PROJECT REPORT

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Objective:-

- Echo Generation
- Echo Cancelation
- Noise Classification

Input:-

- Echo Generation:- An audio file is provided to which echo has to be added.
- Echo Cancelation:- An audio file with echo is provided.
- Noise Classification:- An audio file is provided which contains one of the noises which are:-
  - Ceiling fan
  - Water pump
  - City traffic
  - Pressure cooker
  - Random noise

Problem solving approach:-

- Echo generation:-

  Since, "echo" refers to a delayed and attenuated version of a sound or signal.

  So, to represent an echoed signal the general difference is:

$$y[n] = x[n] + ax[n-n_o]$$

Where a represents the attenuation factor and n_o represents the delay in the signal.

From the above difference equation, we find the generalized transfer function i.e.,

$$H(Z) = 1 + aZ^{-no}$$

We can construct an echoed signal by using the built-in filter(num,den,signal) function with the aid of this transfer function.

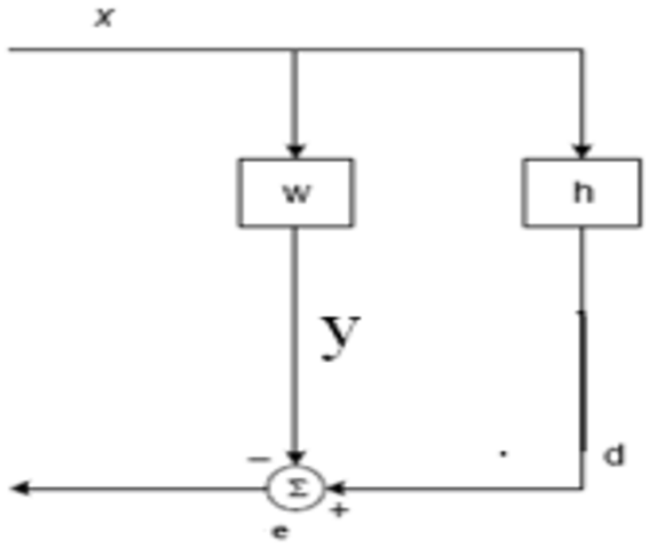
$$y = \text{filter}(num,den,x)$$

Where num = [1, zeros(1,n_o), a) and den = 1 as observed from the transfer function.

We have two parameters n_o and a, which we can vary to generate different echoed signal.

- <u>Echo Cancelation:-</u>

  To cancel/ remove the echo from the given non-uniform echoed signal, we use a method called Least Mean Square Algorithm.



A path that changes the signal x is called h. Transfer function of this filter is not known in the beginning. The task of LMS algorithm is to estimate the transfer function of the filter. The result of the signal distortion is calculated by convolution and is

denoted by d. In this case d is the echo and h is the transfer function of the hybrid.
The adaptive algorithm tries to create a filter w. The transfer function in turn is used for calculating an estimate of the echo. The echo estimate is denoted by y. The signals are added so that the output signal from the algorithm is

$e = d - y$ where e denotes the error signal.

The error signal and the input signal x are used for the estimation of the filter coefficient vector w.
One of the main problems associated with choosing filter weight is that the path h is not stationary. Therefore, the filter weights must be updated frequently so that the adjustment to the variations can be performed. The filter is a FIR filter with the form

$$W = W_0(n) + W_1(n)Z^{-1} + \ldots + W_{N-1}(n)Z^{-(N-1)}$$

With each iteration of the LMS algorithm , the filter tap weights of the adaptive filter are updated according to the following formula

$$W(n+1)=w(n)+2\mu e(n)x(n)$$

Here x(n) is the input vector of time delayed input values,

$$x(n) = [x(n)\ x(n-1)\ x(n-2)\ \ldots..x(n-N+1)^T .$$

The vector $w(n) = [w0(n)\ w1(n)\ w2(n)\ ..\ wN\text{-}1(n)]^T$ represents the coefficients of the adaptive FIR filter tap weight vector at time n.
The parameter μ is known as the step size parameter and is a small positive constant.

