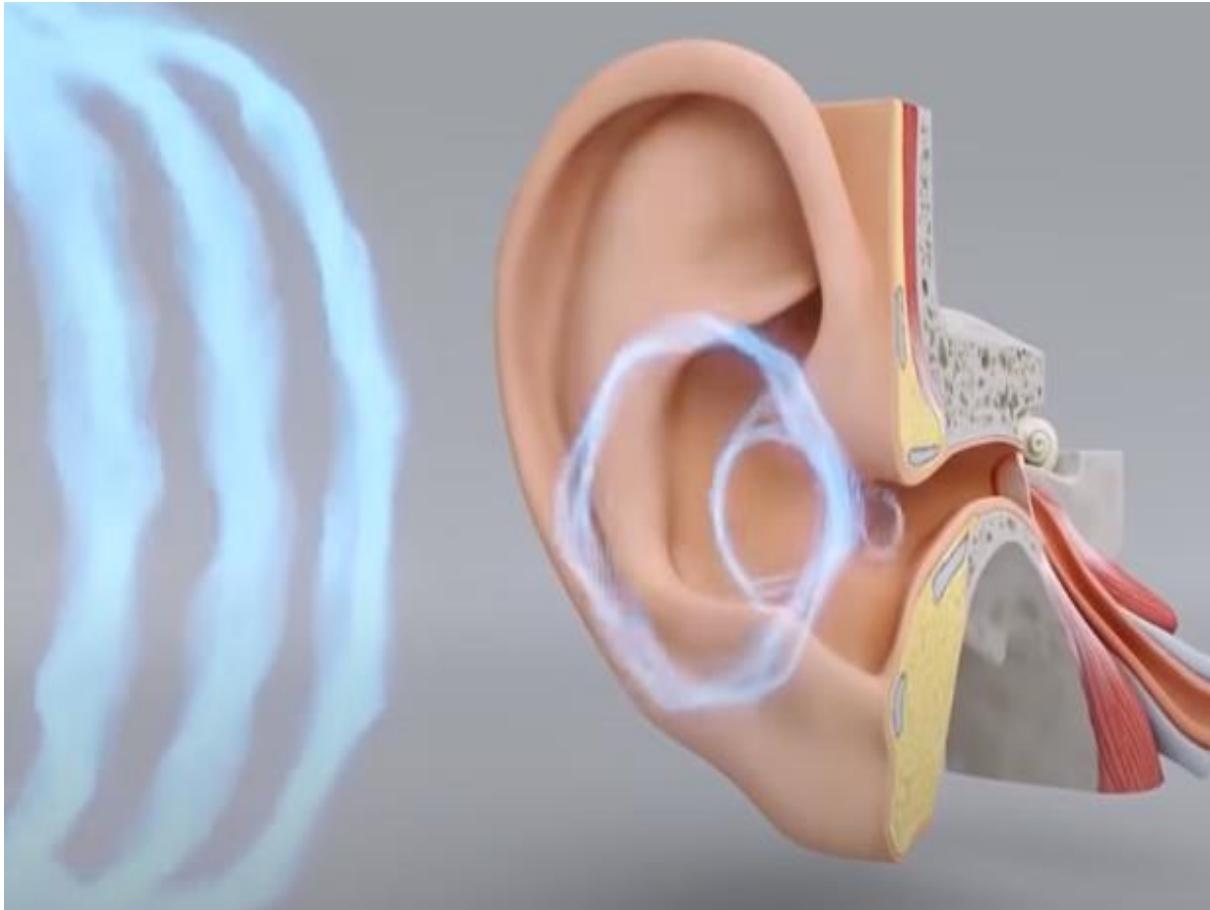
- ✓ Passive noise cancellation: which is when we obstruct the ear canals, an example is when we cover our ears, with the subject of hearing aids it will be possible to understand that it is the pads that said object has that prevents external sound.



- ✓ Active noise cancellation: three important factors are used, the processor, emitter and receiver. What is sought is that through an audio signal captured by a microphone, an inverse phase signal is generated and output through a speaker so that they are destructively combined with the noise signal.



## WHAT IS THE SOUND?



Mechanical waves that propagate in a body or in general in a substance that is vibrating and that as a consequence generates pressure changes in the air.

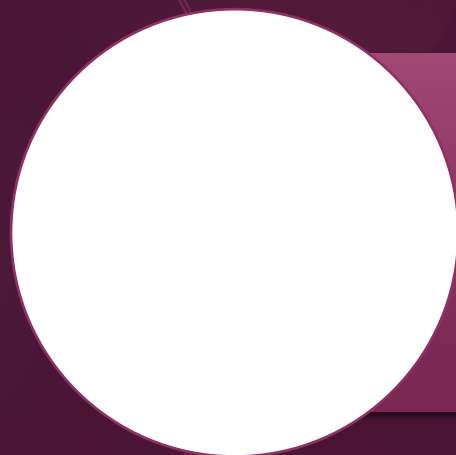
The human ear can hear a frequency range from 20 Hz to 20 KHz.

What happens to the waves when they enter the ear?

It vibrates part of the ear, which is the eardrum (a thin sheet of membranous tissue), where the waves are converted into electrical signals that the brain interprets as sounds.



SECTION



# Structure

# PRINCIPLE OF ACTIVE NOISE CANCELLATION

- The first microphone captures what is the noise signal.
- A filter that will create an acoustic signal inverse phase to the input signal to try to eliminate the noise that propagates.
- The speaker reproduces the signal in reverse phase, causing an overlap.
- The error microphone is used to provide feedback to the system and search for the coefficients that allow minimizing the power of the error.

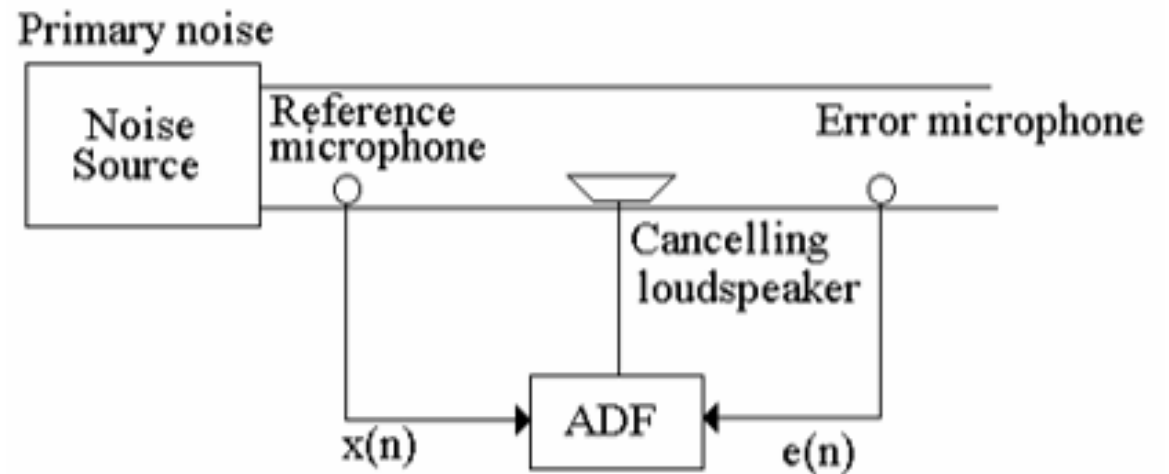


Fig. 4(a) Active noise cancellation using identifier settings



# NECESSARY CALCULATIONS

To start, different small objectives were raised in order to build a final result as shown in the diagram, knowledge of:

- Active noise cancellation principle.
- Structures for active noise cancellation.
- Estimation of the secondary trajectory.

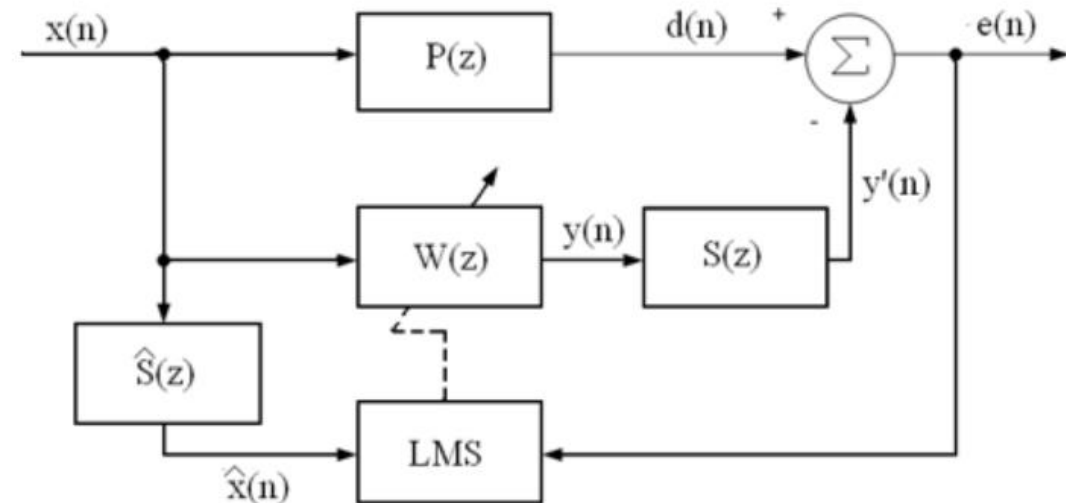


Fig. 4(c). Block diagram of an active noise cancellation scheme using an identifier configuration.

## BLOCKS NEEDED

- $P(z)$  denotes the transfer function of the pipeline.
- $W(z)$  denotes the transfer function of the canceller.
- $S(z)$  is the transfer function of the secondary path.
- $\hat{S}(z)$  is the estimated secondary trajectory.
- $\hat{x}(n)$  duct outlet.
- $y'(n)$  the output signal of the secondary path.