Implemntation of LMS algorithm:-

Each iteration of the LMS algorithm requires 3 distinct steps in this order:

1. The output of the FIR filter, y(n) is calculated using equation:-

$$y(n) = \sum_{i=0}^{N-1} w(n)x(n-i) = \mathbf{w}^T(n)\mathbf{x}(n)$$

2. The value of the error estimation is calculated using equation

$$e(n) = d(n) - y(n)$$

3. The tap weights of the FIR vector are updated in preparation for the next iteration, by equation.

$$w(n+1) = w(n) + \mu(n)e(n)x(n)$$

The main reason for the LMS algorithms popularity in adaptive filtering is its computational simplicity, making it easier to implement than all other commonly used adaptive algorithms.

- Noise Classification:-

We are using the concept of cross-correlation to classify the noise.

1) We made 6 combinations of noises by subtracting the given two pair of noisy signal and obtaining the outputs as follows:

- Water Pump – Fan
- Water Pump – Traffic
- Water Pump – Pressure cooker
- Traffic – Pressure cooker
- Traffic – Fan
- Pressure cooker – Fan

On observing the above, on taking any of the three combinations one of the noise is always present.

2) We performed cross-correlation of these 6-noise signal with the input signal and obtained the peaks from each correlated signal, then we found the max 3 peaks out of the 6.

3) The Common noise present in the 3 peaks will give us noise present in the input signal

Cross Corelation:-

Cross-correlation is a technique used to analyze the similarity between two signals. It involves sliding one signal over another and computing a similarity measure at each step. The result is a correlation function that indicates how well the signals match at different time offsets. Positive peaks in the correlation function suggest alignment, while negative peaks indicate an inverted match.

$$R_{xy}[k] = \sum_{n=-\infty}^{\infty} x[n]y[n-k], \quad (1)$$

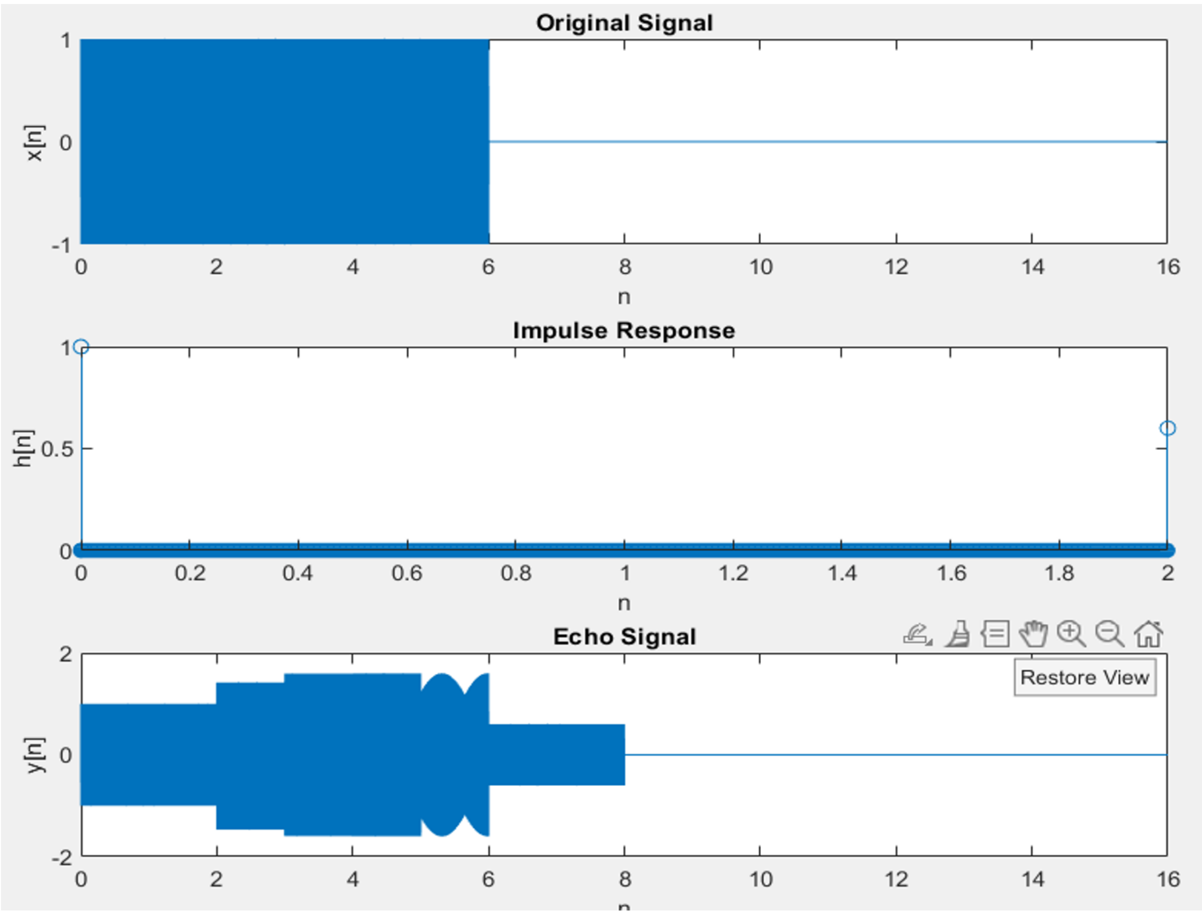
that is, we delay $y[n]$ by k and then multiply the two signals and add them up. One way to think of this is like an infinite version of a dot product. The *autocorrelation* of a signal $x[n]$ is the cross correlation between $x[n]$ and itself:

$$R_{xx}[k] = \sum_{n=-\infty}^{\infty} x[n]x[n-k]. \quad (2)$$

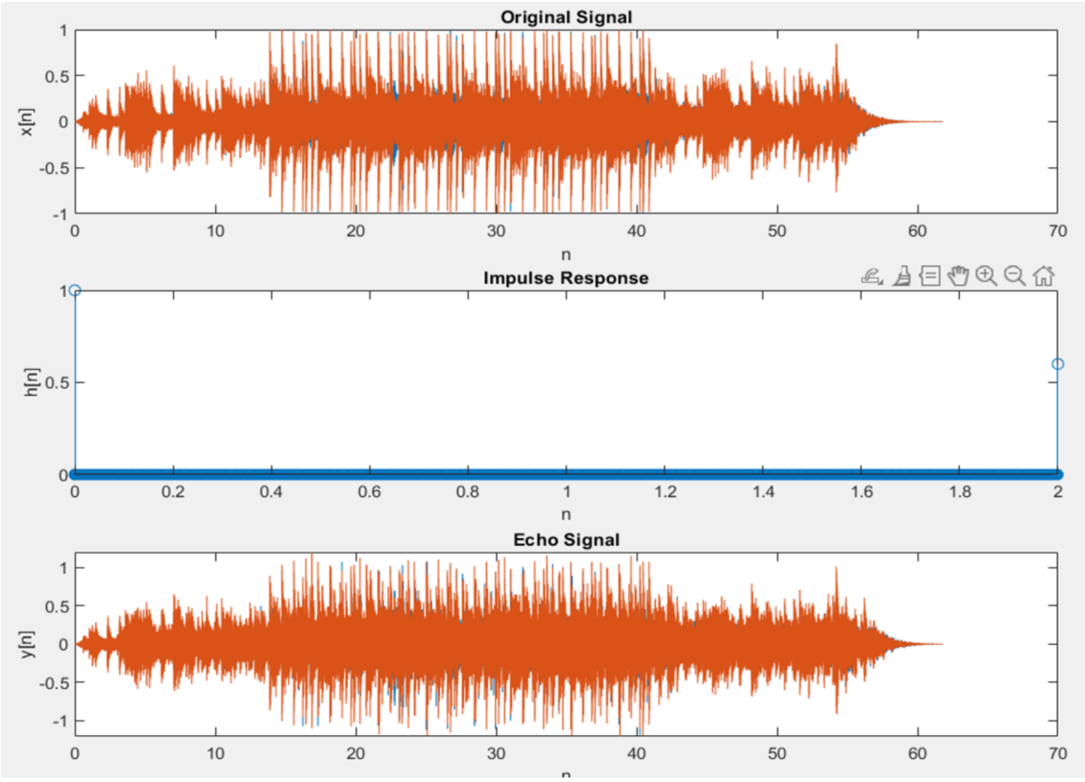
Results:-

- Echo generation:-