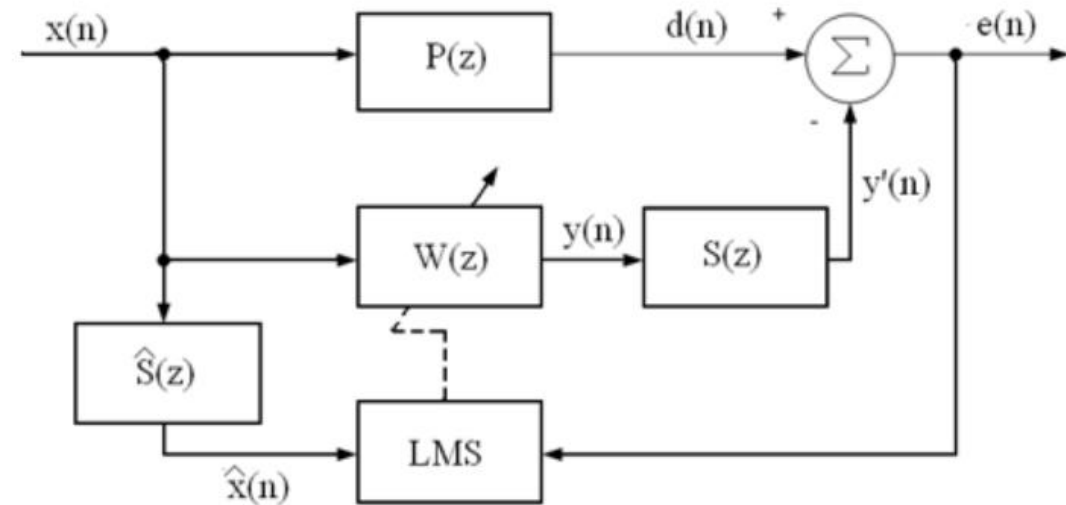


Fig. 4(c). Block diagram of an active noise cancellation scheme using an identifier configuration.