Plot for Q1:-

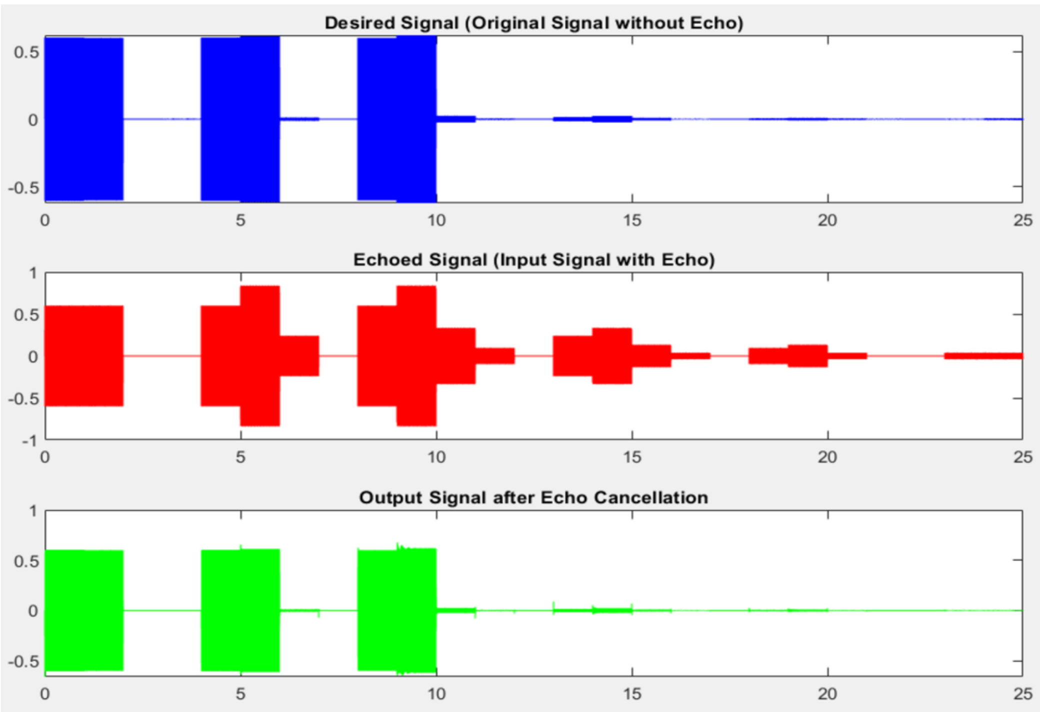


Plot for Q1_hard:-



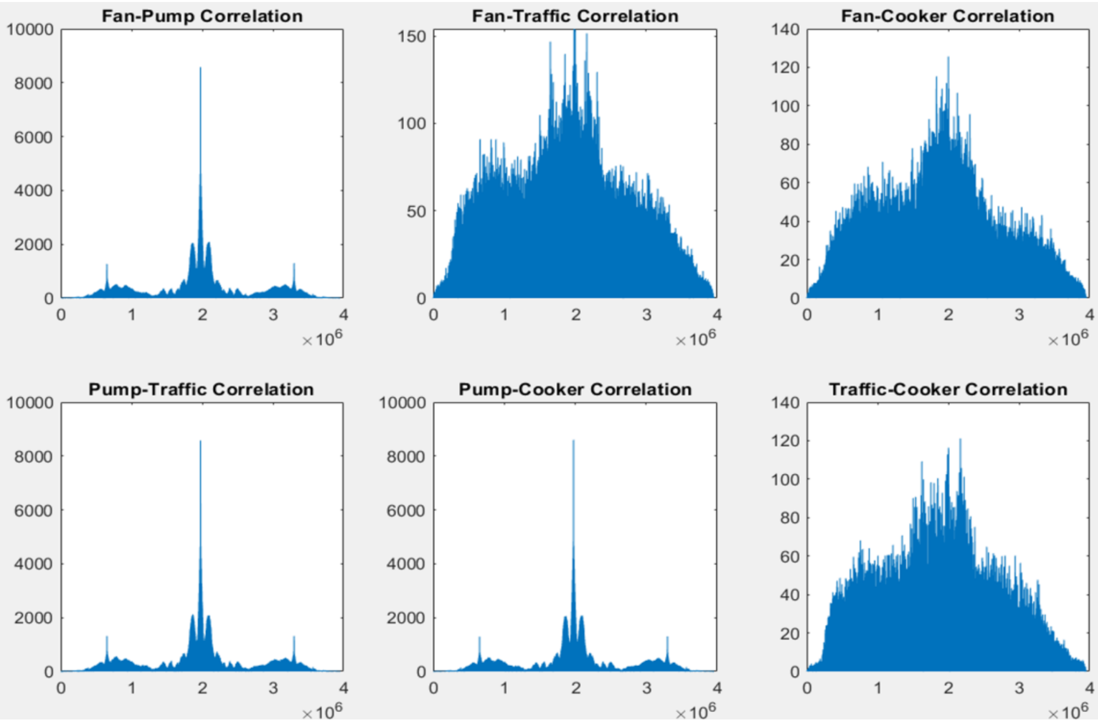
- <u>Echo Cancellation:-</u>

Plot for Q2_not_so_easy:-

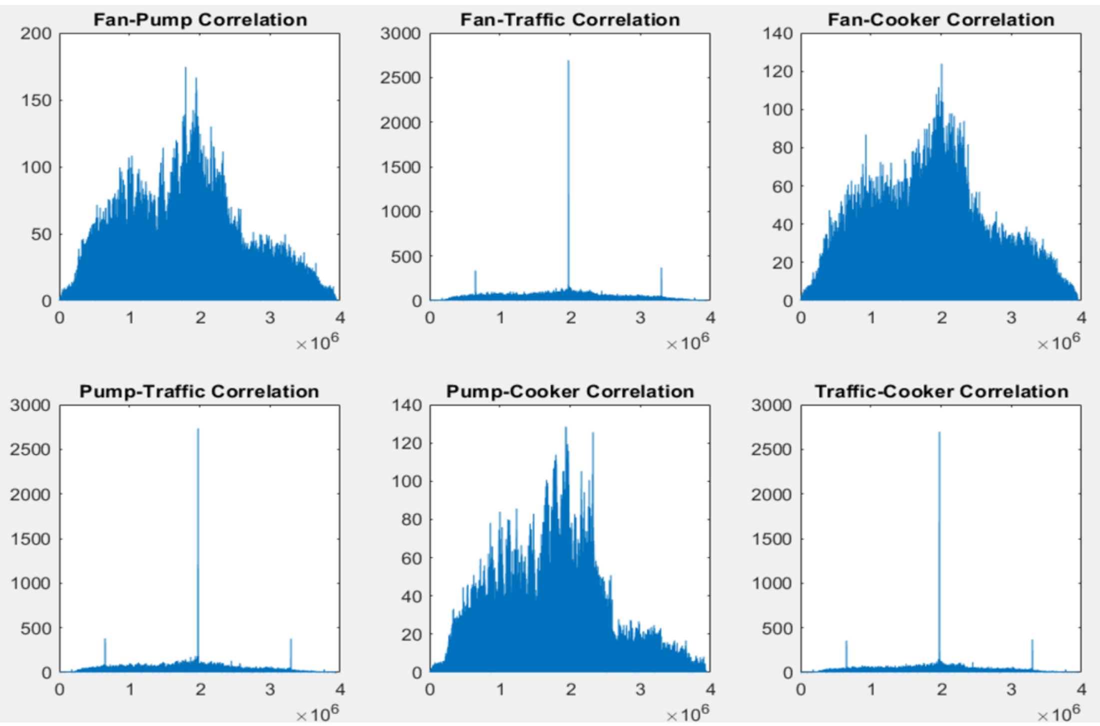


- <u>Noise Classification:-</u>