# DIAGRAM OF THE SECONDARY TRAJECTORY AND HOW IT IS ESTIMATED

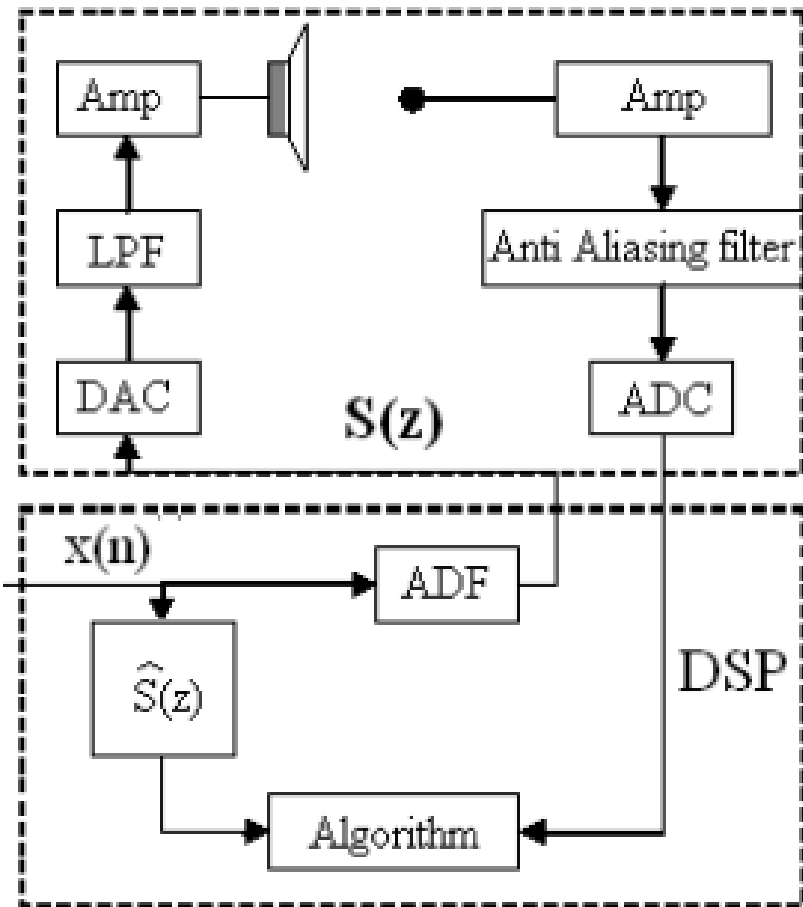


Fig. 4(d) Block diagram of a noise cancellation scheme

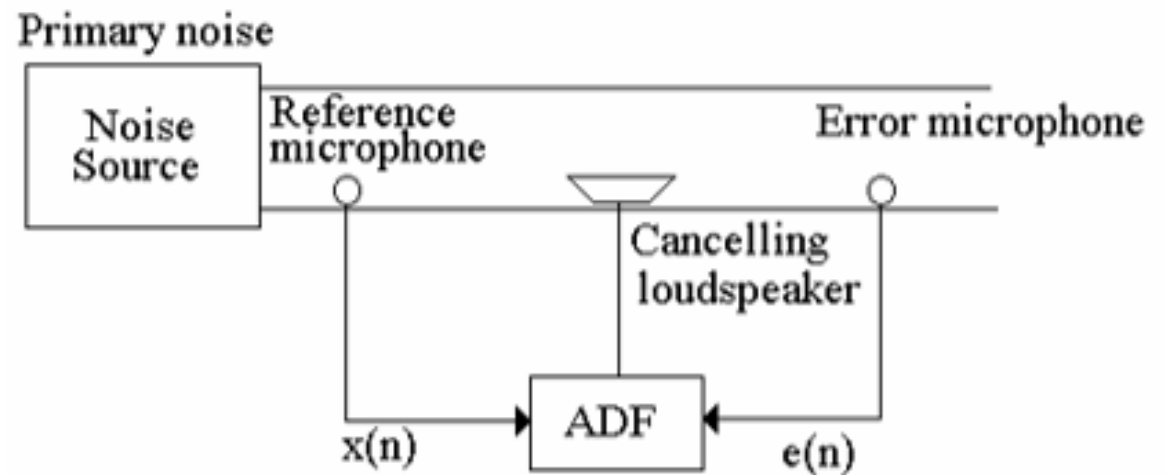


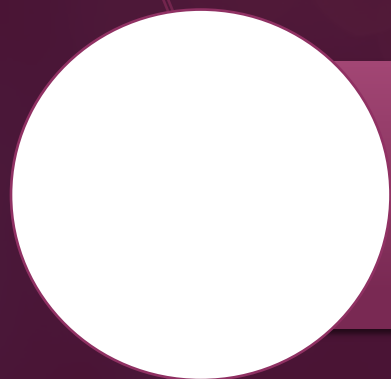
Fig. 4(a) Active noise cancellation using identifier settings

Partes importantes que no están contempladas en el filtro:

- Se comprende de:
- El filtro pasa-bajas se utiliza para evita el traslape.
- Se estima utilizando un filtro.



SECTION



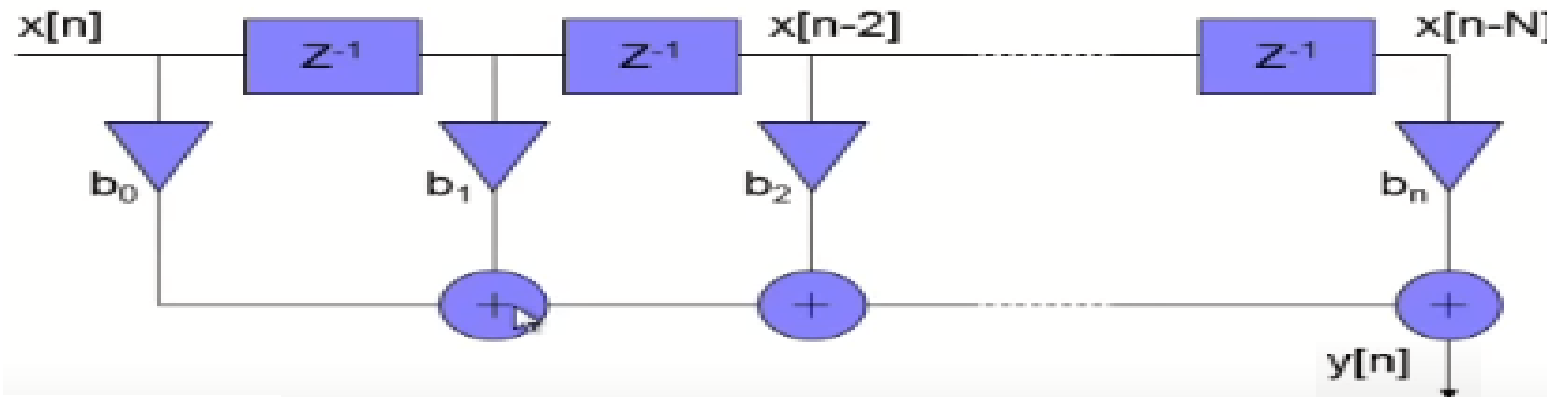
FILTER FIR



# FILTER FIR (FINITE IMPULSE RESPONSE)

In relation to the time domain, the output of the filter is equal to the sum of the inputs at previous times starting from 0 up to the order of the filter, where b will be the coefficients.

$$y[n] = \sum_{k=0}^M b_k x[n-k]$$



$X[n]$ : Input sequence.

$X[n-N]$ : Delayed input sequence N samples.

$y[n]$ : Output stream (filtered)

$B_n$ : Value of the 'b' coefficients.

$Z^{-1}$ : One sample delay

**FIR FILTER:**

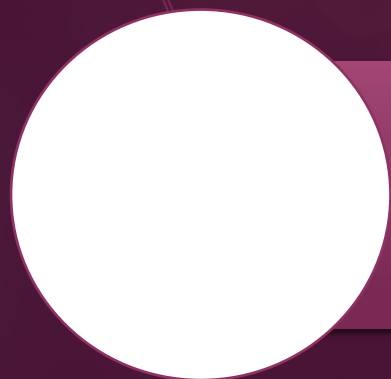
**1. REFERENCE SIGNAL.**

**2. OPTIMAL COEFFICIENTS.**

**3. FILTER OUTLET.**



SECTION



Adaptation algorithm



# ALGORITHM LMS (LEAST MEAN SQUARE)

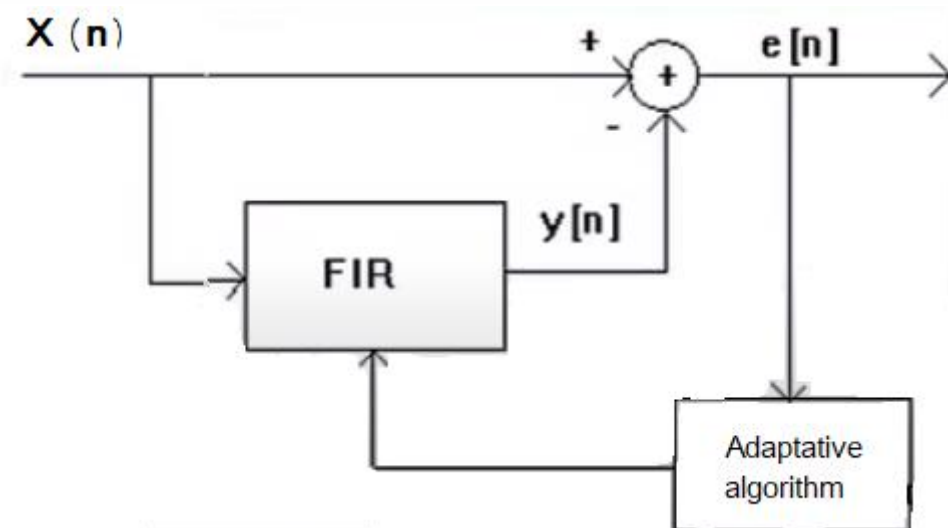
- ✓ It is an algorithm in which a simplification of the calculation of the gradient vector is carried out, modifying the objective function appropriately.
- ✓ The convergence characteristics of the LMS algorithm are examined in order to establish a range for the convergence factor that guarantees stability.
- ✓ The system will be training (W), which are the new coefficients to use in the predictive estimation line.

$$W = W + \mu \cdot \text{error} \cdot Y$$

$$a_{\text{miu}} = (Z_{we}(1))^2 + (Z_{we}(2))^2 + (Z_{we}(3))^2 + \dots + (Z_{we}(N-1))^2 + (Z_{we}(N))^2$$

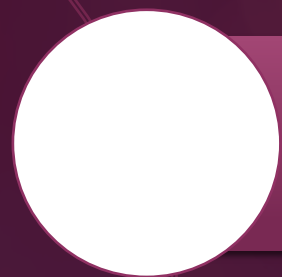
$$\mu = \alpha / a_{\text{miu}}$$

$$0 < \alpha \leq 1$$

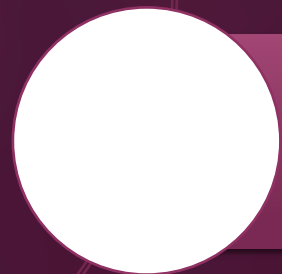




## SECTION



Software



Hardware





## EQUIPMENT TO USE

### Software Python:

It is a high level programming language.

### Main Python libraries used:

- ❑ Numpy: Specialized in numerical calculation and data analysis especially for a large volume of data.
- ❑ WavFile: For parsing .wav audio files and support for mono and stereo types.
- ❑ SCIPY: For the use of analysis of Filters and signals within the frequency domain.

```
#Graficar plt, analizar (np,plt,waves)
import numpy as np
import matplotlib.pyplot as plt
import scipy.io.wavfile as waves
#Grabar
import pyaudio as pa
import wave as wa
import struct
#Reproducir
import simpleaudio as audi
#Hilo ejecución paralela
import threading
```



# EQUIPMENT TO USE

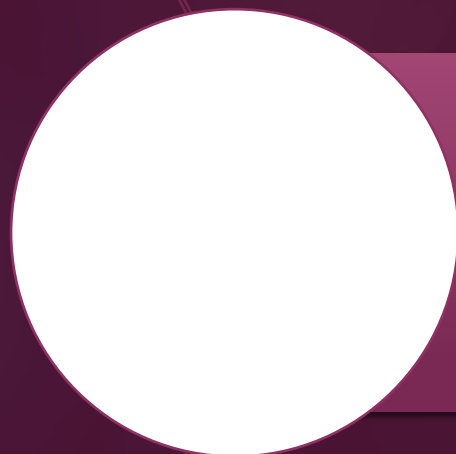
## Tinker Edge R

- ❑ Single-board computer for AI applications, with Arm® big.LITTLE™ technology and Rockchip NPU, easy for Machine-Learning deployment.
- ❑ Arm® big.LITTLE™ A72 + A53 Hexa-core SoC
- ❑ Machine-Learning capability with Rockchip NPU.
- ❑ Multiple MIPI-CSI & DSI / HDMI / Type-C (DP)
- ❑ 12v – 19v DC-in offers stable power delivery
- ❑ Linux & Android