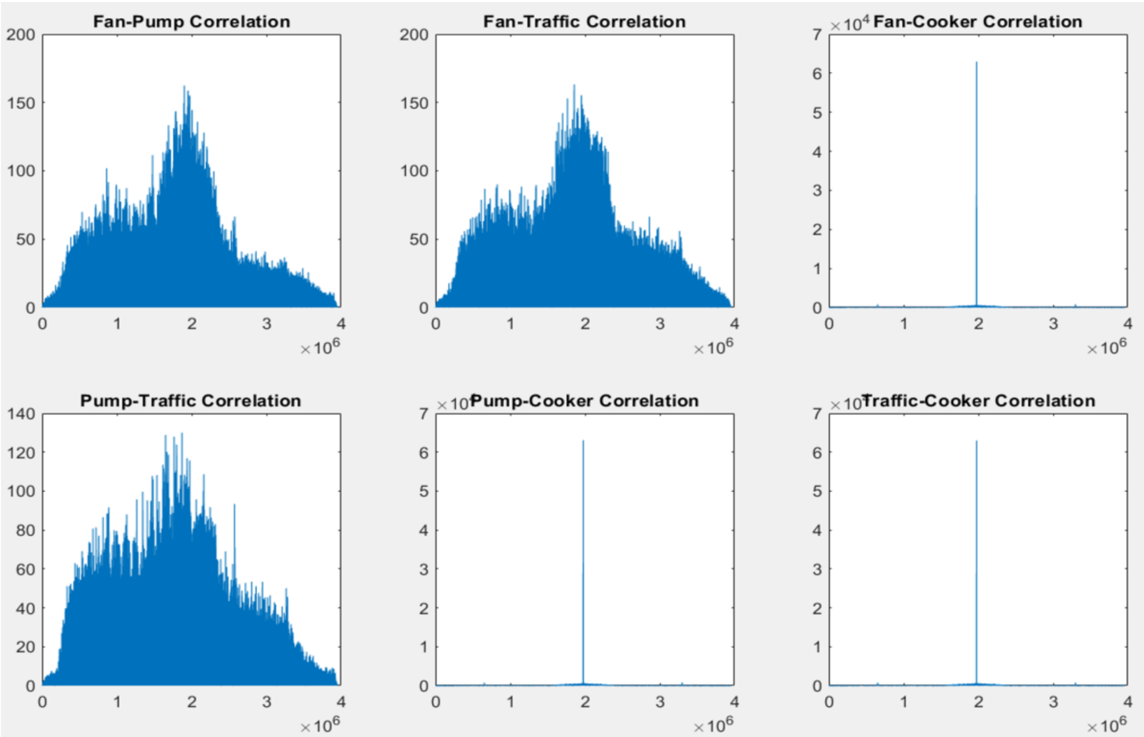
Plots for Cross-Correlation of Music + Pump signal as an input with all noise signal:



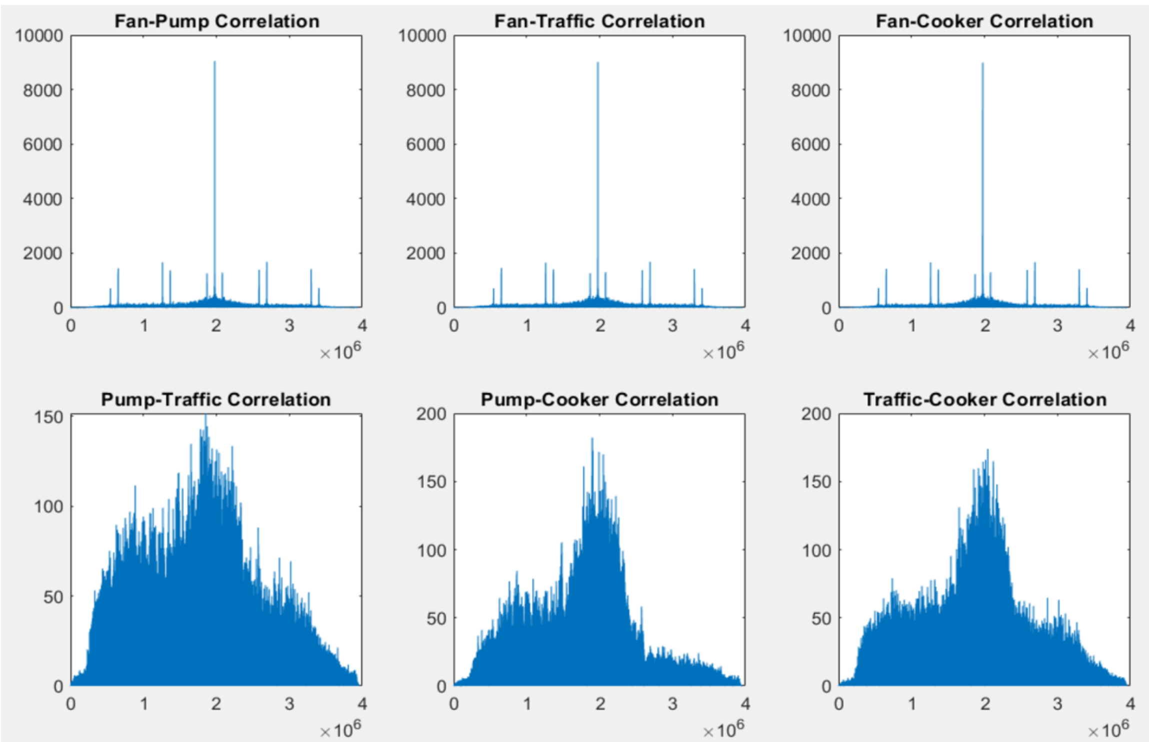
Plots for Cross-Correlation of Music + Traffic signal as an input with all noise signal:



Plots for Cross-Correlation of Music + Pressure Cooker signal as an input with all noise signal:



Plots for Cross-Correlation of Music + Fan signal as an input with all noise signal:



From these four plots, we can see that, of the six correlation plots, three of the peak values share a common noise, and that this common noise is present inside the given signal.

So, we are successfully classifying the background noise present in music recordings without removing the noise. We can accurately identify and categorize the type of noise present in the recording, distinguishing source of origin.

Conclusion:-

- This project has provided a comprehensive overview of the applications of signal processing techniques in real life. We have explored some real-life problems and their solutions while doing the project.
- We get a thorough understanding of how we will apply the techniques discussed during the course to solve real-life problems.
- By understanding and applying signal processing techniques, we gain powerful tools for addressing real-world challenges across various domains.