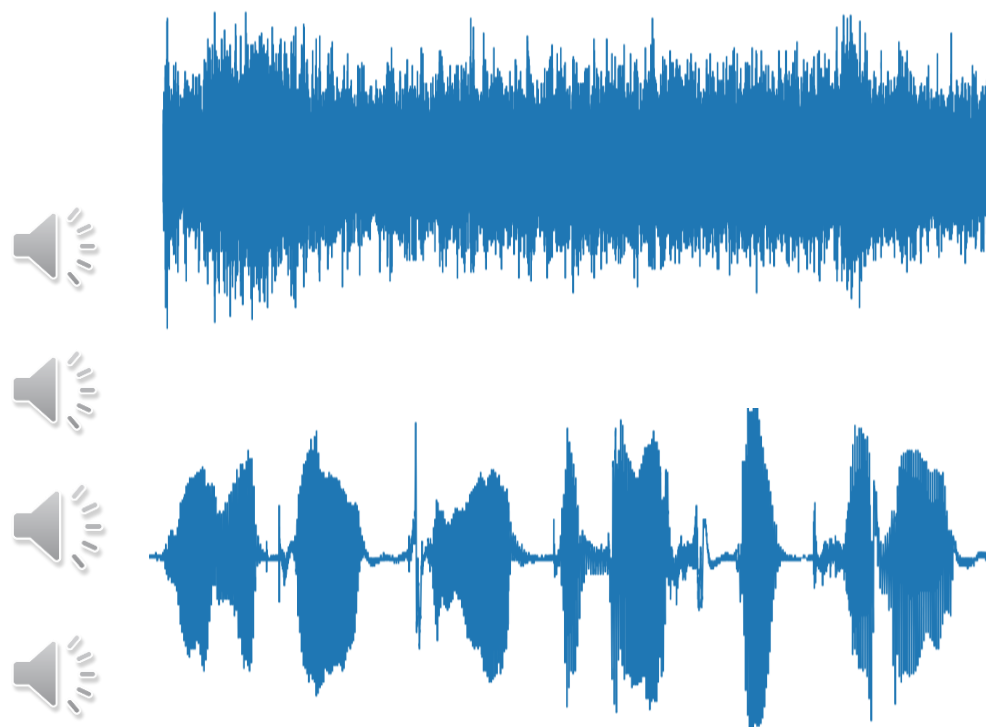
SECTION



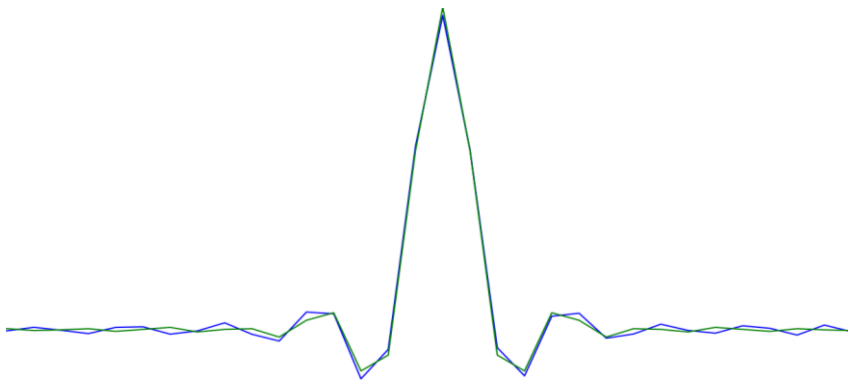
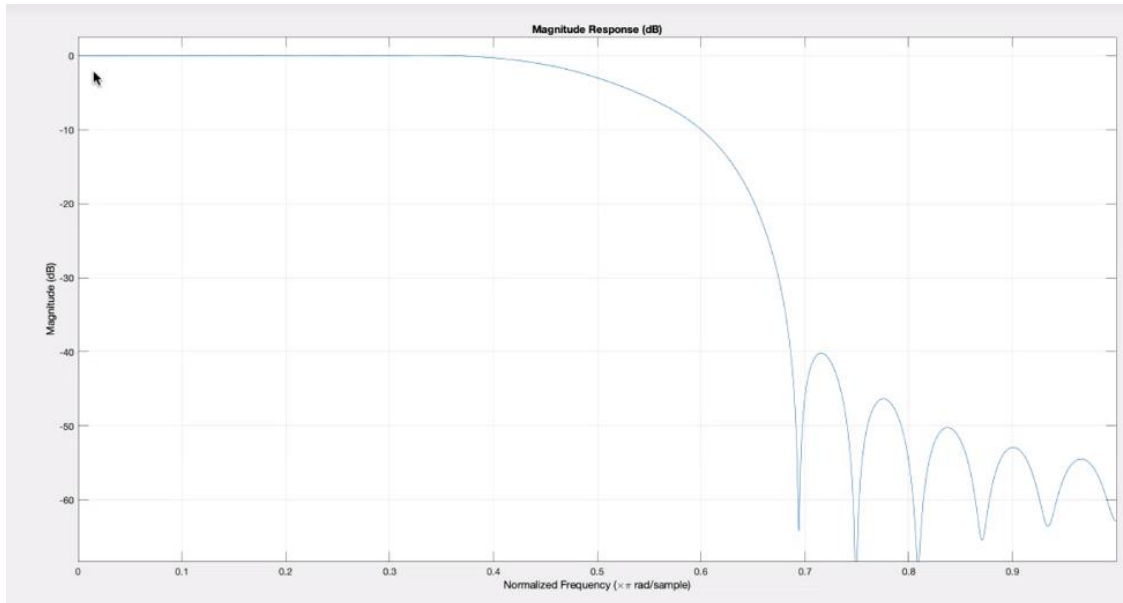
Results

# GENERAL INFORMATION

audio name	Duration in seconds	Maximum amplitude value	Minimum amplitude value	Data to analyze
Plane	30	179	72	232,000
Bells	30	230	36	232,000
Talk	2	13,312	-17,152	22,050
Entertainer	33	25,521	-22,091	1,024,000
Truck	45	7,775	-6,789	1,984,500

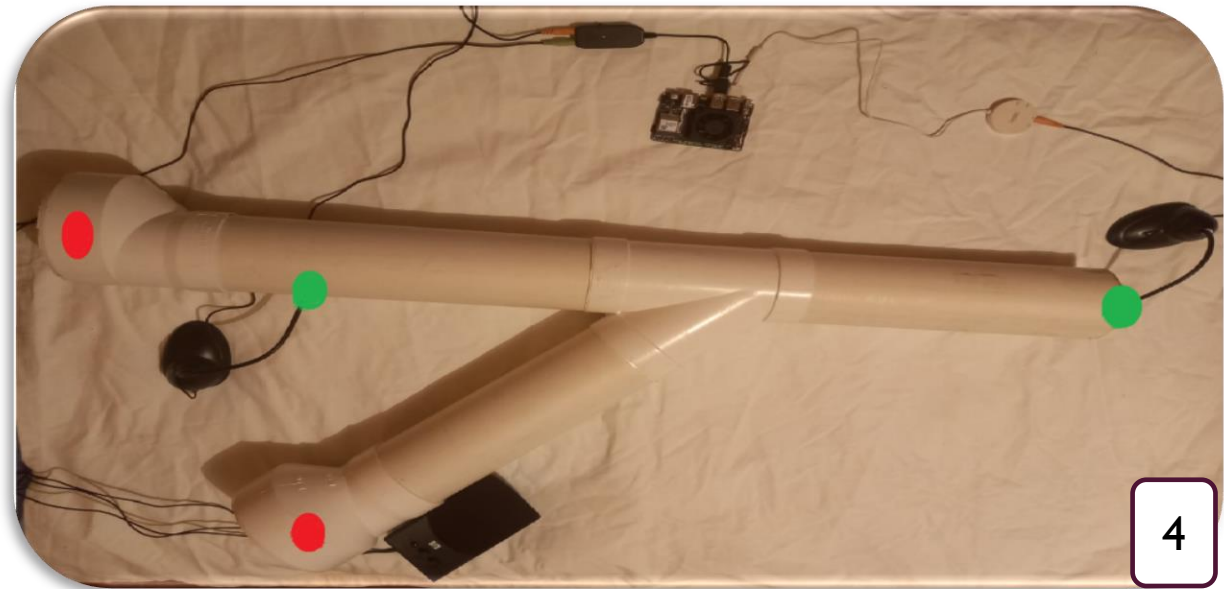
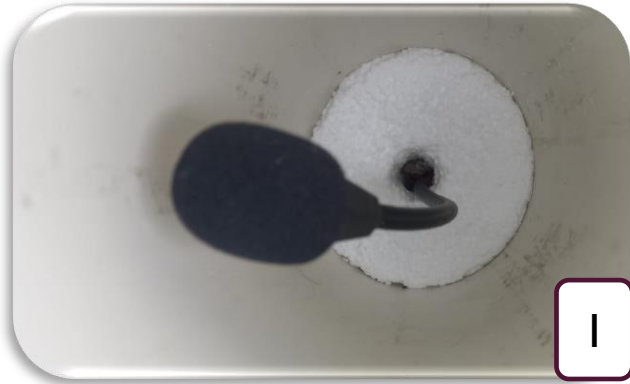


# COEFFICIENTS

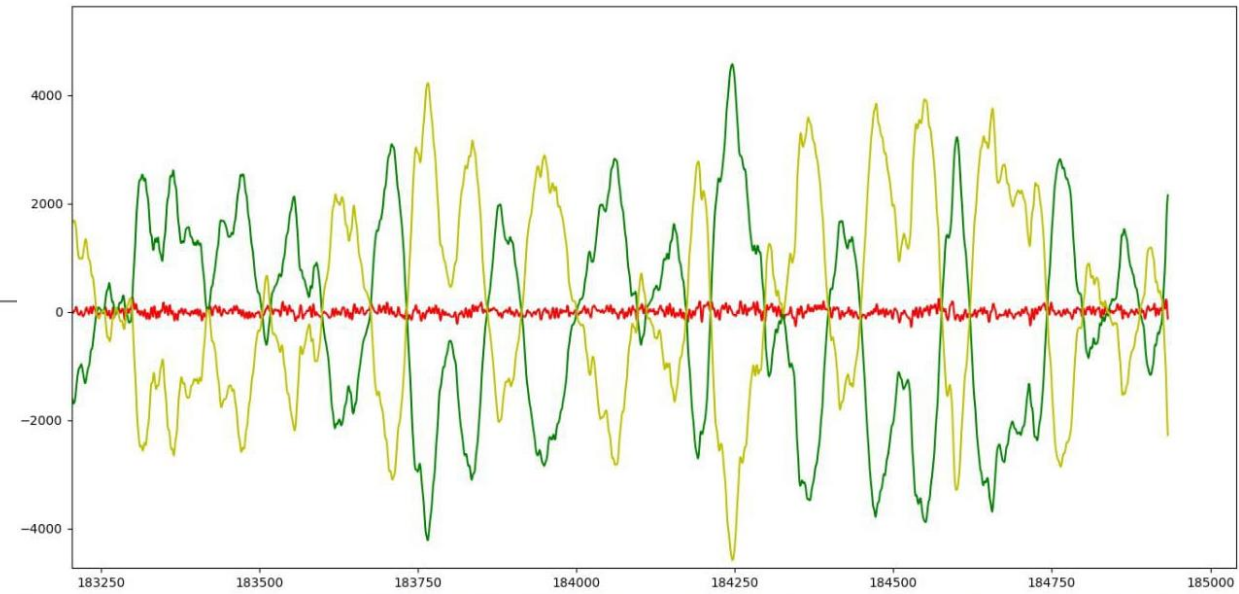
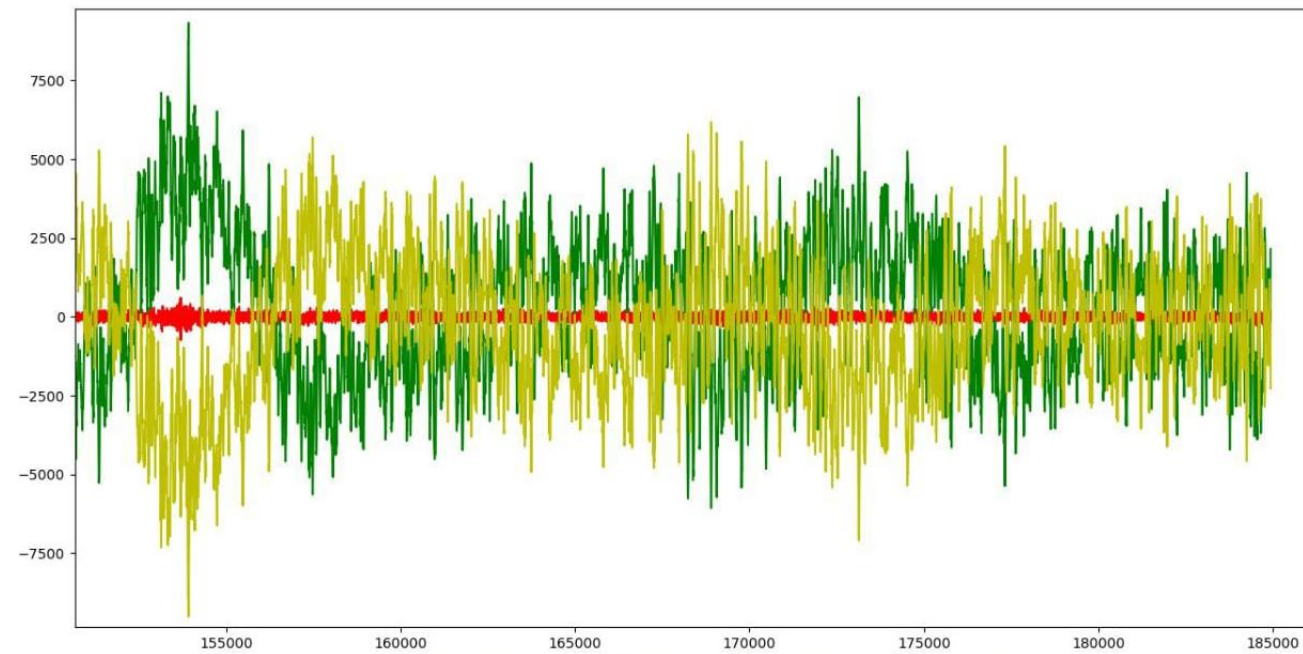


- It has a cutoff frequency (-3 DB).
- Low pass filter used to simulate the noise channel.
- It was applied to audio signals that work with 20.5 KHz, applying the coefficients would have a cutoff at 11.25 KHz.
- 20 LOG filter gain scale.

# EVALUATION OF THE ALGORITHM IN REAL CONDITIONS



# EVALUATION OF THE ALGORITHM IN SIMULATION





## GLOSSARY

<b>ADC</b>	Analog - digital converter
<b>DAC</b>	Digital - analog converter
<b>LMS</b>	Least Mean Square
<b>LPF</b>	Low pass filter
<b>AMP</b>	Amplifier
<b>RLS</b>	Recursive Least Square
<b>FIR</b>	Finite Impulse Response
<b>IIR</b>	Infinite Impulse Response
<b>FFT</b>	Fast Fourier Transform
<b>MSE</b>	Mean Square Error